Virtuoso® Spectre® Circuit Simulator and Accelerated Parallel Simulator RF Analysis User Guide

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Contents

Preface 21

Using License Queuing 23
Suspending and Resuming Licenses 23
Related Documents for Spectre 24
Third Party Tools 24
Typographic and Syntax Conventions 25

1
Introduction 27

Simulation Basics 27
Fundamentals of RF Simulation 27
Shooting Newton and Harmonic Balance algorithm 27
Shooting Method 28
Harmonic Balance Method 28
SpectreRF Analyses 28
Periodic Analyses 28
Quasi-Periodic Analyses 30
Envelope Analysis 32
APS and Parasitic Reduction 32
Co-simulation with Virtuoso AMS Designer 33
Large vs. Small Signal Analysis 34

2
Setting up a SpectreRF Simulation 37

Using Virtuoso ADE 37
A Testbench 37
ADE 37
Analysis form 38
Direct Plot form 38
Netlist Driven Flows 38

3
Frequency Domain Analyses: Harmonic Balance 41
Overview of Simulation Capabilities 41
Large-Signal Harmonic Balance Overview 41
Example 43
Harmonic Balance Solves For Cosines 50
Setting Harmonics Automatically 54
Setting Harmonics Manually 55
Oversample Factor 56
Trading off Harmonics and Oversample Factor 62
Two Input Frequencies 64
Diamond Cut 65
Three Input Frequencies 67
Diamond Cut With Three Frequencies 68
Funnel Cut 69
Axis Cut 71
Aliasing 72
Gain Compression Analysis 79
Sine Representation of Square Wave 81
Convergence 83
Errpreset 83
Iteration Limit 84
Number of Non-Sinusoidal Sources 86
Harmonic Balance Starting Point 86
More Capabilities 87
Save and Recover 87
Writehb and Readhb 88
Sweeps and Restart 88
Itres 89
Freqdivide 89
Oscillators 89
Pinnode 90
Probe-Based Method 91
Oscillator Tuning Mode 91
Semi-Autonomous 91

Things to try to Achieve Convergence (Driven Circuits) 92
Things to try to Achieve Convergence (Oscillators) 93
Implementation in ADE 94
Setting Frequencies, Harmonics, and Oversample 94
Diamond Cut 99
Funnel Cut 100
Sweeps 101
Freqdivide 102
Compression Analysis 103
Oscillator Additions 107
Oscillator Tuning Mode 110
Probe-Based Method 112
Semi-Autonomous 114
Commonly Used Harmonic Balance Options (Driven Circuits and Oscillators) 119
Convergence tab 120
Accuracy Tab 123
Output Tab 125
Reuse Tab 127
Misc Tab 135
Oscillator Options 138
Key Outputs in the Spectre Window 139
Examples 148
ADE Setup for 1-tone Harmonic Balance (hb) Analysis 151
Plotting Spectral Content 155
Plotting dBm 156
Sweeps 159
Plotting Currents 159
Compression Point 162
Loadpull 177
Power-Added Efficiency 190
Oscillators 192
Oscillator Tuning Mode Analysis 201
Oscillator Tuning Mode With Sweeps 205
Two Input Frequencies 210
Compression Point 220
IFFT 223

Compression Vs Pout 227
Example Using the Harmonic Port with Harmonic Balance 228
Three Tone Harmonic Balance 238
Large-Signal IP3 242
Multi-Sinusoid hb Analysis 246
Harmonic Balance AC Analysis (hbAC) 255
Small-signal Versus Large-Signal Analysis for Conversion Gain 255
Example: Conversion Gain (Down conversion) 256
Example: Conversion Gain (Up Conversion) 264
Overview of Simulation Capabilities 266
ADE Implementation 276
Circuit, Input port Setting, and ADE Setup for all the Examples in This Section . . . 276
Using hbAC Analysis for Conversion Gain Measurement 279
Commonly Used HBAC Options 287
hbAC With Multiple hb Inputs 292
Rapid IP3/IP2 302
Triple Beat 308
Compression Distortion Summary 321
IM2 Distortion Summary 327
Modulated hbAC Analysis for an Oscillator 329
Sampled hbAC Analysis for a Mixer 336
Harmonic Balance Noise Analysis (hbnoise) 344
Small-Signal Versus Large-Signal Analysis for Noise 344
Overview of Simulation Capabilities 345
Example 346
Noise Output Near Zero Frequency 352
Noise Output Near The First Harmonic 354
Frequency Sweep 354
Maximum Sideband 355
Setting Harmonics and Sidebands 355
Noise Separation 355
Multiple hbnoise 356
Hbnoise after Multi-Tone hb 356
Modulated for Driven Circuits and Oscillators (AM and PM Noise) 357
Jitter for Driven Circuits 360
Timedomain for Driven Circuits 361

Oscillators 361
Modulated 362
Timedomain 362
FM Jitter 362
PM Jitter 363
Noise Calculations in the Simulator 363
Noise Factor and Noise Figure 363
Input-Referred Noise 365
Noise transfer function 366
Commonly Used Options 366
Cyclo2txtfile Option 369
Interpolation in Noise Files 370
hbXF 373
ADE Implementation 374
Driven Circuits 374
Oscillators 393
Examples 413
Driven Circuit Noise Setup 413
Oscillators 490
hbSP Analysis 522
Overview of hbsp 522
Example 523

4
Single Input Large and Small-signal Analyses 547
Overview of Simulation Capabilities 547
Overview of Periodic Steady-State (pss) Analysis 547
Example 548
Harmonic Balance 554
DC Principles 559
Convergence 561
Skipdc 562
Transient Principles 563
Convergence 563
Relref 563

Integration Methods . . . 568
Controlling Accuracy . . . 571
Shooting PSS . . . 575
Overview . . . 575
Oscillators . . . 576
Delay Time . . . 578
Piecewise Linear Sources for Power Supplies . . . 578
Tstab . . . 578
Beat Frequency . . . 578
Number of Harmonics . . . 579
Accuracy Defaults (errpreset) . . . 579
PSS Shooting Convergence . . . 581
ADE Implementation and Numerical Noise Floor . . . 582
Accuracy and Settings and Trade-Offs . . . 585
Which Engine Should be Selected? . . . 587
Important Outputs in the Spectre.out File . . . 588
Guidelines for Setting up Oscillators for Simulation . . . 591
Beat frequency . . . 591
Tstab . . . 591
Different ways to start the oscillator . . . 591
Saveinit- A way to save the startup waveform . . . 593
Oscillator Node- Used to estimate the frequency for the first pss iteration . . . 593
ADE Implementation . . . 594
Spectre Output File (Oscillators) . . . 597
Oscillator Tuning Mode . . . 599
PSS Options . . . 602
Convergence tab . . . 603
Accuracy Tab . . . 606
Output Tab . . . 622
Reuse Tab . . . 630
Misc Tab . . . 641
Help for Convergence Issues . . . 645
Examples . . . 648
Pss shooting with a single input to an amplifier or the LO applied in a mixer . . . 648
Plotting the output spectrum in dBm . . . 653
Same Example Using Harmonic Balance . . . 655

Loadpull 657
Two Input Frequencies 687
Sweeps 691
Compression Point 695
Voltage Gain 697
Power Gain 700
Power-Added Efficiency 702
Compression Versus Output Power 705
Power Gain Versus Output Power 706
Plotting THD 708
Large-signal IP3 710
Oscillator Simulation 718
Plotting the Oscillator Startup Waveform 721
Plotting the Oscillator Output Spectrum 724
Additions for Oscillator Swept Tuning Voltage 725
Plotting Output Power Versus Tuning Voltage 729
Plotting Output Power Versus Output Frequency 731
Plotting the Output Frequency Versus the Tuning Voltage 734
Plotting the Modulation Sensitivity 737
Oscillator Tuning Mode Analysis 740
Oscillator Tuning Mode With Sweeps 744
Periodic AC Analysis (PAC) 749
Small-signal Versus Large-Signal Analysis for Conversion Gain 749
Example: Conversion Gain (Down Conversion) 750
Example: Conversion Gain (Up Conversion) 758
PAC General Principles 761
Overview of Simulation Capabilities 766
ADE Implementation 778
Circuit, Input port Setting, and ADE Setup for all the Examples in This Section . . . 778
Using PAC Analysis for Conversion Gain Measurement 781
Sampled Circuits and Switched-capacitor Filters 789
Commonly Used PAC Options 790
PAC With Multiple PSS Inputs 794
Rapid IP3/IP2 802
Compression Distortion Summary 808
IM2 Distortion Summary 816

Modulated PAC Analysis 818
Sampled PAC Analysis 826
Periodic Noise Analysis (Pnoise) 834
Small-Signal Versus Large-Signal Analysis for Noise 834
Overview of Simulation Capabilities 835
Example 836
Principles of Pnoise 844
Noise Type=sources 845
Noise Output Near The First Harmonic 846
Frequency Sweep 846
Maximum Sideband 847
Full-Spectrum Pnoise 848
Setting Harmonics and Sidebands 849
Noise Separation 849
Multiple Pnoise 850
Noisetype = Modulated for Driven Circuits and Oscillators (AM and PM Noise) 851
Jitter for Driven Circuits 854
Timedomain for Driven Circuits 855
Oscillators 855
Modulated 856
Timedomain 856
FM Jitter 856
PM Jitter 857
Noise Calculations 857
Phase Noise (Driven Circuit) 857
Noise Factor and Noise Figure 857
Input-Referred Noise 861
Noise transfer function 861
Commonly Used Pnoise Options 862
Interpolation in Noise Files 863
ADE Implementation 874
General Notes 874
Driven Circuits 874
Oscillators 895
Examples 916
Driven Circuit Noise Setup 916

Multiple pnoise 959
Oscillators 998
Periodic Transfer Function Analysis (PXF) 1060
Small-signal Versus Large-Signal Analysis for Conversion Gain 1060
Example: Conversion Gain (Down conversion) 1061
Example: Conversion Gain (Up Conversion) 1070
Overview of Simulation Capabilities 1072
Normal Conversion Gain Measurement (Specialized Analysis = None) 1078
AM to PM measurement (Specialized Analysis = Modulated) 1089
Sampled PXF 1092
ADE Implementation 1093
PXF for a Normal Conversion Gain Measurement 1094
PXF Modulated (AM to PM Conversion) 1096
PXF Sampled 1098
PXF Options 1102
Examples for Driven Circuits 1112
PXF Normal Conversion Gain Measurement 1112
Using PXF Analysis for Conversion Gain Measurement 1114
PXF Modulated (AM to PM conversion) 1123
Sampled PXF 1132
Examples for Oscillators 1143
Conversion Gain From Power Supply to the Output Frequency 1143
Modulated PXF; Measuring AM to PM Conversion and Controlling Spurious Response 1148
Sampled PXF: Measuring Conversion Gain at a Threshold Crossing in a Ring Oscillator 1160
Periodic Stability Analysis (PSTB) 1171
Example: Variation in Loop Gain of an Opamp with Swept Input Signal Amplitude 1172
Overview of Simulation Capabilities 1181
ADE Implementation 1183
PSTB Setup 1183
PSTB Options 1185
The following options are not commonly used: 1186
Oscillator Example 1188
Periodic S-parameter Analysis (PSP) 1196
Example: Down-Conversion Diode Mixer 1196

Overview of Simulation Capabilities 1214
ADE Implementation 1216
PSP Setup 1216
PSP Options 1226
Examples 1230
Swept Input Power In a PA 1246

5
Multiple Input Large and Small-signal Analyses 1259
Quasi-Periodic Steady-State Analysis (QPSS) 1259
Example 1260
QPSS Shooting Concepts 1267
What Happens When you Start a QPSS Analysis? 1275
ADE Implementation 1278
QPSS Options 1280
Example 1293
Quasi-Periodic AC Analysis (QPAC) 1333
Example 1333
Qpac Concepts 1340
ADE Implementation 1351
Example 1357
Quasi Periodic Noise Analysis 1372
Overview 1372
Example 1373
Qpnoise Concepts 1380
Measuring Noise Figure 1382
Frequency Sweep 1382
Noise Separation 1385
Input-Referred Noise 1386
ADE Implementation 1387
Example 1394
Viewing Noise Separation Results 1413
QPXF Analysis 1448
Example: Conversion Gain 1449
Overview of Simulation Capabilities 1457

ADE Implementation 1465
QPXF for a Normal Conversion Gain Measurement 1465
QPXF Options 1467
Example 1471
Conversion Gain as a Function of Blocker Power 1485
QPSP Analysis 1493
Example: Down-Conversion Diode Mixer 1494
Overview of Simulation Capabilities 1513
ADE Implementation 1514
QPSP Setup 1514
QPSP Options 1524
Examples 1527

6
Envelope (ENVLP) Analysis 1541
Example 1542
Envelope Principles 1552
Envelope-Transient Simulation 1558
Transistor Level V/s Fast Envelope 1559
The Passband Model 1559
The Baseband Model 1560
Power Scaling in Baseband Mode 1560
WFreq 1561
When do I use Passband or Baseband? 1561
Fast Envelope Noise 1562
Wireless Mode in Envelope Analysis 1563
Supported Standards 1563
Wireless Analysis Vs. Traditional Envelope Following 1564
Limitation of Wireless Analysis 1564
Setting up the Envelope Analysis in ADE: Wireless Mode 1572
Setting up the Envelope Analysis in ADE: Using your own I and Q Files 1572
Frequency Modulated Input Signals 1599
Commonly Used Options 1605
Autonomous ENVLP Analysis (Oscillators) 1624
Examples 1642

7

Large Signal S-parameter Simulation (LSSP) 1643

Large Signal S Parameters . . . 1643
  Large Signal S parameters for a Two Port Circuit . . . 1644
  Circuit Setup . . . 1645
  ADE Setup . . . 1649
  Analysis Setup . . . 1649
  LSSP Amplitude Sweep . . . 1651
  Plotting the S11 Curve . . . 1653
  Plotting the Input Impedance Magnitude and Phase Curves . . . 1657

8

AnalogLib Components Used in RF Simulation 1659

The Delayline Element . . . 1659
The PORT Element . . . 1661
Capabilities of the port Component . . . 1661
Terminating the Port . . . 1662
  Parameters for the Port Component . . . 1664
  Port Parameters . . . 1664
Using the Harmonic Port with Harmonic Balance . . . 1665
  Source type . . . 1675
  DC Parameters . . . 1676
  DC voltage . . . 1676
  Pulse Waveform Parameters . . . 1677
  PWL Waveform Parameters . . . 1691
  Waveform Entry Method . . . 1693
  Sinusoidal Waveform Parameters . . . 1716
  Modulation Parameters . . . 1719
  Display second sinusoid . . . 1737
  Display multi sinusoid . . . 1739
  Exponential Waveform Parameters . . . 1744
  Bit Waveform Parameters . . . 1746
  Bit Waveform Examples . . . 1749
  PRBS Waveform Parameters . . . 1755

Jitter Generation 1771
PRBS Mode External Triggering 1792
PRBS Source with External Triggering Using a Pulse or PWL Signal 1795
Noise Parameters 1798
Small-Signal Parameters 1802
Temperature Effect Parameters 1804
Simulating Tabulated S-Parameters Using the Nport Component 1806
Convolution-Based Method 1813
Using the nport Component 1814
Nport Compression 1820
Controlling Rational Fit Accuracy 1831
Using the relerr and abserr Parameters 1832
Using the rational order (ratorder) Parameter 1834
Troubleshooting 1834
Assessing the Quality of the Rational Interpolation 1835
Model Reuse 1841
Dcblock, dcfeed, indq, and capq 1843
Dcfeed 1843
Dcblock 1847
indq 1850
capq 1859
Reference: S-Parameter Equations 1866
Network Parameters 1867
Equations for Network Parameters 1867
Two-Port Scalar Quantities 1869
Equations for Two-Port Scalar Quantities 1870
Two-Port Gain Quantities 1872
Equations for Two-Port Gain Calculations 1873
Two-Port Network Circles 1876
Equations for Two-Port Network Circle 1876
Equation for VSWR (Voltage Standing Wave Ratio) 1880
Equation for GD (group delay) 1880
References 1881

9
Cosimulation with MATLAB® and Simulink® 1883
Introduction to Cosimulation with MATLAB 1884
Software Requirements 1884
Setting Up and Running a Cosimulation 1884
Connecting the Coupler Block Into the System-Level Simulink Schematic 1885
Determining How You Want to Start and Run the Cosimulation 1889
Generating a Netlist for the Lower-Level Block 1889
Preparing the Netlist When Using ADE 1889
Preparing the Netlist Without Using a Graphical User Interface 1896
Running the Cosimulation 1898
Starting the Two Applications Separately 1898
Starting SpectreRF Manually and MATLAB Automatically 1899
Starting MATLAB Manually and SpectreRF Automatically 1899
MATLAB Support Matrix 1900

A
Design Workshops 1903
Introduction 1903
Using SpectreRF from the MMSIM Hierarchy 1903
Accessing the Most Current SpectreRF Documentation 1903
Creating a Local Editable Copy of the ExampleLibRF Library 1904
Downloading and Using GPDK180 1904
Starting Virtuoso 1911
Example Circuits 1913
Simulating Low Noise Amplifiers 1914
The lnaSimple Low Noise Amplifier Circuit 1914
Setting Up to Simulate the lnaSimple Low Noise Amplifier 1916
Opening the lnaSimple Circuit in the Schematic Window 1916
Choosing Simulator Options 1919
SP Analysis and Small Signal Gain 1924
Stability 1948
Linear 2-port noise analysis (NF, NFmin) and Noise circles 1959
Summary 1966

- Third-Order Intercept measurement with HB (2 tone HB) 1967
- Choosing Simulator Options 1974
- Measuring IP3 with Rapid IP3 1991
- Summary 1999

Simulating Oscillators 2000
Simulation Methods 2000
Phases of Autonomous PSS/HB Analysis 2001
Phase Noise and Oscillators 2002
Starting and Stabilizing Feedback Oscillators 2002
The Oscillator Circuit 2003
Setting Up to Simulate the Oscillator Circuit 2004
- Setting up ADE-L for Oscillator Simulation 2006

Calculating the Steady-State Solution using PSS Harmonic Balance 2013
- Setting up the PSS Analysis 2013
- Running the PSS analysis 2019

Oscillator Loop Gain Measurement 2026
- Setting up the stb Analysis 2028
- Setting up the PSS Analysis 2038
- Setting up the Pstb Analysis 2043
- Running the PSS, Pstb and stb analysis 2047
- Plotting the results 2047

Phase Noise Measurement and Noise Summary Table 2068
- Setting up the HB and HBnoise Analysis 2070

Oscillator Swept Tuning Range and Phase Noise Measurement 2109
- Setting up the HB and HBnoise Analysis 2111

Ring Oscillator Measurements 2137
- Starting and Stabilization of Ring Oscillators 2137
- The Oscillator Circuit 2138

Simulating the Oscillator Circuit 2138
Calculating the Steady-State Solution using PSS Shooting Analysis 2140
- Setting up the PSS Analysis 2141
- Running the PSS analysis 2154

FM Jitter Measurement using PSS Shooting and Pnoise Jitter Analyses 2163
- Determining FM Jitter 2163
- Setting up the PSS Analysis 2163
- Running the PSS and Pnoise analysis 2183

Calculating the Swept Tuning Range and Phase Noise for the Ring Oscillator 2194
Setting up the PSS Analysis 2194
Running the PSS and Pnoise analysis 2212
Summary 2220
Simulating Mixers 2221
The db_mixer and db_mixer_xmit Mixer Circuits 2221
Setting Up to Simulate the db_mixer Mixer 2223
Opening the db_mixer Mixer Circuit in the Schematic Window 2223
Choosing Simulator Options 2226
Setting Up the Simulation - Setting Design Variables 2232
Setting Up Model Libraries 2283
Setting Design Variables 2284
Setting up hbnoise to measure noise figure 2290
Third-Order Intercept measurement with HB 2313
Mixer Distortion Measurement 2357
Setting Up to Simulate the db_mixer_xmit Mixer 2384
Three Tone Spectral Content and Image Rejection 2387
Three Tone IP3 2414
Signal-to-Noise Ratio 2428
Three Tone HB Analysis Setup 2435
Summary 2462

B
Top-down RF Design Methodology 2465

Top-Down Design of RF Systems 2466
Use Model for Top Down Design 2467
Baseband Modeling 2470
Example Comparing Baseband and Passband Models 2472
rfLib Library Overview 2484
Use Model and Design Example 2488
Opening a New Schematic Window 2489
Opening the Analog Environment 2490
Constructing the Baseband Model for the Receiver 2490
Setting Variable Values for the Receiver Schematic 2528
Setting Up and Running a Transient Analysis 2531

Examining the Results: Eye Diagram, Histogram, and Scatter Plot 2533
Computing Minimized RMS Noise Using the Optimizer 2545
Summarizing the Design Procedure 2562
Creating a Passband View of the Architectural Model 2562
Comparing Baseband and Passband Models 2566
Relationship Between Baseband and Passband Noise 2569
Introduction to Analysis 2570
Preparation Steps for Analyses 2571

C
Documents That Ship in the Software Hierarchy 2579
SpectreRF_simulink_example.pdf 2580
EnvelopeAN.pdf 2580
LSSP_AN.pdf 2581
MatlabWorkshop.pdf 2581
PLL_Jitter_AN.pdf 2582
NS_AN.pdf 2582
PSRR_Drv_AN.pdf 2583
readme.txt 2583
HB_AN.pdf 2583
LTJM_AN.pdf 2583
PerturbationAN.pdf 2584
PSRR_Osc_AN.pdf 2584
RF_Blocks_AN.pdf 2585
JitterAN.pdf 2585
MatlabAN.pdf 2586
PstbAN.pdf 2586

Index 2587

Preface

Virtuoso® Spectre® circuit simulator or Virtuoso® Accelerated Parallel Simulator (APS) base products with the Virtuoso MMSIM RF analysis option provide simulation capabilities for RFIC designers. The simulators:

- Support efficient calculation of the operating point, transfer function, noise, and signal distortion of common RF and communication circuits, such as mixers, LNA, oscillators, sample and holds, and switched capacitor filters.
- Support a multi-technology simulation (MTS) mode that enables the simulation of systems consisting of blocks designed with different processes, such as RF System-in-Package (SIP).

This user guide assumes that you are familiar with:

- RF circuit design
- SPICE simulation
- The Virtuoso® Analog Design Environment (ADE)

The Virtuoso® Spectre® Simulator, the high performance Virtuoso® APS, and the Virtuoso MMSIM RF analysis options are part of the MMSIM (multi-mode simulation) portfolio and are accessible using individual a la carte licenses or MMSIM tokens. Note that mixing of tokens and a la carte license is not allowed. Table -1 on page 21 and Table -2 on page 22 show the capabilities offered with the base products Virtuoso Spectre simulator or Virtuoso APS and the Virtuoso MMSIM RF analysis option.

Table -1 Virtuoso Spectre® Simulator with Virtuoso MMSIM RF Analysis Option

Features
Transient Noise Analysis
DC, small-signal analyses and transient
Monte Carlo, DC mismatch, Parametric sweep
RF Harmonic Balance

Features
RF Shooting Newton
Large-signal Analysis - Periodic and Quasi-periodic (HB, PSS and QPSS)
Noise Analysis - Periodic and Quasi-periodic (pnoise, QPNoise, sampled, jitter)
Small-signal Analysis - Periodic and Quasi-periodic (PAC, PXF, PSP, QPAC, QPXF, QPSP)
Periodic Stability Analysis (PSTB)
RF Transmission Line Library
Envelope Analysis (AM, PM, FM, Autonomous)
Wireless Analysis
Wireless Standard Sources (such as LTE, Zigbee, and 802.11n)
Rapid Distortion Analyses - Perturbation-based IP2 and IP3
Co-simulation with Simulink® from The MathWorks
MMSIM Toolbox for MATLAB® from The MathWorks

Table -2 Virtuoso® Accelerated Parallel Simulator with Virtuoso MMSIM RF Analysis Option

Features
All analysis and features in Virtuoso Spectre
RF Harmonic Balance
RF Shooting Newton
Large-signal Analysis - Periodic and Quasi-periodic (HB, PSS and QPSS)
Noise Analysis - Periodic and Quasi-periodic (pnoise, QPNoise, sampled, jitter)
Small-signal Analysis - Periodic and Quasi-periodic (PAC, PXF, PSP, QPAC, QPXF, QPSP)
Periodic Stability Analysis (PSTB)
RF Transmission Line Library
Envelope Analysis (AM, PM, FM, Autonomous)
Wireless Analysis
Wireless Standard Sources (such as LTE, Zigbee, and 802.11n)

Features
Rapid Distortion Analyses - Perturbation-based IP2 and IP3

Cadence offers multi-core simulation up to 64 cores for RF analysis, enabled with the base product Virtuoso APS along with Virtuoso MMSIM RF analysis and the Virtuoso MMSIM CPU Accelerator options.

Please contact your account representative for more details on the licensing and packaging.

Using License Queuing

You can turn on license queuing by using the `lqtimeout` command line option:

```
spectre +lqtimeout time_in_seconds
```

If a license is not available when you begin a simulation job, the Spectre circuit simulator and APS wait in queue for a license for the specified time. If you specify the value 0 for this option, the Spectre circuit simulator waits indefinitely for a license. The `lqtimeout` option ha s no default value for the standalone Spectre circuit simulator. If you invoke Spectre through the Analog Design Environment, the default value for `lqtimeout` is 900 seconds.

You can use the `lqsleep` option to specify the interval (in seconds) at which the Spectre circuit simulator should check for license availability. The default value for `lqsleep` is 30 seconds.

```
spectre +lqsleep interval
```

For more information on any of the above options, see `spectre -h`.

Suspending and Resuming Licenses

You can direct Spectre and APS to release licenses when suspending a simulation job. This feature is aimed for users of simulation farms, where the licenses in use by a group of lower priority jobs may be needed for a group of higher priority jobs. To enable this feature, simply start Spectre with the `+lsuspend` command line option. In the Solaris environment, press `ctrl+z` to suspend the Spectre license. All licenses are checked in. To resume simulation, press `fg`. These keystrokes may not work if you have changed the default key bindings.

For information on tracking token licensing, see the *Virtuoso® Software Licensing and Configuration Guide*.

In Virtuoso® Analog Design Environment, the `lqtimeout` and `lqsleep` options are controlled by the following options:

```
spectre.envOpts lsuspend boolean t
spectre.envOpts licQueueTimeOut string "900"
spectre.envOpts licQueueSleep string "30"
```

Related Documents for Spectre

This user guide contains information about the functionality. The following documents provide more information about the RF analysis option with Virtuoso Spectre, Virtuoso APS, and related products.

- The Spectre circuit simulator is often run within the analog circuit design environment, under the Cadence design framework II. To see how the Spectre circuit simulator is run under the analog circuit design environment, read the *Virtuoso Analog Design Environment User Guide.*
- To learn more about specific parameters of components and analyses, consult the Spectre online help (`spectre -h`).
- To learn more about the equations used in the Spectre circuit simulator, consult the *Virtuoso Simulator Components and Device Models Manual.*
- The Spectre circuit simulator also includes a waveform display tool, Virtuoso Visualization and Analysis tool, to use to display simulation results. For more information about the tool, see the *Virtuoso Visualization and Analysis User Guide*.
- For more information about using the Spectre circuit simulator with Verilog-A, see the *Verilog-A Language Reference* manual.
- For more information about RF theory, see *Virtuoso Spectre Circuit Simulator RF Analysis Theory*.
- For more information about how you work with the design framework II interface, see *Cadence Design Framework II Help*.
- For more information about specific applications of Spectre analyses, see *The Designer's Guide to SPICE & Spectre*[1].

Third Party Tools

To view any `.swf` multimedia files, you need:

1. Kundert, Kenneth S. *The Designer's Guide to SPICE & Spectre.* Boston: Kluwer Academic Publishers, 1995.

- Flash-enabled web browser, for example, Internet Explorer 5.0 or later, Netscape 6.0 or later, or Mozilla Firefox 1.6 or later. Alternatively, you can download Flash Player (version 6.0 or later) directly from the Adobe website.
- Speakers and a sound card installed on your computer for videos with audio.

Typographic and Syntax Conventions

This list describes the syntax conventions used for the Spectre circuit simulator.

`literal`	Nonitalic words indicate keywords that you must enter literally. These keywords represent command (function, routine) or option names, filenames and paths, and any other sort of type-in commands.
`argument`	Words in italics indicate user-defined arguments for which you must substitute a name or a value. (The characters before the underscore (_) in the word indicate the data types that this argument can take. Names are case sensitive.
|	Vertical bars (OR-bars) separate possible choices for a single argument. They take precedence over any other character.
`[ ]`	Brackets denote optional arguments. When used with OR-bars, they enclose a list of choices. You can choose one argument from the list.
`{ }`	Braces are used with OR-bars and enclose a list of choices. You must choose one argument from the list.
`...`	Three dots (...) indicate that you can repeat the previous argument. If you use them with brackets, you can specify zero or more arguments. If they are used without brackets, you must specify at least one argument, but you can specify more.

The language requires many characters not included in the preceding list. You must enter required characters exactly as shown.

1

Introduction

Simulation Basics

Fundamentally, SpectreRF provides large-signal and small-signal analyses. Large-signal analyses PSS, QPSS, HB, and Envelope solve all the equations, and capture all the harmonics that are produced by the system. Large-signal effects like compression are taken into account. Because all the nonlinearities of the system need to be solved for, the simulation time is longer than the simulation time for small-signal analyses where the large-signal effects like compression are not solved for. The nonlinearity solved for in the large-signal analysis is taken into account, so frequency translation effects caused by the large signal are present in the small-signal analyses, thus conversion gain and noise are accurate for systems that translate frequencies.

Fundamentals of RF Simulation

SpectreRF provides solutions for single or multi tone periodic systems using periodic steady-state (PSS) or quasi-periodic steady state (QPSS), and the small-signal analyses provided for all three. Noise simulations that include frequency translation are provided using Pnoise, hbnoise, or QPnoise. To measure small-signal conversion gain, pac, QPAC, pxf, and QPXF analyses are provided. To measure IP3, rapid IP3 is provided. To measure power gain, PSP and QPSP are provided. This is an extension of S-Parameters for systems that translate frequency. Non-periodic systems can be simulated using envelope

Shooting Newton and Harmonic Balance algorithm

SpectreRF provides a choice of simulation engines between the traditional *Shooting Newton method* (or simply, *Shooting method*) and the *Harmonic Balance method* (HB) with most analyses. The Harmonic Balance engine complements the capabilities of the Shooting method.

The combination of a PSS or QPSS analysis using the Shooting method with a time-varying small-signal analyses is efficient for circuits that respond in a strongly nonlinear manner to the LO or the clock. Consequently, you can use the SpectreRF simulations with the Shooting method to simulate strongly nonlinear circuits, such as switched-capacitor filters, switching mixers, chopper-stabilized amplifiers, PLL-based frequency multipliers, and sample-and-holds. You can use the SpectreRF simulations with the Harmonic Balance method to simulate circuits that produce a reasonably small number of harmonics, such as Low-Noise Amplifiers (LNAs) or Gilbert cell mixers.

Generally, PSS is used for circuits with a single input frequency. Either Shooting or Harmonic Balance can be used. Usually QPSS or HB is used for circuits that have more than one input frequency. QPSS runs much faster and requires less memory when Harmonic Balance is used as the engine. Use Shooting only for very nonlinear circuits like samplers with the sample clock and one or more inputs applied, or switched-capacitor circuits with the switch clock and one or more inputs applied.

Shooting Method

SpectreRF has traditionally used an engine known as the *Shooting method* [kundert90] to implement periodic and quasi-periodic analyses and the envelope analysis. The Shooting method is a time domain method.

Harmonic Balance Method

The Harmonic Balance engine supports frequency domain Harmonic Balance analyses. It provides efficient and robust simulation for circuits that produce a reasonable small number of harmonics. The Harmonic Balance engine is supported on the Solaris, Linux, HP and IBM platforms for both 32 and 64 bit architectures. See *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more information on the Harmonic Balance engine.

SpectreRF Analyses

The Virtuoso® Spectre® circuit simulator RF analysis (SpectreRF) analyses add capabilities to the Virtuoso Spectre circuit simulator, such as direct, efficient computation of steady-state solutions and simulation of circuits that translate frequency. You use the SpectreRF analyses in combination with the Fourier analysis capability of the Spectre circuit simulator and with the Verilog®-A behavioral modeling language.

Periodic Analyses

SpectreRF adds periodic large and small-signal analyses to Spectre simulation.

- **Periodic Steady-State Analysis, PSS (Large-Signal)**

  PSS captures the full non-linear behavior of the circuit usually with a single input applied to the circuit. Shooting or Harmonic Balance can be used for PSS.

- **Periodic AC Analysis, PAC (Small-Signal)**

  PAC allows the small-signal conversion gain and allows the small-signal IP3 to be measured.

- **Periodic S-parameter Analysis, PSP (Small-Signal)**

  PSP allows the small-signal power conversion gain to be measured, and allows the input match at (for example) the RF and the output match at the IF to be measured in the same run.

- **Periodic Transfer Function Analysis, PXF (Small-Signal)**

  PXF can be used to measure small-signal conversion gain to be measured and also PSRR with frequency translations can be measured.

- **Periodic Noise Analysis, Pnoise (Small-Signal)**

  Pnoise allows the noise of systems that translate frequencies to be measured accurately.

- **Periodic Stability Pstb** (**Small-Signal**)

  Pstb measures the open-loop gain and phase for systems where there is a large signal applied. The large signal can cause nonlinearity which might affect the stability of feedback systems.

- **Harmonic balance analysis, HB**

  HB calculates the harmonics and mixing products for all the inputs.

- **Harmonic Balance AC analysis, HBAC**

  HBAC allows the small-signal conversion gain and the small-signal IP3 to be calculated.

- **Harmonic Balance Noise, HBnoise**

  HBnoise allows the Pnoise of systems that translate frequencies to be calculated accurately. Hbxf is provided for the measurement of conversion gain from the sources in the circuit to the output specified for the noise analysis.

- **Harmonic Balance S-Parameter, Hbsp (Small-Signal)**

  Hbsp allows the small-signal power conversion gain to be measured, and allows the input match at (for example) the RF and the output match at the IF to be measured in the same run.

For details on the periodic analyses, see *Virtuoso Spectre Circuit Simulator RF Analysis Theory*.

Periodic Steady-State (PSS) analysis is a large-signal analysis that directly computes the periodic steady-state response of a circuit. With PSS, simulation times are independent of the time constants of the circuit, so PSS can quickly compute the steady-state response of circuits with long time constants, such as high-Q filters and oscillators. You can also sweep frequency or other variables using PSS.

Generally, PSS is used for circuits that have a single input frequency. Although it can simulate circuits with multiple inputs as long as fewer than 50 periods of the highest input frequency or less are simulated, it solves for all of the harmonics produced by the system. If you have a system that has 900MHz and 1GHz, and you want the harmonics though 3.5GHz, this requires 35 harmonics of 100MHz, and is a reasonable problem for PSS to simulate. If the system had 1GHz and 1GHz plus 1MHz, and you wanted harmonics through 3.5GHz, PSS would need to solve for 3,500 harmonics of 1MHz, which would take a very long time, and lots of computer memory. In this case, use QPSS or HB because these analyses only solve for the actual mixing products that are produced by the system. This is a much smaller number of harmonics, thus the runtime is much faster, and it is just as accurate.

After completing a PSS analysis, the SpectreRF simulator can model frequency conversion effects by performing one or more of the periodic small-signal analyses, Periodic AC analysis (PAC) (caused by the nonlinearity from the large signal applied to the circuit), Periodic S-parameter analysis (PSP), Periodic Transfer Function analysis (PXF), Periodic Stability (Pstb), and Periodic Noise analysis (Pnoise). The periodic small-signal analyses are similar to the Spectre L AC, SP, XF, STB, and Noise analyses, but you can apply the periodic small-signal analyses to periodically driven circuits that exhibit frequency conversion. Examples of important frequency conversion effects include conversion gain in mixers, noise in oscillators, and filtering using switched-capacitors.

Therefore, with periodic small-signal analyses you apply a small signal at a frequency that may be non commensurate (not harmonically related) to the small signal fundamental. This small signal is assumed to be small enough so that it is not distorted by the circuit.

Quasi-Periodic Analyses

SpectreRF adds quasi-periodic large and small-signal analyses to Spectre L simulation. Quasi-periodic means that even systems with input signals that don't have a rational number as the fundamental periodicity of the system can still be simulated. If for example, if a system has an input at 1GHz and an input which is a multiple of pi, this system can still be simulated with QPSS.

- **Quasi-Periodic Steady-State Analysis, QPSS (Large-Signal)**

QPSS captures the full non-linear behavior of the circuit with two or more inputs applied to the circuit. For most systems, Harmonic Balance is more efficient.

- **Quasi-Periodic AC Analysis, QPAC (Small-Signal)**

  QPAC allows the measurement of small-signal conversion gain. QPAC takes into account all the harmonics that are calculated by the QPSS analysis.

- **Quasi-Periodic S-parameter Analysis, QPSP (Small-Signal)**

  QPSP allows the small-signal power conversion gain to be measured, and allows the input match at (for example) the RF and the output match at the IF to be measured in the same run. It also allows the measurement of stability factor on amplifiers with large signals present at the input.

- **Quasi-Periodic Transfer Function Analysis, QPXF (Small-Signal)**

  QPXF can be used to measure small-signal conversion gain and also PSRR with frequency translations taken into account can be measured.

- **Quasi-Periodic Noise Analysis, QPnoise (Small-Signal)**

  QPnoise allows the noise of systems that translate frequencies to be measured accurately. All the harmonics of the tones applied in the QPSS are taken into account. Therefore, QPnoise is useful for measuring receiver desensitization when a blocker is applied at the RF input.

- **Quasi-Periodic S-parameter Analysis, QPSP (Small-signal)**

  QPSP allows the measurement of S-Parameters for circuits that have two or more inputs and that translate frequency. For example, S11 is at the input frequency, S22 is at the output frequency, and S21 measures the forward conversion gain.

For details on the quasi-periodic analyses, see *Virtuoso Spectre Circuit Simulator RF Analysis Theory*.

Quasi-Periodic Steady-State (QPSS) analysis, a large-signal analysis, is used for circuits with multiple large tones. With QPSS, you can model periodic distortion and include harmonic effects.

QPSS computes both a large signal, the periodic steady-state response of the circuit, and also the distortion effects of a specified number of moderate signals, including the distortion effects of the number of harmonics that you choose. In Shooting, one signal must be designated large. This signal should be the signal that causes the largest amount of distortion in the system, or the signal that is non-sinusoidal. In Harmonic Balance, tstab should be set to yes for this signal. With QPSS, you can apply one or more additional signals at frequencies not harmonically related to the large signal, and these signals can be large enough to create distortion. Shooting requires that these signals be sinusoidal, but Harmonic Balance allows one additional signal to be a pulse signal.

Quasi-Periodic Noise (QPnoise) analysis is similar to a noise analysis, except that it includes frequency conversion and intermodulation effects. QPnoise analysis is useful for predicting the noise behavior of mixers, switched-capacitor filters and other periodically or quasi-periodically driven circuits. Quasi-periodic systems are systems that have an irrational number as one of the input frequencies. Because of this, there is no exact period of the output waveform.

The Quasi-Periodic AC (QPAC), Quasi-Periodic S-parameter (QPSP) and Quasi-Periodic Transfer Function (QPXF) analyses all work in a similar way as the Spectre L AC, SP and XF analyses except that they include the frequency translations that occur because of a large-signal LO or clock input.

Envelope Analysis

Envelope analysis allows RF circuit designers to efficiently and accurately predict the envelope response of the RF circuits used in communication systems. This allows the ACPR and EVM to be calculated for power amplifier and transmit modulators to be calculated. The new fast envelope capability in release 10.1 can run 100 times faster than traditional envelope with very little loss in accuracy.

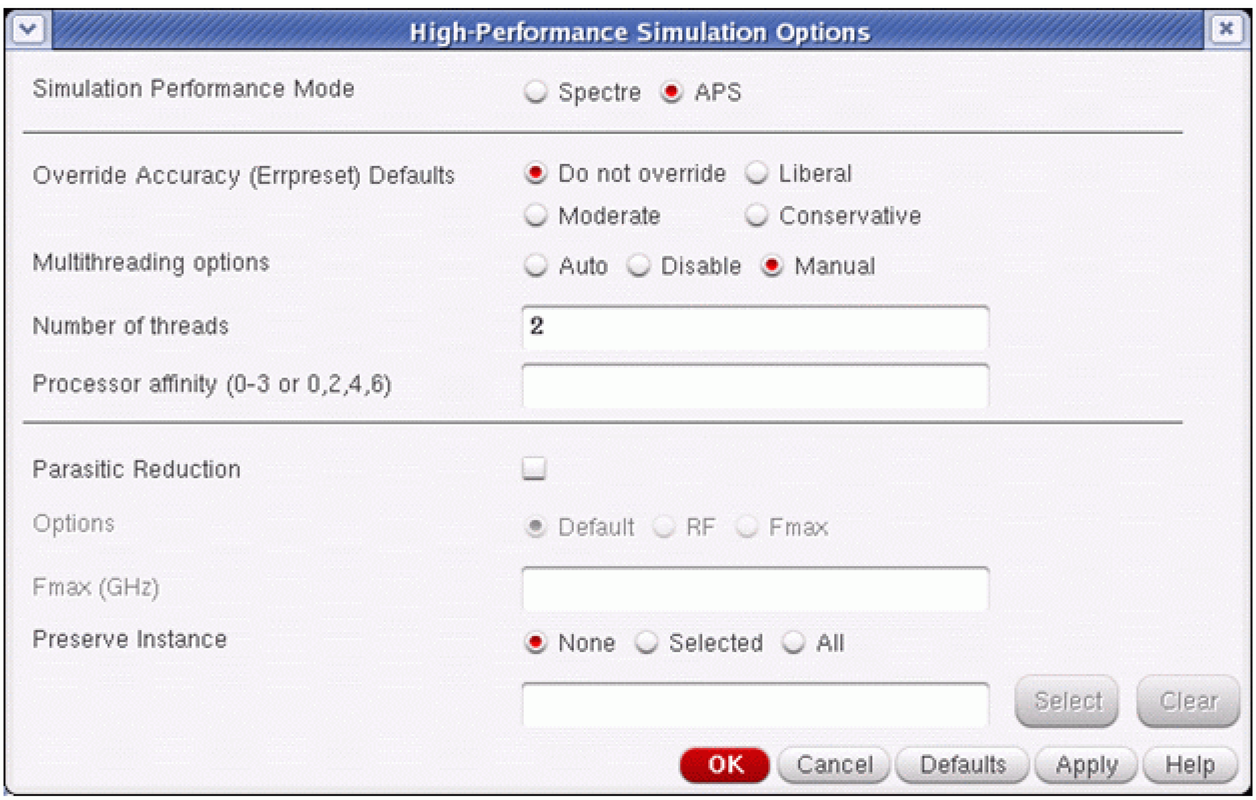
For details on the Envelope analysis, see *Virtuoso Spectre Circuit Simulator RF Analysis Theory.*

APS and Parasitic Reduction

Parasitic reduction is used in post parasitic re-simulations. It is RC reduction only with L and K values are preserved. Reducing the RC networks reduces the number of nodes in the circuit, thereby reducing the number of matrix calculations, so the simulation runs faster. It takes multiple poles at high frequencies and reduces them to a smaller order RC network with lower pole frequencies. When RC reduction is selected with the default settings, the minimum pole frequency in the reduced netlist is 1GHz. This might cause errors for RF circuits because the pole might be below the operating frequency. When RF is selected, the minimum pole frequency is raised to 30GHz. This will result is less reduction, and thus longer simulation times compared to the default setting. Selecting Fmax allows the user to set the minimum pole frequency in GHz. If you want 10GHz as the minimum pole frequency set the `Fmax`(GHz) field to 10, and not 10G. This allows you to make trade-offs between accuracy and runtime for your needs.

APS makes the CMOS models more efficient and allowing multi-threading for the device current evaluation. This will be relatively more efficient on schematic-level circuits and adds the transient APS algorithms for DC and tstab solutions. It also allows multi-threading for all the matrix operations for Harmonic Balance and shooting and their small-signal analyses.

The processor affinity field should be used when you are running two or more jobs on the same multi-core machine. For example, if you have an 8 core machine and you are running two 4-thread jobs; assigning one job to cores 0 through 3 and the other to 4 though 7 saves time by having the operating system always use the same range of cores. This can save some time since the OS has less work to do. Note that if one of the cores you assign the job to already has a job, the entire simulation will run slowly because that one core will need to run two jobs by sharing time between them. The assumption is that all the threads will run on cores that are fully dedicated to the APS job.



Co-simulation with Virtuoso AMS Designer

SpectreRF co-simulation with Virtuoso AMS Designer enables the verification of full chip RF transceivers with mixed-signal baseband and digital control circuits. Using SpectreRF envelope analysis in AMS Designer, you can calculate the transient envelope response of RF circuits while simulating the rest of the system using digital event-driven simulation or mixed-signal behavioral models. Co-simulation allows fast verification of communication systems in which the system architecture is highly dependent on the performance of the individual RF blocks. Using SpectreRF periodic and quasi-periodic steady state analysis in AMS Designer enables fast and accurate simulation of digitally calibrated RF blocks. The technology makes it possible to analyze the effect of digital control on the performance of RF circuits.

For more information about cosimulating with AMS Designer, see *Virtuoso AMS Environment User Guide.*

Large vs. Small Signal Analysis

SpectreRF provides a variety of time-varying small signal analysis for both periodic and quasi-periodic circuits. These small-signal analyses accurately model the frequency translation effects of time-varying circuits. Rather than using traditional small-signal analyses for circuits that exhibit no frequency translation, such as amplifiers and filters, you can simulate circuits that translate frequencies using time-varying small-signal analyses.

Circuits designed to translate from one frequency to another include mixers, samplers, frequency multipliers, phase-locked loops and parametric oscillators. Such circuits are commonly found in wireless communication systems.

Other circuits that translate energy between frequencies as a side effect include oscillators, switched-capacitor and switched-current filters, chopper-stabilized and parametric amplifiers, and sample-and-hold circuits. These circuits are found in both analog and RF circuits.

The quasi-periodic small-signal analyses accurately model the small signal characteristics of circuits with a quasi-periodic operating point, such as mixers with multiple LO frequencies or large RF inputs. The periodic small-signal analyses are more useful for circuits with a single input frequency.

Applying a time-varying small-signal analysis is a two-step process.

First, the simulator ignores the small input or noise signals while performing a PSS, QPSS or Hb analysis to compute the steady-state response to the remaining large-signals, such as the LO or the clock. The PSS or QPSS analysis is the full nonlinear solution for the circuit.

For each subsequent small-signal analysis, the simulator uses the nonlinear operating point computed by the PSS, HB, or QPSS analysis to predict the circuit response to a small sinusoid at an arbitrary frequency. You can perform any number of small-signal analyses after calculating the time-varying large-signal operating point.

The input signals for the small-signal analyses must be sufficiently small that the circuit does not respond to them in a significantly nonlinear fashion. You should use input signals that are at least 10 dB smaller than the 1 dB compression point. This restriction does not apply to the signals you apply in the large-signal analysis.

This two-step process is widely applicable because most circuits that translate frequency react in a strongly nonlinear manner to one stimulus, usually either the LO or the clock, while they react in a nearly-linear manner to other stimuli such as the inputs. A mixer is a typical example. Its noise and conversion characteristics improve if it is discontinuously switched between two states by the LO, yet it must respond linearly to the input signal over a wide dynamic range.

To analyze a mixer with a small RF input and a single LO, you should use a PSS large-signal analysis using shooting or hb as the engine followed by one or more of the PAC, Pnoise, PSP or PXF small-signal analyses.

If the mixer has a small RF input and a large blocker as well as the LO, then a QPSS analysis would be the more appropriate large-signal analysis. Follow the QPSS analysis with one or more of the QPAC, QPNoise, QPSP or QPXF small-signal analyses for the RF input. The basic idea is this: If you have signals in your circuit that cause large-signal effects (like a blocker) this signal must be applied in a large-signal analysis like PSS for one input, or QPSS (or HB) for multiple inputs. For signals that are largely linear (like the conversion product of a receive mixer), you can use small-signal analyses which run faster. If the signals are largely sinusoidal, then hb and the small-signal analyses hbac, hbnoise, and hbsp can also be used.

Some circuits, such as frequency dividers, generate subharmonics. PSS can simulate the large-signal behavior of such circuits if you specify the period T to be that of the subharmonic. For other circuits, such as delta-sigma modulators, the periodically driven circuits respond chaotically, and you must use transient analysis rather than the PSS or QPSS analyses.

With the time-varying small-signal analyses such as QPAC or PXF, unlike traditional small-signal analyses such as AC or XF, there are many transfer functions between any single input and output due to mixing with harmonics. Usually, however, only one or two harmonics provide useful information. For example, when you analyze the down-conversion mixers found in receivers, you want to know about the transfer function that maps the input signal at the RF to the output signal at the IF, which is usually the LO minus the RF.

2

Setting up a SpectreRF Simulation

Using Virtuoso ADE

Many people use ADE as a simulation environment because of the benefits it provides. The netlister netlists directly from the schematic. The direct plot form automates all of the standard RF measurements. Cross-probing to the schematic is provided. Easy set-up of the analyses to be run is provided. In ADE XL, multiple tests can be run at the same time. Monte Carlo and optimization are provided. In ADE GXL, you can estimate the parasitics before the layout is done to get a good estimate of the after-layout performance, and to identify nets that are very sensitive to layout parasitics.

A Testbench

A testbench is the schematic with the sources set up to provide the stimulus, and the different analyses to provide the measurements. In the schematic, make sure that the sources used for inputs and bias are at the top level, and have the circuit instantiated as a symbol (which could have levels of hierarchy below that). This allows these sources to be connected to the circuit but not have any connection to layout. Once the test setup has been made in ADE, that setup can be saved as an ADE state so that you don't have to start from scratch the next time you restart ADE.

ADE

ADE is the simulation environment for your circuit. once the schematic has been entered, ADE is started from the schematic. ADE L looks like the ADE from many years ago. It allows the setup of a single test, and allows parametric plots. It has three main sections: Variables, analysis setups, and calculator expressions to be plotted automatically after the simulation runs. The figure below shows a basic set of variables along with an HB and HBAC simulation setup.

ADE XL and GXL has a very different appearance, but allows the advanced analyses like yield optimization. ADE XL and GXL use several assistants for the different tasks to make the setup easier. These assistants can be moved to your preferred places on the screen and saved. This allows your custom setup to be available later.

Analysis form

The *Choosing Analysis* form is provided for each analysis type you want to run. The *Choosing Analysis* form lists the sources in your circuit at the top. The number of harmonics and oversample factor are set in the edit fields just below the list. Accuracy is set by selecting liberal, moderate, or conservative. In general, to get a quick idea, choose liberal. For most simulations, choose moderate. When a more accurate simulation is desired, select conservative. If you have an oscillator, select the oscillator button. Sweeps can be set up in the form directly, or by using the parametric plot tool in ADE.

Each analysis has its own *Choosing Analysis* form where the appropriate information is entered for that analysis.

Direct Plot form

The *Direct Plot* form automates the plotting of signals. At the top, all the different analyses that were run are listed. Just below are the different things that can be plotted. In SpectreRF, both the time and frequency domain results are provided regardless of whether the simulation is done in the time domain using Shooting, or in the frequency domain using Harmonic Balance. Peak or RMS results are available, and when frequency domain results are present, modifiers are allowed so that dB or phase could be selected for the results.

When a selection is made in the schematic, the waveform tool displays the result.

Netlist Driven Flows

Netlist-driven flows start with a netlist. This might be done manually, or from a netlister that is developed in house. The netlister may or may not add the analysis statements to the netlist. One advantage to running from the netlist is the ability to alter a value and then run again, or to run multiple analyses of the same type with different settings.

To get spectre help for the netlist-driven flow, type spectre -h in a shell window. This gives a list of topics that have help available for, and then you might for example type spectre -h pss. When you do this, you will see the analysis statement and all the options along with all the values that are available.

When the netlist has the proper analysis statements, you can run spectre. Examples of this are shown below:

```
spectre netlist.scs

spectre +aps +mt=2 +log spectre.out netlist.scs (This runs APS with 2 threads and
     a log file spectre.out)
```

When Spectre runs, the results are saved in the <netlist_name>.raw directory. To view the results, type viva& (wavescan& in IC5141) in a shell window, and then plot the waveforms from the Results Browser and the Calculator.

3

Frequency Domain Analyses: Harmonic Balance

Overview of Simulation Capabilities

SpectreRF offers two different modes; Spectre and APS. Spectre uses traditional simulation techniques. APS makes the CMOS models more efficient and allows multi-threading for the device current evaluation. In addition, APS adds full parallel solving of all the matrices, thus speeding up the simulation even more. There is no loss in accuracy because there is no change in the actual equations that are simulated or in the options that control the accuracy of the simulation. These choices are available and are implemented for harmonic balance and periodic steady state simulations. Both large-signal harmonic balance and small-signal hbac, hbsp, and hbnoise analyses have the Spectre and APS implementations.

Large-Signal Harmonic Balance Overview

Harmonic balance is a large-signal analysis that solves in the frequency domain. It calculates the harmonics and mixing products of one or more inputs to the circuit. It takes into account all the large-signal effects in the circuit. Applications include measuring the harmonic content of a single input, calculating the large-signal IP3 of a power amplifier (with 2 input frequencies), and measuring the large-signal IP3 and IP2 of a transmit mixer (with 2 IF tones and an LO input). Harmonic balance can also provide a large-signal solution to the small-signal hbac, hbsp, and hbnoise analyses. More information on small-signal analyses is provided later in the chapter.

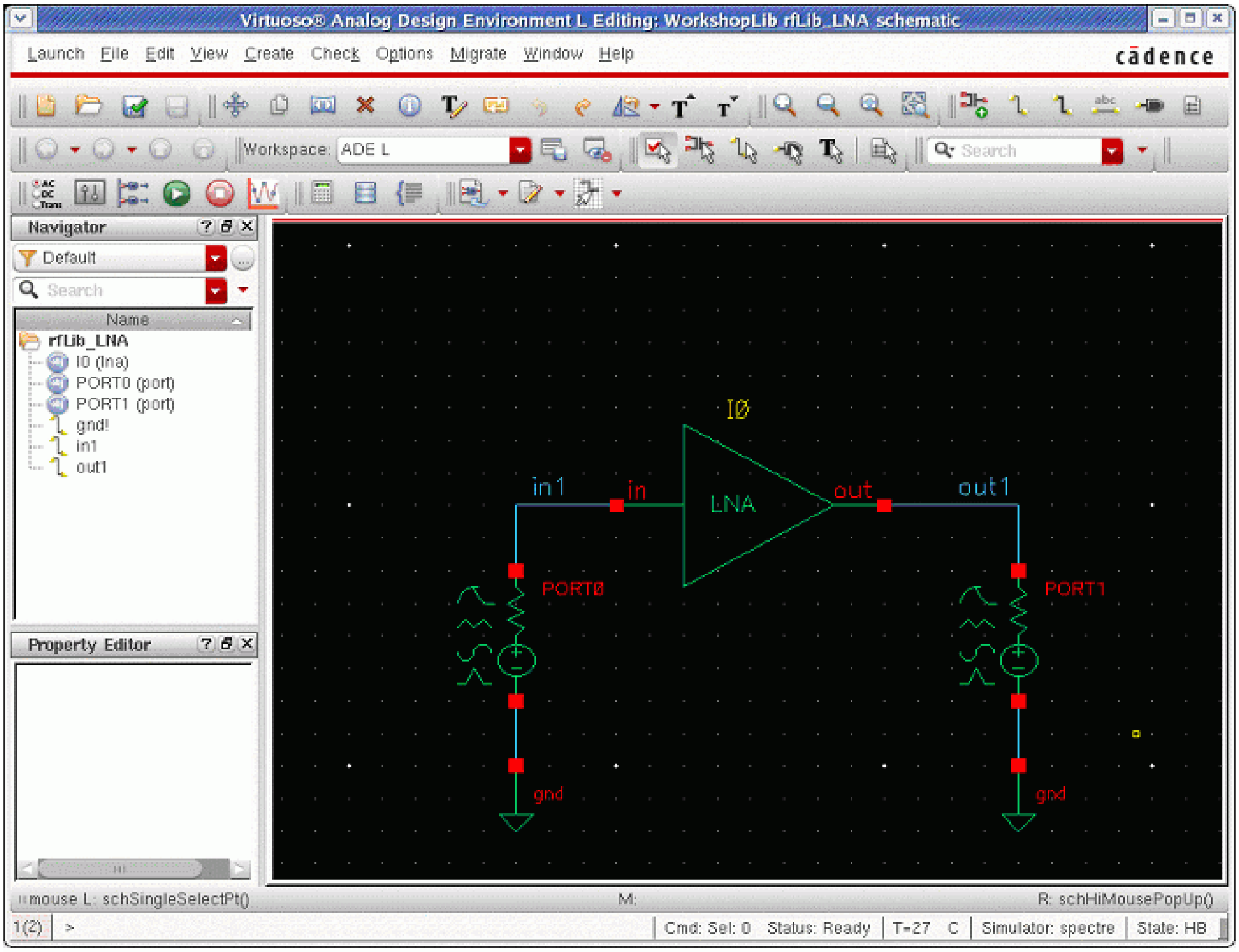
Harmonic balance is chosen for circuits that have near-sinusoidal signals and do not have high speed transitions. Although harmonic balance may work with circuits that have rapid transitions by setting lots of harmonics, it is likely that shooting (available in pss) will run faster. For LC oscillators and especially crystal oscillators, harmonic balance should be used. For ring oscillators, it is likely that shooting pss will be faster and more accurate.

The implementations of harmonic balance in the ADE *Choosing Analyses* form differ somewhat. The *Choosing Analyses* form has three choices that have harmonic balance (hb) as a choice: hb, pss with the engine set to harmonic balance (hb), and qpss with the engine set to harmonic balance (hb). In hb and qpss when harmonic balance (hb) is selected as the engine, the harmonics of only the inputs and the mixing products are calculated. In pss, when harmonic balance is selected as the engine, all the harmonics of the inputs are calculated, even if the power in those harmonics is zero. For example, if there is an input at 2GHz and another at 2.1GHz, the harmonics of 100MHz are calculated. Note that in order to calculate through the fifth harmonic of the input at 2.1GHz, 105 harmonics need to be calculated. For this reason, pss is usually used for a single input frequency, and qpss or hb is usually used for multiple input frequencies. The hb analysis does not require any distinction. It always calculates the harmonics of the inputs, and if multiple inputs are present, it also calculates the mixing frequencies. Also, the hb analysis has an auto mode where all you do is provide the input frequency. Everything else is automatic. Oscillator tuning mode is available where you specify a target frequency and a parameter to be varied, and then hb will tune the oscillator to that frequency. Usually, this is followed by a noise analysis, and this capability allows Monte Carlo simulations where the oscillator is tuned to the frequency, and the noise is measured at that frequency. In addition, hb has a mode where the gain compression is automatically determined. Using this capability speeds up compression analysis in Monte Carlo analyses. This chapter focuses on the hb selection in the ADE *Choosing Analyses* form.

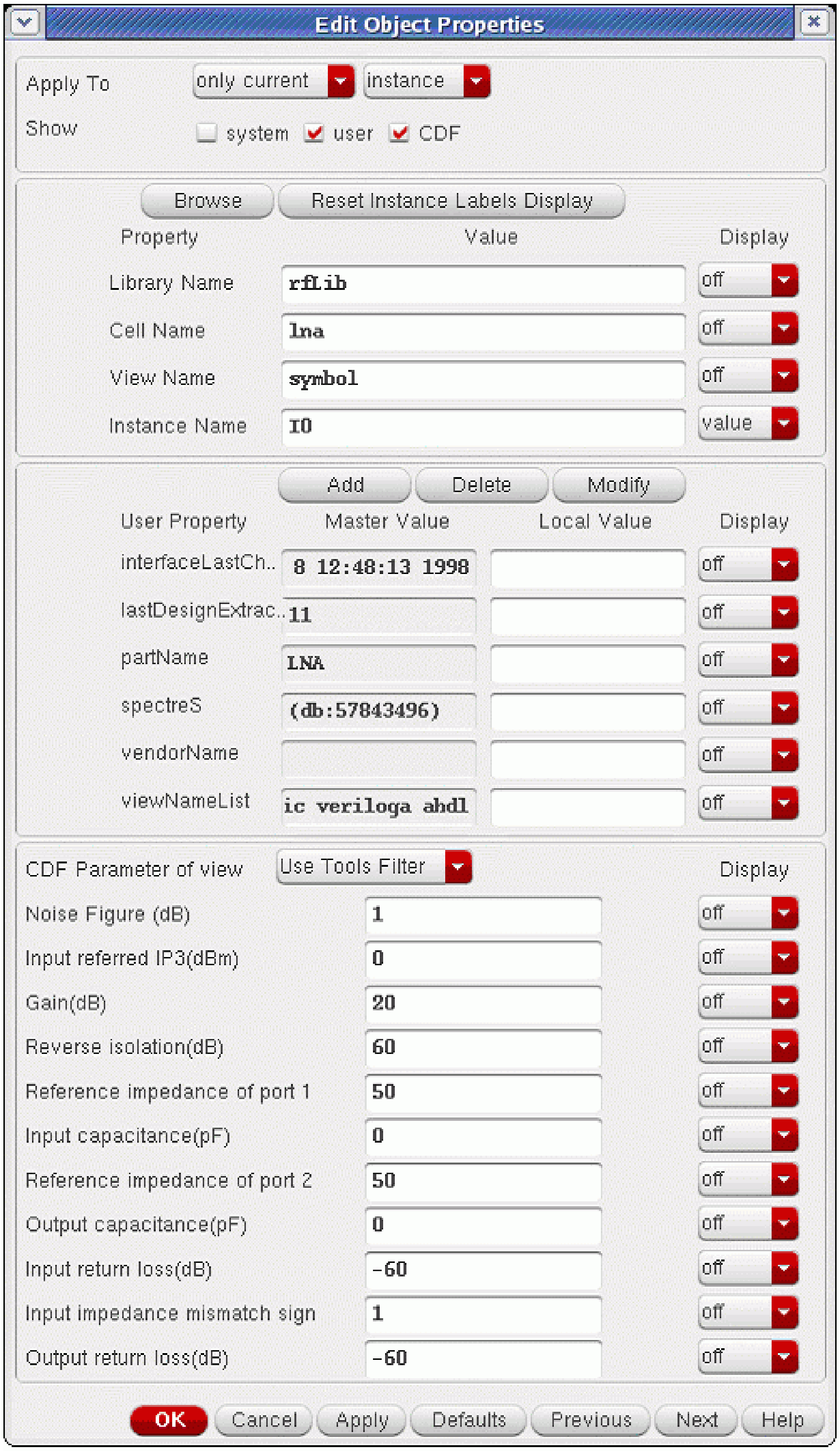
Harmonic Balance provides a solution in the frequency domain. Things that have direct frequency domain representations like capacitors, transmission lines, or S-parameter descriptions go directly into the frequency domain solution. Devices and other nonlinearities need to be evaluated in the time domain, with the ifft, and fft used to translate between the domains. This is an iterative process. Each successive iteration produces a more accurate solution. In other words, an exact solution is never attained. The iterations continue until the answer is "close enough". Close enough is set by choosing an accuracy default in the *Choosing Analyses* form. Liberal, moderate, and conservative accuracy levels are provided. Choose the appropriate accuracy level based on your requirement. If you are calculating very small amplitude harmonics, you might need to choose conservative.

Example

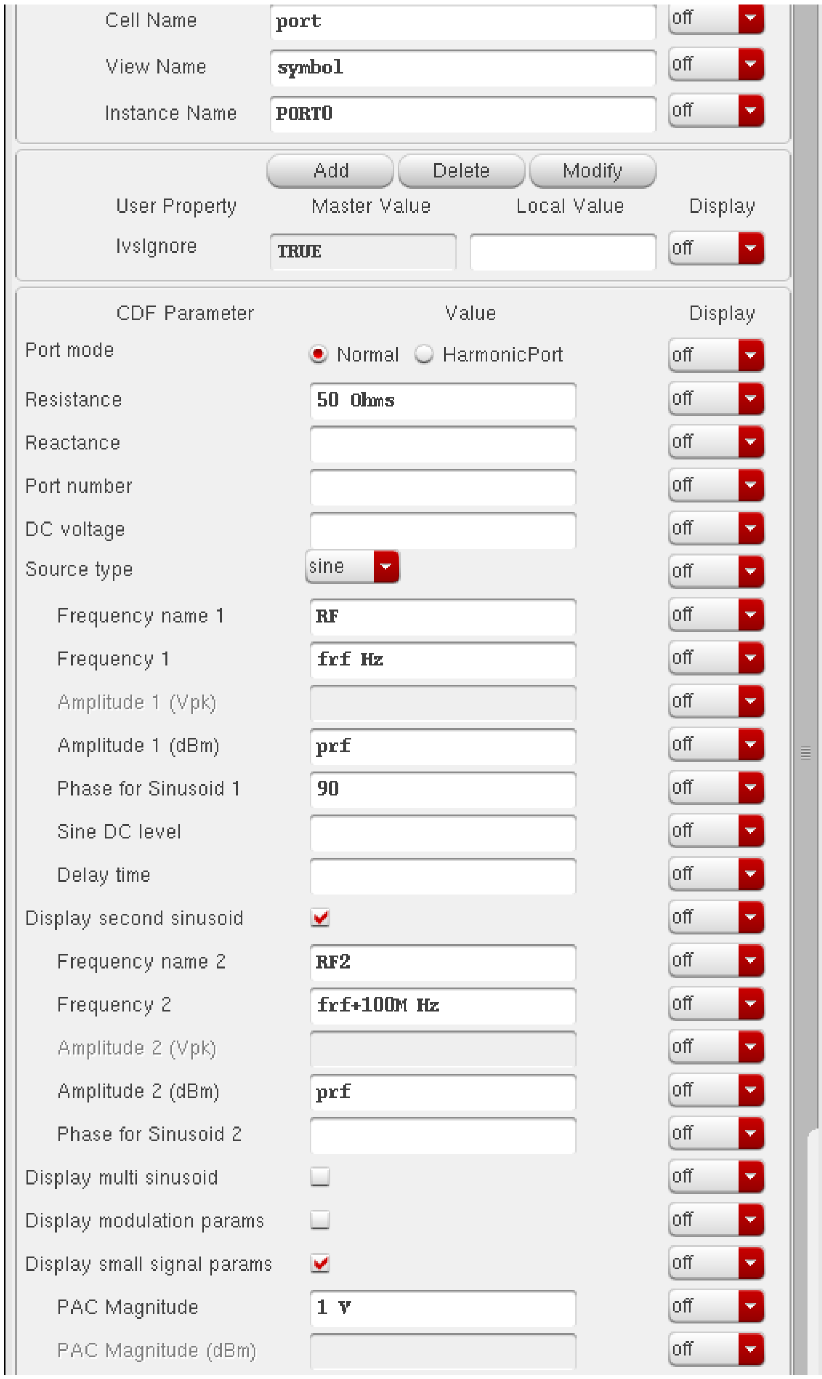
Consider a behavioral low noise amplifier shown below:



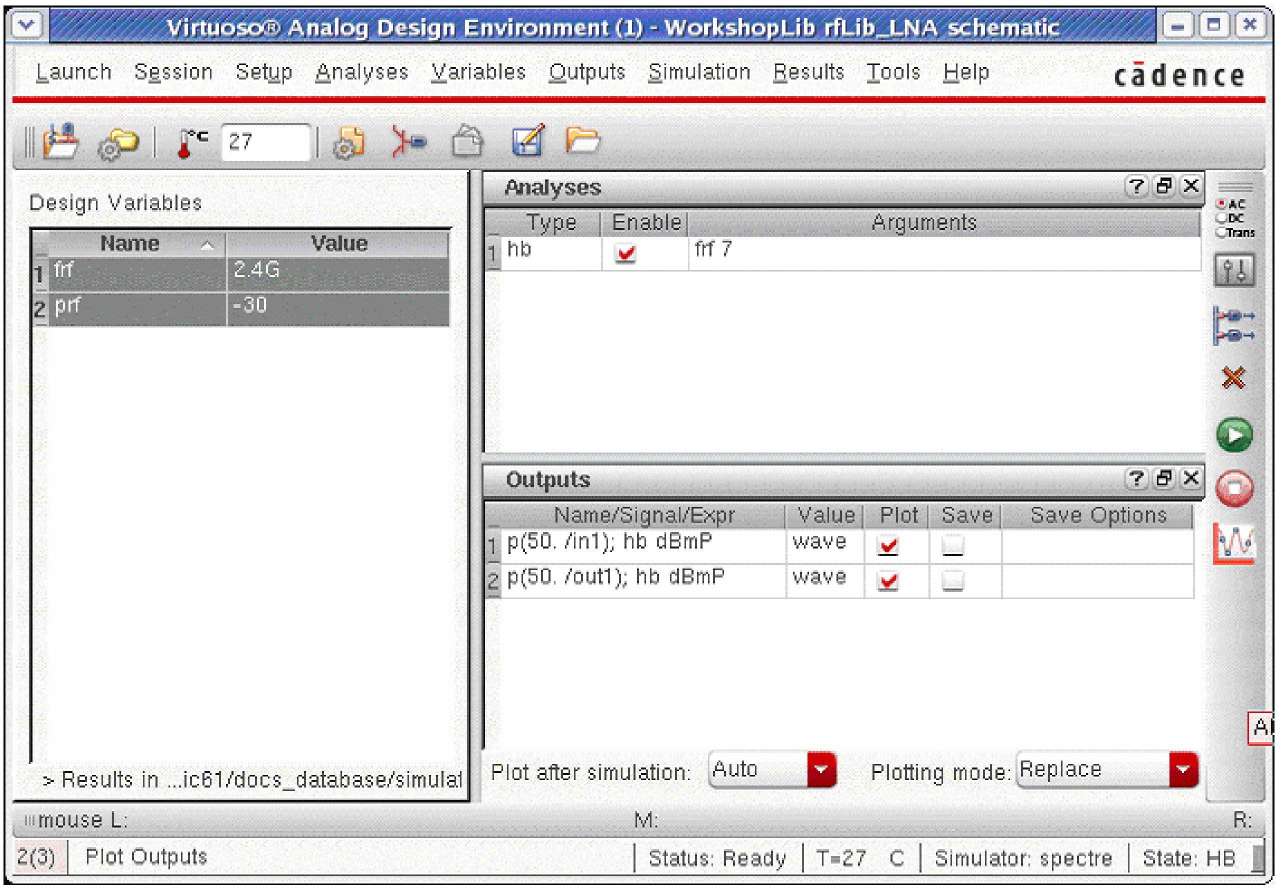
The LNA is a behavioral LNA from rfLib. The properties list is shown below. Note that the gain is 20 dB, and the amplifier is only slightly nonlinear.



The input port (the source on the left side in the above schematic) has the properties shown below.

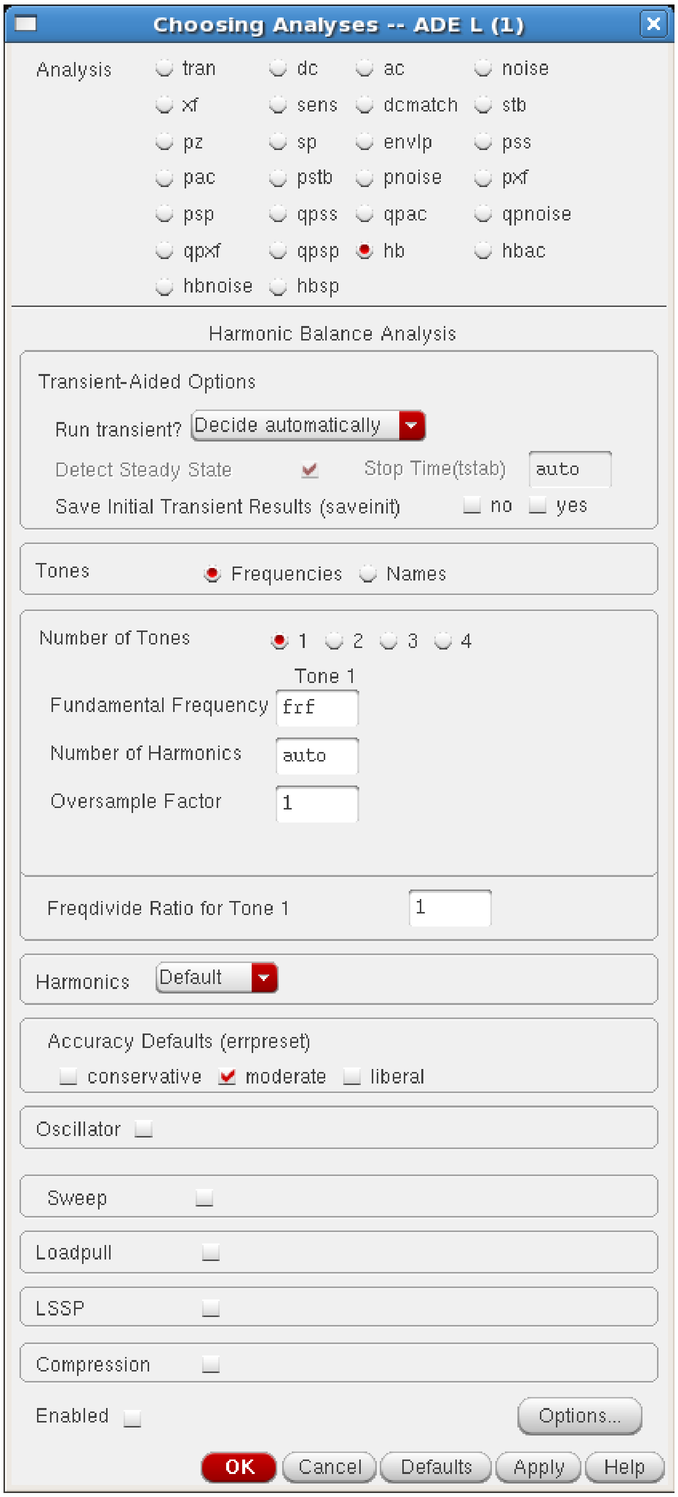


In general, give the signals names in the *Frequency name* property. In the example above, it is RF for the first frequency, and RF2 for the second. Giving names is required if you use the list of signals in the circuit in the harmonic balance Choosing Analyses form instead of specifying the frequencies. The frequency is set to a variable name called frf and the amplitude in dBm is set to the variable name prf. Setting variables in these properties makes it easy to change the frequency or amplitude without needing to change the schematic. The variable frf is set to 2.45G and the variable prf is set to -30 in the variables section of ADE, as shown below.



ADE is used to set up the different analyses by clicking on the *Analysis - Choose* menu or by clicking the AC-DC-Tran icon () located at the upper-right corner of the ADE window.

When you select *Analysis-Choose*, the *Choosing Analyses* form appears. At the top of the form, select the *hb* (harmonic balance) radio button.



In the *Choosing Analyses* form, the frequencies, the number of harmonics, oversample factor and accuracy are set. Oversample factor will be discussed later in the chapter. The *Fundamental Frequency* field can either have a number like 2.4G, or it can have the variable name that is used to set the frequency in the input port as shown above.

Run transient has three settings: Run Automatically, Yes, and No. Run automatically means that the transient will be run before the harmonic balance algorithm. In this mode, the transient starts with a small number of periods of the fundamental frequency for Tone1 as the transient stop time. As the transient analysis runs, the waveform is analyzed to see if steady-state has been reached. When steady-state is reached, a Fourier transform is performed on the last cycle of the transient waveform, and this is used as the starting point of the harmonic balance iterations. If steady-state has not been reached in this small number of periods, the transient is extended in time up to a maximum of 250 periods for a driven circuit, and 500 periods for an oscillator. If steady-state has not been detected in the last period of the transient, other continuation methods will be tried in an attempt to achieve convergence.

Run Automatically also sets the number of harmonics in the *Choosing Analyses* form for Tone1 to auto by default. The transient analysis waveforms at all the nodes in the circuit are analyzed, and based on the waveforms, the number of harmonics that are needed for the simulation to be accurate is set by the simulator. This can be manually overridden by setting a number of harmonics manually.

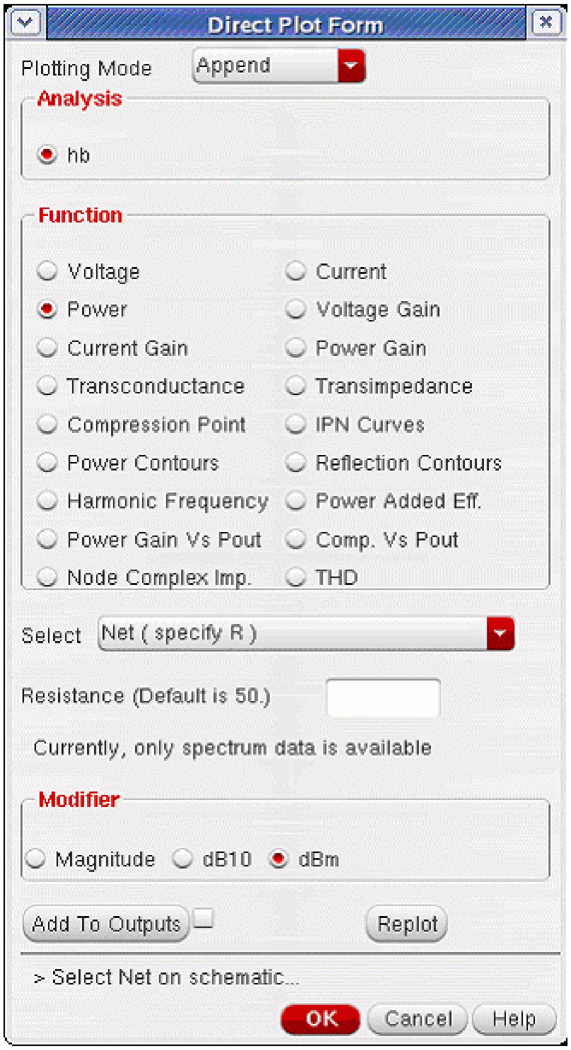
Setting Run transient to Yes requires you to enter a stop time for the initial transient analysis in the Stop Time (tstab) field. This mode allows the steady-state detection to be selected or not. If it is selected, whenever steady-state is detected in the transient analysis, it will terminate, run the fft, and start the harmonic balance iterations. In this mode, the stop time for the transient analysis cannot be automatically extended. If Detect Steady-State is toggled off, the transient will run to the specified stop time without checking for steady-state.

When Run transient is yes, the number of harmonics field remains blank by default. You can either enter the word auto, or you can manually set the number of harmonics by specifying a number. When auto is set, the transient analysis waveforms at all the nodes in the circuit are analyzed, and based on the waveforms, the number of harmonics that are needed for the simulation to be accurate is set by the simulator.

Setting Run Transient to No causes the harmonic balance algorithm to start from the DC solution without transient assist.

When you have completed the *Choosing Analyses* form, click *OK* and run the simulation. You can run the simulation by selecting *Simulation - Netlist and Run*, or by selecting the green arrow on the right side of the ADE window.

When the simulation completes, select *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed.



At any time, the next thing that needs to be accomplished is shown at the bottom of the form.

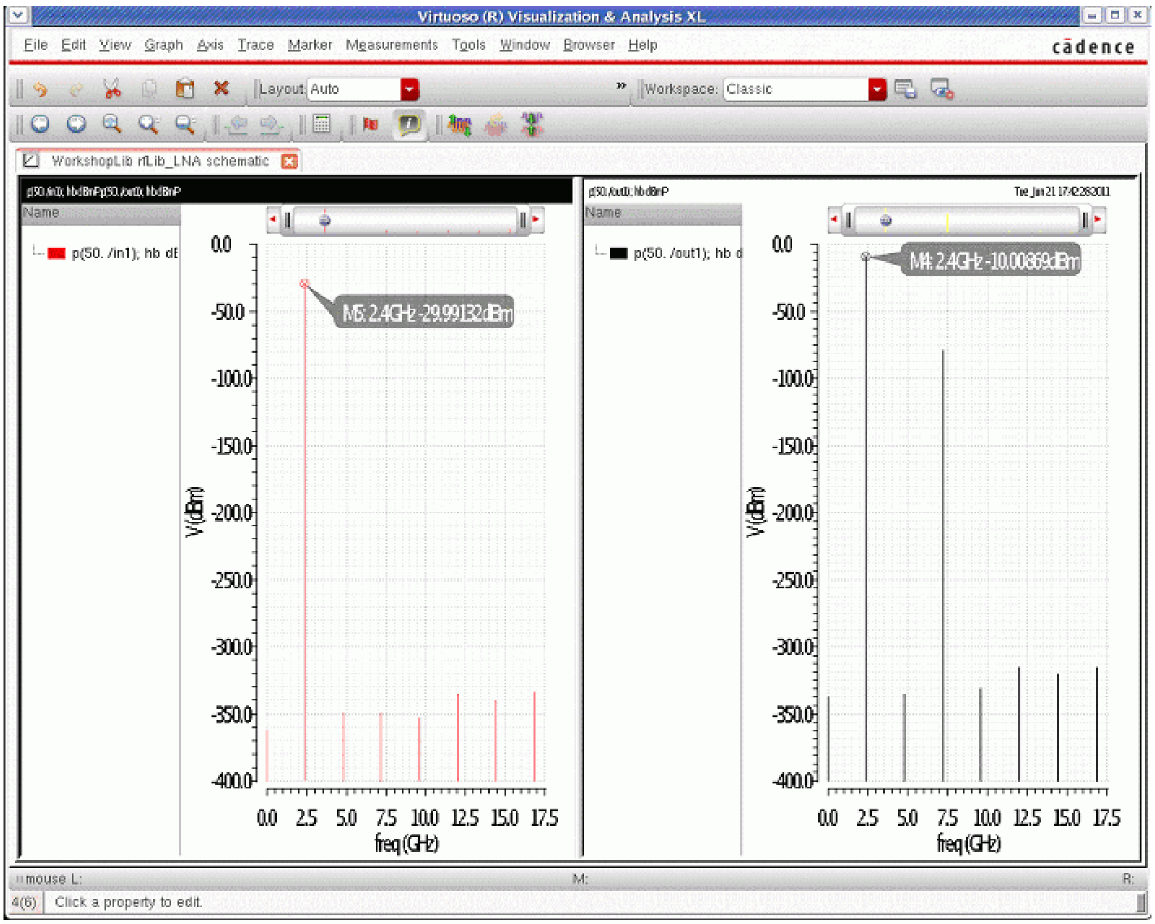
The *Analysis* section at the top of the form displays the list of different analyses that were run. In this example, only *hb* results are available.

The *Function* selection is where the type of data that is desired to be plotted is selected. *Power* is shown above.

To get the power based on the voltage in a net, *Net(specify R)* is chosen from the drop-down list.

Next select Magnitude, which gives power in Watts, dB10, which gives dB with respect to 1 Watt, or dBm, which gives dB with respect to 1 milliwatt. from the *Modifier* section. *dBm* is shown above.

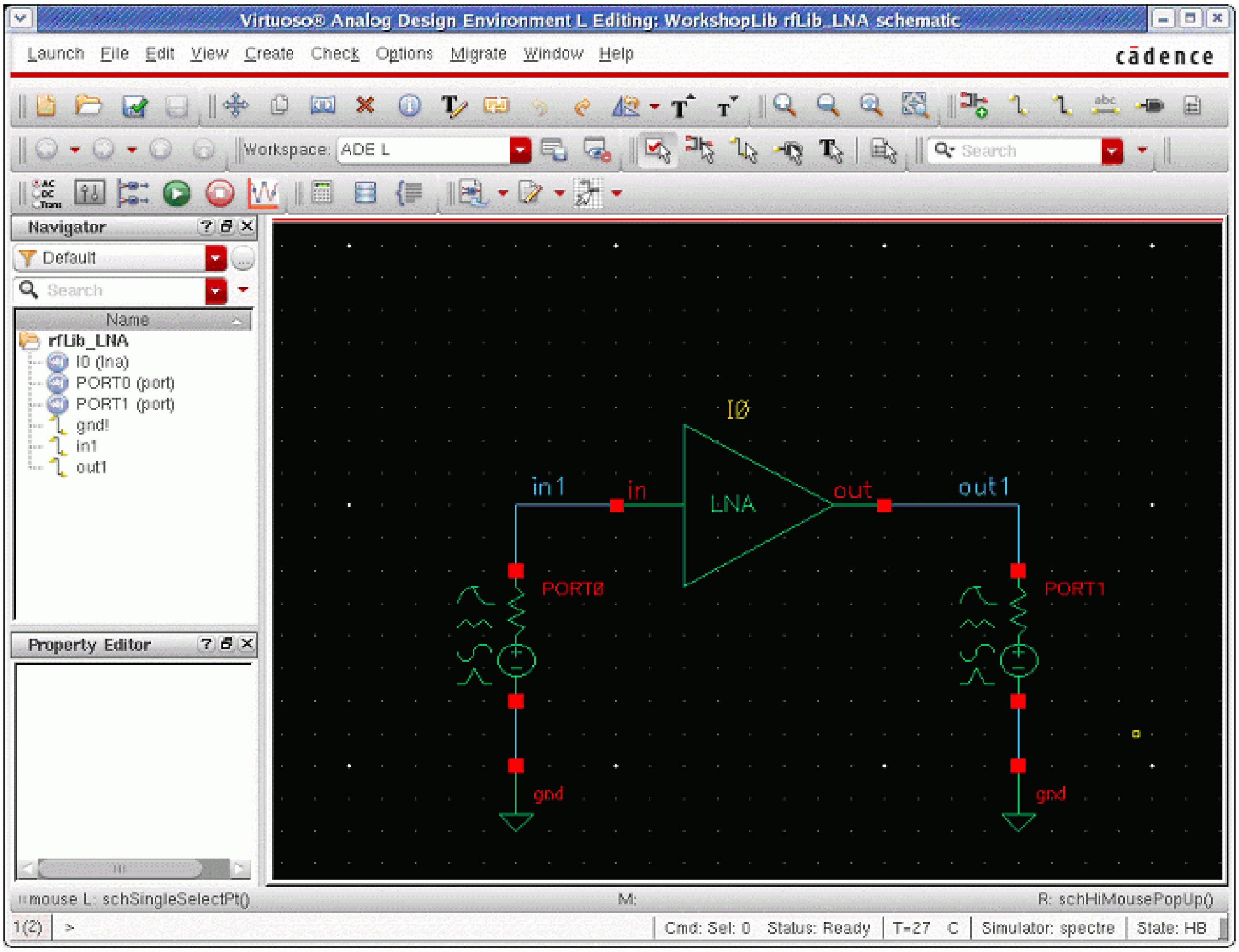
The next step is to select the net in the schematic. The waveform window appears with the selected result in it. Both the input and output nets are shown. A marker has been positioned at the first harmonic in both traces. Note that the input level is almost exactly -30 as set in the ADE, and the output is 20dB larger than the input as set in the behavioral LNA. The harmonics at about -350dBm represent the numerical noise floor of the simulation.



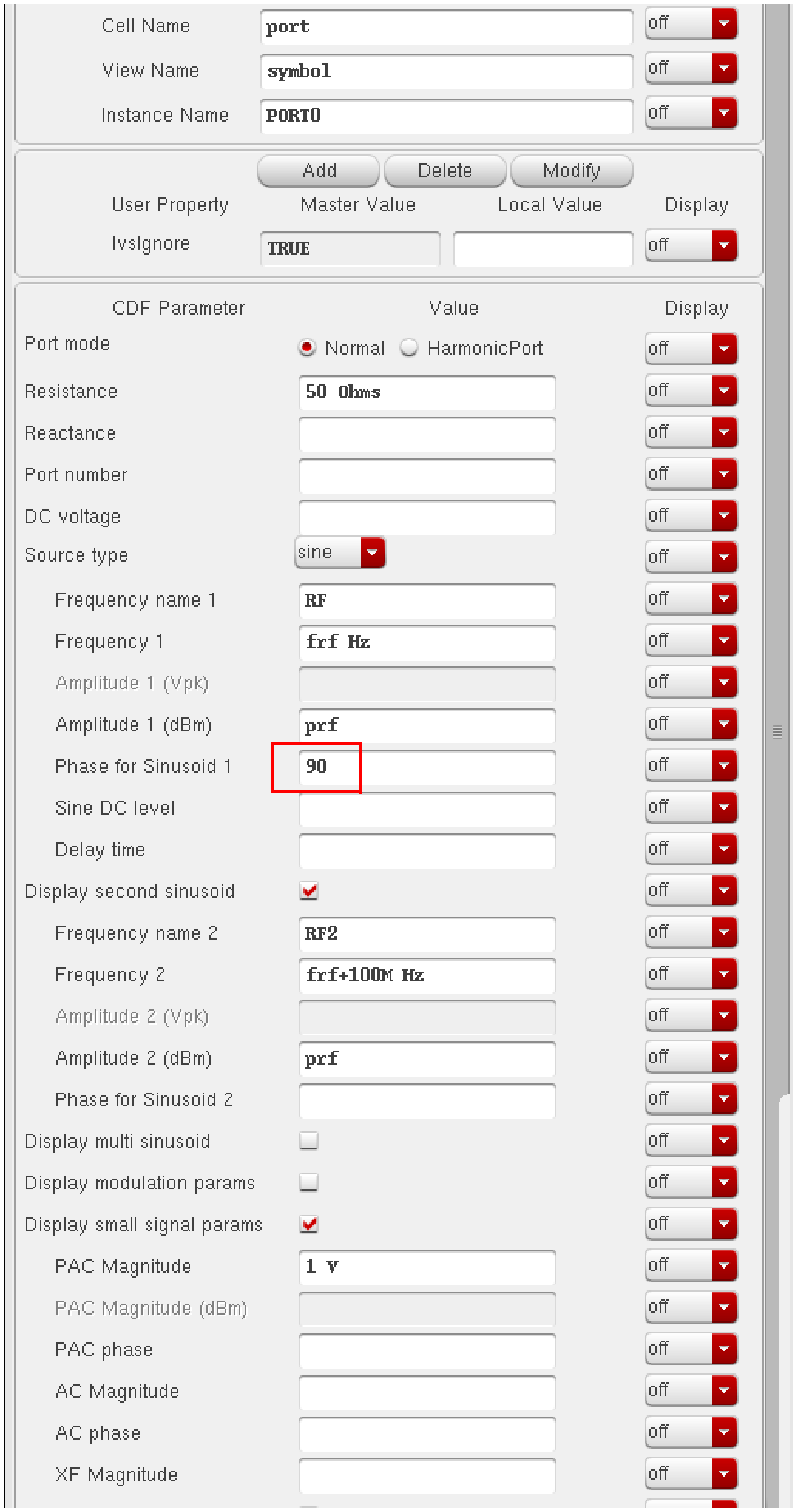
Harmonic Balance Solves For Cosines

In order to be compatible with other harmonic balance simulators, the convention in Spectre harmonic balance is to solve for cosines rather than sines. From a practical point of view, this means that the phase that is solved in harmonic balance is 90 degrees different compared to shooting. It also means that in order to get a zero degree solution in harmonic balance, the phase of the input source needs to be 90 degrees.

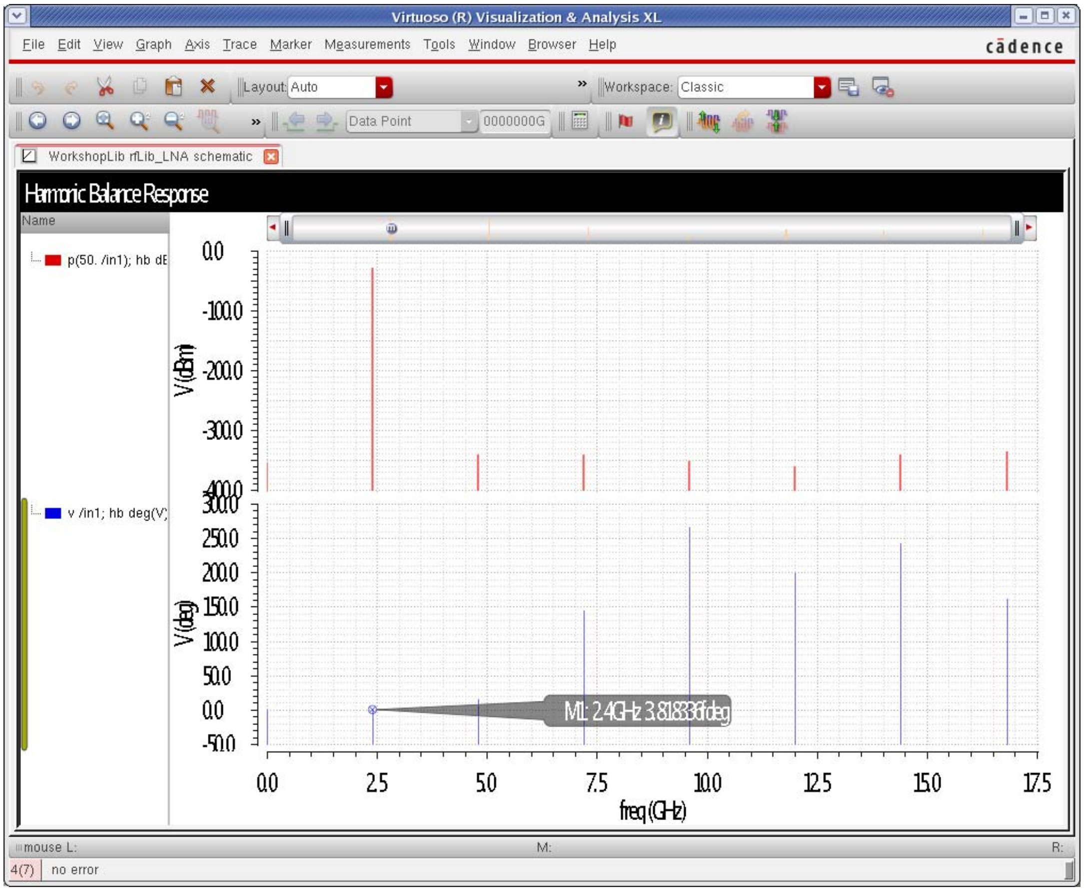
The circuit below is used for this example.



In the properties list for the input port, the phase has been set to 90 degrees. This is the phase of the sinusoid that is produced by the input source.


Cell Name
port
View Name
symbol
Instance Name
PORT0
Add
Delete
Modify
User Property
Master Value
Local Value
Display
lvsIgnore
TRUE
CDF Parameter
Value
Display
Port mode
Normal
HarmonicPort
Resistance
50 Ohms
Reactance
Port number
DC voltage
Source type
sine
Frequency name 1
RF
Frequency 1
frf Hz
Amplitude 1 (Vpk)
Amplitude 1 (dBm)
prf
Phase for Sinusoid 1
90
Sine DC level
Delay time
Display second sinusoid
Frequency name 2
RF2
Frequency 2
frf+100M Hz
Amplitude 2 (Vpk)
Amplitude 2 (dBm)
prf
Phase for Sinusoid 2
Display multi sinusoid
Display modulation params
Display small signal params
PAC Magnitude
1 V
PAC Magnitude (dBm)
PAC phase
AC Magnitude
AC phase
XF Magnitude
off

The signal *in1*, which is the input to the behavioral amplifier has been plotted below. The power in dBm is unaffected by the phase, but you can see that the phase that is calculated by Spectre is zero degrees. In harmonic balance, if you want a reference phase of zero, you must set the phase on the input source to 90 degrees.

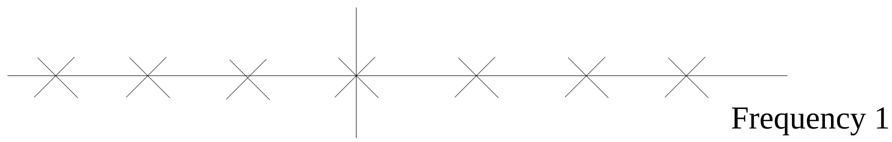


Setting Harmonics Automatically

Harmonic balance can now set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the Run Transient selection in the Transient-Aided Options. This single action will cause a transient analysis to be run until steady-state is detected, and then from the transient analysis, the number of harmonics for Tone1 (when Frequencies is selected) or for the tone that has tstab enabled (when Names is selected).

Setting Harmonics Manually

Consider an amplifier with a single input. In this case the input and its harmonics need to be calculated. In order to get the correct solution, both positive and negative frequencies need to be calculated. With harmonic balance, input is required as to how many harmonics should be calculated. If you set 3 harmonics, then -3. -2, -1, 0, 1, 2, and 3 times the input frequency needs to be calculated, as shown below.



The number of harmonics to be calculated is (2*number of harmonics) +1. The runtime and memory consumption are proportional to the number of unknowns (all the harmonics at all the nodes) that need to be solved. In this case, seven unknowns need to be solved at each node.

In order to make sure that you have enough harmonics, start with an estimate based on the power level and the harmonic content of the input to the circuit. If the input is sinusoidal at fairly low power, try about 3 harmonics. If the amplitude is large, try about 7 harmonics. Next, run the simulation and plot the results. Position a marker at the desired measurement. Now increase the number of harmonics, and re-run the simulation and re-plot the waveform. If the measurement did not change, then the original estimate was enough, and you might be able to reduce the number of harmonics. Use the smallest number of harmonics that are required in order to have a minimum runtime.

If the input is a pulse wave at high amplitude, start at about 15 harmonics and an oversample factor of 4. Oversample factor will be discussed later in the chapter. The simulator takes the fft of the input signal, and uses multiple sources in series with the values and phases set to the value calculated by the fft.

When square waves are present in the circuit, the minimum number of harmonics should be set to the period of the square wave divided by the risetime of the square wave. Very sharp edges require many harmonics in order to be accurate in the time domain. If the time-domain waveform is less important than the frequency domain content, then a smaller number of harmonics can be specified along with an oversample factor of 4 or 8. For more information, refer to the Oversample Factor section of this chapter.

When piecewise linear waveforms are used as power supplies to ramp the power up at time zero, specify only two points. The first point should have time set to zero, and the voltage should be about 80% to 90% of the actual supply voltage. The second point should have the time set equal to about half the period of the operating frequency, and the voltage set to the supply voltage. If the simulation time exceeds the second time in the PWL setup, it will retain that value for the rest of the simulation. More importantly, for any SpectreRF large-signal analysis, the system does not become periodic until after the last timepoint in the PWL file. If you put a point at 1 second and the power supply voltage, the system will need to be simulated for one full second in the tstab interval using the transient algorithm before the harmonic balance simulation can run. This can take a long time.

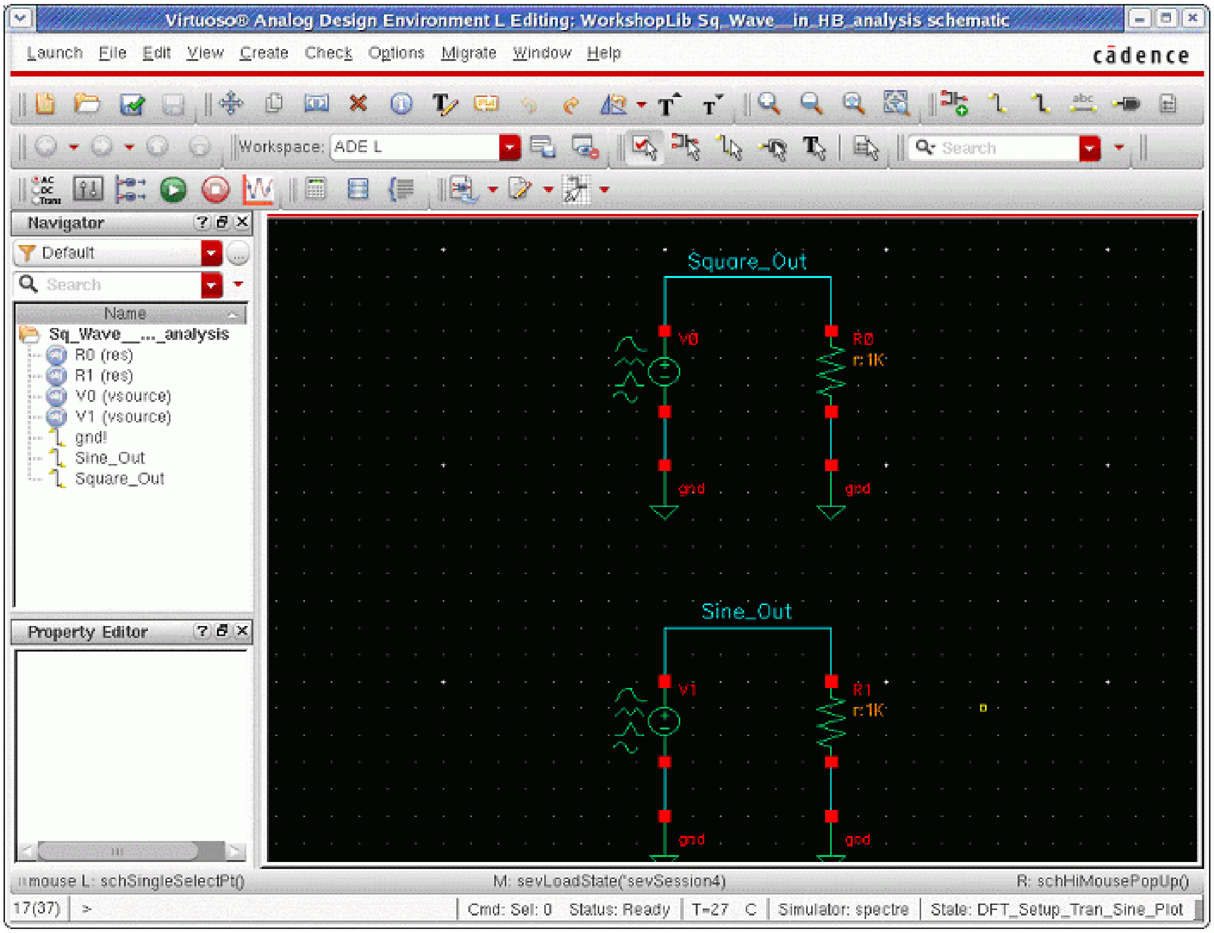
When periodic piecewise linear inputs are used, the more non-sinusoidal the waveform is, the more harmonics you need and also the higher oversample factor needs to be in order to accurately simulate the system. Oversample factor is explained in the following section. Start with an estimate of how many harmonics would be required to represent the waveform in the frequency domain, and then re-run the simulation with more harmonics to see if things changed. Oversample factor also needs to be increased. Sine waves need oversample=1. Square waves need 4 or 8. The more nonsinusoidal the waveform is, the higher the oversample factor needs to be. Note that even with a sinusoid applied, the currents may be very nonsinusoidal. Think of the currents in a diode mixer with a sinusoidal LO input. In this case, an oversample factor of 4 or 8 is likely to be required.

Up to two non-sinusoidal sources are allowed in the circuit. The first can be any periodic signal type, like pulse, exponential, and periodic piecewise linear. The second non-sinusoidal source is limited to being a pulse waveform. The rest of the inputs must be sinusoidal.

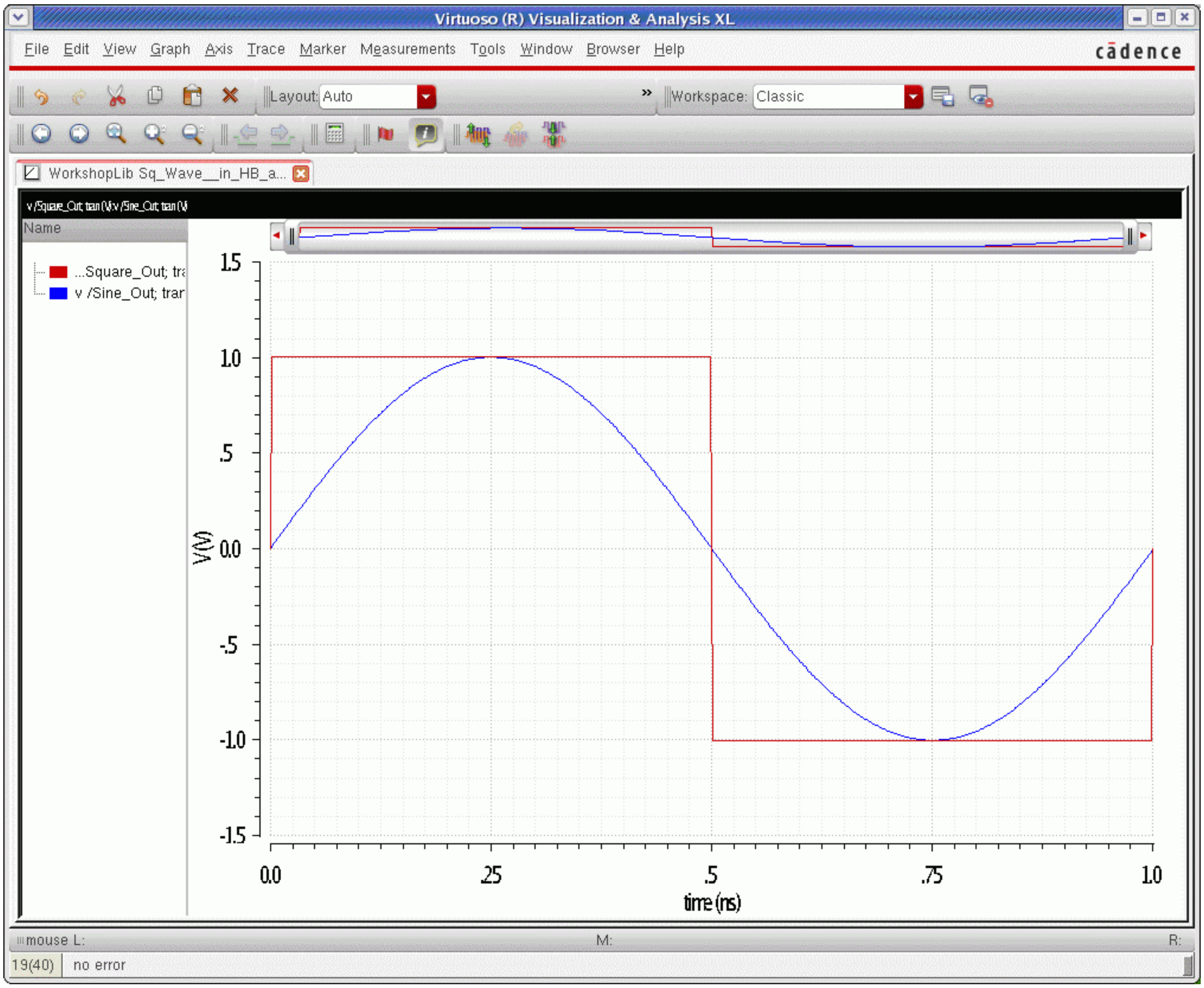
Oversample Factor

Note that fft and ifft are used to translate between time and frequency domains for the nonlinear evaluation. It is the nature of the ifft and fft to require more samples than the absolute minimum required in the time domain to get a correct answer for the harmonics in the frequency domain when the waveform is nonsinusoidal.

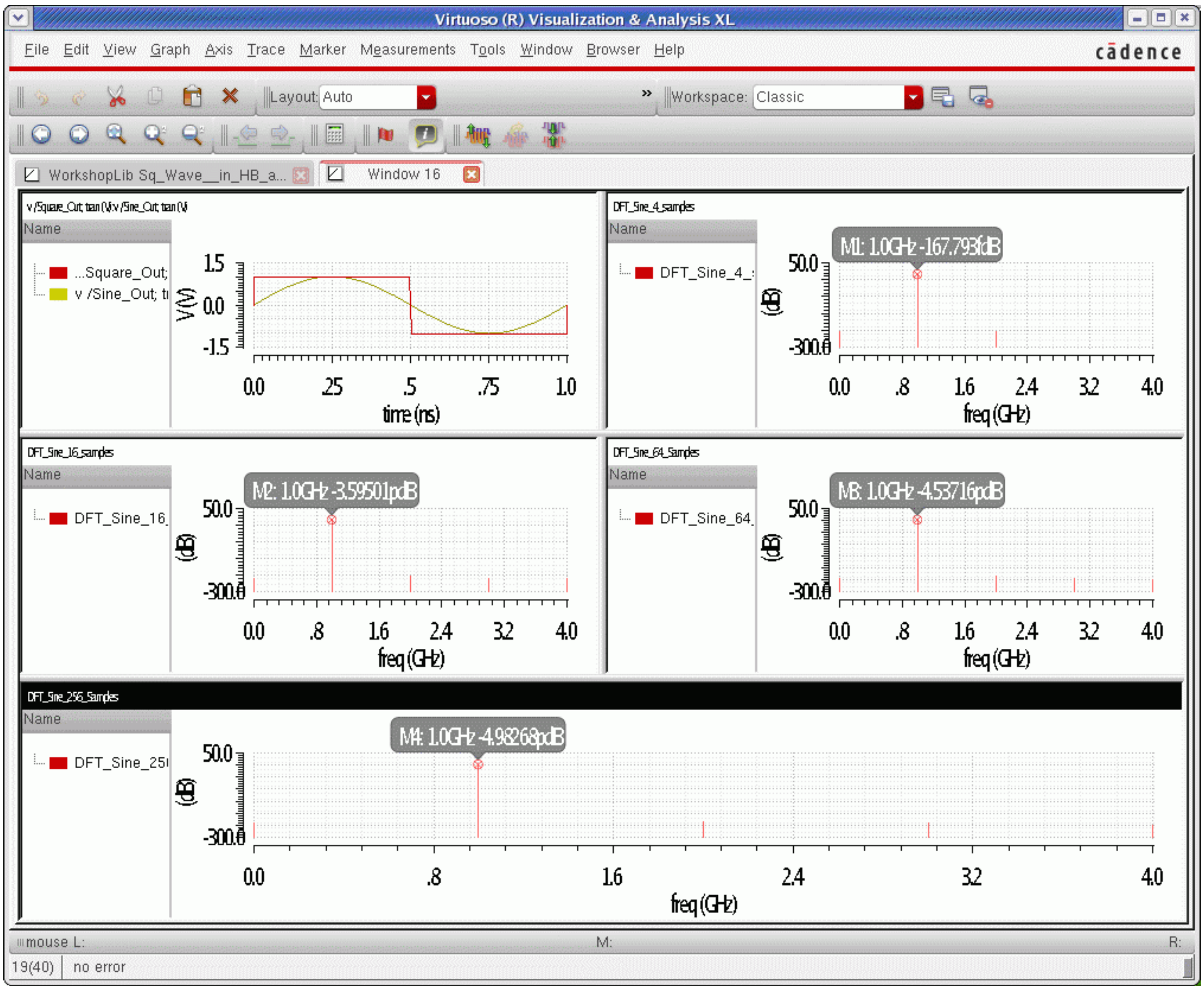
The circuit below has a square wave source on the top, and a sine wave source on the bottom. The load for both is a resistor.



There is no nonlinearity at all in the circuit. Therefore, the harmonics of the sinusoid and the square wave can be analyzed using the transient analysis and the DFT function in the calculator. The DFT function uses an fft algorithm to calculate the peak value of each harmonic. The transient waveforms are shown below. Both signals have exactly 1 volt peak amplitudes.

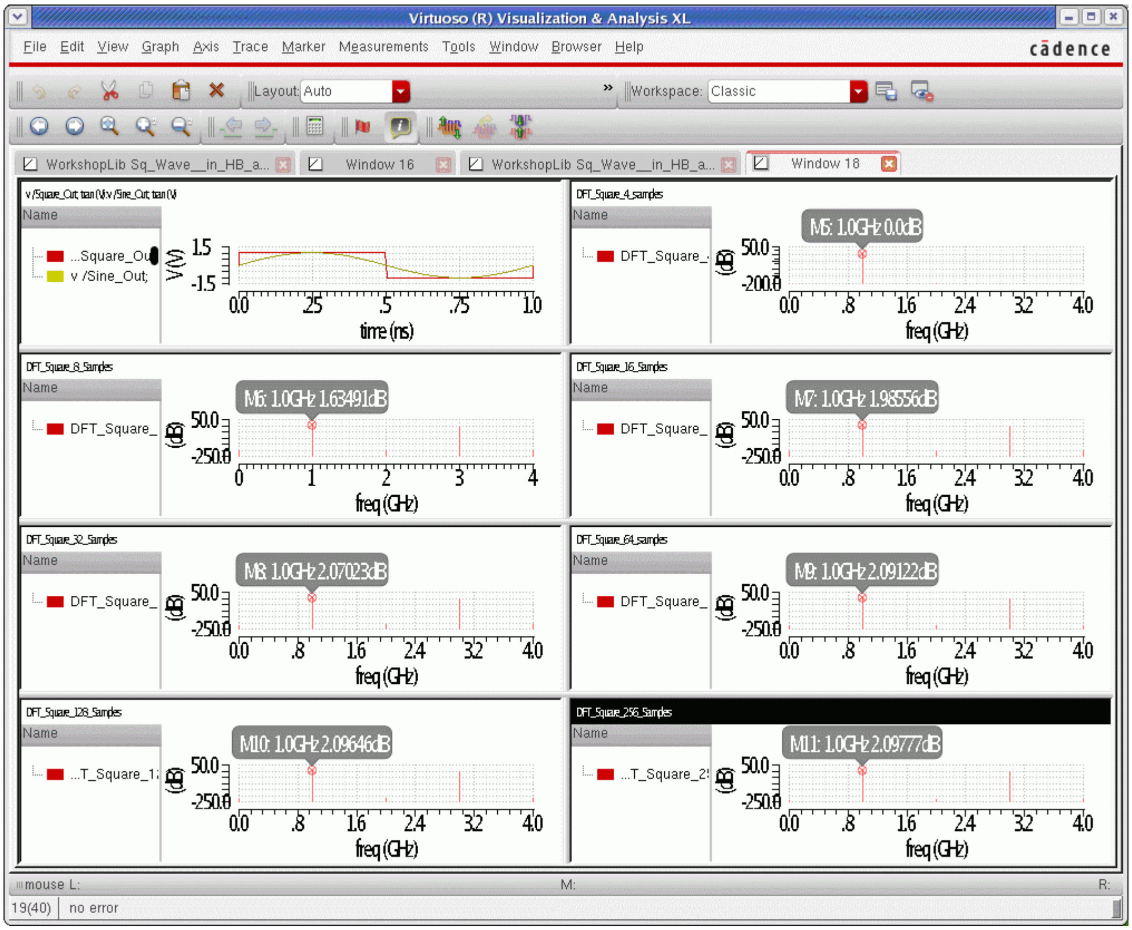


In the following figure, you can see that the fft of the sine wave is essentially the same for 4, 16, 64, and 256 samples of the transient waveform. The waveforms are shown in the top-left sub window. The number of samples can be read in the legend at the top-left corner of each sub window.



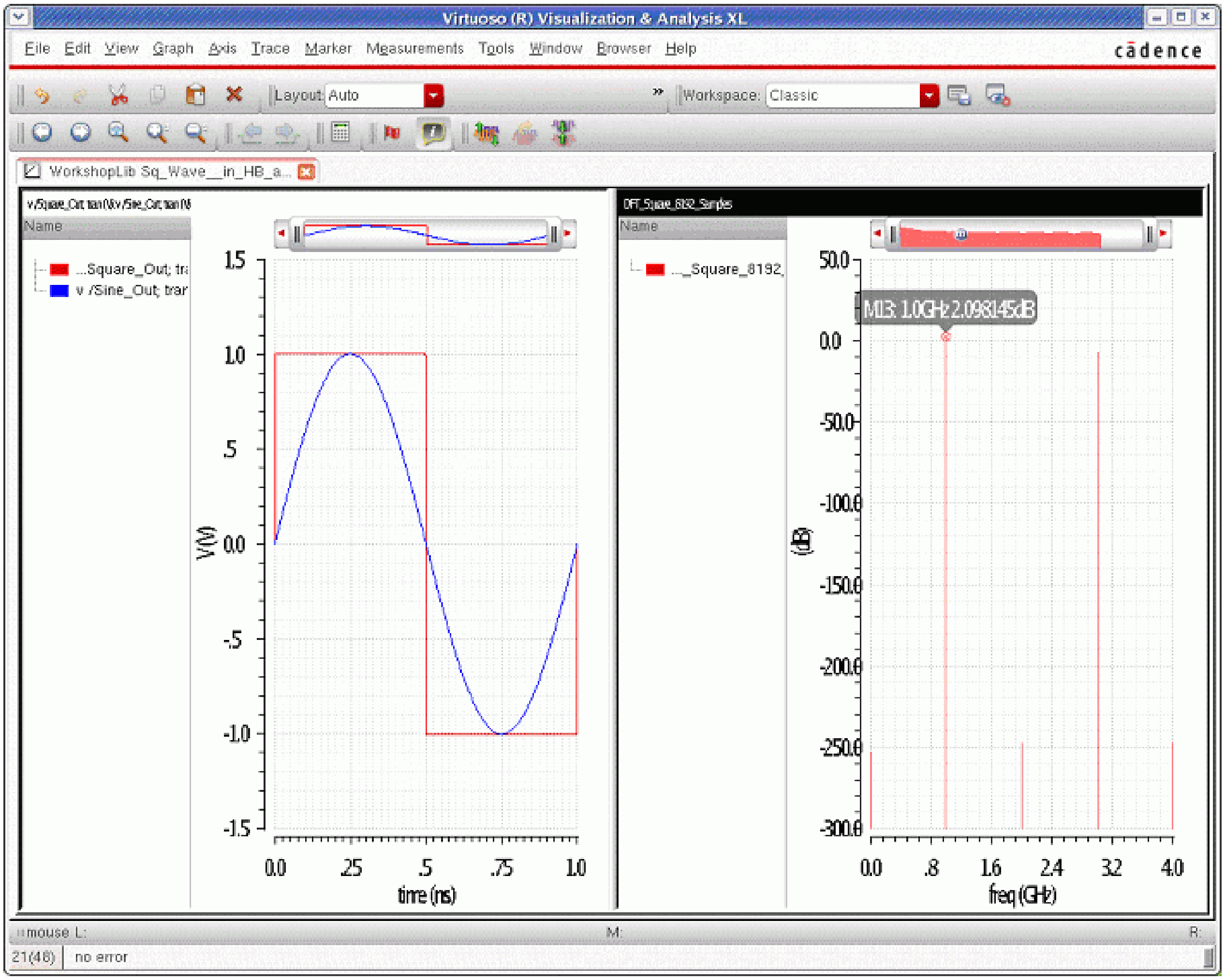
All the readings are in pico-dB or femto-dB. Sine waves do not need extra timepoints to calculate the correct amplitude for the harmonic.

Shown below are the ffts of the square wave with 4, 8, 16, 32, 64, 128, and 256 samples. The number of samples can be read at the upper-left corner of each sub window in the legend.



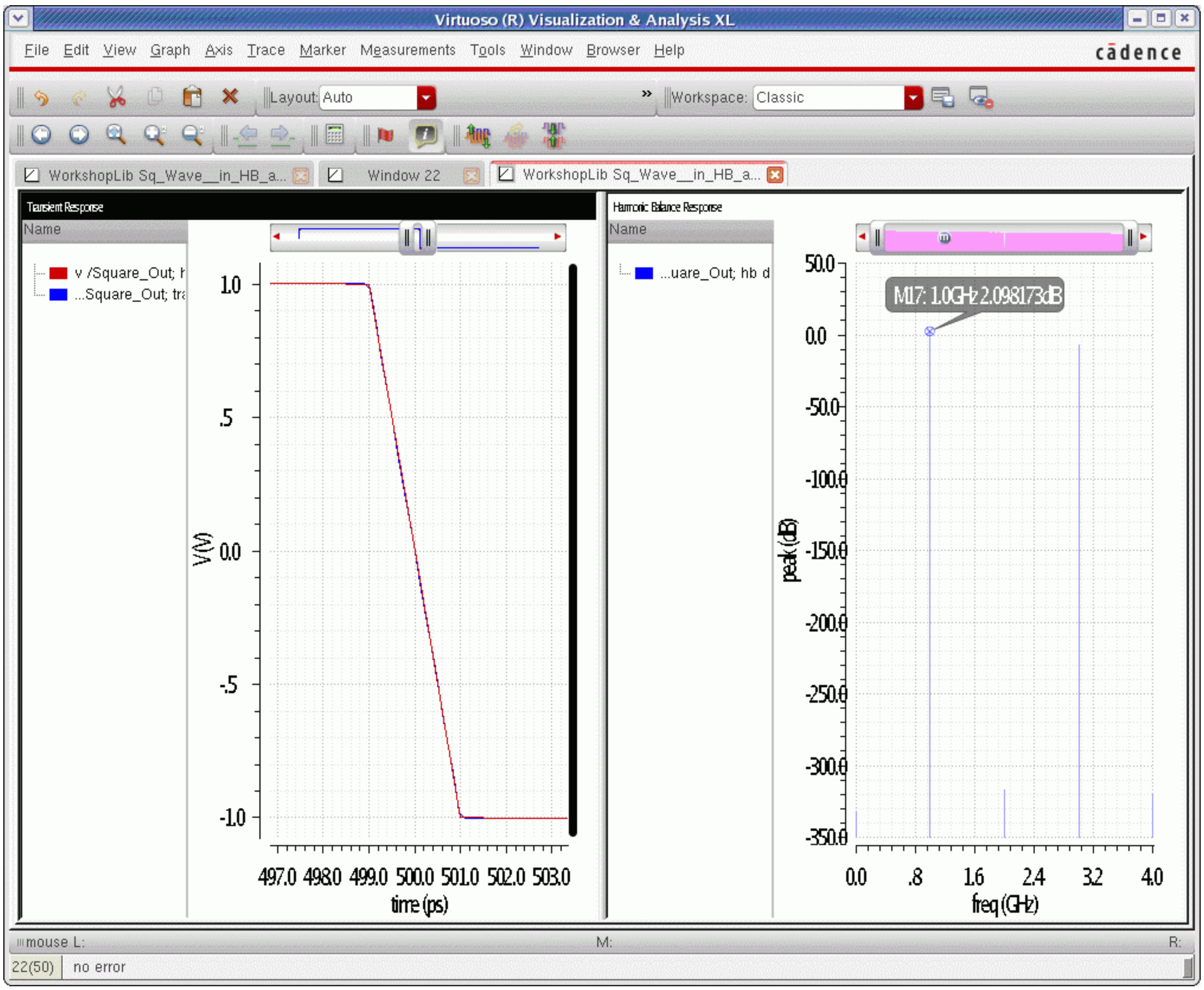
Note that with four samples, the value is 0dBV, which is 1 volt. With 256 samples, the result is 2.09777dB. This is almost 2.1dB larger than with four samples and is obtained just by increasing the number of samples. With 32 samples, you can get almost the right answer. This is a factor of eight times the number of samples required to calculate two harmonics. In other words, if two harmonics are set, an oversample factor of at least eight is required to get the correct answer for the square wave.

When 8192 points are used, only a minor change is observed as shown below.



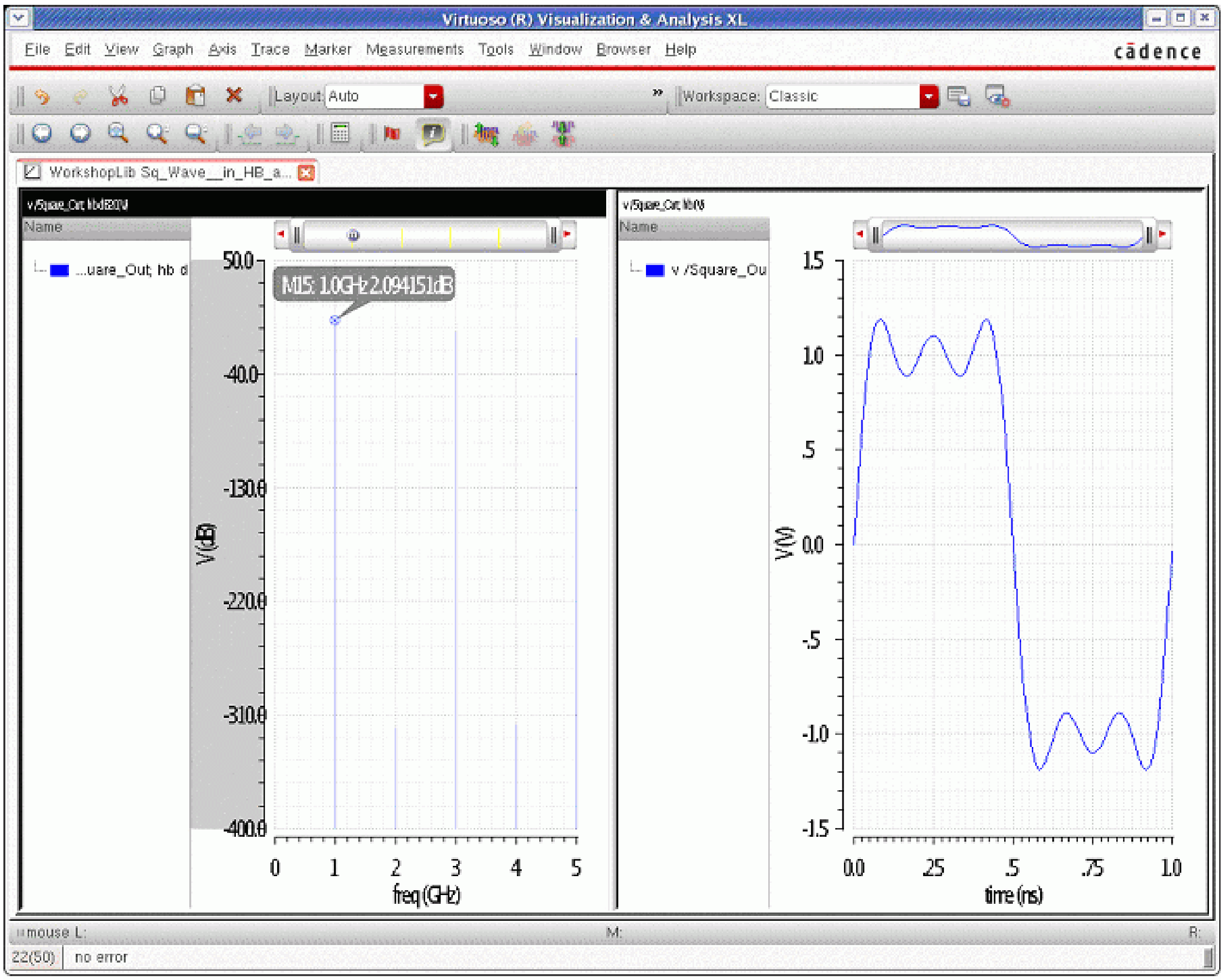
Trading off Harmonics and Oversample Factor

The risetime of the above square wave is 1/500th the period of the square wave. Technically, the period divided by the risetime or 500 harmonics are required to obtain the correct answer. The hb solution for the waveform (from an ifft) and in the frequency domain are shown below.



The large number of harmonics requires a long runtime when real circuits are simulated. Because of the large number of harmonics, shooting pss is likely to run faster than hb for this level of accuracy.

If accuracy in the time domain can be compromised a bit, setting a relatively small number of harmonics with an oversample factor of 8 can get reasonably close to the solution above. In the below example, 5 harmonics with an oversample factor of 8 have been used.



Note that the answer in the frequency domain is only a few thousandths of a dB different than the answer with 500 harmonics. The waveform in the time domain (from an ifft) is very different than the transient waveform. If you only need an answer in the frequency domain, setting a small number of harmonics along with a high oversample factor will result in much faster runtimes with little compromise in accuracy in the frequency domain when non-sinusoidal waveforms are present.

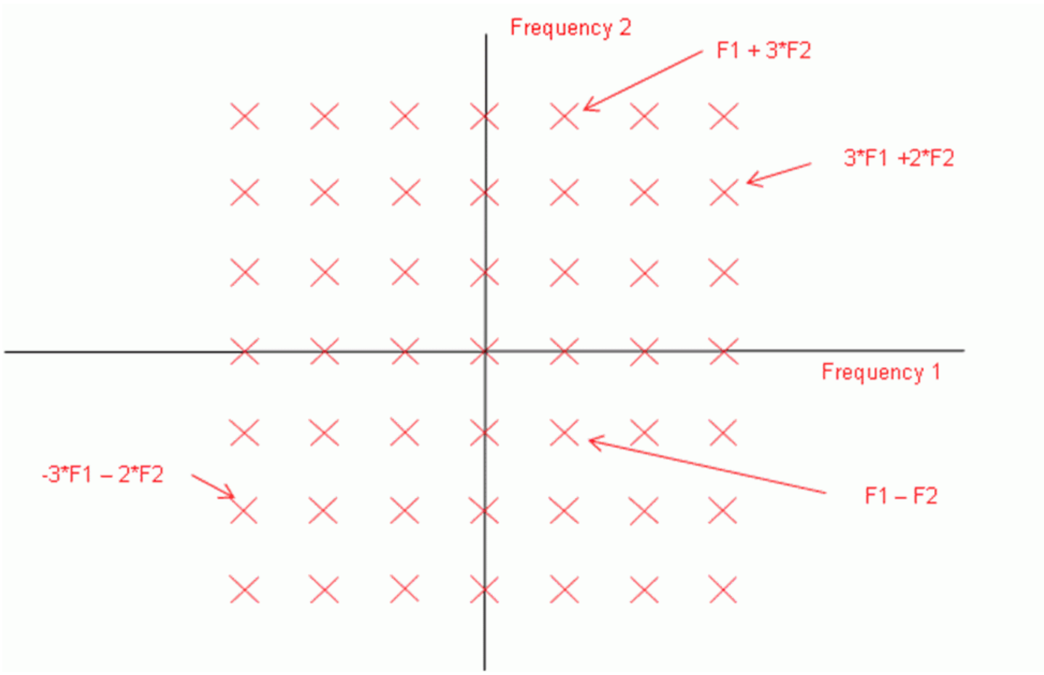
The basic strategy with non-sinusoidal waveforms is to start with an estimate of how many harmonics might be required, and set the oversample factor to 8. Run the simulation. Reduce the number of harmonics by about 50% and run the simulation again. If the answer does not change, reduce the number of harmonics again. Do the same with oversample. You are looking for the lowest runtime that also produces the correct answer, which will usually occur with a relatively large number for oversample factor, and a relatively small number of harmonics.

Every circuit is a little bit different in terms of the number of harmonics and oversample factor, but the principles are the same.

Two Input Frequencies

Consider an amplifier with two input frequencies. Because the circuit is nonlinear, intermodulation distortion is created. A lot more harmonics are produced by the circuit and need to be solved in the simulation. As a result, the runtime is longer and more memory is required. The harmonics to be calculated can be seen graphically, as shown below. Three harmonics are specified for both signals. The horizontal axis represents the first frequency and has symbols at -3*input frequency, -2*input frequency, -1*input frequency, the DC level, +1*input frequency, +2*input frequency, and +3*input frequency. Similarly, the vertical axis has symbols at the frequencies of the harmonics for the second input signal. The first symbol, 45 degrees from the origin, is the mixing product `Frequency1` + `Frequency2`. The figure below shows several different mixing products.

Note that the actual frequencies are not represented well by the chart. If F1 is 1GHz, and F2 is 1.1GHz, then F1+F2 is 2.1GHz and this is the first point at +45 degrees. F1-F2 is 100MHz, and is represented by the first point at -45 degrees.

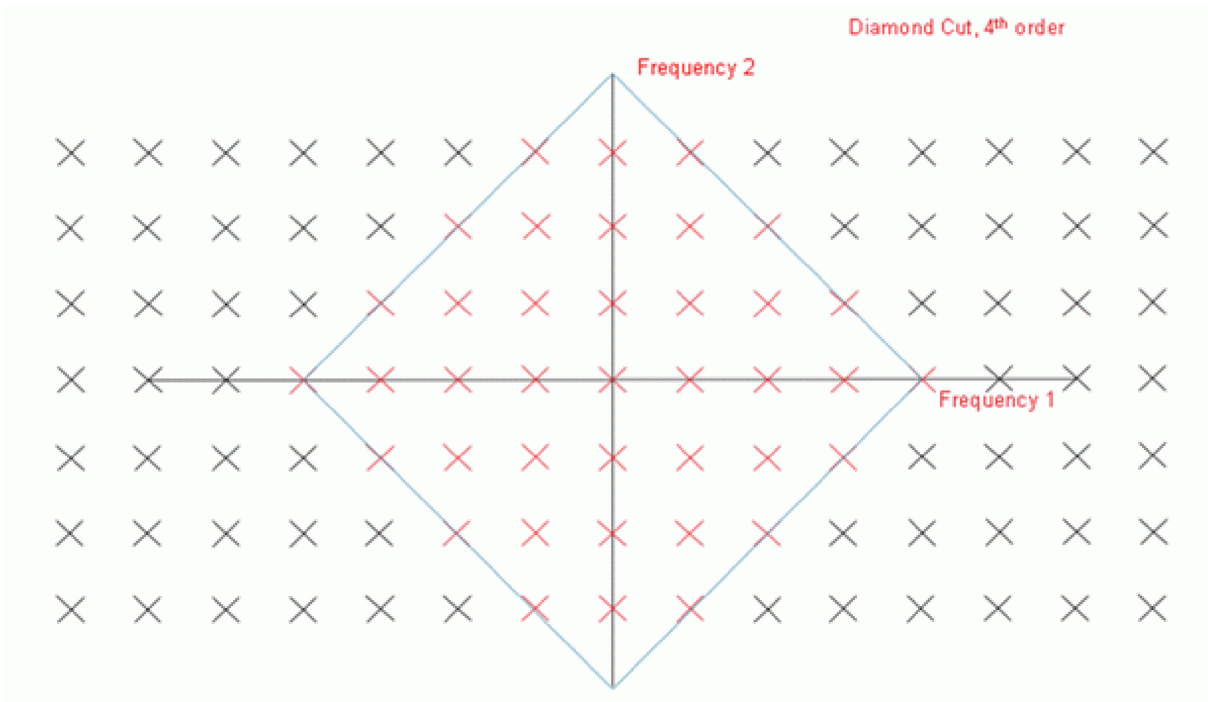


This is called a rectangular cut, which is the default setting in harmonic balance. Note that ((2*number of harmonics on tone 1) +1) * ((2*number of harmonics on tone 2) +1) harmonics need to be calculated. In this case 49 unknowns need to be solved at each node of the circuit.

Note that the diagonal corners are sixth order terms. For example, the upper-right corner is 3*F1 + 3*F2. Since it is a sixth order term, it is likely that the amount of power that the circuit generates is small, and therefore can be removed from the solution space, thereby speeding up the simulation with minimal loss in accuracy.

Diamond Cut

SpectreRF has a diamond cut available where the high order terms can be removed from the simulation. The solution space is shown below for a 4th order diamond cut. The solutions in red are in the solution space. The solutions in black are not. The figure below assumes that 7 harmonics are specified for `Frequency 1`, and 3 for `Frequency 2`.

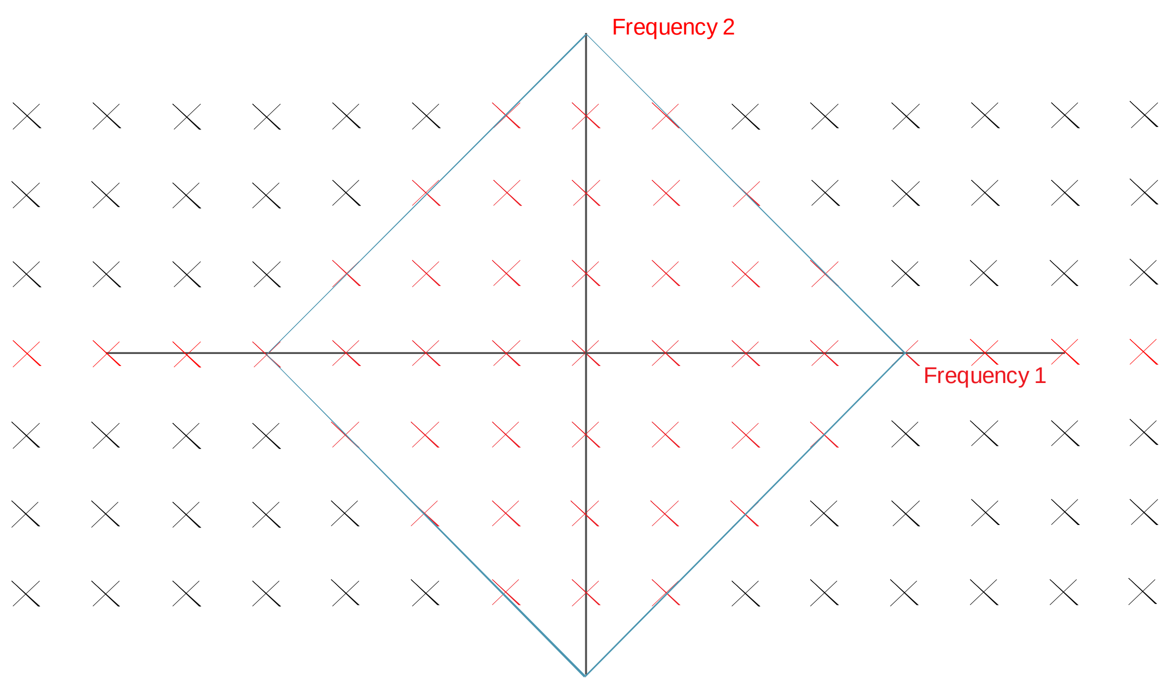


It is called a diamond cut because graphically, it looks like a diamond. The rectangular cut would have 15*7 or 105 harmonics in it. This cut has 39 harmonics in it. Instead of solving 105 unknowns at each node, 39 unknowns are solved at each node, therefore, the simulation runs considerably faster and takes less memory. All the fourth order and lower order terms are in the solution. The fifth order and higher order terms are excluded. As long as there is not much power in those harmonics, the solution will remain accurate.

Diamond cut is useful for circuits where the distortion is similar in the circuit from all the input frequencies. An example is an IP3 simulation for a power amplifier. Because the power level of both inputs is the same, both signals cause similar distortion.

The order of the cut can be set in the ADE *Choosing Analyses* form. Setting the order of the cut is similar to setting the number of harmonics. The more the distortion in the circuit, the larger the order needs to be for accuracy. Start with an estimate of what the circuit might produce, and run the simulation. Take the measurement. Raise the order, re-run, and re-plot. If the measurement changed significantly, then the order needs to be raised again. If the measurement did not change, try a smaller order. Use the smallest order that produces accurate results.

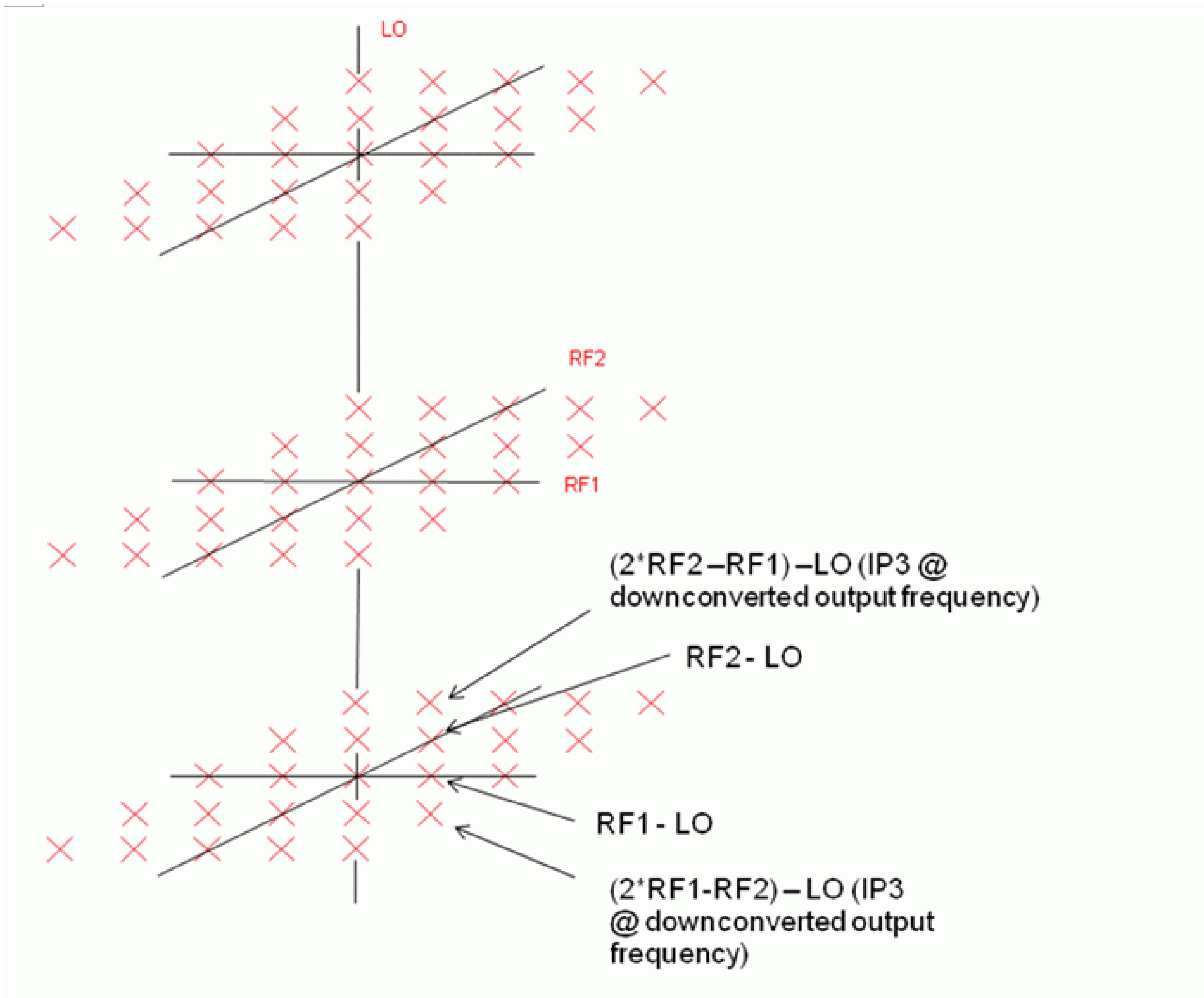
Funnel cut is also available. Funnel cut is a diamond cut with the addition of the harmonics on the axis for all the tones. This cut is useful for systems where there is a difference in the distortion from the different inputs. One example would be a mixer with a large amplitude LO, and an RF tone. In this case, because the LO power is large, all the harmonics of the LO are desired, but the mixing products are limited to lower order. This reduces aliasing with the addition of only a small number of harmonics. A diagram is shown below.



2 input frequencies. Funnel Cut. Red Xs are calculated.

Three Input Frequencies

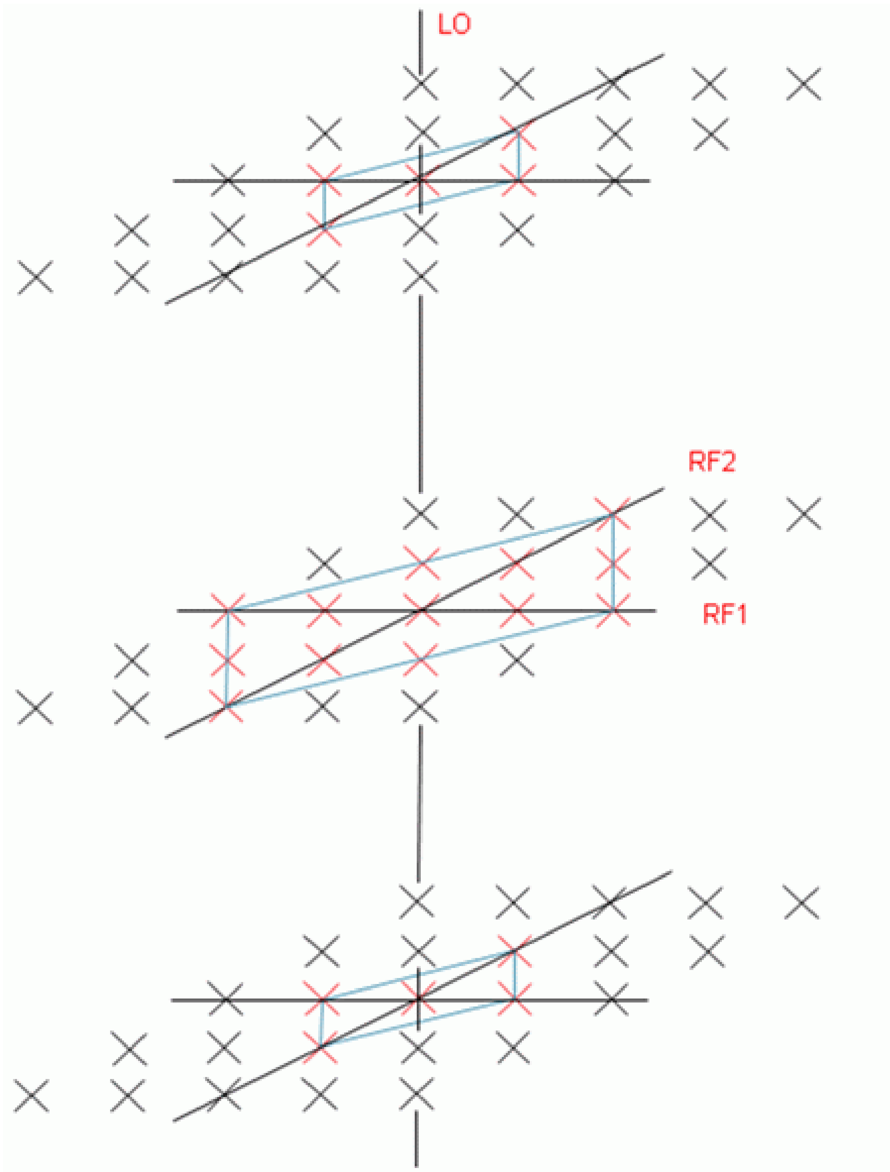
Now imagine a mixer with an LO and two RF signals. Here, there are multiple planes of harmonics that need to be calculated, as shown in the figure below. Only 2 harmonics of the RF tones and one harmonic of the LO are shown in the figure so the planes can be visualized.



Note that ((2*number of harmonics on tone 1) +1) * ((2*number of harmonics on tone 2) +1) * ((2*number of harmonics on tone 3) +1) harmonics need to be calculated. If three harmonics were specified for all 3 tones, 7*7*7 or 343 harmonics need to be calculated at each node. Four input frequencies is a practical maximum for most circuits.

Diamond Cut With Three Frequencies

When the diamond frequency cut is applied, the red solution shown below is calculated when the maximum order is set to two. On the zero plane for the LO, a diamond that goes to the second harmonic of both tones is included. In the -1 and +1 planes for the LO, the diamond is drawn with the order reduced by one. When the number of LO harmonics is larger, which is the general case, on the 0 (zero) LO plane, the diamond goes through the order specified in the `maximorder` parameter. For the +1 and -1 planes, the order is reduced by one. For the +2 and -2 planes, it is reduced by one more. This continues on the rest of the planes.

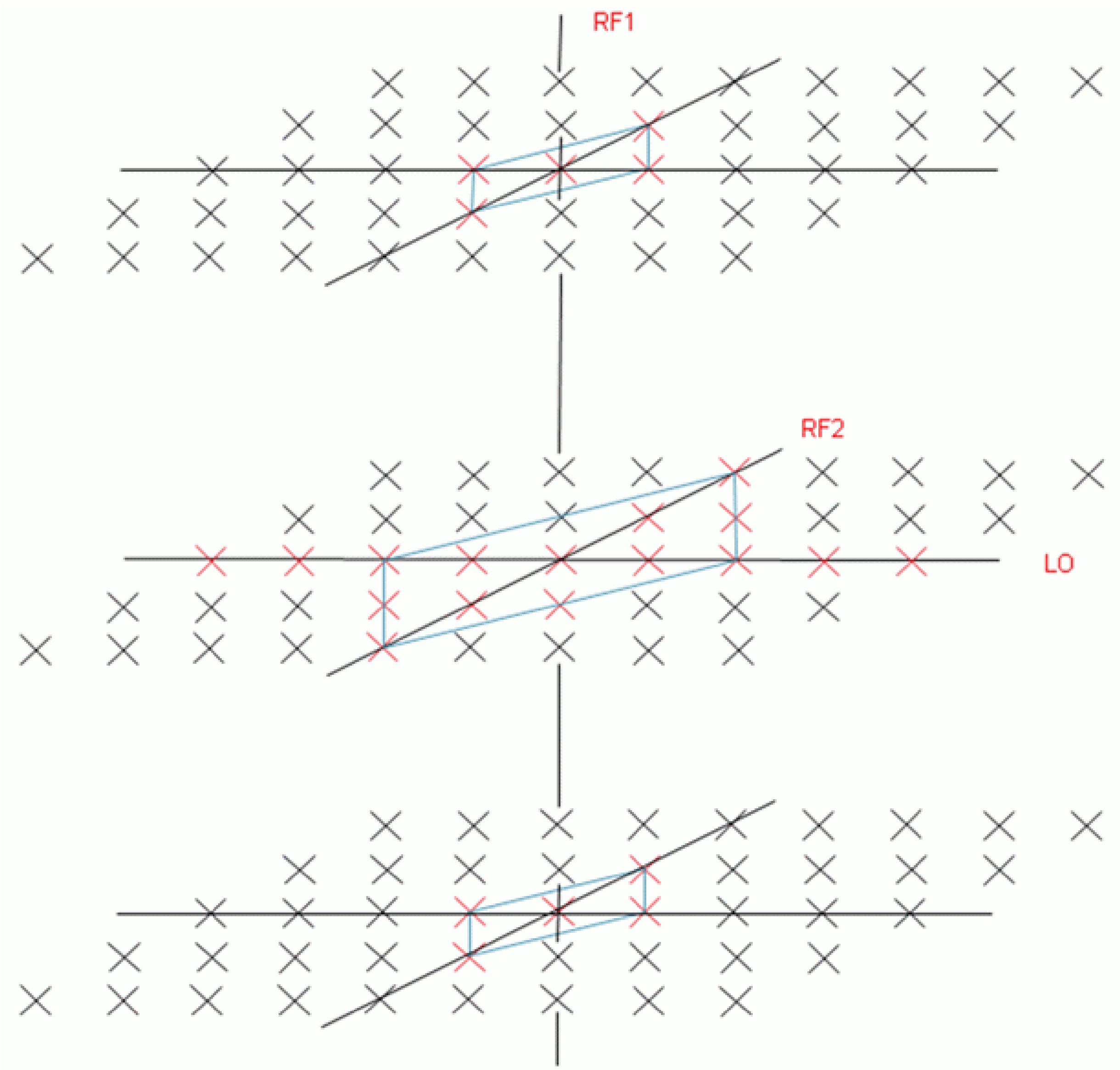


Now imagine that there were more LO harmonics. More planes would be created. If the maximum order was set to 4, on the center plane, the included elements would go through the fourth order. For the first plane above and below, harmonics through the third order would be calculated. For the second plane above and below, harmonics through the second order would be included. For the third plane above and below, only the first harmonic would be calculated. For any other planes, no harmonics would be calculated.

You can see from the figure that fewer harmonics are calculated when the cut is applied and that the diamond is extended vertically. Fewer harmonics are calculated in the upper and lower planes than in the center plane. In this way, many harmonics are pruned from the solution space, and the simulation is sped up in the process. Two harmonics for the LO is artificially small in order to understand the concept. Usually, between 3 and 15 harmonics are chosen in most circuits.

Funnel Cut

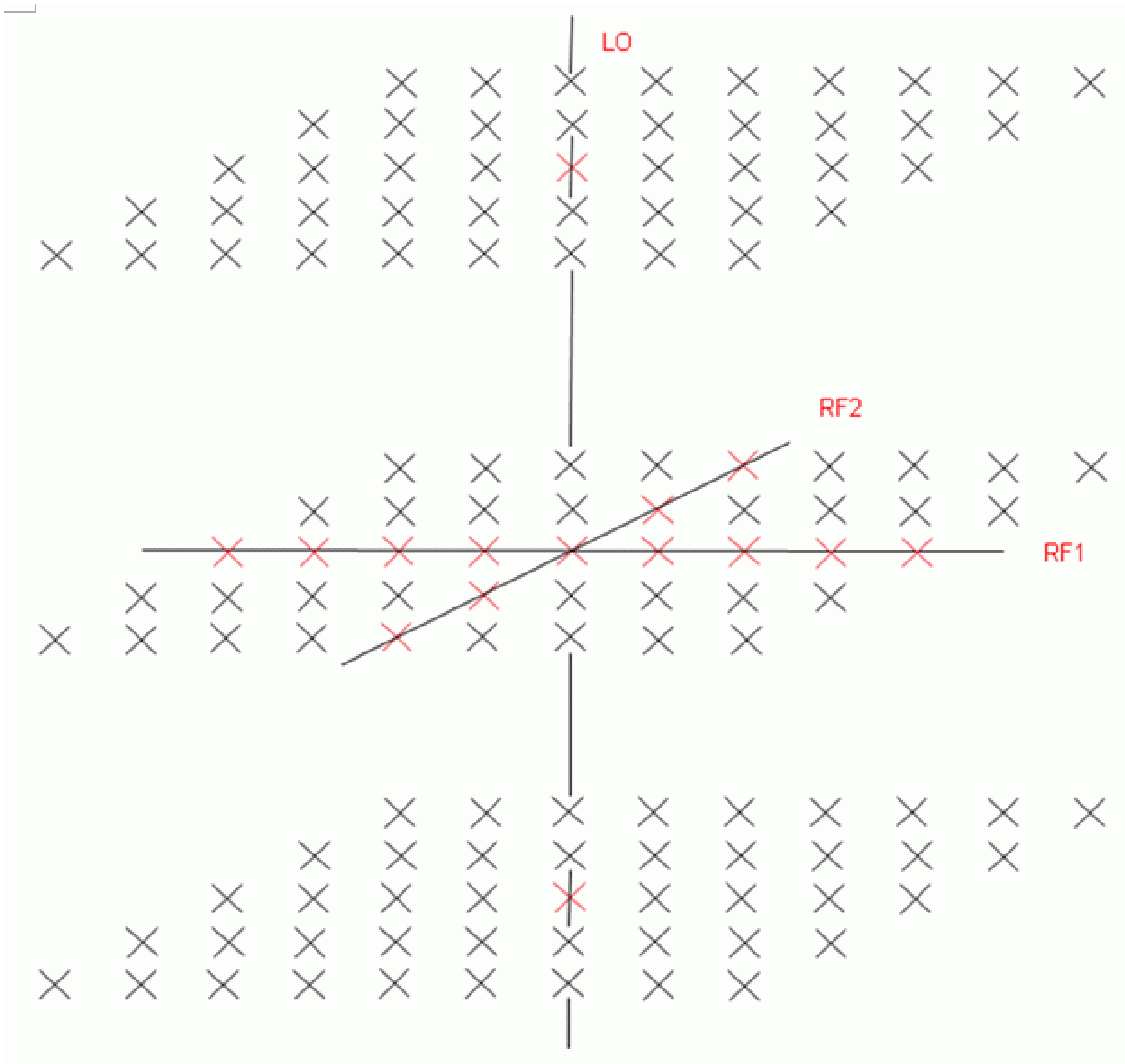
Note that in the following example, because of the high LO power, many LO harmonics would be created in the circuit, and therefore, might cause incorrect results if the higher harmonics of the LO were not included in the solution. Funnel cut takes the diamond cut, and adds harmonics along the axes for all the inputs up to the maximum specified by the number of harmonics for each tone. This is shown in the figure below. Note that the signal names on the axes are changed from the previous figure. This is done to allow the diagram to fit on the page.



Note that the harmonics along the axes only are added. In this case, the LO harmonics are added. The funnel cut is provided for circuits like mixers where the distortion is different from the different inputs. The idea is that the maximum order of the diamond might be 4 or 5 which allows accurate calculation of the mixing products and still calculating enough harmonics of the signal(s) that causes relatively more distortion. If the diamond cut were used, a higher maximum order is necessary to allow the high order harmonics of the high distortion signal to be calculated which includes many more harmonics to be solved than the funnel cut.

Axis Cut

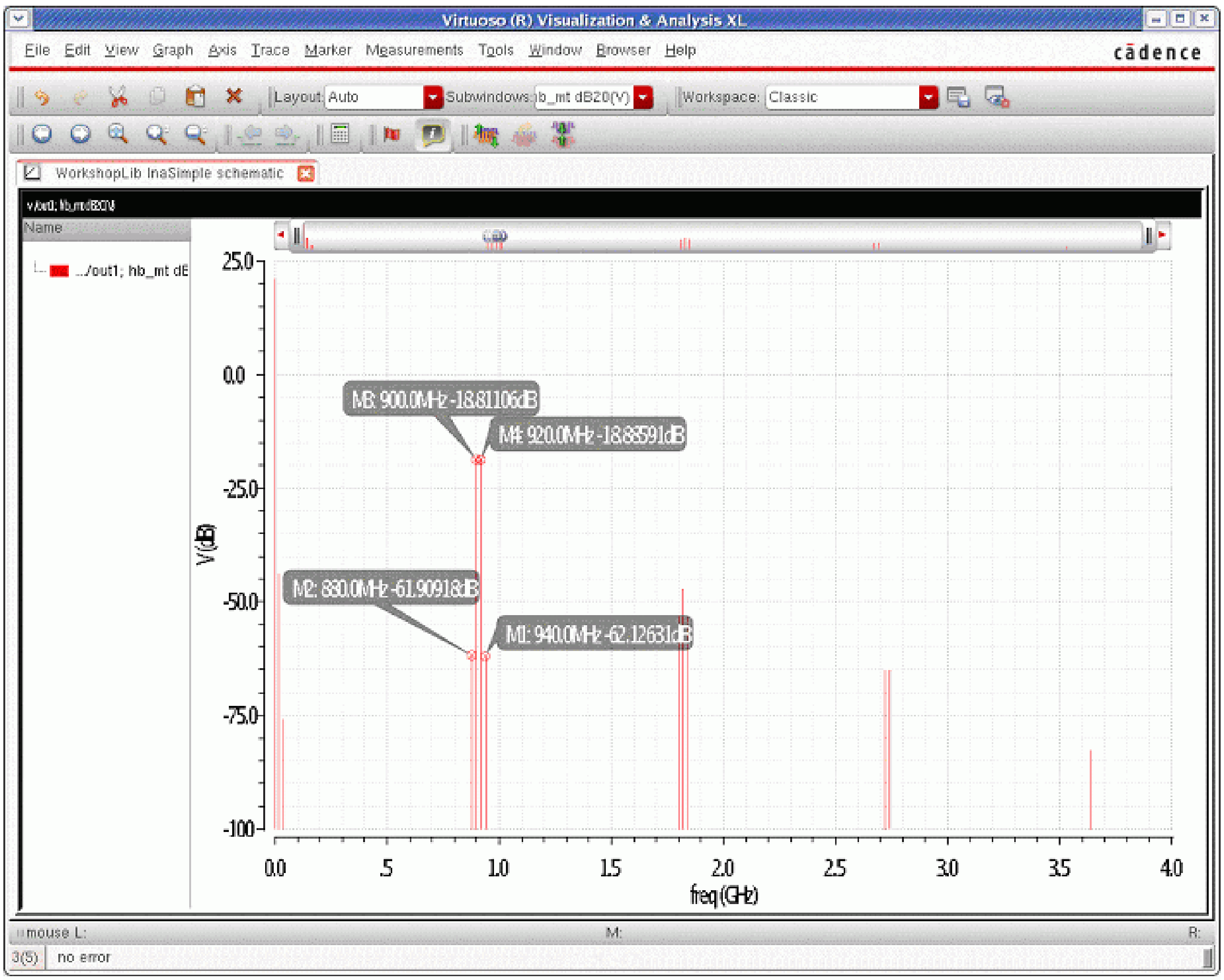
A fourth cut is available called Axis cut that only calculates the harmonics of the inputs themselves. No mixing products are calculated at all. This cut is only useful for getting a quick idea of the amplitudes of the individual tones. It is shown in the figure below.



Aliasing

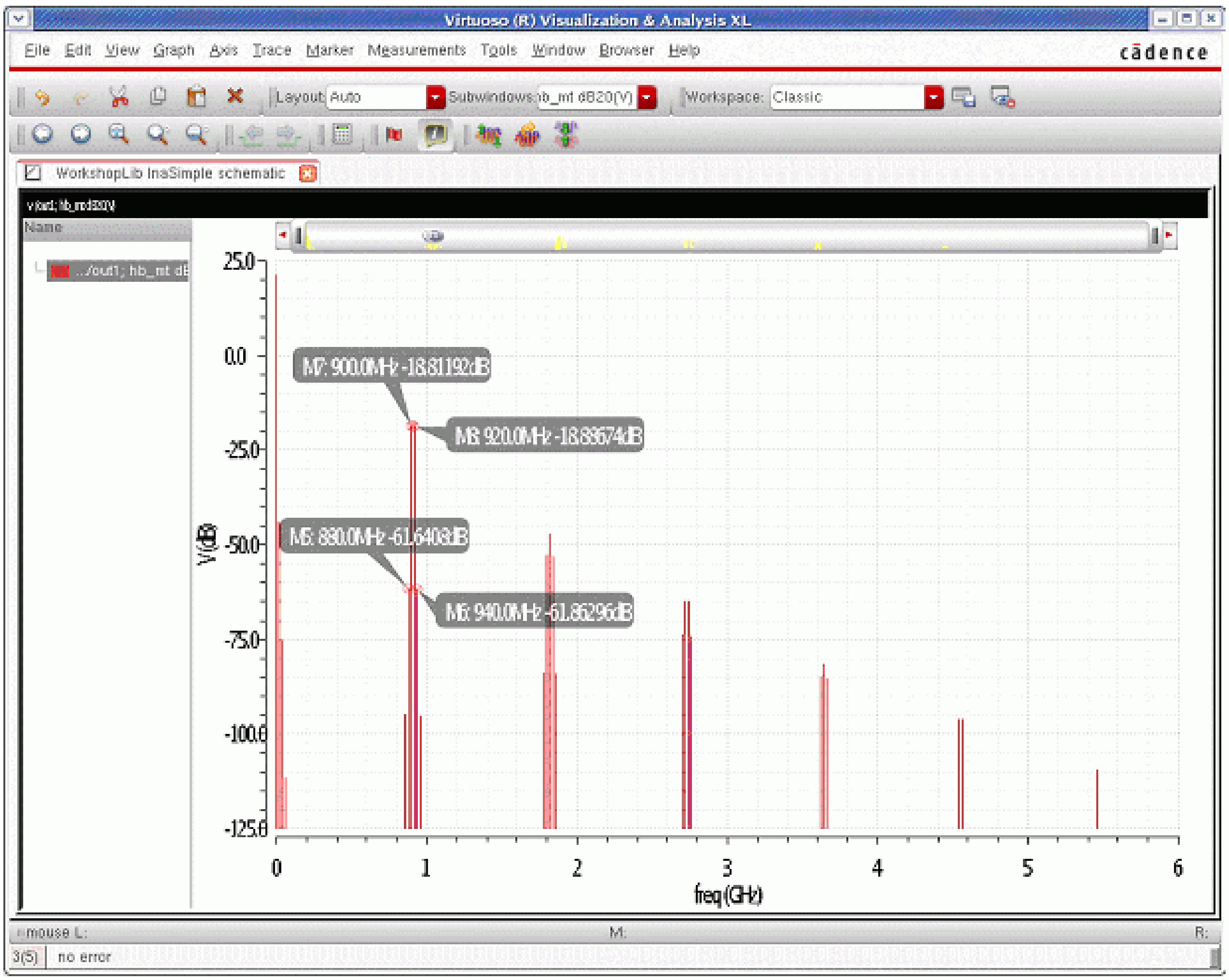
The reason for setting the number of harmonics and the order of different cuts is because of aliasing. If the actual system produces power outside the solution in the higher harmonics, that power will appear in the solution you specified. In the example above with the axis cut, because no mixing products are calculated at all, all that power shows up as increased power somewhere in the harmonics that are actually calculated. It is not predictable where that power may be aliased to.

Aliasing needs to be checked, as can be seen in the following figures.



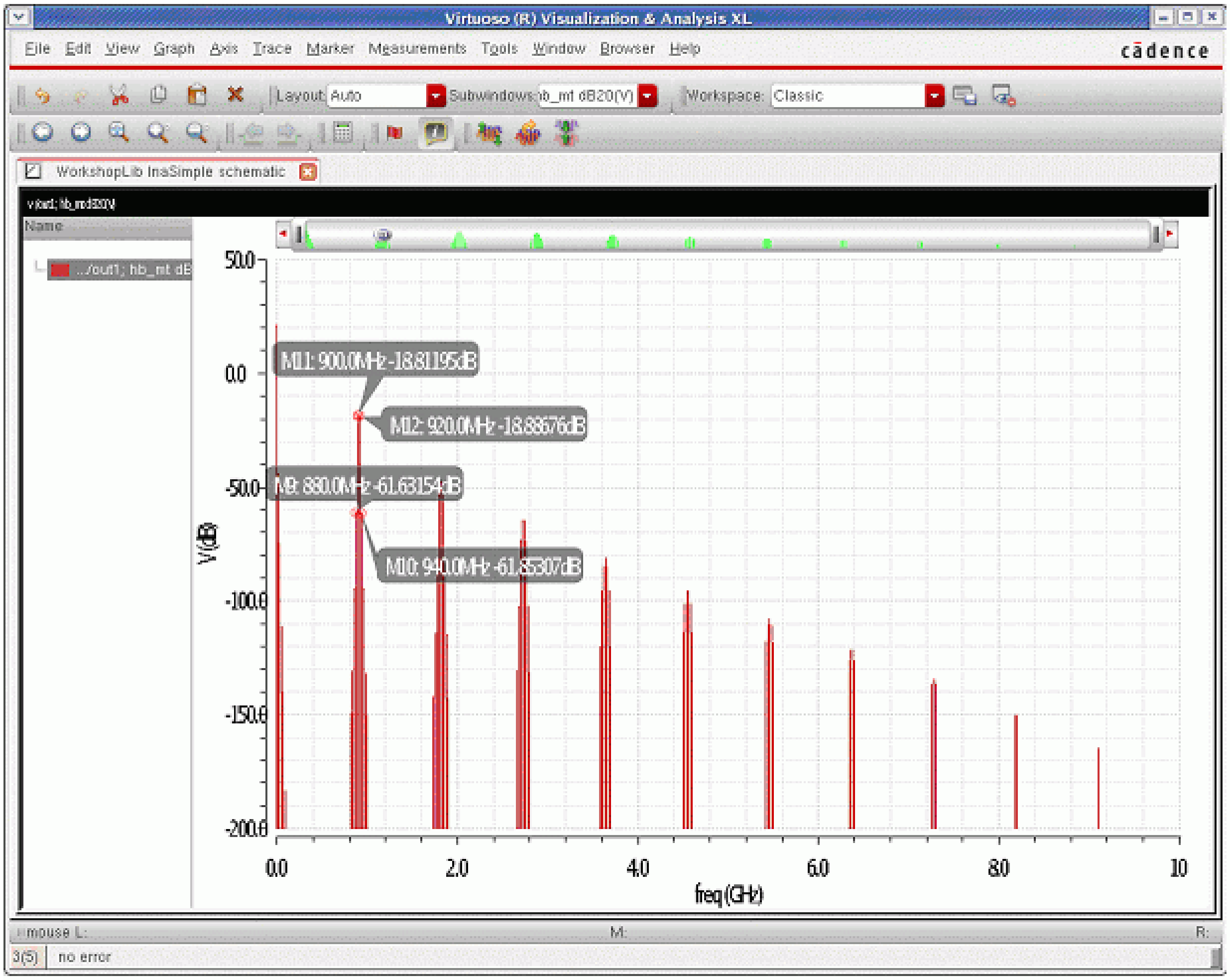
The figure above displays the spectral calculation of an amplifier with 900MHz and 920MHz applied at the same level. Note the level of the harmonics at 880MHz, 900MHz, 920MHz, and 940MHz. In this solution, two harmonics were set for both tones.

When the number of harmonics in the solution is raised to three for each input, the aliasing is less because there is less power in the uncalculated harmonics of the system. Note that in this case, the measurements for the harmonics at 900MHz and 920MHz changed only by 0.01dB. If you only need a measurement of the output power at the main output frequencies, 2 harmonics are enough. The measurements at 880MHz and 940MHz have changed by 0.268dB and 0.263dB respectively. This might or might not be accurate enough for an IP3 measurement depending on your accuracy requirement. The solution is shown below.



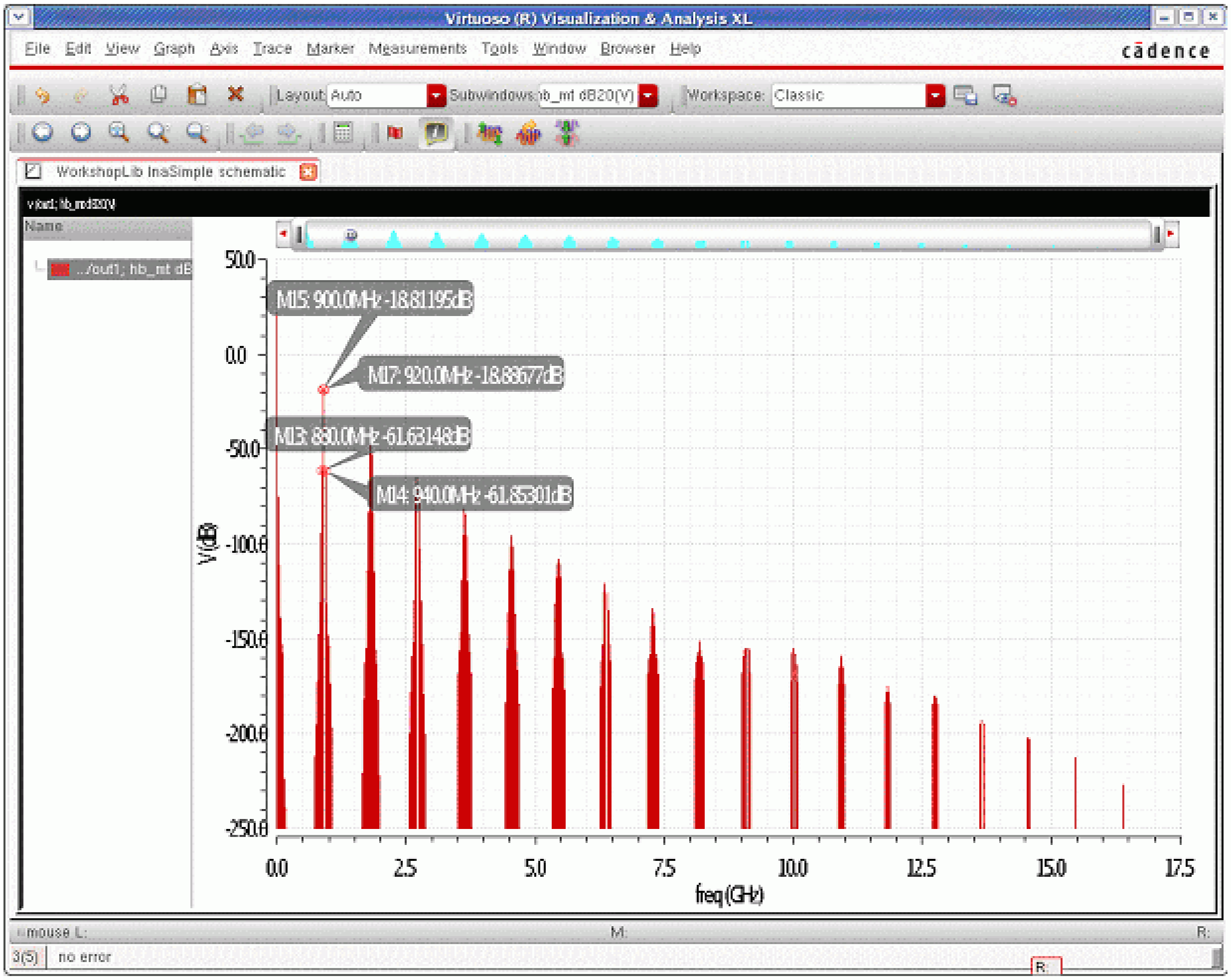
In order to check if the solution has enough harmonics, even more harmonics need to be run. Below is the solution with five harmonics on each tone. Note that the levels have changed by 0.01 dB for both tones at 880MHz and 940MHz. Three harmonics are enough in this case.

Although the solution changed slightly with more harmonics, it is difficult to justify the extra runtime for a 0.01 dB change.



When the harmonics are raised to nine, the solution changes by less than 0.001dB as compared to five. To repeat, three harmonics is enough in this case. There was no significant change when the number of harmonics was raised above three.

The solution with nine harmonics is shown below.

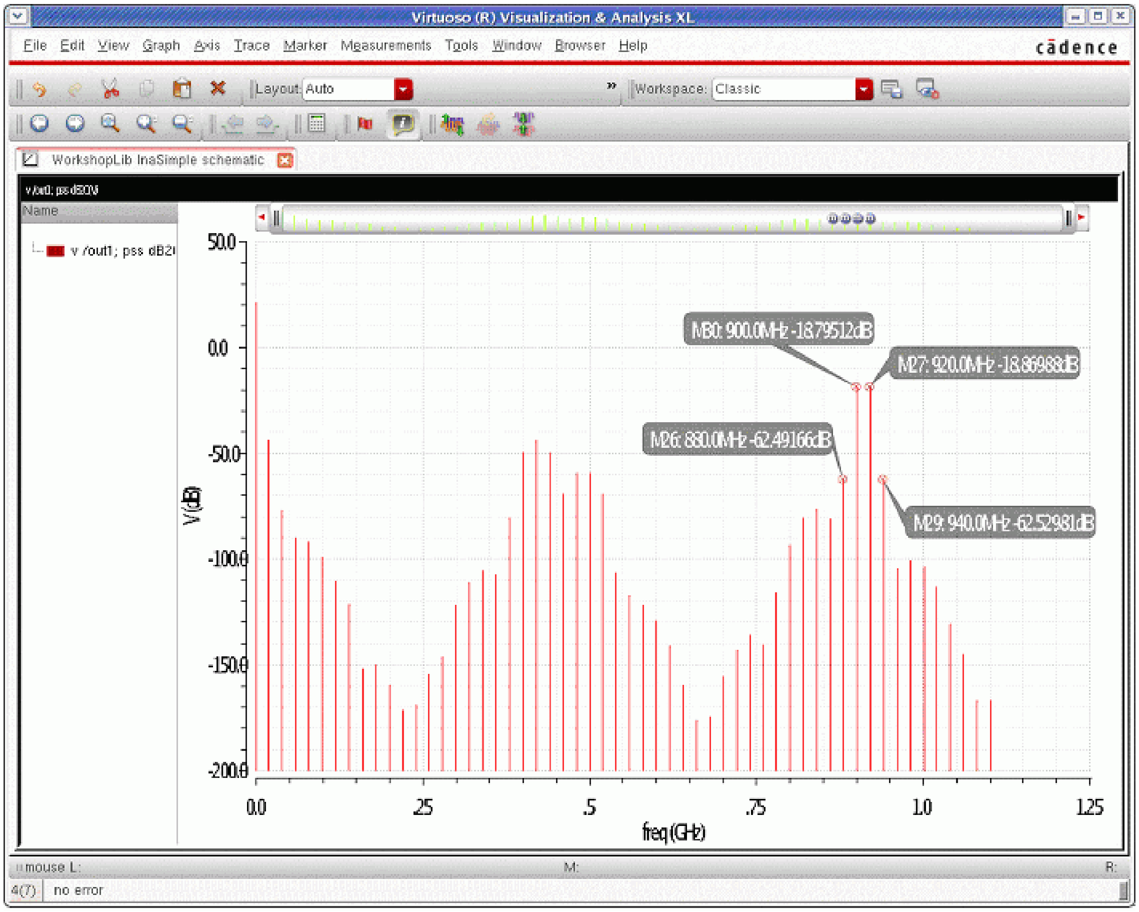


Below is a slightly different view of harmonics. Harmonic balance is supported in pss, where all of the harmonics starting from zero through the highest harmonic are calculated.

Note: This is not suggested for most applications. This is just to understand the concepts only.

The first view shows the solution with slightly more harmonics than the number needed to calculate the upper third order product at 940MHz. The circuit is an amplifier with inputs at 900MHz and 920MHz.

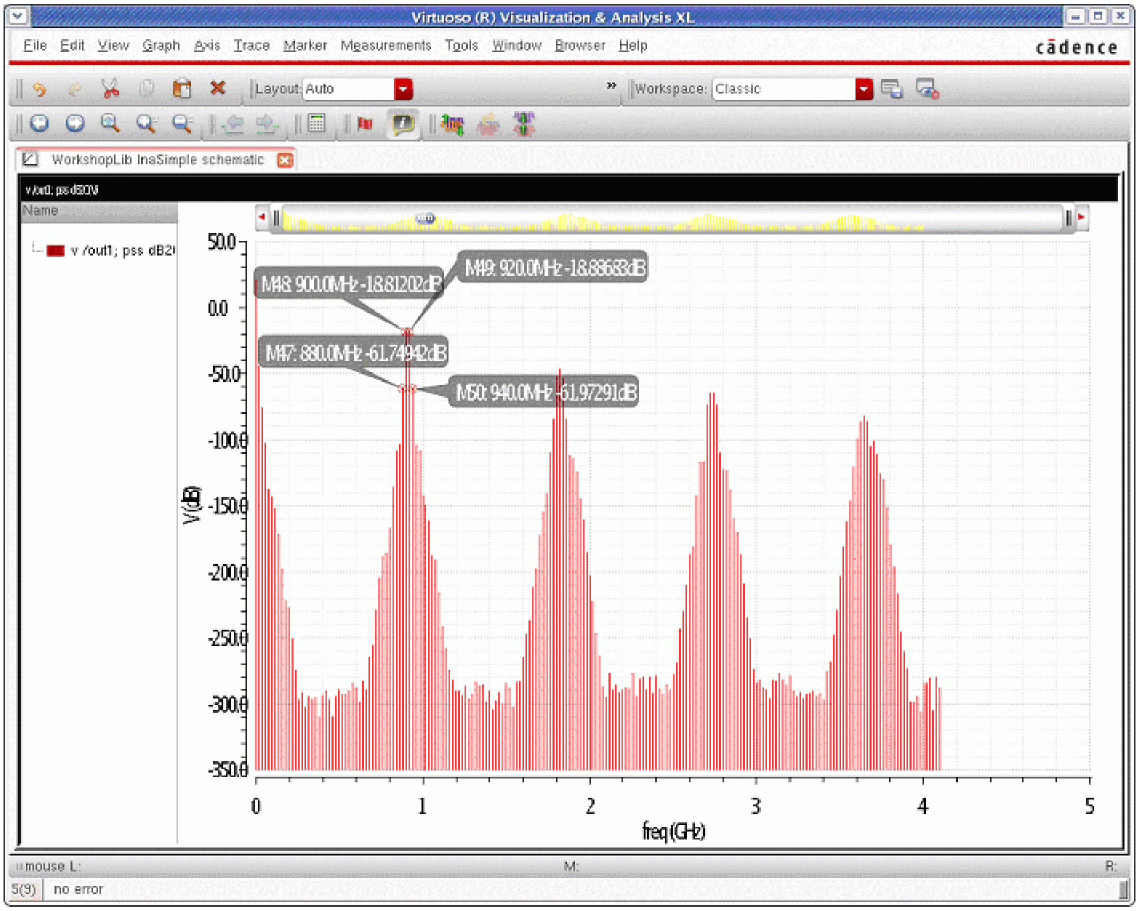
The third order products are at 880MHz and 940MHz.



Note that there are significant harmonics from about 380MHz through 520MHz. These harmonics are not produced by the circuit. These harmonics are produced because the power in higher harmonics is significant, and that power shows up in the solution with the number of harmonics set up for this run. The take away from this is that if you do not calculate enough harmonics in the solution space, random and unpredictable errors are introduced in the solution specified by you.

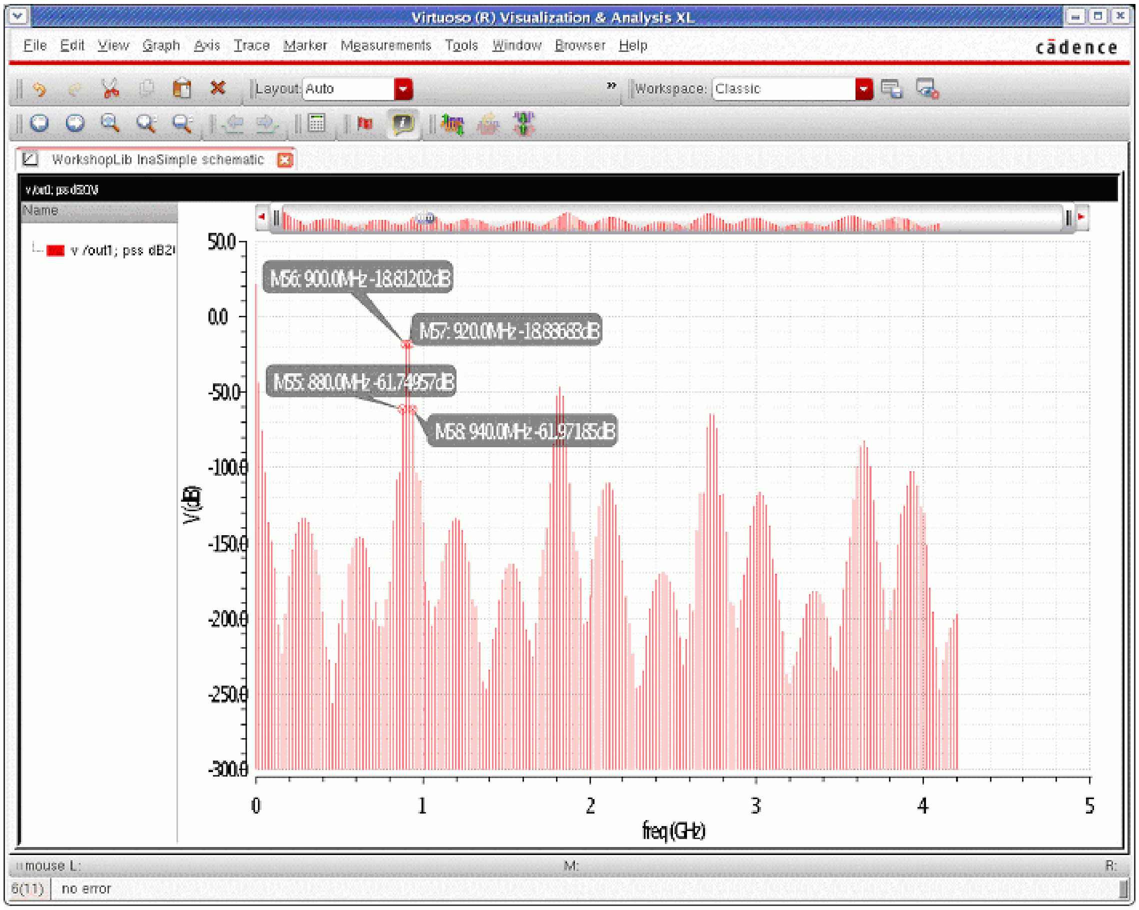
When the number of harmonics are raised, the solution changes. The solution below has 205 harmonics and appears to have no aliasing. Note that the first order terms at 900MHz and

920MHz have only changed slightly, but that the third order terms have changed by a significant amount.

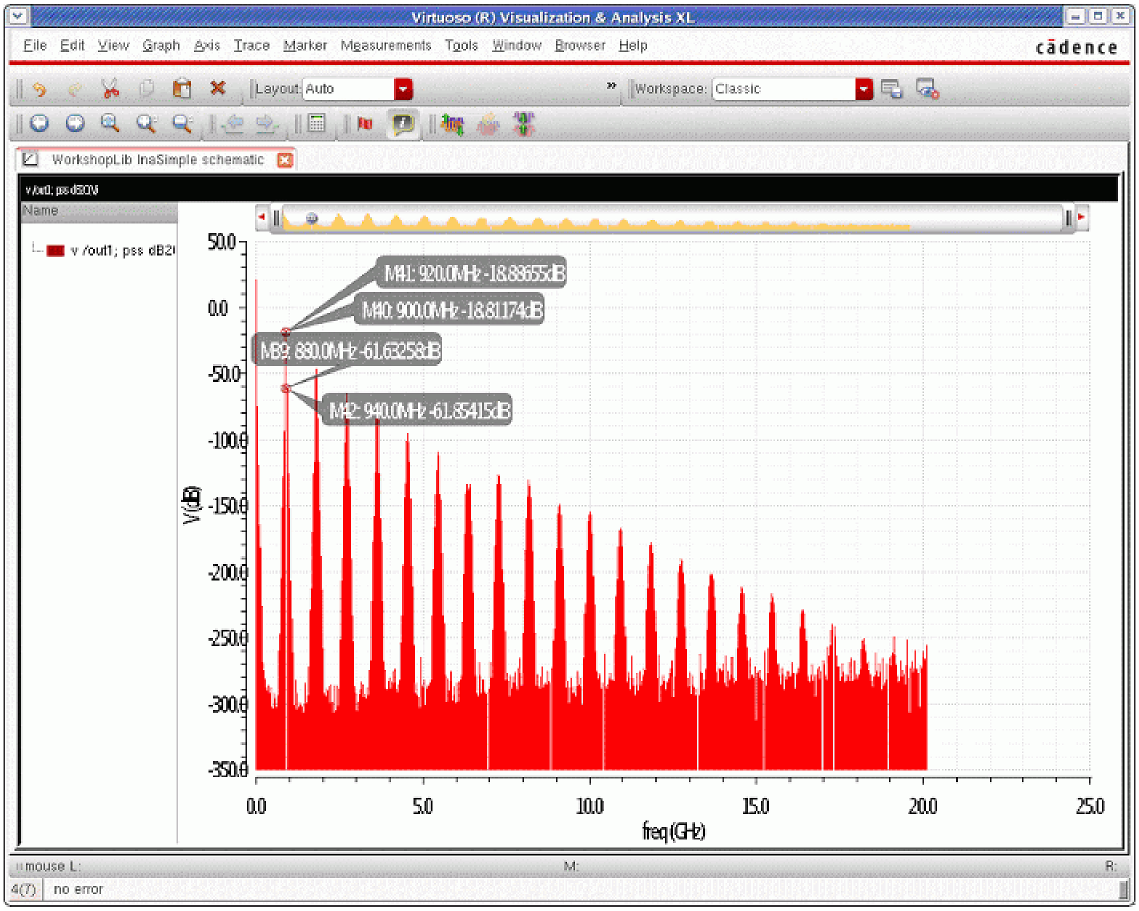


However, when the number of harmonics is raised to 210 (only 5 more harmonics), the figure changes. Here, it is apparent that there aliasing is still going on, and it is a bit difficult to infer which harmonics are real and which are produced by aliasing.

This is shown below.



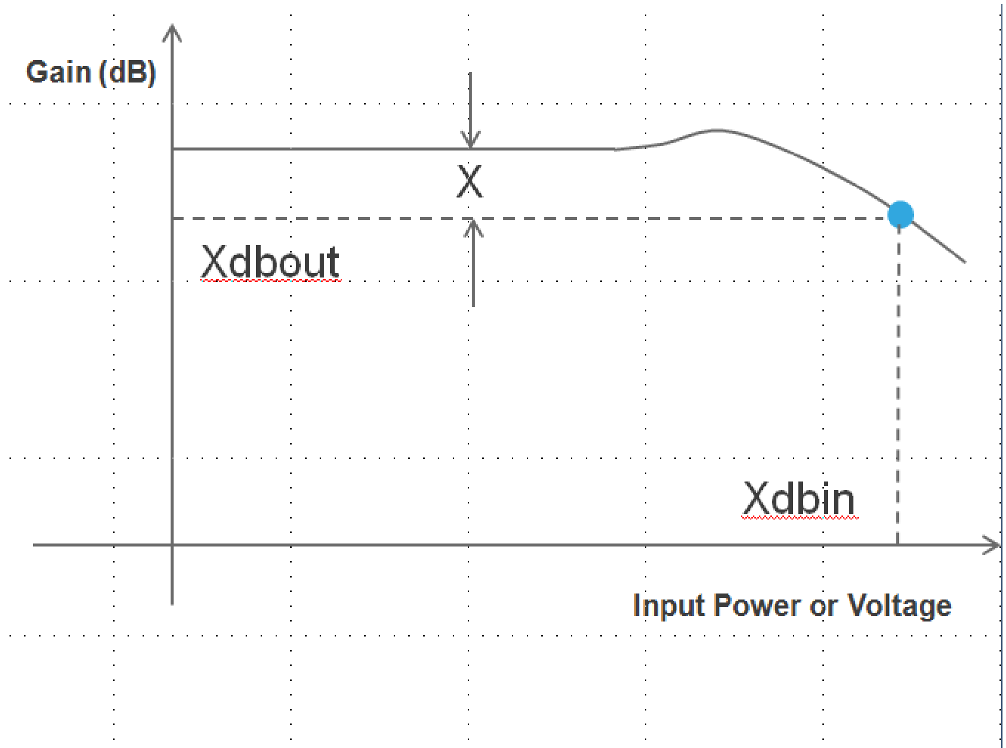
It is only when 1005 harmonics are taken that the aliasing stops because the levels of the higher harmonics are on the order of the numerical noise floor.



Gain Compression Analysis

Gain compression analysis is available in the harmonic balance *Choosing Analyses* form. Compression measurements can be made with either a port or a voltage source as the input to the circuit. Compression analysis does not require manually setting any input sweeps, and has dedicated direct plot functions to plot voltage or power at the compression point and the compression curves.

Gain compression is a measure of gain reduction resulting from circuit nonlinearity. Spectre allows you to measure either transducer power gain or voltage gain compression. Multiple levels of compression can be specified, if desired. In Xdb compression analysis, Spectre sweeps the input power (or voltage) automatically to arrive at the desired compression level. Compression is defined as shown below.



$$G_{power} = dB10\frac{P_{load}(f_{out})}{P_{available}(f_{in})} \qquad G_{voltage} = db20\frac{V_{out}(f_{out})}{V_{source}(f_{in})}$$

The gain reference for compression measurement can be either the gain at small input amplitude as shown above, or the gain reference can be the maximum at the peak above. The diagram above shows the gain, but the actual power (or voltage) in the sweep can also be plotted. Small-signal analyses can also be specified and will be run at the compression point only.

Compression can measure power or voltage compression. For a power compression measurement, the input must be a port, and the output power is measured through a resistor, port or current probe. For a voltage compression measurement, the input is a vsource and The output is taken across a pair of nodes.

Multiple levels of compression can be specified in a run. In the *compression Levels (dB)* field, type a list of compression values separated by spaces.

Rapid Mode is also provided. This mode supports only a single compression level. In this mode, you provide an estimate of the input power or voltage at the compression point. Then, three runs are made. One at low power, one at the level you specified, and a third that is slightly lower then the compression point. Results are then extrapolated based on these three simulations. This mode works well for Monte Carlo and Corners where a large number of simulations are required.

Gain compression analysis stores the following information:

1. The compression point.
2. Gain and power compression curves.
3. The HB simulation results at the compression point.

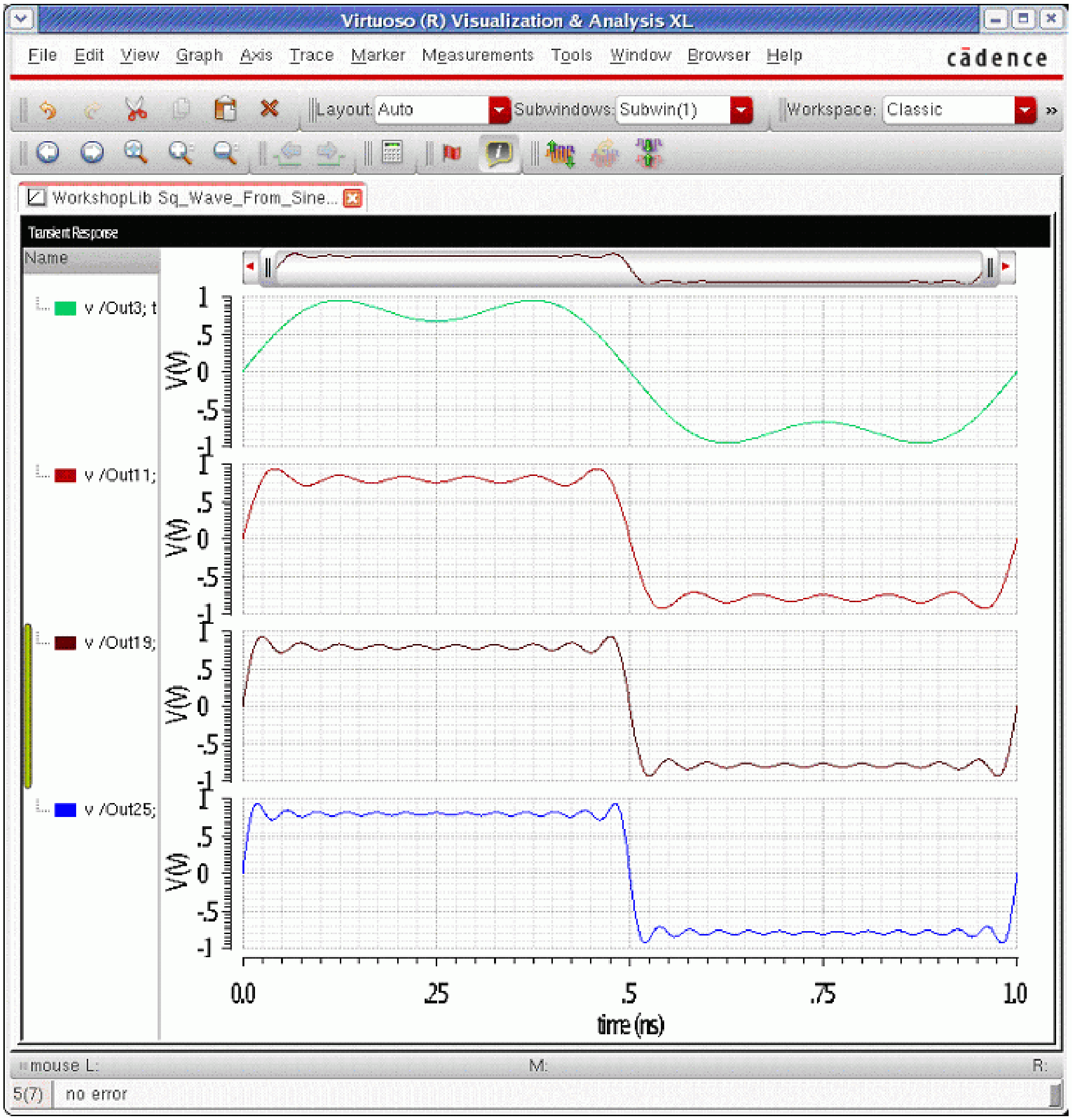
Direct Plot functions are provided for the results.

Small-signal analyses can be set up and run, and will be run only at the gain compression point.

Sine Representation of Square Wave

In harmonic balance, a square wave is approximated by a series of sinusoids at the odd harmonics. The actual waveform that the circuit sees is shown from a transient simulation in the waveform window shown below. The top trace has the first and third harmonic. The second trace contains all the harmonics through the 11th harmonic. The third trace contains all the harmonics through the 19th harmonic. The bottom trace has 25 harmonics. Note that as more harmonics are added, a closer approximation is made to the square wave. The flat part of the waveform gets flatter and the edge gets steeper.

Note that this is done for all non-sinusoidal waveforms by taking the fft of the waveform specified through the harmonic number and then stacking the voltage sources in series with the values set to the value from the fft.



Convergence

Like all harmonic balance simulators, an iterative approach is used to solve the circuit equations. In an iterative approach, there is always another iteration that causes the solution to become more accurate. You stop iterating as soon as the error in the solution becomes small. As a result, the solution is not exact.

To indicate progress in the Spectre output, two metrics are provided.

- `Resd_norm` measures the absolute error in the solution.
- `Delta_norm` measures the change from the last iteration.

Convergence is achieved when the sum of the currents at all the nodes is near zero, and there is only a small motion between the last iteration and the current iteration. When both `resd_norm` and `delta_norm` are less than one, the simulation has converged and the iterations stop.

The absolute accuracy is affected by the product of `reltol * residualtol`. The change that is allowed form iteration-to-iteration is affected by the product of `reltol *lteratio * steadyratio`.

Reltol is a global option, which is an option that affects all analyses that are run. This option can be set in ADE by selecting *Simulation - Options - Analog*. The default for reltol depends on the setting of `liberal`, `moderate`, or `conservative`, whether the circuit is driven or is an oscillator, and the number of inputs to the circuit. The *Residualtol* option is located just below the *reltol* option in the form, and is a multiplier to reltol. Normally residualtol is not used in SpectreRF. Lteratio depends on the setting of `liberal`, `moderate`, or `conservative` for the errpreset option.

Errpreset

Errpreset is shown in the ADE implementation section. Normally, the option *lteratio* is not set away from the default for harmonic balance. *Steadyratio* is a harmonic balance option, and defaults to 1.0. Setting a larger *steadyratio* value allows a larger change from iteration-to-iteration and still allows convergence. Following are the default values for *reltol*, *lteratio*, and *steadyratio*.

1 input driven circuit	Reltol	Lteratio	Steadyratio
Liberal	1e-3	3.5	1.0
Moderate	1e-3	3.5	1.0
Conservative	1e-4	10	1.0

multiple input driven circuit or oscillator	Reltol	Lteratio	Steadyratio
Liberal	1e-3	3.5	1.0
Moderate	1e-4	3.5	1.0
Conservative	1e-5	10	1.0

Note that the values for the *reltol* option are the maximum values for the frequency domain iterations only. For transient assist, the value of the *reltol* option in the global options form is used. If the *reltol* option is set to a value smaller than the values in the table, that value will be used. If the *reltol* option is set to a value higher than the values in the table, the values in the table will be used.

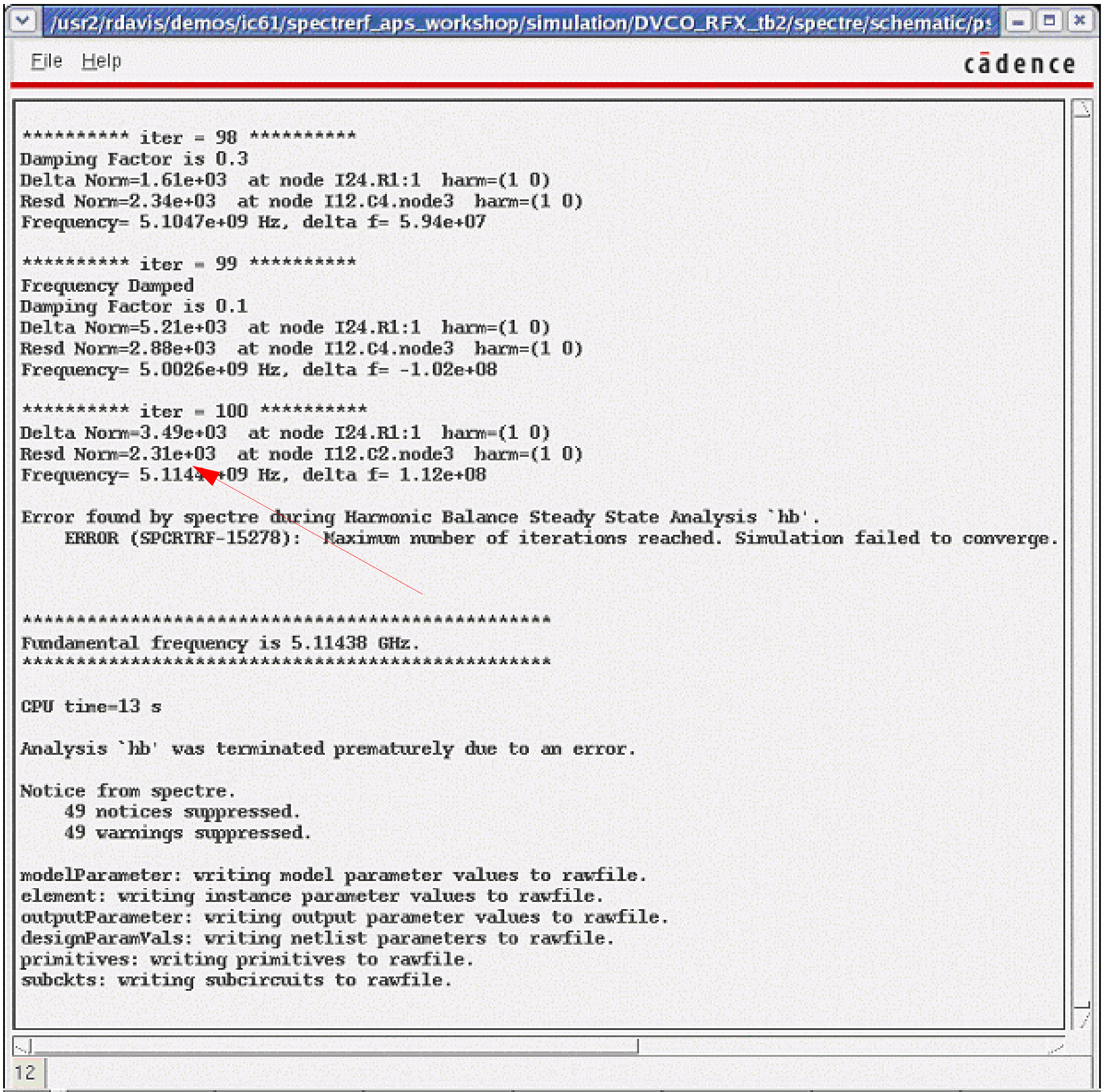
In some cases, the `resd_norm` (a measure of absolute error) in the Spectre output window will be below one, but the delta norm (a measure of the change between iterations) remains above one. In this case, the solution varies from iteration to iteration more than the amount allowed. This is usually caused by the small-amplitude harmonics changing relatively a lot, but since they are small, allowing a large delta is acceptable. Look at the delta norm to see the range it is in, and then set the *steadyratio* option slightly larger than that and re-run the simulation.

Iteration Limit

All iterative approaches have the possibility of not converging. In hb, there is a default limit of 100 iterations. This can be changed by setting the `maxperiods` option.

Note: If the iteration limit is reached without convergence, the solution point for the last iteration is written to the data file and is available for plotting. Make sure that the simulation actually converged by checking the Spectre output window. If you are running a sweep, the sweep will stop when the error is incurred.

The message is located at the end of the output, and is shown below.



You can see from *resd_norm* at the last iteration, that the error is significant. (it is much greater than one.) This answer cannot be trusted. To allow more iterations, set the `maxperiods` option to a number greater than 100.

Number of Non-Sinusoidal Sources

Up to two non-sinusoidal sources are allowed in the circuit. The signal that causes the largest number of harmonics or is periodic piecewise linear in the circuit should be *Tone1* when *Tones* are set to *Frequencies*. *Tones* is a choice in the *Choosing Analyses* form. If *Tones* is set to *Names*, enable tstab in the list for this signal. The second non-sinusoidal source is limited to being a pulse waveform

All the rest of the sources in the circuit must be sinusoidal. When *Frequencies* is selected, there is a limit of four input frequencies in the circuit. *Names* does not have this limitation, but four input frequencies is a practical upper limit because the number of harmonics to solve explodes.

When *Names* is selected, the sources in the circuit need to have entries in the *Frequency name 1* or *Frequency name 2* properties in the signal sources in the circuit. When assigning names, make sure that if you have multiple inputs at a single frequency, like there might be for a differential circuit, the names should be the same. This enables SpectreRF to consider the two signals as one. If you have frequencies that are integer multiples of each other, these should have the same name too. In this case, the actual frequency will be the lower of the two frequencies. Make sure that enough harmonics of the low frequency are specified to incorporate enough harmonics for the high frequency input to prevent aliasing.

Harmonic Balance Starting Point

Harmonic balance has traditionally started from the DC solution for the first iteration in the frequency domain. This is not the default for SpectreRF. The default is to run a short transient until steady-state is reached, and then switch to harmonic balance where the number of harmonics is selected automatically based on the transient analysis waveforms.

Starting from a DC point is a reasonable starting point for driven circuits. The input signal causes harmonics to be created and solved in the frequency domain. Transient aided harmonic balance is also available. This is accomplished by selecting Decide automatically or Yes for run transient. If Yes is selected, setting a simulation time in the stop time property is needed. Transient aided runs a transient analysis on the signal that you select for the time specified, and then runs a single period of the input. A Fourier transform is performed, and the result is used as the starting point for the frequency domain iterations. Because the starting point is closer to the actual solution, it usually requires fewer frequency domain iterations at the expense of running the transient analysis at the start. It is frequently faster to run a short transient and then iterate in the frequency domain. The only way to determine which is faster is to try both ways and see which one finishes first. Only one signal can have transient assist. When *Tones* is set to *Names*, the signal with tstab set to *yes* can have transient assist. Choose the signal that produces the largest amount of distortion for transient assist. When *Tones* is set to *Frequencies*, only *Tone1* can have transient assist. *Tones* is an option in the *Choosing Analyses* form.

Transient assist can also be used to improve convergence. It works because the starting point in the frequency domain is closer to the real solution than the DC solution, which is the default. Anytime an iterative approach is used, the closer the starting point is to the real solution, the higher the probability of convergence.

In some cases, very lengthy transient simulations are needed to produce convergence. There are two ways you might use to shorten this time. The first way is to specify a filename in the *writehb* option on the first run, and then use the *readhb* option on subsequent analyses. The *writehb* option writes out every relevant parameter to a file so that when *readhb* is specified, the simulation can start from the file. The second way to reduce the time is to save the state near the end of the transient assist (tstab) interval, and then use that state to restart the transient assist with a much shorter simulation time before going into the frequency domain iterations. This is accomplished using the *savefile*, *saveperiod*, and *savetime* options.

Both of these ways allow circuit values to be changed when restarted, but the topology of the circuit cannot be altered. When the topology is altered, the transient assist needs to start over. If a value is changed, the change causes an instantaneous discontinuity in the solution and because of this, if the change is too large, there will be convergence failures.

More Capabilities

Save and Recover

Save and recover writes and restores the full internal state of the transient analysis that is used in the tstab interval so that a circuit can be restarted from that state to save time in a simulation that has a long startup time, or in the case where a longer tstab is needed for the transient assist. For now, savefiles that are made using the transient analysis are not usable as a recover file in harmonic balance. The savefile must be created from the tstab interval in harmonic balance or pss.

To save a file, set the *savefile* option to a filename. If you are working from a netlist, the file will be created in the directory you are running from. If you are in the ADE environment, the file is located in the `netlist` directory. This is located in the *simulation/<circuit_name>/<simulator_name>/<view_name>/netlist* directory. The simulation directory defaults to your home directory, but it can also be set by the administrators of the Cadence tools. In most cases, this directory will be shown in the project directory of the *Setup - Simulator/Directory/Host* menu in ADE. If you are unable to locate this directory, contact your system administrator to find the location. Spectre is the simulator name, so if you had a circuit called `abc`, and you started ADE from the schematic view, the `netlist` directory would be located at *~/simulation/abc/spectre/schematic/netlist*.

Next, set either *saveperiod* or type in a list of times in the *savetimes* field. Do not specify both at the same time. If tstab is set to 50n, *saveperiod* is set to 10n, and the *savefile* option is set to *save_file*, when the simulation reaches 10n in the tstab interval the file *save_file* is created in the `netlist` directory. When 20n is reached, *save_file* is overwritten with the data at 20n. This continues until the last 10n interval is reached in the tstab interval.

Alternatively, you can type 10n 25n 40n 50n in the *savetimes* field. This would create four files beginning with *save_file* and ending with the time for that specific point. The exact format of this file changes occasionally, so look in the `netlist` directory to get the actual filename you want to start from.

To recover, you specify the filename you want to recover from, and set tstab to a larger time than the time in the file. You can also change the list of times in the *savetimes* option to save more times after the time you recovered from.

Writehb and Readhb

Setting *writehb* causes the entire harmonic balance solution to be written to a file that can be used later. When readhb is set to a previously saved *writehb* file, the solution is read from the file, and then follow-on small-signal analyses can be performed. Note that *readhb* and *writehb* cannot be set at the same time.

Readhb can also be used as a starting point for when more input frequencies are supplied to the circuit. If you have a 2-tone solution, and then you add a third tone, The two tones in the solution are used as a starting point for the 3-tone solution which must iterate to achieve convergence. Because the starting point contains the solution for two signals, this is a better starting point than using transient aided where the solution to only one tone is available for the first iteration.

Sweeps and Restart

Sweeps are available both in the hb *Choosing Analyses* form and from the parametric tool in ADE. Sweeps automatically use the previous solution as the starting point for the next member of the sweep. If convergence difficulties are encountered, the sweep parameter will be automatically stepped from the value that converged to the value that needs to be measured. When Monte Carlo encounters convergence difficulties, several things are tried to achieve convergence. If you want to disable these automatic features, set the restart option to yes.

Itres

Itres controls the precision of the solution at the first iteration in the frequency domain. Harmonic balance starts with the DC solution as the starting point which causes the solution at the first iteration to be inaccurate. Because of this, the first solution is calculated without much precision (number of digits in the solution) in order to save time. Follow-on iterations have more precision. The final solution has full precision. Note that if you use transient-aided harmonic balance, the starting point in the frequency domain can be quite good, and so itres should be manually set to 1e-2. This forces enough digits in the mantissa to be solved so that the solution differs from a solution with all the digits of the mantissa by less than 1 part in 100. If the circuit is nearly linear (for example an LNA) increasing the precision by setting itres to 1e-2 can also reduce the total number of iterations. Regardless of the setting of itres, as the iterations progress, the precision is automatically increased so that the final solution is accurate. The default is 0.1, which allows a 10% error on the first iteration.

Freqdivide

If the circuit has a frequency divider, specify the divide ratio in the *freqdivide* property. This needs to be an integer. Make sure that oversample factor is set in this case as discussed earlier. Also, make sure that transient-assisted is used, and set the transient part to at least the time needed for the divider to count through its sequence once. The effect of this is that the fundamental frequency is divided down by the freqdivide ratio. Make sure that enough harmonics are specified so that at least the fifth harmonic of the input signal is calculated. If *freqdivide* is set to four, make sure that at least 20 harmonics are specified for that signal.

Oscillators

For oscillators, a DC starting point is not a good starting point. The frequency and amplitude of the harmonics needs to be solved and the DC state has no information at all about this. There is no signal source, so the oscillator usually just does not start. The default in ADE is to estimate the oscillatory frequency and amplitude, and then run a transient analysis until steady-state is reached. Upon reaching steady-state, the number of harmonics needed for the oscillator are calculated by evaluating the waveform at every node in the oscillator, and then running harmonic balance to solve for the frequency and amplitude at each node. This is a good starting point for LC, crystal, or mems oscillators.

If the oscillator is a ring oscillator, apply the initial conditions to force one stage of the ring to a high or low state. Set Run transient to Decide automatically. If desired, turn off *Calculate initial conditions automatically*, which will only work for a conventional feedback oscillator.

When using transient assist manually (called the tstab interval) for oscillators, set *Run Transient* to *Yes*. Specify a time for tstab, on the order of twenty periods of oscillation. Select the check box for Detect Steady State. This will enable the transient, and automatically switch to harmonic balance when steady-state is reached. In this mode, tstab cannot be lengthened automatically like it can be when Decide Automatically is selected.

In many cases, it is desirable to see the startup waveform, which is not saved by default. If you want to see the startup behavior, select *yes* for the *Save Initial Transient Results (saveinit)* option.

When oscillators are simulated, a node or pair of nodes (if it is differential) needs to be declared to the simulator. This pair of nodes is used to get an estimate of the oscillation frequency at the end of the transient assist period. At the end of the transient assist period, 5 periods of the estimated operating frequency are simulated. In these 5 periods, the simulator looks for the frequency divided output at every node in the circuit and if it finds this behavior, it automatically adjusts the frequency. The frequency is estimated based on these last 5 cycles. A Fourier transform is performed on the last single cycle, and this is the starting point in the frequency domain. If frequency dividers are in the circuit, the estimated frequency of the oscillator is recommended to be specified in the *Choosing Analyses* form, and the frequency divide ratio should be specified in the freqdivide option.

Solving the frequency in addition to the amplitudes and phases adds another unknown to the solution set, so it usually takes more iterations to reach a solution compared to a driven circuit.

Pinnode

In a driven circuit, all the phases and amplitudes of the harmonics are defined by the input signal. There is no input for an oscillator, so a pinnode must be selected. The first harmonic phase of that node is pinned (held constant), and then the amplitude of all the nodes and relative phases of all the other nodes are calculated using the pinnode phase being held constant. SpectreRF has an algorithm that automatically selects the pinnode, and it is usually reliable. If convergence is an issue, try setting the pinnode manually in the options form to one of the nodes in the resonator.

To set the pinnode manually, select the *Options* button at the bottom of the hb *Choosing Analyses* form and select the convergence tab (*IC61 only*). In the options field, do not enter a slash(/) at the beginning of the expression. Just enter the hierarchical node name similar to this example: `I12.I1.osc_resonator`. In the schematic, IC61 shows the instance name of the block in the tab for that block, so it is a bit easier to get the hierarchical names.

Probe-Based Method

The normal method of solving for an oscillator allows the amplitudes, phases, and frequencies to be solved simultaneously. A second method called the probe-based method iterates for the frequency solution in the outer loop and the amplitude and phase solution in the inner loop. It places a voltage source between a pair of nodes you specify, which should be in the resonator. If you have a single-ended oscillator, specify one node only. If the second node is left blank, it will be connected to the global ground node automatically. The voltage source converts the oscillator into a driven circuit, which converges easier. The source amplitude, phase, and the frequency are adjusted until the current in the source is near zero. Do not use the probe-based method and the pinnode option together, since the probe-based method also requires a pinnode to be selected. *Harmonic Index* and *Magnitude* do not need to be specified for the probe-based method. The *harmonic index* defaults to 1, which is suggested, and the *Magnitude* defaults to a small amplitude. If you specify *Magnitude*, make sure that the magnitude you specify (in volts RMS) is smaller than the amplitude of the actual oscillations, or the method itself will fail.

Oscillator Tuning Mode

Tuning mode adjusts a parameter in the circuit to produce the set frequency target. This is done automatically when tuning mode is enabled without setting any sweep parameters or interpolation of the resulting curves. When the tuning frequency is reached, any small-signal analyses like noise are run. This allows the simulator to tune the oscillator to a specified frequency and then make a noise measurement. This is useful in Monte Carlo analysis to see how the oscillator performs with process variations. The parameter to be tuned can be a variable, temperature, or a specified device parameter. Oscillator tuning mode is also supported in the pss analysis using the shooting or harmonic balance engines. See Chapter 4, “Single Input Large and Small-signal Analyses,” for details.

In this mode, the target frequency to tune to is the frequency that is specified in the harmonic balance form as the fundamental frequency. You specify a parameter that is to be varied to achieve the desired frequency. This can be a variable, a device parameter, or temperature. When the analysis runs, the oscillator will be tuned tot he desired frequency, and then all the small-signal analyses will be run.

Direct plot functions have been added to plot the tuning parameter.

Semi-Autonomous

Semi-autonomous is provided to allow the simulation of circuits that have periodic sources and oscillators in the same circuit. It is only supported in the hb *Choosing Analyses* form. Typical applications are for when you have a receiver with an RF tone and an oscillator for the

LO so that all the oscillator noise is taken into account for the receiver noise simulation. Another application is for identifying the phase noise and spurious response of an oscillator when power supply ripple is applied.

The simulator combines the driven and oscillator capability in one analysis. To set it up is similar to a multi-tone simulation, just select the *Oscillator* check box and specify the estimate of the oscillator frequency in the *osc!* section of the form. The probe-based method for oscillators is not available for semi-autonomous simulations.

Things to try to Achieve Convergence (Driven Circuits)

Try running transient-assisted by setting the tstab option. Ensure that the signal that causes the most distortion is in *Tone 1* (when *Frequencies* is selected) or has tstab enabled (when *Names* is selected). Transient assist can help on any circuit.

If there is an S-Parameter block in your circuit, make sure that causality is set to *fmax* on every instance of the nport. Starting with MMSIM 11.1 this is the default. The effect of this is to enforce causality for the DC and tstab interval. If this selection is not made, even the DC solution can be affected, sometimes producing results that can be larger than the power supply voltage. Setting causality to *fmax* makes sure that the DC and tstab intervals have a causal model. Refer to the Simulating S-Parameters chapter for a description about the nport component. Also, make sure that every port of the nport has a connection to something in the circuit. Leaving a terminal open can be very bad for convergence.

Look at the startup waveform to see if it is settled. Make sure that tstab is long enough. Look at the starting waveform, and ensure that the amplitude is reasonable for your circuit. To improve convergence, the waveform must be pretty close to settled at the end of the tstab interval where the Fourier transform is made.

Use enough harmonics. If your system produces square waves, make sure that at least five harmonics are calculated. In many cases, many more harmonics than that are required.

Try setting the oversample factor to a higher value. This should be set to four or eight for circuits that have square waves. Setting it to two or four improves the accuracy of the Fourier transforms used in harmonic balance (hb) even for nearly sinusoidal circuits.

Try setting itres to 1e-2. This forces a solution that is more precise on the first iteration, thus improving the chance of convergence.

Try more maxperiods if `resd_norm` is small (near one). This indicates that the circuit is almost converged, and might converge more if more iterations are allowed. The default maxperiods is 100.

If the residual norm is below one, but the delta norm is greater than one, set *steadyratio* just a little higher than the value in which the delta norm is running. The effect is to allow more delta from iteration to iteration and still allow convergence. Only do this if the `resd_norm` is less than one, which indicates that the absolute accuracy check is good.

Try moderate accuracy. Sometimes, loosening the tolerances allows a circuit to converge. Note that this does reduce the accuracy of the solution the simulator solves for.

Things to try to Achieve Convergence (Oscillators)

Try everything listed above. In addition:

Try setting *reltol* to 1e-5 and *vabstol* to 3e-8. These options can be found in the global options form in ADE. Select *Simulation - Options - Analog* to open the global options form. Making *reltol* and *vabstol* smaller makes the initial transient much more accurate by reducing the error caused by iterating to each solution and the numerical integration error. By providing a more accurate estimate of the frequency and amplitude of the waveform, convergence becomes easier.

Try setting a pinnode manually. While the algorithm that is used in SpectreRF is usually reliable, if the oscillator does not converge, set the pinnode to one of the nodes in the resonator.

Select the *Calculate initial conditions automatically* check box if you have an LC oscillator. This causes an initial estimate of the frequency and amplitude to occur at the start of the tstab interval. Add transient assist so that the harmonics can be calculated for the starting point in the frequency domain.

Try the probe based method. This method adds a voltage source to the circuit so it becomes a driven circuit. Because the circuit is driven, convergence is easier. The frequency amplitude and the phase of the source are adjusted so that the current of the source becomes zero. When this happens, the source has no effect on the circuit.

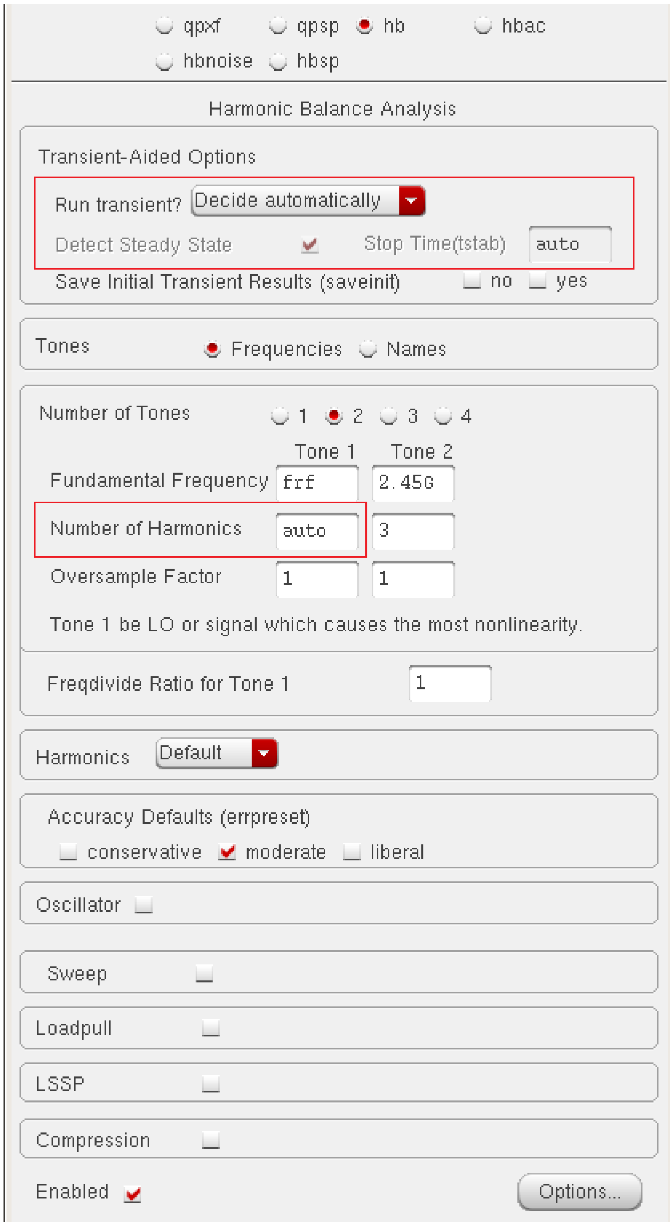
Try setting an initial condition (current) in the inductor of the resonator. Choose a value that approximates the steady-state peak value for the oscillatory condition.

Implementation in ADE

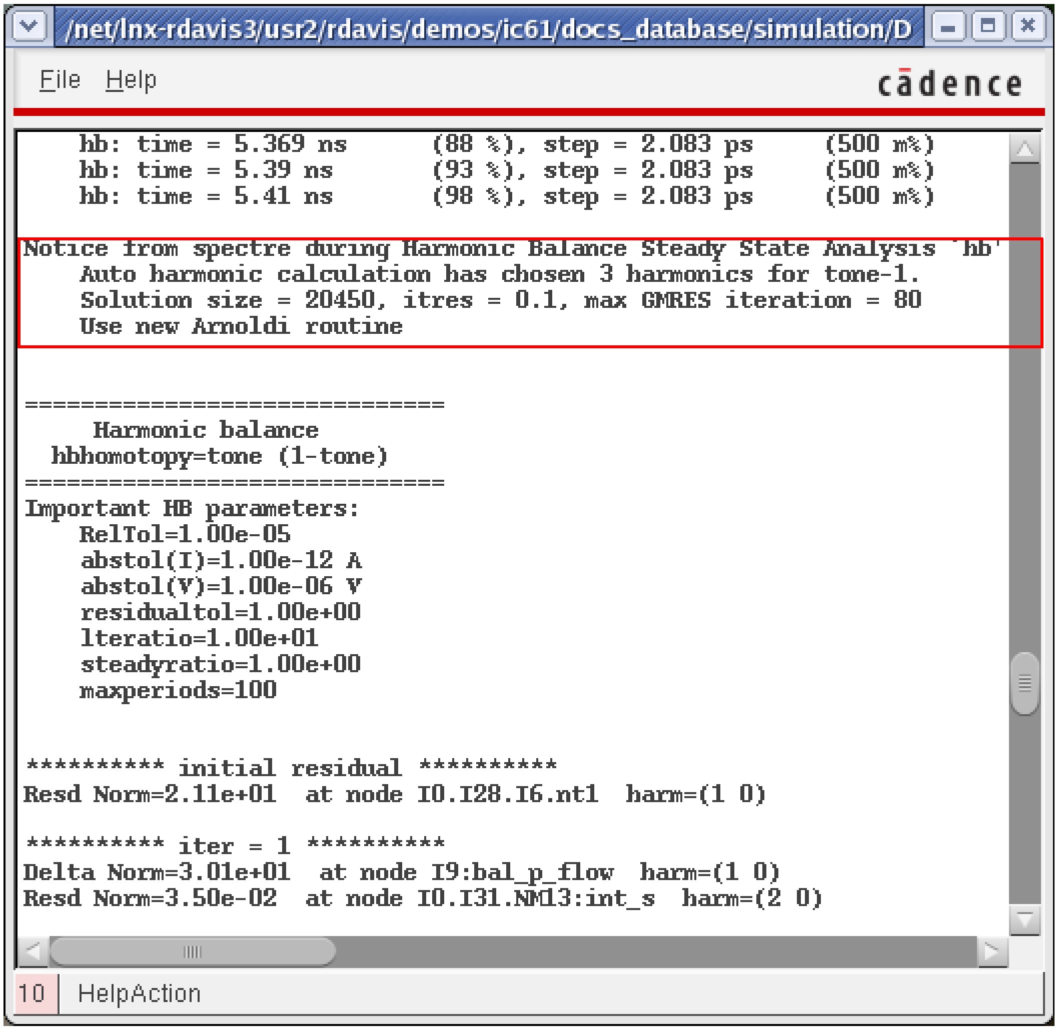
Setting Frequencies, Harmonics, and Oversample

Setting Harmonics and Transient Assist Automatically

The default in ADE is to run transient assist until steady-state is just reached, and then choose the number of harmonics for the first tone automatically, and then proceed with solving in the frequency domain.

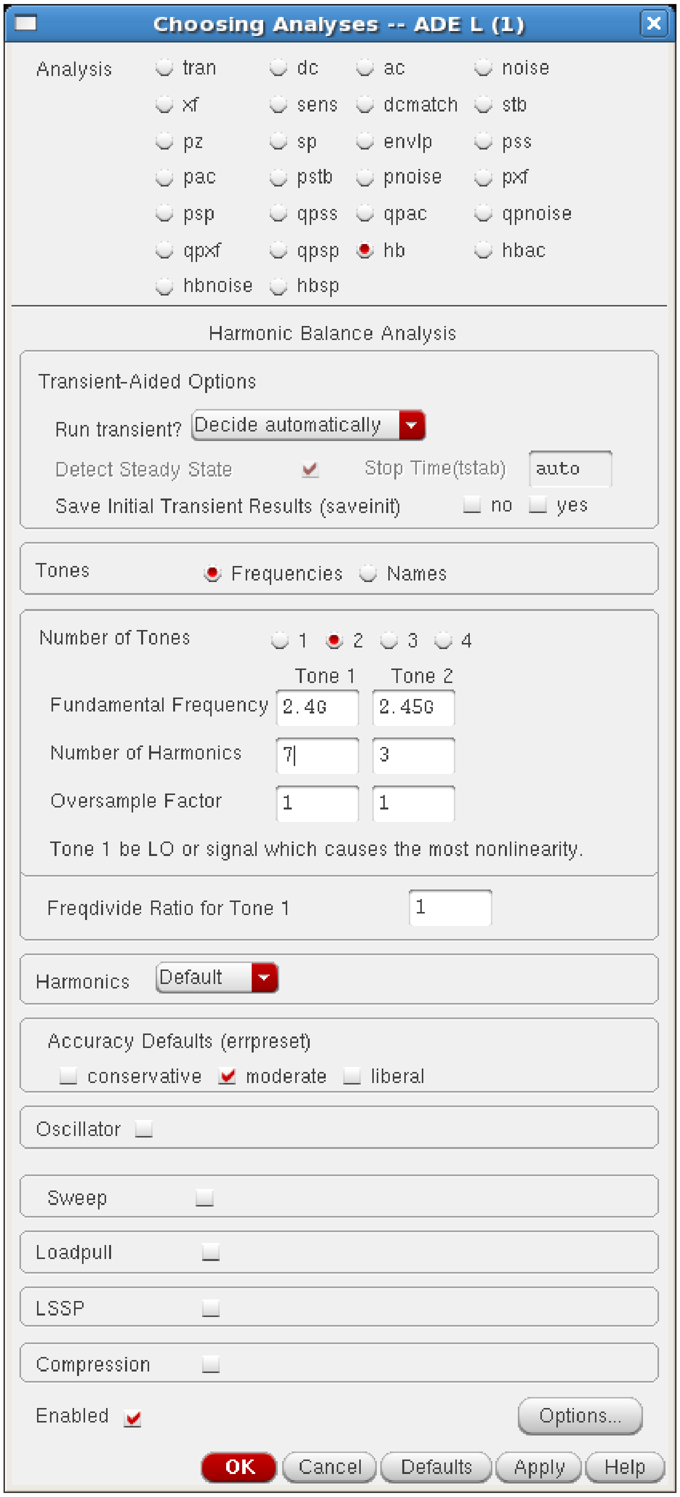

qpxf
qpsp
hb
hbac
hbnoise
hbsp
Harmonic Balance Analysis
Transient-Aided Options
Run transient?
Decide automatically
Detect Steady State
Stop Time(tstab)
auto
Save Initial Transient Results (saveinit)
no
yes
Tones
Frequencies
Names
Number of Tones
1
2
3
4
Tone 1
Tone 2
Fundamental Frequency
frf
2.45G
Number of Harmonics
auto
3
Oversample Factor
1
1
Tone 1 be LO or signal which causes the most nonlinearity.
Freqdivide Ratio for Tone 1
1
Harmonics
Default
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Oscillator
Sweep
Loadpull
LSSP
Compression
Enabled
Options...

The defaults are highlighted above. This configuration is recommended for all hb analyses.



Spectre will calculate the number of harmonics that are needed and report the number in the spectre output file, as highlighted above.

Tones = Frequencies



1. For sinusoidal sources, set *Oversample Factor* to 1.

2. For square waves, set *Oversample Factor* to 2, 4, or 8. Use the smallest number that provides accurate results.

3. If there are multiple sources with a common integer multiple frequency in your circuit, specify the highest common frequency. For example, if 1GHz and 1.5GHz are present in the circuit, set *Tone1* or *Tone2* frequency to 500MHz. In this case, make sure you take enough harmonics of 500MHz to get an accurate result at 1.5GHz.

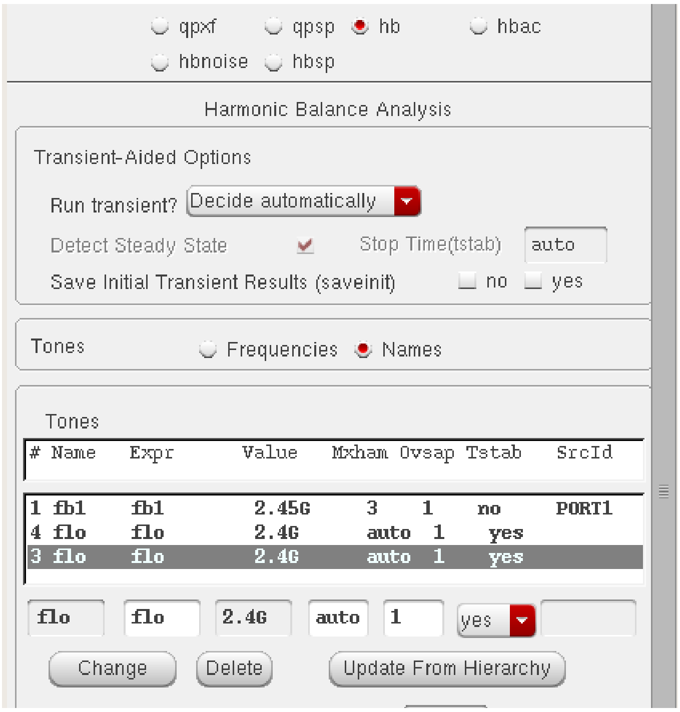
Important

Specify the signal that causes the most distortion in the circuit as Tone1. This is the tone that can have transient assist, and selecting the signal with the most distortion as Tone1 improves the accuracy of harmonic balance.

Tones = Names

1. Select a line in the sources list.

   The fields (highlighted below) are automatically populated with the values of the selected line in the sources list.



2. Change the number of harmonics under the *Mxham* column (auto in the figure above).

3. Change the oversample factor under the *Ovsap* column (1 in the figure above).

4. Set only one of the inputs to have transient assist by setting *Tstab* to *yes*.

Important

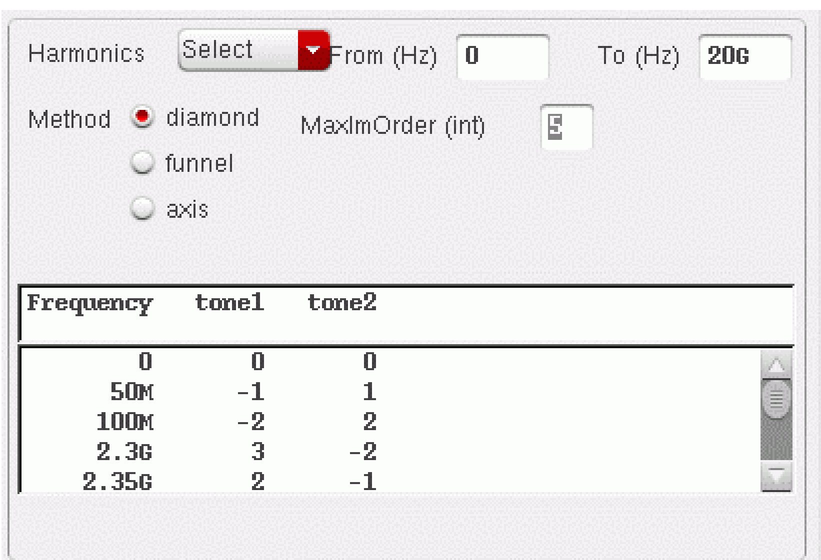
Make sure you enable tstab for the signal that has the largest amount of distortion in the circuit. This will make harmonic balance more accurate.

5. If there are multiple sources in your circuit that have integer frequency relationships, set the *Frequency name 1* property in all the sources in the schematic to the same name. This will cause all the sources to be treated as solving for harmonics of the highest common frequency. For example, if 2.4GHz and 3.6GHz were present in the circuit, a single-tone simulation would be run that solves the harmonics of 1.2GHz. Make sure that you use enough harmonics so that the enough harmonics of 3.6 GHz are solved to prevent aliasing errors.

Diamond Cut

This is used when the number of significant harmonics produced by the circuit are similar for all the inputs. This is usually the case for a large-signal IP3 measurement on an amplifier.

In the *Choosing Analyses* form:



1. Choose *Select* from the Harmonics drop-down list.

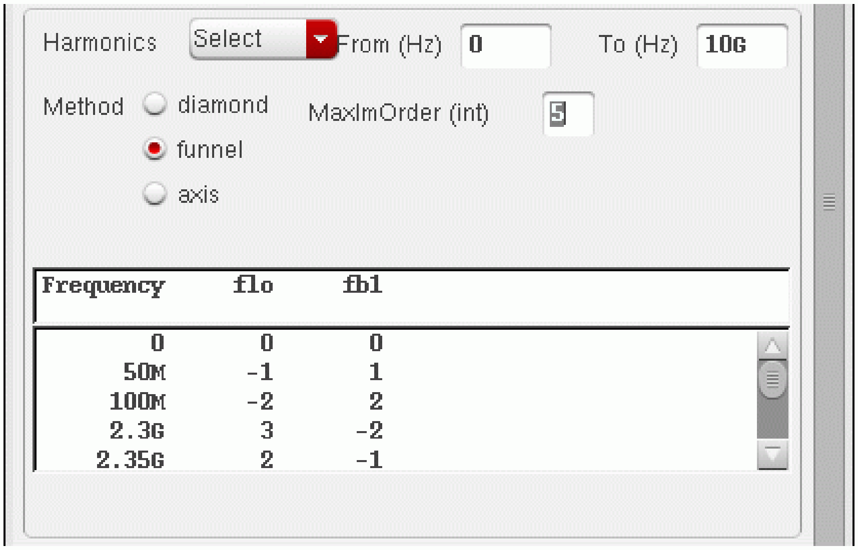
2. Select *diamond*.

3. Type a reasonable value in the *MaximOrder (int)* field. The value is usually between 3 and 7.

Funnel Cut

This is used when the number of significant harmonics produced by the circuit are different for one or more inputs. This is usually the case for a large-signal IP3 measurement on a mixer (The LO usually produces more harmonics than the RF input signals).

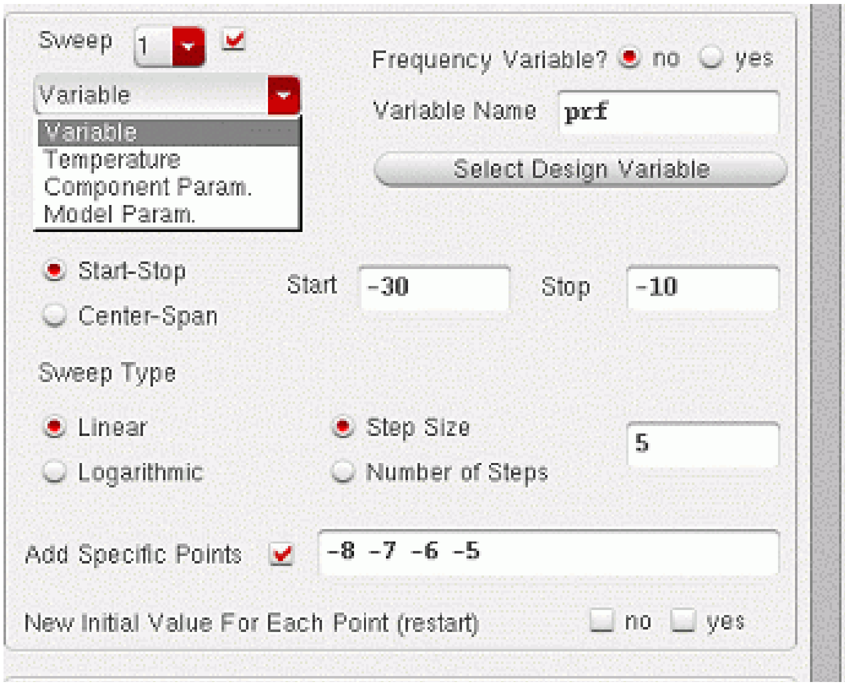
In the *Choosing Analyses* form:



1. Choose *Select* from the Harmonics drop-down list.
2. Select *funnel*.
3. Type a reasonable value in the *MaximOrder (int)* field. The value is usually between 3 and 9.

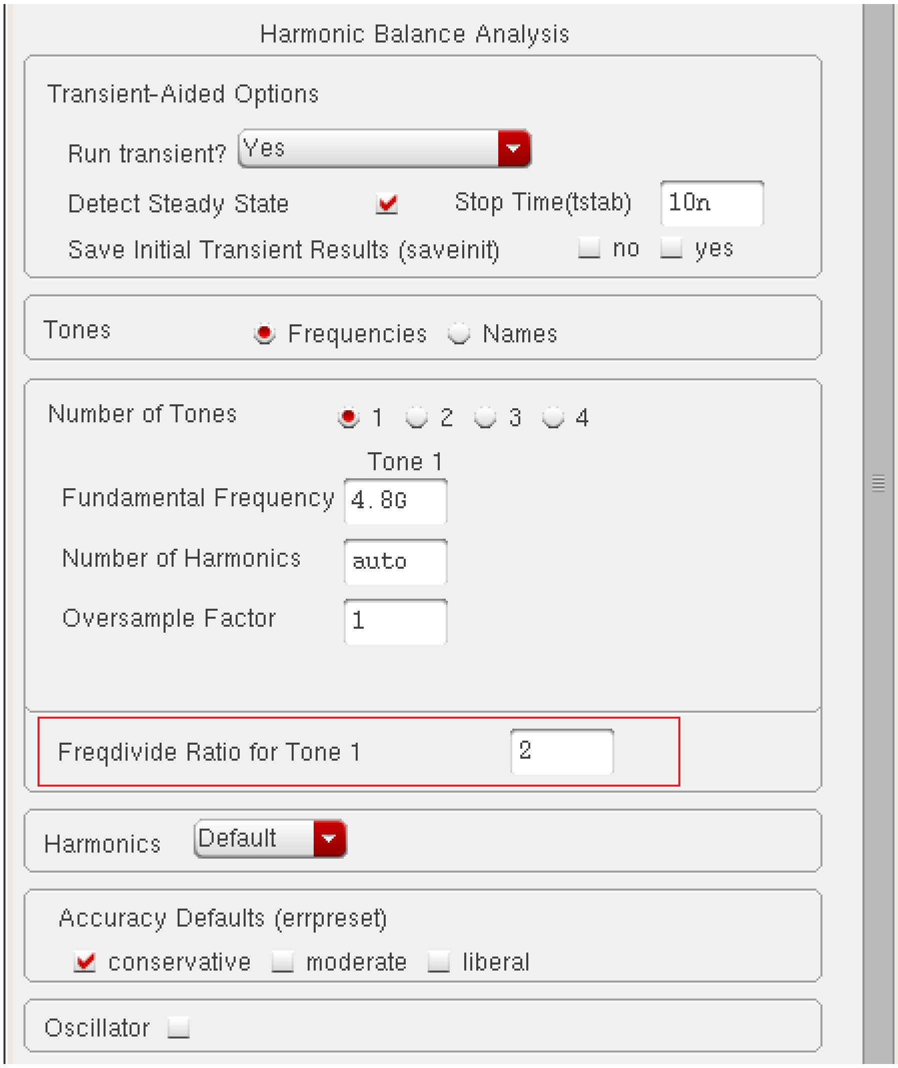
Sweeps

You can sweep up to three things in the hb *Choosing Analyses* form.



- Each can be a variable, temperature, component parameter, or model parameter.
- Choose what you want to sweep from the drop-down list.
- Set the start and stop range for the sweep in the *Start* and *Stop* fields.
- You can add points off the grid in the *Add Specific Points* field.

Freqdivide



If you set freqdivide, make sure that you have at least five harmonics at the input frequency as an absolute minimum. You must also set the *Oversample Factor* between 2 and 8.

Normally, when there are square waves in the circuit, for accurate results for the transient waveform, the number of harmonics needs to be at least the period of the square wave divided by the rise or fall-time, whichever is shorter.

Compression Analysis

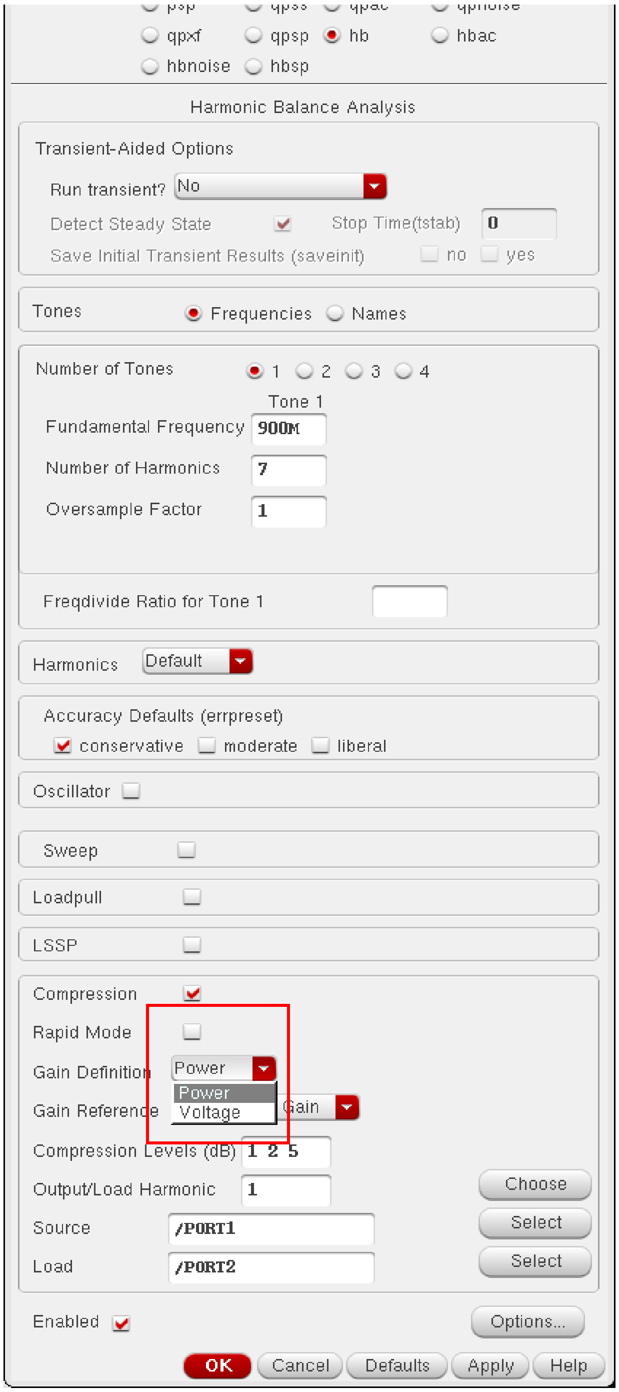
Compression analysis automatically finds the compression point without needing to define a sweep of the input power. Multiple compression levels can be set in the *Compression Level (dB)* field. Separate the levels with a space.

First, set a reasonable number of harmonics that is enough for the simulation to be accurate at the compression point.

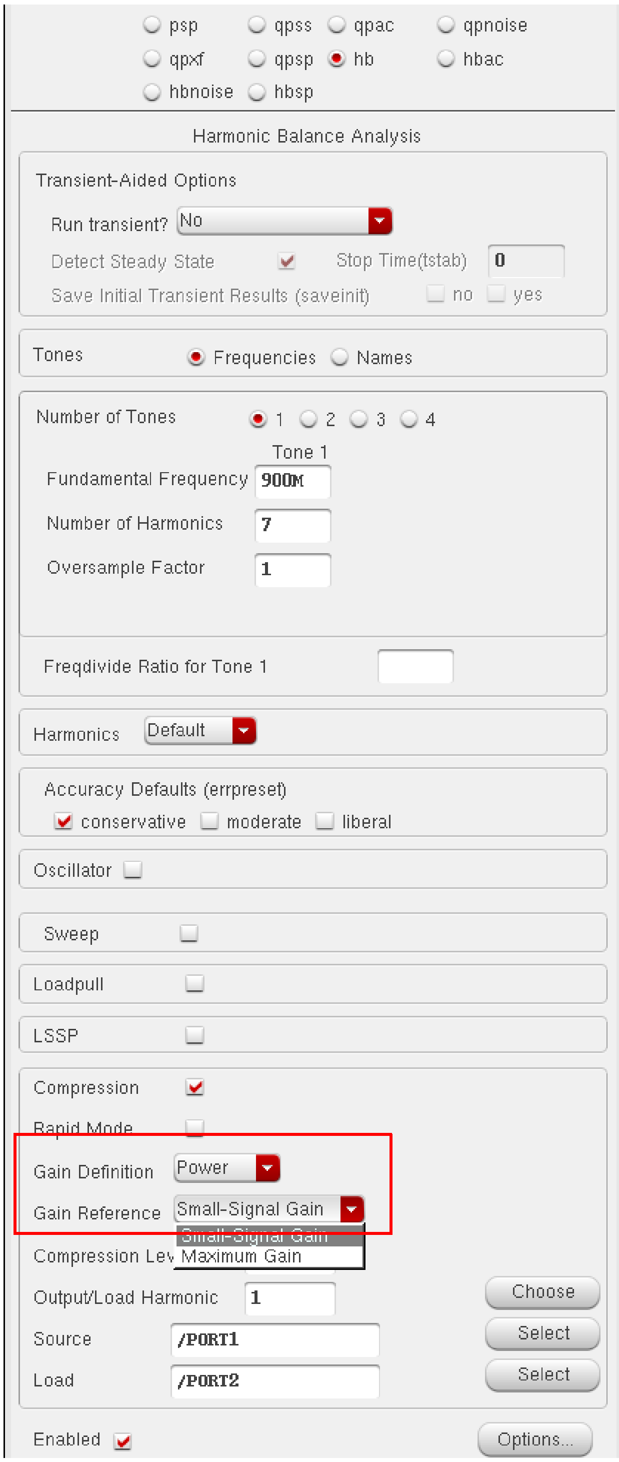
To enable compression analysis:

1. Select the check box to the right of *Compression* in the *Choosing Analyses* form.
2. If desired, select *Rapid Mode*. In this mode you must specify the approximate input referred compression level. This mode is most useful for Monte Carlo and Corners runs where there are many simulations.

3. Select *Power* or *Voltage* from the *Gain Definition* drop-down list.



4. Select either the *Small-Signal Gain* or the *Maximum Gain* as the *Gain Reference*.

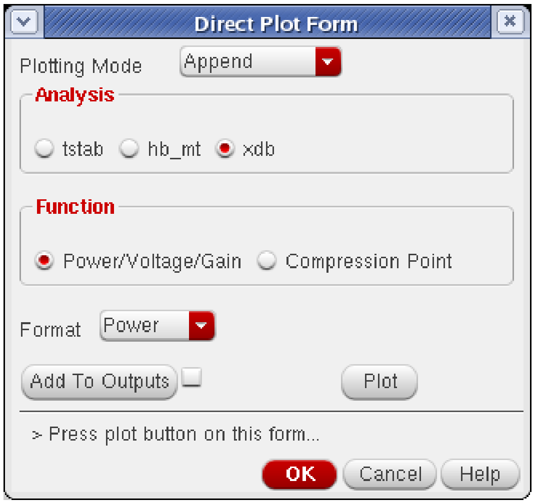


5. Set the compression level. This can be a list of values separated by a space for normal mode, and must be a single value for Rapid Mode.Also set the harmonic number of the

output. To view the harmonic frequencies, click *Choose* to the right of the *Output/Load Harmonic* field.

6. Select the output source and output load and run the simulation.

7. Open the *Direct Plot Form*. To plot the compression results, select *xdb*.

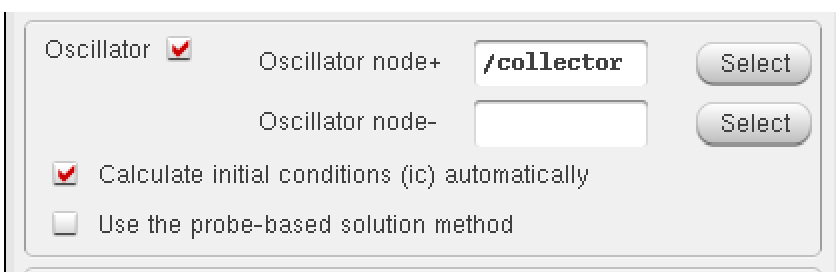


Oscillator Additions

In the *Choosing Analyses* form, select `Names` or `Frequencies`.

Tones = Frequencies

1. If you have an oscillator, select *Oscillator*.

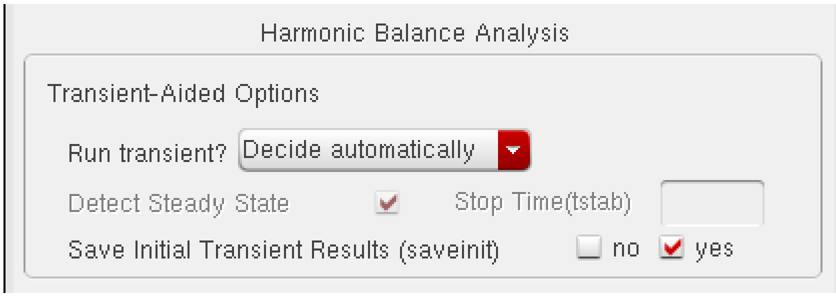


Tone 1 will automatically change to *osc!*.

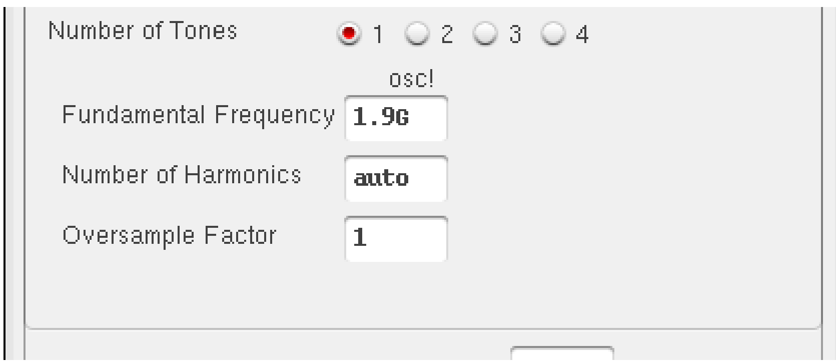
The oscillator node location in the circuit only needs to have a signal on it.

2. Select *Decide automatically* from the *Run transient?* drop-down list. This will cause an estimate of the oscillator frequency and amplitude to be run, then the transient runs in

the tstab interval, but only to the point of steady-state. Once steady-state is reached, a Fourier transform is calculated, and the frequency domain iterations begin.



3. Set the frequencies, number of harmonics, and oversample factor.



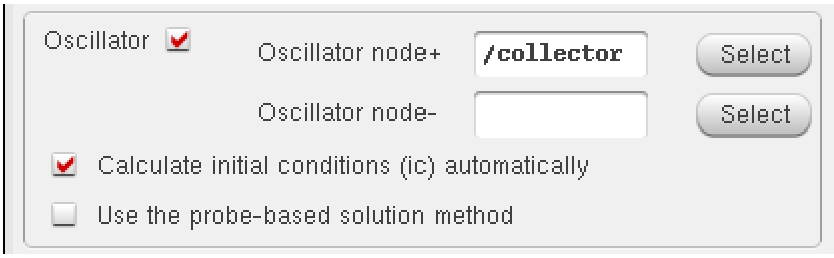
The *Fundamental Frequency* field should have a value between 0.5 and 1.5 times the actual frequency of oscillation. If *Run transient?* is set to *Decide automatically* or *Yes*, *auto* is set for the number of harmonics. At the end of the transient in the tstab interval, the number of harmonics will be set automatically based on the waveforms in the tstab interval.

4. Select *conservative* for oscillators.

Note: You should always select *conservative* for oscillators.

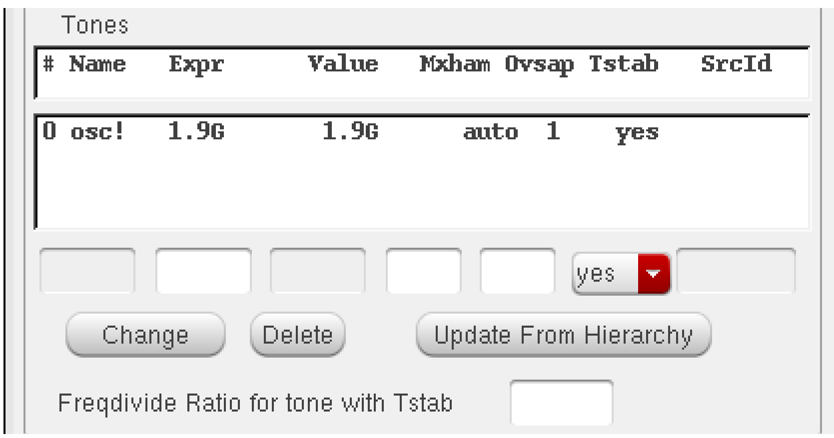
Tones = Names

1. If you have an oscillator, select *Oscillator*. The system will display a popup window that reminds you to provide an estimate for the oscillation field.



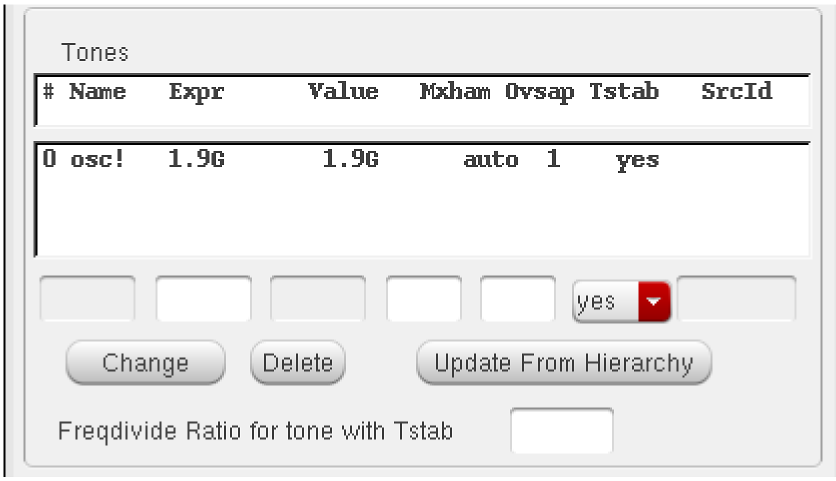
The oscillator node location in the circuit only needs to have a signal on it.

2. Specify a value in the *Expr* field that is between 0.5 and 1.5 times the actual frequency of oscillation.



3. Select *Decide Automatically* from the *Run transient?* drop-down list. This will cause tstab to be run only until steady-state is reached, where a Fourier transform is calculated, and the frequency domain iterations in harmonic balance begin.

4. *Decide automatically* also evaluates the Fourier transform at the end of the tstab interval, and sets the harmonics based on the actual spectrum. The word *auto* appears in the *Mxharm* field.



5. Select *conservative* accuracy for oscillators.

Note: You should always select *conservative* for oscillators.

Oscillator Tuning Mode

Oscillator tuning mode is provided to tune the oscillator to a desired frequency and then run the selected small-signal analyses. This is useful for sweeps and for Monte Carlo simulations. To run an oscillator tuning analysis, do the following:

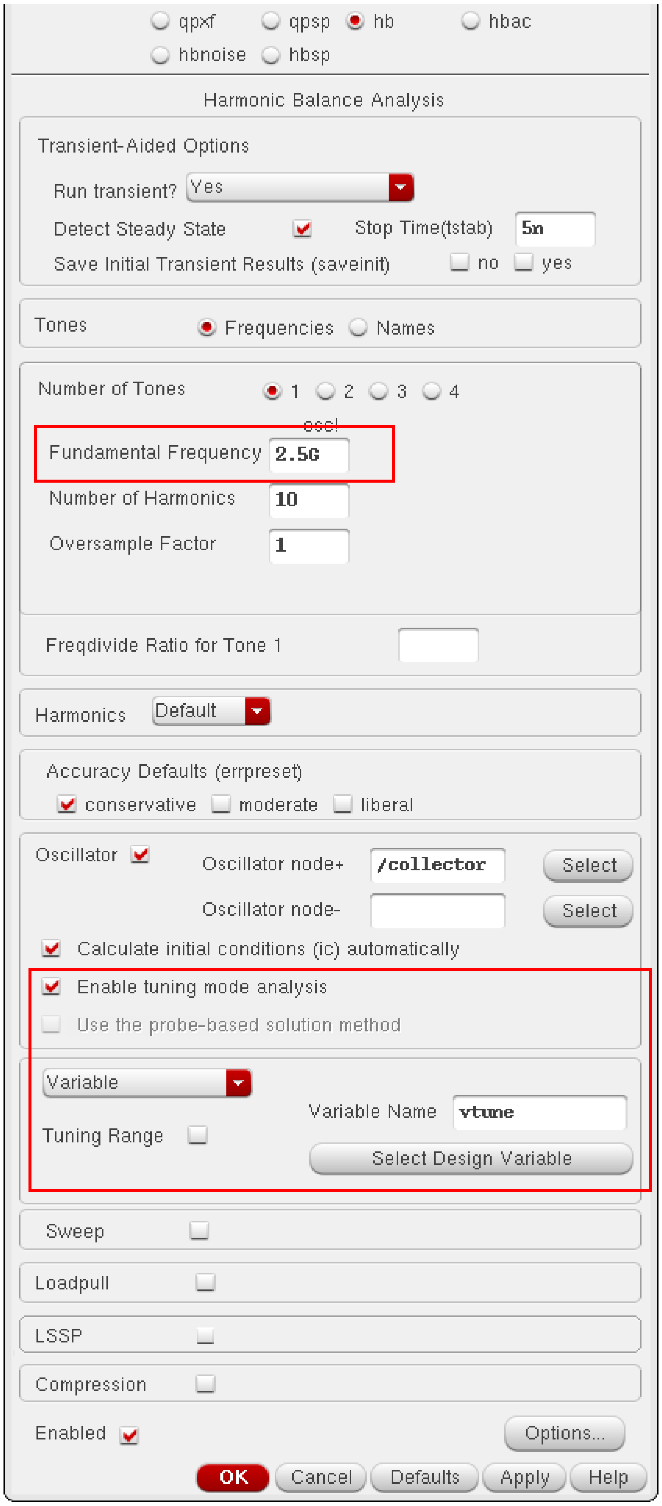
1. Open an oscillator circuit, and start ADE. Open the hb Choosing Analyses form, and select hb analysis.

2. Fill out the form as usual, except for the following:

   a. In the *Fundamental Frequency* field, type the desired frequency target.

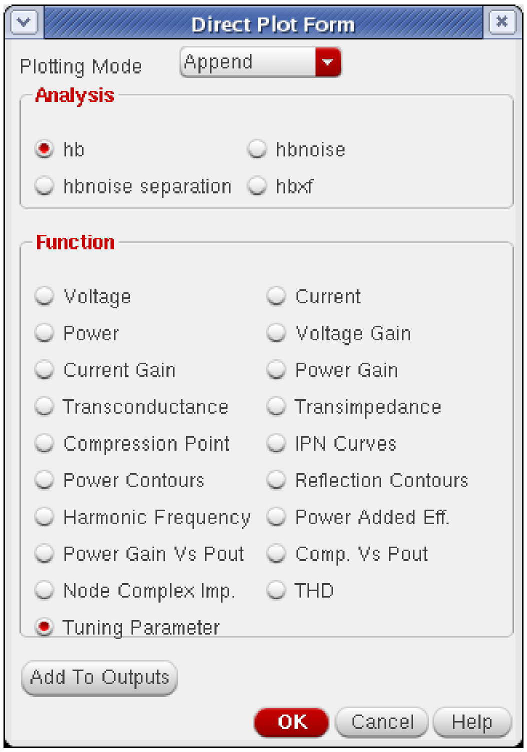
   b. Click the check box next to *Enable tuning mode analysis*.

3. Select whether you want to tune a variable, temperature, or a component parameter. The example below shows a variable.



4. Enable the desired small-signal analyses.

5. Run the analyses. When the simulation completes, open the *Direct Plot Form* from ADE.

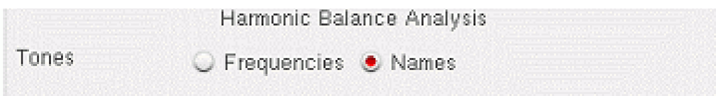


Probe-Based Method

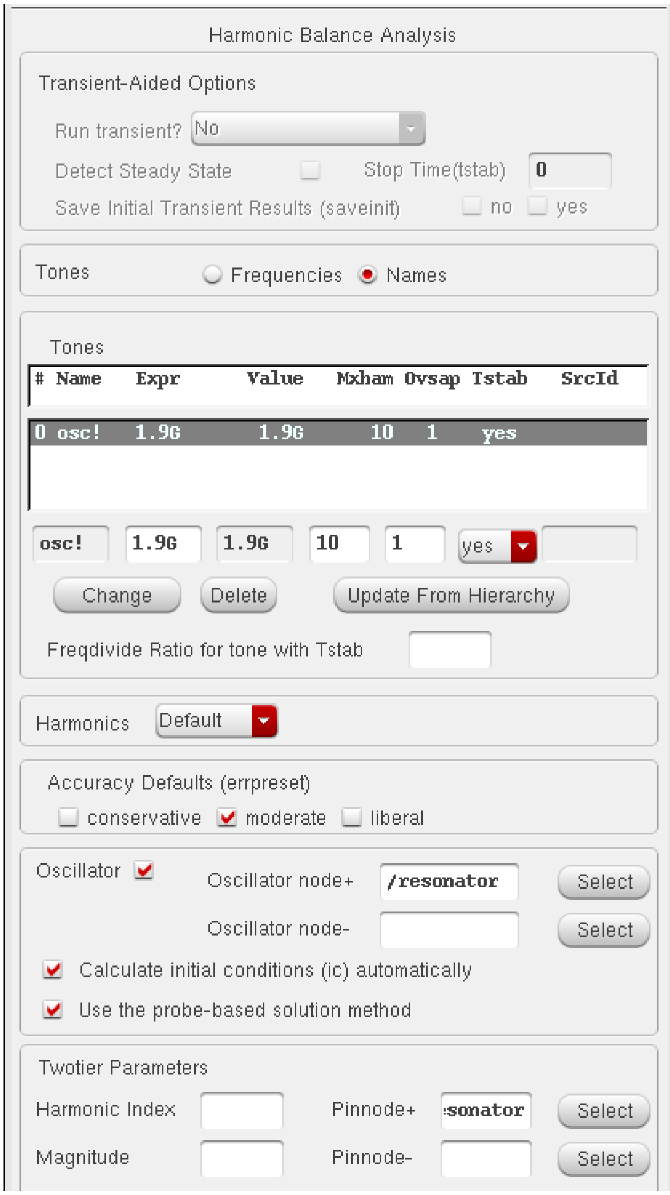
Note: Earlier, this used to be called the two-tier method.

In the *Choosing Analyses* form:

1. Choose *Frequencies* or *Names*.

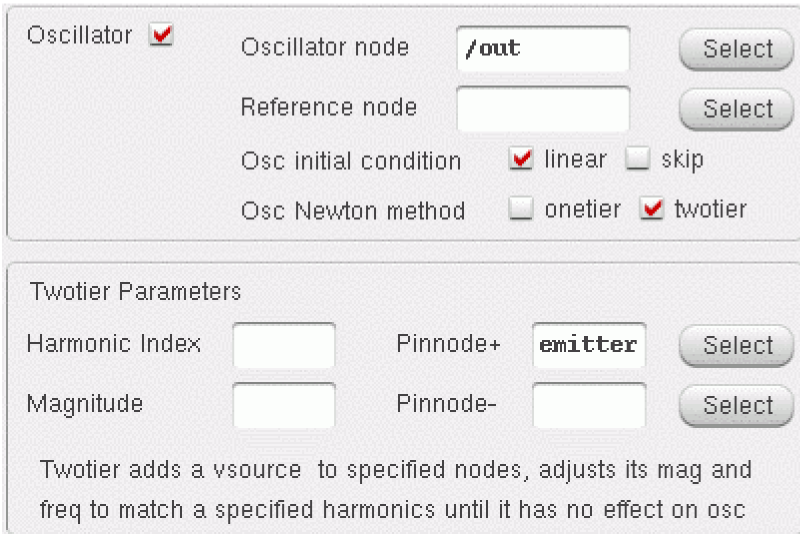


2. Select *Use the probe-based solution method.*



Note: When the probe based method is used, the only choice is to set the *Run transient?* option to *No*.

3. Select an oscillator node that is inside the feedback system for *Pinnode+*.



4. If you have a differential circuit, select the differential node at the mirror location in the feedback system for *Pinnode-*.

5. Leave the *Harmonic Index* and *Magnitude* fields blank.

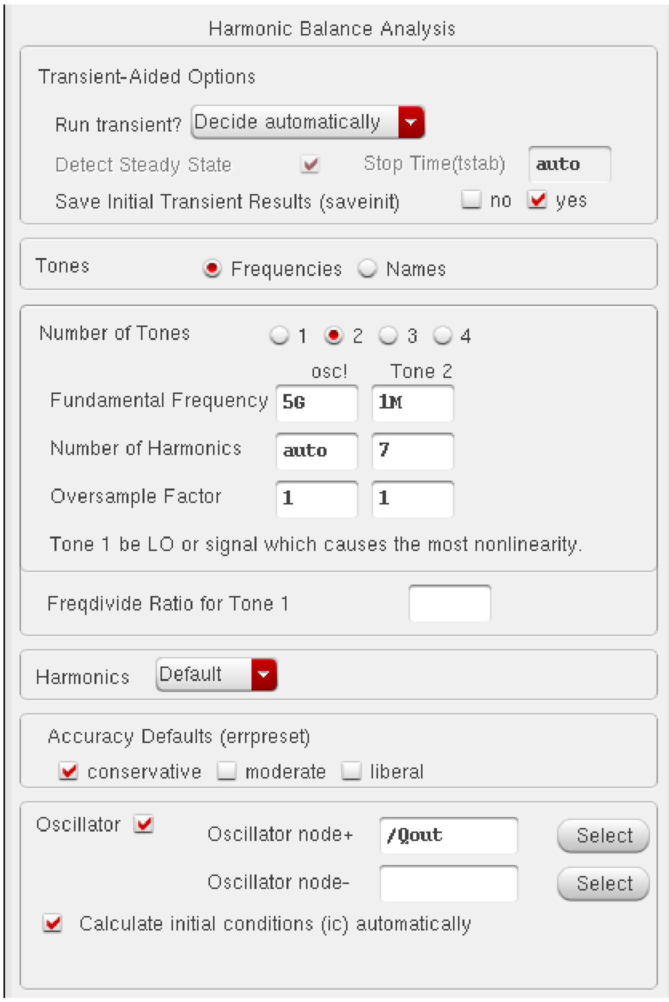
Semi-Autonomous

Semi-Autonomous allows an oscillator and a periodic source to be simulated at the same time.

In the *Choosing Analyses* form, select *Names* or *Frequencies*.

Frequencies

1. Select *Oscillator*.



2. Select the number of tones as desired.

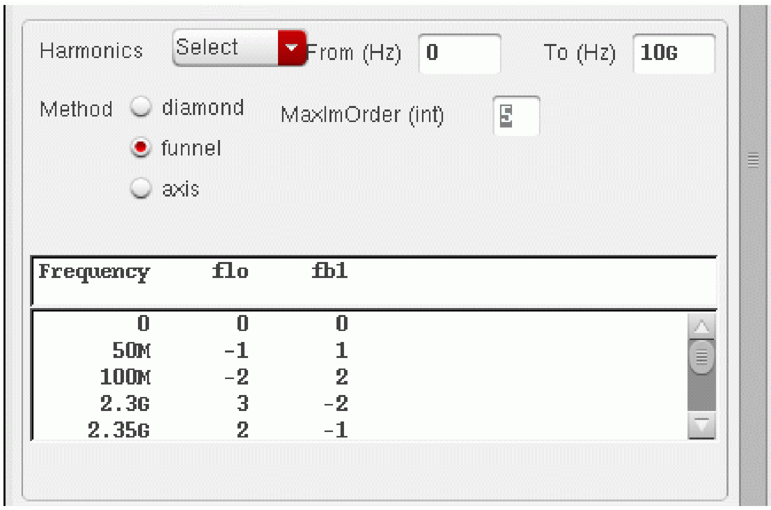
3. Type an estimate for the *Fundamental Frequency* in the *osc!* column.

4. Type the frequencies for the periodic sources in the other columns.

5. Set the number of harmonics and the oversample factor as required. If *Run transient?* is set to *Decide automatically* or *Yes*, then *auto* is entered for the number of harmonics in the first tone only. This can be set manually, if desired. The number of harmonics for the rest of the tones must be set manually.

6. If there is a frequency divider after the oscillator in the circuit, set the divider ratio in the *Freqdivide Ratio for Tone 1* field. Remember to set oversample factor and enough harmonics when this is set.

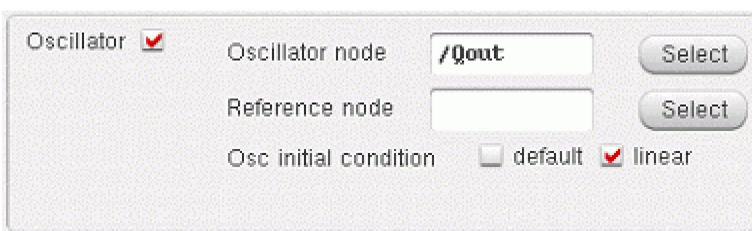
7. If you want to reduce the number of harmonics set, choose *Select* from the *Harmonics* drop-down list and set *MaximOrder*.



8. Select *conservative*.

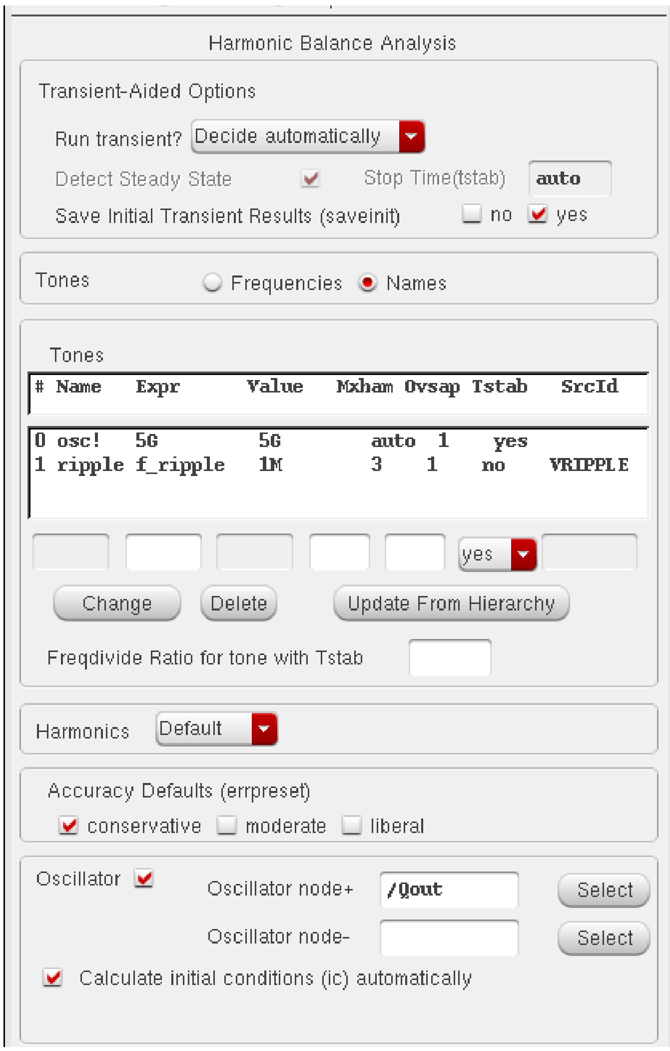
9. If you want to view the startup waveform, select *yes* for *Save Initial Transient Results (saveinit)*.

10. Specify the oscillator nodes in the circuit.

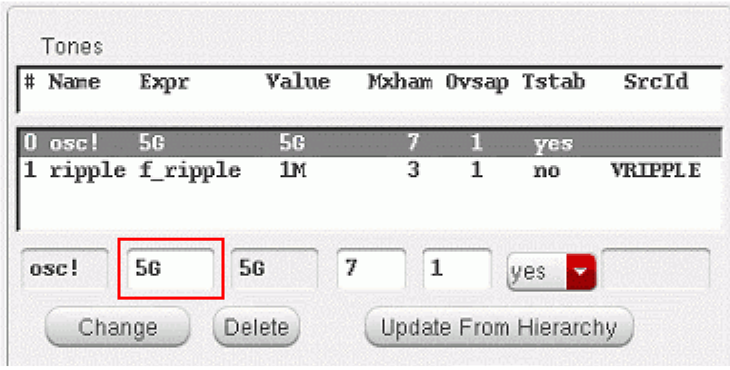


Names

1. Select *Oscillator*. The system will display a popup window that reminds you to supply an estimate for the *Expr* field.



2. Specify the frequency in the *Expr* field.



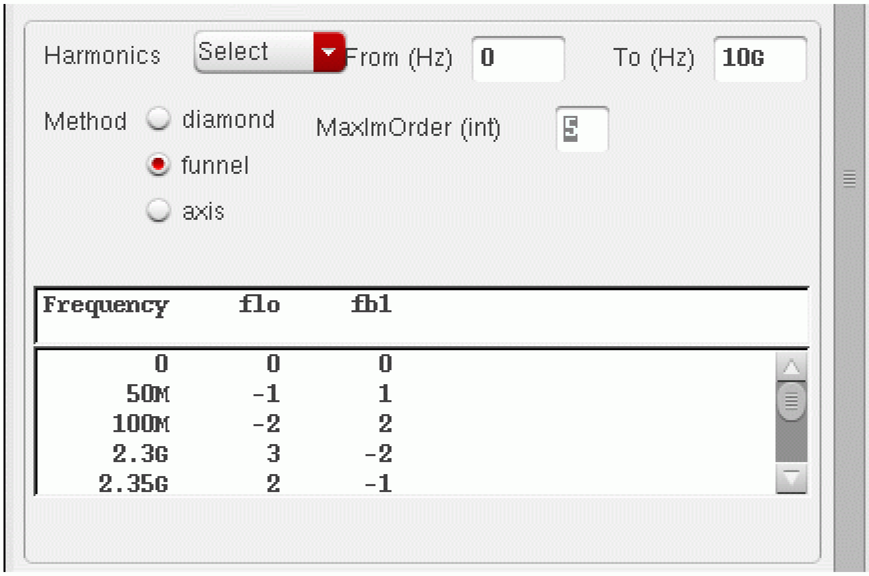
3. Select Decide automatically from the *Run Transient?* drop-down list. This will cause Spectre to run a transient in the tstab interval until steady-state is reached. Then, a Fourier transform is calculated, and the frequency domain iterations begin.

4. Set the harmonics and the oversample factor as required. If *Run transient?* is set to *Decide automatically* or *Yes*, then *auto* is entered for the number of harmonics in the first tone only. This can be set manually, if desired. The number of harmonics for the rest of the tones must be set manually.

5. Click *Change, or* select another field, or use the <*Tab*> key.

6. If there is a frequency divider after the oscillator in the circuit, set the divider ratio in the *Freqdivide Ratio for Tone 1* field. Remember to set oversample factor and enough harmonics when this is set.

7. If you want to reduce the number of harmonics set, choose *Select* from the *Harmonics* drop-down list.



8. Select *Conservative*.
9. If you want to view the startup waveform, select *yes* for *Save Initial Transient Results (saveinit)*.
10. Specify the oscillator nodes in the circuit. This node can be any node in the circuit where the oscillations are present.

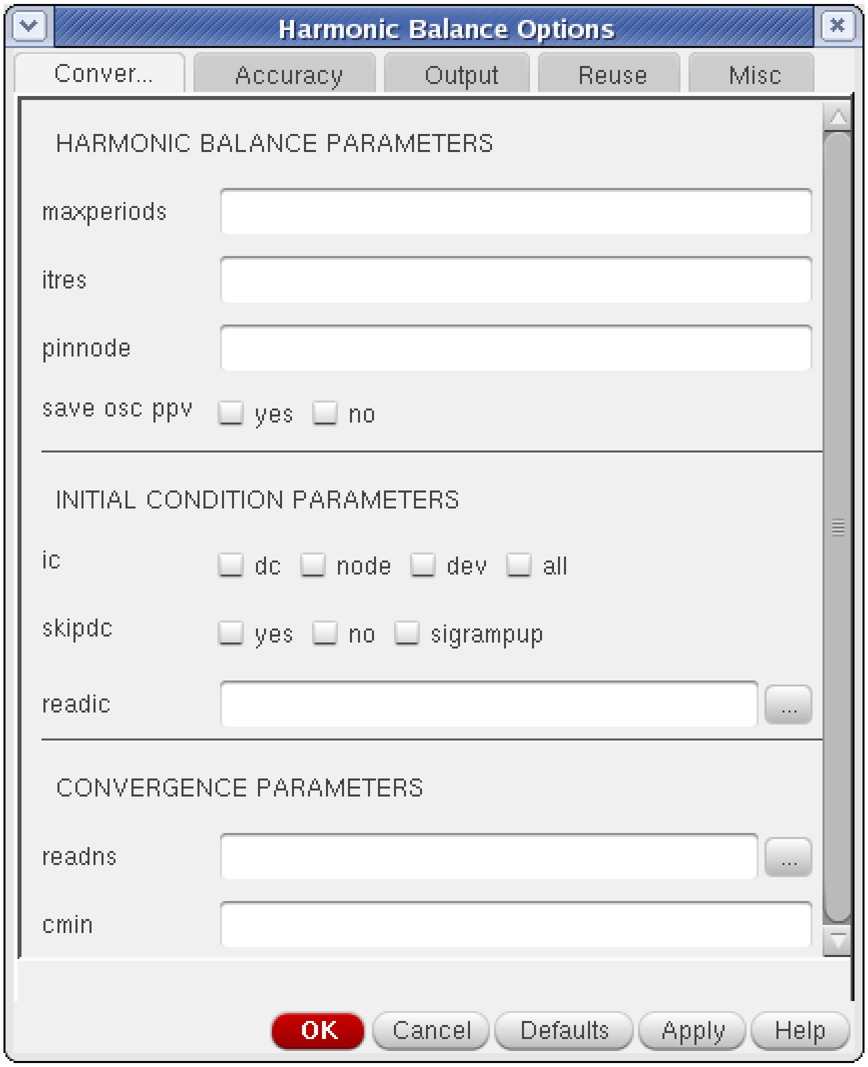
Commonly Used Harmonic Balance Options (Driven Circuits and Oscillators)

The *Harmonic Balance Options* form deliberately omits uncommonly used options. In the unlikely case that you need options that are available in pss-harmonic balance, you will need to determine the `option name=value` pairs and enter this information in the *additionalParams* field in the *Misc* tab. The easiest way to get these `option name=value` pairs is to set the option in a dummy pss analysis, use ADE to create the netlist, and then highlight the text in the netlist reader. This text can be entered in the *additionalParams* field by clicking the center mouse button.

The options are divided into five tabs: *Convergence*, *Accuracy*, *Output*, *Reuse*, and *Misc* (miscellaneous).

The options are shown in the order that they appear in the *Harmonic Balance Options* form in ADE.

Convergence tab

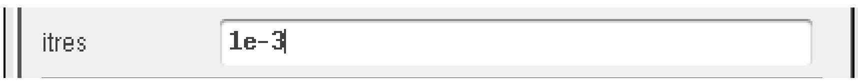


Maxperiods

This specifies the maximum number of iterations for the hb algorithm. The default is 100. If this option is set less than 100, it will be ignored.

Itres

This option specifies the precision for the solution of the first iteration in harmonic balance. For subsequent iterations, the precision is increased so that when the iterations end, full precision is used. The default is 0.1, which allows up to 10% error in the first iteration. If you have a linear or near linear circuit like an LNA, lower itres to the 1e-2 to 1e-4 range so that the solution for the first iteration is more precise. Fewer iterations will be required for the solution to converge.



pinnode

This option is only available for oscillators. Type the pathname to the pinnode in this field. and example is I3.resonator. Do not use a slash (/) at the beginning of the name.

save osc ppv

This option is no longer used.

Ic

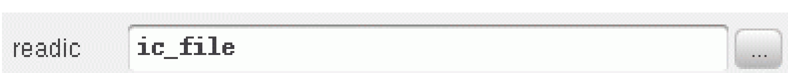
Initial conditions can be specified graphically by selecting *Simulation - Convergence Aids - Initial Condition* in the ADE. Initial conditions can also be specified from a file using the *readic* property. Capacitors and inductors have the initial condition properties in the property list for the component. For capacitors, this is an initial voltage across the capacitor, and for inductors, it is an initial current in the inductor. The default is to observe all the initial conditions in the DC analysis that is used as the time-zero timepoint in the tstab interval. The initial conditions force a voltage or current to be present in the time-zero solution. The initial conditions are released for the rest of the transient simulation in the tstab interval. The *ic* option controls the initial conditions that should be observed in the time-zero timepoint. *all* is the default. *dev* means that only the initial conditions on capacitors and inductors are observed. *node* means that only the initial conditions on a node are observed. *dc* means that no initial conditions are observed. The example below shows the ic option set to *node*.

Skipdc

In some cases, the time-zero timepoint DC analysis does not converge. Instead of stopping the simulation, *skipdc* allows the tstab simulation to continue using an assumed solution for the time-zero timepoint. The default is *no* and a DC analysis is run to get the initial timepoint. *yes* means skip the DC solution, and proceed directly to the tstab simulation. All the nodes with initial conditions specified start at the initial condition value. Nodes with batteries start at the battery voltage. Nodes with no initial conditions start at zero volts. For `skipdc=yes`, the signal sources start as specified immediately in the tstab simulation. *sigrampup* uses the same assumptions for the starting voltages as *yes* does, but the start time is set to negative one tenth of the stop time for the tstab interval. At this time, the signal source time-varying part starts at zero and linearly ramps up to the full value at time = zero. After time = zero, the sources have the full amplitude time-varying part. The example below shows the *skipdc* option set to *sigrampup*.

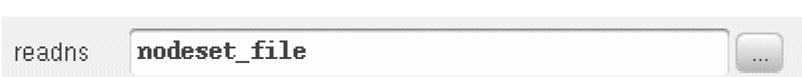
Readic (Shooting and Harmonic Balance)

This specifies an ascii file that contains two columns to be read as initial conditions. The first column is the node name. The second column is the voltage value. If the entry in the *readic* field does not start with / (slash), the entry is located in the netlist directory. To find the netlist directory, select *Setup- Simulator/Directory/Host* in ADE. Look in the *Project Directory* field for the location of the simulation directory. Navigate to that directory and then to the `<Circuit Name>/spectre/<schematic or config>/netlist` directory. You can also click (...) and browse to the directory. An example is shown below.



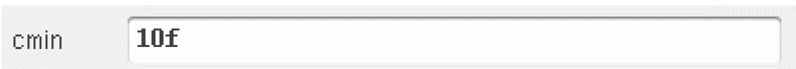
Readns

This specifies an ascii file with the same format as an ic file that is used as nodesets for the time-zero DC solution. Nodesets do not force a voltage to be held for the time-zero solution. Instead, they are a way of speeding up the time-zero calculation. As a suggestion, set the *write* option and the *readns* option to the same filename. The *write* option writes the time-zero solution to a file. When this is used as a starting point, many fewer iterations are needed for the time-zero point to converge. An example is shown below.



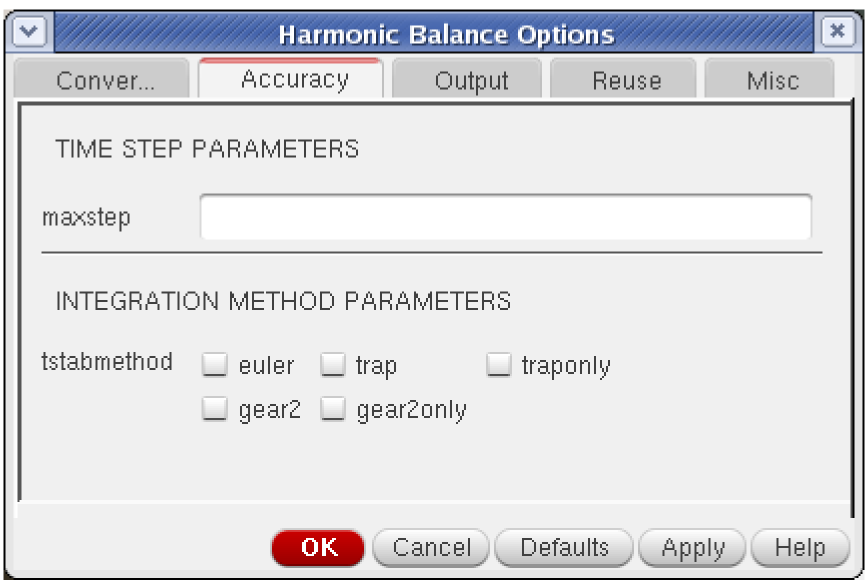
Cmin

cmin can be used to improve convergence in the tstab and shooting intervals for shooting and for the tstab interval for harmonic balance. If a value is set for *cmin*, a capacitor with this value is added to every node with the other terminal of the capacitor connected to the global ground node. If a 10f to 50f capacitor is added, this prevents instantaneous changes from occurring from timepoint to timepoint, thus improving the convergence at the cost of adding non-physical capacitors to the circuit. An example with 10 femtoFarads is shown below. Note that if 10 is entered, a 10 Farad capacitor is added from every node to ground. Remember the multiplier in the entry. An example is shown below.



Accuracy Tab

The *Accuracy* tab is split into two categories, *TIME STEP PARAMETERS* and *INTEGRATION METHOD PARAMETERS*.



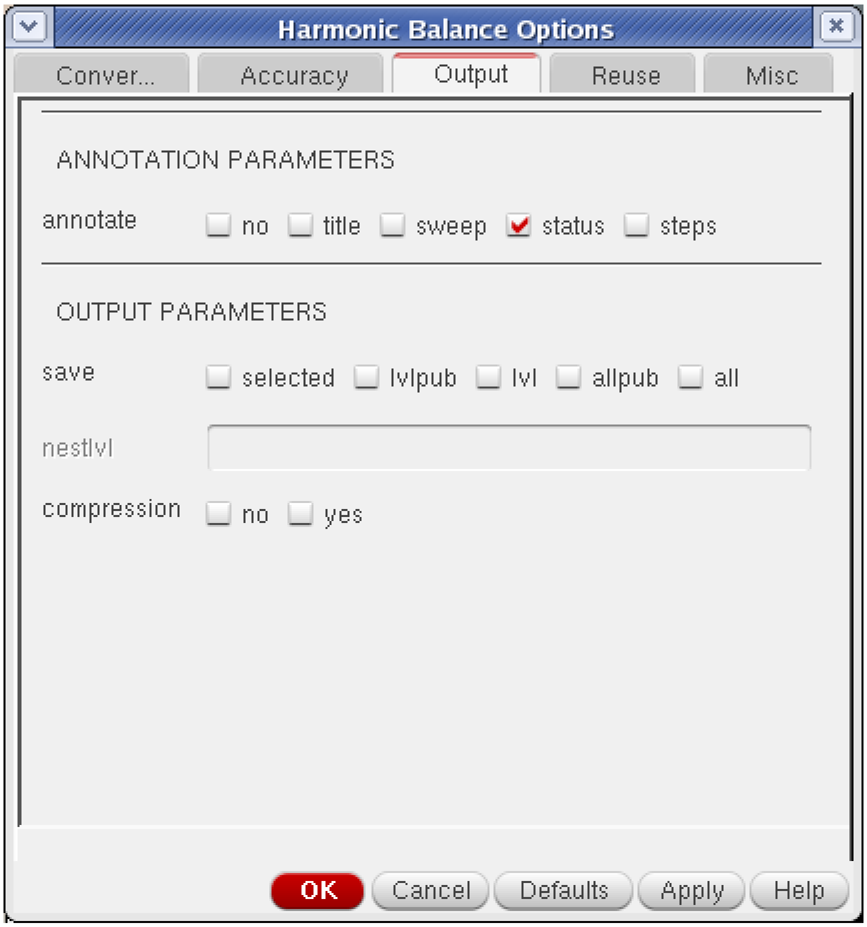
Maxstep

This is the maximum allowable timestep for the transient analysis in the tstab interval.

Tstabmethod

This option controls the integration method in the tstab interval. The default value is determined by the setting of *errpreset* and whether the circuit is driven or is an oscillator. For *moderate* and *conservative* with a driven circuit, the default is *gear2only*. For *liberal* it is *traponly*. Generally, *gear2only* is preferred because of the absence of trapezoidal ringing inherent in the *trap* (trapezoidal) method. The default for *tstabmethod* is *traponly* for oscillators. *traponly* is used for oscillators because with the defaults, the tstab interval uses relatively loose convergence options in order to speed up the simulation. This causes relatively long timesteps which for the *gear2* method cause noticeable numerical damping. *traponly* does not numerically damp, therefore, it provides a better estimate of the oscillating frequency and amplitude. An example with *tstabmethod* set to *gear2only* is shown below.

Output Tab



Annotate

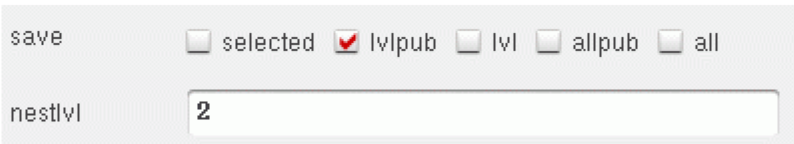
This option is currently not implemented.

Save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished by selecting *Outputs - To Be Saved - Select On Schematic* and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.

Nestlvl (Shooting and Harmonic Balance)

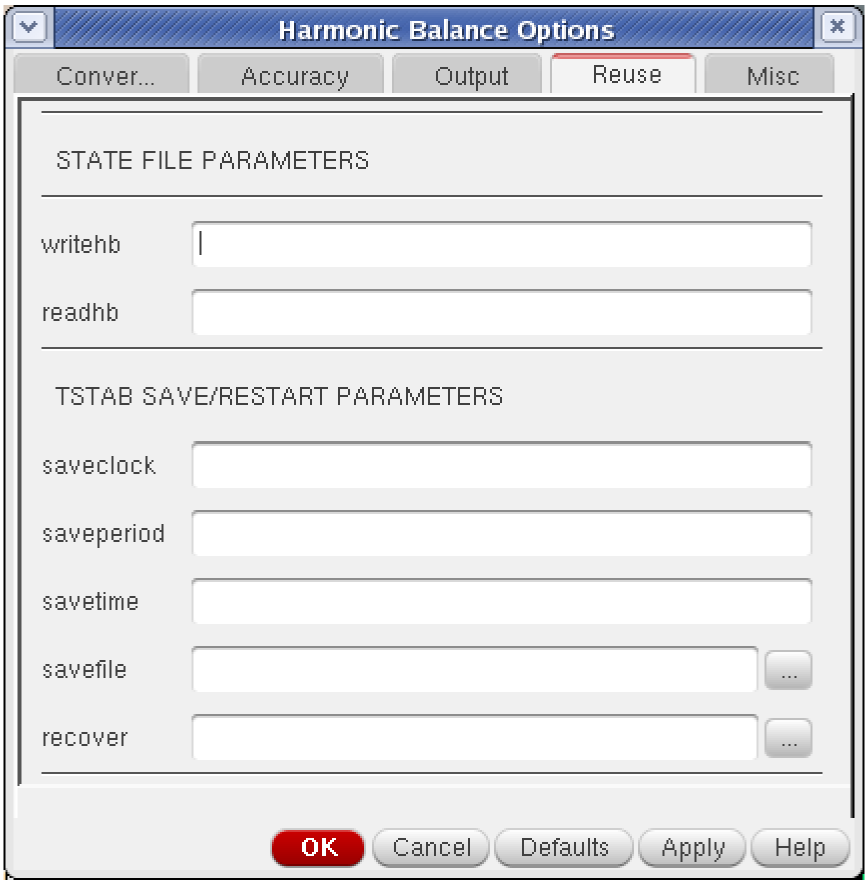
If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer. The example below shows *lvlpub* selected that saves two levels of hierarchy.



Compression (Shooting and Harmonic Balance)

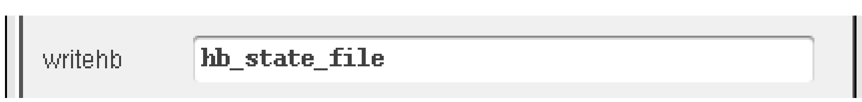
Normally, this should not be used for RF simulation. It is not digital compression. For RF simulation, where the input is sinusoidal, the size of the results file will normally double. It is useful only for circuits that are predominantly square wave.

Reuse Tab



Writehb

This option specifies that the full internal state of the harmonic balance analysis be written out to the file specified in the option. An example is shown below.



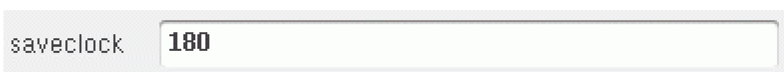
Readhb

This option specifies the file to be read in as a starting point for the next harmonic balance analysis. If nothing is changed in the circuit, then only one iteration is performed and small-signal analyses can be run with a much faster time for the hb analysis. Changes can be made to the circuit or analysis options as long as the topology of the circuit stays the same. The changes introduce a discontinuity at the beginning of the hb analysis, which might be large enough to cause hb not to converge.

An example of *readhb* is shown below.

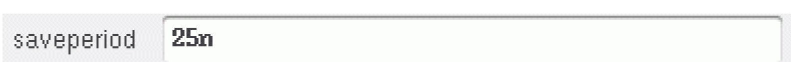
Saveclock

saveclock saves the state of the simulation in the tstab interval at the clock time interval in seconds specified by *saveclock*. When the clock time has passed in the tstab interval, the file specified by the option *savefile* is created. When subsequent clock time intervals have passed, the file is overwritten. Use only one of the save options at a time. The example below shows *saveclock* set to three minutes.



Saveperiod

saveperiod saves the state of the simulation in the tstab interval at the simulation time interval in seconds specified by *saveperiod*. When the simulation time has passed in the tstab interval, the file specified by the option savefile is created. When subsequent simulation time intervals have passed, the file is overwritten. Use only one of the save options at a time. The example below shows *saveperiod* set to 25 nanoseconds.



Savetime

savetime is a list of times in the tstab interval where the state of the simulation is written out. The list should be specified with spaces between the entries. The information is written out with the filename specified in the *savefile* option with extensions for the time after that. Use only one of the save options at a time.

The example below shows four times in the tstab interval to save.

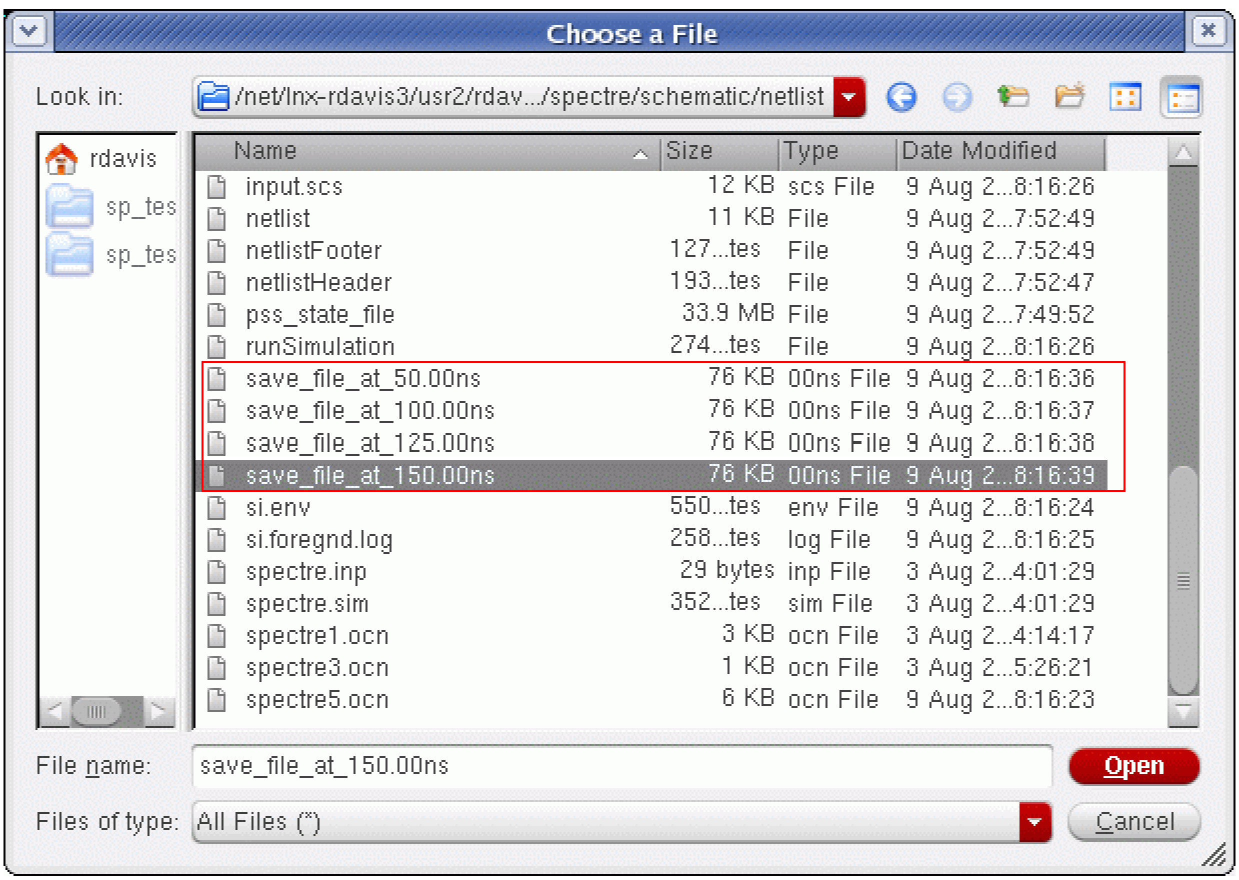
Savefile

This is the file name to write out the tstab state information to. If you do not specify a filename that starts with the slash (/) character, the file is stored in the *netlist* directory. To locate the *netlist* directory, first select *Setup - Simulator/Directory/Host* menu in ADE, and read the path to the project directory. This is the path to the *simulation* directory. In the simulation directory, navigate to the <`circuit_name`>*/spectre/<schematic or config>/netlist* directory.

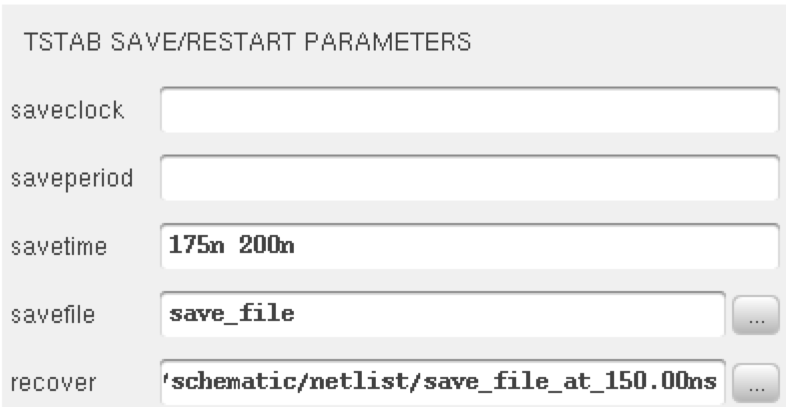
Note: Do not specify a relative pathname like *./save_file/run1*.

Recover (Shooting and Harmonic Balance)

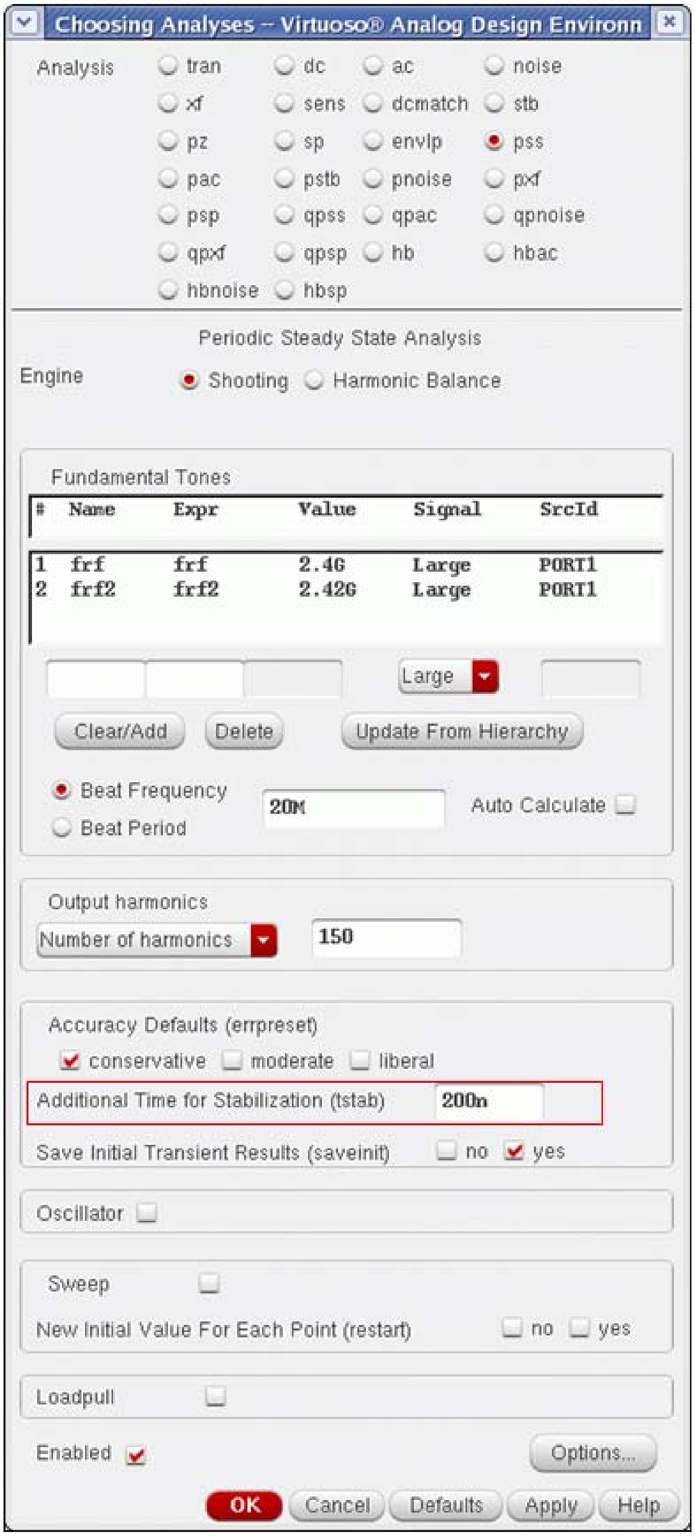
recover specifies the file name to recover the tstab simulation from. If *saveperiod* or *saveclock* are used to make the savefile, then the same name as specified in *savefile* is used. If *savetimes* has a list of times specified, then several files are created at the times specified in the list. Click (...) and browse to the netlist directory where the savefiles are shown.



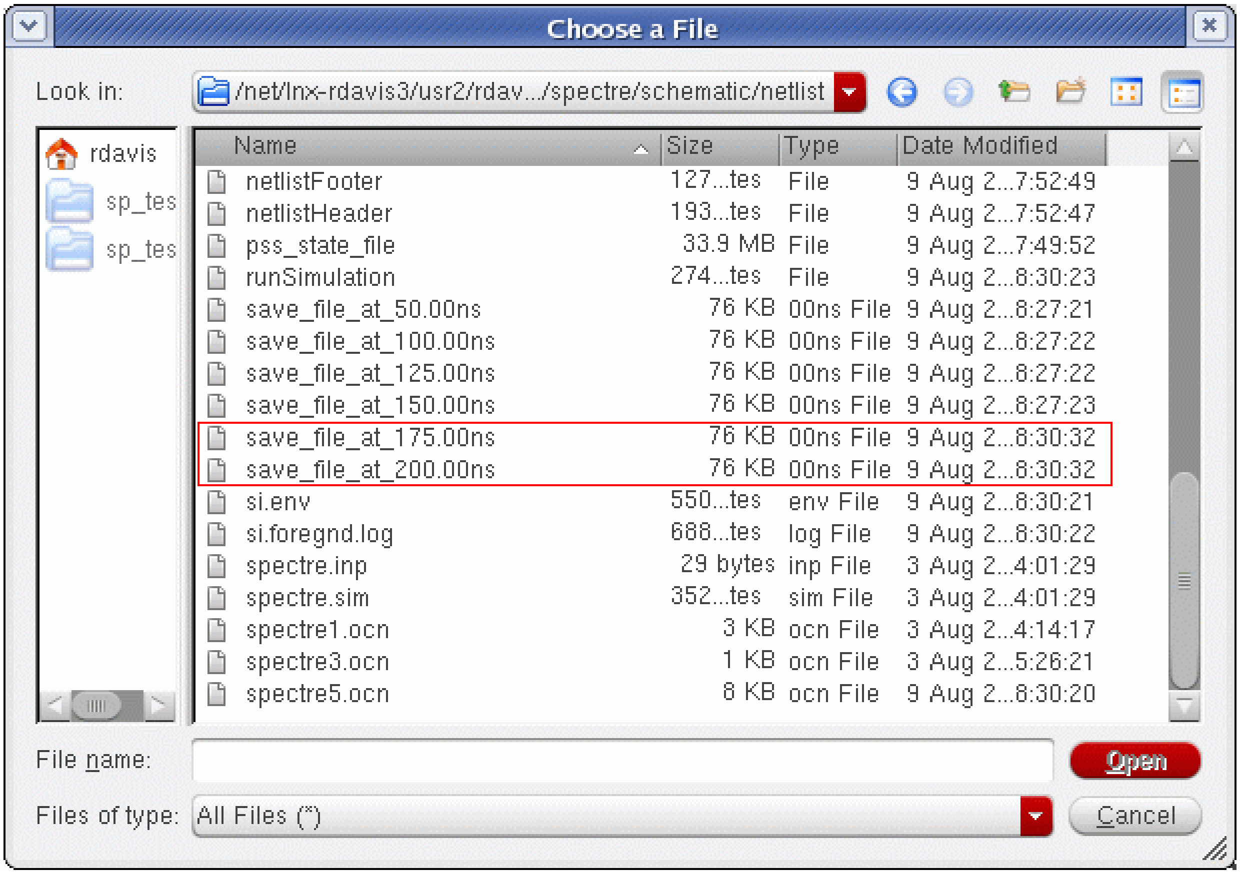
Select one from the list, and click *Open*. This adds the full path to the *recover* option, as shown below. Also note that in the example below, the *savefile* option is still set, and times after the restart times are specified. This is specifically allowed. As long as tstab is 200n or larger, the files at 175n and 200n will be added and can be reused later.



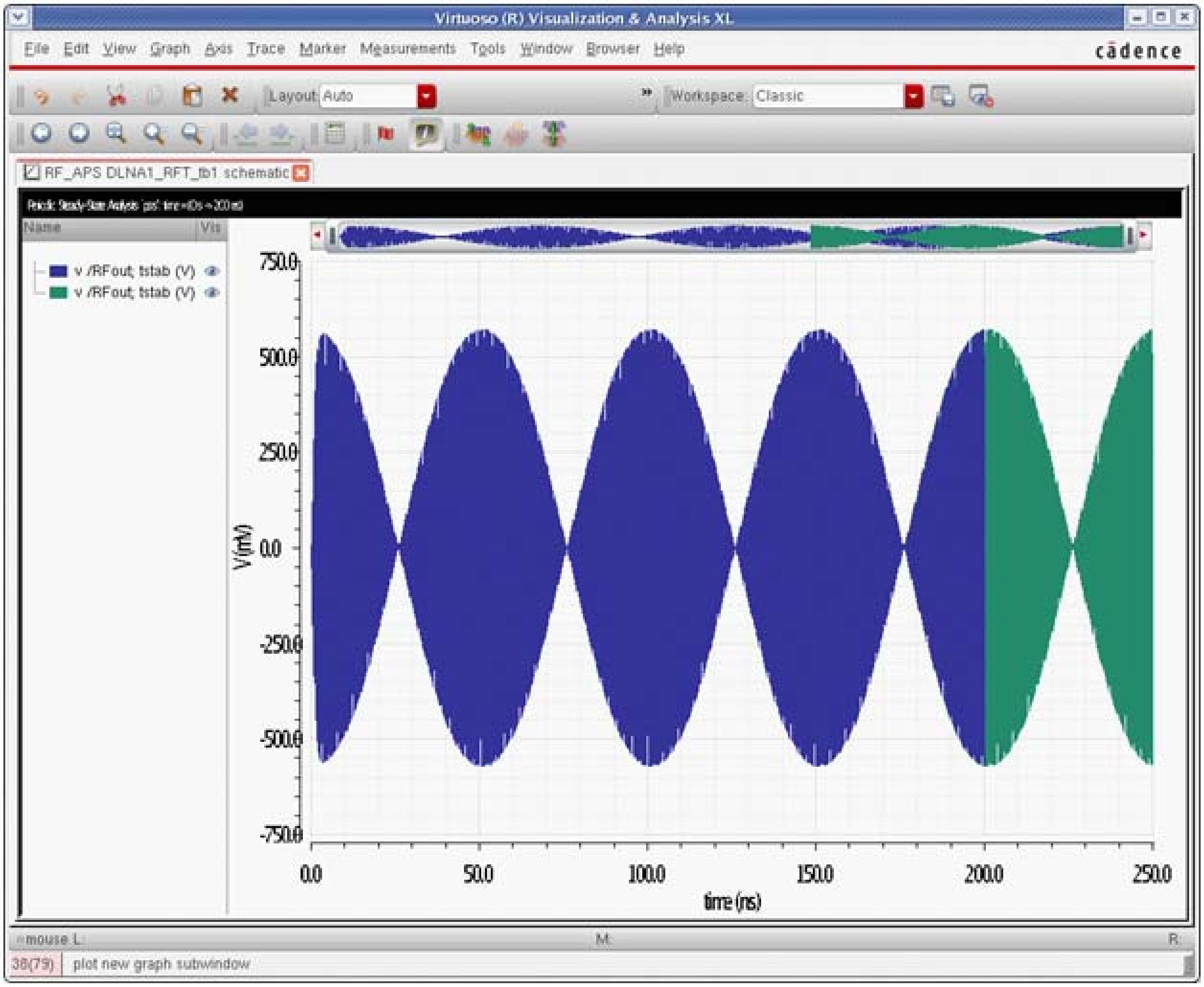
Now change tstab to a later time. In this case, 200n is set.



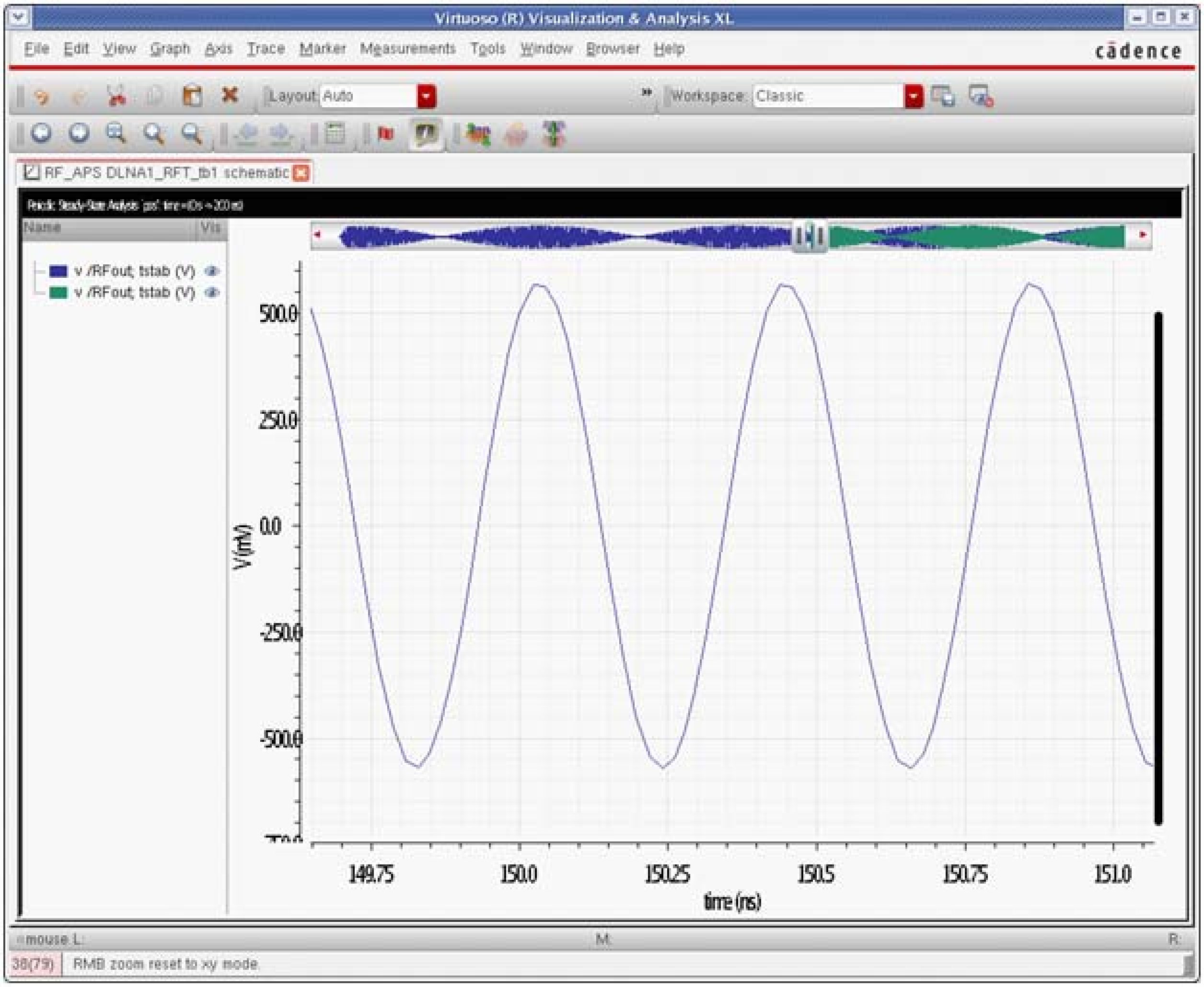
When the simulation runs, the savefiles are created, as shown below.



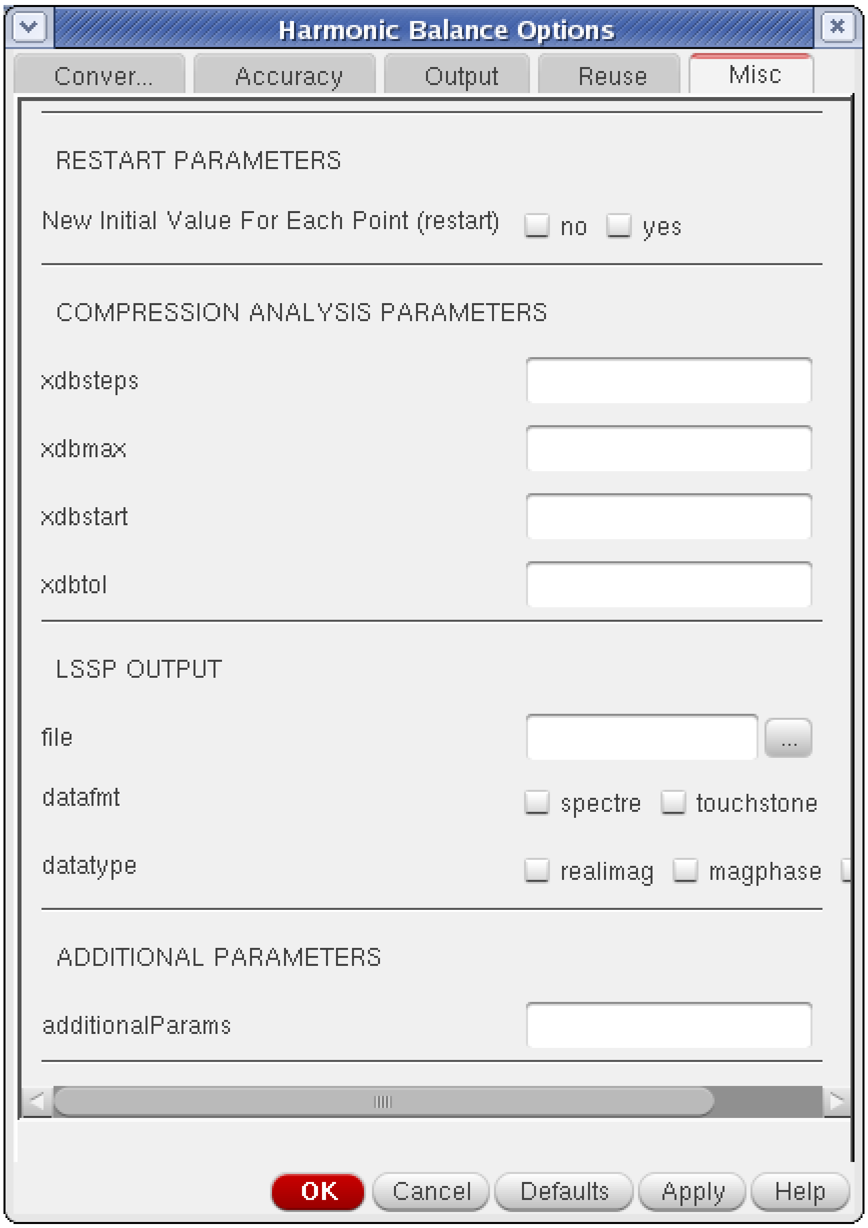
The waveform from the original tstab and the restarted tstab overlay is shown below.



Even in the zoomed-in view, the waveforms overlay, as shown below.



Misc Tab



New Initial Value For Each Point (restart)

Restart is an option that controls the behavior of the simulator in a sweep or in Monte Carlo. If restart is yes, then the simulation restarts from scratch for every sweep point. This can cause excessive runtimes and/or convergence difficulties. When restart is no, the previous member of a sweep or Monte Carlo is used as the starting point for this analysis. If this point does not converge, a variety of methods are applied automatically to try to get convergence.

These methods include parameter sweeping, backing off the source power and then ramping it back up in increments, and running a tstab simulation.

xdbsteps

Normally, this option doesn't need to be set. *xdbsteps* is an option that enables you to specify the maximum number steps for the compression point search. The simulator terminates if the value specified in the `xdbsteps` field is exceeded before the compression point in found. The default value is 100.

xdbmax

Normally, this option does not need to be set. The *xdbmax* option enables you to specify the maximum input power or voltage for the compression point search. The default is +10 dBm when *Gain Definition* is set to *Power* and 100mV when *Gain Definition* is set to *Voltage*.

xdbstart

Normally, this option does not need to be set. The *xdbstart* option enables you to set the starting power or voltage. The default is xdbmax - 50 when *Gain Definition* is set to *Power* and xdbmax/1000 when *Gain Definition* is set to *Voltage*.

xdbtol

Normally, this option does not need to be set. The *xdbtol* option controls the maximum delta between sweep values and sets the maximum delta between the actual harmonic balance last power or voltage level and the largest compression value specified for the compression analysis. The default value for *xdbtol* is 0.01. The actual tolerance between the power or voltage level and the maximum specified power level in the *Choosing Analyses* form for a normal mode compression analysis is +/- (2.5*xdbtol*maximum compression level+0.001). For a *Rapid Mode* compression measurement, the tolerance is doubled. The maximum change in power during a sweep is 10*xdbtol*(xdbmax-xdbstart). The default for *xdbtol* is 0.1, so there will be at least 10 points in the sweep. The starting change in power or voltage is half the maximum change.

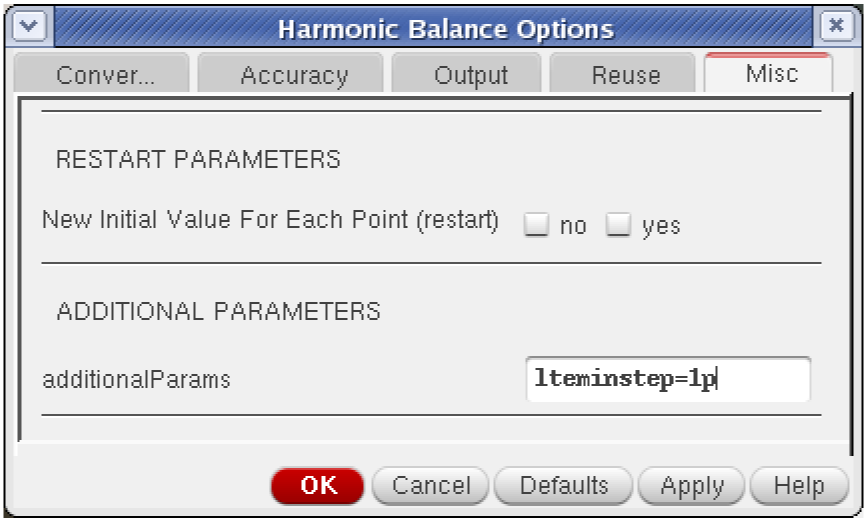
AdditionalParams

This is a field for `<option_name>=value` statements. Multiple `<option_name>=value` statements are allowed with a space between them. Generally, this is used to unlock the beta features, however, there is one case where this field might be useful for an option that is not in the GUI.

In some cases, in the tstab interval, the timestep collapses to near zero. The symptom is that the percent complete stays at the same value for many reporting intervals in the Spectre output window. In this case, there might be a discontinuity in one of the models.

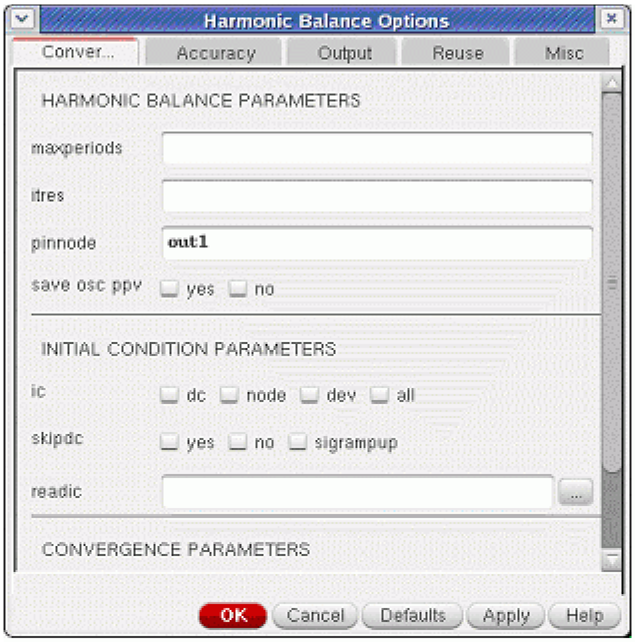
In transient simulation, two things can reduce the timestep; having too much numerical integration error or taking too many iterations at a single timestep can cause the timestep to be cut. Spectre does not report what is causing the timestep to become small. One way to eliminate the possibility of numerical integration error (which is called local truncation error in the simulator) as the cause of reducing the timestep is to set *lteratio* to 1e9. The disadvantage of this is that it disables the normal method of timestep control, therefore, *maxstep* must be set to maintain accuracy.

Another way to accomplish this is to set *lteminstep* to a value of about the stop time divided by 1e5. This allows the normal method of timestep control, but if for some reason the numerical integration error wants to cut the timestep too small, the error is ignored. *lteminstep* is the way to specify this. To do this in the ADE, type `lteminstep=<Stop_Time>/1e5`. If you use `lteratio=1e9`, or if you set *lteminstep* and you still have very small timesteps, try increasing *maxiters* in the *Convergence* tab to between 40 and 100. When *lteratio* or *lteminstep* is set and you still have timestep problems, there is likely a discontinuity in either a device model or in a Verilog-A model. An example is shown below.



Oscillator Options

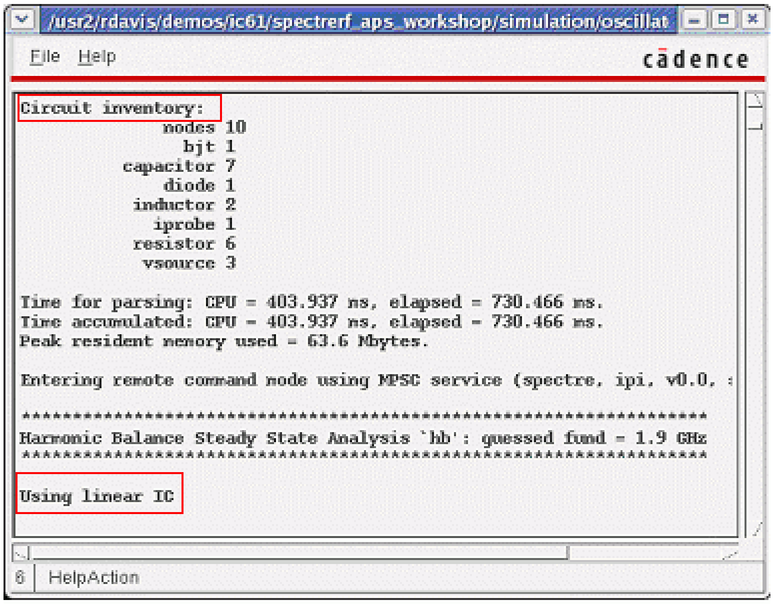
1. *pinnode* can be set manually. Do not start the name with a slash. If you have a hierarchical name, it will usually be in the form *I1.I10.out1*. If you use the probe-based method, do not specify any entry here.



2. The *save osc ppv* option is used with the PLL methodology.

Key Outputs in the Spectre Window

First, the circuit inventory is displayed. In addition, you also get a message that the linear oscic is being used. This is controlled by the *Calculate initial conditions (ic) automatically* check box.



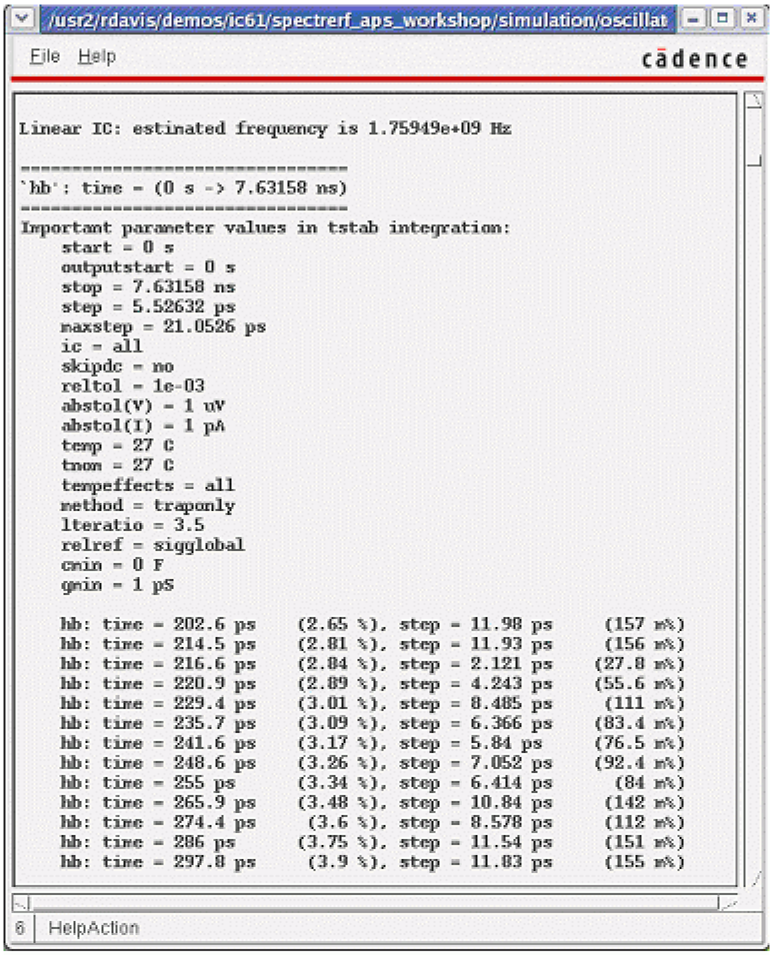
Next, you get the output that either indicates that the linear oscic worked, and the estimate of the frequency, or that it failed, and is running tstab.

The time of the tstab interval is reported. If *Run transient?* is set to *Decide automatically*, you might see one or more messages that indicate the tstab interval is being extended. This usually is only seen for oscillators.

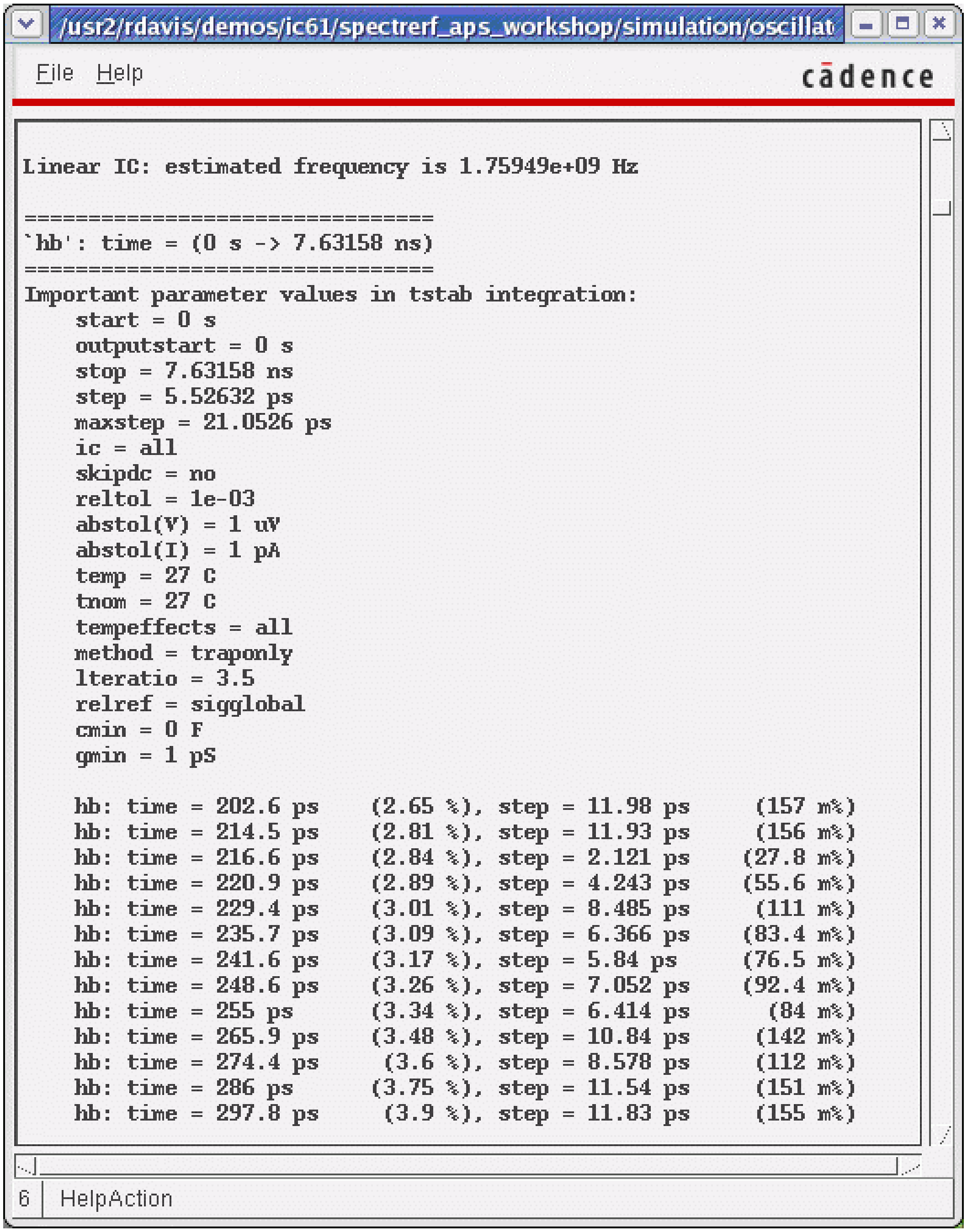
For oscillators, the tstab interval includes the time set in tstab plus five periods of the frequency estimate from linear oscic. If linear oscic is not used, it includes five periods of the frequency you estimated in the *Choosing Analyses* form.

The list of the options that are actually used for this phase is listed.

The transient output from the tstab interval is next.

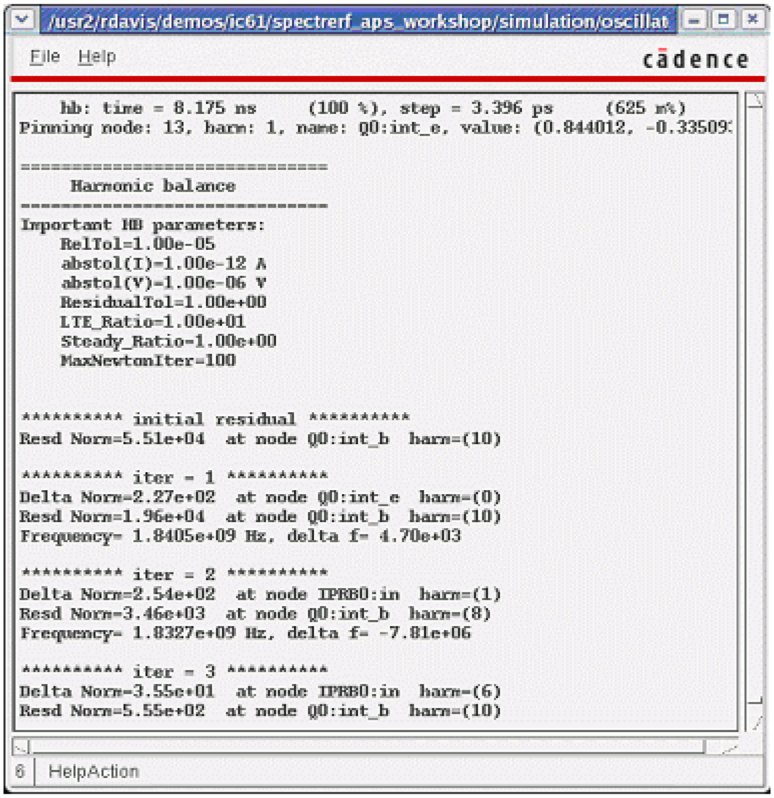


Next, one cycle of the oscillator is simulated.



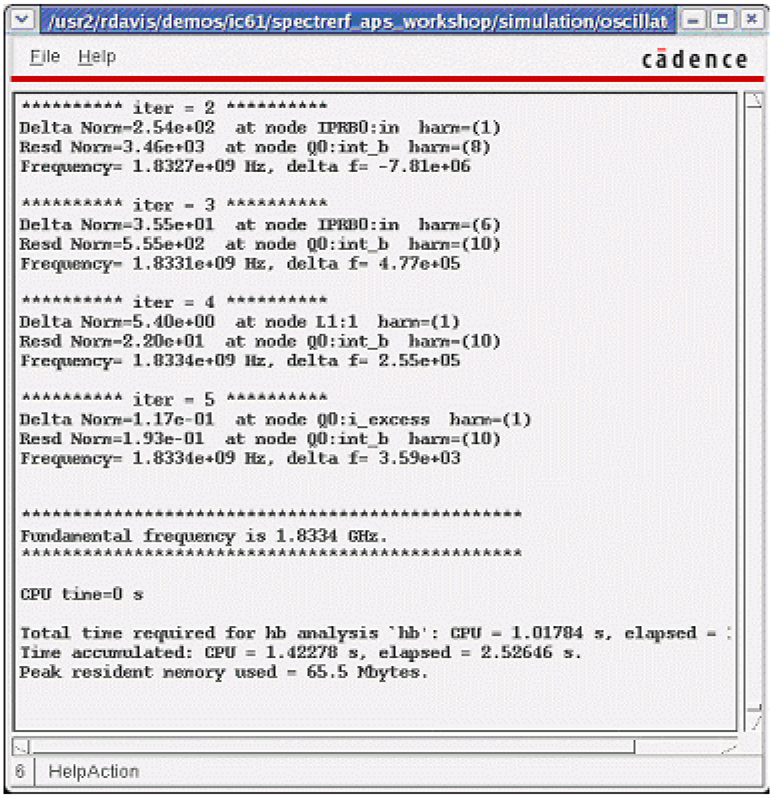
At the end of the one cycle, a Fourier transform is done to serve as the starting point in the frequency domain, and a node is picked for the pinnode.

The frequency domain iterations start.

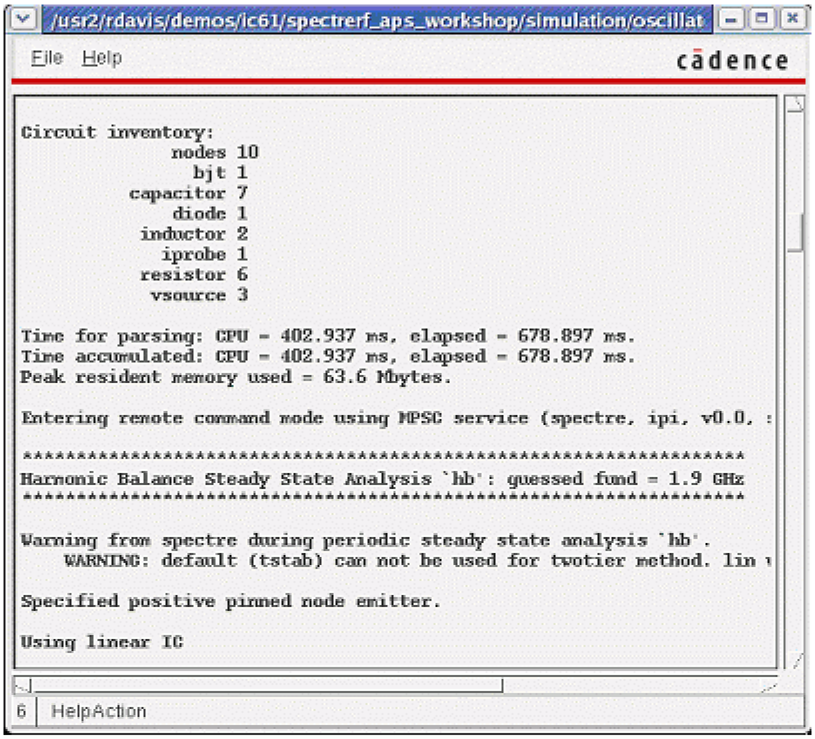


The iterations continue until the delta norm and the residual norm are both less than one.

The frequency is reported. The times and memory use are reported.



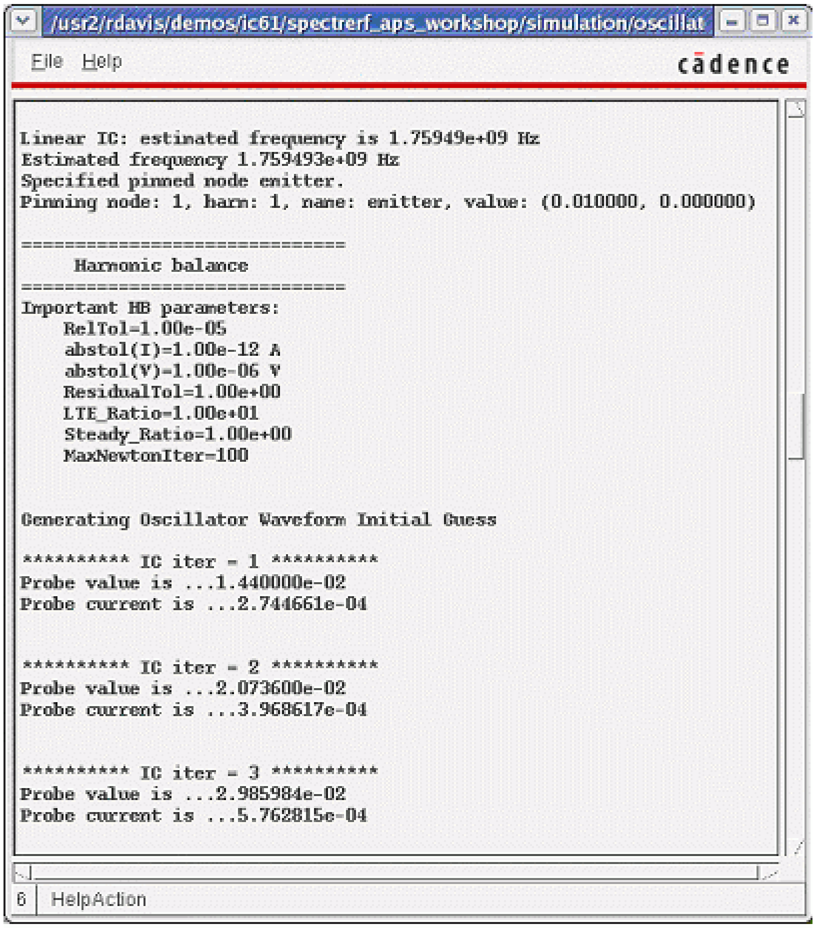
The circuit inventory, pin node, and linear oscic are reported.



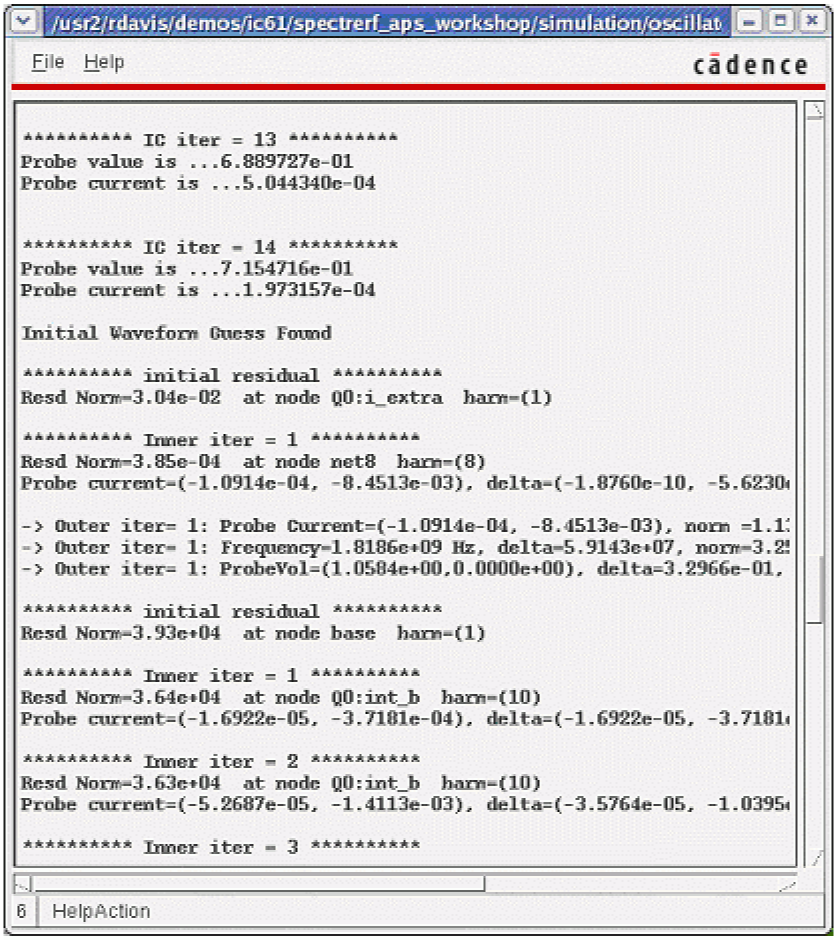
The linear oscic frequency is reported.

The options used for this part of the simulation are reported.

The probe-based method inserts a source connected between the pinnodes. The amplitude and frequency are adjusted until the current is near zero.



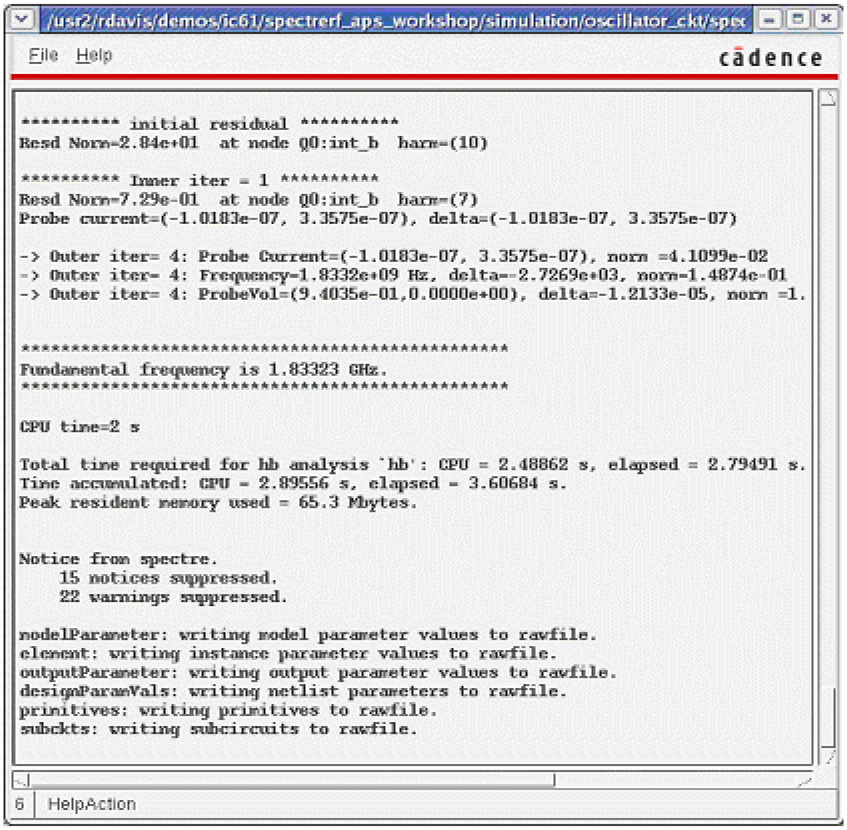
If the probe-based method is used, the initial starting point is determined. The outer loop iterates on the frequency and the inner loop iterates on the amplitude.



The iterations continue until the source current is near zero, the sum of the currents at all the nodes is near zero, and the delta between iterations is near zero.

The tolerances are set by using the errpreset setting (liberal, moderate, conservative).

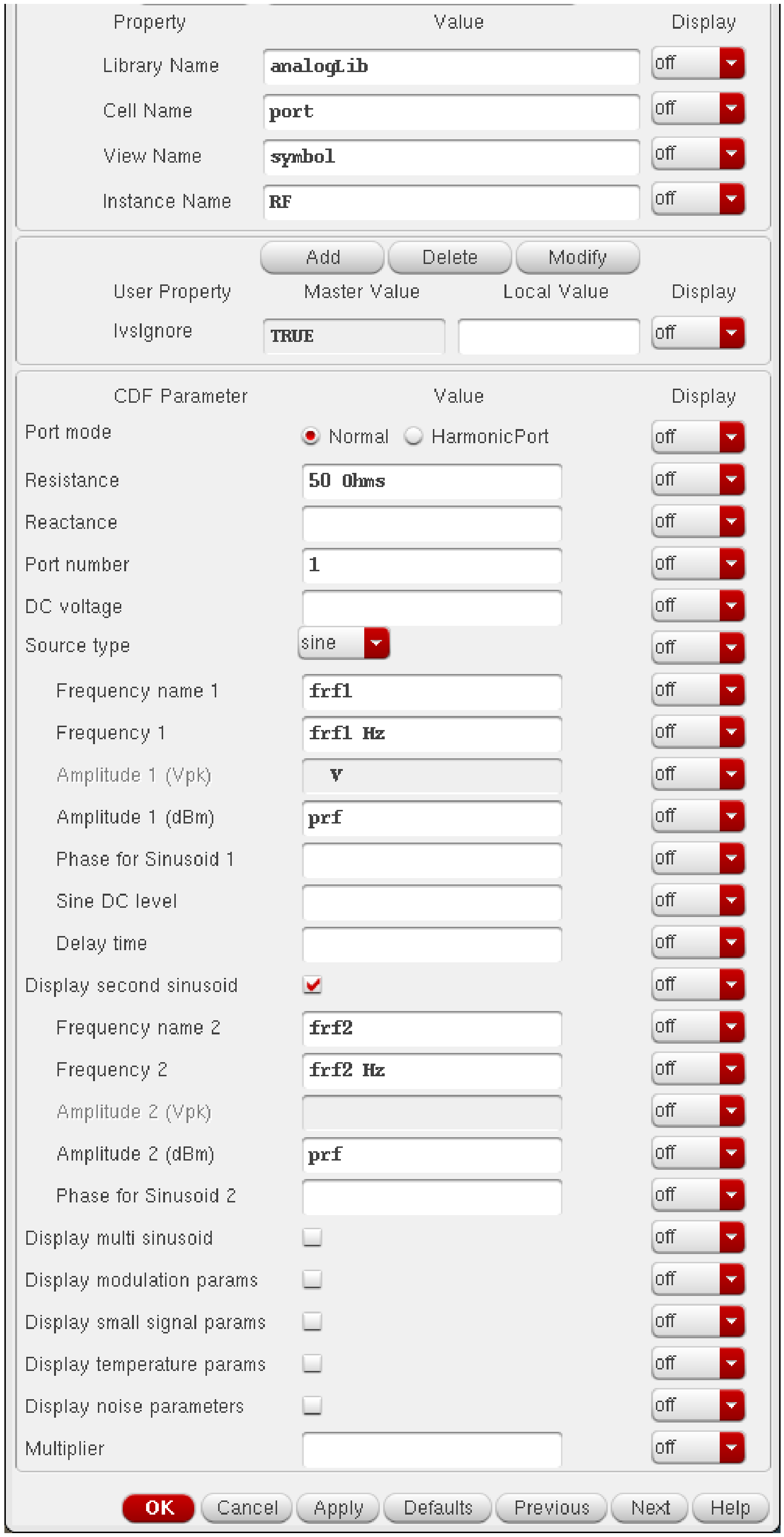
The final frequency is reported. The time and memory statistics are shown.



Examples

RF source setup for all examples in this chapter

The *Edit Object Properties* dialog for the port that is used as the RF source is shown below.


Property
Value
Display
Library Name
analogLib
Cell Name
port
View Name
symbol
Instance Name
RF
Add
Delete
Modify
User Property
Master Value
Local Value
lvsIgnore
TRUE
CDF Parameter
Port mode
Normal
HarmonicPort
Resistance
50 Ohms
Reactance
Port number
1
DC voltage
Source type
sine
Frequency name 1
frf1
Frequency 1
frf1 Hz
Amplitude 1 (Vpk)
V
Amplitude 1 (dBm)
prf
Phase for Sinusoid 1
Sine DC level
Delay time
Display second sinusoid
Frequency name 2
frf2
Frequency 2
frf2 Hz
Amplitude 2 (Vpk)
Amplitude 2 (dBm)
prf
Phase for Sinusoid 2
Display multi sinusoid
Display modulation params
Display small signal params
Display temperature params
Display noise parameters
Multiplier
off
OK
Cancel
Apply
Defaults
Previous
Next
Help

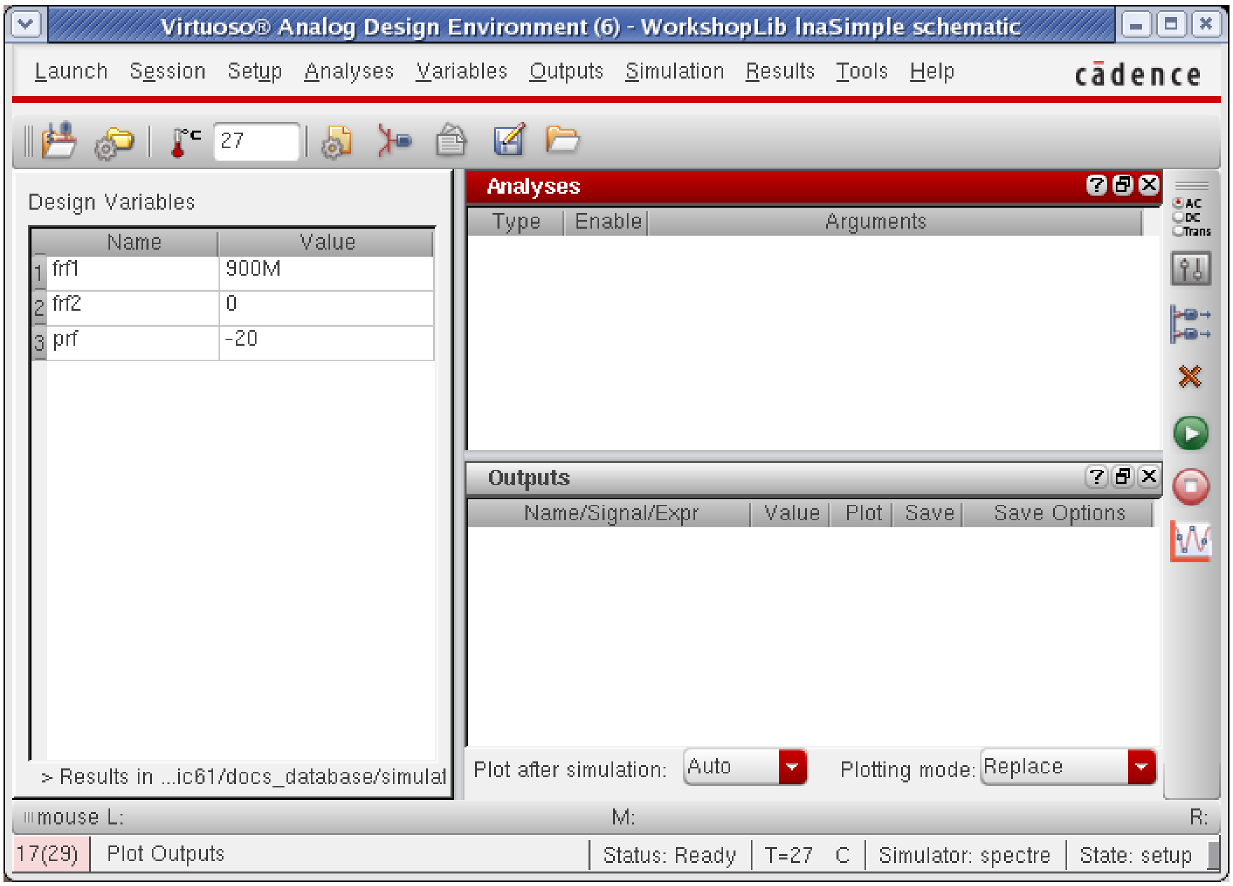
In the *Edit Object Properties* dialog:

1. If you select *Names* in the *Choosing Analyses* form, specify a name in the *Frequency name 1* and *Frequency name 2* fields.
2. Set the first sinusoid frequency to a variable name in the *Frequency 1* field.
3. Set the *Amplitude 1 (dBm)* field to a variable name.
4. Set the *Frequency* 2 field to a variable name.
5. Set the *Amplitude 2(dBm)* field to a variable name used to set the first frequency amplitude, if you want the amplitudes to match. If you do not want the amplitudes to match, you can use a different variable name.
6. Select the *Display small signal params* check box.
7. Set the *PAC Magnitude* field to 1 volt.

Note: Using variables in the entries enables you to change the properties without needing to change the schematic. Although this is useful mostly for ADE L, it works in ADE XL also. ADE XL allows any property in the schematic to be varied directly without changing the schematic.

ADE Setup for 1-tone Harmonic Balance (hb) Analysis

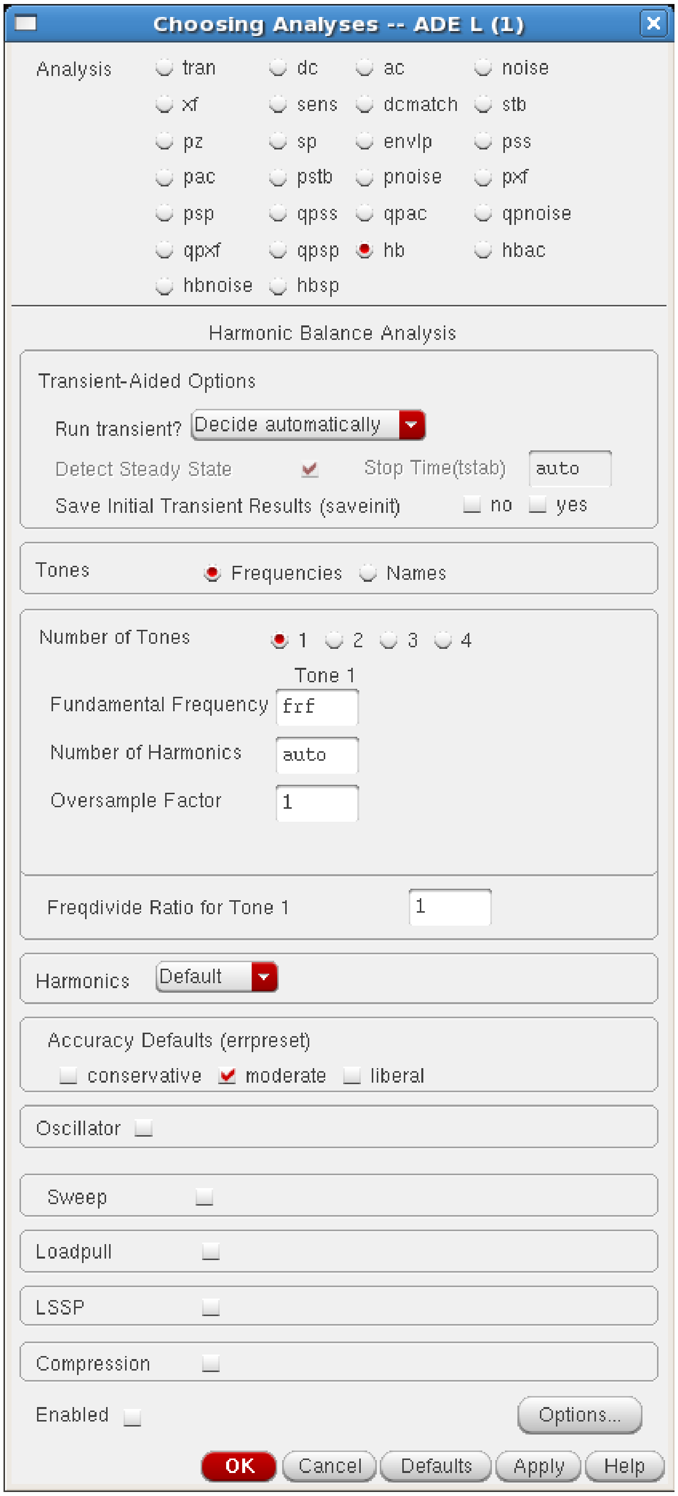
The Analog Design Environment (ADE) window is shown below.



Note: The variable name *frf1* is set to `900M` which sets the first frequency in the input port to 900MHz. The variable *prf* sets the input power for the RF input. The second frequency is zero, which disables that input.

To set the 1-tone hb analysis:

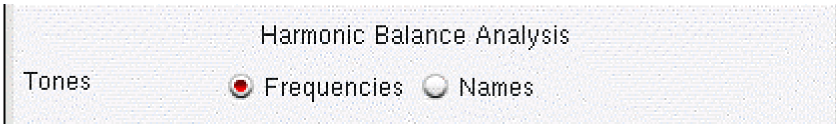
1. Open the *Choosing Analyses* form from the ADE window.



2. Select *hb*.

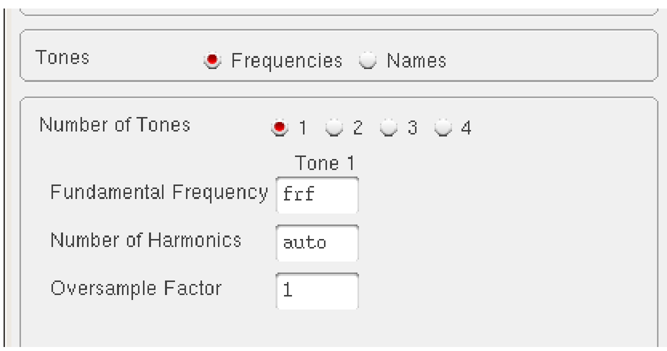
3. You can specify either the frequencies of the inputs, or get a list of the source names and frequencies in the schematic.

   Frequencies is select by default for hb analysis.



4. *Decide automatically* is the default for *Run Transient?*. This is suggested. Except for the frequency, this will set everything to automatic. If you select *Yes*, you will have the option of setting the tstab (transient-aided) time, and the option to automatically manage the tstab stop time.

5. Select *1* to set the number of tones to 1.

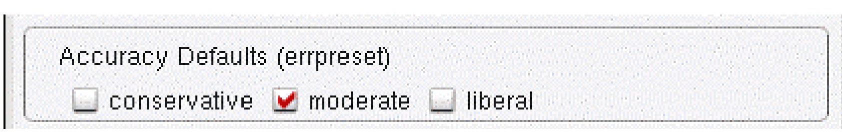


6. Set the frequencies, number of harmonics, and oversample factor.

   auto calculates the number of harmonics that are needed based on the Fourier Transform at the end of the tstab interval, and is available for the *Decide automatically* and *Yes* selections for *Run transient?*. If the circuit has square wave signals and you are setting harmonics manually, set the number of harmonics to at least the period of the signal divided by the risetime and set the oversample factor to between 2 and 8.

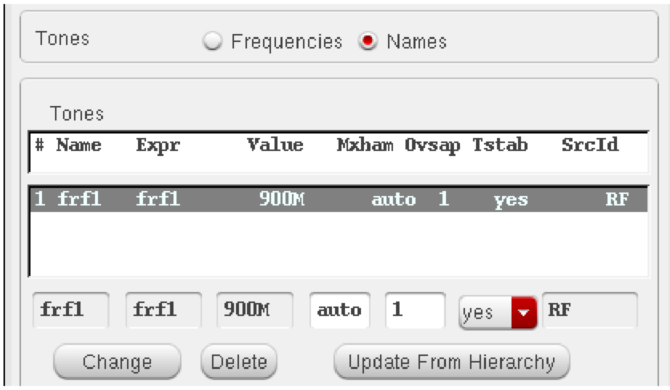
7. Select the *moderate* check box.

   If you need more accuracy, select the *conservative* check box.



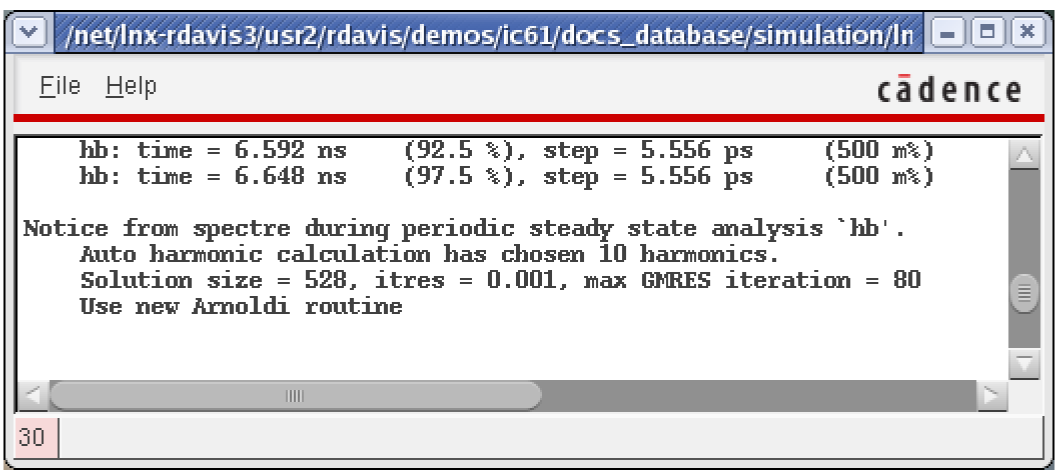
8. When you are done with the setup, run the simulation.

1. When *Names* is selected, all the sources in the circuit are read into the table along with their frequencies. In this case, there is one source with the frequency set to 900M using the variable *frf1*.



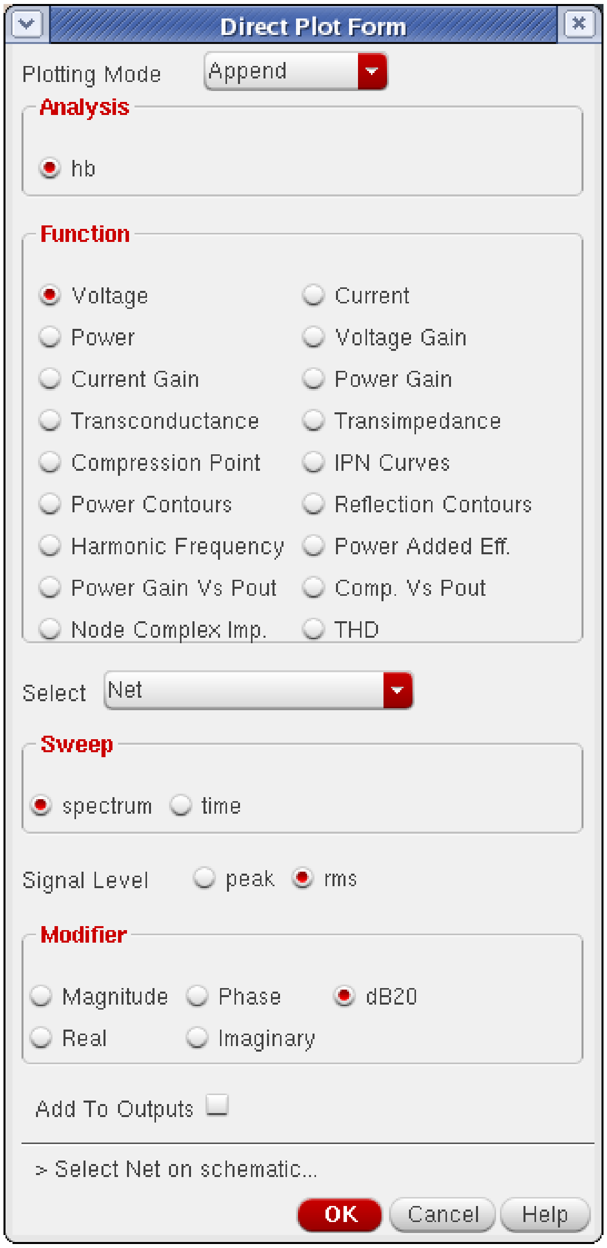
2. To make changes, select the line and set the number of harmonics and the oversample factor in the fields just below.
3. Click *Change,* select another field, or press the <*Tab*> key.
4. When you are done, run the simulation.

   Spectre will calculate the number of harmonics and display the result in the Spectre output window.



Plotting Spectral Content

1. In the ADE window, select *Results - Direct Plot - Main Form*.

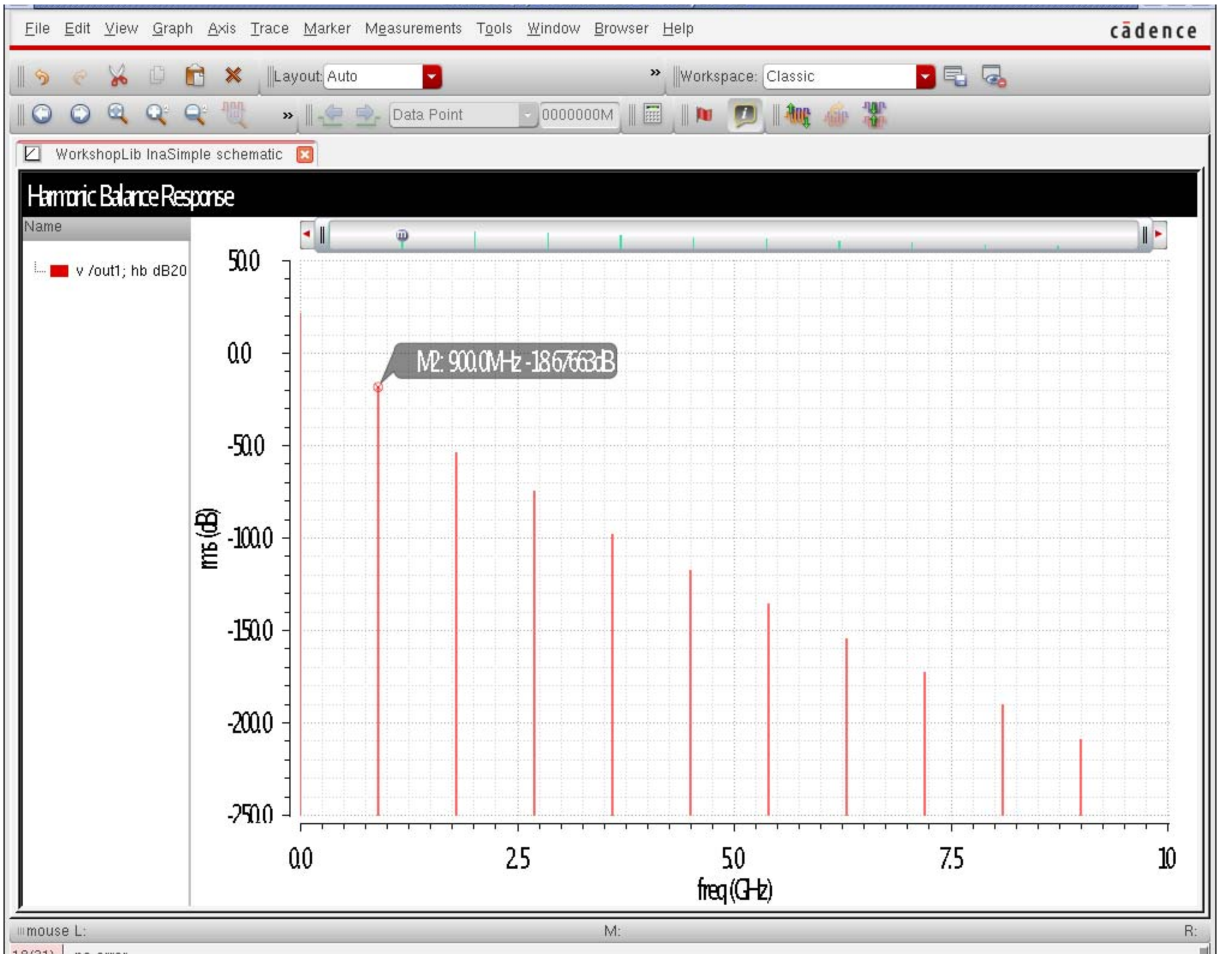


2. Select the *hb* radio button from the *Analysis* section.
3. Select the Voltage radio button from the *Function* section.
4. Select the *spectrum* radio button from the *Sweep* section.
5. Select the *rms* radio button.

6. Select the *dB20* radio button from the *Modifier* section.

7. If you want this signal to plot automatically when a new simulation is run, select *Add To Outputs*.

8. Select the node that you want to plot on the schematic.



The output harmonics are shown in the waveform tool.

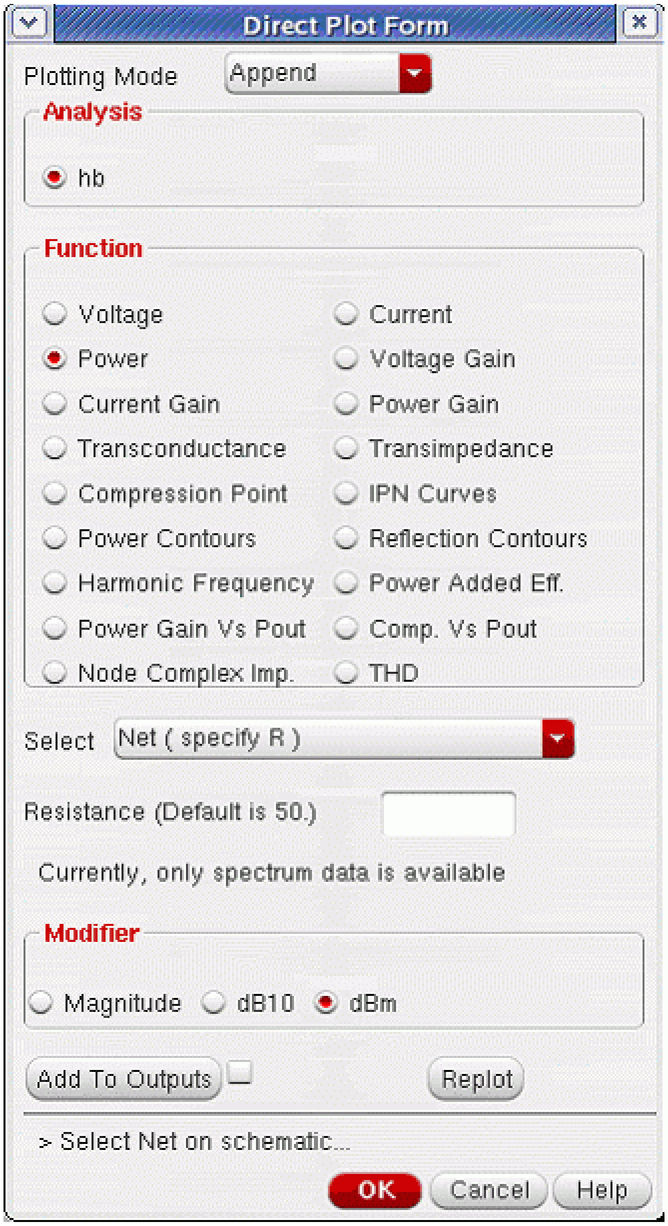
The dB20 modifier is dB with respect to 1 volt, or dBV.

To position the markers, move your cursor near the object you want to read, and type *m*.

Plotting dBm

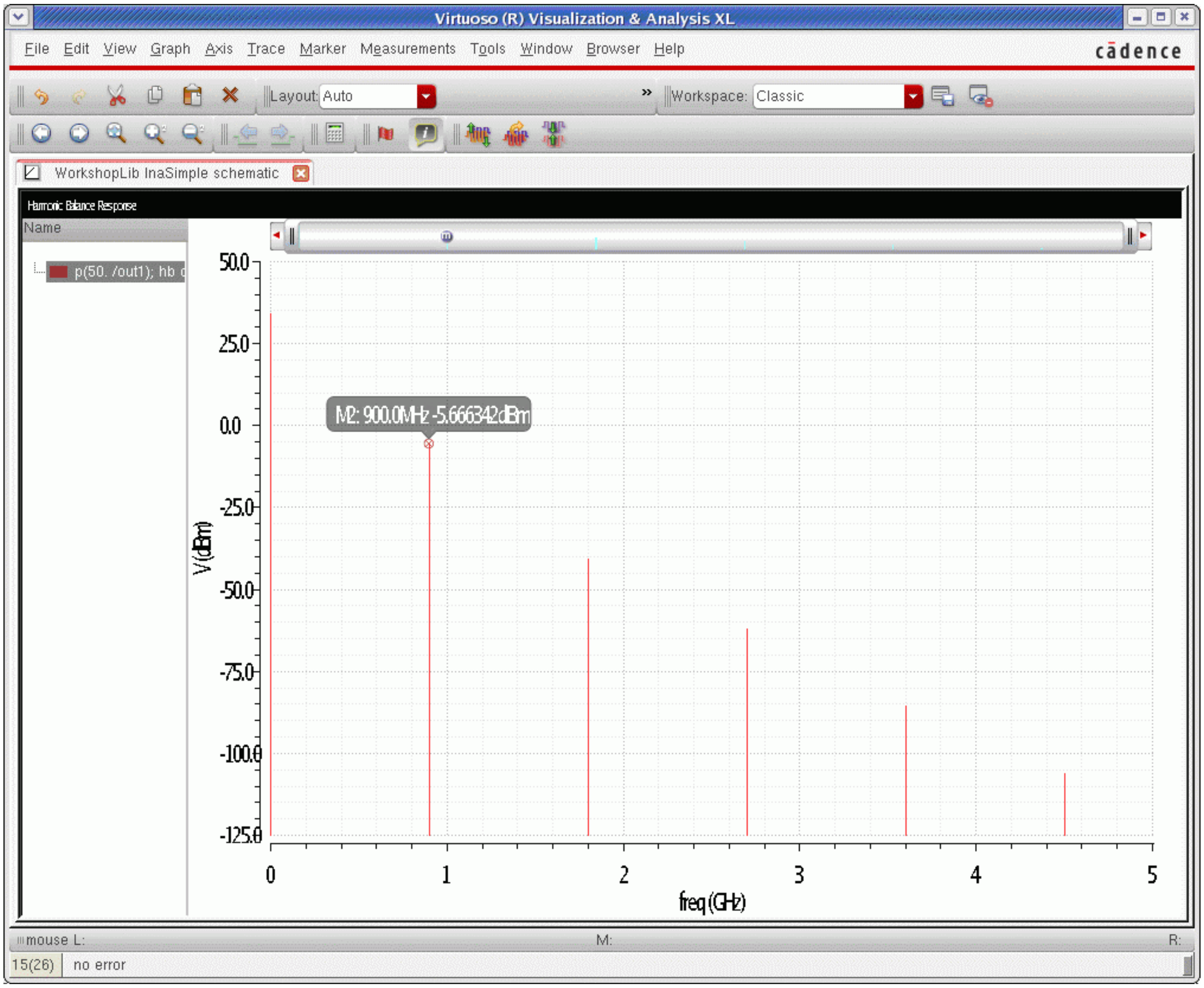
On the *Direct Plot Form*:

1. Select *hb* from the *Analysis* section.



2. Select the *Power* radio button from the *Function* section.
3. Select the *Net (specify R)* option from the drop-down list.
4. Select *dBm* from the *Modifier* section.
5. Select the appropriate net on the schematic.

6. Click *OK*.

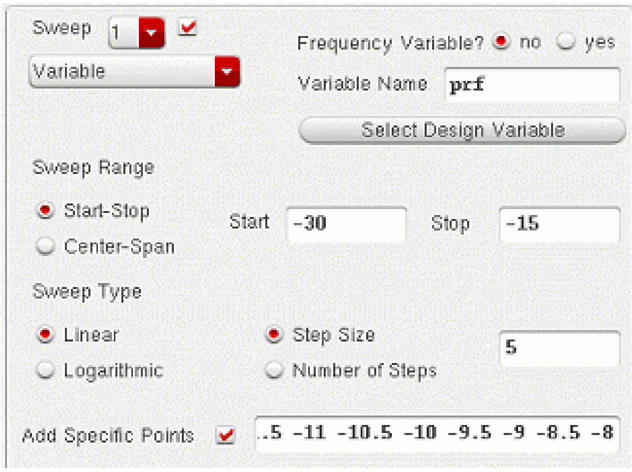


The output spectrum in dBm is displayed.

To position the markers, move your cursor near the thing you want to read, and type *m*.

Sweeps

1. To enable sweeps, select the *Sweep* check box in the *Choosing Analyses* form.



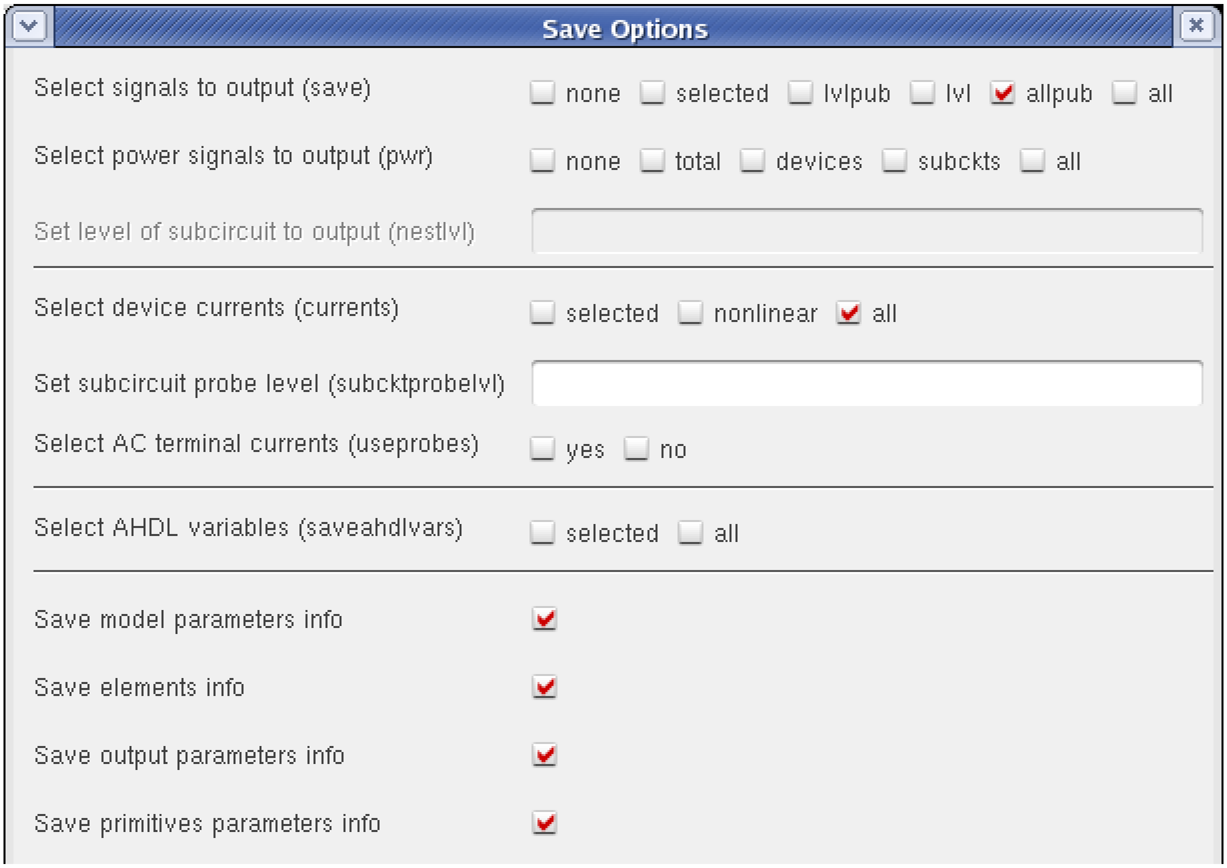
2. Select what you want to sweep from the drop-down list. In the figure above, *variable* is selected.
3. Click *Select Design Variable* and select the variable that you want to sweep.
4. Specify the sweep range in the *Start* and *Stop* fields.
5. If desired, select the *Add Specific Points* check box and specify the additional points.

   Separate the entries with a space.

Plotting Currents

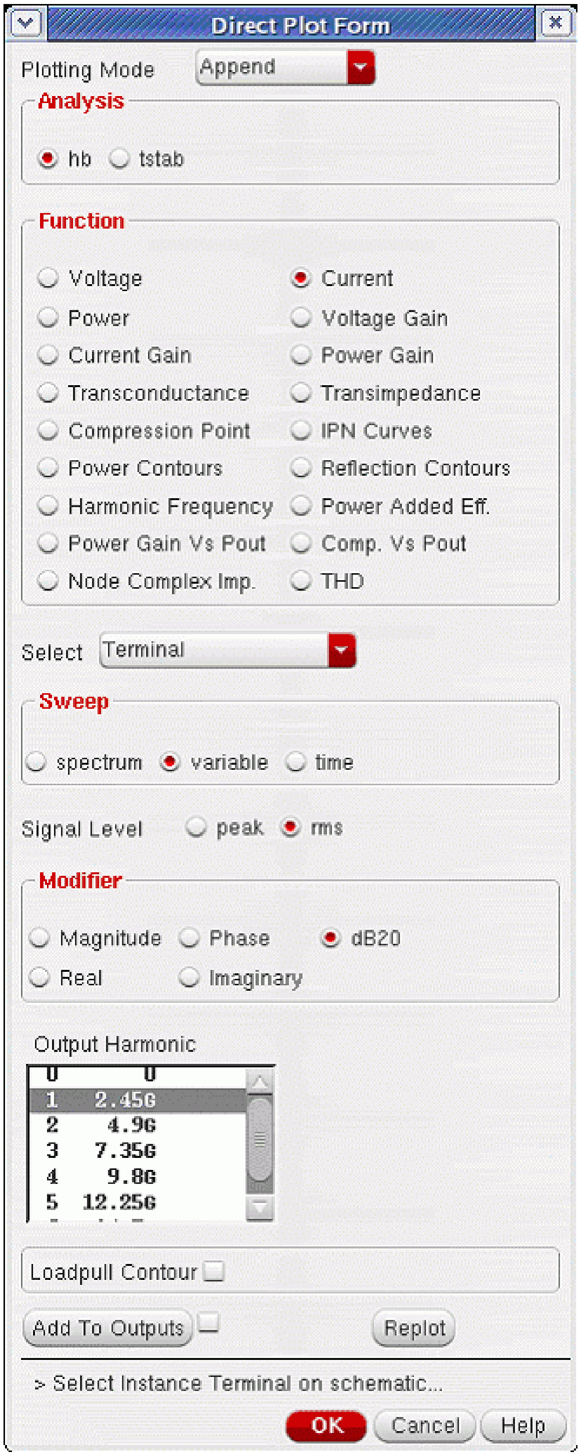
In the past, currents could be plotted from the terminals of the sources in the circuit only. That required adding an iprobe to the circuit in series with the branch. An iprobe is a zero volt voltage source that also allows the current in that branch to be solved.

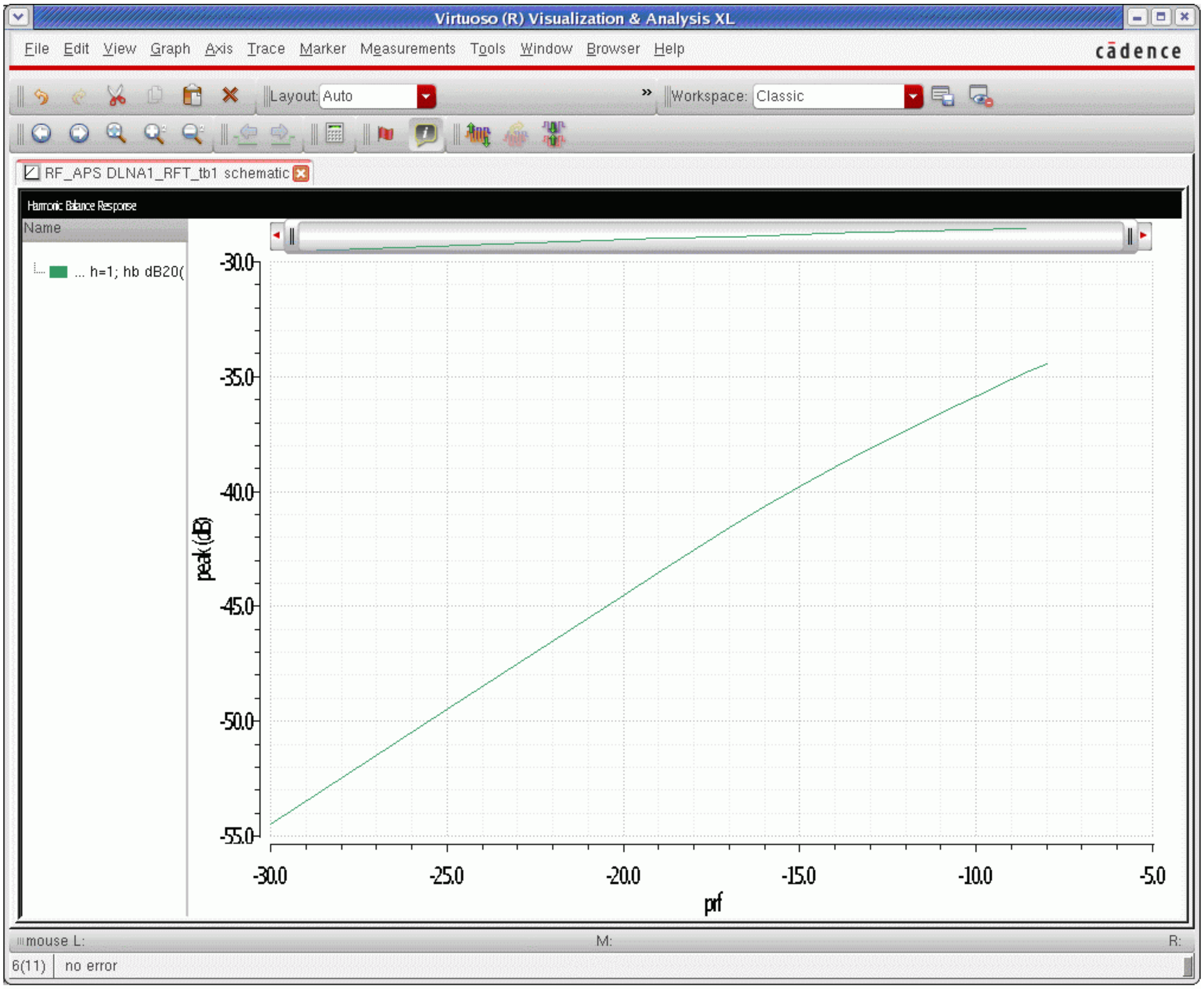
Starting with the MMSIM 12.1.1 release, the normal selection method in ADE for saving currents is supported. If you want to save the currents, select *Outputs - Save All*, and choose *selected*, *nonlinear*, or *all*. This is shown below.



If you choose *selected*, then also select *Outputs - To Be Saved - Select On Schematic*, and make terminal selections on the schematic. Selecting terminals will add expressions in the ADE outputs section to save those selected currents.

To plot the currents, select the *Current* radio button from the *Function* section in the *Direct Plot Form* and click *OK*.





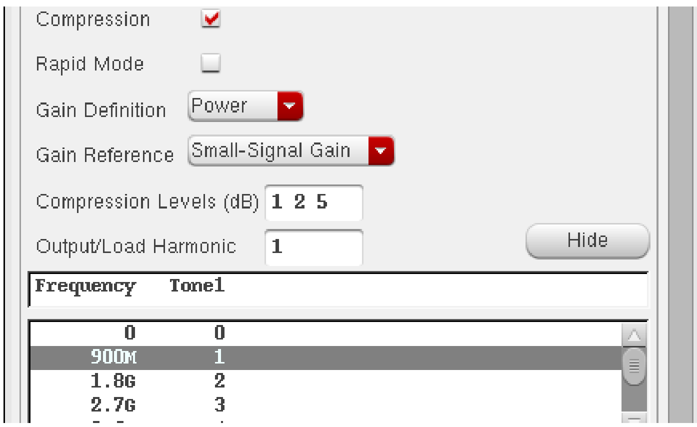
Compression Point

Gain compression enables you to calculate the compression points and compression curves directly, without the need for postprocessing or manual setup of power sweeps. Gain compression calculates voltage and power-based compression points and provides the flexibility to choose the gain reference for compression calculations. With gain compression analysis, the simulator outputs the following:

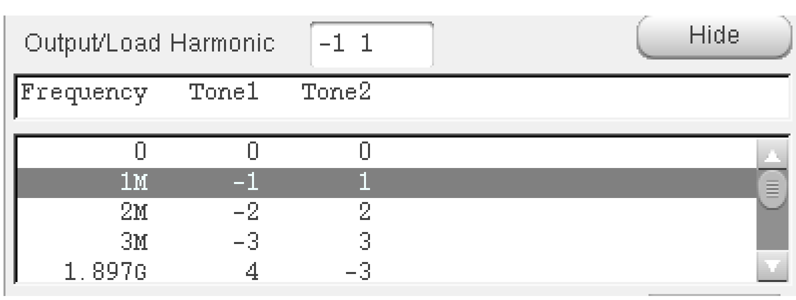
- The power and gain compression curves
- The input-referred power (or voltage) at compression
- The output-referred power (or voltage) at compression

Power Compression and Gain Curves

1. First, set a reasonable number of harmonics. for the simulation to be accurate at the compression point.
2. Select the *Compression* check box.



3. If desired, select *Rapid Mode*. This mode is faster but slightly less accurate, and is suggested for Monte Carlo, sweeps, or corners where a large number of simulations need to be performed. In this mode, you need to supply an estimate of the input-referred compression point.
4. Choose *Power* from the *Gain Definition* drop-down list for compression point calculation.
5. Choose the reference point for gain calculation from the *Gain Reference* drop-down list.
6. Specify the gain compression level in the *Compression Level (dB)* field. This can be a list of compression values separated by spaces in normal mode only. This is shown above in step 3.
7. Specify the output harmonic number in the *Output/Load Harmonic* field. For a two-tone simulation (a mixer), the output harmonic number will be a two-number sequence. Alternatively, you can click *Choose* and select the output harmonic from the table, as shown below.

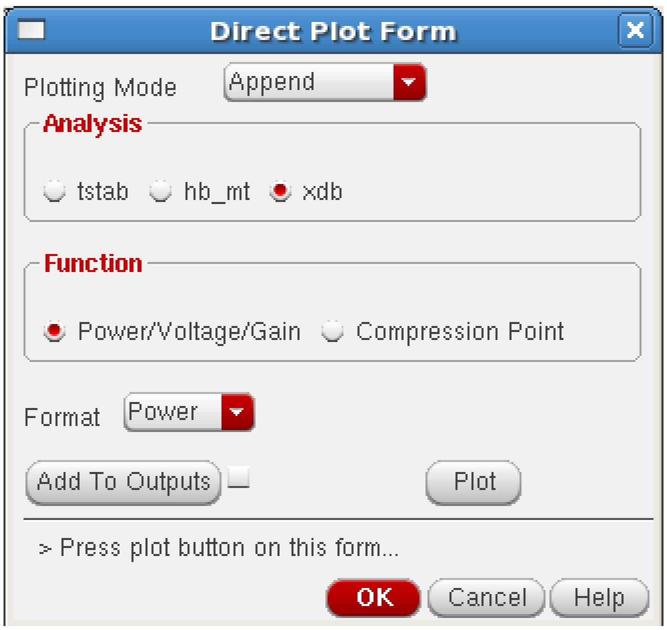


8. Specify the instance name of the input source in the *Source* text field. Alternatively, you can click *Select* and choose the source from the schematic. For power gain, the input source must be a port instance.

9. Specify the instance name of the load termination in the *Load* text field. Alternatively, you can click *Select* and choose the source from the schematic. For load termination, the instance can be a port or a resistor.

10. Run the simulation.

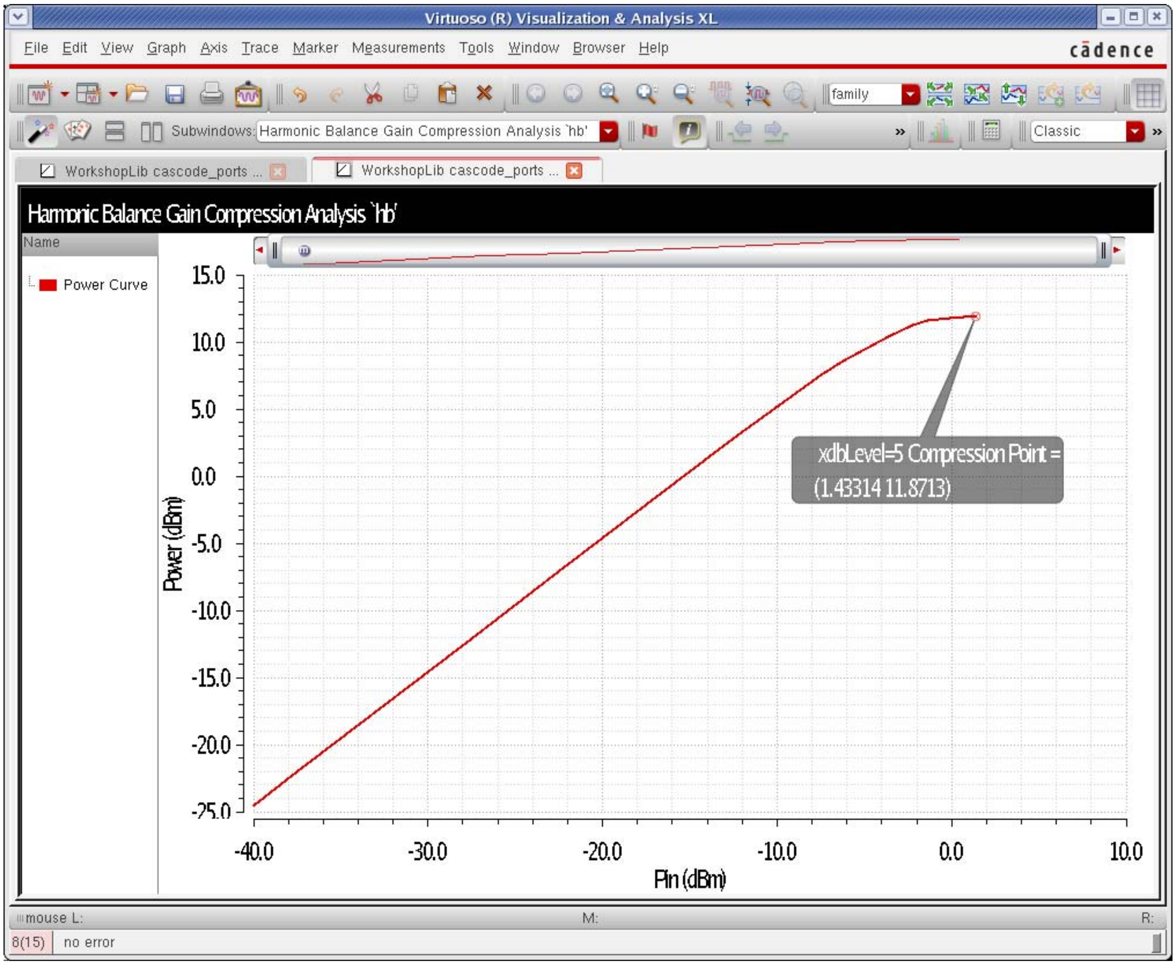
After the simulation completes, choose *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed, as shown below.



In the *Direct Plot Form*:

1. Choose *xdb* from the *Analysis* section.
2. Choose *Power/Voltage/Gain* from the *Function* section.
3. Choose *Power* from the *Format* drop-down list.
4. Click *Plot*.

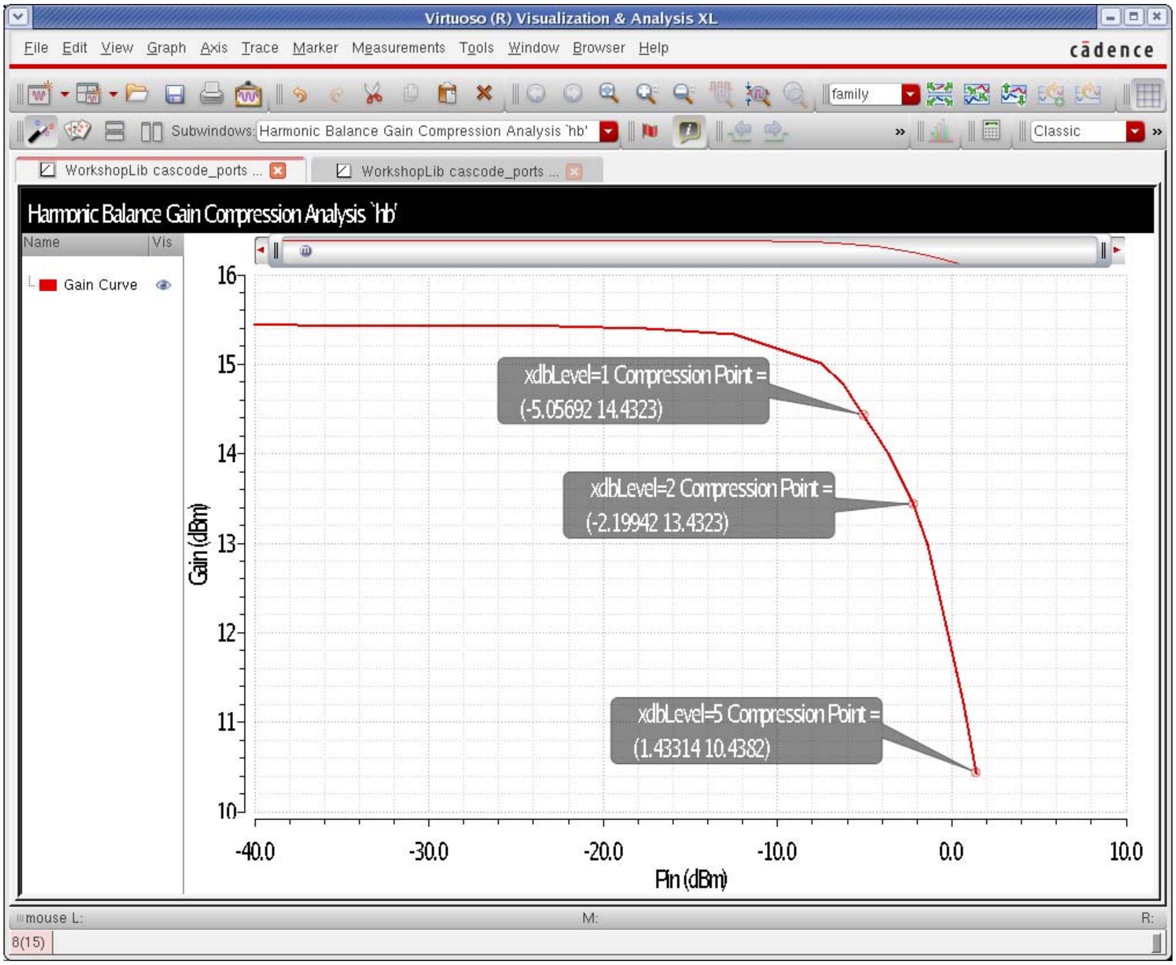
The power curve is plotted in the waveform tool, as shown below.



5. Now choose *Gain* from the *Format* drop-down list.

6. Click *Plot*.

The gain curve is plotted in the waveform tool. This example shows multiple compression levels of 1, 2, and 5dB.

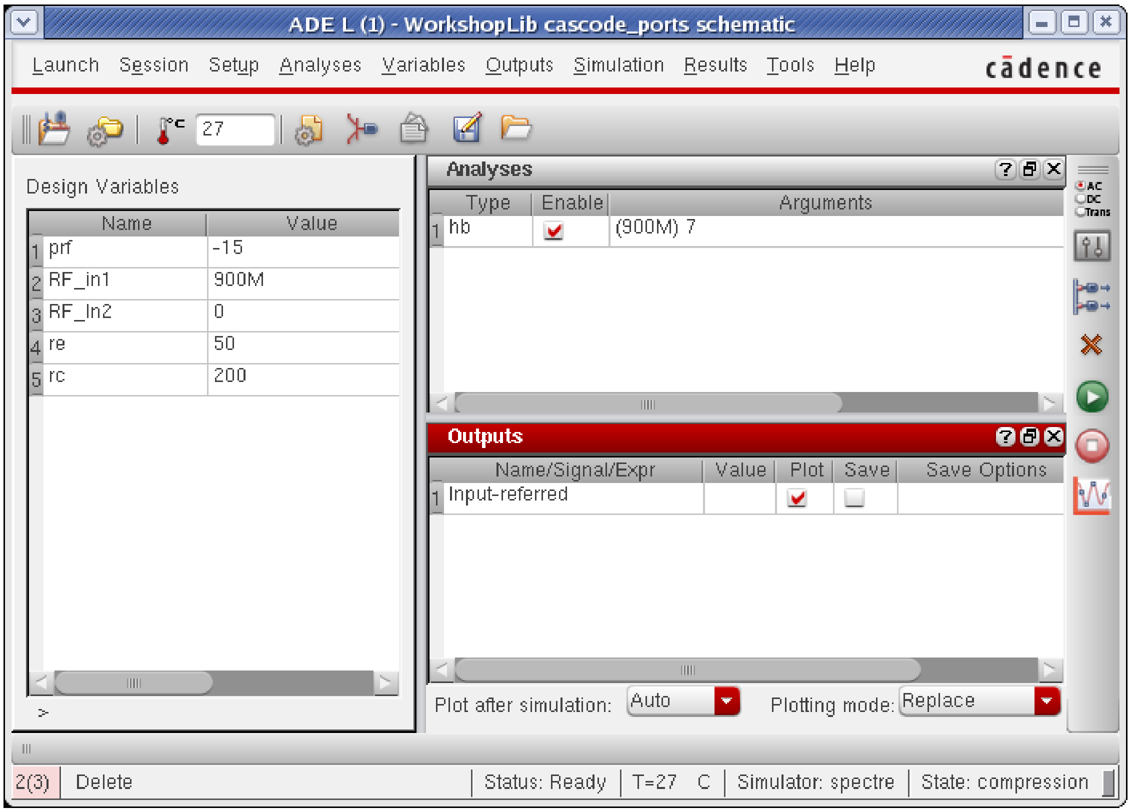


Input-Referred Compression

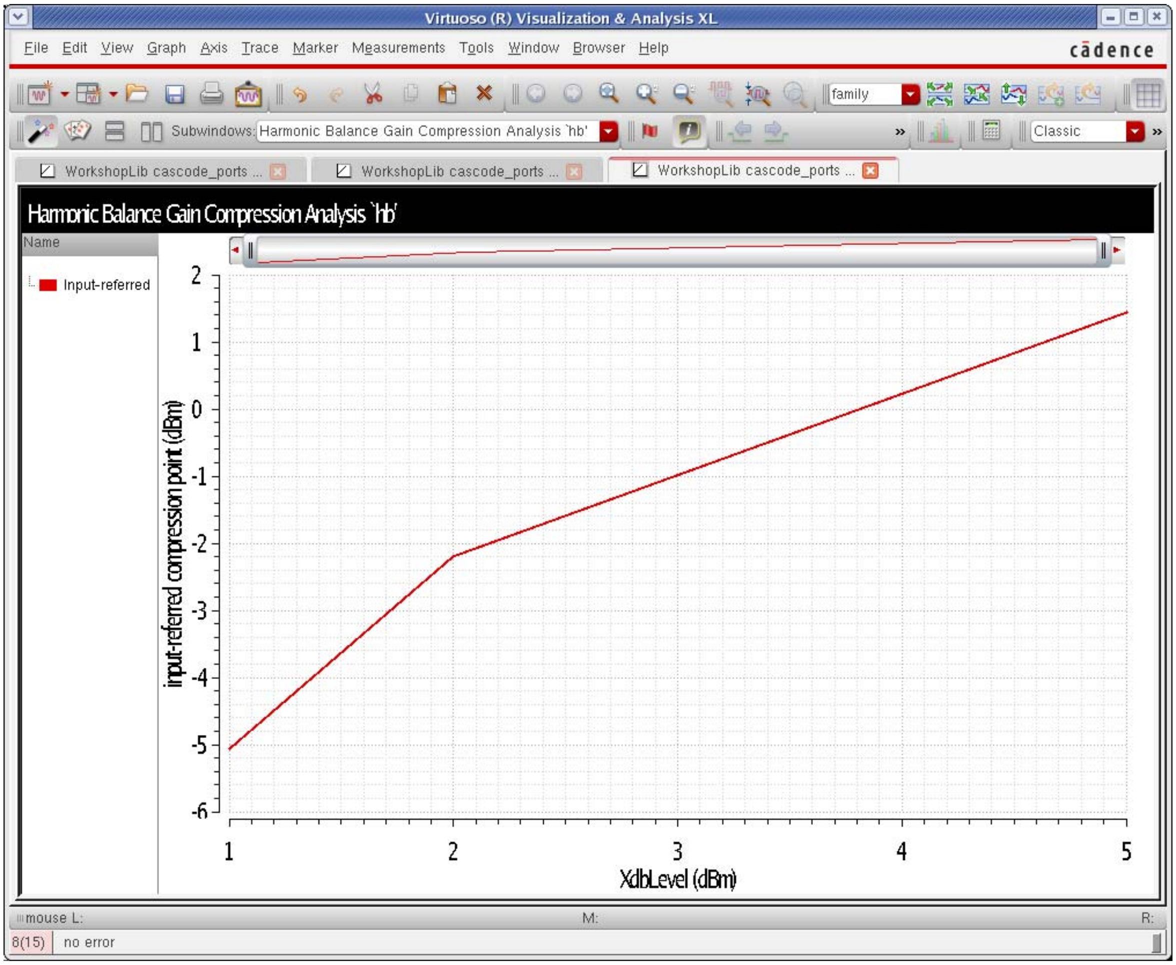
In the *Direct Plot Form*:

1. Choose *xdb* from the *Analysis* section.
2. Choose *Compression Point* from the *Function* section.
3. Choose *Input-referred* from the *Compression Point/Curve* drop-down list.

4. If you want to add the expression to the ADE outputs section so that it plots automatically after a simulation, select *Add To Outputs*.



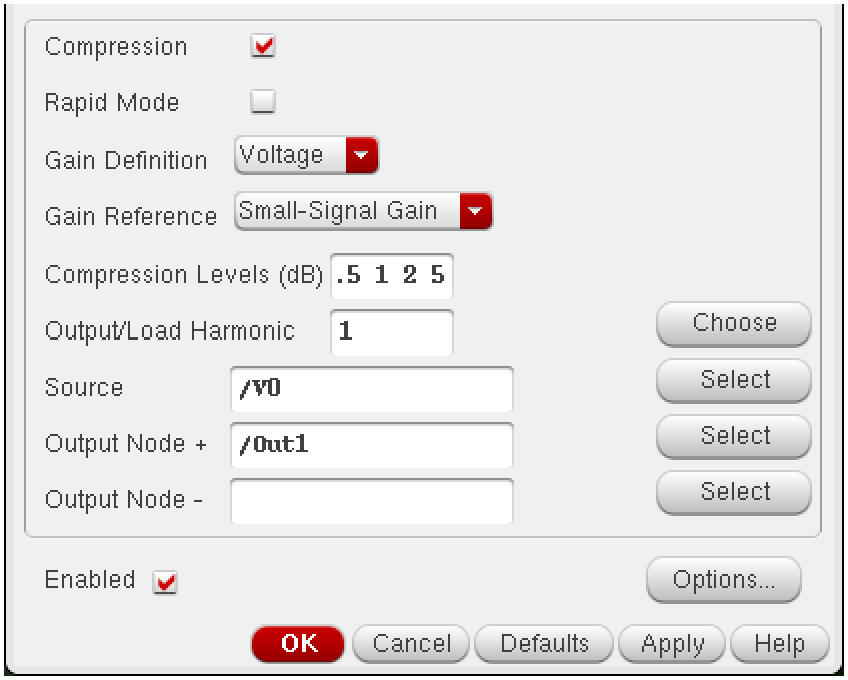
The input-referred compression levels plot. This example shows 1, 2, and 5dB compression levels.



You can also add *Output-referred* compression in the same way.

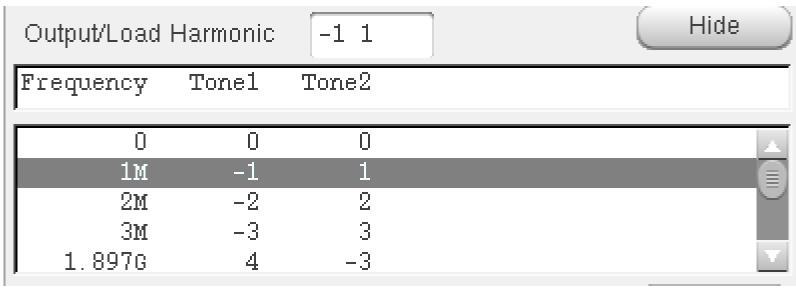
Voltage Gain Compression

1. Select the *Compression* check box.



2. If desired, select *Rapid Mode*. This mode is faster but slightly less accurate, and is suggested for Monte Carlo, sweeps, or corners where a large number of simulations need to be performed. In this mode, you need to supply a n estimate of the input-referred compression point.
3. Choose *Voltage* from the *Gain Definition* drop-down box for compression point calculation.
4. Choose the reference point for gain calculation from the *Gain Reference* drop-down list.
5. Choose the gain compression level in the *Compression Level (dB)* field. A list of values separated by spaces is allowed in normal mode. *Rapid Mode* requires a single entry.
6. Specify the output harmonic number in the *Output/Load Harmonic* field. For a two-tone simulation (a mixer), the output harmonic number will be a two-number sequence.

Alternatively, you can click the Choose button at the right of the Output/Load Harmonic field and select the output harmonic from the table, as shown below.



7. Specify the instance name of the input source in the *Source* text field. Alternatively, you can click *Select* and choose the source from the schematic. For voltage gain, the input source can be a vsource or a port.

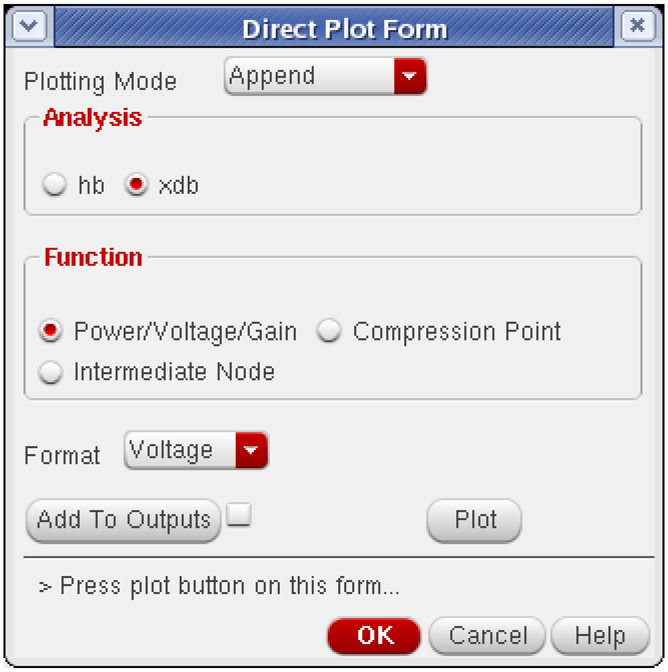
8. Specify the positive output terminal for voltage gain calculation in the *Output Node +* field. If you do not specify a value then the node is considered as global ground.

9. Specify the negative output terminal for voltage gain calculation in the *Output Node +* field. If you do not specify a value then the node is considered as global ground.

10. If data for intermediate nodes in the schematic is desired, then select *Outputs - To Be Saved - Select On Design in ADE*, and click the nets you want to save. These nets can be plotted in the *Direct Plot Form* after the simulation.

11. Run the simulation.

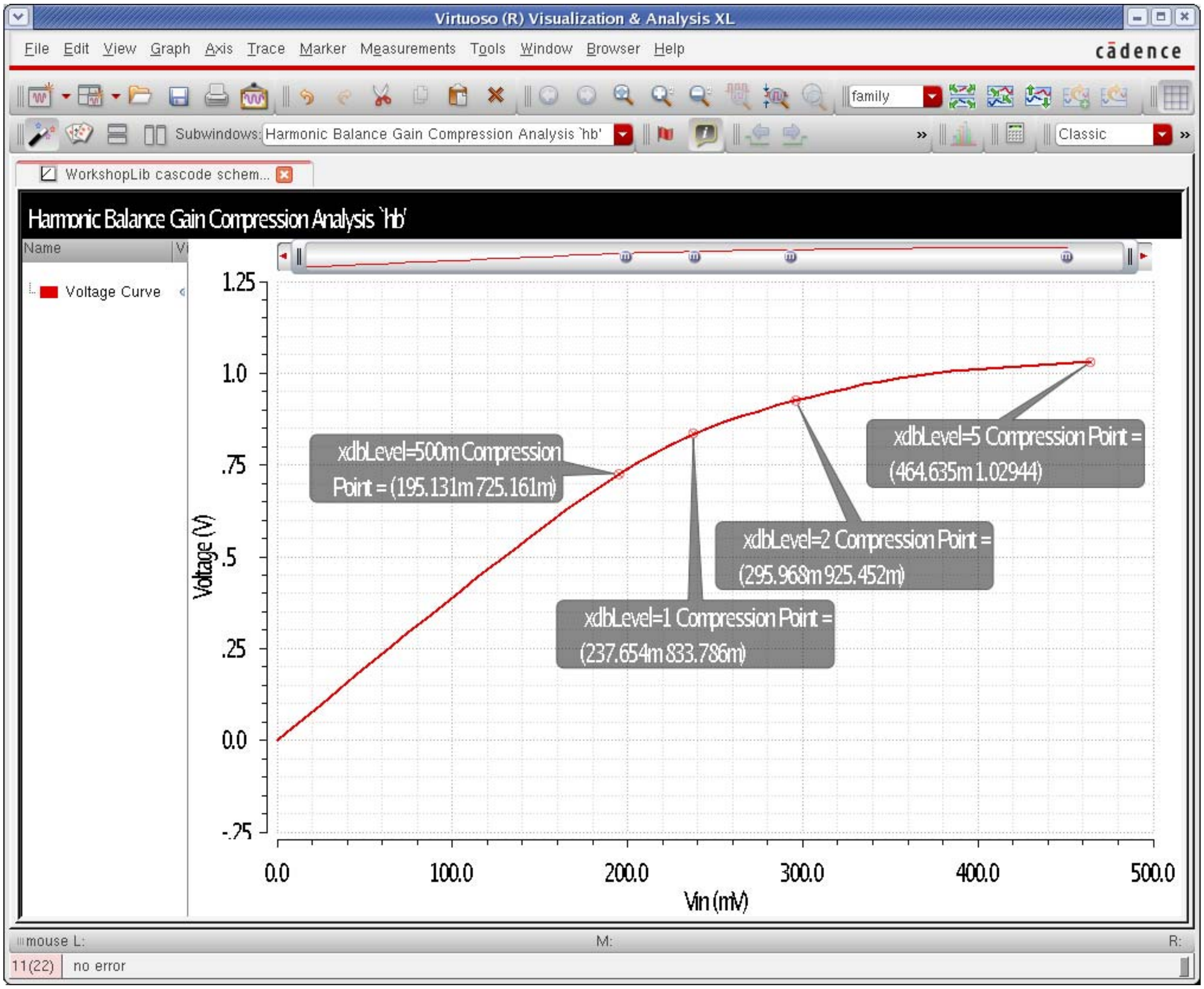
After the simulation completes, choose *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed, as shown below.



In the *Direct Plot Form*:

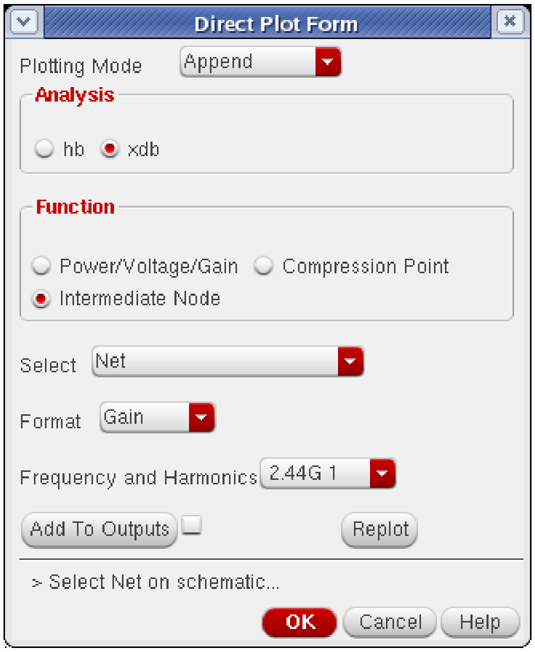
1. Choose *xdb* from the *Analysis* section.
2. Choose *Power/Voltage/Gain* from the *Function* section.
3. Choose *Voltage* from the *Format* drop-down list.
4. If you want to add the expression to the ADE outputs section so that it plots automatically after a simulation, select *Add To Outputs*.
5. Click *Plot*.

The Voltage curve is plotted in the waveform tool, as shown below.



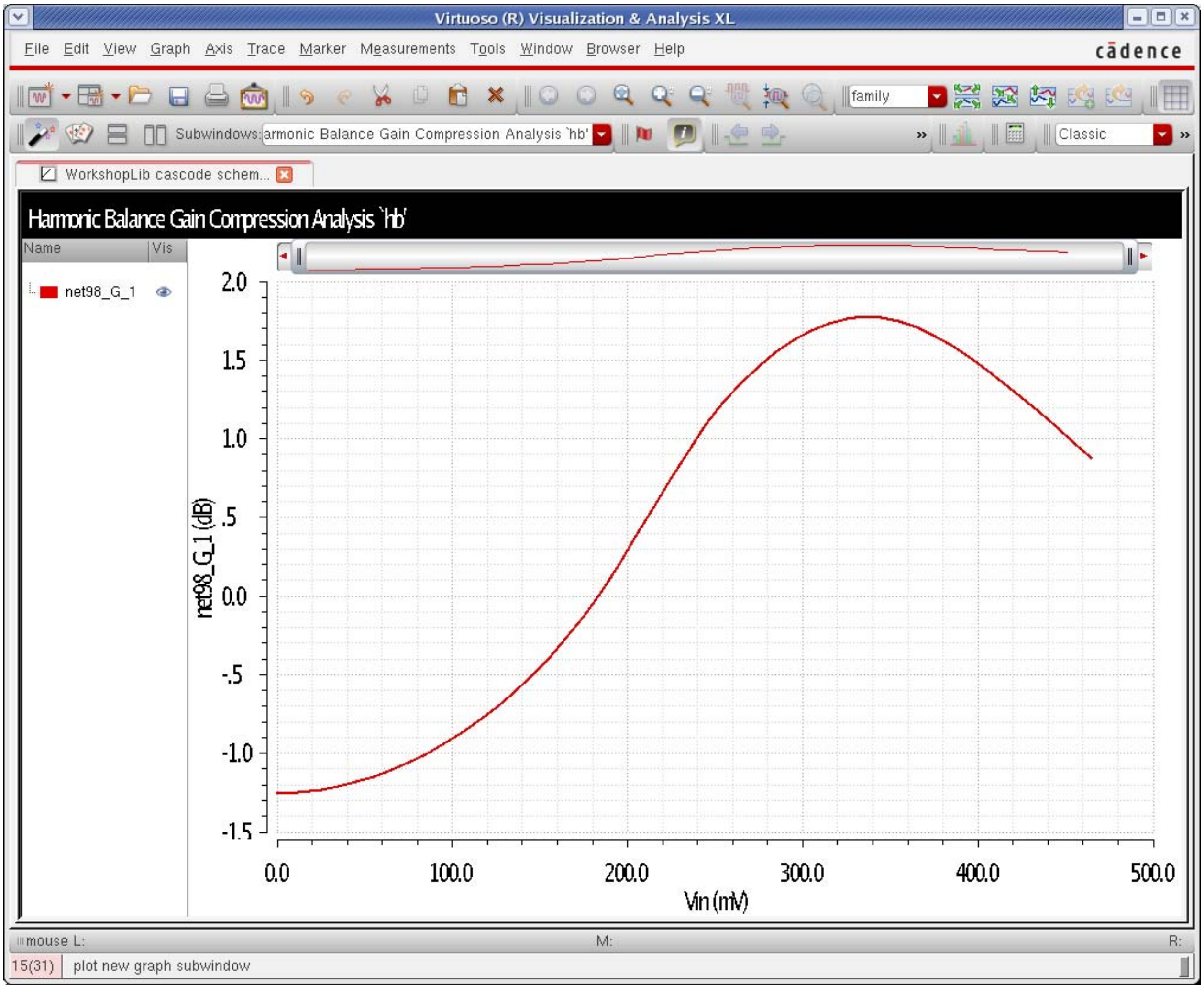
1. To plot the data from intermediate nets, select Intermediate Node from the Function Section.
2. Select *Gain*, *Voltage*, or *Phase* from the *Format* drop-down list. *Gain* is shown in this example.

3. Select the harmonic from the Frequency and Harmonics selection.

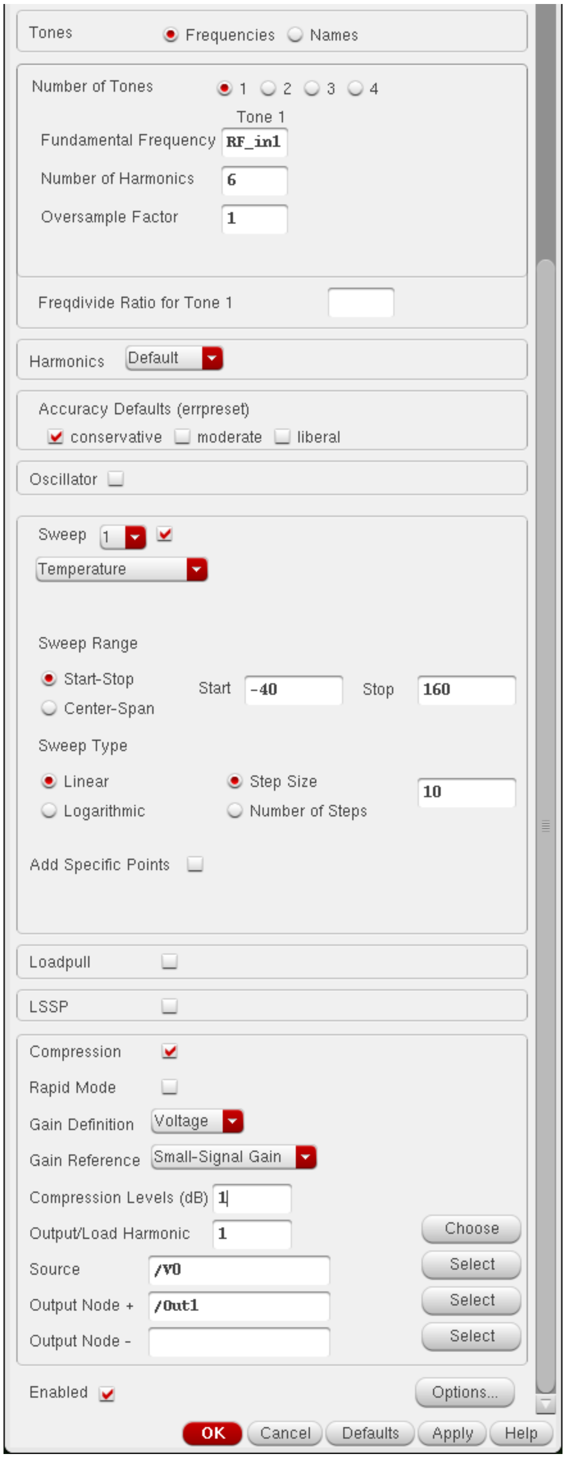


4. Click the net in the schematic.

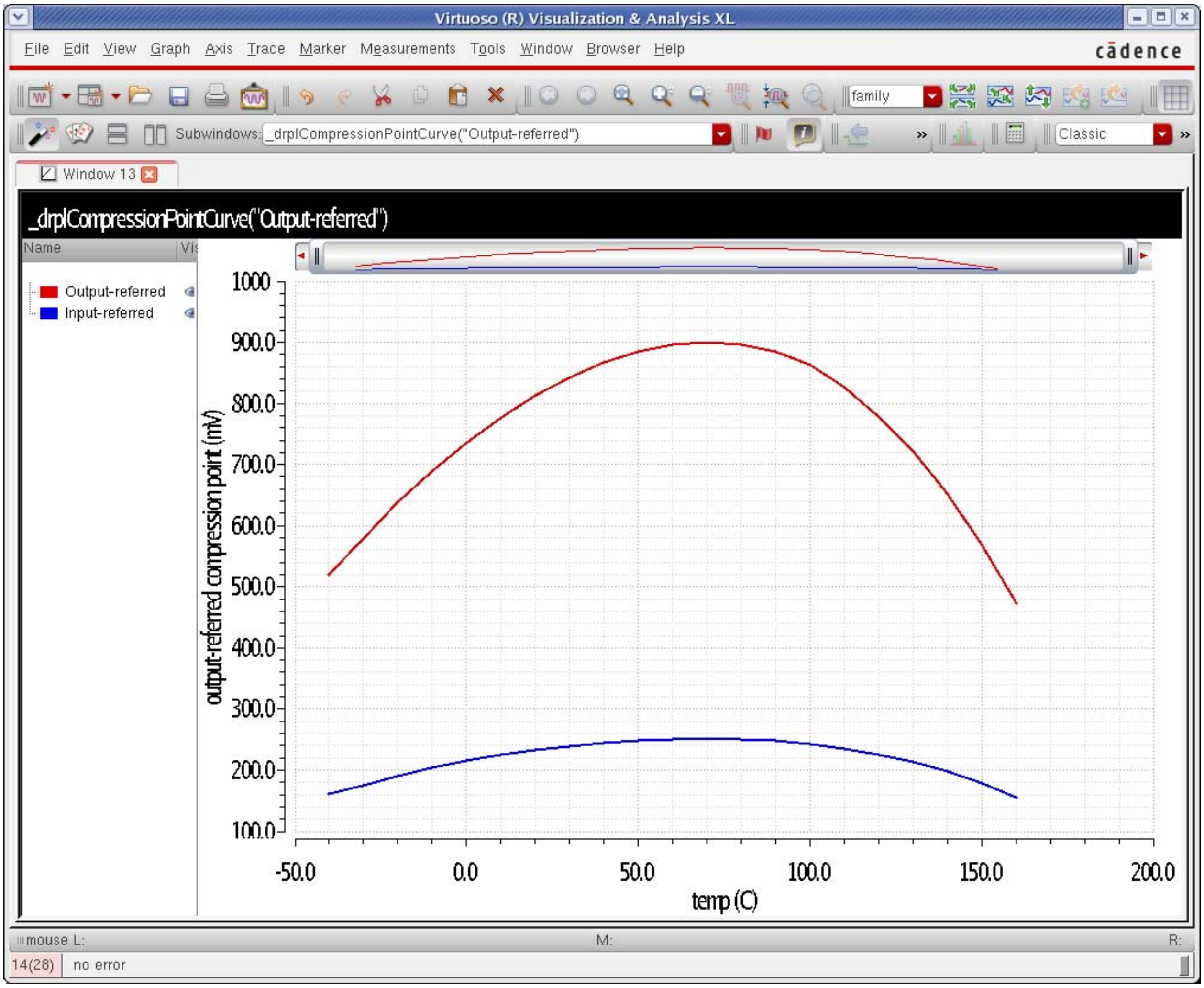
5. The data plots without the compression levels shown.



Compression can be combined with sweeps or monte carlo. To run sweeps, enable the sweep button in the Choosing Analyses form, and set up a sweep. A temperature sweep is shown below.

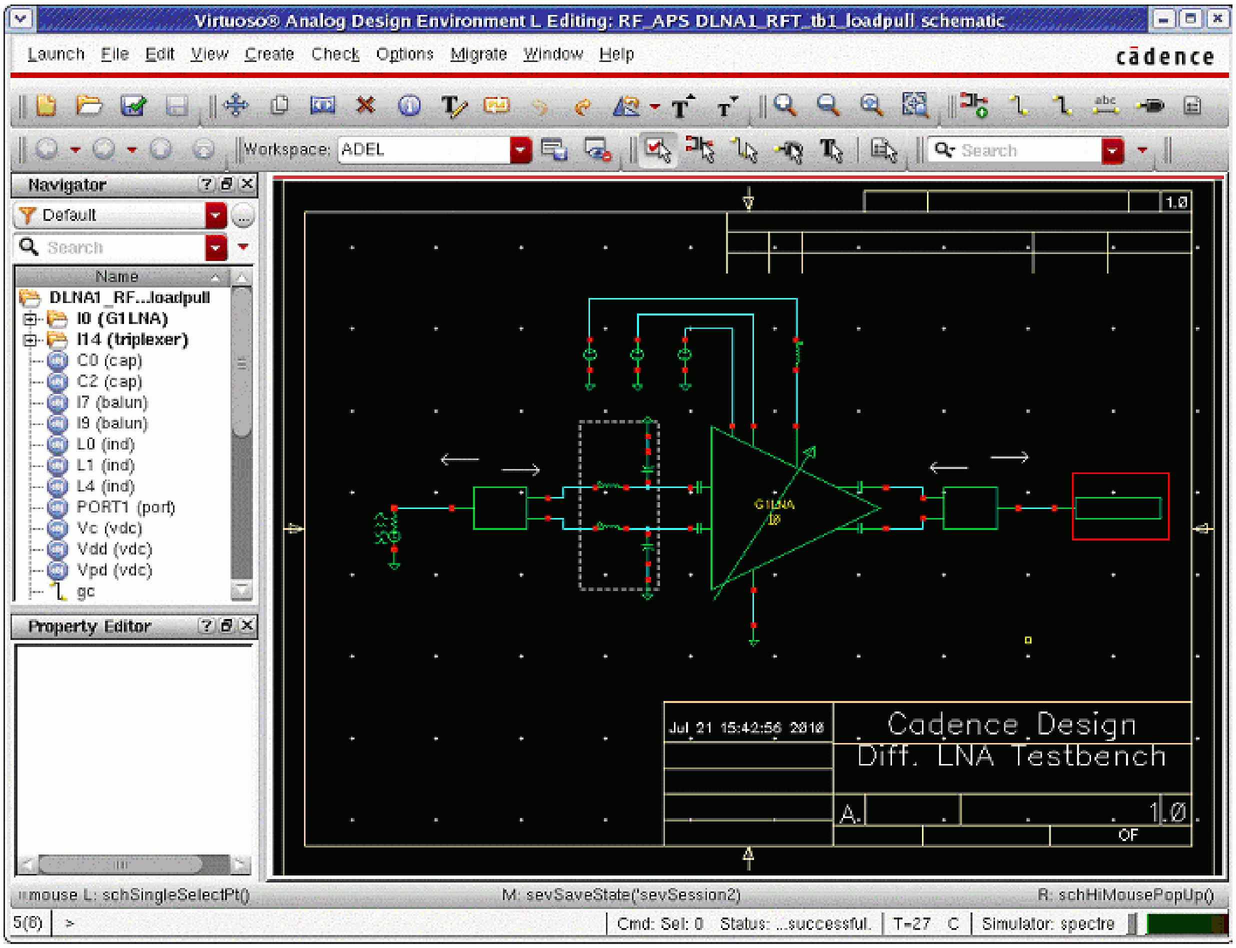


Now run the analysis, and either Direct Plot the compression points, or use the ADE outputs added previously to plot the compression point versus temperature.

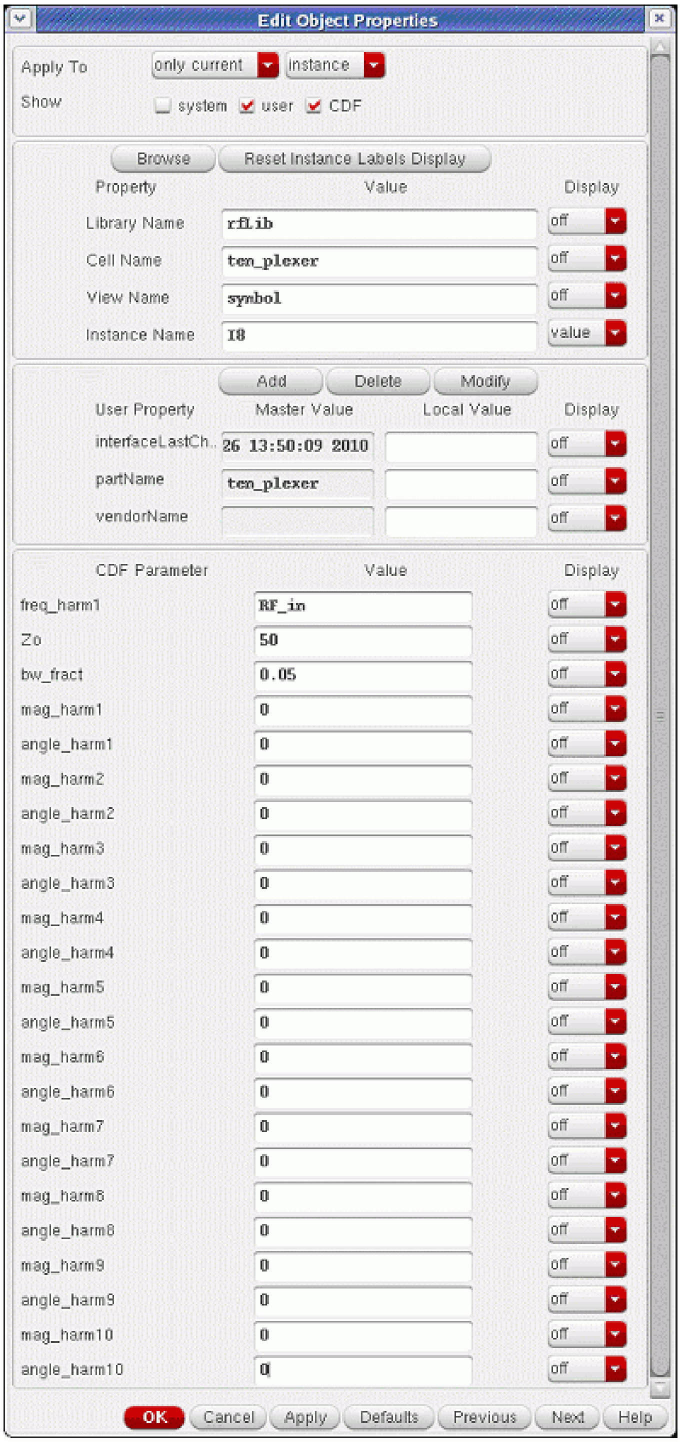


Loadpull

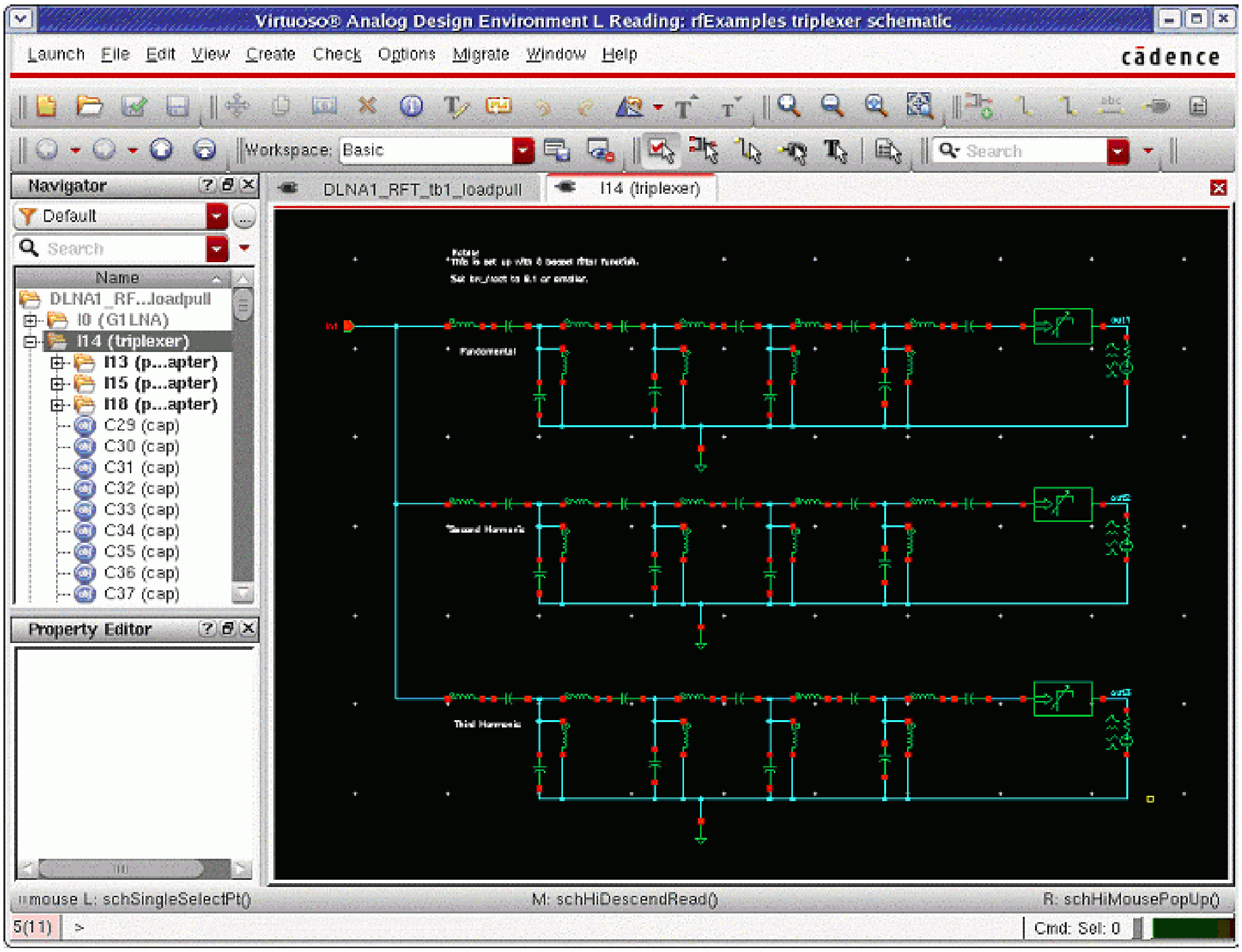
For a loadpull simulation, add a triplexer (for setting the reflection and angle for up to three harmonics) or the ten_plexer (for up to 10 harmonics) as the load to the circuit. These are in the `rfLib` library.



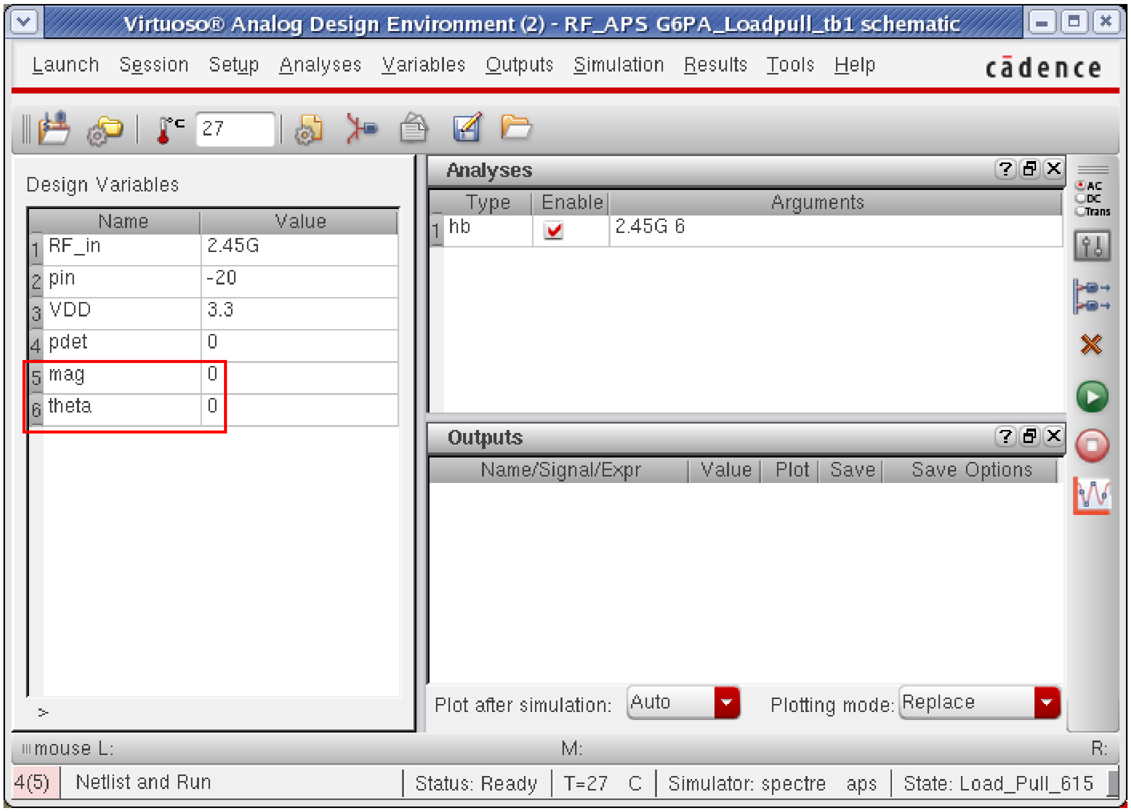
Normally, properties are used to set the reflection coefficient magnitude and phase for the harmonics that are present in the triplexer or ten_plexer. The example below shows the ten_plexer, and has 0 (zero) set for all the reflection coefficients and angles.



Descend-read into the triplexer or ten_plexer. Triplexer is shown. The top bandpass filter is for the first harmonic. The middle is the second harmonic. The bottom is the third harmonic.



1. Add two variables in ADE.

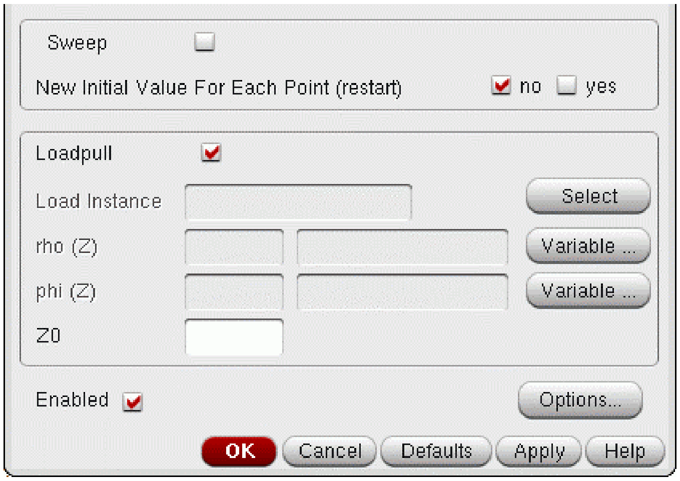


The example above shows `mag` and `theta` variables added. These variables should not be used to set any component values in the circuit.

Triplexer from `rfLib` directory contains three filters for the first, second, and third harmonics and portAdapters with port loads for each of the three filters.

2. Open the pss or hb *Choose Analyses* form.

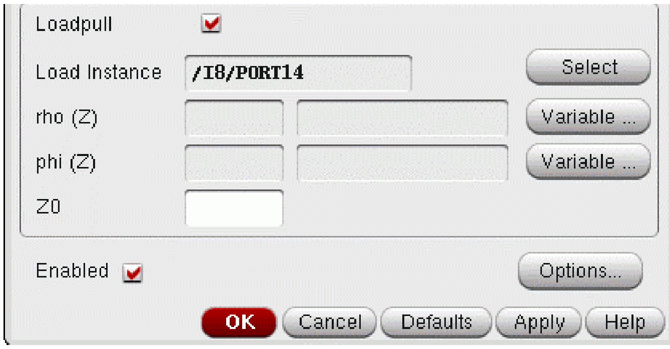
3. Select the *Loadpull* checkbox.



4. Click the *Select* button on the right of the greyed out *Load Instance* field.

5. In the triplexer schematic, select the topmost port on the right side of the schematic. if you want a loadpull for the first harmonic. Select the next port down for a loadpull of the second harmonic. Select the third port down for the third harmonic.

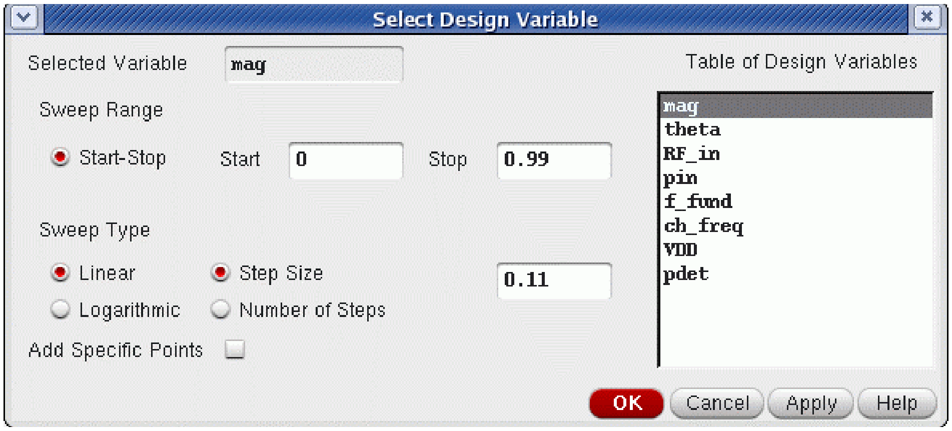
   The instance name is entered in the *Load Instance* field. Direct entry by typing is not allowed for this field.



6. Click the *Variable* button on the right of the *rho (Z)* field. Select the variable you entered in ADE.

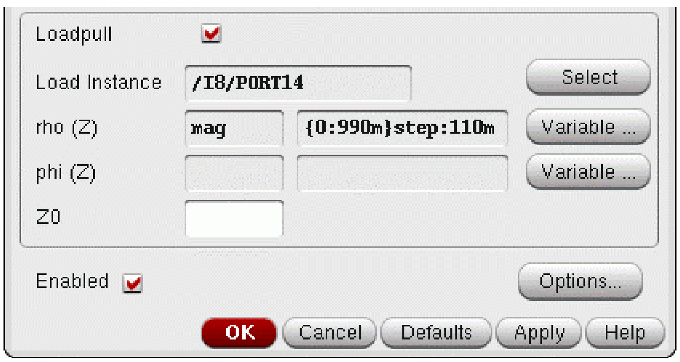
   This variable will be used to set the value of the reflection coefficient in the port.

7. Specify a sweep range, and set the spacing.



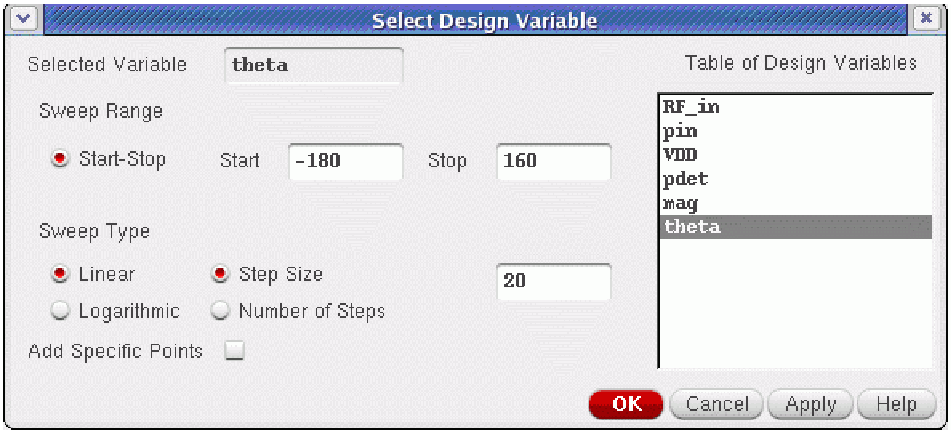
8. When completed, click *OK*.

The values are displayed in the *Choosing Analyses* form.



9. Click the *Variable* button on the right of the *phi (Z)* field. This sets the angle of the reflection coefficient.

10. In the *Select Design Variable* window, select the second variable you entered and specify the sweep limits.

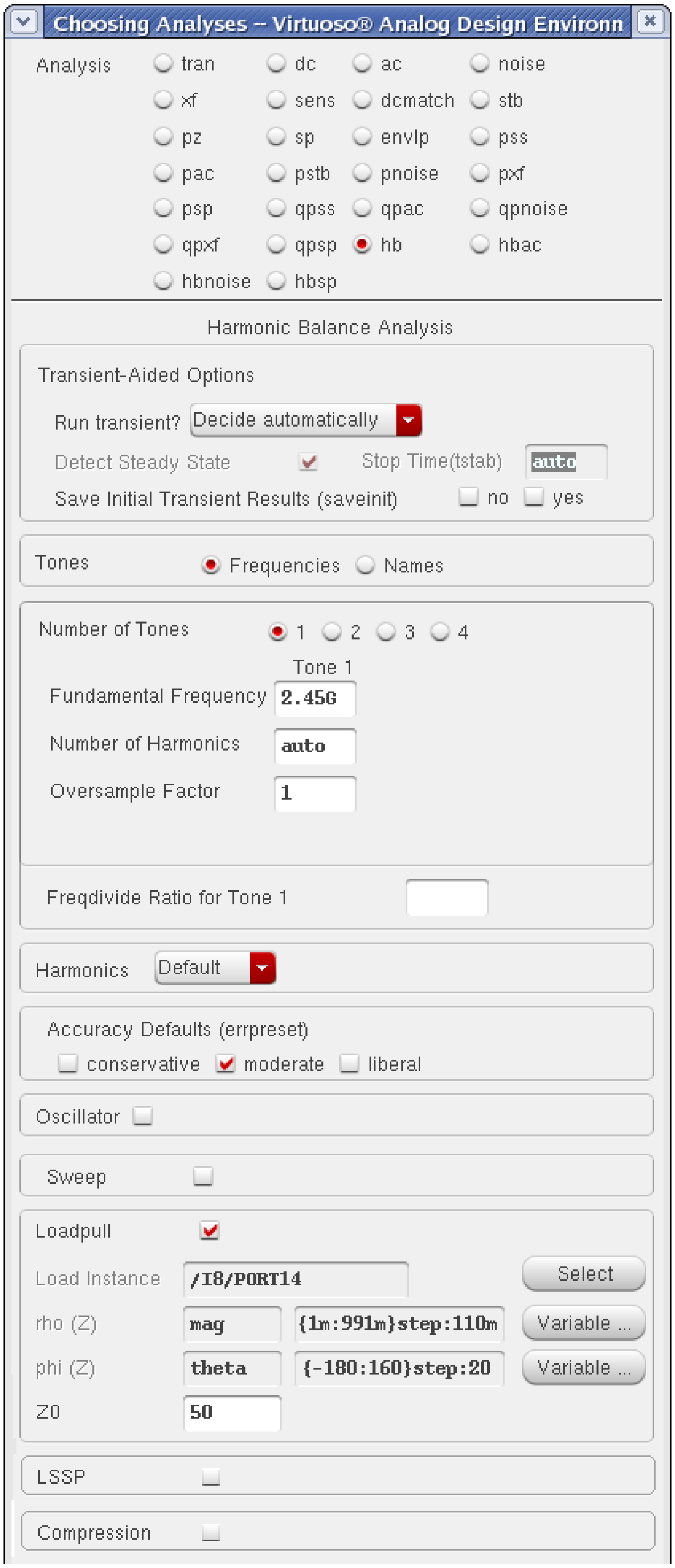


11. When completed, click *OK*.

The *Choosing Analyses* form is updated.

12. Specify the system resistance in the *Z0* field.

The *Choosing Analyses* form should appear similar to the one below.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran dc ac noise
xf sens dcmatch stb
pz sp envlp pss
pac pstb pnoise pxf
psp qpss qpac qpnoise
qpxf qpsp hb hbac
hbnoise hbsp
Harmonic Balance Analysis
Transient-Aided Options
Run transient? Decide automatically
Detect Steady State
Stop Time(tstab) auto
Save Initial Transient Results (saveinit) no yes
Tones Frequencies Names
Number of Tones 1 2 3 4
Tone 1
Fundamental Frequency 2.45G
Number of Harmonics auto
Oversample Factor 1
Freqdivide Ratio for Tone 1
Harmonics Default
Accuracy Defaults (errpreset)
conservative moderate liberal
Oscillator
Sweep
Loadpull
Load Instance /I8/PORT14
Select
rho (Z) mag {1m:991m}step:110m
Variable ...
phi (Z) theta {-180:160}step:20
Variable ...
Z0 50
LSSP
Compression

13. When completed, click *OK* and run the simulation in ADE.

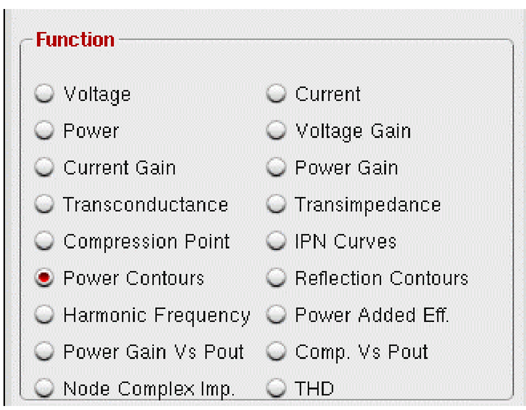
14. In ADE, select *Results - Direct Plot - Main Form*.

There are two methods of plotting loadpull curves.

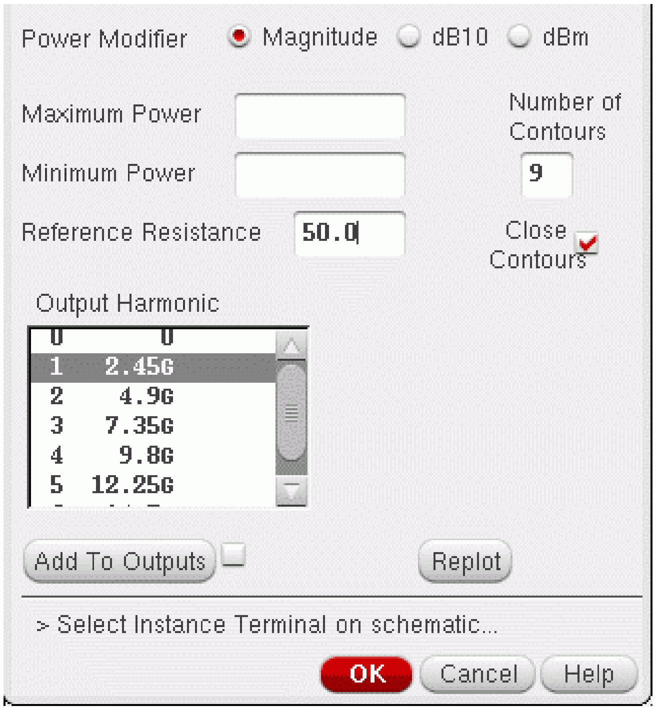
First Method

1. Select *pss or hb results*.

2. Select *Power Contours*.



3. Select *Magnitude*, *dB10*, or *dBm*.



4. Specify 2 more than the number of contour lines you want.

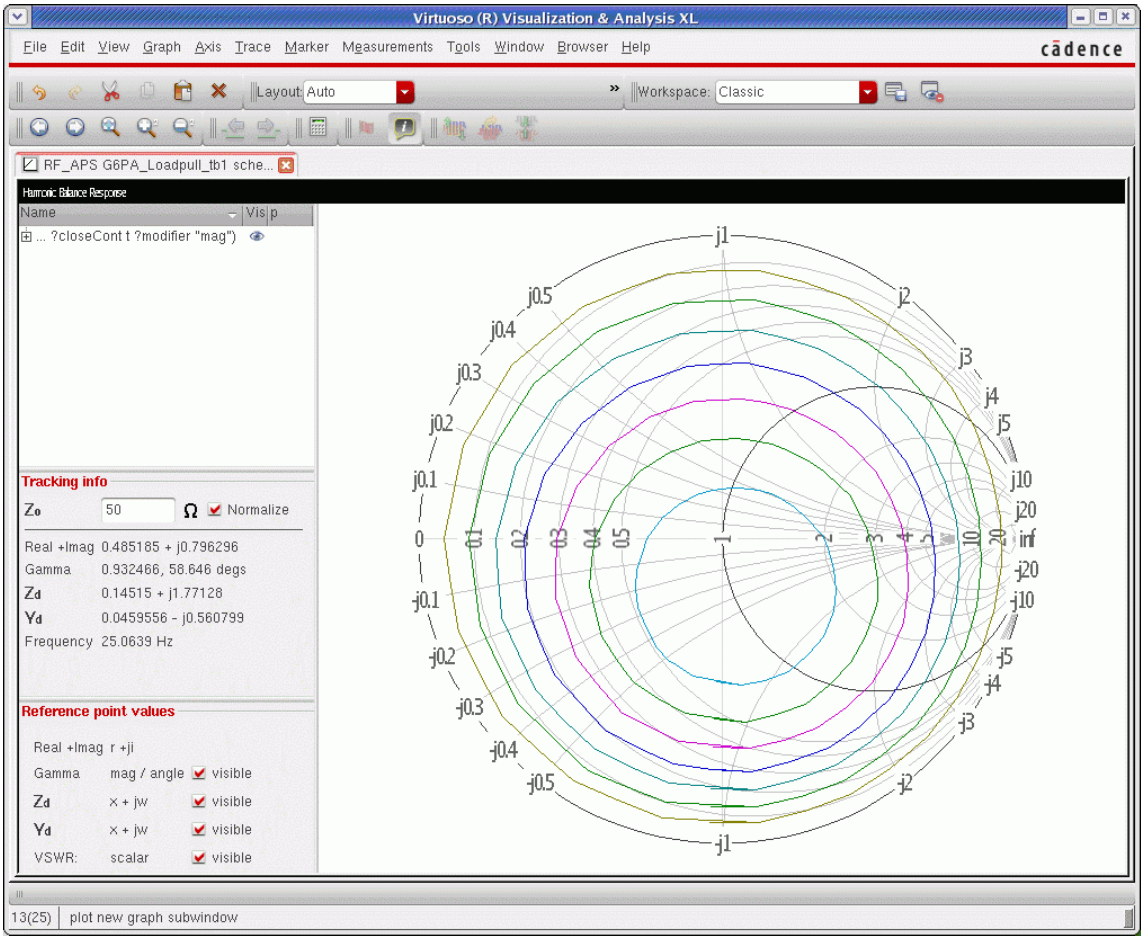
   The smallest and largest power occurs at a single point which is plotted, but are very difficult to see in the waveform tool.

5. Select the *Close Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.

6. Select the first harmonic.

7. Select the top terminal of the top port in the triplexer.

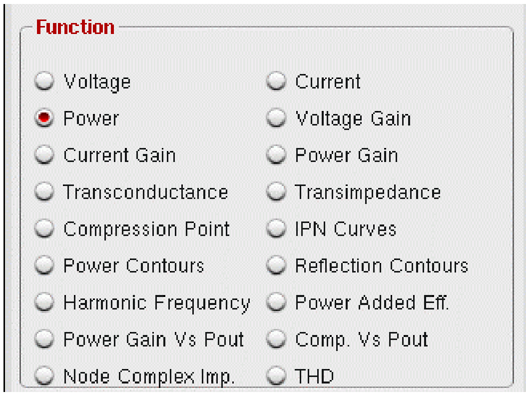
The power contours appear.



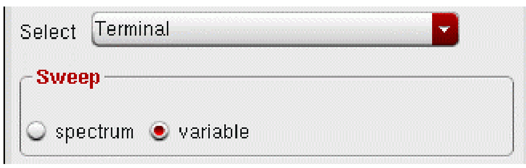
Second Method

1. Select *pss* or *hb results*.

2. Select *Power*.

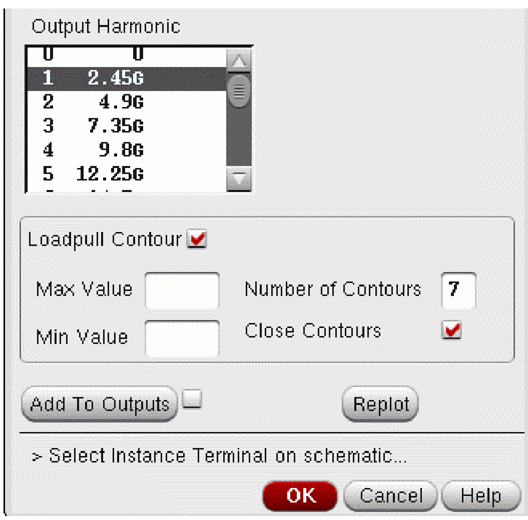


3. Select *variable*.



4. Select *Magnitude*, *dB10*, or *dBm*.

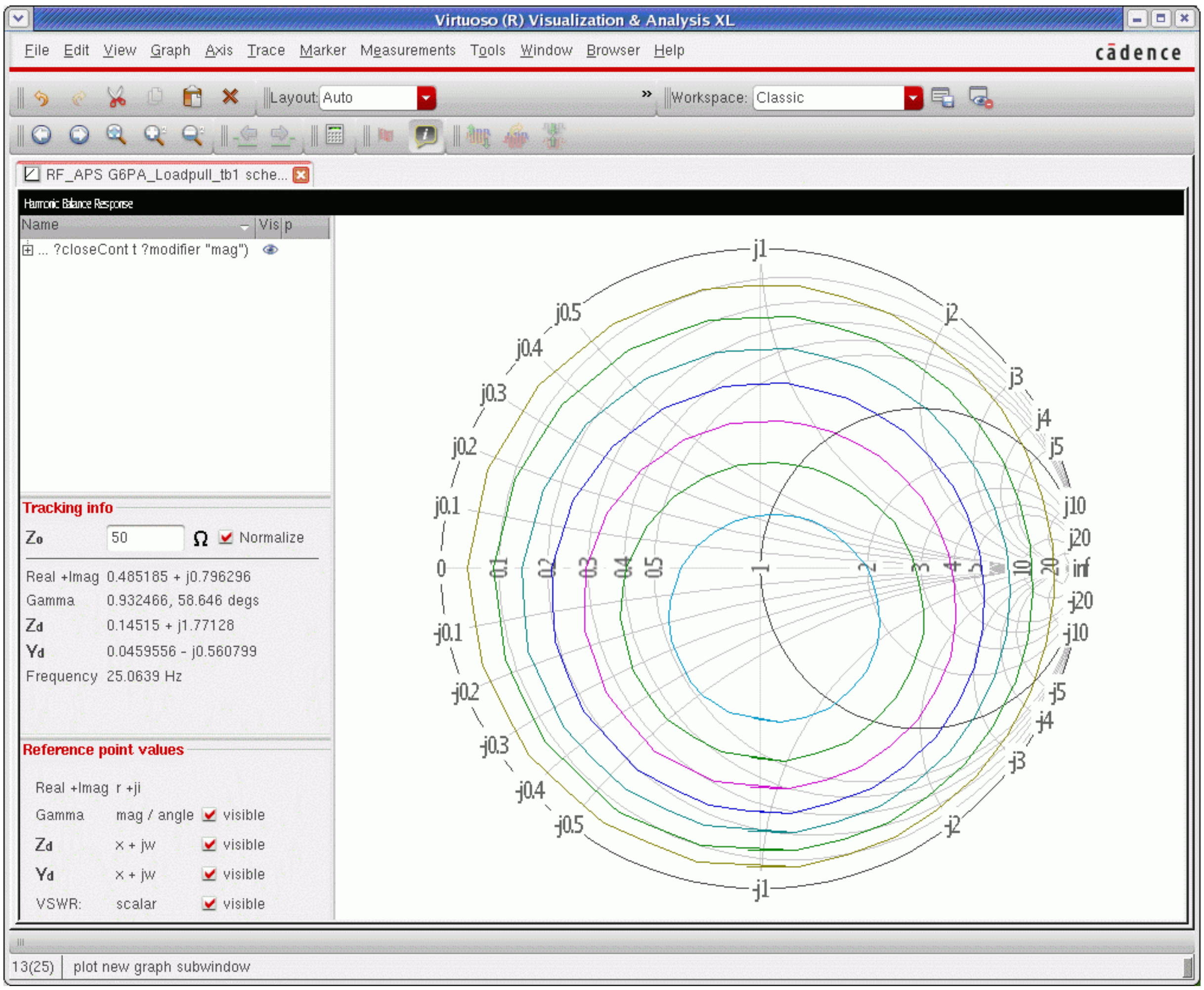
5. Select the *harmonic number*.



6. Select *Loadpull Contour*.

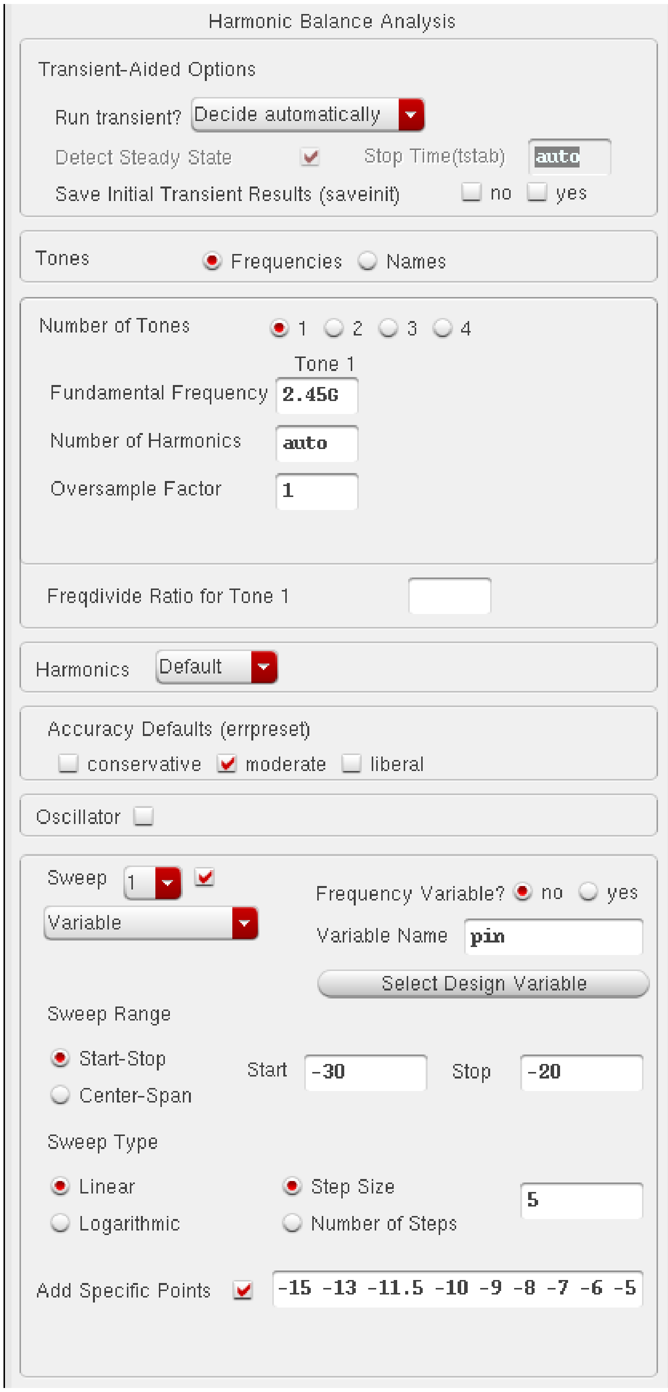
7. Specify 2 more than the number of curves you want.
8. Select the *Closed Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.
9. Select the top terminal of the top port of the triplexer circuit.

   The loadpull curves appear in the waveform tool.



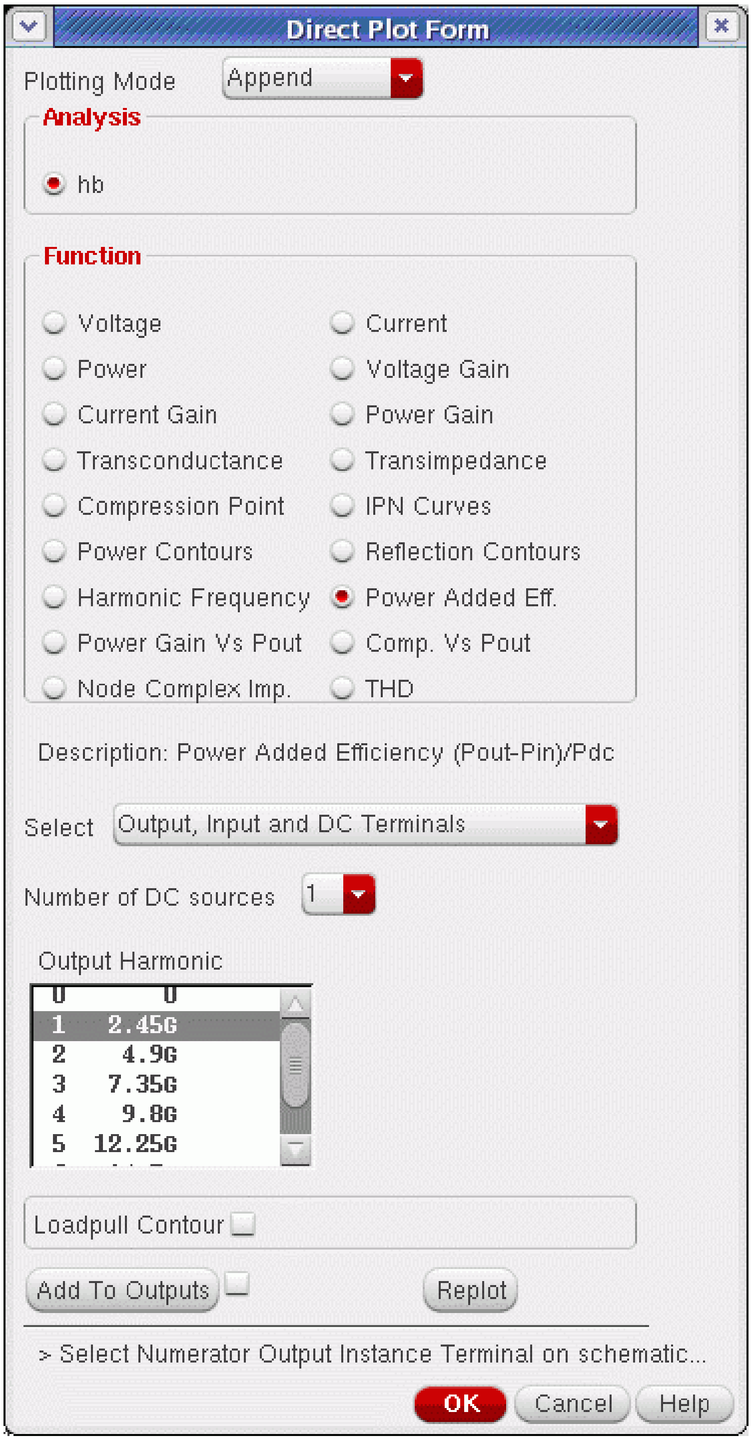
Power-Added Efficiency

1. Set up an hb run as desired. A sweep is shown below.



2. Run the simulation.

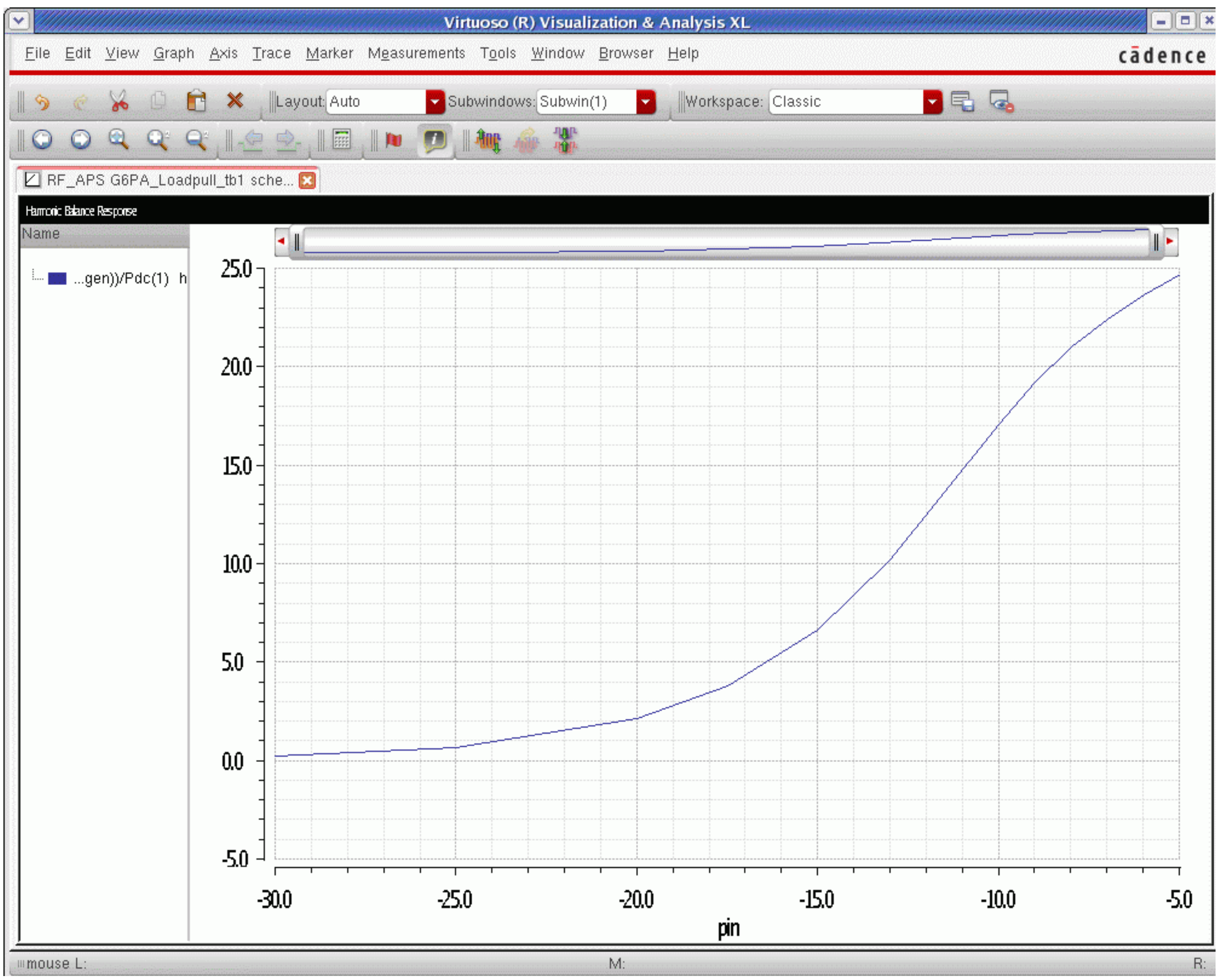
3. When the simulation completes, select *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed.



4. Select *Power Added Eff.*

5. Set the number of DC sources in the circuit you want to include for the PAE measurement.
6. Select the frequency of the output.
7. Select the top terminal of the load in the schematic.
8. Select the top terminal of the input port.
9. Select the positive terminal of the DC source that supplies power to the amplifier. If you set multiple sources, select the positive terminals of all the sources.

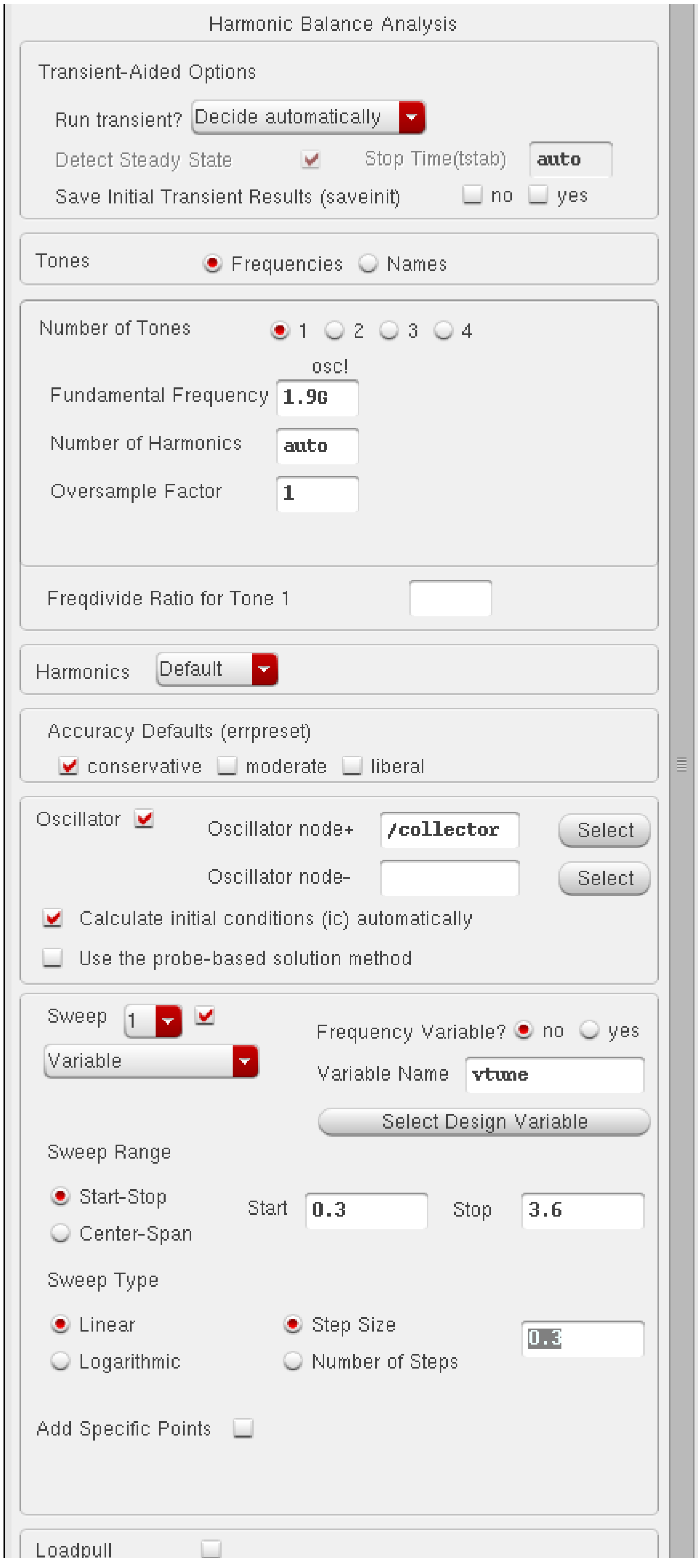
   The PAE is plotted in the waveform tool.



Oscillators

On the *Choosing Analyses* form:

1. Set the *Run Transient?* selection appropriately.



2. For oscillators, selecting the *Detect Steady State* check box is strongly recommended. This will run tstab until steady-state is reached, and then switch to harmonic balance. When *Decide automatically* is set for *Run Transient?*, *Detect Steady State* is always on.

3. Specify a frequency that is within 0.5 times the actual oscillation frequency and 1.5 times this frequency.

4. Set the harmonics and the oversample factor as needed for your oscillator. *auto* harmonics is available when *Run Transient?* is set to *Decide automatically* or *yes*.

5. Always select *conservative*.

   For LC and crystal oscillators, no further action is necessary. For ring oscillators, specify the initial conditions in ADE to force one stage to the high or low state.

6. If you want to see the startup waveform, select *yes* for saveinit.

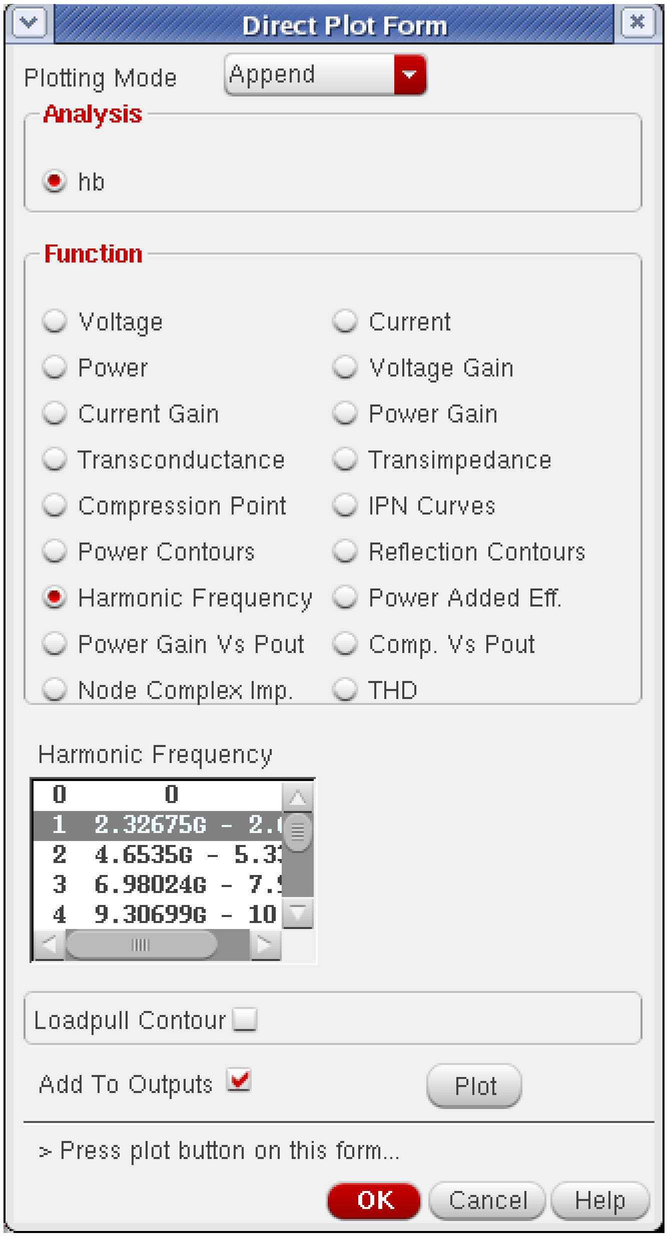
7. Select any net with the oscillator signal present for the oscillator node.

8. If you want the output frequency versus tuning voltage, define an appropriate sweep for your oscillator.

9. Run the simulation.

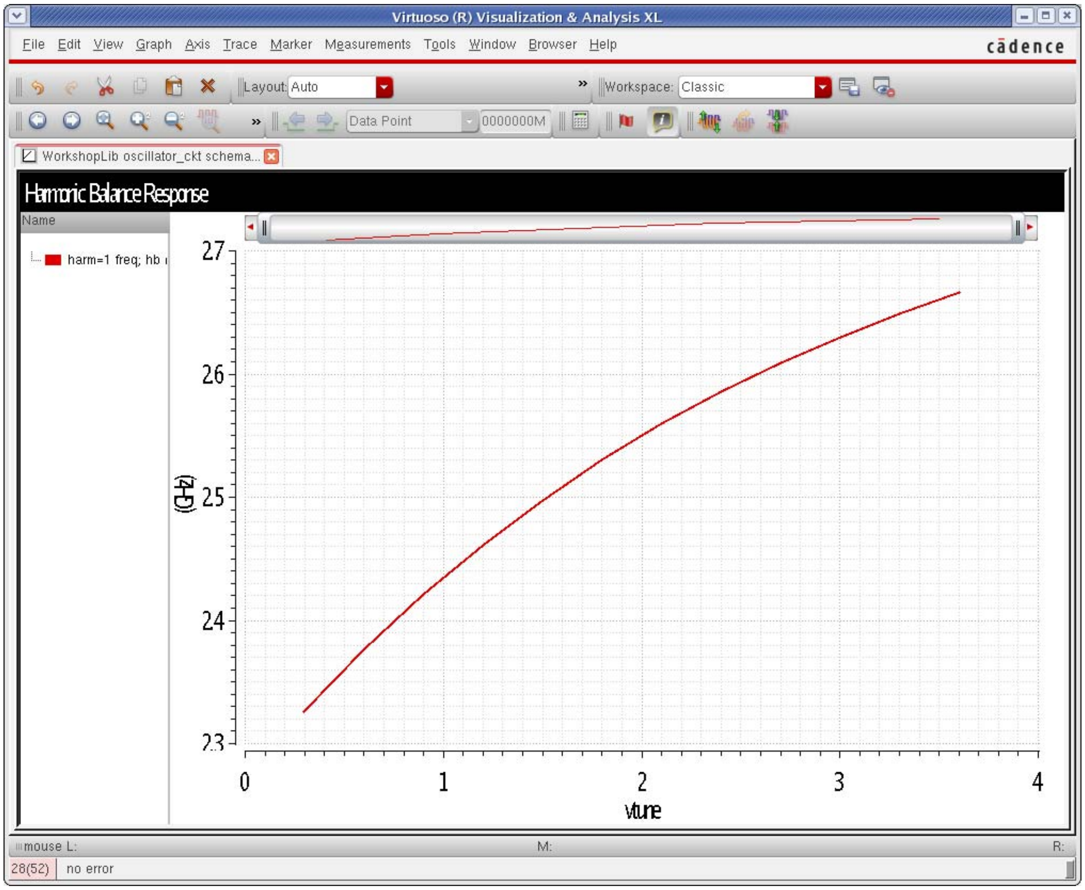
Oscillator Tuning Range

To plot the tuning range, on the *Direct Plot Form*:



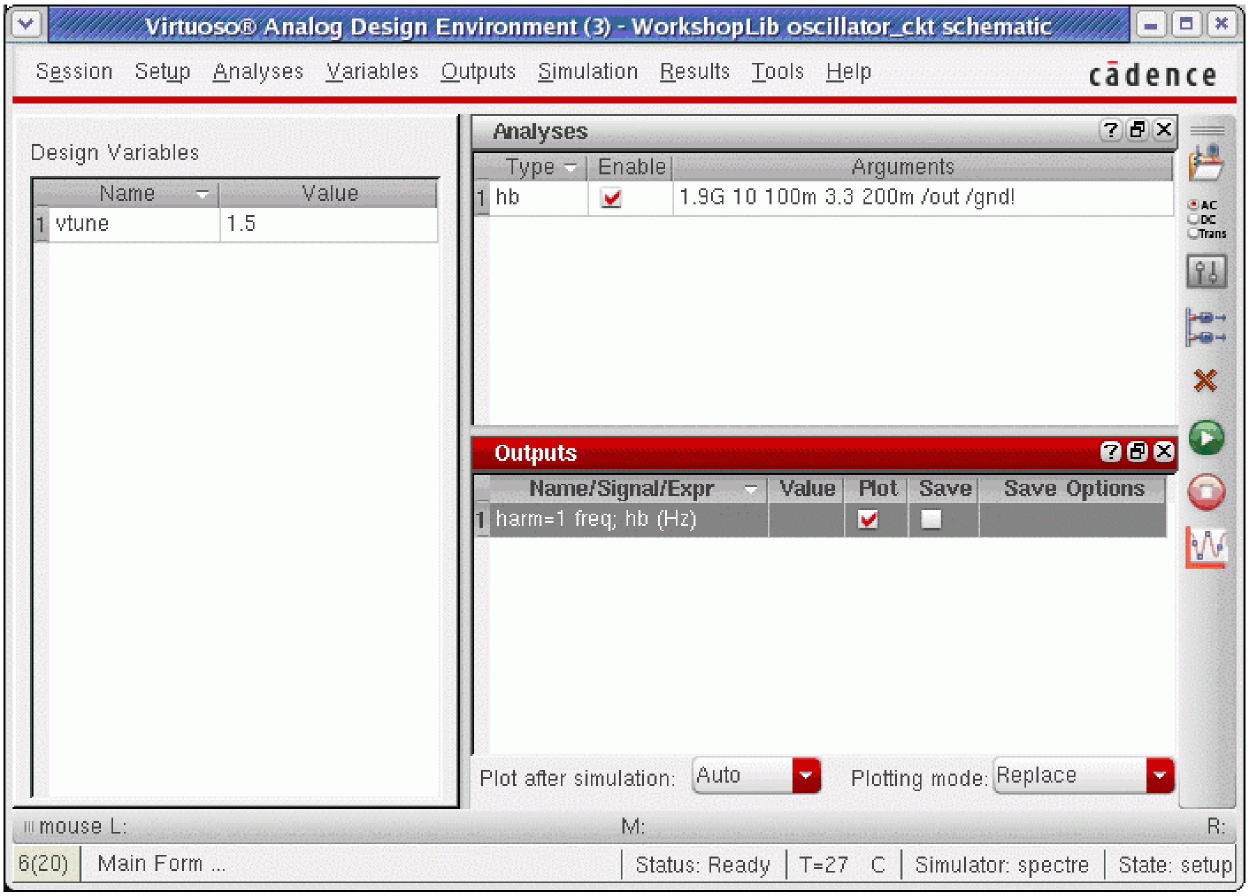
1. Click the *Harmonic Frequency* radio button.
2. Select the *Add to Outputs* check box.
3. Select the desired harmonic number to plot from the *Harmonic Frequency* field.
4. Click *Plot*.

The tuning range plot appears.



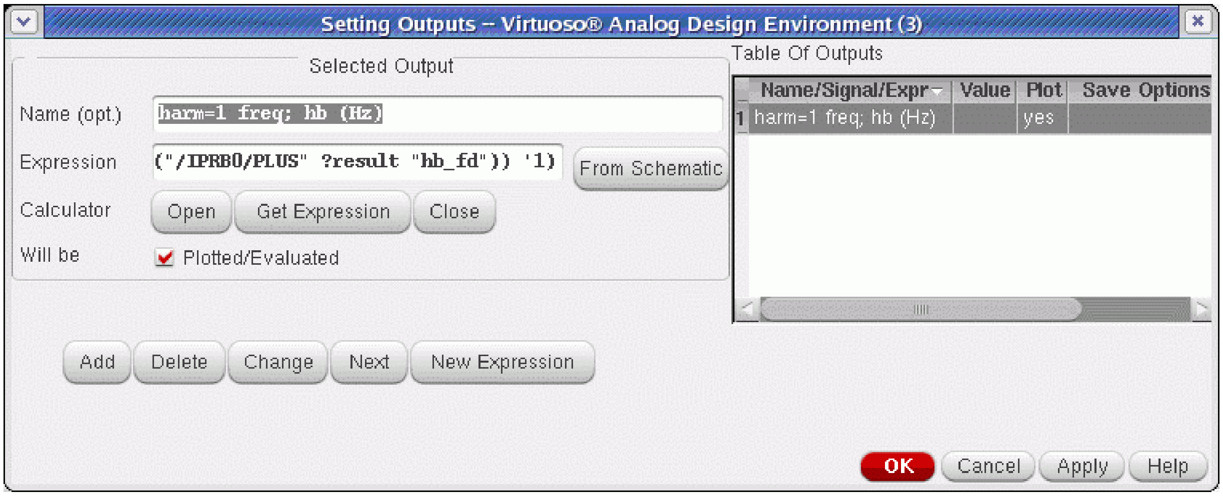
Plotting Modulation Sensitivity

From plotting the oscillator frequency as shown below, an expression is added in the ADE window.



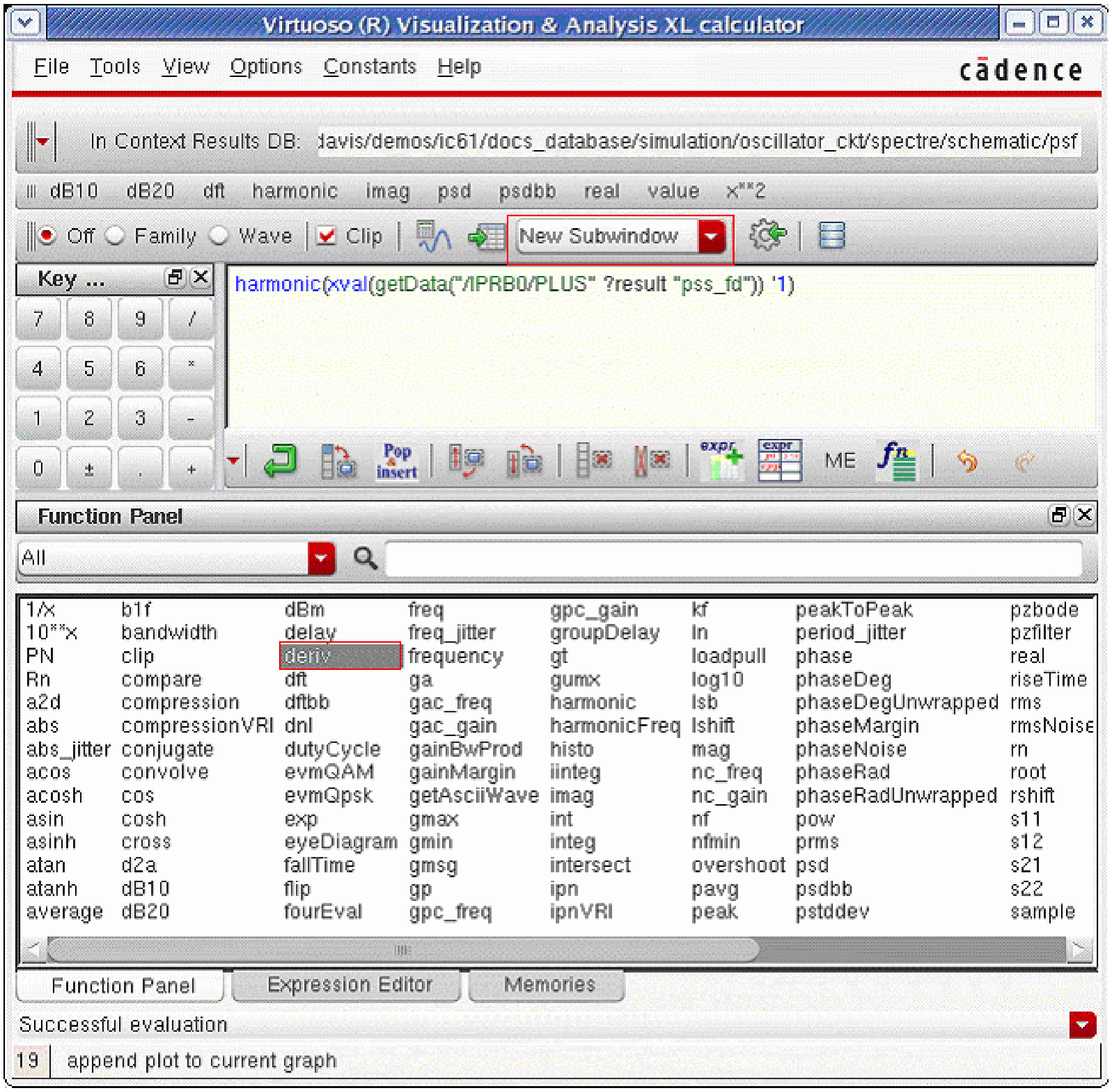
1. Double-click the expression.

The *Setting Outputs* window is displayed.



2. Click *Open*.

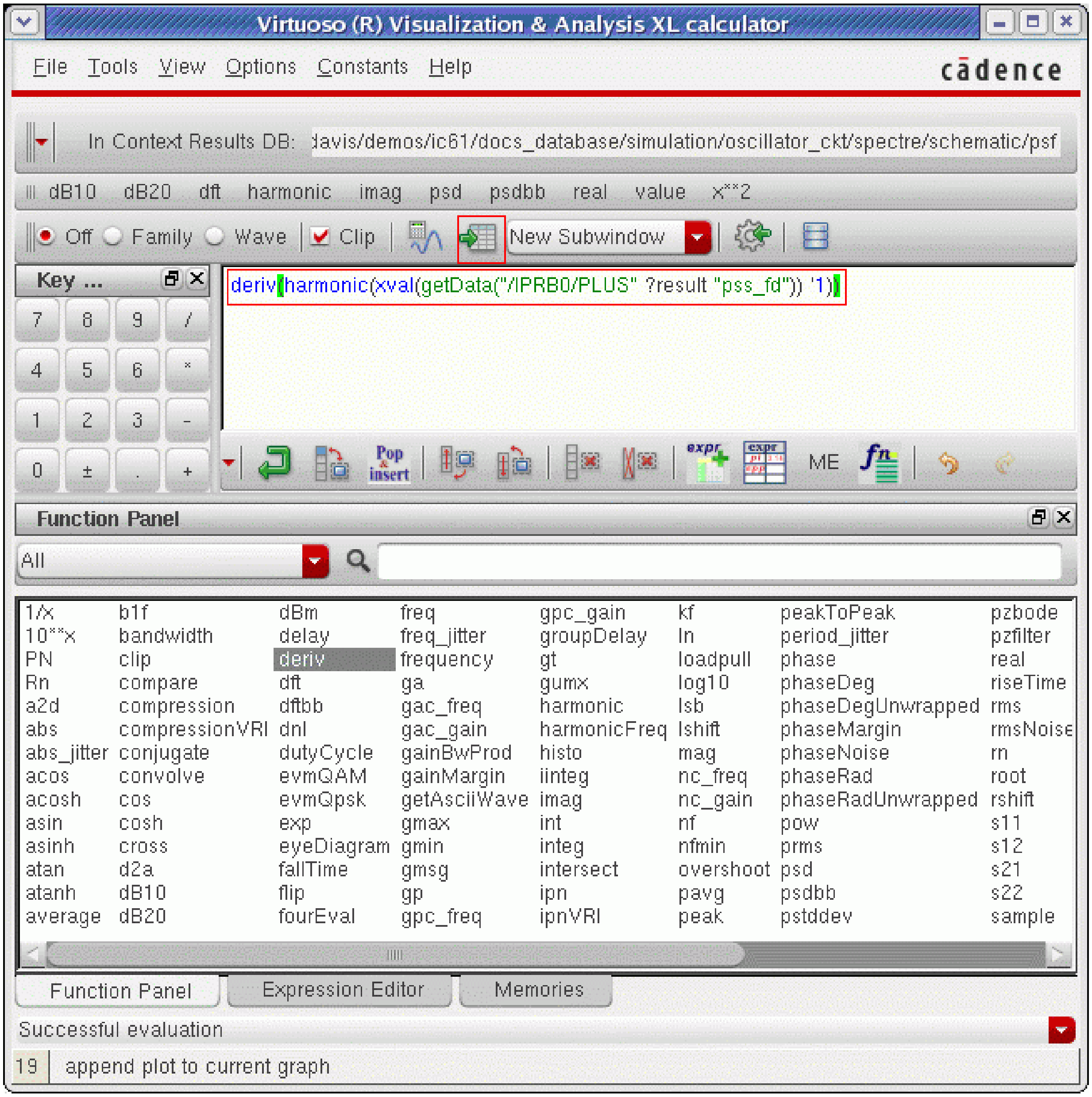
The Calculator starts with the expression in the buffer. This expression will be different for different circuits.



3. Set the plotting mode to *New Subwindow*.

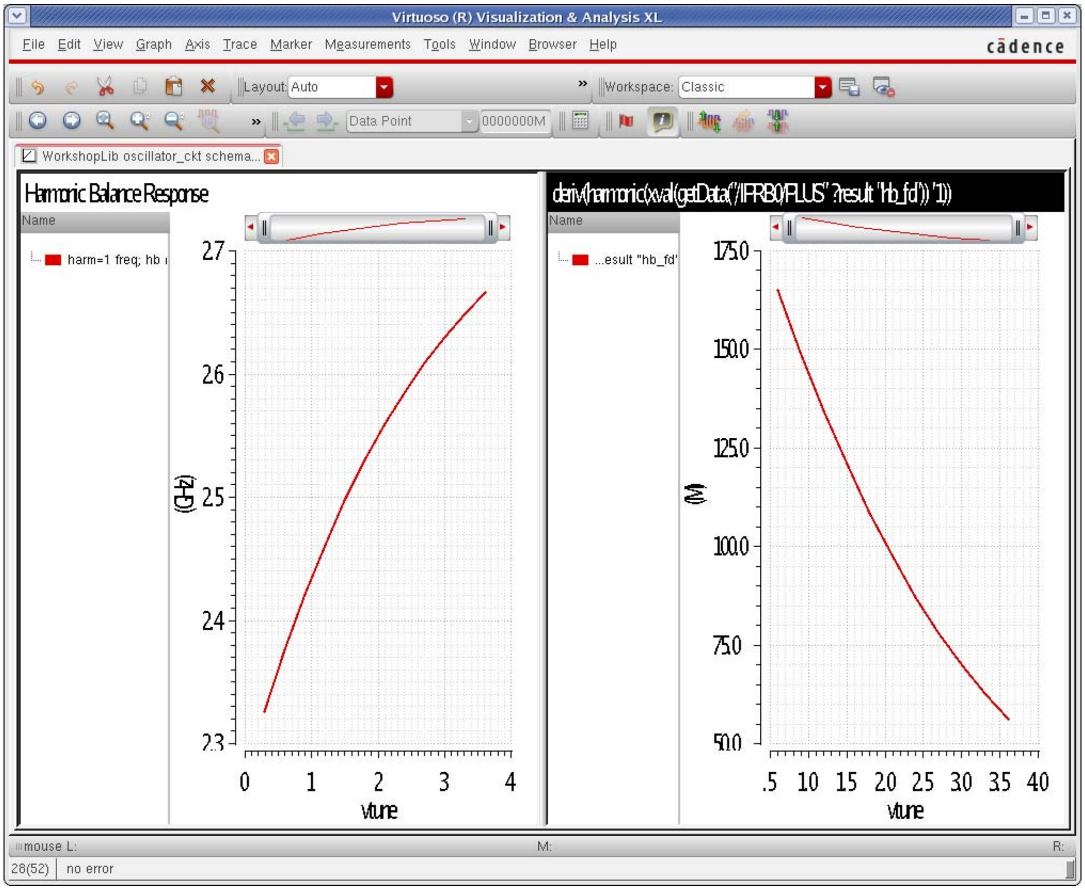
4. Select the *deriv* function.

The expression changes in the buffer.



5. Click the *Plot* icon.

The modulation sensitivity appears in the new subwindow.



Oscillator Tuning Mode Analysis

The oscillator tuning mode analysis tunes the oscillator so that phase noise can be measured at the desired oscillation frequency. It is supported in both harmonic balance and shooting methods and is typically used in Monte Carlo simulations to obtain the phase noise.

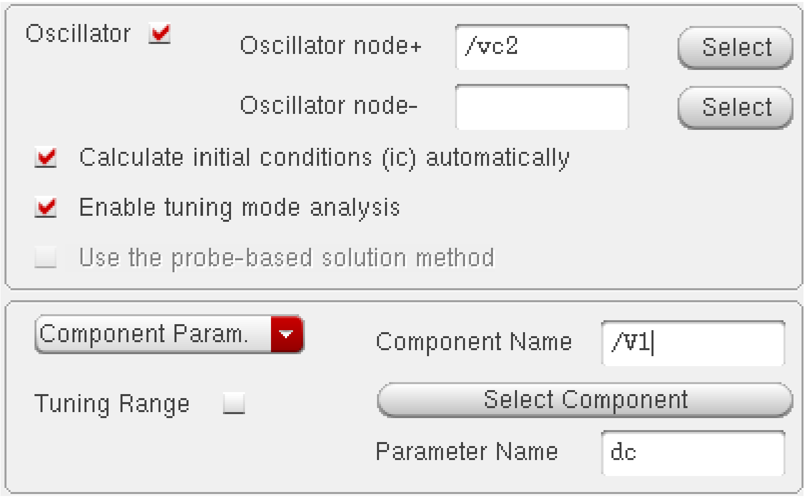
In oscillator tuning mode analysis, the oscillator frequency is treated as the target frequency, and the value of a circuit parameter is obtained to achieve the target frequency. The circuit parameter can be an arbitrary netlist variable or an instance parameter.

1. Open the hb *Choosing Analyses* form.
2. Set up the target frequency in the *Fundamental Frequency* field.
3. Select the *Oscillator* check box.

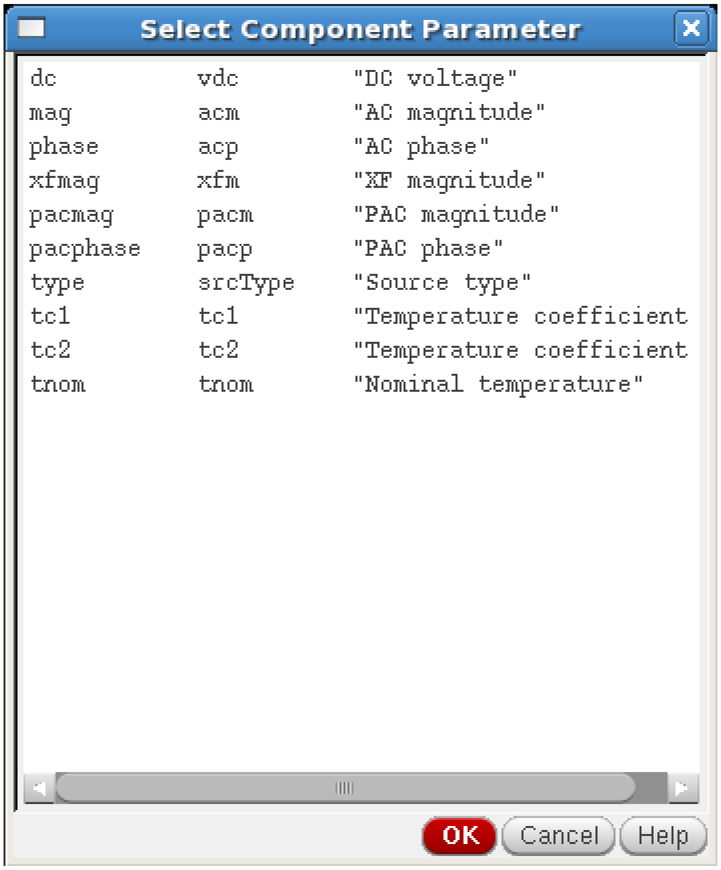
4. Specify the output node from the `Oscillator node+` field. Alternatively, click *Select* and choose the output node from the schematic. Here, `vc2` is selected.

5. Select the *Enable tuning mode analysis* check box.

6. Choose the parameter that needs to be tuned from the drop-down list. Here, *Component Param.* has been selected.

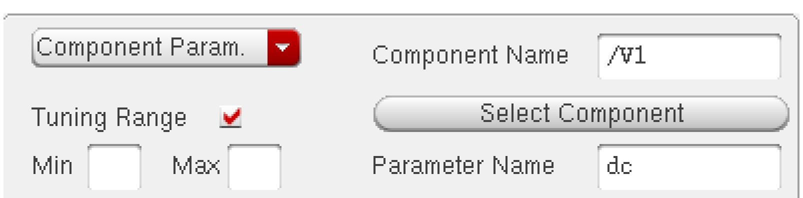


7. Specify the name of the parameter in the *Component Name* field. Alternatively, click *Select Component* and select the parameter from the schematic. When you select a parameter from the schematic a pop-up window appears that contains a list of selectable parameters.



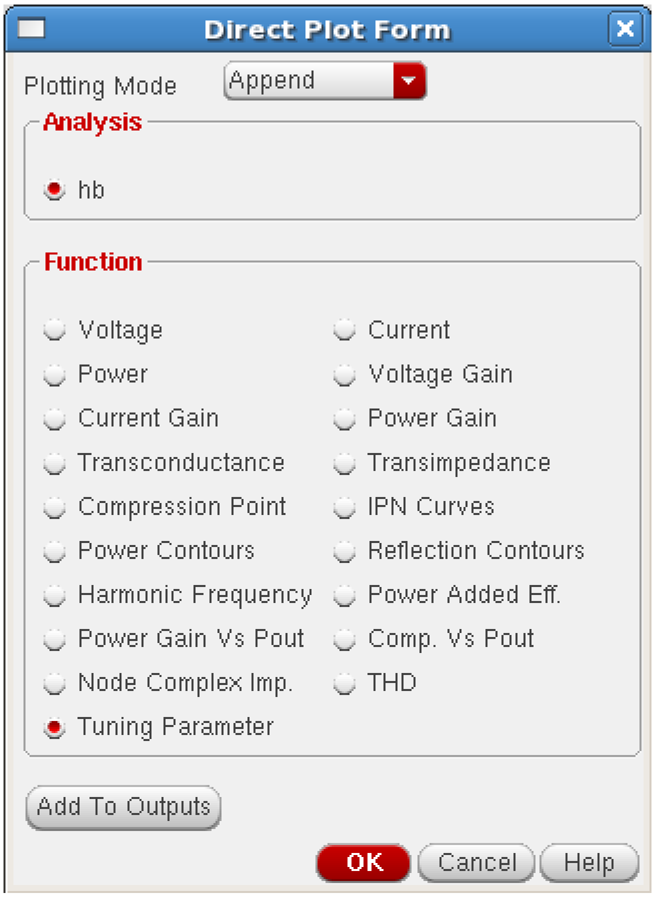
8. Select a parameter from the list. The *Parameter Name* field is automatically populated with the name of the parameter (*dc* in the figure above).

9. Select the *Tuning range* check box and specify the minimum and maximum ranges for the parameter, if required.



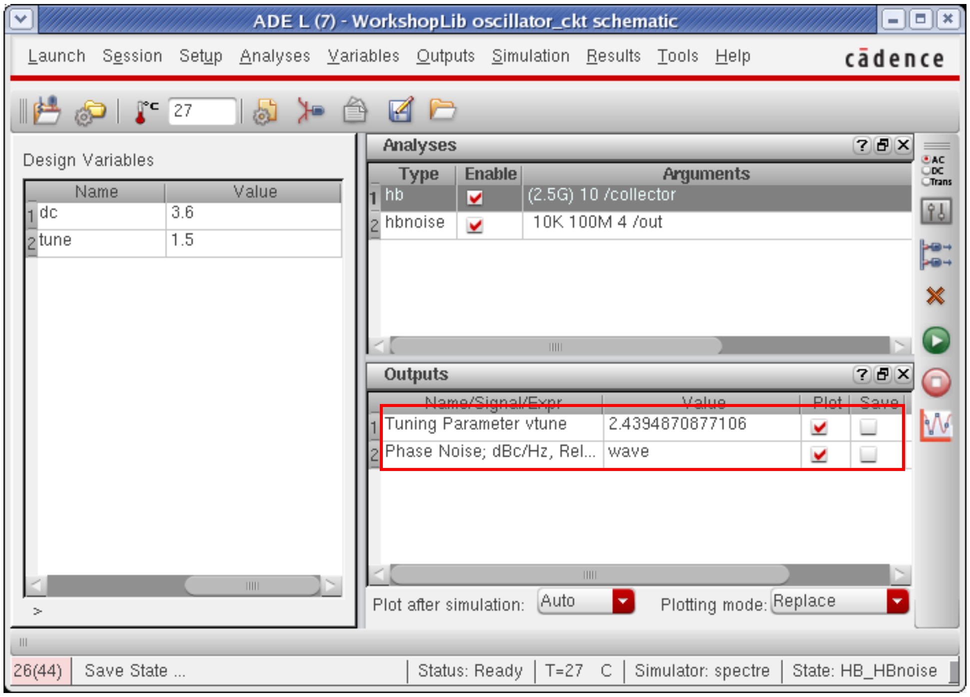
10. Run the simulation.

11. After the simulation completes, click *Results - Direct Plot - Main Form* to display the *Direct Plot Form*.



12. Select *Tuning Parameter* from the *Function* section.
13. Click *Add to Outputs*.
14. Click *OK*.

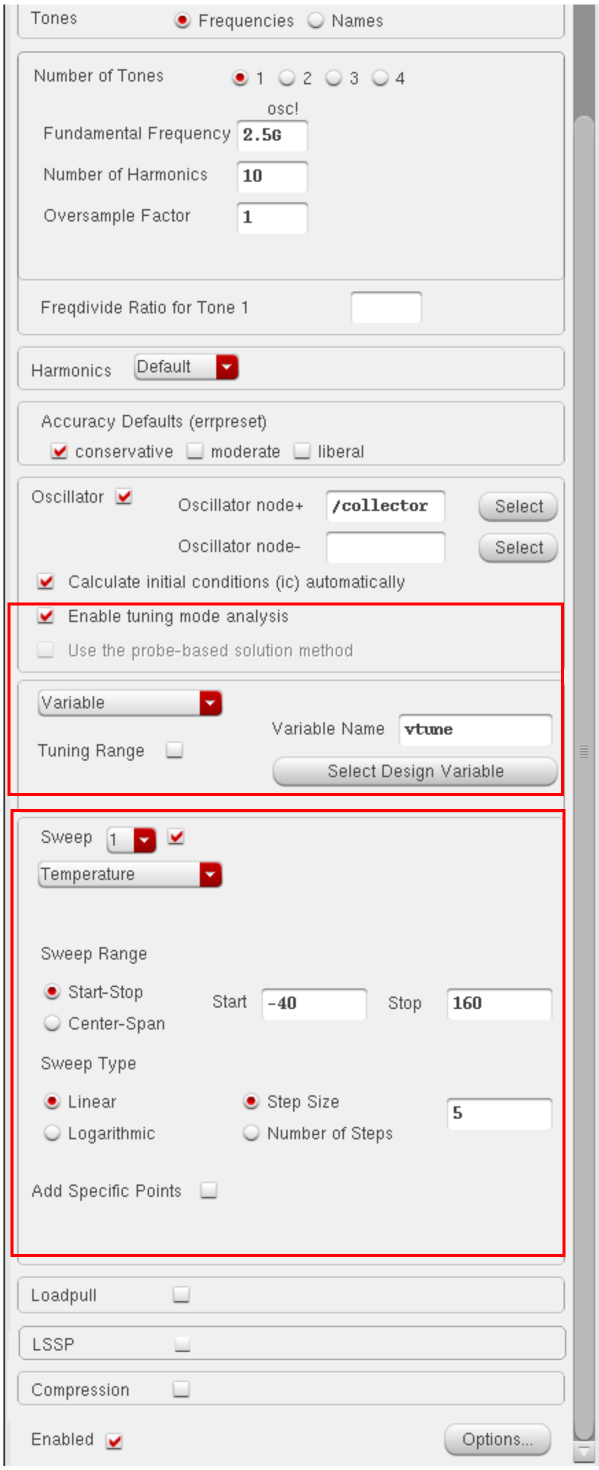
The value of the tuning parameter is added in the *Outputs* section of the ADE window, as shown below.



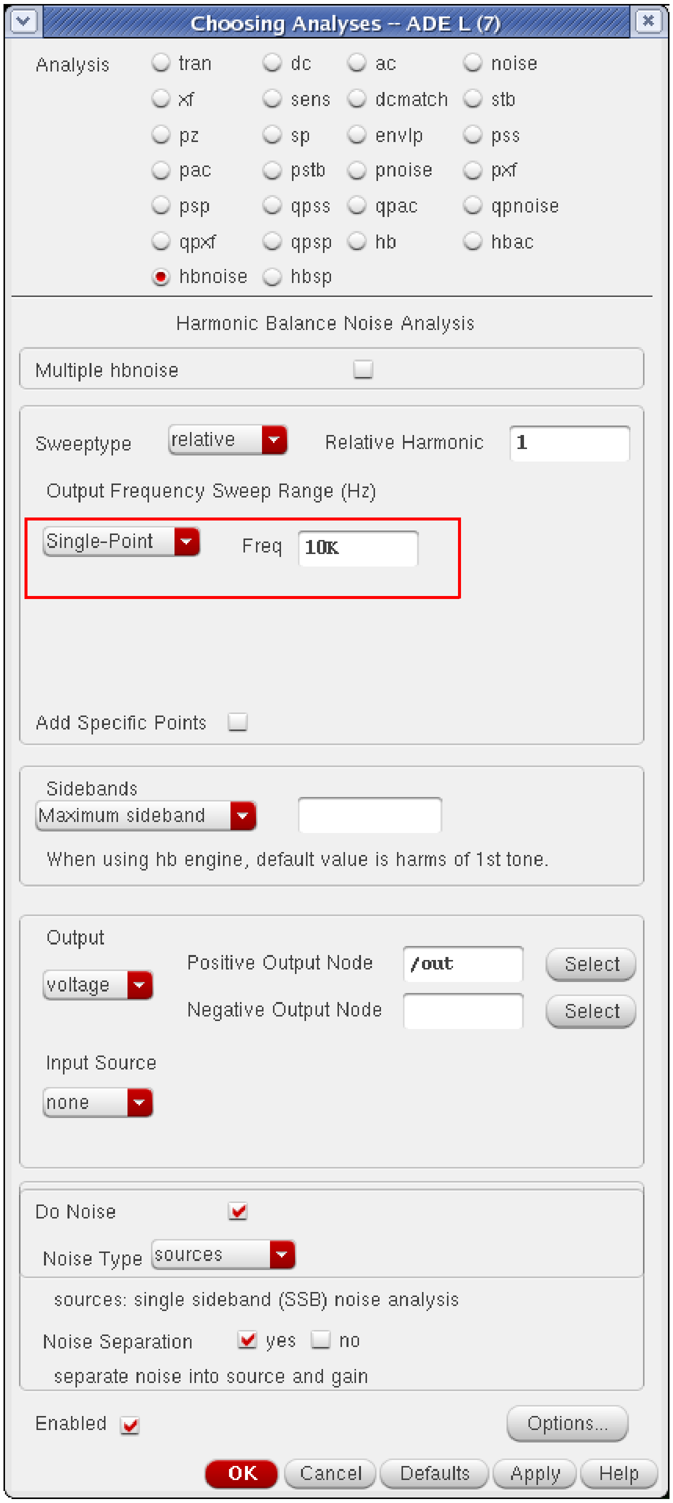
Oscillator Tuning Mode With Sweeps

1. Sweeps and Monte Carlo are also supported with tuning mode. Below is an example of a sweep of temperature with tuning mode and a phase noise measurement.

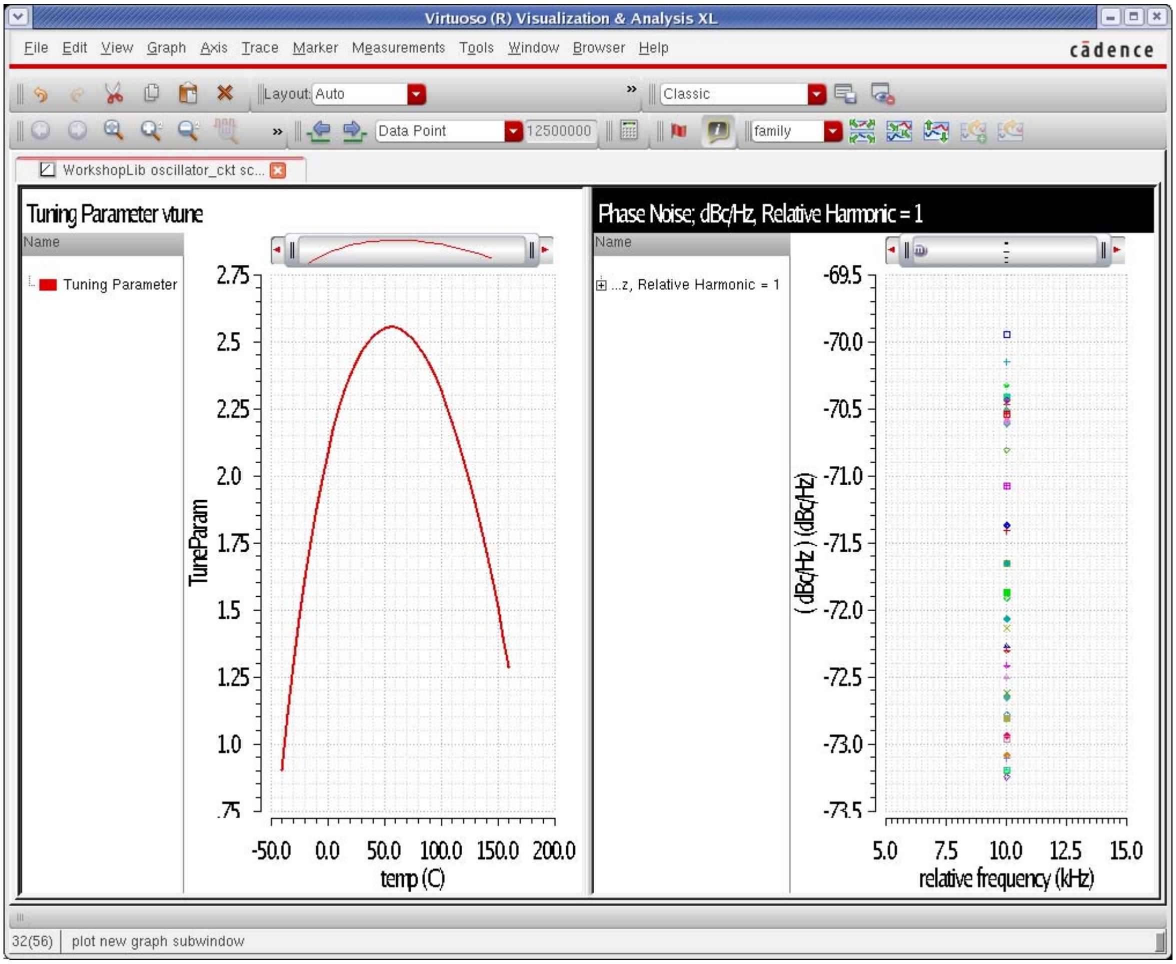
2. Enable sweeps in the hb Choosing Analyses form. In this example, temperature is swept from -40 t0 160C. Tuning mode is enabled as before.



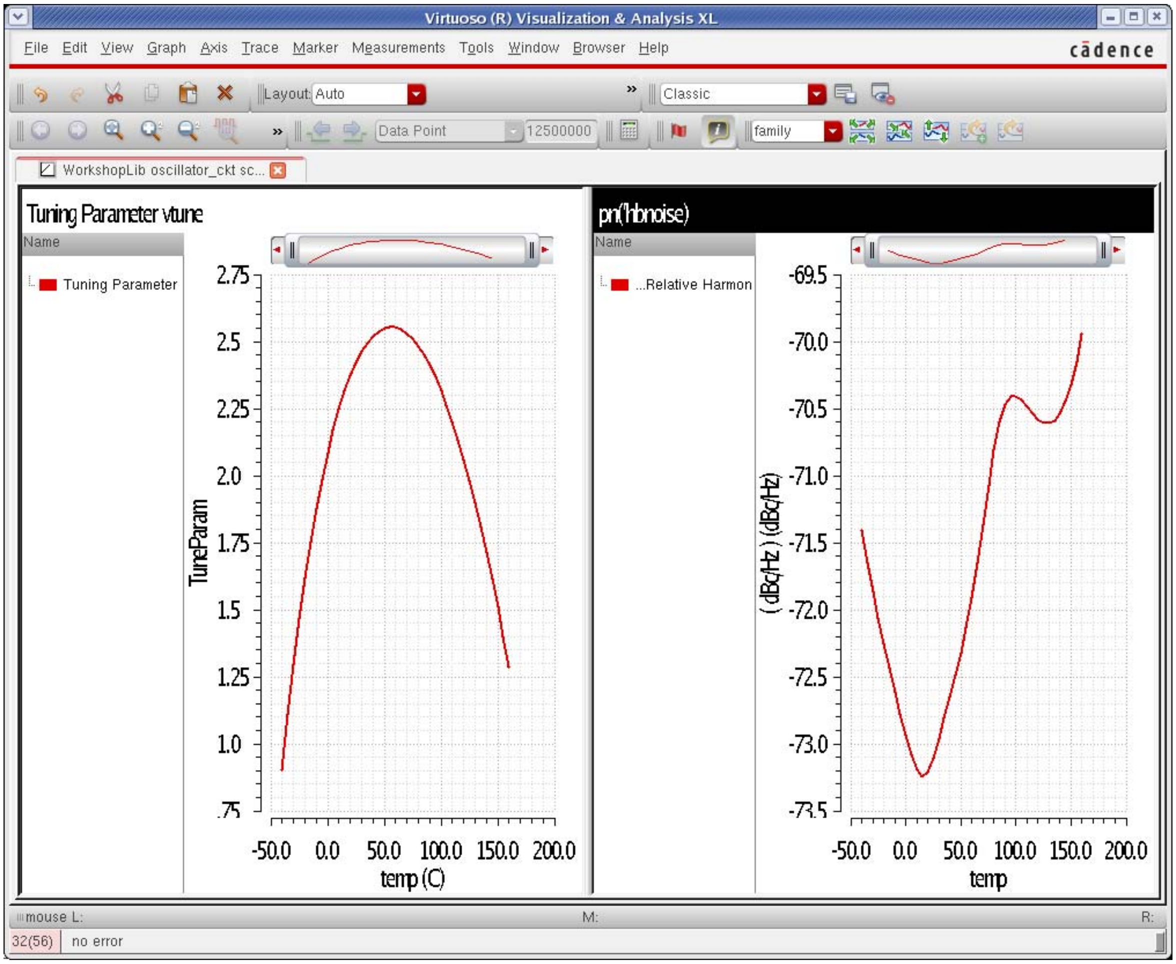
3. Because of the sweep, hbnoise has been changed to a single frequency offset, which simplifies plotting of phase noise after the sweep.



4. Now run the simulation. Because the tuning parameter and phase noise have already been added to the ADE outputs section in the previous section, the tuning voltage and the phase noise are plotted after the simulation.



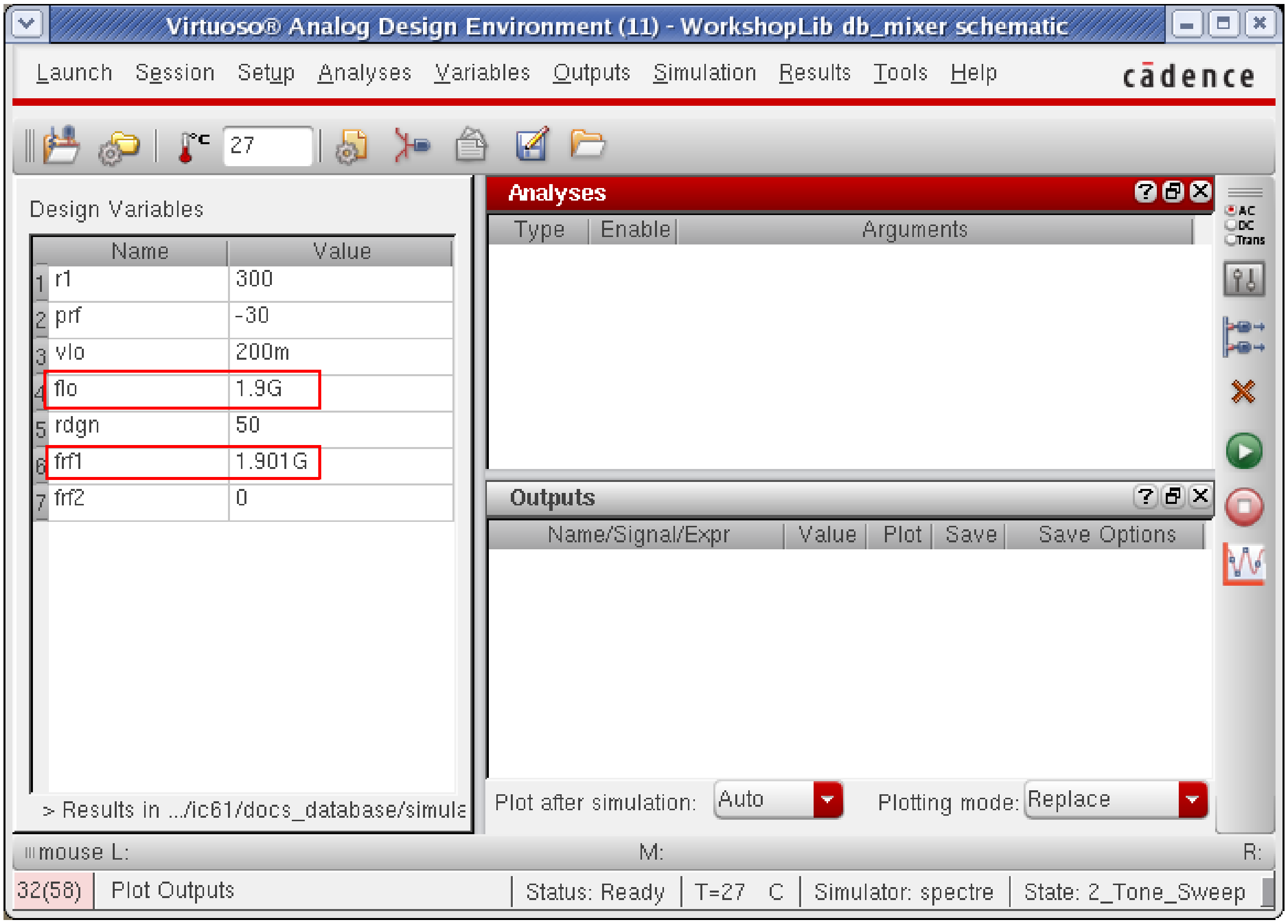
5. To make the phase noise plot more useful, select the X Axis, right-click, and select Swap Sweep Var. In the window that pops up, select OK. Then delete the original phase noise plot.



6. The trend in tuning voltage and phase noise is plotted.

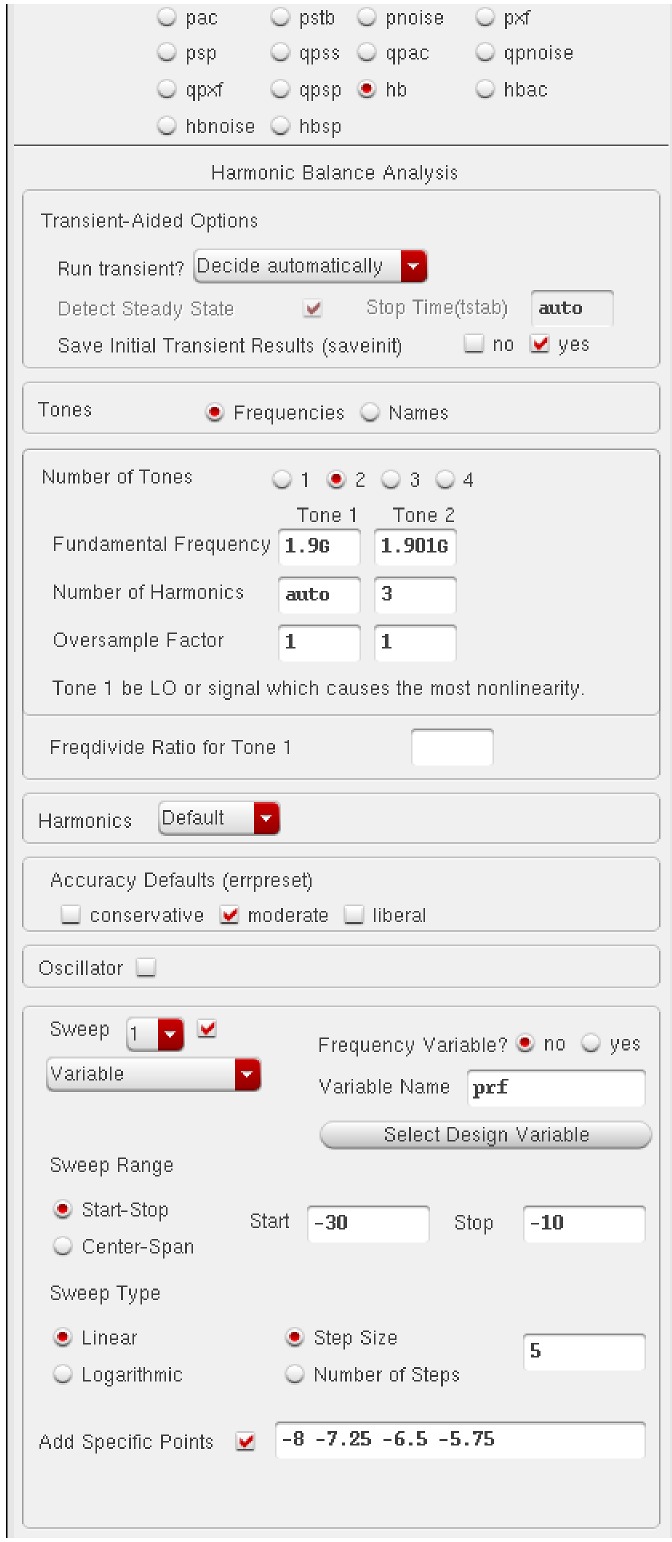
Two Input Frequencies

To enable an additional frequency, set the variable in ADE to the frequency you want instead of 0 (zero). In the window below, the variable frf1 has been set to 1.901G.

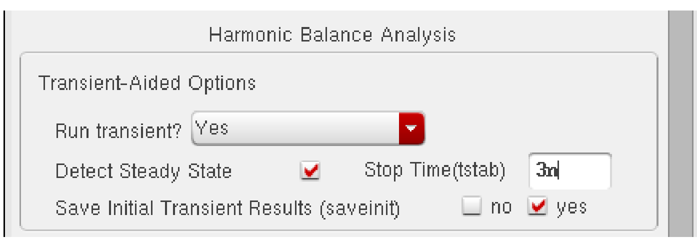


To set the 2-tone hb analysis:

1. Open the *Choosing Analyses* form from the Analog Design Environment (ADE) window.

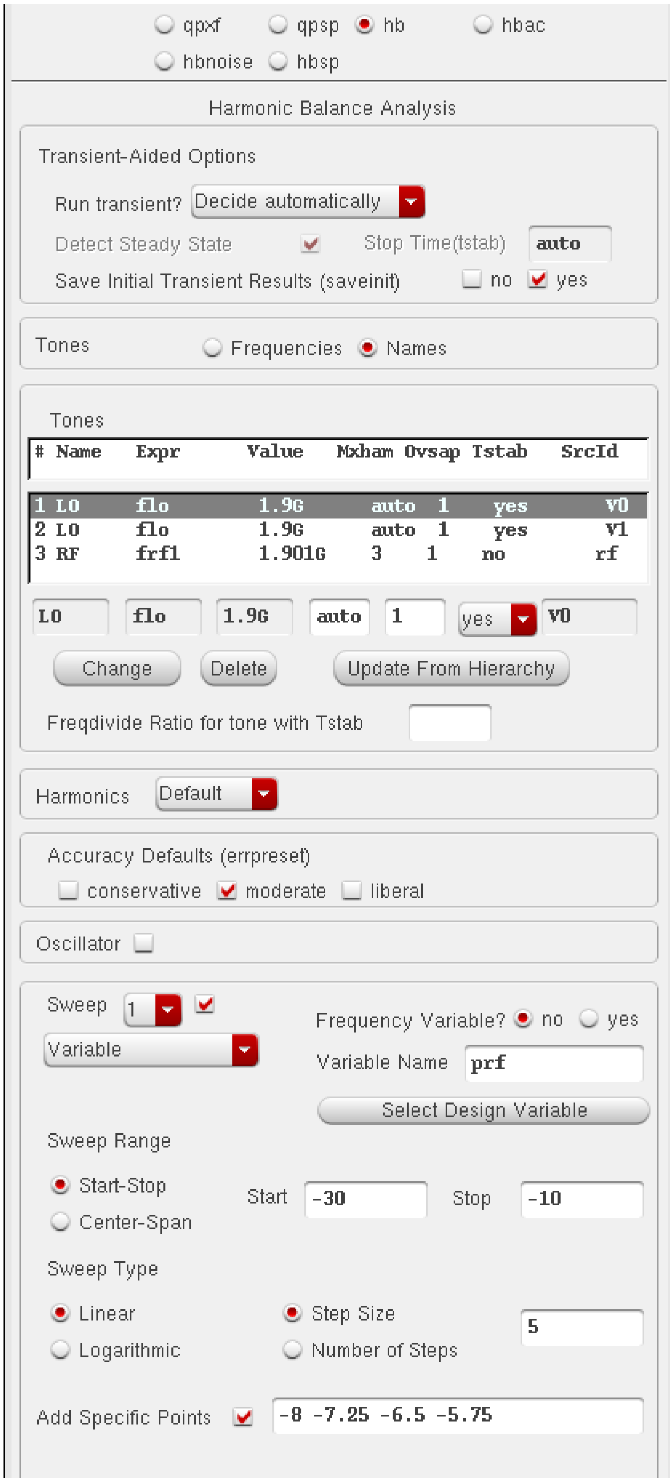

pac pstb pnoise pxf
psp qpss qpac qpnoise
qpxf qpsp hb hbac
hbnoise hbsp
Harmonic Balance Analysis
Transient-Aided Options
Run transient? Decide automatically
Detect Steady State
Stop Time(tstab) auto
Save Initial Transient Results (saveinit) no yes
Tones Frequencies Names
Number of Tones 1 2 3 4
Tone 1 Tone 2
Fundamental Frequency 1.9G 1.901G
Number of Harmonics auto 3
Oversample Factor 1 1
Tone 1 be LO or signal which causes the most nonlinearity.
Freqdivide Ratio for Tone 1
Harmonics Default
Accuracy Defaults (errpreset)
conservative moderate liberal
Oscillator
Sweep 1
Variable
Frequency Variable? no yes
Variable Name prf
Select Design Variable
Sweep Range
Start-Stop
Center-Span
Start -30
Stop -10
Sweep Type
Linear
Logarithmic
Step Size
Number of Steps
5
Add Specific Points -8 -7.25 -6.5 -5.75

2. Decide whether you want to set the harmonics manually or automatically for the first tone. For auto tstab, and auto selection of harmonics for the first tone, select *Decide automatically* from the *Run transient?* drop-down list.
3. You can specify either the frequencies of the inputs, or obtain a list of the source names and frequencies in the schematic. *Frequencies* are shown above.
4. Click the appropriate radio button to select the number of inputs.
5. Set the frequencies, number of harmonics, and oversample factor. For an automatic calculation of the number of harmonics for the first tone, set the number of harmonics to auto in Tone1.
6. If you have a frequency divider in your circuit, set the divide ratio. This assumes that the Tone1 input will be divided in frequency.
7. If you want to enable a frequency cut, choose *Select* from the *Harmonics* drop-down list box and set a value for *MaximOrder*.
8. Set the accuracy you want for the analysis.
9. If you want to manually set transient-aided hb, select *Yes* from the *Run Transient?* drop-down list and set a time for the transient tin the *Stop Time (tstab)* field. If you want to see the startup waveform, select *yes* for saveinit.

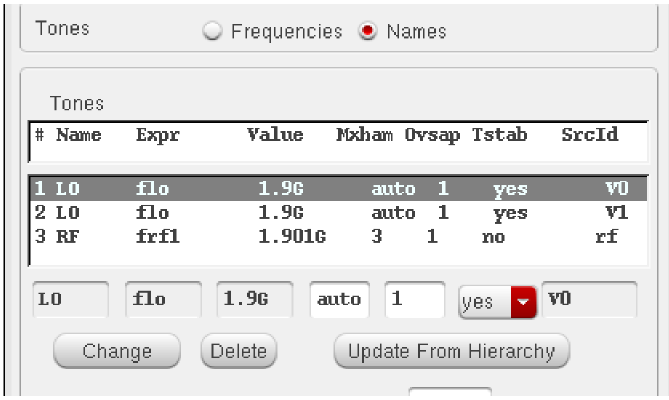


10. Enable sweeps by checking the *Sweep* checkbox, or use the parametric plot tool in ADE. In the *Choosing Analyses* form, you can specify up to three variables. In the parametric plot tool, the number of variables is unlimited.
11. Set the variable name to be swept.
12. Set the sweep range and step information.
13. If you want to run specific points, check the appropriate check box at the left side of the *Add Specific Points* field.

The form below shows the hb *Choosing Analyses* form with *Names* selected. Only the differences are shown from the previous section.



14. When *Names* is selected, ADE reads the schematic and lists all the sources in the circuit.



The *Expr* field lists the entry that sets the frequency, and the values field evaluates the variable expression and lists the value.

Mxharm lists the number of harmonics for that tone. *auto* is allowed if *Decide automatically* or *Yes* are selected from the *Run transient?* drop-down list.

Ovrsap lists the oversample factor for that tone.

Tstab specifies which tone is eligible for transient-aided hb.

Important

Tstab should be enabled for the tone that causes the most distortion in the circuit.

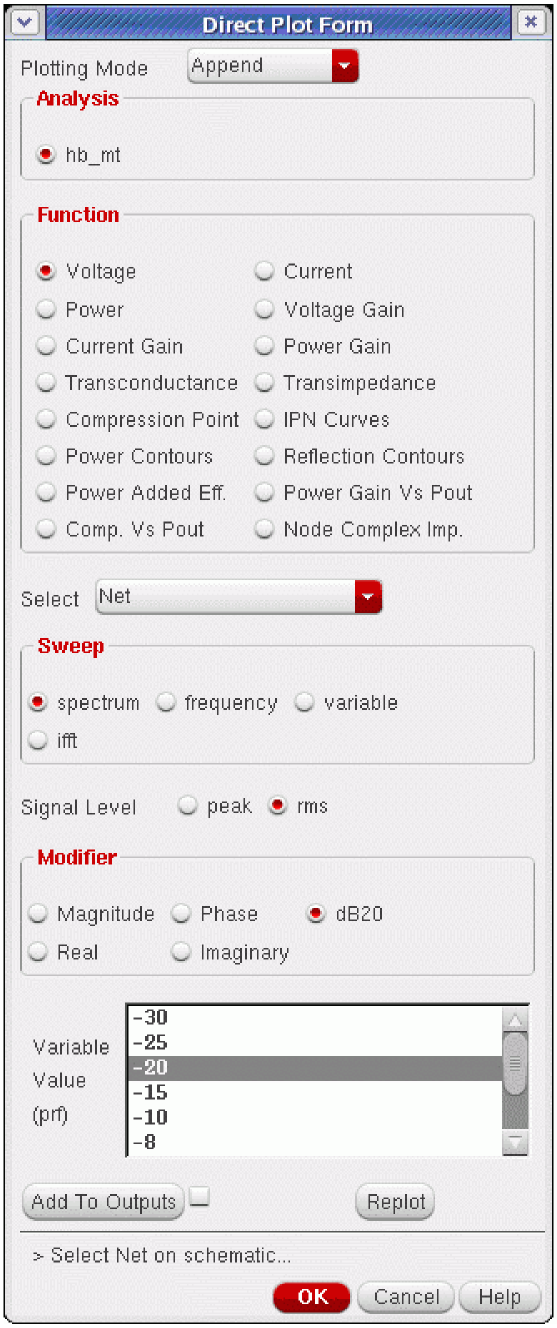
15. To make changes, select the line and then edit the entries in the fields just below.

16. When you are done with your edits, click *Change,* select another field, or press the <*Tab*> key.

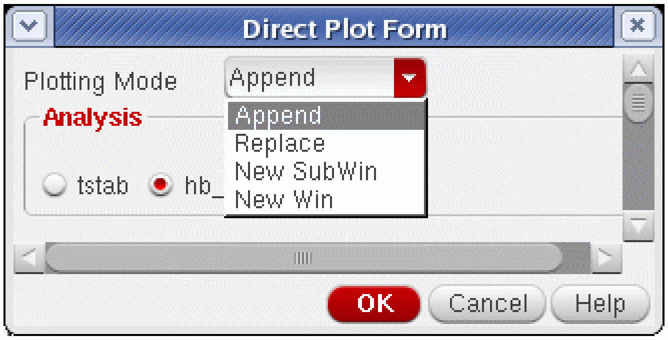
17. When you are done with the setup, run the simulation.

18. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE environment.

The form looks as below:



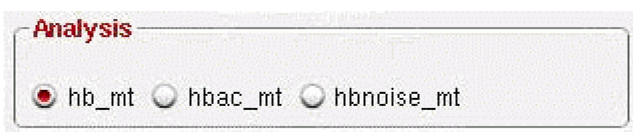
The *Plotting Mode* drop-down list box has the following options:



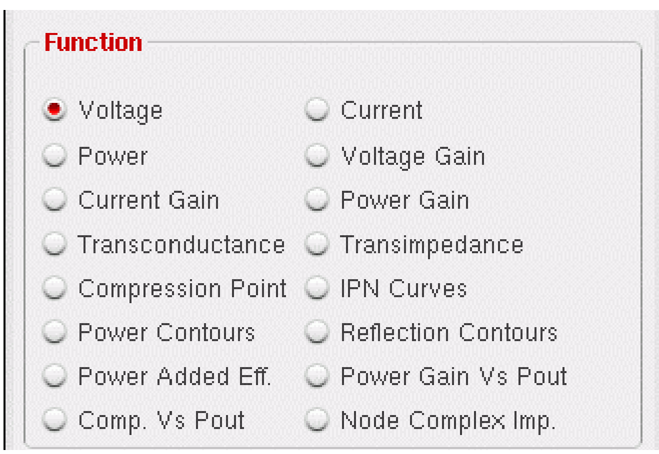
- *Append* - Add the new plot to an existing plot in the selected subwindow.
- *Replace* - Replace the data in the selected subwindow.
- *New SubWin* - Open a new subwindow and plot the data.
- *New Win* - Open a new window and plot the data.

19. Select the analysis type you want to plot from the list.

The figure below shows three different analyses. _*mt* means multiple tone.

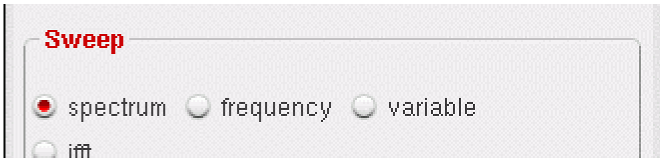


20. Select the specific item you want to plot from the list in the *Function* section. This example shows plotting the voltage on a node.



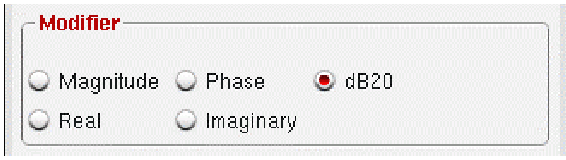
21. Different items can be plotted by using the *Select* drop-down list. You can choose from a number of different things by clicking the *Select* drop-down list. For example, *Differential nets*.

22. If you want to plot the spectrum at a specific power level, select *spectrum* or *frequency*. If you want to see a swept measurement of a specific harmonic, select *variable*. If you want to plot the time-domain waveform at a specific power level, select *ifft*.

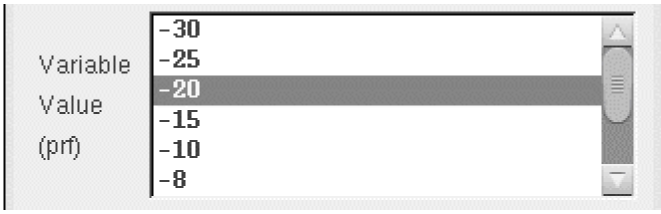


23. For a harmonic plot, select *peak* or *rms*.

24. Select the data type you want to plot. If you want *dBm*, select *Power* in the *Function* section.



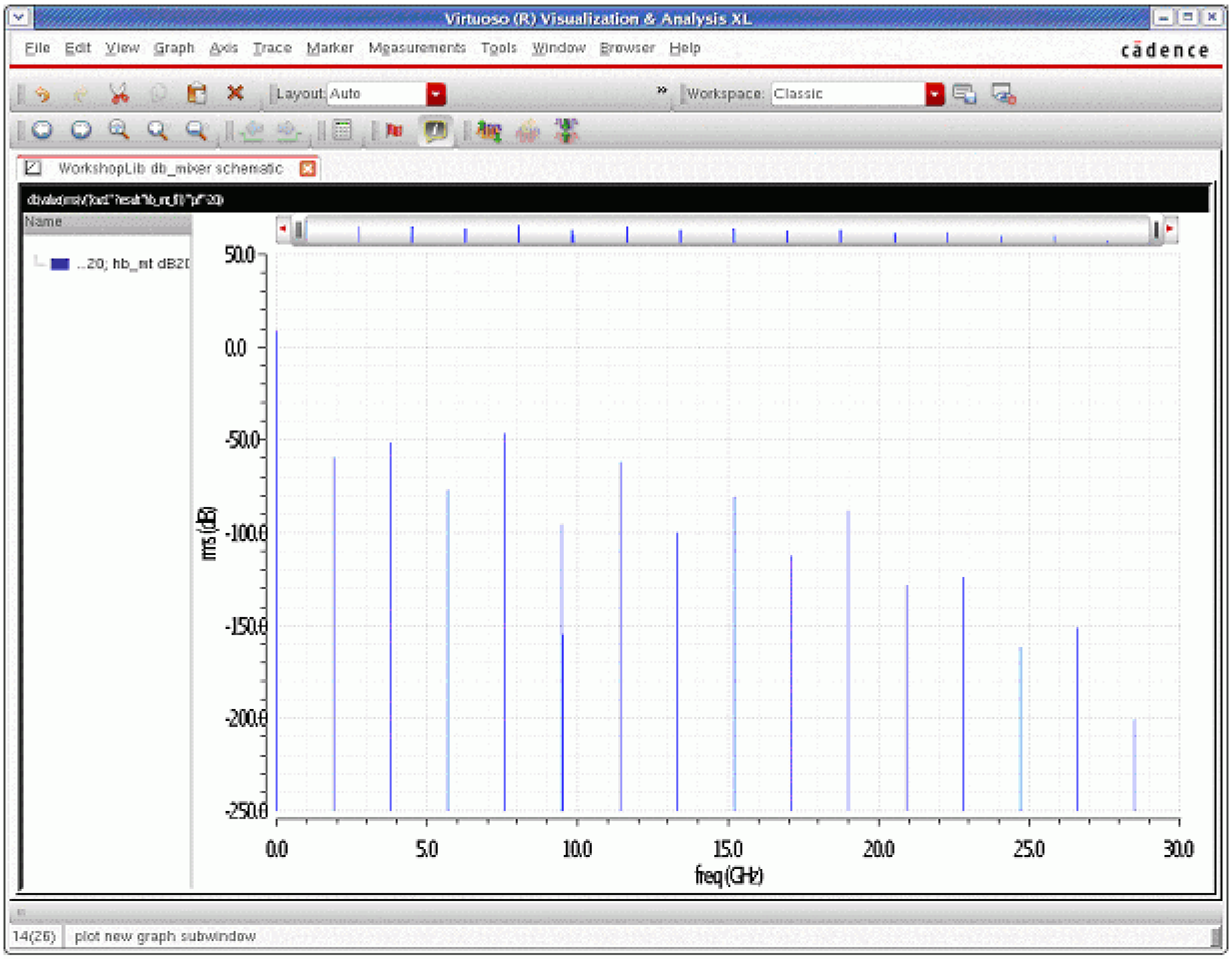
25. Select the desired harmonic or power level (as appropriate for your selection in the sweep section)



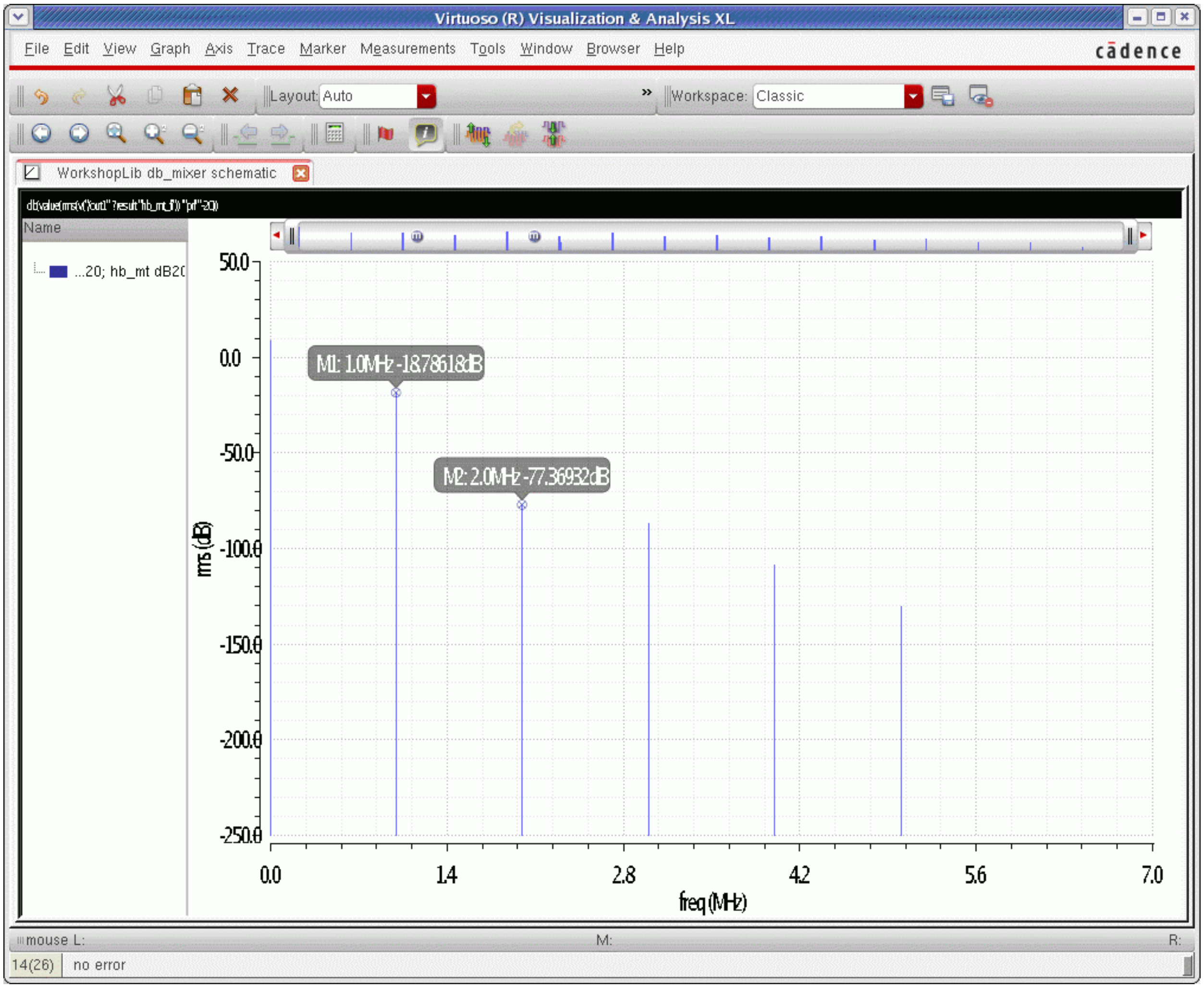
26. If you want to add an expression to the ADE outputs section so that it plots automatically after a simulation, select *Add To Outputs*.

At any time when filling the *Direct Plot Form*, appropriate directions on what to do are given at the bottom of the form. In this example, a variable value needs to be selected in the variables table.

This is the wide range display of all the harmonics. *For more information on the waveform tool, see the documentation for Wavescan (IC5.1) or ViVA (IC6.1).*



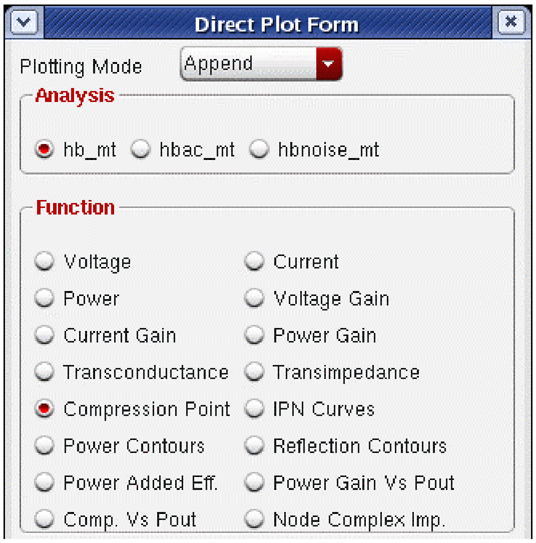
This is the narrow range harmonic display. Click the right mouse button to zoom.



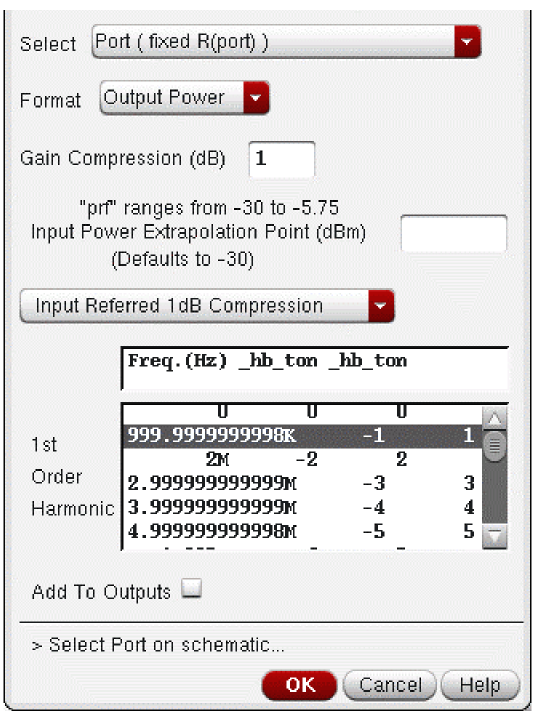
Compression Point

Follow the steps below to get the compression point curve on the *Direct Plot Form*:

1. Select *Compression Point*.



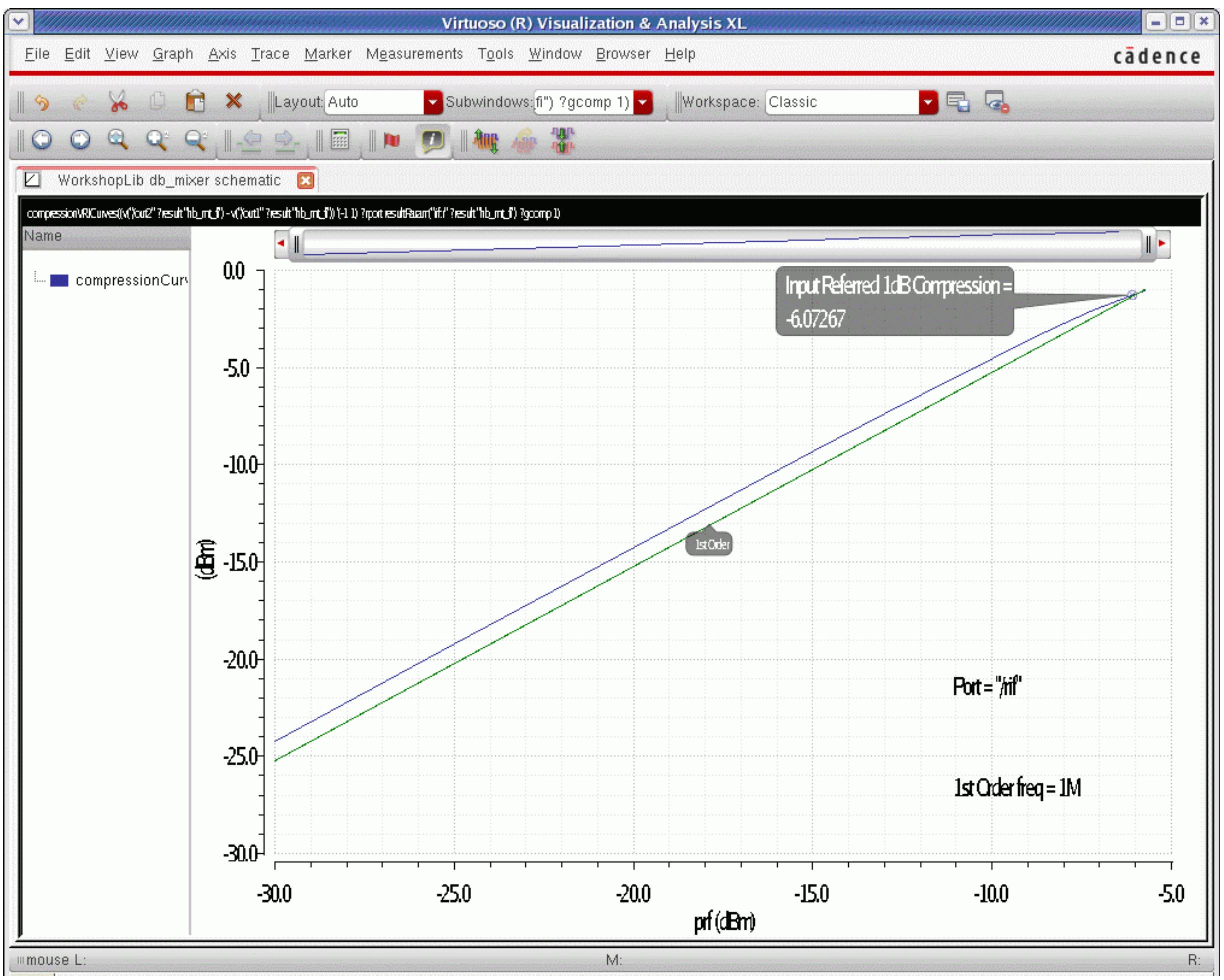
2. Select the desired value (port, net, or differential net) from the *Select* drop-down list.



3. Select whether you want the output power or the gain plotted on the Y axis as a function of the input power from the *Format* drop-down list.

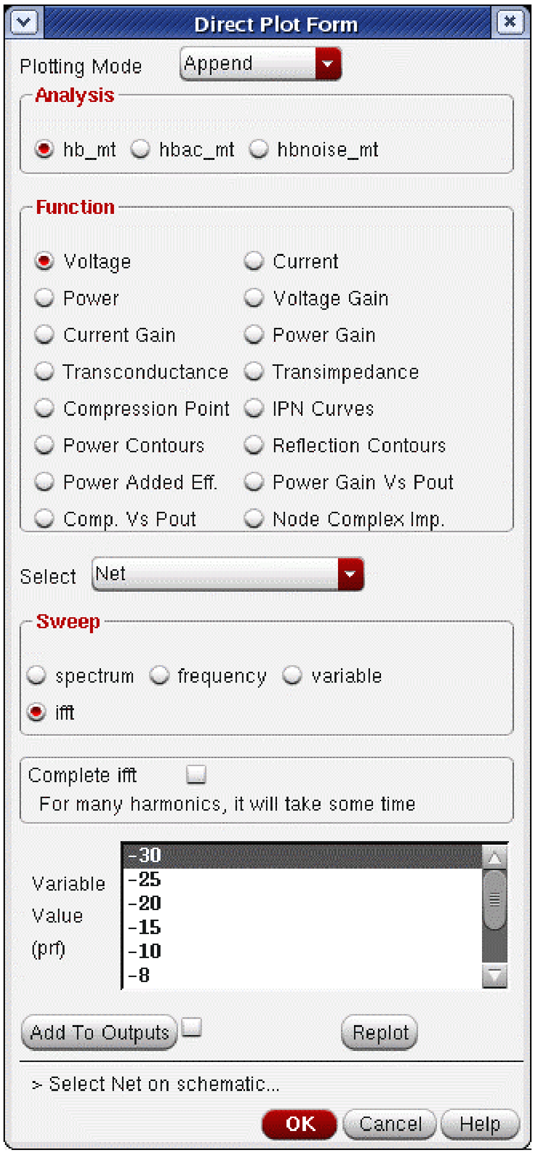
4. Select the desired amount of compression in the *Gain Compression (dB)* field. For small amounts of compression you might need to use conservative in the hb *Choosing Analyses* form.
5. Specify the desired level of input power to begin drawing the 1dB down ideal 1dB per dB curve. The default is to drop down the amount of compression specified at the lowest power point of the sweep.
6. Select the desired output-referred or input-referred compression point.
7. Select the desired frequency of the output harmonic.

This is the compression plot. The ideal 1dB per dB curve is drawn and the compression point is shown at the upper right of the plot.



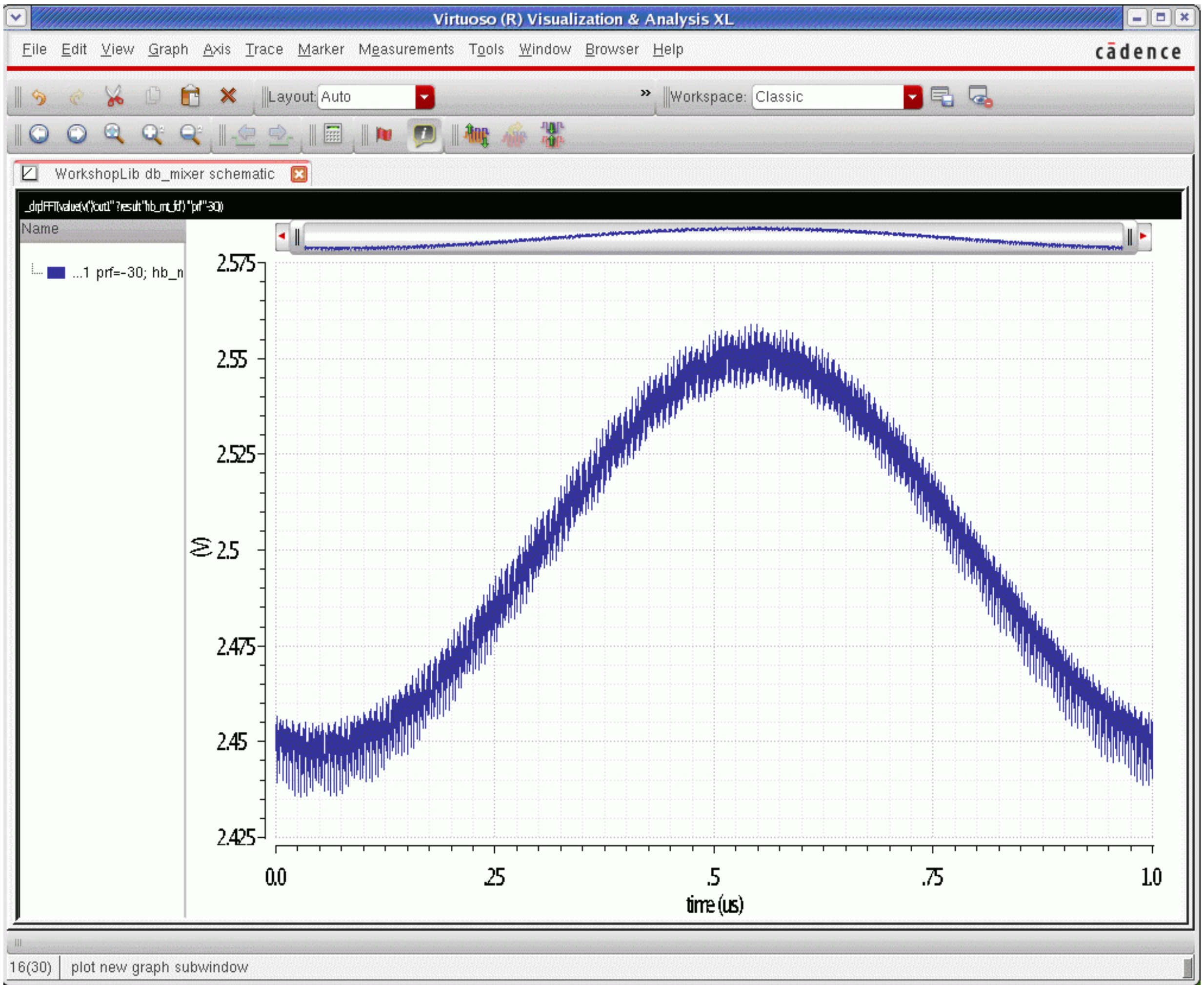
IFFT

To get the time-domain waveform, select the following values on the *Direct Plot Form*:

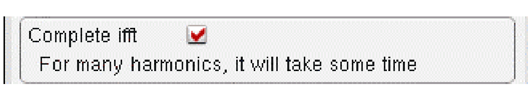


1. Select *ifft*.
2. For multi tone simulations, the default is to filter out the higher harmonics of the hb analysis in order to improve the plotting speed.
3. Select the desired input power for the ifft plot.

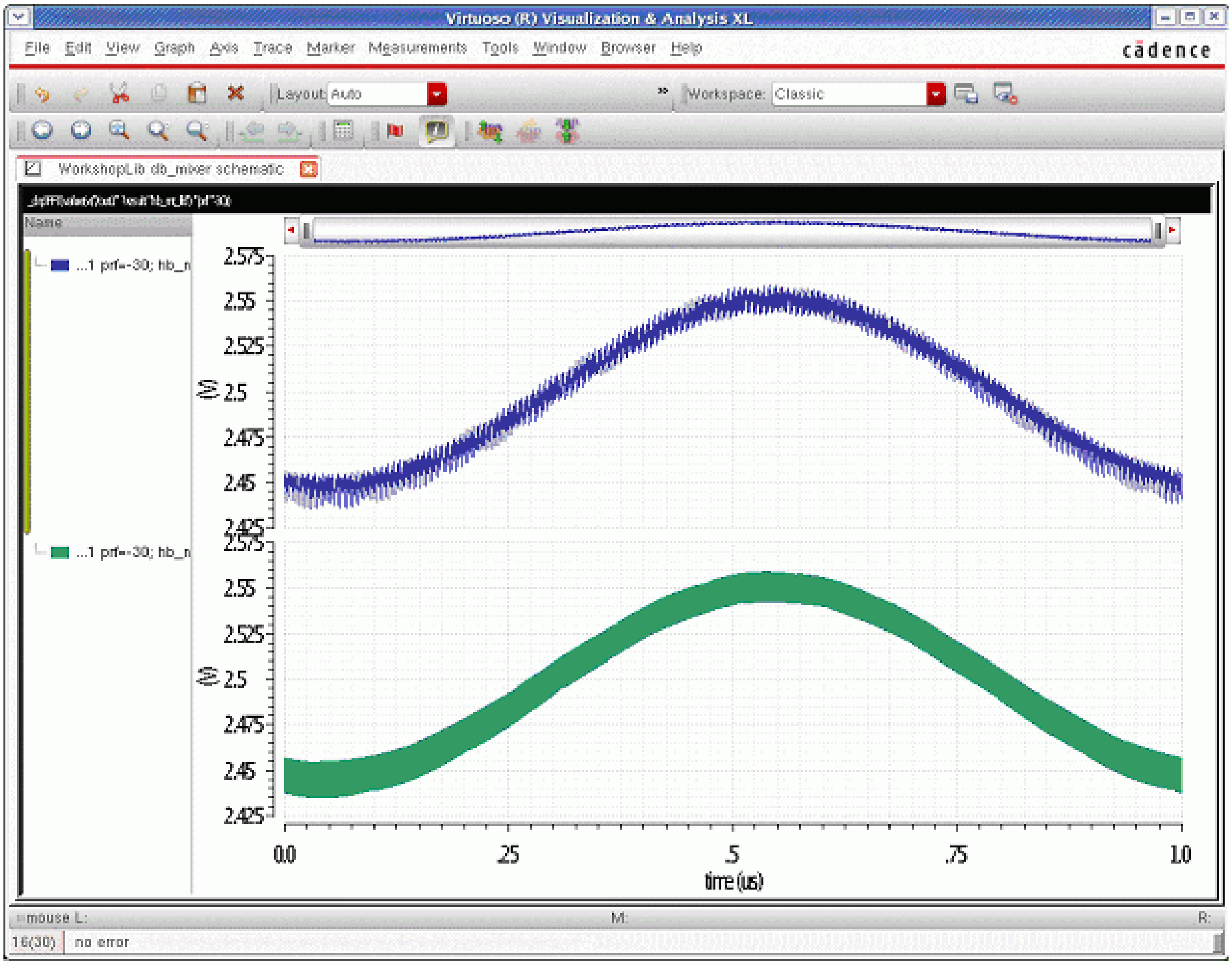
This results in a good waveform that plots quickly. Only the lower frequency harmonics are present in this plot.

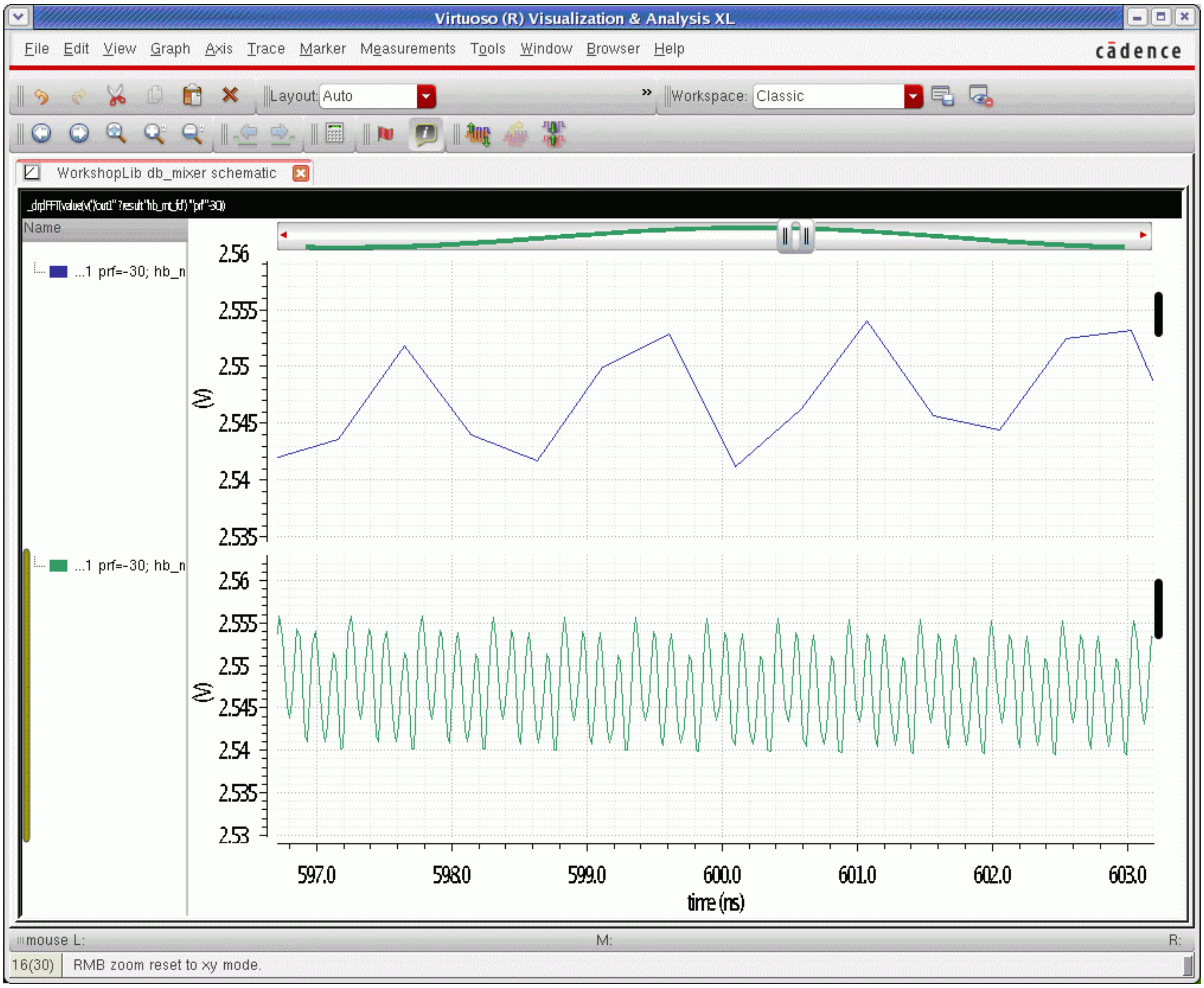


To get the best waveform, select the *Complete ifft* checkbox. The plotting time can be considerable for 3 and 4 tone simulations.



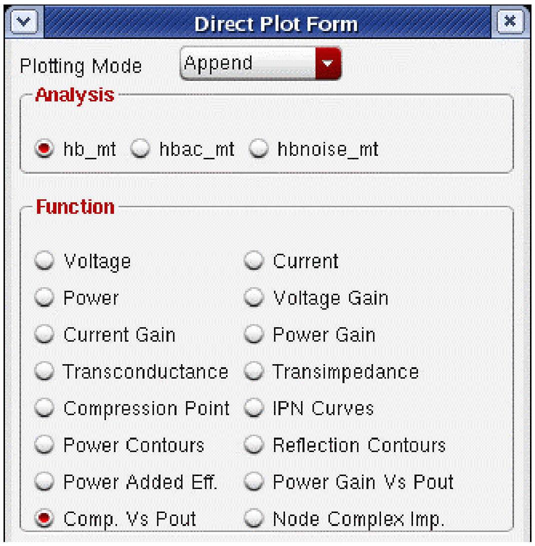
Note that not all the harmonics are present in the upper (default) waveform. You choose if you want a quick idea or the plot with all the harmonics. If you want the higher order harmonics in the waveform, select the *Complete ifft* checkbox in the *Direct Plot Form*. The full waveform and the zoomed-in views are shown below.




Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Layout: Auto
Workspace: Classic
WorkshopLib db_mixer schematic
Name
...1 prf=-30; hb_n
...1 prf=-30; hb_n
2.56
2.555
2.55
2.545
2.54
2.535
2.53
(V)
597.0
598.0
599.0
600.0
601.0
602.0
603.0
time (ns)
mouse L:
M:
R:
16(30)
RMB zoom reset to xy mode.

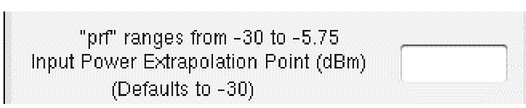
Compression Vs Pout

Another thing you can plot from a sweep is the compression versus the output power. To see this, select *Comp Vs Pout*.

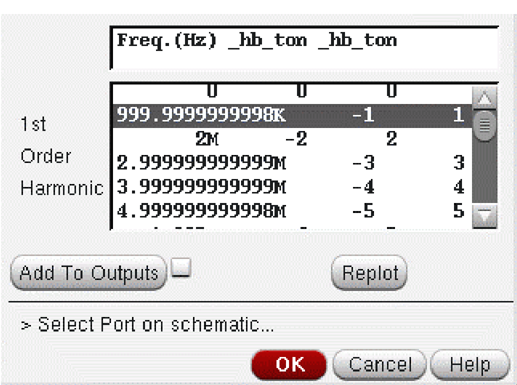


Several different types of data can be selected. For example, a net, or differential nets.

The zero compression input power is set in the field as shown below. The default is the lowest power point of the sweep.

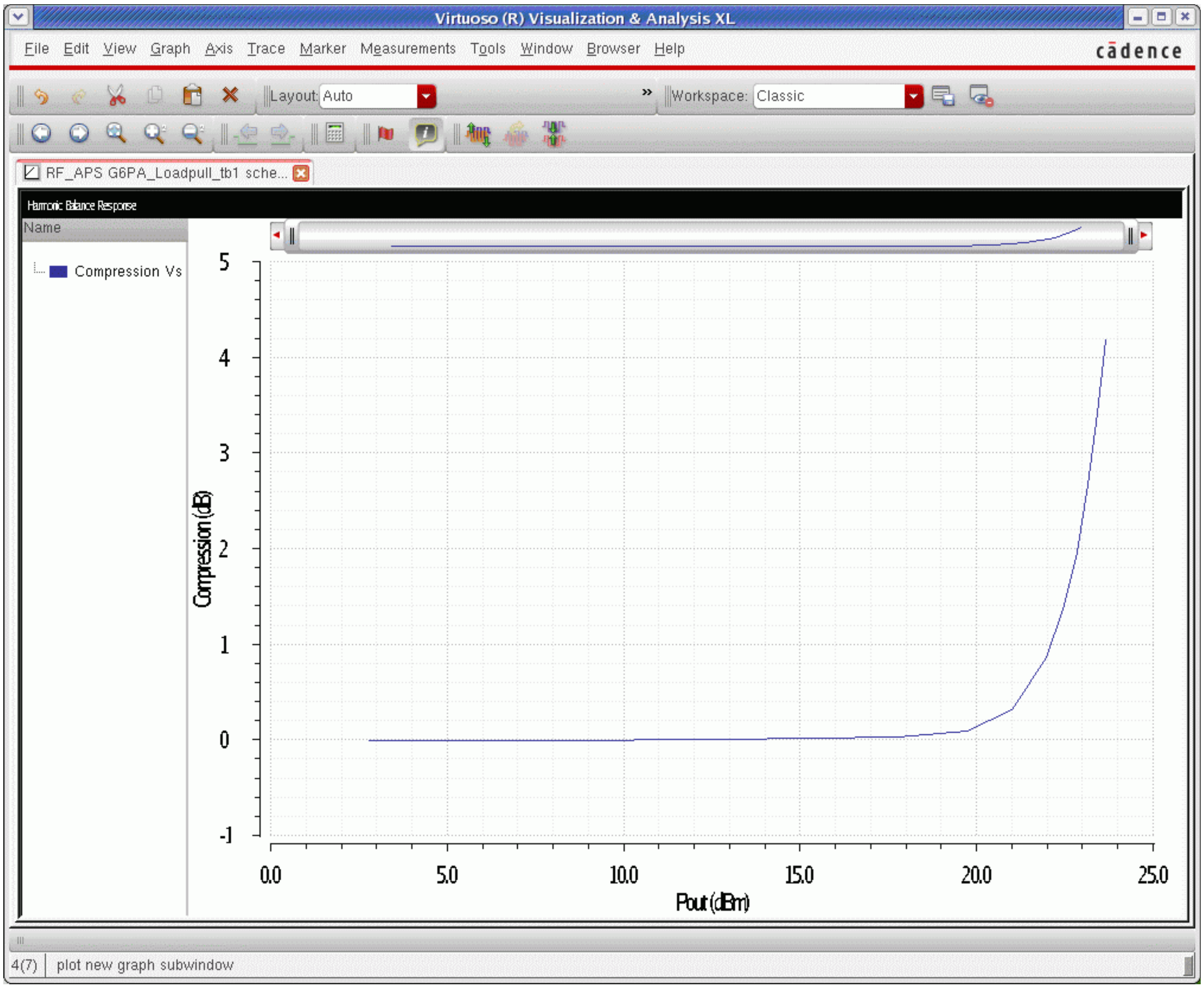


The frequency of the harmonic to be plotted is selected here.



Select the port in the circuit.

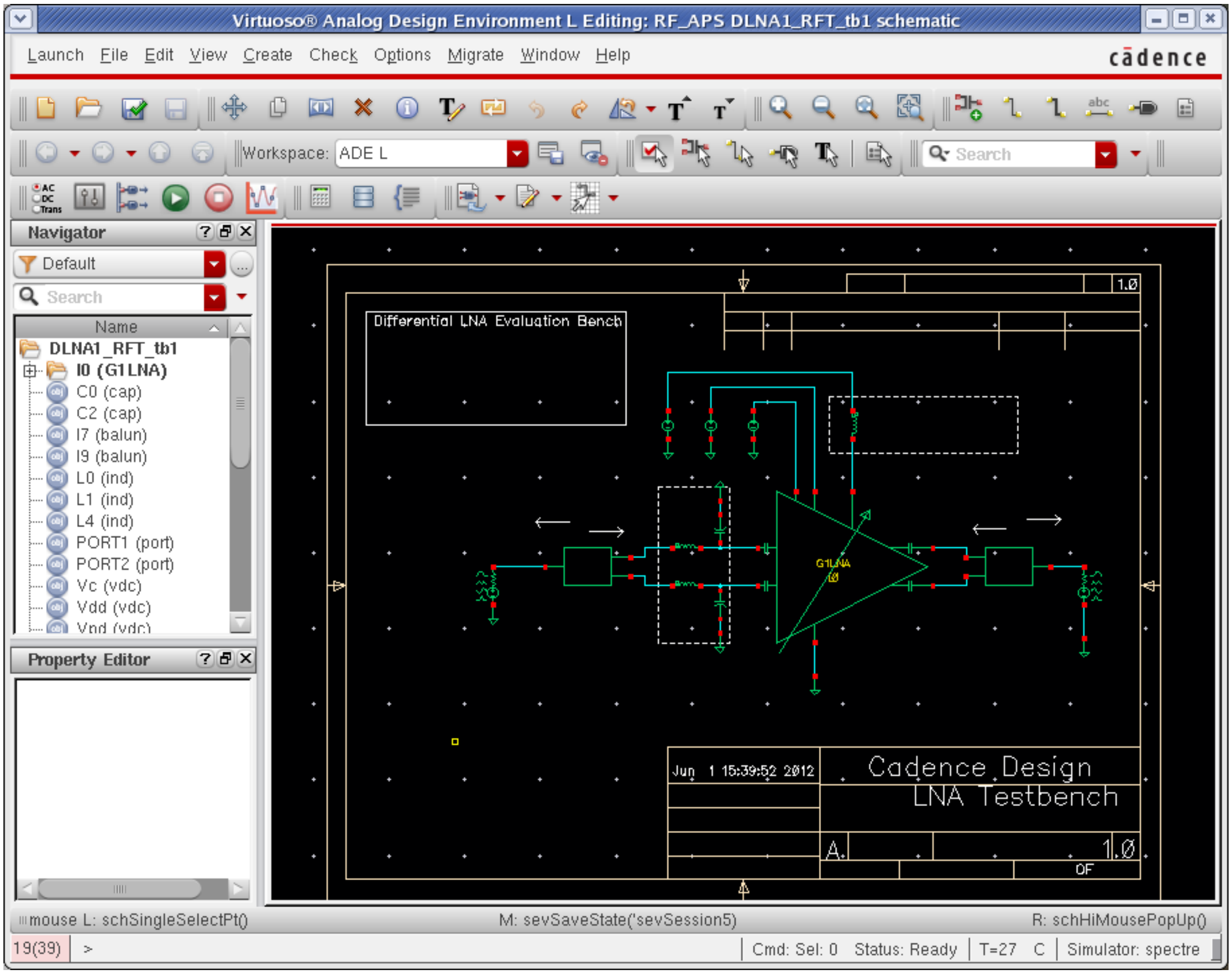
The output is displayed as below.



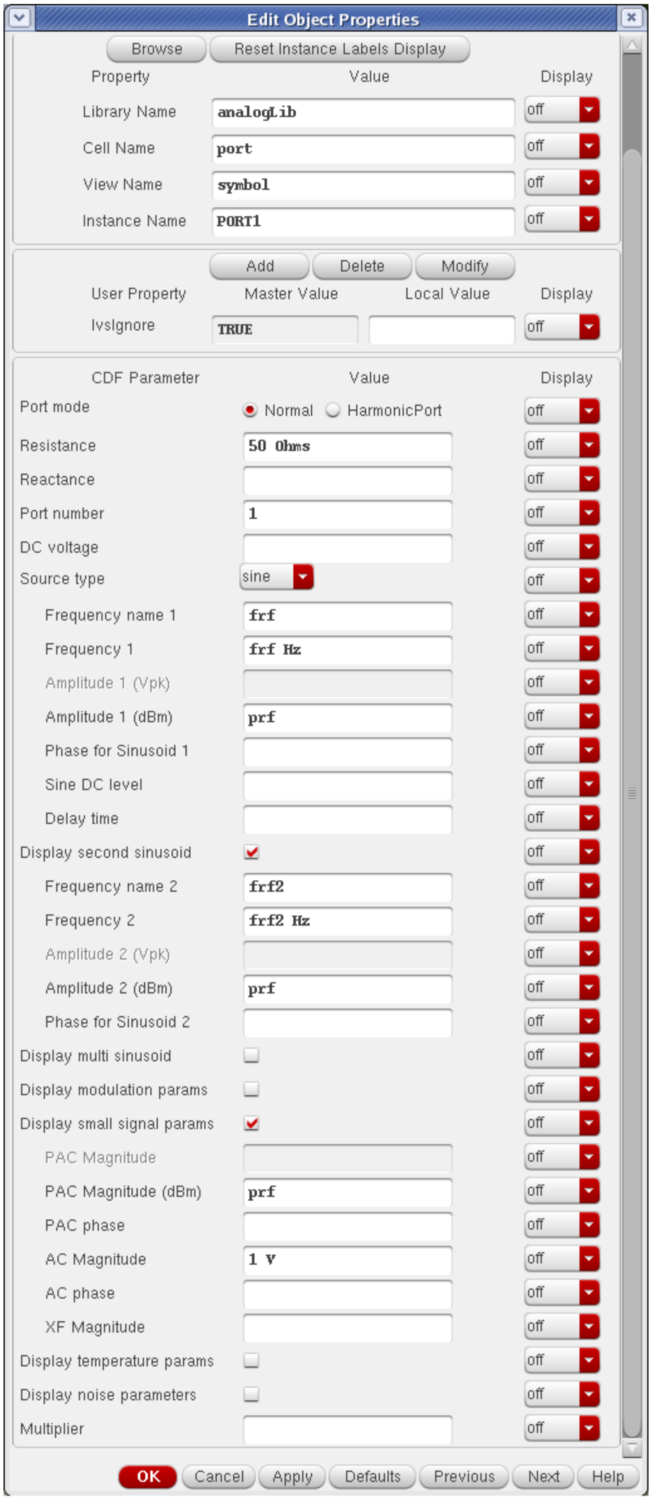
Example Using the Harmonic Port with Harmonic Balance

The port component has a feature called the harmonic port that allows the specification of different impedances at different harmonic numbers. This is available for harmonic balance large-signal analyses only, and does not work with any other analysis.

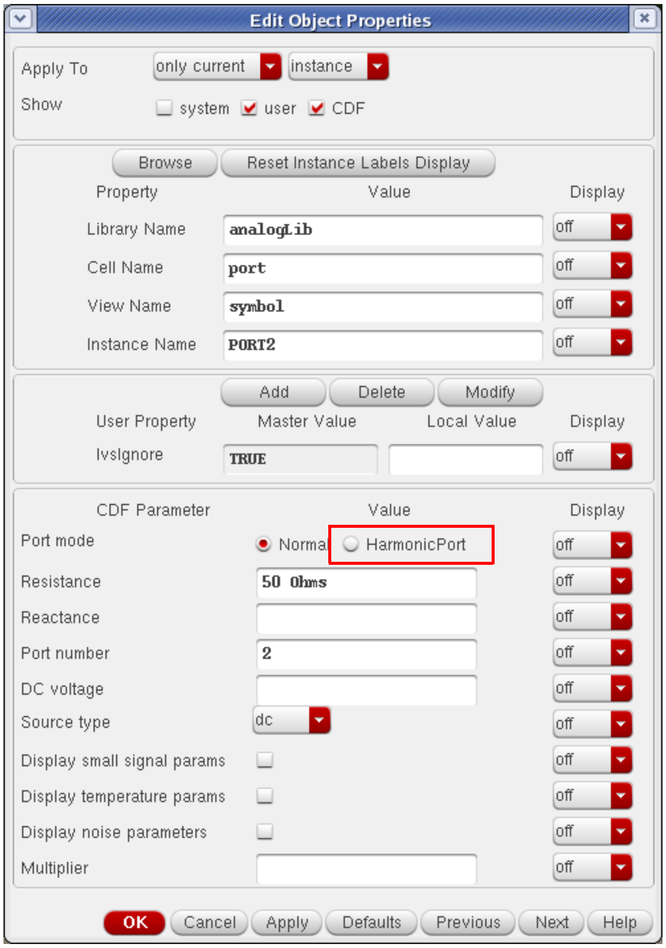
For this example, a 2.4GHz LNA is the example circuit.



The input and output ports are set up normally, as shown in the properties forms below.

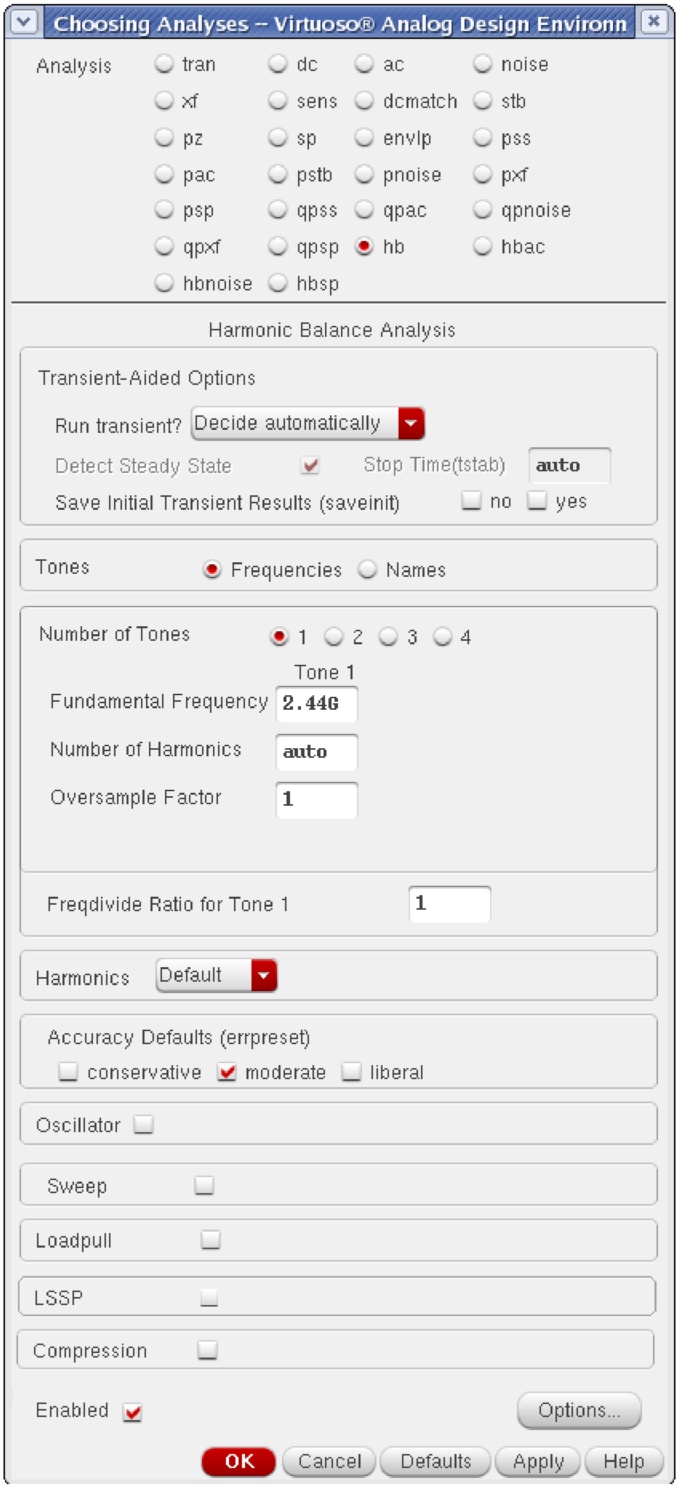


PORT1 is the input and PORT2 is the output.

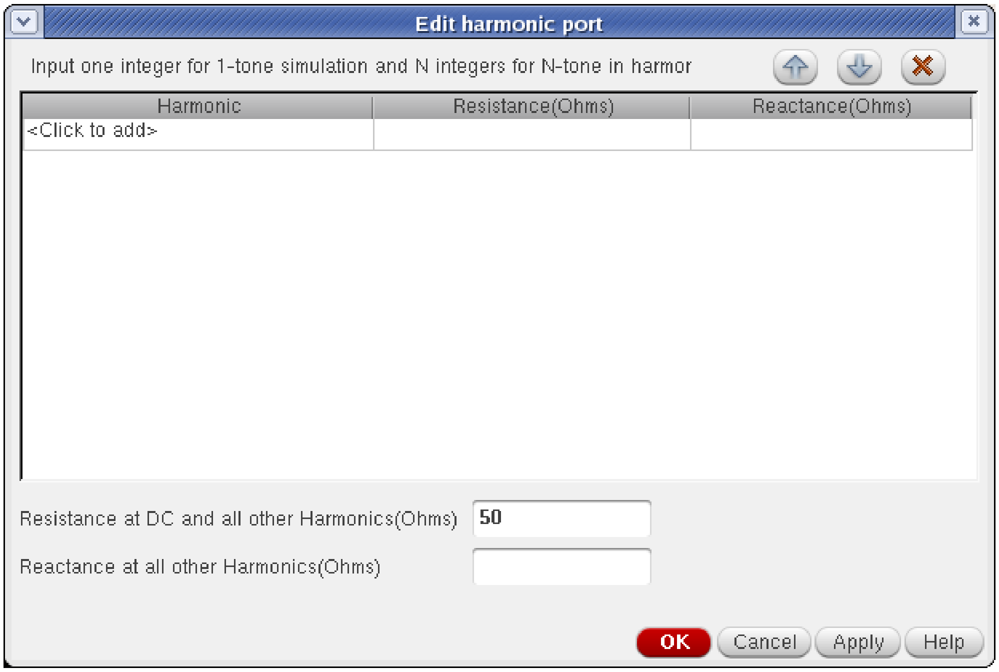


One Input Frequency

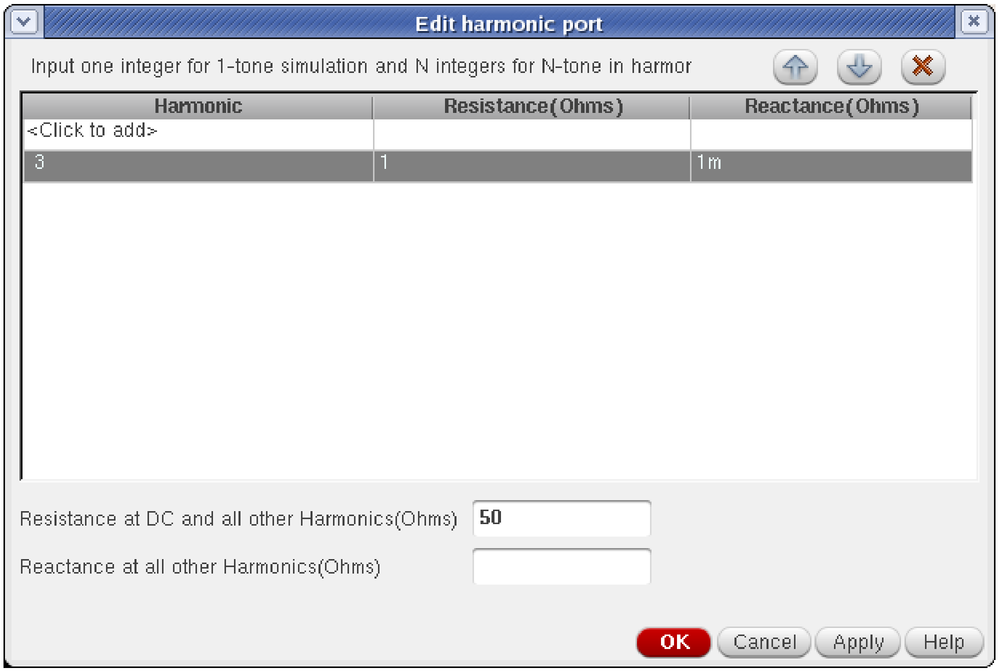
The harmonic balance *Choosing Analyses* form has a single input applied.



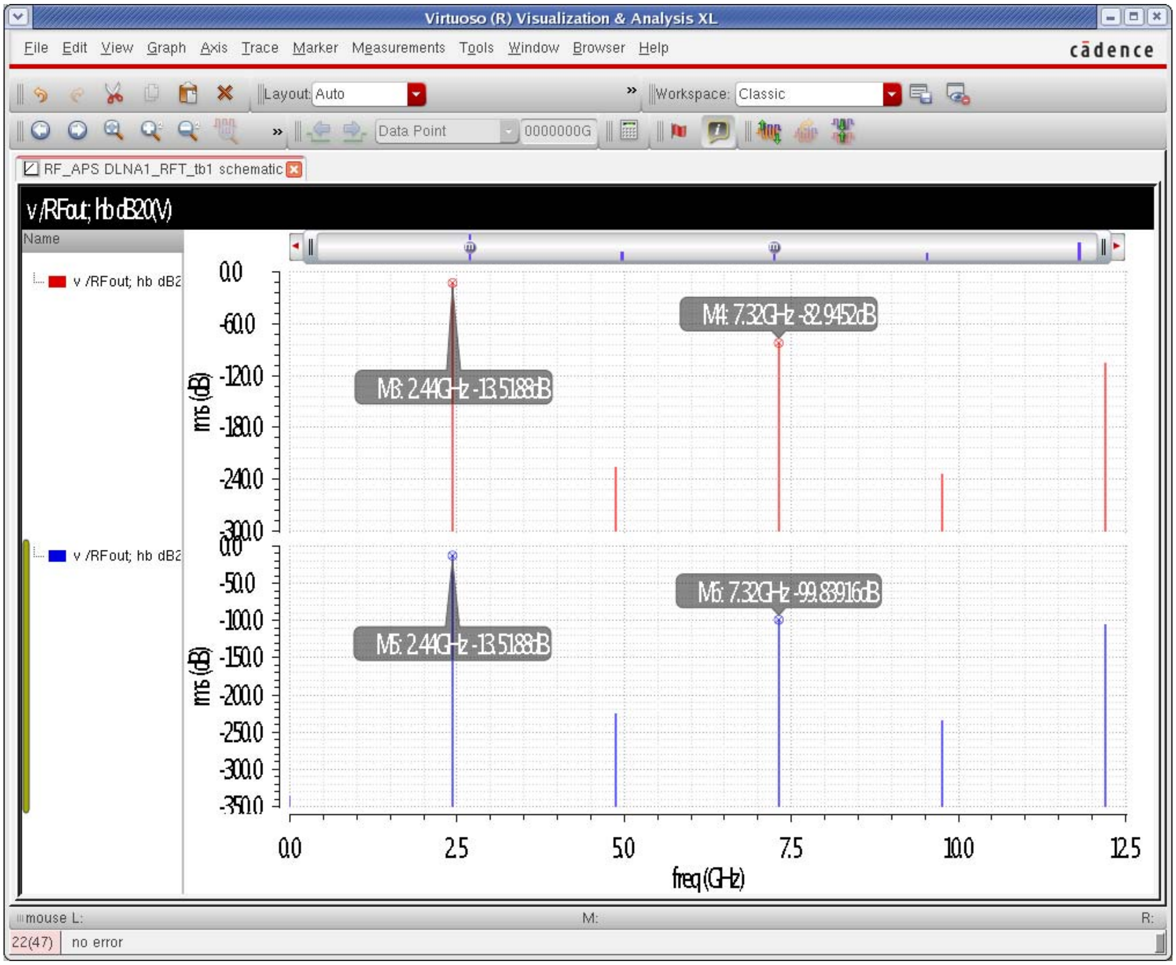
To use the harmonic port, select *Harmonic Port* on the *Edit Object Properties* form. In this case, the port being used as a load in the circuit is selected. The *Edit harmonic port* window is displayed.



For this setup, a single input frequency is applied to the circuit. In this case, harmonic numbers 1, 2, 3, 4, and 5 exist in the simulation result. In the *Harmonic* column, you enter the harmonic number. In the *Resistance* column, you enter the resistance. In the *Reactance* column, you enter the series reactance in ohms. Positive reactance adds inductance, and negative reactance adds capacitance. Multiple entries are allowed, therefore, different impedances can be set for each harmonic. The resistance or reactance can be left blank, which is interpreted as zero resistance or reactance.



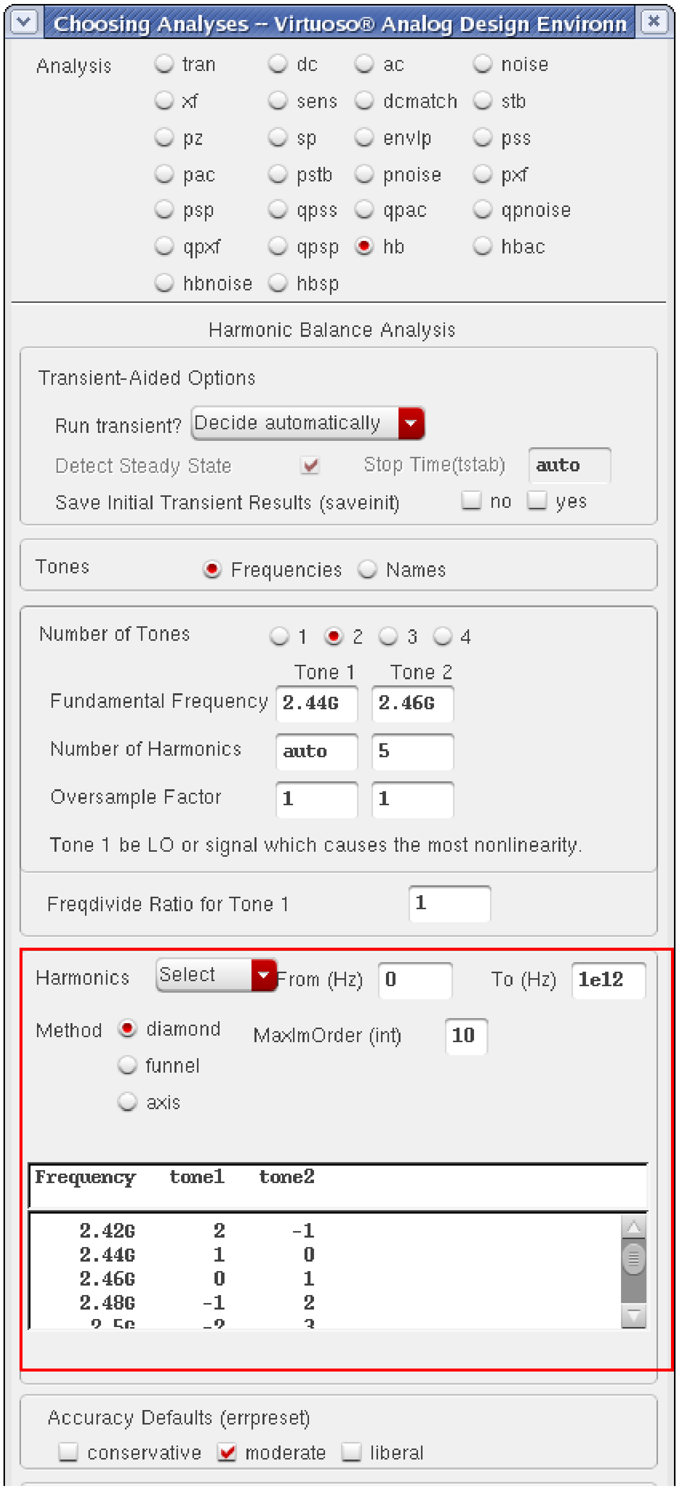
Here is an example where the third harmonic is set to 1 ohm in series with +j1mohm. This should lower the amplitude of the third harmonic.



The top trace is the normal port, and the bottom is the trace with the third harmonic set to 1ohm in series with +j1milliohm. Note that the third harmonic is smaller.

Two Input Frequencies

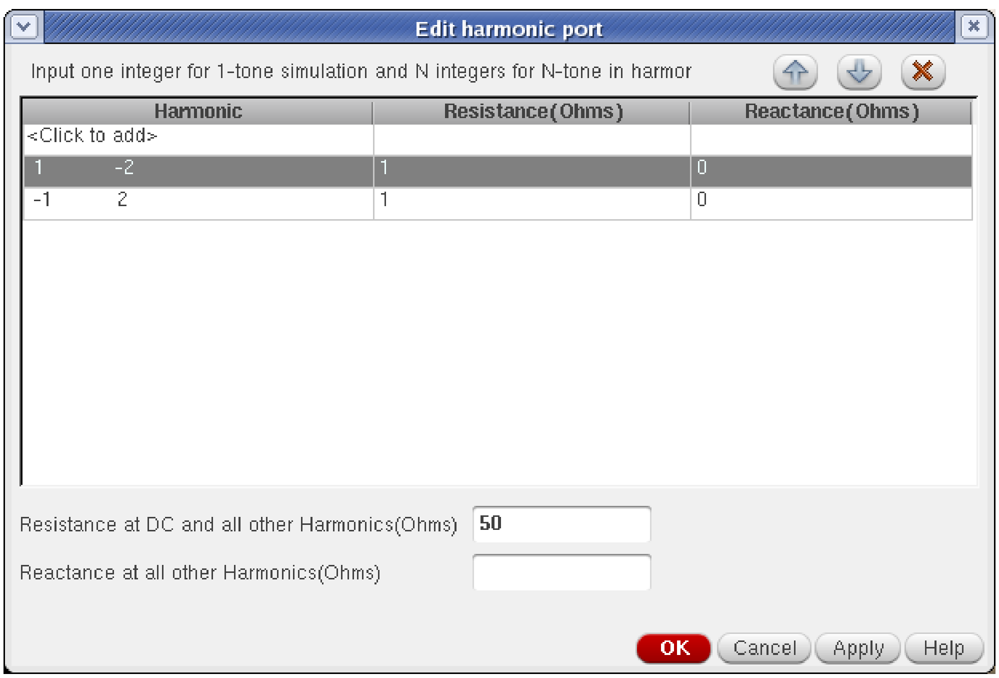
Now the harmonic balance *Choosing Analyses* form has two frequencies.



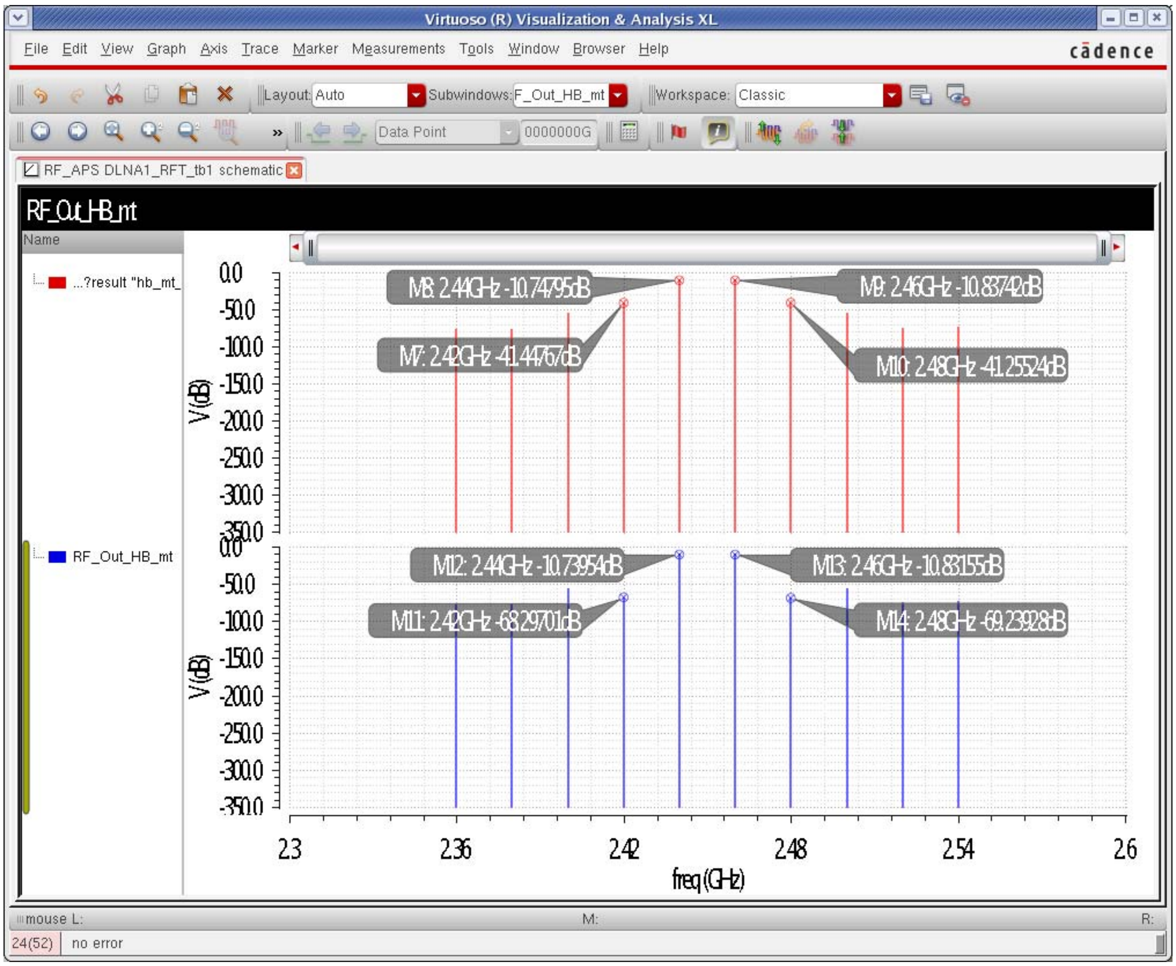
Also, the diamond cut is set with a very large *MaximOrder*. The reason for doing this is to see the harmonic numbers that correspond to the frequencies produced by the circuit. Each frequency has 2 harmonic numbers that indicate which harmonics are mixing with which to produce a specific output frequency. For example, take the 2.44GHz term. The harmonic number is 1 0. The frequency that is produced is 1*2.44G + 0*2.46G. Now look at the intermod at 2.42GHz. This index is 2 -1. The frequency is 2*2.44G - 1*2.46G. These are the harmonic numbers that can be set for the harmonic port. Wild cards (*) are not allowed in the harmonic number specification.

Note that if there were three input frequencies, there would be three numbers that specify every output frequency.

In this case, if the harmonic was specified as 3, there would be an error produced by spectre because there is a single index in the harmonic, but there are 2 indices for all the harmonics of the existing simulation. The number of numbers in the harmonic field must match the number of input freqeuncies in the circuit.



The above example sets the impedance of the third order intermods to be much lower than 50 ohms. If the source is in series, setting the resistance and/or reactance to a large number can block harmonics you do not want. If the port is in parallel like this one, setting small resistance and reactance values can short out an undesired harmonic. This can be used for debugging the circuit, or for setting values to maximize power-added efficiency.



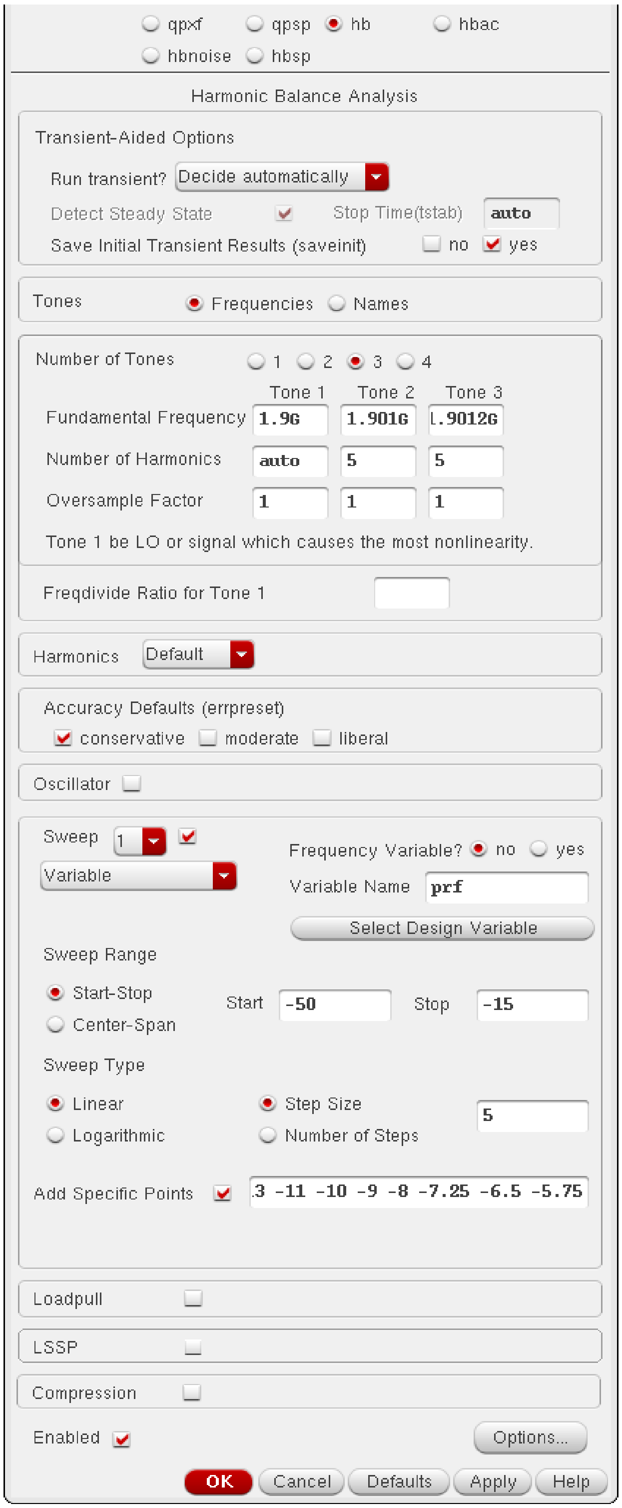
The top trace above is from the normal port, and the bottom trace is from the harmonic port. The third order harmonics are much smaller because they are essentially shorted out.

Three Tone Harmonic Balance

Generally, a 3 tone harmonic balance simulation is used to calculate large-signal IP3 for transmit applications. If you need a small-signal IP3 measurement, see Rapid IP3 in the hbac section of this chapter. For a small-signal IP3 measurement, Rapid IP3 is much faster and is usually more accurate than a 3 tone harmonic balance simulation.

In a 3-tone simulation, first decide if you want to remove the higher order terms from the solution. If you remove the high order harmonics from the simulation, it can run faster because fewer harmonics are calculated.

On the Choosing Analyses form:

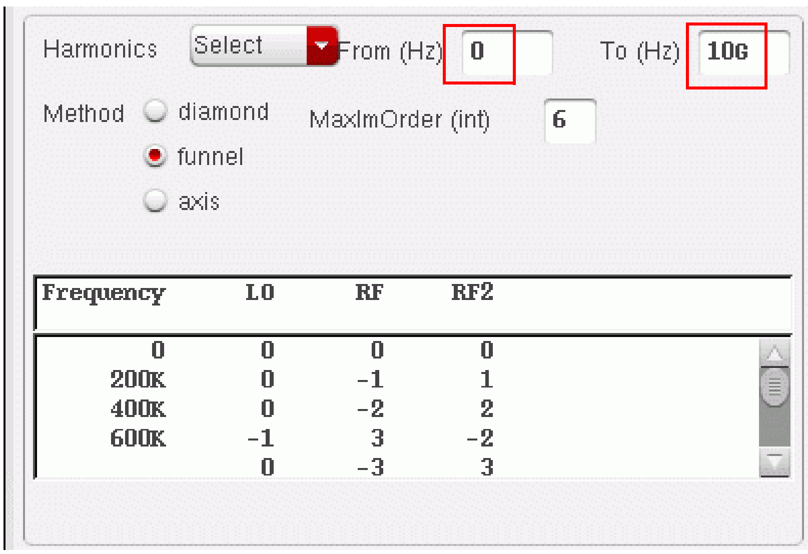


1. To trim the high-order harmonics, choose *Select* from the *Harmonics* drop-down list.

   Funnel is recommended for systems where one or more signals cause a lot of distortion compared to the rest of the signals. Generally, these systems are mixers or sampling circuits that have large amplitude LO or sampling clock signals. *Funnel* calculates the harmonics whose order is less than that specified in *MaximOrder* from the simulation and adds the harmonics of the LO and RF tones but not the higher order mixing products.

   Funnel cuts the harmonics whose order is higher than that specified in *MaximOrder*, but it does not cut the lower-order harmonics of the mixing products. For example, you might set funnel and *MaximOrder* to six. This calculates the sixth order and lower mixing products, but allows the higher order (up to the 10 harmonics specified for Tone1) LO harmonics to be calculated. This is useful for mixer simulations where you know that the LO produces relatively more harmonics than you want with *MaximOrder*. Setting *maximOrder* to six is the recommended starting point for a large-signal IP3 measurement in a down-convert mixer.

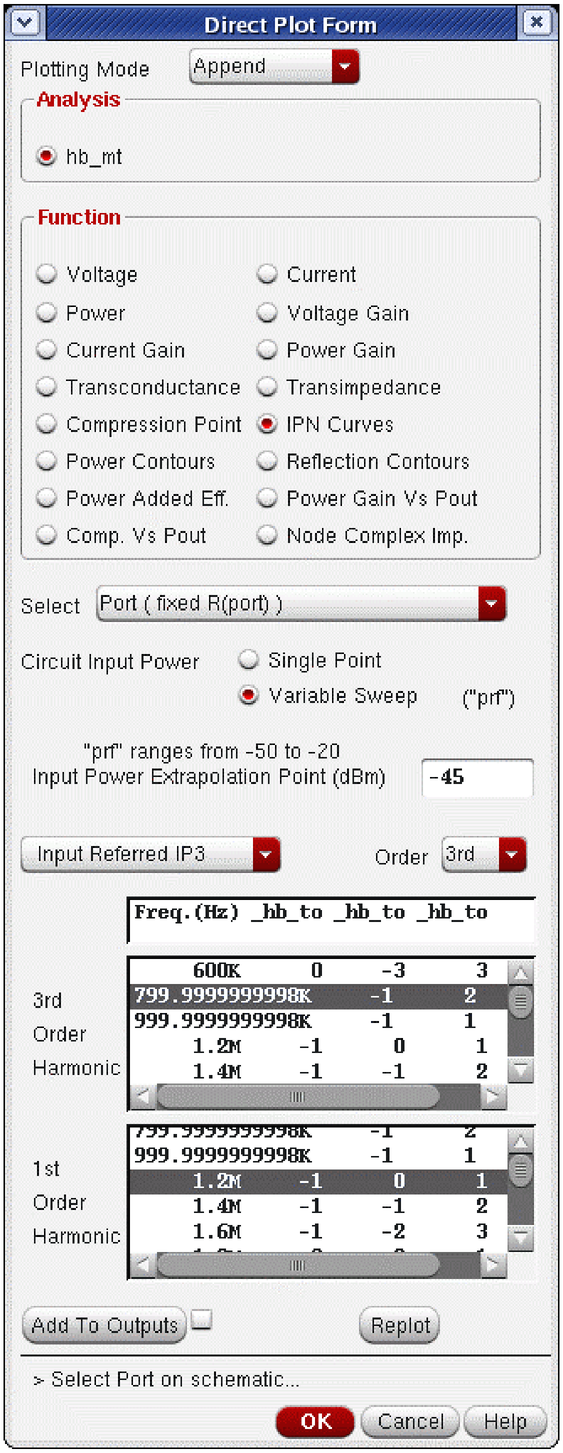
2. To see the actual harmonic frequencies to be run in a list, type a frequency range in the fields next to *Harmonics* drop-down list.



3. Click *OK* and start the simulation.

Large-Signal IP3

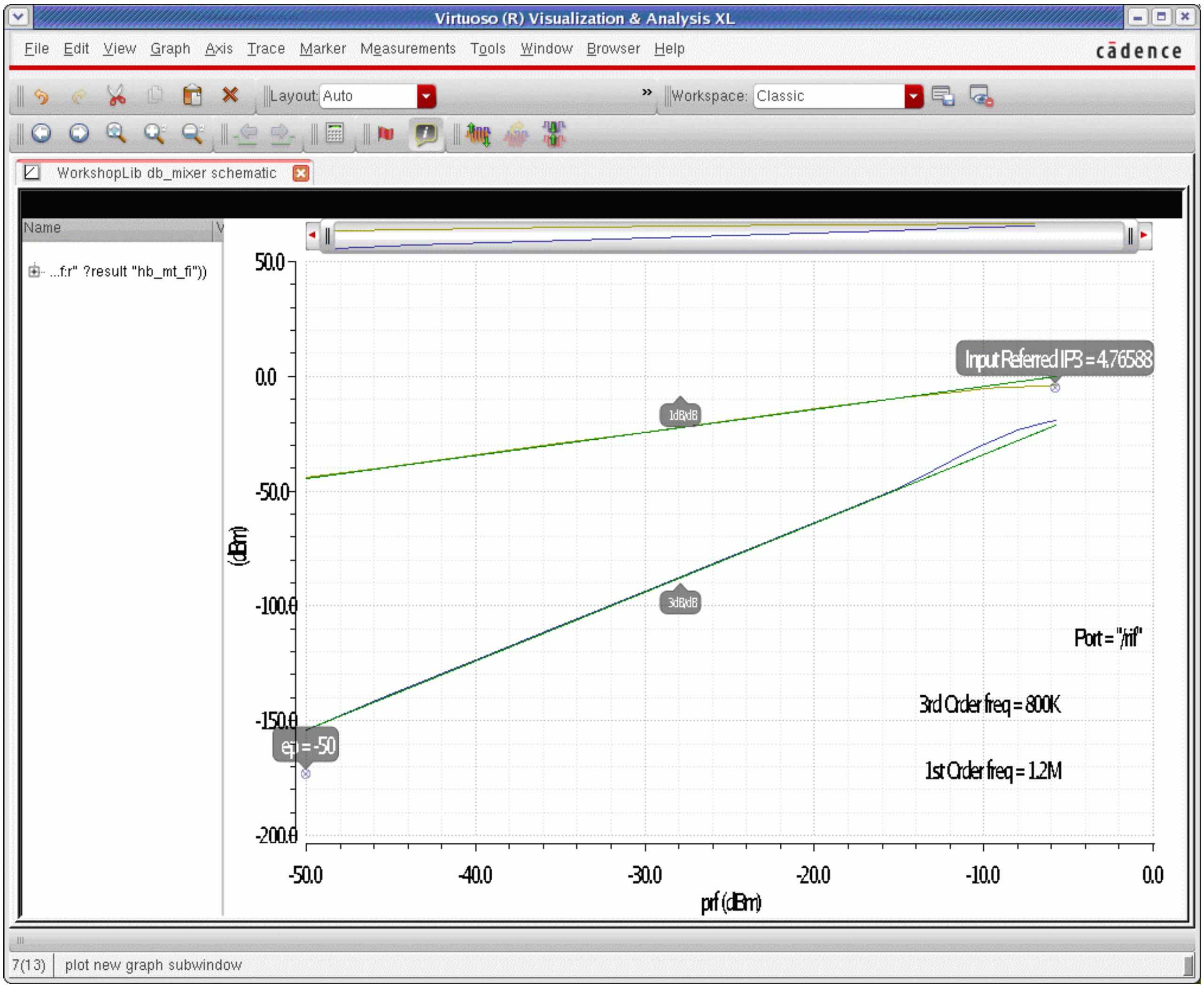
To plot a large-signal IP3, in the *Direct Plot Form*:



1. Select *IPN Curves*.
2. Select the *Variable Sweep* radio button.
3. Select the input power level you want to do the third order projection from. The default is the lowest power of the sweep.
4. Select Input or Output referred from the drop-down list.
5. Select the order from the *Order* drop-down list. *3rd* order is the default.
6. Select the 2nd or 3rd order frequency in the *2nd Order Harmonic or 3rd Order Harmonic* fields.
7. Select the first order frequency in the *1st Order Harmonic field*.

The IP3 calculation (Input Referred IP3 = 4.71 dBm in the plot below) looks like it is tied to the last datapoint of the first order curve. It is tied there so it will be displayed on the screen. It actually calculates the intercept point.

The extrapolation point (ep = -50 in the plot below) is the input power level that the IP3 calculation is done from. This is the input power level where the ideal curves and the measured data overlay.



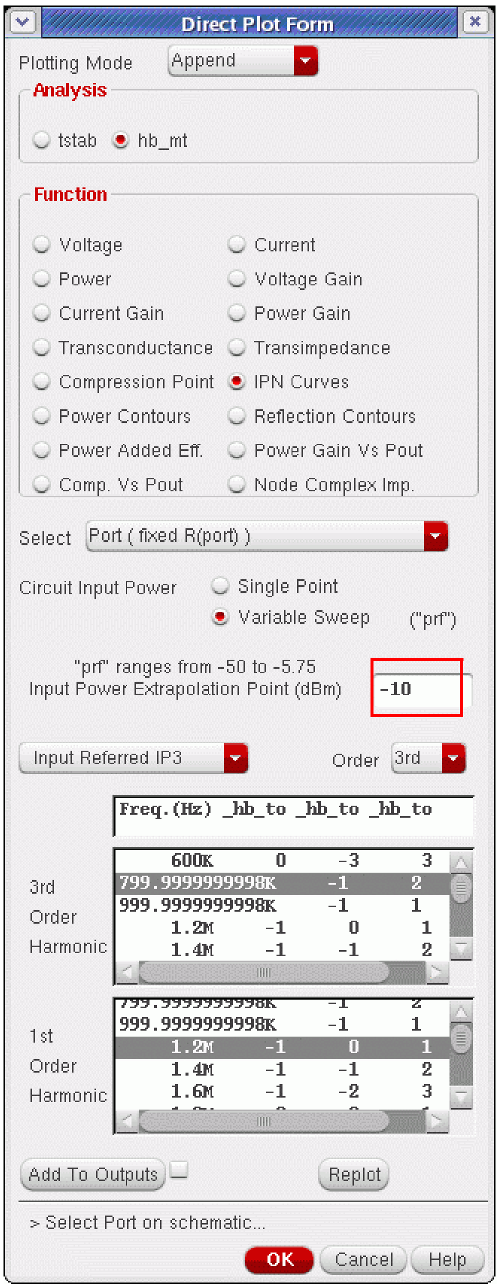
In the IP3 plot for a small-signal IP3 measurement, look for the ideal curves and the data to nearly overlay each other on the third order curve for at least a 10dB range of input power. The plot above shows the data and the ideal curves being essentially the same from -50 to about -15 dBm.

The bottom right corner of the plot shows the following in the order as listed below:

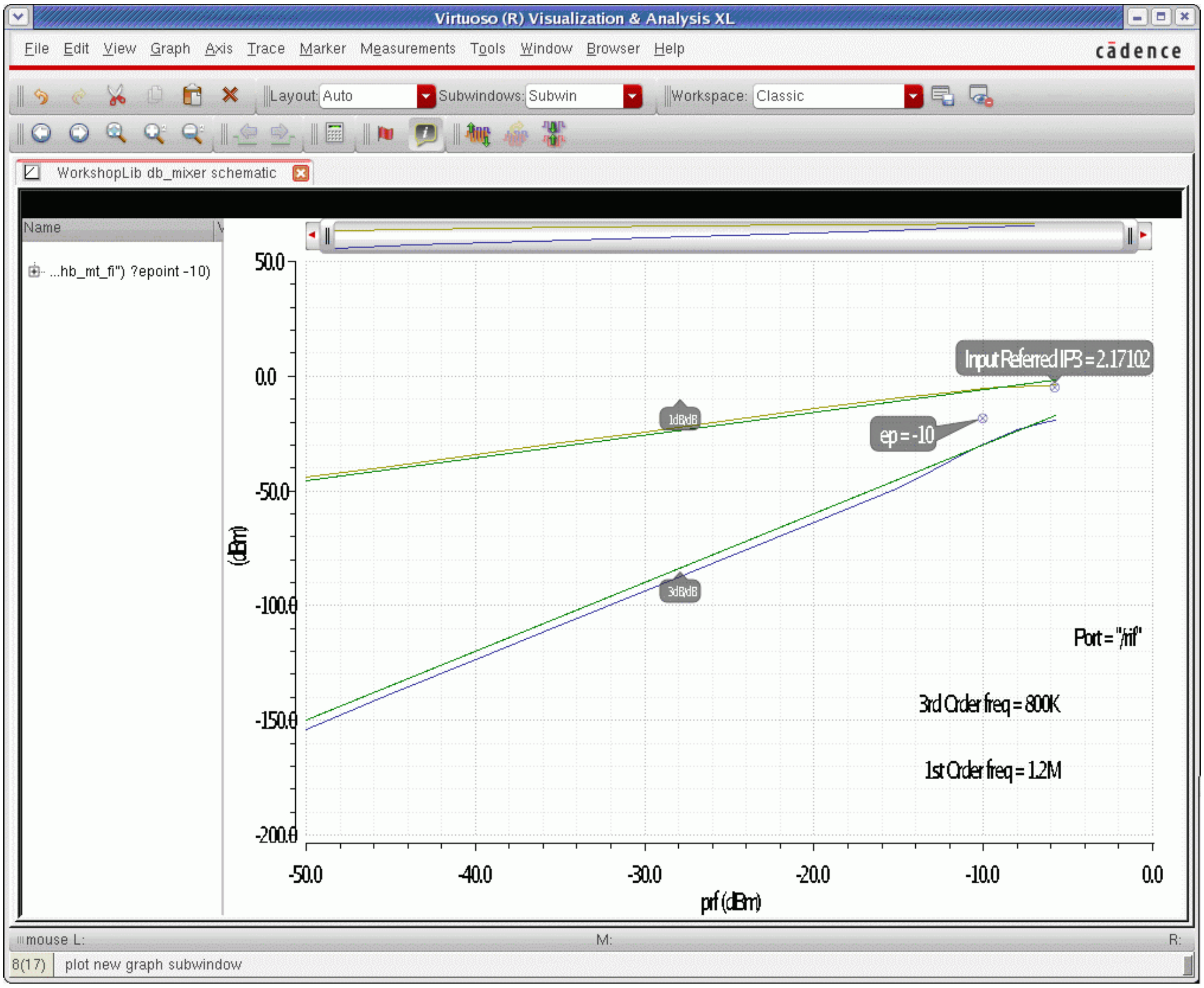
- Output of the circuit.
- Third order frequency.

- First order frequency.

Note that the above IP3 is really a small-signal IP3 measurement. For a much faster IP3 calculation use Rapid IP3 as documented in the hbac and pac sections. For a large-signal IP3 calculation, set the extrapolation point to the desired input power for the large-signal measurement as shown below in the direct plot form.



Now either select replot, or select the output port in the schematic.



Note that the IP3 calculation has changed from 4.76dBm to 2.17dBm.

Multi-Sinusoid hb Analysis

The port component has been enhanced to support the generation of up to nine sinusoids simultaneously. To do this, *Frequency* 2 needs to be set to the spacing frequency. All the sinusoids need to have equal spacing frequencies. *Frequency 1* in the port needs to be set to the center frequency of the multiple output frequencies. The center frequency needs to be one of the frequencies that is generated. For example, imagine that 1GHz, 1.01GHz, and 1.02GHz need to be generated. The center frequency in this case is 1.01GHz and the spacing is 10MHz. Now imagine that 1GHz, 1.01GHz, 1.02GHz, and 1.03GHz need to be generated. Now the center frequency could be either 1.01GHz or 1.02GHz.

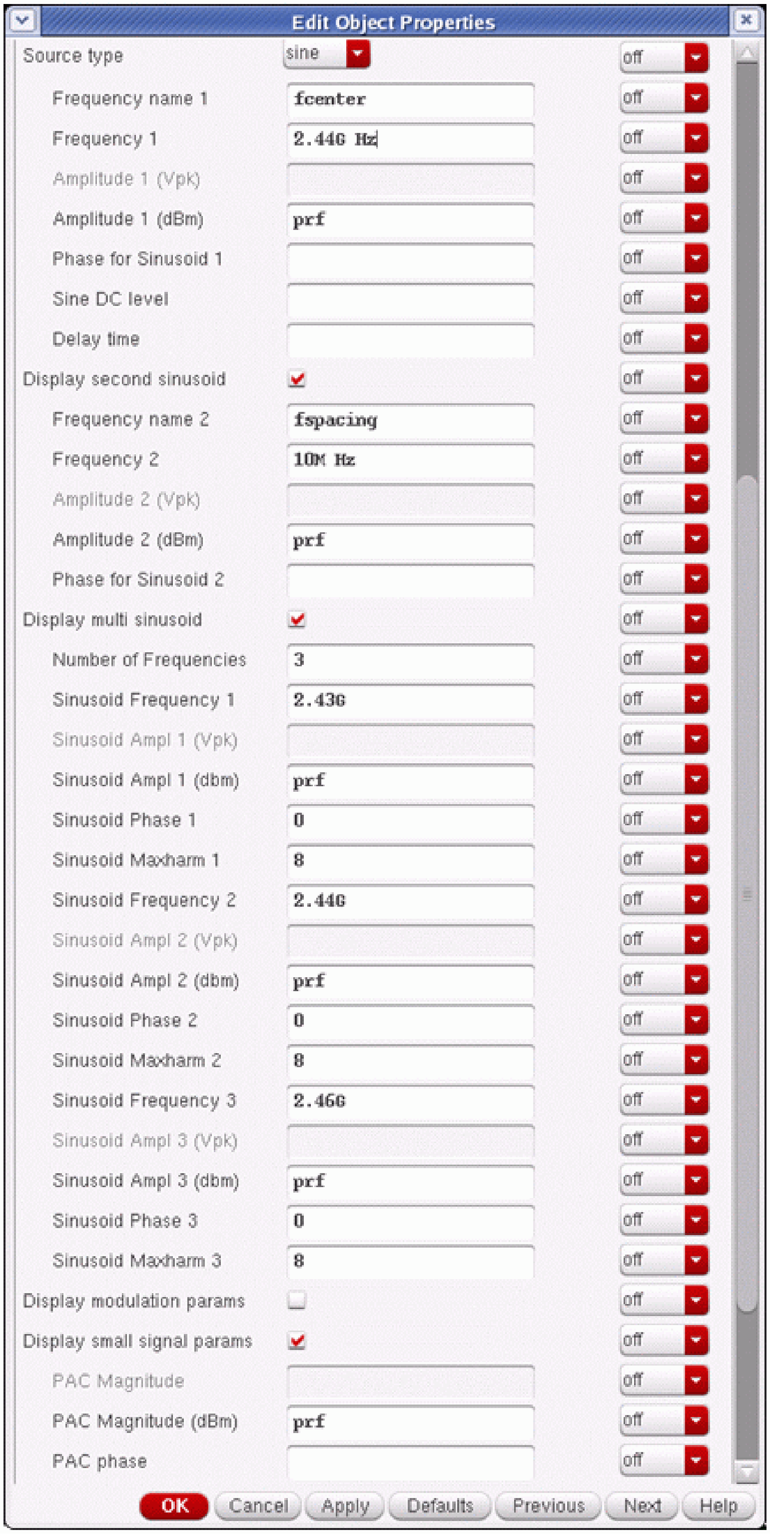
Once the center frequency and spacing are defined, click *Display Multi Sinusoid*. Next, specify the number of carriers you want to generate. There is a maximum limit of nine in one port. If you need more than nine tones, add more ports in series, all with the same center frequency and spacing. Specify the frequencies you need to generate in the additional ports. Divide the port resistance desired by the number of ports you have in series. Note that when you do this, you will need to scale the power in dBm for all the tones in the multi sinusoid because the voltage is set in the port assuming there is a match with the port resistance on the port output. In the case of stacking, the overall resistance is the desired port resistance.

Important

The phase unit for the multi-sinusoid outputs is radians, not degrees.

Once the input frequencies are defined, fields are displayed to specify the frequency and amplitude of each sinusoid to be generated. All the frequencies must be either the center frequency specified in the *Frequency* 1 field, or *Frequency 1* with integer spacing frequency offsets from frequency one.

Below is an example of the port setup to generate three signals.



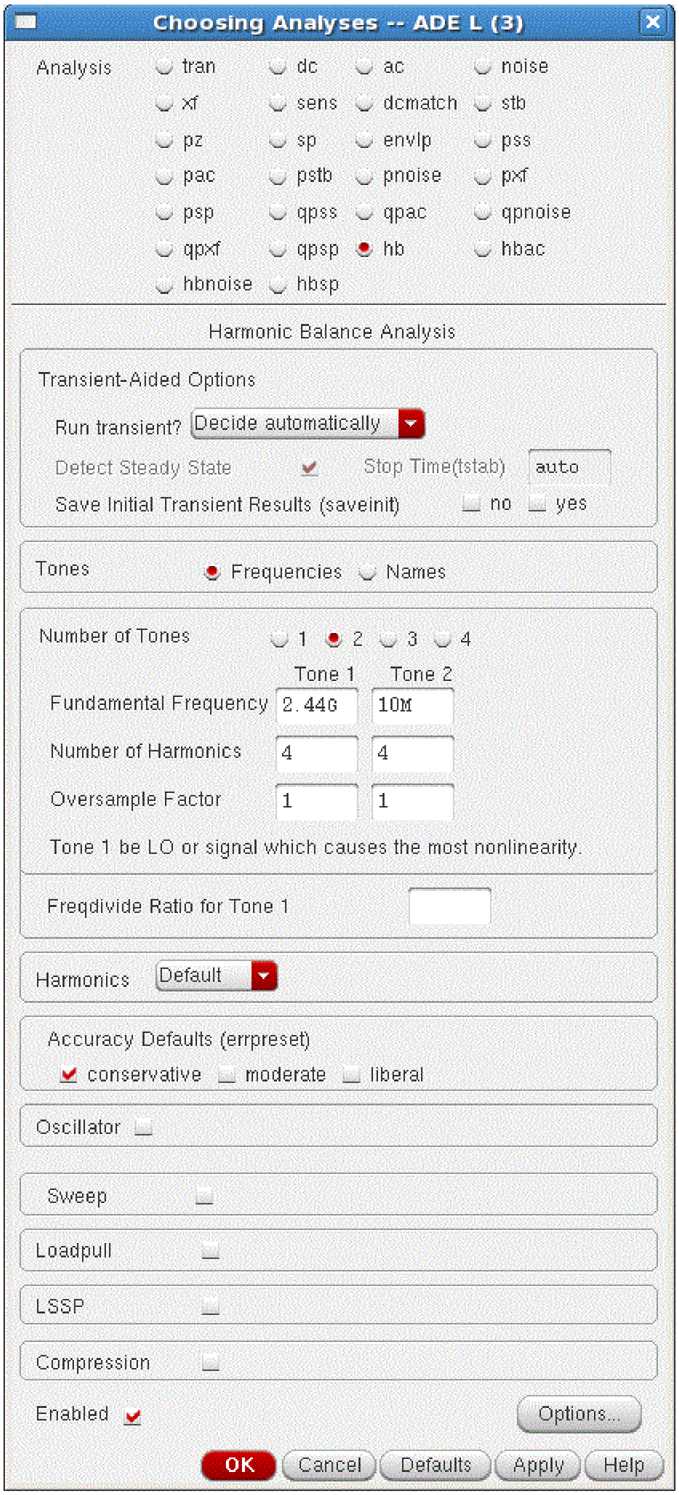
Note that the frequency 2.45G is not generated for this test. For a composite triple beat measurement, one of the frequencies will be missing. For that frequency, just do not specify it.

When the hb analysis is set up in ADE for an amplifier, specify only two input frequencies. The center frequency and the spacing frequency should be specified in the *Fundamental Frequency* field.

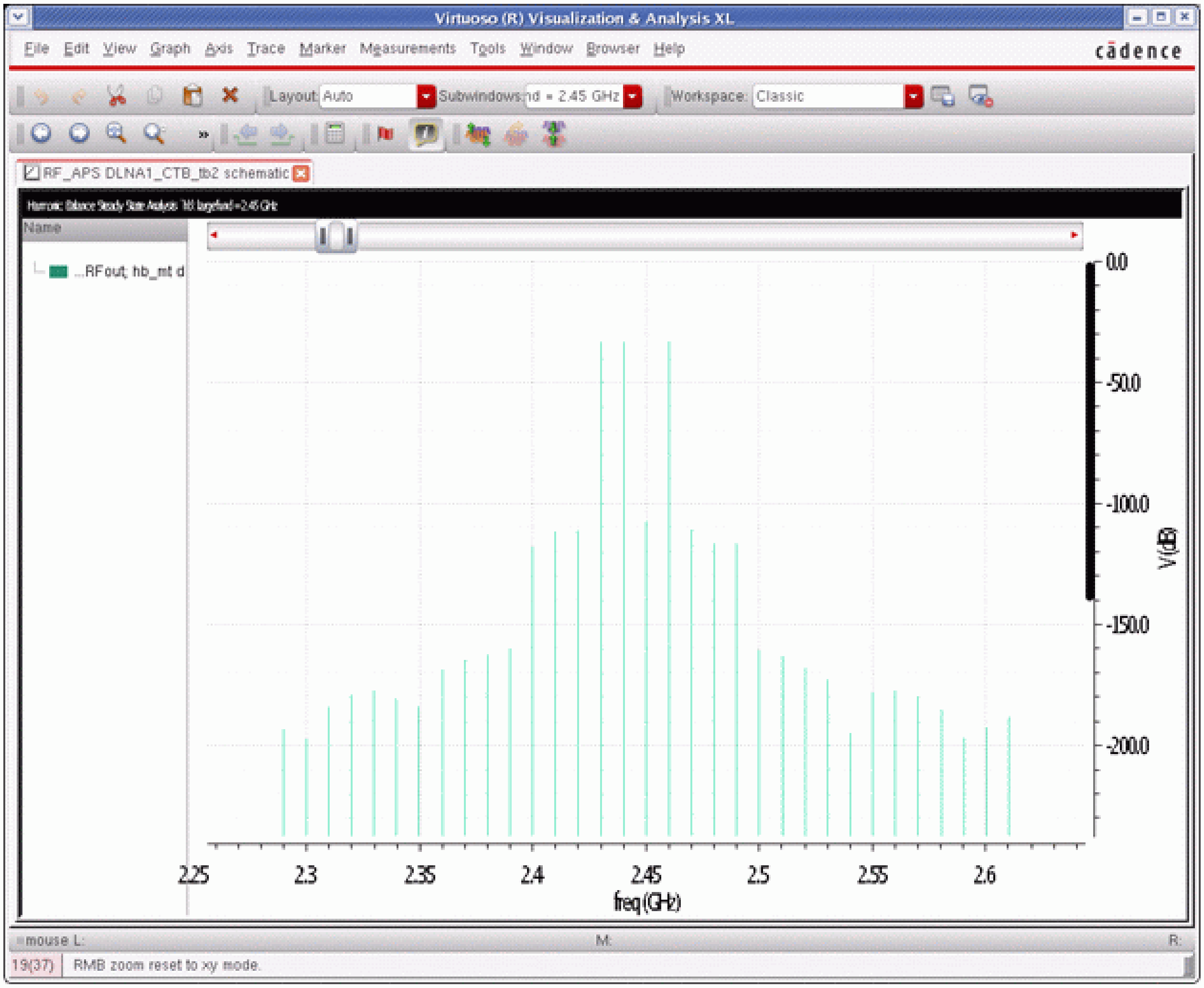
For harmonics, set the total number of possible carriers in the total frequency range. In this case, there are three signals, but there is a fourth possible harmonic at 2.45GHz. In this case, the number of harmonics needs to be four for both tones.

Since all the inputs are sinusoids, *Oversample Factor* is set to one.

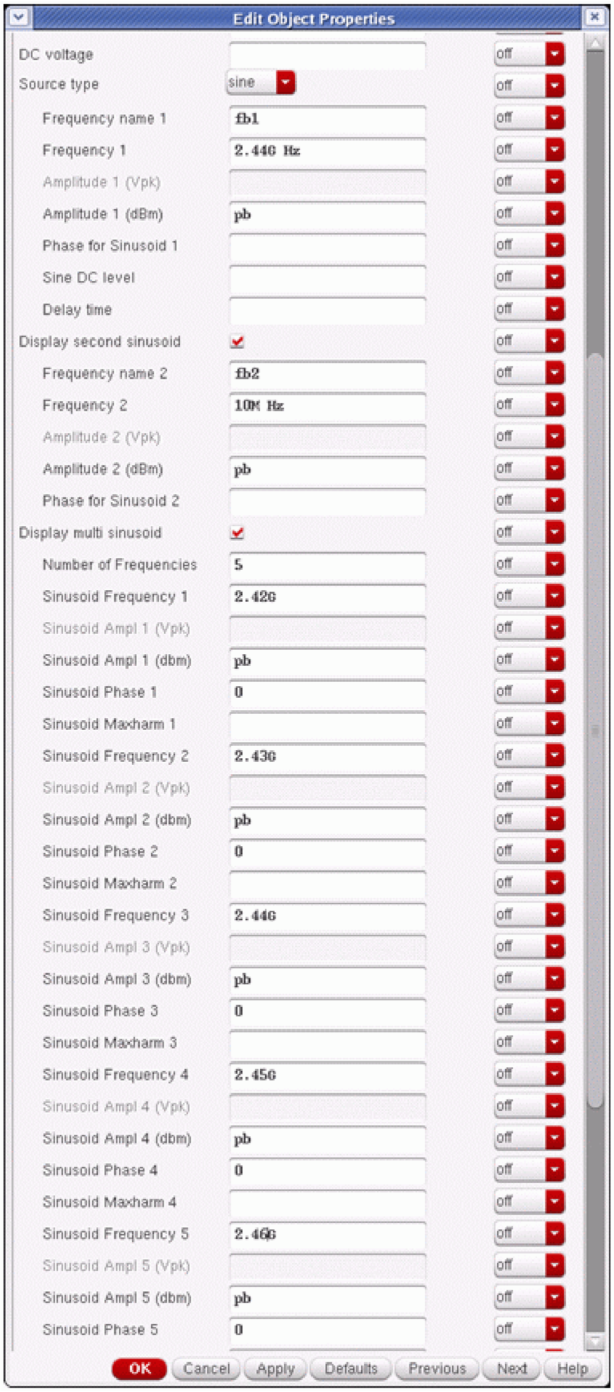
An example is shown below.



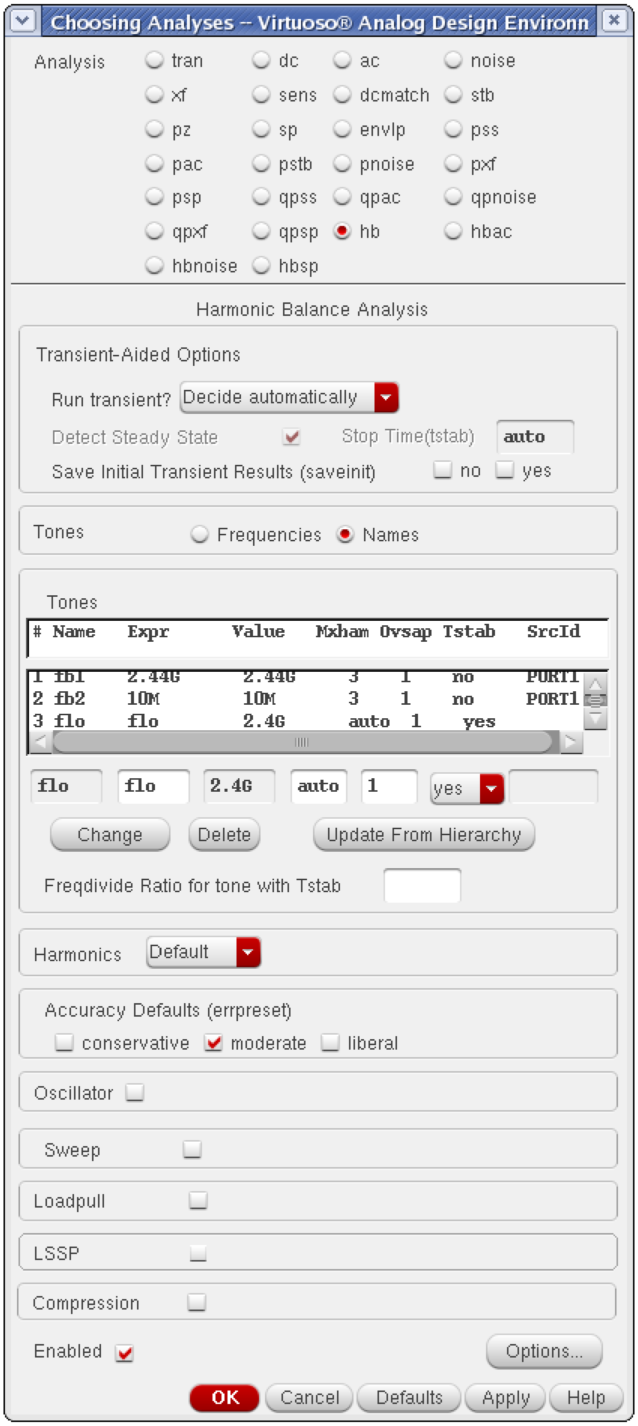
Now run the simulation. The output node of the circuit is plotted.



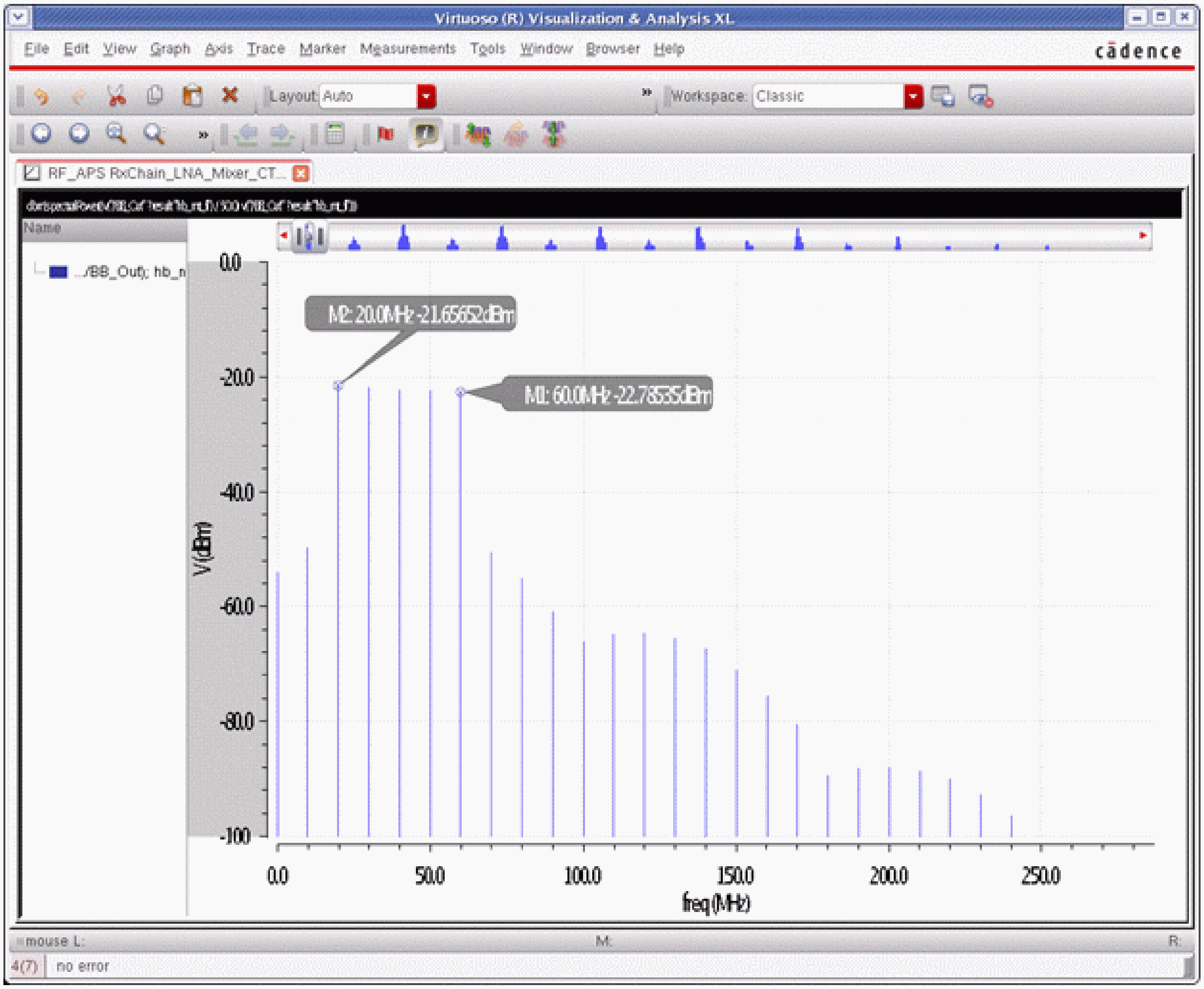
An example with five input frequencies is shown below.



If an LNA-Mixer combination is to be simulated, then there are three tones for this simulation: the center frequency, the spacing frequency, and the LO frequency. For the center and spacing frequencies, five carriers are possible, so five harmonics are set. Ten harmonics are set for the LO. The hb *Choosing Analyses* form is shown below.



Now run the simulation. The down-converted output of the circuit is shown below.



Harmonic Balance AC Analysis (hbAC)

Small-signal Versus Large-Signal Analysis for Conversion Gain

One way to measure conversion gain is to apply the LO and the RF signals in harmonic balance (hb) and calculate the full large-signal response at each RF input frequency, and then plot the output as a function of the sweep frequency. While this works, because all the large-signal effects and many harmonics are calculated, the required time for the simulation can be significant.

The alternative is to use a small-signal approach similar to the AC analysis. The AC analysis runs considerably faster than running the transient at multiple frequencies because it ignores the large-signal effects. In circuits with frequency translation, the AC analysis is unusable because the calculation is based on the DC bias point. In hbac analysis, instead of using the DC bias point, the harmonic balance solution with just the LO is used as the basis of the calculation. The large-signal analysis only calculates the harmonics of a single input, and no mixing products, and then hbac is run after the hb analysis. The overall time savings can be very significant.

The hb solution with the LO calculates the amplitude of the harmonics and it characterizes the nonlinearity of the system. Therefore, in hbac, the output mixing products of the input signal mixing with the LO harmonics can be calculated. Because the signal is considered to be small-signal, harmonics of the hbac test signal are not calculated. Like AC analysis, hbac allows the input to be swept in frequency in order to calculate a frequency response of the output mixing products.

For example, think of a perfectly linear circuit with a single input applied. Because the circuit is perfectly linear, the output spectrum that would be calculated by the hb analysis would have only the first harmonic in it. Because the circuit is linear, if a second input was applied, no frequency translation can occur.

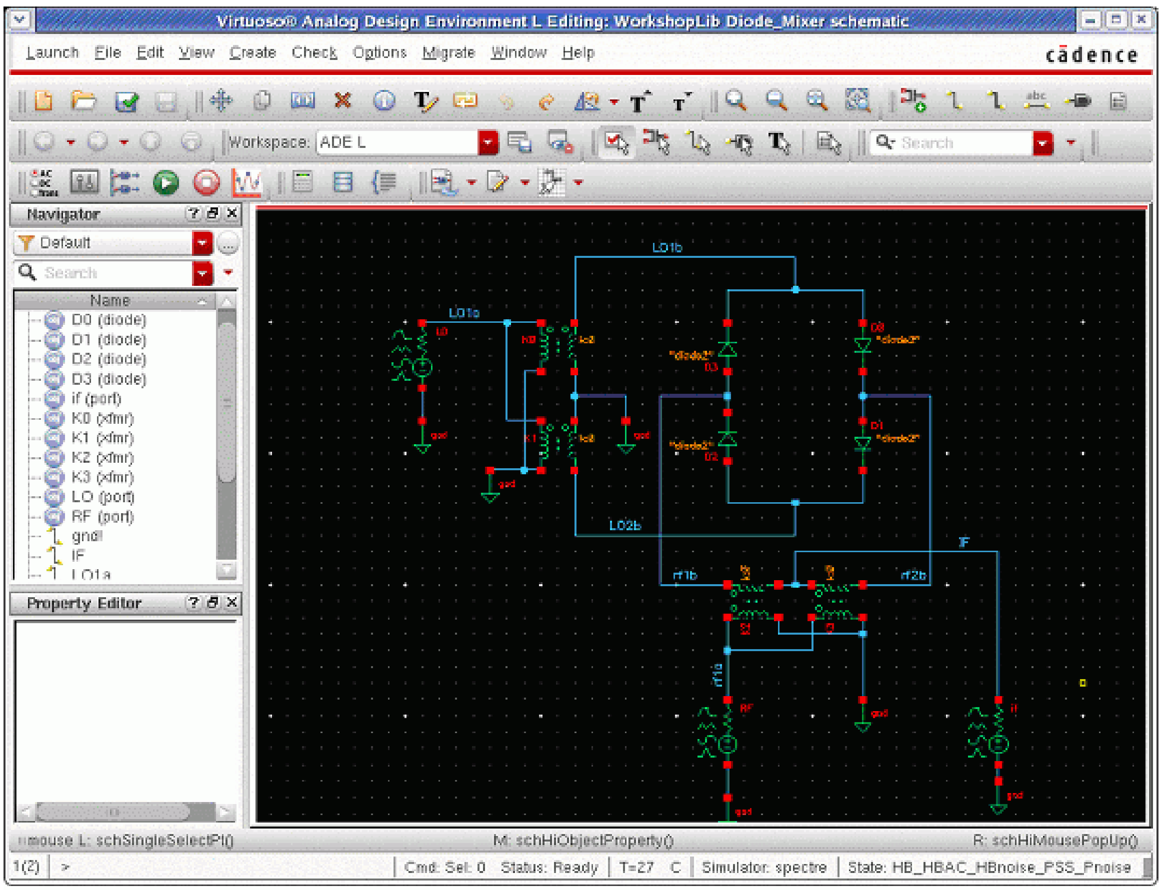
Now imagine a real circuit with a single small input amplitude signal at 1GHz. Because the circuit has nonlinear devices, the output will contain 1GHz and the integer multiples of 1 GHz. Because the input signal is small, the amplitude of the harmonics is small, and if a second input were applied at 1.1 GHz, that signal would be very weakly converted when it is mixed with the harmonics of the circuit. Certainly, 1.1GHz would appear at the output with no mixing. 100MHz and 2.1GHz are produced by mixing with the 1GHz fundamental frequency. 900MHz and 3.1GHz are produced when 1.1GHz mixes with the second harmonic of the 1GHz input. 1.1GHz mixes with the other harmonics as well, producing a wide range of output frequencies from a single input frequency.

Now imagine that the 1GHz signal is large. The output harmonics above the first harmonic are much larger. If the input went up 10dB, the first harmonic goes up 10dB, the second harmonic goes up 20dB, the third harmonic goes up 30dB, and so on. If a second input at 1.1GHz were applied to the circuit, this mixes with the harmonics of 1GHz and more strongly mixes up and down in frequency.

Therefore, the harmonic content of the hb analysis contains information about how strongly a circuit translates frequencies, and this is used by the small-signal analyses hbac, hbnoise, and hbsp in order to calculate the small-signal conversion gain. This section is about hbac analysis.

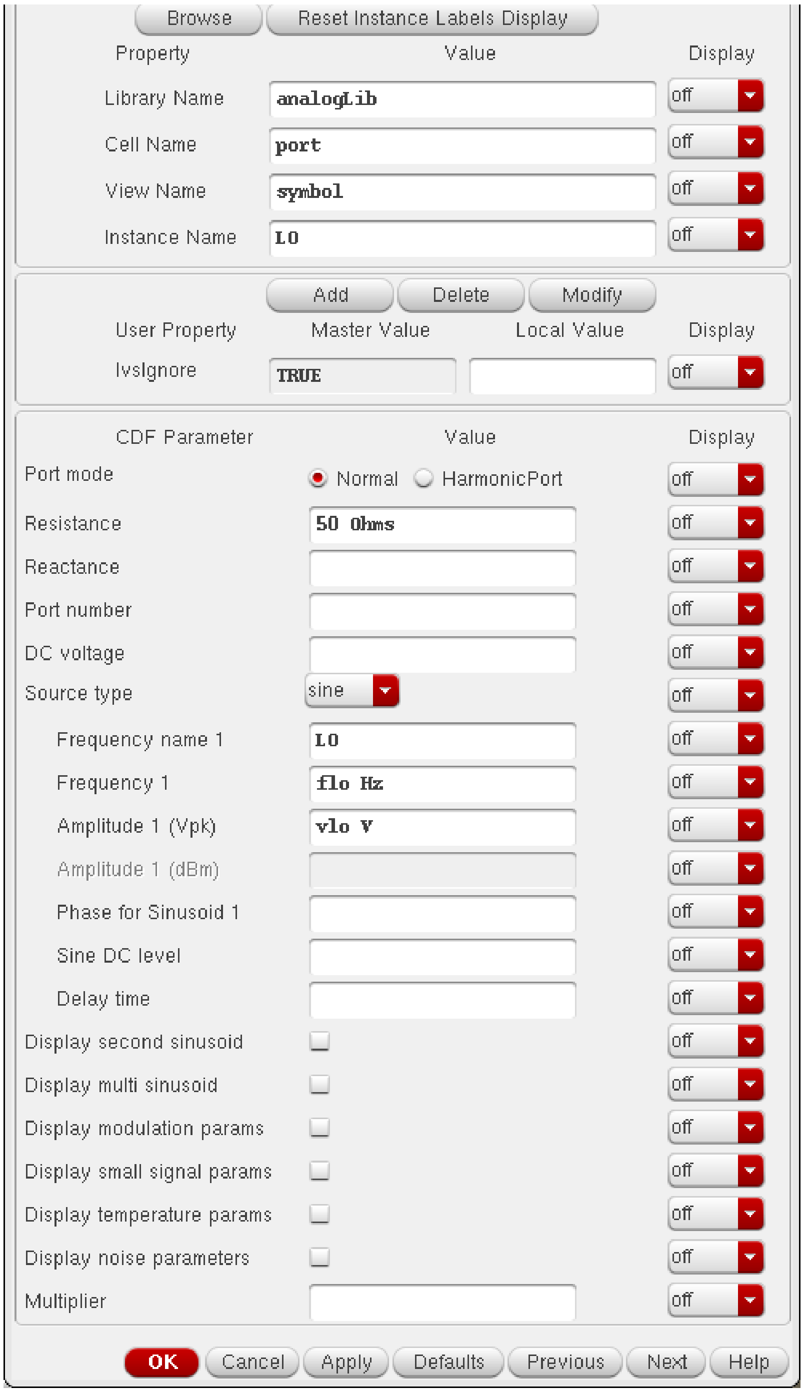
Example: Conversion Gain (Down conversion)

Consider the double-balanced diode mixer shown below.



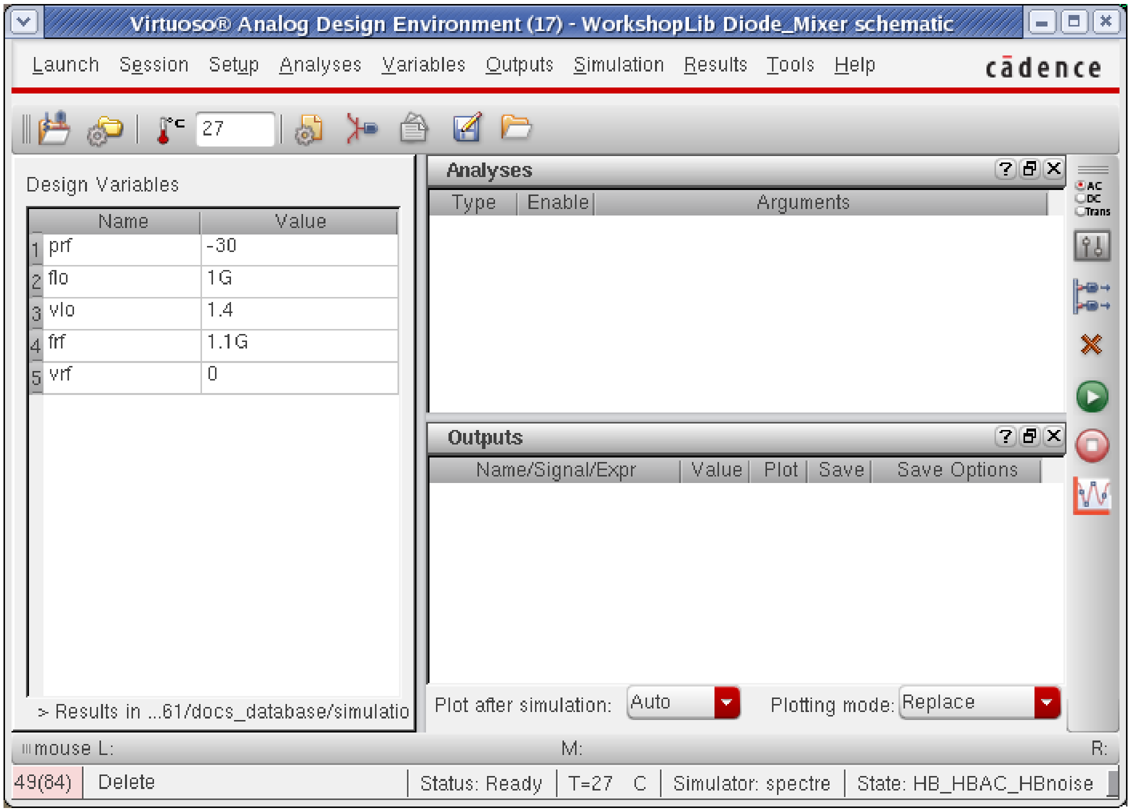
The RF input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed.

The property list is shown below.

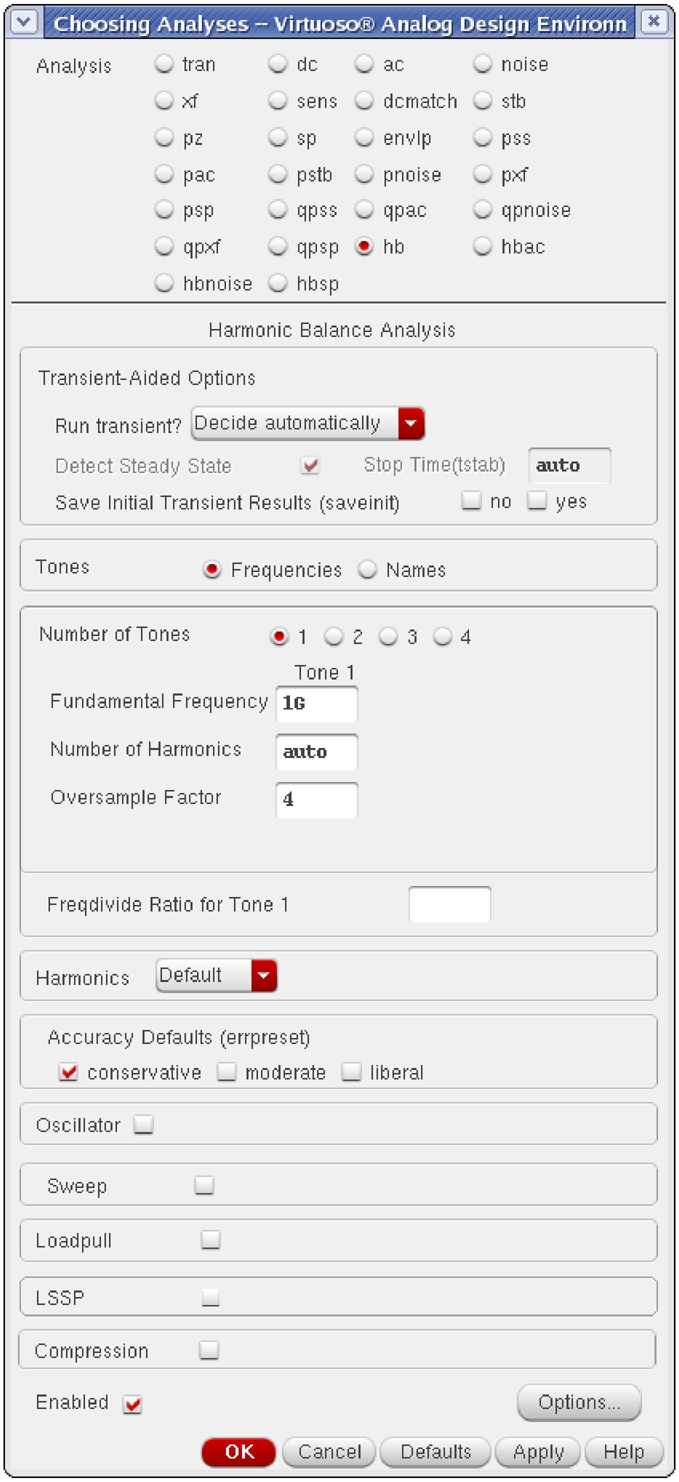


The easiest way to get the variables list into the ADE is to select *Variables -> Copy From Cellview*. To set the values, select the value field to the right of the variable, type the desired value, and press *Enter*.

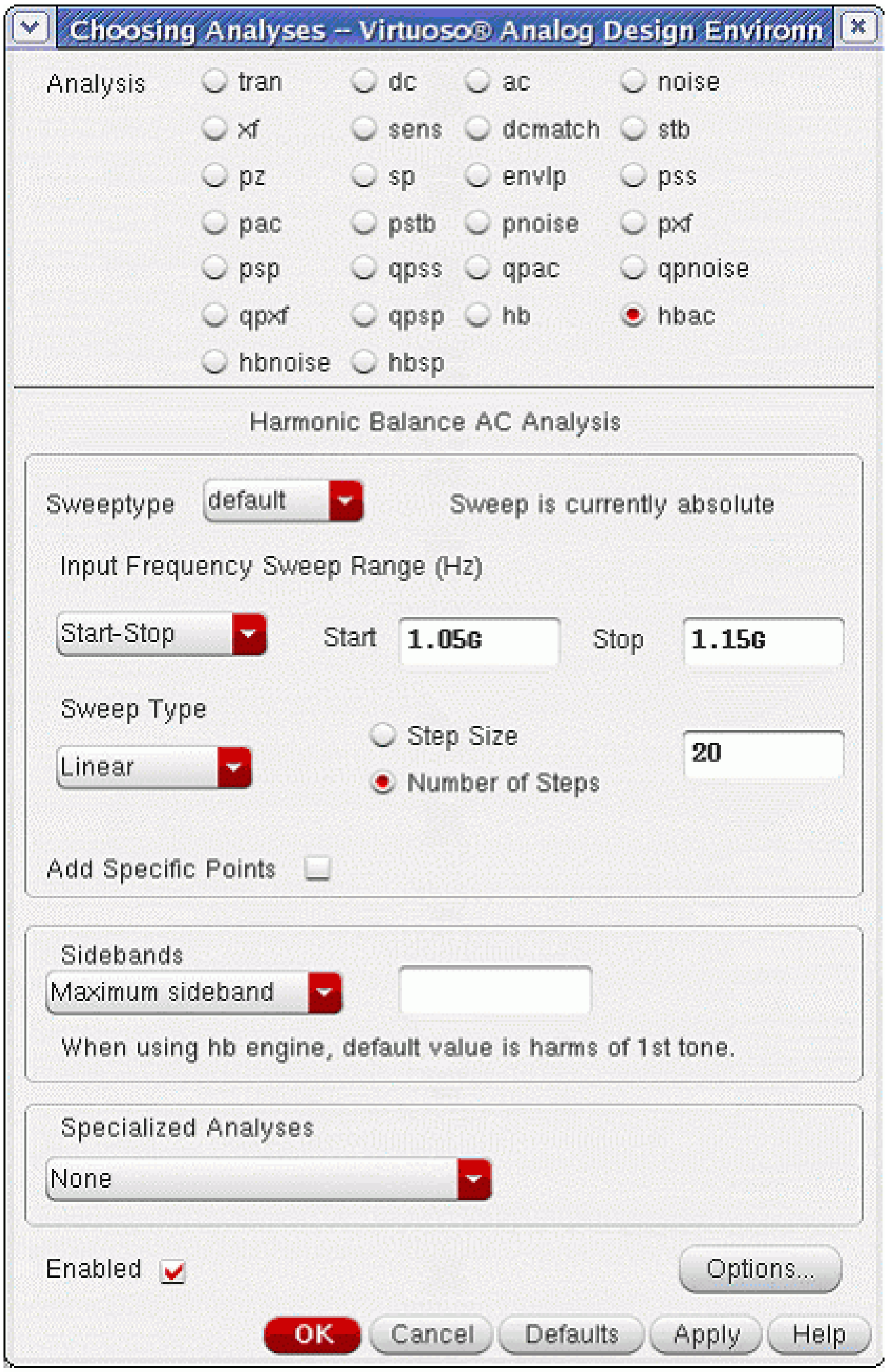
The LO frequency is 1GHz and its amplitude is 1.4 volts peak. This is +13 dBm. The RF frequency is 1.1GHz, but it is disabled because the amplitude is set to 0 (zero). Setting either the frequency or the amplitude to zero disables the production of that waveform from the port.



The hb analysis needs to be set up first. The example below shows a single input at 1GHz with 51 harmonics. For more information, see the harmonic balance section.

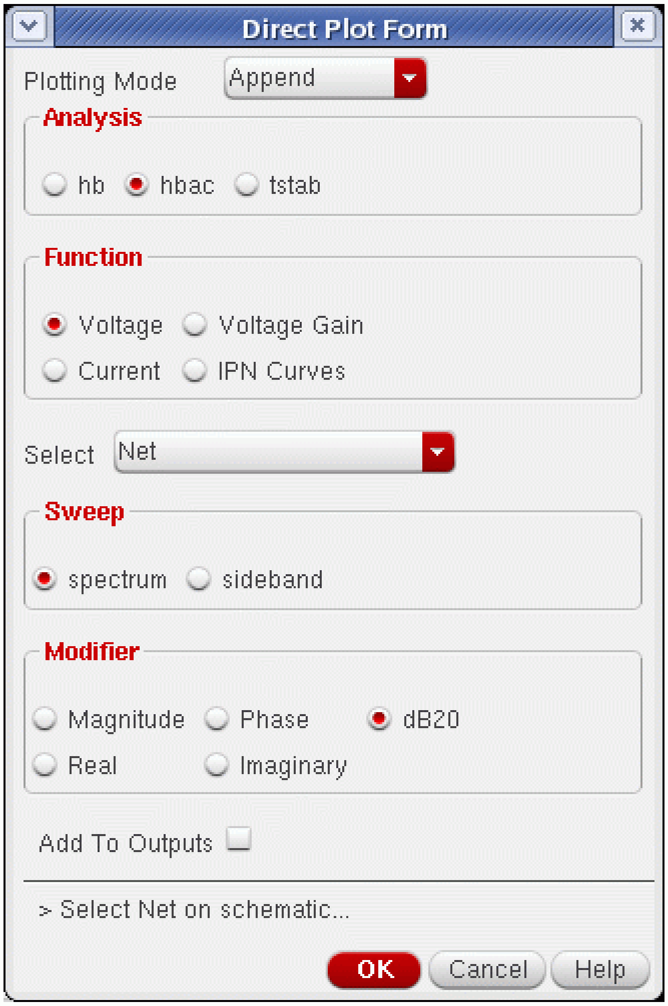


Next, the hbAC analysis needs to be set up. Like AC, the frequency range specified in the form is the input frequency range. *Maximum sideband* should be left blank so that all the mixing products from all the harmonic balance harmonics are present in the output of hbac.



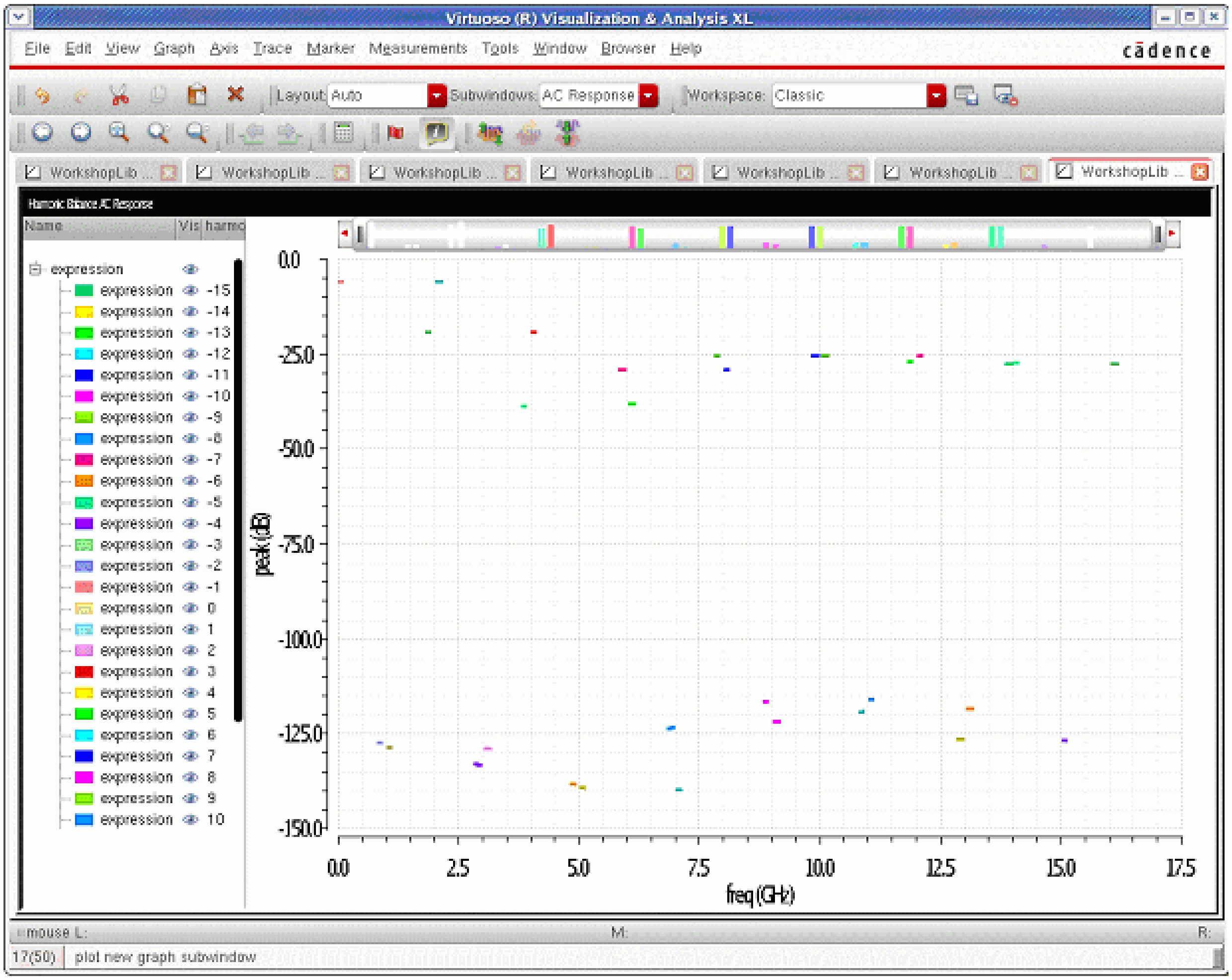
Once the analyses are set up, run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form*.

The example below shows a typical plot.

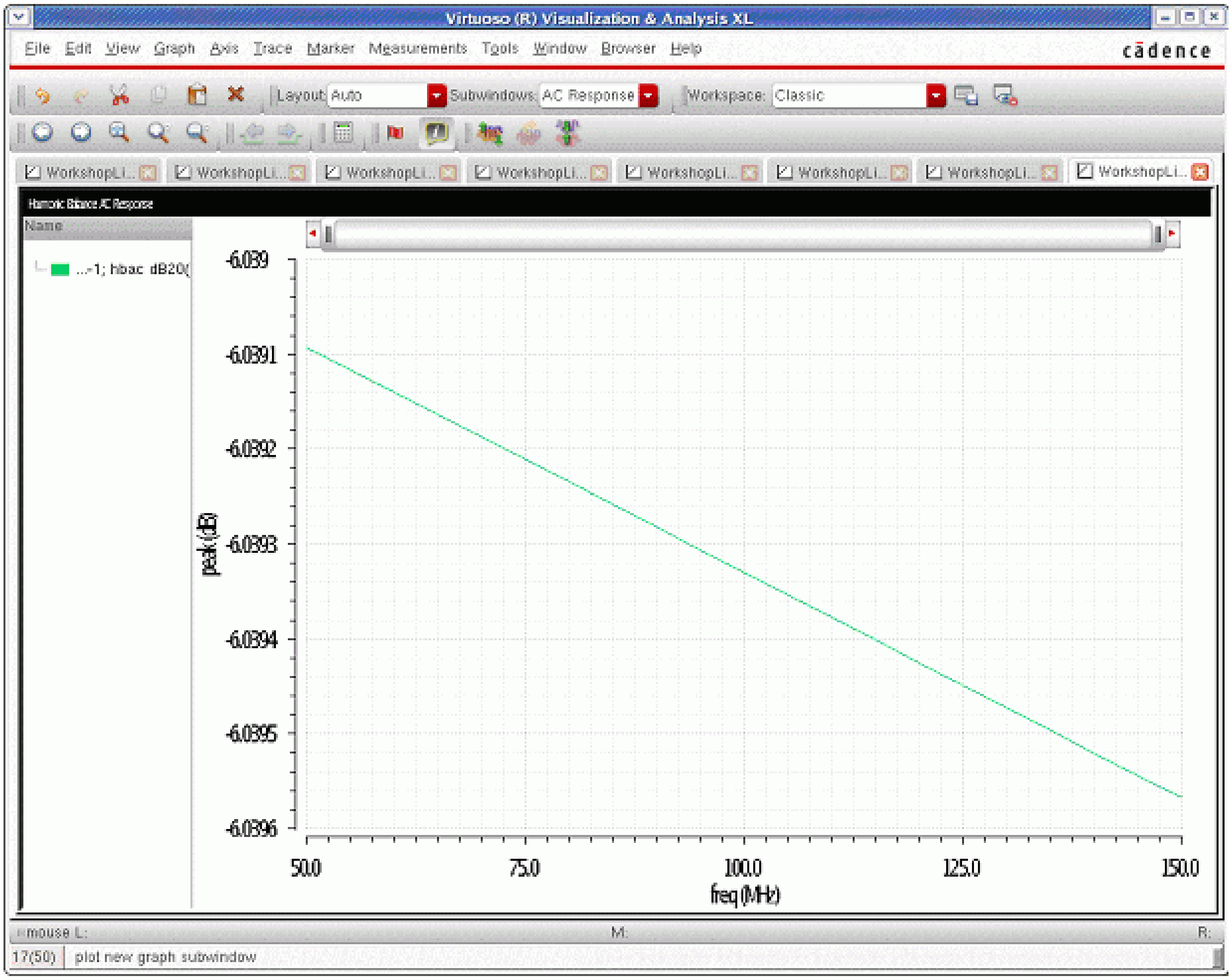


When the net in the schematic is selected, the waveform is displayed.

Note that all the short line segments are the mixing products that are produced when mixed with the harmonics in the harmonic balance large-signal analysis.

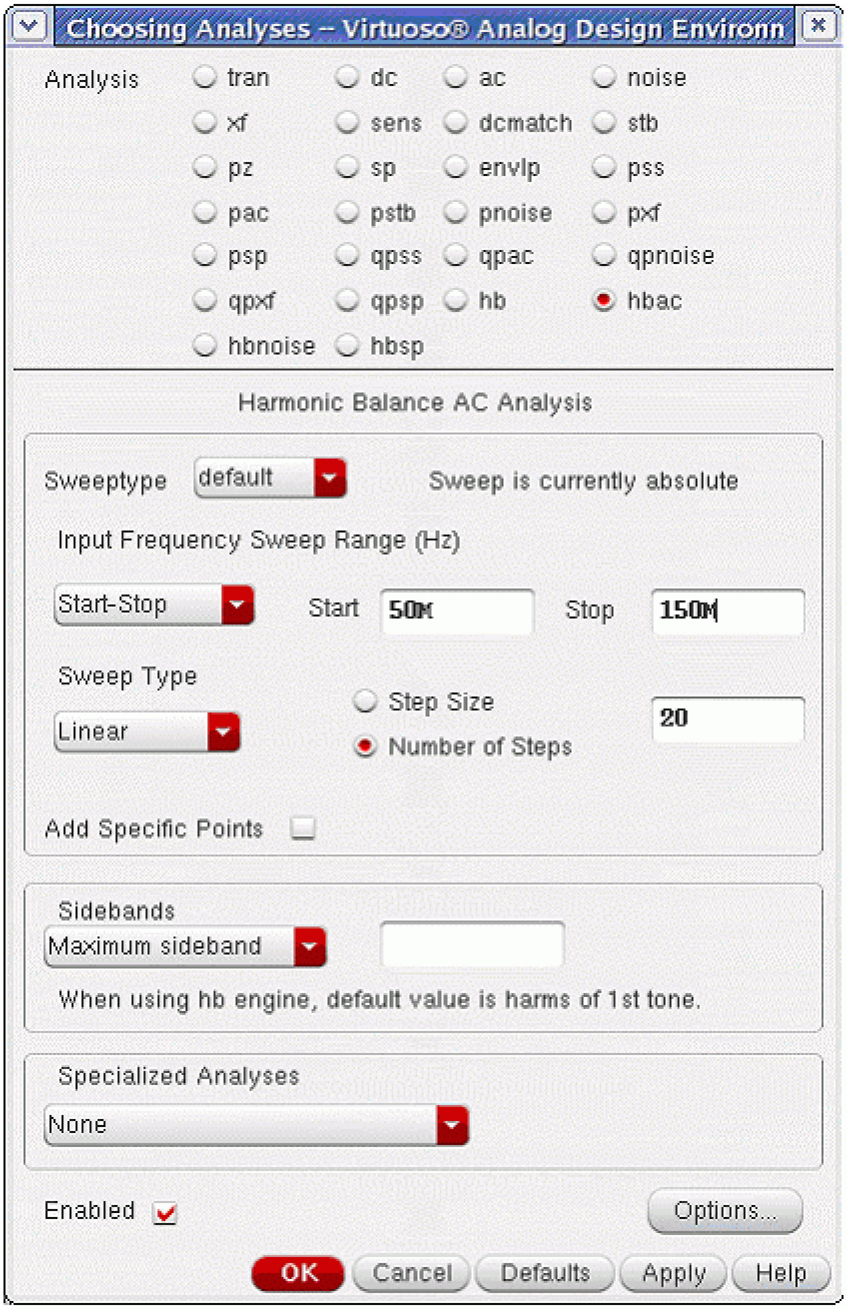


Usually, the frequencies that are produced for the down-converted product are desired. In the *Direct Plot Form*, select *sideband* instead of *spectrum*, and then select the desired frequency range from the list at the bottom of the form. Only that mixing product will be displayed, as shown below. The conversion loss is about 6dB.

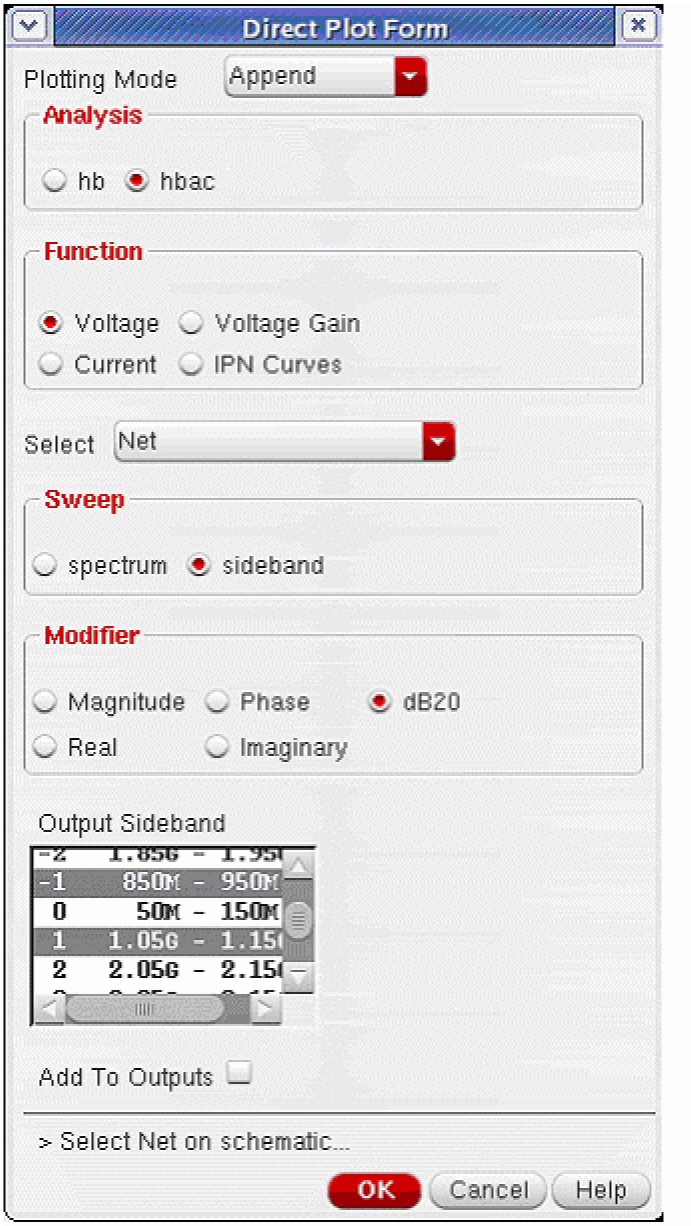


Example: Conversion Gain (Up Conversion)

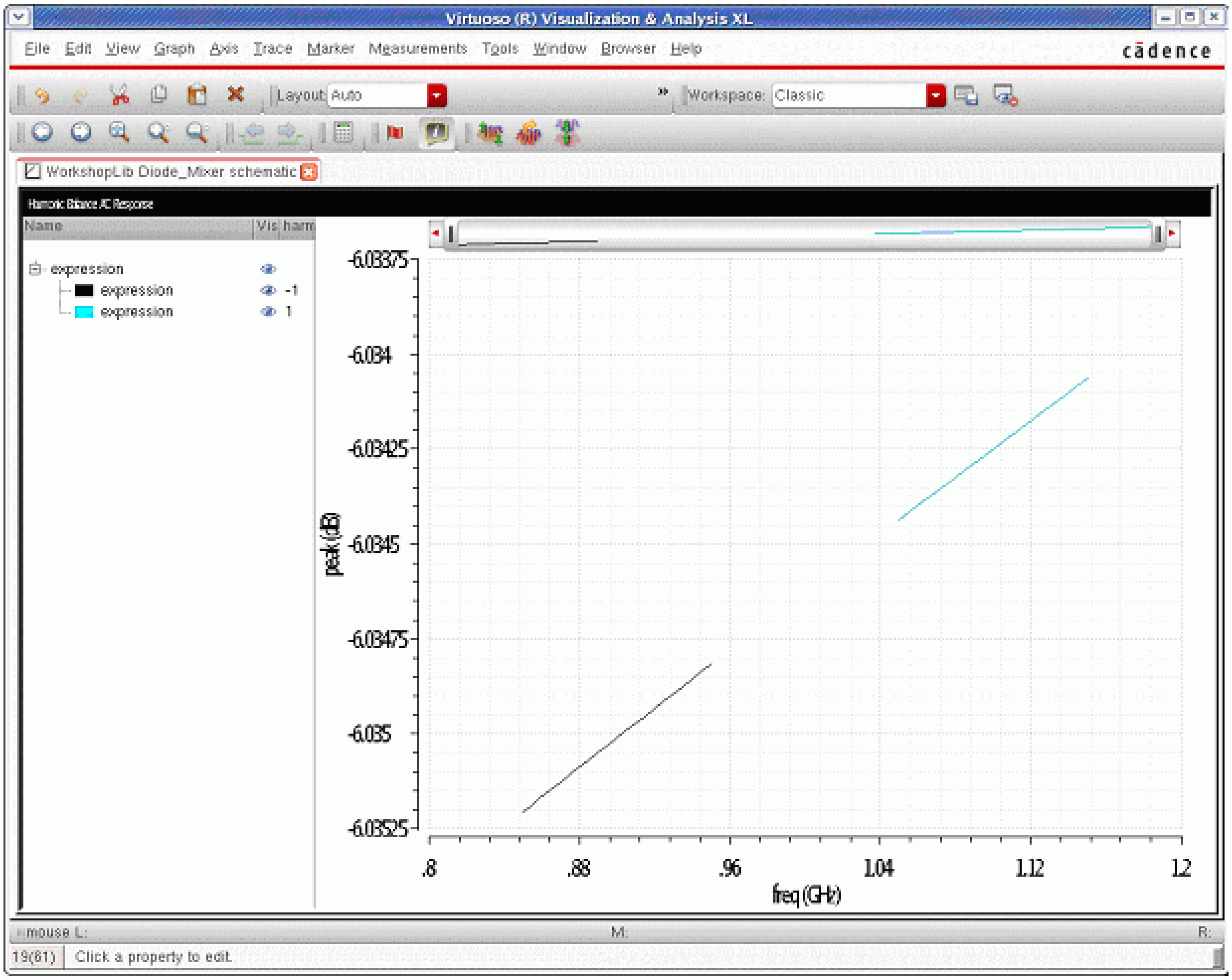
The process is similar for an up-conversion mixer. In this case the input frequency is low, as shown below.



In the *Direct Plot Form*, both the outputs above and below the LO frequency might be desired. Select the first frequency, and click the second while holding the <*Ctrl*> key. Now select the node in the circuit and the results will plot.



Note that the conversion loss is almost identical for both mixing products.

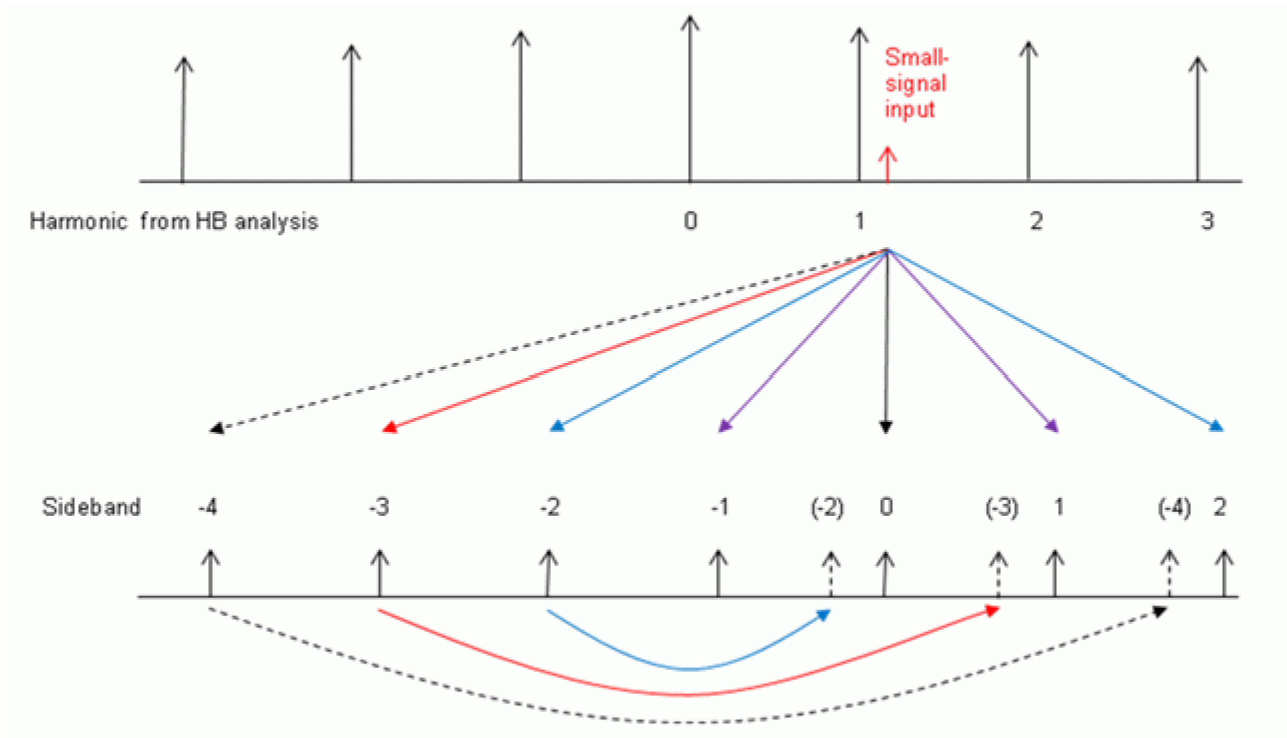


Overview of Simulation Capabilities

Specialized Analysis=None

Down Conversion: Input Near the First harmonic

At the bottom of the hbac *Choosing Analyses* form, there is a selection called *Specialized Analyses*. The default is *None*, and this is for measuring the conversion gain that is the average conversion gain with the large signal applied to the circuit (usually the LO). The mixing products that are produced are called sidebands. The sideband number is the harmonic number that is being mixed with to provide the output. A positive number means that the output mixing product is higher in frequency than the input frequency specified in the *Choosing Analyses* form. A negative number means that the output is lower in frequency than the input frequency. A diagram is shown below.



We tend to think of positive frequencies. The default is to reflect negative frequencies to positive frequencies.

The sideband number is the harmonic that the input mixes with. Negative numbers are downconverted in frequency, and positive numbers are upconverted in frequency.

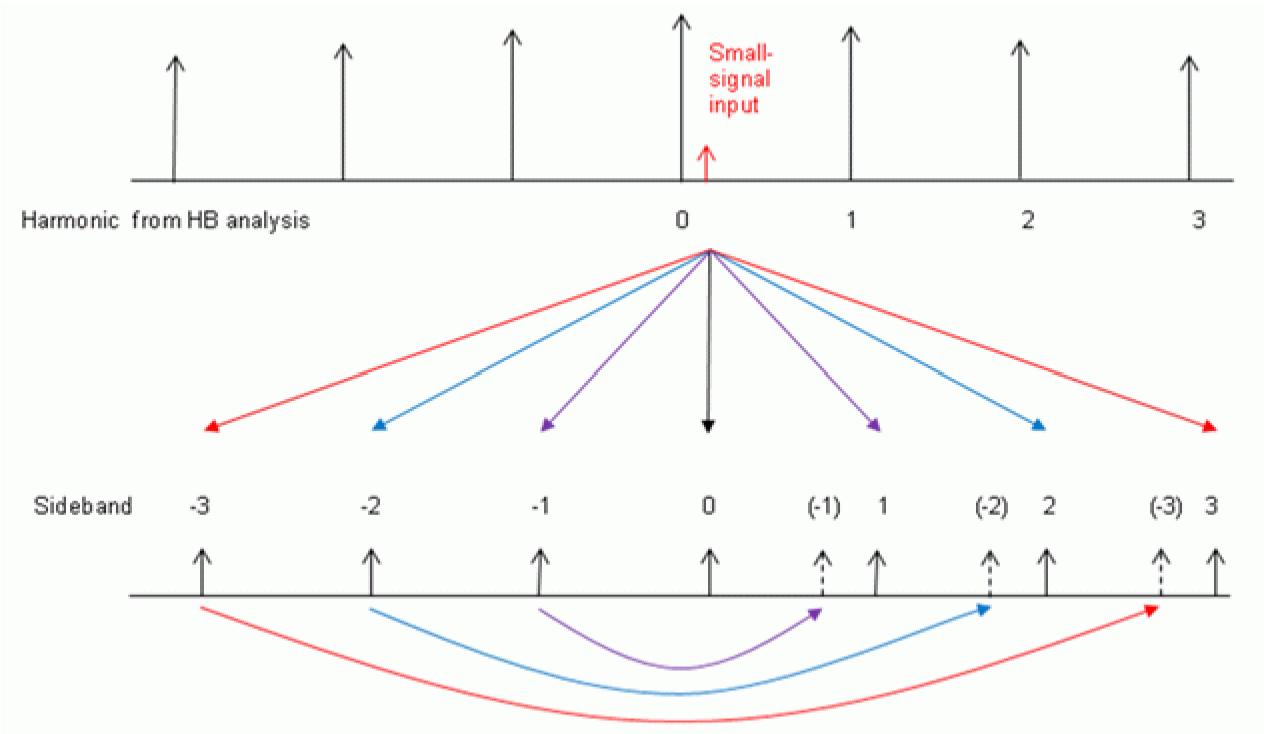
Again, in hbac, a single input frequency is applied at a time, and then the output mixing products are calculated through the highest harmonic of the harmonic balance simulation specified by the *Maximum sideband* parameter. Leaving that parameter blank is recommended, and this calculates the mixing products for all the harmonics in hb. There is no run-time penalty for calculating all the mixing products. There is a runtime penalty for increasing the number of harmonics in the hb analysis.

Note that the input causes harmonics and the output mixing products cause harmonics. Consider a mixer with a 1GHz LO and a 1.1GHz input. All the LO harmonics are calculated by the hb analysis. When 1.1GHz mixes with 1GHz, an output at 100MHz is produced. Because the circuit is nonlinear, harmonics of 100MHz are produced. hbac is a small-signal analysis, and the assumption is that those harmonics do not matter. They are not calculated.

In practical circuits, this limitation is not usually significant because generally, we work with the output frequency itself, not its harmonics.

Up-conversion: Input Near Zero Frequency

In a similar manner, up-conversion is shown below.

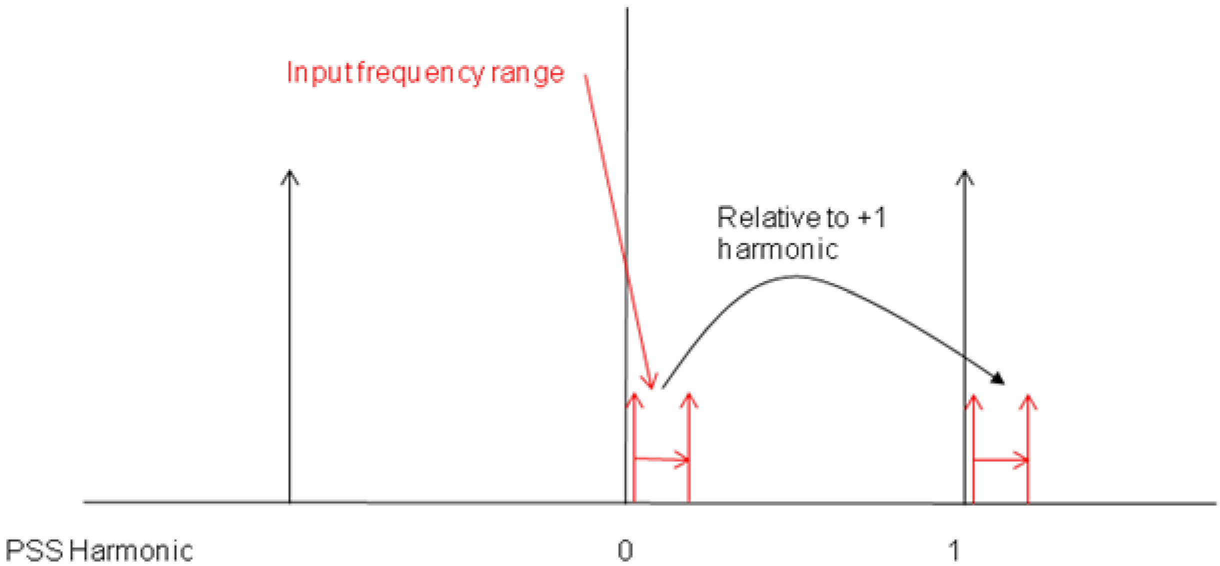


Frequency Sweep

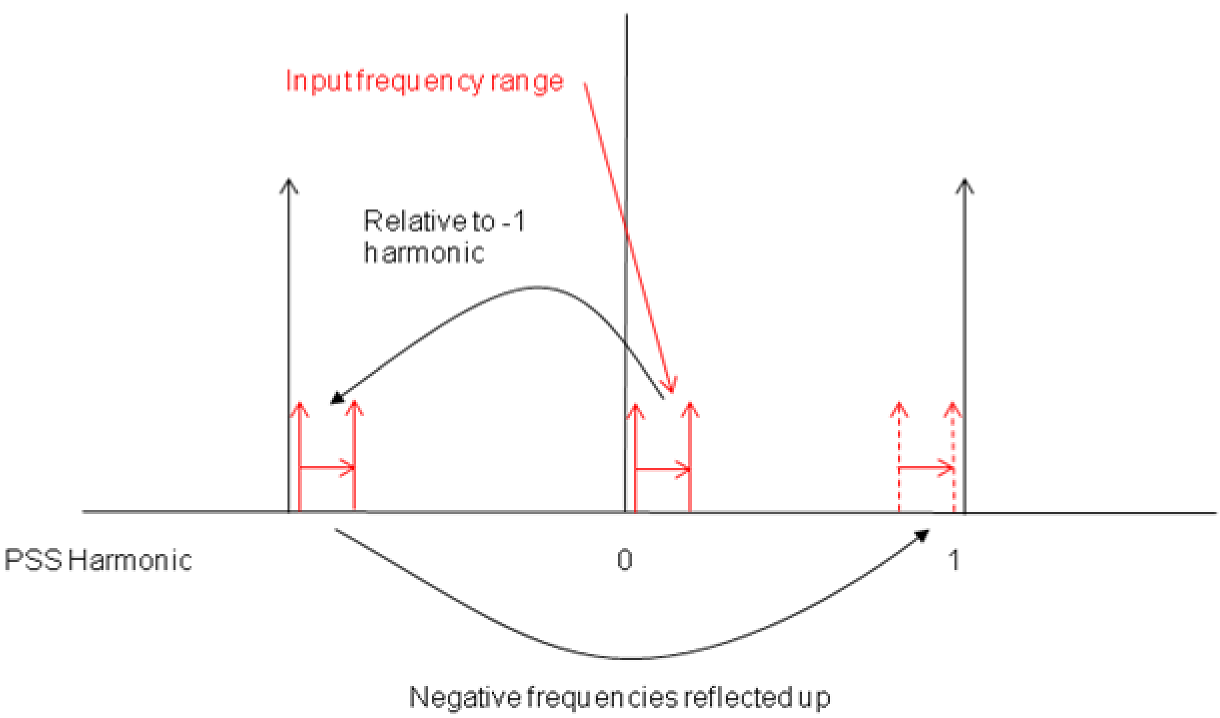
This analysis is like AC, where the input frequency is swept. The input frequency range is specified in the hbac *Choosing Analyses* form.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonic in the harmonic balance analysis. For example, assume that the harmonic balance is at 1GHz, and a log sweep is desired above the first harmonic of the hb analysis. In this case, you need to select *relative* sweep, and specify 1 in the *Relative Harmonic* field. Next, type 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade.

The sweep will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.



To use a log sweep below the first harmonic, use the same frequency range, and type -1 as the relative harmonic. This is illustrated below.



Maximum Sideband

Conceptually, an infinite number of mixing products are produced when a single input is applied to a nonlinear system. From the practical point of view, usually only a small number of mixing products need to be measured.

There is a property in the *Choosing Analyses* form called *Maximum sideband* that is used to define how many mixing products should be calculated. If *Maximum sideband* is zero, the hbac analysis will only contain the frequency that appears without mixing or aliasing of any type. This is not equivalent to a linear AC analysis because hbac takes into account the instantaneously varying nature of the LO signal because of the large signal being present in the hb analysis.

When *Maximum sideband* is one, the outputs without frequency translation and the outputs that mix or alias with the first harmonic are present in the hbac analysis. When *Maximum sideband* is 10, the hbac analysis includes mixing through the 10th harmonic of the harmonic balance analysis. Usually the number of harmonics in the hb analysis and the maximum number of sidebands should agree. This is easily accomplished by leaving the *Maximum sideband* field blank in the *Choosing Analyses* form in ADE.

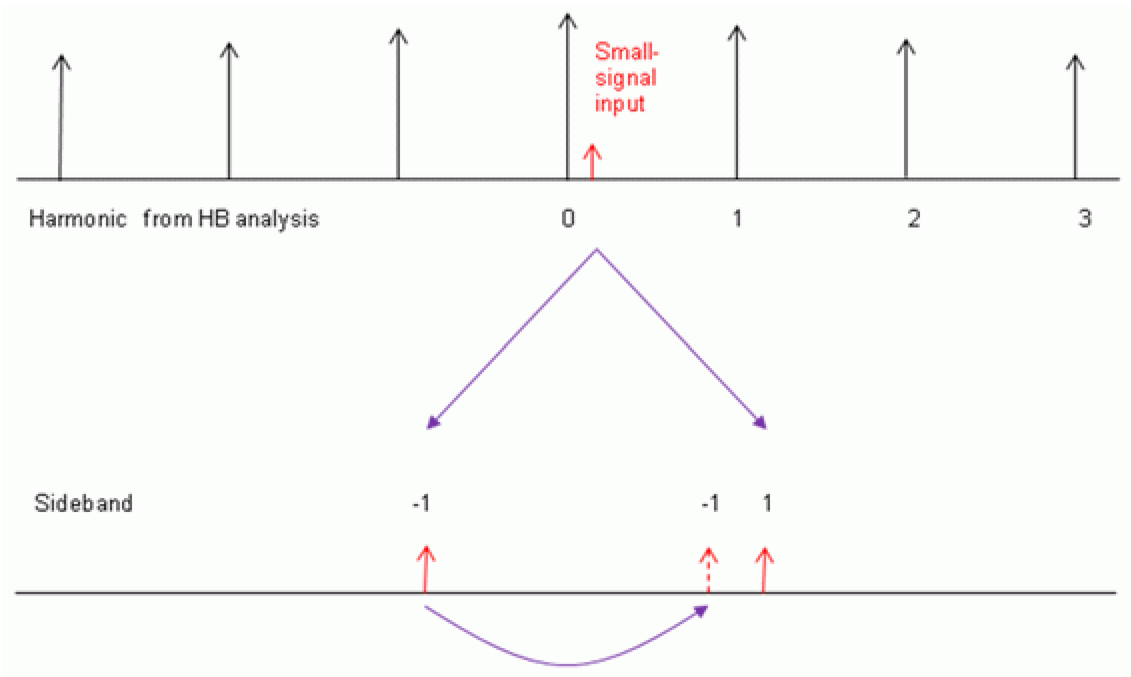
Setting Harmonics and Sidebands

The process of setting harmonics and sidebands is similar to setting the maximum number of harmonics for the harmonic balance simulation. Start with an estimate based on the harmonic content of the input signal(s) and the nonlinearity of the circuit. Then run a simulation. Raise harmonics and sidebands and run again. If the conversion gain measurement did not change, then the original number of harmonics and sidebands might be able to be reduced. If it did change, then more are needed. Getting a stable hbac output may take more or fewer harmonics and sidebands than the harmonic balance simulation by itself.

Modulated

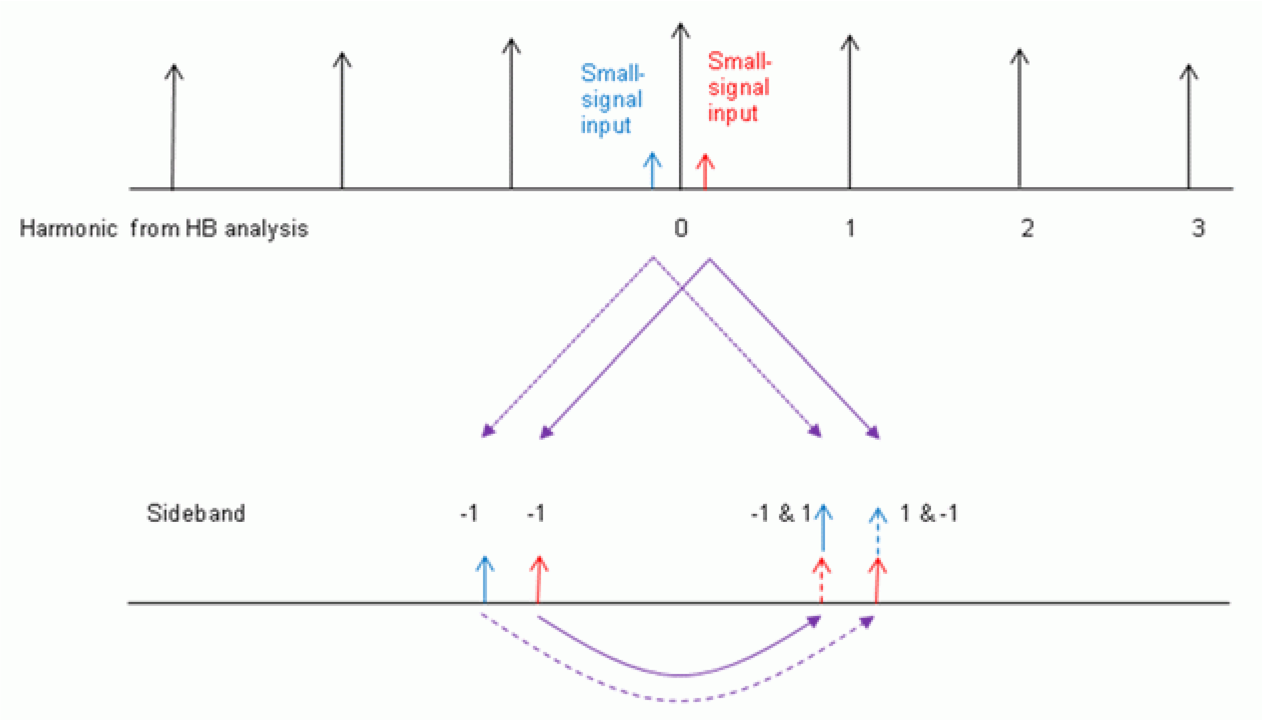
Modulated hbac is typically used for measuring AM to PM measurements in driven circuits and oscillators. In this case, the input is usually the power supply ripple at low frequency, and the desired output is near the first harmonic of the output. Hbac apples a modulated input signal to the circuit, and is able to measure a modulated output signal from the circuit. The types of modulation are Single Side Band (SSB), AM, and PM.

A diagram is shown below for ssb input.



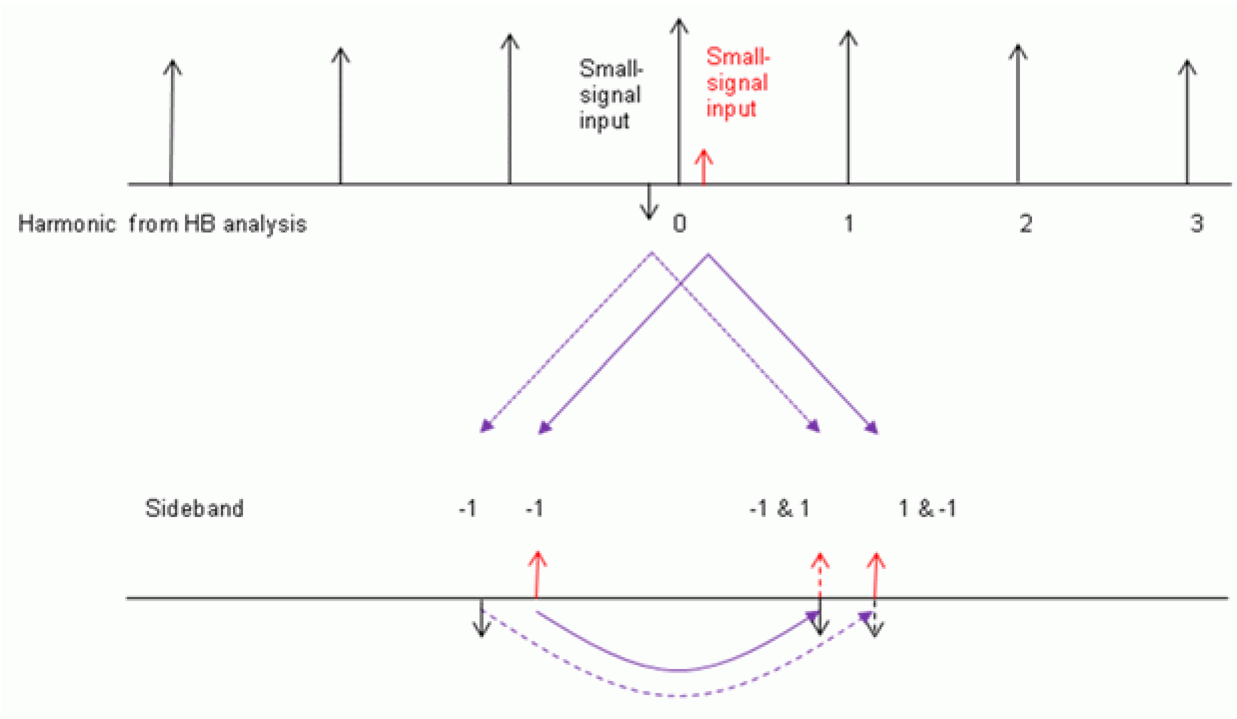
Note that although the input occurs at one frequency, the AM output can be measured directly from the single input.

When AM is applied at the input, the following diagram is shown.



The difference is that in addition to the input at low positive frequency, another input (to form an AM signal) is applied at low negative input. This second input adds to the output signal near the first harmonic.

For PM, the negative input amplitude is negative, as shown below.



By running all three inputs, the SSB/AM/PM input to SSB/AM/PM output can be calculated. This is a setup choice in the *Choosing Analyses* form.

In the *Choosing Analyses* form, the input frequency range, and the output harmonic must also be specified.

All the plot functions are implemented in the *Direct Plot Form*.

Rapid IP2/IP3 (AC and PAC)

Note that Rapid IP2 and Rapid IP3 in hbac can only be selected when a single tone hb analysis is run. This would be the LO signal for a mixer. hbAC and AC add the ability to have multiple input frequencies. The Rapid IP2 and Rapid IP3 that is available in the AC analysis form allows the calculation of IP2 and IP3 for amplifiers, or other circuits where there is no frequency translation. In hbac, Rapid IP3 and Rapid IP2 are provided for circuits that translate frequencies, whether that translation is an up or down conversion. Note that hbac and ac are small-signal analyses, so the limitation is that the measurement is a small-signal IP3. If you need a large-signal IP2 or IP3 measurement, use a large-signal analysis like hb or qpss to make the measurement. Note that when you are using Rapid IP2 or Rapid IP3 in the AC analysis, either a SpectreRF license is required, or an additional mmsim token is required to run the analysis.

hbAC and AC Rapid IP3 also add the ability to calculate the harmonics and mixing products that are generated by the two input frequencies. Make sure that the input level is in the small-signal range. Rapid IP2 and IP3 have exceptionally small numerical noise floors, therefore, it is recommended that you specify a level that is too small, rather than too large.

The plot functions for Rapid IP2 and IP3 are implemented in the *Direct Plot Form*. Select *Results - Direct Plot - Main Form*. In the *Direct Plot Form*, select *ac* or *hbac* from the *Analysis* section. Next, select *Rapid IP2* or *Rapid IP3* (depending on which was set up) from the *Function* section, and click *Plot*. The X and Y points are shown for the intercept. The X value is input-referred IP2/IP3, and the Y is output-referred IP2/IP3.

Triple Beat

Triple Beat is similar to Rapid IP3, except three inputs are allowed. This is provided for the case where there are two small-signal tones closely spaced in frequency from the transmitter that leak in to the receiver along with a small-signal intended RF input signal. In this case, a third order product is produced at the transmitter tone spacing frequency away from the RF input frequency. Note that triple beat is a small-signal analysis. The amplitude of all the tones must be in the small-signal region, or the result will be incorrect.

Compression and IP2 distortion summary

The distortion summaries measure distortion in an amplifier in AC or in circuits that have frequency translation in hbac. The idea is to find which devices in a signal path contribute relatively more or less to the distortion. Since IP3 and compression are related mathematically, the compression distortion summary provides information about which devices contribute the most amount of third order intermodulation distortion. The IP2 distortion summary measures which devices contribute to the second order distortion. Note that when you are using the distortion summaries in the AC analysis, either a SpectreRF license is required, or an additional mmsim token is required to run the analysis.

Both distortion summaries use the *PAC Magnitude* property in the input port. Make sure that the PAC Magnitude (in volts or dBm) is actually in the small-signal range, or the results will not be accurate. This is because the hbac results do not take into account the large-signal limits like slew-rate or power supply voltage.

The distortion summaries provide a list output, not a plot output. Accordingly, instead of selecting *Results - Direct Plot - Main Form*, select *Results - Print - hbAC Distortion Summary*.

For the compression distortion, you get the actual gain with the distortion of each device by itself divided by the ideal small-signal gain provided by the hbac analysis. In addition, the amplitude of the first three harmonics of the linear output frequency is calculated.

For the IM2 distortion summary, you get the total IM2 output amplitude, and the amplitude that results from having only a single device in the circuit.

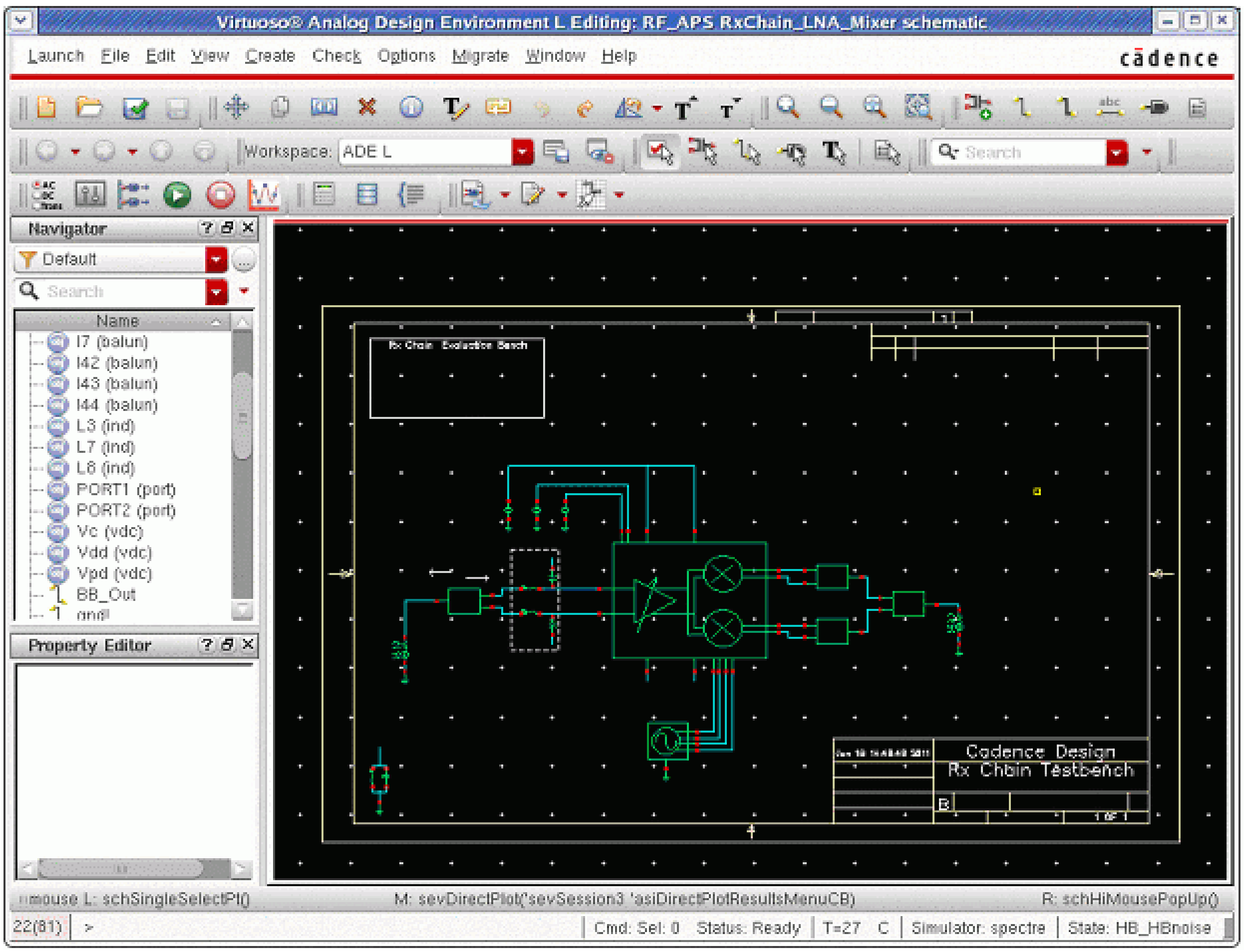
Sampled

Sampled is provided in order to calculate the instantaneous conversion gain at a voltage threshold crossing in the ifft or at specific timepoints in the ifft. This is useful for measuring the conversion gain from the power supply to the oscillator output at a threshold crossing, or at specific times in the LO waveform to the output of a mixer.

ADE Implementation

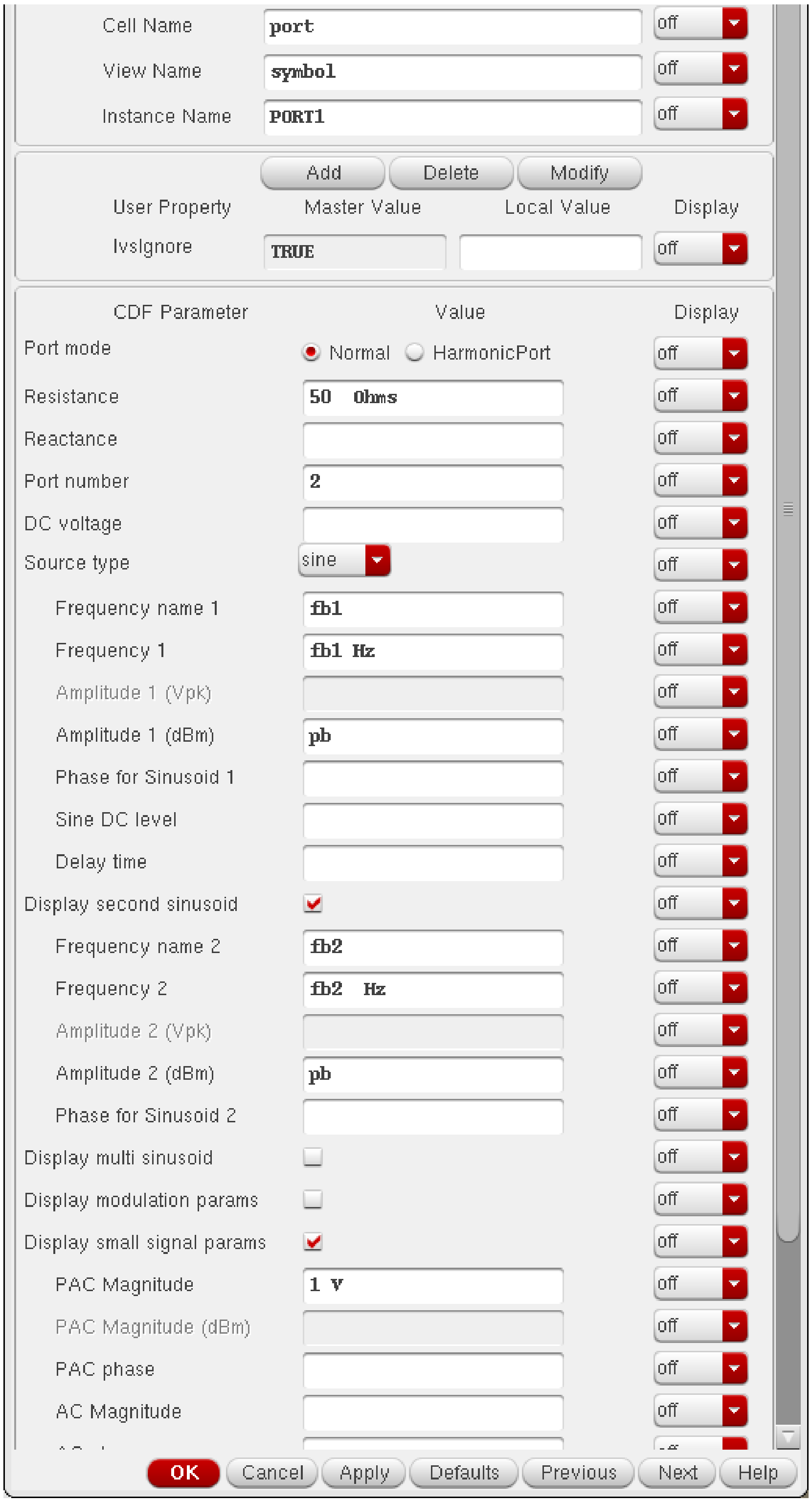
Circuit, Input port Setting, and ADE Setup for all the Examples in This Section

Consider the following circuit:

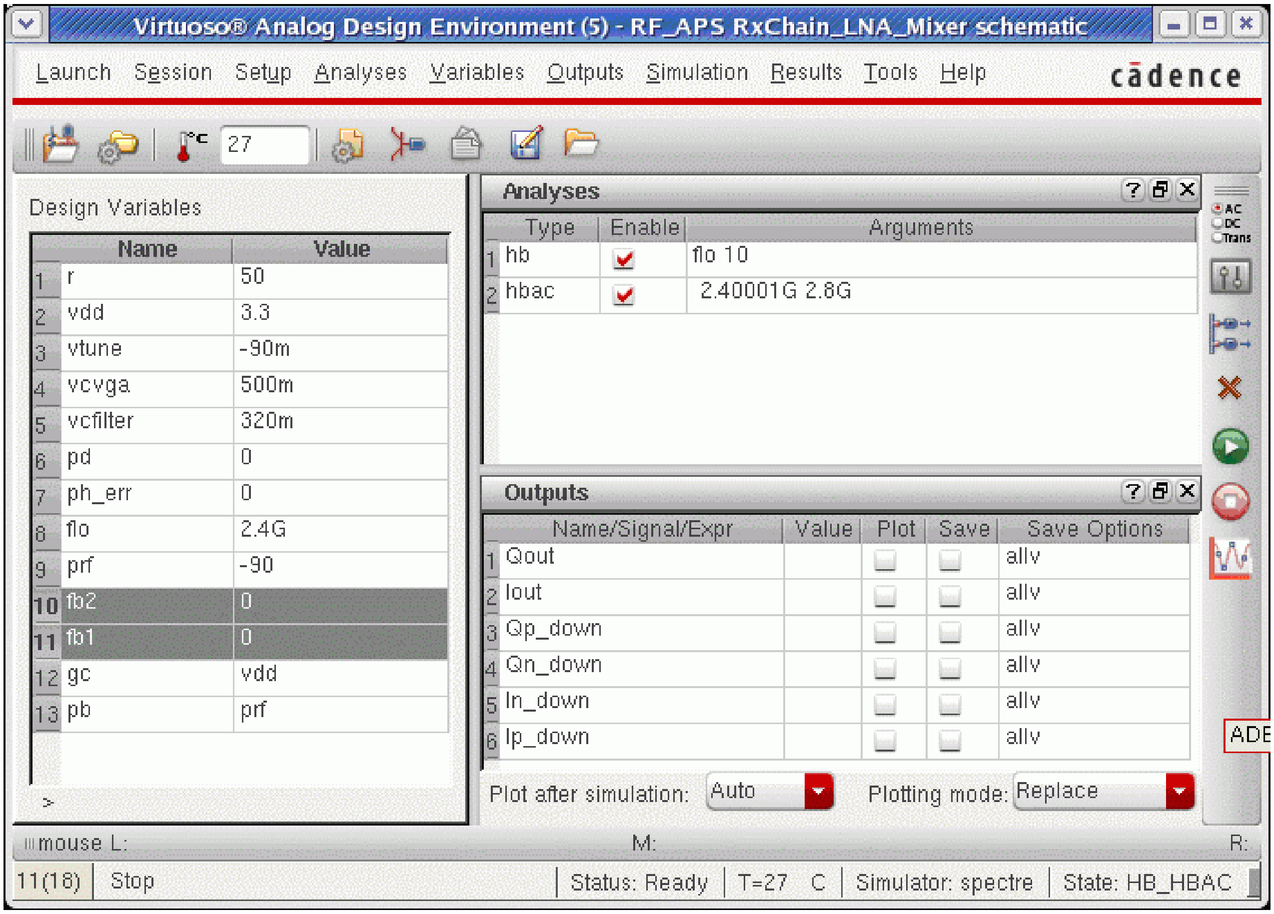


This above circuit is an LNA and a mixer for a 2.4GHz application that is implemented in CMOS. The input source (a port in Cadence terminology) has the values for its frequency and amplitude set to variable names. The PAC magnitude property is set to 1 volt in order to allow easy conversion gain measurements. The reason for the variables is that the frequency and amplitude can be changed in the ADE environment without changing the schematic.

The *Edit Object Properties* form for the input port is shown below.



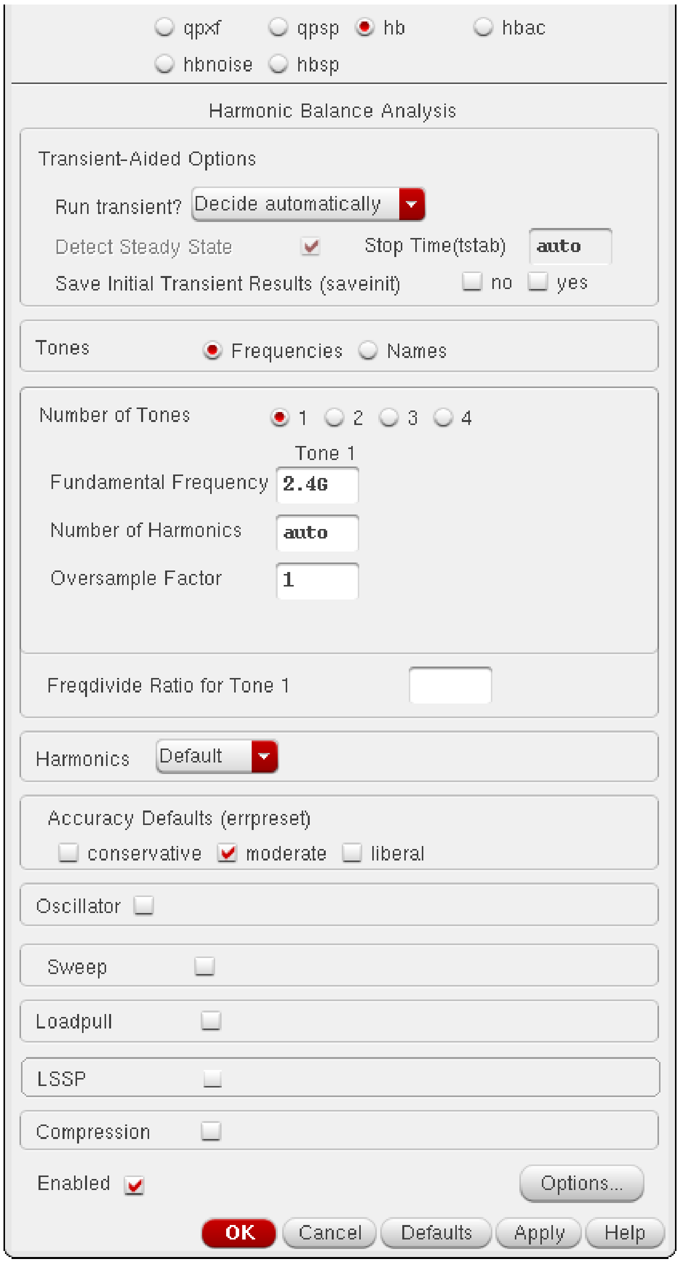
The ADE has the variables set to appropriate values, as shown below.



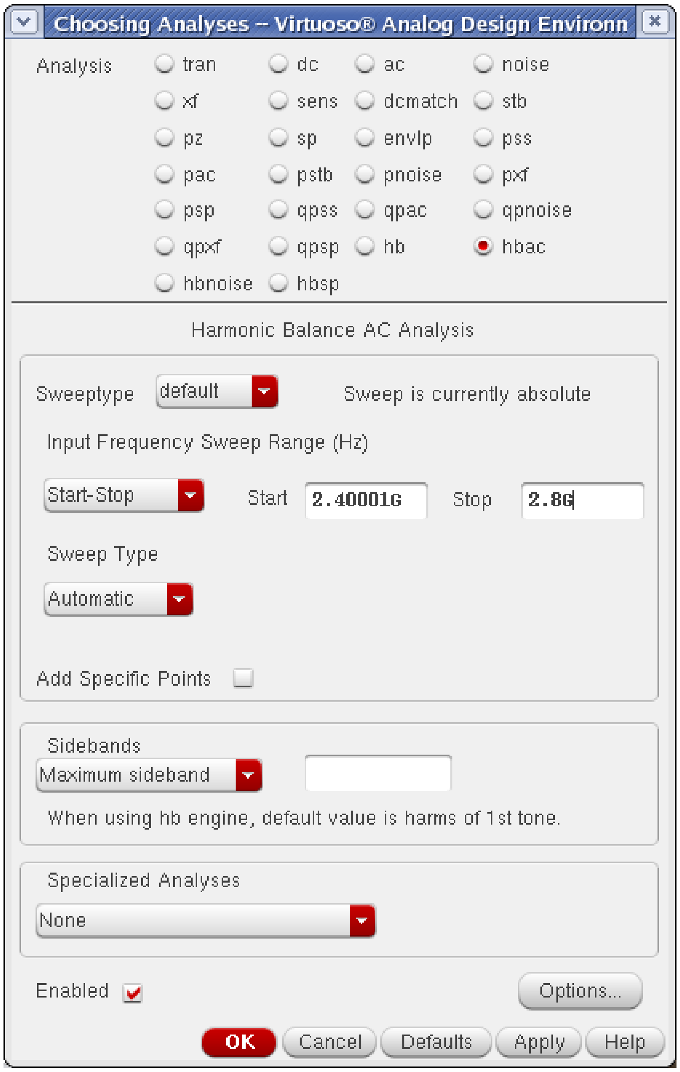
Note that the variables *fb1* and *fb2* (which set the input frequency of the input port) are set to zero, which effectively disables the RF input. Only the LO signal is applied to the circuit.

Using hbAC Analysis for Conversion Gain Measurement

Set up the hb analysis with just the large signal that causes the frequency translation to occur. For more information, see the harmonic balance section.

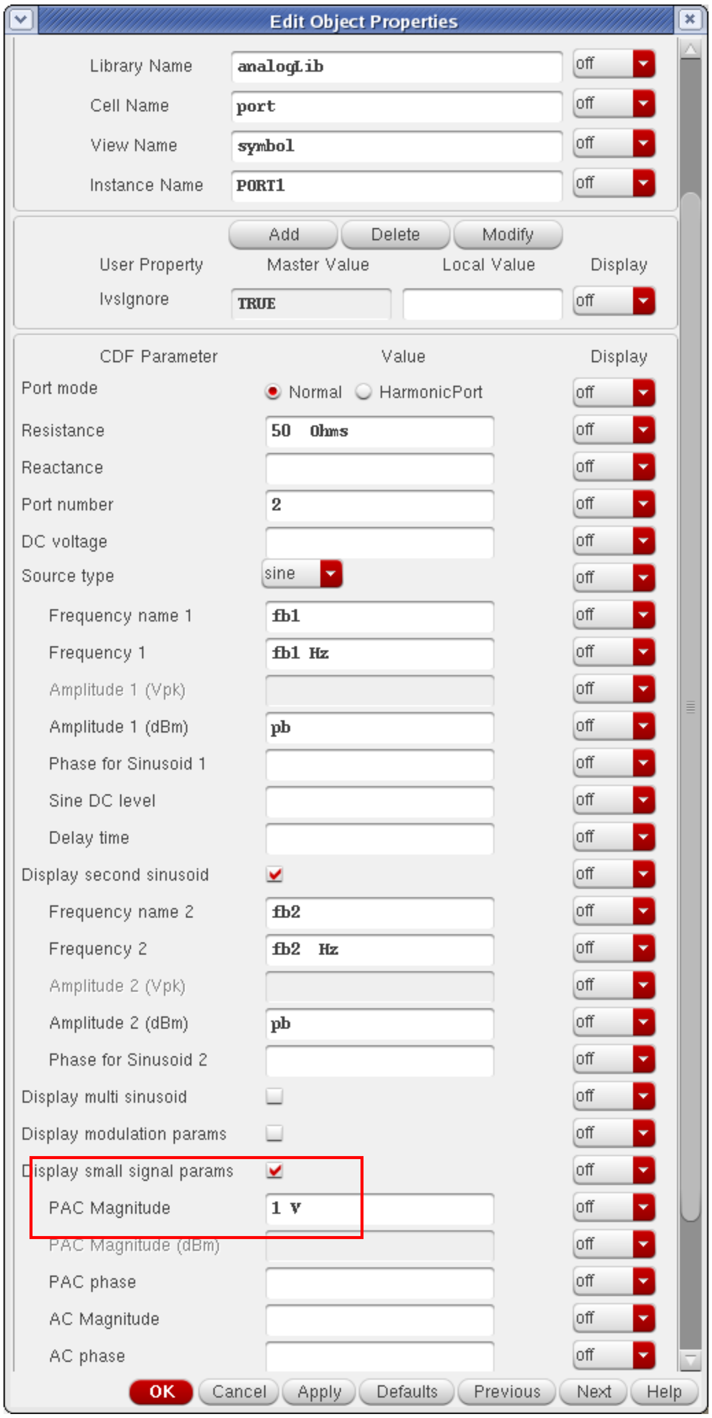


Set up the hbac Choosing Analyses form as follows:



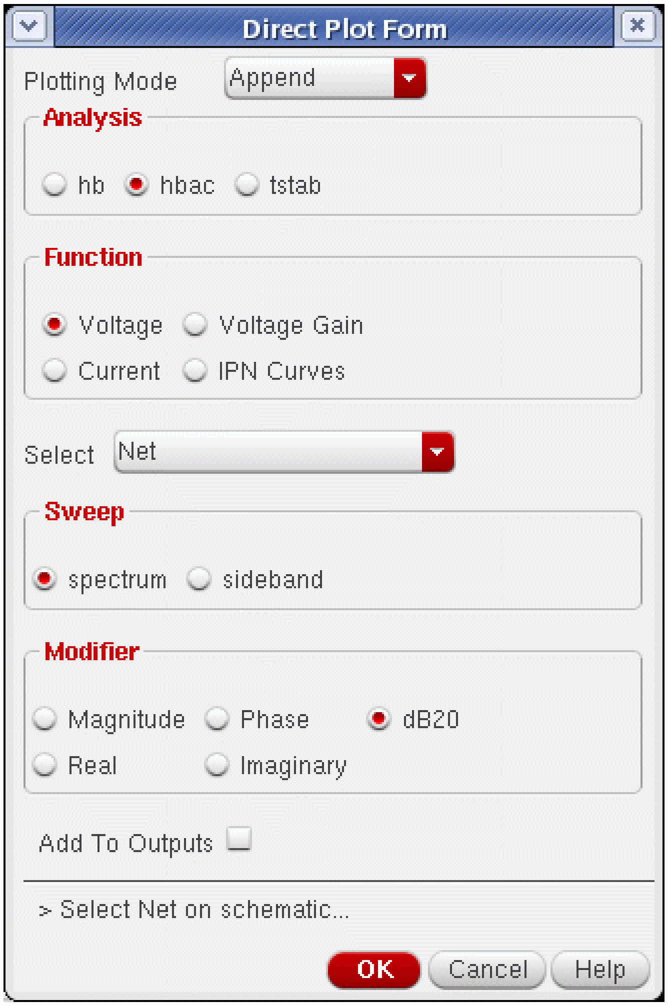
1. Specify the input frequency range.
2. For hbac, or pac where pss is set to the Harmonic Balance engine, the *Maximum Sideband* field should be left blank in which case, all the mixing products are calculated for all the harmonics in the hb analysis.
3. For a normal conversion gain measurement, set the *Specialized Analysis* field to *None*.

For a conversion gain measurement, set the PAC magnitude to 1 volt on the *Edit Object Properties* form for the RF input port as shown earlier. This is always volts peak.



4. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form*.

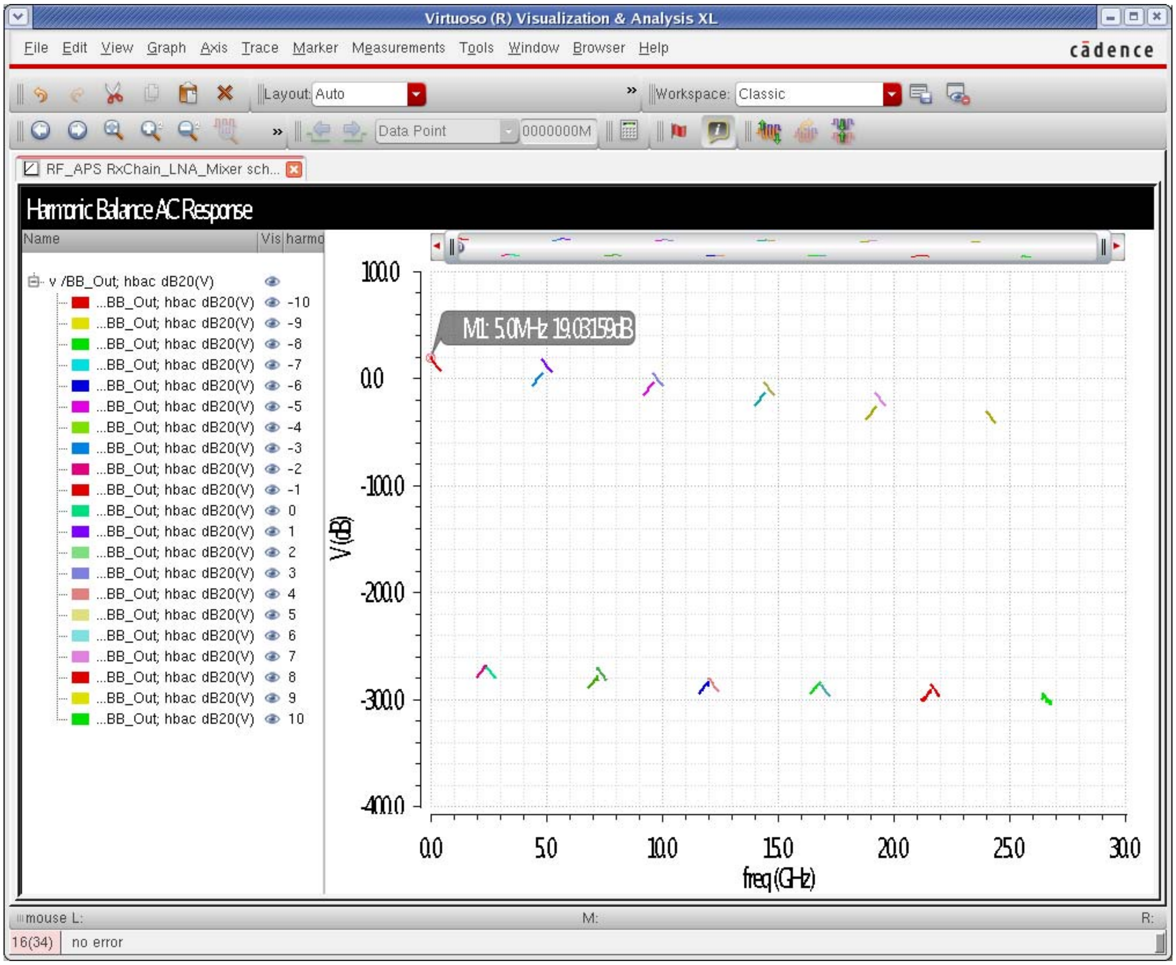
The *Direct Plot Form* is displayed.



The harmonic balance analysis characterizes the nonlinearity of the system as a series of harmonics. Hbac injects an input frequency, and because the system nonlinearity has been characterized by the harmonic balance simulation, the input tone mixes with the harmonics of the LO to produce outputs at multiple frequencies. These mixing products are calculated for mixing the hbAC input frequency range with the hb harmonics. Because the amplitude of the RF input signal is assumed to be small, the harmonics of the hbac input and harmonics of the output mixing products are not calculated.

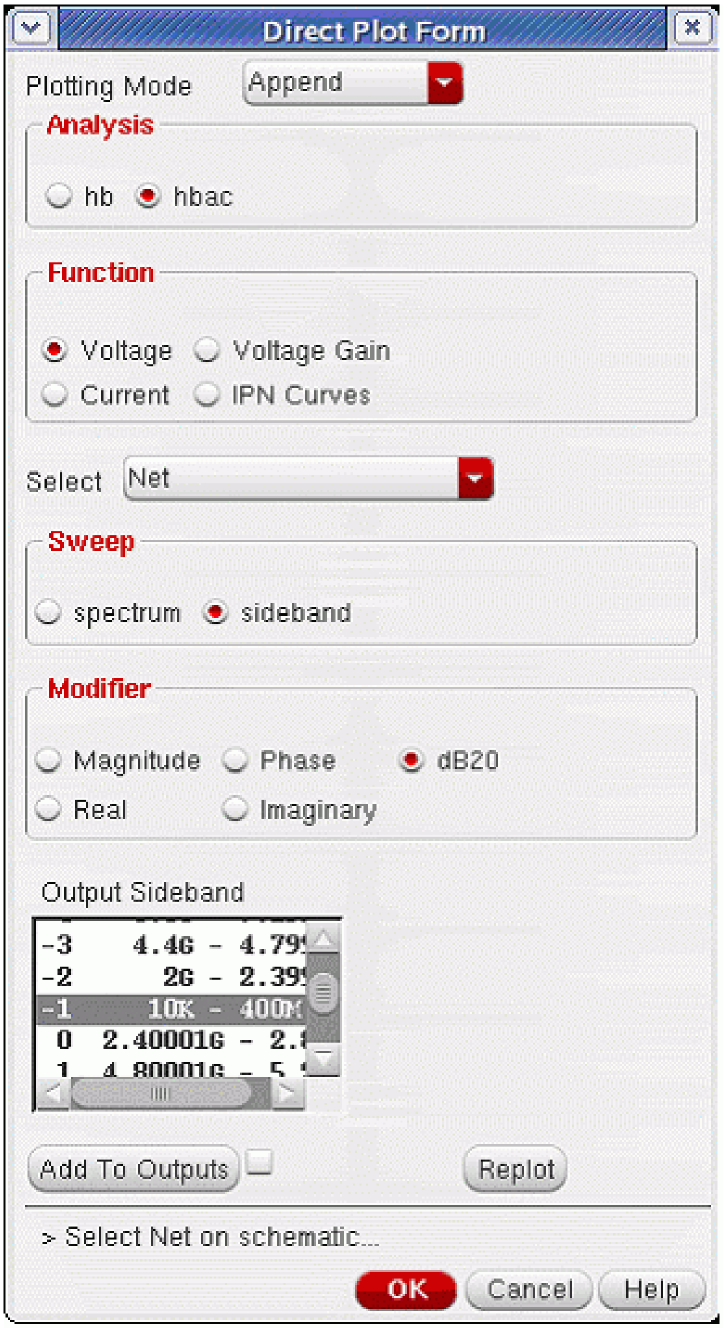
To measure the outputs you desire, place a marker at the desired output frequency of interest and read the conversion gain, as shown below.

The low frequency conversion gain is just over 19dB.

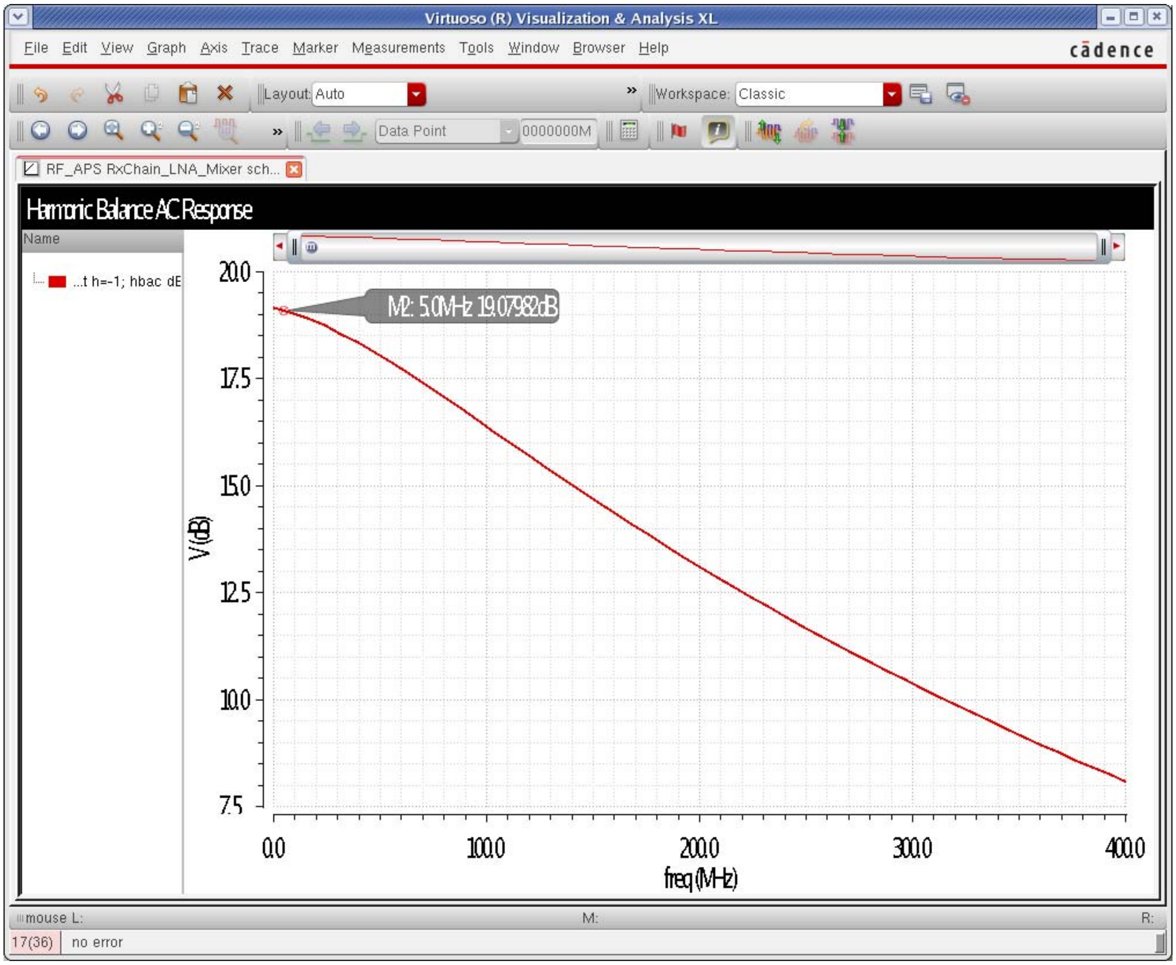


In the *Direct Plot Form*:

1. To plot just the desired output mixing product, select *sideband* instead of *spectrum*, and select the desired output mixing product form the list, as shown below.



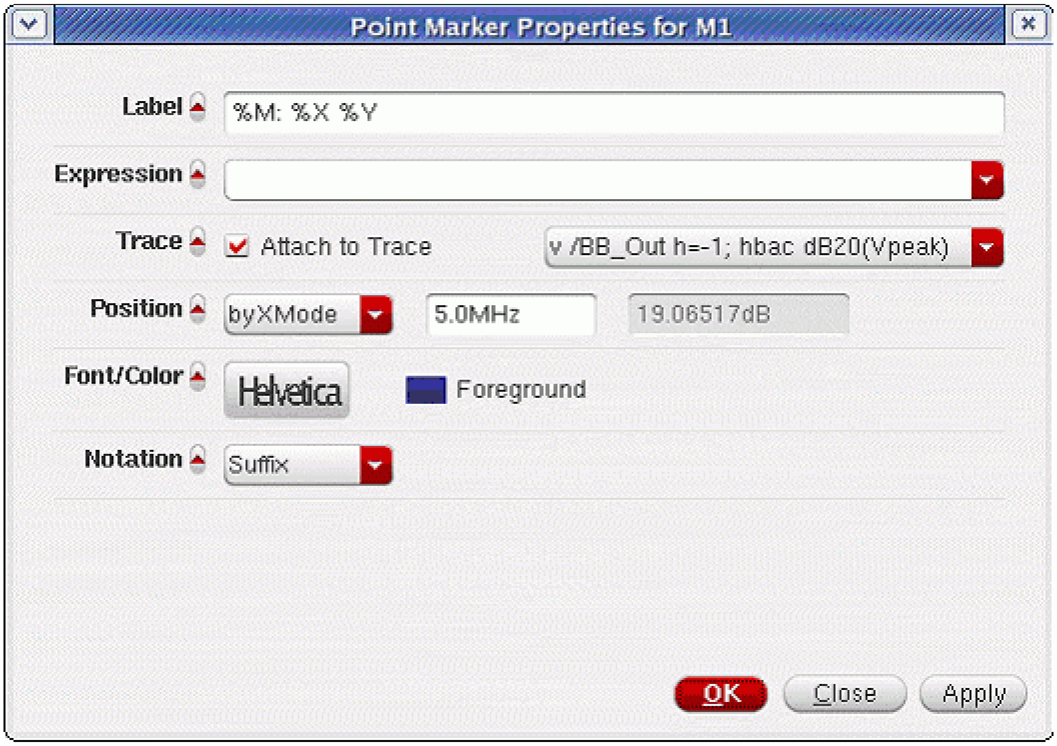
2. Now select the output node, or click the *Replot* button in the *Direct Plot Form*. The waveform tool displays the output, as shown below.



3. Position a marker by moving the cursor to the desired output frequency and type *m*. The marker appears, as shown in the figure above.

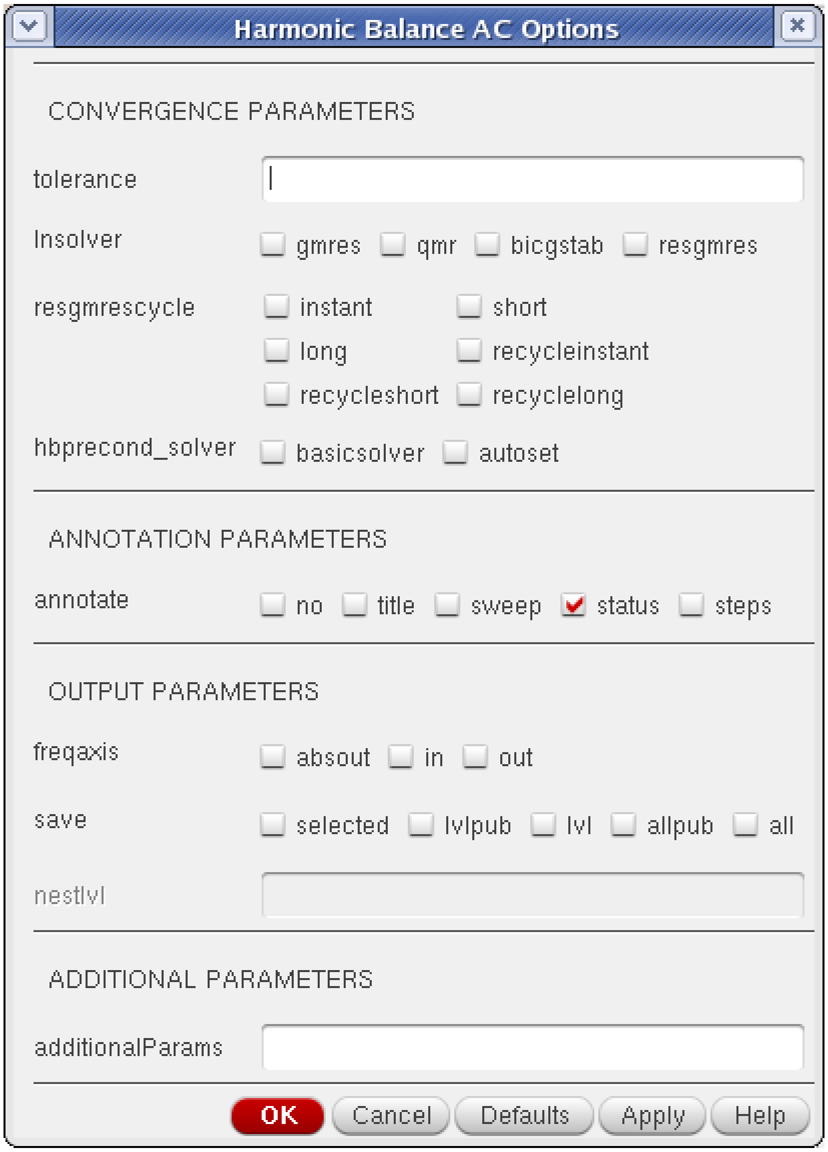
4. To position the marker at an exact frequency, place it as above, and then move your cursor over the intercept point on the trace. Click and hold the right mouse button, move

to *Marker Properties*, and release the mouse button. The *Point Marker Properties* window appears, as shown below.

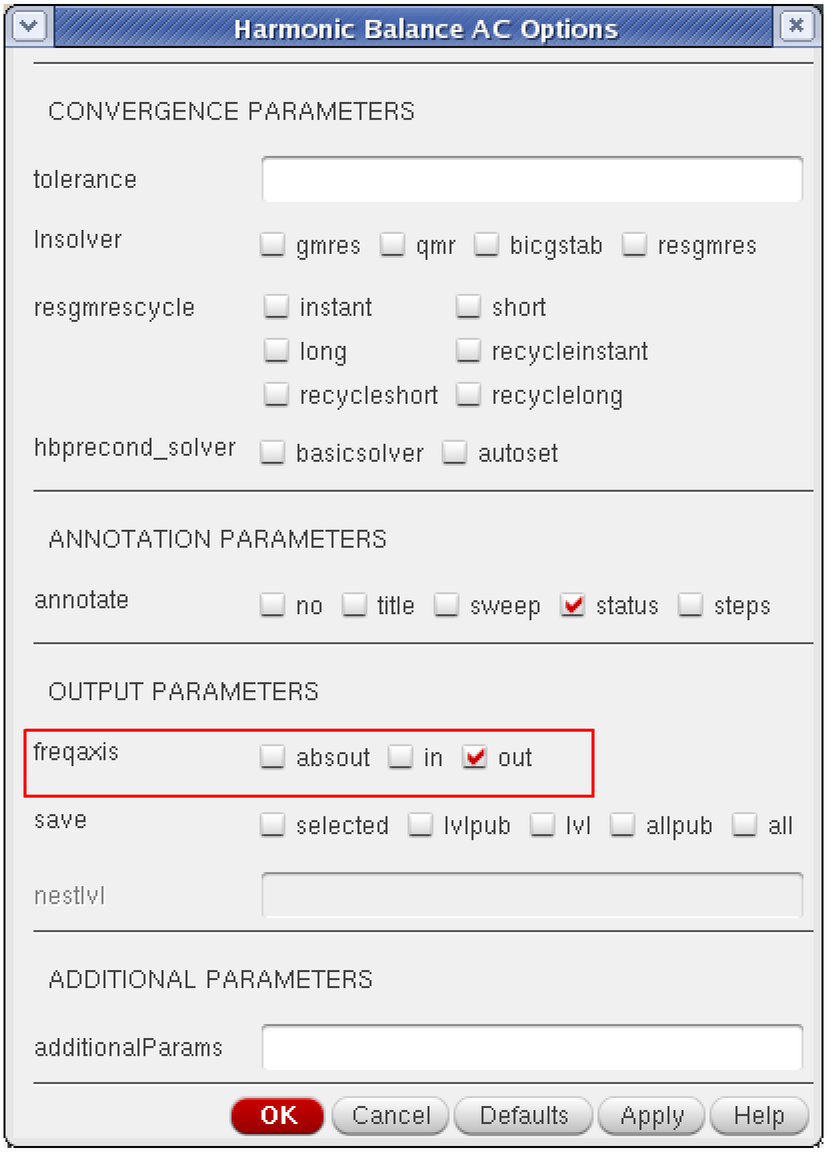


5. Type the frequency in the *Position* field, and click *OK* or *Apply*.

Commonly Used HBAC Options

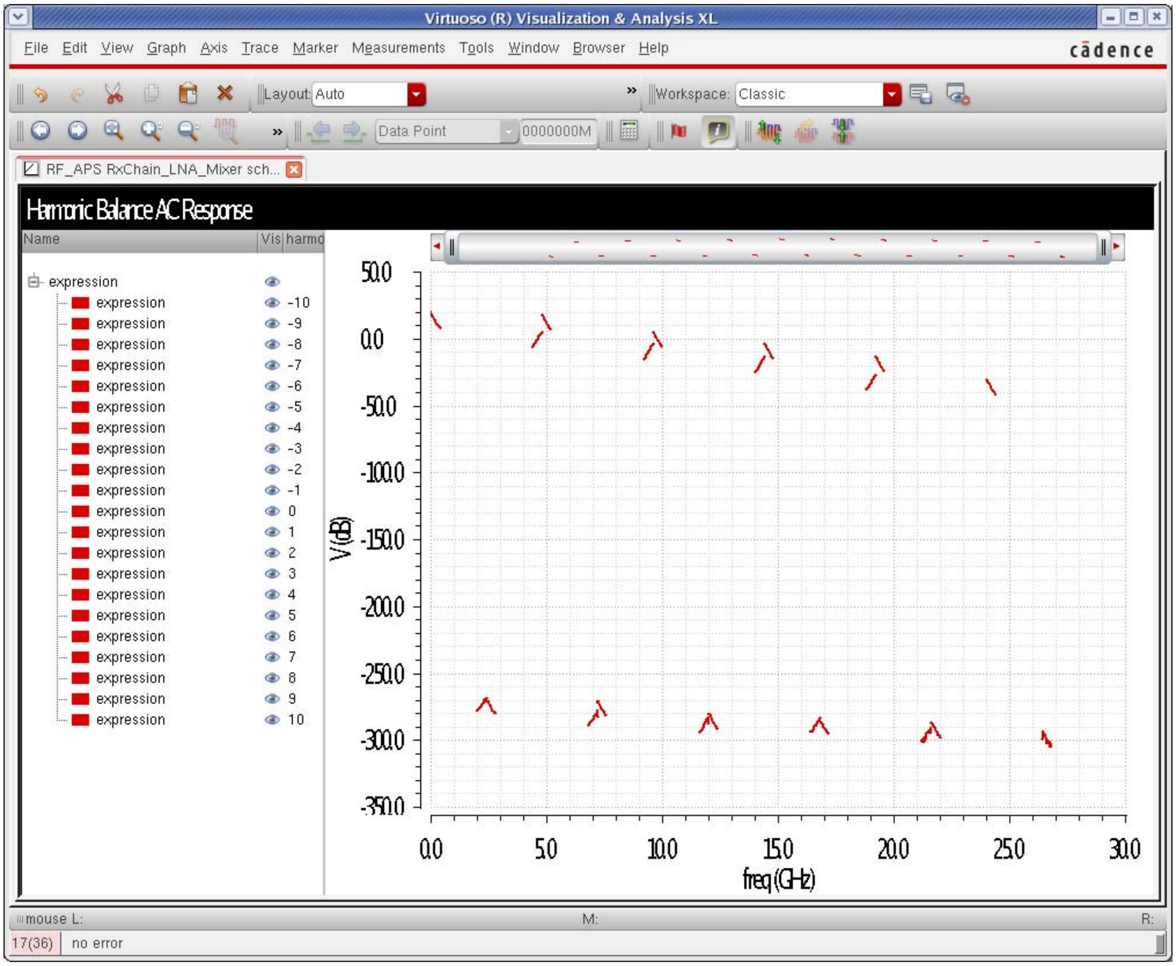


Freqaxis

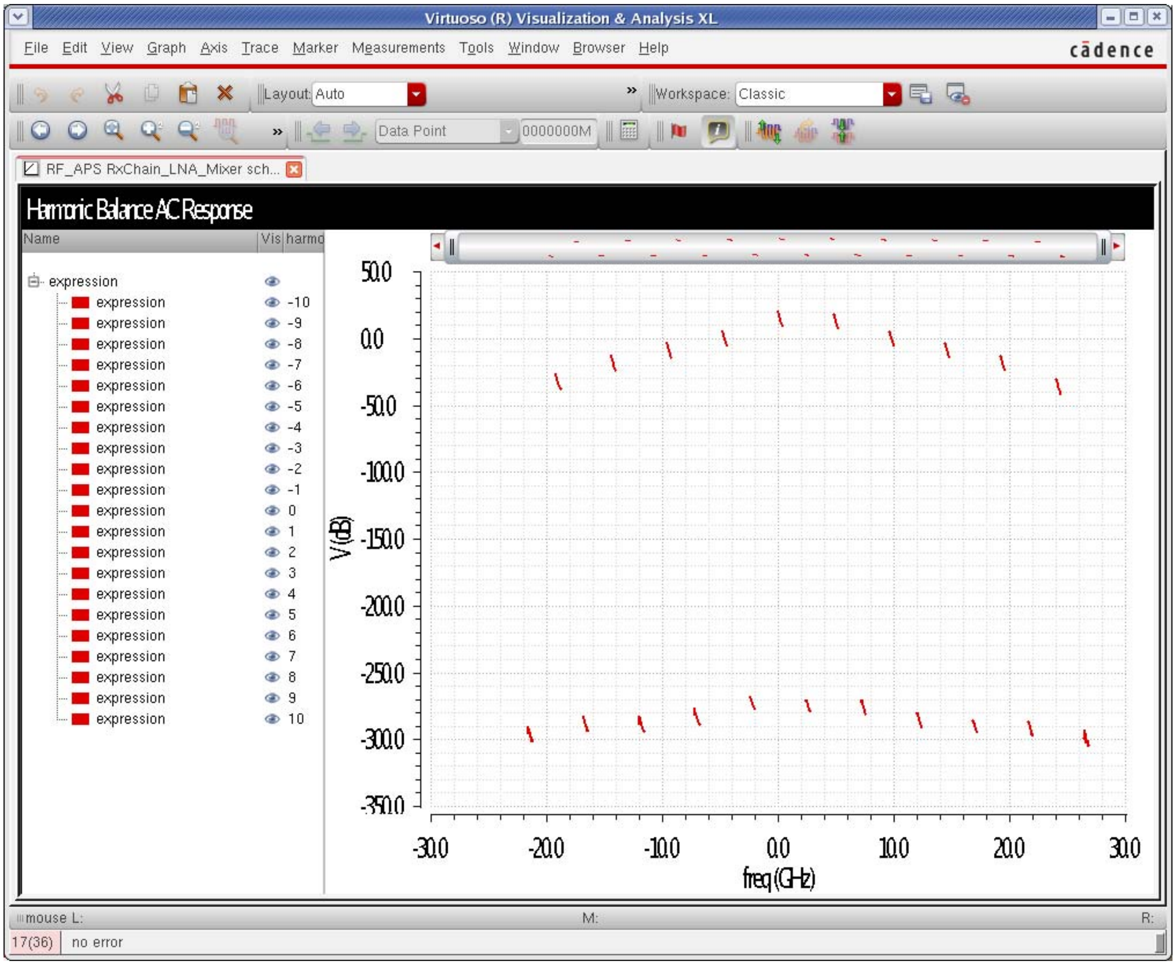


freqaxis specifies whether you want to see the negative frequency axis or not.

The analysis calculates the output frequencies and amplitudes, so *out* and *absout* are reasonable choices. *out* displays the negative frequency axis. *absout* (absolute value of the output) displays positive output frequencies. This is the default. Below is the default of *absout*.



Next is the output with freqaxis set to *out*. This is the same data as the above plot except the negative output frequencies have not been reflected up to the positive frequencies.



If you select *in*, all the outputs at the different frequencies are plotted on the same input frequency range scale. This is not recommended. The default is *absout*.

Tolerance

Leave this option at the default value.

Hbac uses an iterative solver to calculate the output amplitudes. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough, and the tolerance option specifies that accuracy for hbac. For driven circuits, the default is 1e-6. For oscillators, the default is 1e-4.

Lnsolver

Leave this option at the default. Each hbac sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases at the next iteration. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence. Their use is not suggested.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

gmres is the Generalized Minimum RESidual method.

qmr is the Quasi-Minimal Residual method.

bicgstab is the STABilized BI-Conjugate Gradient method.

resgmres is the REStarted Generalized Minimal RESidual method.

Resgmrescycle

Leave this option at the default. This option is used when resgmres is set for the linear solver. For the resgmres linear solver, there are six different options.

Hbprecond_solver

This option has two settings; basicsolver and autoset. Basicsolver is the only solver available in standard Spectre. Autoset is the default solver when APS is used. This solver is faster. If stagnation is detected, Spectre prints a message in the Spectre output window.

Annotate

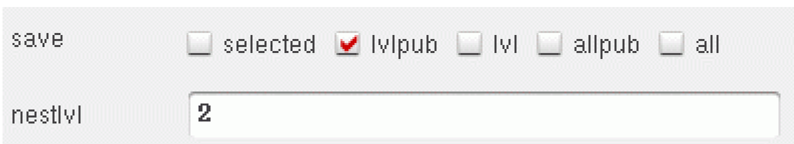
This option controls the amount of output in the spectre output window. Selecting title produces the least amount of detail, and selecting steps gives the most amount of detail. the default is status.

Save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished using the *Outputs - To Be Saved - Select On Schematic* and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.

Nestlvl

If *save* is set to *lvl* or *lvlpub*, this option controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer. The example below shows *lvlpub* selected that saves two levels of hierarchy.

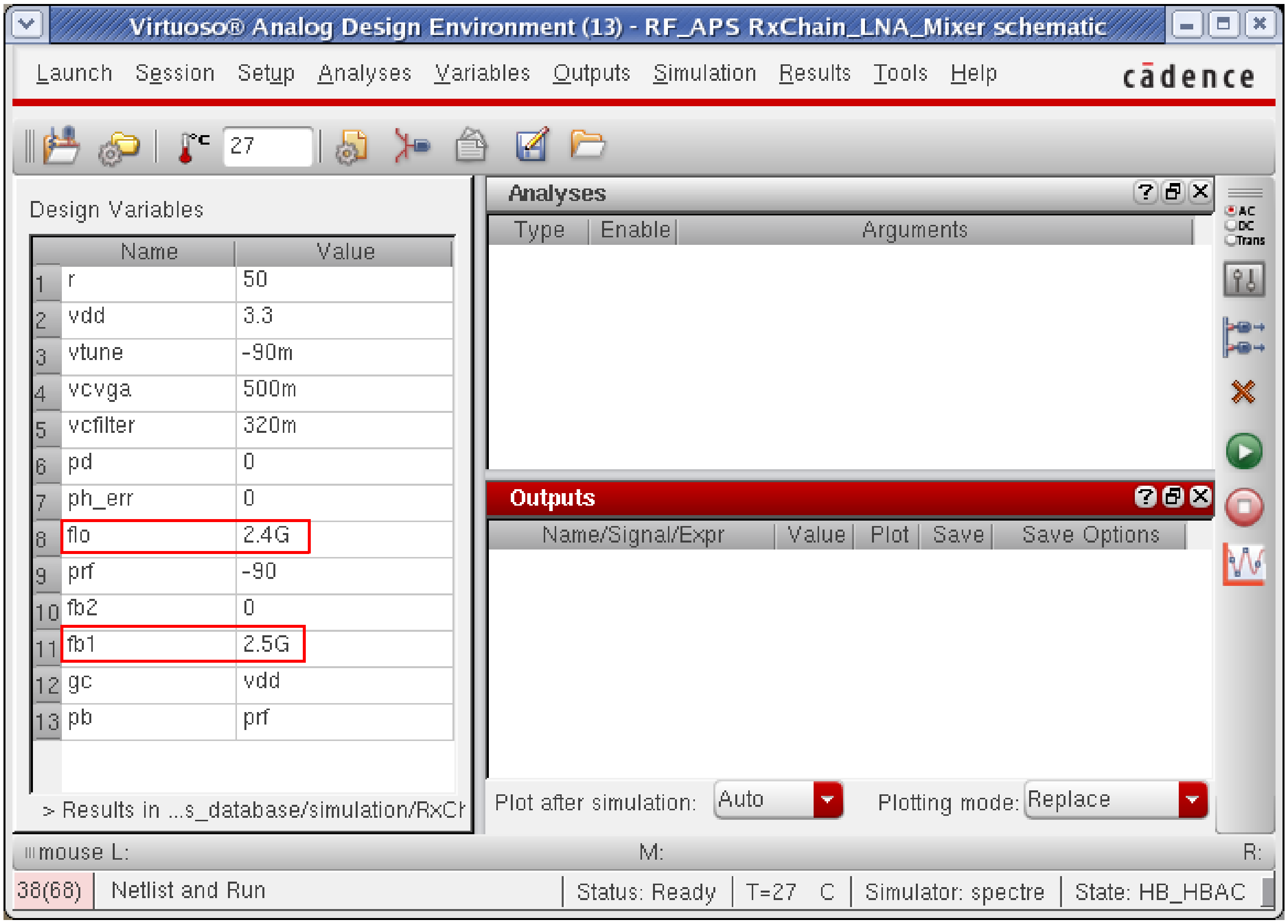


AdditionalParams

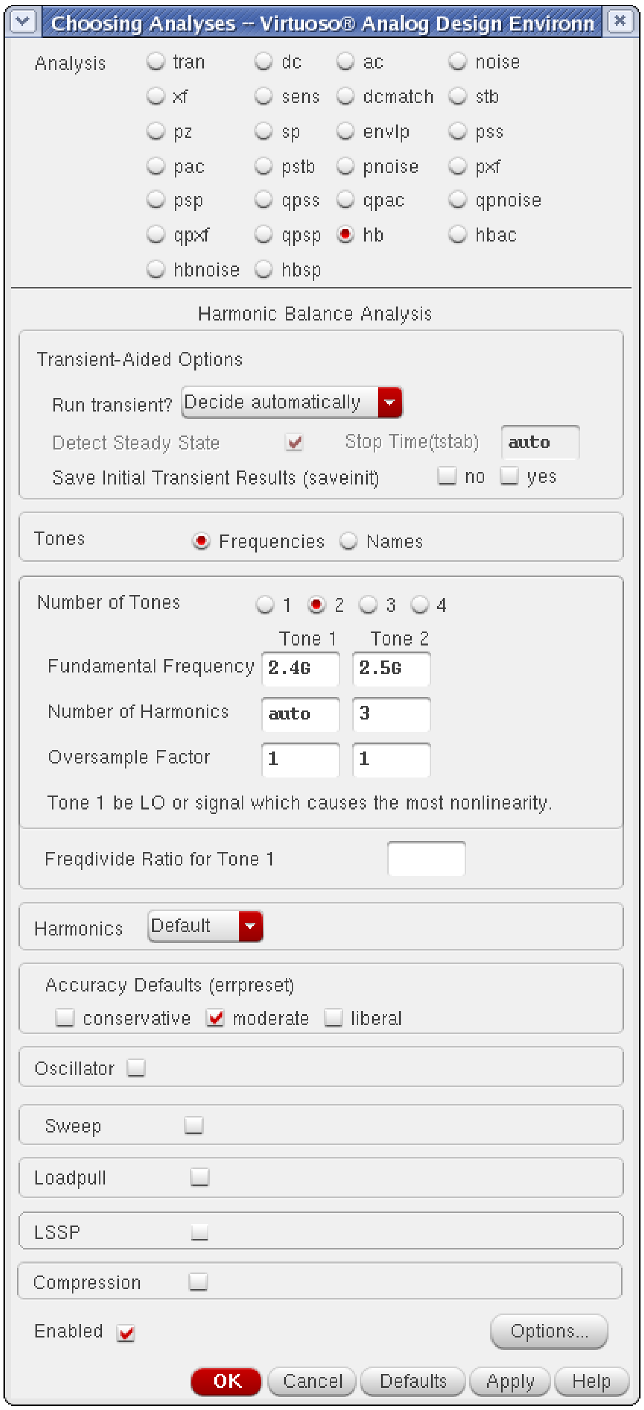
additionalParams is typically used for new features that are being beta tested early in the release cycle where there is not a gui selection available yet.

hbAC With Multiple hb Inputs

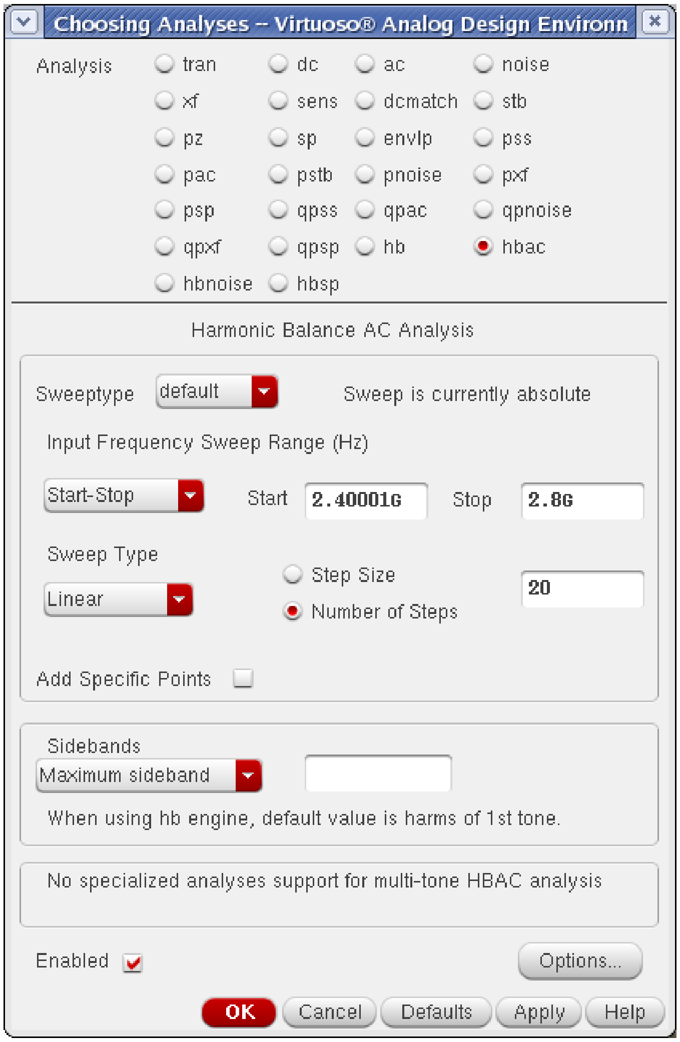
Harmonic balance is available in the hb form and the pss and qpss forms when harmonic balance is selected for the engine. Hbac is available in the hbac form and in the pac and qpac forms when the engine is set to harmonic balance in pss and qpss. In the example below, hb and hbac were used. The LO is set to 2.4GHz as before. To set up the second (RF) input for the harmonic balance analysis, the variable `fb1` was set to `2.5G` in the ADE window, as shown below. This variable sets the frequency of the first source in the RF port component. In this case, the LO is at 2.4GHz and the RF is at 2.5GHz.



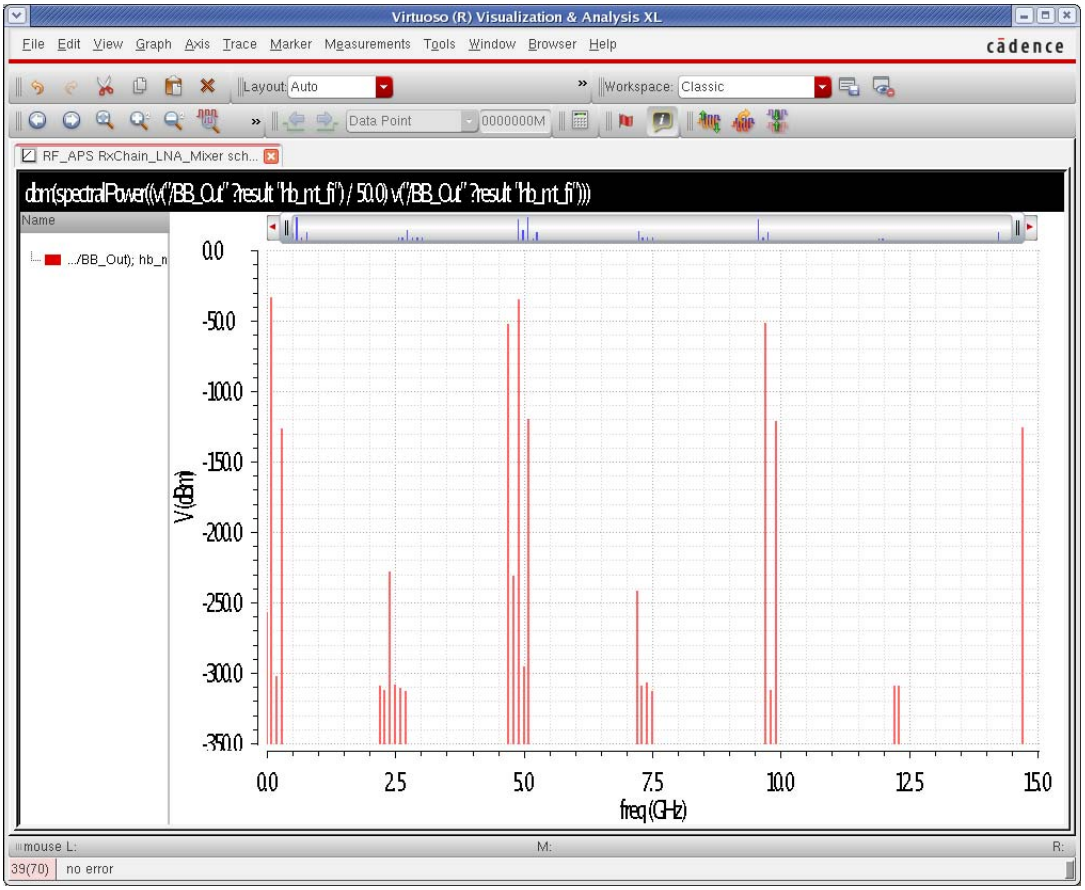
The harmonic balance analysis was set up to calculate the LO harmonics automatically, and three RF harmonics. For more information on Harmonic balance, please see the Harmonic Balance section at the beginning of this chapter.



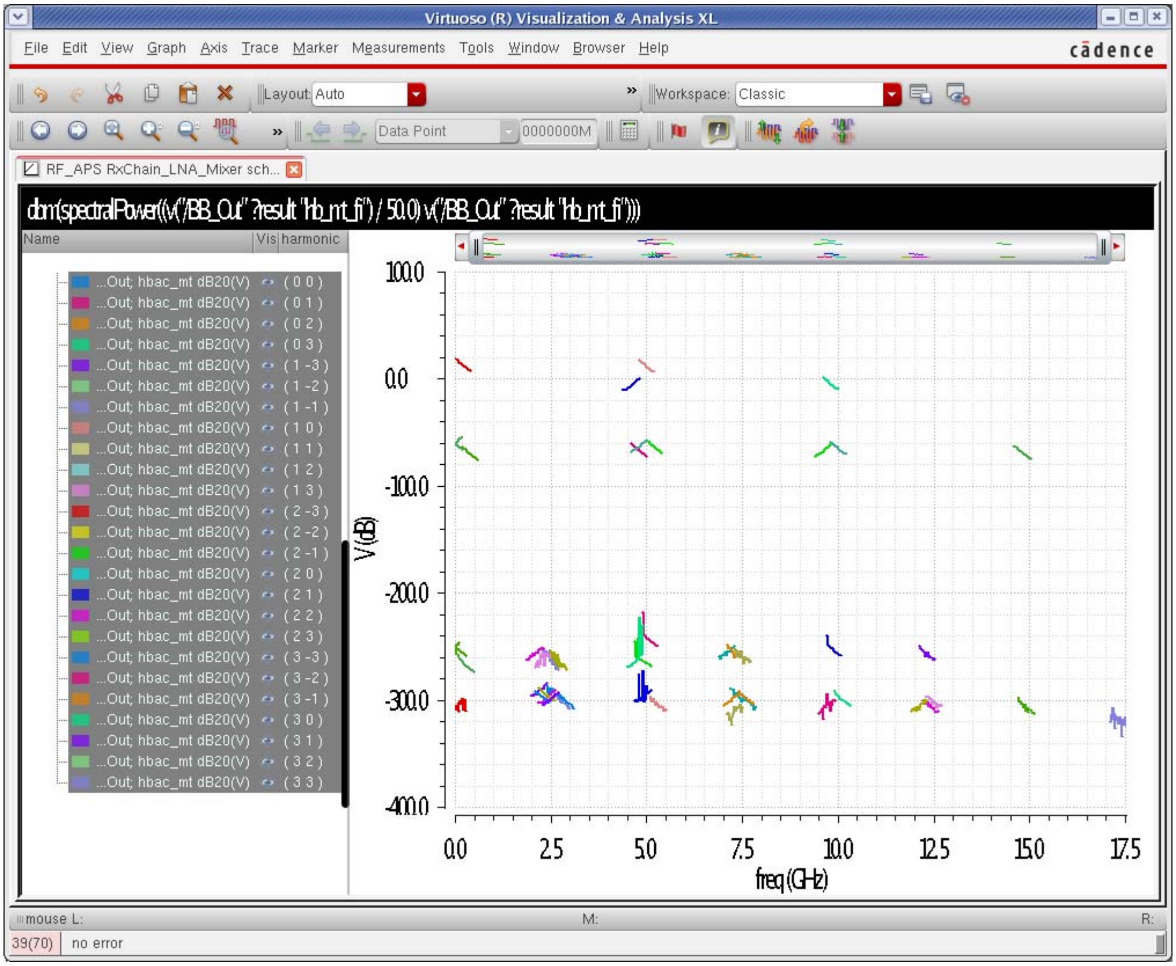
The hbac form is set up exactly the same as the hbac form was for the first example in the ADE implementation section.



Now the analysis was run. The harmonic balance output plot below shows that because there were more input frequencies, many more harmonics are produced by the circuit, which is as expected.

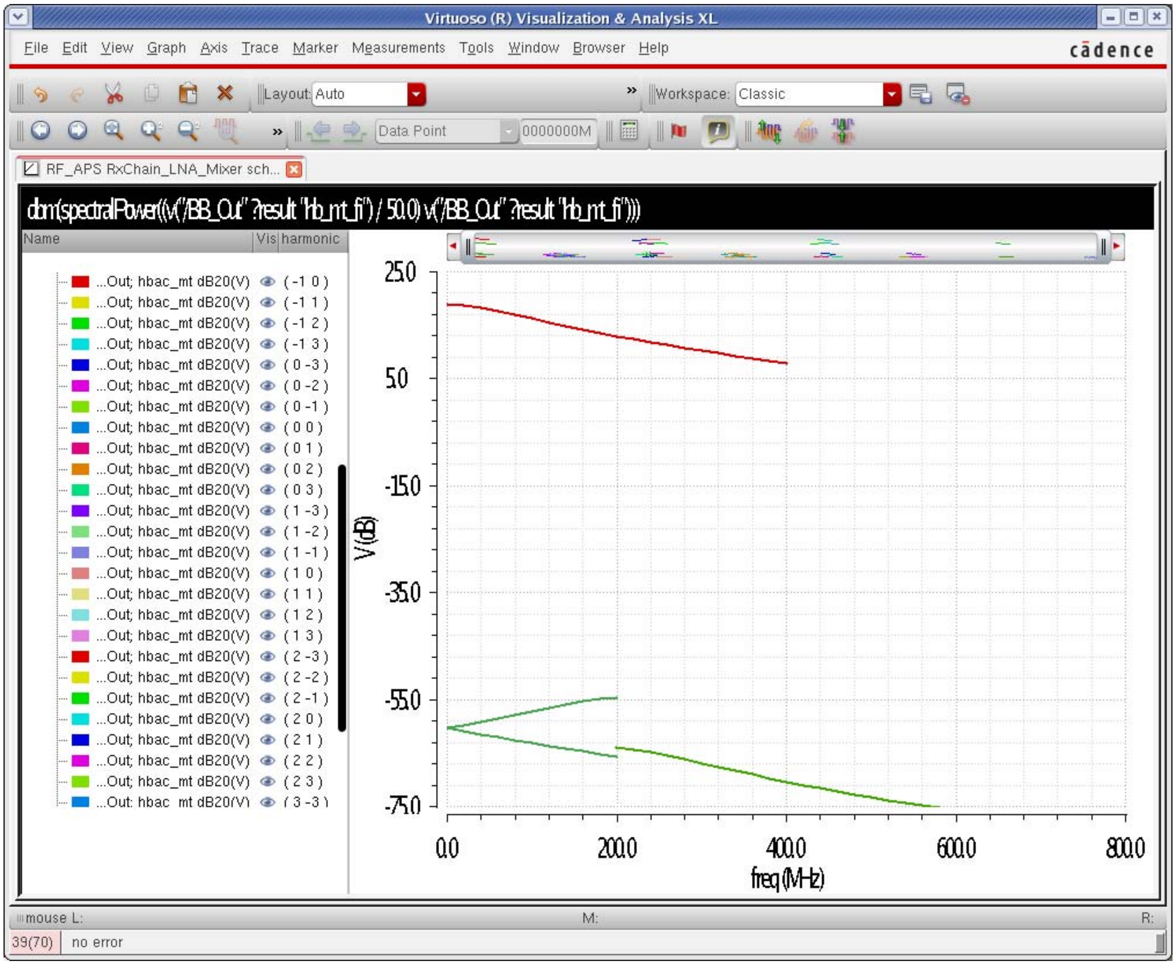


Because there are many more harmonics to mix with, many more mixing products are produced by hbac, as shown below.



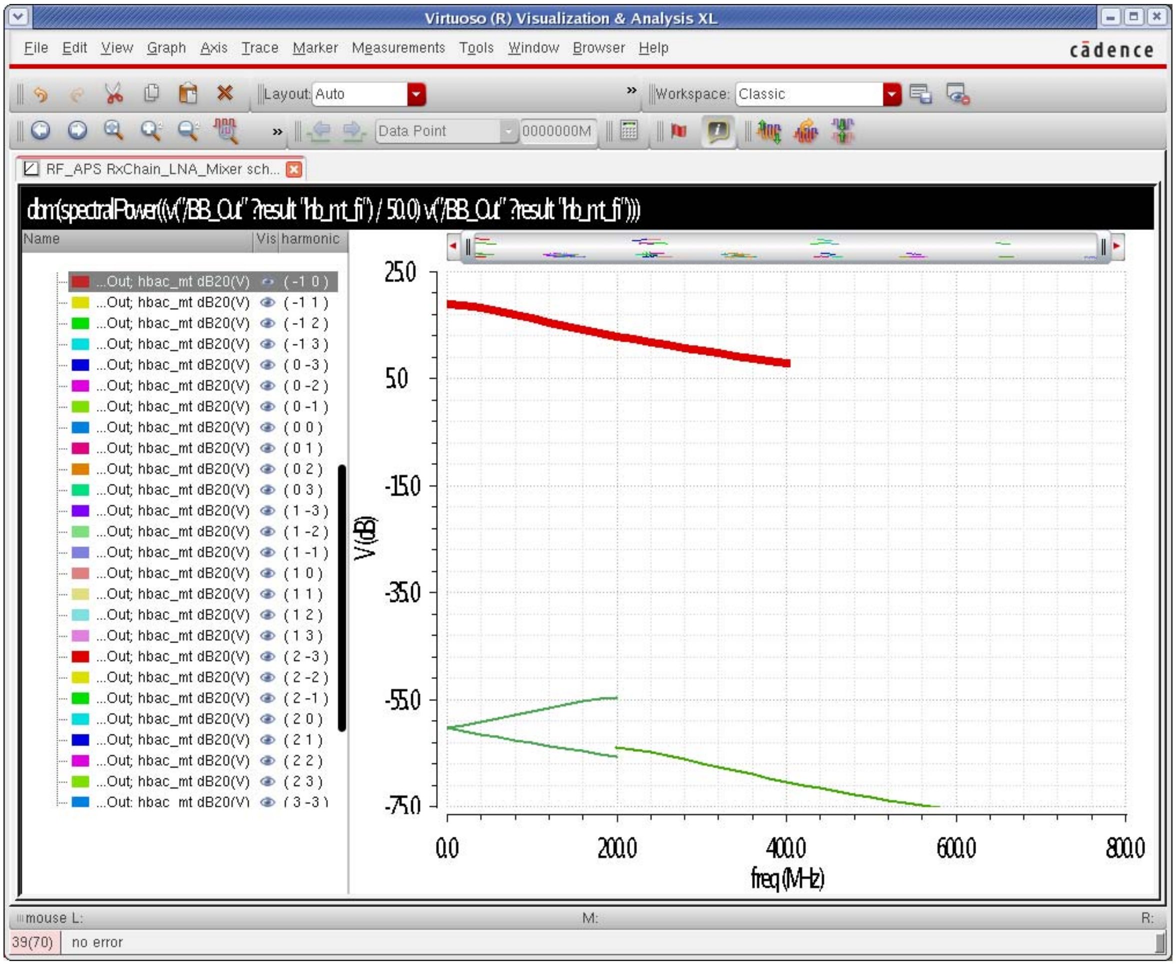
The mixing products below -200dB are at the numerical noise floor of the simulation.

The next figure shows a zoomed in area near zero frequency. Only the mixing products larger than -75 dB are shown.



If the output frequency were 50MHz, note that there are three different values. Which one is correct? Actually, they all are. They are produced by mixing with different harmonics from the large-signal analysis.

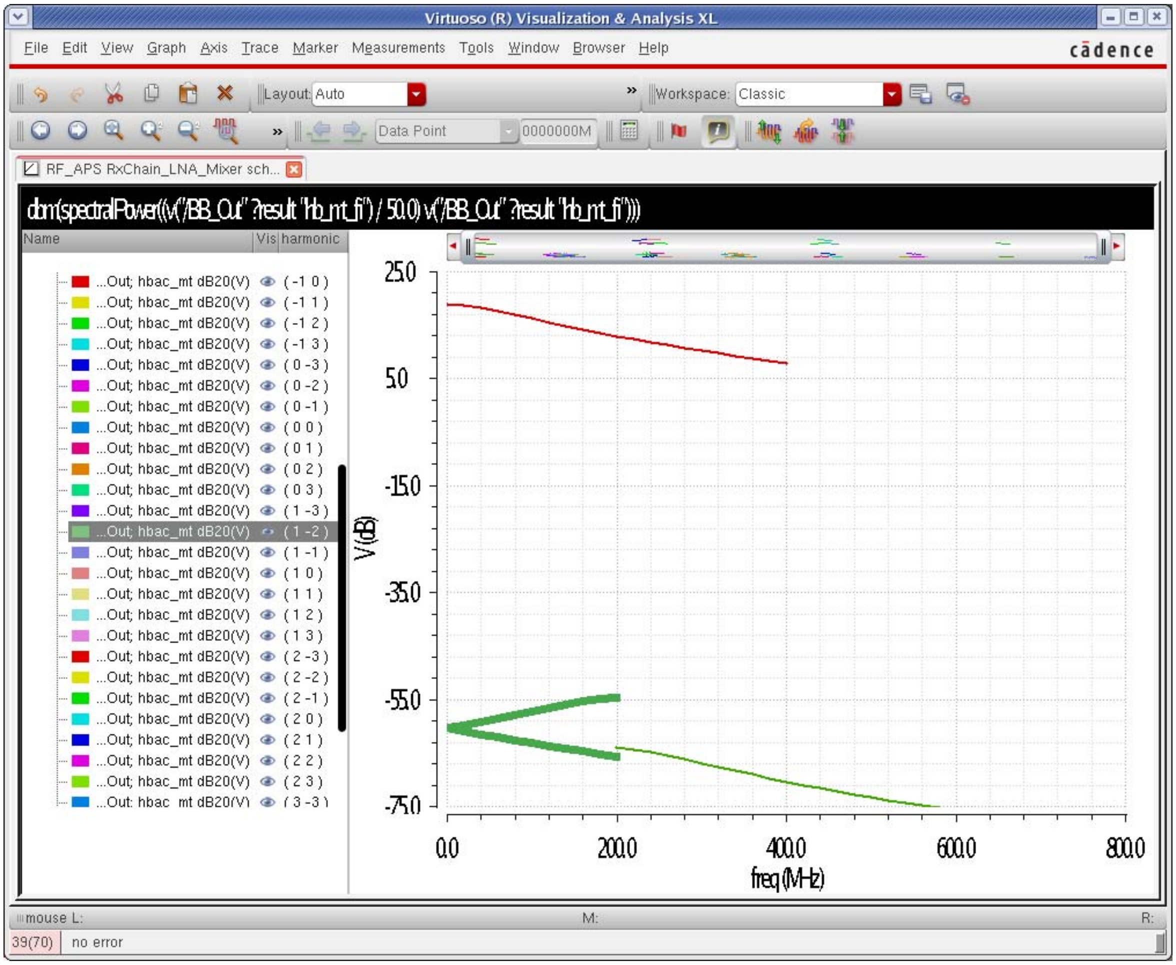
To identify which harmonics are being mixed with, select the top trace. On the left side of the display an entry is highlighted. The example below shows the top trace selected. (It is bold. It is difficult to see in the figure below, but in the waveform tool it is much easier.)



On the right side of the legend, it reads (-1 0) The first number is the harmonic number of the tone with tstab set, or is listed as *Tone1* in the hb *Choosing Analyses* form. This is LO in this example. The second number is the RF input.

-1 0 means that the input frequency is mixed down with the first harmonic of the LO, and not mixed with the RF. The input frequency range in the hbac (qpac) *Choosing Analyses* form is 2.40001G to 2.8G. If you take 2.40001G as a second small-signal input (that is what hbac does) and mix it down with the first harmonic of the LO at 2.4G, you get very near zero. When you mix 2.8G down with 2.4G, you get 400M. So this component starts at near zero, and sweeps up to 400M as the input sweeps from 2.40001G to 2.8G.

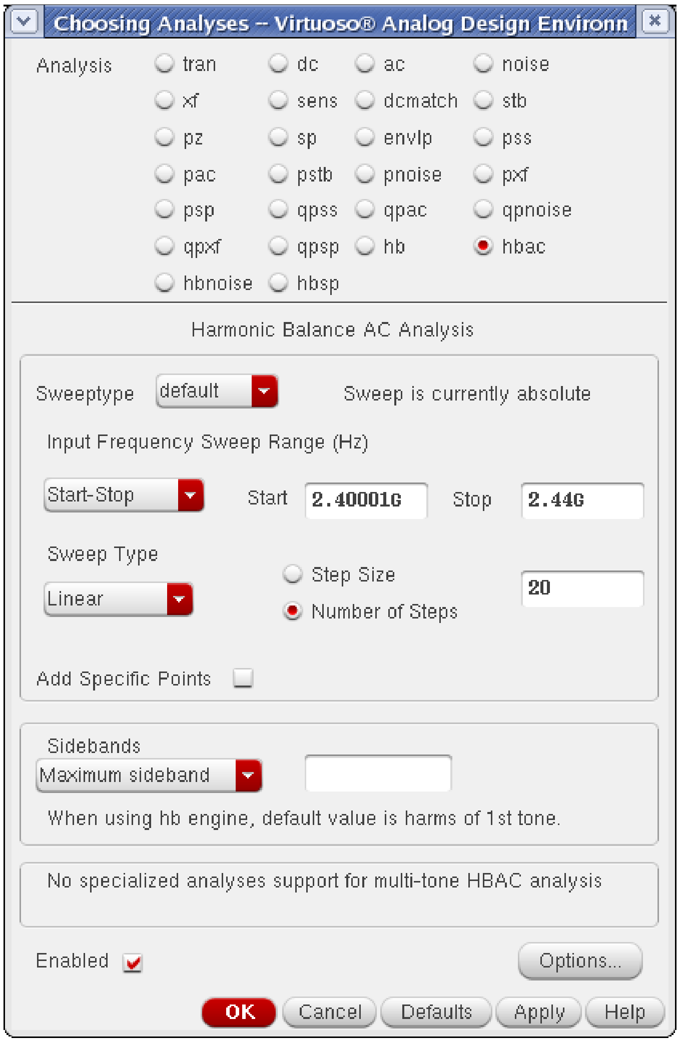
Now select the second trace down, as shown below.

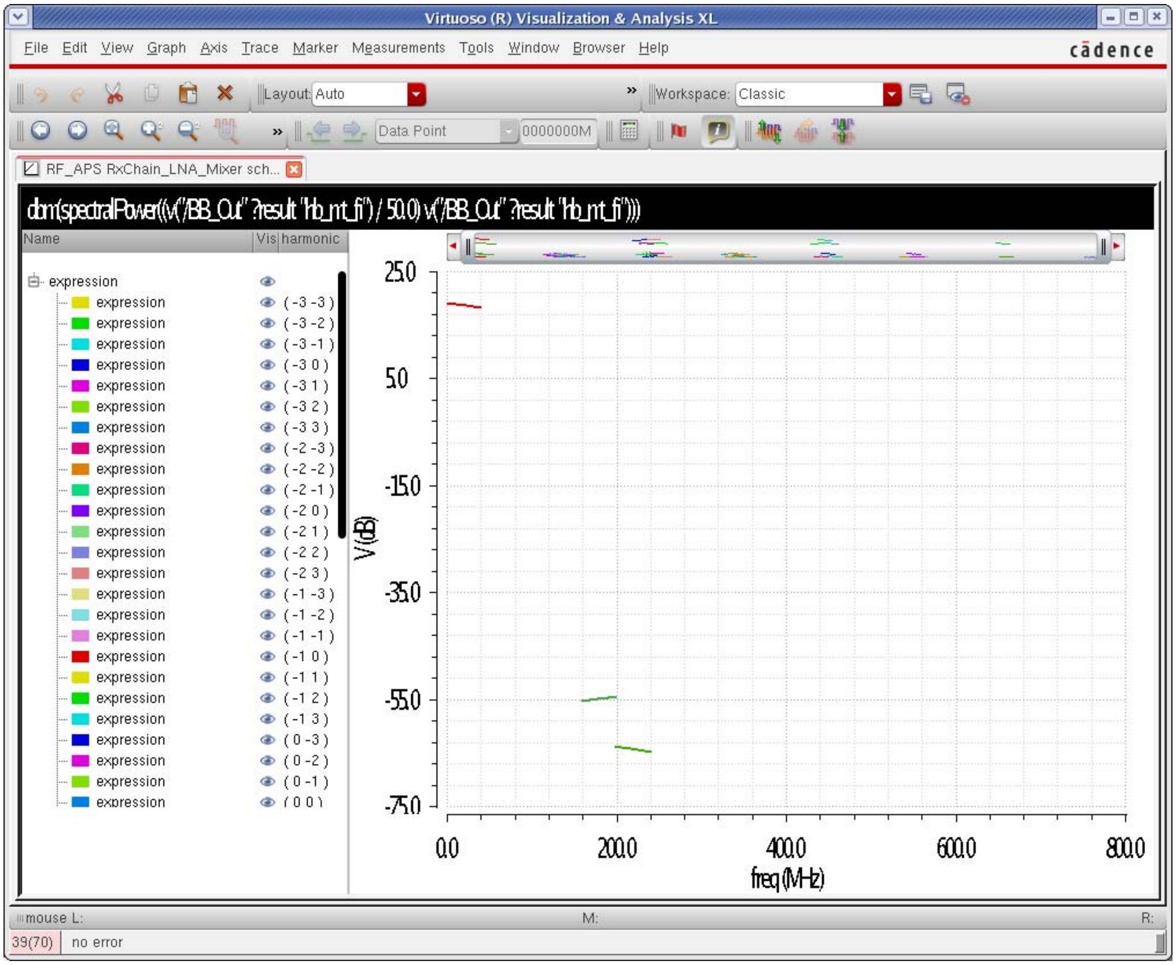


The legend reads 1 -2.

2.40001G mixes up with the first harmonic of the LO, and then mixes down with the second harmonic of the RF. So, 2.40001G + 2.4G - 2*2.5G is very near -200M. As the input sweeps up in frequency, the output becomes less negative, passes through zero, and then sweeps up to +200M. Because the negative frequencies are reflected to the positive frequency domain, the resultant response looks like a less than sign (<) in the above plot.

Note that the ambiguity goes away when the frequency sweep on the input source is restricted to a smaller range.



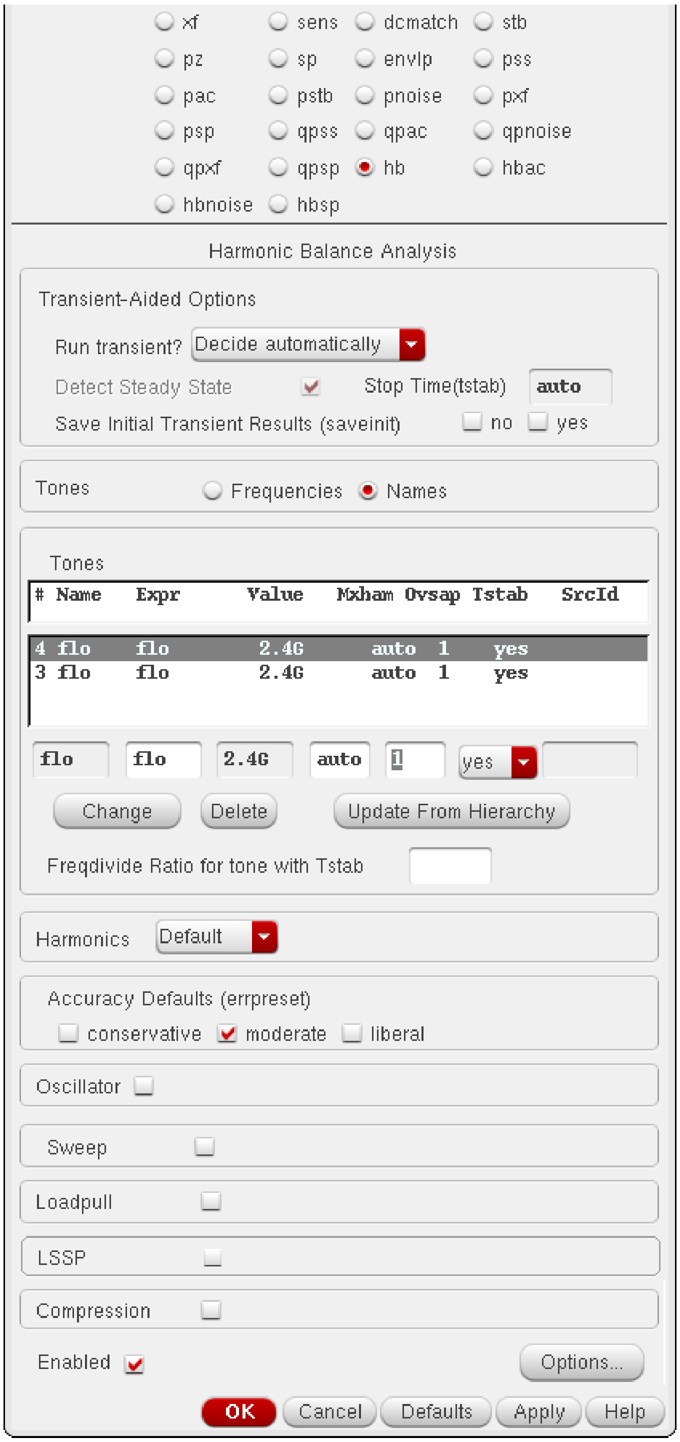


Rapid IP3/IP2

Note: The first discussion is about Rapid IP3.

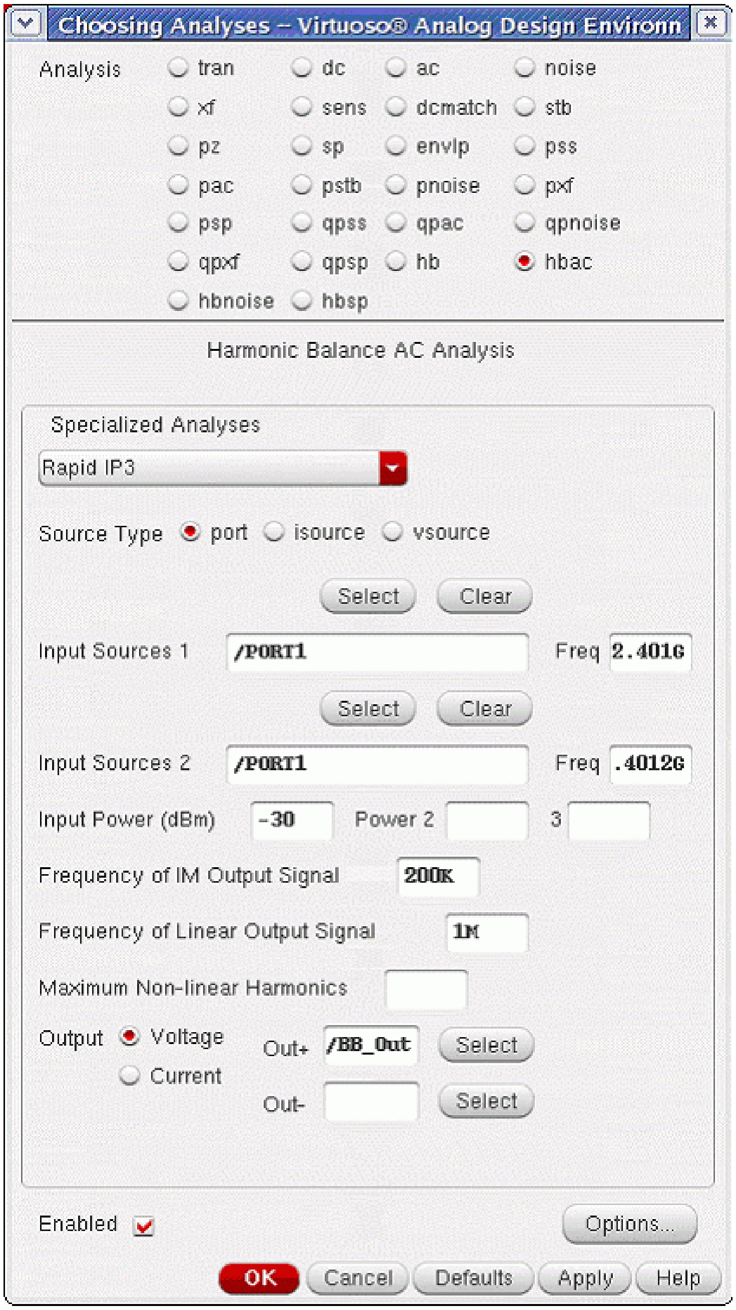
Rapid IP3 and Rapid IP2 are available in the AC and the hbac *Choosing Analyses* forms. Rapid IP3 in the AC form is for measuring IP3 of amplifiers where there is no frequency translation, and Rapid IP3 in the hbac form allows the measurement of IP3 where frequency translations are present. Rapid IP3 and IP2 measure small-signal IP3 and IP2.

1. First set up the hb form with just the LO applied, as shown below. For more information, see the harmonic balance section.



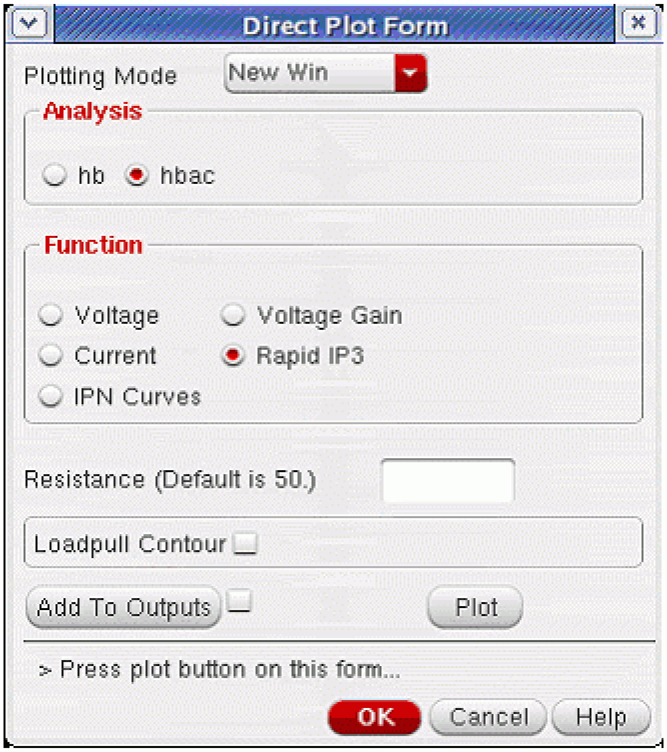
2. Select the hbac analysis. At the bottom of the form, in the *Specialized Analysis* section, select *Rapid IP3* or *Rapid IP2*, depending on what you want to measure. The entries for the forms are identical, except for the frequency of the intermodulation product. For switched FET mixers, make sure that the psp model is used. Incorrect results will be produced for BSIM3 and BSIM4 models due to a limitation in these models.

3. Define the input port, input frequencies, input power, the frequencies of the linear and second or third order products, and the output node. An example setup for Rapid IP3 is shown below. Make sure the input power is in the small-signal range for your circuit.



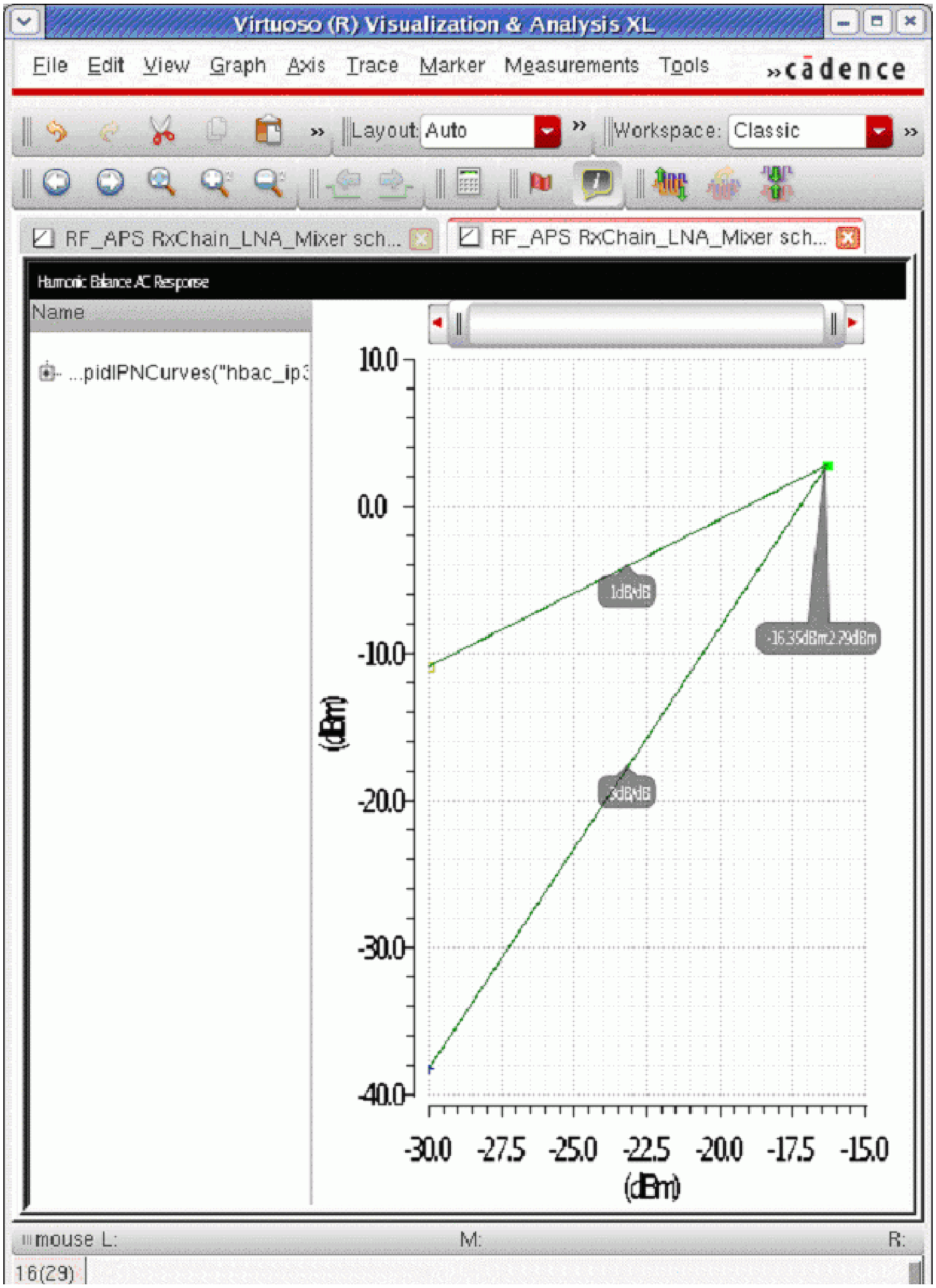
4. Leave the *Maximum Non-linear Harmonics* field blank.

5. Run the simulation. When the simulation is complete, select *Results - Direct Plot - Main Form*.



6. Select *hbac* results.
7. Select *Rapid IP3* or *RapidIP2*, whichever was run.
8. Click *Plot*.

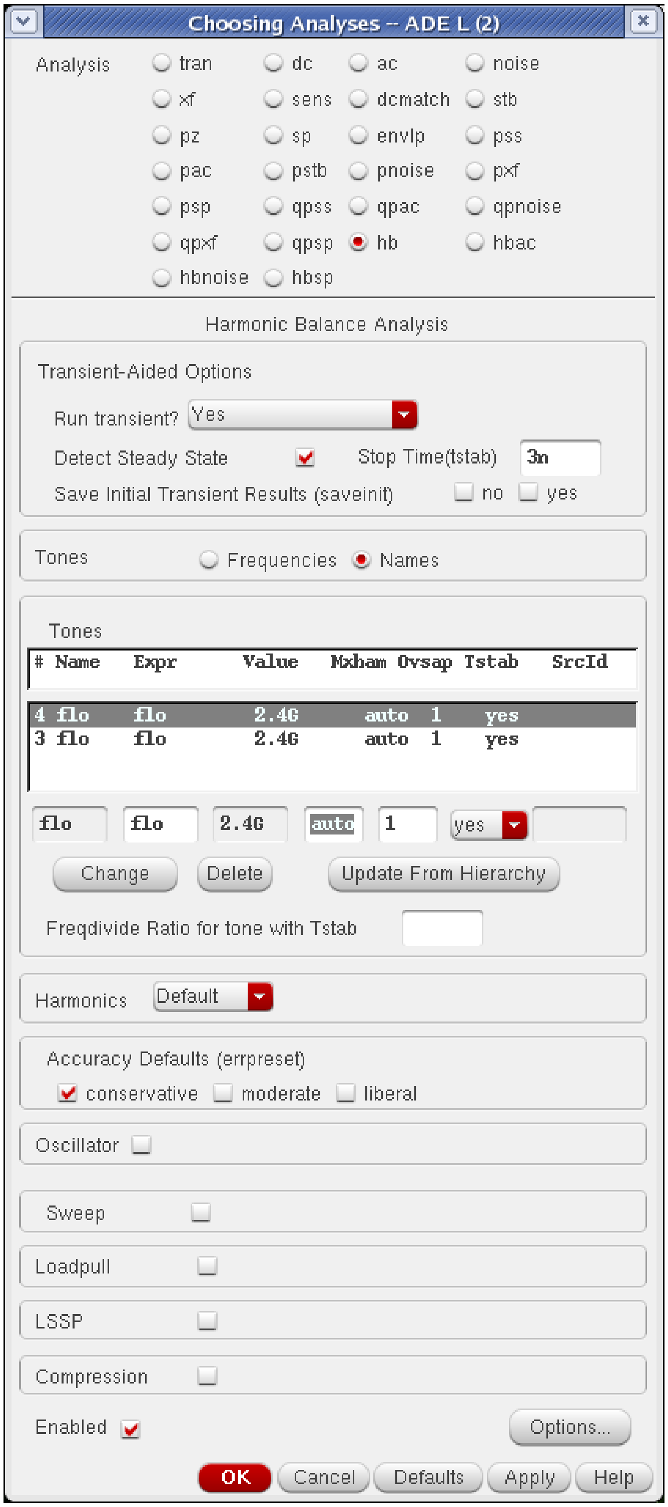
The waveform is displayed. The marker shown at the intercept point shows the input and output-referred IP3 or IP2.



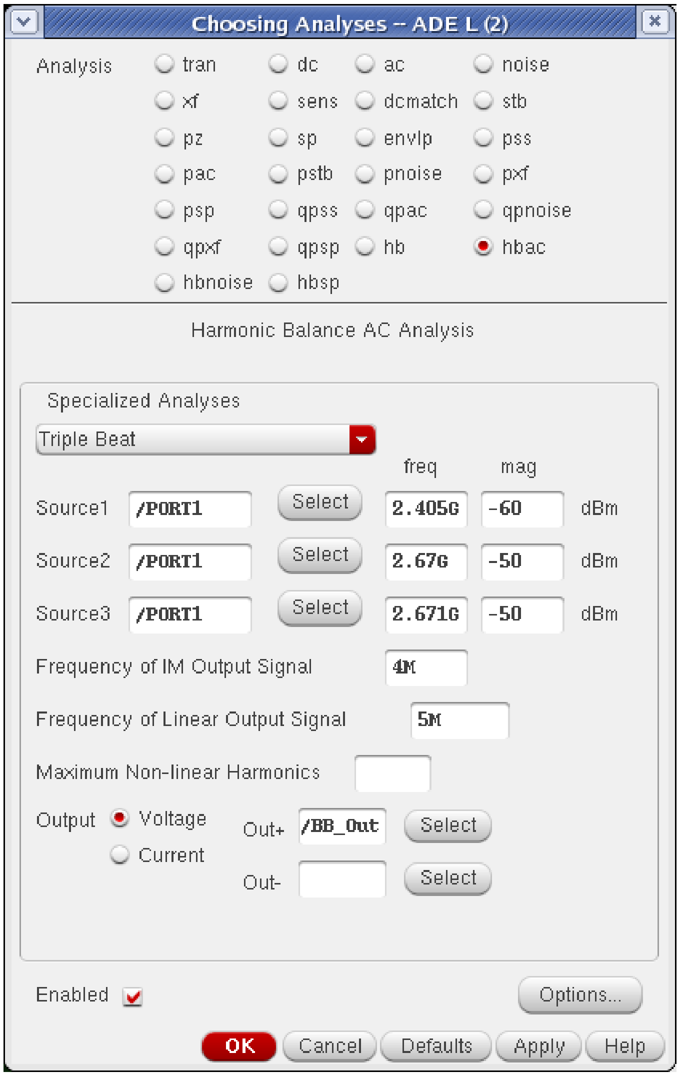
Triple Beat

Triple beat is similar to Rapid IP3 and IP2 except it allows three inputs instead of two. Currently, it is available only in the hbac *Choosing Analyses* form. This capability is provided for the case where two small-signal closely spaced inputs from a transmitter leak into a receiver along with the standard RF input to the receiver. The use model is similar to Rapid IP3. Apply the LO signal in hb, and then use hbac triple beat to simulate the three inputs to the receiver. This is a small-signal analysis. If the input levels for all three tones are not in the small-signal range, the result will be incorrect because hbac ignores large signal effects.

1. To run triple beat, first set up the hb *Choosing Analyses* form with just the LO signal applied.



2. Set up the hbac form with the RF signal and the two leakage signals. In this example, the RF signal is at 2.405G, and the other two signals are 1MHz apart. With the LO at 2.4GHz, this gives an intended baseband output at 5M, and triple beat products at the transmitter spacing (1MHz) above and below the 5MHz output.



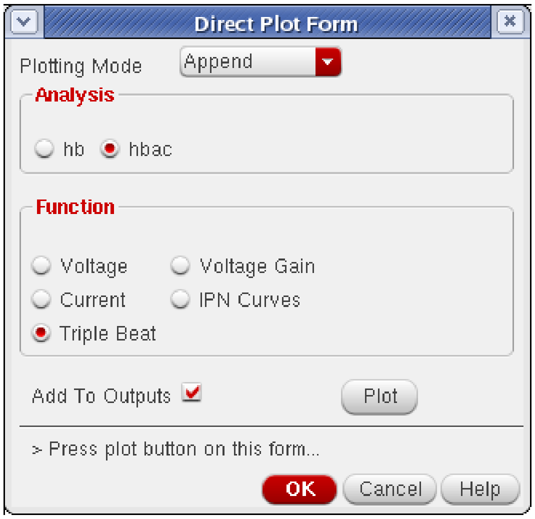
3. Run the simulation,

4. Choose *Results - Direct Plot - Main Form* to open the *Direct Plot Form*.

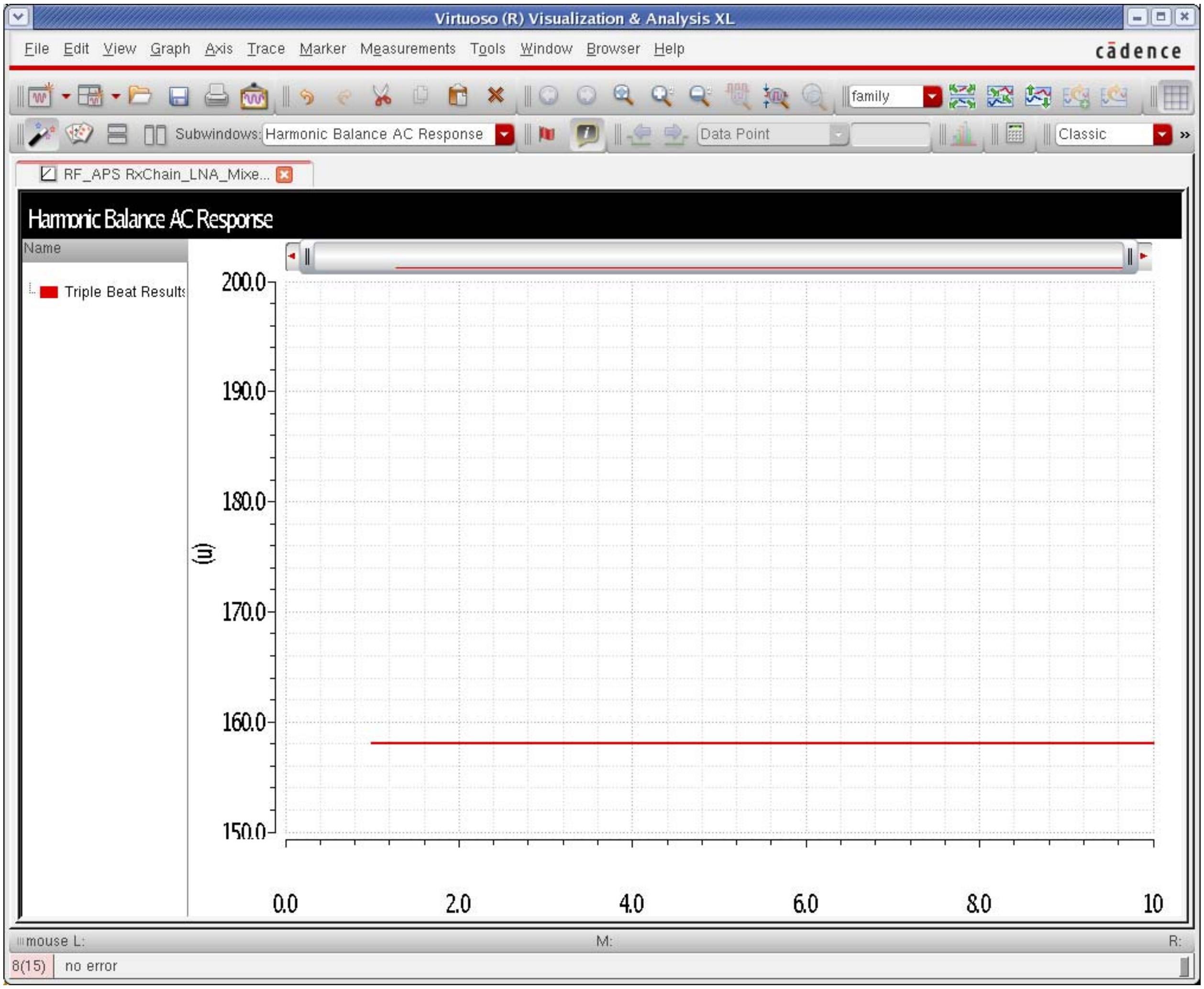
5. Select *hbac* from the *Analysis* section.

6. Select *Triple Beat* from the *Function* section.

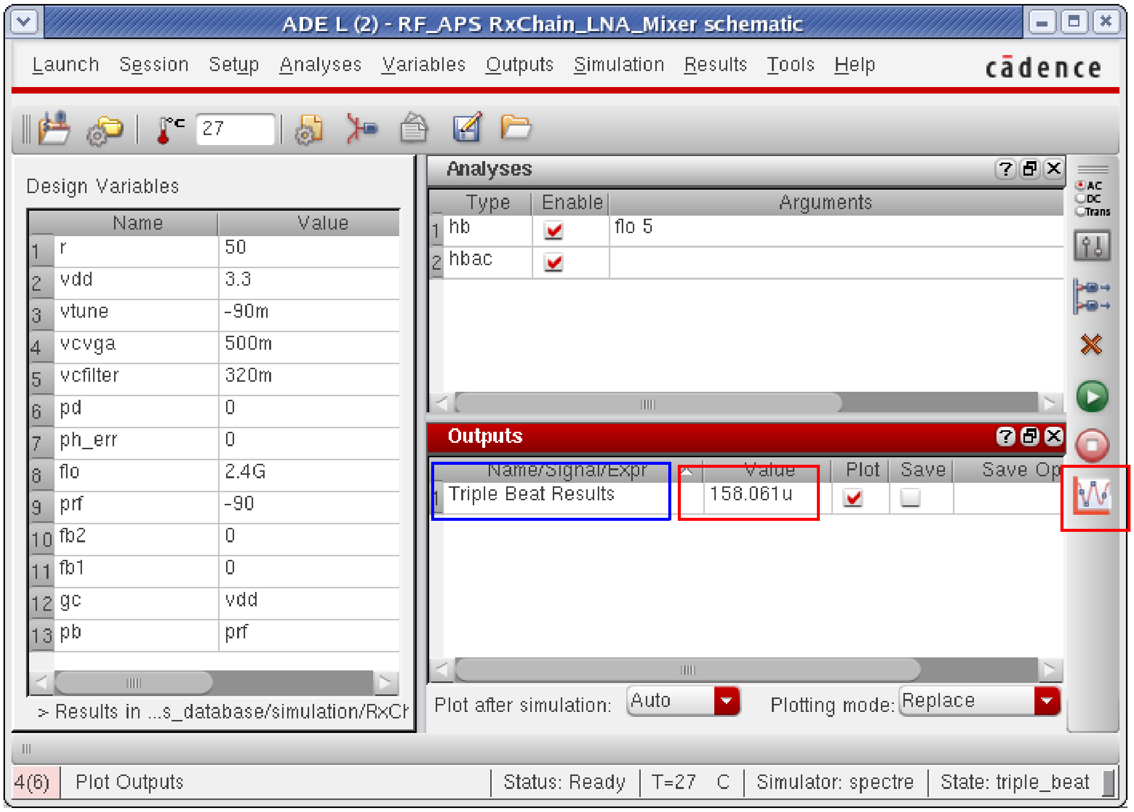
7. If desired, select *Add To Outputs*. This will add an expression in the ADE *Outputs* section.



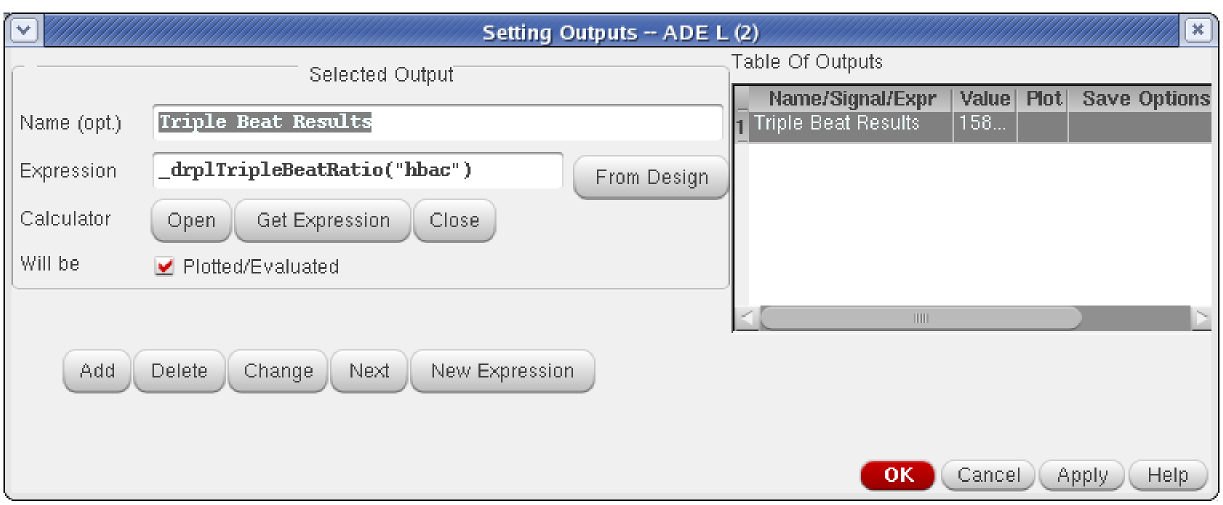
The waveform tool plots the linear ratio of IM3 divided by IM1 with the X axis values from 1 to 10. This function is more useful in the ADE window itself.



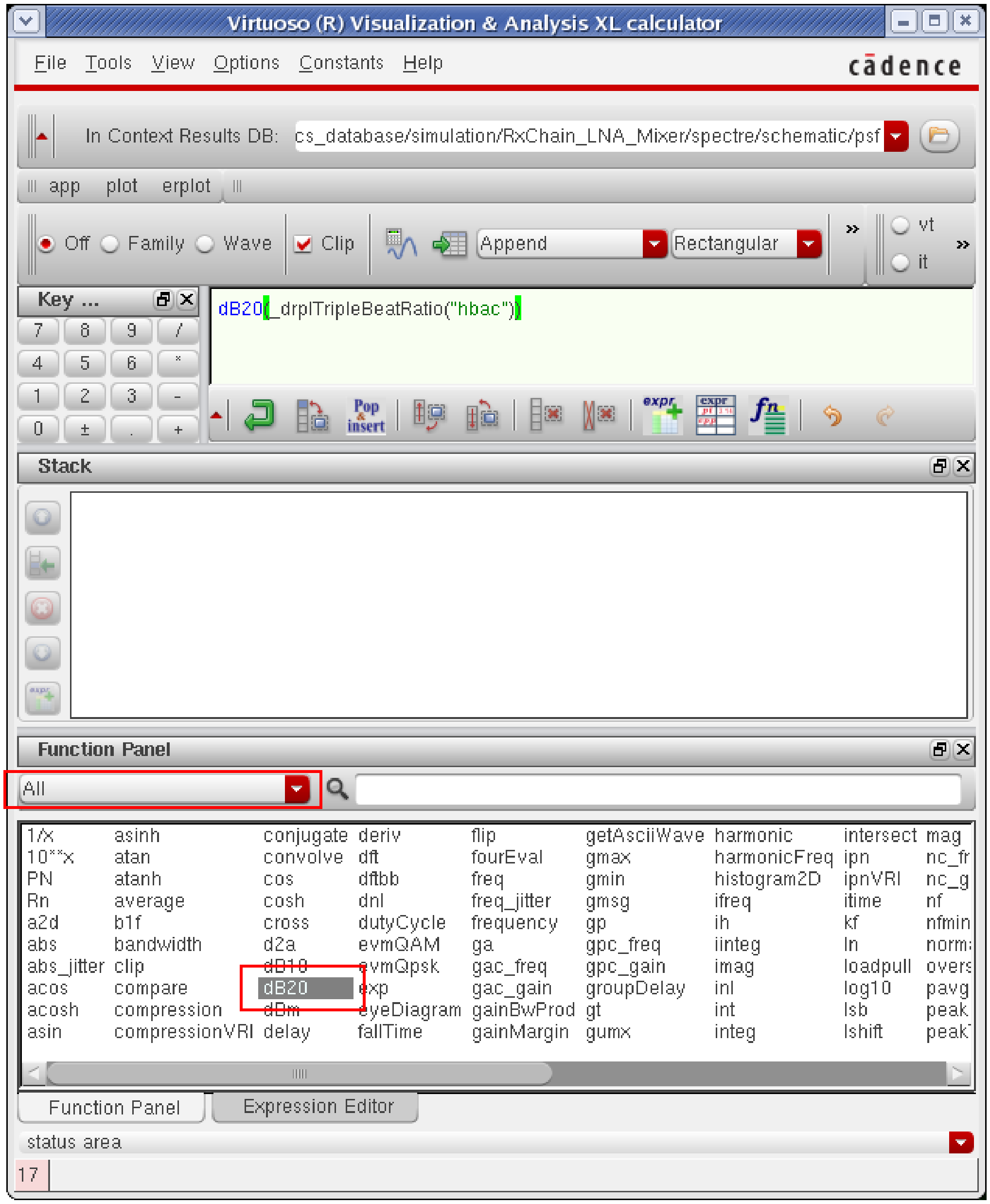
8. In ADE, click the *Plot Outputs* icon. The expression updates with the linear ratio of IM1 divided by IM3
.



9. To convert the output to dB, double-click the expression in the area highlighted by the blue box above. The Setting Outputs window is displayed, as shown below.

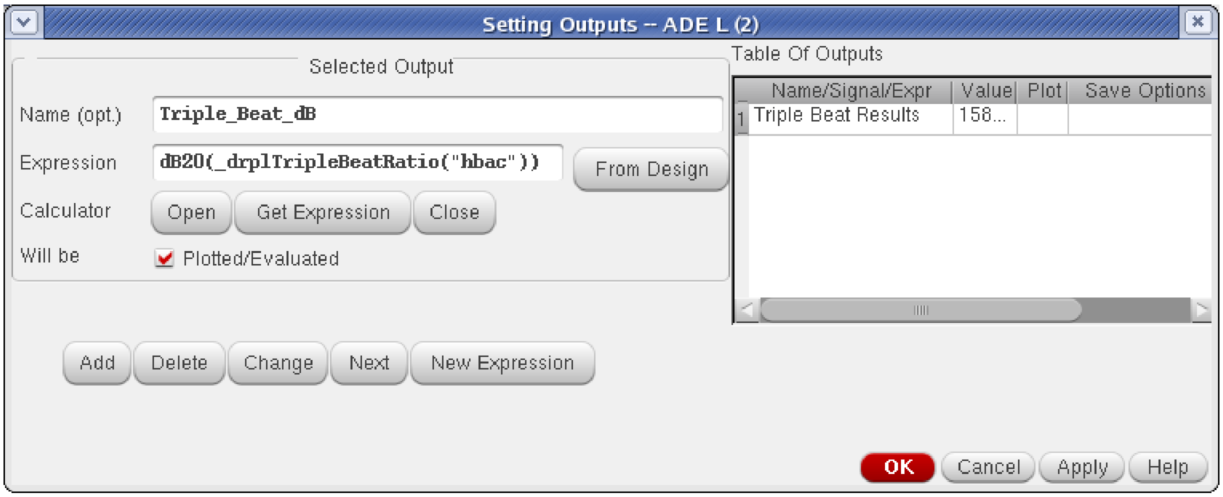


10. Click *Open*. The calculator opens with the expression in the buffer.



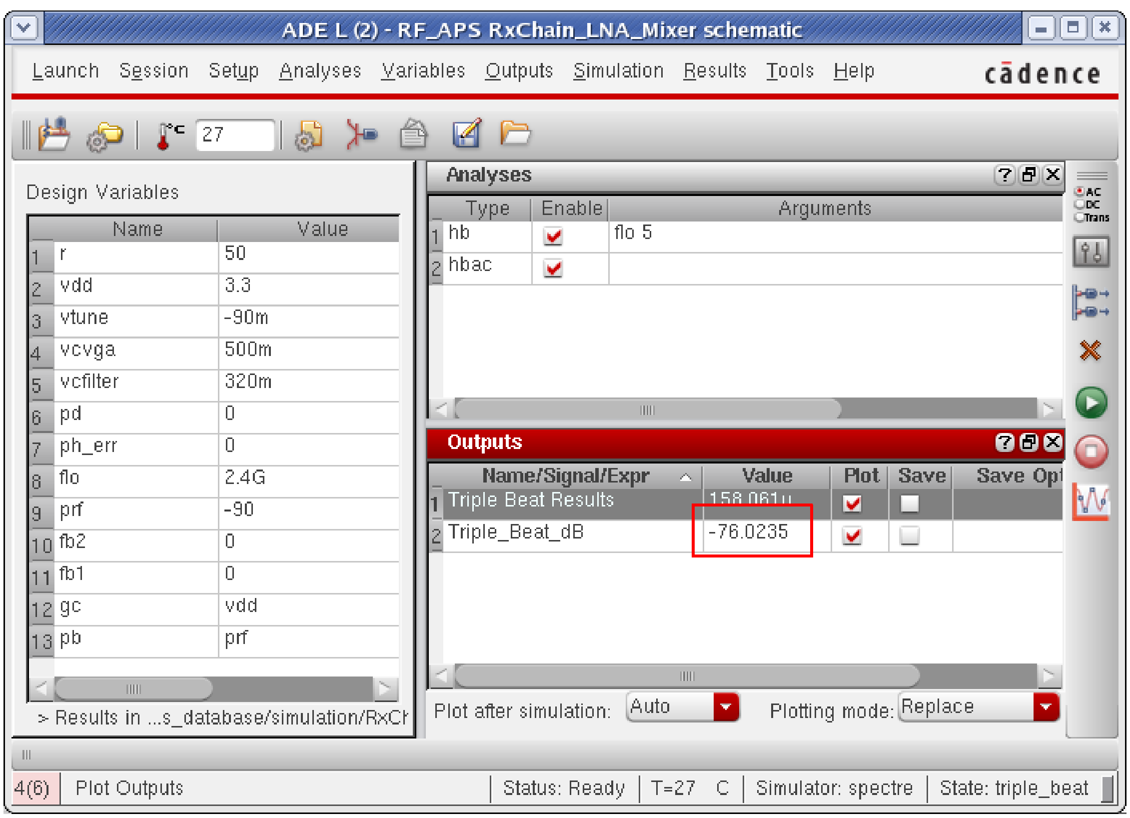
11. If necessary, select *All* just below the *Function* Panel legend. This is highlighted above. Add the *dB20* function.

12. In the Setting Outputs window, click *Next* until the fields are blank. Click *Get Expression*. Add a title, such as *Triple_Beat_dB*.

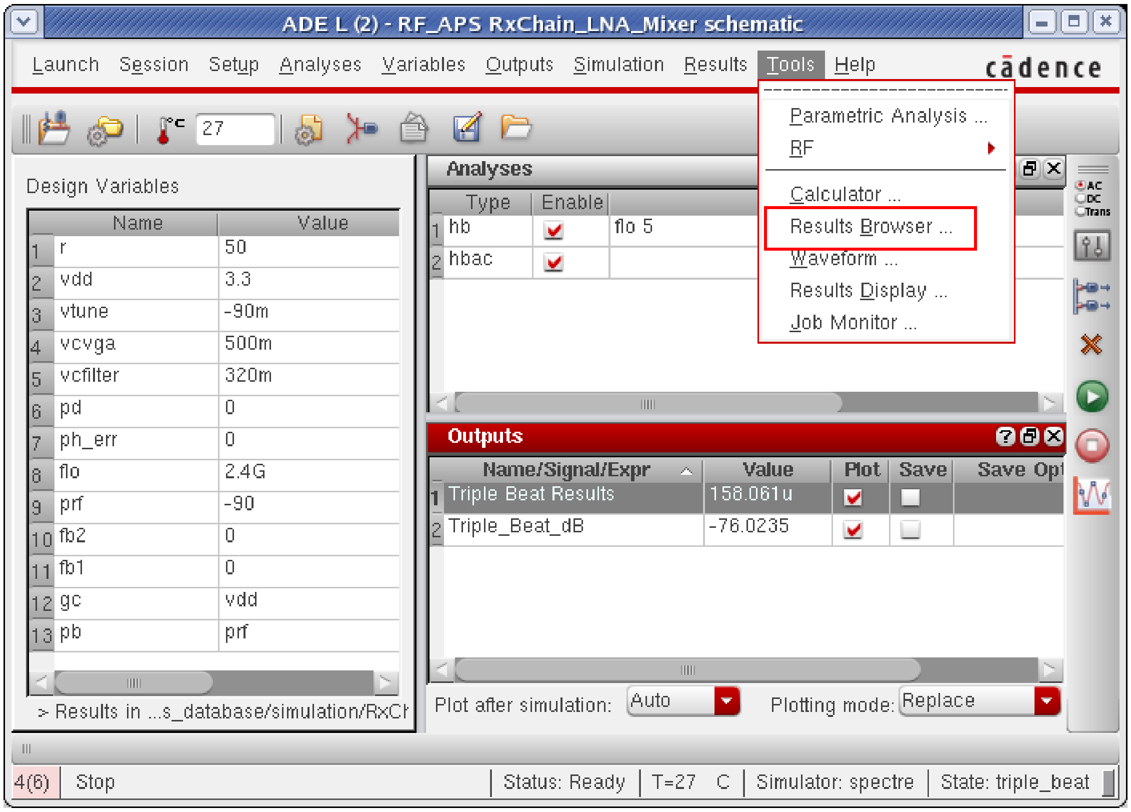


13. Click *Add*. This adds the expression to ADE.

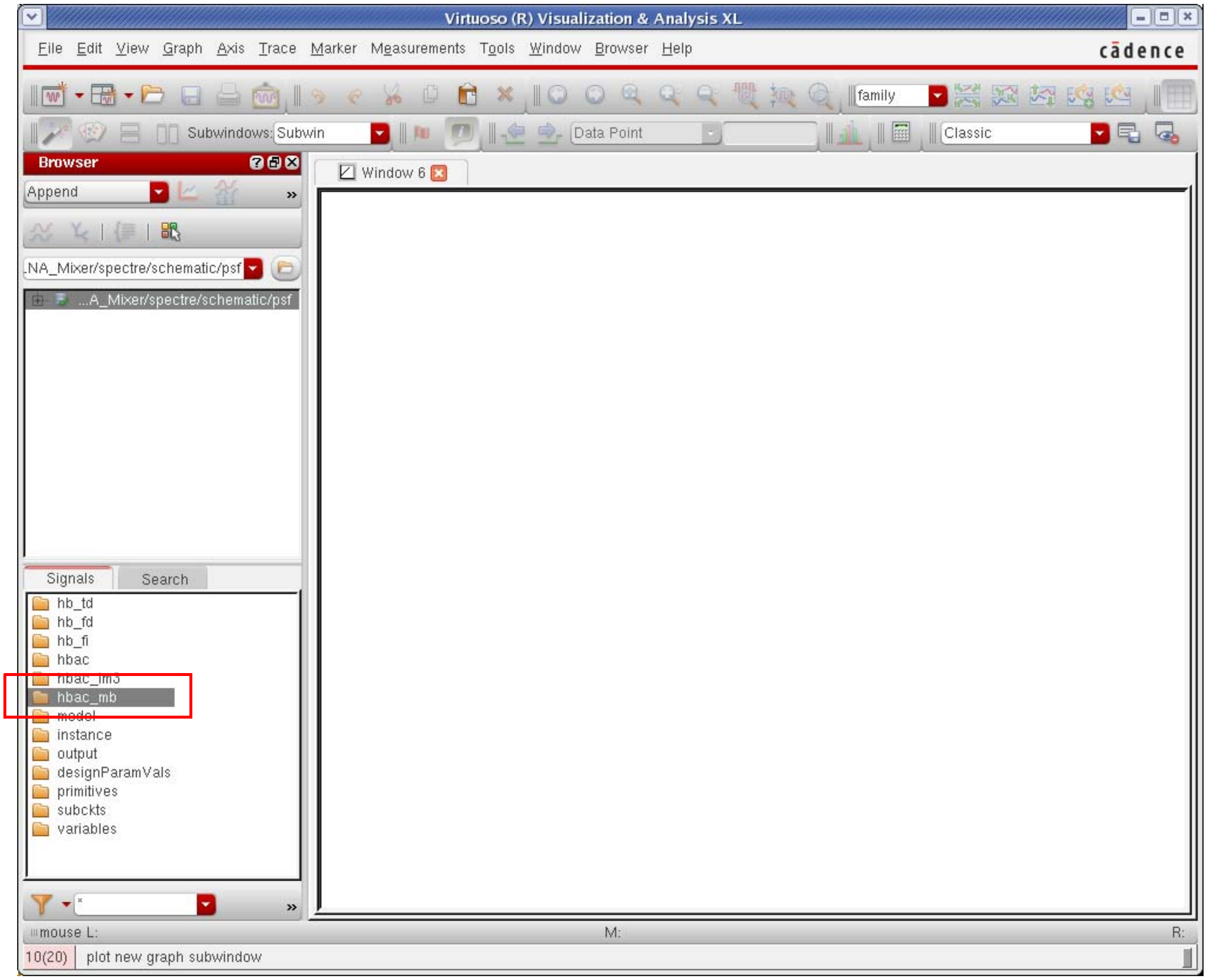
14. In ADE, click the *Plot Outputs* icon. The ratio is now in dB.



15. If the IM1 and IM3 levels are desired, choose *Tools - Results Browser*.

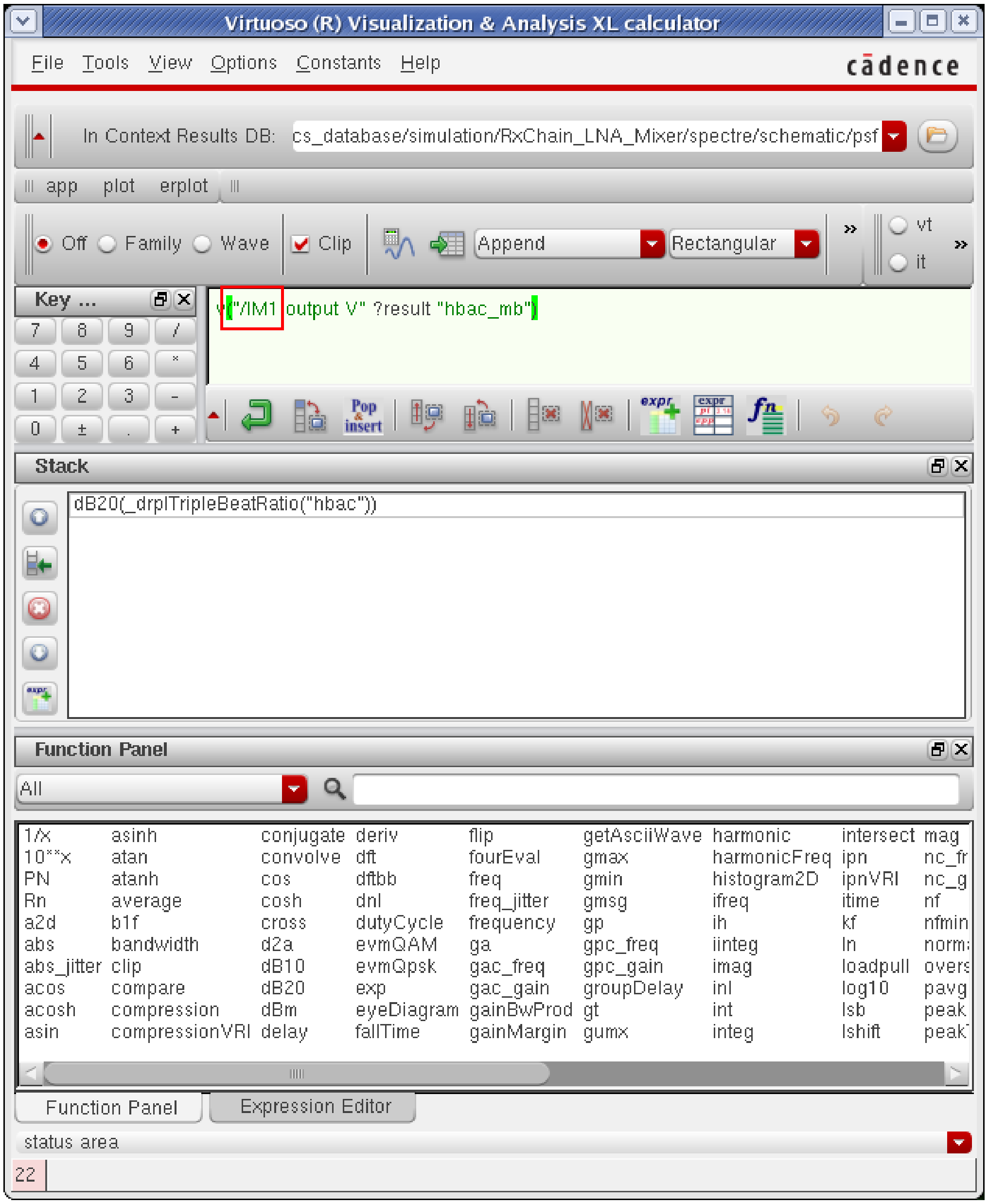


The Virtuoso Visualization and Analysis window is displayed.

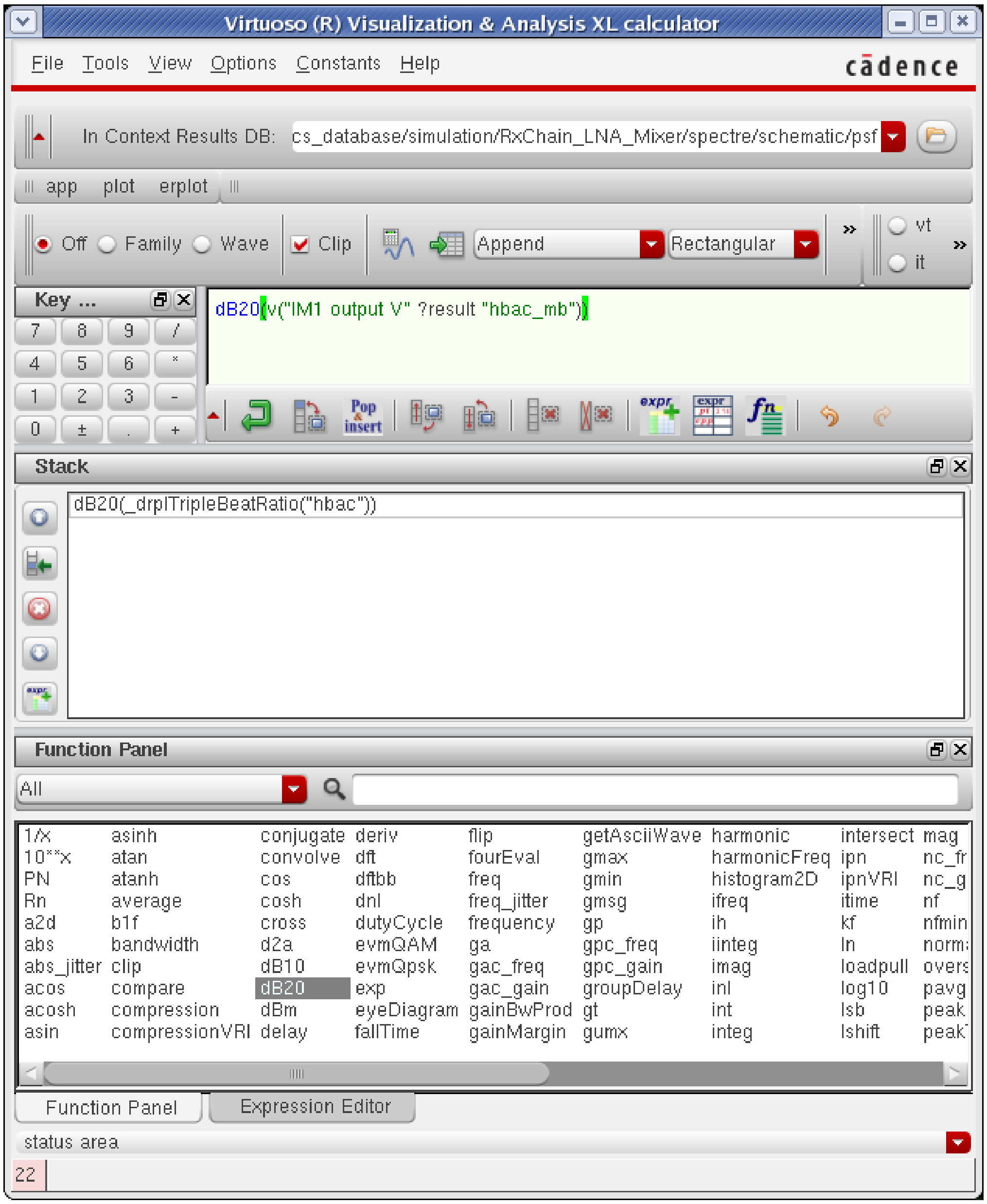


16. Double-click the *hbac_mb* results.
17. Click the *IM1 output V(V)=*<`your result_here`> expression.

18. Click the right mouse button and selet *Calculator*. The calculator opens with the expression in the buffer.

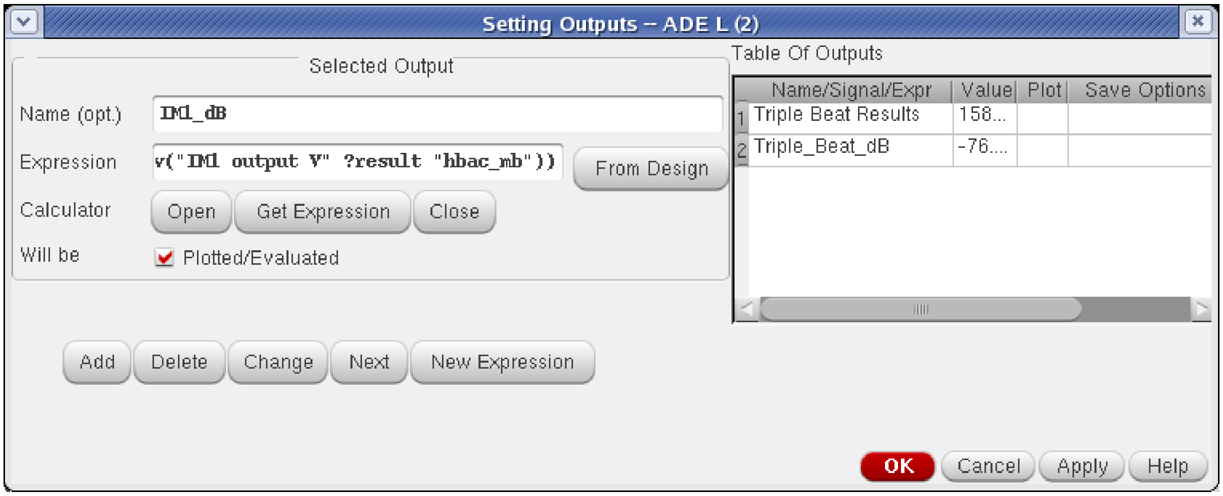


19. In the calculator, examine the expression. If there is a slash before *IM1*, remove it. Refer to the red box above. If desired, add the *dB20* function.



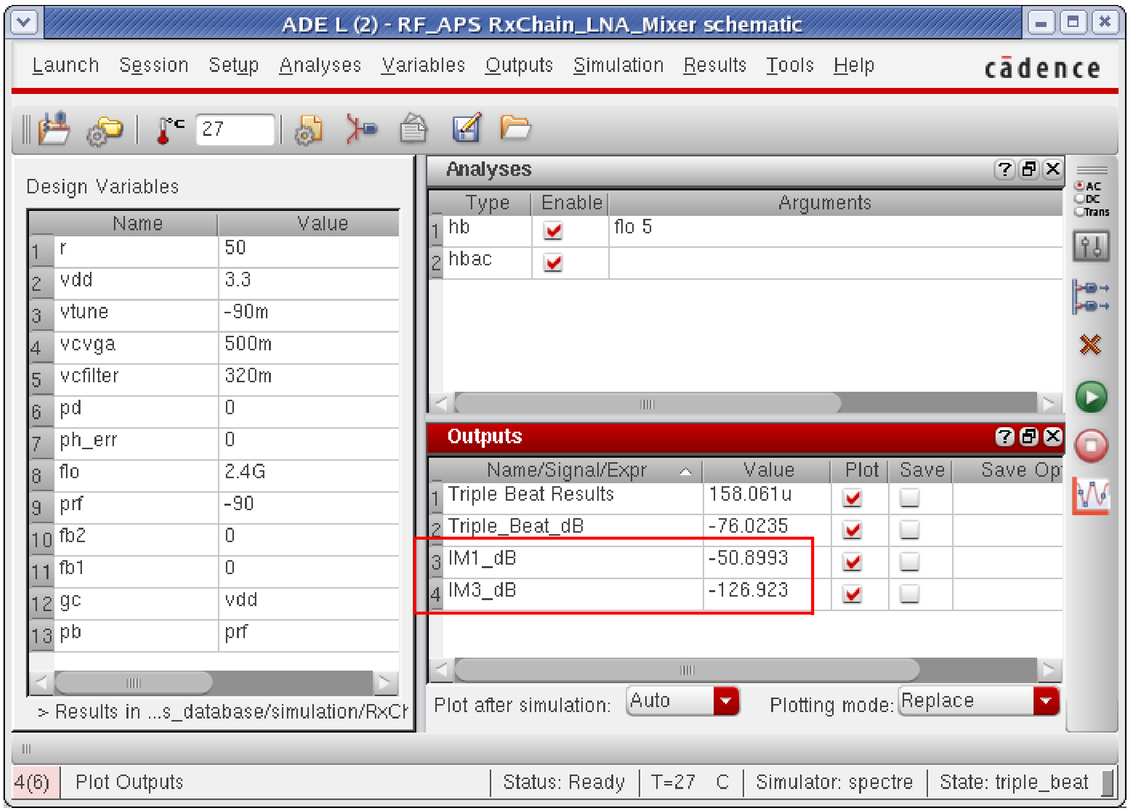
20. In the *Setting Outputs* window, click *Next* until both fields are blank.

21. Click *Get Expression*. Type in a name, such as I*M1_dB*.



22. Click *Add*. This adds the expression to the ADE *Outputs* section.

23. If the level for the IM3 product is desired, follow the previous steps, but send the IM3 expression to the calculator. When you are done, click the *Plot Outputs* icon in ADE. The results are displayed in the ADE *Outputs* section.



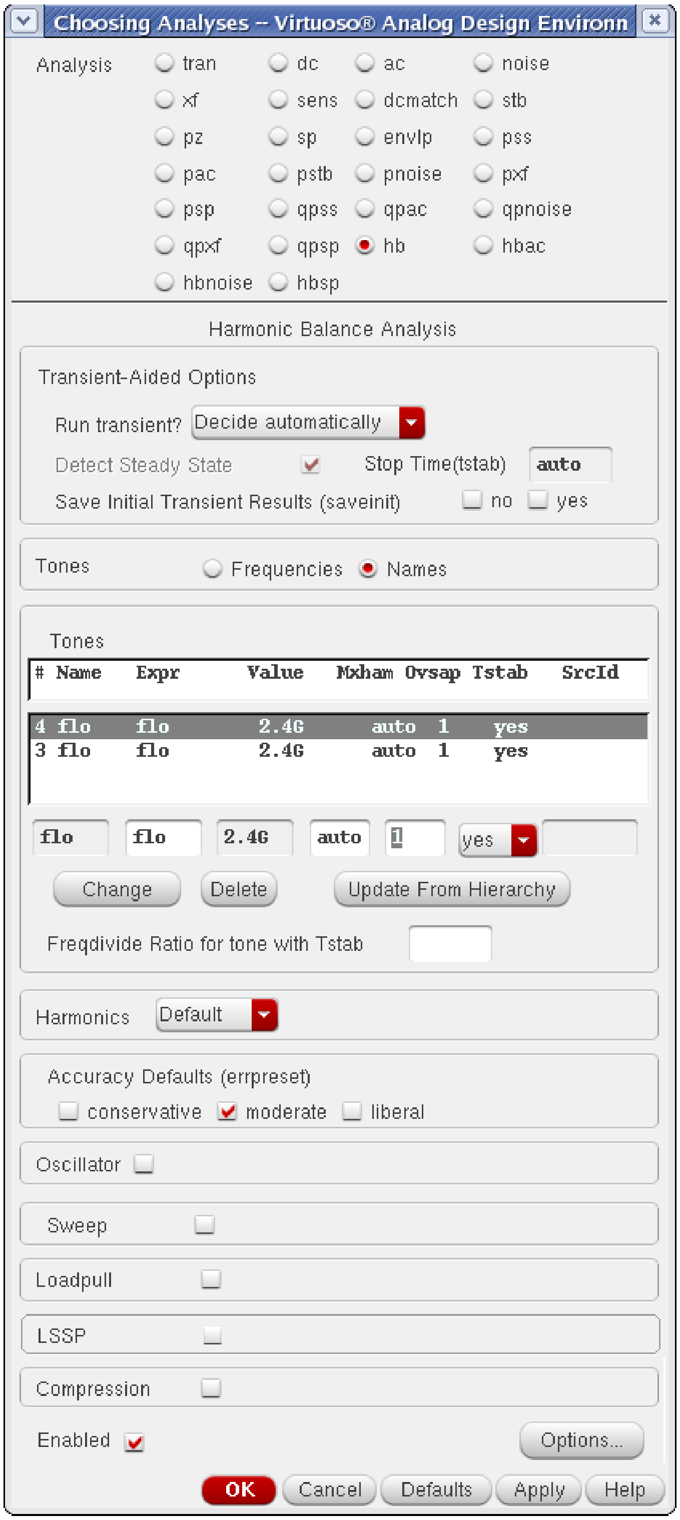
Compression Distortion Summary

Note that the *Compression Distortion Summary* is available in the AC and the hbac *Choosing Analyses* forms. *Compression Distortion Summary* in the AC form is for measuring Compression Distortion of amplifiers where there is no frequency translation, and *Compression Distortion Summary* in the hbac form allows the measurement of Compression Distortion where frequency translations are present.

With the Compression Distortion summary, you can see which components in a frequency translating circuit cause relatively more and less distortion.

Note that compression and IP3 are related mathematically, so this analysis gives information about which components contribute more and less to the IP3 measurement as well as which components contribute to compression.

1. First set up the hb analysis with just the LO applied, as shown below. For more information, see the harmonic balance section.

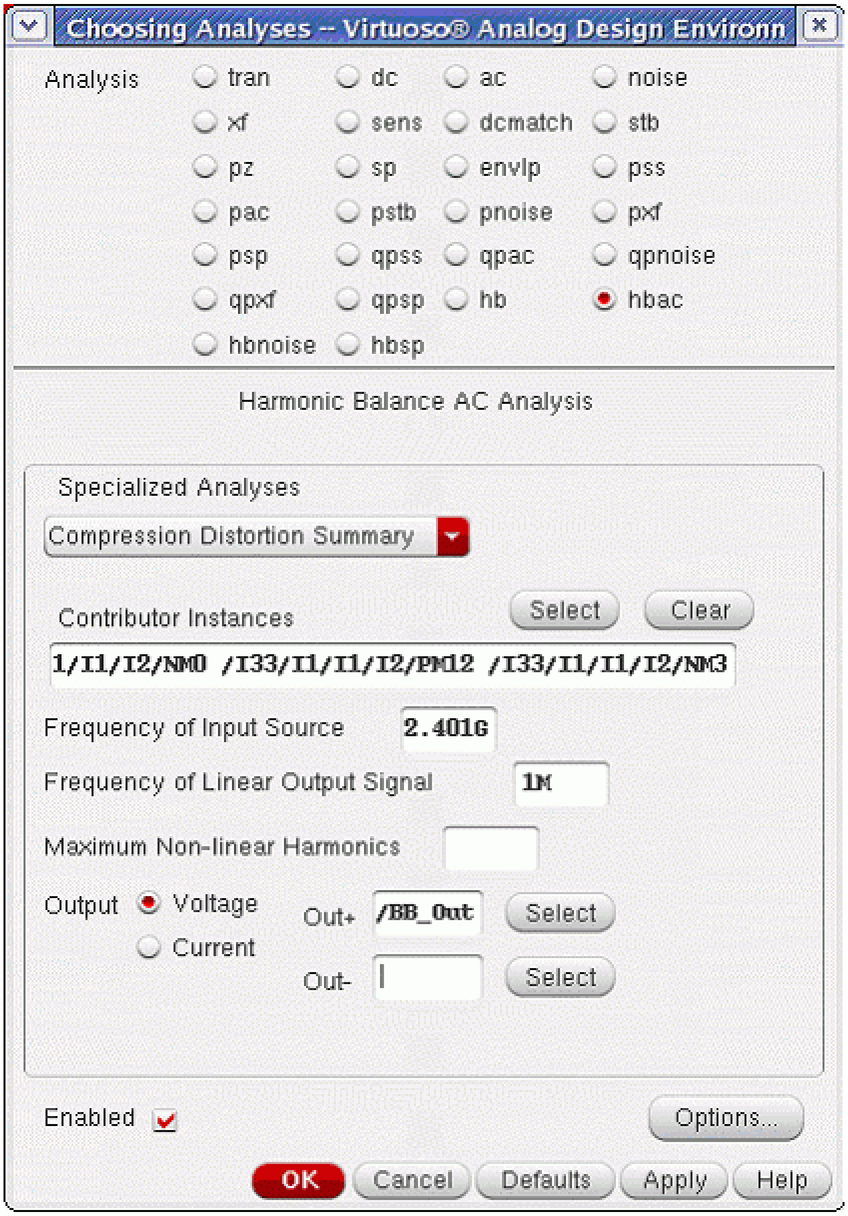


2. Now select the hbac analysis.
3. Select *Compression Distortion Summary* from the *Specialized Analyses* drop-down list.

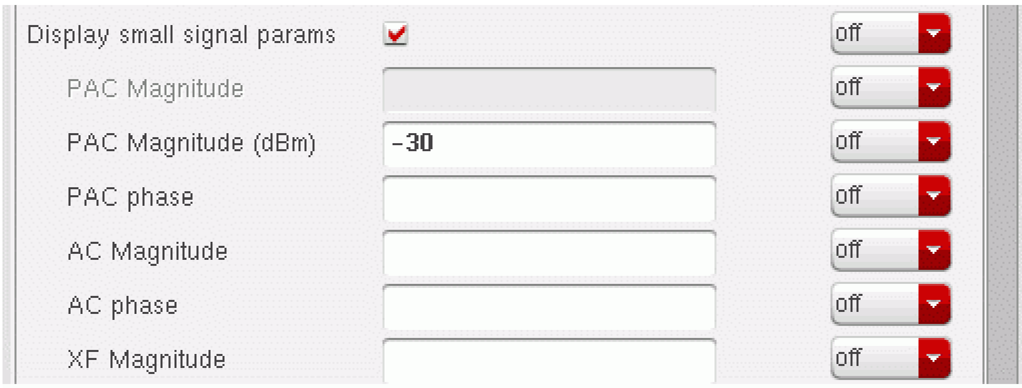
   The following notice appears to remind you to set the PAC amplitude parameter in the source in the small-signal range.

4. If you leave the *Contributor Instances* field blank, all the nonlinear devices in your circuit will be run. If you want a subset, click *Select*, and select the instances that you want on the schematic.
5. Specify the input frequency in the *Frequency of Input Source* field.
6. Specify the output frequency in the *Frequency of Linear Output Signal* field.
7. Leave the *Maximum Non-Linear Harmonics* field blank.
8. Select the output node in the circuit.

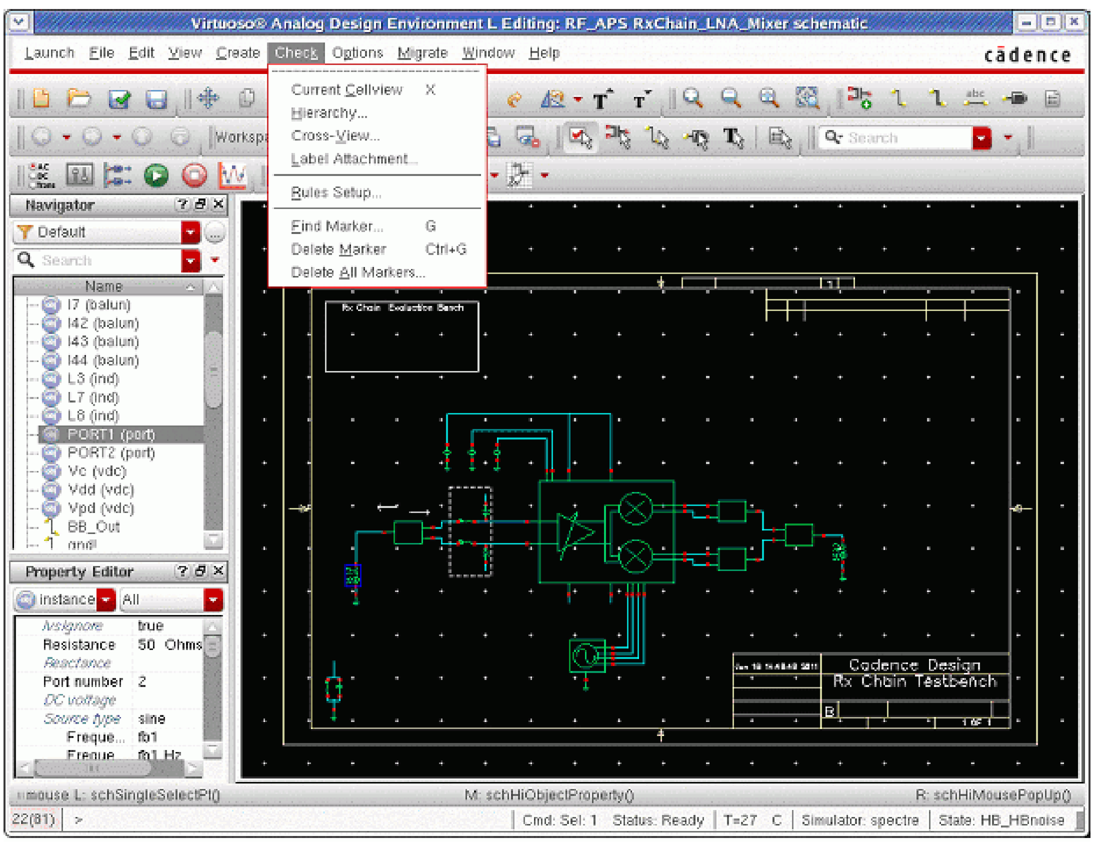
The completed form should appear similar to the one shown below.



9. In the *Edit Object Properties* form, make sure that the *PAC magnitude (dBm)* is set in the small-signal range. In this example, it is set to `-30Bm`.



10. If you make changes to your schematic that you want for the distortion summary, but you do not want to save them to the circuit file, make the desired changes and then select *Check - Current Cellview* in the schematic window. Later, if you want to save them, just select *Check* and *Save*. If you do not want them, quit the schematic window, and do not save the changes.

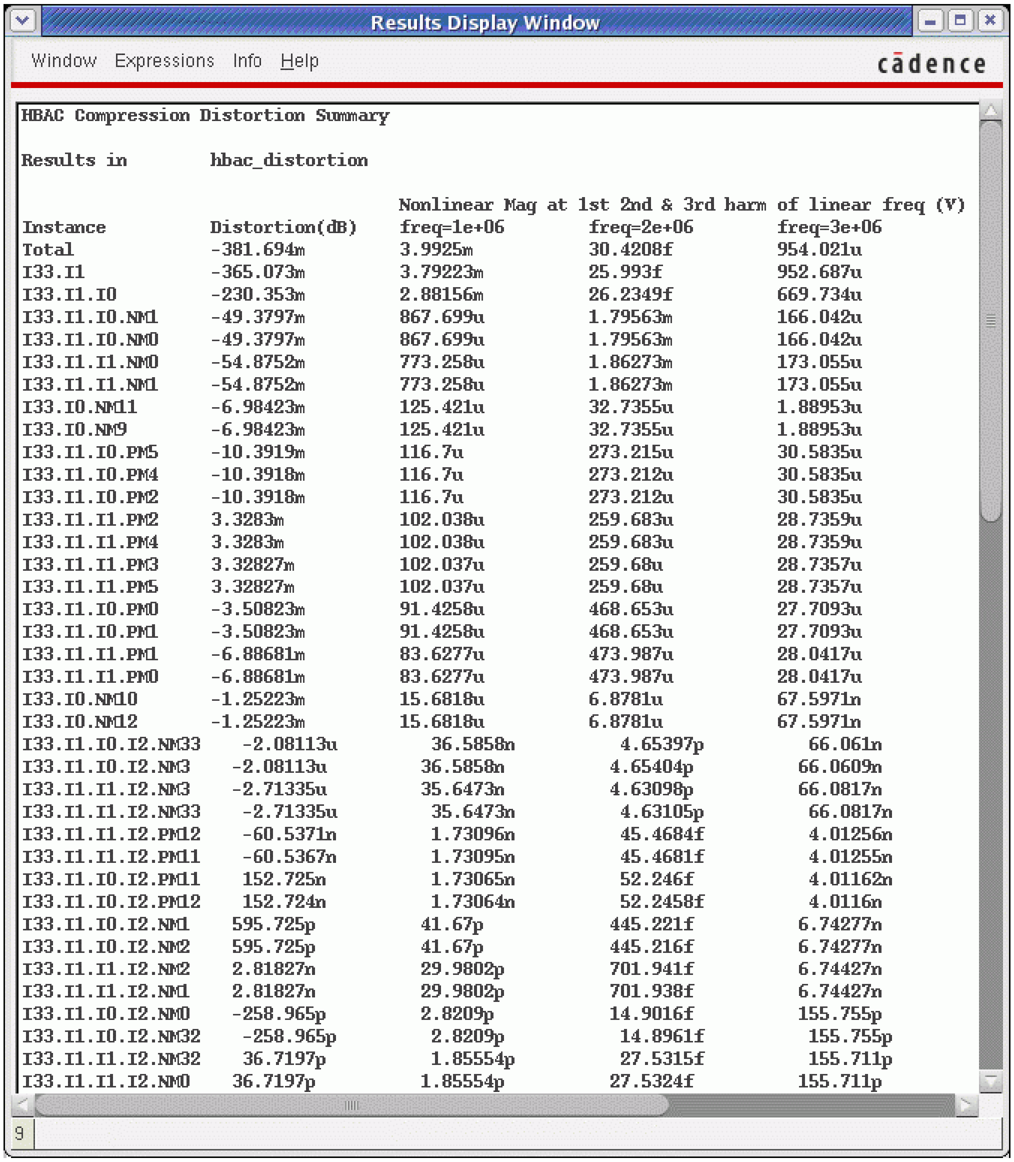


ADE will netlist from the checked view, so it will get the changes.

11. When you are done, run the simulation.

12. When the simulation completes, select *Results - Print - hbAC Distortion Summary*.

The summary is displayed below:



HBAC Compression Distortion Summary

Results in hbac_distortion

		Nonlinear Mag at 1st 2nd & 3rd harm of linear freq (V)		
Instance	Distortion(dB)	freq=1e+06	freq=2e+06	freq=3e+06
Total	-381.694m	3.9925m	30.4208f	954.021u
I33.I1	-365.073m	3.79223m	25.993f	952.687u
I33.I1.I0	-230.353m	2.88156m	26.2349f	669.734u
I33.I1.I0.NM1	-49.3797m	867.699u	1.79563m	166.042u
I33.I1.I0.NM0	-49.3797m	867.699u	1.79563m	166.042u
I33.I1.I1.NM0	-54.8752m	773.258u	1.86273m	173.055u
I33.I1.I1.NM1	-54.8752m	773.258u	1.86273m	173.055u
I33.I0.NM11	-6.98423m	125.421u	32.7355u	1.88953u
I33.I0.NM9	-6.98423m	125.421u	32.7355u	1.88953u
I33.I1.I0.PM5	-10.3919m	116.7u	273.215u	30.5835u
I33.I1.I0.PM4	-10.3918m	116.7u	273.212u	30.5835u
I33.I1.I0.PM2	-10.3918m	116.7u	273.212u	30.5835u
I33.I1.I1.PM2	3.3283m	102.038u	259.683u	28.7359u
I33.I1.I1.PM4	3.3283m	102.038u	259.683u	28.7359u
I33.I1.I1.PM3	3.32827m	102.037u	259.68u	28.7357u
I33.I1.I1.PM5	3.32827m	102.037u	259.68u	28.7357u
I33.I1.I0.PM0	-3.50823m	91.4258u	468.653u	27.7093u
I33.I1.I0.PM1	-3.50823m	91.4258u	468.653u	27.7093u
I33.I1.I1.PM1	-6.88681m	83.6277u	473.987u	28.0417u
I33.I1.I1.PM0	-6.88681m	83.6277u	473.987u	28.0417u
I33.I0.NM10	-1.25223m	15.6818u	6.8781u	67.5971n
I33.I0.NM12	-1.25223m	15.6818u	6.8781u	67.5971n
I33.I1.I0.I2.NM33	-2.08113u	36.5858n	4.65397p	66.061n
I33.I1.I0.I2.NM3	-2.08113u	36.5858n	4.65404p	66.0609n
I33.I1.I1.I2.NM3	-2.71335u	35.6473n	4.63098p	66.0817n
I33.I1.I1.I2.NM33	-2.71335u	35.6473n	4.63105p	66.0817n
I33.I1.I1.I2.PM12	-60.5371n	1.73096n	45.4684f	4.01256n
I33.I1.I1.I2.PM11	-60.5367n	1.73095n	45.4681f	4.01255n
I33.I1.I0.I2.PM11	152.725n	1.73065n	52.246f	4.01162n
I33.I1.I0.I2.PM12	152.724n	1.73064n	52.2458f	4.0116n
I33.I1.I0.I2.NM1	595.725p	41.67p	445.221f	6.74277n
I33.I1.I0.I2.NM2	595.725p	41.67p	445.216f	6.74277n
I33.I1.I1.I2.NM2	2.81827n	29.9802p	701.941f	6.74427n
I33.I1.I1.I2.NM1	2.81827n	29.9802p	701.938f	6.74427n
I33.I1.I0.I2.NM0	-258.965p	2.8209p	14.9016f	155.755p
I33.I1.I0.I2.NM32	-258.965p	2.8209p	14.8961f	155.755p
I33.I1.I1.I2.NM32	36.7197p	1.85554p	27.5315f	155.711p
I33.I1.I1.I2.NM0	36.7197p	1.85554p	27.5324f	155.711p

- ❑ The first column is the instance name in the schematic.
- ❑ The second column is the gain with the signal applied divided by the ideal gain.

Gain compression is shown as a negative number. Gain expansion is shown as a positive number.

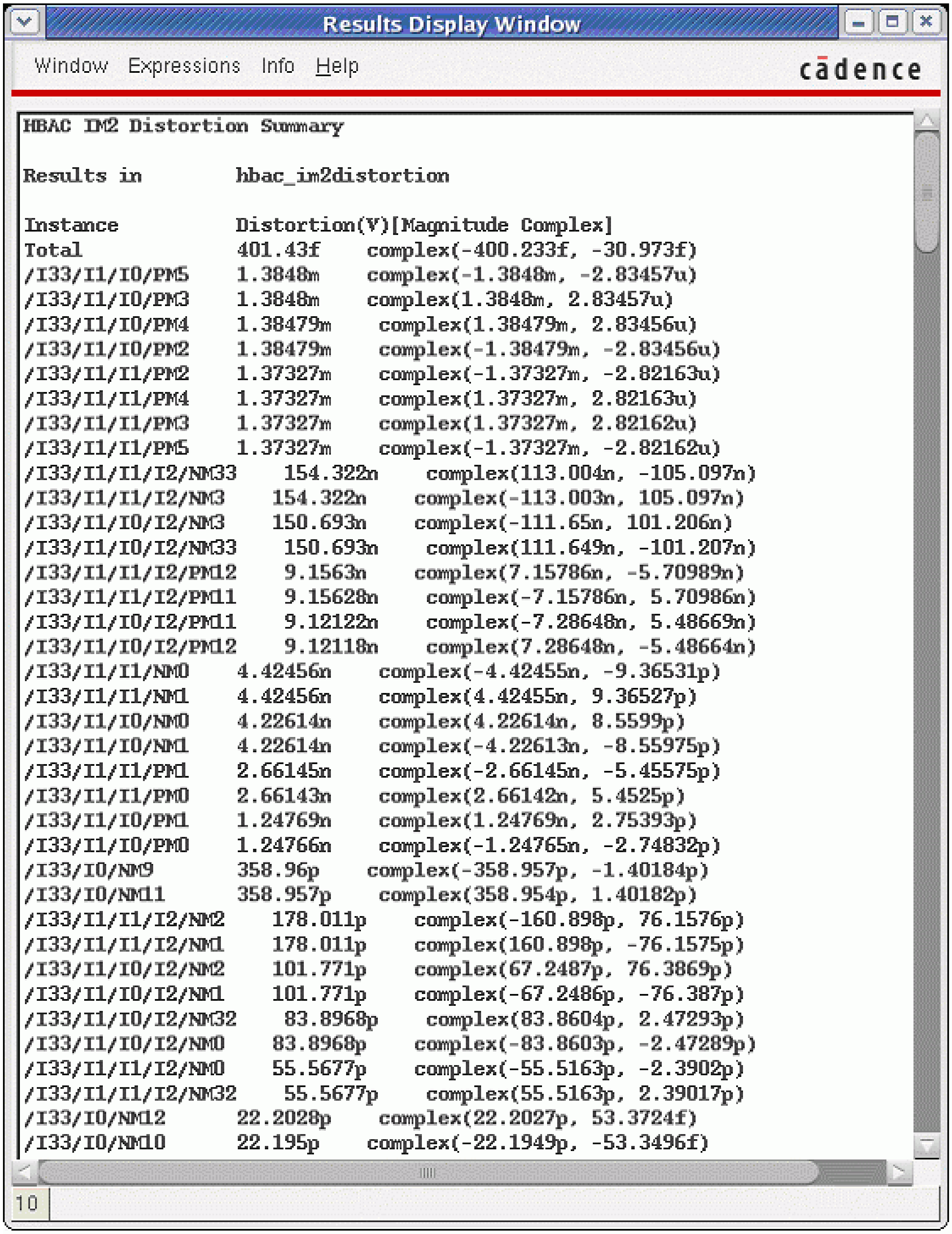
- ❑ The third, fourth, and fifth columns are the amplitudes of the first, second, and third harmonic of the linear output frequency with just the nonlinearity of that single device taken into account.

IM2 Distortion Summary

The steps for the IM2 distortion summary are very similar to the Compression distortion summary in the previous section. The differences are noted below.

- The IM2 distortion summary provides the output voltage (magnitude, real, and imaginary parts) of the 2nd order distortion signal for each component in the schematic by itself.
- The IM2 distortion summary does not provide a selection mechanism for the devices in the circuit. All the devices are run in the IM2 distortion summary.
- The output format is slightly different. The first column is the instance name in the schematic. The second column is the magnitude of the IM2 product that is produced by

that single component in volts. The third column has the real and imaginary parts of the voltage. A sample is shown below.



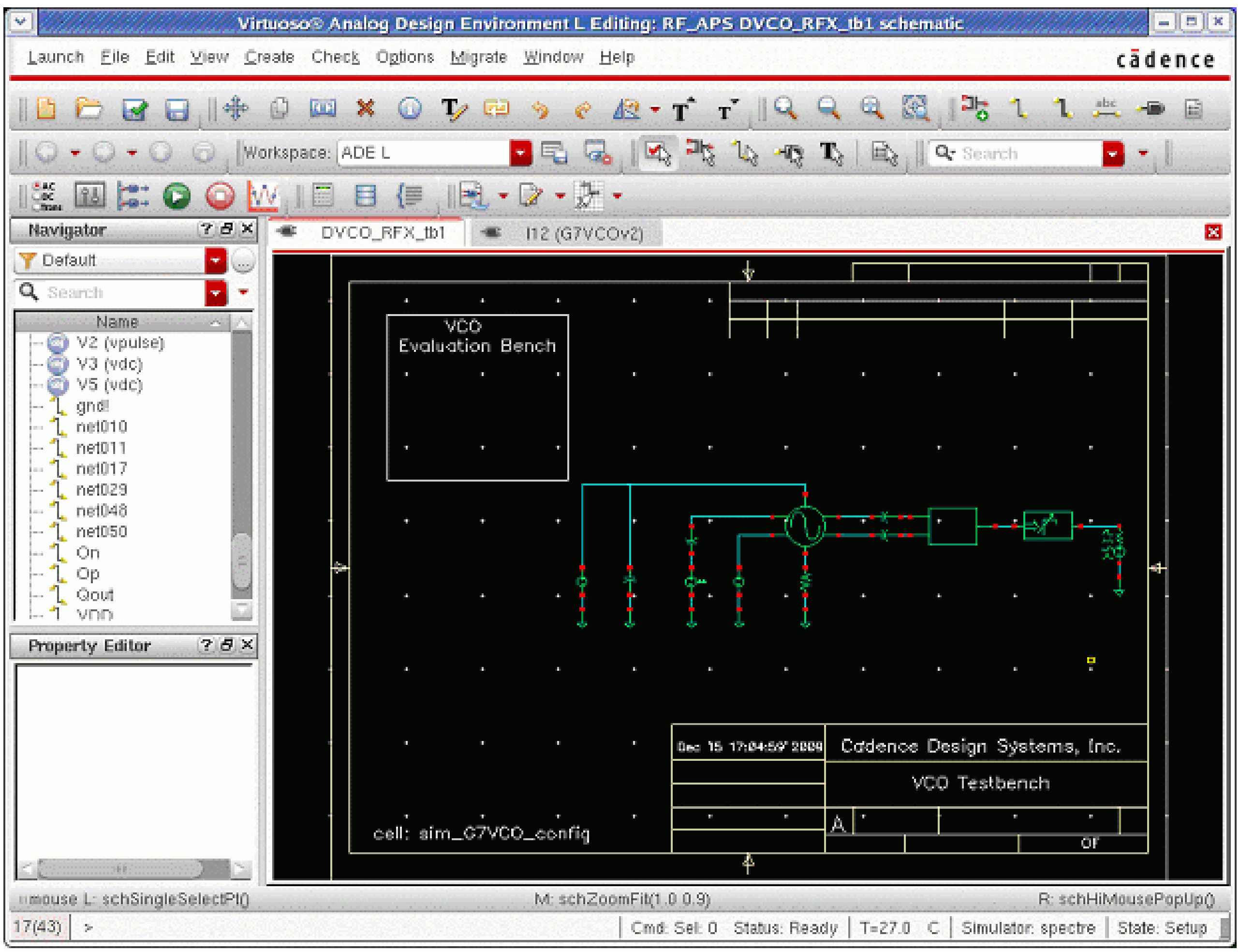

```
HBAC IM2 Distortion Summary

Results in          hbac_im2distortion

Instance            Distortion(V)[Magnitude Complex]
Total               401.43f     complex(-400.233f, -30.973f)
/I33/I1/I0/PM5      1.3848m     complex(-1.3848m, -2.83457u)
/I33/I1/I0/PM3      1.3848m     complex(1.3848m, 2.83457u)
/I33/I1/I0/PM4      1.38479m     complex(1.38479m, 2.83456u)
/I33/I1/I0/PM2      1.38479m     complex(-1.38479m, -2.83456u)
/I33/I1/I1/PM2      1.37327m     complex(-1.37327m, -2.82163u)
/I33/I1/I1/PM4      1.37327m     complex(1.37327m, 2.82163u)
/I33/I1/I1/PM3      1.37327m     complex(1.37327m, 2.82162u)
/I33/I1/I1/PM5      1.37327m     complex(-1.37327m, -2.82162u)
/I33/I1/I1/I2/NM33     154.322n     complex(113.004n, -105.097n)
/I33/I1/I1/I2/NM3     154.322n     complex(-113.003n, 105.097n)
/I33/I1/I0/I2/NM3     150.693n     complex(-111.65n, 101.206n)
/I33/I1/I0/I2/NM33     150.693n     complex(111.649n, -101.207n)
/I33/I1/I1/I2/PM12     9.1563n     complex(7.15786n, -5.70989n)
/I33/I1/I1/I2/PM11     9.15628n     complex(-7.15786n, 5.70986n)
/I33/I1/I0/I2/PM11     9.12122n     complex(-7.28648n, 5.48669n)
/I33/I1/I0/I2/PM12     9.12118n     complex(7.28648n, -5.48664n)
/I33/I1/I1/NM0      4.42456n     complex(-4.42455n, -9.36531p)
/I33/I1/I1/NM1      4.42456n     complex(4.42455n, 9.36527p)
/I33/I1/I0/NM0      4.22614n     complex(4.22614n, 8.5599p)
/I33/I1/I0/NM1      4.22614n     complex(-4.22613n, -8.55975p)
/I33/I1/I1/PM1      2.66145n     complex(-2.66145n, -5.45575p)
/I33/I1/I1/PM0      2.66143n     complex(2.66142n, 5.4525p)
/I33/I1/I0/PM1      1.24769n     complex(1.24769n, 2.75393p)
/I33/I1/I0/PM0      1.24766n     complex(-1.24765n, -2.74832p)
/I33/I0/NM9         358.96p     complex(-358.957p, -1.40184p)
/I33/I0/NM11        358.957p     complex(358.954p, 1.40182p)
/I33/I1/I1/I2/NM2     178.011p     complex(-160.898p, 76.1576p)
/I33/I1/I1/I2/NM1     178.011p     complex(160.898p, -76.1575p)
/I33/I1/I0/I2/NM2     101.771p     complex(67.2487p, 76.3869p)
/I33/I1/I0/I2/NM1     101.771p     complex(-67.2486p, -76.387p)
/I33/I1/I0/I2/NM32     83.8968p     complex(83.8604p, 2.47293p)
/I33/I1/I0/I2/NM0     83.8968p     complex(-83.8603p, -2.47289p)
/I33/I1/I1/I2/NM0     55.5677p     complex(-55.5163p, -2.3902p)
/I33/I1/I1/I2/NM32     55.5677p     complex(55.5163p, 2.39017p)
/I33/I0/NM12        22.2028p     complex(22.2027p, 53.3724f)
/I33/I0/NM10        22.195p     complex(-22.1949p, -53.3496f)
```

Modulated hbAC Analysis for an Oscillator

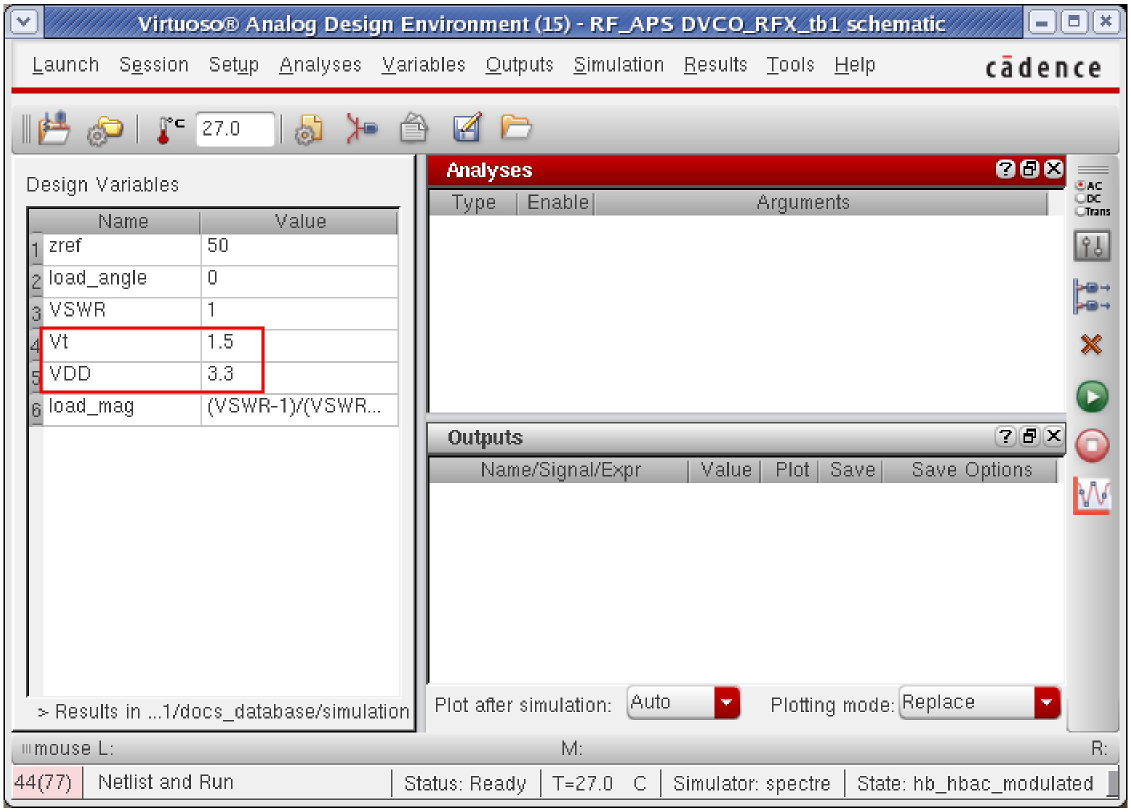
Modulated hbac analysis is usually applied to oscillators to measure AM to PM conversion from low-frequencies on the power supply (ripple) to the output near the first harmonic. It can also be applied to digital circuits for the same purpose. Although the shape of the transfer function is different, the steps are the same for both applications. Note that a relatively large number of harmonics and an oversample factor of at least 4 is needed for simulating the digital circuit.

In the circuit, make sure that any PAC magnitude terms are removed on the input source (if there is one), and set the PAC Magnitude to 1 volt in the source that is used as the power supply. If you do not want to save the change permanently, just select *Check - Current View* in the schematic.

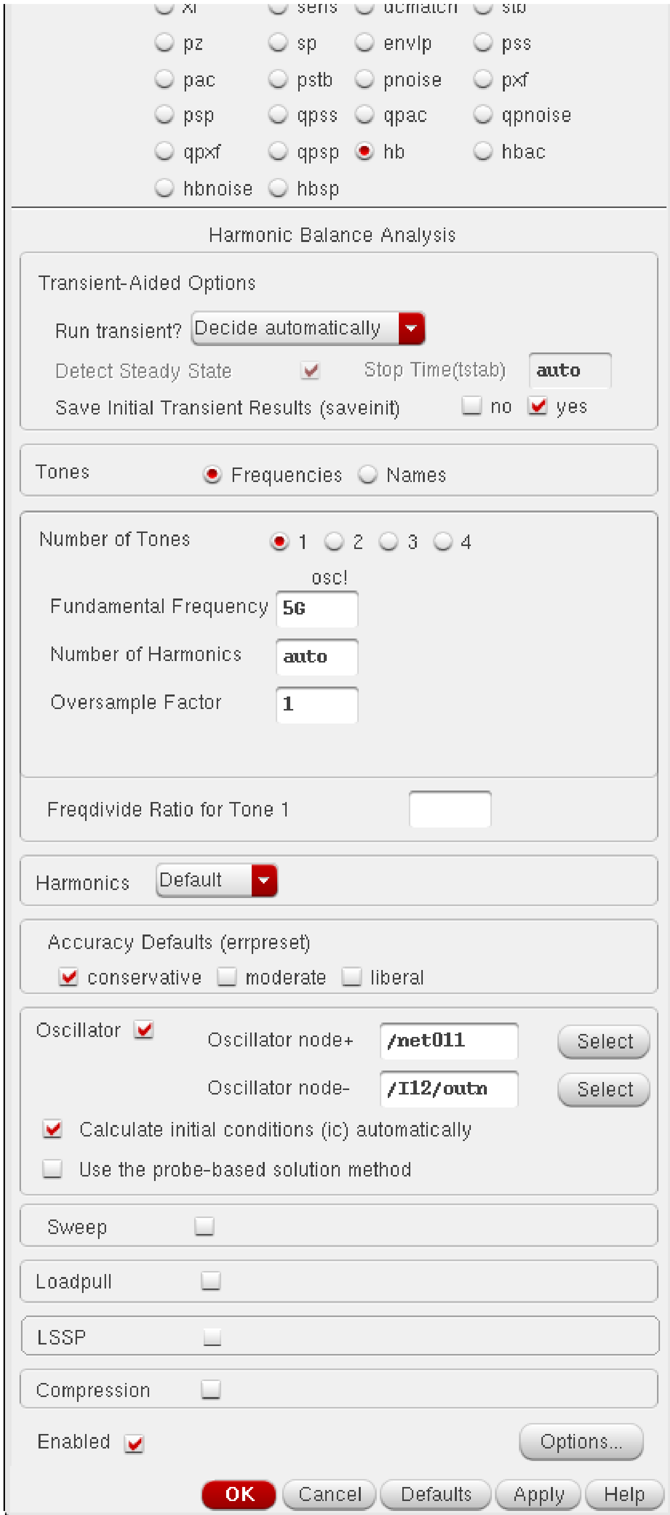
This example is for an oscillator. The circuit is shown below:



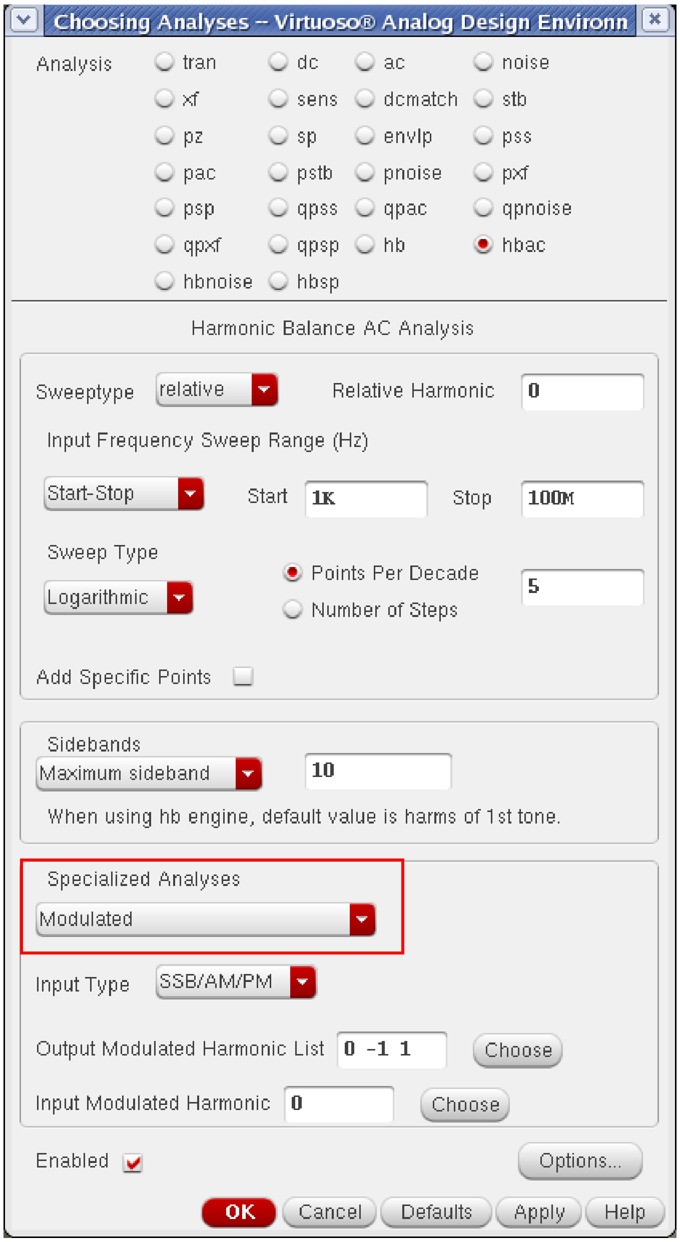
In ADE, set the values for the power supply voltage and for the tuning voltage.



Set up the harmonic balance analysis For more information, see the Harmonic Balance section at the beginning of this chapter.

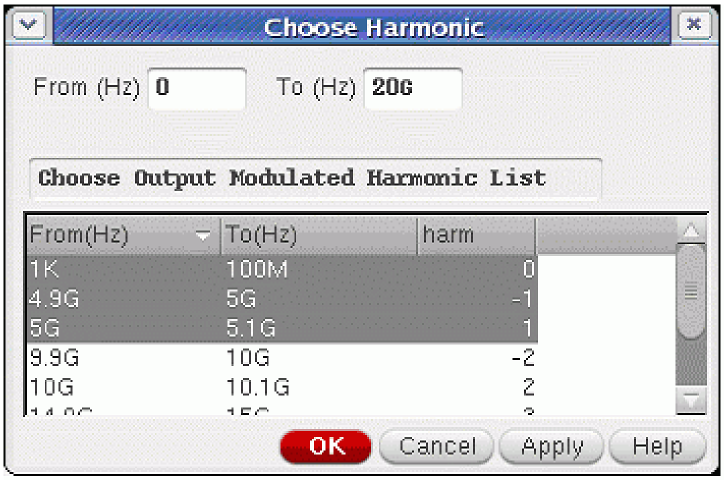


In the hbac *Choosing Analyses* form, first select *Modulated* in the *Specialized Analyses* section, as shown below.

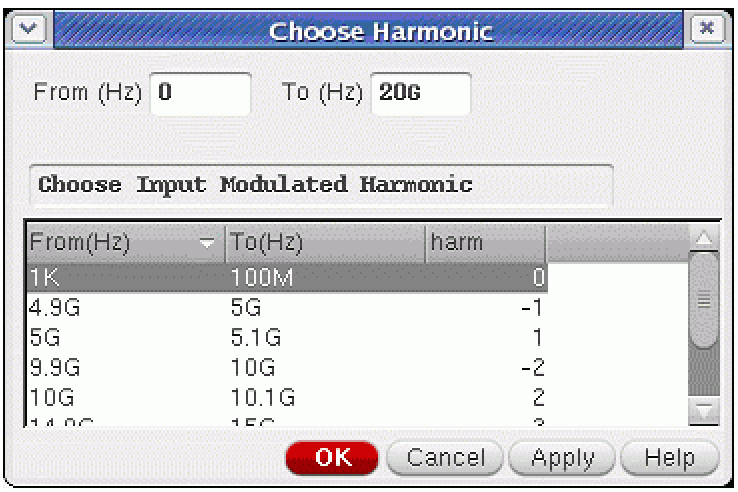


Note that this causes the *Sweep Type* to be set to *relative*. Since the gain from power supply ripple to the output is desired, the relative harmonic number is zero, and then the desired frequency range is entered. *Maximum sideband* can be left blank, or it can match the hb harmonics, as shown above.

Next, the output harmonics can be set. Note that a list of harmonics can be entered. The easiest way is to click the *Choose* button to the right of the field and select the desired frequencies from a list in the *Choose Harmonic* window, as shown below.

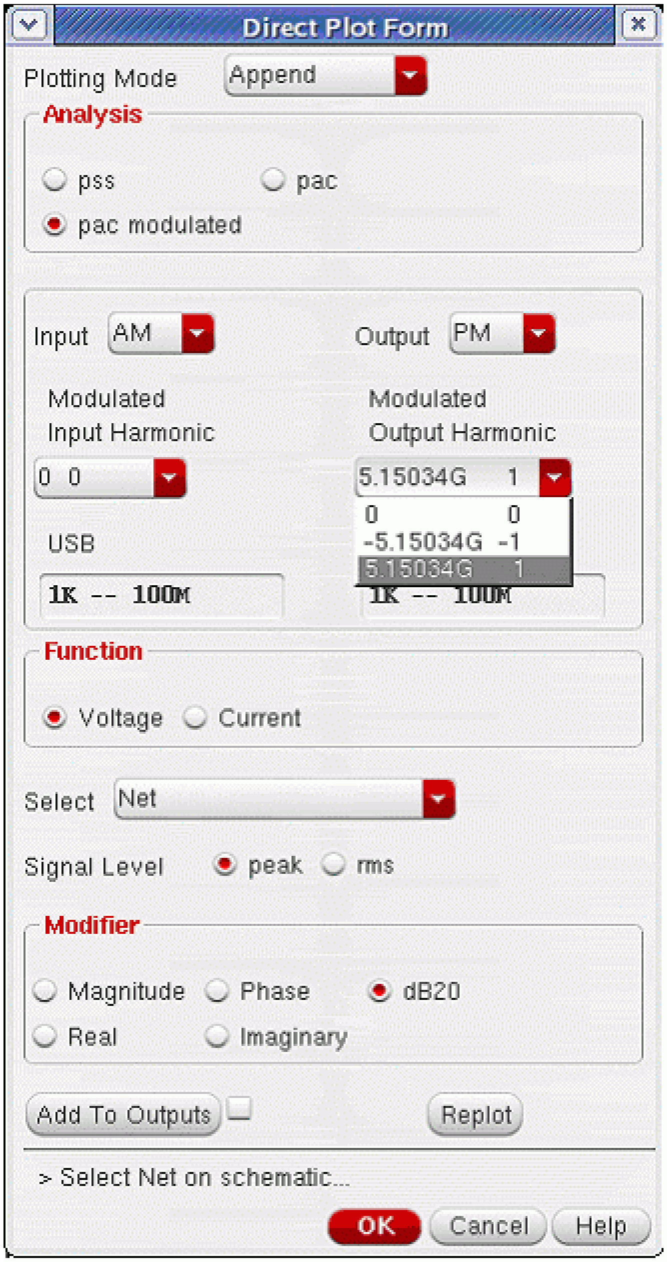


The same process is used to set the input frequency, but only one frequency range is allowed.

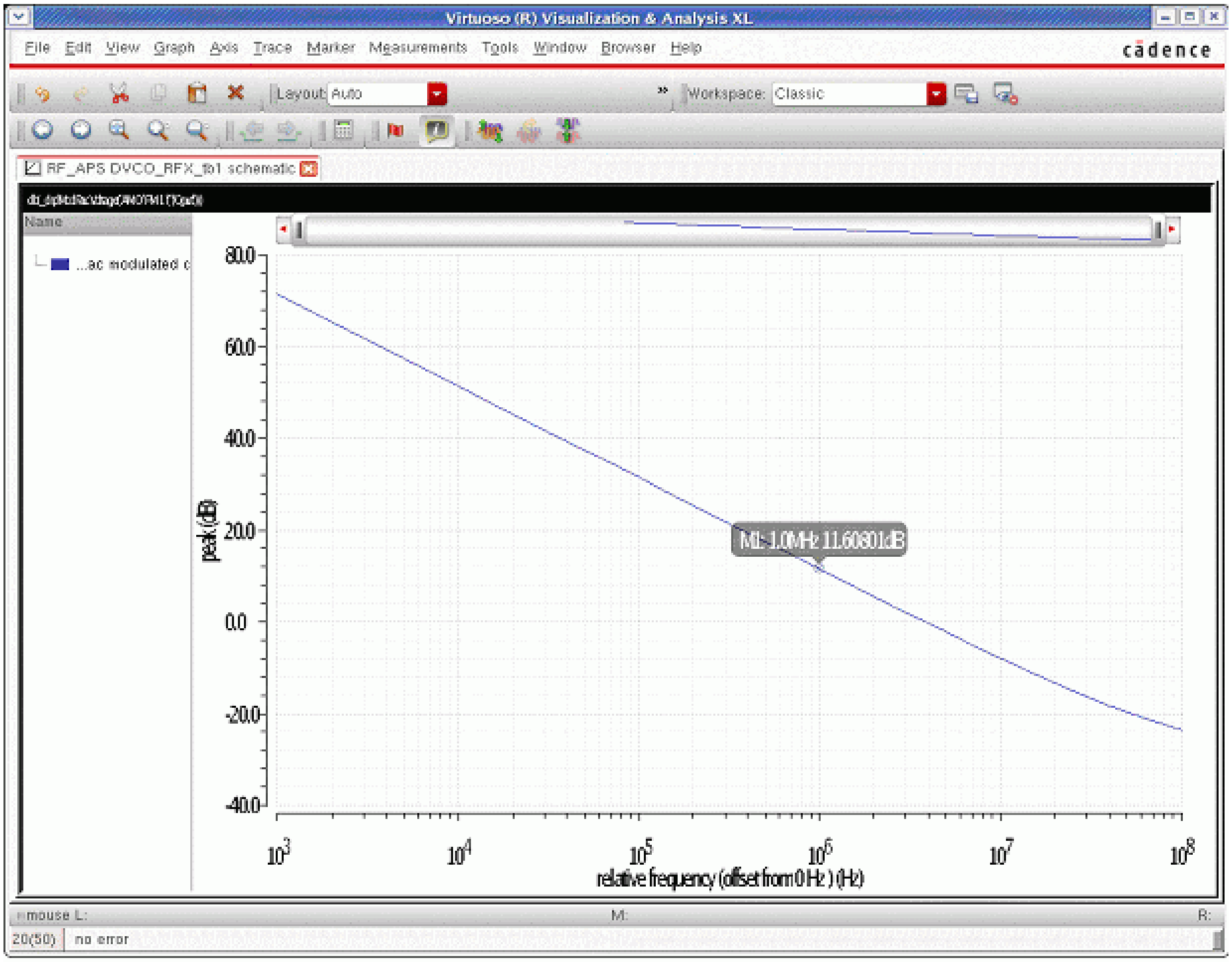


Once the analysis has been set up, run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form*.

The input defaults to `AM`. Select `PM` for the output. Since many output frequencies were calculated, you need to select the one you want to plot. In this example, the output just above the first harmonic is selected. Once this is done, select a net in the schematic.

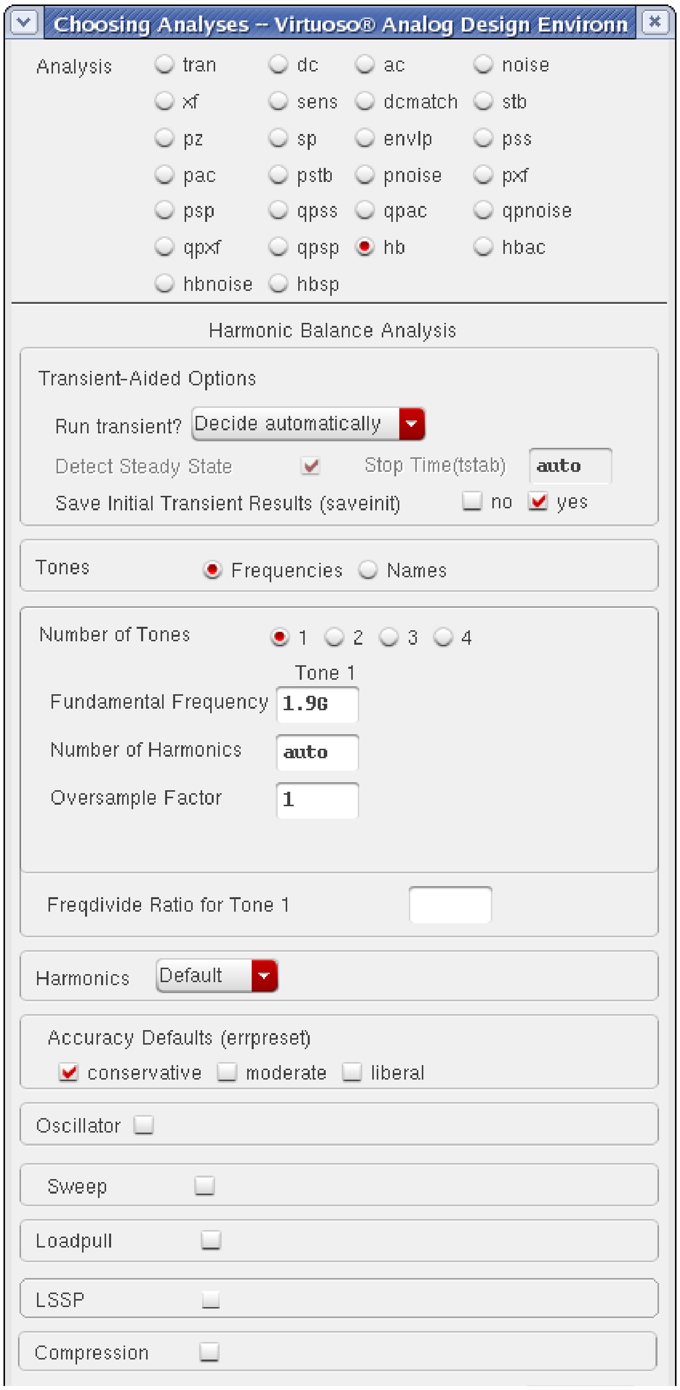


The output is displayed. To make a measurement at a specific frequency, position a marker. The measurement is in radians per volt of input on the supply that has PAC magnitude set.

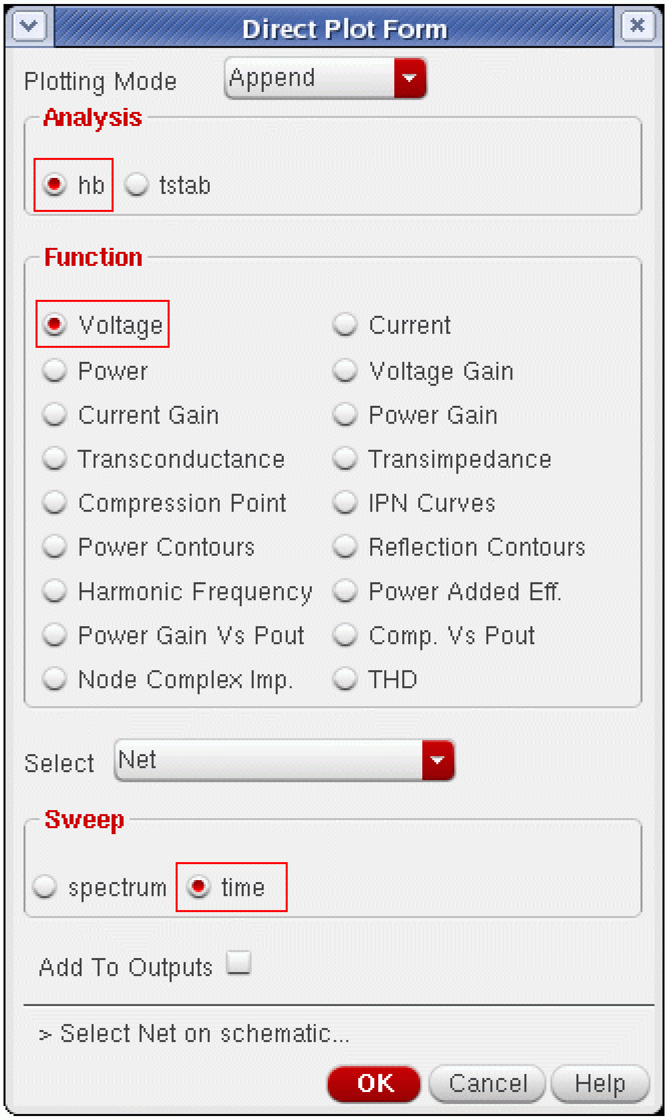


Sampled hbAC Analysis for a Mixer

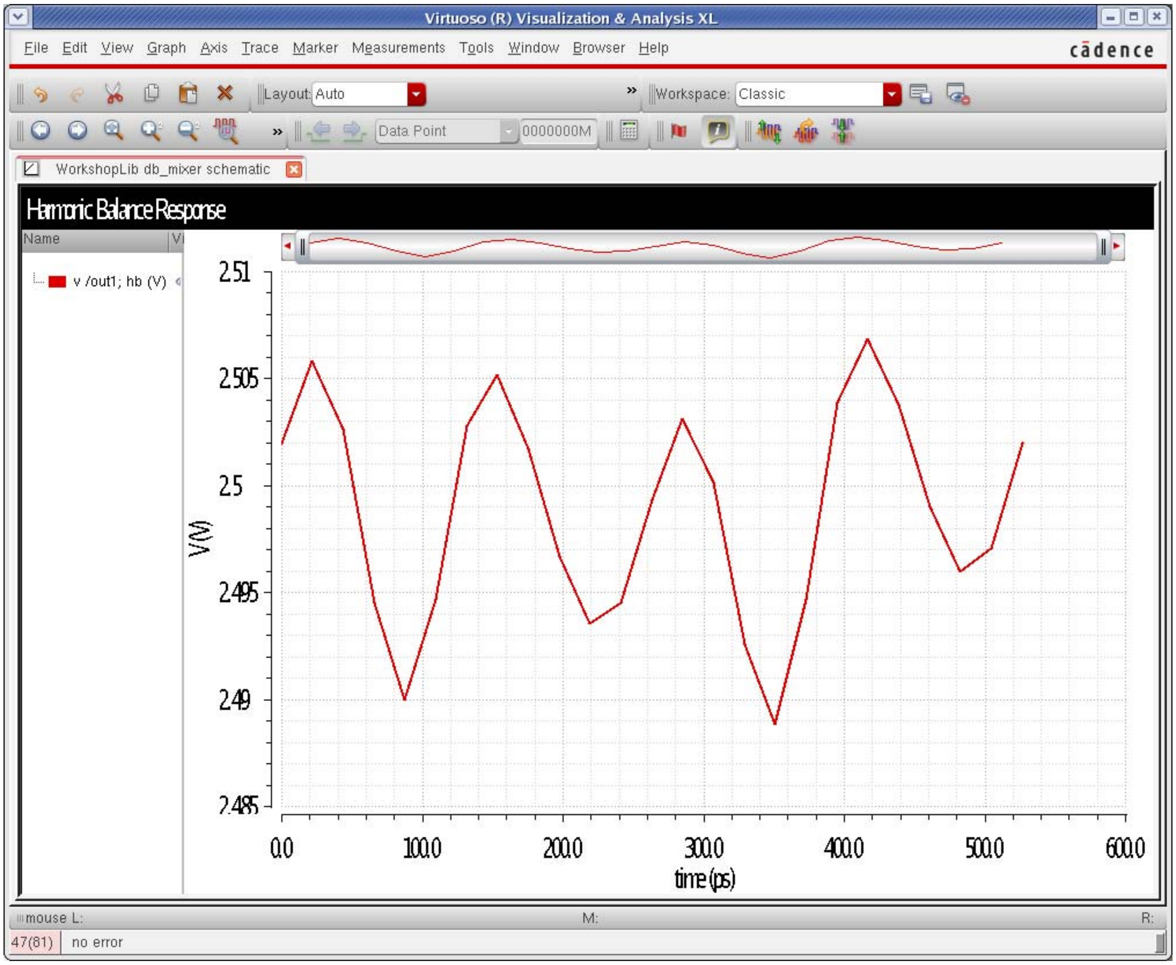
hbAC sampled is for measuring the conversion gain at a specific instantaneous time in the ifft waveform. The first step is to determine an appropriate threshold for the measurement by setting up the hb analysis, as shown below.



Now run the analysis. When the analysis completes, select *Results - Direct Plot - Main Form* in ADE and plot the time-domain waveform, as shown below.

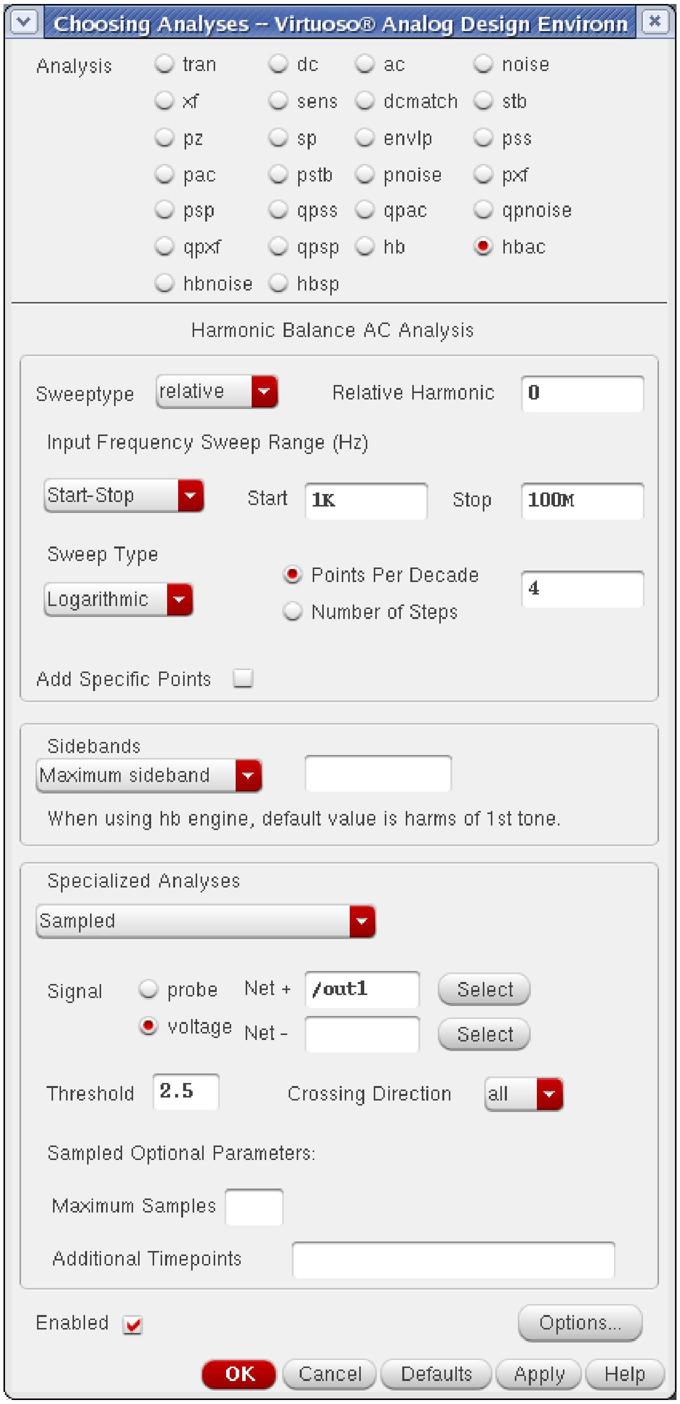


The waveform is shown below.



Note that 2.5 volts is a good threshold voltage.

Now set up the hbac *Choosing Analyses* form.

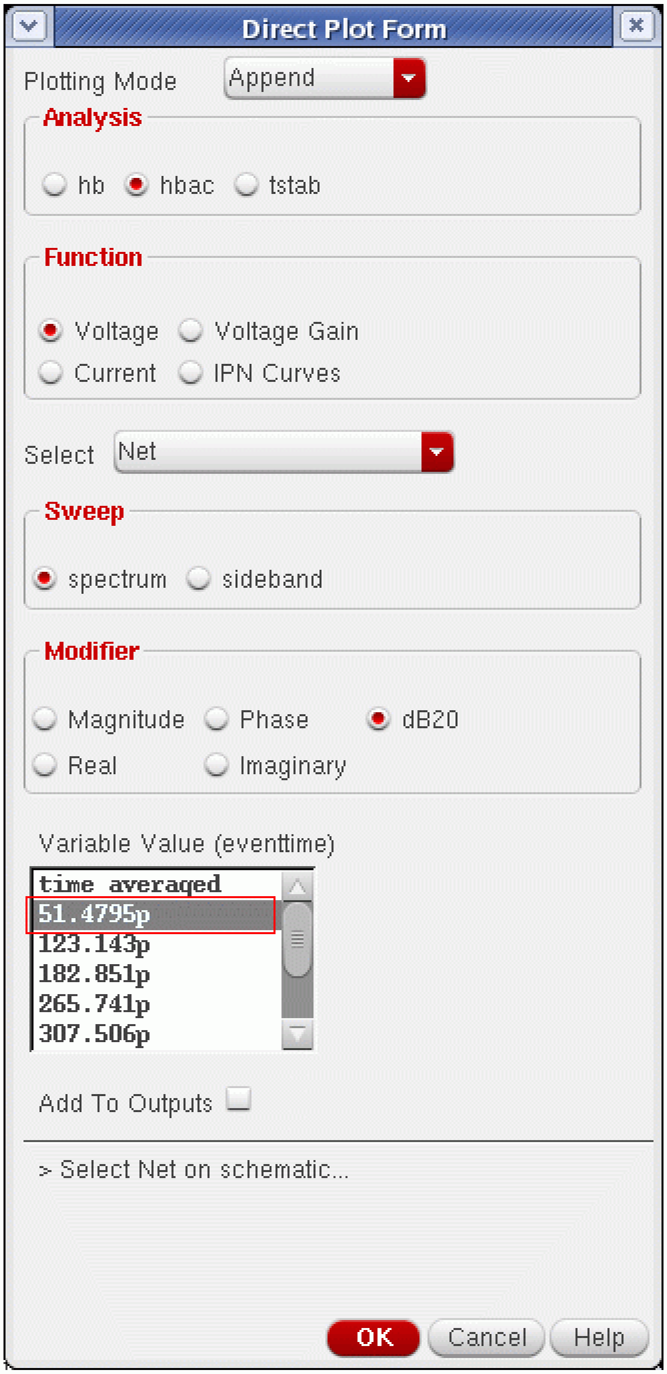


At the top of the form, set up the input frequency range like normal.

1. At the bottom of the form, select *Sampled*. Usually the gain to a node is desired, therefore, select *voltage*.

2. Click *Select* located to the right of the *Net+* field and select the node in the schematic.
3. Specify the desired threshold voltage based on the ifft from the hb analysis
4. Choose an option from the *Crossing Direction* drop-down list. In the example, *all* is selected.
5. If you need more than 16 crossing points, type the number you need in the *Maximum Samples* field.
6. If you want to sample at specific times from the ifft waveform, you can type in the times in the *Additional Timepoints* field.

7. Now run the simulation. When the simulation completes, open the *Direct Plot Form* and set it up, as shown below.



8. Select *hbac* results.

9. Select *Voltage*.

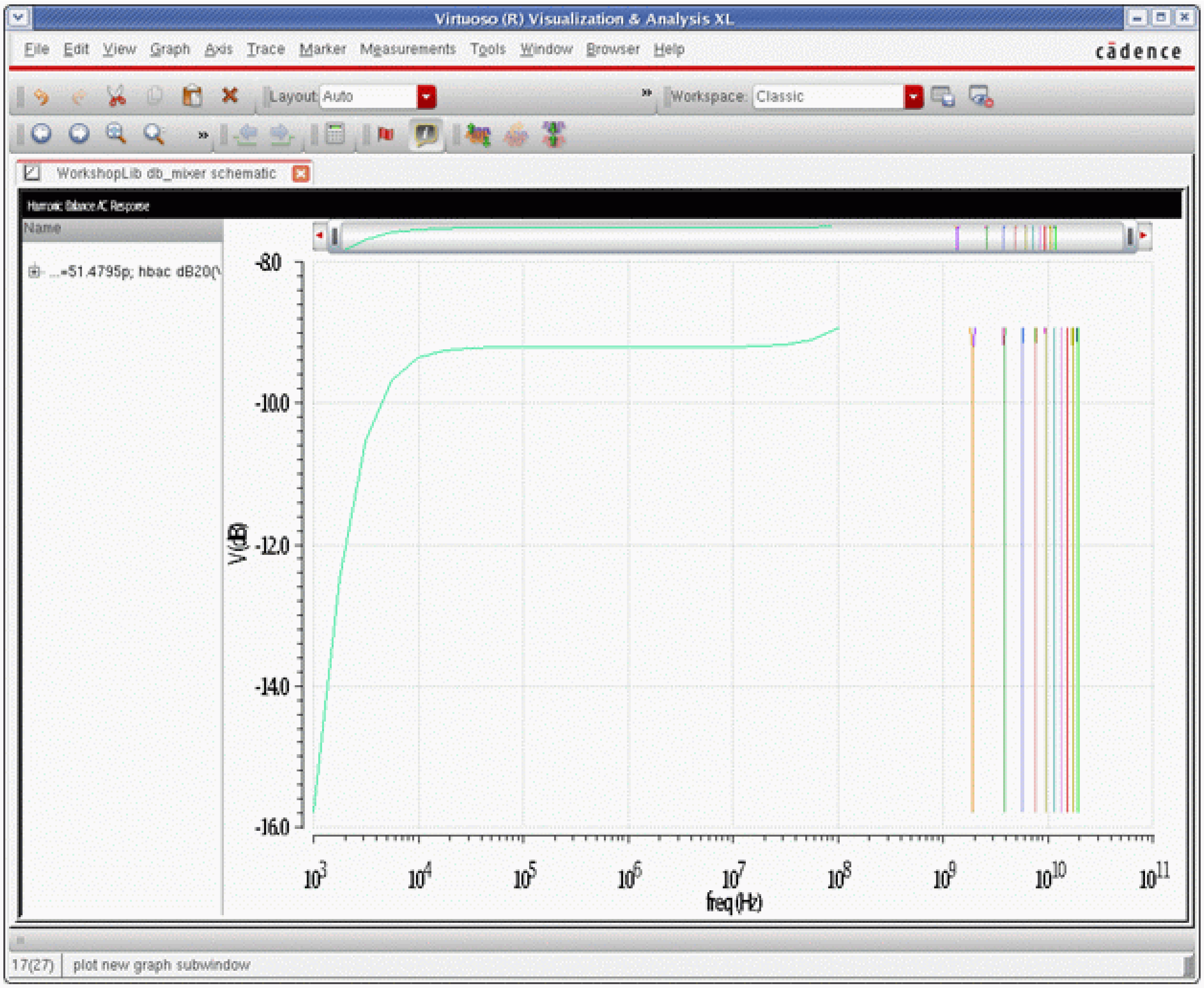
10. Select *spectrum*.

11. Select *dB20*.

12. Select the time you want from the *Variable Value (eventtime)* list.

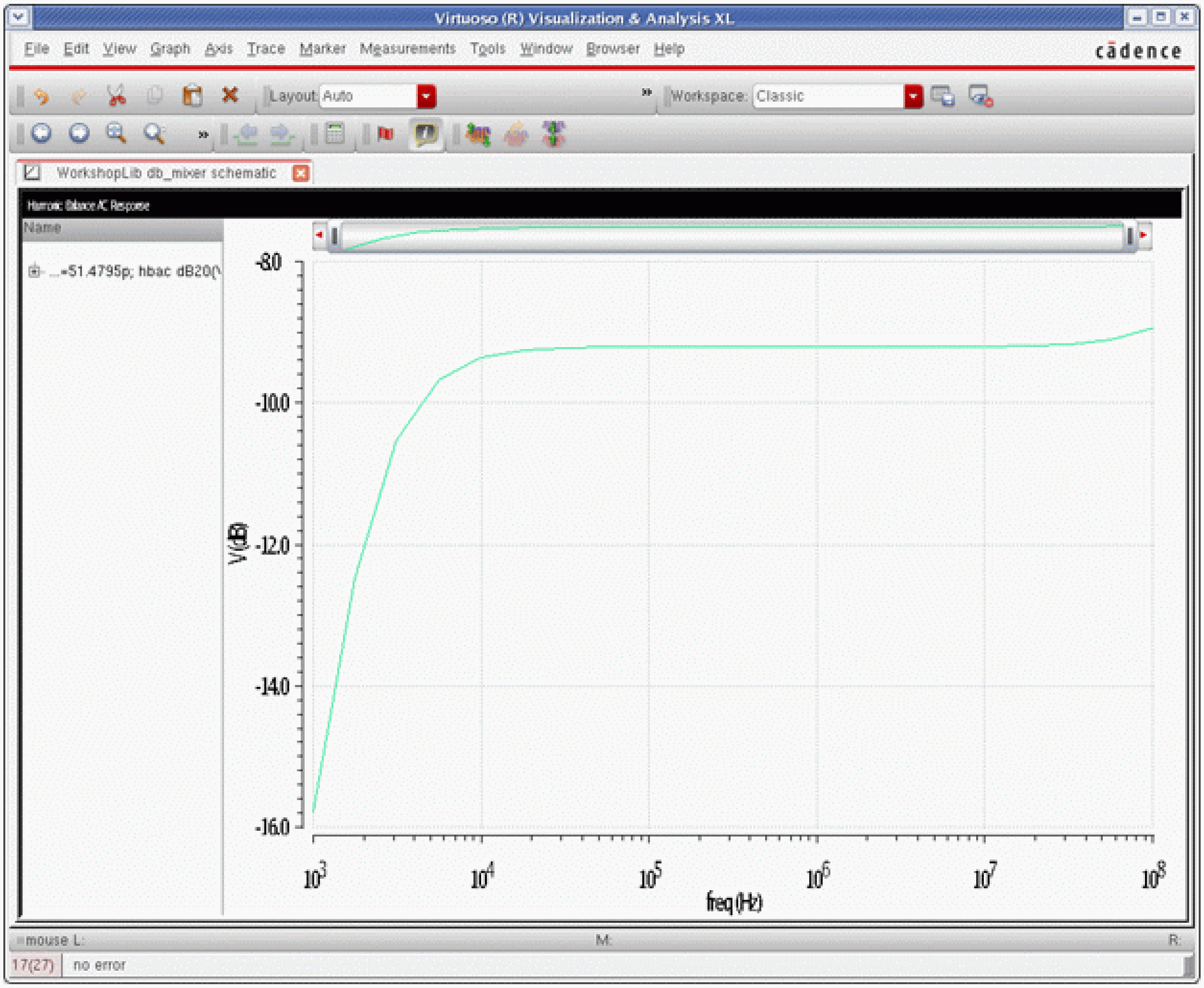
13. Select the output node in the schematic.

The waveform is plotted, as shown below.



Because the sweep was log spaced, the X Axis of the waveform tool is set to log. To zoom in to the frequency of interest, double-click one of the numbers on the X Axis, and type in the output frequency range of interest. This is shown below.

To measure the conversion gain to the output, move the tracking cursor to the output frequency and read the gain.



Harmonic Balance Noise Analysis (hbnoise)

Small-Signal Versus Large-Signal Analysis for Noise

In most cases, noise is a small-signal problem in the real world. Consider an amplifier. One approach is to calculate the small-signal gain from the noise sources to the output, then calculate the frequency response of the noise source, and then add up the noise power at the output. This is done in the linear noise analysis. This runs quickly because the gain calculation just takes the tangent at the DC operating point for everything in the circuit. This ignores large-signal effects like slew-rate limits and clipping. Because noise in most circuits is actually very small compared to the output levels, this approach works well. Because it is a linear approach, frequency translations cannot be calculated.

When the input to the amplifier gets large enough that it starts to create nonlinearity, the frequency translations in the circuit need to be taken into account. Noise near the harmonics can mix up or down to the output frequency. In this case, the large signal needs to be applied in a large-signal analysis like harmonic balance (hb) where the nonlinearity and harmonic levels can be calculated. After the hb runs, hbnoise runs and is similar to linear noise except the frequency translations of the circuit are taken into account. Instead of calculating a single gain from the noise source to the output, both the amplitude of the noise and the gain for the noise sources at the significant harmonics needs to be calculated, and each noise source now has several contributions at the output from several different noise frequencies. The basic strategy is to apply the signals that cause nonlinearity in the hb analysis, and follow that with an hbnoise calculation.

This same approach can be taken for mixers where the LO signal deliberately introduces nonlinearity for the purpose of frequency translation. The LO signal is applied to the mixer in the hb analysis, and hbnoise follows that.

This works with the mixer by itself, or in combination with other circuits, like an LNA, mixer, and a baseband amplifier.

Many designs need to tolerate a large amplitude blocking signal. Apply the LO and the blocking signal (and not the RF signal) in the hb analysis and follow with hbnoise. The blocker power can be swept in the hb analysis and hbnoise run after each sweep value for the blocker power to measure the desensitization.

For large-signal noise problems, or for systems that are not periodic, transient noise is provided. For more information on this, refer to the *Virtuoso® Spectre® Circuit Simulator and Accelerated Parallel Simulator User Guide*. This approach requires specifying a maximum noise frequency to be analyzed, which in RF systems is usually quite high. Two timepoints are created in the period of noisefmax. The transient needs to be run over many noise points in order to characterize the noise present in the circuit and needs post-processing to see the results in the frequency domain. This usually requires much more time than using small-signal noise analyses.

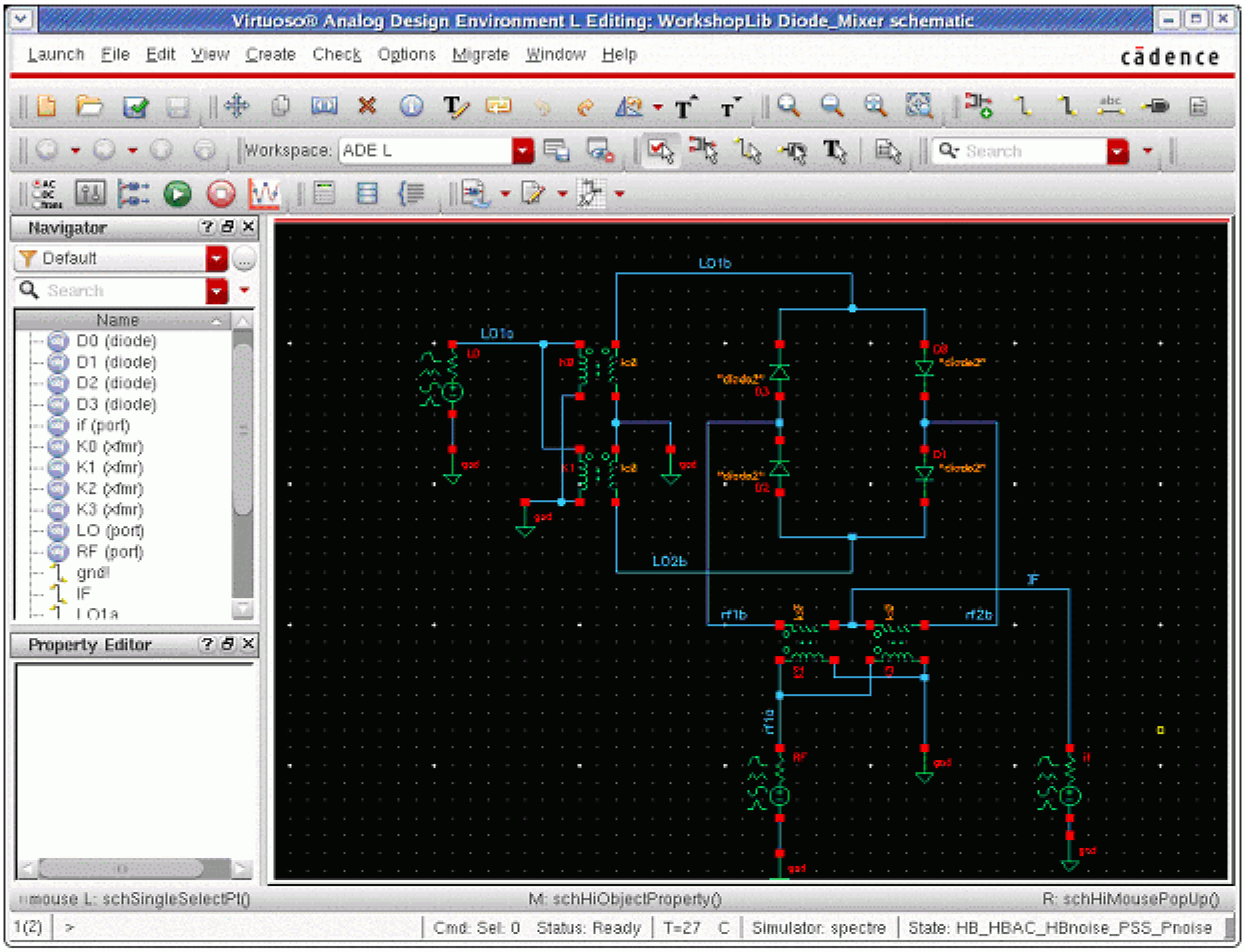
Overview of Simulation Capabilities

The fundamental quantity calculated in hbnoise is the output noise of the circuit with a large signal applied and including the noise translations caused by the nonlinearity of the circuit. In order to run hbnoise, a harmonic balance (hb) analysis needs to be run first in order to calculate the nonlinearity of the circuit and the amplitude of the harmonics that are present to mix with. Hbnoise takes this information from the hb analysis and then for each noise source it calculates the transfer functions from all the frequencies specified in the *Choosing Analyses* form. Then it calculates the frequency response of the noise source itself. Next, it calculates the total noise power at the output for that source at all the noise frequencies that contribute at the output. At the end, it adds up all the noise power from all the sources at all the frequencies. Input-referred calculations require the selection of the input frequency range by setting the reference sideband. This selection is necessary because there are many transfer functions from the input to the output caused by mixing or aliasing from the different harmonics of the system. The passband frequency is needed for the noise figure calculations.

Note that hbnoise is a small-signal calculation that works for periodic systems. If the noise is large-signal, or the system is not periodic, transient noise should be used.

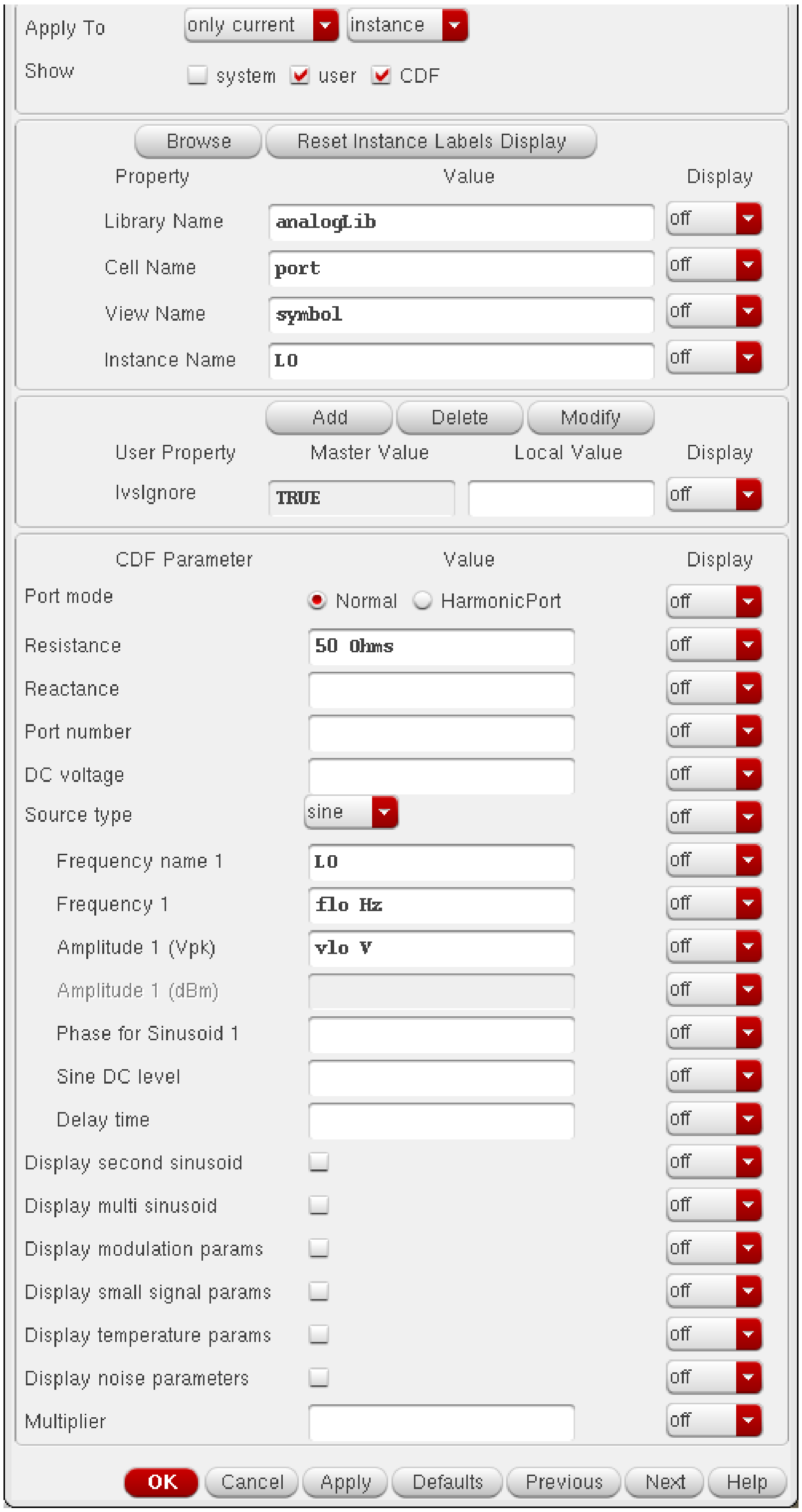
Example

Consider the double-balanced diode mixer below where a single-sideband noise figure measurement is desired.

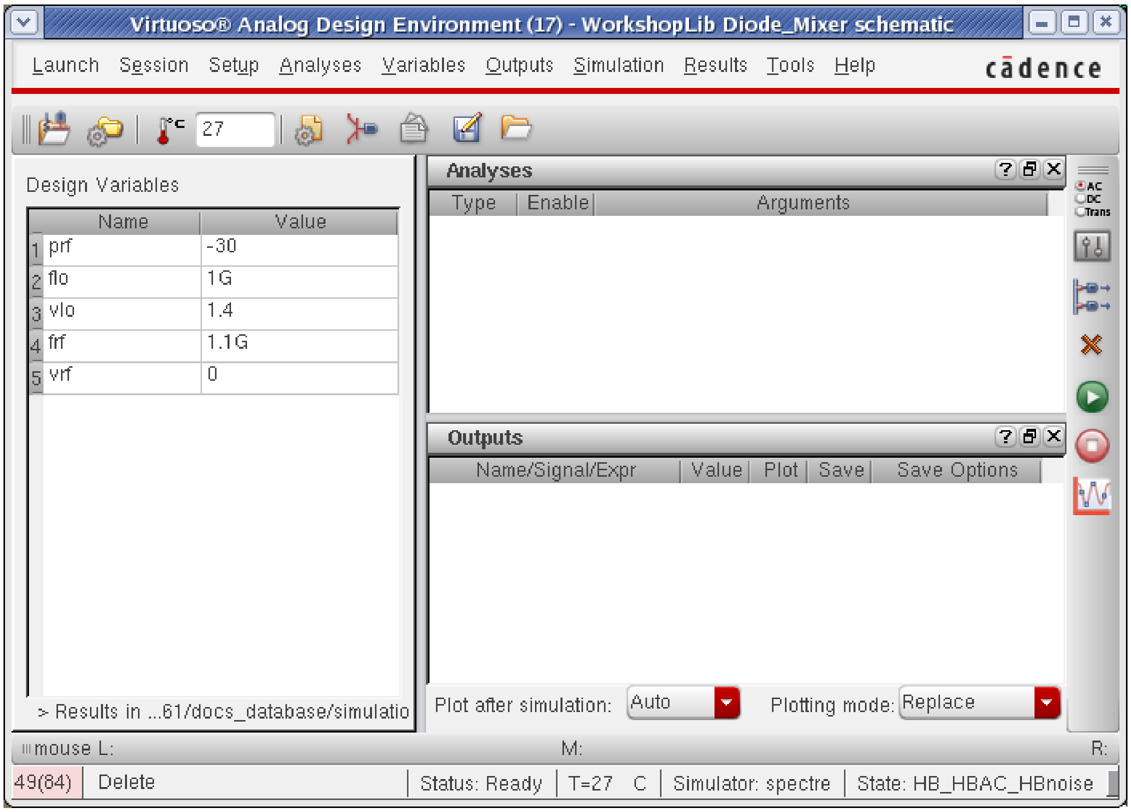


The input source has variables set to define the frequency and amplitude of the two large-signal outputs. This is done so that the frequency and the amplitude can be set in the ADE window without changing the schematic.

The property list for the input port is shown below.

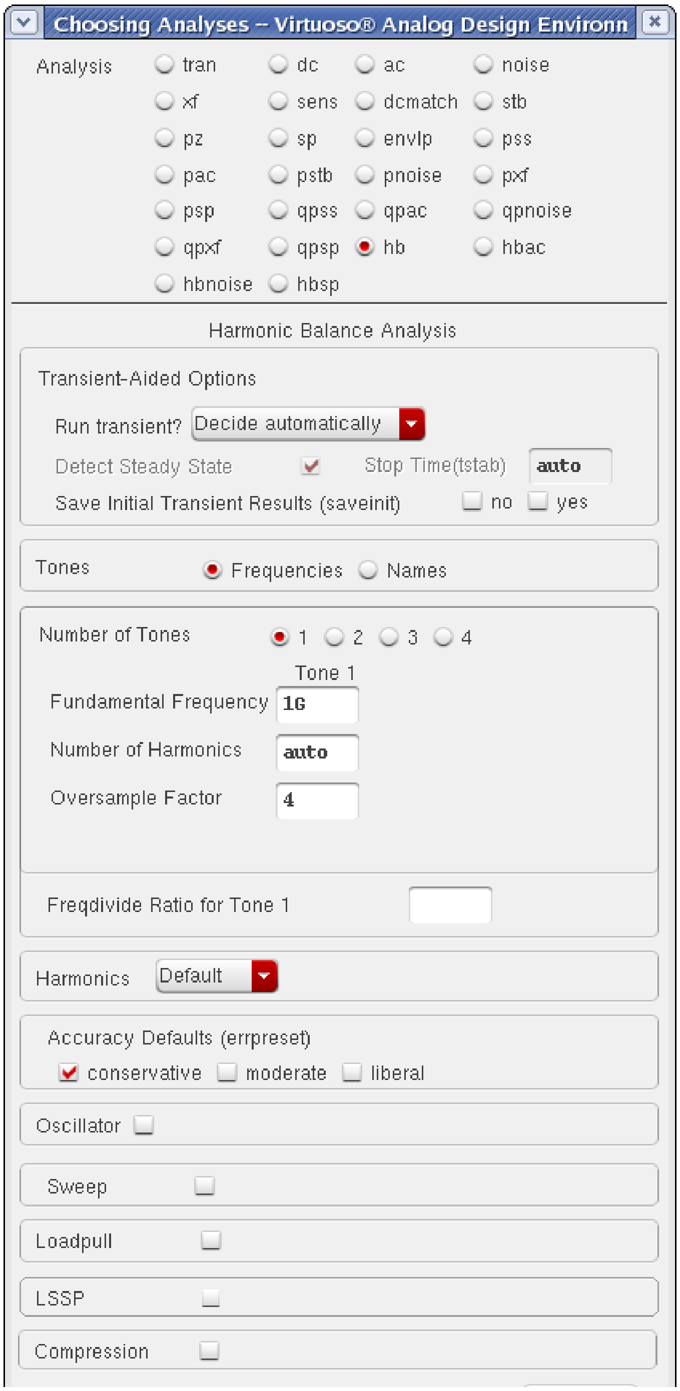


The variables have values set in the ADE variables section. The easiest way to get the variables list is to generate a netlist from the ADE or use *Variables - Copy From CellView*. This will cause all the variable names to be entered in the variables section. Just click in the value field for each variable and enter a value. Press *Enter* when you are done. In the figure below, the LO frequency set by the variable flo is 1GHz. The variable `vrf` which sets the linear amplitude of the large-signal RF inputs is zero. This disables the large-signal waveform. This circuit just has the signal that causes the frequency translations to occur (the LO signal) applied. Note that 1.4 volts peak is +13 dBm, which is a common drive level for a passive diode mixer.



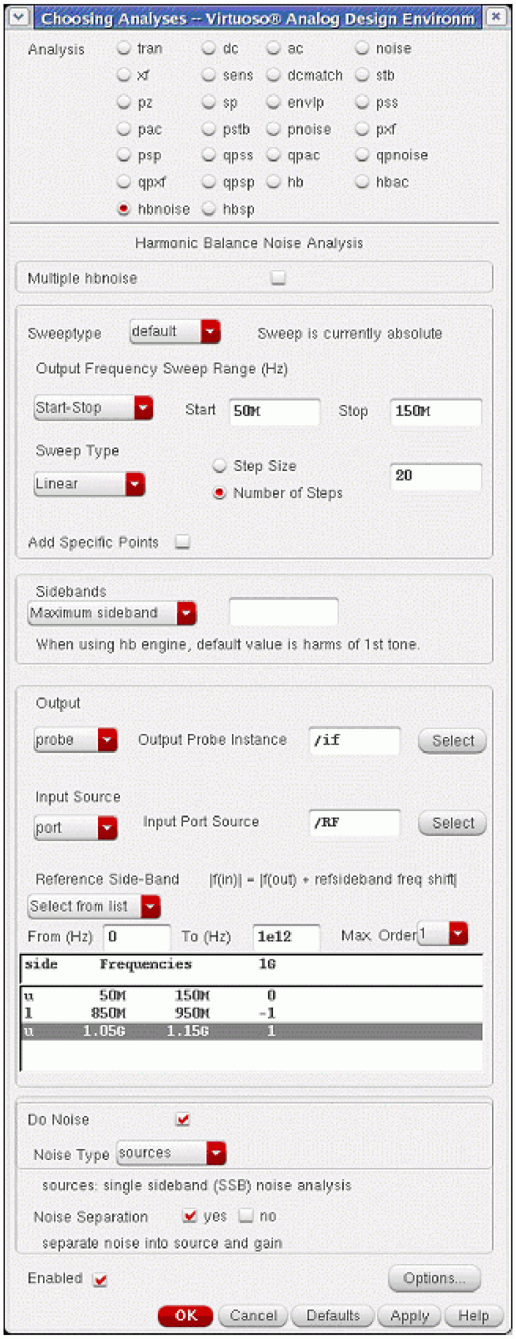
The hb analysis needs to be run first in order to measure the amplitude of the harmonics and to characterize the nonlinearity of the system. From this, the noise folding from all the noise sources can be calculated in the hbnoise analysis. For more information on the setup of the hb form, please refer to the hb large-signal analysis section.

In the form below, the 1GHz LO is solved using automatic harmonics.



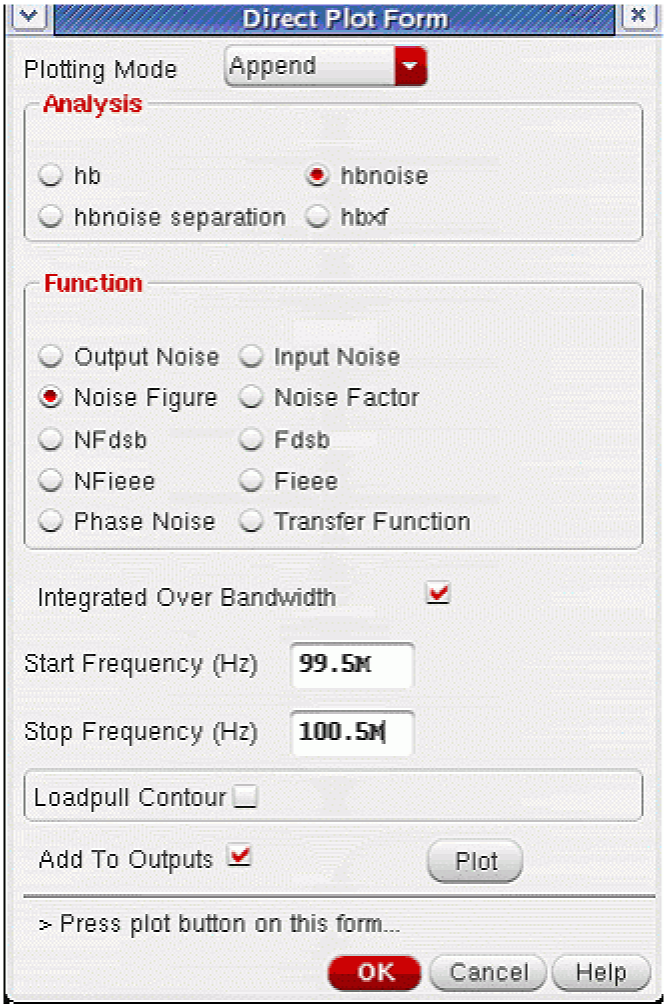
The noise output frequency and the number of noise translations to be calculated are set up in the hbnoise form. In the example below, the output frequency is near baseband. Leaving the *Maximum sideband* field blank causes noise translations from all the harmonics in the hb analysis to be calculated.

The reference sideband selects the passband frequency for the noise figure calculation.



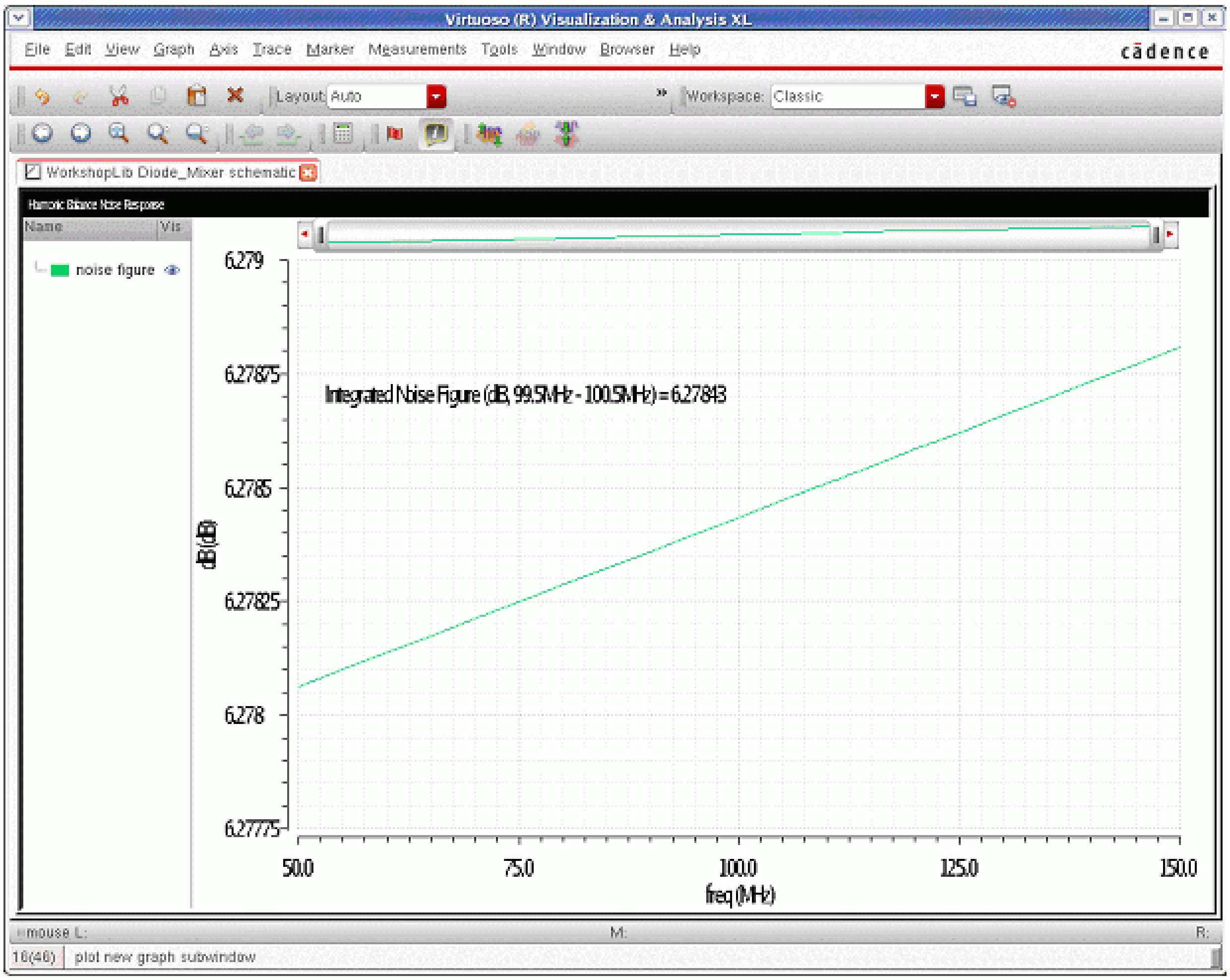
Once both forms are set up, the simulation is run. To plot outputs, select *Results - Direct Plot - Main Form* in the ADE window. The top of the form lists all the analyses that were run.

Select *hbnoise* where the noise figure is calculated. Next, select *NF* because that is what is desired to plot. If the noise figure over a bandwidth is desired, select *Integrated Over Bandwidth*, and specify the frequency range you desire for the measurement.



When you are done, click *Plot*.

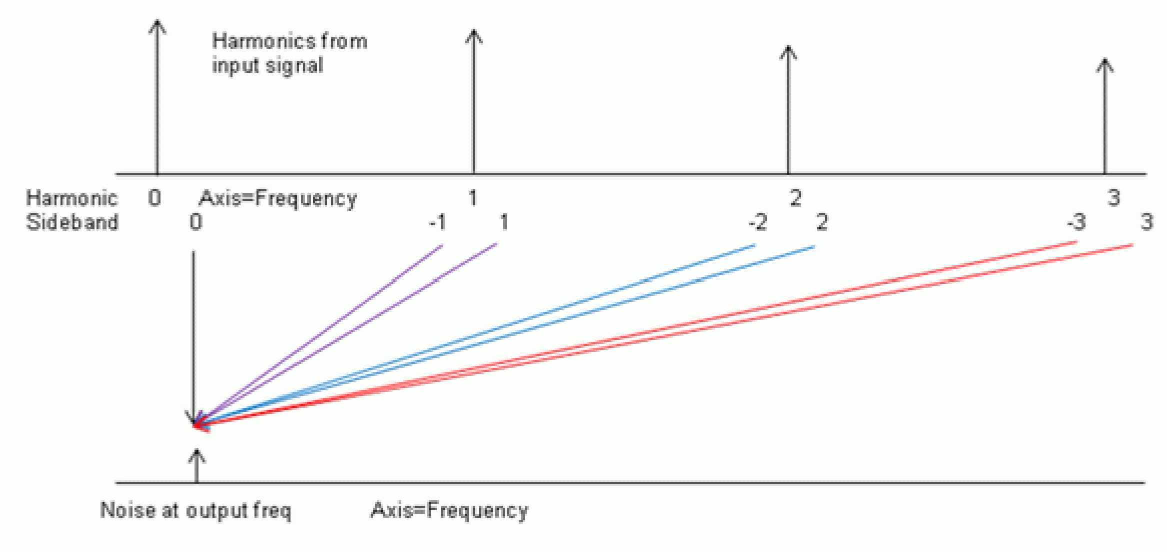
The waveform window plots the noise figure and the noise figure over the range you specified to be displayed, as shown below.



Noise Output Near Zero Frequency

First, the hb analysis is run to calculate the nonlinearity and the harmonics. Then hbnoise is run to calculate the noise. In all systems there is a maximum number of noise translations that need to be taken into account. Either the system harmonics become so small that the amount of noise translation becomes minimal or the devices themselves have finite noise bandwidths.

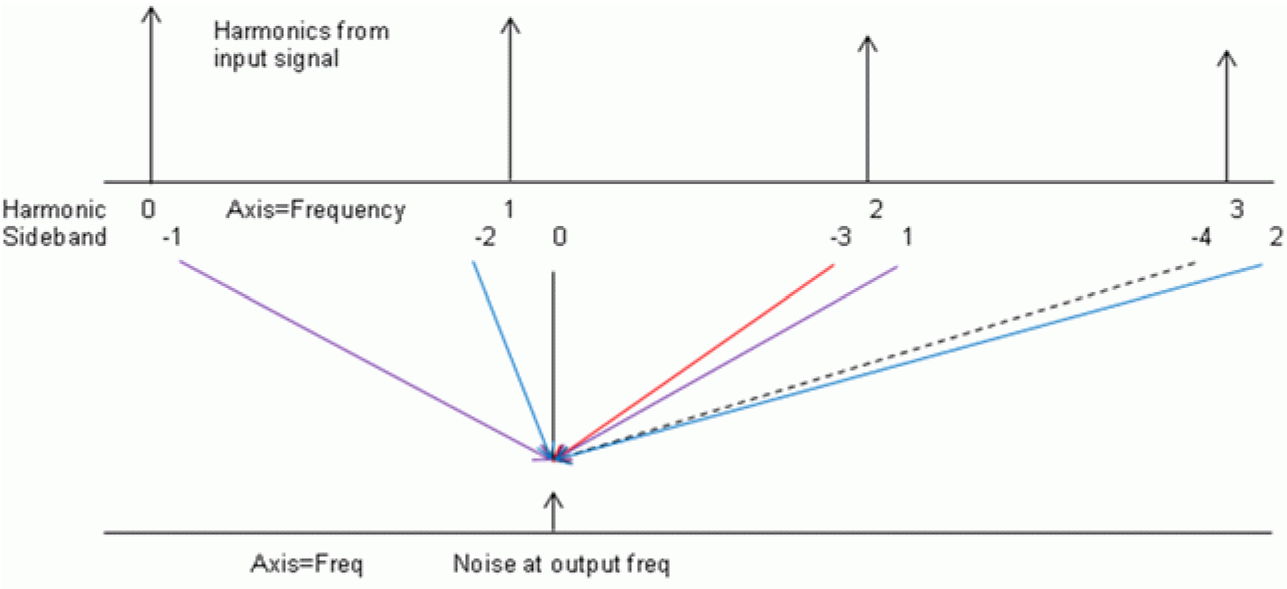
These noise calculations from mixing or aliasing with the different harmonics are called sidebands.



In the diagram above the X axis is frequency. The top represents the harmonics that are calculated by the hb analysis. The bottom represents the noise output frequency. The arrows show the various mixing paths to the output frequency. Hbnoise calculates the noise contribution at the output with all these mixing paths in place. Black is the noise that occurs without mixing. Purple is the noise that mixes with the first harmonic. Blue is the noise that mixes with the second harmonic. Red is the noise that mixes with the third harmonic.

In hbnoise, the output frequency (which is usually a range of frequencies) is specified along with the highest harmonic to calculate the mixing products.

Noise Output Near The First Harmonic



Here the noise output frequency is shifted up to just above the first harmonic frequency. Black is the noise that occurs without mixing. Purple is the noise that mixes with the first harmonic. Blue is the noise that mixes with the second harmonic. Note that the noise frequency just below the first harmonic mixes with the second harmonic to provide the output just above the first harmonic. Red is the noise that mixes with the third harmonic. Note that the second term that mixes with the third harmonic is just above the fourth harmonic of the hb analysis which is not shown. The noise term shown with a dotted line is mixing with the fourth harmonic of the system.

Frequency Sweep

Since the fundamental quantity that is calculated in hbnoise is the output noise, the frequency range defined in the *Choosing Analyses* form defines an output frequency range to calculate the noise for. For driven circuits, the default is to sweep an absolute frequency range. Oscillators will be covered later.

For an up-conversion system, sometimes a log frequency range is desired near one of the harmonics of the system. In that case, select relative sweep, and specify a harmonic number. In the output frequency range, type in the desired offset frequency from that harmonic. Note that this is only available for frequencies above the desired harmonic number because to get this below the harmonic requires taking the log of a negative number, which does not exist.

When automatic is selected for sweep type, 100 total points will be calculated. In most cases, fewer points are needed so it is usually better to select linear or log and then specify the sweep you desire. Typically 3 to 5 points per decade or about 20 total points are needed.

Maximum Sideband

There is a property in the *Choosing Analyses* form called *Maximum sideband* that is used to define how many mixing products should be calculated. If *Maximum sideband* is zero, the noise analysis will only contain the noise that appears without mixing or aliasing of any type. This is not equivalent to a linear noise analysis because hbnoise takes into account the instantaneously varying nature of the noise because of the large signal being present in the hb analysis.

When *Maximum sideband* is one, then the noise without frequency translation and the noise that mixes or aliases with the first harmonic is present in the noise analysis. When *Maximum sideband* is 10, the noise analysis includes noise mixing through the 10th harmonic of the harmonic balance analysis. Usually the number of harmonics in the hb analysis and the maximum number of sidebands should agree. This is easily accomplished by leaving the *Maximum sideband* field blank in the *Choosing Analyses* form in ADE.

Setting Harmonics and Sidebands

The process of setting harmonics and sidebands is similar to setting the maximum number of harmonics for the harmonic balance simulation. Start with an estimate based on the harmonic content of the input signal(s) and the nonlinearity of the circuit. Then run a simulation. Raise harmonics and sidebands and run again. If the noise measurement did not change, then the original number of harmonics and sidebands might be able to be reduced. If it did change, then more are needed. Getting a stable hbnoise output may take more or fewer harmonics and sidebands than the harmonic balance simulation by itself.

Noise Separation

In addition to the total output noise, the individual noise contributions can be plotted if noise separation is selected in the *Choosing Analyses* form. More information about the noise separation will be provided later. Noise separation is available when the noise type is set to sources only. The things that can be plotted are:

1. Total noise at the output from each individual input frequency.
2. The noise at the output from each instance name with all noise mechanisms included for mixing from an individual input frequency.

3. The noise at the output from each instance name with all noise mechanisms broken out separately for mixing from an individual input frequency.
4. The current noise at the noise instance for each instance name with all noise mechanisms included for mixing from an individual input frequency.
5. The noise current in the instance from each instance name with all noise mechanisms broken out separately for mixing from an individual input frequency.
6. The gain to the output from each individual instance with all the individual noise sources broken out separately for mixing from an individual input frequency.

Multiple hbnoise

Some systems contain multiple points in the circuit where the noise measurement is desired. When the *Noise Type* is *sources*, multiple hbnoise analyses are allowed. At the top of the hbnoise *Choosing Analyses* form, an option called *Multiple hbnoise* is provided. When this option is selected, up to six hbnoise analyses can be made after a single hb analysis.

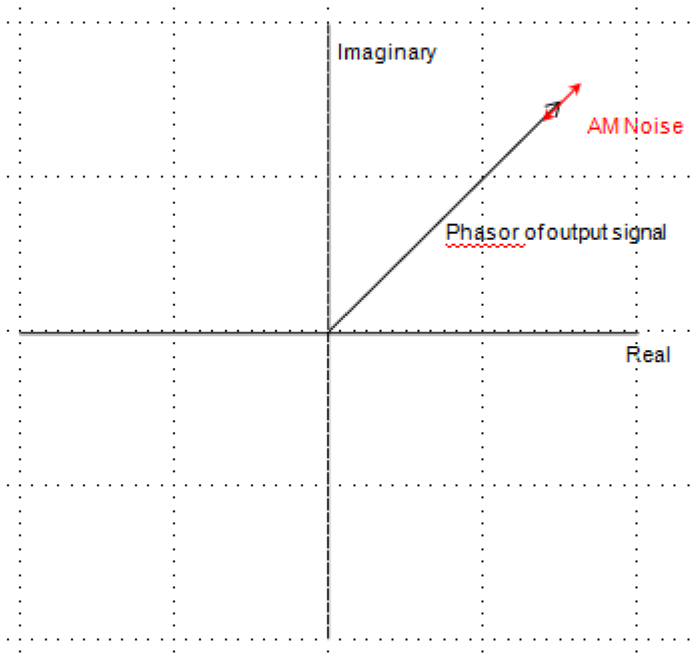
Hbnoise after Multi-Tone hb

When multiple input frequencies are present in the circuit, there are more frequencies in the large-signal analysis to mix with. Not only are the harmonics of both tones present, the intermodulation products are present as well. In this case, the noise analysis becomes more involved because the transfer functions from all the frequencies present in the large-signal have to be calculated for each noise source. Also, the meaning of *Maximum sideband* slightly changes. If 5 were selected, all the noise translations would be calculated through the 5th harmonic of the tone with tstab enabled. In the harmonic balance *Choosing Analyses* form it is *Tone1* when *Frequencies* is selected, and the tone with tstab set to *yes* in the *Names* list. It is recommended that the largest number of harmonics for any tone in the hb list be used as maximum sideband. This can easily be accomplished by leaving the number of sidebands field being left blank.

Note that noise type=sources and noisetype=modulated are the only available choices for a 2-tone driven circuit.

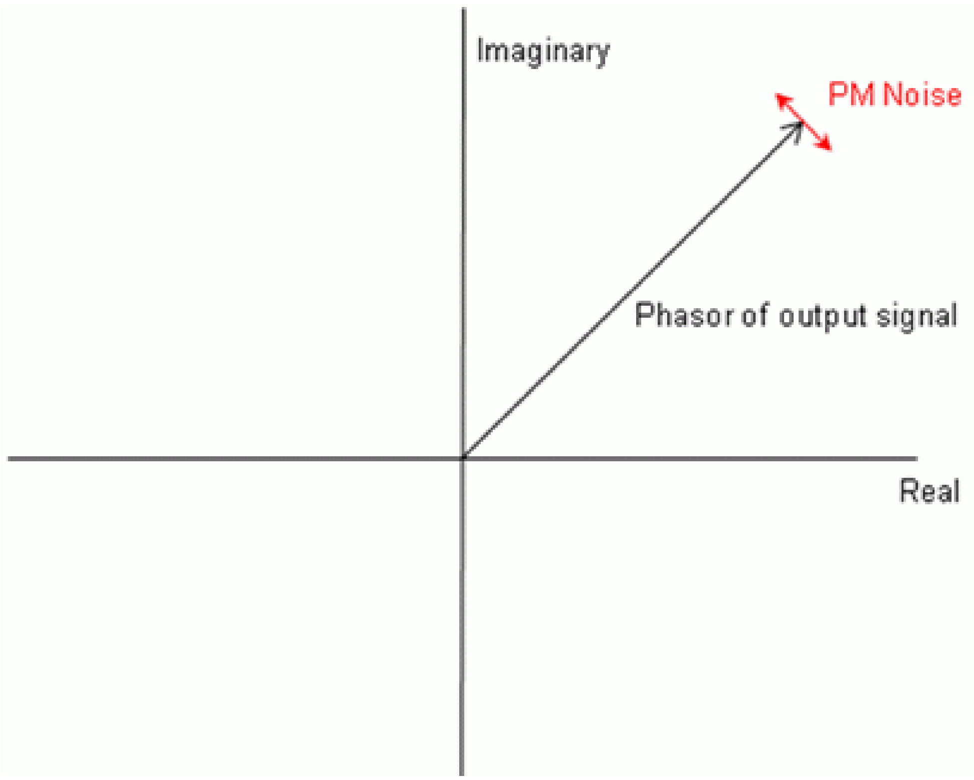
Modulated for Driven Circuits and Oscillators (AM and PM Noise)

Modulated is provided so that the AM and PM components of noise can be separated out. When the *Noise Type* is *sources*, the total noise (AM + PM) is calculated. To illustrate AM noise, see the phasor diagram below.



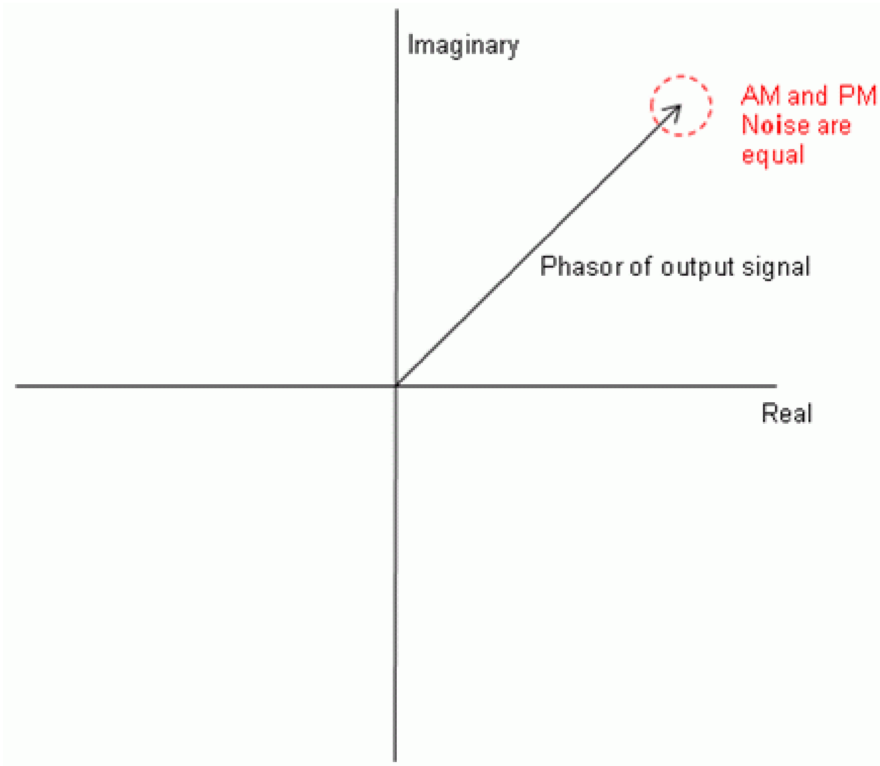
This diagram illustrates a periodic signal with AM noise. The black phasor is the output signal, and AM noise is a random vector that varies the amplitude of the signal.

PM noise is shown in the phasor diagram below.



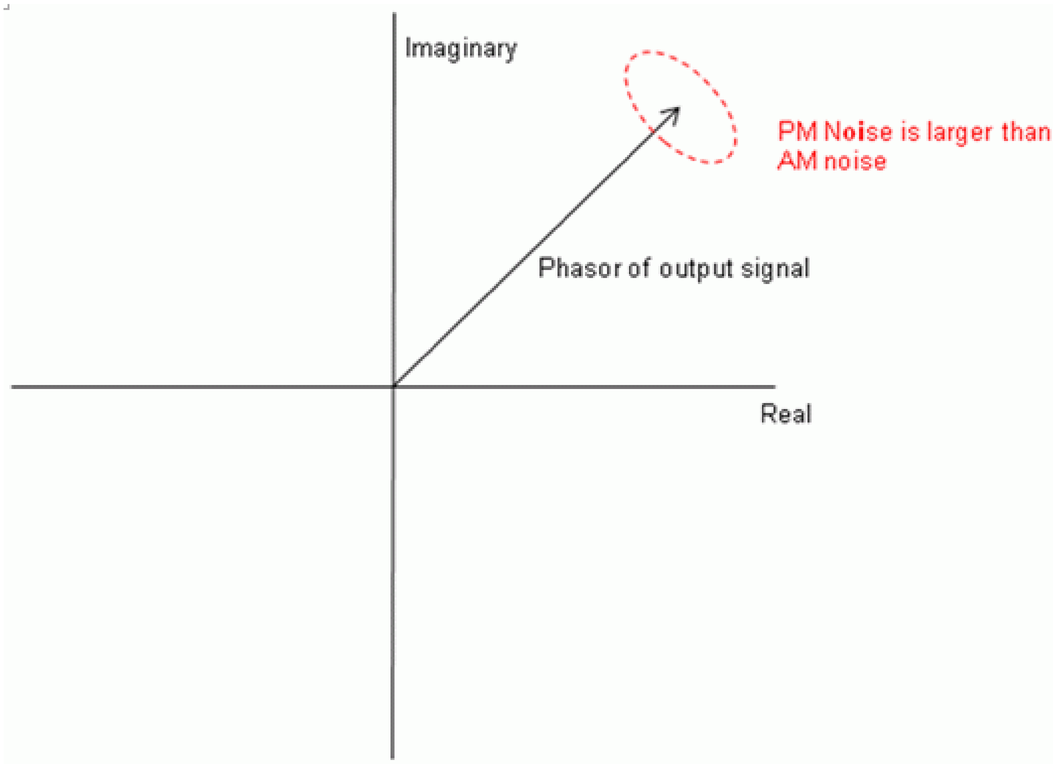
In this case, the amplitude is constant, but the noise is a random vector perpendicular to the signal phasor. The timing of the signal varies a bit from cycle to cycle.

Most circuits produce a combination of AM and PM noise. This concept is shown below.



In this case, a random vector at any angle and amplitude is produced that varies at each instant of time. If the uncertainty creates a round shape, the AM component and the PM component are the same size.

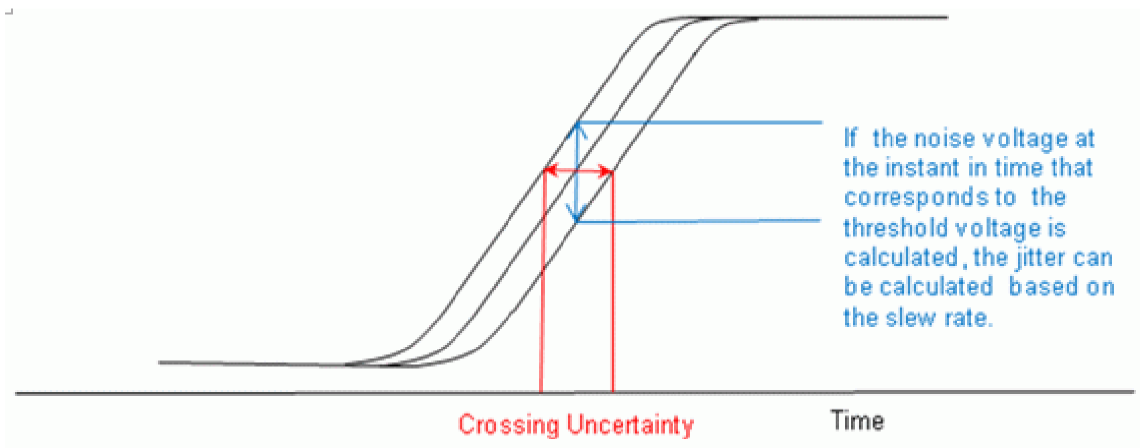
In many circuits, AM or PM noise may dominate. If the PM noise dominates, the circle turns into an ellipse that is large in the perpendicular direction than in the parallel direction. This is shown below.



Modulated also provides the total noise in the frequencies above the output frequency (the USB in the *Direct Plot Form*) and the noise below the output frequency (the LSB in the *Direct Plot Form*).

Jitter for Driven Circuits

Jitter is limited to single input frequency circuits. Jitter is usually applied to digital circuits where the average noise over the entire waveform is not appropriate. The noise in the high state and in the low state does not matter because the output voltage is so far from the voltage that triggers the load that it does not matter. The only time the noise matters is at the exact time where the trigger event is generated. Also, the noise in the low and high state is quite small because one device is completely off, and the other device is driven on. It is basically acting as a resistor. The device is not very good as an amplifier because the output is pulled hard into a rail. As the state changes, both devices are on, and they are relatively better amplifiers, so the noise is likely to be larger in the switching interval. In jitter for driven circuits, the voltage threshold for the noise measurement, and the selection of the rising edge, the falling edge, or both is made. At the threshold voltage specified, a noise measurement is made at that instant. Once the noise voltage is known, the timing uncertainty can be determined from the slew rate. Since the RMS noise voltage is calculated, the timing uncertainty is one standard deviation from the ideal crossing time. This is illustrated below.



Timedomain for Driven Circuits

Timedomain is available for single input frequency circuits only. This is a variation of the jitter analysis explained in the previous section. In this analysis, instead of specifying a voltage, a list of times from the time-domain waveform is supplied. A noise analysis is performed at those instants in time, and the noise voltage is calculated at each timepoint. The conversion from noise voltage to time-domain jitter is a manual step.

Oscillators

Sources

For oscillators, the hbnoise frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the hb form, you should specify the approximate frequency of oscillation for the oscillator itself, and set the freqdivide property to the divider ratio. This has the effect of dropping the first harmonic frequency by the divide ratio. In the hbnoise form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of relative is to take the frequency of harmonic number specified and add to it the frequencies specified in the Choosing Analyses form. If the oscillator had a 1GHz output, and the hbnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

Oscillators do not need noise figure calculations and are not typically driven, so the input source should be set to *none*. The output is usually a voltage, and an output node is typically selected.

Hbnoise is a small-signal analysis and is not limited by large-signal effects such as clipping or slew-rate limits. As a result, at low offset frequency, the phase noise might be significantly greater than 0dBC/Hz. This indicates that the noise is larger than the oscillations, which is not physically possible. If you want to see the phase noise curve level off at low frequency, set the *lorentzian* option to *yes*.

Phase noise is the total output noise divided by the harmonic specified as the relative harmonic for the node specified for the noise measurement. If you want to see the AM and PM components of noise, or you want to measure jitter, or you want to make an instantaneous point in time noise measurement, set the *Noise Type* to *modulated* or *jitter*.

Modulated

Modulated is provided so that the AM and PM components of noise can be separated out. When the *Noise Type* is *sources*, the total noise (AM + PM) is calculated. For a diagram to understand the concepts, see Modulated for Driven Circuits and Oscillators (AM and PM Noise) on page 357

Timedomain

This is usually applied to oscillators that drive digital dividers where the average noise over the entire waveform is not appropriate. The noise above and below the threshold does not matter because the output voltage is so far from the voltage that triggers the load that it does not matter. The only time the noise matters is at the exact time where the trigger event is generated. In time-domain for oscillators, a list of times from the waveform is supplied and a noise measurement is made at that list of times. Once the noise voltage is known, the timing uncertainty can be determined from the slew rate. The noise calculated from this analysis provides a relative measure of the noise at different times. For a quantitative measurement, use PM jitter. For a diagram that illustrates this, see Jitter for Driven Circuits on page 360.

FM Jitter

FM jitter calculates a standard phase noise measurement, and also the AM and PM components from the modulated analysis, and adds the ability to integrate the phase noise curve to calculate the cycle jitter or the cycle-to-cycle jitter. The jitter calculations are integrated into the *Direct Plot Form* in ADE. All the measurements are averaged over the oscillator cycle.

PM Jitter

This is like the time-domain jitter measurement. You specify a threshold voltage and the timing jitter is calculated. In PM Jitter, a quantitative jitter measurement can be made from the direct plot form. For a diagram on how this works, see Jitter for Driven Circuits on page 360.

Noise Calculations in the Simulator

Noise Factor and Noise Figure

When the hbnoise analysis is set up with voltage selected instead of probe for the input, noise figure and noise factor cannot be calculated because the system resistance specified in the port component is not available to the simulator. Using a port as the input probe is required to obtain noise figure and noise factor calculations.

In the *Direct Plot Form*, noise factor and noise figure are both single-sideband measurements.

Noise factor is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the load))/Noise power at the output
from the resistance of the input port from the passband frequency only.
```

In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element. No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the hbnoise *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the Choosing Analyses form.

Noise figure is:

```
10*log(10) (Noise Factor)
```

Noise factor (IEEE) is similar to the noise figure measurement above. If you subtract from the numerator all the noise from the input port at all frequencies except the passband, you have the IEEE noise factor.

As mentioned earlier, in order to calculate the IEEE noise factor, a port component must be used as the input. In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element. No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the *Choosing Analyses* form.

Noise Factor IEEE is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the resistor in the input port at
all frequencies except for the passband) - (Noise power at the output from the
load))/Noise power at the output from the resistance of the input port from the
passband frequency only.
```

Noise figure IEEE is:

```
10*log(10) (IEEE Noise Factor)
```

Noise factor DSB is similar to the single sideband noise factor above, except in the denominator, the noise power from the input port at the image frequency is added.

Noise factor (DSB) is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the load))/Noise power at the output
from the resistance of the input port from the passband and image frequencies only.
```

As mentioned earlier, in order to calculate the DSB noise factor, a port component must be used as the input. In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element. No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency and by inference the image frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the *Choosing Analyses* form.

Noise figure (DSB) is:

```
10*log(10) (Noise Factor (DSB))
```

Summary:

`No` = total output noise
`Ns` = noise at the output due to the input probe at the passband frequency (the source)
`Nsi` = noise at the output due to the image harmonic at the source
`Nso` = noise at the output due to harmonics other than input at the source
`Nl` = noise at the output due to the output probe (the load)
`IRN` = input referred noise
`G` = gain of the circuit
`F` = noise factor
`NF` = noise figure
`Fdsb` = double sideband noise factor
`NFdsb` = double sideband noise figure
`Fieee` = IEEE single sideband noise factor
`NFieee` = IEEE single sideband noise figure

then,

```
IRN = sqrt(No^2/G^2)
F = (No^2 - Nl^2)/Ns^2
NF = 10*log10(F)
Fdsb = (No^2 - Nl^2)/(Ns^2+Nsi^2)
NFdsb = 10*log10(Fdsb)
Fieee = (No^2 - Nl^2 - Nso^2)/Ns^2
NFieee = 10*log10(Fieee)
```

When the hbnoise analysis is set up with voltage selected instead of probe, noise figure cannot be calculated.

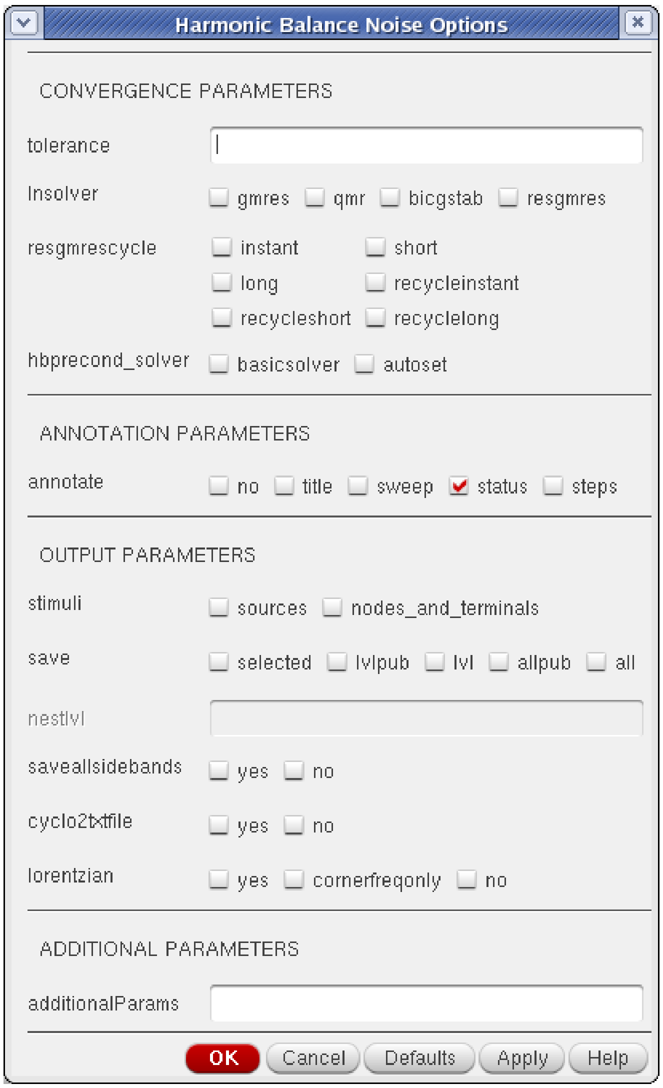
Input-Referred Noise

When the hbnoise analysis is set up with the input source being a port, input-referred noise may not be exact. In this case, the input-referred noise is referred to the voltage source that is inside the port component. The amplitude of the noise is divided by two, which assumes a perfect match. If an accurate input-referred noise is desired, set the *Input Source* to *voltage*, and select the input node. This will disable the noise figure calculation.

Noise transfer function

Hbnoise calculates the forward gain from the passband to the output and saves it. In the *Direct Plot Form*, this is called *Transfer Function*, and is in linear volts at the output divided by volts at the input.

Commonly Used Options



Tolerance

Leave this option at the default value.

Hbnoise uses an iterative solver to calculate the output amplitudes. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough, and the tolerance option specifies that accuracy for hbnoise. For driven circuits, the default is 1e-6. For oscillators, the default is 1e-4.

Lnsolver

Leave this option at the default. Each hbnoise sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases at the next iteration. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence. Their use is not suggested.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

gmres is the Generalized Minimum RESidual method.

qmr is the Quasi-Minimal Residual method.

bicgstab is the STABilized BI-Conjugate Gradient method.

resgmres is the REStarted Generalized Minimal RESidual method.

Resgmrescycle

Leave this option at the default. This option is active when resgmres is set for the lnsolver. For the resgmres linear solver, there are six different options.

Hbprecond_solver

The basic solver is the only solver available in standard Spectre. Autoset is the default solver when APS is used. This solver is faster. If stagnation is detected, Spectre prints a message in the Spectre output window.

Annotate

This option controls the amount of output in the spectre output window. Selecting *title* produces the least amount of detail, and selecting *steps* gives the most amount of detail. The options between produce more details as you move from left to right.

Stimuli

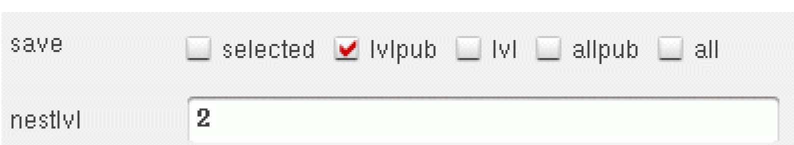
This option is not implemented in the *Direct Plot Form* in ADE. It is used for the hbxf analysis that runs when an hbnoise is run. The default is *sources* which calculates the forward conversion gain in hbxf from the sources in the circuit only. *nodes_and_terminals* also calculates the conversion gain from all the nodes in the circuit to the output. To display these results in ADE, you must use the results browser and the calculator.

Save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished using *Outputs - To Be Saved - Select On Schematic* and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.

Nestlvl

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer. The example below shows *lvlpub* selected that saves two levels of hierarchy.



Saveallsidebands

Leave this option at the default value of *no*. If you want to see the noise contributors from the different noise input frequencies, select *yes* for noise separation in the hbnoise *Choosing Analyses* form.

Lorentzian

This option is documented in the oscillator section. It only affects oscillators.

AdditionalParams

This is a field for `<option_name>=value` statements. Multiple `<option_name>=value` statements are allowed with a space between them. Generally, this is used to unlock the beta features.

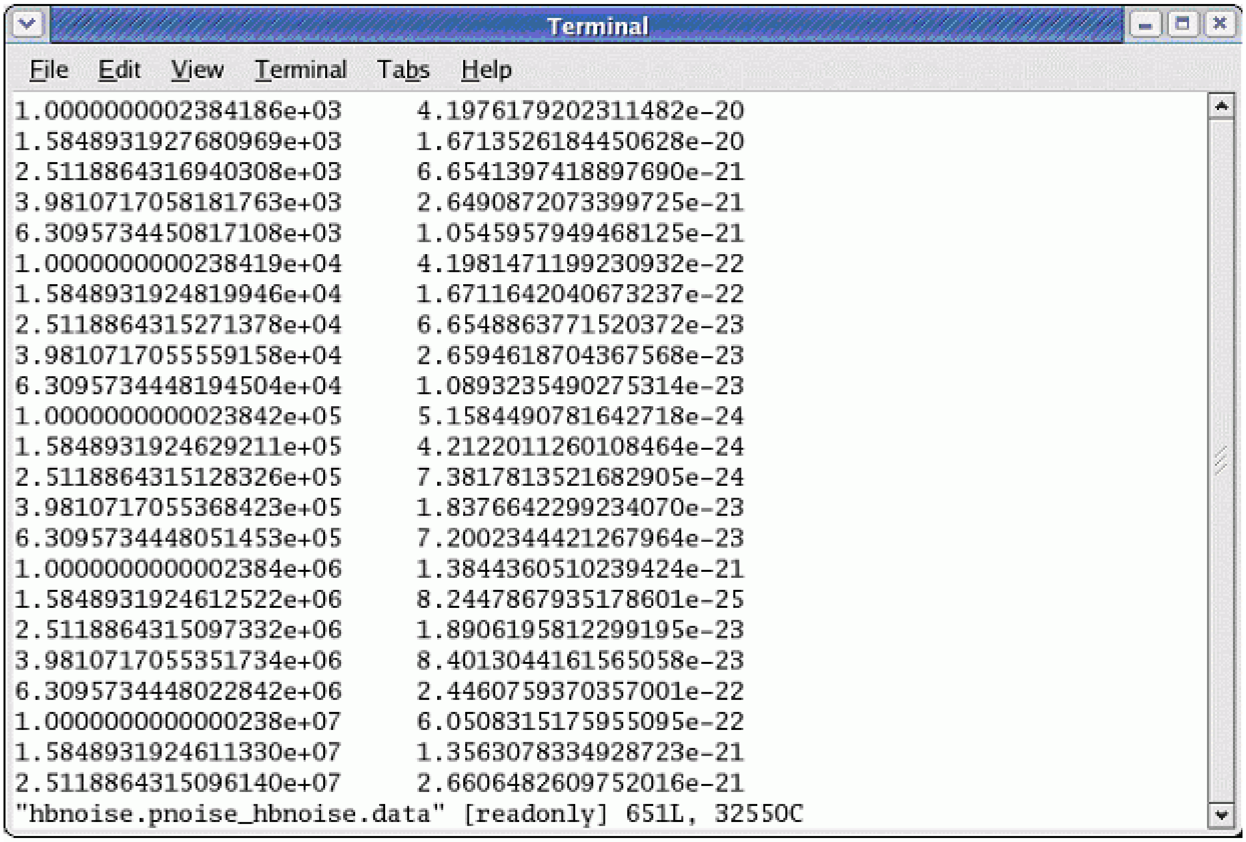
Cyclo2txtfile Option

Select the *cyclo2txtfile* option if you want the simulator to create a file that has all the time-averaged noise information in it. When this option is set, hbnoise runs a noise analysis at all the sideband frequencies using the sweep information from the hbnoise form. The file is created in the `psf` directory and has the filename `hbnoise.pnoise_hbnoise.data`. This option is only available when a single input frequency is applied to the circuit in the hb analysis. Copy this file to another directory if you want to save it. The `psf` directory contents are deleted at the start of each simulation.

The `psf` directory is located either in the `simulation/<circuit_name>/spectre/schematic` directory or in the `<circuit_name>/spectre/config` directory. The simulation directory defaults to your home directory and can be set by your system administrator. Contact your system administrator to find the simulation directory if you are unfamiliar with the location

For more information regarding the other hbnoise options, type `spectre -h hbnoise` in a shell window.

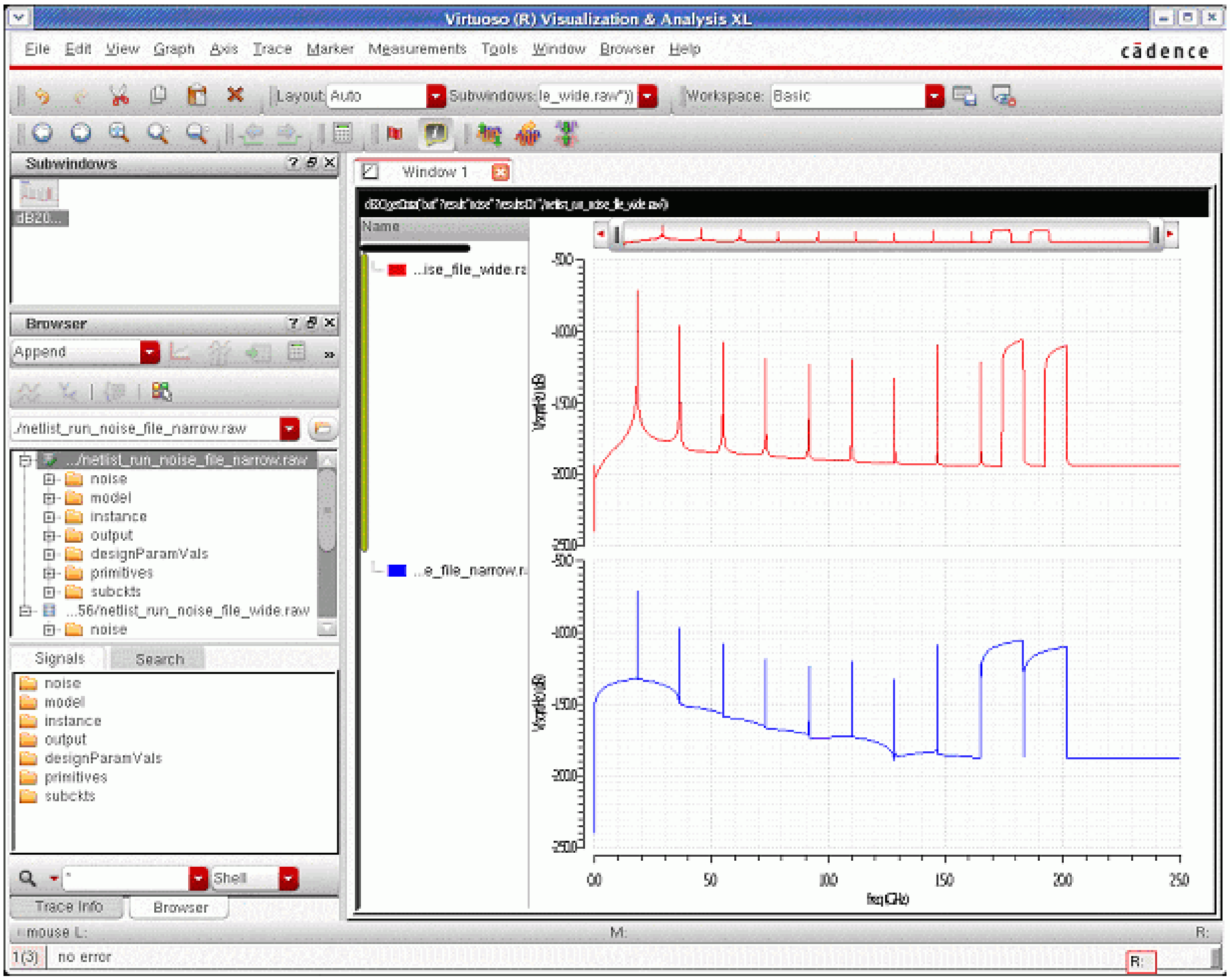
The output file has two columns. The left column is the frequency, and the right column is the noise voltage squared. Below is an example of the file.




```
1.0000000002384186e+03     4.1976179202311482e-20
1.5848931927680969e+03     1.6713526184450628e-20
2.5118864316940308e+03     6.6541397418897690e-21
3.9810717058181763e+03     2.6490872073399725e-21
6.3095734450817108e+03     1.0545957949468125e-21
1.0000000000238419e+04     4.1981471199230932e-22
1.5848931924819946e+04     1.6711642040673237e-22
2.5118864315271378e+04     6.6548863771520372e-23
3.9810717055559158e+04     2.6594618704367568e-23
6.3095734448194504e+04     1.0893235490275314e-23
1.0000000000023842e+05     5.1584490781642718e-24
1.5848931924629211e+05     4.2122011260108464e-24
2.5118864315128326e+05     7.3817813521682905e-24
3.9810717055368423e+05     1.8376642299234070e-23
6.3095734448051453e+05     7.2002344421267964e-23
1.0000000000002384e+06     1.3844360510239424e-21
1.5848931924612522e+06     8.2447867935178601e-25
2.5118864315097332e+06     1.8906195812299195e-23
3.9810717055351734e+06     8.4013044161565058e-23
6.3095734448022842e+06     2.4460759370357001e-22
1.0000000000000238e+07     6.0508315175955095e-22
1.5848931924611330e+07     1.3563078334928723e-21
2.5118864315096140e+07     2.6606482609752016e-21
"hbnoise.pnoise_hbnoise.data" [readonly] 651L, 32550C
```

Interpolation in Noise Files

All independent sources have a property called Noise file name. This is an ascii file with two columns. The left column is frequency, and the right column is noise voltage squared. This file is read for all the small-signal noise analyses. Between the points in the file, an interpolation is made.

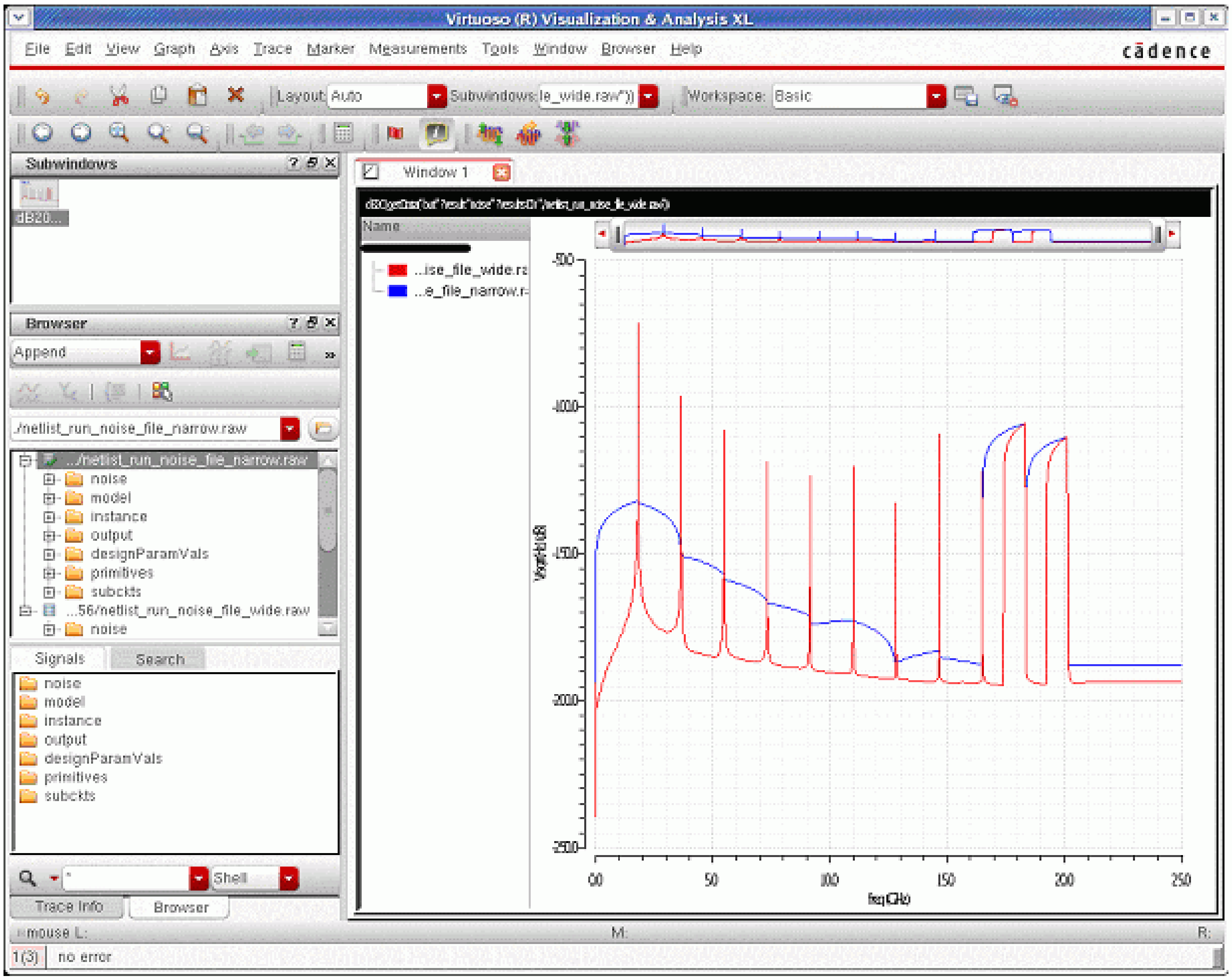


Both curves in the above figure use the `cyclo2txt` output file for an oscillator at about 1.85GHz. Just the noise in the file is plotted. The top curve has a noise sweep to just less than one half the oscillation frequency. The bottom curve has a sweep to 10 MHz relative to the first harmonic. There were 10 harmonics in the harmonic balance simulation and 10 sidebands in the hbnoise analysis.

Because the noise analysis was referenced to the first harmonic, the +10 sideband starts at +10*1.85GHz + 1.85GHz, or 20.35GHz, and goes to about 20.36GHz for the 10MHz sweep, and 21.25GHz for the 900MHz sweep. Note that the data near 20GHz in both starts at about -120dBV and drops. The minus 10 sideband starts at -10*1.85GHz + 1.85GHz, or -16.65GHz to -16.64GHz for the 10MHz sweep and -15.75GHz for the 900MHz sweep. Note that there is no data on the bottom side of the 10th and 11th harmonic. The data in between the frequency points in the file is interpolated. That is the reason for the large rise for the last 2 noise spikes.

The spikes from the 900MHz sweep is narrower because the data goes to a higher frequency near 17GHz and 19GHz.

The figure below shows the curves overlaid on the same strip.

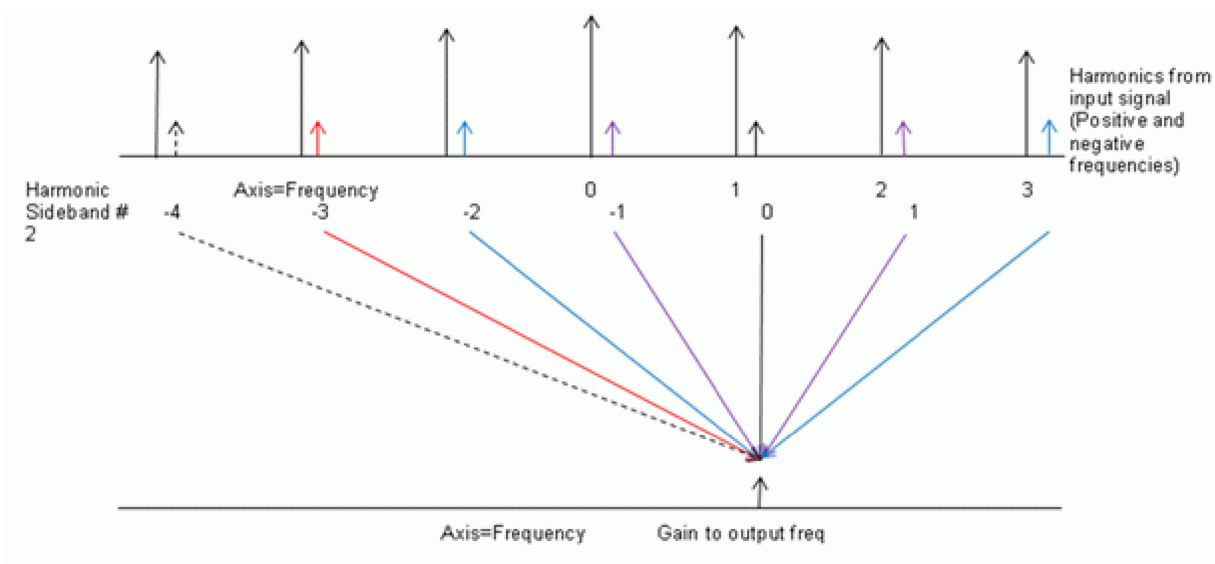


Note that the interpolation above and below the actual data in the 10MHz sweep (the blue trace) causes much more noise than the real system produces. Make sure that your noise sweep is sufficient for your requirements.

hbXF

Note that in the noise simulation, the forward transfer functions need to be calculated for all the noise sources in the circuit at all the noise frequencies that mix to the output frequency. When hb analysis is run and hbnoise is also run, the forward conversion gain from the sources in the circuit to the output specified in the hbnoise form is also calculated and presented in the *Direct Plot Form* as hbxf results.

Note that this is small-signal conversion gain. The large-signal effects like slew-rate and clipping limits are ignored for the hbxf calculation. If you need a large-signal measurement, you must use a large-signal analysis like hb, pss or qpss to do the calculation.



In the figure above, the top spectrum represents the harmonics created by the hb large-signal analysis. The negative and positive frequencies are shown and the DC component is in the center. The arrows show the transfer functions from the different input frequencies to the output frequency that is shown on the bottom axis. hbXF takes the maximum sideband property from the hbnoise setup form, and calculates the specified sidebands, which are really input to output conversion gain transfer functions.

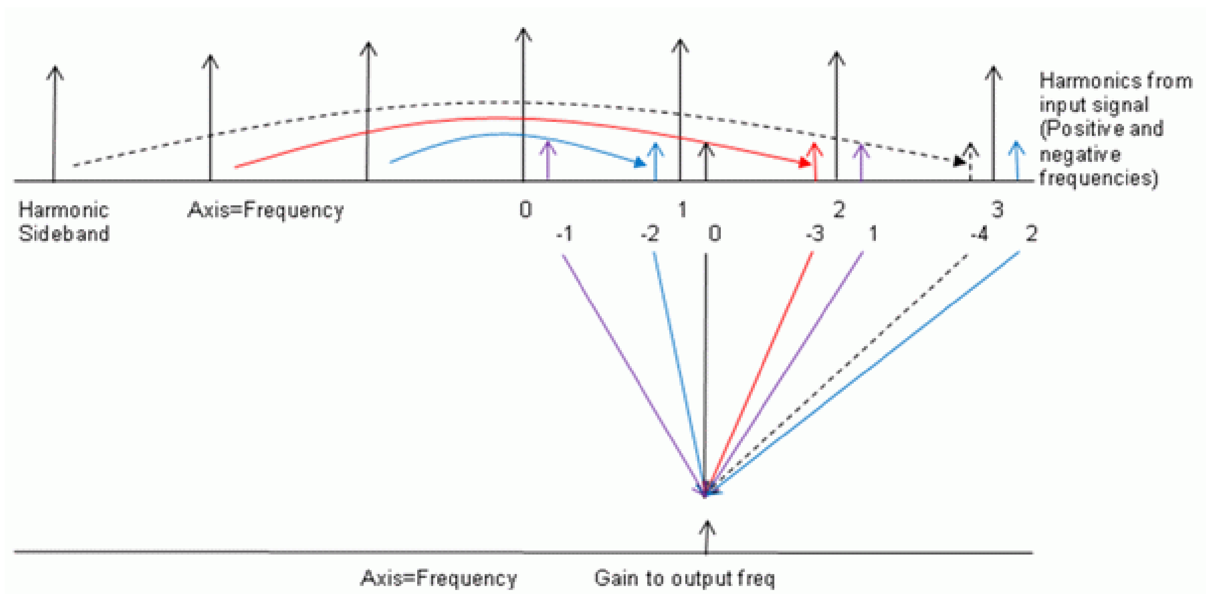
The figure above illustrates a down-conversion case where the output is at low frequency.

The sideband numbers indicate which harmonic number is being mixed for the transfer function. Zero means that there is no frequency translation. One is the mixing product that occurs with a frequency shift of +1 times the hb harmonic frequency. Mixing frequencies to the left have negative numbers because the input frequency that causes the output are more negative than the output frequency.

For example, imagine a large-signal input at 1GHz in the hb analysis and an output frequency of 100MHz to be calculated. With no mixing, there is a gain from the sources in the circuit to the output of the circuit at 100MHz. Only the transfer function from 100MHz to 100MHz is included in this calculation. This is called the zero sideband because there is zero frequency shift from the input to the output.

The input frequencies that mix with the first harmonic at 1GHz to provide an output at 100MHz are at -900MHz and 1.1GHz. The frequency that is below the zero sideband is the -1 sideband and the frequency above the zero sideband is the +1 sideband.

Similarly, there are terms that mix with the second and third harmonics.



Usually, we think of positive frequencies only. This is shown above.

ADE Implementation

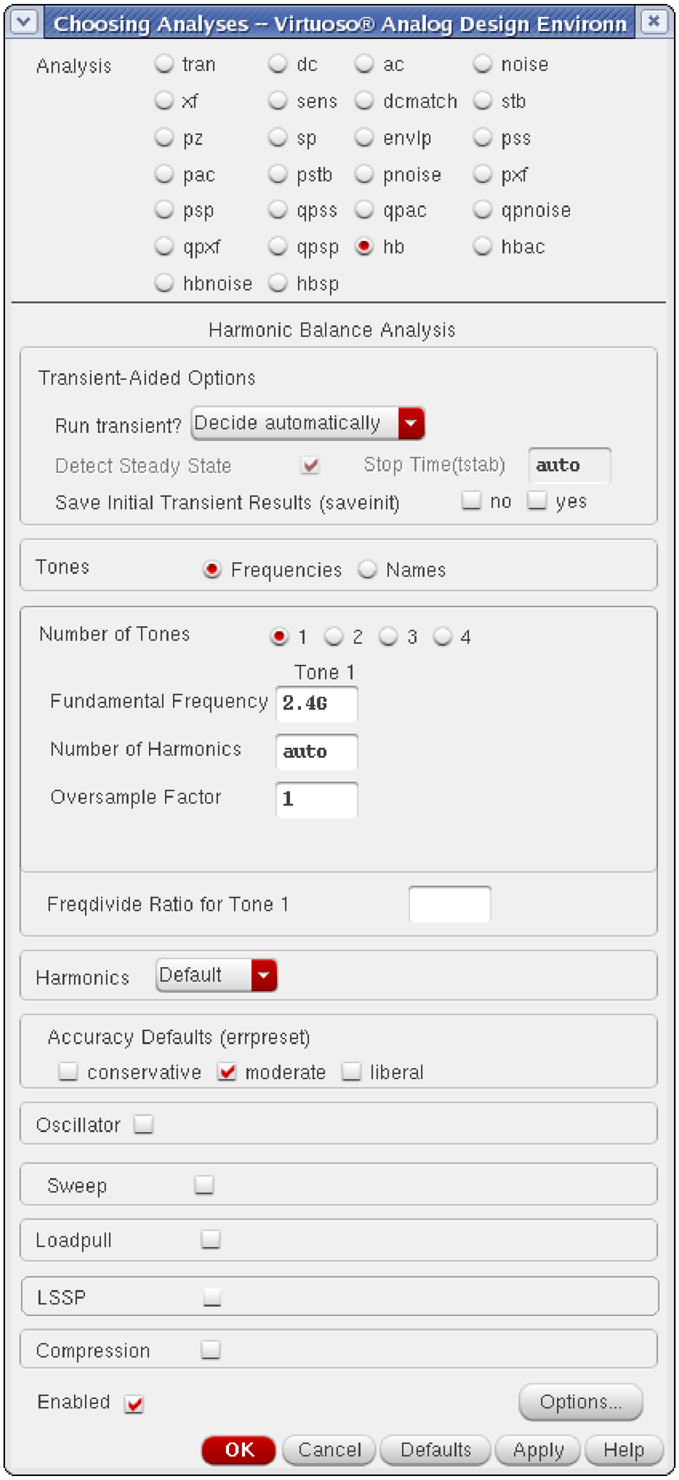
This section jumps from theory to practical with examples on how to use the individual analysis types that have been covered. Driven circuits and oscillators will be shown.

Driven Circuits

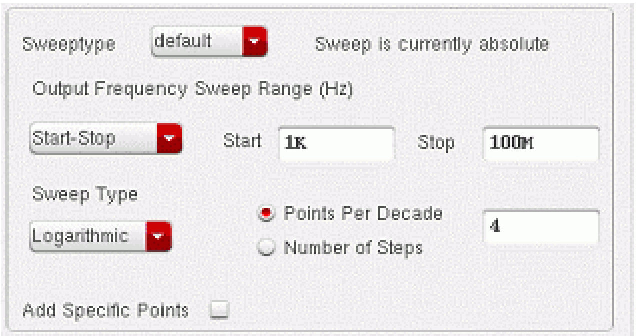
Noise Type = Sources

On the *Choosing Analyses* form:

1. First, set up the harmonic balance analysis. For more information, see the Harmonic Balance section at the beginning of this chapter.



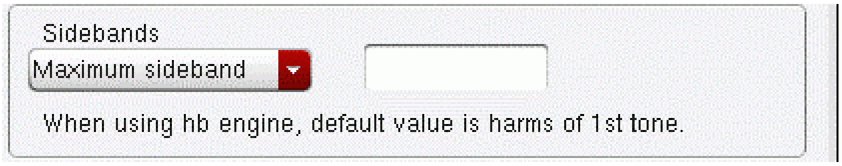
2. In the hbnoise *Choosing Analyses* form, specify the output frequency range of your circuit.



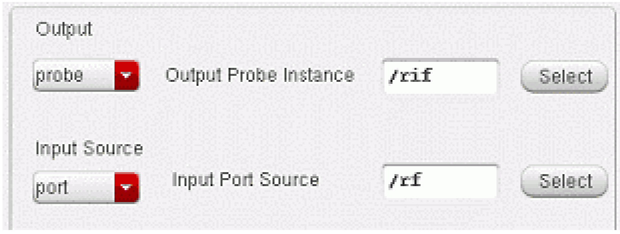
Note: Since the fundamental quantity that is calculated by the noise analysis is the noise at the output, the frequency range is always the output frequency range.

3. Set about 3 to 5 points per decade in the *Points Per Decade* field.

4. Leave the *Maximum sideband* field blank or set it to the same number as the number of harmonics in the hb analysis. This specifies that the noise that mixes with all the hb harmonics is calculated.

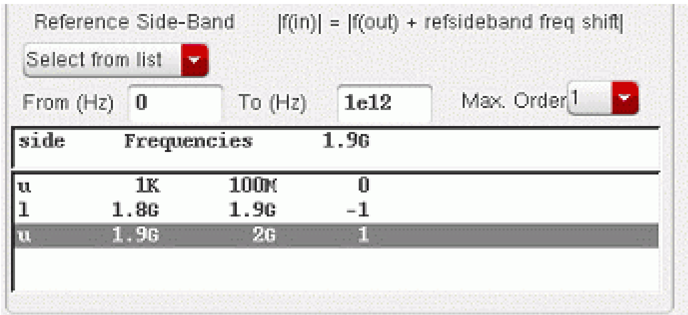


5. Specify the output of the circuit. When the output is set to *probe*, a resistor or a port can be selected and this noise will be subtracted from the noise figure calculation.



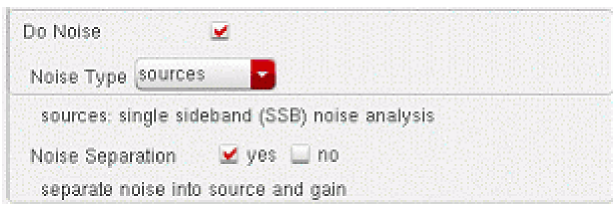
6. Select *port* from the *Input Source* drop-down list for noise figure calculation.

7. Choose *Select from list* from the drop-down list for reference sideband, which specifies the input frequency range for the noise figure calculation.



8. Select the passband frequency range from the list.

9. Select *sources* from the *Noise Type* drop-down list to calculate the average noise power at the output averaged over the entire time of one cycle of the input waveform.



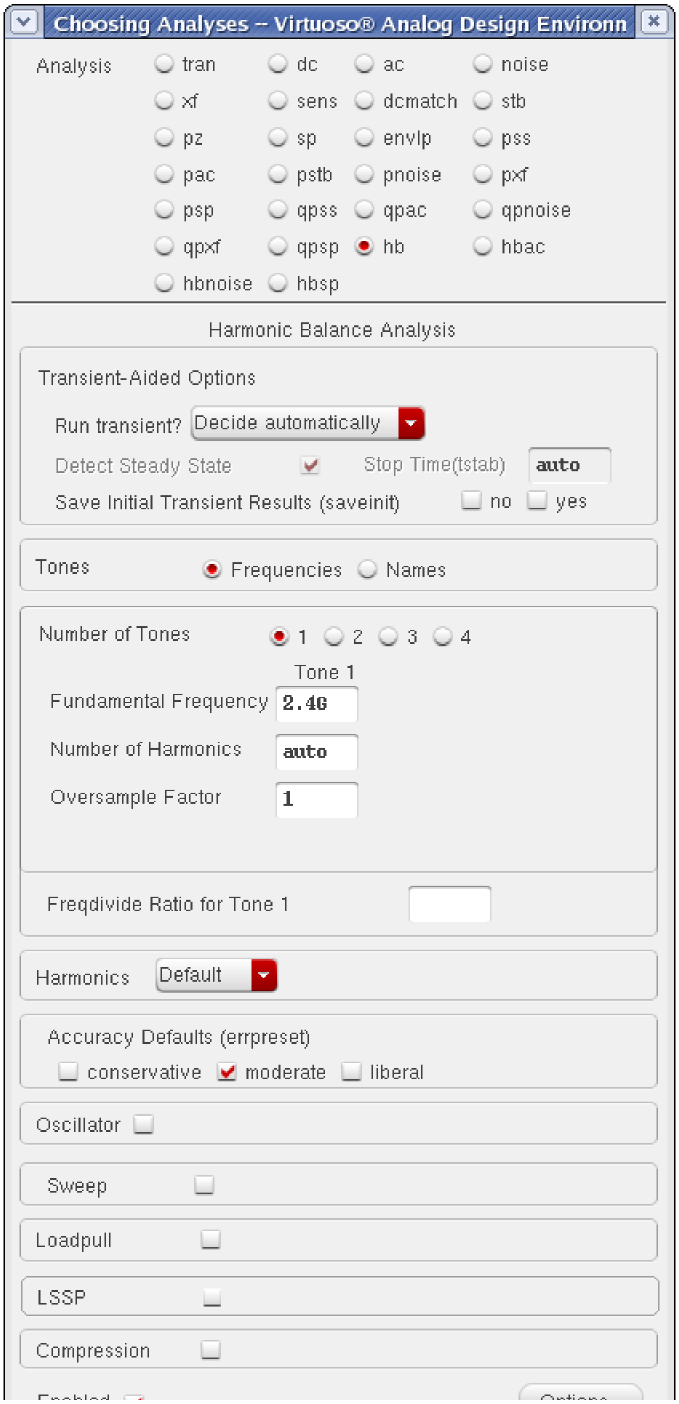
10. Select *yes* for *Noise Separation* to enable plotting of the individual noise contributors in the circuit. Noise separation is only available when *Noise Type* is set to *sources*.

Multiple hbnoise

When *Noise type* is set to *sources*, multiple hbnoise analyses can be run after a single hb analysis. This is useful for LNA-Mixer combinations, or in any circuit where there are multiple points in the circuit where noise is important.

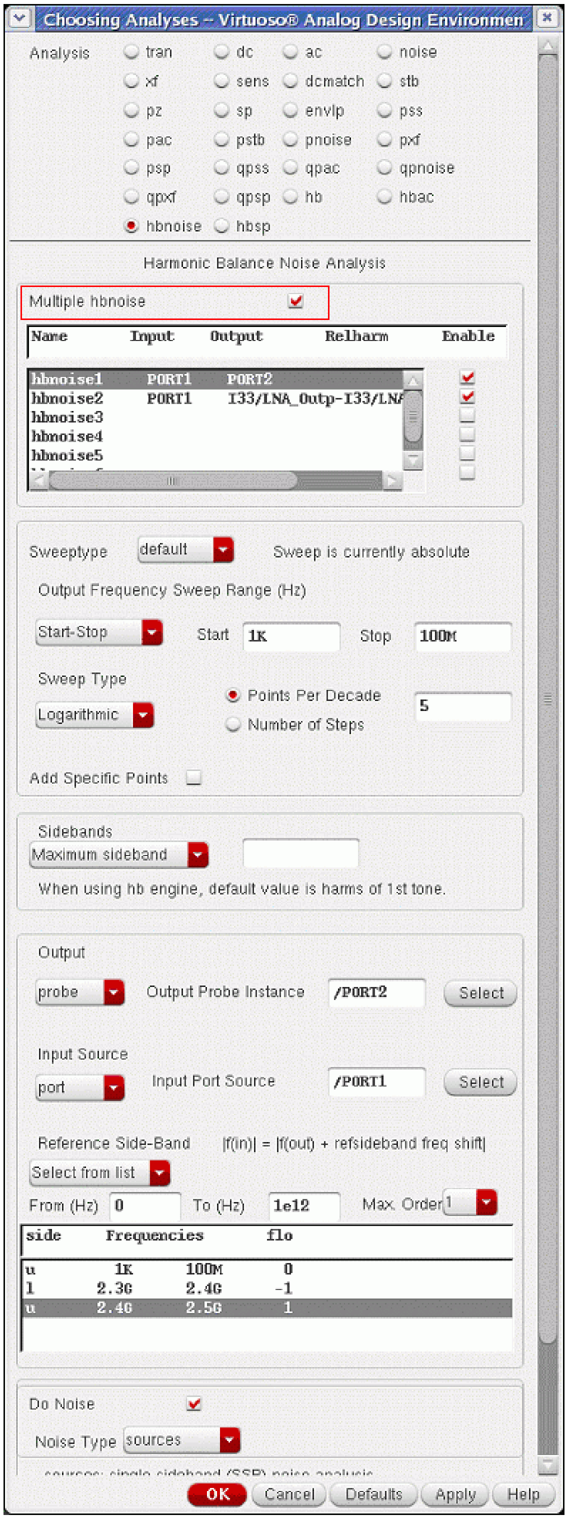
To run multiple analyses after a single hb:

1. First, set up the harmonic balance simulation. for moire information, see the harmonic balance section at the beginning of this chapter.



2. Select the *Multiple hbnoise* check box located at the top of the hbnoise *Choosing Analyses* form.

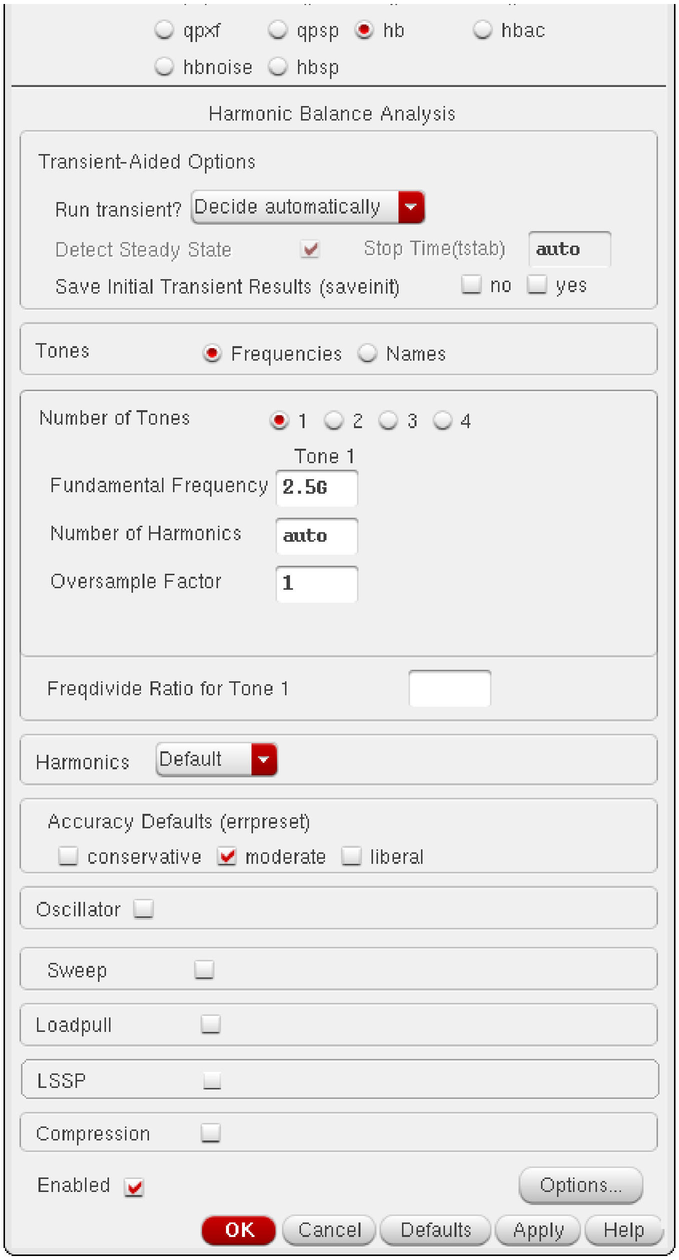
3. Set up all the noise analyses you want to run, as shown below.



Noise Type = Modulated

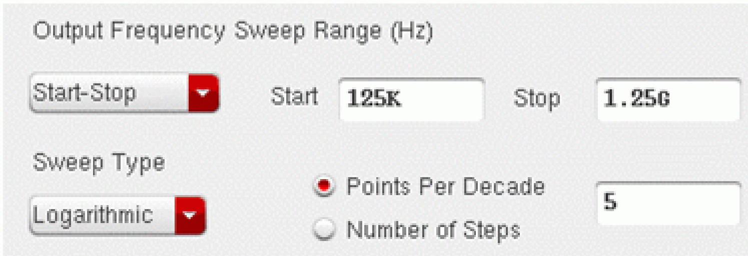
Modulated allows the measurement of the AM and PM components of noise separately.

First, set up the hb analysis as appropriate for your circuit. For more information, see the harmonic balance section of this chapter.



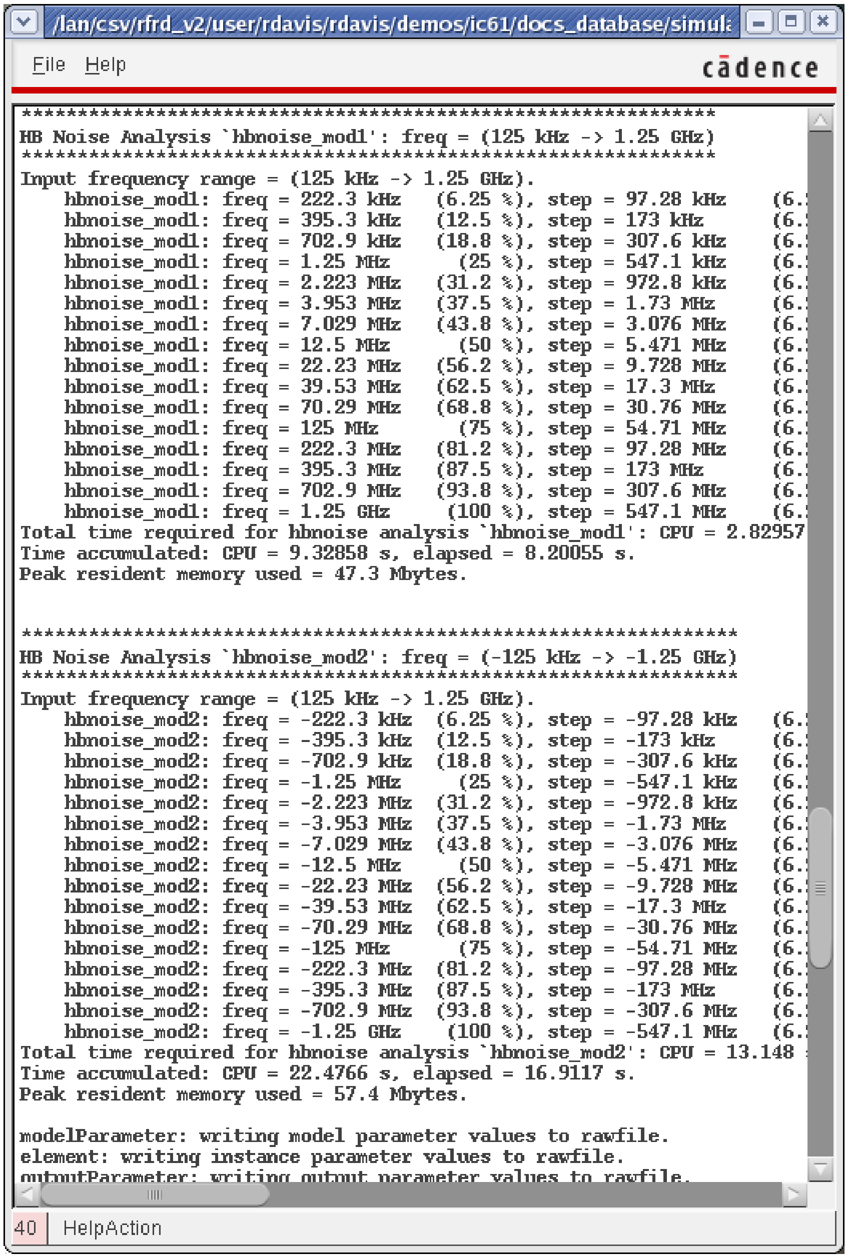
On the hbnoise *Choosing Analyses* form:

1. Set up an appropriate frequency range of interest.



2. Select *modulated* from the *Noise Type* drop-down list.

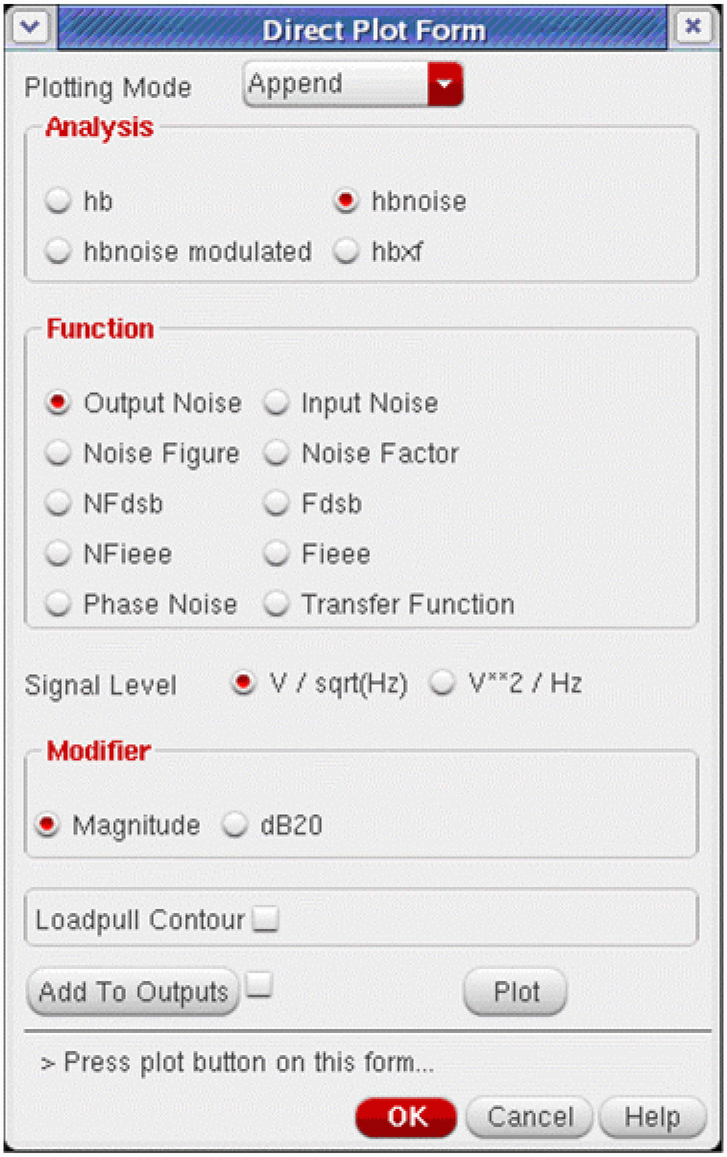
3. Run the simulation. Note that modulated runs both the positive and negative frequencies.



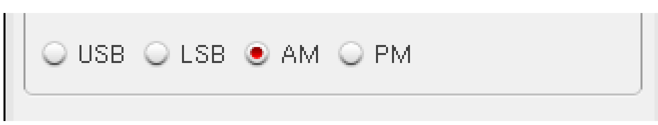
Now plot the Output noise using the *Direct Plot Form*.

On the *Direct Plot Form*:

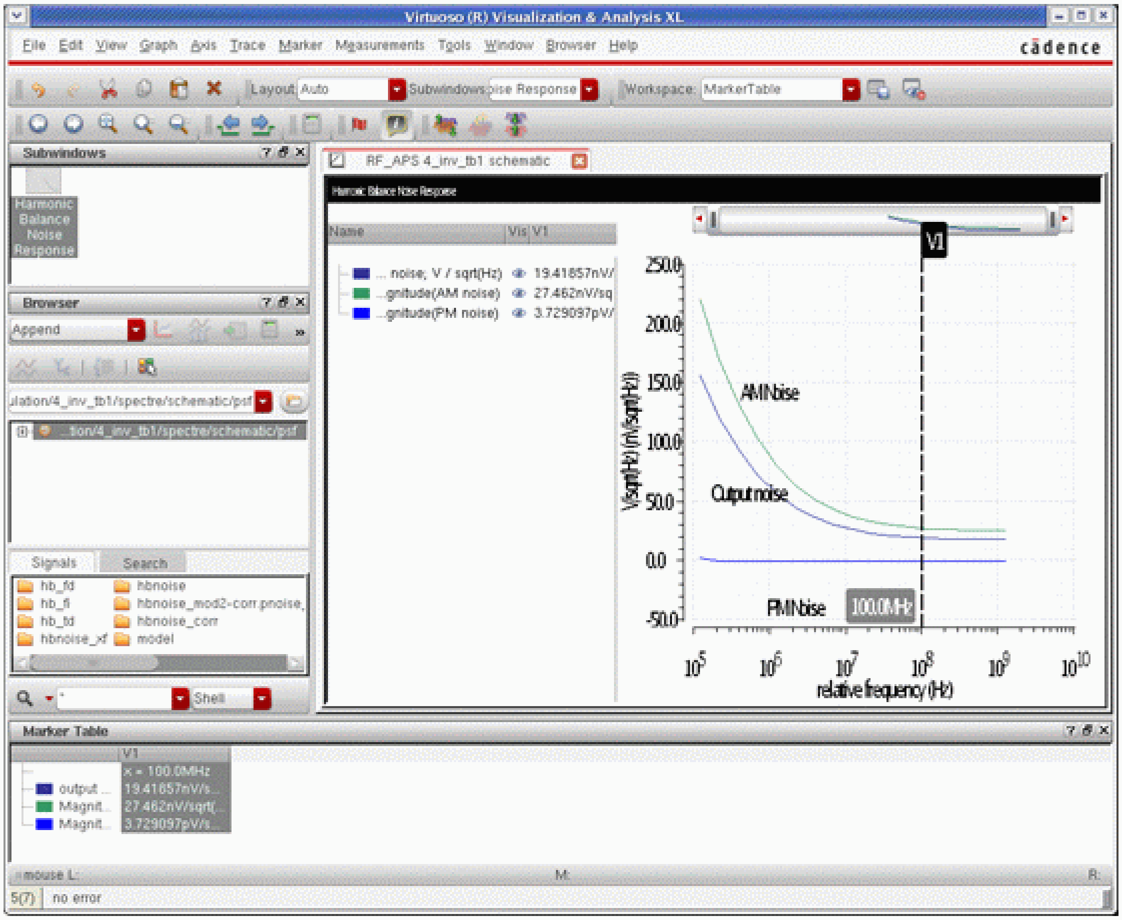
1. Select *Output Noise* from the *Function* section.



2. Select *AM* or *PM* from the *Noise Type* radio buttons. The *AM* noise is shown below.



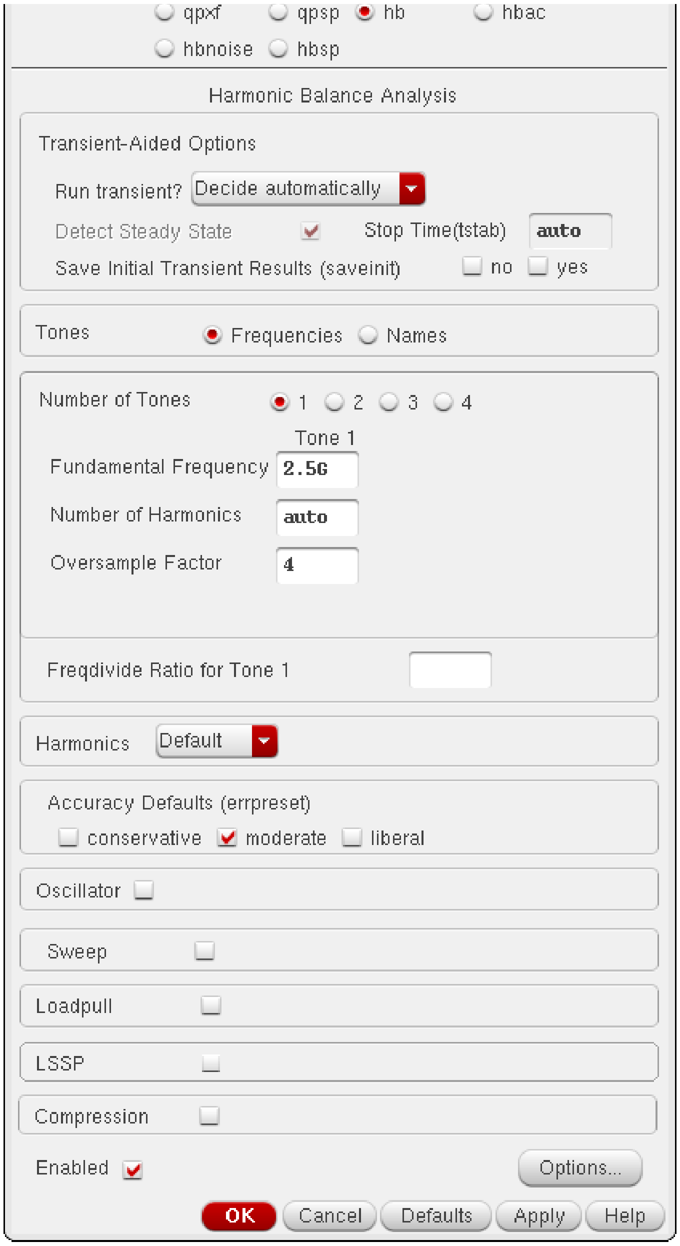
3. Click *Plot*.



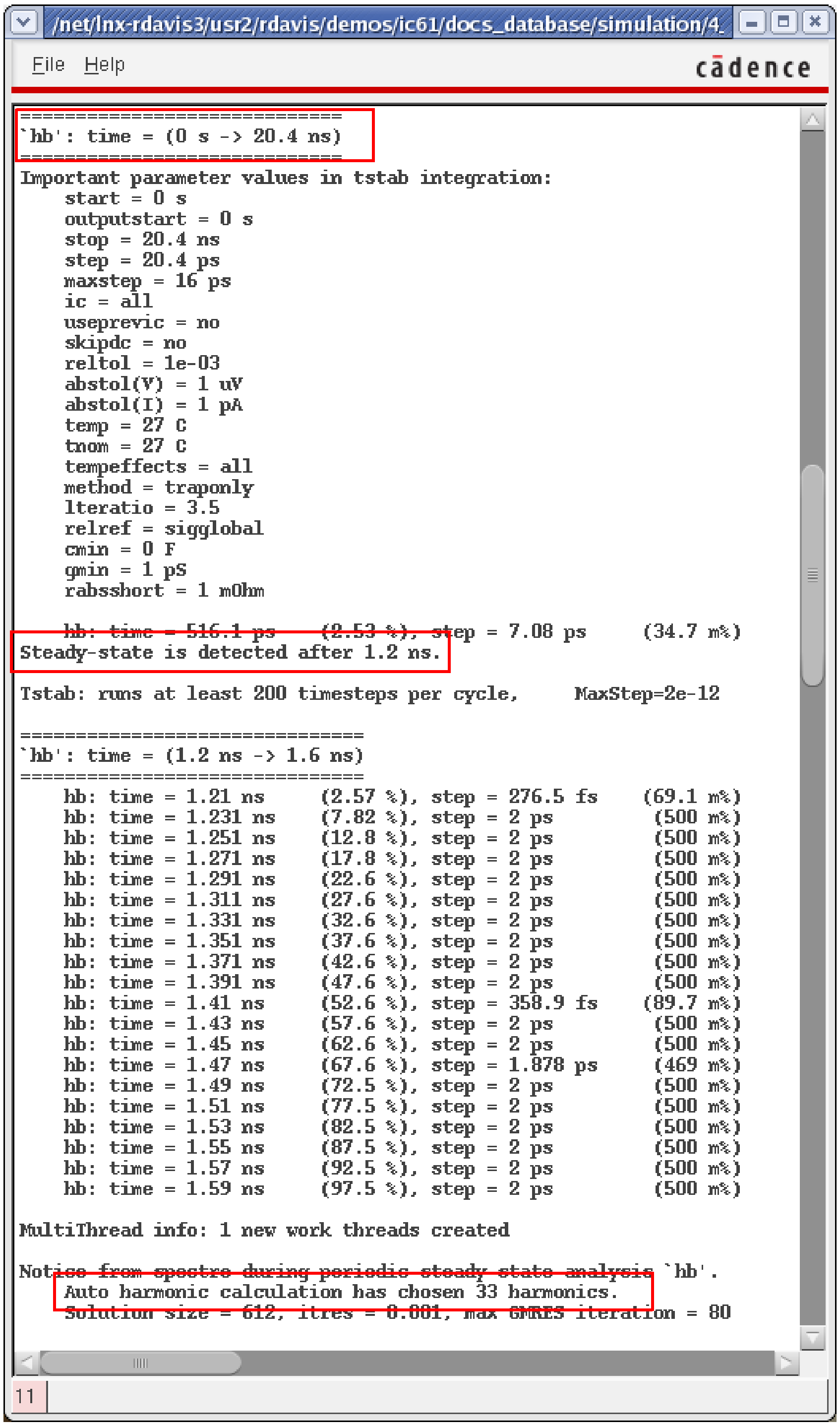
On the plot, note that the AM noise is 3dB larger than the output noise. This is because the output noise is derived from the positive frequencies only, and the AM and PM noise is derived using the positive and negative frequencies.

Noise Type = Jitter

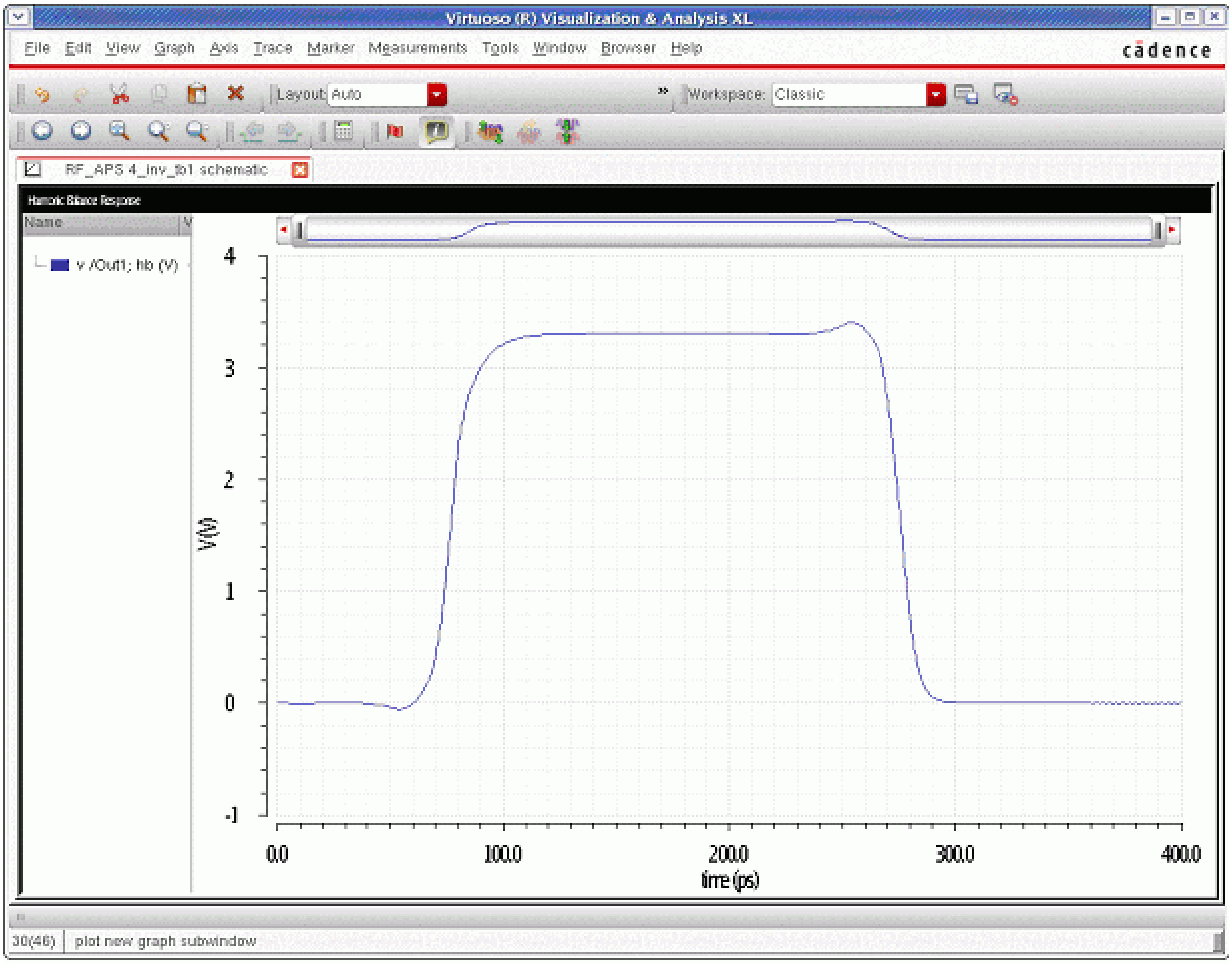
Driven jitter is usually used for measuring the noise in a digital circuit at a specific threshold crossing. Accordingly, the hb *Choose Analyses* form has oversample factor set to 4, as shown below. Note that *Number of Harmonics* is set to *auto*. *auto* sets the number of harmonics that are needed automatically. You do not need to go through the trial and error process of setting harmonics manually.



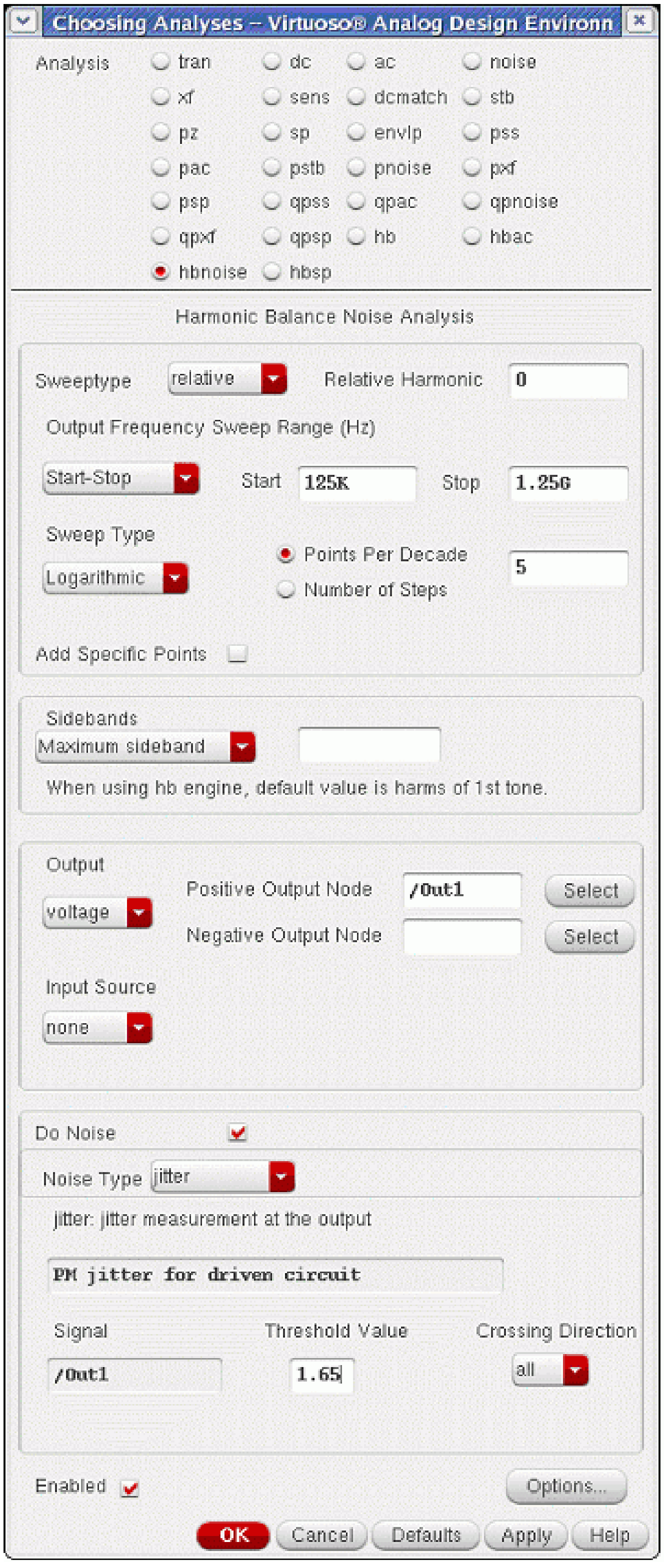
When Spectre runs with *Decide automatically* and *auto harms*, the tstab interval can be remarkably short, as shown below. In addition, the number of harmonics is set high enough to have less than 1% total aliasing power in the result.



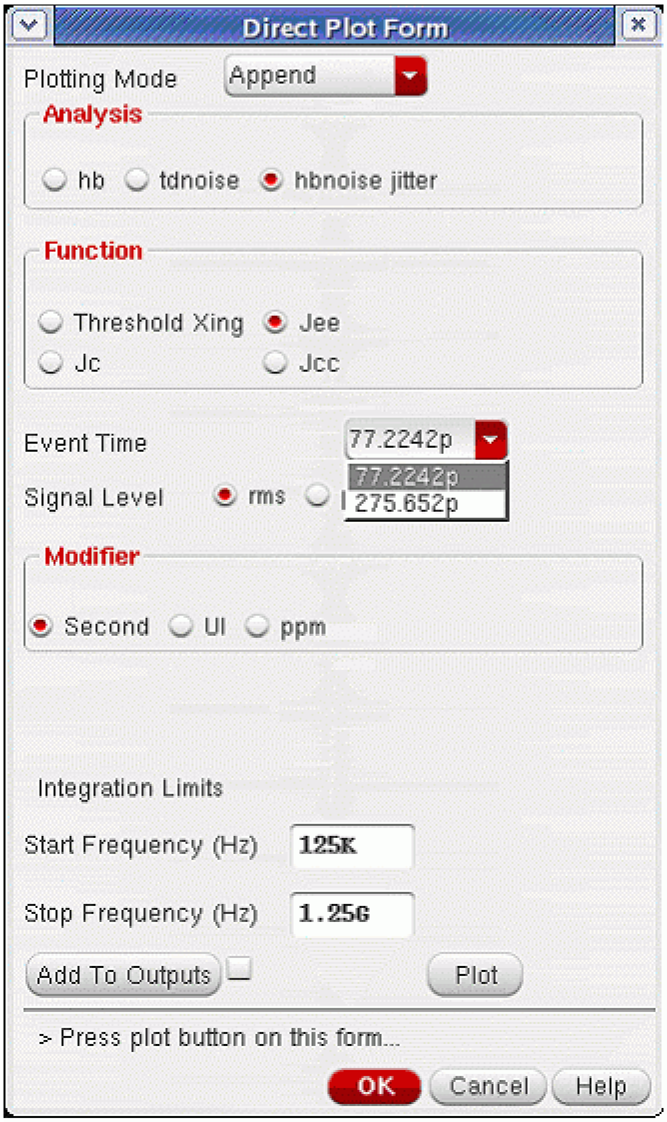
An appropriate threshold is needed, which can be determined by plotting the time domain waveform from the hb *Direct Plot Form*. The waveform is an ifft of the harmonics that were solved in harmonic balance. The waveform below shows an appropriate threshold value to be 1.65V.



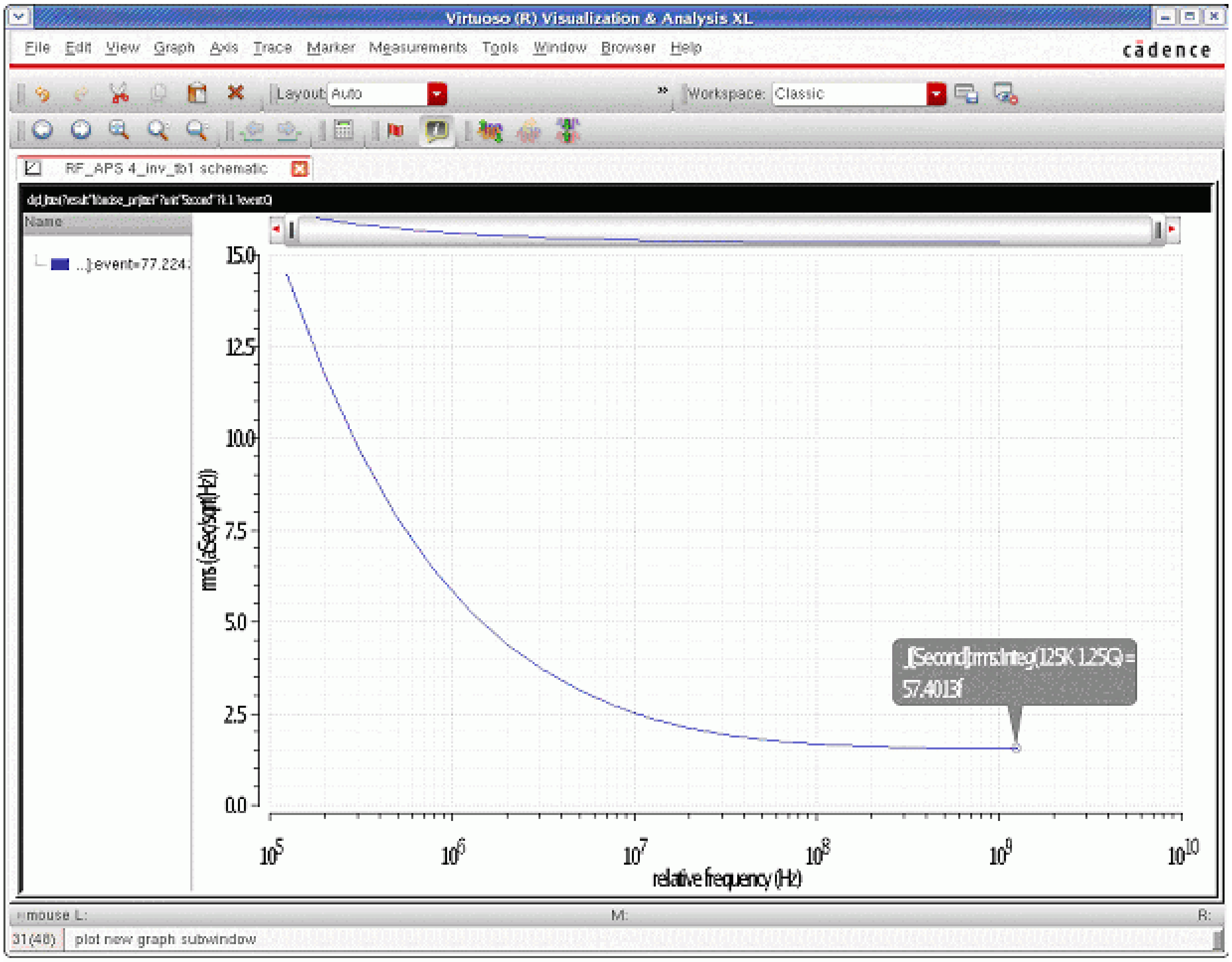
On the hbnoise *Choosing Analyses* form, the stop frequency is half the input frequency. The start frequency is four decades below that. About five points per decade is usually sufficient. *Maximum sideband* should be left blank. The *Output* field is set to *voltage*, and the *Input Source* field is set to *none* because usually, only the output jitter is needed. The threshold is set appropriately. The *Crossing Direction* option can be set to *all*, *rise*, or *fall*.


Choosing Analyses – Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Harmonic Balance Noise Analysis
Sweeptype
relative
Relative Harmonic
0
Output Frequency Sweep Range (Hz)
Start-Stop
Start
125K
Stop
1.25G
Sweep Type
Logarithmic
Points Per Decade
Number of Steps
5
Add Specific Points
Sidebands
Maximum sideband
When using hb engine, default value is harms of 1st tone.
Output
voltage
Positive Output Node
/Out1
Select
Negative Output Node
Select
Input Source
none
Do Noise
Noise Type
jitter
jitter: jitter measurement at the output
PM jitter for driven circuit
Signal
Threshold Value
Crossing Direction
/Out1
1.65
all
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

When the simulation completes, use the *Direct Plot Form* to plot the jitter. For driven circuits, select *Jee*, which is edge-to-edge jitter. The event time is the time in the ifft for the large-signal hb simulation. In the below example, 77p is the rising edge, and 275p is the falling edge.



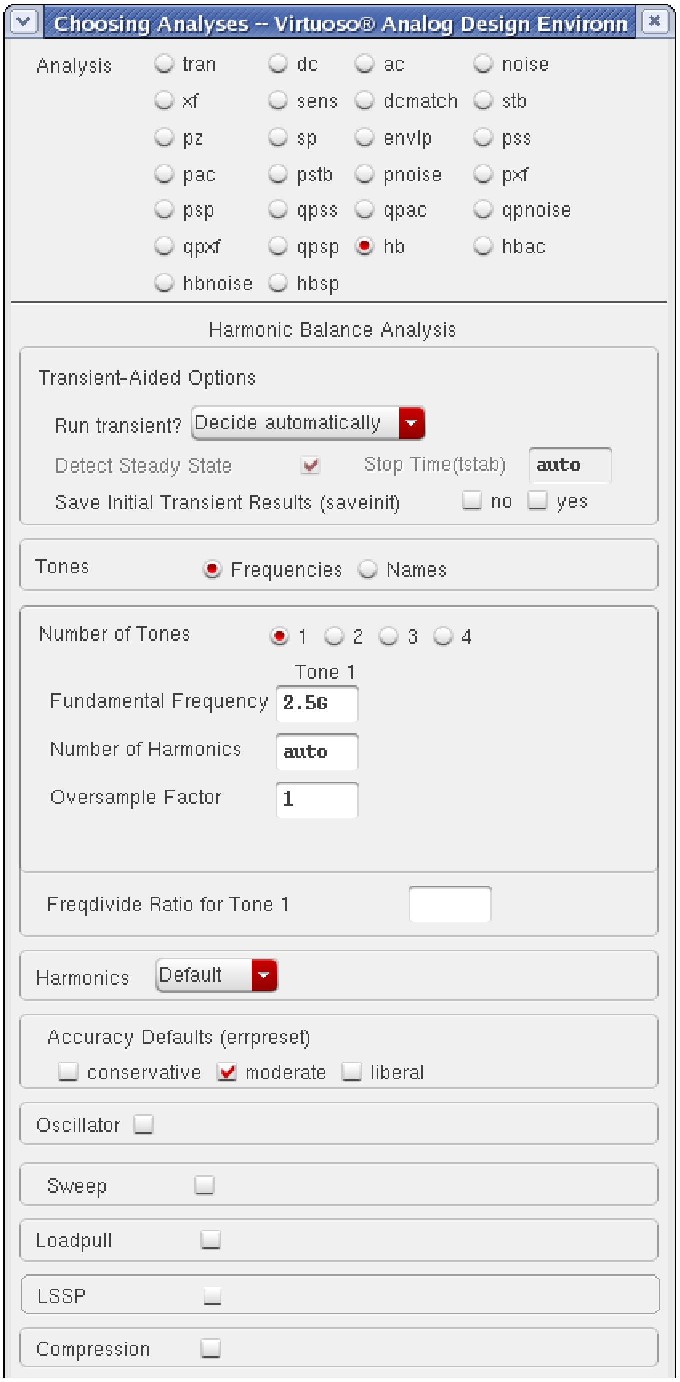
The output noise frequency response and the edge-to-edge jitter are displayed. The jee measurement is one standard deviation for the expected jitter.



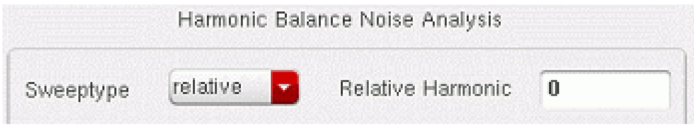
Noise Type = Timedomain

On the hbnoise *Choosing Analyses* form:

1. First, set up the harmonic balance simulation. For more information on harmonic balance, see the Harmonic Balance section at the beginning of this chapter.

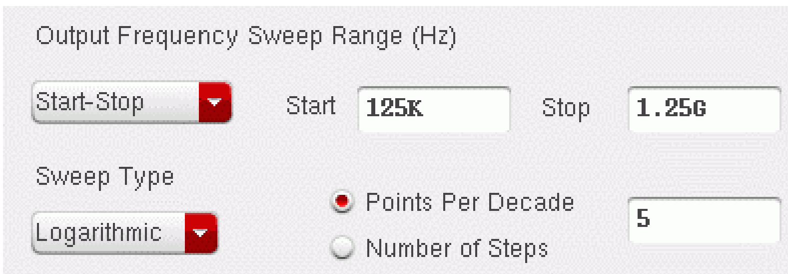


2. In the hbnoise *Choosing Analyses* form, *relative* is forced in the *Sweeptype* drop-down list.



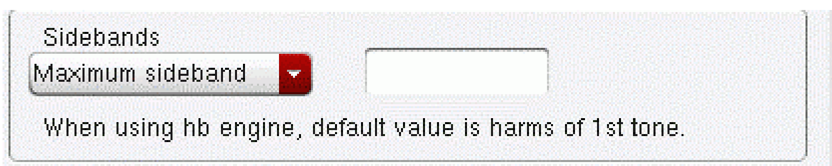
3. Specify 0 in the *Relative Harmonic* field.

4. Set the stop frequency to half the input hb frequency.

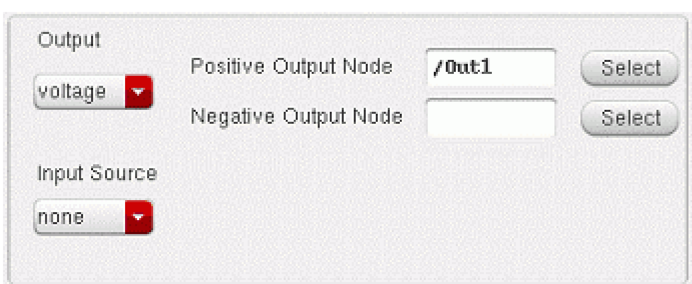


5. Set the start frequency to four decades smaller than the stop frequency.

6. Leave the *Maximum sideband* field blank.



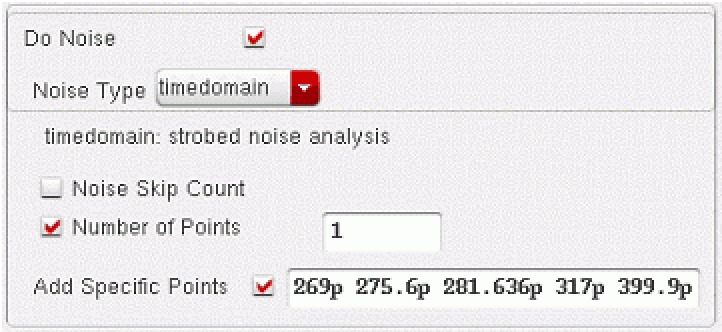
7. Select *voltage* from the *Output* drop-down list.



8. Specify the output node in the *Positive Output Node* field.

9. Select *none* from the *Input Source* drop-down list.

10. Select *timedomain* from the *Noise Type* drop-down list.



11. Select the *Number of Points* check box and specify 1 in the text field.

12. Select the *Add Specific Points* check box and specify the time values for the desired instantaneous noise measurements. Separate the entries with a space.

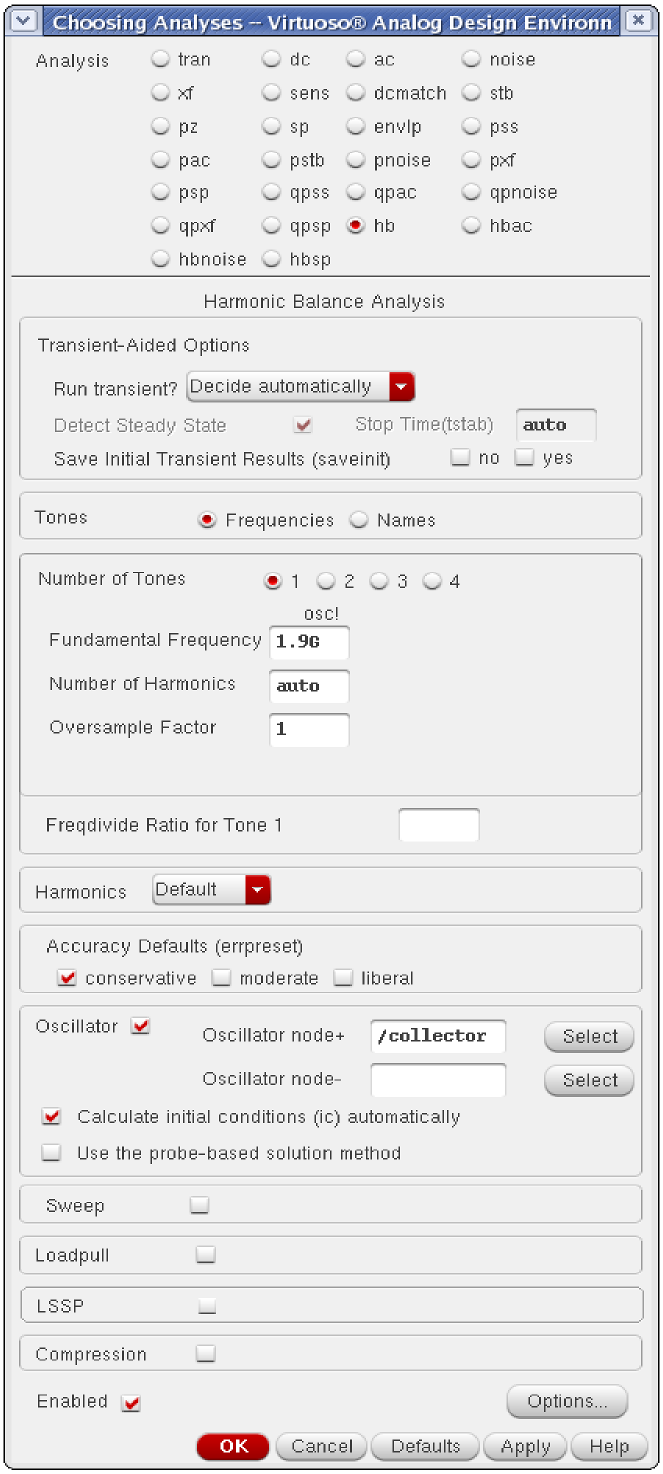
Oscillators

Noise Type = Sources

The settings are the same as driven `noise type = sources` with the exception of being a relative sweep by default. Since the phase noise of the oscillator is near one of the harmonics, *Sweeptype* is set to *relative* by default. A harmonic number is provided for the case where there is a frequency multiplier or a frequency divider on the output of the oscillator. Noise type = sources also has the ability to plot the noise contributors from mixing with each harmonic when noise separation is selected. There is an example of noise separation later in this chapter.

The hbnoise setup is similar to driven except that the relative harmonic is specified.

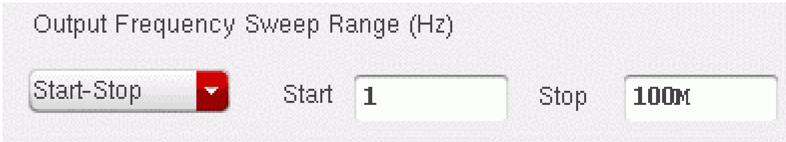
First, set up the harmonic balance analysis. For more information, see the Harmonic Balance section at the beginning of this chapter.



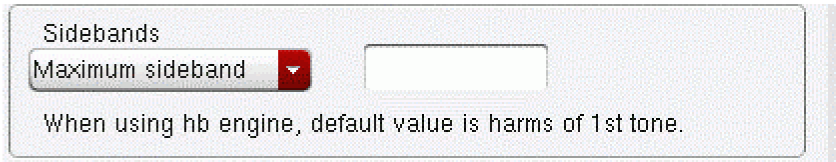
On the hbnoise *Choosing Analyses* form:

1. Select *relative* from the *Sweeptype* drop-down list.

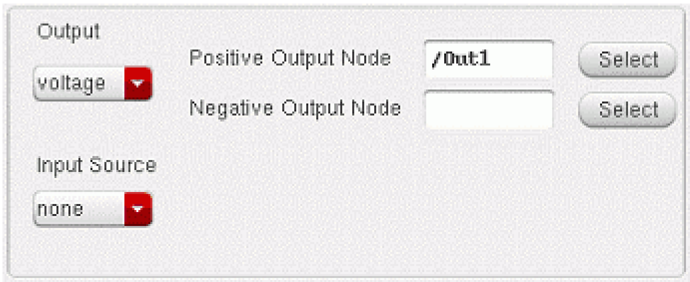
2. Specify the appropriate harmonic number for your design. In this example, the first harmonic is specified in the *Relative Harmonic* field.
3. The frequency range should be the relative frequency range.



4. Leave the *Maximum sideband* field blank.

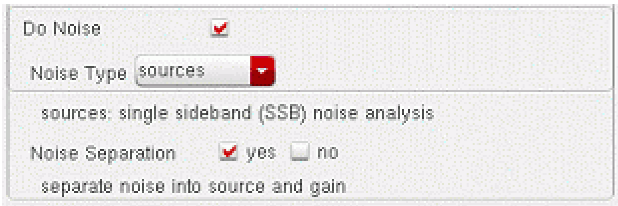


5. Select *voltage* from the *Output* drop-down list.



6. Specify the output node in the *Positive Output Node* field.
7. Select *none* from the *Input Source* drop-down list.

8. Select *sources* from the *Noise Type* drop-down list. This gives an averaged noise measurement.



9. Select *yes* for *Noise Separation* if you want to plot the individual noise contributors.

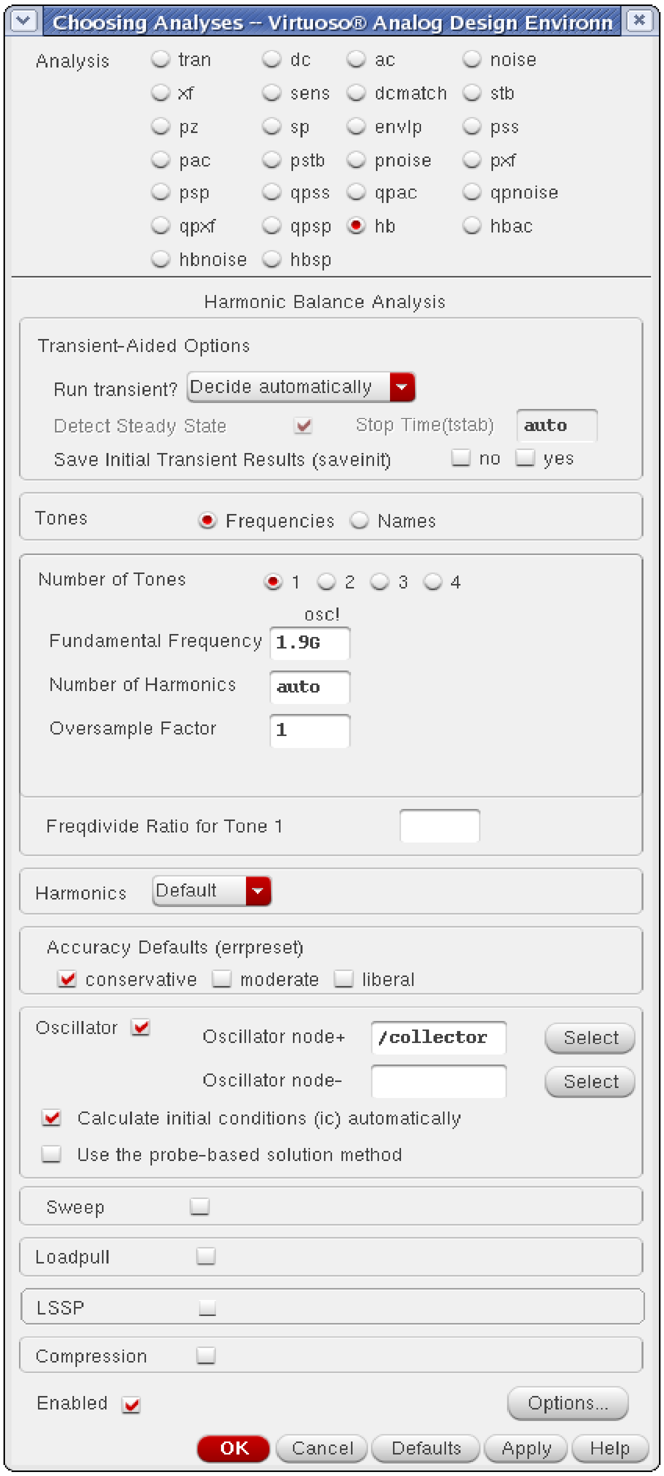
Noise Type = Modulated

This is the same as driven noise type = modulated (see Noise Type = Modulated on page 380). The AM and PM components of phase noise are available for plotting. Generally jitter is the preferred setting because it calculates the modulated result and it allows the calculation of jitter for the oscillator output. For examples, see Oscillators on page 490.

Noise Type = Timedomain

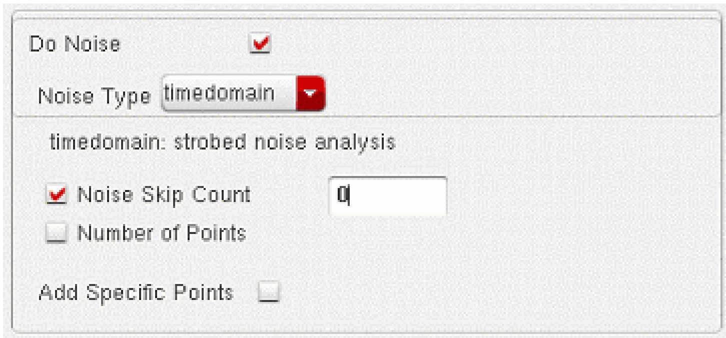
Remember that harmonic balance works in the frequency domain. To calculate a time-domain waveform, an ifft is required to convert the frequency-domain harmonics to a time-domain waveform. `Noise Type = Timedomain` provides a relative measurement of the noise at the individual points of the ifft. For a quantitative measurement, use the PM Jitter measurement.

First, set up the harmonic balance analysis. For more information, see the harmonic balance section at the beginning of this chapter.

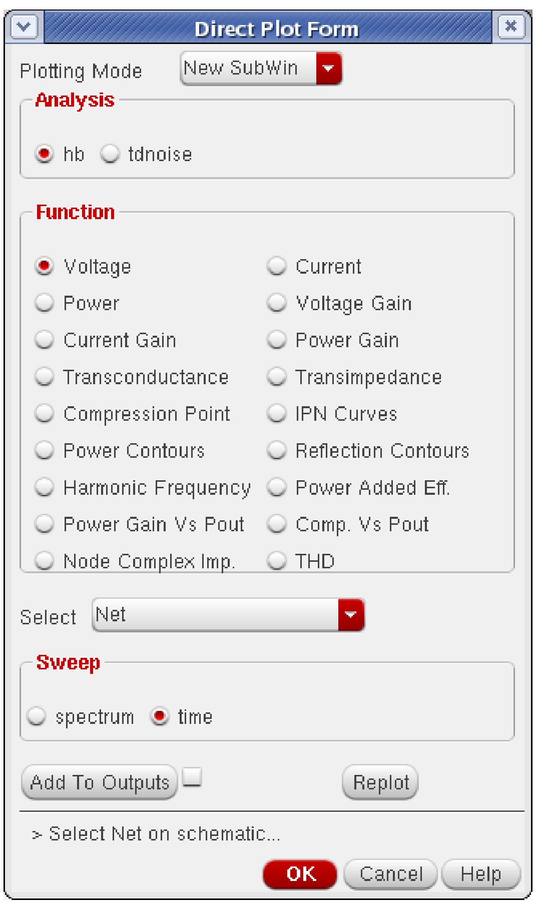


On the hbnoise *Choosing Analyses* form:

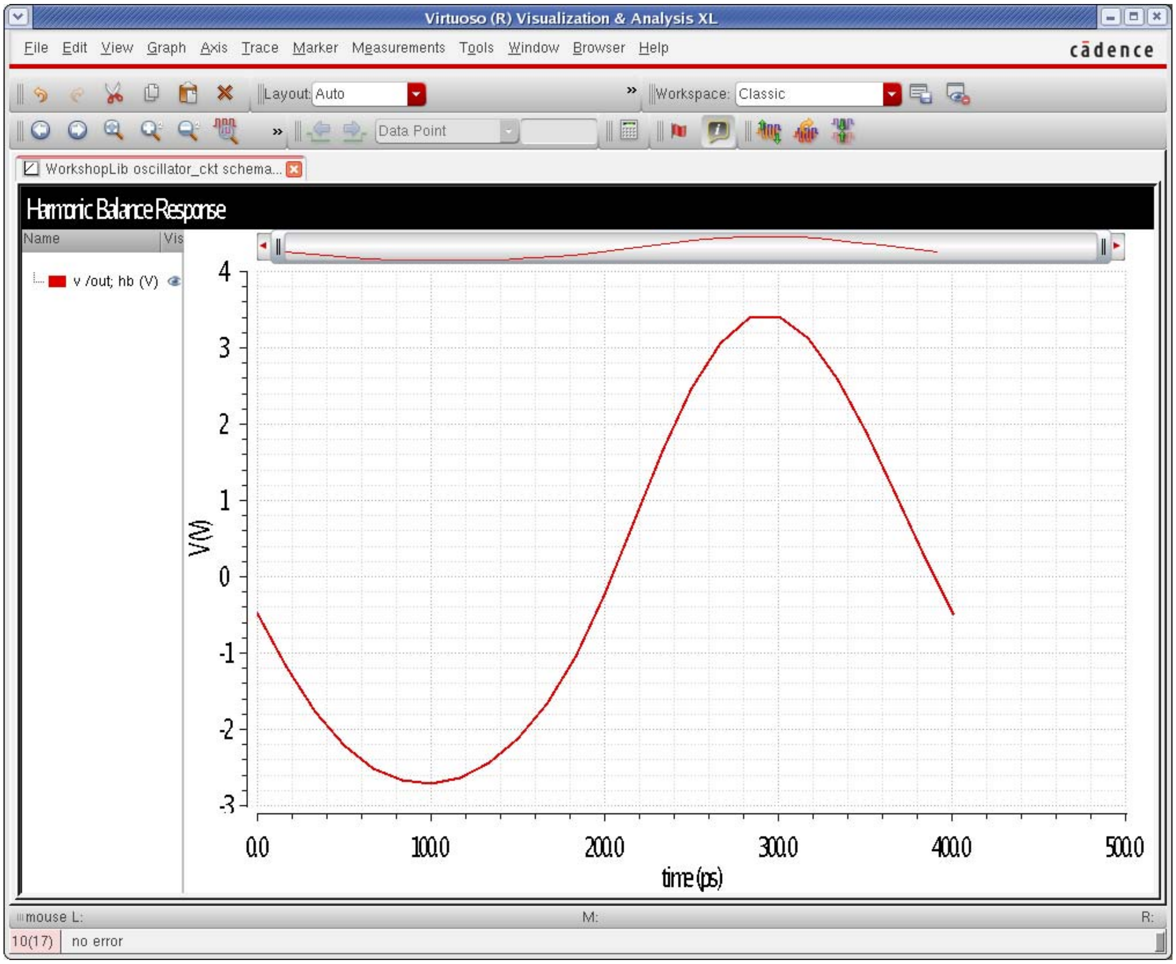
1. Select *timedomain* from the *Noise Type* drop-down list.



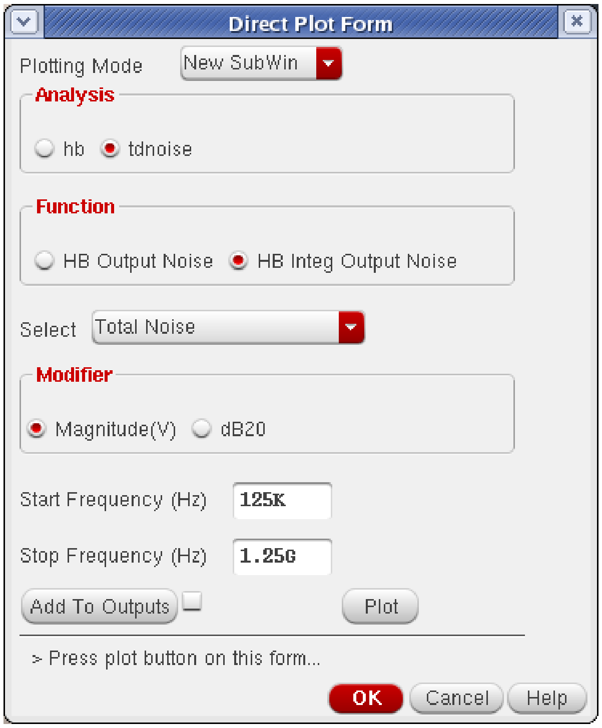
2. Select the *Noise Skip Count* check box and type 0 in the text box.
3. Run the simulation.
4. In ADE, select *Results - Direct Plot - Main Form*.



5. Select hb results.
6. Select Time.
7. Select the output net(s) in the schematic. The output waveform is displayed.



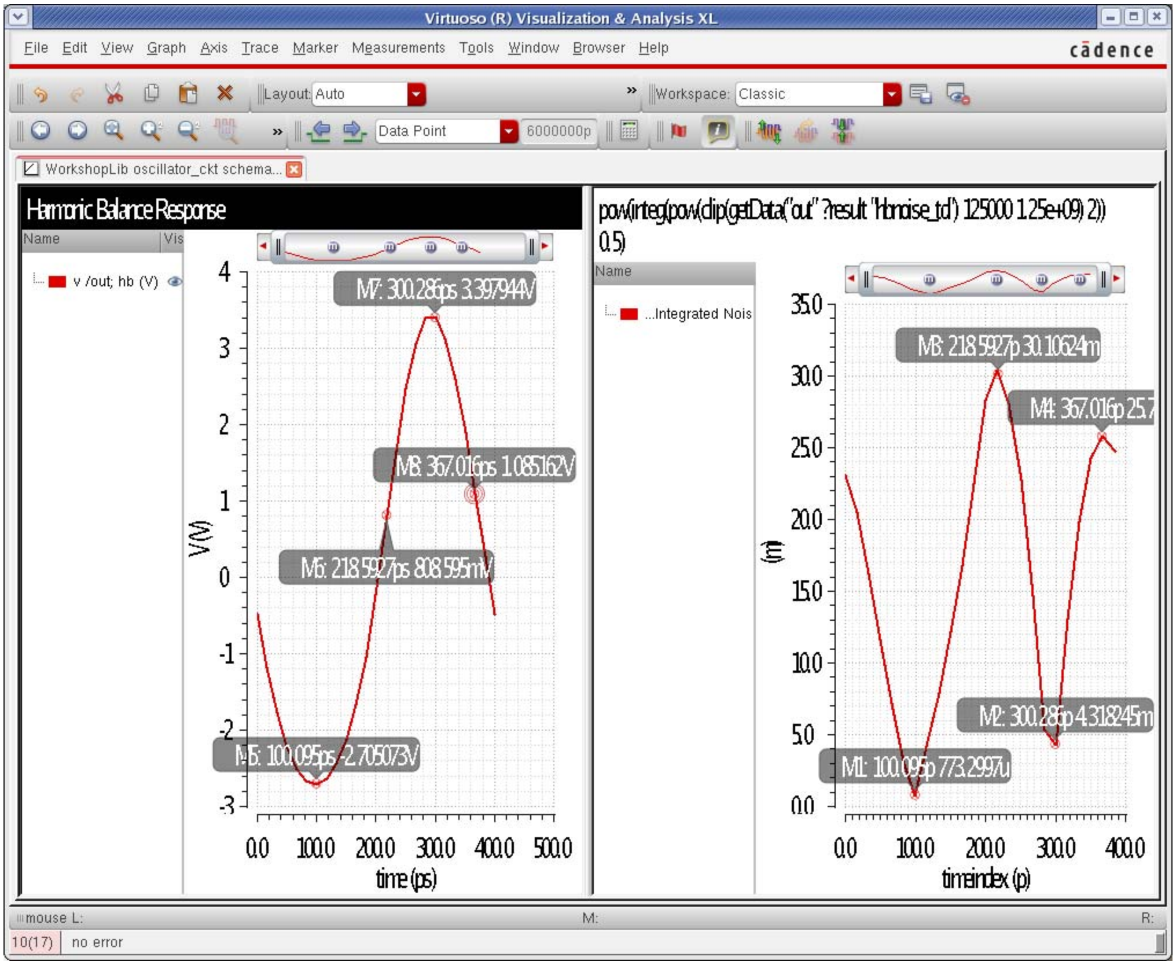
8. Select *tdnoise* results.



9. Select *HB Integ Output Noise*.

10. Type the frequency range form the hbnoise *Choosing Analyses* form.

11. Click *Plot*.



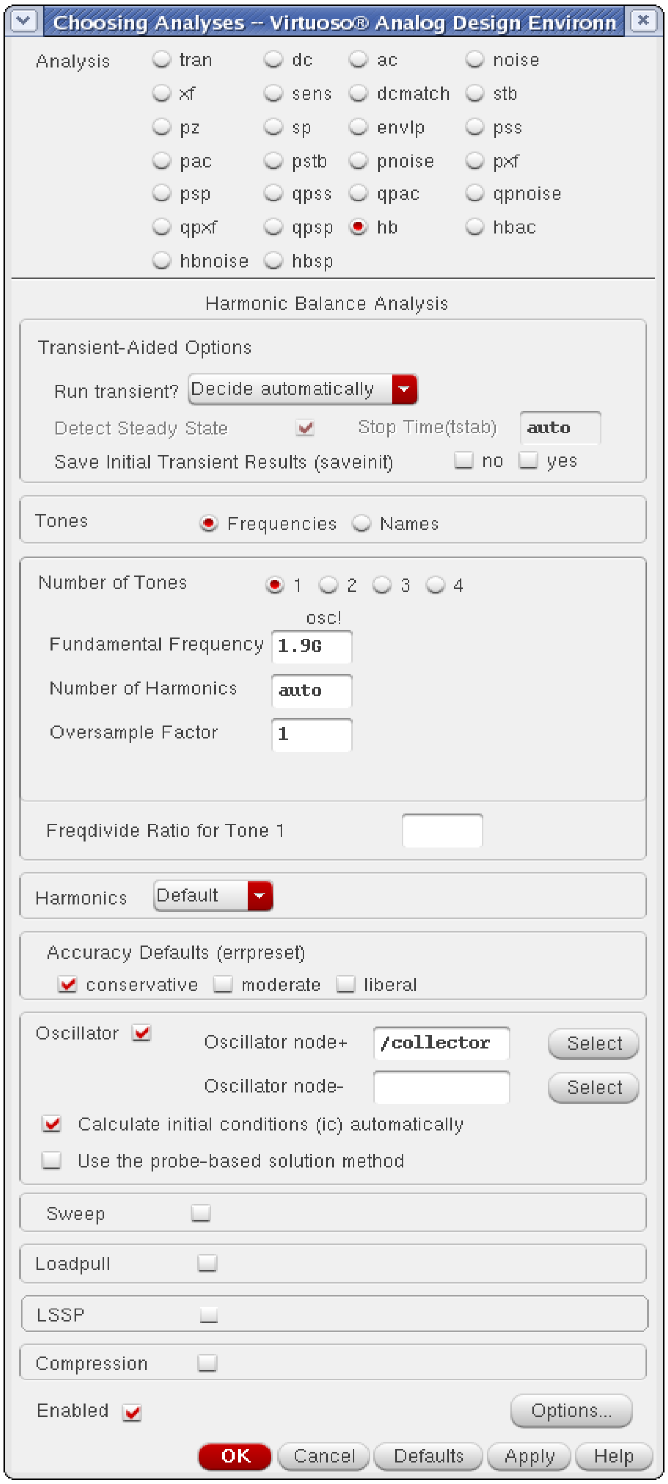
12. Position markers by moving your mouse cursor near the point you want to measure, and type m in the time-domain output as desired. In the above example, markers were placed at the peaks and valleys of the noise curve. Markers were then placed on the waveform to see where the noise is relatively large and relatively small. The largest noise occurs just above the zero crossing of the waveform.

Noise Type = Jitter

Oscillator Phase Noise Measurement (FM Jitter)

This selection allows for measurement of the cycle jitter, the cycle-to-cycle jitter, and the AM and PM components of phase noise. This is an averaged noise measurement over one cycle of the oscillator.

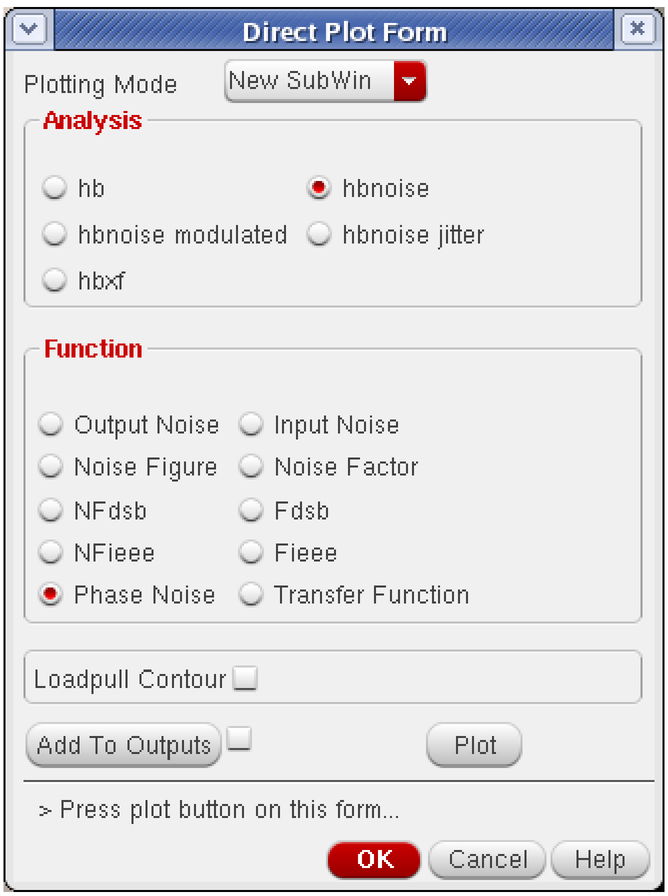
First, set up the harmonic balance analysis. For more information, see the harmonic balance section at the beginning of this chapter.



On the hbnoise *Choosing Analyses* form:

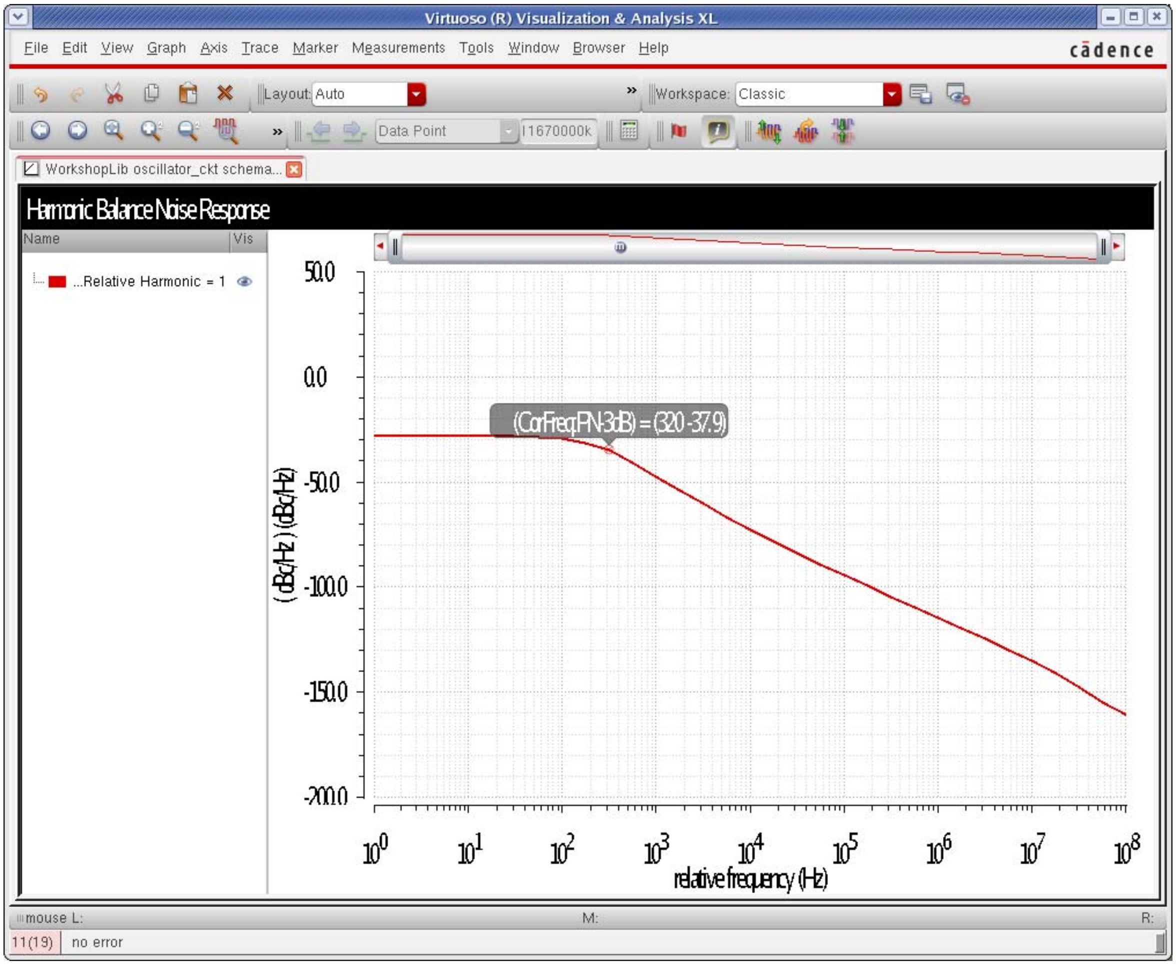
1. Select *Jitter* from the *Noise Type* drop-down list.

2. Select the *FM* radio button.
3. Run the simulation.
4. In the ADE, select *Results - Direct Plot - Main Form*.

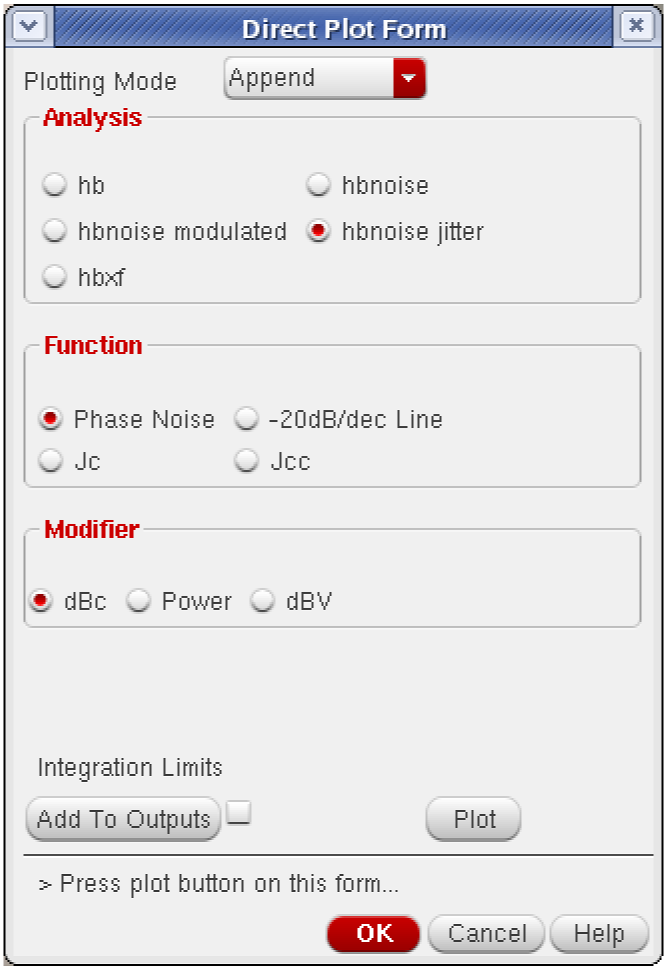


5. Select *hbnoise* results.
6. Select *Phase Noise*.

7. Click *Plot*. The single sideband phase noise is plotted. Note that the Lorentzian option is set to *yes*, which causes the phase noise to flatten out at low offset frequencies.

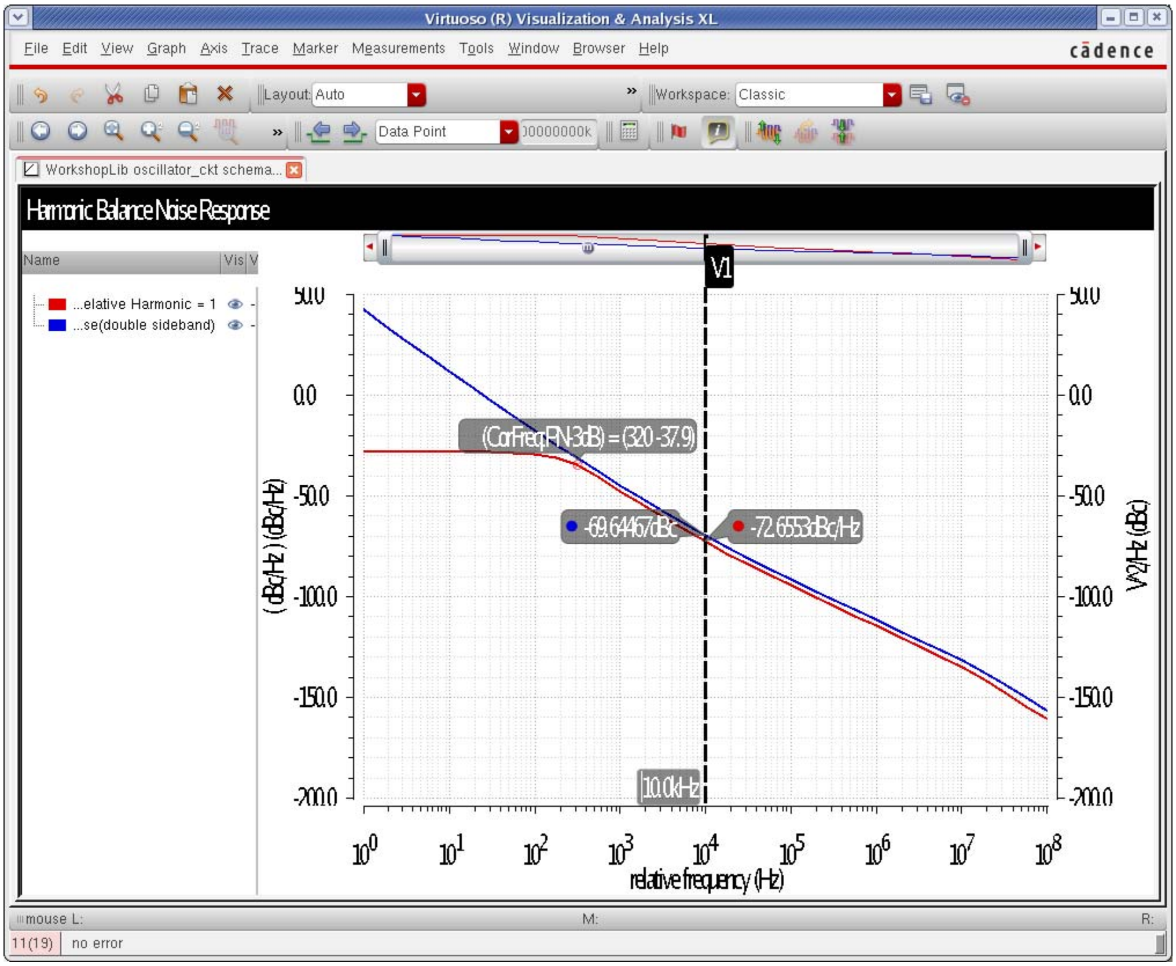


8. Select *hbnoise jitter* results.

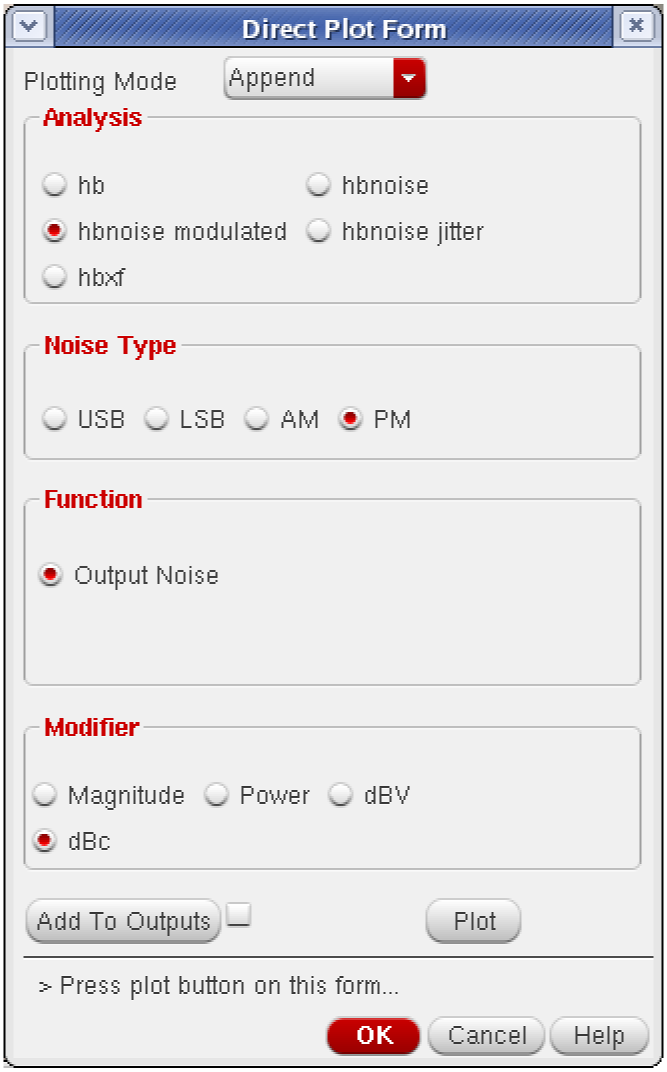


9. Select *Phase Noise*.

10. Click *Plot*. Note that this is double-sideband phase noise, and it does not level off at low frequency.

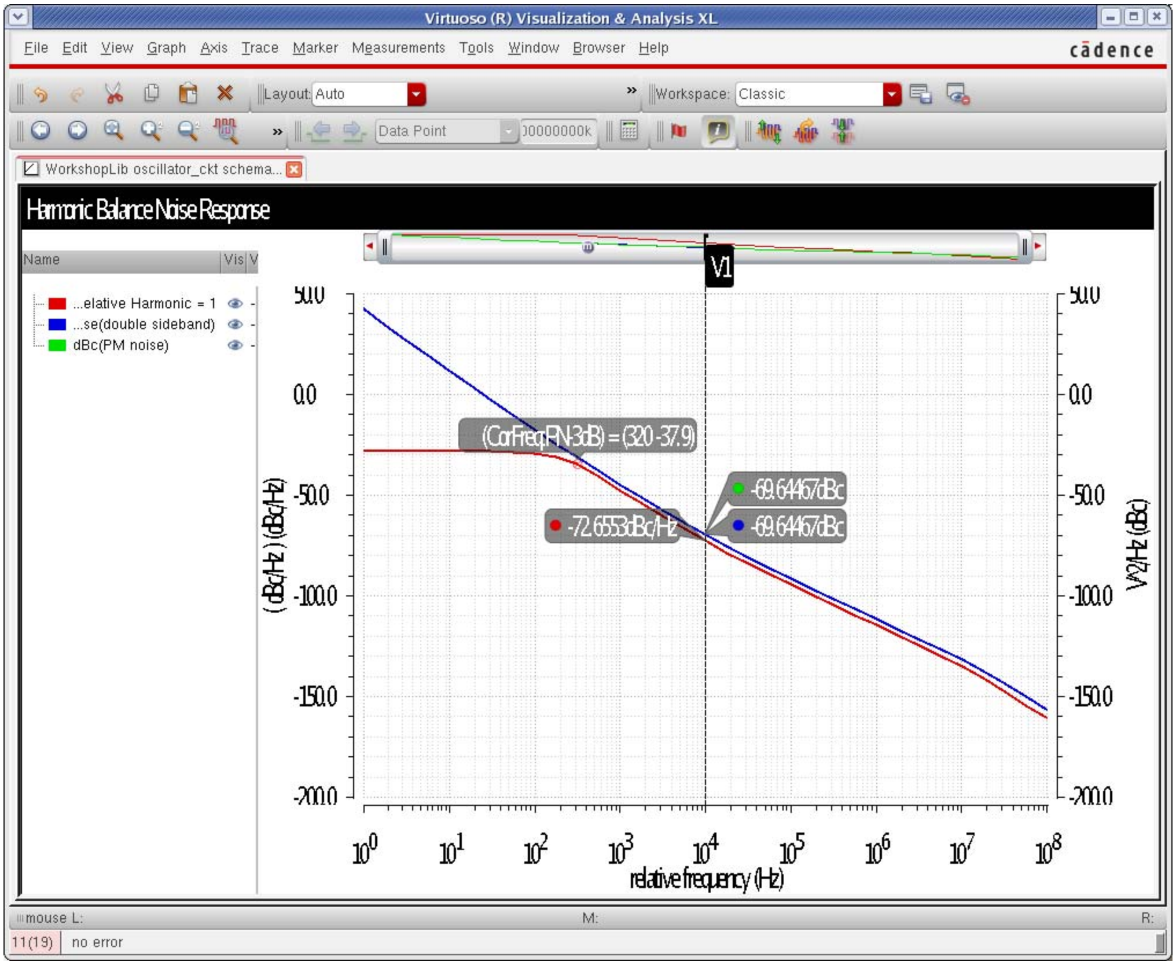


11. Select *hbnoise modulated* results.

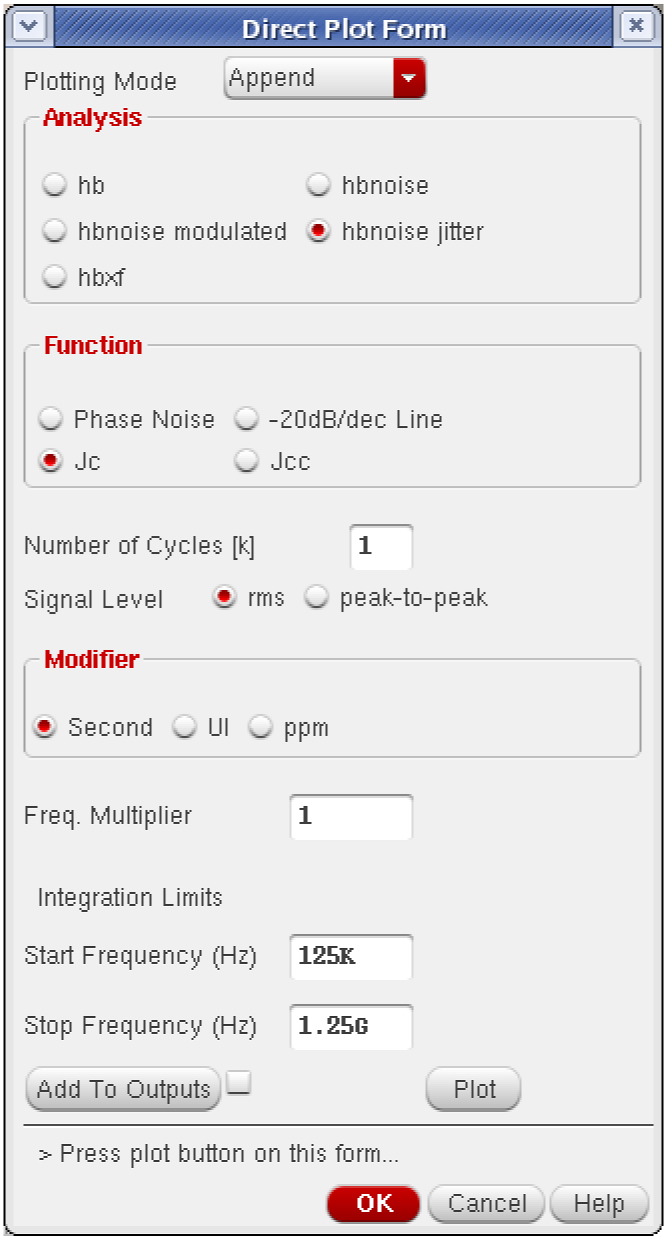


12. Select *PM* from the *Noise Type* section.

13. Click *Plot*. Note that for this oscillator, the PM component overlays the double-sideband phase noise, which indicates that the PM noise dominates.

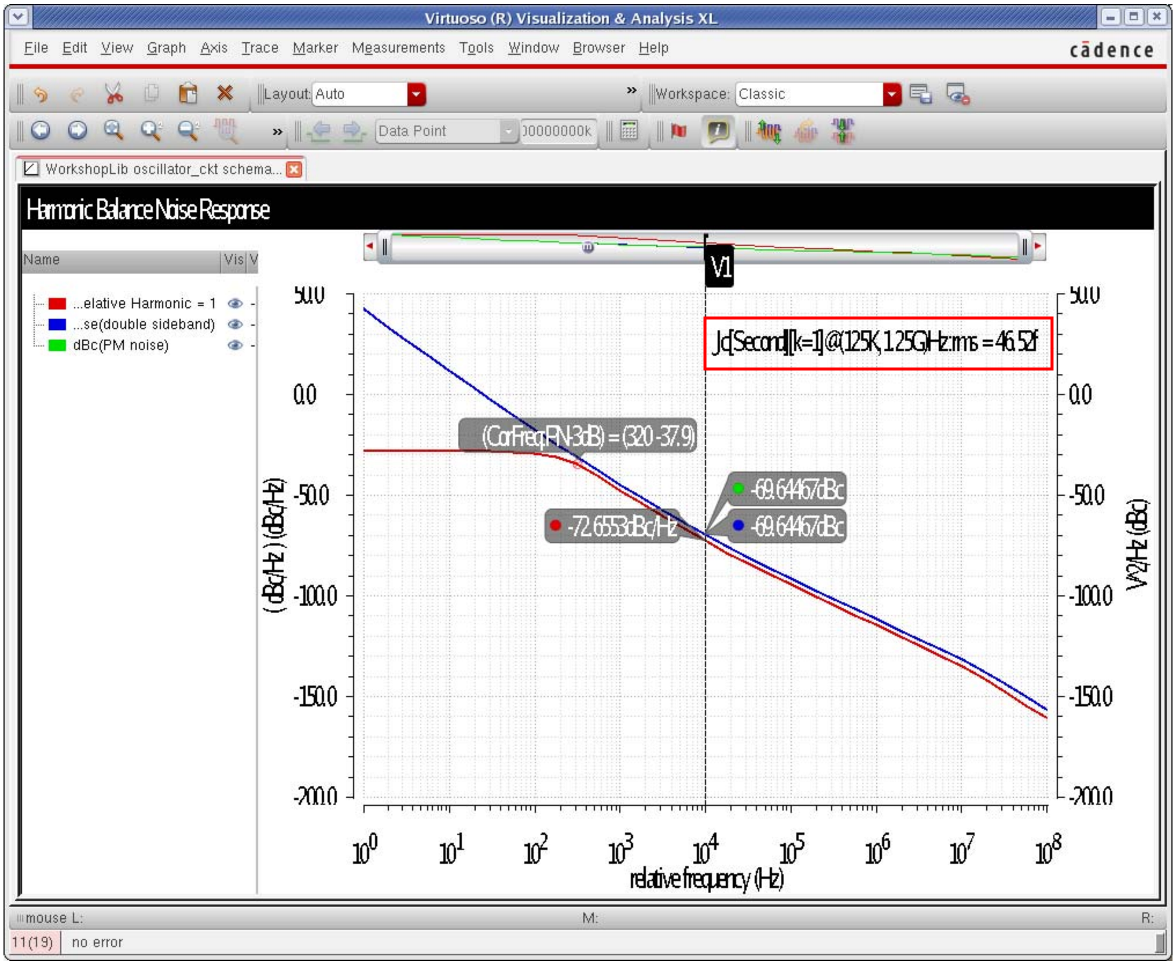


14. To plot the cycle or cycle-to-cycle jitter, select *hbnoise jitter* results.



15. Select *Jc* or *Jcc*.

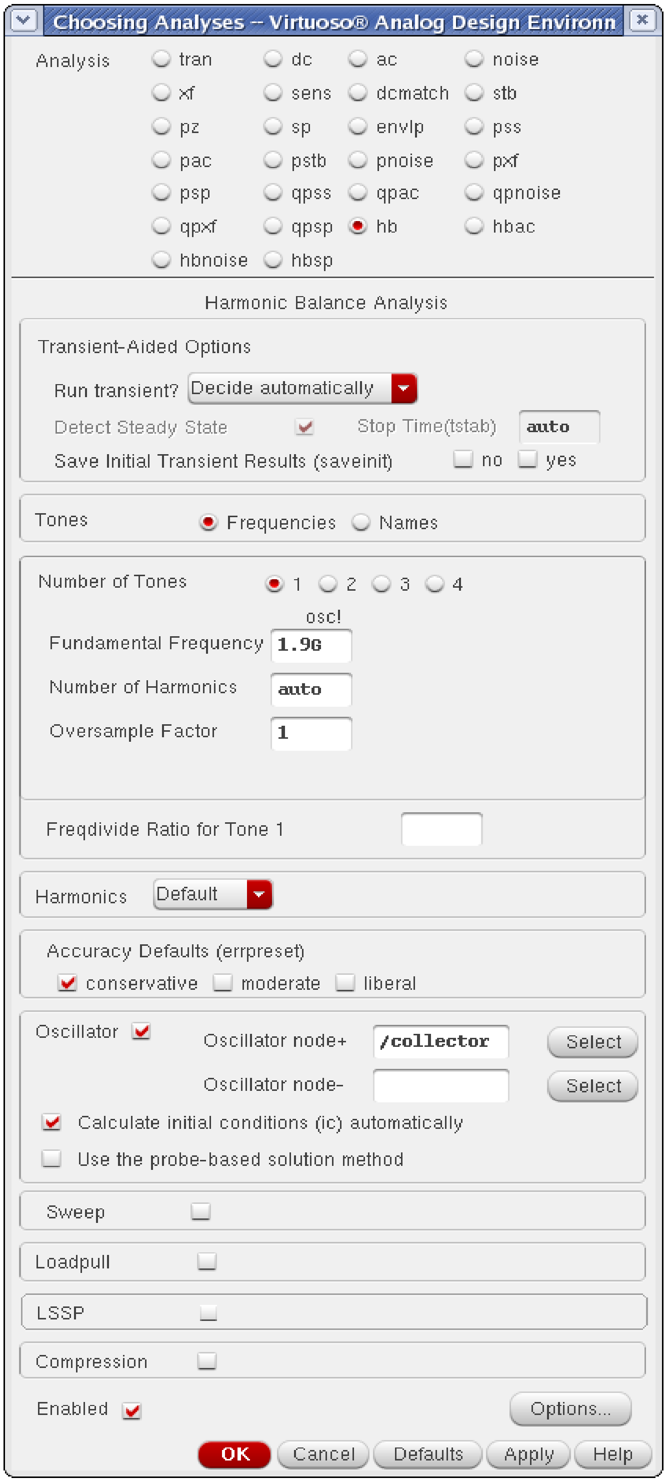
16. *Click* Plot. The cycle jitter result is plotted as a label in the waveform tool.



PM Jitter

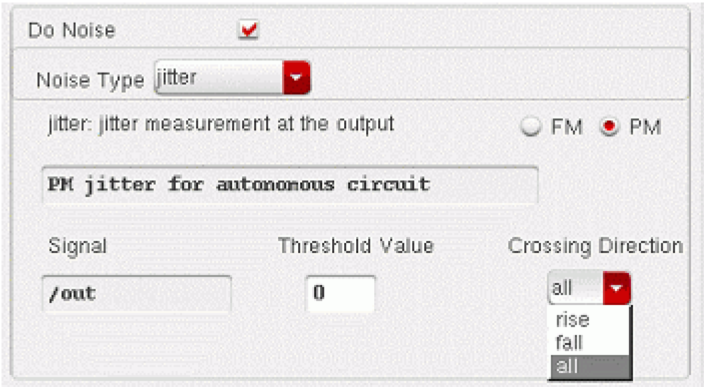
This allows for instantaneous measurement of jitter at individual voltages in the ifft waveform.

First, set up the harmonic balance analysis. For more information, see the harmonic balance section at the beginning of this chapter.



On the hbnoise *Choosing Analyses* form:

1. Select *jitter* from the *Noise Type* drop-down list.



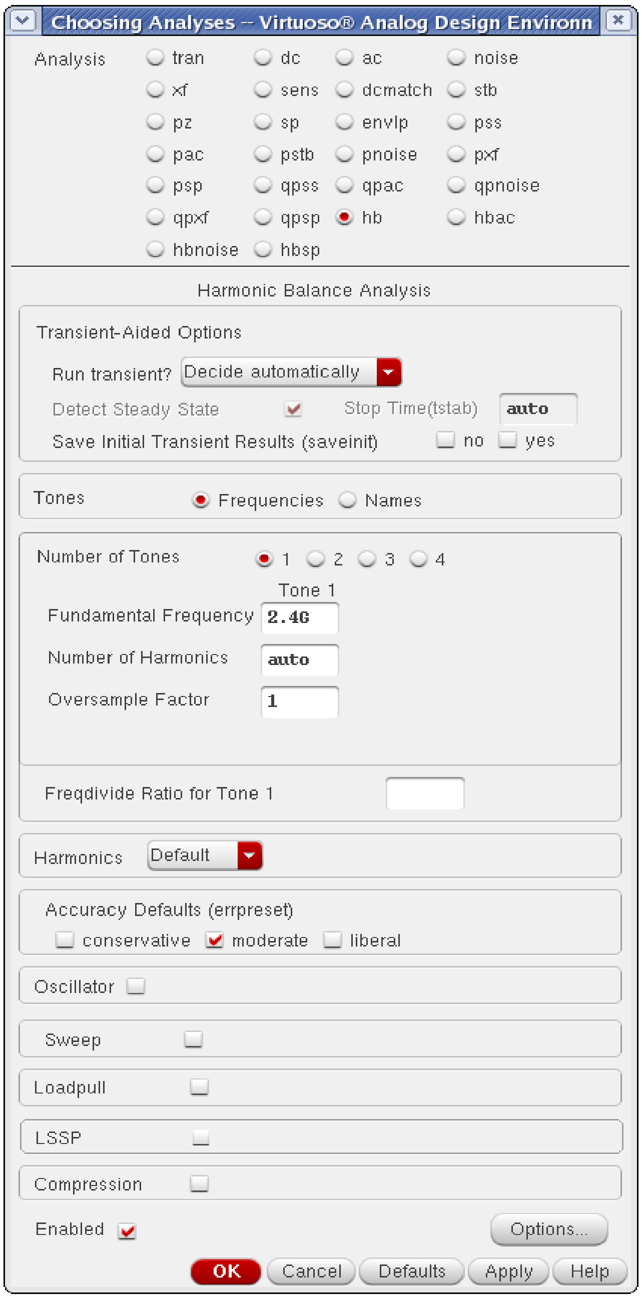
2. Specify the voltage for instantaneous noise measurement in the *Threshold Value* field. If you are not sure of the appropriate voltage, run the hb analysis, and plot the time-domain waveform for the output node(s).

3. Select a value from the *Crossing Direction* drop-down list.

Examples

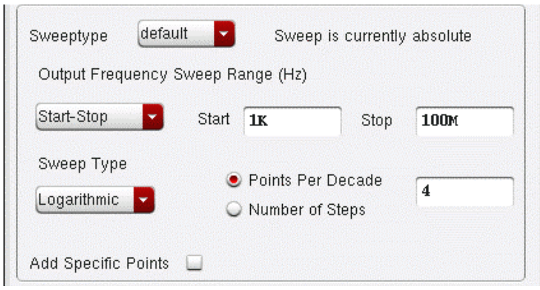
Driven Circuit Noise Setup

First, set up the harmonic balance analysis. For more information, see the Harmonic Balance section at the beginning of this chapter.



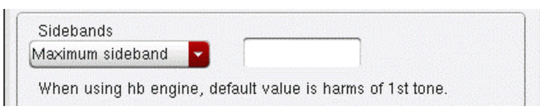
On the hbnoise *Choosing Analyses* form:

1. Specify the output frequency range of your circuit.

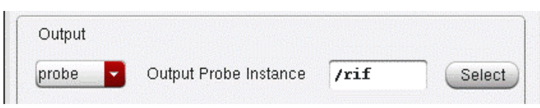


2. If you use log spacing, set about 3 to 5 points per decade in the *Points Per Decade* field.

3. In hbnoise, and pnoise or qpnoise when Harmonic Balance is set for the pss or qpss engine, you can leave the *Maximum sideband* field blank. Then, all the mixing products from the large-signal analysis will be considered.

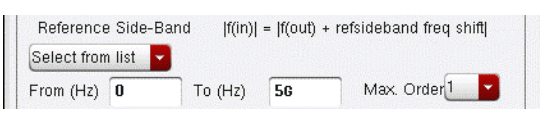


4. Select the resistor or port component used for the load resistance. This excludes the noise from this component for the noise figure calculation, but keeps it for the total noise output calculation.

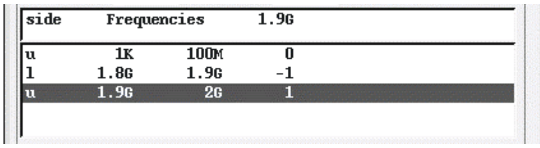


5. If you want a noise figure calculation, you must use a port as the input for the circuit. Set the instance name.

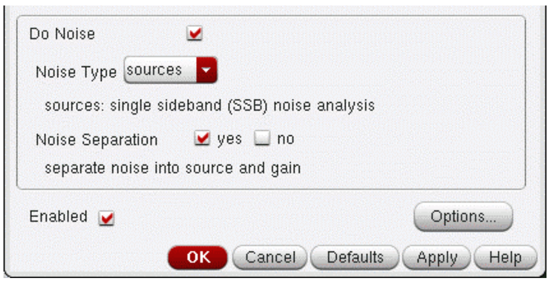
6. The easiest way to select the reference sideband is to set it to choose *Select from list*.



7. Specify the passband frequency range that you want for the noise figure and input-referred noise calculations.



8. If you want to see the noise contributions from the individual frequencies and the different device noise calculations individually on the output of the noise analysis, enable noise separation.



9. When you are done with the setup, run the simulation.

Noise Summary

The noise summary provides information about the devices that contribute noise to the output. This information is available any time an hbnoise simulation completes.

In the ADE, select *Results - Print - Noise Summary*. If noise separation is enabled, there will be two choices at the very top. `Hbnoise_src` has information about the noise currents at the individual noise sources at the noise source. Hbnoise has information about noise at the output of the circuit.

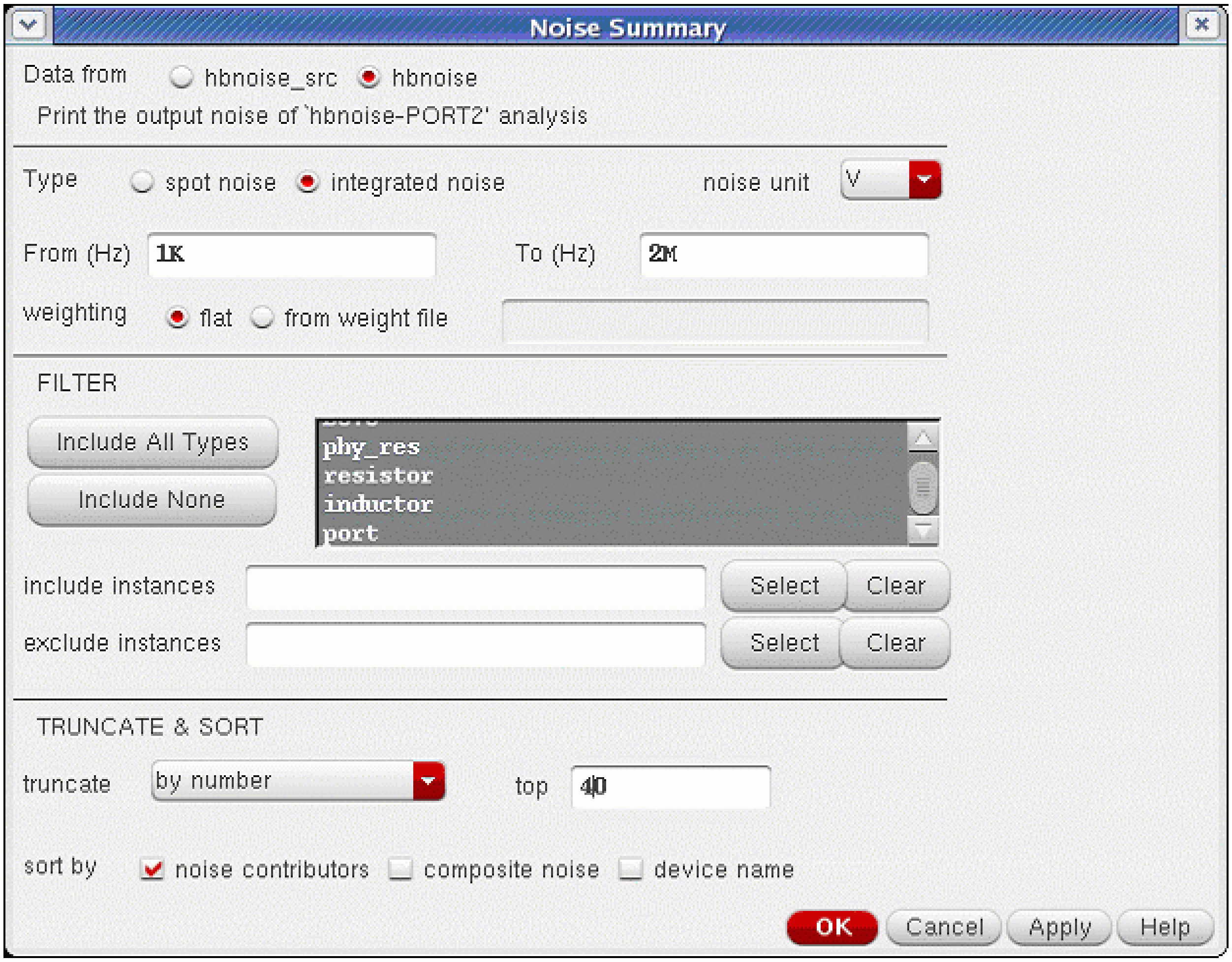
If you want the noise contributors at a single frequency, select *spot noise*. If you want to have the noise integrated over a frequency range, select *integrated noise*.

If you want noise in volts, set the *noise unit* to *V.*

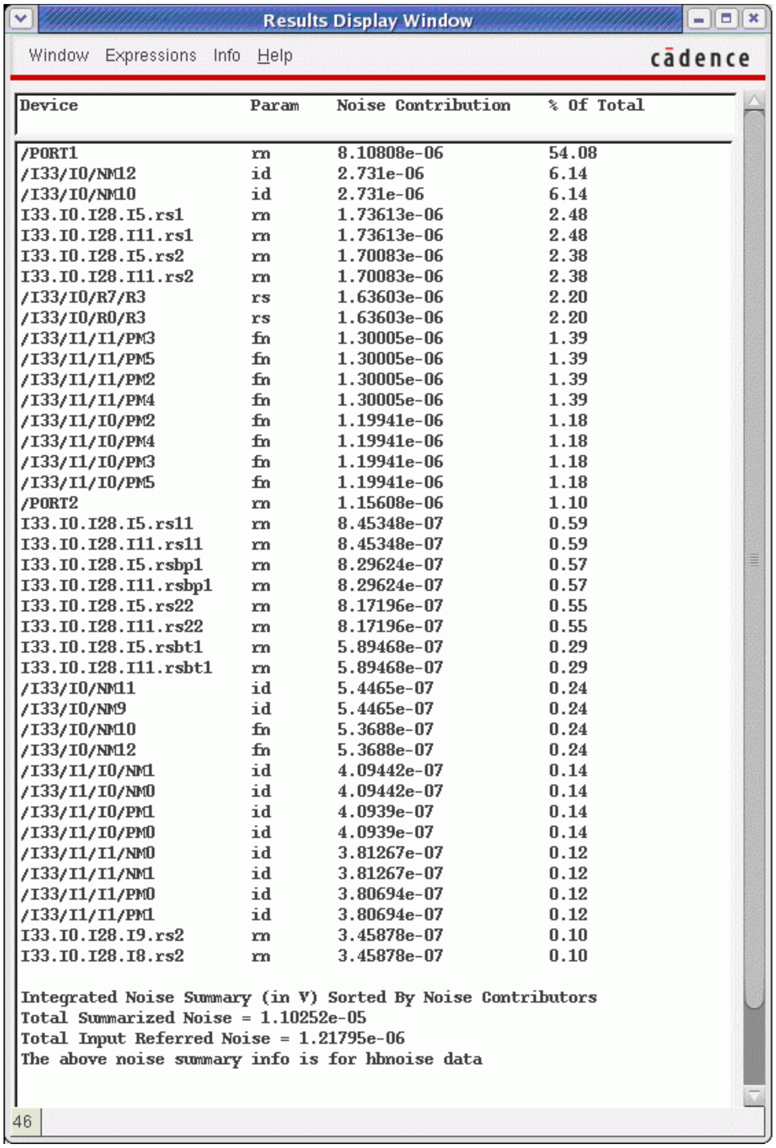
Generally, *Include All Types* should be selected. If you just want the noise from specific types of noise generators, you can select them from the list.

Truncate just applies to the list. The total input and output-referred noise at the bottom of the noise summary output always includes all the noise from everything in the circuit.

The figure below is the noise summary window with the sort being noise contributors. Then there are three windows that show the three different sorted outputs.



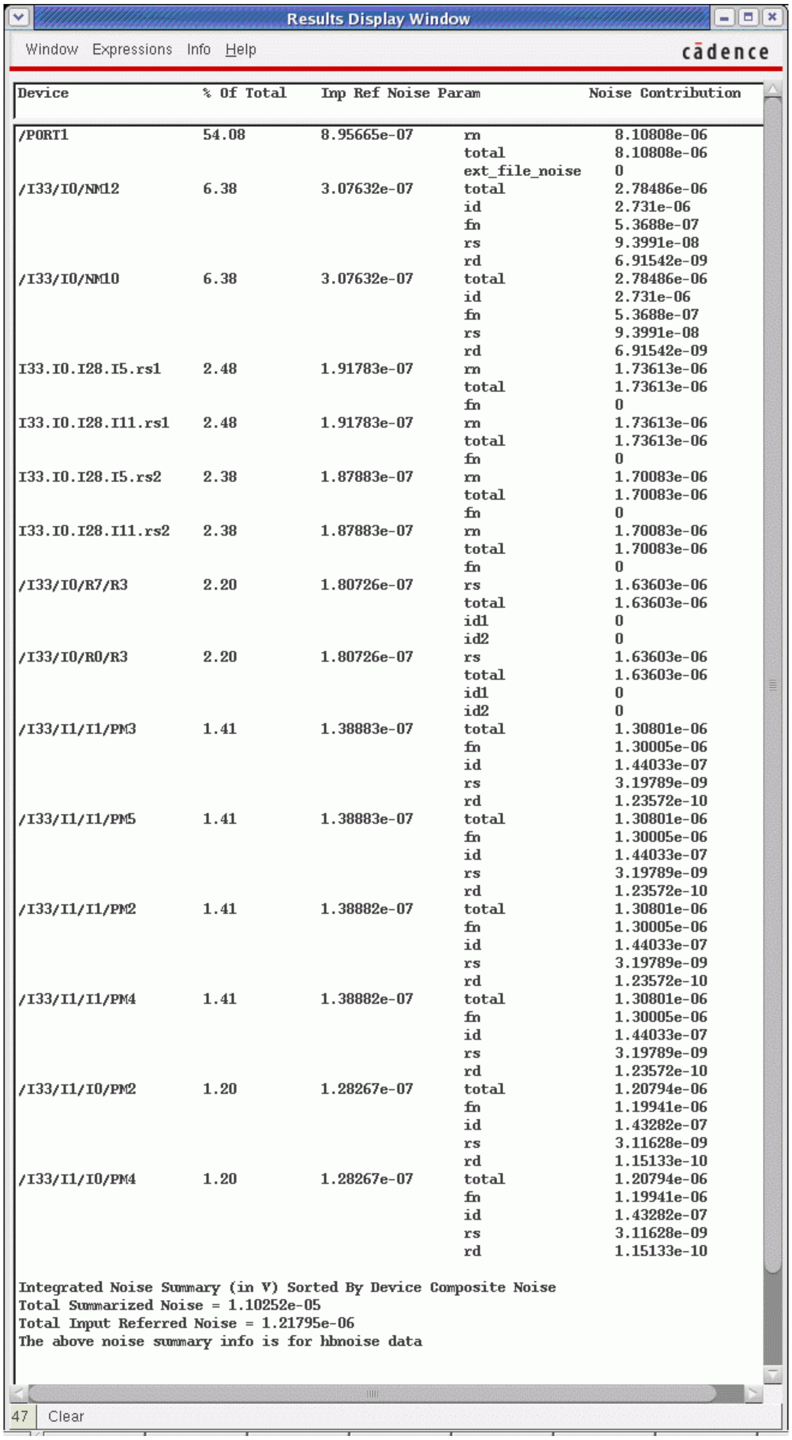
The output below shows the noise summary sorted by noise contributors. The largest noise contributor is on the top of the list.



Device	Param	Noise Contribution	% Of Total
/PORT1	rn	8.10808e-06	54.08
/I33/I0/NM12	id	2.731e-06	6.14
/I33/I0/NM10	id	2.731e-06	6.14
I33.I0.I28.I5.rs1	rn	1.73613e-06	2.48
I33.I0.I28.I11.rs1	rn	1.73613e-06	2.48
I33.I0.I28.I5.rs2	rn	1.70083e-06	2.38
I33.I0.I28.I11.rs2	rn	1.70083e-06	2.38
/I33/I0/R7/R3	rs	1.63603e-06	2.20
/I33/I0/R0/R3	rs	1.63603e-06	2.20
/I33/I1/I1/PM3	fn	1.30005e-06	1.39
/I33/I1/I1/PM5	fn	1.30005e-06	1.39
/I33/I1/I1/PM2	fn	1.30005e-06	1.39
/I33/I1/I1/PM4	fn	1.30005e-06	1.39
/I33/I1/I0/PM2	fn	1.19941e-06	1.18
/I33/I1/I0/PM4	fn	1.19941e-06	1.18
/I33/I1/I0/PM3	fn	1.19941e-06	1.18
/I33/I1/I0/PM5	fn	1.19941e-06	1.18
/PORT2	rn	1.15608e-06	1.10
I33.I0.I28.I5.rs11	rn	8.45348e-07	0.59
I33.I0.I28.I11.rs11	rn	8.45348e-07	0.59
I33.I0.I28.I5.rsbp1	rn	8.29624e-07	0.57
I33.I0.I28.I11.rsbp1	rn	8.29624e-07	0.57
I33.I0.I28.I5.rs22	rn	8.17196e-07	0.55
I33.I0.I28.I11.rs22	rn	8.17196e-07	0.55
I33.I0.I28.I5.rsbt1	rn	5.89468e-07	0.29
I33.I0.I28.I11.rsbt1	rn	5.89468e-07	0.29
/I33/I0/NM11	id	5.4465e-07	0.24
/I33/I0/NM9	id	5.4465e-07	0.24
/I33/I0/NM10	fn	5.3688e-07	0.24
/I33/I0/NM12	fn	5.3688e-07	0.24
/I33/I1/I0/NM1	id	4.09442e-07	0.14
/I33/I1/I0/NM0	id	4.09442e-07	0.14
/I33/I1/I0/PM1	id	4.0939e-07	0.14
/I33/I1/I0/PM0	id	4.0939e-07	0.14
/I33/I1/I1/NM0	id	3.81267e-07	0.12
/I33/I1/I1/NM1	id	3.81267e-07	0.12
/I33/I1/I1/PM0	id	3.80694e-07	0.12
/I33/I1/I1/PM1	id	3.80694e-07	0.12
I33.I0.I28.I9.rs2	rn	3.45878e-07	0.10
I33.I0.I28.I8.rs2	rn	3.45878e-07	0.10

Integrated Noise Summary (in V) Sorted By Noise Contributors
Total Summarized Noise = 1.10252e-05
Total Input Referred Noise = 1.21795e-06
The above noise summary info is for hbnoise data

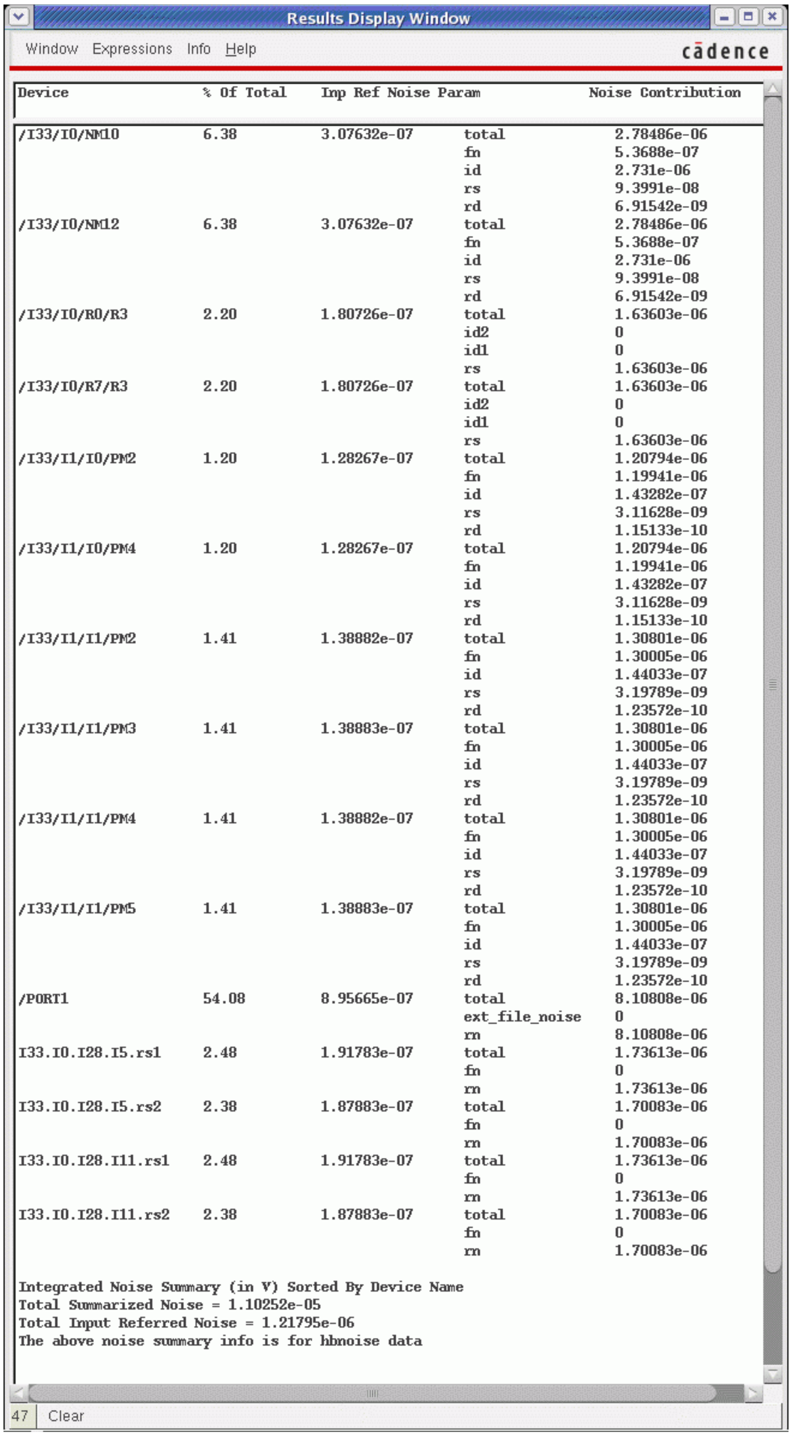
The output below shows the noise summary sorted by composite noise. The largest total noise instance is on the top of the list, and the individual noise contributors for that component are shown below the instance name.



Device	% Of Total	Inp Ref Noise	Param	Noise Contribution
/PORT1	54.08	8.95665e-07	rn	8.10808e-06
			total	8.10808e-06
			ext_file_noise	0
/I33/I0/NM12	6.38	3.07632e-07	total	2.78486e-06
			id	2.731e-06
			fn	5.3688e-07
			rs	9.3991e-08
			rd	6.91542e-09
/I33/I0/NM10	6.38	3.07632e-07	total	2.78486e-06
			id	2.731e-06
			fn	5.3688e-07
			rs	9.3991e-08
			rd	6.91542e-09
I33.I0.I28.I5.rs1	2.48	1.91783e-07	rn	1.73613e-06
			total	1.73613e-06
			fn	0
I33.I0.I28.I11.rs1	2.48	1.91783e-07	rn	1.73613e-06
			total	1.73613e-06
			fn	0
I33.I0.I28.I5.rs2	2.38	1.87883e-07	rn	1.70083e-06
			total	1.70083e-06
			fn	0
I33.I0.I28.I11.rs2	2.38	1.87883e-07	rn	1.70083e-06
			total	1.70083e-06
			fn	0
/I33/I0/R7/R3	2.20	1.80726e-07	rs	1.63603e-06
			total	1.63603e-06
			id1	0
			id2	0
/I33/I0/R0/R3	2.20	1.80726e-07	rs	1.63603e-06
			total	1.63603e-06
			id1	0
			id2	0
/I33/I1/I1/PM3	1.41	1.38883e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM5	1.41	1.38883e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM2	1.41	1.38882e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM4	1.41	1.38882e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I0/PM2	1.20	1.28267e-07	total	1.20794e-06
			fn	1.19941e-06
			id	1.43282e-07
			rs	3.11628e-09
			rd	1.15133e-10
/I33/I1/I0/PM4	1.20	1.28267e-07	total	1.20794e-06
			fn	1.19941e-06
			id	1.43282e-07
			rs	3.11628e-09
			rd	1.15133e-10

Integrated Noise Summary (in V) Sorted By Device Composite Noise
Total Summarized Noise = 1.10252e-05
Total Input Referred Noise = 1.21795e-06
The above noise summary info is for hbnoise data

The output below shows the noise summary sorted by device name. The devices are listed in alphabetical order.



Device	% Of Total	Inp Ref Noise	Param	Noise Contribution
/I33/I0/NM10	6.38	3.07632e-07	total	2.78486e-06
			fn	5.3688e-07
			id	2.731e-06
			rs	9.3991e-08
			rd	6.91542e-09
/I33/I0/NM12	6.38	3.07632e-07	total	2.78486e-06
			fn	5.3688e-07
			id	2.731e-06
			rs	9.3991e-08
			rd	6.91542e-09
/I33/I0/R0/R3	2.20	1.80726e-07	total	1.63603e-06
			id2	0
			id1	0
			rs	1.63603e-06
/I33/I0/R7/R3	2.20	1.80726e-07	total	1.63603e-06
			id2	0
			id1	0
			rs	1.63603e-06
/I33/I1/I0/PM2	1.20	1.28267e-07	total	1.20794e-06
			fn	1.19941e-06
			id	1.43282e-07
			rs	3.11628e-09
			rd	1.15133e-10
/I33/I1/I0/PM4	1.20	1.28267e-07	total	1.20794e-06
			fn	1.19941e-06
			id	1.43282e-07
			rs	3.11628e-09
			rd	1.15133e-10
/I33/I1/I1/PM2	1.41	1.38882e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM3	1.41	1.38883e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM4	1.41	1.38882e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/I33/I1/I1/PM5	1.41	1.38883e-07	total	1.30801e-06
			fn	1.30005e-06
			id	1.44033e-07
			rs	3.19789e-09
			rd	1.23572e-10
/PORT1	54.08	8.95665e-07	total	8.10808e-06
			ext_file_noise	0
			rn	8.10808e-06
I33.I0.I28.I5.rs1	2.48	1.91783e-07	total	1.73613e-06
			fn	0
			rn	1.73613e-06
I33.I0.I28.I5.rs2	2.38	1.87883e-07	total	1.70083e-06
			fn	0
			rn	1.70083e-06
I33.I0.I28.I11.rs1	2.48	1.91783e-07	total	1.73613e-06
			fn	0
			rn	1.73613e-06
I33.I0.I28.I11.rs2	2.38	1.87883e-07	total	1.70083e-06
			fn	0
			rn	1.70083e-06

Integrated Noise Summary (in V) Sorted By Device Name
Total Summarized Noise = 1.10252e-05
Total Input Referred Noise = 1.21795e-06
The above noise summary info is for hbnoise data

For all of the noise summaries, here is a table of the abbreviations used in the *Param* column:

Resistor and Inductor:

Parameter	Meaning
`rn`	resistor thermal noise
`fn`	Flicker noise

Diode:

Parameter	Meaning
`id`	Diode current shot noise
`rs`	Series parasitic resistor noise

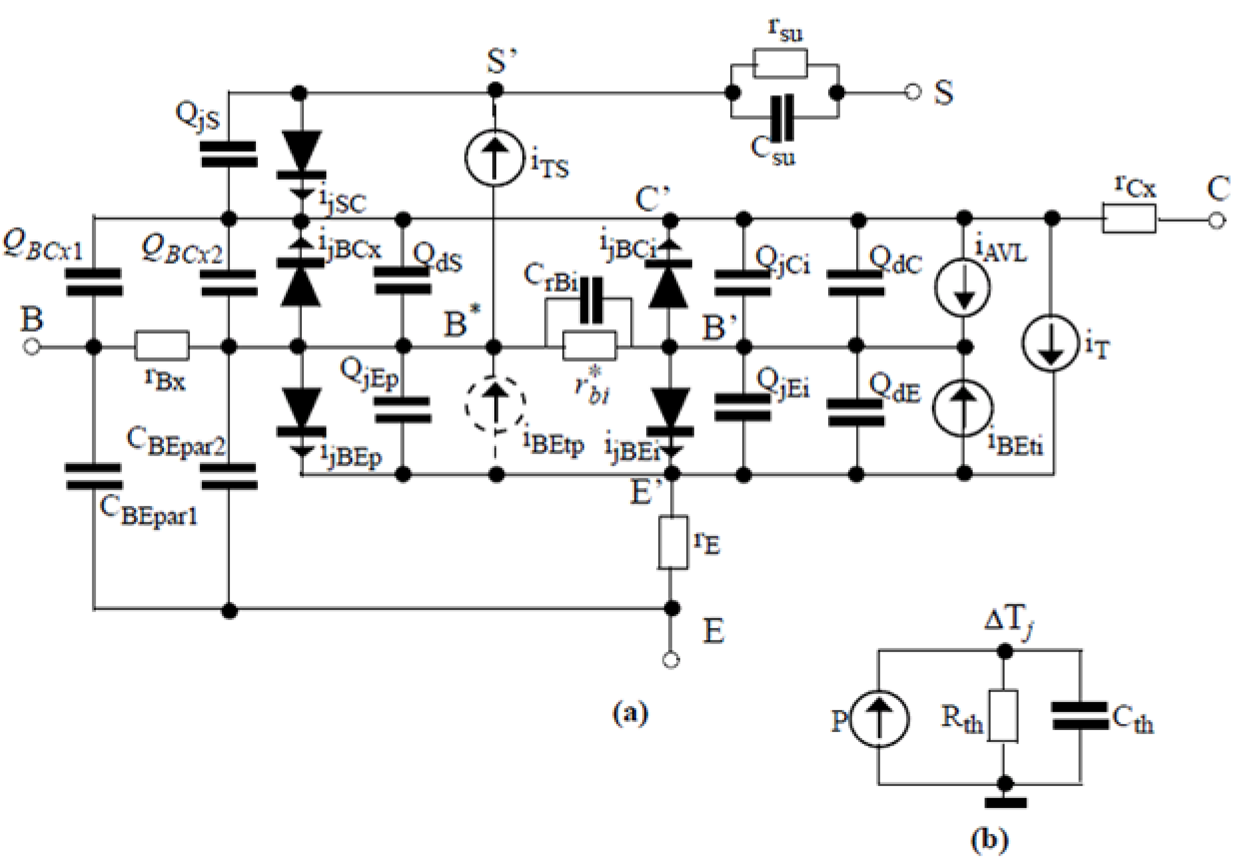
Gummel-Poon BJT:

Parameter	Meaning
`fn`	Flicker noise
`ic`	Collector current shot noise
`ib`	Base current shot noise
`rc`	Thermal noise from the collector parasitic resistor
`rb`	Thermal noise from the base parasitic resistor
`re`	Thermal noise from the emitter parasitic resistor

HiCUM:

Parameter	Meaning
`ib`	Base current shot noise
`ic`	Collector current shot noise
`rbi`	Thermal noise from the internal base parasitic resistor (See diagram below)
`rbx`	Thermal noise from the external base parasitic resistor (See diagram below)
`rcx`	Collector parasitic resistor thermal noise (See diagram below)
`re`	Emitter parasitic resistor thermal noise (See diagram below)
`fn`	Flicker noise

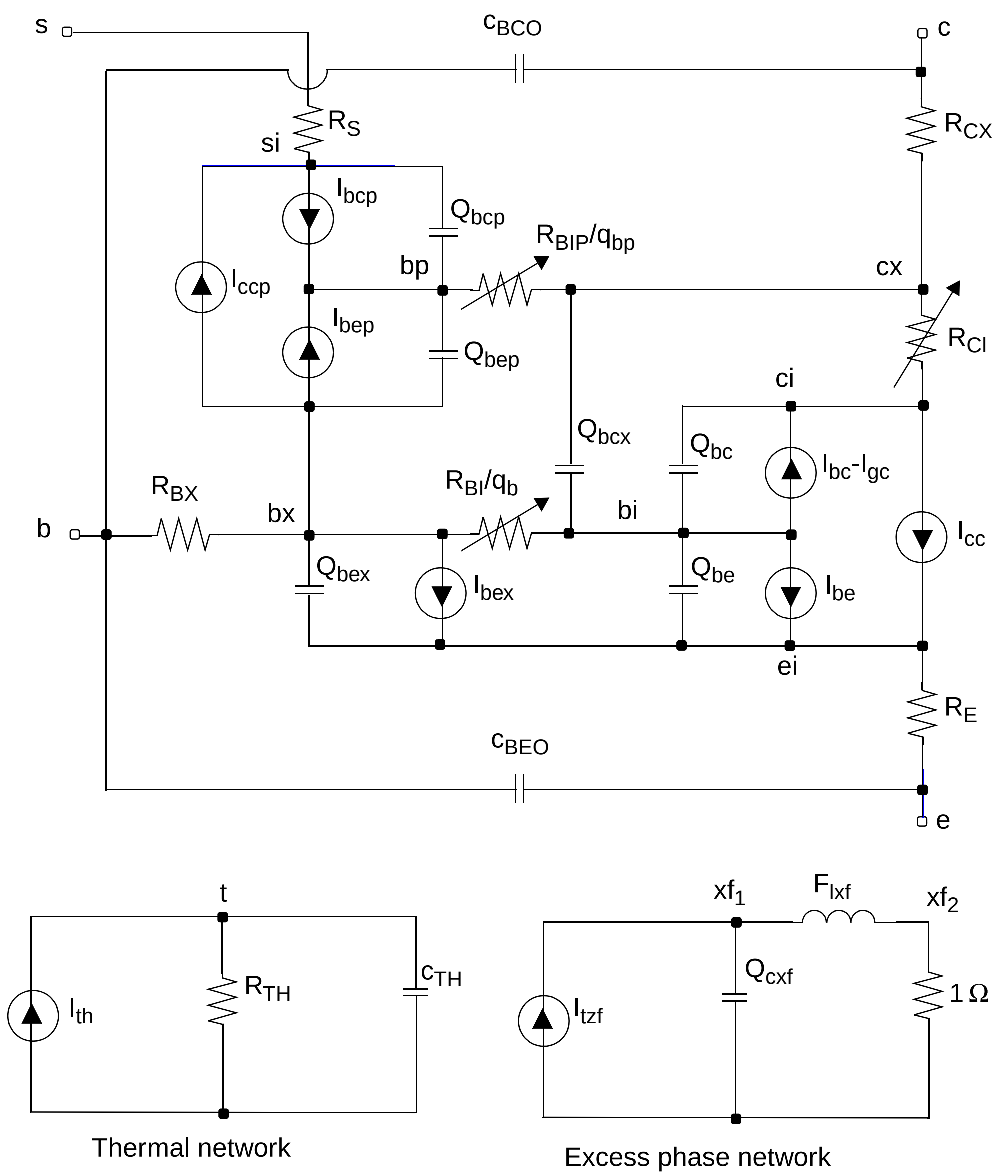
Figure 3-1 HiCUM Schematic Diagram



VBIC:

Parameter	Meaning
`rbi`	Thermal noise in resistor Rbi (See diagram below)
`rbx`	Thermal noise in resistor Rbx (See diagram below)
`rci`	Thermal noise in resistor Rci (See diagram below)
`re`	Thermal noise in resistor Re (See diagram below)
`rs`	Thermal noise in resistor Rs (See diagram below)
`itzf`	Collector current shot noise
`ibe`	Total base current shot noise
`ibex`	Node Bx to E base current shot noise (See diagram below)
`fn`	Ibe flicker noise (See diagram below)
`fnx`	Ibex flicker noise (See diagram below)
`iccp`	Parasitic transport collector to emitter current noise (See diagram below)
`ibep`	Parasitic transport base to emitter current noise (See diagram below)
`fnp`	Ibep flicker Noise (See diagram below)

Figure 3-2 VBIC Schematic Diagram



BSIM3:

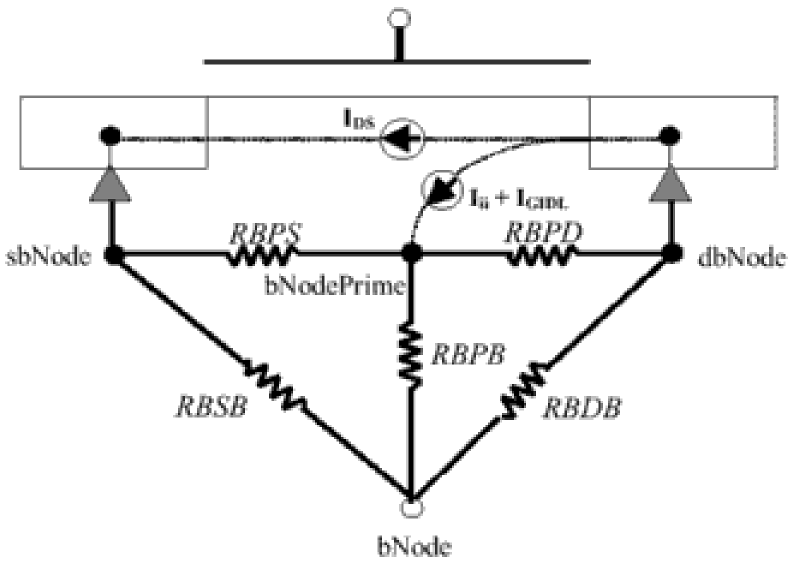
Parameter	**Meaning**
`fn`	Flicker Noise

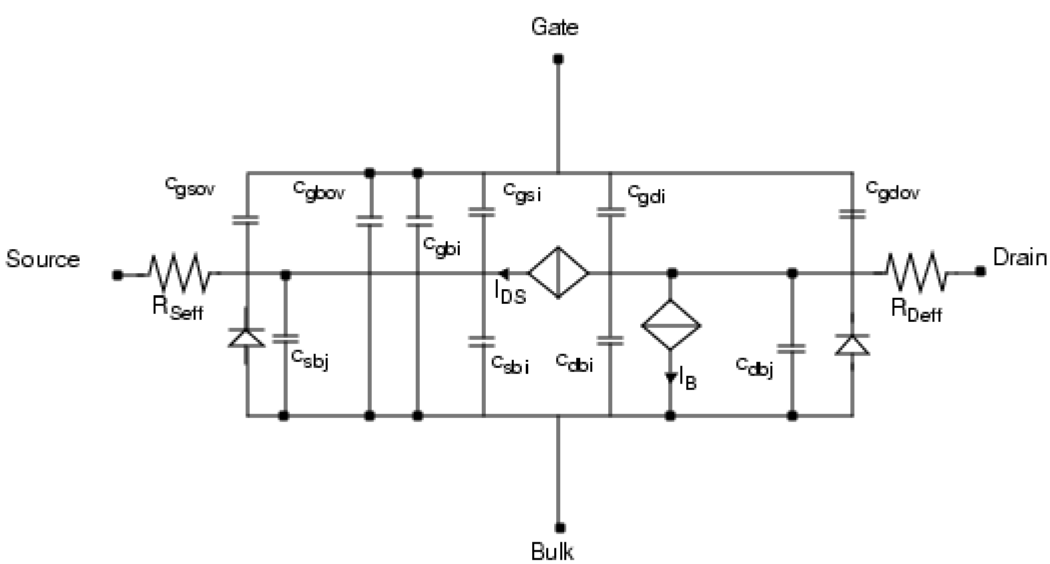
Parameter	Meaning
`id`	Thermal noise from the resistive drain current
`rs`	Thermal noise from the source parasitic resistor
`rd`	Thermal noise from the drain parasitic resistor

BSIM4:

Parameter	Meaning
`fn`	Flicker noise
`id`	Thermal noise from the resistive drain current noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`igd`	Gate-to-drain tunneling current noise
`igs`	Gate-to-source tunneling current noise
`igb`	Gate-to-bulk tunneling current noise
`rbdb`	Resistor thermal noise between dbNode and bNode (See diagram below)
`rbpb`	Resistor thermal noise between bNode and bNodePrime (See diagram below)
`rbpd`	Resistor thermal noise between Bulk and dbNode (See diagram below)
`rbps`	Resistor thermal noise between bulk and bNode prime
`rbsb`	Resistor thermal noise between sbNode and bNode (See diagram below)
`rgbi`	Gate Bias-Independent Resistor thermal noise

Figure 3-3 BSIM4 Schematic Diagrams

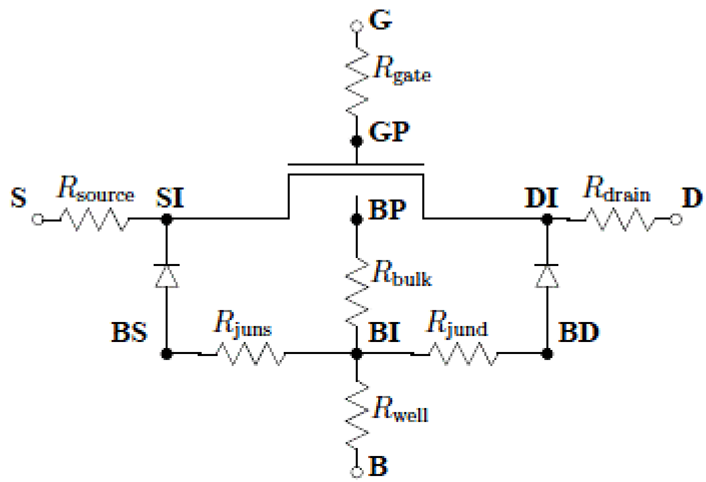




PSP-103

Parameter	Meaning
`sfl`	Channel current flicker noise
`sth`	Channel current thermal noise
`sig`	induced gate noise at 1 Hz
`sigth`	Induced gate noise
`shotgd`	Gate current shot noise from gate to drain
`shotgs`	Gate current shot noise from gate to source
`sjnoise`	Source to Bulk leakage current noise
`djnoise`	Drain to Bulk leakage and avalanche current noise
`rgatenoise`	Gate resistance thermal noise
`rbulknoise`	Bulk resistor thermal noise between node BP and BI (See diagram below)
`rjnoise`	Source side bulk resistor thermal noise between node BI and BS (See diagram below)
`rjundnoise`	Drain side bulk resistor thermal noise between node BI and BD (See diagram below)
`rwellnoise`	Well resistor thermal noise between node BI and B (See diagram below)
`rsourcenoise`	Source parasitic resistor thermal noise
`rdrainnoise`	Drain parasitic resistor thermal noise

Figure 3-4 PSP-103 Schematic Diagram

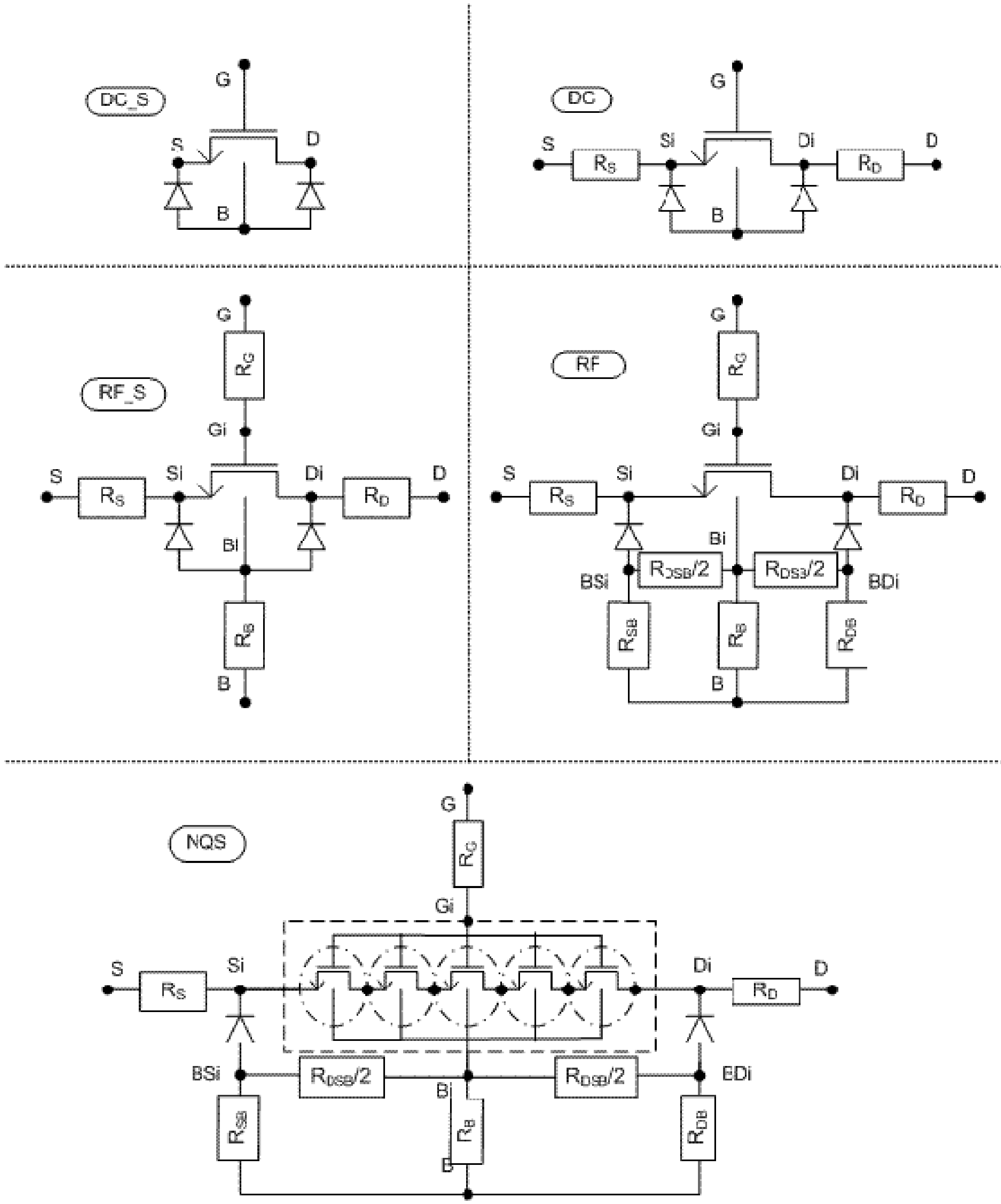


EKV 3.0

Parameter	Meaning
`fn_id`	Channel current flicker noise
`fn_ig`	Gate current flicker noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`rg`	Gate parasitic resistor thermal noise
`rsb`	Thermal noise for the substrate resistor connected between BSi and B. (See diagram below)
`rdb`	Thermal noise for the substrate resistor connected between BDi and B. (See diagram below)
`rbbs`	Thermal noise for the substrate resistor connected between BSi and Bi. (See diagram below where it is labeled R_{DSB}/ 2)

Parameter	Meaning
`rbbd`	Thermal noise for the substrate resistor connected between BDi and Bi. (See diagram below where it is labeled R_{DSB}/ 2)
`id`	Thermal noise of the channel resistor
`cor`	Induced gate noise
`shot_ig`	Gate current shot noise

Figure 3-5 EKV 3.0 Schematic Diagrams



For other model types:

1. In a UNIX/LINUX shell window type `which spectre`.

   This should return a path that ends in `/bin/spectre`.

2. Triple-click the line with the path, and then click the center mouse button to enter it as a command.

3. Use the backspace key, and delete the word `spectre`.

4. Type `cdnshelp` at the end of the expression, and press `Enter`.

   A browser window will be displayed.

5. Close the *Tip of the day* window.

6. At the top, you should see the release number of the simulator you are using.

7. Click the plus sign (+) to the left of the *MMSim* folder.

8. Click the plus sign (+) to the left of the *Virtuoso Simulator Components and Device Models Reference* folder.

9. Select the device type you are using.

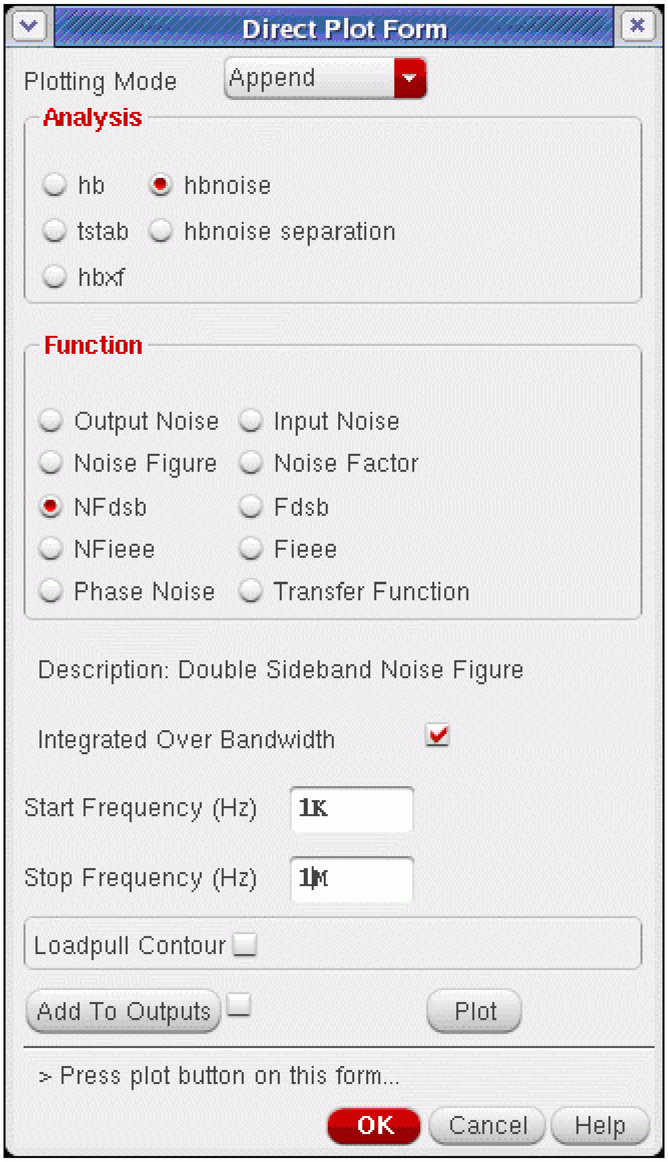
10. At the bottom of that device listing, you will see *Component Statements*. Double-click on this.

11. Enter the name of the parameter in the search box in the right pane of the cdnshelp window.

12. Click the down arrow to the right of the *Find* field until you see the parameter name in the pane on the right side, and then see the definition of that parameter.

Plotting Noise Figure

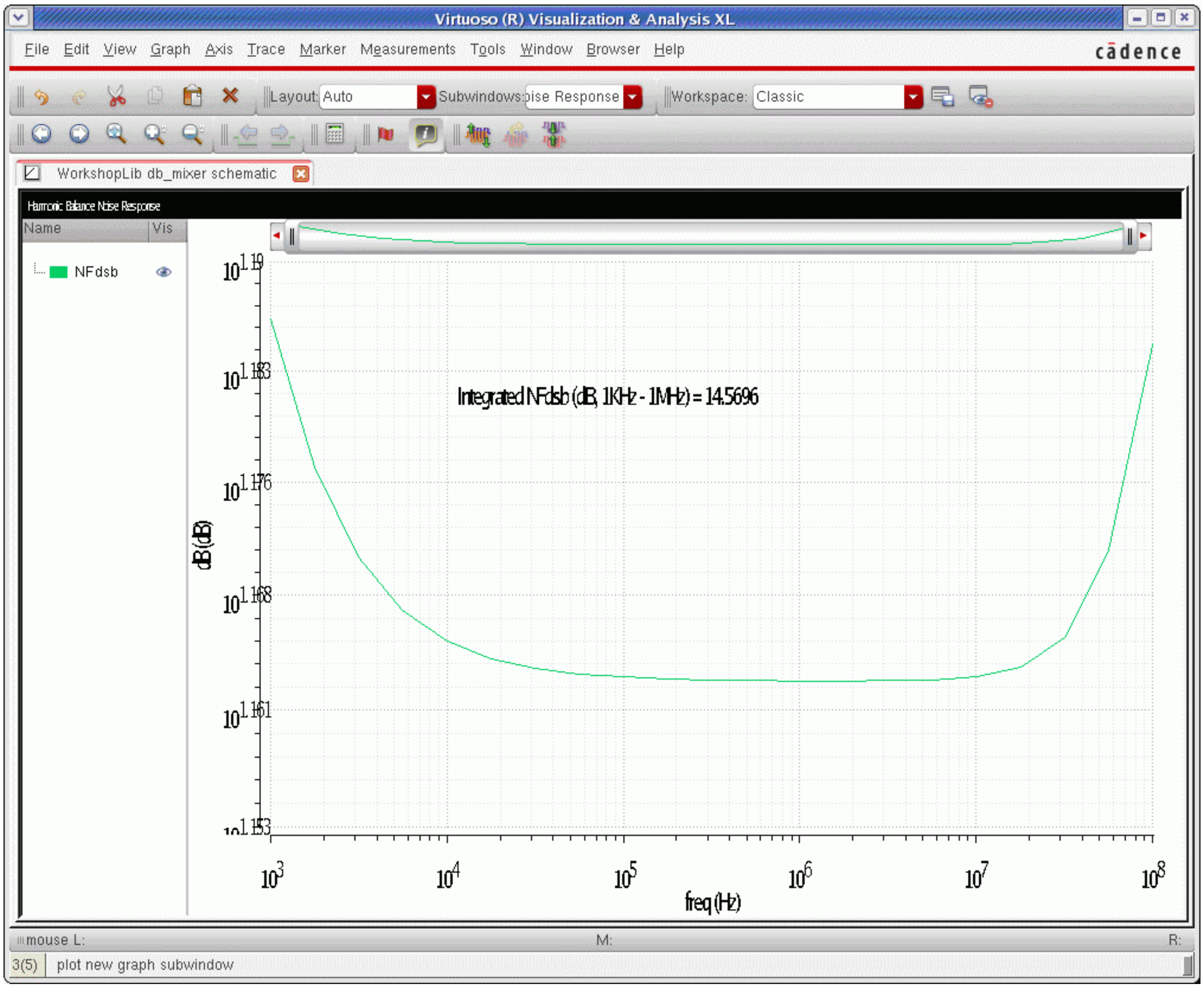
On the *Direct Plot Form*:



1. Select hbNoise from the *Analysis* section.
2. Select the desired type of noise plot from the *Function* section.
3. If you want the noise figure over a specified bandwidth, check the *Integrated Over Bandwidth* checkbox.
4. Specify the frequency range you want for the noise figure calculation.

5. Click *Plot*.

The dsb noise figure and the noise figure calculation over your bandwidth are displayed.

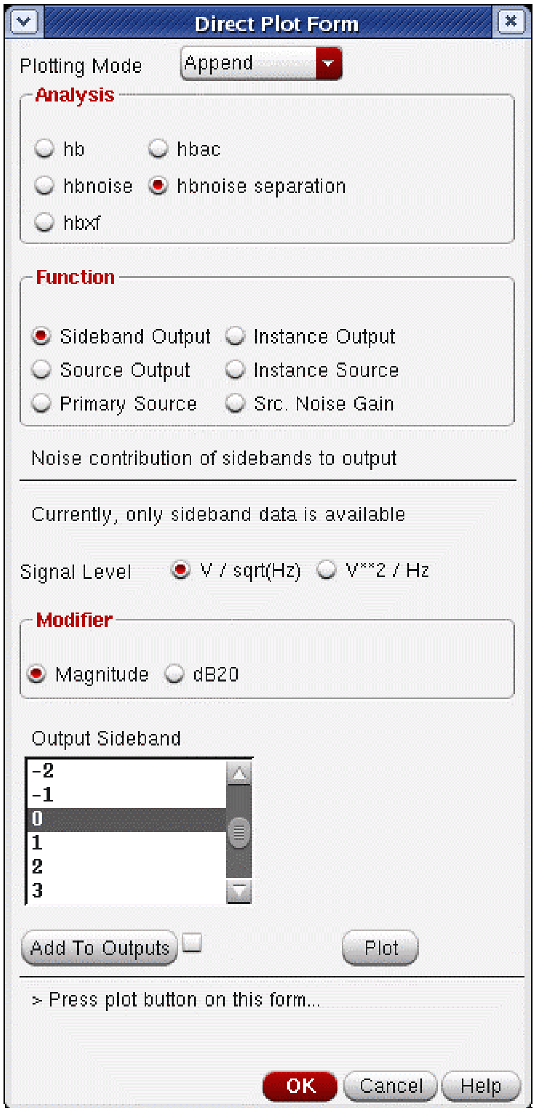


Viewing noise separation results

Viewing the total noise at the output from a single noise input frequency

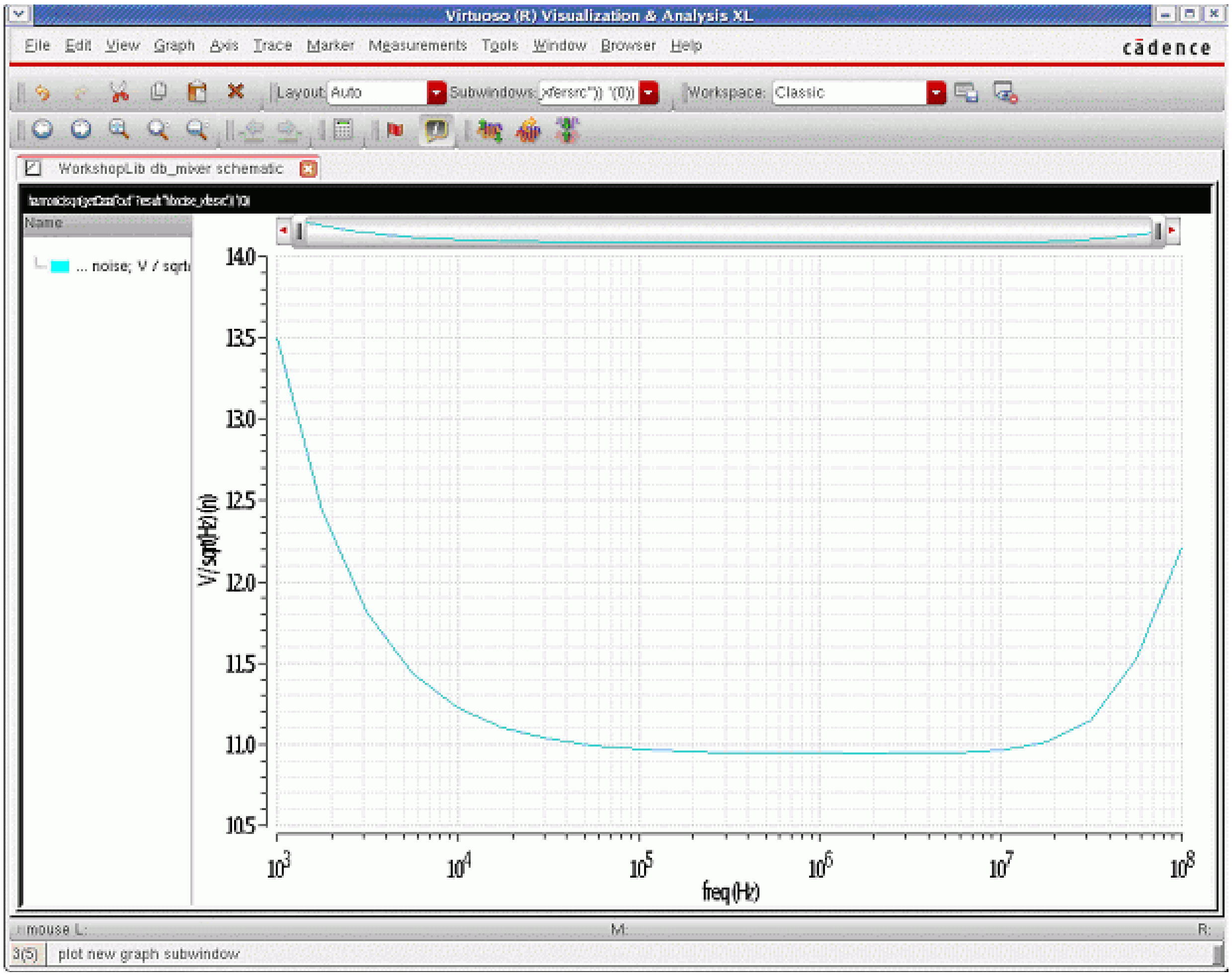
This capability allows the noise from all sources in the circuit at different noise input frequencies to be measured. For example, all the noise that does not change frequency (the zero sideband) can be plotted, along with the other noise input frequencies (sidebands). This plotting capability allows problem frequencies to be identified.

On the *Direct Plot Form*:



1. Select *hbnoise separation* from the *Analysis* section.
2. To view the total noise at the output select *Sideband Output* from the *Function* section.
3. Select *V/sqrt(Hz)* or *v**2/Hz* from the *Signal Level* radio buttons.
4. Select the appropriate modifier from the *Modifier* section.
5. Select the sideband number form the *Output Sideband* list. 0 means no frequency translation.

6. Click *Plot*.

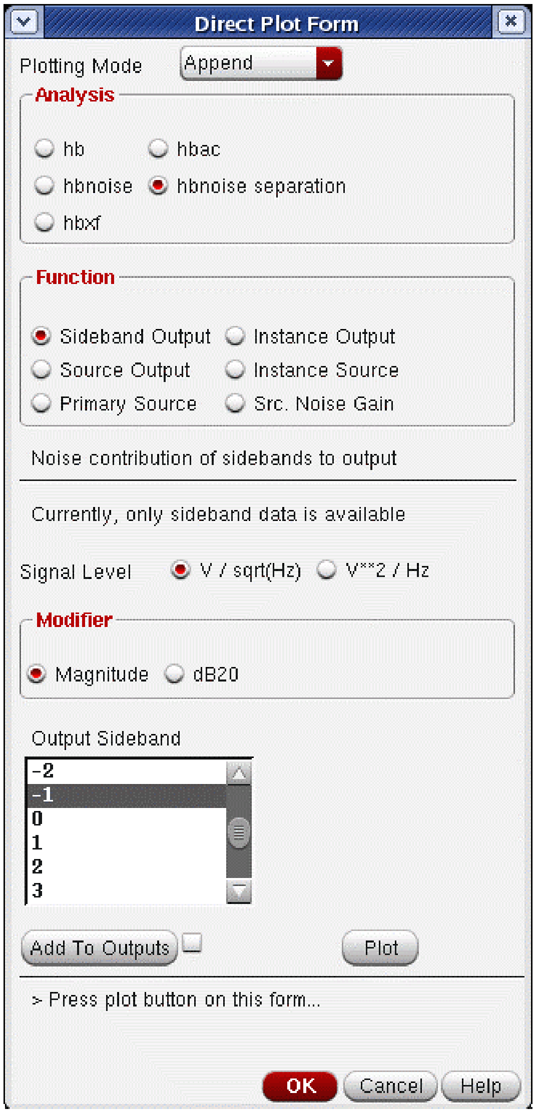


The output noise from sideband zero is plotted. This is the noise that is present at the output with no frequency translations included.

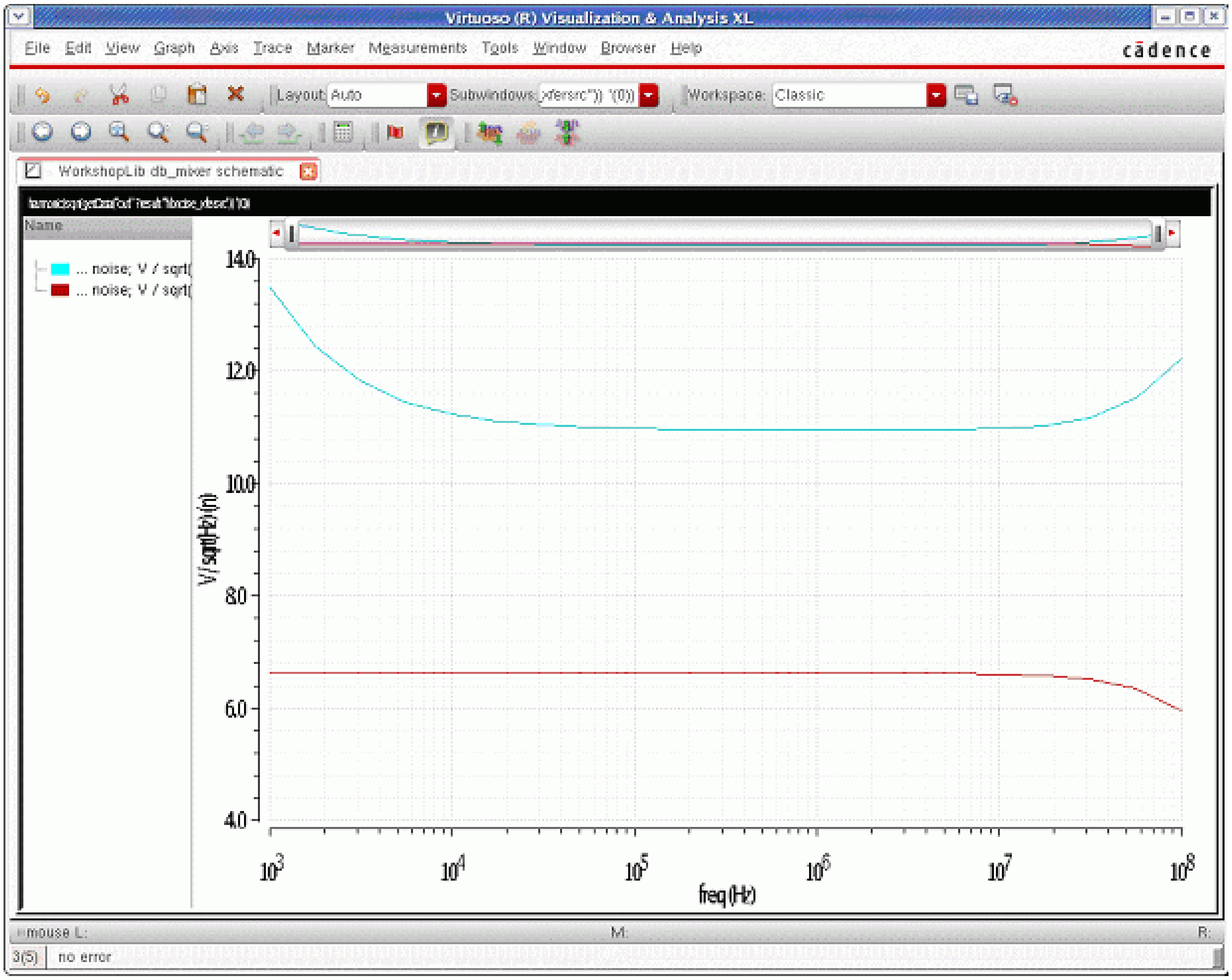
Plotting Noise from Sideband -1

On the *Direct Plot Form*:

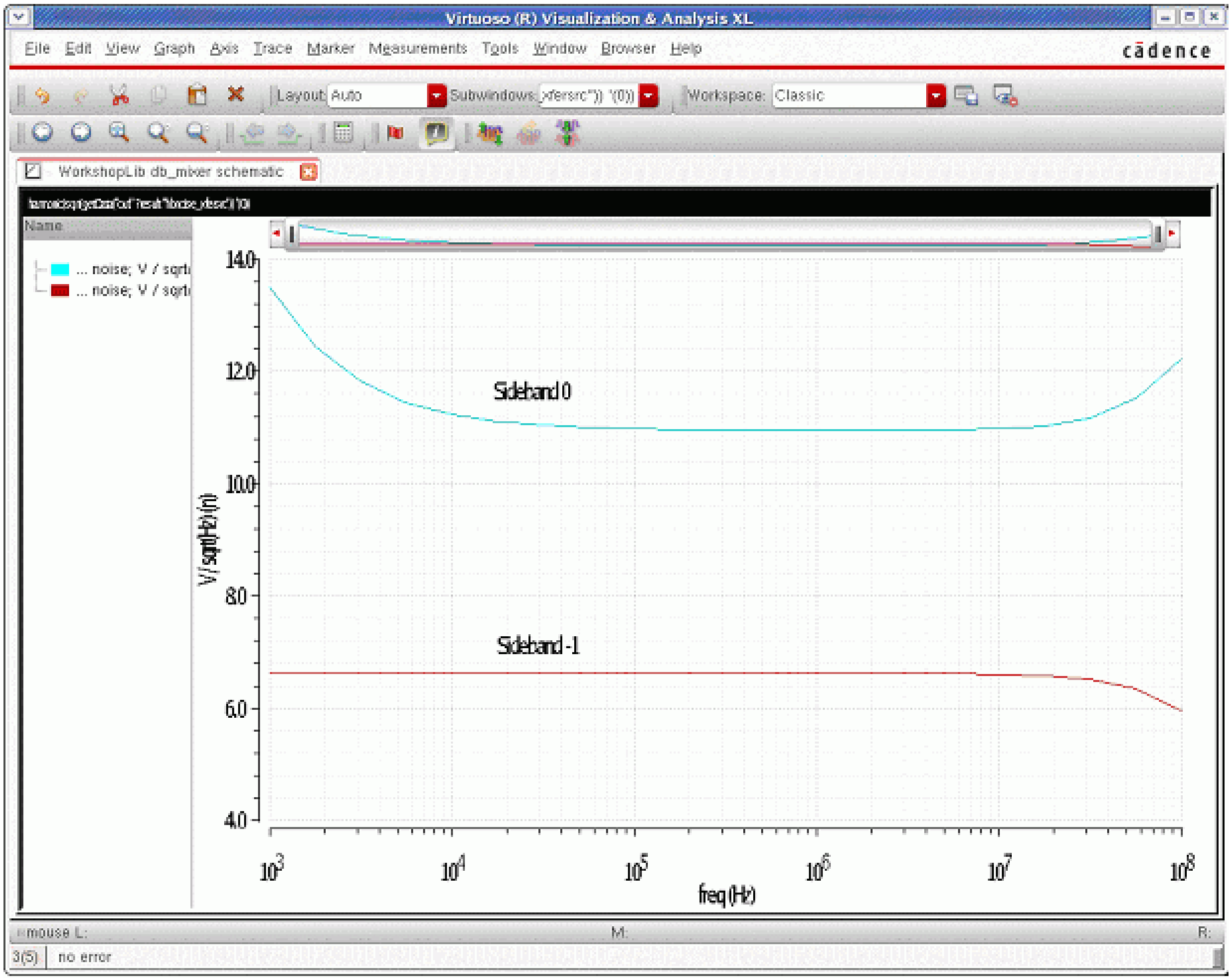
1. Select the -1 sideband from the *Output Sideband* list. -1 means the noise frequency is shifted down in frequency by -1*first harmonic of hb from the frequency that is specified as the noise output frequency in the *Choosing Analyses* form.



2. Click *Plot*.

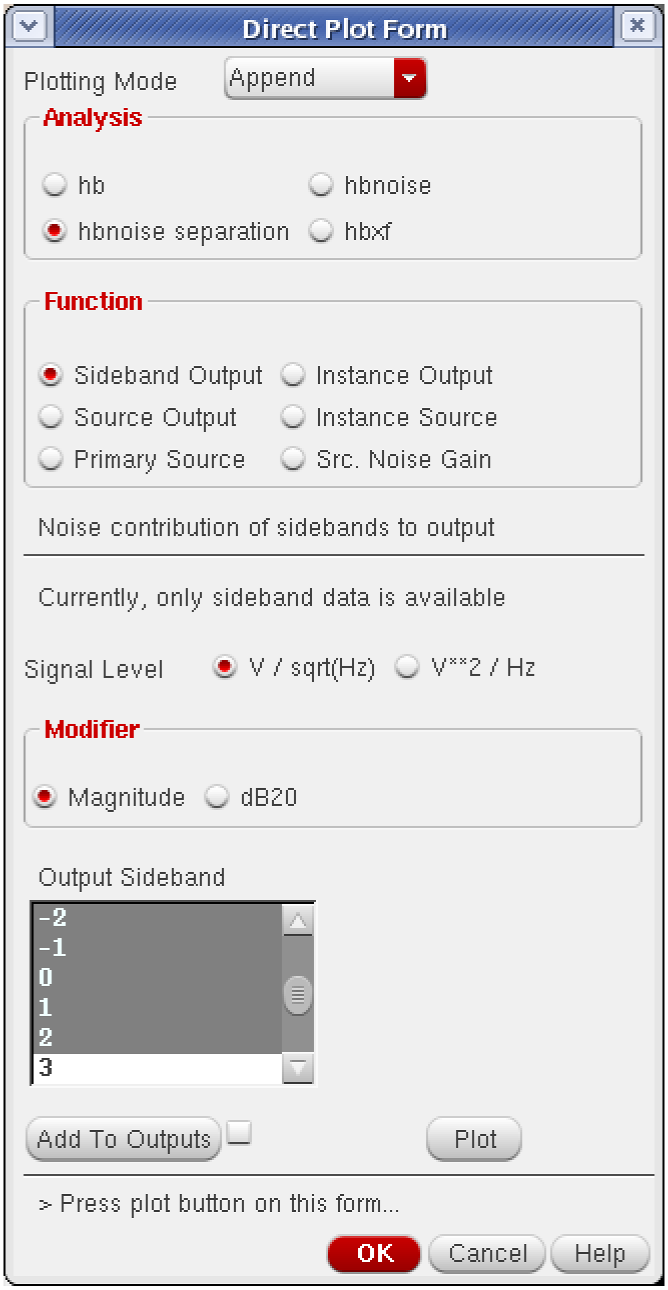

Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Layout: Auto
Subwindows:
Workspace: Classic
WorkshopLib db_mixer schematic
Name
... noise; V / sqrt(
... noise; V / sqrt(
14.0
12.0
10.0
8.0
6.0
4.0
V / sqrt(Hz) (n)
freq (Hz)
mouse L:
M:
R:
3(5)
no error

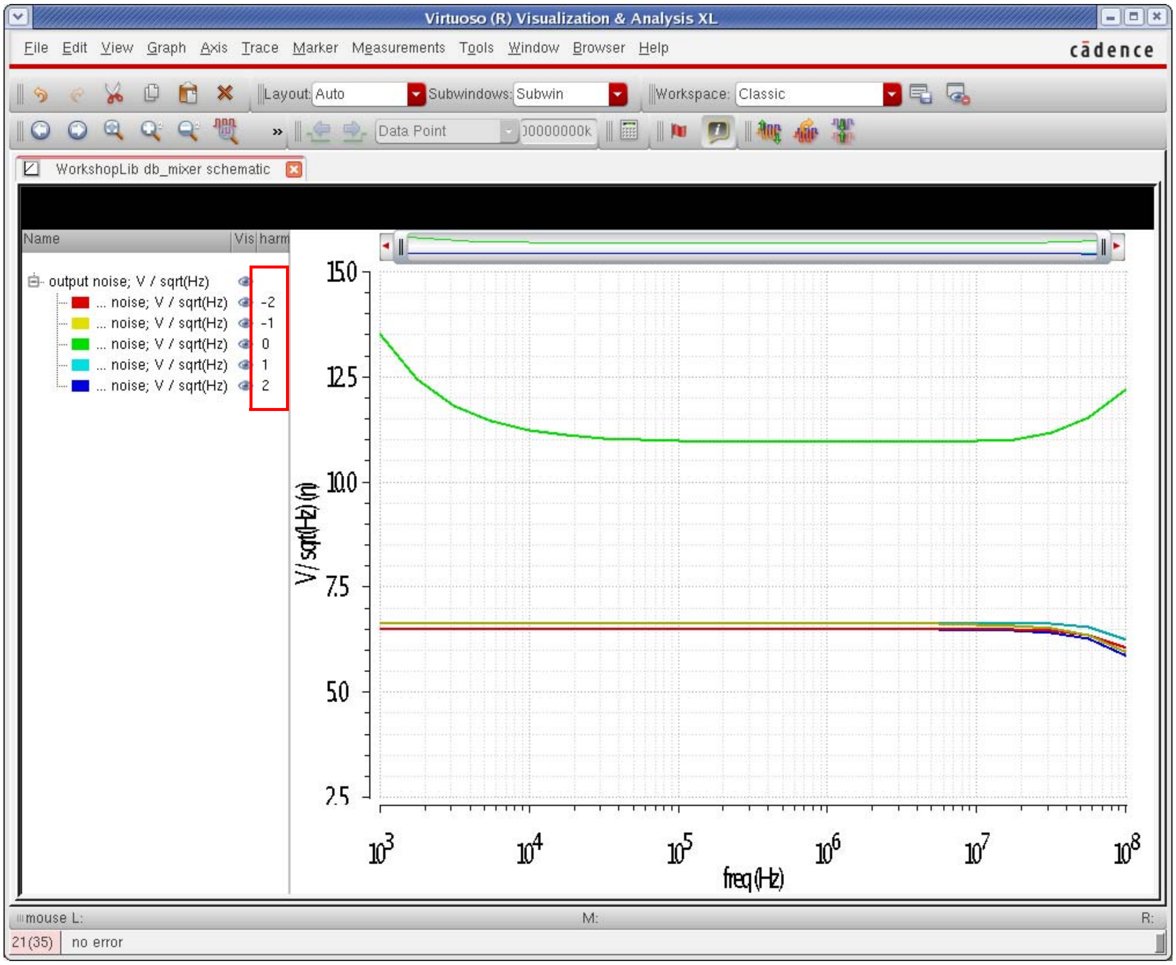
As a suggestion, use the *Graph - Add Label* menu option to place labels in order to identify the traces.



Tip

If you select multiple sidebands to be plotted at the same time, the sideband number will be shown at the right side of the legend in the waveform window.

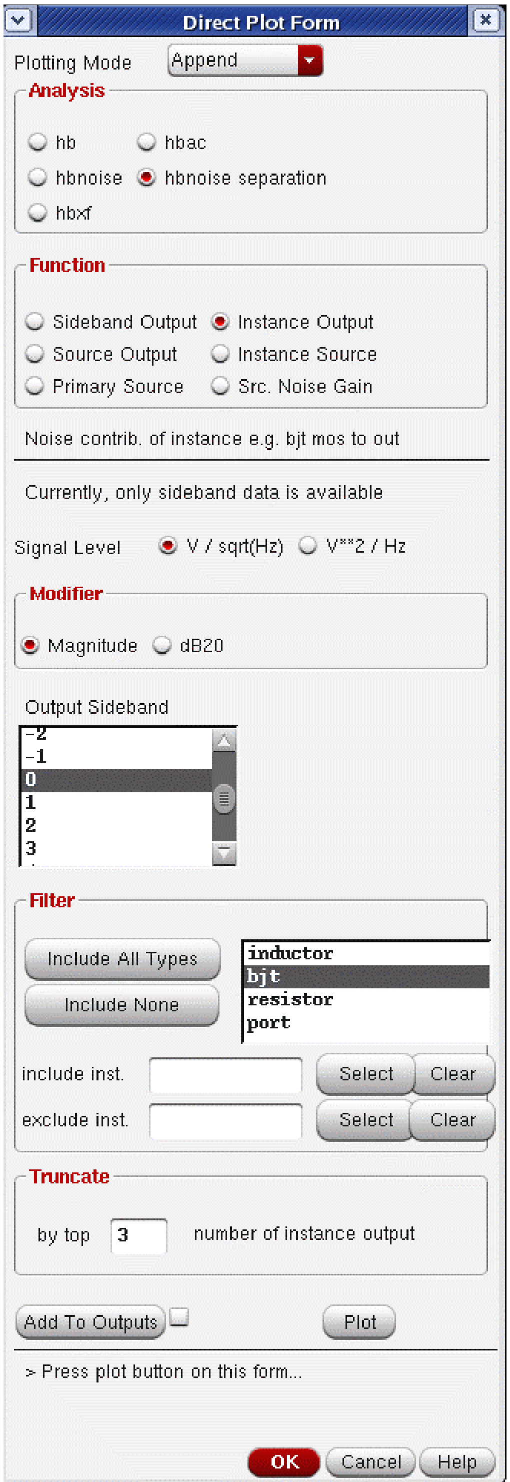




Viewing total noise at the output from each instance

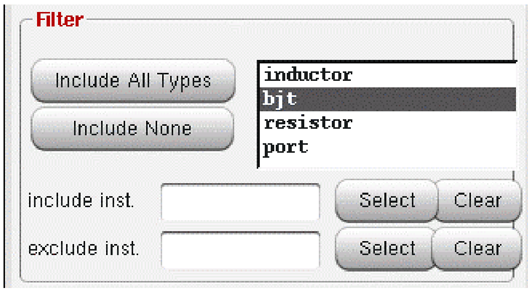
Once the problem frequencies are identified in the plot of *Sideband Output*, the problem components need to be identified. From the previous plot, the zero sideband contributes the largest amount of noise to the overall output noise. Using the *Instance Output* option allows the identification of which components in the zero sideband contribute the most noise. The total noise for all the noise mechanisms within the component is plotted.

On the *Direct Plot Form*:

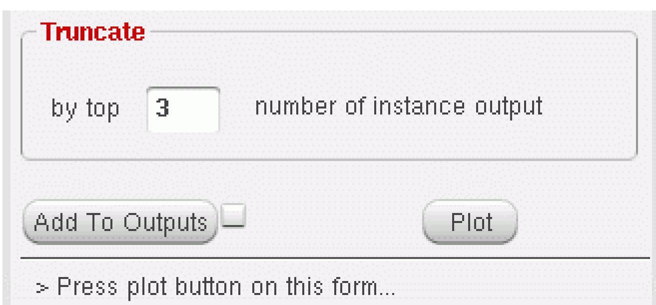


1. Select *Instance Output* from the *Function* section.

2. Select *V/sqrt(Hz)* or *v**2/Hz*.
3. Select the appropriate modifier.
4. Select the sideband number.
5. Select the type of component to display.

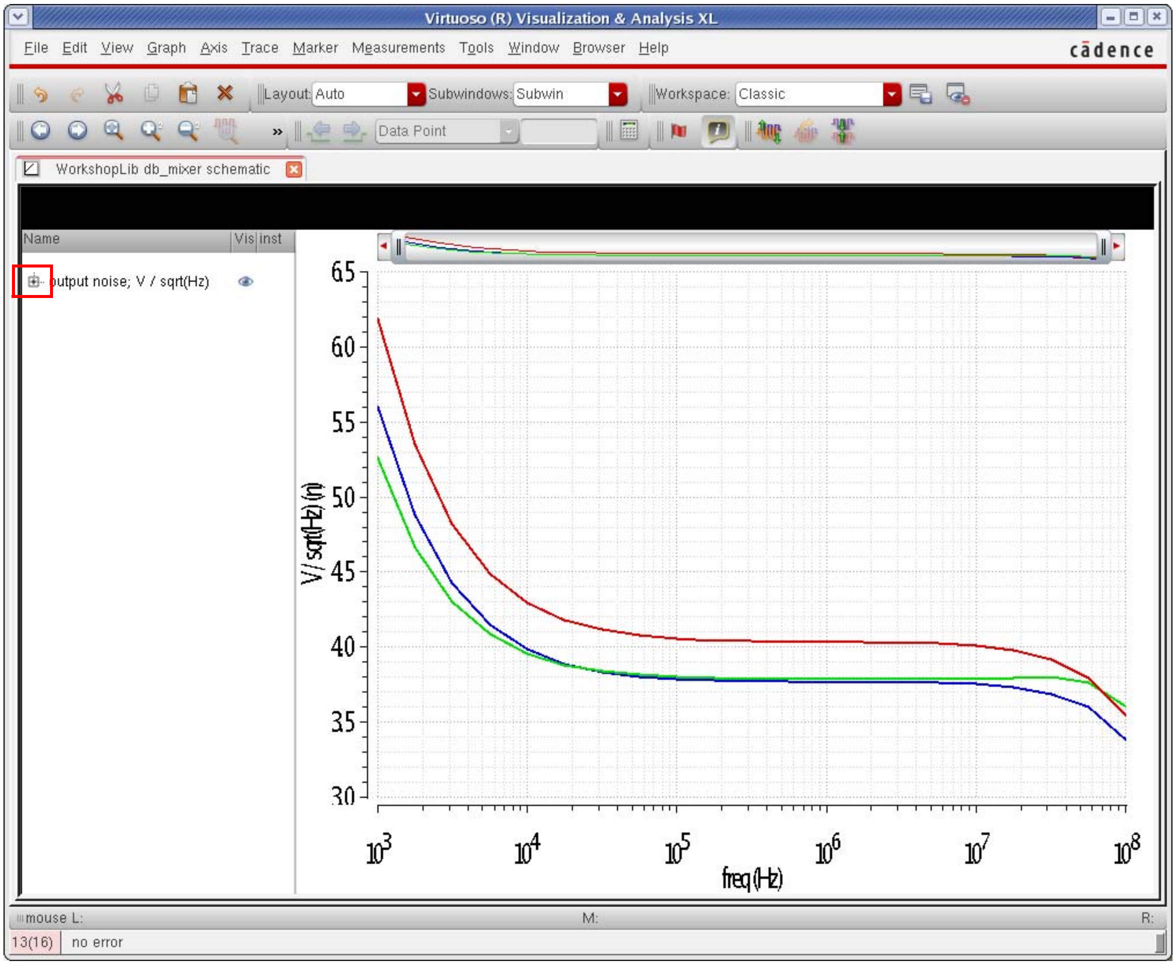


6. If you want every noise source to be considered, select *Include All Types*.
7. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances that you want to exclude from the plot.
8. To include specific instances, click *Select* to the right of the *include inst* field, and select the instances on the schematic.
9. Specify the number of results you want to plot.

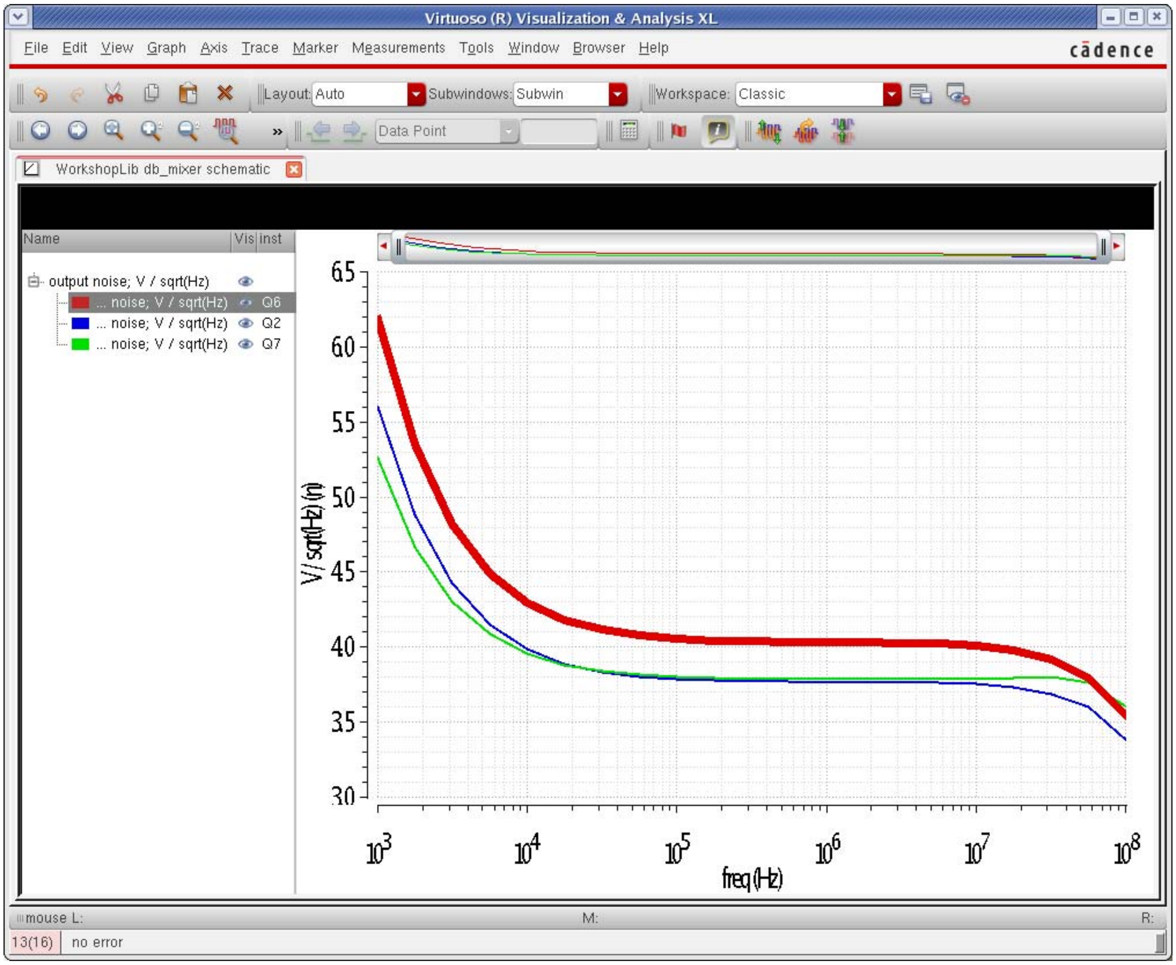


10. Click *Plot*.

The total noise at the output for the top three BJTs is plotted.



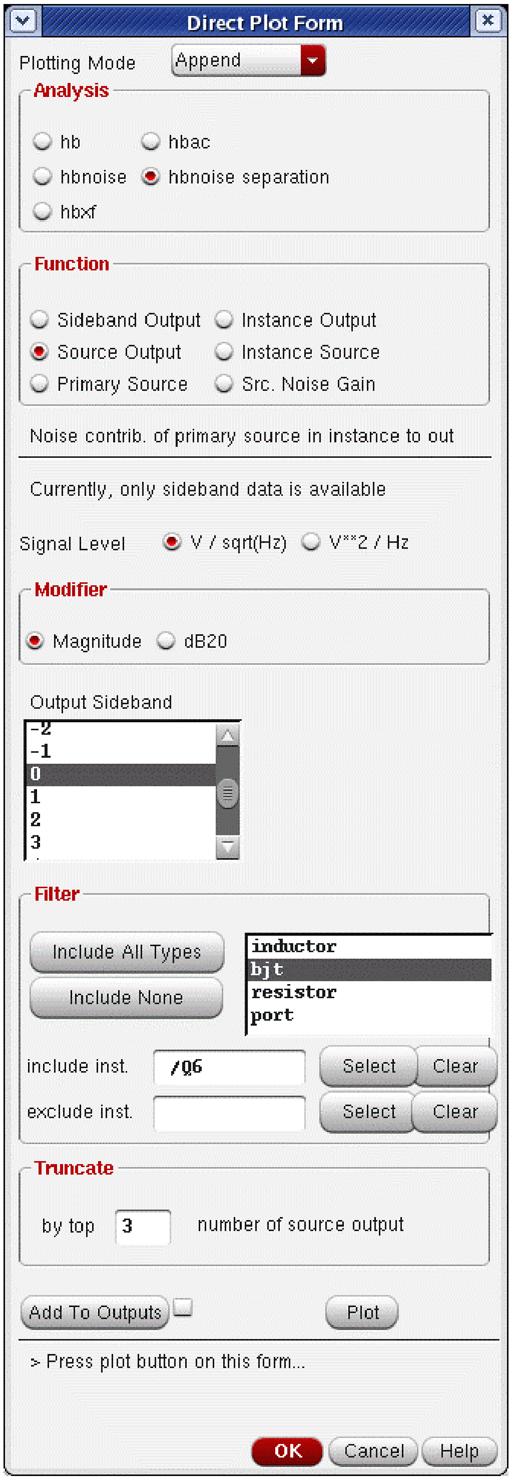
To see which transistor goes with which trace, click the plus sign (highlighted in red on the previous graphic) to the left or the text at the upper left of the display area. The legends expand, and you can see which trace is which in the *Name* field on the left side of the display tool. You can also either select the trace or the individual entry in the *Name* field and the trace and legend will highlight.



Viewing the exact noise mechanism within the devices

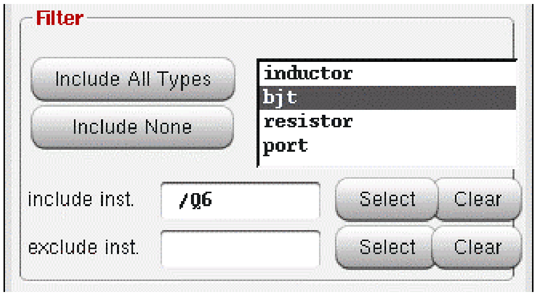
The sideband with the largest contribution was determined by selecting *Sideband Output* and plotting. The specific component was identified by selecting *Instance Output*. The specific noise mechanisms can be identified using *Source output*. This plots the largest individual noise mechanisms for the sideband you select.

On the *Direct Plot Form*:



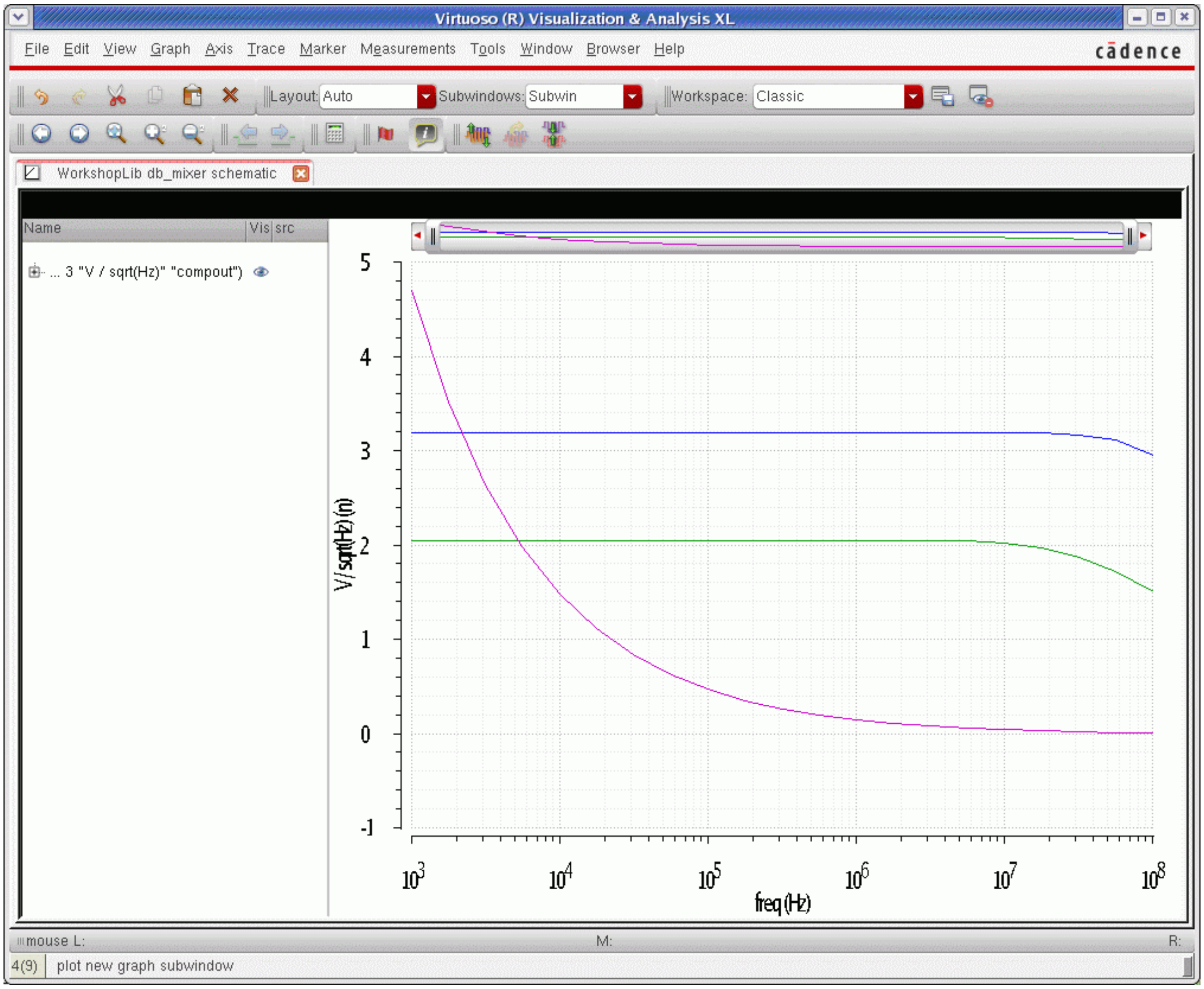
1. Select *Source Output* from the *Function* section.

2. Select *V/sqrt(Hz)* or *v**2/Hz*.
3. Select the appropriate modifier.
4. Select the sideband number.
5. Select the type of component to display.

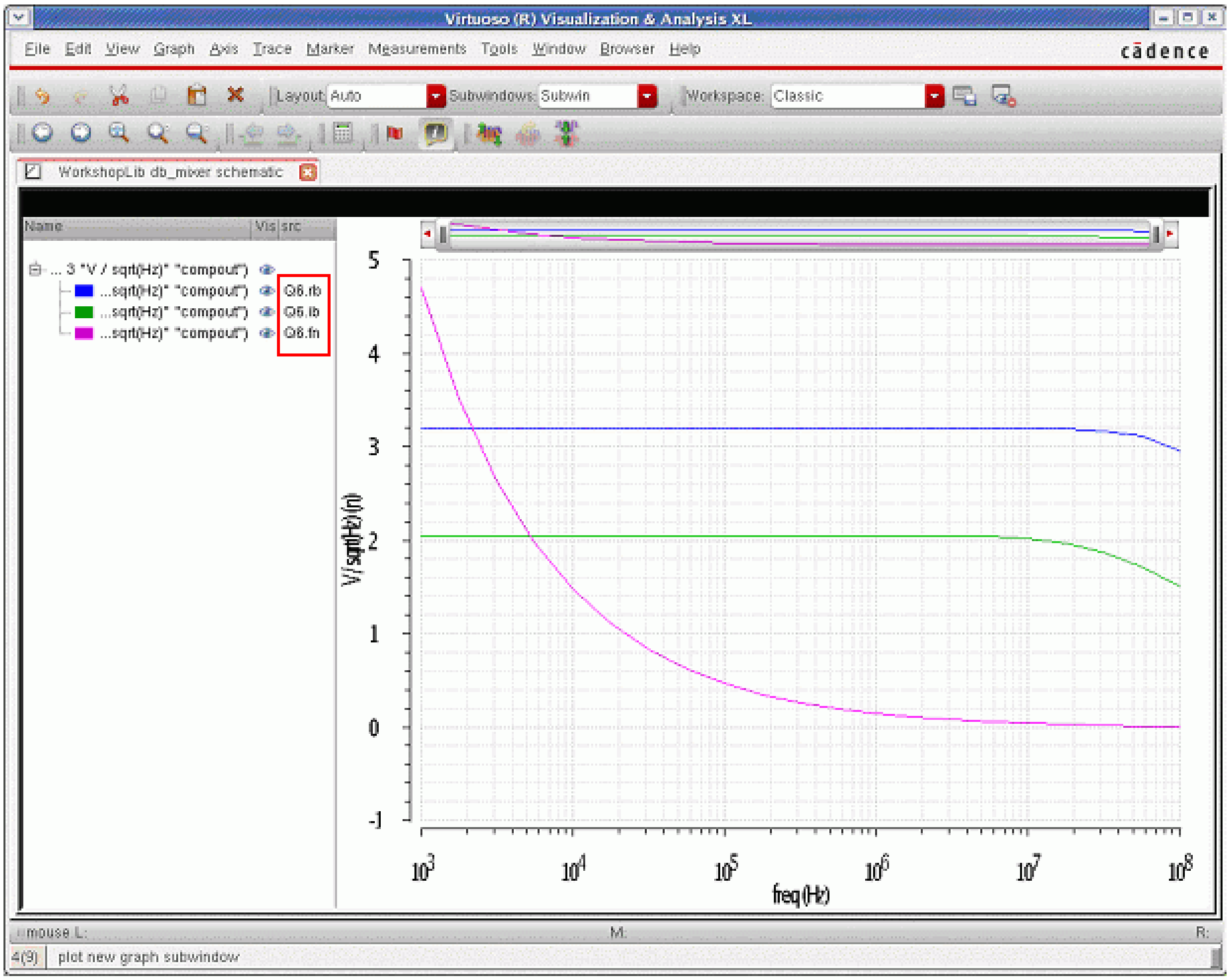


6. If you want every noise source to be considered, select *Include All Types*.
7. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
8. To include specific instances, click *Select* to the right of the *include inst* field, and select the instances on the schematic.
9. If you want to restrict to one device, put its instance name in the *include inst* field. A list of devices separated by a space is also acceptable.
10. If you want to exclude a device (or devices) enter the instance name(s) in the *exclude inst* field.
11. Specify the number of results you want to plot in the *by top* field.
12. Click *Plot.*

The exact noise mechanism within the devices is plotted.

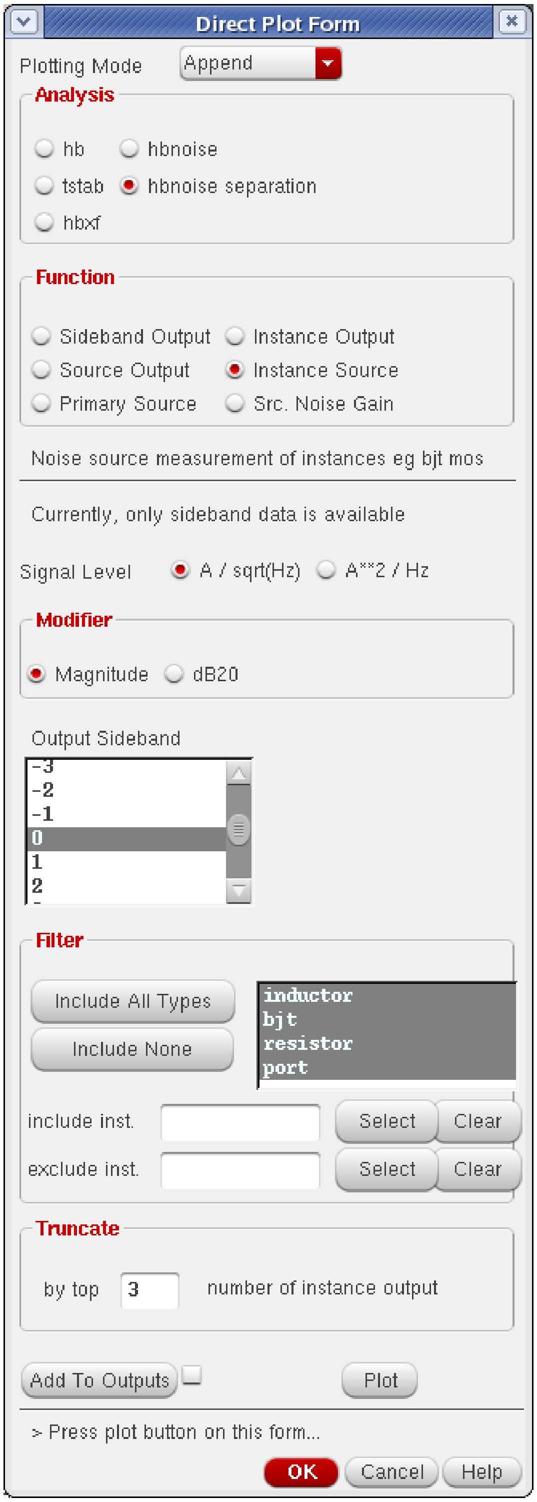


To see which noise source goes with which trace, select the plus sign to the left or the text at the upper left of the display area. The legends will be shown individually in the *Name* field of the waveform tool. The device name and the individual noise contributor will be shown. You can select either the individual legend or the trace and both the legend and the trace will highlight. For a list of the noise parameters for the different device types, see the list beginning just after the noise summary.



- The base current shot noise is in blue.
- The base parasitic resistance is red.
- The flicker noise is pink.

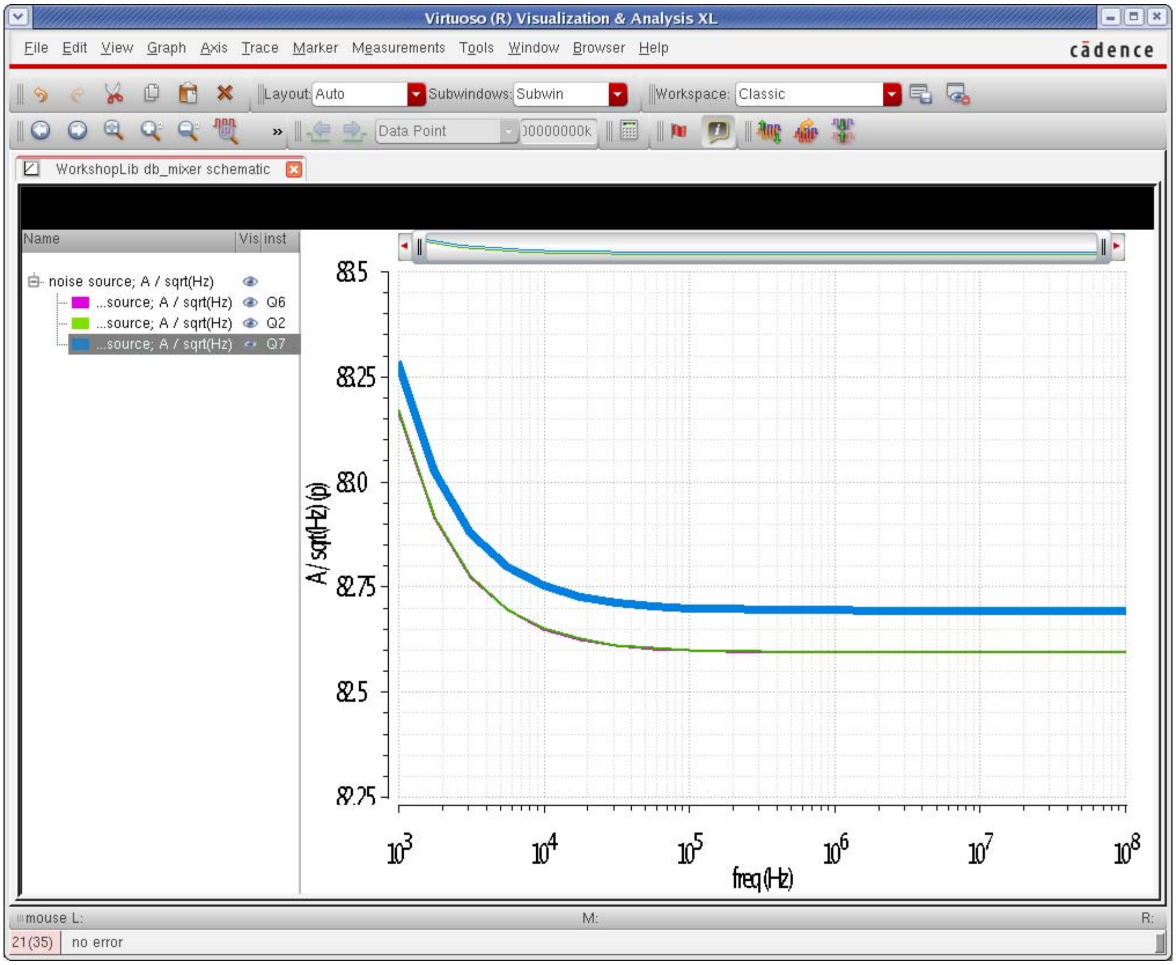
Viewing the total noise currents within the devices



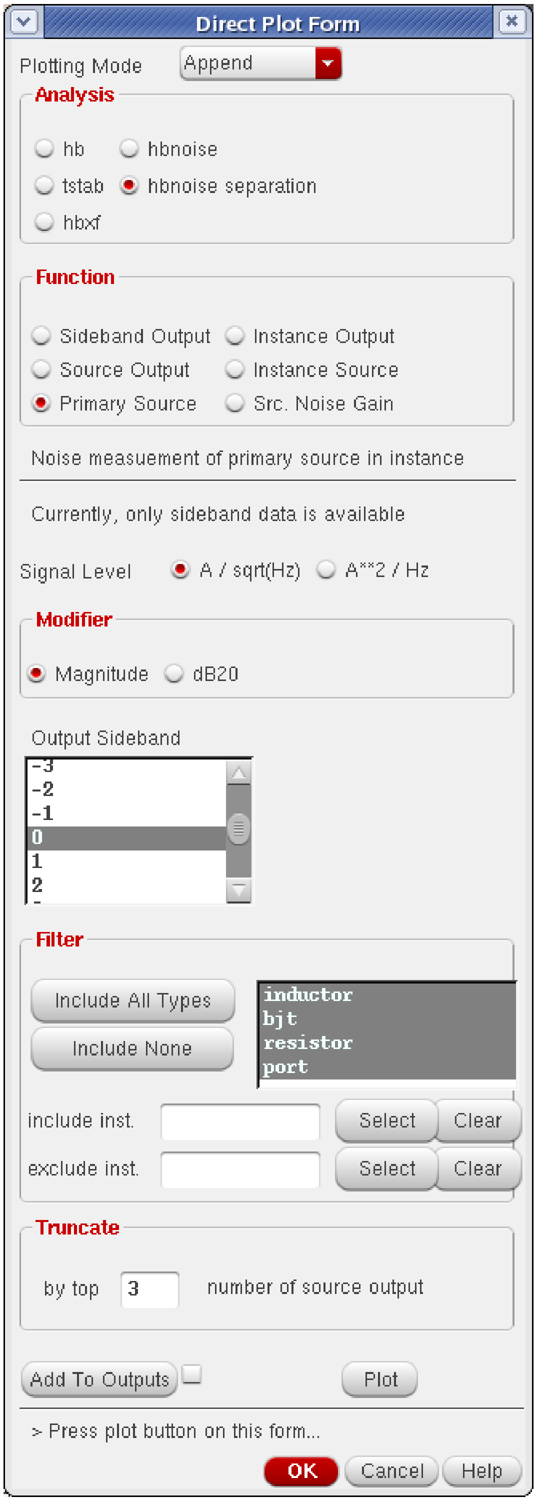
This capability has limited value. All the noise currents from all the noise mechanisms within the instance are added up. No gain function is applied or are available for this measurement. It just adds up everything within that instance.

1. Select *Instance source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the Signal Level selection.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

10. The frequency response curves are displayed.



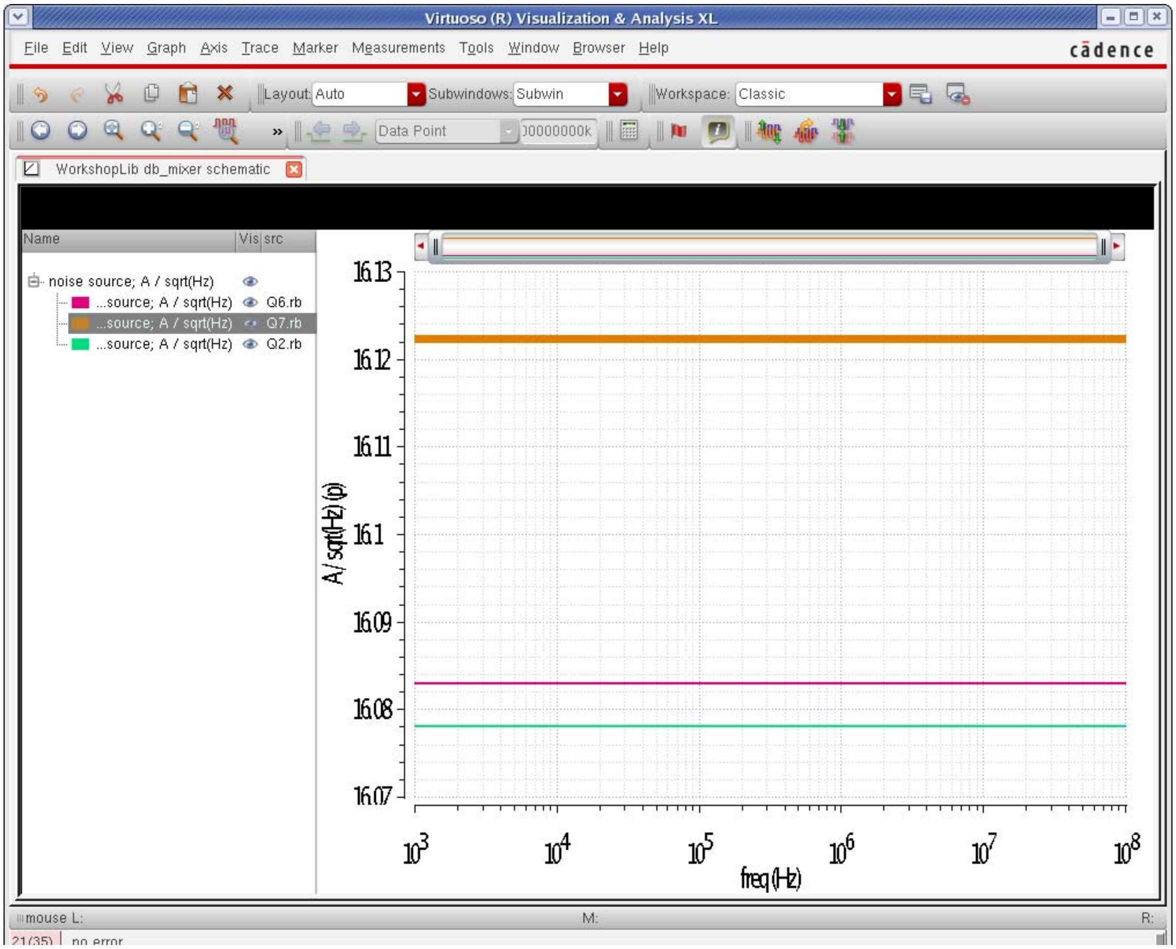
Plotting the individual noise source currents



This capability plots the individual noise currents within the device. it gives you the noise at the source instead of at the output of the simulation. This enables you to identify the largest individual noise sources in the circuit. Note that the largest noise source in the circuit may not contribute the largest noise at the output of the circuit. It depends on the relative gains from the individual sources to the output. These gain functions are available, and are shown in the next section.

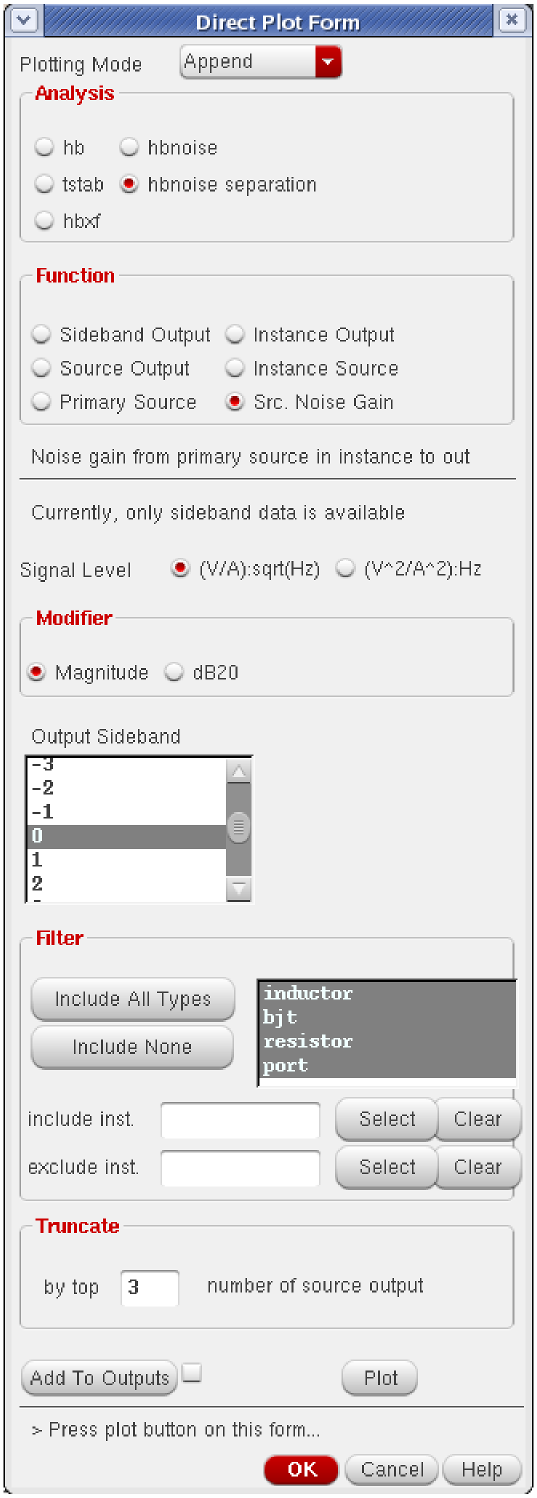
1. Select *Primary Source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* section.
3. Select *Magnitude* or *dB20* in the Modifier section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

10. The frequency response curves are displayed.



For a list of the noise parameter names, please see the table under the section Noise Summary on page 415.

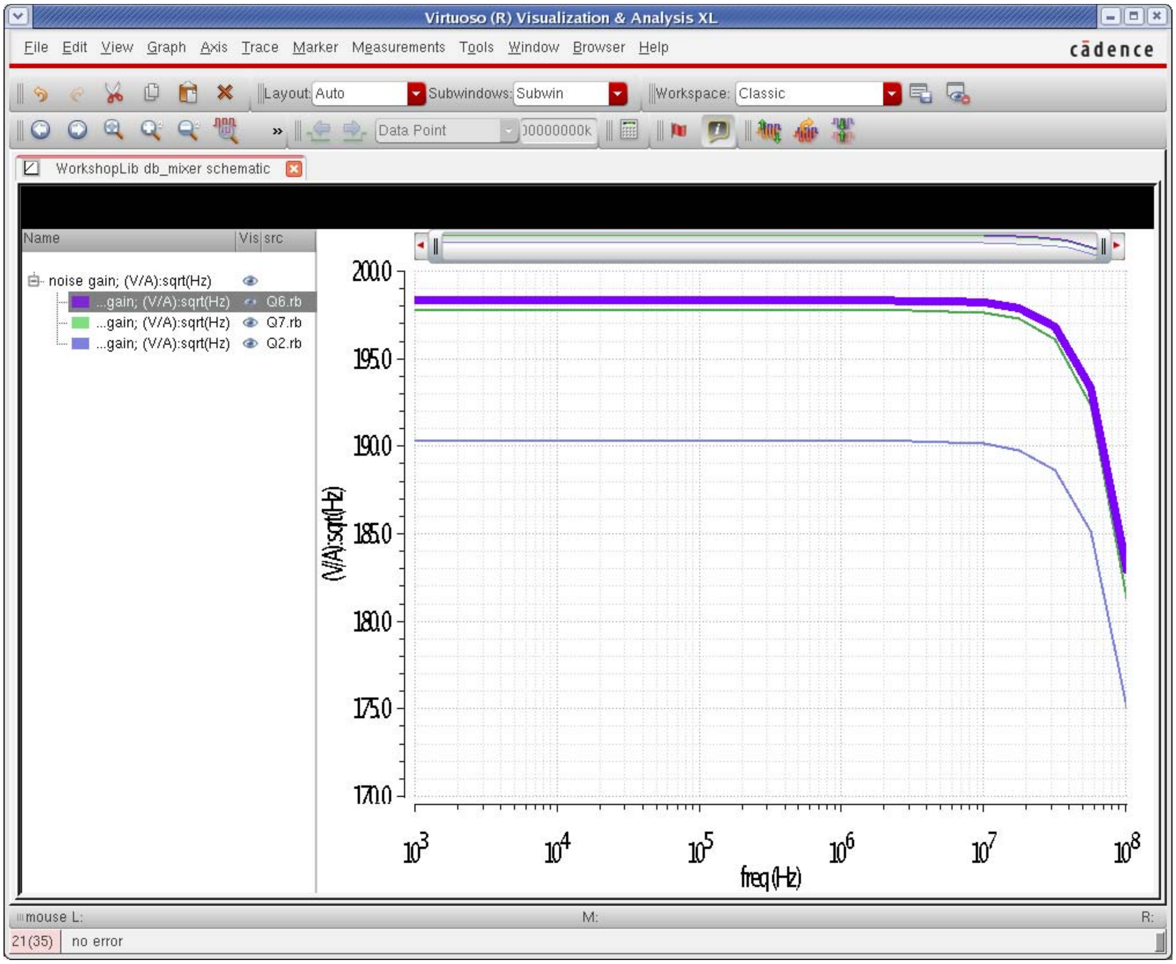
Plotting the gain from the individual noise sources to the output



This capability gives the gain from the individual noise currents to the output of the circuit. The noise current from the previous section times the gain that is plotted in this section gives the noise at the output of the circuit.

1. Select *Source Noise Gain*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* section.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select *Include All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

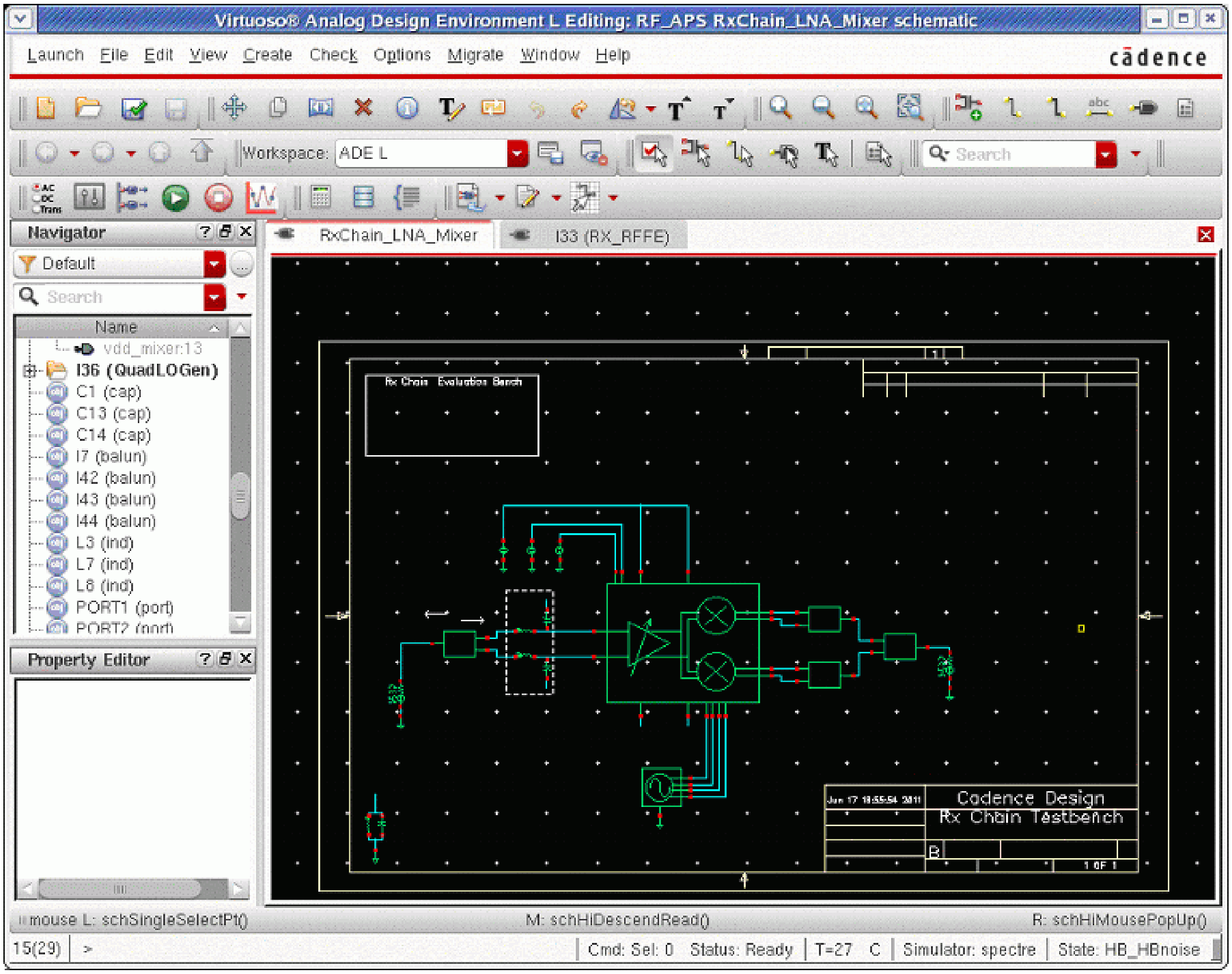
The frequency response curves are displayed. For a list of the abbreviations used for the noise parameters, please see page 123 (77 of 234) of this manual.



Multiple hbnoise

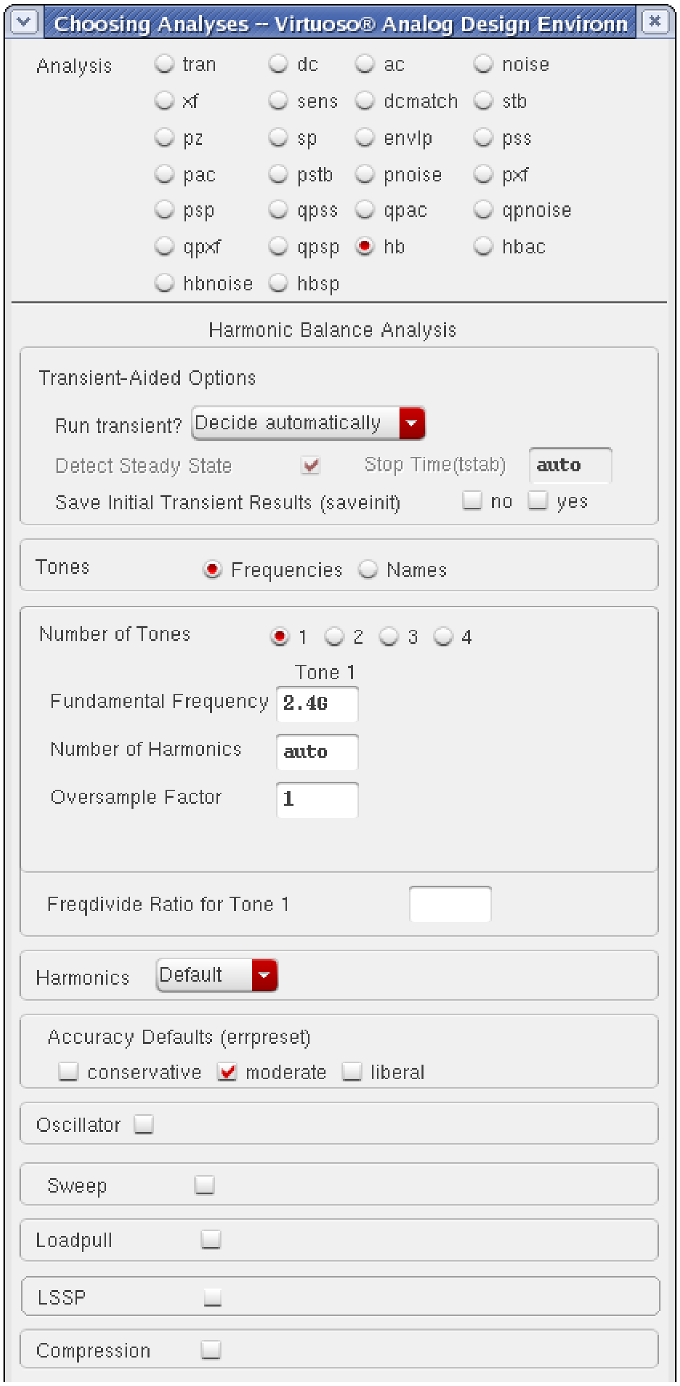
Multiple hbnoise allows multiple pnoise analyses to be run after a harmonic balance simulation so that you can characterize the noise at several points in your circuit. For example, in an LNA-Mixer circuit, you might want to characterize the noise at the LNA output and at the output of the mixer. Multiple hbnoise allows these analyses to be run without needing multiple simulations.

Consider an LNA-Mixer circuit shown below.



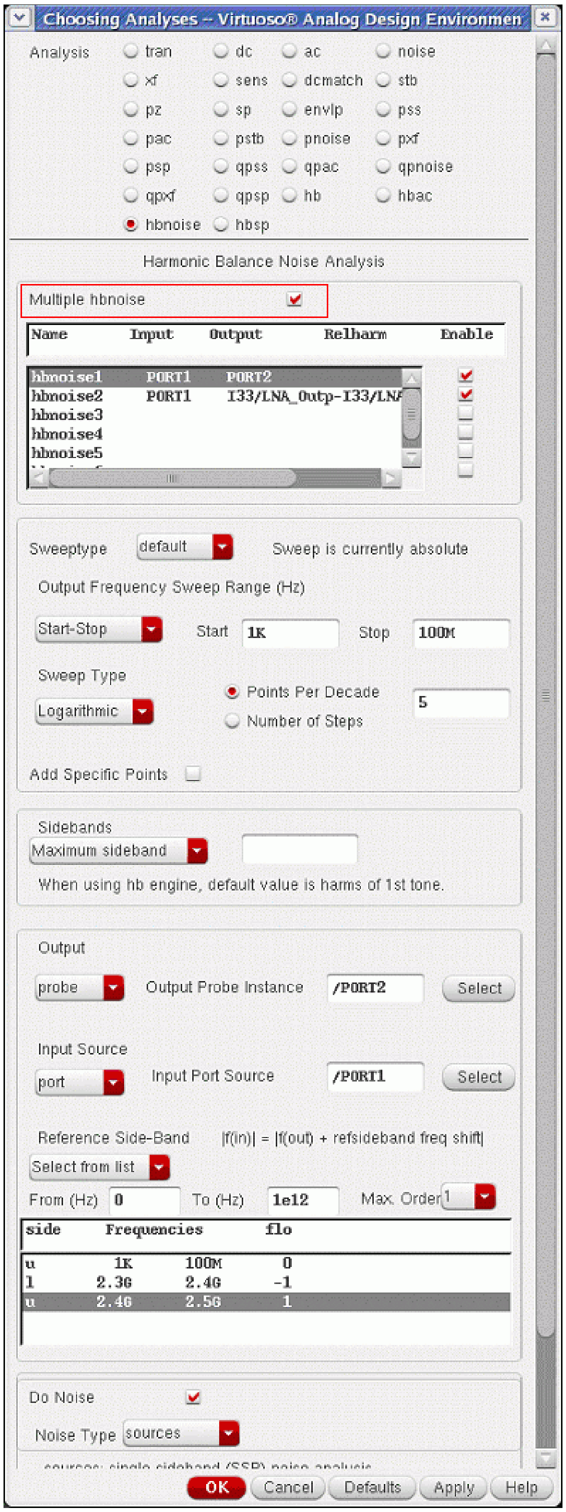
In the above circuit, there are two outputs that need to be measured. The first is at the LNA output, and the second is at the mixer output. The noise frequencies are different at the two outputs.

First, set up the harmonic balance analysis. For more information, see the Harmonic Balance section at the beginning of this chapter.



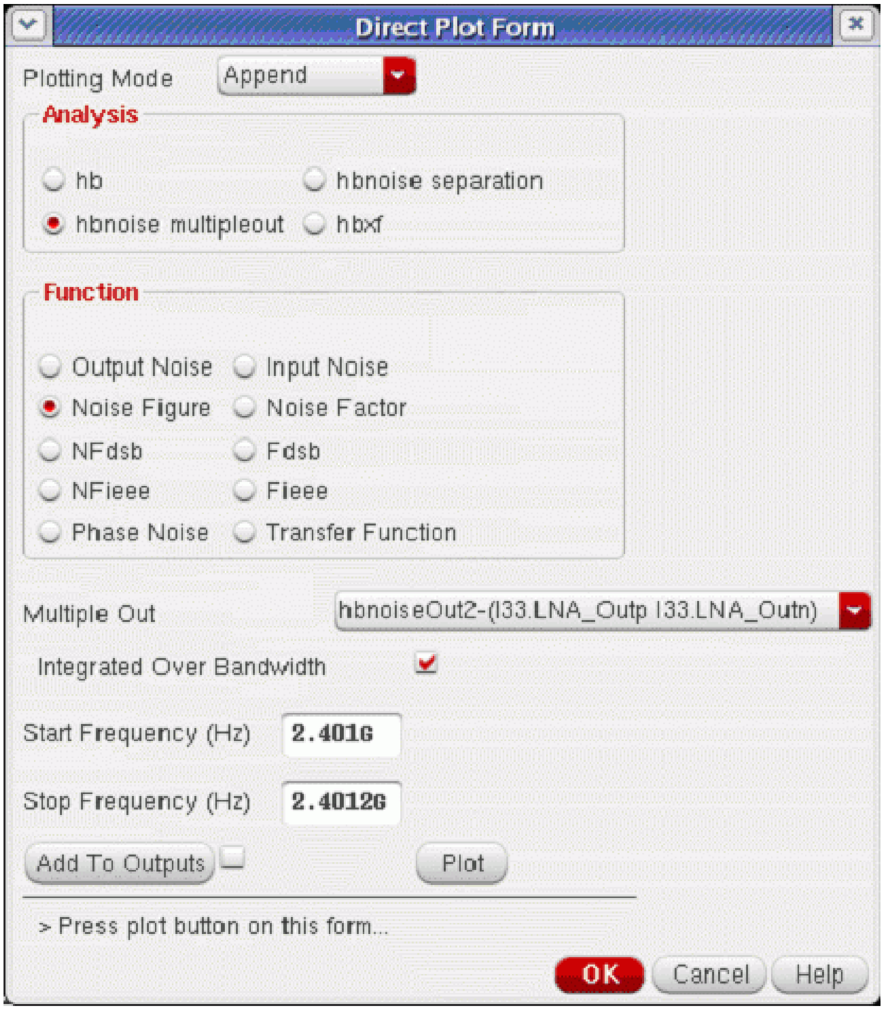
In the hbnoise *Choosing Analyses* form:

1. Select *Multiple hbnoise*, and set up the noise analysis for one of the outputs. The mixer output is shown below.

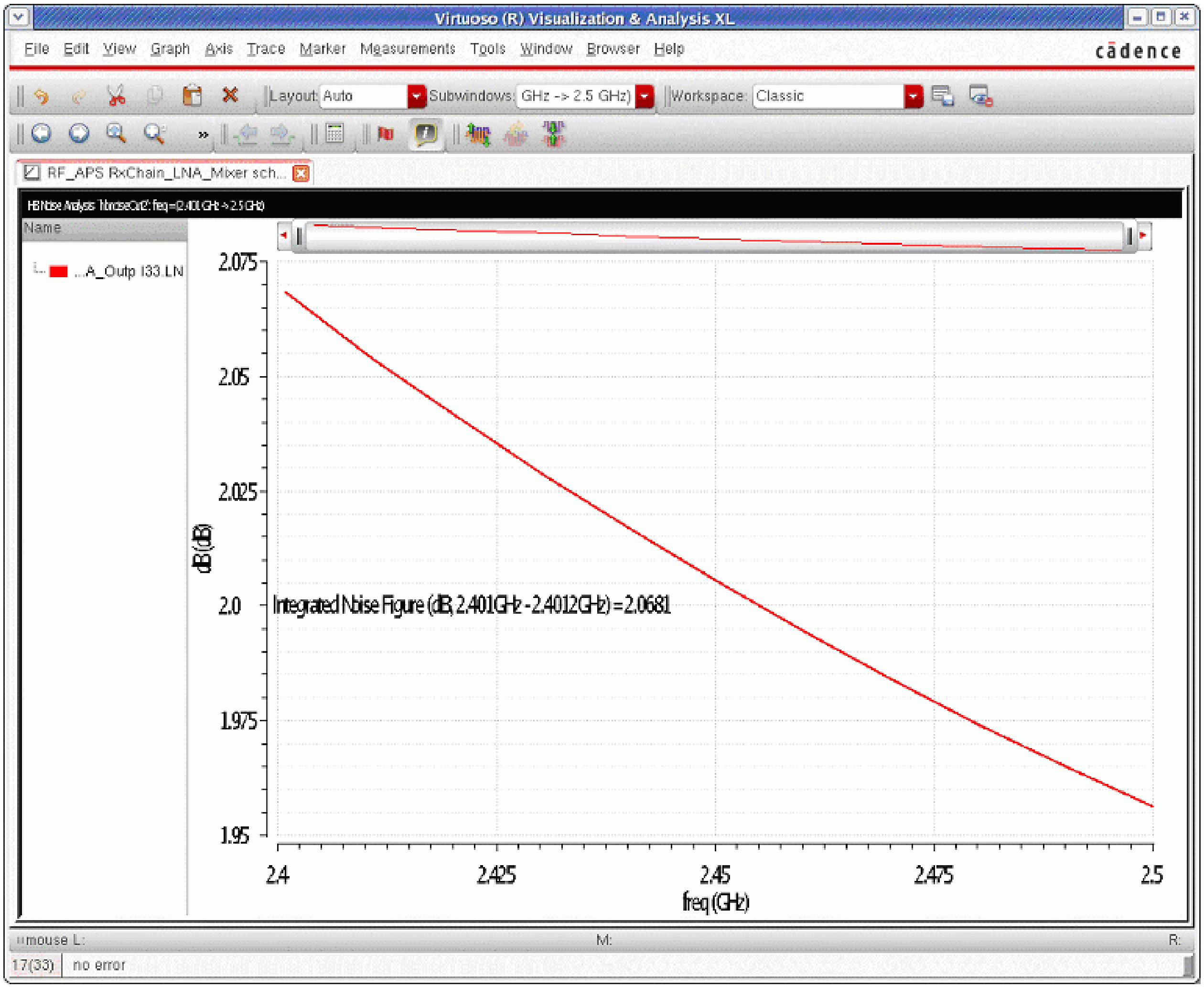


2. Select the second line in the multiple hbnoise section and set up that analysis.

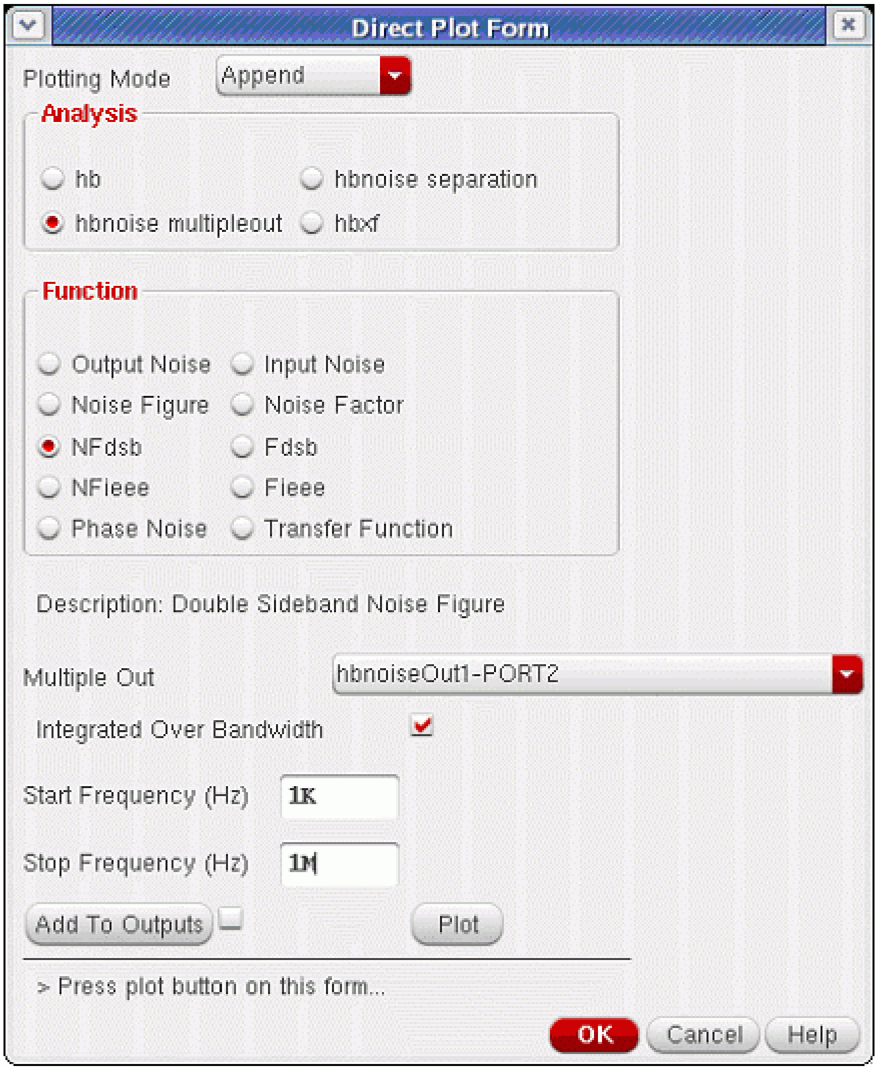
3. Run the simulation. When the simulation completes, select *Results - Direct plot - Main Form*. When multiple hbnoise is run, the *Direct Plot Form* adds the ability to select which hbnoise results to plot. This is the purpose of the Multiple Out selection.



4. Click *Plot*. The single-sideband noise figure measurement for the LNA is shown below.

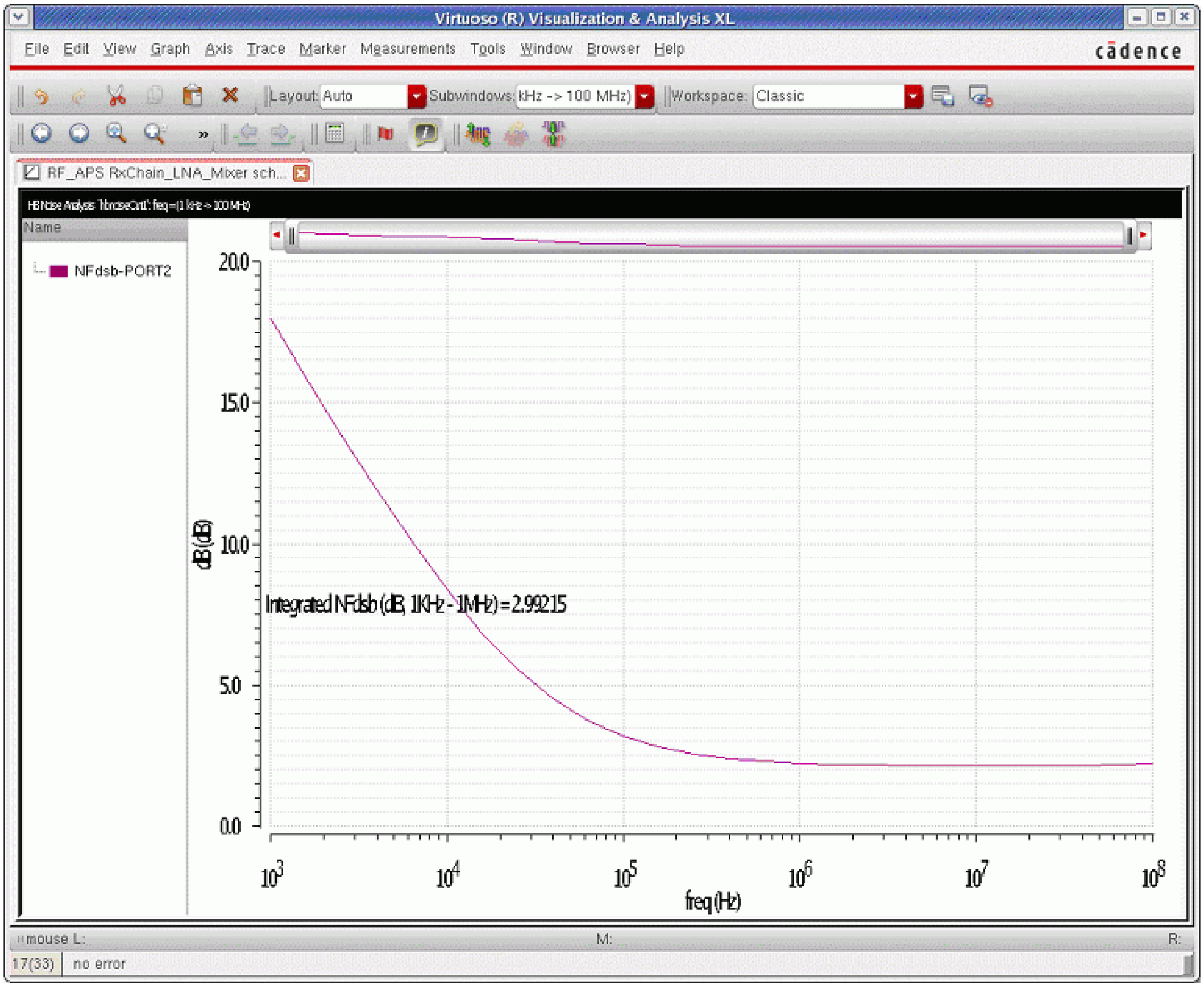


5. To plot the noise figure of the overall circuit, set the *Multiple Out* selection to the Mixer output at Port2. For direct conversion, usually double-sideband noise figure is desired.



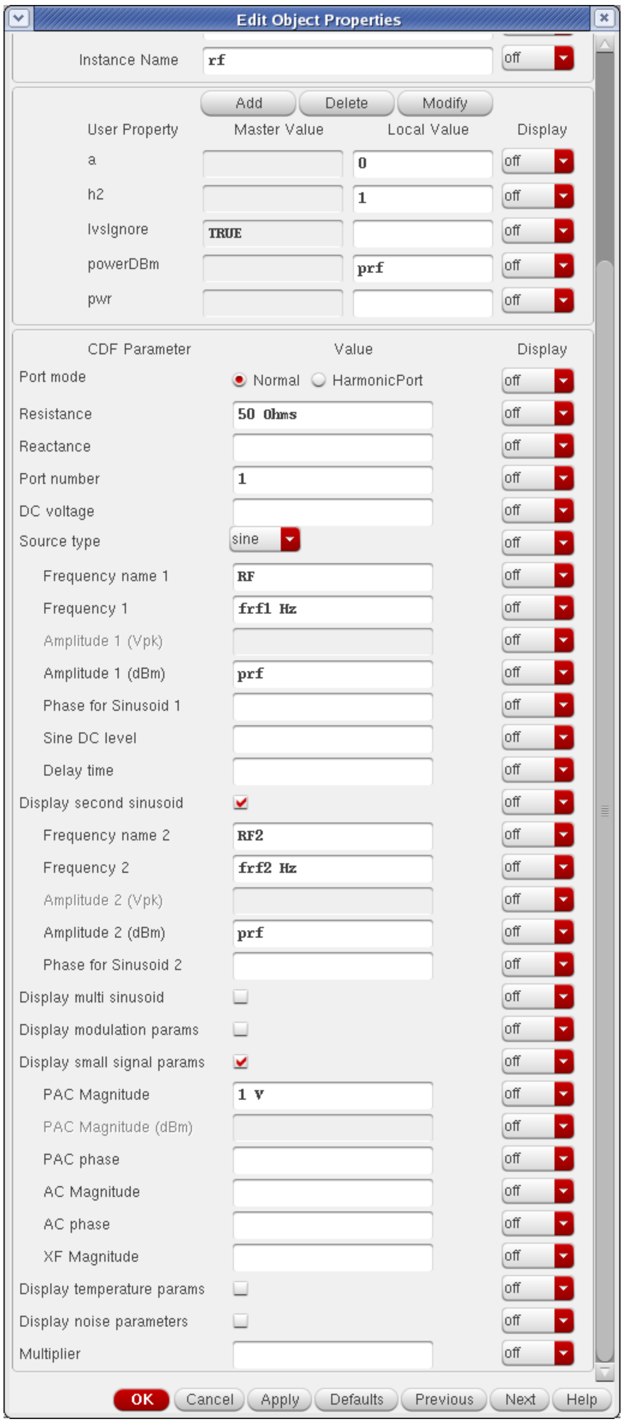
6. Click *Plot*.

The waveform tool shows the noise figure curve and the integrated noise figure to 1MHz.



Noise as a Function of Blocker Power

This is a classic desensitization measurement that measures noise figure as the blocker amplitude gets large. In order to accomplish this, variables need to be added to the RF source so that the input power of the blocker can be easily swept. The RF power in dBm is set by the variable prf in the following properties list.

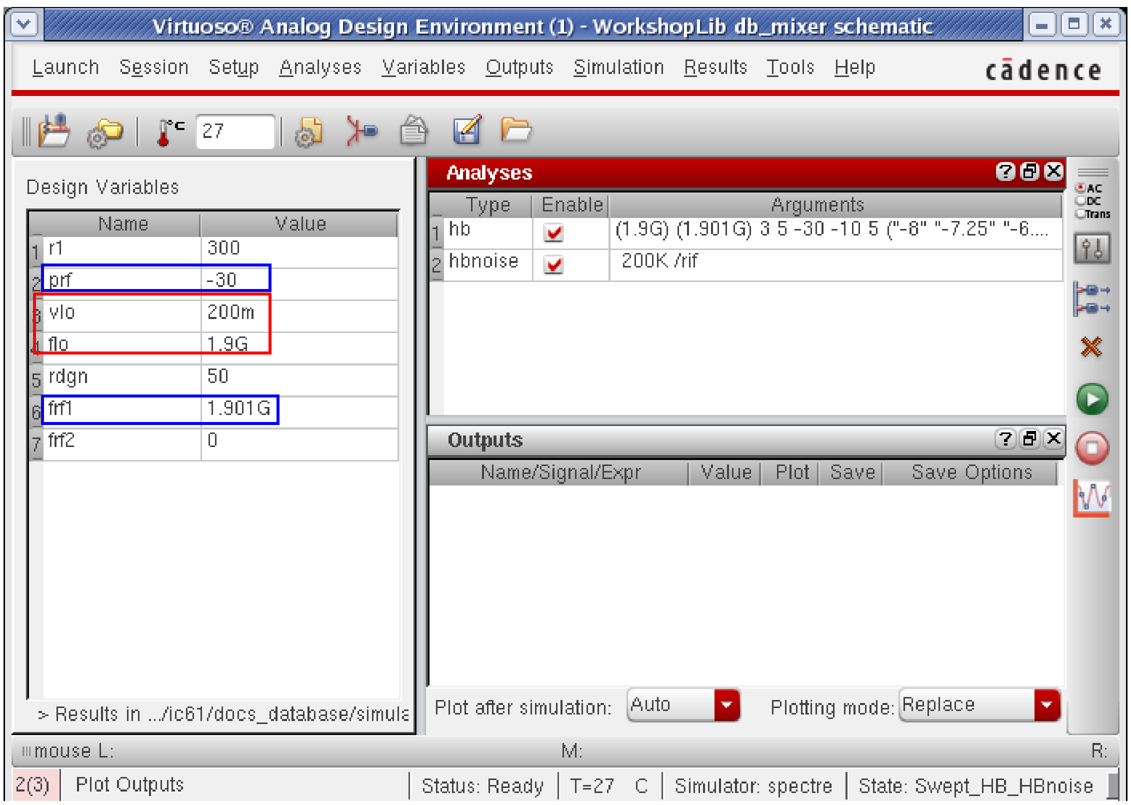


■ Note that the source has two large-signal tones set up.

- The first large-signal tone has the frequency set to the variable `frf1` and the amplitude set to `prf`.
- The second tone has the frequency set to the variable `frf2` and the amplitude set to `prf`. This is the blocker signal.

To measure the noise as a function of blocker power:

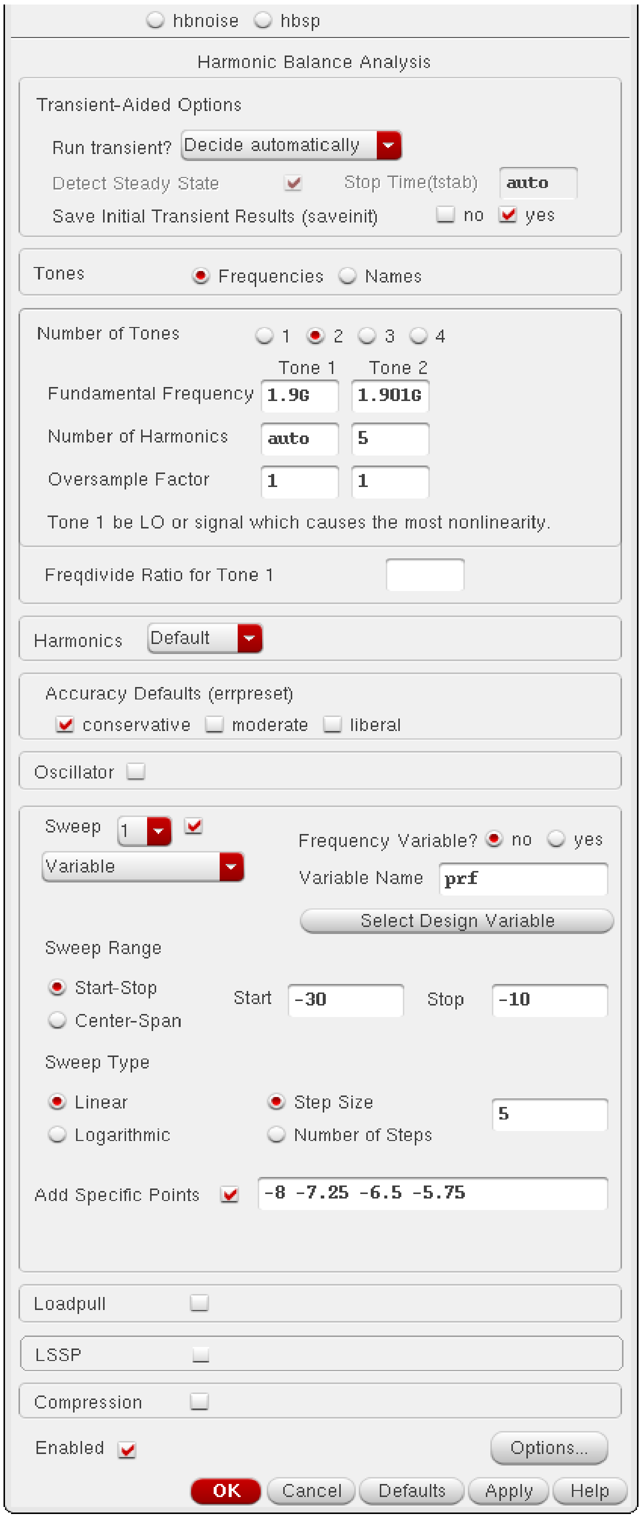
1. Apply the signals that cause the large-signal effects in the harmonic balance (hb) analysis, and then run an hbNoise analysis.



- ❑ The LO is set to 1.9GHz and 200mV peak. This is highlighted in red above.
- ❑ The first RF is set to 1.901G and -30 dBm. This is highlighted in blue above.
- ❑ The second RF tone is disabled by setting its frequency to zero.

  In this case, the frequency set by `frf1` is the blocker. Also in this case, the blocker frequency is set to the RF frequency so that both the gain compression plot from hb and the desensitization can be plotted in the same run. This is done so that plotting compression and plotting the noise figure with respect to RF input power can be done in the same simulation run. For a gain compression plot, the RF input frequency is the design RF input frequency. For a desensitization run, the RF frequency would be set to the blocker frequency which is usually farther away from the LO frequency than the frequency used in this example. The process is identical. Just the frequency is changed.

On the hb *Choosing Analyses* form:



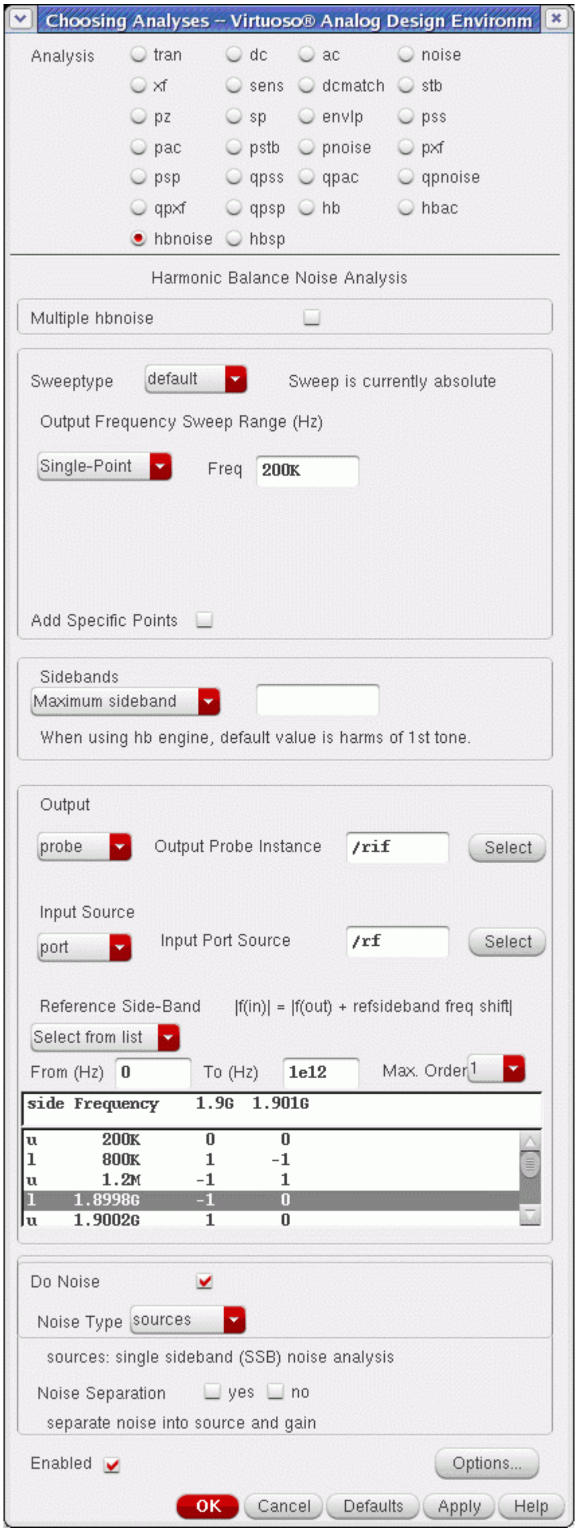
1. Set up the hb analysis appropriately.

   Because the RF power will be quite high, the number of RF harmonics is higher than normal to preserve the accuracy.

2. Sweep the RF power over an appropriate range.

   An appropriate range depends on what you are designing. Generally, you start about 15 to 25dB smaller than the compression point, and stop the sweep about 3 to 5 dB below the compression point. Additional points are defined with closer amplitude spacing so that enough points are taken to produce a smooth gain curve at high input power.

On the hbnoise *Choosing Analyses* form:



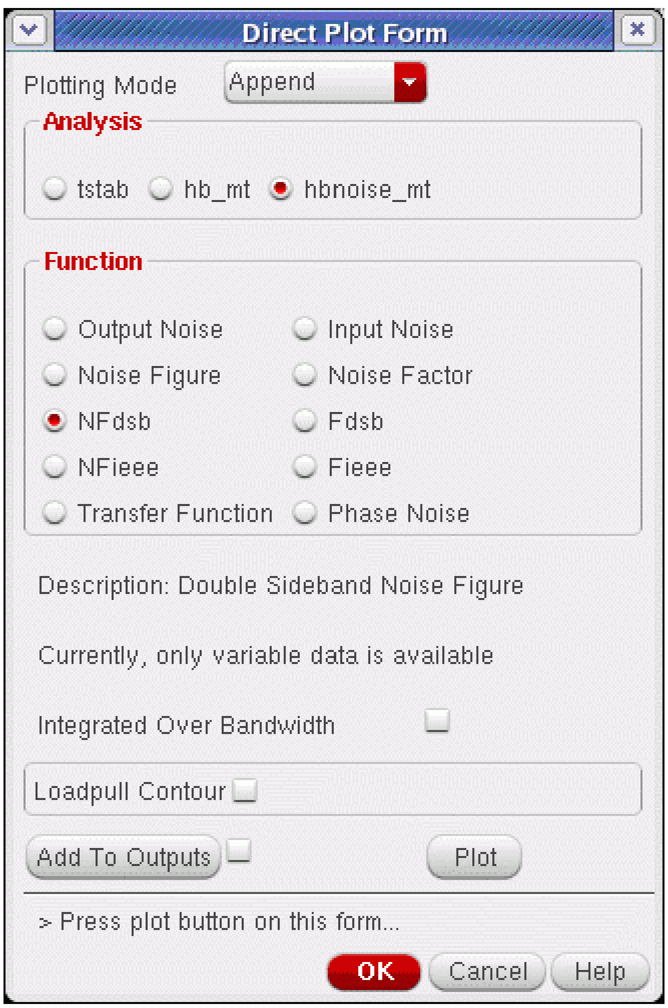
3. Usually, a single frequency is run in hbnoise, when the blocker power is swept. This is done to allow the direct plot functions to work properly after the simulation.

4. Specify the desired output frequency.

5. For *Maximum sideband*, either leave the field blank which causes SpectreRF to calculate all the noise mixing frequencies from all the harmonics set in the hb *Choosing Analyses* form, or specify the same number that is used to set the highest number of harmonics in the hb *Choosing Analyses* form.

6. The rest of the properties are set up as they would be in a normal simulation. Once you have finished setting up the hb and hbnoise *Choosing Analyses* forms, run the simulation.

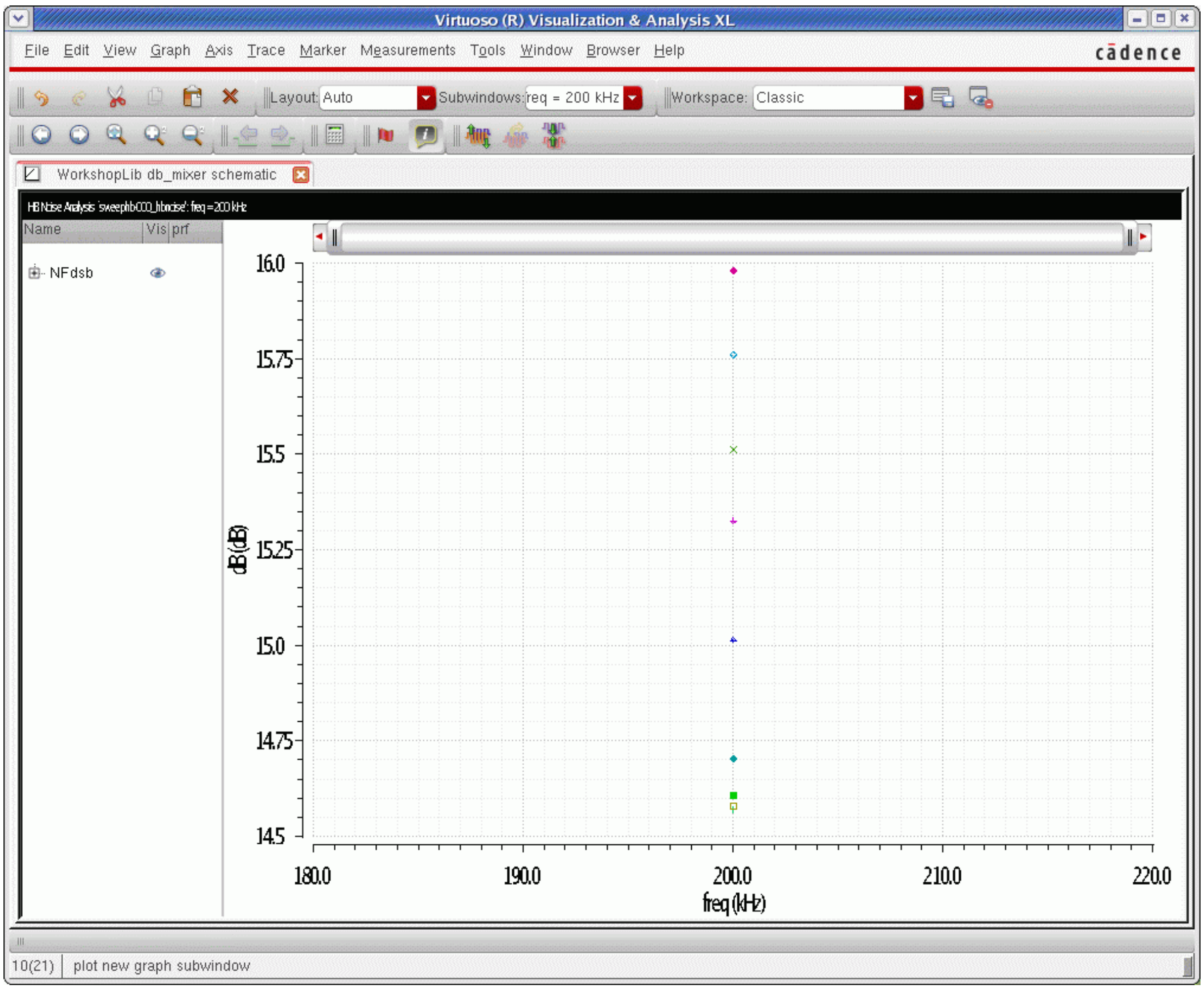
7. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE.



8. Select the desired noise figure measurement.

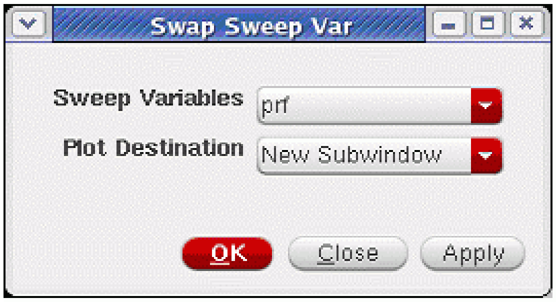
9. Click *Plot*.

The resulting plot appears.



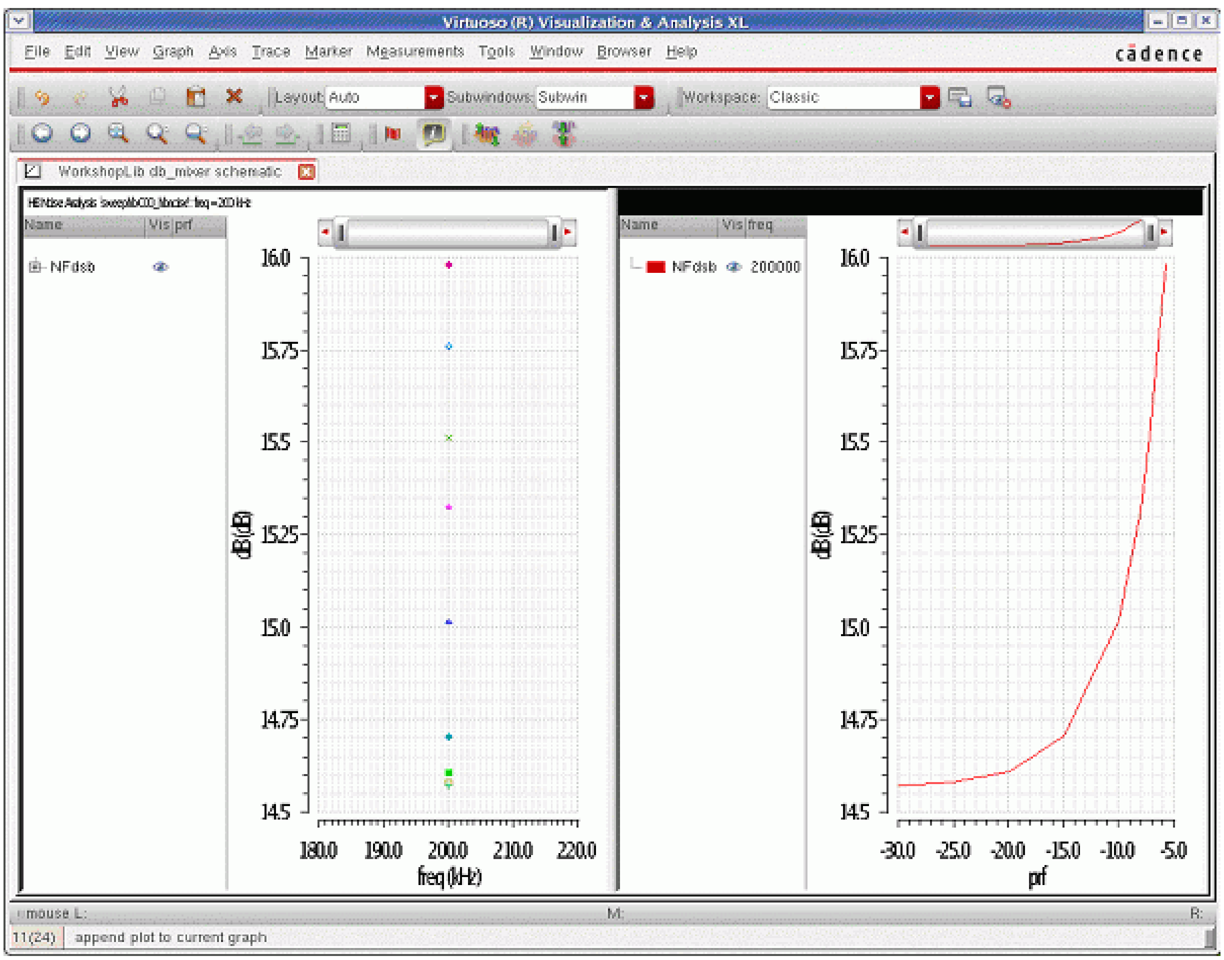
10. To make the plot more useful, right-click one of the numbers on the X Axis and select *Swap Sweep Var* from the context menu.

The *Swap Sweep Var* window is displayed. Note that the *Sweep Variables* field is set to *prf*, which is the variable that was swept.



11. Click *OK*.

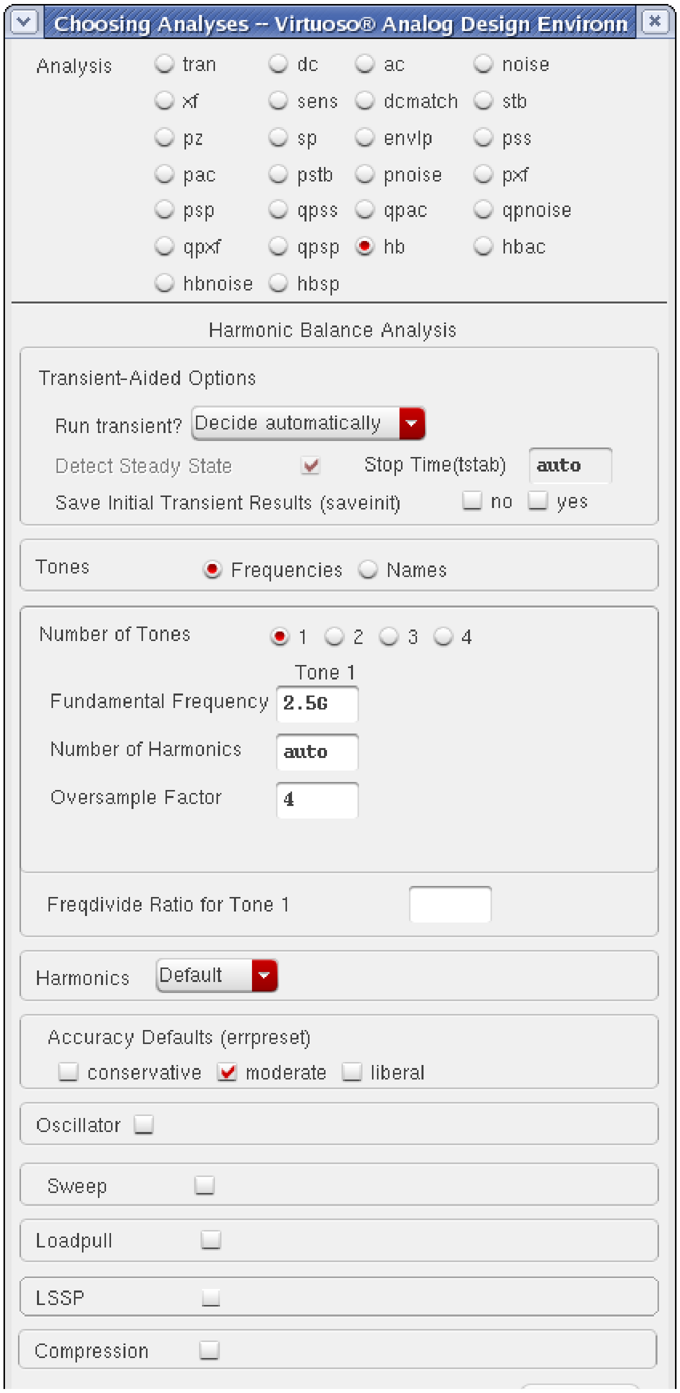
A new subwindow is displayed with the noise figure on the vertical axis, and the blocker power on the X Axis.



Noise Type = Modulated

Modulated allows the measurement of the AM and PM components of noise separately.

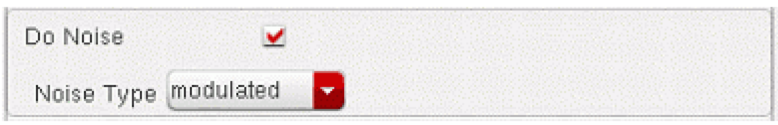
First, set up the hb analysis as appropriate for your circuit. For more information see the harmonic balance section of this chapter.



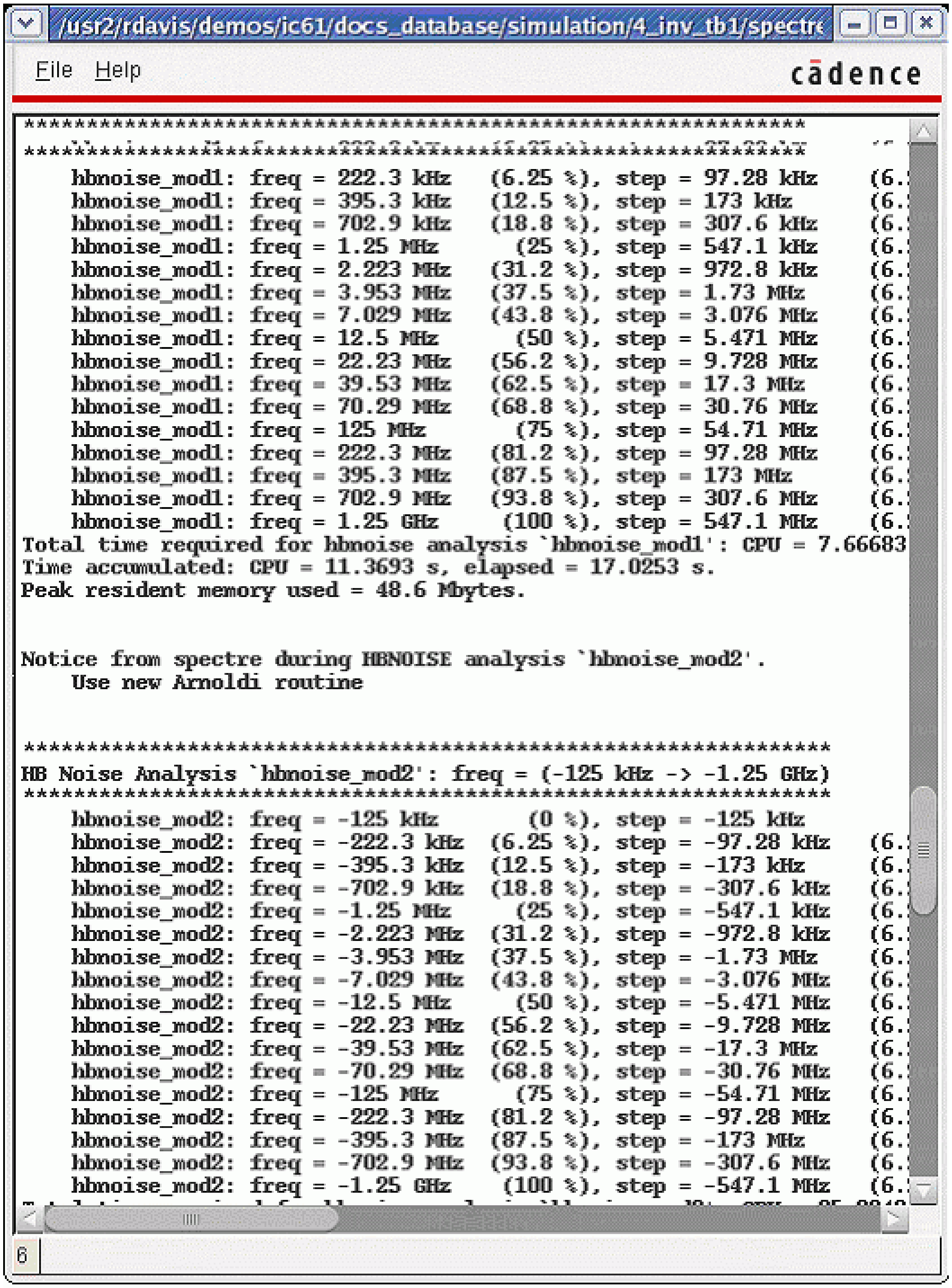
On the hbnoise *Choosing Analyses* form:

1. Set up an appropriate frequency range of interest.

2. Select *modulated* from the *Noise Type* drop-down list.



3. Run the simulation. Note that hbnoise modulated runs both the positive and negative frequencies. This allows both single-sideband and double-side band noise calculations to be plotted.

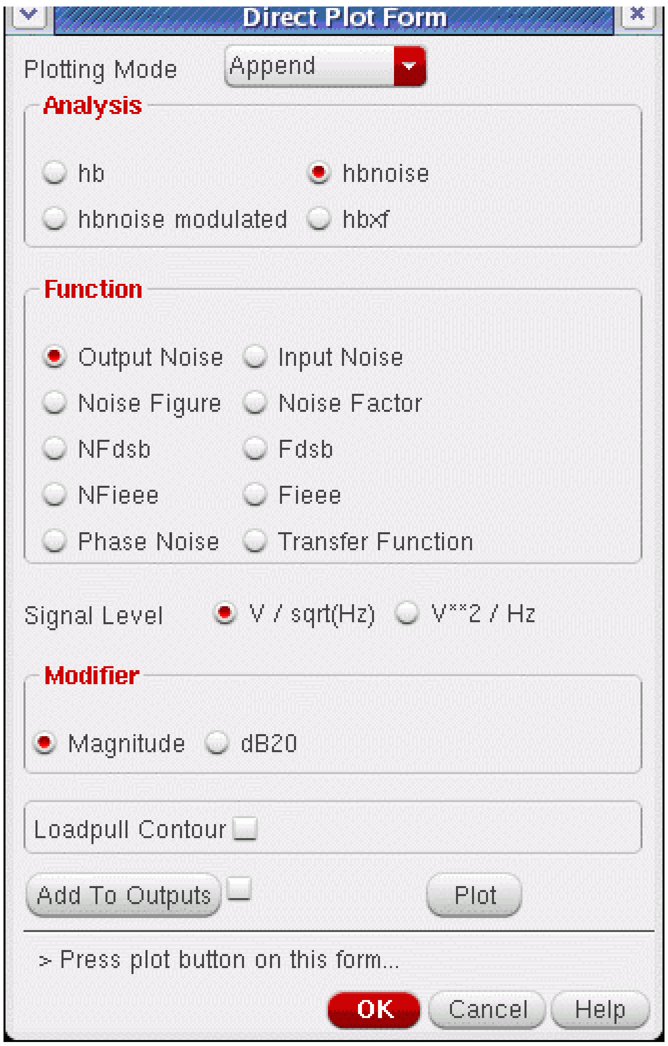


Some measurements are single-sideband and some measurements are double-sideband. The next steps will show the differences.

Now plot the output noise using the *Direct Plot Form*.

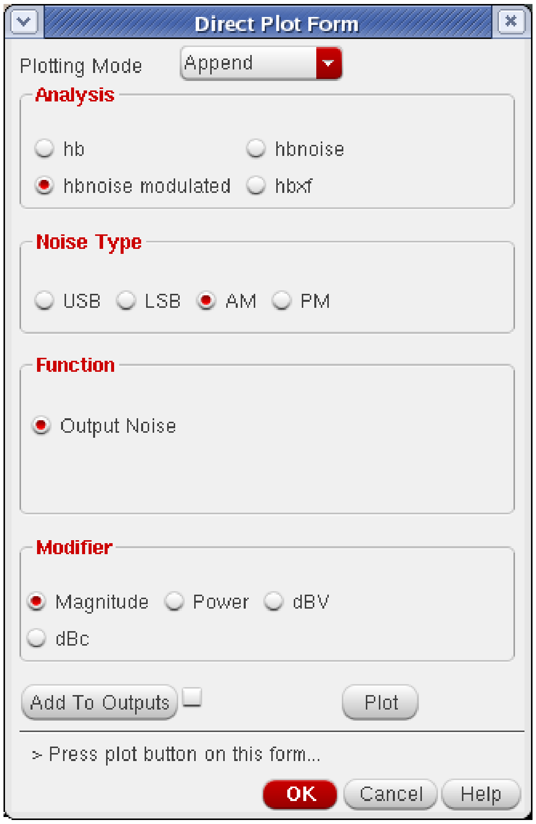
On the *Direct Plot Form*:

1. Select *hbnoise* results. All these measurements are single-sideband.
2. Select *Output Noise* from the *Function* section.

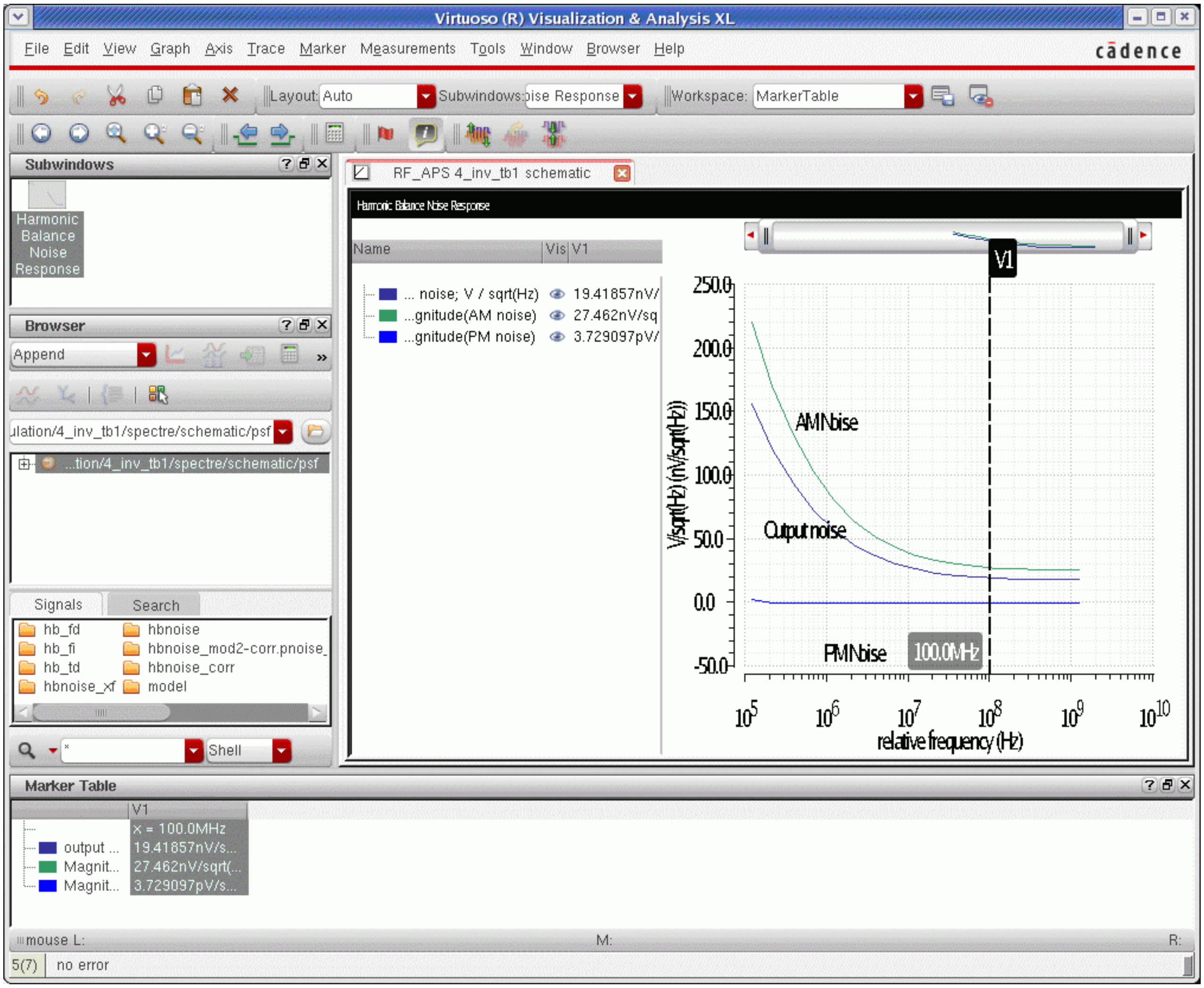


3. Click Plot.
4. Select *hbnoise modulated* results. All these measurements are double-sideband.

5. Select *AM* or *PM* from the *Noise Type* radio buttons. The *AM* noise is selected below.



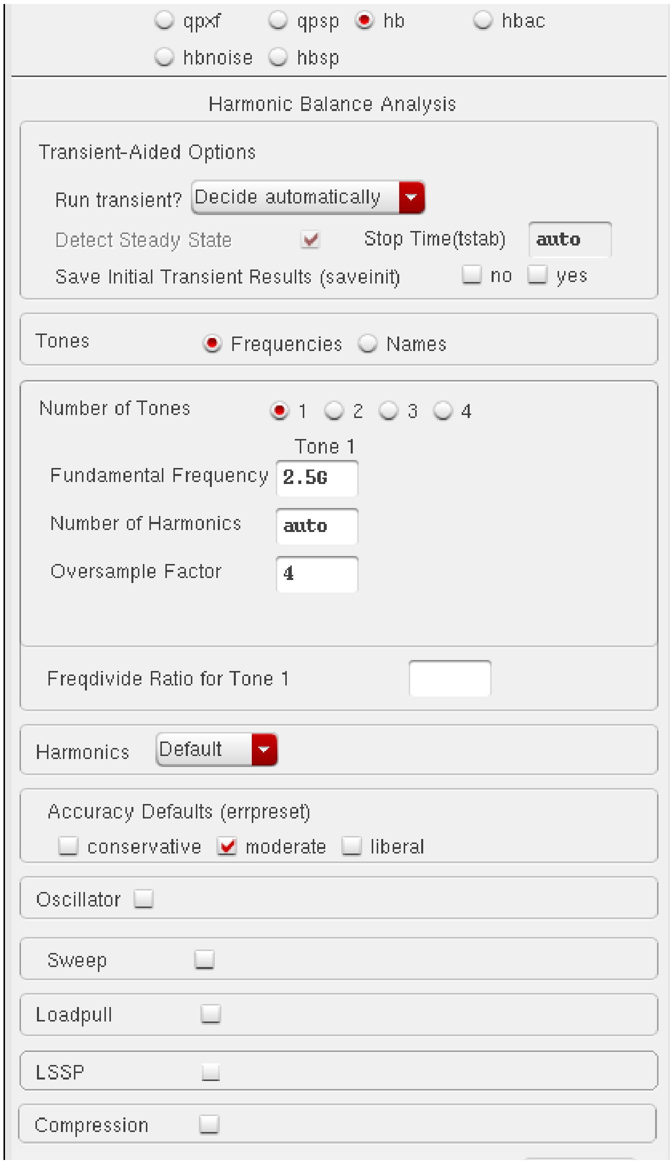
6. Click *Plot*.



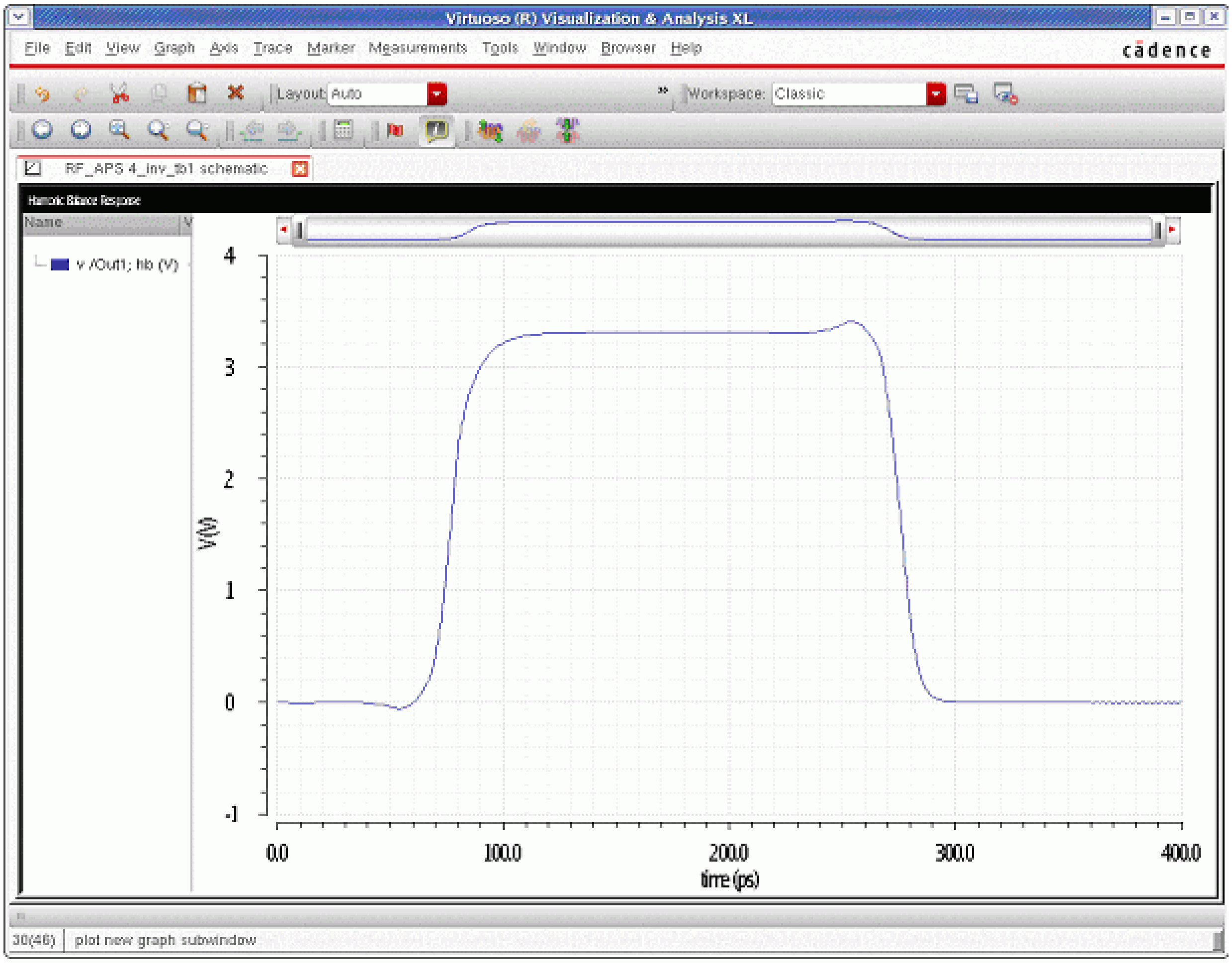
On the plot, note that the AM noise is 3dB larger than the output noise. This is because the output noise is derived from the positive frequencies only, and the AM and PM noise is derived using the positive and negative frequencies.

Noise Type = Jitter

Driven jitter is usually used for measuring the noise in a digital circuit at a specific threshold crossing. This is usually the voltage that causes succeeding stages to be clocked. In this case, *Decide automatically* and *auto* harmonics are suggested. These settings will automatically set the number of harmonics appropriately for the circuit.

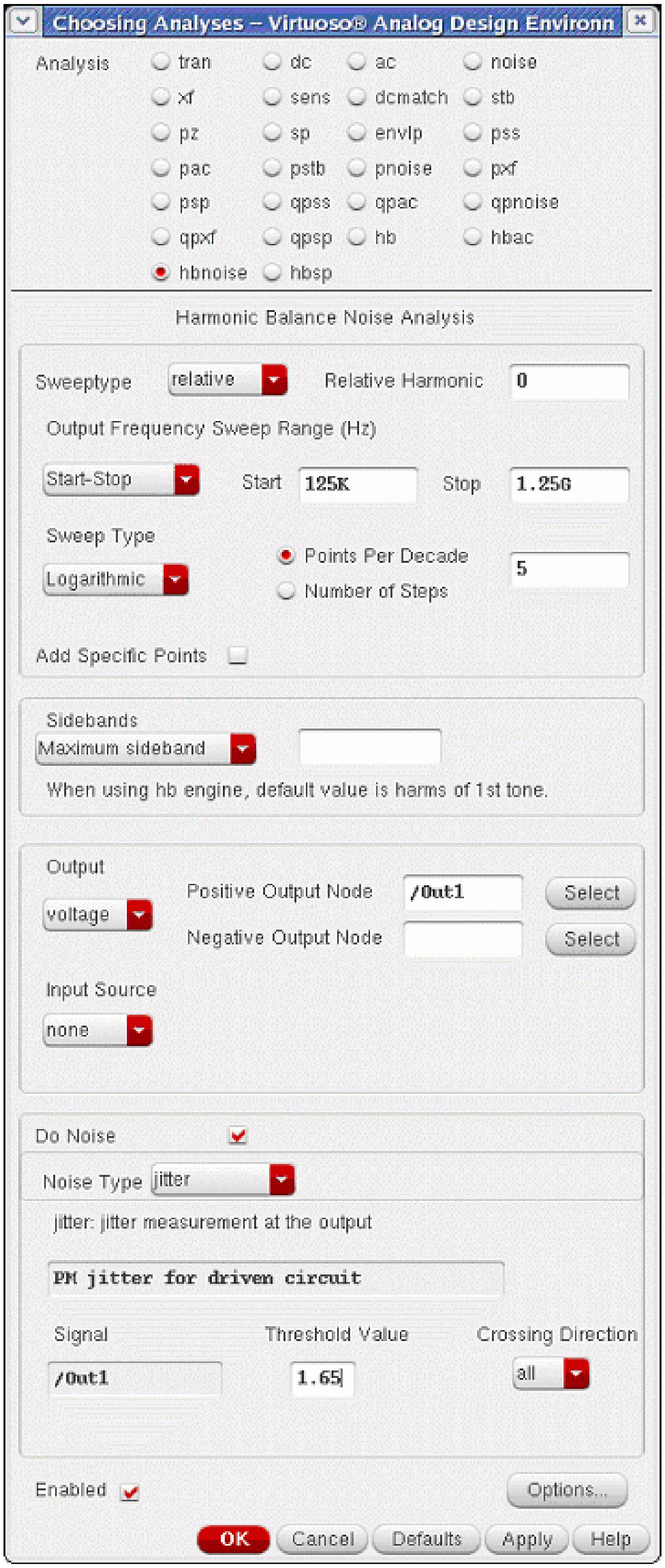


An appropriate threshold is needed, which can be determined by plotting the time domain waveform from the hb *Direct Plot Form*. The waveform is an ifft of the harmonics that were solved in harmonic balance. The waveform below shows an appropriate threshold value to be 1.65V. In most CMOS logic circuits, the clocking voltage is half the supply voltage.

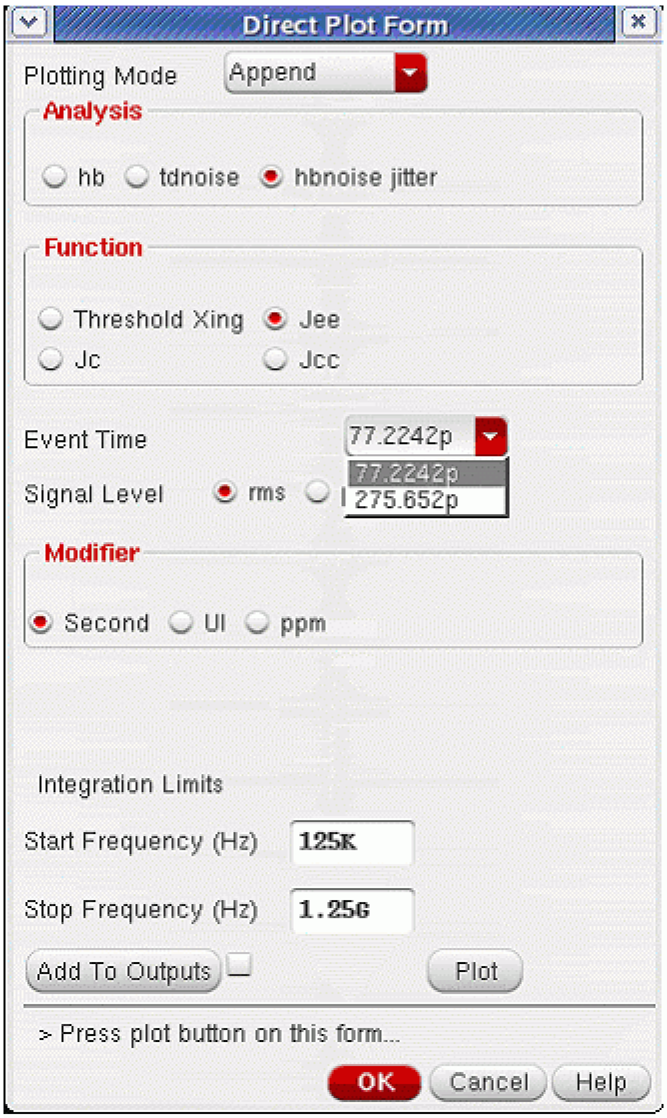


On the hbnoise *Choosing Analyses* form, the stop frequency is half the input frequency. The start frequency is four decades below that. About five points per decade is usually sufficient. *Maximum sideband* should be left blank. The *Output* field is set to *voltage*, and the *Input*

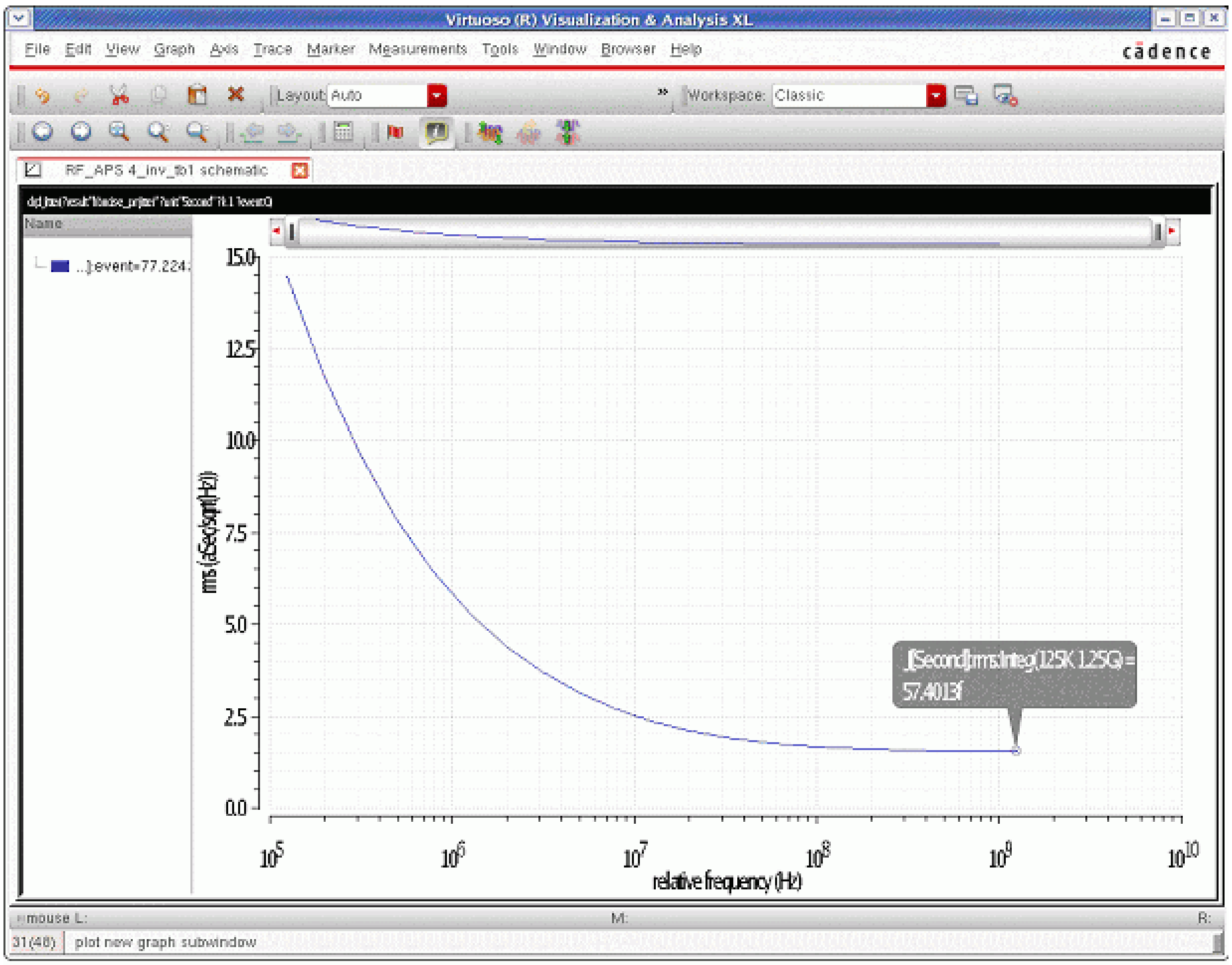
Source field is set to *none* because usually, only the output jitter is needed. The threshold is set appropriately. The *Crossing Direction* option can be set to *all*, *rise*, or *fall*.


Choosing Analyses – Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Harmonic Balance Noise Analysis
Sweeptype
relative
Relative Harmonic
0
Output Frequency Sweep Range (Hz)
Start-Stop
Start
125K
Stop
1.25G
Sweep Type
Logarithmic
Points Per Decade
5
Number of Steps
Add Specific Points
Sidebands
Maximum sideband
When using hb engine, default value is harms of 1st tone.
Output
Positive Output Node
/Out1
Select
voltage
Negative Output Node
Select
Input Source
none
Do Noise
Noise Type
jitter
jitter: jitter measurement at the output
PM jitter for driven circuit
Signal
Threshold Value
Crossing Direction
/Out1
1.65
all
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

When the simulation completes, use the *Direct Plot Form* to plot the jitter. For driven circuits, select *Jee*, which is edge-to-edge jitter. The event time is the time in the ifft from the hb analysis. In the below example, 77p is the rising edge, and 275p is the falling edge. *Event Time* makes the selection of the rising and falling edge. In the *Modifier* section, *Second* gives the Jee calculation in seconds. *UI* is Unit Interval (fractions of one period), and *ppm* is parts per million.



The output noise frequency response and the edge-to-edge jitter are displayed. The jee measurement is one standard deviation for the expected jitter.

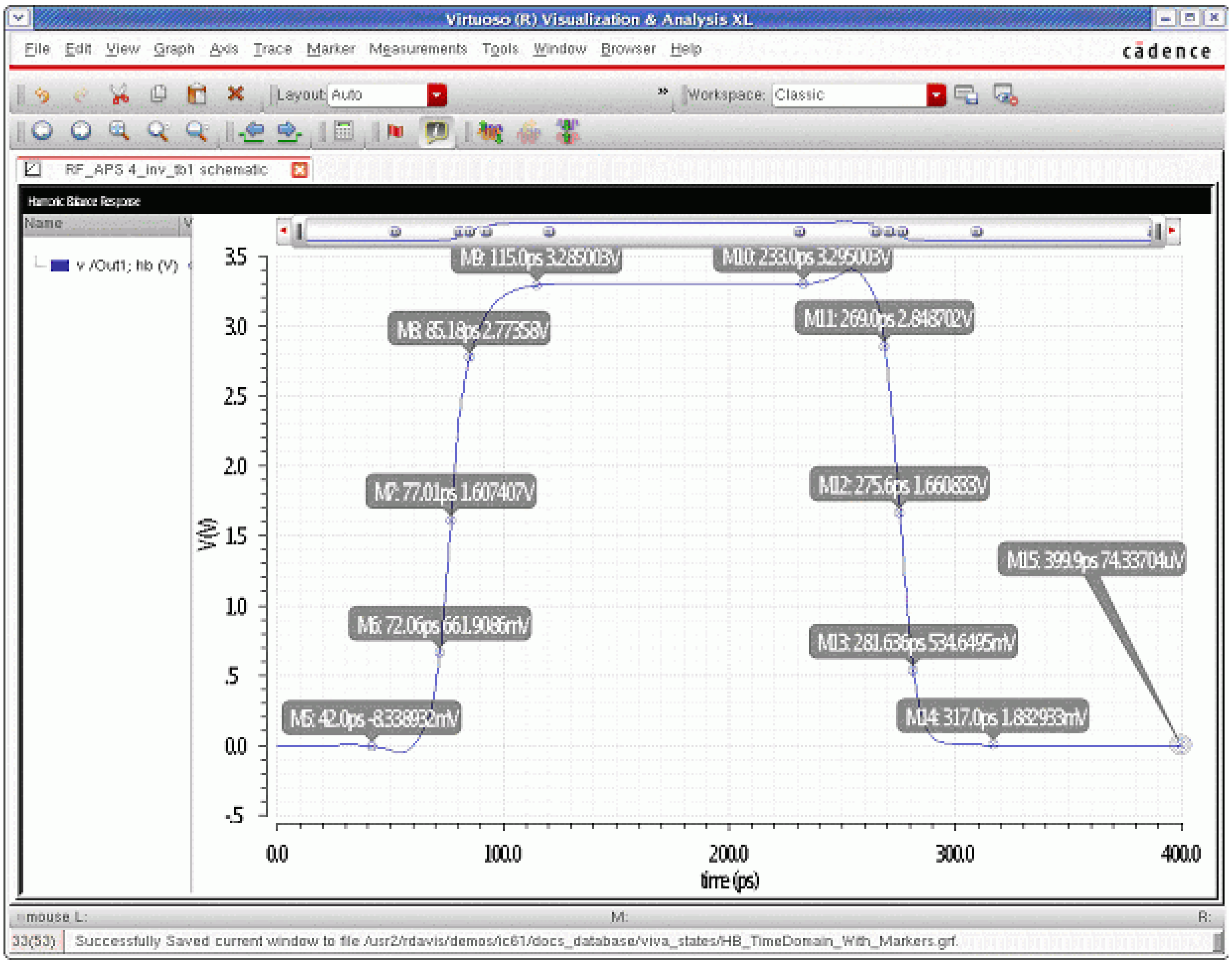


Noise Type = Timedomain

Timedomain noise is used to measure the output noise voltage at several timepoints in the ifft. This capability is similar to `noisetype=jitter`, but instead of being limited to a single threshold value, you can specify multiple times, and instead of having Jee directly in the *Direct Plot Form*, the calculation has to be done manually.

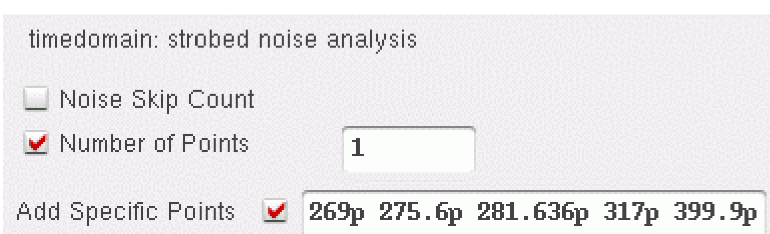
The hb setup is the same as the setup for jitter.

First run the hb analysis and plot the time-domain waveform. Position the markers at interesting timepoints, as shown below.



On the hbnoise *Choosing Analyses* form:

1. Select the *Number of Points* check box and type 1 in the text field. This causes the first noise point at time zero to be calculated.



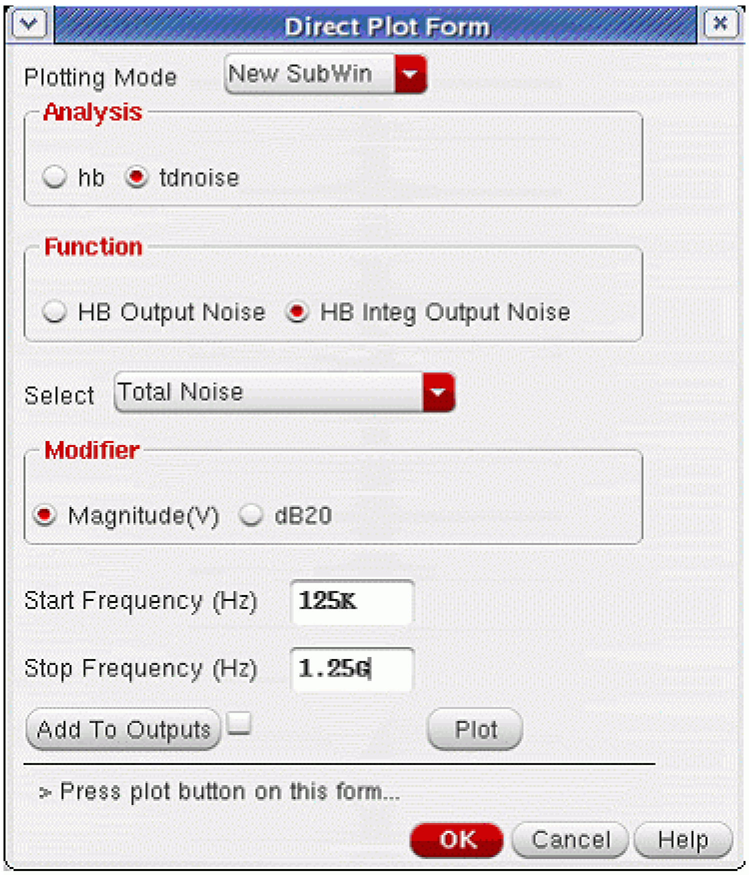
2. Select the *Add Specific Points* check box and enter the time values from the markers positioned in the waveform.

3. Run the simulation.

Because pnoise runs in the frequency-domain, to calculate the noise voltage, we need to integrate the frequency-domain data to calculate the noise. The hbnoise *Choosing Analyses* form defines the frequency range, and the number of frequency translations to consider.

On the *Direct Plot Form*:

1. Select *tdnoise* from the *Analysis* section.



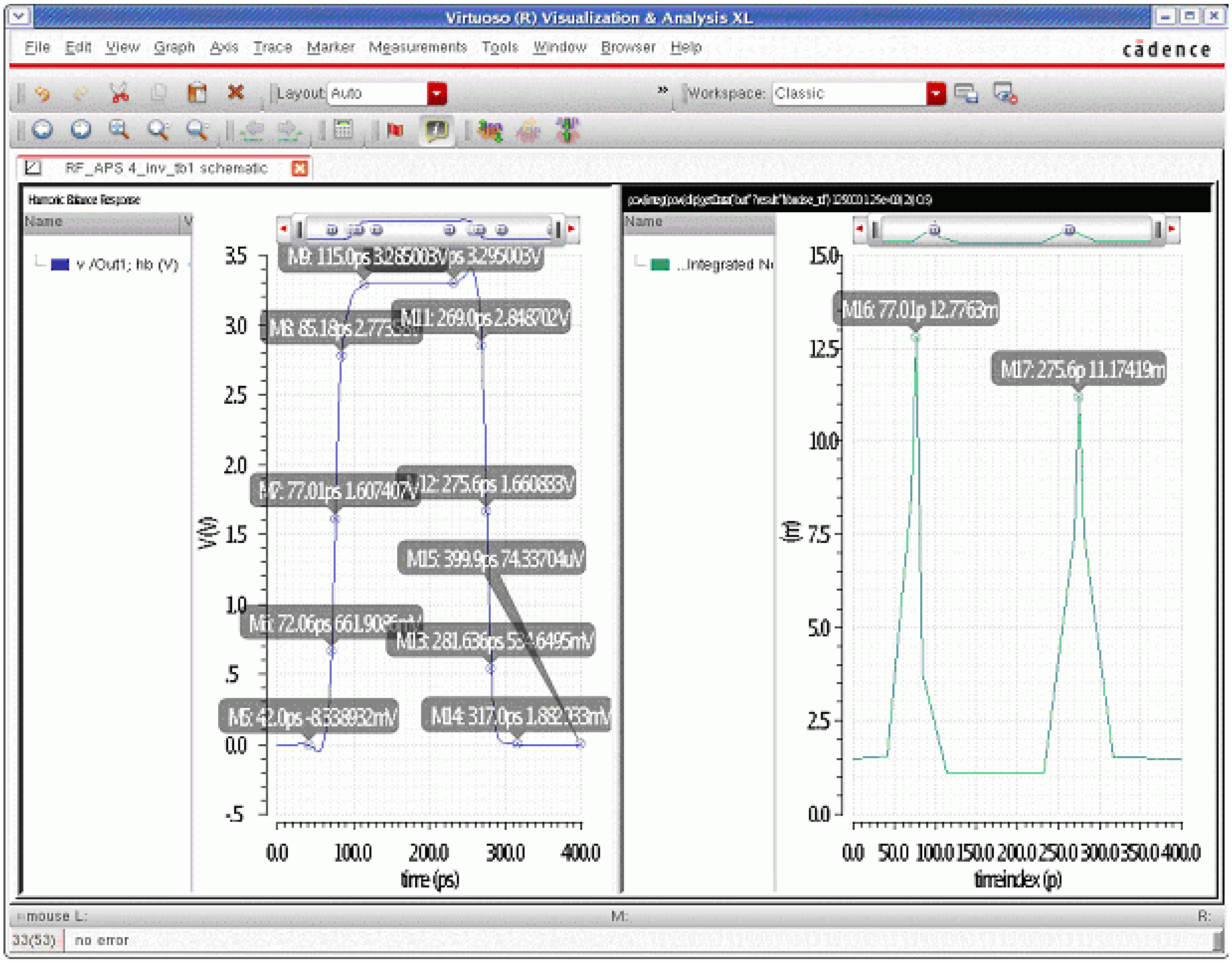
2. Select *HB Integ Output Noise* from the *Function* section.

3. To get the noise voltage, select *Magnitude* from the *Modifier* section.

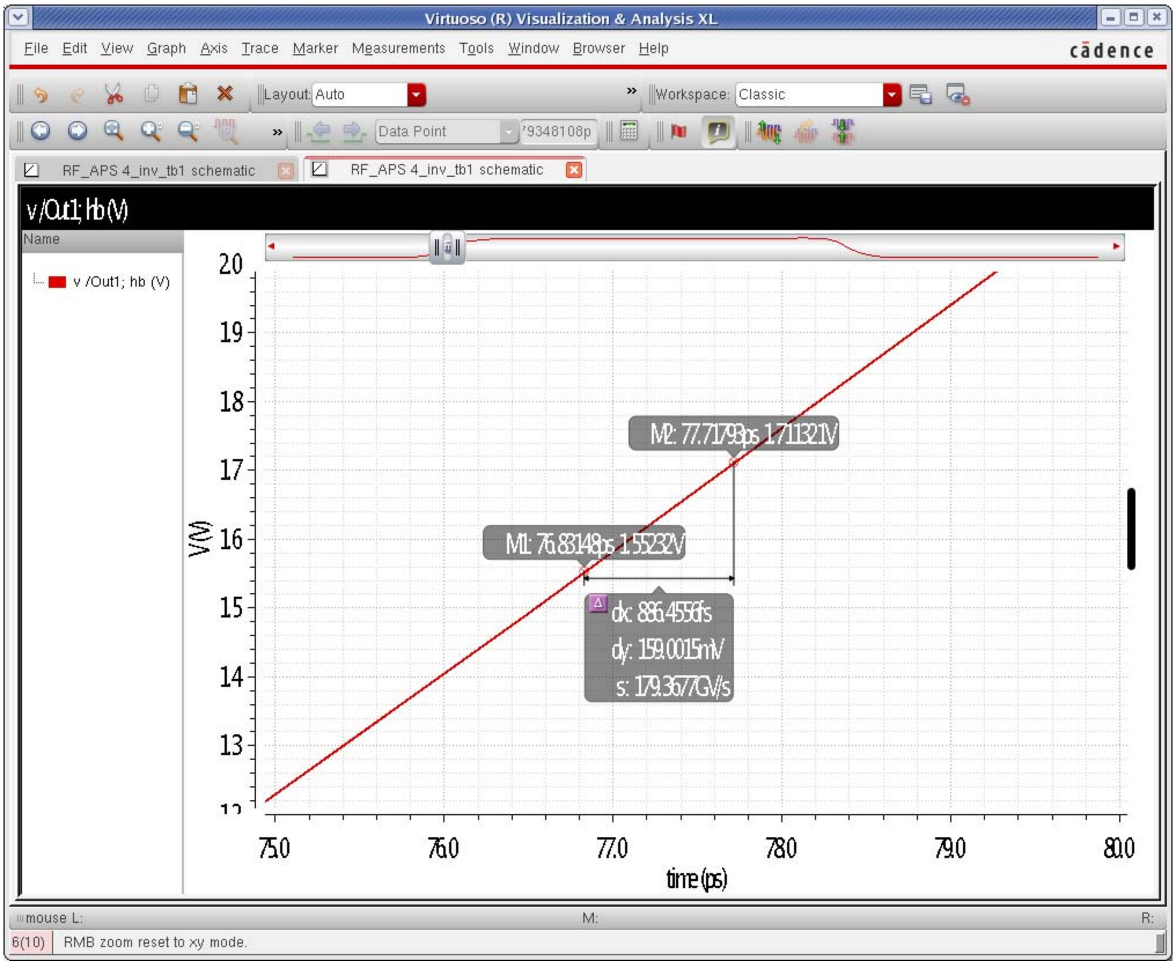
4. The frequency range in the *Direct Plot Form* should be the same as the frequency range defined in the hbnoise *Choosing Analyses* form. This integrates the noise with all the frequency translations over the frequency range defined in the hbnoise setup.

5. Click *Plot*.

The noise voltage is plotted versus time, as shown below. As expected, the maximum noise occurs near the middle of the waveform. Markers have been placed in both subwindows at the same times.



To get the jitter time, take the expression for the integrated noise voltage and divide by the slew rate. To determine the slew rate, zoom in to an area near the voltage you want for the measurement.



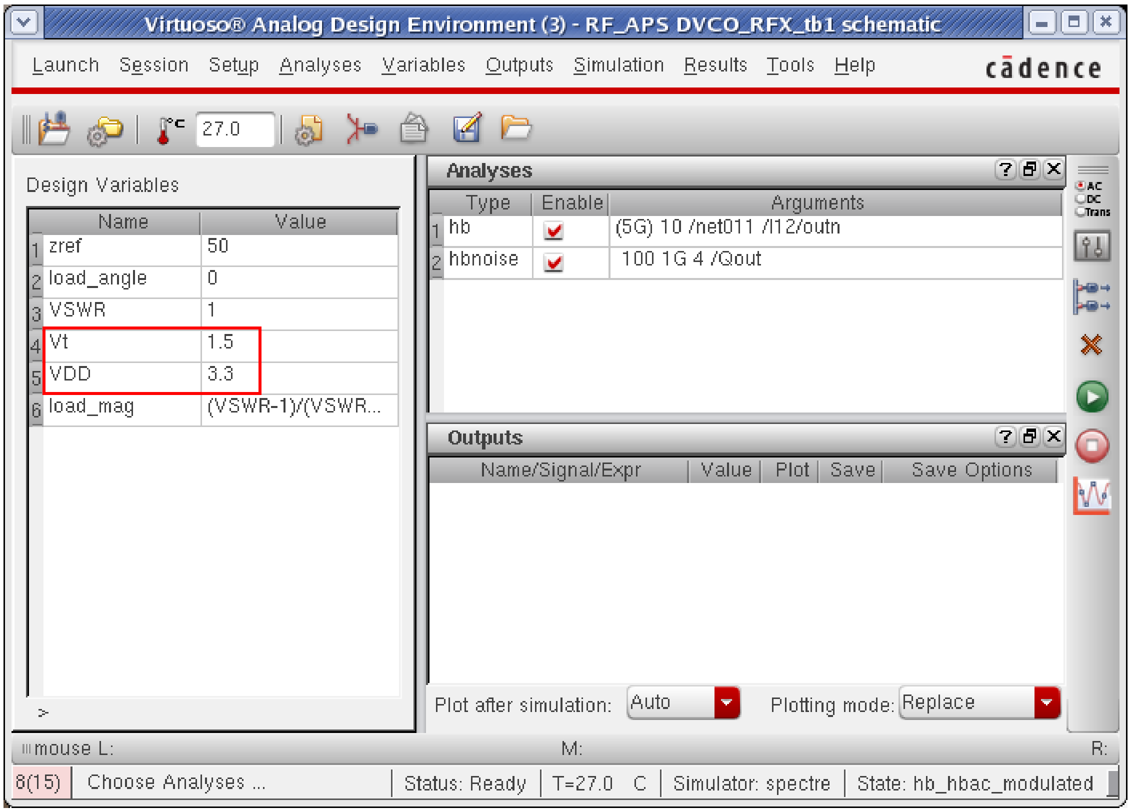
Place a marker at a voltage just below the threshold voltage you desire by moving the mouse near that point, and typing m. Now move to a point just above that voltage, and type d. The slew rate is `dy` divided by `dx`.

Oscillators

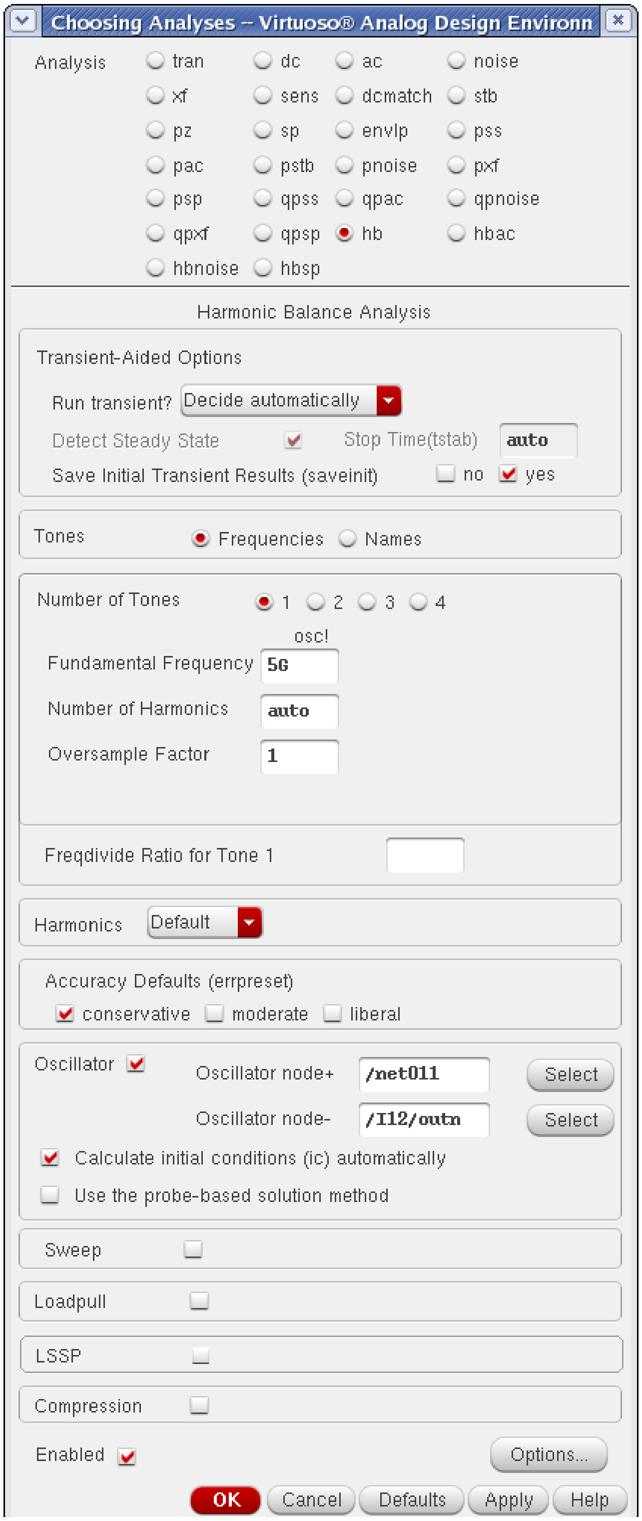
Noise Type = Sources

Noise type=sources is used to measure the phase noise that is averaged over the entire cycle of the oscillator waveform.

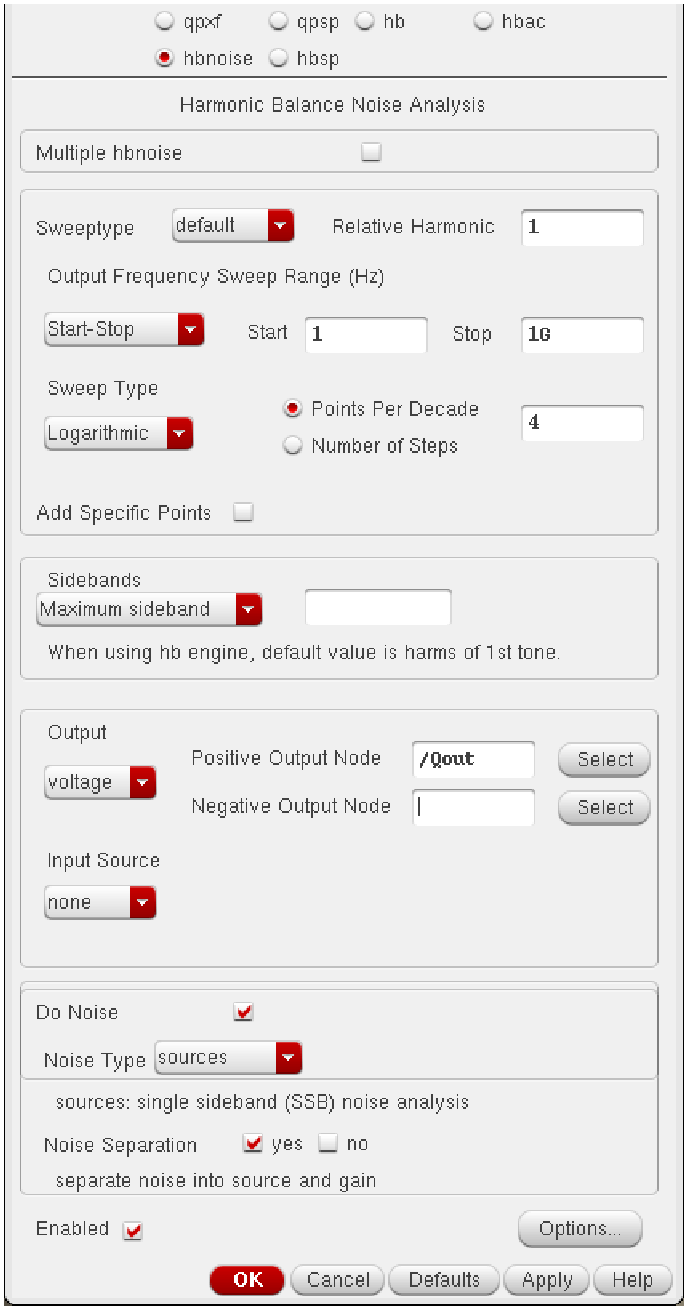
1. First, set variables to define the tuning voltage (vt) and power supply voltages (VDD) in the ADE.



2. Set up the hb form as shown in the following figure. For more information, see the hb section at the beginning of this chapter.



3. Next set up the hbnoise form (More information is provided below).

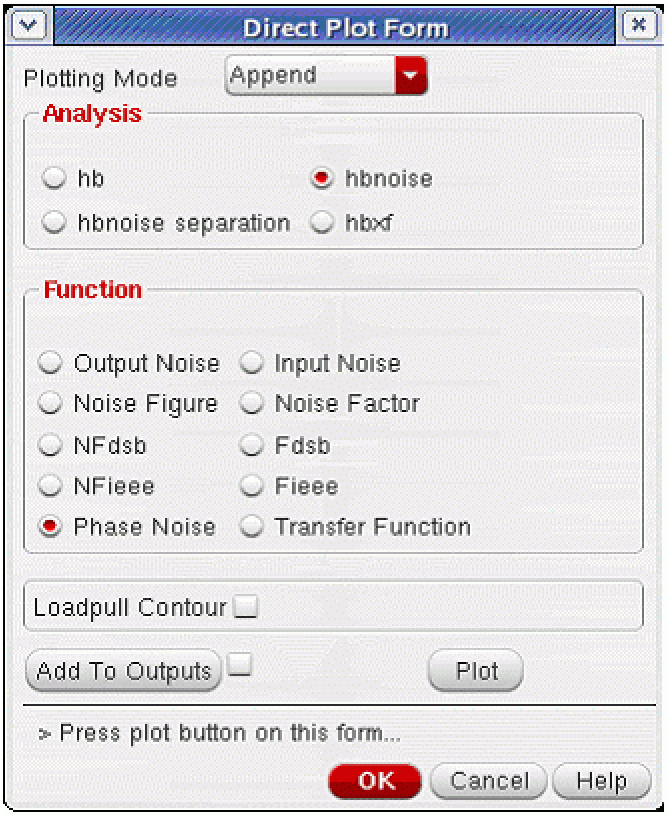


4. The sweep defaults to relative to the frequency of the specified harmonic of the hb analysis. The frequencies specified in the *Choosing Analyses* form are added to the frequency of the specified harmonic when *relative* is selected. In this case, the first harmonic is specified because there no frequency multiplier on the output of the oscillator.

5. Relative to the oscillator frequency, specify a frequency offset.
6. Either specify the number of harmonics for the hb, or leave the *Maximum sideband* field blank, which has the same effect.
7. Set *Output* to *voltage*.
8. Select the output node.
9. Set *Input Sources* to *none*. Input-referred noise is not defined for an oscillator.
10. If you want noise separation, you must first select *sources* from the *Noise Type* drop-down list and then you can select the noise separation.
11. Run the simulation.
12. To plot the single-sideband phase noise, in the ADE, select *Results - Direct Plot - Main Form*.

On the *Direct Plot Form*:

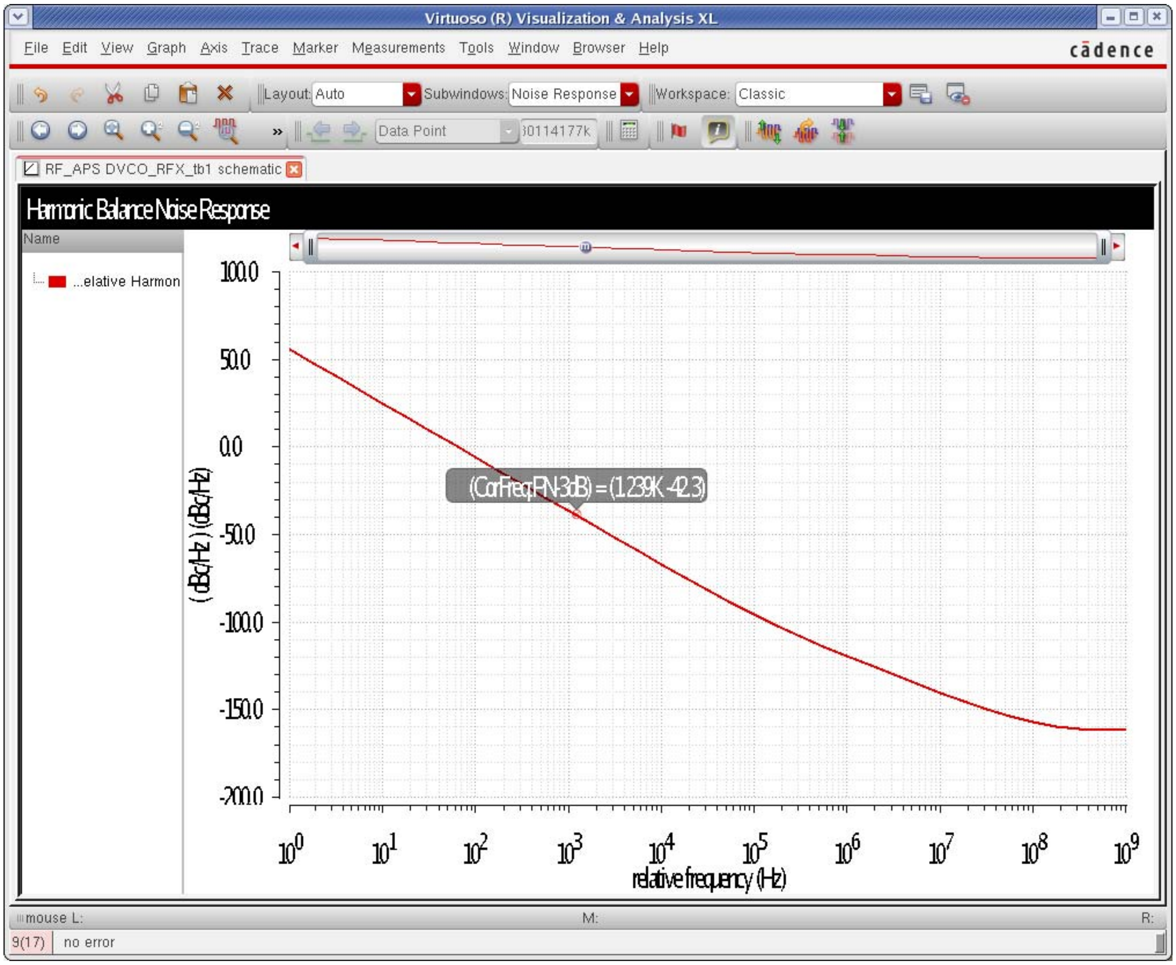
1. Select *hbnoise* from the *Analysis* section.



2. Select *Phase Noise* from the *Function* section.

Note that the phase noise from the hbnoise result set plots the noise that is in the frequencies above the oscillator frequency only (single sideband phase noise). If the double sideband phase noise (which includes the noise from below and above the oscillator output frequency) is desired, set the *Noise Type* to *jitter*. This is documented later.

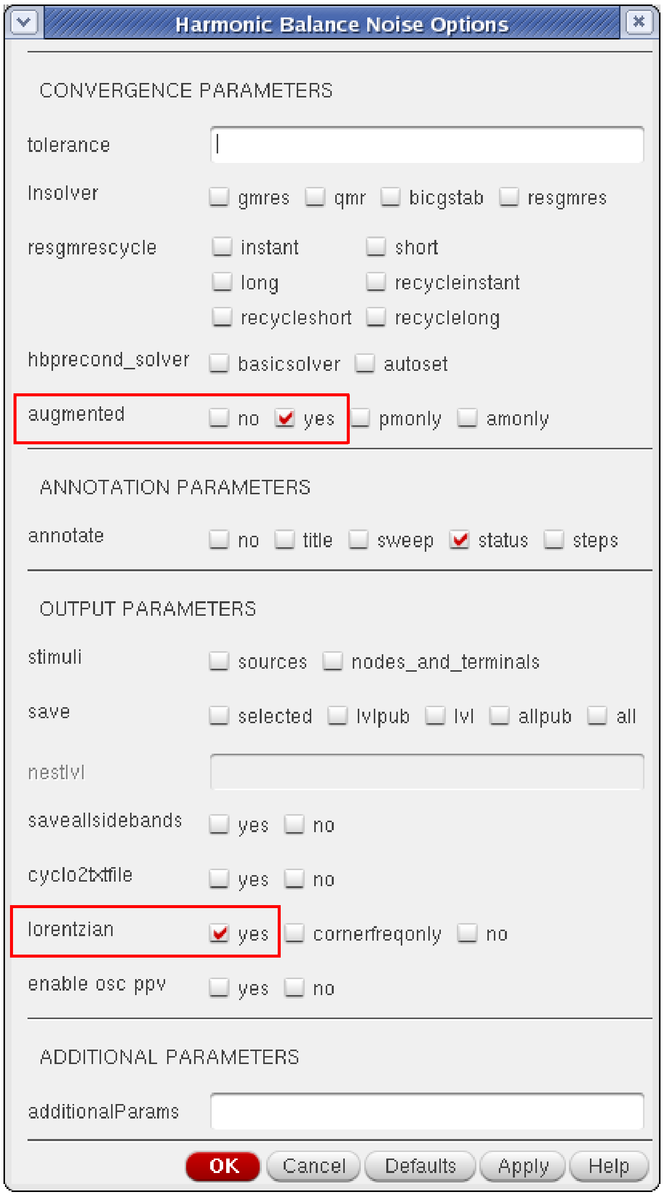
3. Click *Plot*.



Note that the phase noise continues above 0dBC/Hz, which indicates that the phase noise is larger than the amplitude of the oscillator output. This is because hbnoise is a small-signal analysis and it does not recognize the large-signal limits. The CorFreq label gives the frequency where the phase noise curve levels off at low frequency offset.

Commonly Used hbnoise Options for Oscillators

For the rest of the options, please see the Commonly Used hbnoise Options.



Augmented

augmented is the default and is required for most oscillators that have long time constants, perhaps from a Low Drop-Out (LDO) regulator. Either take the default (*yes*) or set *augmented* to *yes* for every circuit. Augmented is a new algorithm that works better than the standard hbnoise algorithm for the measurement of close-in phase noise especially when there are low frequency poles in the circuit.

Lorentzian

There are two ways of looking at phase noise. One way is to think of measuring the spectral output of the oscillator with a network analyzer. In this case, as the frequency gets close to the oscillator frequency, the amplitude starts going up, levels off, and then drops as the oscillator frequency is passed. In this case, the noise cannot be larger than the oscillations themselves, which is 0 dBc. If this is how you view phase noise, set the *lorentzian* option to *yes*.

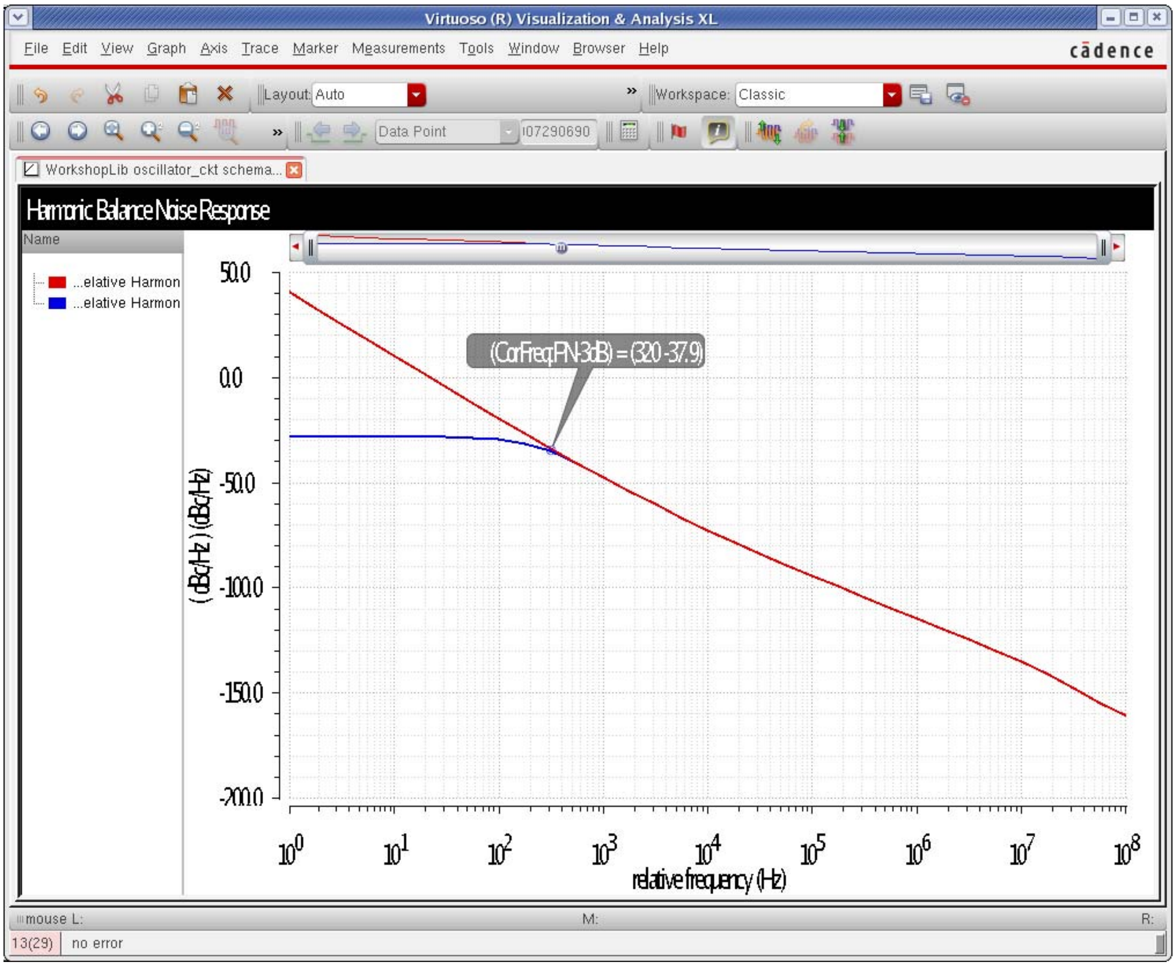
If you think in terms of jitter, in one cycle of the oscillator output, because of the noise, you can calculate one standard deviation of the timing. If you go two cycles, and you integrate the noise with respect to time, you get twice as much noise power, or about 1.4 times the noise voltage or jitter. When you get to an infinite number of oscillator cycles, the jitter also becomes infinite. As you increase the time, you are lowering the noise frequency, and in this case, as the time becomes large, the frequency becomes very low. At infinite time, you get infinite jitter, and this occurs at zero offset from the carrier. Jitter is just another way of looking at phase noise. In other words, the phase noise can easily exceed the carrier amplitude (0 dBc) as the offset frequency becomes small because it is heading to infinity at zero frequency offset. If this is how you view phase noise, set the *lorentzian* option to *no*.

Lorentzian=yes means calculate the leveling off. *Lorentzian=cornerfreqonly* calculates the continuously rising phase noise, but it places a marker on the phase noise curve at the frequency where the phase noise would level off. *no* causes the phase noise to continue to rise as the offset frequency approaches zero.

Comparison Between Lorentzian Yes and No

As noted above, lorentzian causes leveling off of the phase noise at low frequency.

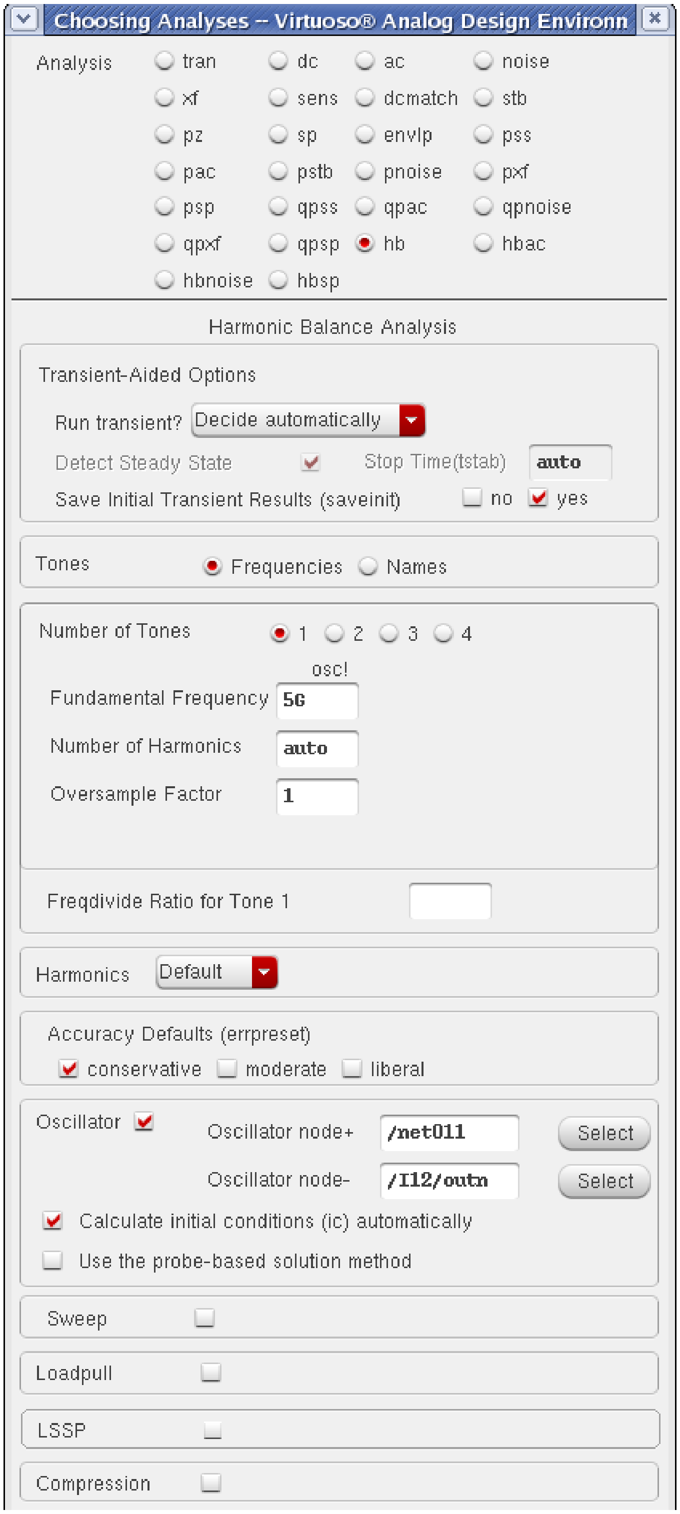
Note that when *lorentzian=no*, the phase noise curve goes above 0 dBc. This would mean that the noise is larger than the amplitude of the oscillator. This is because hbnoise is a small-signal analysis that does not recognize the large-signal limits of the circuit like clipping. It is similar to AC where you can specify an input magnitude of 1 megavolt, and see 10 megavolts on the output.



Noise Type = Modulated

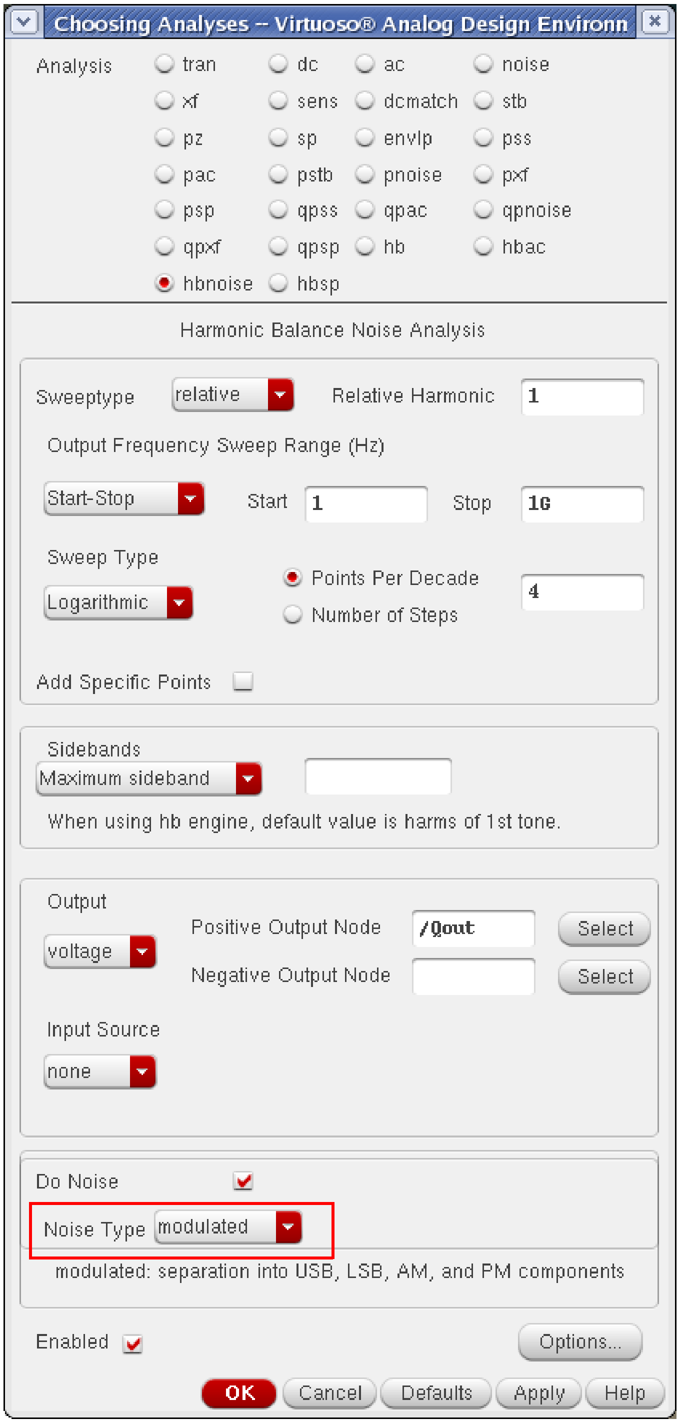
1. First set the hb form, as shown in the following figure. For more information, see the Harmonic Balance section at the beginning of this chapter.

Because your circuit is different, you might need a different estimate of the oscillation frequency and number of harmonics.



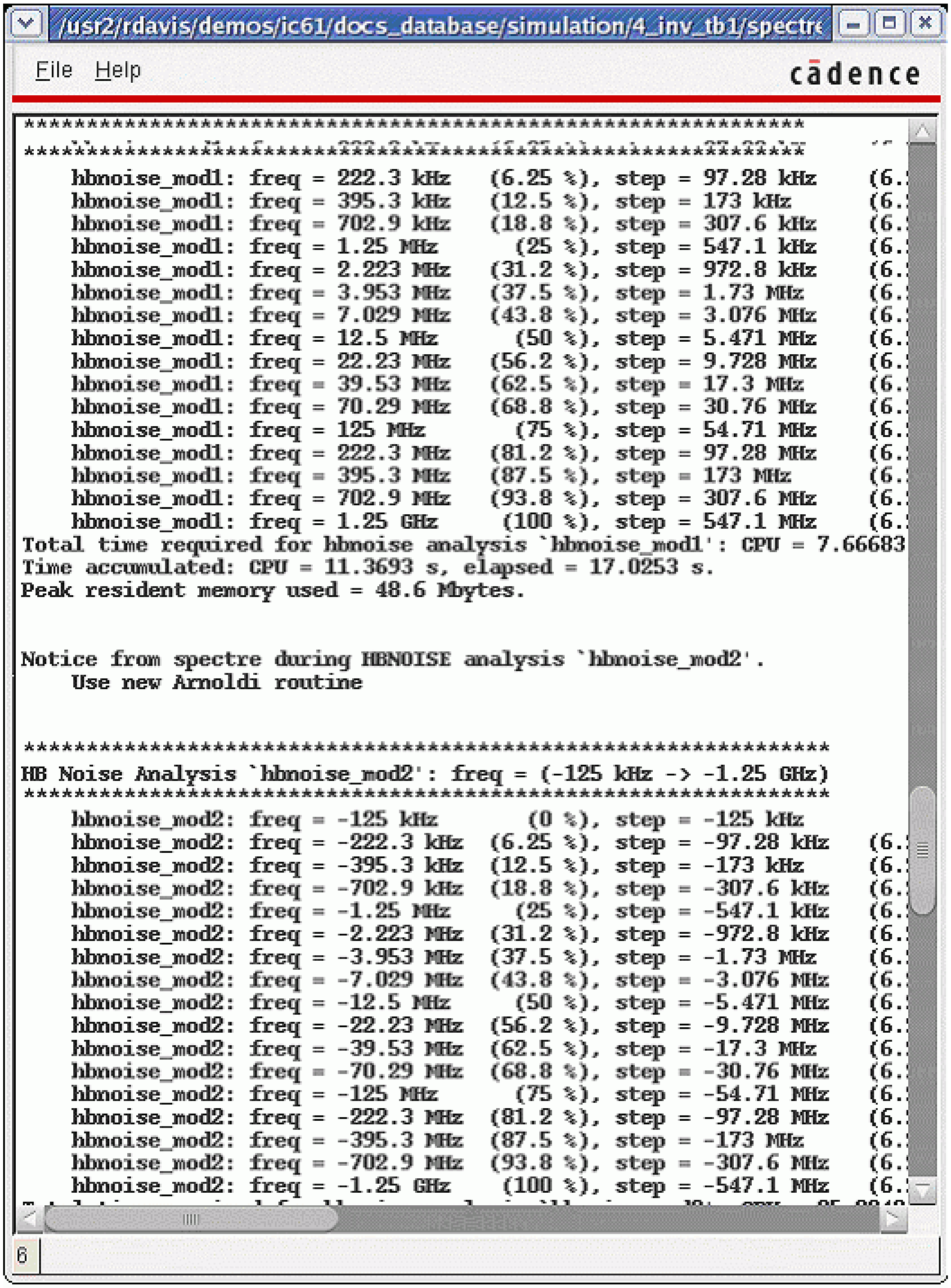
Next, set the hbnoise form.

2. Select *modulated* from the *Noise Type* drop-down list.



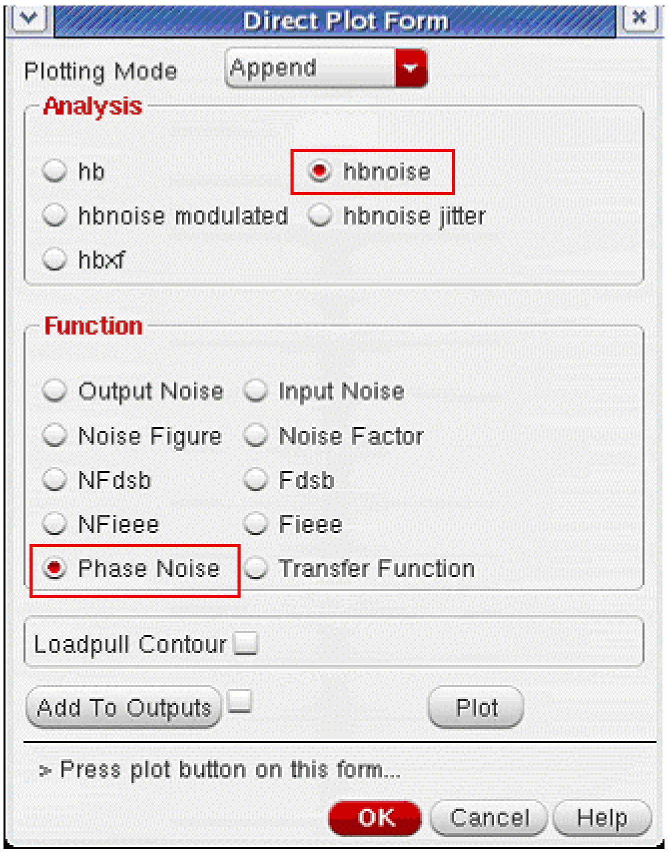
3. Run the simulation.

Note that positive and negative frequencies are run in hbnoise. Since *relative* was chosen for the *Sweep Type*, the frequencies above and below the first harmonic are analyzed for noise.

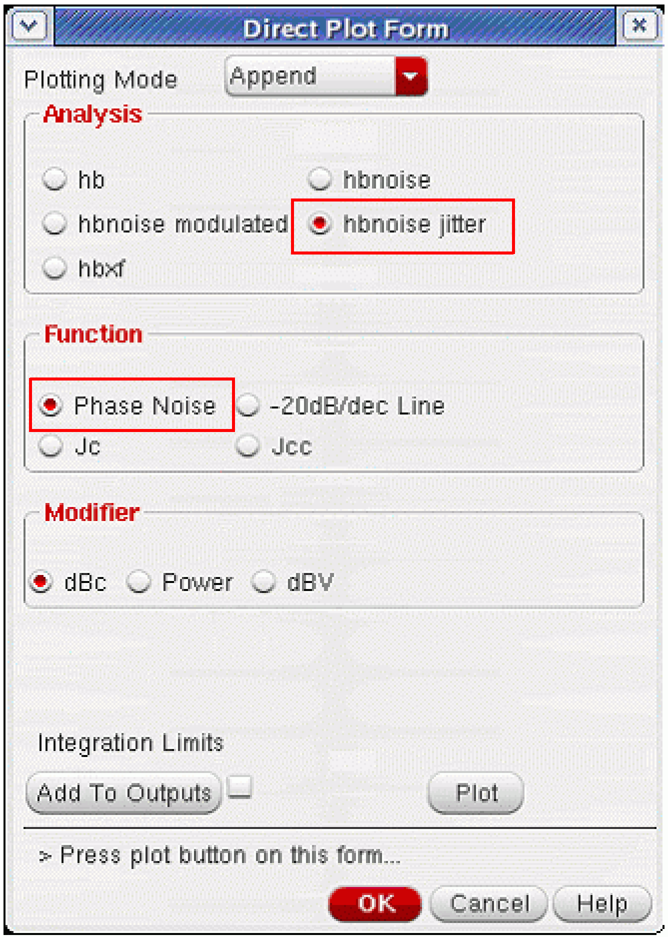



```
*****************************************************************
*****************************************************************
    hbnoise_mod1: freq = 222.3 kHz   (6.25 %), step = 97.28 kHz    (6.
    hbnoise_mod1: freq = 395.3 kHz   (12.5 %), step = 173 kHz      (6.
    hbnoise_mod1: freq = 702.9 kHz   (18.8 %), step = 307.6 kHz    (6.
    hbnoise_mod1: freq = 1.25 MHz      (25 %), step = 547.1 kHz    (6.
    hbnoise_mod1: freq = 2.223 MHz   (31.2 %), step = 972.8 kHz    (6.
    hbnoise_mod1: freq = 3.953 MHz   (37.5 %), step = 1.73 MHz     (6.
    hbnoise_mod1: freq = 7.029 MHz   (43.8 %), step = 3.076 MHz    (6.
    hbnoise_mod1: freq = 12.5 MHz      (50 %), step = 5.471 MHz    (6.
    hbnoise_mod1: freq = 22.23 MHz   (56.2 %), step = 9.728 MHz    (6.
    hbnoise_mod1: freq = 39.53 MHz   (62.5 %), step = 17.3 MHz     (6.
    hbnoise_mod1: freq = 70.29 MHz   (68.8 %), step = 30.76 MHz    (6.
    hbnoise_mod1: freq = 125 MHz       (75 %), step = 54.71 MHz    (6.
    hbnoise_mod1: freq = 222.3 MHz   (81.2 %), step = 97.28 MHz    (6.
    hbnoise_mod1: freq = 395.3 MHz   (87.5 %), step = 173 MHz      (6.
    hbnoise_mod1: freq = 702.9 MHz   (93.8 %), step = 307.6 MHz    (6.
    hbnoise_mod1: freq = 1.25 GHz     (100 %), step = 547.1 MHz    (6.
Total time required for hbnoise analysis `hbnoise_mod1': CPU = 7.66683
Time accumulated: CPU = 11.3693 s, elapsed = 17.0253 s.
Peak resident memory used = 48.6 Mbytes.


Notice from spectre during HBNOISE analysis `hbnoise_mod2'.
    Use new Arnoldi routine


*****************************************************************
HB Noise Analysis `hbnoise_mod2': freq = (-125 kHz -> -1.25 GHz)
*****************************************************************
    hbnoise_mod2: freq = -125 kHz       (0 %), step = -125 kHz
    hbnoise_mod2: freq = -222.3 kHz  (6.25 %), step = -97.28 kHz   (6.
    hbnoise_mod2: freq = -395.3 kHz  (12.5 %), step = -173 kHz     (6.
    hbnoise_mod2: freq = -702.9 kHz  (18.8 %), step = -307.6 kHz   (6.
    hbnoise_mod2: freq = -1.25 MHz     (25 %), step = -547.1 kHz   (6.
    hbnoise_mod2: freq = -2.223 MHz  (31.2 %), step = -972.8 kHz   (6.
    hbnoise_mod2: freq = -3.953 MHz  (37.5 %), step = -1.73 MHz    (6.
    hbnoise_mod2: freq = -7.029 MHz  (43.8 %), step = -3.076 MHz   (6.
    hbnoise_mod2: freq = -12.5 MHz     (50 %), step = -5.471 MHz   (6.
    hbnoise_mod2: freq = -22.23 MHz  (56.2 %), step = -9.728 MHz   (6.
    hbnoise_mod2: freq = -39.53 MHz  (62.5 %), step = -17.3 MHz    (6.
    hbnoise_mod2: freq = -70.29 MHz  (68.8 %), step = -30.76 MHz   (6.
    hbnoise_mod2: freq = -125 MHz      (75 %), step = -54.71 MHz   (6.
    hbnoise_mod2: freq = -222.3 MHz  (81.2 %), step = -97.28 MHz   (6.
    hbnoise_mod2: freq = -395.3 MHz  (87.5 %), step = -173 MHz     (6.
    hbnoise_mod2: freq = -702.9 MHz  (93.8 %), step = -307.6 MHz   (6.
    hbnoise_mod2: freq = -1.25 GHz    (100 %), step = -547.1 MHz   (6.
```

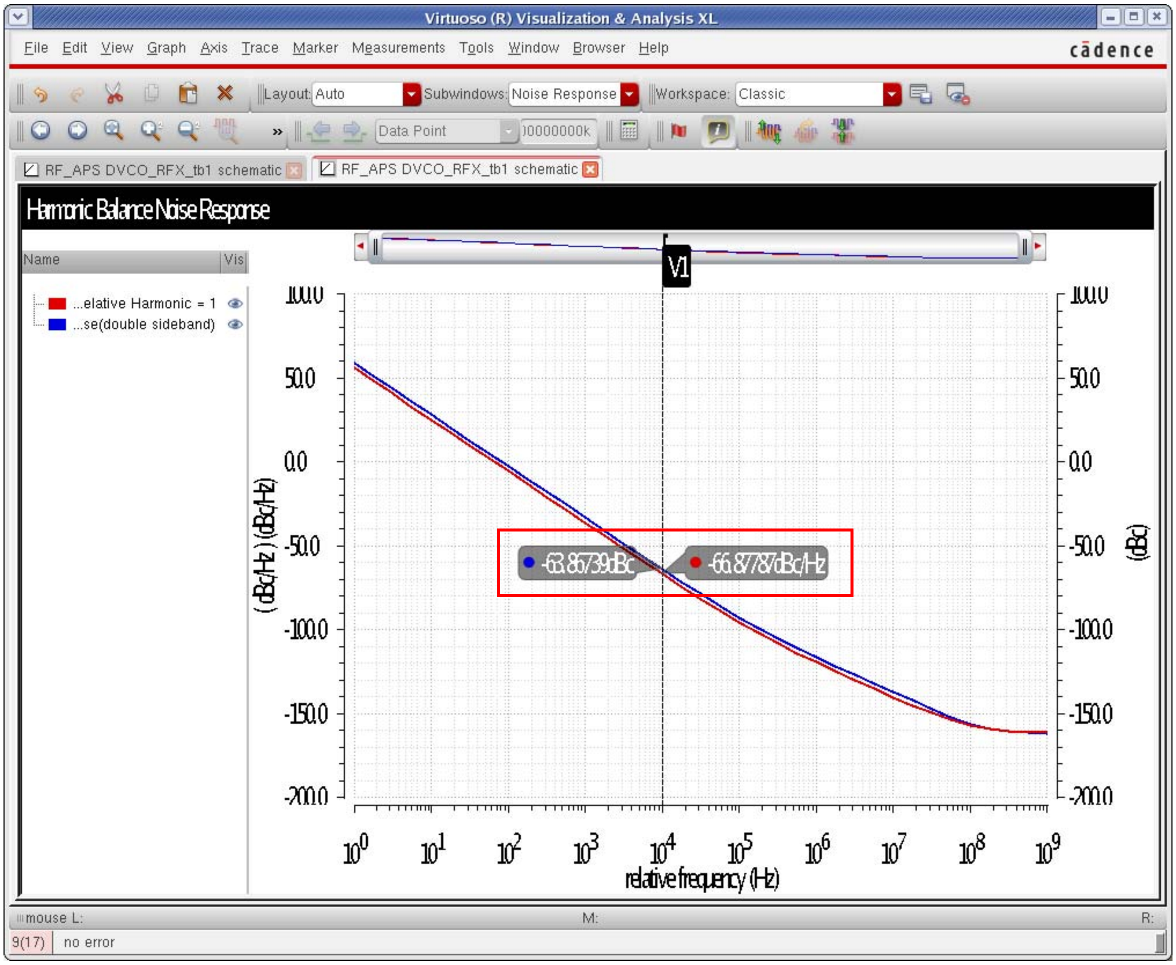
4. To plot the phase noise, in the ADE window, select *Results - Direct Plot - Main Form*.
5. On the *Direct Plot Form,* plot the phase noise from the *hbnoise* results. Note that hbnoise results are always single-sideband measurements. Only the frequencies above the oscillator output frequency are considered for all the available plots in the hbnoise result set.



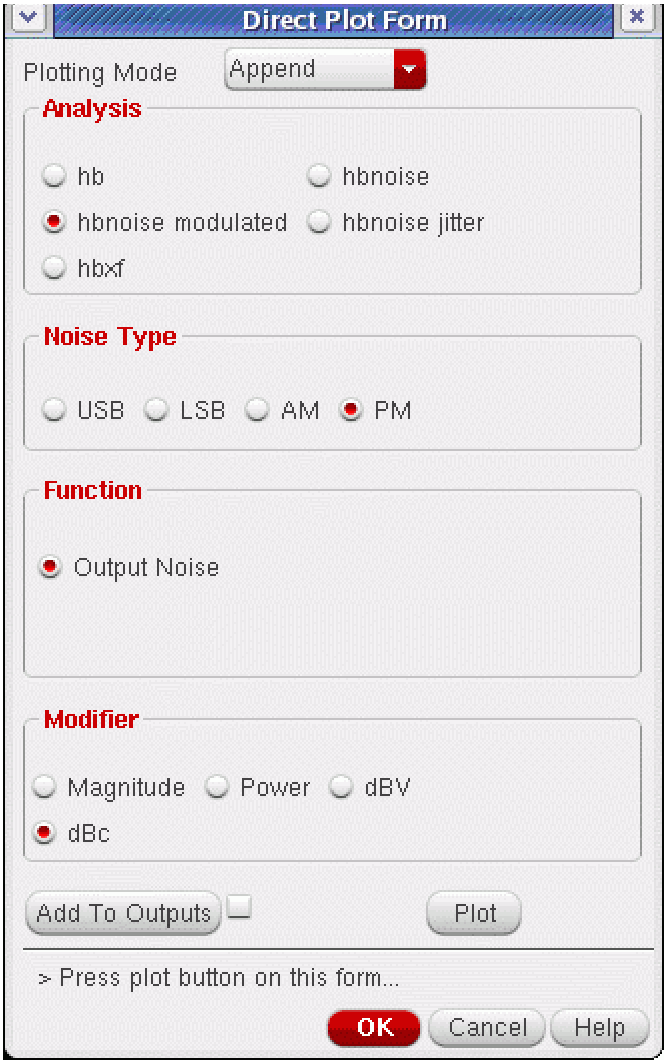
6. Next, plot the phase noise from the *hbnoise jitter* result set. All the plottable items in the hbnoise jitter result set are always double-sideband measurements.



Note that the phase noise from the pnoise result contains the noise in the frequencies above the oscillator frequency only, and the phase noise from the hbnoise jitter results contains the noise power from above and below the oscillator frequency. This is the reason for the 3.01 dB difference.



7. Select *hbnoise modulated* from the *Function* section.

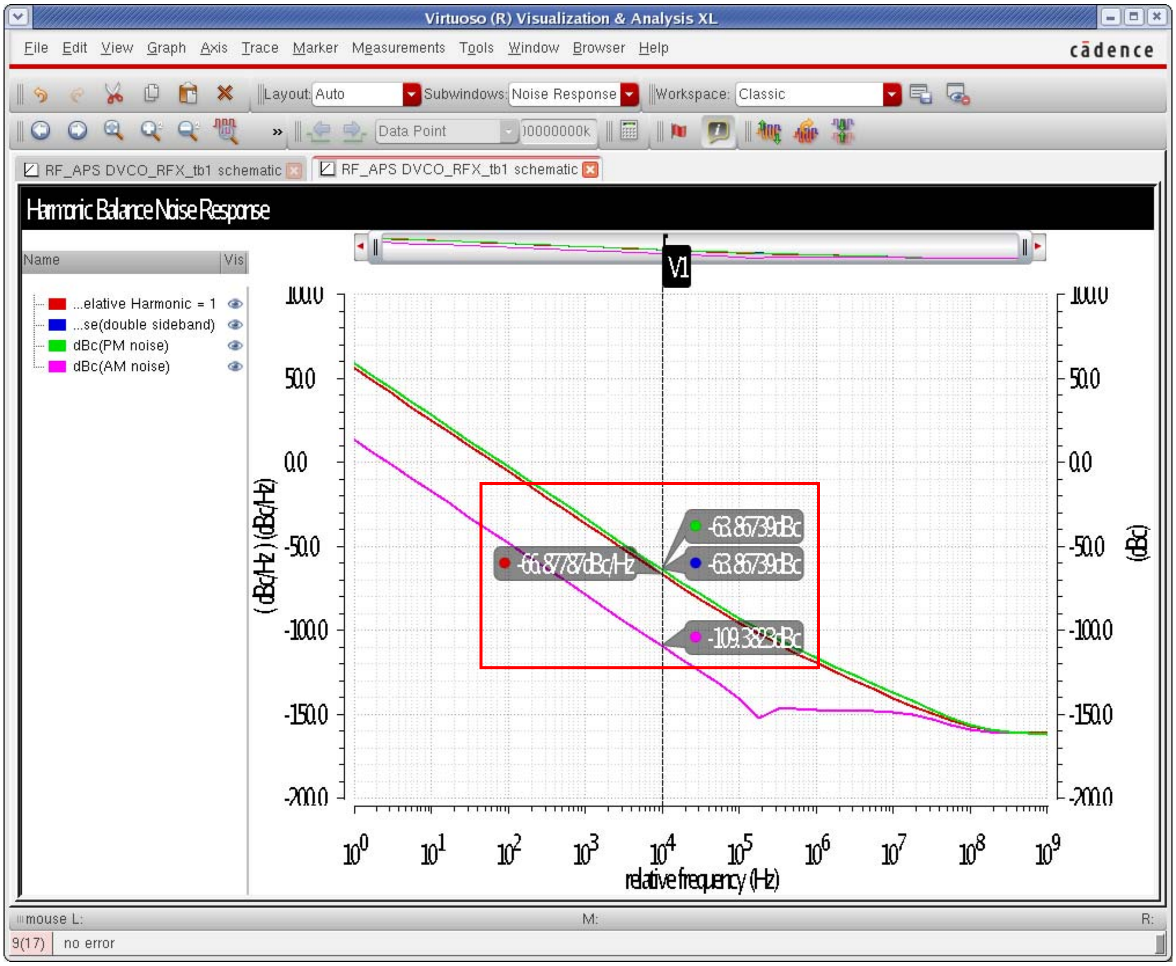


8. Select *AM* or *PM* from the *Noise Type* section.

   Note that the AM and PM noise are also double sideband measurements.

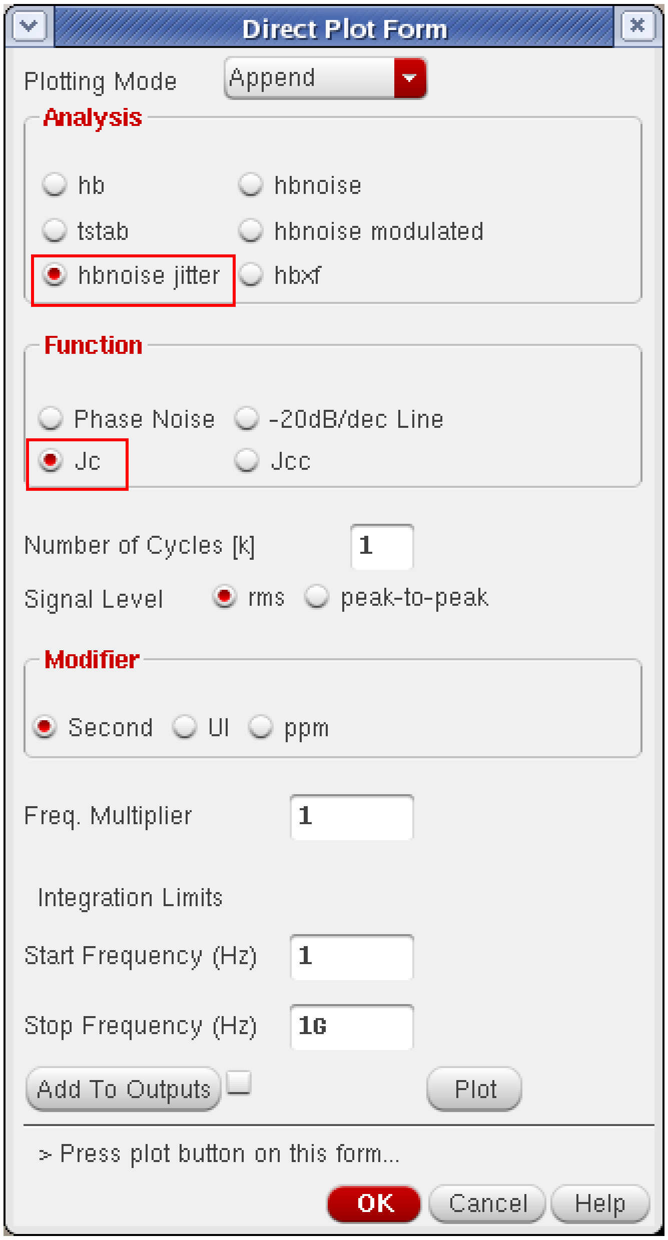
9. Click *Plot*. Both the AM and PM plots have been added to the phase noise plots from before in the plot below. The PM component and the dsb phase noise are almost

identical. This shows that the AM and PM components are double-sideband measurements.



To plot cycle jitter or cycle-to-cycle jitter:

1. On the *Direct Plot Form*, select *hbnoise jitter* from the *Analysis* section.



2. Select *Jc* or *Jcc* from the *Function* section.

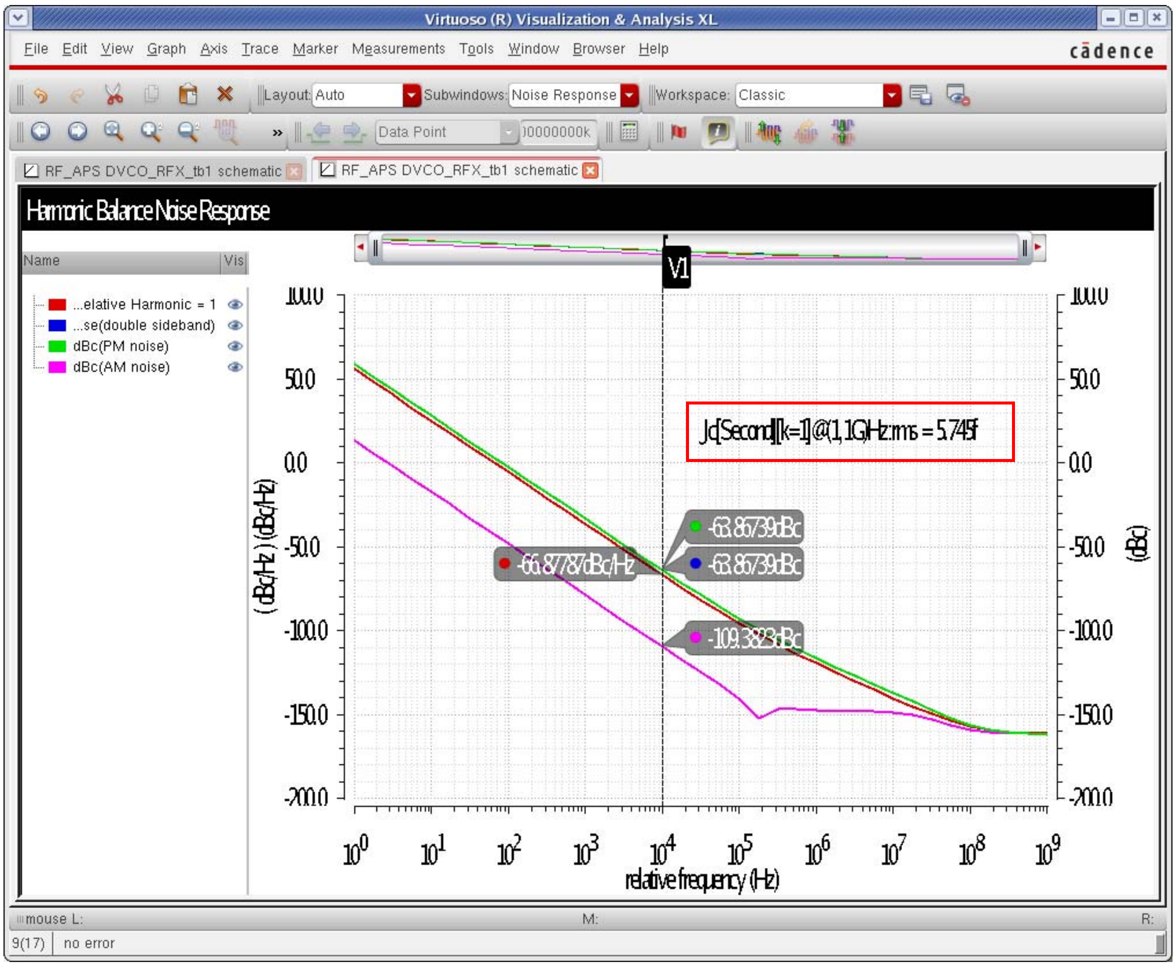
3. The frequency range will default to the range set in the *Choose Analyses* form.

   Cycle jitter (sometimes called period jitter) and cycle-to-cycle jitter are calculated by integrating the noise in the phase noise curve. Cycle jitter fixes the first threshold in time, and measures one standard deviation of the variation of the crossing of the second edge.

Cycle-to-cycle jitter measures one standard deviation of the variation in the period of the waveform.

4. Click *Plot*.

   The Jc or Jcc calculation is shown as a label that is added to the waveform tool.



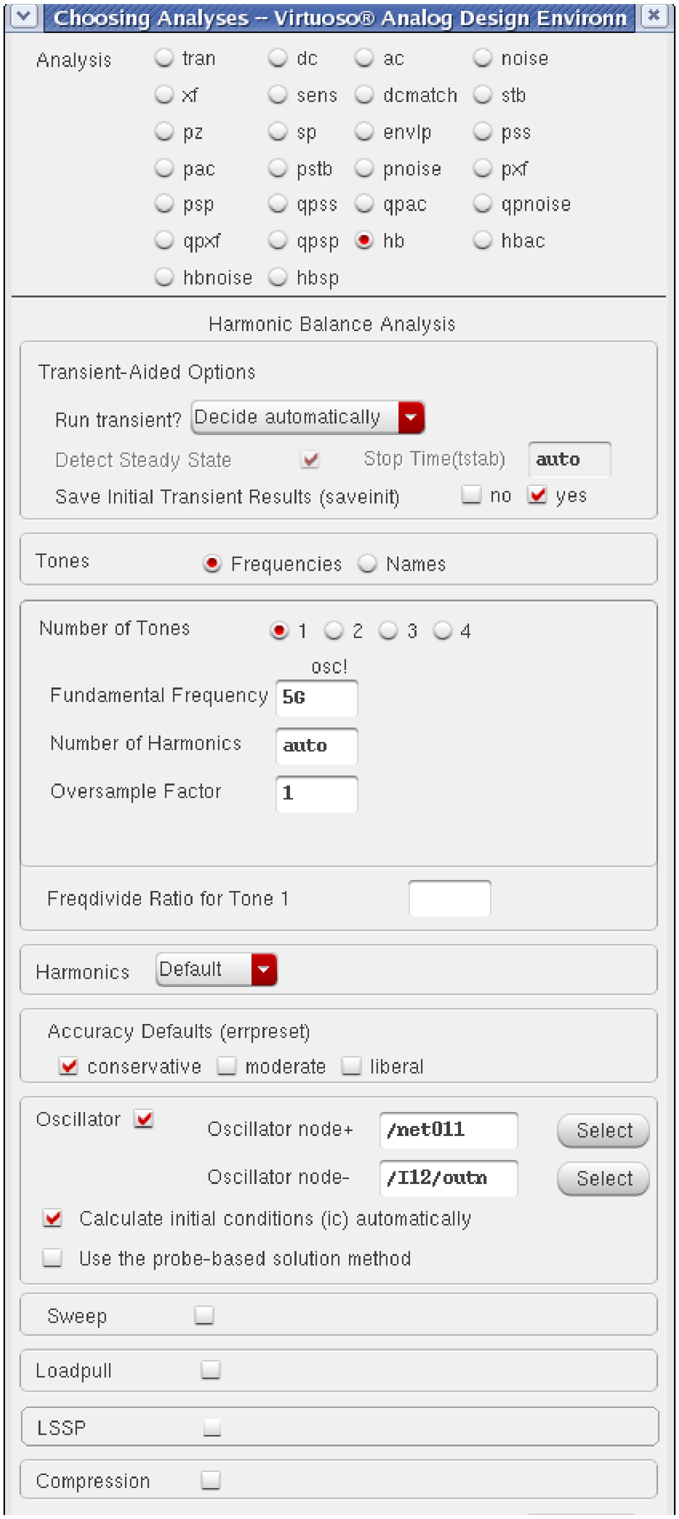
Noise Type = Jitter (FM)

Noise Type = Jitter (FM) produces the same output as Noise Type = Modulated. See Noise Type = Modulated on page 498.

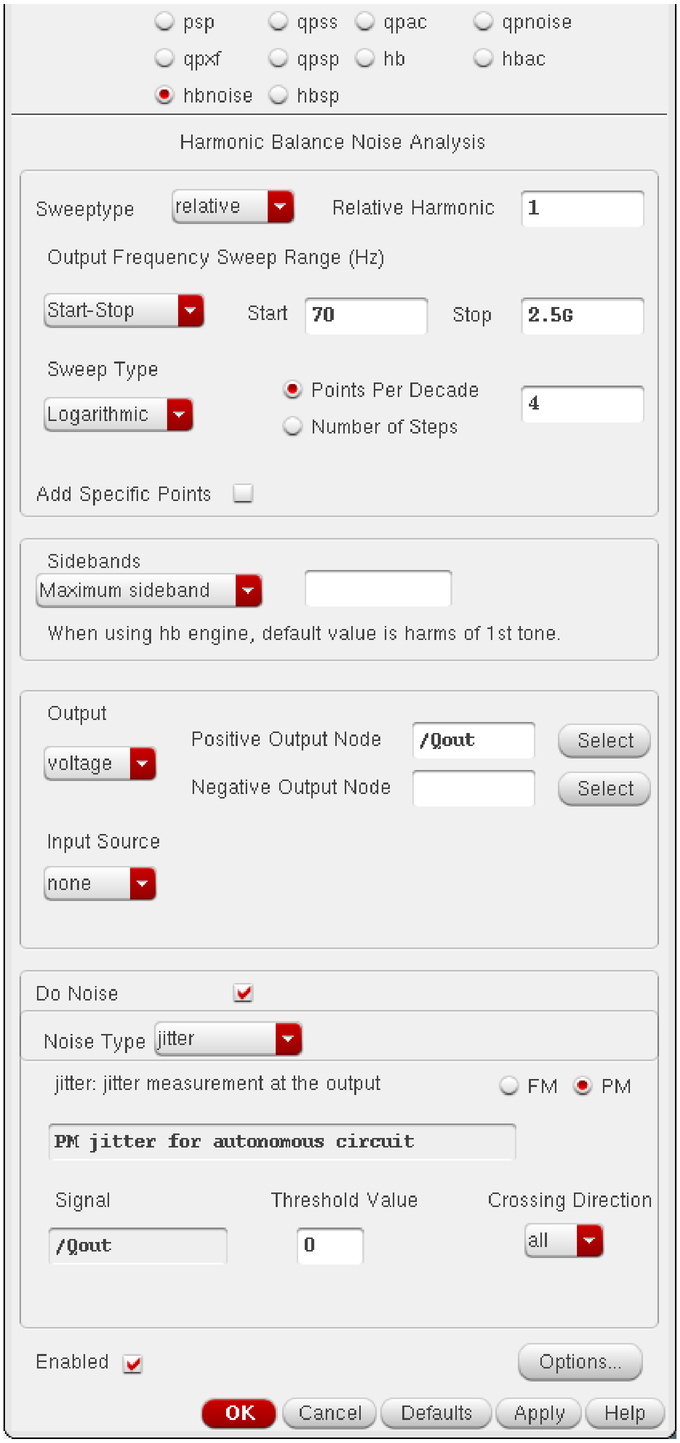
Noise Type = Jitter (PM)

This analysis gives an instantaneous measurement of jitter at a threshold voltage on the timedomain waveform. This is useful for measuring the jitter of an oscillator that is driving a digital component. This is the oscillator equivalent of *noisetype=jitter* for driven circuits. Both of these are variations of *noisetype=timedomain*. Timedomain noise calculates just the instantaneous noise. FM Jitter also calculates the noise, however, it also determines the slew rate of the output, and does the noise divided by the slew rate calculation to calculate the actual time-domain jitter.

1. First, set up the hb analysis. For more information on this, see the harmonic balance section in this chapter.



2. Next set up the hbnoise *Choosing Analyses* form.



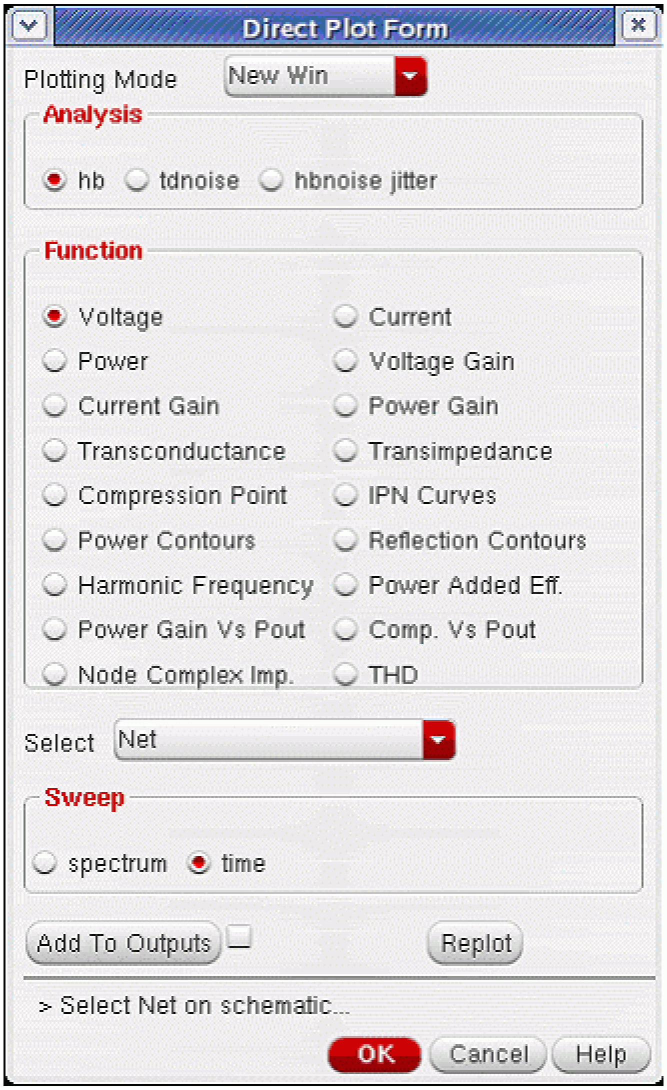
In the *Choosing Analyses* form:

❑ Select *jitter* from the *Noise Type* drop-down list.

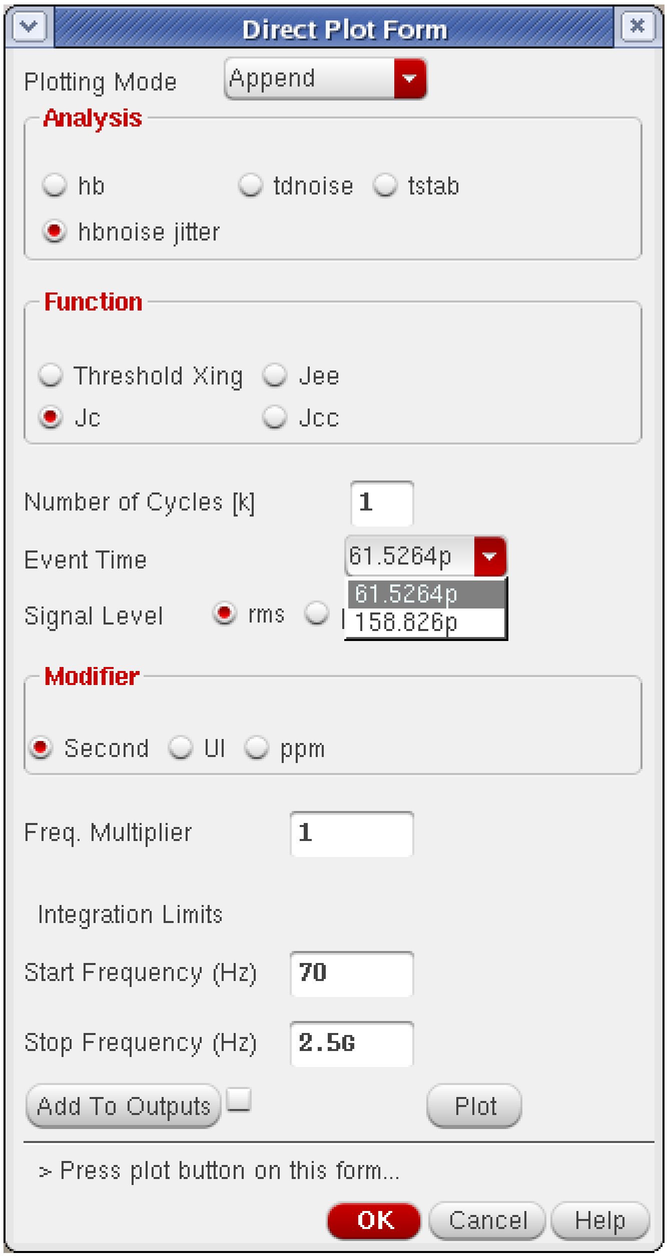
- Select the *PM* radio button.
- Specify a threshold voltage for the time-domain waveform in the *Threshold Value* field.
- Specify a crossing direction. In this case, *all* is selected so that all the threshold crossings are calculated.
- The stop frequency should be very close to half the oscillator frequency.
- The start frequency should be approximately the frequency where the phase noise reaches 0 dBc.

3. Run the simulation.
4. In the ADE window, select *Results - Direct Plot - Main Form*.

On the *Direct Plot Form*, first plot the timedomain result.

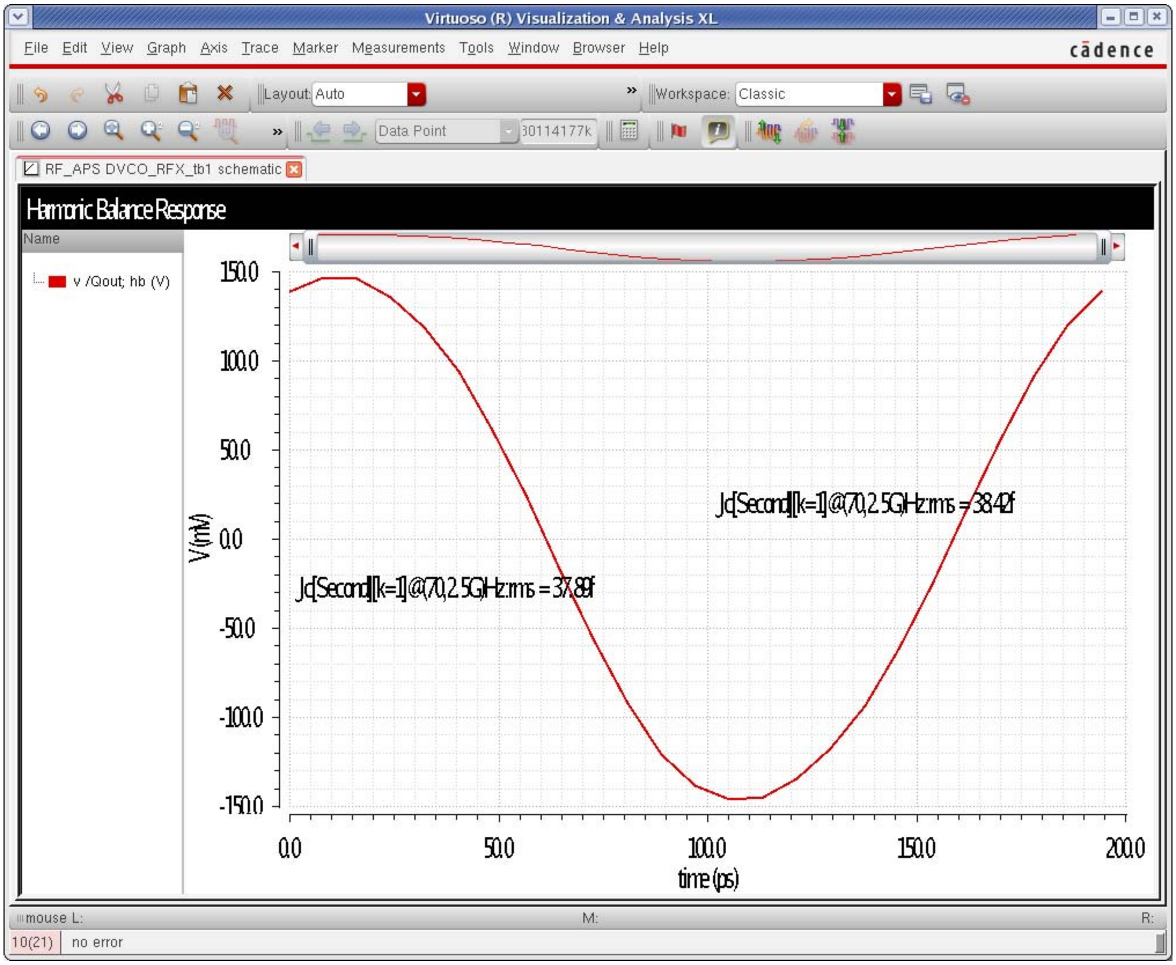


Now plot the cycle jitter (Jc) or the cycle-to-cycle jitter.



Note that there is a selection of event times if you selected *all* for *Crossing Direction*. This is the time on the waveform where the calculation was made. Note that this will not plot unless there is at least one waveform displayed in the waveform tool.

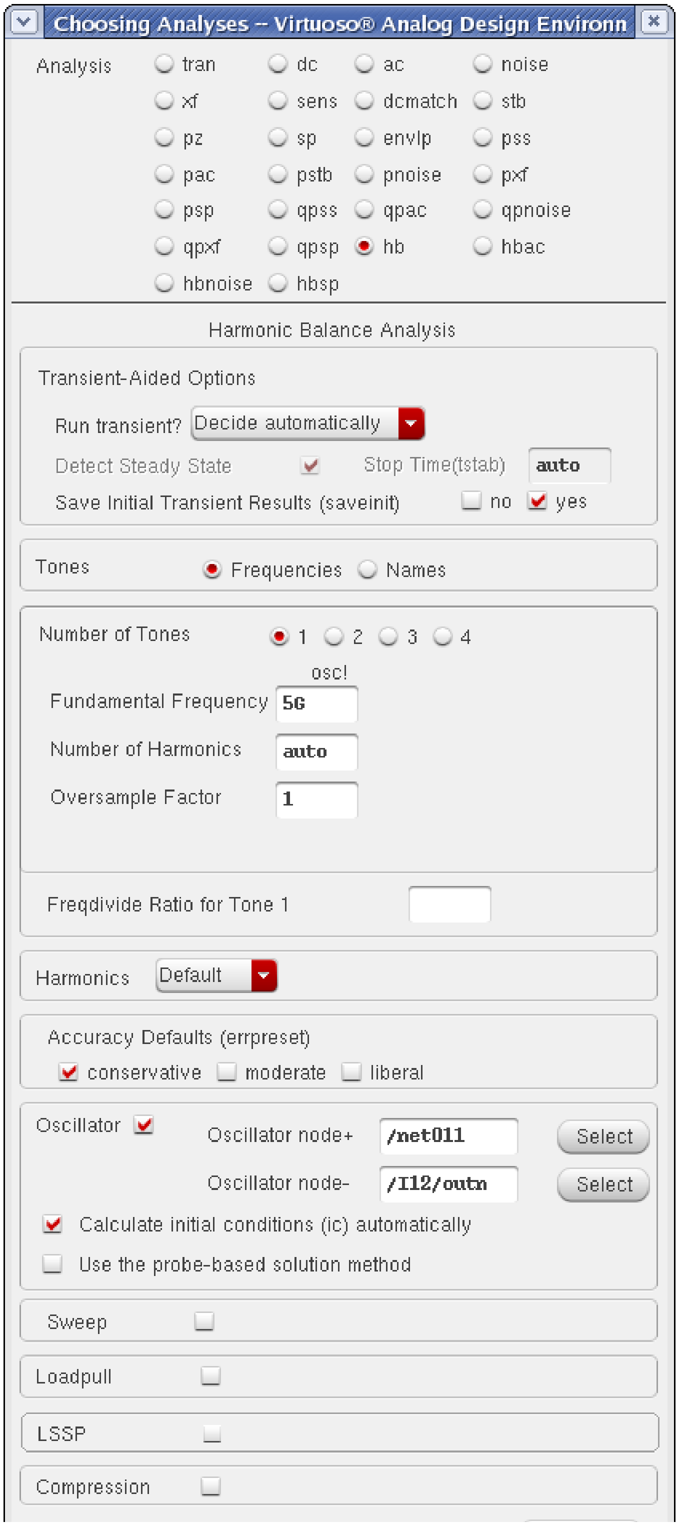
The figure below shows the time-domain waveform, and the jitter calculations for the falling edge and the rising edge.



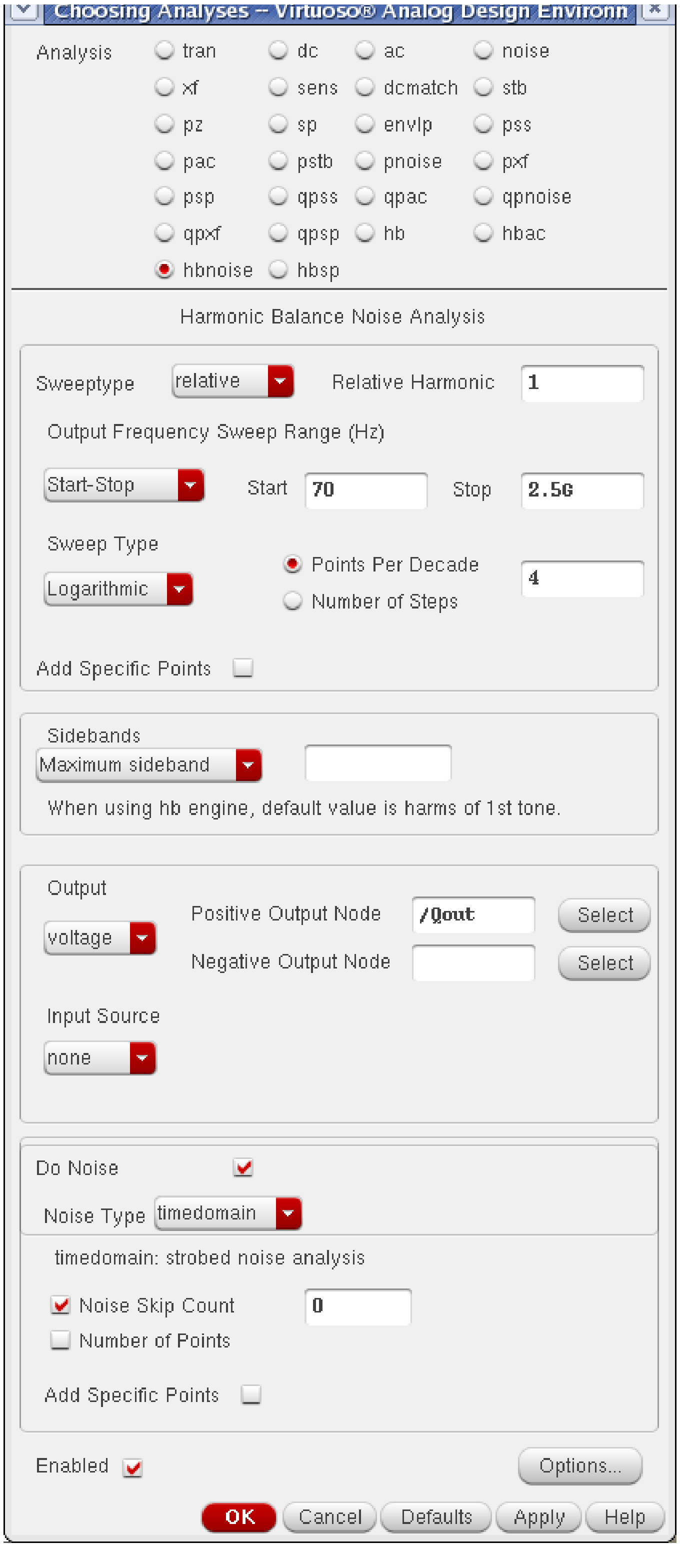
Noise Type = Timedomain

This analysis type provides a relative measurement of the noise at each time in the ifft. Note that if you have a lot of harmonics and/or a large oversample factor, there will be many timepoints in the ifft. A method will be presented to reduce the number of hbnoise points.

1. First, set up the hb analysis for the oscillator. For more information, see the harmonic balance section at the beginning of this chapter.



2. Next, set up the hbnoise analysis.



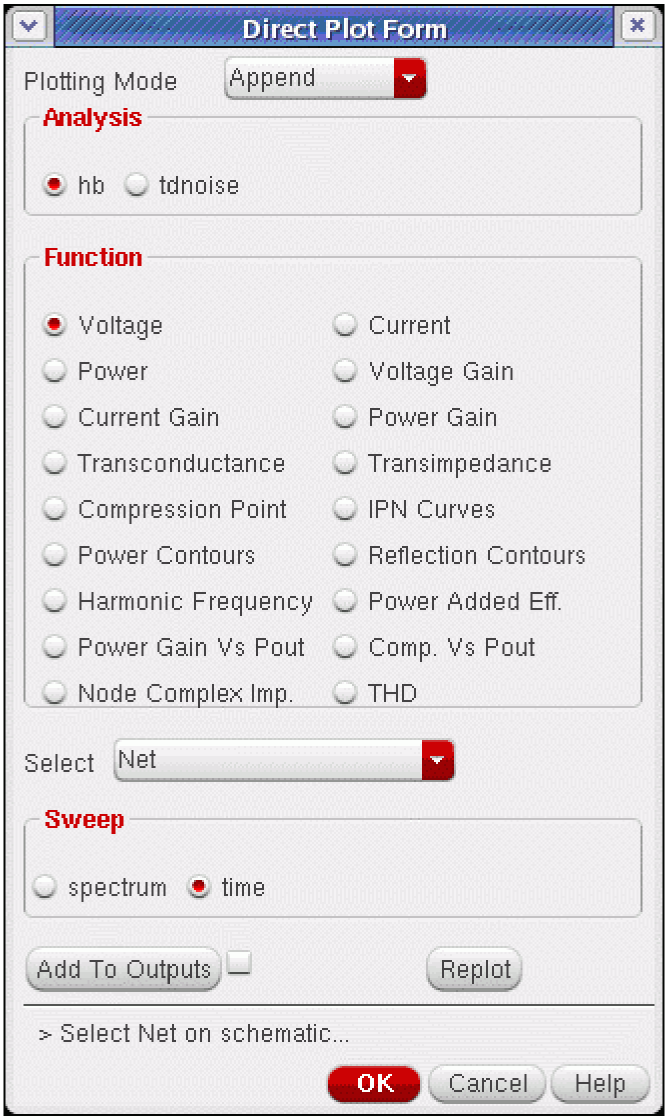
In the hbnoise *Choosing Analyses* form:

- ❑ The stop frequency should be near one half the frequency of the oscillator.
- ❑ The start frequency should be near the point where the phase noise curve reads 0 dBc.
- ❑ Select *timedomain* from the *Noise Type* drop-down list.
- ❑ The default *Noise Skip Count* is 0 (zero).

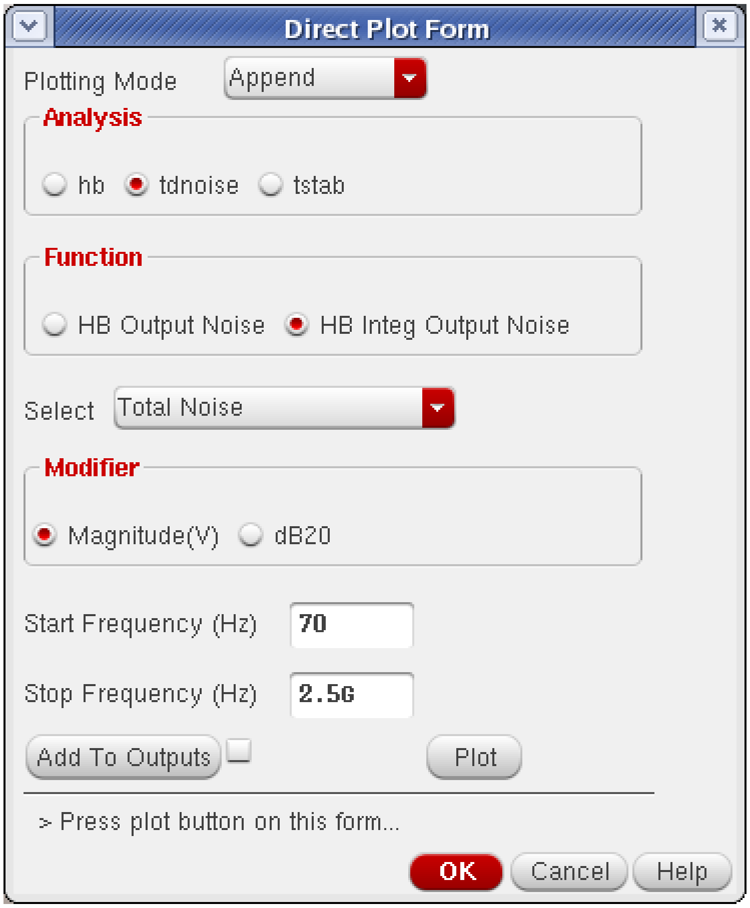
  An hbnoise run will always be run at time = 0. A noise skip count of 2 means run the timepoint at 0 seconds, then skip 2 ifft points, then run the next one, then skip 2, and so on.

3. Run the simulation.
4. In the ADE window, select *Results - Direct Plot - Main Form*.

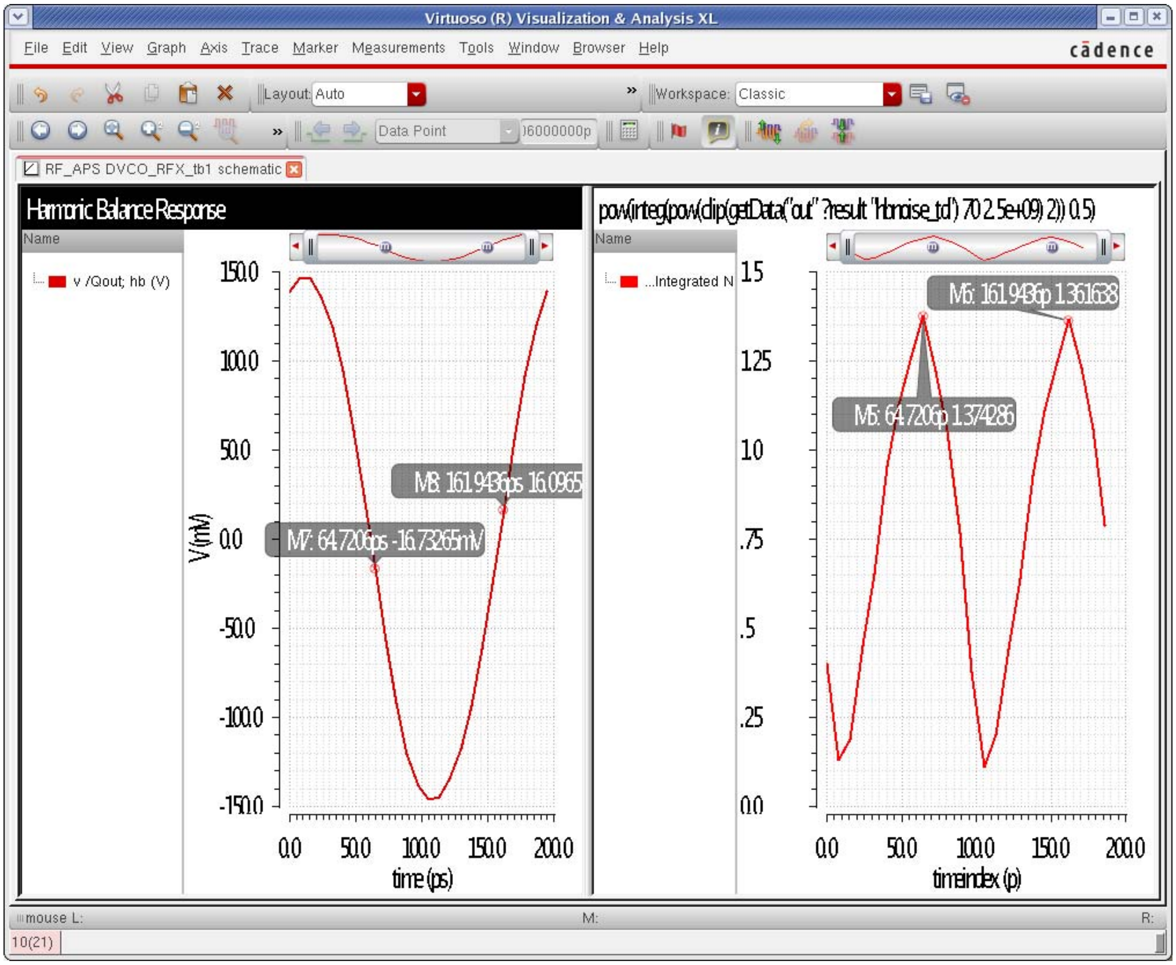
5. Plot the output time-domain waveform (the ifft).



6. Now plot the integrated output noise.



The waveform tool displays both results. Markers have been placed at the largest noise points, and at the times for those maximums.



The noise is highest near the zero crossing, and lowest at the peaks and valleys of the waveform.

hbSP Analysis

Overview of hbsp

Hbsp is an extension to the idea of linear S-Parameters, so that S-Parameters can be measured for circuits that translate frequency. In linear S-Parameters, a wave is launched from the input and that wave is usually partially reflected at the circuit connection, and it propagates to the output where it can be measured. Then a wave is launched at the output source, some of it is reflected back to the output, and a wave appears at the input. Using this idea, linear S-Parameters can be used effectively to characterize many linear systems.

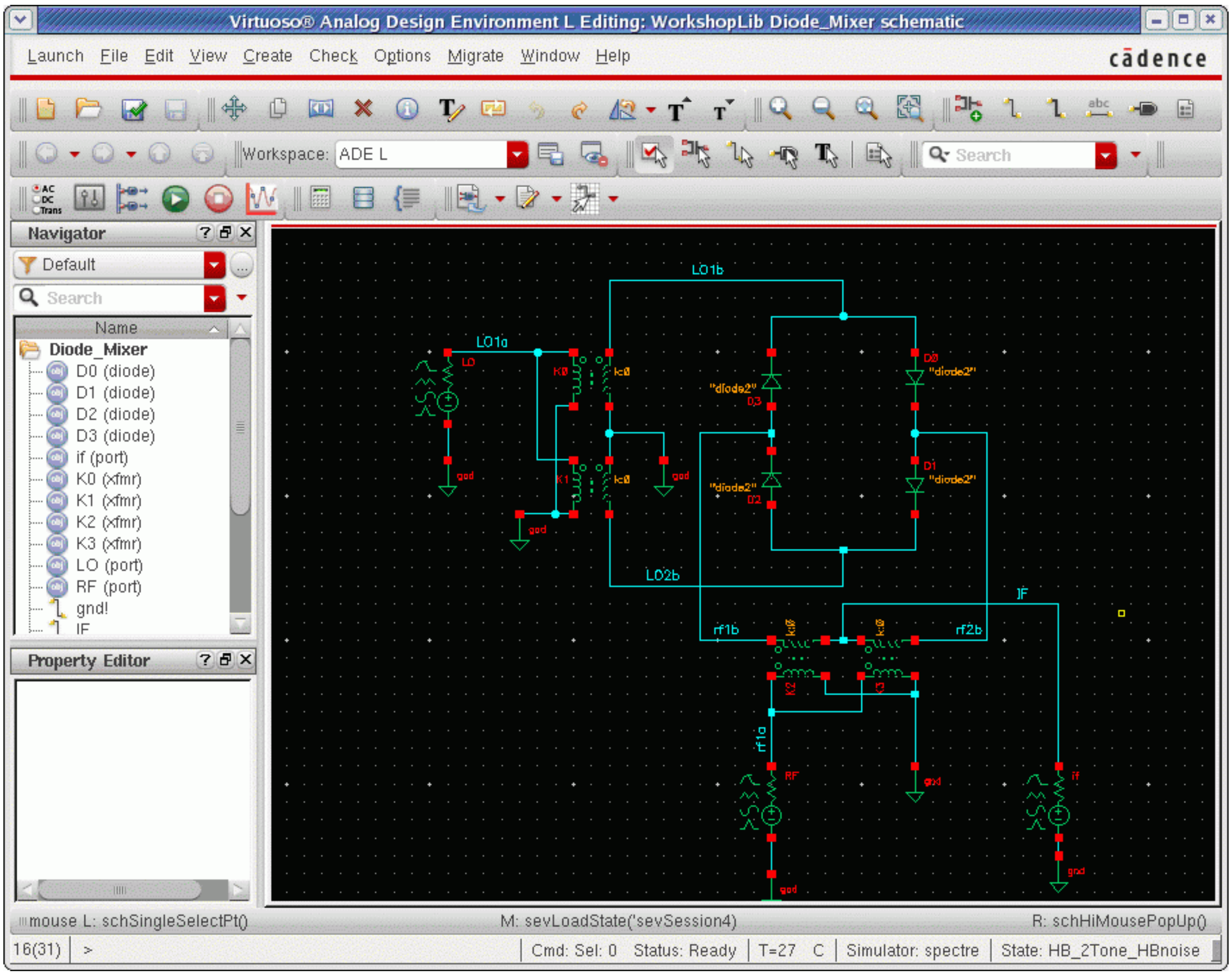
With hbsp, a wave is launched at the input just like in S-Parameter analysis. Some of this wave is reflected back to the input, depending on the input match. Because the system is nonlinear, this wave can mix with harmonics of the periodic source that is driving the circuit to cause waves to appear at the output at many different mixing frequencies. Hbsp supports a 2-port simulation where one of the output mixing products can be measured at a time.

As an example, imagine a mixer that has a 1.1GHz RF input, and a 1GHz Local Oscillator signal. The forward wave is launched at 1.1GHz, and at the output of the mixer 1.1GHz is produced with no frequency translation. This measures the RF to IF isolation. 100MHz and 2.1GHz are the mixing products produced by mixing with the first harmonic of the LO signal. For a down conversion mixer, the 100MHz term is desired to measure the conversion gain. One of these waves can be measured in hbsp. Like linear S-Parameters, hbsp measures the power gain of the system.

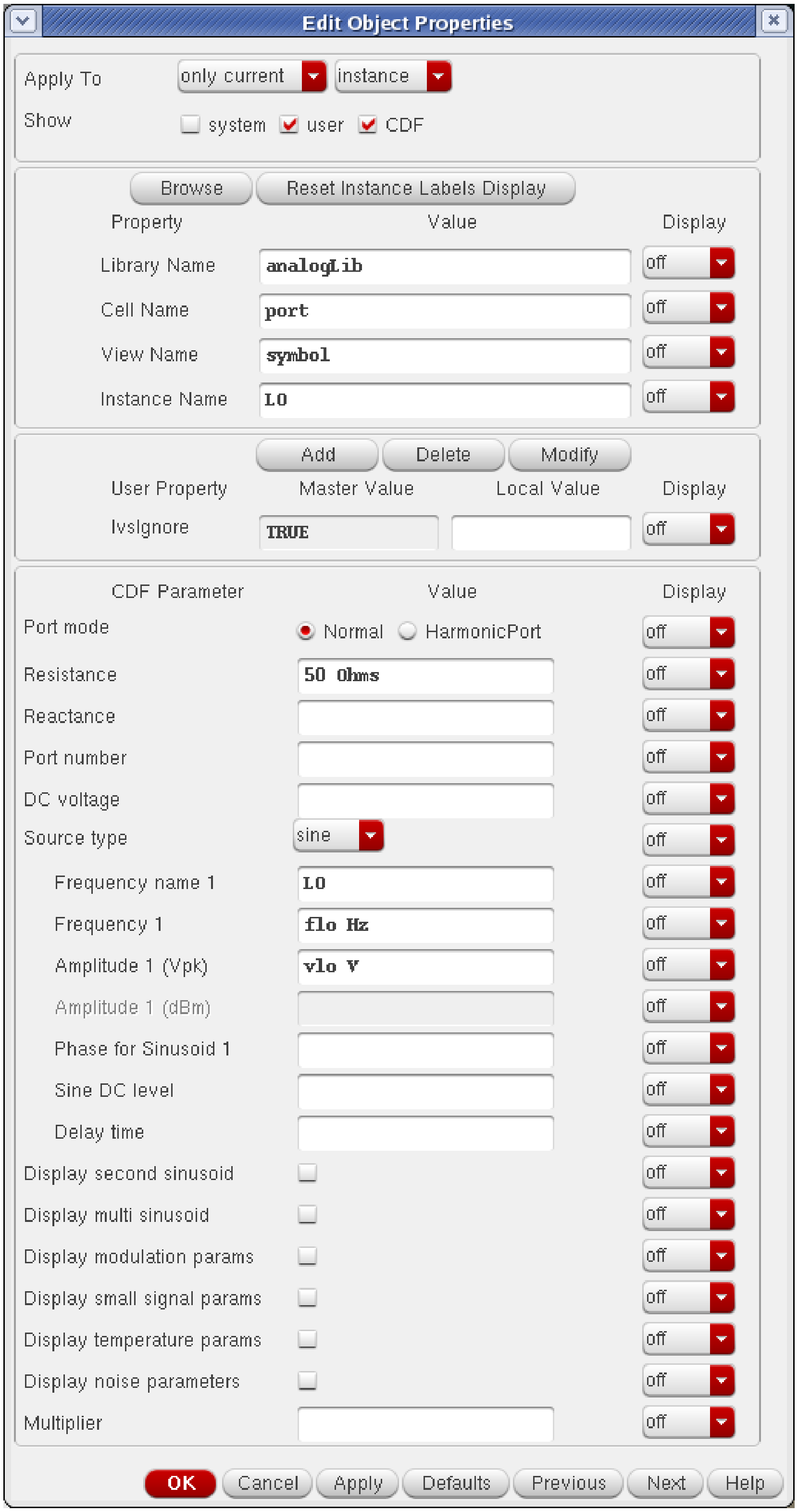
For the reverse gain and match calculation, the input is at the output frequency of the mixer, and the output is at the input frequency of the mixer. The forward gain path is from 1.1GHz to 100MHz, and the reverse gain path is from 100MHz to 1.1GHz. Therefore, S11 is a measure of the input match at the RF frequency, and S22 is the output match at the IF frequency. S21 measures the forward conversion gain, and S12 measures the reverse conversion gain.

Example

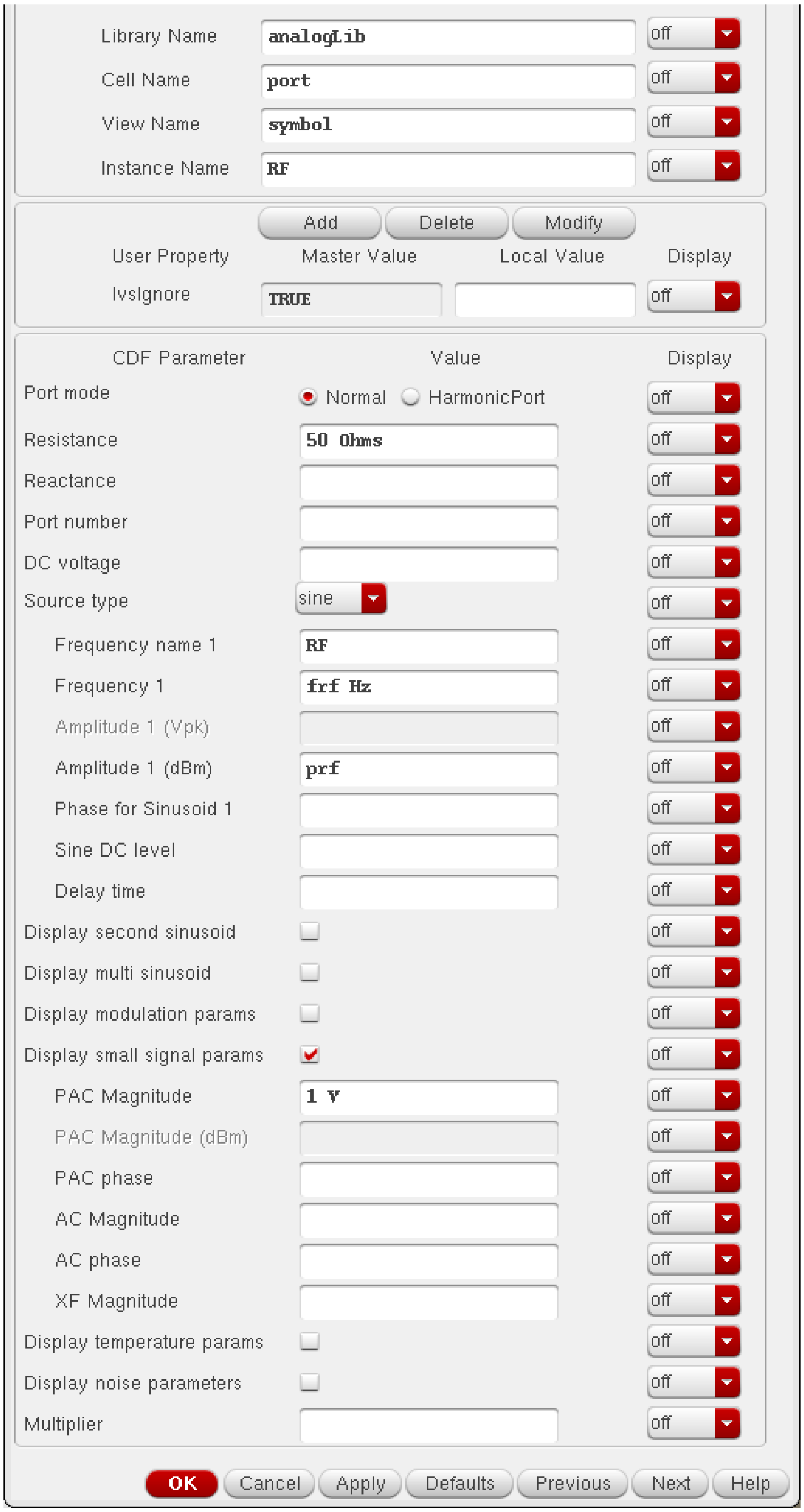
Consider a double-balanced diode mixer, as shown below.



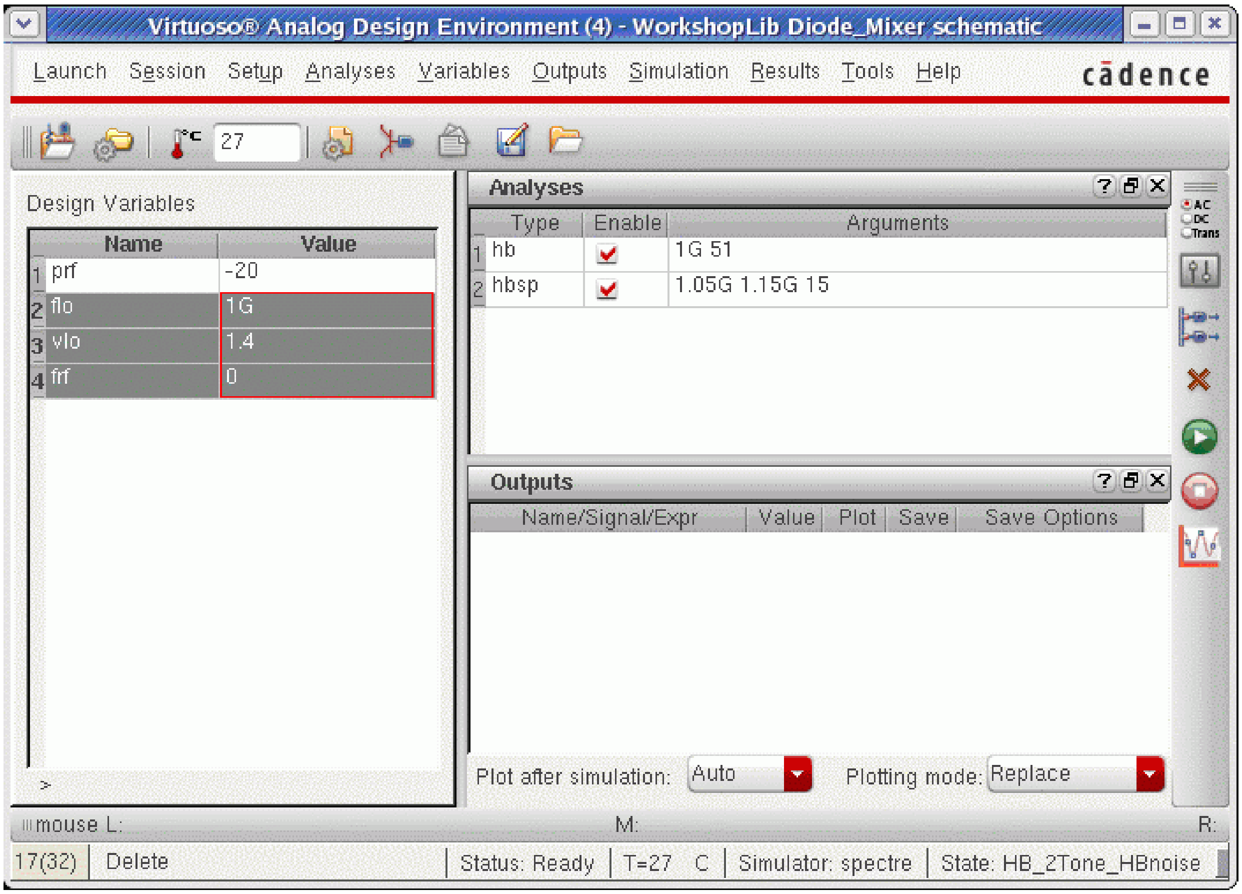
The properties of the LO source are as follows. The frequency and amplitude are set to variable names so they can be changed in the ADE without changing the schematic.


Edit Object Properties
Apply To
only current
instance
Show
system
user
CDF
Browse
Reset Instance Labels Display
Property
Value
Display
Library Name
analogLib
Cell Name
port
View Name
symbol
Instance Name
LO
Add
Delete
Modify
User Property
Master Value
Local Value
lvsIgnore
TRUE
CDF Parameter
Port mode
Normal
HarmonicPort
Resistance
50 Ohms
Reactance
Port number
DC voltage
Source type
sine
Frequency name 1
LO
Frequency 1
flo Hz
Amplitude 1 (Vpk)
vlo V
Amplitude 1 (dBm)
Phase for Sinusoid 1
Sine DC level
Delay time
Display second sinusoid
Display multi sinusoid
Display modulation params
Display small signal params
Display temperature params
Display noise parameters
Multiplier
off
OK
Cancel
Apply
Defaults
Previous
Next
Help

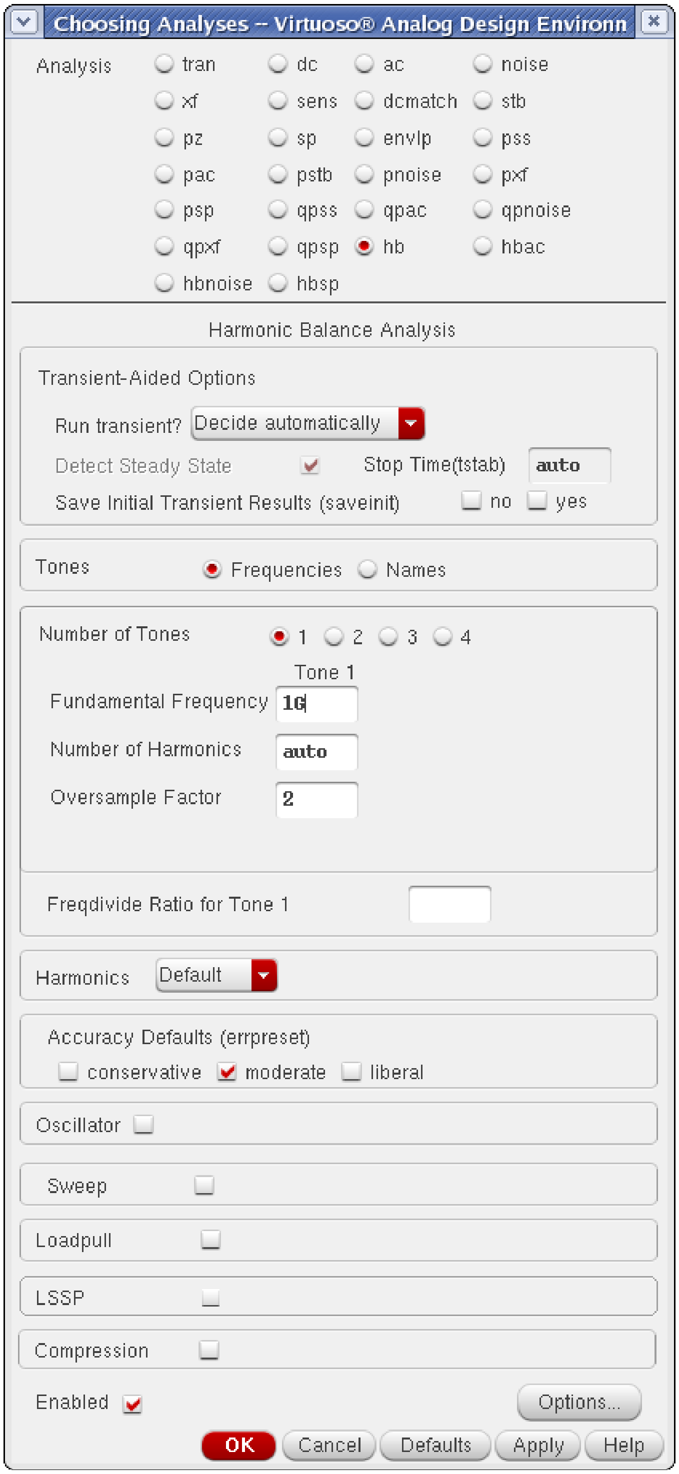
The RF source is similar, but with different variable names.



The variables are set in the ADE. The *frf* variable is set to 0 (zero) in order to disable the large-signal waveform. Only the *LO* is applied at 1GHz. 1.4 volts is +13 dBm.



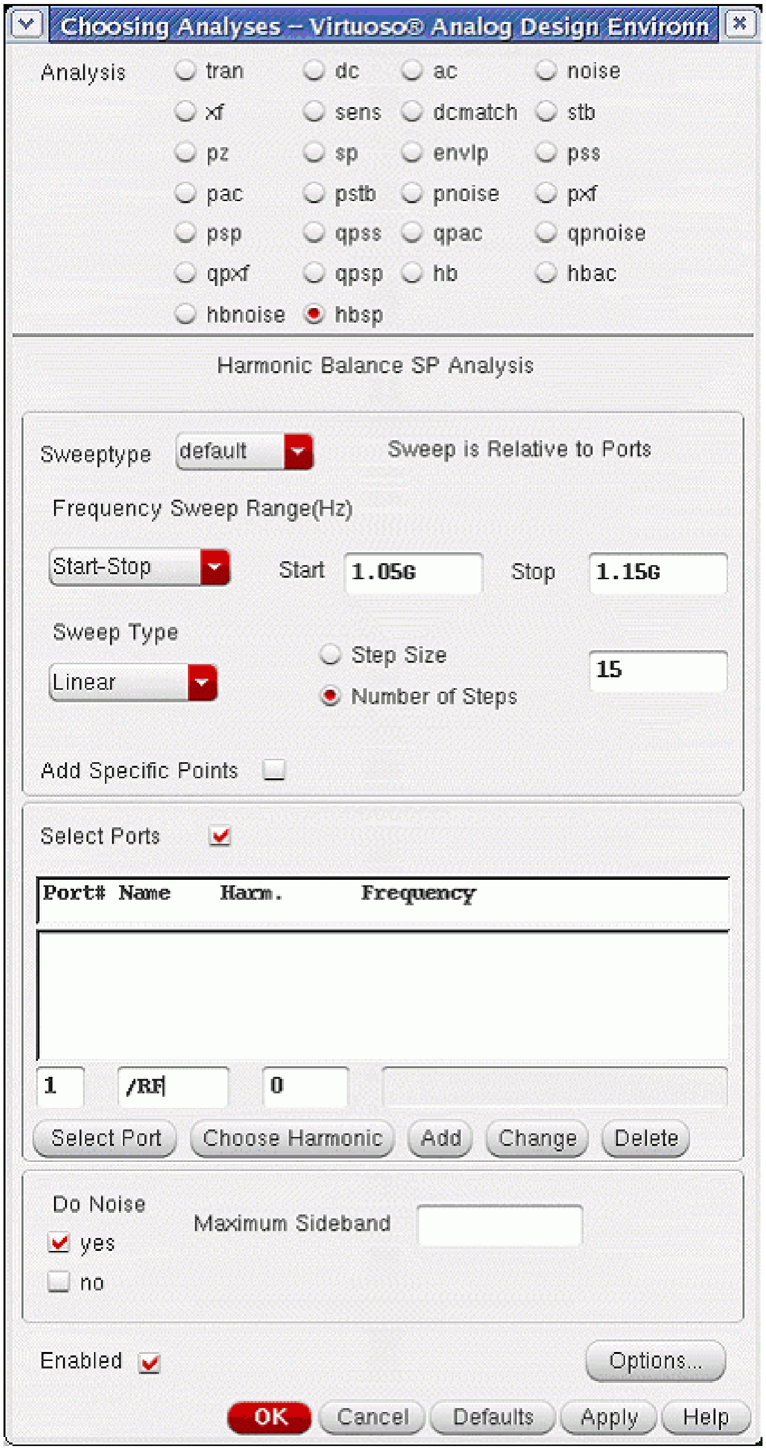
Next, set up the hb *Choosing Analyses* form. The hb analysis is used to characterize the nonlinearity of the system and measures the amplitude of the harmonics.



Harmonics are set to auto so that the simulator will choose the number of harmonics automatically, and oversample factor is raised above one because the currents in the diodes are very nonsinusoidal. For more information on setting the oversample factor, refer to the harmonic balance section at the beginning of this chapter.

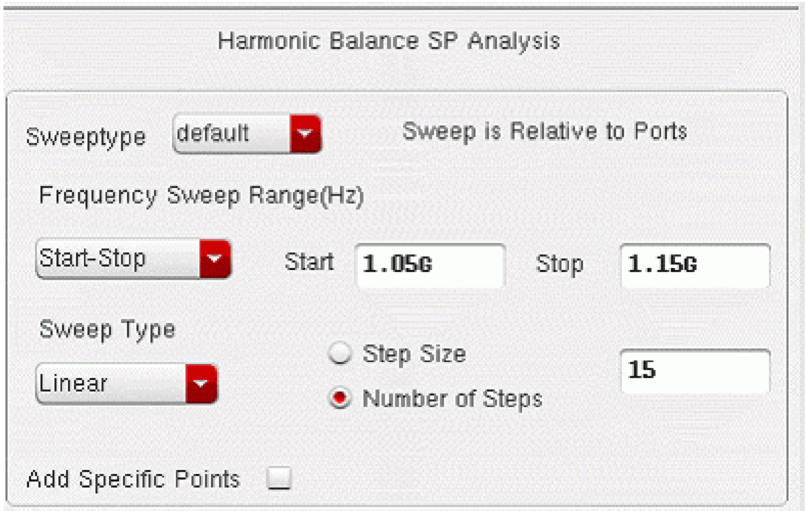
Down Conversion and RF to IF Isolation hbsp Setup

1. Open the hbsp *Choosing Analyses* form from the ADE window.

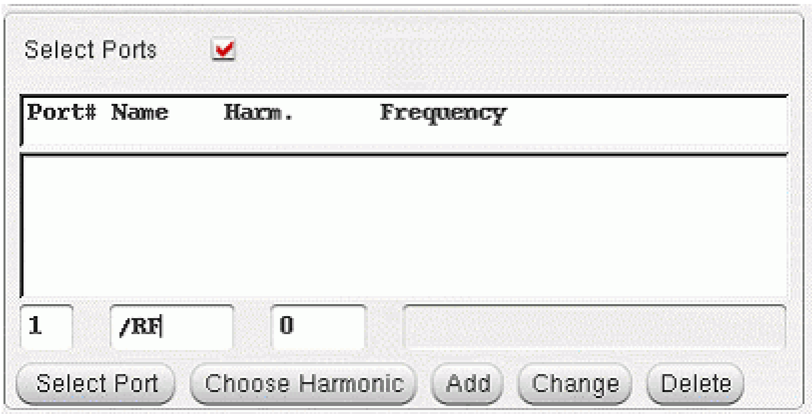


2. Specify the input or output frequency range in the *Start* and *Stop* fields.

   Both up conversion and the down conversion will be shown.The LO frequency is 1GHz as was shown in the harmonic balance *Choosing Analyses* form. In the figure below, the frequency range chosen is 1.05G to 1.15G. For down conversion, this is the input frequency range. The down-converted output is from 50M to 150M.

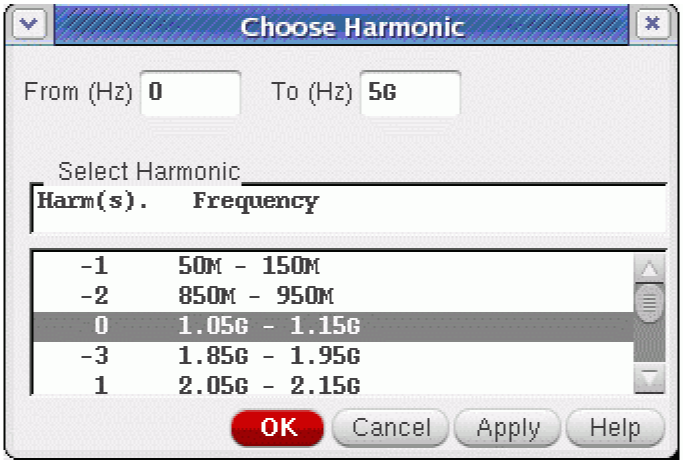


3. Enter 1 in the *Port #* column.



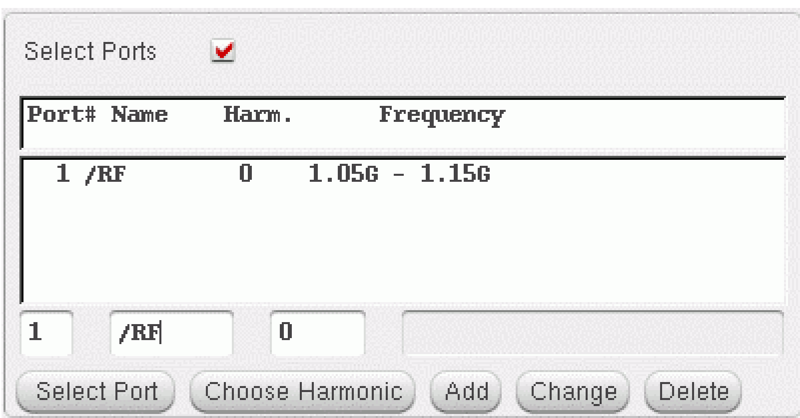
4. Click *Select Port* and select the input port in the circuit.
5. Click *Choose Harmonic*.

The Choose Harmonic window is displayed.

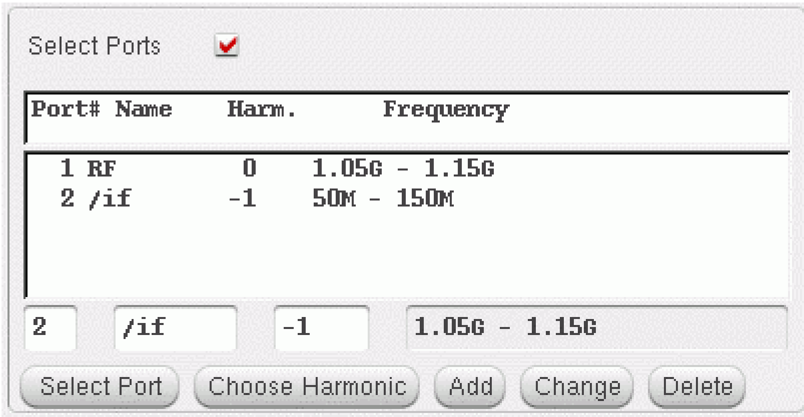


6. In the *Choose Harmonic* window, select the input frequency range.
7. Click *OK*.
8. Click *Add*.

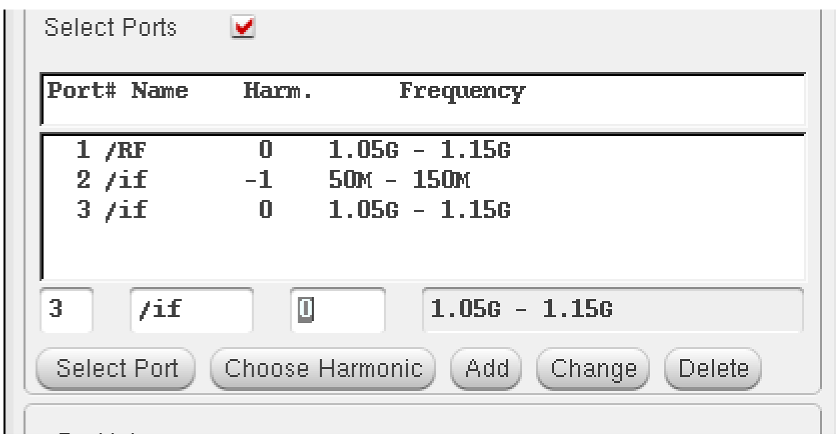
   Port 1 is added to the list with the input frequency range.



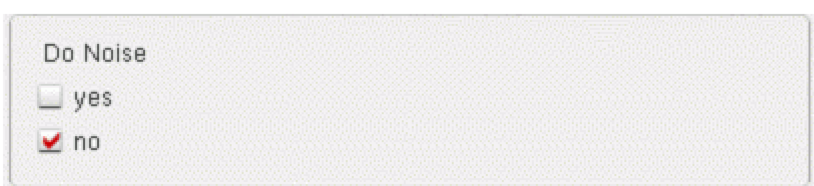
9. Follow the steps from 2 through 8 to add a second IF output port at the downconverted frequency. The second port is added, as shown below.



10. Follow the steps from 2 through 8 to add a third IF output port at the non-converted frequency. This will measure the RF to IF isolation. The third port is added, as shown below.



The noise analysis is the same as the noise analysis in hbnoise. Refer to the hbnoise section for details. In the figure below, noise has been disabled because we will not be performing noise analysis.

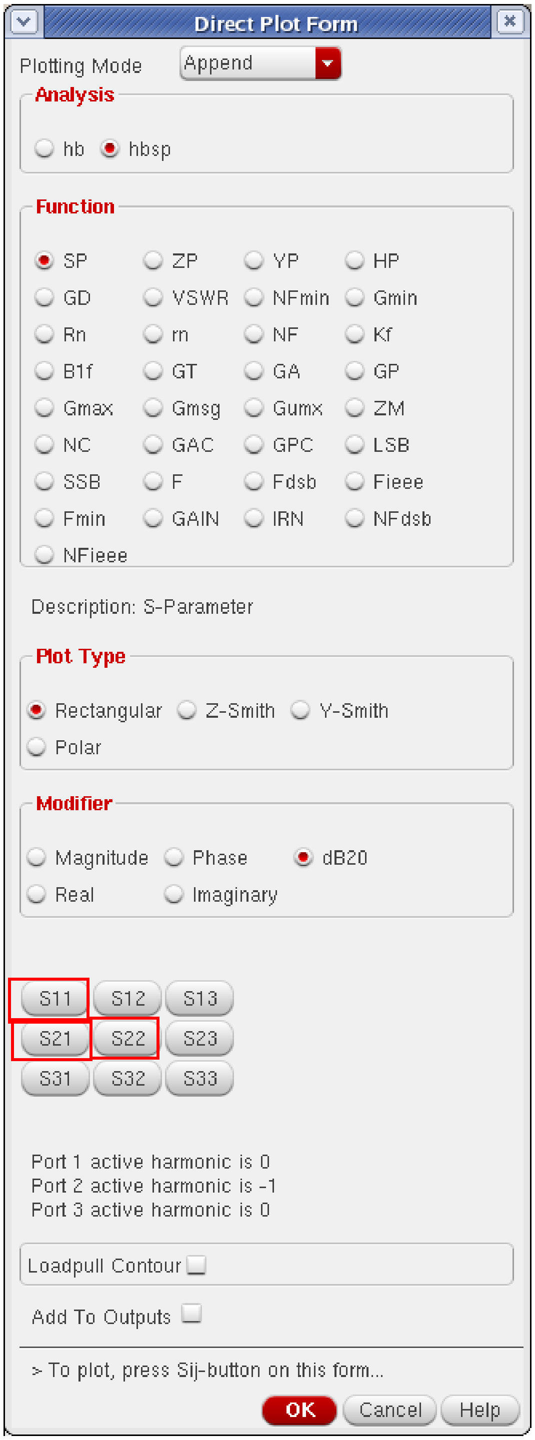


11. Run the simulation.

12. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window.

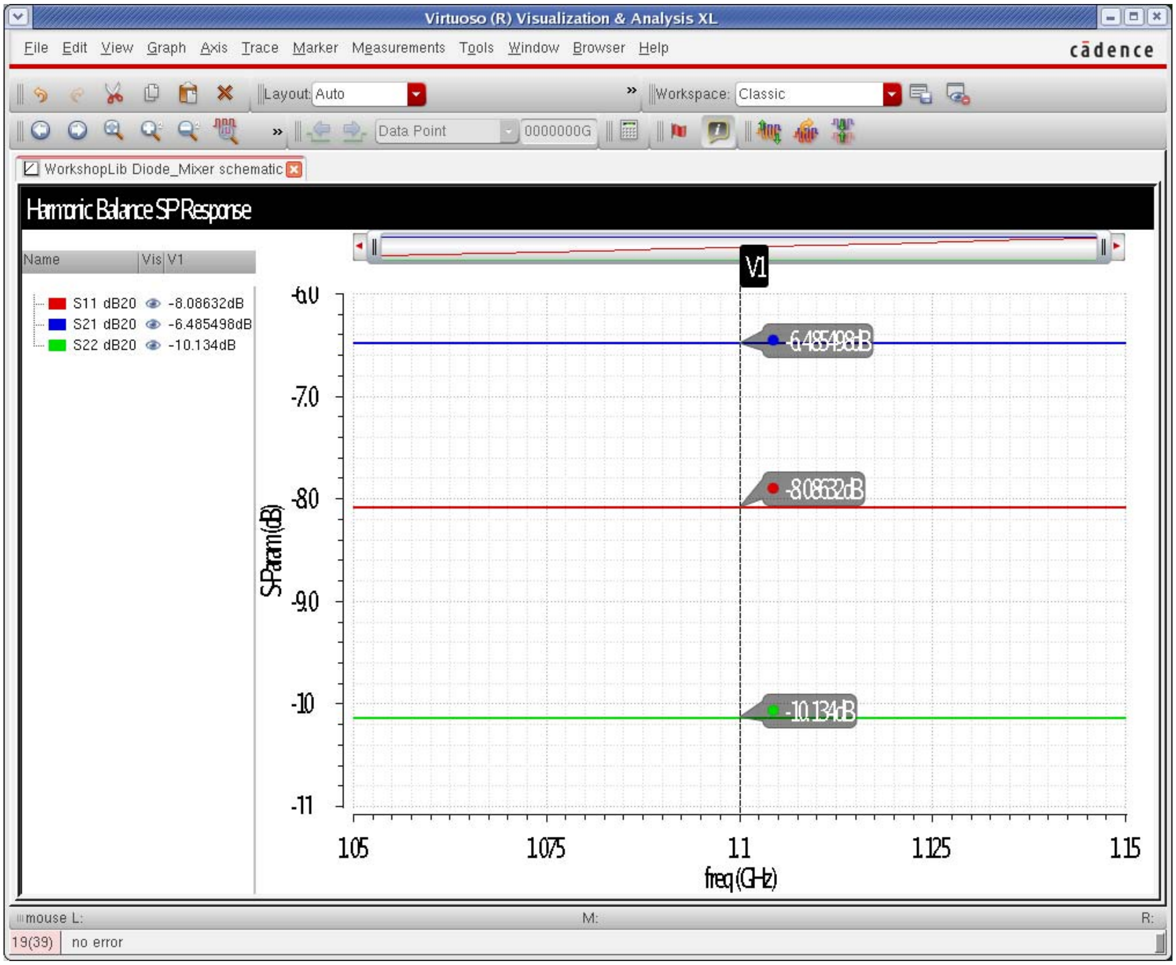
The *Direct Plot Form* is displayed.

13. In the *Direct Plot Form*, select *hbsp* from the *Analysis* section.

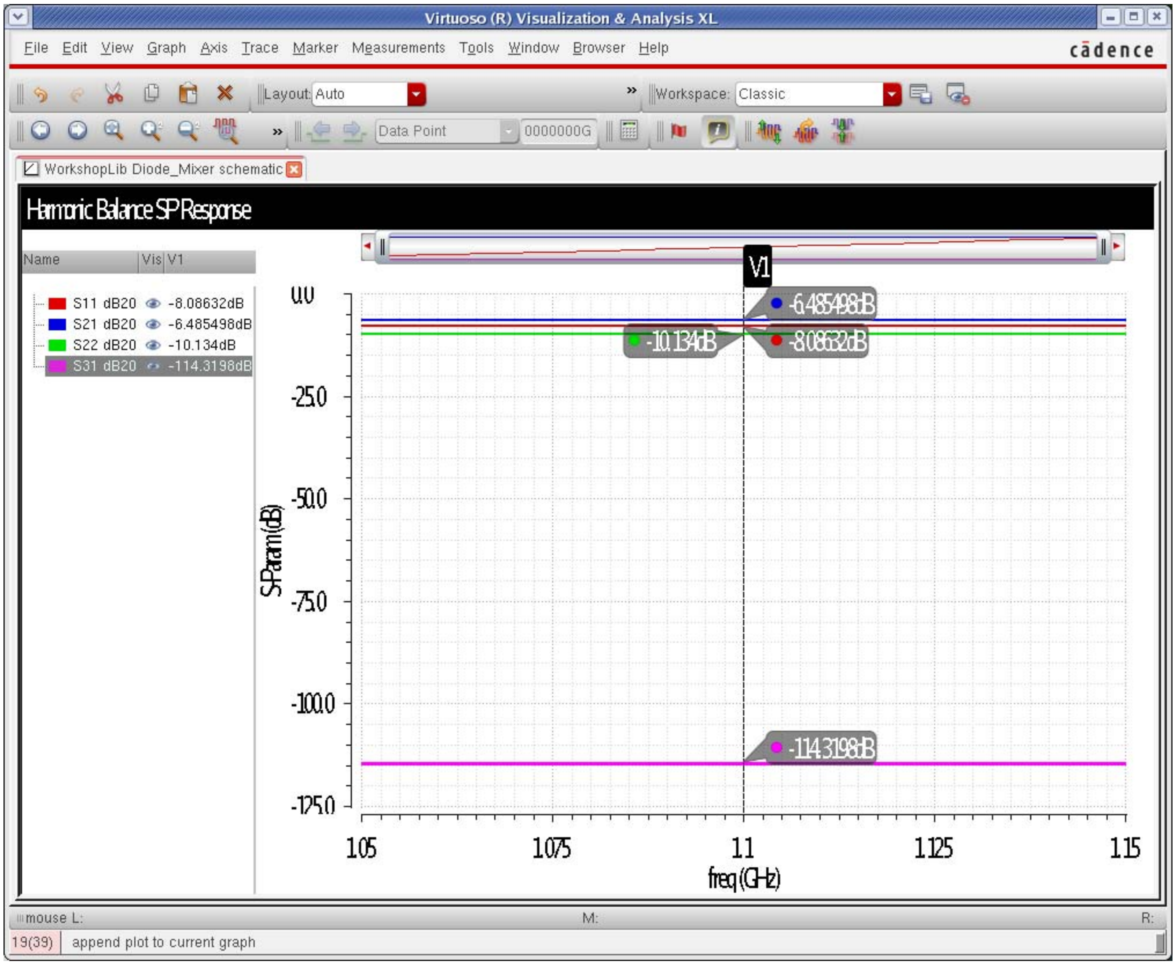


14. Now plot S11, S21, and S22 by clicking the buttons near the bottom of the *Direct Plot Form*.

The waveform window is displayed, as shown below.



Now plot the RF to IF isolation by selecting S31.



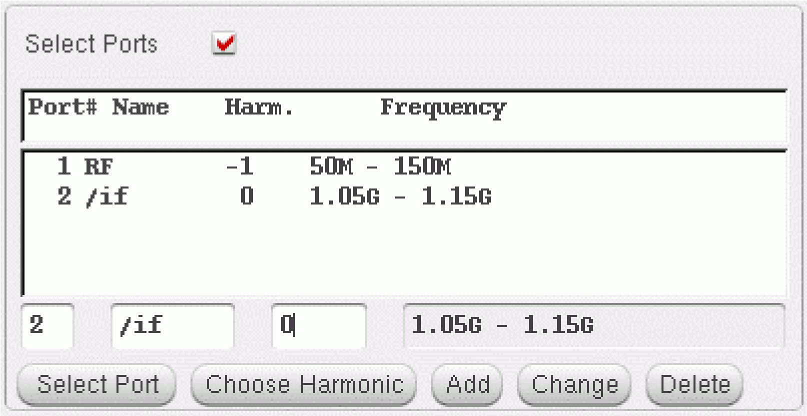
The RF to IF isolation in the ideal case is quite high at just over 114dB.

UP Conversion hbsp Setup

Remember that the LO is at 1GHz as it was before. Now we will reverse the input and output frequencies. The input is from 50M to 150M, and there are two outputs. One output is from 850M to 950M and the other output is from 1.05G to 1.15G. We will focus on the output at 1.p05G to 1.15G in this example.

The frequency sweep range is unchanged at 1.05G to 1.15G.

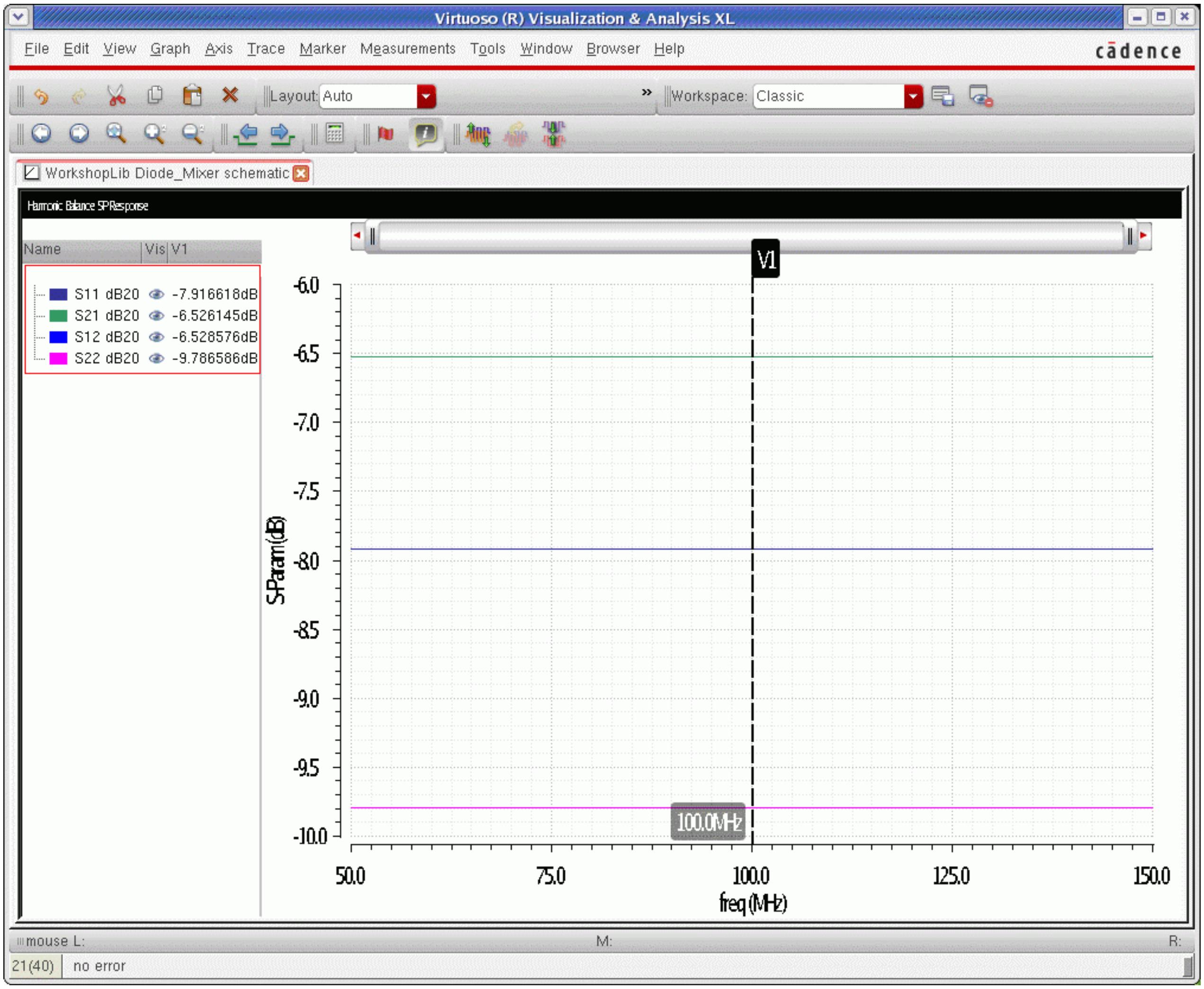
For up-conversion, reset the hbsp *Choosing Analyses* form and follow the same steps to specify the frequency for the two ports, as shown in the following figure.



Note that the input and output frequencies have changed. Now the input is from 50M to 150M, and the output is from 1.05G to 1.15G. This is done using the same *Choose Harmonic* button as before.

Also, since the gain with no frequency translation from RF to IF is not made because measuring it at 100MHz to 100MHz has little value. The 50M to 150M term at the output of an up-conversion mixer is always filtered out anyway.

The plots of the S-Parameters are shown below.



All the calculations that are available in sp are available in hbsp. Some things like the stability factor Kf are only defined when the input frequency and the output frequency are the same. When parameters become invalid because of frequency differences between the input and output, the *Direct Plot Form* displays an error, and the parameter does not plot.

The difference between sp and hbsp is that in sp, there is no signal applied to the circuit. In hbsp, a signal is applied to the circuit. The S-Parameters are measured with that signal applied. Also, many times the input and output frequencies are different in hbsp, and by definition, linear S-Parameters cannot change frequencies.

In the example above, the signal in the harmonic balance large-signal analysis was an LO signal for a mixer, but in the example below, the signal in harmonic balance is the RF input to a power amplifier. In the next example, the input and the output are at the same frequency and the input power is swept.

Amplifier with Swept Input Power

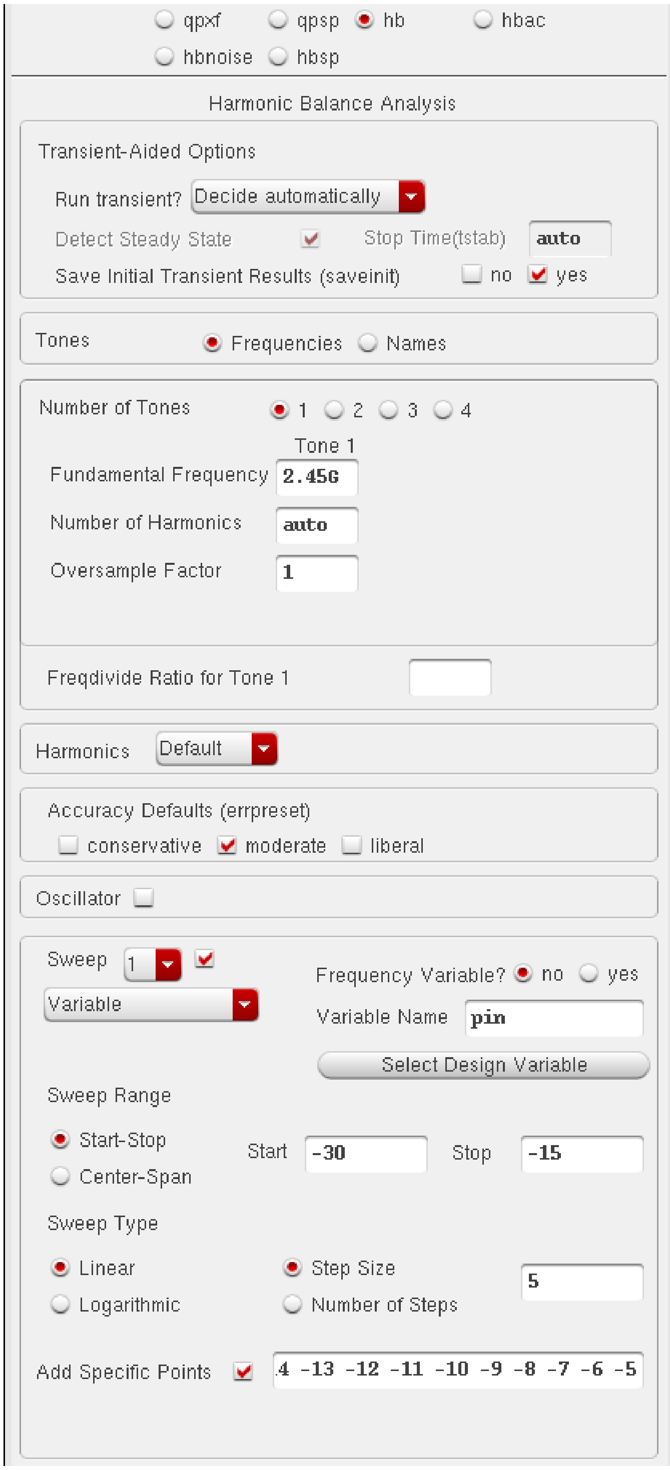
Imagine an amplifier with a tiny amplitude RF input signal at 1GHz. In this case, the amplifier is very linear, and if an S-parameter measurement is done at a small offset frequency, for example, 1.01GHz, the same measurement is obtained as a linear S-Parameter measurement at 1.01GHz.

Now imagine that the input signal at 1GHz is near the compression point of the amplifier. Because the system is much more nonlinear, if a small-signal S-Parameter measurement is done at 1.01GHz, the S-Parameters would be different. Because the input signal is large, the instantaneous input impedance changes, and this changes the input match at 1.01GHz. Similarly, the other S-Parameter measurements would be affected.

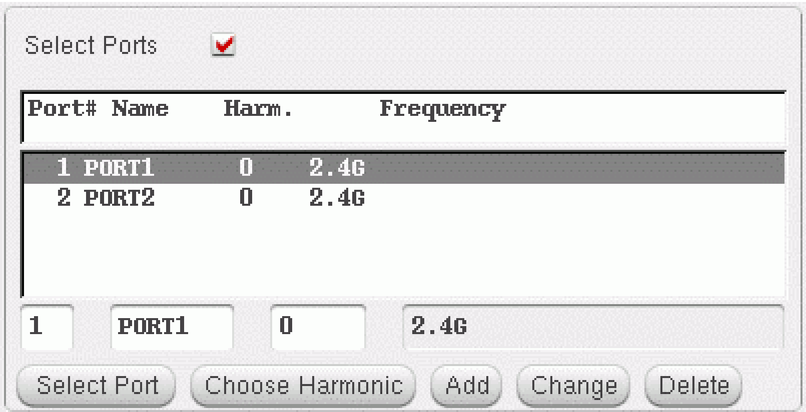
Hbsp runs after hb, so it can be used to characterize the small-signal performance with the large signal present. This is not a large-signal S-parameter measurement, but it is close. If a true large-signal S-Parameter measurement is needed, use LSSP to run that simulation. Only S11, S22, S21, and S12 can be measured with the LSSP.

1. Set the hb *Choosing Analyses* form as appropriate for your amplifier, as shown below. The delta-power at high input power is small by adding the additional points to the

Choosing Analyses form. At high power, things change quickly, so more resolution is needed.



2. In the hbsp *Choosing Analyses* form, set a single frequency point, as shown below.

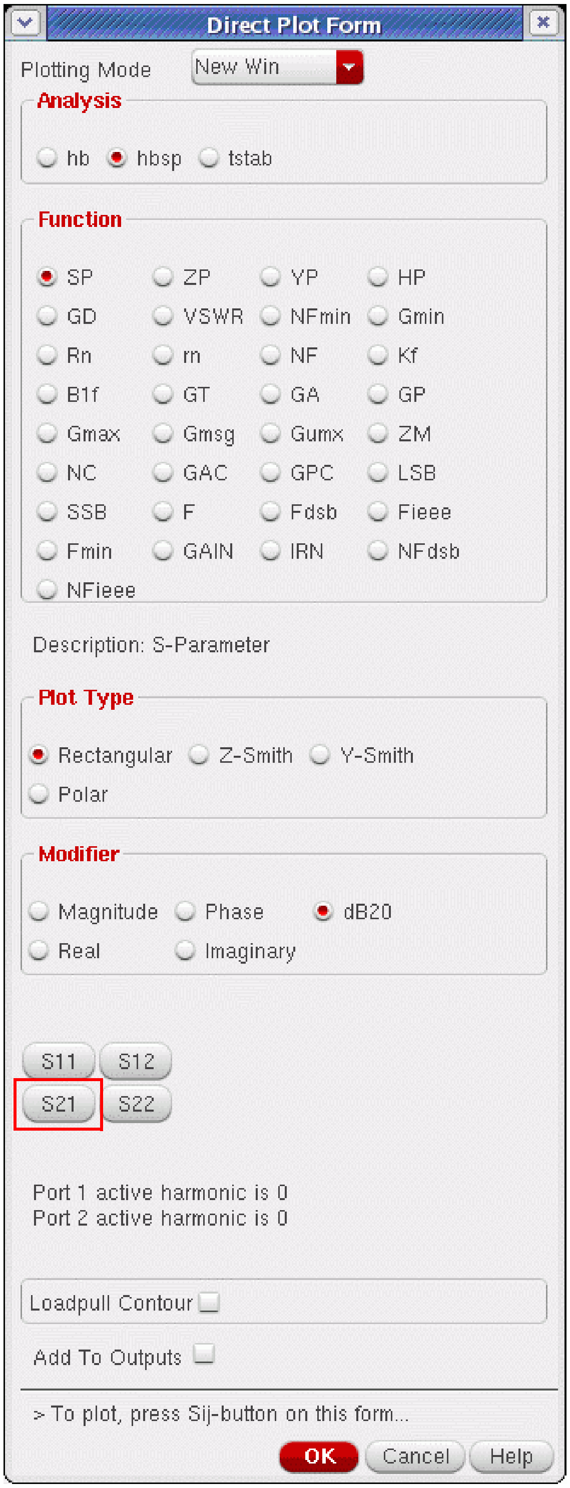


Note that both *PORT1* and *PORT2* are set to the same frequency. Use a single frequency point so the gain and stability functions return a single point in the sweep. You can use multiple points if you are familiar with the calculator.

3. Run the simulation.

4. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window.

The *Direct Plot Form* is displayed.

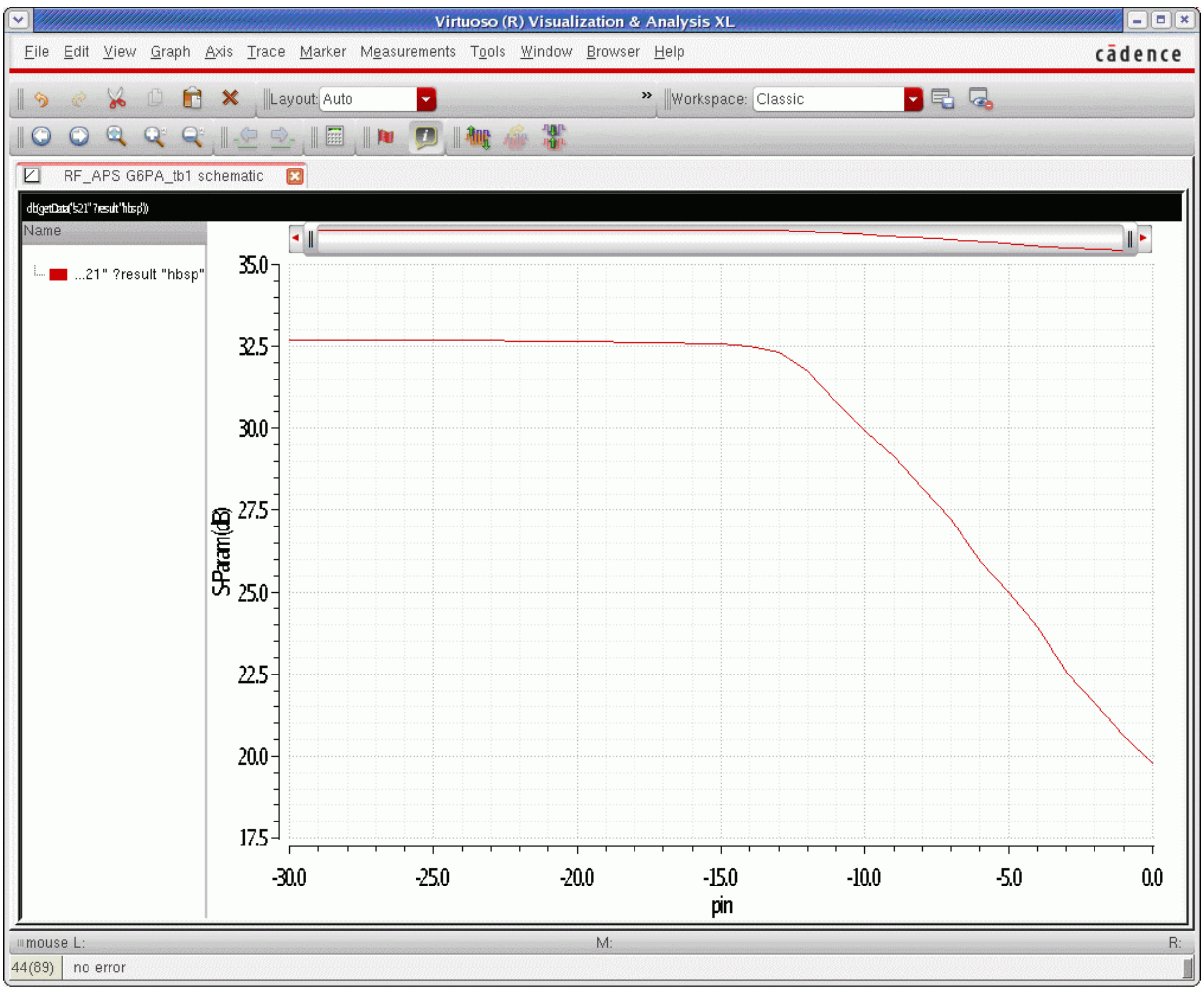


Plotting S21

In the *Direct Plot Form*:

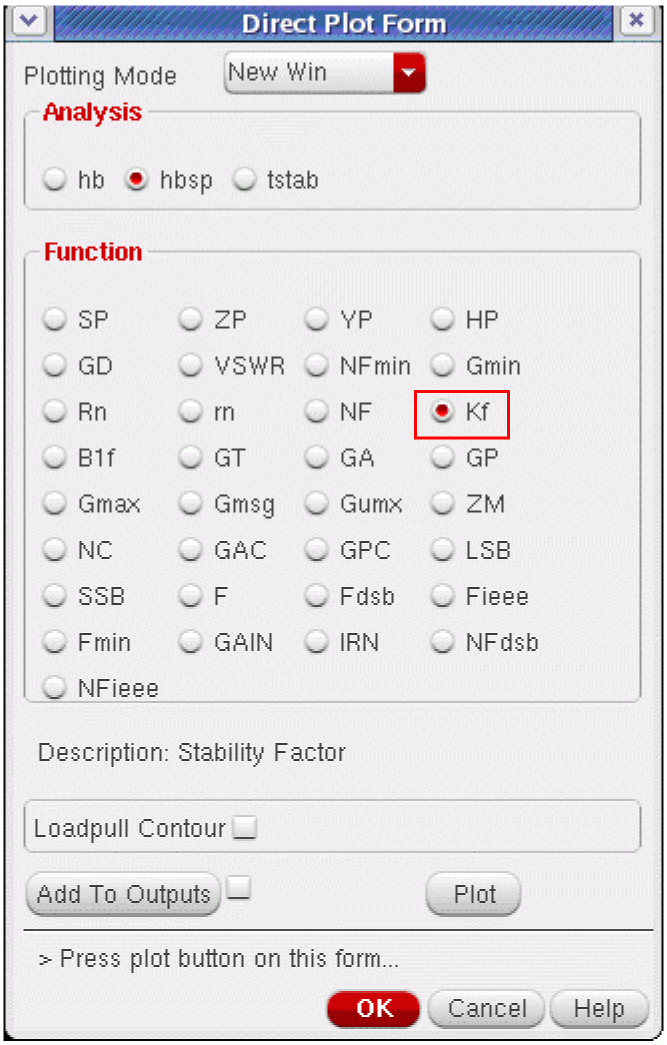
1. Select the *hbsp* radio button from the *Analysis* section.
2. Select the *SP* radio button from the *Function* section.
3. Select the *dB20* radio button from the *Modifier* section.
4. Click *S21*.

   The waveform displays *S21* as a function of input power.

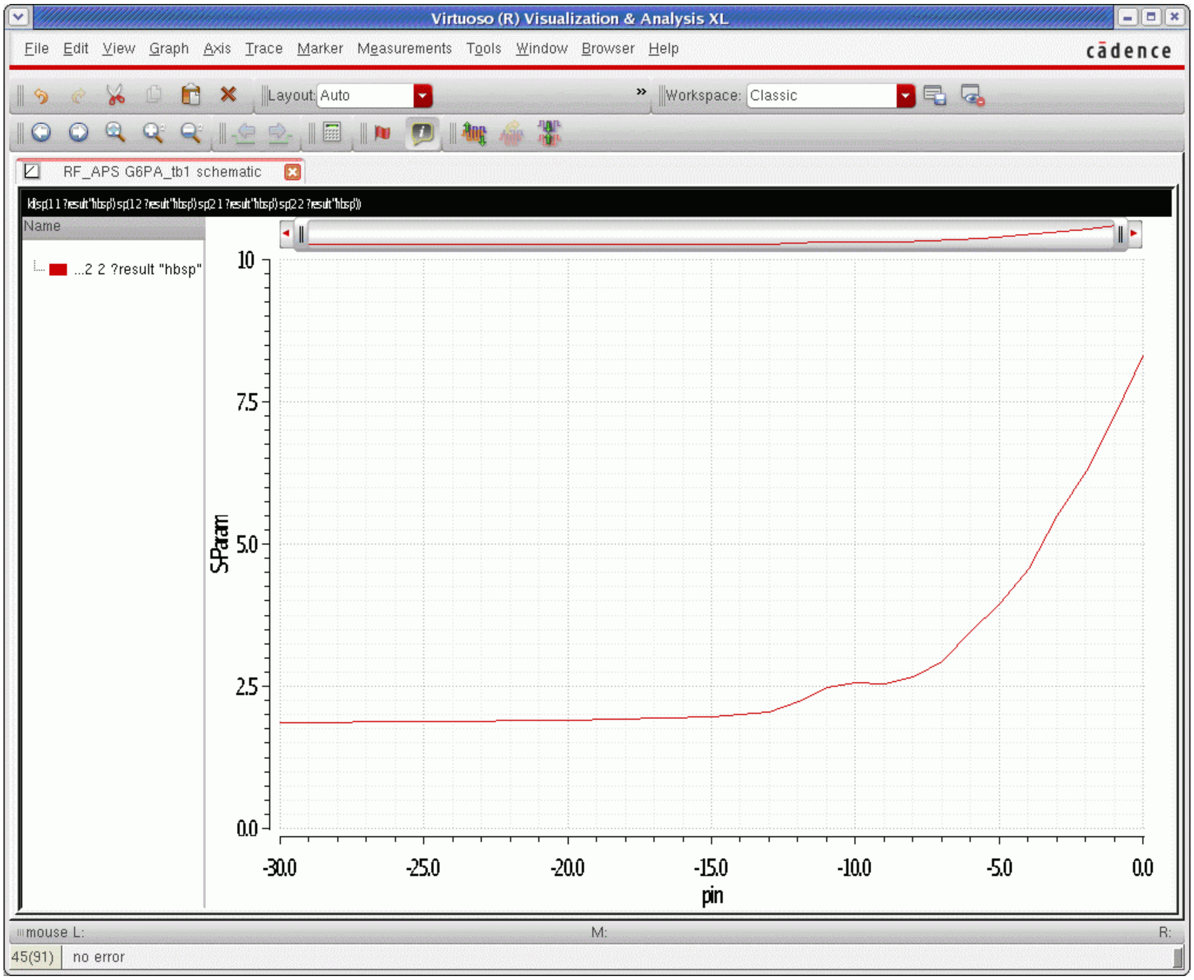


Plotting Kf

To plot the stability factor Kf, select the *Kf* radio button in the *Function* section and click *Plot*, as shown below. For the actual equations used for the different measurements below, refer to Chapter 8, "AnalogLib Components Used in RF Simulation,".

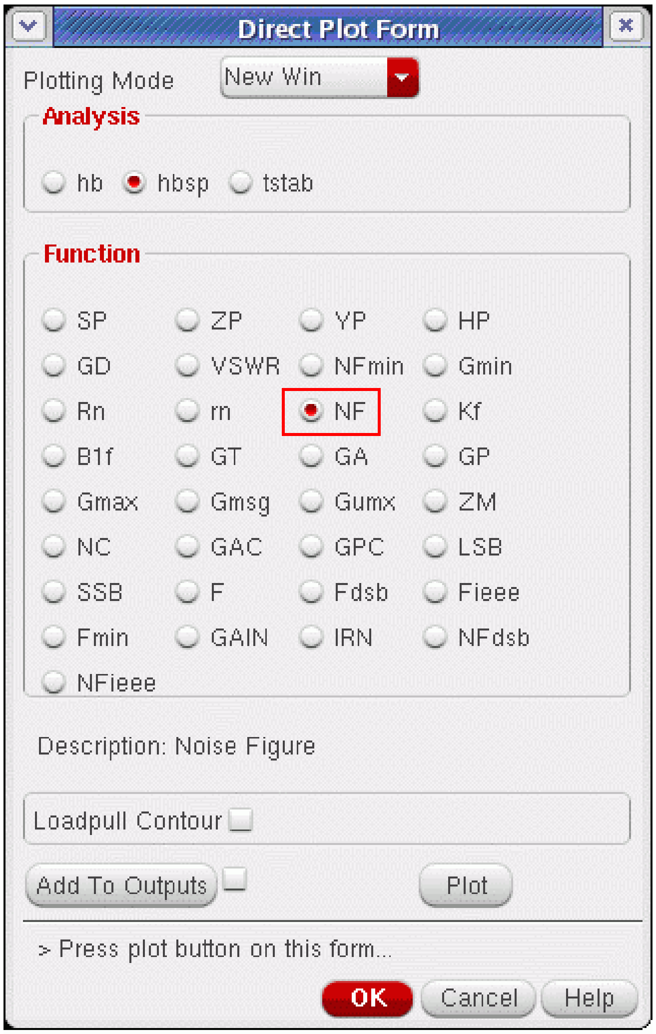


The waveform window displays Kf as a function of input power.

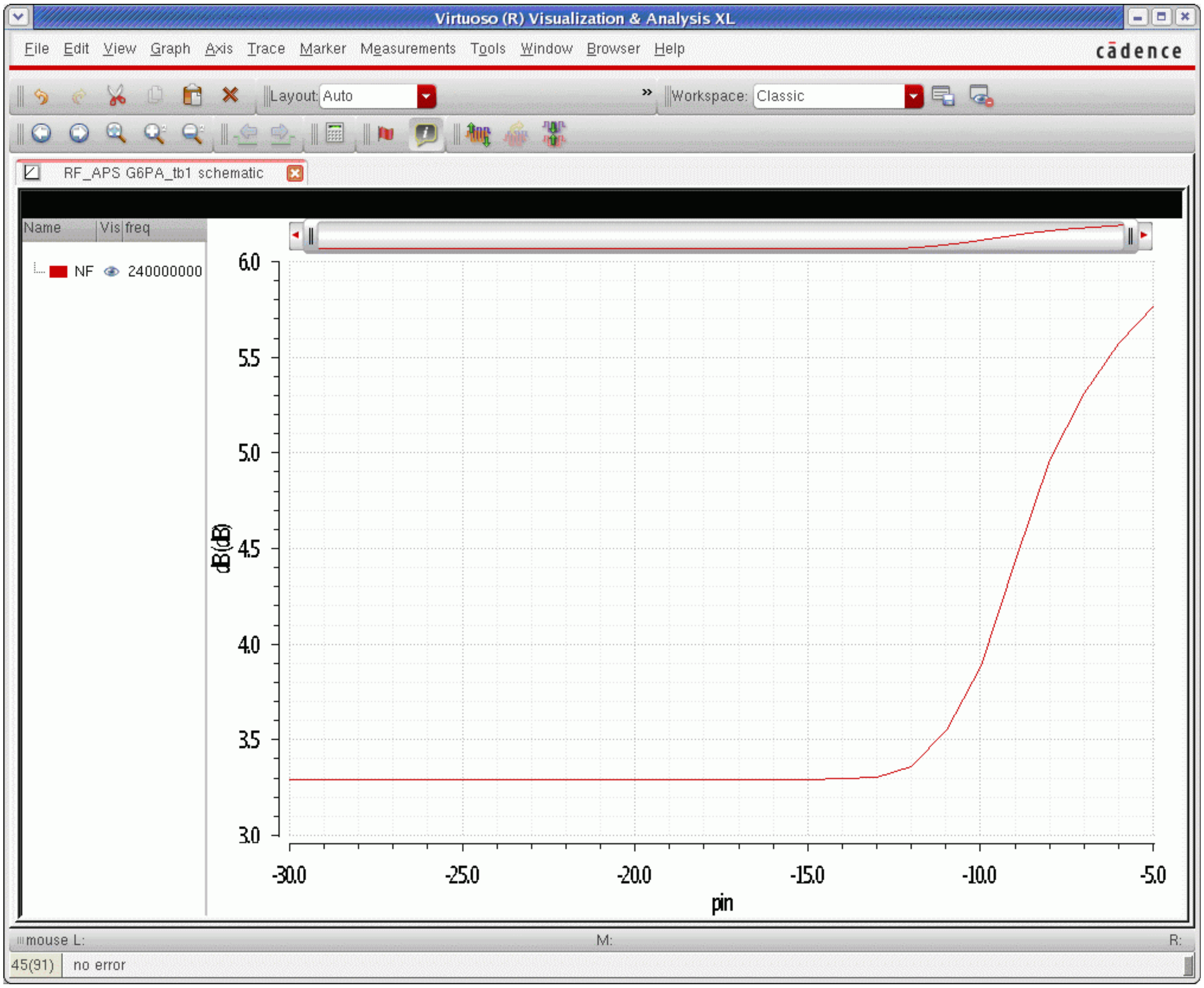


Plotting Noise Figure

To plot the Noise Figure, select the *NF* radio button in the *Function* section and click *Plot*, as shown below.



The waveform window displays noise figure as a function of RF input power.



Note that this is a power amplifier and not an LNA. Most LNAs have a much lower noise figure in the low RF input power region.

4

Single Input Large and Small-signal Analyses

Overview of Simulation Capabilities

SpectreRF offers two different modes, Spectre and APS. Spectre uses traditional simulation techniques. APS makes the CMOS models more efficient and allows multi-threading for the device current evaluation. In addition, APS adds full parallel solving of all the matrices, thus speeding up the simulation even more. There is no loss in accuracy because there is no change in the actual equations that are simulated, or in the options that control the accuracy of the simulation. These choices are available and are implemented for periodic steady-state (pss) shooting and pss harmonic balance. Both large-signal and small-signal `pac`, `pxf`, `psp`, `pstb`, and `pnoise` analyses have the Spectre and APS implementations.

Overview of Periodic Steady-State (pss) Analysis

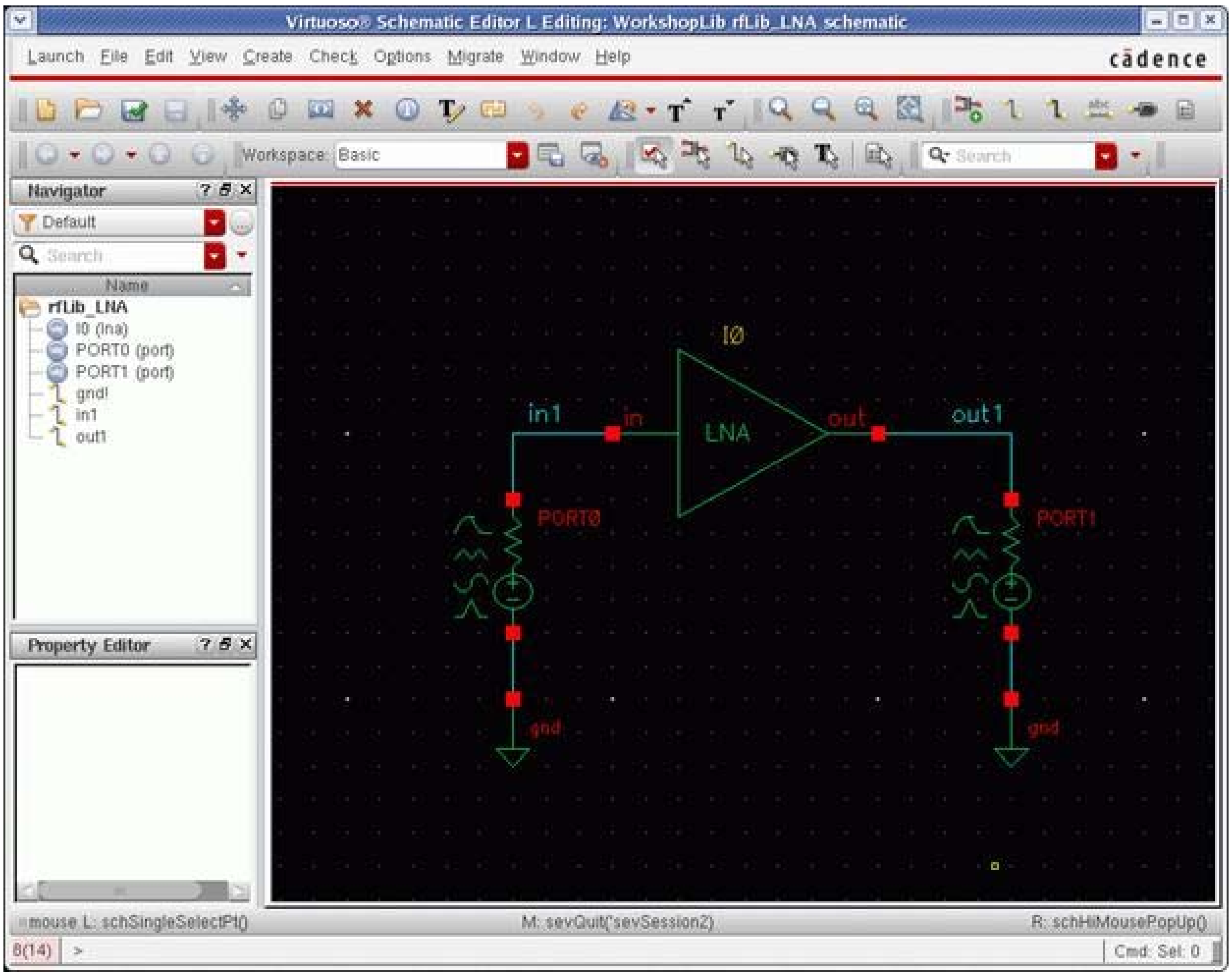
Pss calculates the settled time and frequency domain response of your system. Harmonic balance and shooting engines are available. Harmonic balance solves in the frequency domain and uses an ifft to calculate the time-domain response. Shooting calculates the settled time-domain waveforms and calculates the frequency-domain content using a Fourier transform.

All the harmonics of the beat frequency are calculated in the pss analysis up to the maximum harmonic specified in the *Choosing Analyses* form.

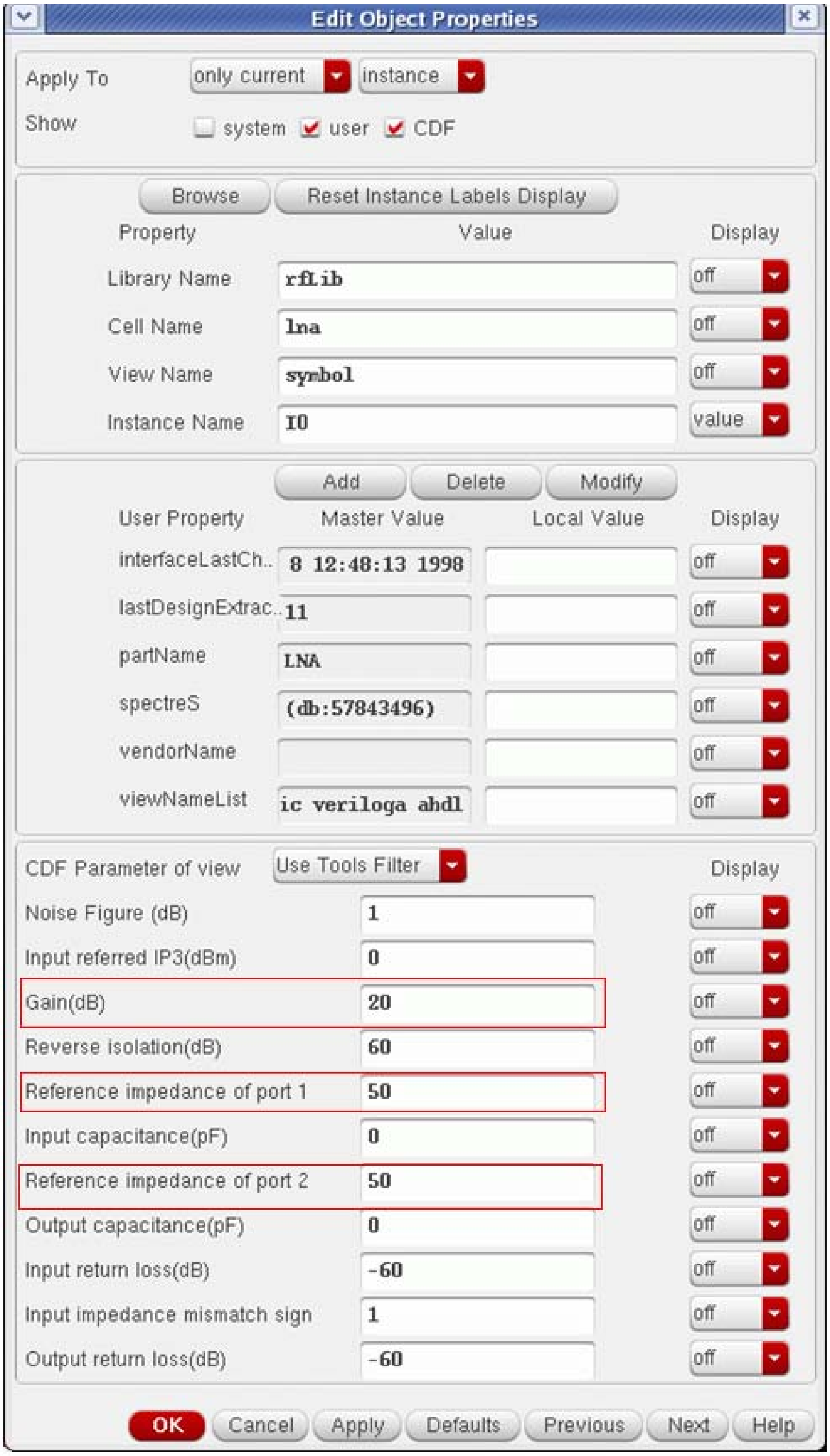
Pss needs to simulate an integer number of cycles. This limits the practical application of pss to single-input frequency circuits, or multiple-input frequency circuits that have a relatively high common multiple frequency, called the beat frequency. For example, if 1GHz and 1.1GHz frequencies are applied, the beat frequency is 100M. Solving for the fifth harmonic of the 1.1GHz input requires that 55 harmonics be calculated. This is a reasonable problem for pss. If the inputs are 1GHz and 1.001GHz, 5005 harmonics are needed to calculate through the fifth harmonic of 1.001GHz. This is not a good application for pss. If you need to simulate this kind of system, use qpss-harmonic balance or hb analysis instead.

Example

A behavioral amplifier circuit is shown below.

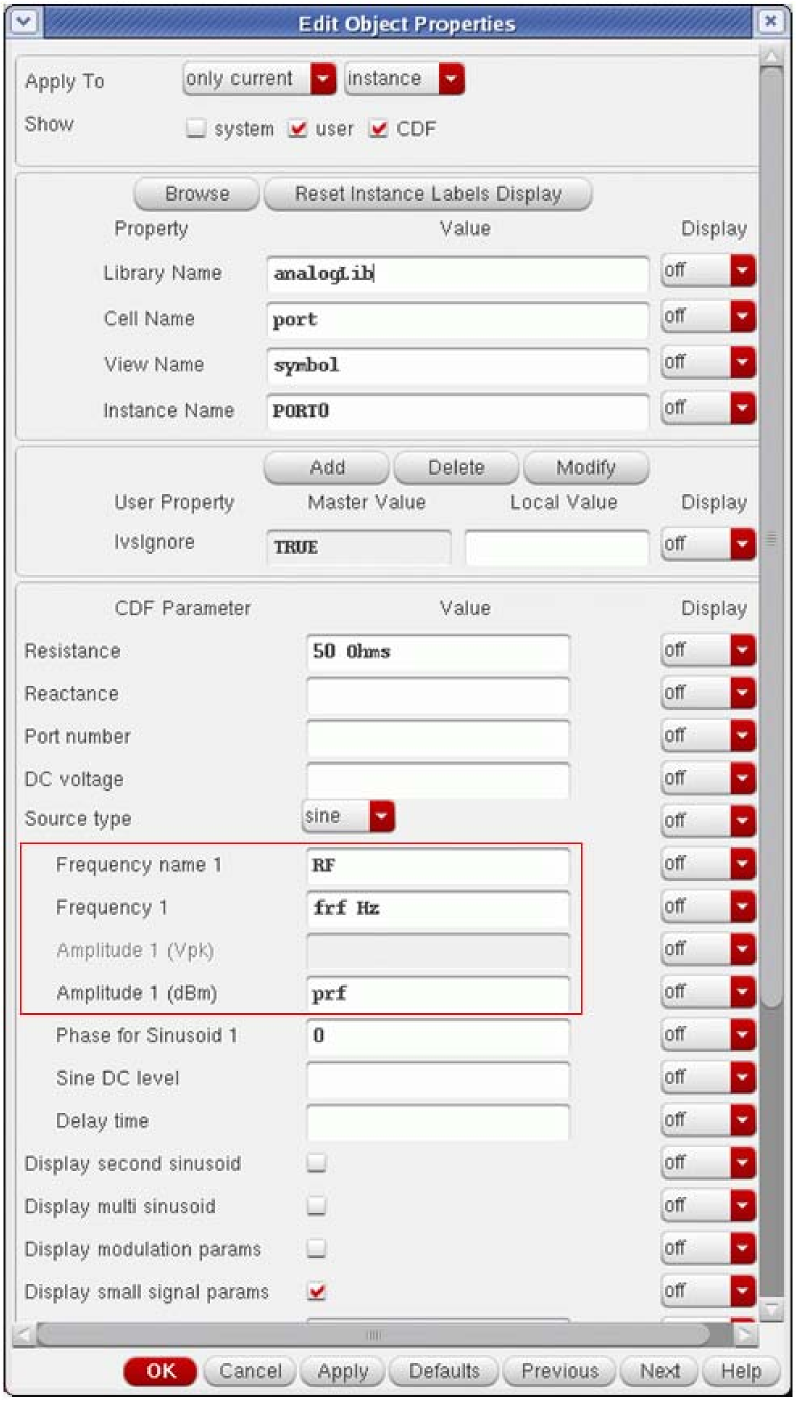


The properties for the amplifier are shown below.

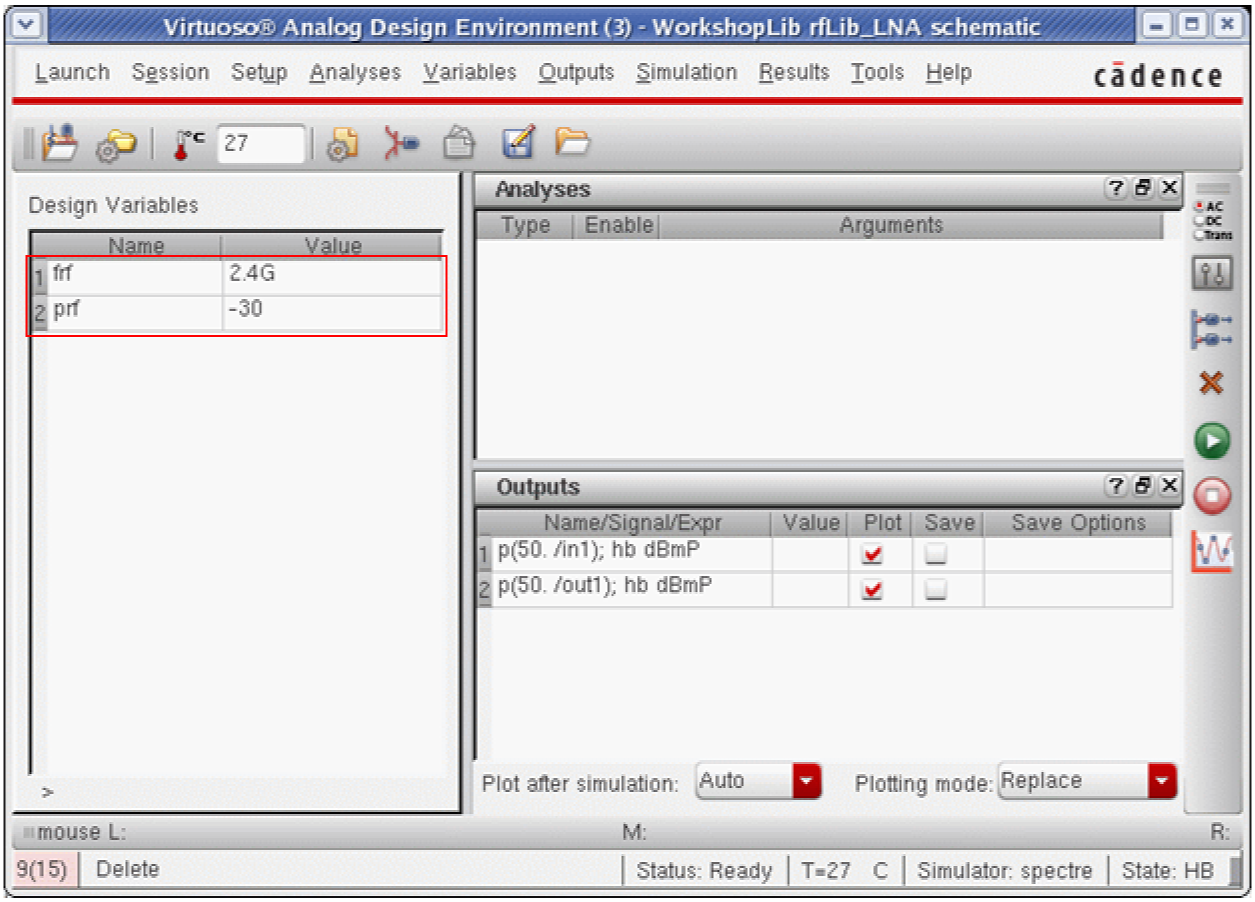


The amplifier has input and output impedances of 50 ohms and the gain is 20dB.

The input source on the left of the schematic is called a port. The properties of the port are shown below.

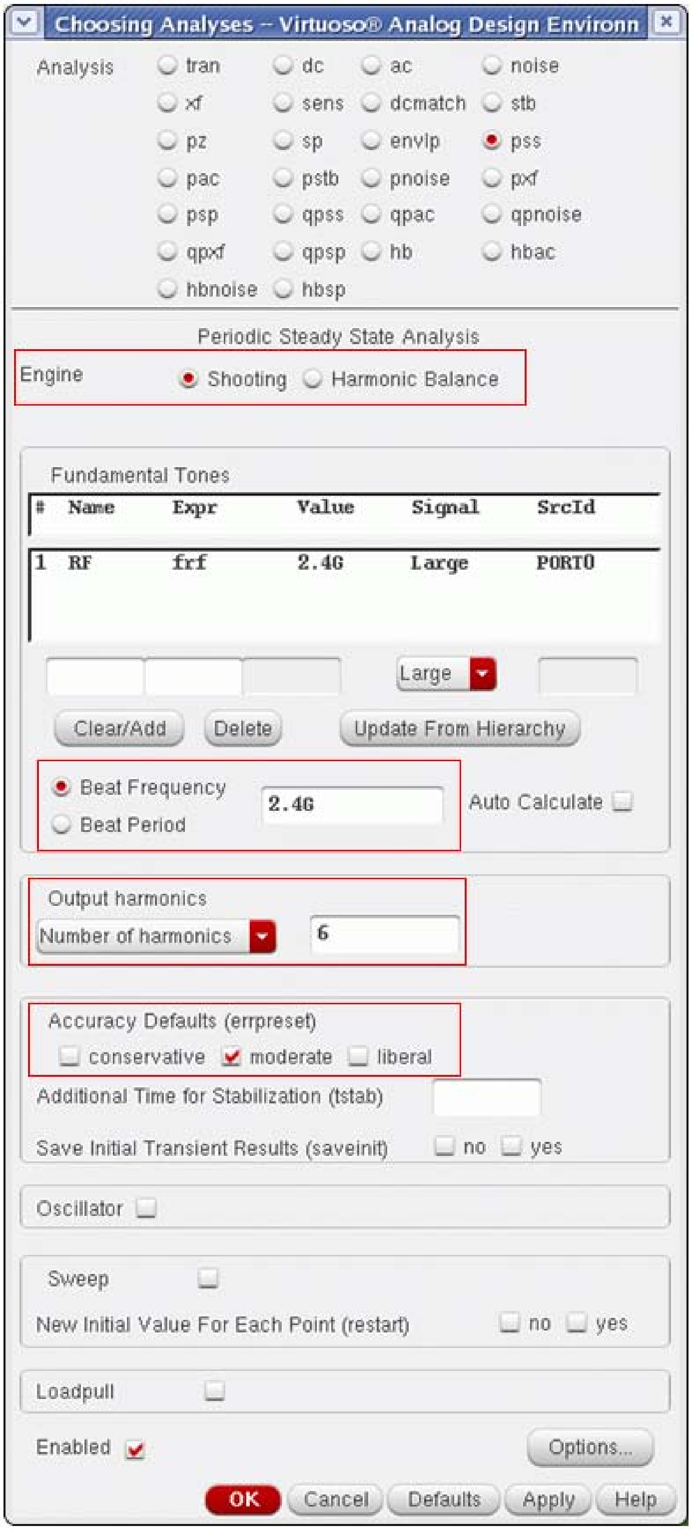


The port is set to generate sinusoids, and one signal is defined. For the pss analysis, you must enter a name for the signal in the *Frequency name 1* field. Variable names are used to set the frequency and amplitude so that the frequency and amplitude can be changed in the analog design environment (ADE) without needing to change the schematic. The ADE setup is shown below.



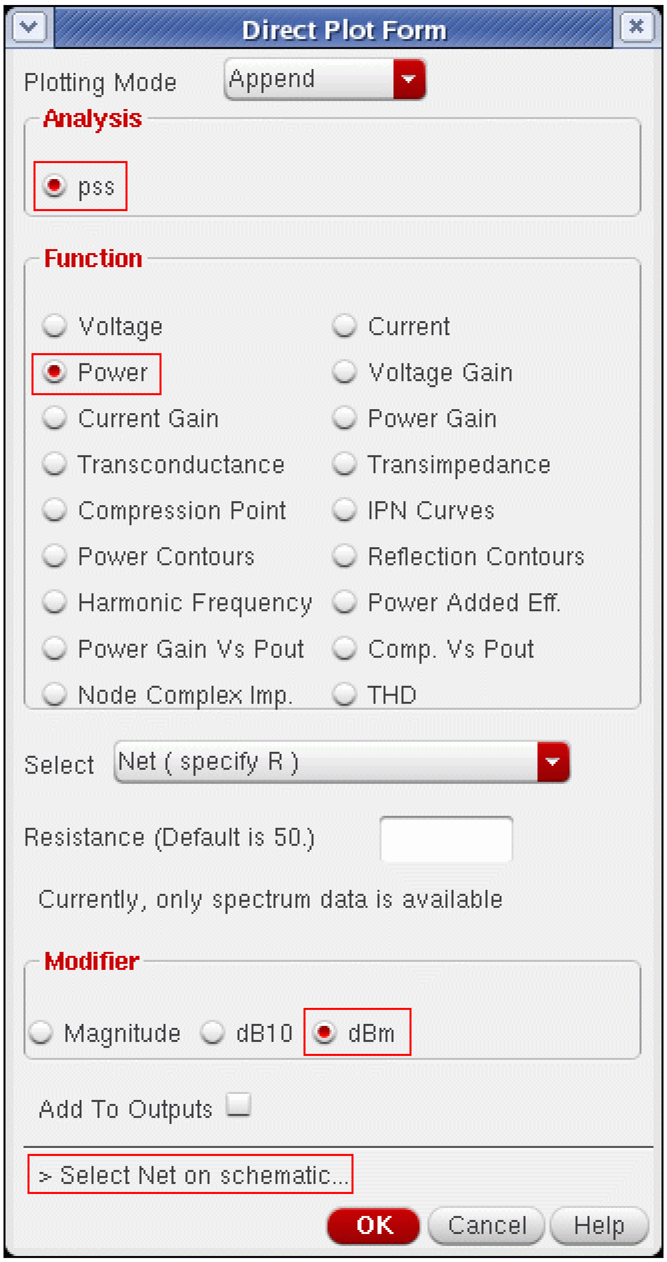
The input frequency is 2.4GHz and the input power is -30 dBm.

Next, the pss analysis is set up, as shown below.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
1 RF frf 2.4G Large PORT0
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
Beat Period
2.4G
Auto Calculate
Output harmonics
Number of harmonics
6
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

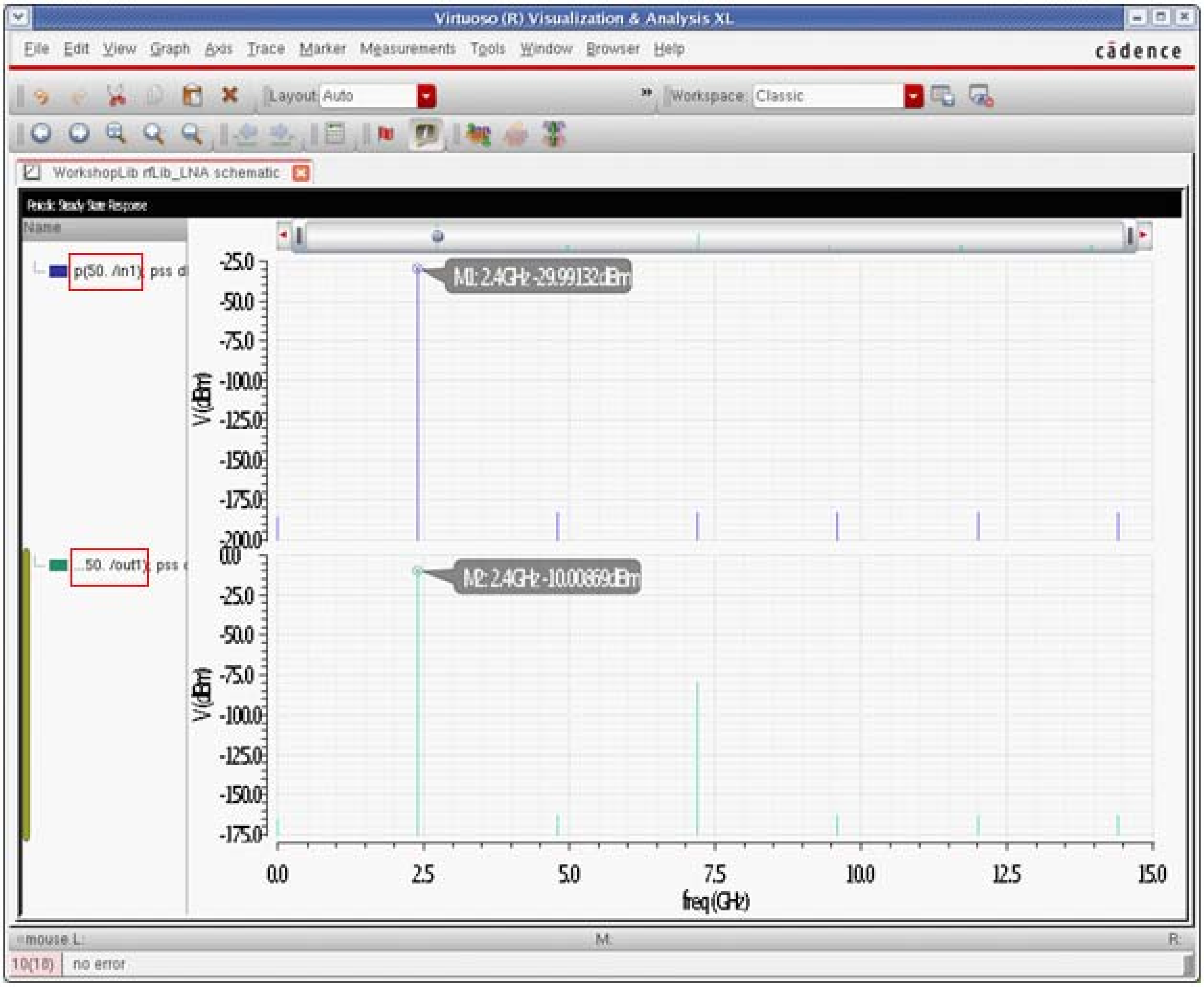
The pss *Choosing* Analyses form is set to a single input at 2.4GHz with 6 harmonics and moderate accuracy.

The simulation is run, and then in the ADE, *Results - Direct Plot - Main Form* is selected. The *Direct Plot Form* is displayed, as shown below.



Here, dBm is calculated based on the voltage on a node. At the bottom of the *Direct Plot Form* are directions. In this case, the net on the schematic needs to be selected.

The waveform window is shown below.



The basic display shows the harmonics that were calculated by pss. Markers have been placed on the first harmonics of the input and output. The input is -30dBm, and the output is -10dBm.

Harmonic Balance

There are three ways to run Harmonic Balance simulations in SpectreRF. These are:

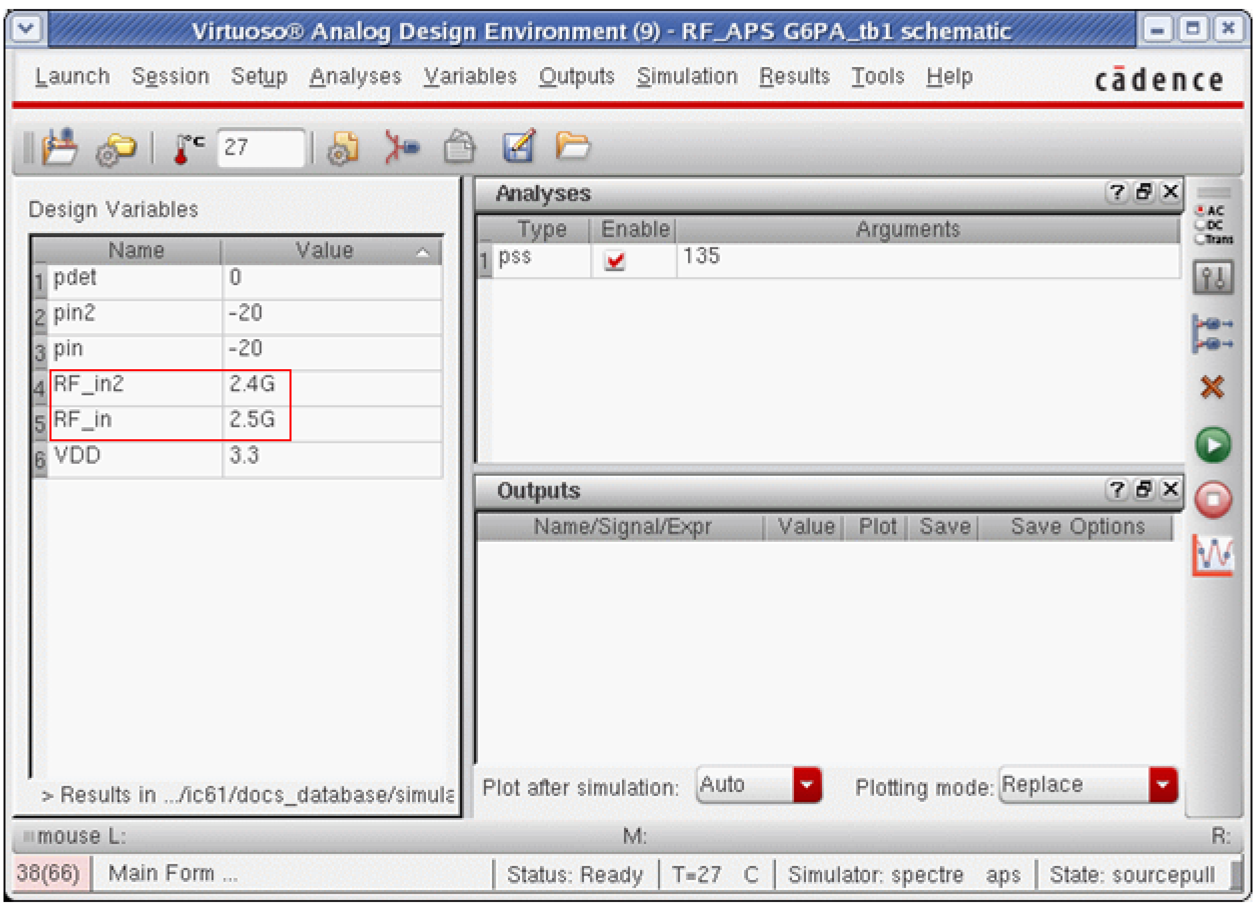
- pss-hb
- qpss-hb
- hb

Qpss with the harmonic balance engine selected and hb are the exact same analysis, however the hb *Choosing Analyses* form has all the latest improvements, and the qpss form does not. If you are considering using qpss-harmonic balance, it is strongly recommended that you use the hb *Choosing Analyses* form instead.

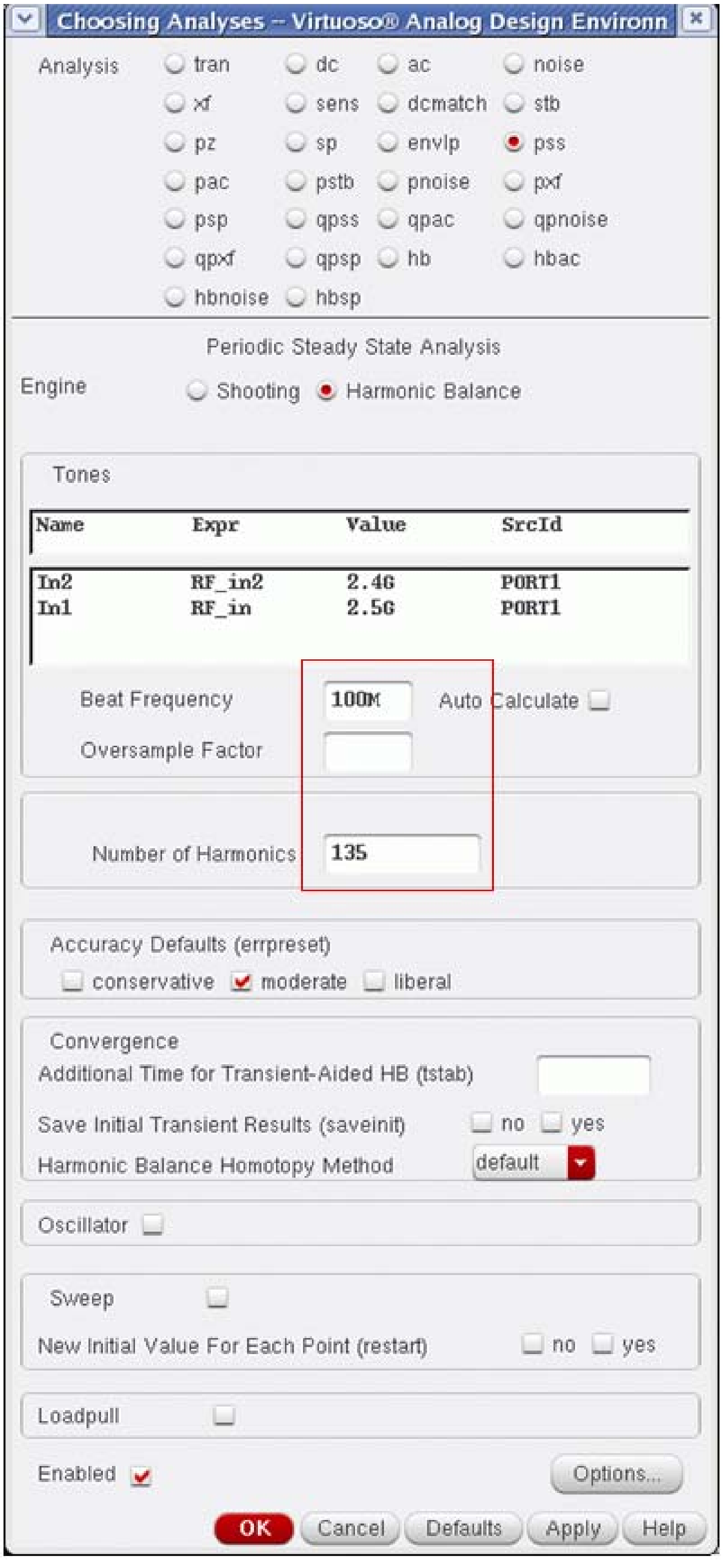
Harmonic balance in the pss analysis is slightly different than in hb and qpss harmonic balance. In pss, it is always a single-tone simulation. This means that all the harmonics of the beat frequency are calculated by pss instead of just the harmonics of the signals and the mixing products.

Also, there are some analysis options that are available directly in the ADE GUI for pss-harmonic balance that are not in the ADE GUI for hb. Although you do not require these options, you can still use an option that is not in the ADE GUI for hb by typing `option_name=value` in the *AdditionalParams* field for the option that you want to use in hb.

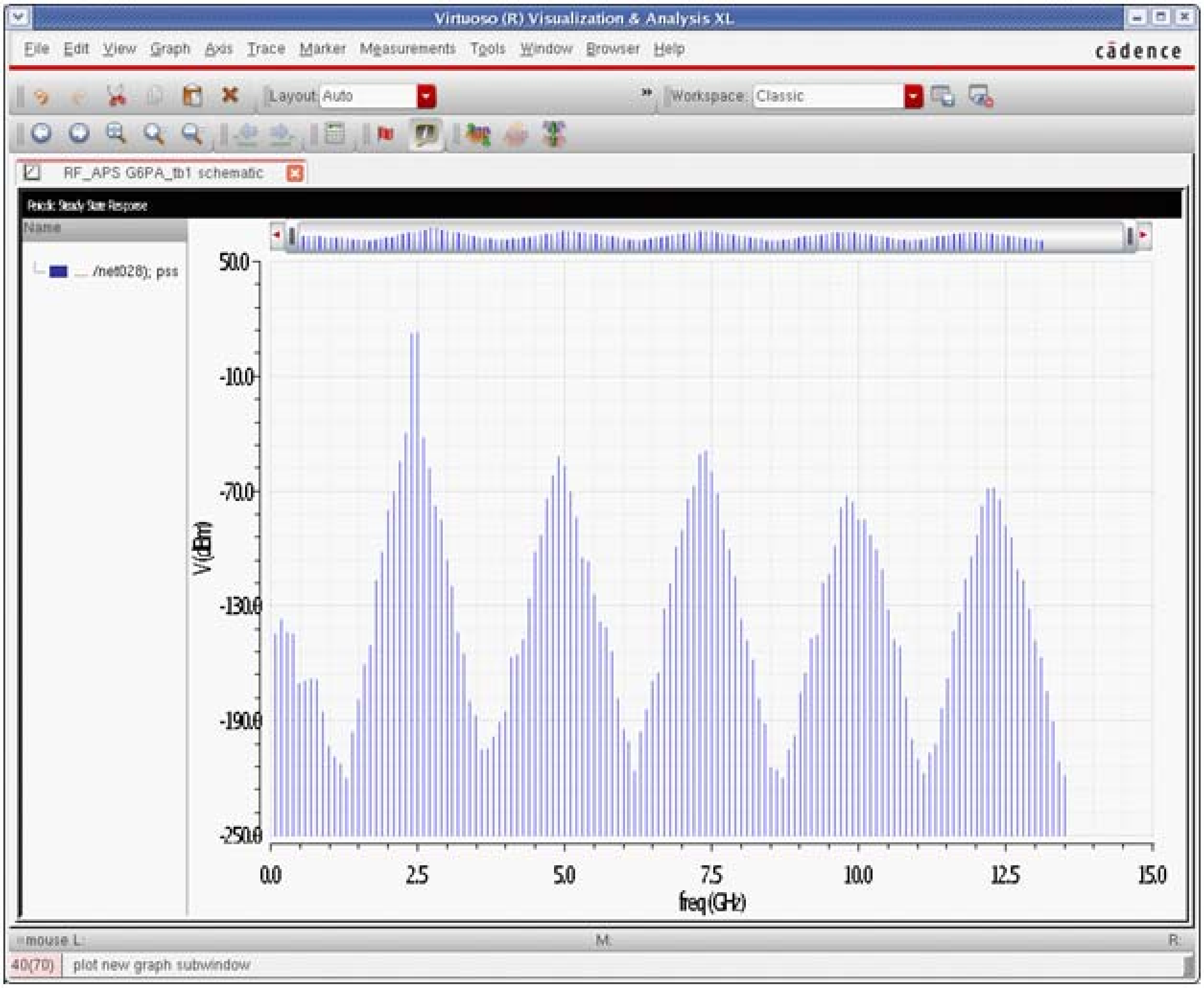
In the example below, there are two input frequencies at 2.4GHz and 2.5GHz.



The simulation was run with 135 harmonics so that the fifth harmonic of 2.5GHz was captured. Usually, running hb or qpss-harmonic balance is faster because fewer harmonics need to be calculated. This example is for reference only.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Tones
Name
Expr
Value
SrcId
In2
RF_in2
2.4G
PORT1
In1
RF_in
2.5G
PORT1
Beat Frequency
100M
Auto Calculate
Oversample Factor
Number of Harmonics
135
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Convergence
Additional Time for Transient-Aided HB (tstab)
Save Initial Transient Results (saveinit)
no
yes
Harmonic Balance Homotopy Method
default
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

The output spectrum shows that all the harmonics of 100MHz are calculated.



Again note that hb or qpss-harmonic balance requires fewer harmonics to be calculated because just the significant harmonics that the circuit actually produces are calculated.

Other than this difference, pss-harmonic balance is exactly the same as hb. For more information on harmonic balance, refer to Chapter 3, “Frequency Domain Analyses: Harmonic Balance” .

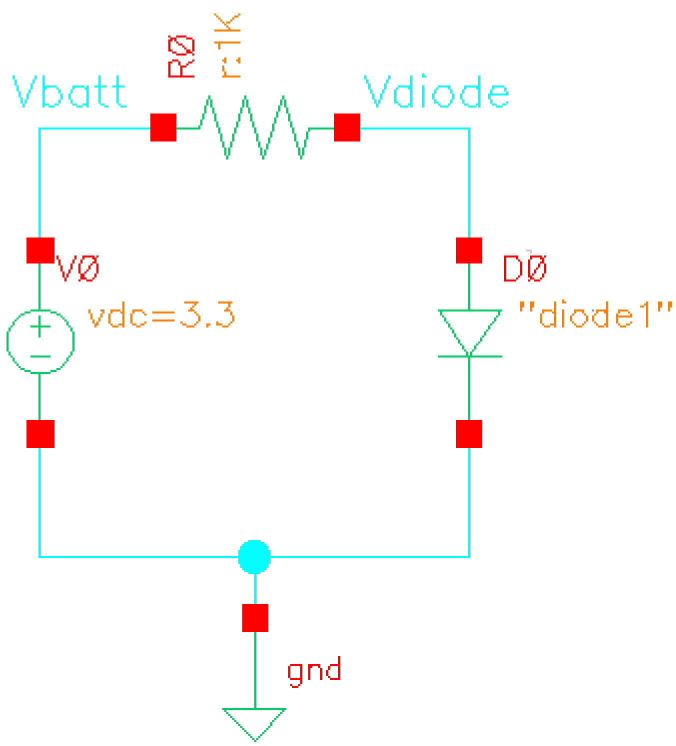
DC Principles

In order to understand the pss analysis, the fundamentals of simulation need to be discussed. Since the transient starts from a DC analysis that provides the time-zero timepoint, the DC Principles are reviewed. Since shooting pss is an extension to the transient analysis, Transient Principles on page 563 are also provided as a review.

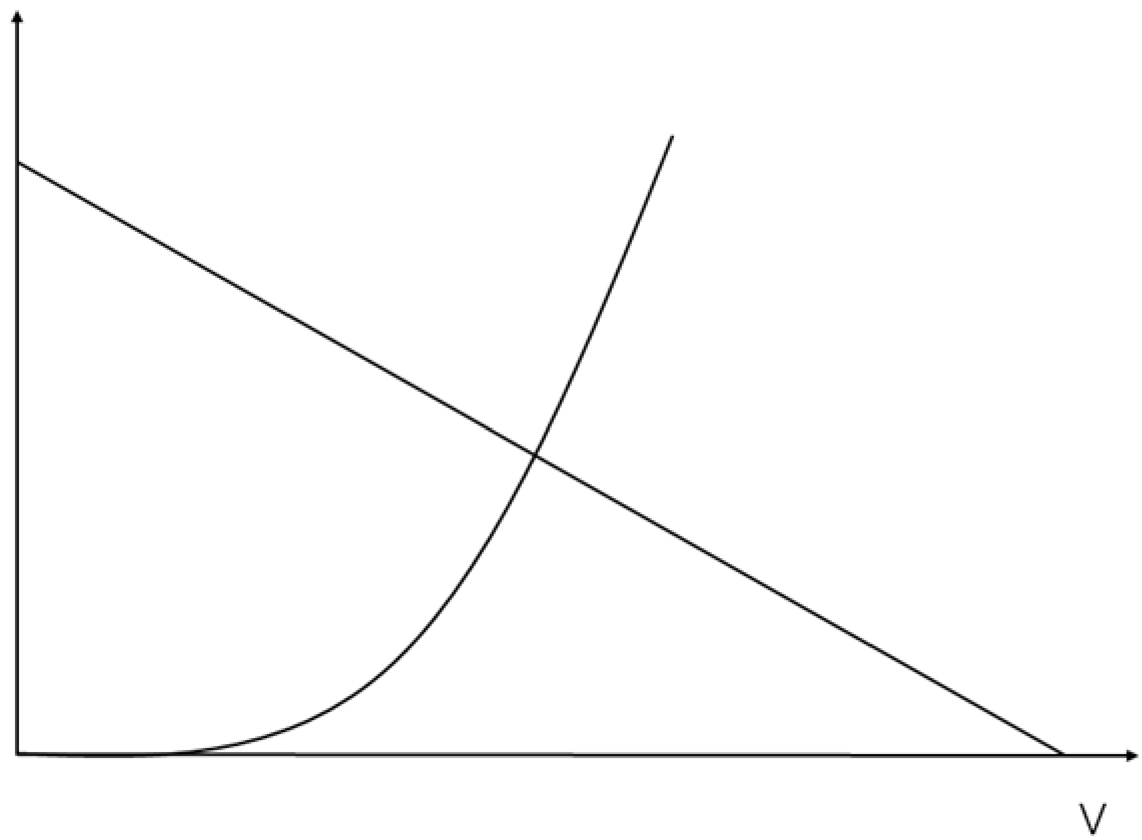
In any simulation, there has to be an equal number of simultaneous equations as unknowns in the circuit. In simulation, the KCL equations at each node are used. The fundamental assumption is that the sum of the currents at each node has to be zero.

The capacitors and inductors can be removed from the circuit because they have no effect on the DC solution. The resistors have linear current and voltage relationships and go directly into a linear algebra matrix. Sources also go directly into the matrix. These values do not change from iteration to iteration. The nonlinearities do not have constant resistance with respect to the applied input voltage.Therefore, they are replaced by resistors and current sources which go into the matrix. Now all the entries in the matrix are linear, and the matrix can be solved. An iterative technique is used to calculate the DC bias point.

Consider the circuit shown below.



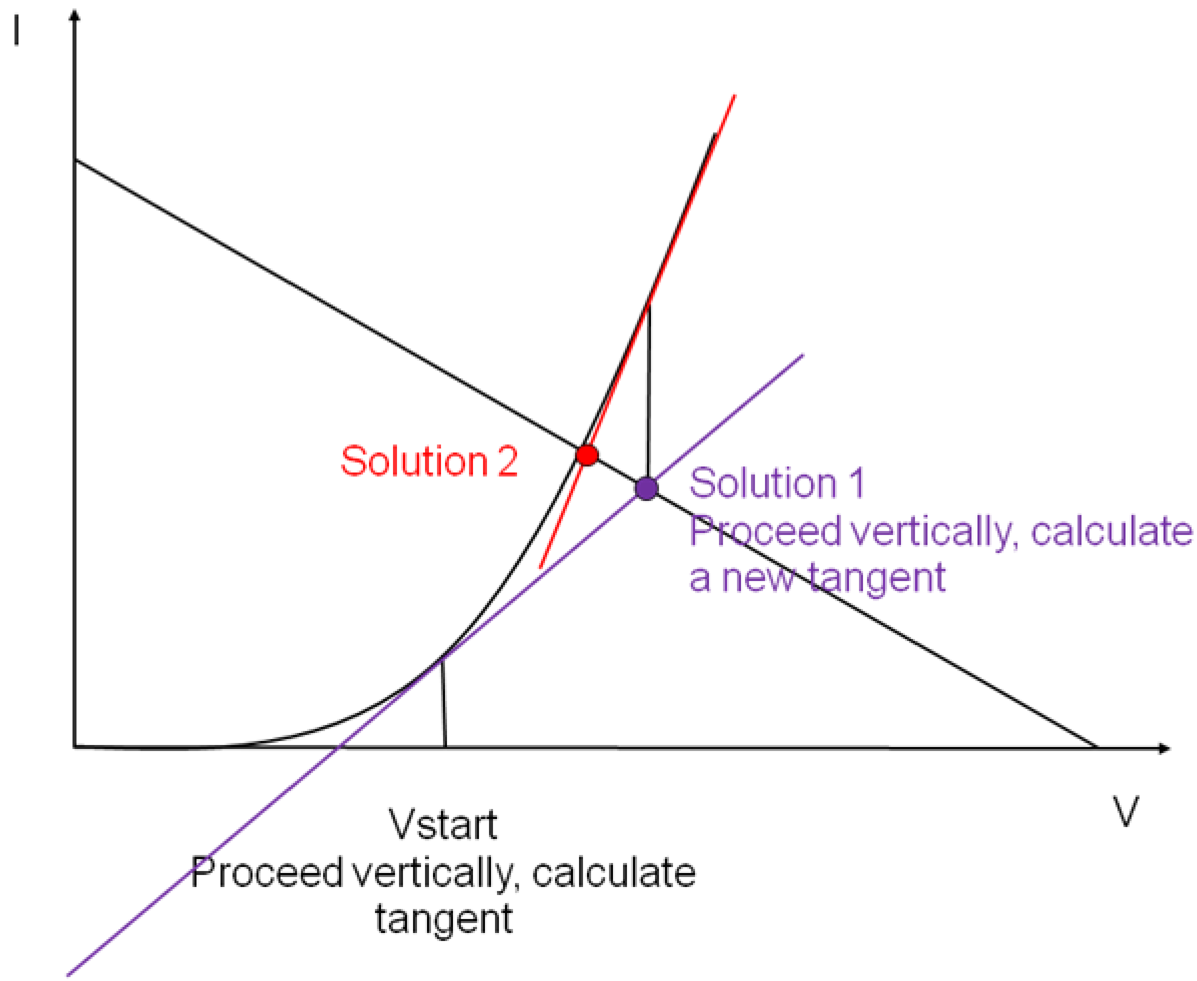
On a current versus voltage plot, the system looks, as shown below.



The intercept where the battery/resistor current coming in is equal to the diode current going out is the exact solution to the problem. Since computers cannot solve algebraic equations, an iterative approach is used to calculate the solution.

A starting voltage is assumed for each nonlinearity in the circuit. The default is to assume that the device is on. For a diode, that means about 0.7 volts across the device.

In the device model, the I versus V curve is available, and the ability to calculate the tangent at any voltage is also provided. For the first iteration, a line is drawn vertically from the starting point, and at the point where the vertical line intersects the I-V curve, the tangent is calculated. The nonlinear junction is replaced with a resistor and a current source. The current source value is the negative Y-Axis intercept of the tangent, and the conductance of the resistor is the slope of the tangent. The reason for doing this is that the resistor and current sources go directly into a linear algebra matrix that can be inverted and solved for voltages on each node. From that solution, a new tangent is calculated, that value is placed in the matrix, and it is solved again. This is shown below for the first two iterations.



Note that the iterations can continue forever, always getting closer to the correct answer, but never achieving the correct answer.

This process is used for every nonlinearity in the circuit. Note that for a more complicated device, there may also be nonlinear-controlled sources to model the output current.

Convergence

The iterations stop (and the circuit has converged) when the answer is close enough. Close enough is defined by the settings of the convergence options `reltol`, `vabstol`, and `iabstol`. Just after the linear solution is calculated, the current from the linear solution is compared with the nonlinear current from the nonlinear equation for all the nonlinearities in the circuit. When the difference in current is less than `reltol` * nonlinear current + `iabstol` for all the devices, a second check is made. When the change in voltage from the last iteration to this one for all nodes in the circuit is less than `reltol` * (the larger of the last or this iteration voltage) + `vabstol`, then a KCL check is made for all the nodes. When the sum of the currents at each node is less than `reltol` * (largest current into the node) + `iabstol`, the solution has converged. All three criteria need to be met everywhere in the circuit to have the iterations stop. When this happens, the circuit converges and the iterations stop. Again, note that the solution is not exact. To get a more accurate solution, `reltol` and `vabstol` need to be reduced from the defaults. More accuracy in DC usually costs a few more iterations.

The default for `reltol` is `1e-3` (0.1%). The default for `vabstol` is `1e-6` (1 microvolt). The default for `iabstol` is `1e-12` (1 picoamp). By default, when the answer is within 0.1% plus a little bit, the simulator stops iterating.

With the default values, note that when the solution is 1 millivolt, the error that is allowed is 0.1% of 1 millivolt + 1 microvolt or 2 microvolts. When the solution is larger than 1 millivolt, the relative term dominates the error tolerance. Similarly, when the current reaches 1 nanoamp, the relative tolerance becomes dominant. Because most circuits have voltages and currents larger than these, the relative term (`reltol`) usually dominates the error tolerances. The absolute terms (`vabstol` and `iabstol`) apply just to a very small region near zero.

The voltage that exists from the sources in the circuit at time zero is applied, and the solution of the DC algorithm is used as the first timepoint.

Skipdc

The default for `skipdc` is `no`. Normally, a DC solution is run for the initial transient solution at time zero.

In some cases, the time zero timepoint does not converge. Using the `skipdc` option allows the simulation to proceed to the transient without a DC solution for the time zero point. In this case, an assumed solution is used as the starting point. For the nodes that have batteries connected to ground, the assumed solution is the battery voltage. For nodes that have initial conditions specified, the initial condition is used. For all other nodes, zero volts is assumed.

When `skipdc` is set to `yes`, the assumed solution is used at time zero and the transient sources start immediately. Sometimes, this causes problems for the transient, therefore, setting `skipdc` to `sigrampup` causes the time-varying to start at minus 10 percent of the stop time in the `tstab` and ramp up from zero to the final value at zero seconds. When the default for `skipdc` causes problems for the transient, easing the input signal can help the transient get started.

Transient Principles

In the transient, the input voltage is allowed to vary as a function of time, and the capacitor and inductors are added to the circuit. The capacitors are modeled using a resistor and a current source in parallel, and the inductors are modeled as a battery in series with a resistor. The values of the resistors, batteries, and current sources are adjusted at each timepoint to accurately model the correct behavior of the capacitor and inductor. Note that the matrix is still scalar (No real and imaginary part). Keeping the matrix scalar speeds up the solution at each timepoint.

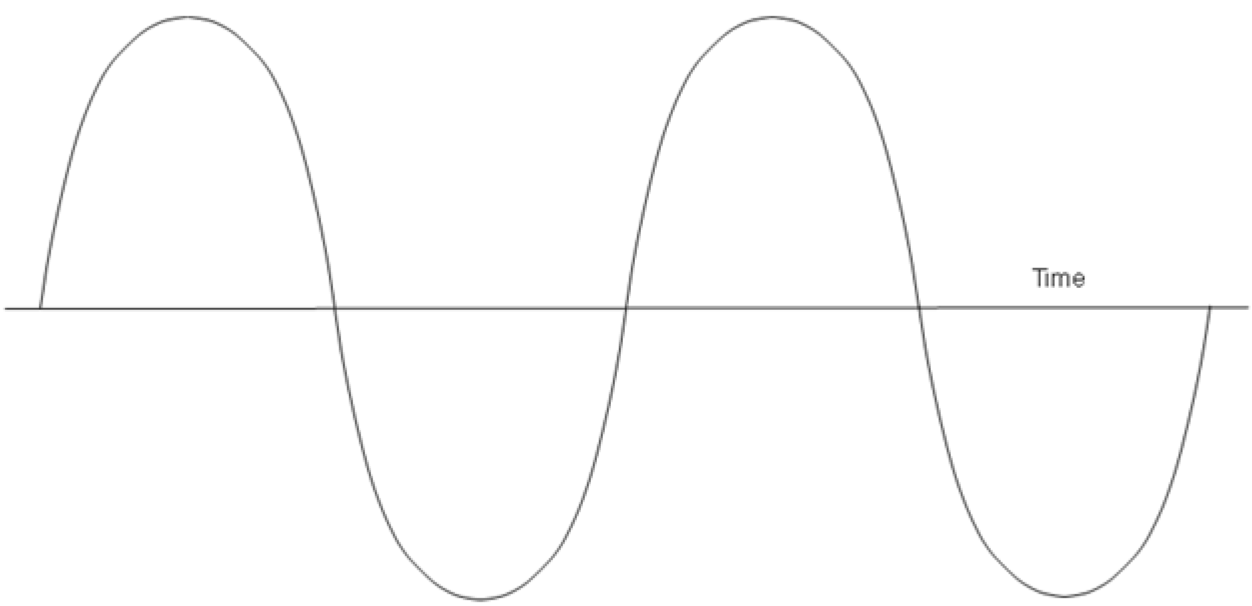
The system is broken into individual timepoints, and a solution is made at each timepoint.

Convergence

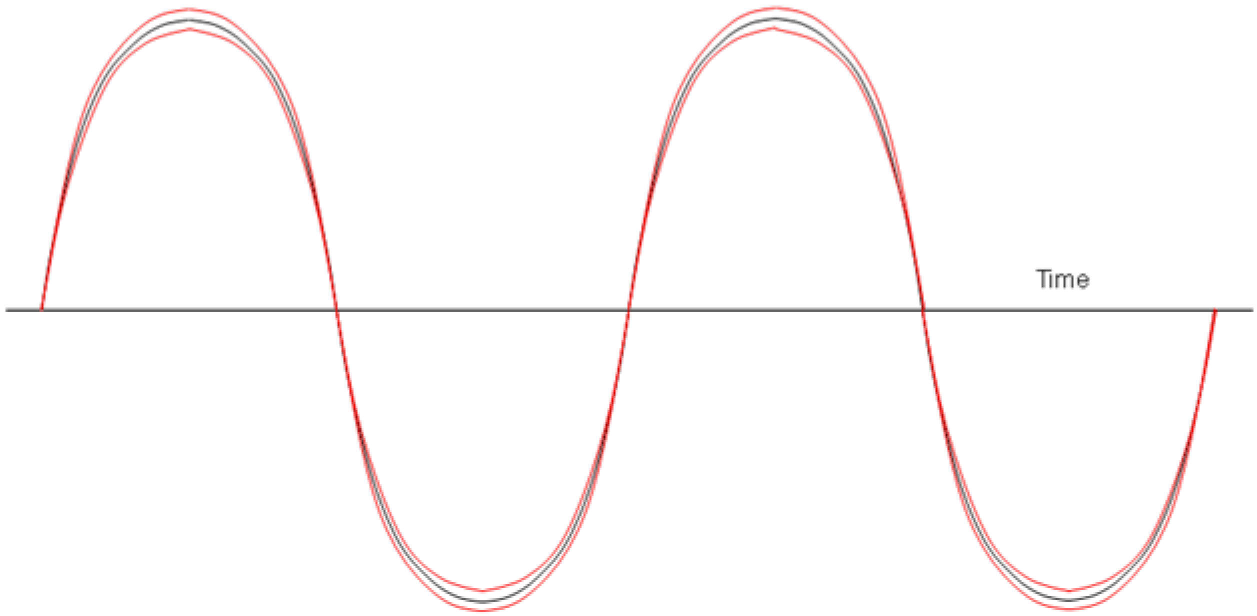
At each timepoint, the system is solved for using an iterative process, just like the DC solution. The difference is that in the transient, history is used to predict the movement of the circuit, so that the number of iterations at each timepoint is reduced. The same convergence criteria is used in the transient as in the DC analysis with the exception of where the `reltol * voltage` or `reltol * current` terms come from in the circuit. This is controlled by the `relref` option.

Relref

Consider a sine wave, as shown below.

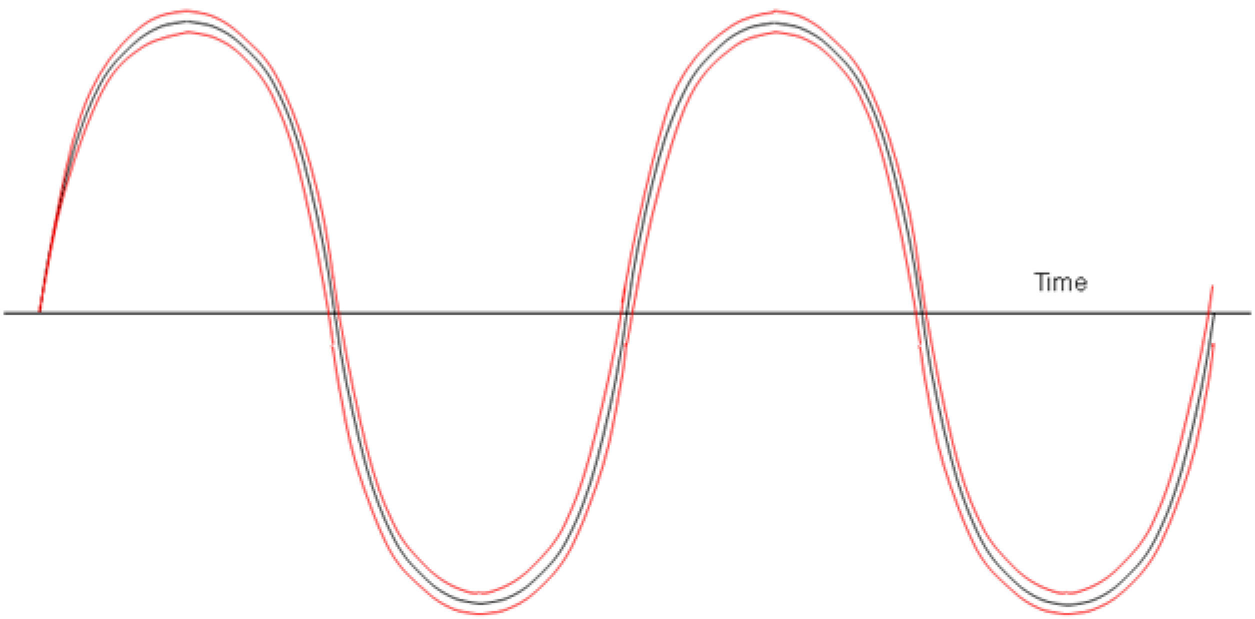


- Pointlocal criteria is `reltol` * the voltage or current at this node at this iteration or the last iteration, whichever is larger. Note that as convergence is attained, the solution is almost exactly the sinusoid, so when the voltage is small, the uncertainty is small. When the voltage is large, the uncertainty is large. This is shown below.



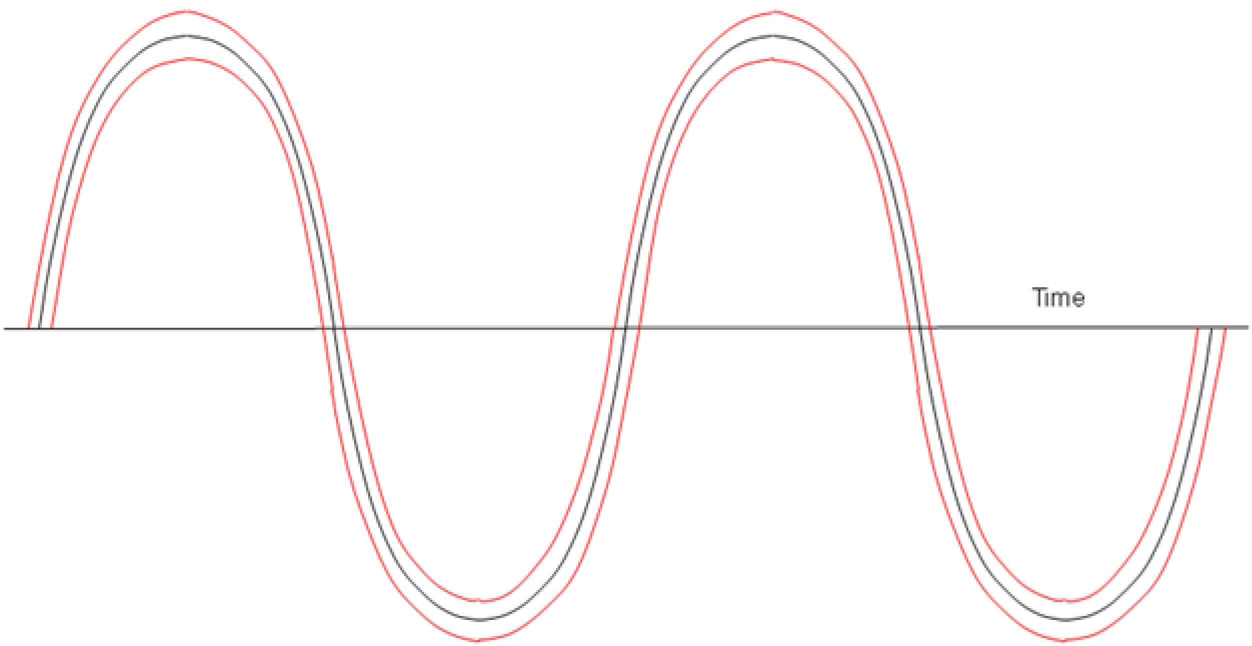
  Although this is the most accurate solution, more time is required for the simulation because of the very small error tolerances near zero. Usually, the local criteria, which is shown next is preferable.

- Local criteria is `reltol` * the voltage or `reltol` * `current` at this node at this or the previous iteration, or at any solution in the past. With a sinusoid starting at zero, the uncertainty is zero and gets larger as the voltage gets larger. Once the voltage reaches the peak, the uncertainty is constant after that. This is because `reltol` * peak voltage is always larger than `reltol`* the current solution.



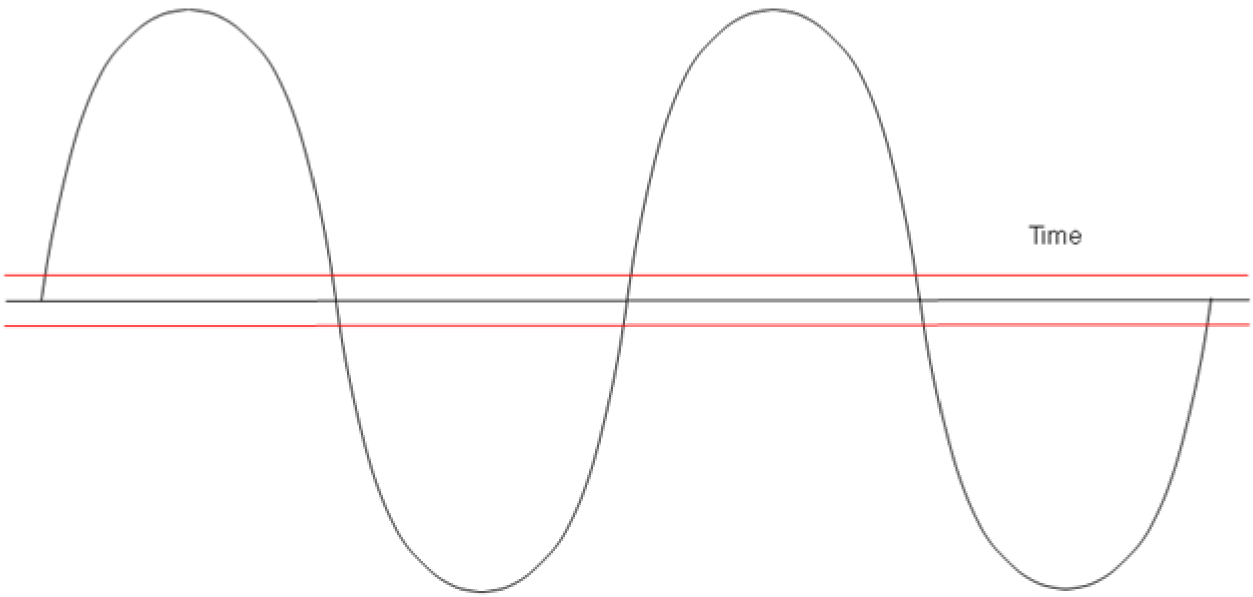
The error tolerance is larger after the first peak compared to `pointlocal`, but it is never larger than the value at the peak. This is a good compromise for accuracy of the simulation.

- Global criteria is `reltol` * largest voltage anywhere in the circuit now or at any time in the past. For most circuits, the largest voltage in the circuit is the power supply voltage and the largest current is the power supply current. Since the power supply voltage and current is usually larger than the node voltages and currents, the error is larger, as shown below.



Note that when global criteria are used, the error tolerance gets much larger for low voltage nodes and low current branches. Because of this, the accuracy degrades when using global criteria.

- Note that for `pointlocal` and for local criteria at `time = zero`, when the voltage is zero, `reltol` * voltage is also zero. Since every iteration produces a different solution, an absolute term needs to be added to the relative term so that convergence can be obtained when the voltage or current signal goes through zero. For voltages, this is set by the option `vabstol`, which defaults to 1 microvolt. (`1e-6`) For currents, this is set by the option `iabstol` which defaults to 1 picoamp (`1e-12`). This small absolute tolerance is added to the relative term at all timepoints. The absolute term is shown below.



The `relref` option affects the transient-based analysis only (tran, pss shooting, qpss shooting, and envelope shooting). It uses different combinations of criteria (`pointlocal`, `local`, `global`) for the different settings of `relref`.

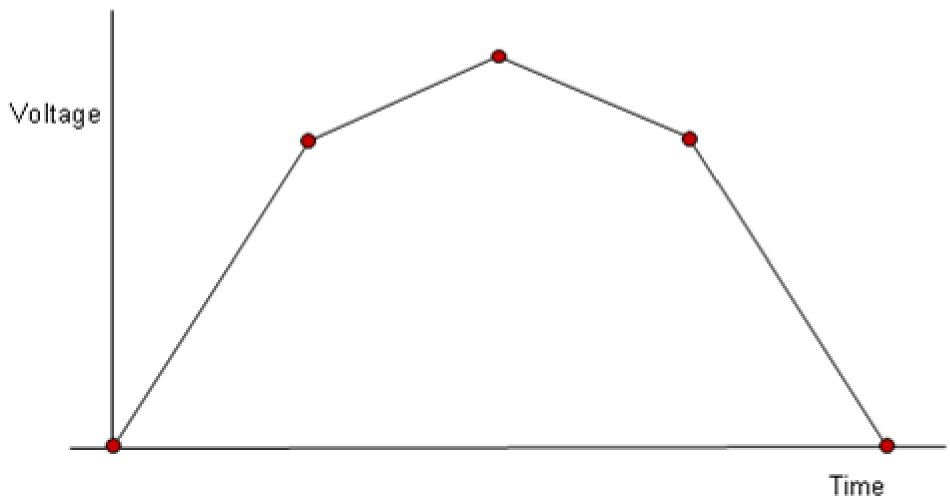
	Pointlocal	**Alllocal**	**Sigglobal**	**Allglobal**
Device Current Check	Pointlocal	Local	Global	Global
Delta-V Check	Pointlocal	Local	Global	Global
KCL Check	Pointlocal	Local	Local	Global

Note that in the order of most accurate to least accurate, the `relref` settings are `pointlocal`, `alllocal`, `sigglobal`, and `allglobal`. The default for most SpectreRF simulations is `alllocal`.

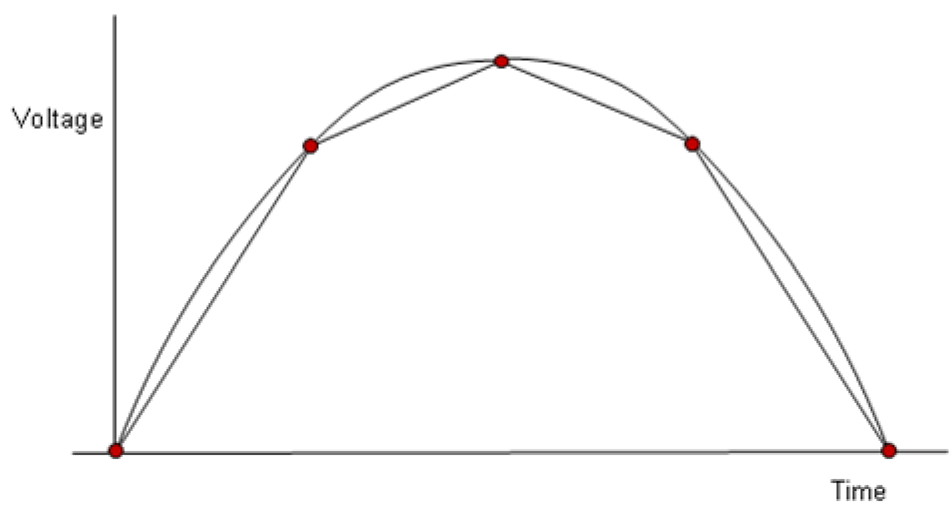
Integration Methods

Trapezoidal Method

In the transient, numerical integration is used for integrating current to get voltage on capacitors and for integrating voltage to get current in inductors. One method is to assume that the current is constant at the average current that flows during the interval. Because the current is constant, the voltage is a ramp during the timestep. The waveform that might be produced is shown below.

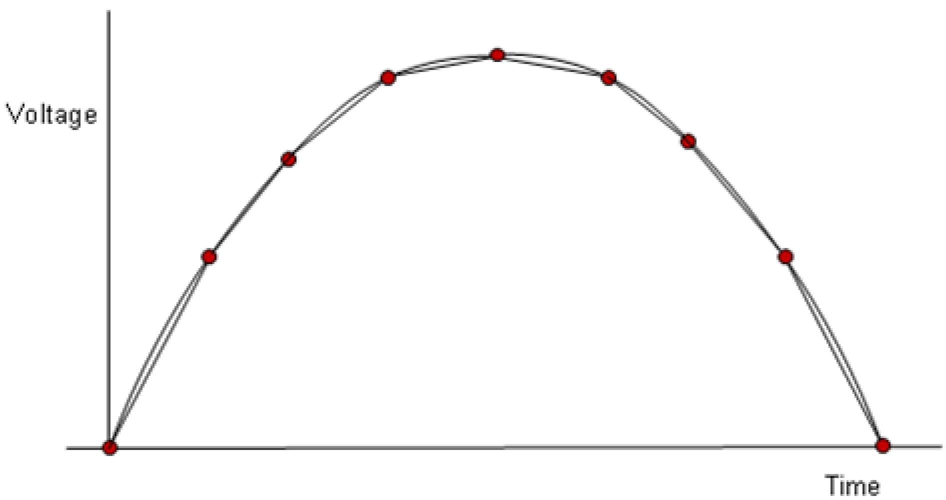


Note that the real waveform is usually a curve. This is shown below.



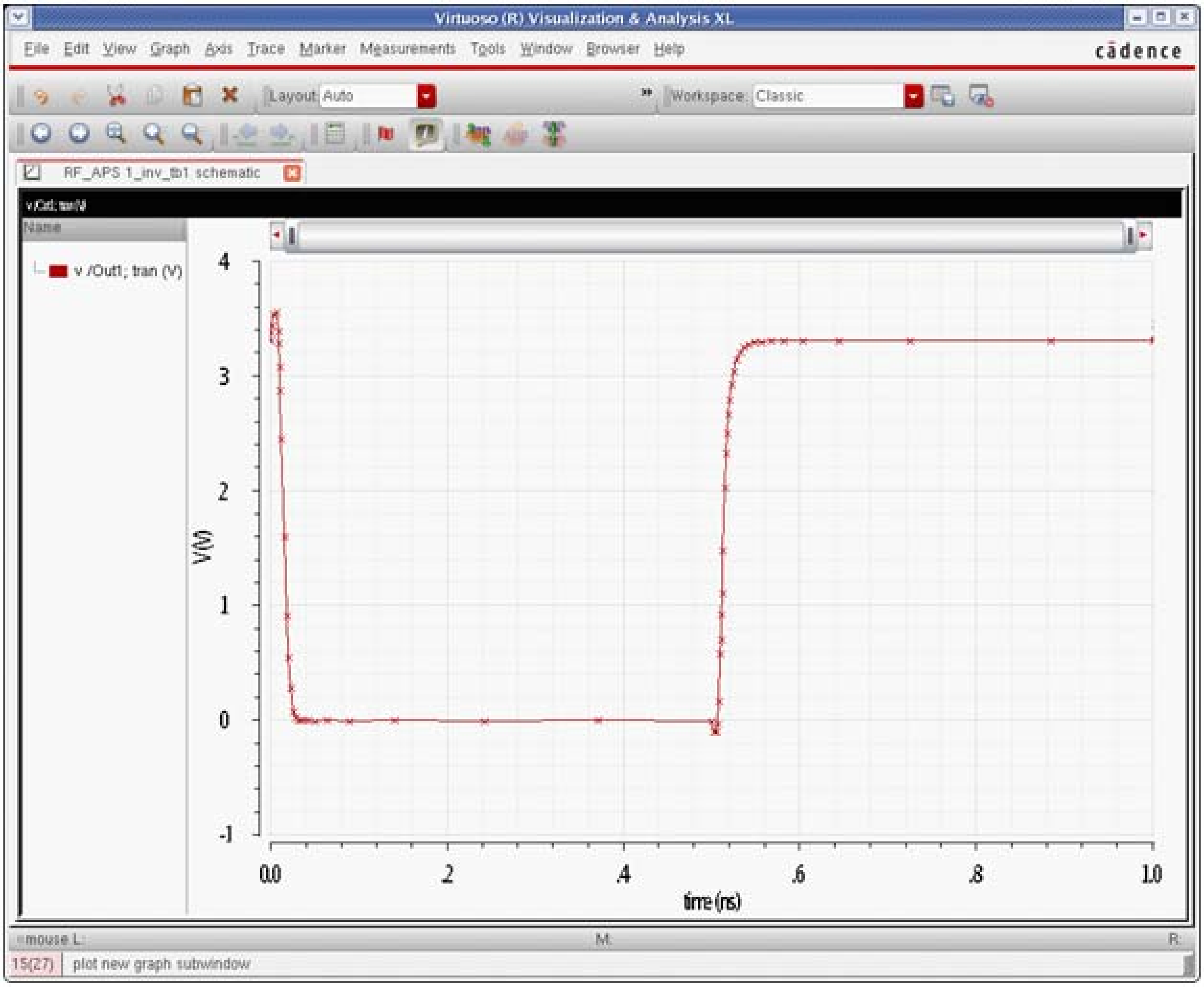
Note that the area between the straight line and the curve is the charge that is in the physical system, but not in the simulation. Errors are introduced in the simulation because of a finite timestep that is in the simulation. The timestep cannot go infinitely small because it would take infinitely long to run the simulation. As a consequence, numerical integration errors are inherent in the time-domain simulation because of the finite timestep.

This error can be controlled by making the timestep smaller, as shown below.



Note that by taking twice as many timepoints, the sum of the errors is much less than the previous case. One way to accomplish this is to set a small maximum timestep. Although this produces the most accurate waveform, the runtime cost is usually quite high because the maximum timestep is used everywhere in the simulation.

When the waveform is at a constant value, there is no numerical integration error. The place where the error occurs is the curvature that is produced by the circuit. Spectre controls the timestep based on the curvature that is produced by the circuit. Because of this, the timestep is inherently a variable, as shown in the below waveform. The X symbols on the waveform indicate the place where a timepoint exists in the transient output data.



Note that where the waveform is flat, the timestep bumps out, and where there is a high curvature, lots of points are taken. In this way, the accuracy of the waveform is preserved with the least possible number of timepoints. This is done to reduce the runtime and is the normal method of timestep control.

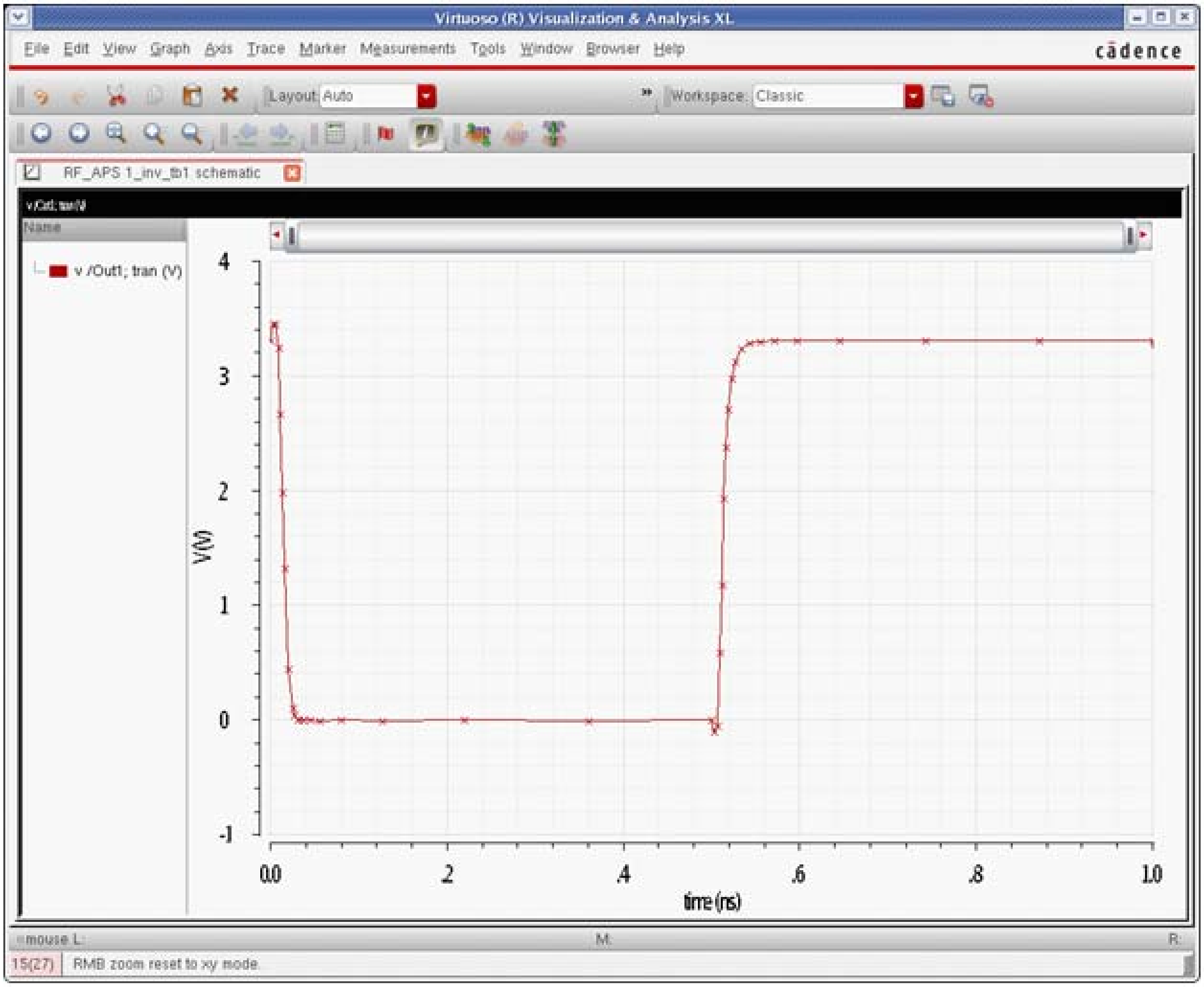
The trapezoidal method is generally good for normal transient simulations because the method does not exaggerate or damp any ringing that the circuit might produce. Although it is not visible in the waveform above, the trapezoidal method has ringing. At each point, there is an error term that alternates its sign at each timepoint. First a bit up, then a bit down.

Gear2 method

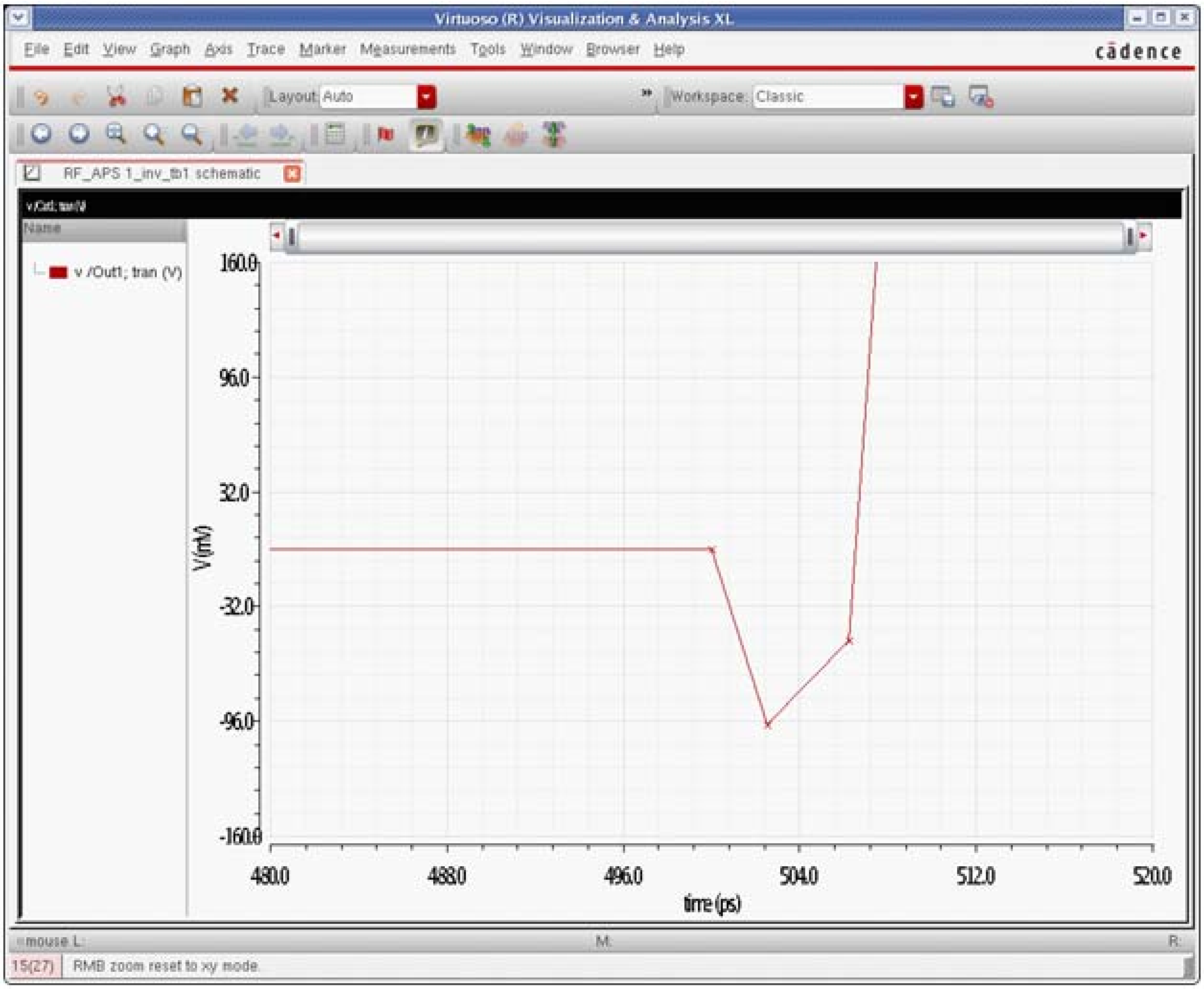
In the Gear2 method, the current is assumed to be a polynomial fit in the timestep. This requires the simulation history to determine the curve. Because of this, the simulation takes about 5% longer to run compared to trapezoidal. Because the current is more accurately represented, the method is a bit more accurate than trapezoidal. It also slightly numerically damps any ringing that the circuit might produce. This effect disappears completely when the timesteps are small, which is usually accomplished by setting `reltol` to `1e-4` or smaller. For very high Q circuits, small numerical integration errors cause large errors in the voltage in the resonator. For that reason, harmonic balance is usually preferable to shooting for high Q circuits. If you must simulate a high Q circuit using the transient analysis, always use the `traponly` numerical integration method.

Controlling Accuracy

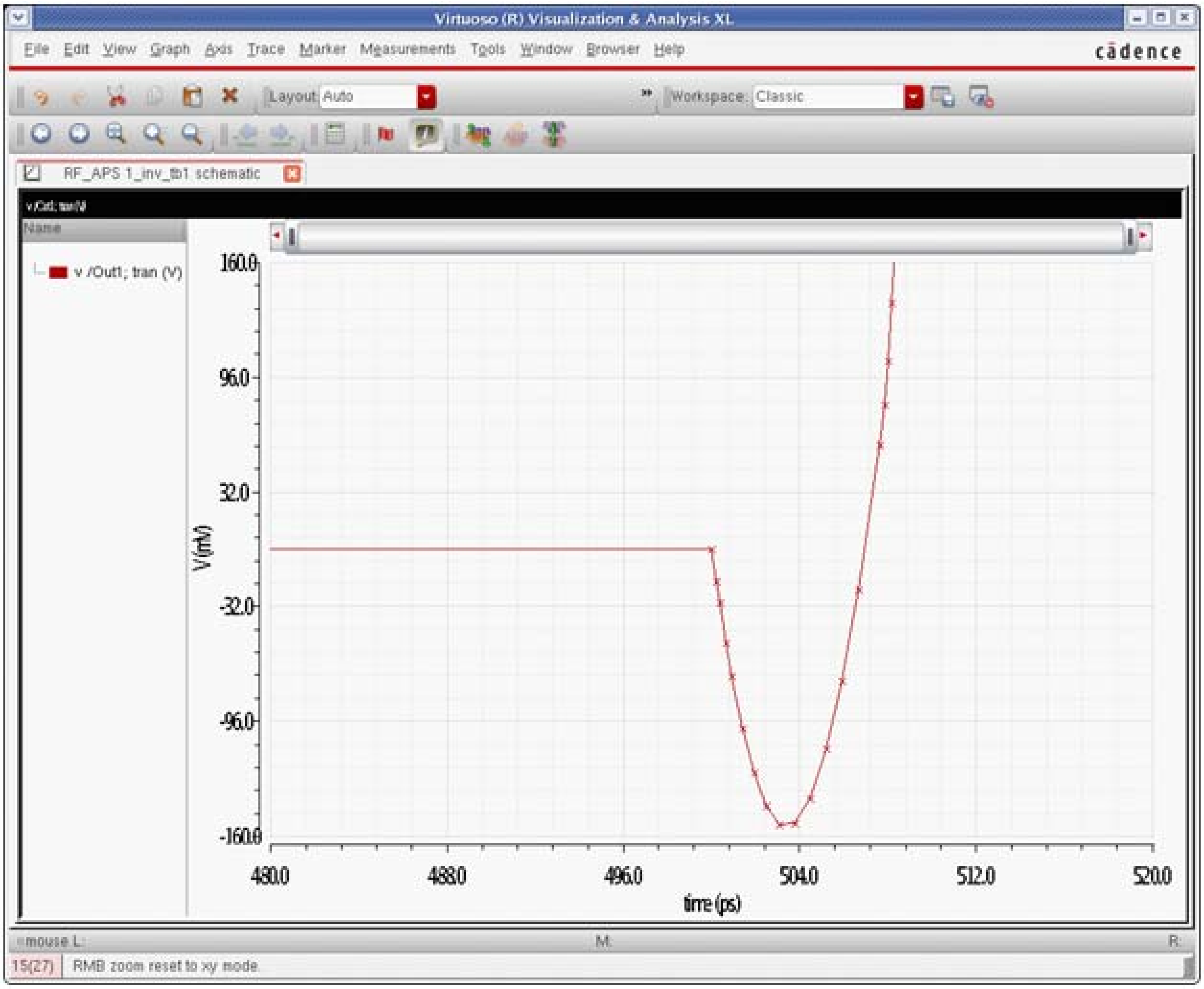
The simulator works by setting a limit for the curvature. This limit is controllable by setting `lteratio` and/or `reltol`. The product of `reltol` times `lteratio` sets the actual limit. Increasing `lteratio` sets a higher amount of curvature and leads to more errors when evaluating the numerical integration. The output with `lteratio` set to 10 times the default, is shown below.



The waveshape is substantially similar. In this case, it is OK to allow more numerical integration errors. If the details of a small portion of the waveform need to be accurate, zoom in to the time scale that is needed, as shown below.



Note the obvious piecewise linear behavior in the waveform. This is an indicator that at this time scale, the error is unacceptable. `Lteratio` and `reltol` were reduced for the following waveform.



The waveshape is substantially changed. This is because the numerical errors were significant at the time/voltage scale that we needed for the measurement.

There is still a bit of piecewise linear behavior. Reducing `reltol` and `lteratio` further will only be slightly more accurate and the runtime will be much longer. In simulation, a trade-off of accuracy versus runtime needs to be made using `reltol`, `vabstol`, and `lteratio` to get the accuracy that is needed without excessive runtimes.

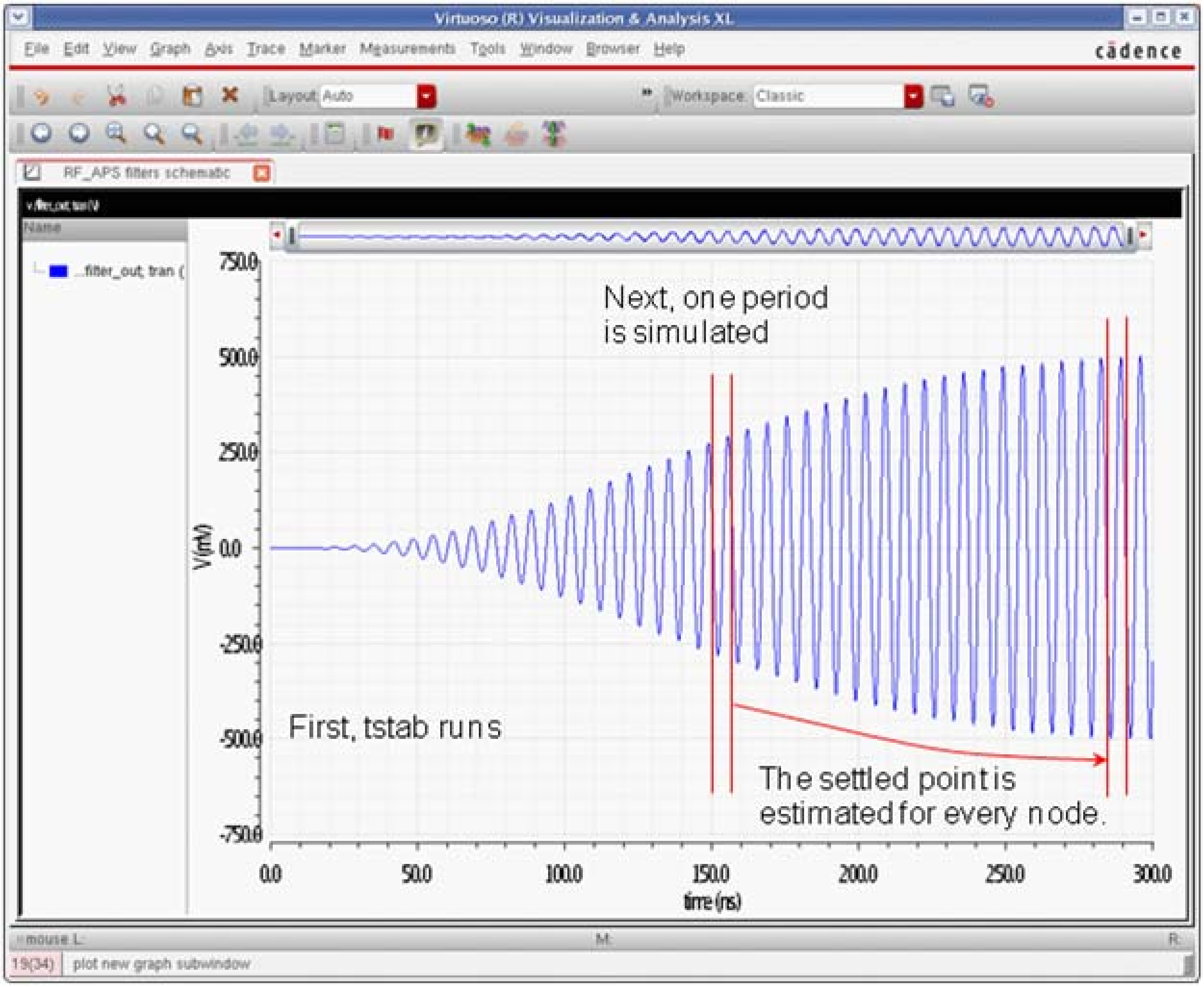
Shooting PSS

Overview

Shooting is a way of calculating the steady-state waveforms of a periodic system. Any waveshape is allowed as an input with the limitation that the waveform must be periodic. First, a transient is run for the duration set by the additional time for stabilization (`tstab`) option. Next, one period of the pss beat frequency is simulated while still using the normal transient algorithm. Then, the transient continues for one period of the waveform with the addition of saving all the solution matrices at each timestep. This is called the shooting window.

At the end of simulating this period of the input, the beginning and ending states are compared. If all the voltages and currents are equal, the steady-state solution has been achieved, and the simulation stops. In general, because the transient analysis starts from a settled DC solution at time zero, there is still settling behavior going on at the end of the first time through the shooting window. This causes the solutions to be different at the beginning and ending of the shooting window. The simulator now looks through all the solution matrices which contain information about the behavior of the circuit and the nonlinear resistors, capacitors, and inductors in the circuit, Using this information, an estimate is made of the final settled voltage for every node in the circuit. This is brought back as the starting point for the shooting window, and one period is run again. At the end, the voltages and currents are again compared to see if they are equal. This process continues until the beginning and ending states are equal. This process is called the shooting Newton method.

Graphically, this is shown below.



The estimate for every node is brought back as the starting point, and the period just after 15 nsec is simulated again. This continues until the beginning and ending voltages and currents almost match.

Note that pss shooting is really an extension to the transient analysis. All the concepts and options that were discussed before in the transient review section apply to shooting pss as well.

Oscillators

The algorithm is slightly different for oscillators. First, the tstab interval is simulated. Next, a single period of the estimated frequency from the *Choosing Analyses* form is simulated. So far, this is exactly the same as driven pss. To achieve reliable convergence, the oscillator must reach near the steady-state behavior during this phase.

Because the oscillator may also have a frequency divider at the output, additional four periods of the estimated frequency in the pss *Choosing Analyses* form are simulated. During this time, Spectre looks for a frequency divided output.

Next, one period of the lowest frequency in the circuit is determined. If the oscillator does not have a frequency divider, then one period of the oscillator output is solved for. If the oscillator has a divider, one period of the divided down frequency is solved for.

Once one period is determined, then the steady-state waveforms for all the nodes and the period are solved for so that the beginning and ending voltages and currents almost match, and the period is stable for at least two iterations. Because the period is an additional variable to be solved for compared to a driven circuit where the period is known, oscillators typically take more iterations to converge to the solution.

Oscillator Tuning Mode

Pss has the ability to tune an oscillator to a specified frequency, and then run the small-signal analyses (usually noise) that are specified. This is typically used in sweeps and in Monte Carlo simulations where the phase noise is desired at a specified frequency.

Tuning mode adjusts a parameter in the circuit to produce the set frequency target. This is done automatically when tuning mode is enabled without setting any sweep parameters or interpolation of the resulting curves. When the tuning frequency is reached, any small-signal analyses like noise are run. This allows the simulator to tune the oscillator to a specified frequency and then make a noise measurement. This is useful in Monte Carlo analysis to see how the oscillator performs with process variations. The parameter to be tuned can be a variable, temperature, or a specified device parameter. Oscillator tuning mode is also supported in the hb analysis. See Chapter 3, “Frequency Domain Analyses: Harmonic Balance,” for details.

In this mode, the target frequency to tune to is the frequency that is specified in the pss *Choosing Analyses* form as the fundamental frequency. You specify a parameter that is to be varied to achieve the desired frequency. This can be a variable, a device parameter, or temperature. When the analysis runs, the oscillator will be tuned to the desired frequency, and then all the small-signal analyses will be run.

Direct plot functions have been added to plot the tuning parameter.

Delay Time

All the signal sources have a property called delay time. During the delay time, the source stays at a constant value, and then the waveform starts. The system does not become periodic until the longest delay time has elapsed. Delay times longer than one period of the output waveform of a source should be avoided. The beginning of the tstab interval is the longest delay time in the sources of the circuit.

Piecewise Linear Sources for Power Supplies

In some cases, nonperiodic piecewise linear sources are used. This is one way to start an oscillator by giving the power supply a small bump just after the transient analysis starts. In this case, set the first point at zero seconds and 80% to 90% of the full supply voltage, and specify a second point about half period of the operating frequency at the full supply voltage. Do not specify a point at one second (or some other fairly large time) and the supply voltage. Spectre will maintain the voltage of a source if the simulation time exceeds the last point from a piecewise linear source. If there is a one second point, then something that is usually at a high frequency has to simulate for one second before the circuit becomes periodic. Tstab would start at the last point in the piecewise linear waveform.

Tstab

In the pss Choosing Analyses form, there is a property called *Additional Time for Stabilization (tstab)*. In this manual, this will be called tstab. The default for tstab is zero, but if there is a value defined for the parameter, then a normal transient will be run to a stop time equal to tstab. For driven circuits, it is seldom necessary to set tstab, but oscillators usually require it to be set.

Beat Frequency

The beat frequency is the periodicity of the system. If a single input frequency is applied, it is the frequency of the input signal. For frequencies that have inconvenient frequencies, but easy periods, a selection is available to set the period instead of the frequency.

If there are two input frequencies, the highest common multiple frequency must be set as the beat frequency. The limitation is that an integer number of cycles must be simulated for all the input frequencies. This is required because pss checks the differences between the currents and voltages at the beginning and end of the shooting window and this difference must be zero. This forces an integer number of cycles for all the sources in the circuit to be simulated.

As an example, if a 1 GHz frequency is applied to the system, the beat frequency (periodicity) of the system is 1 GHz. If 1 GHz and 1.1 GHz frequencies are applied, the beat frequency is 100 MHz. In this system, 10 cycles of 1GHz and 11 cycles of 1.1 GHz need to be simulated to get the steady-state waveform. If 1 GHz and 1.001GHz need to be simulated, the beat frequency is 1 MHz. This is not a practical pss simulation because 1000 cycles of 1 GHz and 1001 cycles of 1.001 GHz need to be simulated, and at least 3003 harmonics need to be calculated to measure the third harmonic of 1.001 GHz. The time required for the simulation and the memory requirement is excessive for this case. In this case, using qpss or hb is much faster and requires much less memory.

For oscillators, an estimate of the oscillator frequency is required in the *Beat Frequency* field. If the linear oscillator initial condition is used, the frequency range needs to be from 0.5 to 1.5 times the actual oscillation frequency. If the default oscillator initial condition is used, then the frequency must be between 0.25 and four times the actual frequency of oscillation.

Number of Harmonics

In the case of pss shooting, this is a post-processing specification that might have an effect on the minimum number of timepoints in the solution. This parameter specifies the number of harmonics to calculate in the Fourier transform that is run after the time-domain simulation to calculate the harmonics of the large-signal response. The Fourier transform is not an FFT that requires exactly spaced samples of the waveform. Rather, it is Fourier integral-based, which uses the timepoints that were actually calculated by the pss analysis as the input. Since the waveform that the Fourier-integral based algorithm sees are the actual timepoints, all the harmonics specified are accurate, even if you only specify one or two harmonics. Zero (0) is a perfectly legal value for the number of harmonics, and it just means that no Fourier transform will be run after the time-domain solution is calculated.

If the number of harmonics is raised above 10, then more timepoints will be forced in the pss solution in order to keep the accuracy of the frequency-domain results very high. 20 timepoints will be run in the period of the highest harmonic. Therefore, if maximum harmonics is set to 20, a minimum of 400 timepoints are forced in the pss analysis. If a large number of harmonics is specified, runtime and memory requirements get large. Unless you really need a lot of harmonics, do not set a large number.

Accuracy Defaults (errpreset)

Errpreset sets the default values for six options in order to tune the simulator for your needs. Use `conservative` if you need high accuracy. `Moderate` is a general-purpose setting, and should be used for most simulations. `Liberal` is not recommended.

The following chart summarizes the options and the values for driven circuits.

errpreset	Reltol[1]	Relref	Method	Lteratio[2]	Steadyratio	Maxstep
Liberal	1e-3	Sigglobal	Traponly	3.5	0.001	Period/50
Moderate	1e-3	Alllocal	Gear2only	3.5	0.001	Period/200
Conservative	1e-4	Alllocal	Gear2only	10 or 3.5	0.01	Period/200

1. Reltol: This value is used as a maximum value in the shooting window. The value used in the tstab interval is set by the option reltol.
2. Lteratio: For conservative, 10 is used in the shooting window unless the option reltol is set to 2.85e-4 or smaller. If 2.85e-4 or smaller is used, lteratio defaults to 3.5. The value used for lteratio in the tstab interval is the same as that used in the shooting window.

For example, if `errpreset` is set to `moderate` and `reltol` is set to `3e-4`, `reltol` during the tstab interval is `3e-4`, and `reltol` in the shooting window is also `3e-4` because it is smaller than `1e-3`, which is the maximum value that can be used in the shooting window. `Lteratio` is 3.5 for the tstab interval and for the shooting window.

If `errpreset` is `conservative` and `reltol` is set to `3e-4`, then `reltol` in the tstab interval is `3e-4`, and in the shooting window it is 1`e-4`. This is because the maximum value for `reltol` in the shooting window for `conservative` is `1e-4`. `Lteratio` is `10` for both the tstab interval and the shooting window.

For oscillators, the defaults are slightly different.

errpreset	Reltol[1]	Relref	Method	Lteratio[2]	Steadyratio	Maxstep
Liberal	1e-3	Sigglobal	Traponly	3.5	0.001	Period/50
Moderate	1e-4	Alllocal	Gear2only	3.5	0.001	Period/200
Conservative	1e-5	Alllocal	Gear2only	10	0.1	Period/200

1. Reltol: This value is used as a maximum value in the shooting window. The value used in the tstab interval is set by the option reltol.
2. Lteratio: For conservative, 10 is used in the shooting window unless the option reltol is set to 2.85e-4 or smaller. If 2.85e-4 or smaller is used, lteratio defaults to 3.5. The value used in the tstab interval is the same as that used in the shooting window.

PSS Shooting Convergence

As discussed earlier, the waveform is iteratively solved for during the shooting window. This means that exact solutions are never obtained because there is another iteration that will produce a closer answer. The iterations must stop when the solution is accurate enough. This is set using the convergence options `reltol`, `vabstol`, `iabstol`, `lteratio`, `errpreset`, and an option called `steadyratio`. The iterations stop when all the voltages in all the nodes and all the currents in all the branches match at the beginning and end within the following tolerance:

```
(Convergence Criteria) * Lteratio * Steadyratio
```

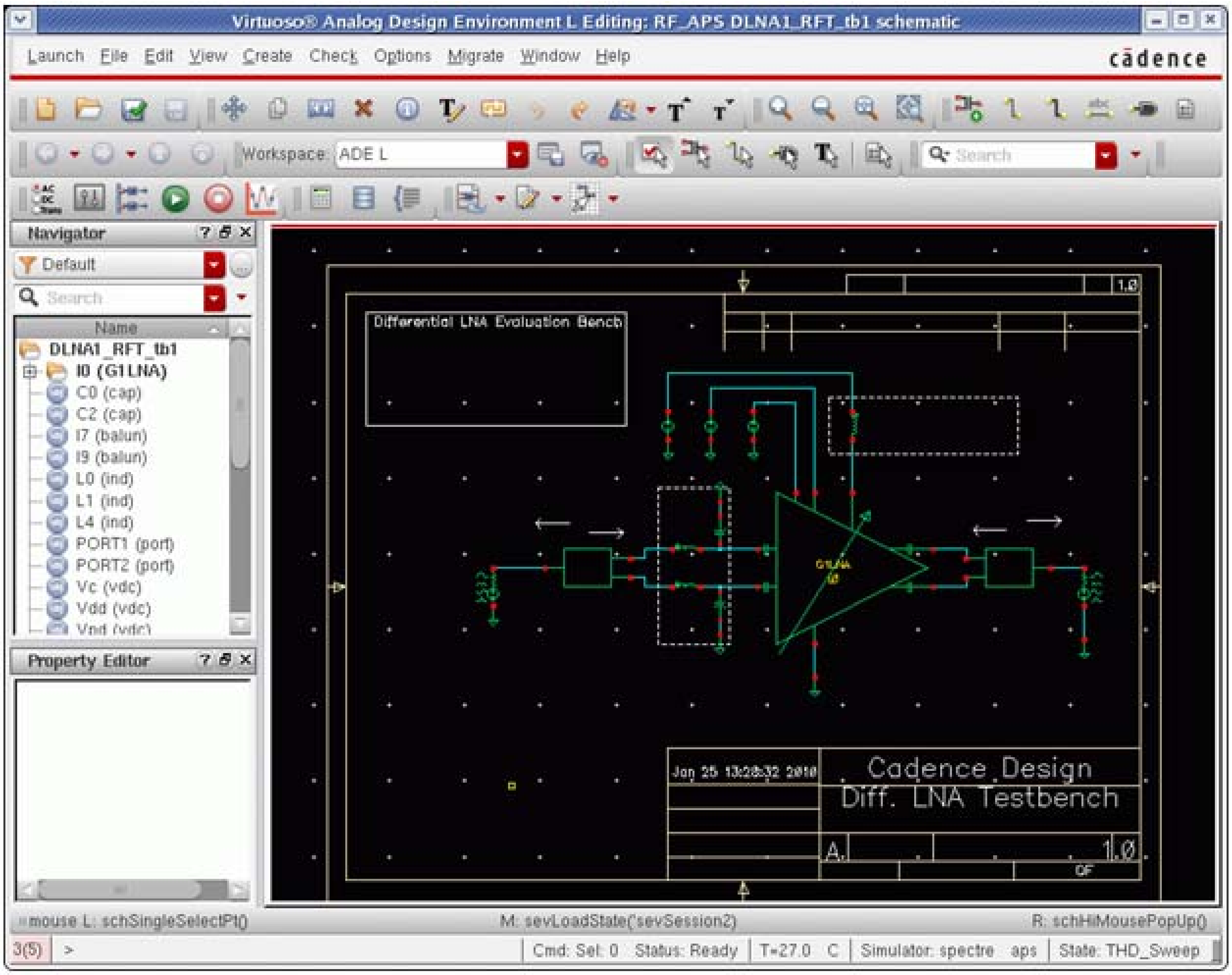
The convergence criteria depend on the setting of `reltol`, `vabstol`, `iabstol`, `errpreset`, and the setting of `relref`. `Lteratio` defaults to `3.5` or `10`, depending on the setting of `reltol` and `errpreset`.

`Steadyratio` depends on the setting of `errpreset`. For more details regarding this option see Steadyratio (Shooting Only) on page 610.

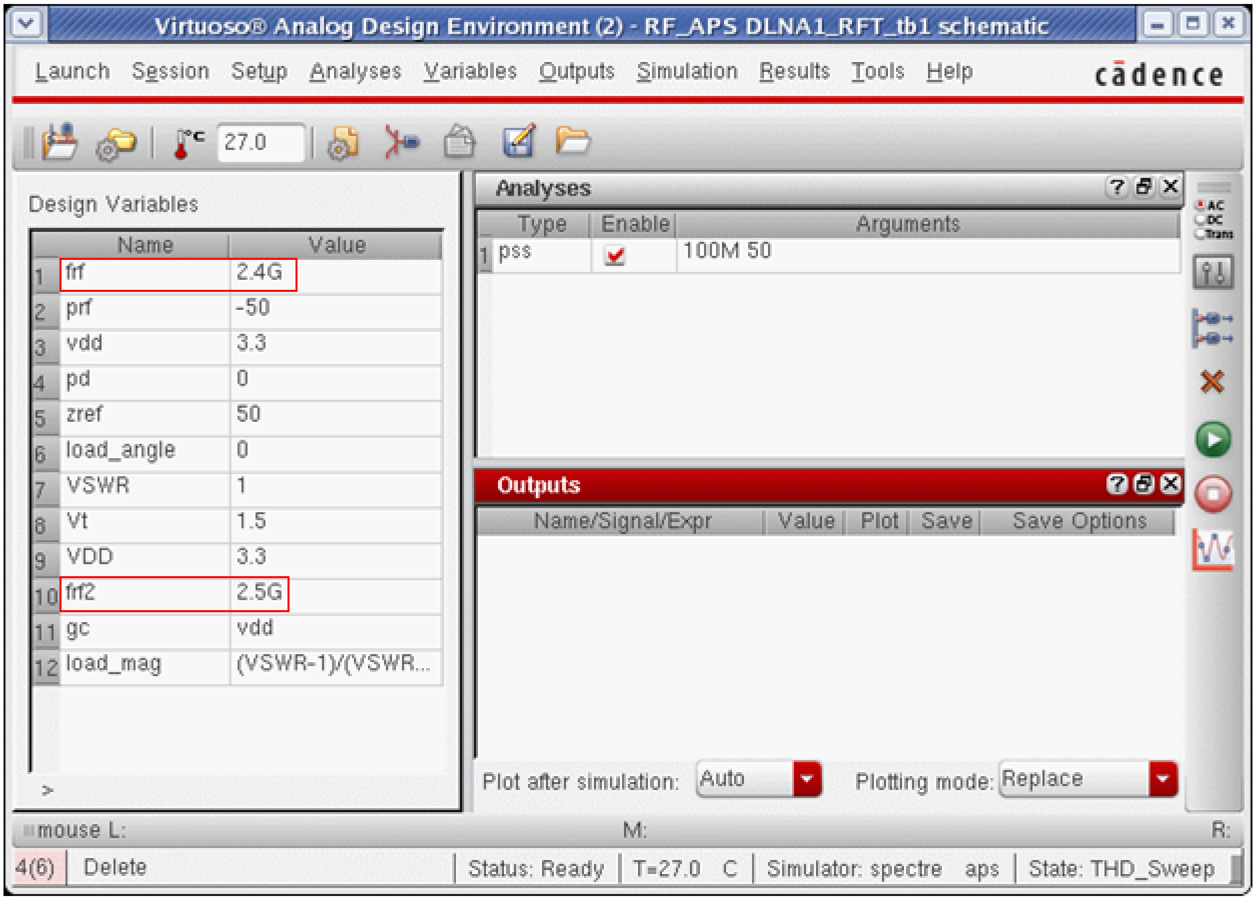
ADE Implementation and Numerical Noise Floor

Next, we will discuss how SpectreRF is implemented into ADE and also provide information about how the simulation accuracy is affected by setting a number of simulator options.

A transistor-level LNA is shown below.

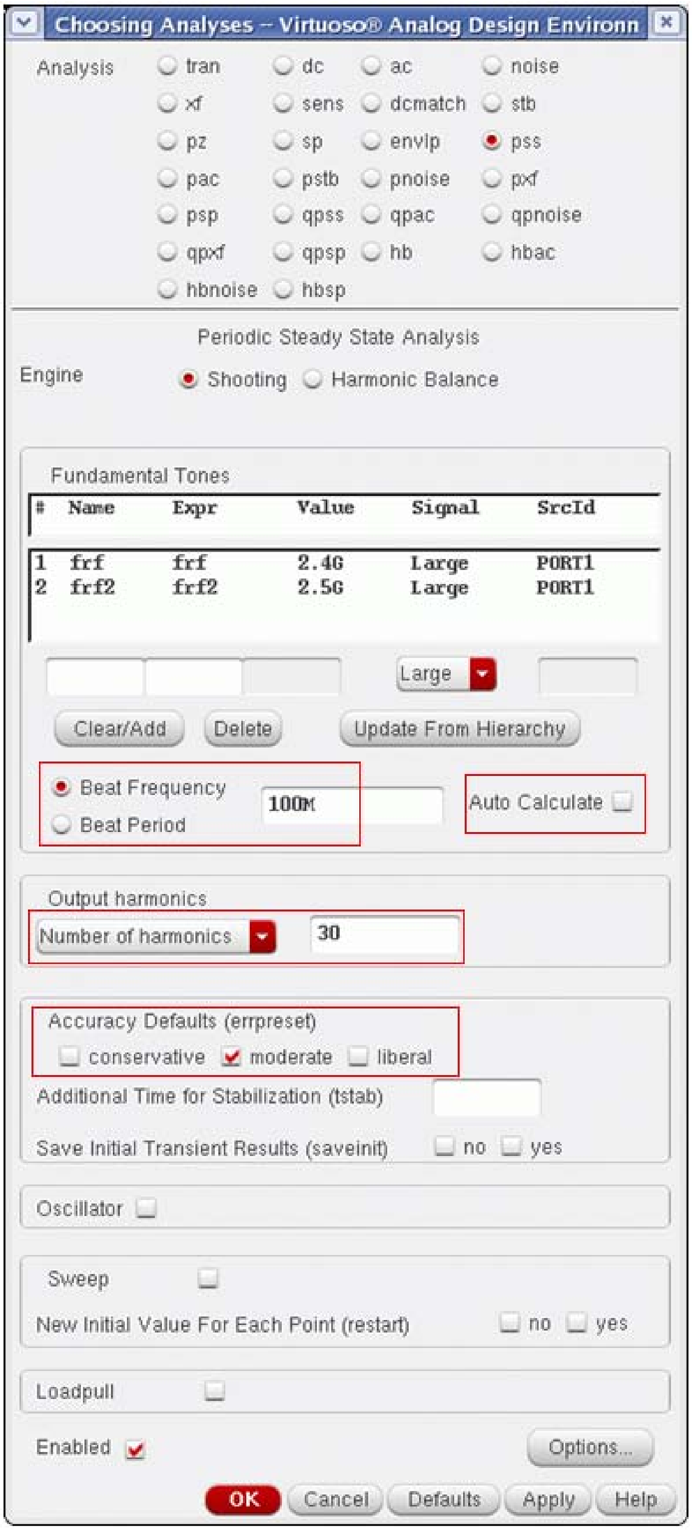


The input frequencies for the transistor-level LNA are set to 2.4 GHz and 2.5 GHz, as shown in the ADE window below.



Note that this setup is approaching the limit of what pss can do because 25 periods of the higher input frequency need to be simulated. If the frequencies were closer together, qpss or hb would be much more efficient.

The pss *Choosing Analyses* form is shown below.



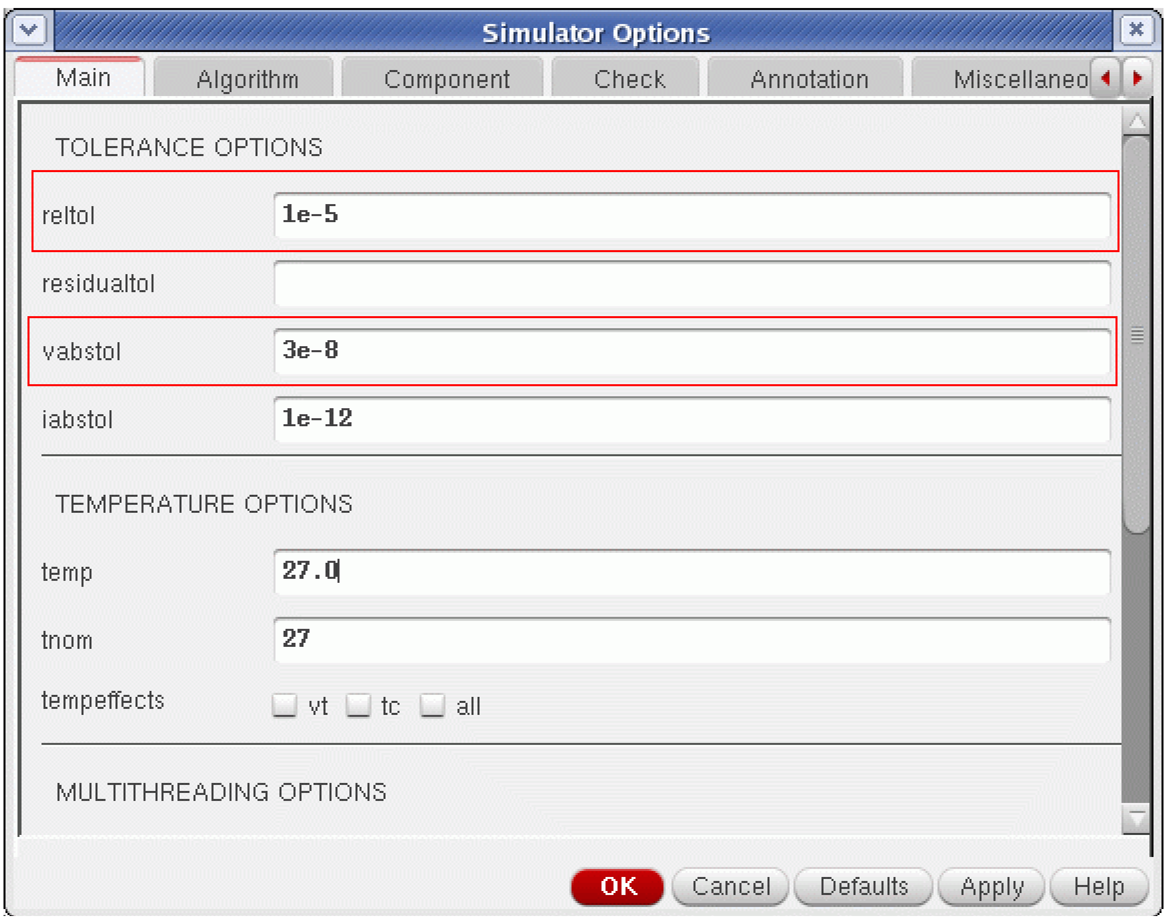
All the sources in the circuit are read in to the *Fundamental Tones* table at the top of the *Choosing Analyses* form. When the *Auto Calculate* option is selected, ADE calculates the periodicity of the circuit and displays it in the grayed-out *Beat Frequency* field.

Note: When *Auto Calculate* is selected, manual editing of the *Beat Frequency* field is not allowed.

Accuracy and Settings and Trade-Offs

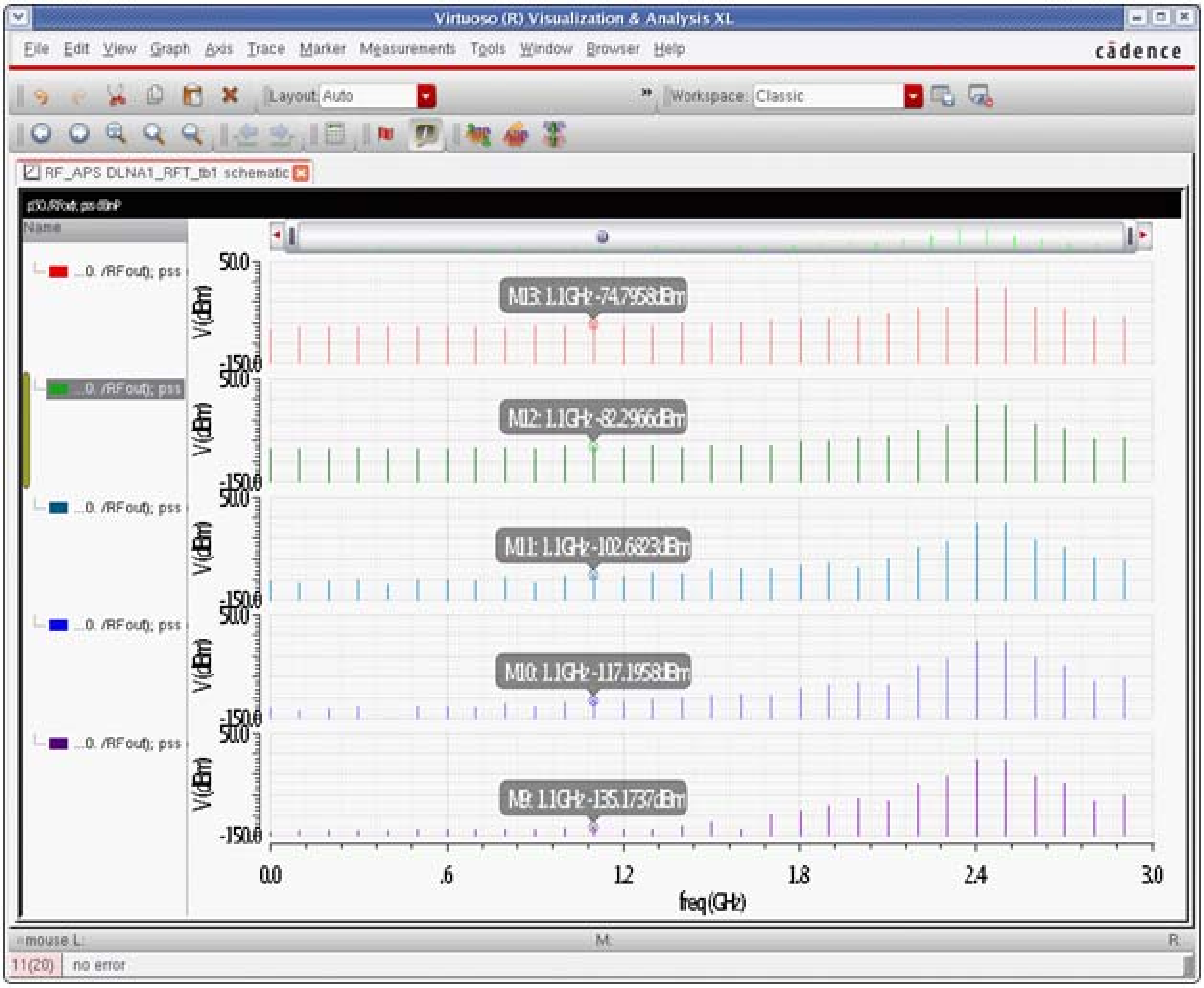
A comparison is made for several settings. Moderate and conservative settings are used.

After that, values for *reltol* and *vabstol* are specified, as shown below in the *Simulator Options* form. This form can be opened by selecting *Simulation - Options - Analog in the ADE*.



First, `reltol=1e-5` and `vabstol=3e-8` are used, and then `reltol=1e-6` and `vabstol=1e-8` are used. Finally, a lot of timepoints are forced by setting the maximum harmonics to 1000. For this test, `reltol` and `vabstol` are kept at `1e-6` and `1e-8`.

The spectral result from all the simulations is shown below.



A marker is placed at 1.1 GHz in order to measure the numerical noise floor of the simulation.

The top is with `errpreset` set to `moderate`.

The second is `conservative`.

The third is `reltol=1e-5` and `vabstol=3e-8`.

The fourth is `reltol=1e-6` and `vabstol=1e-8`.

The final one is with 1000 harmonics set. The display is scaled to show only the first 30 harmonics.

Note that the numerical noise floor drops with the steps that are taken to increase accuracy. The runtime and memory requirement also increases for each successive level of accuracy.

This is a good general series of steps to improve the accuracy of a shooting simulation. The basic process is to set up and run the simulation, make the accuracy better, and run the simulation again. If the answer does not change, then the original setting is good enough. If the answer changes, increase the accuracy again. Continue until you find the setting of the options that cause the answer to be accurate without excessive runtime.

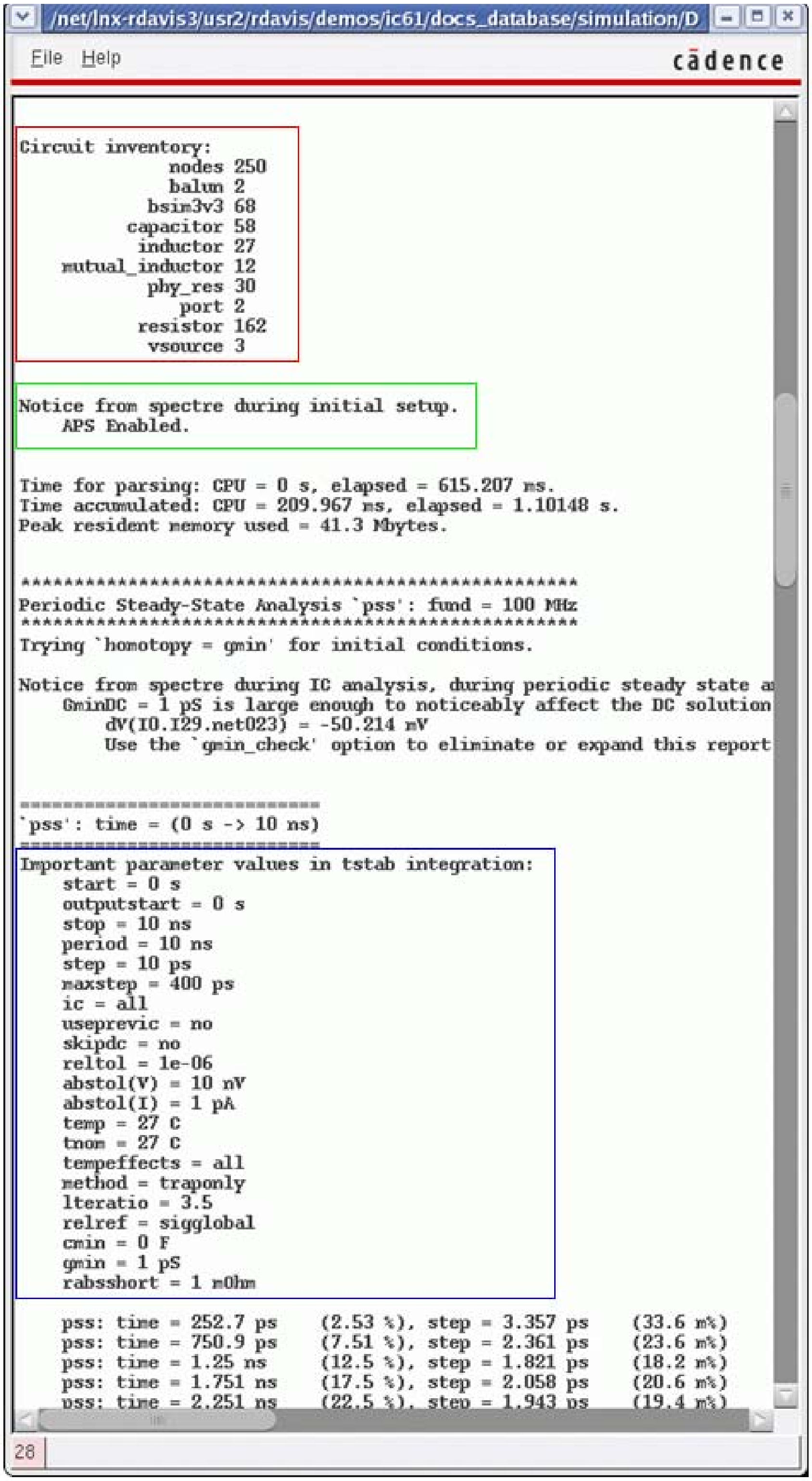
Note that there is a practical minimum numerical noise floor at about -135dBm. Although it might be possible to lower the noise floor a bit more by taking even more timepoints in the simulation, the time and memory requirements usually are not worth the lower noise floor. Still, -135dBm is a pretty good numerical noise floor of about 3e-8 volts RMS.

Additionally, you might try using the *highorder* option to improve accuracy. This is shown in the pss options section. This option requires significantly longer to run but produces a more accurate result.

Which Engine Should be Selected?

Conceptually, shooting and harmonic balance produce the same answer as long as enough harmonics and/or a high enough oversample factor are used in hb and the accuracy of shooting is controlled well enough for your requirement. In general, if the accurate waveform is desired for a system with square waves in the solution, then shooting should be selected. If the system is very nonlinear like a sample-and-hold or a switched-capacitor filter, shooting should always be used. If the waveforms are nearly sinusoidal, then harmonic balance should be selected. If it is something in between, then try both engines and use the one that produces the fastest runtime. Note that even for square wave systems, harmonic balance can be used effectively as long as the time-domain waveform is not desired, and oversample factor is set to at least four. For more details on harmonic balance, see Chapter 3, “Frequency Domain Analyses: Harmonic Balance”.

Important Outputs in the Spectre.out File

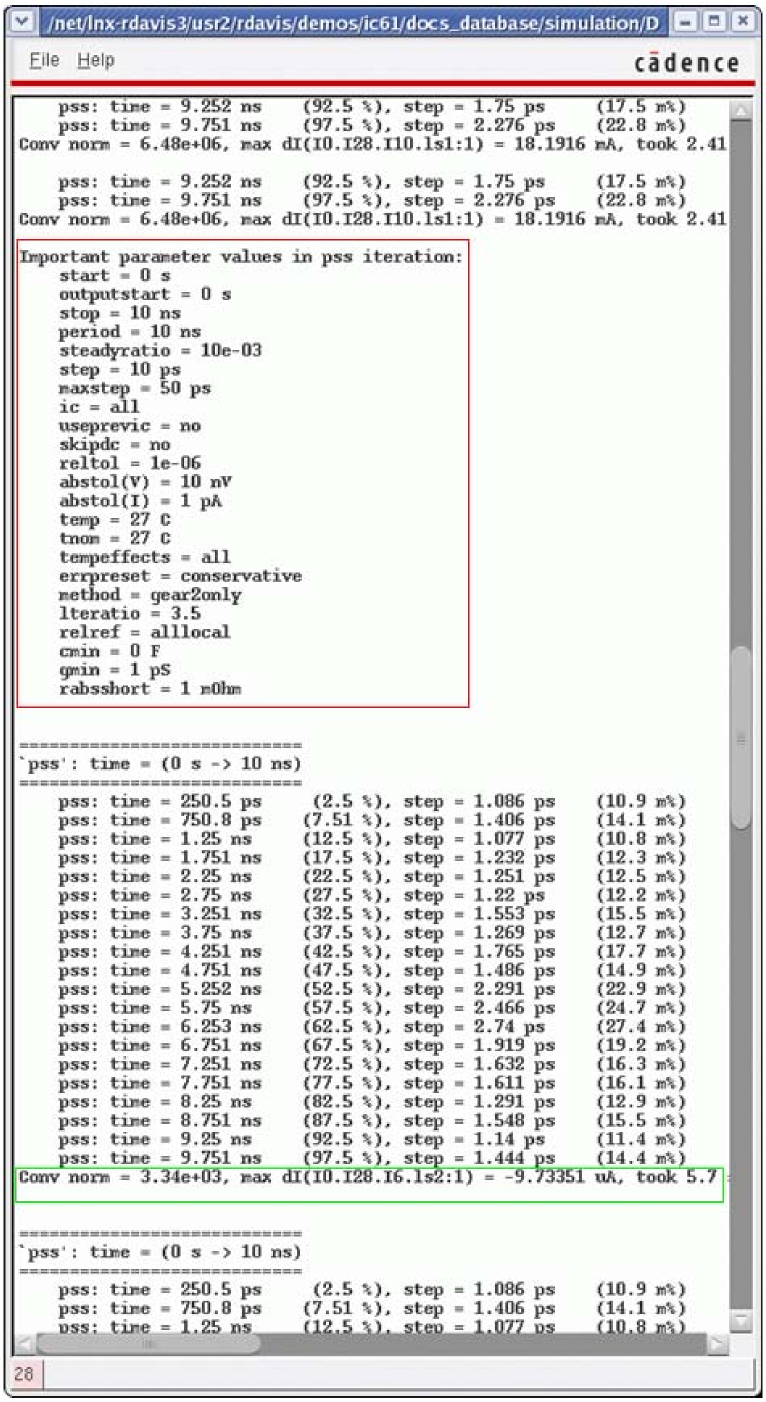


```
/net/lnx-rdavis3/usr2/rdavis/demos/ic61/docs_database/simulation/D
File  Help                                                 cādence

Circuit inventory:
              nodes 250
              balun 2
            bsim3v3 68
          capacitor 58
           inductor 27
    mutual_inductor 12
            phy_res 30
               port 2
           resistor 162
            vsource 3

Notice from spectre during initial setup.
    APS Enabled.

Time for parsing: CPU = 0 s, elapsed = 615.207 ms.
Time accumulated: CPU = 209.967 ms, elapsed = 1.10148 s.
Peak resident memory used = 41.3 Mbytes.

****************************************************
Periodic Steady-State Analysis `pss': fund = 100 MHz
****************************************************
Trying `homotopy = gmin' for initial conditions.

Notice from spectre during IC analysis, during periodic steady state a
    GminDC = 1 pS is large enough to noticeably affect the DC solution
        dV(I0.I29.net023) = -50.214 mV
        Use the `gmin_check' option to eliminate or expand this report

============================
`pss': time = (0 s -> 10 ns)
============================
Important parameter values in tstab integration:
    start = 0 s
    outputstart = 0 s
    stop = 10 ns
    period = 10 ns
    step = 10 ps
    maxstep = 400 ps
    ic = all
    useprevic = no
    skipdc = no
    reltol = 1e-06
    abstol(V) = 10 nV
    abstol(I) = 1 pA
    temp = 27 C
    tnom = 27 C
    tempeffects = all
    method = traponly
    lteratio = 3.5
    relref = sigglobal
    cmin = 0 F
    gmin = 1 pS
    rabsshort = 1 mOhm

    pss: time = 252.7 ps    (2.53 %), step = 3.357 ps    (33.6 m%)
    pss: time = 750.9 ps    (7.51 %), step = 2.361 ps    (23.6 m%)
    pss: time = 1.25 ns     (12.5 %), step = 1.821 ps    (18.2 m%)
    pss: time = 1.751 ns    (17.5 %), step = 2.058 ps    (20.6 m%)
    pss: time = 2.251 ns    (22.5 %), step = 1.943 ps    (19.4 m%)
28
```

The circuit inventory (highlighted in red above) lists the contents of the circuit.

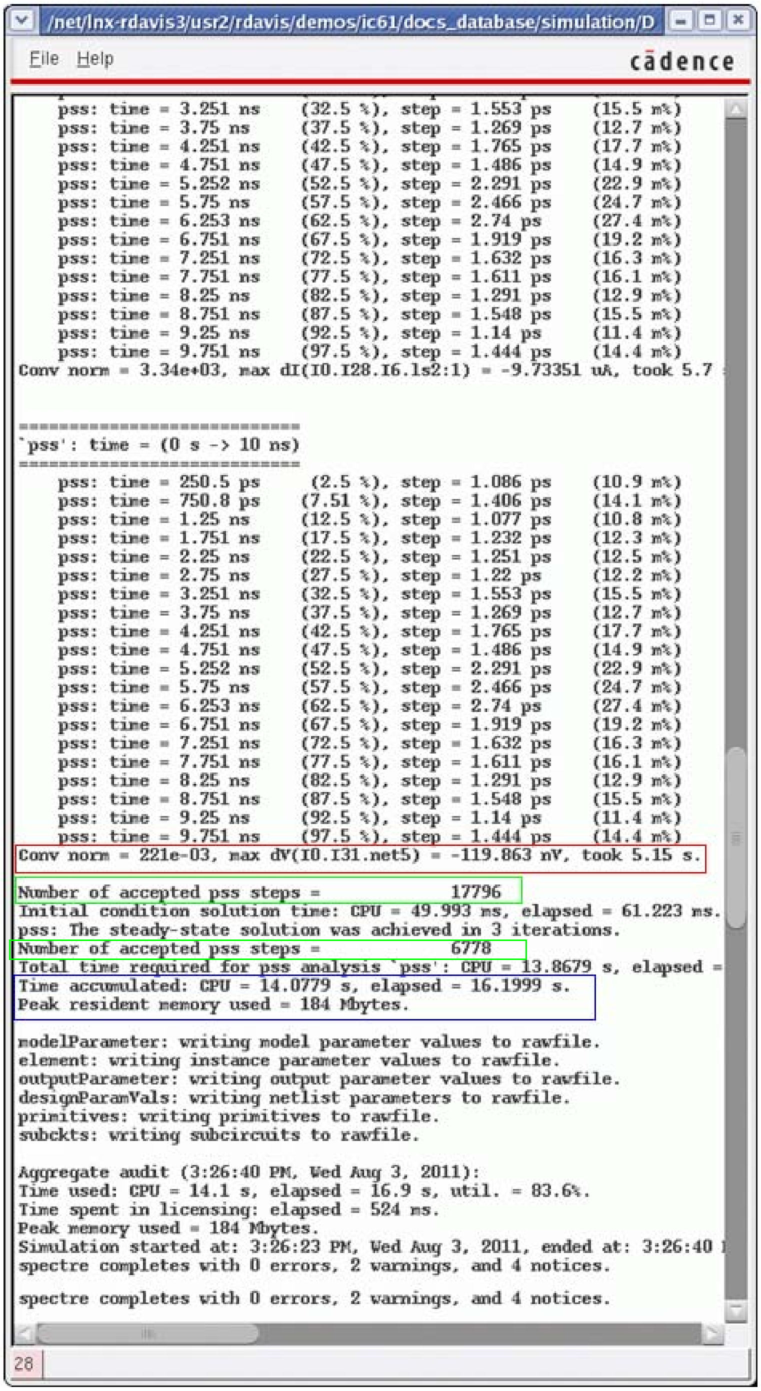
If APS is used, it is noted in the section highlighted in green above.

The section highlighted in blue above contains the option settings for tstab simulation.



The section highlighted in red above lists the options for shooting iterations.

When the `Conv_Norm` number gets less than one, the iterations stop. In this case, the error is 3.34e-3 times the maximum allowed, and the largest mismatch is I0.I28.I6.Is2 terminal 1, and the mismatch is 9.73351 microamps (highlighted in green above).



When the Conv_Norm is less than one, the iterations stop (highlighted in red above).

The total number of timesteps in the solution is listed (highlighted in green above). The first listing is the number of timesteps in the tstab interval. The second listing is the number of timepoints in the shooting interval.

CPU time, elapsed (clock) time, and memory statistics are listed (highlighted in blue above). When APS is used with multi-threading, the CPU time is generally larger than the elapsed time. CPU time totals the usage of all CPUs.

Guidelines for Setting up Oscillators for Simulation

Beat frequency

Enter a frequency that is between 0.25 * actual frequency to 4 * actual frequency. As long as the specified frequency is within this range, the solution should be possible.

Tstab

Tstab is the time interval that is used to start the oscillator and allows the oscillations to build up to a near steady-state amplitude.

Different ways to start the oscillator

The first thing to do when simulating oscillators is to reliably start the oscillator. Different methods are used for different types of oscillators. These are:

- **Moderate Q Feedback Oscillators**: If the oscillator is a feedback oscillator, then the linear oscillator initial condition (oscic) is highly recommended. When this is selected, the beat frequency needs to be within 0.5 * actual frequency to 1.5 * actual frequency. This performs a variation of the linear stability analysis after the DC analysis for the first timepoint in the tstab interval. It provides a very good estimate of the oscillating frequency and amplitude at every node and this estimate is used at the beginning of the tstab interval. This allows the duration of the tstab period to be about 5 to 10 periods of the oscillator frequency for normal Q circuits. In many cases, the tstab time can be set to zero. Doing this, causes the pss analysis to start with the estimated frequency and amplitude from the linear oscic.

  Another way to start the oscillator is to use a piecewise linear source as the power supply. At time zero, the voltage is about 80% to 90% of nominal. The second timepoint should be about one half the period of the oscillator and the full power supply voltage. The bump in voltage causes a voltage transient that starts things up. For differential oscillators, do this on one side of the differential system.

A third way is to add a current pulse to a point in the circuit that causes the oscillation to start using an ipulse component from *analogLib*. Specify a single pulse that is large enough to get things started. This is accomplished by adding an ipulse component from *analogLib* and specifying the zero value to zero, the one value to be about a milliamp, the rise and fall times to about one tenth of a period, and the pulse width to about three tenths of a period. Leave the period field blank to indicate that only a single pulse should be generated. The ipulse is an ideal open when the current is zero, so it does not load the circuit.

For all the different methods other than the linear oscic, tstab must be long enough for the oscillator to reach near steady-state operation. For most oscillators, this is about 100 periods of the oscillation frequency.

- **High Q Oscillators**: For high Q oscillators, harmonic balance is recommended. The reason for this is that in harmonic balance, the motional equivalent circuit (electrical representation of the crystal) goes directly into the frequency domain representation of the circuit. In shooting, there is always a small numerical integration error in the solution. For high Q circuits, this error causes uncertainty in the pss result that can cause the pss analysis to become inaccurate.
- **Ring Oscillators**: For ring oscillators, start the oscillation by specifying initial conditions at the output of one stage of the ring. Force the output of this stage to low or high with the initial conditions. This will force the oscillator into a known state at time zero, and then at the beginning of the transient, the node(s) will begin the switch to the other state, thereby starting the oscillator. Tstab should be set to about 1 or 2 periods.

For all the different methods other than the linear oscic, tstab must be long enough for the oscillator to reach near steady-state operation. For most oscillators, this is about 100 periods of the oscillation frequency. When using linear oscic, this can usually be shortened to about 10 periods or even zero.

As usual for any pss analysis, first the tstab interval is simulated. Then one period of the frequency supplied in the pss *Choosing Analyses* form is simulated. For oscillators, an additional four periods of the specified frequency in the pss setup is simulated. In the last interval, Spectre checks for possible divided-down frequencies. If these frequencies are found, Spectre adjusts the frequency/period to correspond to the new lower frequency and tries to solve for that lower frequency.

Saveinit- A way to save the startup waveform

For many oscillators, it is desirable to see the actual startup waveform in order to verify that the startup is robust. To see the startup waveform, select *yes* for the *Save Initial Transient Results (saveinit)* property in the *Choosing Analyses* form. This will write the initial waveforms to the output data for viewing.

Oscillator Node- Used to estimate the frequency for the first pss iteration

When linear oscic is used, the oscillator node must be a node that is inside the feedback system for the oscillator. This is used to estimate the initial oscillating frequency. If the oscillator is differential, specify a pair of nodes in a symmetrical location in the oscillator.

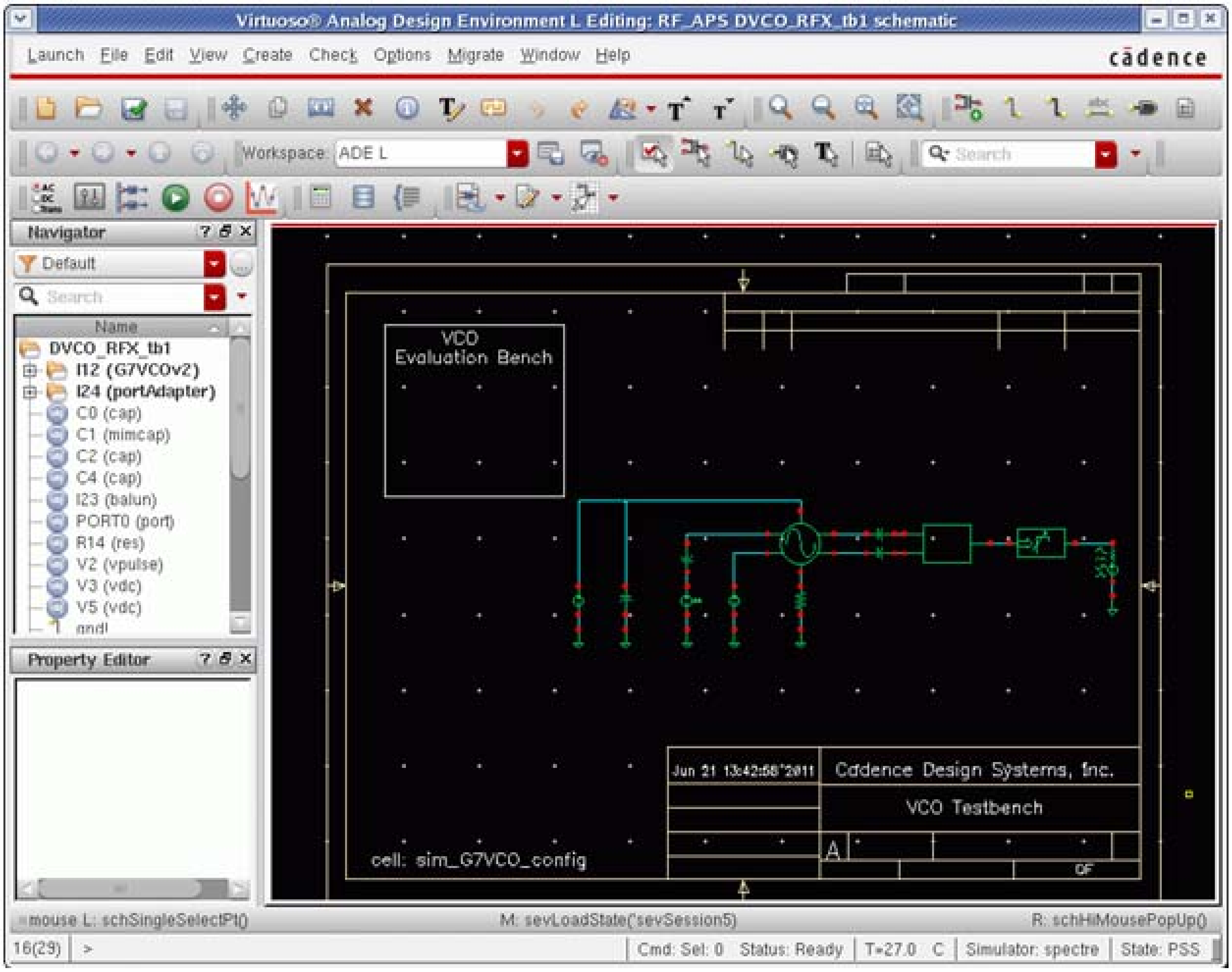
Regardless of the setting for oscic, the oscillator node is used at the end of the tstab interval to determine an initial estimate of the actual oscillation frequency.

If the amplitude on this node drops to less than 1mV peak, an error is displayed indicating that the amplitude of the oscillator node is too small to properly estimate the frequency, and the simulation stops. If the frequency gets to 1000 * frequency specified, an error is displayed indicating that the frequency is too high to be reliable.

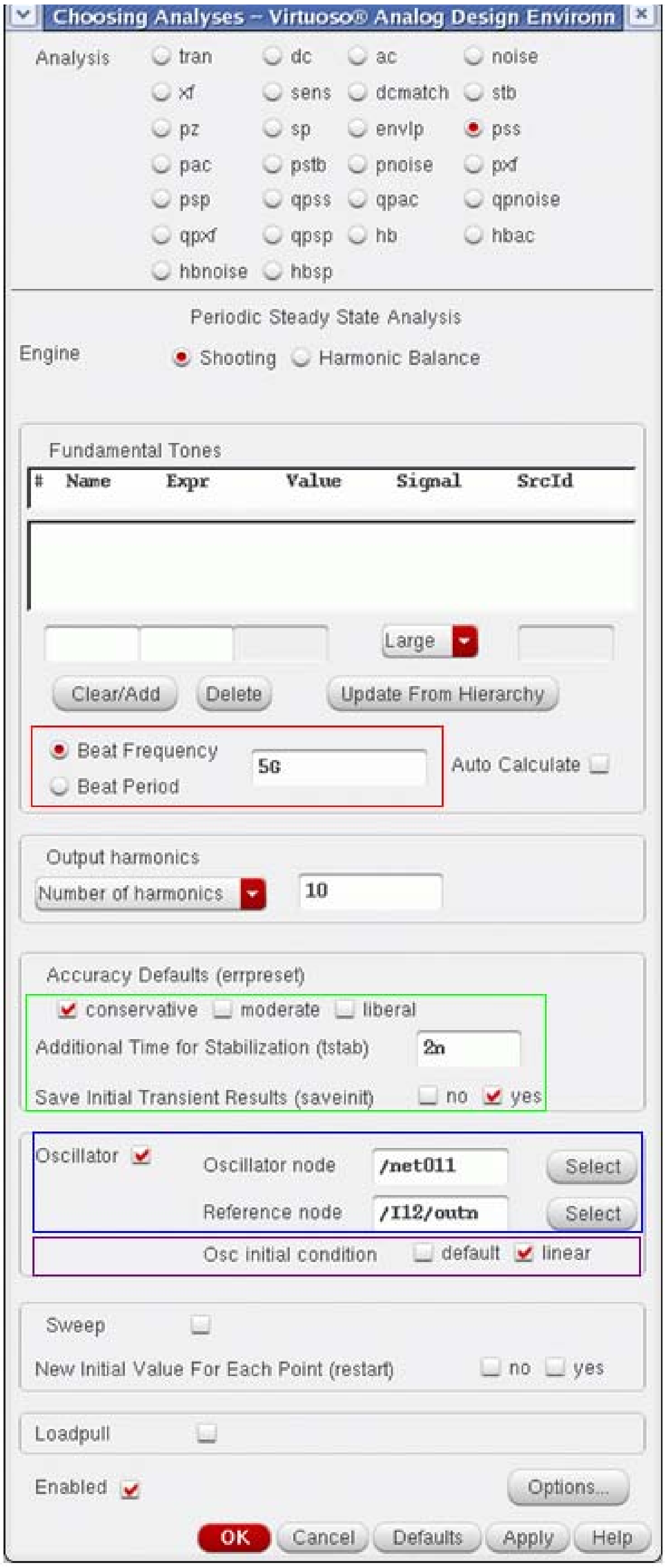
ADE Implementation

Below is an example ADE setup for an oscillator circuit. Let us set up a PSS analysis, run the simulation, and plot the output waveform.

Consider an oscillator shown below.



1. Select *Analyses - Choose* to open the *Choosing Analyses* form.



2. Specify an estimate of the oscillation frequency (highlighted in red above). If you use *linear* oscic, the estimate should be between 0.5 * actual oscillating frequency to 1.5 * actual oscillating frequency. Otherwise, it can be between 0.25 and 4 times the actual operating frequency.

3. Select *Conservative* (recommended) for oscillators (highlighted in green above).

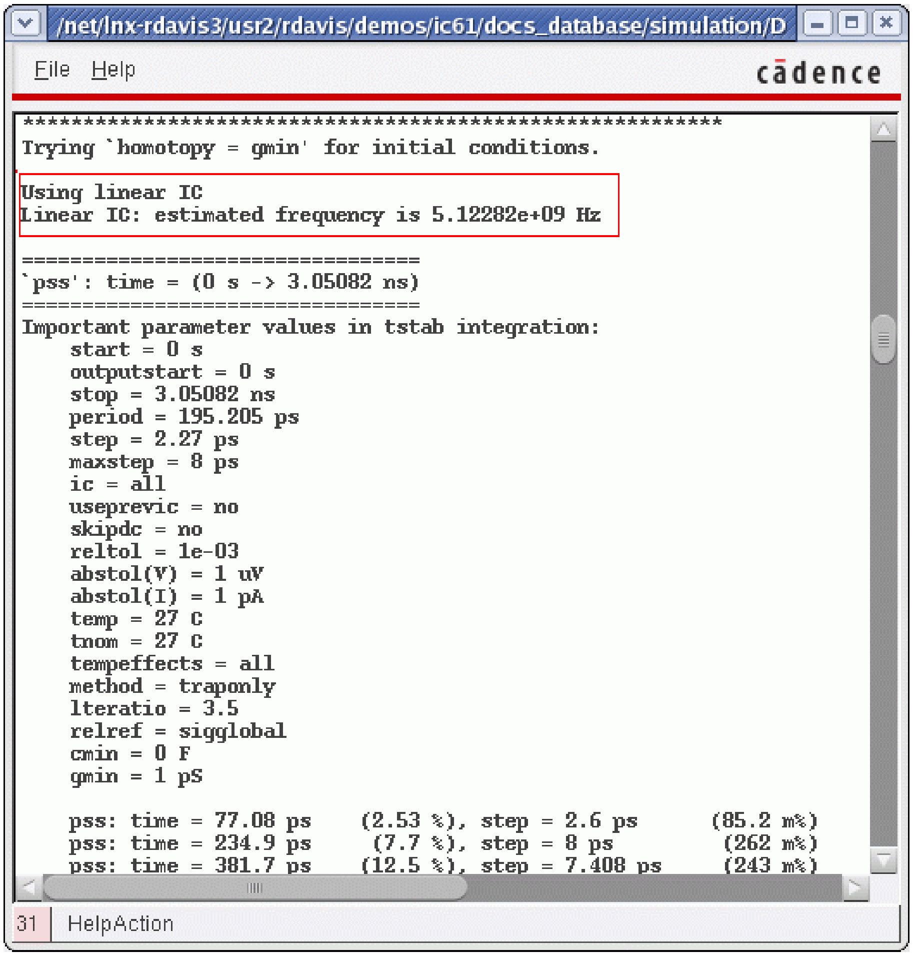
4. The oscillator node should be inside the feedback system if *linear* oscic is used (highlighted in blue above). If the *default* oscic is used, the node just needs a signal on it.

5. *Select linear* oscic (recommended) for feedback oscillators (highlighted in purple above).

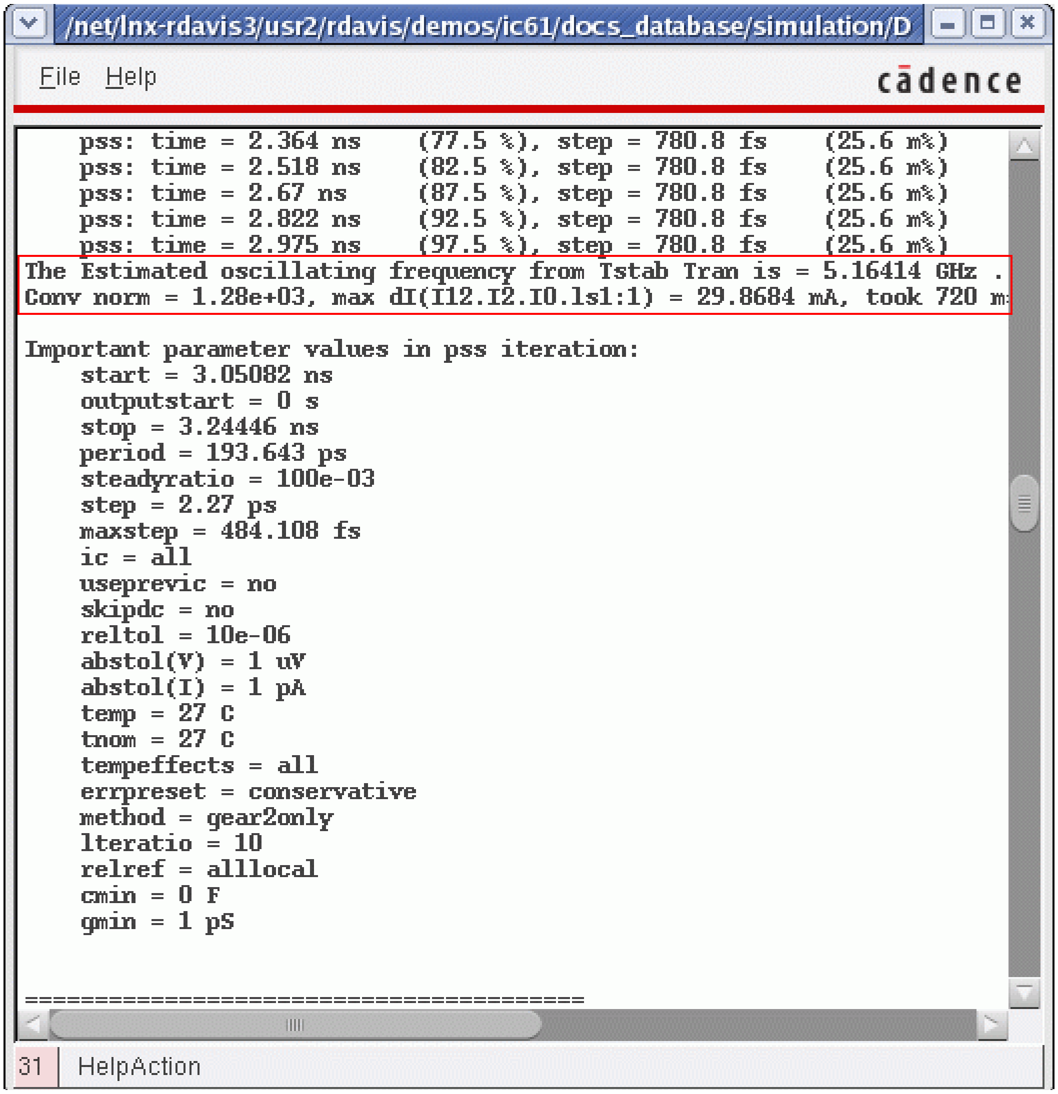
   default oscic starts from the time zero timepoint in the tstab interval.

Note: For a driven circuit, the process is the same except that the oscillator node and the oscillator initial condition fields are not shown.

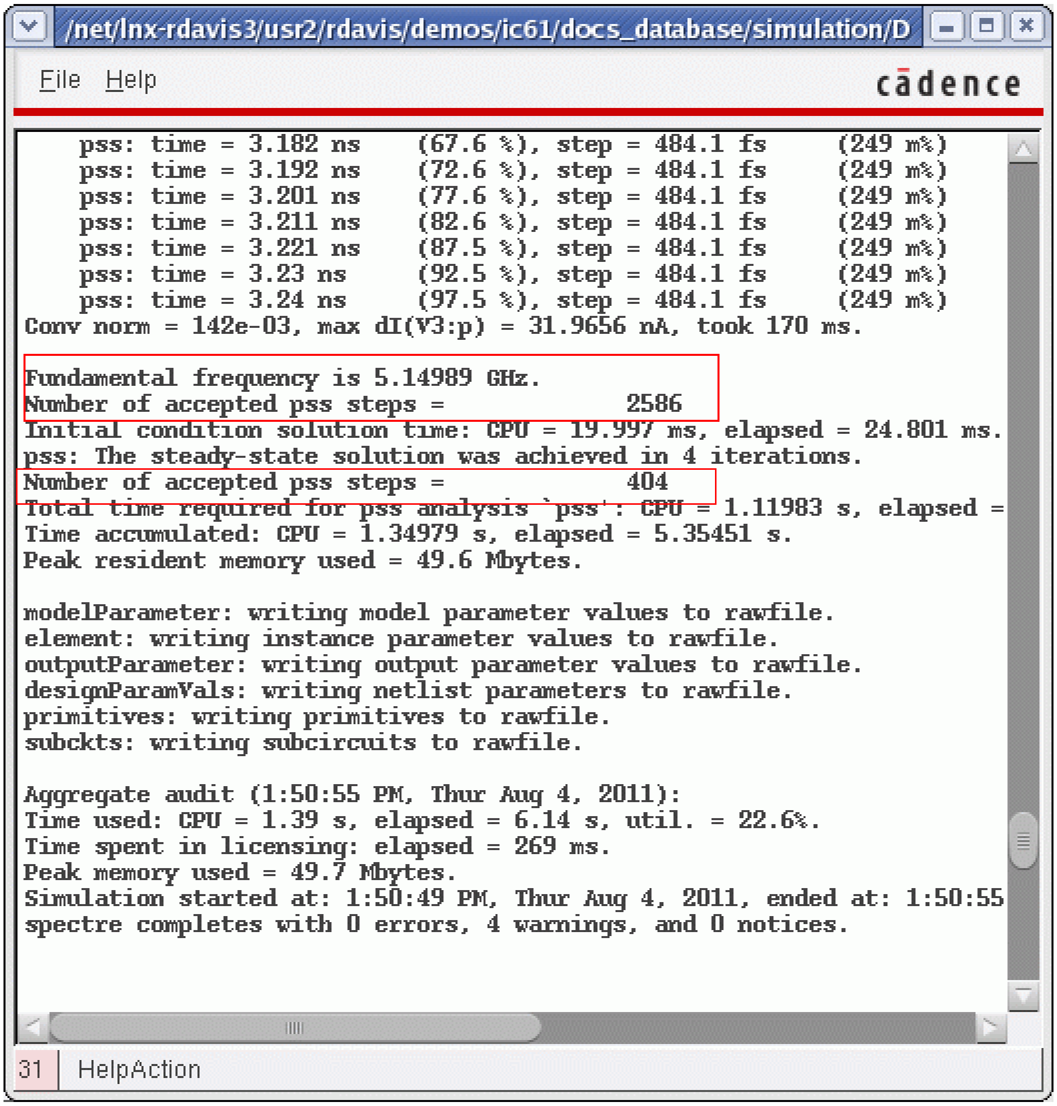
Spectre Output File (Oscillators)



The estimate of the oscillation frequency from the linear oscic (highlighted in red above) is shown first.



The frequency from the end of the tstab interval (highlighted in red above) is displayed next.



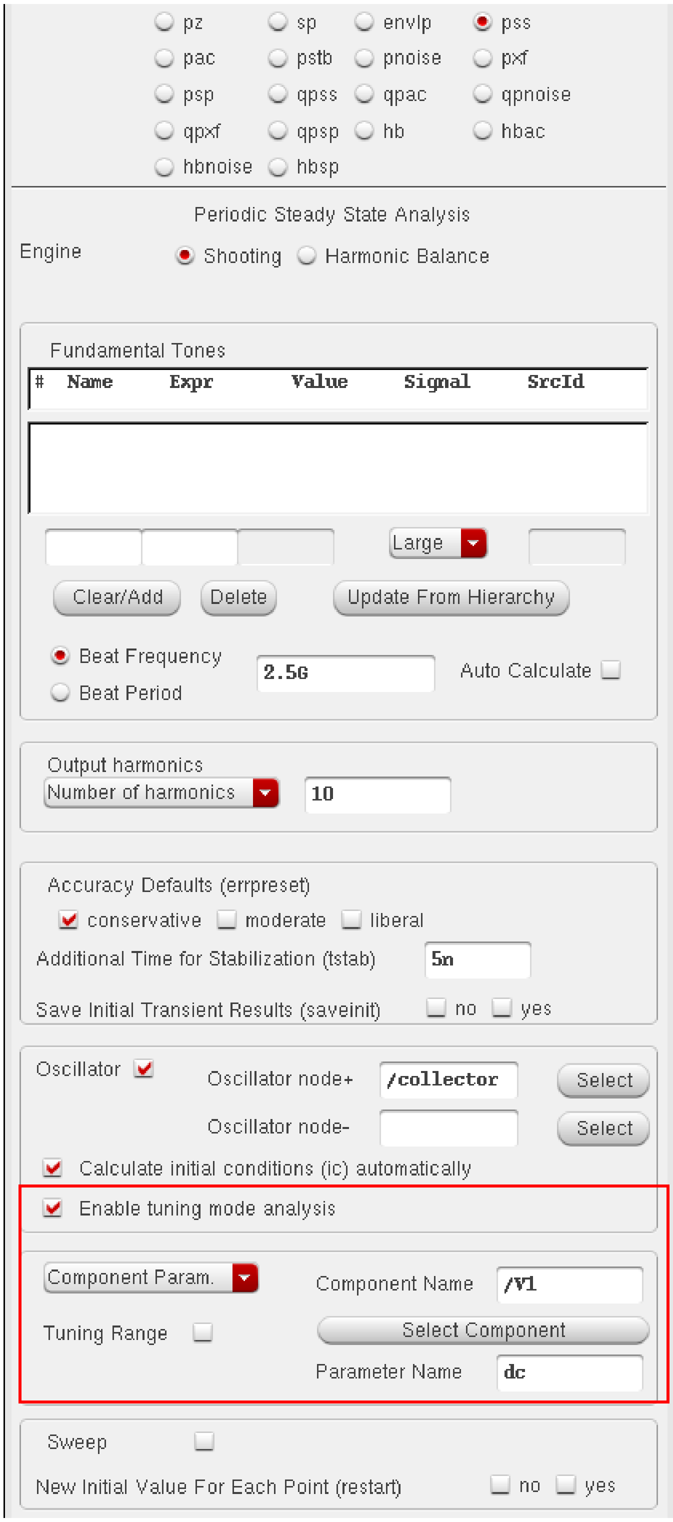
The solution frequency and the number of timesteps (highlighted in red above) are shown. The first listing of *accepted pss timesteps* is the number of timesteps in the tstab interval. The second number is the number in the shooting interval.

Oscillator Tuning Mode

Oscillator tuning mode is provided to tune the oscillator to a desired frequency and then run the selected small-signal analyses. This is useful for sweeps and for Monte Carlo simulations. To run an oscillator tuning analysis, perform the following:

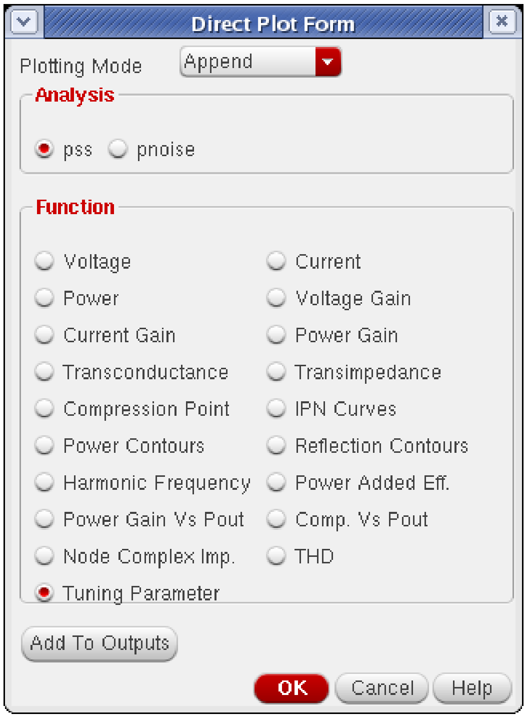
1. Open an oscillator circuit, and start ADE. Open the *Choosing Analyses* form, and select *pss* analysis.
2. Fill out the form as usual, except for the following:
   - **a.** In the *Fundamental Frequency* field, type the desired frequency target.
   - **b.** Click the check box next to *Enable tuning mode analysis*.

c. Select whether you want to tune a variable, temperature, or a component parameter. the example below shows a component parameter.



d. Enable the desired small-signal analyses.

e. Run the analyses. When the simulation completes, open the *Direct Plot Form* from ADE.

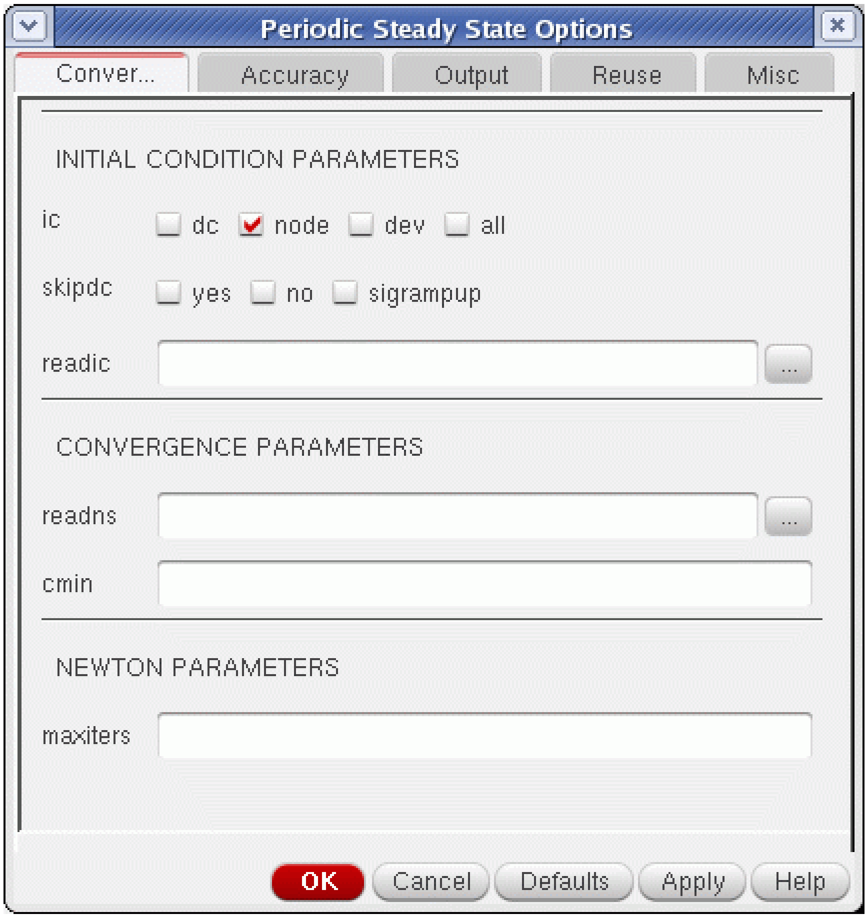


PSS Options

The options are divided into five tabs: *Convergence*, *Accuracy*, *Output*, *Reuse*, and *Misc* (miscellaneous).

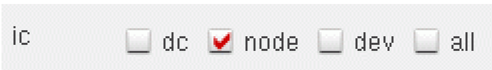
The options are shown in the order that they occur in the options form in ADE.

Convergence tab



Ic (Shooting and Harmonic Balance)

Initial conditions can be specified graphically by selecting *Simulation - Convergence Aids - Initial Condition* in the ADE. Initial conditions can also be specified from a file using the *readic* property. Capacitors and inductors have the initial condition properties in the property list for the component. For capacitors, this is an initial voltage across the capacitor, and for inductors, it is an initial current in the inductor. The default is to observe all the initial conditions in the DC analysis that is used as the time-zero timepoint. The initial conditions force a voltage or current to be present in the time-zero solution. The initial conditions are released for the rest of the transient simulation in the tstab interval. The *ic* option controls the initial conditions that should be observed in the time-zero timepoint. *all* is the default. *dev* means that only the initial conditions on capacitors and inductors are observed. *node* means that only the initial conditions on a node are observed. *dc* means that no initial conditions are observed. The example below shows the ic option set to *node*.

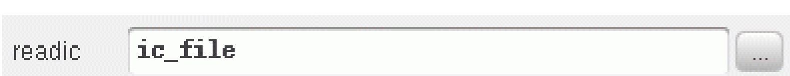


Skipdc (Shooting and Harmonic Balance)

In some cases, the time-zero timepoint DC analysis does not converge. Instead of stopping the simulation, *skipdc* allows the tstab simulation to continue using an assumed solution for the time-zero timepoint. The default is *no* and a DC analysis is run to get the initial timepoint. *yes* means skip the DC solution, and proceed directly to the tstab simulation. All the nodes with initial conditions specified start at the initial condition value. Nodes with batteries start at the battery voltage. Nodes with no initial conditions start at zero volts. For `skipdc=yes`, the signal sources start as specified immediately in the tstab simulation. *sigrampup* uses the same assumptions for the starting voltages as *yes* does, but the start time is set to negative one tenth of the stop time for the tstab interval. At this time, the signal source time-varying part starts at zero and linearly ramps up to the full value at time = zero. After time = zero, the sources have the full amplitude time-varying part. The example below shows the *skipdc* option set to *sigrampup*.

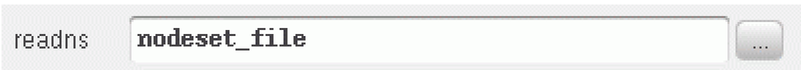
Readic (Shooting and Harmonic Balance)

This specifies an ascii file that contains two columns to be read as initial conditions. The first column is the node name. The second column is the voltage value. If the entry does not start with / (slash), the entry is located in the netlist directory. To find the netlist directory, select *Setup- Simulator/Directory/Host* in ADE. Look in the *Project Directory* field for the location of the simulation directory. Navigate to that directory and then to the `<Circuit Name>/spectre/<schematic or config>/netlist` directory. You can also click () and browse to the directory. An example is shown below.



Readns (Shooting and Harmonic Balance)

This specifies an ascii file with the same format as an ic file that is used as nodesets for the time-zero DC solution. Nodesets do not force a voltage to be held for the time-zero solution. Instead, they are a way of speeding up the time-zero calculation. As a suggestion, set the *write* option and the *readns* option to the same filename. The *write* option writes the time-zero solution to a file. When this is used as a starting point, many fewer iterations are needed for the time-zero point to converge. An example is shown below.



Cmin (Shooting and Harmonic Balance)

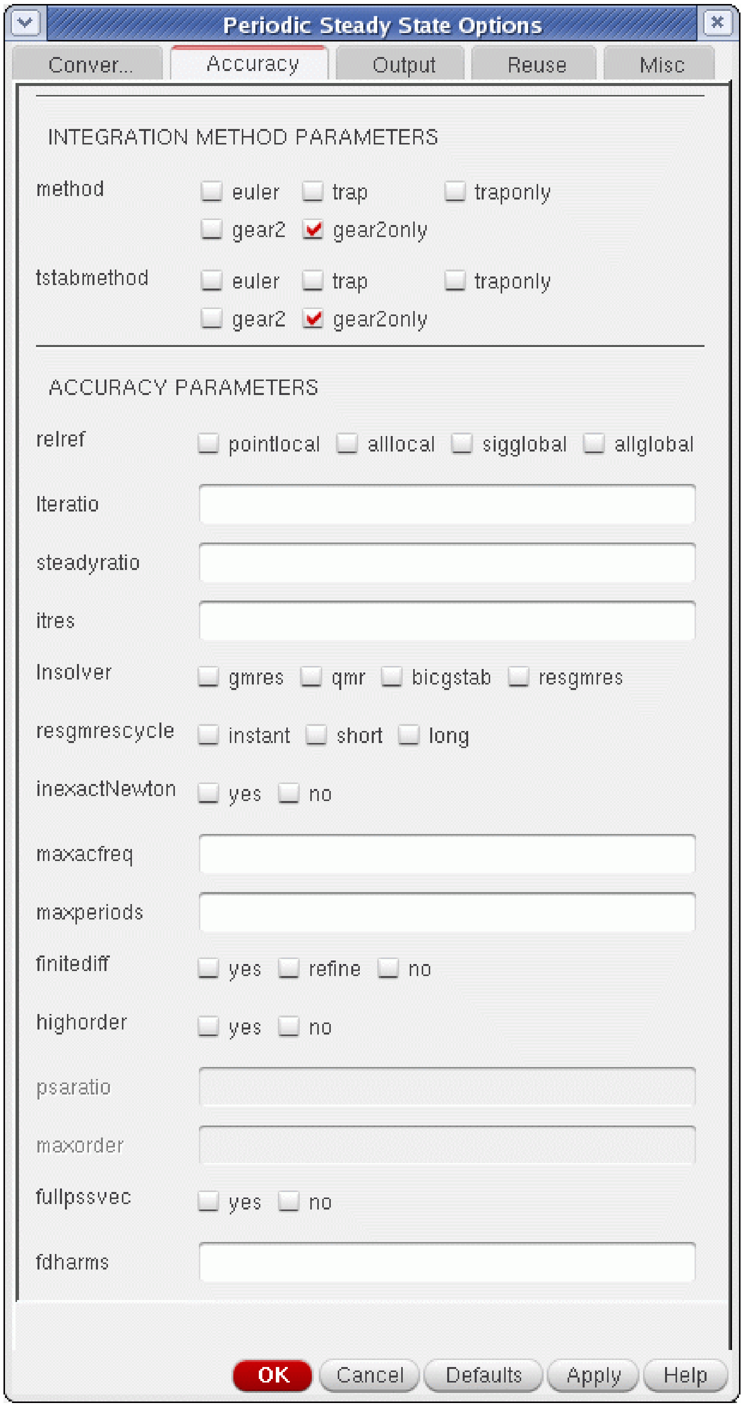
cmin can be used to improve convergence in the tstab and shooting intervals for shooting and for the tstab interval for harmonic balance. If a value is set for *cmin*, a capacitor with this value is added to every node with the other terminal of the capacitor connected to the global ground node. If a 10f to 50f capacitor is added, this prevents instantaneous changes from occurring from timepoint to timepoint, thus improving the convergence at the cost of adding non-physical capacitors to the circuit. An example with 10 femtoFarads is shown below. Note that if 10 is entered, a 10 Farad capacitor is added from every node to ground. Remember the multiplier in the entry. An example is shown below.

Maxiters (Shooting and Harmonic Balance)

The transient analysis for tstab (for both shooting and hb) and shooting interval iterate to a solution at each timepoint. *maxiters* specifies the maximum number of iterations before the timestep is cut for another try at convergence. In some cases, model parameters can cause discontinuities in the device current or capacitance. If this occurs, the change in the circuit condition can be large enough to require more than the five iterations that are the default. Specifying *maxiters* to the 40 to 100 range usually allows the simulator to converge in spite of the discontinuity. An example with *maxiters* set to 40 is shown below.

Accuracy Tab

The *Accuracy* tab is split into two categories: INTEGRATION METHOD PARAMETERS and ACCURACY PARAMETERS. For each parameter, the most commonly used settings are discussed.


Periodic Steady State Options
Conver...
Accuracy
Output
Reuse
Misc
INTEGRATION METHOD PARAMETERS
method
euler
trap
traponly
gear2
gear2only
tstabmethod
euler
trap
traponly
gear2
gear2only
ACCURACY PARAMETERS
relref
pointlocal
alllocal
sigglobal
allglobal
lteratio
steadyratio
itres
lnsolver
gmres
qmr
bicgstab
resgmres
resgmrescycle
instant
short
long
inexactNewton
yes
no
maxacfreq
maxperiods
finitediff
yes
refine
no
highorder
yes
no
psaratio
maxorder
fullpssvec
yes
no
fdharms
OK
Cancel
Defaults
Apply
Help

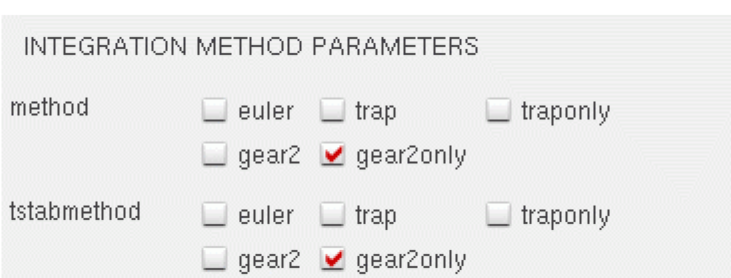
Method (Shooting only)

This option controls the integration method for the shooting window. Note that harmonic balance does not have the *method* option because the solver works in the frequency domain. The default value for the shooting window is determined by the setting of *errpreset*. For *moderate* and *conservative*, the default is *gear2only*. For *liberal* it is *traponly*. Generally *gear2only* is preferred because of the absence of trapezoidal ringing inherent in the *trap* (trapezoidal) method. Note that *liberal* is not recommended for RF simulation.

In the case where you are simulating a crystal oscillator with shooting, the method option should be set to *traponly*. The ringing that is inherent in the trapezoidal method is present and may cause convergence failure because of the unpredictable amplitude of the ringing, which depends on the accumulated total numerical integration error. The benefit of having a method that neither damps nor emphasizes the oscillations is required when simulating high Q oscillators. Even though `trap` neither emphasizes nor damps the oscillations, the small numerical integration errors caused by the finite timestep can still cause some inaccuracy in the simulation result. Using *gear2only* is not recommended even though convergence is more likely because the numerical damping inherent in the *gear2only* method causes unacceptable damping in the resonator, which causes smaller than real amplitude inside the resonator. Because of this, the overall solution will result in an error and can cause inaccurate measurement of the phase noise.

Tstabmethod (Shooting and Harmonic Balance)

This option controls the integration method in the tstab interval. The default value is determined by the setting of *errpreset* and whether the circuit is driven or is an oscillator. For *moderate* and *conservative* with a driven circuit, the default is *gear2only*. For *liberal* it is *traponly*. Generally *gear2only* is preferred because of the absence of trapezoidal ringing inherent in the *trap* (trapezoidal) method. The default for *tstabmethod* is *traponly* for oscillators. *traponly* is used for oscillators because with the defaults, the tstab interval uses relatively loose convergence options in order to speed up the simulation. This causes relatively long timesteps which for the *gear2* method would cause noticeable numerical damping. *traponly* does not numerically damp, therefore, it provides a better estimate of the oscillating frequency and amplitude. An example with *tstabmethod* set to *gear2only* is shown below.



Relref (Shooting (tstab and shooting window) and Harmonic Balance (tstab only))

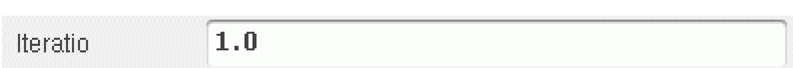
In the order of most to least accurate, the *relref* settings are *pointlocal*, *alllocal*, *sigglobal*, *allglobal*. The default for *moderate* and *conservative* is *alllocal* which is preferred. *relref* is used in the transient analysis in the tstab interval for both shooting and harmonic balance and in the shooting window for shooting. *relref* is used in hb only for the tstab interval because the solver works in the frequency domain. In some cases, the timestep can collapse to near zero. The symptom is that in the Spectre output window, the percent done number remains the same for many reporting intervals. The solution to this is to set *relref* to *sigglobal*, which is slightly less accurate. In the Fourier transform, the noise floor usually degrades by 3 to 6 dB. Another possibility is to use the *lteminstep* option as described in the *AdditionalParams* option of the *Misc* tab.

Refer to Relref on page 563 for a detailed discussion on *relref*.

An example with *relref* set to *sigglobal* is shown below.

Lteratio Shooting ((tstab and shooting window) and Harmonic Balance (tstab only))

lteratio is a multiplier for the allowable numerical integration error in the tstab interval and the shooting window. The default is 3.5 or 10 depending on the setting of *reltol* and *errpreset*. See Controlling Accuracy on page 571 for details. *lteratio* cannot be set smaller than 1.0 and is normally between 3.5 and 100. In some cases, the timestep collapses to near zero and setting *relref* to *sigglobal* does not fix the problem. In this case, there might be a model discontinuity. To test this, disable numerical integration timestep control by setting *lteratio* to 1e9. Note that this is an extreme measure not to be used under normal simulations. If the timestep still collapses, there is a discontinuity in one of the models in the circuit. This could be a device model or a Verilog-A model. If it is necessary to set *lteratio* very large to get the simulation to complete, you must also set *maxstep* small enough to preserve the accuracy of the simulation. An example with *lteratio* set to the smallest value of *1.0* is shown below. Note that *lteminstep* might also be used to treat this condition. See the AdditionalParams (Shooting and Harmonic Balance) on page 645 section in the *Misc* tab on how to use this option.



Steadyratio (Shooting Only)

The maximum mismatch between the beginning and ending values for the voltages and currents in the shooting window is determined by the *convergence criteria* * *lteratio* * *steadyratio*. In some cases, with oscillators, *conv_norm* drops to a small value and then bumps up and down. However, it never gets less than one, therefore, the convergence fails. In this case, setting *steadyratio* to 1.0 will cause convergence if the *conv_norm* number gets to 10 or less. While larger settings of *steadyratio* may cause the oscillator to converge, more mismatch from beginning to end of the shooting window is allowed, which allows much more startup behavior to be in the solution. Setting *steadyratio* larger than one is not recommended. An example with *steadyratio* set to 1.0 is shown below.

Steadyratio (Harmonic Balance Only)

steadyratio in harmonic balance is used as a multiplier for the maximum delta that is allows from iteration to iteration. The default is 1.0. In some cases, *resd_norm* falls below one which indicates that the KCL check at each node is passing, but the Delta Norm is greater than one, which indicates that the motion from iteration-to-iteration is larger than the maximum allowed. Only in this case should *steadyratio* be set larger than one. Set it just larger than the value of the *resd_norm*. In this case, the smaller harmonics are changing relatively a lot, but because they are small, they do not affect the accuracy of the solution.

Itres (Shooting Only)

Use the default for this option. *itres* controls the precision (number of digits solved for) in the first iteration. The default is 1e-4 which causes 4 digits of resolution in the first iteration. Subsequent iterations are solved with more precision.

Itres (Harmonic Balance Only)

The default for *itres* for harmonic balance is 0.9, which allows up to 90% difference from the exact solution on the first iteration. Although this seems large, solving exactly for the first iteration can frequently cause non-convergence. As the iterations progress, the difference is reduced so that at the solution, full accuracy is achieved. In the case of very linear systems like an amplifier, tightening *itres* to the 1e-2 to 1e-4 range can reduce the number of frequency domain iterations and runtime. Also, when transient assist is used, the fft provides an excellent starting point for the first iteration. Therefore, setting *itres* to the 1e-2 to 1e-4 range can also reduce the number of frequency domain iterations and the runtime.

Lnsolver (Shooting Only)

Leave this option at the default. At the end of each pss iteration, a large matrix is formed and solved to calculate the new starting value in the shooting window. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

You need to have a good understanding of the linear algebra theory to understand the differences between these methods.

Gmres is the Generalized Minimum RESidual method.

Qmr is the Quasi-Minimal Residual method.

Bicgstab is the STABilized BI-Conjugate Gradient method.

Resgmres is the REStarted Generalized Minimal Residual method.

Lnsolver (Harmonic Balance Only)

Leave this option at the default. The frequency domain matrices are calculated with an iterative solver. *lnsolver* controls the algorithm that is used for the iterative solver. *gmres* is the default, and is a good general-purpose solver because it usually has higher accuracy for each iteration. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

You need to have a good understanding of the linear algebra theory to understand the differences between these methods.

Gmres is the Generalized Minimum RESidual method.

Qmr is the Quasi-Minimal Residual method.

Bicgstab is the STABilized BI-Conjugate Gradient method.

Resgmres is the REStarted Generalized Minimal Residual method.

Resgmrescycle (Shooting and Harmonic Balance)

Leave this option at the default. For the resgmres linear solver, there are several different options.

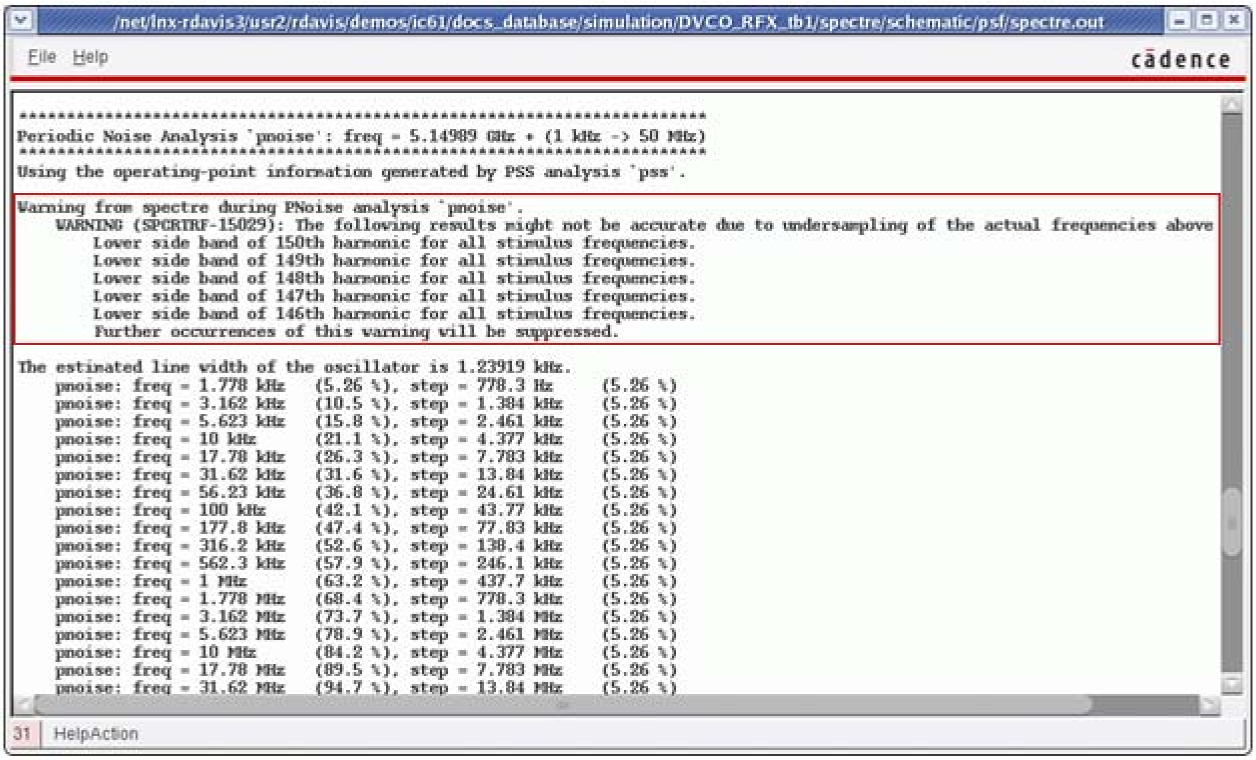
Inexact Newton (Shooting Only)

In the simulation time between iterations, an estimate is made for the starting point for the next iteration (shooting) or the harmonic content for the next iteration (hb). *inexactNewton* causes an inexact solution for the matrix that calculates the next starting point to be calculated. In some cases, solving the matrix exactly on the first iteration can cause the iterative solver to need many iterations to achieve convergence. Setting i*nexactNewton* causes a different series of iterations, which may speed up the simulation. The inexactness is reduced as the iterations progress. Try this option when the time spent between iterations is excessive.

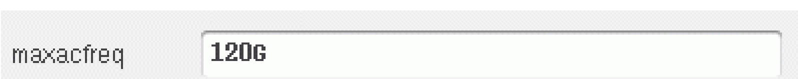
Maxacfreq (Shooting Only)

If maximum sideband in *pac*, *pnoise*, *pxf*, or *psp* is set larger than 40, then the *pss* option *maxacfreq* needs to be set to the pss beat frequency times the largest maximum sideband number in the small-signal analyses. For example, if *pac* has 20 for maximum sidebands, *pnoise* has 70, *pxf* has 10, and *psp* has 50, then *maxacfreq* needs to be set to the pss frequency times 70.

Spectre may issue a warning at the beginning of the small-signal analysis like the following one from pnoise if *maxacfreq* is not set and more than 40 sidebands are specified. If you see this warning in a small-signal analysis, set the pss option *maxacfreq* to the pss beat frequency times the number of sidebands used in the small-signal analysis. In the example below, *maxacfreq* should be set to 150 * pss frequency.

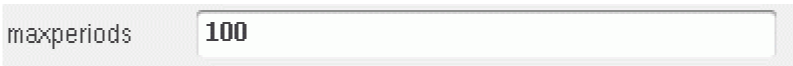


An example of *maxacfreq* set to 120 GHz is shown below.



Maxperiods (Shooting and Harmonic Balance)

Any iterative method may fail to converge, and therefore needs an iteration limit. Maxperiods sets the maximum number of times the waveform in the shooting window can be simulated or the number of hb iterations. The default is 20 times for a driven circuit, and 50 times for an oscillator in shooting, and 100 for hb. This is a minimum allowed value for this option. Setting a number less than this default will be ignored by Spectre. This should be set higher only after watching the conv_norm during the pss iterations. If the conv_norm is trending down with no sudden large increase, setting more iterations may allow convergence. If the conv_norm trends down, and then gets large quickly, setting more iterations is very unlikely to help. An example with 100 is shown below.



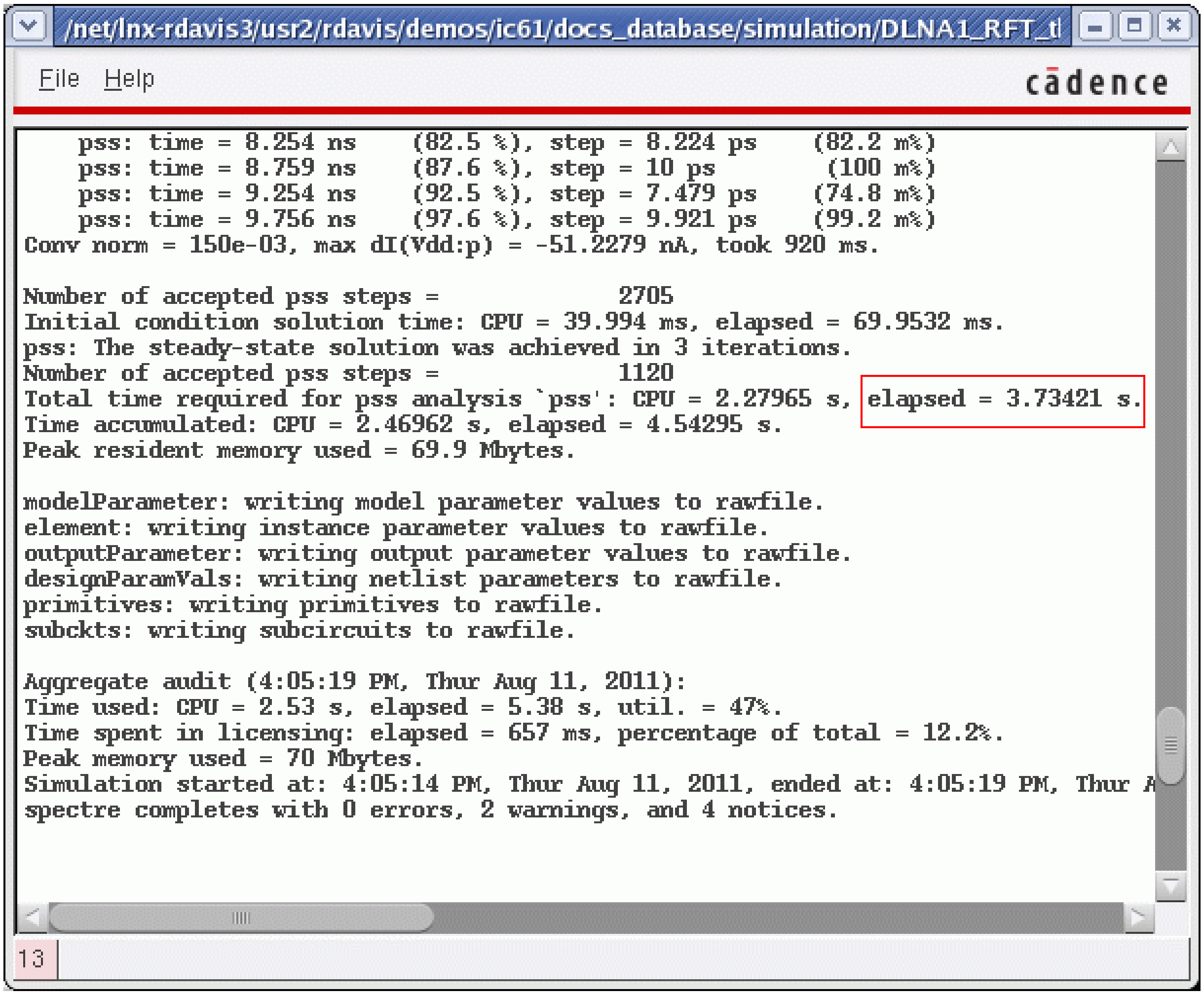
Finitediff (Shooting Only)

Leave this option at the default. This is set automatically when *readpss* is set. It allows small changes in circuit values when the *readpss* option is used.

Highorder (Shooting Only)

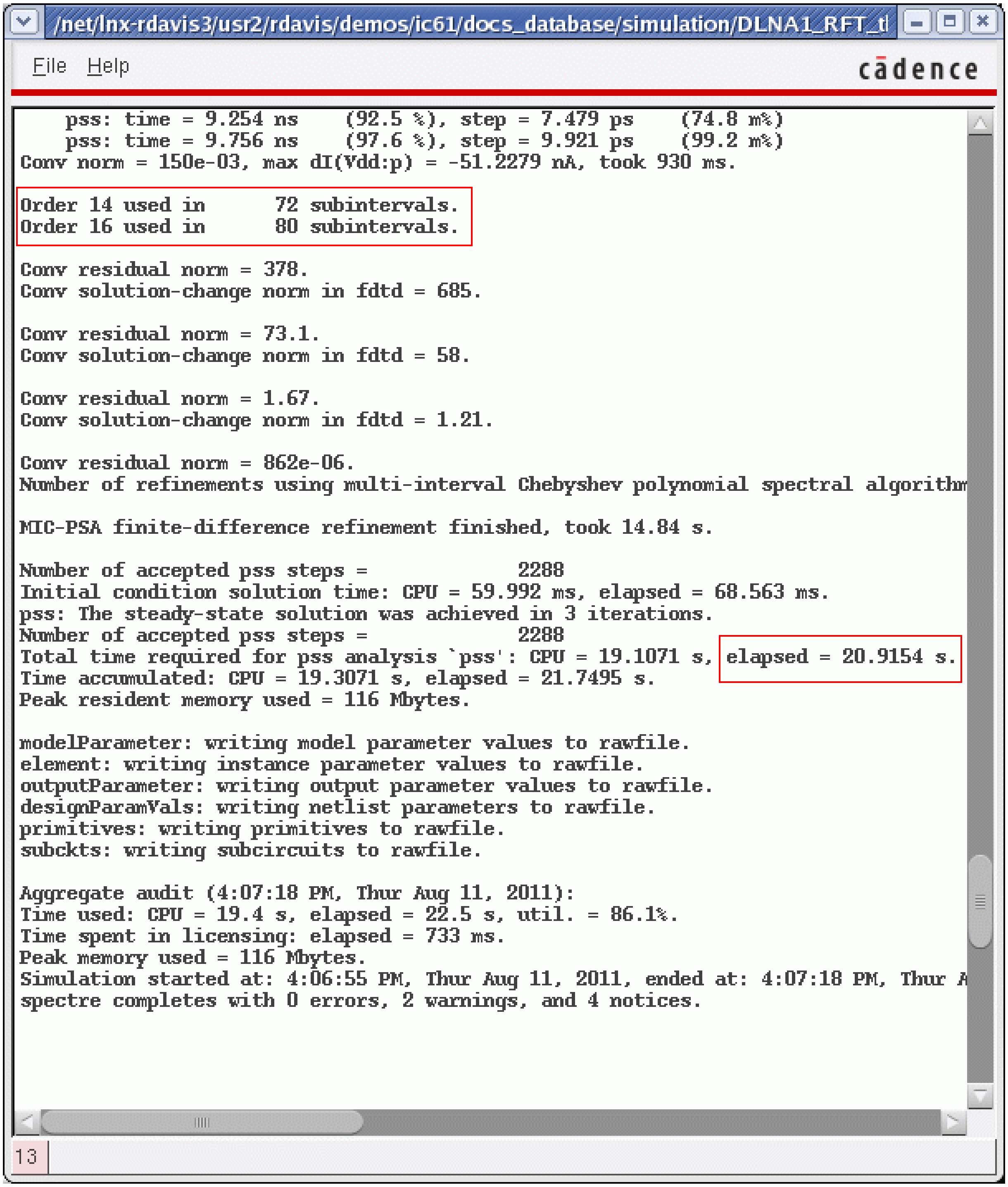
The normal integration method in the shooting window is *gear2only* which is a second order integration method. *highorder* allows the integration method to have up to 16th order numerical integration, thus making the numerical noise floor of the simulation much smaller. When *highorder* is set to *yes*, a normal pss is run until convergence is achieved, and then that solution is refined using a higher order integration method. In practice this has value for very linear systems like LNAs, and has smaller value for more complicated blocks like mixers. When *highorder* is set to *yes*, there can be considerable time when the highorder algorithm is running, but there is no output to the spectre output window. This is normal for highorder. An example of `highorder=yes` is shown below.

Below is the runtime for `highorder = no`.



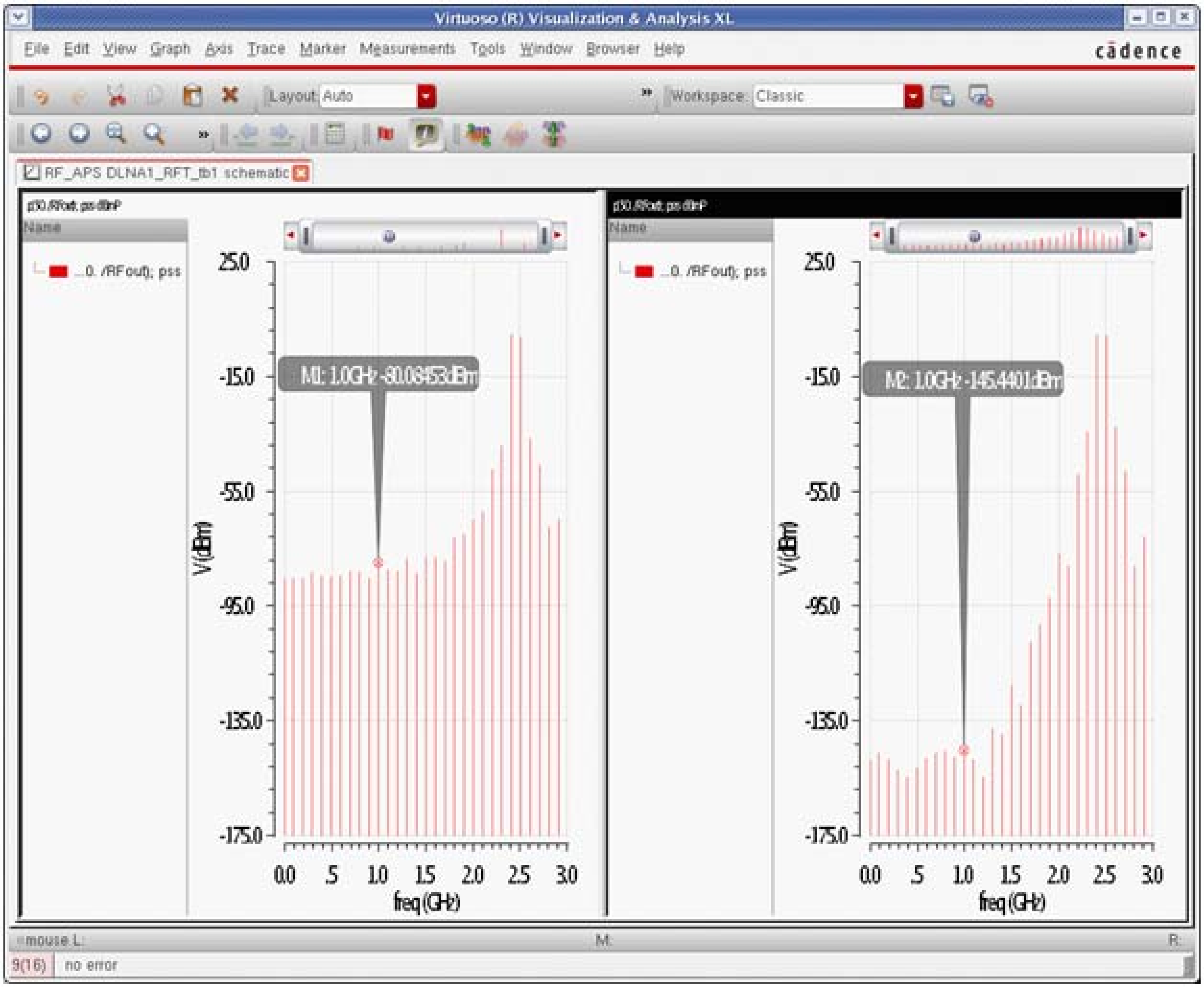

```
    pss: time = 8.254 ns    (82.5 %), step = 8.224 ps    (82.2 m%)
    pss: time = 8.759 ns    (87.6 %), step = 10 ps        (100 m%)
    pss: time = 9.254 ns    (92.5 %), step = 7.479 ps    (74.8 m%)
    pss: time = 9.756 ns    (97.6 %), step = 9.921 ps    (99.2 m%)
Conv norm = 150e-03, max dI(Vdd:p) = -51.2279 nA, took 920 ms.

Number of accepted pss steps =              2705
Initial condition solution time: CPU = 39.994 ms, elapsed = 69.9532 ms.
pss: The steady-state solution was achieved in 3 iterations.
Number of accepted pss steps =              1120
Total time required for pss analysis `pss': CPU = 2.27965 s, elapsed = 3.73421 s.
Time accumulated: CPU = 2.46962 s, elapsed = 4.54295 s.
Peak resident memory used = 69.9 Mbytes.

modelParameter: writing model parameter values to rawfile.
element: writing instance parameter values to rawfile.
outputParameter: writing output parameter values to rawfile.
designParamVals: writing netlist parameters to rawfile.
primitives: writing primitives to rawfile.
subckts: writing subcircuits to rawfile.

Aggregate audit (4:05:19 PM, Thur Aug 11, 2011):
Time used: CPU = 2.53 s, elapsed = 5.38 s, util. = 47%.
Time spent in licensing: elapsed = 657 ms, percentage of total = 12.2%.
Peak memory used = 70 Mbytes.
Simulation started at: 4:05:14 PM, Thur Aug 11, 2011, ended at: 4:05:19 PM, Thur A
spectre completes with 0 errors, 2 warnings, and 4 notices.
```

Below is the runtime and the report of the order that was used in the simulation when *highorder* was set to *yes*. The runtime is much longer.

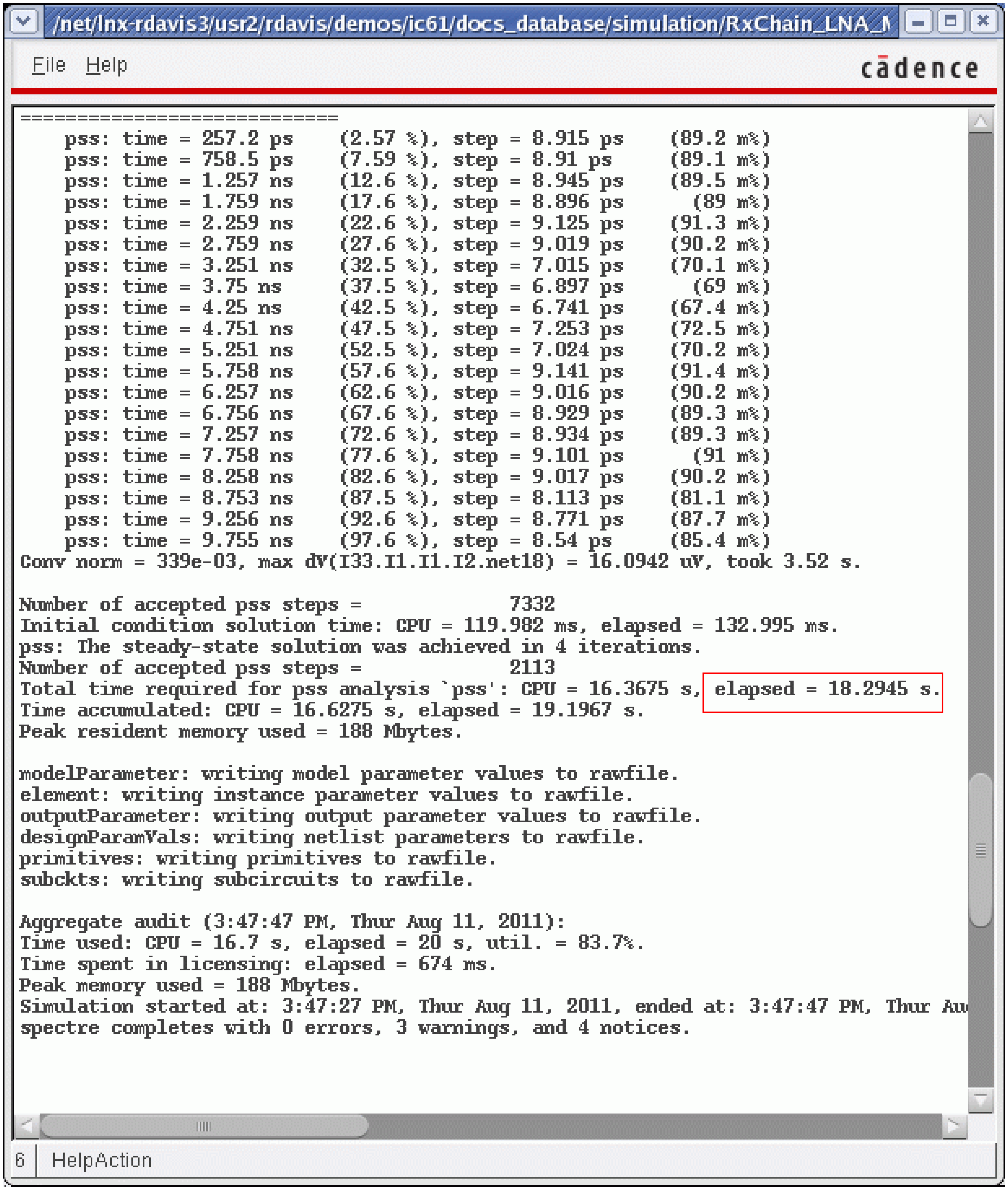


```
/net/lnx-rdavis3/usr2/rdavis/demos/ic61/docs_database/simulation/DLNA1_RFT_tl
File  Help                                                        cādence
    pss: time = 9.254 ns    (92.5 %), step = 7.479 ps    (74.8 m%)
    pss: time = 9.756 ns    (97.6 %), step = 9.921 ps    (99.2 m%)
Conv norm = 150e-03, max dI(Vdd:p) = -51.2279 nA, took 930 ms.

Order 14 used in      72 subintervals.
Order 16 used in      80 subintervals.

Conv residual norm = 378.
Conv solution-change norm in fdtd = 685.

Conv residual norm = 73.1.
Conv solution-change norm in fdtd = 58.

Conv residual norm = 1.67.
Conv solution-change norm in fdtd = 1.21.

Conv residual norm = 862e-06.
Number of refinements using multi-interval Chebyshev polynomial spectral algorithm

MIC-PSA finite-difference refinement finished, took 14.84 s.

Number of accepted pss steps =                2288
Initial condition solution time: CPU = 59.992 ms, elapsed = 68.563 ms.
pss: The steady-state solution was achieved in 3 iterations.
Number of accepted pss steps =                2288
Total time required for pss analysis `pss': CPU = 19.1071 s, elapsed = 20.9154 s.
Time accumulated: CPU = 19.3071 s, elapsed = 21.7495 s.
Peak resident memory used = 116 Mbytes.

modelParameter: writing model parameter values to rawfile.
element: writing instance parameter values to rawfile.
outputParameter: writing output parameter values to rawfile.
designParamVals: writing netlist parameters to rawfile.
primitives: writing primitives to rawfile.
subckts: writing subcircuits to rawfile.

Aggregate audit (4:07:18 PM, Thur Aug 11, 2011):
Time used: CPU = 19.4 s, elapsed = 22.5 s, util. = 86.1%.
Time spent in licensing: elapsed = 733 ms.
Peak memory used = 116 Mbytes.
Simulation started at: 4:06:55 PM, Thur Aug 11, 2011, ended at: 4:07:18 PM, Thur A
spectre completes with 0 errors, 2 warnings, and 4 notices.
13
```

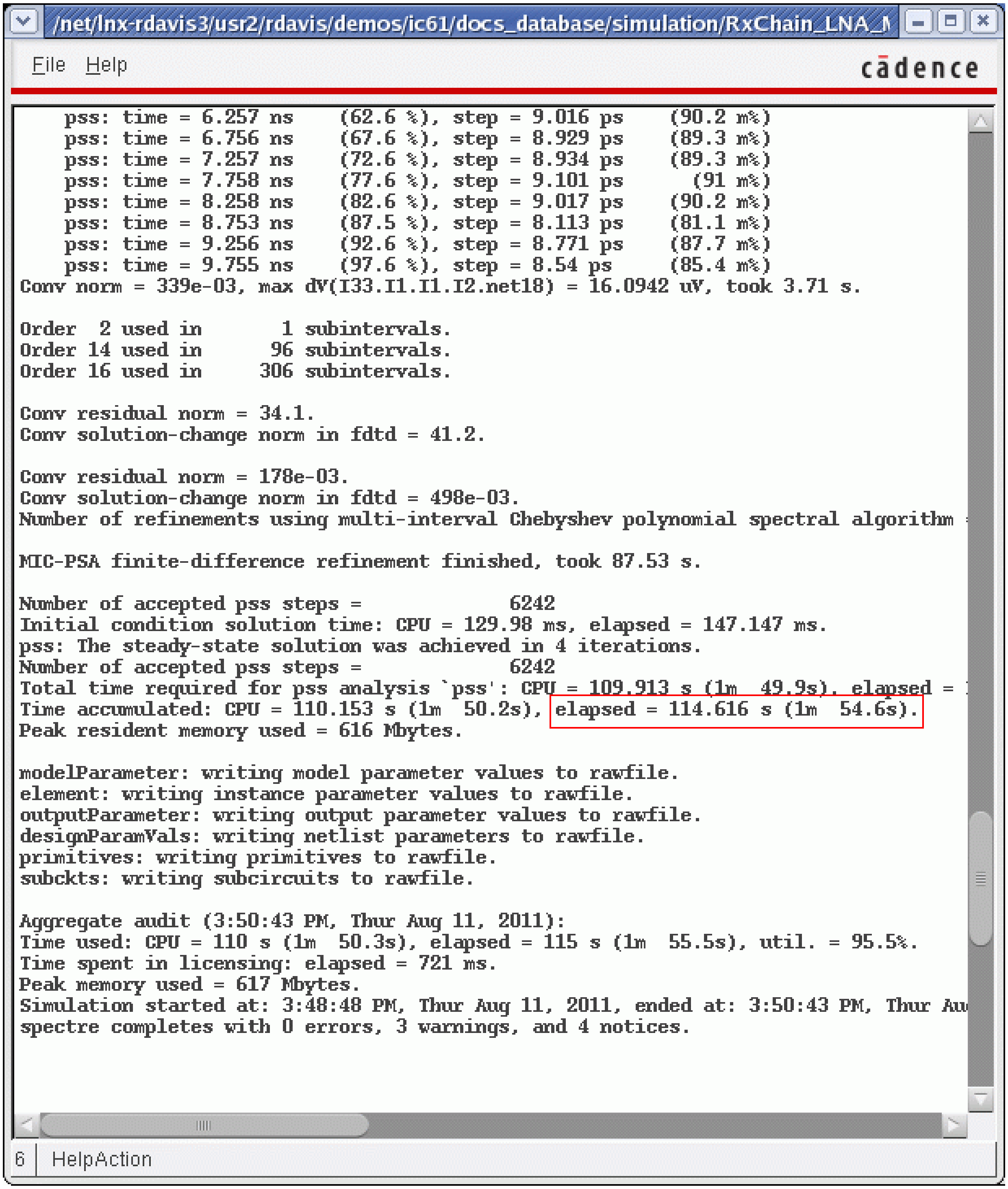
Below is a spectral comparison between standard pss and *highorder* set to *yes*. The convergence options are at the default and *conservative* is selected. The results are for an LNA. The numerical noise floor at 1GHz dropped from -80dBm to -145 dBm.



Below is a comparison of runtimes for an LNA-Mixer circuit. The first is with *highorder* set to *no*, and the second is with *highorder* set to *yes*.

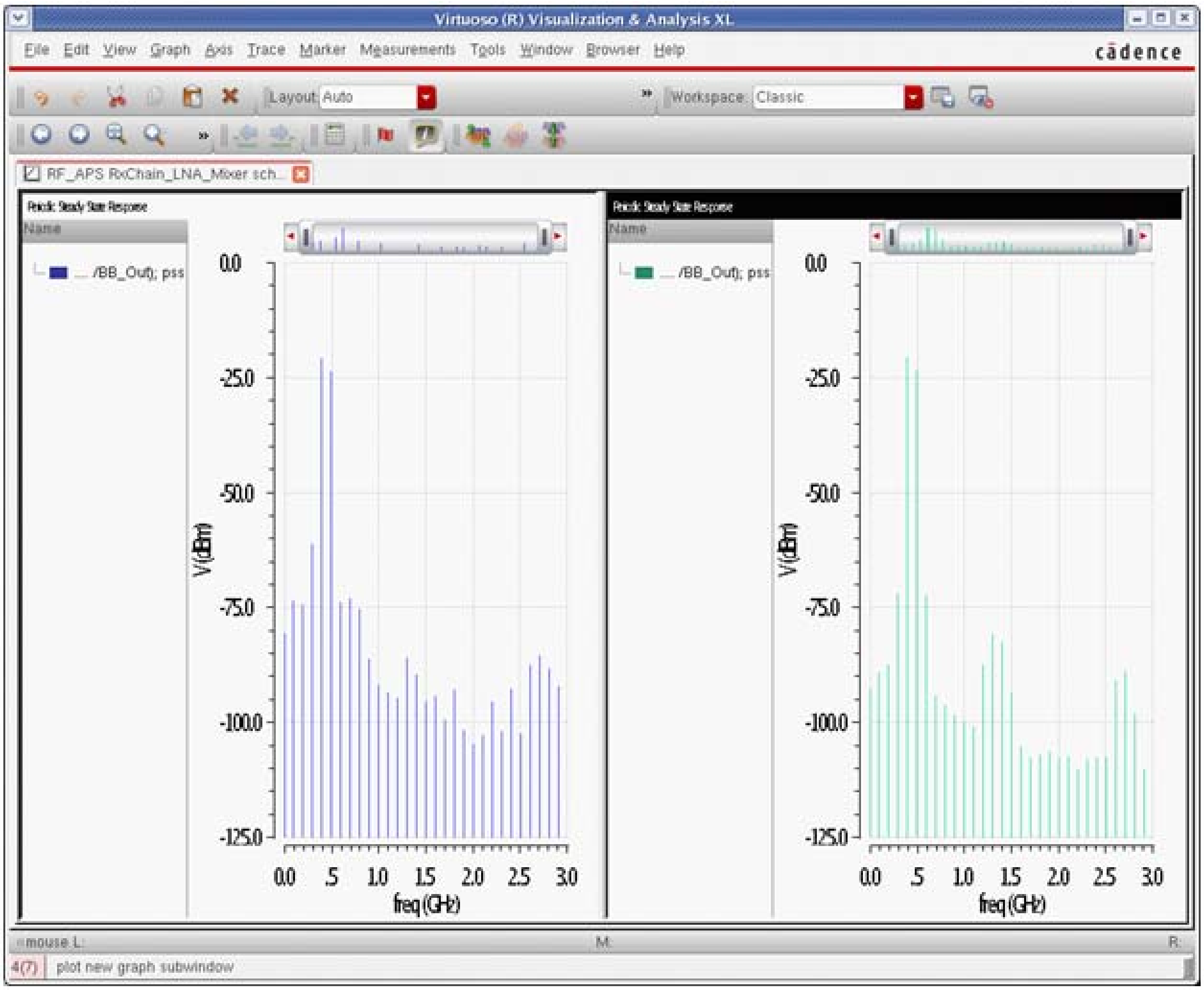



```
============================
    pss: time = 257.2 ps    (2.57 %), step = 8.915 ps    (89.2 m%)
    pss: time = 758.5 ps    (7.59 %), step = 8.91 ps     (89.1 m%)
    pss: time = 1.257 ns    (12.6 %), step = 8.945 ps    (89.5 m%)
    pss: time = 1.759 ns    (17.6 %), step = 8.896 ps      (89 m%)
    pss: time = 2.259 ns    (22.6 %), step = 9.125 ps    (91.3 m%)
    pss: time = 2.759 ns    (27.6 %), step = 9.019 ps    (90.2 m%)
    pss: time = 3.251 ns    (32.5 %), step = 7.015 ps    (70.1 m%)
    pss: time = 3.75 ns     (37.5 %), step = 6.897 ps      (69 m%)
    pss: time = 4.25 ns     (42.5 %), step = 6.741 ps    (67.4 m%)
    pss: time = 4.751 ns    (47.5 %), step = 7.253 ps    (72.5 m%)
    pss: time = 5.251 ns    (52.5 %), step = 7.024 ps    (70.2 m%)
    pss: time = 5.758 ns    (57.6 %), step = 9.141 ps    (91.4 m%)
    pss: time = 6.257 ns    (62.6 %), step = 9.016 ps    (90.2 m%)
    pss: time = 6.756 ns    (67.6 %), step = 8.929 ps    (89.3 m%)
    pss: time = 7.257 ns    (72.6 %), step = 8.934 ps    (89.3 m%)
    pss: time = 7.758 ns    (77.6 %), step = 9.101 ps      (91 m%)
    pss: time = 8.258 ns    (82.6 %), step = 9.017 ps    (90.2 m%)
    pss: time = 8.753 ns    (87.5 %), step = 8.113 ps    (81.1 m%)
    pss: time = 9.256 ns    (92.6 %), step = 8.771 ps    (87.7 m%)
    pss: time = 9.755 ns    (97.6 %), step = 8.54 ps     (85.4 m%)
Conv norm = 339e-03, max dV(I33.I1.I1.I2.net18) = 16.0942 uV, took 3.52 s.

Number of accepted pss steps =             7332
Initial condition solution time: CPU = 119.982 ms, elapsed = 132.995 ms.
pss: The steady-state solution was achieved in 4 iterations.
Number of accepted pss steps =             2113
Total time required for pss analysis `pss': CPU = 16.3675 s, elapsed = 18.2945 s.
Time accumulated: CPU = 16.6275 s, elapsed = 19.1967 s.
Peak resident memory used = 188 Mbytes.

modelParameter: writing model parameter values to rawfile.
element: writing instance parameter values to rawfile.
outputParameter: writing output parameter values to rawfile.
designParamVals: writing netlist parameters to rawfile.
primitives: writing primitives to rawfile.
subckts: writing subcircuits to rawfile.

Aggregate audit (3:47:47 PM, Thur Aug 11, 2011):
Time used: CPU = 16.7 s, elapsed = 20 s, util. = 83.7%.
Time spent in licensing: elapsed = 674 ms.
Peak memory used = 188 Mbytes.
Simulation started at: 3:47:27 PM, Thur Aug 11, 2011, ended at: 3:47:47 PM, Thur Au
spectre completes with 0 errors, 3 warnings, and 4 notices.
```



```
/net/lnx-rdavis3/usr2/rdavis/demos/ic61/docs_database/simulation/RxChain_LNA_N
File  Help                                                        cadence
    pss: time = 6.257 ns     (62.6 %), step = 9.016 ps     (90.2 m%)
    pss: time = 6.756 ns     (67.6 %), step = 8.929 ps     (89.3 m%)
    pss: time = 7.257 ns     (72.6 %), step = 8.934 ps     (89.3 m%)
    pss: time = 7.758 ns     (77.6 %), step = 9.101 ps       (91 m%)
    pss: time = 8.258 ns     (82.6 %), step = 9.017 ps     (90.2 m%)
    pss: time = 8.753 ns     (87.5 %), step = 8.113 ps     (81.1 m%)
    pss: time = 9.256 ns     (92.6 %), step = 8.771 ps     (87.7 m%)
    pss: time = 9.755 ns     (97.6 %), step = 8.54 ps      (85.4 m%)
Conv norm = 339e-03, max dV(I33.I1.I1.I2.net18) = 16.0942 uV, took 3.71 s.

Order  2 used in       1 subintervals.
Order 14 used in      96 subintervals.
Order 16 used in     306 subintervals.

Conv residual norm = 34.1.
Conv solution-change norm in fdtd = 41.2.

Conv residual norm = 178e-03.
Conv solution-change norm in fdtd = 498e-03.
Number of refinements using multi-interval Chebyshev polynomial spectral algorithm

MIC-PSA finite-difference refinement finished, took 87.53 s.

Number of accepted pss steps =             6242
Initial condition solution time: CPU = 129.98 ms, elapsed = 147.147 ms.
pss: The steady-state solution was achieved in 4 iterations.
Number of accepted pss steps =             6242
Total time required for pss analysis `pss': CPU = 109.913 s (1m  49.9s), elapsed =
Time accumulated: CPU = 110.153 s (1m  50.2s), elapsed = 114.616 s (1m  54.6s).
Peak resident memory used = 616 Mbytes.

modelParameter: writing model parameter values to rawfile.
element: writing instance parameter values to rawfile.
outputParameter: writing output parameter values to rawfile.
designParamVals: writing netlist parameters to rawfile.
primitives: writing primitives to rawfile.
subckts: writing subcircuits to rawfile.

Aggregate audit (3:50:43 PM, Thur Aug 11, 2011):
Time used: CPU = 110 s (1m  50.3s), elapsed = 115 s (1m  55.5s), util. = 95.5%.
Time spent in licensing: elapsed = 721 ms.
Peak memory used = 617 Mbytes.
Simulation started at: 3:48:48 PM, Thur Aug 11, 2011, ended at: 3:50:43 PM, Thur Au
spectre completes with 0 errors, 3 warnings, and 4 notices.

6  HelpAction
```

The runtime is much longer for `highorder=no`, however, it allows the intermodulation products to be resolved. *highorder* set to *yes* is shown in the right subwindow below.



Psaratio (Shooting Only)

This is active when *highorder* is set to *yes*. The default is *one*. Setting smaller values produce more accurate solutions at the expense of runtime. Normally, the default should be used. In the example below, *psaratio* is set to 0.2 and a comparison of the numerical noise floor at 1GHz is made for the LNA used above. The noise floor dropped about 11 dBm when *psaratio* is lowered.

Maxorder (Shooting Only)

Leave this option at the default of 16. The order is set automatically in the highorder algorithm.

This is active when *highorder* is set to *yes*. This controls the maximum order used for the highorder method.

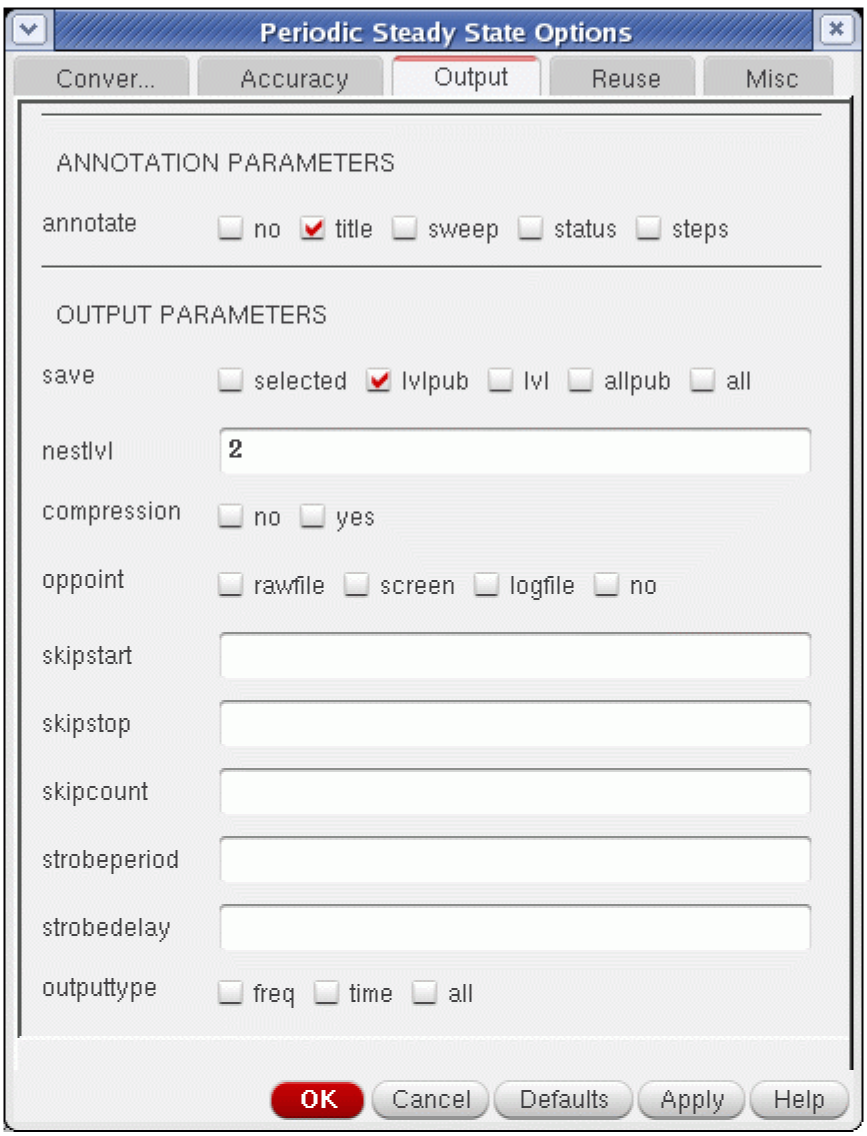
Fullpssvec (Shooting Only)

Leave this option at the default. At the end of each pss iteration, the settled steady-state point needs to be estimated. Because of the inherently variable timestep, not all the timepoints in the pss solution need to be used to estimate the next starting point. Setting *fullpssvec* to *yes* causes all the timepoints to be read in to calculate the starting voltages for the next iteration.

Fdharms (Shooting Only)

fdharms controls the number of harmonics of the pss beat frequency that are exposed to the `nport`, `mtline`, and `tline` components. This should be set to the number of harmonics that are set for maximum harmonics in the pss *Choosing Analyses* form.

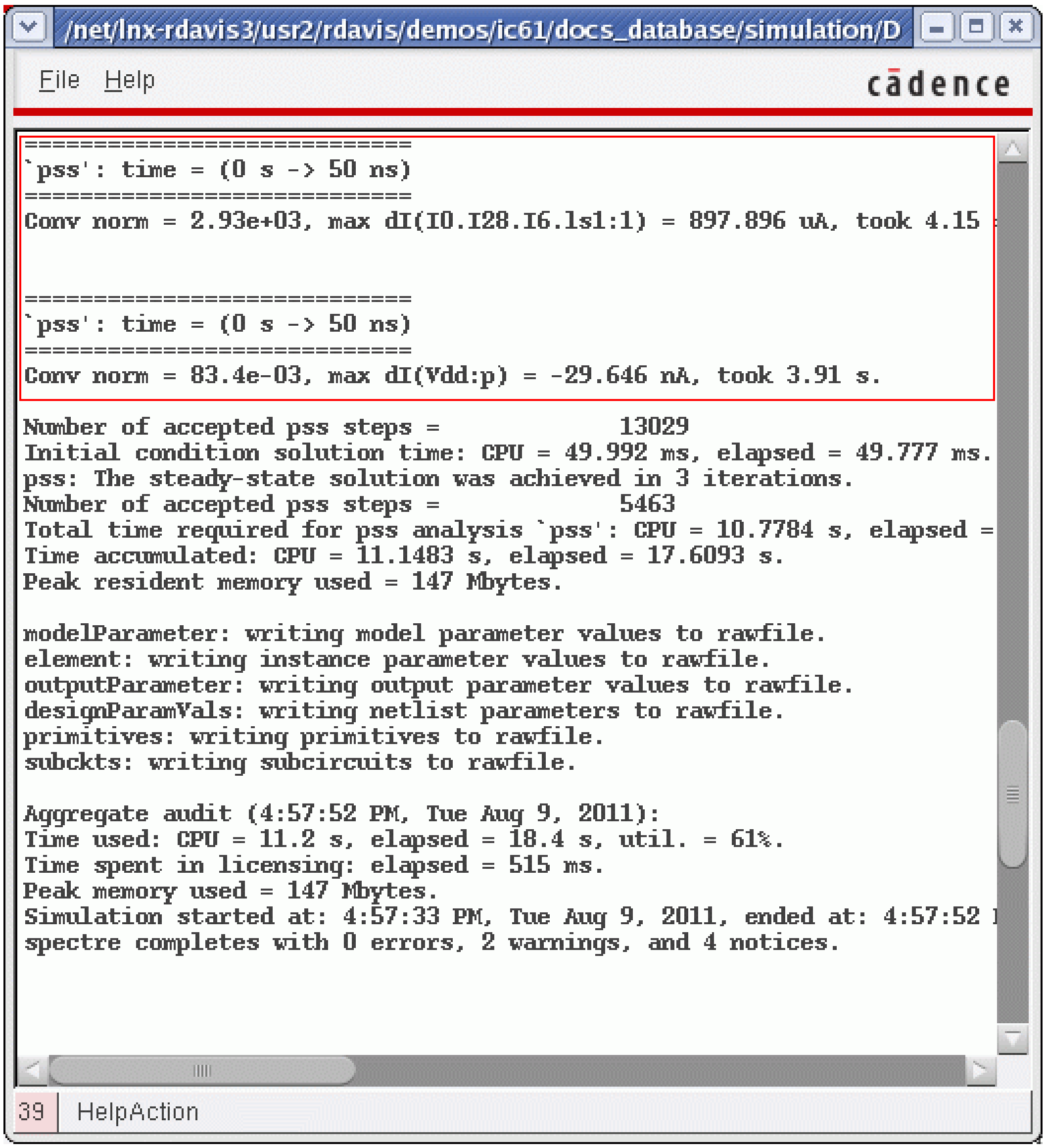
Output Tab



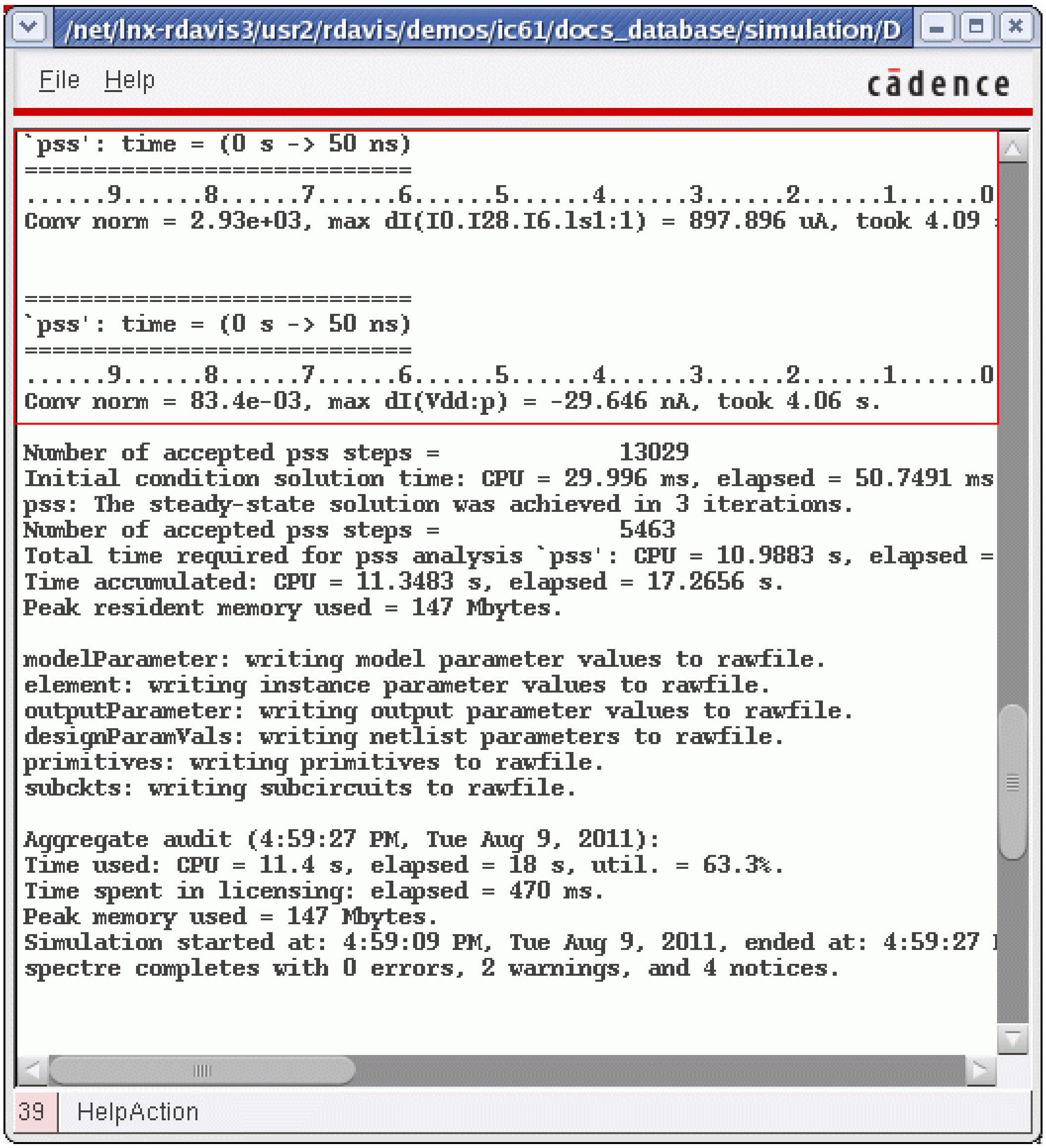
Annotate (Shooting and Harmonic Balance)

This option controls the level of detail in the output log. The default is *status*. *no* produces no output. *title* produces only the conv_norm number at each iteration. *sweep* counts down from 10 to zero at each iteration and displays conv_norm. *status* provides output at 2.5% and then at every 5% of the way to the end at each iteration and the conv_norm number. *steps* provides output at each timestep in the simulation and the conv_norm number at each iteration. An example is shown below with *title* selected.

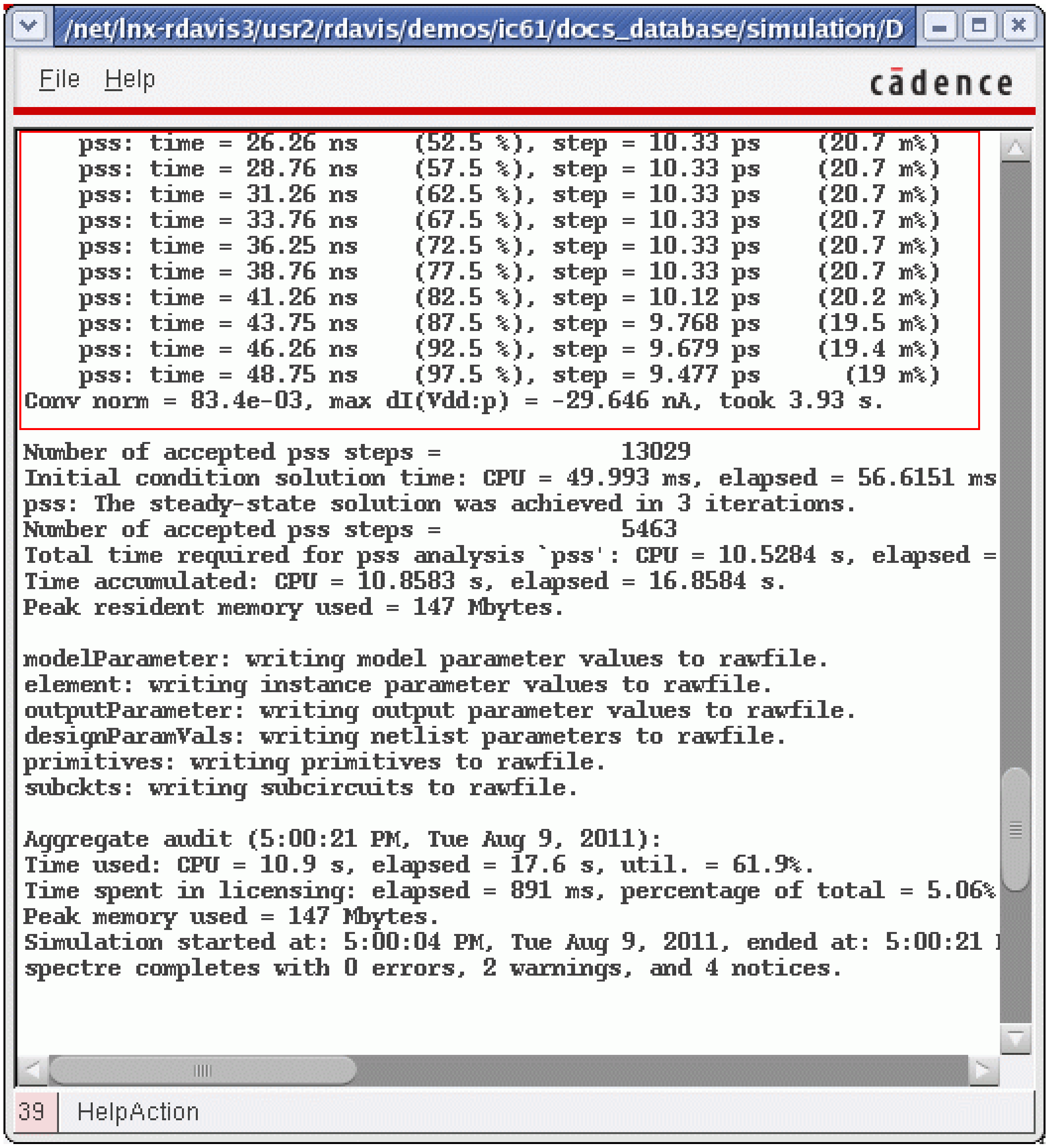
Below is the Spectre output with *title* selected.



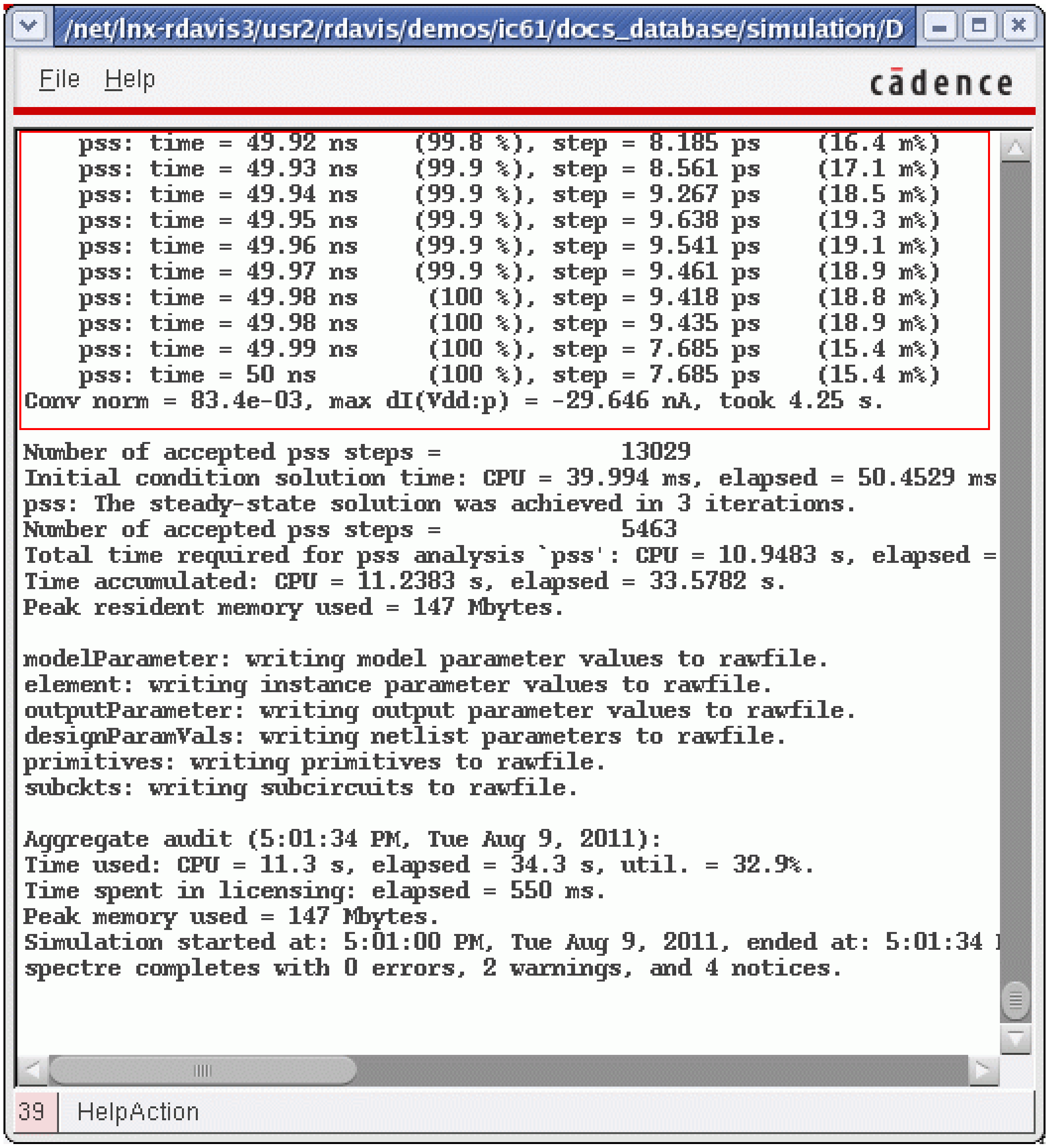
Below is the output with *sweep* selected.



Below is the output with status *selected*.



Below is the output with *steps* selected.

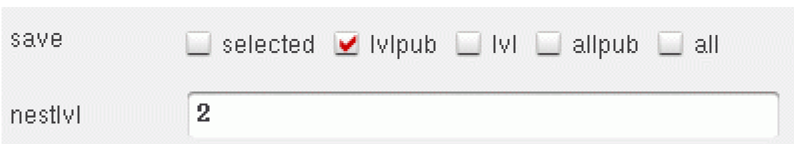


Save (Shooting and Harmonic Balance)

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished using the *Outputs - To Be Saved - Select On Schematic* and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.

Nestlvl (Shooting and Harmonic Balance)

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer. The example below shows *lvlpub* selected that saves two levels of hierarchy.



Compression (Shooting and Harmonic Balance)

Normally, this should not be used for RF simulation. It is not digital compression. For RF simulation, where the input is sinusoidal, the size of the results file will normally double. It is useful only for circuits that are predominantly square wave.

Oppoint (Shooting and Harmonic Balance)

Normally, this should be left at the default of *rawfile*. It controls where the time-zero operating point solution should be saved to.

Skipstart (Shooting and Harmonic Balance)

Normally, this is not used for RF simulation unless there is a very long tstab. It is a way of reducing the amount of data in the output file from the tstab interval by not writing many of the timepoints to the output file. Skipstart controls the simulation time where the skipping is to start. The default is 0 (zero).

Skipstop (Shooting and Harmonic Balance)

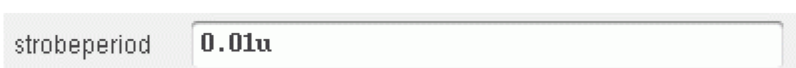
Normally, this is not used for RF simulation unless there is a very long tstab. This specifies the simulation time in the tstab interval where the skipping is to stop. The default is the stop time in the tstab interval.

Skipcount (Shooting and Harmonic Balance)

If *skipstart* and *skipstop* are set either *skipcount* or *strobeperiod* should be set, but not both. *skipcount* saves one out of every skipcount points to the output data file. A *skipcount* of 3 saves one, then skips two points. Skipcount of 10 saves one in 10 timepoints to the output file. The default is 1, which saves every point. An example is shown below. If *tstab* is 1u, the first 5% and the last 5% has all the datapoints saved, and in between, one out of ten points is saved.

Strobeperiod (Shooting and Harmonic Balance)

If *skipstart* and *skipstop* are set either *skipcount* or *strobeperiod* should be set, but not both. *strobeperiod* forces simulation datapoints between *skipstart* and *skipstop* at the interval of *strobeperiod*. Below is an example where the first 5% and the last 5% have all the datapoints, and in between there are timepoints at 0.01usecond intervals. Note that the simulator may actually calculate many points between the *skipstart* and *skipstop* times, but only the data at the strobe interval is saved to the output file.



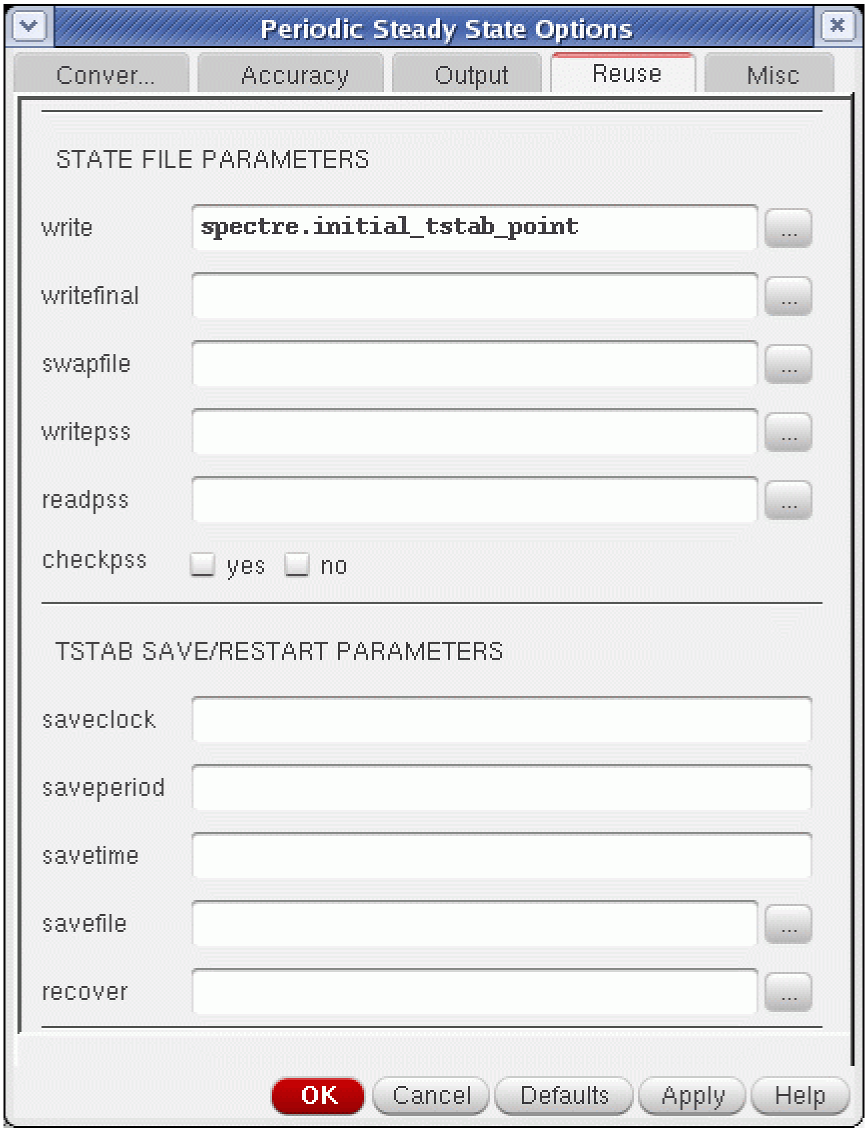
Strobedelay (Shooting and Harmonic Balance)

strobedelay specifies a time after the skipstart point for the beginning of the strobing. *strobedelay* must be between 0 (zero) and the time set in *strobeperiod*.

Outputtype (Shooting Only)

This controls what type of information is to be saved in the output file. The default of *all* saves both the time and frequency data, but the data can be restricted to either time or frequency domain data.

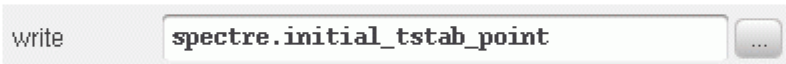
Reuse Tab



Write (Shooting Only)

write specifies a filename to which the DC solution used as the first timepoint be written to. If the name does not start with slash (/), the file will be written in the *netlist* directory. To determine the netlist directory location, select *Setup - Simulator/Directory/Host* in ADE. The *Project Directory* field lists the location of the simulation directory. From this directory, the netlist directory is in <`Circuit Name`>/<`simulator name`>/<*schematic or config*>*/netlist*.

An example is shown below.



Writefinal (Shooting Only)

writefinal writes the final pss timepoint from the shooting window (not the final timepoint of the tstab interval) to the filename specified. See the description for *write* for the location. An example is shown below.

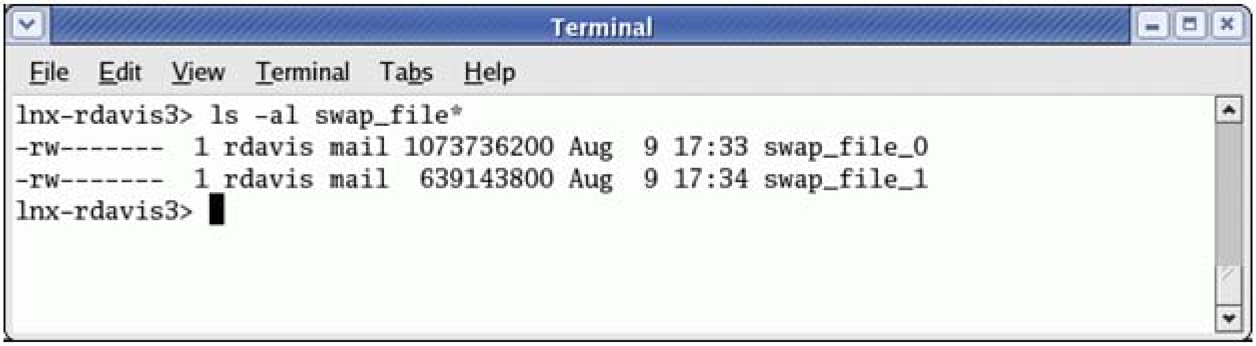
Swapfile (Shooting Only)

During the shooting interval, the solution matrices need to be saved in order to calculate the settled voltages of every node for the next iteration. The solution matrices are the internal solution for all the simultaneous equations for the circuit. The solution vector is also saved, and this is the solution voltages and currents. Normally, this data is kept in memory which makes the Spectre process get larger, starting with the second iteration of pss. On 32-bit systems, a maximum process-size of slightly less than 4GB limits the size of the circuit that can be simulated. Even on 64-bit systems, there is a limit set by the amount of memory that is installed on the system that is running Spectre.

Swapping that is done by the operating system is inefficient because there is no logical relationship for the swapped memory pages. Setting swapfile to a name causes Spectre to write all the solution matrices to the disk in sequential cylinders so that the read time from the disk is faster than using swapping by the operating system. Disk accesses are inherently much slower than memory accesses, however, using swapfile is considerably faster than swapping in the OS. Using swapfile reduces the size of the Spectre process so that it fits in the memory installed in the machine, or allows larger circuits to be simulated. There is a large amount of data that typically is written to the disk. Because of this, the best choice is to write the data in a disk that is local to the machine. If the data is written over a high-latency network, it might actually be better to use the local swap space. Only a trial can determine which is better. Below is an example of setting *swapfile*.

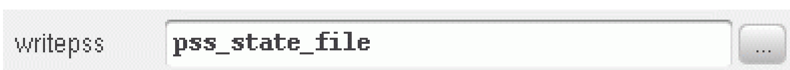
swapfile writes to the netlist directory by default. To locate the simulation directory, select *Setup - Simulator/Directory/Host* and look in the *Project Directory* field. Navigate to that directory, and then navigate to the <*Circuit_Name*>*/spectre/*<*schematic_or_config*>*/*

netlist directory. If the size of the swapfile gets larger than 1GB, it splits into multiple files, as shown below.

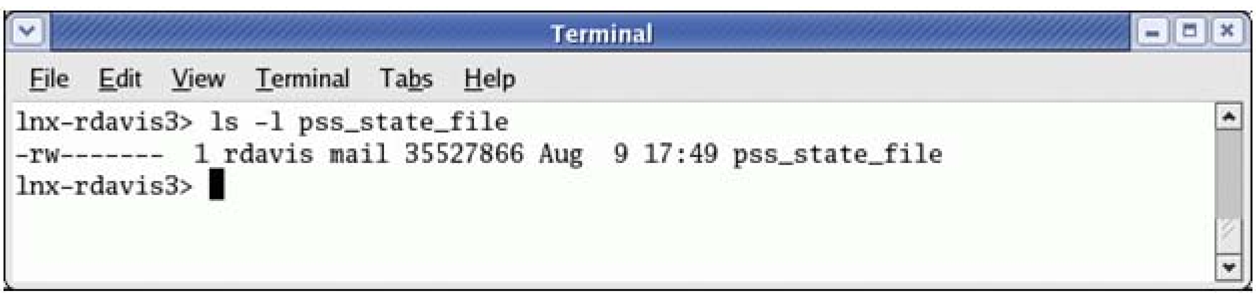


Writepss (Shooting and Harmonic Balance)

This option specifies that the full internal state of the pss analysis be written out to the file specified in the option. An example is shown below.



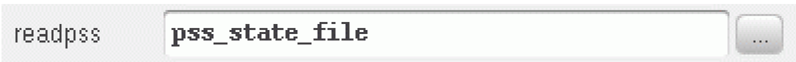
For large circuits, the file can be quite large.



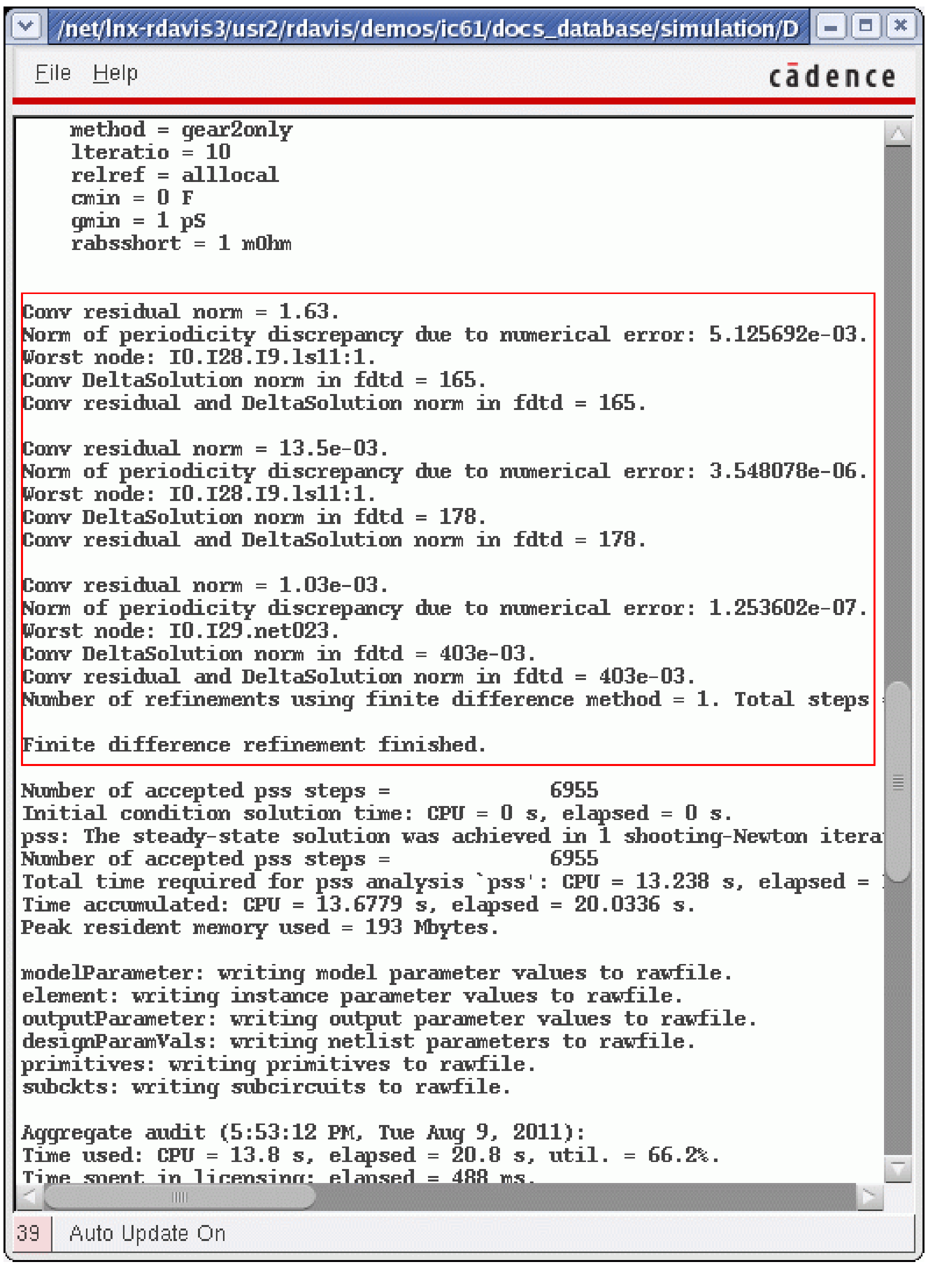
Readpss (Shooting and Harmonic Balance)

This option specifies the file to be read in as a starting point for the next pss analysis. If nothing is changed in the circuit, then only one iteration is performed and small-signal analyses can be run with a much faster time for the pss analysis. Changes can be made to the circuit or analysis options as long as the topology of the circuit stays the same. Changes introduce a discontinuity at the beginning of the pss analysis, which might be large enough to cause the pss to not converge.

An example of *readpss* is shown below.



When a circuit value is changed, a refinement is done automatically. Basically, the pss analysis is re-run. Instead of starting from scratch, it starts from the solution that is in the *writepss* file. This is shown below.

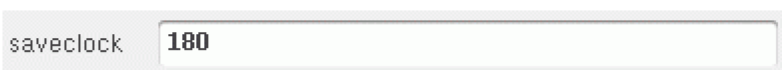


Checkpss (Shooting Only)

Leave this option at the default of *yes*. This option is used in conjunction with the *readpss* option and the *highorder* option. If *readpss* is set to a filename, (The pss is read from a file), and if *highorder* is set to *yes,* only then is *checkpss* read. If *checkpss* is set to *yes* (default), then any changes in the circuit are observed, and the *readpss* file serves as the starting point for the pss iterations. If the *checkpss* option is set to *no*, then none of the changes to the circuit elements are used. The readpss file is used directly with no changes.

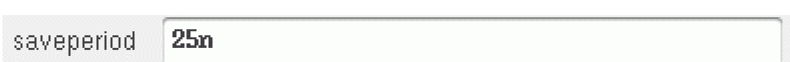
Saveclock (Shooting and Harmonic Balance)

saveclock saves the state of the simulation in the tstab interval at the time interval in seconds specified by *saveclock*. When the clock time has passed in the tstab interval, the file specified by the option *savefile* is created. When subsequent clock time intervals have passed, the file is overwritten. Use only one of the save options at a time. The example below shows *saveclock* set to three minutes.



Saveperiod (Shooting and Harmonic Balance)

saveperiod saves the state of the simulation in the tstab interval at the simulation time interval in seconds specified by *saveperiod*. When the simulation time has passed in the tstab interval, the file specified by the option savefile is created. When subsequent simulation time intervals have passed, the file is overwritten. Use only one of the save options at a time. The example below shows *saveperiod* set to 25 nanoseconds.



Savetime (Shooting and Harmonic Balance)

savetime is a list of times in the tstab interval where the state of the simulation is written out. The list should be specified with spaces between the entries. The information is written out with the filename specified in the *savefile* option with extensions for the time after that. Use only one of the save options at a time.

The example below shows four times in the tstab interval to save.

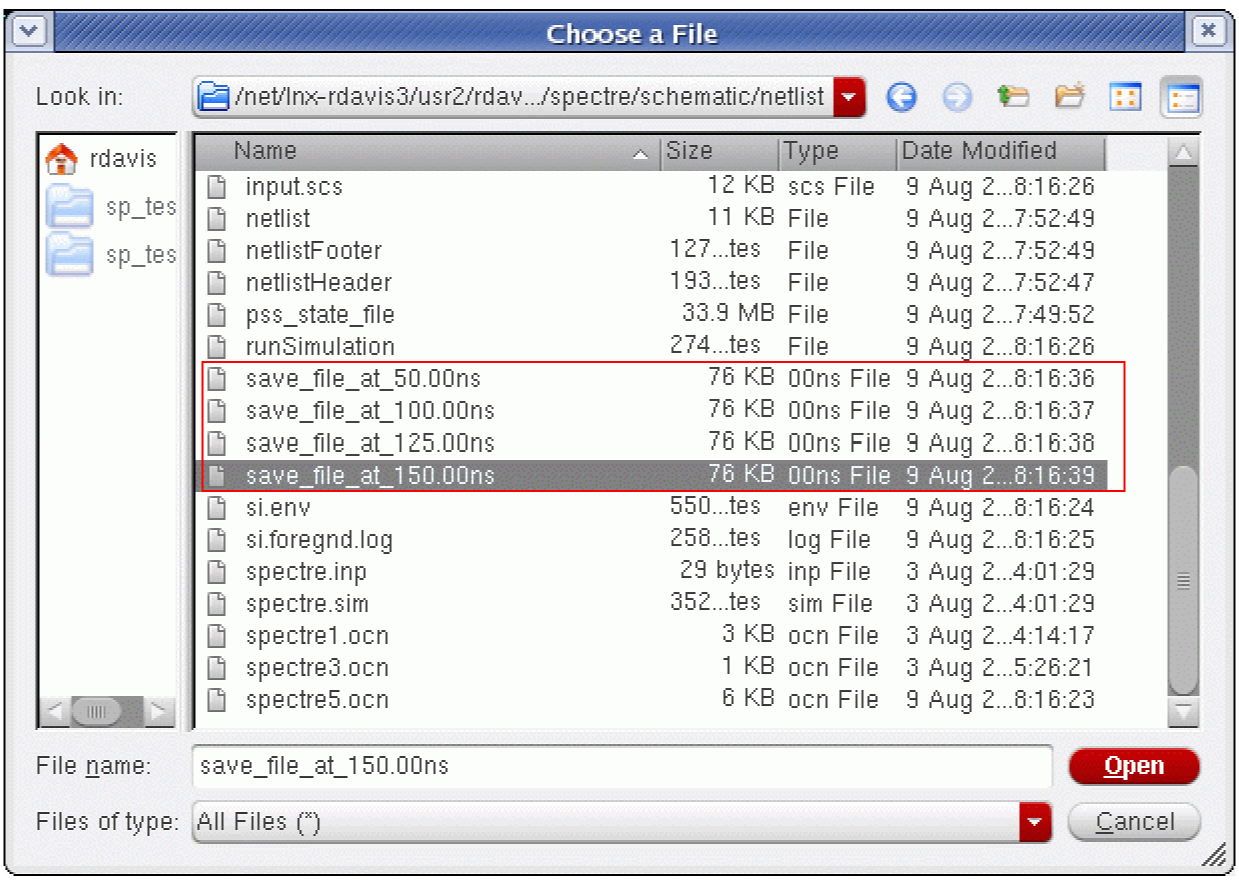
Savefile (Shooting and Harmonic Balance)

This is the file name to write out the tstab state information to. If you do not specify a filename that starts with the slash (/) character, the file is stored in the *netlist* directory. To locate the *netlist* directory, first select *Setup - Simulator/Directory/Host* menu in ADE, and read the path to the project directory. This is the path to the *simulation* directory. In the simulation directory, navigate to the <`circuit_name`>*/spectre/<schematic or config>/netlist* directory.

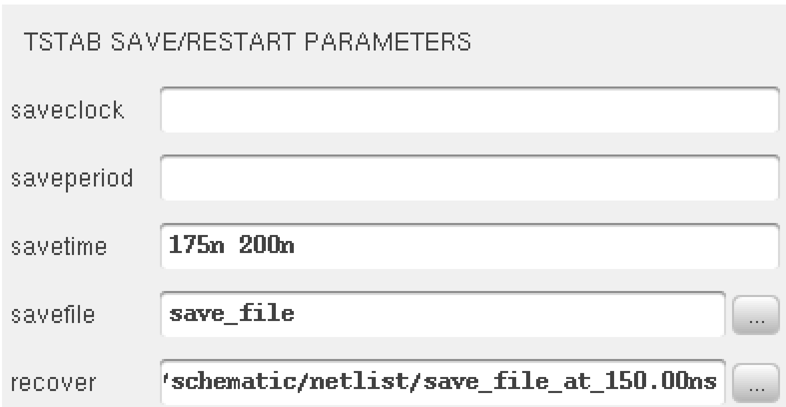
Note: Do not specify a relative path name like *./save_file/run1*.

Recover (Shooting and Harmonic Balance)

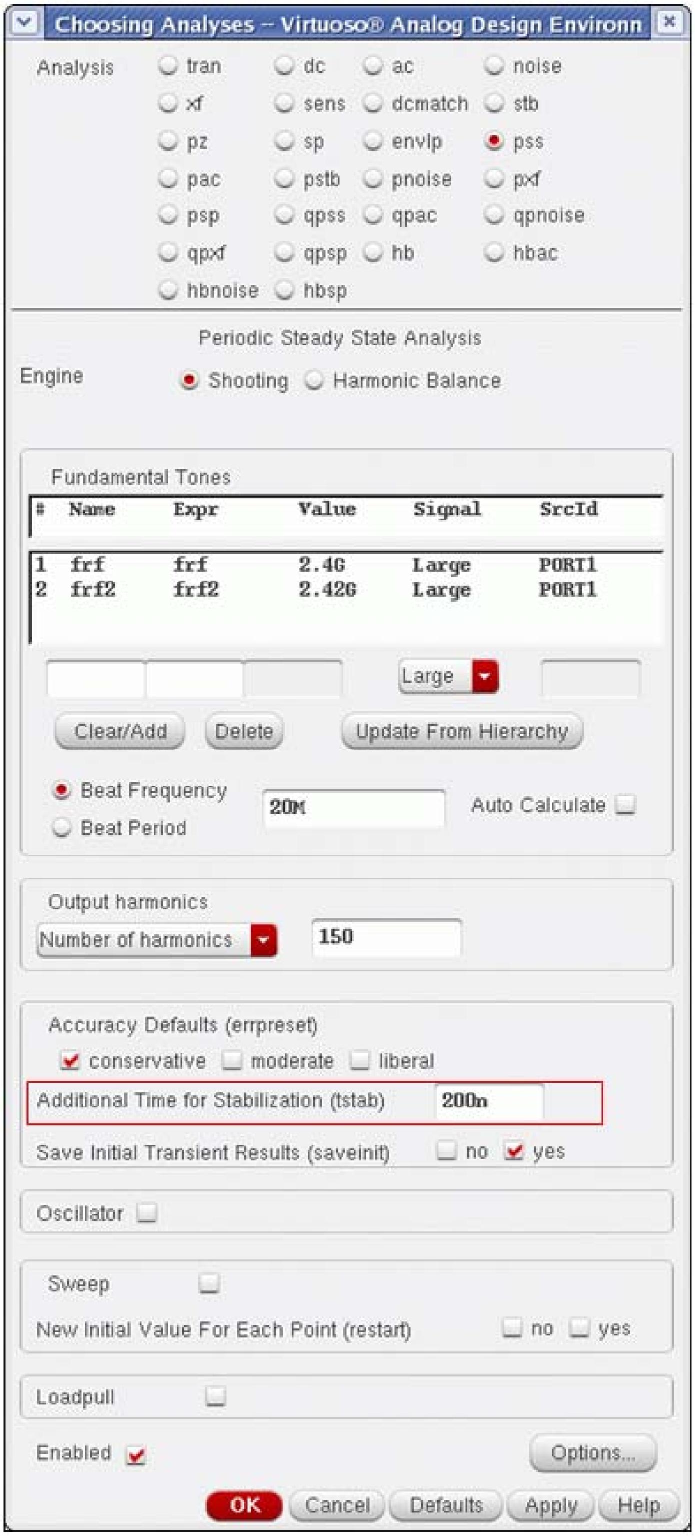
recover specifies the file name to recover the tstab simulation from. If *saveperiod* or *saveclock* are used to make the savefile, then just the same name as specified in *savefile* is used. If *savetimes* has a list of times specified, then several files are created at the times specified in the list. Click (...) and browse to the netlist directory where the savefiles are shown.



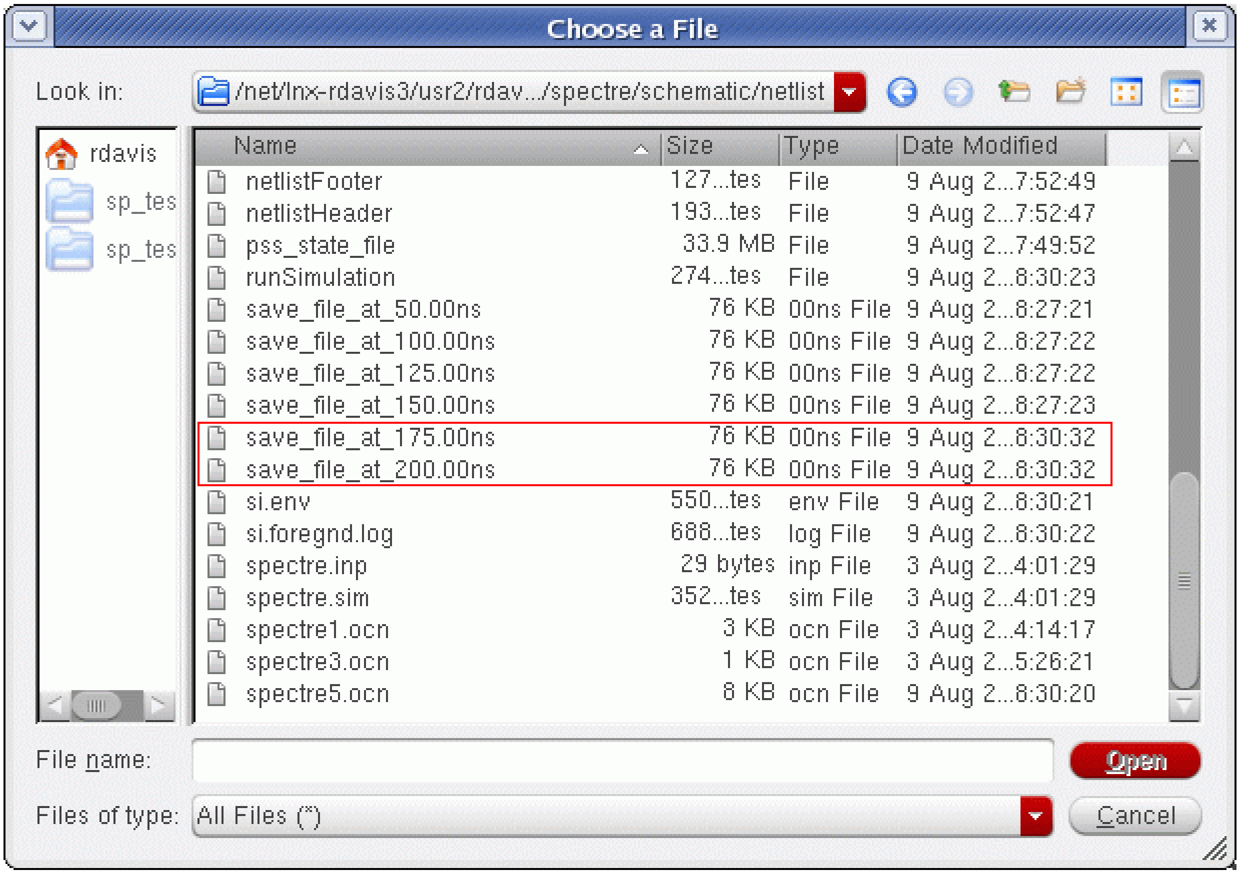
Select one from the list, and click *Open*. This adds the full path to the *recover* option, as shown below. Also note that in the example below, the *savefile* option is still set, and times after the restart times are specified. This is specifically allowed. As long as tstab is 200n or larger, the files at 175n and 200n will be added and can be reused later.



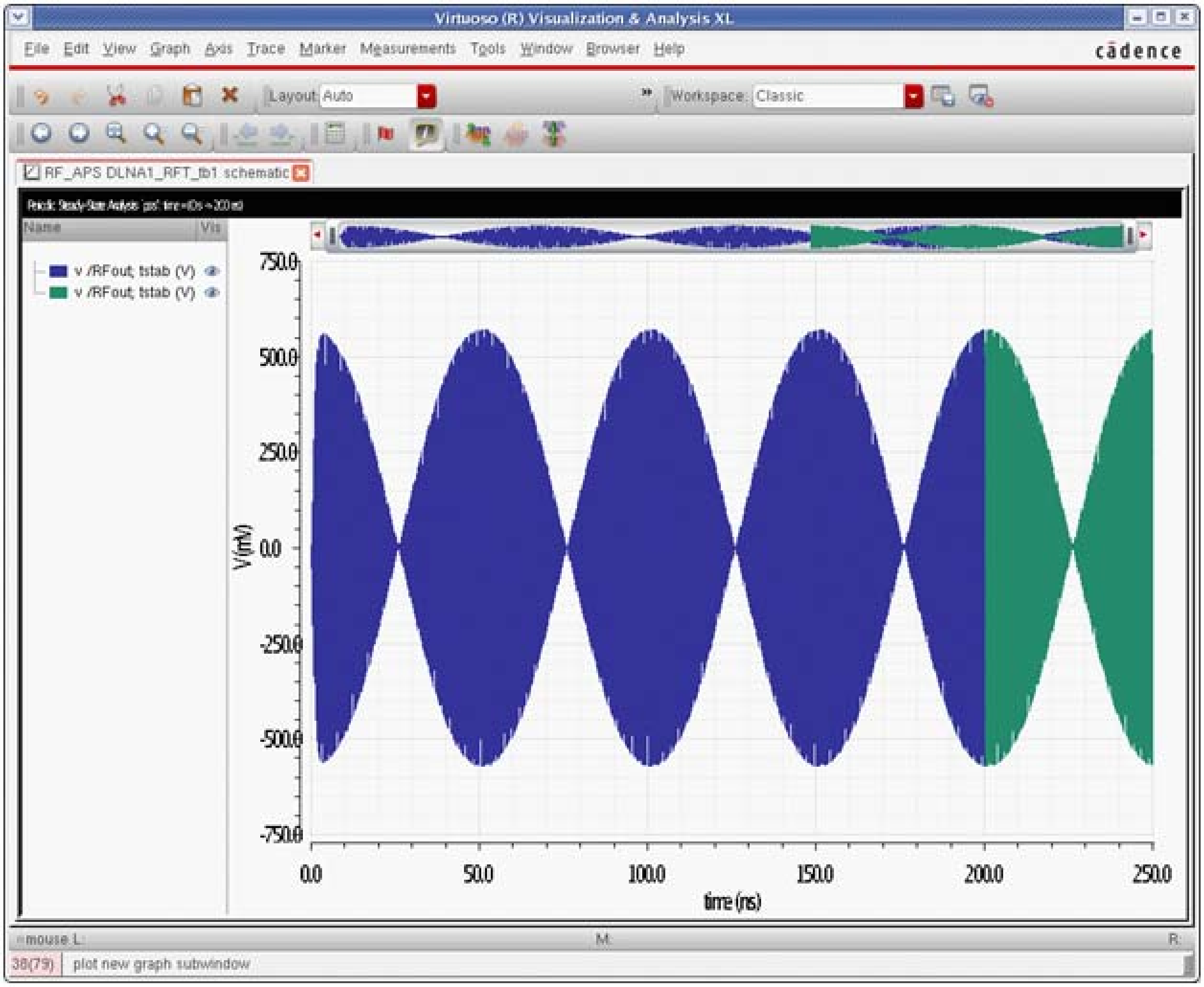
Now change tstab to a later time. In this case 200n was set.



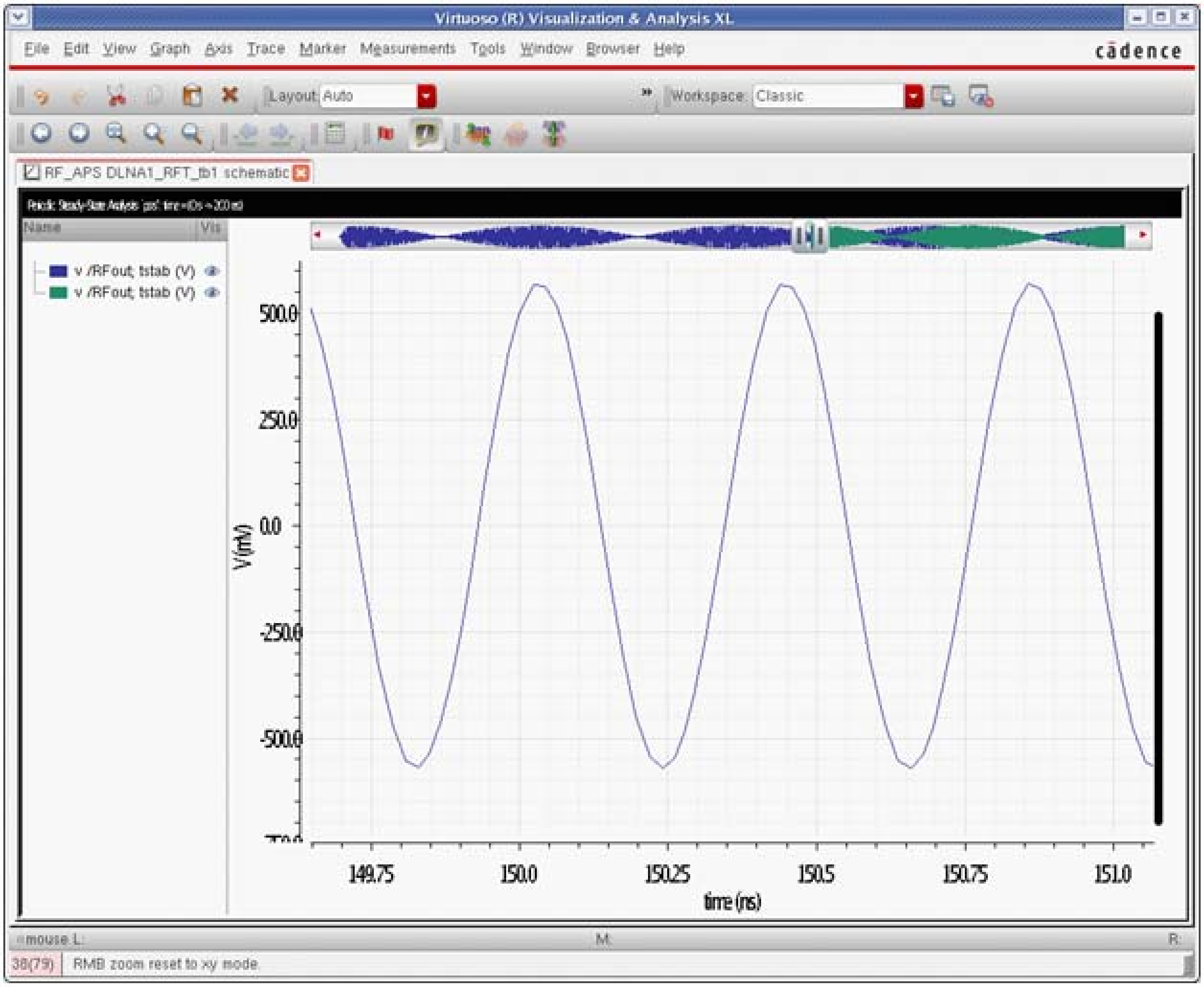
When the simulation runs, the savefiles are created, as shown below.



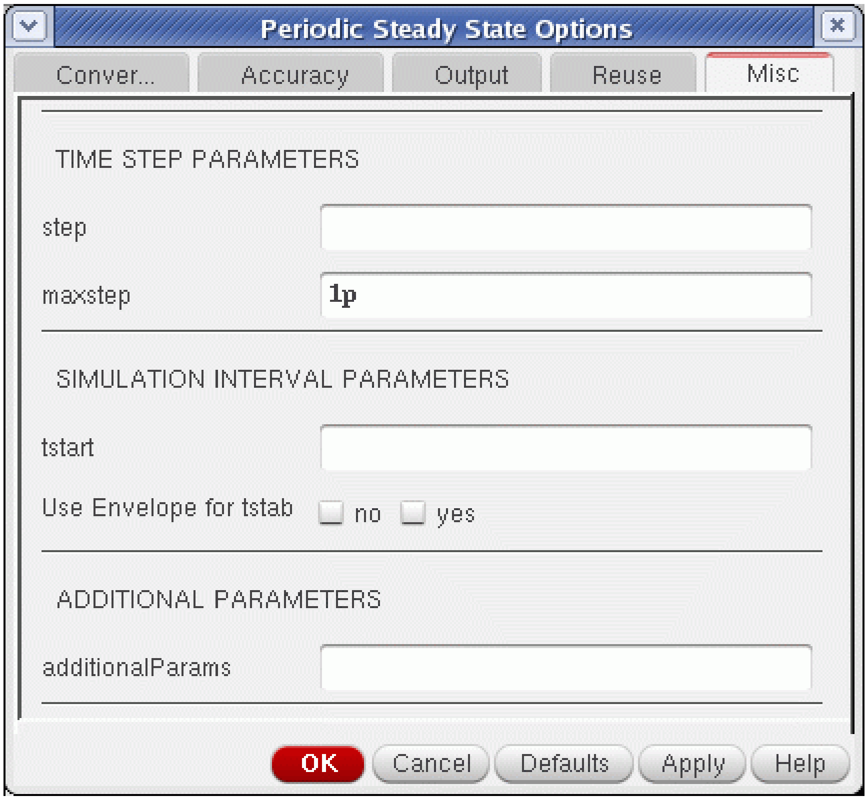
The waveform from the original tstab and the restarted tstab overlay is shown below.



Even in the zoomed-in view, the waveforms overlay, as shown below.



Misc Tab



Step (Shooting Only)

Leave this option at the default. All the nodes are checked for curvature in the local truncation error (Numerical integration error) check. If that node has a capacitor or inductor connected to it, the timestep should be controlled to reduce the numerical integration error. If the node does not have a capacitor or inductor connected to it and it has a lot of curvature, the timestep can only be reduced to the value specified in the *step* option. The default is the simulation interval divided by 1000.

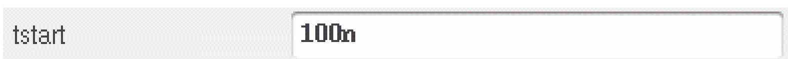
Maxstep (Shooting and Harmonic Balance)

In shooting, *maxstep* is observed in the tstab interval and in the shooting window. In harmonic balance, it is observed in the tstab interval only. Normally, *maxstep* should be left at the default and either setting *errpreset* or changing *reltol* is used to control the accuracy. In some cases, extreme accuracy is needed, and this is accomplished by setting a small maxstep value. A small constant timestep produces maximum accuracy at the cost of runtime and memory consumption.

An example with *maxstep* set to 1 picosecond set is shown below.

Tstart (Shooting and Harmonic Balance)

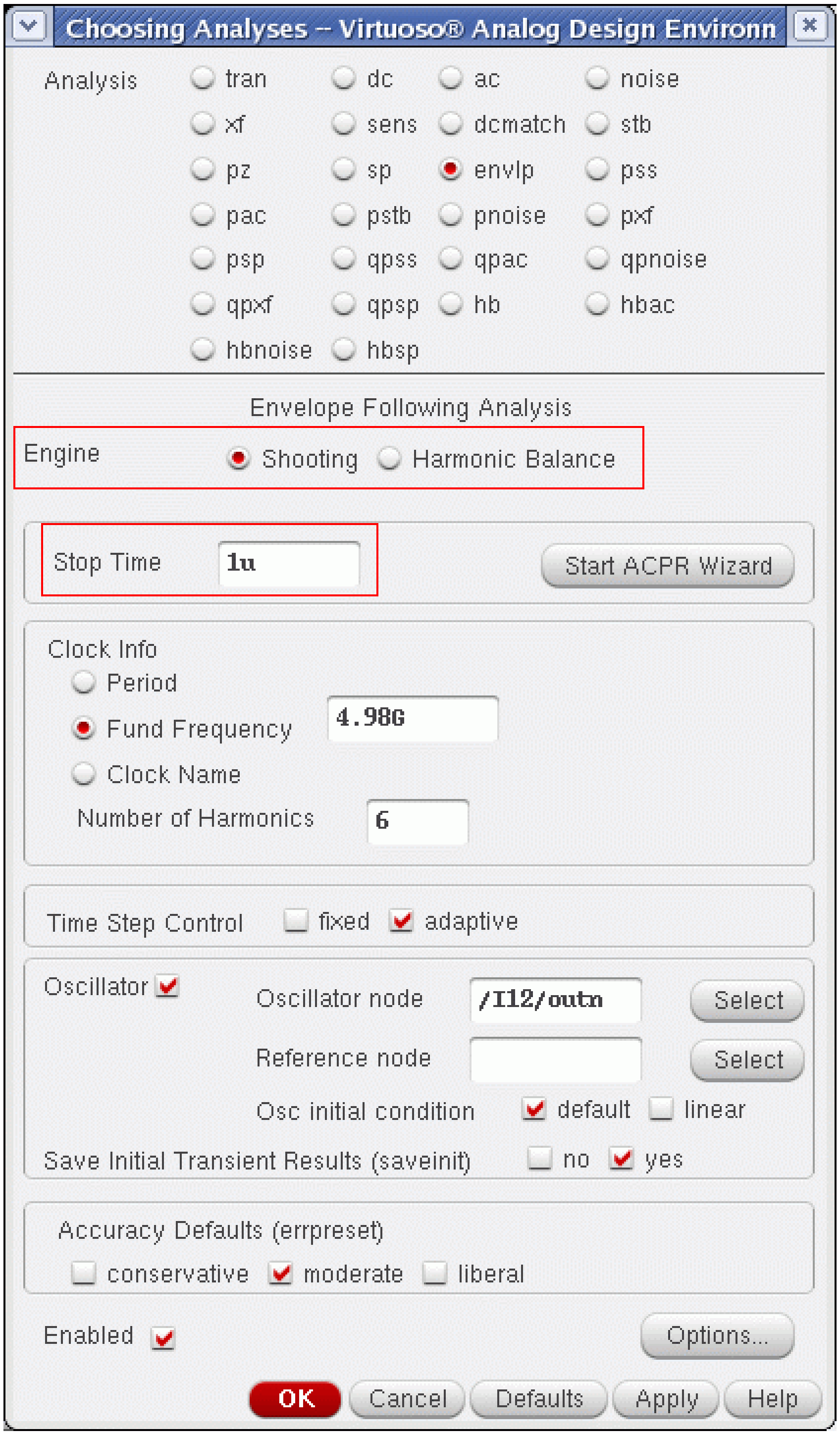
Normally, this is not used. *tstart* specifies the starting time for the tstab interval and defaults to 0 (zero). A negative, 0, or positive *tstart* is allowed. An example with *tstart* set to 100n is shown below.

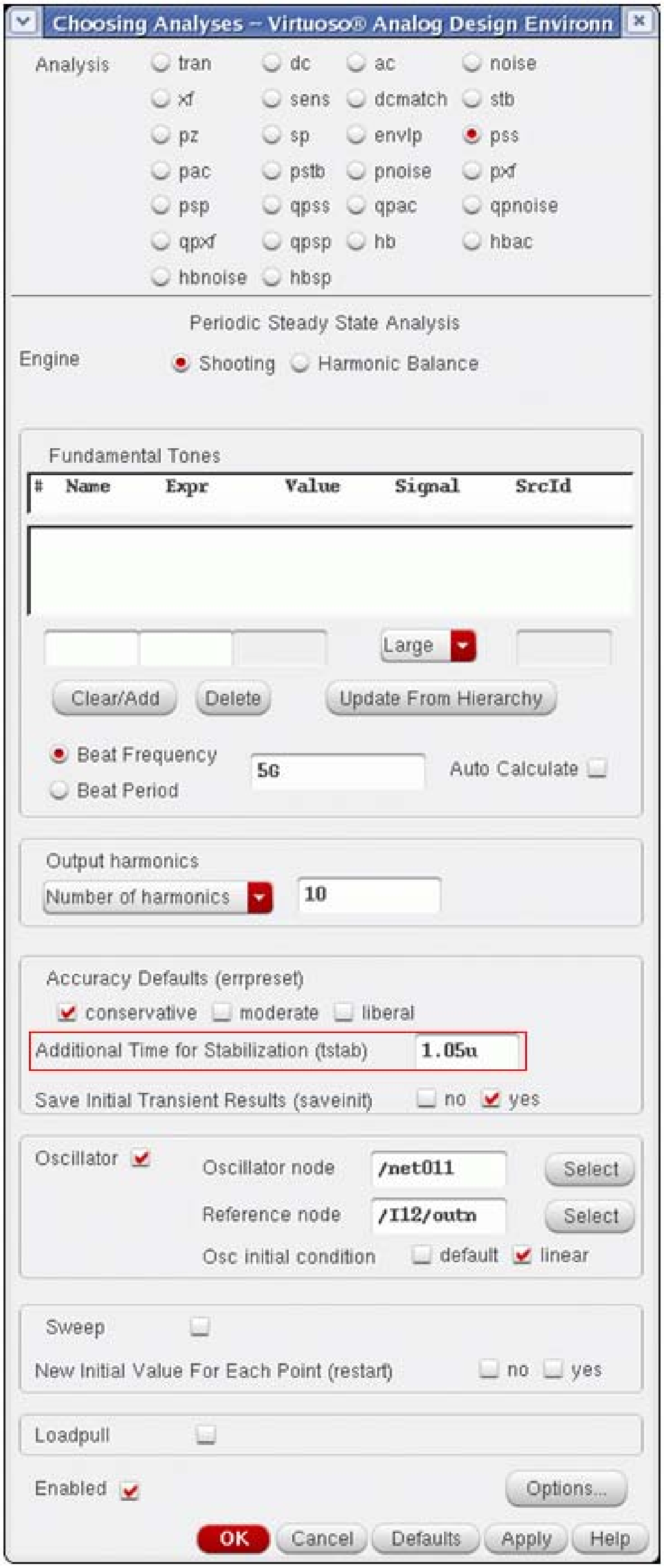


Use envelope for tstab (Shooting and Harmonic Balance)

This option is not recommended because we are considering ending support for this feature. Normally, a transient analysis is used for the tstab interval. In the case of circuits that have high frequencies and long time constants like oscillator amplitude control or crystal oscillators, the time required for the transient analysis can be excessive. In this case, select *yes* for *Use Envelope for tstab*, and set up a shooting envelope analysis. In the pss *Choosing Analyses* form, set tstab slightly larger than the stop time for the envelope analysis. The example below shows the *Use Envelope for tstab* option set to *yes*. For more information on the envelope analysis, see Chapter 6, “Envelope (ENVLP) Analysis”.

The envlp and pss *Choosing Analyses* forms are shown below.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Envelope Following Analysis
Engine
Shooting
Harmonic Balance
Stop Time
1u
Start ACPR Wizard
Clock Info
Period
Fund Frequency
4.98G
Clock Name
Number of Harmonics
6
Time Step Control
fixed
adaptive
Oscillator
Oscillator node
/I12/outn
Select
Reference node
Select
Osc initial condition
default
linear
Save Initial Transient Results (saveinit)
no
yes
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Enabled
Options...
OK
Cancel
Defaults
Apply
Help


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
Beat Period
5G
Auto Calculate
Output harmonics
Number of harmonics
10
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
1.05u
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Oscillator node
/net011
Select
Reference node
/I12/outn
Select
Osc initial condition
default
linear
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

AdditionalParams (Shooting and Harmonic Balance)

This is a field for `<option_name>=value` statements. Multiple `<option_name>=value` statements are allowed with a space between them. Generally, this is used to unlock the beta features, however, there is one case where this field might be useful for an option that is not in the GUI.

In some cases, either in the tstab interval or in the shooting window, the timestep collapses to near zero. The symptom is that the percent complete stays at the same value for many reporting intervals. In this case, there might be a discontinuity in one of the models.

In transient simulation, two things can reduce the timestep; having too much numerical integration error or taking too many iterations at a single timestep can cause the timestep to be cut. Spectre does not report what is causing the timestep to become small. One way to eliminate the possibility of numerical integration error (which is called local truncation error in the simulator) as the cause of reducing the timestep is to set *lteratio* to 1e9. The disadvantage of this is that it disables the normal method of timestep control, therefore, *maxstep* must be set to maintain accuracy.

Another way to accomplish this is to set *lteminstep* to a value of about the stop time divided by 1e5. This allows the normal method of timestep control, but if for some reason the numerical integration error wants to cut the timestep too small, the error is ignored. *lteminstep* is the way to specify this. To do this in the ADE, type `lteminstep=<Stop_Time>/1e5`. If you use `lteratio=1e9`, or if you set *lteminstep* and you still have very small timesteps, try increasing *maxiters* in the *Convergence* tab to between 40 and 100. When *lteratio* or *lteminstep* is set and you still have timestep problems, there is likely a discontinuity in either a device model or in a Verilog-A model. An example is shown below.

Help for Convergence Issues

Spectre can have difficulties in three areas:

- **Trouble with the time-zero timepoint**

  Occasionally, Spectre begins the simulation process, reports the inventory, and then reports that it is starting the tstab interval, and then appears to hang. If you have access to a CPU monitor, you will find that the CPU is at nearly 100% utilization. The likely cause is that the voltages from the signal sources at time=zero are non-zero, and the DC algorithm for calculating the time-zero timepoint is stuck either in `homotopy=source` or `gmin`. In this case, use the global options form (*Simulation - Options - Analog* in the

ADE) and set the *homotopy* option in the *Algorithm* tab to *dptran*. This is highly likely to resolve the problem. If this is ineffective, try setting *try_fast_op* to *no* in the global options *Algorithm* tab. If the problem persists, set `skipdc=sigrampup` and skip the time-zero calculation entirely.

- **Trouble in the tstab interval**

  The typical problem here is that in some cases, the timestep collapses to near zero, and as a result, the simulation appears to be stuck at a certain point. If this is the case, first try setting *relref* to *sigglobal*. This causes the accuracy to degrade a bit, but it allows the simulation to get past the troublesome point. If this fails, in the pss options form set the *lteminstep* option to about the tstab time divided by 1e5. If tstab is 100n, then type `lteminstep=100n/1e5` in the *additionalParams* field, and try again. If this does not fix the problem, there is likely a discontinuous model in the circuit. To be sure, set `maxiters=100`, `lteratio=1e9`, and `maxstep` about equal to `tstab/1e4`. Now run the simulation. If the problem persists, there is very likely a discontinuous model somewhere in the circuit. Use the Spectre output on which the nodes do not converge to determine which model has the problem.

  If you see the same problem in the shooting window, use the same series of steps.

- **Trouble in the shooting interval**

  Driven Circuits

  - Check that the option *iabstol* is not set smaller than 1e-12.
  - If there is a frequency divider in the circuit, remember that the input frequency will be divided down. Specify the divided-down frequency as the pss frequency. Remember that pss must simulate one cycle of the periodicity of the circuit in order to converge.
  - Set a longer tstab. A longer tstab allows the circuit to get closer to steady-state and thus converge better.
  - Check the tstab stopping point. For a sine wave, you want to avoid having the tstab waveform stop at a peak or valley of the waveform. For a square wave, the best point to stop at in the tstab waveform is just before the next transition where most of the voltages have relatively settled.
  - Try changing the tstab time by a fraction of a period. Just moving to a different point can help convergence. Generally, the longer the tstab, the higher the probability of convergence.
  - Do not over-tighten *reltol* and *vabstol*. A practical minimum for *reltol* is 1e-6 and for *vabstol*, 1e-8. Although some circuits may allow smaller values, many would not.

  Oscillators

- ❑ Make sure that the oscillator is starting by selecting *yes* for *saveinit* and then viewing the startup waveform. If the waveform looks like a DC level for a while and then starts near the end of the tstab interval, try one of the methods listed in Different ways to start the oscillator on page 591 to start the oscillator more robustly.
- ❑ In some cases, Spectre will find a node that looks like it is a frequency divider output even though there is no divider in the circuit. When this happens frequently, convergence is negatively affected. Look at the starting and stopping times during the shooting iterations to see if this is happening and if you are sure that there is no frequency divider in the circuit, set the estimated frequency in the *Choosing Analyses* form to about 3.5 times the actual frequency of oscillation. The reason this works is that at the end of tstab, one period of the estimate is simulated, and then four periods of the estimated frequency is simulated, and during this interval, Spectre looks for a frequency divided output. Setting the estimated frequency to 3.5 times the real oscillation frequency causes just over one cycle of the real oscillation frequency to be simulated, and so Spectre cannot find a frequency-divided output.
- ❑ If there is a frequency divider in the circuit, make sure that the oscillator node is on the output of the frequency divider and the pss estimated frequency is the divided down frequency.
- ❑ Change tstab by a fraction of a period so that the starting and ending point are different. For sinusoidal oscillators, avoid peaks and valleys. For square wave oscillators, try to end tstab just before a transition. When using this strategy, note that the save and recover options can prevent the need to resimulate from time zero each time a different tstab is used.
- ❑ Try a longer tstab. Sometimes there are long time constants in the bias circuitry. When using this strategy, note that the *save* and *recover* options can prevent the need to resimulate from time zero each time a different tstab is used.
- ❑ Try setting `reltol=1e-5` and `vabstol=3e-8`. Setting tighter tolerances makes the amplitude and frequency estimate from the tstab interval more accurate.
- ❑ Try setting *maxacfreq* to about the frequency times 1000. This forces more timepoints, which makes the simulation more accurate. More time and more memory are required when this is set.
- ❑ If the *conv_norm* number is trending down with no large increases from iteration to iteration, try setting *maxperiods* to the 200 range. If *conv_norm* generally trends down during the additional iterations, set maxperiods to a higher value. If the *conv_norm* trends down, and then suddenly has a large increase, it is unlikely that setting *maxperiods* to a higher value will succeed.

Examples

Generally speaking, this section shows examples using shooting. The reason is that harmonic balance has Chapter 3, which is dedicated to the hb *Choosing Analyses* form. All the capabilities shown are available for shooting and harmonic balance unless otherwise noted.

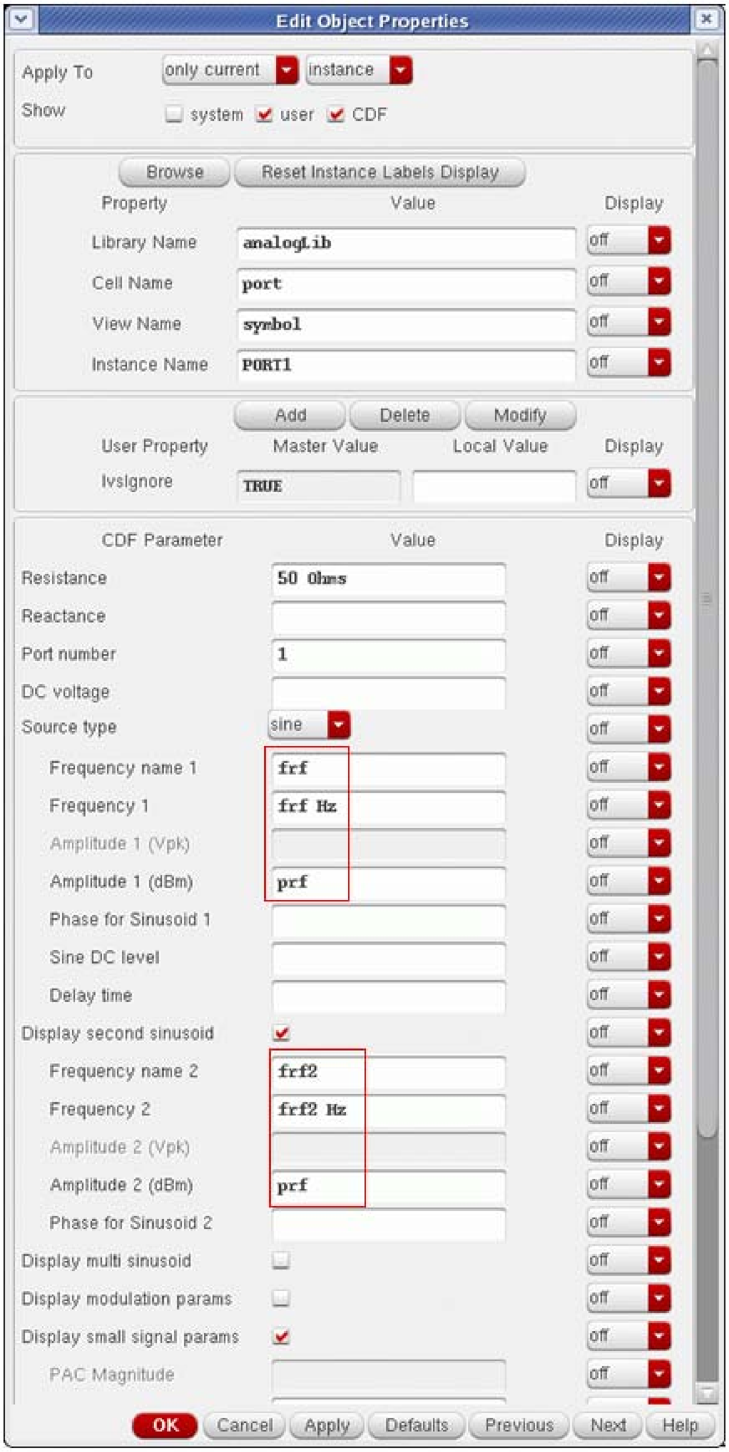
Pss shooting with a single input to an amplifier or the LO applied in a mixer

The example below shows an amplifier.

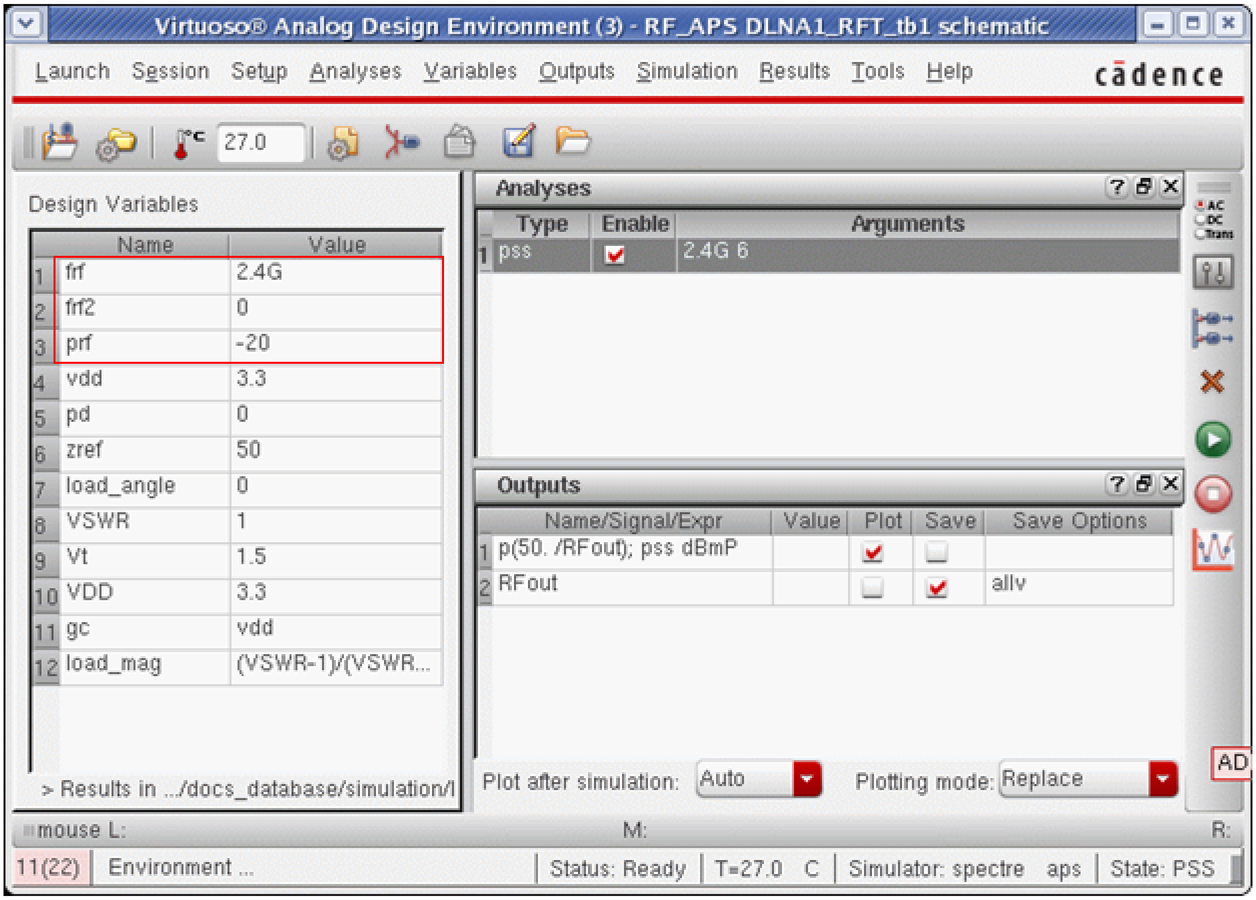
1. In the schematic, for the RF port assign names for the sources in the *Frequency name 1* and *Frequency name 2* fields.

   This is just a name for you to use to remember the function of that signal. In this case, the name and the variable that sets the frequency are the same, but they do not need to agree. It is perfectly acceptable to have a name like RF1 and a frequency variable freq1. Additionally, the amplitude of both RF signals is set to the same variable name. This is not required either. In the case of measuring a large-signal IP3 where you deliberately want to match the amplitudes of both inputs, it is convenient to use the same name in both amplitudes.

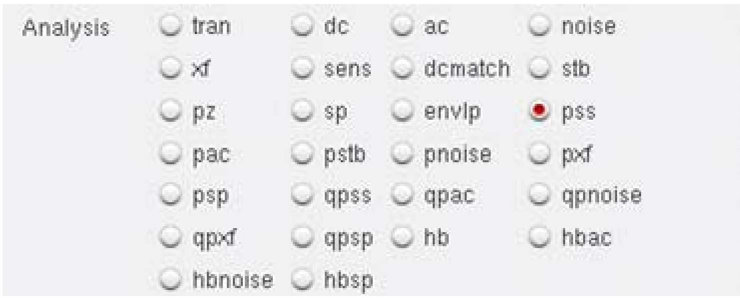
In the case where you want the flexibility to introduce a blocking signal in addition to an RF signal, it makes sense to use two different variables because the blocker usually has a very high amplitude compared to the RF signal.



2. In ADE, assign values for the variables that are used in the input port. In this example, the first RF frequency is set to 2.4GHz, and the second RF frequency is set to 0, which disables that frequency from being generated.

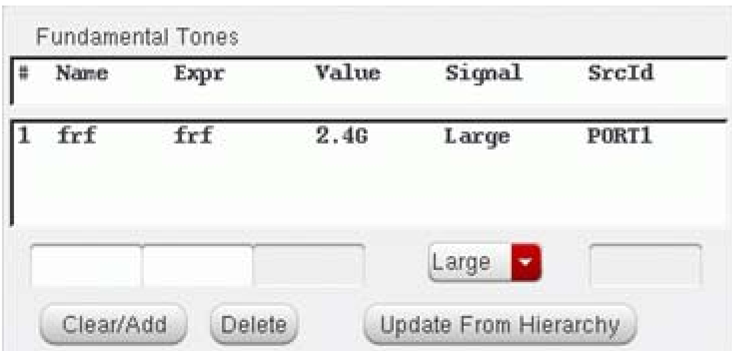


3. In the ADE *Choosing Analyses* form, select pss in the *Analysis* section.

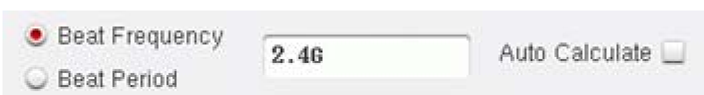


4. Select an engine that you want to use. *Shooting* and *Harmonic Balance* are both available in pss.

5. All the sources in the top-level circuit are read in to the list in the *Fundamental Tones* section. If you have signal sources lower in the level of the hierarchy, select *Update From Hierarchy*.



6. For the beat frequency, either select *Auto Calculate*, or type the frequency for the periodicity of the circuit.



If you are simulating a frequency divider, selecting *Auto Calculate* is inappropriate because it bases the calculation on the frequency of the source in the circuit. In this case, specify the frequency of the divider output.

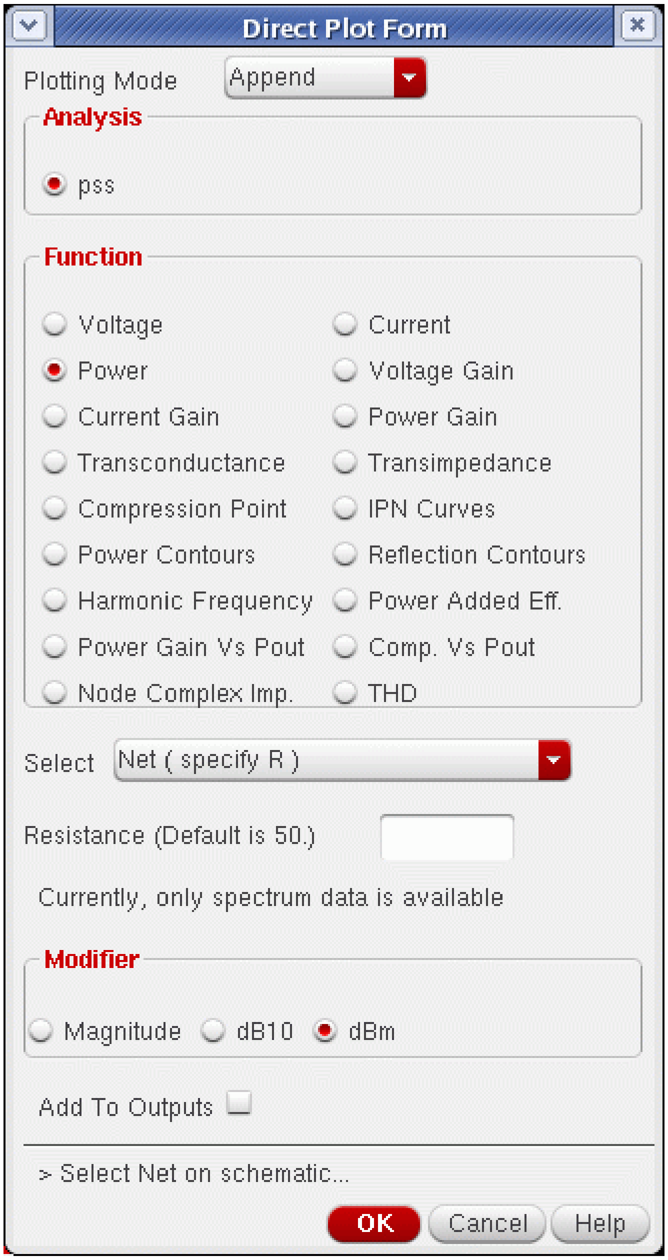
7. Specify the number of harmonics that you want to calculate from the waveform in the *Number of harmonics* field.

If you do not want harmonics to be calculated, type 0 (zero) in the *Number of harmonics* field. When the number of harmonics is raised above 10, more timepoints are forced in the pss which raise the simulation time and the memory required for the simulation.

8. In most cases, moderate accuracy is enough. If you want more accuracy, select *conservative*. If this is not enough, refer to the section Accuracy and Settings and Trade-Offs on page 585.

9. Run the simulation.

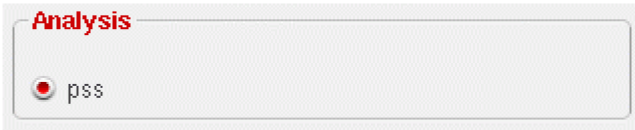
10. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE. The *Direct Plot Form* is displayed, as shown below.



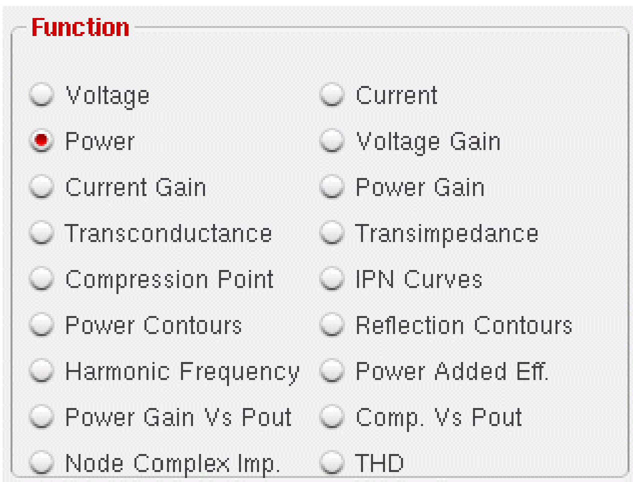
Plotting the output spectrum in dBm

In the Direct Plot form:

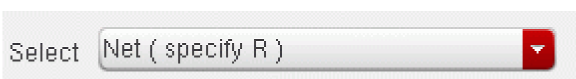
1. Select *pss* results.



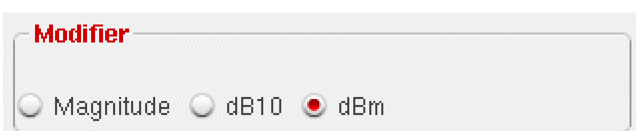
2. Select *Power*.



3. To plot the power in dBm from a voltage on a net with an assumed resistor on that net, select *Net (specify R)* from the *Select* drop-down list.

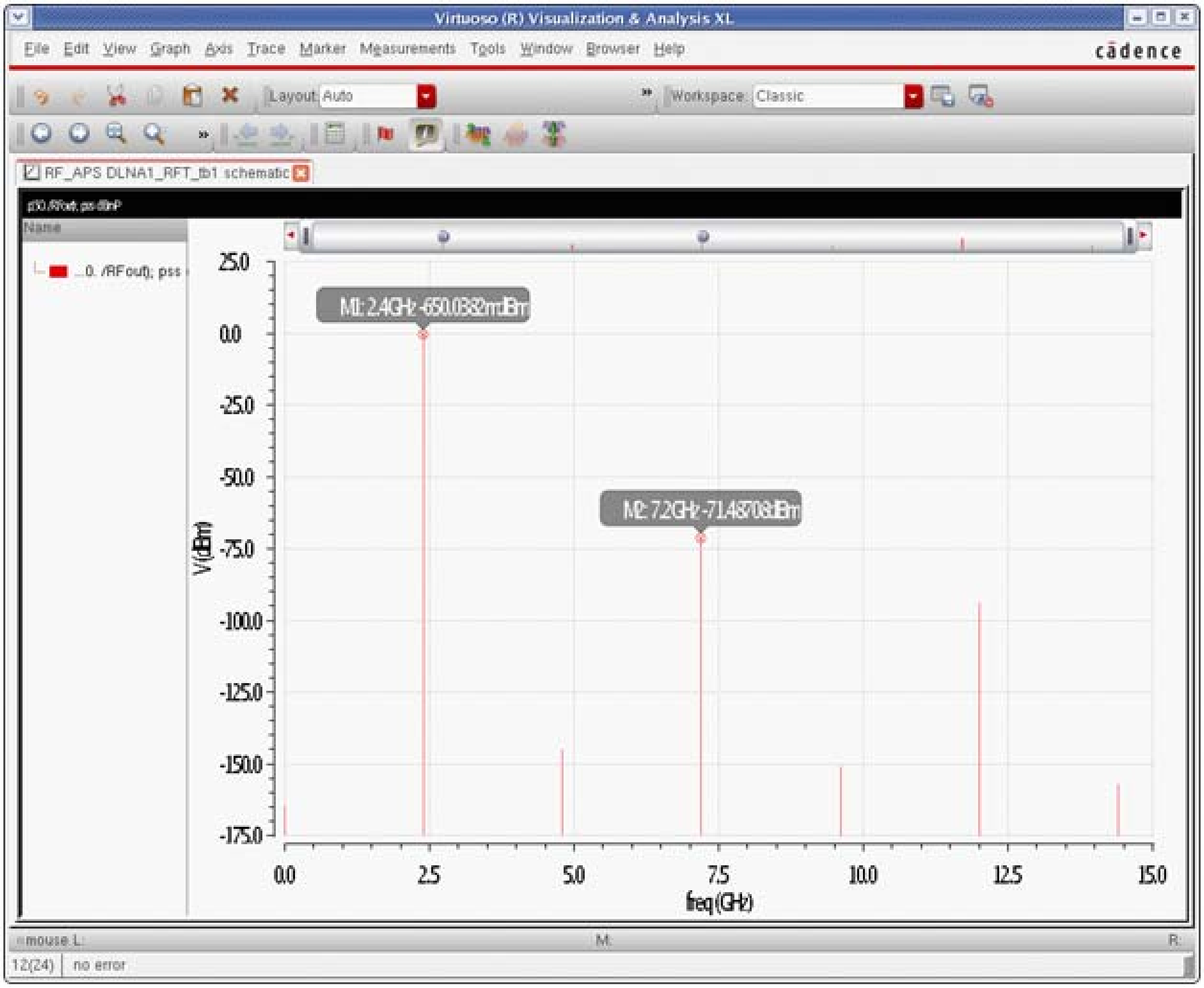


4. Select *dBm*.



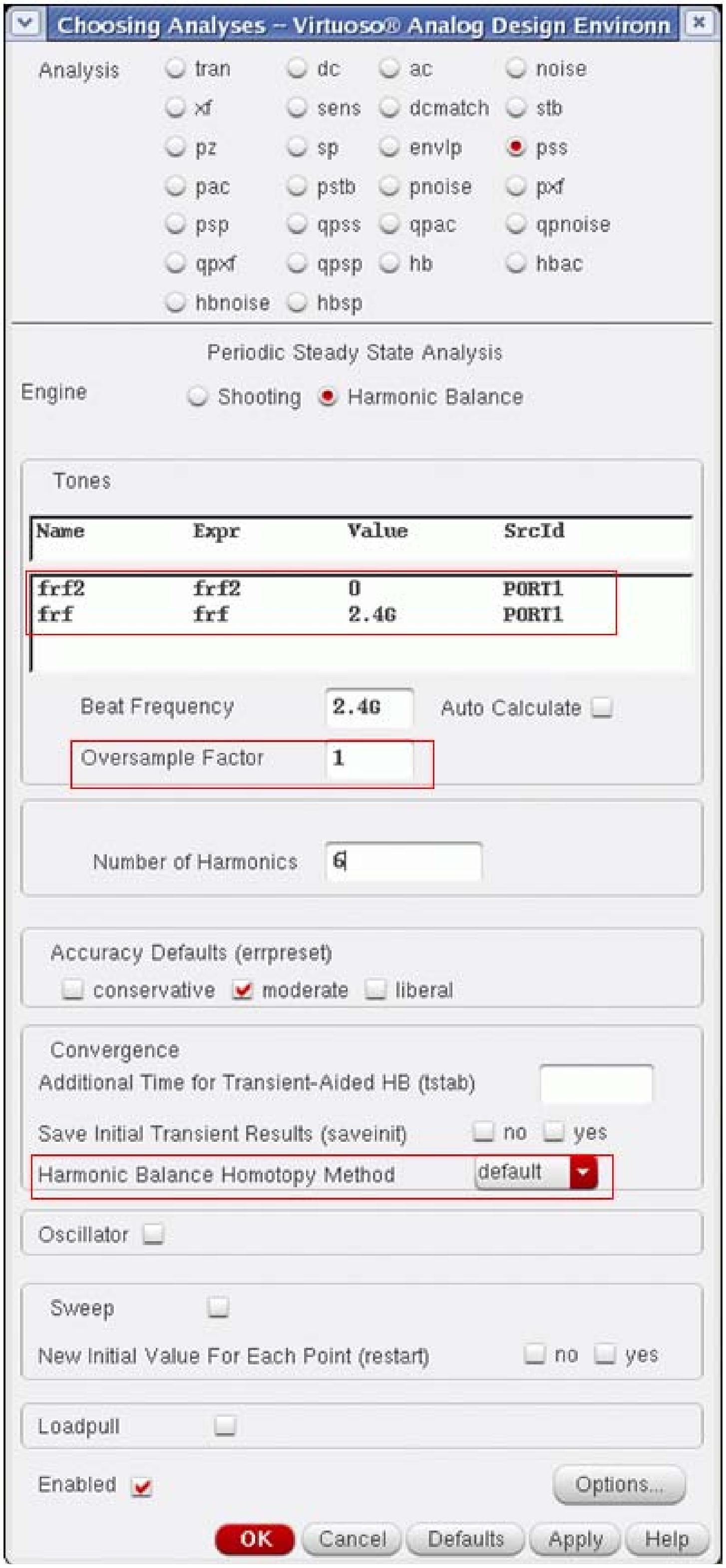
5. Select the net in the schematic you want to plot.

The spectral plot is shown in the waveform tool.

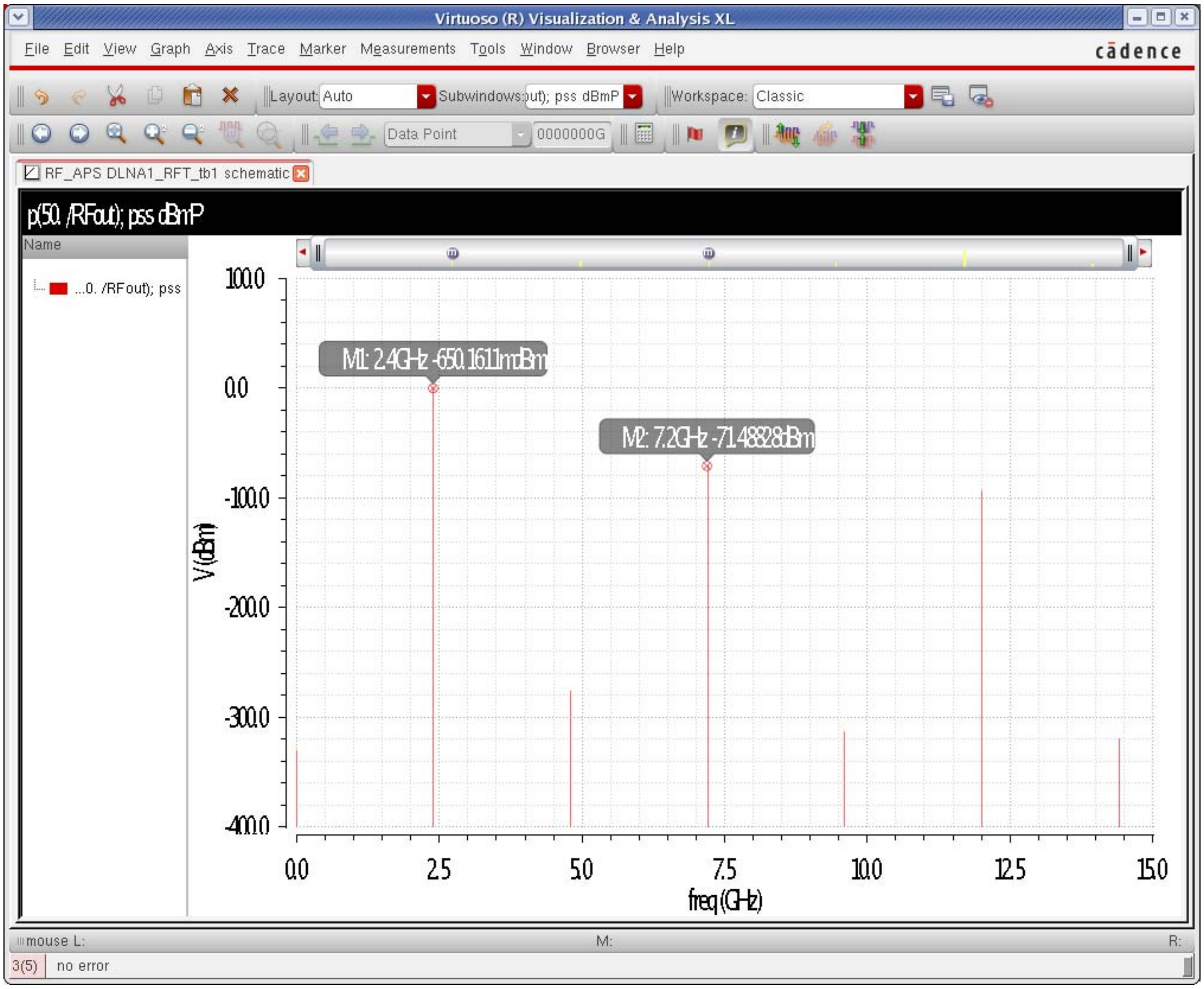


Same Example Using Harmonic Balance

The hb *Choosing Analyses* form is only slightly different from the shooting form. The list of sources in the *Tones* section lists all the sources including the ones that have the frequency set to zero. There is an *Oversample Factor* field. There is also an hbhomotopy choice. For more information on harmonic balance, see Chapter 3, "Frequency Domain Analyses: Harmonic Balance".


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Tones
Name
Expr
Value
SrcId
frf2
frf2
0
PORT1
frf
frf
2.4G
PORT1
Beat Frequency
2.4G
Auto Calculate
Oversample Factor
1
Number of Harmonics
6
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Convergence
Additional Time for Transient-Aided HB (tstab)
Save Initial Transient Results (saveinit)
no
yes
Harmonic Balance Homotopy Method
default
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

The waveform results agree for shooting and harmonic balance, as shown in the waveform tool.

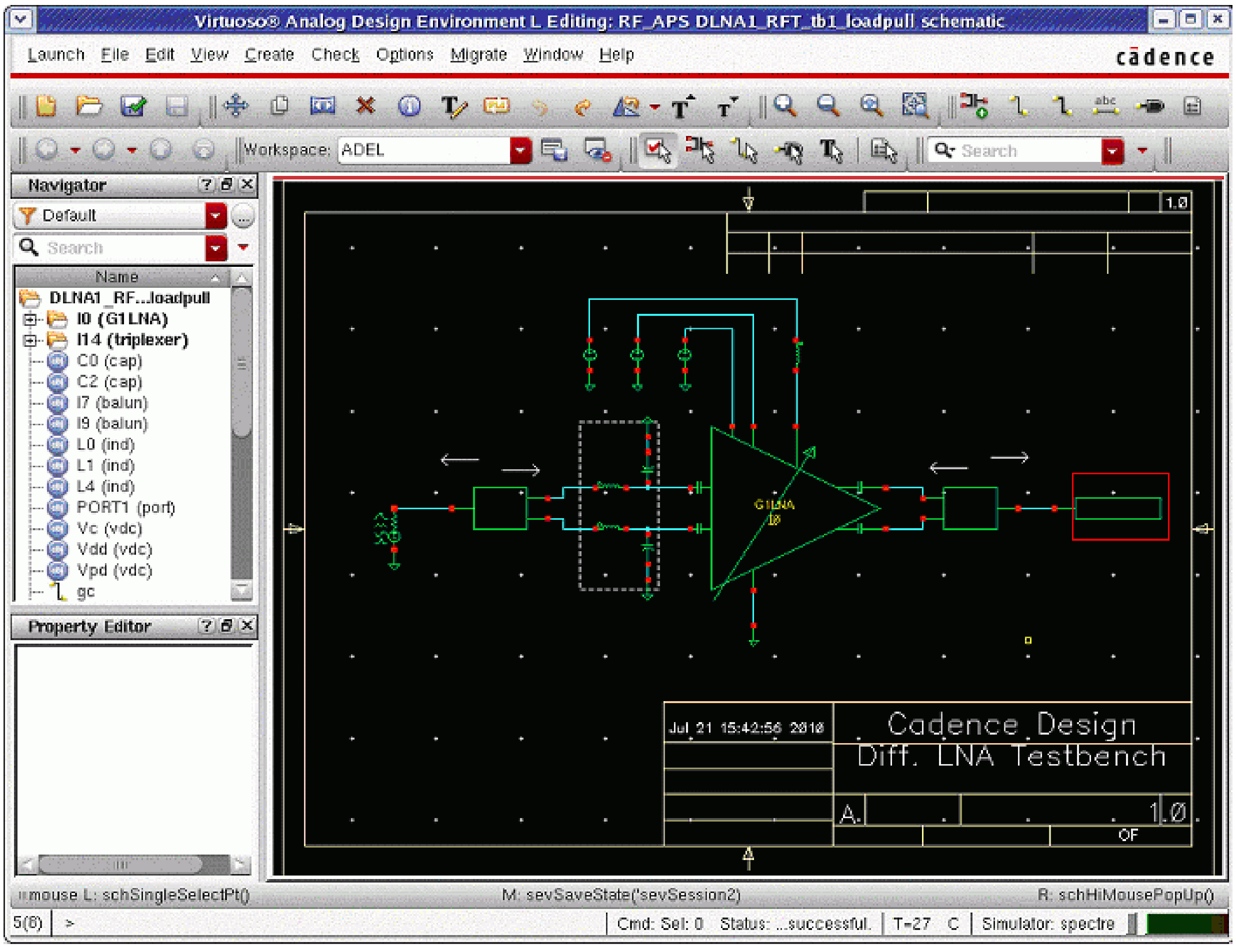


Loadpull

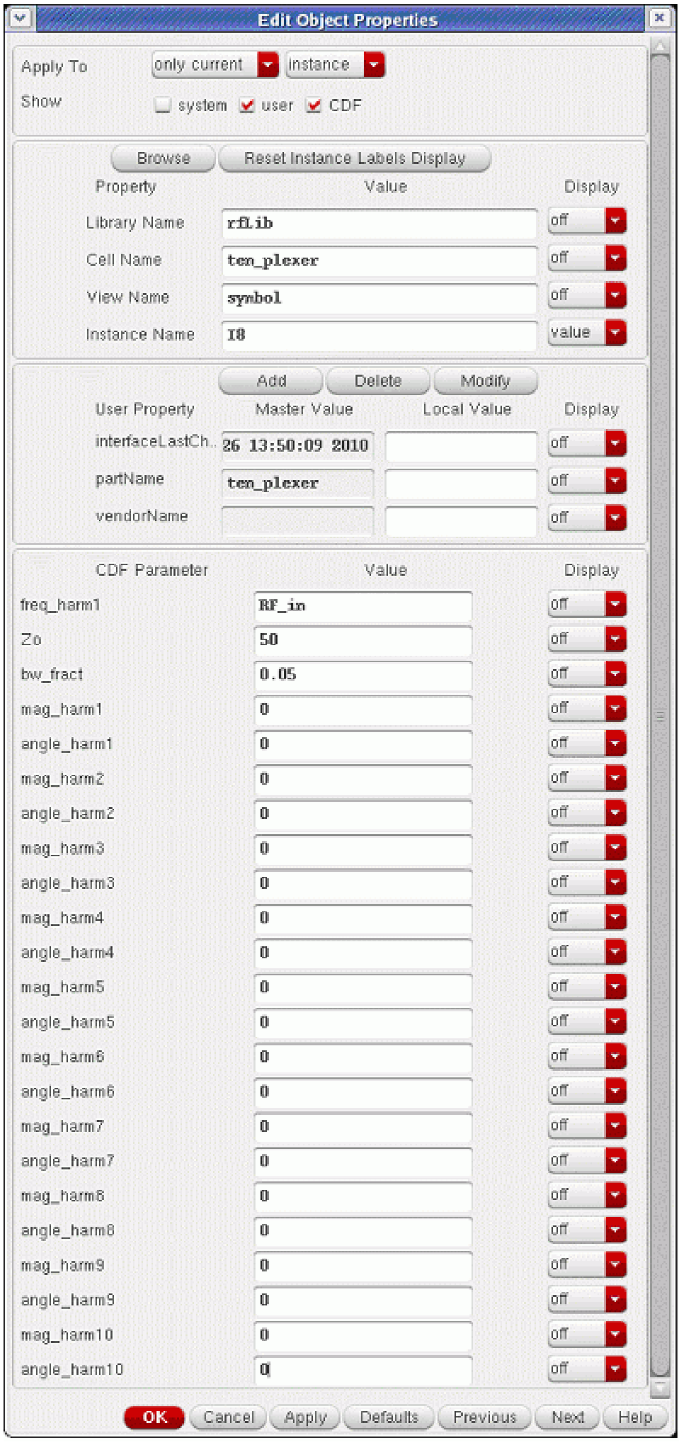
Generally, loadpull is performed using harmonic balance. There are several reasons, but the biggest reason is that harmonic balance is almost always faster than shooting for a loadpull simulation.

PSS Harmonic Balance

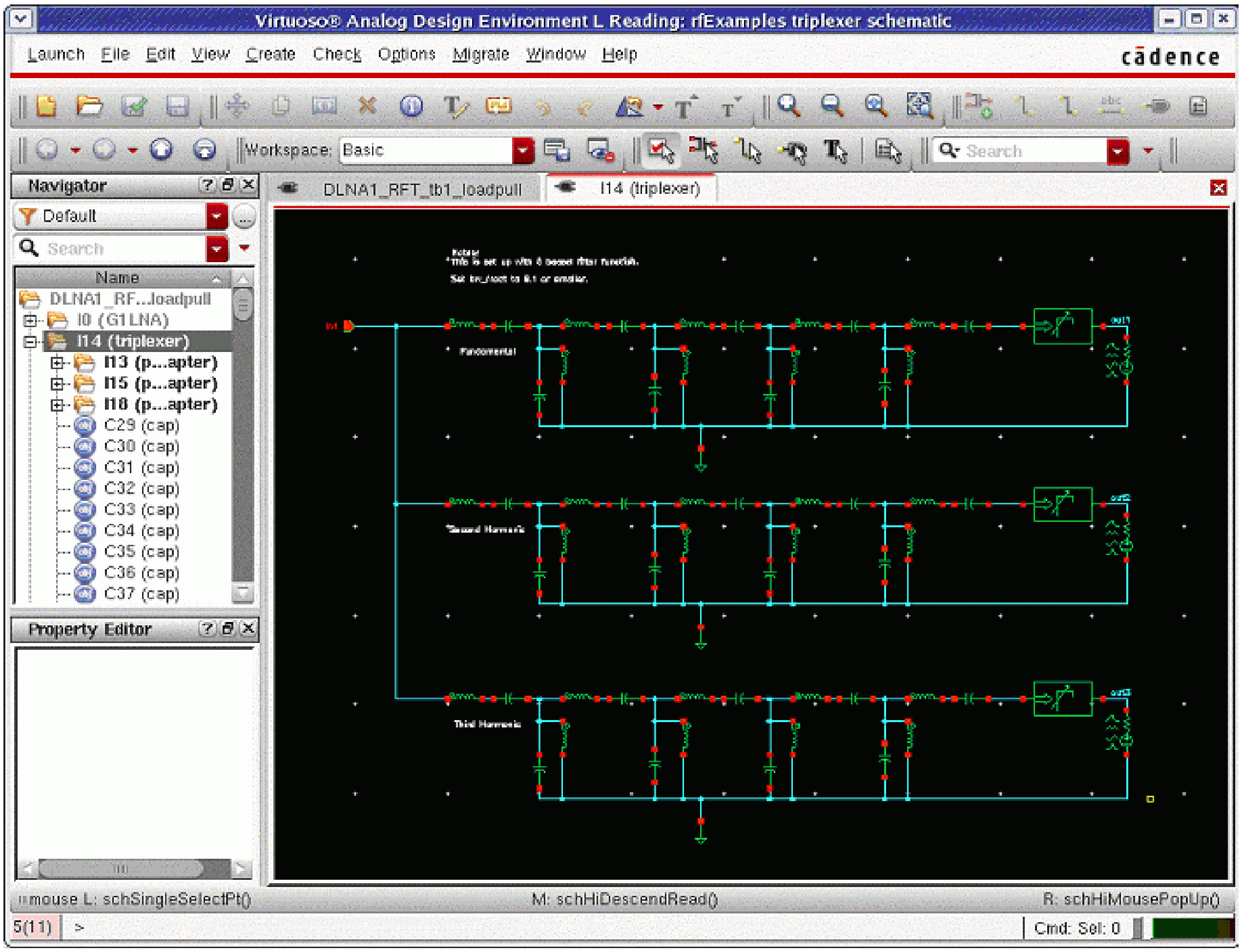
For a loadpull simulation, add a triplexer (for setting the reflection and angle for up to three harmonics) or the ten_plexer (for up to 10 harmonics) as the load to the circuit. These are present in the `rfLib` library.



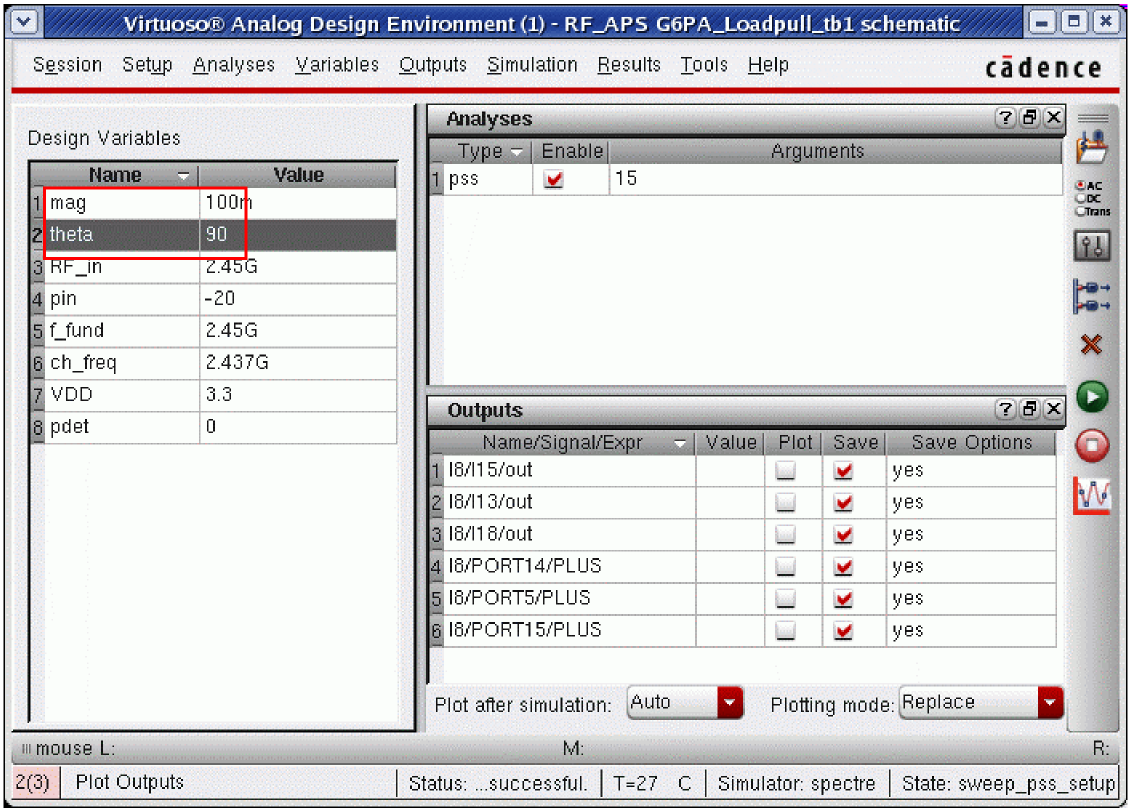
Normally, properties are used to set the reflection coefficient magnitude and phase for the harmonics that are present in the triplexer or ten_plexer. The example below shows the ten_plexer, and has 0 (zero) set for all the reflection coefficients and angles.



Descend-read into the triplexer or ten_plexer. Triplexer is shown. The top bandpass filter is for the first harmonic. The middle is the second harmonic. The bottom is the third harmonic.



1. Add two variables in ADE.

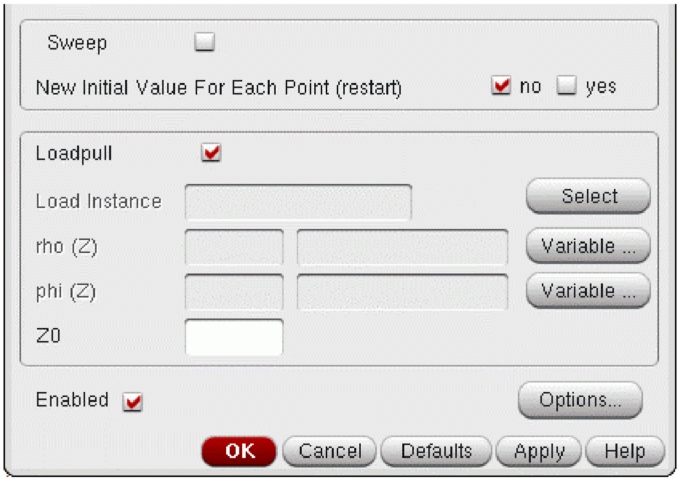


The example above shows `mag` and `theta` variables added. These variables should not be used to set any component values in the circuit.

Triplexer from `rfLib` directory contains three filters for the first, second, and third harmonics and portAdapters with port loads for each of the three filters.

2. Open the PSS or HB *Choose Analyses* form.

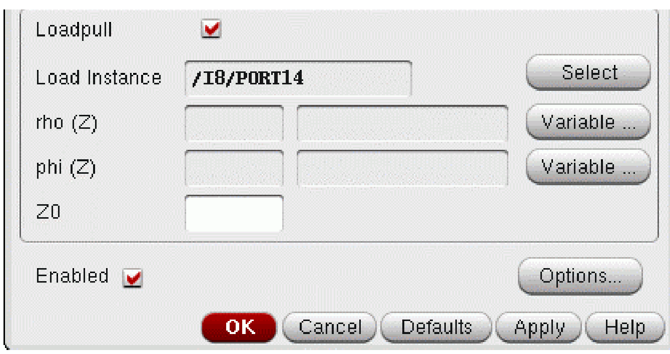
3. Select the *Loadpull* checkbox.



4. Click *Select* on the right of the greyed out *Load Instance* field.

5. In the triplexer schematic, select the topmost port on the right side of the schematic if you want a loadpull for the first harmonic. Select the next port down for a loadpull of the second harmonic. Select the third port down for the third harmonic.

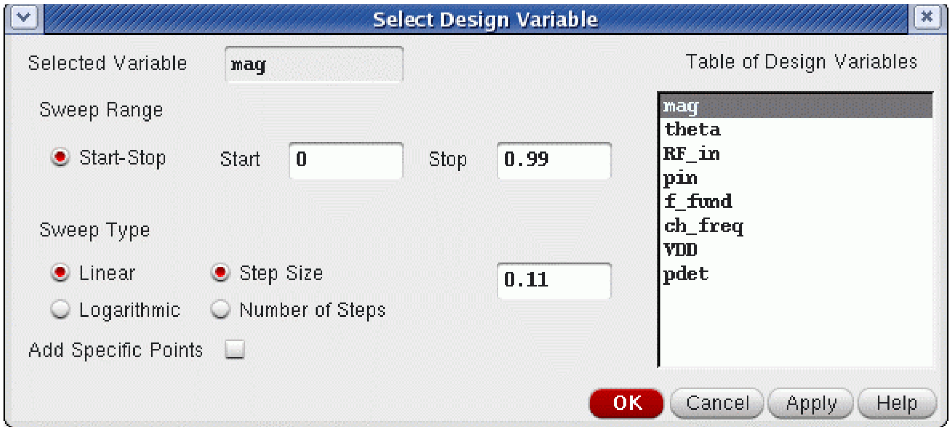
The instance name is entered in the *Load Instance* field. Direct entry by typing is not allowed for this field.



6. Click the *Variable* button on the right of the *rho (Z)* field. Select the variable you entered in ADE.

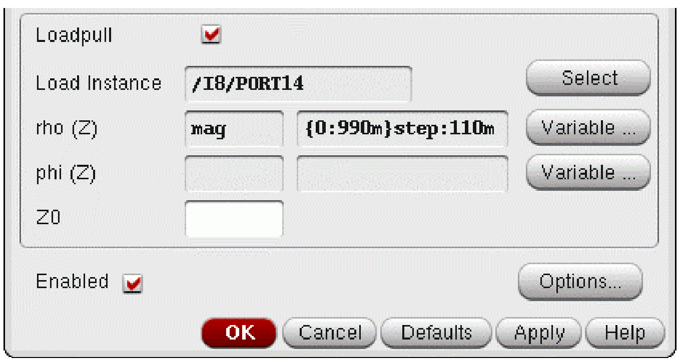
This variable will be used to set the value of the reflection coefficient in the port.

7. Specify a sweep range, and set the spacing.



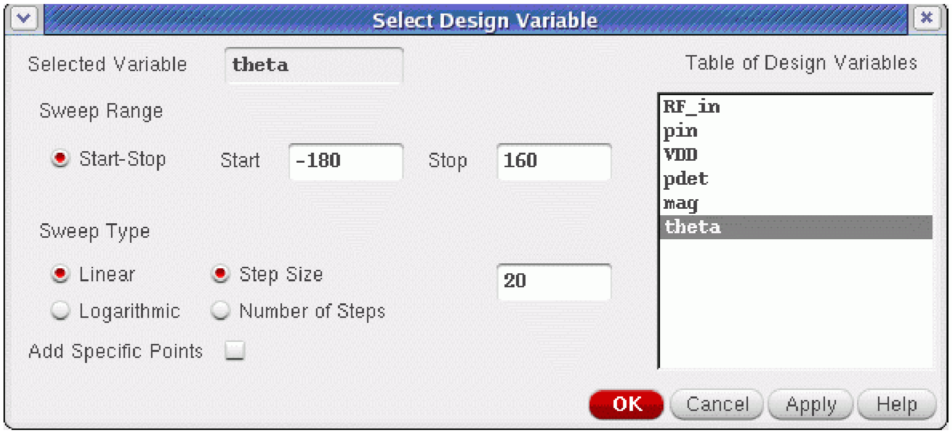
8. When completed, click *OK*.

The values are displayed in the *Choose Analyses* form.



9. Click the *Variable* button on the right of the *phi (Z)* field. This sets the angle of the reflection coefficient.

10. In the *Select Design Variable* window, select the second variable you entered and specify the sweep limits.

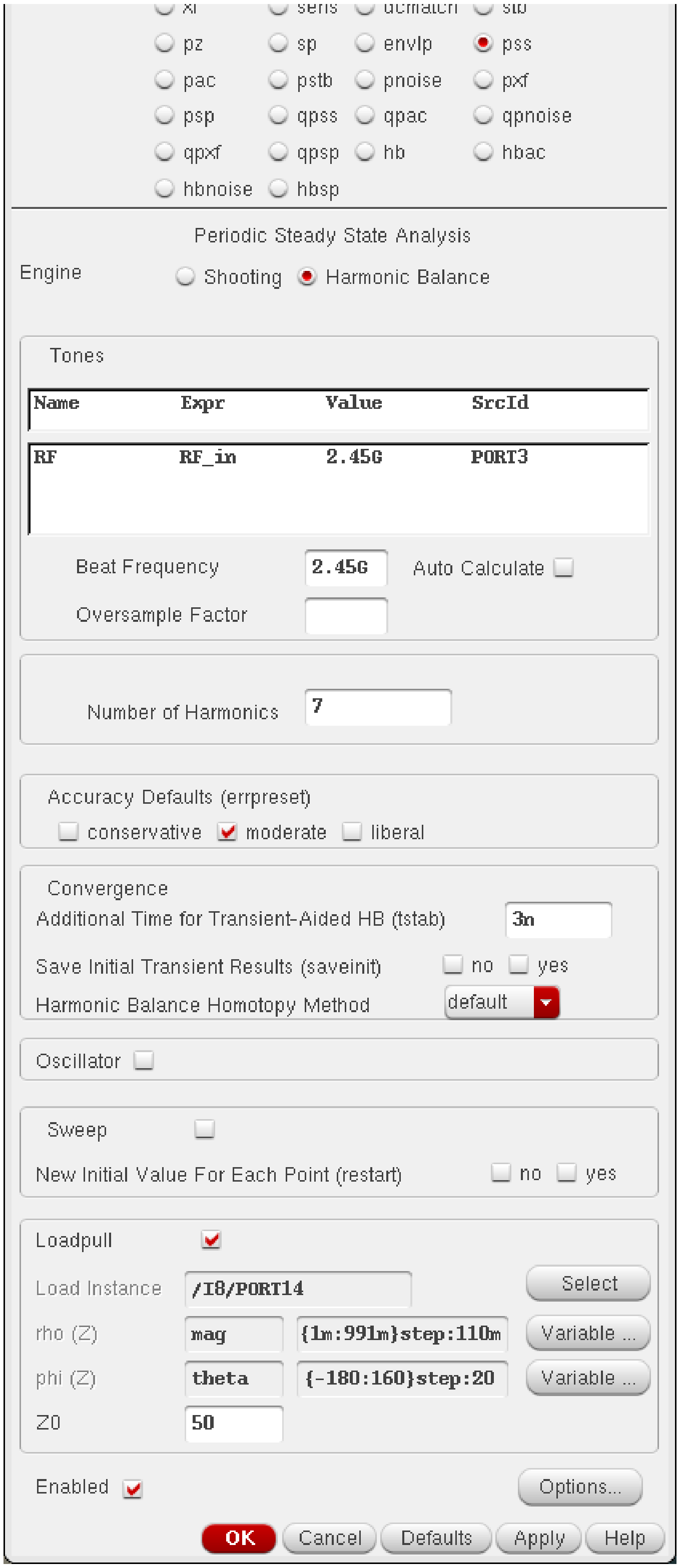


11. When completed, click *OK*.

The *Choose Analyses* form is updated.

12. Specify the system resistance in the *Z0* field.

The *Choosing Analyses* form should appear similar to the one below.


pz sp envlp pss
pac pstb pnoise pxf
psp qpss qpac qpnoise
qpxf qpsp hb hbac
hbnoise hbsp
Periodic Steady State Analysis
Engine Shooting Harmonic Balance
Tones
Name Expr Value SrcId
RF RF_in 2.45G PORT3
Beat Frequency 2.45G Auto Calculate
Oversample Factor
Number of Harmonics 7
Accuracy Defaults (errpreset)
conservative moderate liberal
Convergence
Additional Time for Transient-Aided HB (tstab) 3n
Save Initial Transient Results (saveinit) no yes
Harmonic Balance Homotopy Method default
Oscillator
Sweep
New Initial Value For Each Point (restart) no yes
Loadpull
Load Instance /I8/PORT14 Select
rho (Z) mag {1m:991m}step:110m Variable ...
phi (Z) theta {-180:160}step:20 Variable ...
Z0 50
Enabled Options...
OK Cancel Defaults Apply Help

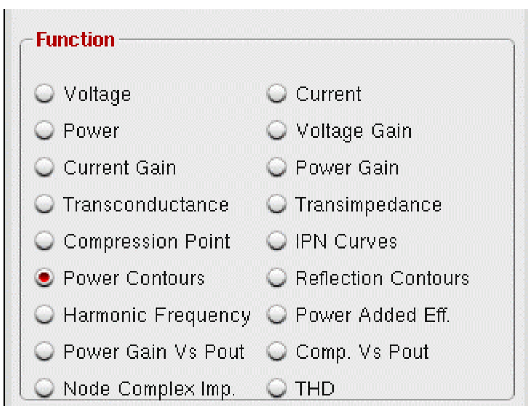
13. When completed, click *OK* and run the simulation in ADE.
14. In ADE, select *Results - Direct Plot - Main Form*.

There are two methods to plot the loadpull curves.

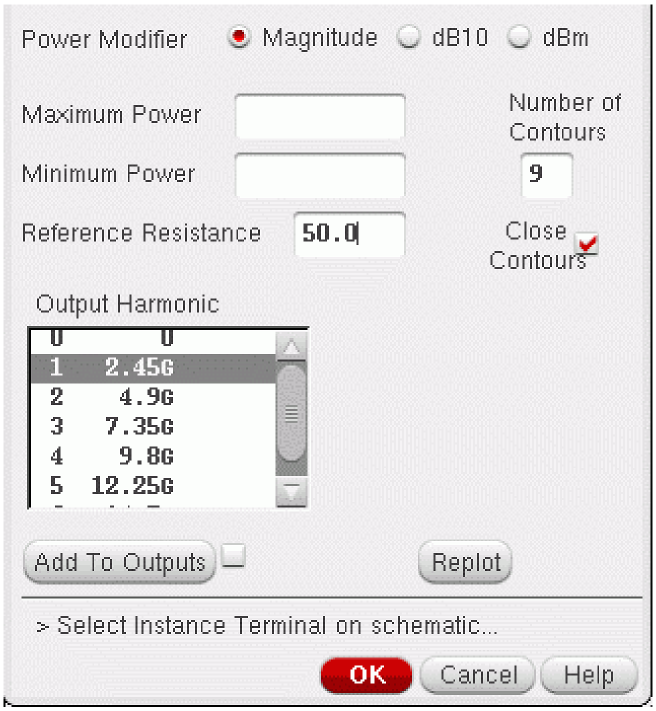
First method

1. Select *pss or hb results*.

2. Select *Power Contours*.



3. Select *Magnitude*, *dB10*, or *dBm*.



4. Specify 2 more than the number of contour lines you want.

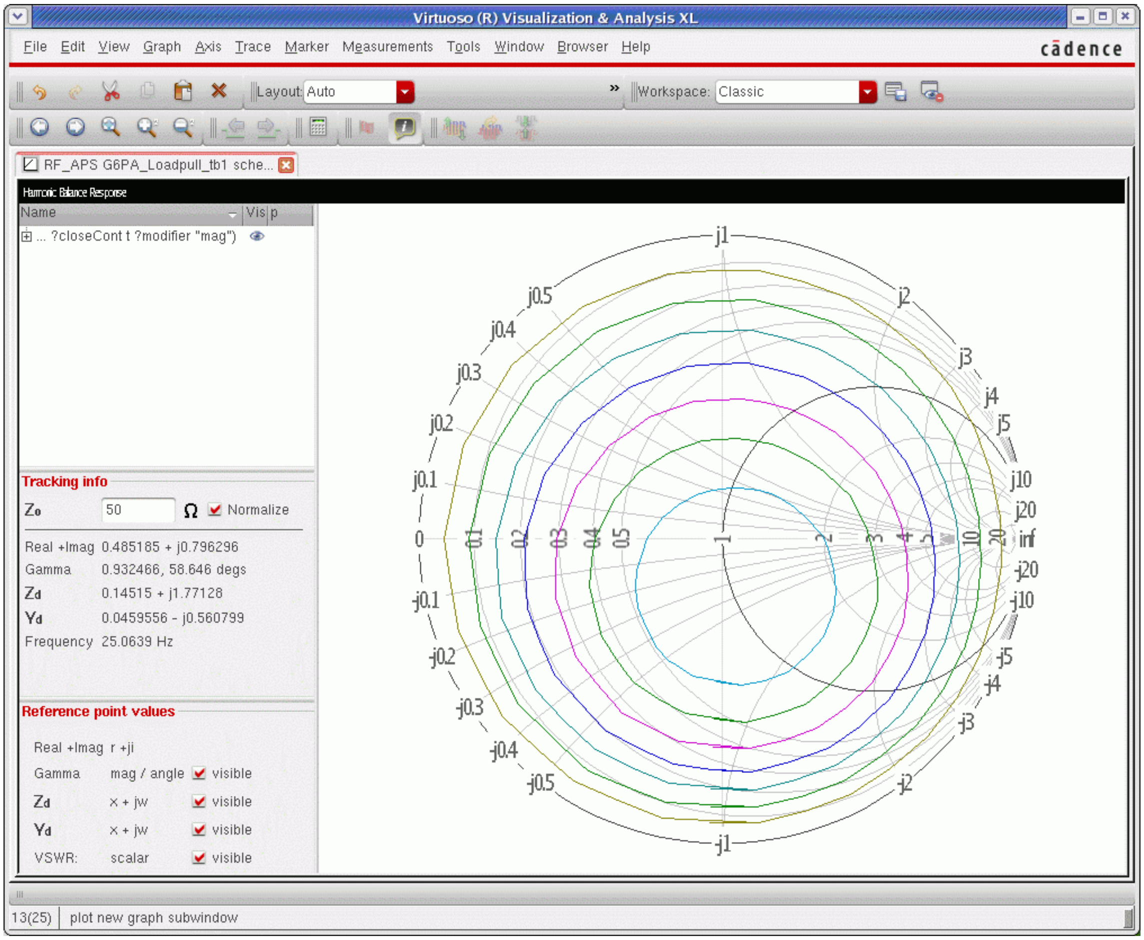
The smallest and largest power occurs at a single point which is plotted, but are very difficult to see in the waveform tool.

5. Select the *Close Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.

6. Select the first harmonic.

7. Select the top terminal of the top port in the triplexer.

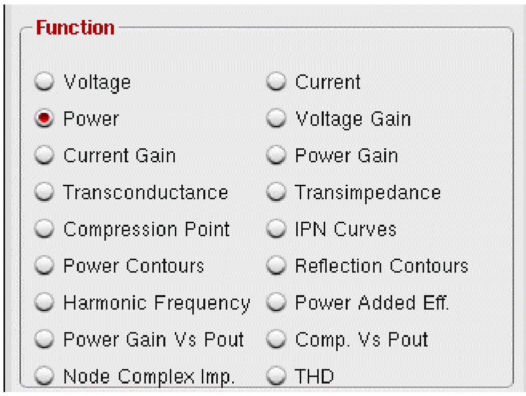
The power contours appear.



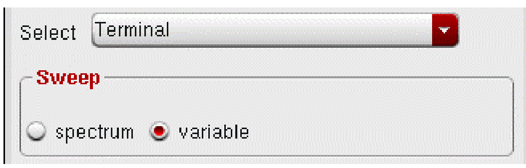
Second method

1. Select *pss* or *hb results*.

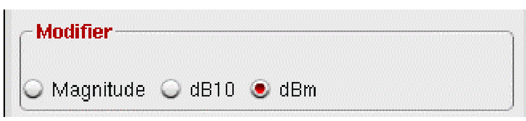
2. Select *Power*.



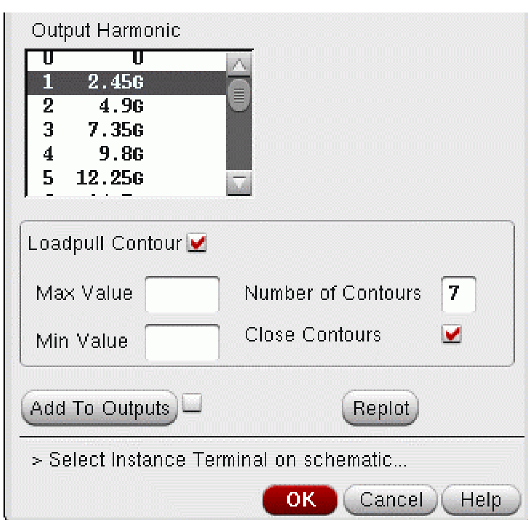
3. Select *variable*.



4. Select *Magnitude*, *dB10*, or *dBm*.



5. Select the *harmonic number*.



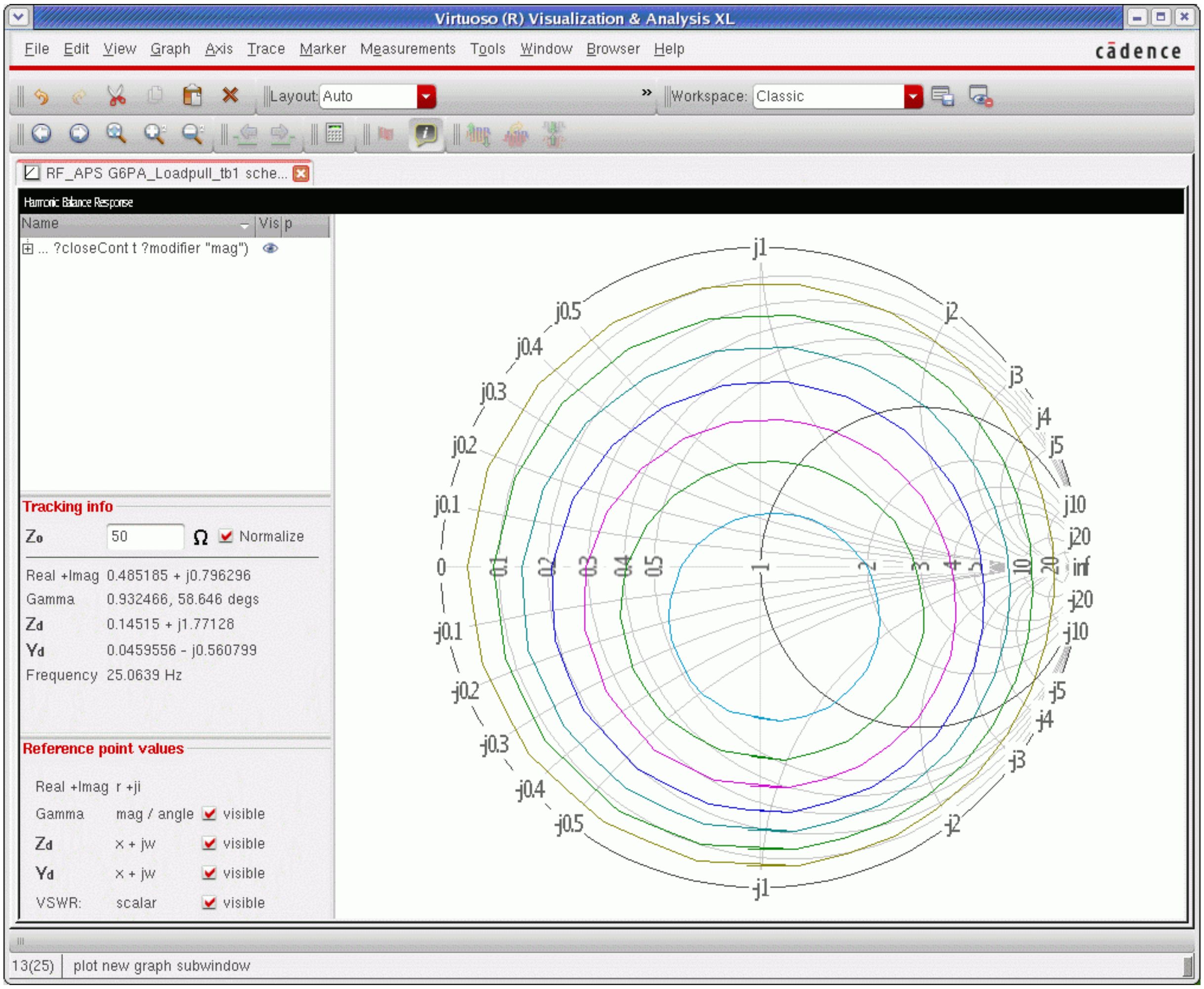
6. Select *Loadpull Contour*.

7. Specify 2 more than the number of curves you want.

8. Select the *Closed Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.

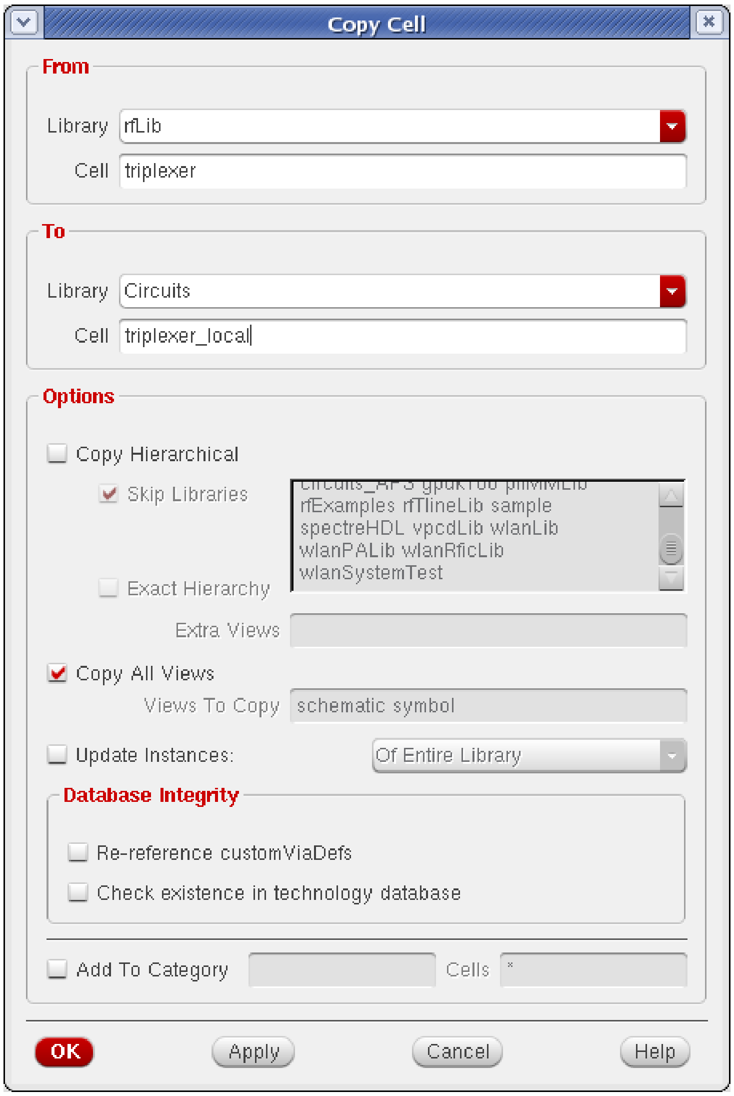
9. Select the top terminal of the top port of the triplexer circuit.

   The loadpull curves appear in the waveform tool.

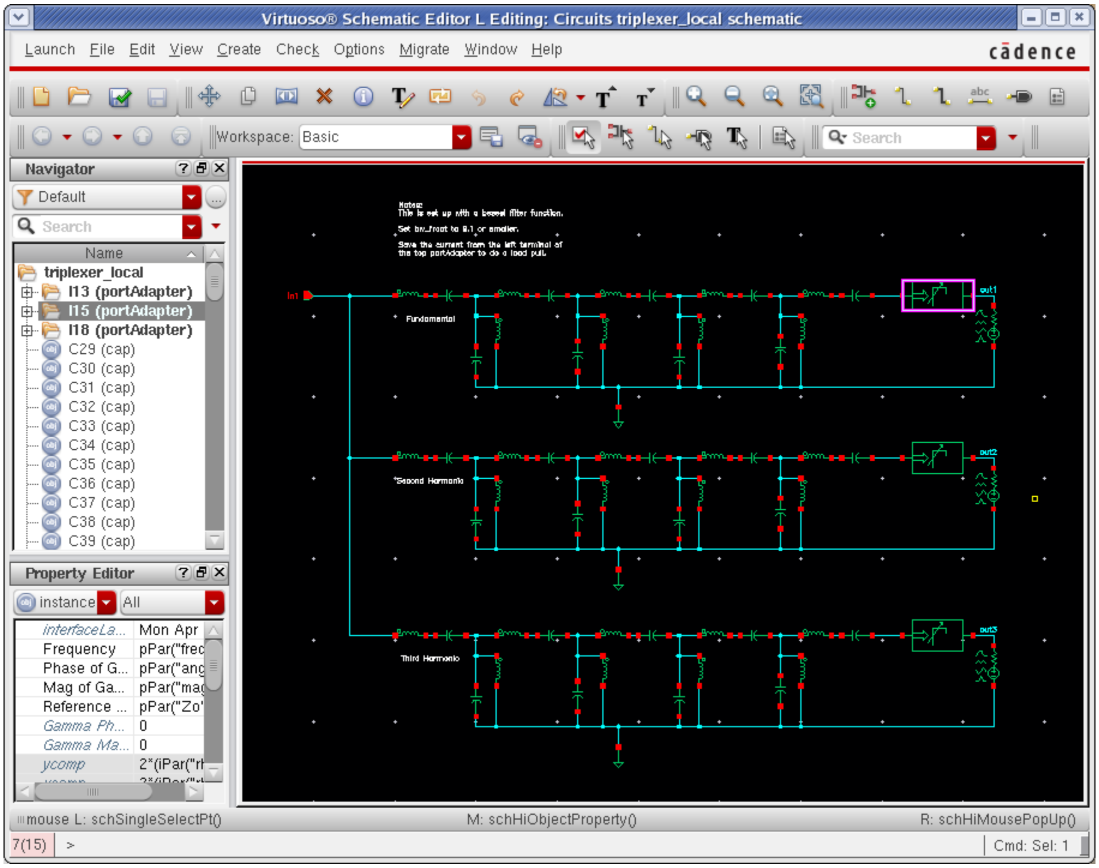


PSS Shooting Loadpull

1. For a loadpull simulation, copy a a triplexer (for setting the reflection and angle for up to three harmonics) or the ten_plexer (for up to 10 harmonics) into a local writable library, and rename the cell. These components are located in the `rfLib` library.

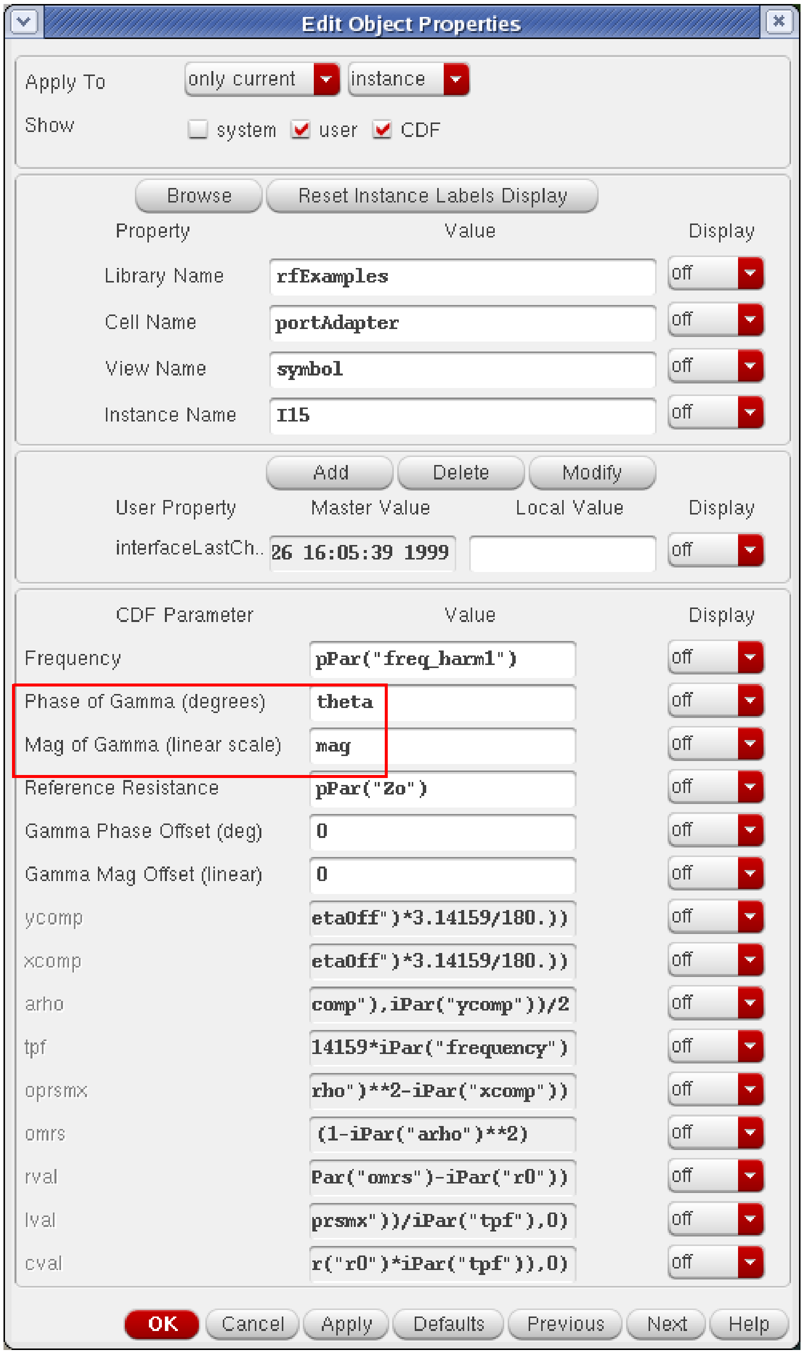


2. Open the triplexer_local schematic, and select the portAdapter for the desired harmonic. The top portAdapter is for the first harmonic, and the harmonic numbers increase as you move down in the schematic. The top portAdapter is highlighted below.



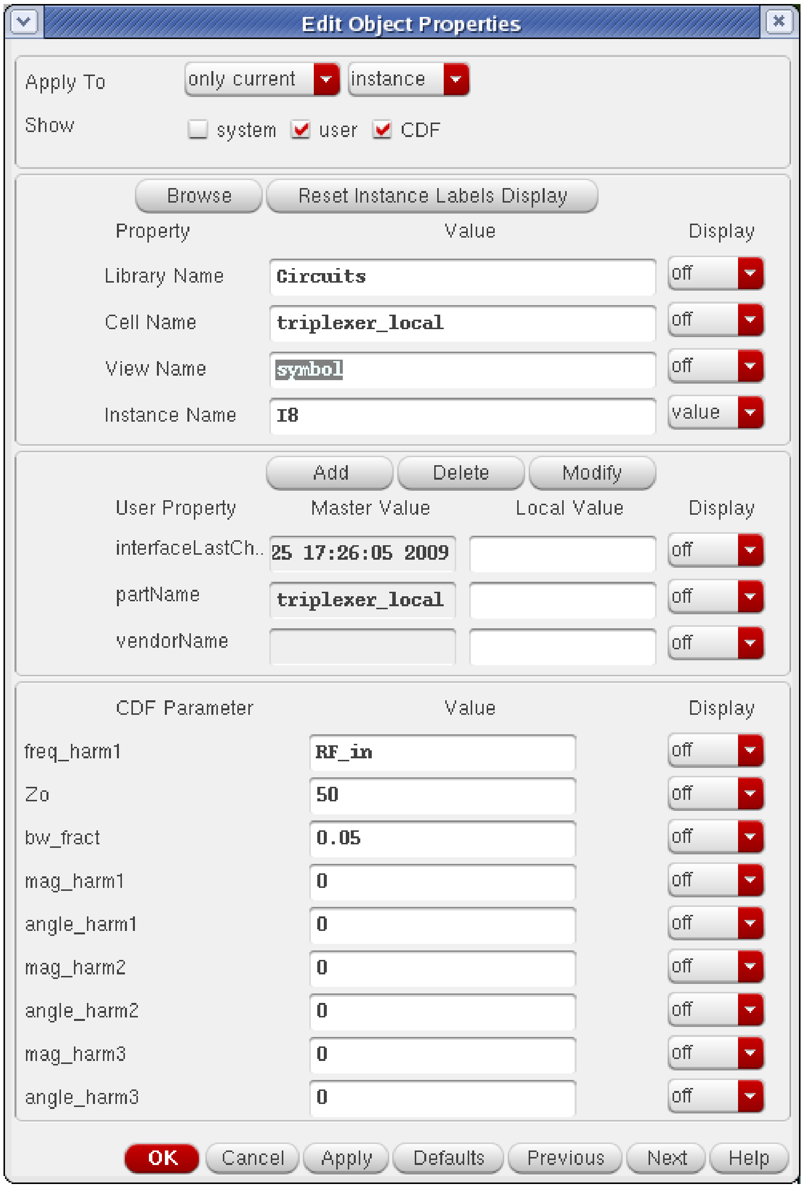
The properties list is set as follows:

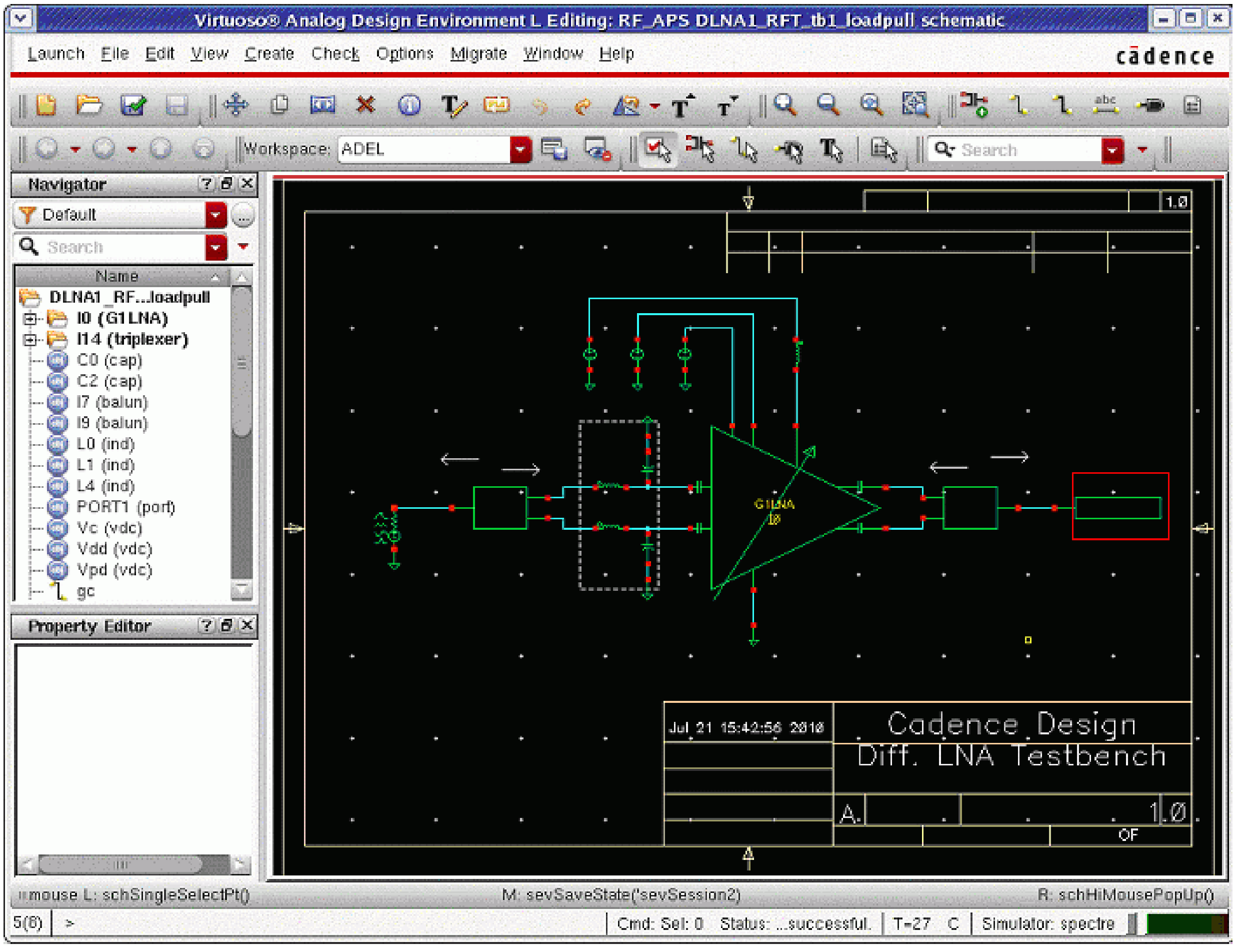
3. Change *Phase of Gamma (degrees)* and *Mag of Gamma (linear scale)* to variable names. Note that the phase is above the magnitude in the properties list.



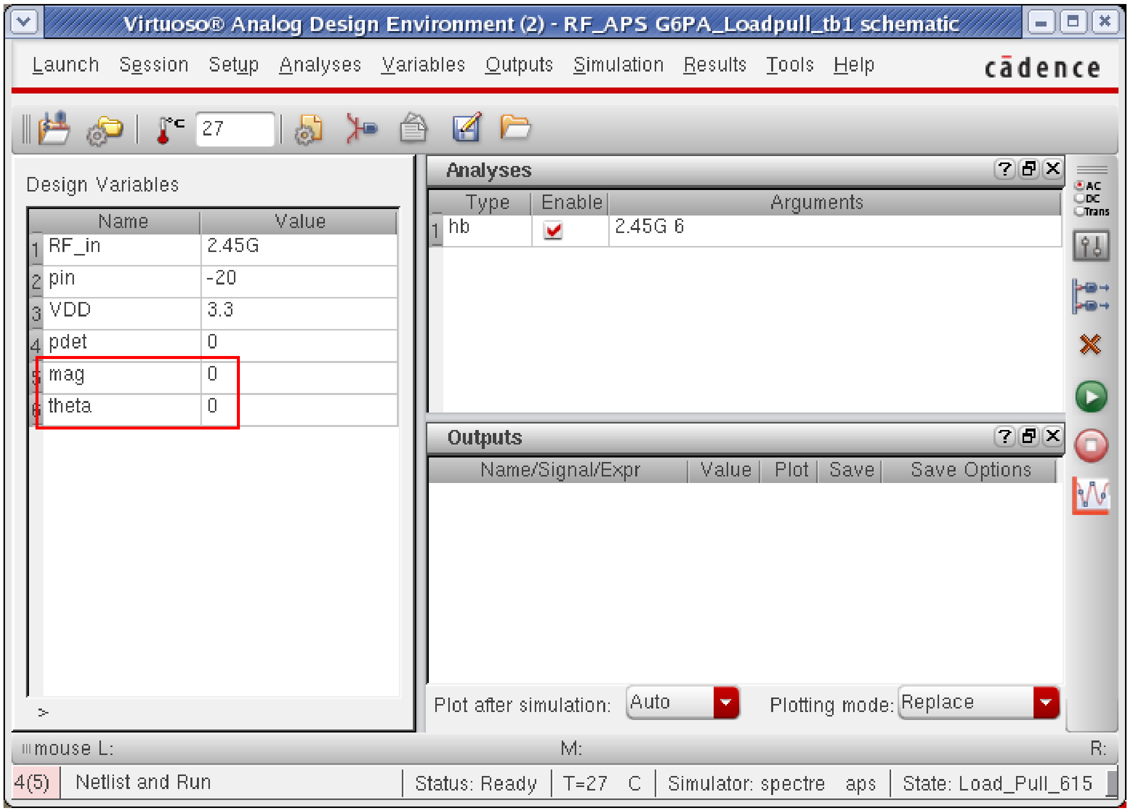
4. Save the local copy.

5. Add the local copy of the triplexer or ten_plexer as the load in the schematic. Save the schematic.




Virtuoso® Analog Design Environment L Editing: RF_APS DLNA1_RFT_tb1_loadpull schematic
Launch File Edit View Create Check Options Migrate Window Help
cadence
Workspace: ADEL
Navigator
Default
Name
DLNA1_RF...loadpull
I0 (G1LNA)
I14 (triplexer)
C0 (cap)
C2 (cap)
I7 (balun)
I9 (balun)
L0 (ind)
L1 (ind)
L4 (ind)
PORT1 (port)
Vc (vdc)
Vdd (vdc)
Vpd (vdc)
gc
Property Editor
Jul 21 15:42:56 2010
Cadence Design
Diff. LNA Testbench
mouse L: schSingleSelectPt()
M: sevSaveState('sevSession2)
R: schHiMousePopUp()
Cmd: Sel: 0 Status: ...successful. T=27 C Simulator: spectre

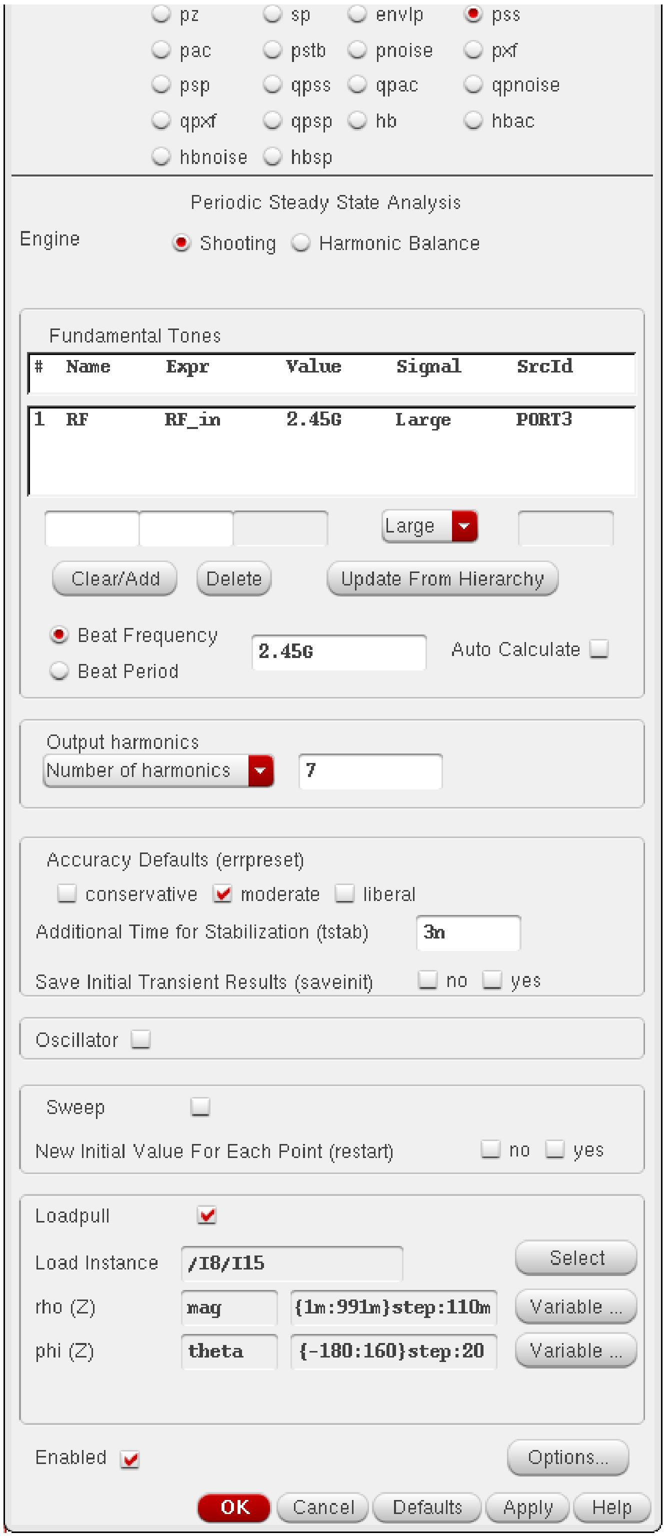
6. Add the two variables you used in the local triplexer or ten_plexer into ADE.



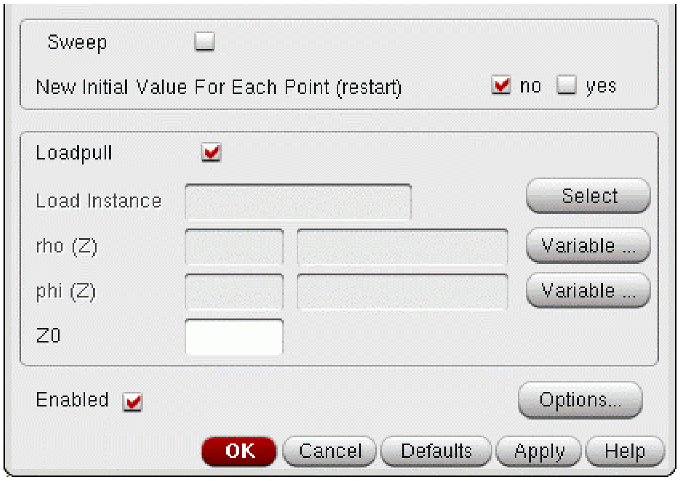
The example above shows `mag` and `theta` variables added. These variables should not be used to set any component values in the circuit.

Triplexer from your local library directory contains three filters for the first, second, and third harmonics and portAdapters with port loads for each of the three filters.

7. Open the PSS *Choose Analyses* form.

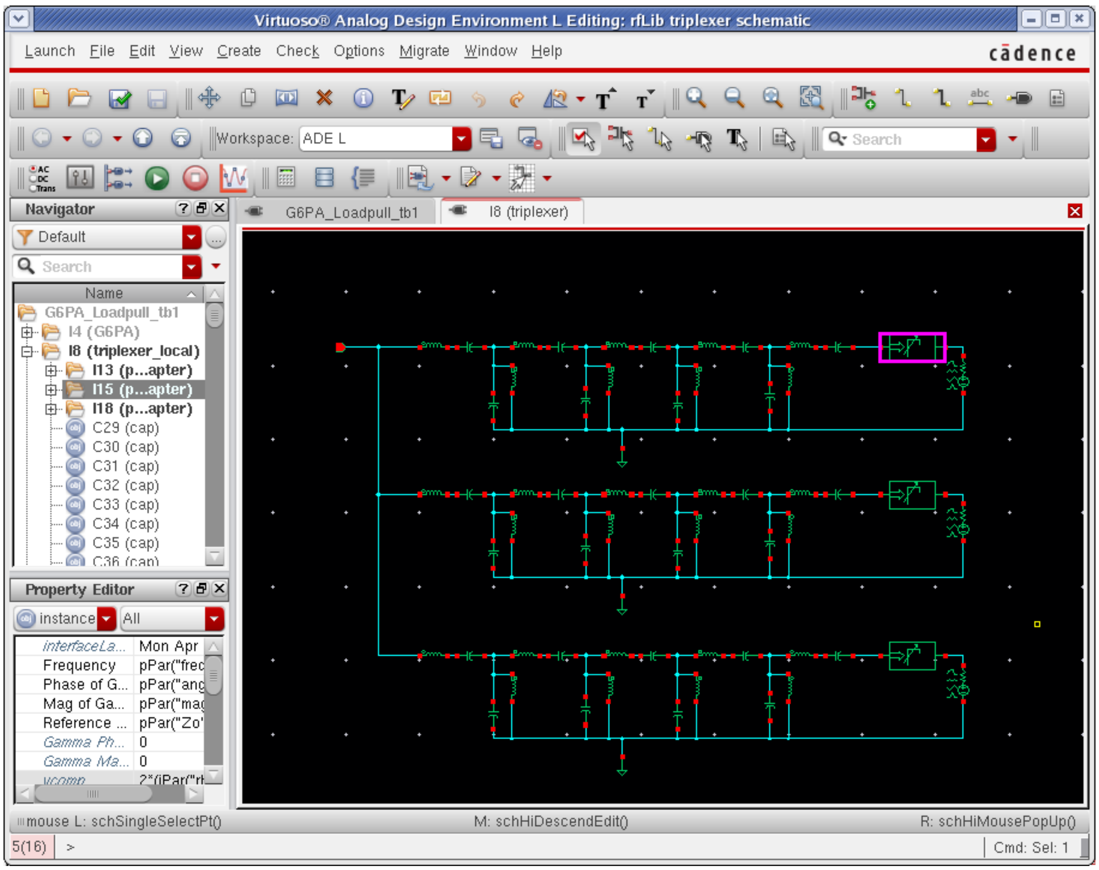


8. Set up the top of the form as usual.
9. Select the *Loadpull* checkbox.

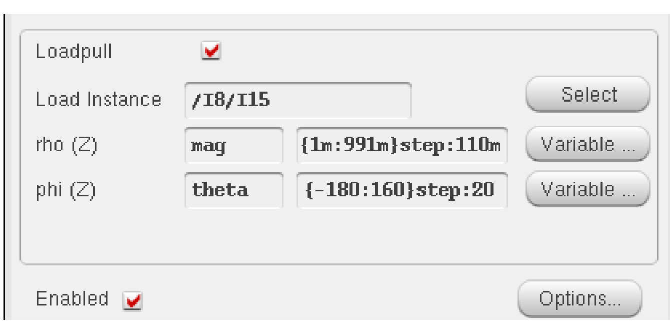


10. Click *Select* on the right of the greyed out *Load Instance* field.
11. In the triplexer_local schematic, select the topmost portAdapter on the right side of the schematic if you want a loadpull for the first harmonic. Select the next portAdapter down

for a loadpull of the second harmonic. Select the third portAdapter down for the third harmonic. The first harmonic portAdapter is highlighted below.



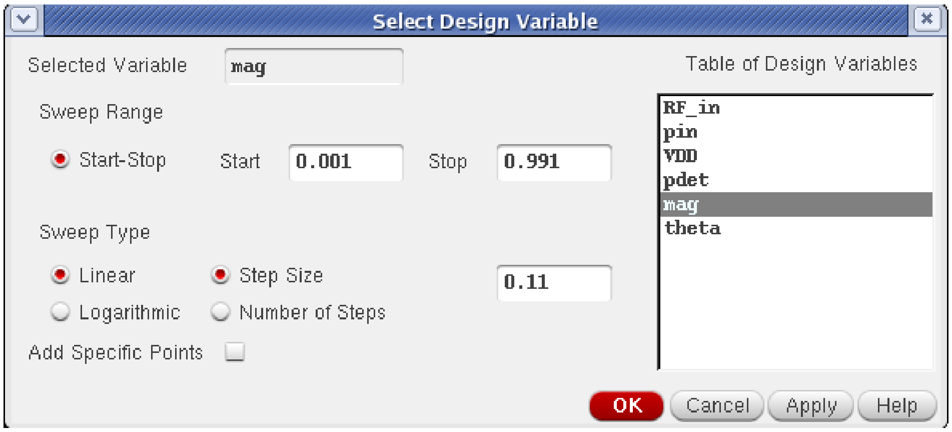
The instance name is entered in the *Load Instance* field. Direct entry by typing is not allowed for this field.



12. Click the *Variable* button on the right of the *rho (Z)* field. Select the variable you entered in ADE.

This variable is used to set the value of the reflection coefficient in the port.

13. Specify a sweep range, and set the spacing.

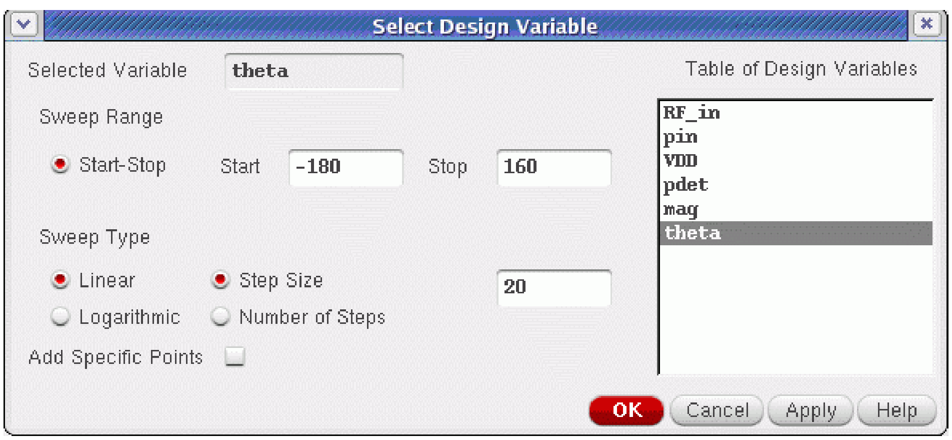


14. When completed, select *OK*.

    The values are displayed in the *Choose Analyses* form.

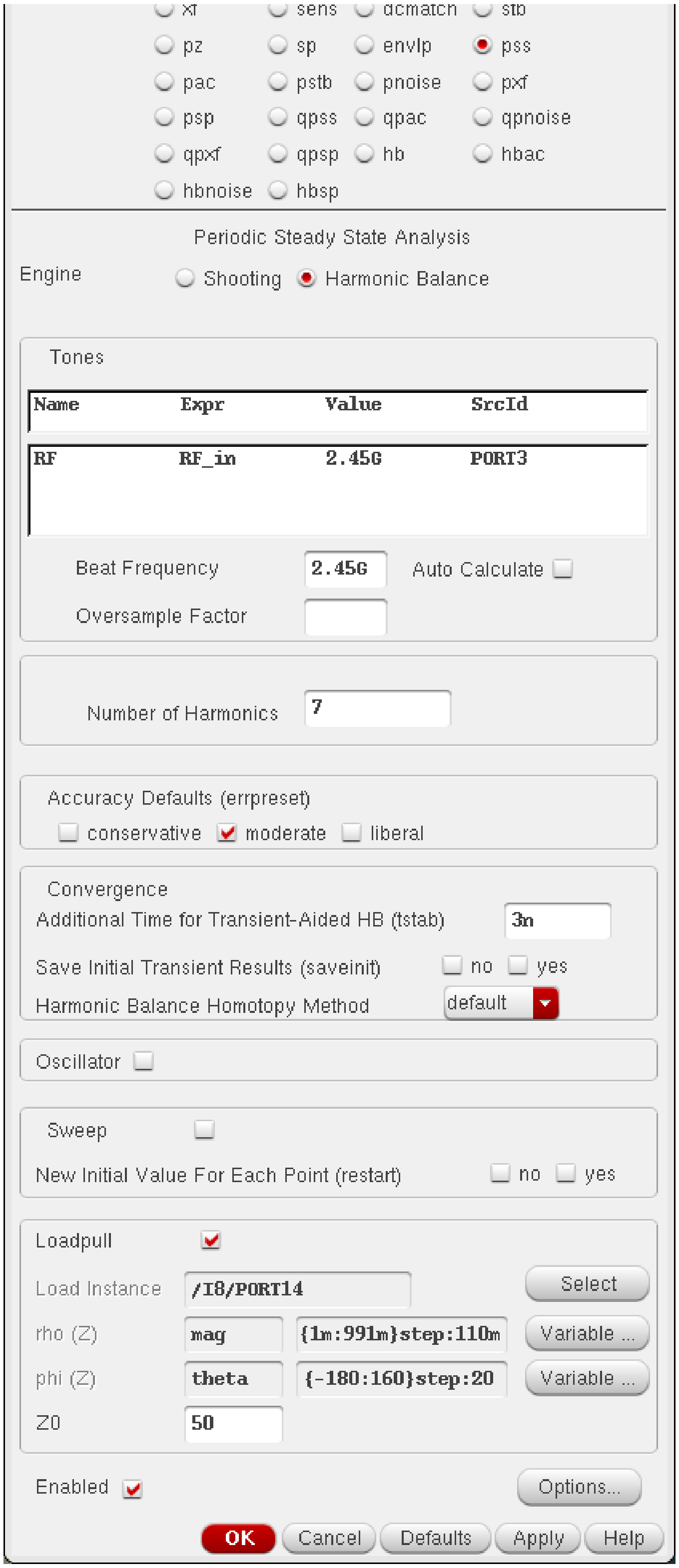
15. Click the *Variable* button on the right of the *phi (Z)* field. This sets the angle of the reflection coefficient.

16. In the *Select Design Variable* window, select the second variable you entered and specify the sweep limits.



17. When completed, click *OK*.

    The *Choose Analyses* form is updated.


xf sens acmatch stb
pz sp envlp pss
pac pstb pnoise pxf
psp qpss qpac qpnoise
qpxf qpsp hb hbac
hbnoise hbsp
Periodic Steady State Analysis
Engine Shooting Harmonic Balance
Tones
Name Expr Value SrcId
RF RF_in 2.45G PORT3
Beat Frequency 2.45G Auto Calculate
Oversample Factor
Number of Harmonics 7
Accuracy Defaults (errpreset)
conservative moderate liberal
Convergence
Additional Time for Transient-Aided HB (tstab) 3n
Save Initial Transient Results (saveinit) no yes
Harmonic Balance Homotopy Method default
Oscillator
Sweep
New Initial Value For Each Point (restart) no yes
Loadpull
Load Instance /I8/PORT14 Select
rho (Z) mag {1m:991m}step:110m Variable ...
phi (Z) theta {-180:160}step:20 Variable ...
Z0 50
Enabled Options...
OK Cancel Defaults Apply Help

18. When completed, click *OK* and run the simulation in ADE.

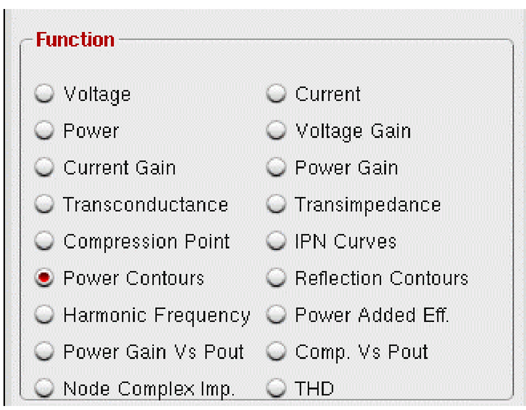
19. In ADE, select *Results - Direct Plot - Main Form*.

There are two methods of plotting loadpull curves.

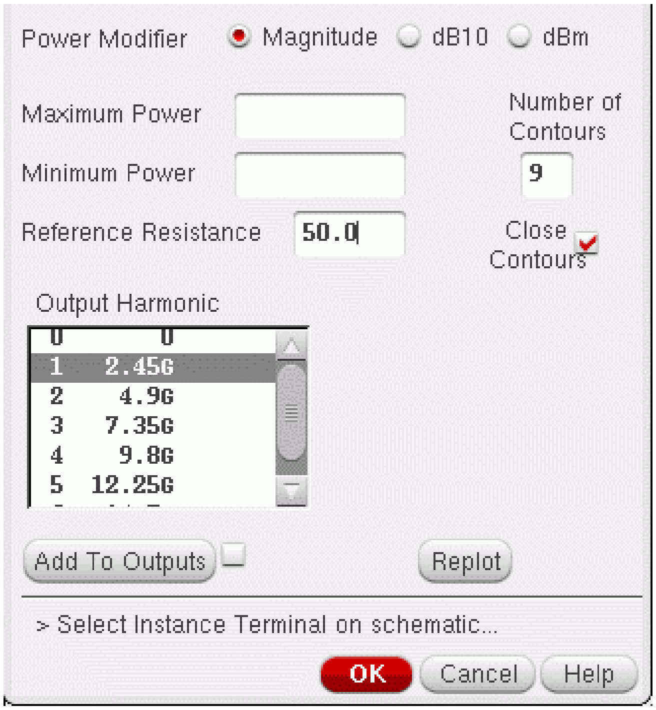
First method

1. Select *pss results*.

2. Select *Power Contours*.



3. Select *Magnitude*, *dB10*, or *dBm*.



4. Specify 2 more than the number of contour lines you want.

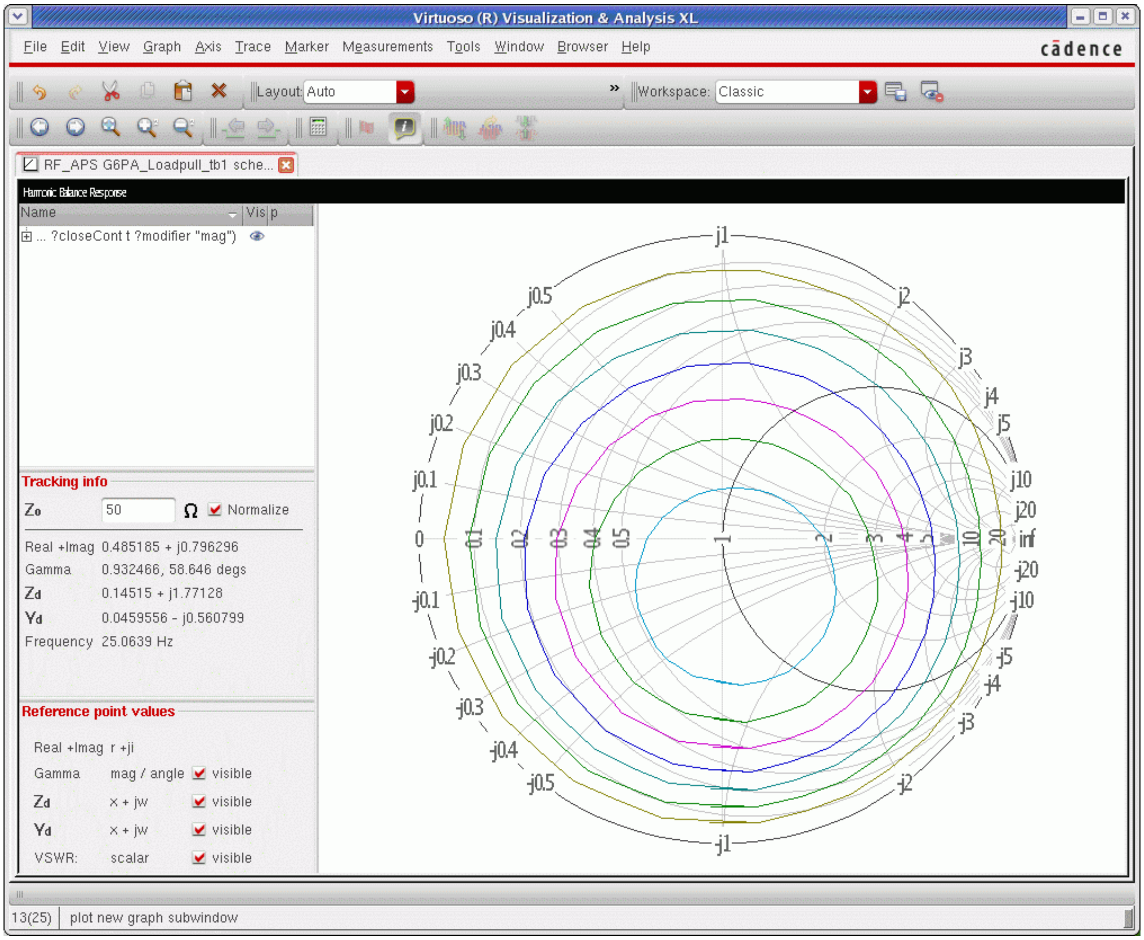
The smallest and largest power occurs at a single point which is plotted, but are very difficult to see in the waveform tool.

5. Select the *Close Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.

6. Select the first harmonic.

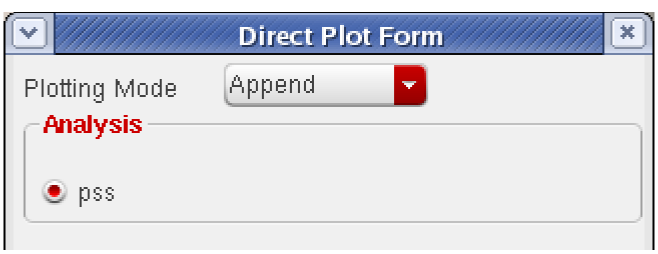
7. Select the top terminal of the top port in the triplexer.

The power contours appear.

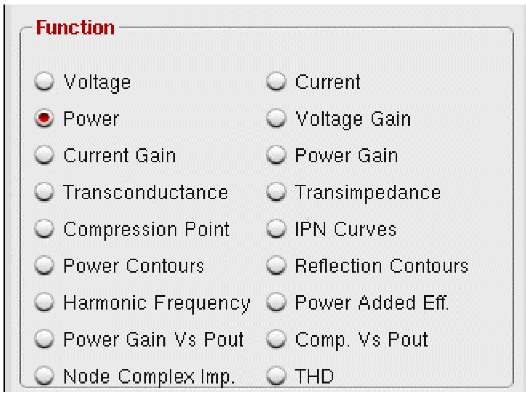


Second method

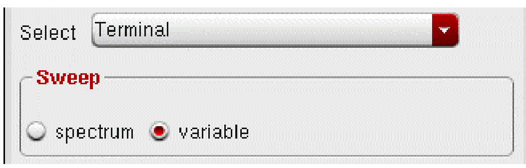
1. Select *pss results*.



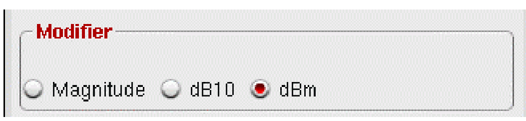
2. Select *Power*.



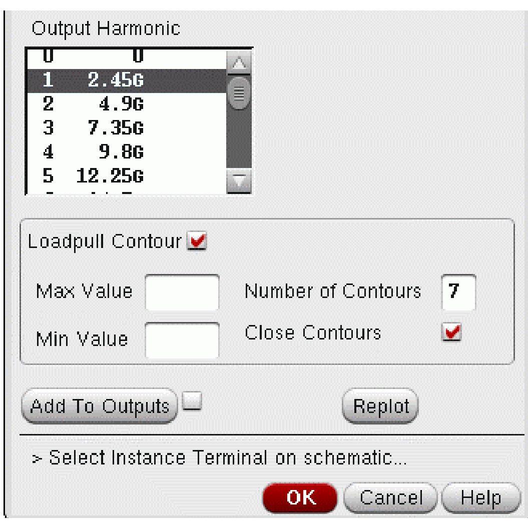
3. Select *variable*.



4. Select *Magnitude*, *dB10*, or *dBm*.



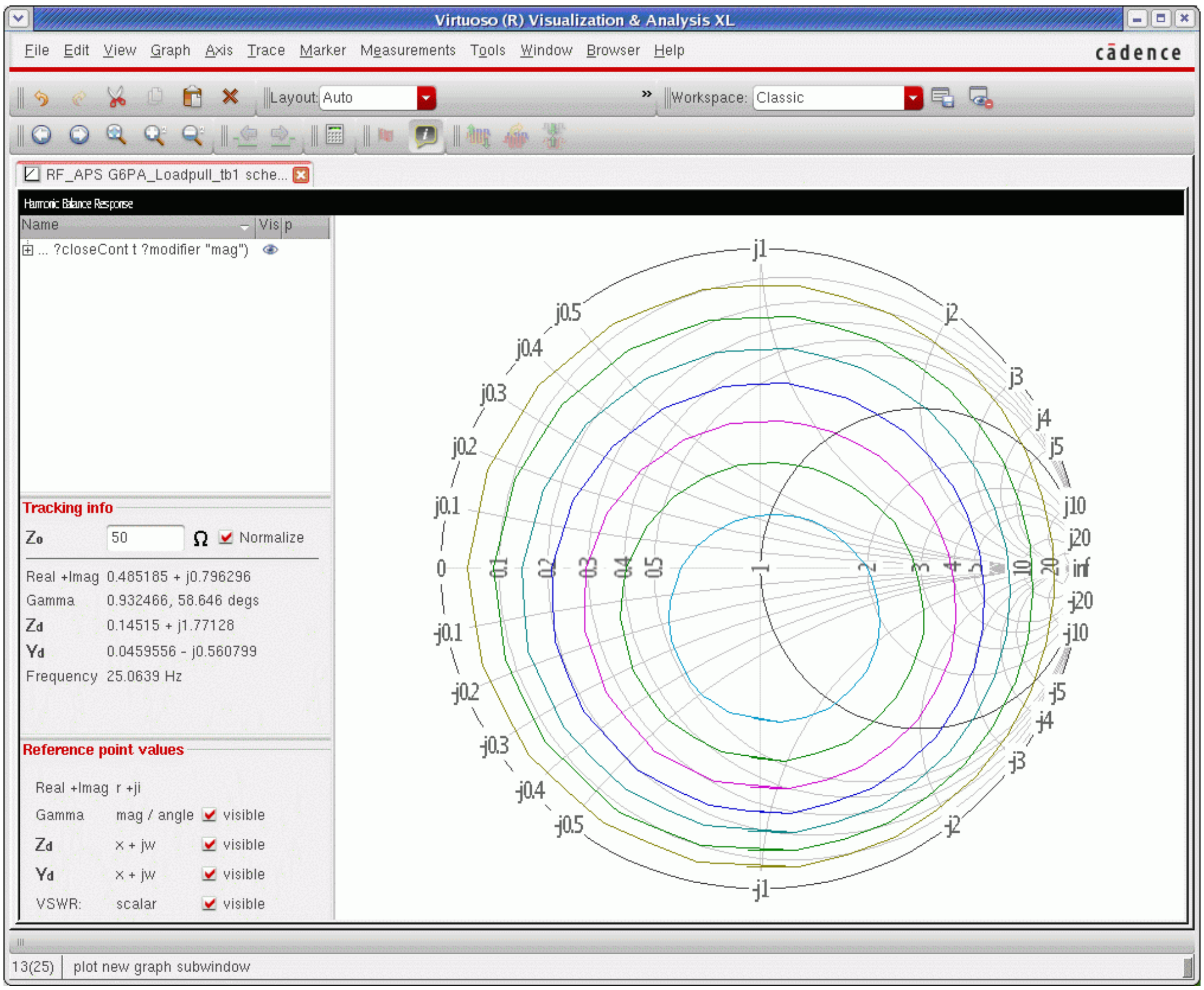
5. Select the *harmonic number*.



6. Select *Loadpull Contour*.

7. Specify 2 more than the number of curves you want.
8. Select the *Closed Contours* check box. This causes the contours from the loadpull to be displayed as circles or ellipses, depending on the distortion of the circuit.
9. Select the top terminal of the top port of the triplexer circuit.

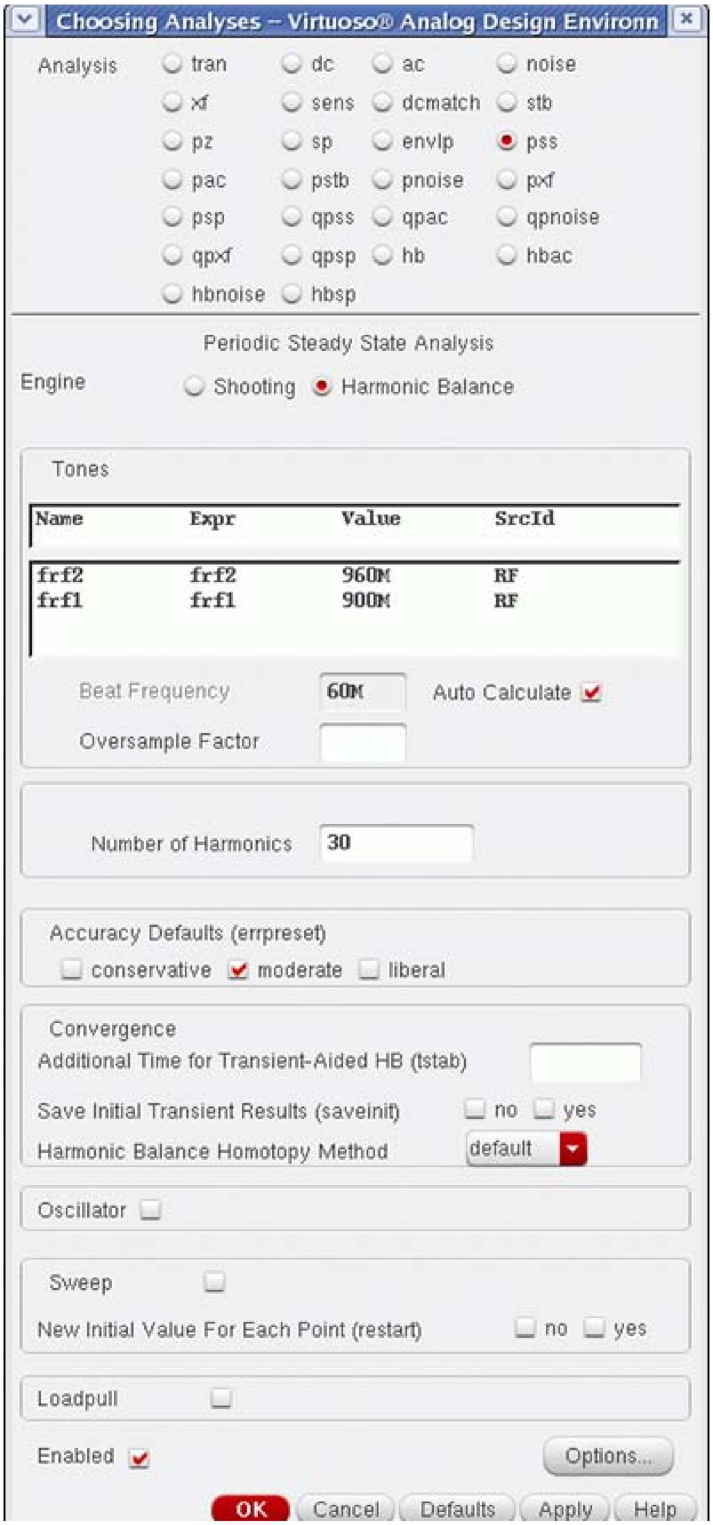
   The loadpull curves appear in the waveform tool.



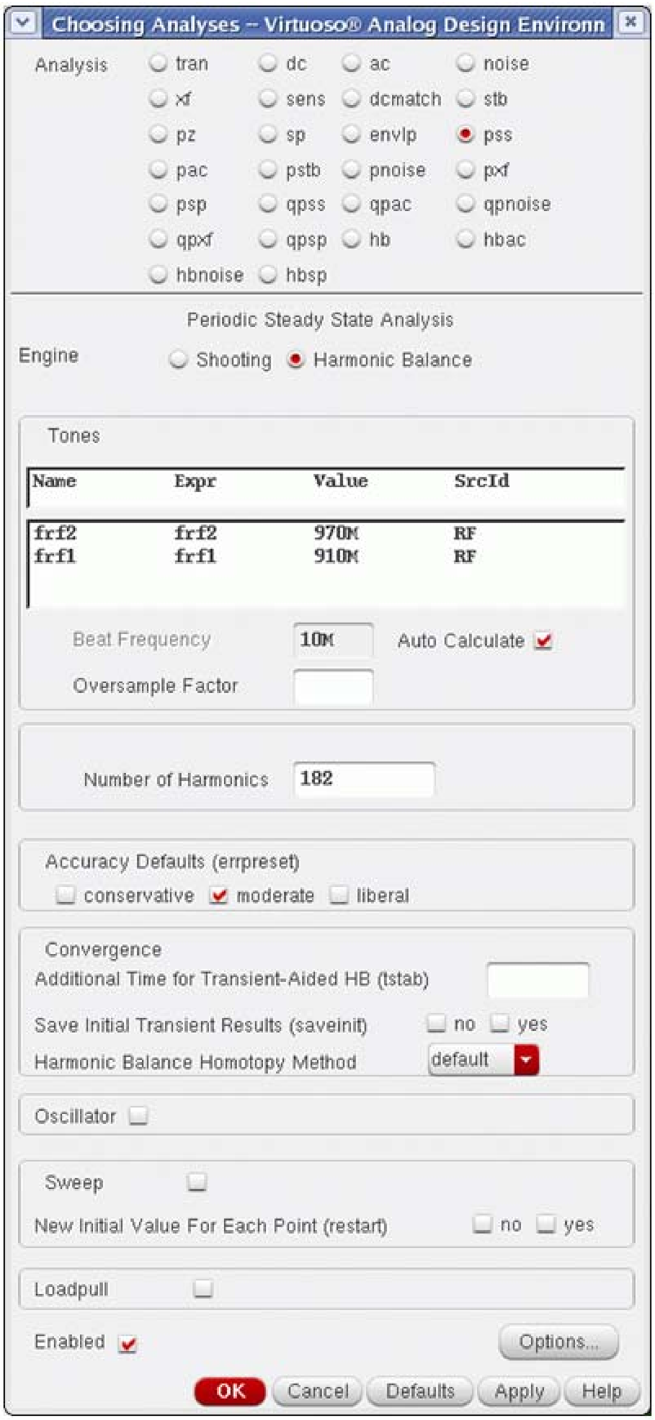
Two Input Frequencies

Two input frequencies can be simulated as long as the beat frequency is high compared to the frequencies that are applied to the circuit. In general, if the pss beat frequency is smaller than about 1/25th the highest input frequency, the simulation is faster, if you use qpss or hb.

In the example below, 30 periods of 900MHz and 31 periods of 960 MHz need to be calculated. This is on the edge of acceptability for pss. For this simulation, it might be faster to simulate using qpss or hb because fewer harmonics need to be solved, which requires less memory and time.



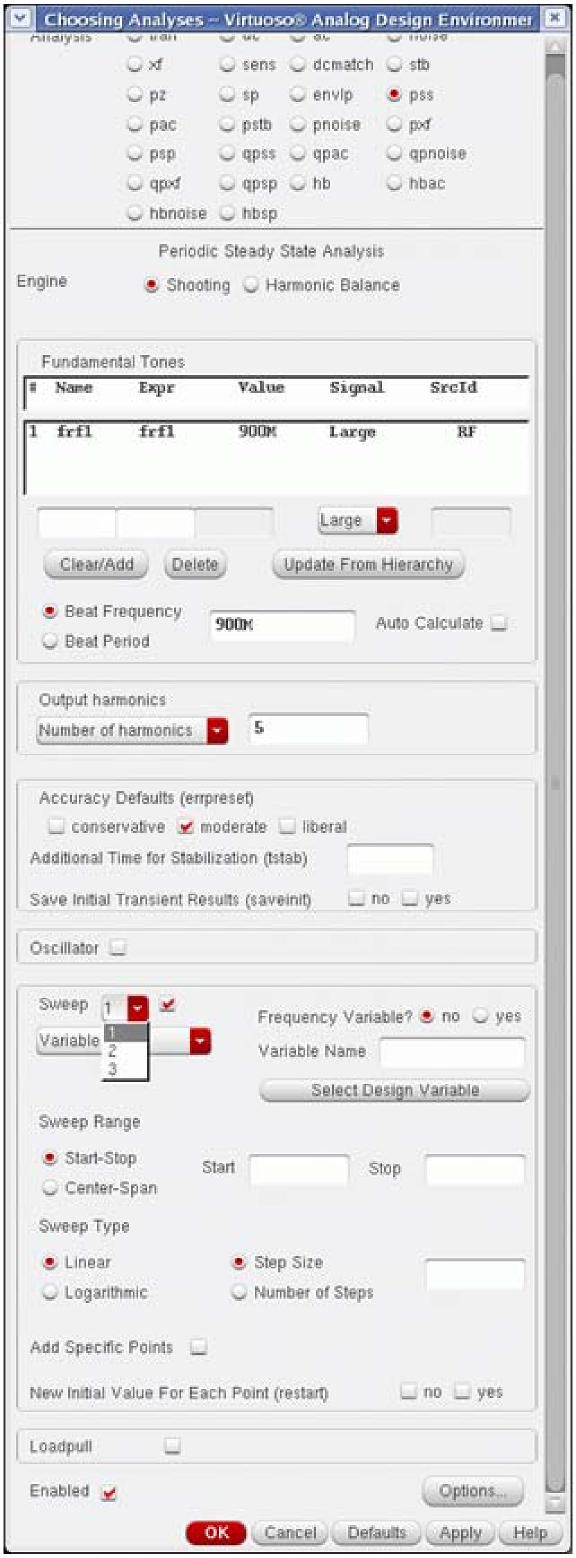
Note that when the frequencies are moved slightly to 910M and 970M, the beat frequency drops to 10MHz. In this case, it would definitely be faster to use hb or qpss. These analyses are much faster than pss. Also, since the pss frequency is lower, more harmonics need to be calculated in order to see the desired mixing products.



Sweeps

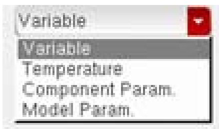
For the next plots, a sweep is set up in the pss *Choosing Analyses* form. A sweep is needed in order to measure compression.

Sweeps can be made either in the parametric plot tool in ADE where the number of swept things is not limited, or in the pss *Choosing Analyses* form, where the number of swept things is limited to three. This example shows using the pss *Choosing Analyses* form shown below. For documentation on the parametric plot tool, please refer to the ADE documentation.

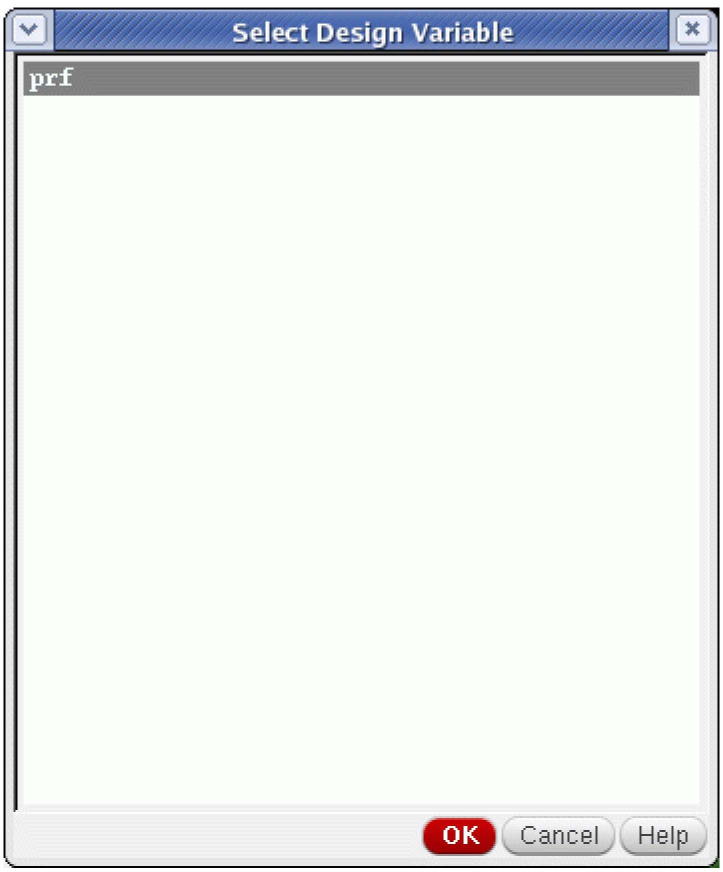

Choosing Analyses -- Virtuoso® Analog Design Environmen
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
1 frf1 frf1 900M Large RF
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
Beat Period
900M
Auto Calculate
Output harmonics
Number of harmonics
5
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Sweep
1
Variable
2
3
Frequency Variable?
no
yes
Variable Name
Select Design Variable
Sweep Range
Start-Stop
Center-Span
Start
Stop
Sweep Type
Linear
Logarithmic
Step Size
Number of Steps
Add Specific Points
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

1. Select a sweep number from the *Sweep* drop-down list.

2. Select the type of parameter that is being varied from the *Variable* drop-down list.

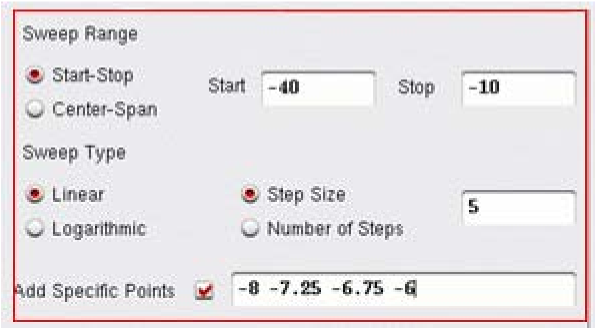


3. Type the variable name in the *Variable Name* field. You can also select the variable from a list. To do this, click *Select Design Variable*. and the list is displayed.

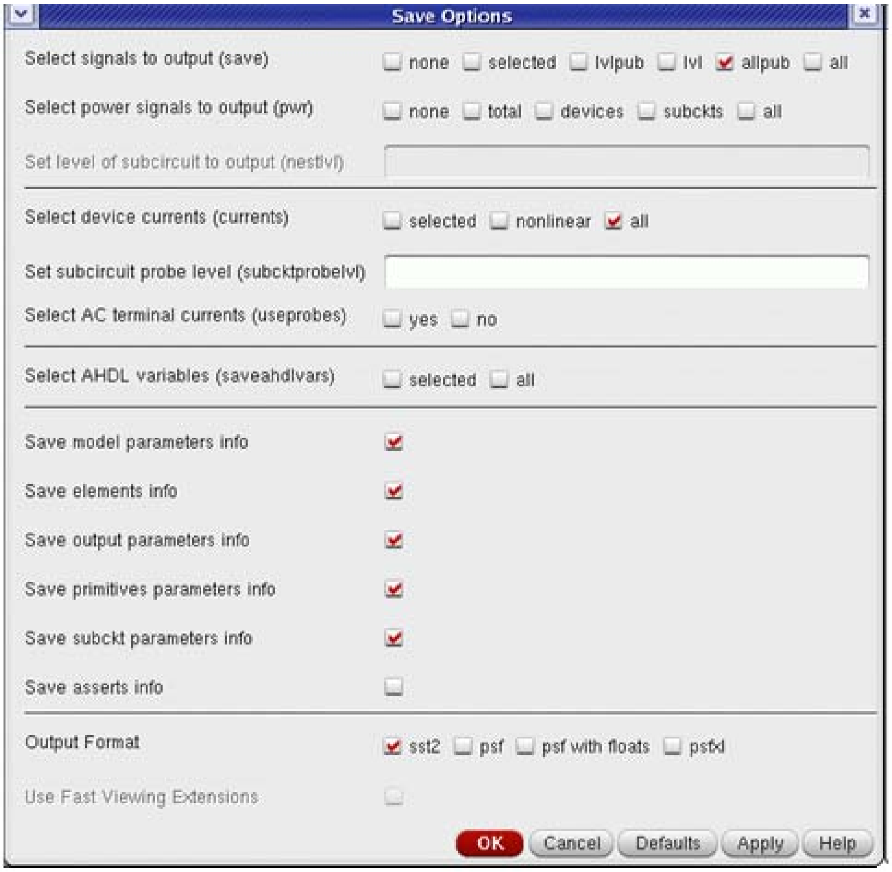


Select the desired variable, and click *OK* to enter the variable.

4. Define the sweep range. Set up a sweep range, and if you desire values that are in between the sweep values, or above or below the sweep values, select *Add Specific Points*. Type a list of values separated by spaces. This is shown below.



For many power measurements, currents need to be saved. If you are using Shooting, these currents can be saved, To save all the currents from the ADE environment. Select *Outputs - Save All*, and then select *all* for *Select Device Currents (currents)*. This takes the currents that are calculated from every component in the circuit and adds them into the simulator result file.



For hb, port, and voltage source, only currents are saved, and this save is automatic. The ADE *Save Options* form shown above has no effect in hb. If you need to measure a current other than in a port or voltage source, add an iprobe in series with the branch.

If all the currents need to be saved when harmonic balance is used as the engine, select *yes* for the *Select AC terminal currents (useprobes)* option above. This adds a zero volt battery in series with every branch in the circuit. Although this results in saving all the currents, it also adds entries in the harmonic balance matrices for each voltage source. As a result, the hb matrices get much larger, therefore, more time and memory are required to solve the equations.

In addition, there is a very small chance of causing unresolvable numerical issues and the simulator issues a singular matrix error message. If you see this error, turn off the *Select AC terminal currents (useprobes)* option and run the simulation again.

Note: The *Select AC terminal currents (useprobes)* option is not recommended if you run a small-signal analysis after the large-signal shooting pss analysis.

Compression Point

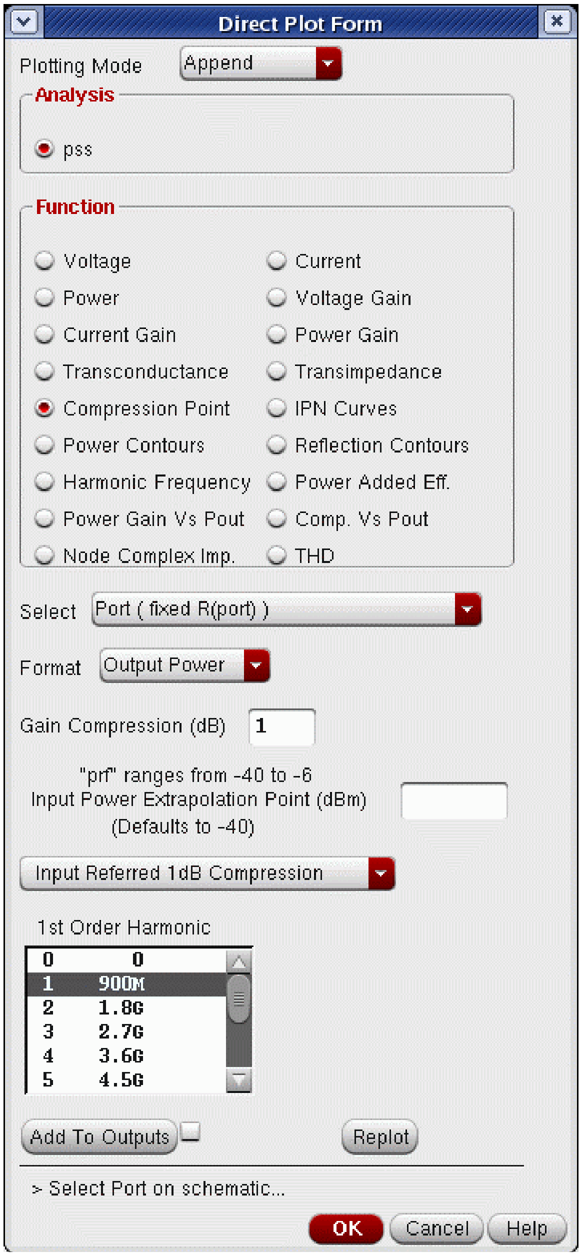
This requires a swept input power setup as shown in the section Loadpull on page 657.

Make sure that the power at the first point in the sweep is low enough so that there is no compression in the output at that power level. The first point in the sweep is used to establish the uncompressed power for the compression measurement.

To plot a compression point measurement:

1. Select *Results - Direct Plot - Main Form* in the ADE.

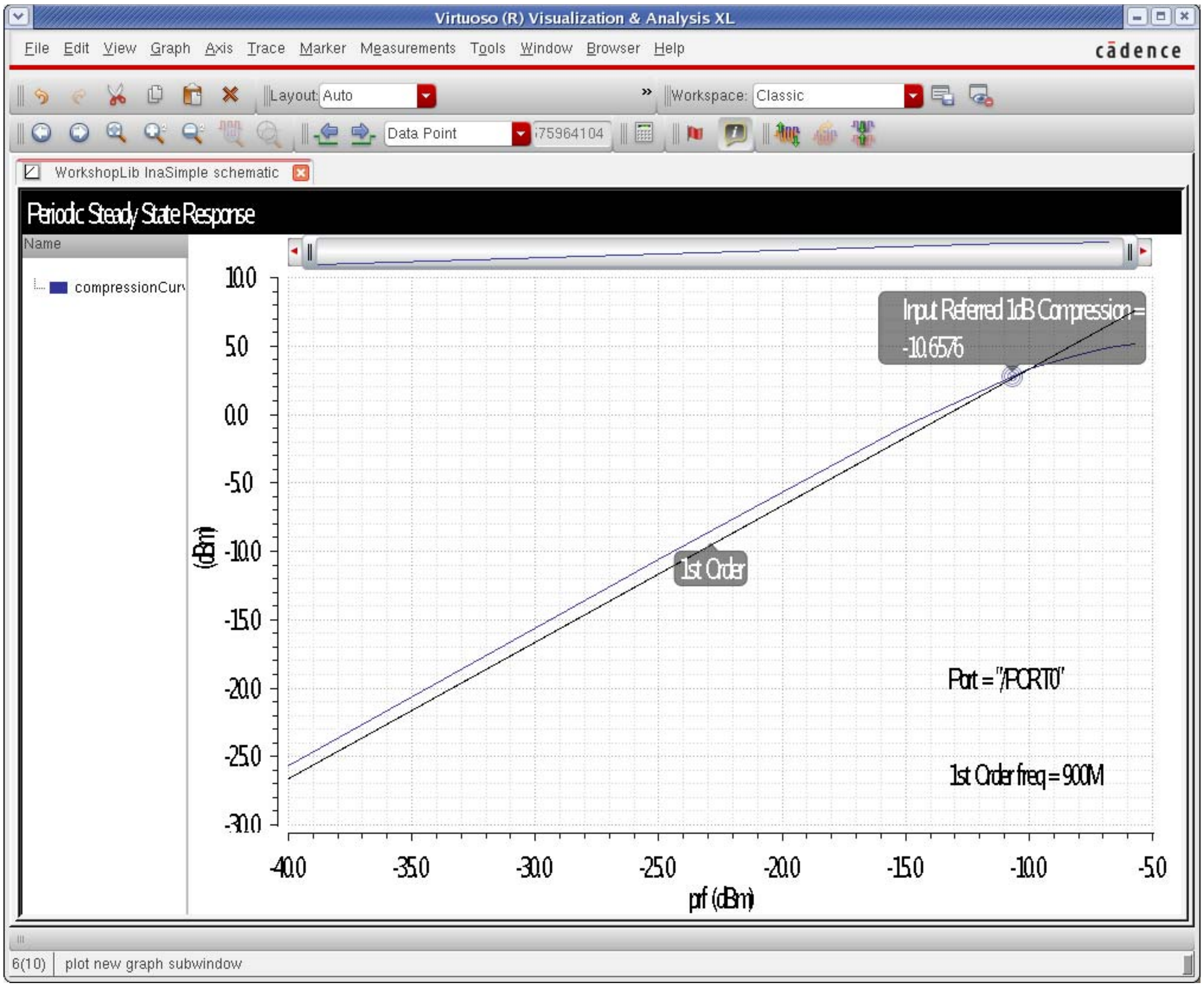
The Direct Plot Form is displayed, as shown below.



2. Select *Compression Point* from the *Function* section.
3. Select the type of thing that needs to be probed in the circuit from the *Select* drop-down list.
4. Specify the compression level.
5. Specify input or output-referred compression amount.

6. Select the output harmonic to be plotted.
7. Make the selection that is shown at the bottom of the *Direct Plot Form*. In this example, a port in the schematic that is used as a load should be selected.

   The compression curve is displayed.

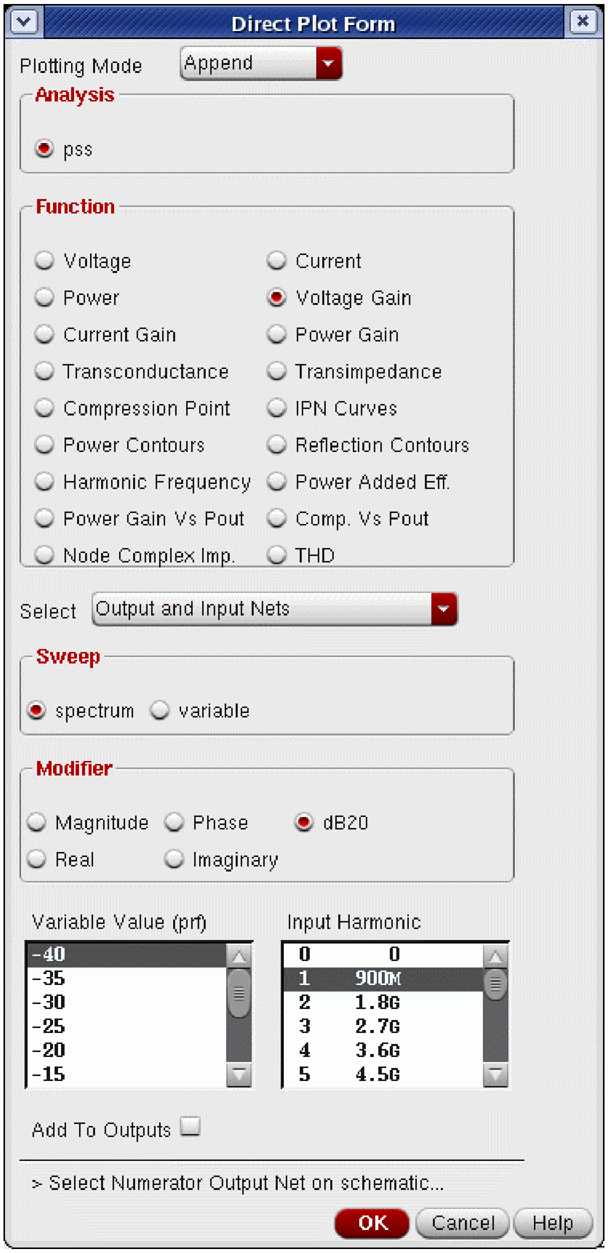


Voltage Gain

This example shows a swept measurement. To set this up, see Loadpull on page 657. A sweep is not required for this measurement.

The voltage gain direct plot function provides the output spectrum divided by the amplitude of the input at the specified input frequency. Note that the input amplitude at the input frequency has a single value. All the output harmonics at all the calculated output frequencies are scaled by the single voltage at the input.

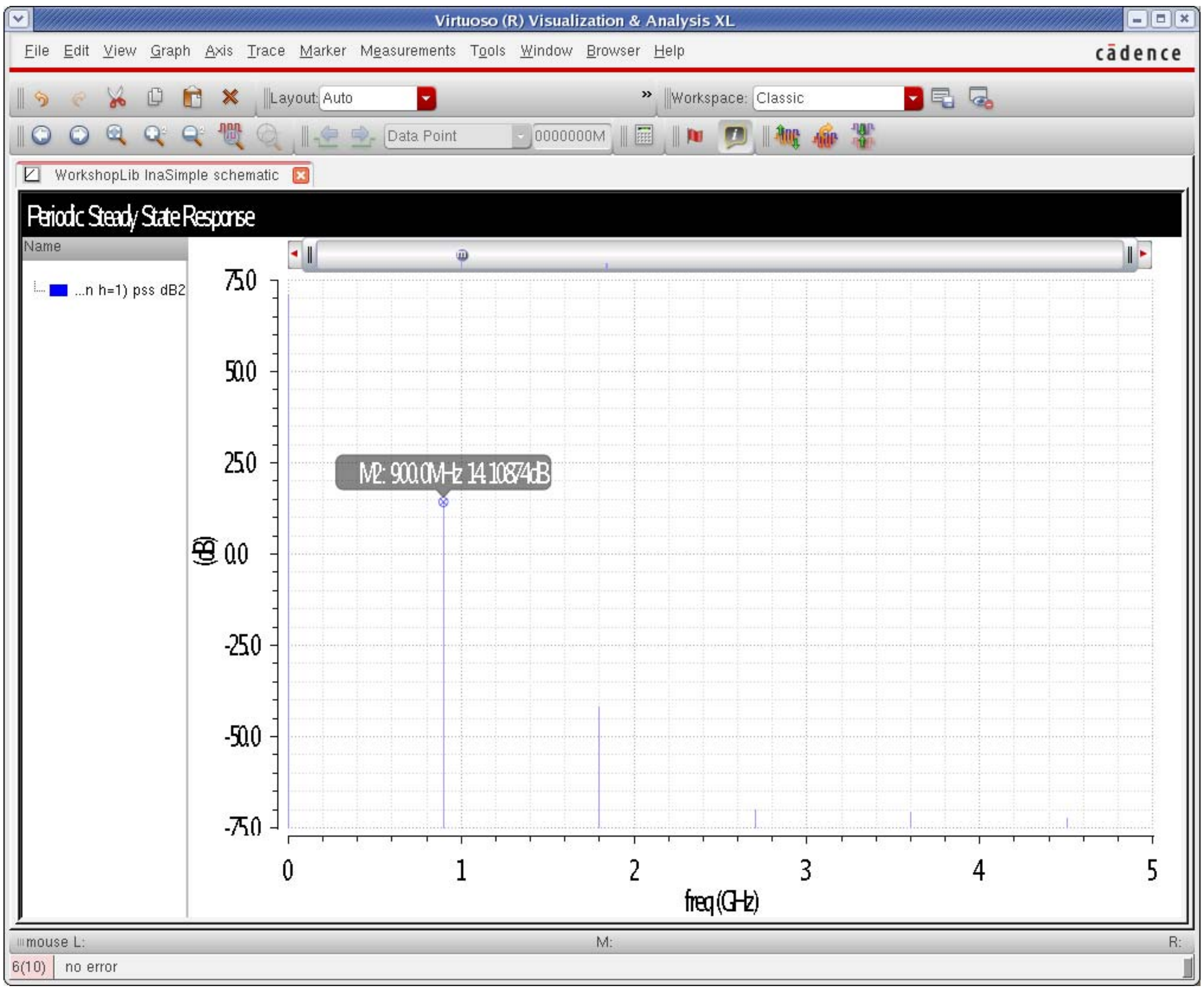
To plot the voltage gain:



1. Select *Voltage Gain* from *Function* section.
2. Select the appropriate modifier.
3. Select the power level of the sweep.

4. Select the input harmonic frequency.
5. Select the output node in the schematic.
6. Select the input node in the schematic.

Below is the output spectrum of the circuit, divided by the amplitude of the input at the input frequency.

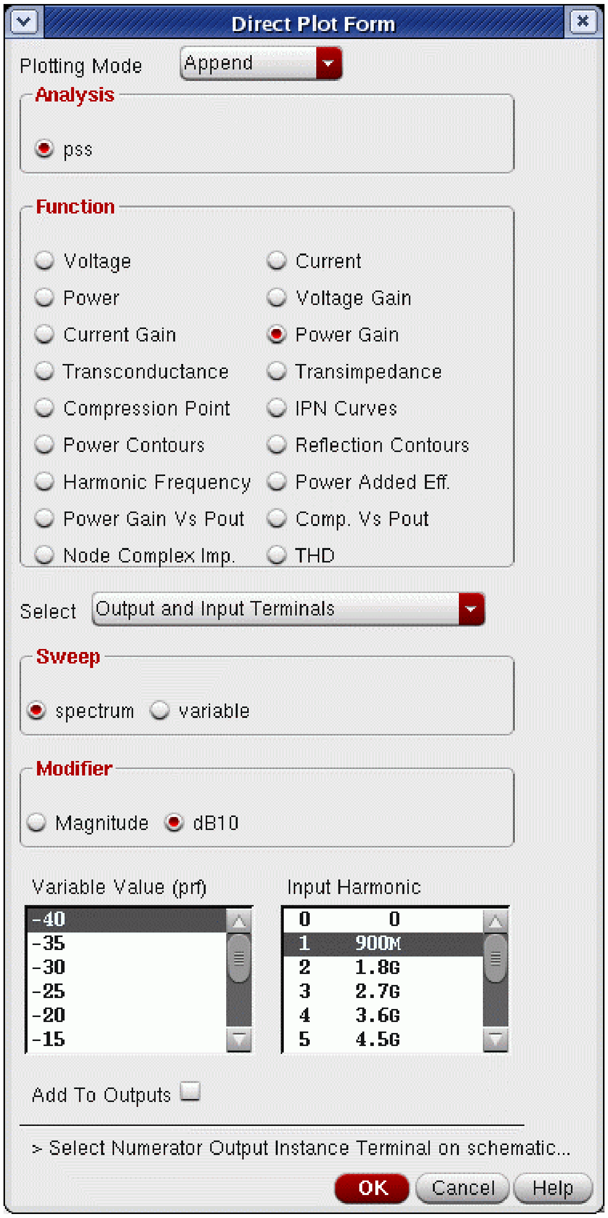


Power Gain

This example shows a swept measurement. To set this up, see Loadpull on page 657. A sweep is not required for this measurement.

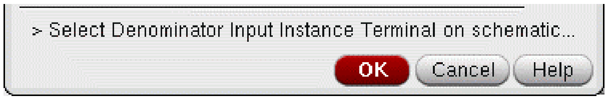
The power gain direct plot function provides the output spectral power divided by the power of the input at the specified input frequency. Note that the input power at the input frequency has a single value. All the output harmonics at all the calculated output frequencies are scaled by the single voltage at the input.

To plot the power gain of a circuit:

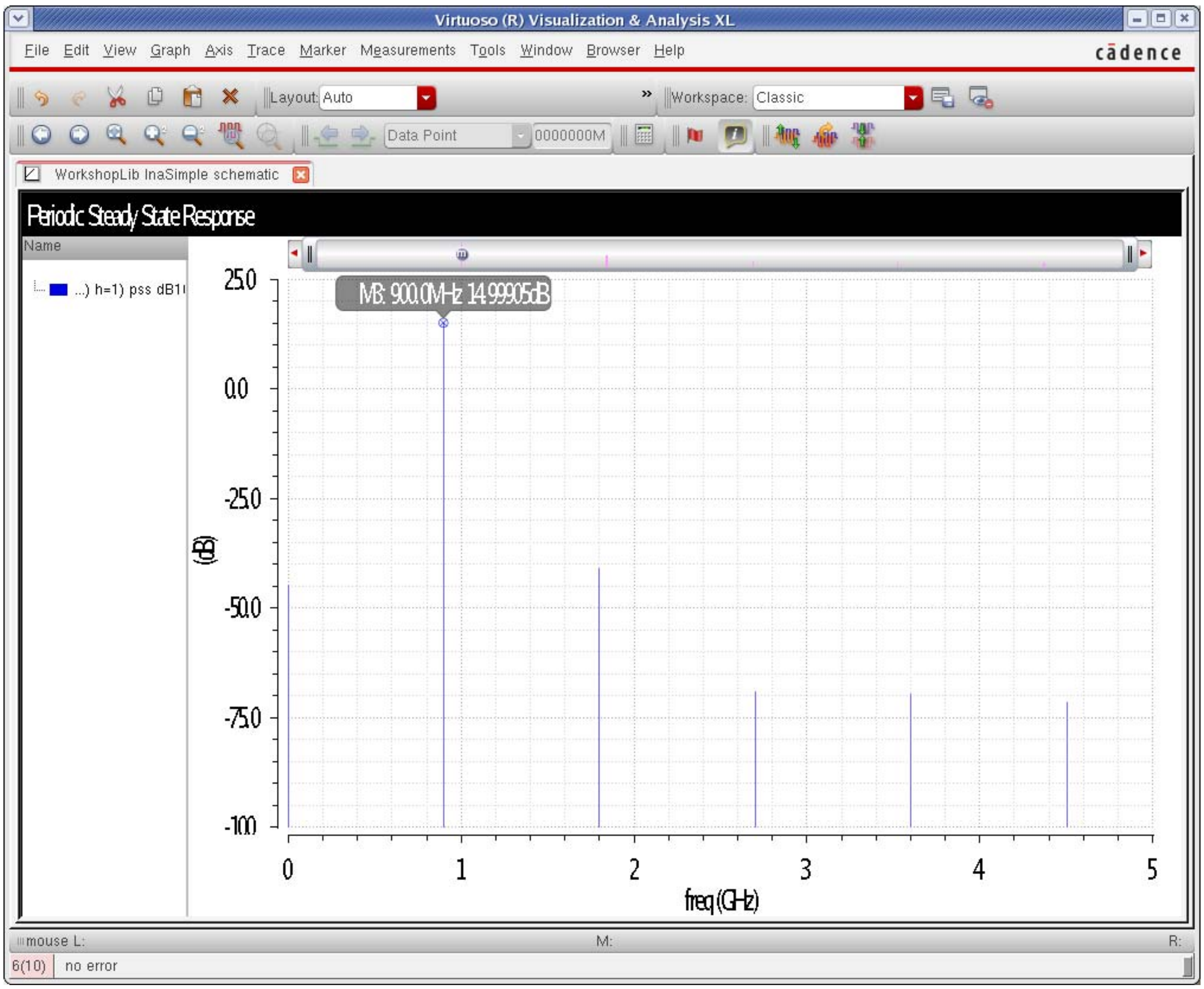


1. Select *Power Gain* from the *Function* section.
2. Select the appropriate modifier.
3. Select the input power.
4. Select the frequency of the input.
5. Select the terminal of the load.

6. Select the terminal of the input source.



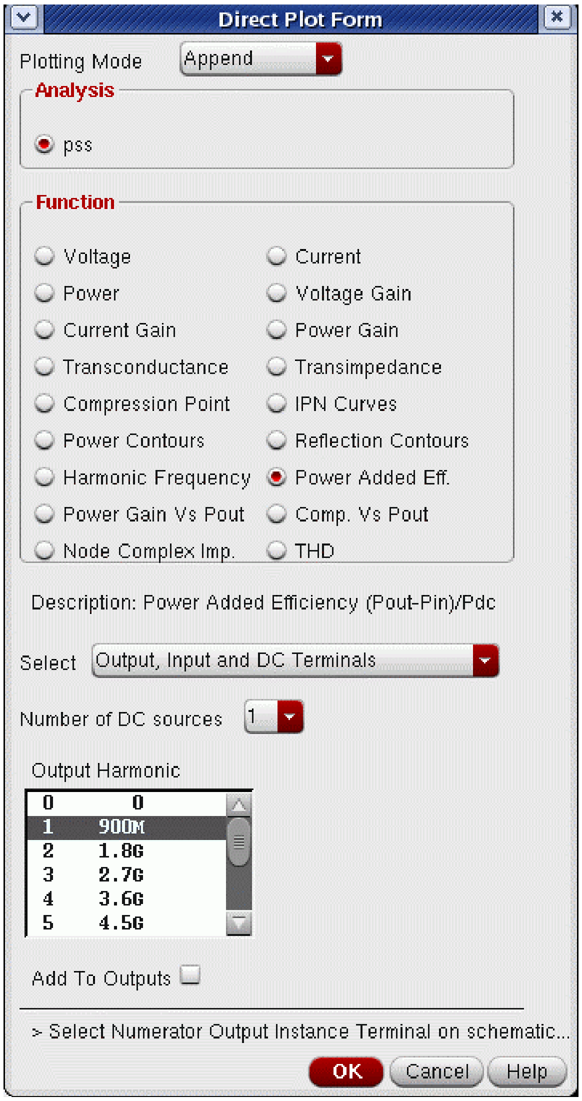
Below is the output power spectrum, divided by the magnitude of the input power at the input frequency.



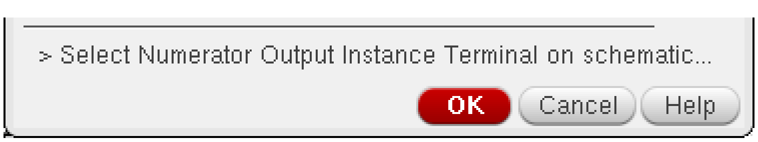
Power-Added Efficiency

This example shows a swept measurement. To set this up, see Loadpull on page 657. A sweep is not required for this measurement.

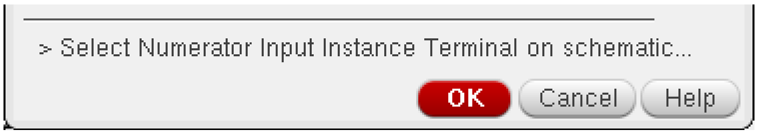
To plot power-added efficiency:



1. Select *Power Added Eff* from the *Function* section.
2. Select the frequency of the input.
3. Select the number of DC sources that power the circuit.
4. Select the terminal of the output load.

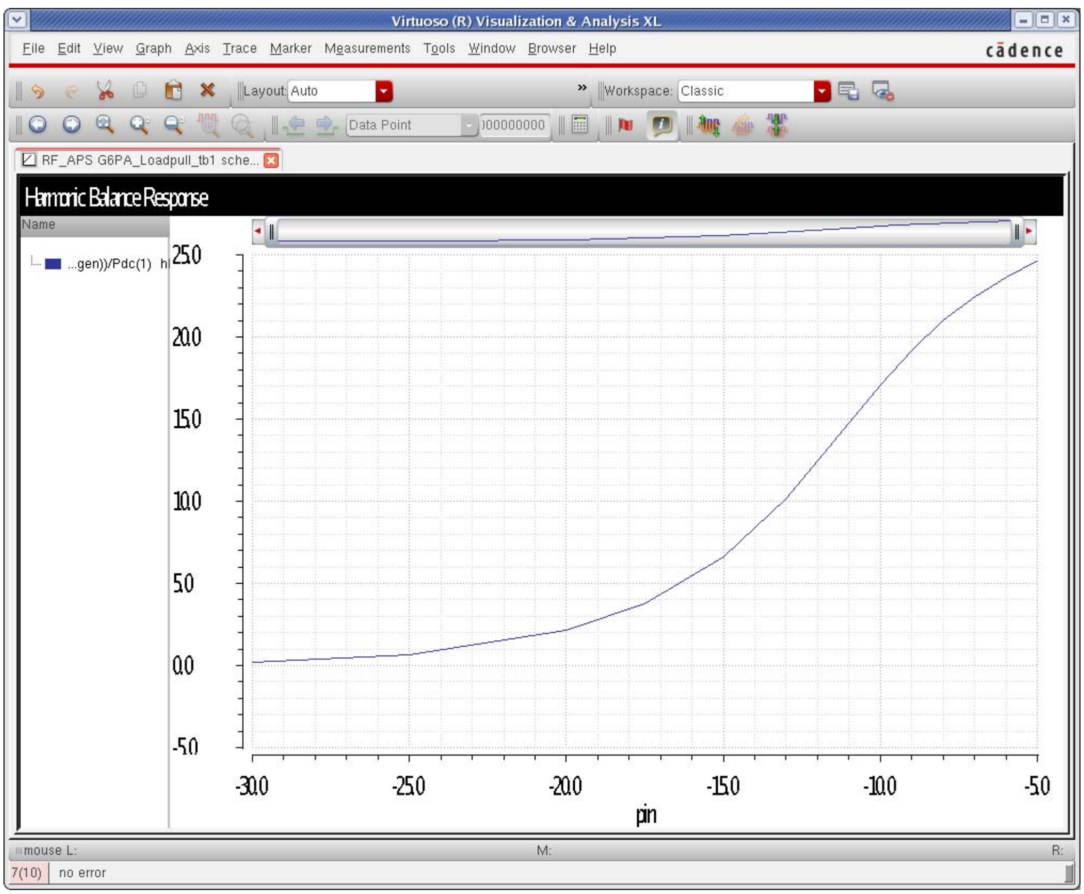


5. Select the terminal of the input source.



6. Select the terminal(s) of the DC sources.

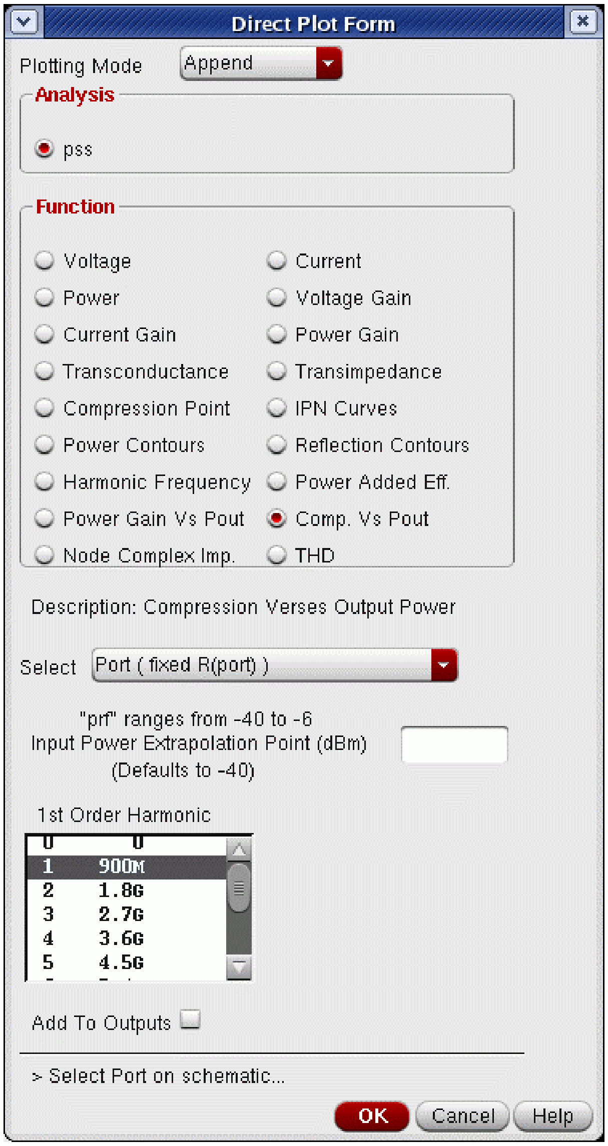
The swept PAE curve is displayed.



Compression Versus Output Power

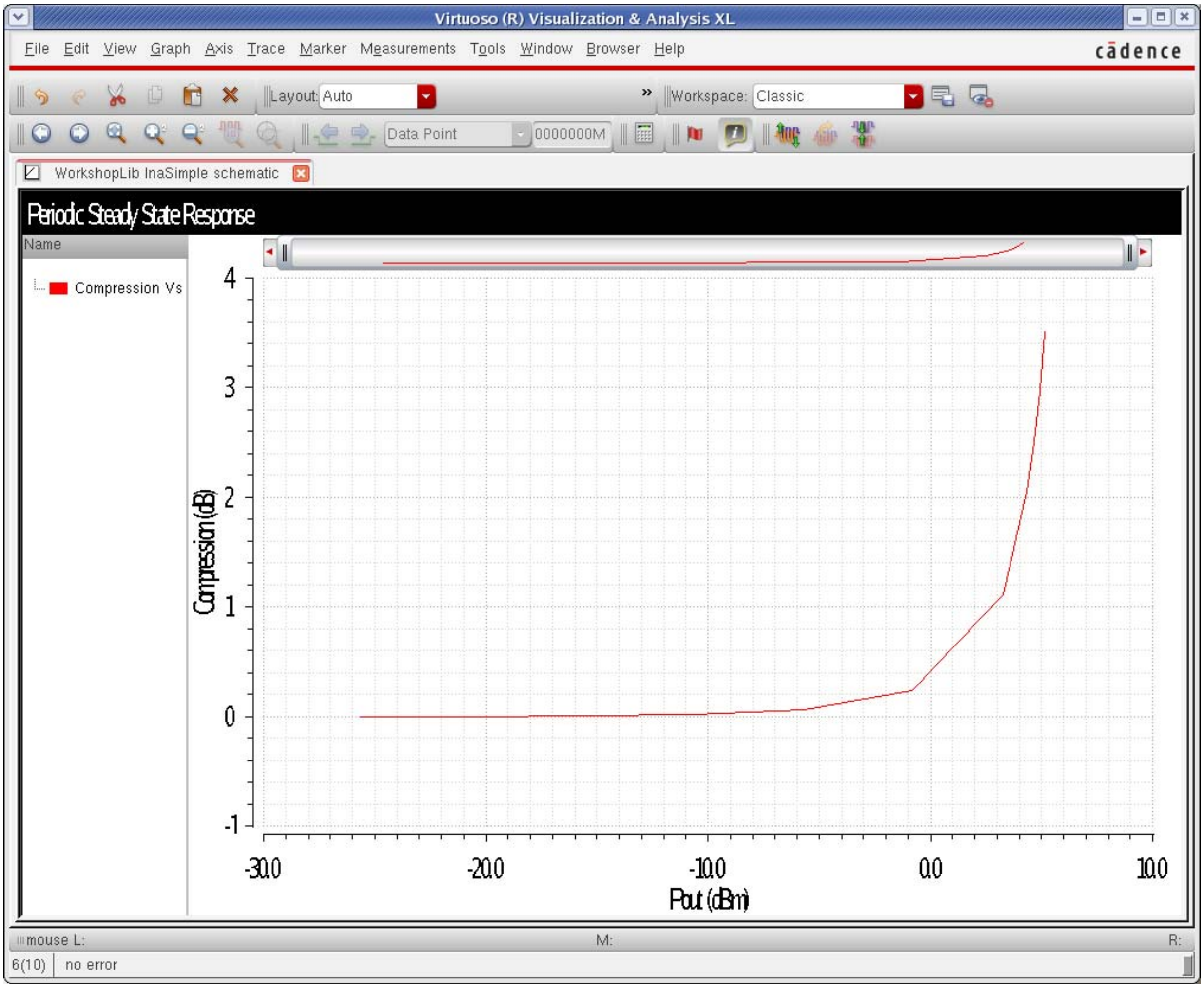
This example shows a swept measurement. To set this up, see Loadpull on page 657.

To plot compression versus output power:



1. Select *Comp Vs Pout* from the *Function* section.
2. Select the output frequency.
3. Select the load port in the circuit.

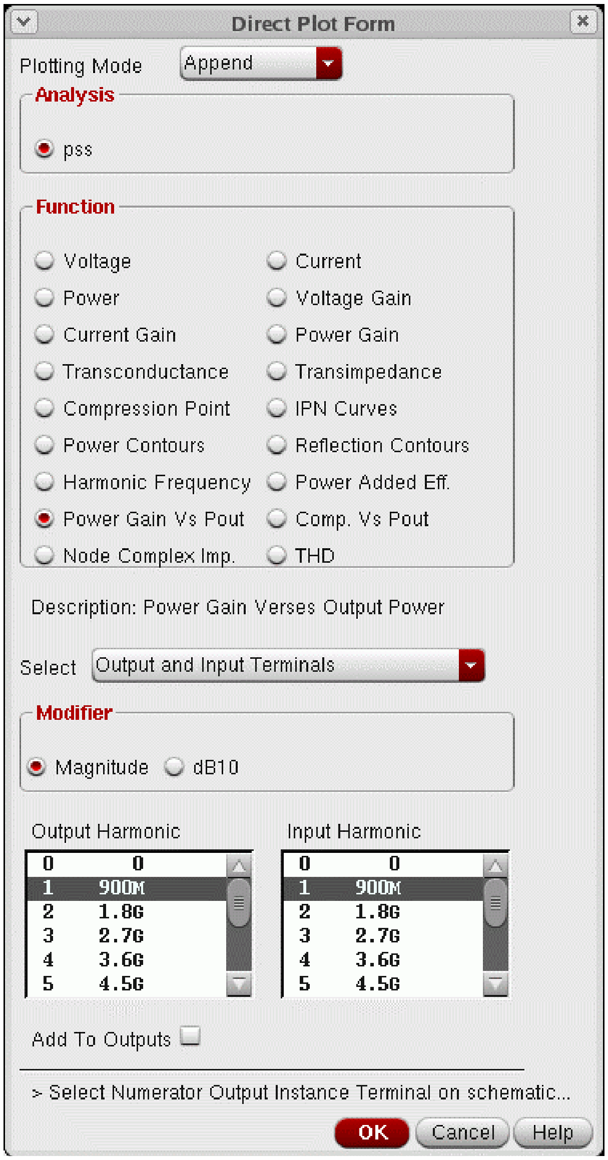
The compression curve is displayed. The assumption is that the first point in the sweep has no compression.



Power Gain Versus Output Power

This example shows a swept measurement. To set this up, see Loadpull on page 657.

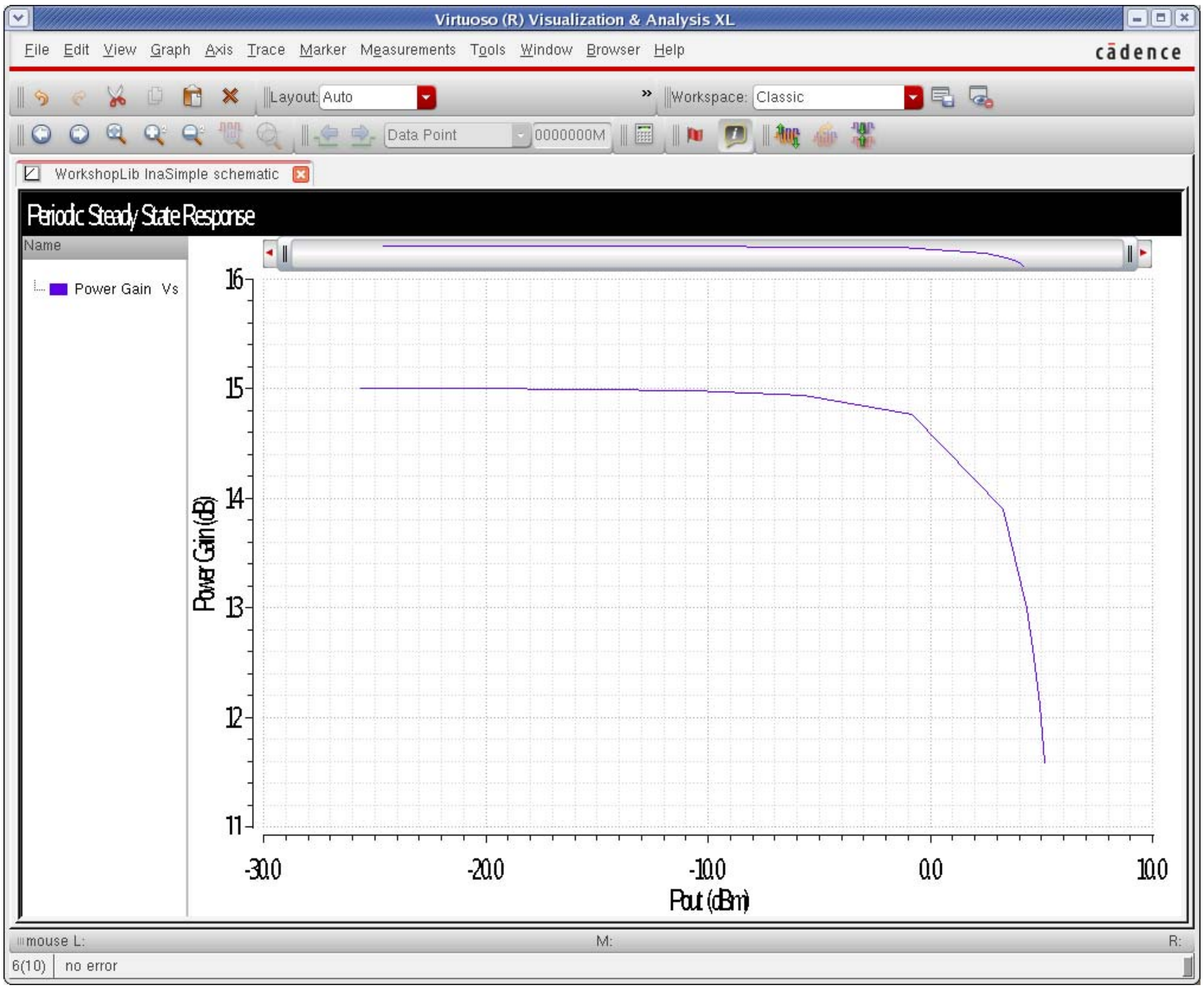
To see the power gain versus the output power:



1. Select *Power Gain vs Pout* from the *Function* section.
2. Select the appropriate modifier.
3. Select the output frequency.
4. Select the input frequency.
5. Select the terminal of the load.

6. Select the terminal of the input source.

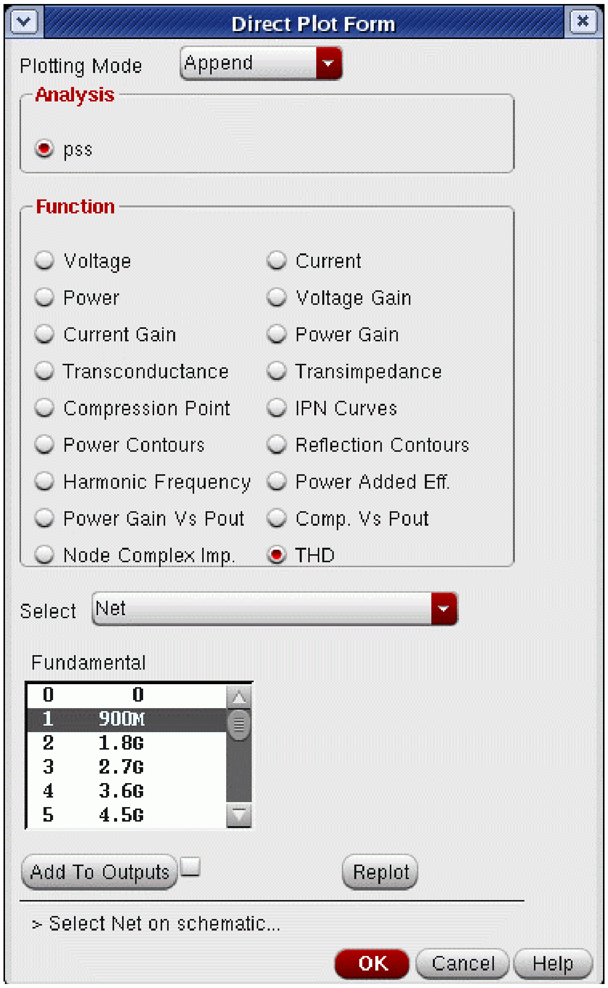
The plot is displayed. The vertical axis is the power gain. The horizontal axis is the swept input power.



Plotting THD

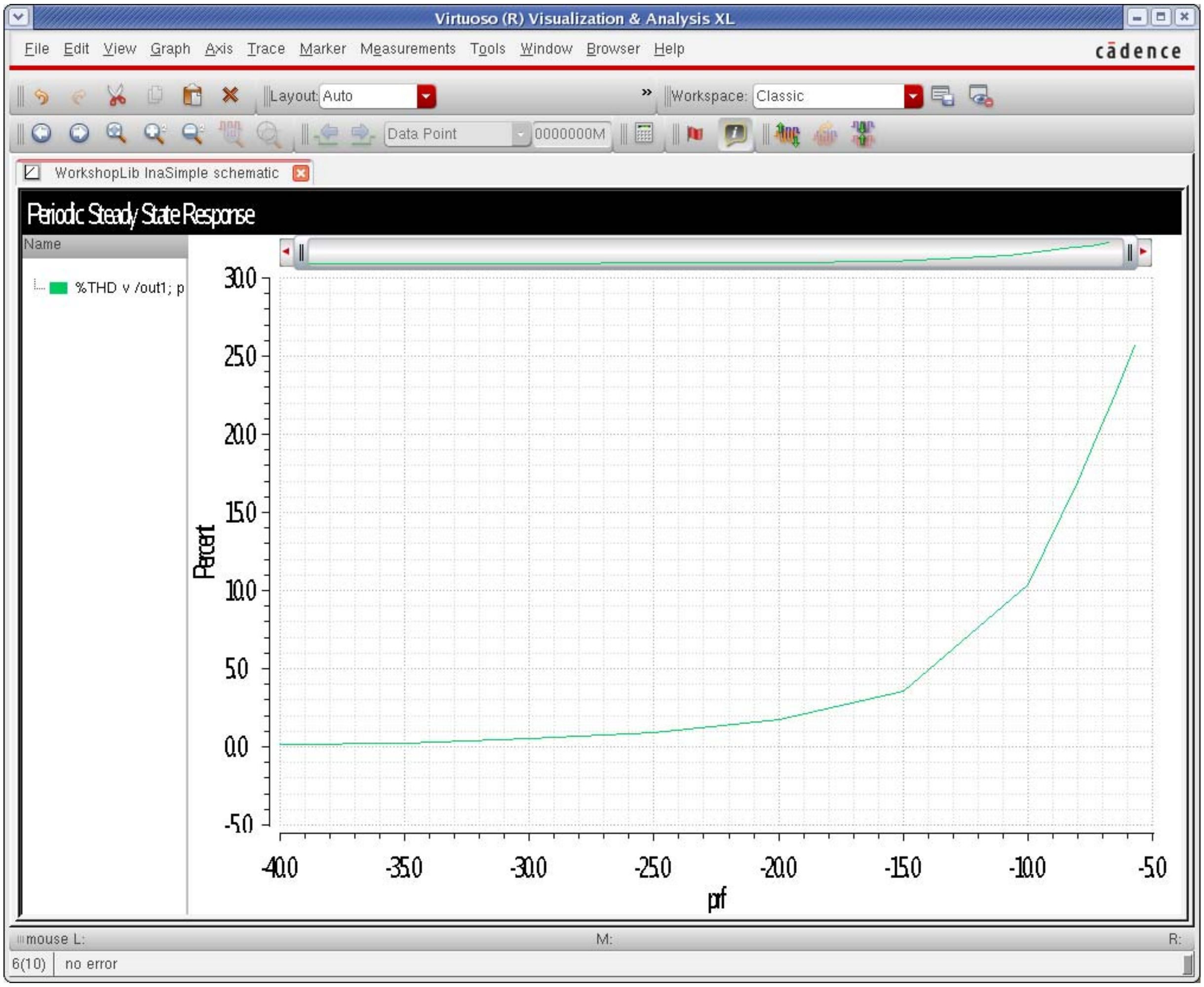
This example shows a swept measurement. To set this up, see Loadpull on page 657. A sweep is not required for this measurement.

To plot the total harmonic distortion:



1. Select *THD* from the *Function* section.
2. Select the output frequency.
3. Select the net in the schematic.

The THD versus input power plot is displayed.

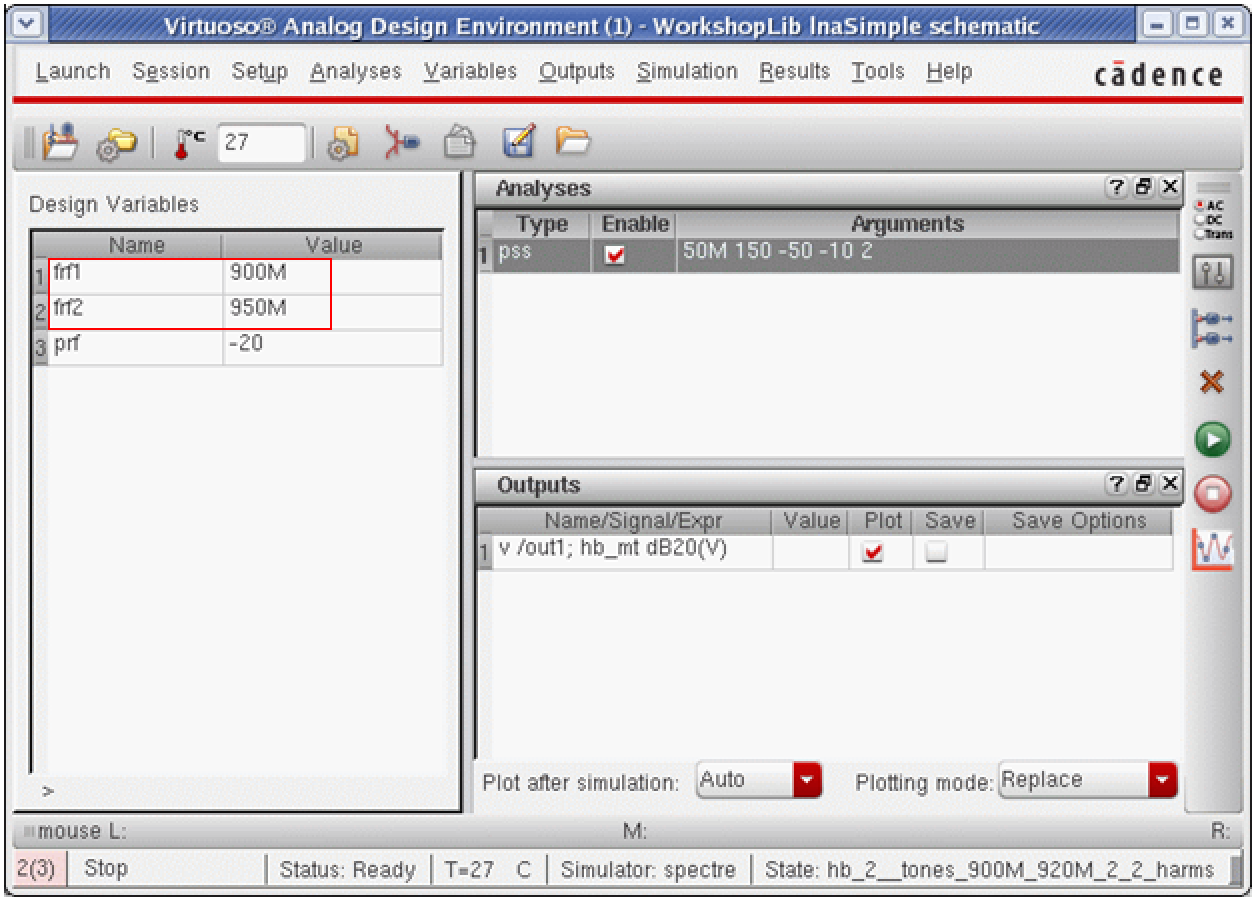


Large-signal IP3

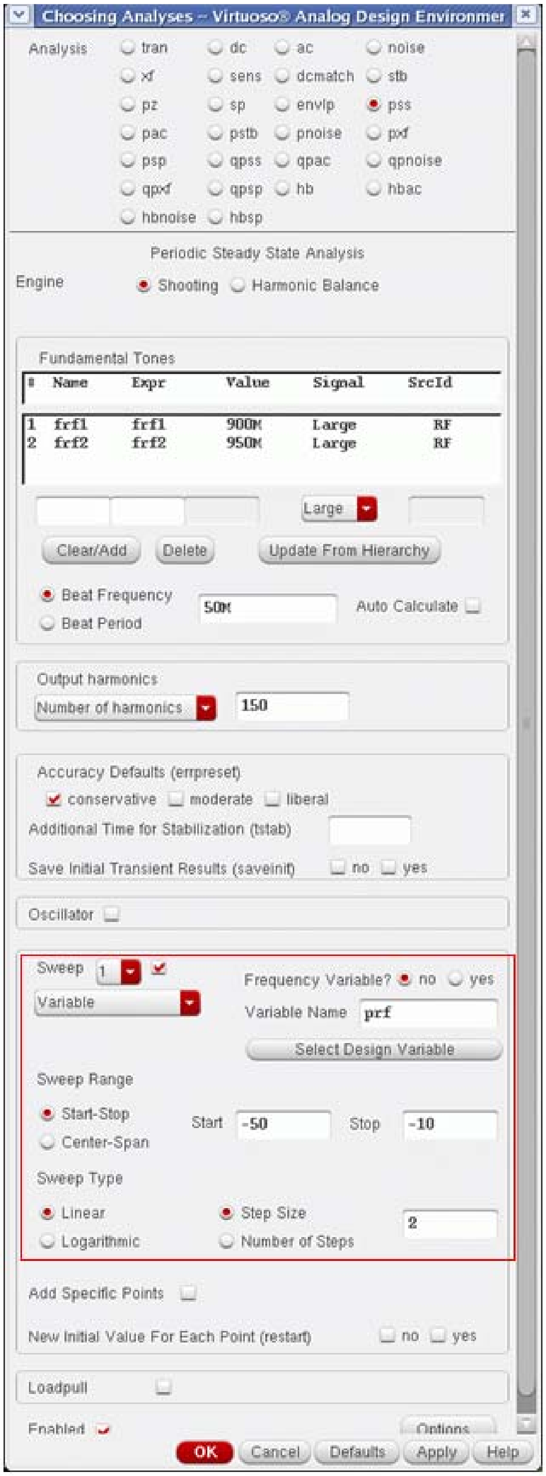
This example shows a swept measurement. To set this up, see Loadpull on page 657.

In pss, all the harmonics of the input waveform are calculated. If the input frequencies get close together, the pss beat frequency becomes small compared to the highest input frequency. If you need to simulate more than 25 periods of the highest input frequency, switch to hb or qpss which run faster than pss.

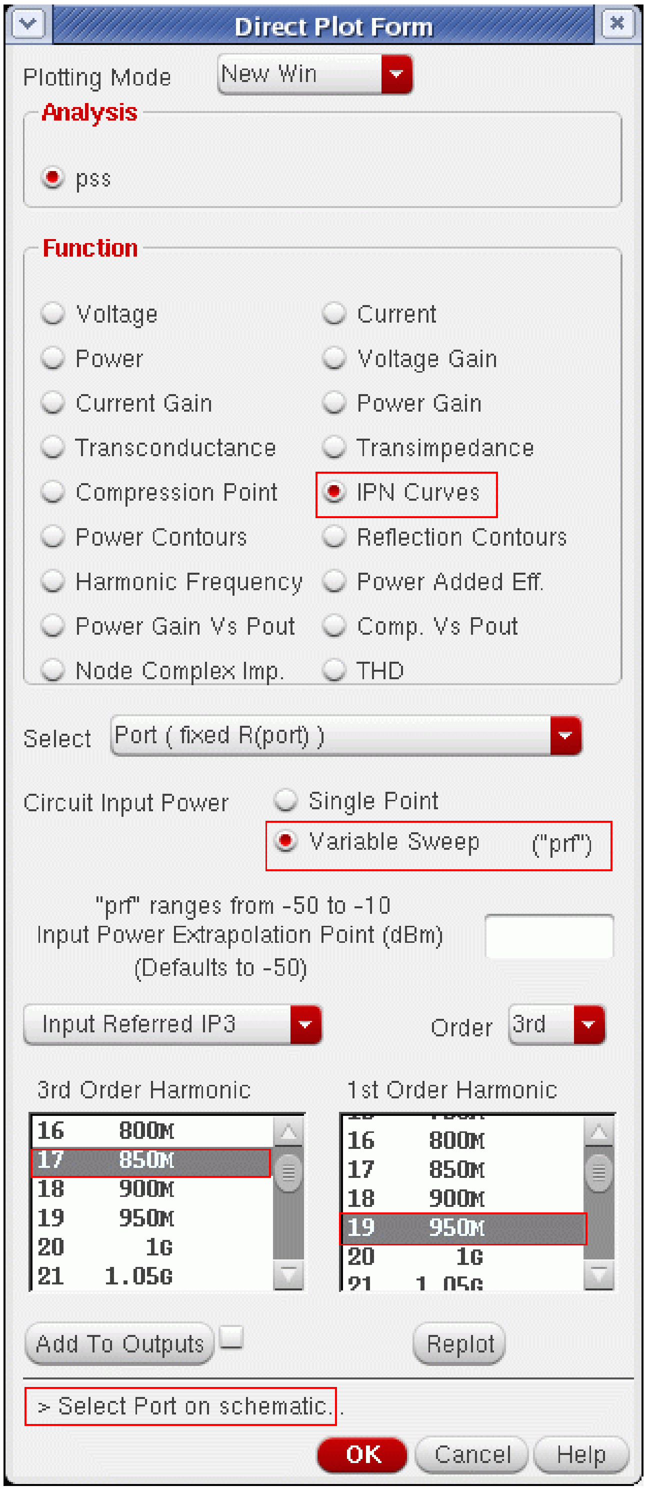
To plot a large-signal IP3, both inputs are enabled in the ADE window by specifying the values in the *Variables* section of the ADE for the frequency and amplitude of both RF sources. Note that in this case, the highest frequency (950MHz) divided by the beat frequency (50MHz) is 19. This is a reasonable problem for pss.



The input is swept over a reasonable range of power levels. If the lower boundary of the sweep is too small, the noise floor of the simulator will be present in the result. If the input power gets too high, the inter-modulation product gets unpredictable. In this example, the power is deliberately started at a very small value and the stopping power is in a very nonlinear region. This is done so the proper way to plot an IP3 can be shown later,

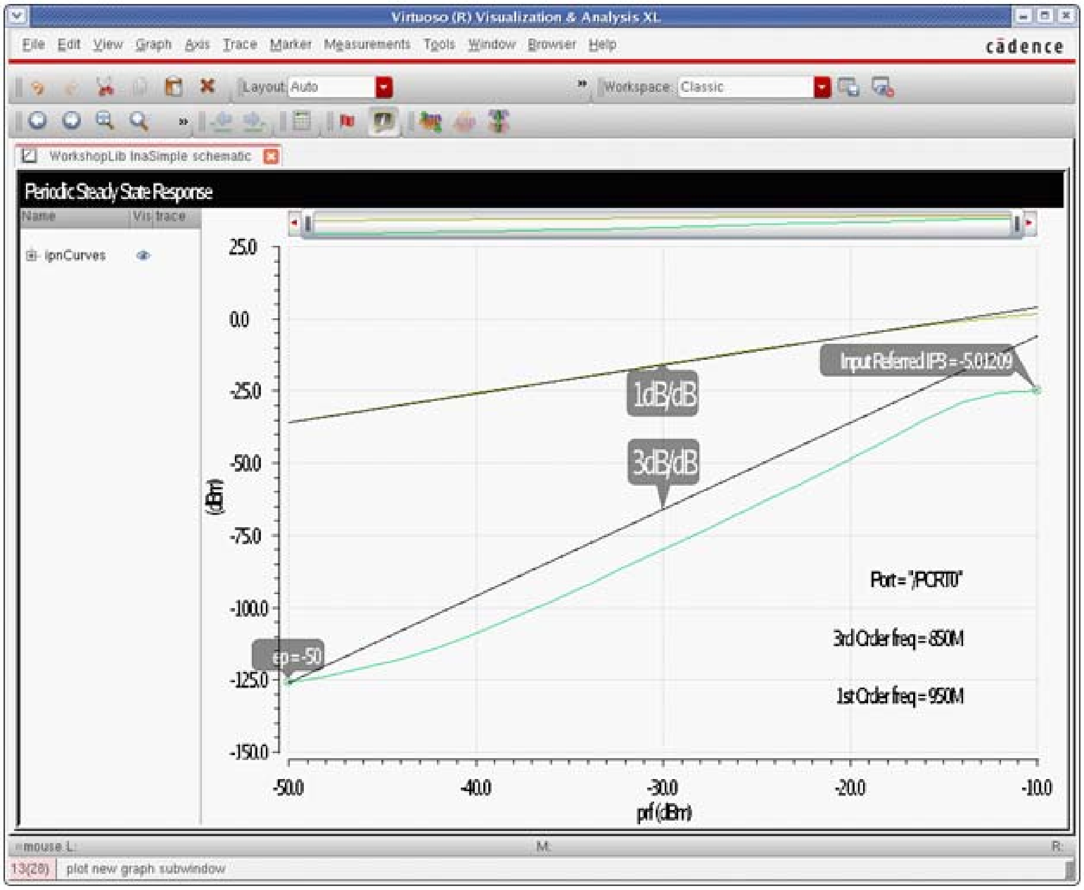


1. When the setup is complete, run the simulation.
2. When the simulation completes, select *Results - Direct Plot - Main Form* in ADE.



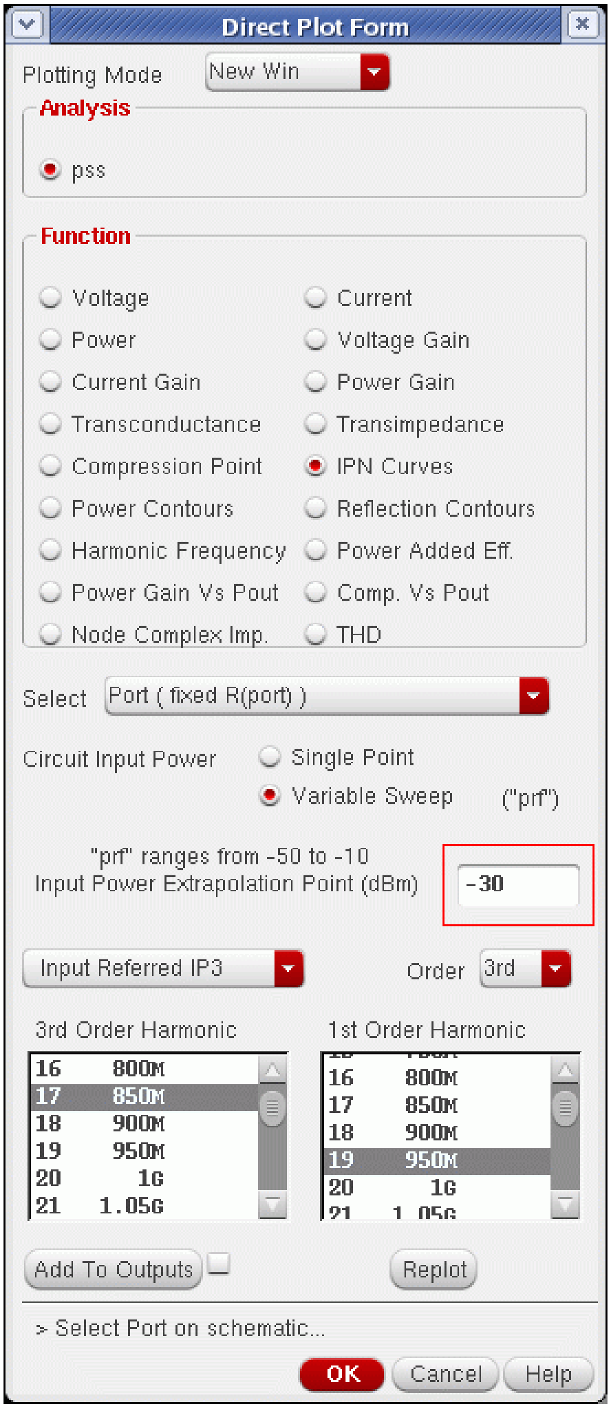
3. Select *IPN Curves* from the *Function* section.
4. Select the type of output signal to plot (or Port) The default is to select a port in the schematic.
5. Select *Variable Sweep*.
6. Set the order of the curves to be plotted. Usually, the order is 2nd or 3rd.
7. Select whether you want input or output-referred IP3 or IP2.
8. Leave the extrapolation point blank for the first plot (explained later).
9. Select the third or second-order frequency from the 2nd or *3rd Order Harmonic* field.
10. Usually, you select the frequency that has the highest amplitude, which is often the lower frequency term.
11. Select the first-order frequency from the *1st Order Harmonic* field. Usually you want the smallest amplitude, which is often the higher frequency term.
12. Select the output port in the schematic.

The swept large-signal IP3 is plotted, as shown below.

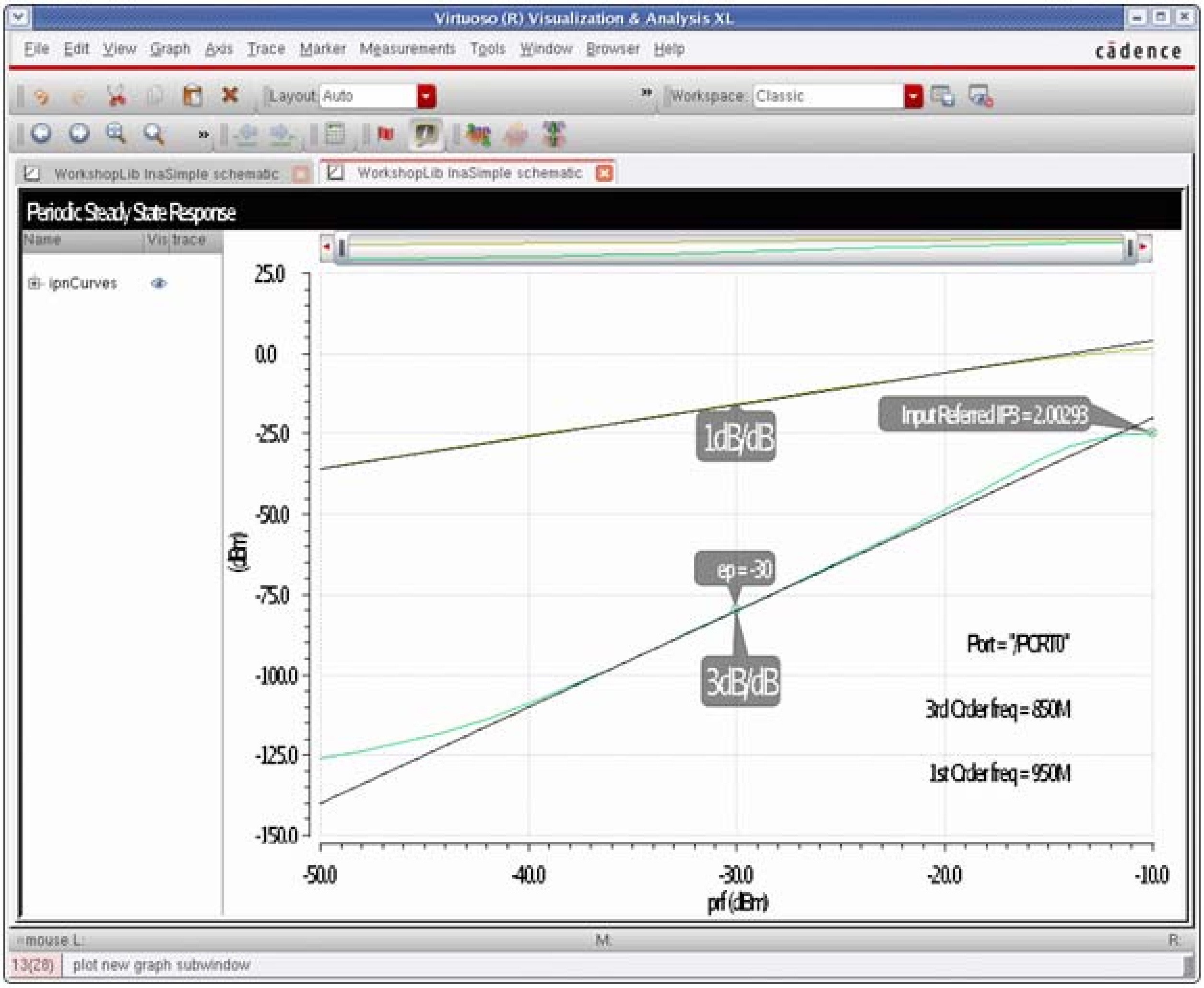


- Plotting IP3 always involves setting an extrapolation point. First plot the curve using the default for the extrapolation point.
- At high power, the third-order curve becomes unpredictable. At low power, the simulation has a noise floor. The data from the intermodulation product starts to level off.
- Set the extrapolation point above the noise floor, but below the large-signal region.
- The actual data for the intermodulation product should follow the ideal curve for at least one data point in the sweep above and below the extrapolation point.

- When -30 is set as the extrapolation point, the data for the third-order frequency follows the ideal curve above and below the extrapolation point, as shown below.



- Note that the IP3 calculation has increased by about 7dB. Make sure that the extrapolation point is set correctly to get the correct measurement.

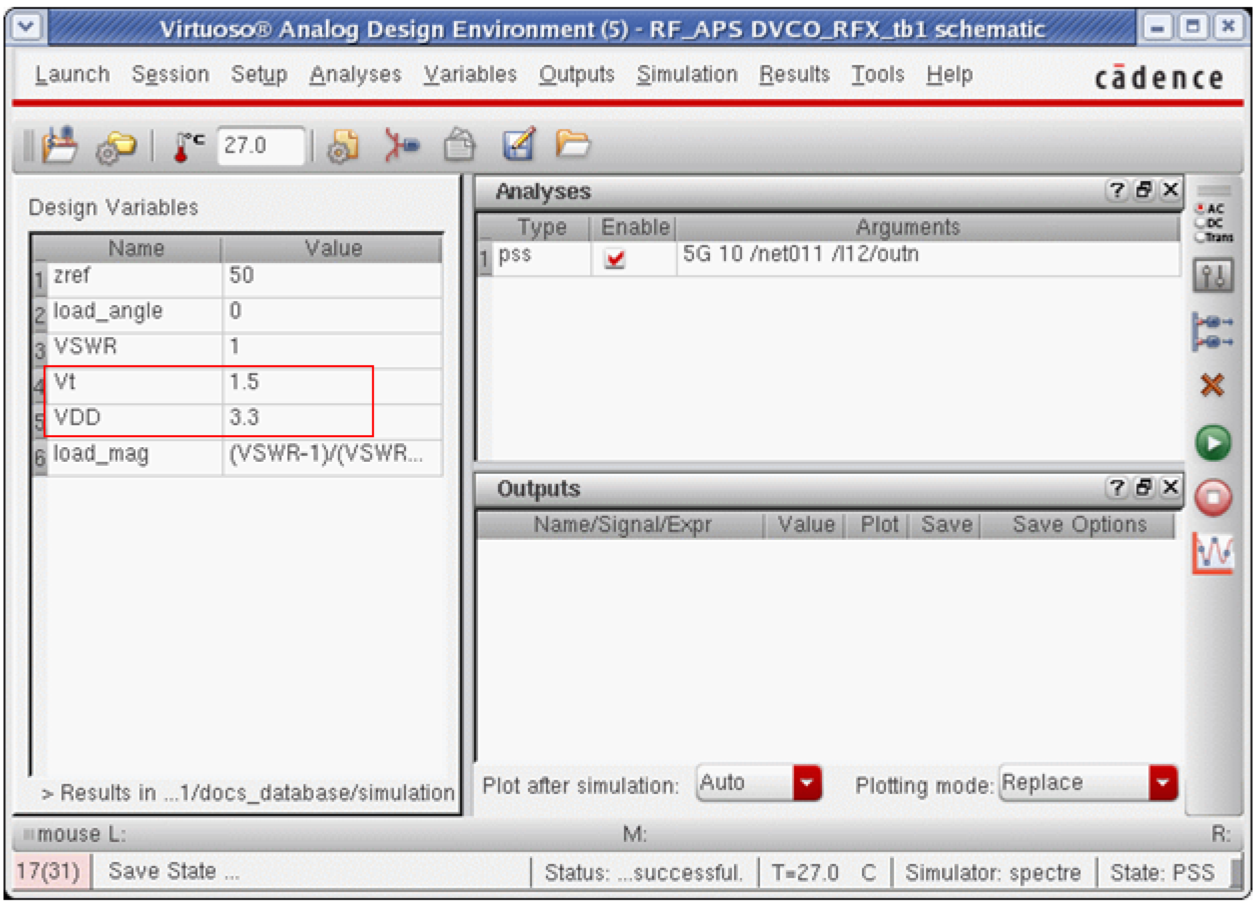


- The above plot is actually a small-signal IP3 measurement. If this is your objective, using rapid IP3 or IP2 in the AC *Choosing Analyses* form for amplifiers and in the PAC *Choosing Analyses* form for circuits that translate frequencies will run much faster and will provide a solution that is just as accurate.
- Note that `bsim3` and `bsim4` models have a limitation that causes the IP3 measurement to become inaccurate for switched-FET mixers whether it is done in rapid IP3, pss, qpss, or hb. In this case, the psp model is highly recommended. If this model is not available from the foundry, use pss, qpss, or hb at as high an input amplitude as possible without causing large-signal behavior in the third-order product.

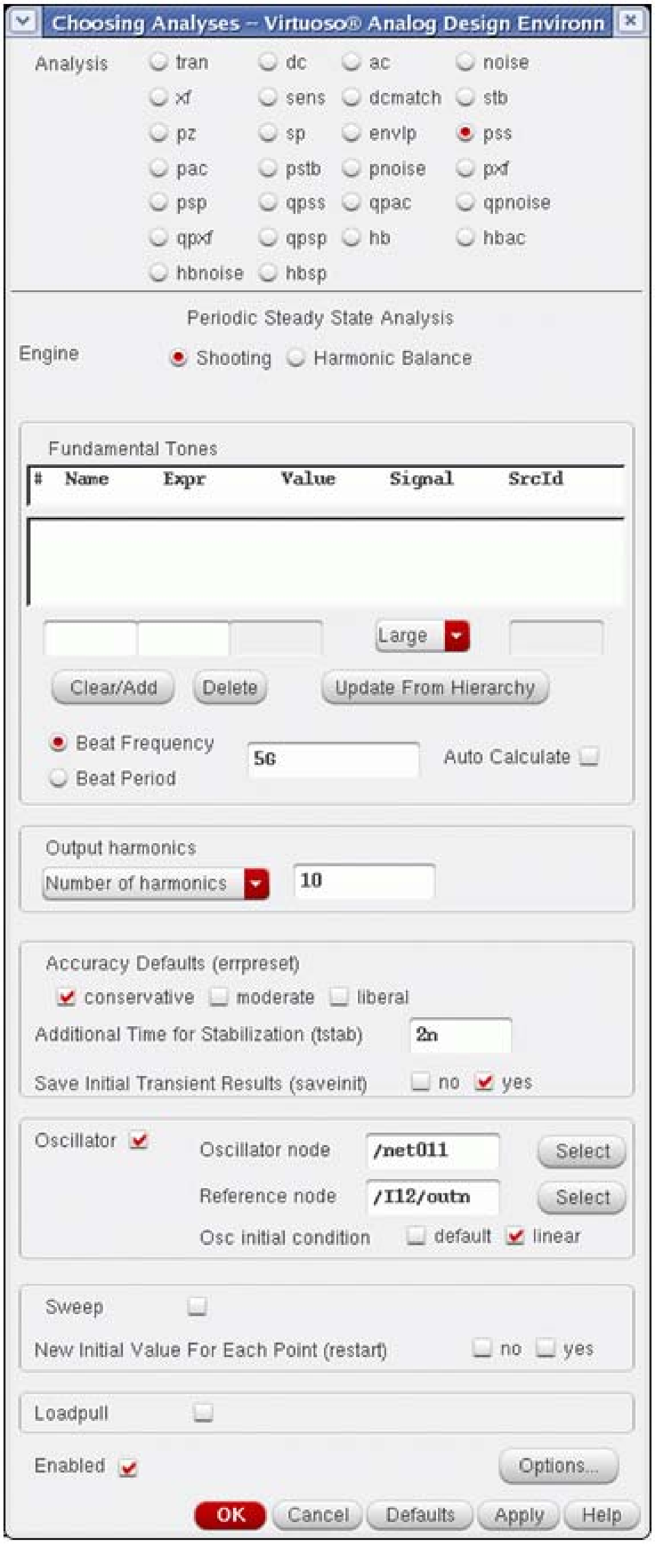
Oscillator Simulation

There are no input signals for an oscillator. In the pss analysis, the waveform needs to be calculated. To do this, both the frequency and the amplitudes need to be solved. This requires more iterations to calculate the solution.

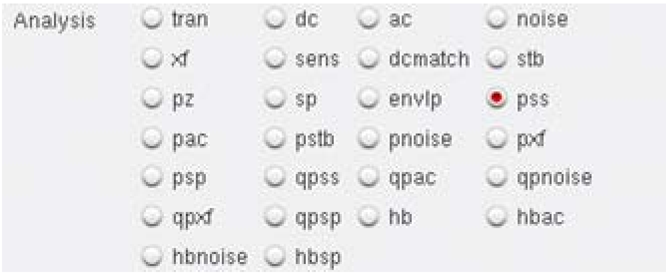
In the ADE, variables are used to set the tuning voltage and the power supply voltage.



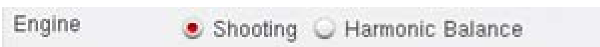
In the pss *Choosing Analyses* form:



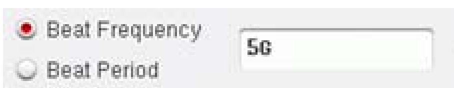
1. Select *pss*.



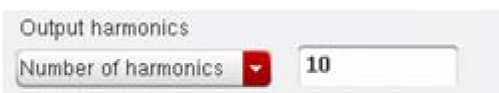
2. Select the engine that you want to use.



3. Specify a frequency estimate that is between 0.5 and 1.5 times the actual oscillation frequency if you are using the linear oscillator initial condition, or from 0.25 to four times the actual oscillating frequency if you use the default oscic.



4. Set the number of harmonics you want for the spectral calculation.



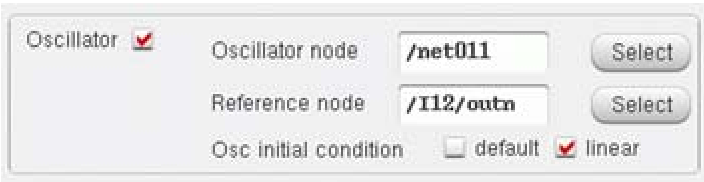
5. Select the accuracy. *conservative* accuracy is recommended for oscillators.

6. Specify about zero to five periods for tstab when the linear oscic is used and about 100 periods if the default oscic is used on a feedback oscillator.

If you have a ring oscillator, set about three periods for tstab.

7. If you want to see the startup waveform, select yes for *saveinit*.

8. Select *Oscillator*. If you have a single-ended oscillator, specify one node inside the feedback system for the *Oscillator node*. If you have a differential oscillator, specify two nodes at mirror locations in the *Oscillator node* and the *Reference node*.



9. If you have a feedback oscillator, linear oscic is recommended because tstab can be drastically shortened, thus saving simulation time.

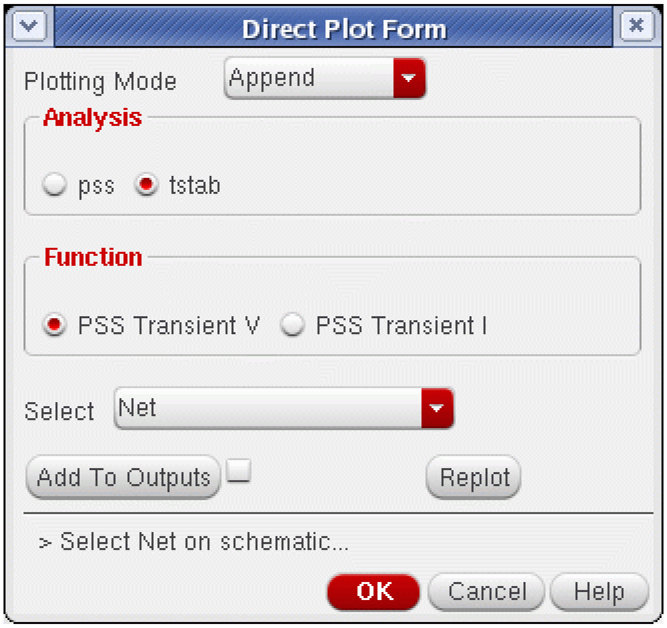
10. If you have a ring oscillator, set one stage high or low using initial conditions, and use the default oscic.

11. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main form* in ADE.

Plotting the Oscillator Startup Waveform

To plot the startup waveform, select tstab results and then select a node in the circuit. If you do not see the tstab results, go back to the *Choosing Analyses* form and select *yes* for *saveinit*.

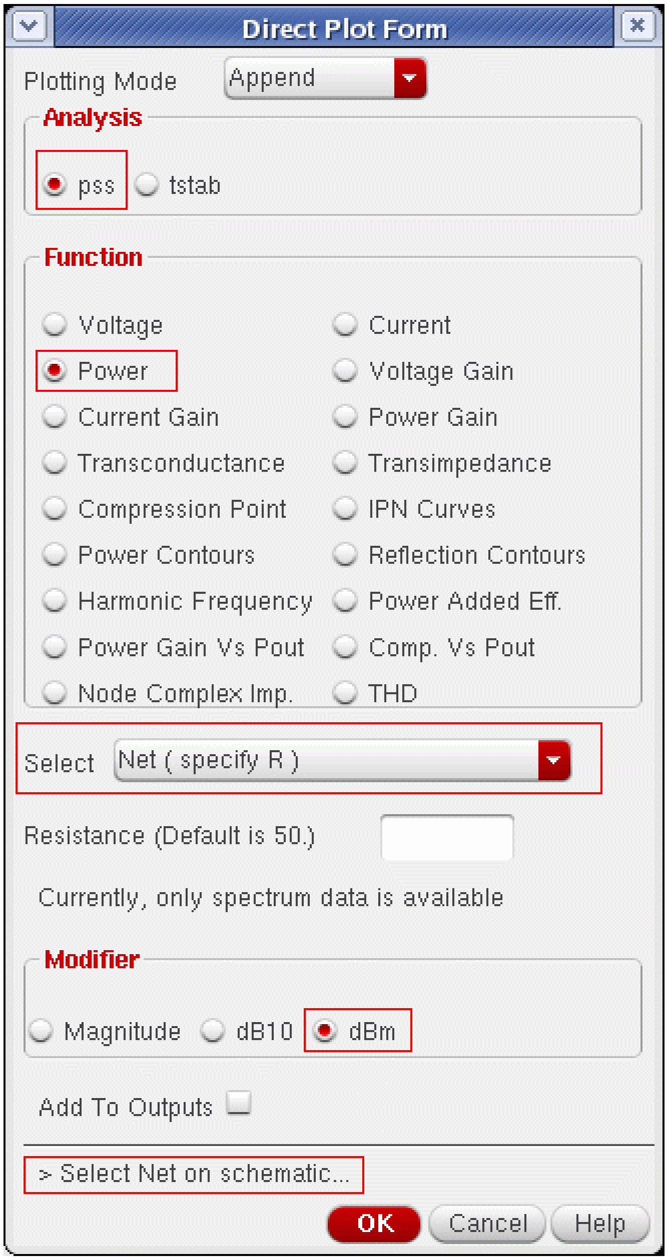
When the linear oscic is used, the startup characteristics of the waveform are not very evident. If you want to see the startup waveform to evaluate how robustly your oscillator starts up, use the default oscic and provide a very small starting kick at time zero plus so that the waveform starts small in amplitude and builds up. In this case, it is likely that tstab needs to be longer to allow the oscillator to reach a near steady-state operation before going into the pss solver.



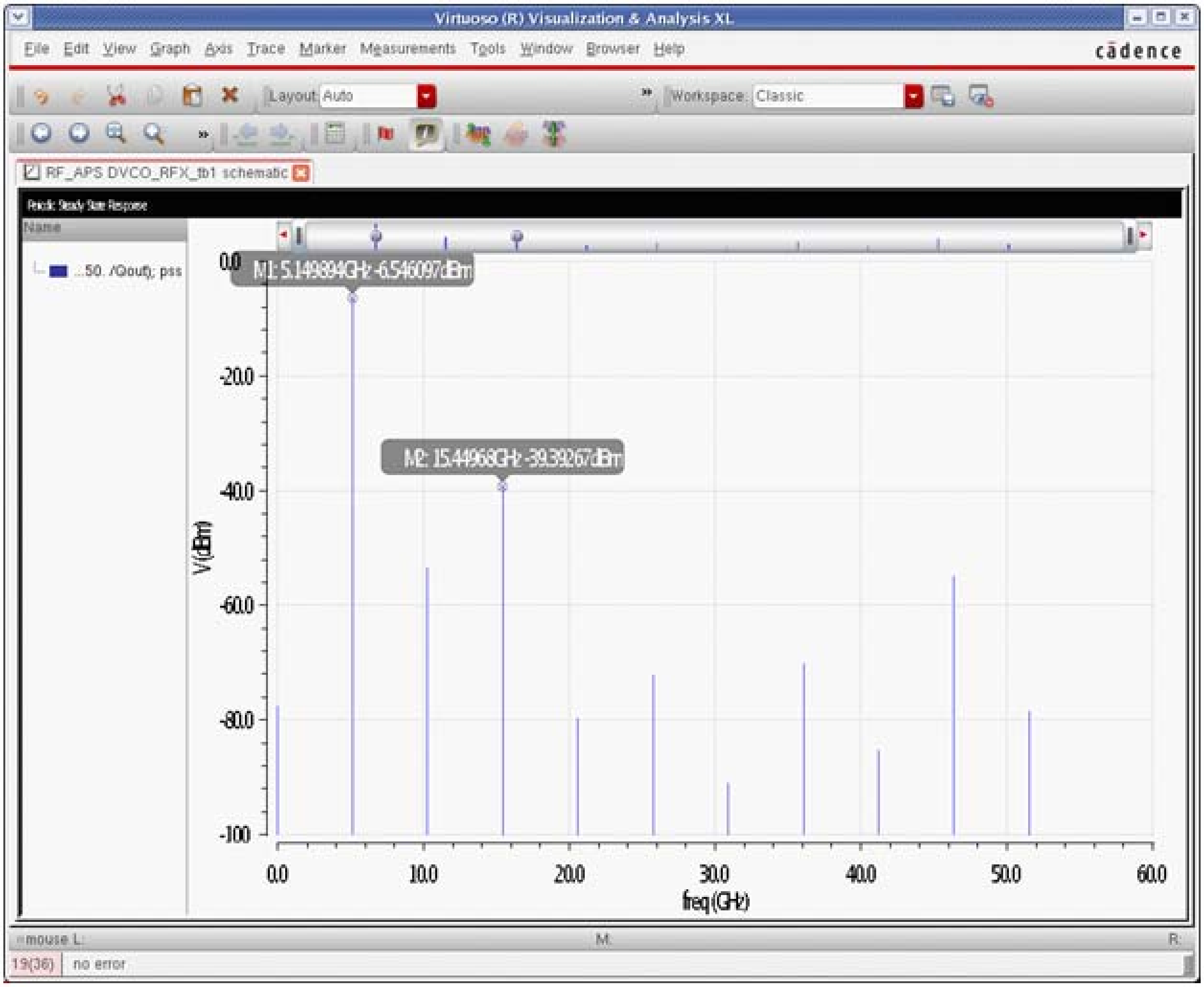
The result is displayed in the waveform tool. Note that with the linear oscic, the oscillator starts at time = zero with the estimate of the frequency and amplitude from the very beginning. This allows a very short tstab. Also note that the simulation continues after the setting of tstab. During this time, Spectre searches the oscillator for a divided-down output frequency.

Plotting the Oscillator Output Spectrum

To plot the output spectrum, select *pss*, and then select either *Voltage* or *Power*.



When a node is selected in the schematic, the power in dBm is plotted in the waveform tool, as shown below.

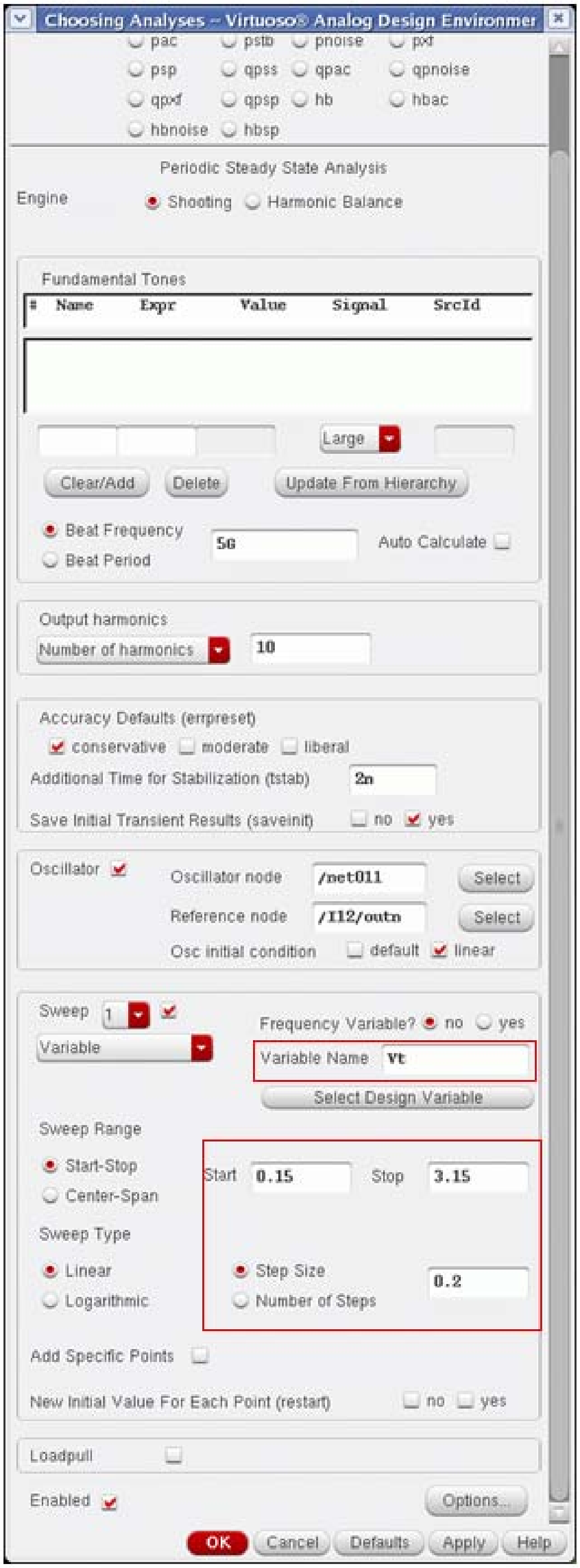


Additions for Oscillator Swept Tuning Voltage

In the *Choosing Analyses* form:

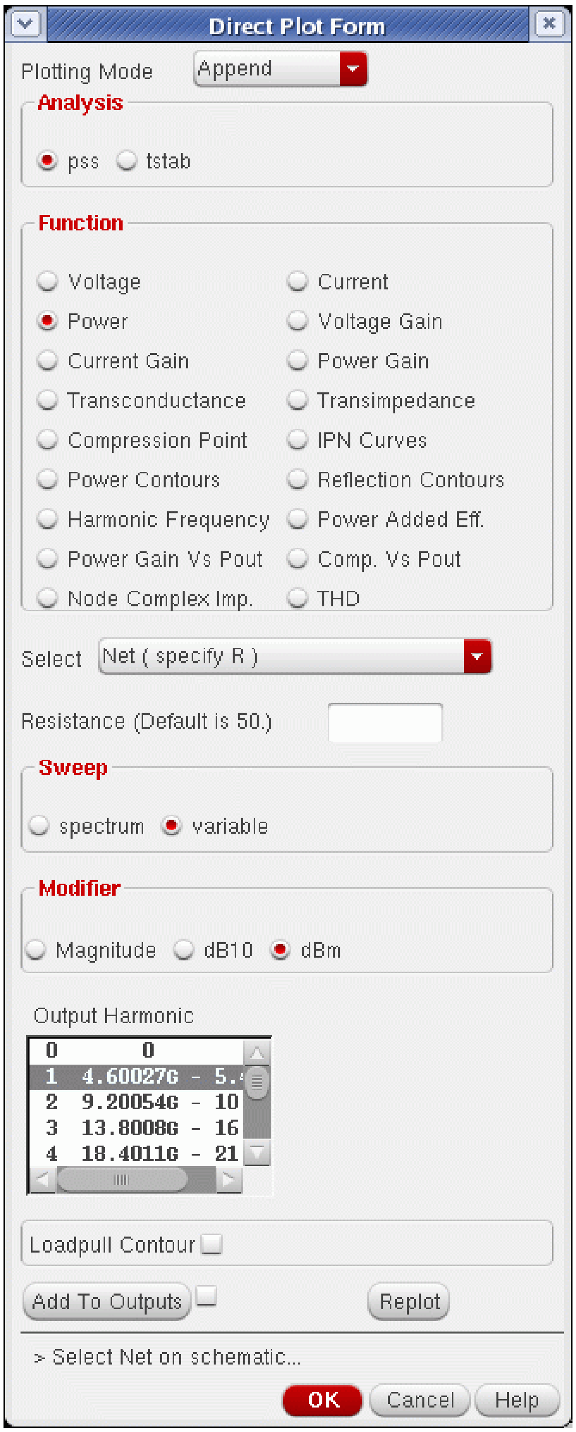
1. Select *Sweep*.
2. Type the variable name that sets the tuning voltage in the *Variable Name* field.

3. Set a reasonable sweep range for your oscillator.


Choosing Analyses -- Virtuoso® Analog Design Environmer
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
5G
Auto Calculate
Beat Period
Output harmonics
Number of harmonics
10
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
2n
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Oscillator node
/net011
Select
Reference node
/I12/outn
Select
Osc initial condition
default
linear
Sweep
1
Frequency Variable?
no
yes
Variable
Variable Name
Vt
Select Design Variable
Sweep Range
Start-Stop
Start
0.15
Stop
3.15
Center-Span
Sweep Type
Linear
Step Size
0.2
Logarithmic
Number of Steps
Add Specific Points
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

4. Run the simulation.

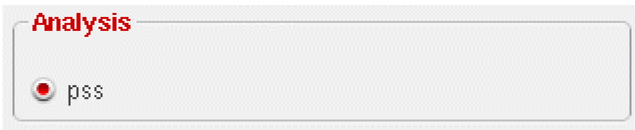
5. When the simulation completes, select Results - *Direct Plot - Main Form* in ADE. The *Direct Plot Form* is displayed.



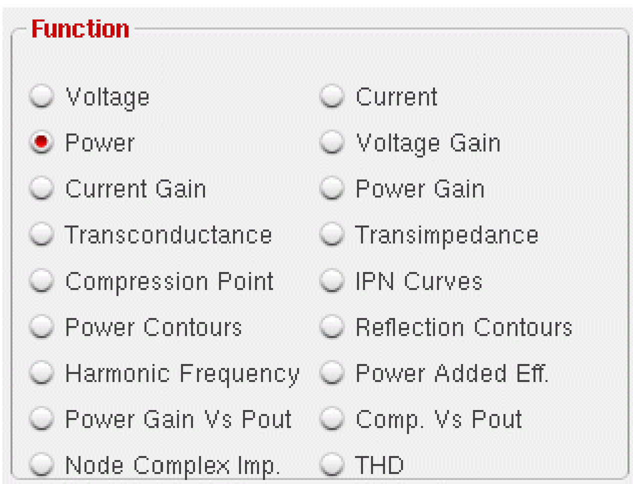
Plotting Output Power Versus Tuning Voltage

To plot the power as a function of the tuning voltage:

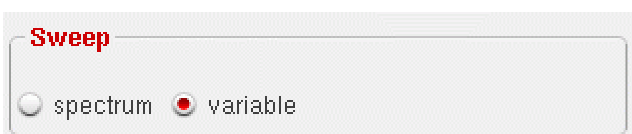
1. Select *pss* results.



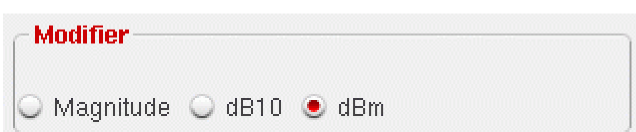
2. Select *Power*.



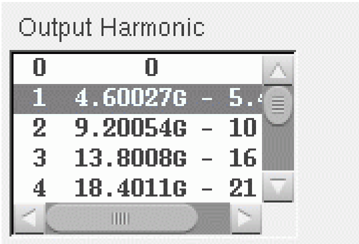
3. Select *variable* sweep.



4. Select *dBm*.

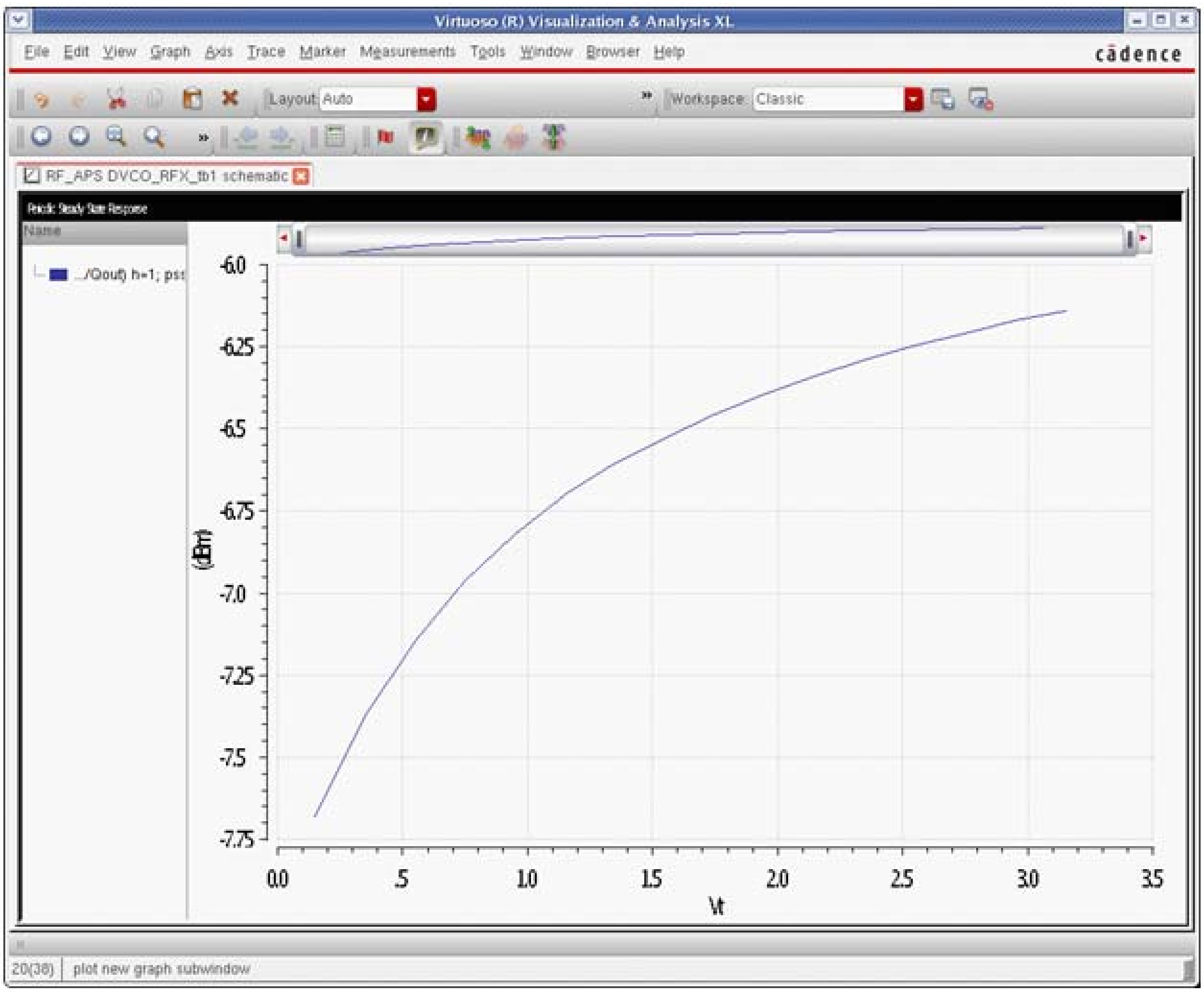


5. Select the harmonic number. The frequency range is also shown.



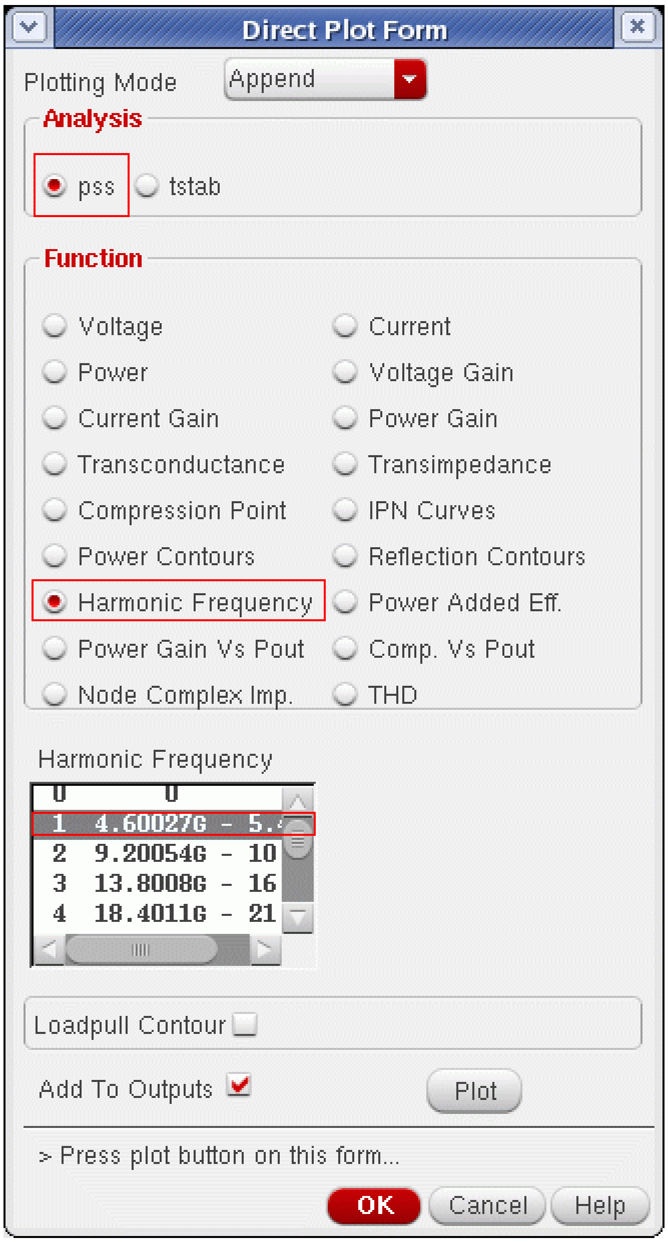
6. Select the node in the schematic you want to plot.

The output power versus the tuning voltage is plotted, as shown below.

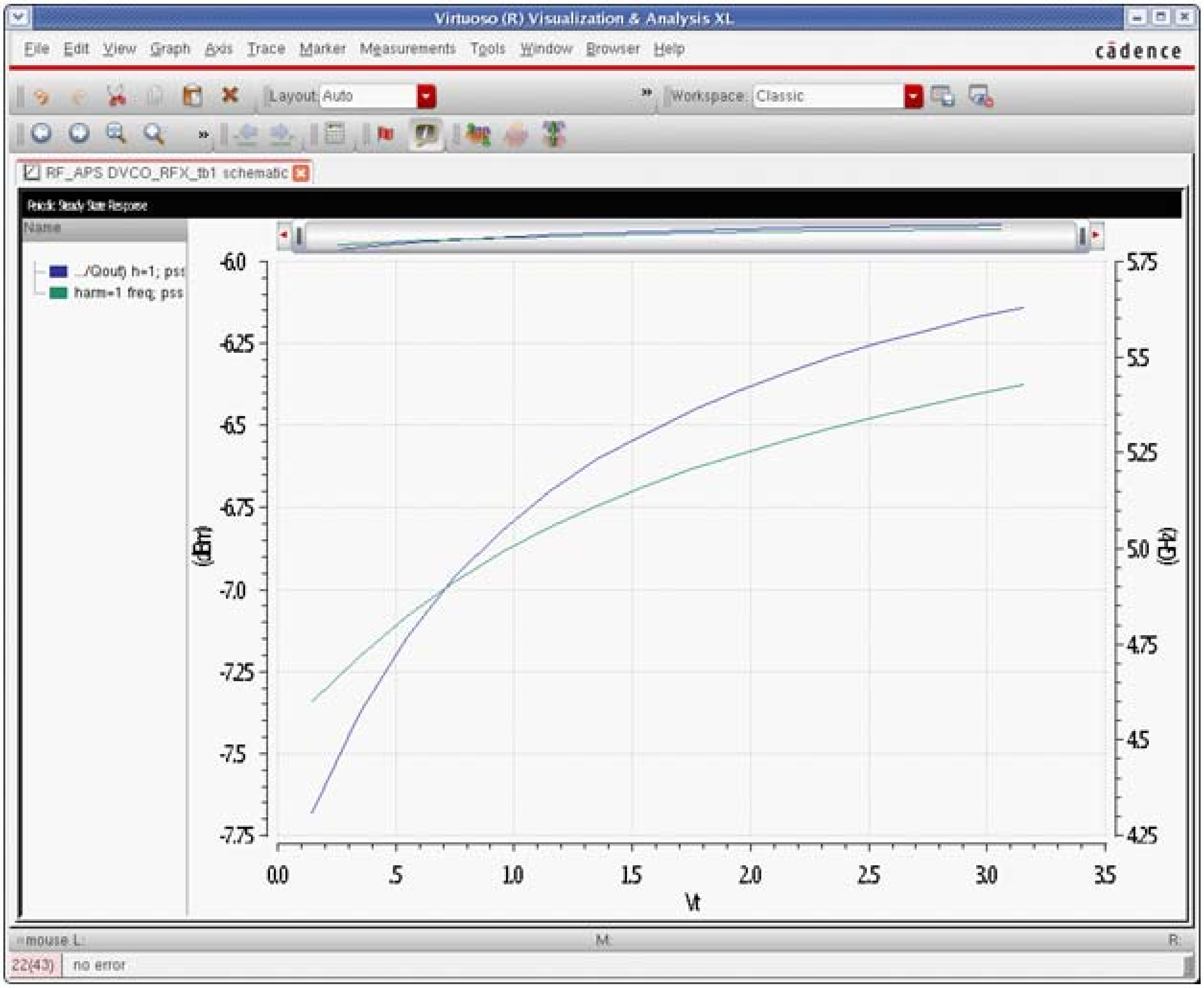


Plotting Output Power Versus Output Frequency

Now plot the harmonic frequency on the same plot.

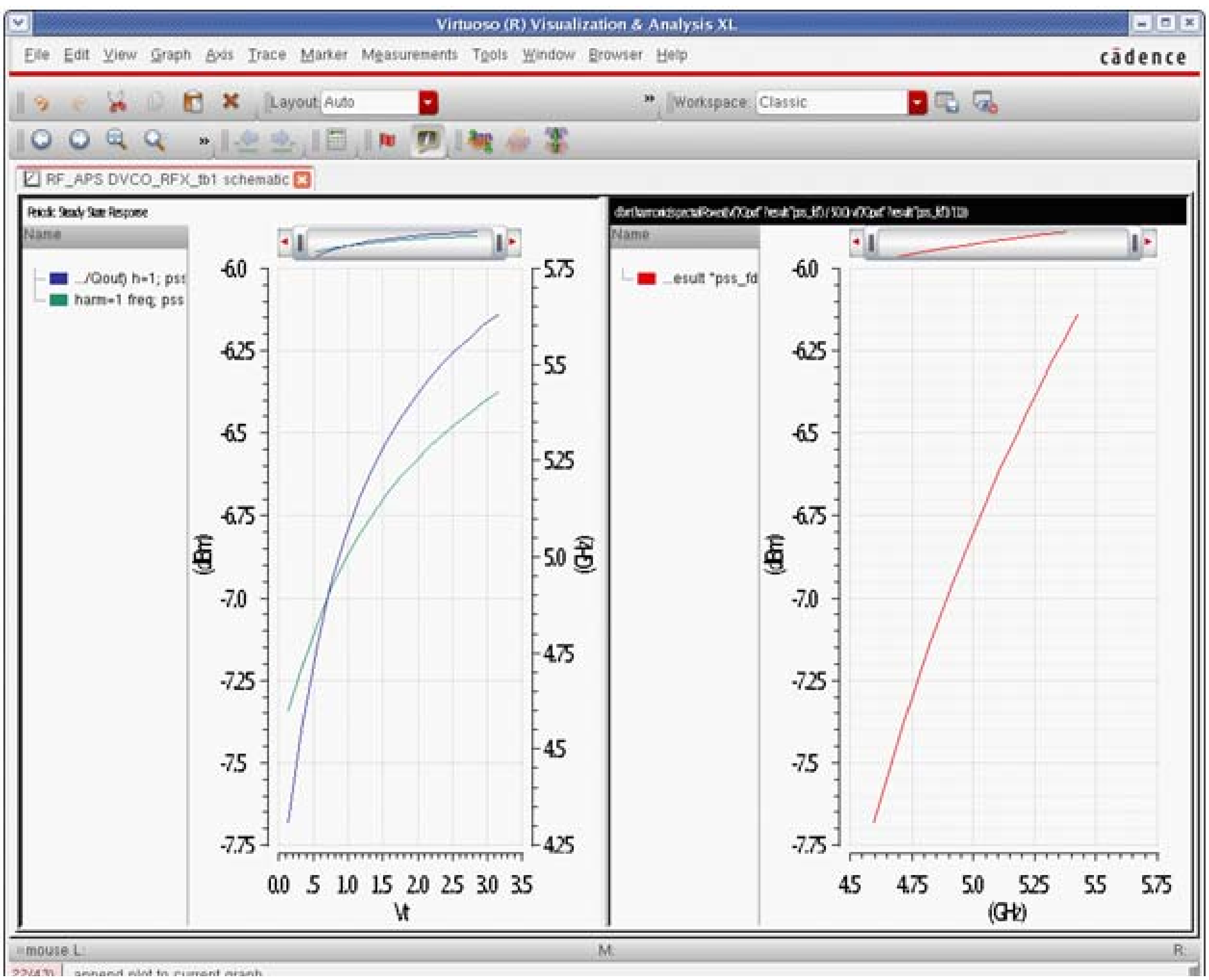


The frequency is plotted on the same subwindow. A second vertical axis is used.



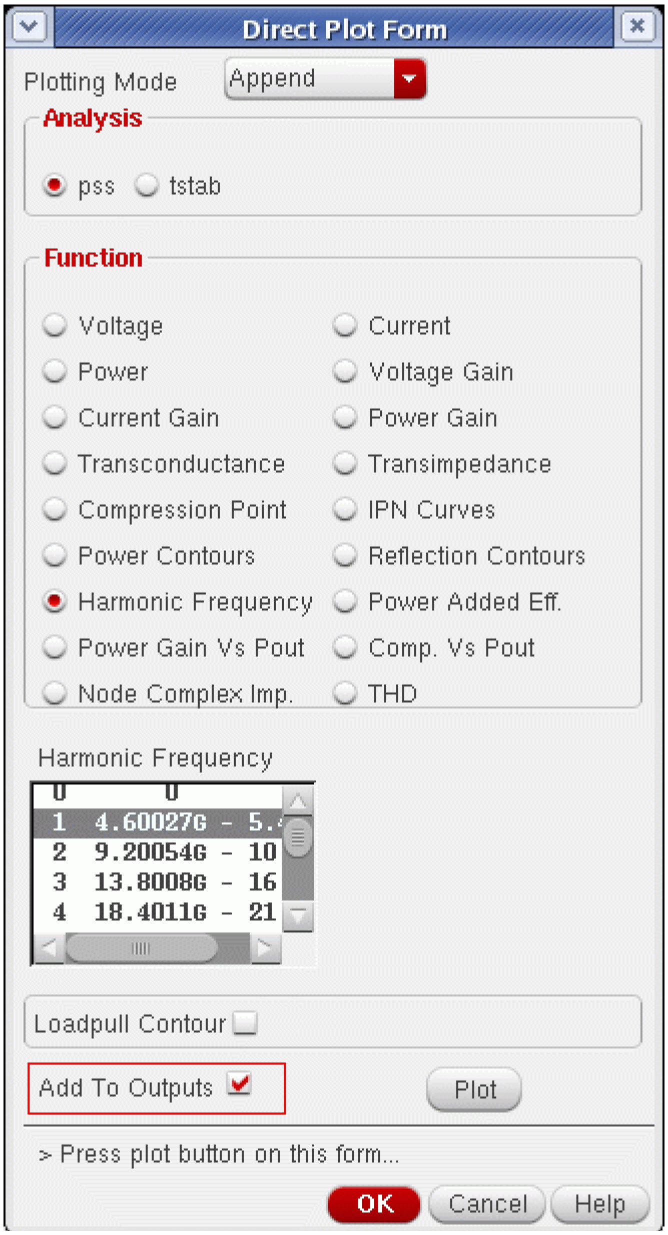
To plot the output power versus the frequency, right click on one of the numbers on the X Axis, and select Y vs. Y. The *Y vs Y* window appears. Select the frequency for the X Axis from the *Select a Trace* drop-down list and click *OK*.

The plot of the output power versus the output frequency is plotted in a new subwindow.

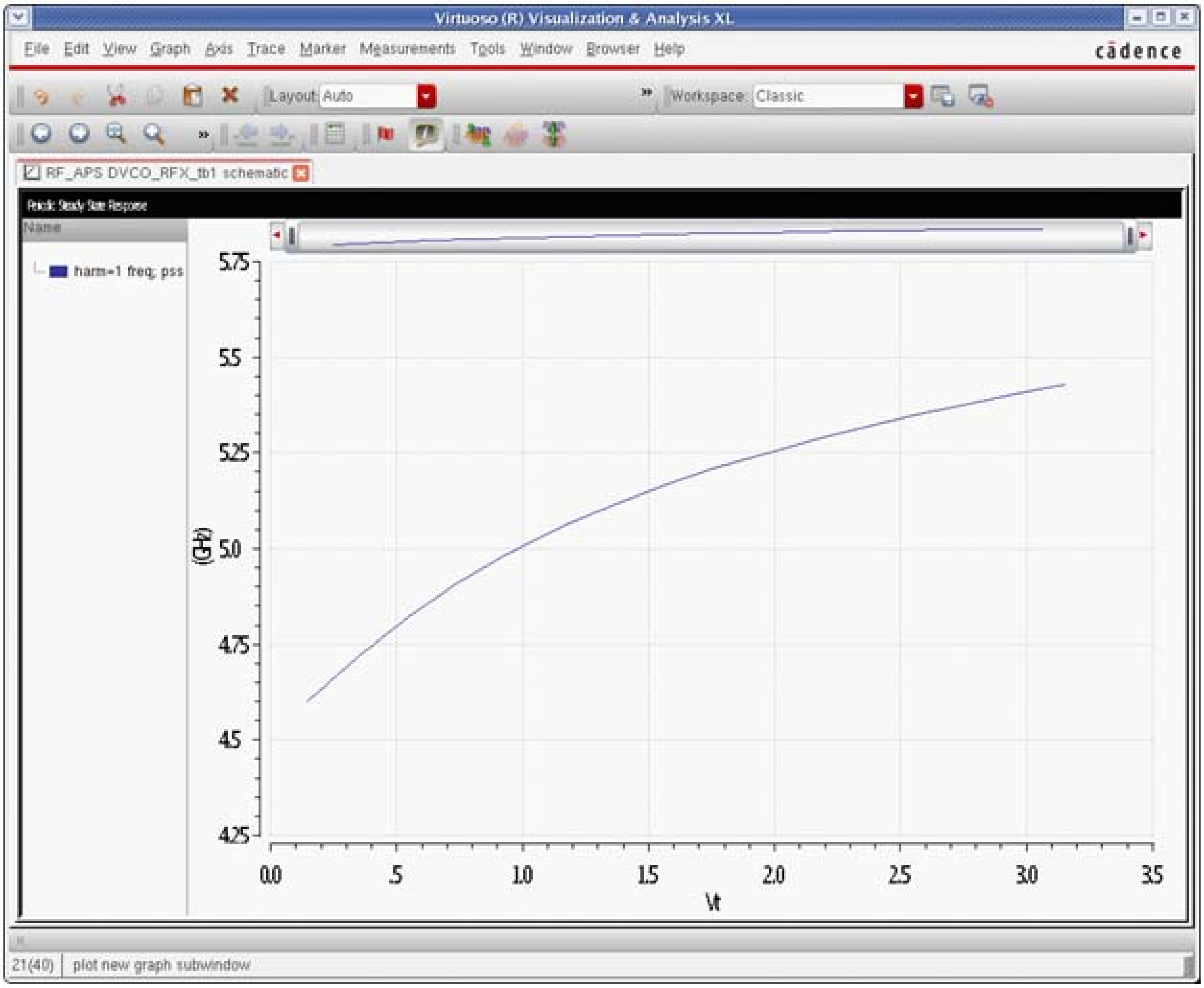


Plotting the Output Frequency Versus the Tuning Voltage

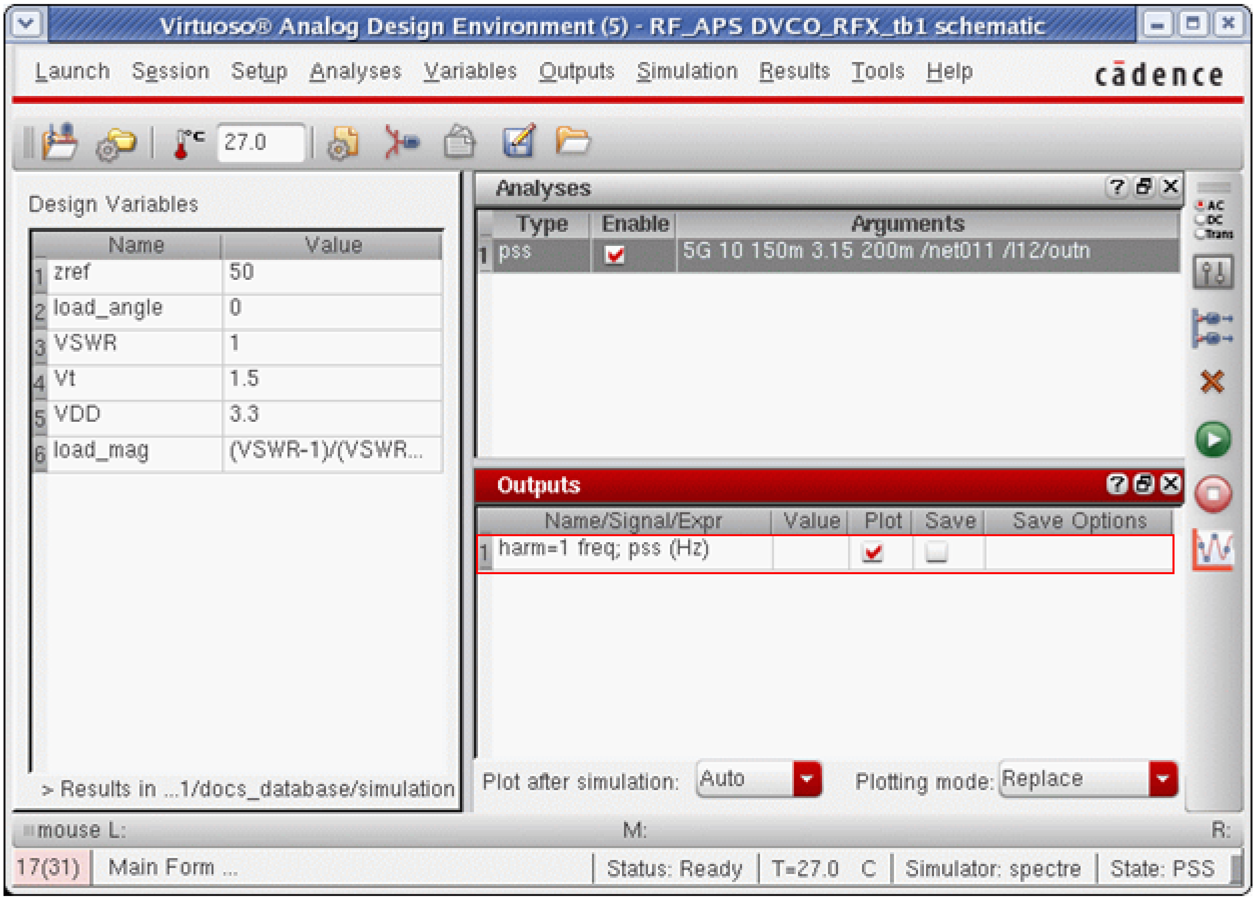
Now quit the waveform window, and select *Add To Outputs* in the *Direct Plot Form*, and plot the harmonic frequency again.



The tuning curve is plotted again, as shown below.



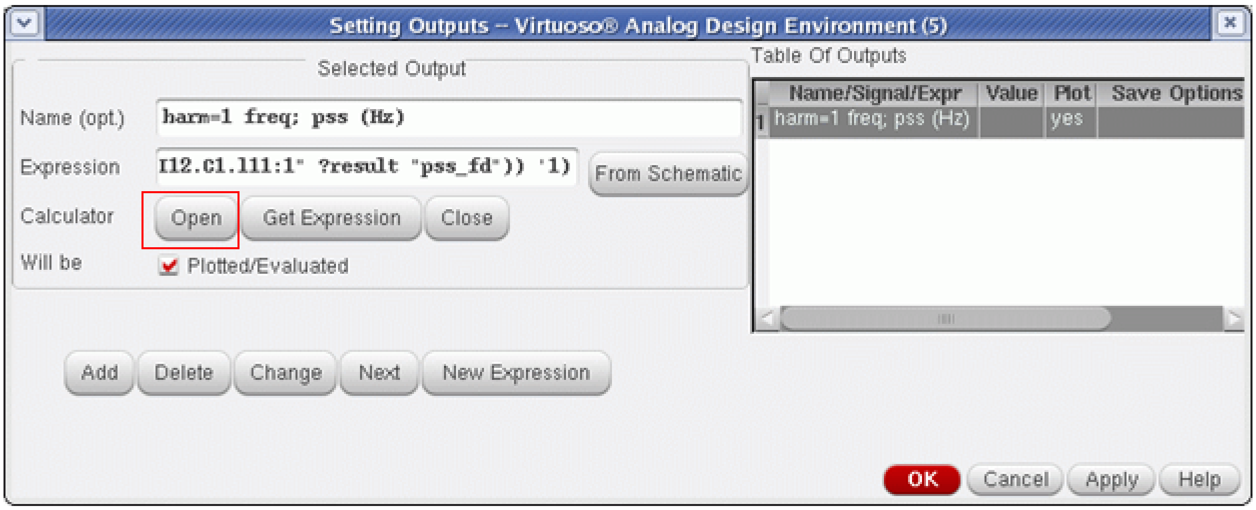
In the ADE, a new expression is added to the *Outputs* section.



Plotting the Modulation Sensitivity

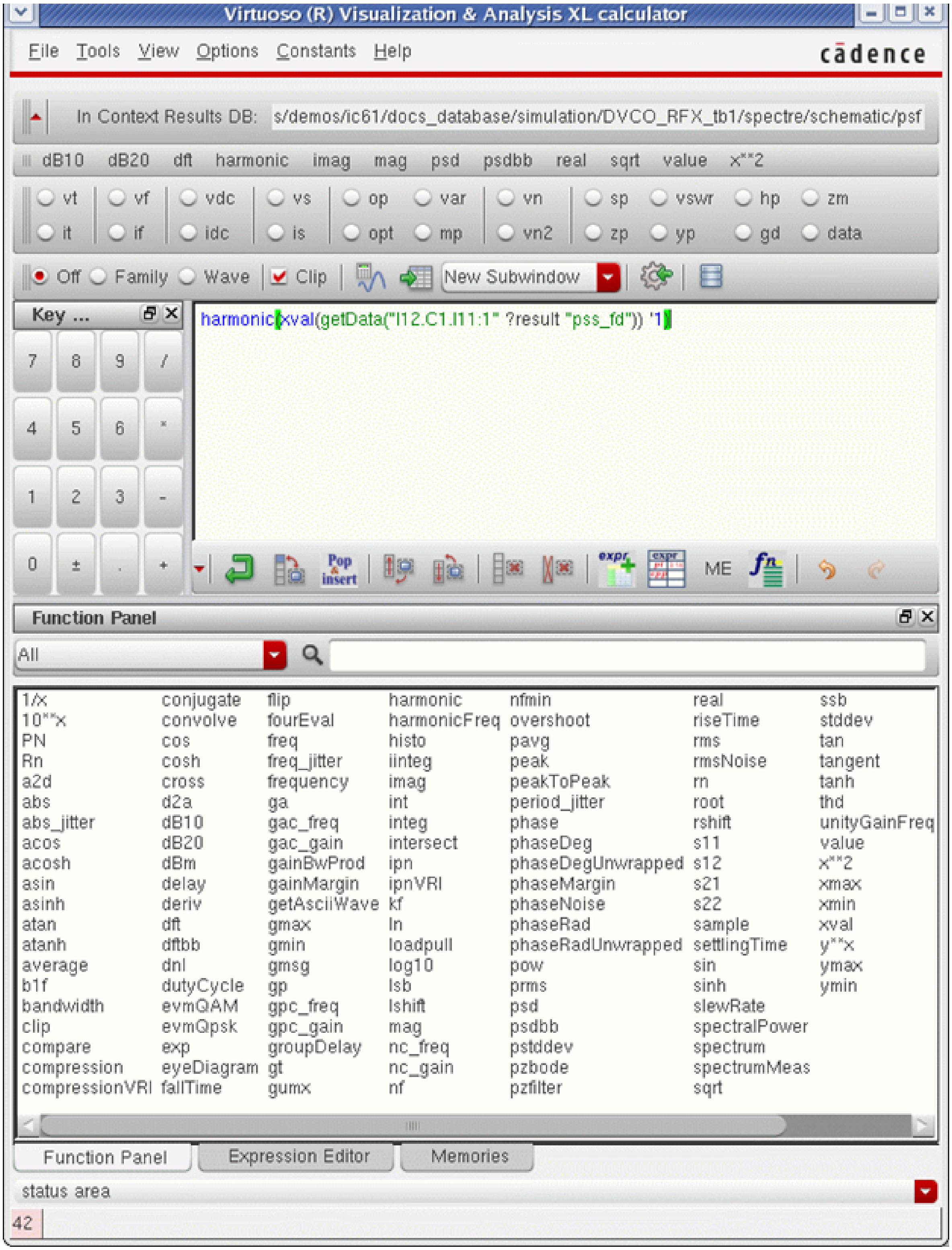
1. Double-click the expression in the *Outputs* section.

   The *Setting Outputs* window is displayed. The expression is different for every oscillator.

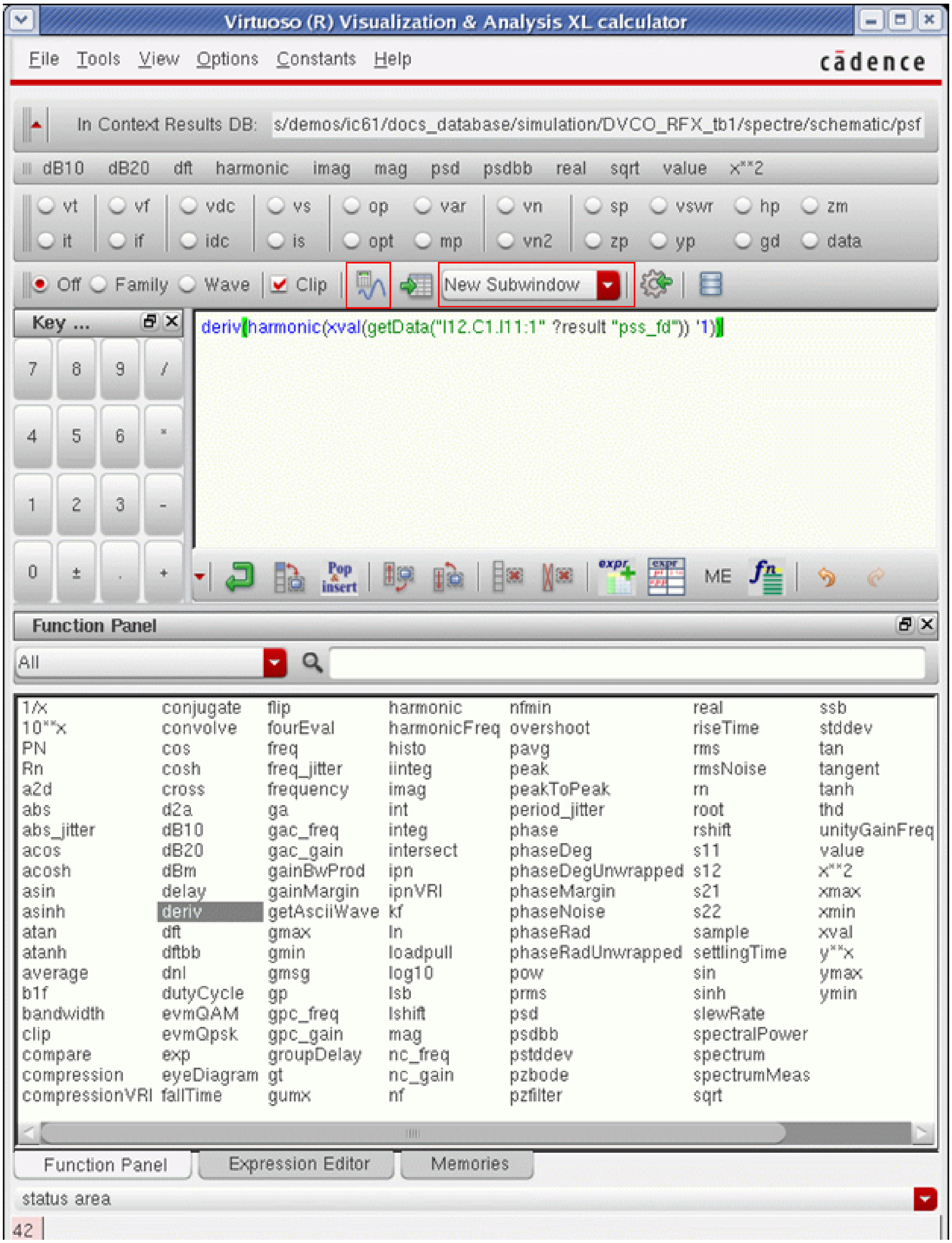


2. Click *Open*.

The calculator opens with the expression in it.

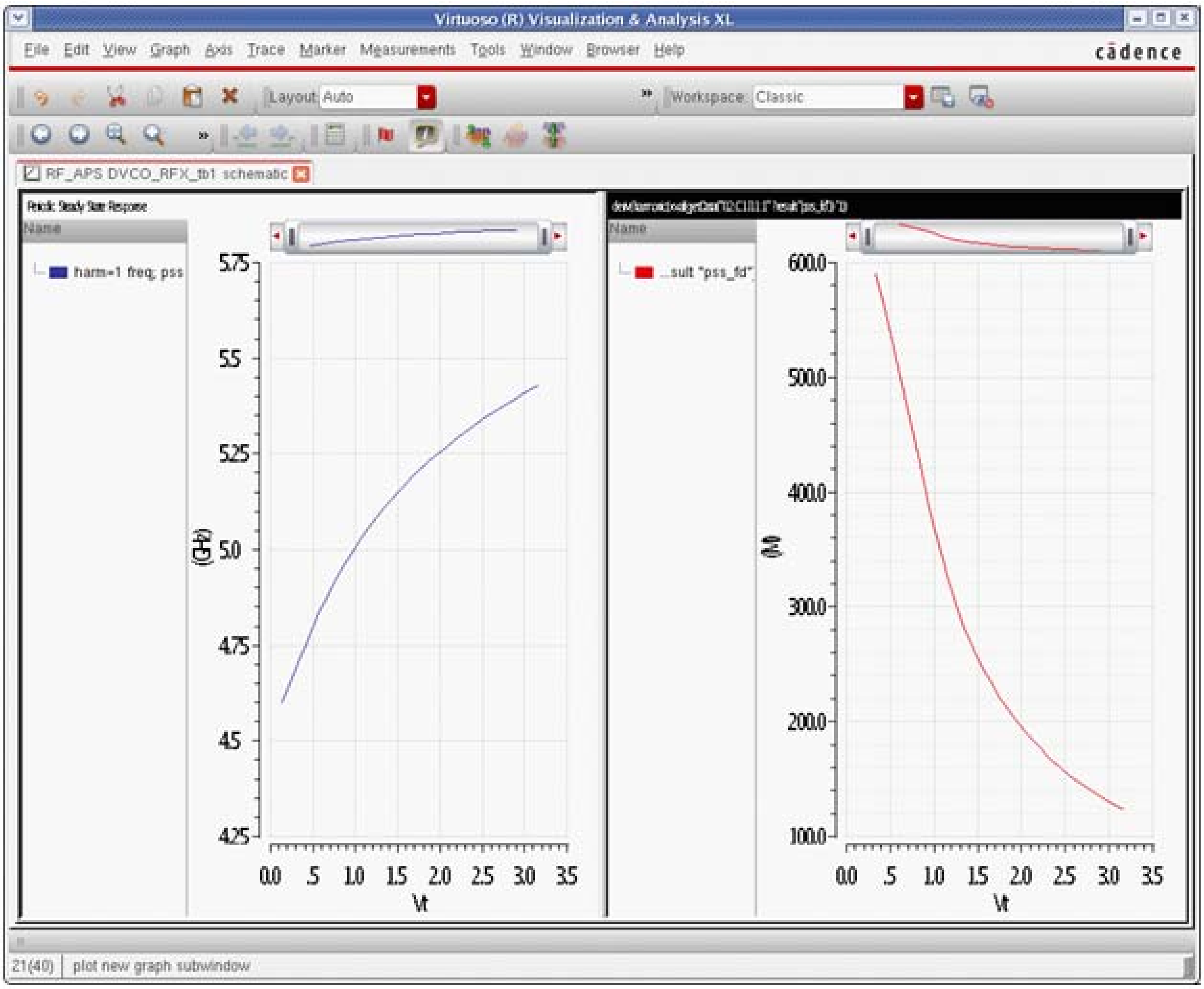


3. Select the derivative function. The expression changes.



4. Set the plotting mode to *New Subwindow*.
5. Click the *Plot* icon.

The modulation sensitivity in MHz/Volt is plotted in the subwindow on the right.

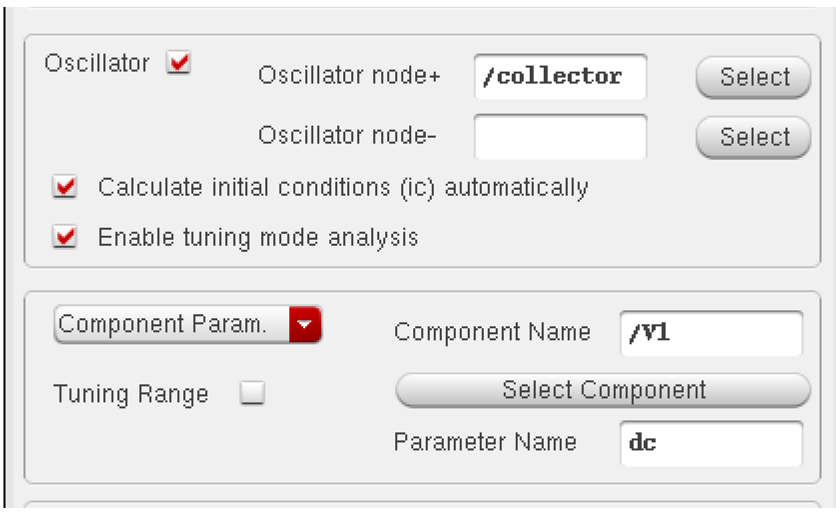


Oscillator Tuning Mode Analysis

The oscillator tuning mode analysis tunes the oscillator so that phase noise can be measured at the desired oscillation frequency. It is supported in both harmonic balance and shooting methods and is typically used in Monte Carlo simulations or sweeps to obtain the phase noise.

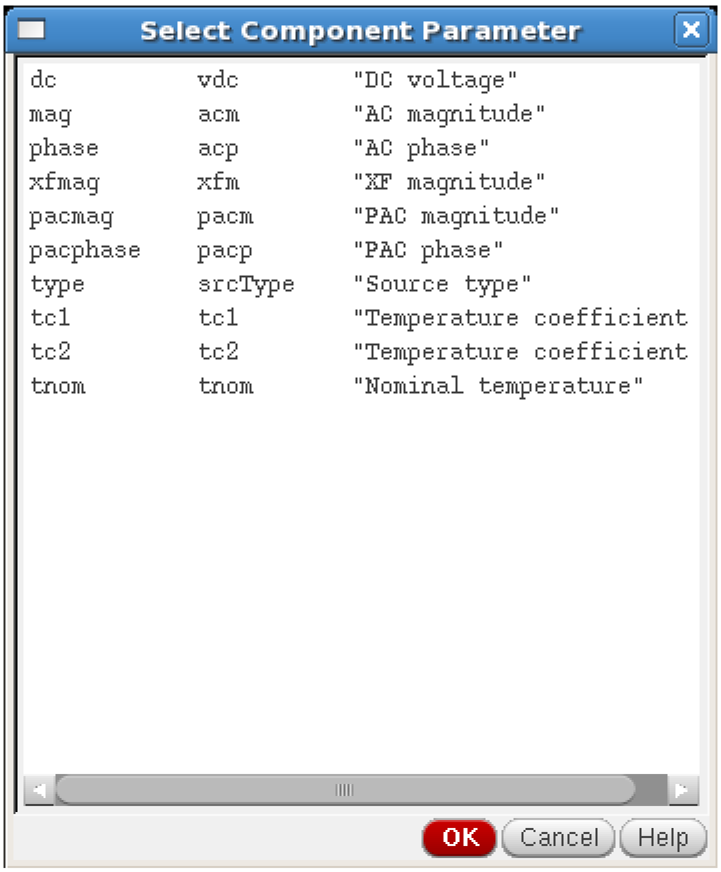
In oscillator tuning mode analysis, the oscillator frequency is treated as the target frequency, and the value of a circuit parameter is obtained to achieve the target frequency. The circuit parameter can be an arbitrary ADE variable or an instance parameter.

1. Open the pss *Choosing Analyses* form.
2. Set up the target frequency in the *Beat Frequency* field.
3. Select the *Oscillator* check box.
4. Specify the output node from the `Oscillator node+` field. Alternatively, click *Select* and choose the output node from the schematic. Here, `vc2` is selected.
5. Select the *Enable tuning mode analysis* check box.
6. Choose the parameter that needs to be tuned from the drop-down list. Here, *Component Param.* has been selected.



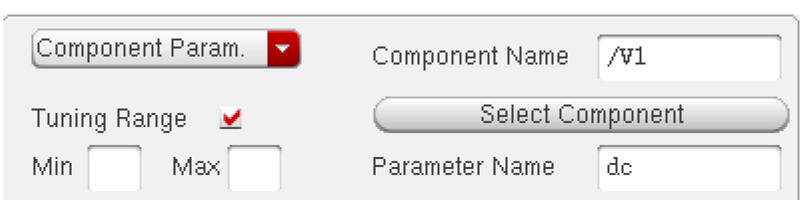
7. Specify the name of the parameter in the *Component Name* field. Alternatively, click *Select Component* and select the parameter from the schematic. When you select a

parameter from the schematic a pop-up window appears that contains a list of selectable parameters.



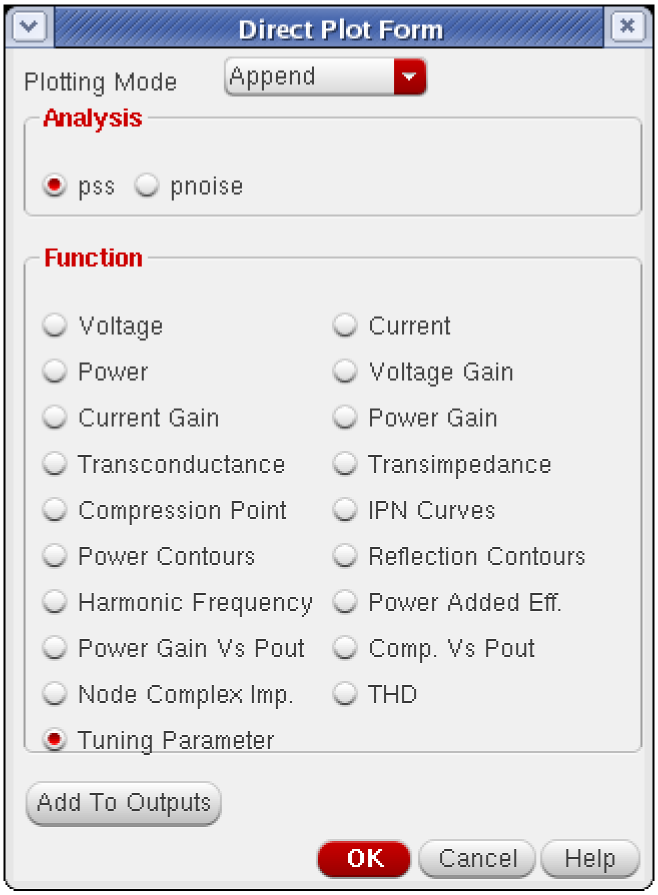
8. Select a parameter from the list. The *Parameter Name* field is automatically populated with the name of the parameter (*dc* in the figure above).

9. Select the *Tuning range* check box and specify the minimum and maximum ranges for the parameter, if required.



10. Run the simulation.

11. After the simulation completes, click *Results - Direct Plot - Main Form* to display the *Direct Plot Form*.

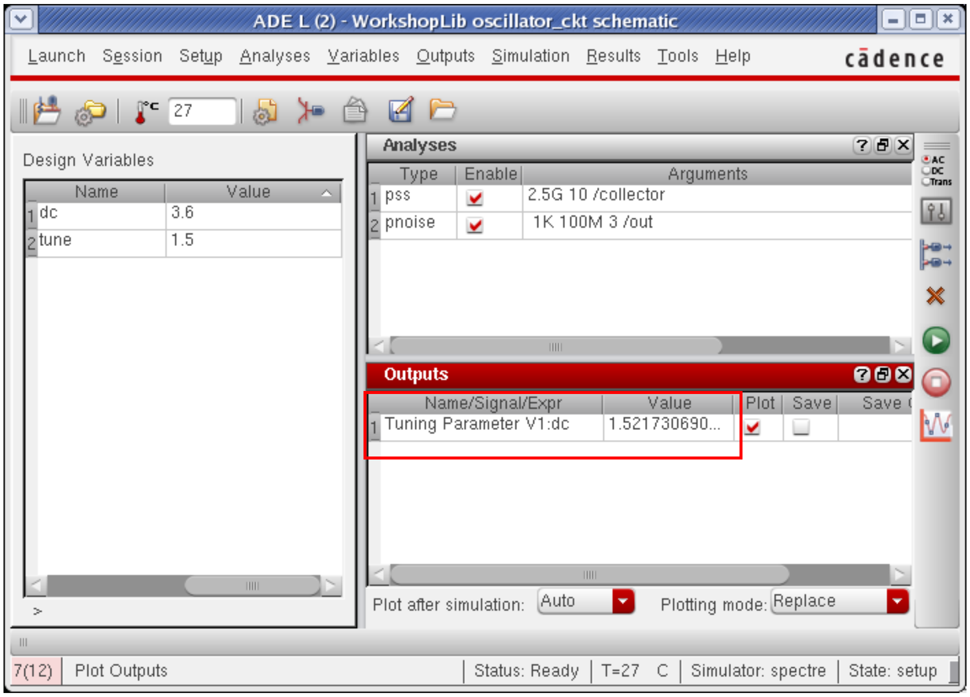


12. Select *Tuning Parameter* from the *Function* section.

13. Click *Add to Outputs*.

14. Click *OK*.

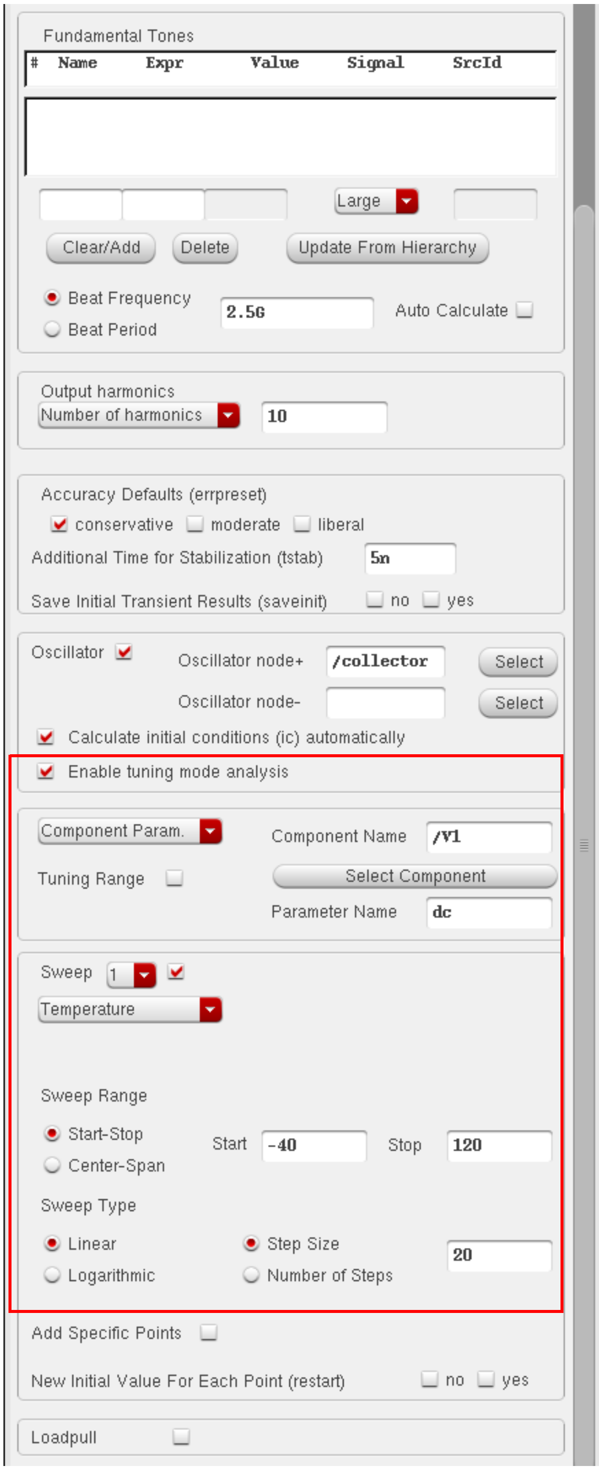
The value of the tuning parameter is added in the *Outputs* section of the ADE window, as shown below.



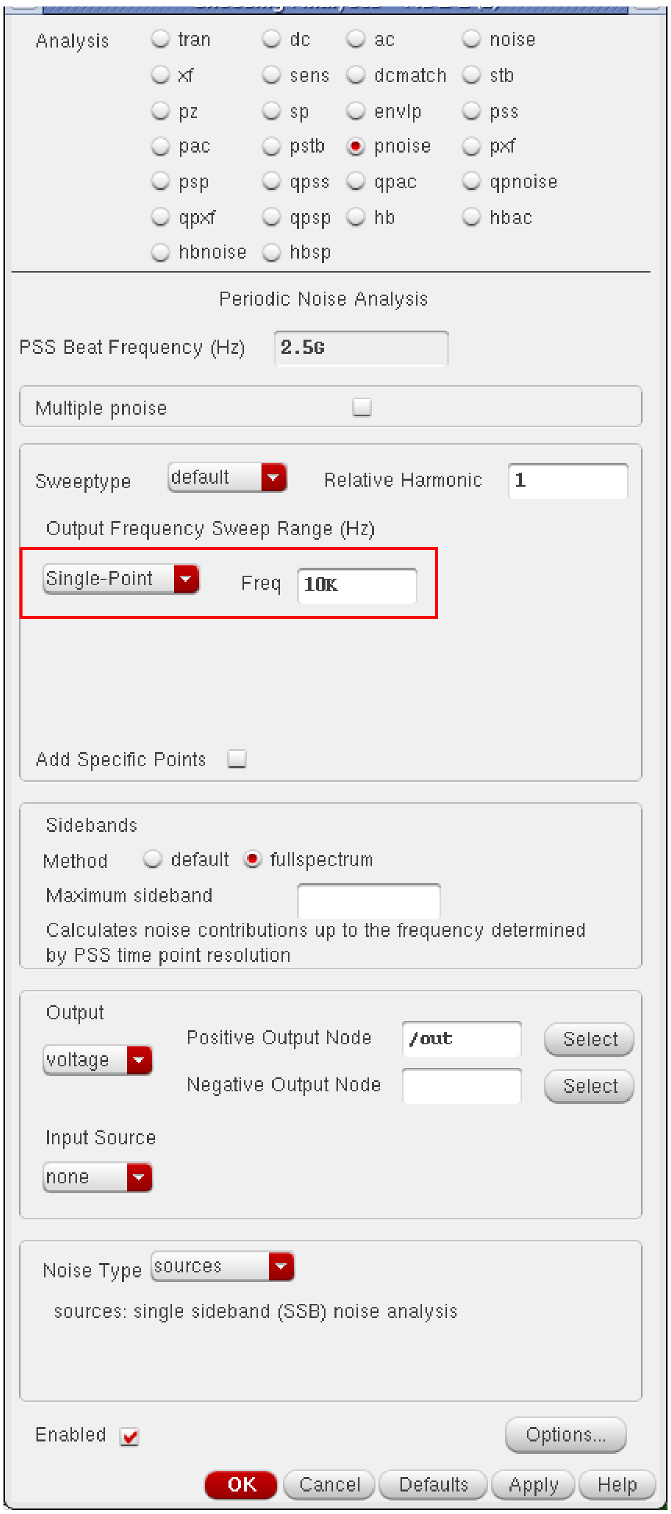
Oscillator Tuning Mode With Sweeps

Sweeps and Monte Carlo are also supported with tuning mode. Below is an example of a sweep of temperature with tuning mode and a phase noise measurement.

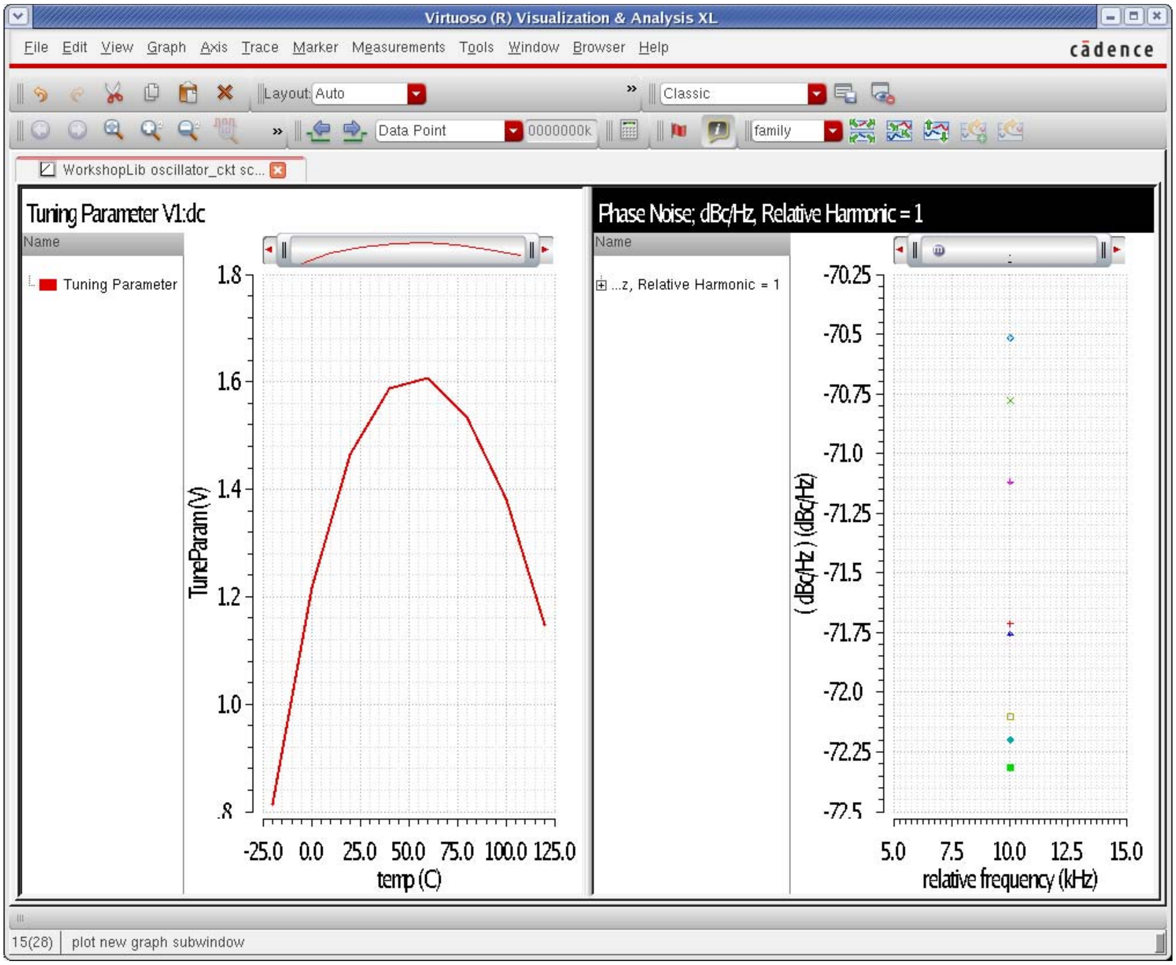
1. Enable sweeps in the pss *Choosing Analyses* form. In this example, temperature is swept from -20 t0 120C. Tuning mode is enabled as before.



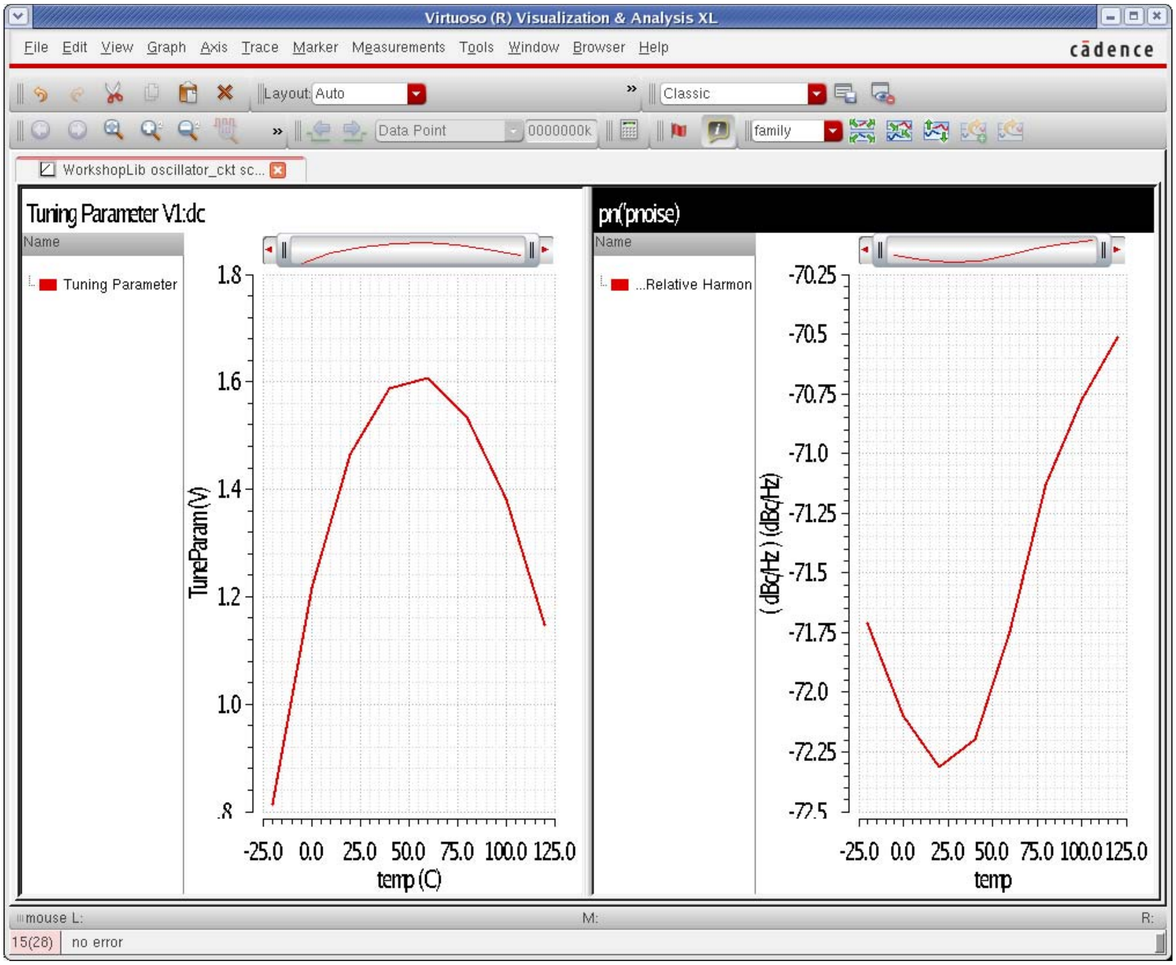
2. Because of the sweep, pnoise has been changed to a single frequency offset, which simplifies plotting of phase noise after the sweep.



3. Now run the simulation. Because the tuning parameter and phase noise have already been added to the ADE outputs section in the previous section, the tuning voltage and the phase noise are plotted after the simulation.



4. To make the phase noise plot more useful, select the X Axis, click the right mouse button, and select *Swap Sweep Var*. In the window that pops up, click *OK*. Then delete the original phase noise plot.



5. The trend in tuning voltage and phase noise is plotted.

Periodic AC Analysis (PAC)

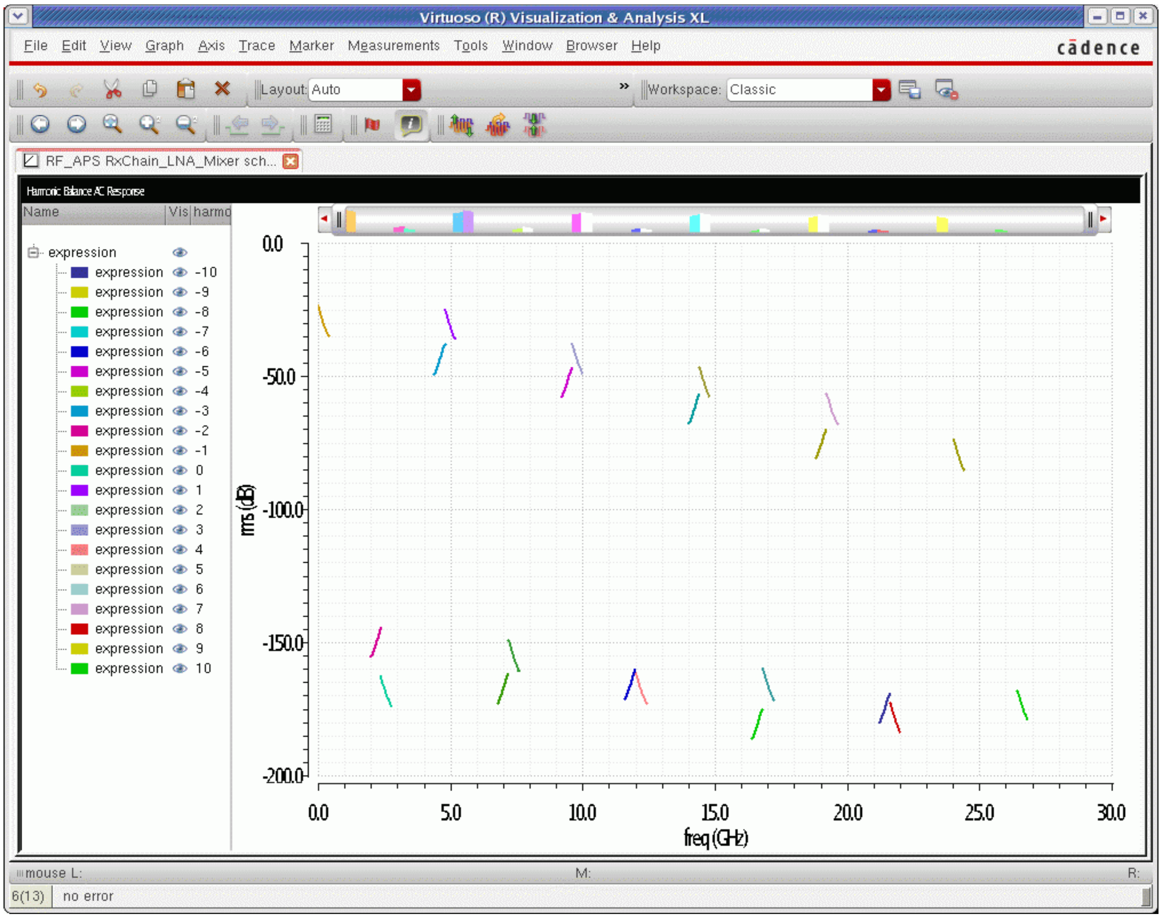
Small-signal Versus Large-Signal Analysis for Conversion Gain

One way to measure conversion gain is to apply the LO and the RF signals in harmonic balance (hb) or quasi-periodic steady-state (qpss) and calculate the full large-signal response at each RF input frequency, and then plot the output as a function of the sweep frequency. While this works, because all the large-signal effects and many harmonics are calculated, the required time for the simulation can be significant.

The alternative is to use a small-signal approach similar to the AC analysis. The AC analysis runs considerably faster than running the transient at multiple frequencies because it ignores the large-signal effects. In circuits with frequency translation, the AC analysis is unusable because the calculation is based on the DC bias point. In hb Periodic AC (PAC) analysis, instead of using the DC bias point, the harmonic balance periodic steady-state (pss) solution with just the LO is used as the basis of the calculation. The large-signal analysis only calculates the harmonics of a single input, and no mixing products, and then hb pac is run after the hb pss analysis and calculates the outputs that are produced by mixing the pac input with the harmonics that are in the pss analysis. The overall time savings can be very significant.

Example: Conversion Gain (Down Conversion)

Consider the double-balanced diode mixer shown below.

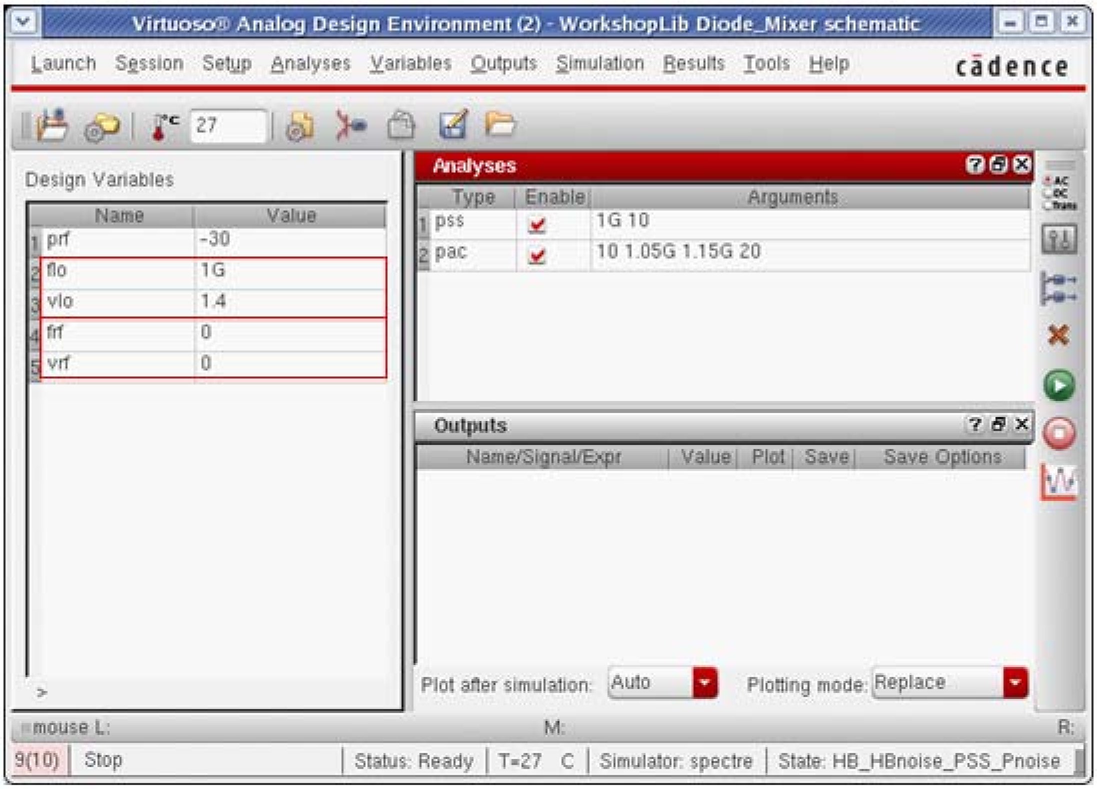


The RF input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. Also note that the pac magnitude term is set to 1 volt. This makes it easy for a conversion gain measurement because the input signal is 0dB when the *dB20* modifier is used in the *Direct Plot Form* for the pac result.

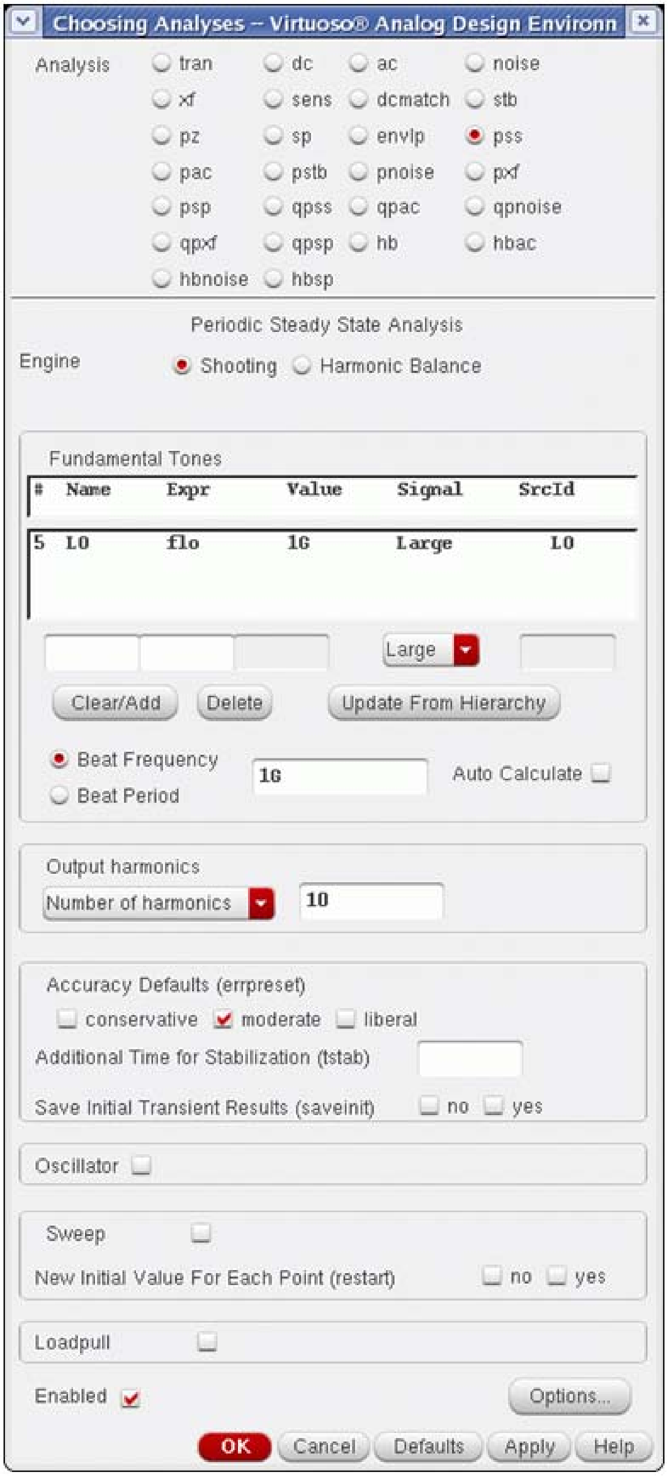
The property list is shown below.

The easiest way to get the variables list into the ADE is to select *Variables - Copy From Cellview*. To set the values, select the value field to the right of the variable, type the desired value, and press *Enter*.

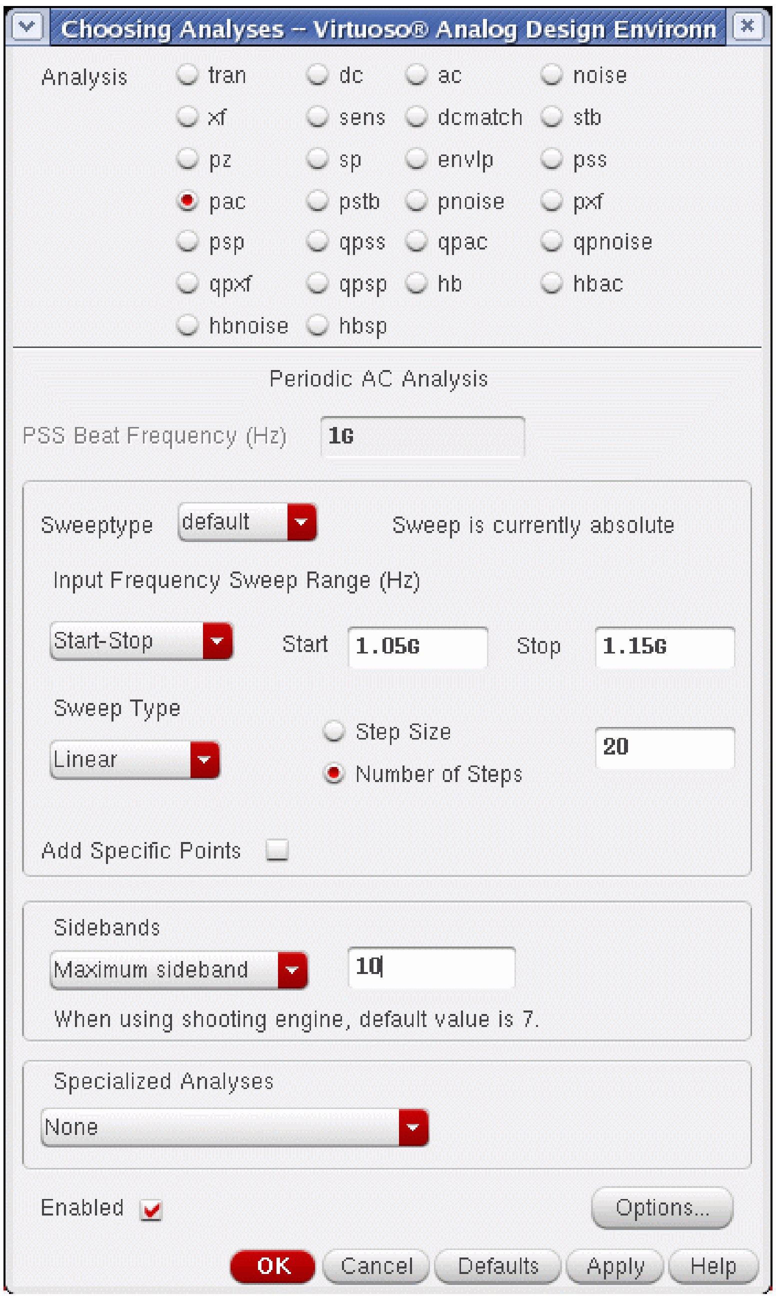
The LO frequency is 1GHz and its amplitude is 1.4 volts peak. This is +13 dBm. (+13 dBm is a common drive level for diode mixers.) The RF frequency is 1.1GHz, but it is disabled because the amplitude is set to 0 (zero). Setting either the frequency or the amplitude to zero disables the production of that waveform from the port. This provides an easy way to enable or disable signals without needing to change the schematic.



The pss analysis needs to be set up first. Note that both shooting and harmonic balance are available in pss for the engine. The example below shows a single input at 1GHz with 10 harmonics using the shooting engine. For more information, see Overview of Periodic Steady-State (pss) Analysis on page 547. Diode mixers have rapid changes in the diode currents. Because of this rapid change, shooting is likely to be faster and more accurate than harmonic balance.

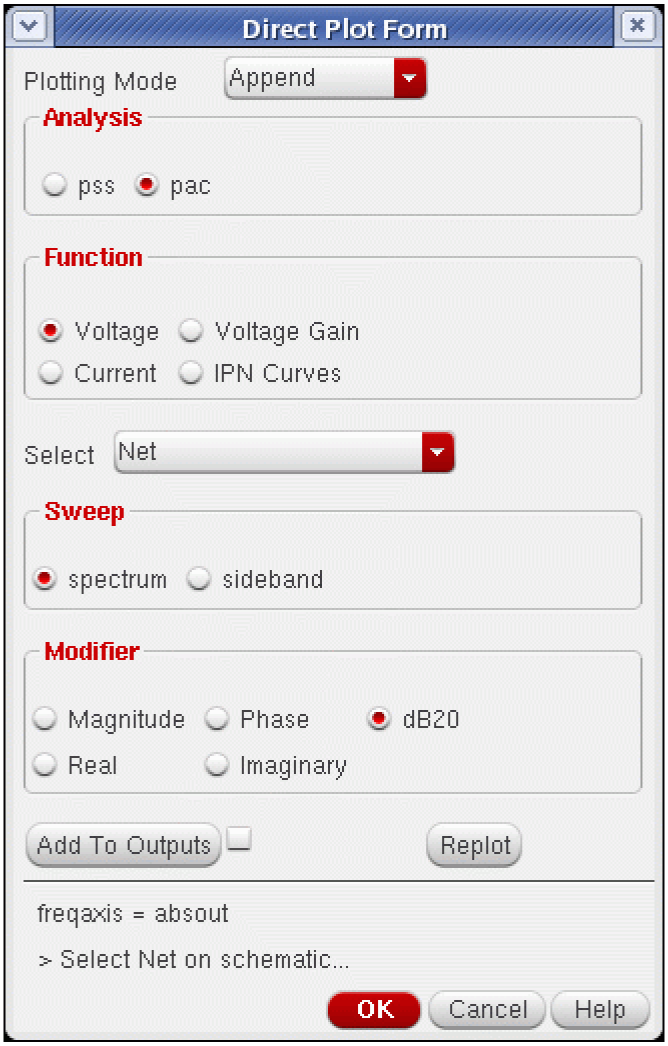


Next, the PAC analysis needs to be set up. Like AC, the frequency range specified in the form is the input frequency range. When shooting is used for the pss engine, *Maximum sideband* should be set somewhere less than four times the number of harmonics set in the pss analysis. Setting 10 sidebands calculates the mixing products from the input frequency in PAC with the harmonics through the tenth harmonic in the pss analysis. The actual number chosen depends on the mixing products that are important in your application. When harmonic balance is chosen for the engine, leave the *Maximum sideband* field blank.



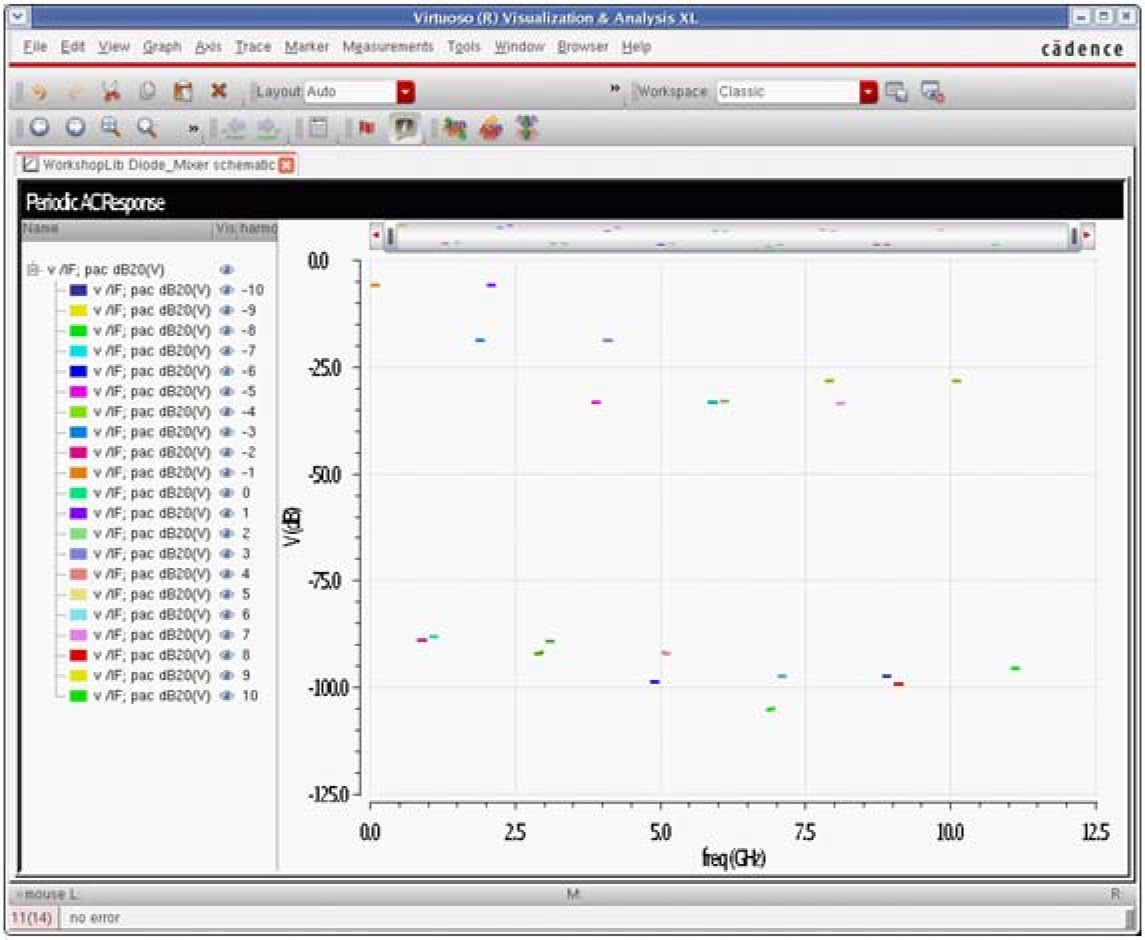
Once the analyses are set up, the simulation can be run by selecting *Simulation - Netlist and Run* in the ADE. When the simulation finishes, select *Results - Direct Plot - Main Form*.

The example below shows a typical *Direct Plot Form* setup to measure the conversion gain.

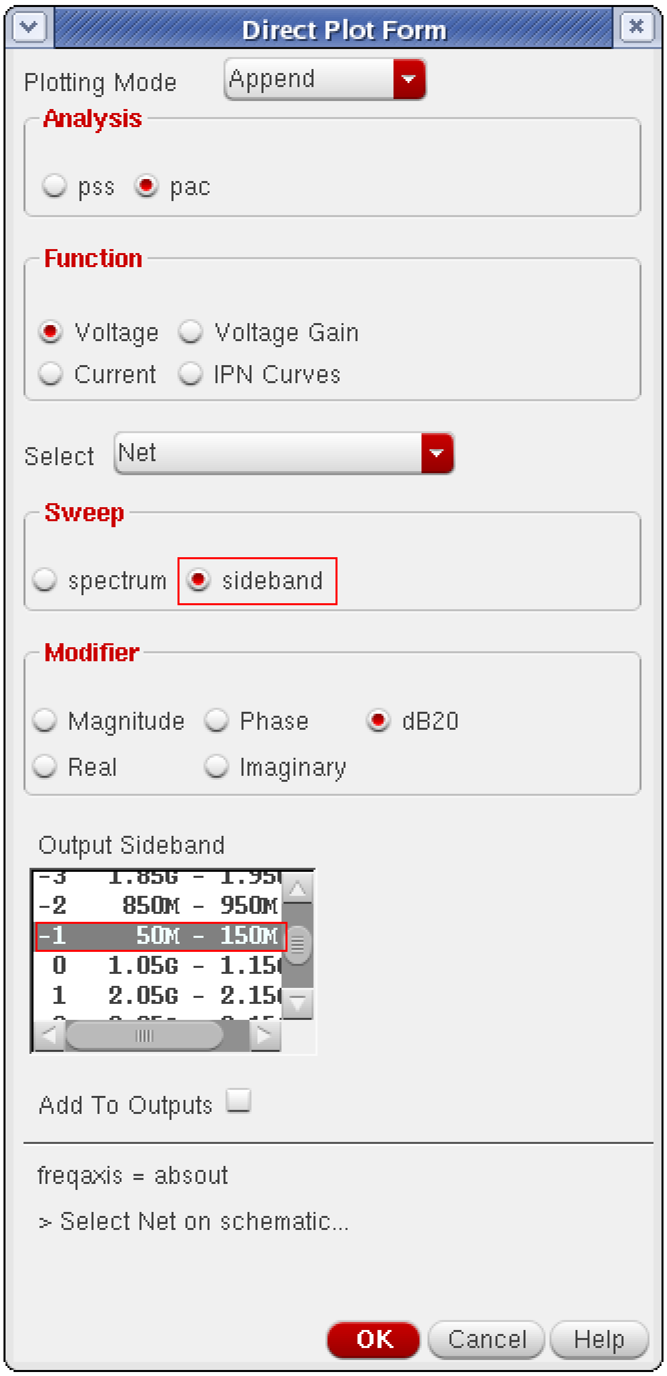


When the net in the schematic is selected, the waveform is displayed.

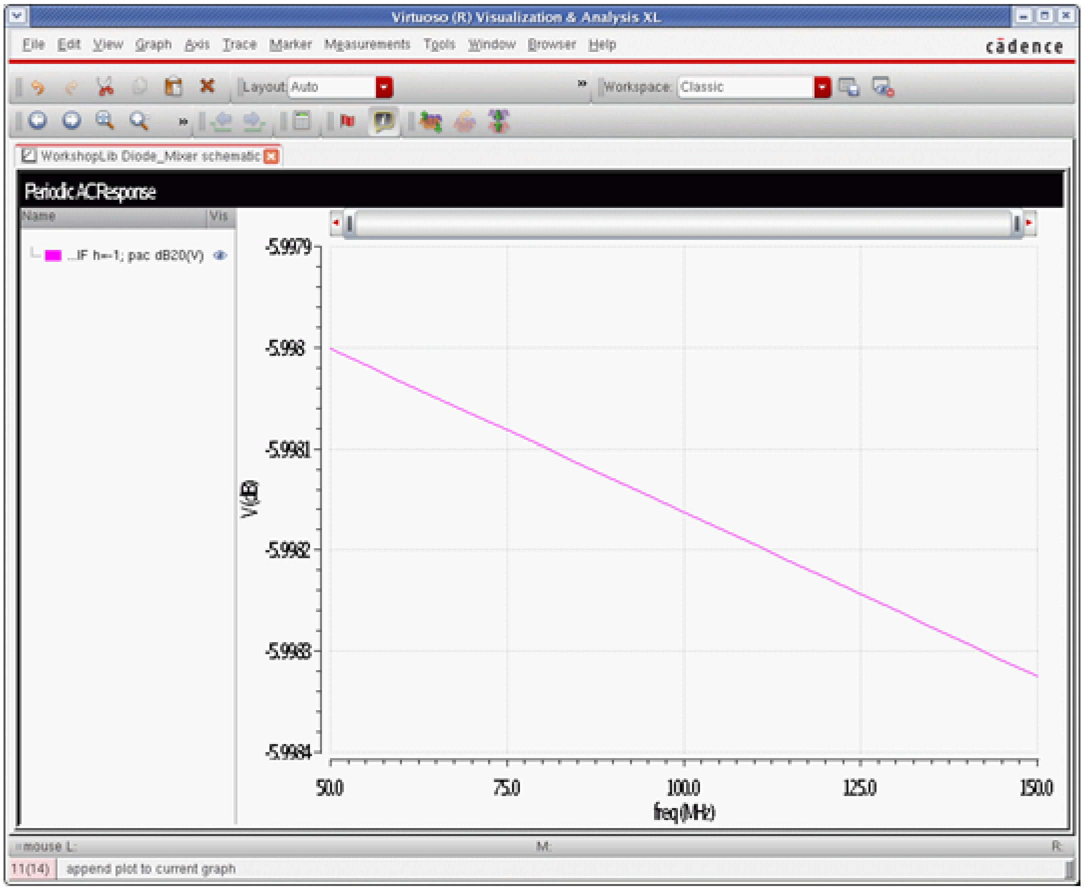
Note that all the short line segments are the mixing products that are produced when mixed with the harmonics in the periodic steady-state (pss) large-signal analysis.



In the down-conversion case, the frequencies that are produced for the down-converted output product are desired. In the *Direct Plot Form*, select *sideband* instead of *spectrum*, and then select the desired frequency range from the list at the bottom of the form. Only that mixing product will be displayed, as shown below.



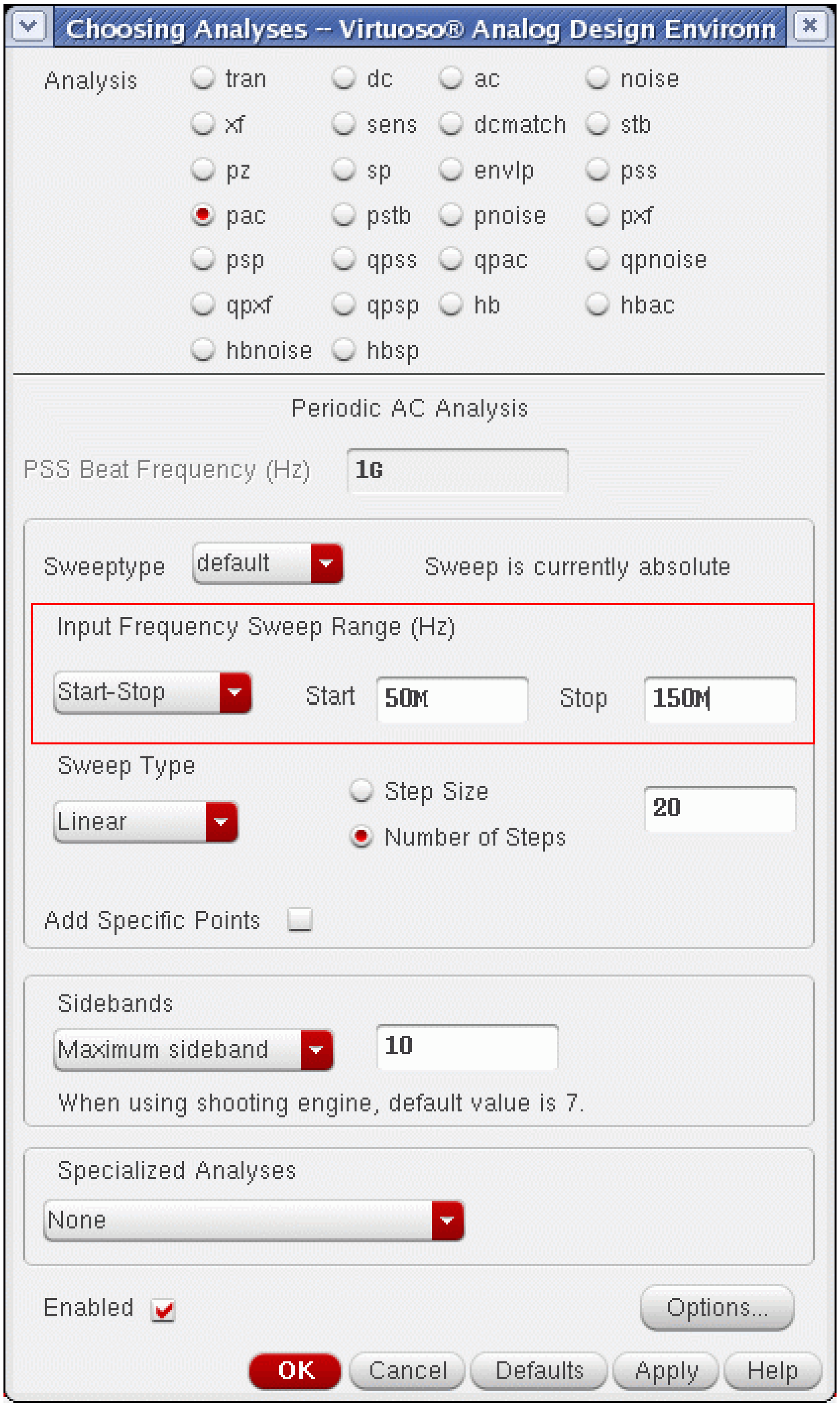
The conversion loss is about 6dB.



Example: Conversion Gain (Up Conversion)

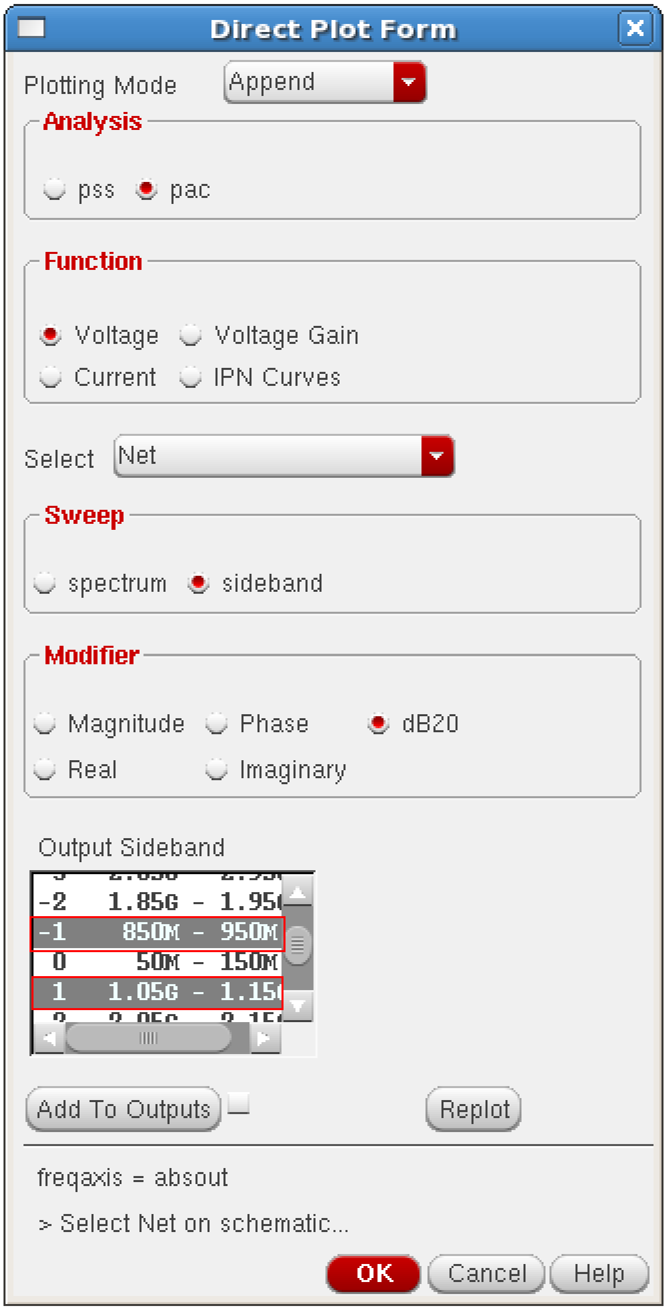
The process is similar for an up-conversion mixer. In this case, the input frequency is low, as shown below. The LO remains at 1GHz, as it was in the previous section.

The RF outputs are 50MHz through 150MHz above and below the 1 GHz LO.

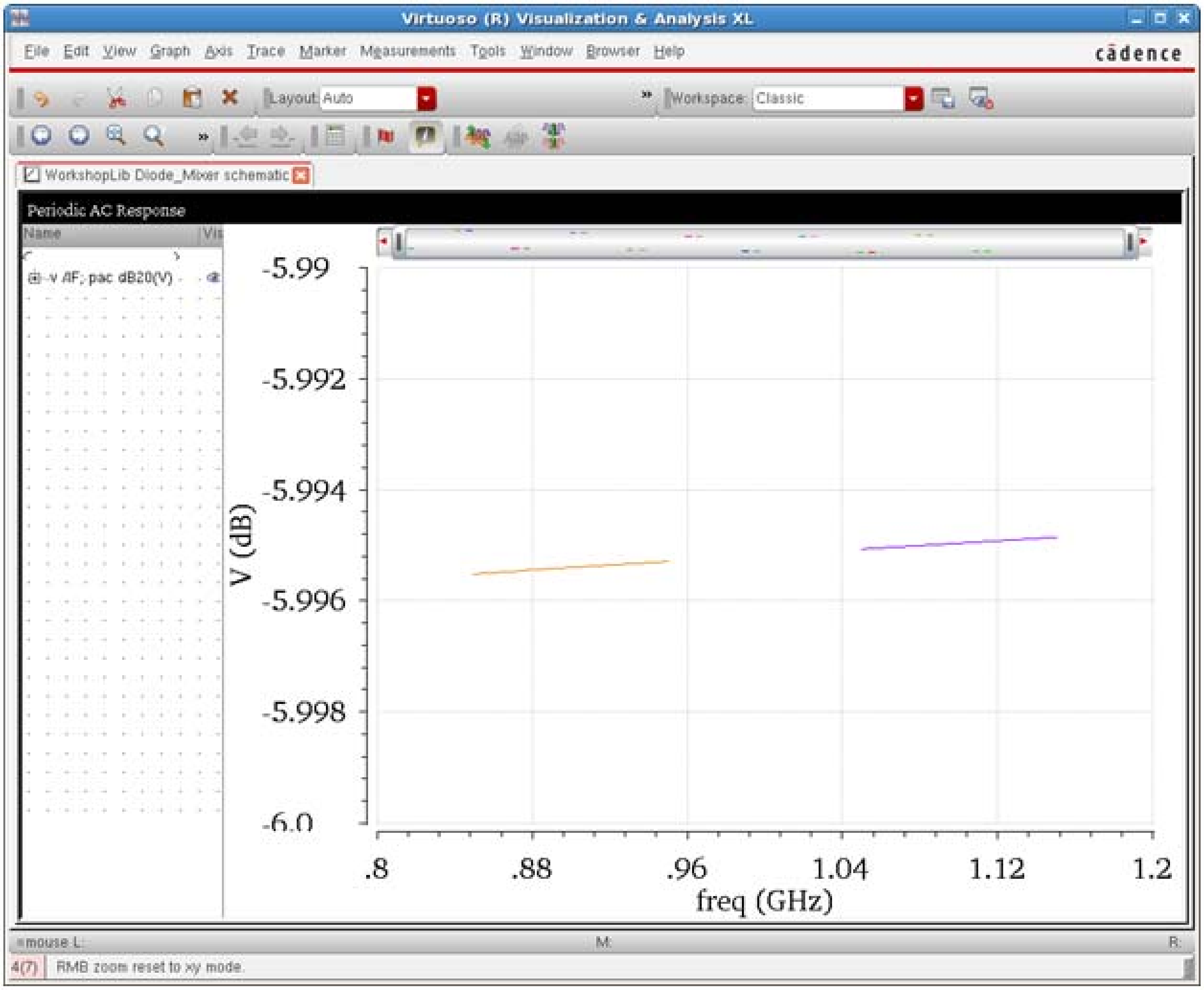


In the *Direct Plot Form*, both the outputs above and below the LO frequency might be desired.

Select the first frequency, and click the second while holding the *Ctrl* key. Now select the output net in the circuit and the results will plot.



Note that the conversion loss is almost identical for both mixing products.



PAC General Principles

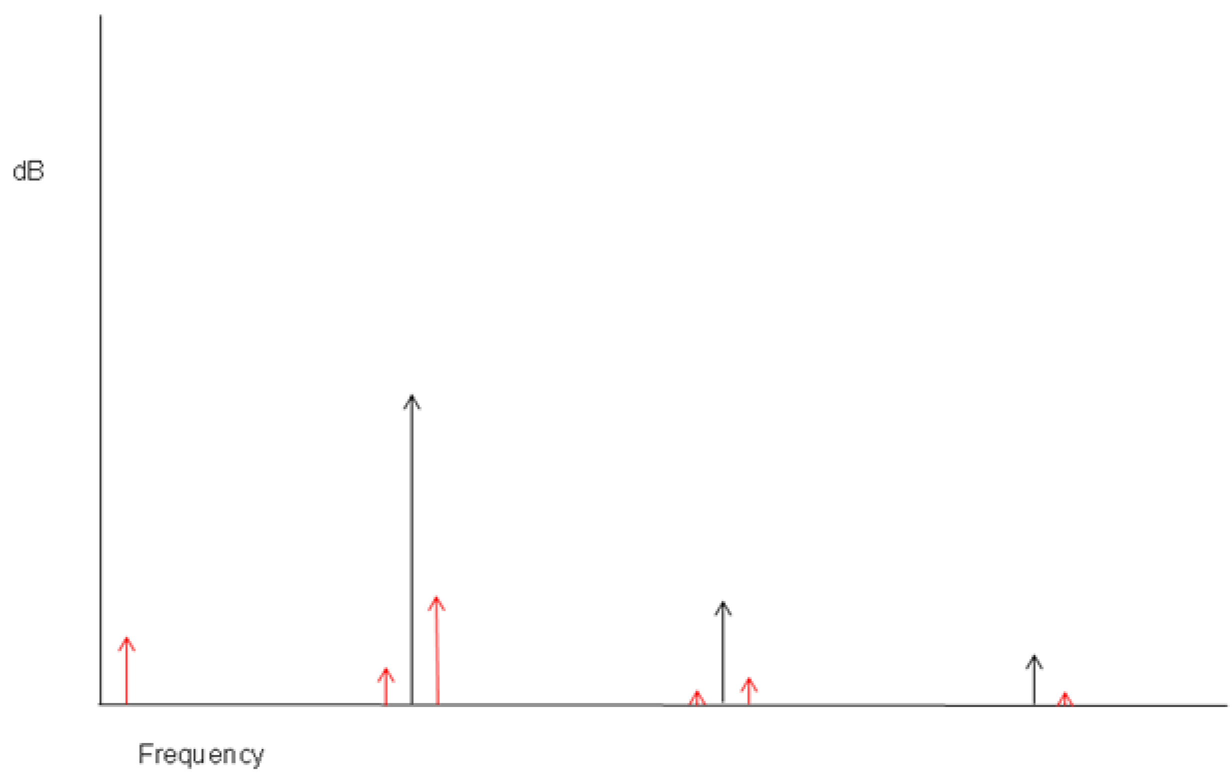
The pss solution with LO calculates the amplitude of the harmonics and characterizes the nonlinearity of the system. Therefore, in pac, the output mixing products of the input signal mixing with the LO harmonics can be calculated. Like ac analysis, pac allows the input to be swept in frequency to calculate a frequency response of the output mixing products. Also, like ac, when pac is used for conversion gain measurements, the conversion gain is the same regardless of the input amplitude. The small-signal conversion gain is calculated and is a good representation of the gain until compression is approached or exceeded in the real circuit. If a measurement of compression is required, use a large-signal analysis like pss, hb, or qpss.

PAC used with Harmonic Balance PSS

When harmonic balance is used for the pss analysis, the harmonics of the circuit are calculated directly by the harmonic balance engine.

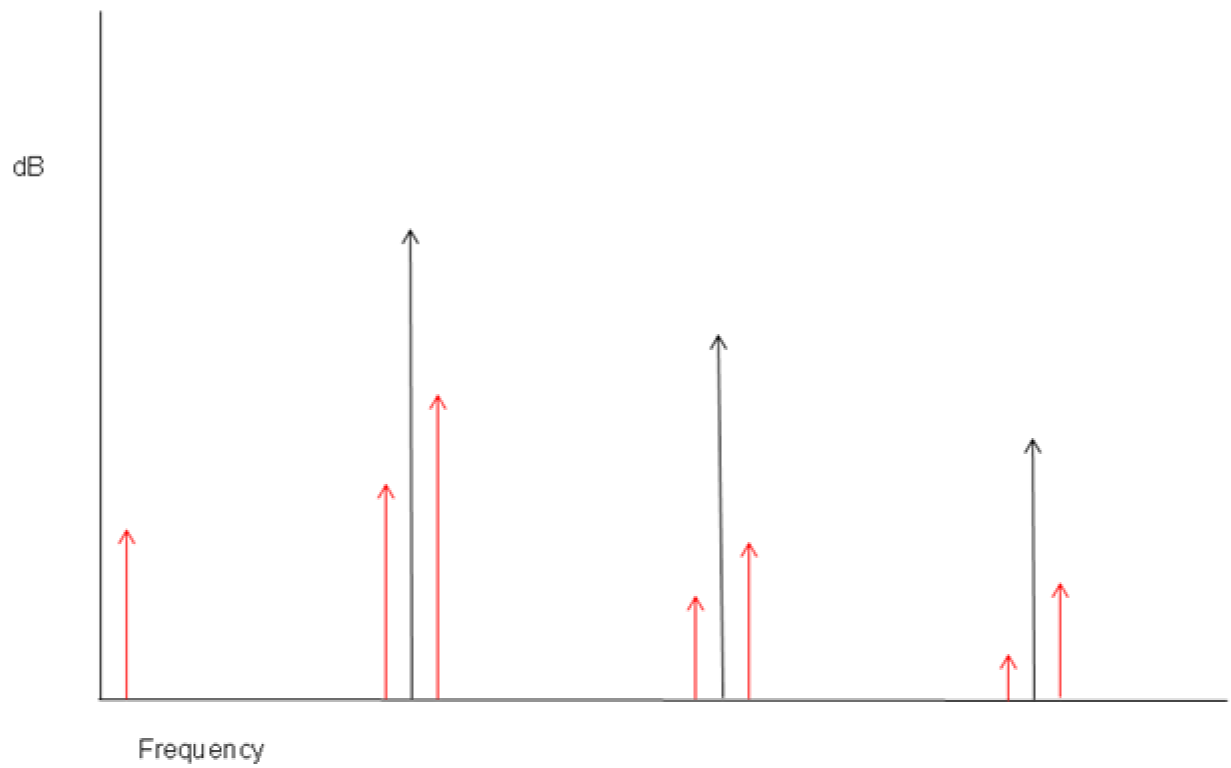
Think of a perfectly linear circuit with a single input applied. Because the circuit is perfectly linear, in the pss analysis when harmonic balance is used, the frequency domain solution only has the harmonic at the input frequency at every net in the circuit. No other harmonics are created because the system is perfectly linear. Because the circuit is linear, if a second input is applied, no frequency translation can occur.

Now imagine a real circuit with a single small input amplitude signal at 1GHz. In the frequency domain with harmonic balance selected in pss, the first harmonic is present along with the higher harmonics. Because the input signal is small, the signals at the harmonics of 1GHz are very small. If a second input is applied at 1.1 GHz, that signal is weakly converted when it is mixed with the harmonics of the circuit. Certainly, 1.1GHz would appear at the output with no mixing. 100MHz and 2.1GHz are produced by mixing with the 1GHz fundamental frequency. 900MHz and 3.1GHz are produced when 1.1GHz mixes with the second harmonic of the 1GHz input. 1.1GHz mixes with the other harmonics as well, producing a wide range of output frequencies from a single input frequency. The harmonic output might look like the figure below. Only the first three harmonics are shown. The black harmonics are the pss harmonics, and the red mixing products are pac outputs, which are called sidebands.



Note that there is an output mixing product that is created at 100MHz. Because the real circuit is nonlinear, harmonics of 100MHz are created in the circuit. In pac, the assumption is that all the signals are small, and with this assumption, the harmonics of 100MHz do not need to be calculated. This assumption also applies to all the other outputs in pac. Therefore, the PAC output is a partial output from the circuit with only the main mixing products displayed. Since in most cases, these are the desired outputs, conversion gain can still be obtained by using pac. In other words, pac calculates the main mixing products that are created by mixing with the pss harmonics, but the harmonics of the pac outputs are not calculated.

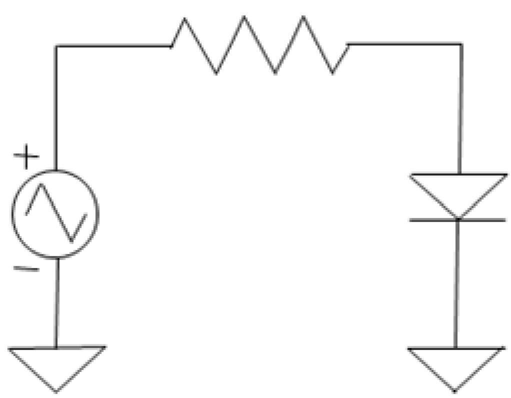
Now imagine that the 1GHz signal is larger. If the input goes up 10dB, the first harmonic goes up 10dB, In the frequency domain, if harmonic balance is selected in pss, the second harmonic would come up 20dB, the third harmonic would come up 30dB, and so on. If a second input at 1.1GHz is applied to the circuit, this mixes with the fundamental and the harmonics of 1GHz and more strongly mixes up-and-down in frequency. This is illustrated below. The black harmonics are the pss harmonics, and the red mixing products are pac outputs, which are called sidebands.



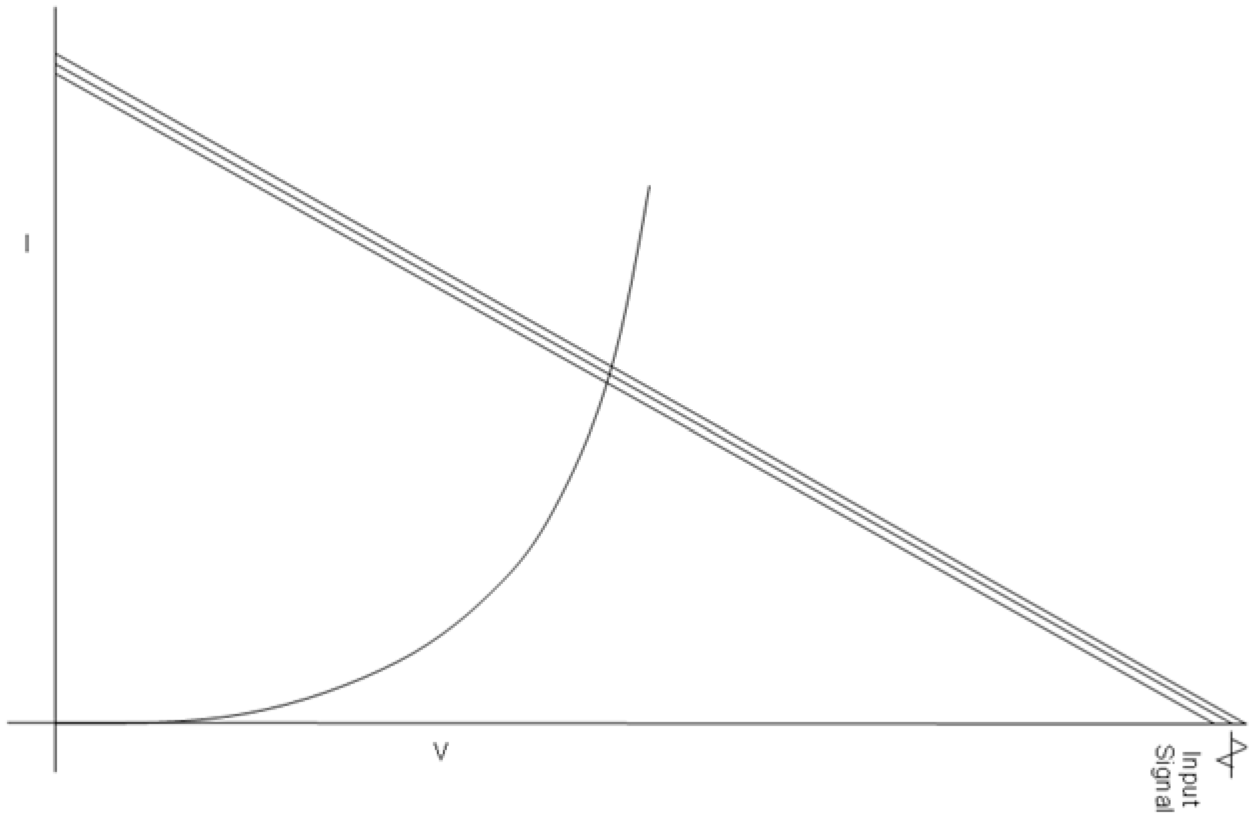
PAC used with Shooting PSS

Shooting is a variation of the transient analysis where the individual timepoints are calculated, and the entire waveform is iterated to calculate the steady-state waveform. The transient algorithm works by using all the equations at all the nodes in the circuit where the assumption is made that the sum of the currents is zero at every node and at every timepoint. Ohm's law is observed, and the equations look like `I=V*G` where `G` is the conductance. In general, the matrix form of the equations is constructed, so the system of simultaneous equations becomes `[G]*[V]=[I]` where `[G]` is a square matrix, and `[V]` and `[I]` are columns. To solve for [`V]`, the `[G]` matrix is inverted and this matrix multiplies the `[I]` matrix to solve for all the voltages on all the nodes. At the end of the pss analysis, all the solution matrices are saved as data to be passed to the small-signal analyses. The result is that the conductance is calculated and saved for every nonlinear component in the circuit at every timepoint.

For the following circuit:



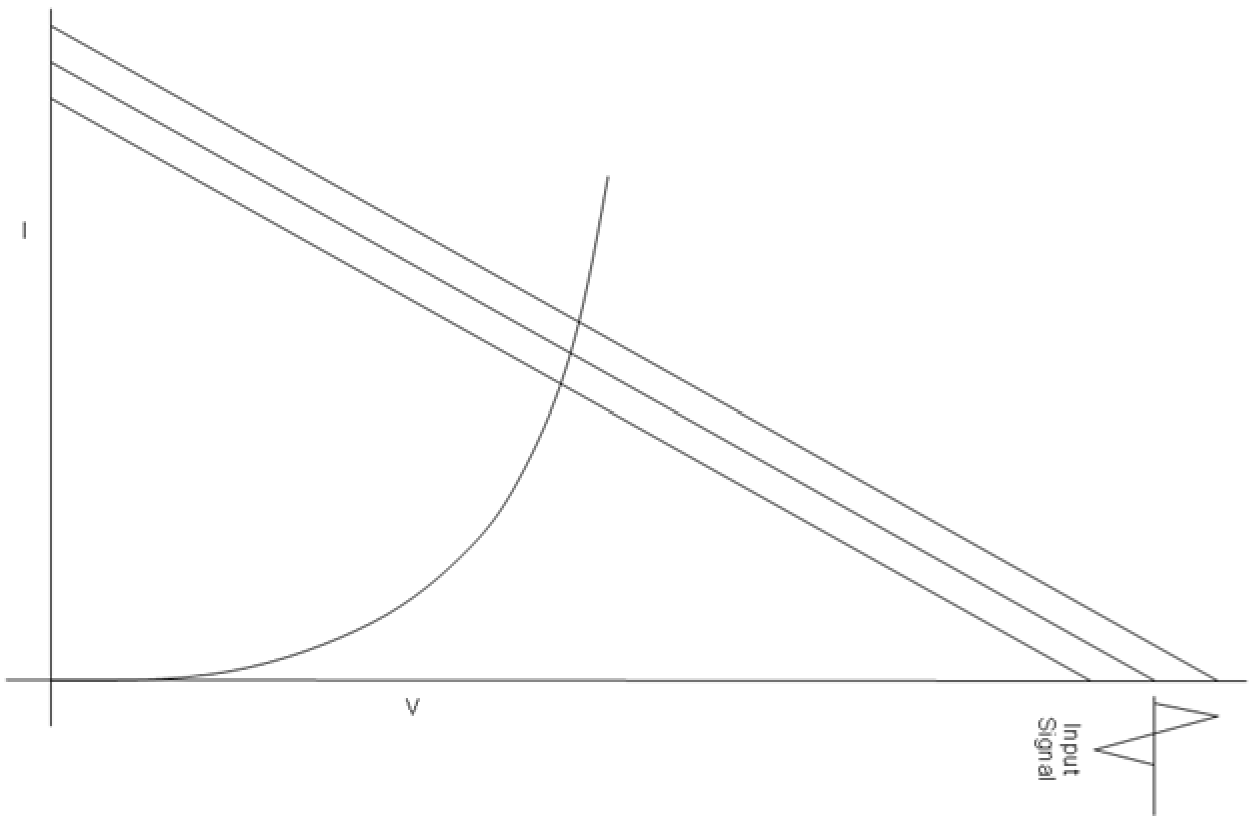
Below is the I-versus-V curve with 3 different values for the input voltage. The center, positive peak of the input, and the negative peak of the input are shown. The input amplitude is small.



In the above figure, the amplitude of the 1 GHz signal causes small variations in the slope of the tangent of the nonlinear curve as the input signal is varied in amplitude. The slope is the conductance. Because the circuit has a nonlinear device, the conductance of the device varies slightly from timepoint to timepoint in the pss analysis. Because the input signal at 1.0GHz is small, the variation in conductance from timepoint to timepoint is small. The slope changes just a little bit as the input signal is moved over its range.

Now imagine that the 1GHz signal is 10dB larger. A 10dB change is a linear factor of about 3.16. The larger amplitude of the input causes a larger portion of the nonlinear device curves to be traversed, and thus the variation in conductance becomes larger. If a second input at

1.1GHz were applied to the circuit, this mixes with the fundamental and the harmonics of 1GHz and more strongly mixes up and down in frequency. This is illustrated below.



Therefore, the variation in the conductance of the devices from timepoint to timepoint when shooting is selected in pss, or the harmonic content if harmonic balance is selected in pss contains information about how strongly a circuit translates frequencies, and this is used by the pac small-signal analysis to calculate the small-signal conversion gain.

Overview of Simulation Capabilities

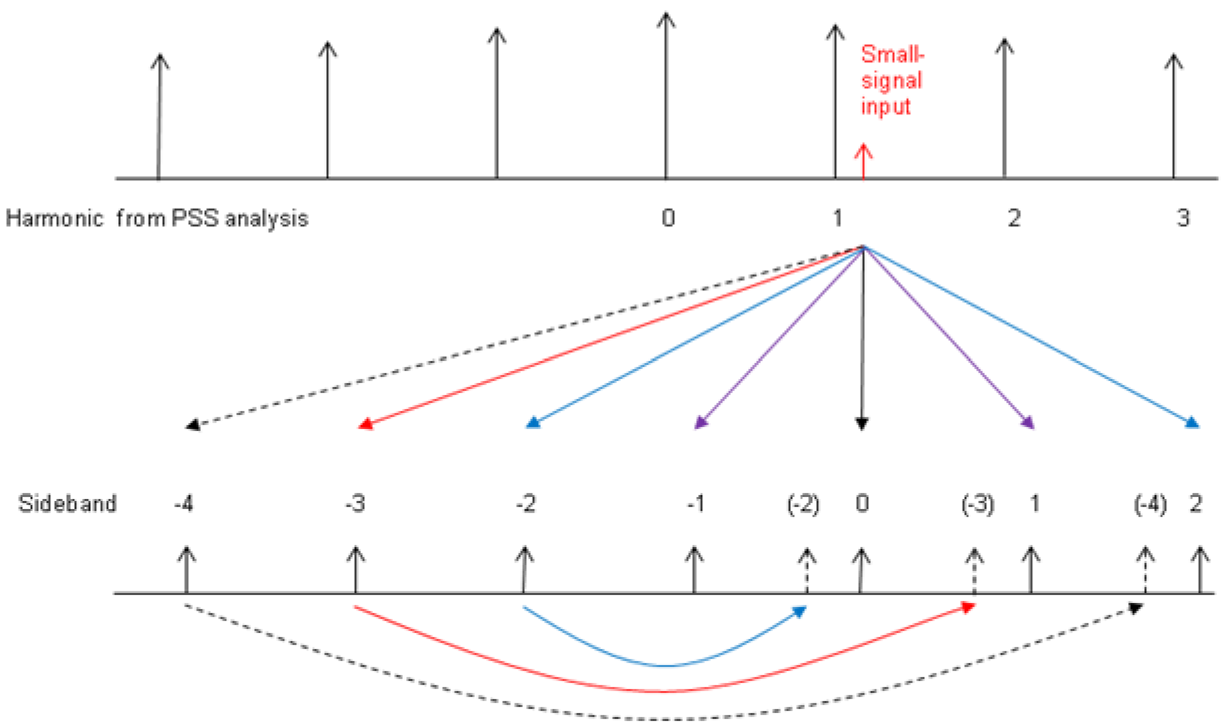
Pac has a choice called specialized analysis which is located near the bottom of the *Choosing Analyses* form. This sets the mode of the measurement to be either averaged over the whole pss waveform or at a specific timepoint in the pss waveform. Many different types of analyses can be performed, like conversion gain, distortion contribution, and instantaneous gain measurements. Below is a description of the capabilities for each specialized analysis setting.

Also note that in this section, just the concepts are discussed without any mention of how to do the setup in the Analog Design Environment (ADE). To see the ADE setup, see ADE Implementation on page 778.

Specialized Analysis=None

Down Conversion: Input Near the First harmonic

At the bottom of the pac *Choosing Analyses* form, there is a selection called *Specialized Analyses*. The default is *None*, and this is for measuring the conversion gain that is the average conversion gain with the large signal applied to the circuit (usually the LO or the sample clock) applied to the circuit. The mixing products that are produced are called sidebands. The sideband number is the harmonic number that is being mixed with to provide the output. A positive number means that the output mixing product is higher in frequency than the input frequency specified in the *Choosing Analyses* form. A negative number means that the output is lower in frequency than the input frequency. A diagram is shown below.

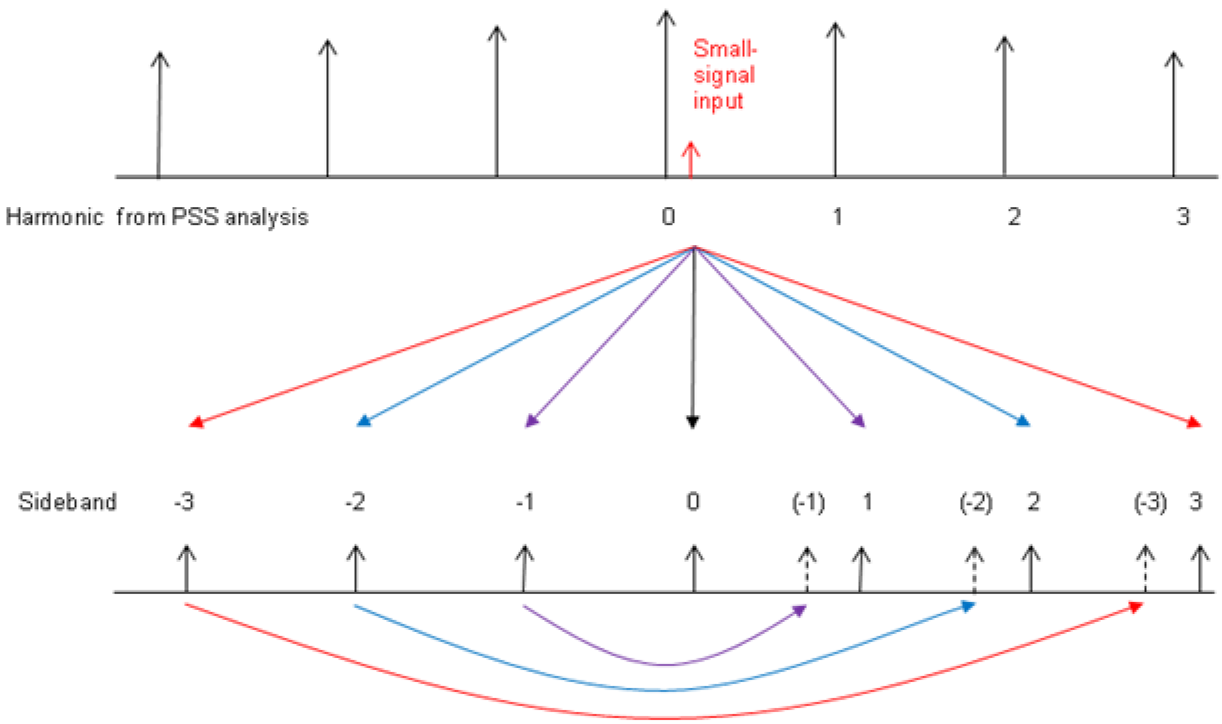


We tend to think of positive frequencies. The default is to reflect negative frequencies to positive frequencies.

Again, in pac, a single input frequency is applied at a time, and then the output mixing products are calculated through the highest harmonic of the periodic steady-state (pss) simulation specified by the *Maximum sideband* parameter. *Maximum sideband* defaults to seven for shooting and the total number of harmonics specified in the pss analysis for hb. Usually *Maximum sideband* is set to calculate the desired mixing products. This can be accomplished by setting *Maximum sideband* or by using the select from range capability in the pac *Choosing Analyses* form.

Up-conversion: Input Near Zero Frequency

In a similar manner, up-conversion is shown below.

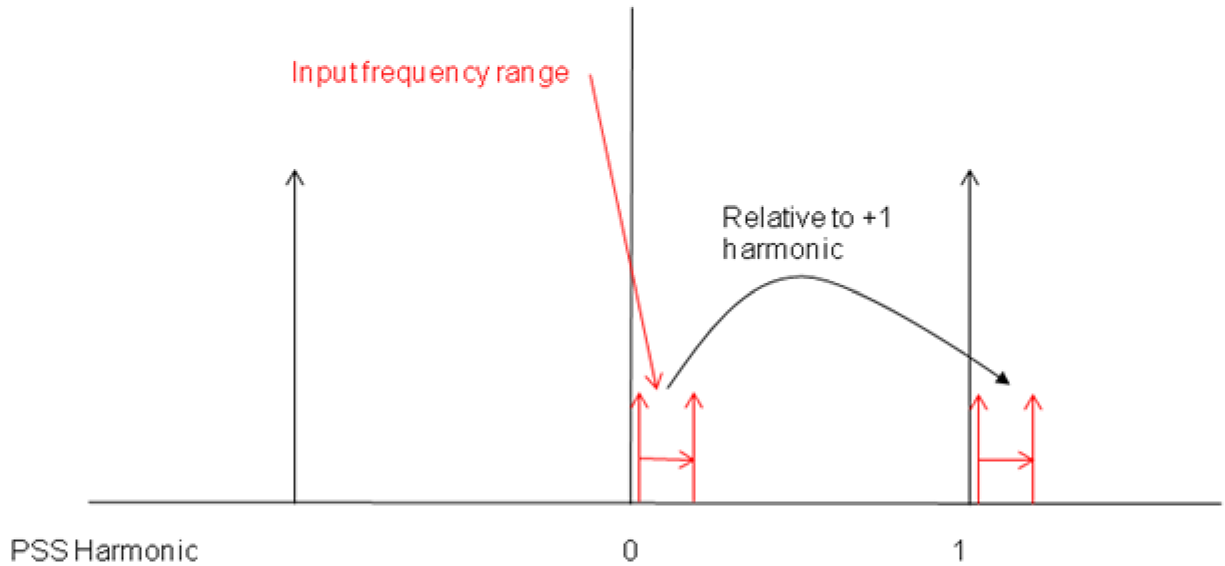


Frequency Sweep

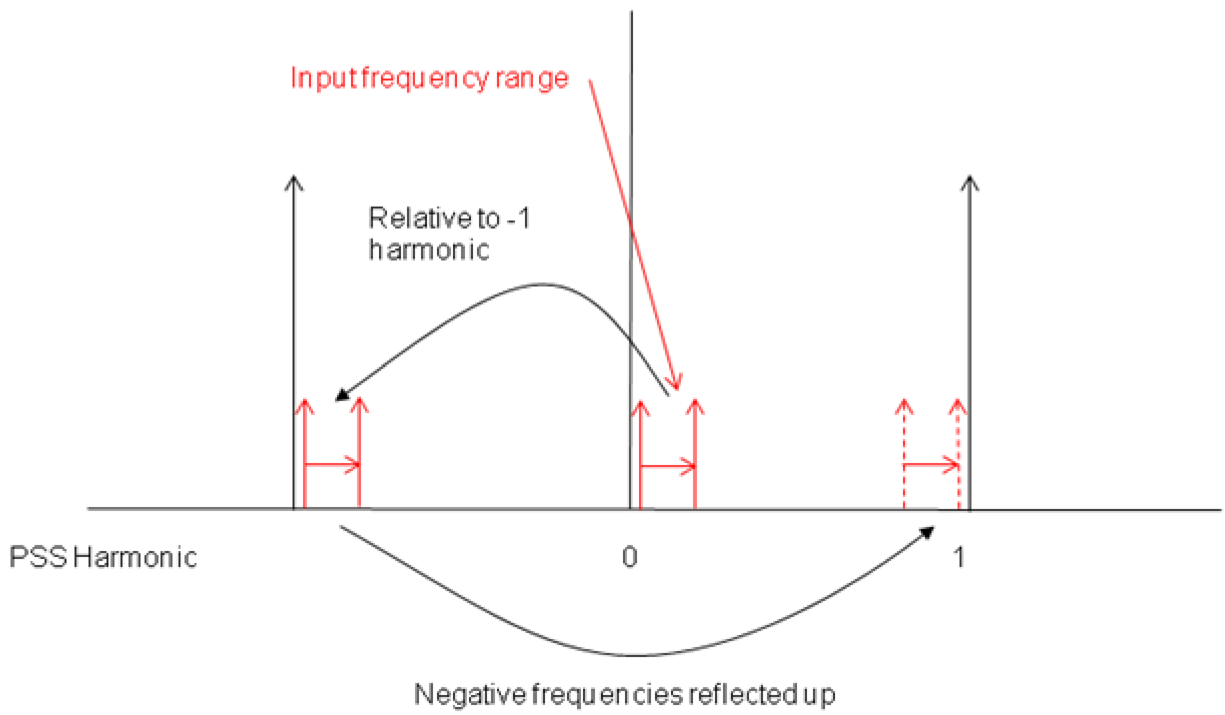
This analysis is like AC, where the input frequency is swept. The input frequency range is specified in the pac *Choosing Analyses* form.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonic in the pss analysis. For example, assume that the pss is at 1GHz, and a log sweep is desired above the first harmonic of the pss analysis. In this case, you need to select *relative* sweep, and specify 1 in the *Relative Harmonic* field. Next, type in 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade.

The sweep will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.



To use a log sweep below the first harmonic, use the same frequency range, and type in -1 as the relative harmonic. This is illustrated below.



Maximum Sideband

Sideband is the name of the PAC output mixing product. The sideband number is the pss harmonic number that is mixed with. If the sideband number is negative, the output frequency is mixed down in frequency from the input. If the sideband is positive, the output is mixed up in frequency.

Conceptually, an infinite number of mixing products are produced when a single input is applied to a nonlinear system. From the practical point of view, usually only a small number of mixing products need to be measured.

There is a property in the *Choosing Analyses* form called *Maximum sideband* that is used to define how many mixing products should be calculated. If *Maximum sideband* is zero, the pac analysis will only contain the frequency that appears without mixing or aliasing of any type. This is not equivalent to a linear AC analysis because pac takes into account the instantaneously varying nature of the LO or sample clock signal because of the large signal being present in the pss analysis.

When *Maximum sideband* is one, the outputs without frequency translation and the outputs that mix or alias with the first harmonic of the pss analysis are present in the pac analysis. When *Maximum sideband* is 10, the pac analysis includes mixing through the 10th harmonic of the pss analysis. When shooting is used for the pss analysis, the number of sidebands can be any number up to four times the number of harmonics specified in the pss analysis. When harmonic balance is selected, up to the number of harmonics in the pss analysis can be selected.

Setting Harmonics and Sidebands

Shooting

The pss analysis, when shooting is selected, has a minimum of 200 timepoints, so it inherently has frequency domain content through the 100th harmonic of the pss beat frequency. Maximum harmonic is a pss post-processing property that specifies how many harmonics to calculate in the Fourier transform of the waveform in pss. If the number of harmonics is zero, it just means that no frequency domain information should be calculated in the simulation. If the number of harmonics is 10 or less, there is no effect on the PSS waveform. When maximum harmonics gets larger than 10, the minimum number of timepoints in the pss analysis is increased. 20 timepoints at the frequency of the highest harmonic are forced. Thus, if maximum harmonics is raised, the pss waveform gets more accurate at the cost of longer runtimes and more memory required to run the pss simulation because more timepoints are forced in the pss simulation.

In order for the pac analysis to correctly measure conversion gain, the wave shape at the highest mixing harmonic needs to be accurate, so with the default settings in shooting pss, up to 40 sidebands can be accurately calculated in the pac analysis. In most cases, only a small number of sidebands needs to be calculated, and so with shooting, you can count on the pac result to be accurate.

If you need more than 40 sidebands when you are using shooting pss, you need to set the *maxacfreq* option in the pss analysis to the pss beat frequency times the number of sidebands requested in the pac analysis, or alternatively, set the number of pss harmonics to the number of sidebands divided by four. Either method works to set the maximum timestep in the pss analysis as required for the pac analysis to be accurate.

Harmonic Balance

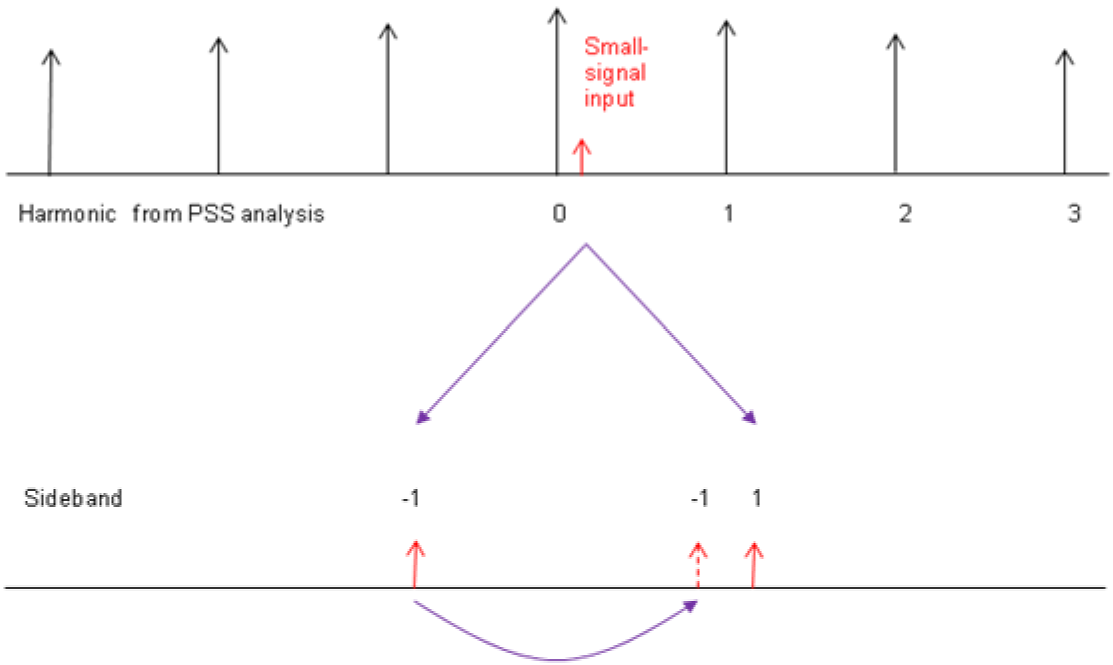
When *Harmonic Balance* is selected in the pss analysis, only the harmonics specified in the pss *Choosing Analyses* form exist in the solution. In the pac analysis, because only the harmonics specified in the pss-hb *Choosing Analyses* form exist, the maximum sideband term can only be the number of harmonics specified in the pss analysis or less. Usually when *Harmonic Balance* is selected in the pss analysis, the *Maximum sideband* field in pac usually should be left blank which calculates all the output mixing products.

Note that with harmonic balance, the number of harmonics (and oversample factor if you have non-sinusoidal waveforms) in the pss analysis needs to be determined. Start with a number of harmonics in the pss based on an estimate of how many harmonics the circuit actually produces with your input waveform, and run the pss and pac analyses. Remember that for non-sinusoidal waveforms, oversample factor also needs to be set greater than one. Run the analysis and do the desired measurement. Now increase the number of harmonics by roughly 50% and run the simulation again. If the answer does not change, then you had enough harmonics originally, and you might actually be able to decrease the number of harmonics from the original setting. If you have non-sinusoidal waveforms, increase the oversample factor before increasing the number of harmonics to find the minimum value for both oversample factor and harmonics.

Modulated

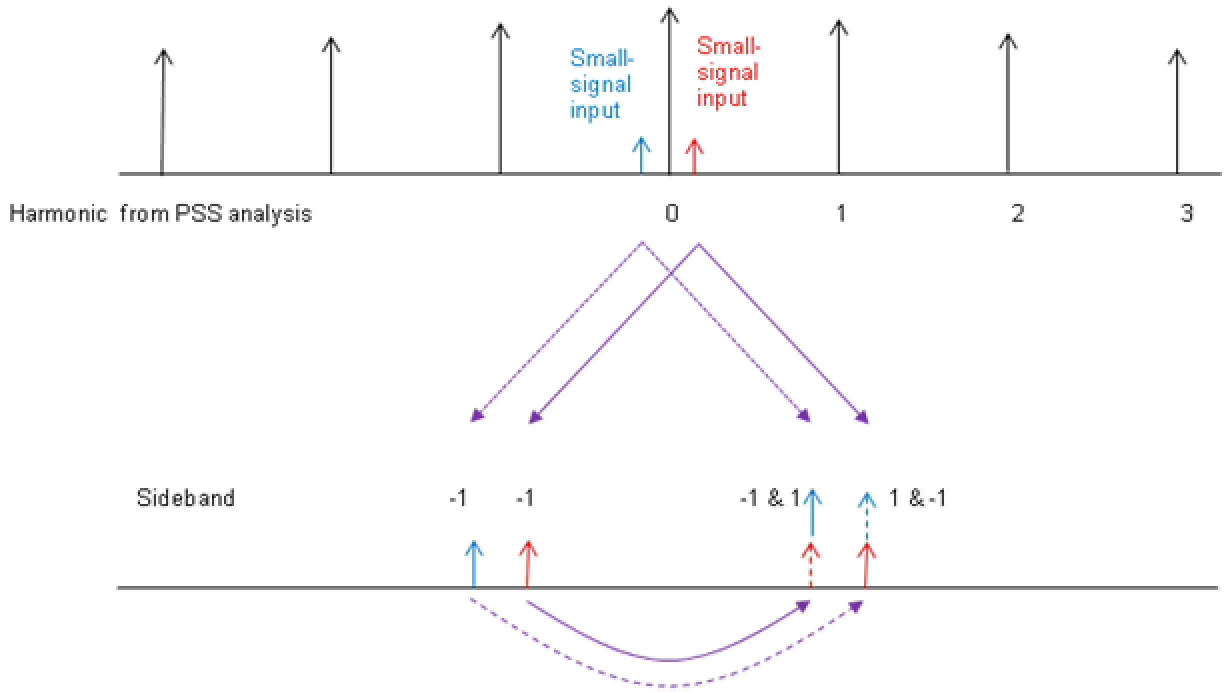
Modulated pac is typically used for measuring AM to PM measurements in driven circuits and oscillators. In this case, the input is usually the power supply ripple at low frequency, and the desired output is near the first harmonic of the output. Modulated is an averaged measurement over the whole pss waveform (shooting) or all the harmonics (harmonic balance). Pac apples a modulated input signal to the circuit, and is able to measure a modulated output signal from the circuit. The types of modulation are Single Side Band (SSB), AM, and PM.

A diagram is shown below for SSB input.



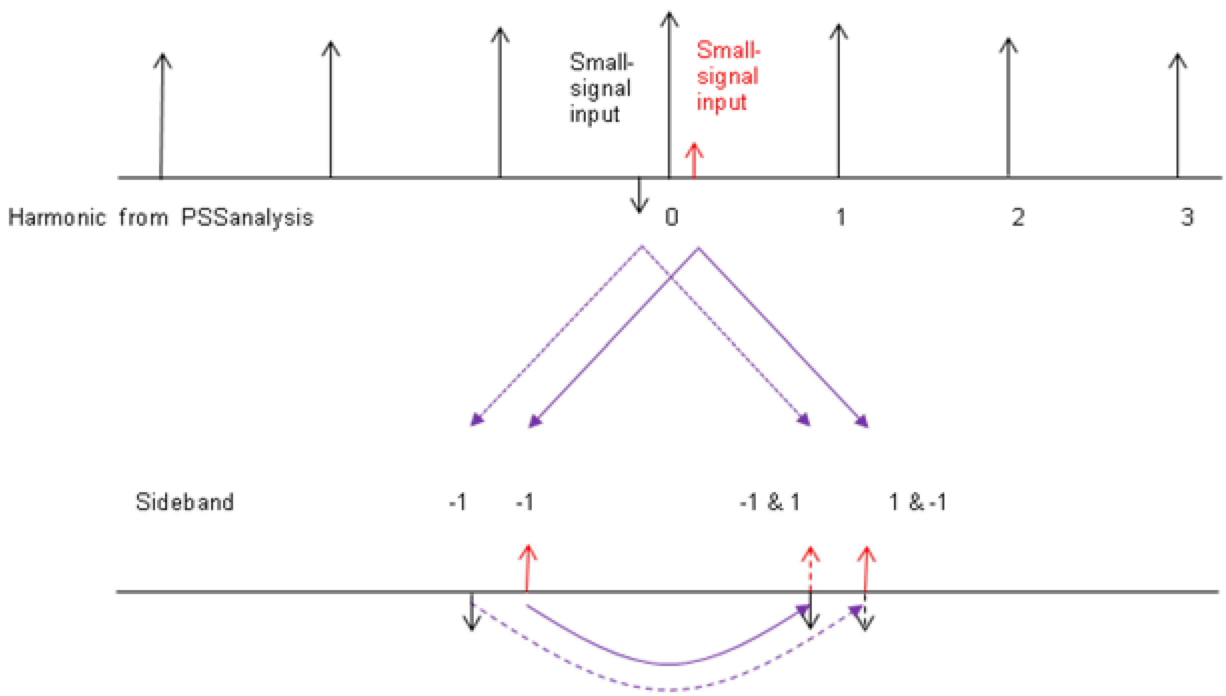
Note that although the input occurs at one frequency, the AM output can be measured directly from the single input.

When AM is applied at the input, the following figure is shown.



The difference is that in addition to the input at low positive frequency, another input (to form an AM signal) is applied at a low negative input. This second input adds to the output signal near the first harmonic.

For PM, the negative input amplitude is negative, as shown below.



By running all three inputs, the SSB/AM/PM input to SSB/AM/PM output can be calculated. This is a setup choice in the *Choosing Analyses* form.

In the *Choosing Analyses* form, the input frequency range, and the output harmonic must also be specified.

All the plot functions are implemented in the *Direct Plot Form*.

Rapid IP2/IP3 (AC and PAC)

Rapid IP2 and Rapid IP3 have been added to the ac and pac analyses. First, a provision has been made to allow two input frequencies. Second, the ac and pac analyses have been enhanced so that harmonics of those signals and the intermodulation distortion can be calculated. Doing this in a small-signal analysis is much faster than doing this in a large-signal analysis because the large-signal effects are ignored.

The Rapid IP2 and Rapid IP3 that is available in the ac *Choosing Analyses* form allows the calculation of IP2 and IP3 for amplifiers. In pac, Rapid IP3 and Rapid IP2 are provided for circuits that translate frequencies, whether that translation is an up or down conversion. In this case, the signal that is causing the frequency translation (the LO or the sample clock) is applied in the pss analysis first in order to characterize the nonlinearity of the system so that the correct IP2/IP3 measurement can be made. Note that pac and ac are small-signal analyses, so the limitation is that the measurement is a small-signal IP3 or IP2. If you need a large-signal IP2 or IP3 measurement, use a large-signal analysis like hb or qpss to make the measurement. Note that when you are using Rapid IP2 or Rapid IP3 in the ac analysis, either a SpectreRF license is required, or an additional mmsim token is required to run the analysis.

Make sure that the input level for the IP3 or IP2 measurement is in the small-signal range. Rapid IP2 and IP3 have exceptionally small numerical noise floors, therefore, it is recommended that you specify a level that is too small, rather than too large. Rapid IP2 and Rapid IP3 work by evaluating the device equations for nonlinearity. Do not specify one volt as the input amplitude because this is generally in a large-signal range, and usually causes the devices to reach points where the device currents are deliberately linearized, thus making the IP2/3 measurement more linear than it actually is.

The plot functions for Rapid IP2 and Rapid IP3 are implemented in the *Direct Plot Form*.

1. Select *Results - Direct Plot - Main Form*. In the Direct Plot Form.
2. Select *ac* or *pac* from the *Analysis* section.
3. Select *Rapid IP2* or *Rapid IP3* (depending on which was set up) from the *Function* section, and click *Plot*.

The X and Y points are shown for the intercept.

The X value is input-referred IP2/IP3, and the Y is output-referred IP2/IP3.

Note that if you have a switched-FET mixer and you are using the BSIM3 or BSIM4 models, due to a model limitation, the IP3 calculation from rapid IP2/IP3 will be incorrect. In this case, the PSP model is recommended. If a PSP model is not available, use hb or qpss with an input amplitude that is as large as possible while still being in the small-signal range for your circuit or switch to a 1dB gain compression measurement, The compression measurement is much more accurate than the IP3 measurement for switched FET mixers.

Compression and IP2 distortion summary

The distortion summaries measure distortion for an amplifier in ac or in circuits that have frequency translation in pac. The idea is to find which devices in a signal path contribute relatively more or less to the distortion. Since IP3 and compression are related mathematically, the compression distortion summary provides information about which devices contribute the most amount of third order intermodulation distortion. The IP2 distortion summary measures which devices contribute to the second order distortion. Note that when you are using the distortion summaries in the ac analysis for amplifiers, either a SpectreRF license is required, or an additional mmsim token is required to run the analysis.

Both distortion summaries use the *PAC Magnitude* property in the input port. Make sure that the *PAC Magnitude* (in volts or dBm) is actually in the small-signal range, (At least 10 dB smaller than the 1dB gain compression point) or the results will not be accurate. This is because the pac results do not take into account the large-signal limits like slew-rate or power supply voltage and the device models deviate from the ideal equations at very large amplitude by just continuing a linear slope above a certain amplitude. That amplitude depends on the device model parameters and is likely to be reached with a 1 volt input amplitude. The compression distortion summaries have an extremely low numerical noise floor, so it is better to set an input amplitude that is too small, than to set it too large.

The distortion summaries provide a list output, not a plot output. Accordingly, instead of selecting *Results - Direct Plot - Main Form*, select *Results - Print - PAC Distortion Summary*.

For the compression distortion, you get the actual gain with the distortion of each device by itself divided by the ideal small-signal gain provided by the pac analysis. In addition, the amplitude of the first three harmonics of the linear output frequency is calculated.

For the IM2 distortion summary, you get the total IM2 output amplitude that results from having only a single device in the circuit.

Sampled

Sampled is provided in order to calculate the instantaneous conversion gain at a voltage threshold crossing in the pss waveform or at specific timepoints in the pss simulation. Note that this analysis is fundamentally different than the analyses discussed earlier. All the previous analyses measured the average conversion gain that is produced when all the harmonics are taken into account (hb) or all the timepoints are taken into account (shooting). In sampled, an instantaneous measurement at a single point in time (or a list of times) of the LO or clock waveform is made.

Note that when harmonic balance is selected, sampled pac is the same as in shooting, but the time-domain waveform is the ifft of the harmonics calculated by the harmonic balance analysis. This is useful for measuring the conversion gain from the power supply to the oscillator output at a threshold crossing, or at specific times in the LO waveform to the output of a mixer.

When shooting is selected, a minimum of 200 timepoints exist in the solution waveform, and shooting automatically makes the timestep smaller in areas of rapid change, thus, there is likely a datapoint in the pss analysis that is very near the voltage level you specify as a threshold. In harmonic balance, the timepoints in the ifft depend on the number of harmonics and the oversample factor. Set a large enough number of harmonics and/or set a large enough oversample factor to make sure that there is a timepoint in the ifft that is very near the actual threshold value you specify in the *Choosing Analyses* form.

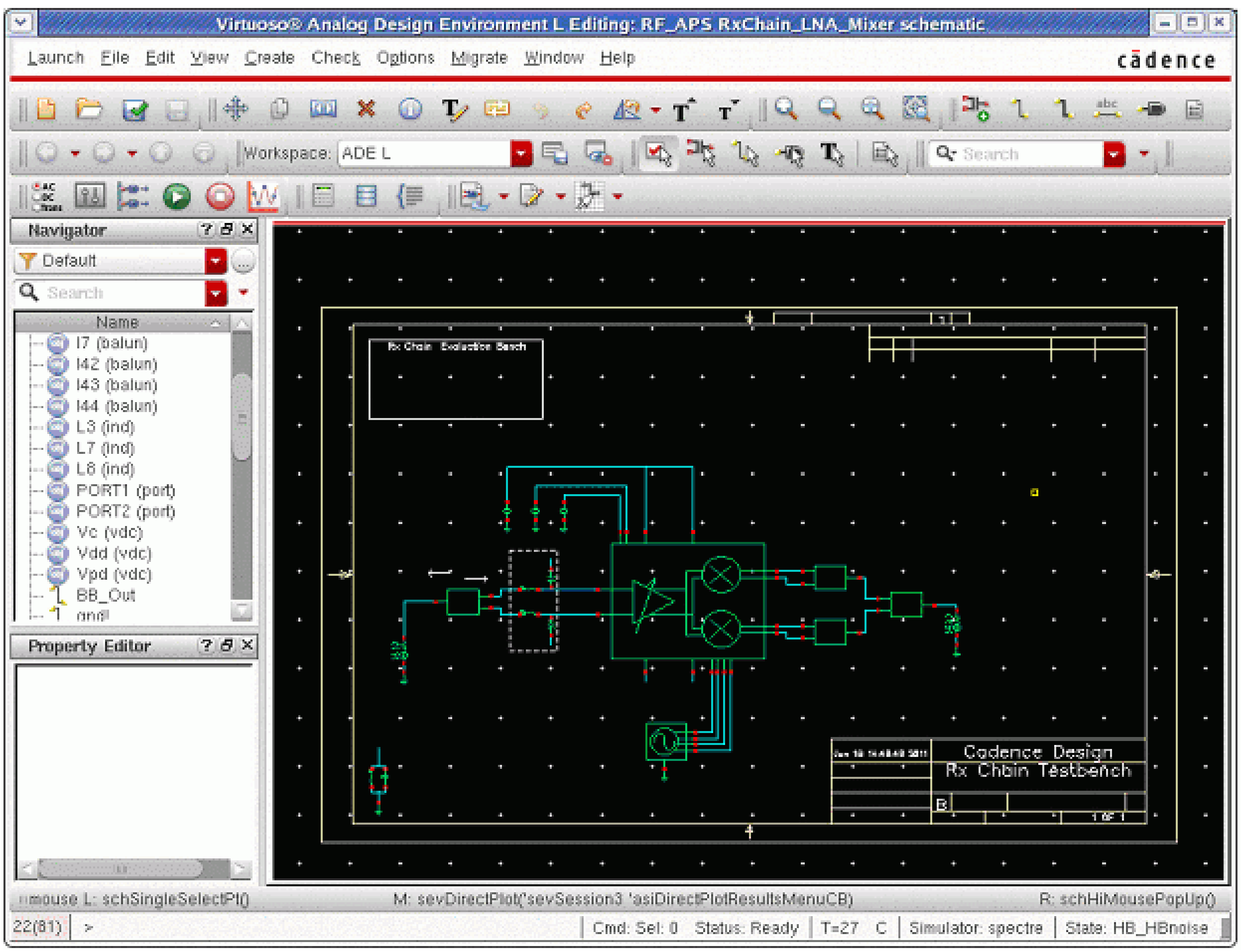
ADE Implementation

Note: In general, the examples in this section show shooting pss. The examples work for harmonic balance pss unless otherwise noted. This is done mostly to show shooting. Harmonic balance has its own section in Chapter 3, "Frequency Domain Analyses: Harmonic Balance.".

The settings in the PAC form can be slightly different for shooting and harmonic balance, and the method for both is covered in each section.

Circuit, Input port Setting, and ADE Setup for all the Examples in This Section

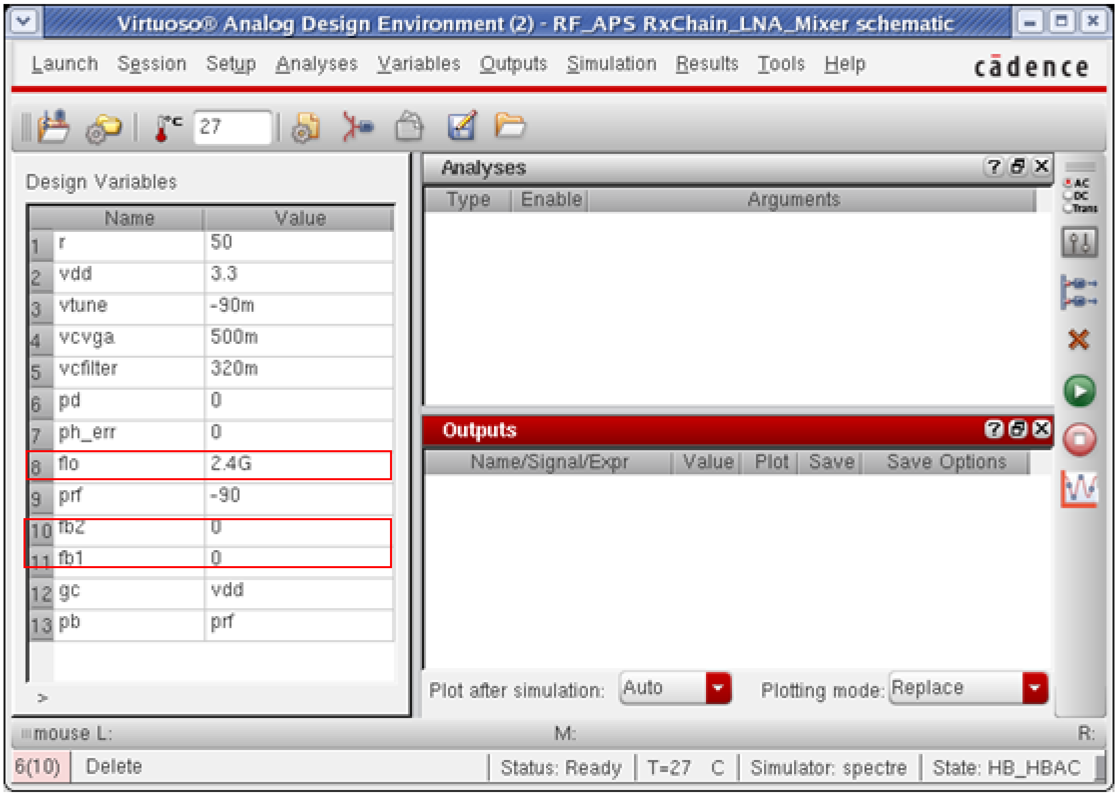
Consider the following circuit:



This above circuit is an LNA and a mixer for a 2.4GHz application that is implemented in CMOS. The input source (a port in Cadence terminology) has the values for its frequency and amplitude set to variable names. The PAC magnitude property is set to 1 volt in order to allow easy conversion gain measurements. The reason for the variables is that the frequency and amplitude can be changed in the ADE environment without changing the schematic.

The *Edit Object Properties* form for the input port is shown below.

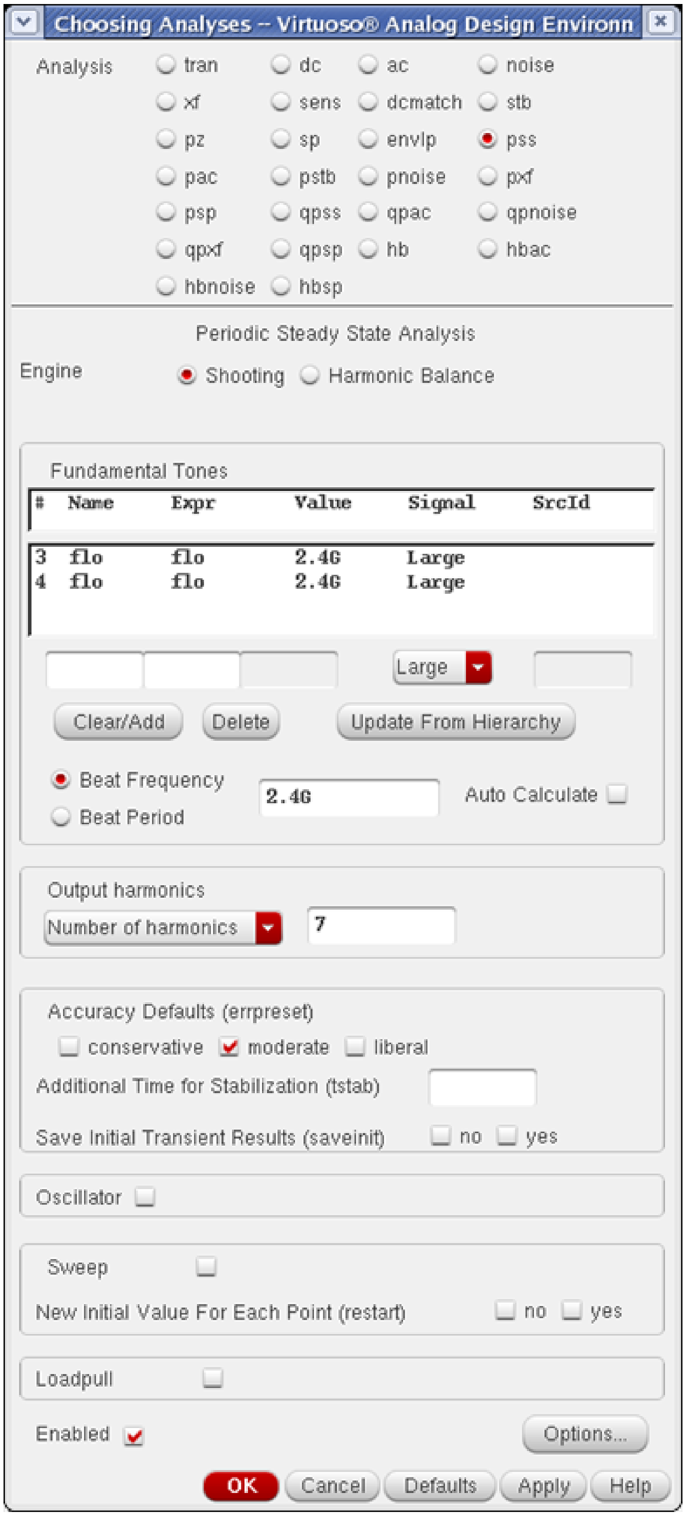
The ADE has the variables set to appropriate values, as shown below.



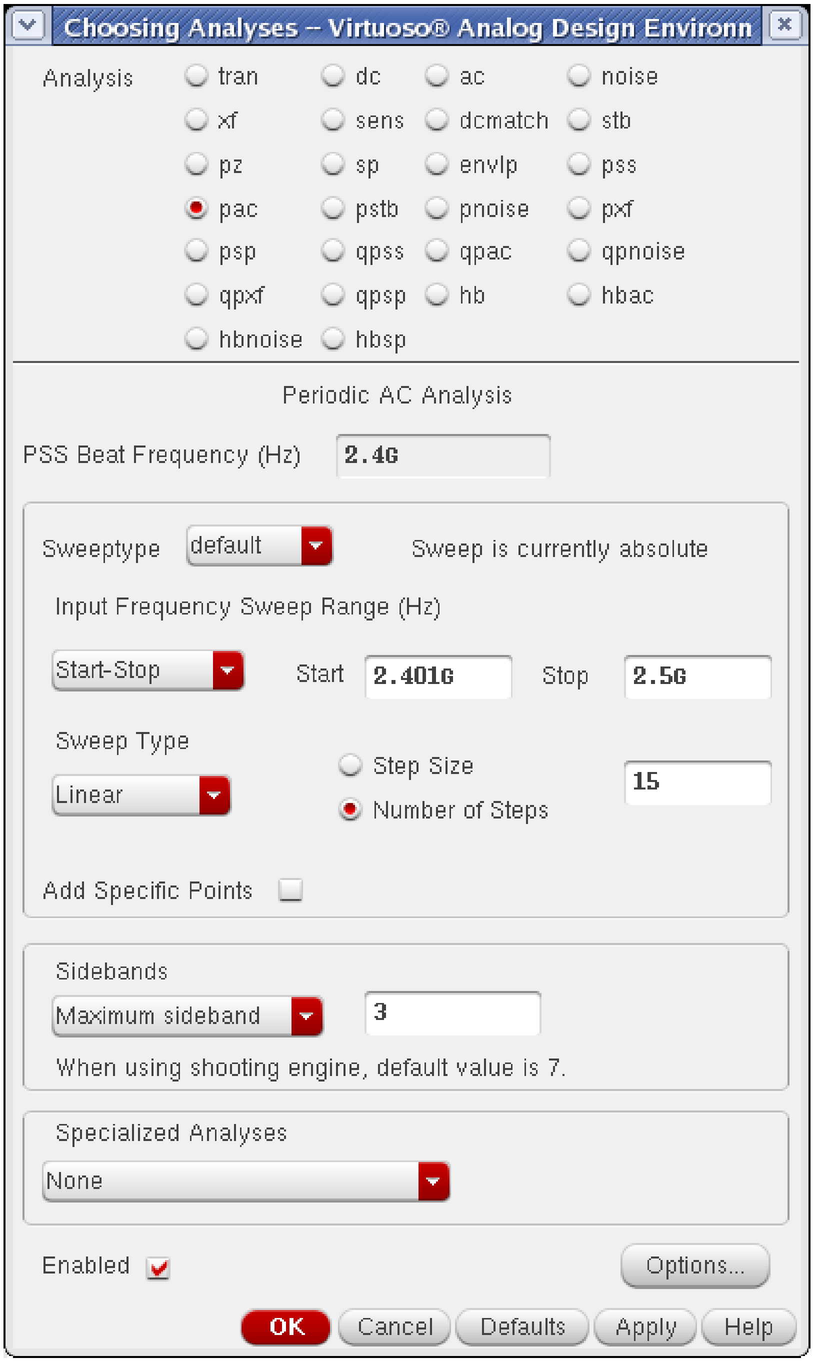
Note that the variables fb1 and fb2 (which set the input frequency of the input port) are set to zero, which effectively disables the RF input. Only the LO signal is applied to the circuit for the pss analysis.

Using PAC Analysis for Conversion Gain Measurement

Set up the pss analysis with just the large signal that causes the frequency translation to occur, which is generally the LO signal or the sample clock. For more information, refer to Overview of Periodic Steady-State (pss) Analysis on page 547. Regarding whether to use shooting or harmonic balance, pick the analysis that produces the smallest runtime. In general, shooting is recommended for systems that have sharp edges, and harmonic balance is recommended for systems that are more sinusoidal.



Set up the pac *Choosing Analyses* form as follows:



1. Specify the input frequency range. In this case, the *Sweeptype* is set to *absolute*, which means that there is no frequency shift applied to the input frequency. If a log sweep above the first harmonic is desired, set the *Sweeptype* to *relative*, and specify one as the relative harmonic number. The input frequency range would be 1MHz to 100MHz to get

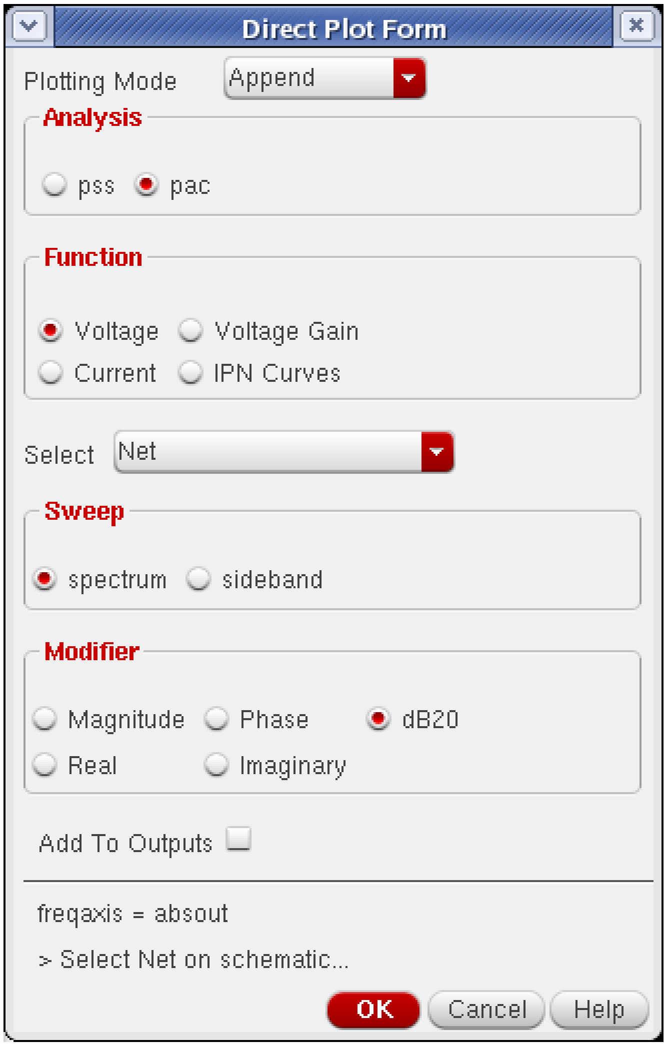
the same frequency range as in the example above. Relative shifts the frequency specified the harmonic number times the pss frequency.

2. For pac, the *Maximum sideband* field should be set to the highest harmonic of the pss analysis that mixing products are desired to be calculated. This is generally determined by the application. If you only need one or two output mixing product, choose *Select from range*, and highlight the sidebands you want to calculate. Setting *Maximum sideband* for pac is usually not critical as long as enough sidebands are calculated to measure the desired output mixing product (called a sideband). The *Direct Plot Form* allows the plotting of individual output mixing products as will be shown later. As a general recommendation, set the number of sidebands equal to the number of pss harmonics.

3. For a conversion gain measurement, set the *PAC Magnitude* to 1 volt on the *Edit Object Properties* form, as shown earlier. This is always volts peak.

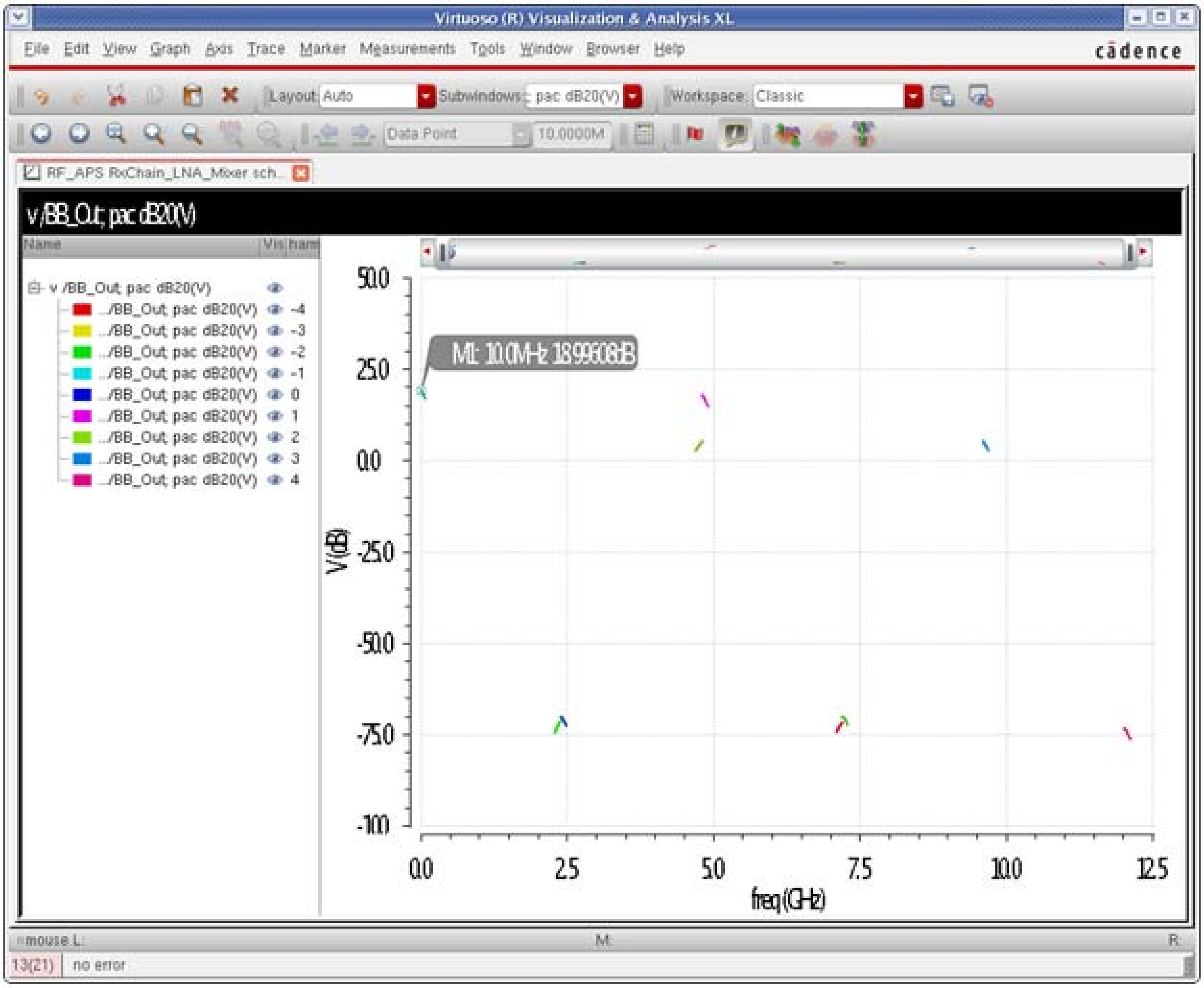
4. Now run the simulation. In the Analog Design Environment (ADE), select *Simulation - Netlist and Run*.

The *Direct Plot Form* is displayed.



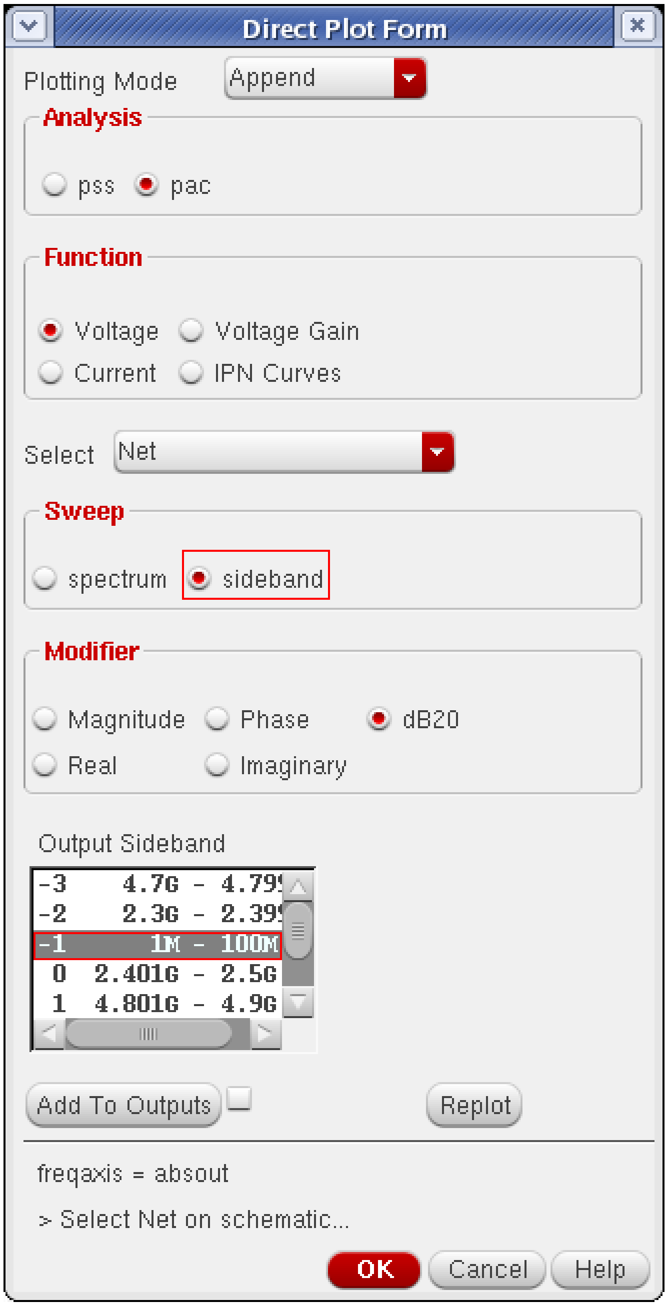
Mixing products are calculated for mixing the pac input frequency range with the harmonics calculated in the pss analysis. Harmonics of the pac input and harmonics of the output mixing products are not calculated because pac is a small-signal analysis. Because it is small-signal, the harmonics of the input and the harmonics of the mixing products are so small that they do not matter.

To measure the outputs you desire, place a marker at the desired output frequency of interest and read the conversion gain, as shown below.

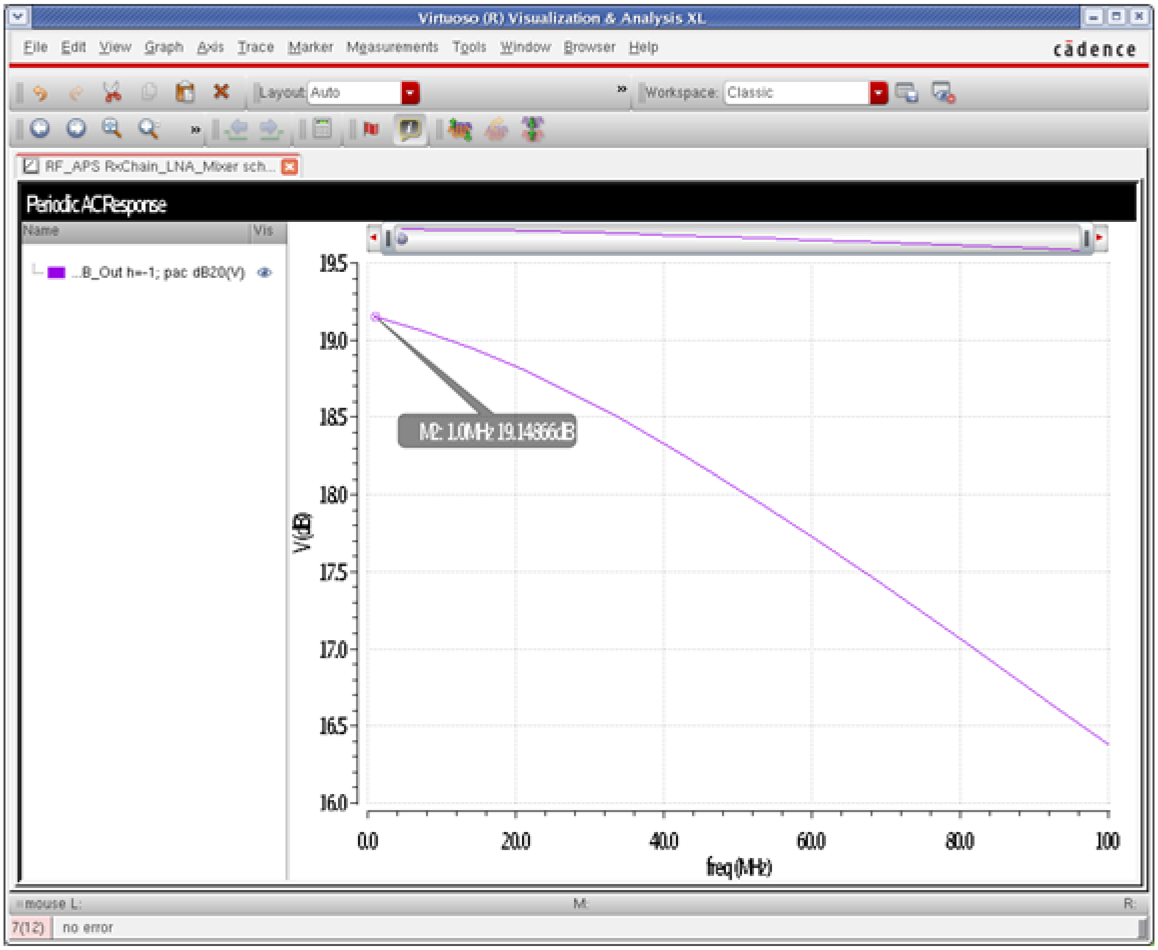


To plot a single output sideband (mixing product), in the *Direct Plot Form*:

1. Select *sideband* instead of *spectrum*, and select the desired output mixing product form the list as shown below.

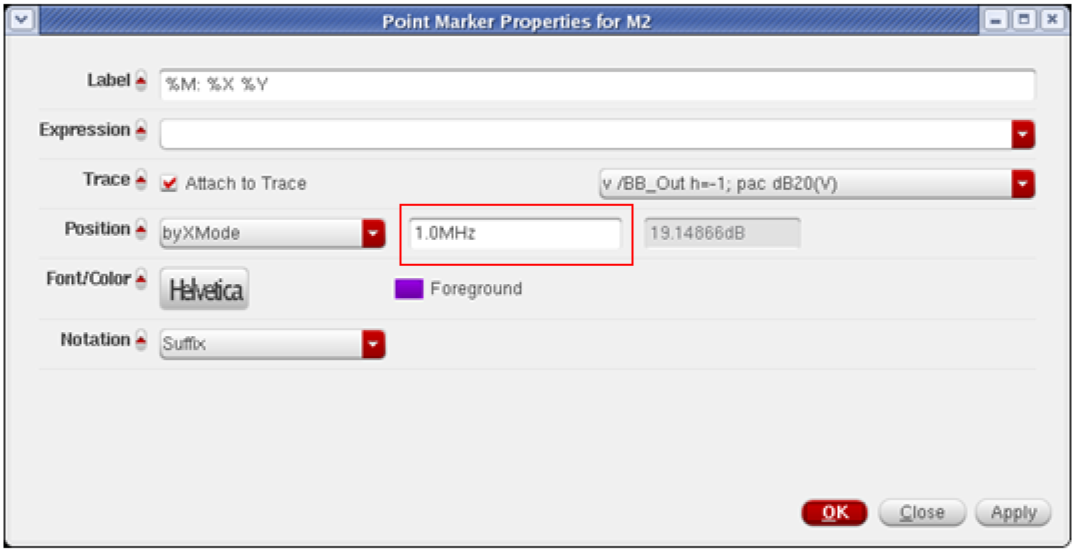


2. Select the output net in the schematic, or click the *Replot* button in the *Direct Plot Form*. The waveform tool displays the output, as shown below.



3. Position a marker by moving the cursor to the desired output frequency and type `m`. The marker appears, as shown in the figure above.

4. To position the marker at an exact frequency, place it as above, and then move your cursor over the intercept point on the trace. Click and hold the right mouse button, move to *Marker Properties*, and release the mouse button. The *Point Marker Properties* window appears, as shown below.



5. Type the frequency in the *Position* field, and click *OK* or *Apply*.

Sampled Circuits and Switched-capacitor Filters

For sampled circuits or for switched-capacitor circuits, always use shooting pss because the clocking waveforms have rapid transitions. Shooting captures these transitions in a natural and accurate way.

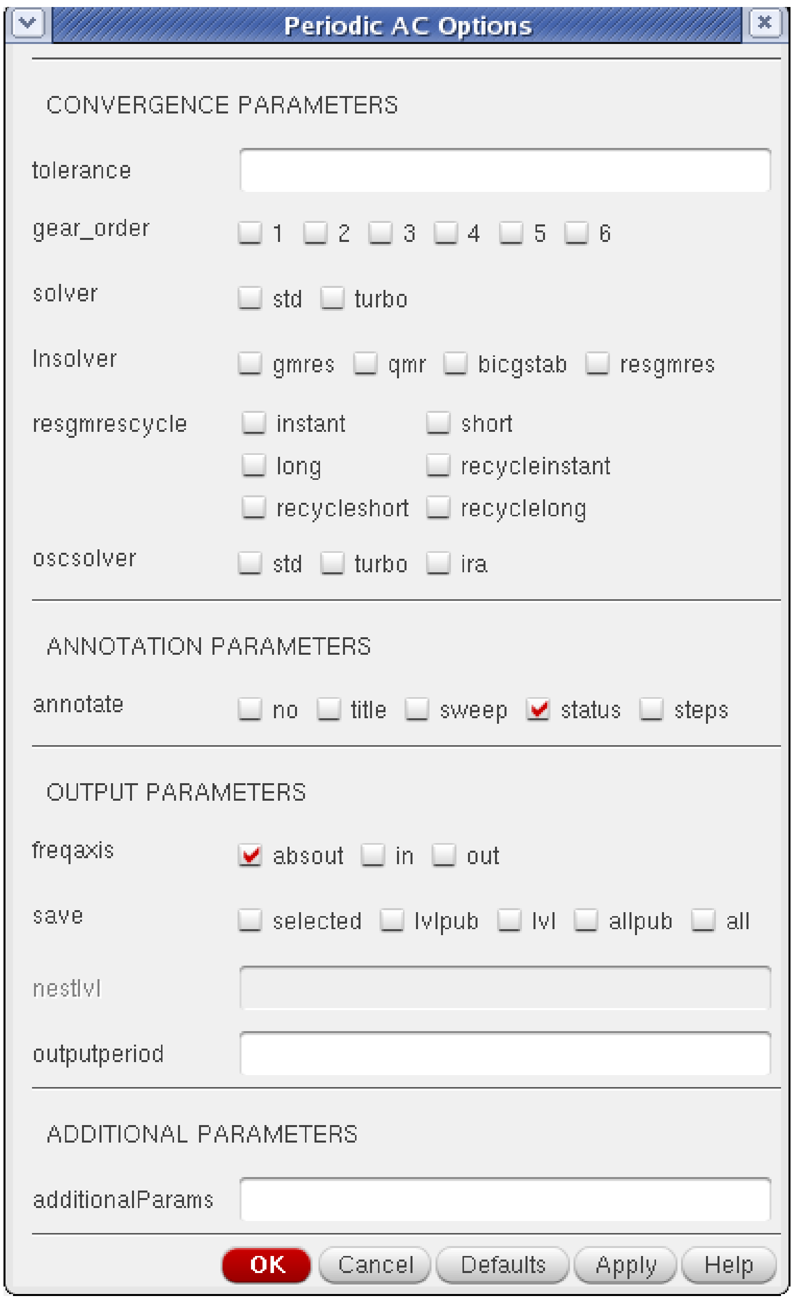
In pac, generally only the sideband with zero frequency translation is desired, so *Maximum sideband* can be set to zero. A warning is produced in the Spectre output window stating that the number of sidebands is lower than anticipated. This warning can be ignored.

Note that the average gain is determined by pac. In the sampling circuit, this is generally very near unity gain, but for the switched-capacitor circuit without a sample and hold, the gain is generally a bit less than half amplitude which is about -6dB. This is because the output waveform follows the input waveform for a little less than half the time, and the capacitors are reset by the switches the other half the time.

If you desire to see the aliasing at different harmonics of the clock, use pxf analysis. For example, imagine a sampling frequency of 1MHz. In pac, with the zero sideband only, the input and output frequencies are always the same. Therefore, if the input is at 1.1MHz in pac, the output is also at 1.1MHz. Generally, the term that is interesting is the term that aliases down from 1.1MHz to 100KHz which is not measured by the pac analysis (The input and output frequencies are the same in the zero sideband.) To see the aliasing at the different harmonics, use pxf with an output frequency of near zero to 500KHz, and specify *Maximum sideband* to the harmonic number desired. If the *Maximum sideband* is greater than 40, either set the pss option *maxacfreq* to the number of sidebands times the pss beat frequency, or set the number of harmonics to the number of sidebands divided by four.

Commonly Used PAC Options

Freqaxis

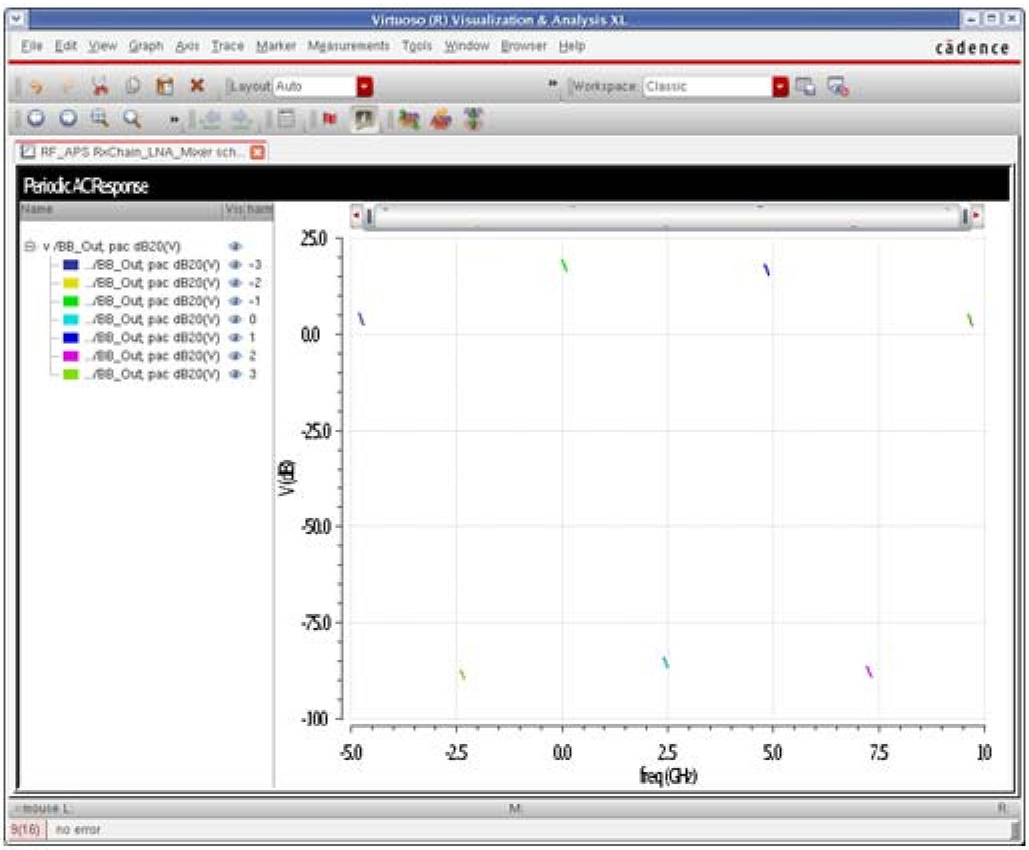


freqaxis specifies whether you want to see the negative frequency axis or not.

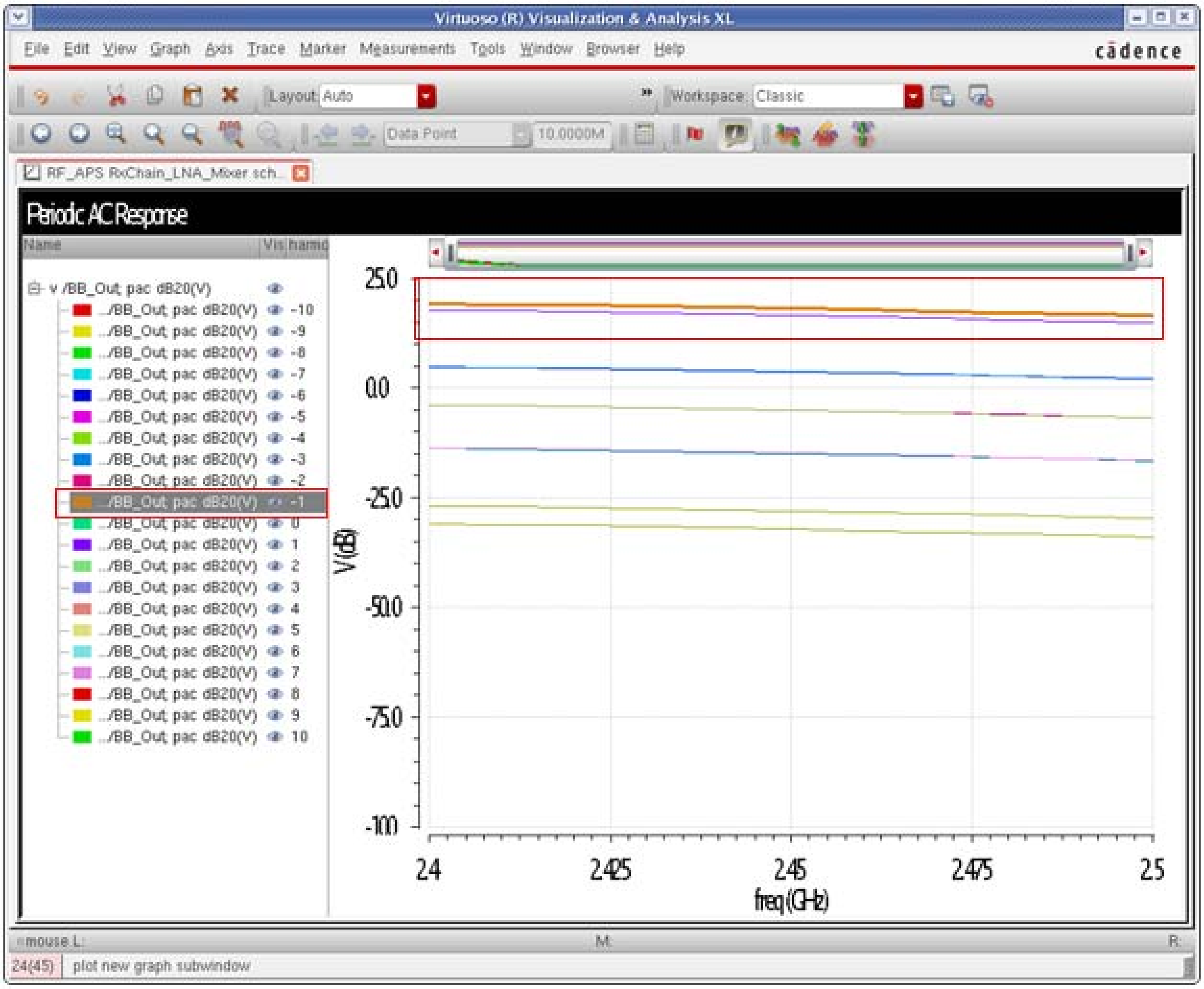
The analysis calculates the output frequencies and amplitudes, so *out* and *absout* are reasonable choices. *out* displays the negative frequency axis. *absout* (absolute value of the output) displays positive output frequencies. This is the default.

If you select *in*, all the outputs at the different frequencies are plotted on the same input frequency range scale. This is not recommended. The default is *absout*.

Below is an example of *freqaxis* set to *out*. Note that the data is the same as before. It is just displayed with the negative frequency axis present. The negative frequencies have not been reflected up to the positive frequency domain.



Below is an example with the *freqaxis* option set to *in*. All the outputs at different frequencies are plotted on the input frequency scale. The frequency can be calculated by selecting the appropriate sideband from the list on the left side of the waveform tool, and then adding the sideband number times the pss beat frequency to the frequency displayed on the X axis.



Tolerance

Leave this option at the default value.

Pac uses an iterative solver to calculate the output amplitudes. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough, and the tolerance option specifies that accuracy for pac. For shooting, the default tolerance is 1e-9. For driven circuits where HB is the pss engine, the default is 1e-6. For oscillators where hb is selected for the pss engine, the default is 1e-4.

Solver

The solver option is used only for driven circuits. The default solver is the turbo solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pac input frequency is very close to the frequency of one of the harmonics in the pss, warning messages will appear in the pac output warning that the accuracy might not be good enough. If you see these messages, select the std solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but which takes longer to run than the turbo solver.

Hbprecond_solver

This option is only available only when harmonic balance is selected in PSS.

The basic solver is the only solver available in standard Spectre. Autoset is the default solver when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window.

Oscsolver

The oscsolver option is used only for oscillators. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pac input frequency is very close to the frequency of one of the harmonics in the pss, warning messages appear in the pac output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but takes longer to run than the *turbo* solver.

For very large oscillator designs, the ira oscsolver may take less memory and run faster with no loss in accuracy, however, it is less robust for convergence compared to *turbo* or *std*.

Gear_order

Do not change this option.

Lnsolver

Leave this option at the default. Each pac sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

gmres is the Generalized Minimum RESidual method.

qmr is the Quasi-Minimal Residual method.

bicgstab is the STABilized BI-Conjugate Gradient method.

resgmres is the REStarted Generalized Minimal Residual method.

Resgmrescycle

Leave this option at the default. For the resgmres linear solver, there are several different options.

Outputperiod

Do not use this option.

AdditionalParams

additionalParams is typically used for new features that are being beta tested.

For more information about the other options, type `spectre -h pac` at the command prompt in a Unix shell window

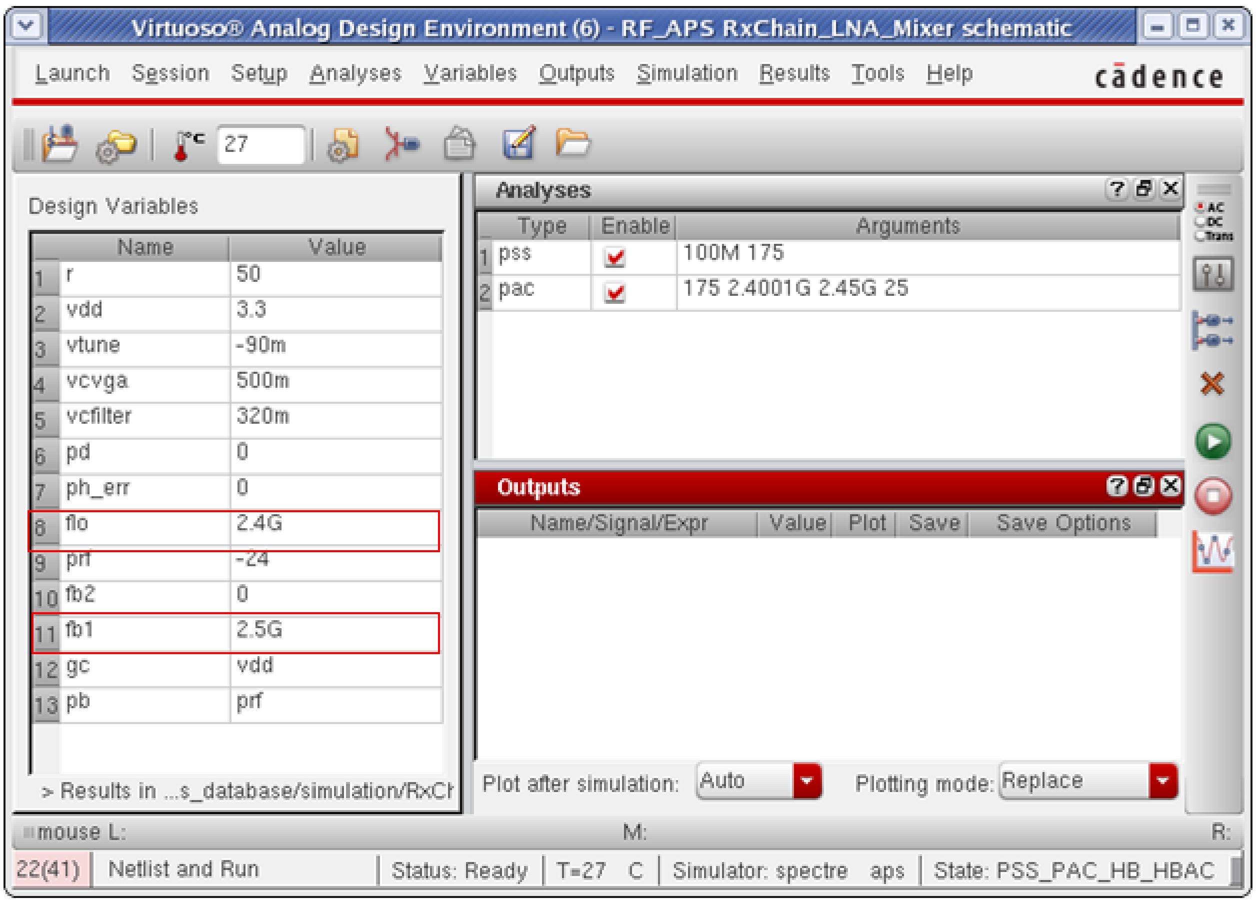
PAC With Multiple PSS Inputs

This is generally used when the conversion gain is desired with a large blocker signal present at the RF input. The basic strategy is to apply the signals that cause the nonlinearity to occur (the LO and the blocking signal) in the pss, and then use pac to measure the conversion gain when the blocker is present. In many cases, the blocker power is swept in order to measure the degradation in conversion gain when the blocker gets large.

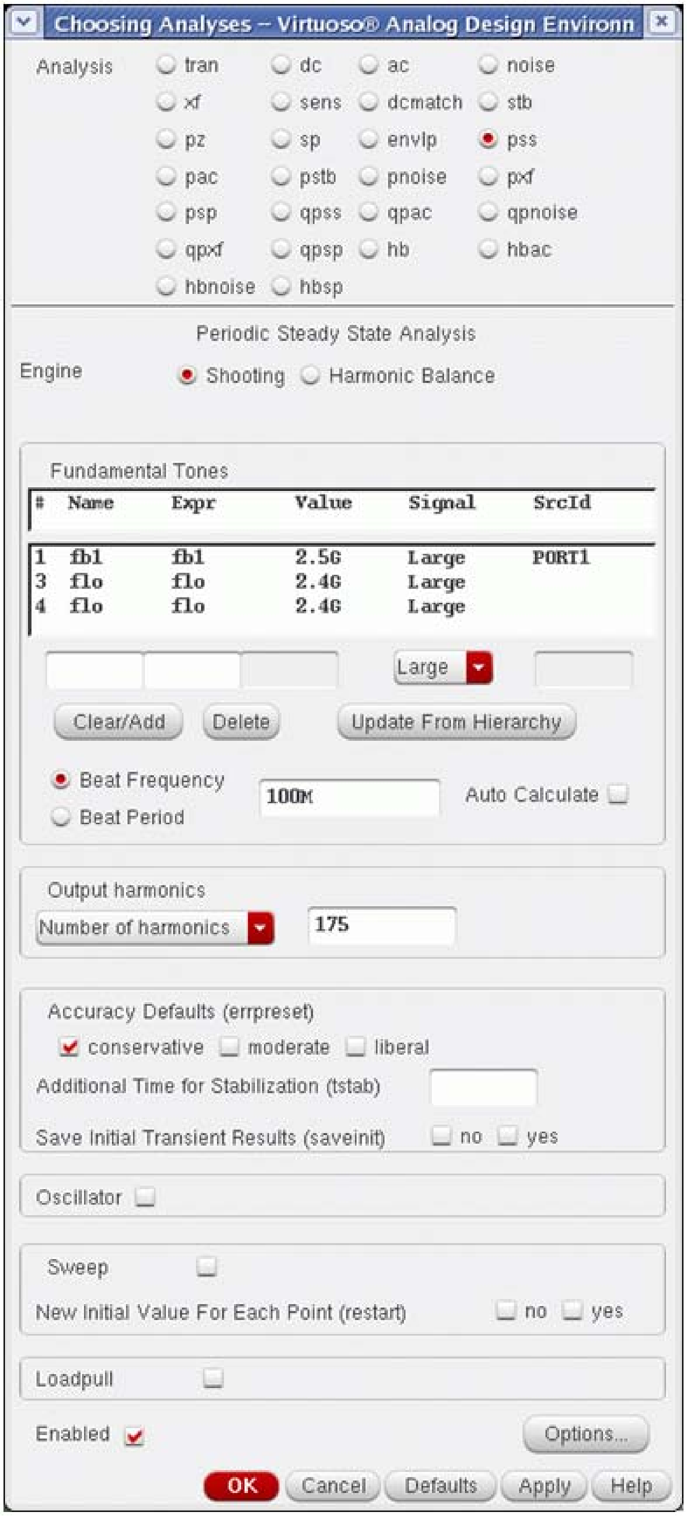
Multiple input frequencies might be able to be run in the PSS, depending on the ratio between the highest input frequency divided by the pss beat frequency. If this ratio is about 25 or less, then pss can be used for the simulation. If the ratio is greater than 25, consider using harmonic balance (hb) or qpss to run the simulation because it is very likely to take less time to run.

To set up the second input, the variable fb1 was set to 2.5G in the ADE window, as shown

below. This variable sets the frequency of the first source in the port component that supplies the RF signal. In this case, the LO is at 2.4GHz and the RF is at 2.5GHz.

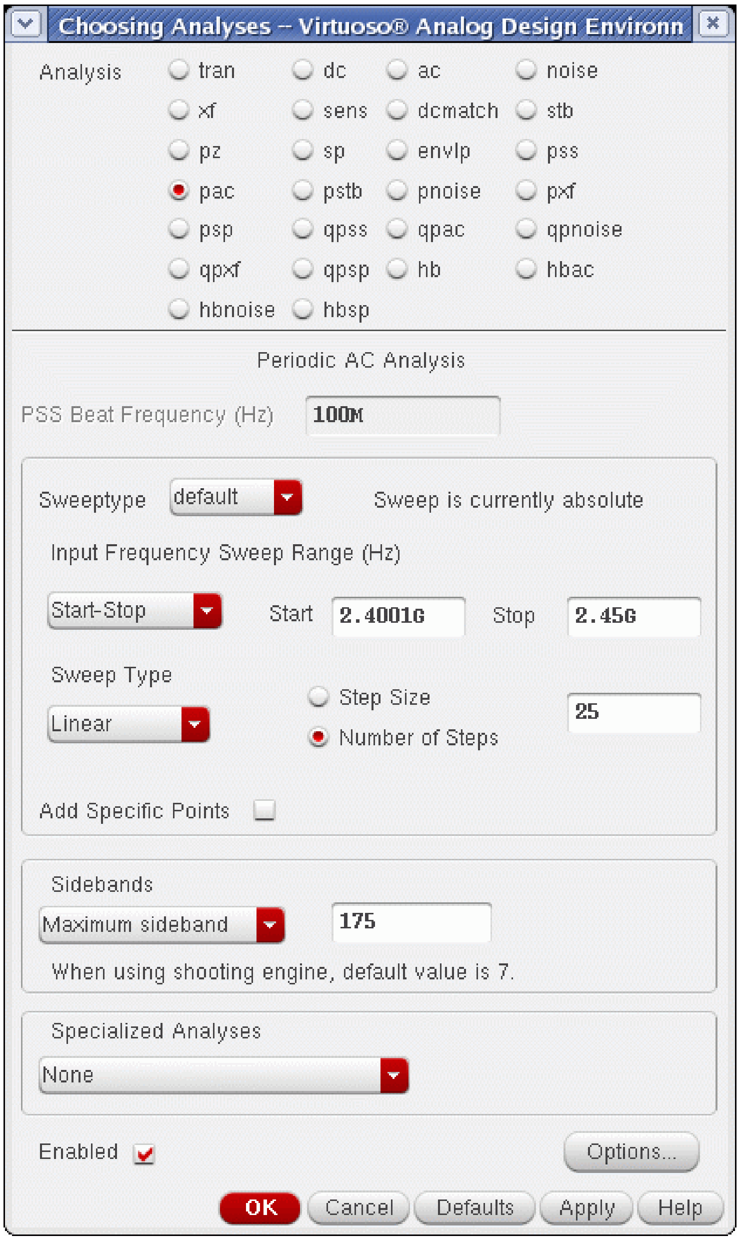


The pss was set up with 175 total harmonics of 100MHz to allow harmonics through the seventh harmonic of the LO signal at 2.5GHz to be calculated. For more information on pss, Overview of Periodic Steady-State (pss) Analysis on page 547.

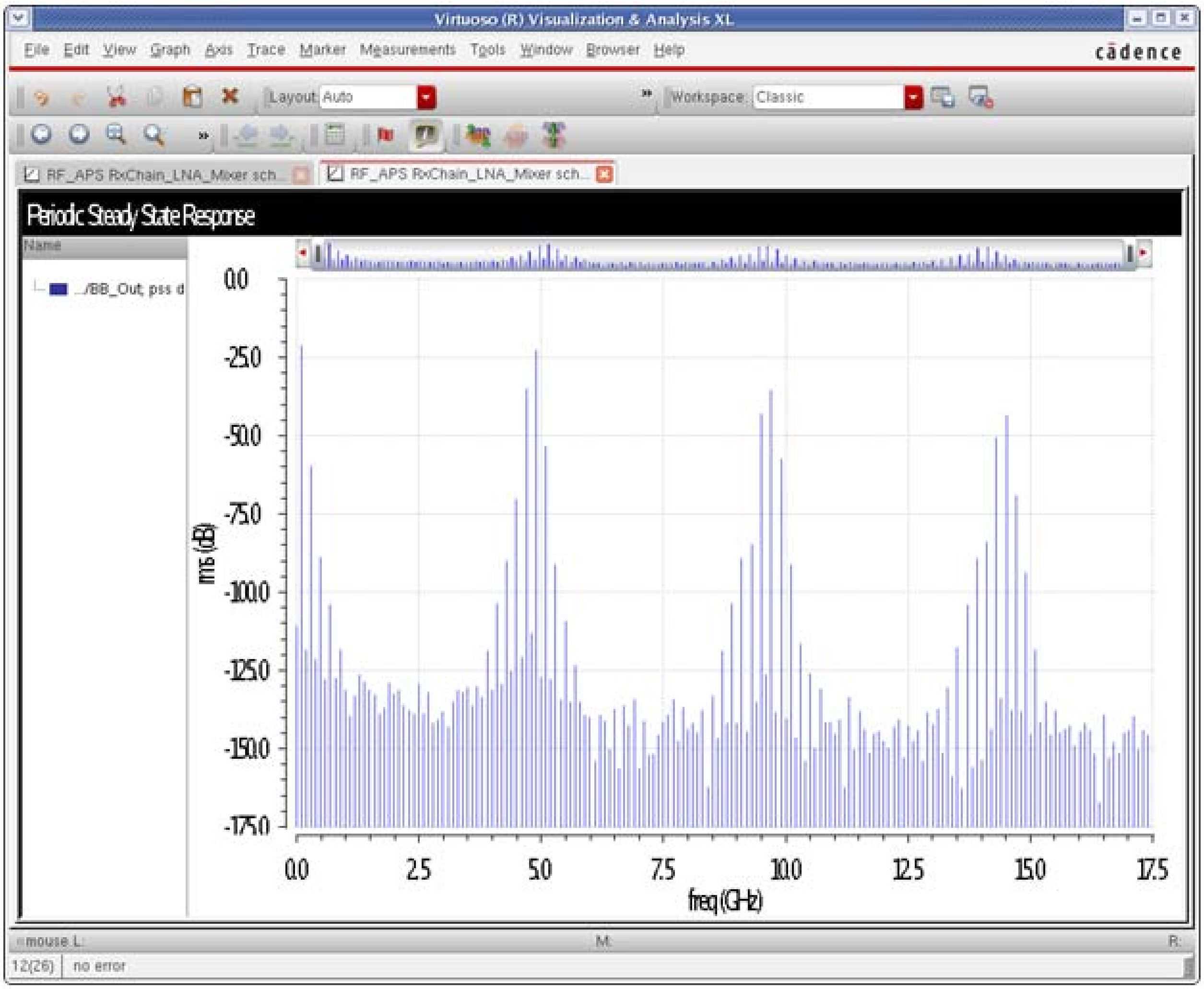

Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
1 fb1 fb1 2.5G Large PORT1
3 flo flo 2.4G Large
4 flo flo 2.4G Large
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
100M
Auto Calculate
Beat Period
Output harmonics
Number of harmonics
175
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

Remember that the frequency range is the input frequency range. In this example, the frequencies just above the first harmonic of the LO are swept.

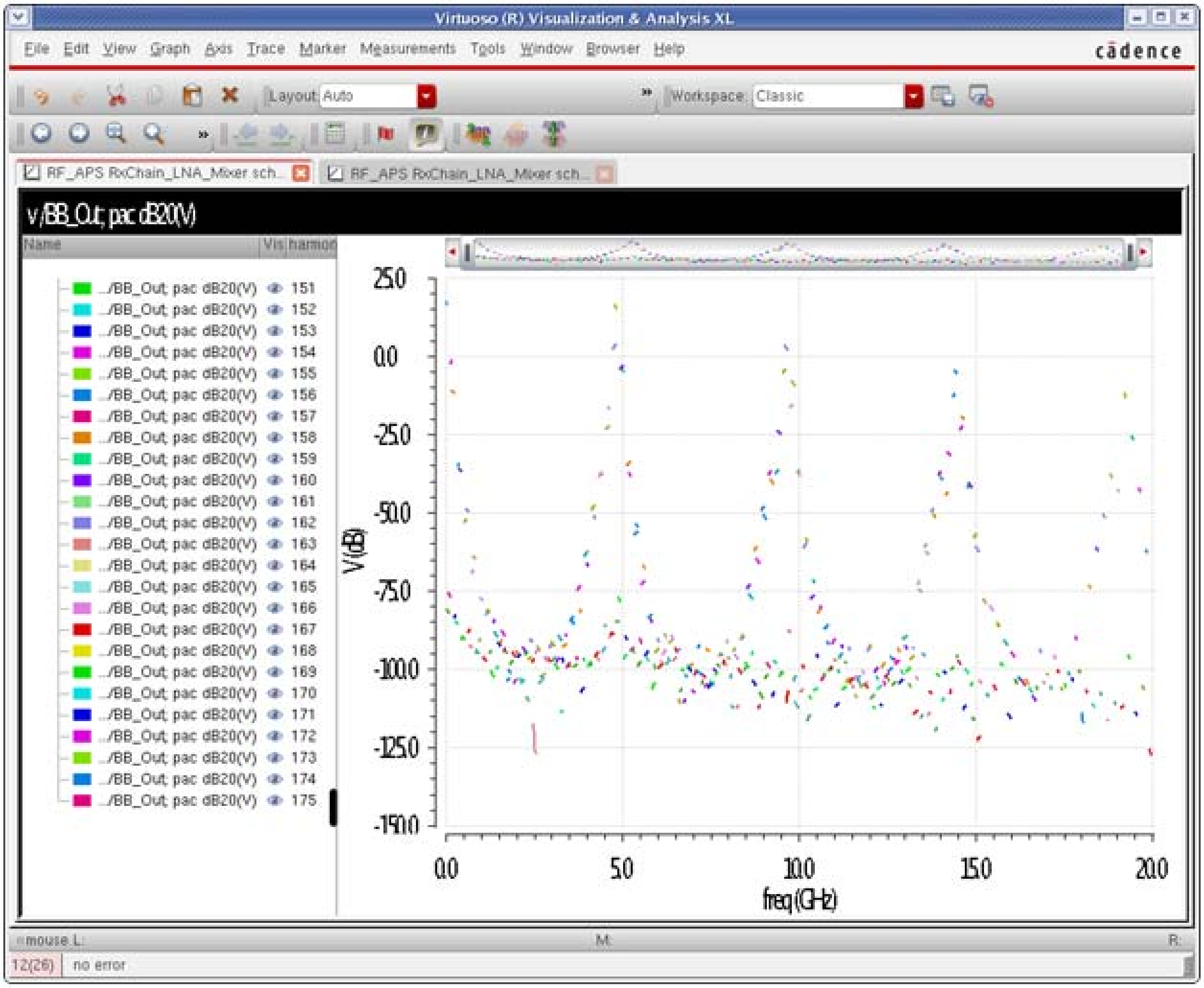
The pac form is set up to calculate the mixing products for all the harmonics that are calculated in the pss analysis. When shooting is used in the pss analysis, up to four times as many sidebands can be specified in pac as harmonics are specified in pss. When harmonic balance is used in pss, *Maximum sideband* can be set up to the number of harmonics in the pss analysis. Sidebands are the mixing products that are produced when the input signal mixes with the harmonics in the pss analysis.



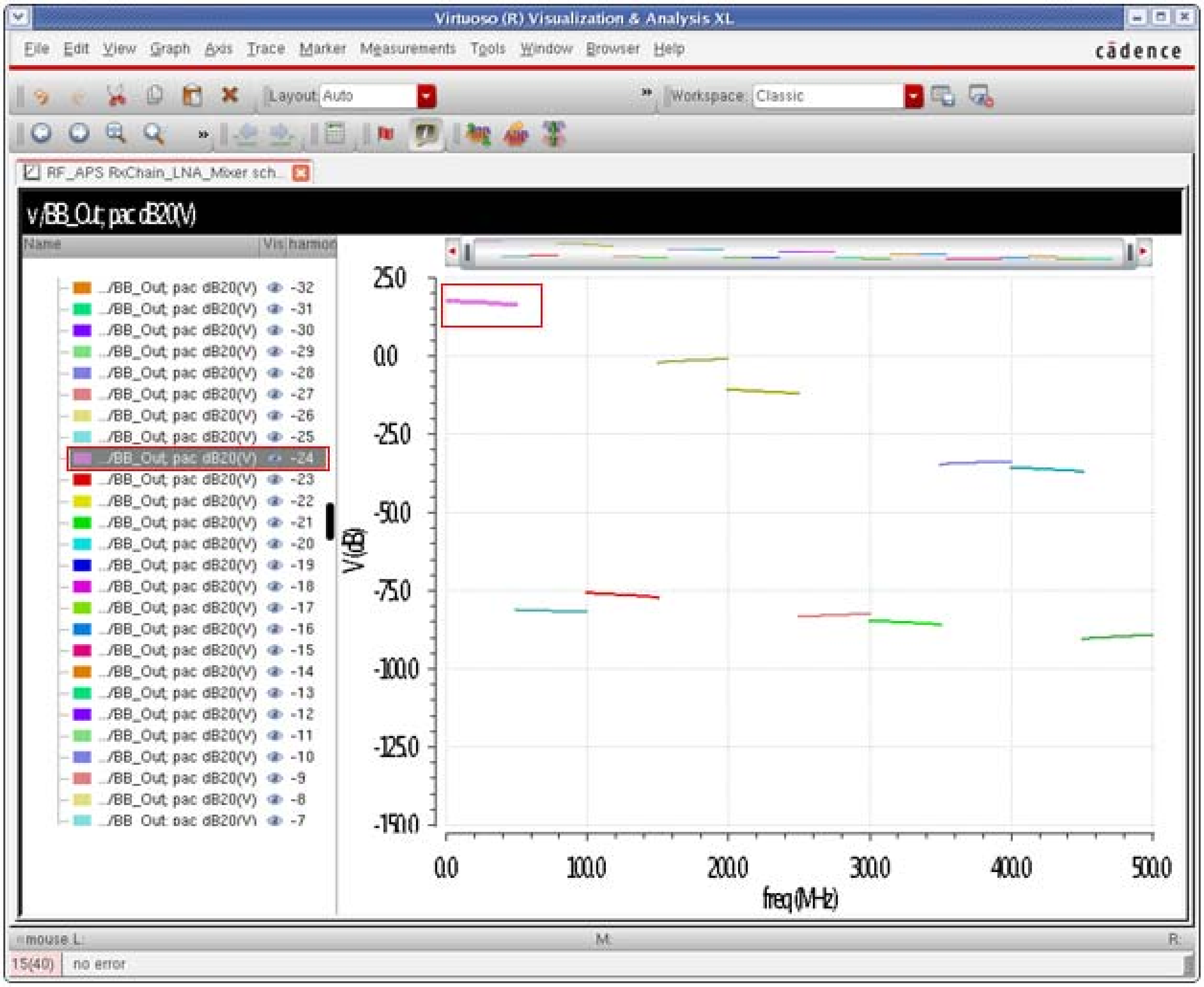
Now run the analysis. The spectral plot below from the pss analysis shows that because there were more input frequencies in the circuit, many more harmonics are produced by the circuit, which is as expected.



As shown below, because there are many more pss harmonics to mix with, many more mixing products (sidebands) are produced by pac.

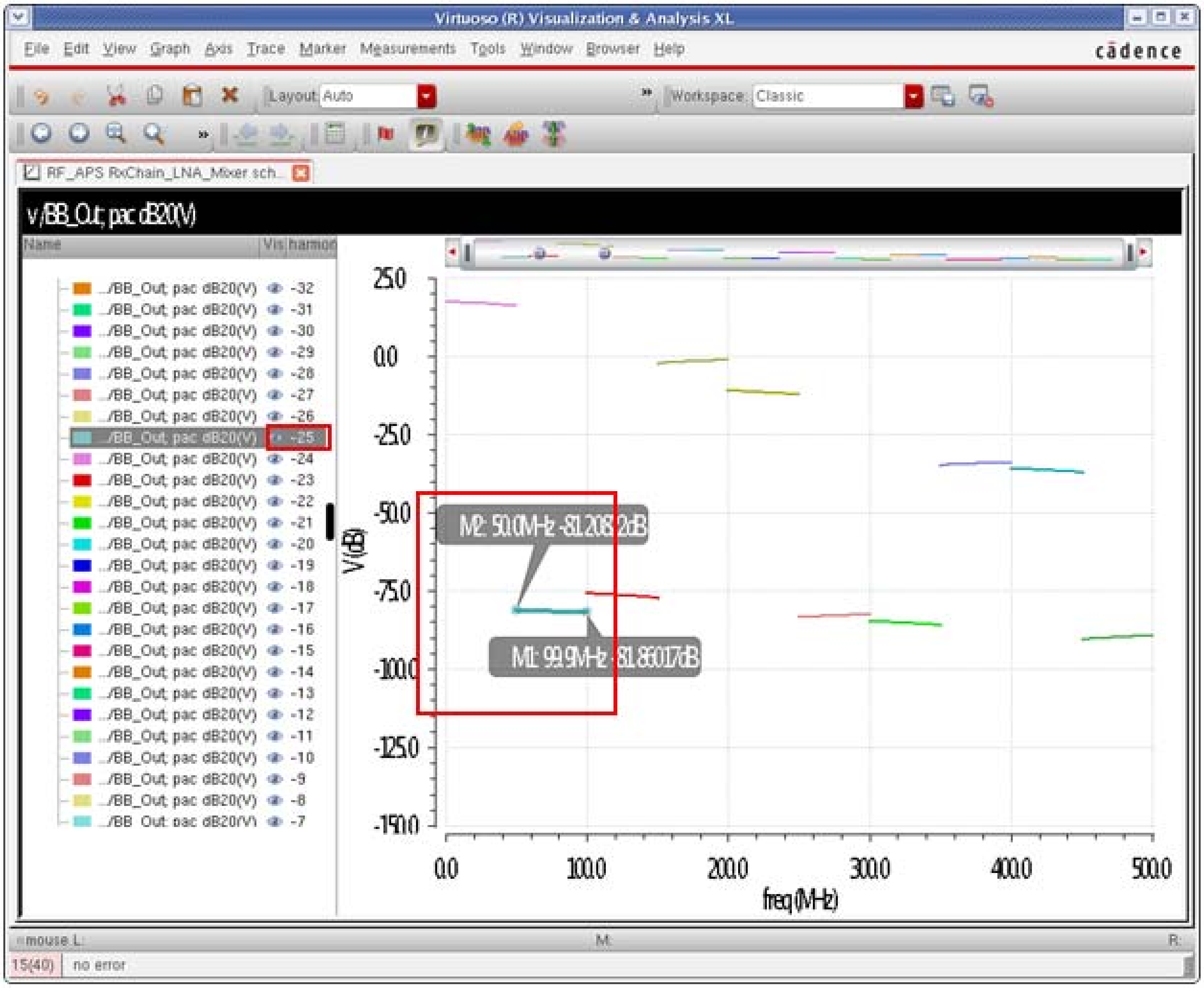

Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Layout: Auto
Workspace: Classic
RF_APS RxChain_LNA_Mixer sch...
RF_APS RxChain_LNA_Mixer sch...
v /BB_Out; pac dB20(V)
Name
Vis harmon
../BB_Out; pac dB20(V) 151
../BB_Out; pac dB20(V) 152
../BB_Out; pac dB20(V) 153
../BB_Out; pac dB20(V) 154
../BB_Out; pac dB20(V) 155
../BB_Out; pac dB20(V) 156
../BB_Out; pac dB20(V) 157
../BB_Out; pac dB20(V) 158
../BB_Out; pac dB20(V) 159
../BB_Out; pac dB20(V) 160
../BB_Out; pac dB20(V) 161
../BB_Out; pac dB20(V) 162
../BB_Out; pac dB20(V) 163
../BB_Out; pac dB20(V) 164
../BB_Out; pac dB20(V) 165
../BB_Out; pac dB20(V) 166
../BB_Out; pac dB20(V) 167
../BB_Out; pac dB20(V) 168
../BB_Out; pac dB20(V) 169
../BB_Out; pac dB20(V) 170
../BB_Out; pac dB20(V) 171
../BB_Out; pac dB20(V) 172
../BB_Out; pac dB20(V) 173
../BB_Out; pac dB20(V) 174
../BB_Out; pac dB20(V) 175
25.0
0.0
-25.0
-50.0
-75.0
-100.0
-125.0
-150.0
V (dB)
0.0
5.0
10.0
15.0
20.0
freq (GHz)
mouse L:
M:
R:
12(26)
no error

The next figure shows a zoomed-in area near zero frequency. Only the mixing products larger than -100 dB are shown.



Note that the index in the pac analysis is the frequency shift from the input frequency to the output frequency in multiples of the pss beat frequency. The output highlighted above mixes with the 24th harmonic of 100MHz, which is the LO signal in the circuit. This is the main output mixing product when RF mixes with LO.

Note that the RF input in the pac analysis at 2.4001G to 2.45G can also mix with 2.5G (The additional RF signal.) When the RF input is at 2.4001G the IF output is 99.9MHz when it mixes with the other RF tone at 2.5GHz. As the input frequency in the pac sweeps up, the output frequency sweeps down to 50MHz. Since it is mixing with 2.5GHz, the output frequency is downshifted by -25*100MHz, thus the sideband number shown in the legend at the left is -25. This is shown below.

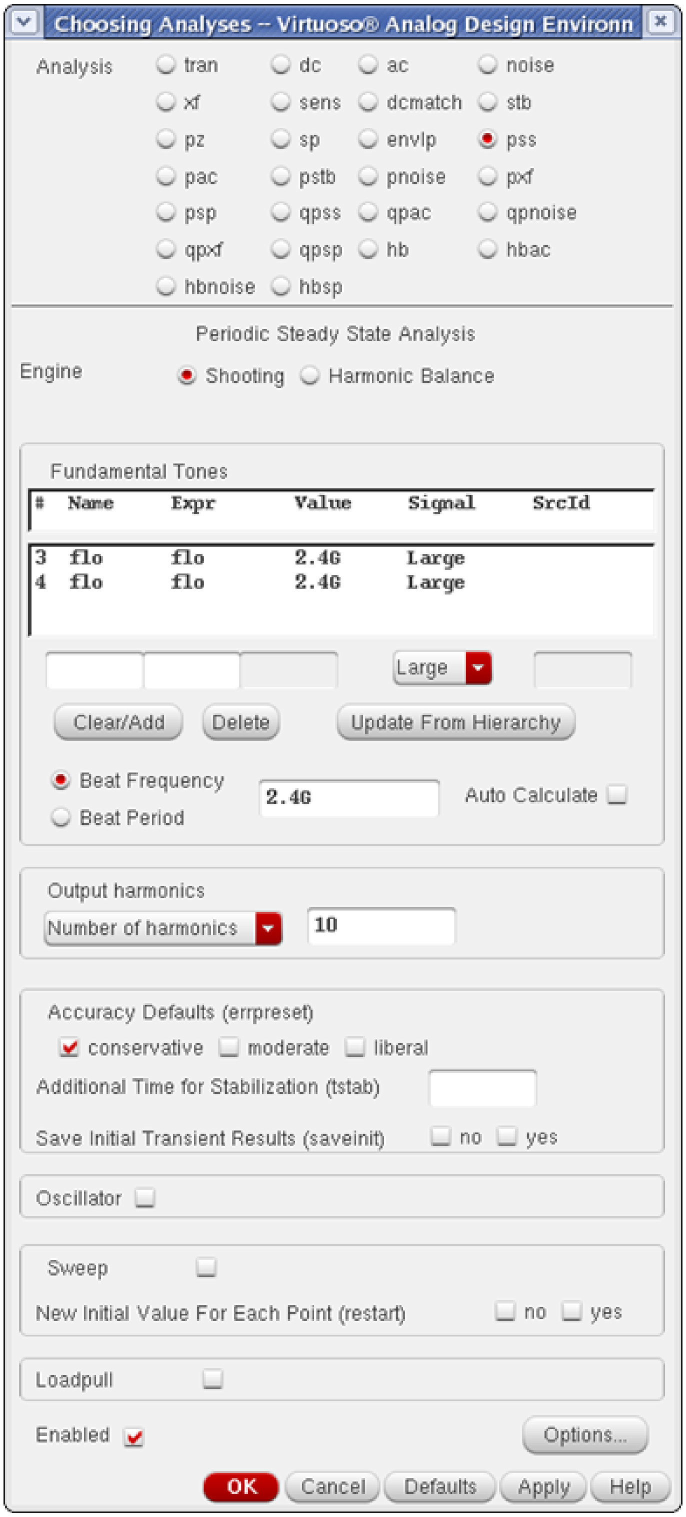


Rapid IP3/IP2

Rapid IP3 and Rapid IP2 are available in the ac and the pac *Choosing Analyses* forms. Rapid IP3 in the ac form is for measuring IP3 of amplifiers where there is no frequency translation, and Rapid IP3 in the pac form allows the measurement of IP3 where frequency translations are present. Rapid IP3 and IP2 measure small-signal IP3 and IP2.

The example below is for an LNA-mixer circuit.

1. First set up the pss form with just the LO applied, as shown below. For more information, see Overview of Periodic Steady-State (pss) Analysis on page 547. Both the shooting and harmonic balance pss engines are supported in pac.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
3 flo flo 2.4G Large
4 flo flo 2.4G Large
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
Beat Period
2.4G
Auto Calculate
Output harmonics
Number of harmonics
10
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

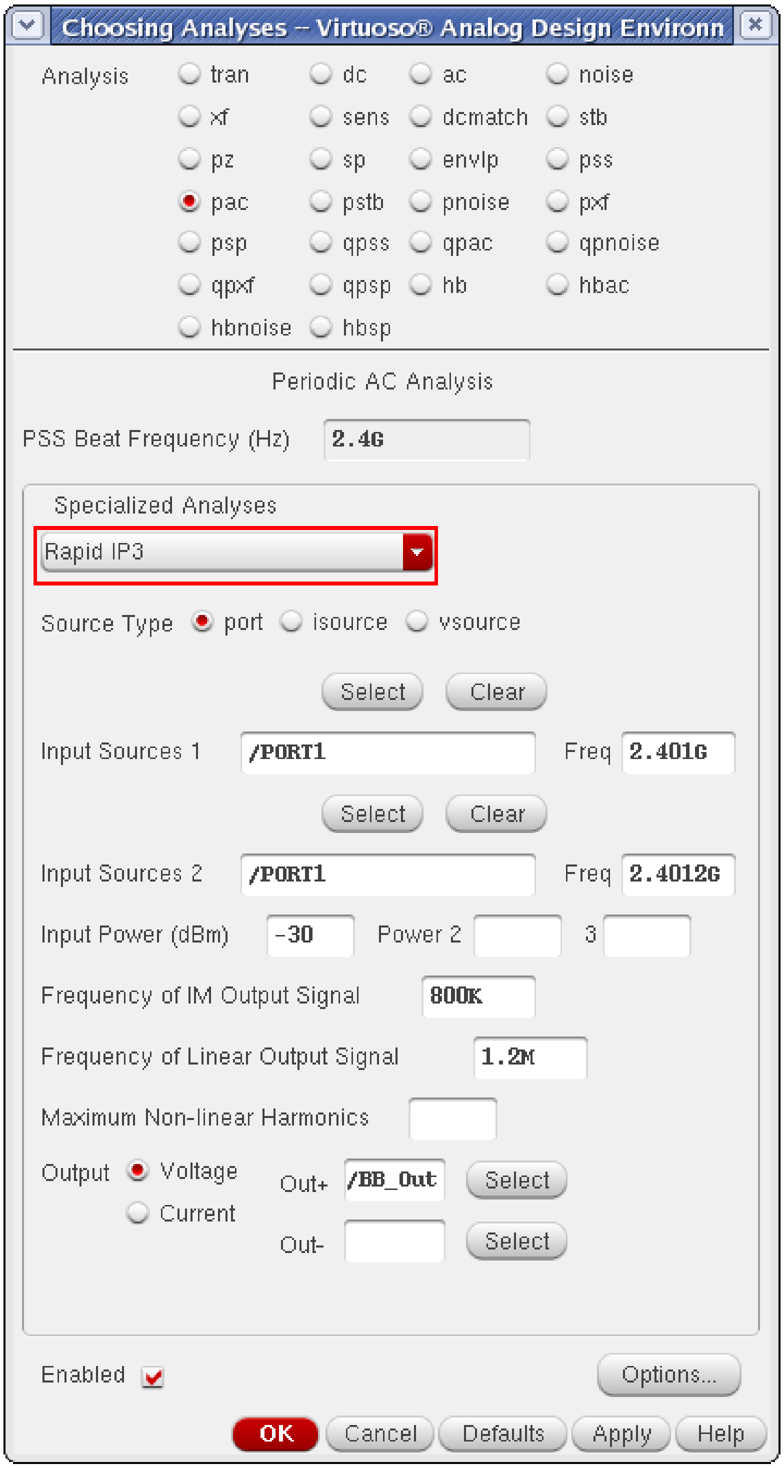
In the example below, the LO is at 2.4GHz, and the RF inputs are at 2.401GHz and 2.4012GHz. This makes the main mixing products at the output of the mixer at 1MHz and 1.2MHz. The third order intermods are at 800KHz and 1.4MHz. Because the frequencies are similar, either the 1 or 1.2 MHz products can be set as the linear frequency, and either the 800KHz or 1.4MHz term can be used for the third order frequency. Generally, the smallest amplitude linear term and the largest amplitude third order term are used for the IP3 measurement. If IP2 was desired, the second order product is at the difference frequency between the 2 RF inputs (200KHz in this example). The first order terms are the same as IP3.

2. Select the *pac* analysis. At the bottom of the form in the *Specialized Analysis* section, select *Rapid IP3* or *Rapid IP2*, depending on what you want to measure. The entries for the forms are identical, except for the frequency of the intermodulation product. For switched FET mixers, make sure that the psp model is used. Incorrect results are produced for BSIM3 and BSIM4 models due to a limitation in the model. If you need to measure IP3 with a BSIM3 or 4 model, use hb analysis with as large an amplitude as possible while maintaining the amplitude in the small-signal region.

3. Define the input port, input frequencies, input power, the frequencies of the linear and second or third order products, and the output node. An example for Rapid IP3 is shown below. Make sure that the input power is in the small-signal range. (At least 10dB smaller than the compression point)

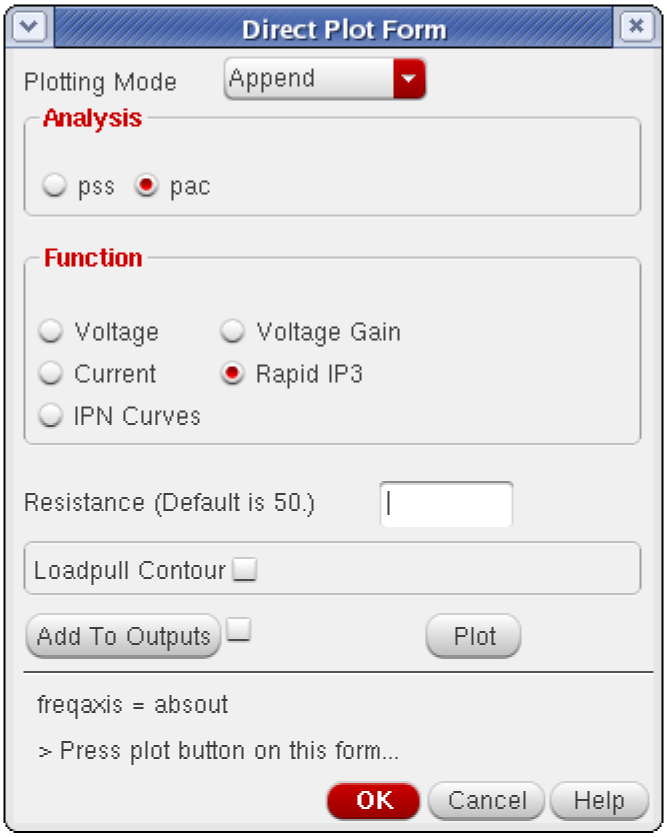
4. Select the output net in the circuit.

Note that for amplifiers, the pss analysis does not need to be set up. Just use the ac *Choosing Analyses* form which has the same appearance as the pac *Choosing Analyses* form shown below. The linear output frequency can be either of the two input frequencies. The third-order frequency can be the frequency spacing above or below the RF input frequencies.



5. Leave the *Maximum Non-linear Harmonics* field blank. The default value is appropriate for the Rapid IP2/IP3 calculation.

6. Run the simulation. When the simulation is complete, select *Results - Direct Plot - Main Form*.



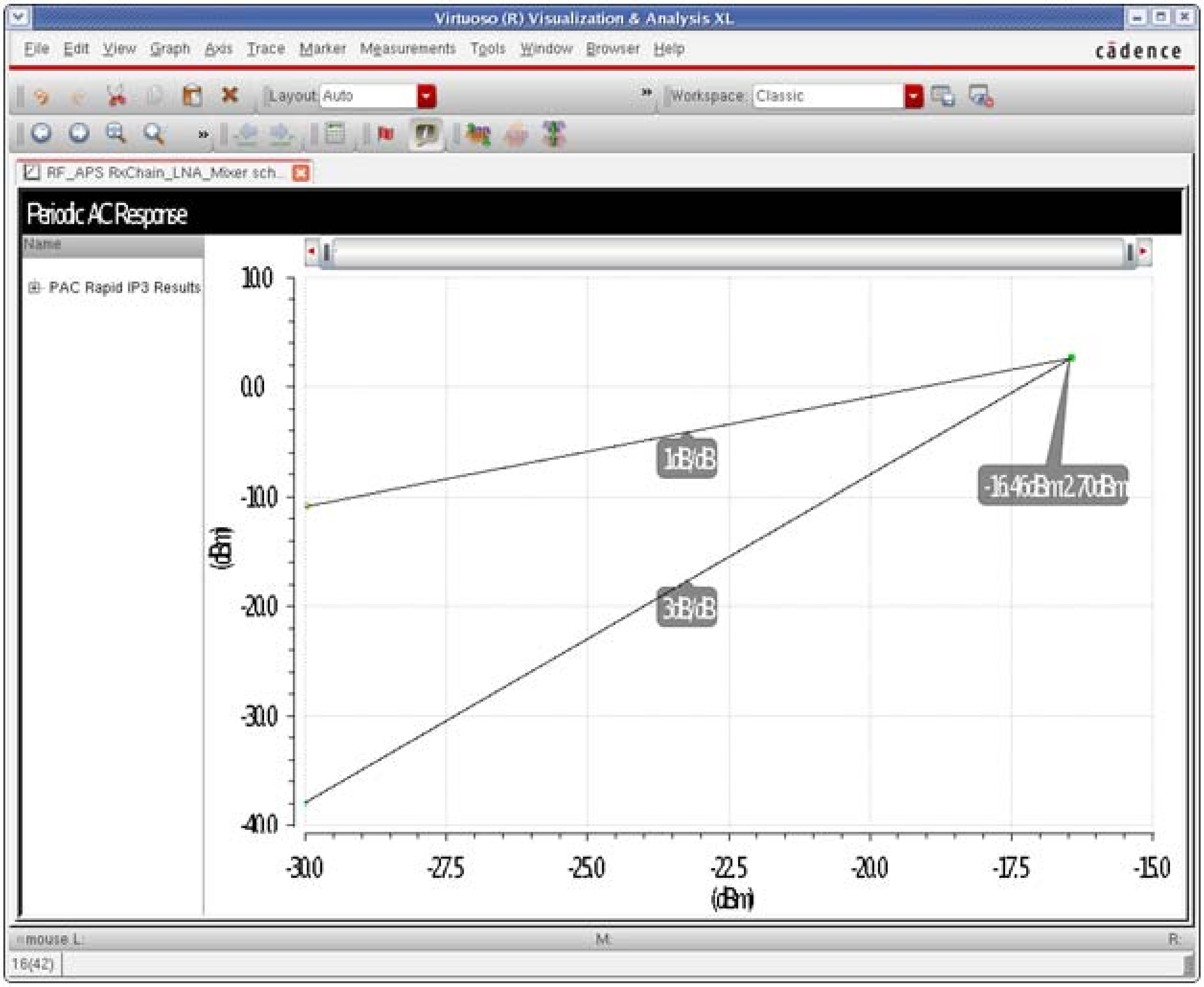
7. Select *pac* results.

8. Select *Rapid IP3*.

Rapid IP2 is shown if an IP2 analysis is run.

9. Click *Plot*.

The waveform is displayed. The marker shown at the intercept point shows the input and output-referred IP3 or IP2.



Compression Distortion Summary

The compression distortion summary is provided in order to get a relative measure of which components in a gain path contribute more or less compression. Because compression and IP3 are related mathematically, the compression distortion provides information about which components contribute to IP3.

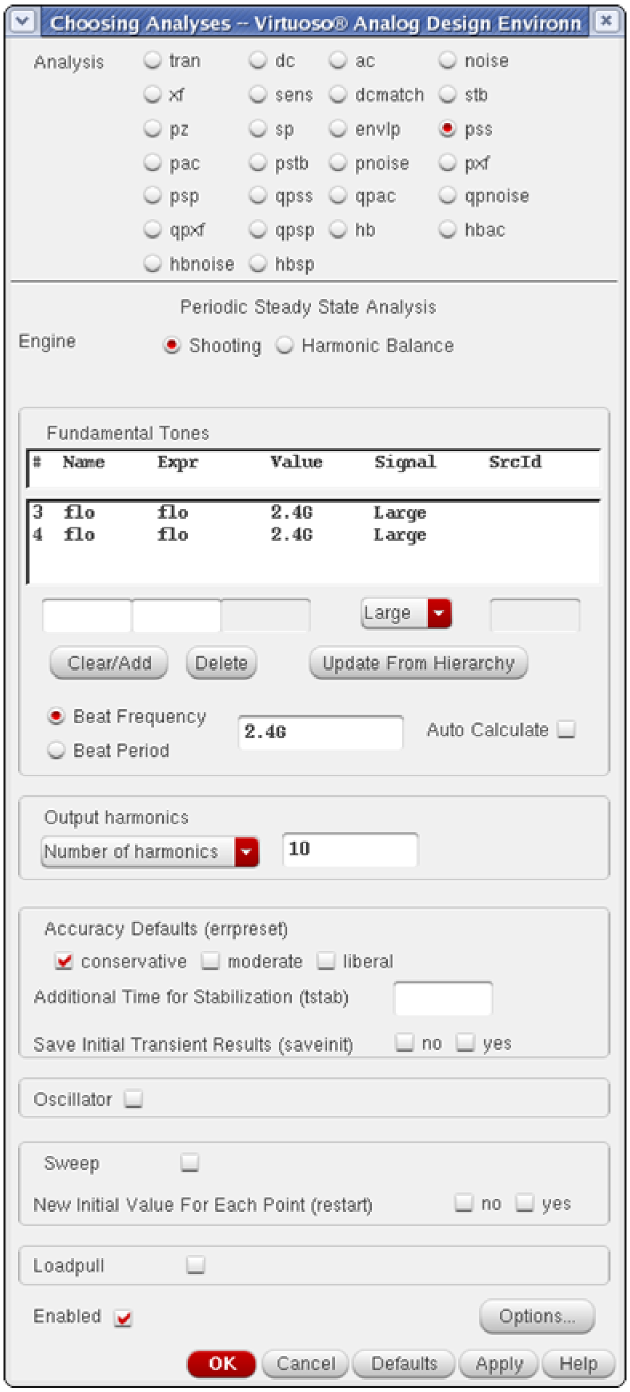
Note that the Compression Distortion Summary is available in the ac and the pac *Choosing Analyses* forms. Compression Distortion Summary in the ac form is for measuring which components add compression in amplifiers where there is no frequency translation, and Compression Distortion Summary in the pac form allows the measuring of components that contribute to compression where frequency translations are present in the circuit.

Note that for sampling circuits or for switched-capacitor filters, the sample clock(s) are applied in the pss, and the small-signal input Is applied in the pac analysis.

The *Contributor Instances* field is provided so that you can select which components in the circuit should be considered in the analysis. If the *Contributor Instances* field is left blank, all the nonlinearities in your circuit are run.

The following example is for an LNA-Mixer circuit.

1. First set up the pss analysis with just the LO or sample clock applied, as shown below. For more information, see Overview of Periodic Steady-State (pss) Analysis on page 547.



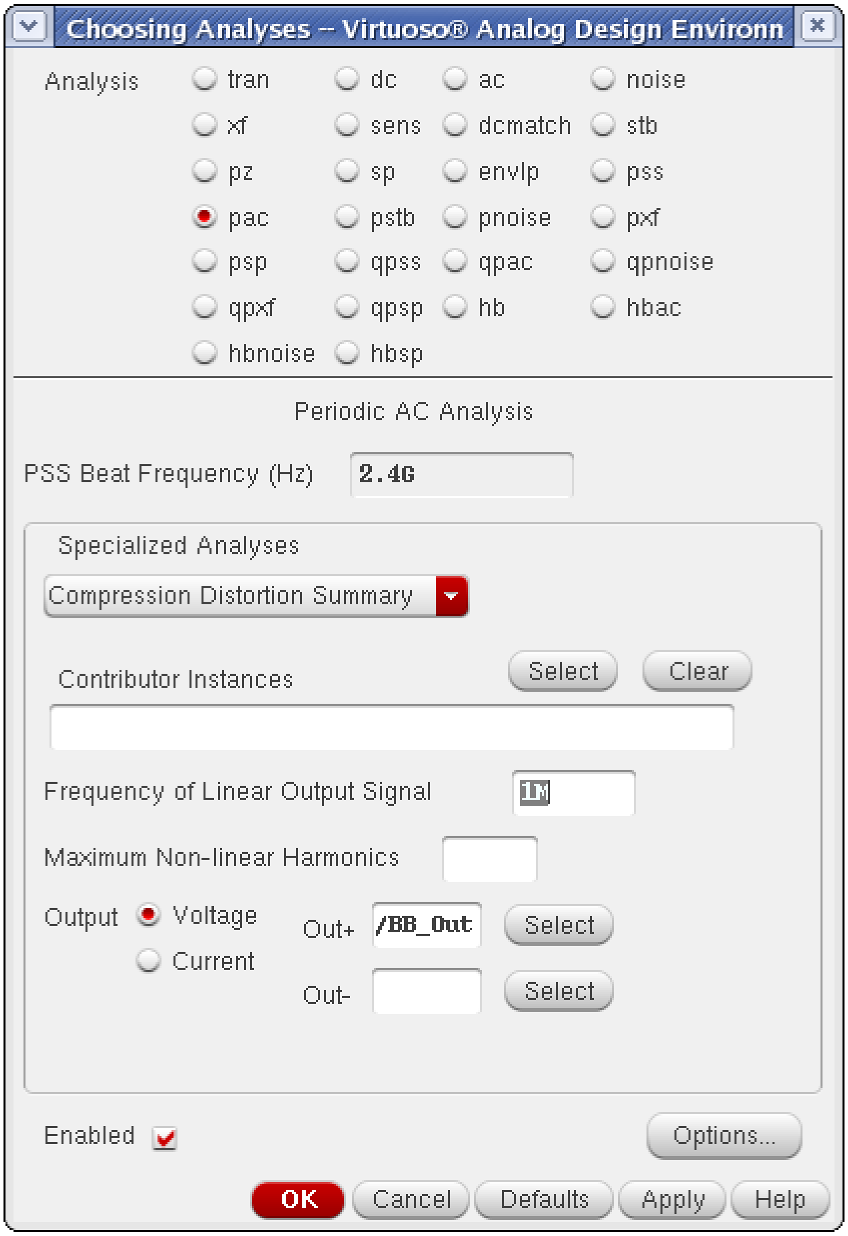
2. Now select pac analysis.

3. Select *Compression Distortion Summary* from the *Specialized Analyses* drop-down list.

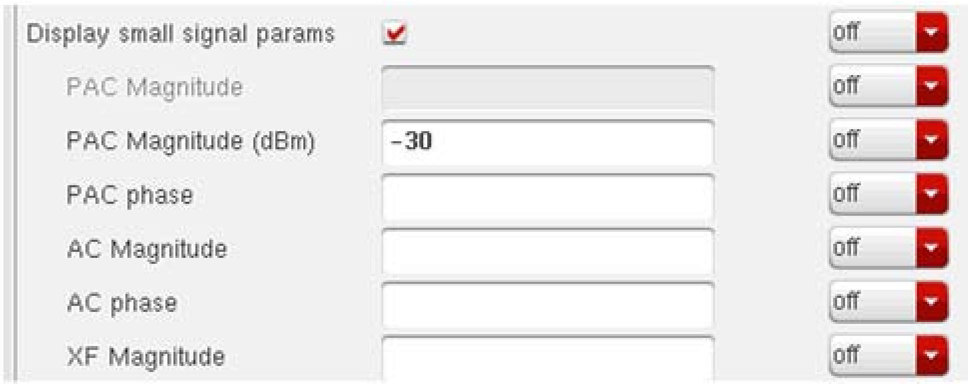
   The following notice appears to remind you to set the PAC amplitude parameter in the source in the schematic to a value somewhere in the small-signal range. This is generally at least 10dB smaller than the 1dB gain compression point.

4. If you leave the *Contributor Instances* field blank, all the nonlinear devices in your circuit will be run. If you want a subset, click *Select*, and select the instances you want on the schematic.
5. Specify the input frequency in the *Frequency of Input Source* field.
6. Specify the output frequency in the *Frequency of Linear Output Signal* field.
7. Leave the *Maximum Non-Linear Harmonics* field blank. The default value is *optimum* for the compression distortion measurement.
8. Select the output node in the circuit. When the *Out-* field is left blank, ADE automatically assigns this to the global ground node.

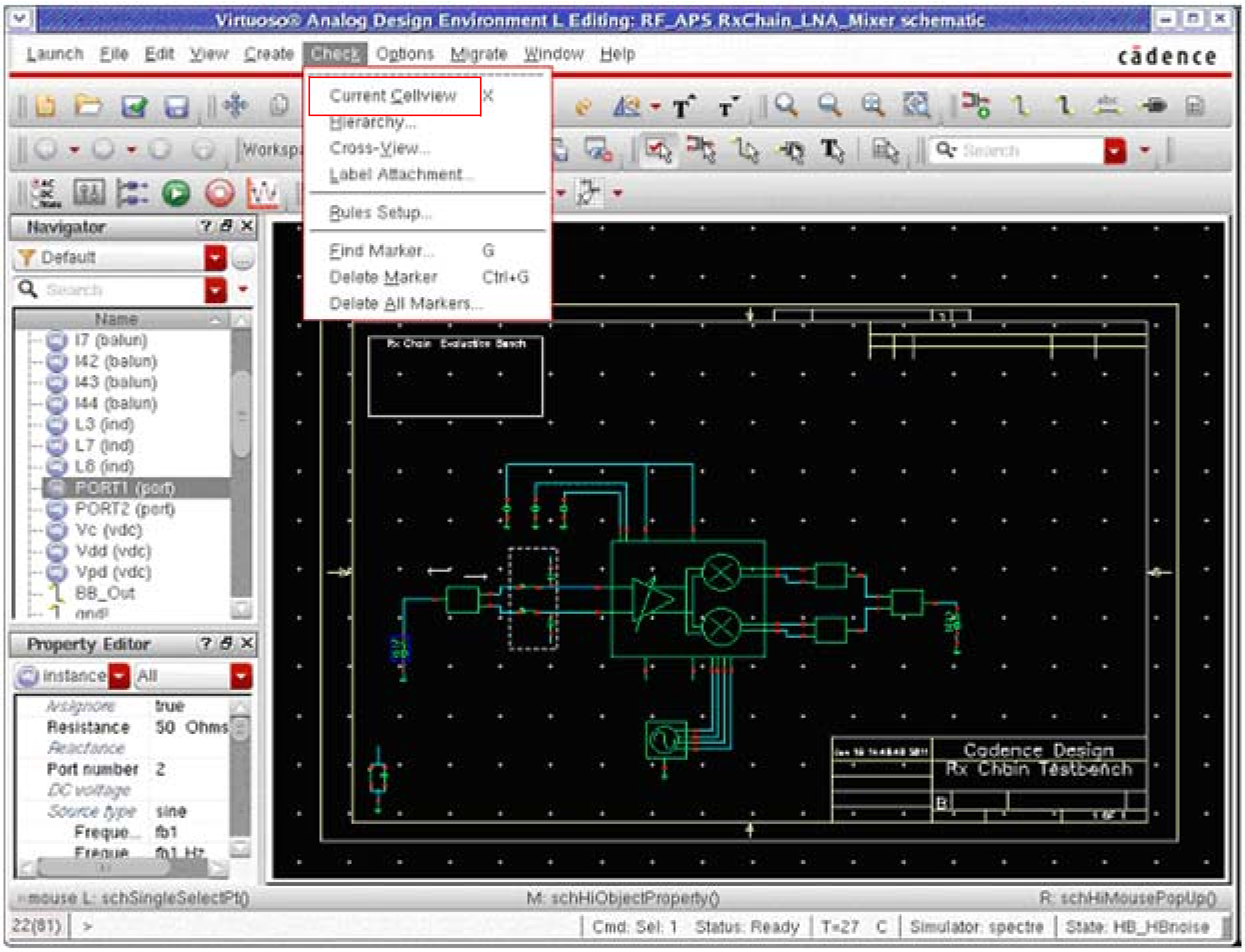
The completed form appears similar to the one shown below.



9. In the *Edit Object Properties* form, make sure that the *PAC magnitude (dBm)* is set in the small-signal range. In this example, it is set to -30Bm.



10. If you make changes to your schematic that you want for the distortion summary, but you do not want to save them to the circuit file, make the desired changes and then select *Check - Current Cellview* in the schematic window. Later, if you want to save them, just select *Check* and *Save*. If you do not want them, quit from the schematic window, and do not save the changes.

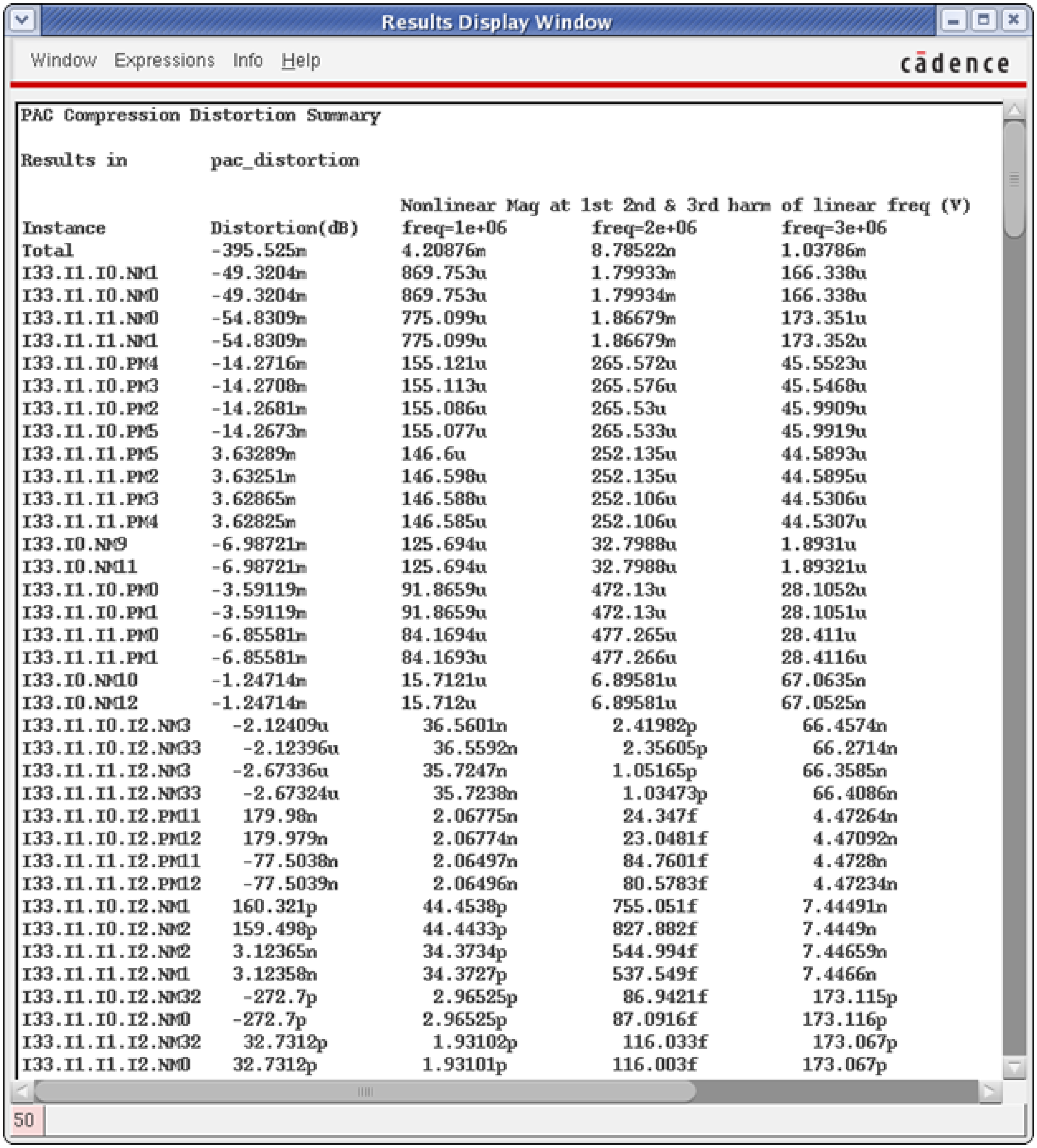


ADE will netlist from the checked view, so it will get the changes.

11. When you are done, run the simulation.

12. When the simulation completes, select *Results - Print - PAC Distortion Summary*.

The summary is displayed below.



PAC Compression Distortion Summary

Results in pac_distortion

		Nonlinear Mag at 1st 2nd & 3rd harm of linear freq (V)		
Instance	Distortion(dB)	freq=1e+06	freq=2e+06	freq=3e+06
Total	-395.525m	4.20876m	8.78522m	1.03786m
I33.I1.I0.NM1	-49.3204m	869.753u	1.79933m	166.338u
I33.I1.I0.NM0	-49.3204m	869.753u	1.79934m	166.338u
I33.I1.I1.NM0	-54.8309m	775.099u	1.86679m	173.351u
I33.I1.I1.NM1	-54.8309m	775.099u	1.86679m	173.352u
I33.I1.I0.PM4	-14.2716m	155.121u	265.572u	45.5523u
I33.I1.I0.PM3	-14.2708m	155.113u	265.576u	45.5468u
I33.I1.I0.PM2	-14.2681m	155.086u	265.53u	45.9909u
I33.I1.I0.PM5	-14.2673m	155.077u	265.533u	45.9919u
I33.I1.I1.PM5	3.63289m	146.6u	252.135u	44.5893u
I33.I1.I1.PM2	3.63251m	146.598u	252.135u	44.5895u
I33.I1.I1.PM3	3.62865m	146.588u	252.106u	44.5306u
I33.I1.I1.PM4	3.62825m	146.585u	252.106u	44.5307u
I33.I0.NM9	-6.98721m	125.694u	32.7988u	1.8931u
I33.I0.NM11	-6.98721m	125.694u	32.7988u	1.89321u
I33.I1.I0.PM0	-3.59119m	91.8659u	472.13u	28.1052u
I33.I1.I0.PM1	-3.59119m	91.8659u	472.13u	28.1051u
I33.I1.I1.PM0	-6.85581m	84.1694u	477.265u	28.411u
I33.I1.I1.PM1	-6.85581m	84.1693u	477.266u	28.4116u
I33.I0.NM10	-1.24714m	15.7121u	6.89581u	67.0635n
I33.I0.NM12	-1.24714m	15.712u	6.89581u	67.0525n
I33.I1.I0.I2.NM3	-2.12409u	36.5601n	2.41982p	66.4574n
I33.I1.I0.I2.NM33	-2.12396u	36.5592n	2.35605p	66.2714n
I33.I1.I1.I2.NM3	-2.67336u	35.7247n	1.05165p	66.3585n
I33.I1.I1.I2.NM33	-2.67324u	35.7238n	1.03473p	66.4086n
I33.I1.I0.I2.PM11	179.98n	2.06775n	24.347f	4.47264n
I33.I1.I0.I2.PM12	179.979n	2.06774n	23.0481f	4.47092n
I33.I1.I1.I2.PM11	-77.5038n	2.06497n	84.7601f	4.4728n
I33.I1.I1.I2.PM12	-77.5039n	2.06496n	80.5783f	4.47234n
I33.I1.I0.I2.NM1	160.321p	44.4538p	755.051f	7.44491n
I33.I1.I0.I2.NM2	159.498p	44.4433p	827.882f	7.4449n
I33.I1.I1.I2.NM2	3.12365n	34.3734p	544.994f	7.44659n
I33.I1.I1.I2.NM1	3.12358n	34.3727p	537.549f	7.4466n
I33.I1.I0.I2.NM32	-272.7p	2.96525p	86.9421f	173.115p
I33.I1.I0.I2.NM0	-272.7p	2.96525p	87.0916f	173.116p
I33.I1.I1.I2.NM32	32.7312p	1.93102p	116.033f	173.067p
I33.I1.I1.I2.NM0	32.7312p	1.93101p	116.003f	173.067p

- The first column is the instance name in the schematic.

- The second column is the gain with the nonlinearity of that single device divided by the ideal gain from the ac or pac analysis.

  Gain compression is shown as a negative number. Gain expansion is shown as a positive number.

- The third, fourth, and fifth columns are the amplitudes of the first, second, and third harmonic of the linear output frequency with just the nonlinearity of that single device taken into account.

Sampled Circuits and Switched-capacitor Filters

For sampled circuits or for switched-capacitor circuits, always use shooting pss because the clocking waveforms have rapid transitions. Shooting captures these transitions in a natural and accurate way.

In pac, set up the compression distortion form just as you would for the mixer example above. Note that the linear output frequency is the input frequency. Otherwise, the setup is exactly the same as the setup shown above.

IM2 Distortion Summary

The steps for the IM2 distortion summary are very similar to the Compression Distortion Summary in the previous section. The differences are noted below.

- The IM2 distortion summary provides the output voltage (magnitude, real, and imaginary parts) of the second order distortion signal for each component in the schematic by itself.
- The IM2 distortion summary does not provide a selection mechanism for the devices in the circuit. All the devices are run in the IM2 distortion summary.
- The output format is slightly different. The first column is the instance name in the schematic. The second column is the magnitude of the IM2 product that is produced by

that single component in volts. The third column has the real and imaginary parts of the voltage. A sample is shown below.


```
PAC IM2 Distortion Summary

Results in          pac_im2distortion

Instance            Distortion(V)[Magnitude Complex]
Total               56.8138n     complex(56.8132n, 279.356p)
/I33/I1/I0/PM4      1.45004m     complex(1.45003m, -2.96617u)
/I33/I1/I0/PM3      1.45003m     complex(1.45003m, -2.96618u)
/I33/I1/I0/PM2      1.45002m     complex(-1.45001m, 2.96629u)
/I33/I1/I0/PM5      1.45001m     complex(-1.45001m, 2.96628u)
/I33/I1/I1/PM4      1.43768m     complex(1.43768m, -2.95021u)
/I33/I1/I1/PM3      1.43768m     complex(1.43767m, -2.95019u)
/I33/I1/I1/PM2      1.43767m     complex(-1.43767m, 2.95025u)
/I33/I1/I1/PM5      1.43767m     complex(-1.43767m, 2.95024u)
/I33/I1/I1/I2/NM3     144.198n     complex(-97.8239n, -105.941n)
/I33/I1/I1/I2/NM33     144.197n     complex(97.8234n, 105.94n)
/I33/I1/I0/I2/NM3     140.951n     complex(-96.7777n, -102.476n)
/I33/I1/I0/I2/NM33     140.951n     complex(96.7775n, 102.476n)
/I33/I1/I1/I2/PM11     10.0589n     complex(-7.44012n, -6.7695n)
/I33/I1/I1/I2/PM12     10.0589n     complex(7.44012n, 6.76949n)
/I33/I1/I0/I2/PM11     10.0337n     complex(-7.6139n, -6.53482n)
/I33/I1/I0/I2/PM12     10.0337n     complex(7.6139n, 6.53481n)
/I33/I1/I1/PM1      5.07326n     complex(-5.07326n, 4.89652p)
/I33/I1/I0/PM0      4.36043n     complex(4.36039n, 18.022p)
/I33/I1/I1/PM0      3.67355n     complex(3.67349n, -21.7191p)
/I33/I1/I1/NM0      3.00179n     complex(-3.00172n, 20.0364p)
/I33/I1/I1/NM1      2.61043n     complex(2.61009n, -41.8165p)
/I33/I1/I0/NM0      1.95443n     complex(-1.95432n, -20.0203p)
/I33/I1/I0/NM1      1.64469n     complex(1.64469n, 387.776f)
/I33/I1/I0/PM1      753.377p     complex(753.304p, 10.4897p)
/I33/I0/NM9         266.846p     complex(-266.8p, -4.96938p)
/I33/I1/I1/I2/NM2     226.574p     complex(-212.697p, -78.0755p)
/I33/I1/I1/I2/NM1     226.573p     complex(212.694p, 78.079p)
/I33/I0/NM12        165.221p     complex(165.216p, 1.39871p)
/I33/I0/NM10        126.074p     complex(125.993p, -4.53295p)
```

Sampled Circuits and Switched-capacitor Filters

For sampled circuits or for switched-capacitor circuits, always use shooting pss because the clocking waveforms have rapid transitions. Shooting captures these transitions in a natural and accurate way.

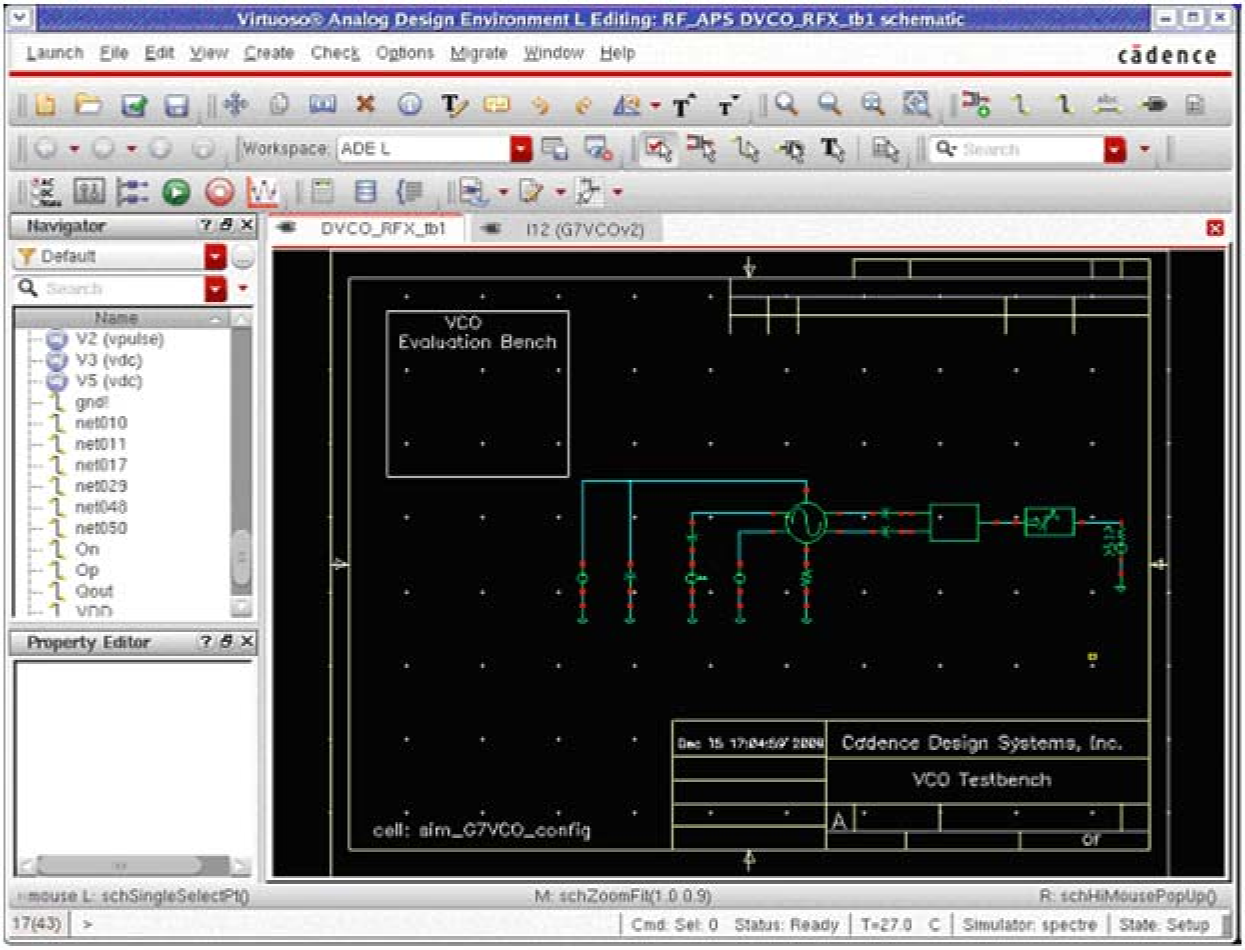
In pac, set up the IP2 distortion form just as you would for the mixer example above. Note that the linear output frequency is one of the input frequencies and the intermodulation frequency is the frequency spacing. Otherwise, the setup is exactly the same as the setup shown above.

Modulated PAC Analysis

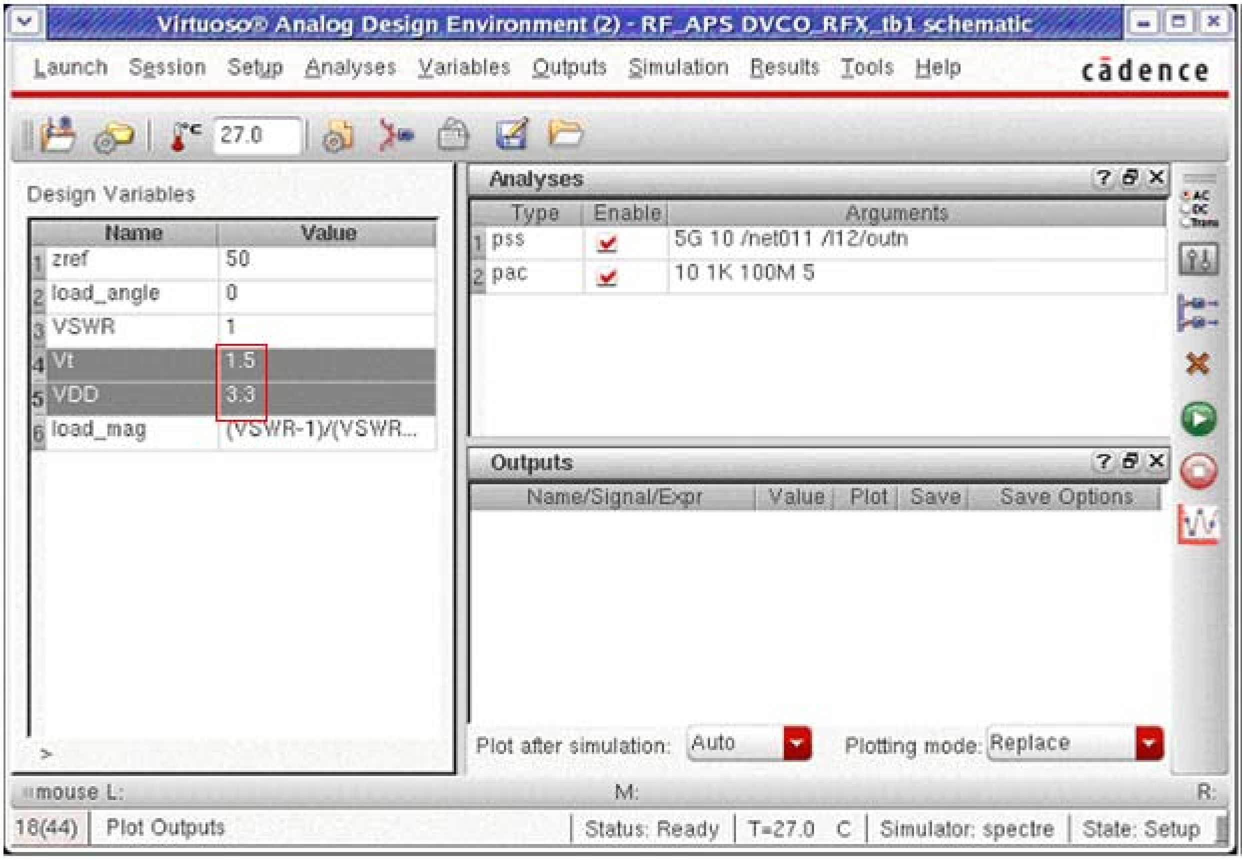
Modulated pac analysis is usually applied to oscillators to measure AM-to-PM conversion from low-frequencies on the power supply (ripple) to the output near the first harmonic. It can also be applied to digital circuits for the same purpose. Although the shape of the transfer function is different, the steps are the same for both applications.

In the circuit, make sure that any PAC magnitude terms are removed on the input source (if there is one), and set the PAC Magnitude to 1 volt in the source that is used as the power supply. If you do not want to save the change permanently, just select *Check - Current View* in the schematic.

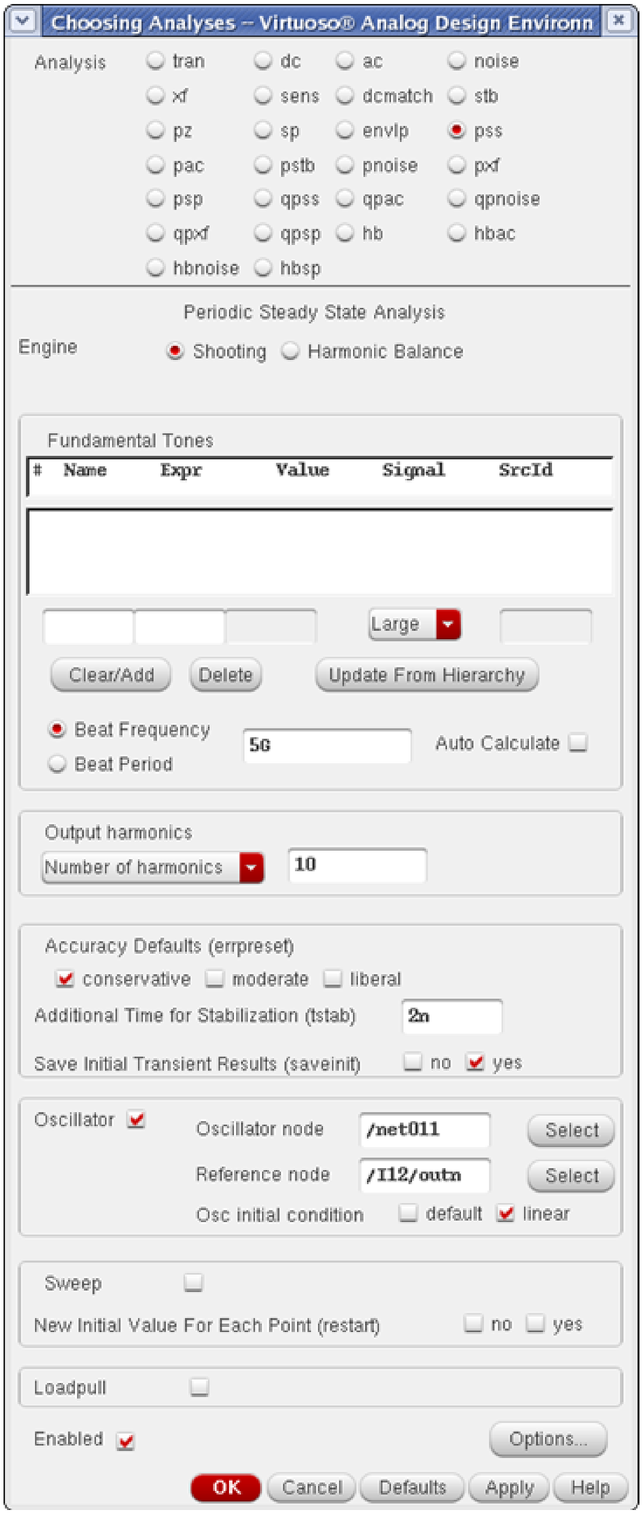
This example is for an oscillator. The circuit is shown below:



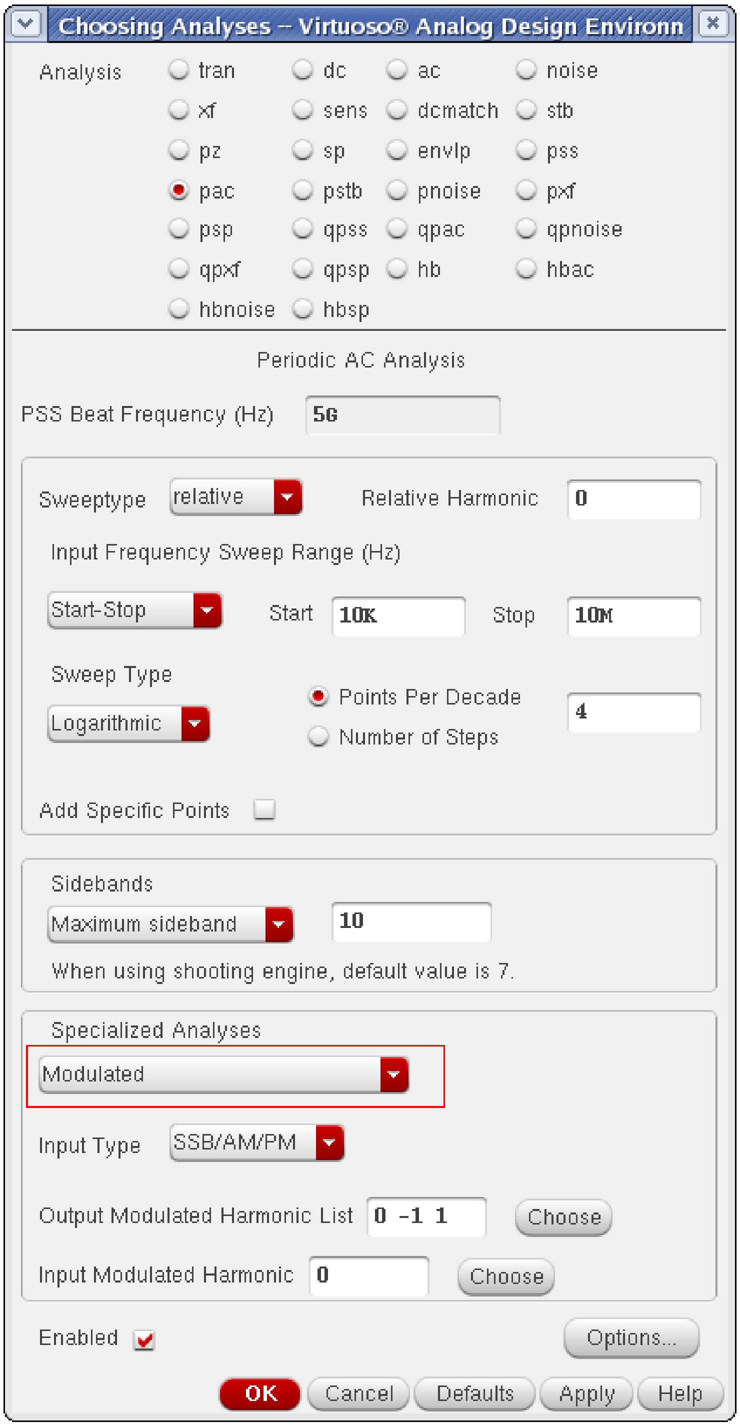
In the ADE, set values for the power supply voltage and for the tuning voltage.



Set up the pss analysis. For more information on pss, see Overview of Periodic Steady-State (pss) Analysis on page 547.



In the pac *Choosing Analyses* form, first select *Modulated* in the *Specialized Analyses* section, as shown below.

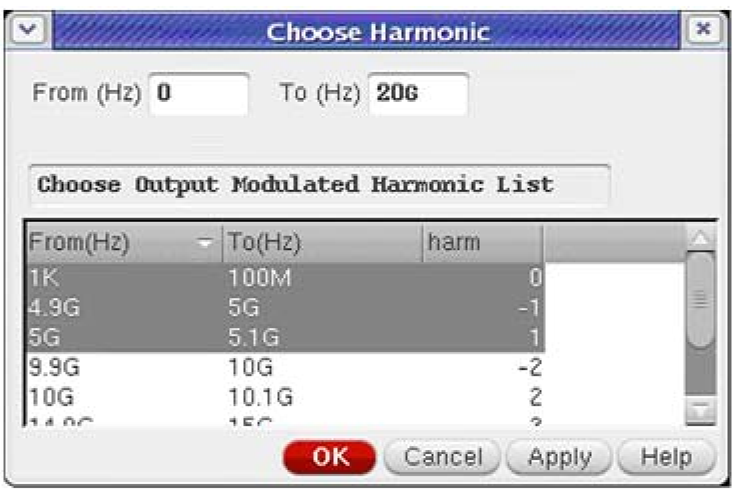


Note that selecting *Modulated* causes the *Sweep type* to be set to *relative*. Since the gain from power supply ripple to the output is desired, the relative harmonic number is zero, and then the desired frequency range is entered. If the shooting engine is used, maximum sideband can be up to 4 times the number of harmonics in the pss analysis harmonics as shown above. Setting the *Sweeptype* to *relative* allows a frequency shift in multiples of the pss beat frequency to be applied to the input frequency. In this example, when the relative harmonic is zero, the input frequency range is not shifted at all in frequency.

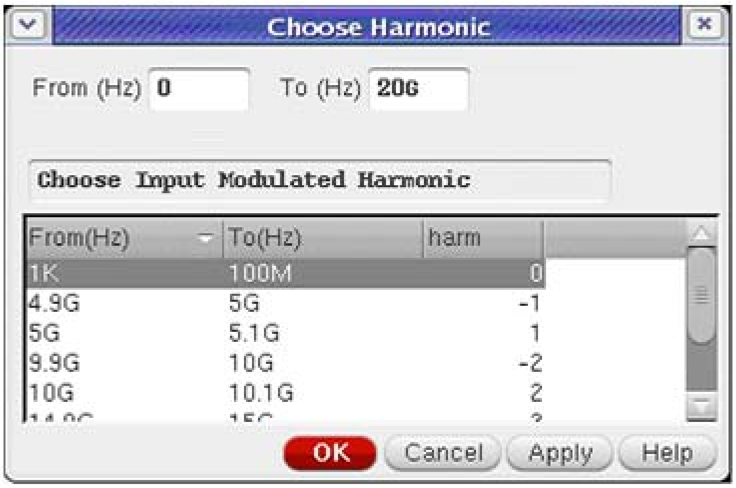
Maximum sideband needs to be set larger than is needed for measuring the output frequency. In this case, the input signal is the ripple at low frequency, and the output is near the first harmonic. Because of this, *Maximum sideband* needs to be one or larger. In this case, the setting of *Maximum sideband* is set to the same number as the maximum harmonics setting in pss.

Input Type is set to *SSB/AM/PM* so that all the combinations of SSB, AM, and PM to SSB, AM, or PM can be calculated.

Next, the output frequency ranges to be calculated can be set. Note that a list of harmonics can be entered. The easiest way is to click the *Choose* button to the right of the field, and select the desired frequencies from a list in the *Choose Harmonic* window, as shown below.

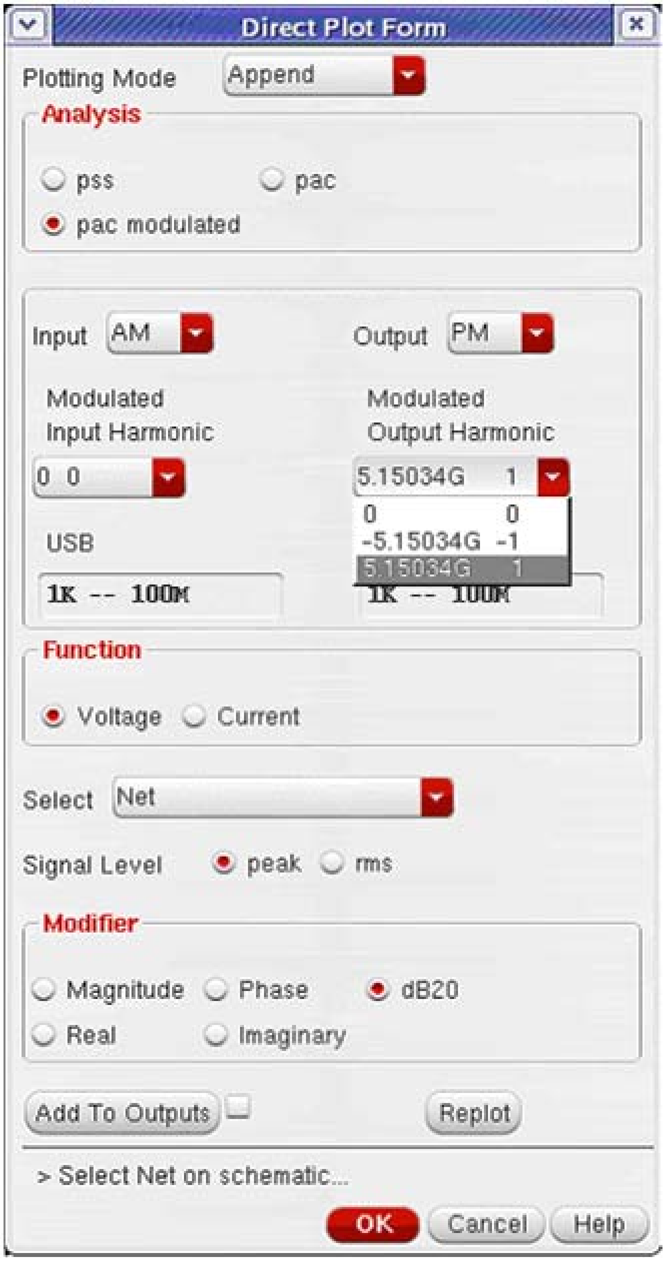


The same process is used to set the input frequency, but only one frequency range is allowed.

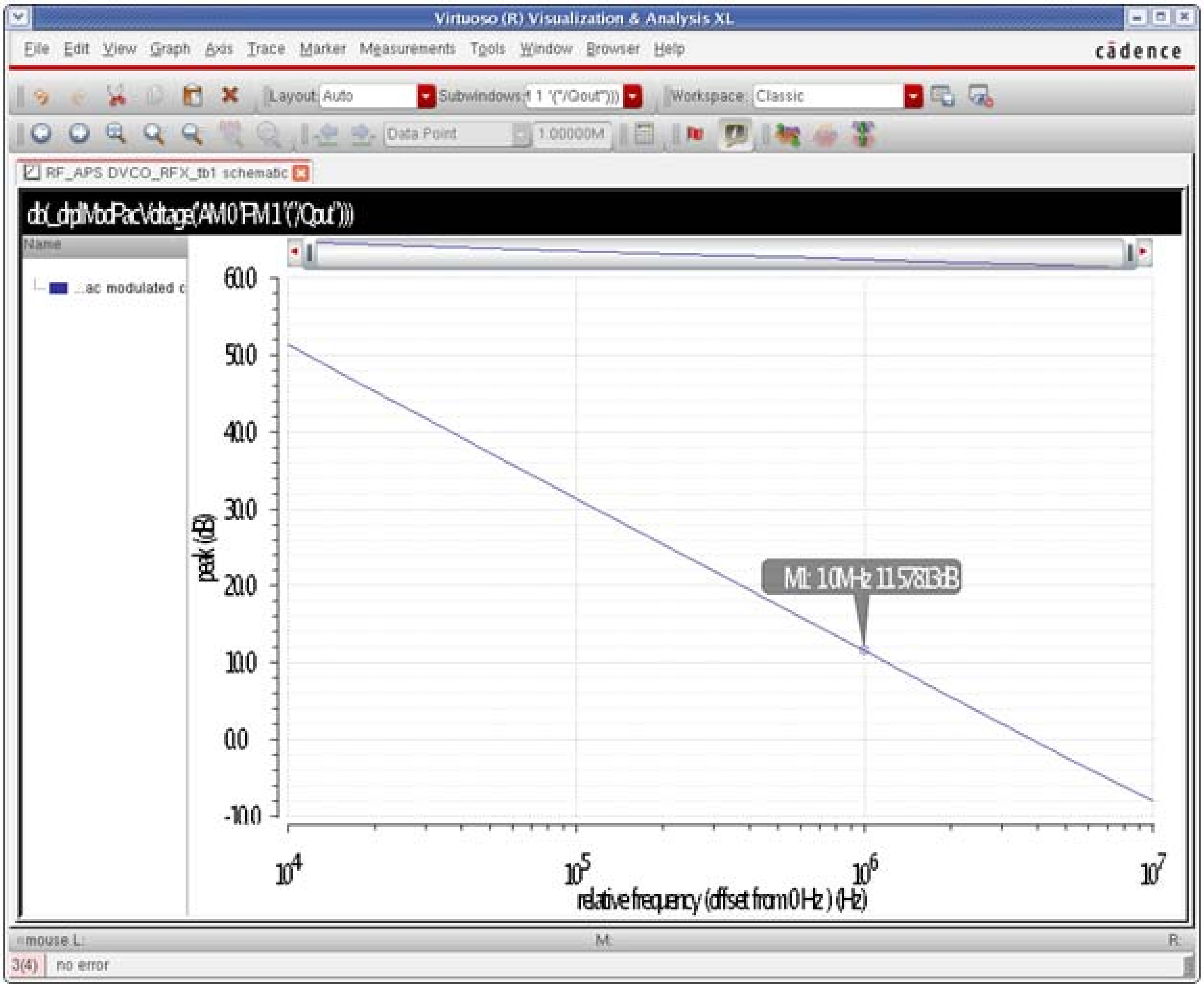


Once the analysis has been set up, run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form*.

The input defaults to *AM*. Select *PM* for the output. Since many output frequencies were calculated, you need to select the one you want to plot. In this example, the output just above the first harmonic is selected. Once this is done, select a net in the schematic.

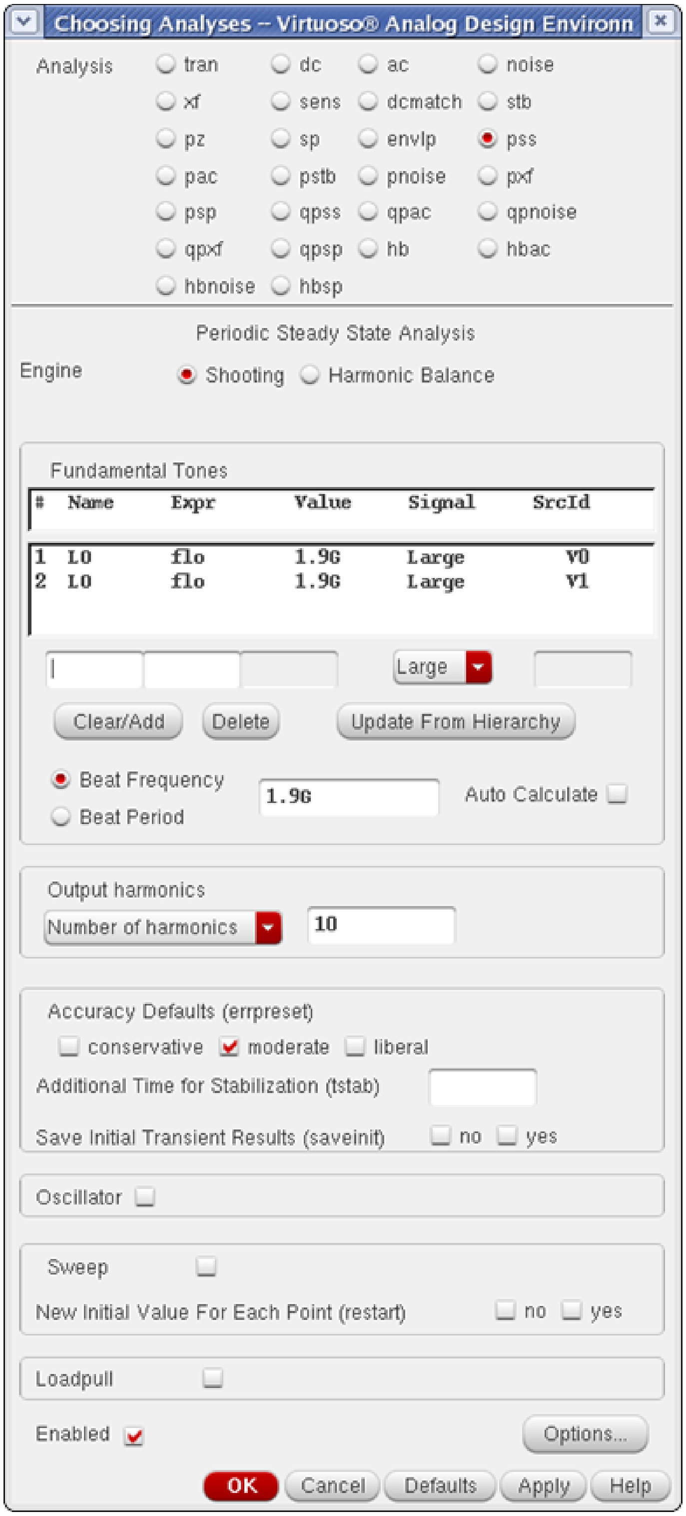


The output is displayed. To make a measurement at a specific frequency, position a marker. The measurement is in radians per volt of input on the supply that has PAC magnitude set.



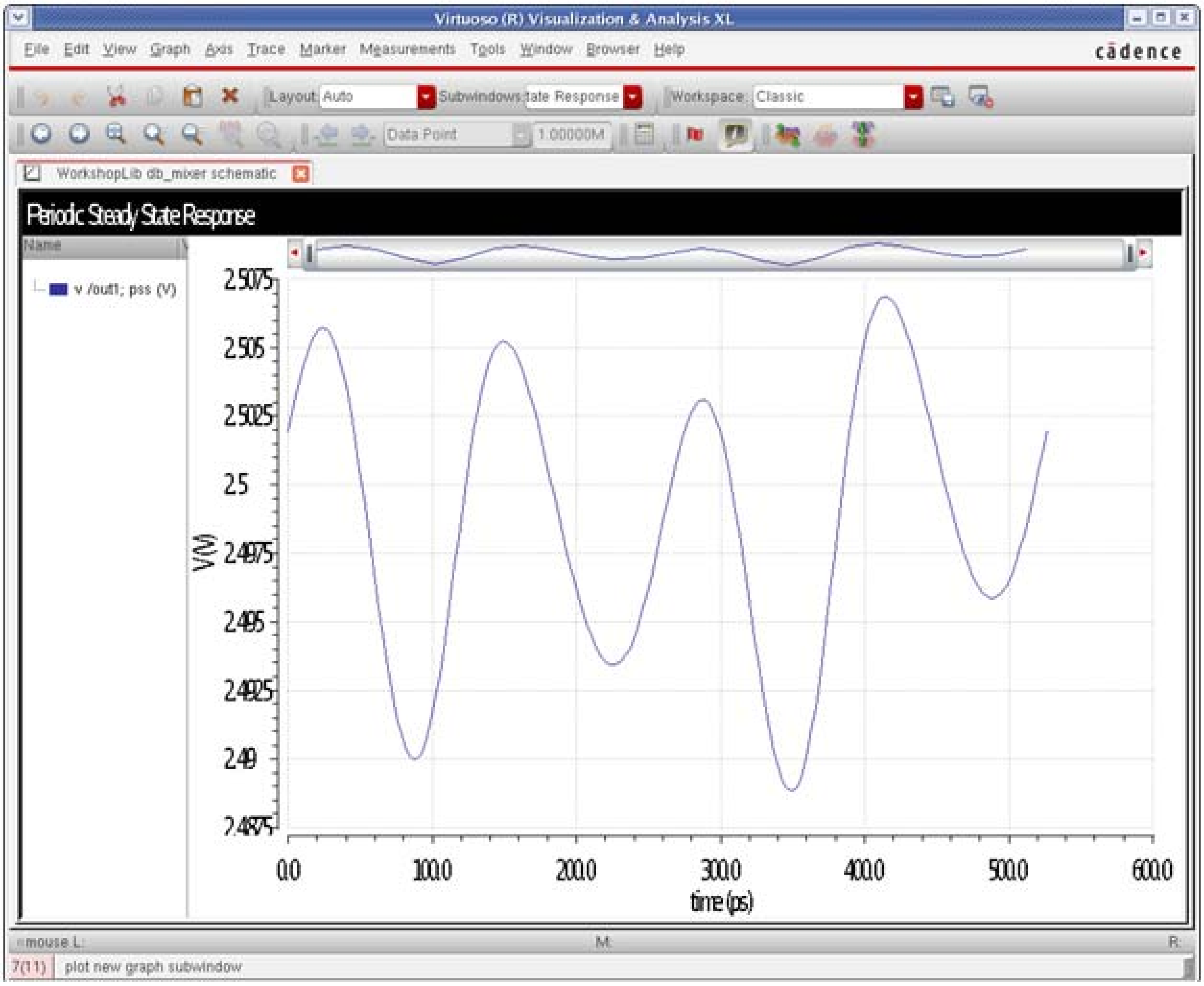
Sampled PAC Analysis

Pac sampled is for measuring the conversion gain at a specific instantaneous time in the pss waveform. The first step is to determine an appropriate threshold for the measurement by setting up the pss analysis, as shown below. Shooting is shown below, but this is also available in harmonic balance. For harmonic balance, the waveform is an ifft of the harmonics.



Now run the analysis. When the analysis completes, select *Results - Direct Plot - Main Form* in the ADE and plot the time-domain waveform on the output net, as shown below.

The waveform is shown below.



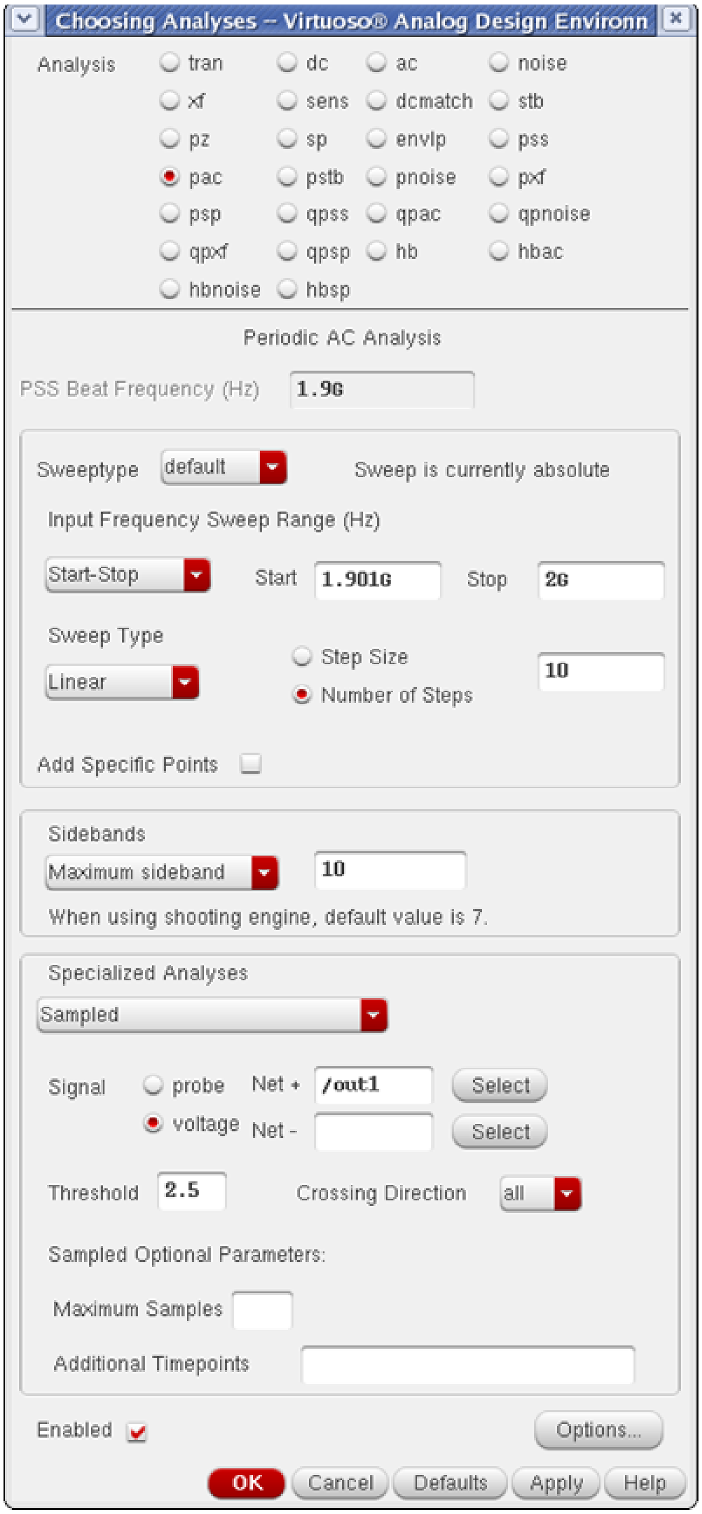
Note that 2.5 volts is a value near the middle of the waveform. This is chosen as the threshold for the calculation. Pick a threshold that is appropriate for your system.

Now set up the pac *Choosing Analyses* form.

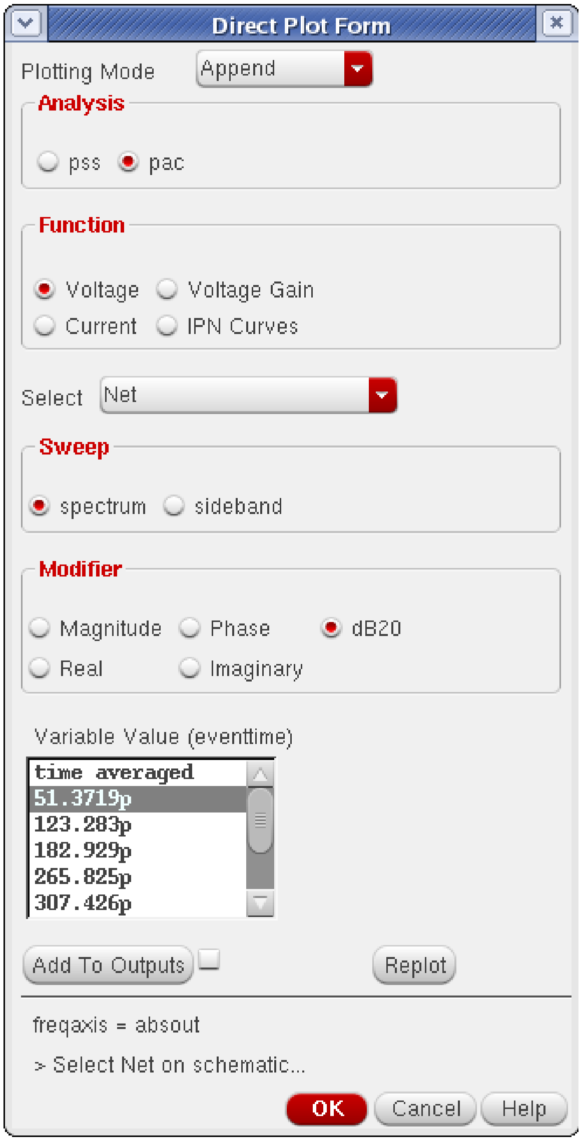
1. At the top of the form, set up the input frequency range like normal.
2. At the bottom of the form, select *Sampled*. Usually the gain to a node is desired, so select voltage.

3. Click *Select* to the right of the *Net+* field and select the node in the schematic. Note that if the *Net-* field is left blank, the global ground node is used.
4. Specify the desired threshold voltage based on the time-domain waveform from the pss analysis
5. Choose an option from the *Crossing Direction* drop-down box. In the example, *all* is selected.
6. If you need more than 16 crossing points, type the number you need in the *Maximum Samples* field.

7. If you want to sample at specific times from the pss waveform, you can type in the times in the *Additional Timepoints* field.



8. Now run the simulation. When the simulation completes, open the *Direct Plot Form* and set it up, as shown below.



9. Select *pac* results.

10. Select *Voltage*.

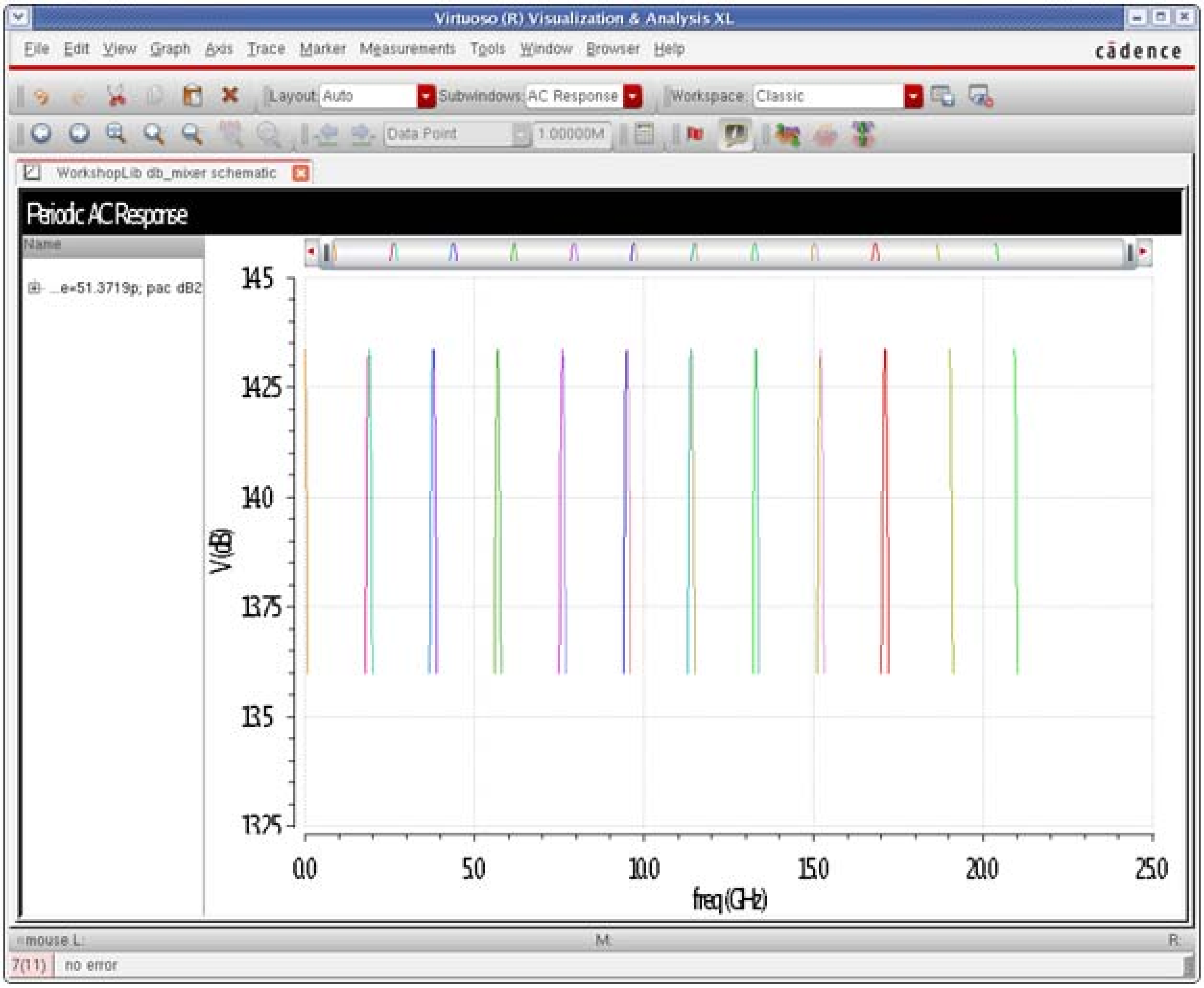
11. Select *spectrum*.

12. Select *dB20*.

13. Select the time you want from the list.

14. Select the output node in the schematic.

The waveform is plotted as shown below



To zoom in to the frequency of interest, double-click one of the numbers on the X Axis, and type in the output frequency range of interest. This is shown below.

To measure the conversion gain to the output, move the tracking cursor to the output frequency and read the gain or place a point marker by typing `m`.


Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cadence
Layout: Auto
Subwindows: AC Response
Workspace: Classic
Data Point
50.0000M
WorkshopLib db_mixer schematic
Periodic AC Response
Name
Vis
time=51.3719p; pac dB20(V)
-10
-9
-8
-7
-6
-5
-4
-3
-2
-1
0
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3
4
5
6
7
8
9
10
M1: 50.0MHz 141.5165dB
V (dB)
144
143
142
141
140
139
138
137
136
135
10
20.8
40.6
60.4
80.2
100
freq (MHz)
mouse L:
M:
R:
7(11)
RMB zoom reset to xy mode.

Periodic Noise Analysis (Pnoise)

Small-Signal Versus Large-Signal Analysis for Noise

In most cases, noise is a small-signal problem in the real world. Consider an amplifier. One approach is to calculate the small-signal gain from the noise sources to the output, then calculate the frequency response of the noise source, and then add up the noise power at the output. This is done in the linear noise analysis. This runs quickly because the gain calculation just takes the tangent at the DC operating point for everything in the circuit. This ignores large-signal effects like slew-rate limits and clipping. Because noise in most circuits is actually very small compared to the output levels, this approach works well. Because it is a linear approach, frequency translations cannot be calculated.

When the input to the amplifier gets large enough that it starts to create nonlinearity, the frequency translations in the circuit need to be taken into account. Noise near the harmonics can mix up or down to the output frequency. In this case, the large signal needs to be applied in a large-signal analysis like periodic steady-state (pss) where the nonlinearity and harmonic levels can be calculated. After the pss runs, pnoise runs and is similar to linear noise except the frequency translations of the circuit are taken into account. Instead of calculating a single gain from the noise source to the output, both the amplitude of the noise and the gain for the noise sources at the significant harmonics needs to be calculated, and each noise source now has several contributions at the output from several different noise frequencies that mix or alias to the output frequency. The basic strategy is to apply the signals that cause nonlinearity in the pss analysis, and follow that with a pnoise calculation.

This same approach can be taken for mixers where the LO signal deliberately introduces nonlinearity for the purpose of frequency translation. The LO signal is applied to the mixer in the pss analysis, and pnoise follows that. This works with the mixer by itself, or in combination with other circuits, like an LNA and a baseband amplifier.

Many designs need to tolerate a large amplitude blocking signal. The basic strategy is to apply the LO and the blocking signal (and not the RF signal) in the pss analysis and follow with pnoise. Note that depending on the frequencies of the LO and the blocker, pss may not be appropriate. If the highest frequency divided by the periodicity of the system (the pss beat frequency) is about 25 or less, pss-pnoise is a reasonable approach. If the highest input frequency is 2.5GHz and the periodicity is 1MHz, use hb-hbnoise or qpss-qpnoise instead because they will require much less time and memory to run. This is because pss calculates all the harmonics of the pss beat frequency, and 2500 harmonics would be needed just to calculate the fundamental frequency of the 2.5GHz input. Both hb and qpss calculate just the mixing frequencies that are actually produced by the circuit, which is a much smaller number of harmonics. The blocker power can be swept in the pss analysis and pnoise is run after each sweep value for the blocker power to measure the increase in noise with higher blocker power (desensitization).

For large-signal noise problems, or for systems that are not periodic, transient noise is provided. For more information on this, refer to the *Virtuoso® Spectre® Circuit Simulator and Accelerated Parallel Simulator User Guide*. This approach requires specifying a maximum noise frequency to be analyzed, which in RF systems is usually quite high. Two timepoints are created in the period of noisefmax. The transient needs to be run over many noise points in order to characterize the noise present in the circuit and needs post-processing to see the results in the frequency domain. This usually requires much more time than using small-signal noise analyses.

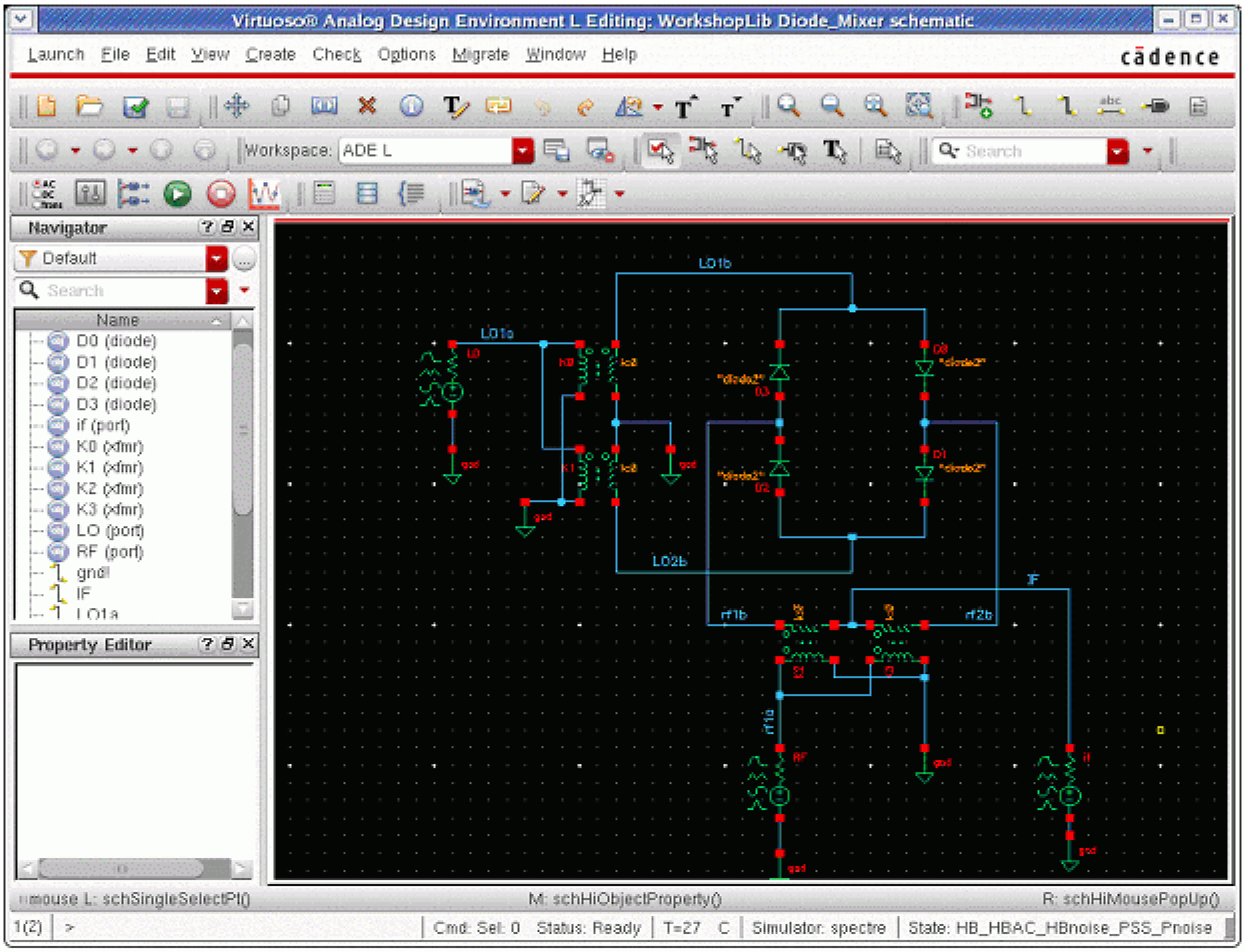
Overview of Simulation Capabilities

The fundamental quantity calculated in pnoise is the output noise of the circuit with a large signal applied and including the noise translations caused by the nonlinearity of the circuit. In order to run pnoise, a periodic steady-state (pss) analysis needs to be run first in order to calculate the nonlinearity of the circuit and the amplitude of the harmonics that are present to mix with. Pnoise takes this information from the pss analysis and then for each noise source, it calculates the transfer functions from all the frequencies specified by the maximum sideband field in the *Choosing Analyses* form. Then it calculates the frequency response of the noise source itself. Next, it calculates the total noise power at the output for that source at all the noise frequencies that contribute at the output. Finally, it adds up all the noise power from all the sources at all the frequencies. Input-referred calculations require the selection of the input frequency range by setting the reference sideband. This selection is necessary because there are many transfer functions from the input to the output caused by mixing or aliasing from the different harmonics of the system. The passband frequency is also needed for the noise figure calculations.

Note that pnoise is a small-signal calculation that works for periodic systems. If the noise is large-signal, or the system is not periodic, transient noise should be used.

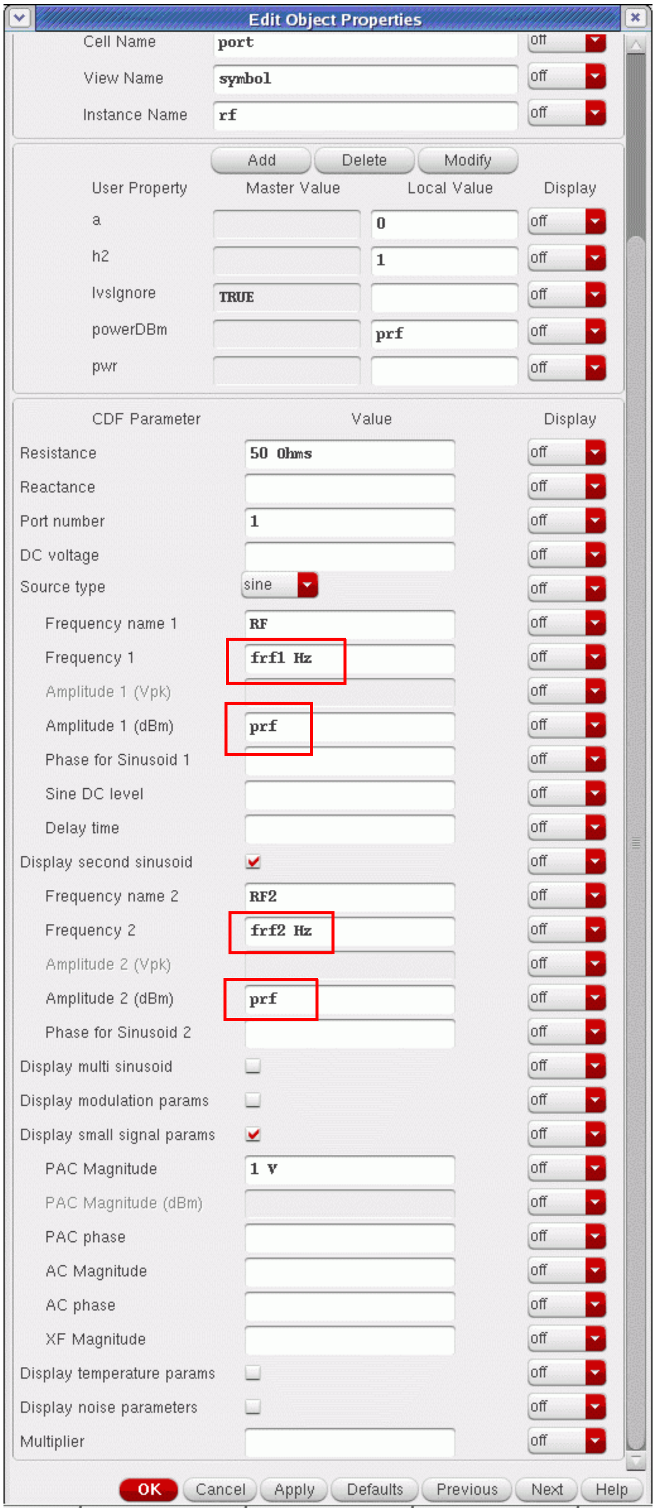
Example

Consider the double-balanced diode mixer below where a single-sideband noise figure measurement is desired.

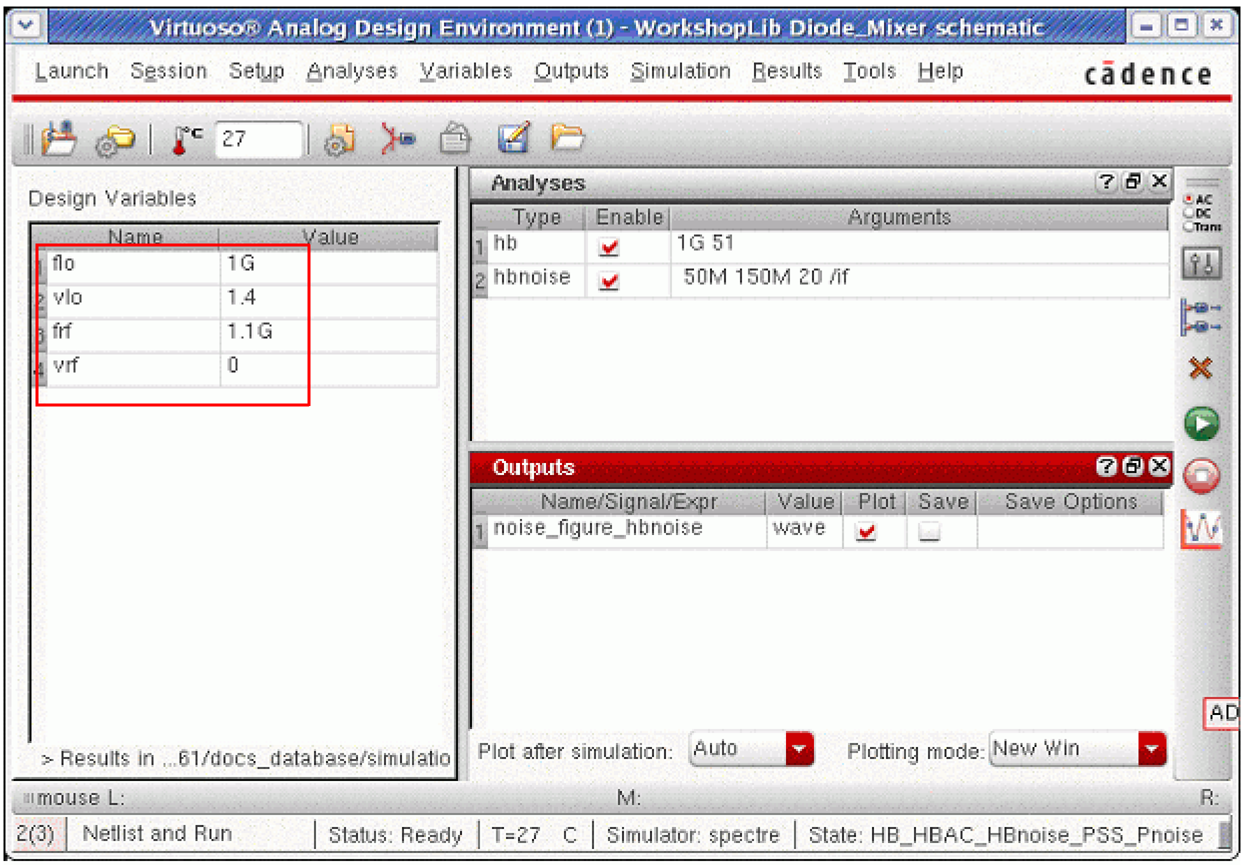


The input source has variables set to define the frequency and amplitude of the two large-signal outputs. This is done so that the frequency and the amplitude can be set in the ADE window without changing the schematic.

The property list is shown below.

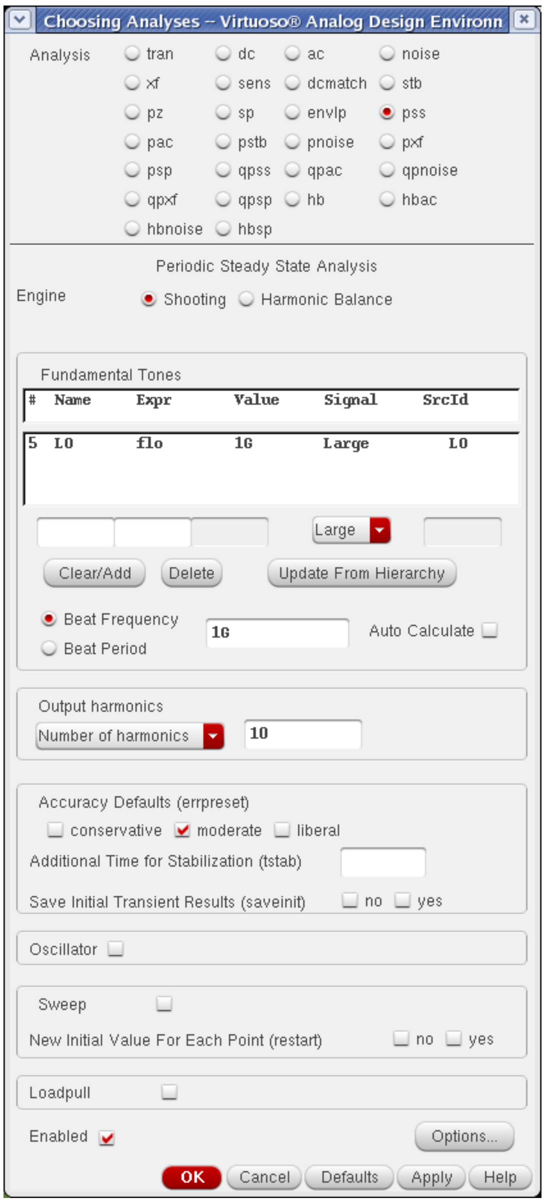


The variables have values set in the ADE variables section. The easiest way to get the variables list is to generate a netlist from the ADE or use *Variables - Copy From CellView*. This causes all the variable names to be entered in the variables section. Just click the value field for each variable and enter a value. Press <Enter> when you are done. In the figure below, the LO frequency set by the variable flo is 1GHz. The variables *fb1* and *fb2* which set the frequency of the large-signal RF inputs is zero. This disables the large-signal waveform. This circuit just has the signal that causes the frequency translations to occur (the LO signal) applied. Note that 1.4 volts peak is +13 dBm. This is a common drive level for diode mixers.



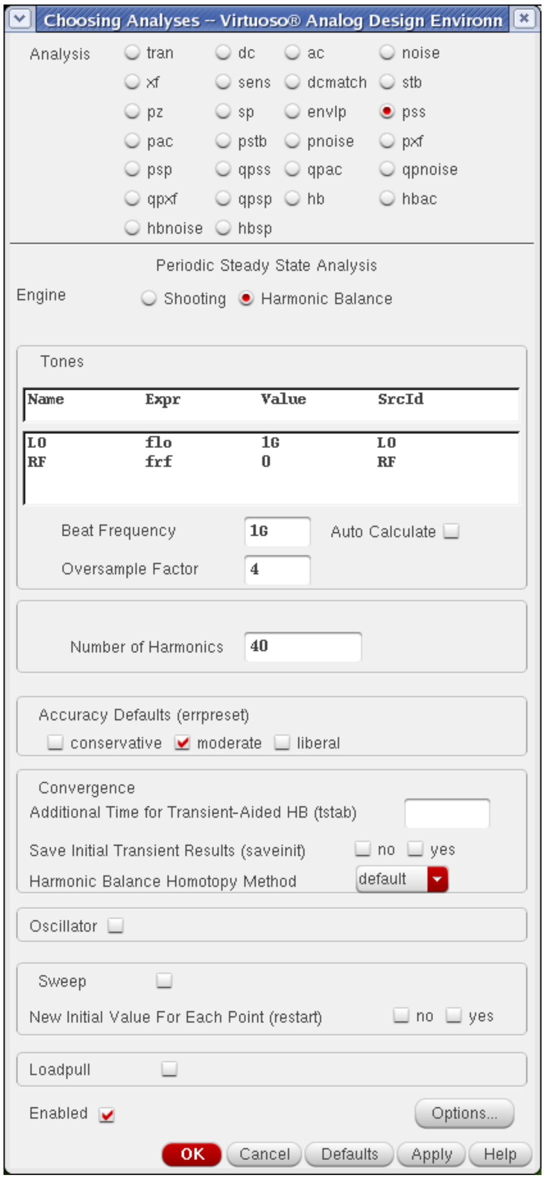
The pss analysis needs to be run first in order to measure the amplitude of the harmonics and to characterize the nonlinearity of the system. From this, the noise folding from all the noise sources can be calculated in the pnoise analysis. For more information on the setup of the pss form, please refer to Overview of Periodic Steady-State (pss) Analysis on page 547.

Pss can be set to use either the shooting engine or the harmonic balance engine. The example below uses the shooting engine. Because shooting is a variation of the transient which automatically reduces the timestep in areas or rapid transition (where the diodes switch) and because the pss includes at least 200 timepoints in the solution, the pss shooting solution itself contains information to the 100th harmonic of 1GHz. For the pnoise to be accurate, at least 5 timepoints are needed at the highest frequency that is being calculated in the pnoise. Thus, with the defaults, frequency translations through the 40th harmonic of 1GHz can be accurately calculated when shooting is selected as the engine.

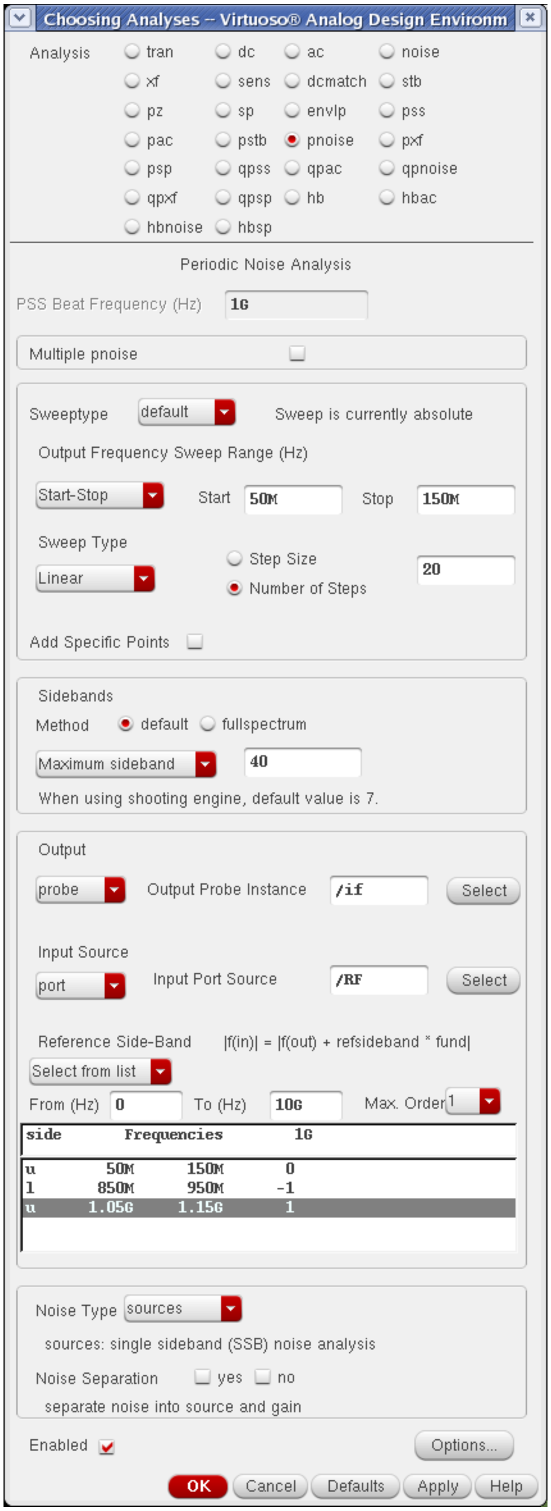


When harmonic balance is selected as the engine, the number of harmonics and the oversample factor need to be set appropriately. For more information, see Chapter 3,

“Frequency Domain Analyses: Harmonic Balance,” Below is an example of harmonic balance in the pss *Choosing Analyses* form.



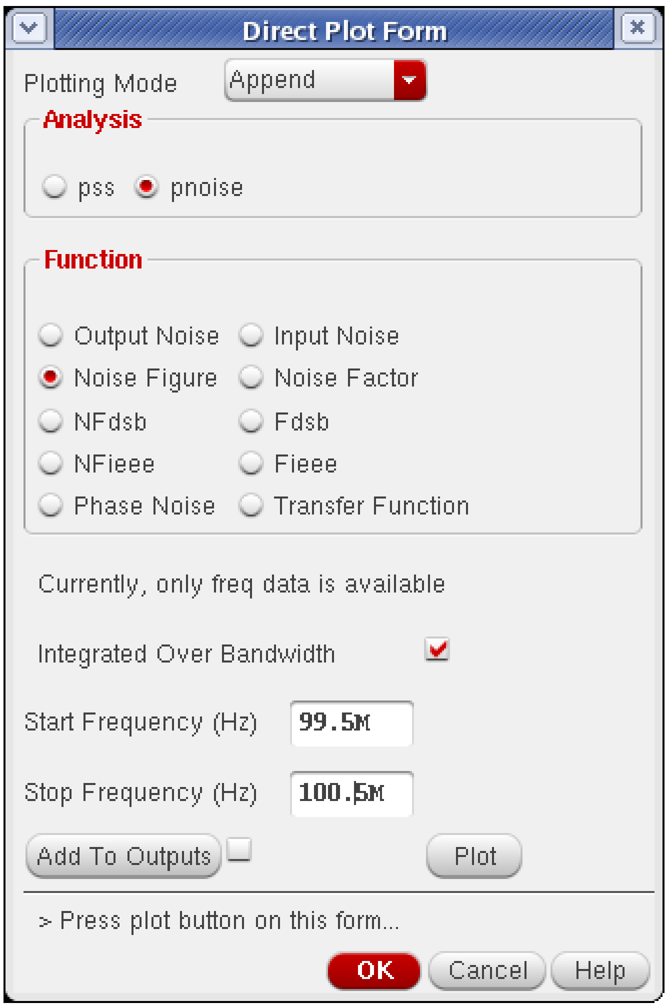
Regardless of the engine that is used in the pss *Choosing Analyses* form, the noise output frequency and the number of noise translations to be calculated are set up in the pnoise form. When harmonic balance is selected for the engine in pss, leave the maximum sideband field blank or set it to the number of harmonics in the pss-hb *Choosing Analyses* form, which calculates the noise translations for all the harmonics that are calculated in the pss analysis. When shooting is selected, you must specify the number of noise translations you want to calculate in the maximum sidebands field. Up to 40 noise translations can be calculated with the default settings in pss. The example below shows 40 sidebands, which is the number of harmonics used in the pss-hb analysis that is shown in the preceding figure.


Choosing Analyses -- Virtuoso® Analog Design Environm
Analysis
tran dc ac noise
xf sens dcmatch stb
pz sp envlp pss
pac pstb pnoise pxf
psp qpss qpac qpnoise
qpxf qpsp hb hbac
hbnoise hbsp
Periodic Noise Analysis
PSS Beat Frequency (Hz) 1G
Multiple pnoise
Sweeptype default Sweep is currently absolute
Output Frequency Sweep Range (Hz)
Start-Stop Start 50M Stop 150M
Sweep Type
Linear
Step Size
Number of Steps 20
Add Specific Points
Sidebands
Method default fullspectrum
Maximum sideband 40
When using shooting engine, default value is 7.
Output
probe Output Probe Instance /if Select
Input Source
port Input Port Source /RF Select
Reference Side-Band |f(in)| = |f(out) + refsideband * fund|
Select from list
From (Hz) 0 To (Hz) 10G Max. Order 1
side Frequencies 1G
u 50M 150M 0
l 850M 950M -1
u 1.05G 1.15G 1
Noise Type sources
sources: single sideband (SSB) noise analysis
Noise Separation yes no
separate noise into source and gain
Enabled
Options...
OK Cancel Defaults Apply Help

The reference sideband setting is used to specify the passband frequency for the noise figure calculation.

In the example above, the output frequency is near baseband.

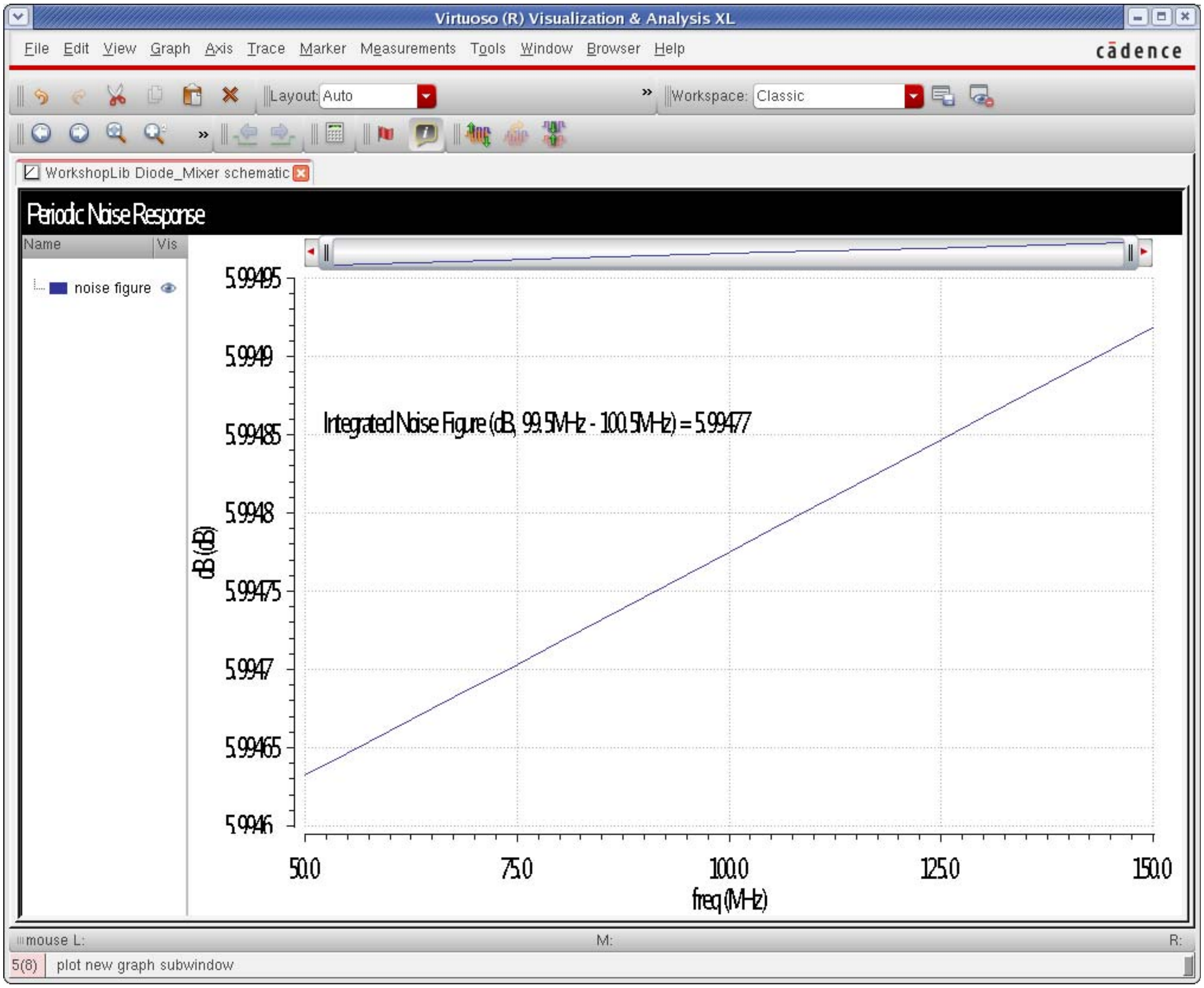
The reference sideband selects the passband frequency for the noise figure calculation. Once both forms are set up, the simulation is run. To plot outputs, select *Results - Direct Plot - Main Form* in the ADE window.



The top of the form lists all the analyses that were run. Select *pnoise* where the noise figure is calculated. Next, select *NF* because that is what is desired to be plotted. If the noise figure over a bandwidth is desired, select *Integrated Over Bandwidth*, and specify the frequency range you desire for the measurement.

When you are done, click *Plot*.

The waveform window plots the noise figure curve and the integrated noise figure calculation over the frequency range you specified in the direct plot form. This is shown below.



Principles of Pnoise

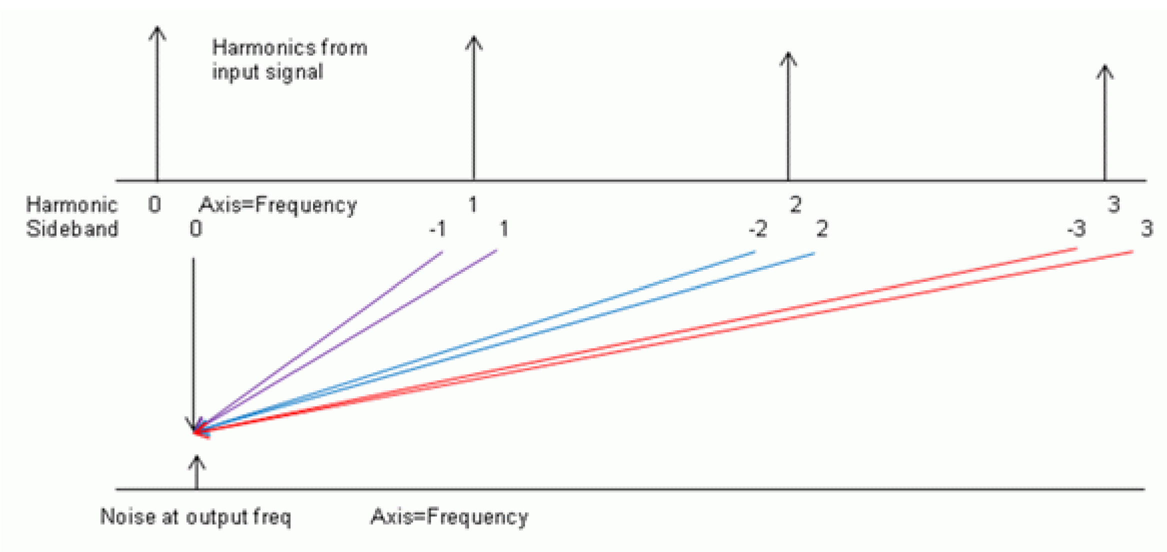
Pnoise has three basic types of noise calculation. The first mode calculates the average noise power that occurs with the variations of the LO or clock signal taken into account. The noise is averaged over the LO or clock cycle. This is calculated when the noisetype parameter is set to sources. The AM and PM components of the noise can be calculated when the noisetype is set to modulated. When the noisetype is set to jitter for driven circuits, an instantaneous measurement of the noise at the threshold crossing in the pss analysis is calculated. This allows jitter measurements for digital outputs on either the rising, falling, or both edges of the clock. For oscillators, when jitter is selected, either an averaged measurement can be made for phase noise, or an instantaneous measurement can be made at one threshold voltage of the oscillator output voltage or current. There is a fourth setting in the *Choosing Analyses* form which is called timedomain. Conceptually, this does the same noise measurement as noisetype=jitter, but it requires you to do the jitter calculation manually. It also allows multiple places in the rising and/or falling edges to be measured.

Noise Type=sources

Noise Output Near Zero Frequency

First, the pss analysis is run to calculate the nonlinearity and the harmonics. Then pnoise is run to calculate the noise. In all systems, there is a maximum number of noise translations that need to be taken into account. Either the system harmonics become so small that the amount of noise translation becomes minimal or the devices themselves have finite noise bandwidths.

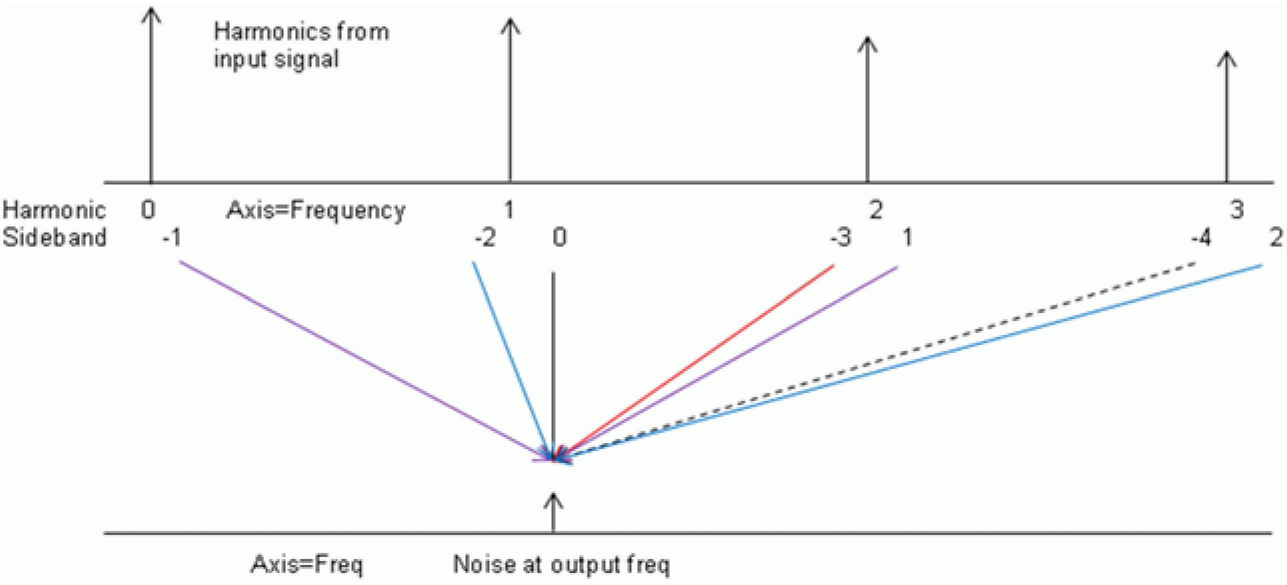
These noise calculations from mixing or aliasing with the different harmonics are called sidebands.



In the diagram above the X axis is the frequency. The top represents the harmonics that are calculated by the pss analysis. The bottom represents the noise output frequency. The arrows show the various mixing paths to the output frequency. Pnoise calculates the noise contribution at the output with all these mixing paths in place. Black is the noise that occurs without mixing or aliasing. Purple is the noise that mixes with the first harmonic. Blue is the noise that mixes with the second harmonic. Red is the noise that mixes with the third harmonic.

In pnoise, the output frequency range is specified along with the highest harmonic to calculate the mixing products.

Noise Output Near The First Harmonic

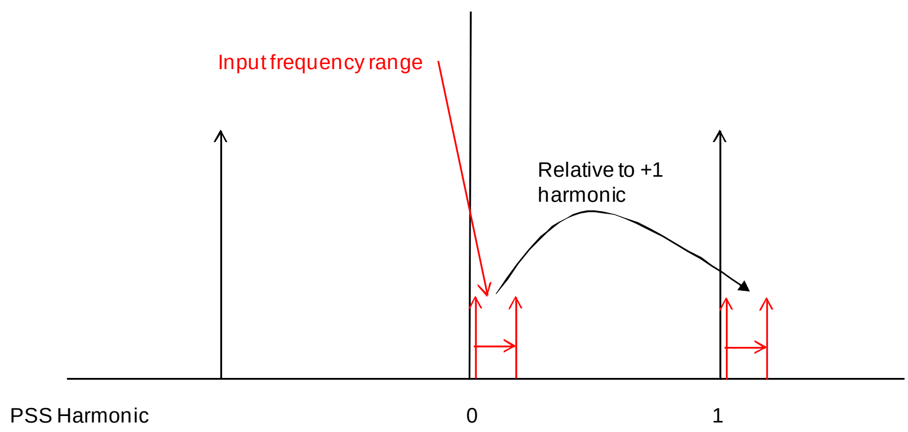


Here, the noise output frequency is just above the first harmonic frequency. Black is the noise that occurs without mixing. Purple is the noise that mixes with the first harmonic. Blue is the noise that mixes with the second harmonic. Note that the noise frequency just below the first harmonic mixes with the second harmonic to provide the output just above the first harmonic. Red is the noise that mixes with the third harmonic. Note that the second term that mixes with the third harmonic is just above the fourth harmonic of the pss analysis which is not shown. The noise term shown with a dotted line is mixing with the fourth harmonic of the system.

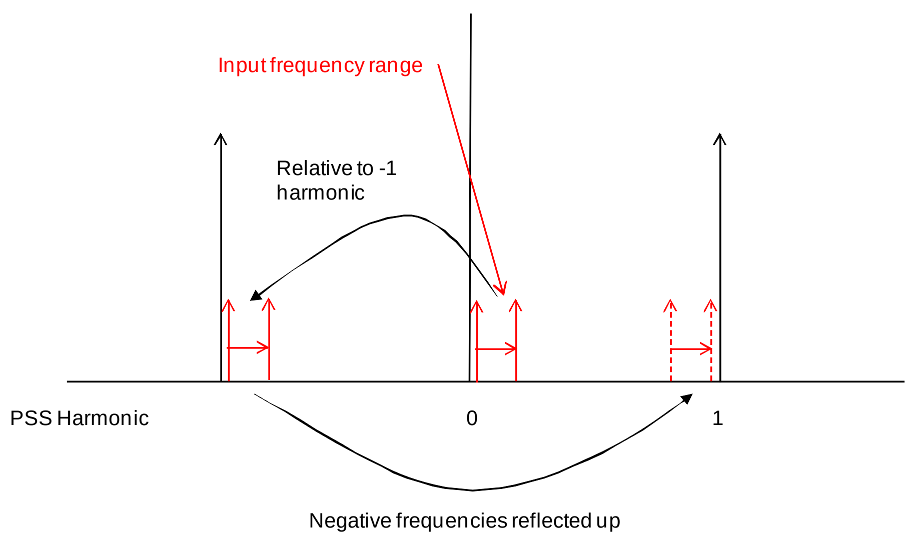
Frequency Sweep

Since the fundamental quantity that is calculated in pnoise is the output noise, the frequency range defined in the *Choosing Analyses* form defines an output frequency range to calculate the noise for. For driven circuits, the default is to sweep an absolute frequency range. Oscillators will be covered later in this chapter.

For an up-conversion system, sometimes, a log frequency range is desired near one of the harmonics of the system. In that case, select relative sweep, and specify a harmonic number. In the output frequency range, type the desired offset frequency from that harmonic. Below is a diagram that shows relative to the first harmonic, which calculates the frequencies just above the first harmonic.



When a log sweep is desired below the first harmonic, set the relative harmonic to -1 as shown in the diagram below.



When automatic is selected for sweep type, 100 total points will be calculated. In most cases, fewer points are needed, so it is usually better to select linear or log and then specify the sweep you desire. Typically, 3 to 5 points per decade or about 20 total points are needed.

Maximum Sideband

There is a property in the *Choosing Analyses* form called *Maximum sideband* that is used to define how many mixing products should be calculated. If *Maximum sideband* is zero, the noise analysis will only contain the noise that appears without mixing or aliasing of any type.

This is not equivalent to a linear noise analysis because pnoise takes into account the instantaneously varying nature of the noise because of the large signal being present in the pss analysis.

When *Maximum sideband* is one, then the noise without frequency translation and the noise that mixes or aliases with the first harmonic of the pss is present in the noise analysis. When *Maximum sideband* is 10, the noise analysis includes noise mixing through the 10th harmonic of the pss analysis. When harmonic balance is selected as the pss engine, the number of harmonics in the pss analysis and the maximum number of sidebands should be the same. This is easily accomplished by leaving the *Maximum sideband* field blank in the *Choosing Analyses* form in ADE. When shooting is selected for the pss engine, up to 40 sidebands can be set in the maximum sideband field by default. If you need more than 40 sidebands in pnoise, consider using full-spectrum pnoise.

Full-Spectrum Pnoise

Full-Spectrum pnoise is useful for circuits like switched-capacitor filters or sampling circuits where aliasing occurs through very high harmonics of the clock. The runtime advantages are large with no loss in accuracy of the result. Full-Spectrum pnoise is available when shooting is selected for the pss engine and APS is selected in the *Setup - High Performance Simulation* menu in ADE. Selecting *fullspectrum* in the pnoise form forces APS to be selected in ADE. If you are running from the command line without using aps, fullspectrum will not be used. In normal pnoise when shooting is selected, you need to set the maximum sideband term. In full-spectrum pnoise you do not except in cases where the pss beat frequency is 100KHz or less. In this case, set maximum sidebands to the 1/F noise corner frequency divided by the pss beat frequency. Pnoise calculates all the noise translations it can, based on the maximum timestep in the pss analysis. The easiest way to change the maximum timestep in pss is to increase the number of harmonics above 10 in the pss *Choosing Analyses* form. Full-spectrum pnoise runs faster than normal pnoise when the maximum number of sidebands in pnoise is about 50 or greater. The larger the number of sidebands, the larger the speedup for fullspectrum.

Fullspectrum pnoise is available for all available noisetypes. Noise separation is not supported when fullspectrum is selected.

Setting Harmonics and Sidebands

Harmonic Balance as the PSS engine

The process of setting harmonics and sidebands is similar to setting the maximum number of harmonics for the harmonic balance simulation. Start with an estimate based on the harmonic content of the input signal(s) and the nonlinearity of the circuit. Then run a simulation. Raise harmonics and sidebands and run the simulation again. If the noise measurement did not change, then the original number of harmonics and sidebands may be reduced. If it did change, then more are needed. For more details on this, refer to the section Harmonic Balance Noise Analysis (hbnoise) on page 344.

Shooting as the PSS Engine

For full-spectrum pnoise, start with 10 pss harmonics, Select full-spectrum, and select APS. Run the simulation, and plot the noise result. Now increase the number of harmonics in the pss analysis, which forces more pss timepoints, and run the simulation again. If the pnoise result did not change, then you had enough harmonics to begin with. If the noise result did change, then increase the number of pss harmonics and run the simulation again.

For regular pnoise, estimate the number of sidebands based on the pss harmonic plot. As a coarse estimate, find the harmonic where the LO or clock signal drops to about 40 dB below the amplitude of the LO or clock first harmonic. For example, if this is the 10th harmonic, set maximum sideband to 10. Run the simulation and plot the noise measurement. Increase the number of sidebands until the noise result does not change anymore. Use the smallest number of sidebands that produces a stable result. If the number of sidebands is greater than about 50, consider using full-spectrum pnoise because it is likely to run faster than normal pnoise.

Noise Separation

In addition to the total output noise, the individual noise contributions can be plotted if noise separation is selected in the *Choosing Analyses* form. More information about the noise separation will be provided later. Noise separation is available when the noise type is set to sources only and is not available for fullspectrum pnoise. The things that can be plotted are:

1. Total noise at the output from each individual noise input frequency. This allows the identification of which noise frequencies are causing the noise problem.
2. The noise at the output from each instance name with all noise mechanisms included for mixing from an individual noise input frequency. Once the noise frequency is identified in point 1 above, this capability plots the total noise from each component at the

troublesome noise frequency. This allows the identification of which component in the circuit is causing the problem.

3. The noise at the output from each instance name with all noise mechanisms broken out separately for mixing from an individual noise input frequency. This capability splits out the individual noise sources at the troublesome noise frequency. For example, if a parasitic resistor was found to be the problem, the component might be resized to reduce the noise.

4. The current noise at the noise instance for each instance name with all noise mechanisms included for mixing from an individual noise input frequency. This capability has limited value. For this plot, all the noise source currents for all the noise mechanisms within the component are added together.

5. The noise current in the instance from each instance name with all noise mechanisms broken out separately for mixing from an individual noise input frequency. This capability allows the identification of which noise sources are the largest in the circuit. The measurement is at the source, not at the output.

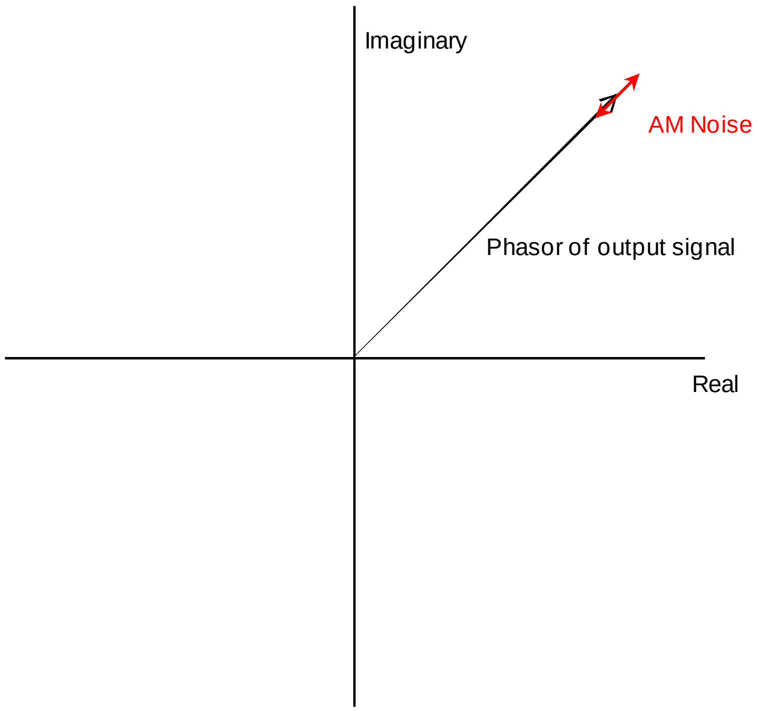
6. The gain to the output from each individual instance with all the individual noise sources broken out separately for mixing from an individual input frequency. This capability plots the gain from the noise sources in point 5 above to the output of the circuit.

Multiple Pnoise

Some systems contain multiple points in the circuit where the noise measurement is desired. When the *Noise Type* is *sources*, multiple pnoise analyses are allowed. At the top of the pnoise *Choosing Analyses* form, an option called *Multiple pnoise* is provided. When this option is selected, up to six pnoise analyses can be made after a single pss analysis.

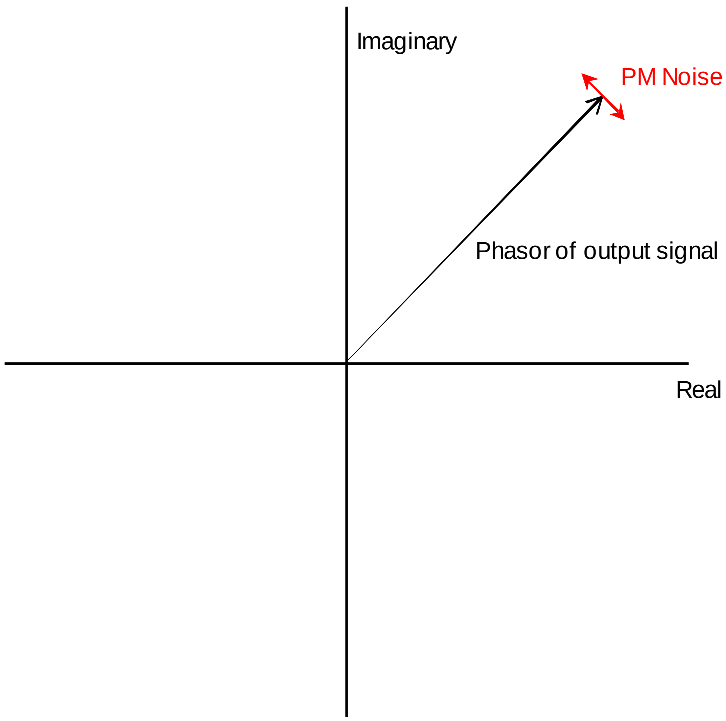
Noisetype = Modulated for Driven Circuits and Oscillators (AM and PM Noise)

Modulated is provided so that the AM and PM components of noise can be separated. When the *Noise Type* is *sources*, the total noise (AM + PM) is calculated. To illustrate AM noise, see the phasor diagram below.



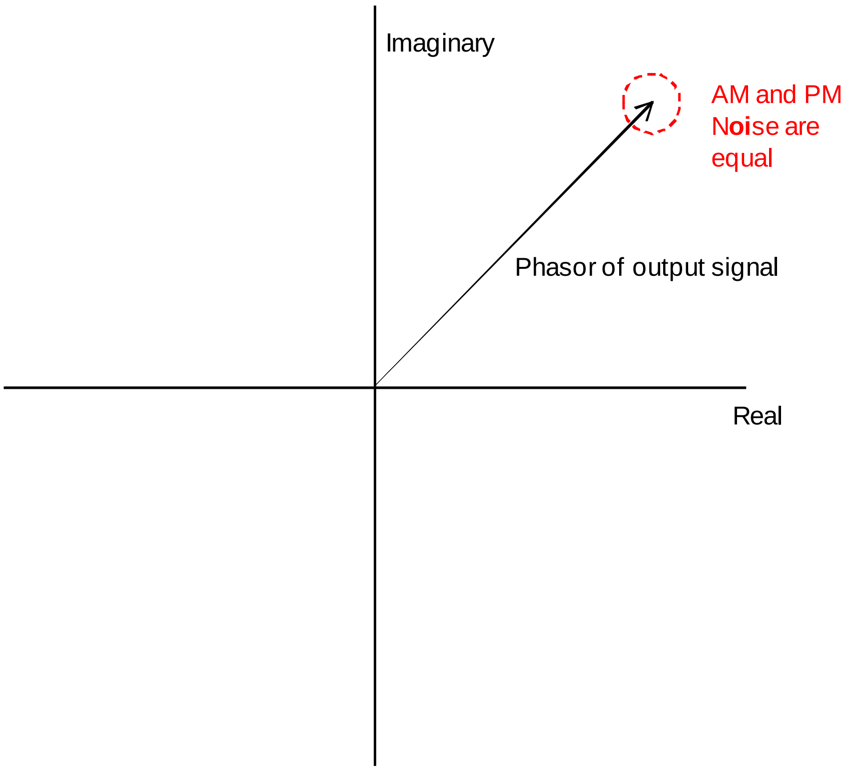
This diagram illustrates a periodic signal with AM noise. The black phasor is the output signal, and AM noise is a random vector that varies the amplitude of the signal.

PM noise is shown in the phasor diagram below.



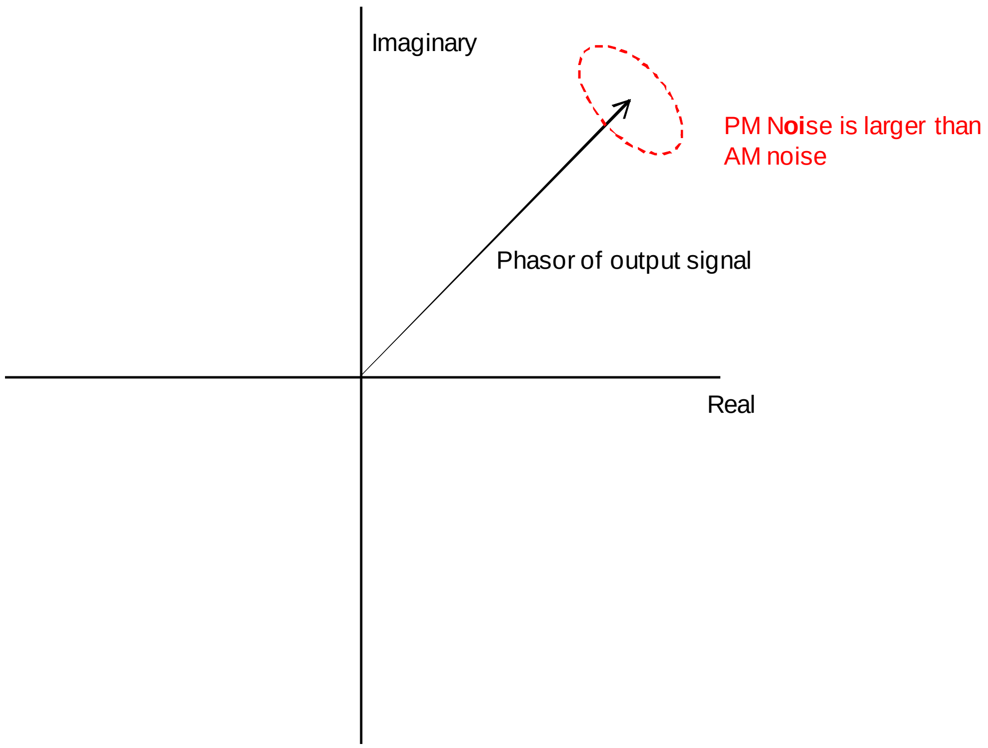
In this case, the amplitude is constant, but the noise is a random vector perpendicular to the signal phasor. The timing of the signal varies a bit from cycle to cycle.

Most circuits produce a combination of AM and PM noise. This concept is shown below.



In this case, a random vector at any angle and amplitude is produced that varies at each instant of time. If the uncertainty creates a round shape, the AM component and the PM component are the same size.

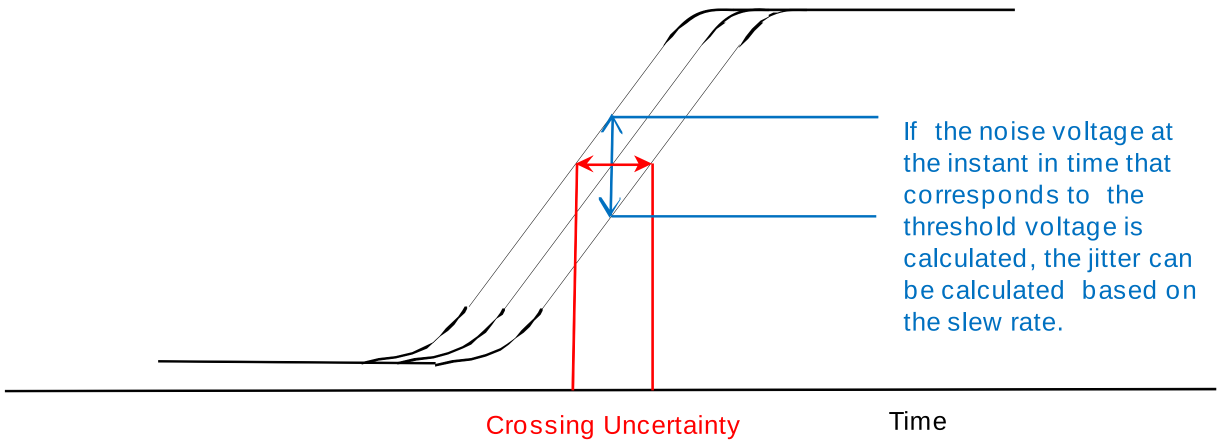
In many circuits, AM or PM noise may dominate. If the PM noise dominates, the circle turns into an ellipse that is larger in the perpendicular direction than in the parallel direction. This is shown below.



Modulated also provides the total noise in the frequencies above the output frequency (the USB in the *Direct Plot Form*) and the noise below the output frequency (the LSB in the *Direct Plot Form*).

Jitter for Driven Circuits

Jitter is usually applied to digital circuits where the average noise over the entire waveform is not appropriate. The noise in the high state and in the low state does not matter because the output voltage is so far from the voltage that triggers the load that it does not matter. The only time the noise matters is at the exact time where the trigger event is generated. Also, the noise in the low and high state is quite small because one device is completely off, and the other device is driven on. It is basically a resistor. The device is not very good as an amplifier because the output is pulled hard into a rail. As the state changes, both devices are on, and they are relatively better amplifiers, so the noise is likely to be larger in the switching interval. In jitter for driven circuits, the voltage for the noise measurement, and the selection of the rising edge, the falling edge, or both is made. At the threshold voltage specified, a noise measurement is made at that instant in time. Once the noise voltage is known, the timing uncertainty can be determined from the slew rate. Since the RMS noise voltage is calculated, the timing uncertainty is one standard deviation from the ideal crossing time. This is illustrated below.



Timedomain for Driven Circuits

This is a variation of the jitter analysis explained in the previous section. In this analysis, instead of specifying a threshold voltage, a list of times from the time-domain waveform is supplied. A noise analysis is performed at those instants in time, and the noise voltage is calculated at each timepoint. The conversion from noise voltage to time-domain jitter is a manual step. Divide the noise voltage by the slew rate to get the jitter.

Oscillators

Noise Type = Sources

For oscillators, the pnoise frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the pss form, you should specify the approximate frequency of oscillation for the frequency-divided signal. In the pnoise form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of relative is to take the frequency of the harmonic number specified and add to it the frequencies specified in the *Choosing Analyses* form. If the oscillator had a 1GHz output, and the pnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

Oscillators do not need noise figure calculations and are not typically driven, so the input source should be set to *none*. The output is usually a voltage, and an output node is typically selected.

Pnoise is a small-signal analysis and is not limited by large-signal effects, such as clipping or slew-rate limits. As a result, at low offset frequency, the phase noise might be significantly greater than 0dBc/Hz. This indicates that the noise is larger than the oscillations, which is not physically possible. If you want to see the phase noise curve level off at low frequency, set the *lorentzian* option to *yes*. More information is provided in the lorentzian option discussion.

Phase noise is the total output noise (the AM plus the PM component of noise) divided by the harmonic specified as the relative harmonic for the node specified for the noise measurement. If you want to see the AM and PM components of noise, or you want to measure jitter, or you want to make an instantaneous point in time noise measurement, set the *Noise Type* to *modulated* or *jitter*.

Modulated

Modulated is provided so that the AM and PM components of noise can be separated out. When the *Noise Type* is *sources*, the total noise (AM + PM) is calculated. For a diagram to understand the concepts, see Noisetype = Modulated for Driven Circuits and Oscillators (AM and PM Noise) on page 851

Timedomain

This is usually applied to oscillators that drive digital dividers where the average noise over the entire waveform is not appropriate. The noise above and below the threshold does not matter because the output voltage is so far from the voltage that triggers the load that it can never cross the threshold. The only time the noise matters is at the exact time where the trigger event is generated. In timedomain for oscillators, a list of times from the waveform is supplied and a noise measurement is made at that list of times. Once the noise voltage is known, the timing uncertainty can be determined from the slew rate. The noise calculated from this analysis provides a relative measure of the noise at different times. For a quantitative measurement, use PM jitter. For a diagram that illustrates this, see Jitter for Driven Circuits on page 854.

FM Jitter

FM jitter calculates a standard averaged phase noise measurement, and also the AM and PM components from the modulated analysis, and adds the ability to integrate the phase noise curve to calculate the cycle jitter or the cycle-to-cycle jitter. The jitter calculations are integrated into the *Direct Plot Form* in ADE. All the measurements are averaged over the oscillator cycle.

PM Jitter

This is like the timedomain jitter measurement. You specify a threshold voltage and the timing jitter is calculated. In PM Jitter, a quantitative jitter measurement can be made from the direct plot form. For a diagram on how this works, see Jitter for Driven Circuits on page 854.

Noise Calculations

Phase Noise (Driven Circuit)

For a driven circuit with a single input frequency, phase noise is the total noise at the output divided by the amplitude of the first harmonic at the output node specified in the pnoise analysis. Note that phase noise from the pnoise results selection is a single-sideband phase noise. Only the frequencies above the oscillator frequency are included in the measurement. To get a double-sideband phase noise measurement, set the noise type to jitter, and select the pnoise jitter result set.

Noise Factor and Noise Figure

When the pnoise analysis is set up with voltage selected instead of probe for the input, noise figure and noise factor will not have the noise of the load subtracted automatically from the measurement. Using a port as the input probe is required to obtain noise figure and noise factor calculations.

In the *Direct Plot Form*, noise factor and noise figure are both single-sideband measurements.

Noise Factor

Noise Factor is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the load))/Noise power at the output
from the resistance of the input port from the passband frequency only.
```

Note that in order to calculate the noise factor, a port component must be used as the input. In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element.

No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the pnoise *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the *Choosing Analyses* form.

Noise Figure

Noise figure is:

```
10*log(10) (Noise Factor)
```

Noise factor (IEEE) is similar to the noise figure measurement above. If you subtract from the numerator all the noise from the input port at all frequencies except the passband, you have the IEEE noise factor.

Note that in order to calculate the IEEE noise factor, a port component must be used as the input. In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element. No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the *Choosing Analyses* form.

Noise Factor IEEE

Noise Factor IEEE is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the resistor in the input port at
all frequencies except for the passband) - (Noise power at the output from the
load))/Noise power at the output from the resistance of the input port from the
passband frequency only.
```

Noise Figure IEEE

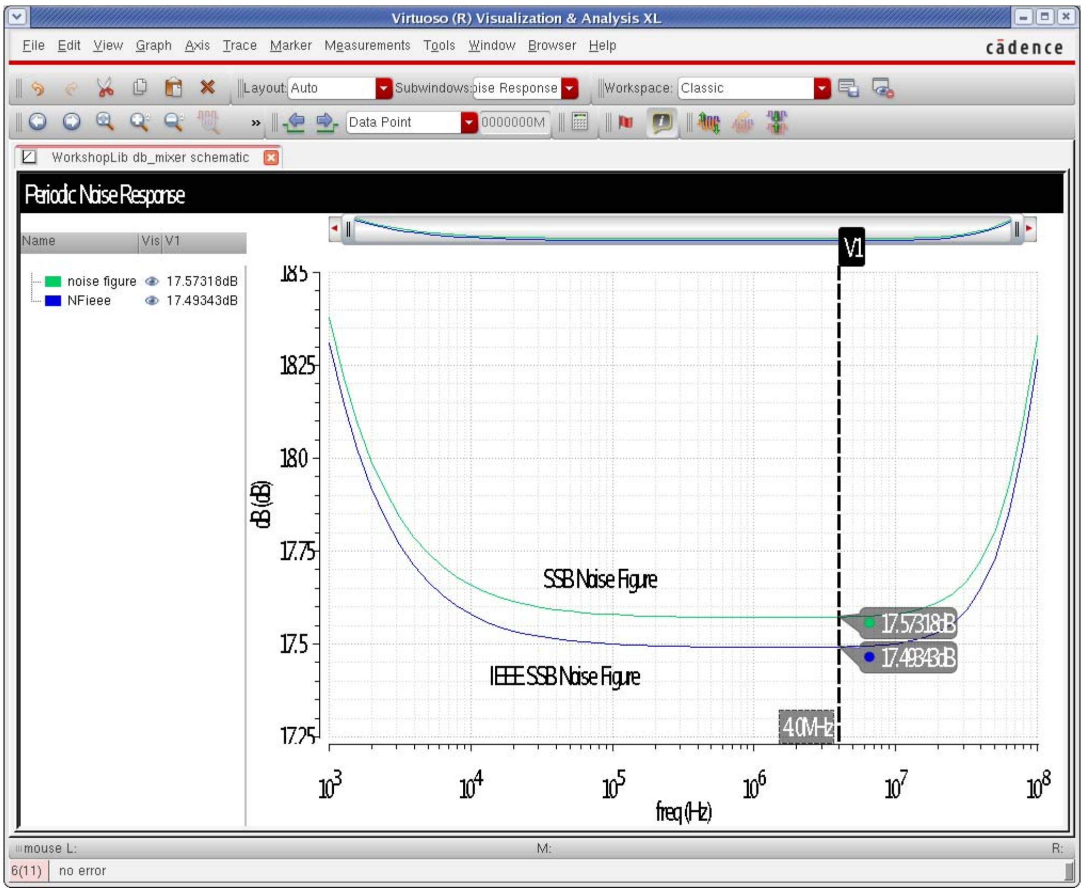
Noise figure IEEE is:

```
10*log(10) (IEEE Noise Factor)
```

Noise factor DSB is similar to the single sideband noise factor above, except in the denominator, the noise power from the input port at the image frequency is added.

Note that Noise Figure and NFiee are both single-sideband measurements. The difference between the measurements is that in noise figure, all the noise frequencies from the input port that contribute noise at the output are included, and in NFieee, only the noise at the passband frequency is included. As a result, NFieee will produce a lower noise figure than Noise Figure will The difference between the measurements depends on the circuit.

.



Noise Factor DSB

Noise factor (DSB) is:

```
((All the noise power at the output from all the components with all the frequency
translations) - (Noise power at the output from the load))/Noise power at the output
from the resistance of the input port from the passband and image frequencies only.
```

Note that in order to calculate the DSB noise factor, a port component must be used as the input. In the *Choosing Analyses* form, the input port is identified by the instance name. The port component is a voltage source in series with a resistor and is considered as a single element. No access is provided to the results for the internal node where the resistor and voltage source join.

In order to determine the passband frequency, the reference sideband must be supplied. This specifies the passband frequency and by inference the image frequency for the noise figure calculation. The easiest way to select the reference sideband is to use the *Select from list* feature in the *Choosing Analyses* form.

The noise from the load is normally defined by selecting the port or resistor that is serving as the load for the circuit in the *Choosing Analyses* form.

Noise Figure DSB

Noise figure (DSB) is:

```
10*log(10) (Noise Factor (DSB))
```

Summary:

`No` = total output noise
`Ns` = noise at the output due to the input probe at the passband frequency (the source)
`Nsi` = noise at the output due to the image harmonic at the source
`Nso` = noise at the output due to harmonics other than input at the source
`Nl` = noise at the output due to the output probe (the load)
`IRN` = input referred noise
`G` = gain of the circuit
`F` = noise factor
`NF` = noise figure
`Fdsb` = double sideband noise factor
`NFdsb` = double sideband noise figure
`Fieee` = IEEE single sideband noise factor
`NFieee` = IEEE single sideband noise figure

then,

```
IRN = sqrt(No^2/G^2)
F = (No^2 - Nl^2)/Ns^2
NF = 10*log10(F)
Fdsb = (No^2 - Nl^2)/(Ns^2+Nsi^2)
NFdsb = 10*log10(Fdsb)
Fieee = (No^2 - Nl^2 - Nso^2)/Ns^2
NFieee = 10*log10(Fieee)
```

When the pnoise analysis is set up with voltage selected instead of probe, noise figure cannot be calculated.

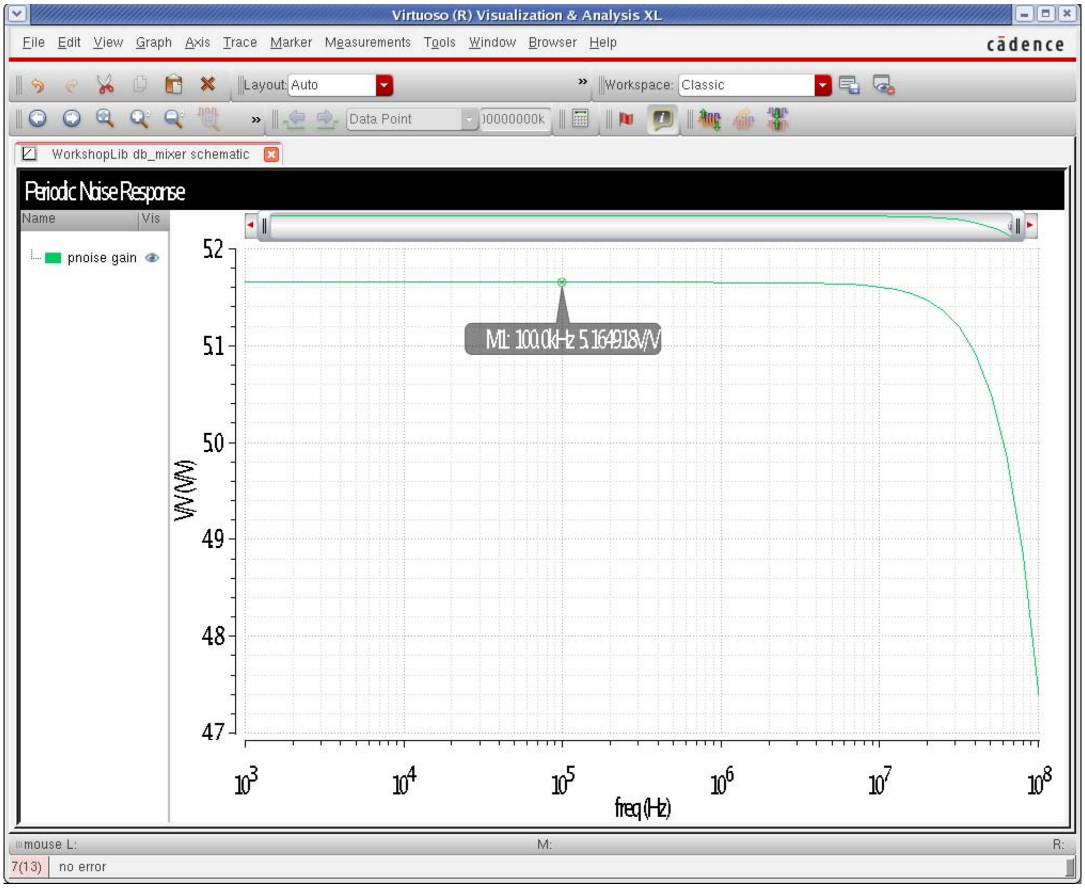
Input-Referred Noise

Important

When the pnoise analysis is set up with the input source being a port, input-referred noise may not be exact. In this case, the input-referred noise is referred to the voltage source that is inside the port component. The amplitude of the noise is divided by two, which assumes a perfect match. If an accurate input-referred noise is desired, set the *Input Source* to *voltage*, and select the input node. This will disable the noise figure calculation.

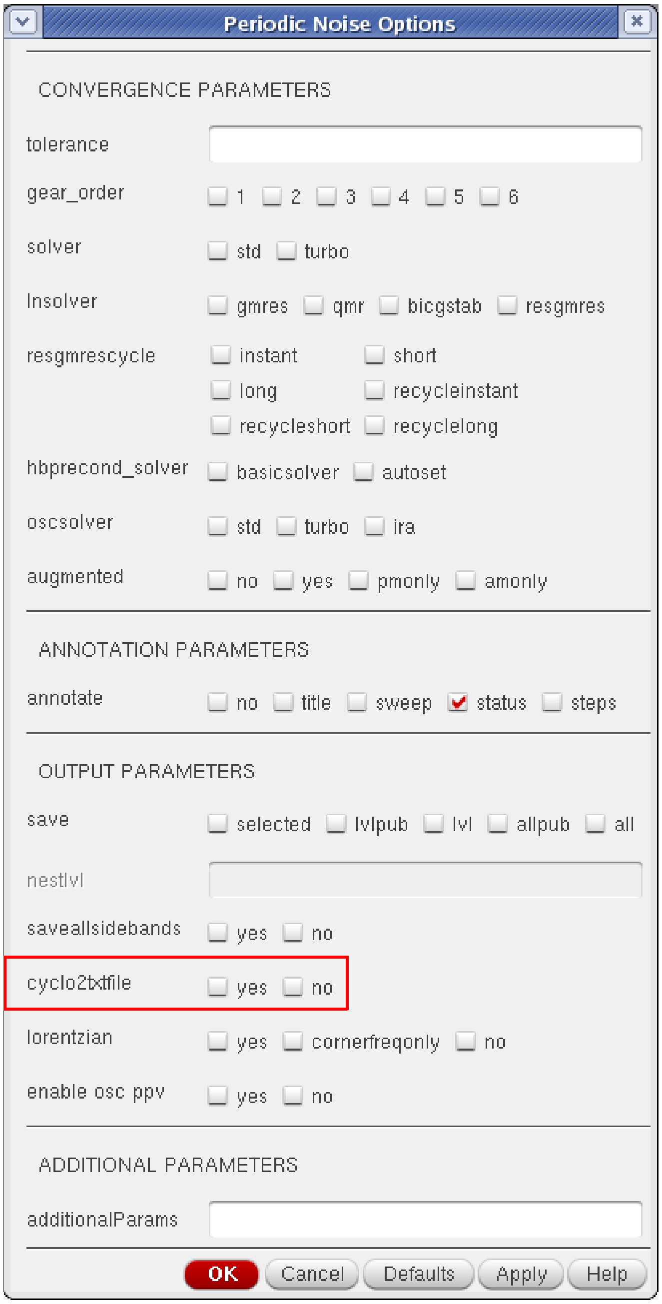
Noise transfer function

Pnoise calculates the forward gain from the input passband to the output and saves it. In the *Direct Plot Form*, this is called *Transfer Function*, and is in linear volts at the output divided by volts at the input.



Commonly Used Pnoise Options

Cyclo2txtfile



Select the `cyclo2txtfile` option if you want the simulator to create a file that has all the noise information in it. When this option is set, pnoise runs a noise analysis at all the sideband frequencies using the sweep information from the pnoise form. The file is created in the `psf` directory and has the filename pnoise.pnoise.data. If you intend to keep this file for use in a voltage source with the noise file set to a name, make sure you copy this file to another directory. The psf directory contents are deleted at the start of each simulation.

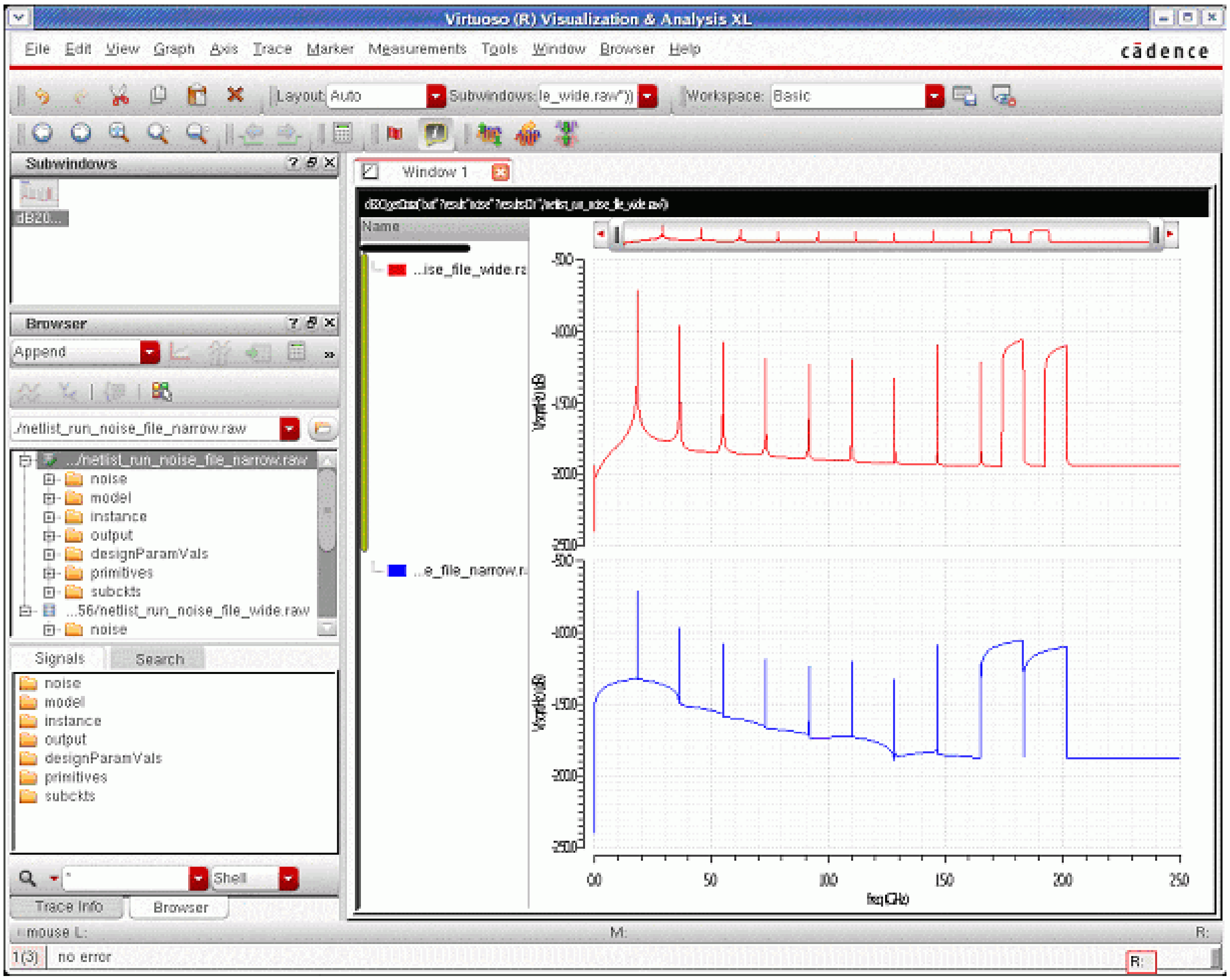
The psf directory is located either in the simulation/<circuit_name>/spectre/schematic directory or in the <circuit_name>/spectre/config directory. The simulation directory defaults to your home directory and can be set by your system administrator. Contact your system administrator to find the simulation directory if you are unfamiliar with the location.

The output file from cyclo2txtfile has two columns. The left column is the frequency, and the right column is the noise voltage squared, normalized to a 1Hz noise bandwidth. Below is an example of the file.

```
Terminal
File  Edit  View  Terminal  Tabs  Help
1.0000000002384186e+03     4.1976179202311482e-20
1.5848931927680969e+03     1.6713526184450628e-20
2.5118864316940308e+03     6.6541397418897690e-21
3.9810717058181763e+03     2.6490872073399725e-21
6.3095734450817108e+03     1.0545957949468125e-21
1.0000000000238419e+04     4.1981471199230932e-22
1.5848931924819946e+04     1.6711642040673237e-22
2.5118864315271378e+04     6.6548863771520372e-23
3.9810717055559158e+04     2.6594618704367568e-23
6.3095734448194504e+04     1.0893235490275314e-23
1.0000000000023842e+05     5.1584490781642718e-24
1.5848931924629211e+05     4.2122011260108464e-24
2.5118864315128326e+05     7.3817813521682905e-24
3.9810717055368423e+05     1.8376642299234070e-23
6.3095734448051453e+05     7.2002344421267964e-23
1.0000000000002384e+06     1.3844360510239424e-21
1.5848931924612522e+06     8.2447867935178601e-25
2.5118864315097332e+06     1.8906195812299195e-23
3.9810717055351734e+06     8.4013044161565058e-23
6.3095734448022842e+06     2.4460759370357001e-22
1.0000000000000238e+07     6.0508315175955095e-22
1.5848931924611330e+07     1.3563078334928723e-21
2.5118864315096140e+07     2.6606482609752016e-21
"hbnoise.pnoise_hbnoise.data" [readonly] 651L, 32550C
```

Interpolation in Noise Files

All independent sources have a property called Noise file name. This is an ascii file with two columns. The left column is the frequency, and the right column is the noise voltage squared normalized to a 1 Hz bandwidth. This file is read for all the small-signal noise analyses. Between the points in the file, an interpolation is made.

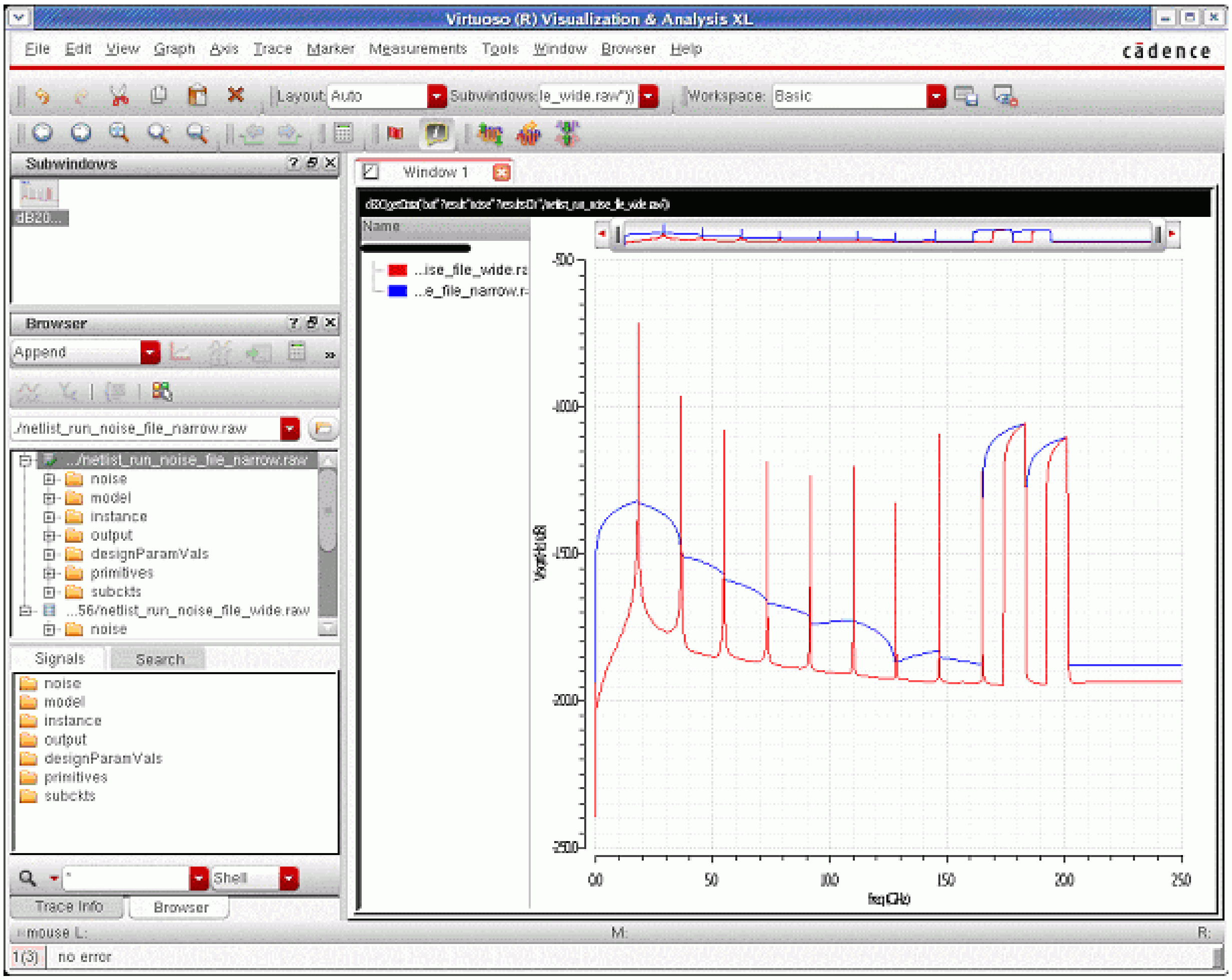


Both curves in the above figure use the `cyclo2txt` output file for a circuit at about 1.85GHz. Just the noise in the file is plotted. The top curve has a noise sweep to just less than one half the oscillation frequency. The bottom curve has a sweep to 10MHz relative to the first harmonic. There were 10 harmonics in the pss simulation and 10 sidebands in the pnoise analysis.

Because the noise analysis was referenced to the first harmonic, the +10 sideband starts at +10*1.85GHz + 1.85GHz, or 20.35GHz, and goes to about 20.36GHz for the 10MHz sweep, and 21.25GHz for the 900MHz sweep. Note that the data near 20GHz in both starts at about -120dBV and drops. The minus 10 sideband starts at -10*1.85GHz + 1.85GHz, or -16.65GHz to -16.64GHz for the 10MHz sweep and -15.75GHz for the 900MHz sweep. Note that there is no data on the bottom side of the 10th and 11th harmonic. The data in between the frequency points in the file is interpolated. That is the reason for the large rise for the last 2 noise spikes.

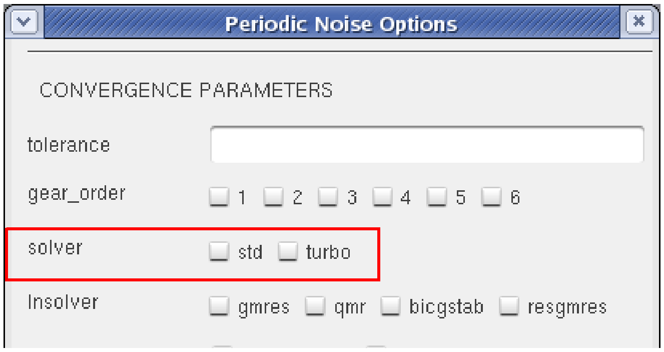
The spikes from the 900MHz sweep is narrower because the data goes to a higher frequency near 17GHz and 19GHz.

The figure below shows the curves overlaid on the same strip.



Note that the interpolation above and below the actual data in the 10MHz sweep (the blue trace) causes much more noise than the real system produces. Make sure that your noise sweep is sufficient for your requirements.

Solver

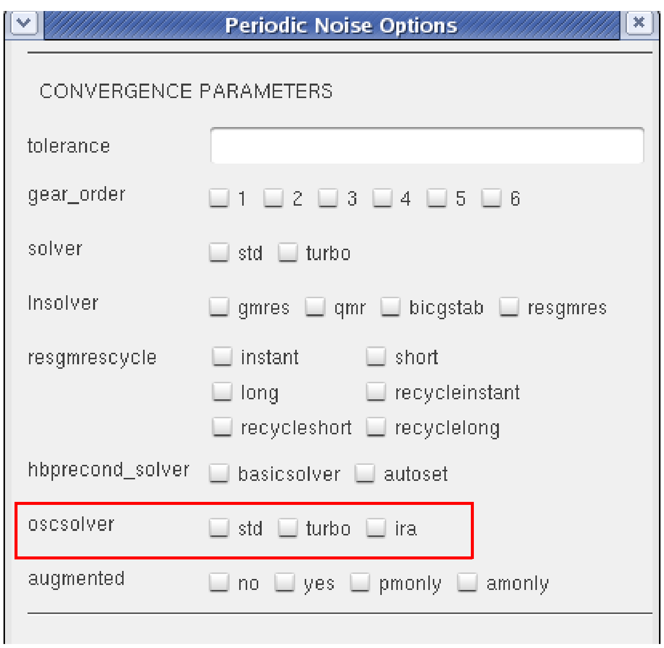


This option is used for driven circuits only.

The default is *turbo*.

For harmonic balance pnoise, leave this option at the default value. For shooting pnoise, sometimes you will see warning messages that the residual is larger than the tolerance specified. This usually occurs at low offset frequencies. If you see this message, then select the *std* solver, which is more able to handle low offset frequencies. The runtime will increase when the std solver is selected.

Oscsolver



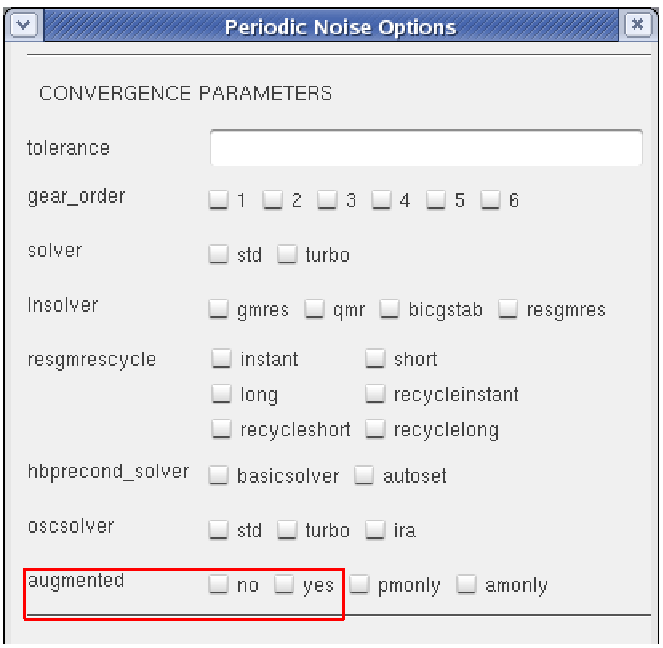
This option is used for oscillators only.

The default is *turbo*.

For harmonic balance pnoise, leave this option at the default value. For shooting pnoise, sometimes, you will see warning messages that the residual is larger than the tolerance specified. This usually occurs at low offset frequencies. If you see this message, then select the std solver, which is more able to handle low offset frequencies. The runtime will increase when the std solver is selected.

The ira solver may use less memory than turbo or std, but it is less robust for convergence.

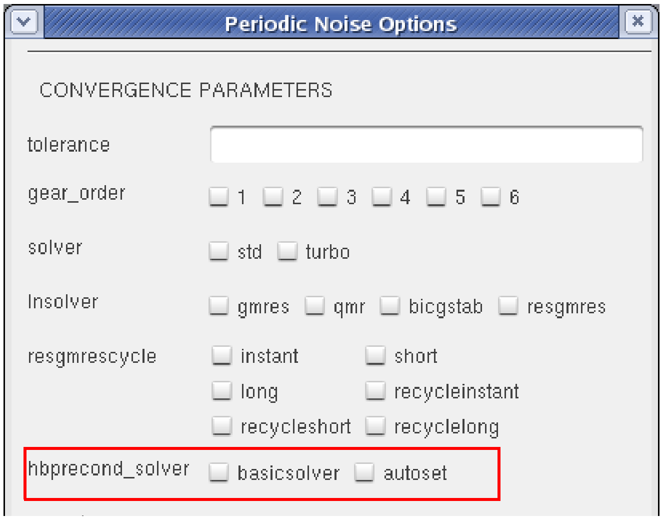
Augmented



Leave this option at the default. (augmented=yes)

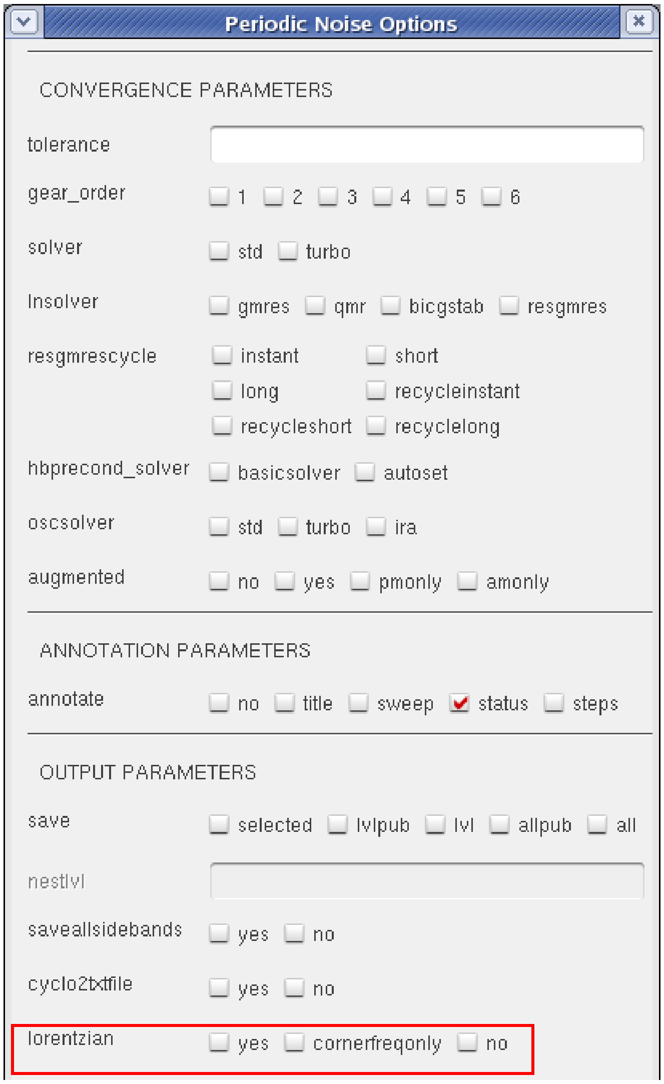
If your circuit has low frequency poles, (usually from a biasing circuit) the default augmented solver is much more accurate than the standard solver. Pmonly and amonly should not be selected.

hbprecond_solver



This option is only available when harmonic balance is set for the pss solver. When APS is not used, only the basic solver is available. When APS is used, first a much faster preconditioner is used. Occasionally, this preconditioner will stagnate, and when it does, it issues a warning that it is switching to the basic solver, and it reverts back to the basic solver. A preconditioner is a mathematical algorithm that makes the iterative matrix solving process faster.

Lorentzian

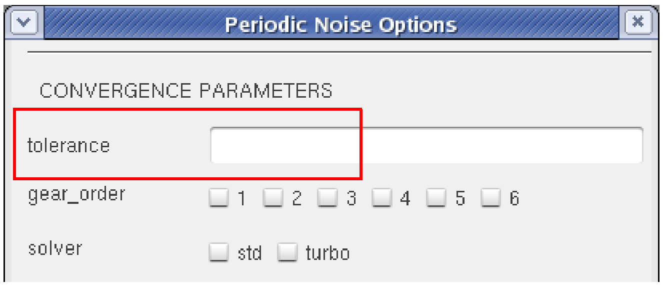


There are two ways of looking at phase noise. One way is to think of measuring the spectral output of the oscillator with a network analyzer. In this case, as the frequency gets close to the oscillator frequency, the amplitude starts going up, levels off, and then drops as the oscillator frequency is passed. In this case, the noise cannot be larger than the oscillations themselves, which is 0 dBc. If this is how you view phase noise, set the *lorentzian* option to *yes*.

If you think in terms of jitter, in one cycle of the oscillator output, because of the noise, you can calculate one standard deviation of the timing. If you go two cycles, and you integrate the noise with respect to time, you get twice as much noise power, or about 1.4 times the noise voltage or jitter. When you get to an infinite number of oscillator cycles, the jitter also becomes infinite. As you increase the time, you are lowering the noise frequency, and in this case, as the time becomes large, the frequency becomes very low. At infinite time, you get infinite jitter, and this occurs at zero offset from the carrier. Jitter is just another way of looking at phase noise. In other words, the phase noise can easily exceed the carrier amplitude (0 dBc) as the offset frequency becomes small because it is heading to infinity at zero frequency offset. If this is how you view phase noise, set the *lorentzian* option to *no*.

Lorentzian=yes means calculate the leveling off. Lorentzian=cornerfreqonly calculates the continuously rising phase noise, but it places a marker on the phase noise curve at the frequency where the phase noise would level off. No causes the phase noise to continue to rise as the offset frequency approaches zero.

Tolerance



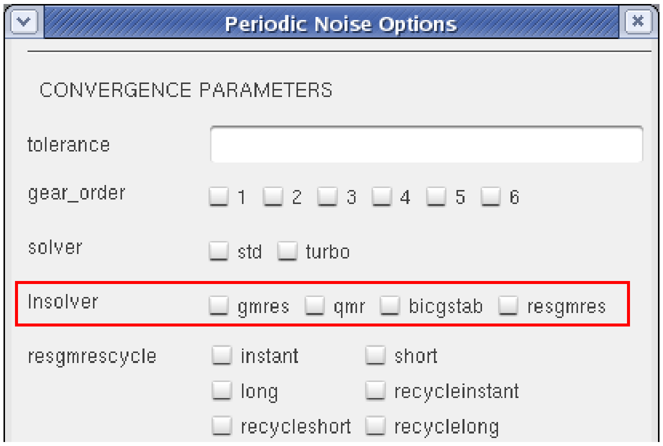
Leave this option at the default.

Pnoise uses an iterative solver to calculate the noise, and any iterative solver needs a tolerance term to specify when the system has been solved accurately enough. The default for shooting pnoise is 1e-9. Harmonic balance driven circuit pnoise tolerance is 1e-6 and 1e-4 for oscillators.

Gear_Order

Do not set this option.

Lnsolver



Leave this option at the default, which is *gmres*. To calculate the conversion gain from the noise sources to the output, many matrices are formed and solved to calculate the noise at the output of the system. These matrices are solved using an iterative solver. Several different algorithms are provided for the iterative solver. Gmres is the default because the accuracy of each iteration inherently increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, seek graduate-level textbooks that describe linear algebra theory.

Gmres is the Generalized Minimum RESidual method.

Qmr is the Quasi-Minimal Residual method.

Bicgstab is the STABilized BI-Conjugate Gradient method.

Resgmres is the REStarted Generalized Minimal Residual method.

Resgmrescycle

Leave this option at the default, which is *short*. For the resgmres solver, there are several different options.

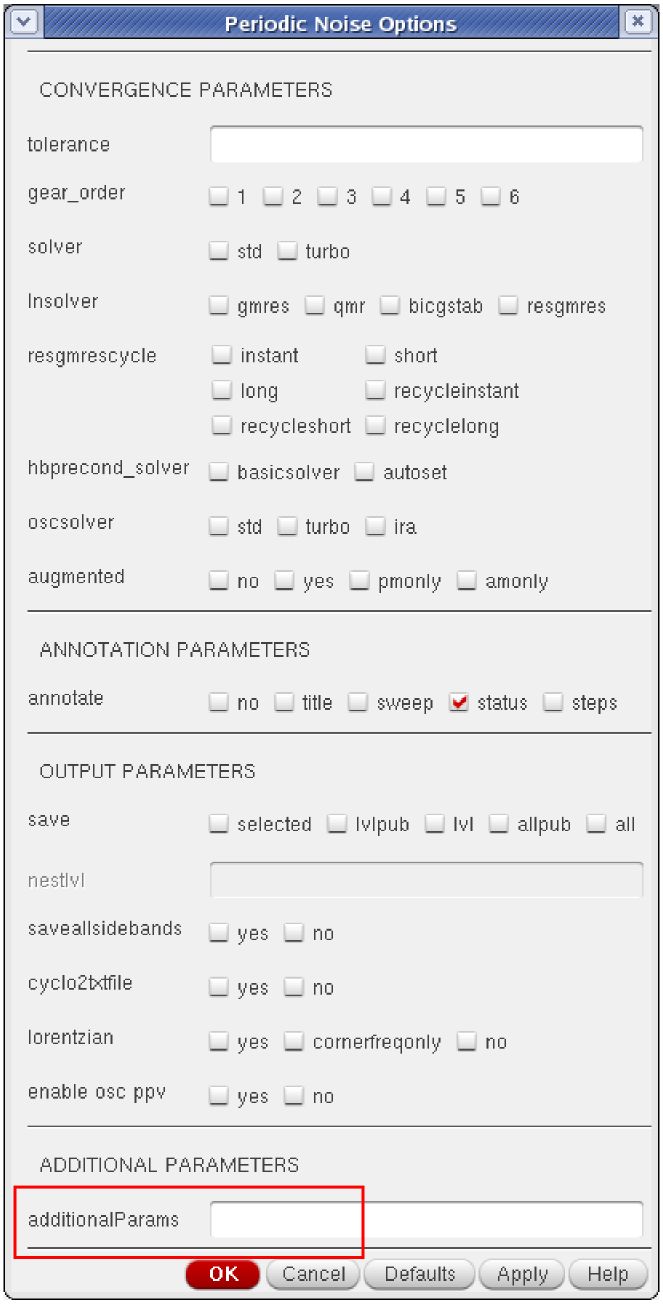
Saveallsidebands

Leave this option at the default value of *no*. If you want to see the noise contributors from the different noise input frequencies, select *yes* for noise separation in the pnoise main *Choosing Analyses* form.

Enable Osc PPV

This option is used for the PLL noise methodology only, and does not need to be set as long as you use the PLL Noise Wizard from ADE.

AdditionalParams



This is typically used for customers who are evaluating new features. If you know the option name and the value, you can type it here. For example, you can type solver=std in this field instead of using the check box in the options form.

ADE Implementation

General Notes

All pnoise types are supported whether the pss engine is shooting or harmonic balance. Because Chapter 3 in this manual talks about harmonic balance exclusively, in this Chapter most of the examples will reference shooting pss. Where there are differences between shooting and harmonic balance, those differences will be noted. Otherwise, shooting and harmonic balance are completely interchangeable, and will provide the same answer as long as enough harmonics are used in harmonic balance.

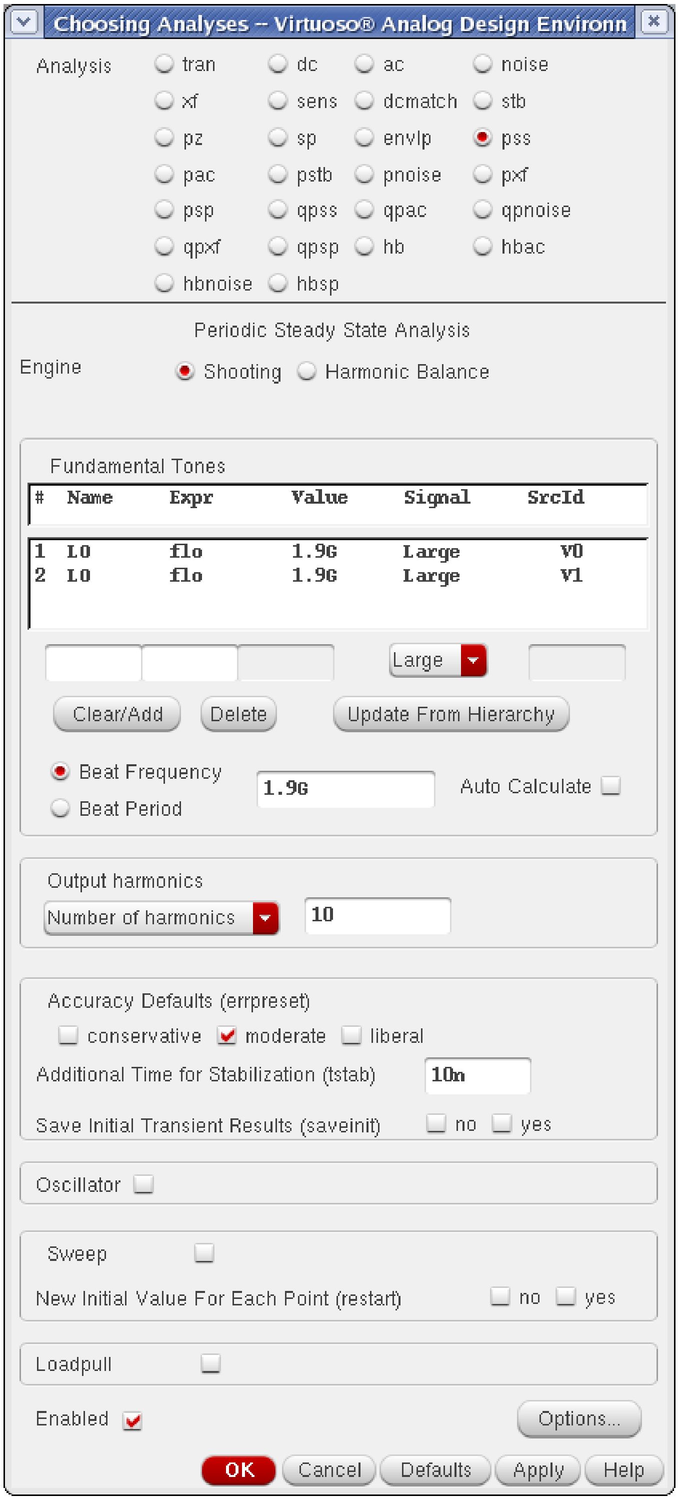
Because shooting takes a minimum of 200 timepoints in the pss, the pss data inherently contains data through the 100th harmonic of the pss. For this reason, unless there is a requirement to calculate noise translations through more than the 40th harmonic of the pss, it is seldom required to force more harmonics in shooting pss. Also with shooting pss, fullspectrum is available, which does not require setting sidebands at all in most cases. However, if the pss beat frequency is 100KHz or less, then set maximum sideband to the 1/F noise corner frequency divided by the pss beat frequency. In harmonic balance, only the harmonics that are solved in the pss exist. When harmonic balance is selected, if you set maximum sideband larger than the number of pss harmonics, a warning message will be entered in the Spectre output log, and maximum sideband will be reset to the number of pss harmonics.

Driven Circuits

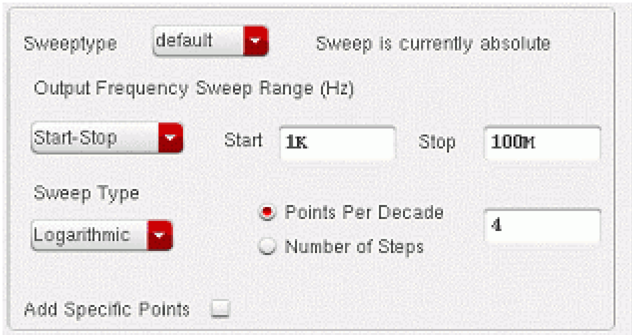
Noise Type = Sources

On the *Choosing Analyses* form:

1. First, set up the pss analysis. For more information on the pss analysis, see the beginning of this chapter.



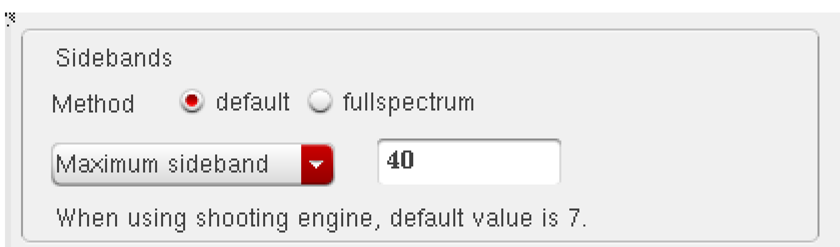
2. In the pnoise Analysis selection, specify the output frequency range of your circuit.

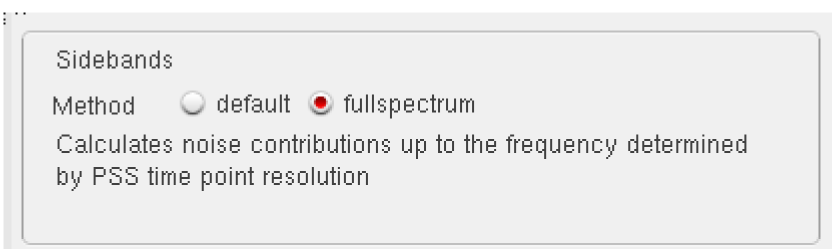


Note: Since the fundamental quantity that is calculated by the noise analysis is the noise at the output, the frequency range is always the output frequency range.

3. Set about 3 to 5 points per decade in the *Points Per Decade* field.

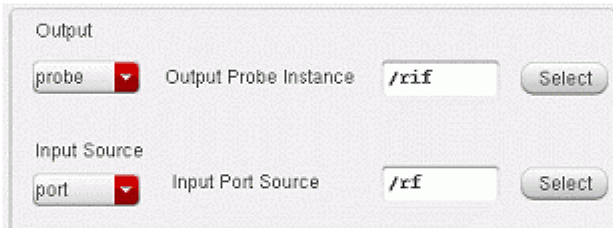
4. When hb is selected as the engine in the pss form, leave the *Maximum sideband* field blank or set it to the same number as the number of harmonics in the HB analysis. This specifies that the noise that mixes with all the hb harmonics is calculated. When the pss engine is set to shooting, first decide whether or not to use full-spectrum pnoise, and if the default method is used, set a number of sidebands that is appropriate for your circuit. APS is selected automatically and is required when using full-spectrum pnoise.





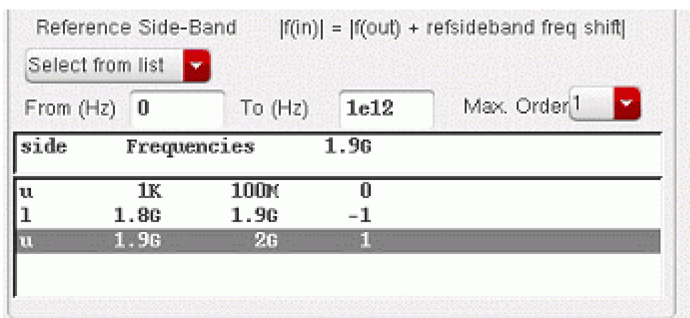
5. Specify the output of the circuit. When output is set to *probe*, a resistor or a port can be selected and this noise will be subtracted from the noise figure calculation. This is an easy way to get the correct answer for a noise figure calculation.

   The nodes that the probe is connected to also define the output nodes for the circuit. If you connect the probe across a differential circuit, the noise measurement is for that differential circuit. Also note that when you select a probe, it is only used in the noise figure calculation. The output noise includes the noise of the selected element.



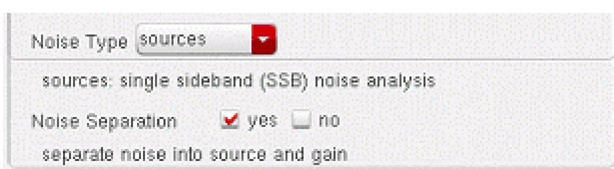
6. Select *port* from the *Input Source* drop-down list for noise figure calculation.

7. Choose *Select from list* from the drop-down list for reference sideband, which specifies the input frequency range (also called the passband frequency range) for the noise figure calculation.



8. Select the passband frequency range from the list.

9. Select *sources* from the *Noise Type* drop-down list to calculate the average noise power at the output averaged over the entire time of one cycle of the input waveform.



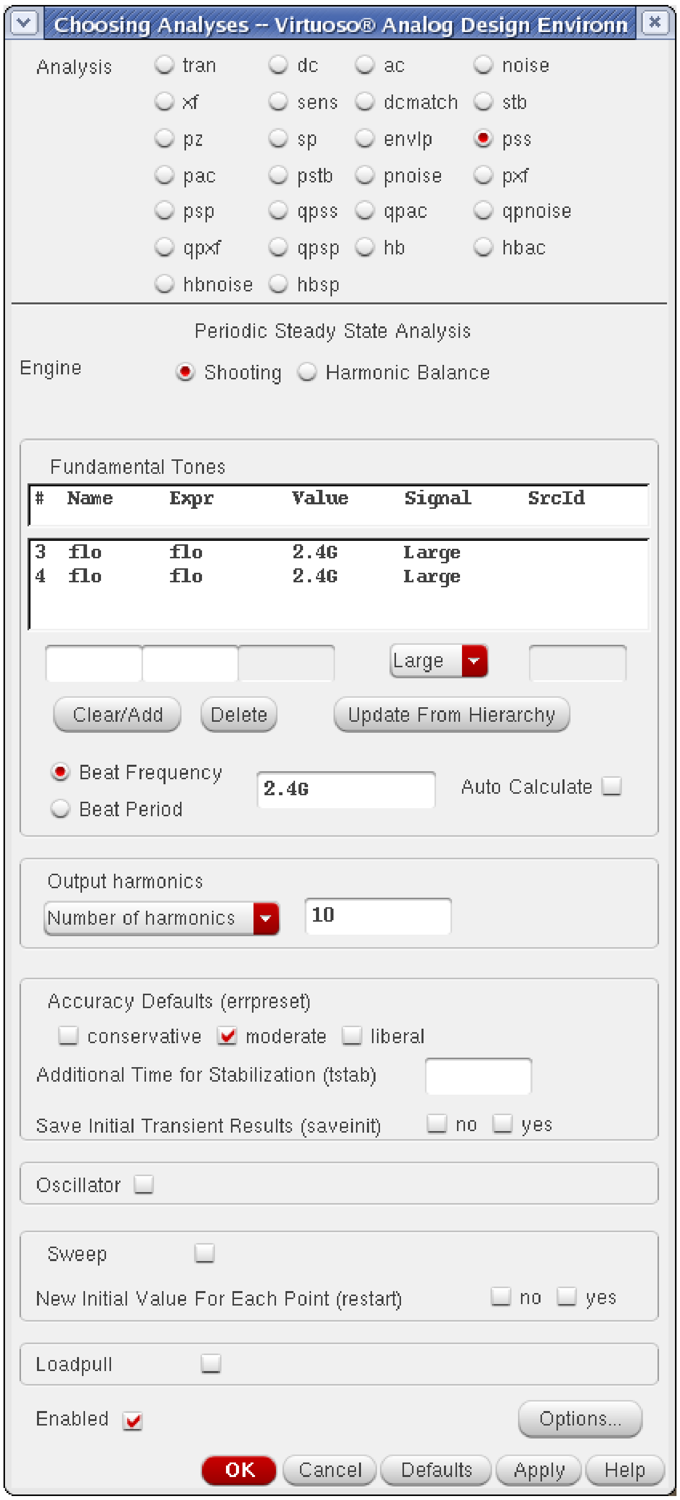
10. Select *yes* for *Noise Separation* to enable plotting of the individual noise contributors in the circuit. Noise separation is only available when *Noise Type* is set to *sources*.

Multiple pnoise

When *Noise type* is set to *sources*, multiple pnoise analyses can be run after a single pss analysis. This is useful for LNA-Mixer combinations, or in any circuit where there are multiple points in the circuit where noise is important.

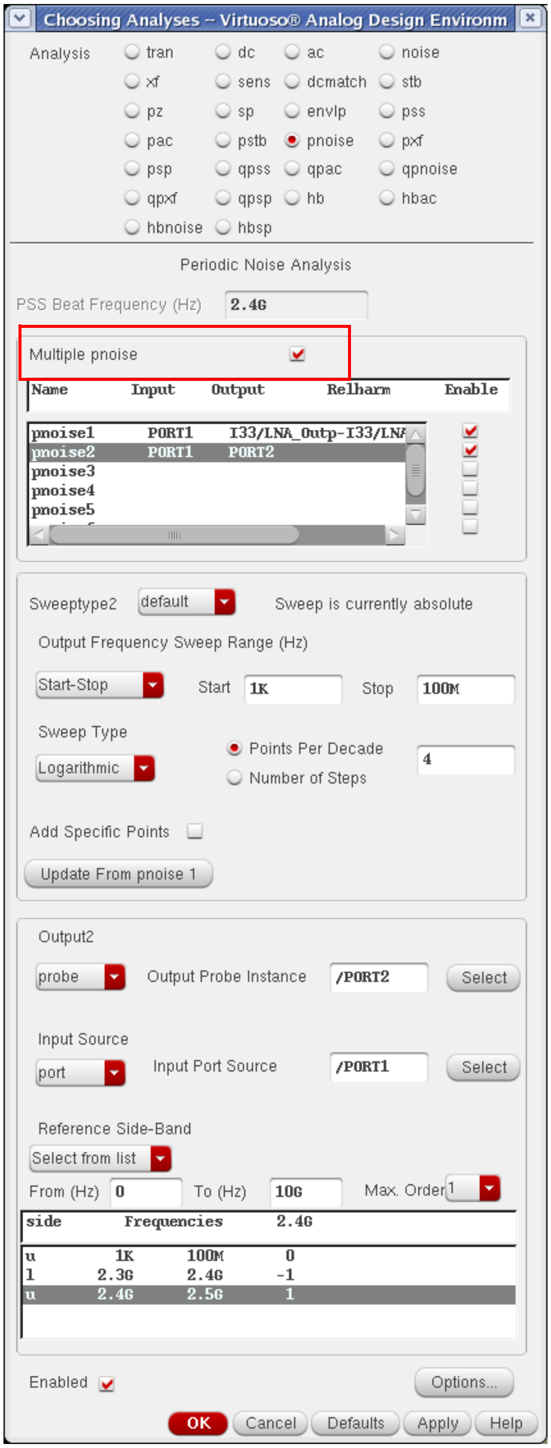
To run multiple analyses after a single pss:

1. First, set up the pss analysis as appropriate for your circuit. For more information, lease see the pss section at the beginning of this chapter.



2. Select the *Multiple pnoise* check box located at the top of the pnoise *Choosing Analyses* form.

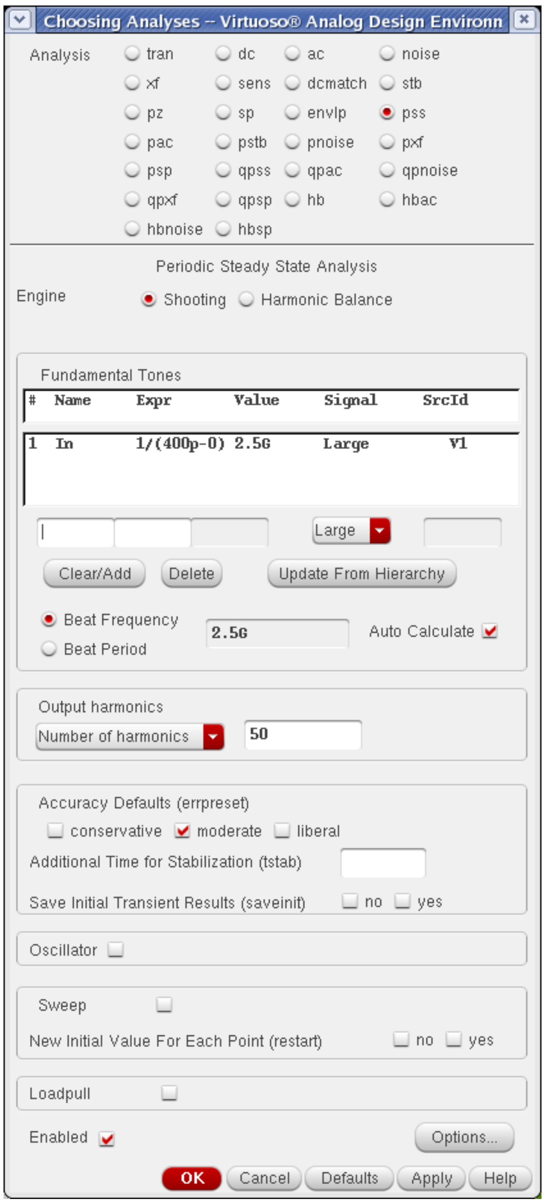
3. Set up all the noise analyses you want to run, as shown below.



Noise Type = Modulated

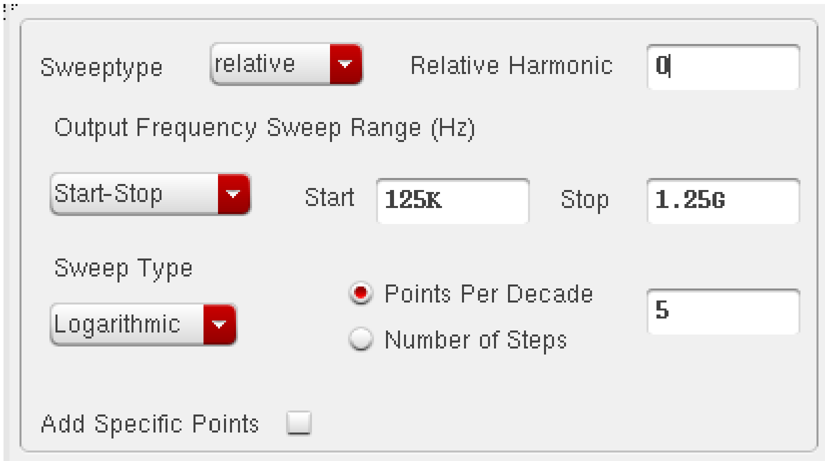
Modulated allows the measurement of the AM and PM components of noise separately.

First, set up the pss analysis as appropriate for your circuit. For more information, see the periodic steady-state section at the beginning of this chapter.

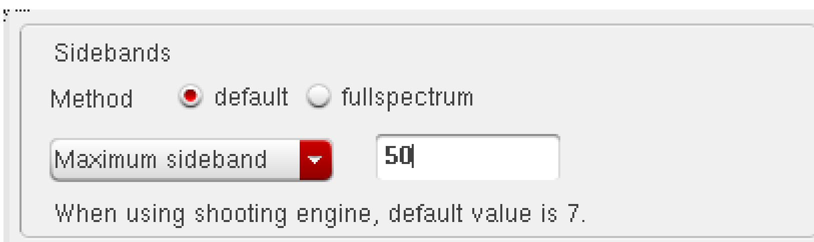


On the pnoise *Choosing Analyses* form:

1. Set up an appropriate frequency range of interest.

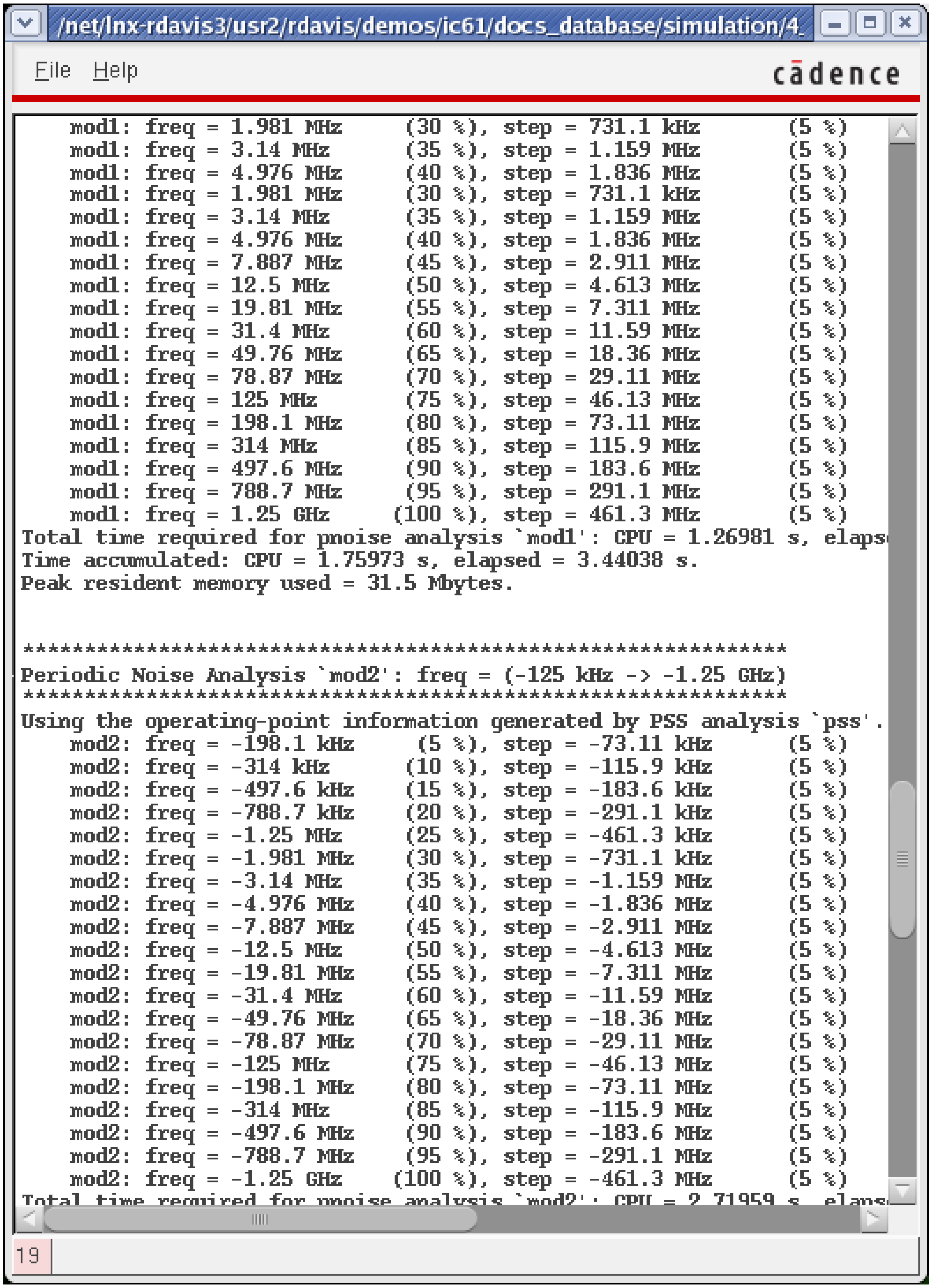


2. Set an appropriate number of sidebands. When harmonic balance is selected for the pss analysis, this field should be left blank. Fullspectrum is available for shooting only, and will be faster than the default pnoise when the number of noise sidebands is about 50 or greater with no loss in accuracy.



3. Select *modulated* from the *Noise Type* drop-down list.

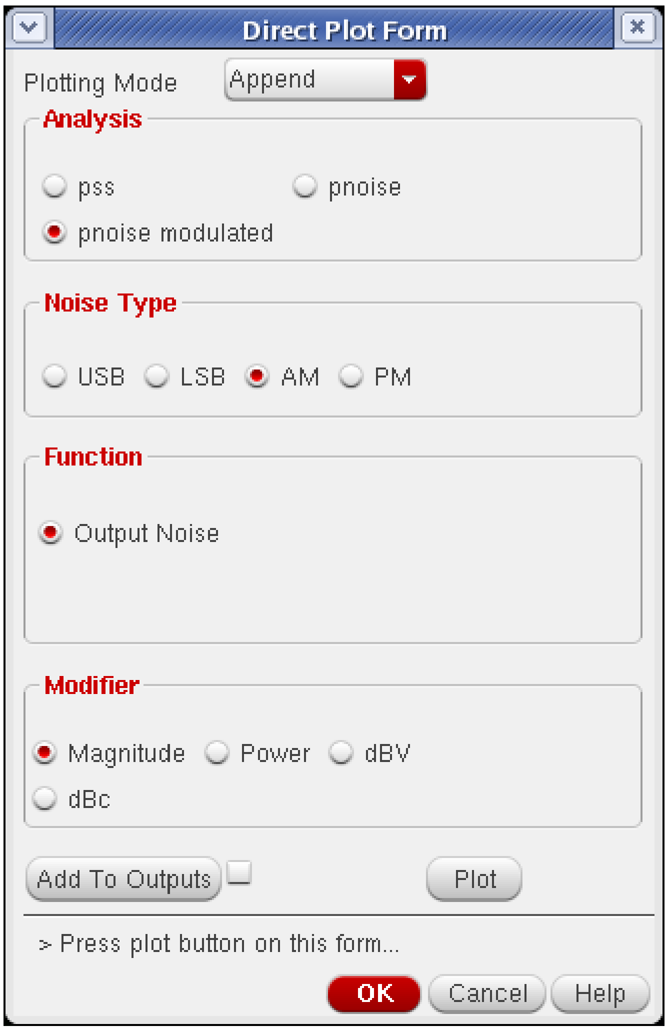
4. Run the simulation. Note that modulated runs both the positive and negative frequencies. This is required to calculate the AM and PM components.



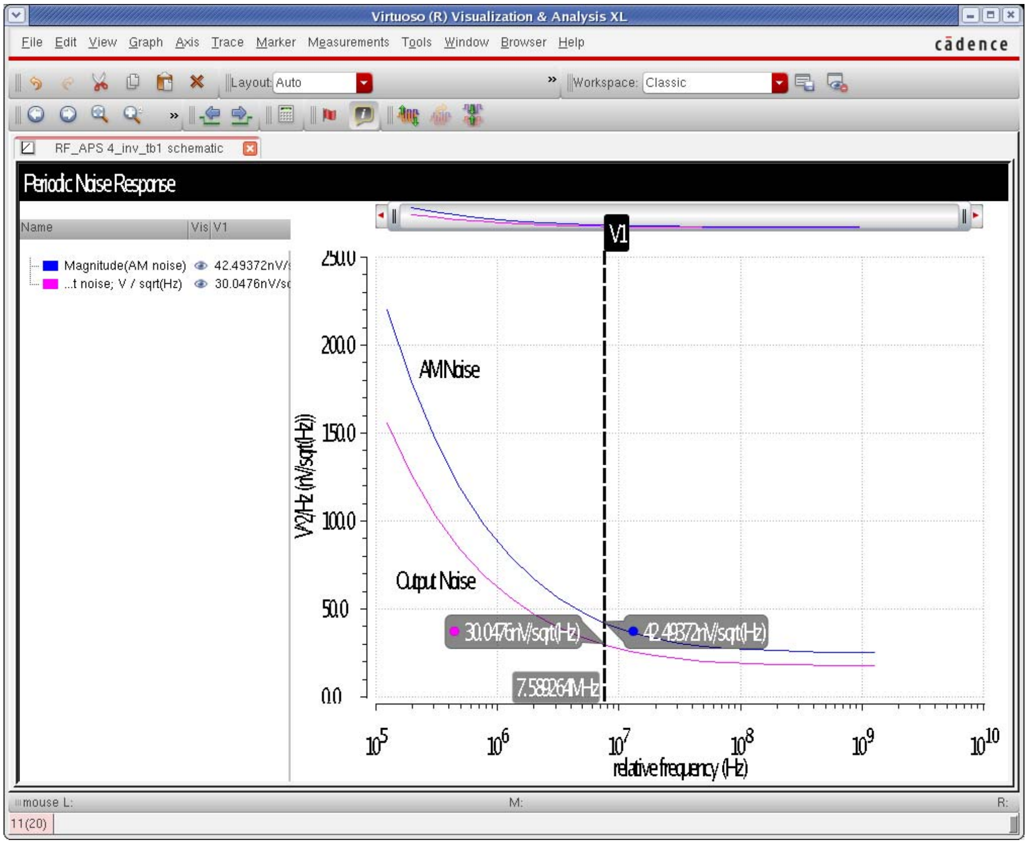
Now plot the Output noise using the Direct Plot Form.

On the *Direct Plot Form*:

1. Select *pnoise modulated* from the *Analysis* section.
2. Select *AM* or *PM* from the *Noise Type* section. *AM* is selected here.
3. Select *Magnitude* or *dBc* in the *Modifier* section.



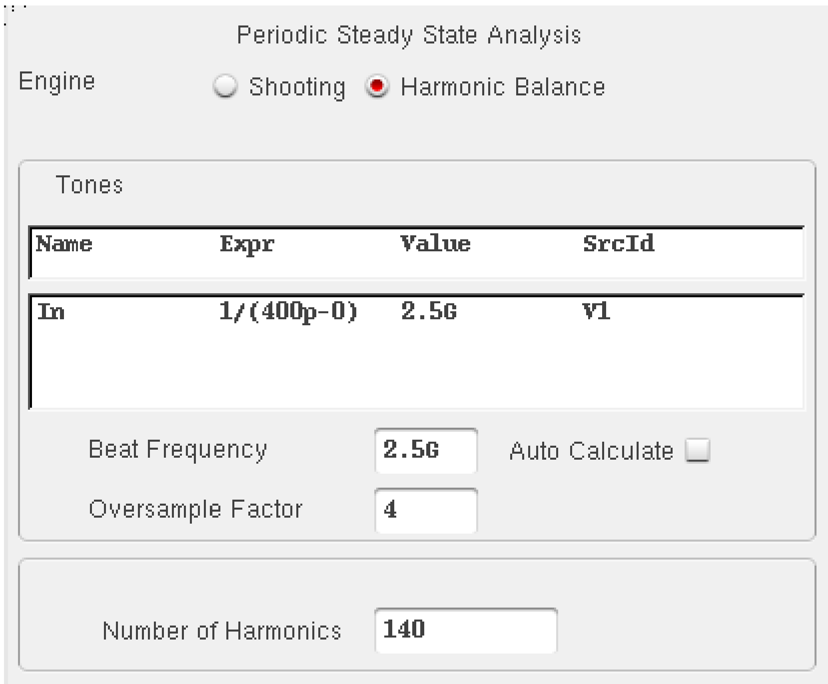
4. Click *Plot*.



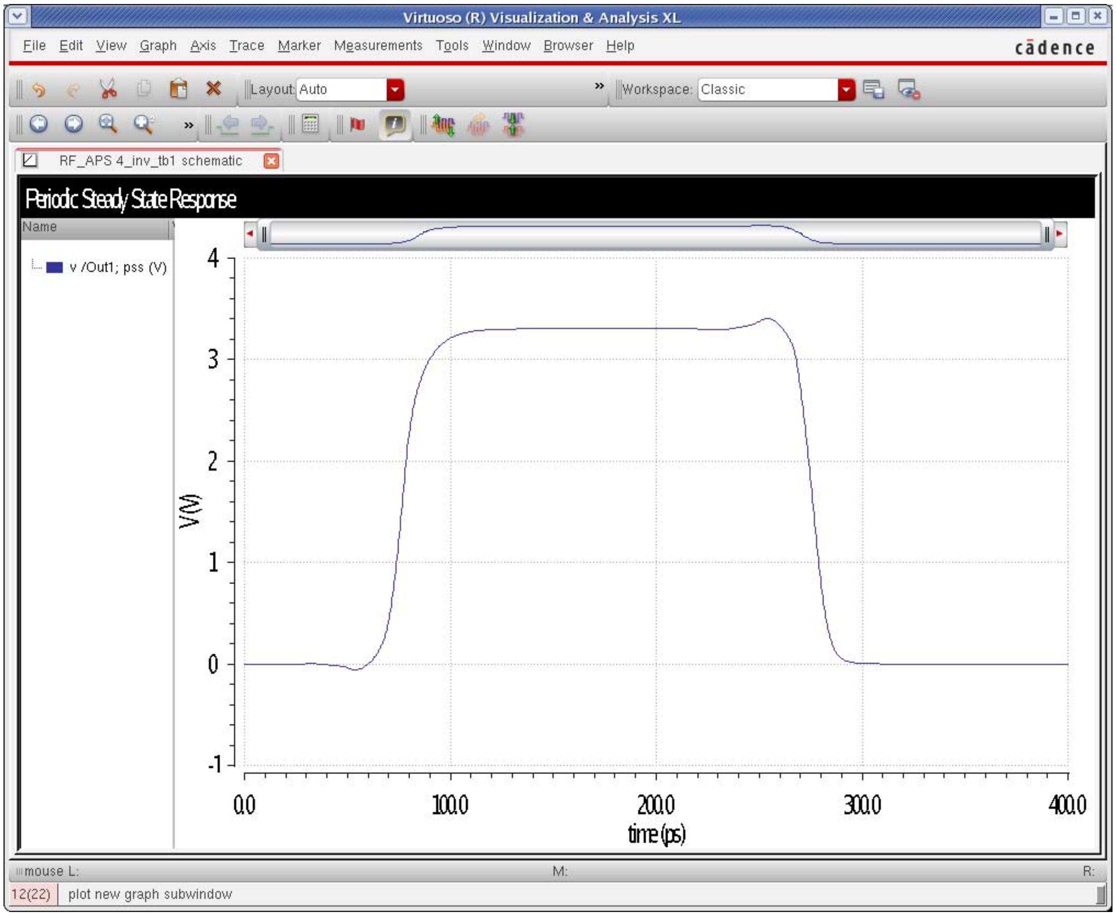
On the plot, note that the AM noise is 3dB larger than the output noise. This is because the output noise is derived from the positive frequencies only (single-sideband), and the AM and PM noise is derived using the positive and negative frequencies (double-sideband).

Noise Type = Jitter

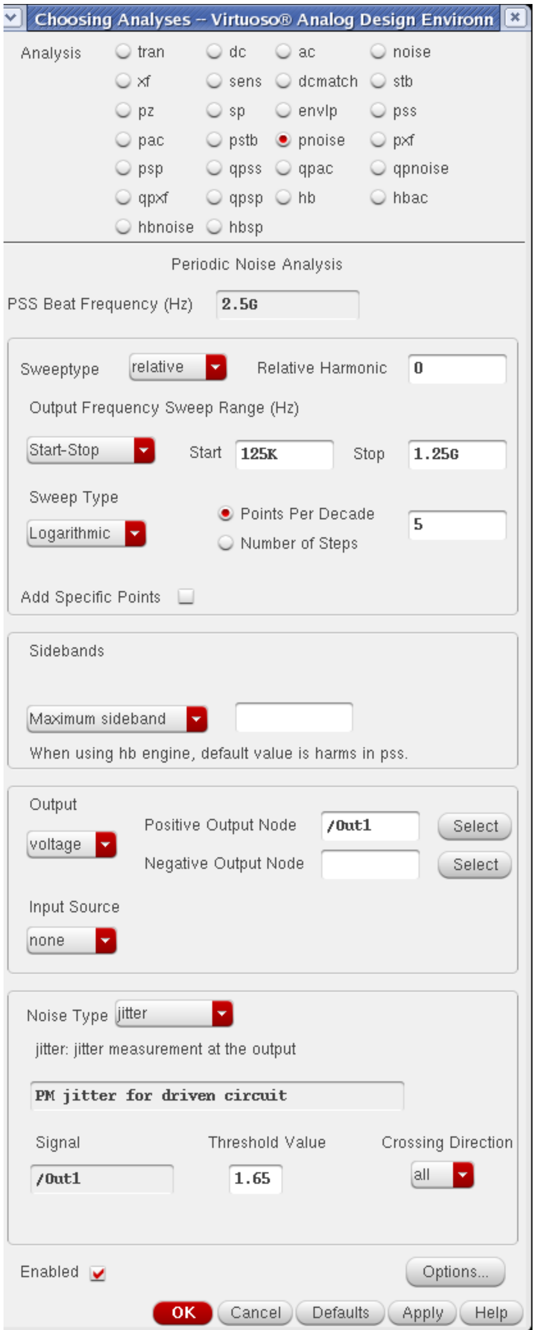
Driven jitter is usually used for measuring the noise in a digital circuit at a specific threshold crossing. Usually, shooting pss is used to capture the waveform, but this capability is also implemented in harmonic balance. This shows the harmonic balance example. The pss *Choosing Analyses* form has a lot of harmonics and the oversample factor is set to 4, as shown below. Because the digital output is usually a square wave, a high number of harmonics and a large oversample factor need to be set. For more information, please see the harmonic balance section at the beginning of Chapter 3.



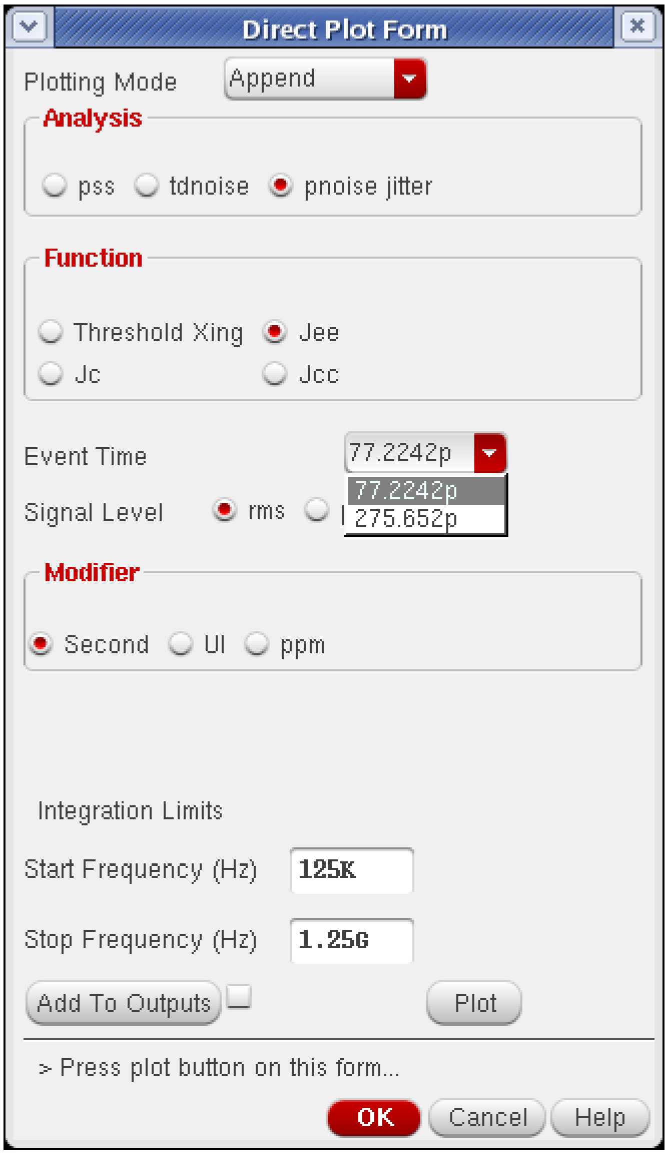
An appropriate threshold is needed, which can be determined by plotting the time domain waveform from the pss *Direct Plot Form*. The waveform below shows an appropriate threshold value to be 1.65V.



On the pnoise *Choosing Analyses* form, the stop frequency is half the input frequency. The start frequency is 4 decades below that. Log sweep with about 5 points per decade is usually sufficient. *Maximum sideband* should be left blank when harmonic balance is selected for the pss engine. When the pss engine is set to shooting, the number of sidebands should be about two to four times the number of pss harmonics. The *Output* field is set to *voltage*, and the *Input Source* field is set to *none* because usually, only the output jitter is needed. The threshold is set appropriately. The *Crossing Direction* option can be set to *all*, *rise,* or *fall*.

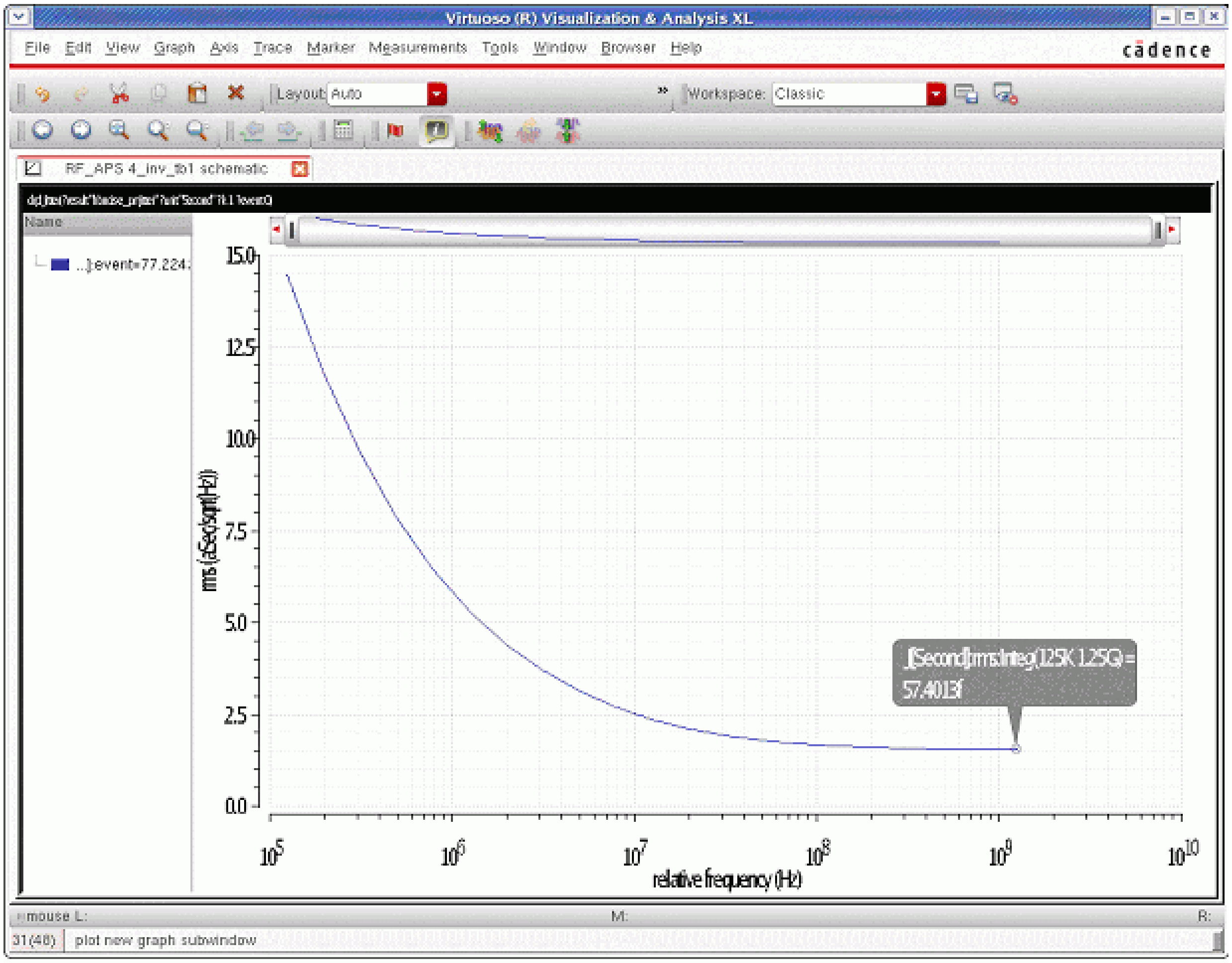


When the simulation completes, use the *Direct Plot Form* to plot the jitter. For driven circuits, select *Jee*, which is an edge-to-edge jitter. The event time is the time in the time-domain waveform for the large-signal pss simulation. In the below example, in the *Event Time* selection, 77p is the rising edge, and 275p is the falling edge. If you are in doubt, plot the time-domain waveform from pss and go to the time listed in the *Event Time* field.



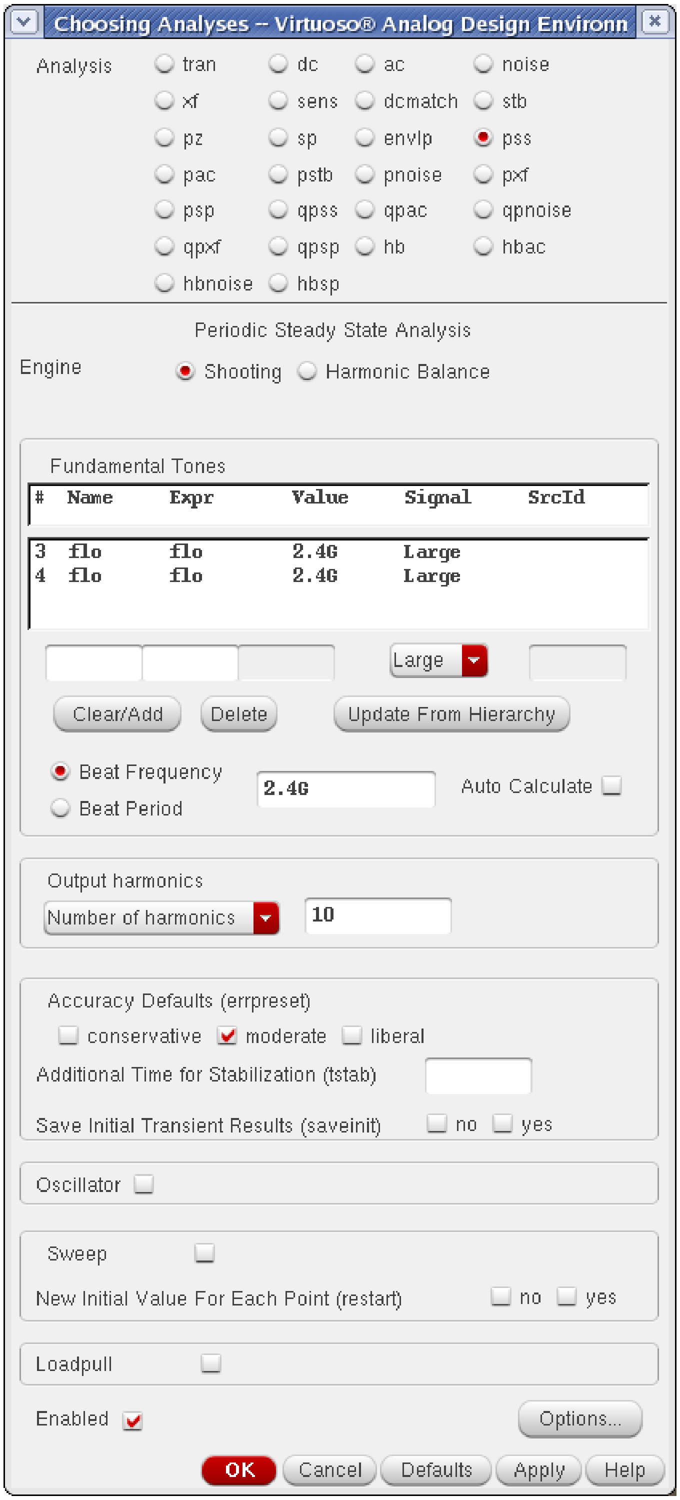
In the Modifier section, *UI* is unit interval, and *ppm* is parts per million.

The output noise frequency response and the edge-to-edge jitter are displayed. The jee measurement is one standard deviation for the expected jitter.

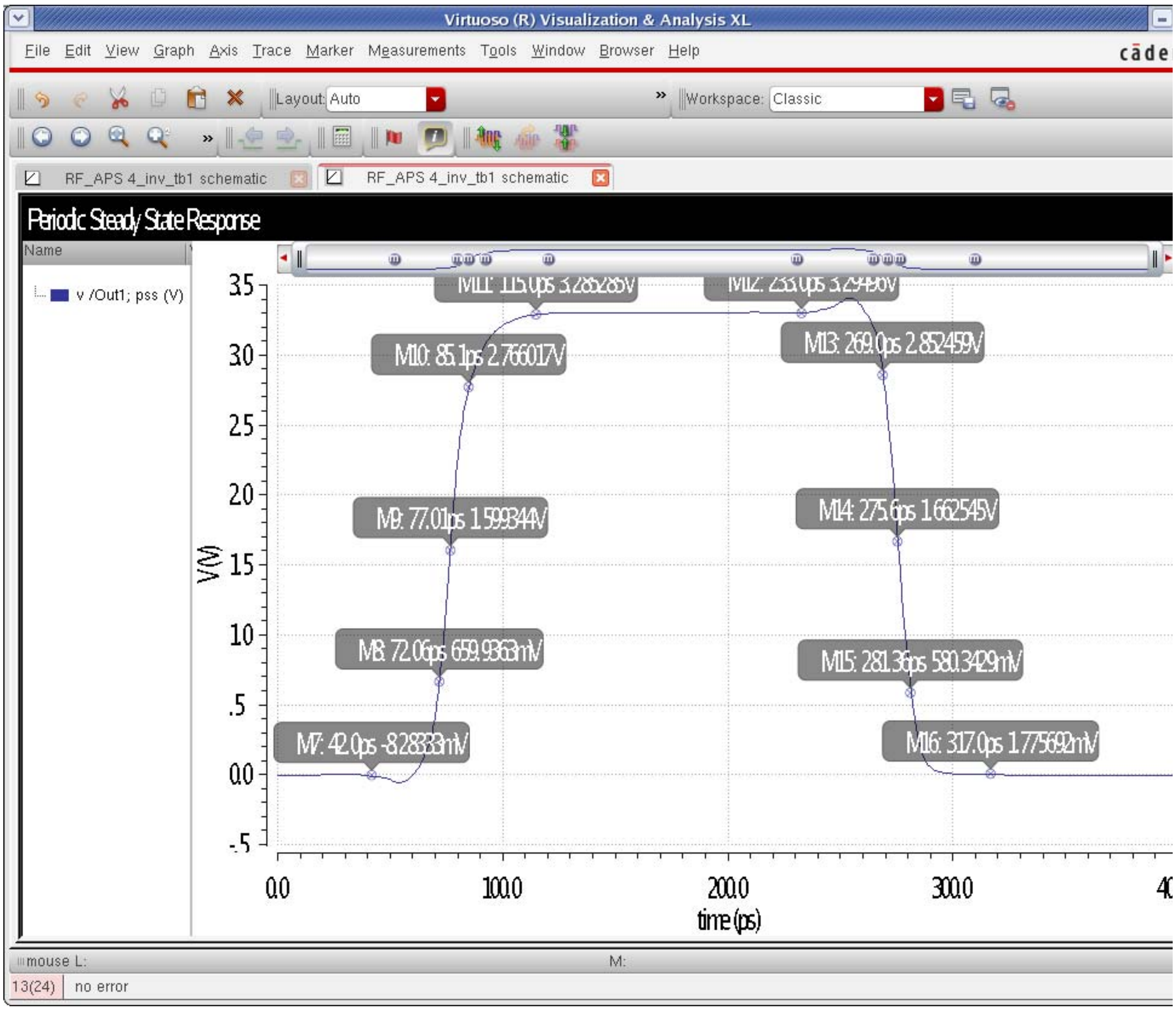


Noise Type = Timedomain

1. First, set up the pss analysis as appropriate for your circuit. For more information, see the periodic steady-state section at the beginning of this chapter.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Steady State Analysis
Engine
Shooting
Harmonic Balance
Fundamental Tones
Name Expr Value Signal SrcId
3 flo flo 2.4G Large
4 flo flo 2.4G Large
Large
Clear/Add
Delete
Update From Hierarchy
Beat Frequency
2.4G
Auto Calculate
Beat Period
Output harmonics
Number of harmonics
10
Accuracy Defaults (errpreset)
conservative
moderate
liberal
Additional Time for Stabilization (tstab)
Save Initial Transient Results (saveinit)
no
yes
Oscillator
Sweep
New Initial Value For Each Point (restart)
no
yes
Loadpull
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

2. For timedomain noise, first plot the time-domain waveform from the pss analysis, and place markers at the different voltages of interest in the waveform as shown below.

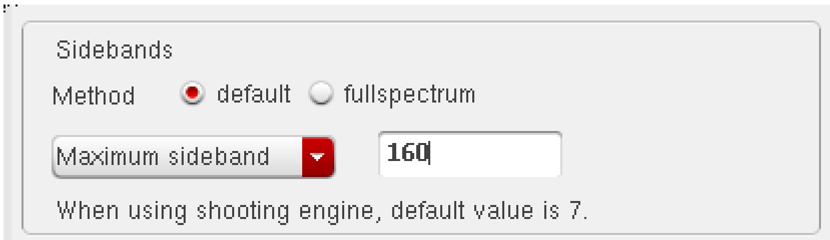


On the pnoise *Choosing Analyses* form:

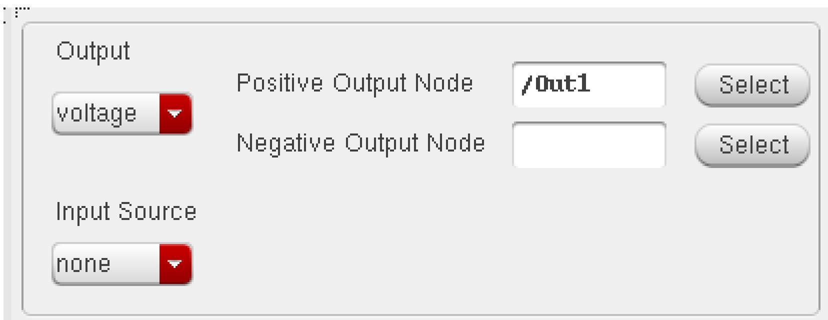
1. Set the stop frequency to half the pss frequency.

2. Set the start frequency to 4 decades smaller than the stop frequency.

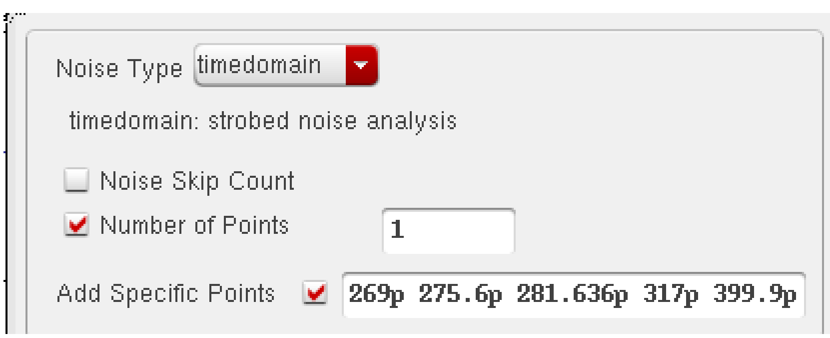
3. Leave the *Maximum sideband* field blank when using harmonic balance as the pss engine. When shooting is used, specify two to four times the number of sidebands as harmonics in the pss analysis. Shooting is shown below.



4. Select *voltage* from the *Output* drop-down list.



5. Specify the output node in the *Positive Output Node* field. When the negative node is left blank, the global ground node is used as the reference node.
6. Select *none* from the *Input Source* drop-down list.
7. Select *timedomain* from the *Noise Type* drop-down list.



8. Select the *Number of Points* check box and specify 1 in the text field. This tells time-domain noise to run only at the first timepoint.

9. Select the *Add Specific Points* check box and specify the time values for the desired instantaneous noise measurements. Separate the entries with a space. This list of times will be analyzed as well as the time=zero point.

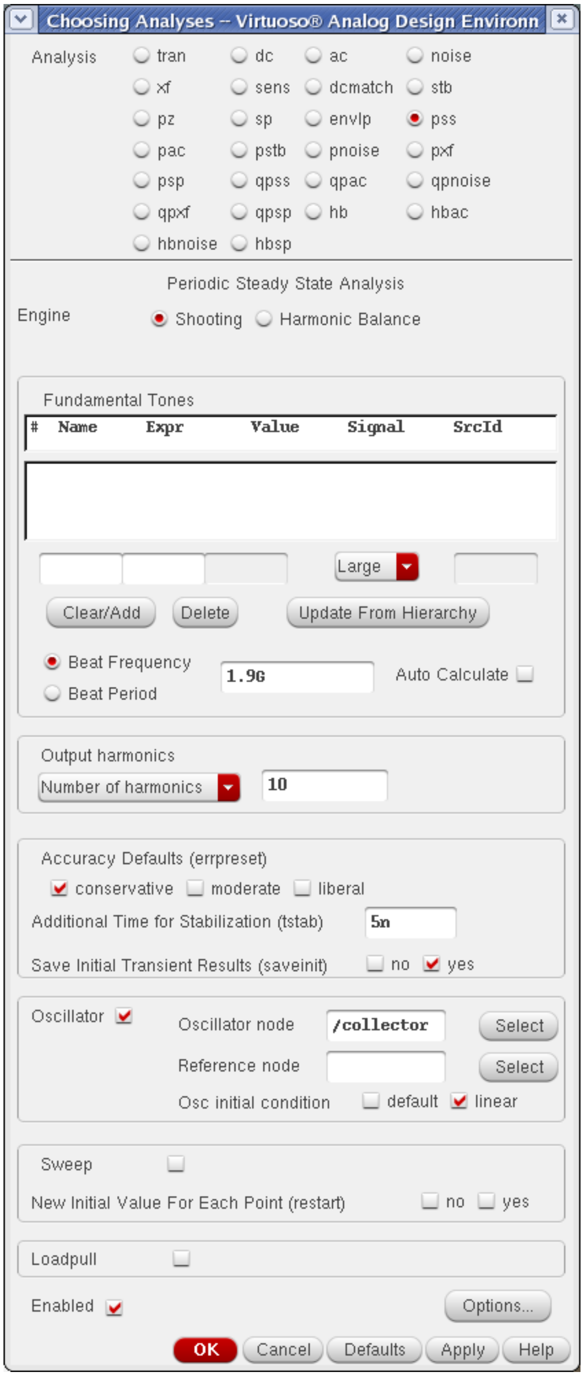
Oscillators

Noise Type = Sources

The settings are the same as driven `noise type = sources` with the exception of being a relative sweep by default. Since the phase noise of the oscillator is near one of the harmonics, *Sweeptype* is set to *relative* by default. A harmonic number is provided for the case where there is a frequency multiplier or a frequency divider on the output of the oscillator. Noise type = sources also has the ability to plot the noise contributors from mixing with each harmonic when noise separation is selected. There is an example of noise separation later in this chapter.

The pnoise setup is similar to driven except that the relative harmonic is specified. The frequency of the relative harmonic number will be added (if the harmonic number is positive) or subtracted (if the harmonic number is negative) to the frequencies in the Frequency Sweep Range field.

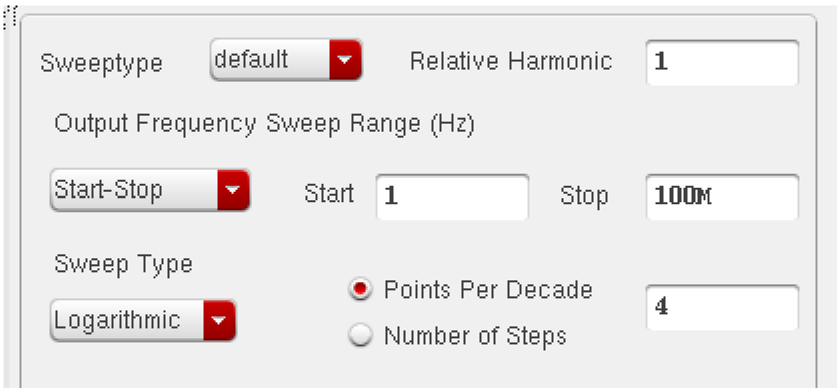
1. First, set up the pss analysis as appropriate for your circuit. For more information on the pss analysis, see the periodic steady-state analysis section at the beginning of this chapter.



On the pnoise *Choosing Analyses* form:

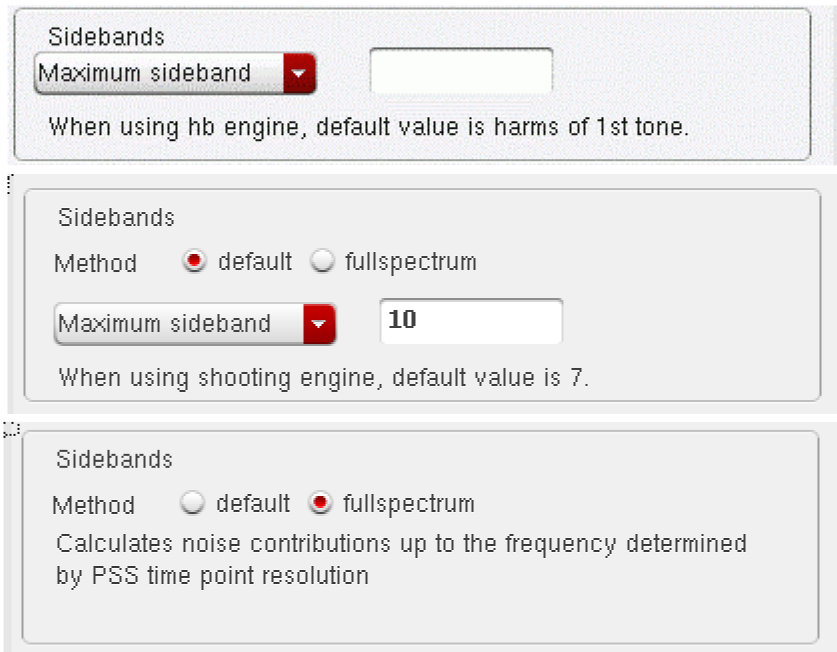
2. Specify the appropriate harmonic number for your design. In this example, the first harmonic is specified in the *Relative Harmonic* field. If you are designing an oscillator by itself, the harmonic number is one. If you have an oscillator with a frequency doubler on the output, the harmonic number is two.

3. The frequency range is added to the frequency of the harmonic number that is specified.

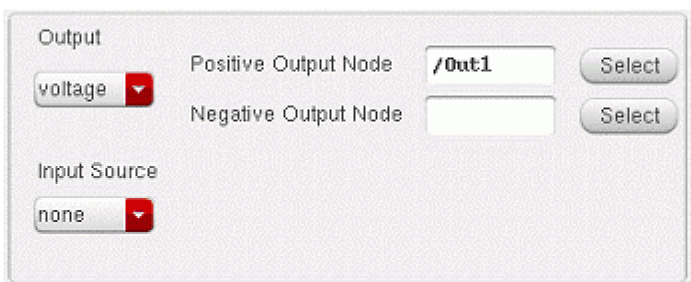


4. If the pss engine is harmonic balance, leave the *Maximum sideband* field blank. If the pss engine is shooting, select whether the default method or the full-spectrum method should be used. Full-spectrum is faster than the default method when more than about 50 sidebands are required for your circuit and it also does not require the maximum sideband field to be set. When the default method is selected, specify the number of desired frequency translations in the *Maximum sideband* field. This will always be specific to the design. For a rough estimate, plot the frequency domain content, and find the harmonic number where the amplitude has dropped about 40dB. Try this number in maximum sidebands. To verify that you have enough sidebands, raise the number by about 50% and run again. If the noise result did not change, then you had enough sidebands to begin with. Up to 40 sidebands can be specified by default. If you need more

than 40 sidebands, set the number of pss harmonics to the number of sidebands divided by two to four.



5. Select *voltage* from the *Output* drop-down list.

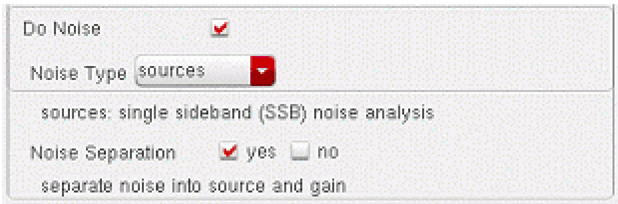


6. Specify the output node in the *Positive Output Node* field. When the negative node is left blank, the global ground node is used.

7. Select *none* from the *Input Source* drop-down list.

8. Select *sources* from the *Noise Type* drop-down list. This gives a noise measurement that is the average noise power at the output with the large signal applied in the pss

analysis and the noise folding specified by the maximum sideband parameter in the *Choosing Analyses* form.



9. Select *yes* for *Noise Separation,* if you want to plot the individual noise contributors. Fullspectrum does not support noise separation.

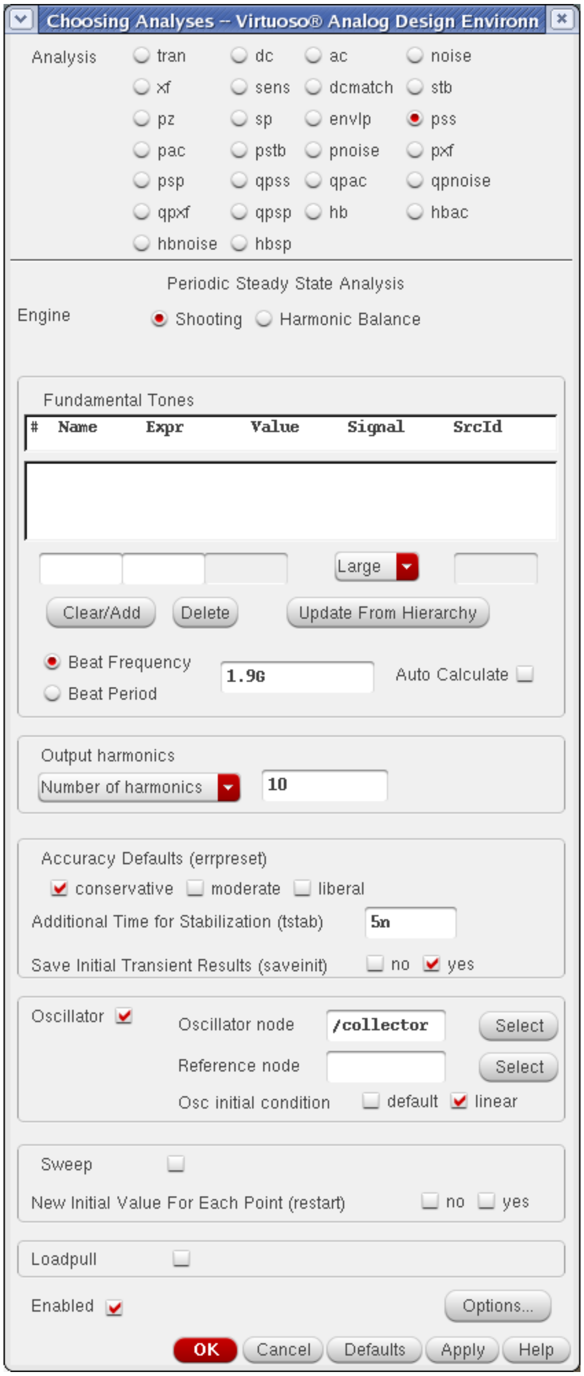
Noise Type = Modulated

This is the same as driven noise type = modulated (see Noise Type = Modulated on page 881). The AM and PM components of phase noise are available for plotting. Generally jitter is the preferred setting because it calculates the modulated result and it allows the calculation of jitter for the oscillator output. For examples, see Oscillators on page 998.

Noise Type = Timedomain

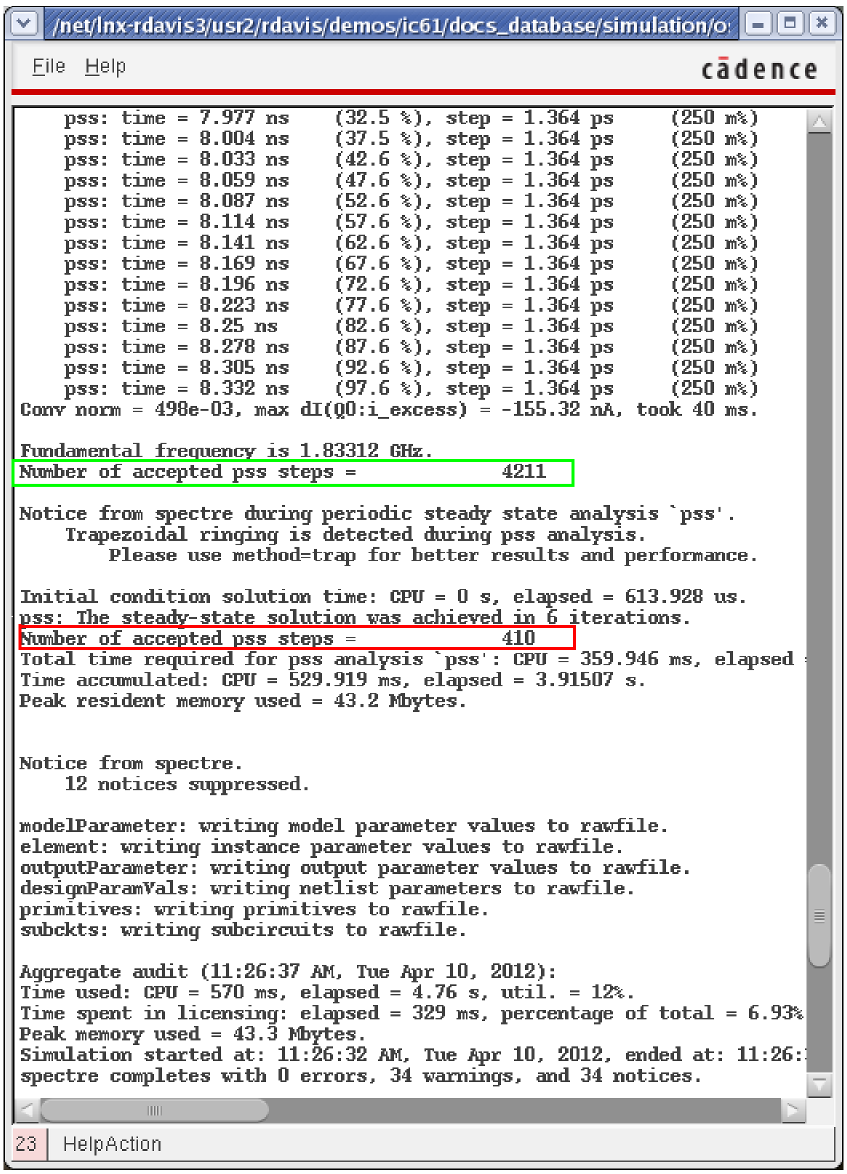
This provides a relative measurement of the noise at the individual points of the time-domain waveform. This is supported in both harmonic balance and in shooting. For a quantitative measurement, use the PM Jitter measurement.

1. First, set up the pss analysis as appropriate for your circuit. For more information, see the periodic steady-state analysis at the beginning of this chapter.



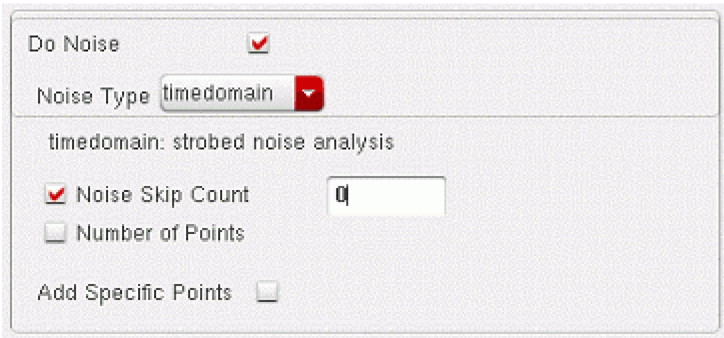
2. Run the pss analysis, and note the number of timepoints in the shooting window. Note that the first timepoint report in the spectre output window is for the tstab interval, and the

second is for the shooting window. The default behavior of timedomain noise is to run a pnoise analysis at each pss timepoint, which is usually prohibitive in terms of runtime. Determine a number of desired points where you want a pnoise analysis to be run. The larger the noiseskipcount, the larger the number of pss timepoints are skipped by the pnoise analysis. The number of timepoints in the tstab interval is shown in the green box below. The number of timepoints in the shooting window is shown in the red box below.

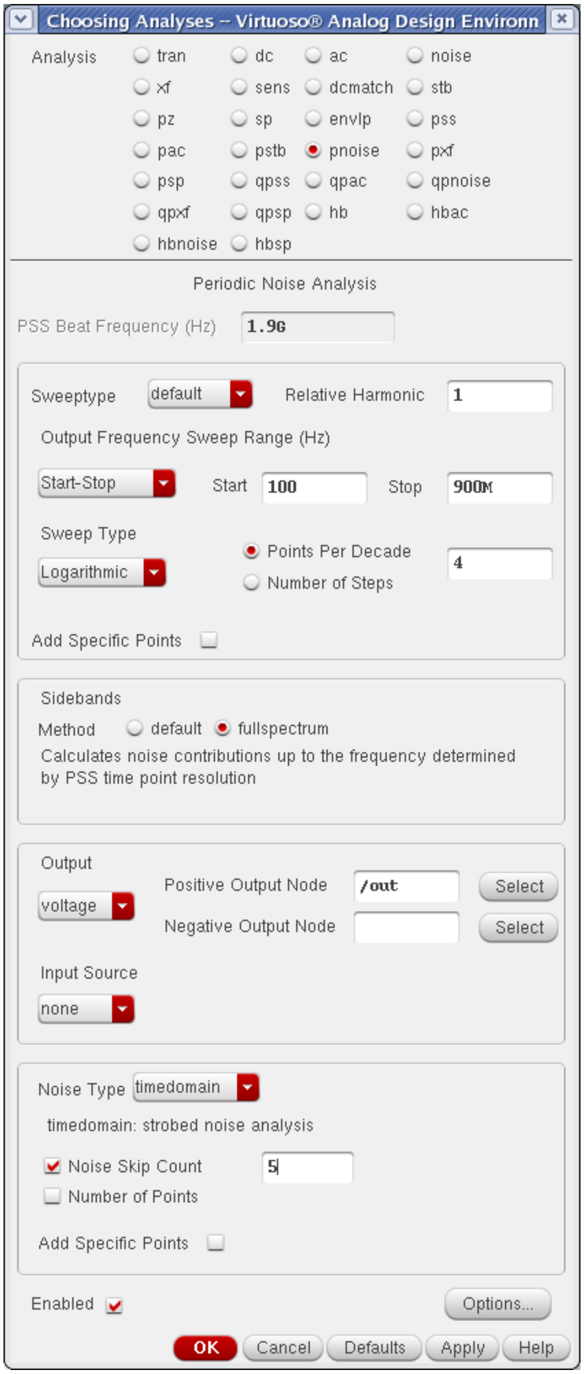



```
    pss: time = 7.977 ns    (32.5 %), step = 1.364 ps     (250 m%)
    pss: time = 8.004 ns    (37.5 %), step = 1.364 ps     (250 m%)
    pss: time = 8.033 ns    (42.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.059 ns    (47.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.087 ns    (52.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.114 ns    (57.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.141 ns    (62.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.169 ns    (67.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.196 ns    (72.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.223 ns    (77.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.25 ns     (82.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.278 ns    (87.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.305 ns    (92.6 %), step = 1.364 ps     (250 m%)
    pss: time = 8.332 ns    (97.6 %), step = 1.364 ps     (250 m%)
Conv norm = 498e-03, max dI(Q0:i_excess) = -155.32 nA, took 40 ms.

Fundamental frequency is 1.83312 GHz.
Number of accepted pss steps =             4211

Notice from spectre during periodic steady state analysis `pss'.
    Trapezoidal ringing is detected during pss analysis.
        Please use method=trap for better results and performance.

Initial condition solution time: CPU = 0 s, elapsed = 613.928 us.
pss: The steady-state solution was achieved in 6 iterations.
Number of accepted pss steps =             410
Total time required for pss analysis `pss': CPU = 359.946 ms, elapsed
Time accumulated: CPU = 529.919 ms, elapsed = 3.91507 s.
Peak resident memory used = 43.2 Mbytes.


Notice from spectre.
    12 notices suppressed.

modelParameter: writing model parameter values to rawfile.
element: writing instance parameter values to rawfile.
outputParameter: writing output parameter values to rawfile.
designParamVals: writing netlist parameters to rawfile.
primitives: writing primitives to rawfile.
subckts: writing subcircuits to rawfile.

Aggregate audit (11:26:37 AM, Tue Apr 10, 2012):
Time used: CPU = 570 ms, elapsed = 4.76 s, util. = 12%.
Time spent in licensing: elapsed = 329 ms, percentage of total = 6.93%
Peak memory used = 43.3 Mbytes.
Simulation started at: 11:26:32 AM, Tue Apr 10, 2012, ended at: 11:26:
spectre completes with 0 errors, 34 warnings, and 34 notices.
```

3. On the pnoise *Choosing Analyses* form, Select *timedomain* from the *Noise Type* drop-down list.



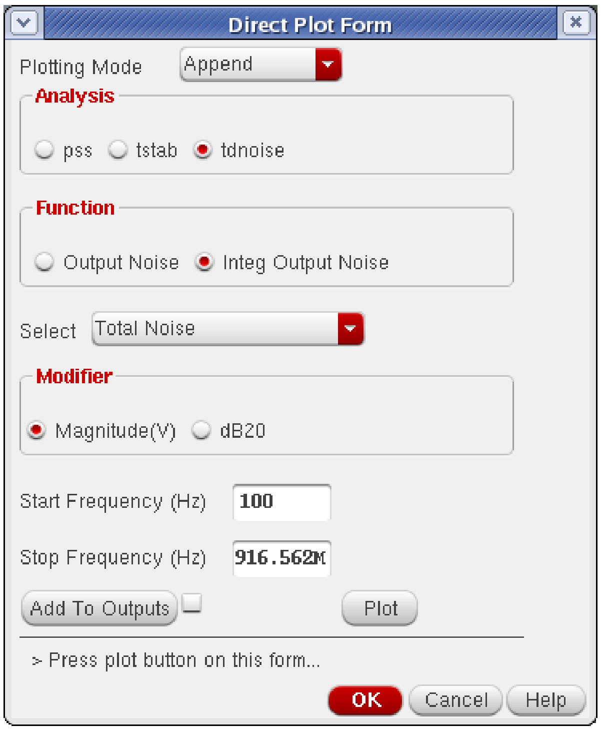
4. Select the *Noise Skip Count* check box and type the number determined above in the text box. This example shows 5, which means run pnoise at the first timepoint, then skip 6, and run again.



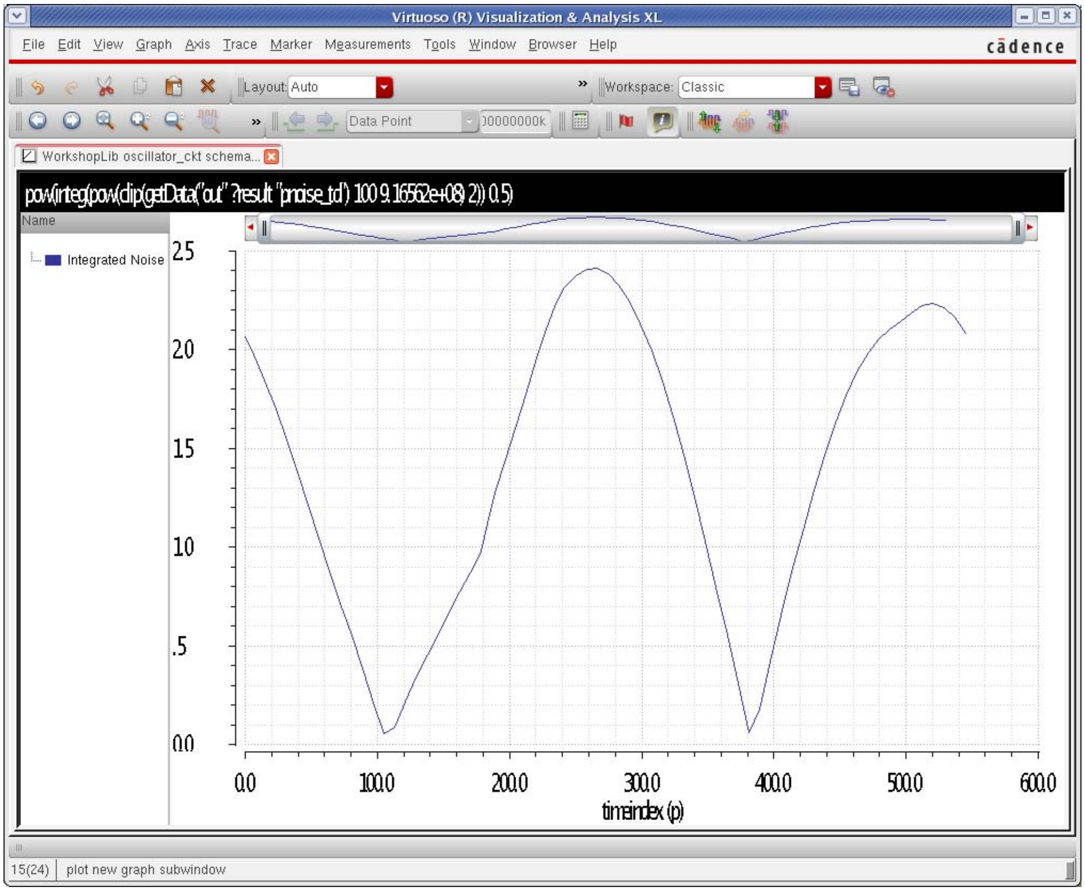
5. Run the analysis.

6. In ADE, select *Results - Direct Plot - Main Form*.
7. Select *tdnoise* results.
8. Select *Integrated Output Noise*.
9. Specify the start frequency from the pnoise *Choosing Analyses* form.
10. Click *Plot.*

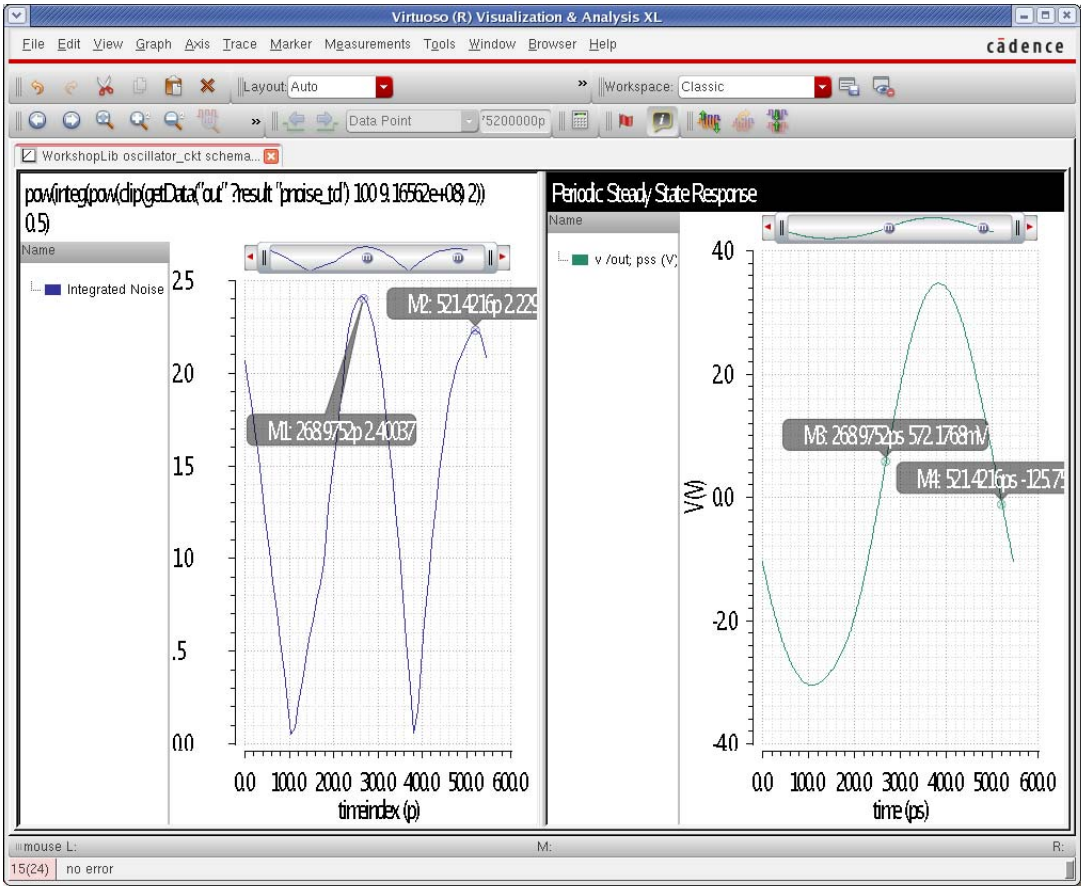
.



11. The integrated noise curve is plotted.



12. To determine which points on the oscillator output waveform have the most noise, plot the pss output waveform and place markers at the appropriate points.

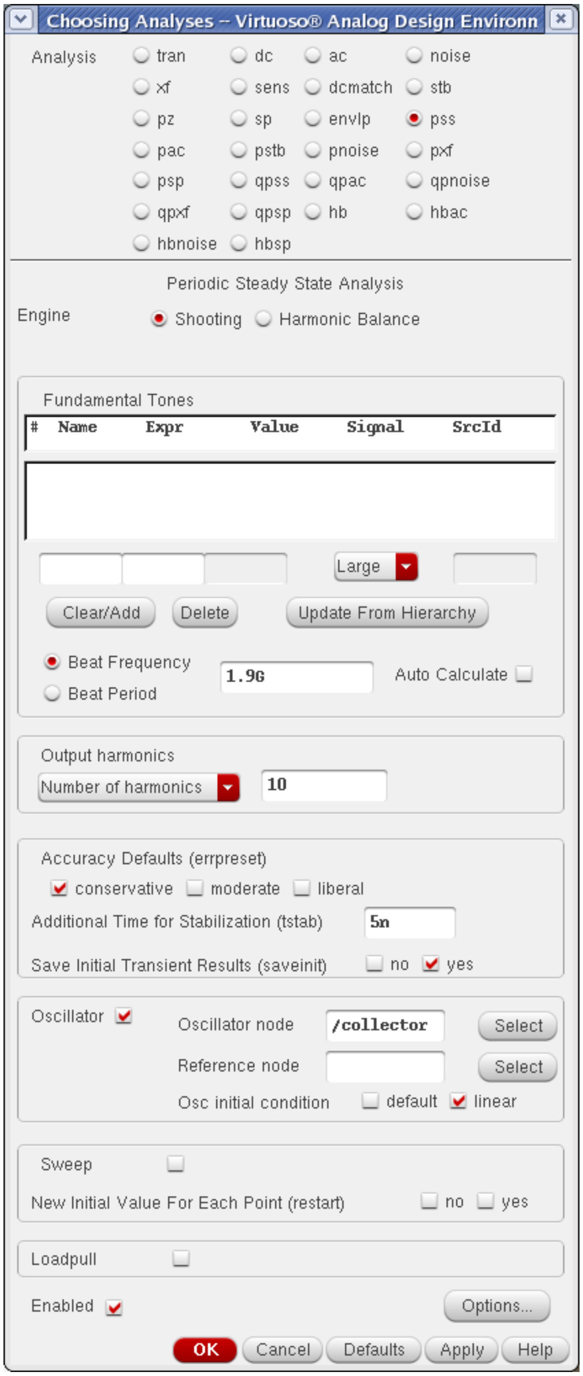


Noise Type = Jitter

FM Jitter

This selection allows for measurement of the cycle jitter, the cycle-to-cycle jitter, and the AM and PM components of phase noise. This is an averaged noise measurement over one cycle of the oscillator.

1. First, set up the pss form as appropriate for your circuit. For more information on the pss analysis, please see the periodic steady-state section at the beginning of this chapter.

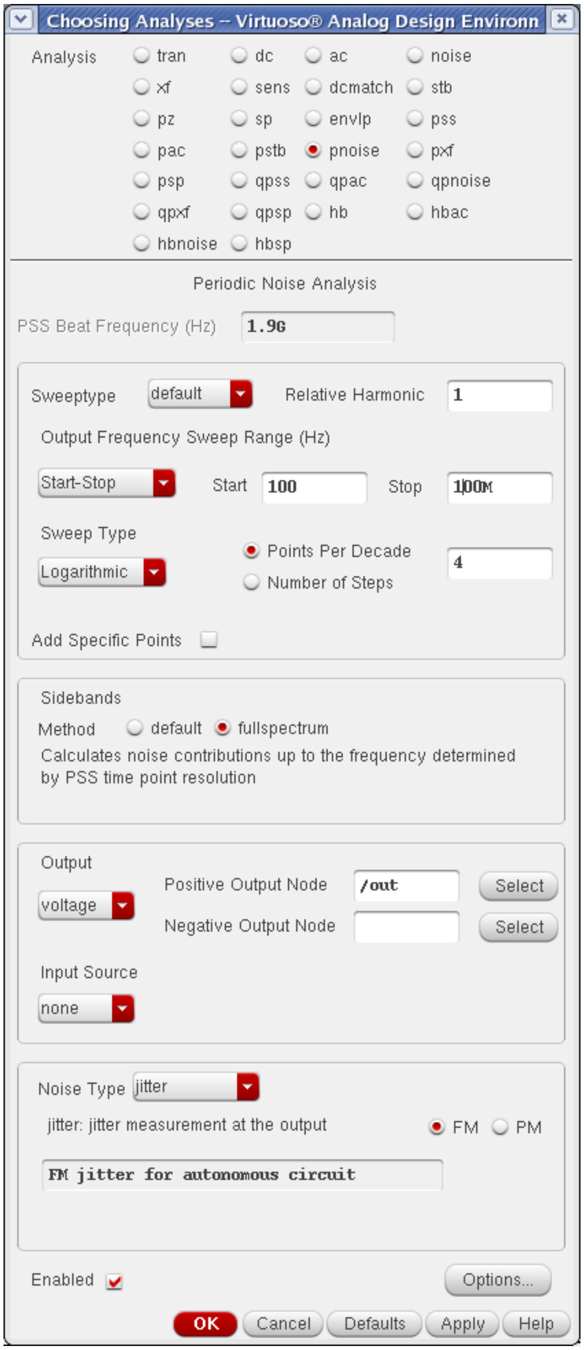


2. On the pnoise *Choosing Analyses* form, select *Jitter* from the *Noise Type* drop-down list.

3. Select the *FM* radio button.

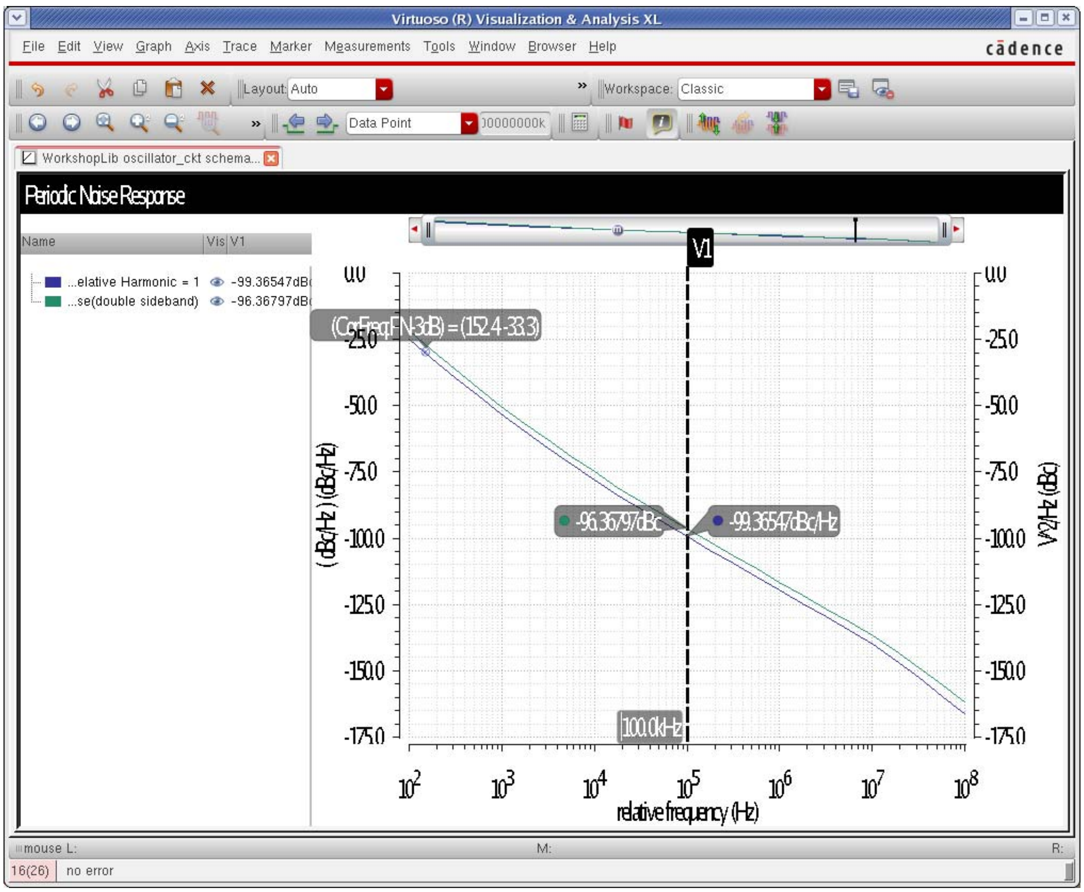
4. Fill out the form as it would be when noisetype is set to sources. When Jitter is selected, the noise analysis will be run above and below the harmonic number selected in the

pnoise form. This is done so that the noise from above and below the oscillator output frequency can be included in the noise calculation.

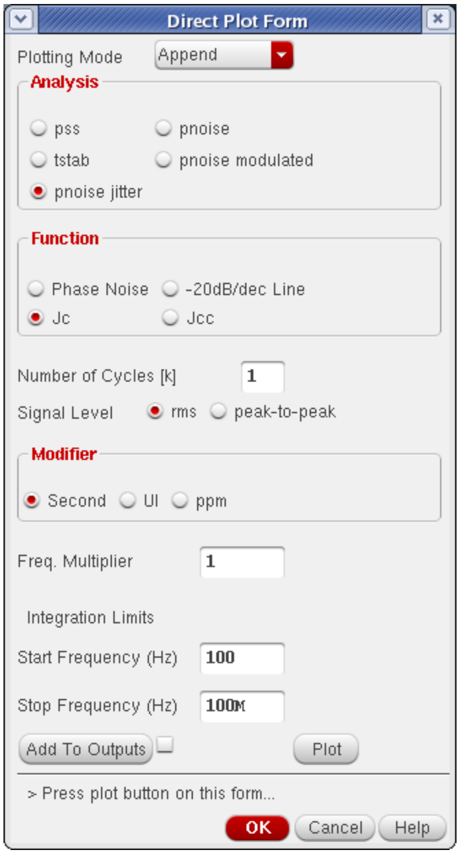


5. In ADE, select *Results - Direct Plot - Main Form*.
6. Select *pnoise* in the *Analysis* section.

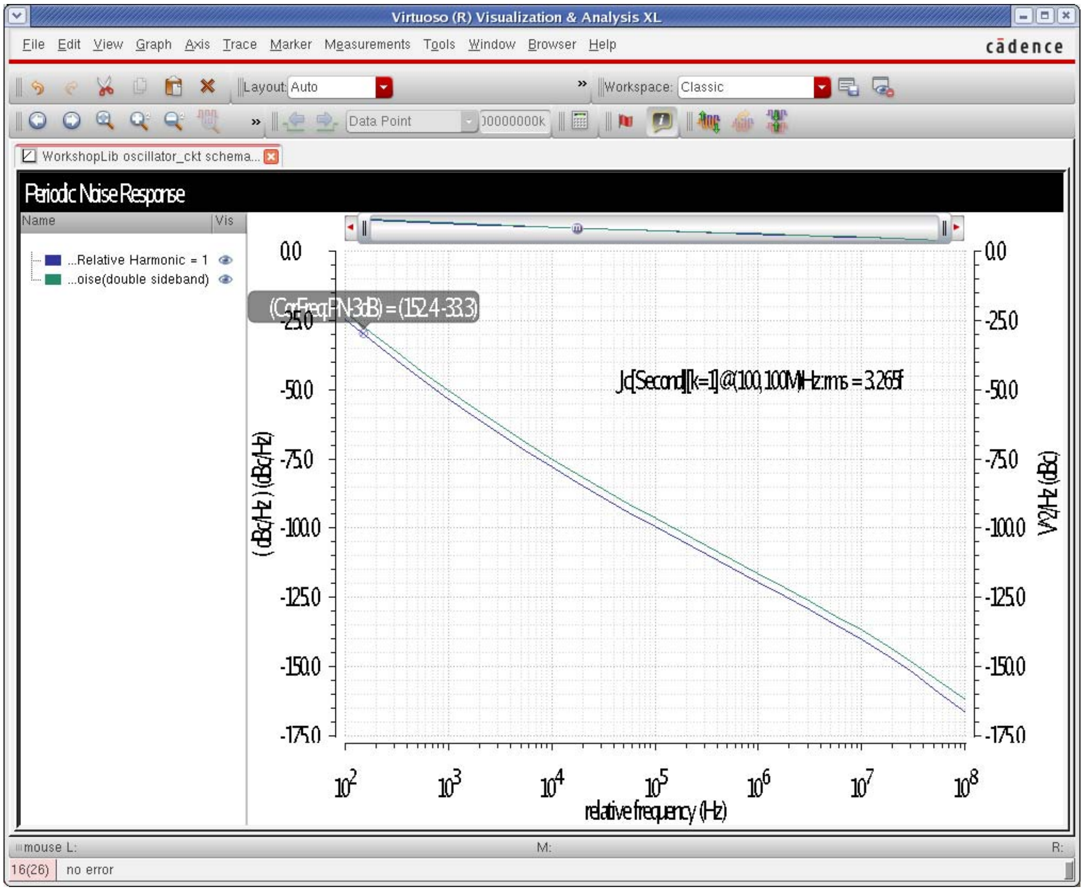
7. Select *Phase Noise*.
8. Select *Plot*.
9. Select *Pnoise Jitter* in the *Analysis* section.
10. Click *Plot*. Note that Phase Noise from the pnoise result set is a single sideband measurement. Phase Noise from the pnoise jitter result set is a double-sideband measurement. As such, it is usually 3dB larger than Phase Noise from the pnoise result set.



11. Select Jc or Jcc. In the modifier section, UI is Unit interval, and ppm is parts per mission.



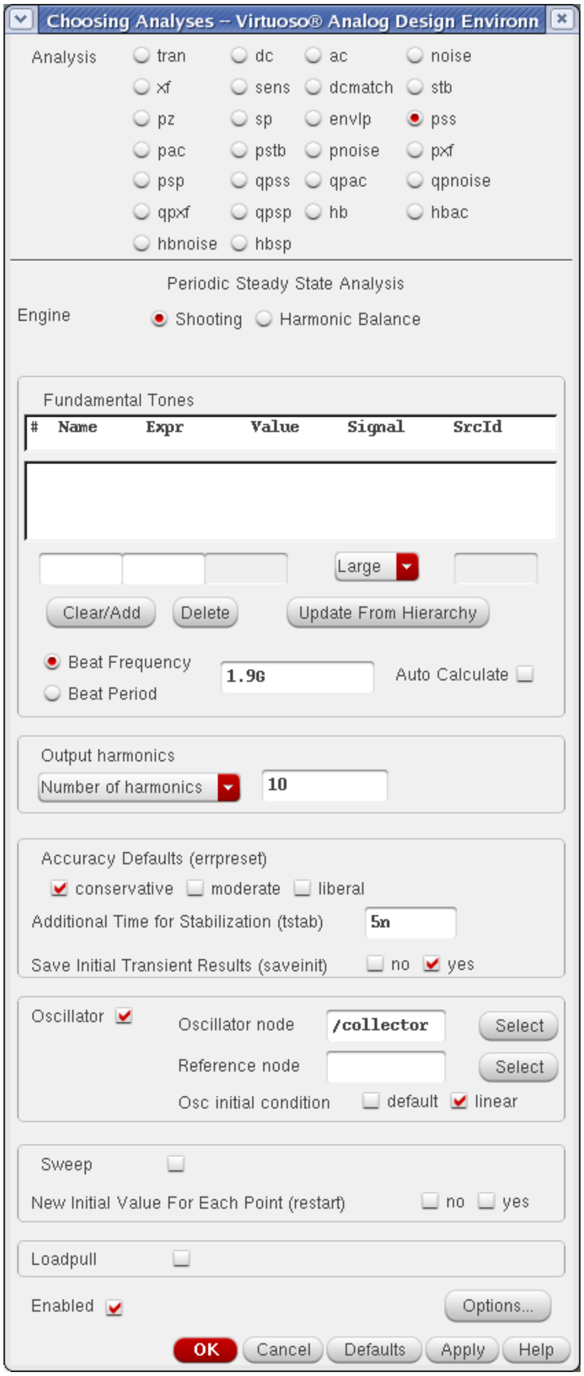
12. Click *Plot*. The Jc or Jcc calculation is plotted as text in the waveform window.



PM Jitter

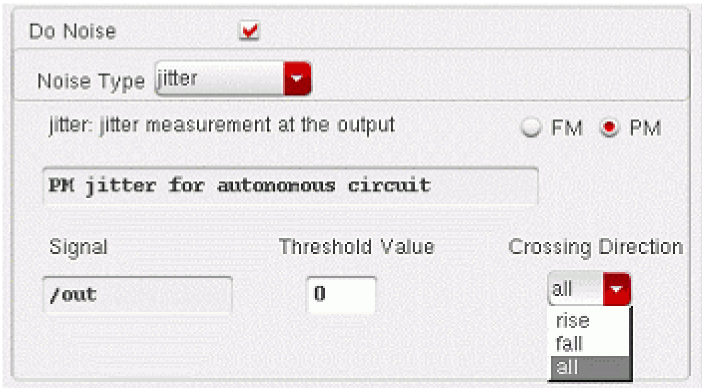
This allows for instantaneous measurement of jitter at individual voltages in the pss waveform. This is supported in both shooting and harmonic balance.

1. Set up the pss analysis as appropriate for your circuit. For more information, please see the periodic steady-state analysis at the beginning of this chapter.



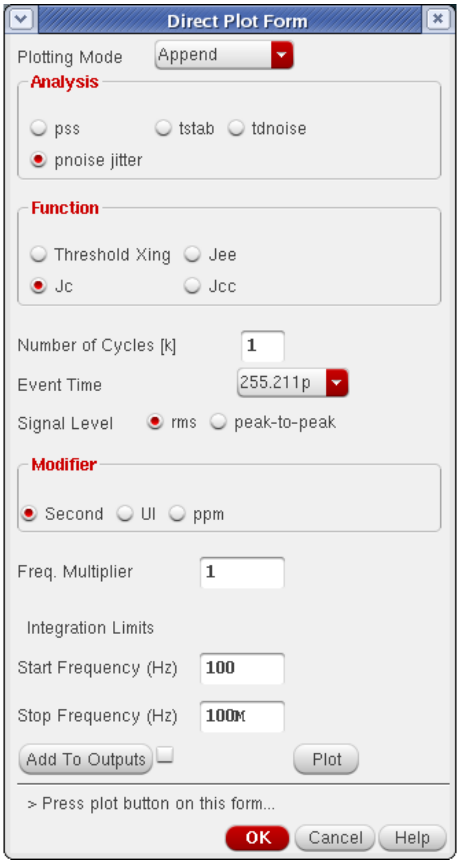
2. On the pnoise *Choosing Analyses* form, select *Jitter* from the *Noise Type* drop-down list.

3. Select *PM* as the selection for *jitter: jitter measurement at the output*.

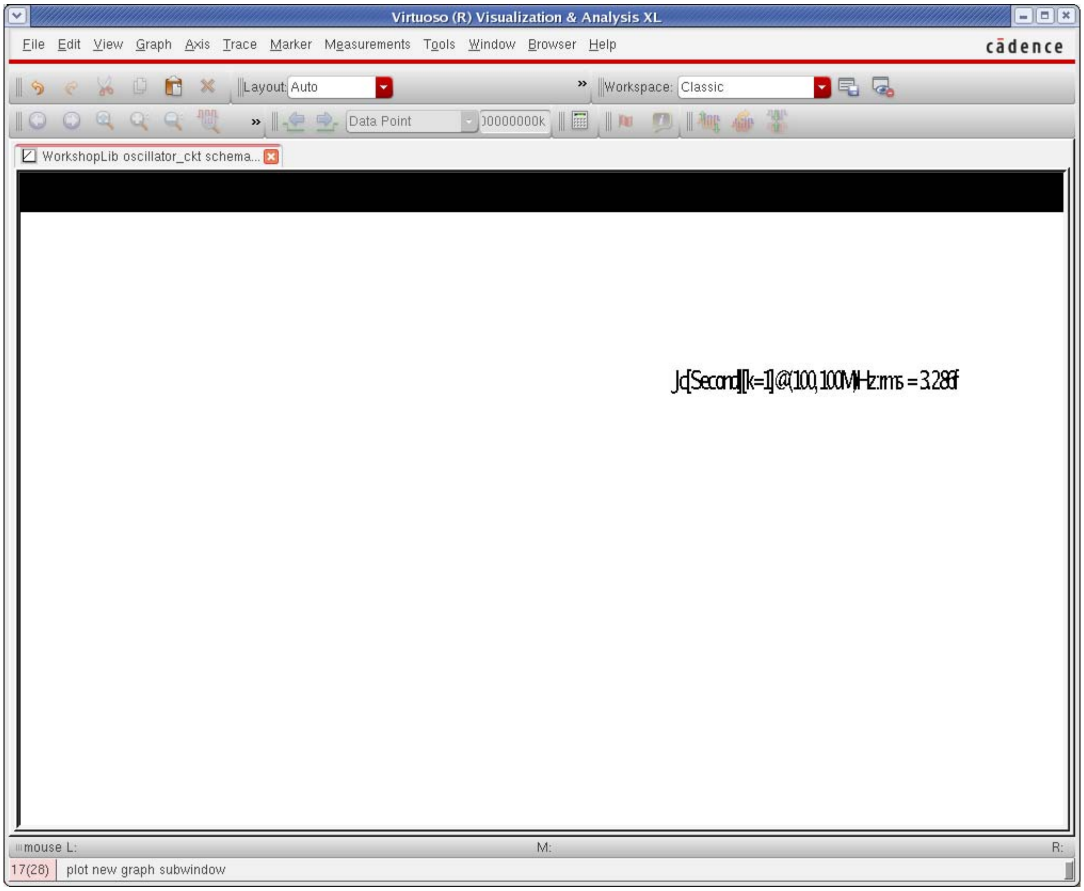


4. Specify the voltage for instantaneous noise measurement in the *Threshold Value* field.
5. Select a value from the *Crossing Direction* drop-down list.
6. Run the simulation.
7. In ADE, select *Results - Direct Plot - Main Form*.
8. Select *pnoise jitter*.

9. Select Jc or Jcc.



10. Click *Plot*. The jitter calculation appears as a label in the waveform tool.

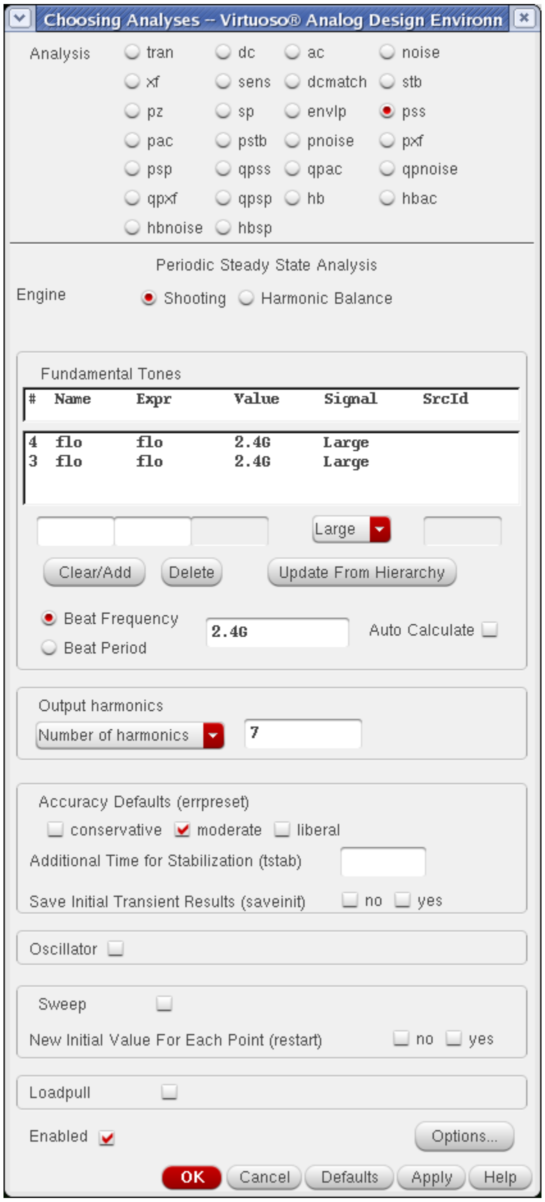


Examples

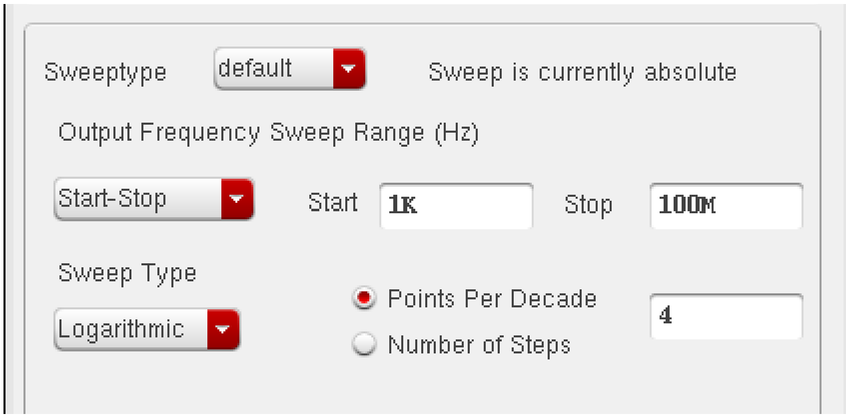
Driven Circuit Noise Setup

On the *Choosing Analyses* form:

1. First, set up the pss analysis for your circuit. For more information see the pss section at the beginning of this chapter. Shooting is shown in the example below.

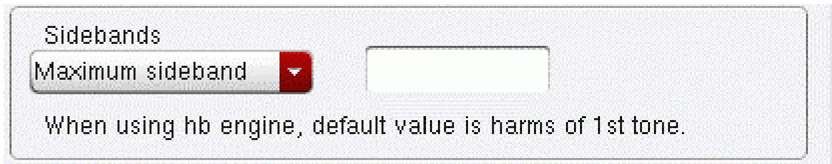


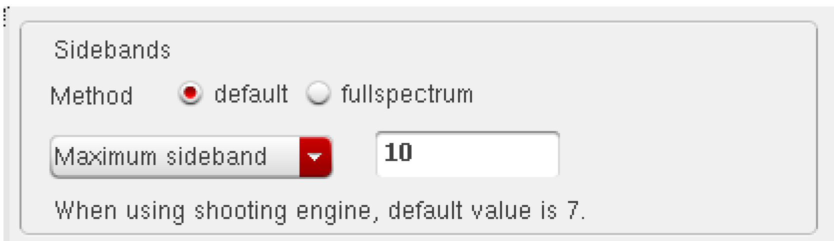
2. In the pnoise *Choosing Analyses* form, specify the output frequency range of your circuit.

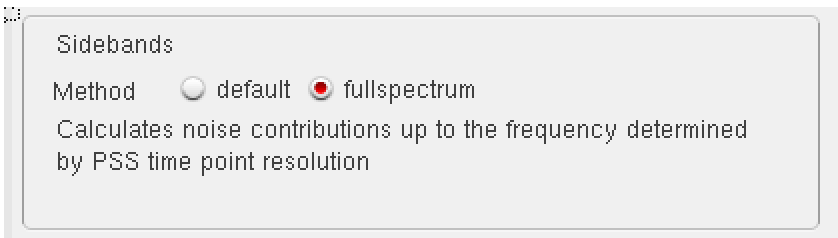


3. If you use log spacing, set about 3 to 5 points per decade in the *Points Per Decade* field.

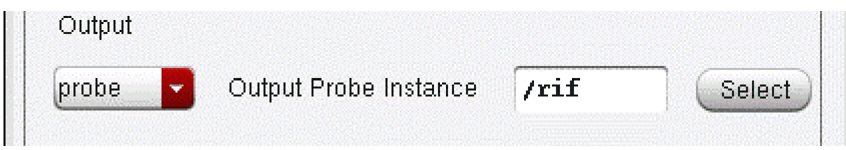
4. When Harmonic Balance is set for the pss engine, you can leave the *Maximum sideband* field blank. All the mixing products from the large-signal analysis will be considered. If the pss engine is shooting, select whether the default method or the full-spectrum method should be used. Full-spectrum is faster than the default method when many sidebands are required for your circuit and it does not require the sidebands to be set. When the default method is selected, specify the number of the desired frequency translations in the *Maximum sideband* field. Up to 40 sidebands can be specified by default.





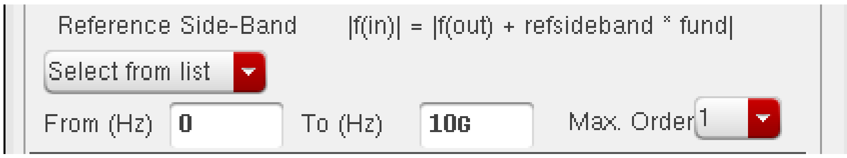


5. Select the resistor or port component used for the load resistance. This excludes the noise from this component for the noise figure calculation, but keeps it for the total noise output calculation.

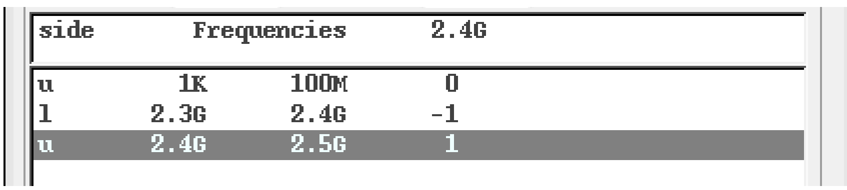


6. If you want a noise figure calculation, you must use a port as the input for the circuit. Set the instance name.

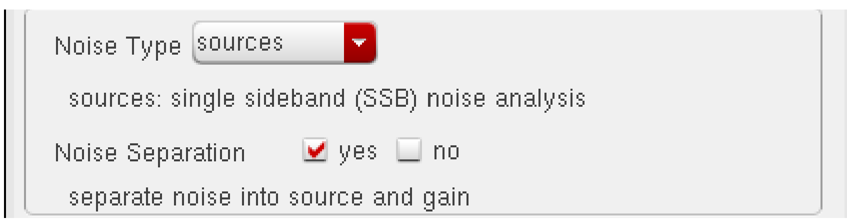
7. The easiest way to select the reference sideband is to set it to *Select from list*.



8. Select the passband frequency range you want for the noise figure and input-referred noise calculations.



9. If you want to see the noise contributions from the individual frequencies and the different device noise calculations individually on the output of the noise analysis, enable noise separation. This is not supported for the full-spectrum method.



10. When you are done with the setup, run the simulation.

Noise Summary

The noise summary provides information about the devices that contribute noise to the output. This information is available any time a pnoise simulation completes.

In the ADE, select *Results - Print - Noise Summary.* If noise separation is enabled, there will be two choices at the very top. `Pnoise_src` has information about the noise currents at the individual noise sources in the circuit. Pnoise has information about noise at the output of the circuit.

If you want the noise contributors at a single frequency, select *spot noise*. If you want to have the noise integrated over a frequency range, select *integrated noise*.

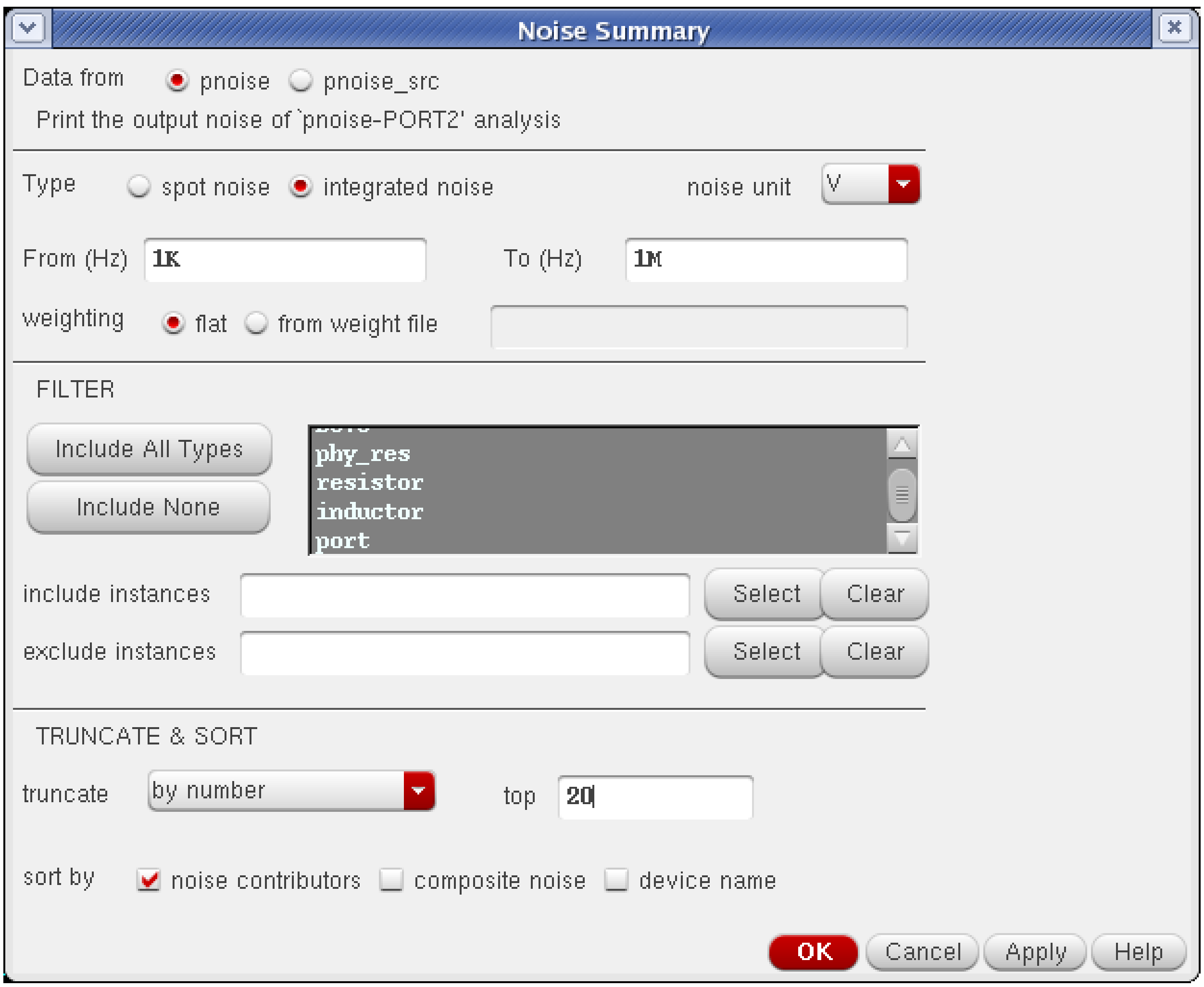
If you want noise in volts, set the *noise unit* to *V.*

Generally, *Include All Types* should be selected. If you just want the noise from specific types of noise generators, you can select them from the list.

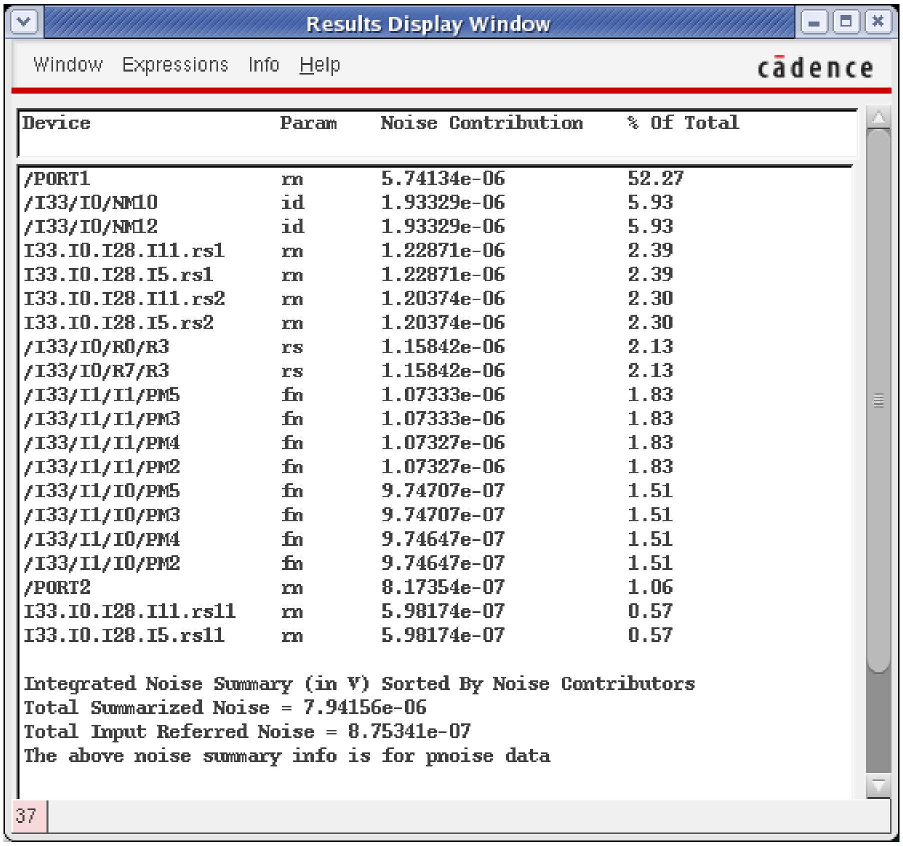
If you want to specifically include or exclude instances in the list, use the select button to the right side of the include instances and exclude instances fields and then select the components in the schematic.

Truncate just applies to the list. The total input and output-referred noise at the bottom of the noise summary output always includes all the noise from everything.

The figure below is the noise summary window with the sort function being set to noise contributors. Then there are three windows that show the three different sorted outputs.



An example of the noise summary output sorted by noise contributors.

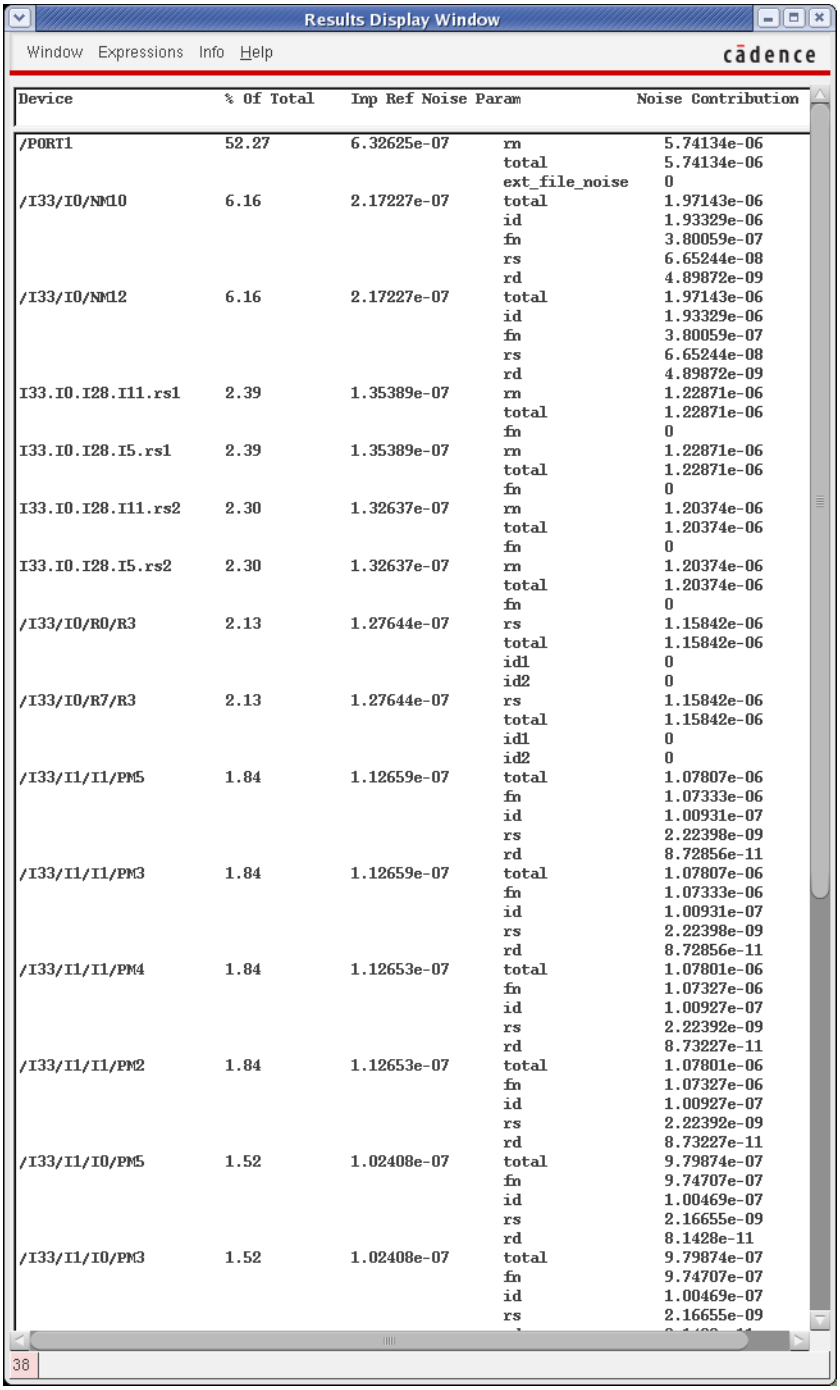


Device	Param	Noise Contribution	% Of Total
/PORT1	rn	5.74134e-06	52.27
/I33/I0/NM10	id	1.93329e-06	5.93
/I33/I0/NM12	id	1.93329e-06	5.93
I33.I0.I28.I11.rs1	rn	1.22871e-06	2.39
I33.I0.I28.I5.rs1	rn	1.22871e-06	2.39
I33.I0.I28.I11.rs2	rn	1.20374e-06	2.30
I33.I0.I28.I5.rs2	rn	1.20374e-06	2.30
/I33/I0/R0/R3	rs	1.15842e-06	2.13
/I33/I0/R7/R3	rs	1.15842e-06	2.13
/I33/I1/I1/PM5	fn	1.07333e-06	1.83
/I33/I1/I1/PM3	fn	1.07333e-06	1.83
/I33/I1/I1/PM4	fn	1.07327e-06	1.83
/I33/I1/I1/PM2	fn	1.07327e-06	1.83
/I33/I1/I0/PM5	fn	9.74707e-07	1.51
/I33/I1/I0/PM3	fn	9.74707e-07	1.51
/I33/I1/I0/PM4	fn	9.74647e-07	1.51
/I33/I1/I0/PM2	fn	9.74647e-07	1.51
/PORT2	rn	8.17354e-07	1.06
I33.I0.I28.I11.rs11	rn	5.98174e-07	0.57
I33.I0.I28.I5.rs11	rn	5.98174e-07	0.57

Integrated Noise Summary (in V) Sorted By Noise Contributors
Total Summarized Noise = 7.94156e-06
Total Input Referred Noise = 8.75341e-07
The above noise summary info is for pnoise data

An example of the noise summary output sorted by composite noise.

The below output is sorted by device name.



Device	% Of Total	Inp Ref Noise	Param	Noise Contribution
/PORT1	52.27	6.32625e-07	rn	5.74134e-06
			total	5.74134e-06
			ext_file_noise	0
/I33/I0/NM10	6.16	2.17227e-07	total	1.97143e-06
			id	1.93329e-06
			fn	3.80059e-07
			rs	6.65244e-08
			rd	4.89872e-09
/I33/I0/NM12	6.16	2.17227e-07	total	1.97143e-06
			id	1.93329e-06
			fn	3.80059e-07
			rs	6.65244e-08
			rd	4.89872e-09
I33.I0.I28.I11.rs1	2.39	1.35389e-07	rn	1.22871e-06
			total	1.22871e-06
			fn	0
I33.I0.I28.I5.rs1	2.39	1.35389e-07	rn	1.22871e-06
			total	1.22871e-06
			fn	0
I33.I0.I28.I11.rs2	2.30	1.32637e-07	rn	1.20374e-06
			total	1.20374e-06
			fn	0
I33.I0.I28.I5.rs2	2.30	1.32637e-07	rn	1.20374e-06
			total	1.20374e-06
			fn	0
/I33/I0/R0/R3	2.13	1.27644e-07	rs	1.15842e-06
			total	1.15842e-06
			id1	0
			id2	0
/I33/I0/R7/R3	2.13	1.27644e-07	rs	1.15842e-06
			total	1.15842e-06
			id1	0
			id2	0
/I33/I1/I1/PM5	1.84	1.12659e-07	total	1.07807e-06
			fn	1.07333e-06
			id	1.00931e-07
			rs	2.22398e-09
			rd	8.72856e-11
/I33/I1/I1/PM3	1.84	1.12659e-07	total	1.07807e-06
			fn	1.07333e-06
			id	1.00931e-07
			rs	2.22398e-09
			rd	8.72856e-11
/I33/I1/I1/PM4	1.84	1.12653e-07	total	1.07801e-06
			fn	1.07327e-06
			id	1.00927e-07
			rs	2.22392e-09
			rd	8.73227e-11
/I33/I1/I1/PM2	1.84	1.12653e-07	total	1.07801e-06
			fn	1.07327e-06
			id	1.00927e-07
			rs	2.22392e-09
			rd	8.73227e-11
/I33/I1/I0/PM5	1.52	1.02408e-07	total	9.79874e-07
			fn	9.74707e-07
			id	1.00469e-07
			rs	2.16655e-09
			rd	8.1428e-11
/I33/I1/I0/PM3	1.52	1.02408e-07	total	9.79874e-07
			fn	9.74707e-07
			id	1.00469e-07
			rs	2.16655e-09

For all of the noise summaries, here is a table of the abbreviations used in the Param column:

Resistor and Inductor:

Parameter	Meaning
`rn`	resistor thermal noise
`fn`	Flicker noise

Diode:

Parameter	Meaning
`id`	Diode current shot noise
`rs`	Series parasitic resistor noise

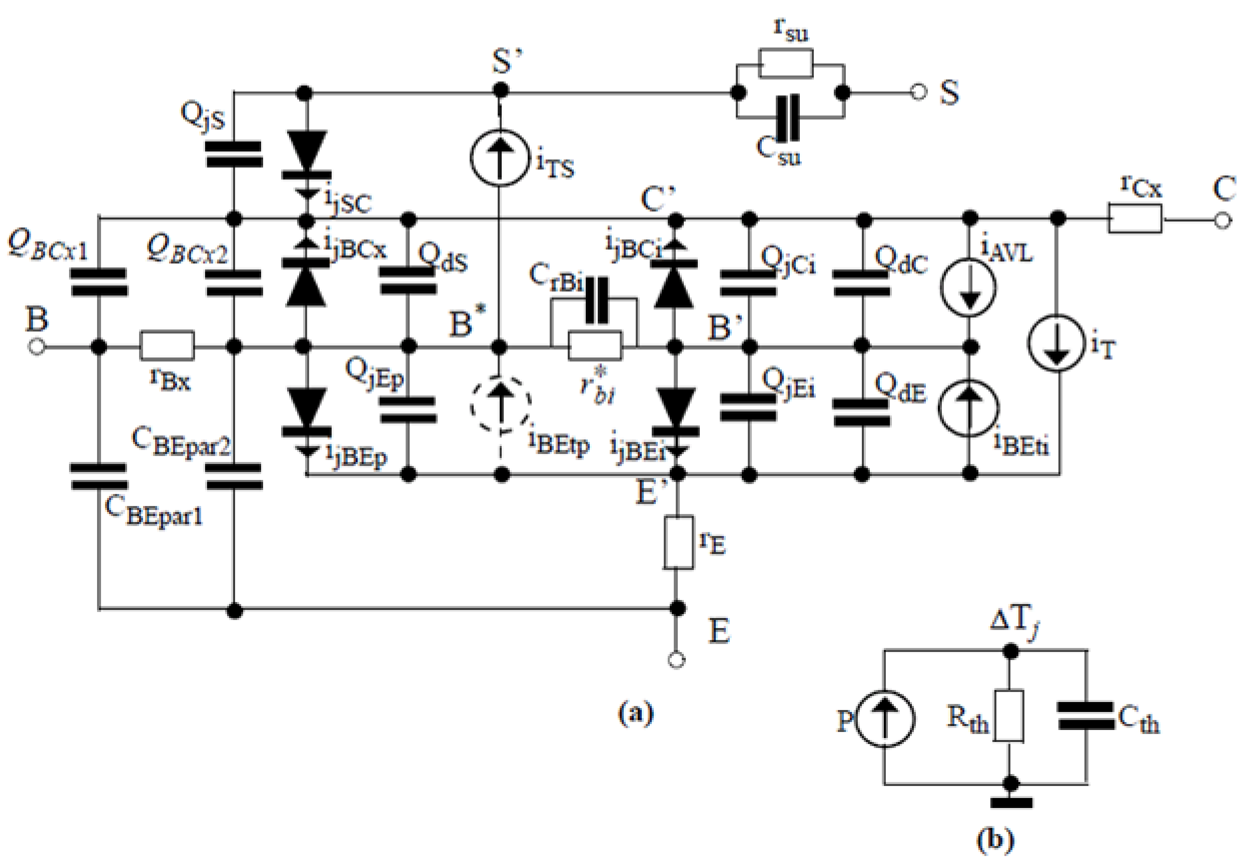
Gummel-Poon BJT:

Parameter	Meaning
`fn`	Flicker noise
`ic`	Collector current shot noise
`ib`	Base current shot noise
`rc`	Thermal noise from the collector parasitic resistor
`rb`	Thermal noise from the base parasitic resistor
`re`	Thermal noise from the emitter parasitic resistor

HiCUM:

Parameter	Meaning
`ib`	Base current shot noise
`ic`	Collector current shot noise
`rbi`	Thermal noise from the internal base parasitic resistor (See diagram below)
`rbx`	Thermal noise from the external base parasitic resistor (See diagram below)
`rcx`	Collector parasitic resistor thermal noise (See diagram below)
`re`	Emitter parasitic resistor thermal noise (See diagram below)
`fn`	Flicker noise

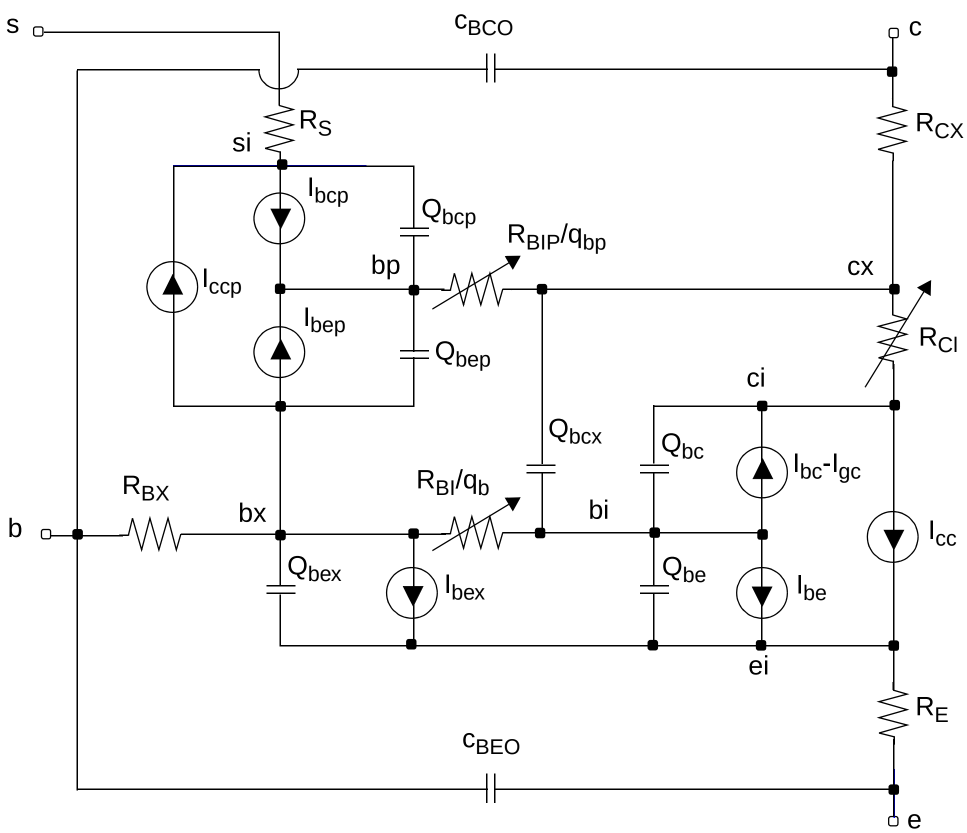
Figure 4-1 HiCUM Schematic Diagram

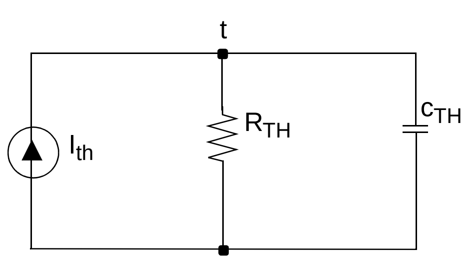


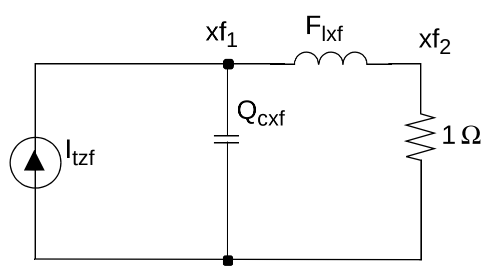
VBIC:

Parameter	Meaning
`rbi`	Thermal noise in resistor Rbi (See diagram below)
`rbx`	Thermal noise in resistor Rbx (See diagram below)
`rci`	Thermal noise in resistor Rci (See diagram below)
`re`	Thermal noise in resistor Re (See diagram below)
`rs`	Thermal noise in resistor Rs (See diagram below)
`itzf`	Collector current shot noise
`ibe`	Total base current shot noise
`ibex`	Node Bx to E base current shot noise (See diagram below)
`fn`	Ibe flicker noise (See diagram below)
`fnx`	Ibex flicker noise (See diagram below)
`iccp`	Parasitic transport collector to emitter current noise (See diagram below)
`ibep`	Parasitic transport base to emitter current noise (See diagram below)
`fnp`	Ibep flicker Noise (See diagram below)

Figure 4-2 VBIC Schematic Diagram







BSIM3:

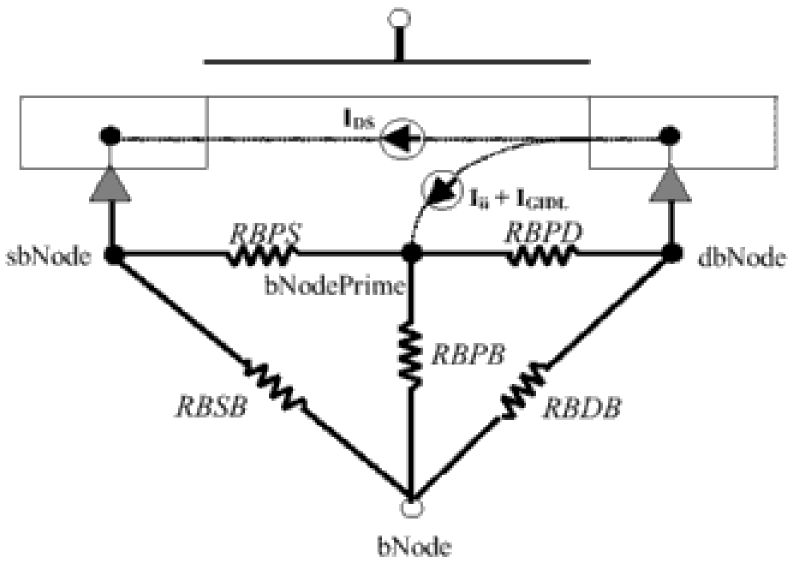
Parameter	Meaning
`fn`	Flicker Noise

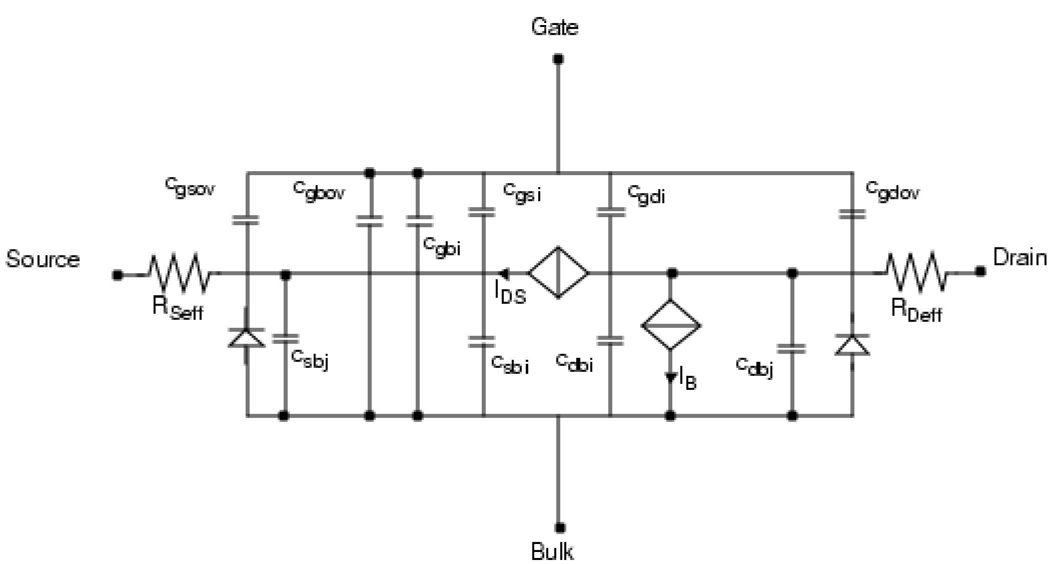
Parameter	Meaning
`id`	Thermal noise from the resistive drain current
`rs`	Thermal noise from the source parasitic resistor
`rd`	Thermal noise from the drain parasitic resistor

BSIM4:

Parameter	Meaning
`fn`	Flicker noise
`id`	Thermal noise from the resistive drain current noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`igd`	Gate-to-drain tunneling current noise
`igs`	Gate-to-source tunneling current noise
`igb`	Gate-to-bulk tunneling current noise
`rbdb`	Resistor thermal noise between dbNode and bNode (See diagram below)
`rbpb`	Resistor thermal noise between bNode and bNodePrime (See diagram below)
`rbpd`	Resistor thermal noise between Bulk and dbNode (See diagram below)
`rbps`	Resistor thermal noise between bulk and bNode prime
`rbsb`	Resistor thermal noise between sbNode and bNode (See diagram below)
`rgbi`	Gate Bias-Independent Resistor thermal noise

Figure 4-3 BSIM4 Schematic Diagrams

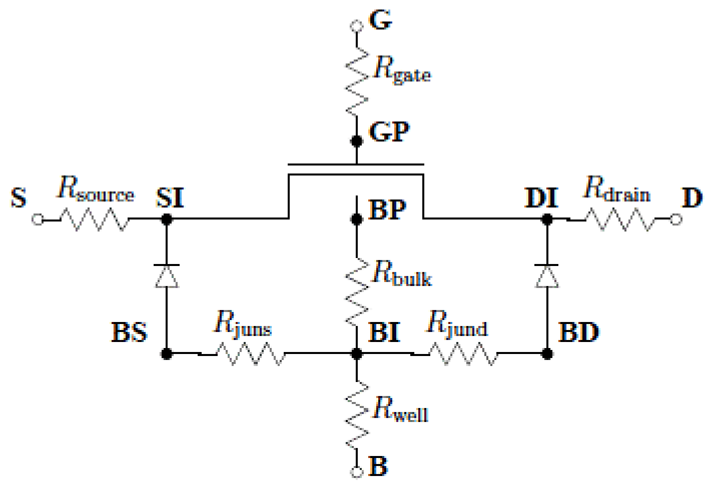




PSP-103

Parameter	Meaning
sfl	Channel current flicker noise
sth	Channel current thermal noise
sig	induced gate noise at 1 Hz
sigth	Induced gate noise
shotgd	Gate current shot noise from gate to drain
shotgs	Gate current shot noise from gate to source
sjnoise	Source to Bulk leakage current noise
djnoise	Drain to Bulk leakage and avalanche current noise
rgatenoise	Gate resistance thermal noise
rbulknoise	Bulk resistor thermal noise between node BP and BI (See diagram below)
rjnoise	Source side bulk resistor thermal noise between node BI and BS (See diagram below)
rjundnoise	Drain side bulk resistor thermal noise between node BI and BD (See diagram below)
rwellnoise	Well resistor thermal noise between node BI and B (See diagram below)
rsourcenoise	Source parasitic resistor thermal noise
rdrainnoise	Drain parasitic resistor thermal noise

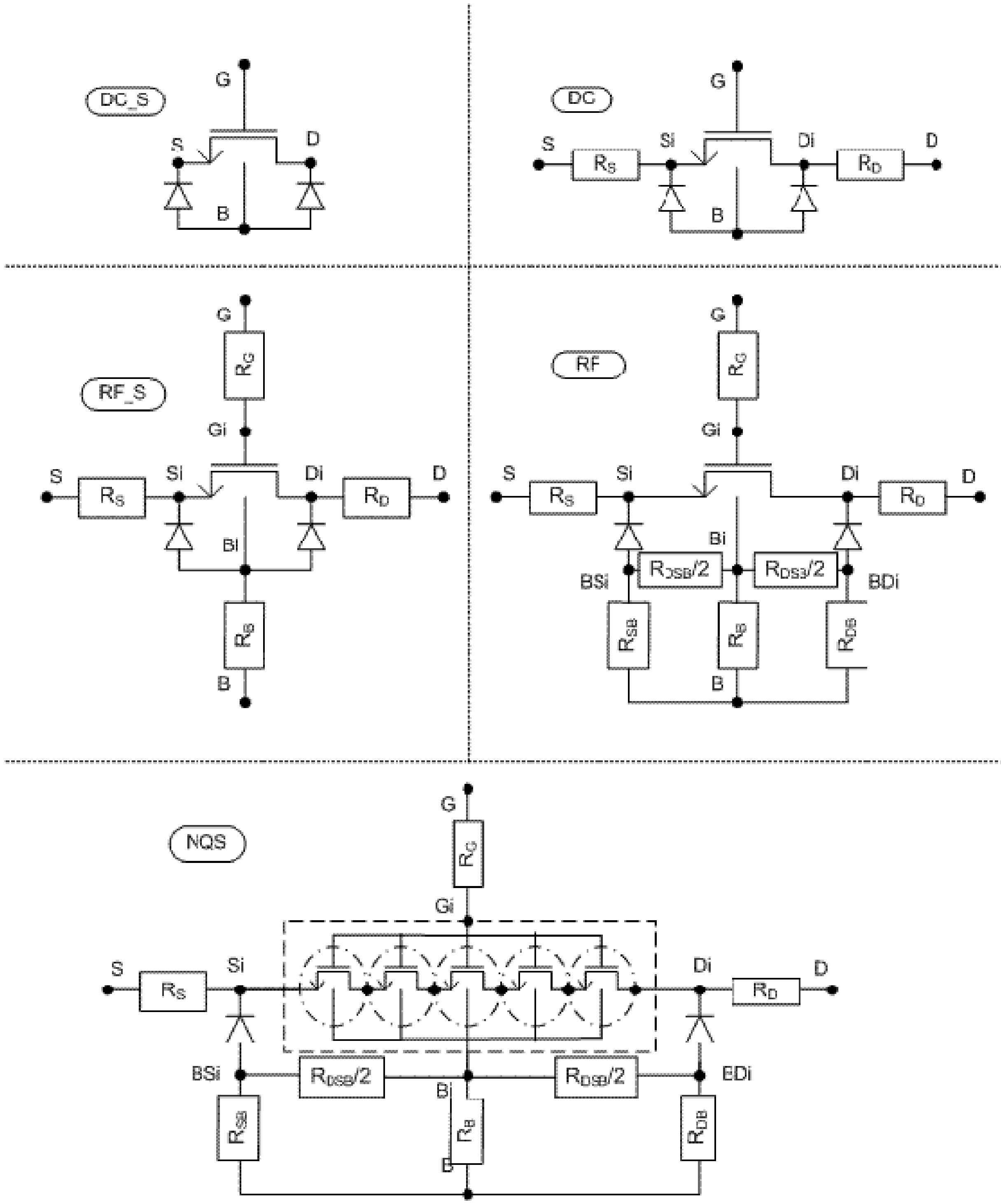
Figure 4-4 PSP-103 Schematic Diagram



EKV 3.0

Parameter	Meaning
`fn_id`	Channel current flicker noise
`fn_ig`	Gate current flicker noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`rg`	Gate parasitic resistor thermal noise
`rsb`	Thermal noise for the substrate resistor connected between BSi and B. (See diagram below)
`rdb`	Thermal noise for the substrate resistor connected between BDi and B. (See diagram below)
`rbbs`	Thermal noise for the substrate resistor connected between BSi and Bi. (See diagram below where it is labeled $R_{DSB}/2$)
`rbbd`	Thermal noise for the substrate resistor connected between BDi and Bi. (See diagram below where it is labeled $R_{DSB}/2$)
`id`	Thermal noise of the channel resistor
`cor`	Induced gate noise
`shot_ig`	Gate current shot noise

Figure 4-5 EKV3.0 Schematic Diagrams



For other model types:

1. In a unix/linux shell window type `which spectre`.

   This should return a path that ends in `tools/bin/spectre`.

2. Triple-click the line with the path, and then click the center mouse button to enter it as a command.

3. Use the backspace key, and delete the word `spectre`.

4. Type `cdnshelp` at the end of the expression, and press `Enter`.

   A browser window will be displayed.

5. Close the *Tip of the day* window.

6. At the top, you should see the release number of the simulator you are using.

7. Click the plus sign (+) to the left of the MMSim folder.

8. Click the plus sign (+) to the left of the *Virtuoso Simulator Components and Device Models Reference* folder.

9. Select the device type you are using.

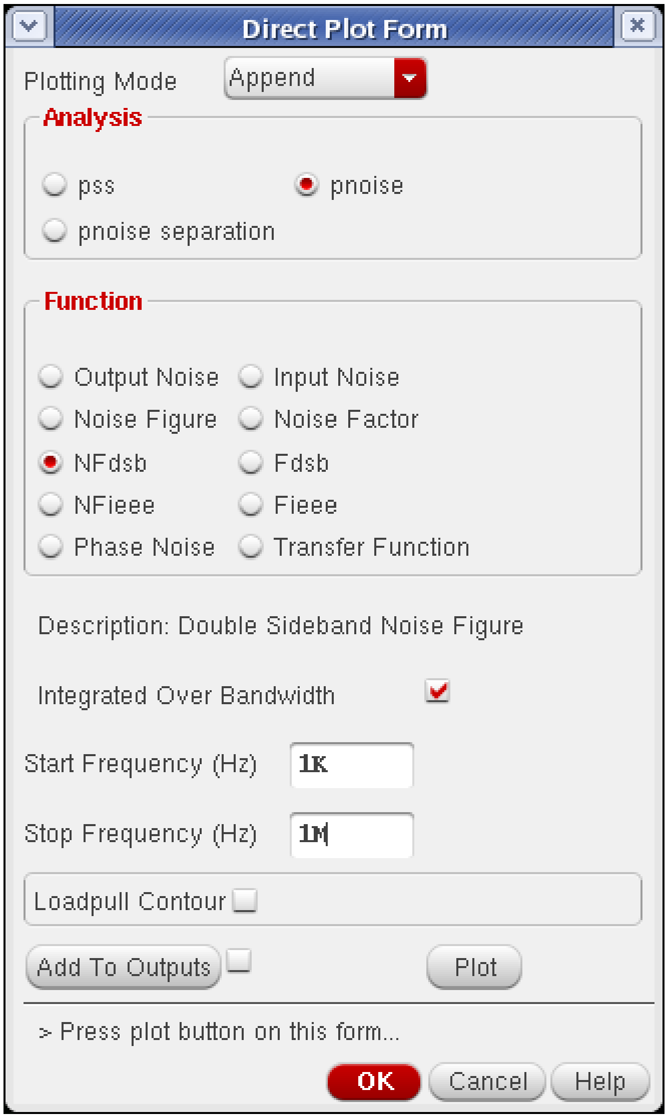
10. At the bottom of that device listing, you will see *Component Statements*. Double-click on this.

11. Enter the name of the parameter in the search box in the right pane of the cdnshelp window.

12. Click the down arrow to the right of the *Find* field until you see the parameter name in the pane on the right side, and then see the definition of that parameter.

Noise Figure

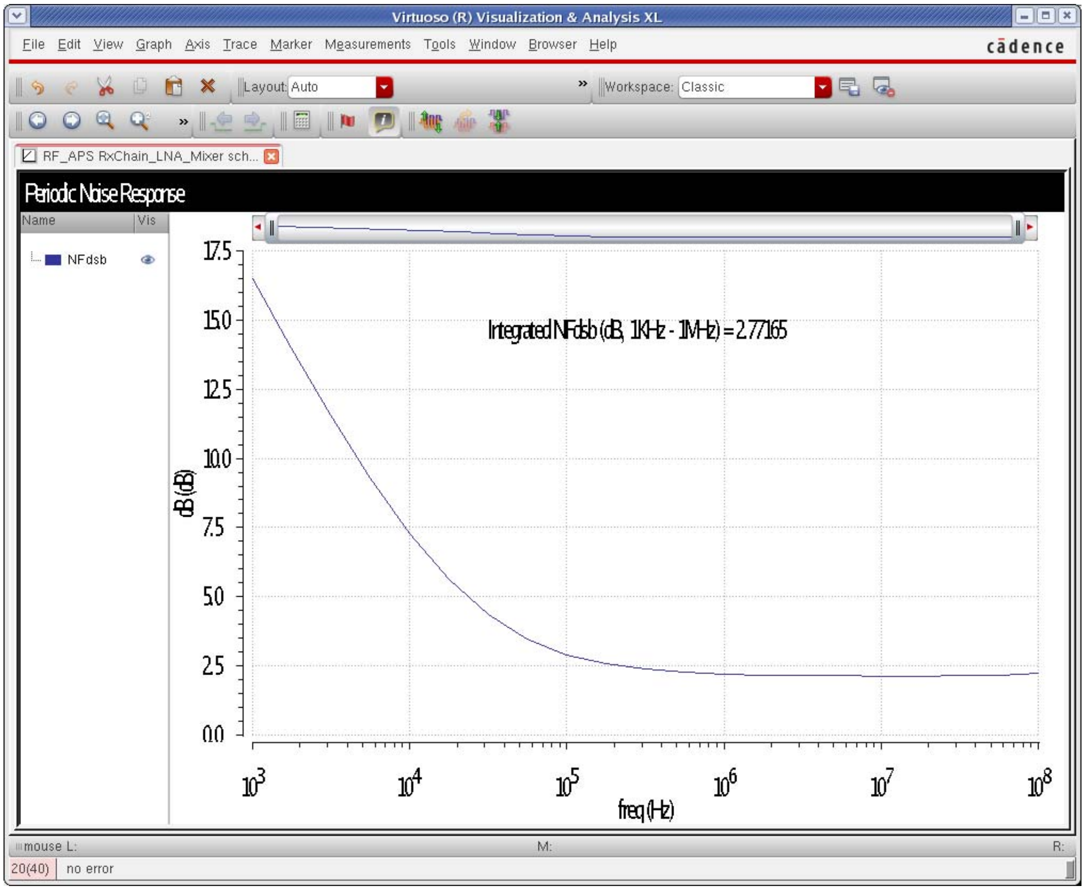
On the *Direct Plot Form*:



1. Select *pnoise* from the *Analysis* section.
2. Select the desired type of noise plot from the *Function* section. *Noise Figure* is single-sideband noise figure.
3. If you want the noise figure over a specified bandwidth, check the *Integrated Over Bandwidth* checkbox.
4. Specify the frequency range you want for the noise figure calculation.

5. Click *Plot*.

The dsb noise figure and the noise figure calculation over your bandwidth are displayed.

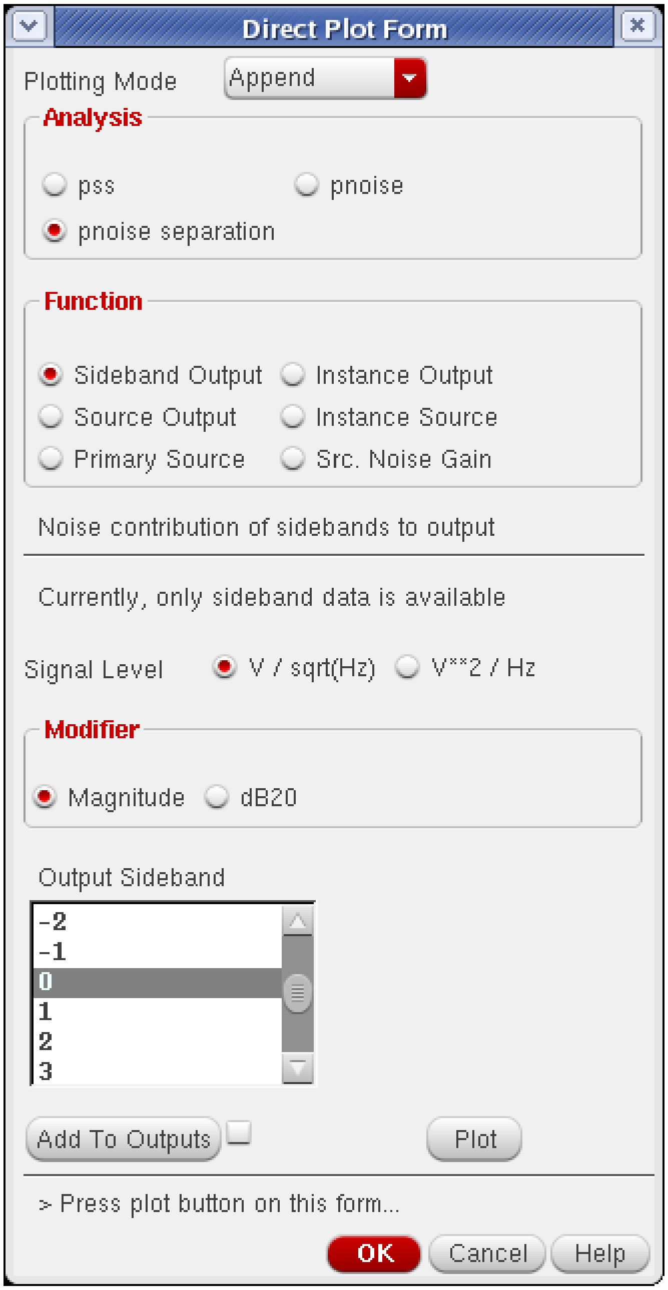


Viewing Noise Separation Results

Viewing the Total Noise at the Output from a Single Noise Input Frequency

This capability allows the noise from all sources in the circuit at different noise input frequencies to be measured. For example, all the noise that does not change frequency (the zero sideband) can be plotted, along with the other noise input frequencies (sidebands) This plotting capability allows problem frequencies to be identified.

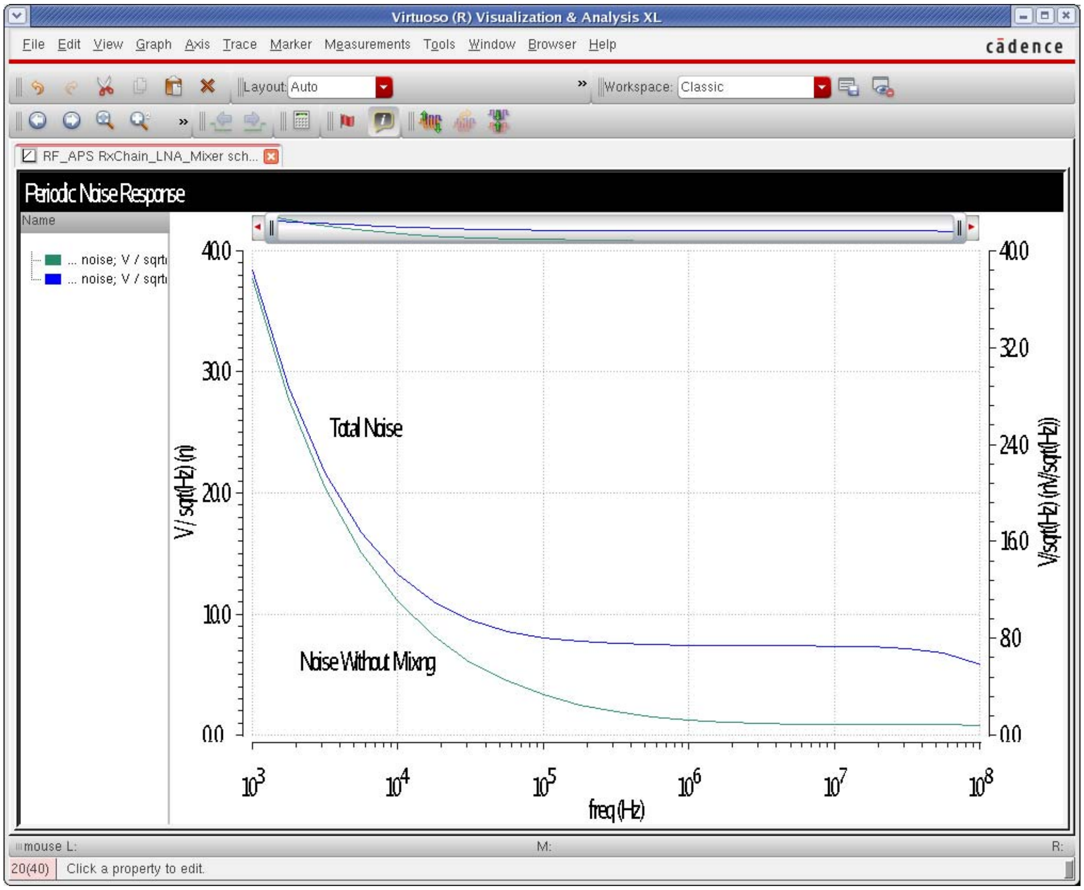
On the Direct Plot Form:



1. Select *pnoise separation* from the *Analysis* section.
2. To view the total noise at the output select *Sideband Output* from the *Function* section.
3. Select *V/sqrt(Hz)* or *v**2/Hz* from the *Signal Level* radio buttons.
4. Select the appropriate modifier from the *Modifier* section.

5. Select the sideband number form the *Output Sideband* list. 0 means no frequency translation.

6. Click *Plot*.



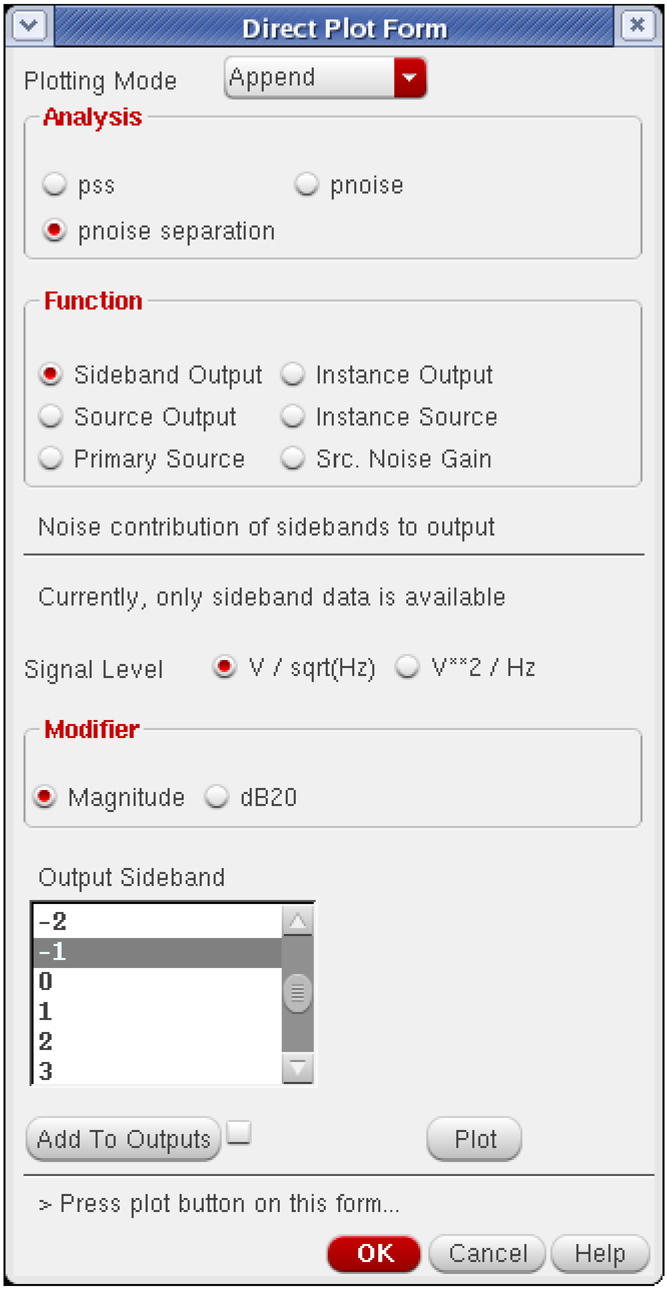
The output noise from sideband zero is plotted. This is the noise that is present at the output with no frequency translations included.

Plotting Noise from Sideband -1

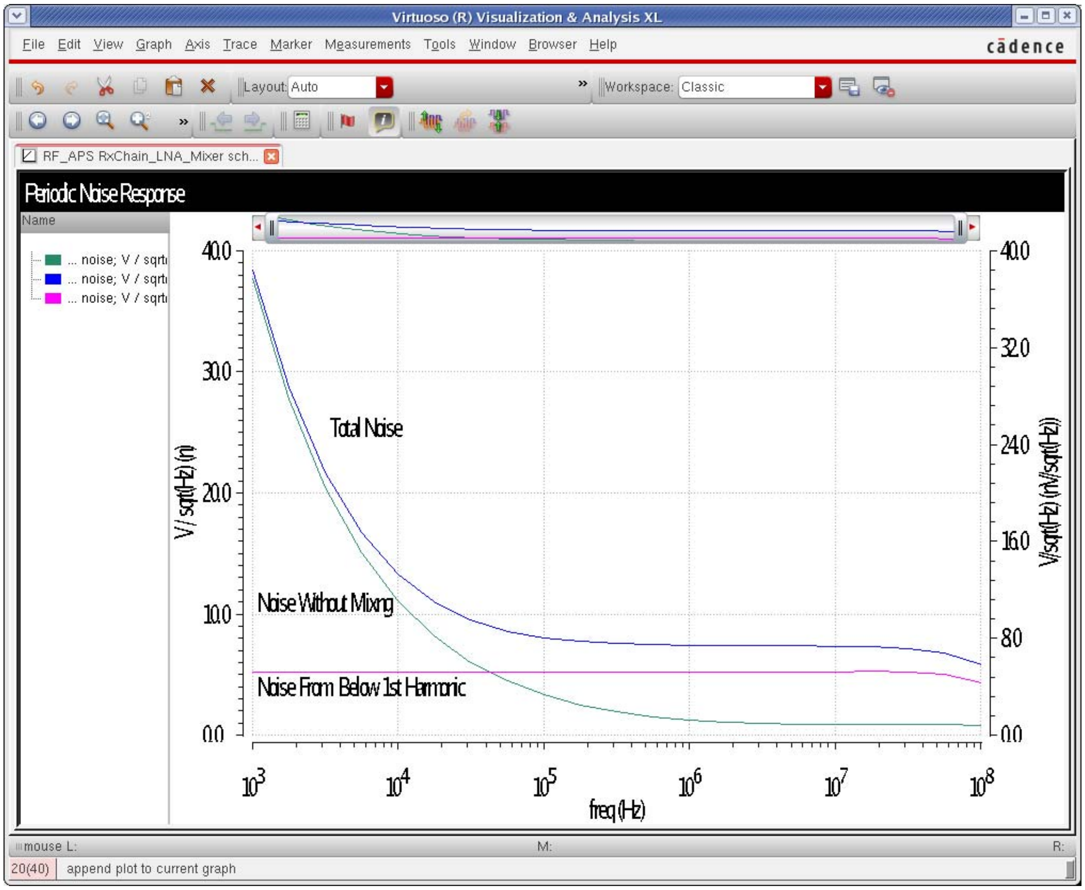
On the *Direct Plot Form*:

1. Select the -1 sideband from the *Output Sideband* list. Selecting the -1 sideband means the noise frequency is shifted down in frequency by -1* first harmonic of pss from the frequency that is specified as the noise output frequency in the *Choosing Analyses*

form. For example, if the pss frequency is 1.9G, and the noise output frequency is 1M, the noise input frequency is -1.899GHz.

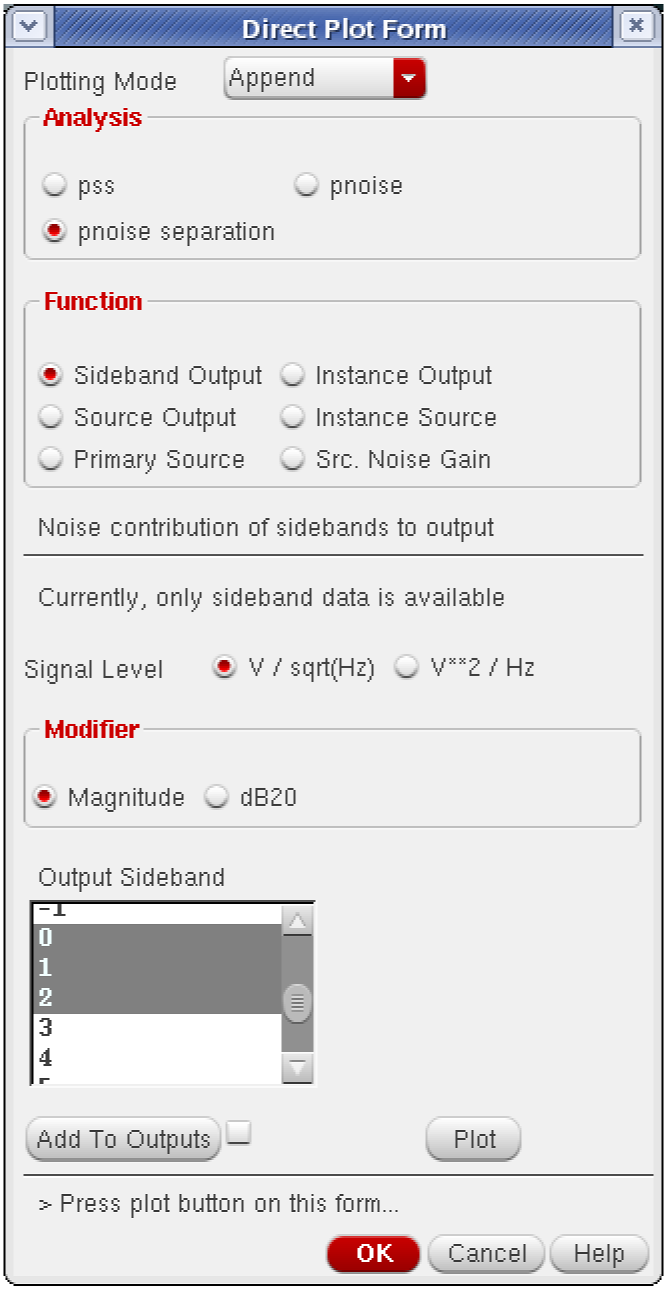


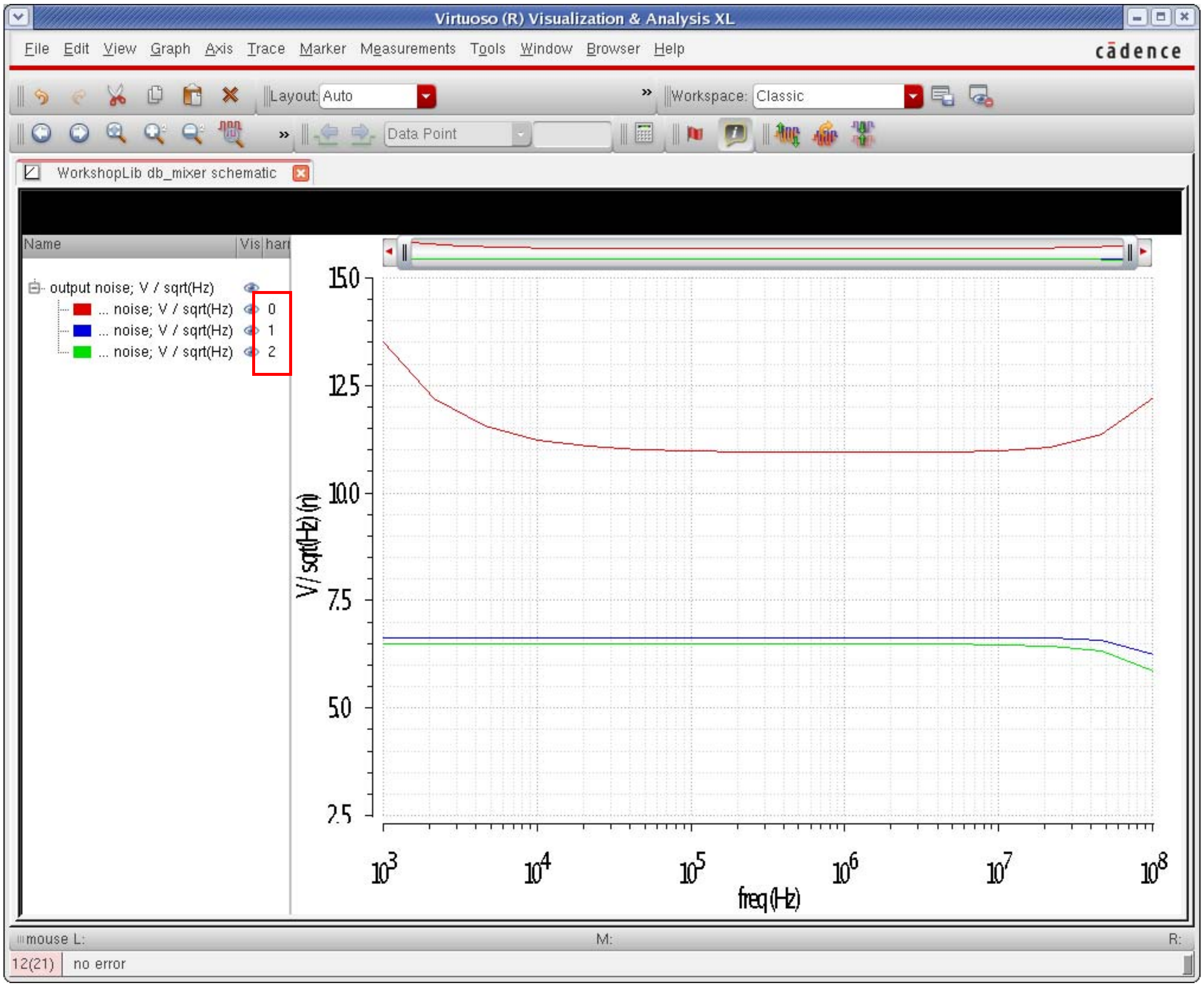
2. Click *Plot*.



As a suggestion, use the *Graph - Add Label* menu option to place labels to identify the traces, as shown in the figure above.

If you select multiple sidebands to be plotted at the same time, the sideband number will be shown at the right side of the legend in the waveform window.

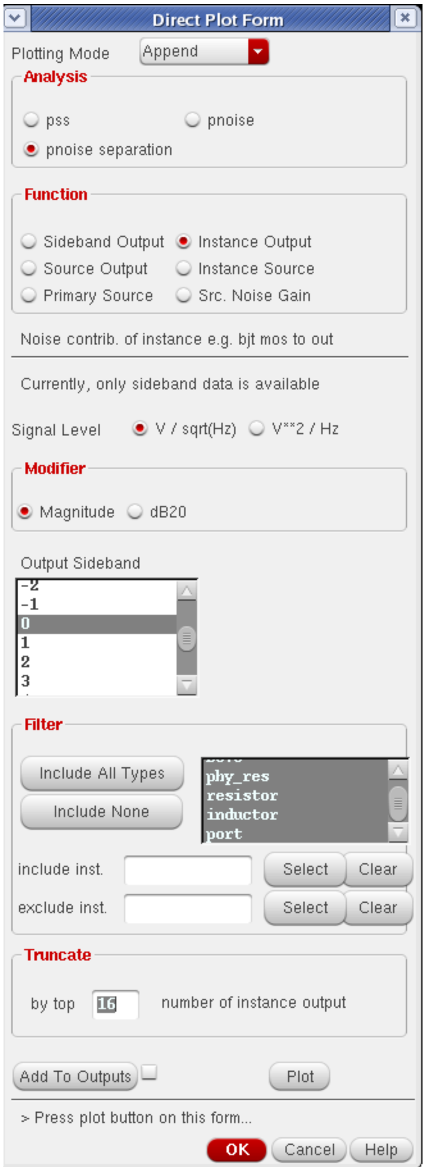




Viewing Total Noise at the Output from each Instance

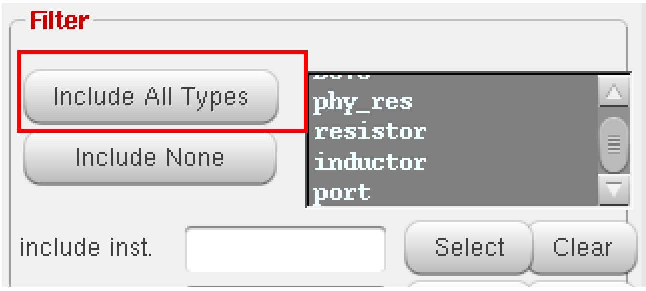
Once the problem frequencies are identified in the plot of Sideband Output, the problem components need to be identified. From the previous plot, the zero sideband contributes the largest amount of noise to the overall output noise. Using the Instance Output allows the identification of which components in the zero sideband contribute the most noise. The total noise for all the noise mechanisms within the component is plotted.

On the *Direct Plot Form*:

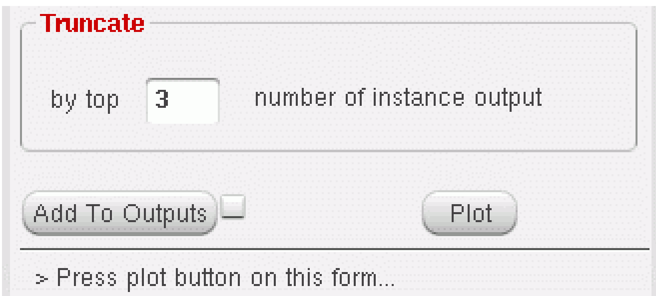


1. Select *Instance Output* from the *Function* section.
2. Select *V/sqrt(Hz)* or *V**2/Hz*.
3. Select the appropriate modifier.
4. Select the sideband number. In this example, the zero sideband was determined to contribute the most noise at the output of the circuit.

5. Select the type of component to display. If you want all of the types, choose *Include All Types*.

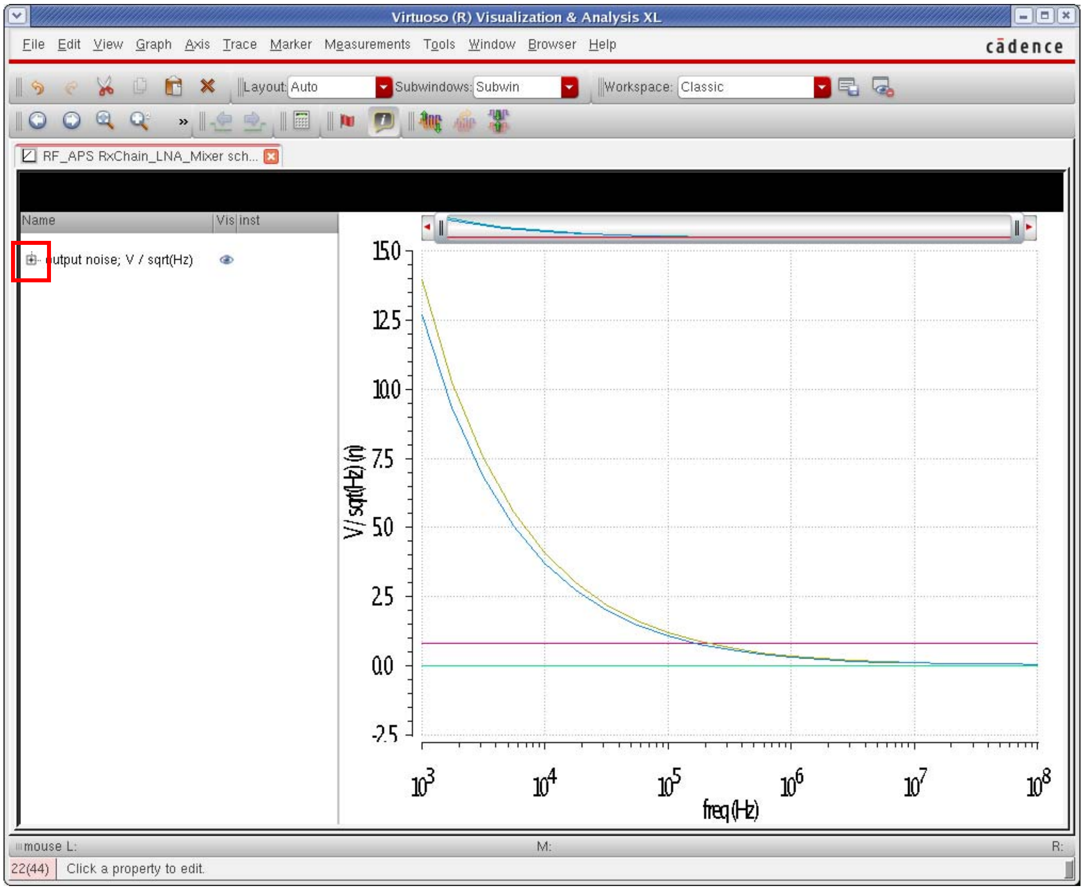


6. Specify the number of results you want to plot.

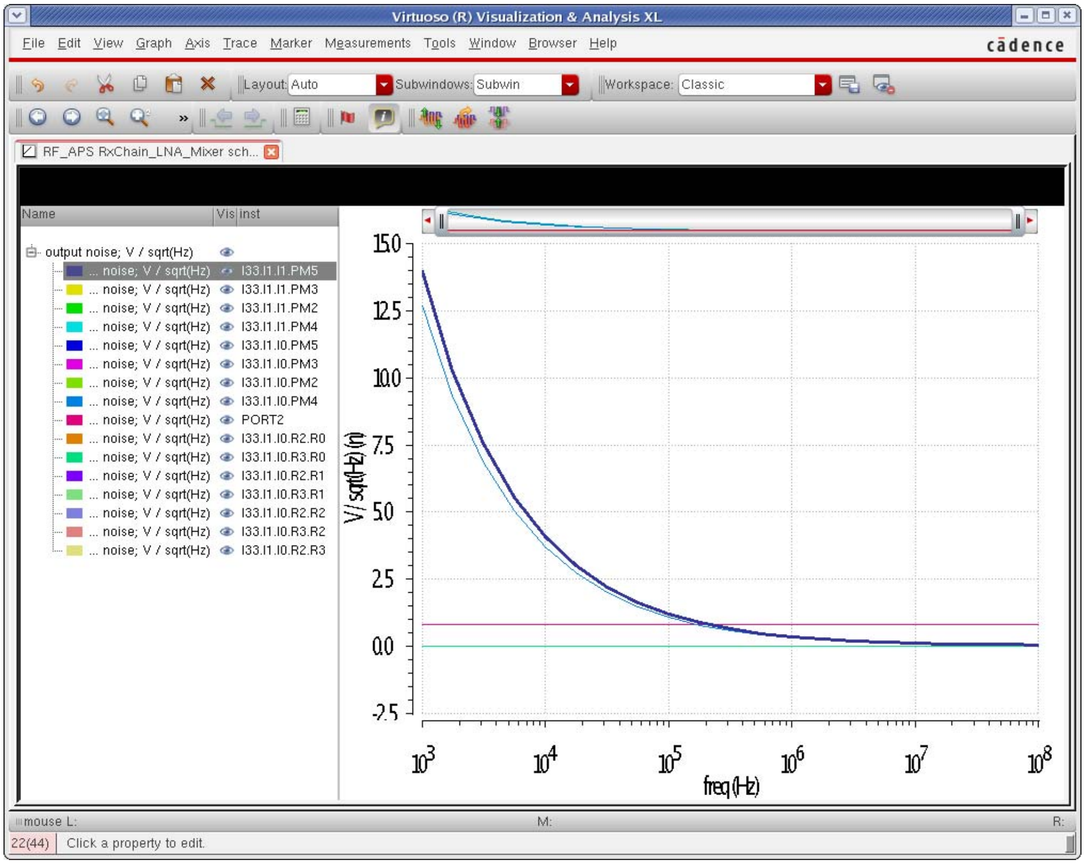


7. Click *Plot*.

The total noise at the output for the top three components is plotted.



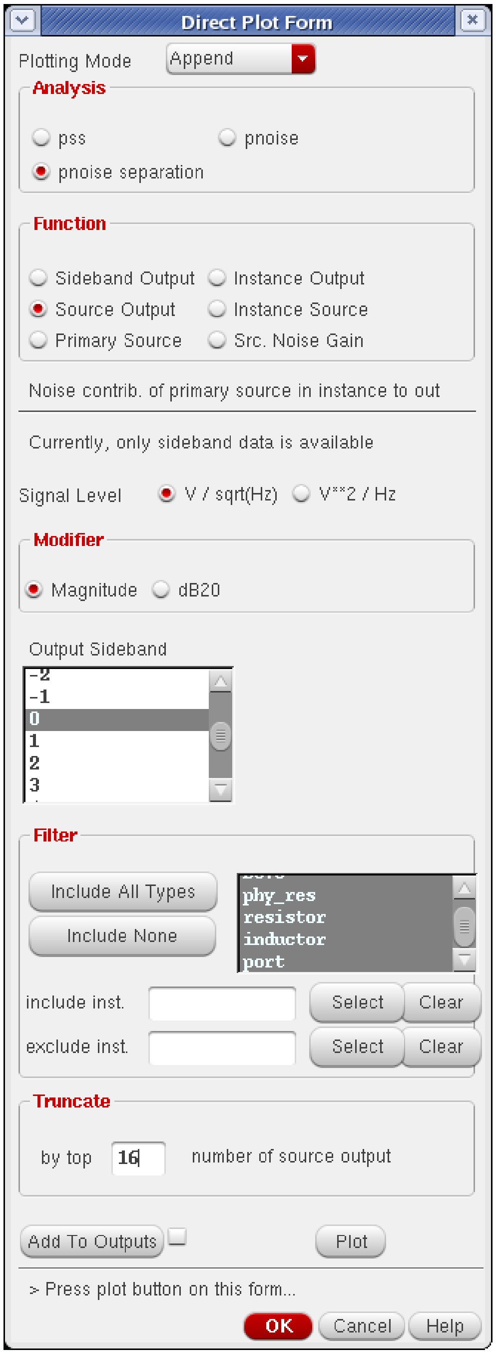
To see which component goes with which trace, select the plus sign (highlighted in red on the previous graphic) to the left or the text at the upper left of the display area. The legends expand, and you can see which trace is which in the Name field on the left side of the display tool. You can also either select the trace or the individual entry in the Name field and the trace and legend will highlight. The instance name is shown on the right side of the legend. You may need to change the size of the legend area by moving your mouse cursor over the line that separates the two areas in the waveform tool. The cursor will change shape. Now click-and-hold the mouse button, move to a new location, and release the mouse button.



Viewing the Exact Noise Mechanism Within the devices

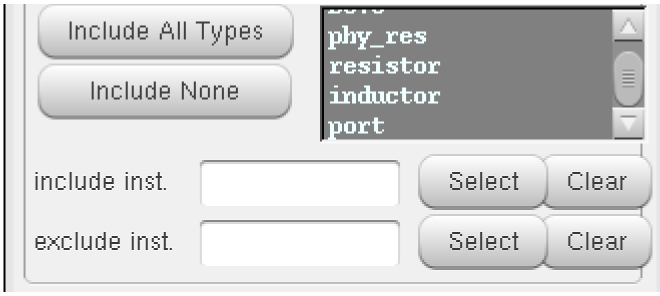
The sideband with the largest contribution was determined by selecting Sideband Output and plotting. The specific component was identified by selecting Instance Output. The specific noise mechanisms can be identified using Source output. This plots the largest individual noise mechanisms for the sideband you select.

On the *Direct Plot Form*:



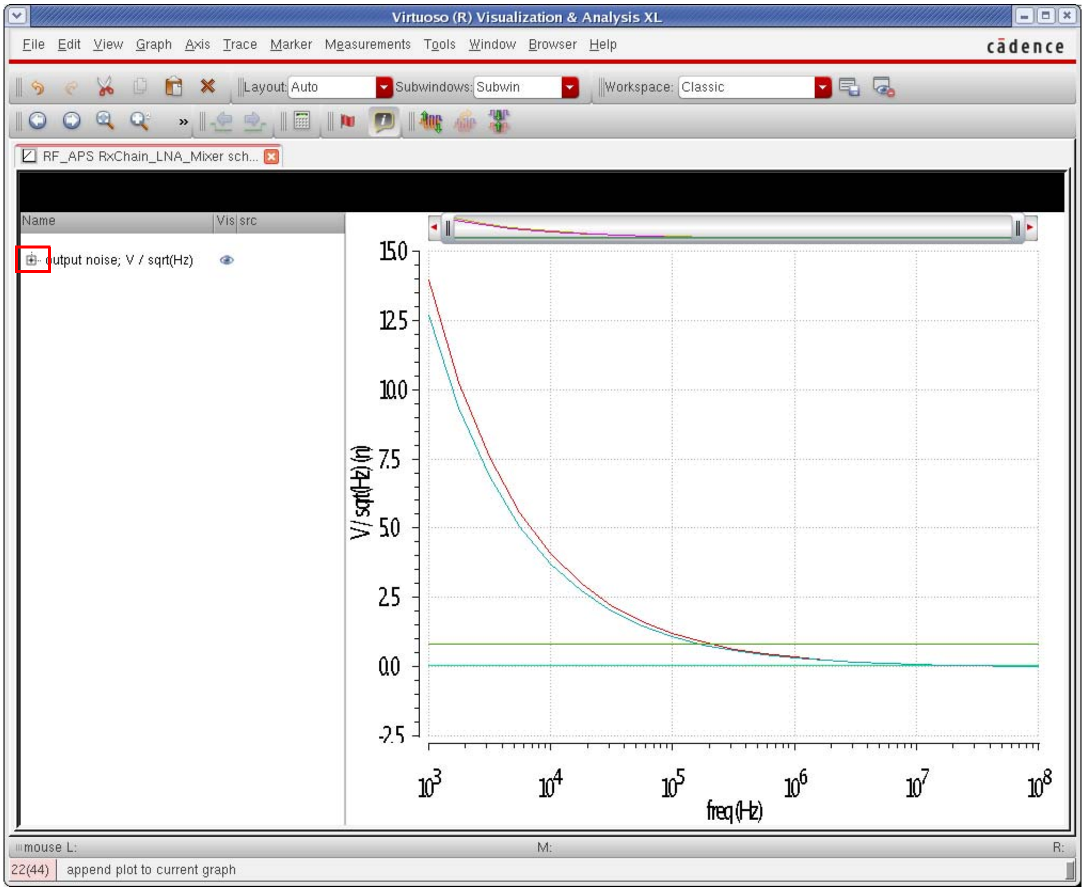
1. Select *Source Output* from the *Function* section.
2. Select *V/sqrt(Hz)* or *v**2/Hz*.
3. Select the appropriate modifier.

4. Select the sideband number.
5. Select the type of component to display. If you want all of the types, choose *Include All Types*.

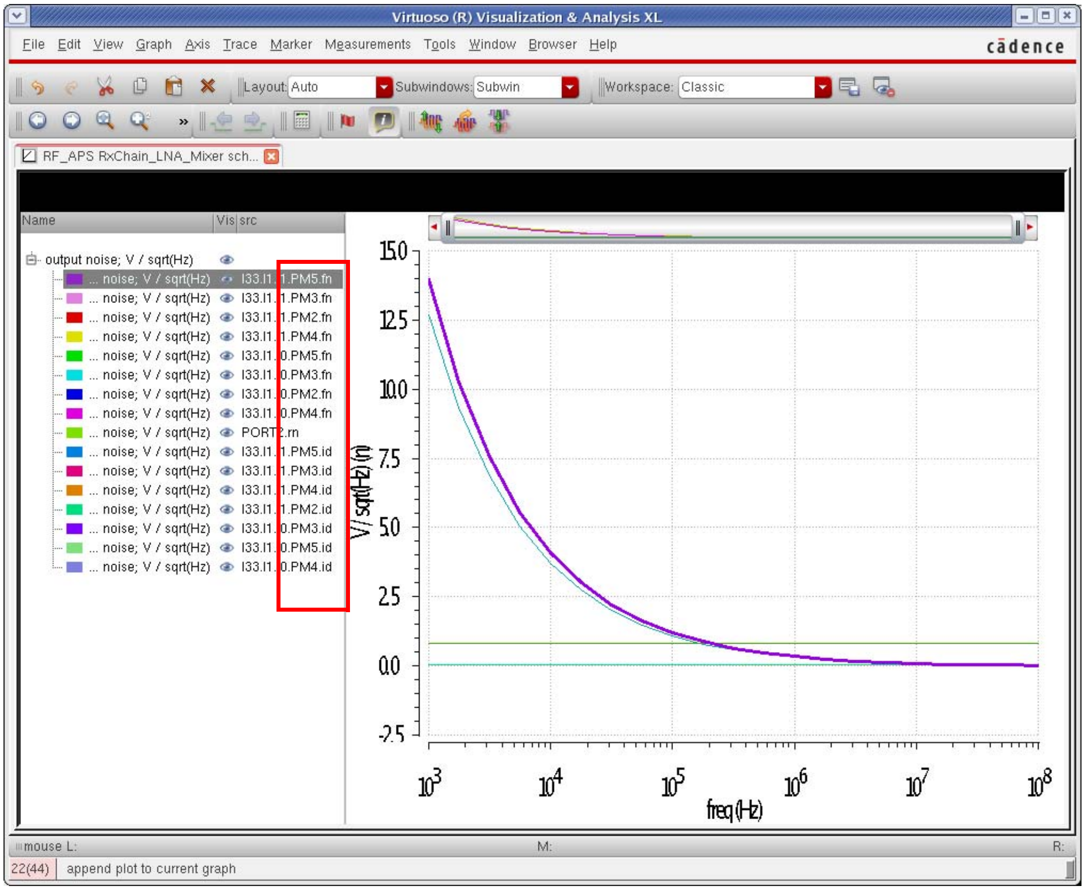


6. If you want to restrict to one device, put its instance name in the *include inst* field. A list of devices separated by a space is also acceptable.
7. If you want to exclude a device (or devices) enter the instance name(s) in the *exclude inst* field.
8. Specify the number of results you want to plot in the *by top* field.
9. Click *Plot.*

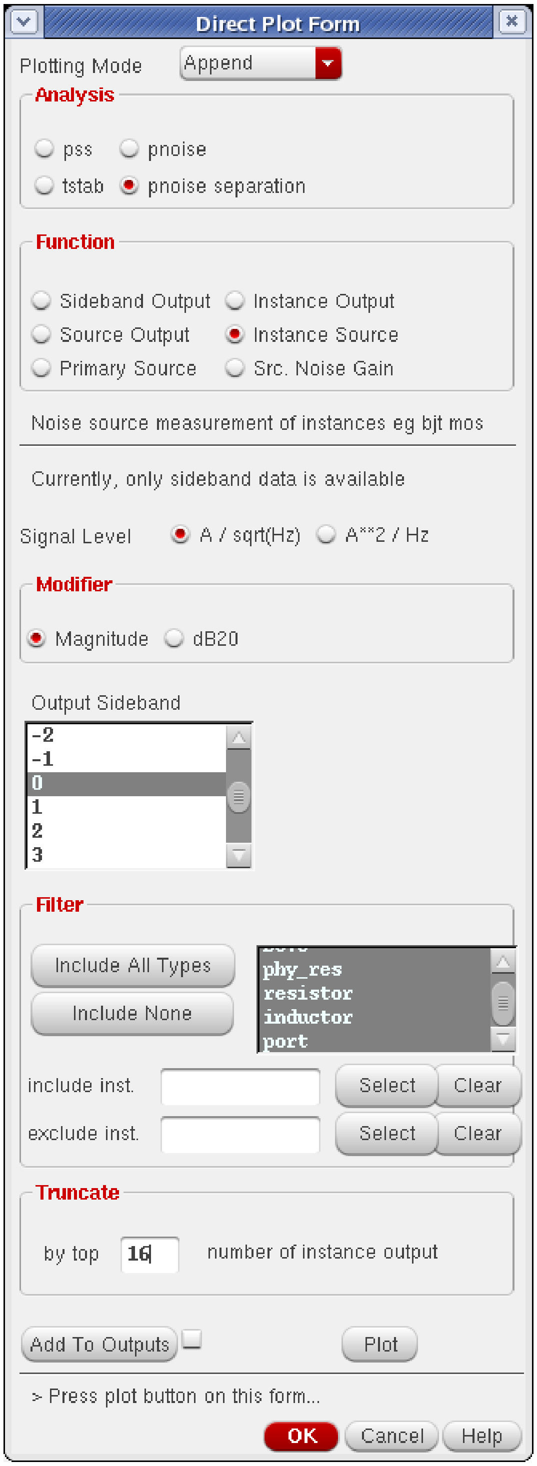
The exact noise mechanism within the devices is plotted.



To see which noise source goes with which trace, select the plus sign to the left or the text at the upper left of the display area. The legends will be shown individually in the Name field of the waveform tool. The device name and the individual noise contributor will be shown. You can select either the individual legend or the trace and both the legend and the trace will be highlighted. The first part is the instance name, and the second part is the specific noise mechanism within that instance. For a list of the abbreviations used for the noise parameter, see the list that starts on page 123 (77 of 234) In the example below, fn is flicker noise, and id is the channel resistance thermal noise.



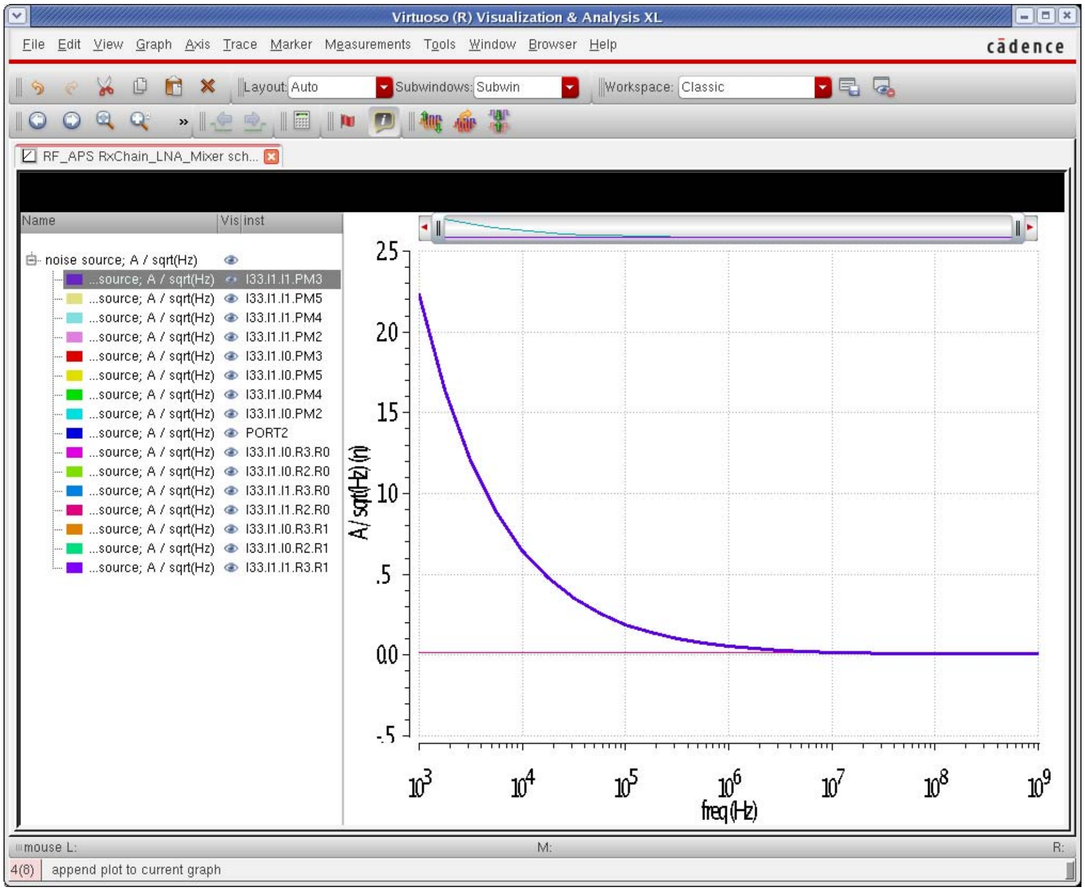
Viewing the total noise currents within the devices



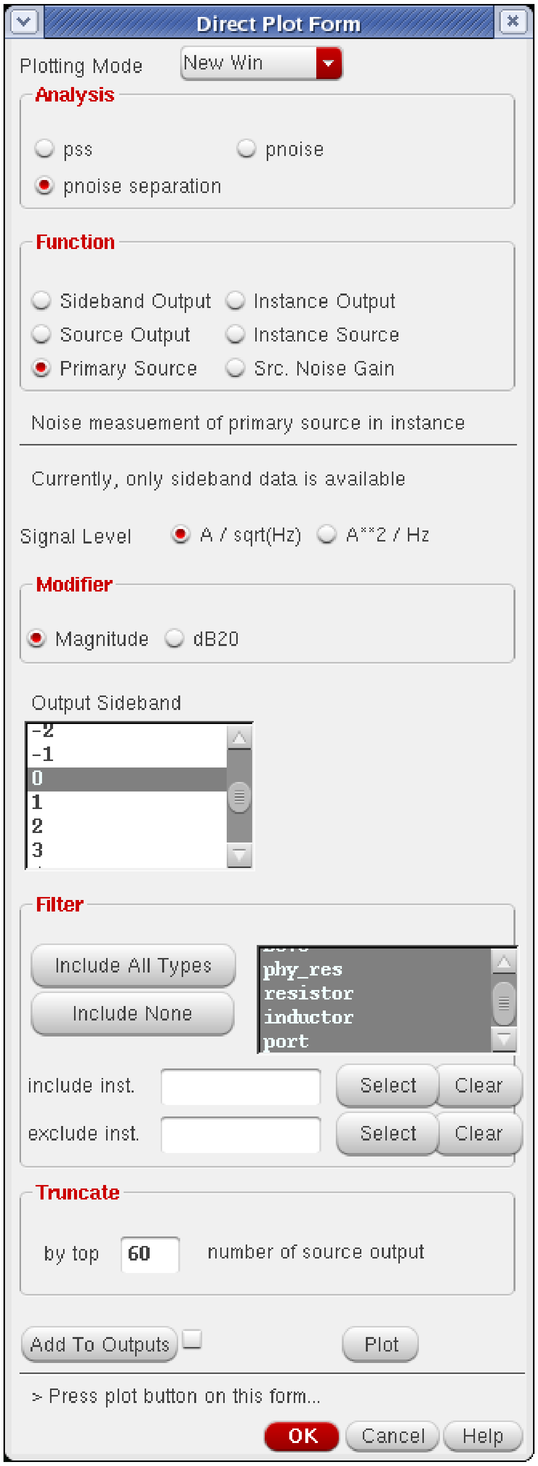
This capability has limited value. All the noise currents from all the noise mechanisms within the instance are added up. No gain function is applied or are available for this measurement. It just adds up everything within that instance.

1. Select *Instance source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the Signal Level selection.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

10. The frequency response curves are displayed.



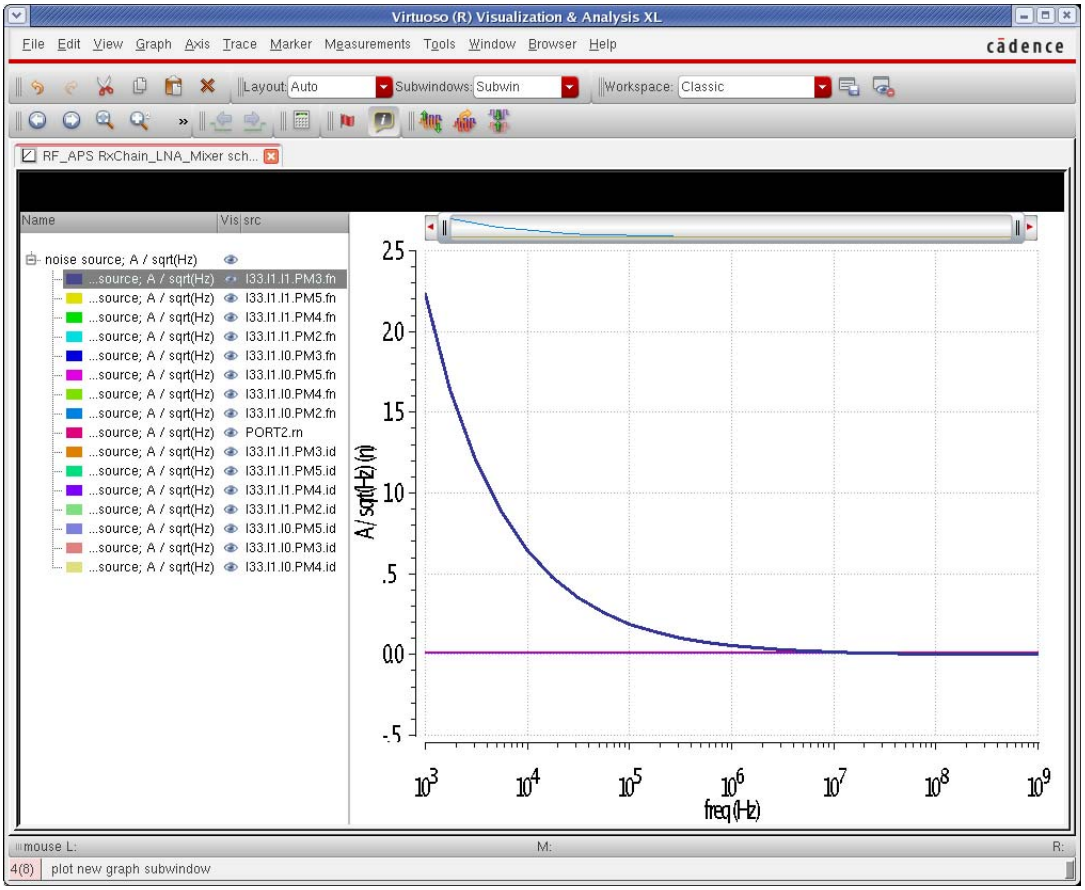
Plotting the individual noise source currents



This capability plots the individual noise currents within the device. This capability gives you the noise at the source instead of at the output of the simulation.

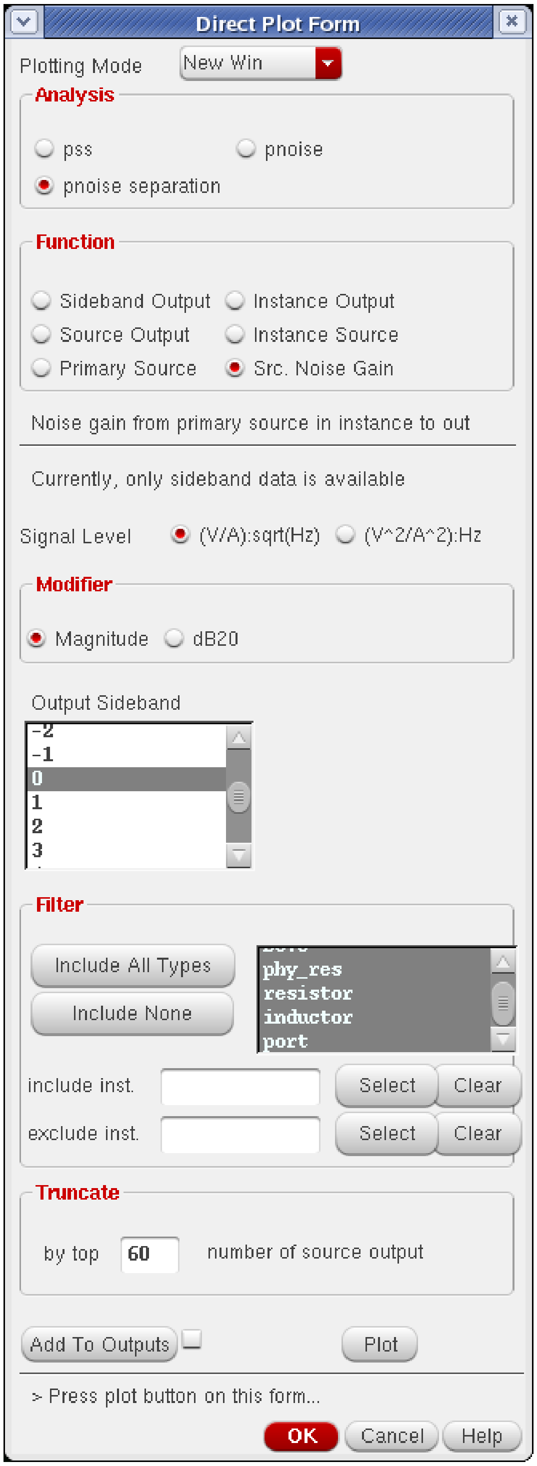
1. Select *Primary Source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* selection.
3. Select *Magnitude* or *dB20* in the Modifier section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

10. The frequency response curves are displayed.



For a list of the noise parameter names, please see page 123 (77 of 234) of this chapter.

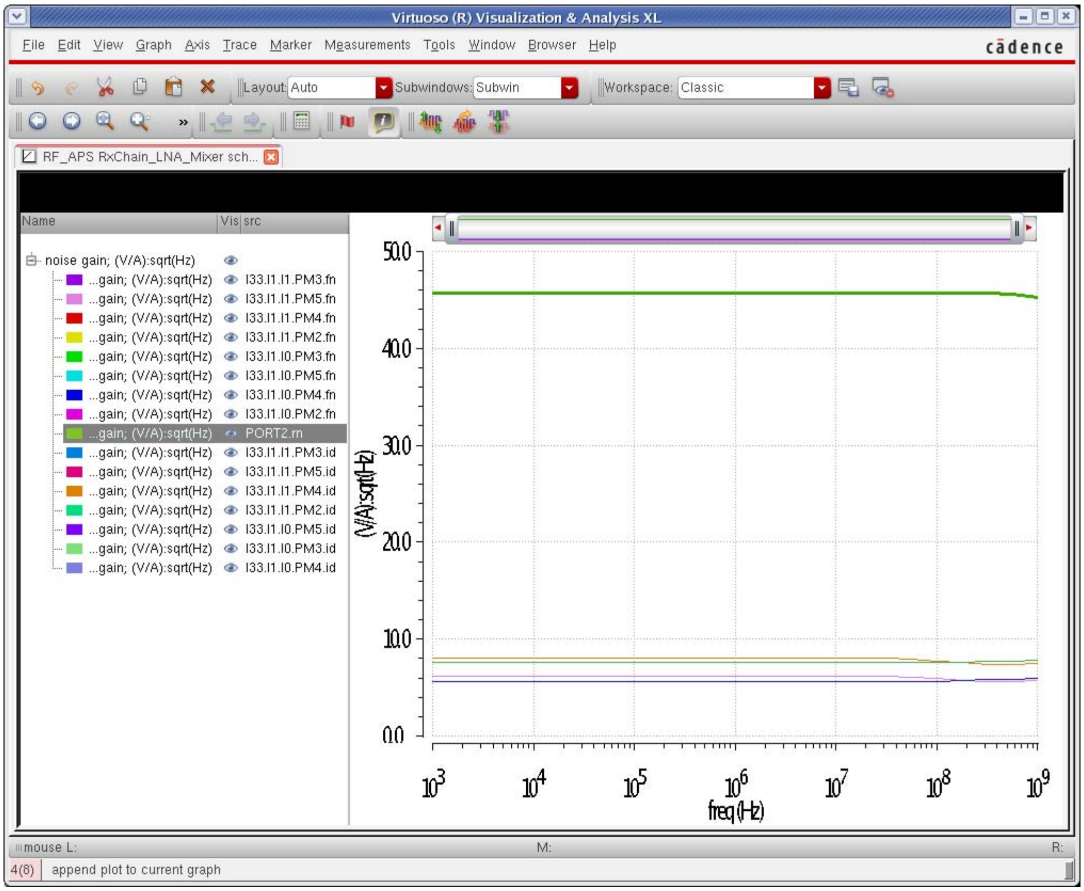
Plotting the gain from the individual noise sources to the output



This capability gives the gain from the individual noise currents to the output of the circuit.

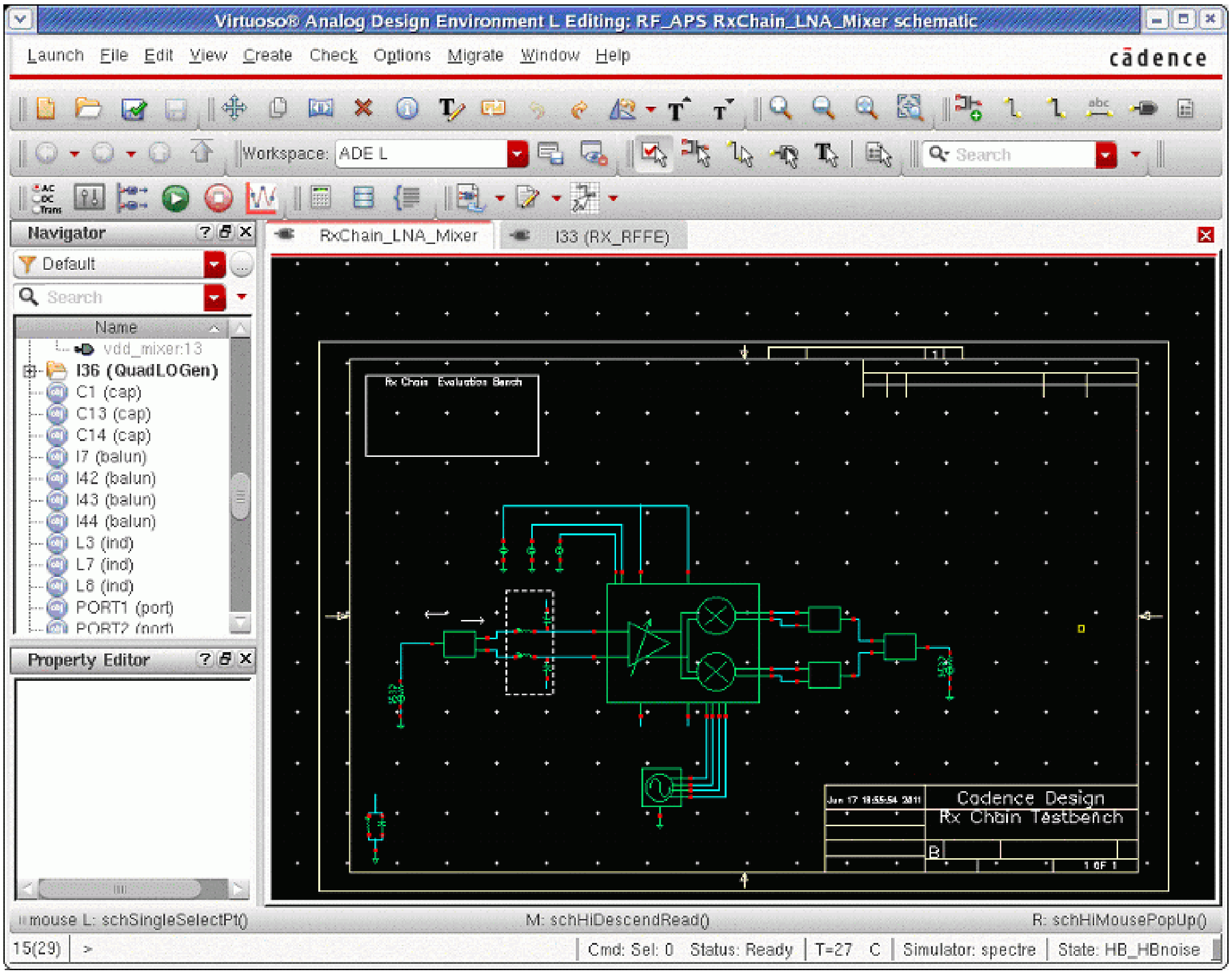
1. Select *Source Noise Gain*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* selection.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select *Include All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.

The frequency response curves are displayed. For a list of the abbreviations used for the noise parameters, please see page 123 (77 of 234) of this manual.



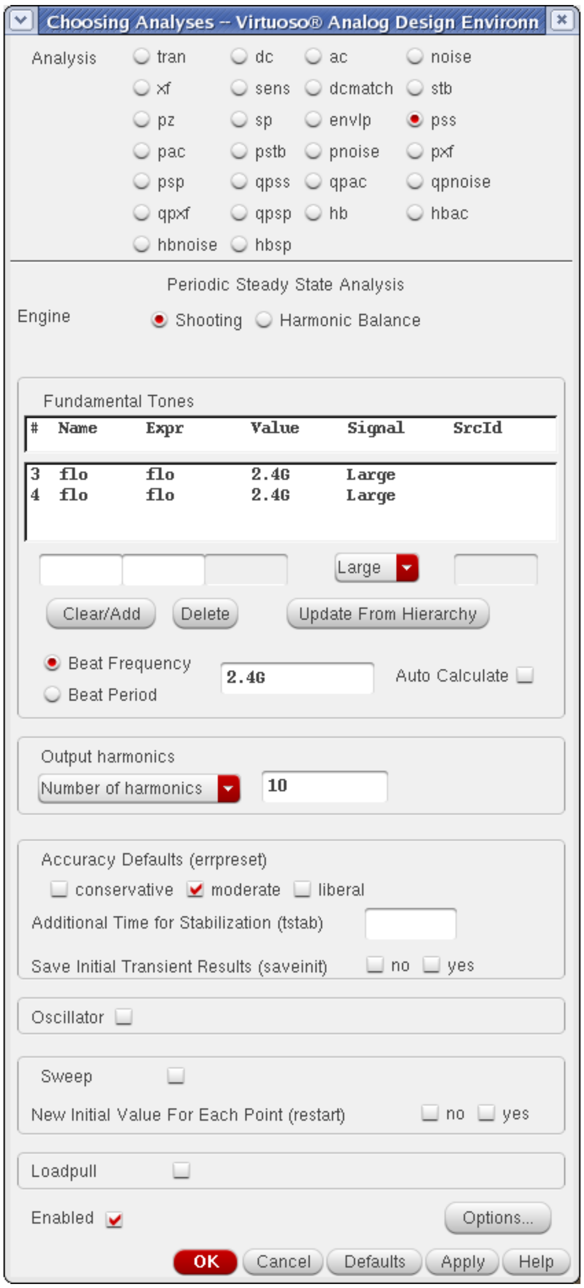
Multiple pnoise

Consider an LNA-Mixer circuit shown below.

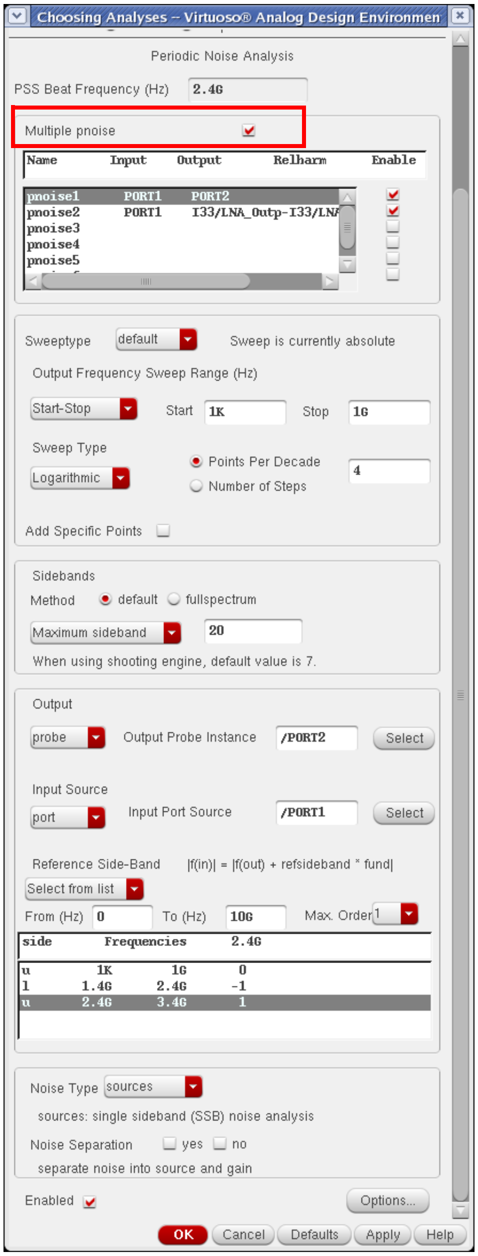


In the above circuit, there are two outputs that need to be measured. The first is at the LNA output, and the second is at the mixer output. The noise frequencies are different at the two outputs.

1. First, set up the pss form as appropriate for your design. For more information on the pss analysis, please see the periodic steady-state analysis at the beginning of this chapter.

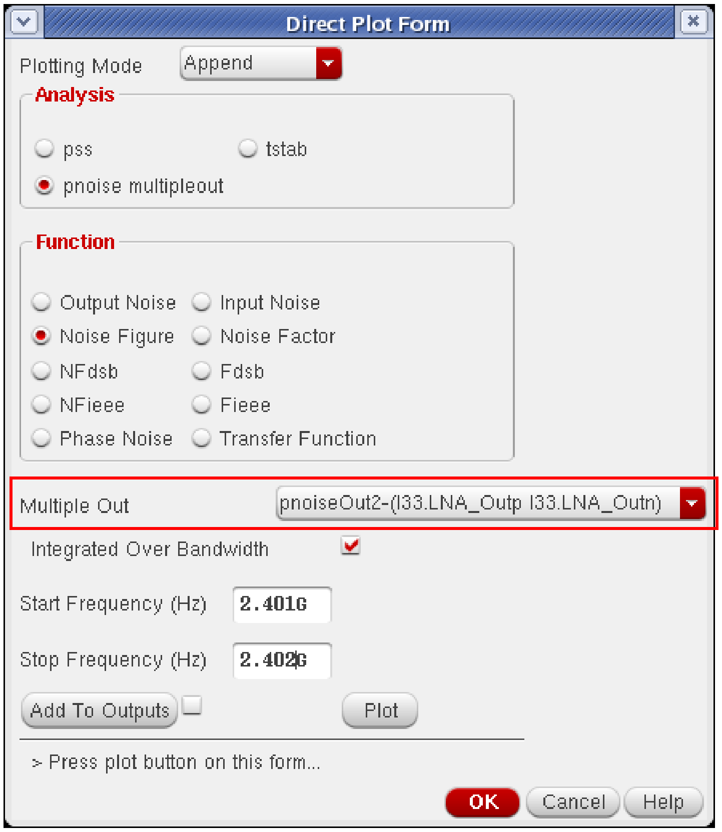


2. In the pnoise *Choosing Analyses* form, select *Multiple pnoise*, and set up the noise analysis for one of the outputs. The mixer output is shown below.

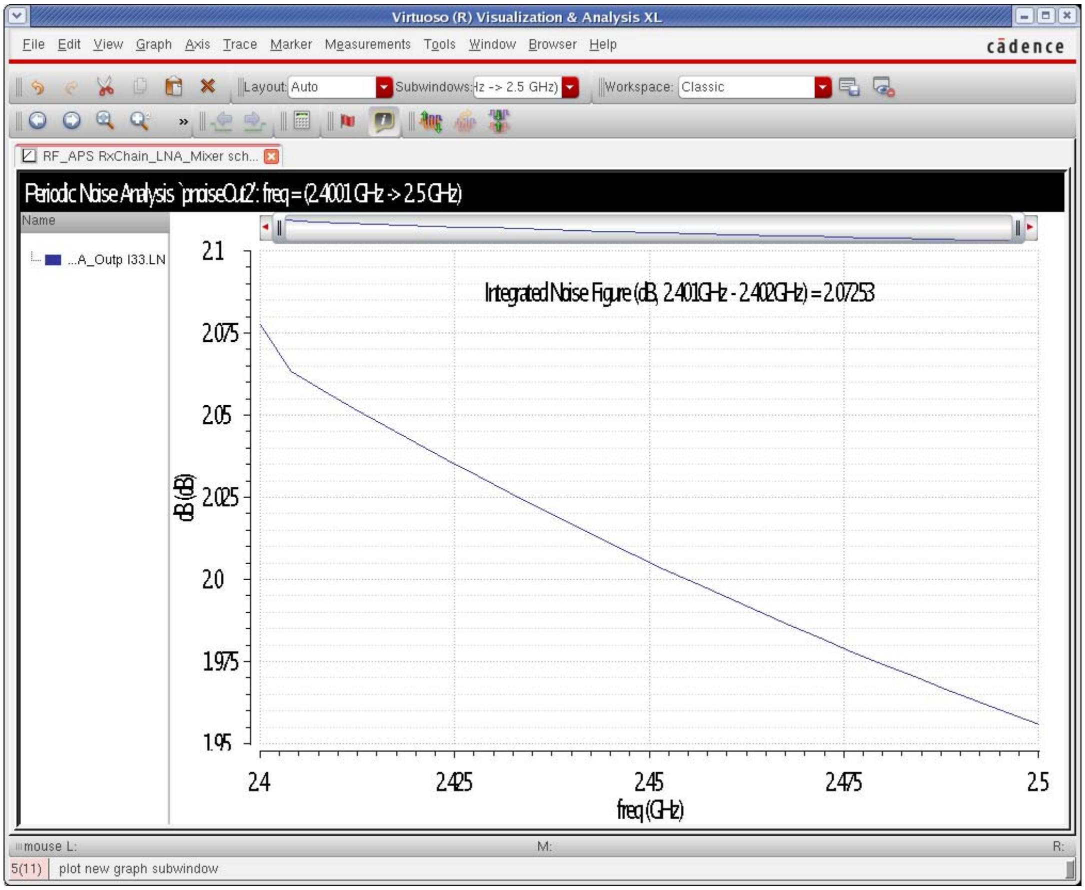


3. Select the second line in the *Multiple pnoise* section and set up that analysis.

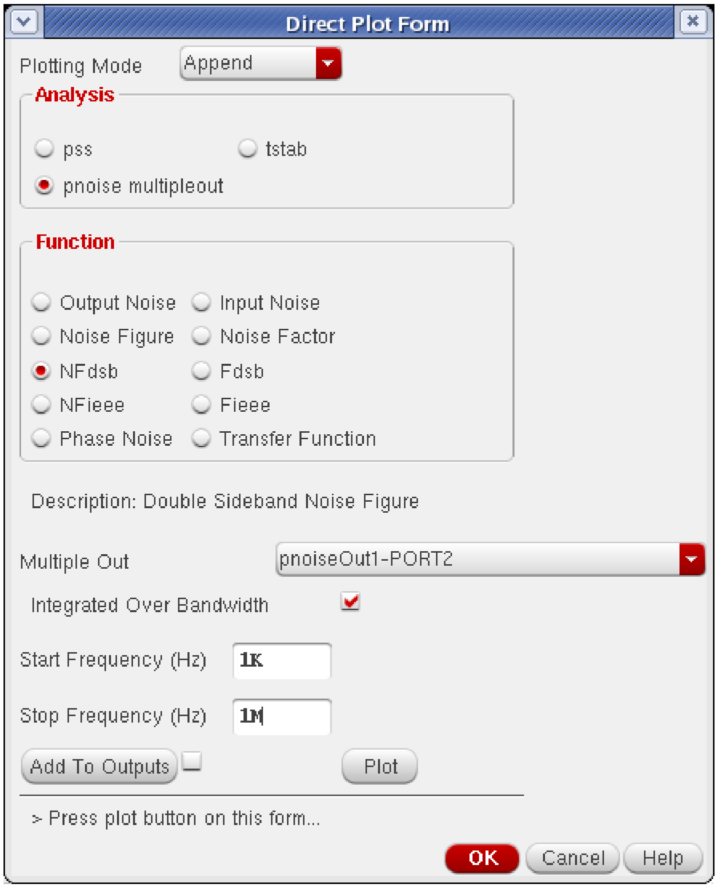
4. Run the simulation. When the simulation completes, select *Results - Direct plot - Main Form*. The single-sideband noise figure measurement for the LNA is shown below. The selection mechanism for the noise result is called Multiple Out and is highlighted below.



5. Click *Plot*.

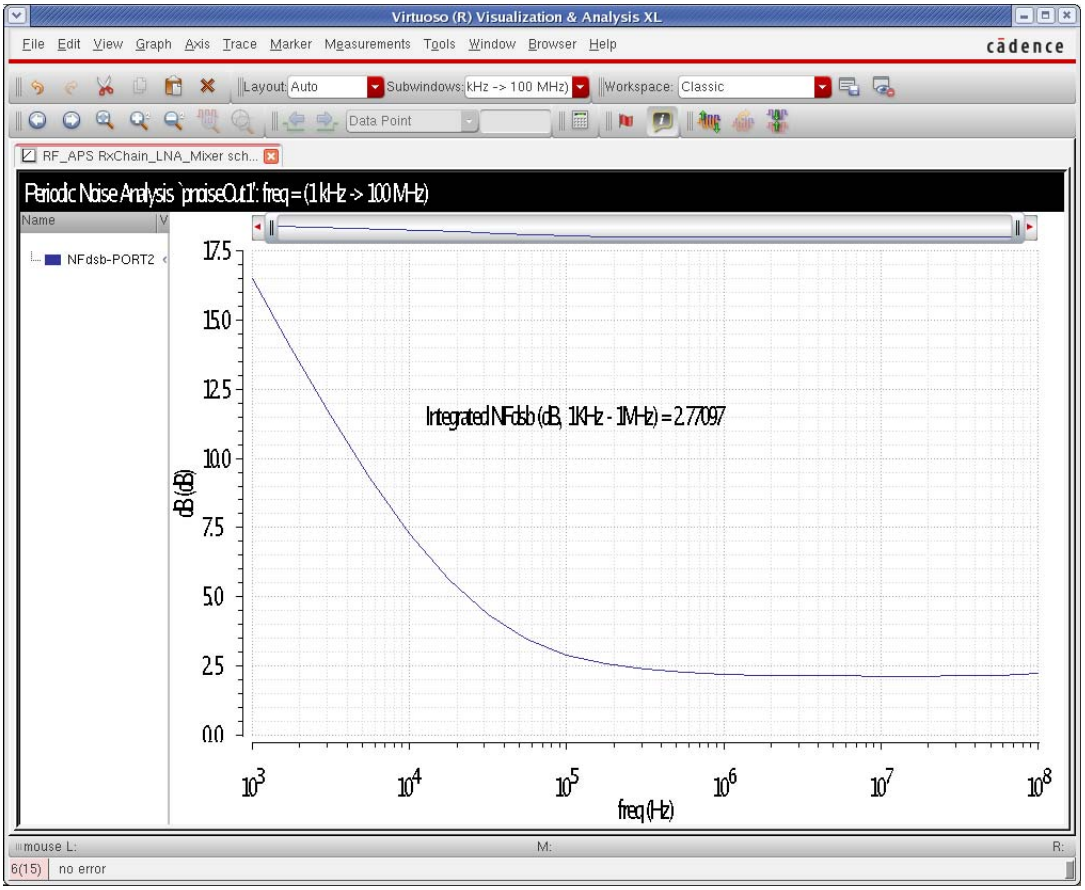

Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Layout: Auto
Subwindows
Workspace: Classic
RF_APS RxChain_LNA_Mixer sch...
Periodic Noise Analysis `pnoiseOut2': freq = (2.4001 GHz -> 2.5 GHz)
Name
...A_Outp I33.LN
Integrated Noise Figure (dB, 2.401GHz - 2.402GHz) = 2.07253
dB (dB)
freq (GHz)
mouse L:
M:
R:
5(11)
plot new graph subwindow

6. To plot the noise figure of the overall circuit, set the *Multiple Out* selection to the Mixer output at Port2. For direct conversion, usually double-sideband noise figure is desired.

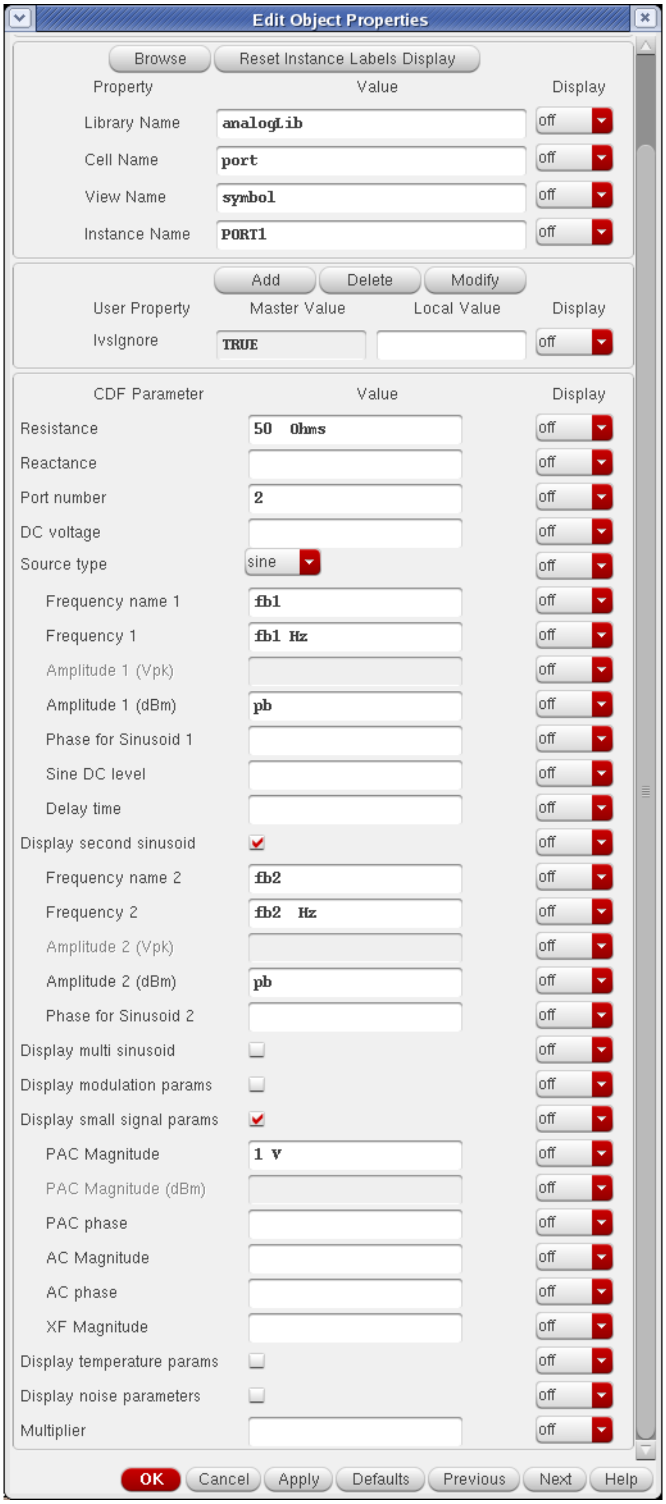


7. Click *Plot*.

The waveform tool shows the noise figure curve and the integrated noise figure to 1MHz.



Noise as a Function of Blocker Power



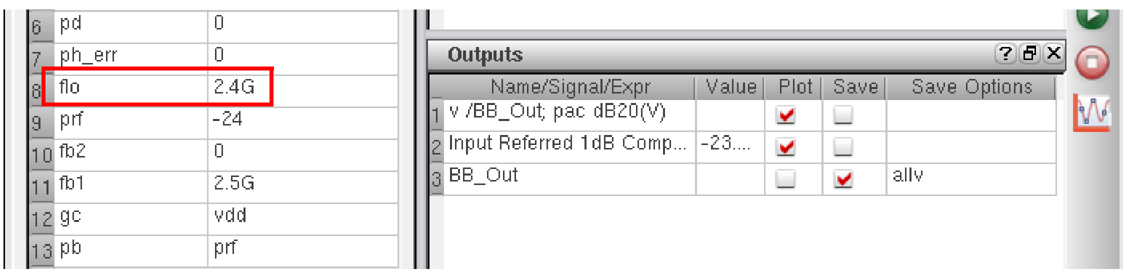
- Note that the input source has 2 large-signal tones set up.

- The first large-signal tone has the frequency set to the variable `fb1` and the amplitude set to `pb`.
- The second tone has the frequency set to the variable `fb2` and the amplitude set to `pb`. This is the blocker signal.

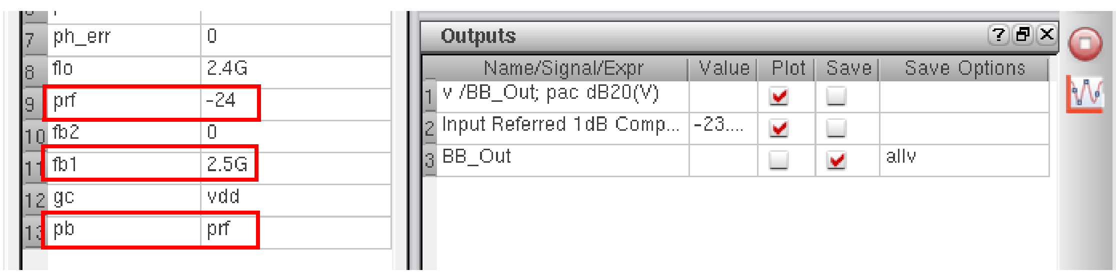
To measure the noise as a function of blocker power:

1. Apply the signals that cause the large-signal effects in the pss analysis, and then run a pnoise analysis. In this case, the LO and the RF signal should be applied in the pss analysis. Note that if the highest input frequency divided by the PSS beat frequency is above about 25, it is very likely that hb or qpss would run faster than pss because hb and qpss only calculate the actual frequencies produced by the circuit, while pss calculates all the harmonics of the beat frequency.

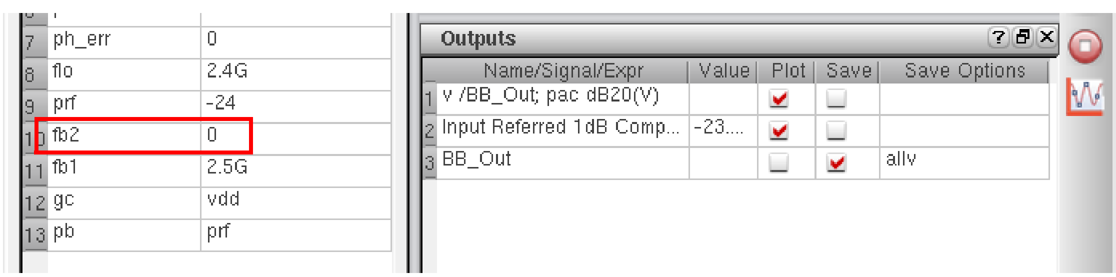
- ❑ The LO is set to 2.4GHz.



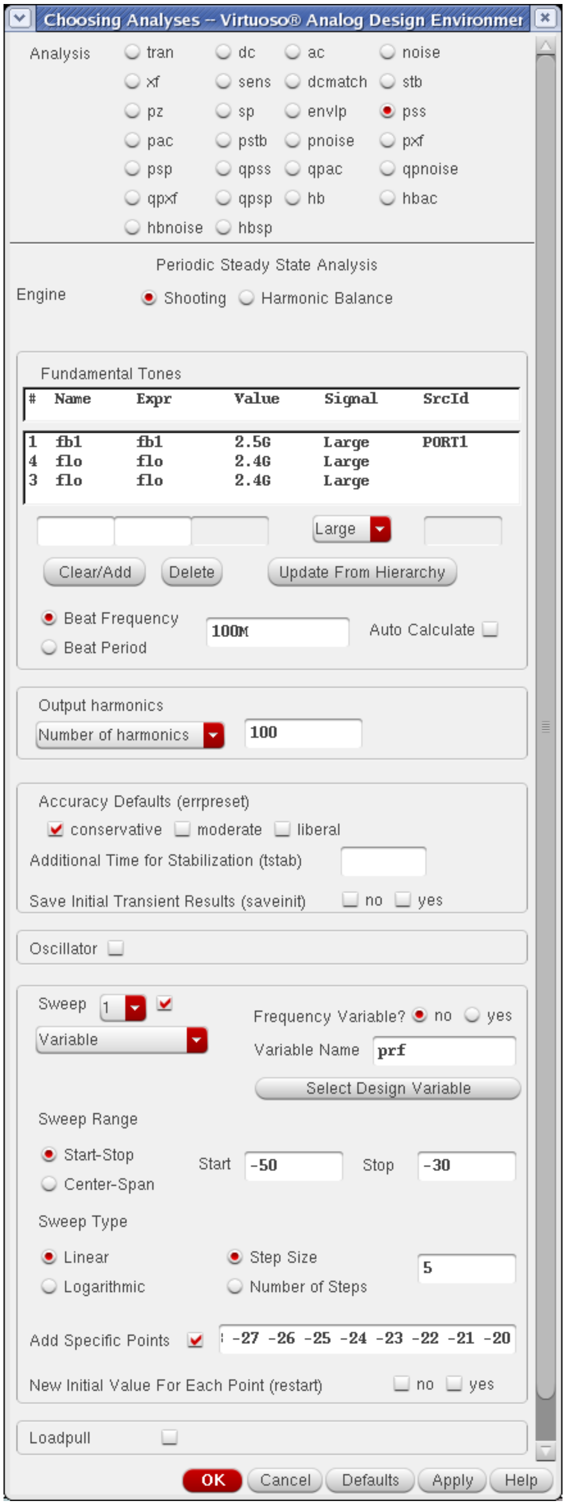
- ❑ The first RF is set to 2.5G and -24 dBm.



- ❑ The second RF tone is disabled by setting its frequency to zero.

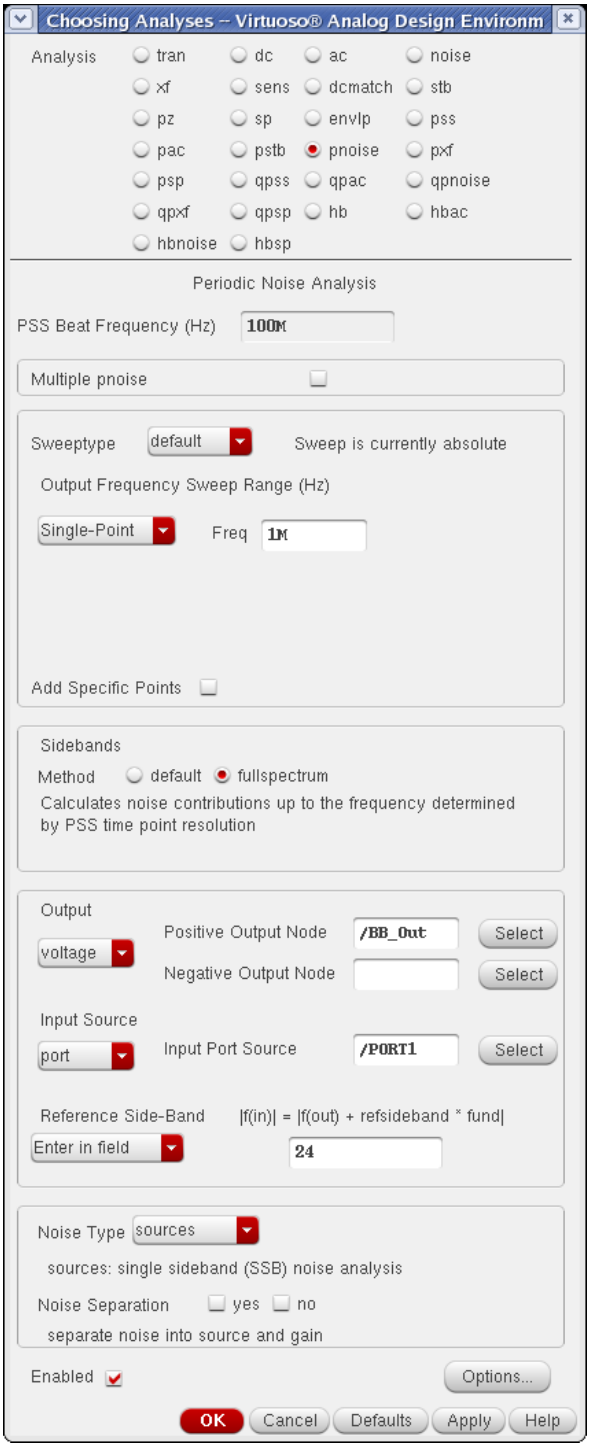


On the pss *Choosing Analyses* form:



1. Set up the pss analysis appropriately. For more information, see the pss section at the beginning of the chapter.
2. Sweep the RF power over an appropriate range. Generally speaking, the power should start about 20 to 40 dB below the below the 1dB gain compression point, and sweep through the power level that is needed to meet the specification for the communication system you are working on. In general, this higher level would be near the compression point, or perhaps slightly above it.

On the pnoise *Choosing Analyses* form:



1. Usually, a single frequency is run in the pnoise, when the blocker power is swept.

2. Specify the desired output frequency.

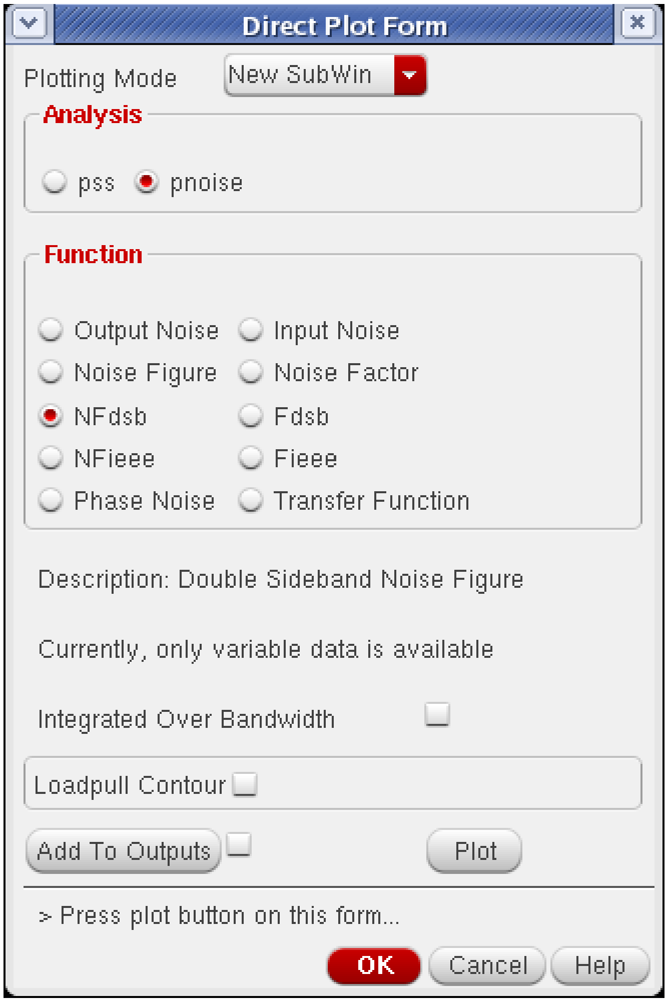
   Shooting: Because to go through the fifth harmonic of 2.5GHz requires that sidebands be set to 125, (5*25th harmonic of 100MHz), fullspectrum pnoise is chosen so the simulation time is reduced. When fullspectrum is chosen, the number of frequency translations depends on the maximum step in the pss analysis. The maximum step of the pss analysis depends on how the maximum harmonic field is set. In this case, to verify that enough sidebands are being used, first set the maximum harmonics term to a bit less than the number anticipated for pnoise. Run the simulation, and plot the noise figure. Roughly double the number of pss harmonics, and run the simulation again. If the noise figure plot did not change, you had enough to start with, and try reducing the number of harmonics in the pss analysis. Reducing the number of harmonics below 10 will have no effect on the simulation. If the noise figure increased, then double the number of harmonics again and re-run the simulation. Use the smallest number of harmonics that produces a stable pnoise result. If the default pnoise method is used, start with an estimate of the number of frequency translations you need based on the nonlinearity of the system. Set that as the number of sidebands, Divide this number by 4, and set that as the number of pss harmonics. Run the simulation, and plot the noise result. Now double the number of pss harmonics and the number of pnoise sidebands. Rerun the simulation and replot the noise result. If the noise result did not change, you had enough harmonics and sidebands to begin with. If it did change, double again, and replot. Use the smallest number of harmonics and sidebands that give a stable result.

   Harmonic Balance: Leave the *Maximum sideband* field blank, and estimate the number of required harmonics in the pss analysis. Run the simulation and plot the noise. Now increase the number of harmonics in pss by about 50%, and run the simulation again. If the noise changed, increase the pss harmonics and run again. If the noise did not change, reduce the number of pss harmonics an run again. Use the smallest number of harmonics you can in pss that maintains the stable noise output in pnoise.

3. The rest are set as the preceding section.

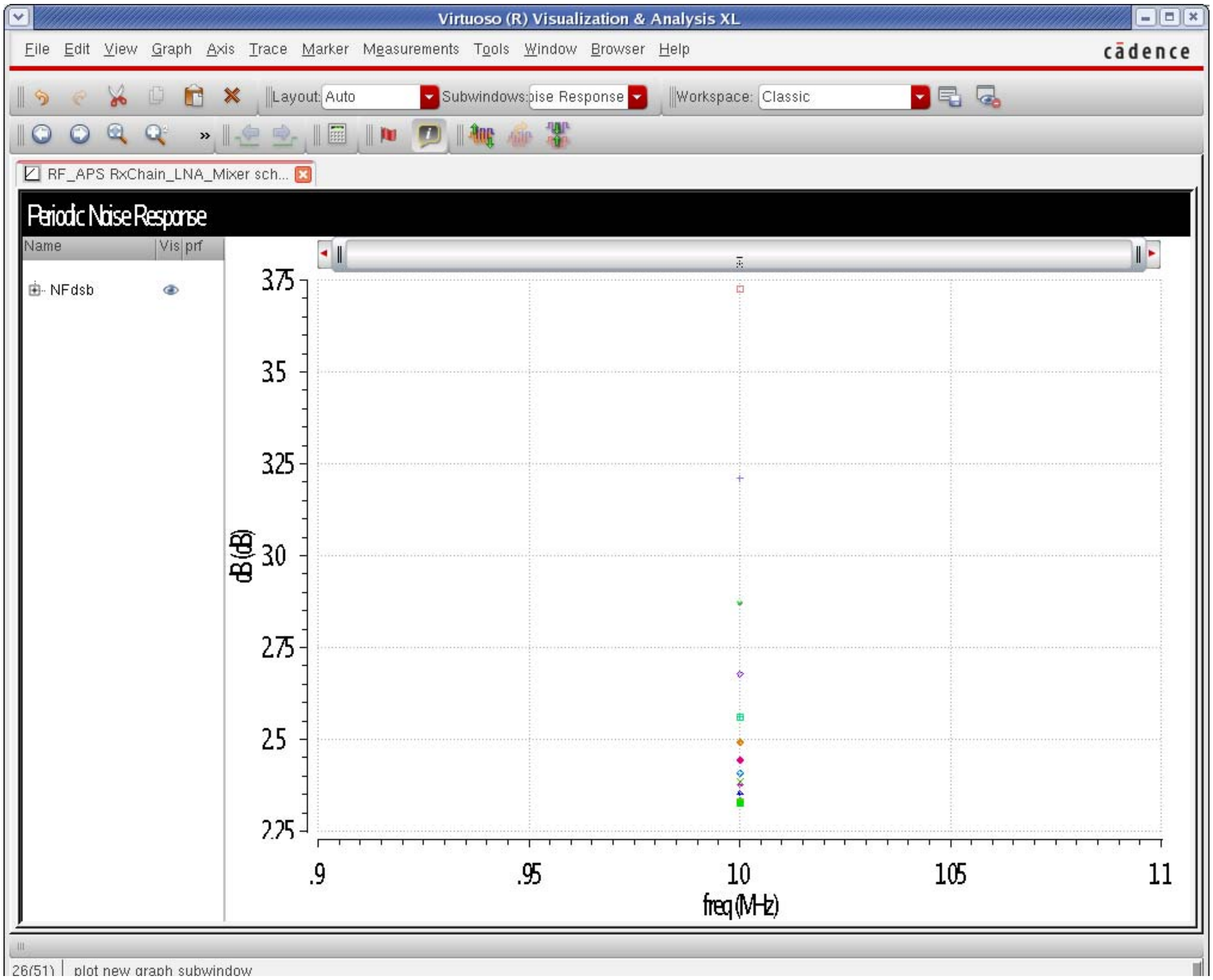
4. Run the simulation when you are done.

5. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE.

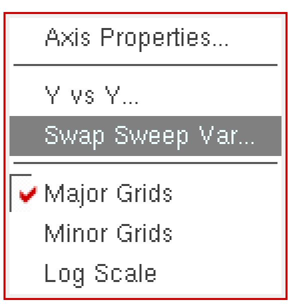


6. Select *pnoise*.
7. Select the desired noise figure measurement.
8. Click *Plot*.

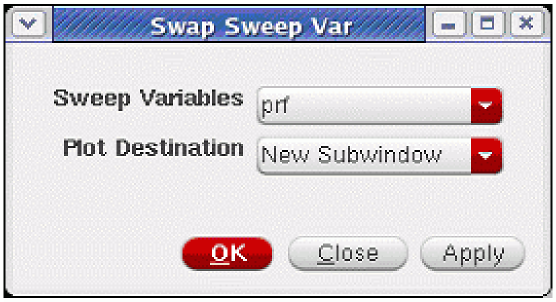
The resulting plot appears.



9. To make the plot more useful, right-click one of the numbers on the X Axis and select *Swap Sweep Var* from the context menu.

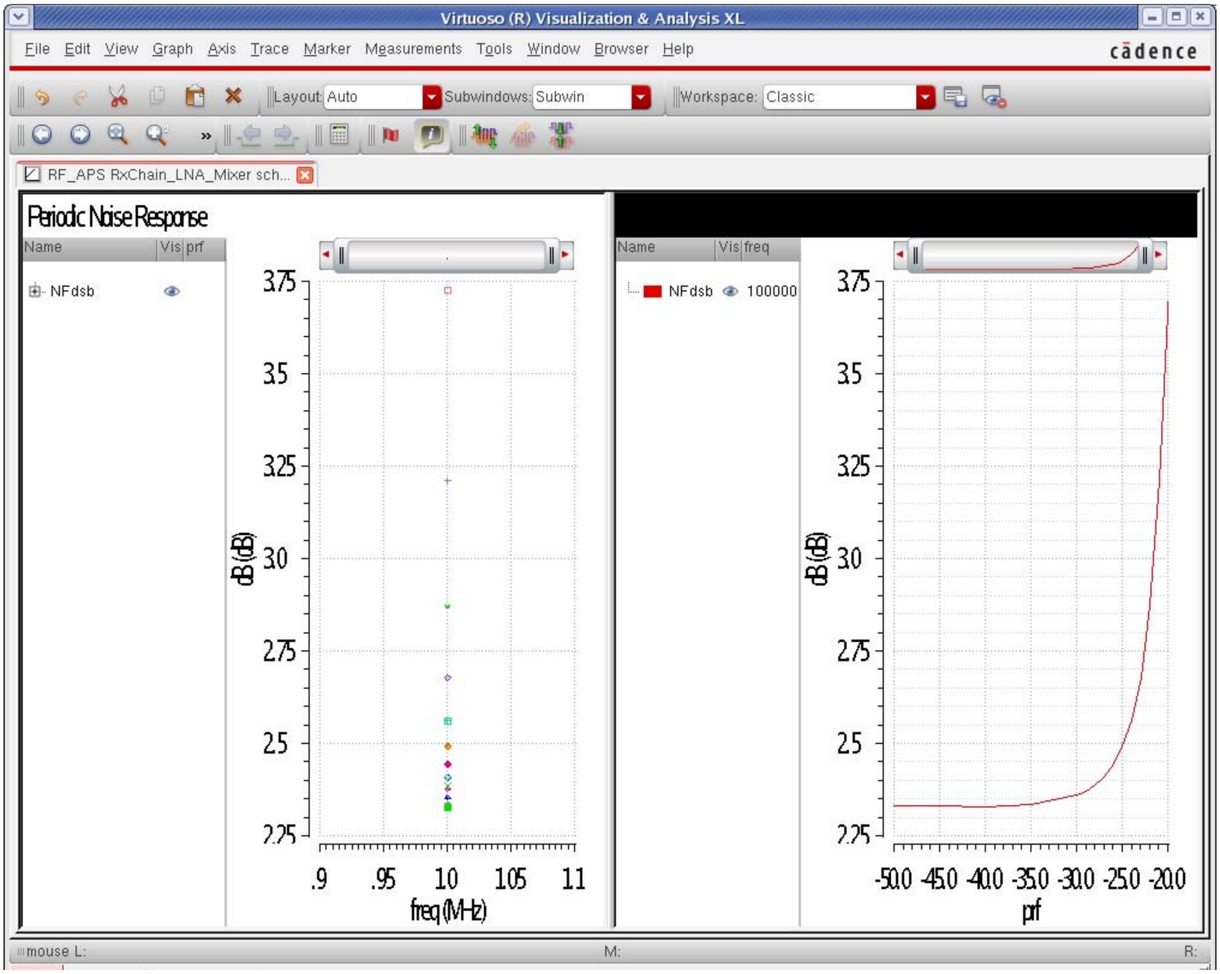


The *Swap Sweep Var* window is displayed. Note that the *Sweep Variables* field is set to *prf*, which is the variable that was swept.



10. Click *OK*.

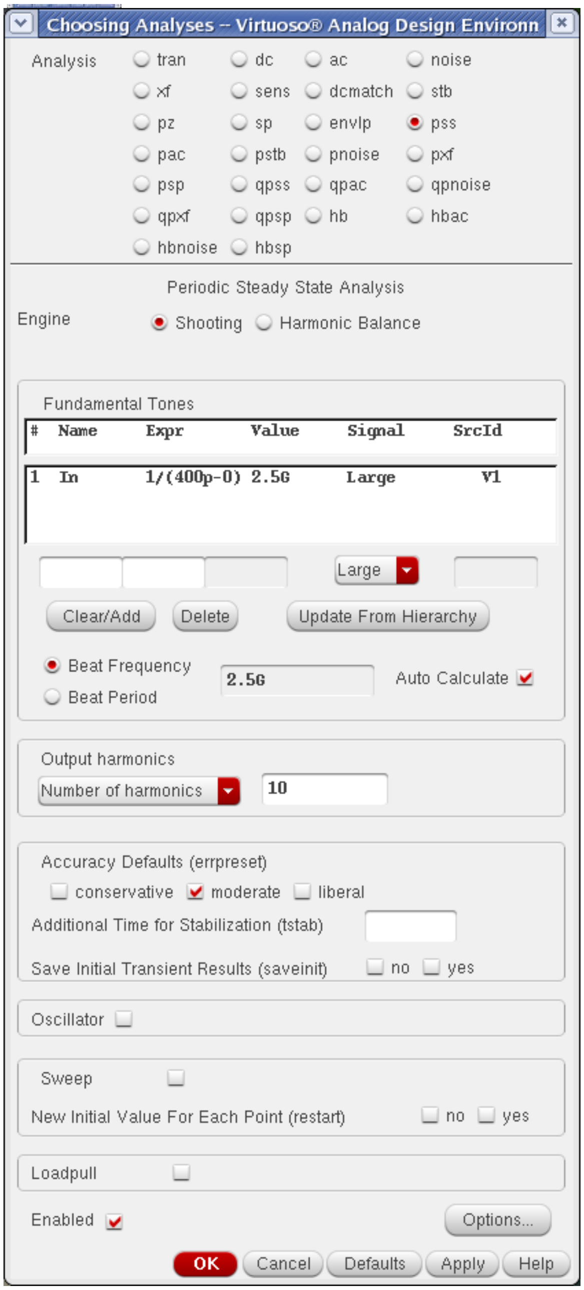
A new subwindow is displayed with the noise figure on the Y axis, and the blocker power on the X Axis.



Noise Type = Modulated

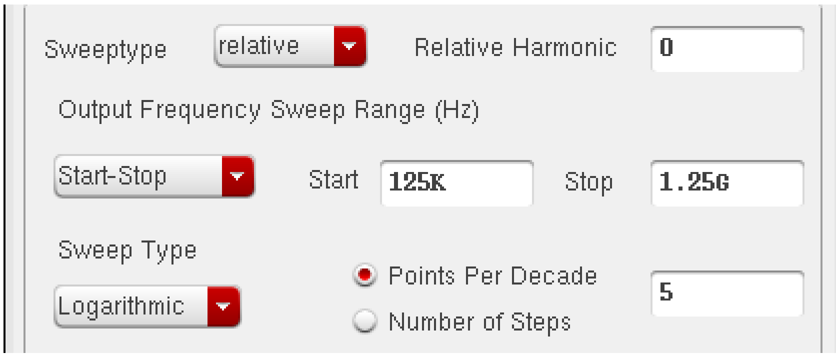
Modulated allows the measurement of the AM and PM components of noise separately. This can be applied to any driven circuit or oscillator. This section is for driven circuits. See the oscillator section if you have an oscillator. Note that this is an averaged measurement over the full cycle of the pss waveform, not an instantaneous measurement at a single threshold voltage.

First, set up the pss analysis as appropriate for your circuit. For more information, refer to the section Overview of Periodic Steady-State (pss) Analysis on page 547.



On the pnoise *Choosing Analyses* form:

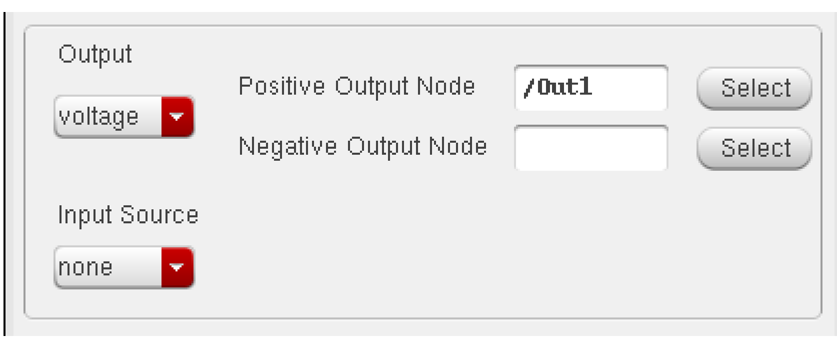
1. Set up an appropriate frequency range of interest.



Shooting: Set maximum sideband up to 4 times the number of pss harmonics. Up to 40 sidebands can be set using the defaults. Fullspectrum is not available for harmonic balance.

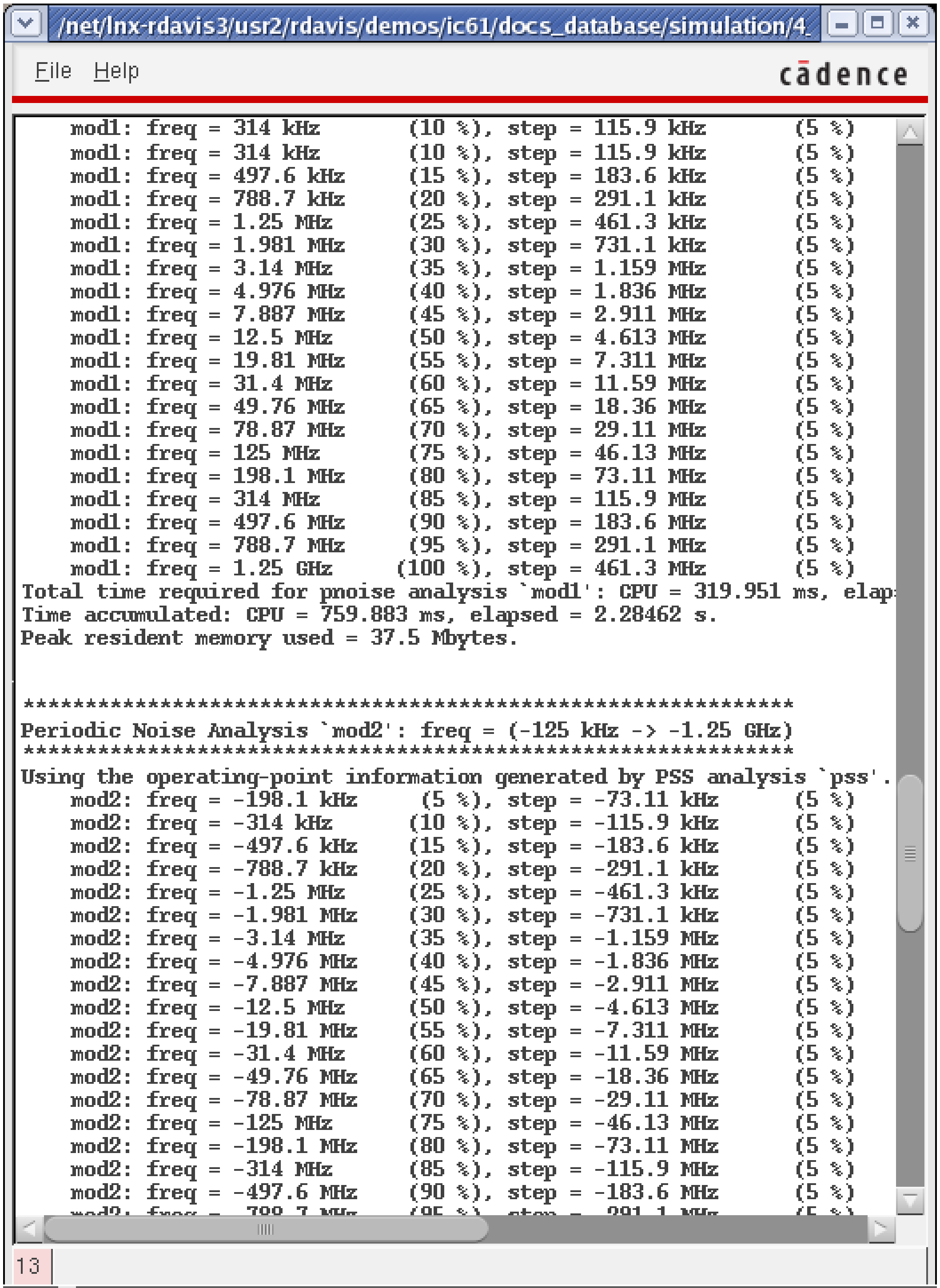
2. **Harmonic balance**: Leave the *Maximum sideband* field blank. This calculates noise translations for all the harmonics in the pss-harmonic balance analysis.

3. Select the output net or nets in the schematic. If the *Negative Output Node* field is left blank, the global ground net is used as the entry.



4. Select *modulated* from the *Noise Type* drop-down list.

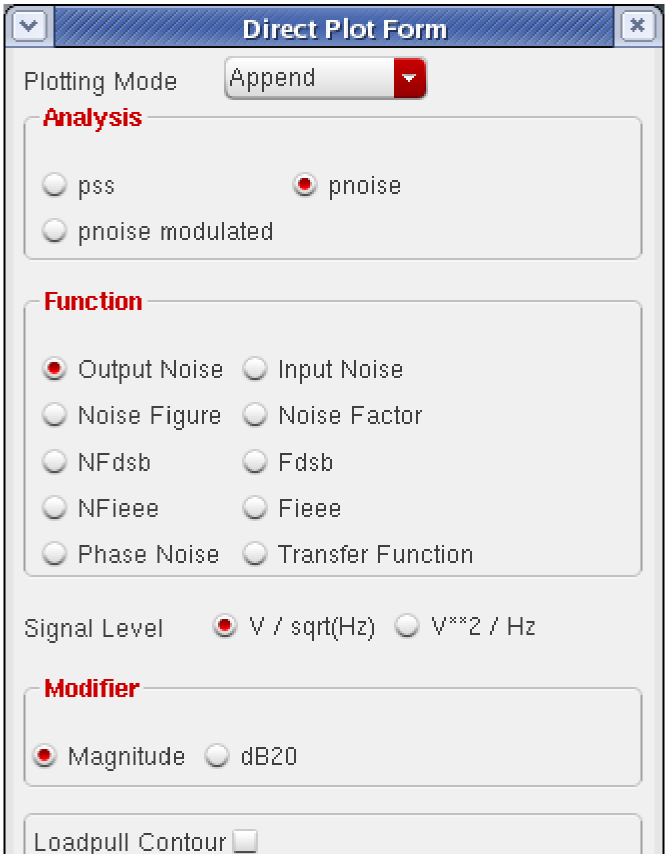
5. Run the simulation. Note that modulated runs both the positive and negative frequencies. These frequencies are above and below the harmonic specified in the pnoise *Choosing Analyses* form.



Now plot the output noise using the *Direct Plot Form*.

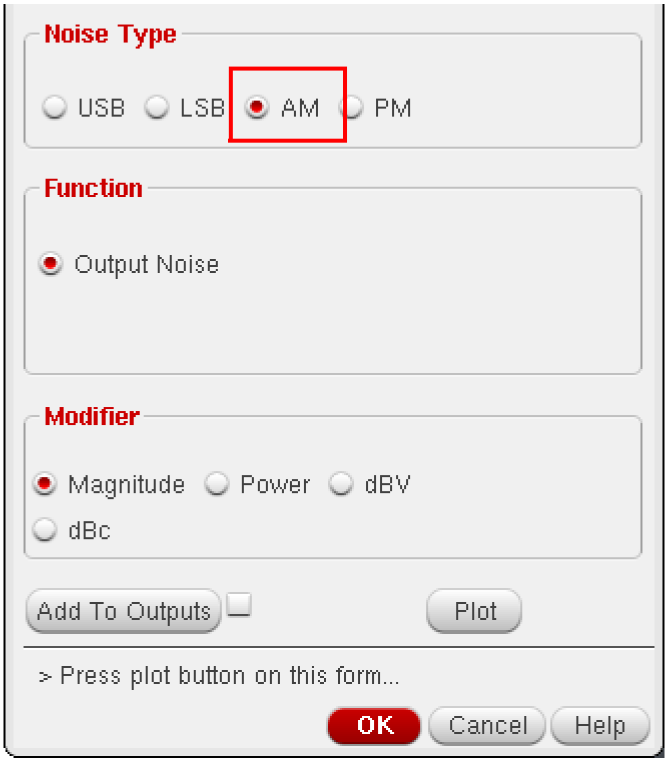
On the *Direct Plot Form*:

1. Select pnoise.
2. Select *Output Noise* from the *Function* section.

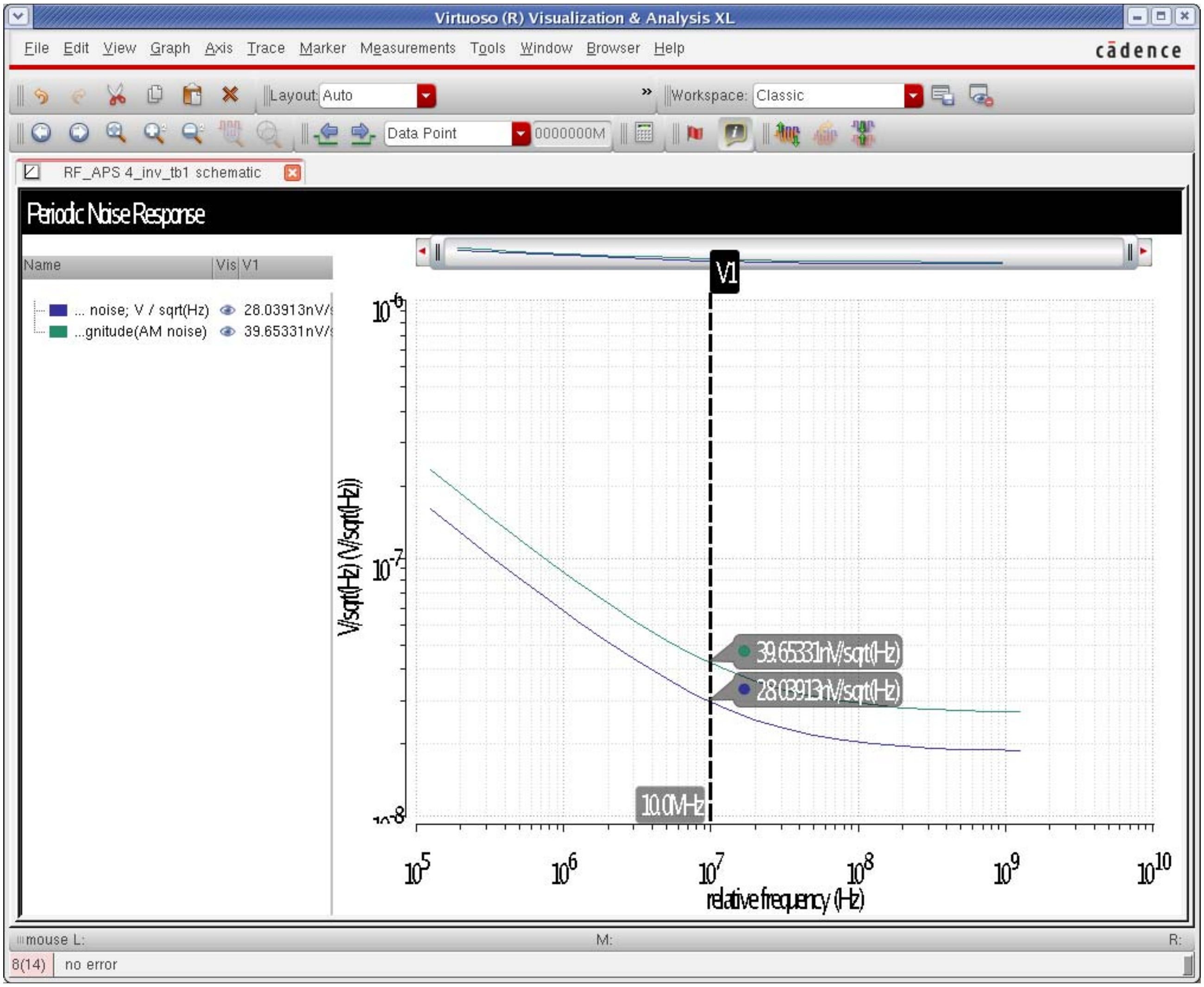


3. Click *Plot*.
4. Select *pnoise modulated*.

5. Select *AM* or *PM* from the *Noise Type* radio buttons.



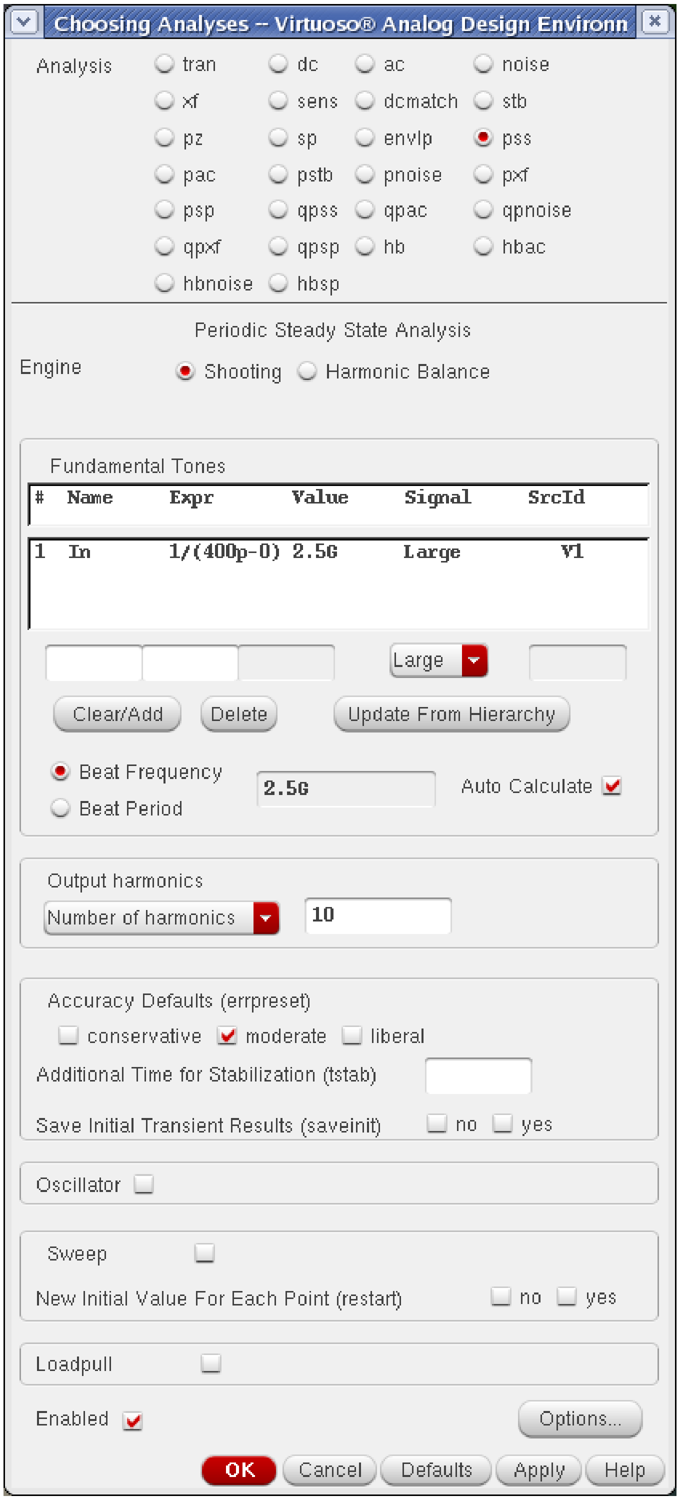
6. Click *Plot*.



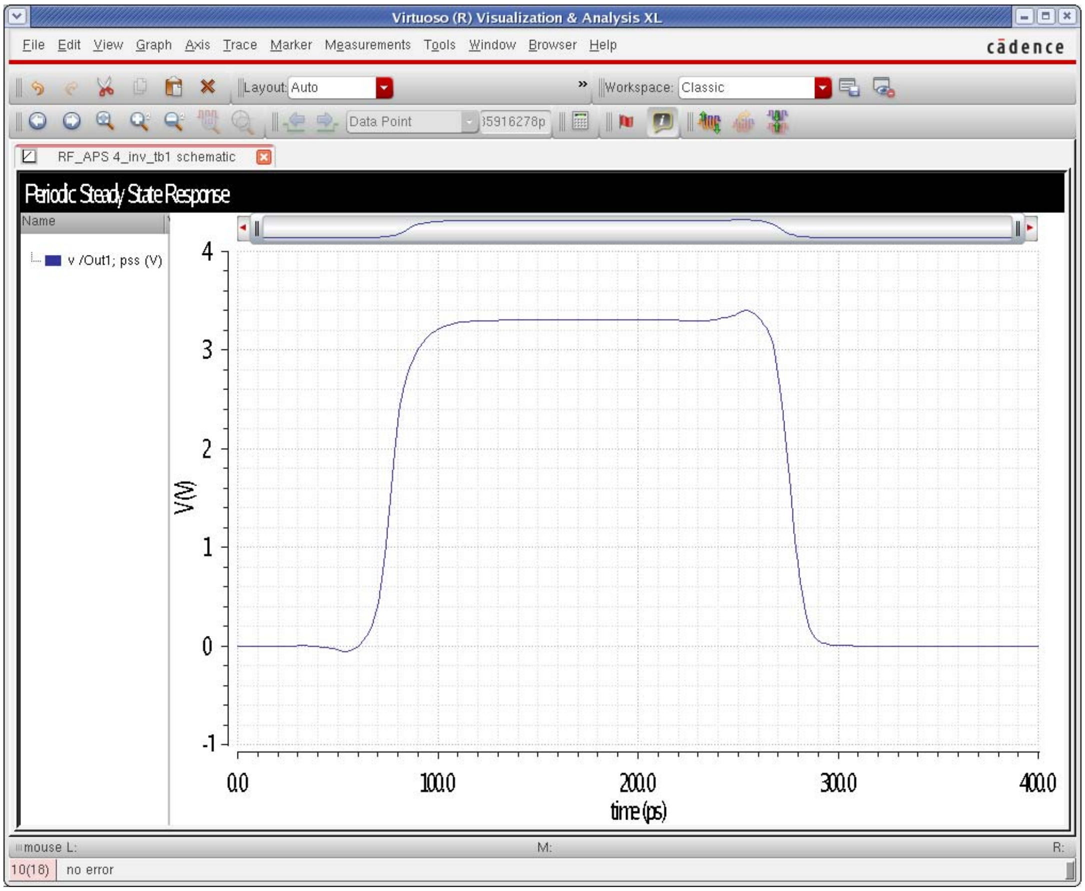
On the plot, note that the AM noise is 3dB larger than the output noise. 3dB is a linear factor of 1.414. This is because the output noise is derived from the positive frequencies only (single sideband), and the AM and PM noise is derived using the positive and negative frequencies (double sideband).

Noise Type = Jitter

Driven jitter is usually used for measuring the noise in a digital circuit at a specific threshold crossing.This is an instantaneous measurement at one or more points in the time-domain waveform at a single threshold voltage. Accordingly, the pss *Choose Analyses* form uses shooting, as shown below. For more information on the pss analysis, please see Overview of Periodic Steady-State (pss) Analysis on page 547. This is also supported for harmonic balance. If harmonic balance is used, make sure you set oversample factor to 4 or 8, and use enough harmonics to capture the waveform accurately.



An appropriate threshold is needed, which can be determined by plotting the time domain waveform from the pss *Direct Plot Form*. The waveform below shows an appropriate threshold value to be 1.65V for this circuit, which is a 3.3 Volt logic circuit.



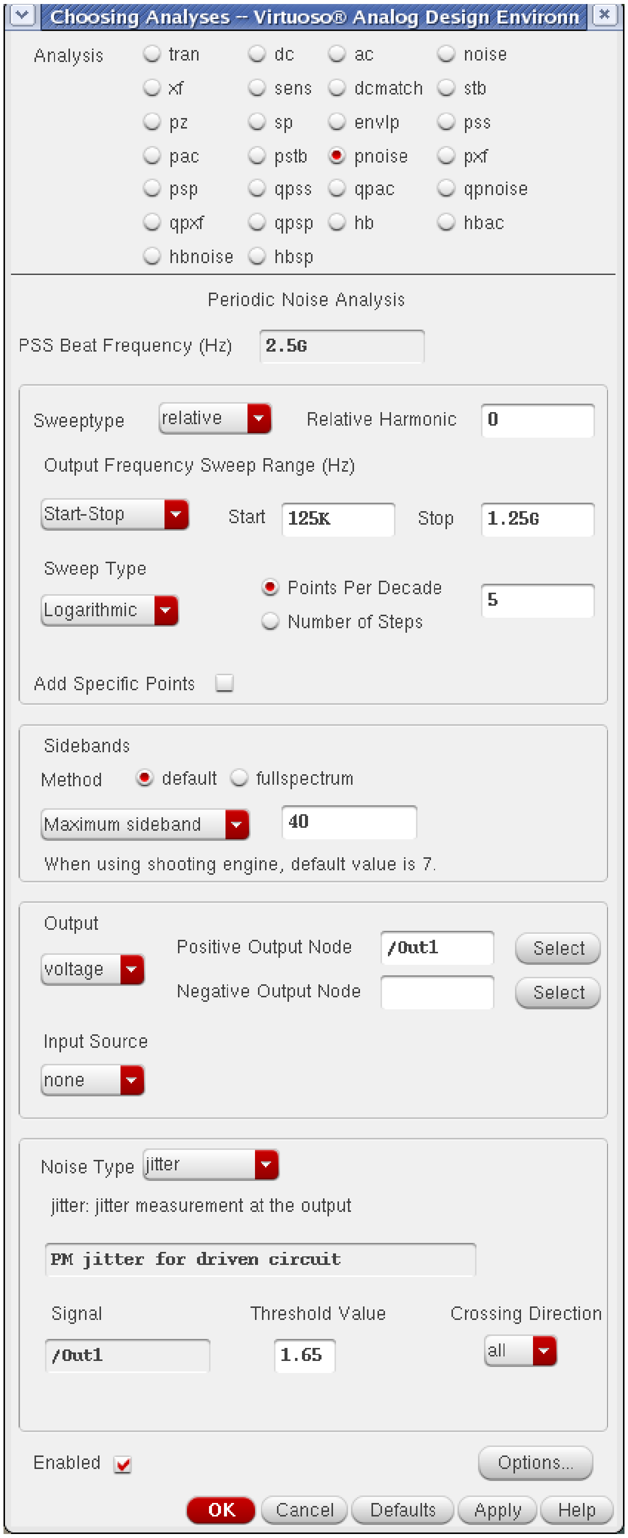
On the pnoise *Choosing Analyses* form:

1. The stop frequency is half the input frequency.
2. The start frequency is 4 decades below that.
3. About 5 points per decade is usually sufficient.
4. *Maximum sideband* should be set to 40 for shooting, and left blank for harmonic balance. Some systems need more than 40 sidebands. To verify that 40 is enough, first run 40 sidebands and plot the noise. Next, double the number of PSS harmonics and the number of pnoise sidebands and run the simulation again. If the results are the same, then 40 sidebands is enough. If the results change, you need more harmonics and sidebands. If *fullspectrum* is chosen, start with 10 harmonics and plot the noise. Then double to 20 harmonics, re-run the simulation, and re-plot the noise. If the noise did not

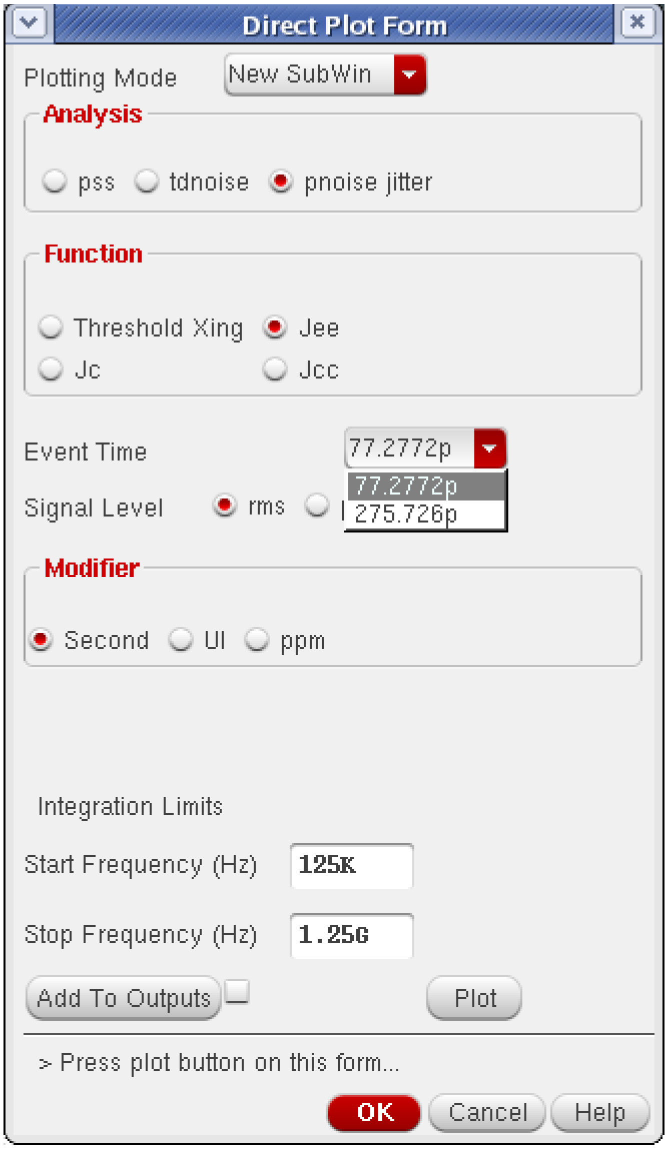
change, go back to 10 harmonics. If it did change, continue to increase the number of pss harmonics until the noise measurement no longer changes.

5. The *Output* field is set to *voltage*, and the *Input Source* field is set to *none* because usually, only the output jitter is needed.
6. The *Noise Type* is set to *Jitter*.

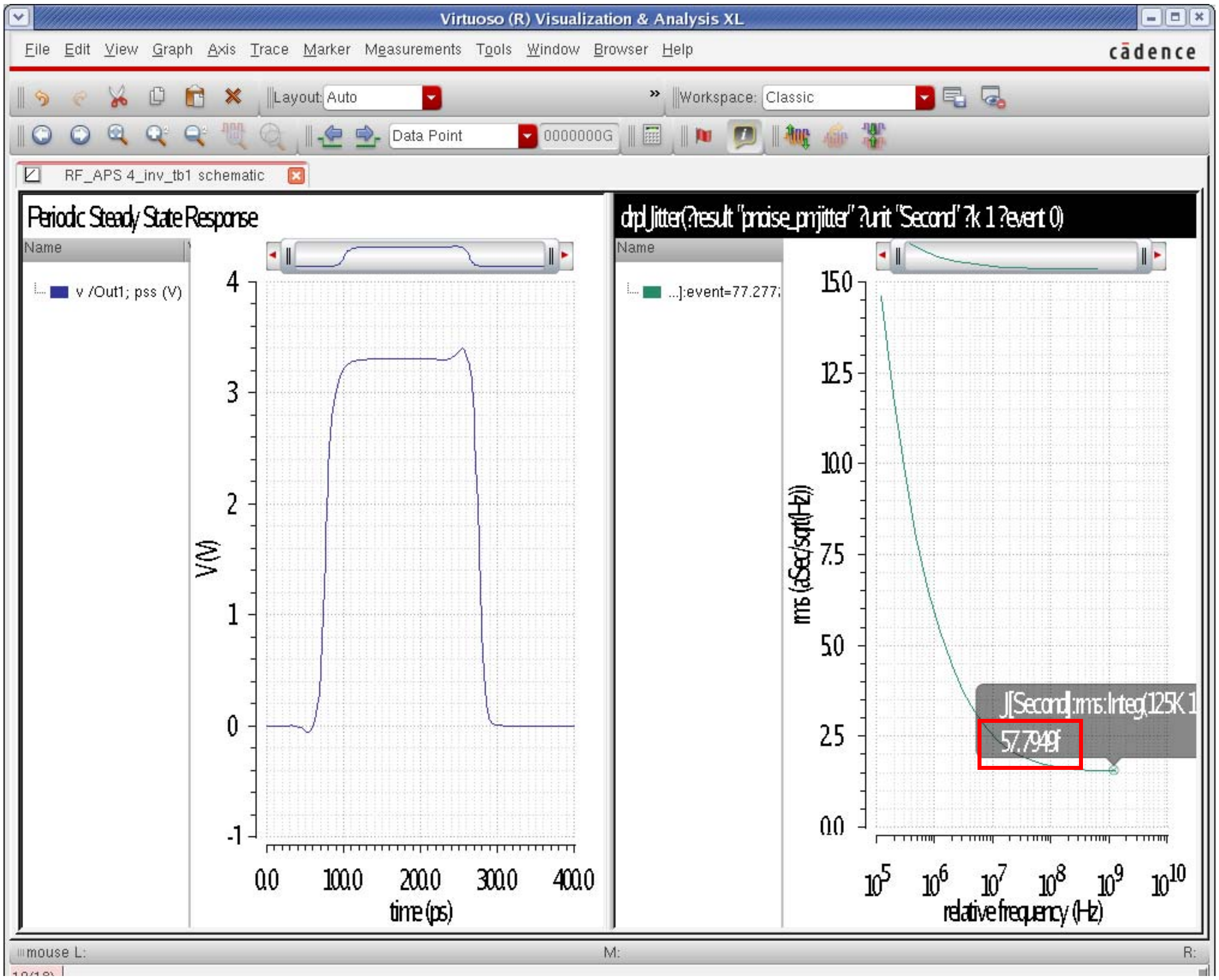
7. The threshold is set appropriately. The *Crossing Direction* option can be set to *all*, *rise* or *fall*. In the example below, it is set to *all*, which allows jitter measurements on the rising and falling edges of the digital output.


Choosing Analyses -- Virtuoso® Analog Design Environn
Analysis
tran
dc
ac
noise
xf
sens
dcmatch
stb
pz
sp
envlp
pss
pac
pstb
pnoise
pxf
psp
qpss
qpac
qpnoise
qpxf
qpsp
hb
hbac
hbnoise
hbsp
Periodic Noise Analysis
PSS Beat Frequency (Hz)
2.5G
Sweeptype
relative
Relative Harmonic
0
Output Frequency Sweep Range (Hz)
Start-Stop
Start
125K
Stop
1.25G
Sweep Type
Logarithmic
Points Per Decade
5
Number of Steps
Add Specific Points
Sidebands
Method
default
fullspectrum
Maximum sideband
40
When using shooting engine, default value is 7.
Output
voltage
Positive Output Node
/Out1
Select
Negative Output Node
Select
Input Source
none
Noise Type
jitter
jitter: jitter measurement at the output
PM jitter for driven circuit
Signal
/Out1
Threshold Value
1.65
Crossing Direction
all
Enabled
Options...
OK
Cancel
Defaults
Apply
Help

Now run the simulation. When the simulation completes, use the *Direct Plot Form* to plot the jitter. For driven circuits, select *Jee*, which is edge-to-edge jitter. The event time is the time in the pss waveform. In the below example, 77p is the rising edge, and 275p is the falling edge.



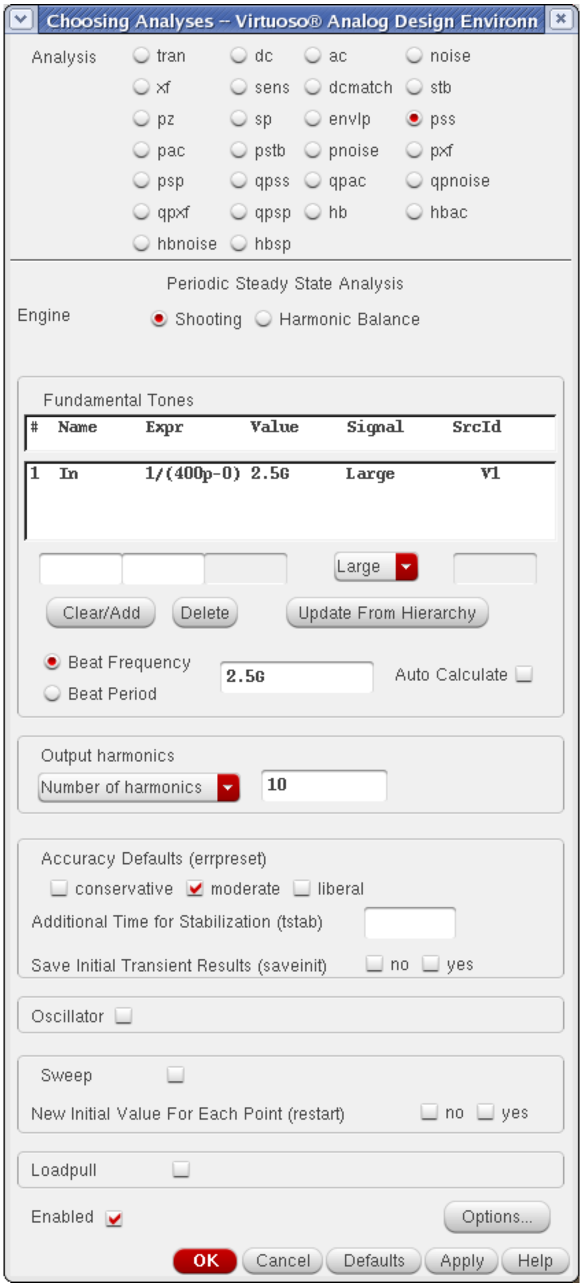
The output noise frequency response and the edge-to-edge jitter are displayed. The jee measurement is one standard deviation for the expected jitter.



Noise Type = Timedomain

This is used to measure the output noise voltage at several timepoints in the pss waveform. The circuit and the pss setup is the same as the setup for jitter in the section just before this one.

First set up and run the pss analysis and plot the time-domain waveform.

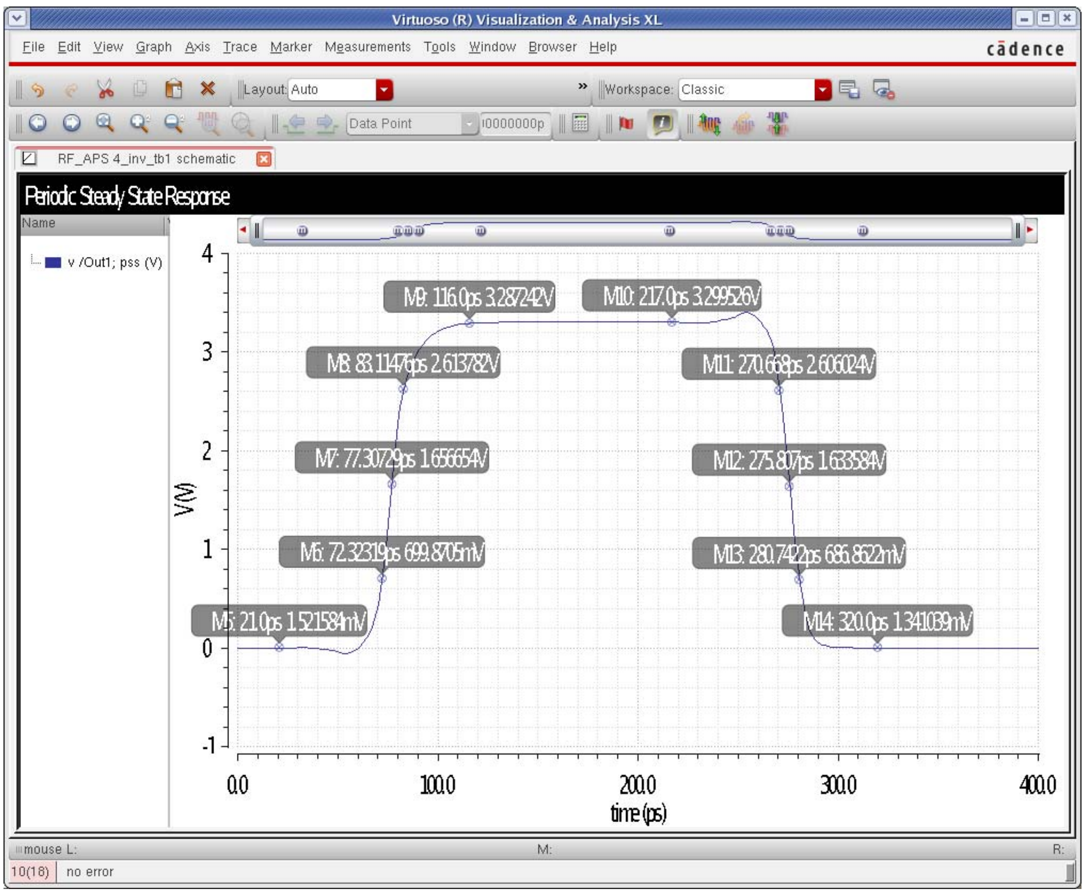


Position the markers at interesting timepoints, as shown below.

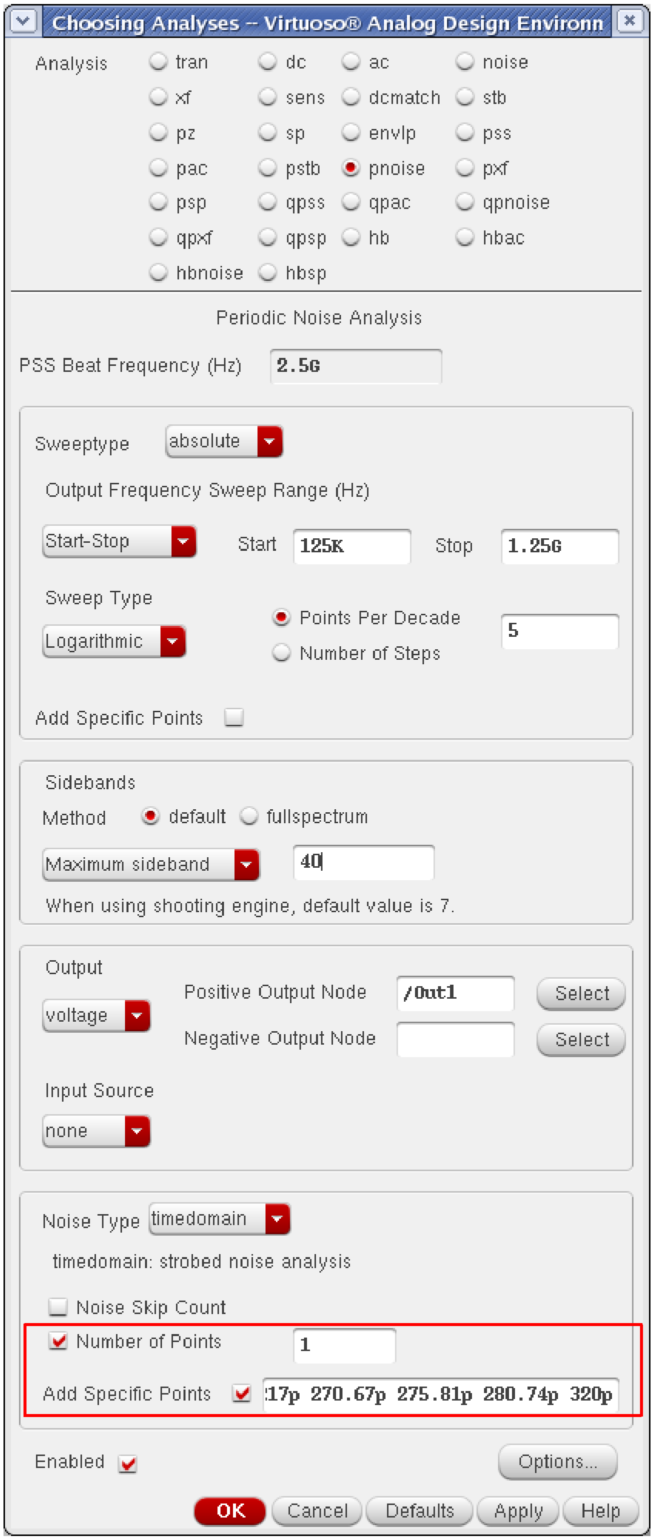
To plot this waveform,

1. In ADE, select *Results - Direct Plot - Main Form*.

2. Select pss.
3. In the *Sweep* section, select *time*.
4. Select the desired net in the schematic.
5. Move your mouse cursor to the point in the waveform where you want a marker, and type `m`.
6. You can move the markers by moving to the dot that highlights the position of the marker on the trace, click and hold the mouse button, move to the desired location, and release the mouse button.



On the pnoise *Choosing Analyses* form:



1. Set the *Sweeptype* to *absolute*.
2. Set the *Stop* frequency to one half the PSS frequency.
3. Set the *Start* frequency 4 decades below the *Stop* frequency.
4. Select log sweep, and set 5 points per decade.
5. For shooting, select *default* or *fullspectrum*. If you select *default*, set the number of sidebands to 40.

Some systems require more frequency translations to be used. To check for this if you used fullspectrum, double the number of pss harmonics to 20, and run the simulation again. If the noise measurement did not change, you can go back to 10 pss harmonics. If it did change, double the number of harmonics, and rerun the simulation. Use the smallest number of harmonics that produce a stable result.

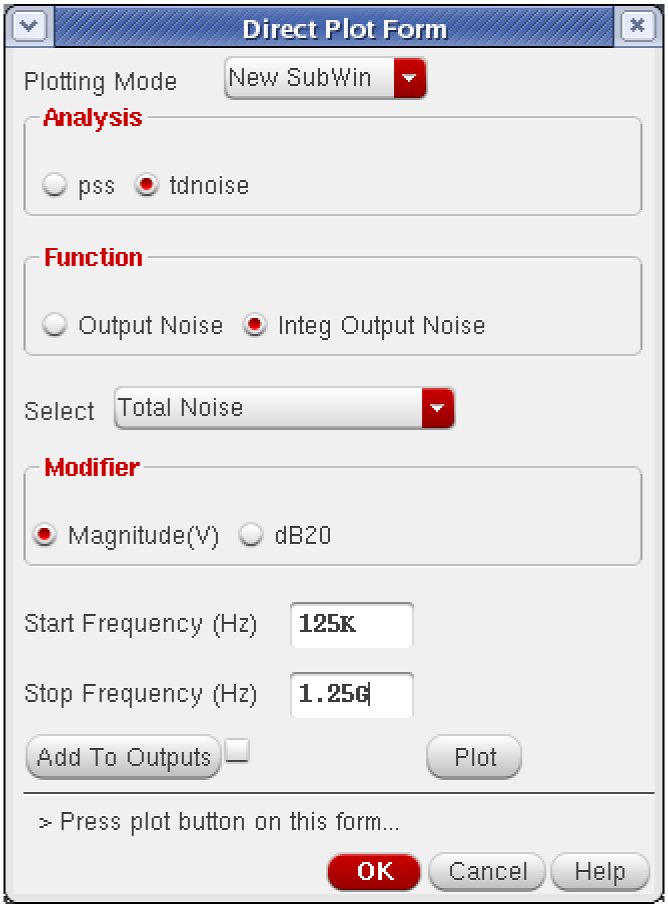
When you are using the default method, double the number of sidebands to 80, and set pss harmonics to 20. Rerun the simulation, and use the above method, except both the number of sidebands and the pss harmonics need to be changed for each run.

For harmonic balance, leave the number of pnoise sidebands blank. Start with an estimate of the number of harmonics you need based on the nonlinearity of the system and the input waveform. For square wave input, you should also set oversample factor to 2 or 4. Run the simulation. and plot the result. Now double the number of pss harmonics and run the simulation again. If the noise changed, double the number of harmonics again. Use the smallest number of harmonics that produce a stable noise measurement.

1. Select the *Number of Points* check box and type 1 in the text field. The default in timedomain noise is to run a pnoise analysis at every timepoint in the pss analysis. Because noise is usually only required at several timepoints, setting the number of points to 1 causes a single point at the first timepoint in the pss analysis to be run in pnoise.
2. Select the *Add Specific Points* check box and enter the time values from the markers positioned in the waveform. Use the space character to separate the entries.
3. Run the simulation.

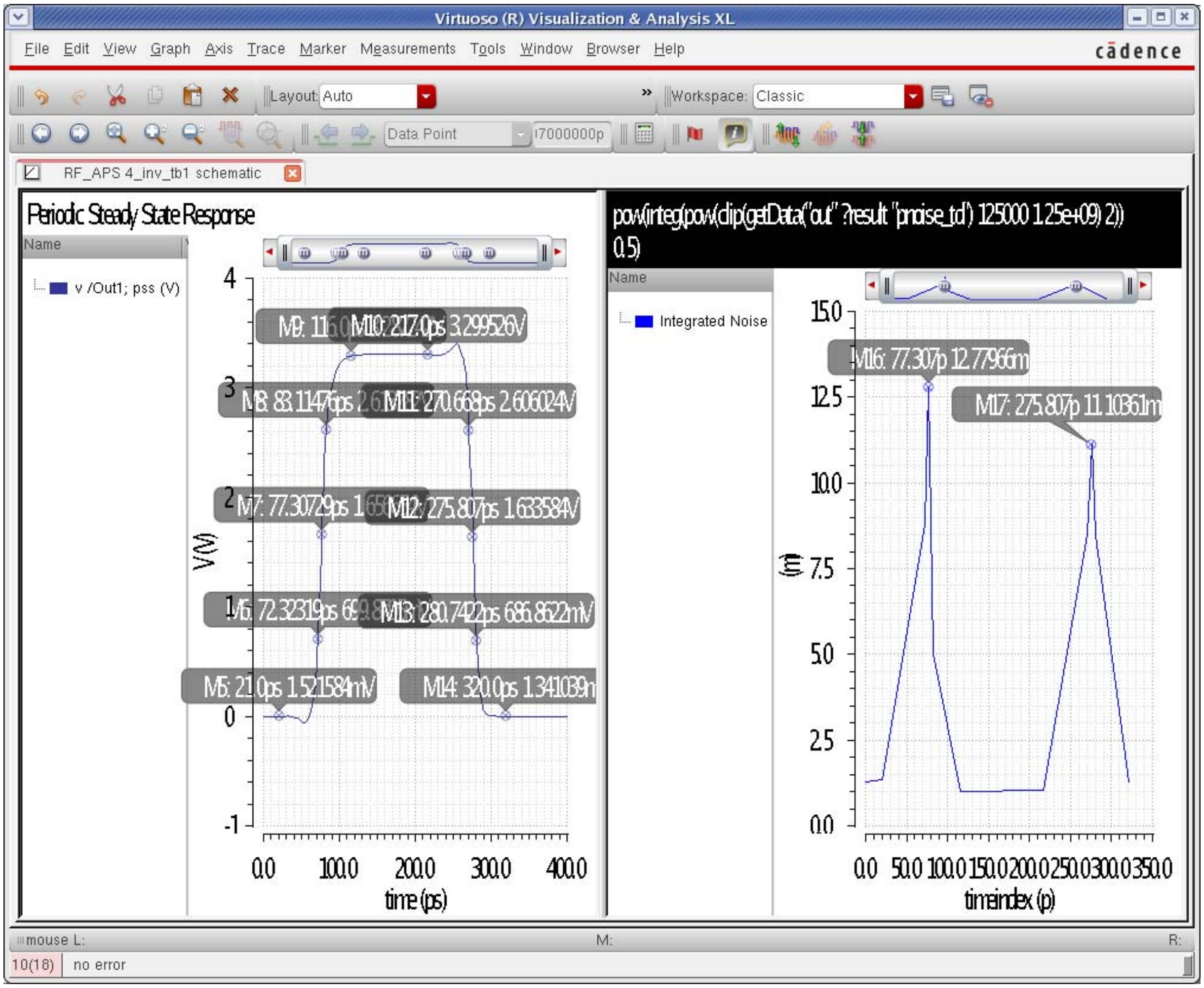
On the Direct Plot Form:

1. Select *tdnoise* from the *Analysis* section.

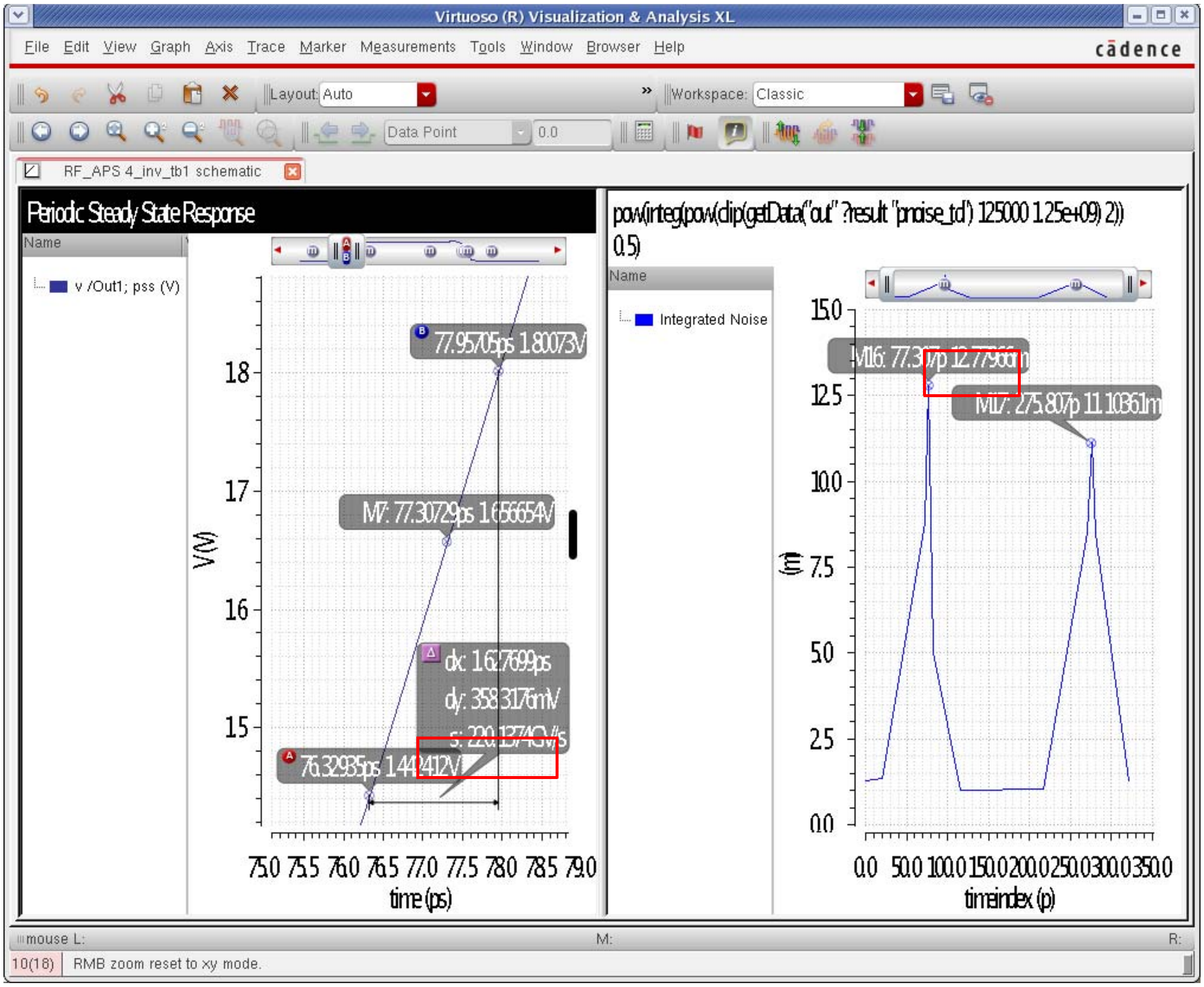


2. Select *Integ Output Noise* from the *Function* section.
3. Select *Add to Outputs*.
4. Click *Plot*.

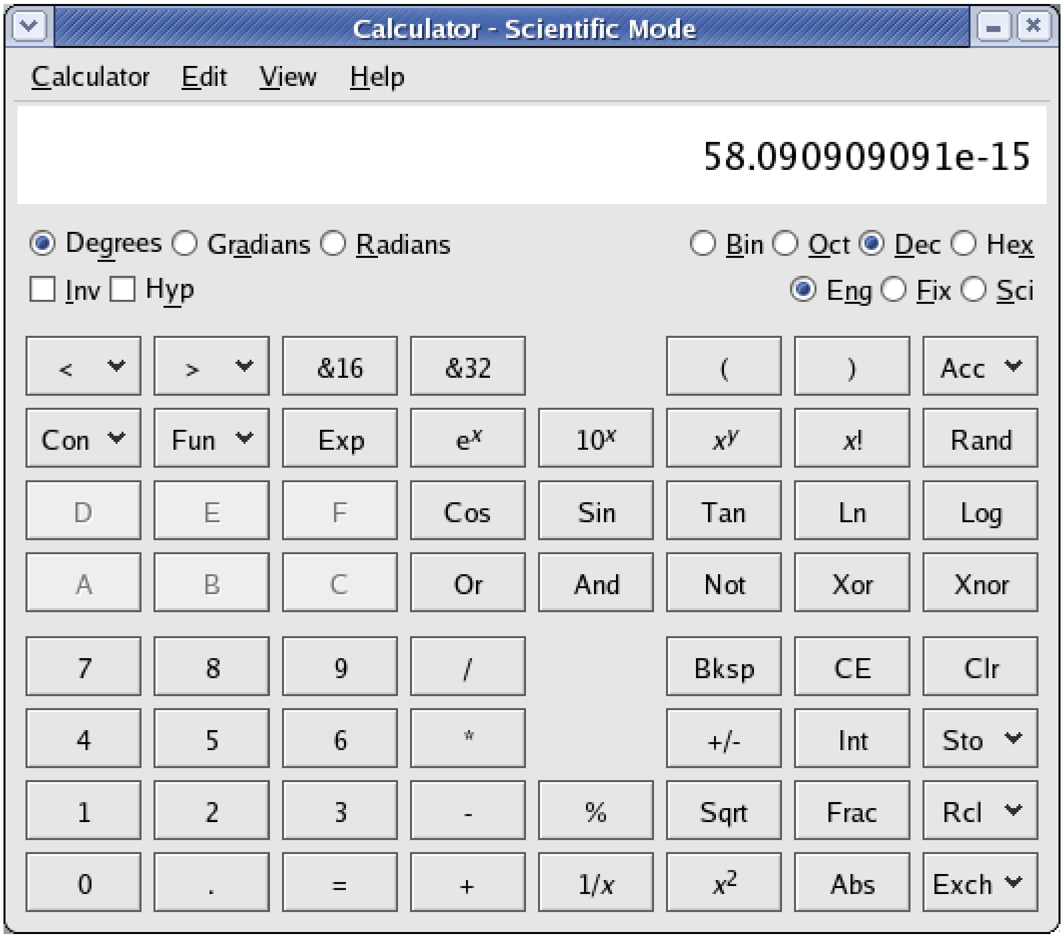
The noise voltage is plotted as shown below. As expected, the maximum noise occurs near the 1.65 volt level of the waveform. This is shown in the right subwindow below.



To get the jitter in seconds, take the integrated noise voltage and divide by the slew rate. To determine the slew rate, place marker A and marker B near the threshold crossing at 77.3 picoseconds. The slew rate is calculated for you in the display. The slew rate is shown in the left subwindow below, and the integrated noise is taken from the subwindow on the right.



The result is shown in the calculator below.



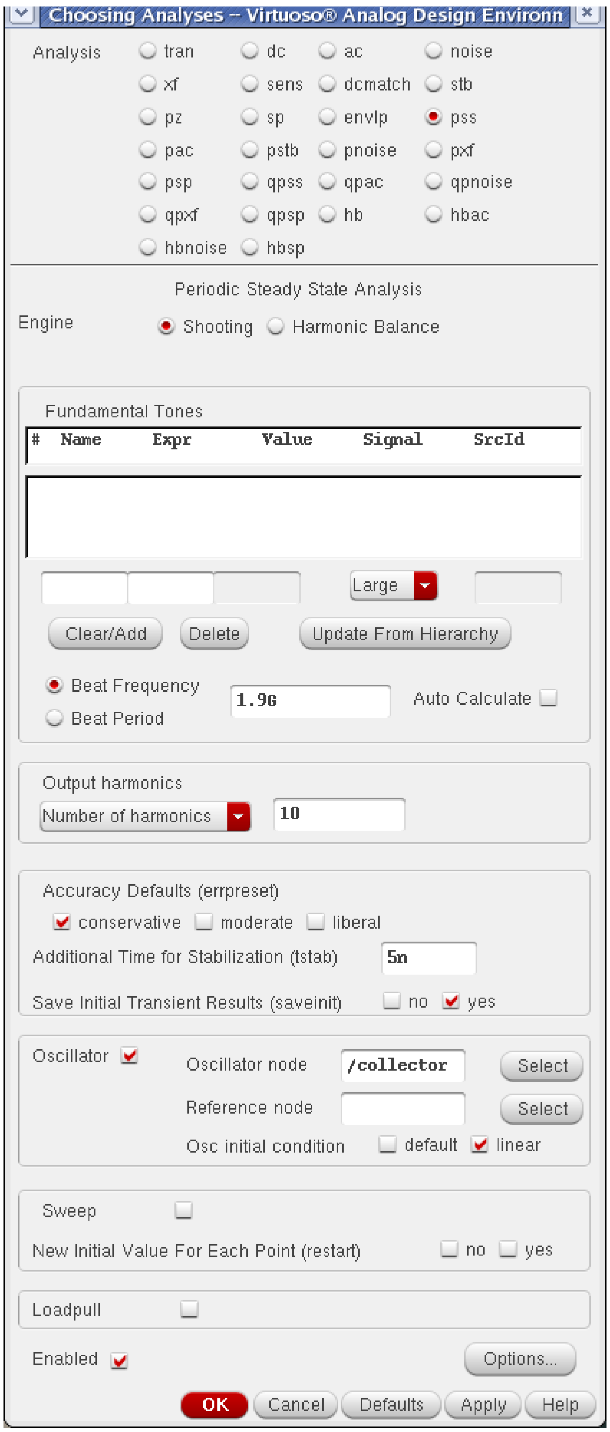
This compares well with the Jee calculation of 57.79 femtoseconds from the section just before this one. Note that this is one standard deviation for the jitter.

Oscillators

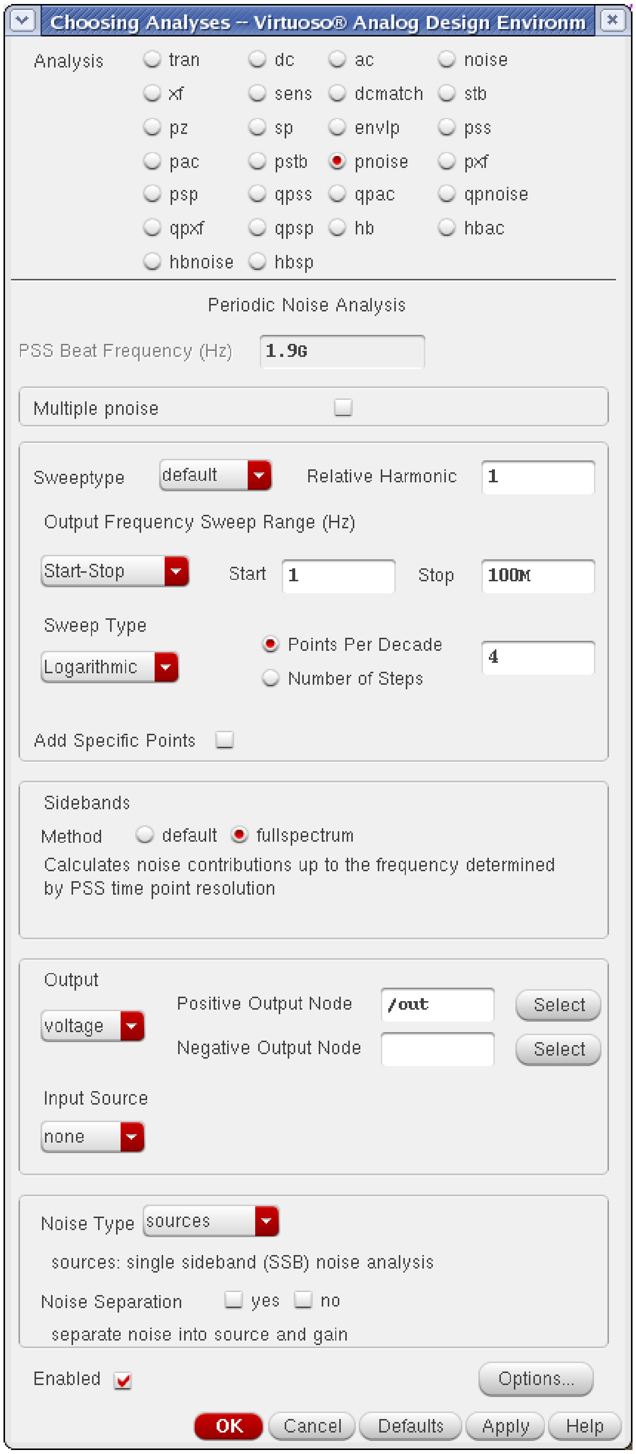
Noise Type = Sources

Noise type=sources is used to measure the single sideband phase noise that is averaged over the entire cycle of the oscillator waveform.

1. First, set up the pss form as shown in the following figure. For more information see the pss section at the beginning of this chapter of the user guide.



2. Next, set up the pnoise form as shown below. For the oscillator, you may need to change the start and stop frequencies and the output node. Select *Sources* from the *Noise Type* drop-down list to measure the average noise of the oscillator.



The sweep defaults to relative to the frequency of the specified harmonic of the pss analysis. The frequencies specified in the *Choosing Analyses* form are added to the frequency of the specified harmonic when *relative* is selected. In this case, the first harmonic is specified because there is no frequency multiplier on the output of the oscillator.

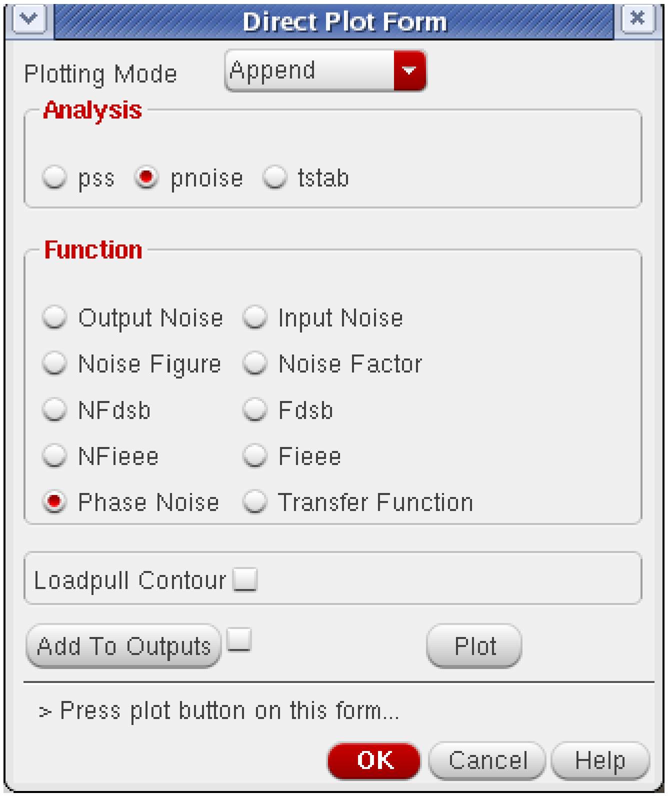
1. Set the relative harmonic number. For an oscillator by itself, the relative harmonic is one. For an oscillator with a prescaler and you desire the output of the prescaler, the harmonic number is one. If you want the noise of the VCO without the prescaler, then the harmonic number is the divide ratio. If you have an oscillator with a frequency multiplier and you want the noise at the output of the frequency multiplier, enter the frequency multiplication ratio. If you want the noise at the output of the VCO by itself, enter one.
2. Relative to the output frequency, specify a frequency offset.
3. Generally, three to five points per decade in a log sweep are sufficient.
4. Setting sidebands depends on whether shooting or harmonic balance has been chosen as the pss engine.
   a. When harmonic balance is selected, leave the maximum sideband field blank, which causes the noise translations for all the pss harmonics to be calculated. Fullspectrum is not available for harmonic balance. Experiment with the number of harmonics and oversamplefactor to find the most efficient pss-pnoise runtime. Overview of Periodic Steady-State (pss) Analysis on page 547 for more information on setting harmonics.
   b. When shooting is the pss engine, fullspectrum is available and recommended. Fullspectrum calculates the number of noise translations to include based on the maximum timestep in the pss analysis. By default, frequency translations through the 20th harmonic are taken into account by fullspectrum pnoise. In the unusual case where the noise calculation needs to go through a very high harmonic of the pss analysis, increase the number of harmonics in the pss form to the minimum number of harmonics that causes the noise result to not increase with a larger number of pss harmonics. Generally speaking, these will be switched-capacitor circuits or sampling circuits where aliasing can go through a very high harmonic of the clock. Fullspectrum on this type of circuit will run much faster than the default pnoise simulation. The larger the number of sidebands, the larger the speed differential.
5. Note that when *fullspectrum* pnoise is chosen, APS will be enabled in ADE. Full-spectrum will only be run in APS. It is not available in standard Spectre. APS requires

additional tokens or different licensing to run compared to standard Spectre. This will be shown with a pop-up window when the ADE environment is used.

6. Set *Output* to *voltage*.
7. Select the output node.
8. Set *Input Sources* to *none*. Input-referred noise is not defined for an oscillator.
9. If you want noise separation, you must first select *sources* from the *Noise Type* drop-down list and then you can select the noise separation. Noise separation allows plotting of where specifically the noise is coming from.
10. Run the simulation.
11. In the ADE, select *Results - Direct Plot - Main Form*.

On the *Direct Plot Form*:

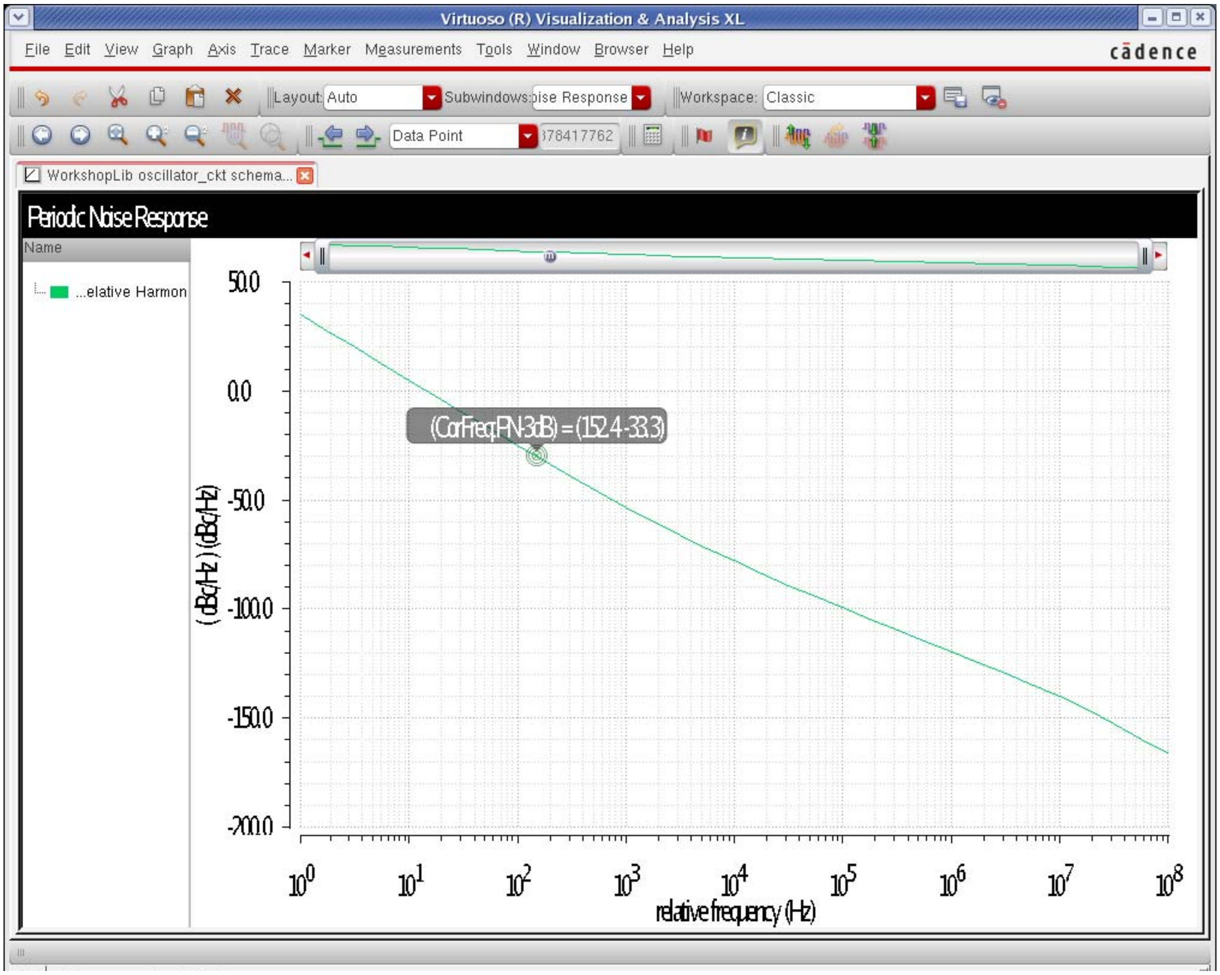
1. Select *pnoise* from the *Analysis* section.



2. Select *Phase Noise* from the *Function* section.

   Note that the phase noise from the pnoise result set plots the noise that is in the frequencies above the oscillator frequency only. If the double sideband phase noise, (which includes the noise from below and above the oscillator output frequency) is desired, set the *Noise Type* to *jitter*. This is documented later in this chapter.

3. Click *Plot*.



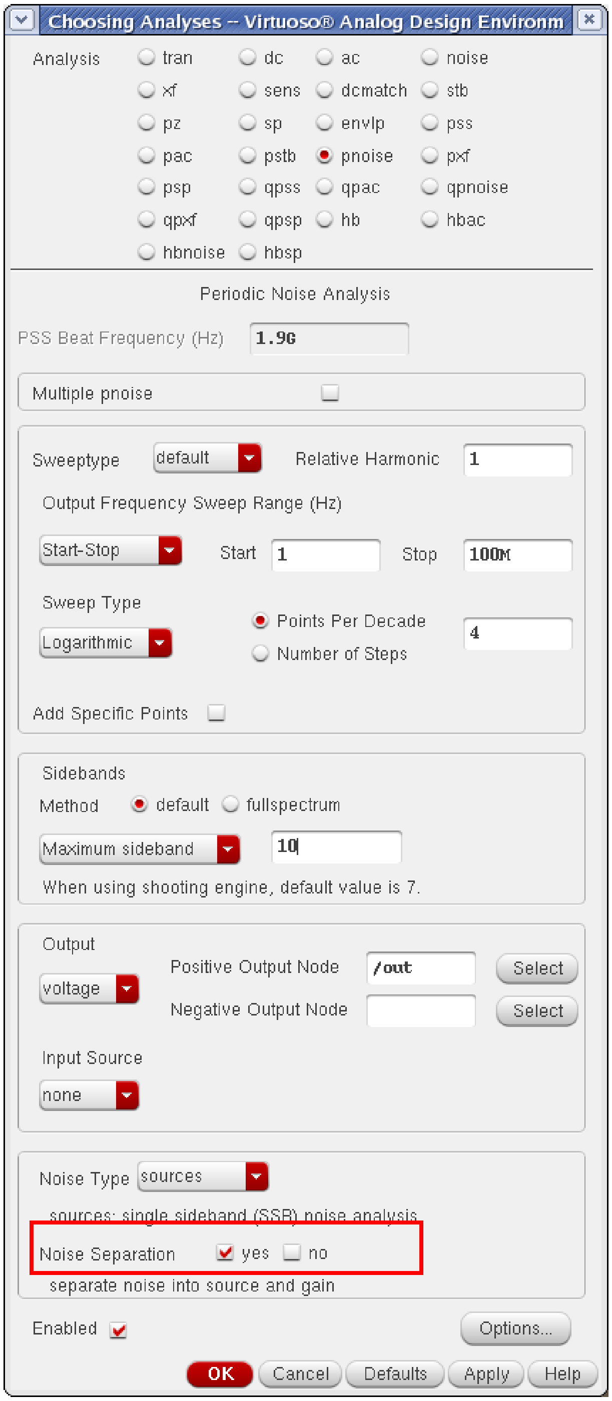
Note that the phase noise continues above 0dBc/Hz, which indicates that the phase noise is larger than the amplitude of the oscillator output. This is because pnoise is a small-signal analysis and it does not recognize the large-signal limits. The *CorFreq* label gives the frequency where the phase noise curve levels off at low frequency offset.

Noise Separation

One of the benefits of simulation is that since the noise terms are all calculated individually, we can store the data in a way to make these individual calculations available for viewing. To see the different noise separations, select *yes* for *Noise Separation* in the pnoise *Choosing Analyses* form.

The default in pnoise is to calculate all the noise translations that are specified in the *Maximum sideband* property in the *Choosing Analyses* form, add all these contributions at the output, and display that total noise term. Enabling noise separation allows you to see which components and/or which frequencies contribute the most noise at the output. Once you know where the noise is coming from, you can design a solution to the problem.

Noise separation is available for shooting and harmonic balance, but not for full spectrum shooting pnoise.

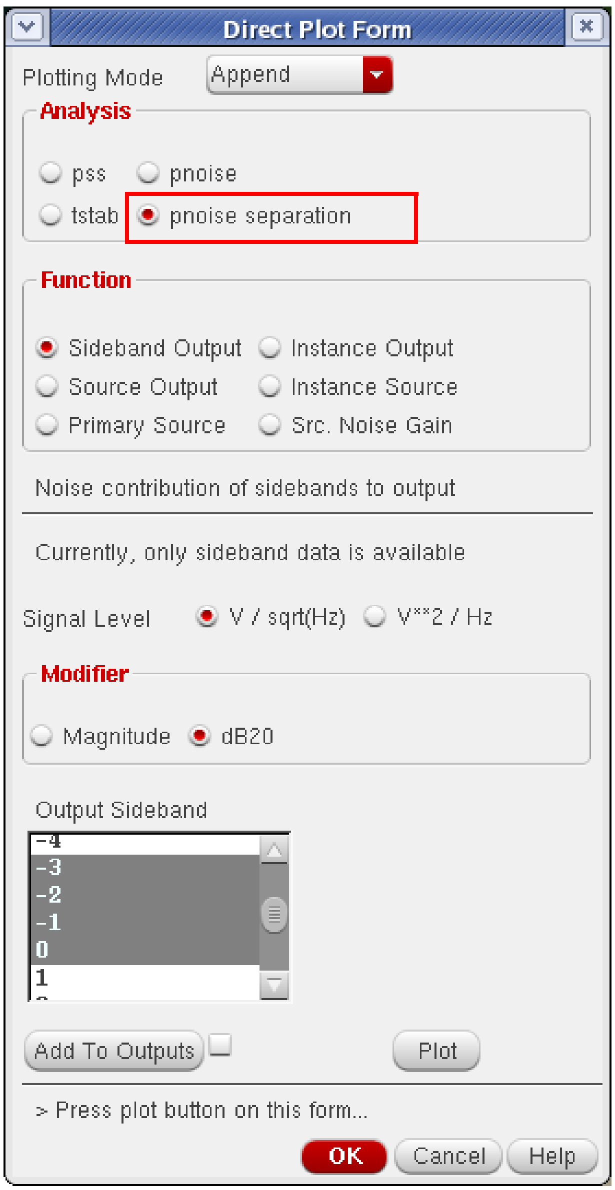


When the choice is made and the simulation is run, select *Results - Direct Plot - Main Form* in ADE.

Viewing the total noise at the output from each sideband

This allows the measurement of the noise at the output from the individual noise input frequencies in the circuit.

When noise separation is set to *yes* in the pnoise *Choosing Analyses* form, you can also plot subsets of the total noise in order to identify where the noise is coming from. In the *Direct Plot Form*, select the *pnoise separation* result set.



To view the total noise from mixing with different pss harmonics:

1. Select *Sideband Output*.

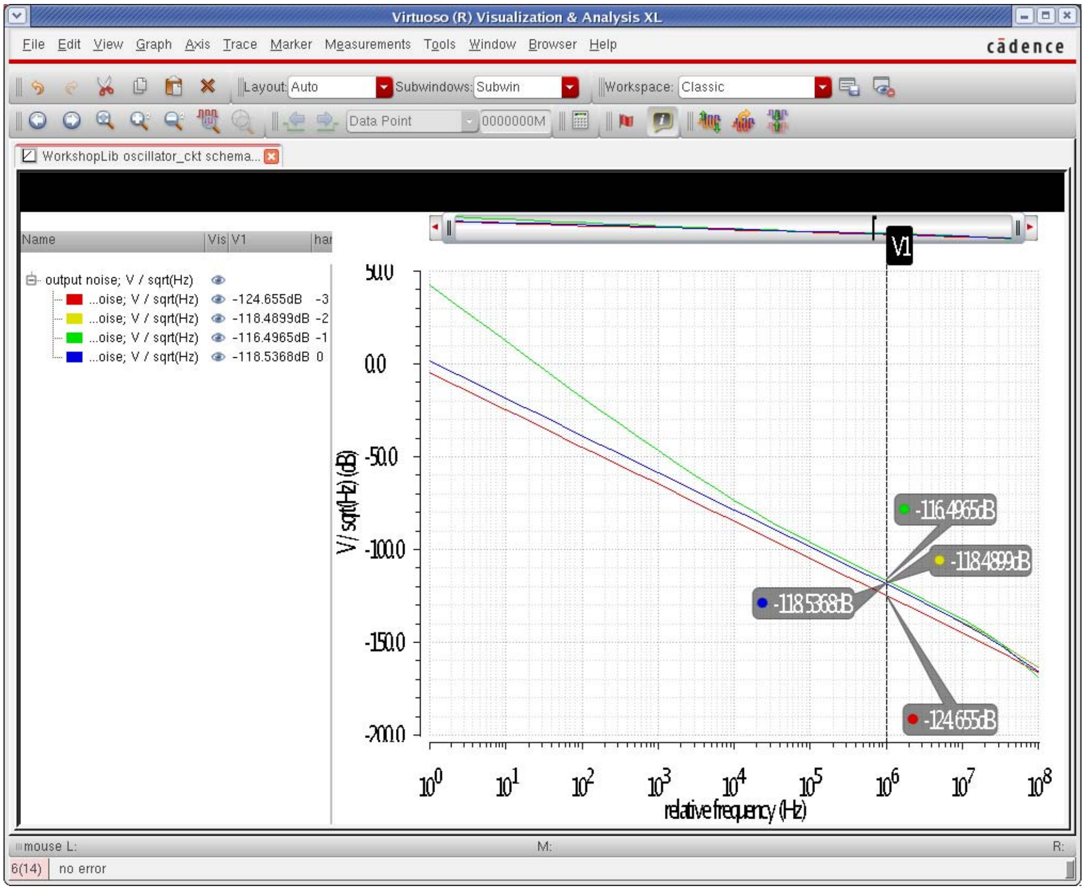
   Remember that the zero sideband means that the noise input is the same frequency as the noise output. Selecting the -1 sideband means that the noise that is below the output frequency that mixes with the first harmonic is plotted. -2 means that the noise frequency is below the noise output frequency, and it mixes with the second harmonic of the pss analysis. +1 means that the noise frequency is above the noise output frequency, and mixes with the first harmonic of the pss analysis.

   Example:

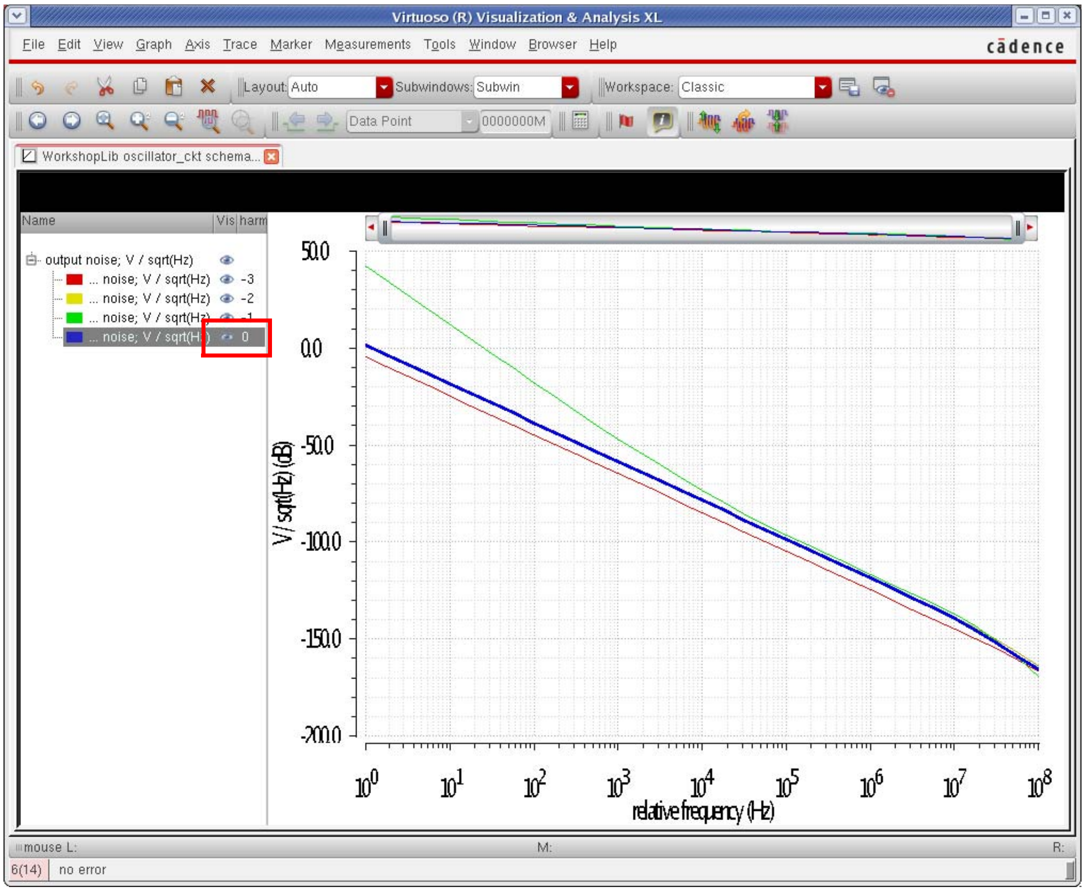
   In this case, the PSS frequency is 1.9G. In the pnoise form, *relative* is selected, and the relative harmonic number is 1. This means that the output noise just above the first harmonic frequency is being analyzed. When the noise output frequency is 1MHz, the zero sideband is 1MHz above the first harmonic of the pss analysis. The -1 sideband is (-1 * 1.9GHz) + 1MHz + 1.9GHz), or 1MHz. The +1 sideband is at (+1 * 1.9G) + 1MHz + 1.9GHz, or 3.801GHz.

2. Select *V / sqrt(Hz)* or *V** 2 / Hz*.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select one or more sidebands from the *Direct Plot Form*.

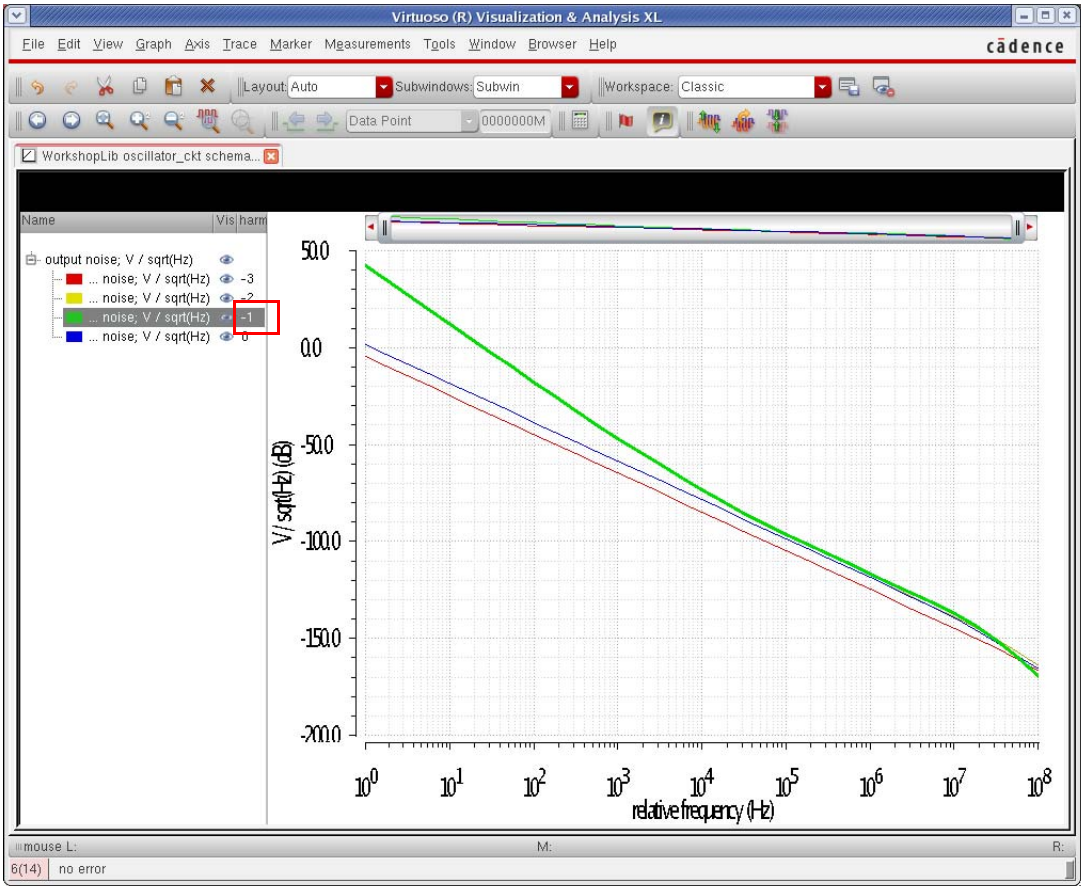
5. Click *Plot*. The results display in the waveform tool. A vertical marker has been placed so the individual results at 1MHz can be read out.



6. To see the individual contributions, select the legend at the upper left for the sideband number you want to see. To see the noise at the output with no frequency translation, select the zero sideband as shown below. The legend and the curve will highlight.

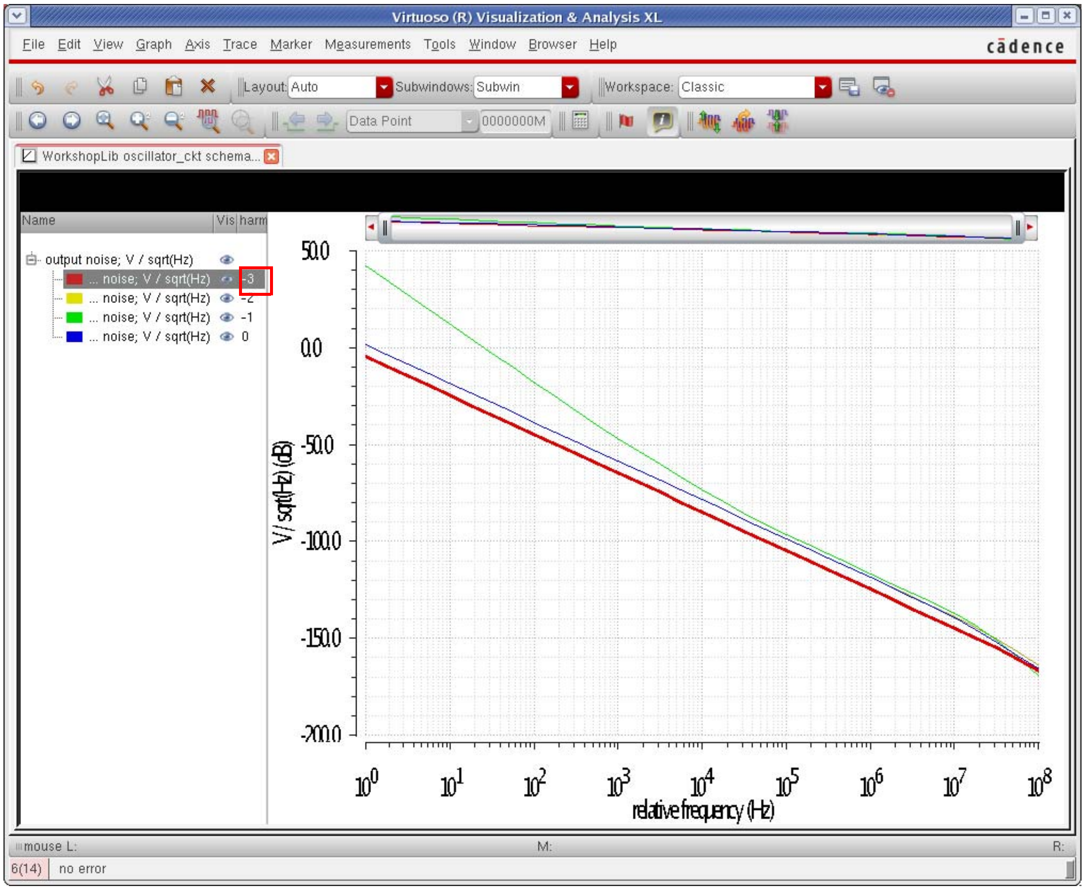


7. To see the noise that is upconverted from near zero frequency, select the -1 sideband. Selecting the -1 sideband means that the noise frequency is one times the pss frequency (the oscillator frequency) below the noise output frequency. This includes the 1/F noise that upconverts to the output.



8. To see the noise that appears by mixing with the second harmonic, select the -2 sideband.

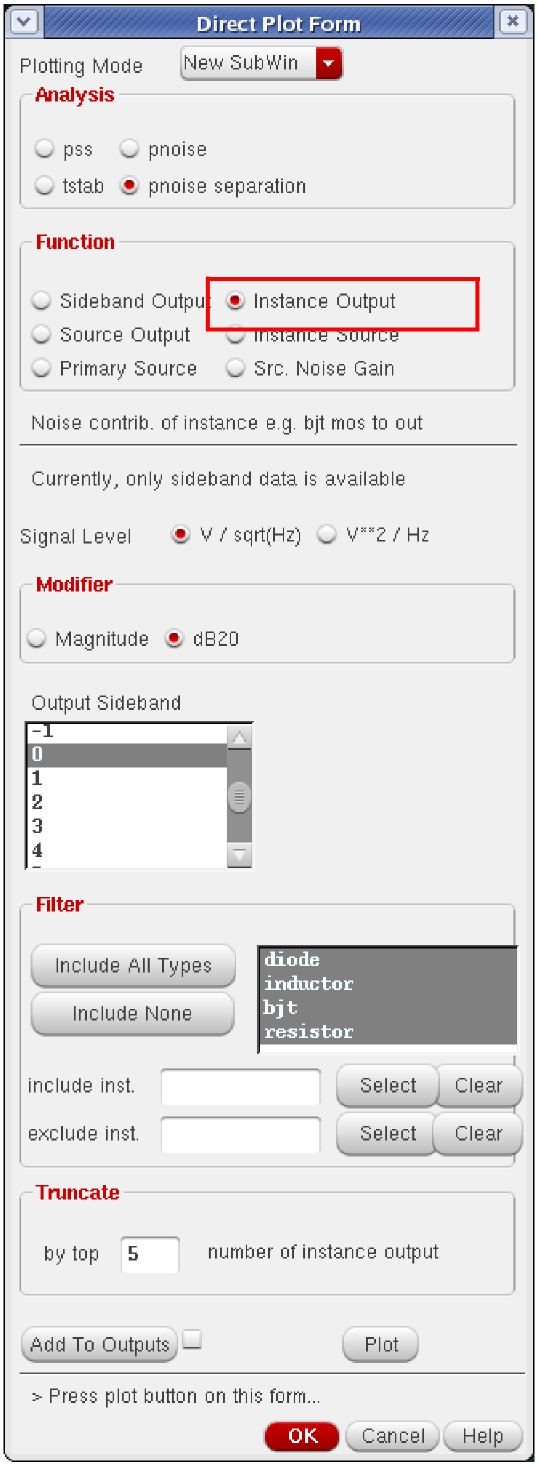
9. To see the noise converted by mixing with the third harmonic, select the -3 sideband.



Viewing the total noise at the output for all the noise mechanisms in each noise device for one noise sideband

This capability allows you to plot the total noise from each device at each separate input frequency. This gives you an idea of not only the frequencies that are troublesome, but also the specific devices that are troublesome.

1. Select Instance Output.



2. Select *V / sqrt(Hz)* or *V**2 / Hz*.

3. Select linear *Magnitude*, or *dB20*.

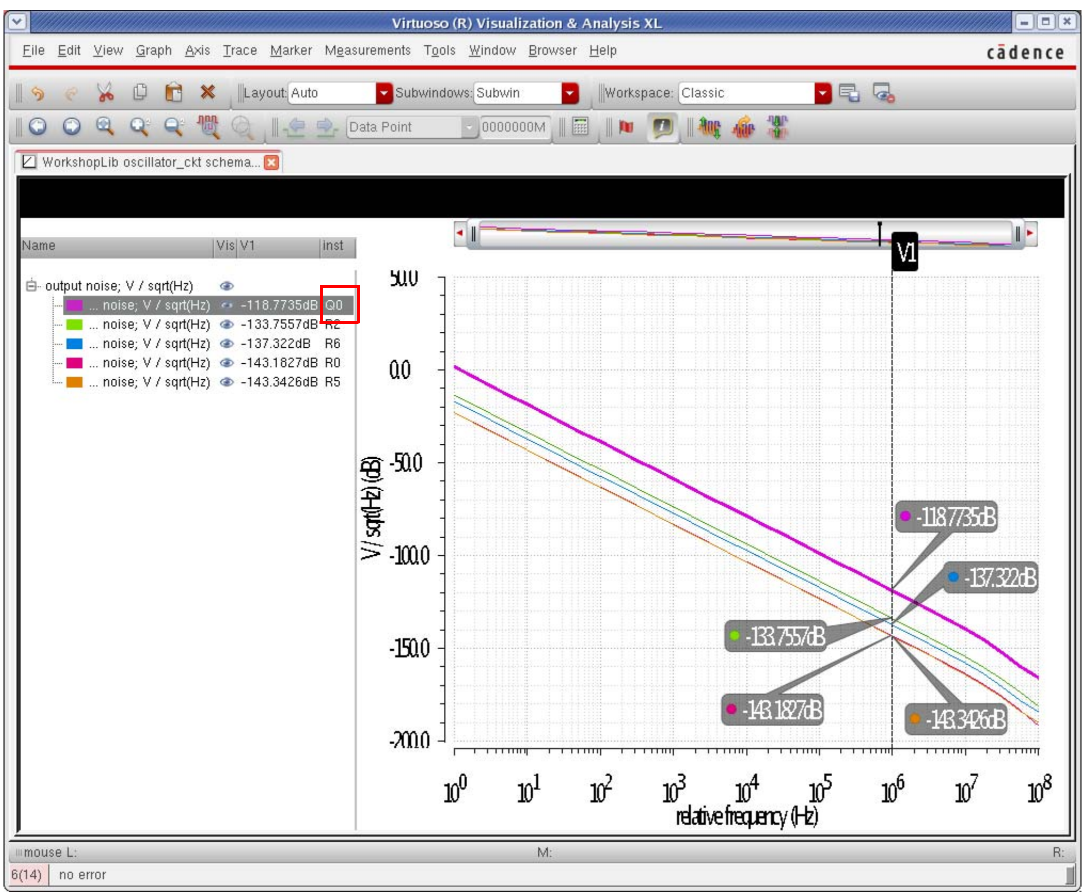
4. Select only one noise sideband.

5. To plot the noise from all the different device types, select *Include All Types*.

6. To make individual selections, choose each individual device type in the list. To select more than one device type at a time, press and hold the <Ctrl> key while you choose an additional category.

7. To include only the devices you select from the schematic included, select I*nclude None*, and click the *Select* button to the right of *Include Inst*. Now select the instances you want to include from the schematic. As you select the instances, the instance name will appear in the list. Select to the right of *Include inst*. Now select the instances you want in the schematic. As you click the instances, the instance Name will appear in the list.

8. In the Truncate field, type in the number of traces you want to view. The instances with the largest noise contributions will be plotted.

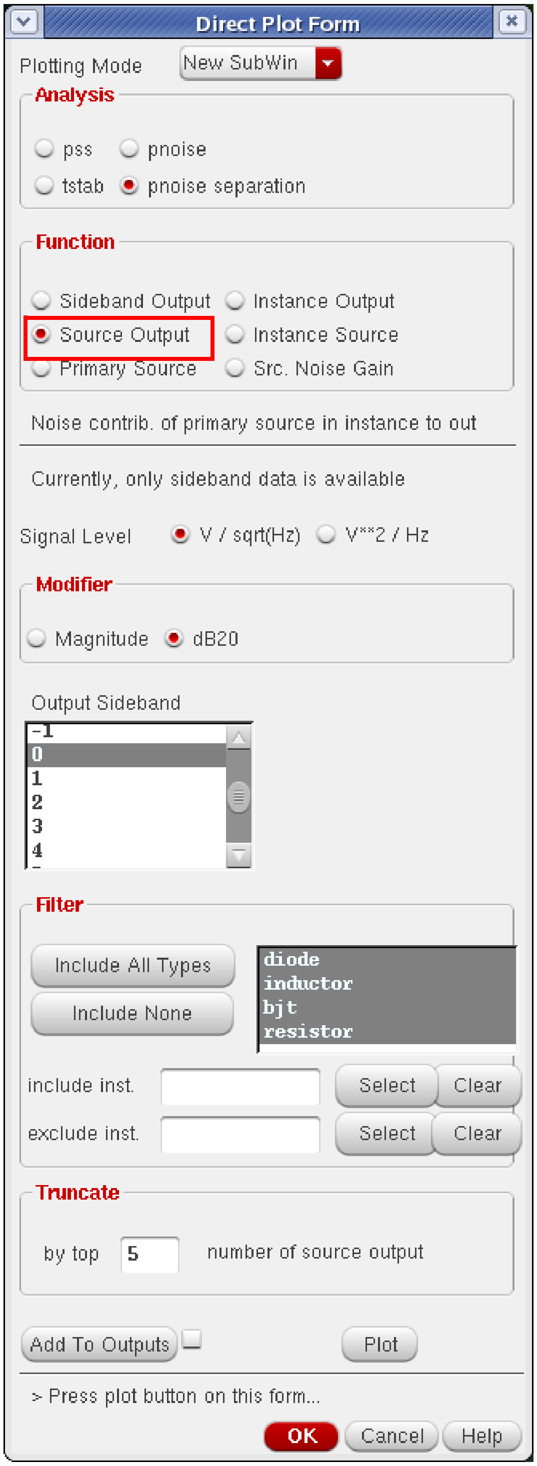


The highlighted trace above is a plot of all the noise contributors in *Q0* at the output of the circuit.

To plot the individual noise contributors form each instance from a single noise sideband

This plot splits the noise into each specific noise generator at each individual noise input frequency.

1. Select Source Output.



2. Select *V / sqrt(Hz)* or *V**2 / Hz*.

3. Select linear *Magnitude*, or *dB20*.

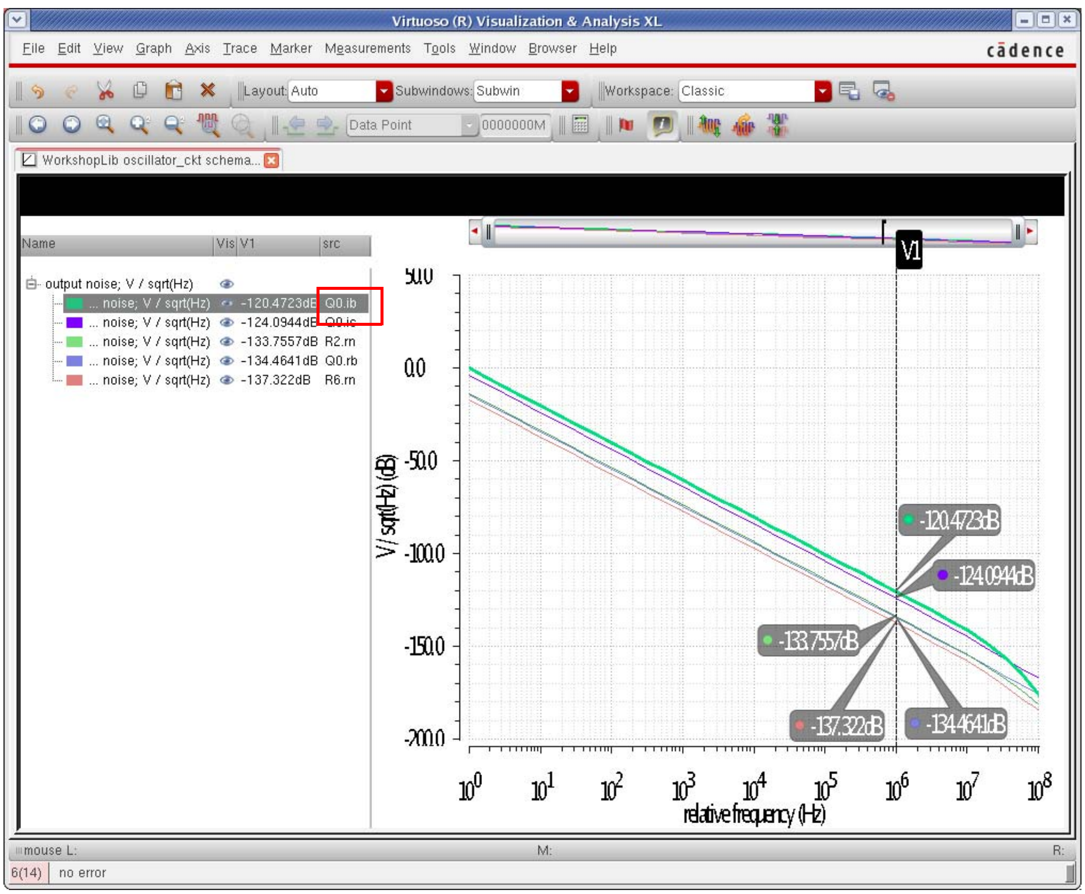
4. Select only one noise sideband.

5. To plot the noise from all the different device types, select *Include All Types*.

6. To make individual selections, choose each individual device type in the list. To select more than one device type at a time, press and hold the <Ctrl> key while you choose an additional category.

7. To include only the devices you select from the schematic included, select *Include None*, and click the *Select* button to the right of *Include Inst*. Now select the instances you want to include from the schematic. As you select the instances, the instance name will appear in the list. Select to the right of *Include inst*. Now select the instances you want in the schematic. As you click the instances, the instance Name will appear in the list.

8. In the Truncate field, type in the number of traces you want to view. The instances with the largest noise contributions will be plotted.

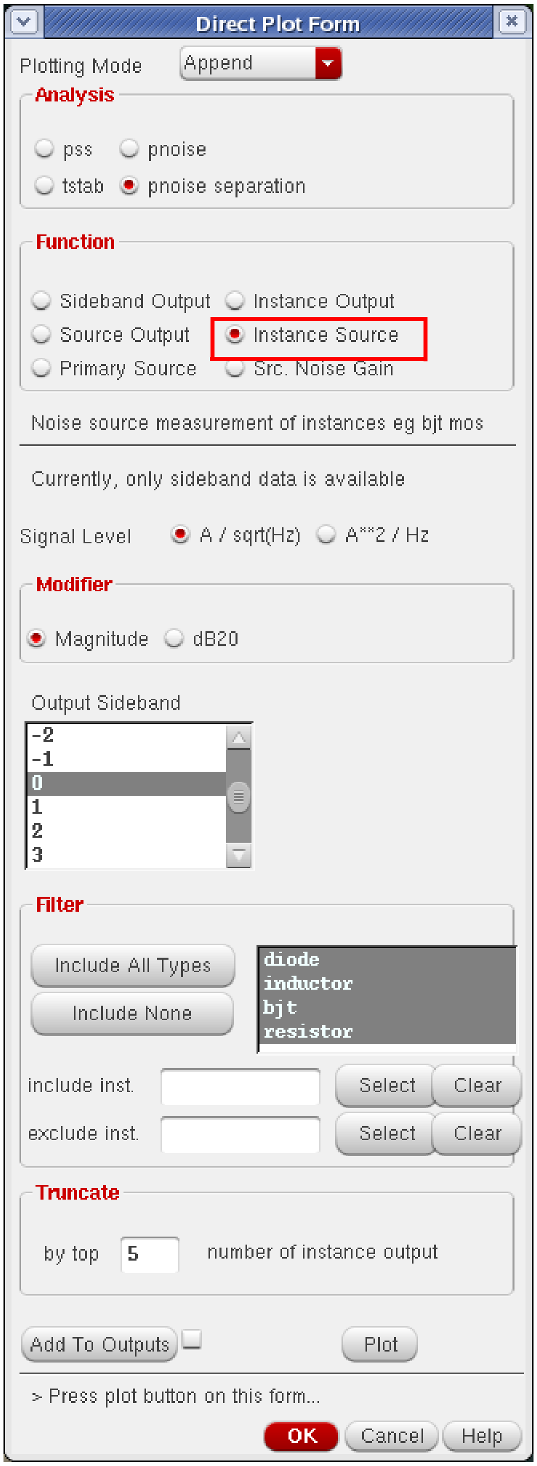


The instance name and the noise parameter are displayed in the waveform tool. See the red square in the above figure. For a table of the noise parameters, see page 123 (77 of 234) of this document.

Viewing the total noise current from each instance at the source of the noise

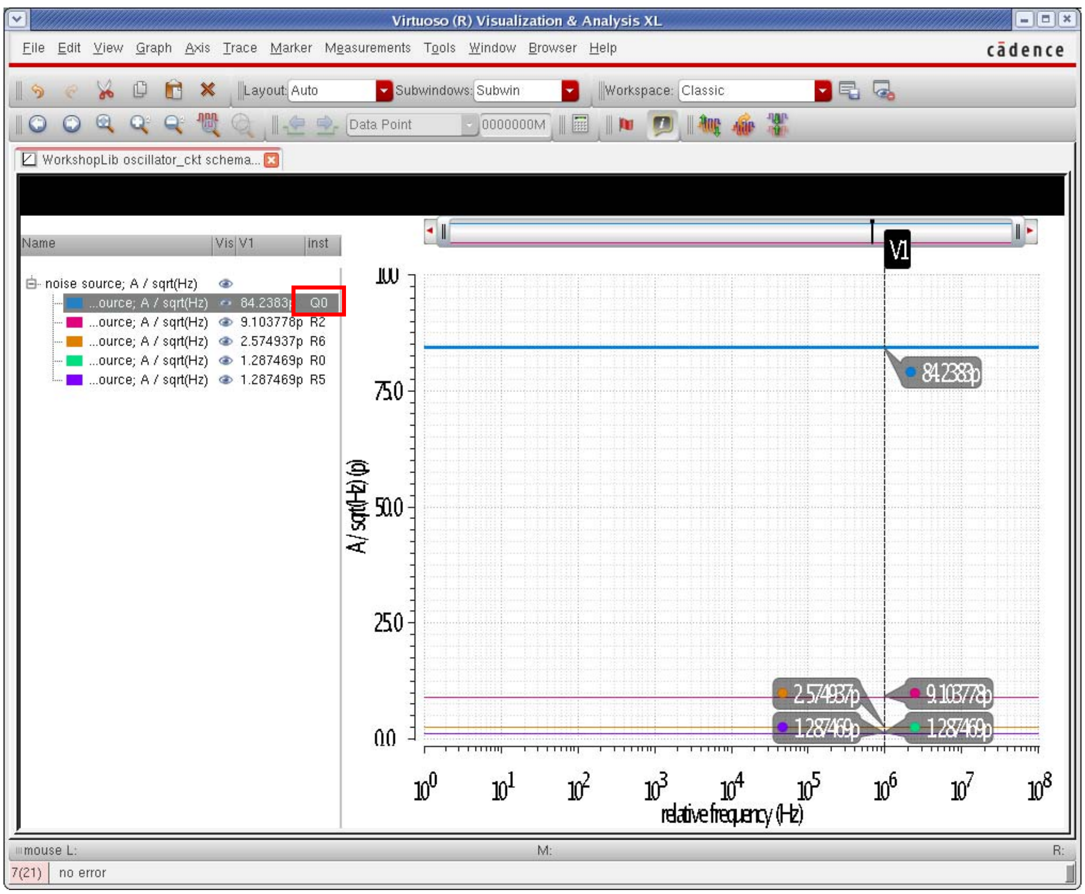
Instance source changes the focus from the output to the noise source itself, and plots the actual noise currents at the source for each individual noise frequency. All the noise from all the individual noise sources combined for each device is available.

1. In the *Direct Plot Form*, select *Instance Source*.



2. Select *V / sqrt(Hz)* or *V**2 / Hz*.

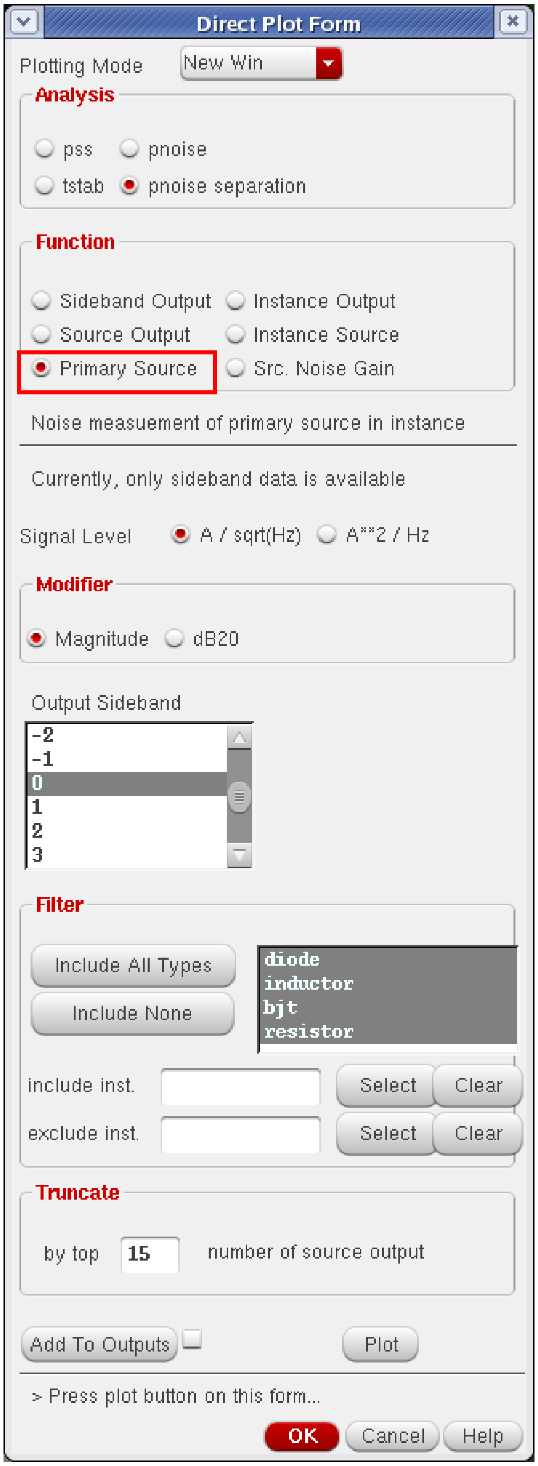
3. Select linear *Magnitude*, or *dB20*.
4. Select only one noise sideband.
5. To plot the noise from all the different device types, select *Include All Types*.
6. To make individual selections, choose each individual device type in the list. To select more than one device type at a time, press and hold the Control key while you choose an additional category.
7. To include only the devices you select from the schematic included, select *Include None*, and click the *Select* button to the right of *Include Inst*. Now select the instances you want to include from the schematic. As you select the instances, the instance name will appear in the list. Select to the right of *Include inst*. Now select the instances you want in the schematic. As you click the instances, the instance Name will appear in the list.
8. In the *Truncate* field, type in the number of traces you want to view. The instances with the largest noise contributions will be plotted. The total noise from all sources in Q0 at 1MHz offset is about 84 picoAmps per root hertz.



Viewing the individual noise current from each noise source within each instance at the source of the noise

This splits out the individual noise currents at the source for each individual noise source in the circuit.

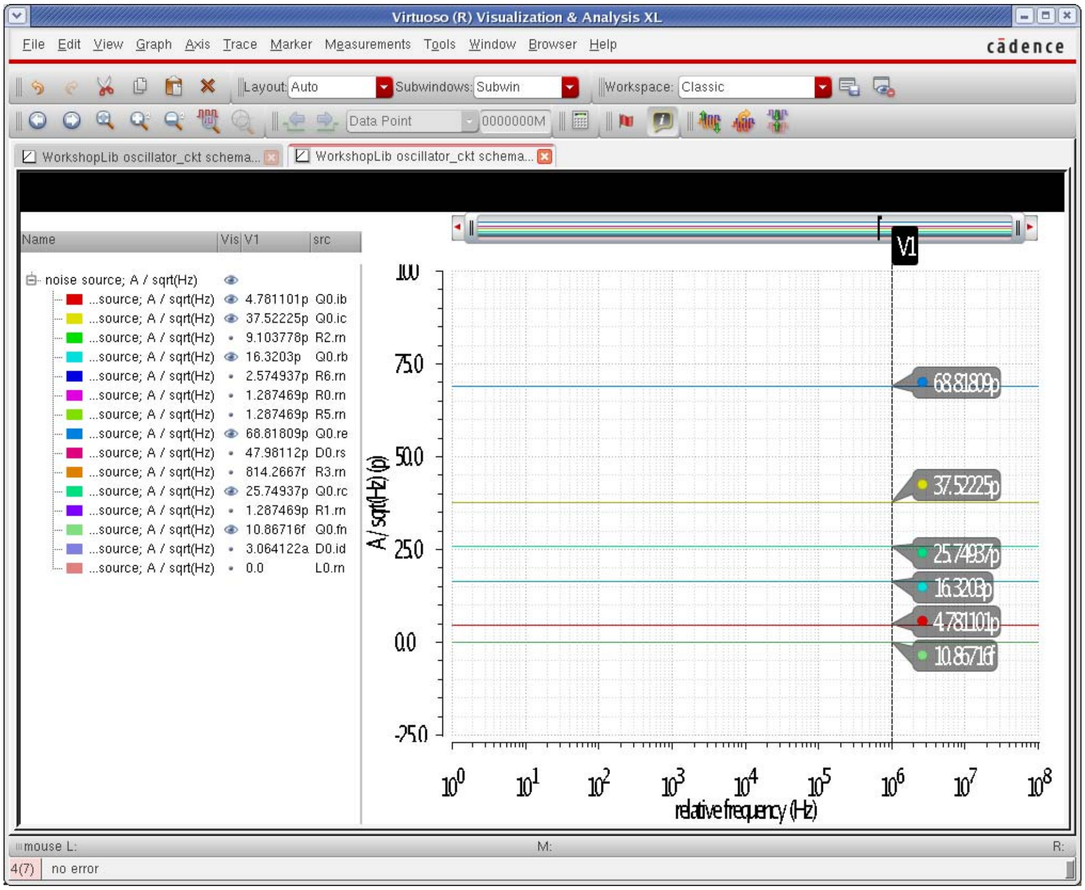
1. In the *Direct Plot Form*, select *Primary Source*.



2. Select *V / sqrt(Hz)* or *V**2 / Hz*.

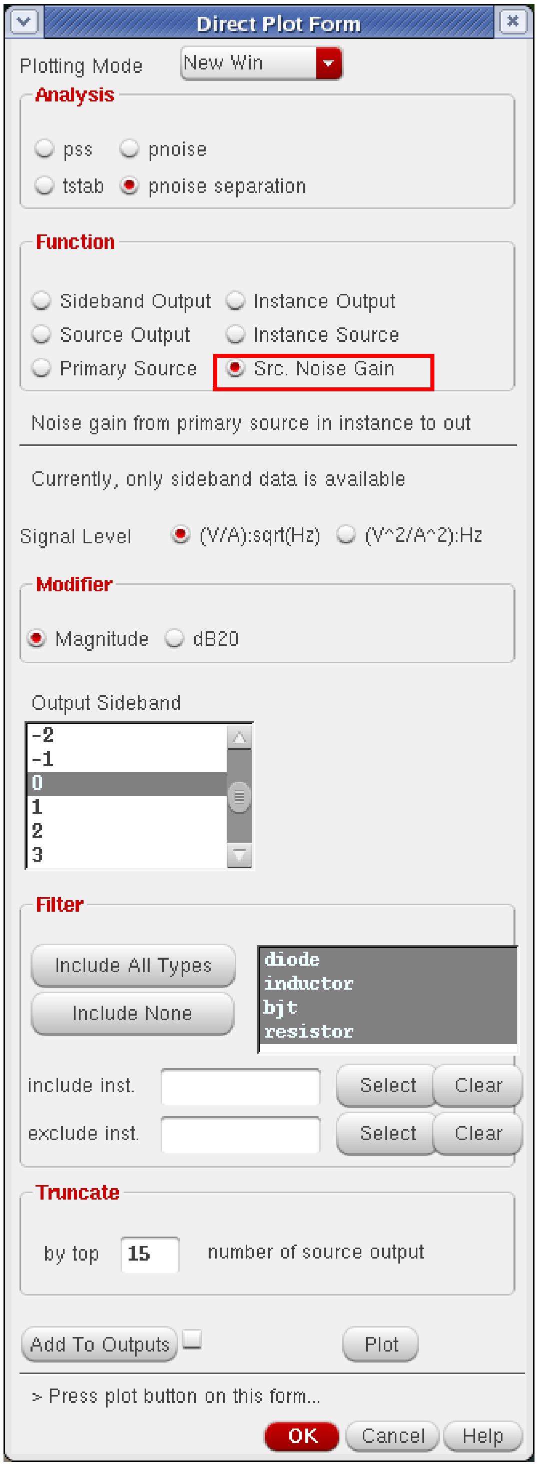
3. Select linear *Magnitude*, or *dB20*.
4. Select only one noise sideband.
5. To plot the noise from all the different device types, select *Include All Types*.
6. To make individual selections, choose each individual device type in the list. To select more than one device type at a time, press and hold the Control key while you choose an additional category.
7. To include only the devices you select from the schematic included, select *Include None*, and click the *Select* button to the right of Include Inst. Now select the instances you want to include from the schematic. As you select the instances, the instance name will appear in the list. Select to the right of *Include inst*. Now select the instances you want in the schematic. As you click the instances, the instance Name will appear in the list.
8. In the *Truncate* field, type in the number of traces you want to view. The instances with the largest noise contributions will be plotted. The individual noise from all the different sources in Q0 at 1MHz offset is shown below. To compare the noise with the total noise

shown in the previous section, add the noise power (not the voltage) from the individual noise sources by taking the square root of the sum of the squares.



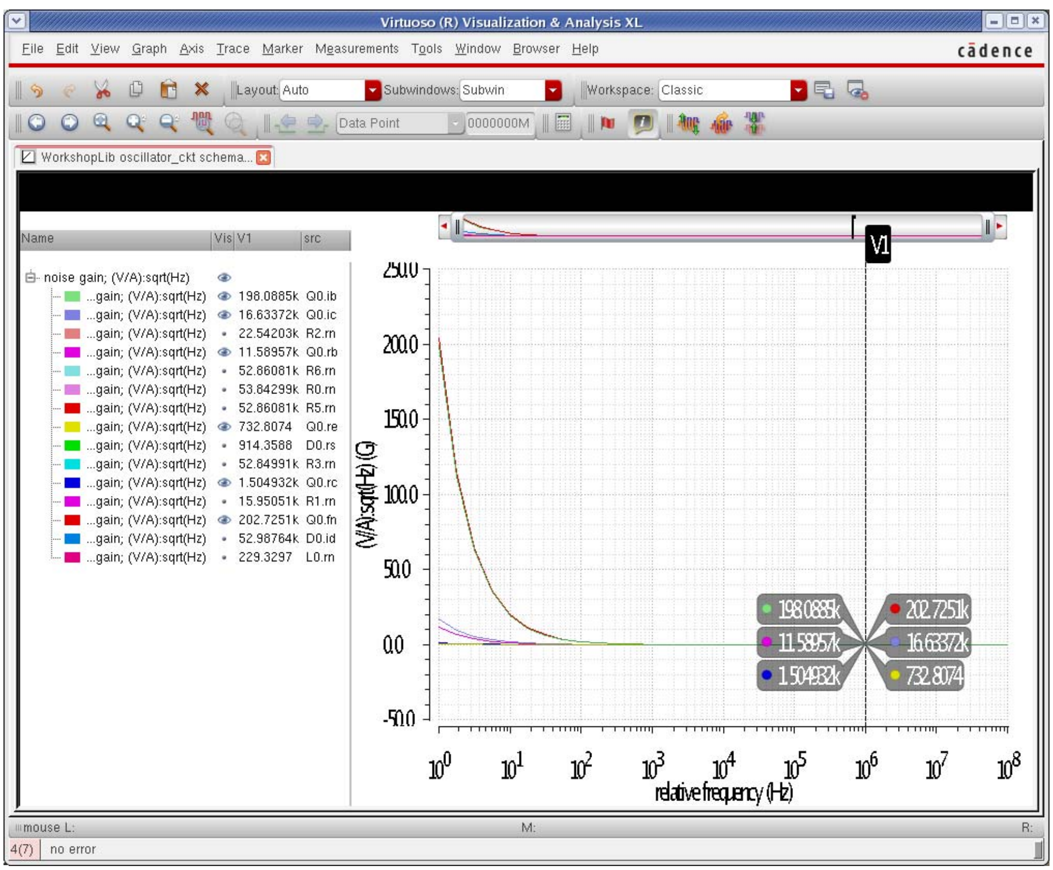
Viewing the transfer function from the individual sources of the noise to the output of the circuit

1. In the *Direct Plot Form*, select *Src Noise Gain*.

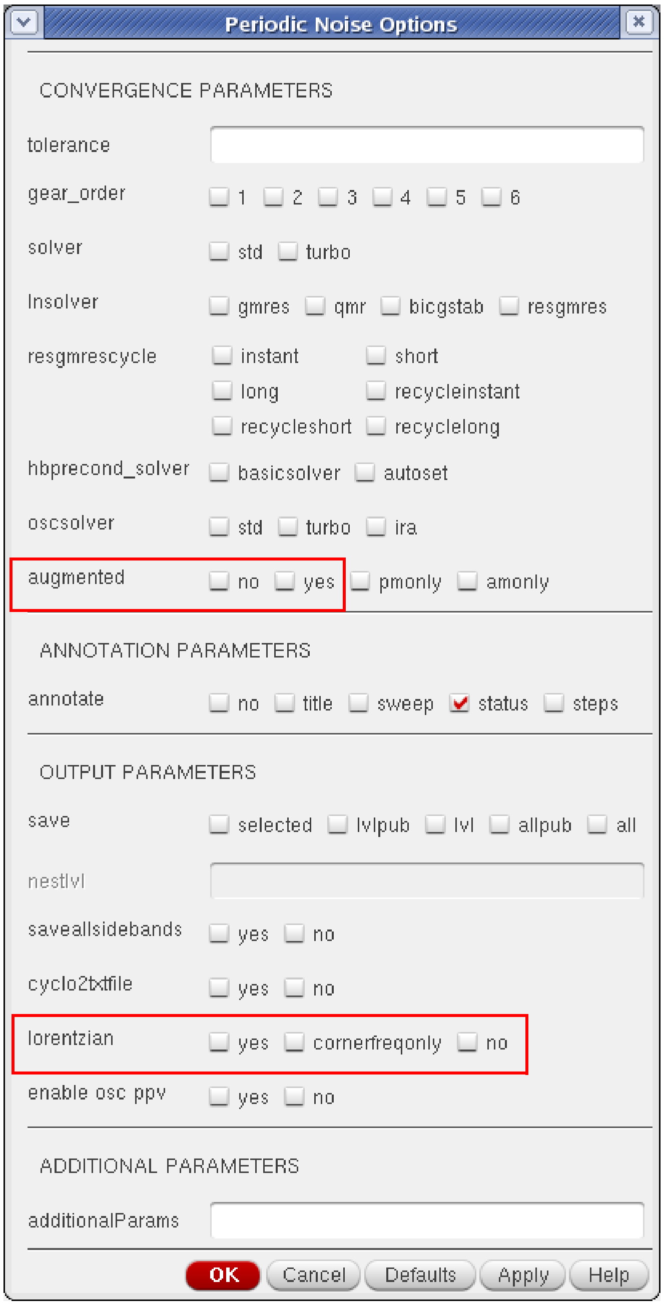


2. Select *V / sqrt(Hz)* or *V**2 / Hz*.

3. Select linear *Magnitude*, or *dB20*.
4. Select only one noise sideband.
5. To plot the noise from all the different device types, select *Include All Types*.
6. To make individual selections, choose each individual device type in the list. To select more than one device type at a time, press and hold the Control key while you choose an additional category.
7. To include only the devices you select from the schematic included, select *Include None*, and click the *Select* button to the right of *Include Inst*. Now select the instances you want to include from the schematic. As you select the instances, the instance name will appear in the list. Select to the right of *Include inst*. Now select the instances you want in the schematic. As you click the instances, the instance Name will appear in the list.
8. In the *Truncate* field, type in the number of traces you want to view. The instances with the largest noise contributions will be plotted. The gain from all the noise sources to the output is plotted. Only the noise sources from Q0 are visible. For a table that lists the noise parameter for each device type, please see page 123 (77 of 234) of this document.



Commonly Used Pnoise Options



augmented=yes: is the default and is required for most oscillators that have long time constants, perhaps from a Low Drop-Out (LDO) regulator. Either take the default or set *augmented* to *yes* for every circuit.

Lorentzian

There are two ways of looking at phase noise. One way is to think of measuring the spectral output of the oscillator with a network analyzer. In this case, as the frequency gets close to the oscillator frequency, the amplitude starts going up, levels off, and then drops as the oscillator frequency is passed. In this case, the noise cannot be larger than the oscillations themselves, which is 0 dBc. If this is how you view phase noise, set the *lorentzian* option to *yes*.

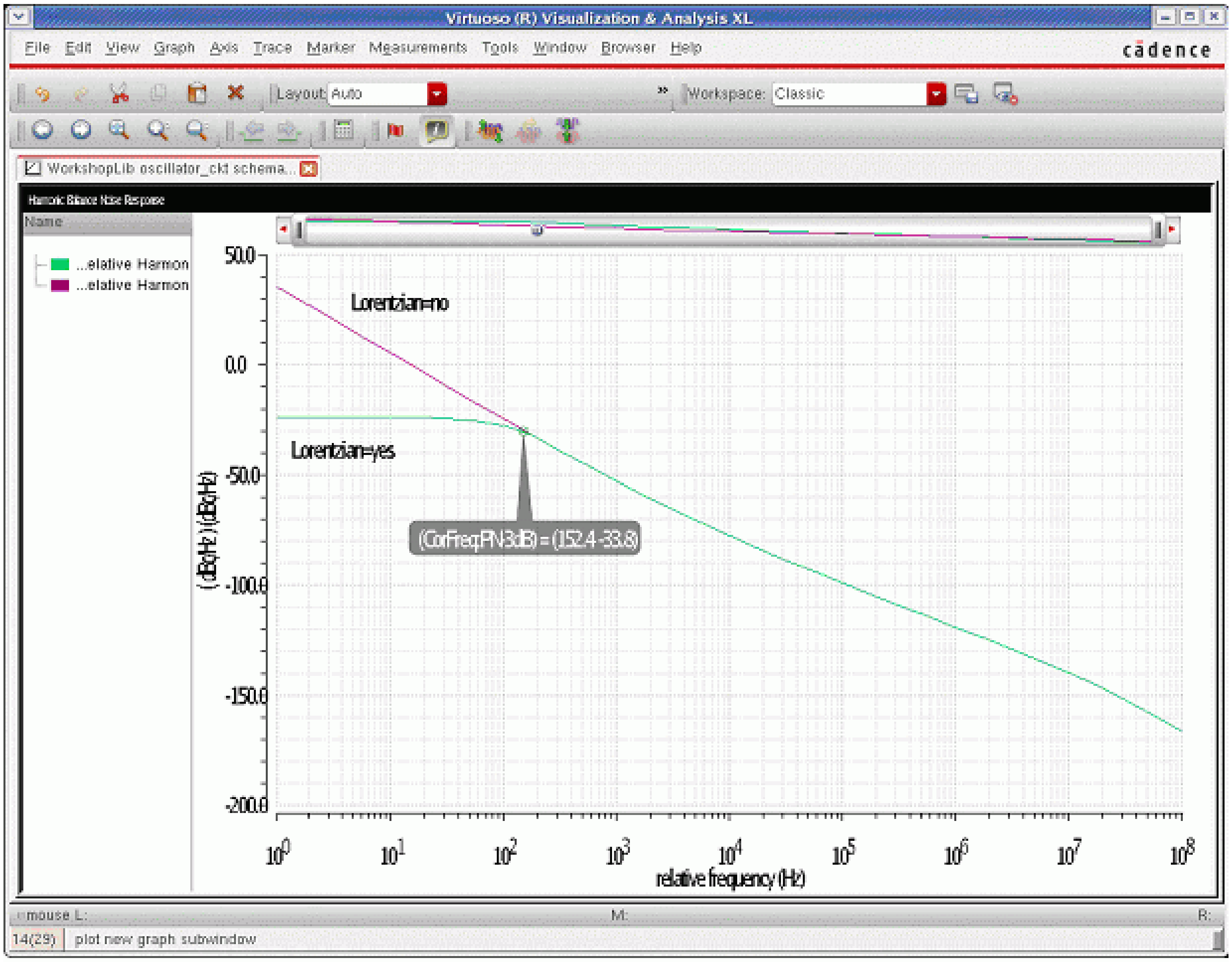
If you think in terms of jitter, in one cycle of the oscillator output, because of the noise, you can calculate one standard deviation of the timing. If you go two cycles, and you integrate the noise with respect to time, you get twice as much noise power, or about 1.4 times the noise voltage or jitter. When you get to an infinite number of oscillator cycles, the jitter also becomes infinite. As you increase the time, you are lowering the noise frequency, and in this case, as the time becomes large, the frequency becomes very low. At infinite time, you get infinite jitter, and this occurs at zero offset from the carrier. Jitter is just another way of looking at phase noise. In other words, the phase noise can easily exceed the carrier amplitude (0 dBc) as the offset frequency becomes small because it is heading to infinity at zero frequency offset. If this is how you view phase noise, set the *lorentzian* option to *no*.

lorentzian=yes means calculate the leveling off. **lorentzian=cornerfreqonly** calculates the continuously rising phase noise, but it places a marker on the phase noise curve at the frequency where the phase noise would level off. *no* causes the phase noise to continue to rise as the offset frequency approaches zero.

Comparison Between Lorentzian Yes and No

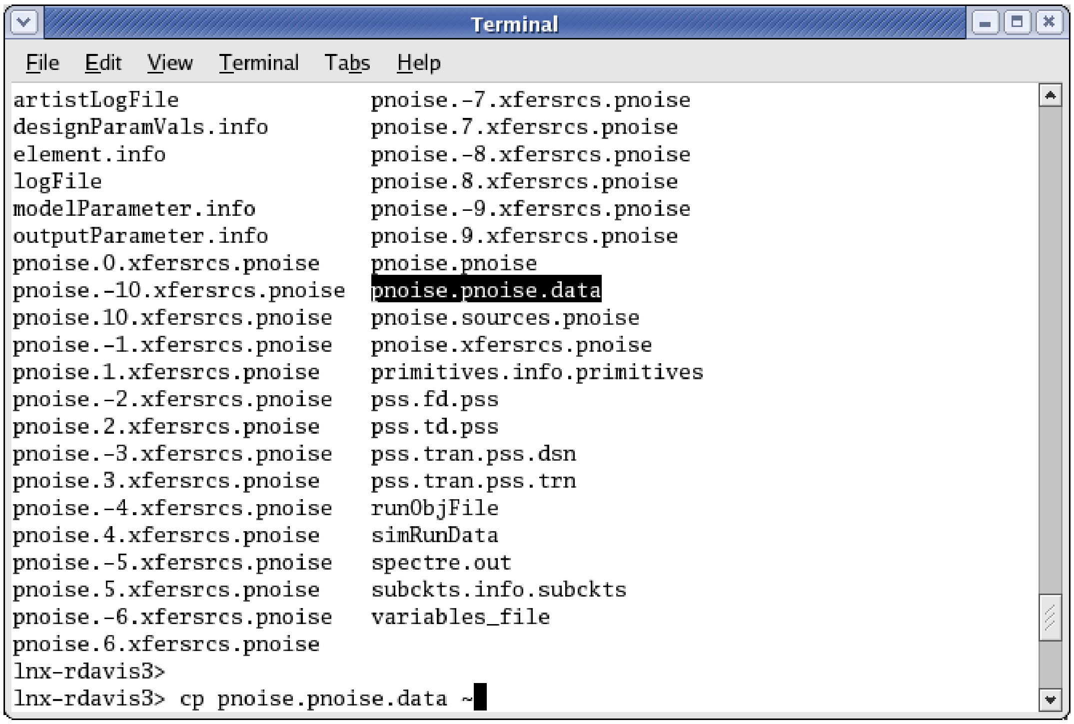
As noted above, lorentzian causes leveling off of the phase noise at low frequency.

Note that when *lorentzian=no*, the phase noise curve goes above 0 dBc. This would mean that the noise is larger than the amplitude of the oscillator, which is not physical. This is because pnoise is a small-signal analysis that does not recognize the large-signal limits of the circuit like clipping. It is similar to AC where you can specify an input magnitude of 1 megavolt, and see 10 megavolts on the output.



Cyclo2txtfile=yes: Setting this option causes pnoise to be run with all the noise frequencies and sidebands specified in the *Choosing Analyses* form. It creates a text file with the sorted frequencies and the volts-squared per hertz values that result for your circuit. This file is in the psf directory, and is called *pnoise.pnoise.data*.

To locate the `psf` directory, first select *Setup - Simulator/Directory/Host* in ADE. The path to the simulation directory is shown in the `Project Directory` field. In that directory, browse to the simulation/<Circuit_Name>/spectre/<View_Name>/psf directory. View_Name is typically schematic or config. Do a list, and look for the file pnoise.pnoise.data.



If you intend to keep this file for use in a voltage source with the noise file set to a name, make sure to copy that file to another directory to save it. The psf directory contents are deleted at the beginning of each simulation.

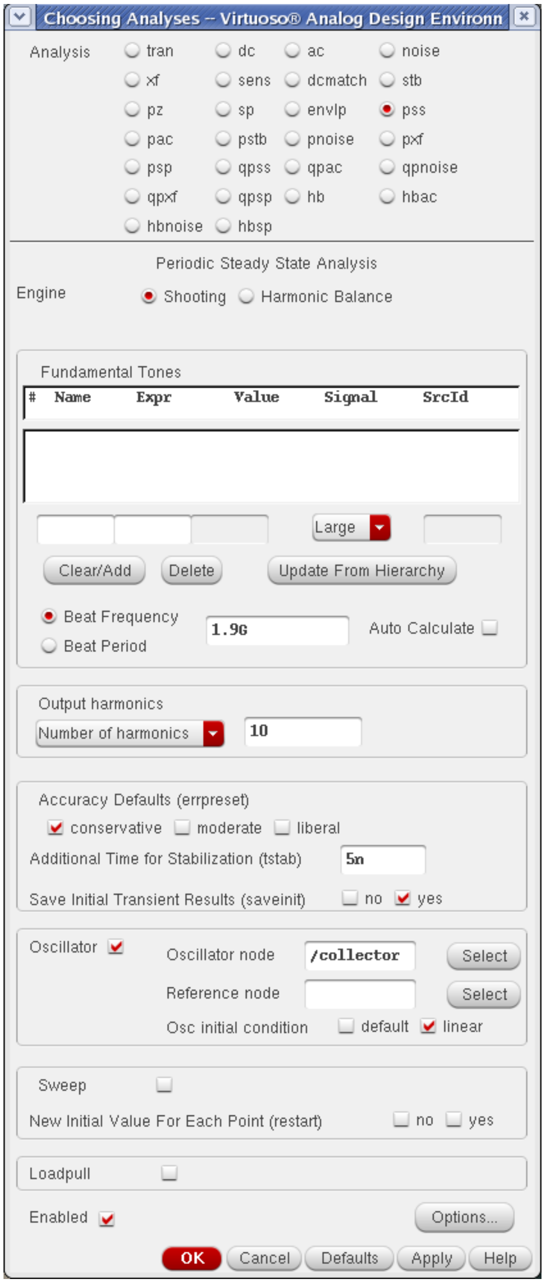
For more information regarding the other pnoise options, refer to the commonly used pnoise options section starting on page 81 (35 of 242) (of this document.

Noise Type = Modulated

Modulated allows the measurement of the AM and PM components of noise, along with the noise response above and below the harmonic number specified in the pnoise *Choosing Analyses* form.

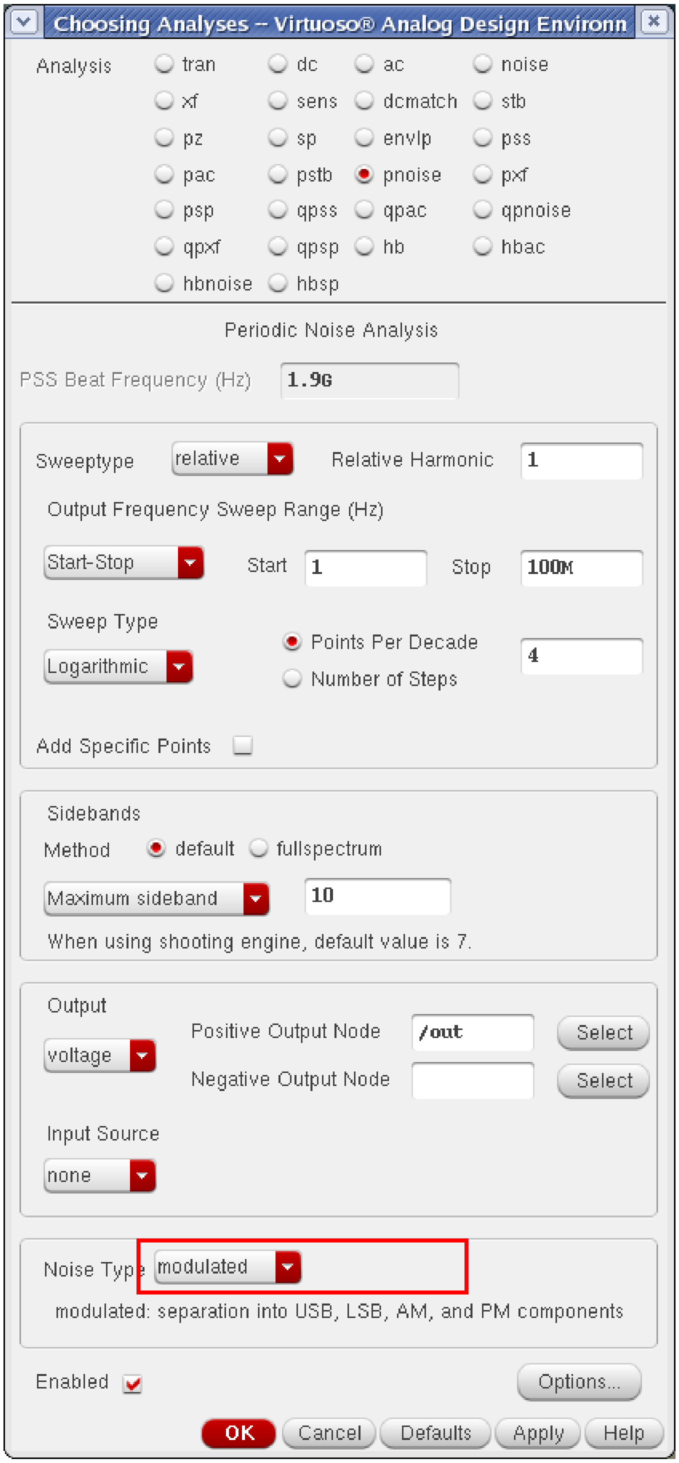
1. First set the pss form, as shown in the following figure.

You need an estimate of the oscillation frequency and the number of harmonics for your circuit. For more information regarding the pss setup, see Overview of Periodic Steady-State (pss) Analysis on page 547.



2. Next set up the pnoise form.

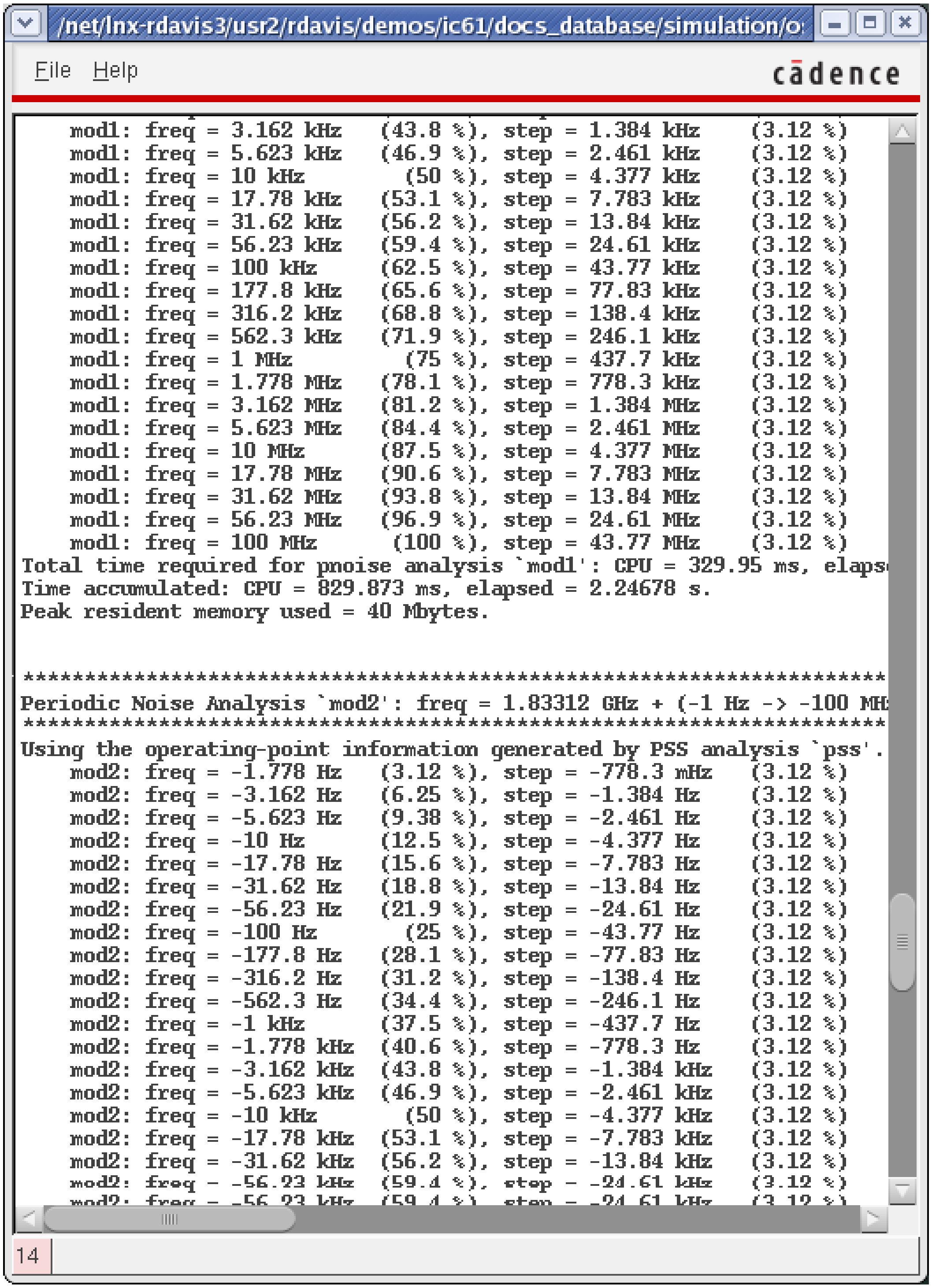
3. Select *modulated* from the *Noise Type* drop-down list.



Note that the rest of the fields are the same as for when noise=sources as shown in the preceding section.

4. Run the simulation.

Note that positive and negative frequencies are run in pnoise. Since *relative* was chosen for the *Sweep Type*, the frequencies above and below the first harmonic are analyzed for noise.



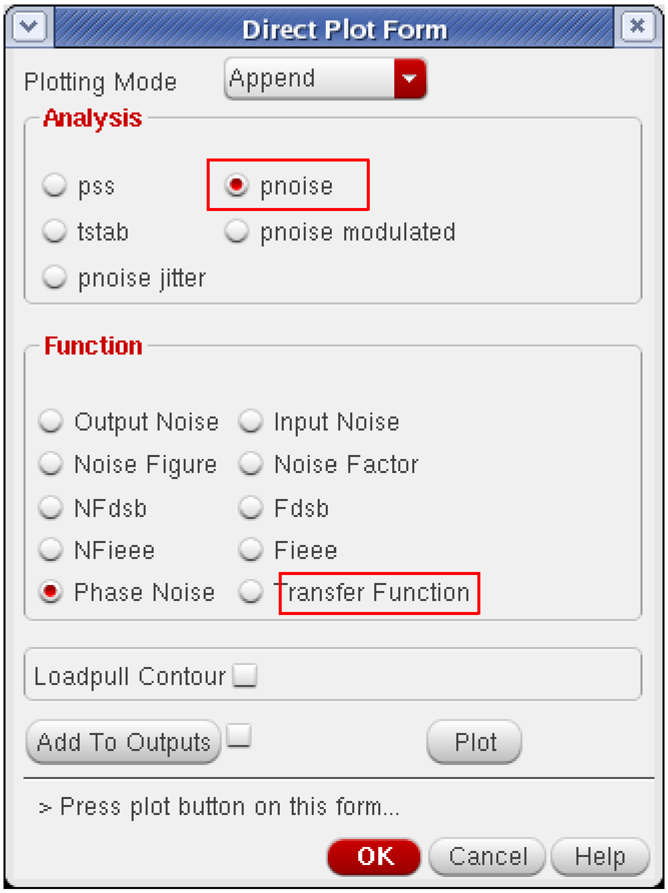
/net/lnx-rdavis3/usr2/rdavis/demos/ic61/docs_database/simulation/o

File Help

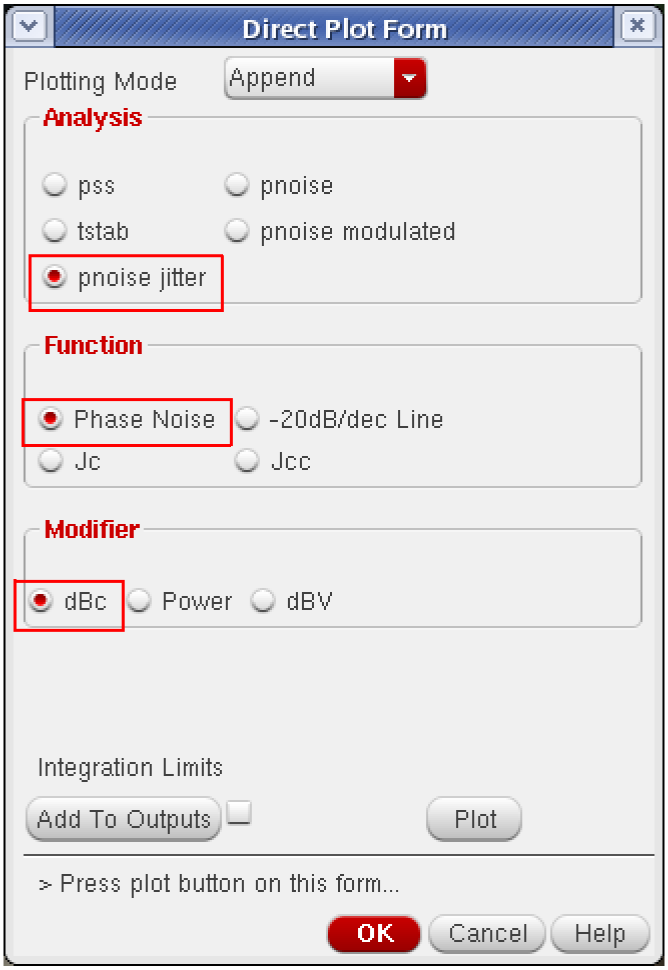
```
    mod1: freq = 3.162 kHz   (43.8 %), step = 1.384 kHz    (3.12 %)
    mod1: freq = 5.623 kHz   (46.9 %), step = 2.461 kHz    (3.12 %)
    mod1: freq = 10 kHz        (50 %), step = 4.377 kHz    (3.12 %)
    mod1: freq = 17.78 kHz   (53.1 %), step = 7.783 kHz    (3.12 %)
    mod1: freq = 31.62 kHz   (56.2 %), step = 13.84 kHz    (3.12 %)
    mod1: freq = 56.23 kHz   (59.4 %), step = 24.61 kHz    (3.12 %)
    mod1: freq = 100 kHz     (62.5 %), step = 43.77 kHz    (3.12 %)
    mod1: freq = 177.8 kHz   (65.6 %), step = 77.83 kHz    (3.12 %)
    mod1: freq = 316.2 kHz   (68.8 %), step = 138.4 kHz    (3.12 %)
    mod1: freq = 562.3 kHz   (71.9 %), step = 246.1 kHz    (3.12 %)
    mod1: freq = 1 MHz         (75 %), step = 437.7 kHz    (3.12 %)
    mod1: freq = 1.778 MHz   (78.1 %), step = 778.3 kHz    (3.12 %)
    mod1: freq = 3.162 MHz   (81.2 %), step = 1.384 MHz    (3.12 %)
    mod1: freq = 5.623 MHz   (84.4 %), step = 2.461 MHz    (3.12 %)
    mod1: freq = 10 MHz      (87.5 %), step = 4.377 MHz    (3.12 %)
    mod1: freq = 17.78 MHz   (90.6 %), step = 7.783 MHz    (3.12 %)
    mod1: freq = 31.62 MHz   (93.8 %), step = 13.84 MHz    (3.12 %)
    mod1: freq = 56.23 MHz   (96.9 %), step = 24.61 MHz    (3.12 %)
    mod1: freq = 100 MHz      (100 %), step = 43.77 MHz    (3.12 %)
Total time required for pnoise analysis `mod1': CPU = 329.95 ms, elaps
Time accumulated: CPU = 829.873 ms, elapsed = 2.24678 s.
Peak resident memory used = 40 Mbytes.


**********************************************************************
Periodic Noise Analysis `mod2': freq = 1.83312 GHz + (-1 Hz -> -100 MH
**********************************************************************
Using the operating-point information generated by PSS analysis `pss'.
    mod2: freq = -1.778 Hz   (3.12 %), step = -778.3 mHz   (3.12 %)
    mod2: freq = -3.162 Hz   (6.25 %), step = -1.384 Hz    (3.12 %)
    mod2: freq = -5.623 Hz   (9.38 %), step = -2.461 Hz    (3.12 %)
    mod2: freq = -10 Hz      (12.5 %), step = -4.377 Hz    (3.12 %)
    mod2: freq = -17.78 Hz   (15.6 %), step = -7.783 Hz    (3.12 %)
    mod2: freq = -31.62 Hz   (18.8 %), step = -13.84 Hz    (3.12 %)
    mod2: freq = -56.23 Hz   (21.9 %), step = -24.61 Hz    (3.12 %)
    mod2: freq = -100 Hz       (25 %), step = -43.77 Hz    (3.12 %)
    mod2: freq = -177.8 Hz   (28.1 %), step = -77.83 Hz    (3.12 %)
    mod2: freq = -316.2 Hz   (31.2 %), step = -138.4 Hz    (3.12 %)
    mod2: freq = -562.3 Hz   (34.4 %), step = -246.1 Hz    (3.12 %)
    mod2: freq = -1 kHz      (37.5 %), step = -437.7 Hz    (3.12 %)
    mod2: freq = -1.778 kHz  (40.6 %), step = -778.3 Hz    (3.12 %)
    mod2: freq = -3.162 kHz  (43.8 %), step = -1.384 kHz   (3.12 %)
    mod2: freq = -5.623 kHz  (46.9 %), step = -2.461 kHz   (3.12 %)
    mod2: freq = -10 kHz       (50 %), step = -4.377 kHz   (3.12 %)
    mod2: freq = -17.78 kHz  (53.1 %), step = -7.783 kHz   (3.12 %)
    mod2: freq = -31.62 kHz  (56.2 %), step = -13.84 kHz   (3.12 %)
    mod2: freq = -56.23 kHz  (59.4 %), step = -24.61 kHz   (3.12 %)
```

14

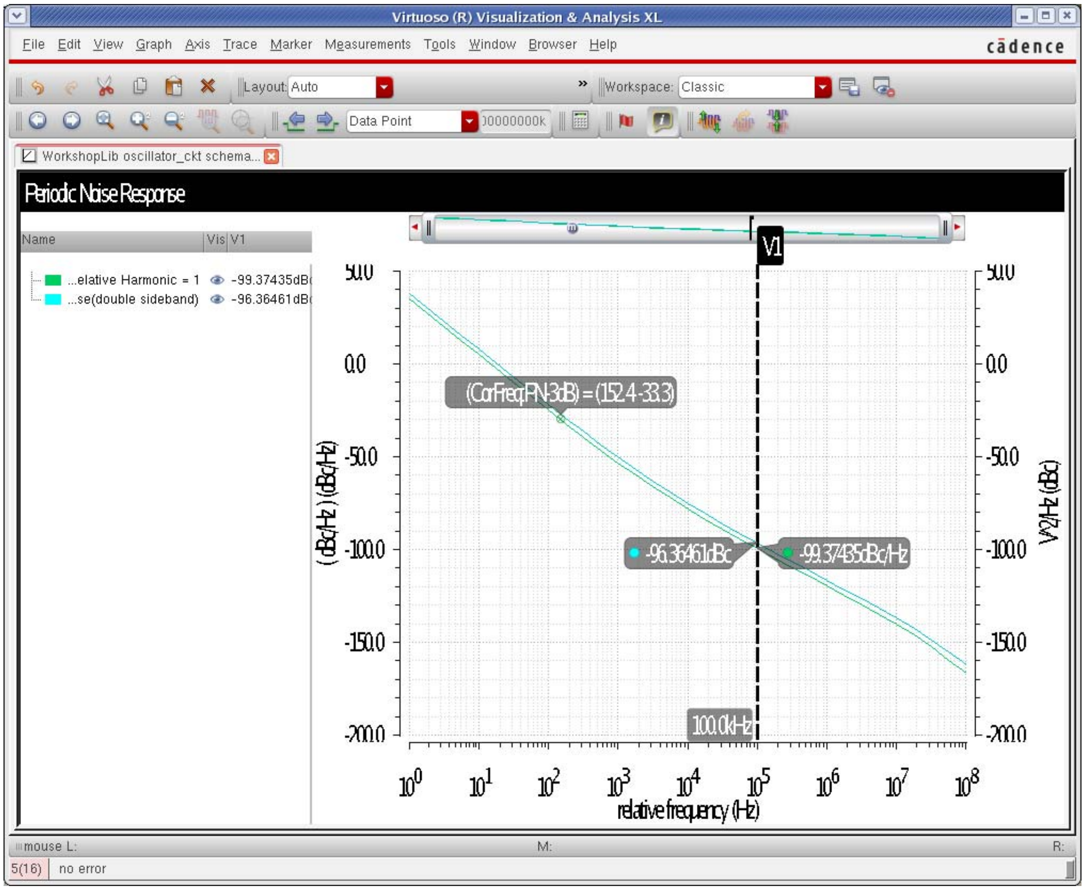
5. In the ADE window, select *Results - Direct Plot - Main Form*.
6. On the *Direct Plot Form,* plot the phase noise from the *pnoise* results.



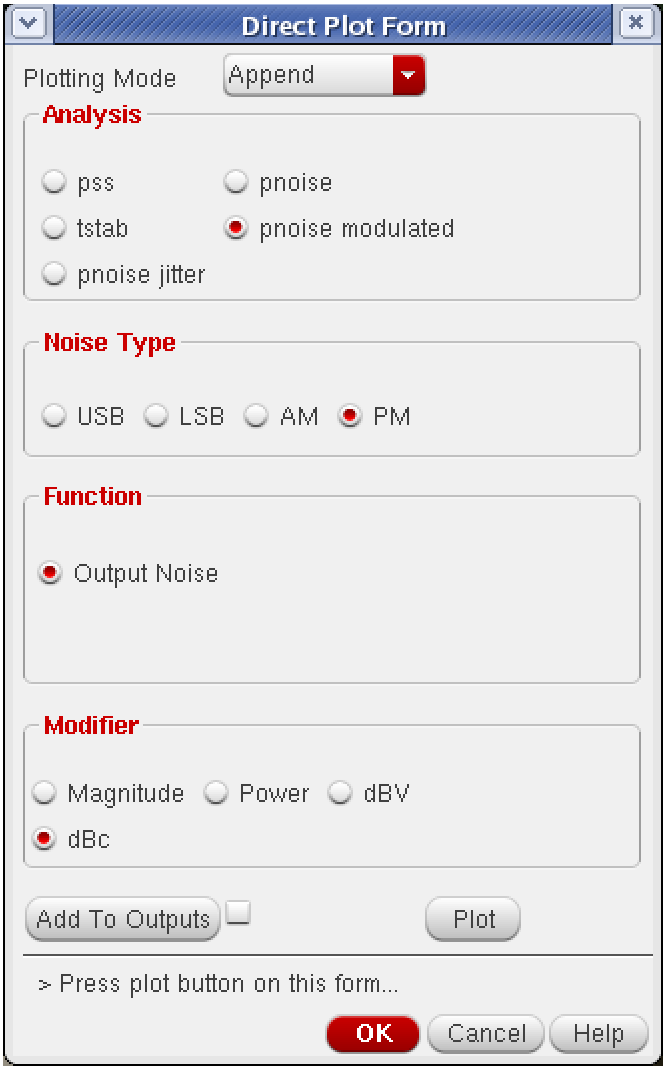
7. Next, plot the phase noise from the *pnoise jitter* result set.



Note that the phase noise from the pnoise result contains the noise in the frequencies above the oscillator frequency only, and phase noise from the pnoise jitter results contain the noise power from above and below the oscillator frequency. This is the reason for the 3.01 dB difference.



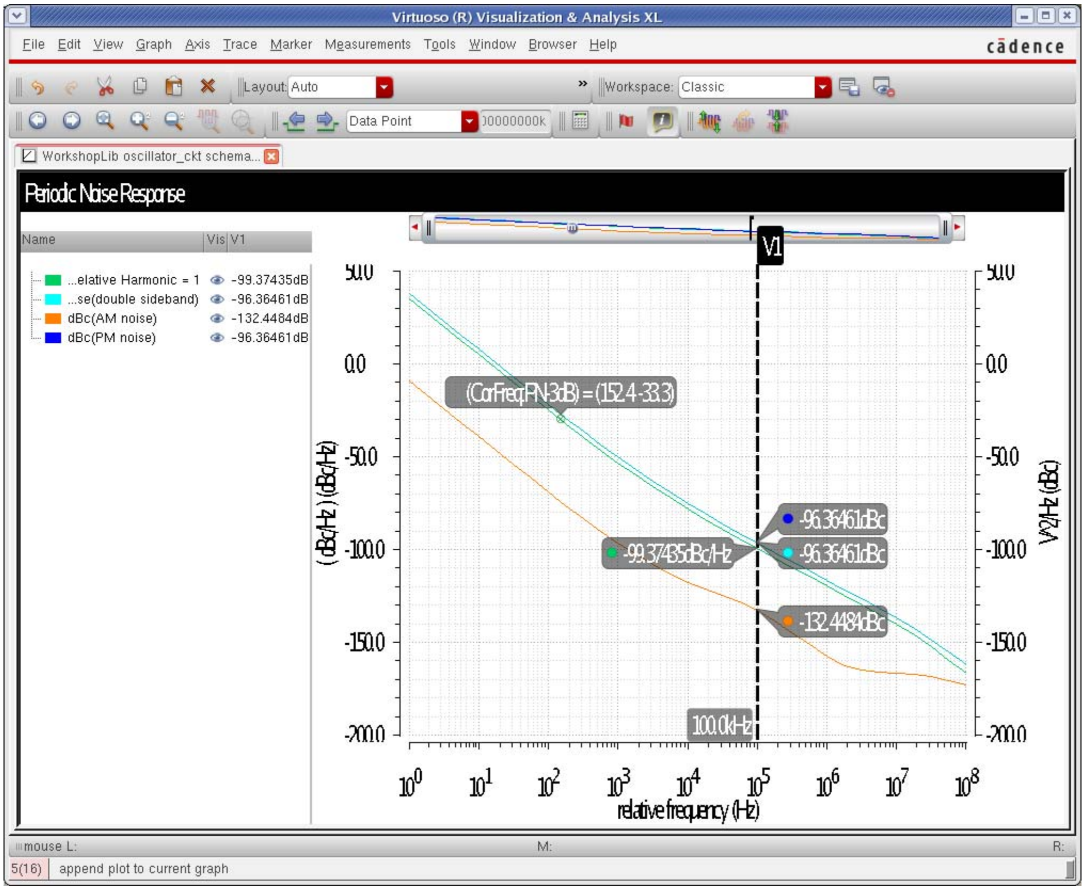
8. Select *pnoise modulated* from the *Analysis* section.



9. Select *AM* or *PM* from the *Noise Type* section.

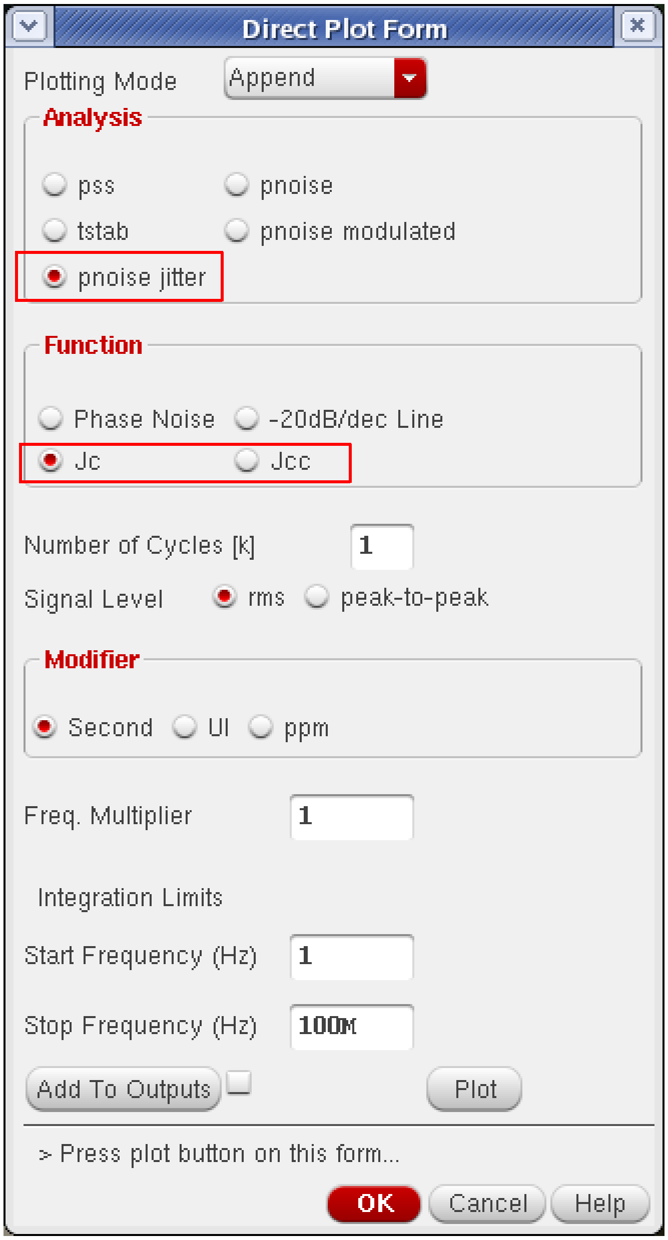
Note that the AM and PM noise are also double sideband measurements. This can be seen in the result by noting that the phase noise from the pnoise jitter result set is equal to the PM component of phase noise.

10. Click *Plot*.



To plot cycle jitter or cycle-to-cycle jitter

1. On the *Direct Plot Form*, select *pnoise jitter* from the *Analysis* section.

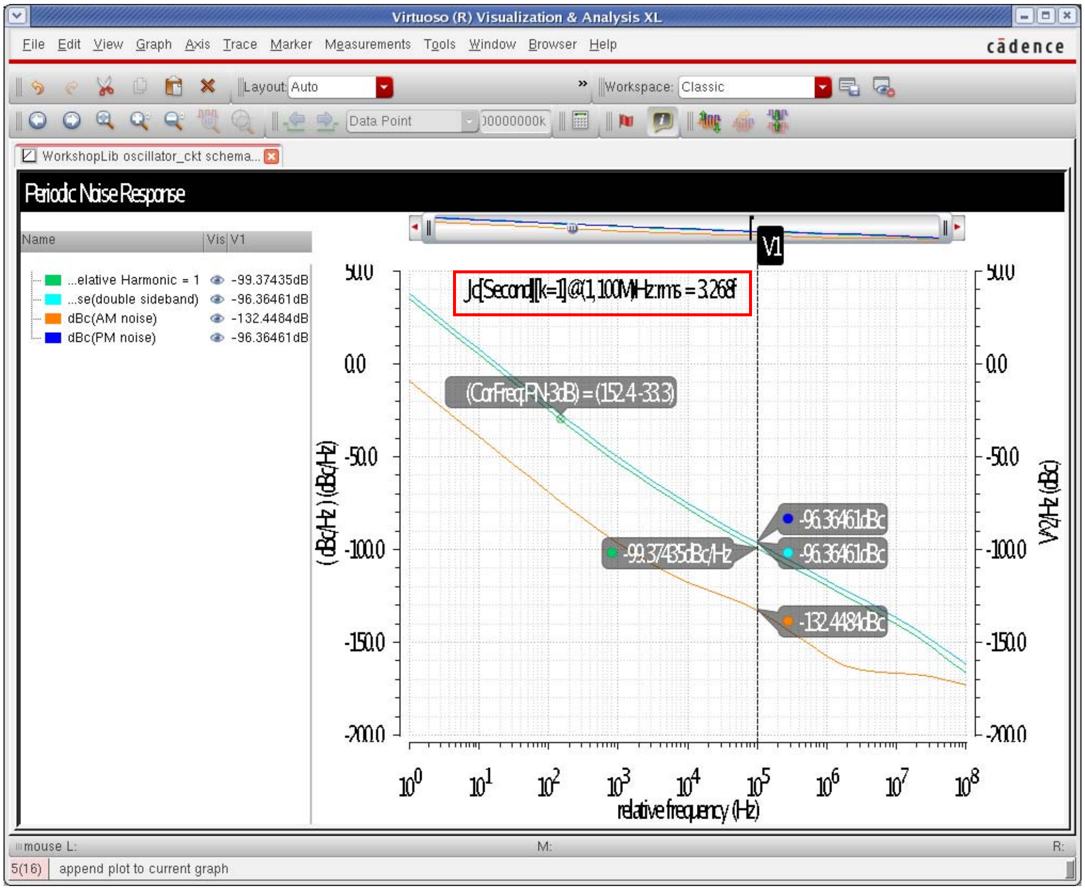


2. Select *Jc* or *Jcc* from the *Function* section.

   The frequency range will default to the range set in the *Choose Analyses* form.

3. Click *Plot*.

The jc or jcc calculation is shown as a label that is added to the waveform tool.



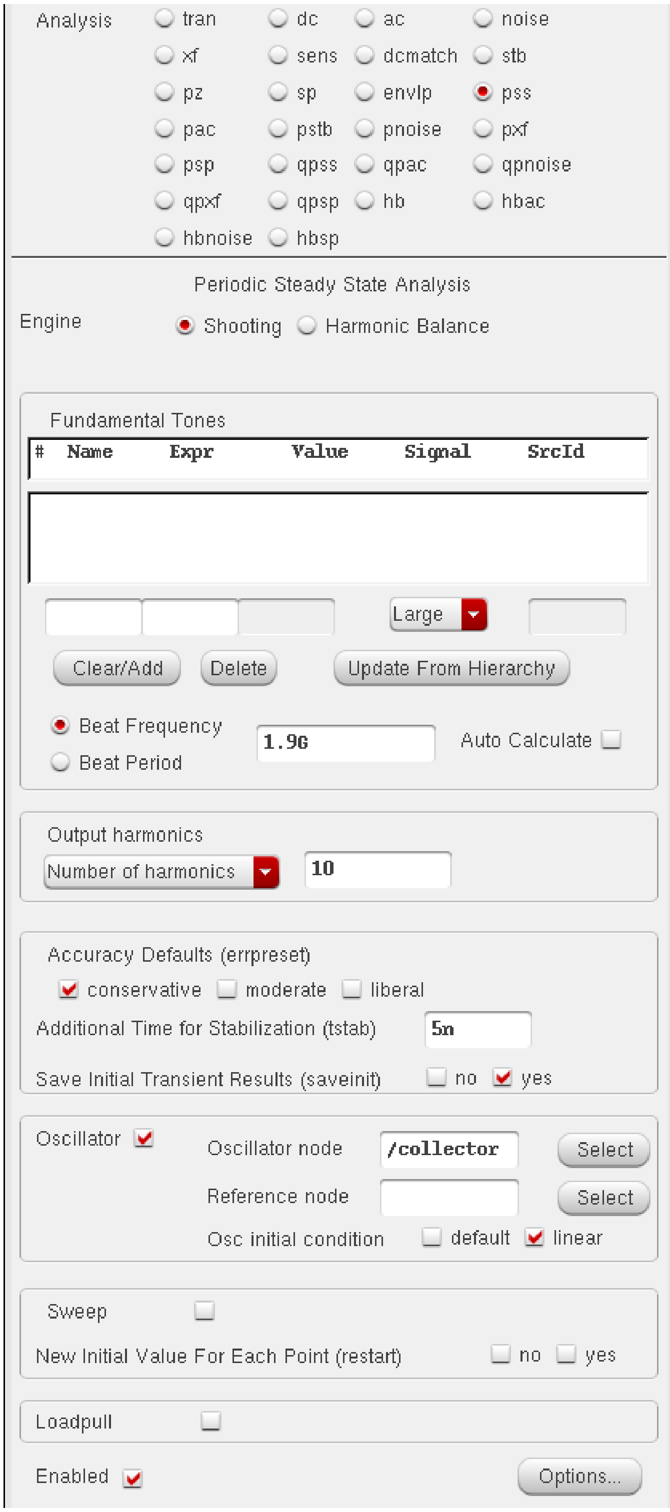
Noise Type = Jitter (FM)

Noise Type = Jitter (FM) produces the same output as Noise Type = Modulated. See Noise Type = Modulated on page 1032.

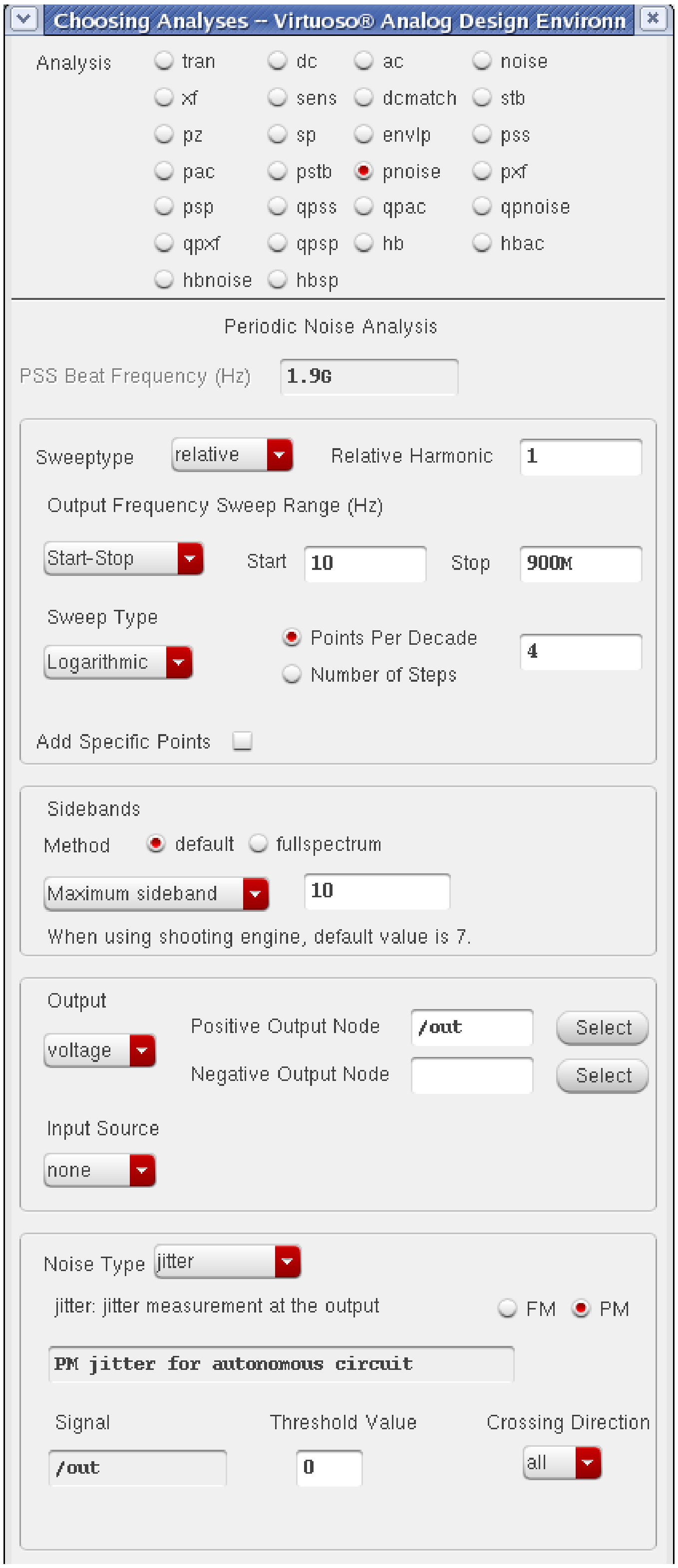
Noise Type = Jitter (PM)

This analysis gives an instantaneous measurement of jitter at a threshold voltage on the timedomain waveform. This is useful for measuring the jitter of an oscillator that is driving a digital component.

1. First, set up the pss analysis. For more information on this, see Overview of Periodic Steady-State (pss) Analysis on page 547.



2. Next set up the pnoise *Choosing Analyses* form, as shown below.



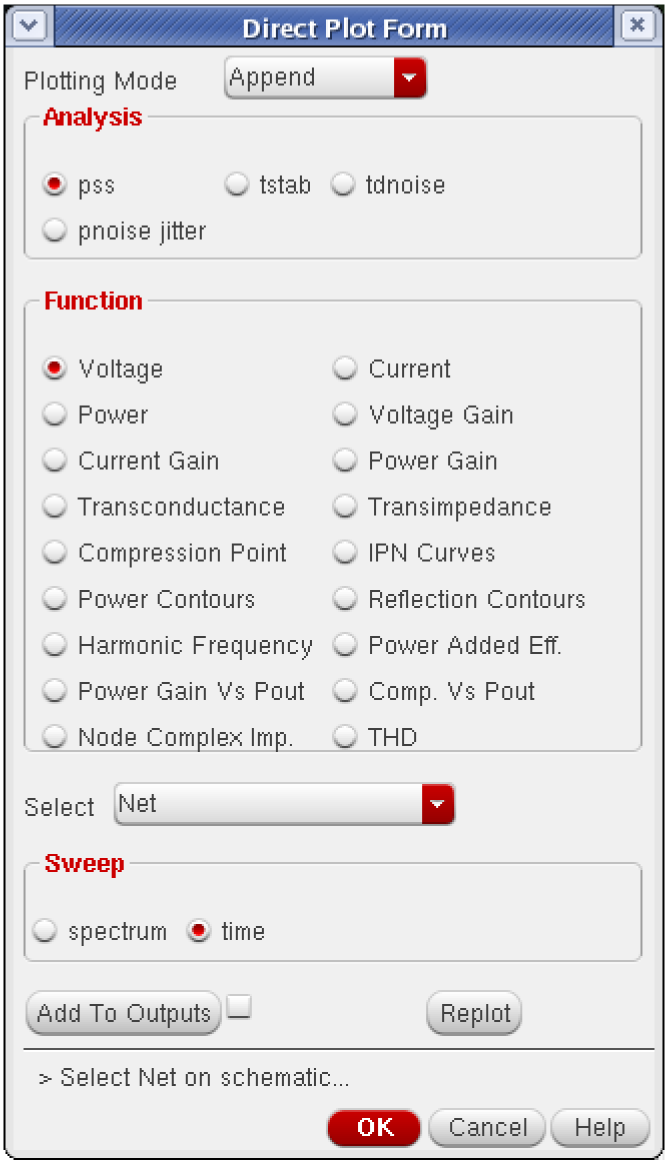
In the *Choosing Analyses* form:

- ❑ Select *jitter* from the *Noise Type* drop-down list.
- ❑ Select the *PM* radio button.
- ❑ Specify a threshold voltage for the time-domain waveform in the *Threshold Value* field.
- ❑ Specify a crossing direction.
- ❑ The stop frequency should be very close to half the oscillator frequency.
- ❑ The start frequency should be approximately the frequency where the phase noise reaches 0 dBc.

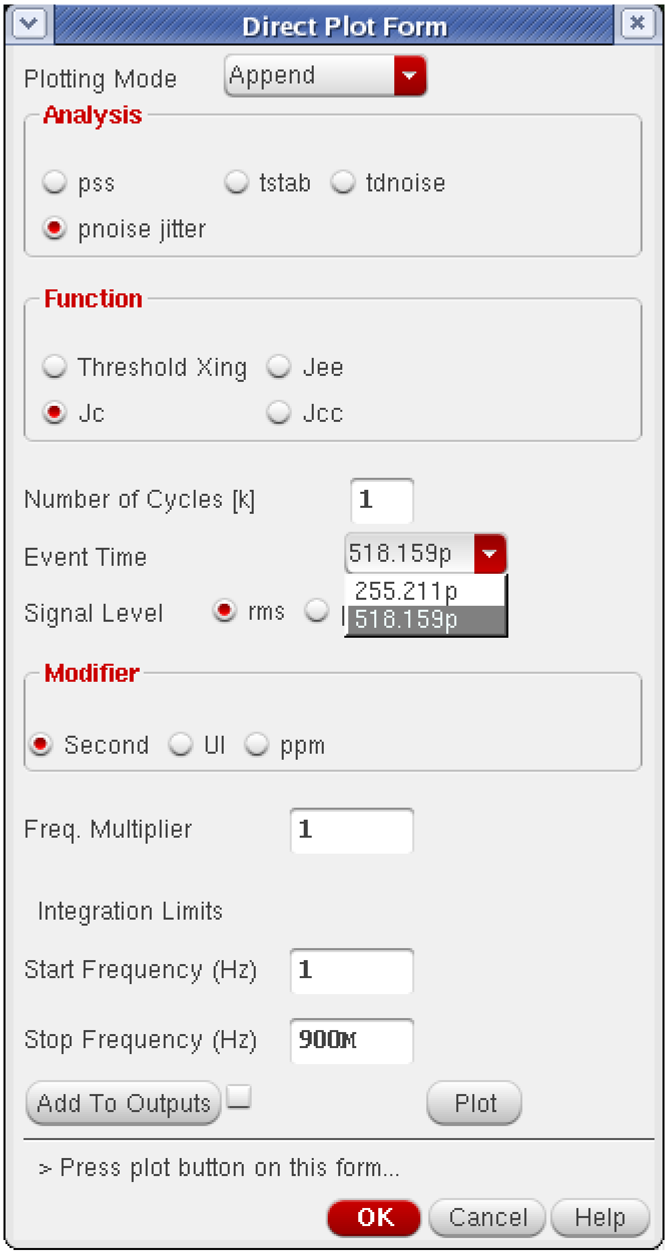
3. Run the simulation.

4. In the ADE, select *Results - Direct Plot - Main Form*.

On the *Direct Plot Form*, first plot the pss time-domain result.

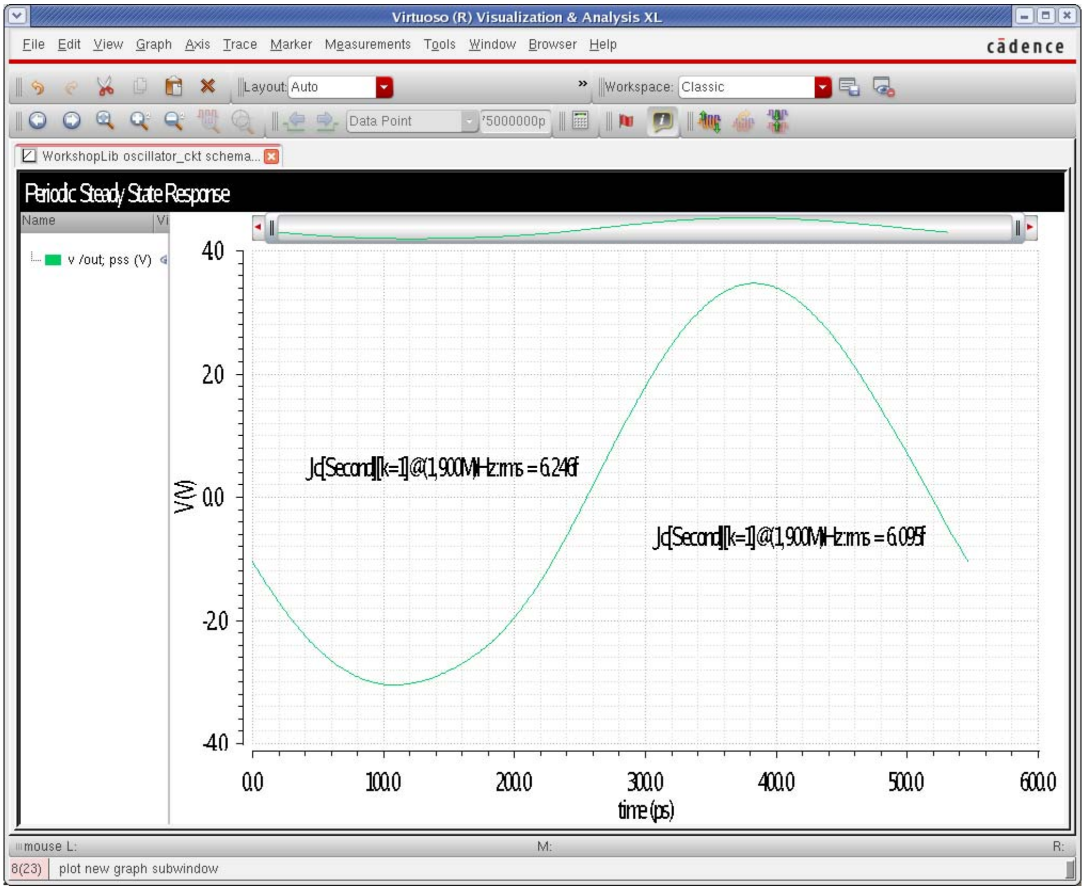


Now plot the cycle jitter (Jc) or the cycle-to-cycle jitter.



Note that there is a selection of event times if you selected *all* for *Crossing Direction*. This is the time on the waveform where the calculation was made.

Following are the calculations for the falling edge and the rising edge.



Noise Summary

The noise summary provides information about the devices that contribute noise to the output. This information is available any time an pnoise simulation completes.

In the ADE, select *Results - Print - Noise Summary*. If noise separation is enabled, there will be two choices at the very top. *pnoise_src* has information about the noise currents at the individual noise sources at the noise source. *pnoise(pmjitter)* has information about noise at the output of the circuit.

If you want the noise contributors at a single frequency, select *spot noise*. If you want to have the noise integrated over a frequency range, select *integrated noise*.

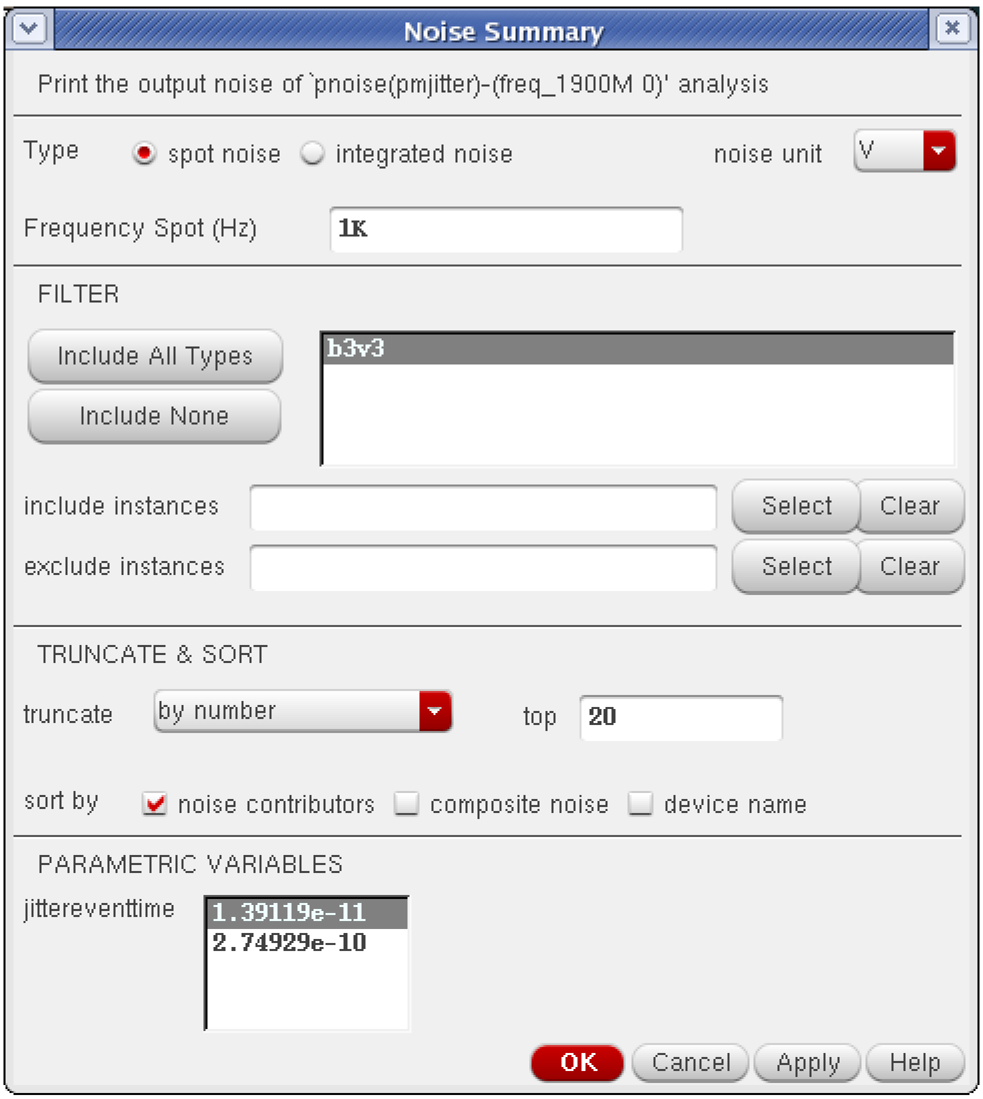
If you want noise in volts, set the *noise unit* to *V.*

Generally, *Include All Types* should be selected. If you just want the noise from specific types of noise generators, you can select them from the list.

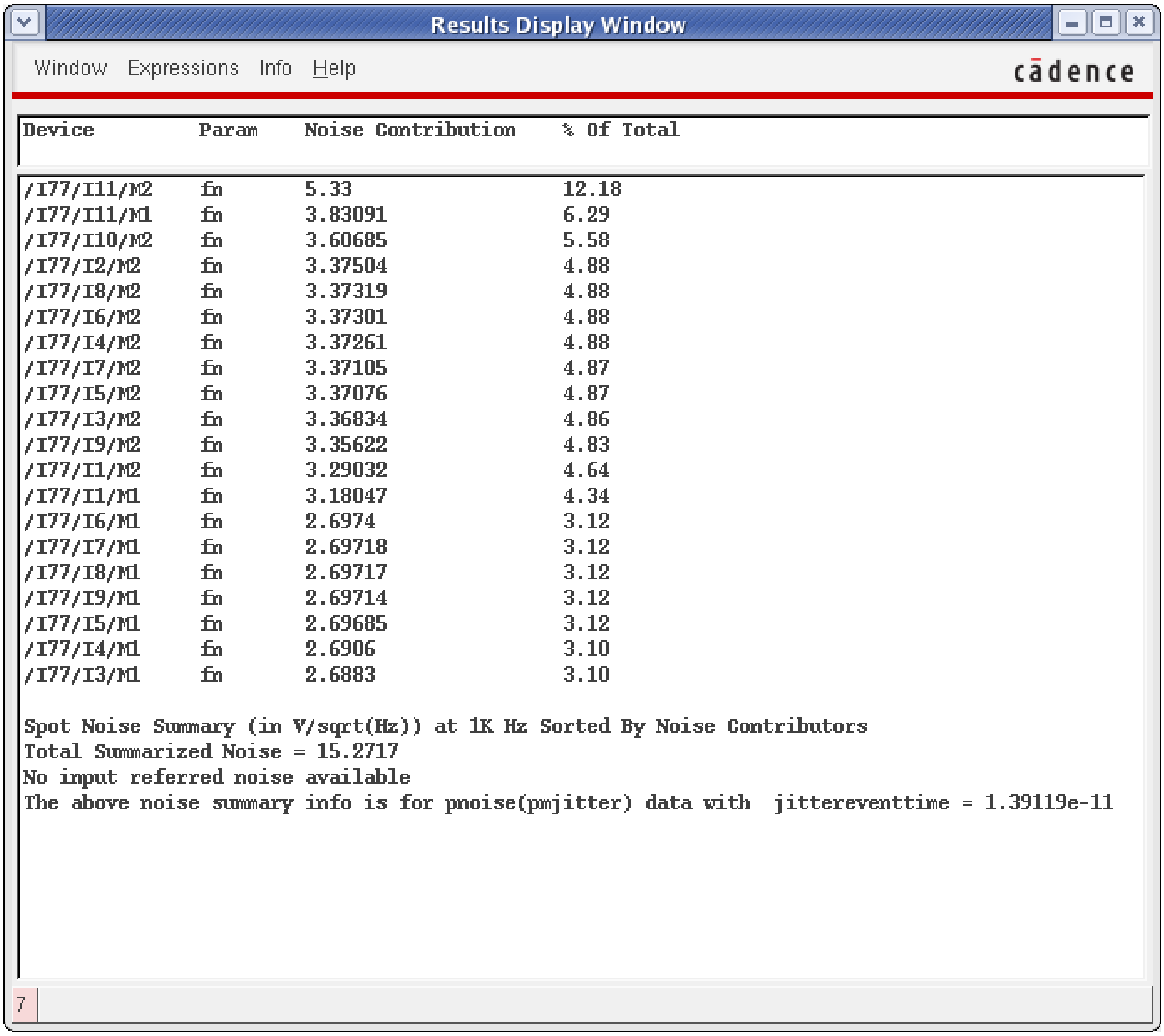
Truncate just applies to the list. The total input and output-referred noise at the bottom of the noise summary output always includes all the noise from everything in the circuit.

Jittereventtime - This is a list of the times in the PSS waveform where the noise calculations were made. To see whether the times are in the PSS waveform, plot the time-domain waveform from PSS. In many cases, only a single time is in the list.

The figure below is the noise summary window with the sort being noise contributors.



The output below shows the noise summary sorted by noise contributors. The largest noise contributor is on the top of the list.



Device	Param	Noise Contribution	% Of Total
/I77/I11/M2	fn	5.33	12.18
/I77/I11/M1	fn	3.83091	6.29
/I77/I10/M2	fn	3.60685	5.58
/I77/I2/M2	fn	3.37504	4.88
/I77/I8/M2	fn	3.37319	4.88
/I77/I6/M2	fn	3.37301	4.88
/I77/I4/M2	fn	3.37261	4.88
/I77/I7/M2	fn	3.37105	4.87
/I77/I5/M2	fn	3.37076	4.87
/I77/I3/M2	fn	3.36834	4.86
/I77/I9/M2	fn	3.35622	4.83
/I77/I1/M2	fn	3.29032	4.64
/I77/I1/M1	fn	3.18047	4.34
/I77/I6/M1	fn	2.6974	3.12
/I77/I7/M1	fn	2.69718	3.12
/I77/I8/M1	fn	2.69717	3.12
/I77/I9/M1	fn	2.69714	3.12
/I77/I5/M1	fn	2.69685	3.12
/I77/I4/M1	fn	2.6906	3.10
/I77/I3/M1	fn	2.6883	3.10

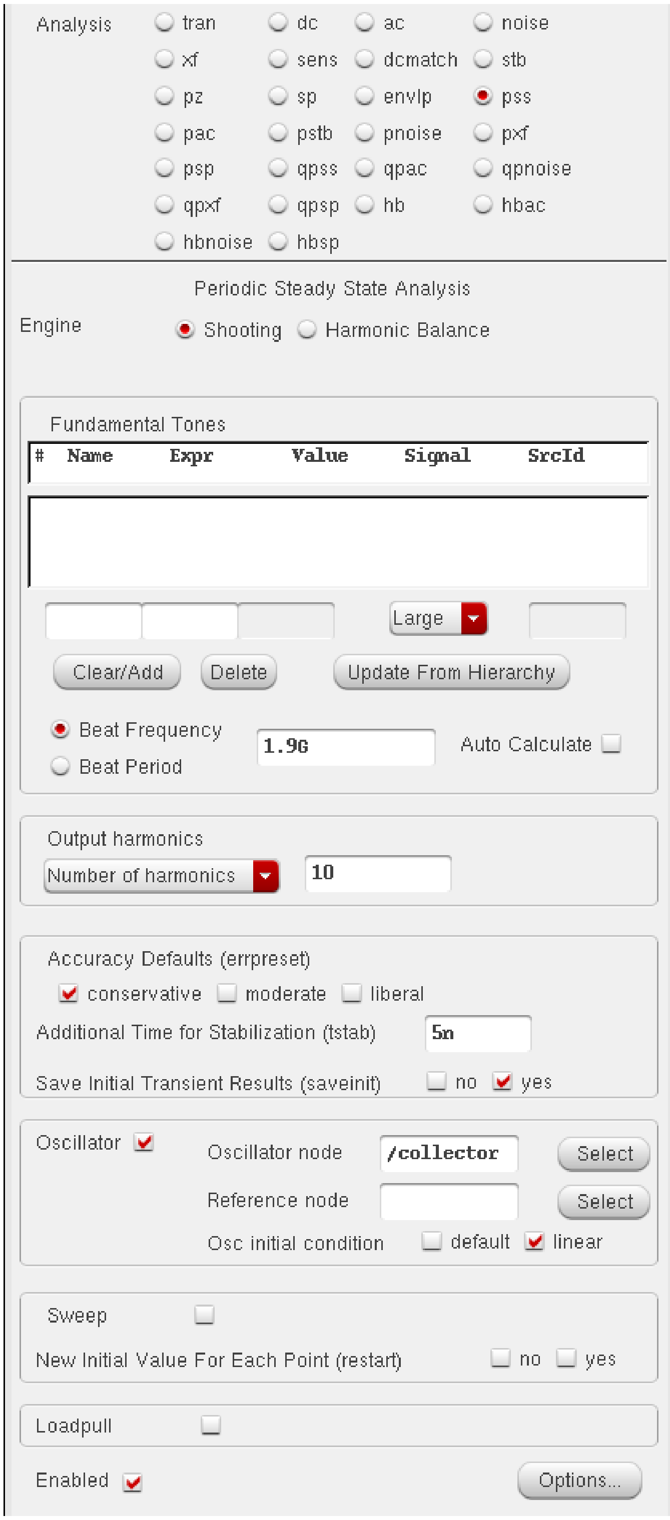
Spot Noise Summary (in V/sqrt(Hz)) at 1K Hz Sorted By Noise Contributors
Total Summarized Noise = 15.2717
No input referred noise available
The above noise summary info is for pnoise(pmjitter) data with jittereventtime = 1.39119e-11

For the abbreviations used in the Param column, refer to the table under the section Noise Summary on page 920.

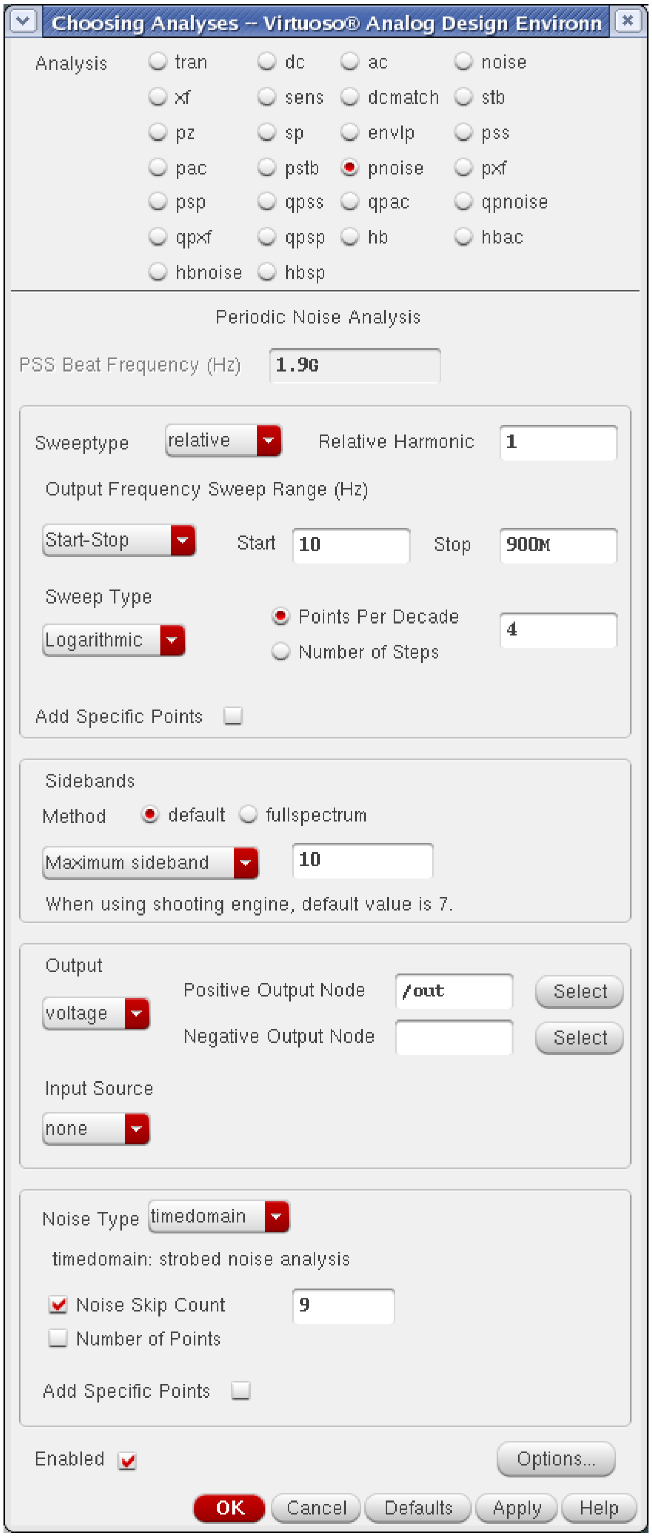
Noise Type = Timedomain

This analysis type provides a relative measurement of the noise at each time in the pss output waveform. Note that if you have a lot of harmonics and/or a large oversample factor, there will be many timepoints in the ifft in harmonic balance. In the shooting output, there are at least 200 timepoints. A method will be presented to reduce the number of pnoise points.

1. First, set up the pss analysis for the oscillator. For more information, see Overview of Periodic Steady-State (pss) Analysis on page 547.



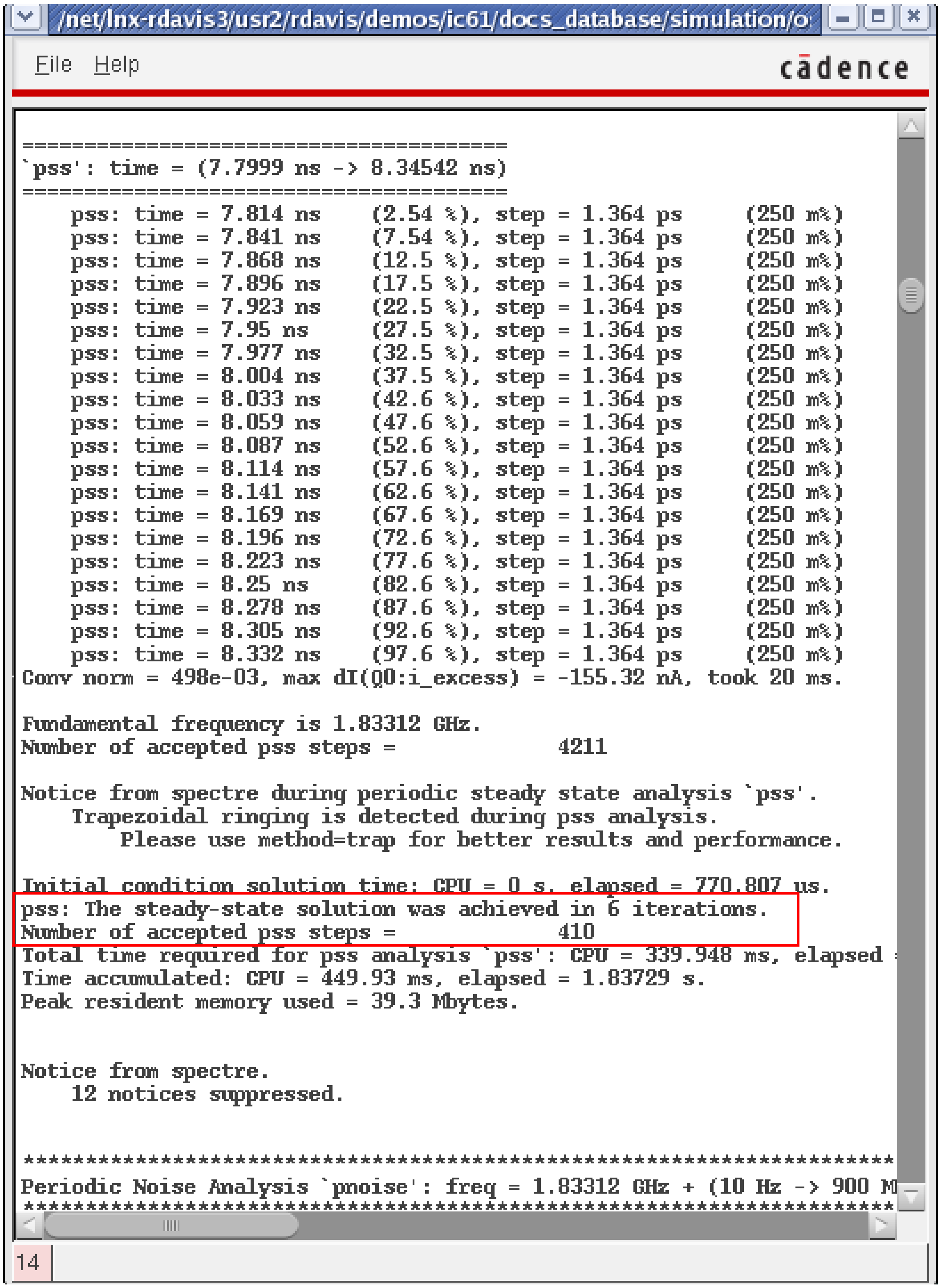
2. Next, set up the pnoise analysis, as shown below.



3. In the pnoise *Choosing Analyses* form:

- ❑ The stop frequency should be near one half the frequency of the oscillator.
- ❑ The start frequency should be near the point where the phase noise curve reads 0dBc.
- ❑ Select *timedomain* from the *Noise Type* drop-down list.
- ❑ The default *Noise Skip Count* is 0 (zero).

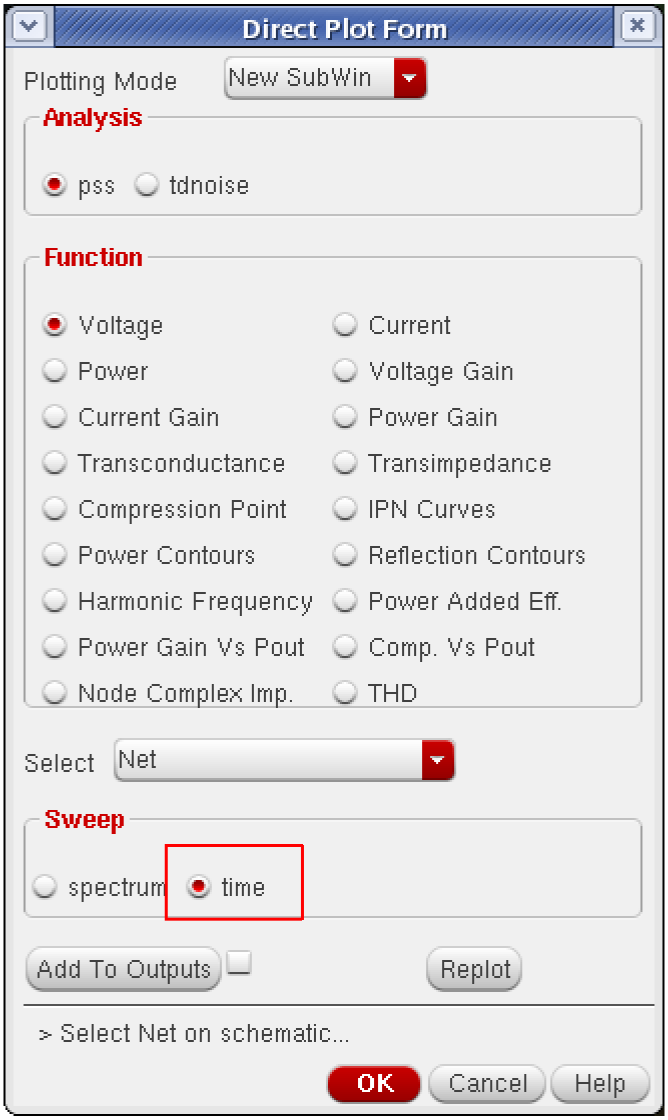
A pnoise run will always be run at time = 0. A noise skip count of 2 means run the timepoint at 0 seconds, then skip 2 pss timepoints, then run the next one, then skip 2, and so on. Pick an appropriate number for noise skip count so that the pnoise runtime is not excessive. In harmonic balance, there will be 2*number of harmonics*oversample timepoints. In shooting, first run pss by itself and then you can read from the Spectre output file how many timepoints there are.



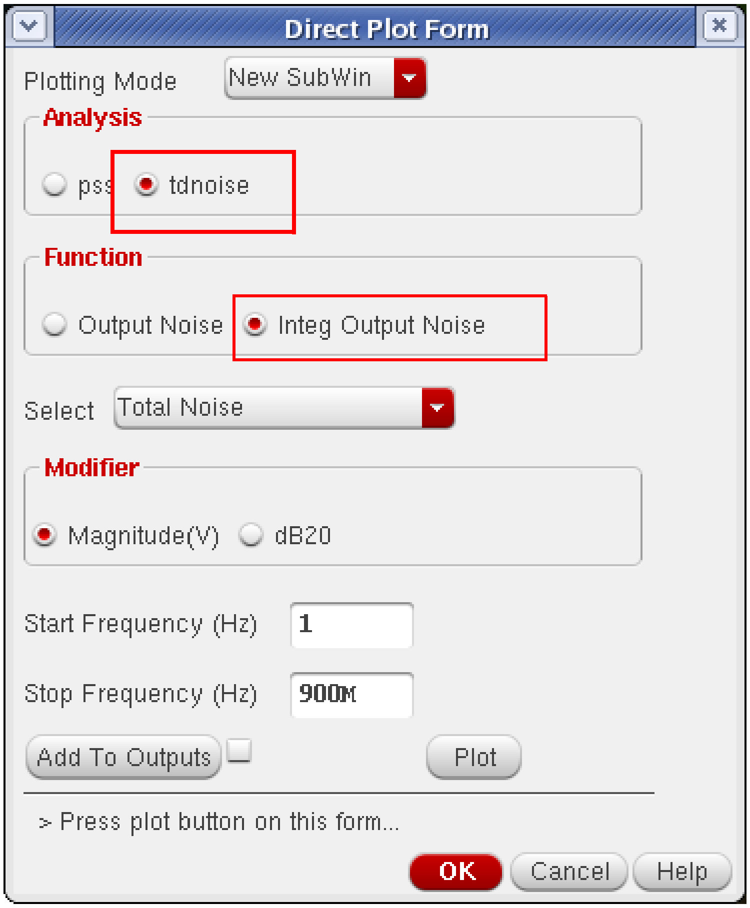
4. Run the simulation.

5. In ADE, select *Results - Direct Plot - Main Form*.

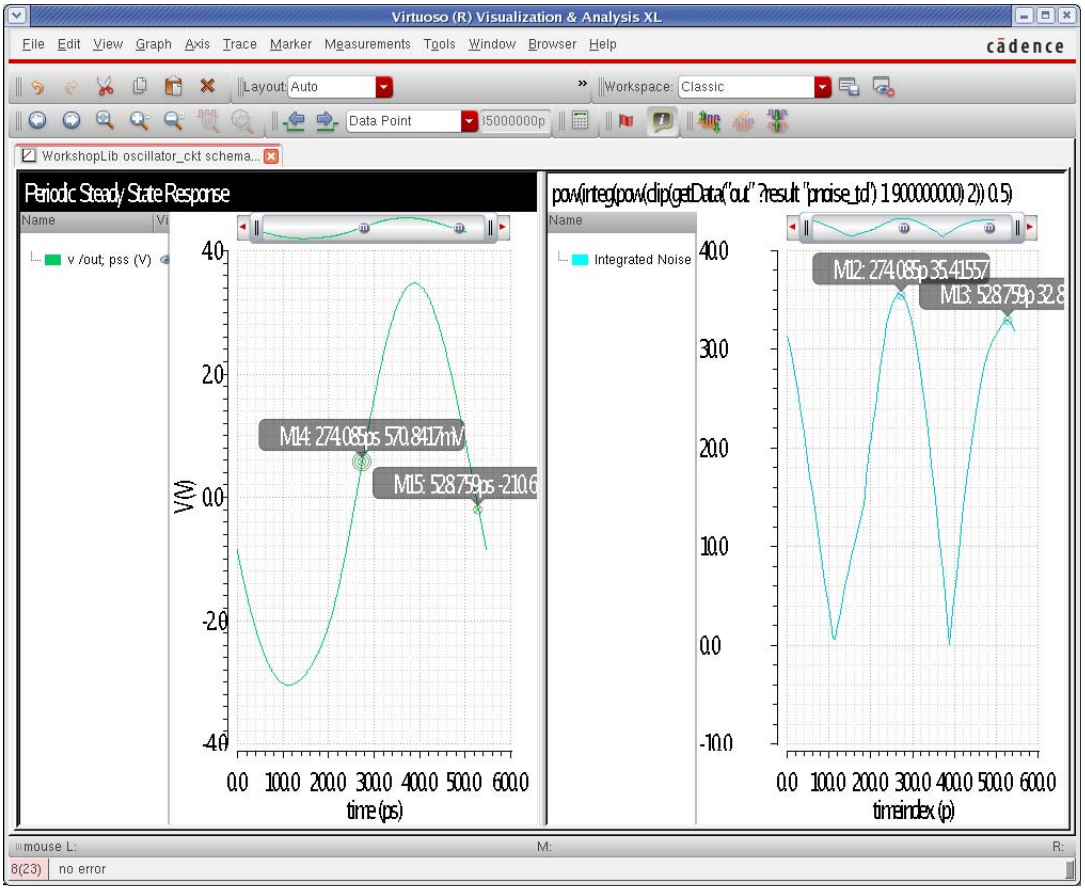
6. Plot the output waveform.



7. Now plot the integrated output noise.



The waveform tool displays both results. Markers have been placed at the largest noise points, and at the times for those maximums. Remember that this is a relative measurement for comparing the noise voltage at different points in time only. The noise is near zero at the valleys, and near the maximum at the peaks.



The noise is highest near the zero crossing, and lowest at the peaks and valleys of the waveform.

Periodic Transfer Function Analysis (PXF)

Small-signal Versus Large-Signal Analysis for Conversion Gain

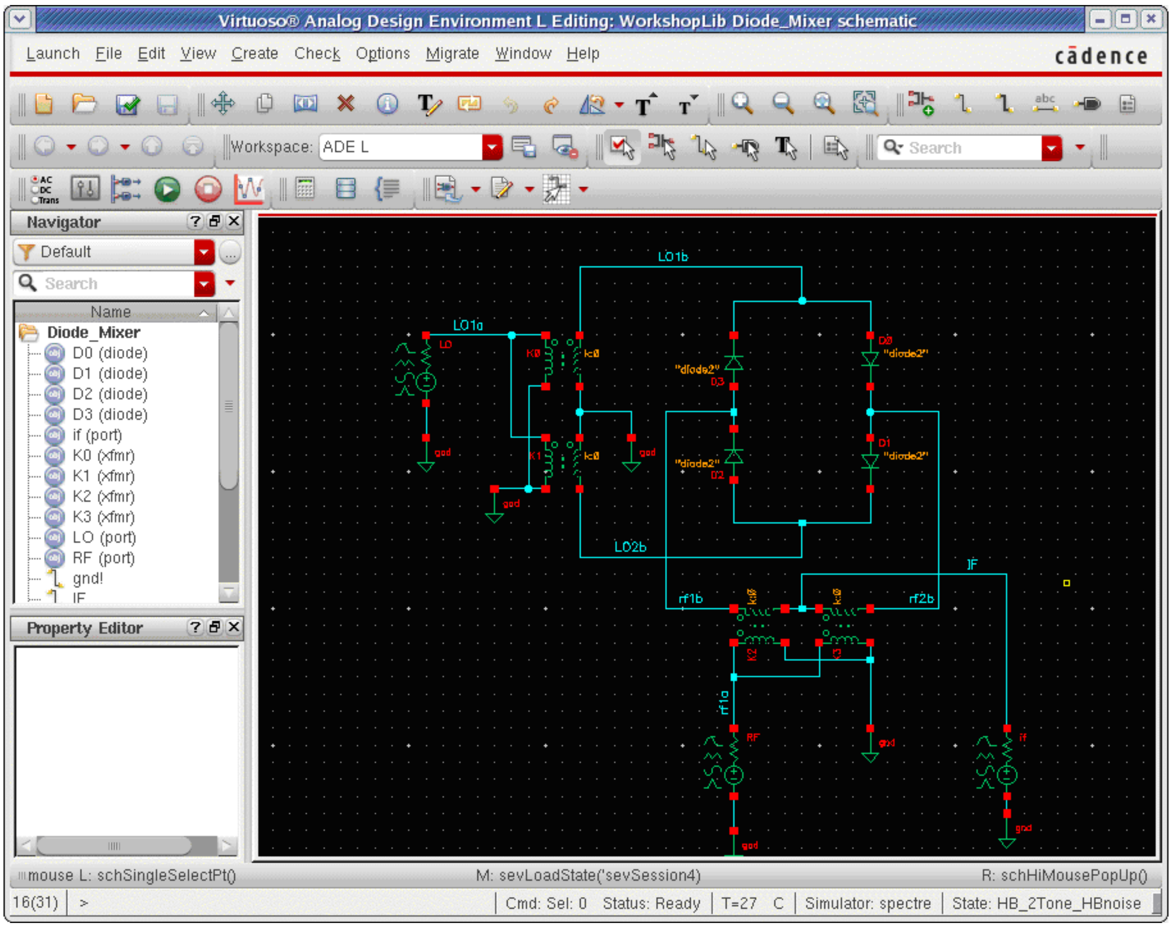
One way to measure conversion gain is to apply the LO and the RF signals in harmonic balance (hb) or quasi-periodic steady-state (qpss) analysis and calculate the full large-signal response at each RF input frequency, and then plot the output as a function of the sweep frequency. While this works, because all the large-signal effects and many harmonics are calculated, the required time for the simulation can be significant.

The alternative is to use a small-signal approach similar to the xf analysis. The xf analysis runs considerably faster than running the transient at multiple frequencies because it ignores the large-signal effects. In circuits with frequency translation, the xf analysis is unusable because the calculation is a linear calculation based on the DC bias point. In pxf analysis, instead of using the DC bias point, the pss solution with just the LO is used as the basis of the calculation. The large-signal analysis calculates the harmonics of the single input that causes the frequency translation to occur, (usually the LO signal or sample clock) and then pxf is run after the pss analysis to calculate the conversion gain. The conversion gain is calculated based on the nonlinearity in the pss solution. The overall time savings can be very significant.

In pac, we apply the LO signal (or the sample clock) in the pss analysis, and then drive the input with the small-signal pac analysis and measure the mixing products that come out. While this is very intuitive, the output display can be very confusing to read when the conversion gain near several pss harmonics is desired. Pxf changes the focus to the output. One output frequency at a time is considered. Pxf is a two step process. First, the input frequencies that cause the output frequency are calculated. For example, imagine a system with a 1GHz LO signal, and a 1MHz output signal. In PXF, we specify 1MHz as the output frequency. In this system, inputs at 1MHz, 999MHz, 1001MHz, 1999MHz, 2001MHz, and so on will cause 1MHz to be produced at the output. Next, the forward conversion gain from all those input frequencies is calculated. Since the input frequencies are different, the conversion gain is different for every input frequency. Using pxf, you can measure the conversion gain near a number of LO harmonics (or clock harmonics) at the same time.

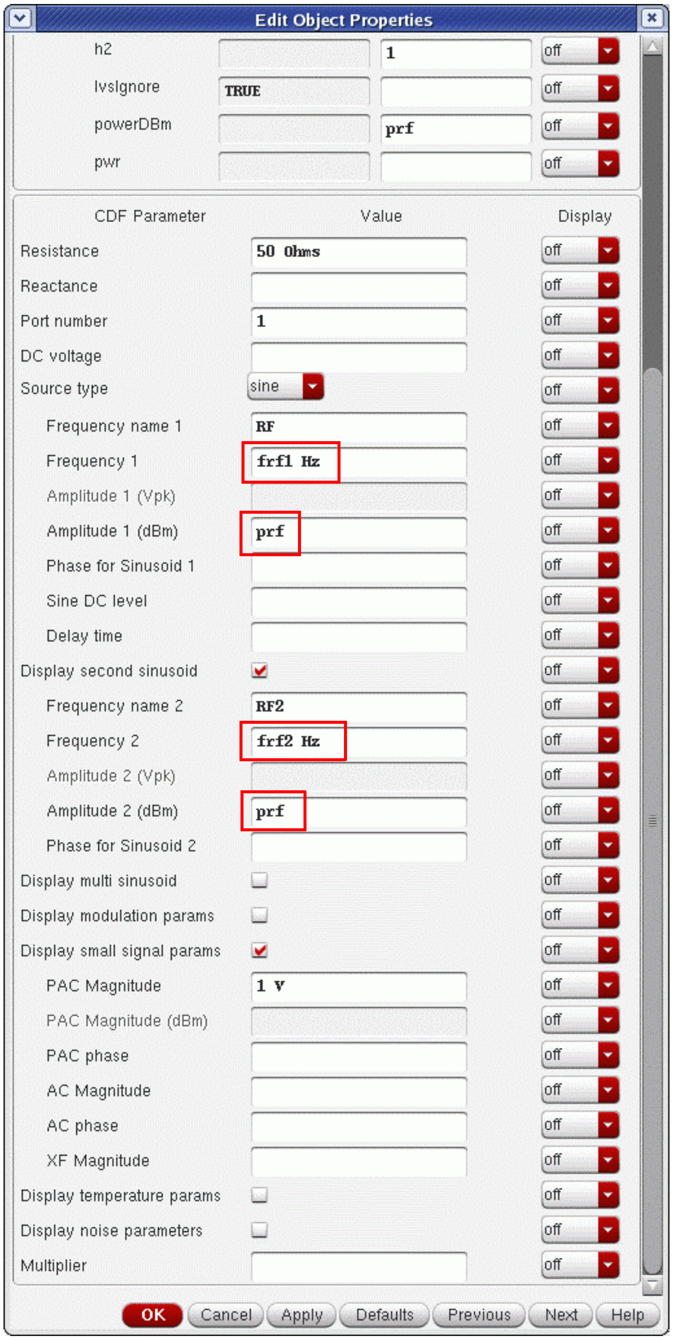
Example: Conversion Gain (Down conversion)

Consider the double-balanced diode mixer shown below.



The RF input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. This allows changes in frequency or amplitude from the ADE environment without the need to change the schematic itself.

The property list is shown below.

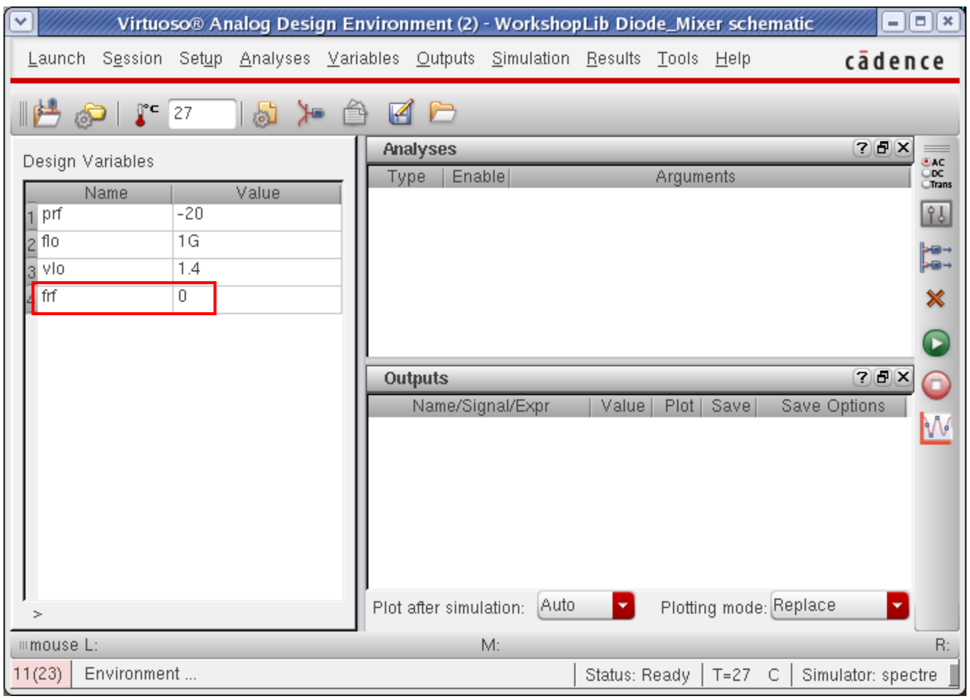


The easiest way to get the variables list into the ADE is to select *Variables - Copy From Cellview*. To set the values, select the value field to the right of the variable, type in the desired value, and press *Enter*. When you have set the variables, if you want to add the

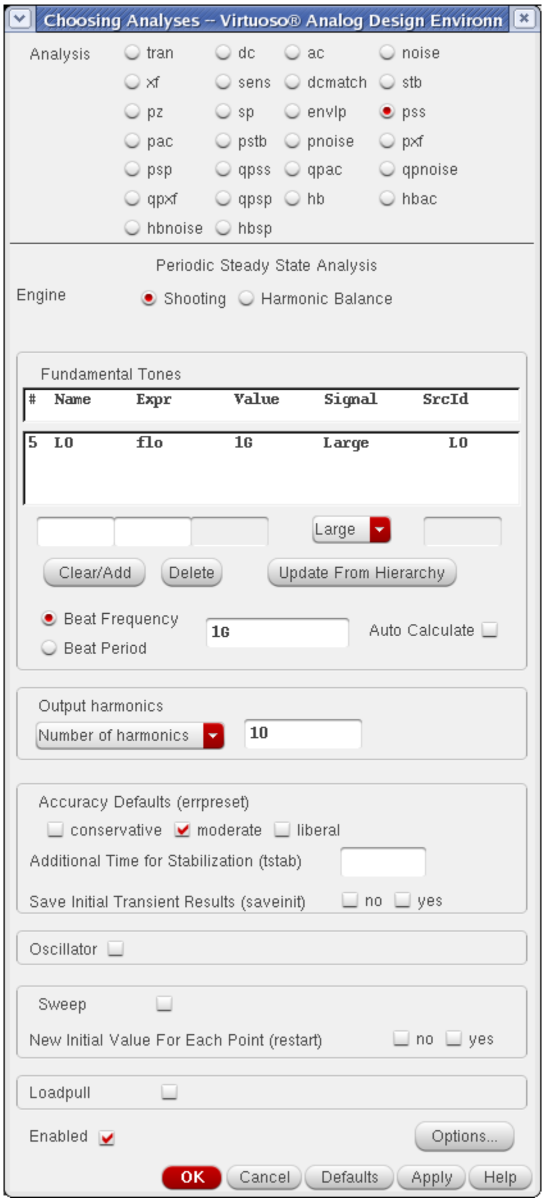
values back into the cellview, in the ADE window, select *Variables - Copy To Cellview*, and then perform a *Check and Save* in the schematic.

In this example, the LO frequency is 1GHz and its amplitude is 1.4 volts peak. This is +13 dBm, which is a common drive level for diode mixers. The design RF input frequency is 1.1GHz, but it is disabled because the amplitude is set to 0 (zero). Setting either the frequency or the amplitude to zero disables the production of that waveform from the port.

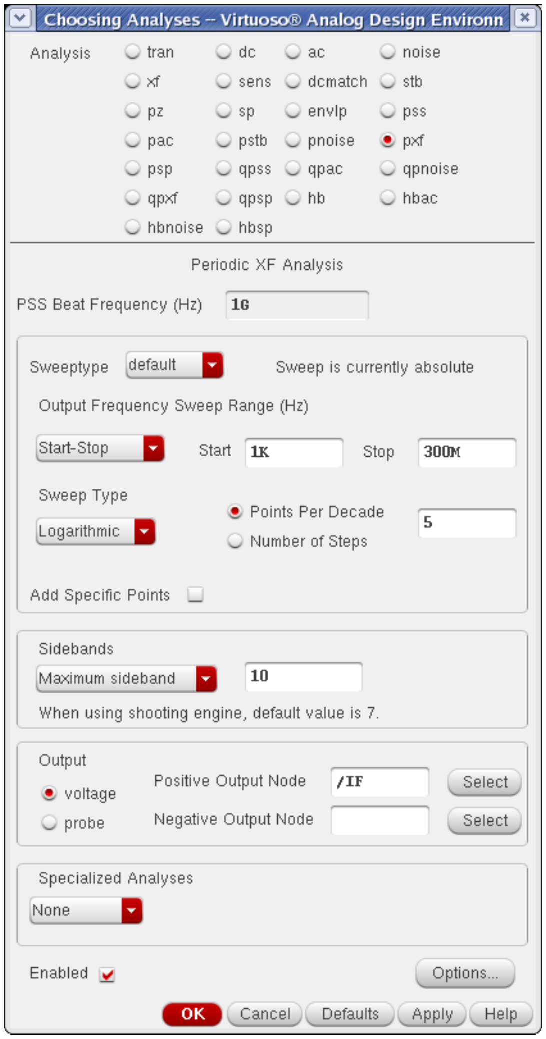
The basic strategy is to apply the signal that causes the frequency translation in the pss analysis, and then measure the forward conversion gain using pxf analysis.



The pss analysis needs to be set up first. The example below shows a single input at 1GHz with 10 harmonics. For more information on setting up the pss analysis, see the pss section at the beginning of this chapter.

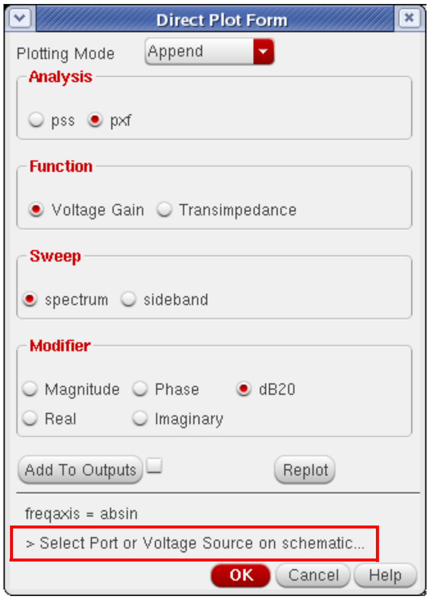


Next, the pxf analysis needs to be set up. Like xf, the frequency range specified in the *Choosing Analyses* form is the **output** frequency range. *Maximum sideband* in this example is set to the number of harmonics in the pss analysis so that all the conversion gain terms from all the pss harmonics are present in the output of pxf.



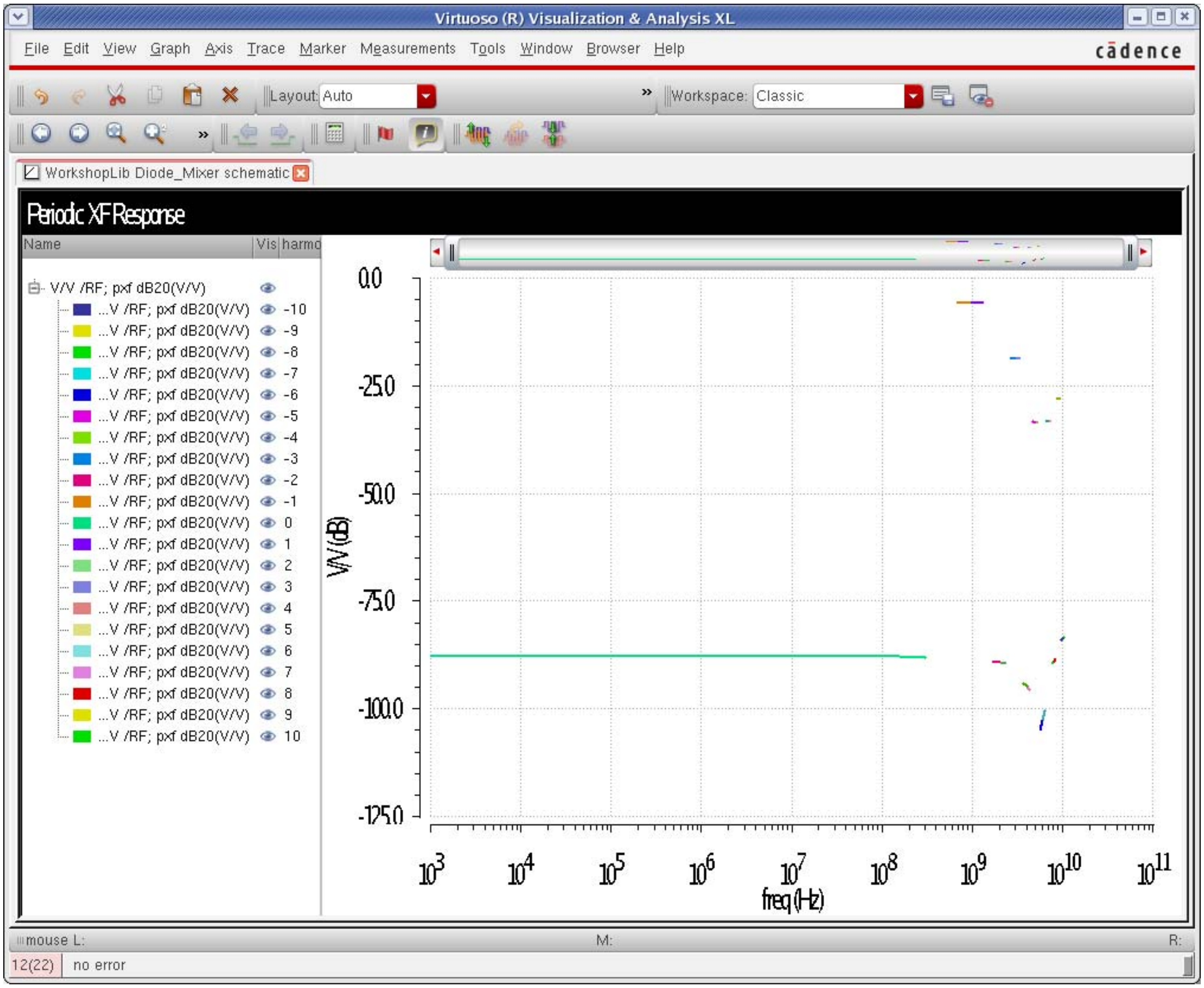
Once the analyses are set up, the simulation can be run. When the simulation finishes, select *Results - Direct Plot - Main Form* in the ADE window.

The example below shows a typical *Direct Plot Form*.

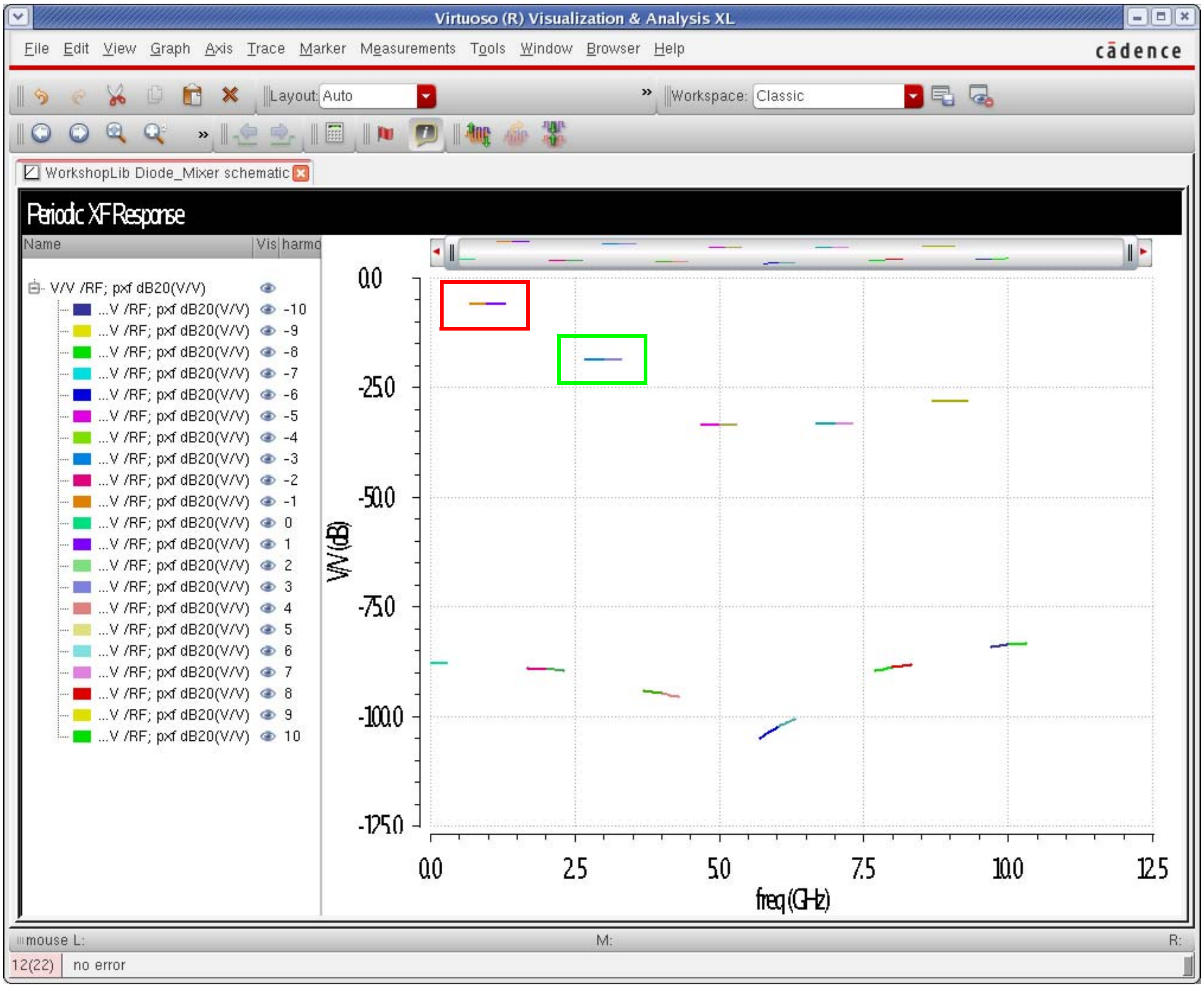


The pxf *Direct Plot Form* is a bit non-intuitive. The *Choosing Analyses* form specifies the output frequency range and the output node. Pxf calculates the transfer functions from all the independent sources in the circuit to that output node. Therefore, to plot, you select the source in the circuit that you want the forward conversion transfer functions to be displayed. When the source in the schematic is selected, the waveform is displayed.

Note that all the short line segments are the forward conversion gain from the selected source (the input source in this case) to the output. Because the sweep was a log sweep, the data is displayed on a log plot.



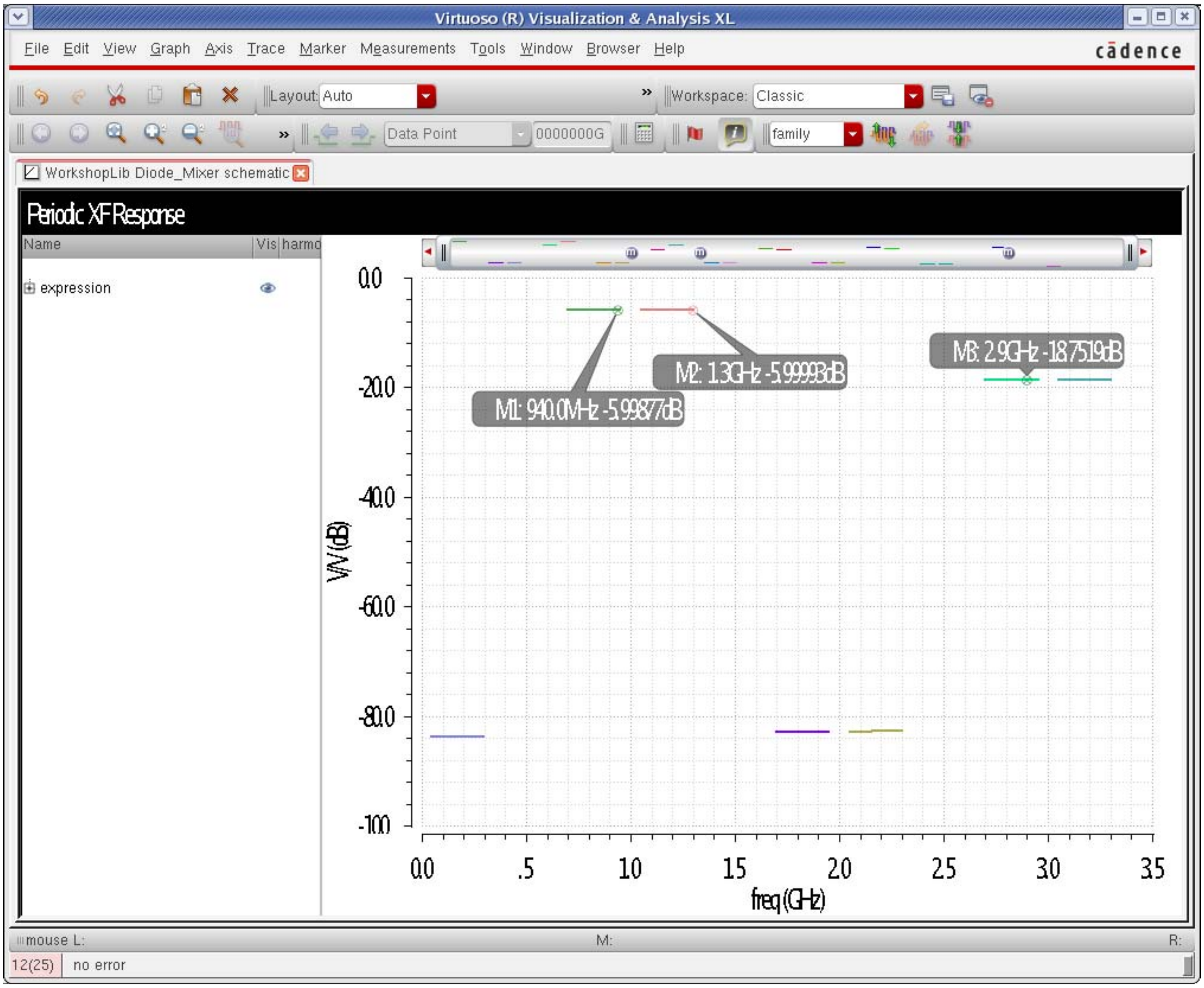
To make the plot more useful, set the X-Axis to linear. Move the mouse cursor over one of the numbers on the X Axis, click the right mouse button, and deselect *Log Scale*.



The LO frequency is 1GHz. To measure the conversion gain, go to the **input** frequency on the X Axis, and read the gain from the display. Note that the gain above and below 1GHz is just less than -5dB. The rectangle in red above shows where to look. The gain near 3GHz is just above -20dB. The green rectangle above show where to look. The gain near 5GHz is about negative 35dB. The gain near the even harmonics is quite small.

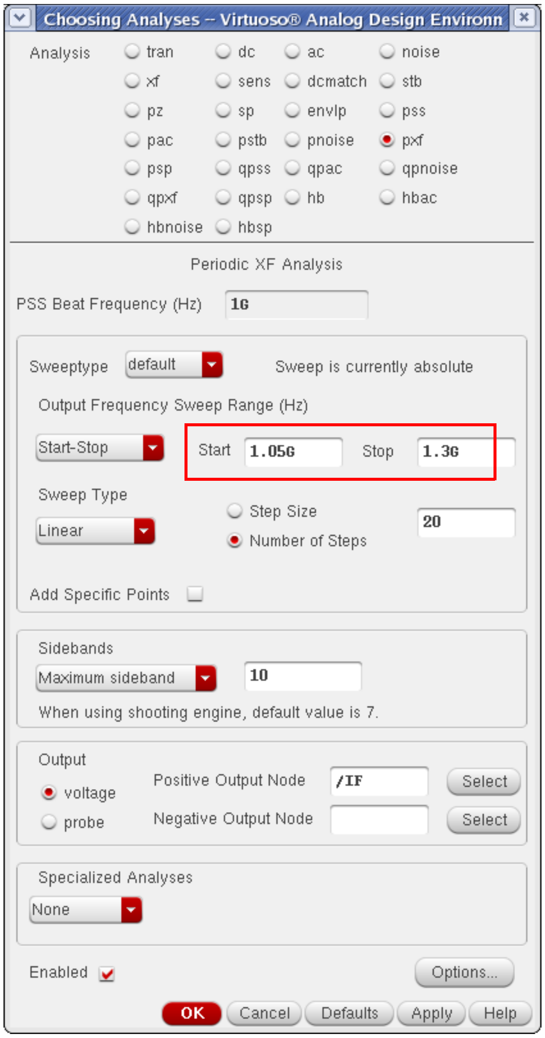
Now position the markers at the desired input frequency for the conversion gain measurement. To do this, move the mouse cursor near the trace at the desired input frequency, and type `m`.

As shown in the figure below, the conversion gain is about -6dB above and below the 1GHz LO frequency. This is very common for diode mixers.

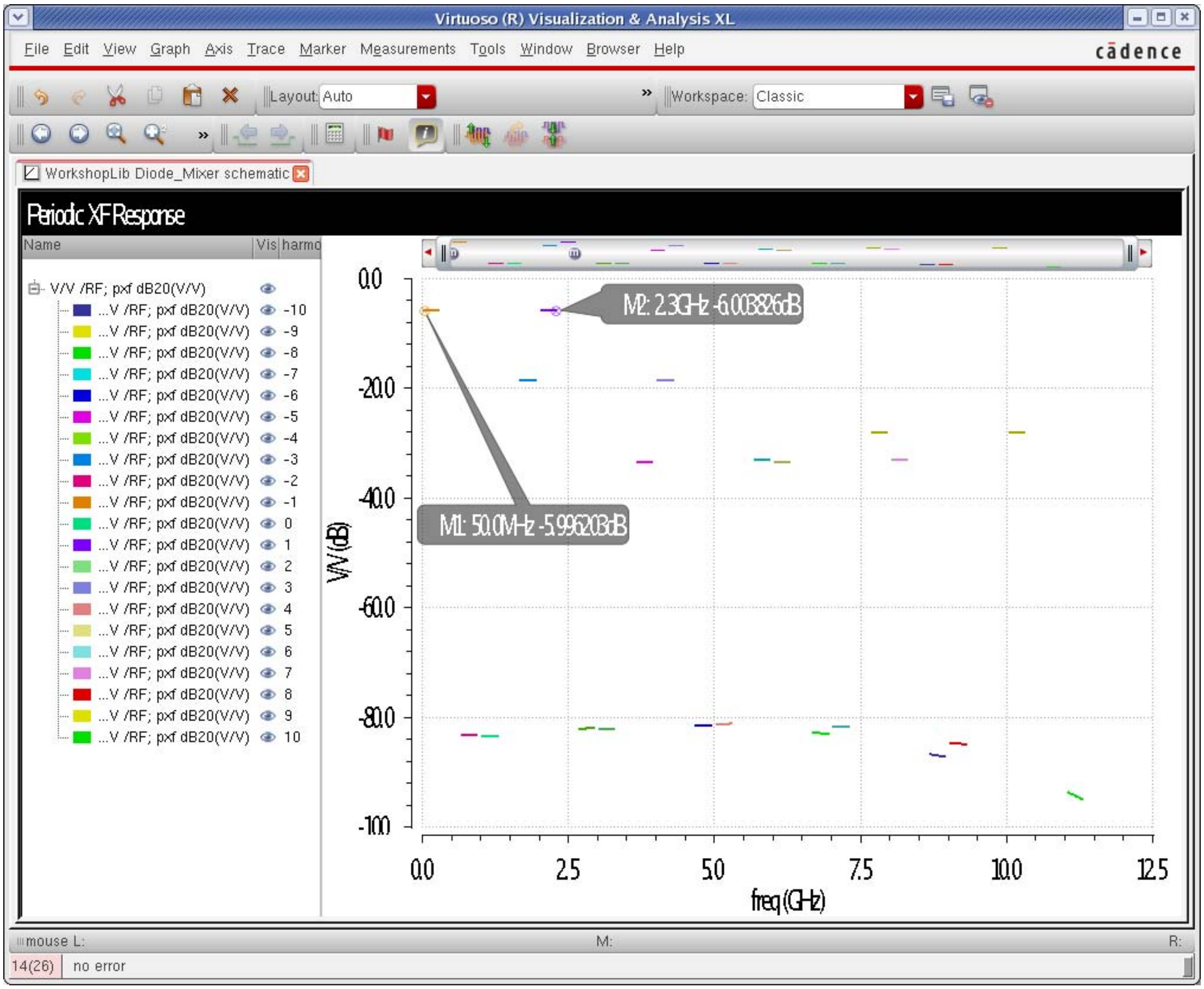


Example: Conversion Gain (Up Conversion)

The process is similar for an up-conversion mixer. In this case, the output frequency is just above the first harmonic of the pss analysis, as shown below.



The conversion gain plot is shown below. Note that the gain is very similar from near 0 Hz and near the image at just above 2GHz.



Overview of Simulation Capabilities

The primary application of pxf is to measure conversion gain. Pxf is a small-signal analysis. It calculates the small-signal conversion gain. If you need a large-signal measurement of conversion gain, then use a large-signal analysis like harmonic balance (hb) or quasi-periodic steady-state (qpss) analysis. Pxf measures gain in a way that requires some thought for most engineers who are accustomed to connecting an input, driving the system, and measuring what comes out. This is the idea of pac, where we capture the large-signal operating point in pss, and then drive a small-signal input, and measure the mixing products that are produced. If you think of this methodology, to measure the conversion gain around several harmonics usually requires a separate simulation near each harmonic.

Pxf was created to allow the measurement of the conversion gain transfer functions near all the pss harmonics in the same simulation run. In pac, we have one input frequency at a time, and output mixing products at multiple output frequencies are calculated. In pxf, we focus on one output mixing product at a time, and measure the input frequencies that cause that output frequency, and measure the forward conversion gain from all those input frequencies to the single output frequency. Multiple output frequencies are usually analyzed in a pxf analysis. Unlike pac where the output can be seen at every node in the circuit, pxf calculates the gain only from all the independent sources in the circuit. Therefore, to measure a conversion gain, you select the source instance from the *Direct Plot Form* instead of selecting a net.

For example, imagine a mixer with an LO frequency of 1GHz. Now, imagine the down conversion case where the output is 1MHz. Given the LO frequency, inputs at 1MHz, 999MHz, 1.001GHz, 1.999GHz, 2.001GHz, 2.999GHz, 3.01GHz, and so on cause an output at 1MHz. Depending on the circuit, all those different input to output paths have different gain from each input frequency to the output at 1MHz. In pxf, you specify the highest pss harmonic to calculate the transfer functions from in the maximum sideband field in the *Choosing Analyses* form. All these different input frequencies are called sidebands in pxf.

Pxf has three different modes, which are set by the selection in the *Specialized Analysis* section of the *Choosing Analyses* form. *Specialized Analysis = None* measures the conversion gain that occurs given the nonlinearity created by large-signal in the circuit. *Modulated* allows the measurement of AM to PM conversion for circuits, and is useful for measuring the power supply ripple to output phase transfer function. *Sampled* is seldom used, and supplies a measurement of the instantaneous gain when the gain is repeatedly measured at a single phase (point in time) of the periodic large signal.

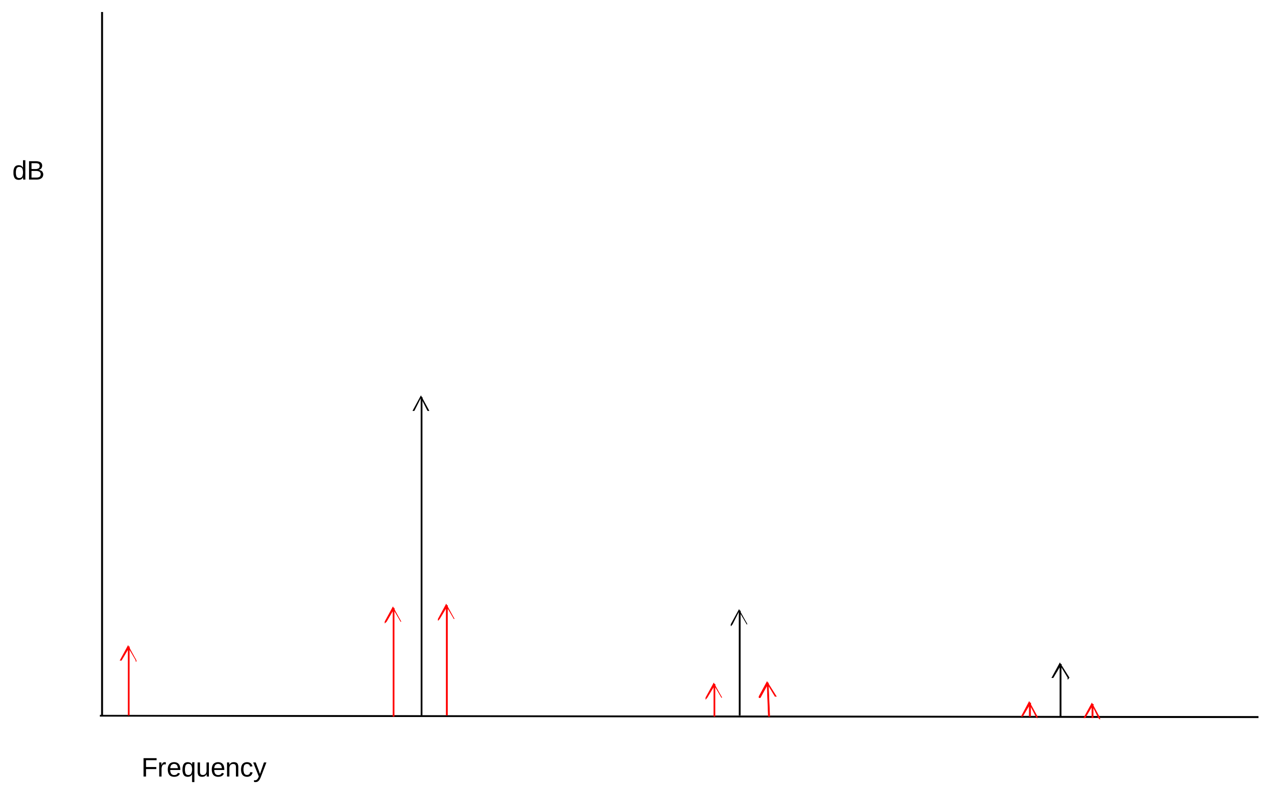
PXF used with Harmonic Balance PSS

When harmonic balance is used for the pss analysis, the harmonics of the circuit are calculated directly by the harmonic balance engine.

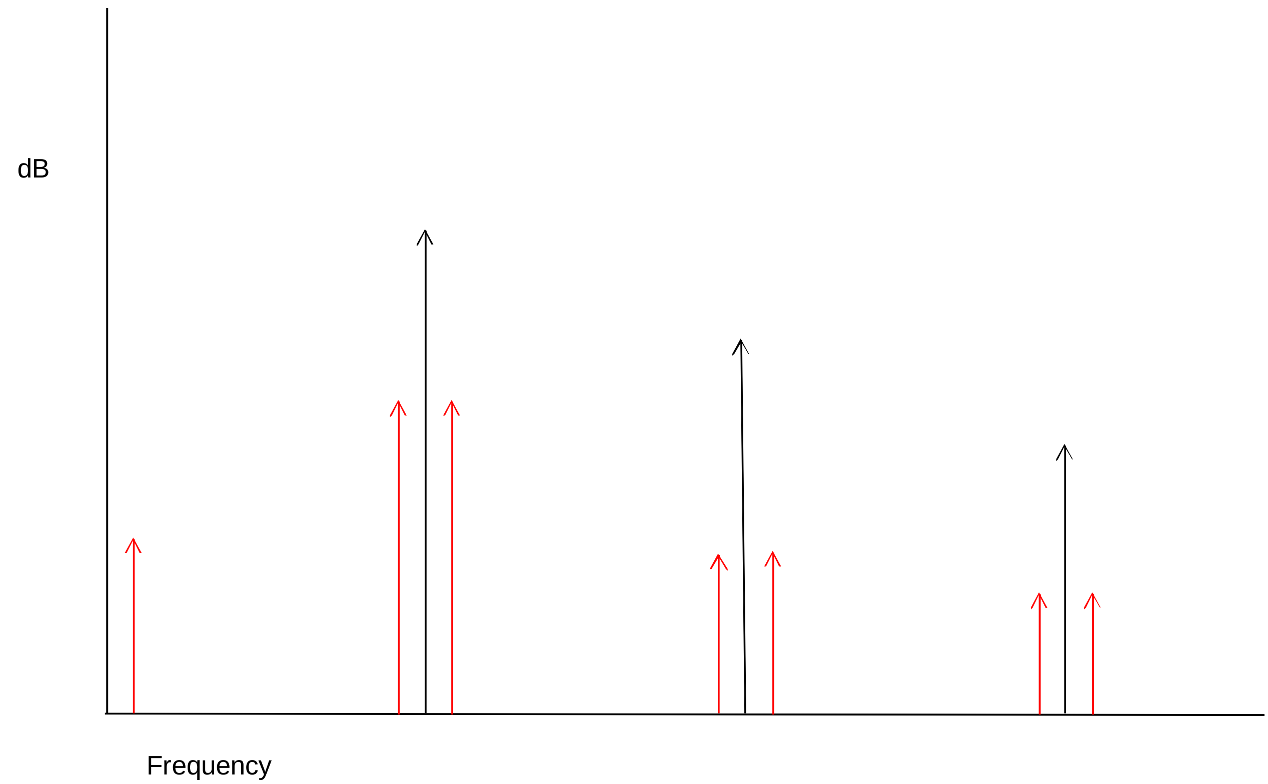
Think of a perfectly linear circuit with a single output frequency to be analyzed. Because the circuit is perfectly linear, in the pss analysis when harmonic balance is used, the frequency domain solution only has the harmonic at the input frequency at every net in the circuit. No other harmonics are created because the system is perfectly linear. Because the circuit is linear, a gain function exists for the case where there is no frequency translation. Because the circuit is perfectly linear, no frequency translation can occur. Therefore, all the conversion gain terms have zero gain.

Now imagine a real circuit with a single small input amplitude signal at 1GHz. In the frequency domain with harmonic balance selected in pss, 1GHz is present along with 2GHz, 3GHz and the other higher order harmonics. Because the input signal is small, the signals at the harmonics of 1GHz in pss are very small. If 100MHz is set as the output frequency, the conversion gain is small from 900MHz and 1.1GHz. Since the higher harmonics are even smaller than the amplitude of 1GHz, the conversion gain form 1.9 and 2.1HGHz along with the higher harmonics plus and minus 100MHz are also very small.

The conversion gain might look like the figure below. Only the first three harmonics are shown. The black harmonics are the pss harmonics, and the red mixing products are pxf conversion gain measurements, which are called sidebands.



Now imagine that the 1GHz signal is larger. If the input goes up 10dB, the first harmonic goes up 10dB, In the frequency domain, if harmonic balance is selected in pss, the second harmonic would come up 20dB, the third harmonic would come up 30dB, and so on. If the conversion gain is calculated to 100MHz, the conversion gain is larger because all the harmonics have become larger. The circuit is more nonlinear. This is illustrated below. The black harmonics are the pss harmonics, and the red conversion gain calculations are pxf input frequencies with conversion gain measurements, which are called sidebands.

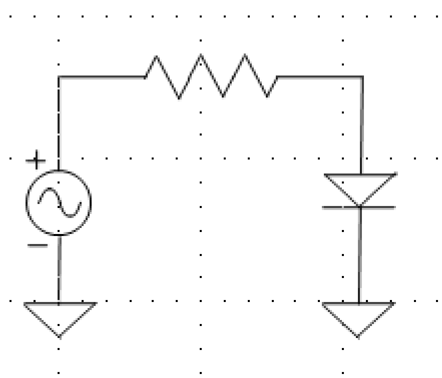


Thus, the harmonic content from pss harmonic balance is a measure of the nonlinearity of the circuit, and forms the basis of the calculation for the conversion gain in pxf.

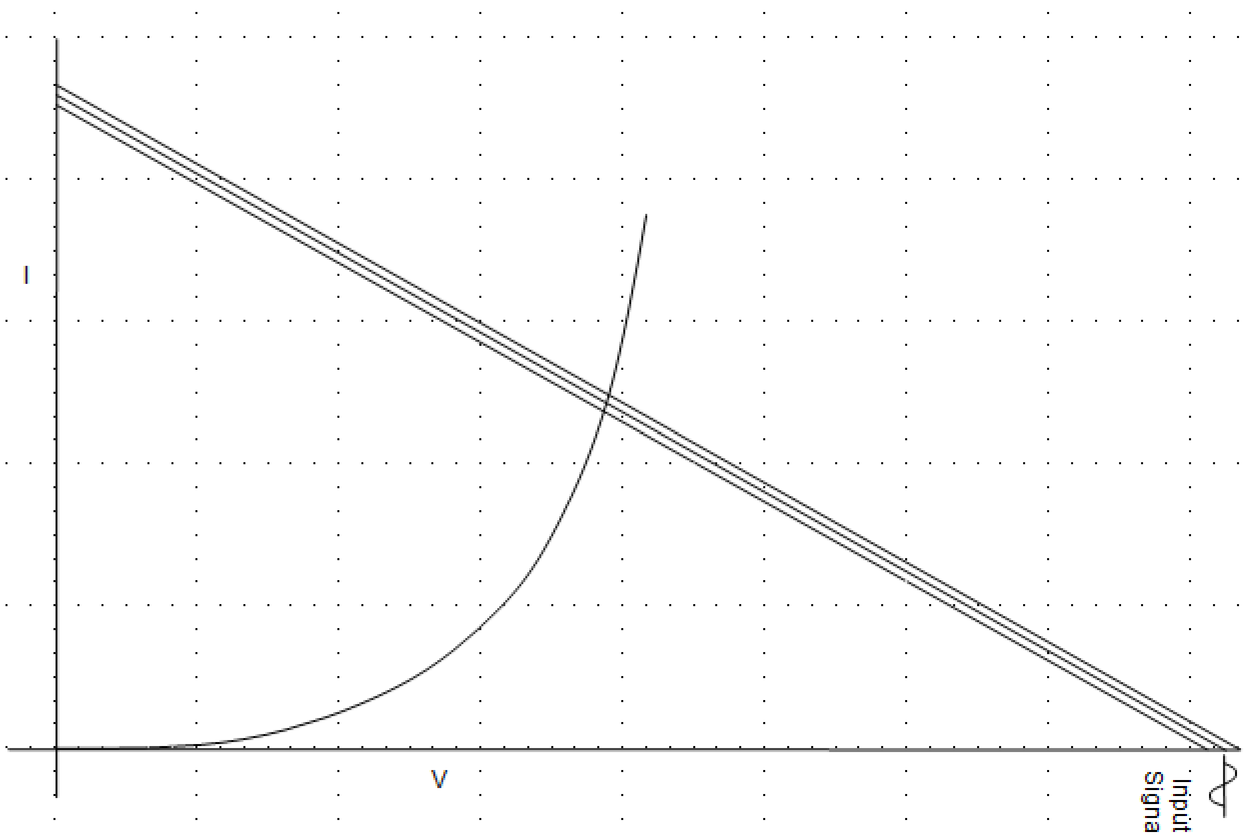
PXF used with Shooting PSS

Shooting is a variation of the transient analysis where the individual timepoints are calculated, and the entire waveform is iterated to calculate the steady-state waveform. The transient algorithm works by using all the equations at all the nodes in the circuit where the assumption is made that the sum of the currents is zero at every node and at every timepoint. Ohm's law is observed, and the equations look like `I=V*G` where `G` is the conductance. In general, the matrix form of the equations is constructed, so the system of simultaneous equations becomes `[G]*[V]=[I]` where `[G]` is a square matrix, and `[V]` and `[I]` are column matrices. To solve for [`V]`, the `[G]` matrix is inverted and this matrix multiplies the `[I]` matrix to solve for all the voltages on all the nodes. At the end of the pss analysis, all the solution matrices are saved as data to be passed to the small-signal analyses. The result is that the conductance is calculated and saved for every nonlinear component in the circuit at every timepoint.

For the following circuit:

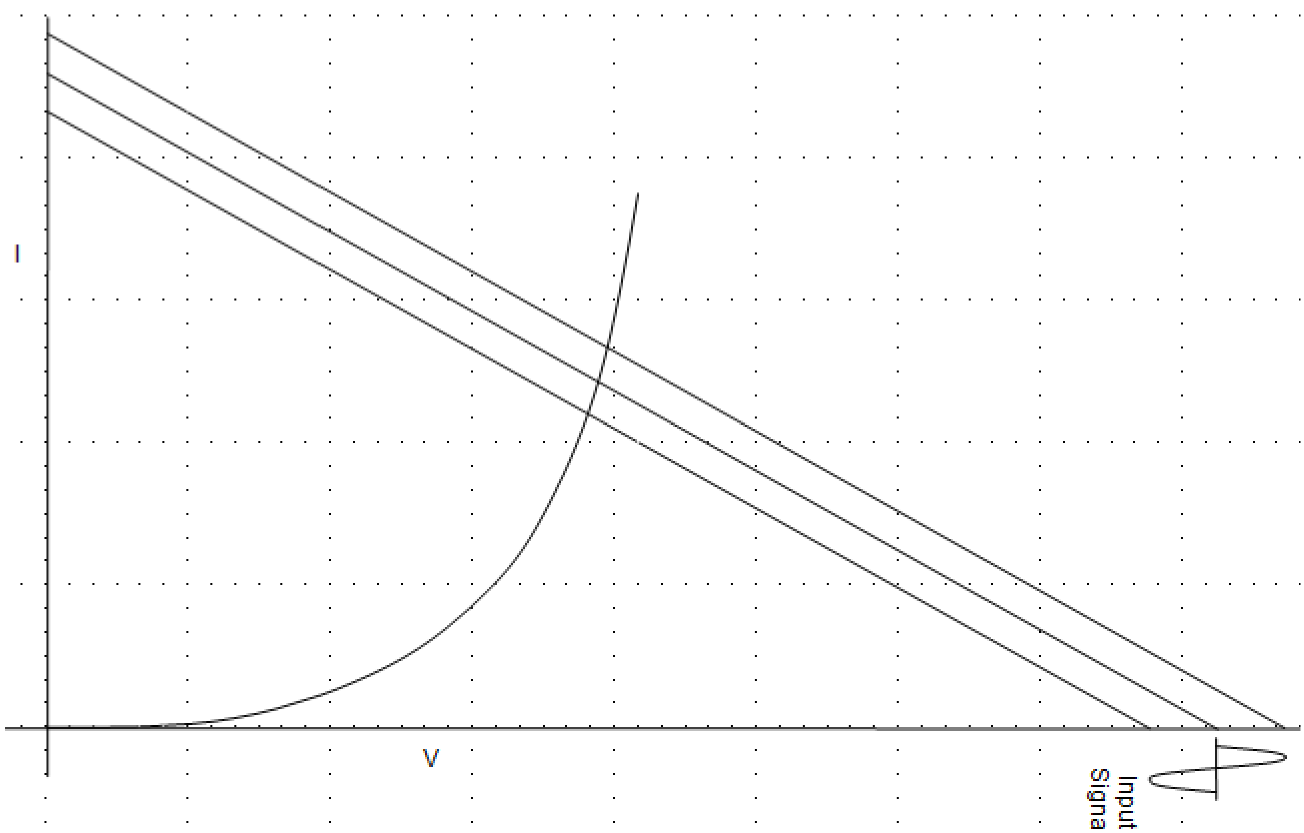


Below is the I-versus-V curve with three different values for the input voltage. The center, positive peak of the input, and the negative peak of the input are shown. The input amplitude is small.



In the above figure, the amplitude of the 1 GHz signal causes small variations in the slope of the tangent of the nonlinear curve as the input signal is varied in amplitude. The slope is the conductance. Because the circuit has a nonlinear device, the conductance of the device varies slightly from timepoint to timepoint in the pss analysis. Because the input signal at 1.0GHz is small, the variation in conductance from timepoint to timepoint is small. The slope changes just a little bit as the input signal is moved over its range.

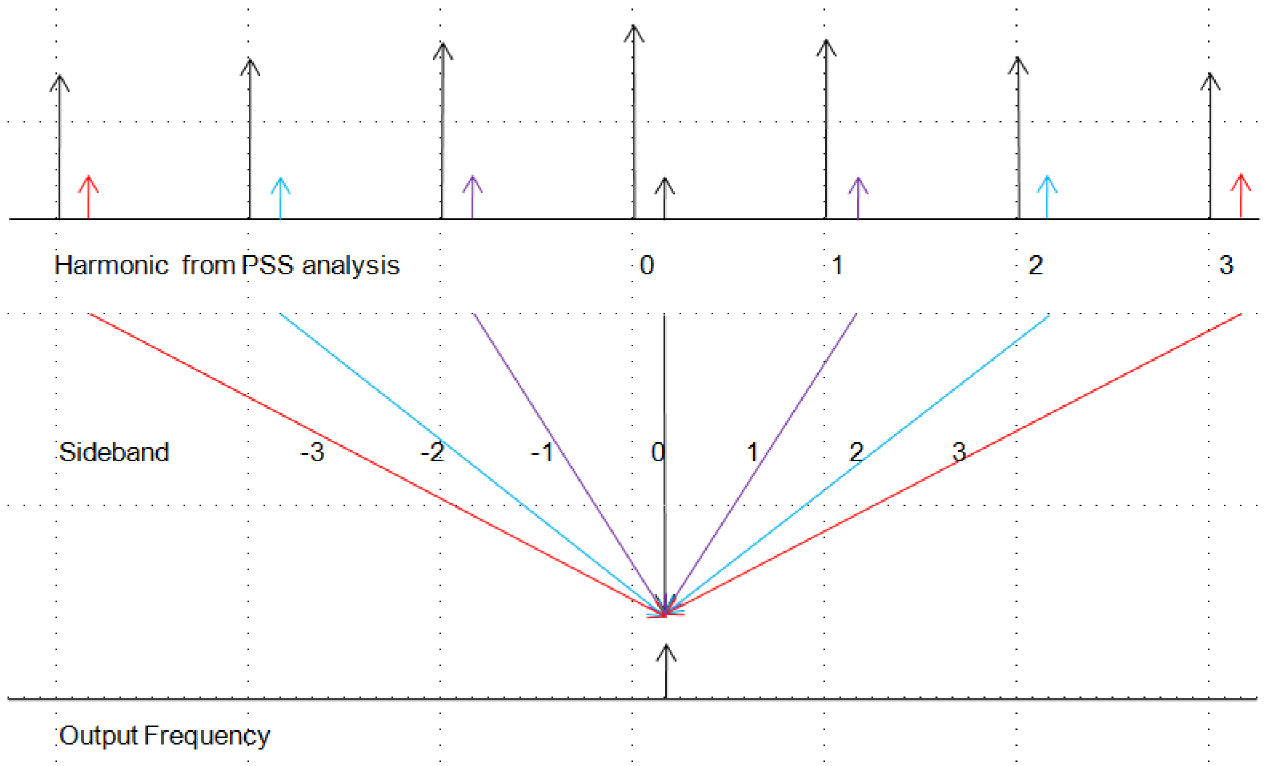
Now imagine that the 1GHz signal is 10dB larger. A 10dB change is a linear factor of about 3.16. The larger amplitude of the input causes a larger portion of the nonlinear device curves to be traversed, and thus the variation in conductance becomes larger. The circuit becomes more nonlinear, and the conversion gain is larger. This is illustrated below.



Therefore, the variation in the conductance of the devices from timepoint to timepoint in pss shooting contains information about how strongly a circuit translates frequencies, and this is used by the pxf small-signal analysis to calculate the small-signal conversion gain.

Normal Conversion Gain Measurement (Specialized Analysis = None)

This mode is used to make a normal conversion gain measurement.



The figure above illustrates a down-conversion case where the output is just above zero Hz.

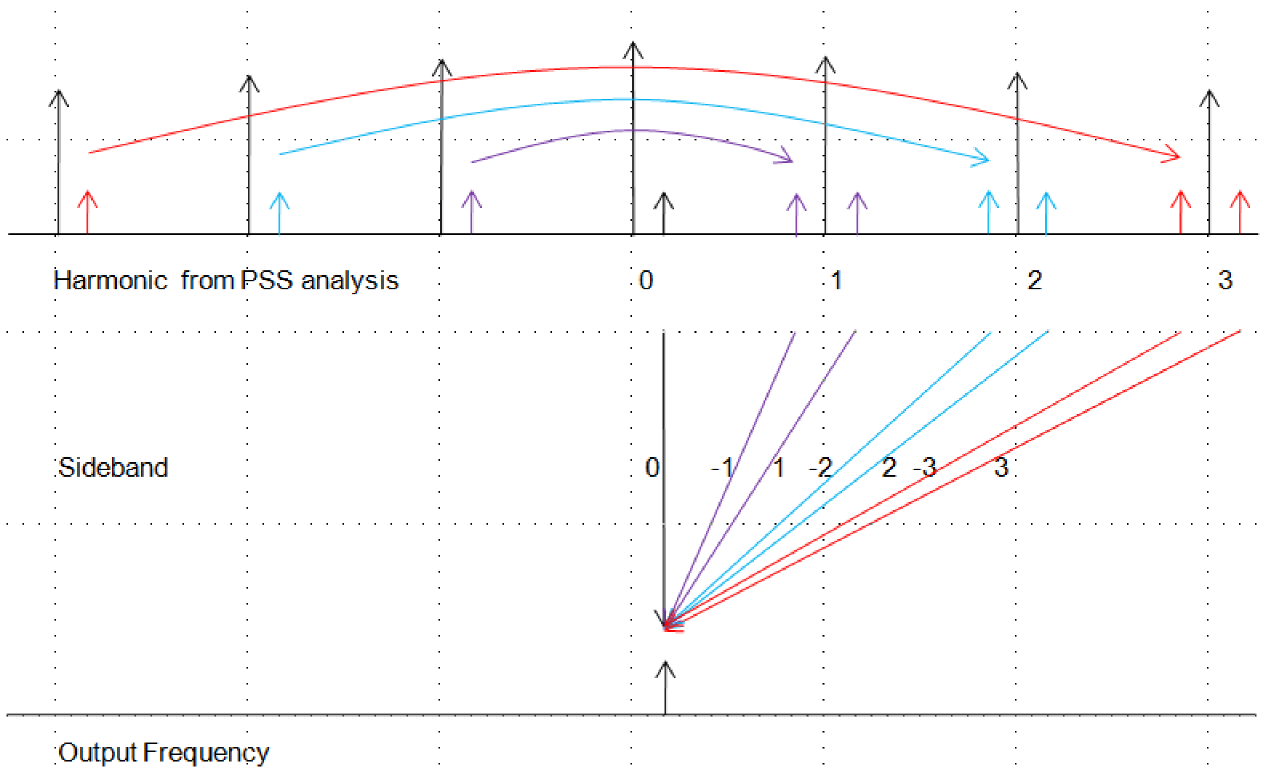
In the figure above, the top spectrum in black represents the harmonics created by the pss large-signal analysis. The negative and positive frequencies are shown and the DC component is in the center. The colored arrows show the transfer functions from the different input frequencies to the output frequency that is shown on the bottom axis. Pxf takes the maximum sideband property from the pxf *Choosing Analyses* form, and calculates the specified sidebands, which are input to output conversion gain transfer functions. In this example, maximum sideband is set to three.

The sideband numbers indicate which harmonic number is being mixed with for the transfer function calculation. Zero means that there is no frequency translation. One is the mixing product that occurs with an input frequency shift of +1 times the pss beat frequency. Mixing frequencies to the left have negative numbers because the input frequency that causes the output is more negative than the output frequency.

For example, imagine a large-signal input at 1GHz in the pss analysis and an output frequency of 100MHz to be calculated. With no mixing, there is a gain from the sources in the circuit to the output of the circuit at 100MHz. This is called the zero sideband because there is zero frequency shift from the input to the output. This is not the same measurement as a measurement in the ac analysis because the ac only includes the DC operating point, and the pxf analysis takes into account the circuit with the LO or clock signal applied.

The input frequencies that mix with the first harmonic at 1GHz to provide an output at 100MHz are at -900MHz and 1.1GHz. The frequency that is below the zero sideband is the -1 sideband and the frequency above the zero sideband is the +1 sideband.

Similarly, there are terms that mix with the second and third harmonics.



Usually, we think of positive frequencies only. This is shown above. The negative frequency inputs are reflected up into the positive frequencies. This is the default for pxf.

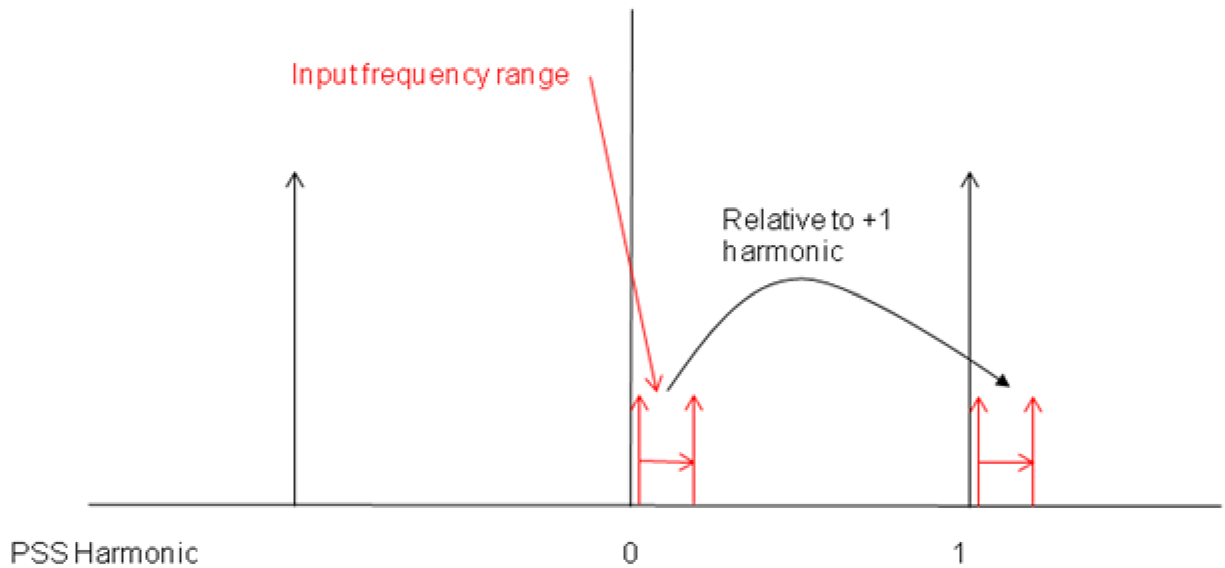
Your application sets the number of sidebands that you need. If you have specs through the fifth harmonic, you need to set maximum sidebands to 5.

Frequency Sweep

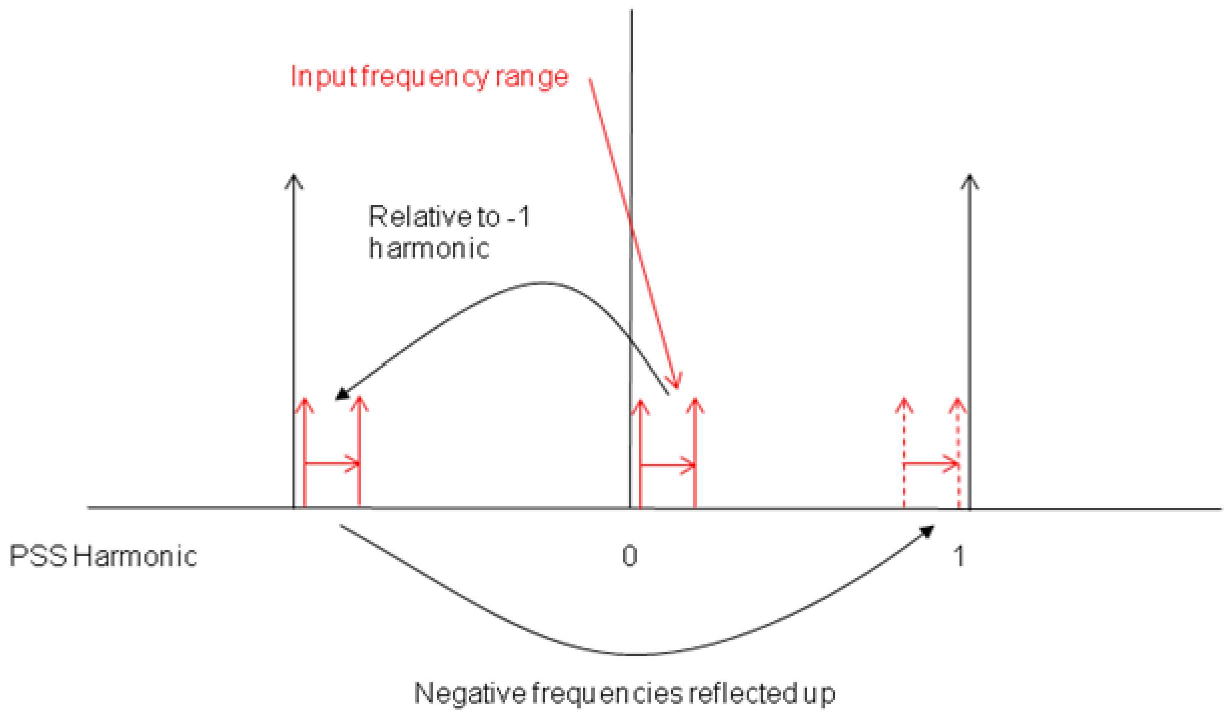
This analysis is like xf, where the **output** frequency is varied. The output frequency range to be analyzed is specified in the pxf *Choosing Analyses* form. Pxf then calculates the conversion gain terms.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonic in the pss analysis. For example, assume that the pss is at 1GHz, and a log sweep is desired above the first harmonic of the pss analysis. In this case, you need to select *relative* sweep, and specify 1 in the *Relative Harmonic* field. Next, type in 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade.

The sweep will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.



To use a log sweep below the first harmonic, use the same frequency range, and type in -1 as the relative harmonic. This is illustrated below.



Maximum Sideband

Sideband is the name of the pxf conversion gain terms. The sideband number is the pss harmonic number that is mixed with. If the sideband number is negative, the input frequency is more negative than the output frequency. If the sideband is positive, the input is more positive than the output frequency.

Conceptually, an infinite number of conversion gain terms exist when a single output frequency is analyzed in a nonlinear system. From the practical point of view, usually only a small number of conversion gain terms need to be measured near the LO frequency and a small number of harmonics.

There is a property in the *Choosing Analyses* form called *Maximum sideband* that is used to define how many mixing products should be calculated. If *Maximum sideband* is 0, the pxf analysis will only contain the frequency that appears without mixing or aliasing of any type. This is not equivalent to a linear AC analysis because pxf takes into account the instantaneously varying nature of the LO or sample clock signal because of the large signal being present in the pss analysis.

When *Maximum sideband* is 1, the inputs without frequency translation and the inputs that mix or alias with the first harmonic of the pss analysis are present in the pxf analysis. When *Maximum sideband* is 10, the pxf analysis includes mixing through the 10th harmonic of the pss analysis. When shooting is used for the pss analysis, the number of sidebands can be any number up to four times the number of harmonics specified in the pss analysis. When harmonic balance is selected, up to the number of harmonics in the pss analysis can be specified.

Setting Harmonics and Sidebands

Shooting

The pss analysis, when shooting is selected, has a minimum of 200 timepoints, so it inherently has frequency domain content through the 100th harmonic of the pss beat frequency. Maximum harmonic is a pss post-processing property that specifies how many harmonics to calculate in the Fourier transform of the waveform in pss. If the number of harmonics is zero, it just means that no frequency domain information should be calculated in the simulation. If the number of harmonics is 10 or less, there is no effect on the PSS waveform. When maximum harmonics gets larger than 10, the minimum number of timepoints in the pss analysis is increased. 20 timepoints in the period of the highest harmonic are forced. Thus, if maximum harmonics is raised above ten, the pss waveform gets more timepoints which increases the accuracy of the waveform, at the cost of longer runtimes and more memory required to run the pss simulation.

In order for the pxf analysis to correctly measure conversion gain, the wave shape at the highest mixing harmonic needs to be accurate, so with the default settings in shooting pss, up to 40 sidebands can be accurately calculated in the pxf analysis. In most cases, only a small number of sidebands needs to be calculated, and so with shooting, you can usually count on the pxf result to be accurate.

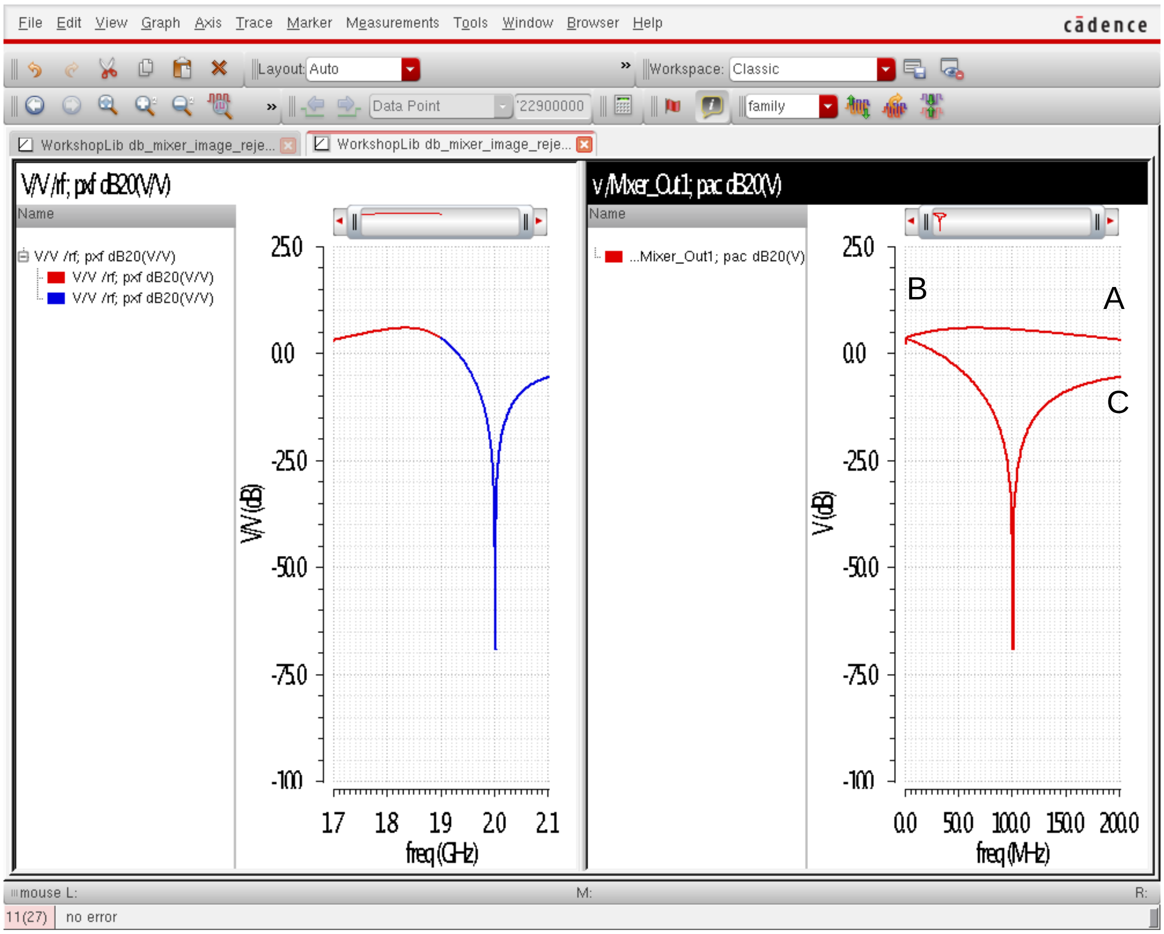
Although this is very unusual, if you need more than 40 sidebands when you are using shooting pss, you need to set the *maxacfreq* option in the pss analysis to the pss beat frequency times the number of sidebands requested in the pxf analysis, or alternatively, set the number of pss harmonics to the number of sidebands in pxf divided by four. Either method works to set the maximum timestep in the pss analysis as required for the pxf analysis to be accurate.

Harmonic Balance

When *Harmonic Balance* is selected in the pss analysis, only the harmonics specified in the pss *Choosing Analyses* form exist in the solution. In the pxf analysis, because only the harmonics specified in the pss-hb *Choosing Analyses* form exist, the maximum sideband term can only be the number of harmonics specified in the pss analysis or less. Usually, when *Harmonic Balance* is selected in the pss analysis, the *Maximum sideband* field in pxf usually should be left blank which calculates all the conversion gain calculations from all the harmonics in the pss analysis. The *Direct Plot Form* is then used to display just the conversion gain terms you want.

Note that with harmonic balance, the number of harmonics (and oversample factor if you have non-sinusoidal waveforms) in the pss analysis needs to be determined. Start with a number of harmonics in the pss based on an estimate of how many harmonics the circuit actually produces with your input waveform, and run the pss and pxf analyses. Remember that for non-sinusoidal waveforms, oversample factor also needs to be set greater than one. For more information, see the pss section at the beginning of this chapter or the hb analysis at the beginning of Chapter 3. Run the analysis and make the desired measurement. Now increase the number of harmonics by roughly 50% and run the simulation again. If the answer does not change, then you had enough harmonics originally, and you might actually be able to decrease the number of harmonics from the original setting. If you have non-sinusoidal waveforms, increase the oversample factor to between four and eight before increasing the number of harmonics to find the minimum value for both oversample factor and harmonics.

Pxf is usually preferred to pac when you need to measure both the passband and the image. Below is a plot of the pxf and pac output for an image reject mixer with a 1.9GHz LO and a 100MHz IF. The passband is at 1.8GHz and the image is at 2GHz.

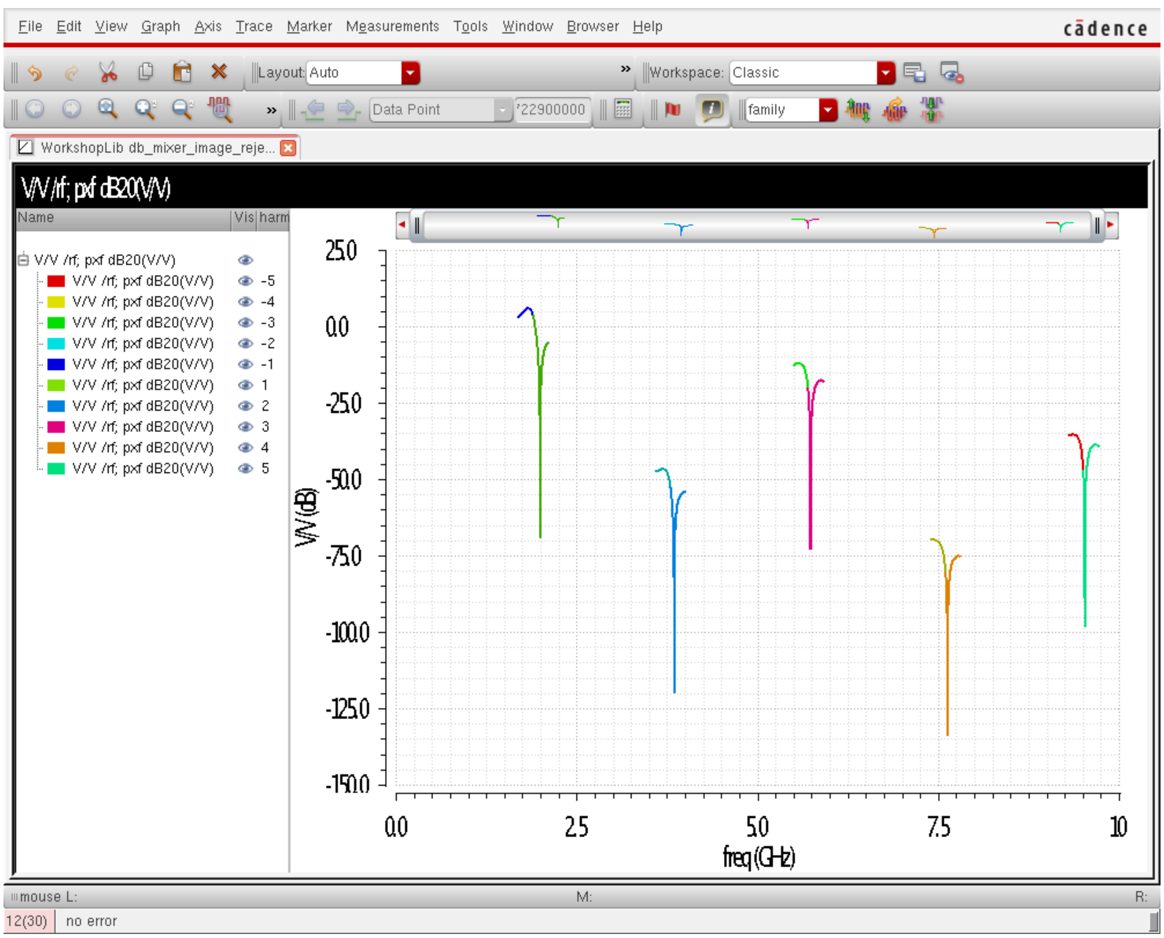


On the left is a pxf plot when the input port is probed, thus measuring the gain from the input to the output of the image reject mixer. To read the gain, go to the **input** frequency, and read the gain on the vertical scale. At the passband frequency of 1.8GHz the gain is about 8dB. In the image at 2GHz, the gain is about -70 dB. The LO frequency is at the junction of the red and blue traces.

To measure passband and image gain in pac, the input frequency starts at 1.7GHz and stops at 2.1GHz. At 1.7GHz, the output frequency is -200MHz, but pxf reflects the negative frequencies to the positive frequency domain by default. The trace starts at point A above, and sweeps down to zero at point B. The input frequency is 1.9GHz and the output is zero Hz. As the input continues above 1.9GHz, the output frequency becomes positive, and follows the curve to point C. Most people find the curve from pac to be more confusing than the curve from pxf, so if you need to measure both the passband and the image, or if you need to measure the conversion gain near the pss harmonics, pxf is recommended.

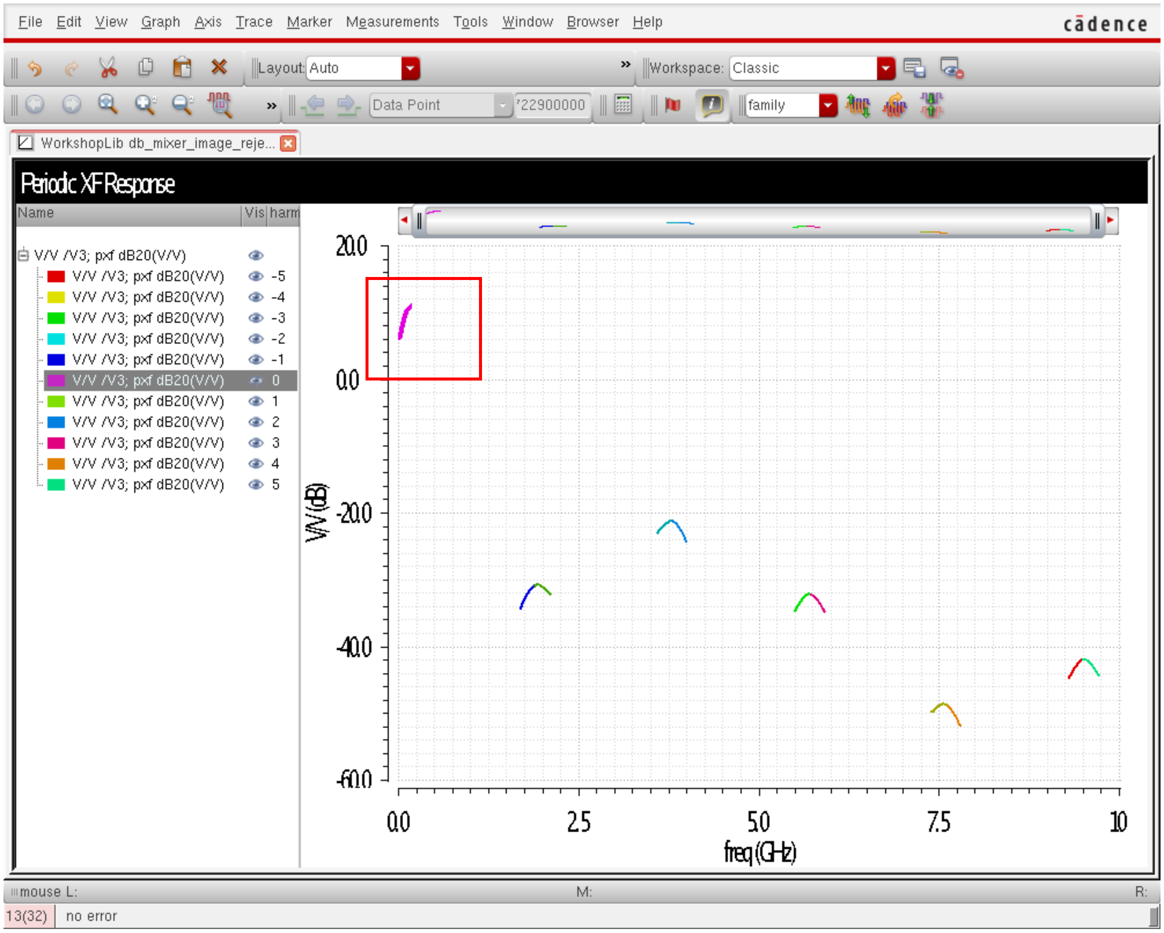
Note that in the pac result above, you cannot just add the gain values at a specific frequency. If the frequency was 100MHz, you do have two different gain values of about +5dB and -70 dB. The output frequency for the +5dB curve is -100M, and the output for the -70dB curve is +100MHz. Since the frequencies are different, you cannot just add the two gain numbers.

Pxf can measure the gain near may harmonics at the same time by increasing the maximum sideband parameter in the *Choosing Analyses* form. Below is an example of conversion gain through the fifth harmonic. Your application will determine the number of sidebands to set.

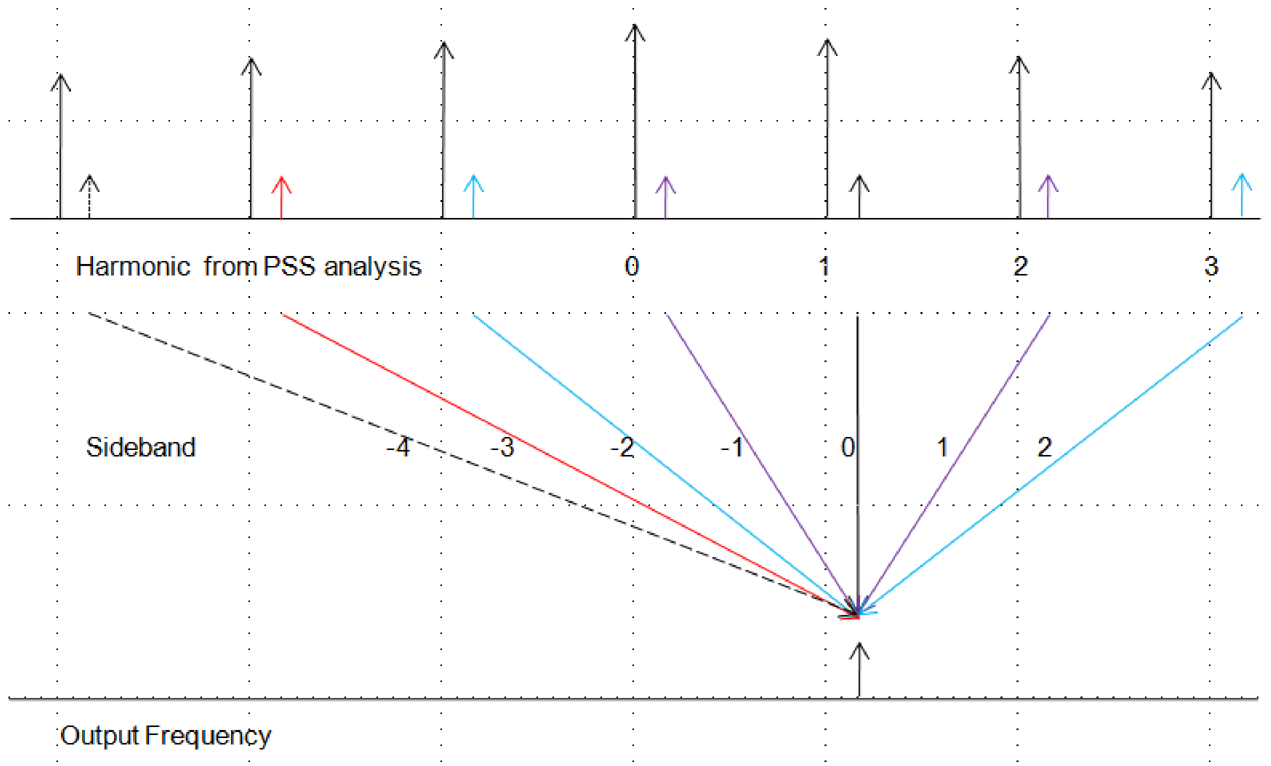


Note that the conversion gain is still relatively high near the third harmonic.

Pxf can also be used to measure the conversion gain from the battery that supplies power to the output. All the gain and conversion gain terms should be below unity, or a potential oscillator is created. In the plot below, the gain from the bias source to the output at the IF frequency of 100MHz is above 0dB. This circuit is a potential oscillator.

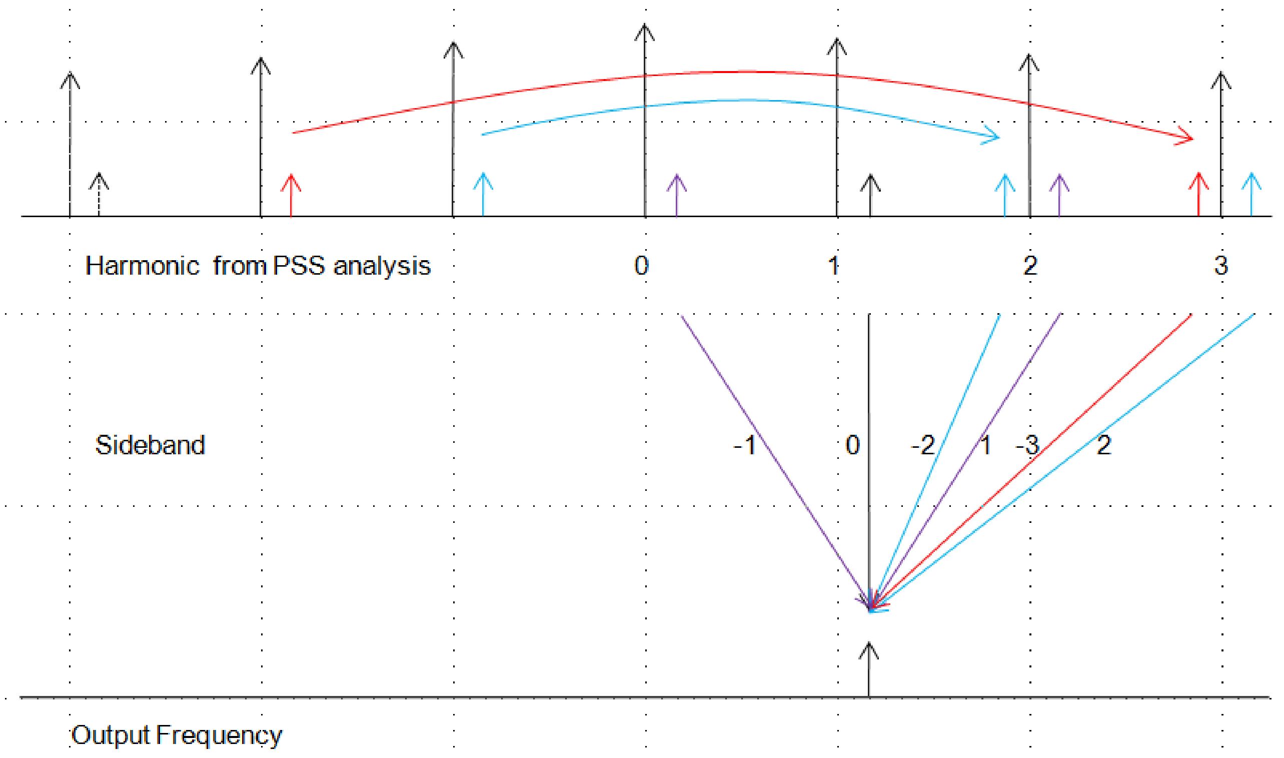


Pxf is useful for up-conversion applications as well, with the limitation that a small-signal gain measurement is made. This is shown below.



The difference is that the output frequency is just above the first harmonic. The rules for the sidebands are the same. -1 means that the input frequency is shifted down from the output frequency by -1 * pss beat frequency.

Usually, we think in terms of the positive frequency domain, which is shown below.

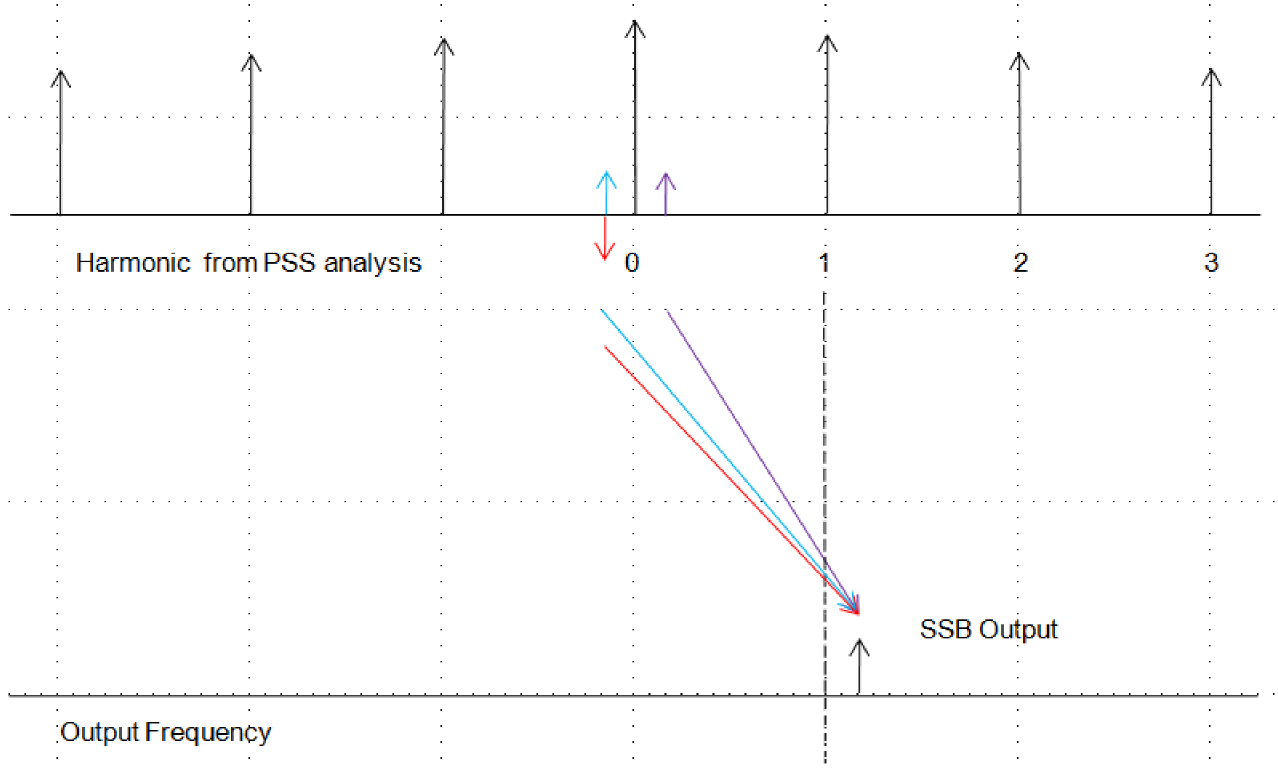


AM to PM measurement (Specialized Analysis = Modulated)

This is used to measure the am to pm conversion. Normally, it is used for oscillators to measure the am to pm conversion from the power supply, but it can also be used for driven circuits. In both cases, remember that pxf is a small-signal analysis. If you need a large-signal measurement, use the transient, hb, or pss analysis.

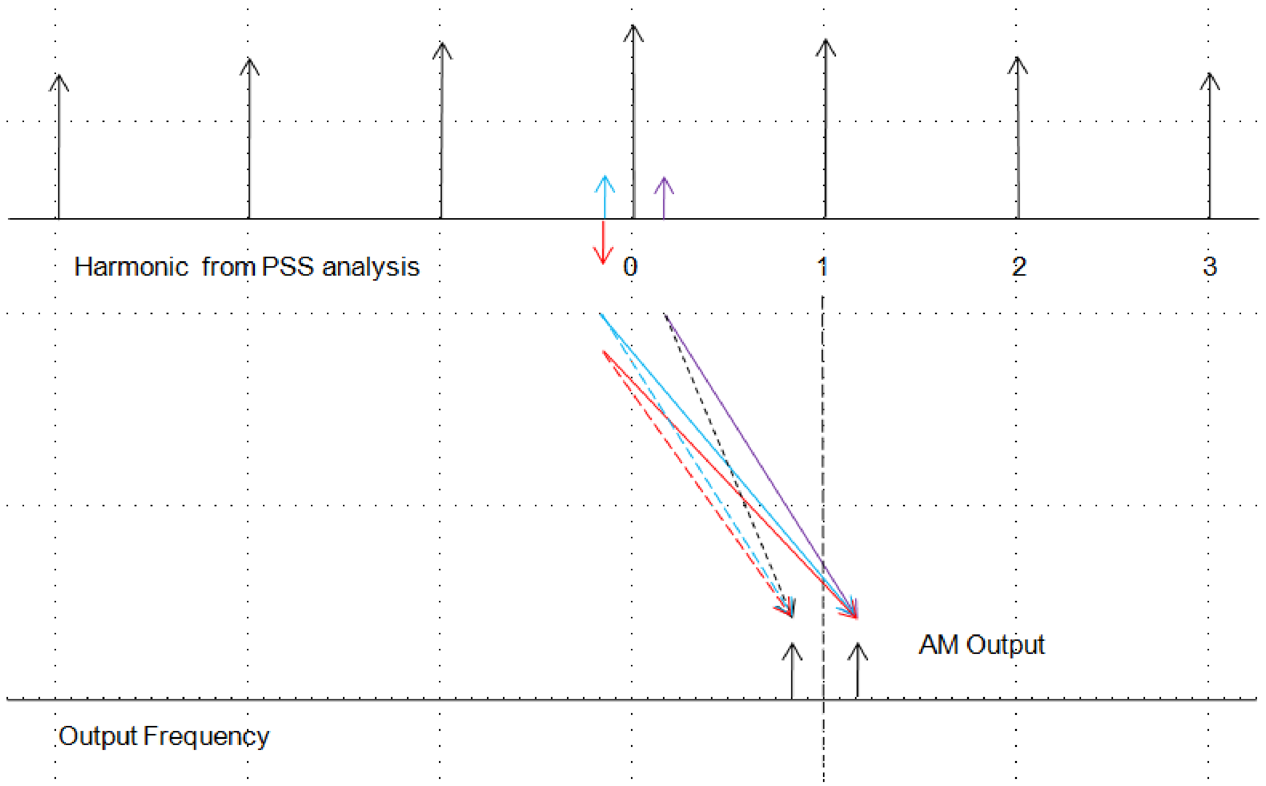
In modulated analysis, the output type should be `ssb`/`am`/`pm` so that all the gain terms can be calculated. Generally, the AM to PM gain is desired.

Conceptually, if the **output** is ssb (single sideband) then the PM output cannot be measured.

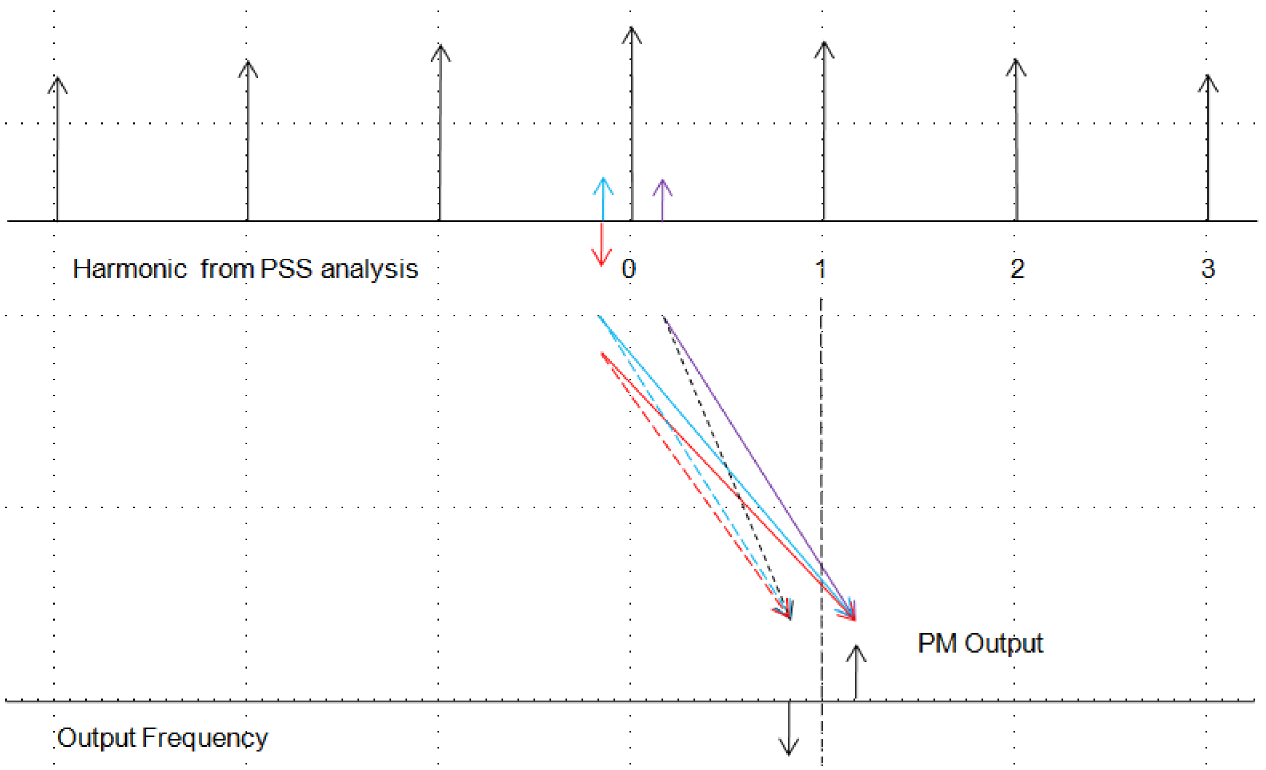


The gain from SSB, AM, or PM inputs to the output frequency is the only gain measurement available.

When the output is AM, both the sidebands above and below the chosen harmonic are calculated.



This can measure SSB, AM, or PM to an AM output.



When the output is PM, the SSB, AM, or PM to PM gain can be measured. Generally, the AM to PM gain is desired.

Sampled PXF

This analysis is seldom used.

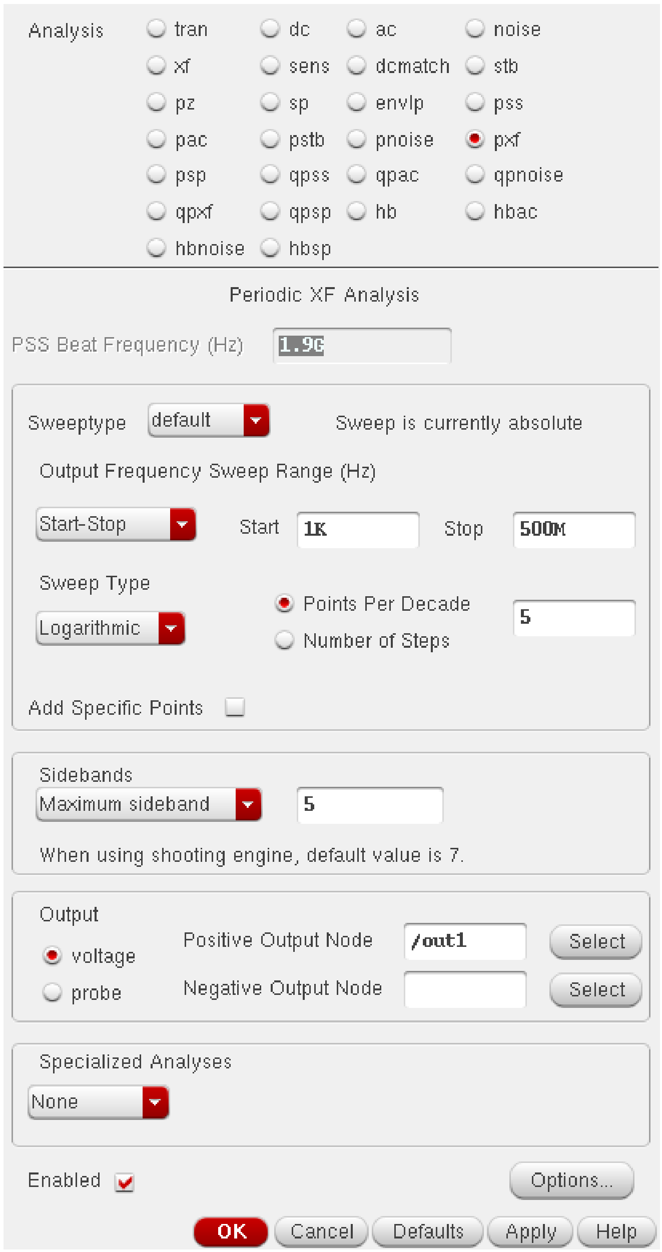
The basic idea is to measure the conversion gain as if the conversion gain was repeatedly measured at the same phase (instant in time) in the periodic large signal that is driving the circuit over an infinite number of cycles of the large signal. This measurement is impossible to duplicate in the lab. The primary application is for measuring the gain from the power supply to the output which is driving a digital circuit or a comparator that is triggered at a specific voltage level.

ADE Implementation

This section jumps from the theoretical to the practical with examples of how to use the individual analysis types we just covered. The focus is only on the pxf *Choosing Analyses* form so you can see where the individual settings are made. For examples with all the steps shown, see the examples section that follows this section.

PXF for a Normal Conversion Gain Measurement

Here is an example of the *Choosing Analyses* form for a normal conversion gain measurement.



1. At the top of the form the PSS beat frequency of 1.9GHz is shown.

2. Sweeptype can be relative or absolute. The default for driven circuits is absolute, and for oscillators it is relative. The selection that is being used is shown to the right of the sweeptype selection.

3. The frequency sweep range is always an **output** frequency range. It is not a sweep in the traditional sense of connecting an input, sweeping over a frequency range, and measuring the output. Instead, it is an output frequency range to be analyzed. At each output frequency in the sweep, the input frequencies that will mix with the pss harmonics to cause that output frequency are calculated, and the forward conversion gain is calculated.

4. Linear and log spacings are provided. Generally, *Auto* should be avoided because it will always calculate 50 frequency points. Usually, a much smaller number is needed, so it is recommended that you set *linear* or *log*, and control the resolution yourself.

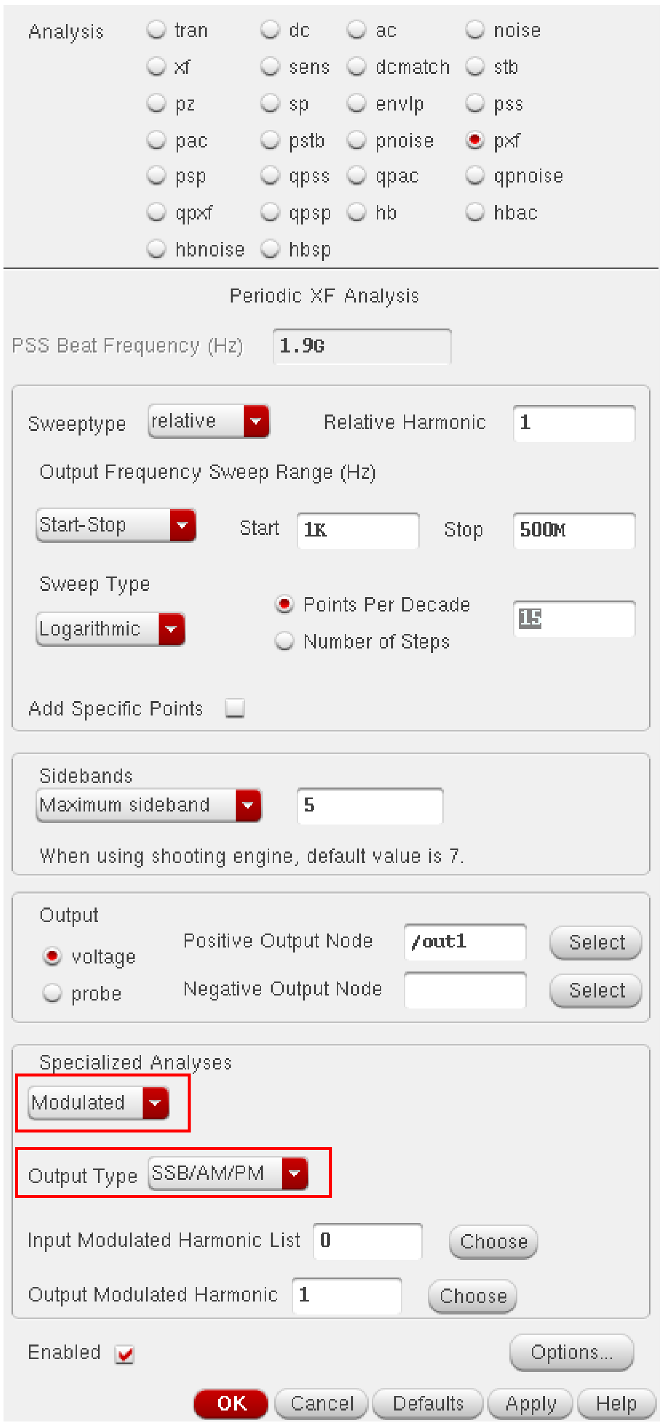
5. Although there are choices for the Sidebands selection, it is usually better to leave the setting at *Maximum sideband*, and if you need a small subset of the output results, use the *Direct Plot Form* to filter the results. For shooting, you can use up to 40 sidebands with no additional settings. If you need more than 40 sidebands, either set the number of pss harmonics to the number of sidebands divided by 4, or set the pss option *maxacfreq* to the pss beat frequency times the number of sidebands in pxf. For harmonic balance, choose a number of harmonics in the pss form, and leave the *Maximum sideband* field blank. This will calculate the conversion gain from all the harmonics in the pss analysis. For harmonic balance only, run the simulation, and make the measurement in pxf. Now increase the number of pss harmonics by about 50% and run again. If the pxf result changed significantly, then you need more harmonics in pss. If it did not change, use the smallest number of harmonics that results in an stable measurement.

6. The output can measure either a voltage or a current. The voltage measurement is the default. Specify the output node or nodes in the circuit. If the output is a current select probe for the output, add an iprobe from analogLib (a current probe) in series with the output in the schematic, and specify the instance name in the *Choosing Analyses* form.

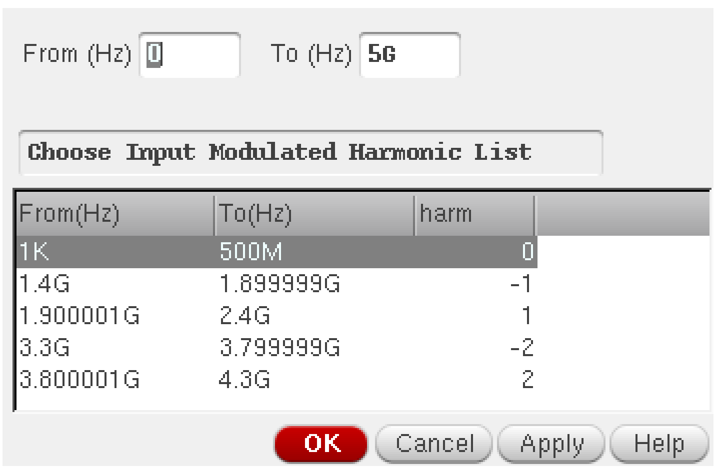
7. When *Specialized Analysis* is set to *None*, a normal conversion gain measurement is made.

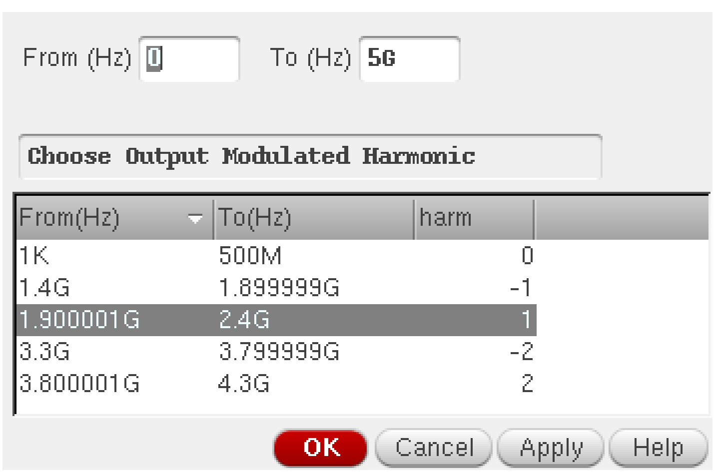
PXF Modulated (AM to PM Conversion)

Here is an example of a modulated *Choosing Analyses* form.



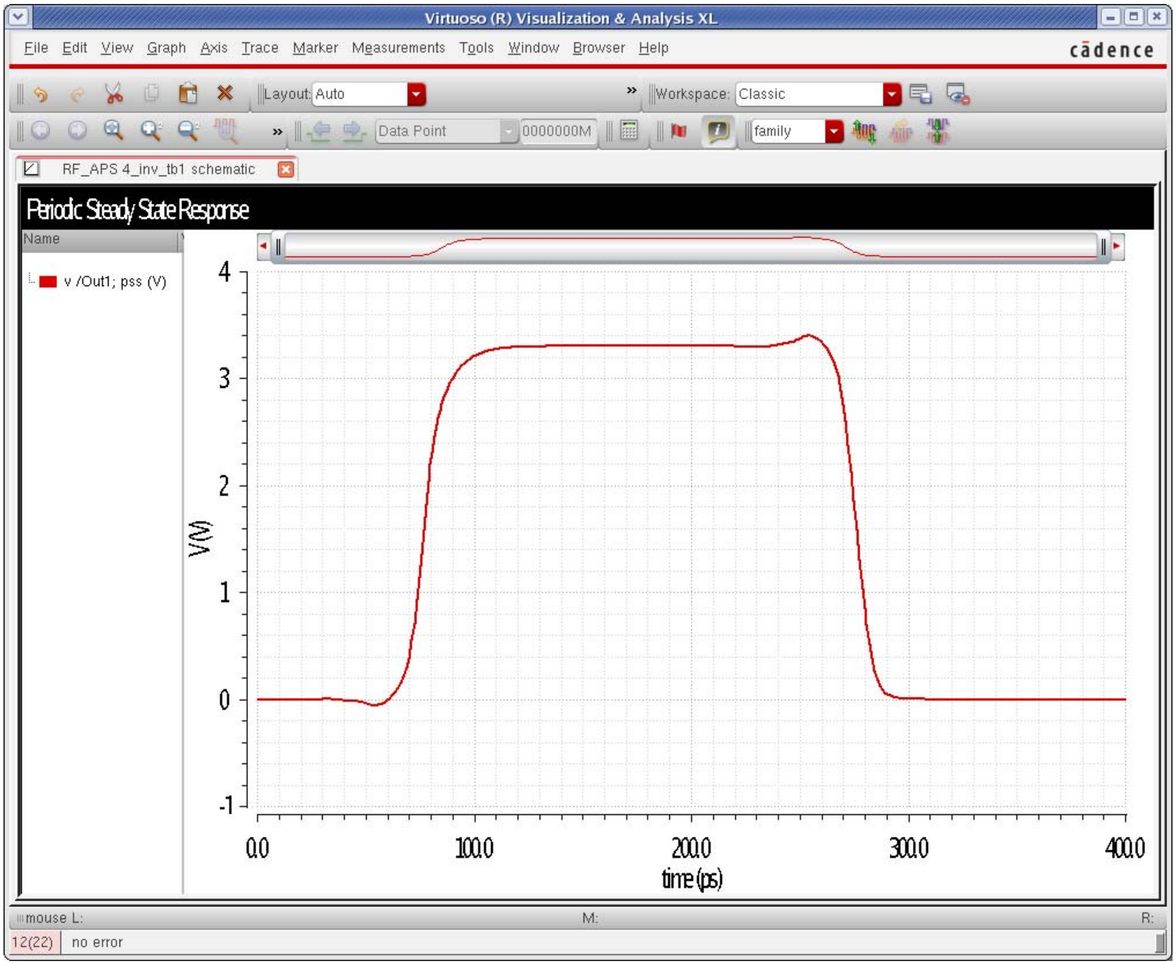
1. The functions of the form above the *Specialized Analyses* section are the same as for a normal conversion gain measurement.
2. In the *Specialized Analyses* section, select *Modulated*. The bottom part of the form redraws.
3. In general, set the *Output Type* to *SSB/AM/PM*. This allows all the different transfer functions to be calculated, with only a small performance penalty.
4. In modulated, the input and output frequencies need to be identified. The best way to do this is to click the *Choose* button to the right of the field, select the input frequency from the list of available frequencies, and click *OK*. Then repeat this action for the output frequency range, as shown below.





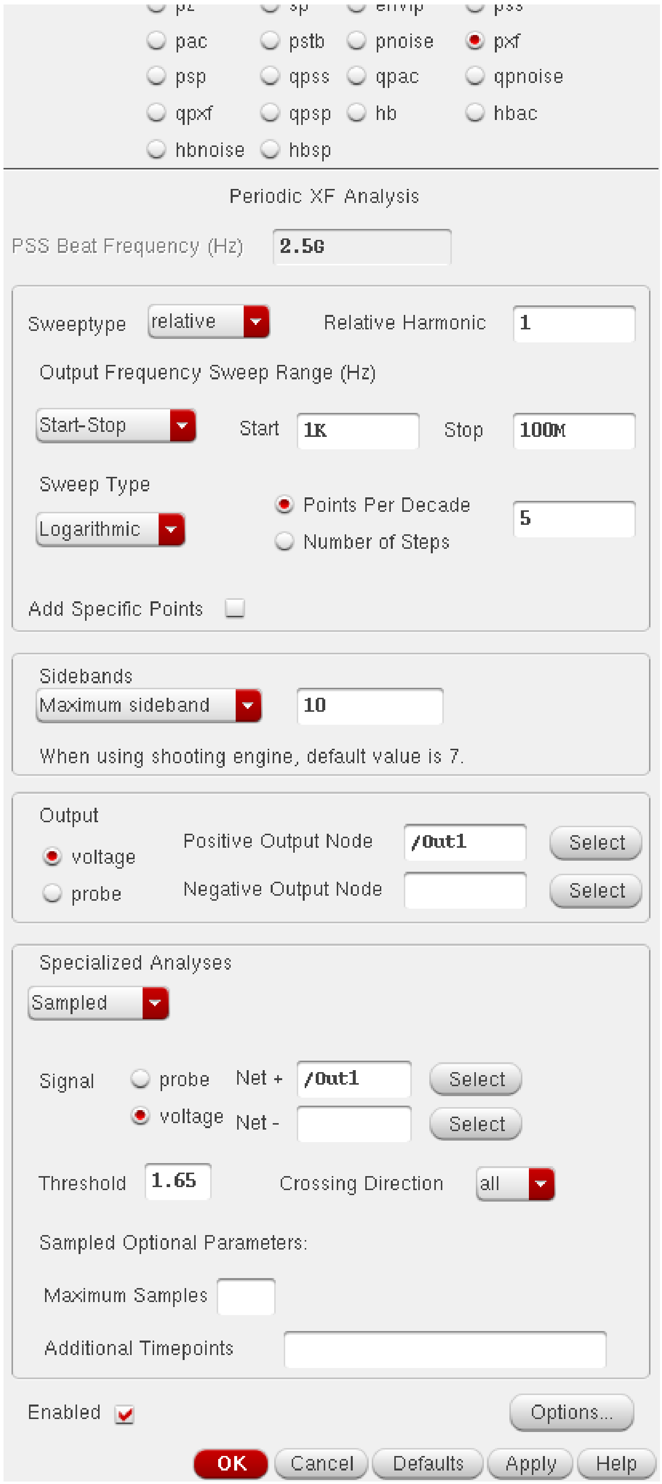
PXF Sampled

Sampled pxf is used to measure the conversion gain at a specific point in time of the periodic signal during the pss analysis. In general, this analysis is applied to digital circuits so that the instantaneous conversion gain from the power supply to the output can be measured. Below is the time-domain waveform of a 3.3 volt digital logic circuit.



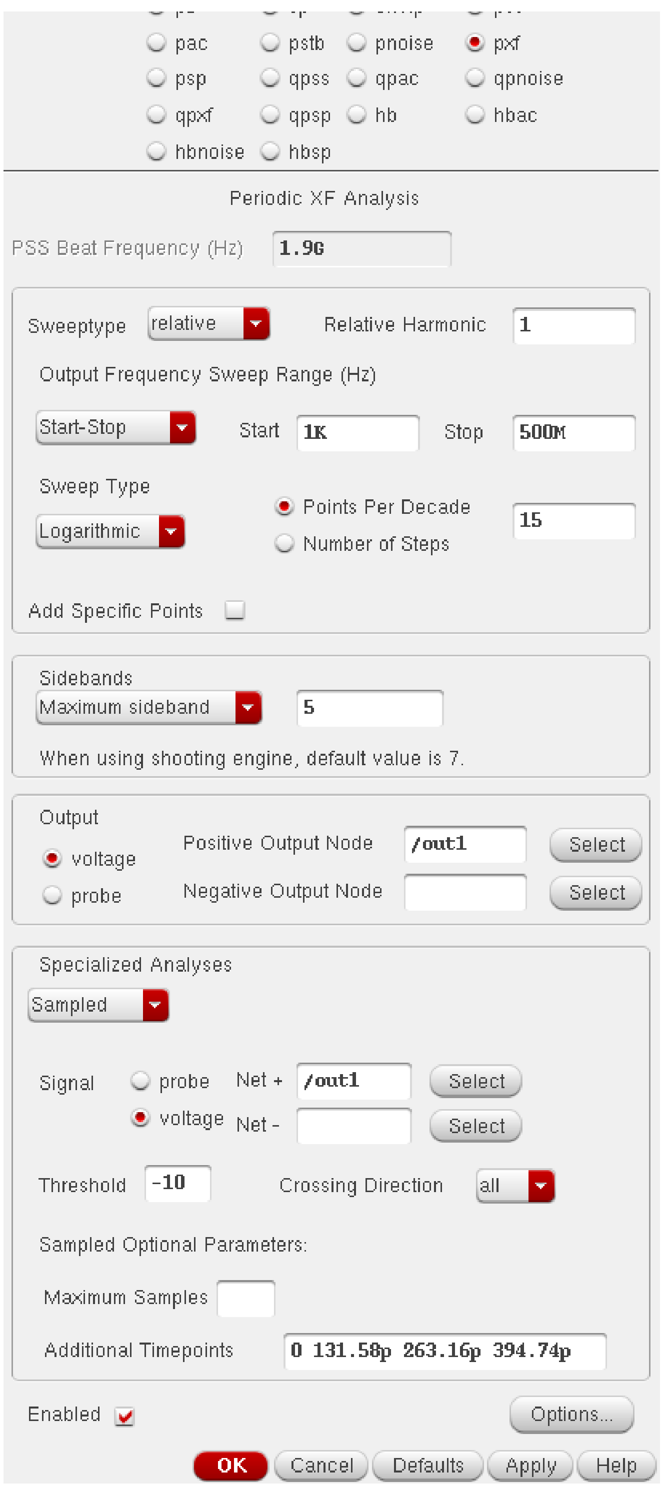
In this case, a threshold of 1.65 volts would be appropriate.

Below is an example of a sampled setup for a 3.3 volt logic circuit.



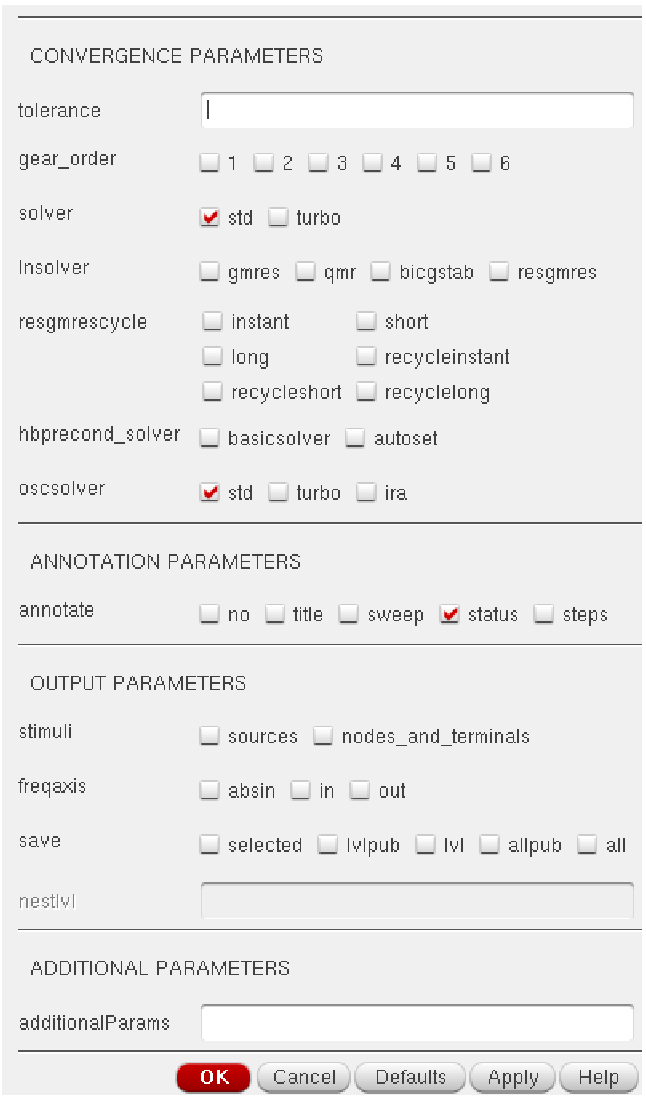
1. The functions of the form above the *Specialized Analyses* section are the same as for a normal conversion gain measurement.
2. In the *Specialized Analyses* section, select *Sampled*. The bottom part of the form redraws. If your output is a current, add an iprobe (current probe) from analogLib in series with your output in the schematic, and select the iprobe as the probe selection. If you have a voltage, select voltage, and specify a node or nodes in the schematic.
3. The normal use of the form is to specify a voltage threshold for the measurement in the threshold field, and not specify additional timepoints. The example above is for a 3.3 volt logic circuit where the switching threshold is 1.65 Volts.
4. The *Crossing Direction* can be set to *all*, *rise*, or *fall*.
5. In the case of multiple crossings, as shown in the first pss waveform shown above, all the crossings will be calculated with a maximum of 16 by default. If you have more than 16 crossing events, set the number of crossings you have in the *Maximum Samples* field. In this example, there are two crossing events, so no entries are required in the *Maximum Samples* field.
6. If you have specific pss times you want to measure, specify a list of pss simulation times in the *Additional Timepoints* field. Separate the entries with a space. An example of using only the timepoints is shown below. A threshold of -10V is well outside the

anticipated voltage on the output node which disables the timepoint selection using a voltage threshold.



PXF Options

Below is the Pxf options form.



The below options are occasionally used.

Solver

The solver option is used only for driven circuits. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pxf output frequency is very close to the frequency of one of the harmonics in the pss result, warning messages will appear in the pxf output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but which takes longer to run than the *turbo* solver.

Oscsolver

The oscsolver option is used only for oscillators. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pxf output frequency is very close to the frequency of one of the harmonics in the pss result, warning messages appear in the pxf output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but takes longer to run than the *turbo* solver.

For very large oscillator designs, the *ira* oscsolver may take less memory and run faster with no loss in accuracy, however, it is less robust for convergence compared to *turbo* or *std*.

Hbprecond_solver

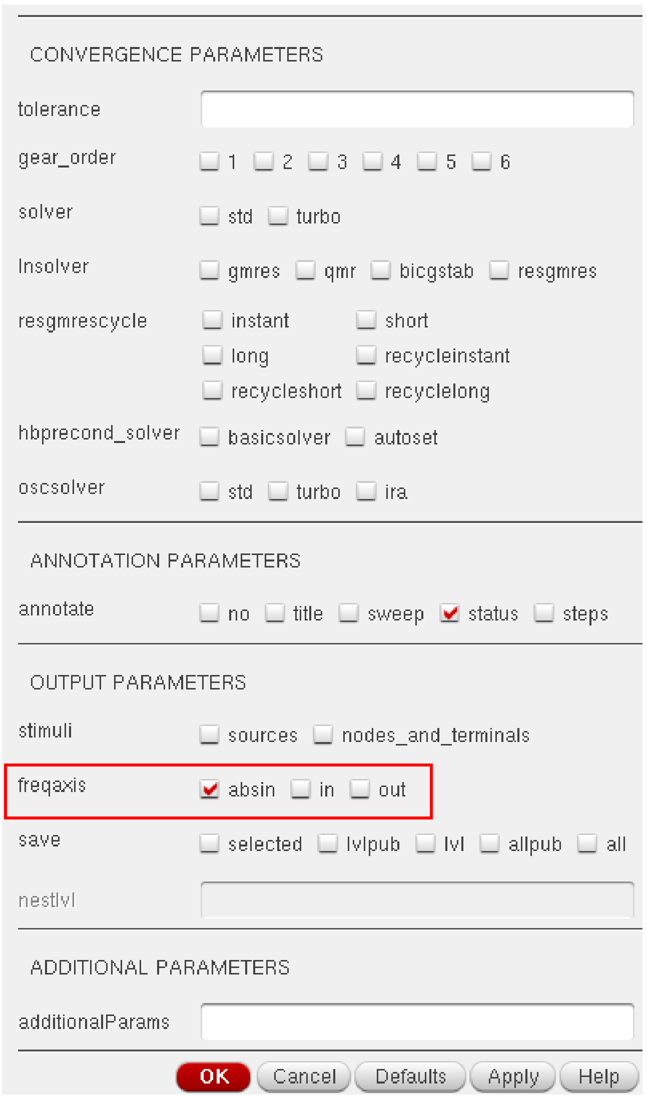
This option is only available only when harmonic balance is selected in PSS and APS is used.

The basic solver is the only solver available in standard Spectre. *autoset* is the default solver when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window. If you have stagnation, it will save a small amount of time to set this option to *basicsolver*.

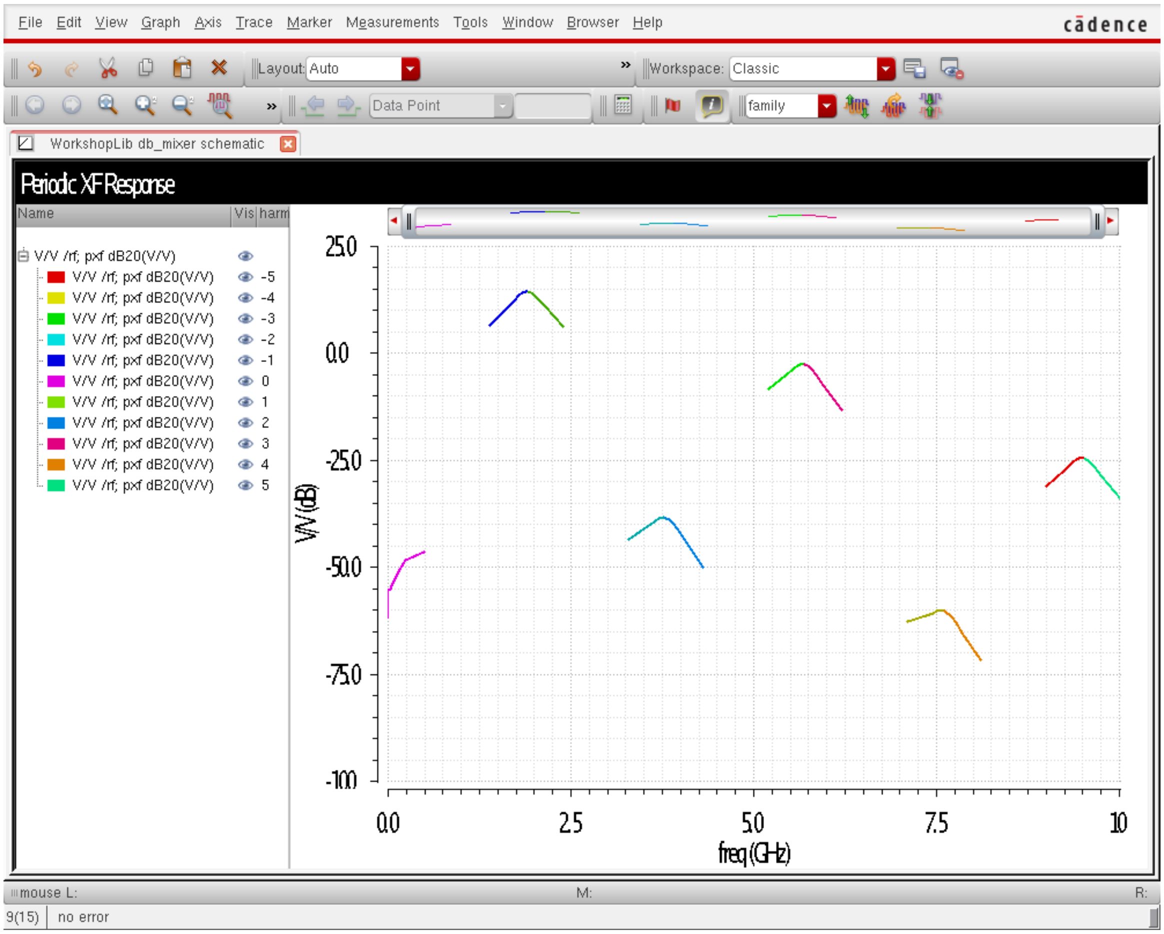
Freqaxis

freqaxis specifies whether you want to see the negative frequency axis or not.

The analysis calculates the input frequencies and forward conversion gain, so *in* and *absin* are reasonable choices. *in* displays the negative frequency axis. *absin* (absolute value of the input) displays positive input frequencies. This is the default.

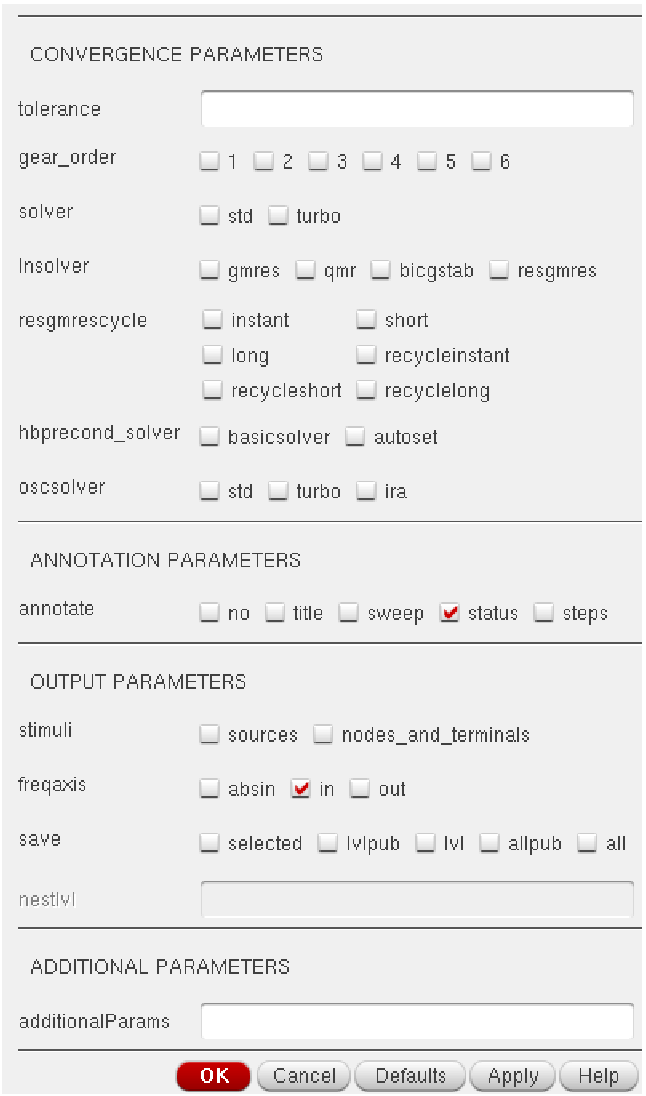


The input frequencies are plotted on the positive frequency axis.

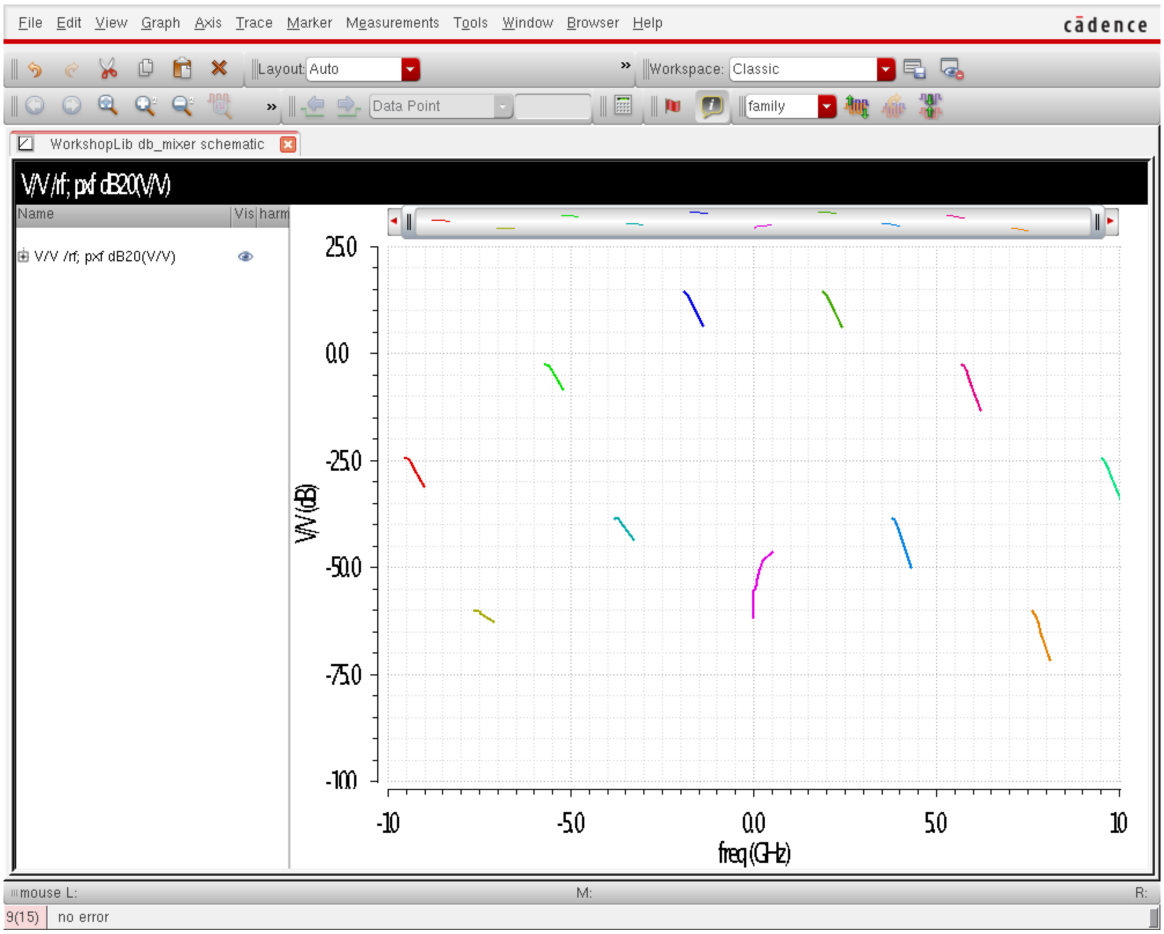


To read the conversion gain, go to the **input** frequency on the curve, and measure the conversion gain.

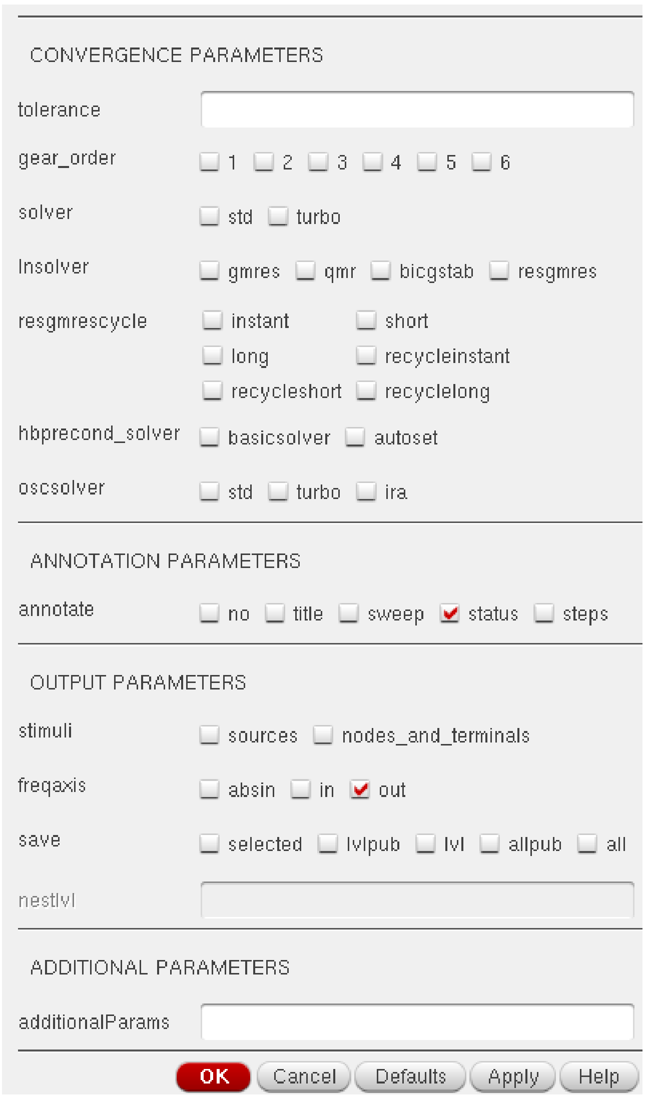
Below is an example of *freqaxis* set to *in*.



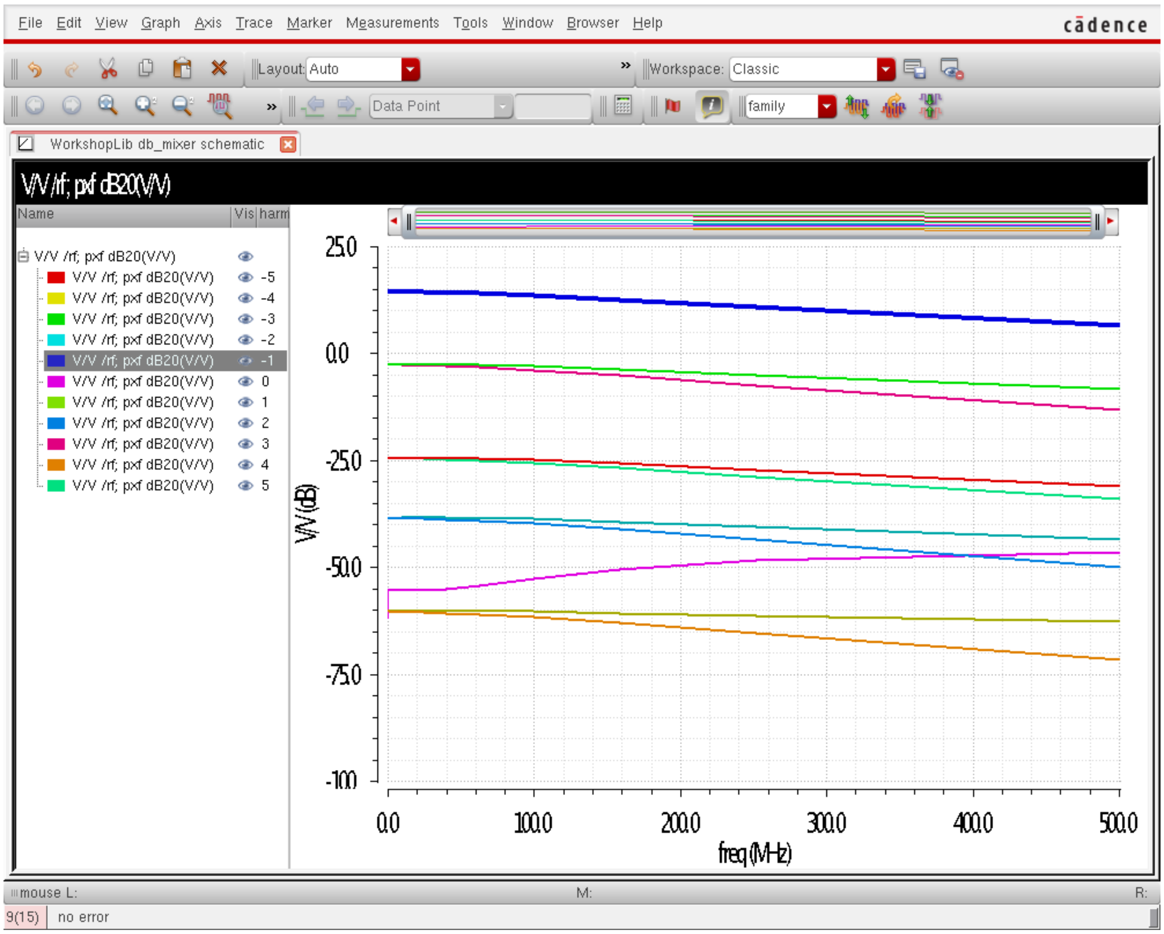
Note that the data is the same as before. It is just displayed with the negative frequency axis present. The negative frequencies have not been reflected up to the positive frequency domain. The waveform display shows the negative frequency axis now.



If you select *out*, all the inputs at the different frequencies are plotted on the same output frequency range scale. This is not recommended.



The waveform tool shows all the different input frequencies on the same output frequency scale. Again, this is not recommended.



The following options are not normally used.

Tolerance

Leave this option at the default value.

Pxf uses an iterative solver to calculate the conversion gain terms. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough. The tolerance option specifies that accuracy for pxf. For shooting, the default tolerance is 1e-9. For driven circuits where HB is the pss engine, the default is 1e-6. For oscillators where hb is selected for the pss engine, the default is 1e-4.

Gear_order

Do not change this option.

Lnsolver

Leave this option at the default.

Each pxf sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases with more iterations. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

Resgmrescycle

Leave this option at the default. For the resgmres linear solver, there are several different options.

Annotate

This option controls the level of detail in the Spectre output log. Less detail is on the left. More detail is on the right.

Stimuli

This option is not implemented at the current time.

Save

This option is not implemented at the current time.

Nestlvl (Shooting and Harmonic Balance)

This option is not implemented at the current time.

AdditionalParams

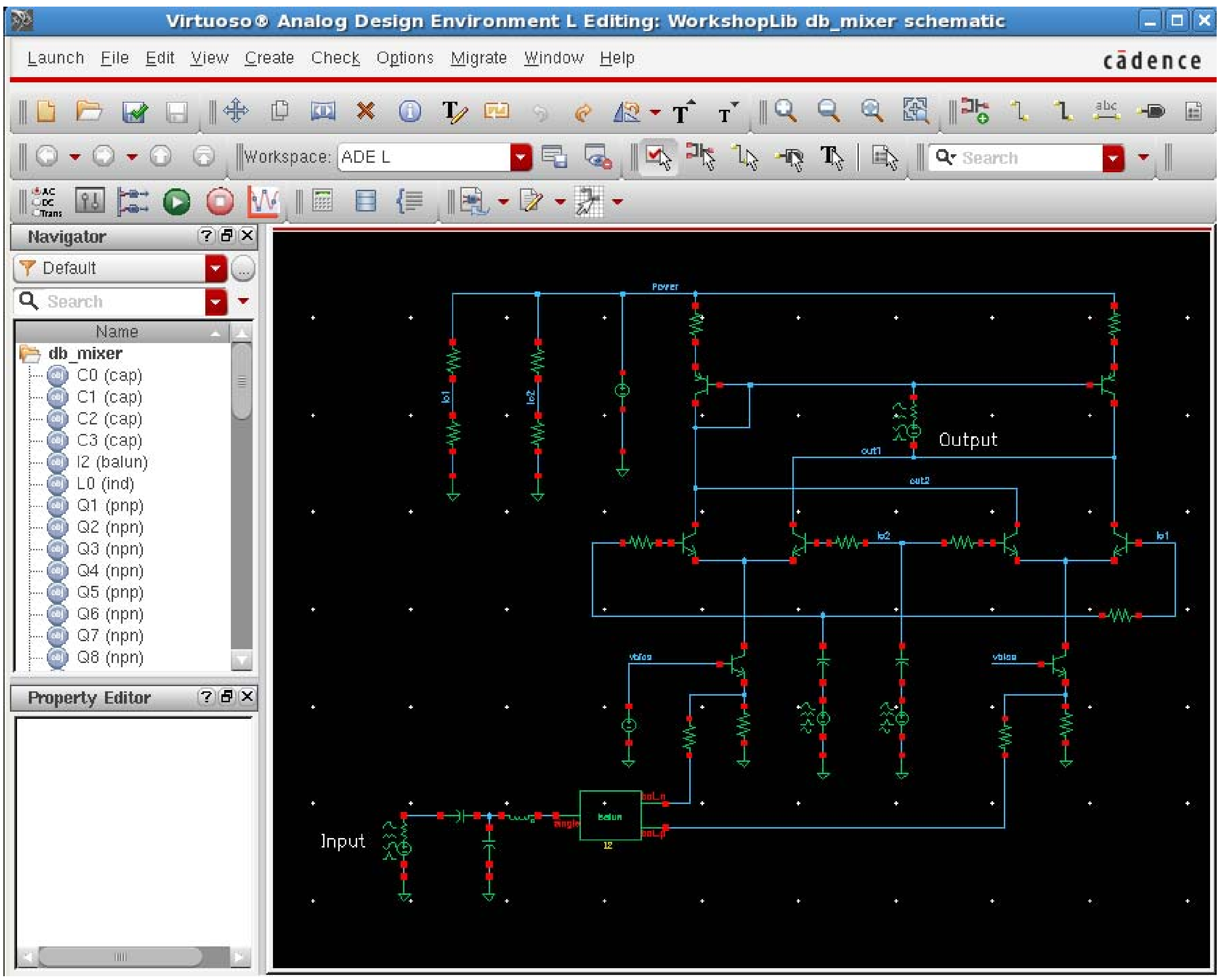
additionalParams is typically used for new features that are being beta tested. This field expects `keyword=value` pairs.

For more information about the other options, type `spectre -h pxf` at the command prompt in a Unix shell window.

Examples for Driven Circuits

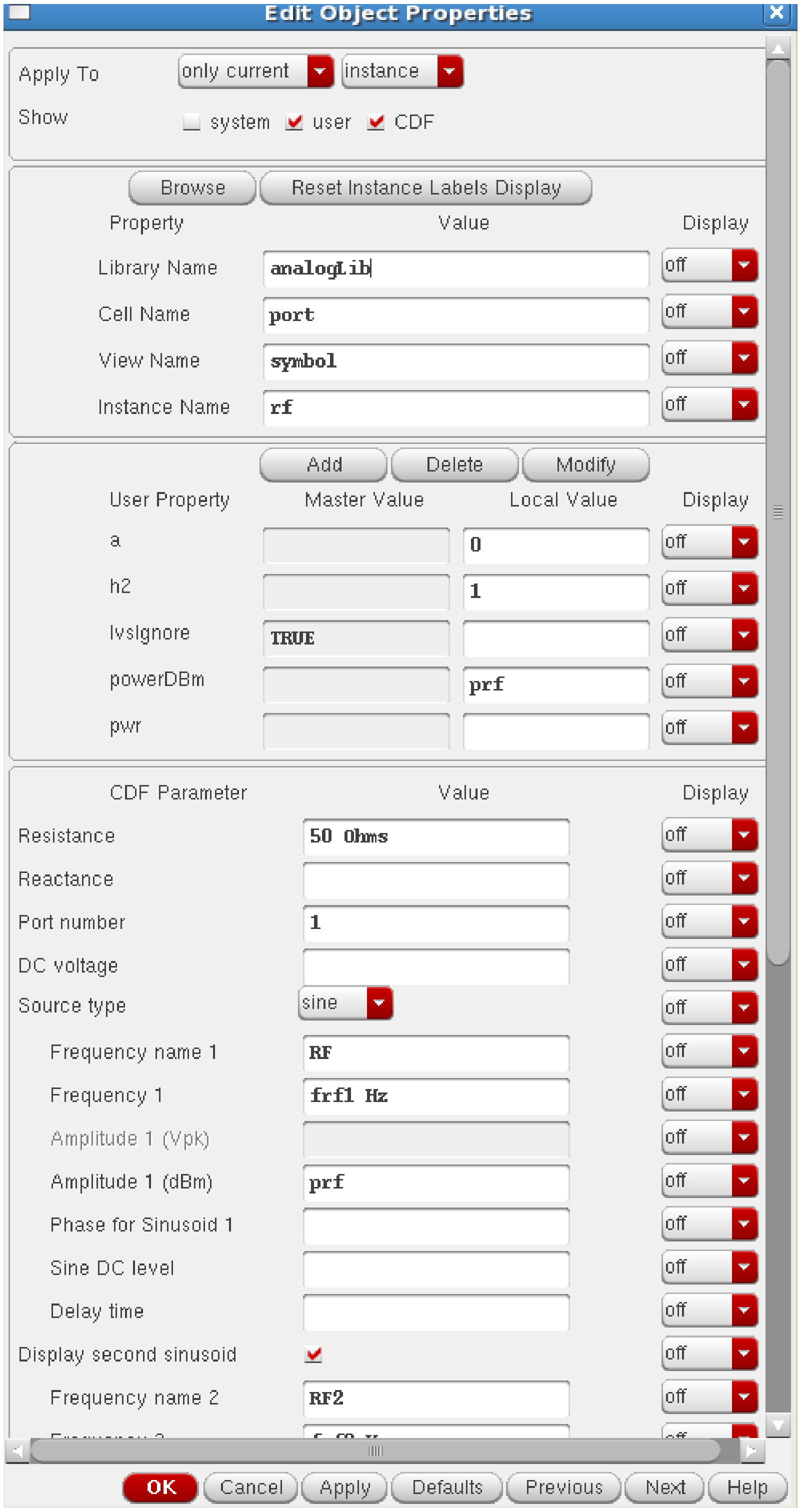
PXF Normal Conversion Gain Measurement

Consider the following circuit:



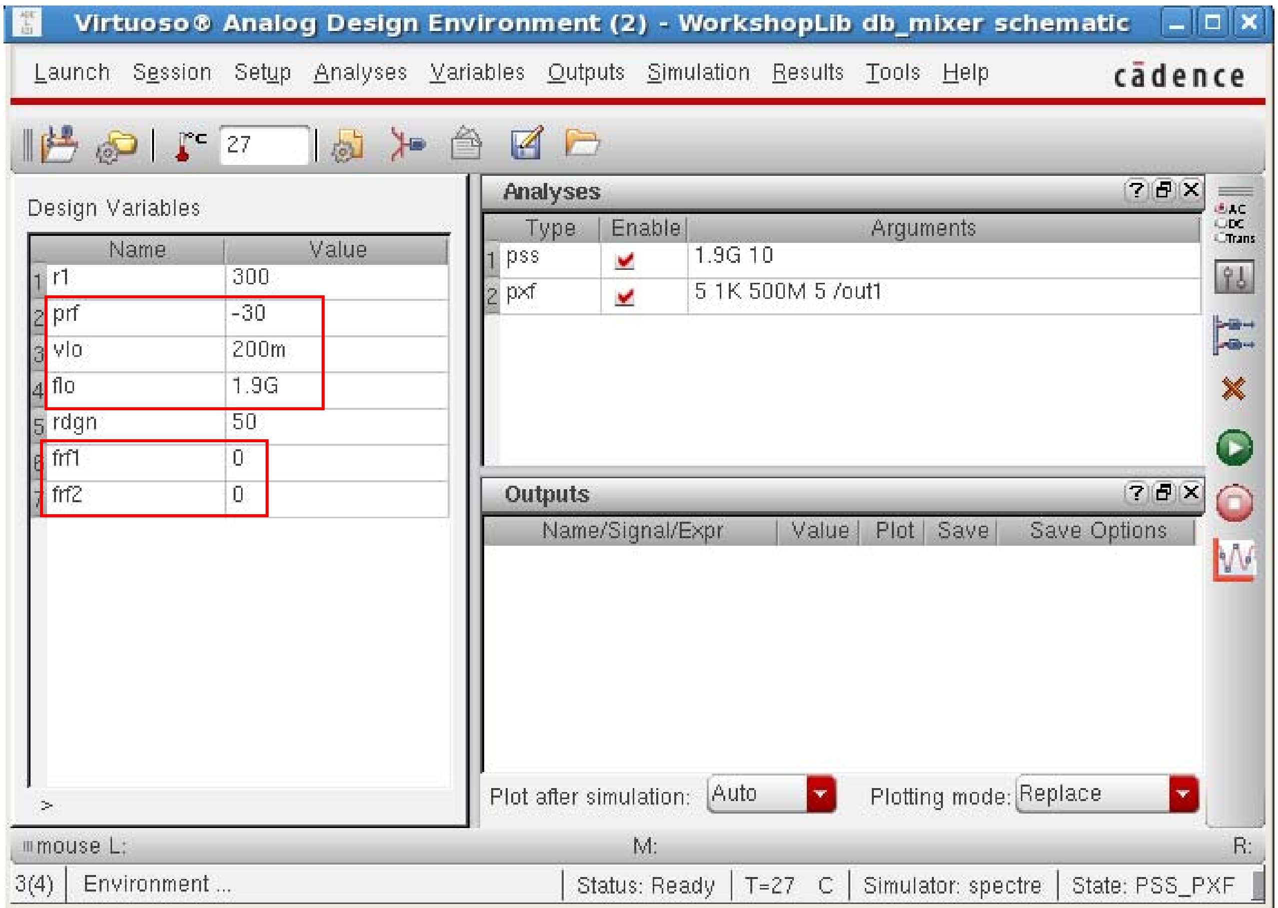
The circuit is a double-balanced down-conversion mixer with a 1.9GHz LO.

The *Edit Object Properties* form for the input port is shown below.



Using variables to set the frequencies and amplitudes is suggested because you can change the circuit setup by changing variable values in ADE without needing to change the circuit itself.

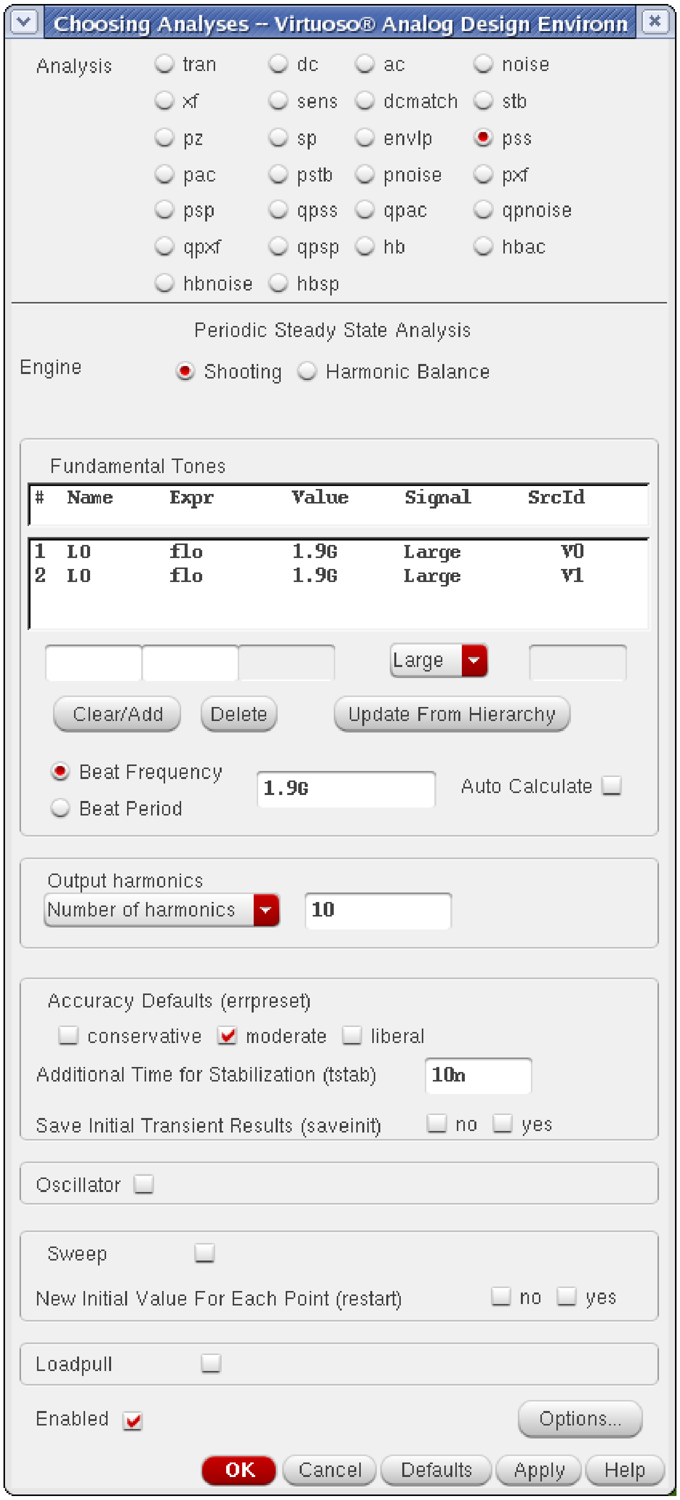
ADE has the variables set to appropriate values, as shown below.



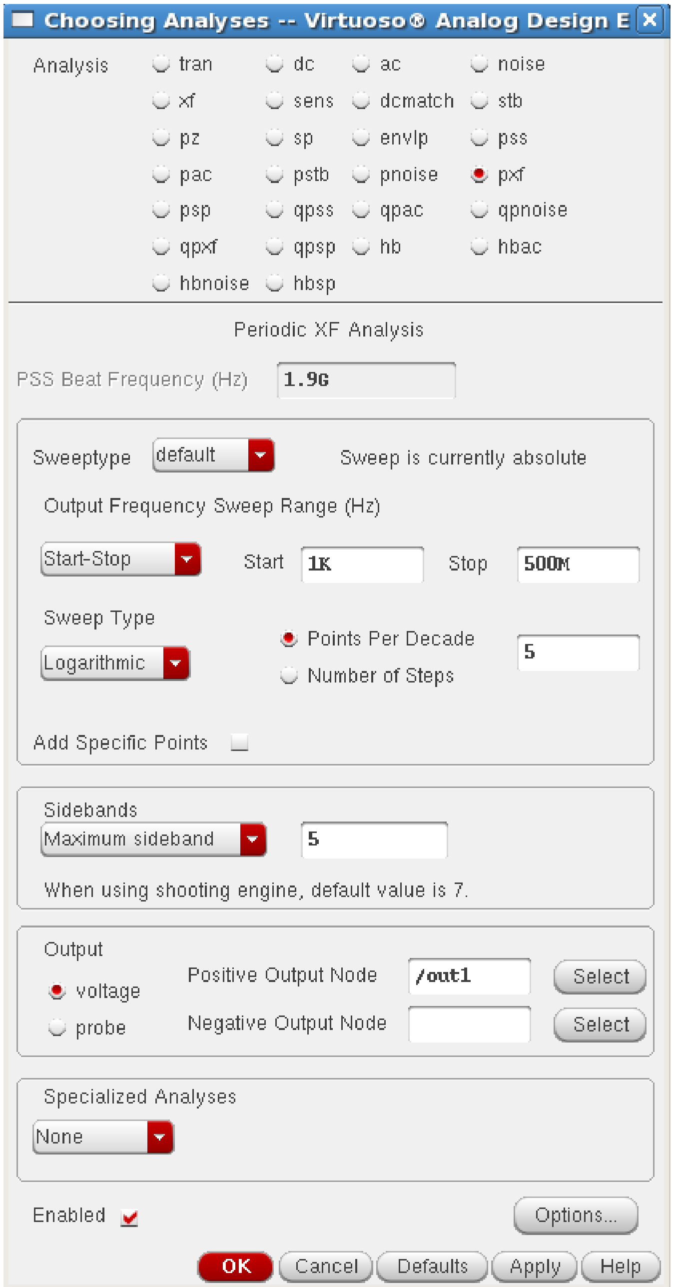
This setup causes the LO to be driven, and both RF inputs to be turned off for the pss analysis by setting the RF frequencies to zero. You can disable a signal for the pss analysis by setting its frequency to zero.

Using PXF Analysis for Conversion Gain Measurement

Set up the pss analysis with just the large signal that causes the frequency translation to occur, which is generally the LO signal or the sample clock. For more information, refer to *Overview of Periodic Steady-State (pss) Analysis* section at the beginning of this chapter. Regarding whether to use shooting or harmonic balance, pick the analysis that produces the smallest runtime or best convergence. In general, shooting is recommended for systems that have sharp edges, and harmonic balance is recommended for systems that are more sinusoidal.



Set up the PXF analysis, as follows.



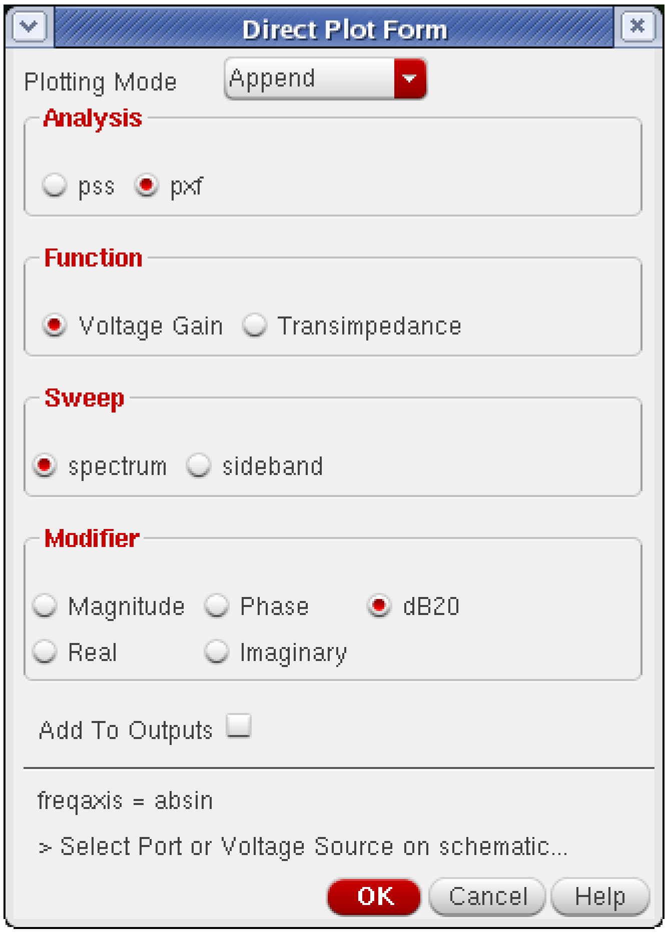
1. Specify an appropriate **output** frequency range for your circuit.
2. Generally, either linear or log spacing should be used. Auto usually takes more frequency point than are required, and thus extends the runtime.

3. For the pxf analysis, as a general recommendation, set the *Maximum sideband* term to the same value you uses for the maximum number of pss harmonics. Alternatively, the *Maximum sideband* field could be set to the highest harmonic of the pss analysis that mixing products are desired to be calculated. This is generally determined by the application. If you only need one or two output mixing product, and you are using shooting in the pss analysis, choose *Select from range*, and highlight the sidebands you want to calculate. Setting *Maximum sideband* for pxf is usually not critical as long as enough sidebands are calculated to measure the desired input frequency and forward conversion gain (called a sideband). The *Direct Plot Form* allows the plotting of the individual input sidebands as will be shown later.

4. Set the output nodes, as shown in the example. If the *Negative Output Node* field is left blank as shown above, the global ground node is used by the simulator for the reference node. If you have an output current, add an iprobe from analogLib in series with the output, and click probe in the output section. Then specify the iprobe in the *Output Probe Instance* field.

5. Now run the simulation.

6. After the simulation completes, select *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed.



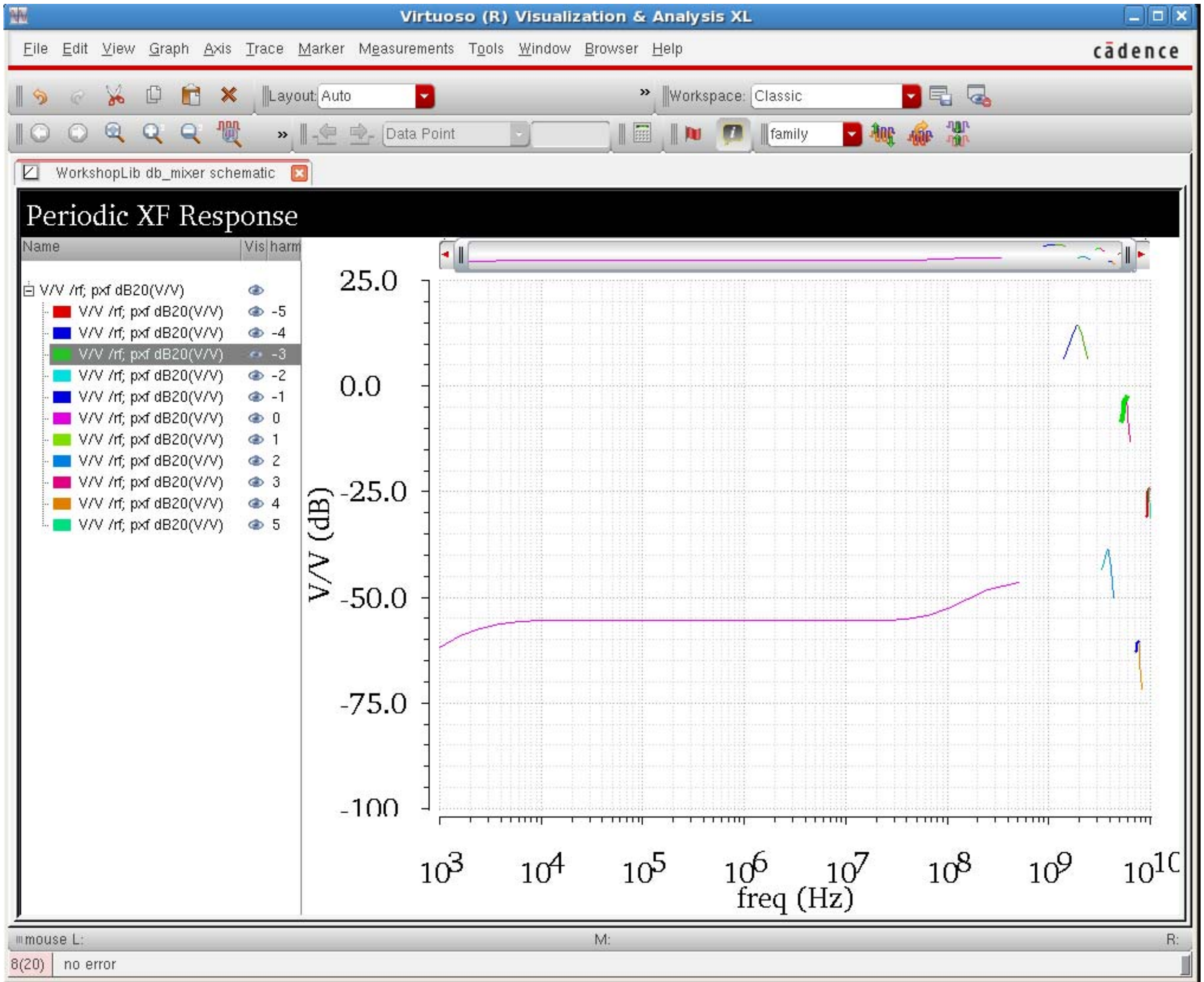
In the *Direct Plot Form*:

7. Select *pxf* from the *Analysis* section.

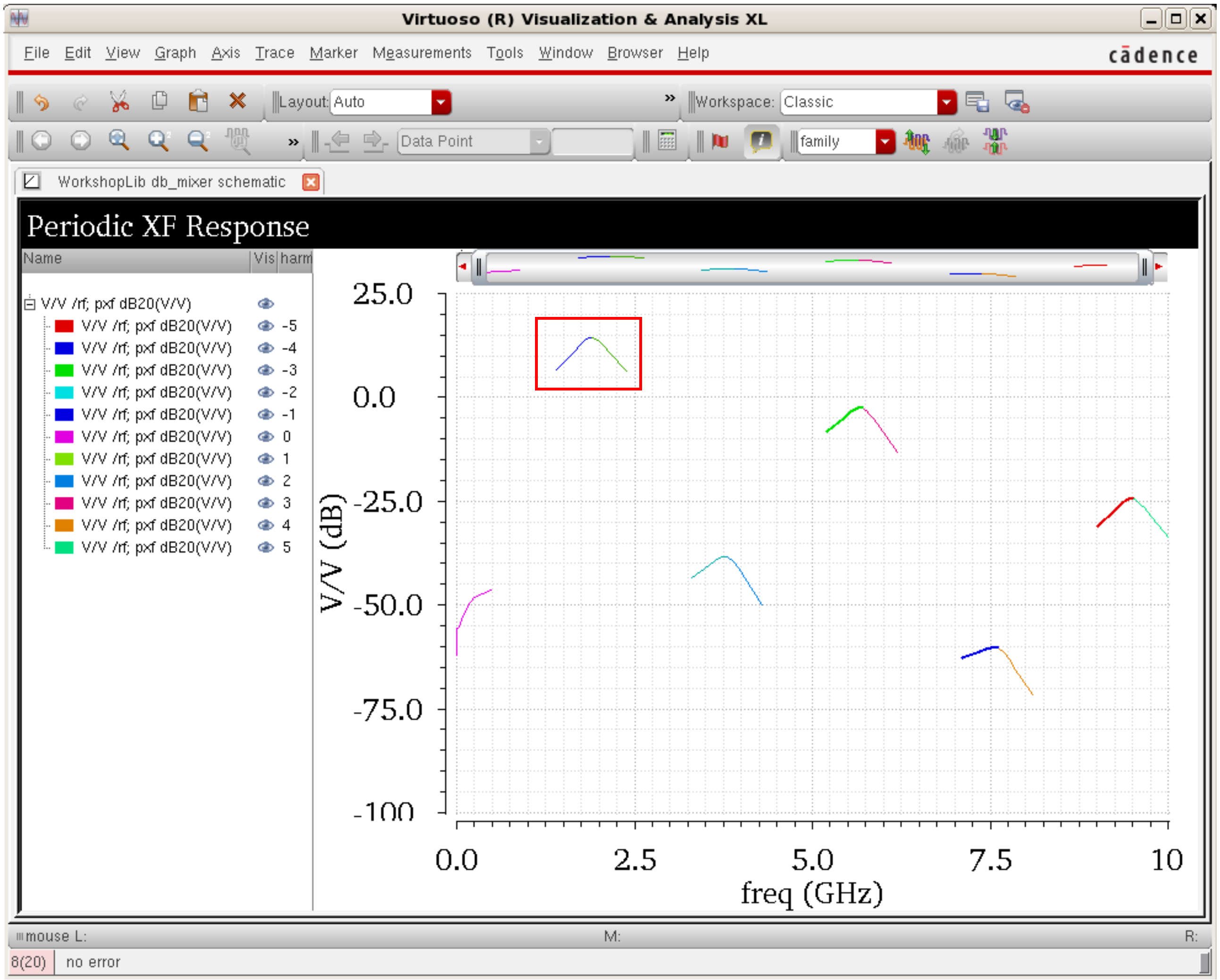
8. Select *dB20* from the *Modifier* section.

9. Select the voltage or current source from the schematic. Note that when you plot pxf results, you are selecting an instance, not a node. You will get the conversion gain results for the source you select in the circuit. The example below shows the conversion gain from the input source in the circuit.

The waveform window is displayed, as shown below.

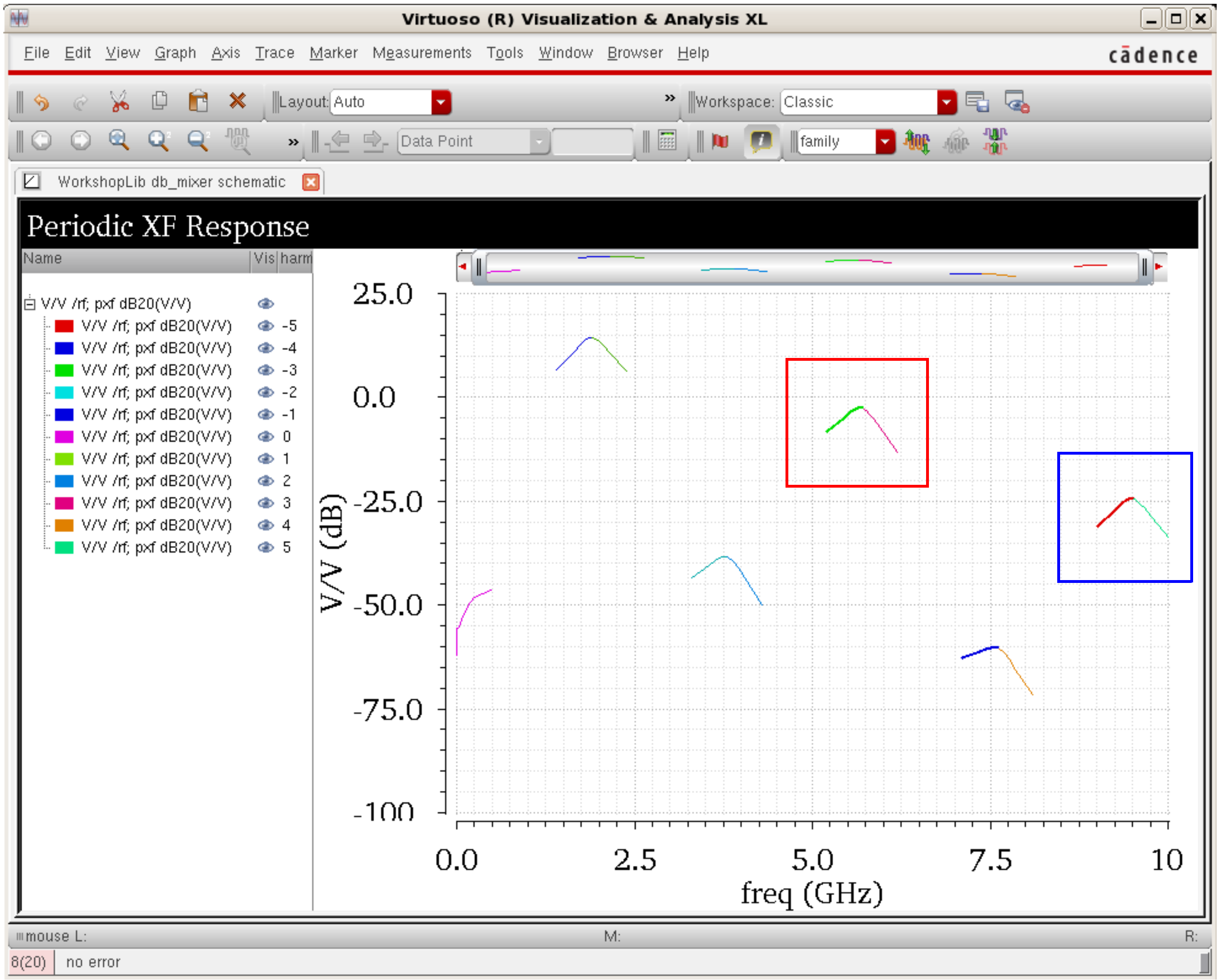


Since the *Sweeptype* was *log*, the data is displayed with a log axis in the waveform tool. In some cases, a linear X Axis is easier to read. To accomplish this, move the mouse cursor over one of the numbers on the X Axis and right-click. Move to Log Scale and click to deselect log. This is shown below.



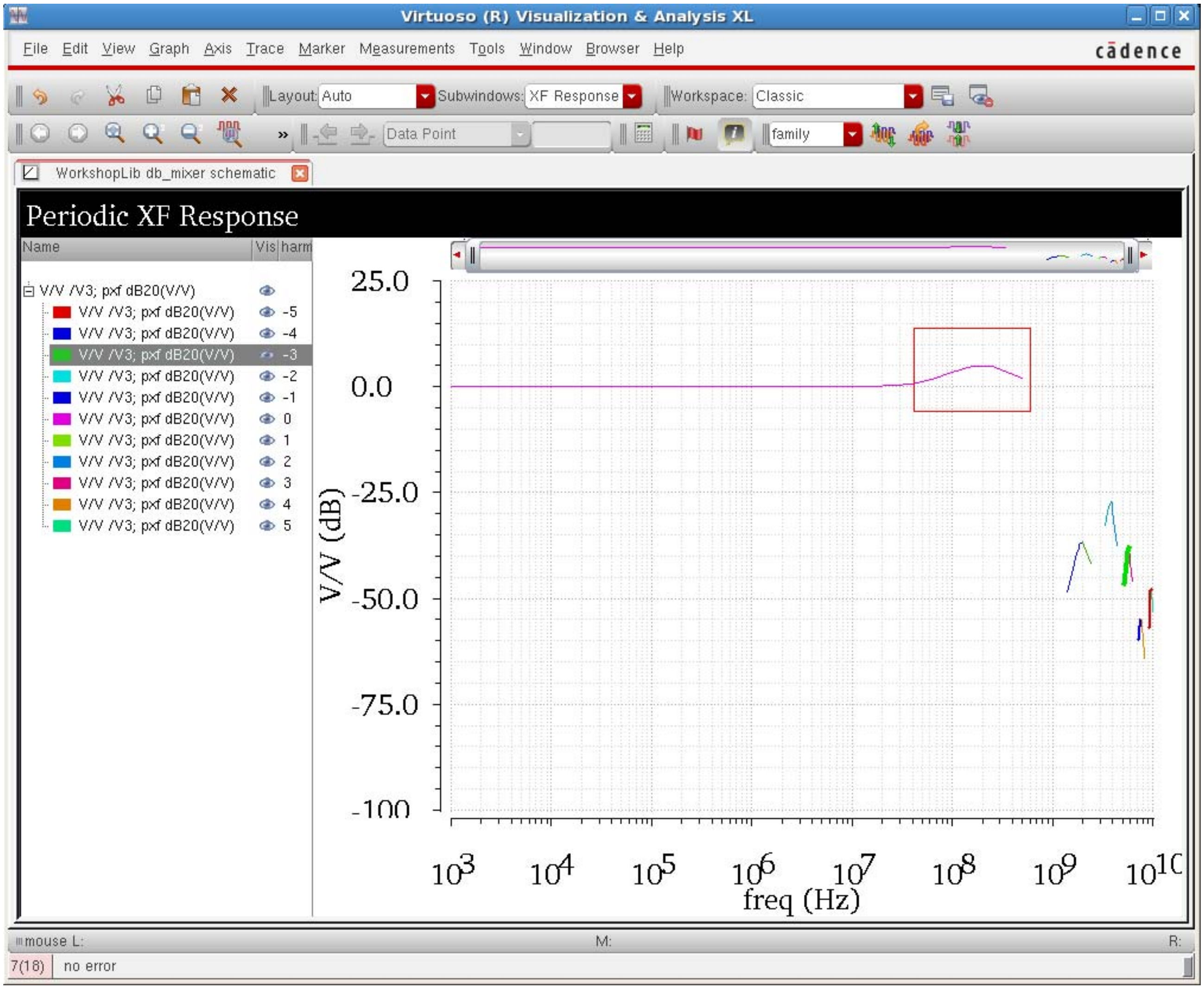
To read the conversion gain, move to the desired **input** frequency, and read the gain off the Y axis. The gain at 1.91GHz is about 15dB. The place to look is in the red rectangle above. Alternatively, position your mouse cursor on the trace at the input frequency, and read the value in the cursor readout. You can also place a marker at the input frequency to read out the conversion gain. To do this, move your mouse cursor near the trace at the input frequency, and type m.

10. The conversion gain near the third and fifth harmonics of the LO is also reasonable high as well. The gain near the third harmonic is near -5dB, and the gain near the third harmonic is near -25 dB. See the red and blue boxes below.



11. In the *Direct Plot Form*, select *New Win* from the *Plotting mode* drop-down list and select the DC source that is supplying power to the circuit.

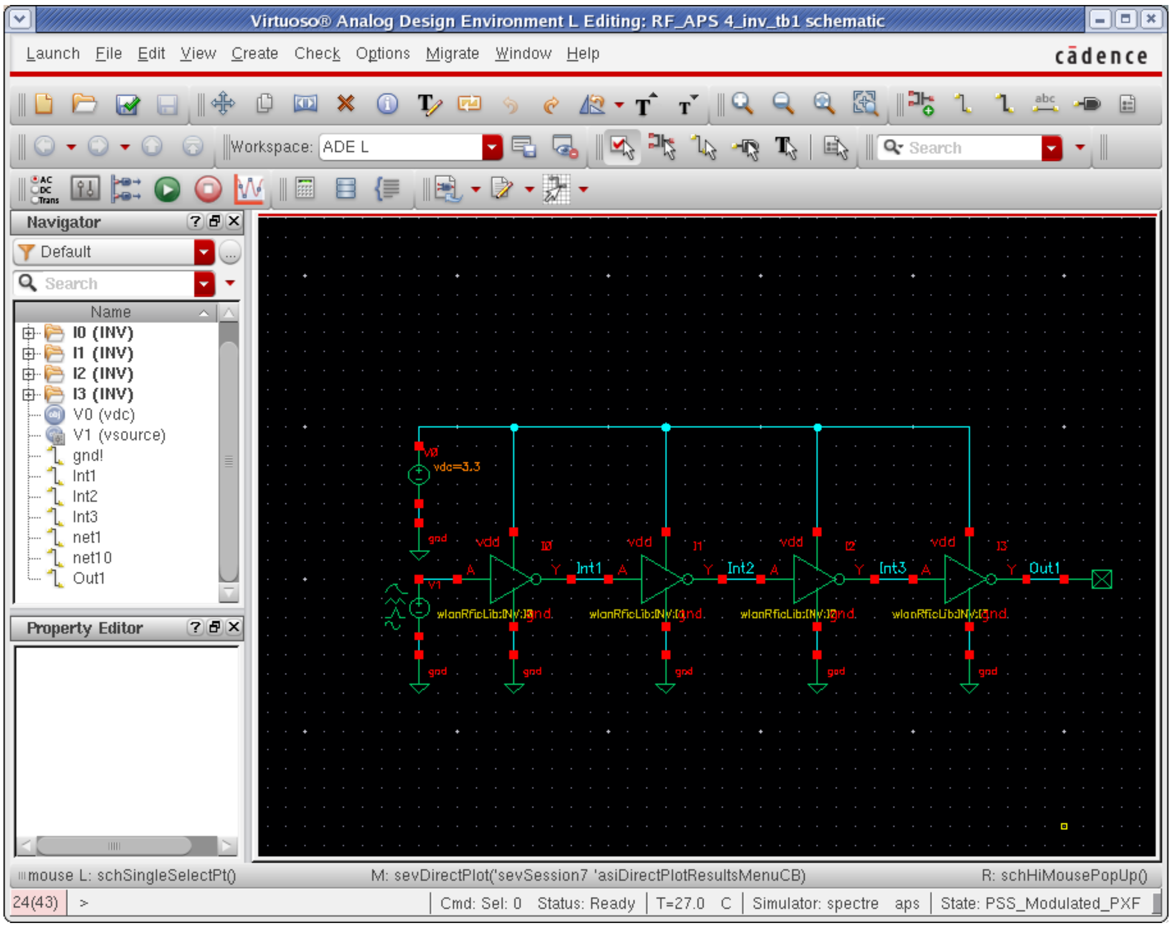
The waveform window is displayed, as shown below.



Note that the gain is about +5dB at 150MHz. This circuit is a potential oscillator.

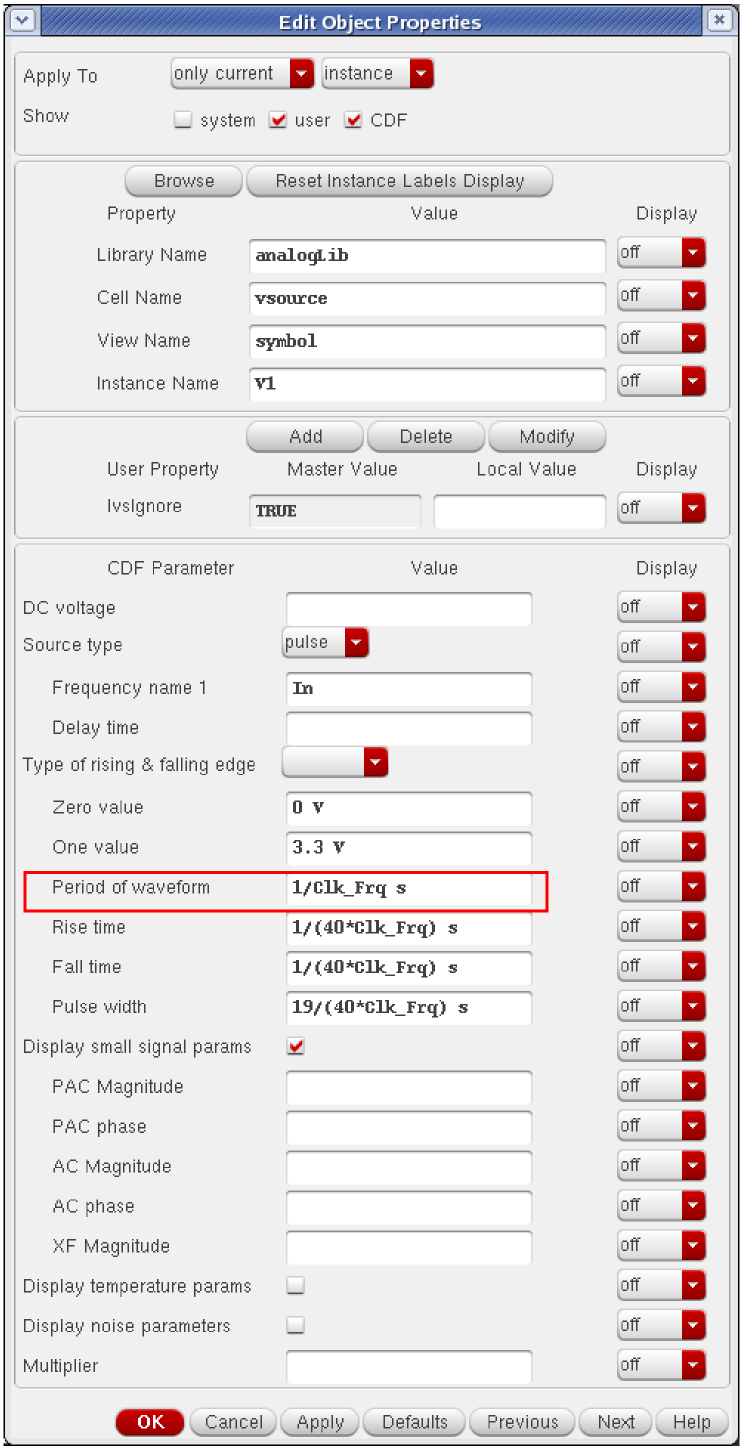
PXF Modulated (AM to PM conversion)

Consider the digital circuit below.

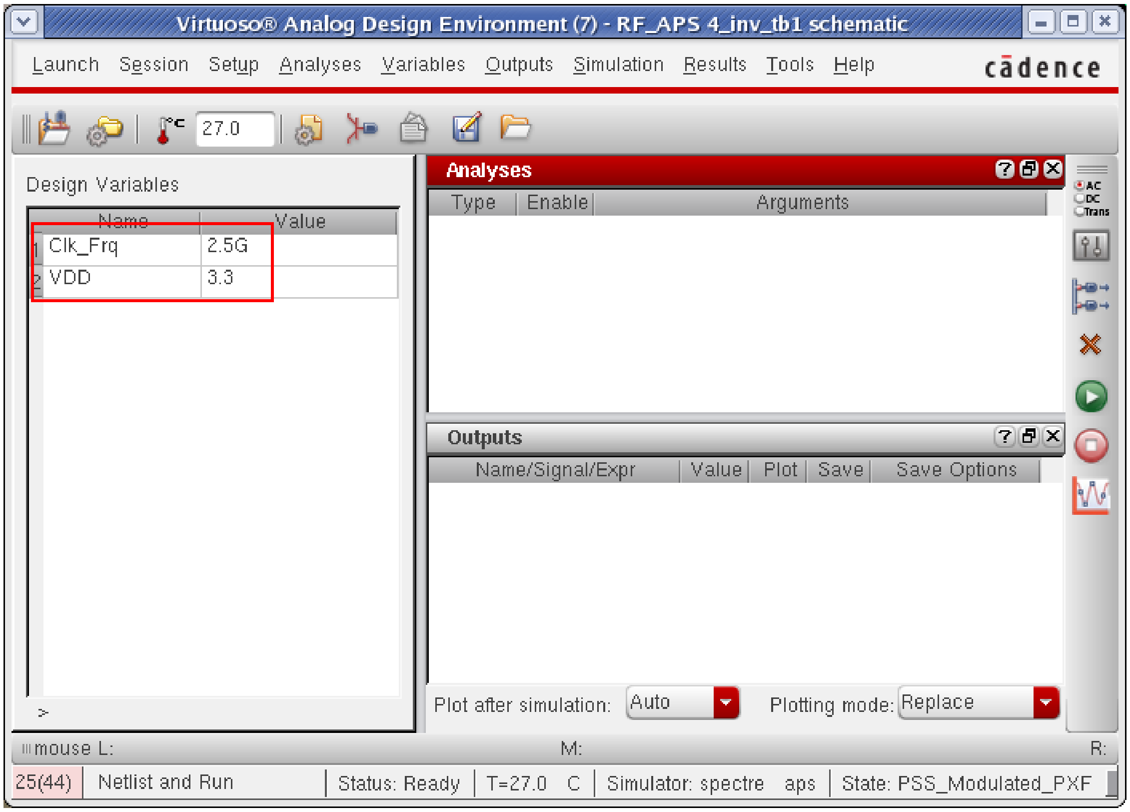


1. The input source has variables to define the frequency. This allows the frequency to be changed form the ADE environment without needing to change the circuit. The properties list is shown below. The rise and fall times are set to a small fraction of the period of the signal. This is done so the rise and fall times do not produce nonphysical, excessively fast

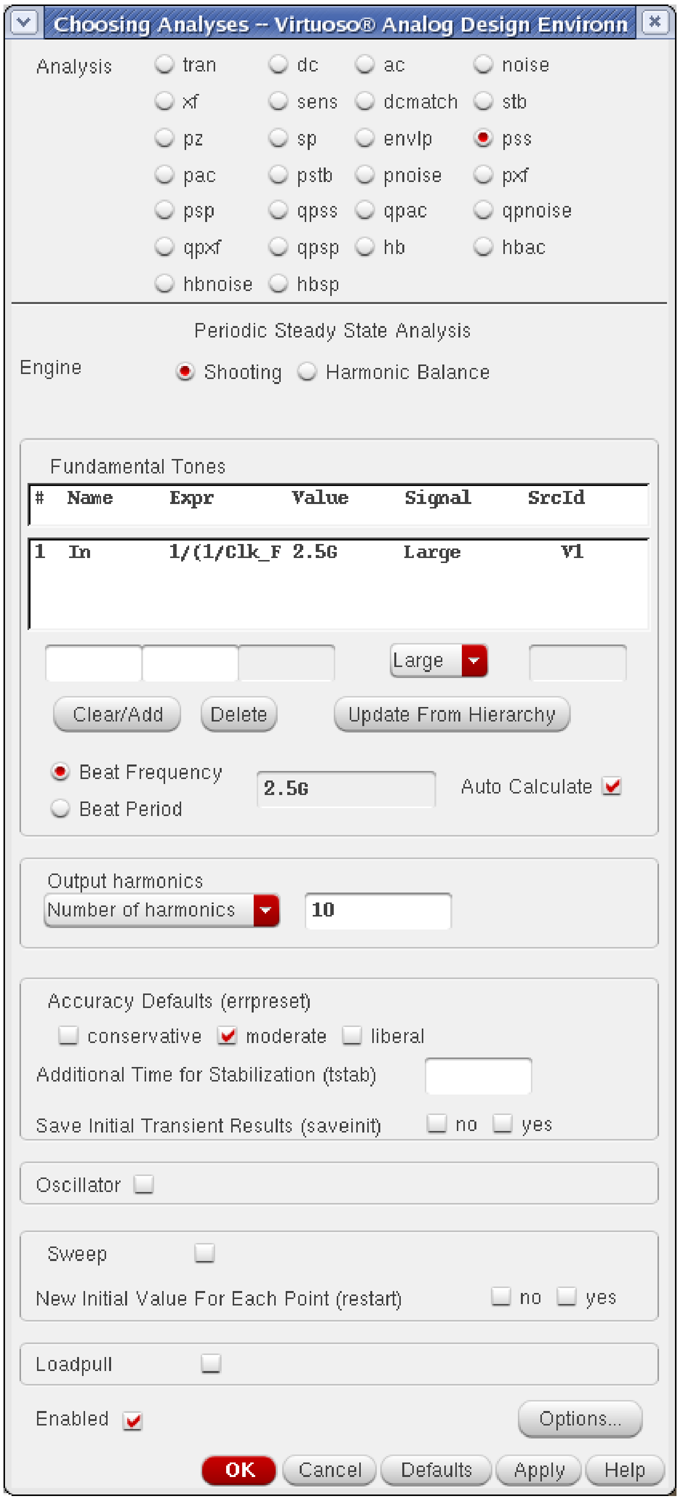
risetimes, which cannot be realistically produced in a circuit anyway. The pulse width is set to produce a 50% duty cycle square wave.



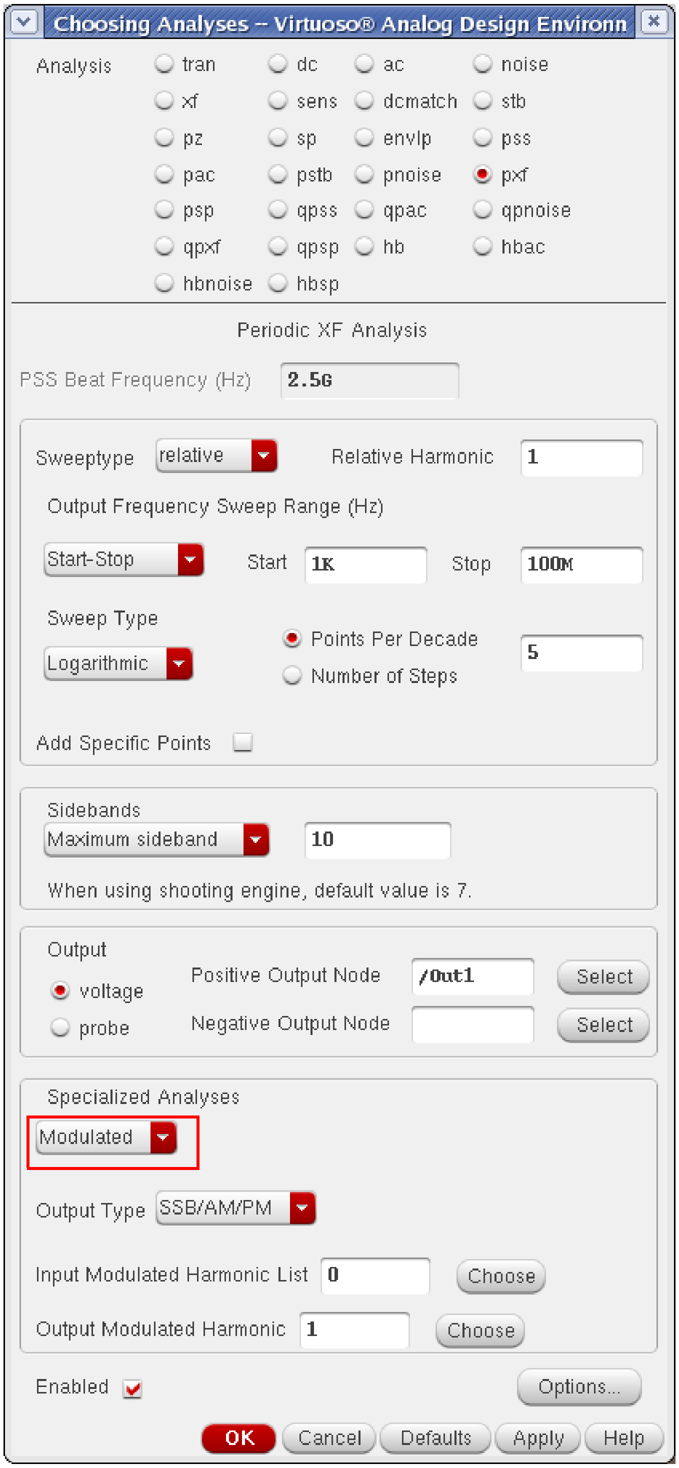
2. The default values for the variables are set in ADE. The input frequency is 2.5GHz.



3. To run a pxf analysis, first a pss analysis needs to be run. For more information, see the pss section at the beginning of this chapter.



4. Now set up the pxf modulated analysis.



5. First, select *Modulated* at the bottom of the *Choosing Analyses* form. This forces relative Sweeptype.

6. The objective is to measure AM to PM conversion on the 2.5GHz output when there is power supply ripple. Remember that in pxf, the frequency range is an **output** frequency range. Accordingly, the relative harmonic is 1, and the output frequency is analyzed just above the output at 2.5GHz.

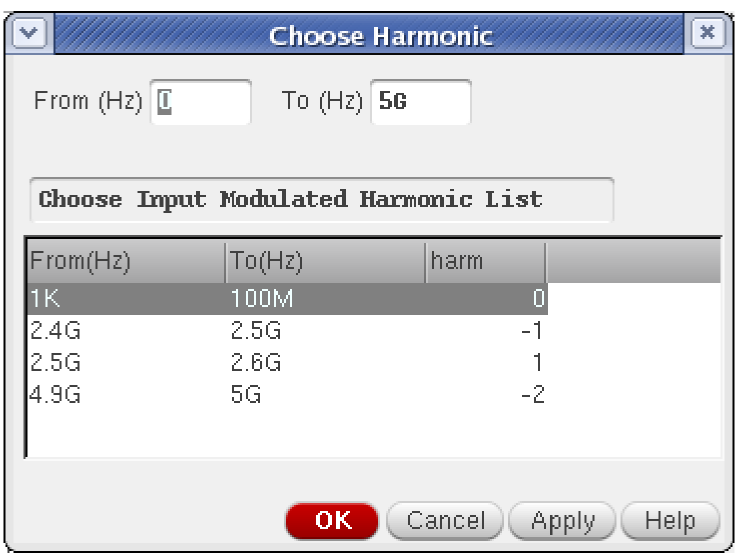
7. Usually, it is better to select linear or log spacing, and set your own sweep resolution. Automatic usually takes more points than are needed, thus extending the runtime.

8. Set the number of sidebands equal to the number of pss harmonics.

9. Specify an output node or nodes, as usual in pxf. If you have an output current, add an iprobe in series with the output, and click probe in the *Output* section. Then specify the instance name of the iprobe in the *Output Probe Instance* field.

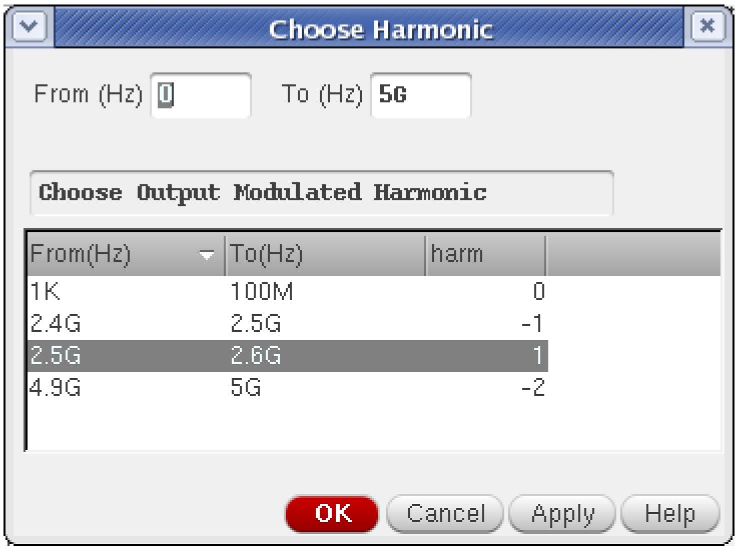
10. Always set the *Output Type* to *SSB/AM/PM*. This allows all the SSB/AM/PM to SSB/AM/PM components to be plotted.

11. Click the *Choose* button to the right of the *Input Modulated Harmonic List* field. Select the input frequency as shown below. This example measures the power supply ripple at low frequency to PM output near the 2.5GHz carrier, so the input frequency range is 1K to 100M.

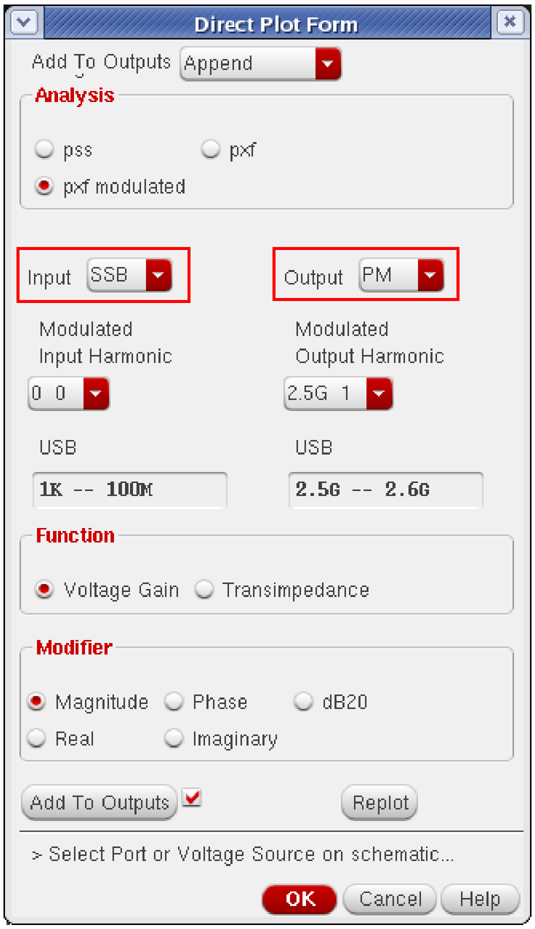


12. Click the *Choose* button to the right of the *Output Modulated Harmonic* field. Select the output frequency, as shown below. This example measures the power supply ripple

at low frequency to PM output near the 2.5GHz carrier, so the output frequency range is 2.5G to 2.6G.



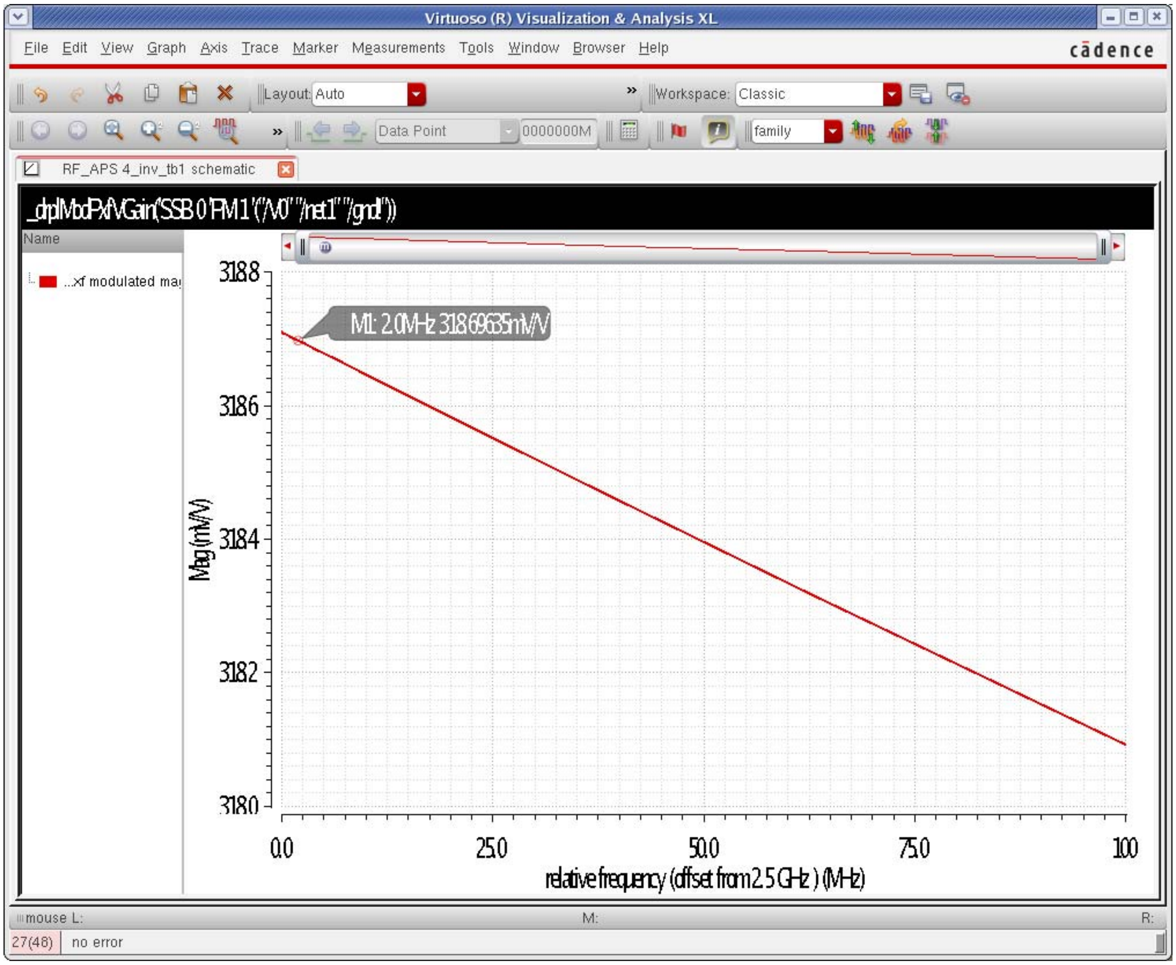
13. Click *OK*, and run the simulation. When the analysis completes, select *Results - Direct Plot - Main Form* from the ADE window.



In the *Direct Plot Form*:

1. Select *pxf modulated* from the *Analysis* section.
2. At the current time, to measure AM to PM conversion when the input is near zero Hz, the *Input* option should be set to *SSB*.
3. Set the *Output* option to *PM*.
4. Select *Voltage Gain* from the *Function* section
5. Select *Magnitude* from the *Modifier* section.

6. Click the DC source in the circuit that supplies power to the circuit. The measurement plots.

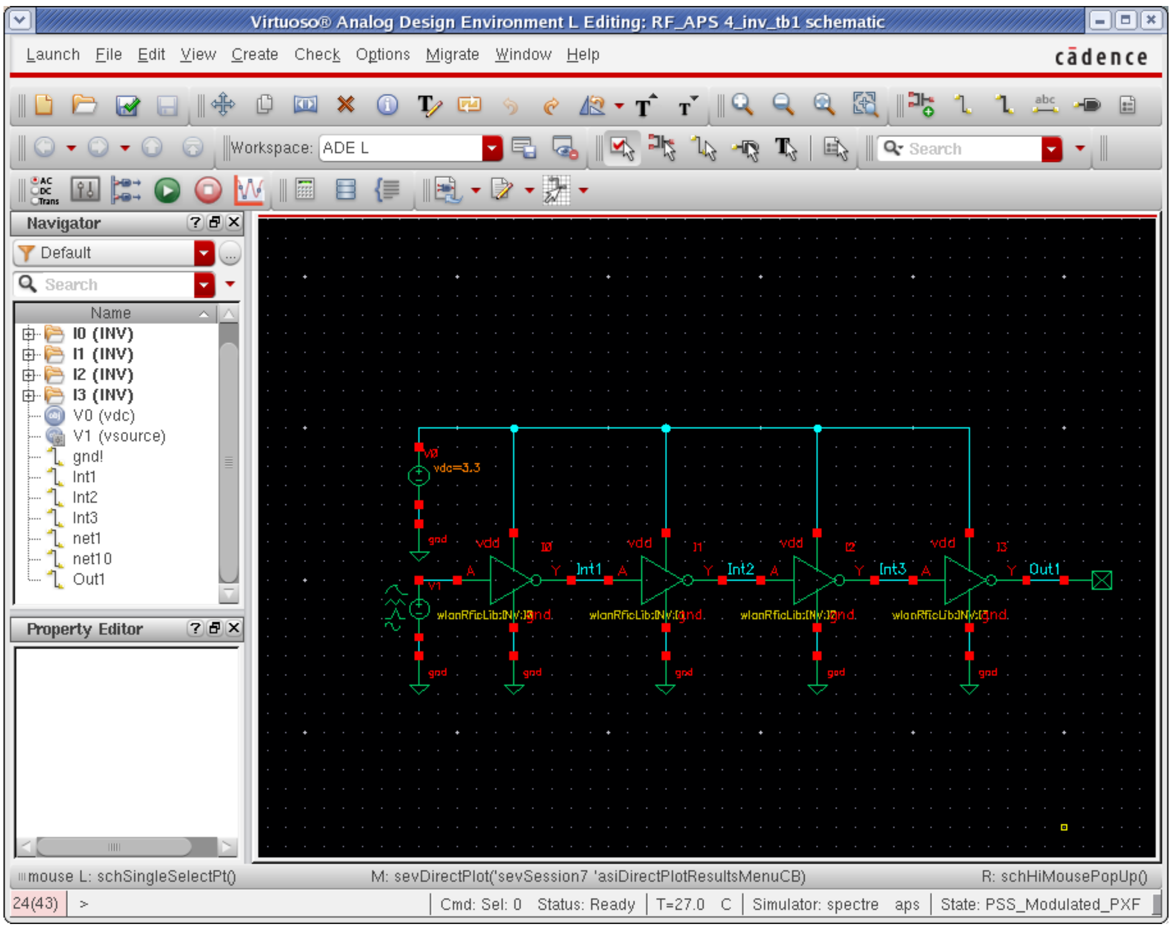


7. Although the vertical axis is labeled mV/V, it is actually plotting milliRadians of phase change per volt of power supply change. 318 milliRadians is about 18 degrees of phase change for a 1 volt change on the power supply. [(0.318/6.28)*360] Note that pxf is a small-signal measurement, and this applies only for small variations of the power supply voltage.

Sampled PXF

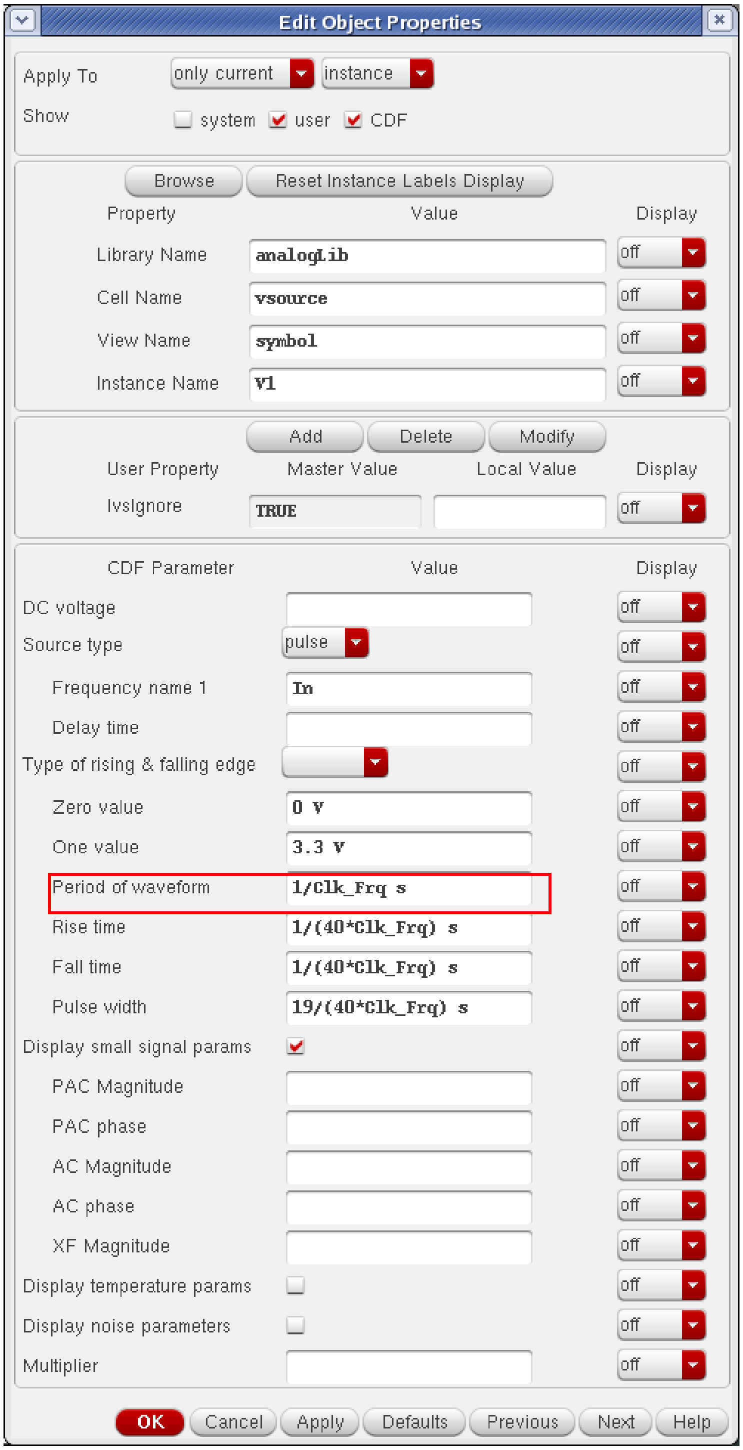
Sampled pxf is seldom used. It is provided so that conversion gain measurements from the power supply to a digital output that is periodic, and sampled at the threshold point every cycle. The typical threshold is the VDD/2 switching point for further logic circuits.

Consider the digital circuit below.

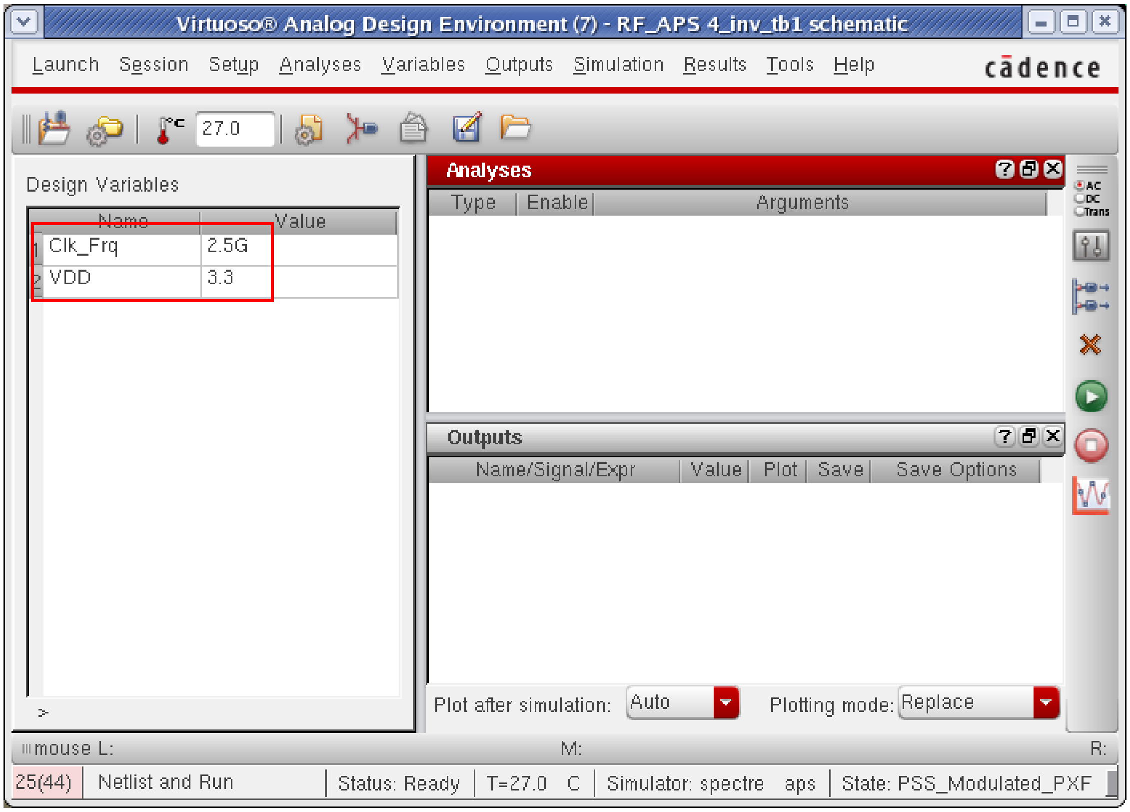


1. The input source has variables to define the frequency. This allows the frequency to be changed from the ADE environment without needing to change the circuit. The properties list is shown below. The rise and fall times are set to a small fraction of the period of the signal. This is done so the rise and fall times do not produce nonphysical, excessively fast

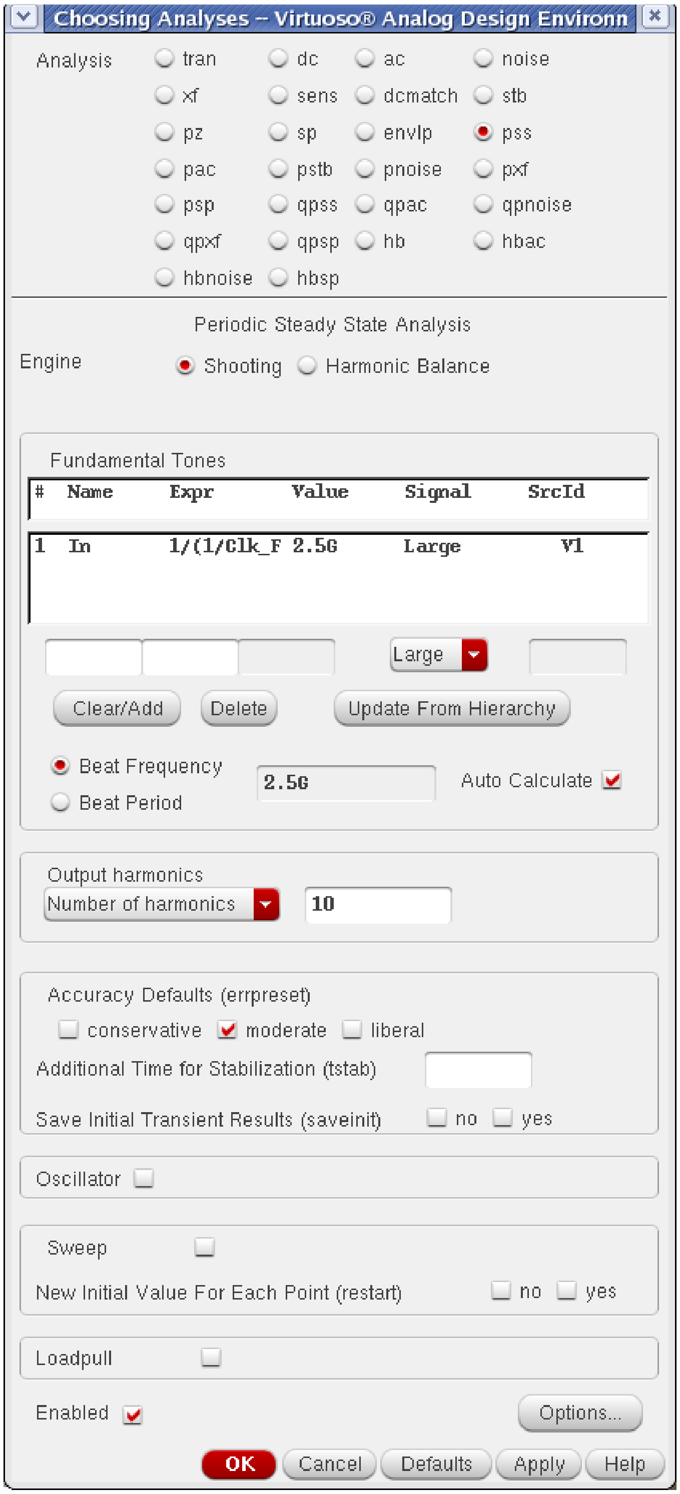
risetimes, which cannot be realistically produced in a circuit anyway. The pulse width is set to produce a 50% duty cycle square wave.



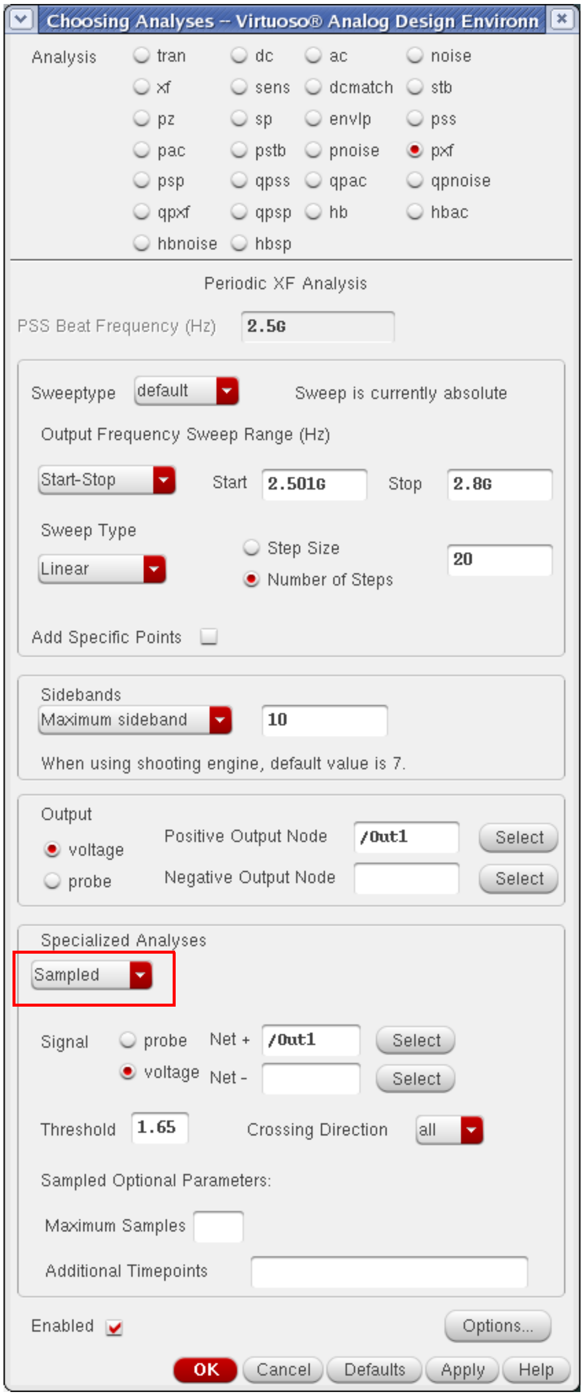
2. The default values for the variables are set in ADE. The input frequency is 2.5GHz.



3. To run a pxf analysis, first a pss analysis needs to be run. For more information, see the pss section at the beginning of this chapter.



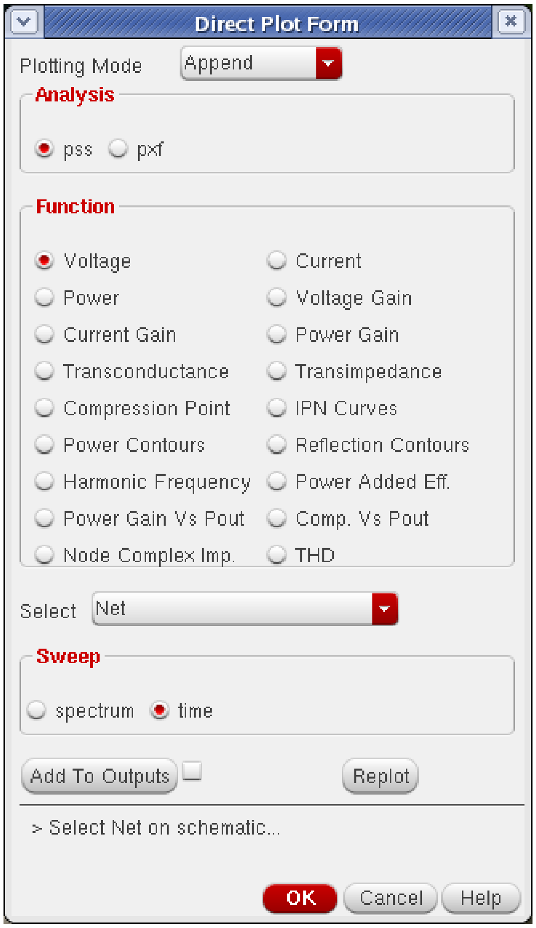
4. Now set up the pxf sampled analysis.



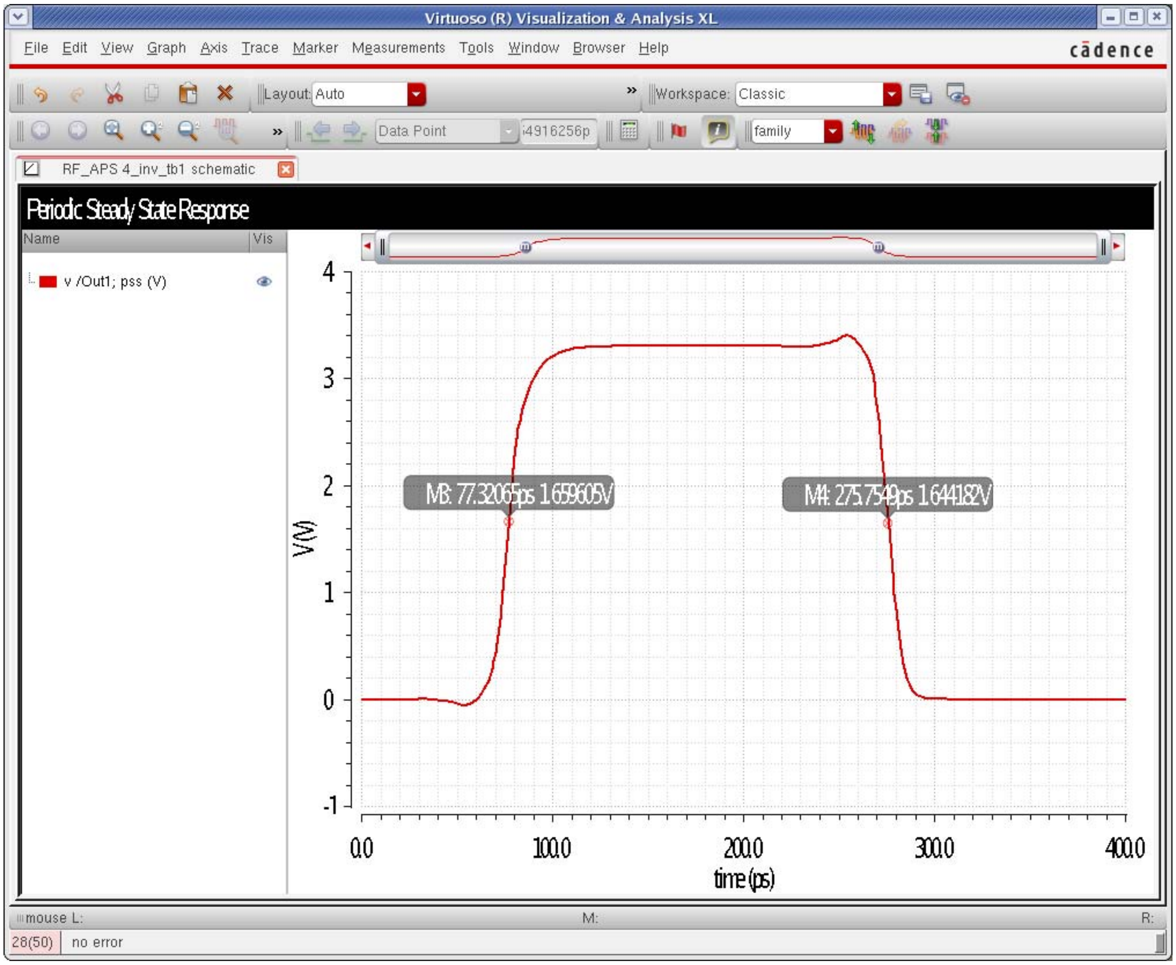
5. First, select *Sampled* at the bottom of the *Choosing Analyses* form.

6. The objective is to measure the conversion gain on the 2.5GHz output at the threshold crossing of the digital output when there is power supply ripple. Remember that in pxf, the frequency range is an **output** frequency range. Accordingly, the output frequency is analyzed just above the output at 2.5GHz.
7. Usually, it is better to select linear or log spacing, and set your own sweep resolution. Automatic usually takes more points than are needed, thus extending the runtime.
8. Set the number of sidebands equal to the number of pss harmonics.
9. Specify an output node or nodes, as usual in pxf. If you have an output current, add an iprobe in series with the output, and click probe in the *Output* section. Then specify the instance name of the iprobe in the *Output Probe Instance* field.
10. Click *Select* to the right of the *Net+* field and select the output net in the schematic.
11. Type in the threshold voltage. In 3.3 volt logic, 1.65 volts is appropriate.
12. Choose the crossing direction. Rising, falling, and all crossing events are selectable. This example shows all, which will calculate the transfer functions for both the rising and falling edges.
13. If you want specific times in the pss waveform to be measured, type those times into the *Additional Timepoints* field. Separate the times with a space. This is not shown in the example.

14. Click *OK* and run the simulation. When the analysis completes, select *Results - Direct Plot - Main Form* from the ADE window.

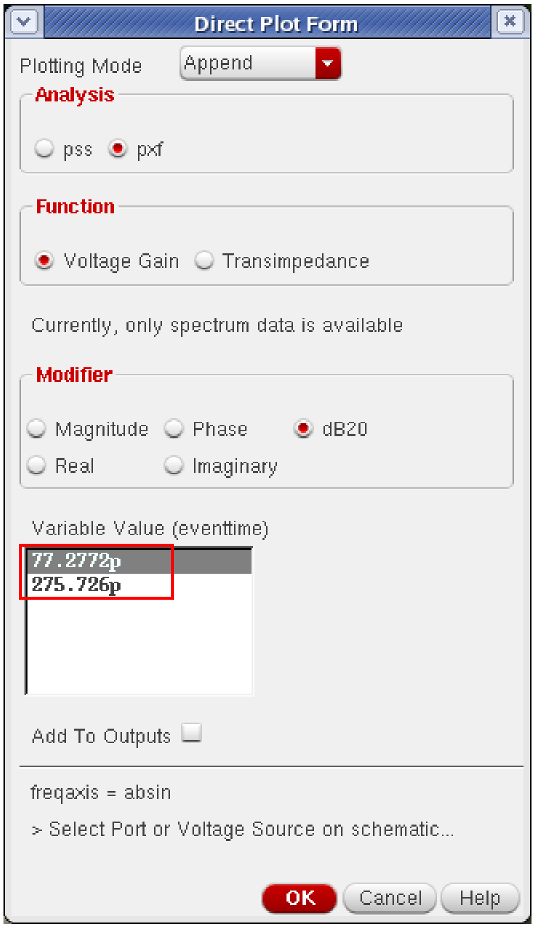


15. Select pss results, and plot the time-domain waveform. This is shown in the previous graphic. Plotting the waveform establishes which time corresponds to which edge.

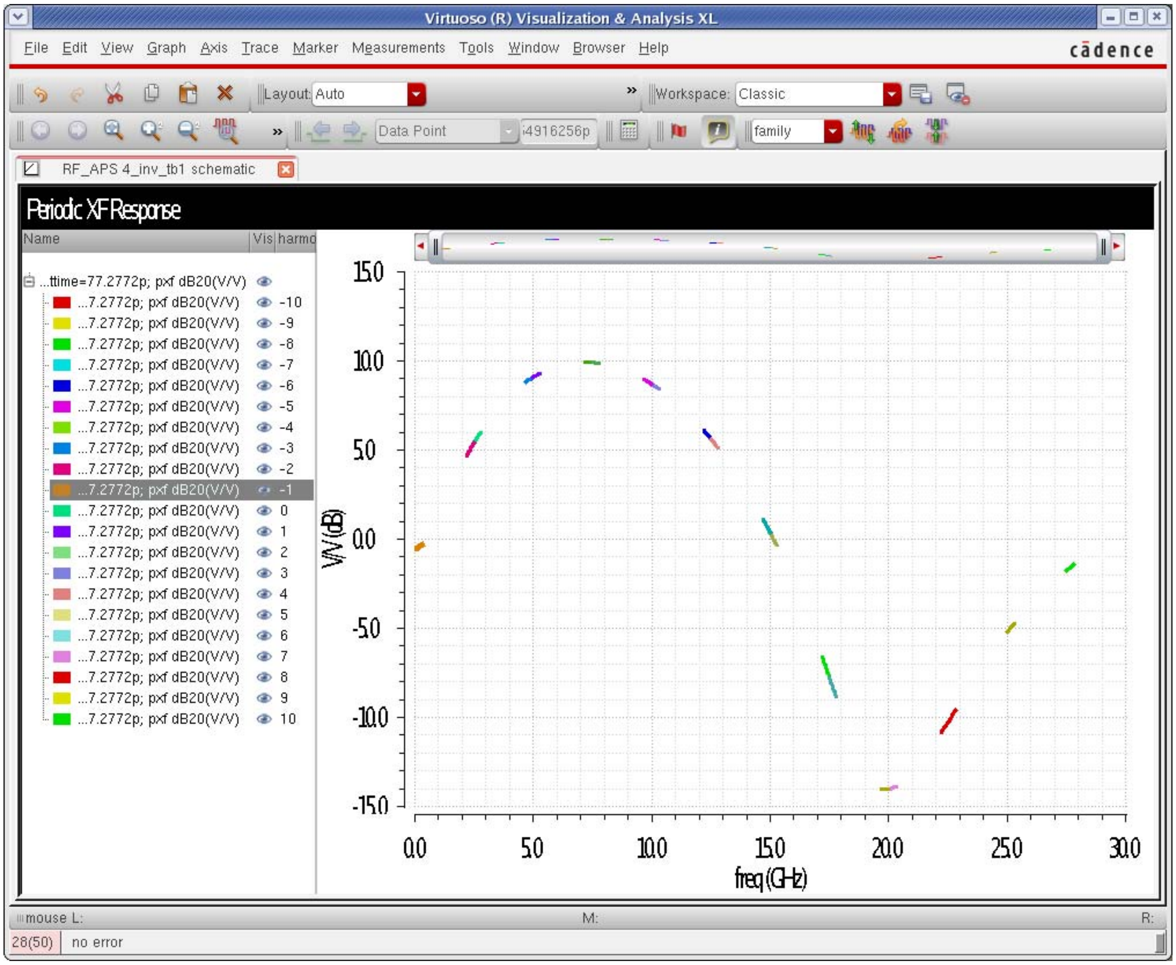


16. For this example, the rising edge is about 77nsec, and the falling edge is about 275nsec.

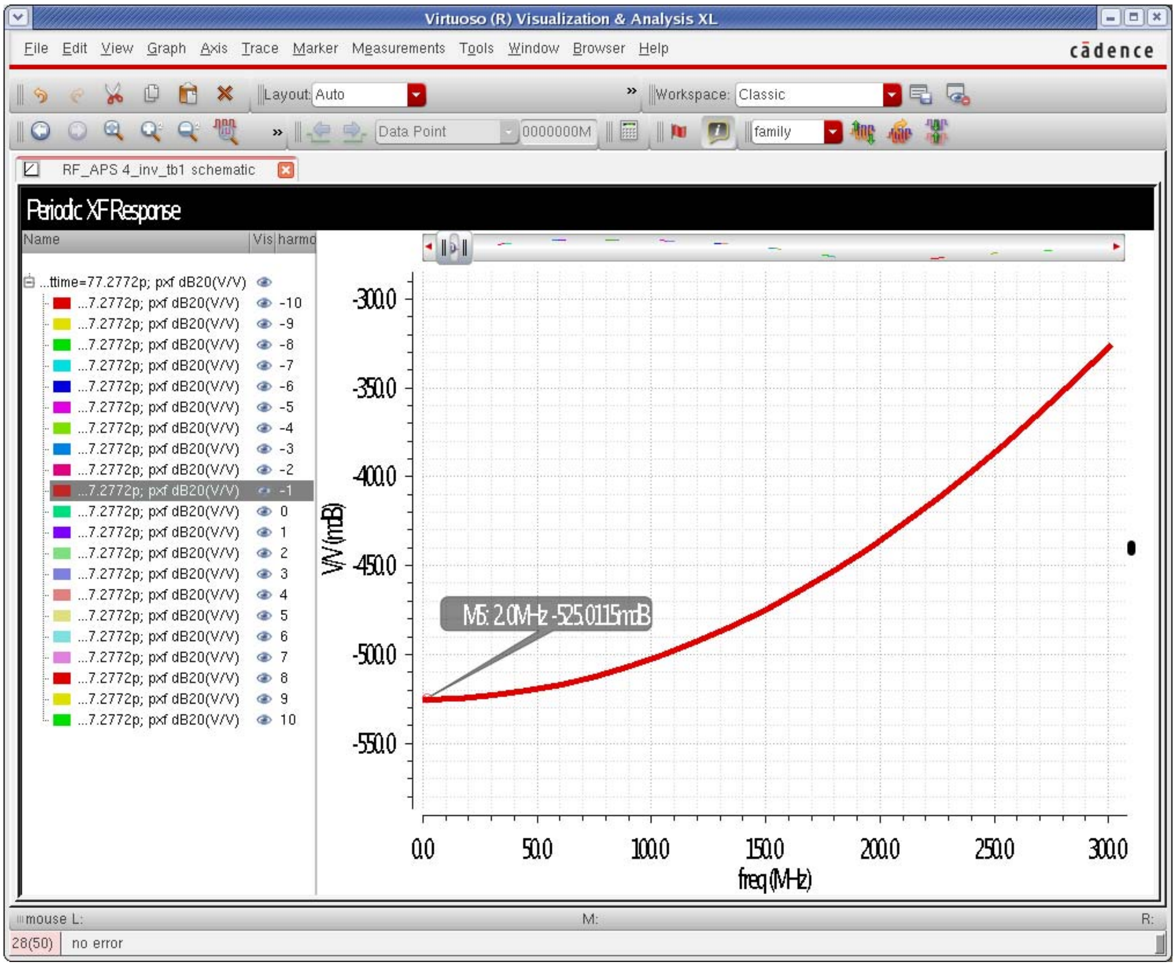
17. Now select pxf results. If desired, select dB20. Select the crossing time from the list at the bottom. The example shows the rising edge.



18. Click the DC source in the circuit that supplies power to the circuit. The measurement plots.



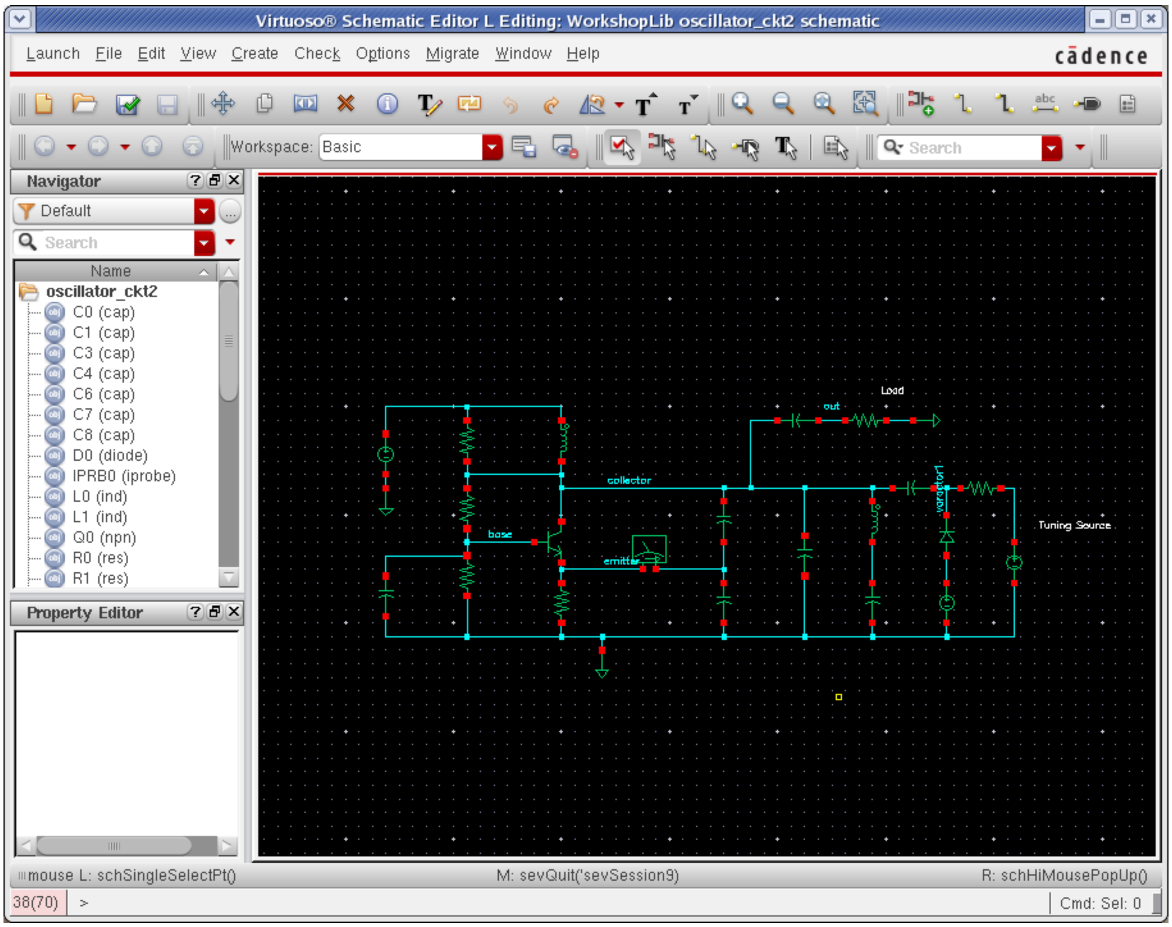
19. To measure the conversion gain, go to the **input** frequency on the plot, and make the measurement. Below, the plot is zoomed in to the results from low frequency, like power supply ripple. At 2MHz, the gain is about -0.5dB.



Examples for Oscillators

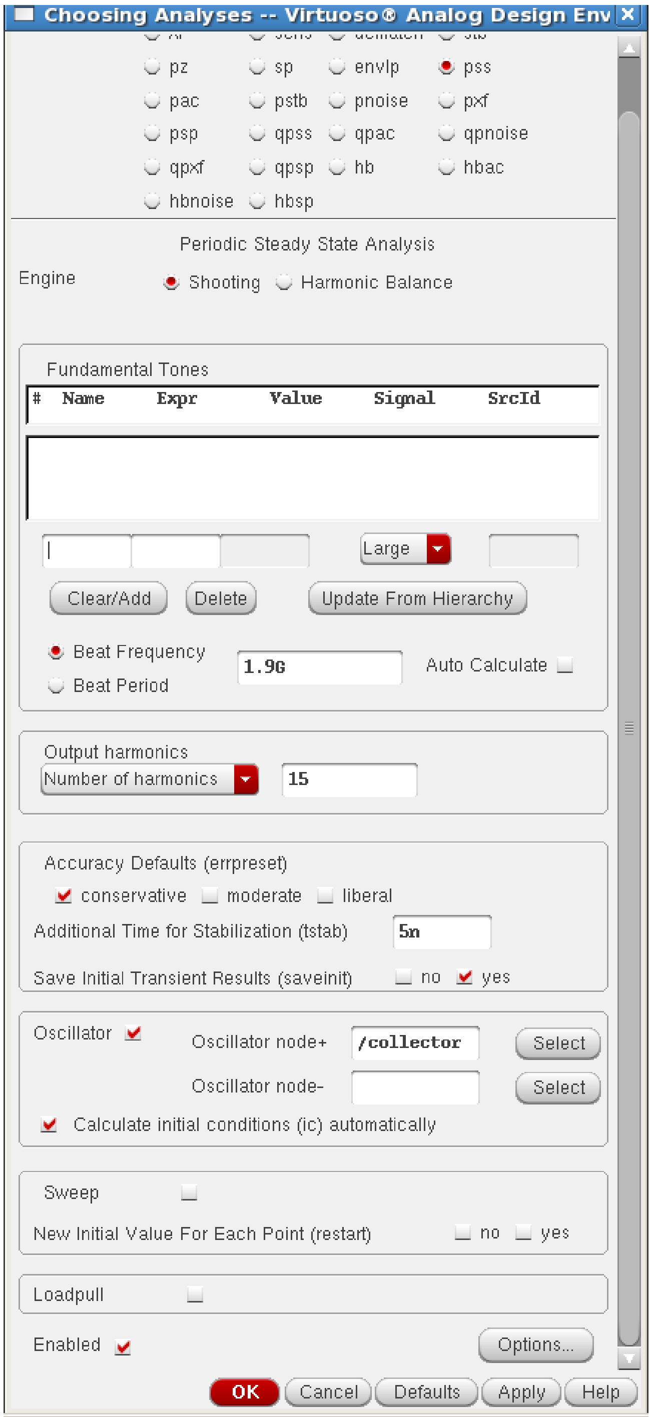
Conversion Gain From Power Supply to the Output Frequency

Consider the circuit shown below.

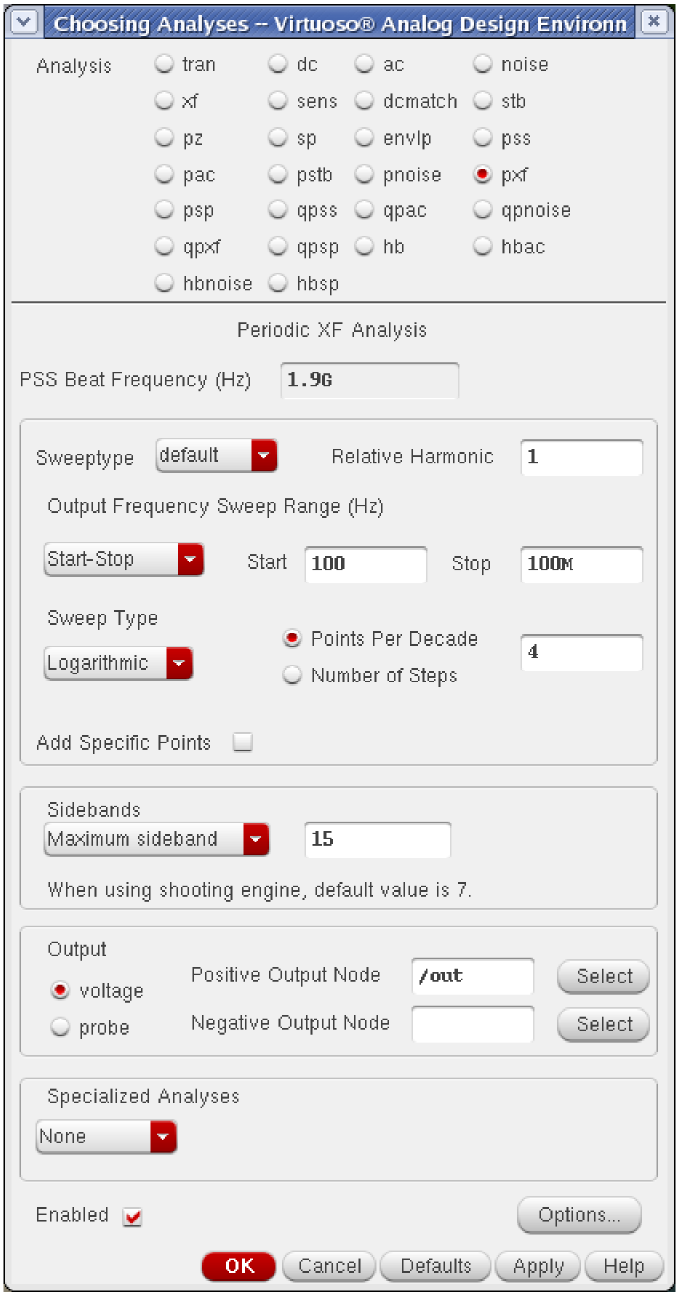


On the *Choosing Analyses* form:

1. Set up the pss analysis, as shown below. For more information, refer to *Overview of Periodic Steady-State (pss) Analysis* at the beginning of this chapter.



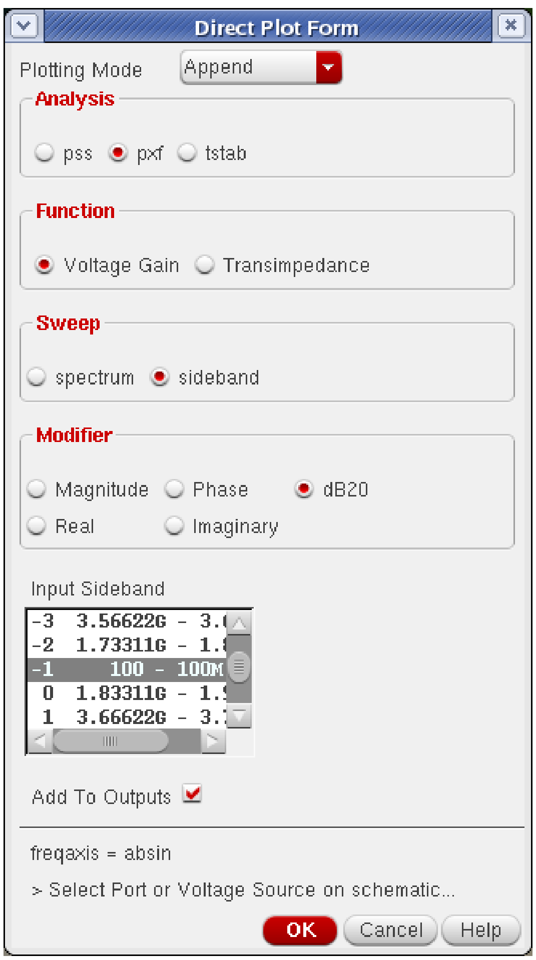
2. Set up the pxf analysis, as shown below.



3. R*elative* from the *Sweeptype* drop-down list is the default when an oscillator simulation is set in the pss analysis.

4. Specify the output harmonic in the *Relative Harmonic* field.

5. Set the frequency range to a frequency range that is relevant for your needs. In this example, the frequency range is from 100Hz to 100MHz above the first harmonic of the oscillator frequency.
6. Select *logarithmic* from the *Sweep Type* drop-down list.
7. Specify 3 to 5 in the *Points Per Decade* field.
8. Set the *Maximum sideband* option to the same value as you have set for the maximum harmonics in the pss analysis.
9. Click *OK*.
10. Run the simulation.
11. When the simulation completes, select *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed.



In the *Direct Plot Form*:

12. Select *pxf* in the *Analysis* section.

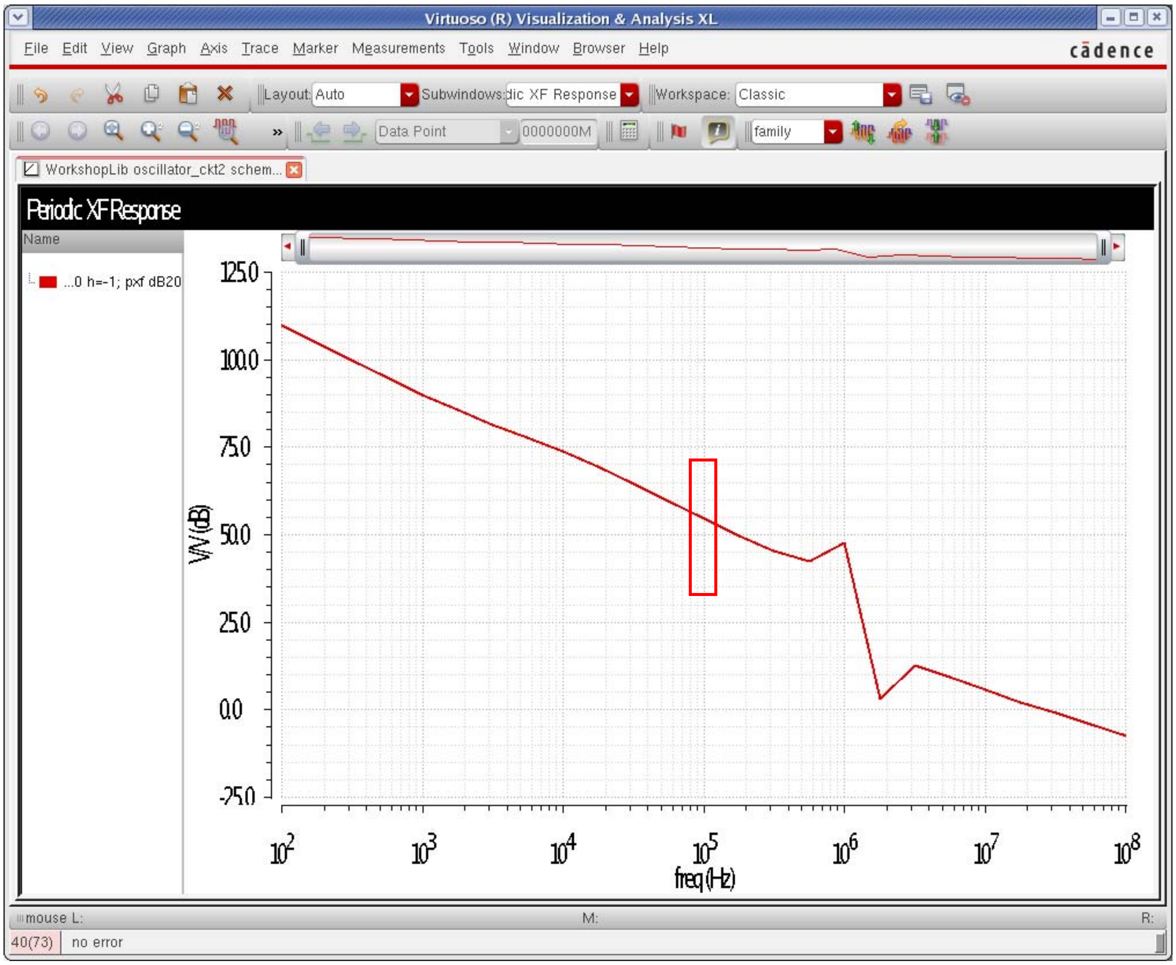
13. Select *sideband* from the *Sweep section*.

14. Select dB20.

15. Select your **input** frequency range in the list in the *Input Sideband* section.

16. Select the power supply source in the Schematic window.

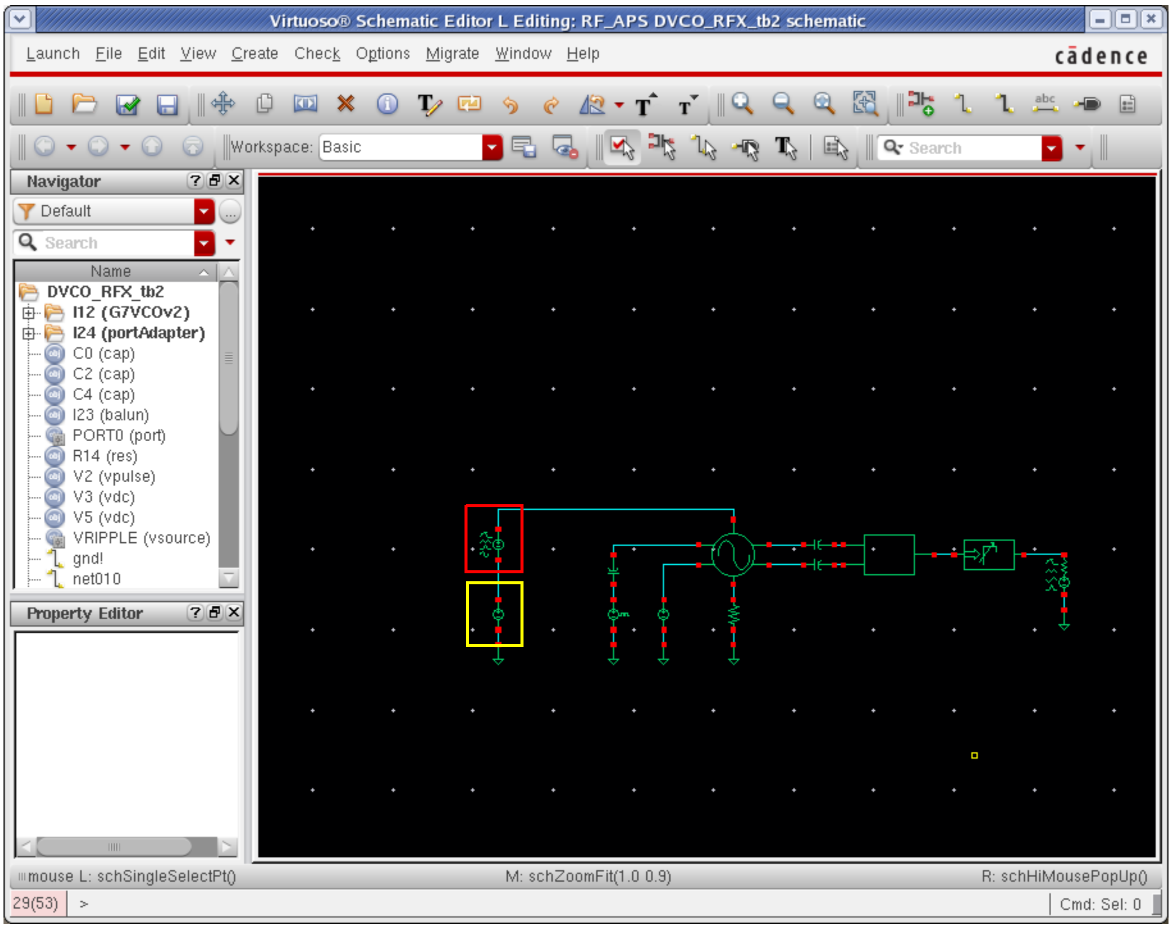
The waveform is plotted, as shown below.



To read the gain, go to the input frequency, and read the gain from the Y axis. The gain is about 55dB at 100KHz in this example.

Modulated PXF; Measuring AM to PM Conversion and Controlling Spurious Response

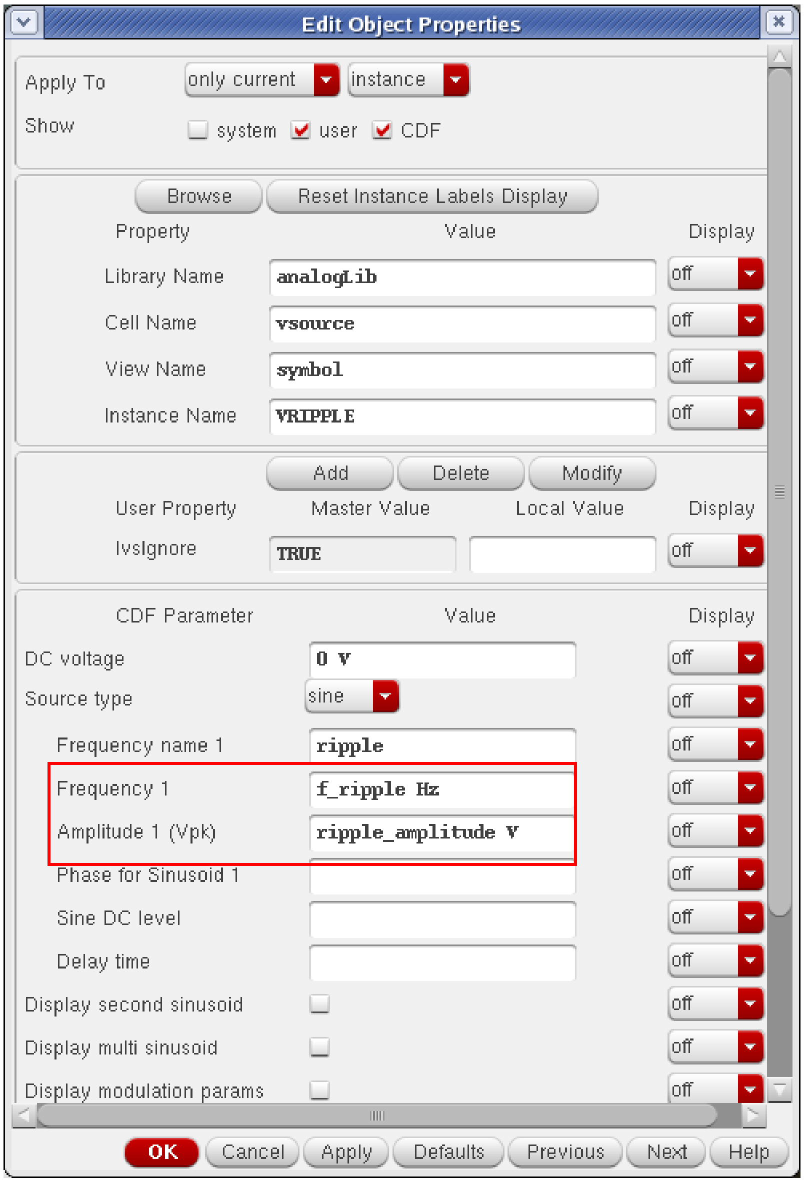
The focus of this section is controlling the spurious response caused by the power supply ripple. Consider an oscillator shown below.



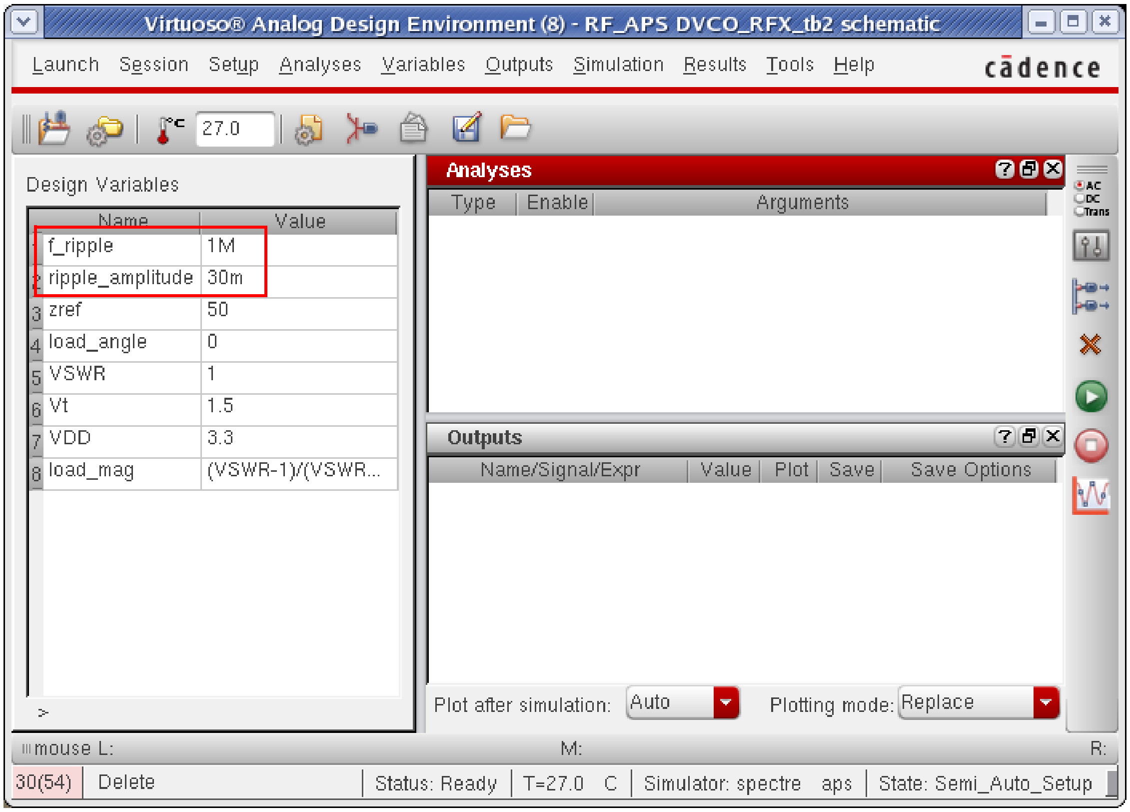
The first part of this section uses hb, which is equivalent to pss. The reason hb is used is because hb has the ability to run an oscillator with ripple so the spurious output of the oscillator can be measured. Pss does not have this capability. Using hb to solve for the spurious output usually takes longer than a pss with a modulated pxf analysis. Pss-pxf modulated will be used to measure the conversion gain from the power supply ripple to the output phase of the oscillator. Once the spurious output is measured with hb, then running pss and modulated pxf can be used to reduce the conversion gain from the power supply to the output, thus reducing the spurious output. Using this combination of analyses saves time.

1. On the left are two sources. The bottom source is the DC source that supplies power to the circuit. This source is in the yellow rectangle above. The upper source is a source that injects ripple into the simulation. This source is in the red rectangle above.

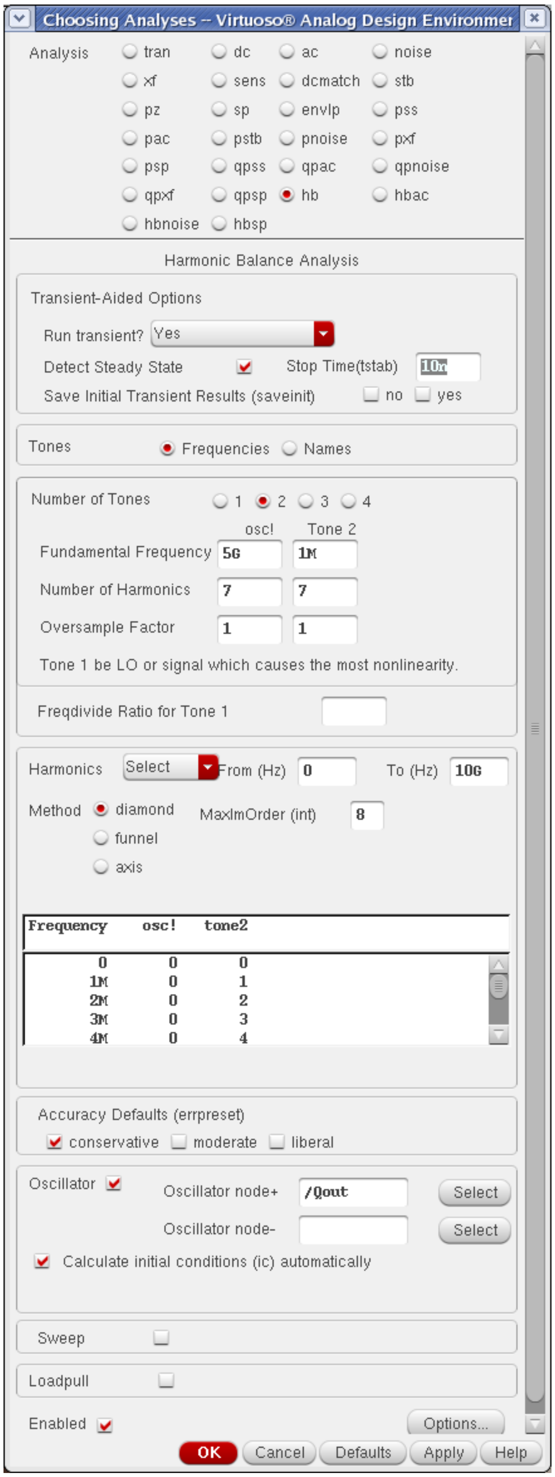
2. The ripple frequency and amplitude are set using variable names so the values can be changed in the ADE environment without needing to change the circuit. The ripple source properties are shown below.



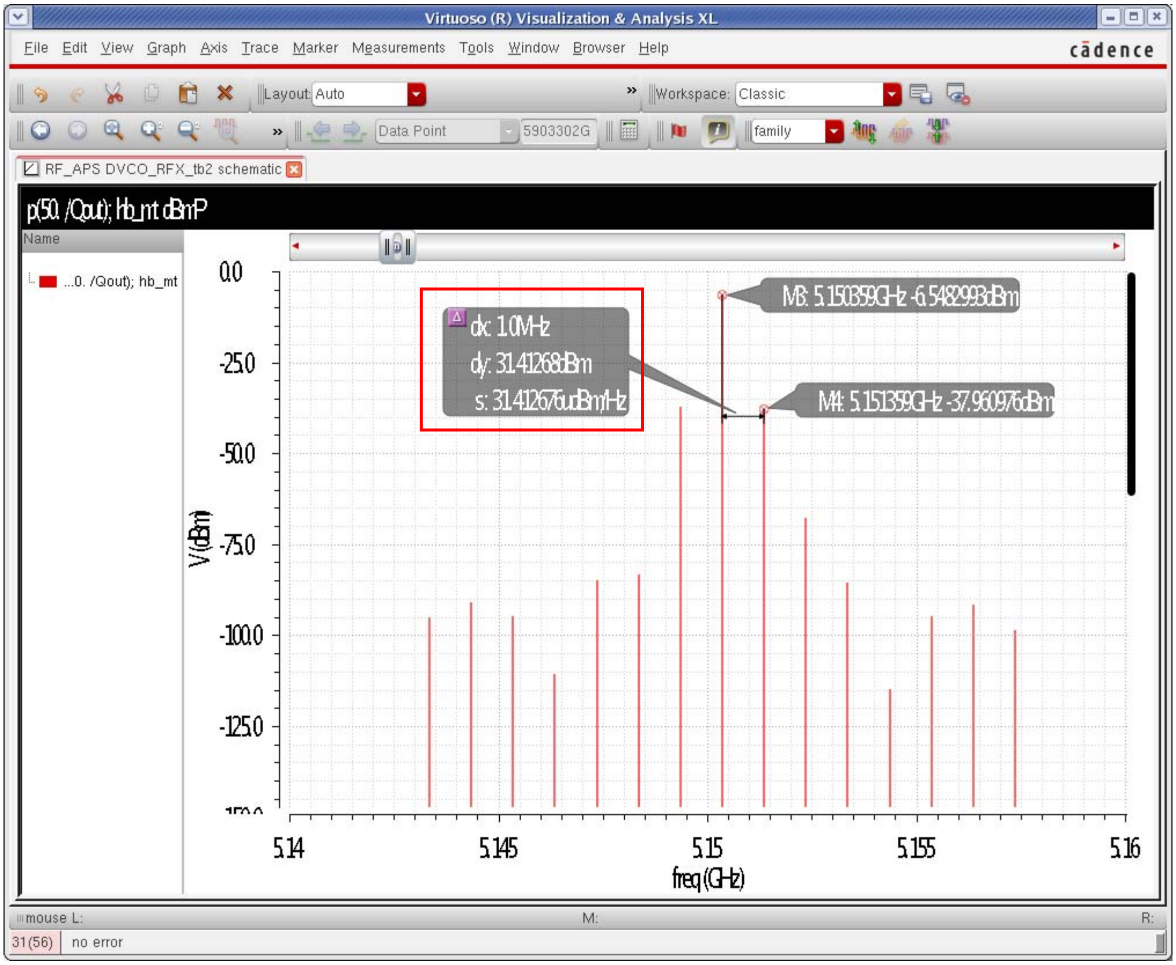
3. ADE sets the frequency to 1MHz and the amplitude to 30mV peak.



4. First, measure what is happening with the ripple applied in the circuit. See the hb section for more details. The hb setup is as below.



5. Run the simulation, and plot the output spectrum.



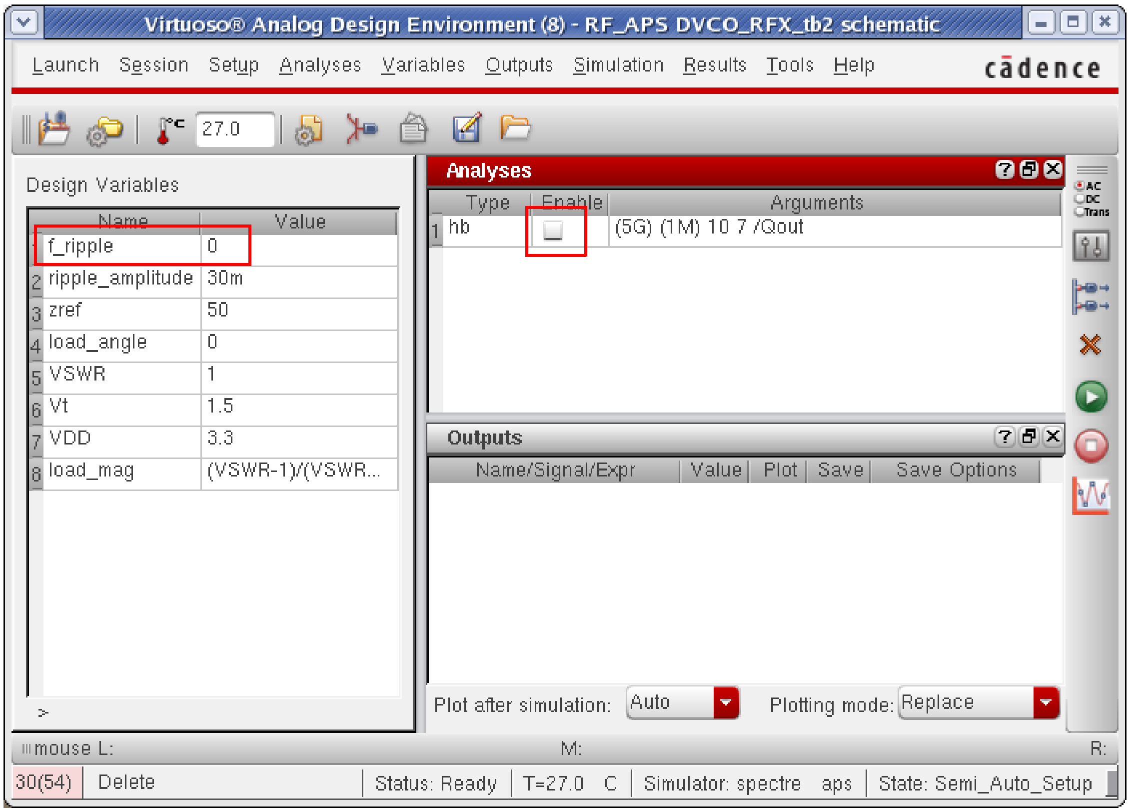
6. Place a marker by moving your mouse cursor over the tip of the oscillator frequency, and typing m. Now move your mouse cursor over one of the spurs, and type d. This places another marker, and causes the delta readout shown above to be displayed.

The spurious output is about 31.4dB down from the oscillator output frequency.

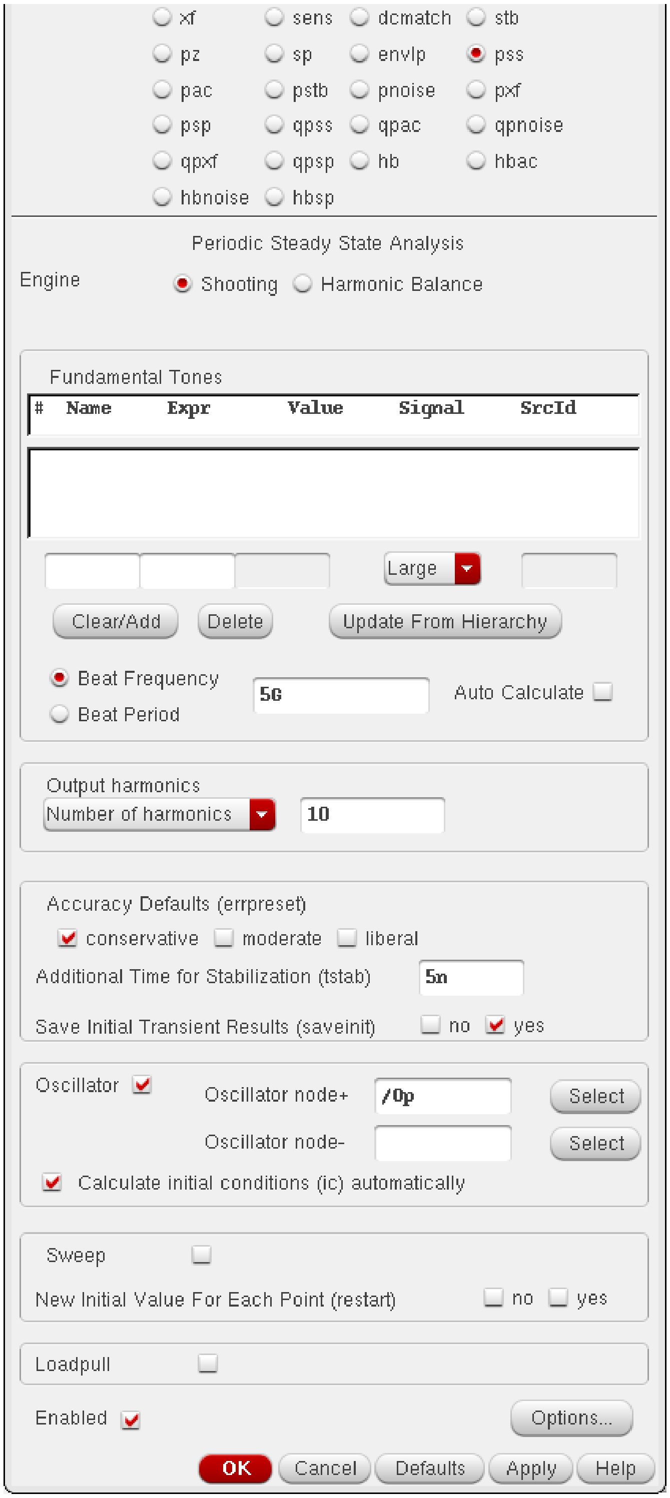
7. Now that we know what we really have, and how much this is above the specification, we can switch to a faster method to reduce the spurious output. This will be done using pss and modulated pxf.

8. Now measure the AM to PM conversion using pss and pxf modulated. This will tell you the gain from the power supply to the PM output.

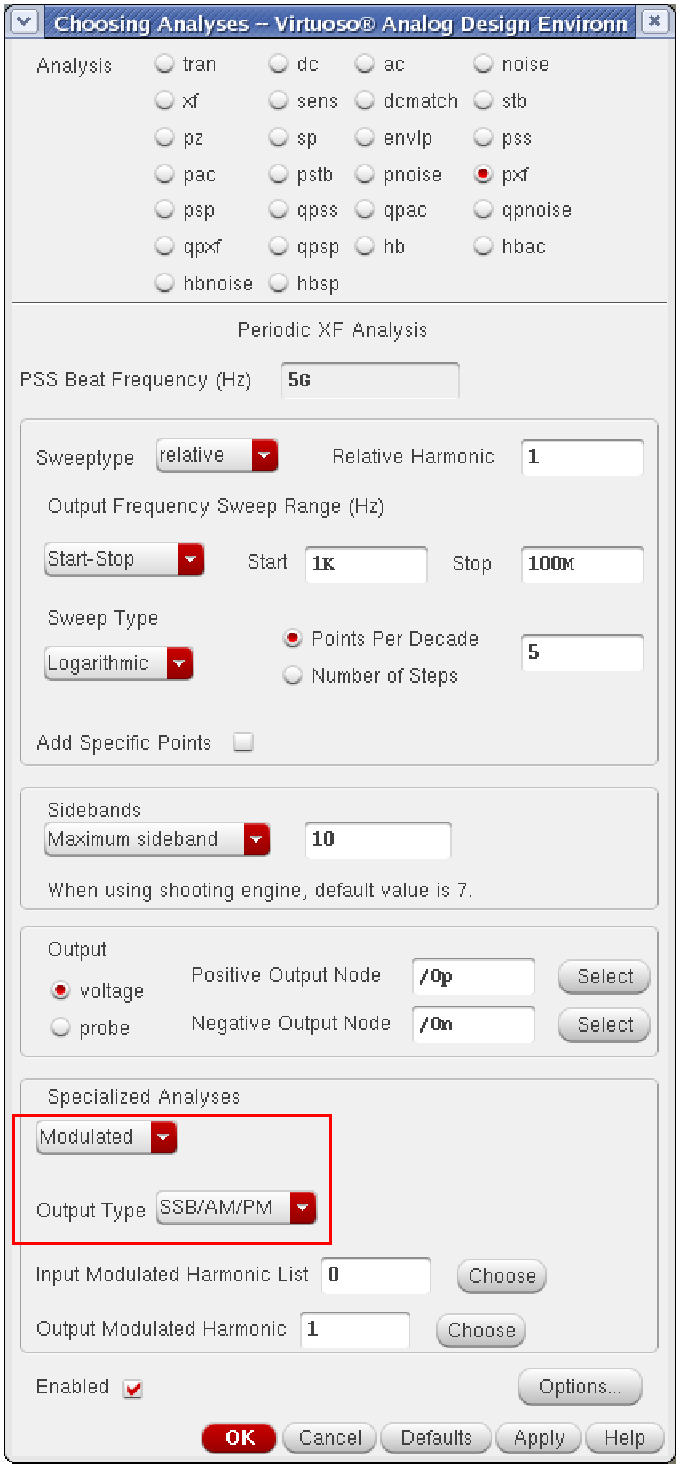
9. First, disable the ripple source by setting its frequency to 0 in ADE, and disable the hb analysis used to measure the spurious response.



10. Set up a pss analysis with just the oscillator signal. For more information, see the pss section at the beginning of this chapter.



11. Now set up a pxf modulated analysis, as follows:



12. At the bottom of the form, set *Specialized Analysis* to *Modulated*, and set the *Output Type* to *SSB/AM/PM*.

13. At the top of the form, *Modulated* forces a relative frequency sweep. The frequency sweep range is always an output frequency range in pxf. Since the gain from the power supply ripple to the output above the oscillator frequency is desired, the output harmonic is 1 and the frequency range is just above the first harmonic.

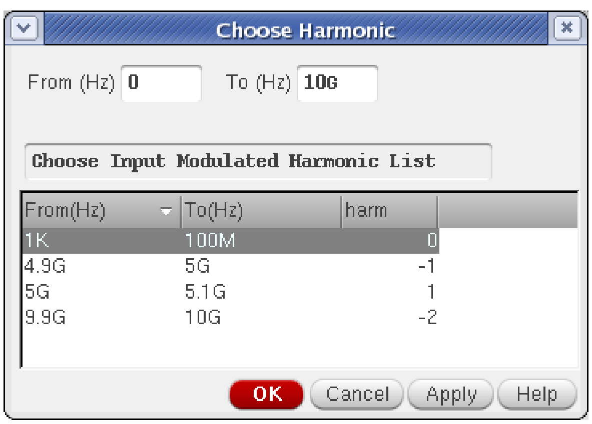
14. It is usually better to select either a linear or a log sweep, and define your own resolution because automatic usually takes more points and therefore a longer runtime than is usually needed.

15. Set *Maximum sideband* to the same value that you used for harmonics in the pss analysis.

16. Specify the output node or current as usual. If you have an output current, add an iprobe in series with the output, and click probe in the Output section. Then specify the instance name of the iprobe in the Output Probe Instance field.

17. Now, set the input frequency by clicking *Choose* to the right of the *Input Modulated Harmonic List* field.

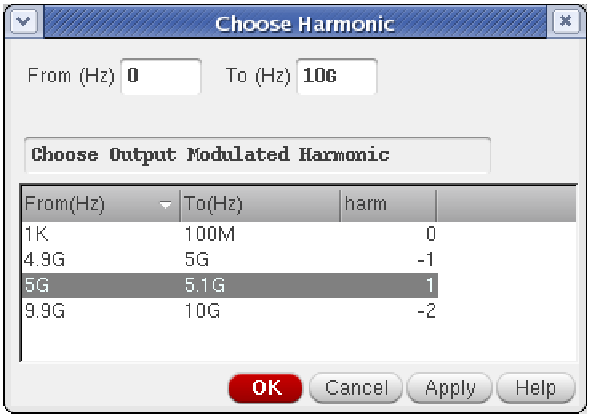
The Choose Harmonic window is displayed.



18. In the Choose Harmonic window, select the frequency range and click *OK*.

19. To set the output frequency, click *Choose* to the right of the *Output Modulated Harmonic* field.

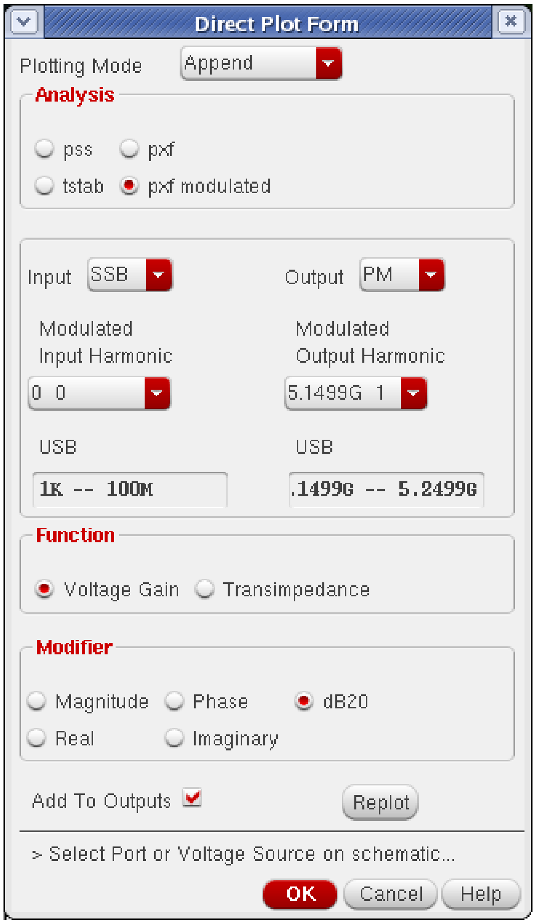
The *Choose Harmonic* window is displayed.



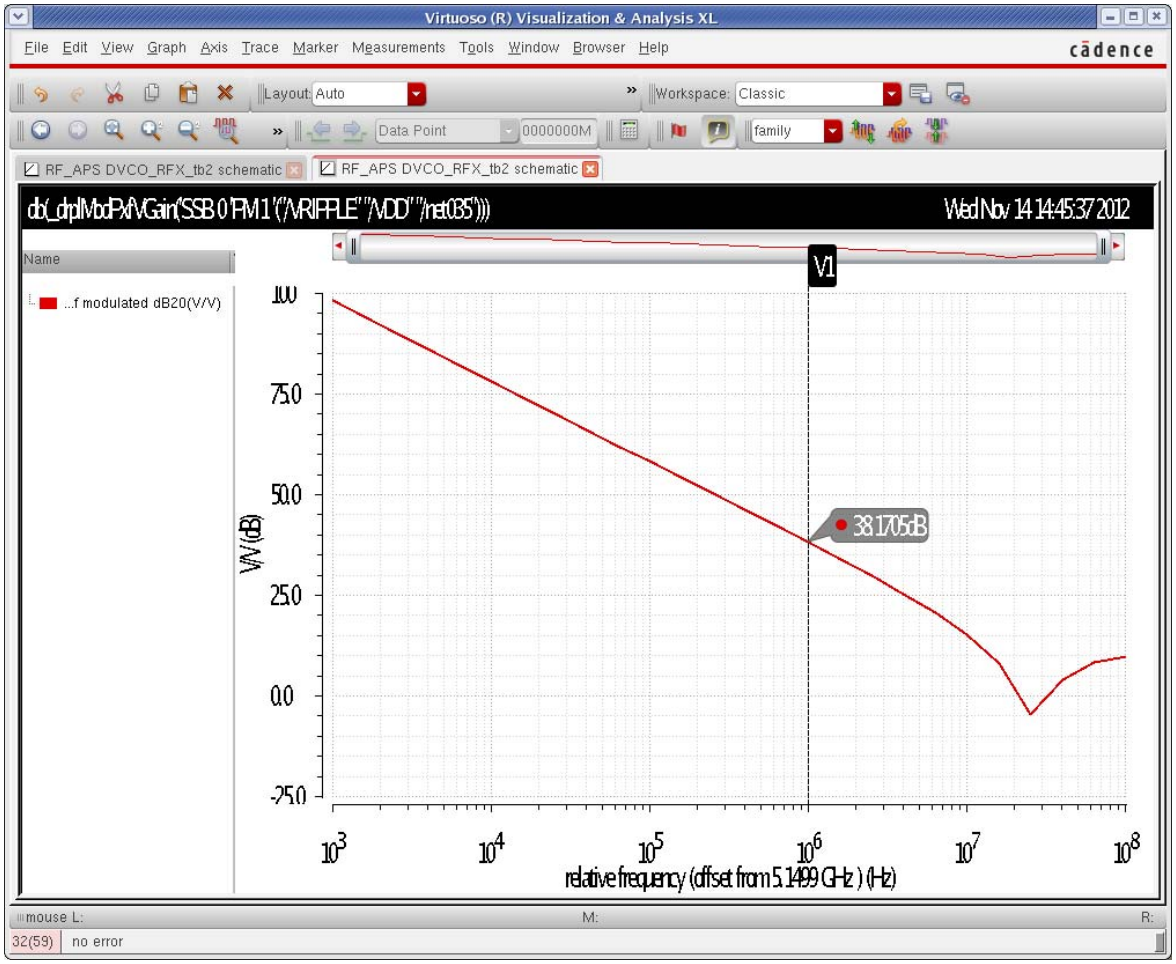
20. In the *Choose Harmonic* window, select the output frequency range and click *OK*.

21. Click *OK* in the *Choosing Analyses* form and run the simulation.

22. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window.



23. Currently, to measure the AM to PM conversion, you set *Input* to *SSB*, and *Output* to *PM*. Select *dB20,* if desired. Select the ripple source in the schematic. The AM to PM gain is plotted.

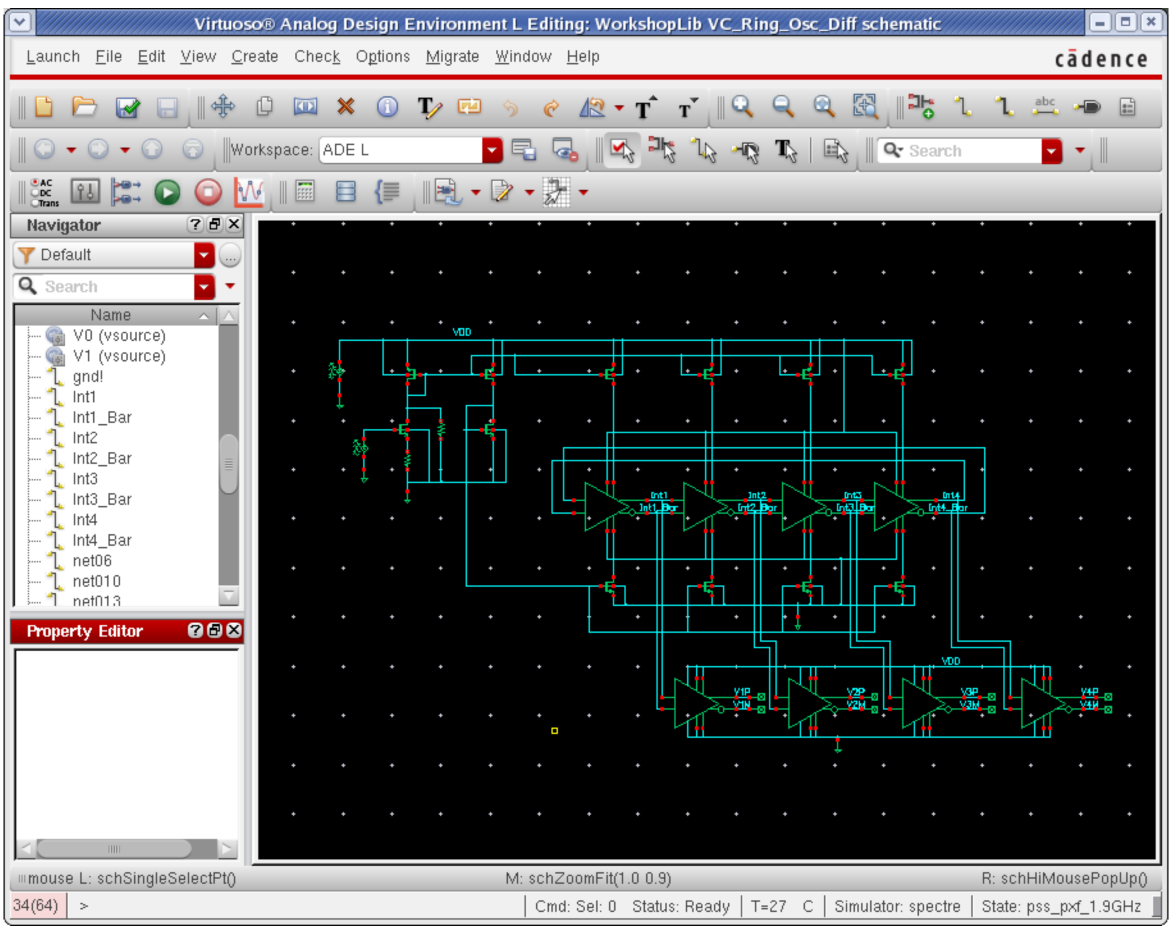


24. Position a marker at the ripple frequency to measure the gain. The gain in this example is about 38dB. As you implement circuitry to reduce the ripple, you can run the pss-pxf modulated analysis in less time than it takes to run the hb analysis with both the oscillator and the ripple. When you have a ripple reduction circuit in place, you can re-run the two-tone hb analysis to verify the fix works.

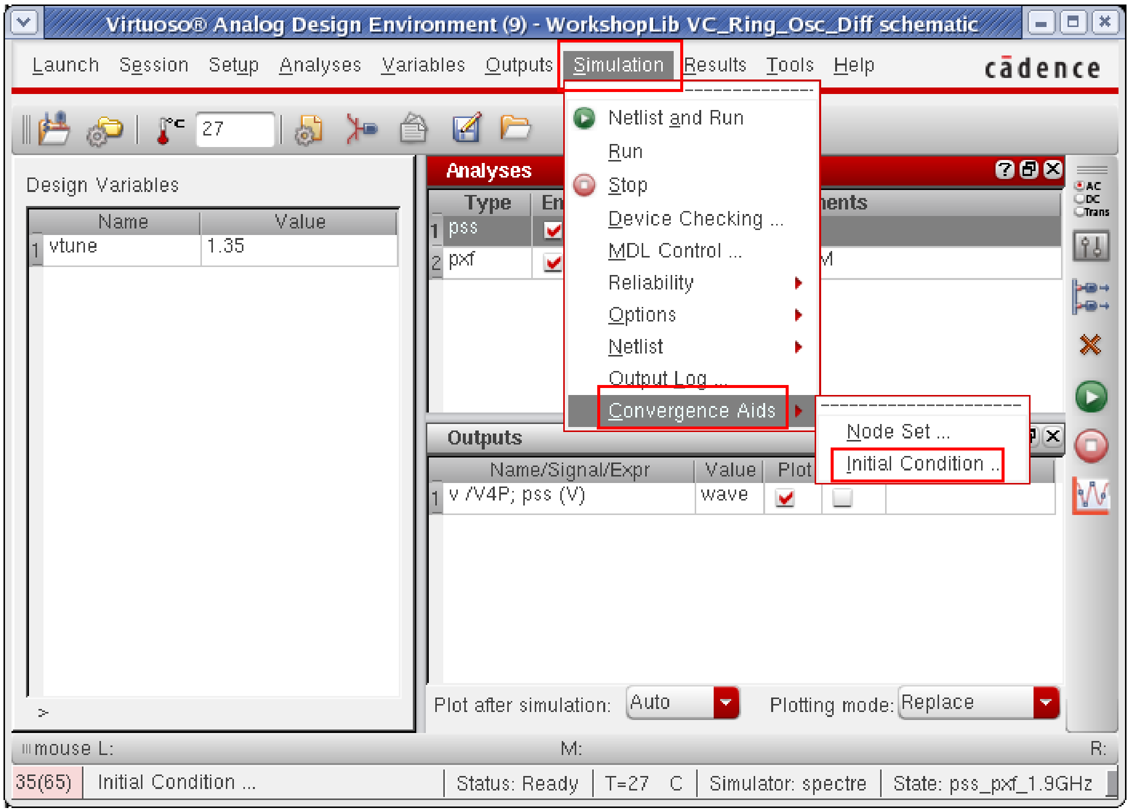
Sampled PXF: Measuring Conversion Gain at a Threshold Crossing in a Ring Oscillator

Sampled pxf can be used to calculate the gain from the power supply to the output when the output is defined as the periodic point where the gain is measured at the same phase (time) of the oscillator as if it were sampled repeatedly at exactly the same phase (time) over an infinite number of cycles. This is the conversion gain from the power supply to a single point in the output waveform, as opposed to the averaged measurements from before. Usually, this is applied to ring oscillators where relatively sharp edges are present in the output, and the output is driving some kind of digital logic. In this case, only the point where the threshold for the digital logic on the output matters. All the rest of the points in the waveform are either above or below the threshold, and thus do not influence the threshold event time.

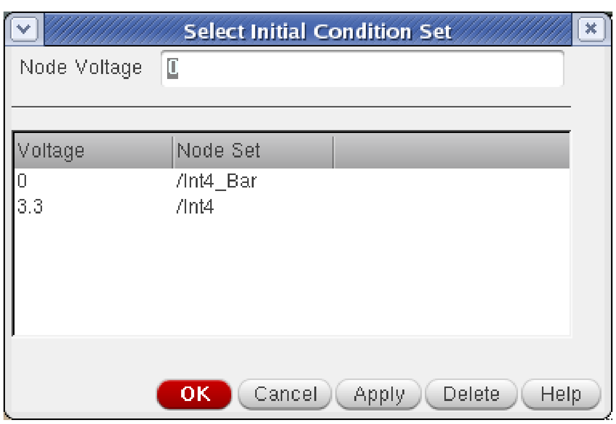
1. Consider the 4-stage differential voltage-controlled ring oscillator below.



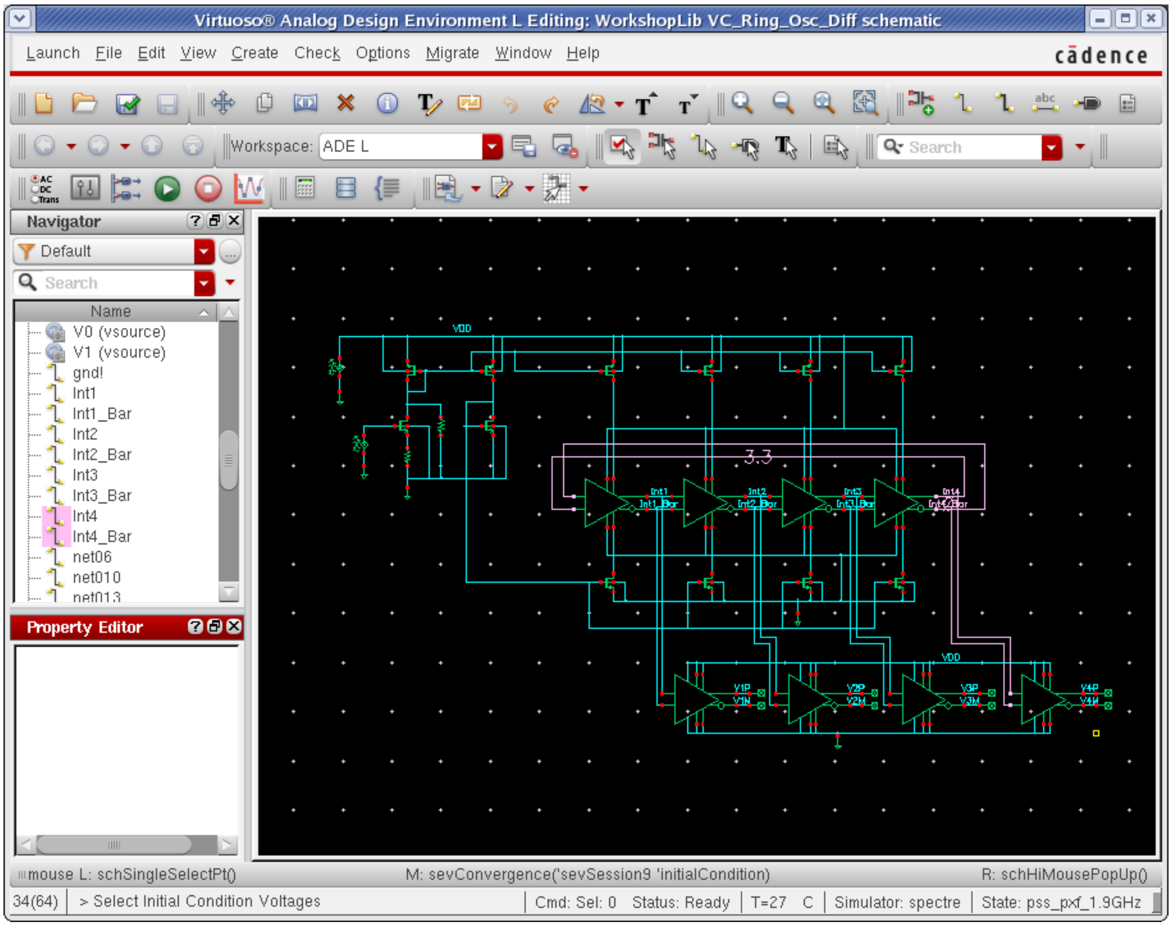
2. First, set initial conditions on one of the stages of the ring to force a DC operating point. In the ADE window, select *Simulation - Convergence Aids - Initial condition*.



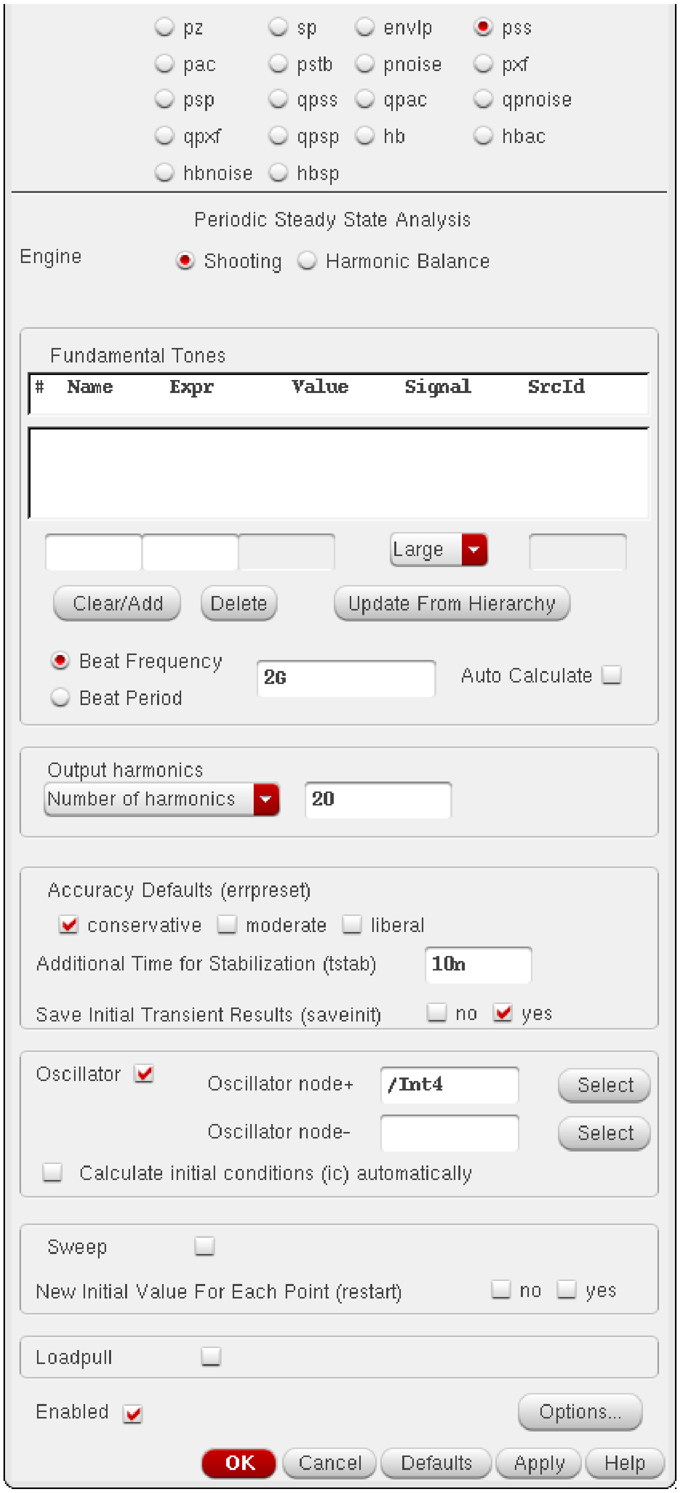
3. The *Select Initial Condition Set* window appears. Since this is a differential oscillator, there are two outputs from one stage. One of those outputs is set low, and the other is set high. This forces a starting condition for the tstab interval.



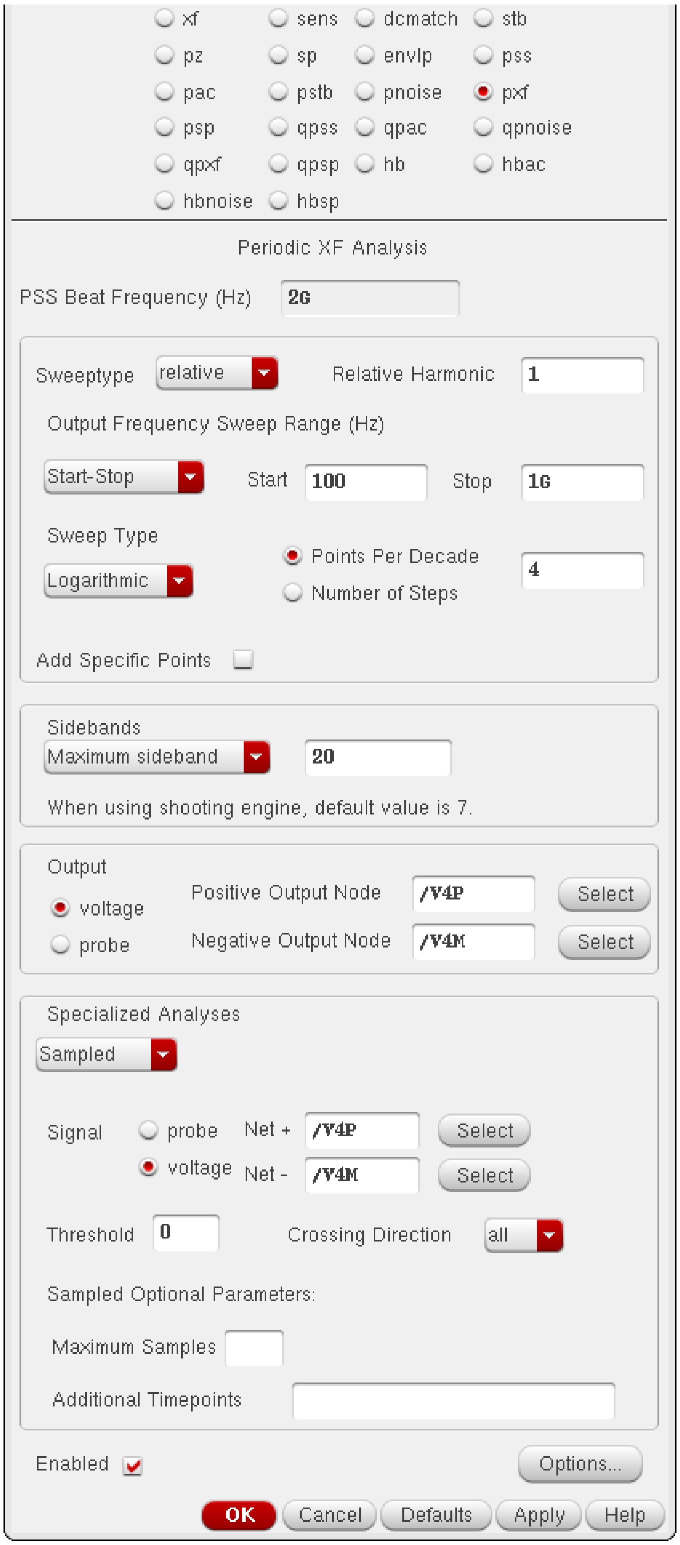
4. When the *Select Initial Condition Set* window is displayed, the nodes with the IC defined are highlighted in the schematic, as shown below. The fourth stage output is set.



5. Now set up the pss analysis. For more information, please see the pss section at the beginning of this chapter.

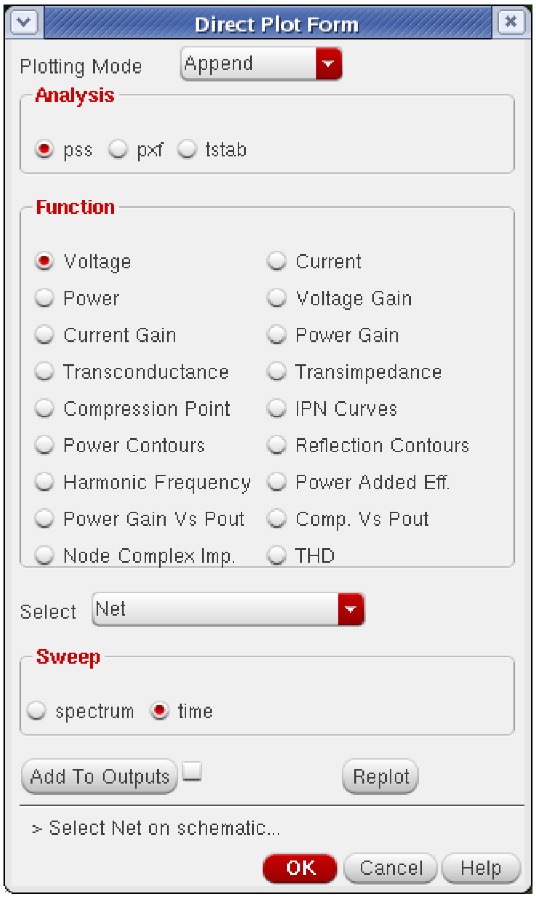


6. Set up the sampled pxf analysis, as shown below.

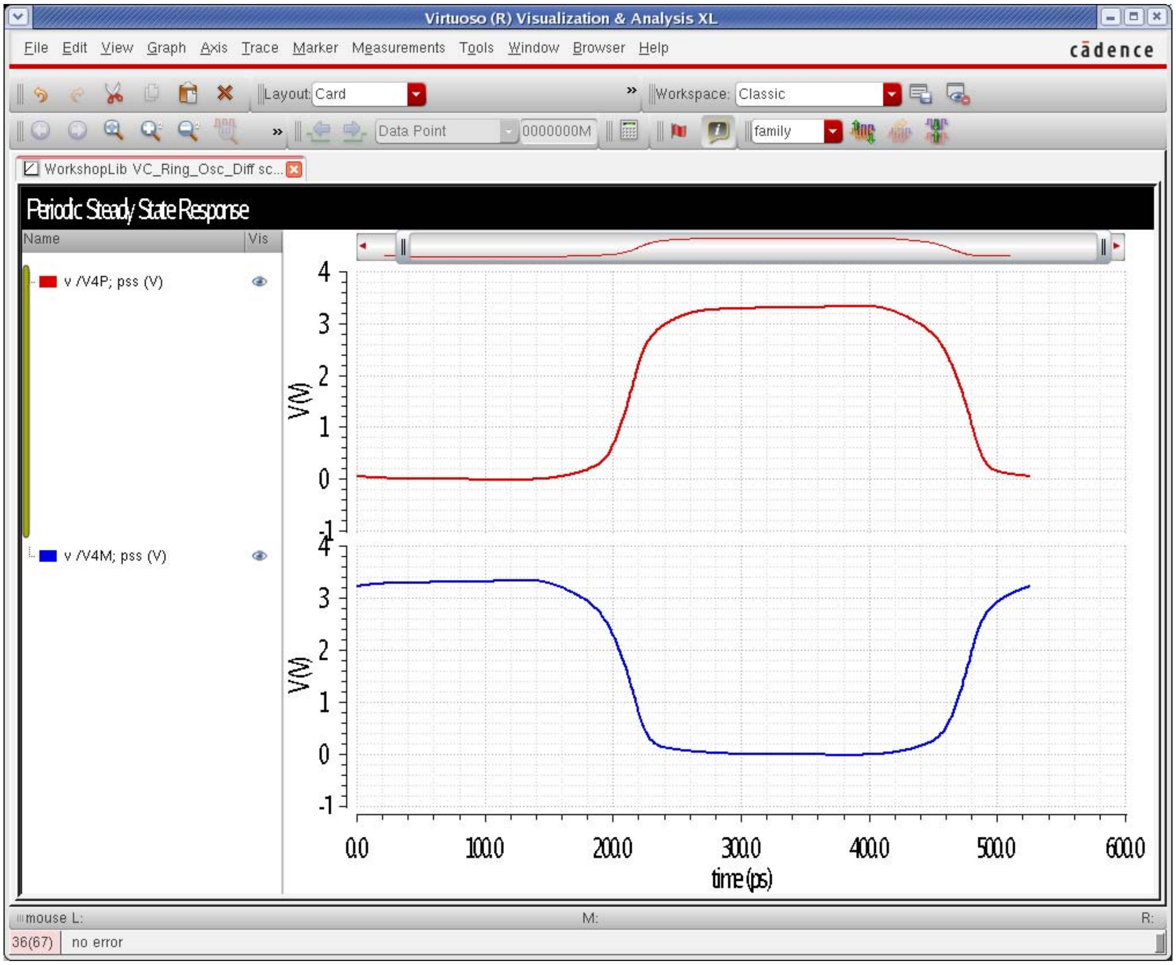


7. Select *Sampled* from the *Specialized Analyses* drop-down list.
8. Because the pss is set for an oscillator, relative is set for pxf. The **output** frequency is just above the first harmonic of the oscillator in this example.
9. Usually, it is better to select Linear or log, and define your own sweep. Auto usually takes more frequency point than are needed, thus extending the pxf runtime.
10. Set *Maximum sideband* to the same value used for pss harmonics.
11. Select the output net or current, as usual. If you have an output current, add an iprobe in series with the output, and click probe in the *Output* section. Then specify the instance name of the iprobe in the *Output Probe Instance* field.
12. Below the *Sampled* selection, define the output type, and set the output again.
13. Specify an appropriate threshold. This ring oscillator has a differential output, so 0 volts is an appropriate threshold.
14. If desired, set the crossing direction. This example calculates all the crossings of 1.65 volts.

15. Click *OK* in the *Choosing Analyses* form and run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form*. The *Direct Plot Form* appears.



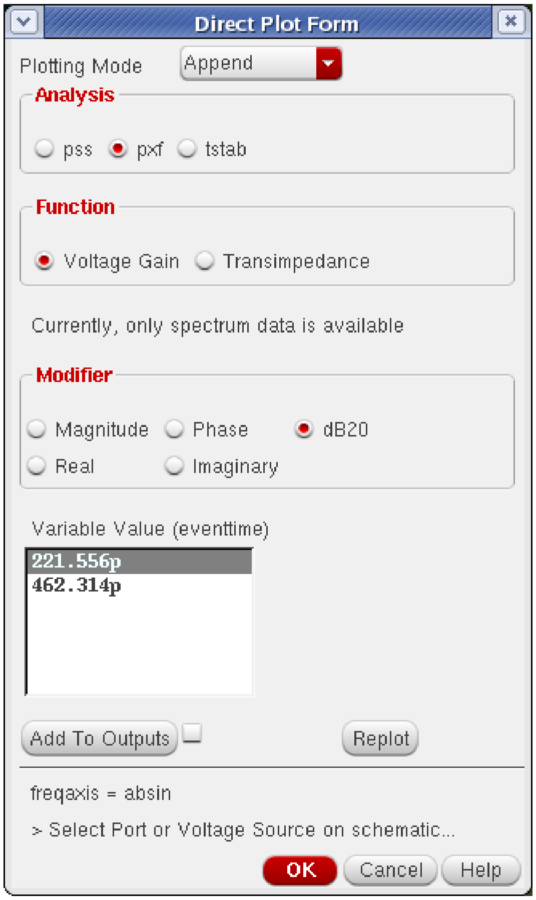
16. Plot the pss time-domain output waveform(s) so you can see the times of the transitions .



17. For the V4P signal, the rising edge is about 213psec, and the falling edge is about 472psec.

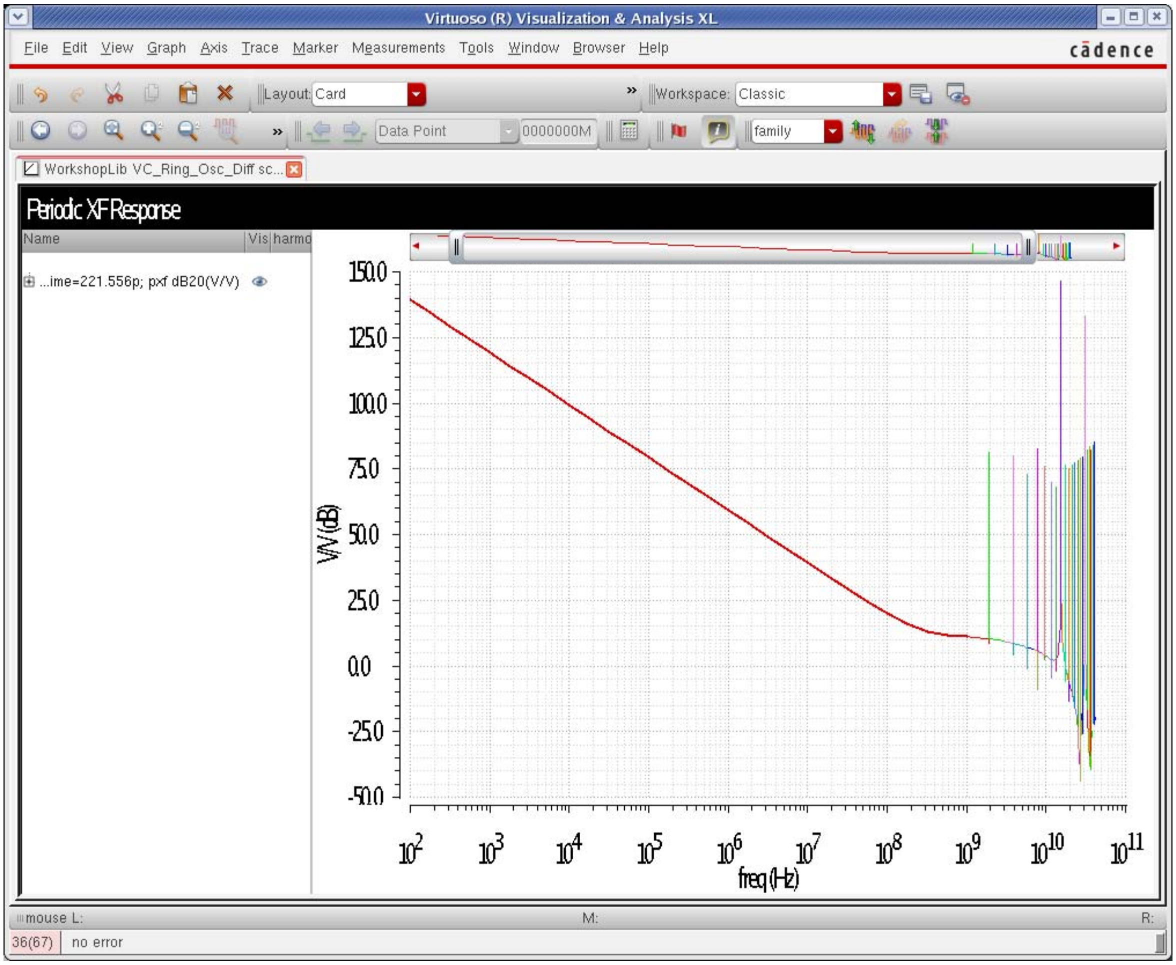
18. Now plot the pxf sampled results.

19. If desired, select *dB20*.

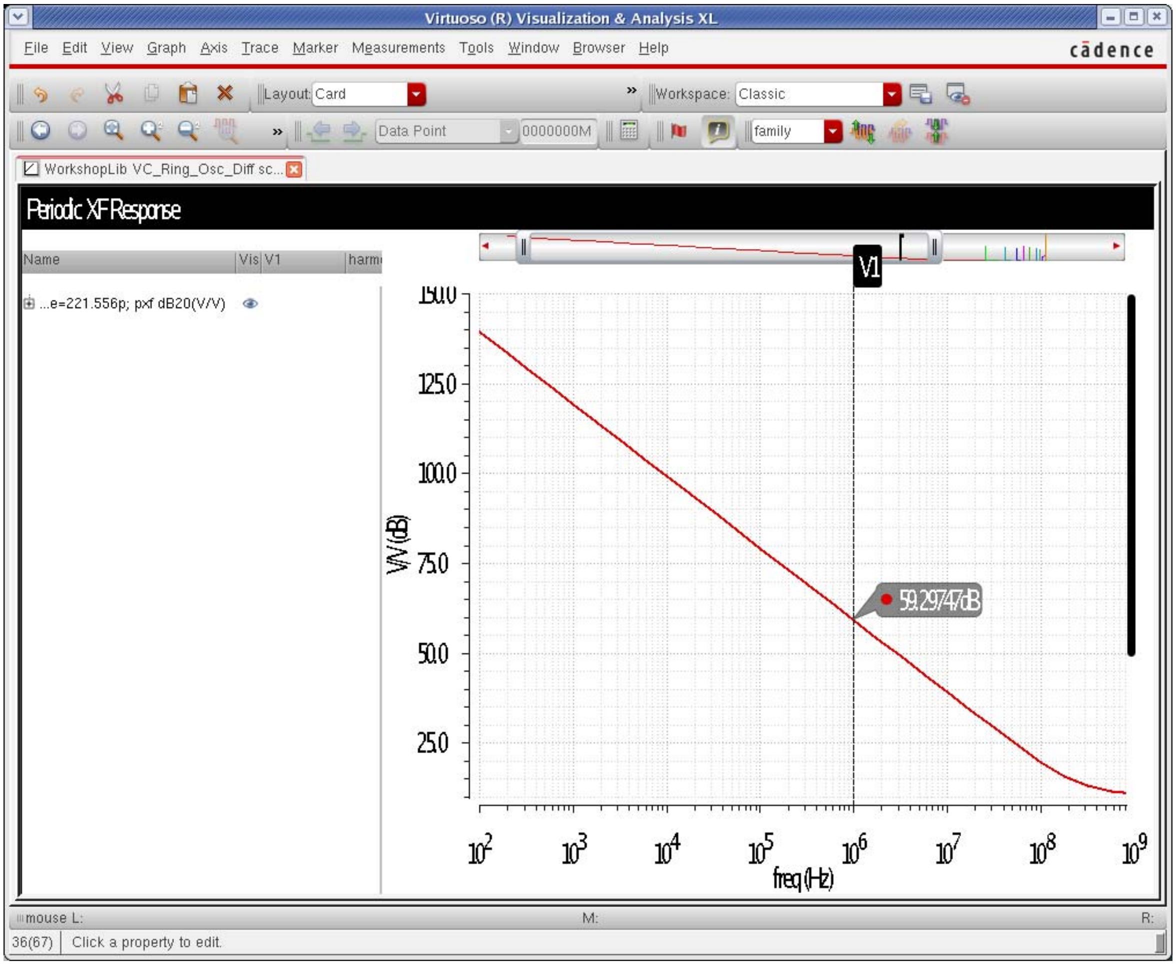


20. Select the event time.

21. Now select the power supply source in the schematic. The results plot, as shown below.



22. To read the result, go to the input frequency on the pxf plot, and read the gain. This is the voltage gain from the power supply to the output just above the first harmonic of the oscillator at a single timepoint in the waveform. If there is voltage gain from the power supply to the output, it shows up as ripple on the output, which will affect the timing of the output crossing. The gain at 1MHz input frequency is shown below.



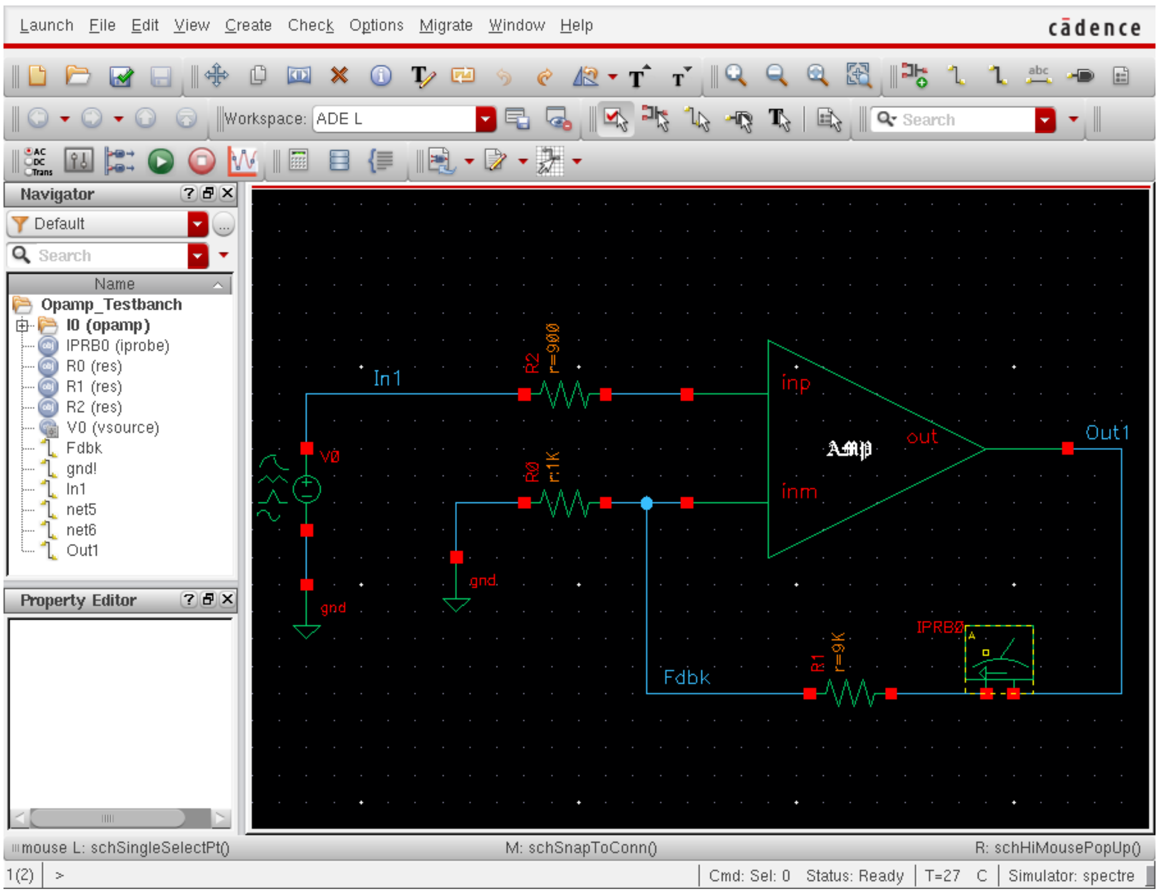
Periodic Stability Analysis (PSTB)

Pstb is provided to measure the small-signal loop gain in the presence of a large signal. As an example, think of an amplifier. When the input signal is small, the amplifier is largely linear, and stb and pstb should agree. However, when the input signal becomes large, the amplifier becomes more nonlinear. This will change the loop gain of the feedback system. Pstb measures the loop gain with the large signal present in the system.

An additional application is for measuring the loop gain of oscillators. In most oscillators, the power supplies define the output level. The nonlinearities created by the oscillation cause the loop gain to shift slightly from the linear stb analysis results. We have seen cases where multiple stable oscillation frequencies resulted from the nonlinearities introduced by the oscillations themselves, and this showed up in the pstb result. The large signal oscillation is solved in the pss analysis, and the small-signal stability analysis is run in the pstb analysis.

Example: Variation in Loop Gain of an Opamp with Swept Input Signal Amplitude

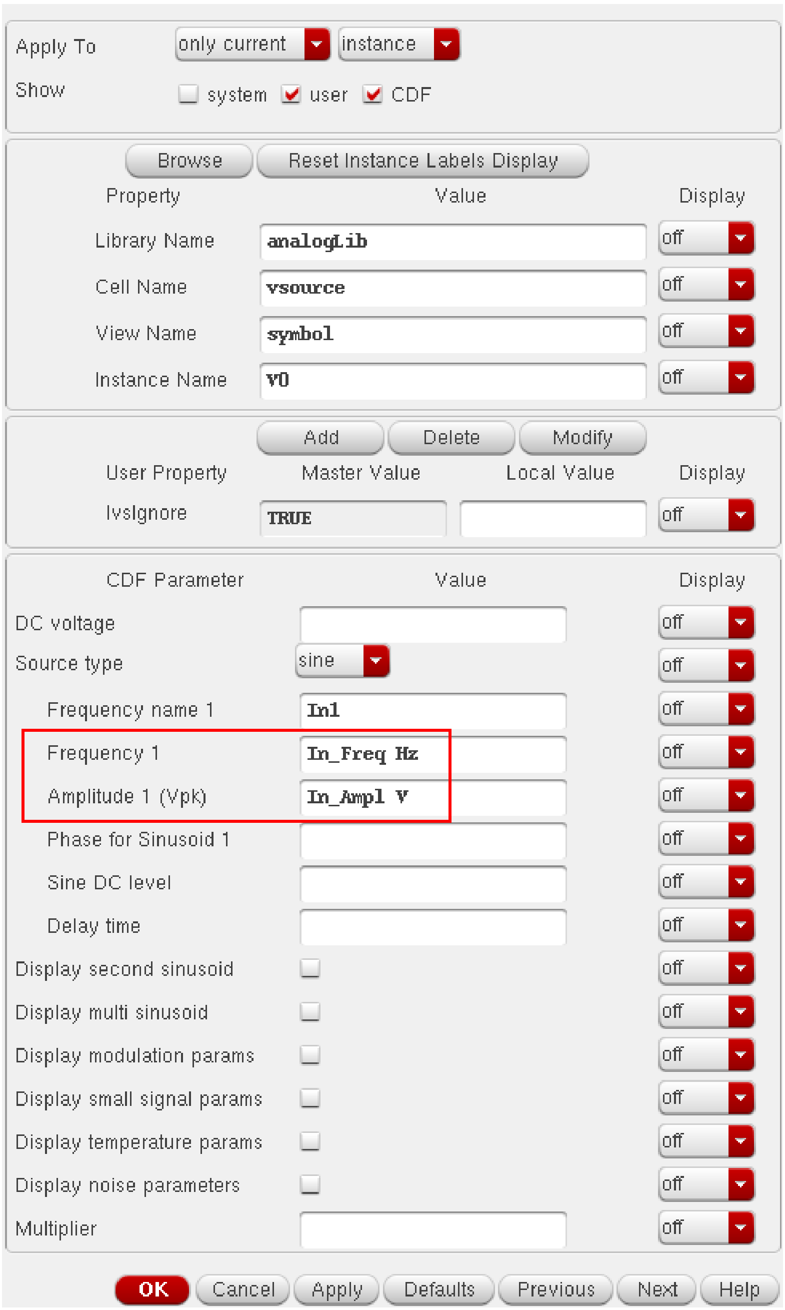
Consider the opamp circuit below.



To use pstb, an `iprobe` from `analogLib` (single ended) or a `diffstbprobe` from `analogLib` (differential) needs to be added in series with the feedback path. Note the arrow in the current probe is in the same direction as the direction of the feedback.

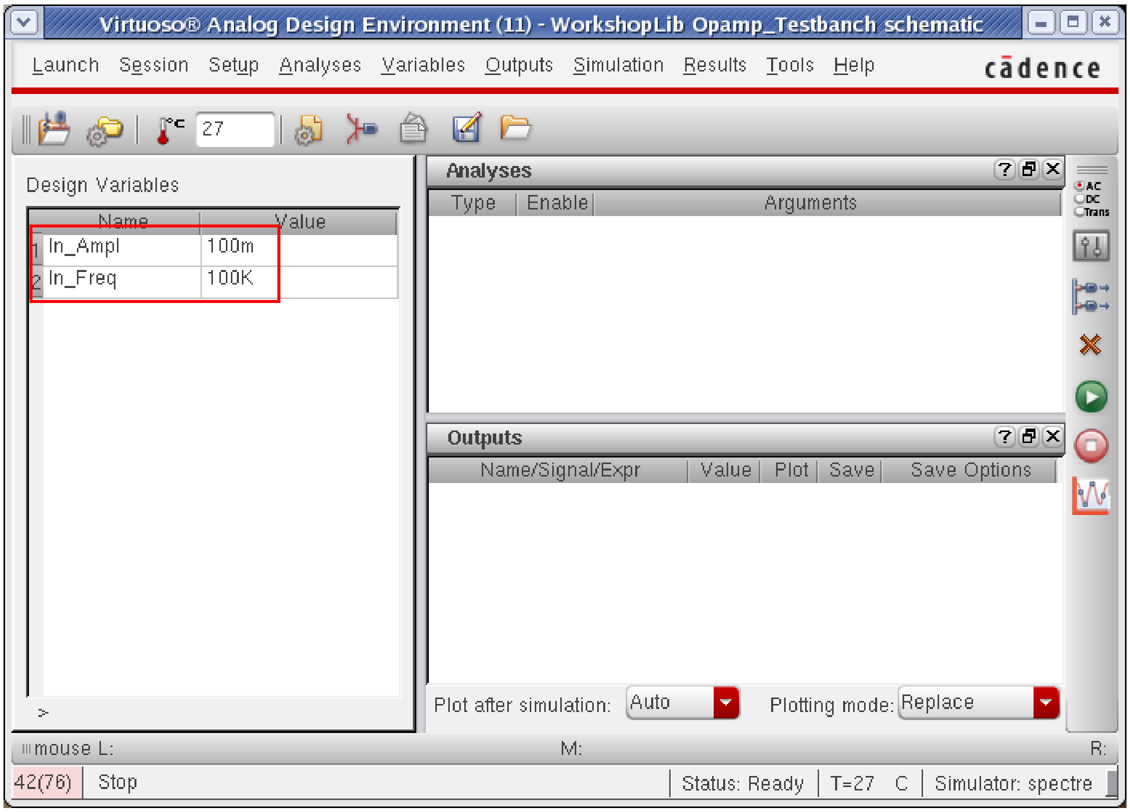
The input source to the circuit has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. This allows changes in frequency or amplitude from the ADE environment without needing to change the schematic itself.

The property list for the input source is shown below.



The easiest way to get the variables list into the ADE is to select *Variables -> Copy From Cellview*. To set the values, select the value field to the right of the variable, type in the desired value, and press *Enter.* When you have set the variables, if you want to add the

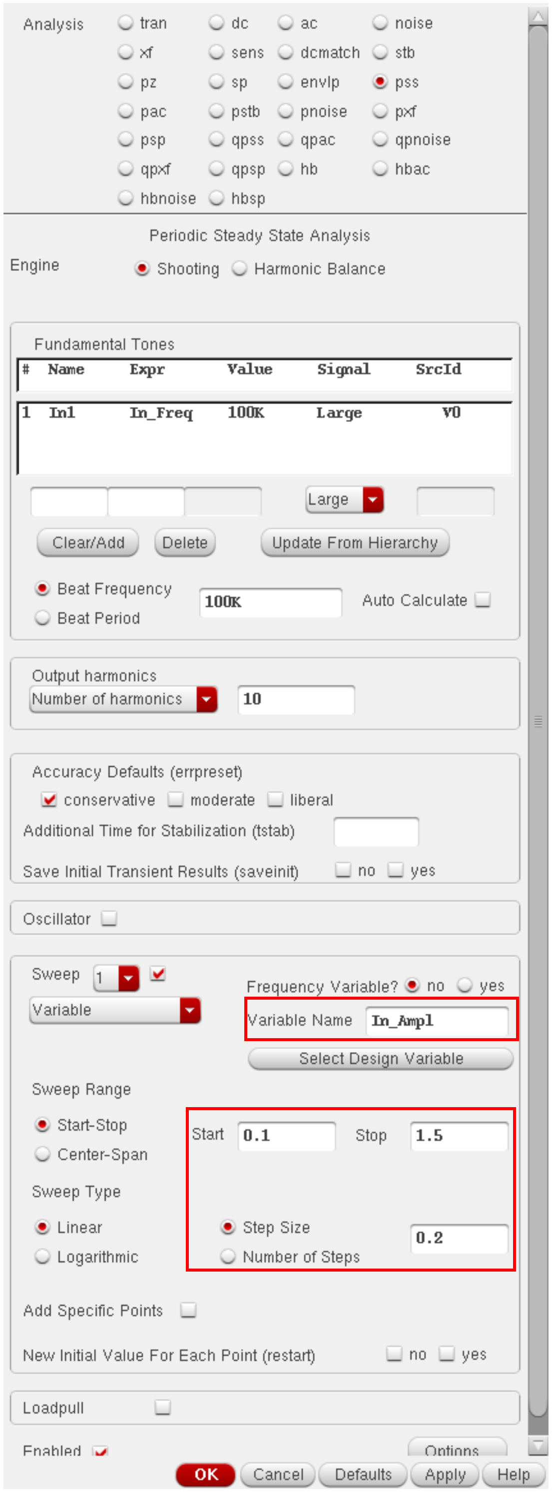
values back into the cellview, in the ADE window, select *Variables - Copy To Cellview*, and then perform a *Check and Save* in the schematic.



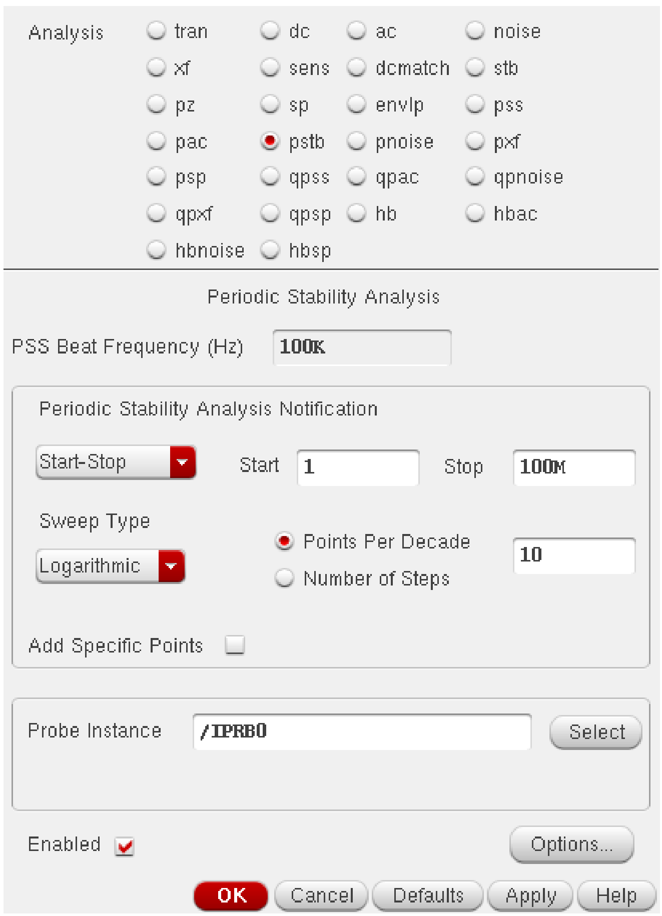
The input frequency for this example is 100KHz and its amplitude is 100 millivolts peak.

The basic strategy is to apply the signal that causes the distortion in the pss analysis, and then measure the loop gain using the pstb analysis.

The pss analysis needs to be set up first. The example below shows a single input at 100KHz with 10 harmonics. The amplitude is swept from 0.1 volts to 1.5 volts peak with a step value of 0.2 volts. For more information, see the pss section at the beginning of this chapter. This will characterize the amplifier at a range of different input amplitudes. At each amplitude, both the pss and pstb analyses are run.

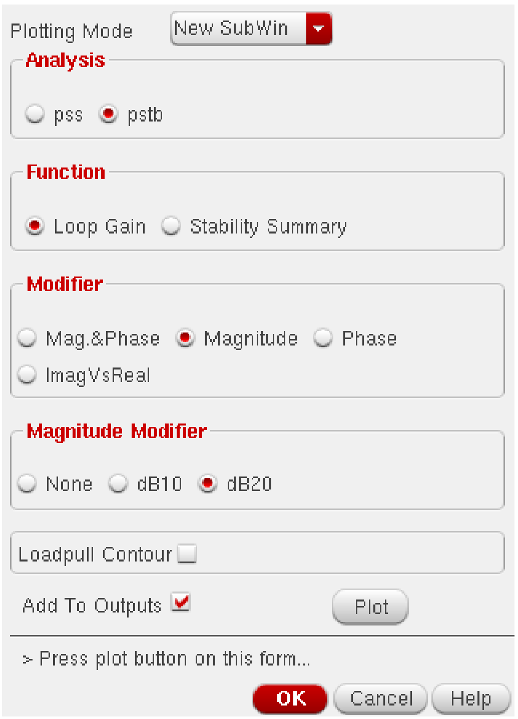


Next, the pstb analysis needs to be set up. The frequency range depends on your application. The *Probe Instance* is the iprobe from `analogLib` in the circuit. If you have a differential circuit, use `diffstbprobe` from `analogLib`. `diffstbprobe` allows the measurement of common-mode and differential-mode loop gain.



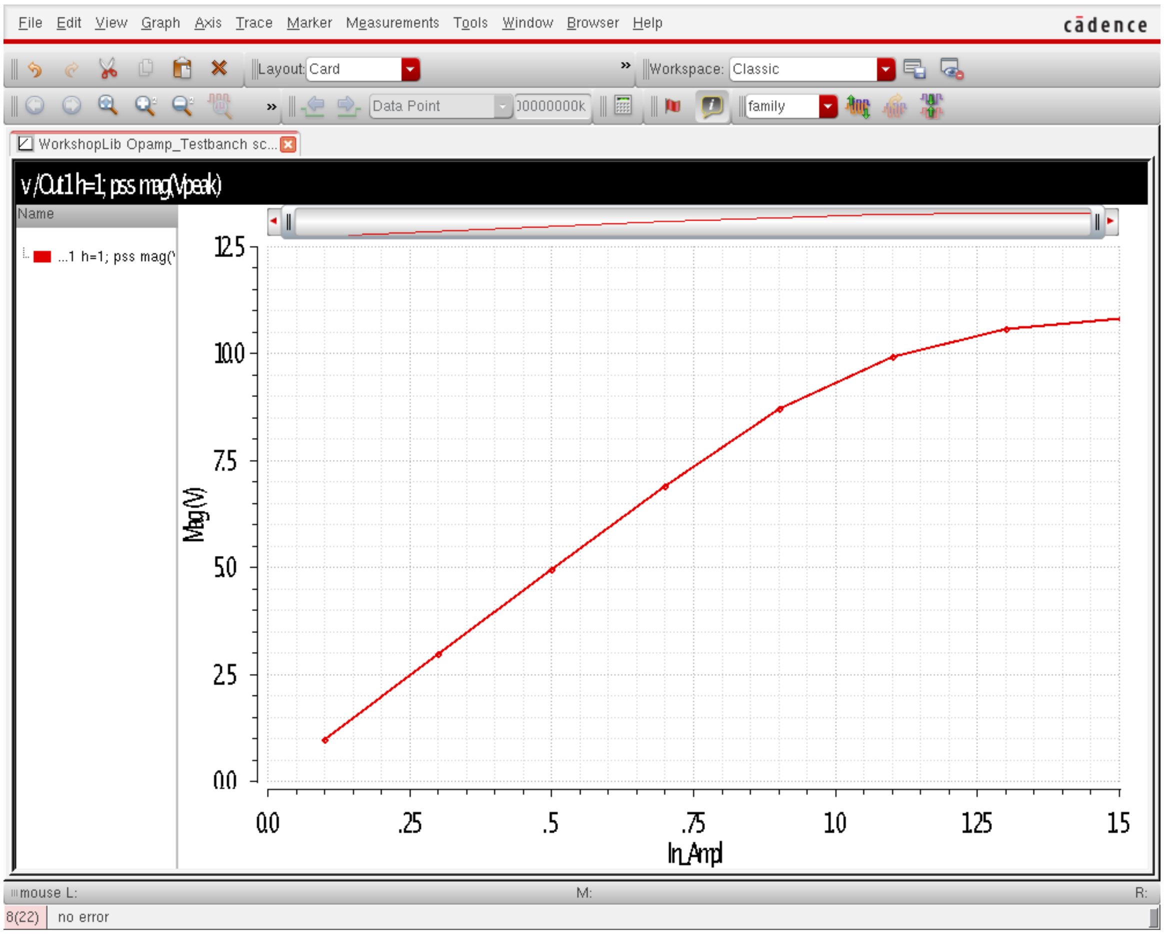
Once the analyses are set up, the simulation can be run. When the simulation finishes, select *Results - Direct Plot - Main Form* in the ADE window.

The example below shows the *Direct Plot Form*. If you want dB for the loop gain amplitude plot, plot the magnitude and phase separately This example shows dB for the loop gain magnitude plot.

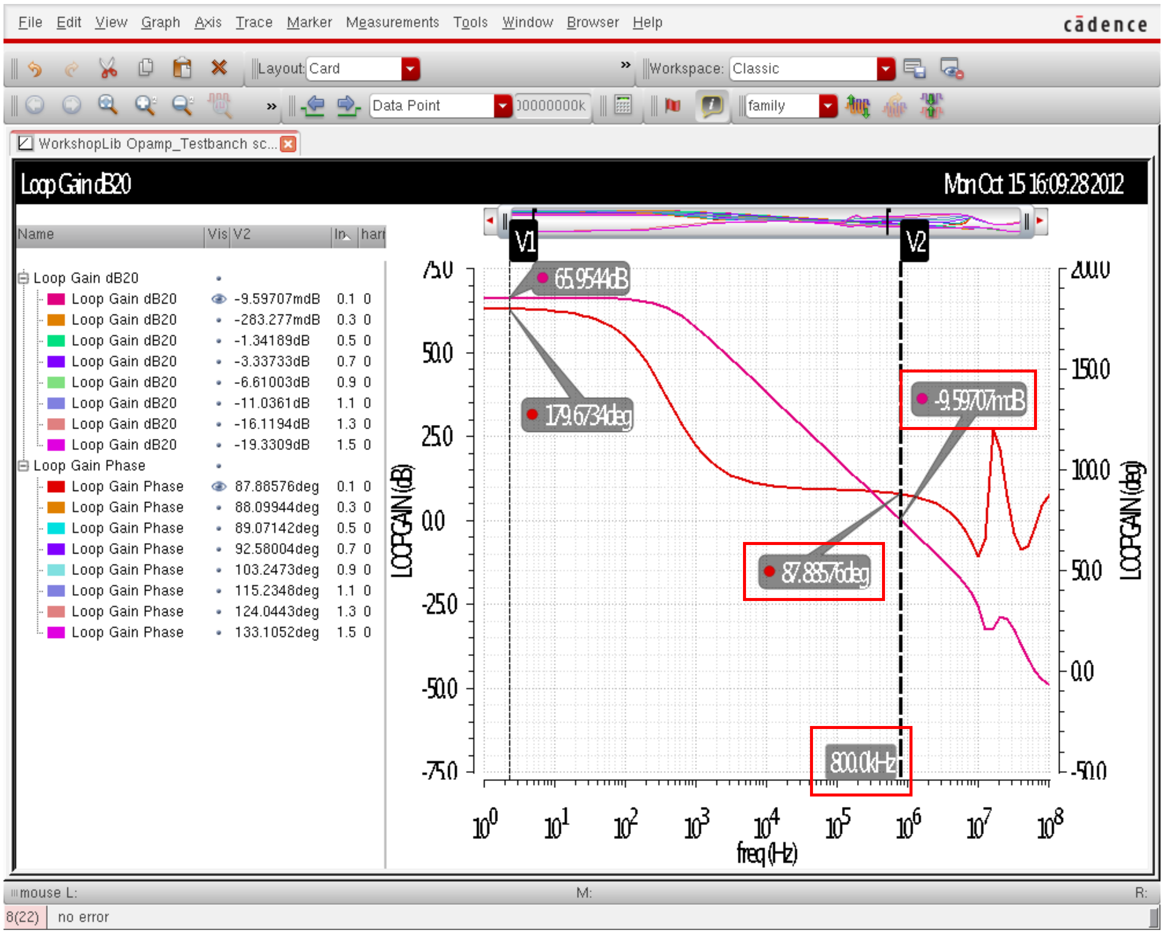


If you select *Add To Outputs*, an expression is entered in the ADE outputs section so that the next time you simulate your circuit, this measurement plots automatically.

Here is the output voltage of the opamp versus input voltage for the sweep. The curve begins to compress at 1.1 volts peak.



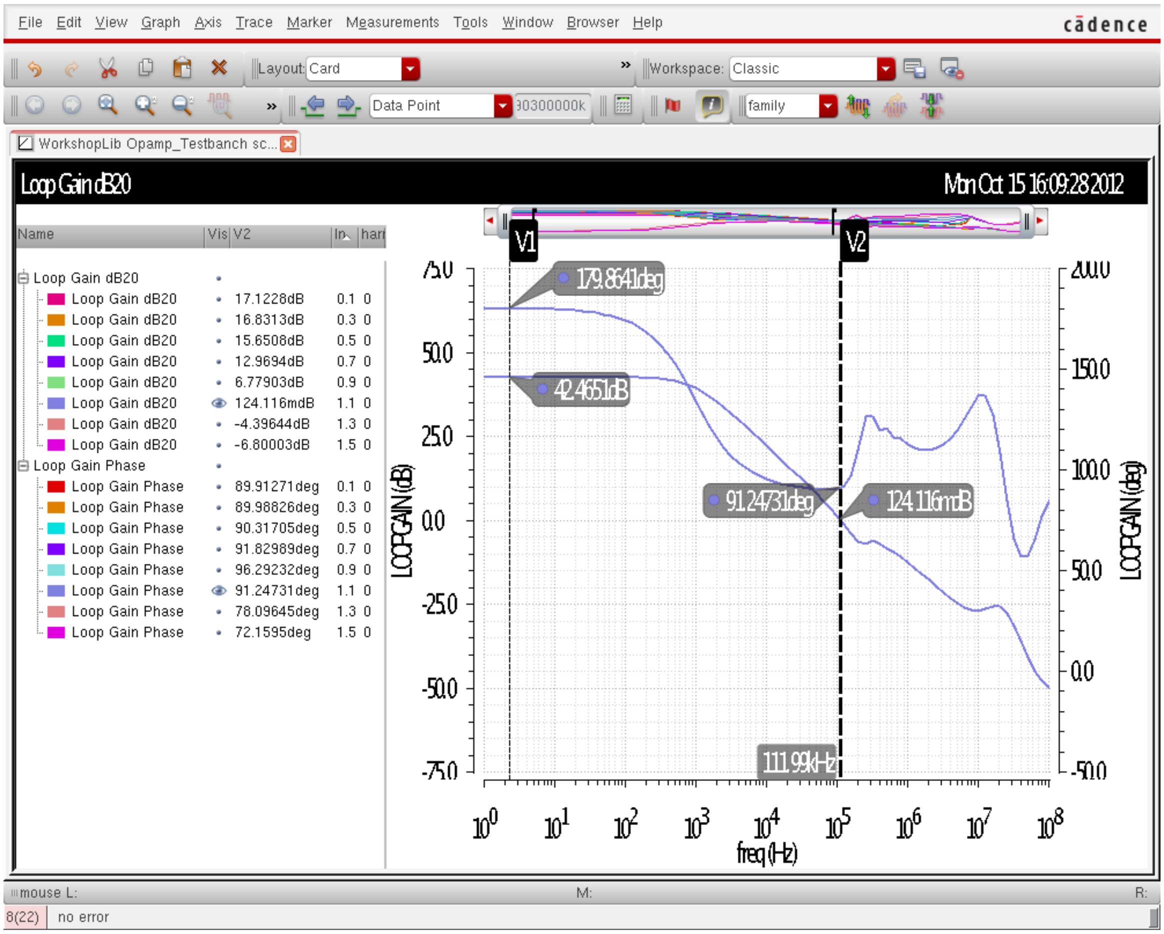
Here is the loop gain and phase curve for the 0.1 volt input.



Note that the DC phase is 180, and not zero. From feedback theory, we would expect zero. So what causes this difference? In feedback theory, we always assume there is a forward and reverse transfer function, and an inverting summing junction with a gain of unity. In feedback theory, we plot the forward and reverse transfer functions only. When the phase is zero, and then applied through the inverting input, the phase becomes 180. In the circuit, we cannot separate out the summing junction. Thus, the condition of instability is a linear magnitude of 1 and a phase of zero, or multiples of 360 degrees.

Vertical marker V2 has been placed at (nearly) unity gain (0dB). The phase is about 87.8 degrees. The distance to zero phase is 87.8 degrees. This is the phase margin for this circuit at low input amplitude. The unity gain frequency is about 800KHz

Below is the plot with a 1.1 volt input.



The DC gain has dropped from about 66dB to about 42 dB. The unity gain frequency has dropped from about 800KHz to about 112KHz. The phase margin has actually improved.

Overview of Simulation Capabilities

Pstb is used to measure the loop gain with a large signal applied to the circuit. First, the large signal is applied in pss. An `iprobe` (single-ended) or a `diffstbprobe` (differential) is placed in the circuit in series with the feedback path. Both probes are in `analogLib`. Pstb applies a voltage to the source(s) in the probe, and measures the amplitude and phase of the signal across the probe. This is the loop gain of the circuit. `diffstbprobe` allows the measurement of common-mode and differential-mode gain in the differential circuit.

Pstb is supported for both shooting and harmonic balance pss.

In some cases, placing the probe in the circuit can break some local feedback paths. If this happens, the loop gain calculation will be incorrect. If you are unsure if this could be happening in your circuit, place the probe in a different position in the loop. If the loop gain stays the same, both results are correct. If it changed, try a third point in the circuit. When the two points agree, you have the loop gain.

Setting Harmonics and Sidebands

Shooting

The pss analysis, when shooting is selected, has a minimum of 200 timepoints, so it inherently has frequency domain content through the 100th harmonic of the pss beat frequency. Maximum harmonic is a pss post-processing property that specifies how many harmonics to calculate in the Fourier transform of the waveform in pss. If the number of harmonics is zero, it just means that no frequency domain information should be calculated in the simulation. If the number of harmonics is 10 or less, there is no effect on the PSS waveform. When the maximum harmonics gets larger than 10, the minimum number of timepoints in the pss analysis is increased. 20 timepoints in the period of the highest harmonic are forced. Thus, if the maximum harmonics is raised, the pss waveform gets more timepoints which improves the accuracy of the waveform at the cost of longer runtimes and more memory required to run the pss simulation.

In order for the pstb analysis to correctly measure conversion gain, the wave shape at the highest analysis frequency needs to be accurate, so with the default settings in shooting pss, up to 40 times the pss beat frequency can be used as the maximum frequency in pstb. If you need a higher stop frequency, either set the number of harmonics to maximum frequency in pstb/4*pss beat frequency, or set the pss option *maxacfreq* to the stop frequency in pstb.

Harmonic Balance

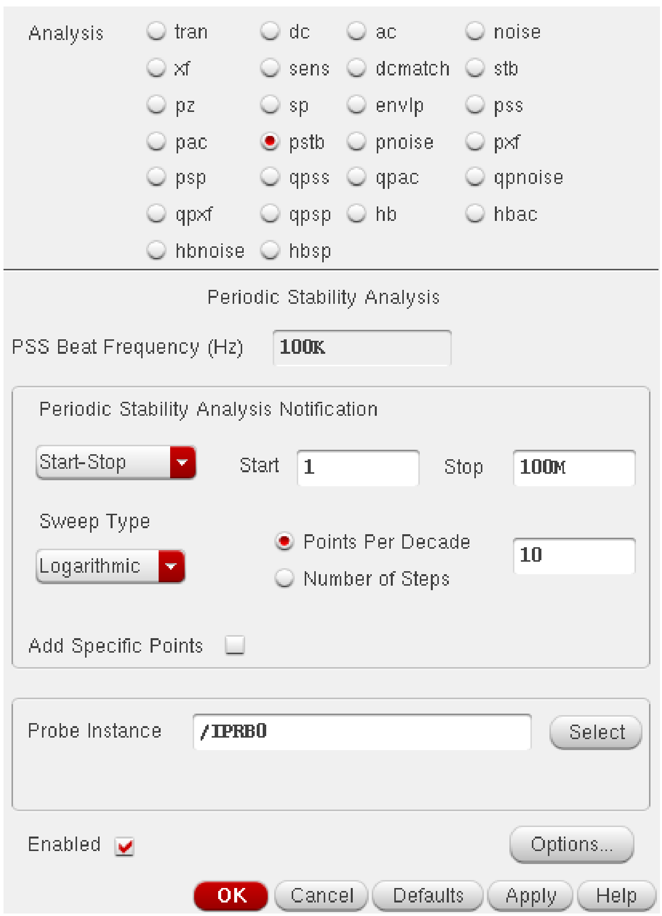
When *Harmonic Balance* is selected in the pss analysis, only the harmonics specified in the pss *Choosing Analyses* form exist in the solution. In the pstb analysis, because only the harmonics specified in the pss-hb *Choosing Analyses* form exist, the maximum frequency term can only be as high as the highest frequency harmonic specified in the pss analysis. If you need a higher maximum frequency in pstb, you need to specify more harmonics in the pss analysis.

ADE Implementation

This section jumps from the theoretical to the practical with examples of how to use pstb. The focus is only on the pstb *Choosing Analyses* form so you can see where the individual settings are made. For examples, with all the steps shown, see the examples section that follows this section.

PSTB Setup

Here is an example of the Choosing Analyses form for a normal loop gain measurement
.



1. At the top of the form the PSS beat frequency of 100KHz is shown.
2. The frequency sweep range specifies the limits and spacing of the loop gain measurement. Note that with this maximum frequency of 100M, either the pss option *maxacfreq* needs to be set to 100M, or the number of pss harmonics needs to be set to 250.

Tip

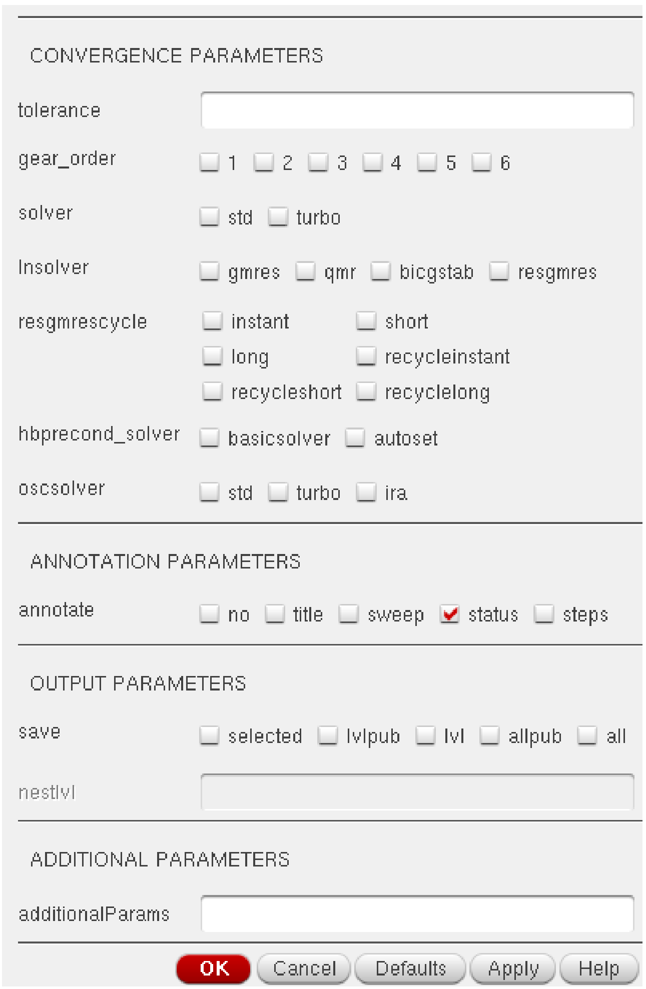
Maxacfreq forces 5 timepoints in the period of the frequency specified for maxacfreq. PSS harmonics above 10 forces 20 timepoints in the period of the highest harmonic. In this example, the pss beat frequency is 100K, and the highest frequency in pstb is 100M, or a ratio of 1000. Since pss harmonics force four times as many timepoints as maxacfreq does, to get equivalent runs, you need 1000/4 or 250 harmonics in pss.

3. Linear and log sweeps are provided. Generally, log sweeps are used. The number of points depends on how rapidly the loop gain changes with frequency in your circuit. High Q circuits will need more resolution than shown in this example.

4. A probe must be used in the circuit and specified in the *Probe Instance* field. For single-ended circuits, an `iprobe` from `analogLib` should be used. For differential circuits, a `diffstbprobe` should be used. When `diffstbprobe` is used, you will have an additional choice in the *Choosing Analyses* form that allows the selection of differential or common-mode measurements of loop gain.

PSTB Options

Below is the pstb options form.



Solver

The solver option is used only for driven circuits. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pstb analysis frequency is very close to the frequency of one of the harmonics in the pss result, warning messages will appear in the pstb output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but which takes longer to run than the turbo solver.

Oscsolver

The *oscsolver* option is used only for oscillators. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the pstb analysis frequency is very close to the frequency of one of the harmonics in the pss result, warning messages appear in the pstb output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but takes longer to run than the *turbo* solver.

For very large oscillator designs, the ira oscsolver may take less memory and run faster with no loss in accuracy, however, it is less robust for convergence compared to *turbo* or *std*.

Hbprecond_solver

This option is only available only when harmonic balance selected in PSS and APS is used.

The basic solver is the only solver available in standard Spectre when harmonic balance is selected. *autoset* is the default solver in harmonic balance when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window. If you have stagnation, it will save a small amount of time to set this option to *basicsolver*.

The following options are not commonly used:

Tolerance

Leave this option at the default value.

Pstb uses an iterative solver to calculate the loop gain at the specified frequencies. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough. The tolerance option specifies that accuracy for pstb. For shooting, the default tolerance is 1e-9. For driven circuits where HB is the pss engine, the default is 1e-6. For oscillators where hb is selected for the pss engine, the default is 1e-4.

Gear_order

Do not change this option.

Lnsolver

Leave this option at the default.

Each pstb frequency point loop gain solution is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases with increasing iterations. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

Resgmrescycle

Leave this option at the default. For the *resgmres* linear solver, there are several different options.

Annotate

This option controls the level of detail in the Spectre output log. Selections to the left in the form produce less detail. Selections to the right in the form add increasing detail.

Save

This option is not implemented at the current time.

Nestlvl (Shooting and Harmonic Balance)

This option is not implemented at the current time.

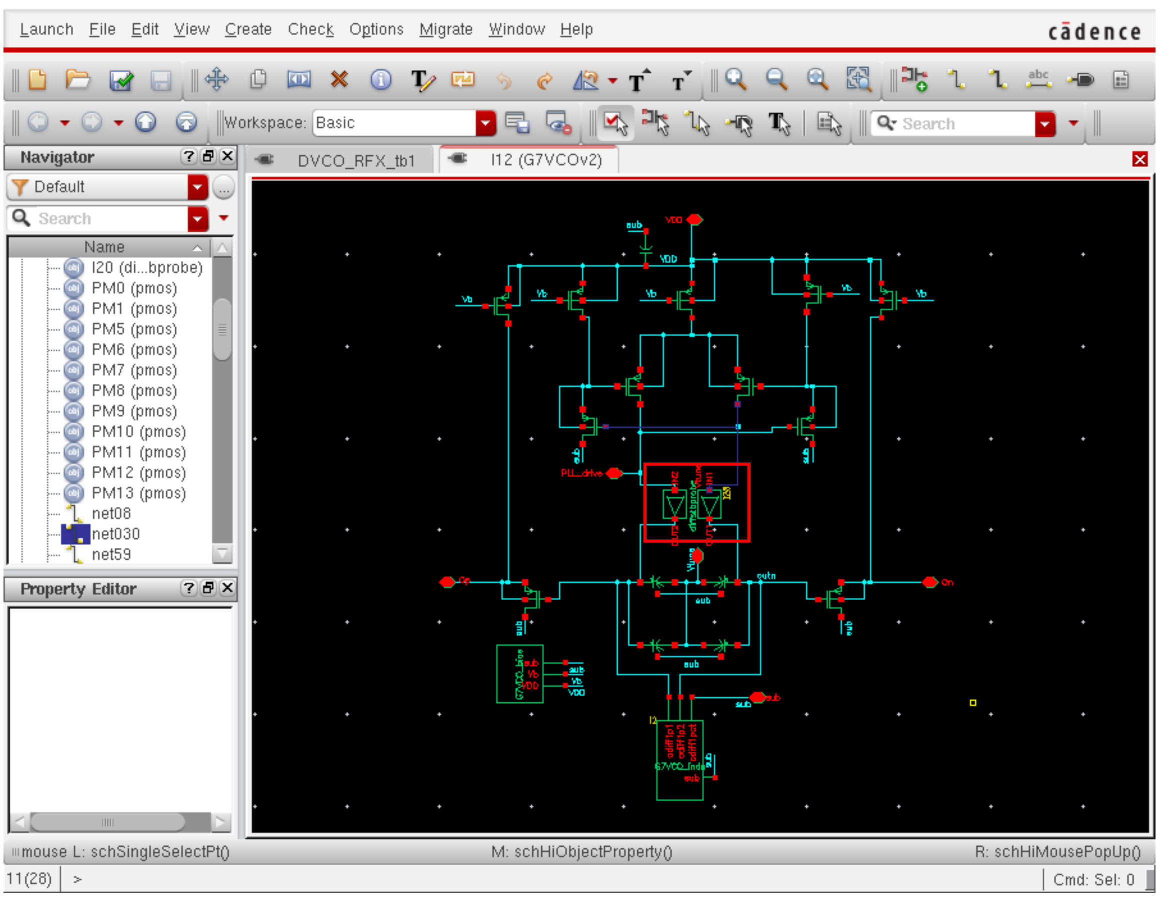
AdditionalParams

additionalParams is typically used for new features that are being beta tested. `keyword=value` pairs are expected in this field.

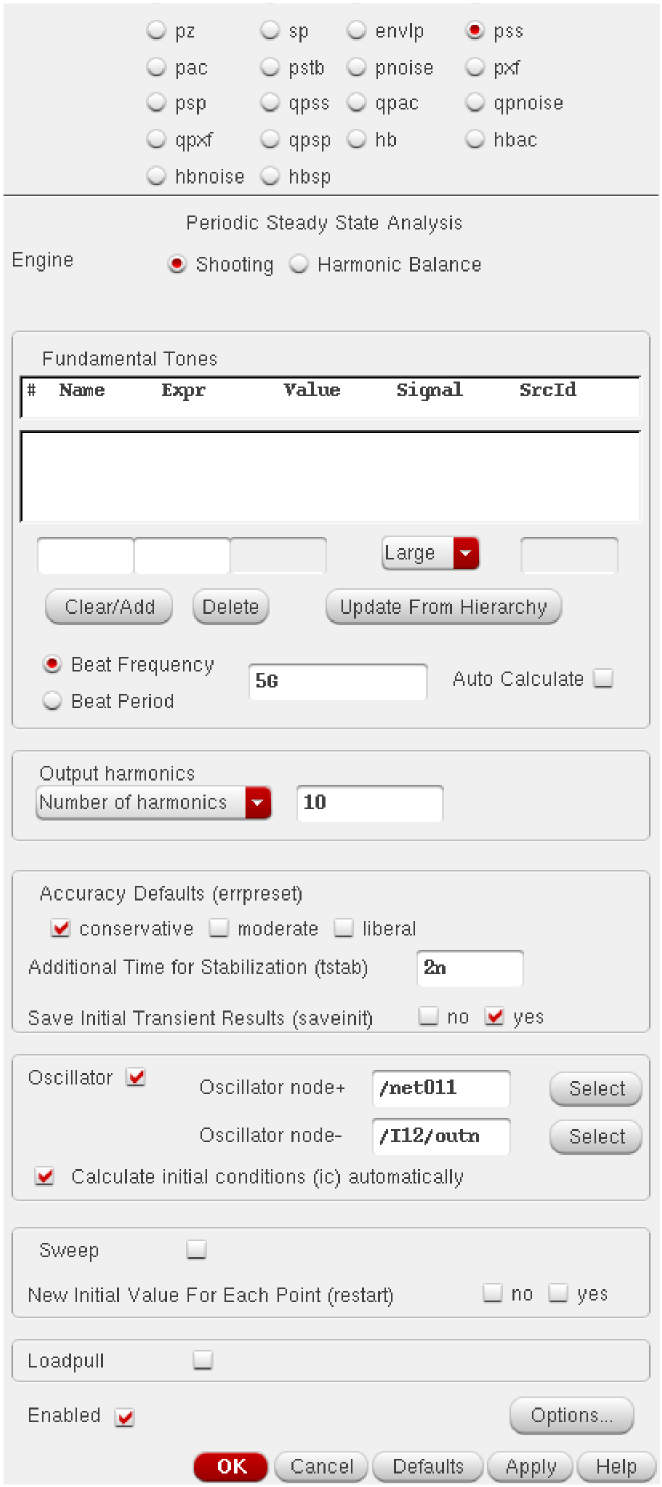
For more information about the other options, type `spectre -h pstb` at the command prompt in a Unix shell window.

Oscillator Example

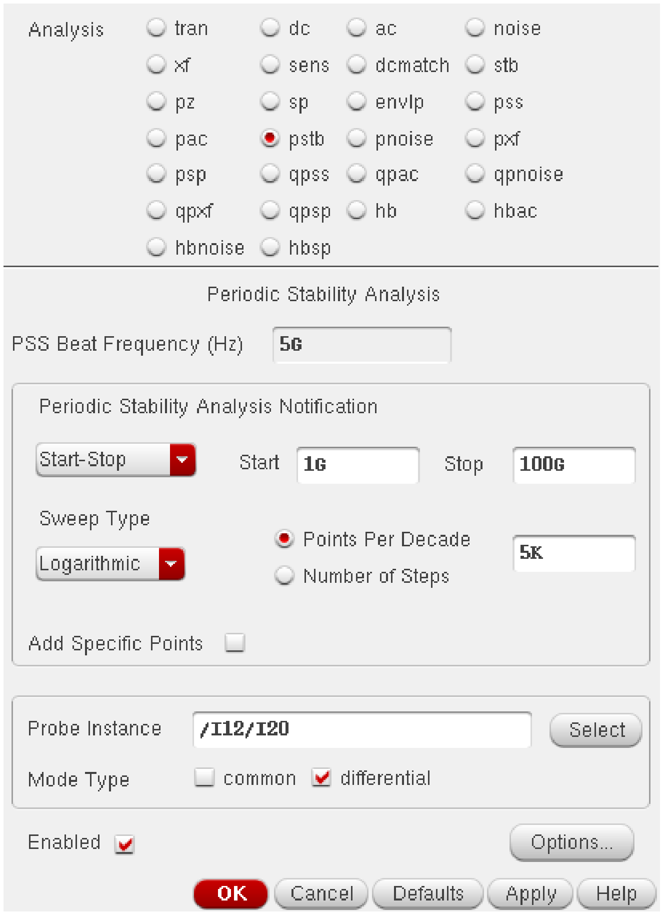
For the differential oscillator circuit below, a `diffstbprobe` has been placed at mirror locations in the differential circuit. This is shown in the red box below.



1. First, a pss analysis needs to be set up. For more information, refer to the pss section at the beginning of this chapter.



2. Now set up the pstb analysis. The resolution of this sweep is quite high.

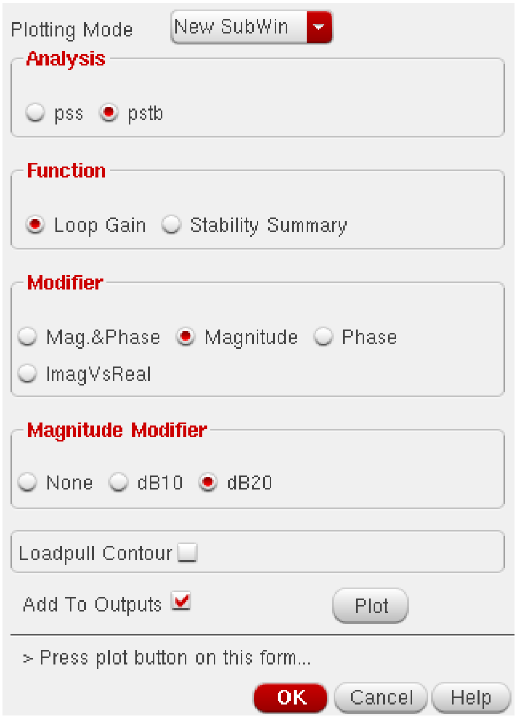


The frequency sweep range, and the number of points will be determined by your circuit. In this case, from linear stb analysis, all the interesting variations started at 1GHz. Narrow peaks were detected, which forced many points in order to resolve the amplitude of the peaks.

3. Differential mode is selected because this is a differential oscillator. The common mode loop gain will be checked later to make sure that only the differential mode should produce oscillations.

4. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window.

5. Now the loop gain and phase are plotted using the *Direct Plot Form*.

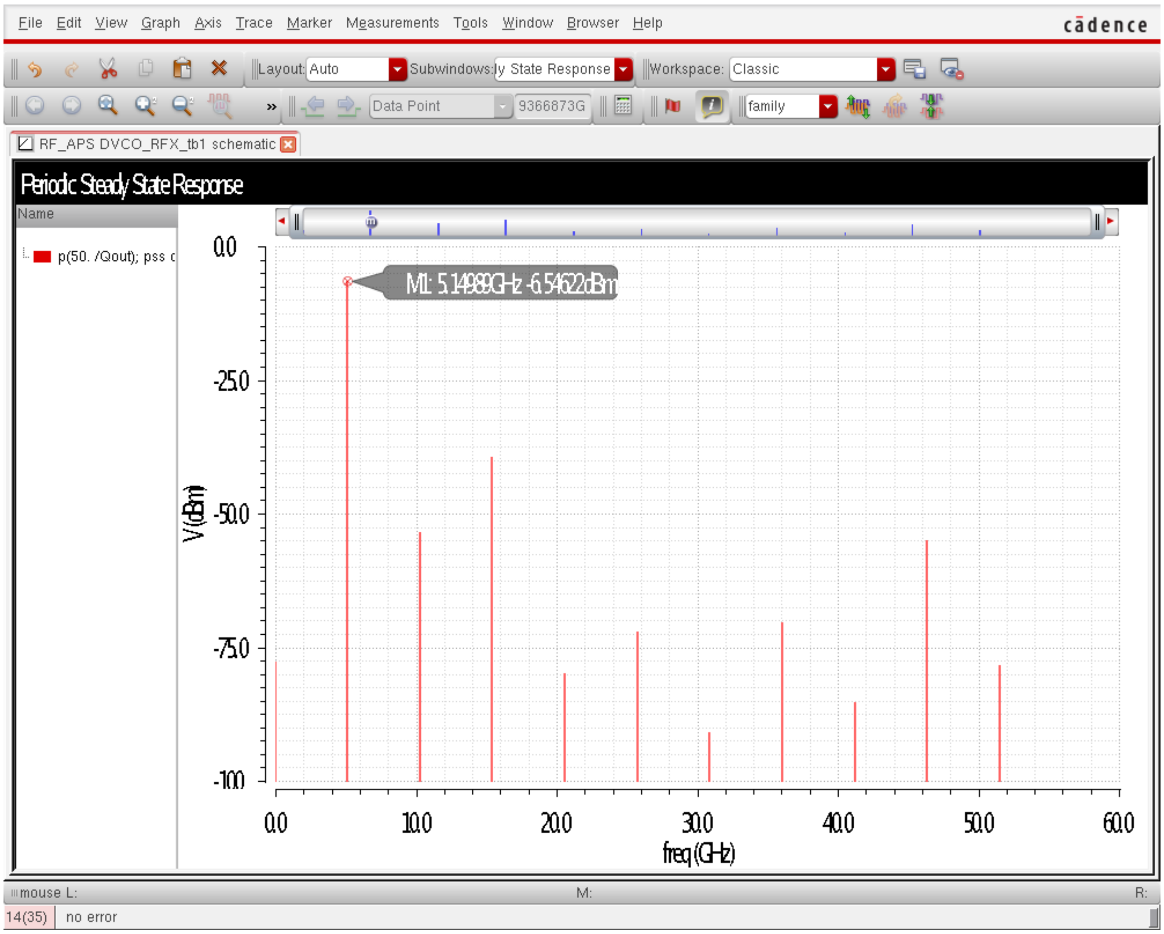


6. Now, the plot of the loop gain and phase is shown. The red trace is loop gain, and the blue trace is the phase.

7. A vertical marker has been placed at the zero degree point on the phase curve. Note that the gain is very close to unity, and the phase is very close to zero. Remember that in pstb, the summing junction is included in the analysis, so the condition for oscillation is unity gain and zero or multiples of 360 degrees of phase. The frequency readout shows 5.1499GHz.

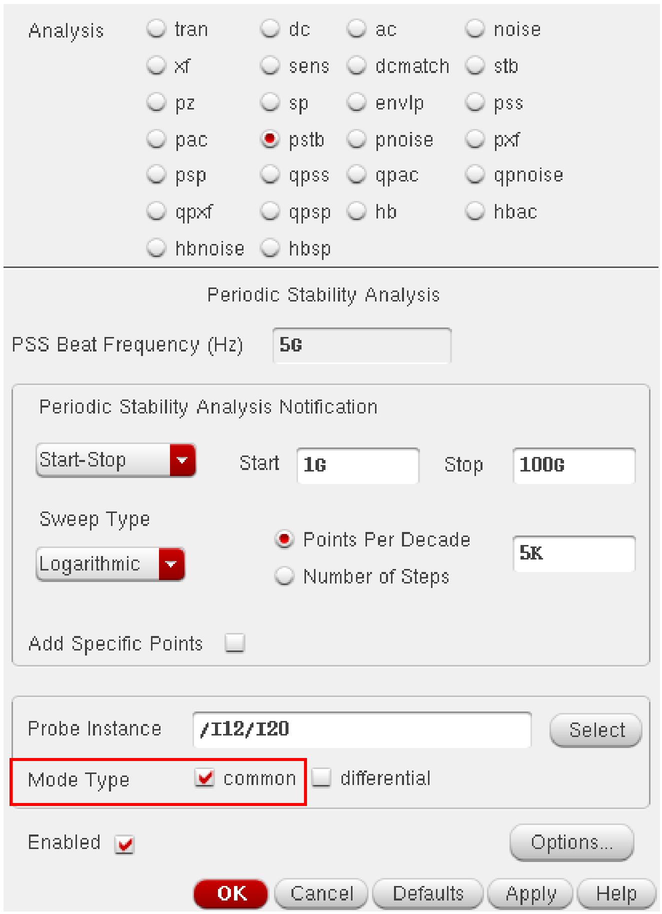
8. For an oscillator, you should see almost exactly unity gain and a phase of almost exactly zero or multiples of 360 degrees. Small differences are to be expected because of finite resolution in the frequency sweep and/or interpolation error in the markers.

9. The plot of the output spectrum is shown below.

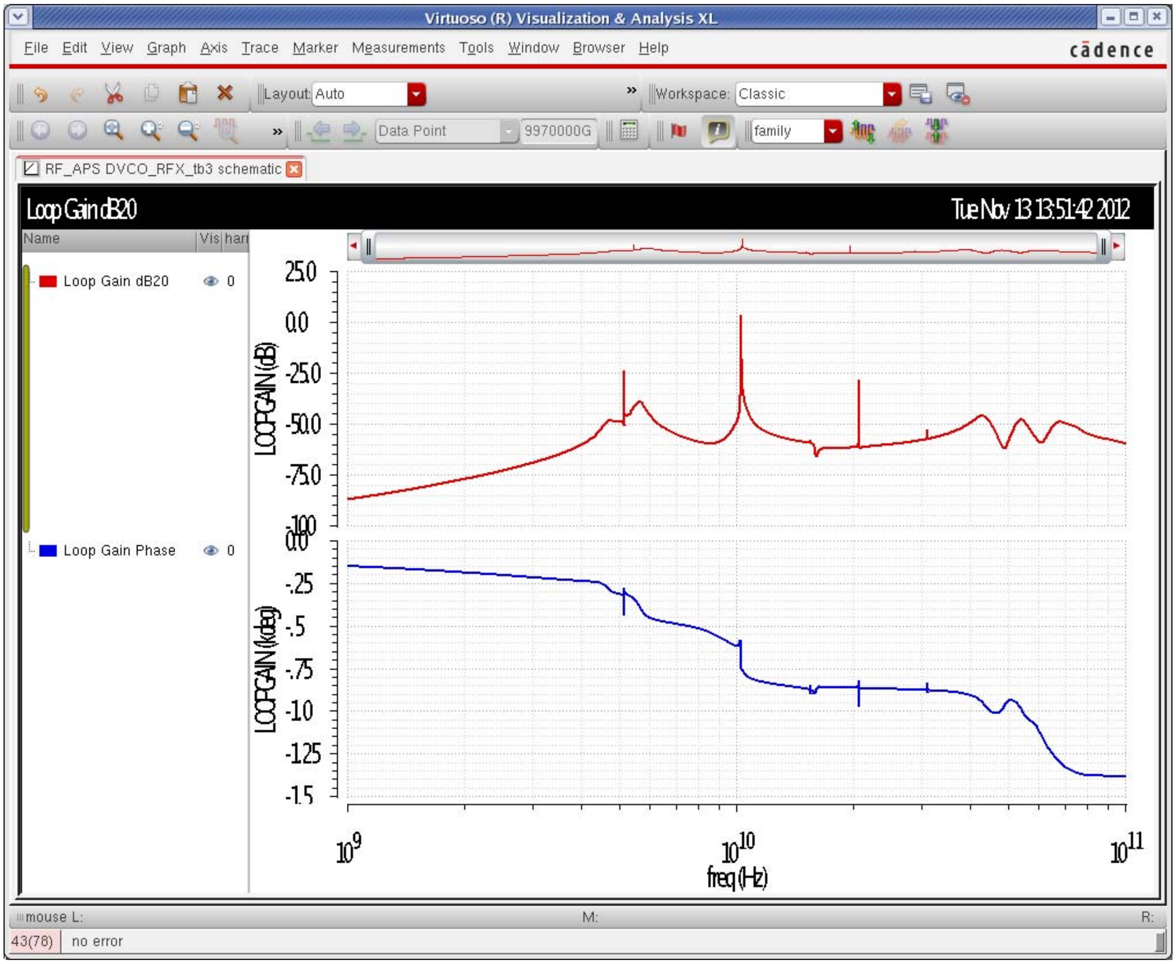


10. Note that the frequency in the large-signal pss is the same as the small-signal pstb analysis. The nonlinearities of the oscillator are taken into account in the pstb simulation.

Whenever you have a differential circuit, the common-mode loop gain should be checked as well. Select common mode in the *Choosing Analyses* form shown below.



11. The common-mode loop gain and phase is shown below.



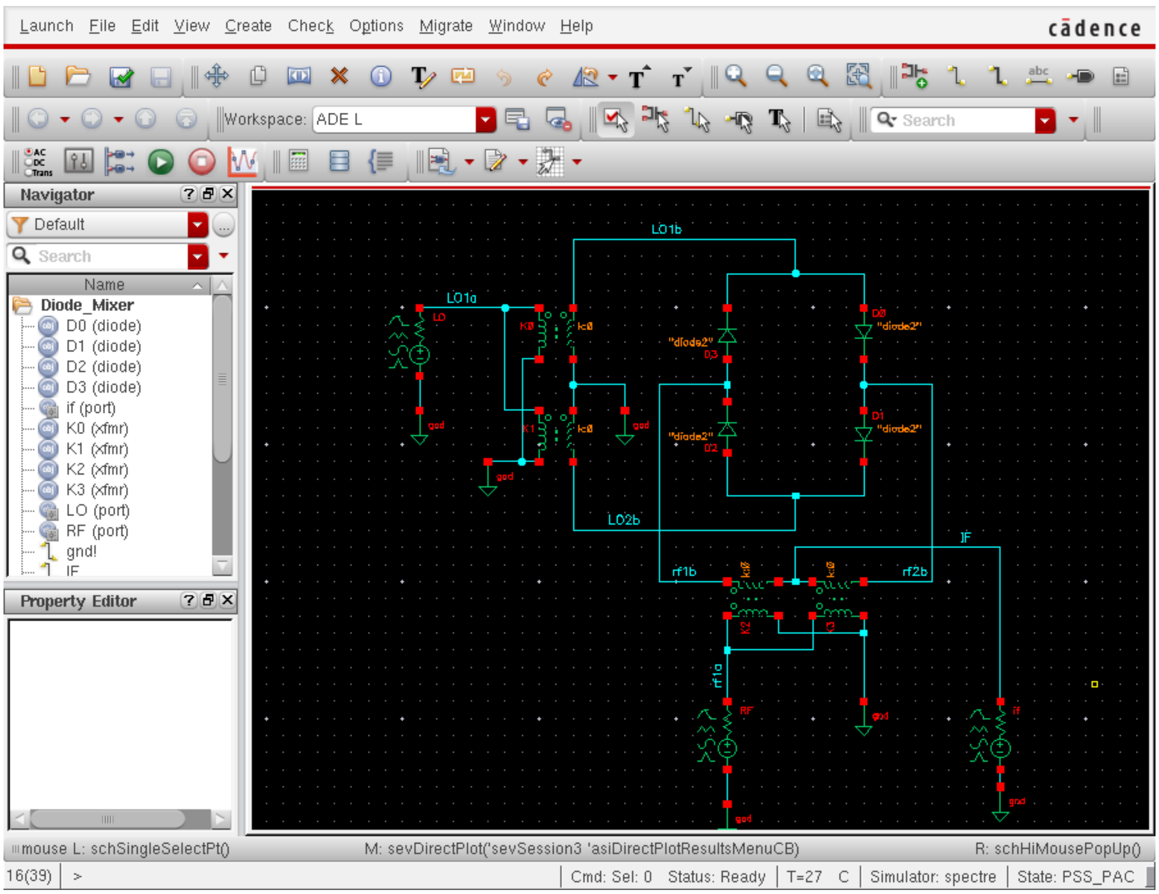
12. The loop gain is above unity near 10GHz. This needs to be checked in more detail for the possibility of a common-mode oscillation.

Periodic S-parameter Analysis (PSP)

Psp is provided to measure S-Parameters for circuits that translate frequencies. All the basic assumptions of S-Parameters apply to psp, with the addition of having different input and output frequencies. Psp can be used for up or down-conversion applications.

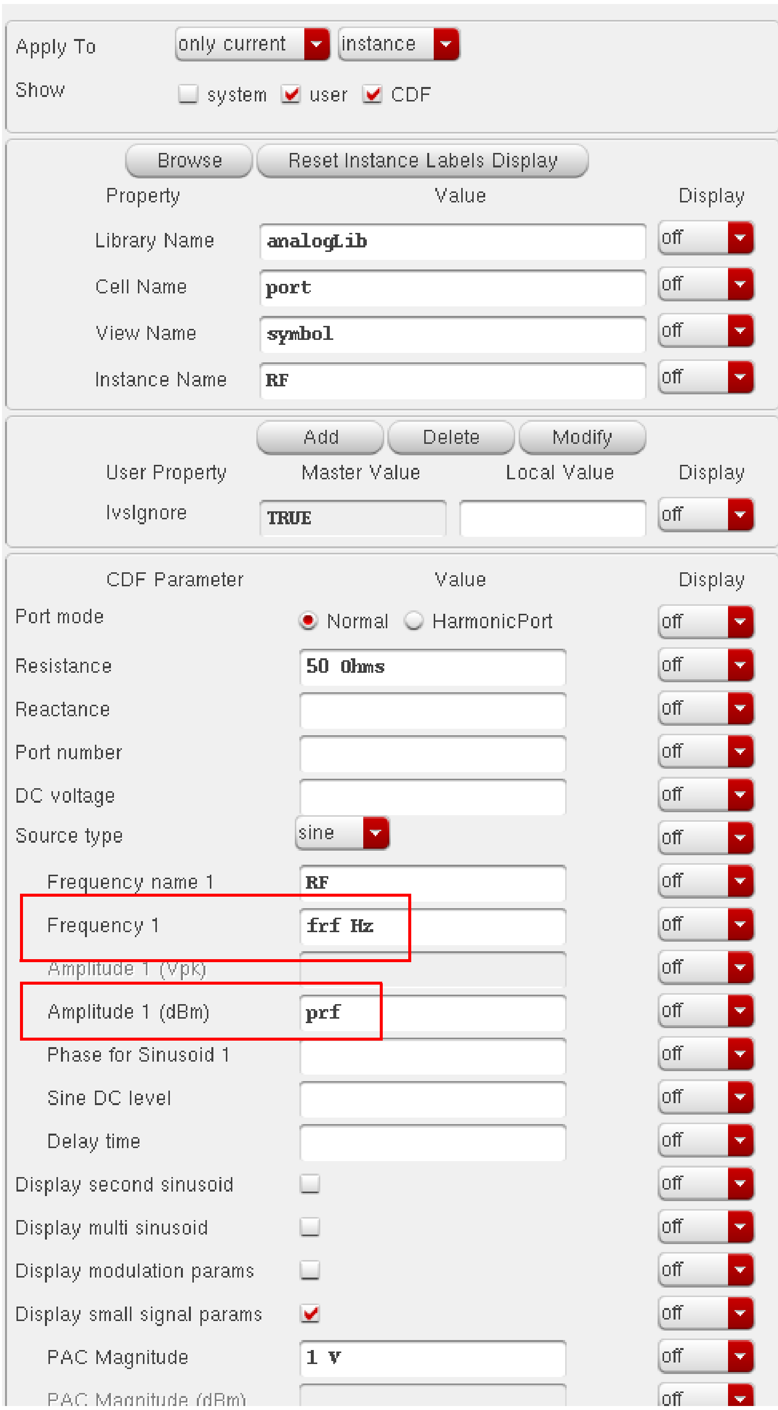
Example: Down-Conversion Diode Mixer

Consider the double-balanced diode mixer circuit below.



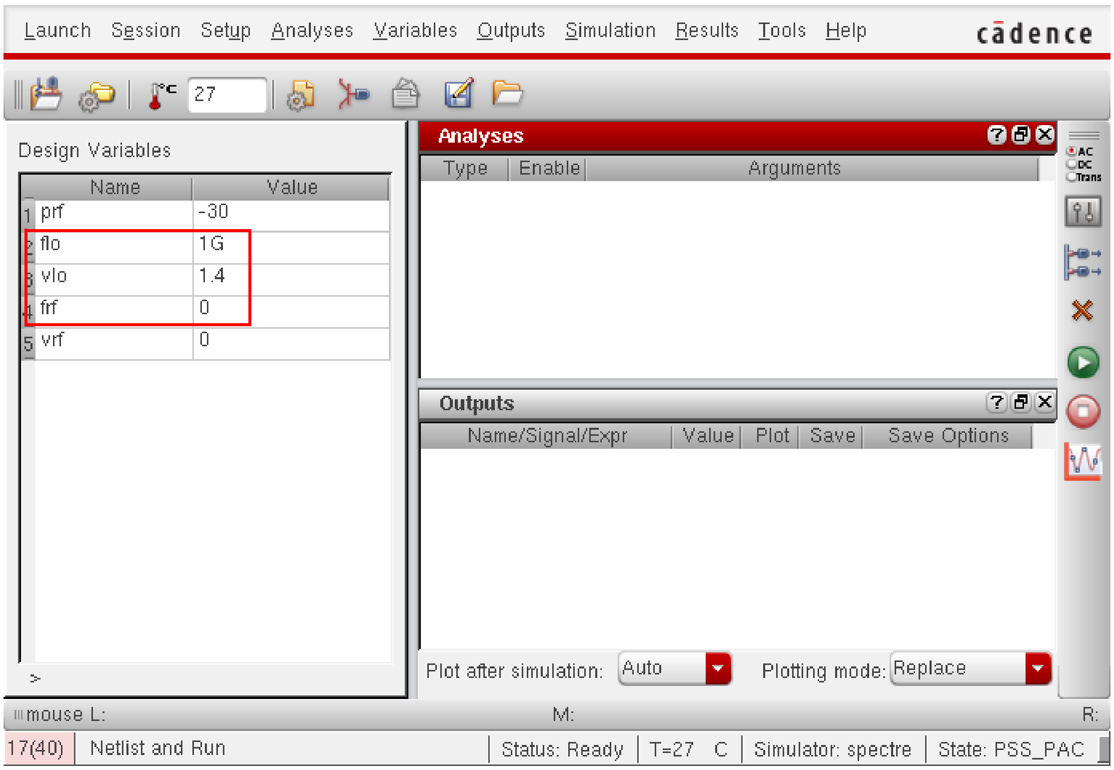
The input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. This allows changes in frequency or amplitude from the ADE environment without needing to change the schematic itself.

The property list for the RF input port is shown below.



The same strategy is used for the LO port.

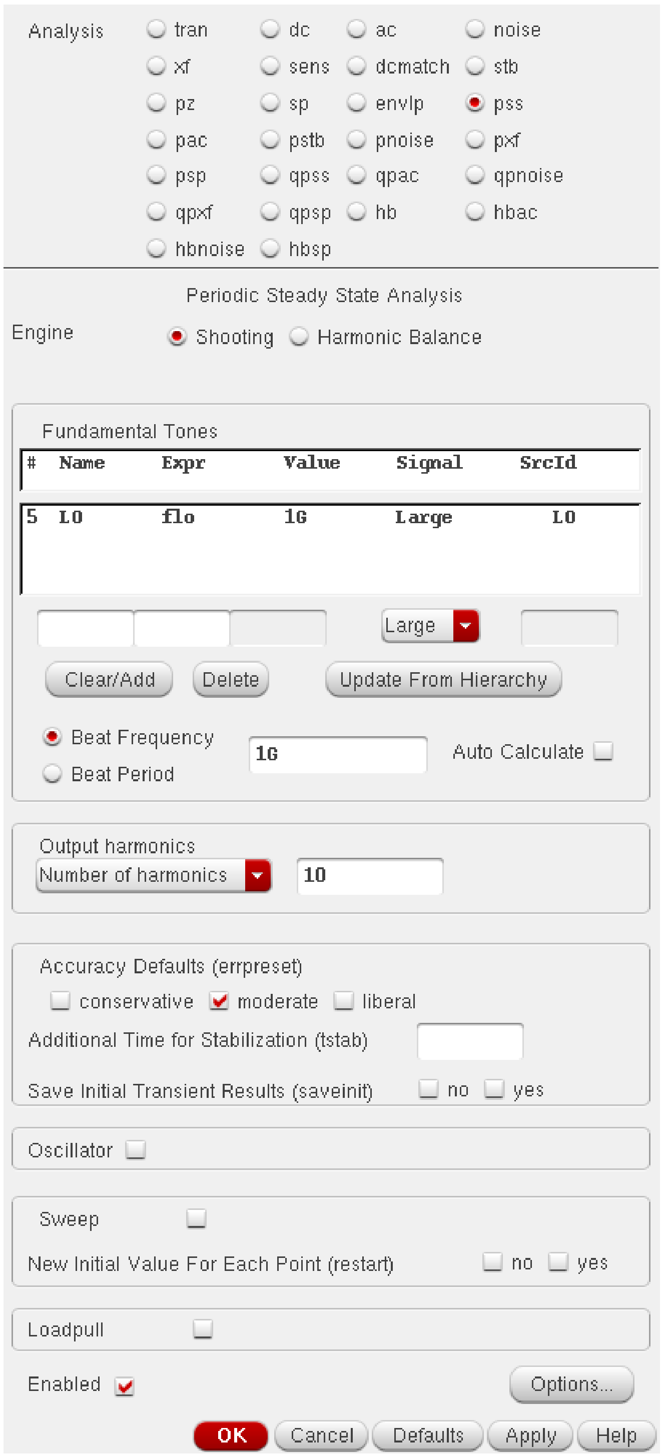
The easiest way to get the variables list into the ADE is to select *Variables -> Copy From Cellview*. To set the values, select the value field to the right of the variable, type in the desired value, and press *Enter.* When you have set the variables, if you want to add the values back into the cellview, in the ADE window, select *Variables - Copy To Cellview*, and then perform a *Check and Save* in the schematic.



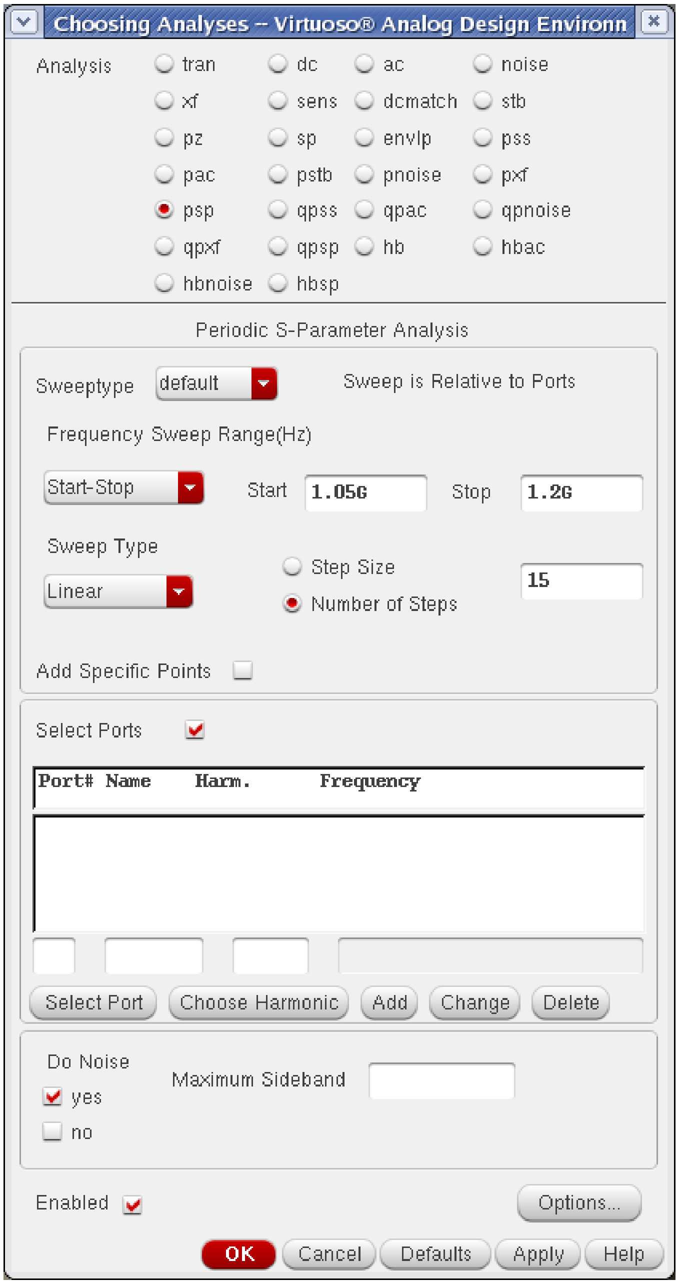
The RF input frequency is disabled (by setting the frequency to zero), and the LO is at 1GHz with a 1.4 volt peak signal. This is +13 dBm, which is a common drive level for a diode mixer.

The basic strategy is to apply the signal that causes the frequency translation in the pss analysis, and then measure the frequency-shifted S-Parameters using psp. Doing this allows measuring the input match at the input frequency, and the output match at the output frequency in the same analysis. For the forward and reverse gain terms, the specified frequency translation is taken into account.

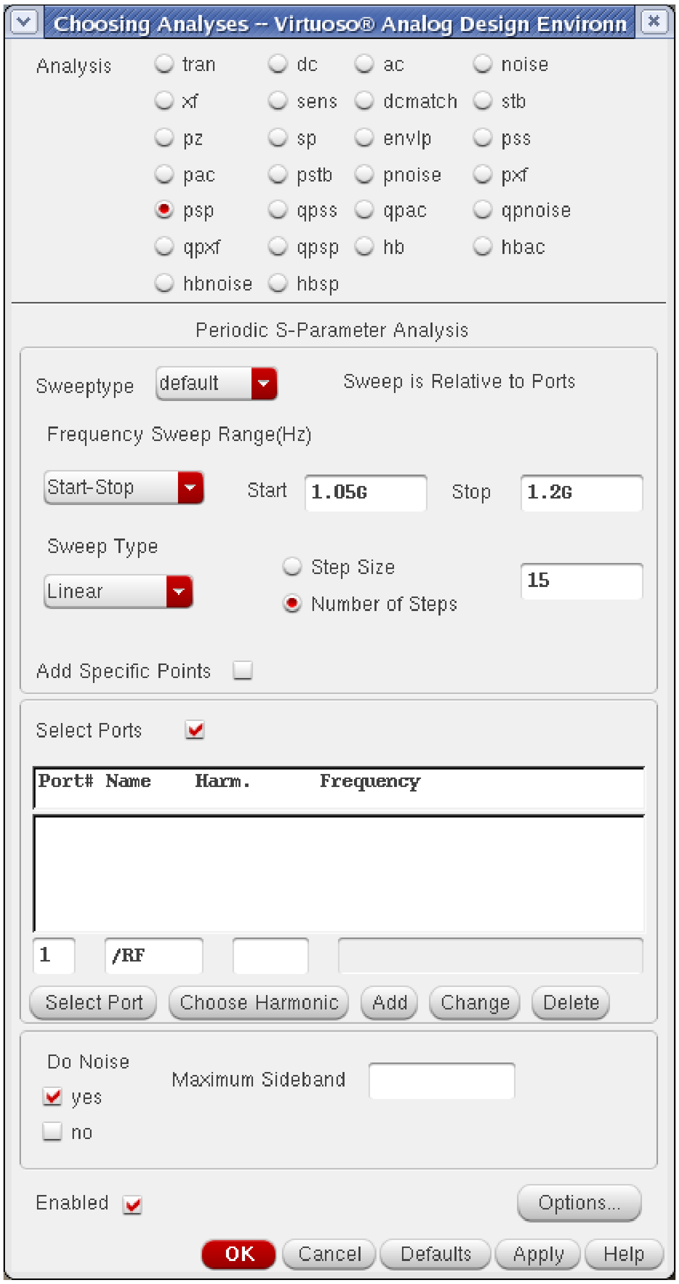
The pss analysis needs to be set up first. The example below shows a single input at 1GHz with 10 harmonics. For more information, see the pss section at the beginning of this chapter.



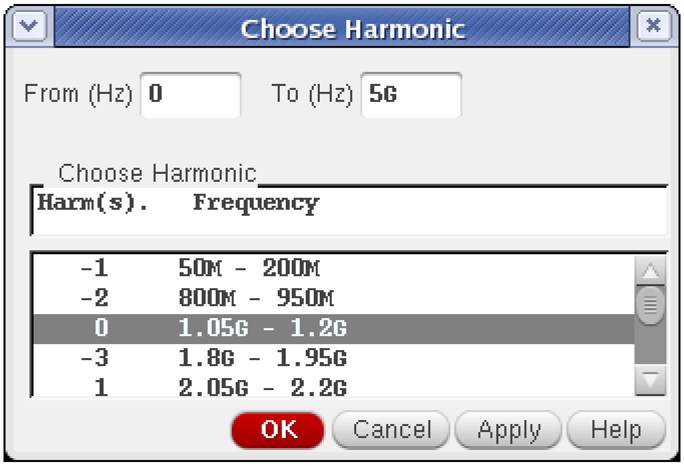
Next, the psp analysis needs to be set up. The frequency range depends on your application, and can be either an input frequency range, or an output frequency range. The actual frequency for each port-frequency definition will be set up later. In this example, the frequency range is just above the LO frequency at 1GHz.



Start with Port 1. In the *Port #* column, type 1. In the *Name* column, either type the instance name, or use the *Select Port* button just below the edit line to select the port in the circuit. In this case, Port 1 is the RF port.

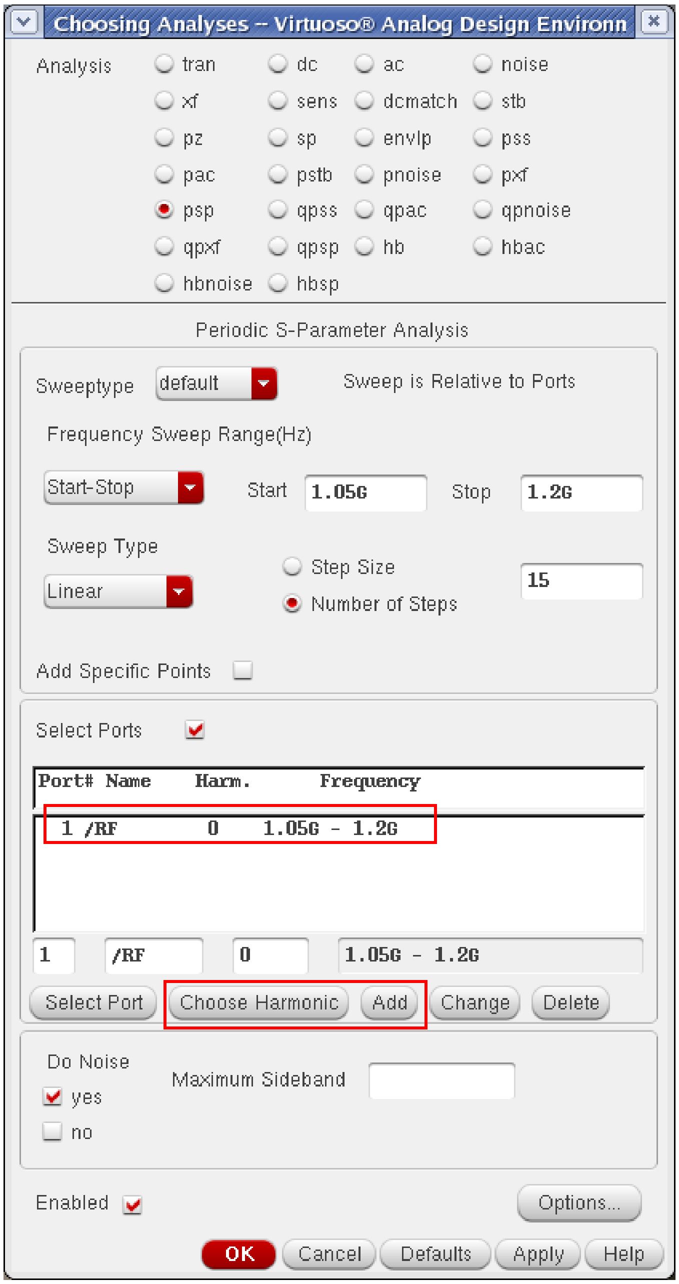


Now click *Choose Harmonic*. In the *Choose Harmonic* window, select the frequency range you want associated with that port number. For the down-conversion application, The input is just above the LO frequency of 1GHz.



Click OK.

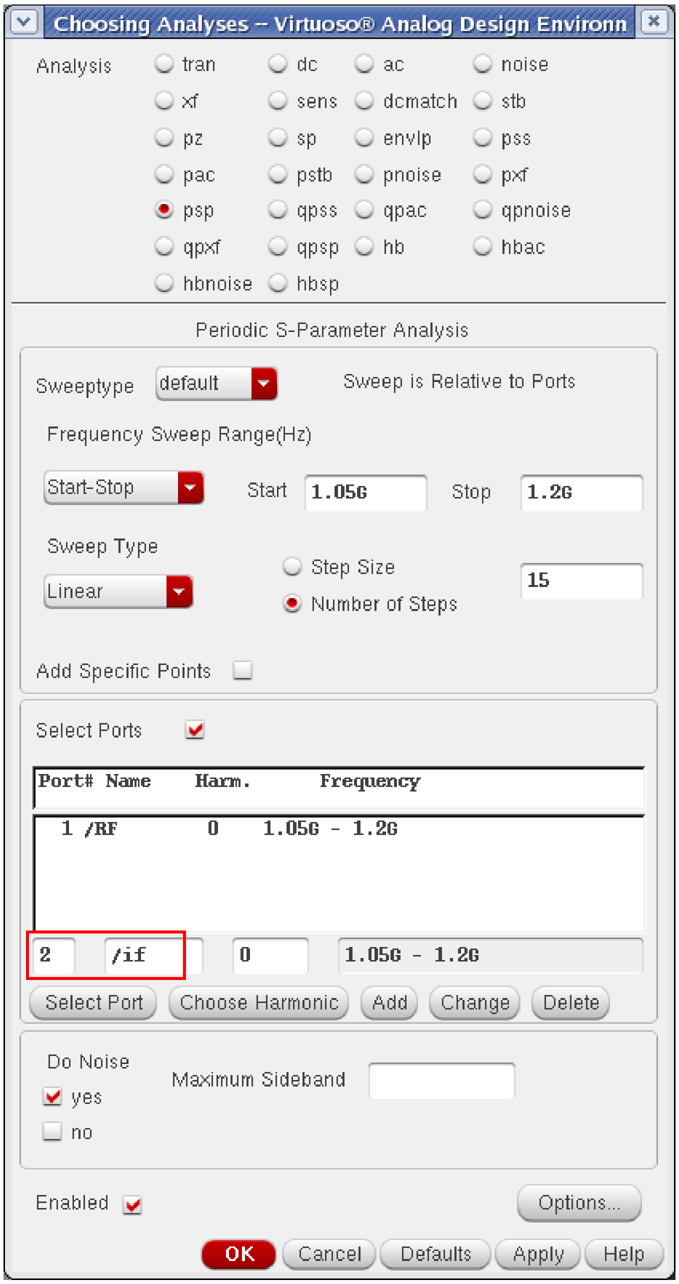
Click *Add* located to the right of the *Choose Harmonic* button. This adds that port and frequency definition to the list for the analysis. By convention, Port 1 defines the input frequency.



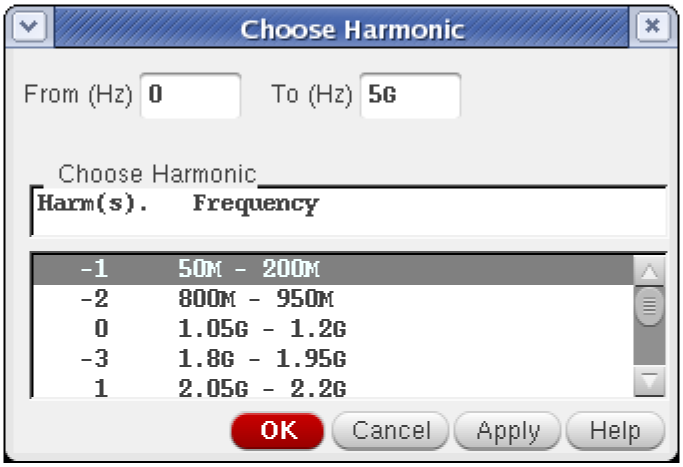
Note that Port 1 has both a port and a frequency definition associated with it. In psp, both are necessary for a port definition.

In psp, multiple port-frequency definitions can be run at one time.

The next step is to measure RF to IF conversion gain. Type 2 in the *Port #* column. Click *Select Port*, and select the IF port.

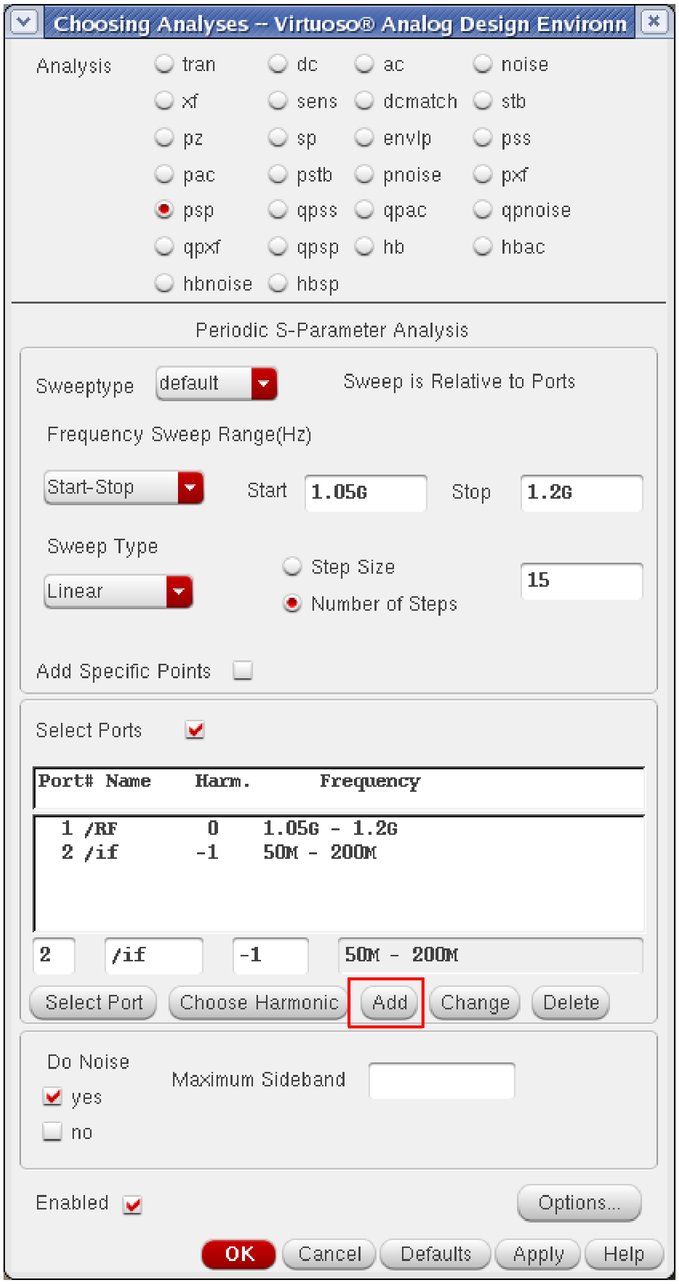


Now define the frequency range for that port. Click *Choose Harmonic*, and select the IF frequency from the list.

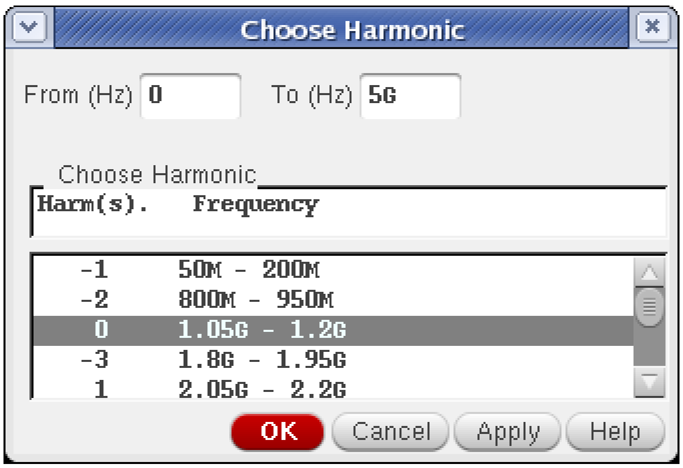


Click OK.

Click *Add* located to the right of the *Choose Harmonic* button. The second port-frequency definition is added to the list. By convention, Port 2 defines the output frequency range.

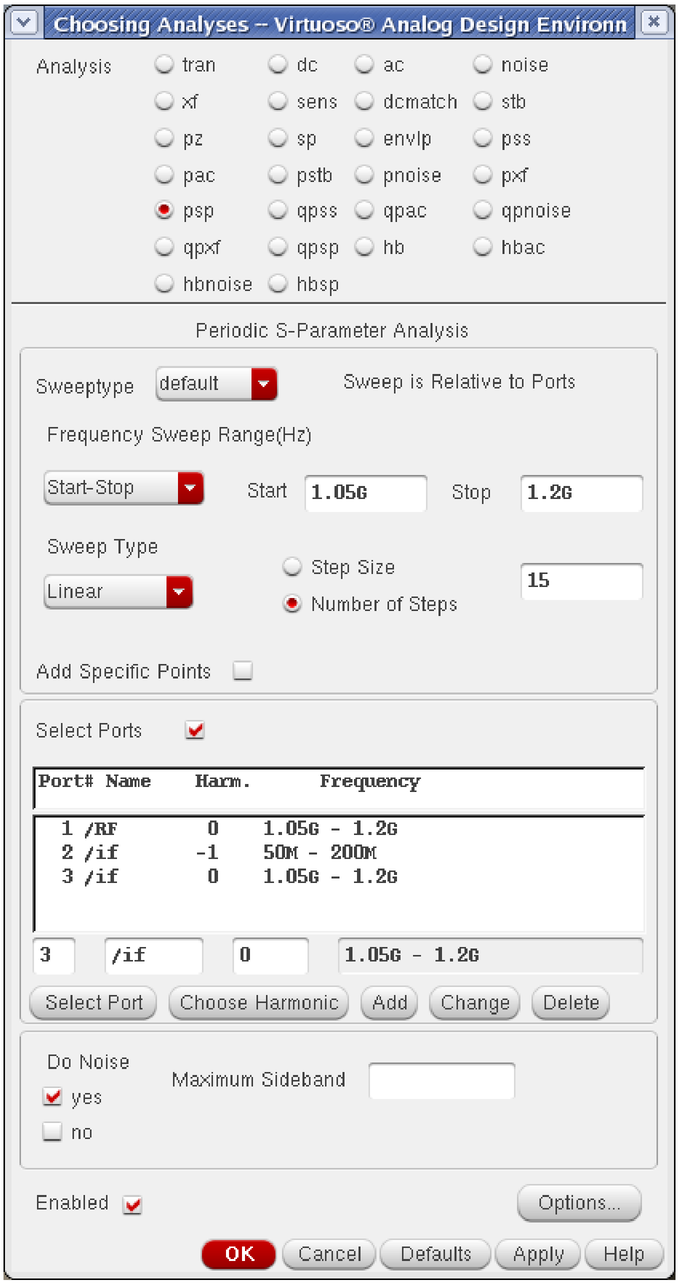


Now measure the RF to IF isolation. Type 3 in the *Port #* column. Click *Select Port*, and again select the IF output port. Click *Choose Harmonic*, and select the input frequency range.

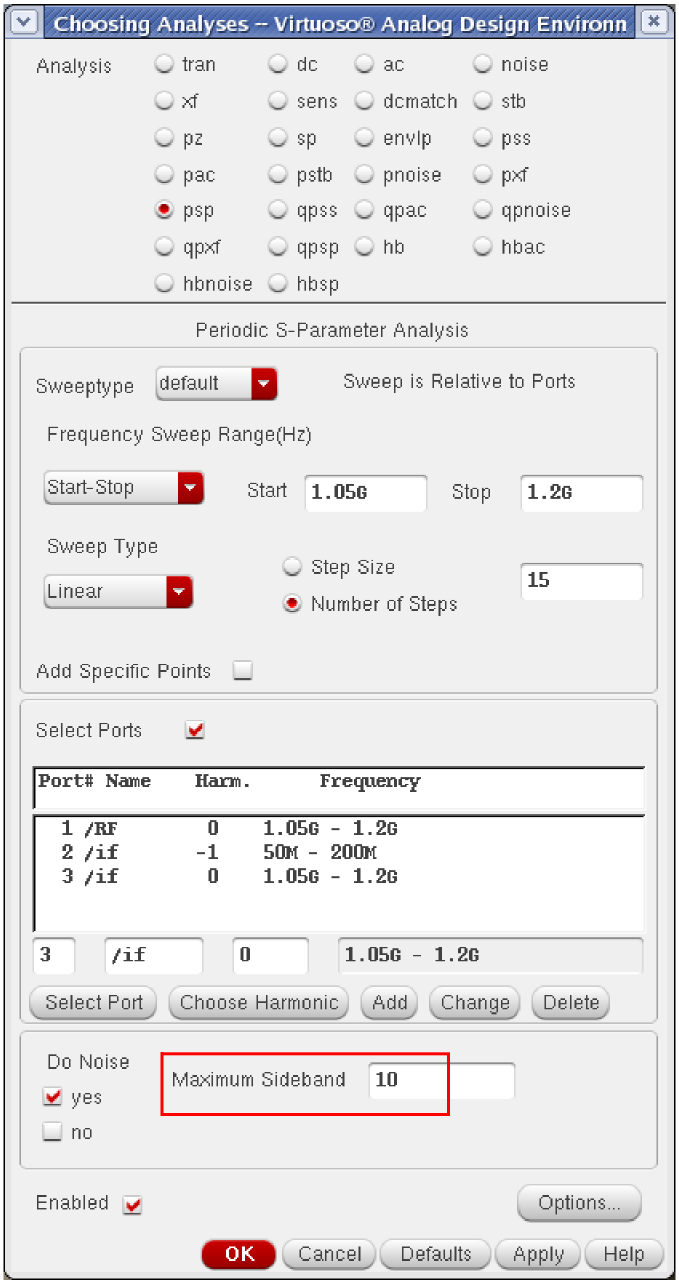


Click OK.

Click *Add* located to the right of the *Choose Harmonic* button. The third port-frequency definition is added.



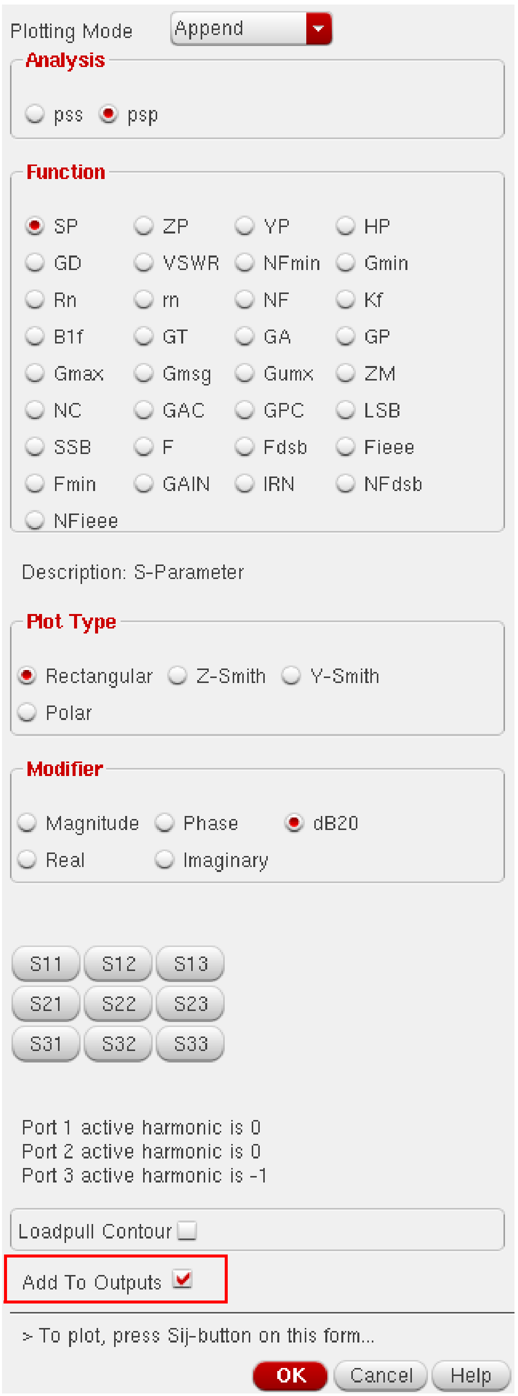
If a noise simulation is desired, set the *Maximum sideband* field to the same number of harmonics used in the pss analysis.



With the above port-frequency definitions, S_{11} measures the input match at the RF frequency. S_{22} measures the output match at the IF frequency. S_{31} measures the RF to IF isolation, and s_{21} measures the RF to IF conversion gain.

Once the analyses are set up, the simulation can be run. When the simulation finishes, select *Results - Direct Plot - Main Form* in the ADE window.

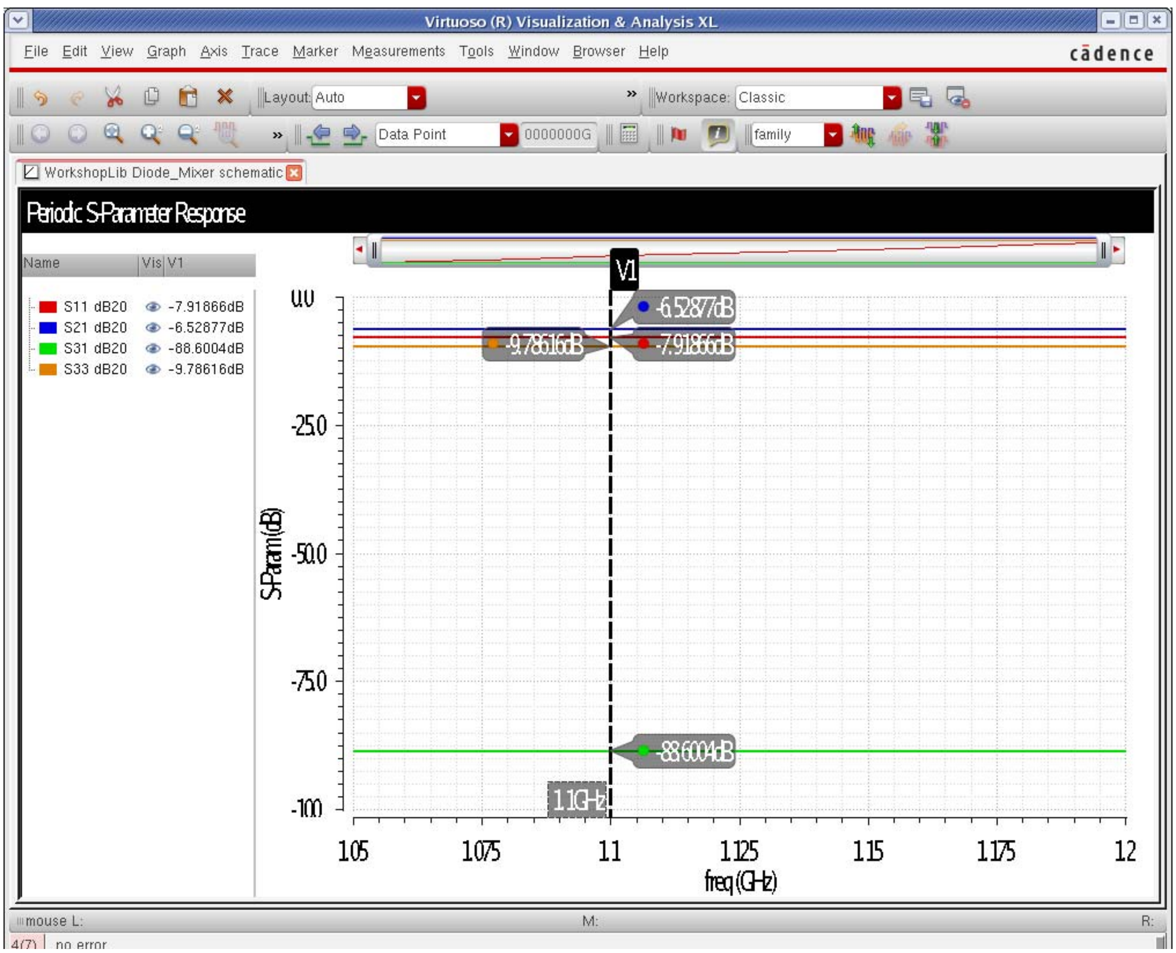
The *Direct Plot Form* is displayed, as shown below.



Note: Many of the measurements in the *Direct Plot Form*, like Kf, and B1f are only defined for two-port networks. In the *Direct Plot Form*, if you select one of those items for plotting, you will see a message that says the data is not 2 X 2. Also note that Kf and B1f are only defined for amplifiers where there is no frequency translation. If you run a two-port simulation on a mixer like this, and you select Kf to be plotted, you will see a message that the harmonic of the two ports are not the same, and as a result, this cannot be plotted.

If you click *Add To Outputs*, an expression is entered in the ADE outputs section so that the next time you simulate your circuit, this measurement plots automatically.

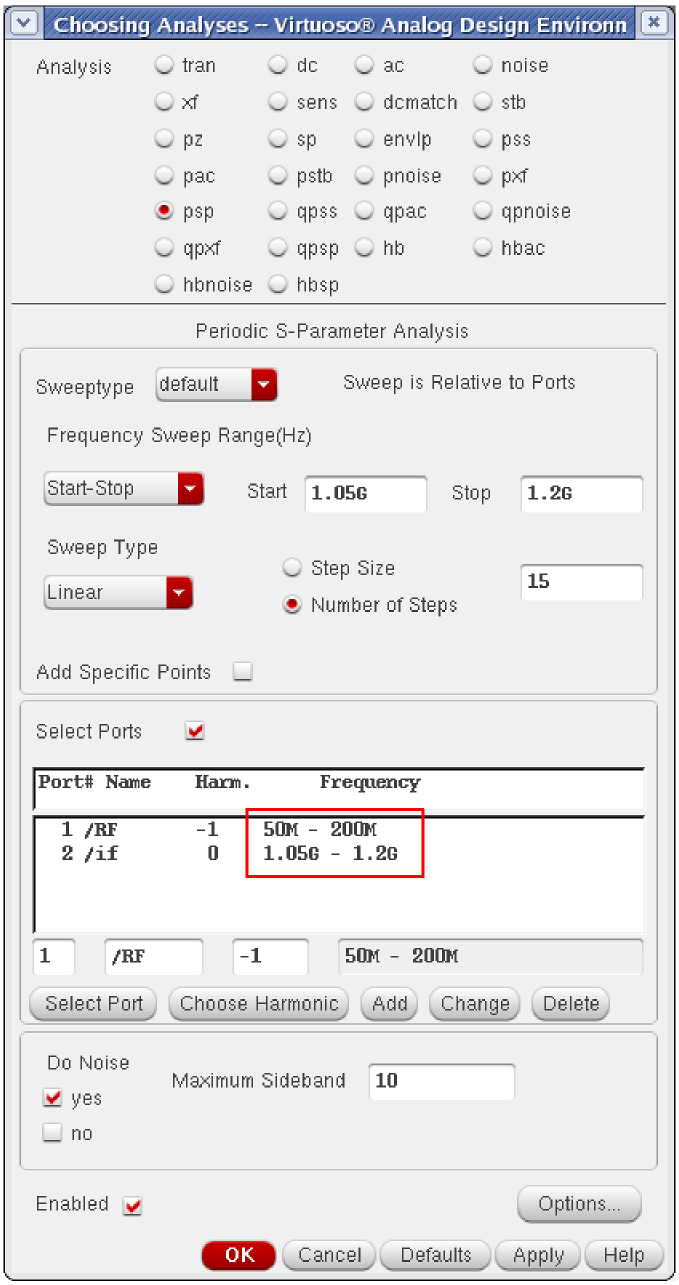
Here are the S11, S21, S31, and S33 plots.



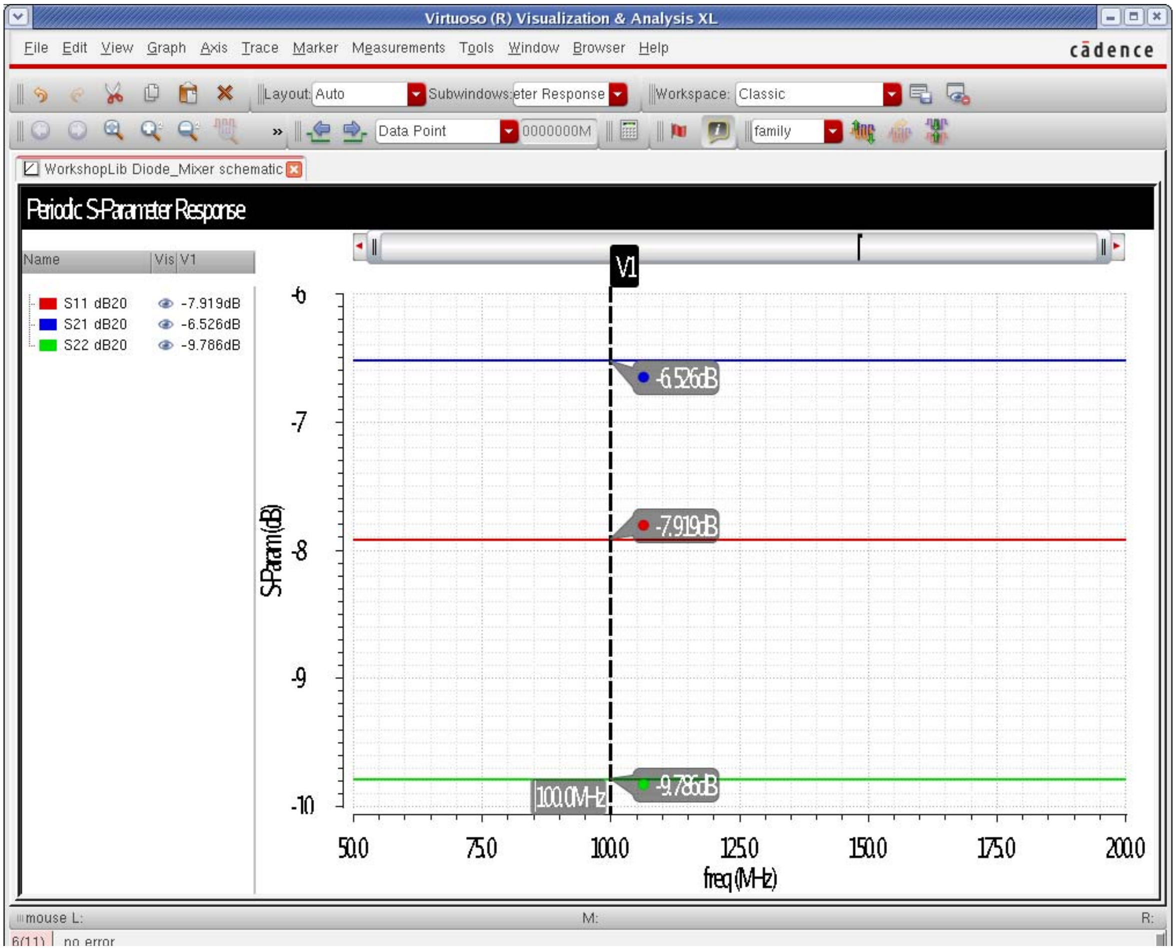
The conversion loss is about 6.5 dB. The input match is about -7.9dB. The output match is about -9.8dB. The RF to IF isolation is about 88 dB.

Note that these measurements can be plotted on a Smith Chart as well. The Smith Chart capability is only available from the *Direct Plot Form*. Expressions in the ADE outputs section will always plot on a rectangular plot. The Smith Chart has not been shown here because this circuit is inherently very wide band. The curve on the Smith Chart would be tiny. An example using a Smith Chart is shown later on in the Examples section.

Psp can be used for up conversion as well. To do that, just change the frequency definitions for the ports. Since IF to RF gain is usually not needed, only a two-port measurement has been set up. Note the frequency difference for Ports 1 and 2.



Now run the analysis. S11, S21, and S22 are plotted, as shown below.



Overview of Simulation Capabilities

Psp is similar to sp in that power gain measurements are made. Sp is based on the DC operating point, so it can only produce results that are linear. All the frequencies are the same for all the S-Parameter measurements in the sp analysis.

Psp uses the pss analysis results as the basis for making the S-Parameter measurement. Because the PSS result is the operating point solution for psp, the input can mix with any of the harmonics that are present in the pss analysis for the frequency-translated terms. The term SXX is not enough to describe the system. We need not only port numbers, but also the frequency needs to be defined for each port definition.

In the example in the preceding section, the LO was at 1GHz and the specified frequency was just above the first harmonic of the LO. In the psp definition for each port, both the port name and the frequency that is associated with that measurement need to be defined.

Note: Some measurements like Kf are only defined for systems that do not translate frequency. In psp, because the calculation is based on the pss large-signal solution, you can sweep the input power to an amplifier, and measure the stability factor as it changes with the input amplitude. In this case, the input and output frequencies are the same, so Kf is defined. If you have different frequencies on the input and output, Kf cannot be plotted because it is not defined. The *Direct Plot Form* will say that term cannot be plotted. Otherwise, all the parameters that can be plotted in sp are available in psp. Also, note that Kf and B1f are only defined for 2-port networks.

Noise analysis is available in psp, and is the same as pnoise. Please see the pnoise chapter for noise information.

Setting Harmonics and Sidebands

Shooting

The pss analysis, when shooting is selected, has a minimum of 200 timepoints, so it inherently has the frequency domain content through the 100th harmonic of the pss beat frequency. Maximum harmonic is a pss post-processing property that specifies how many harmonics to calculate in the Fourier transform of the waveform in pss. If the number of harmonics is zero, it just means that no frequency domain information should be calculated in the simulation. If the number of harmonics is 10 or less, there is no effect on the PSS waveform. When maximum harmonics gets larger than 10, the minimum number of timepoints in the pss analysis is increased. 20 timepoints in the period of the highest harmonic are forced. Thus, if maximum harmonics is raised, the pss waveform gets more timepoints which increases the accuracy of the pss waveform at the cost of longer runtimes and more memory required to run the pss simulation.

In order for the psp analysis to correctly measure conversion gain, the wave shape at the highest analysis frequency needs to be accurate, so with the default settings in shooting pss, up to 40 times the pss beat frequency can be used as the maximum frequency in psp. This will cover the vast majority of cases.

Harmonic Balance

When *Harmonic Balance* is selected in the pss analysis, only the harmonics specified in the pss *Choosing Analyses* form exist in the solution. If the harmonic balance analysis has enough harmonics to accurately capture the behavior of your circuit, the psp result will be accurate. To see if the result is accurate, increase the number of pss harmonics by about 50% and run again. If the psp result did not change, then you had enough harmonics to begin with, and might be able to be reduced. If it did change, add more harmonics and run again. For the fastest simulation, use the smallest number of harmonics that produce a stable psp result.

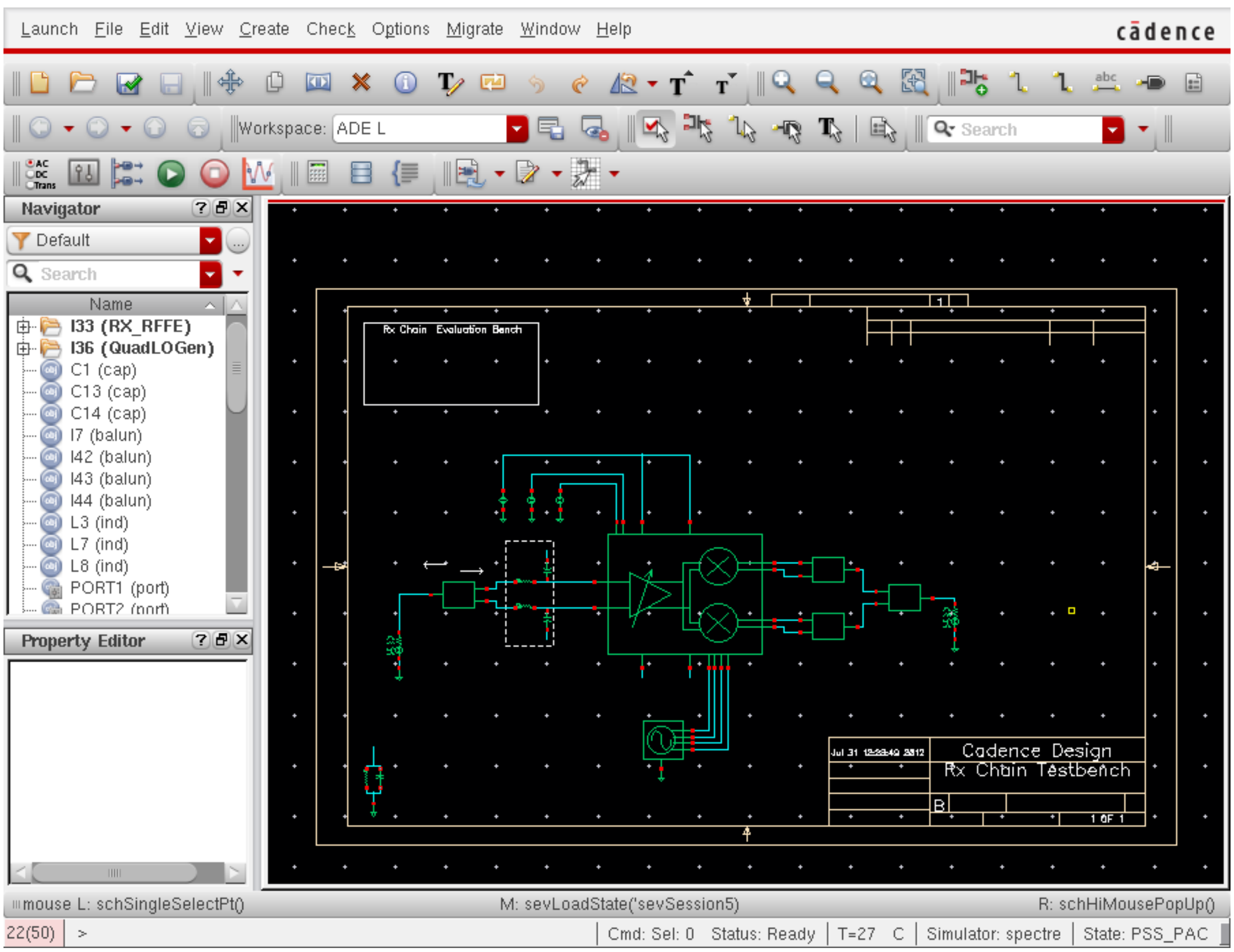
In the psp analysis, because only the harmonics specified in the pss-hb *Choosing Analyses* form exist, the maximum frequency term can only be as high as the number of harmonics specified in the pss analysis times the pss beat frequency.

ADE Implementation

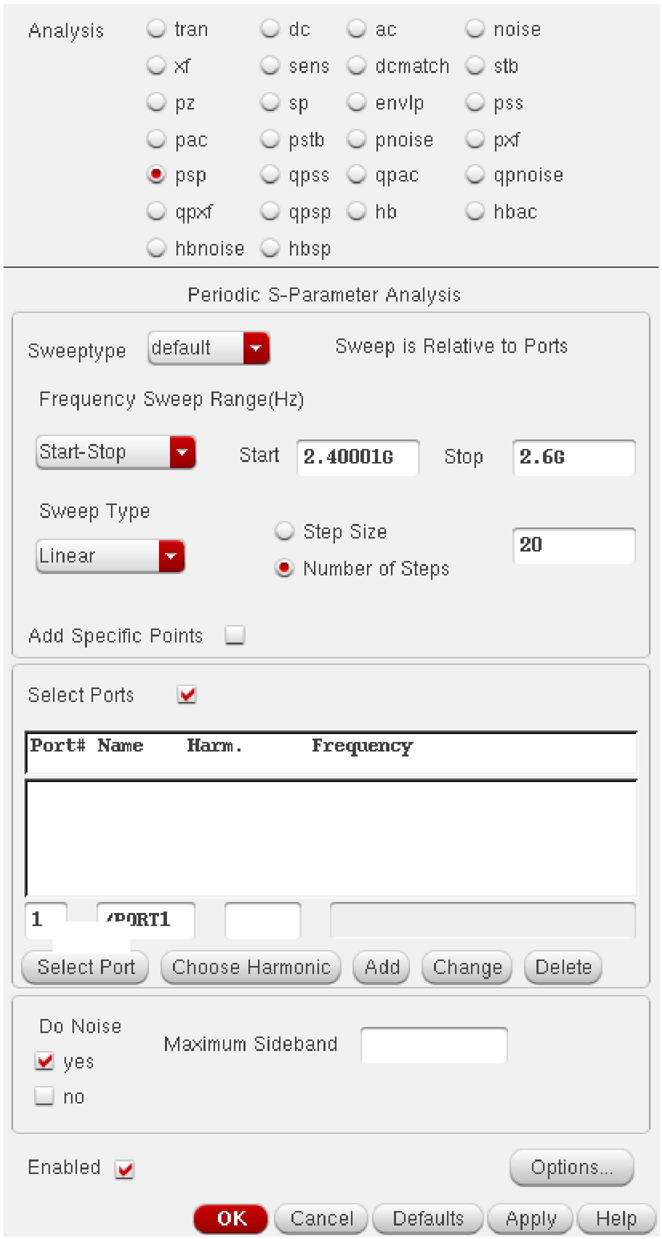
This section jumps from the theoretical to the practical with examples of how to use psp. The focus is only on the psp *Choosing Analyses* form so you can see where the individual settings are made. For examples with all the steps shown, see the examples section that follows this section.

PSP Setup

For this example, consider an LNA-Mixer circuit shown below.



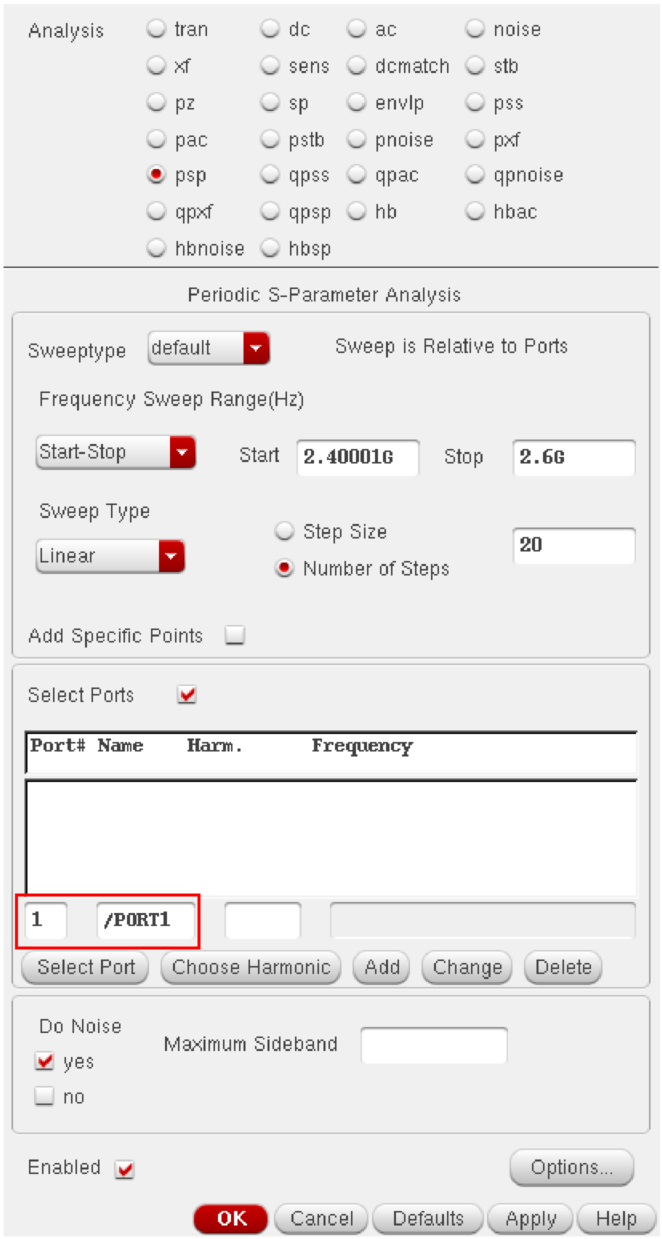
1. First, define either an input or output frequency range in the frequency sweep range fields. It is usually easier to define an input frequency range, as shown below for a 2.4GHz application.



2. The frequency sweep range specifies the limits and spacing for the input frequency.

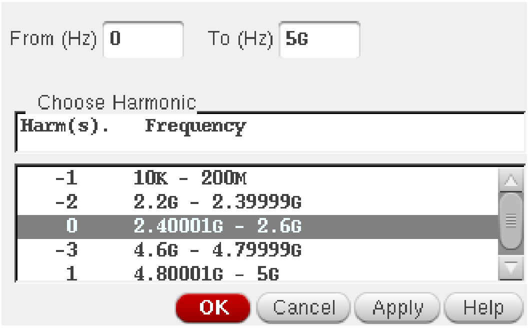
3. Linear and log sweeps are provided. It is usually better to select linear or log, and specify the number of steps instead of using automatic. Automatic will run 50 frequency points, which usually is not needed for most applications.

4. Now port and frequency definitions need to be made.



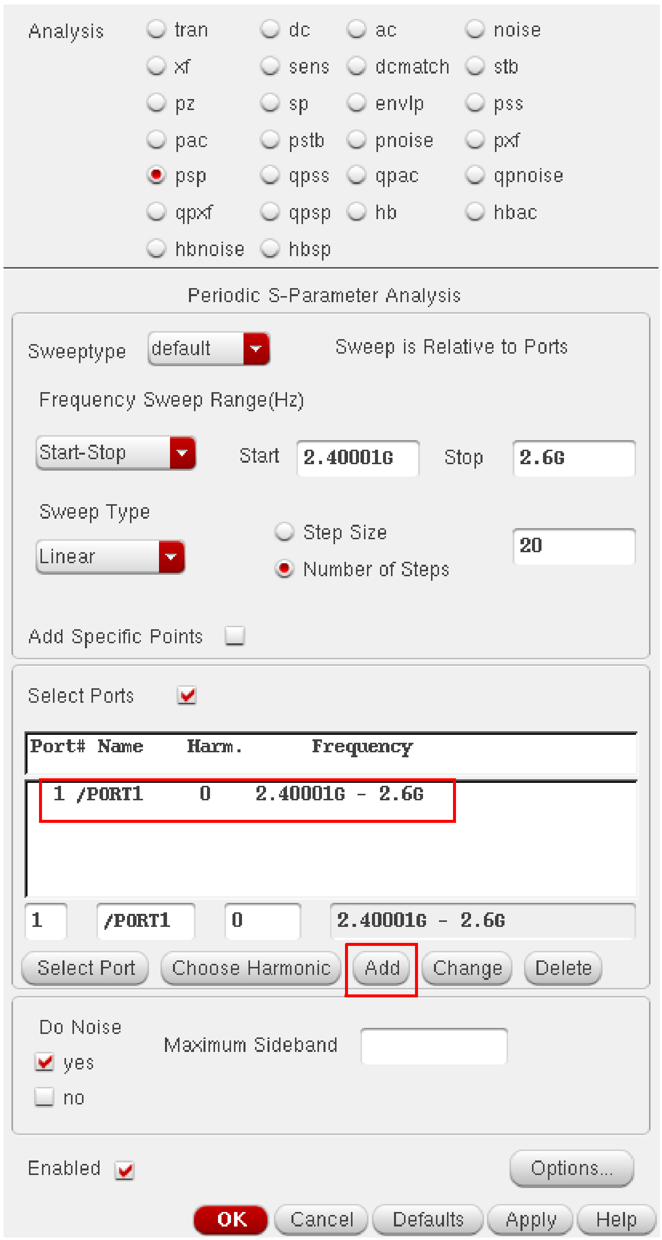
5. In the text field just above the *Select Port* button, type 1 in the *Port #* field. Click *Select Port*, and select the input port.

6. Click *Choose harmonic*. In the window that appears, select the input frequency range.

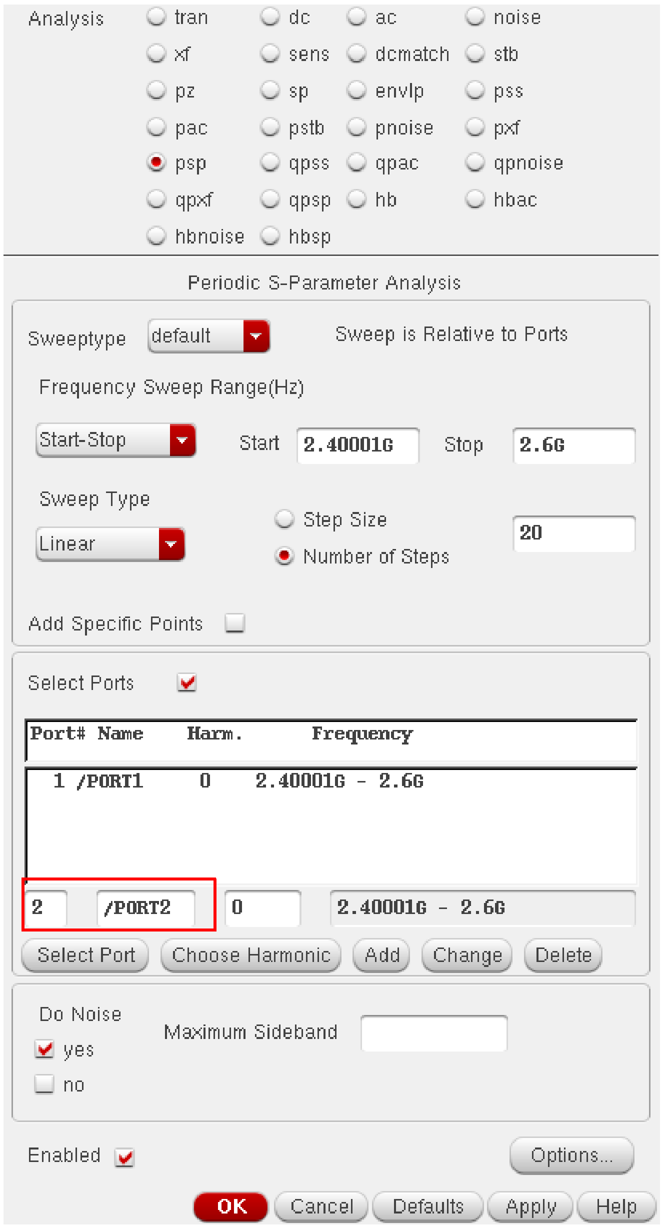


7. Click *OK*.

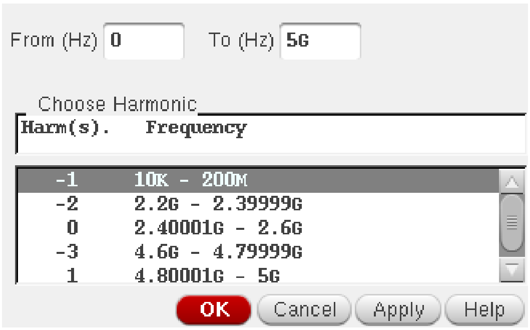
8. Click *Add*. The port-frequency combination is added to the list. By convention, Port 1 sets the input frequency for psp.



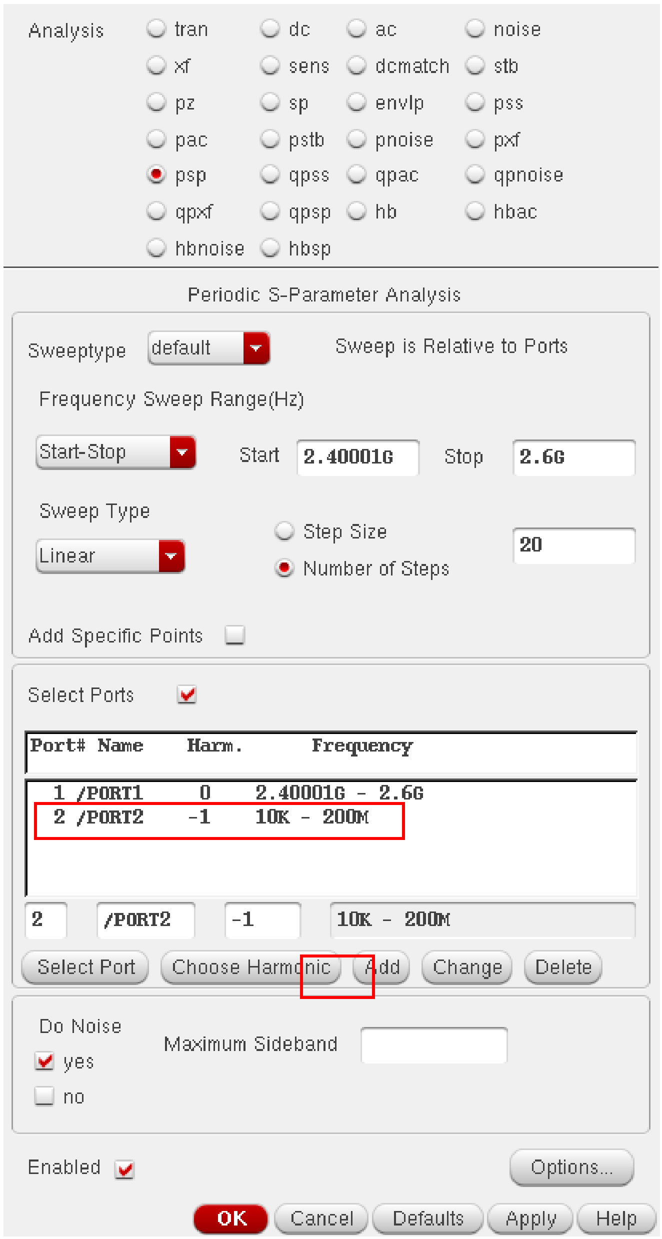
9. Now define the output port-frequency combination as Port 2.



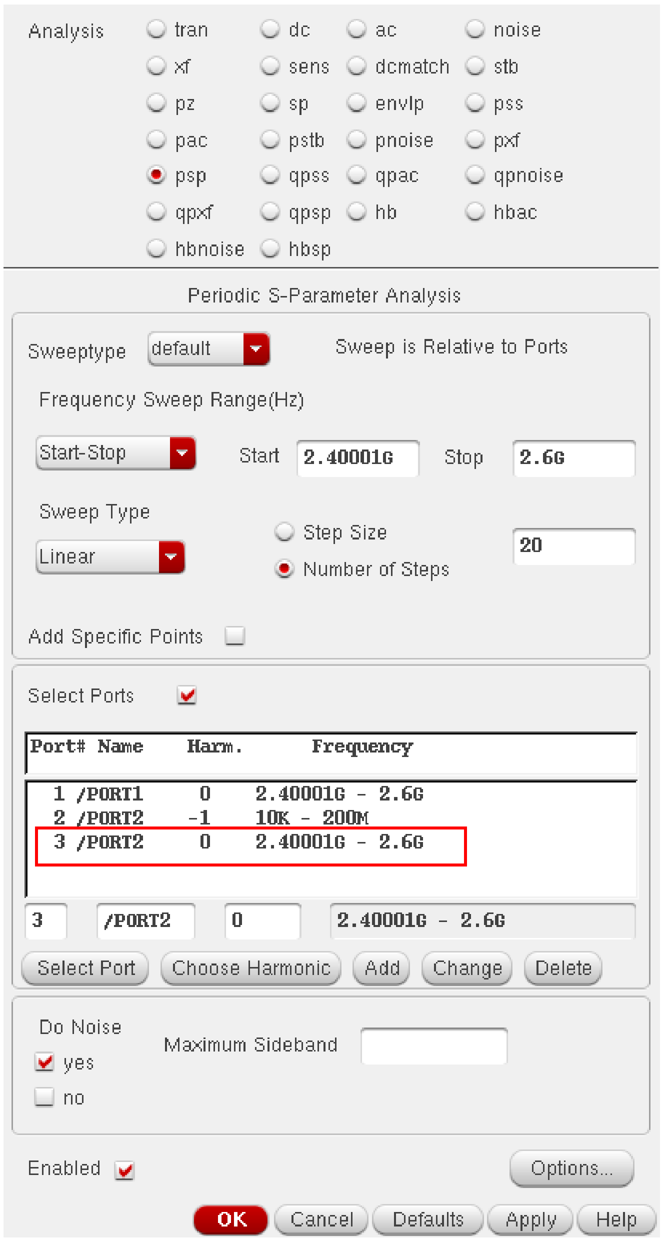
10. Note that the frequency was selected by clicking *Choose Harmonic*, and selecting the output frequency range as shown below.



11. Next, click *Add*. This adds the port/frequency combination as Port 2 in the list. By convention, Port 2 sets the output frequency.

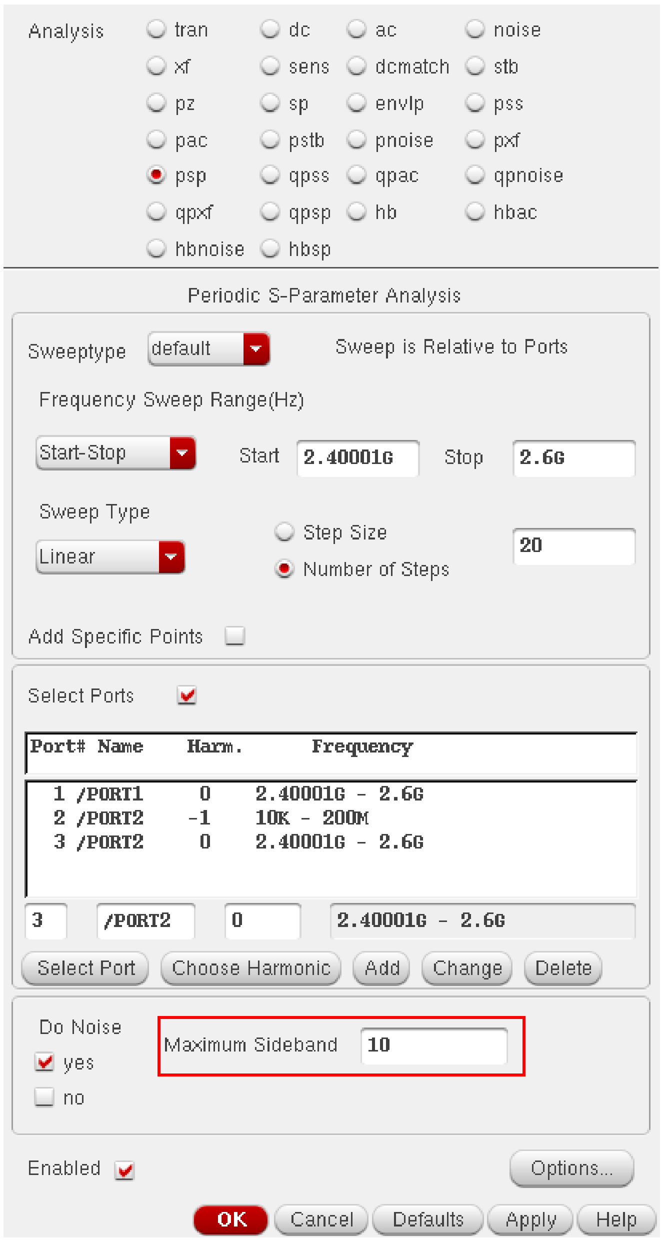


12. Repeat the process to define Port 3 as the gain from RF to IF with no frequency translation. (The RF to IF isolation.)



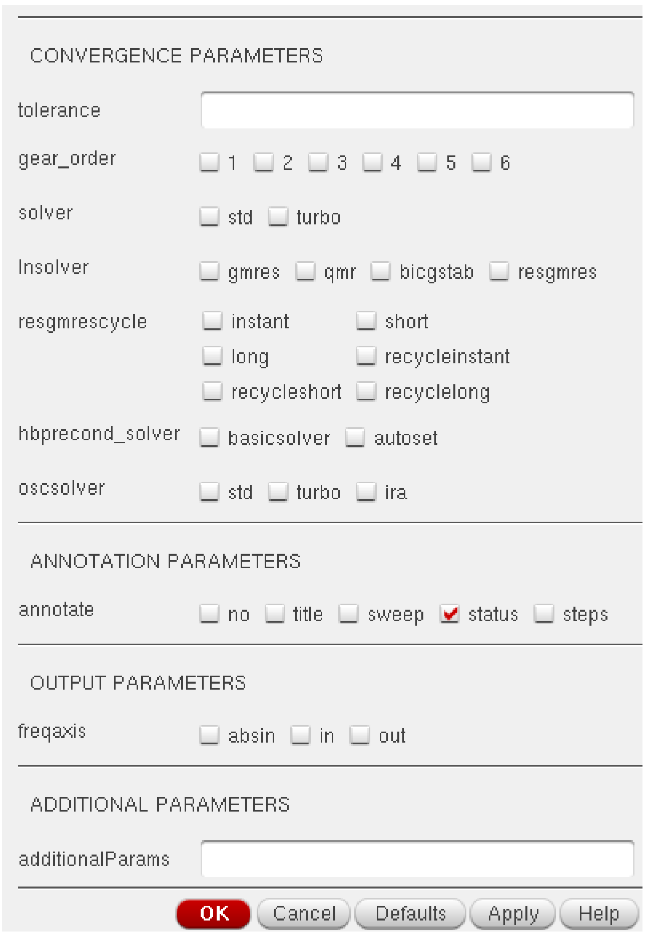
13. Click *Add* to add the last port-frequency definition to the list.

14. If you want a noise simulation, set *Maximum sideband* to the number of LO or clock harmonics. This analysis is the same as pnoise. For more information, see the pnoise section in this chapter.



PSP Options

Below is the psp options form.



None of the options are commonly used.

Solver

The solver option is used only for driven circuits. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the psp analysis frequency is very close to the frequency of one of the harmonics in the pss, warning messages will appear in the psp output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but which takes longer to run than the *turbo* solver.

Oscsolver

The *oscsolver* option is used only for oscillators. The default solver is the *turbo* solver.

When hb is used as the engine in pss, leave this option at the default.

When shooting is used, sometimes when the psp analysis frequency is very close to the frequency of one of the harmonics in the pss, warning messages appear in the psp output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the pss, but takes longer to run than the *turbo* solver.

For very large oscillator designs, the ira oscsolver may take less memory and run faster with no loss in accuracy, however, it is less robust for convergence compared to *turbo* or *std*.

Hbprecond_solver

This option is only available only when harmonic balance is selected in PSS and APS is used.

The basic solver is the only solver available in standard Spectre when harmonic balance is selected. *autoset* is the default solver in harmonic balance when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window. If you have stagnation, it will save a small amount of time to set this option to *basicsolver*.

Tolerance

Leave this option at the default value.

Psp uses an iterative solver to calculate the output amplitudes. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough. The tolerance option specifies that accuracy for psp. For shooting, the default tolerance is 1e-9. For driven circuits where HB is the pss engine, the default is 1e-6. For oscillators where hb is selected for the pss engine, the default is 1e-4.

Gear_order

Do not change this option.

Lnsolver

Leave this option at the default.

Each psp frequency point solution is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases with increasing iterations. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

Resgmrescycle

Leave this option at the default. For the *resgmres* linear solver, there are several different options.

Annotate

This option controls the level of detail in the Spectre output log. Selections to the left in the form have less detail in the Spectre output log. Selections to the right in the form have increasing levels of detail in the output log.

Freqaxis

Psp displays all the results on the same frequency scale. The default is *absin* (absolute value of the input frequency) If there is a negative input frequency, *in* will show that frequency on the negative frequency axis. *out* shows the output frequency range on the X axis.

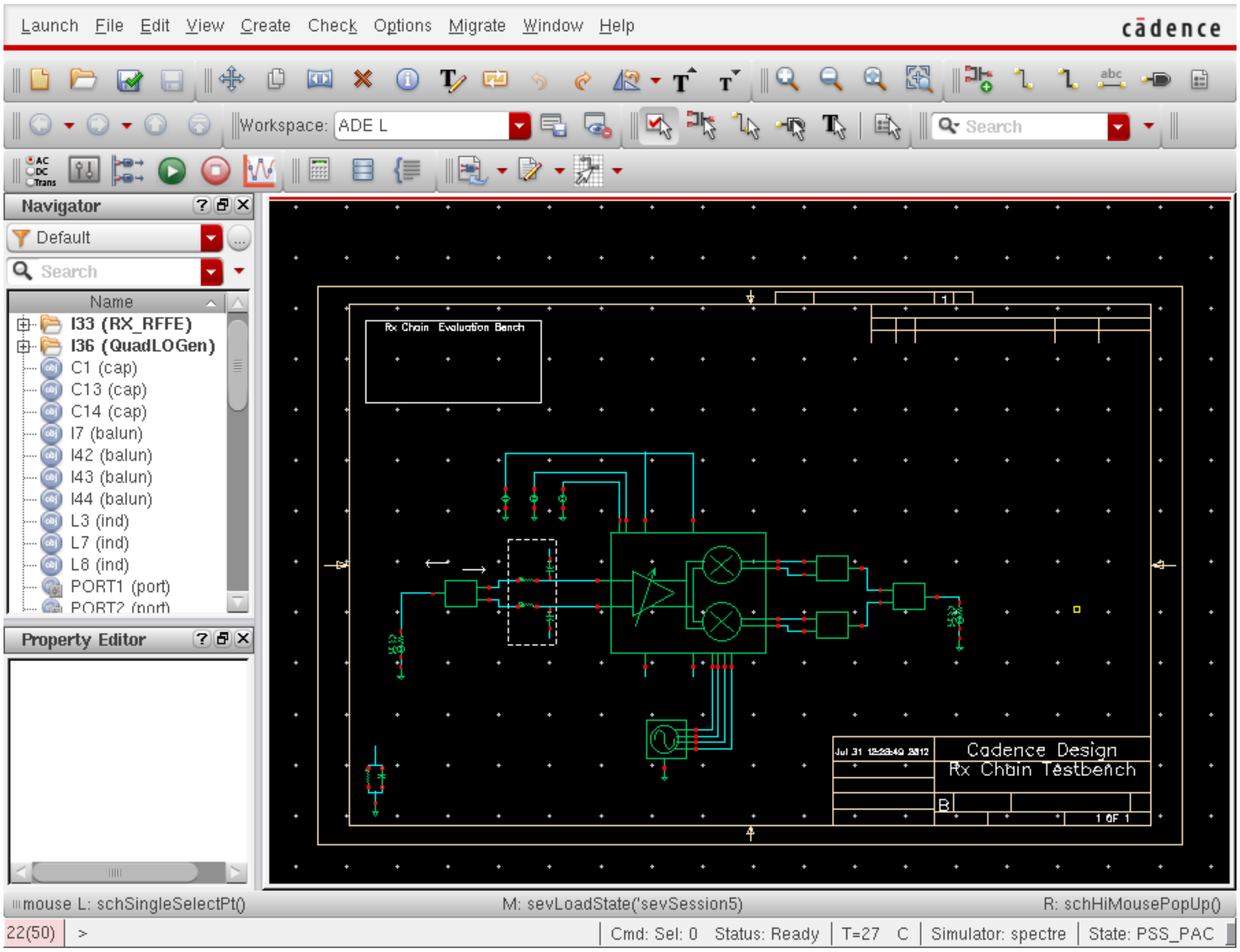
AdditionalParams

additionalParams is typically used for new features that are being beta tested. Entries in this field are `keyword=value` pairs.

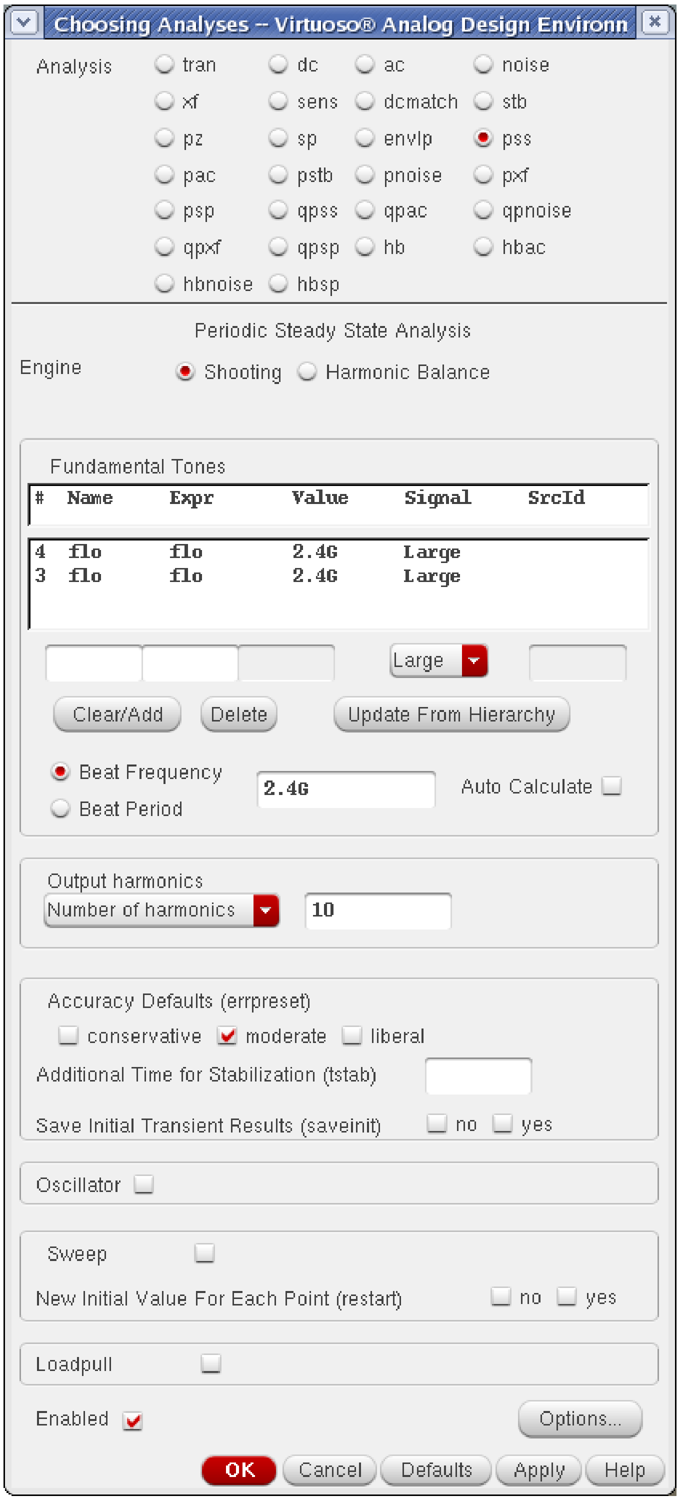
For more information about the other options, type `spectre -h psp` at the command prompt in a Unix shell window.

Examples

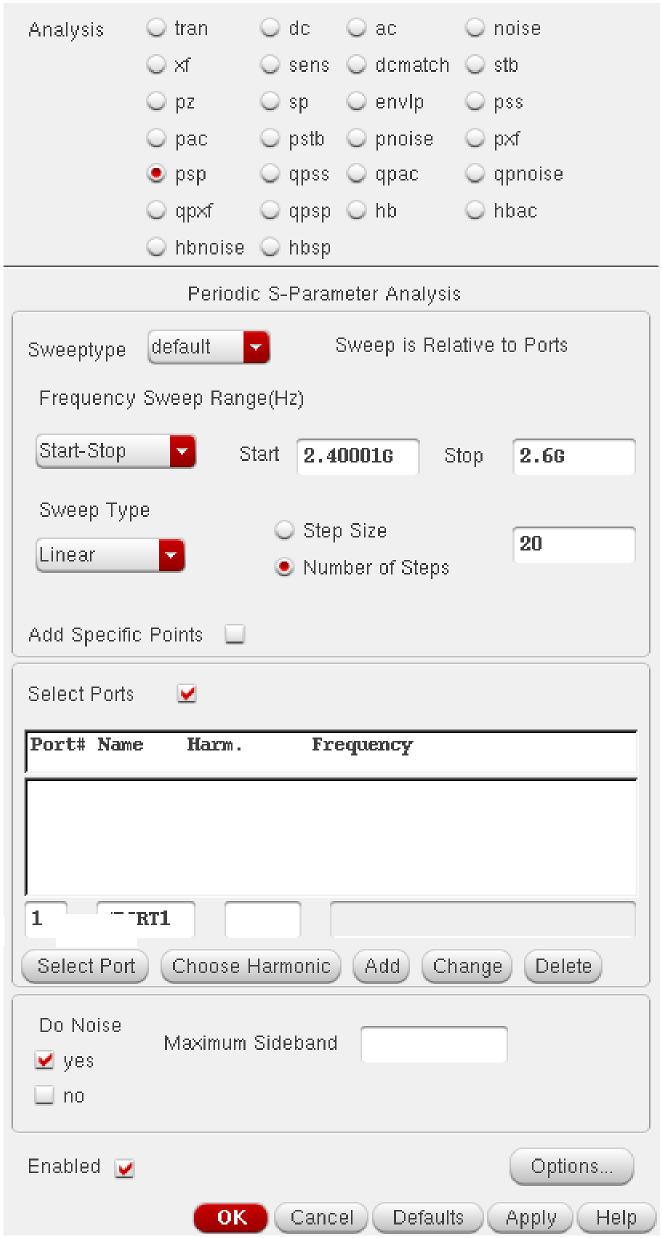
For this example, consider an LNA-Mixer circuit shown below.



1. Set up the pss analysis. For more detail, see the pss section at the beginning of this chapter.



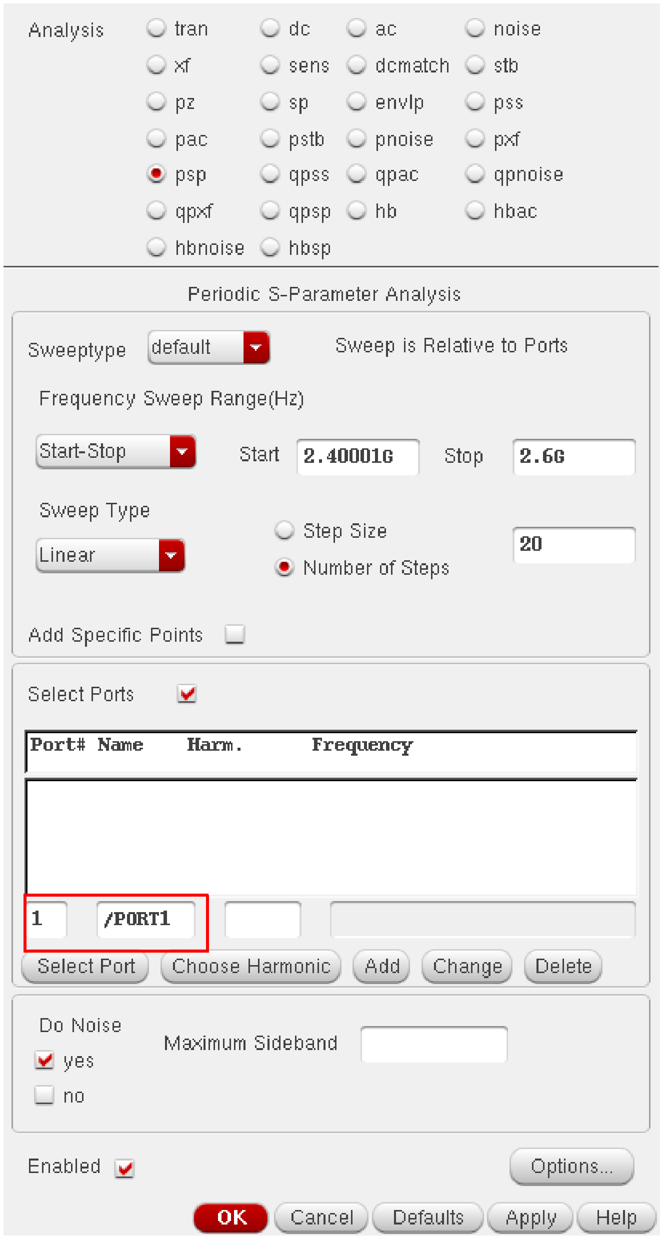
2. In the psp *Choosing Analyses* form, define either an input or output frequency range in the frequency sweep range fields. It is usually easier to define an input frequency range, as shown below for a 2.4GHz application.



3. It is usually easier to define the frequency sweep range to be the input frequency range.

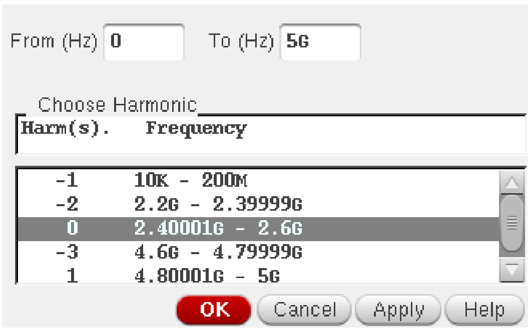
4. Linear and log sweeps are provided. It is usually better to select linear or log, and specify the number of steps instead of using automatic. Automatic will run 50 frequency points, which usually is not needed for most applications.

5. Now port and frequency definitions need to be made.

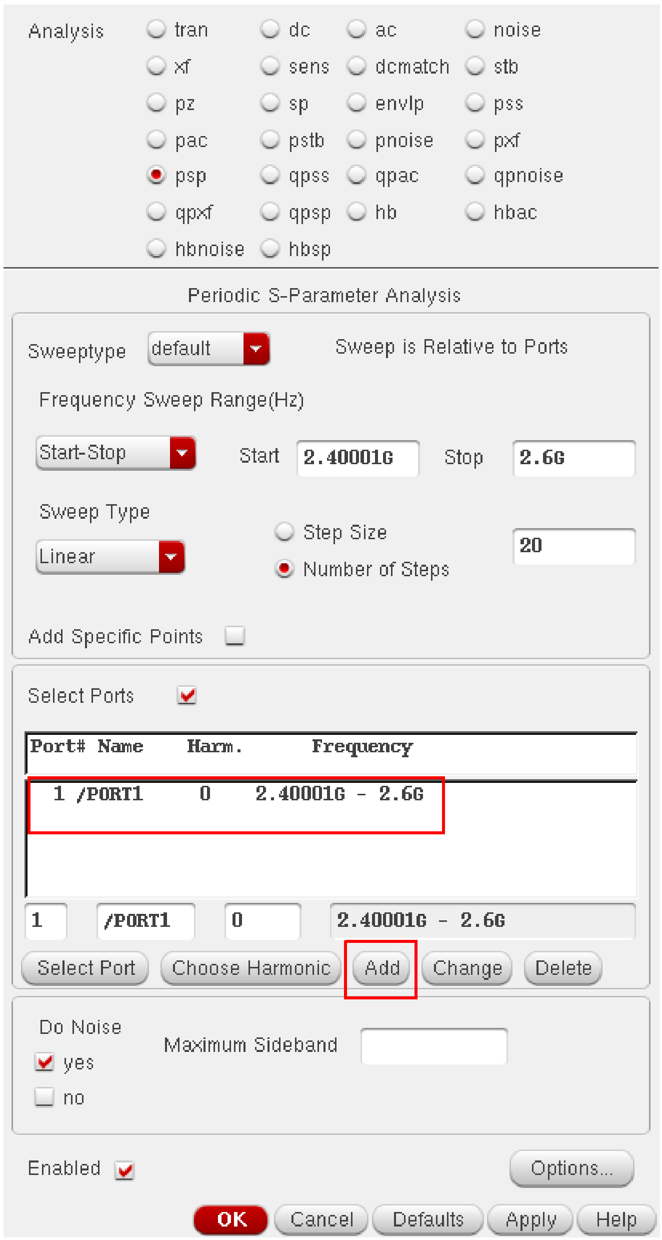


6. In the field just above the `Select Port` button, type 1 in the *Port #* field. Click *Select Port*, and select the input port.

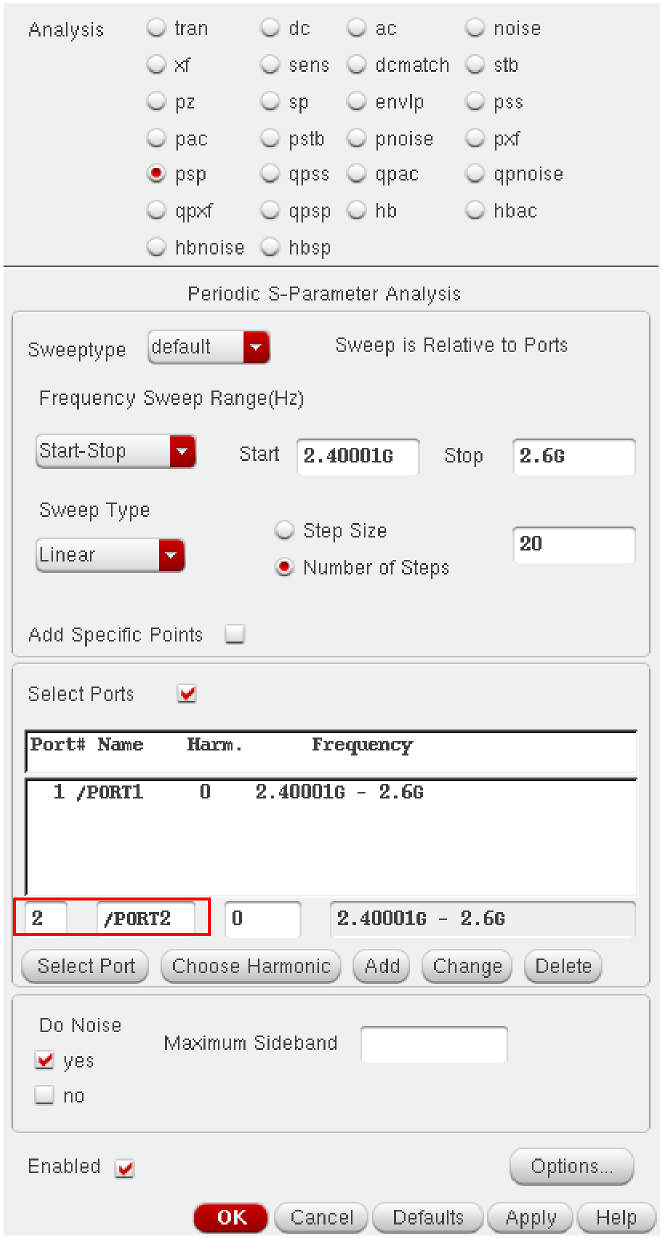
7. Click *Choose harmonic*. In the window that appears, select the input frequency range. and click OK.



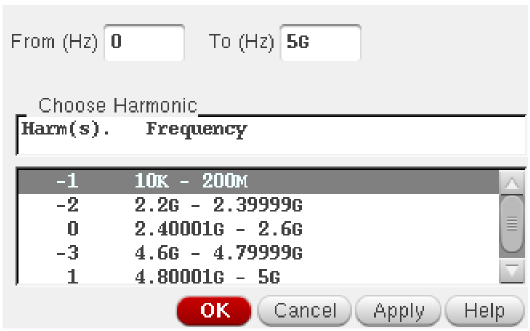
8. Click *Add*. The port-frequency combination is added to the list. By convention, Port 1 defines the input frequency range for the psp analysis.



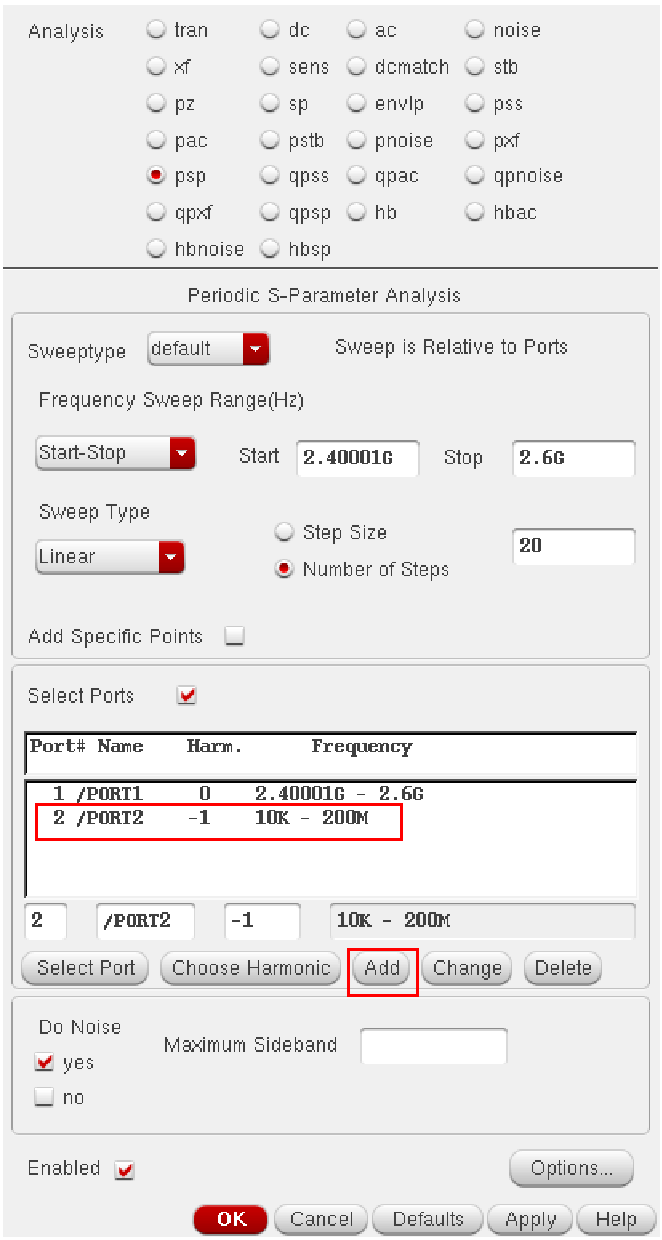
9. Now define the output port-frequency combination as Port 2.



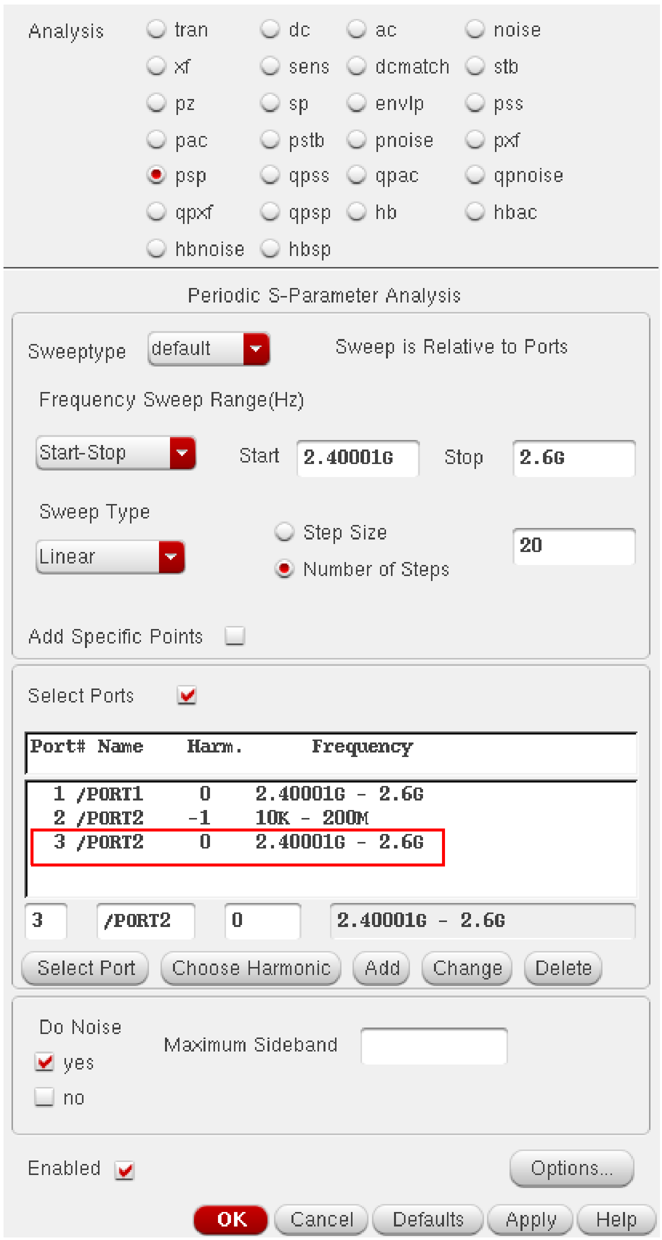
10. Note that the frequency was selected by clicking *Choose Harmonic*, and selecting the output frequency range as shown below.



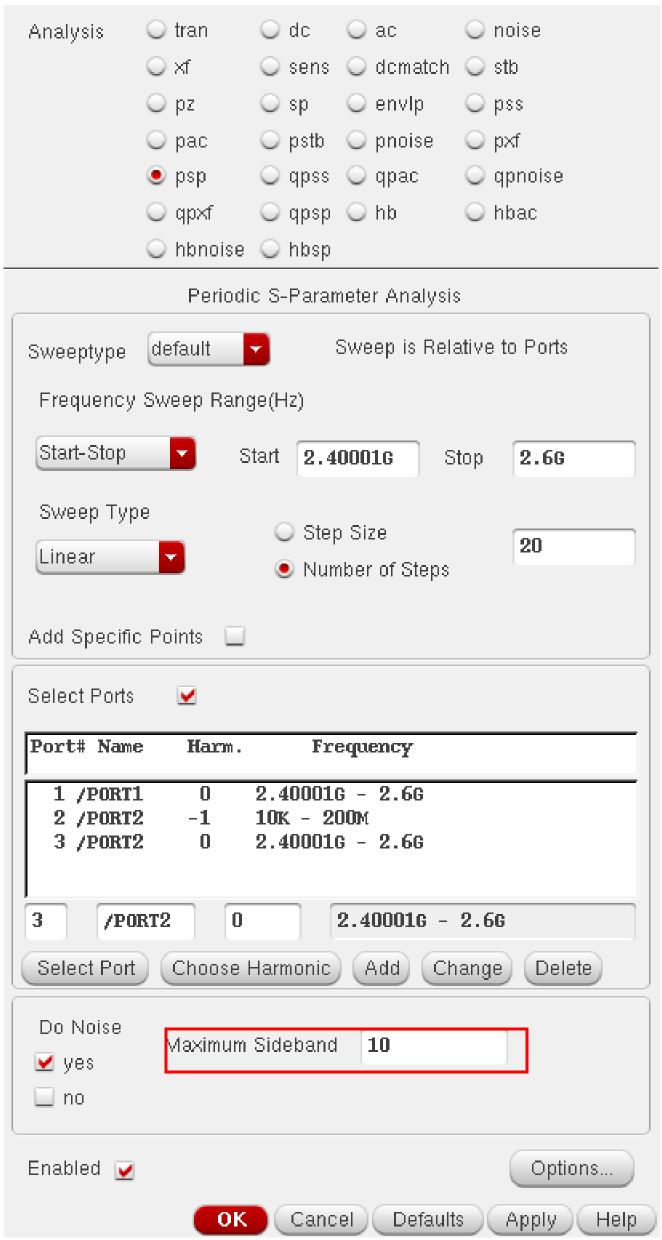
11. Next, click *Add*. This adds the port-frequency combination as Port 2 in the list. By convention, Port 2 defines the output frequency range for the psp analysis



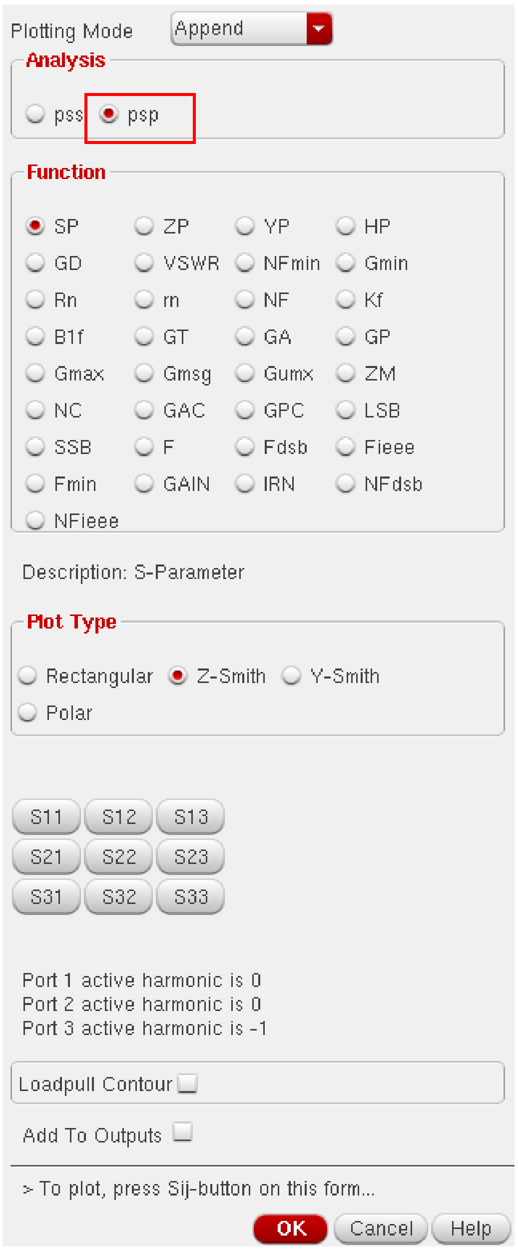
12. Repeat the process to define Port 3 as the gain from RF to IF with no frequency translation. (The RF to IF isolation).



13. If you want a noise simulation, set *Maximum sideband* to the same value that was used for the LO or clock harmonics in the pss analysis. This is the same analysis as pnoise. For more information about pnoise, see the pnoise section in this chapter.

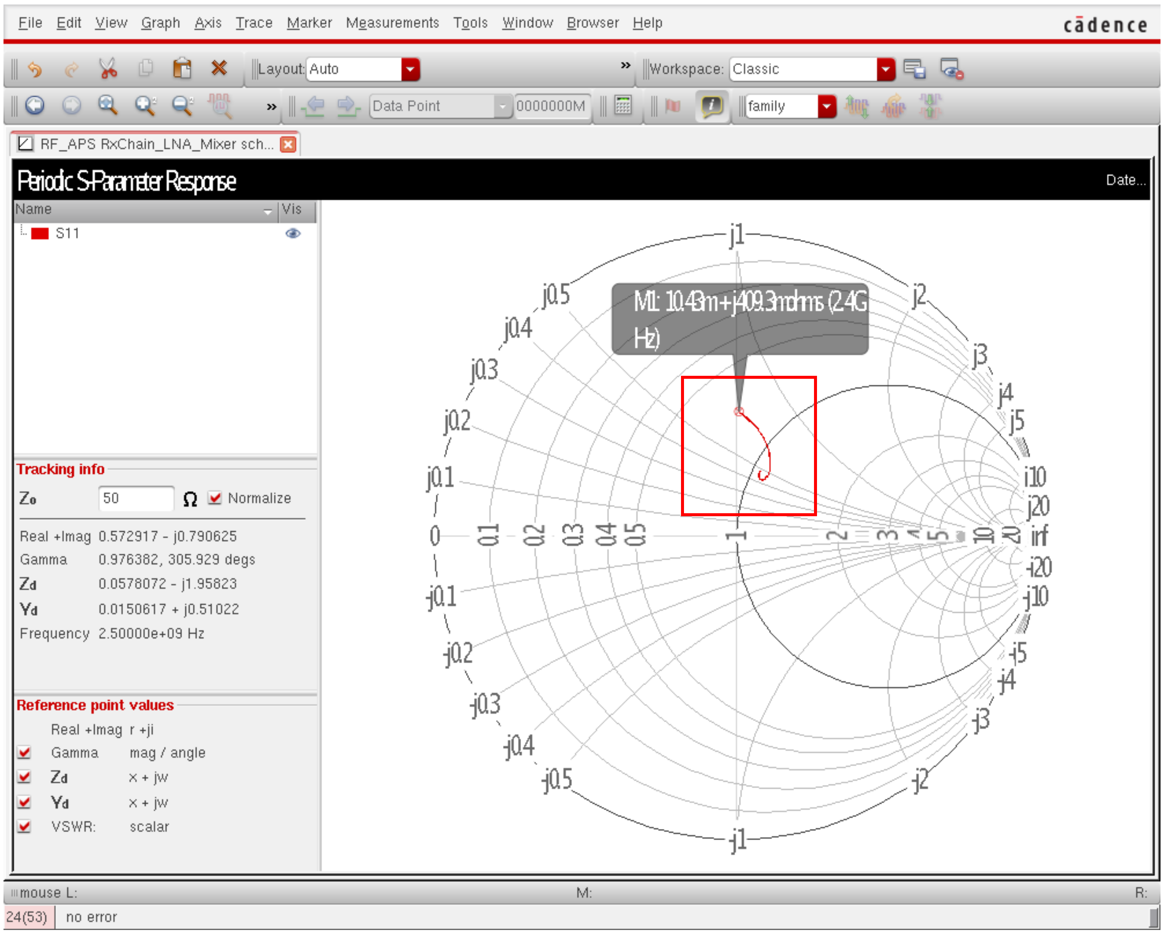


14. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window.



15. Select *psp* from the *Analysis* section.

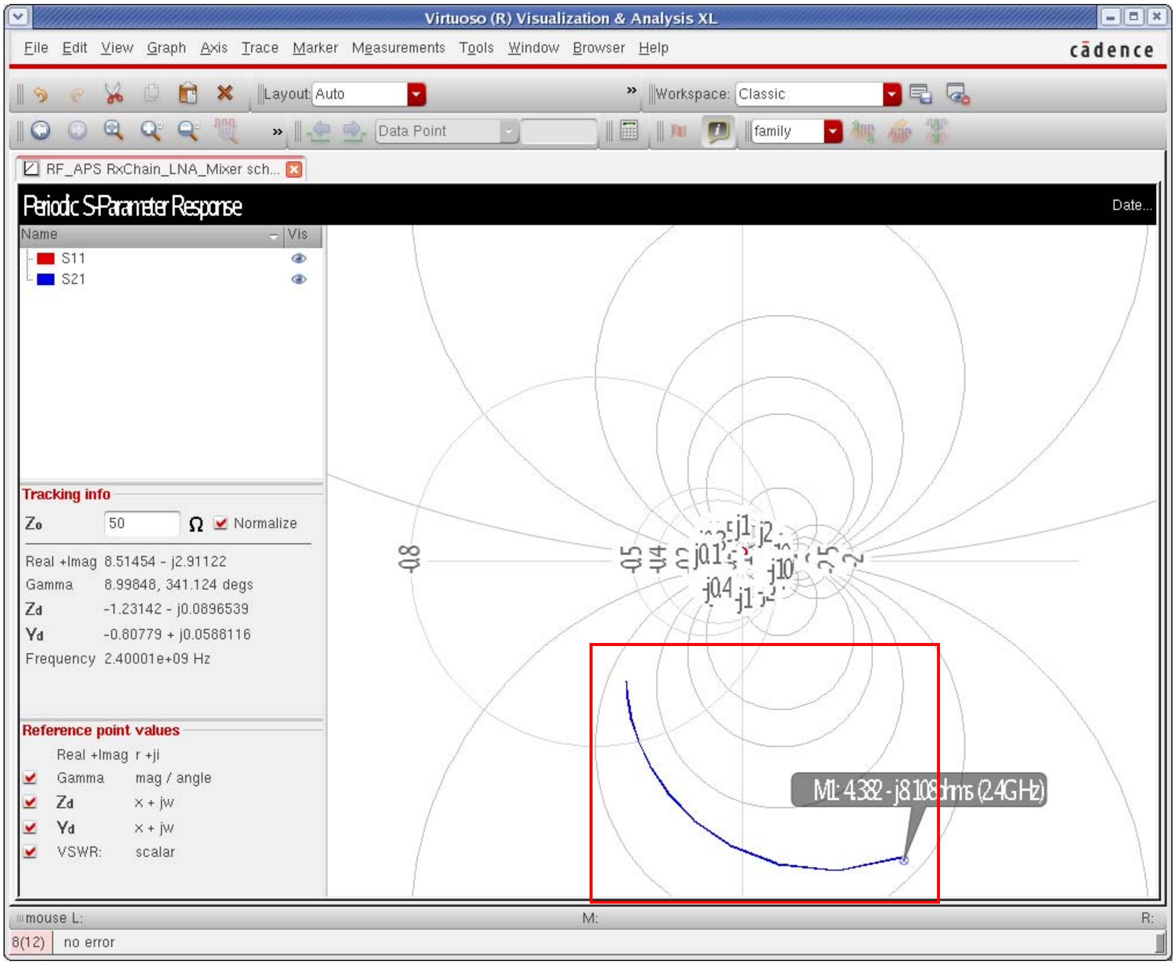
16. To plot S-Parameters, select *SP* from the *Function* section. In this case, Port 1 is the input port at the RF frequency, Port 2 is the output port at the IF frequency, and Port 3 is the output port at the RF frequency. In the *Plot Type* section, several choices are available. The input match (S11) can be plotted on the Smith Chart. Now, click the *S11* button. S11 appears on the Smith Chart.



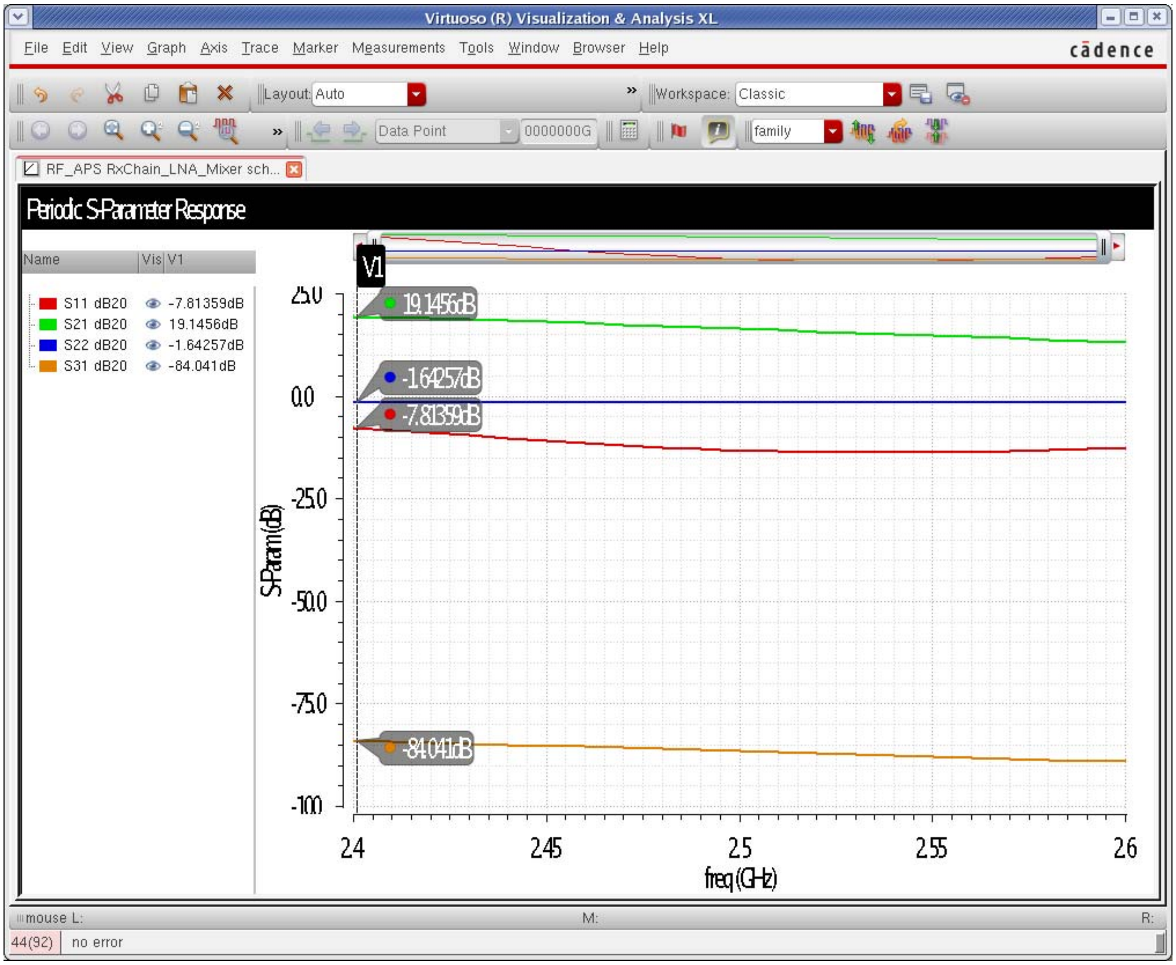
The length of the vector from the center of the circle to the curve is the magnitude of the reflection coefficient. In the above plot, the input match becomes better at high frequency because the vector is shorter. Most LNAs are deliberately mismatched because a lower noise figure can typically be obtained with a small mismatch.

17. There are two useful zoom functions for the Smith Chart. typing the f key fits the actual data to the Smith Chart. <Shift>-F fits the unit circle, as shown above.

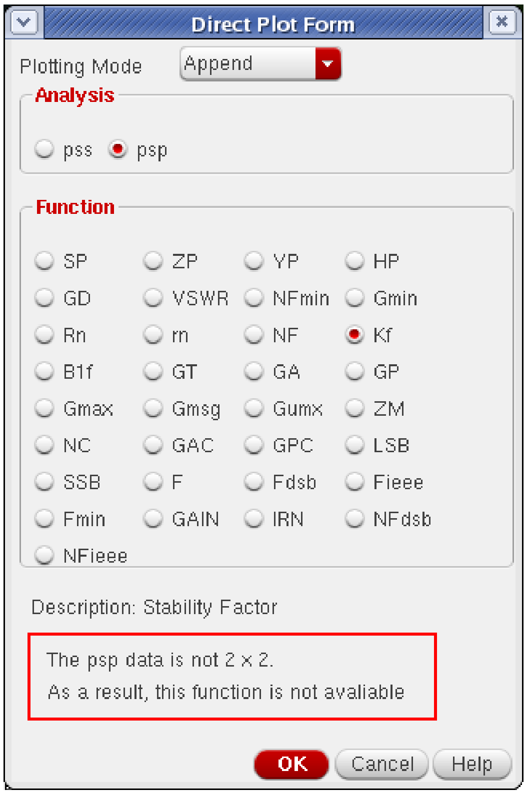
18. Now plot S21 on the Smith Chart. It displays outside of the unit circle.



19. Now plot S11, S21, S22, and S31 on a rectangular plot. The conversion gain is about 19.1dB. The input match is questionable. The output match is not good, but it does not matter because the output is at low frequency. The RF to IF isolation is about 84dB.

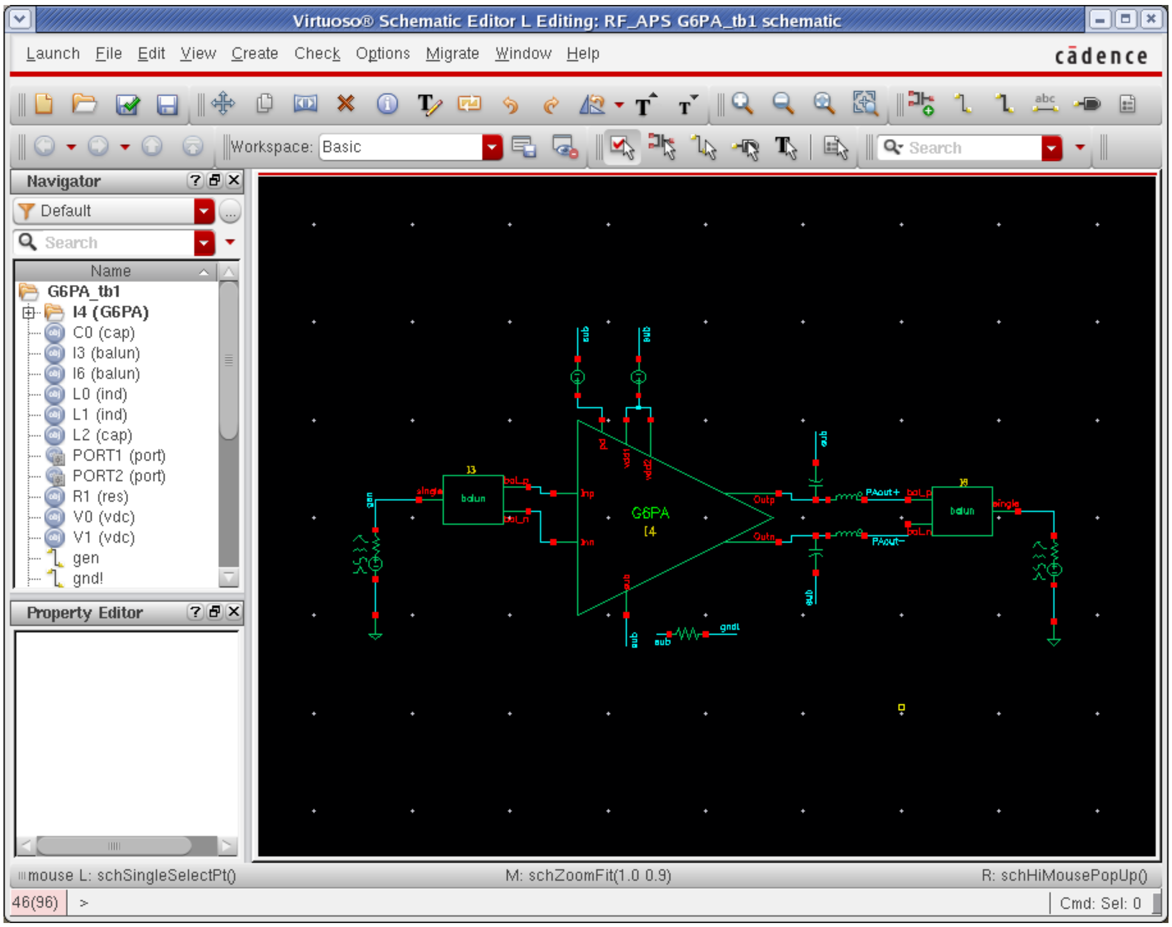


20. When you select Kf, because this is a 3-port analysis, the *Direct Plot Form* shows that Kf can not be plotted. Kf is defined for a 2-port simulation only. If you removed the RF to IF S-parameter measurement in this example, and converted to a 2-port simulation, you still would not be able to plot the data. Kf and B1f are defined only for amplifiers where the input and output frequency are the same.

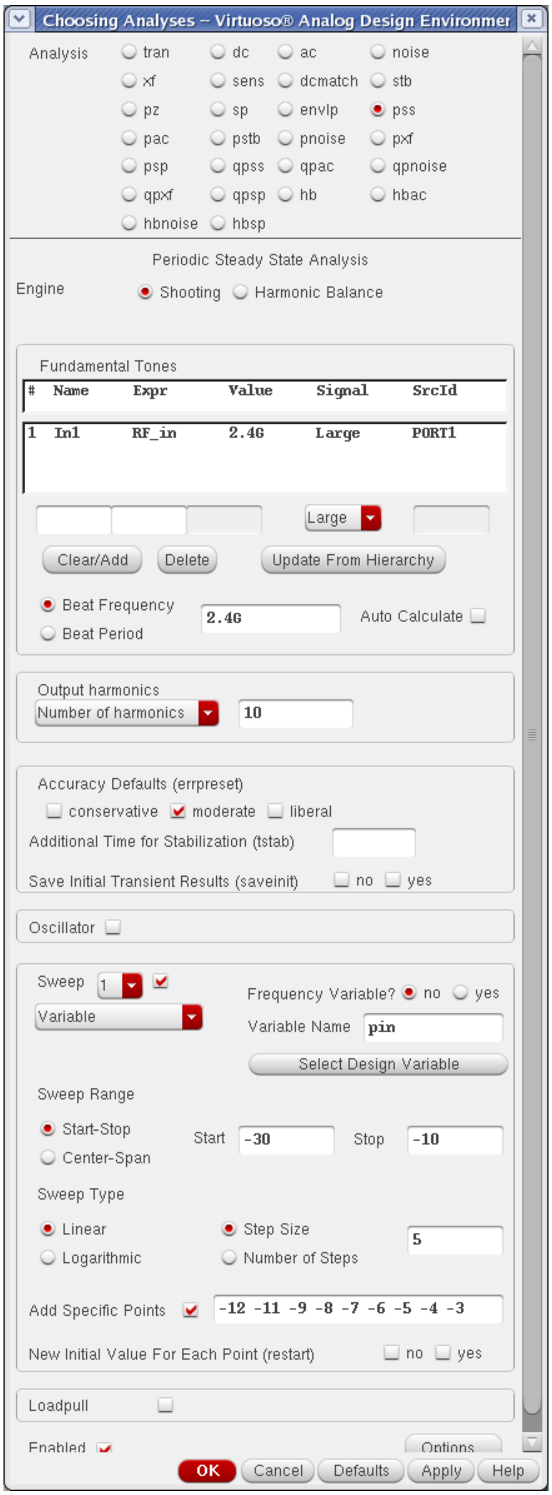


Swept Input Power In a PA

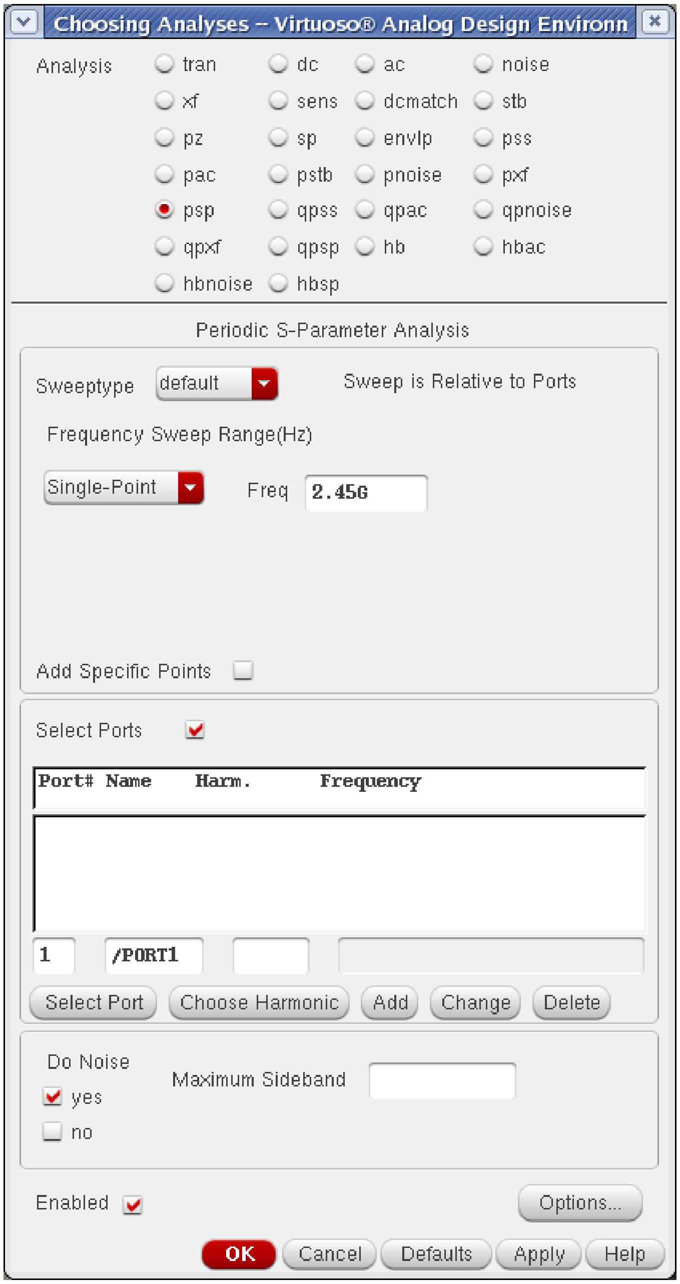
For the next example, consider a power amplifier.



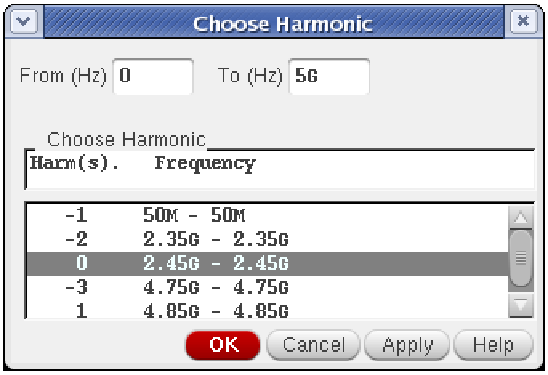
1. First, set up the pss analysis. In this case, the input power is swept. For more information, see the pss section at the beginning of this chapter.



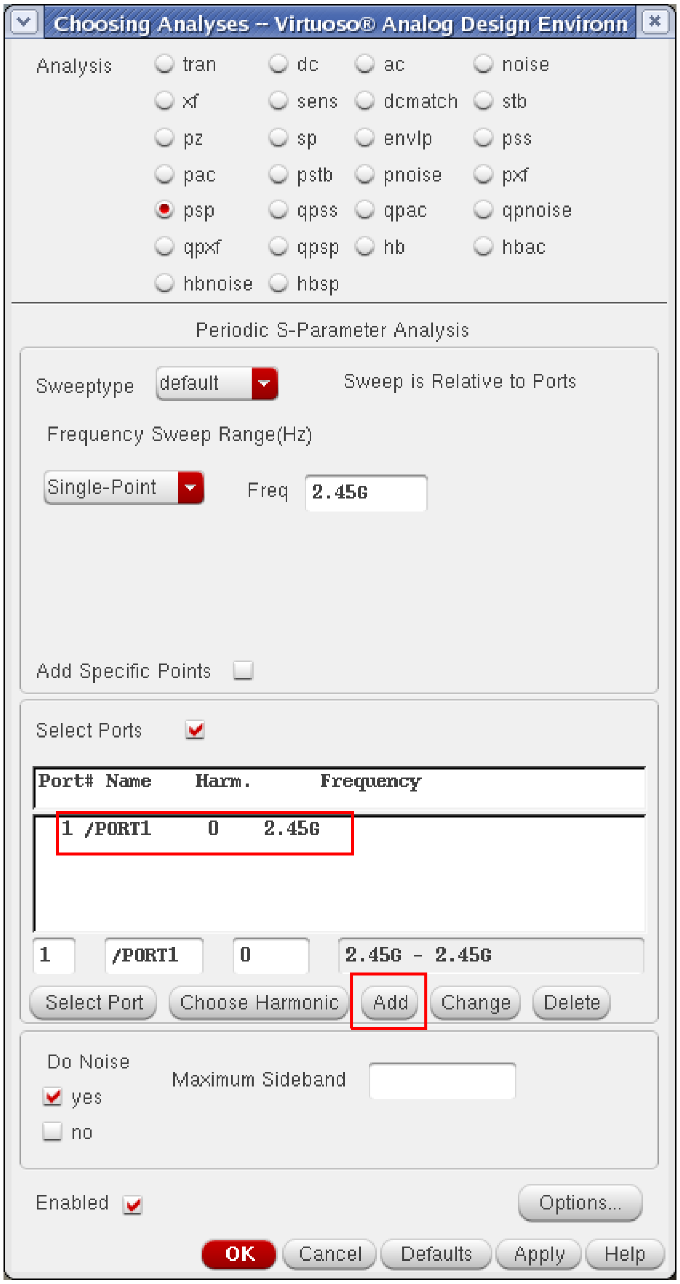
2. Now define the input frequency and input port for psp. When a sweep is run in pss, the psp frequency sweep will switch to single point automatically.



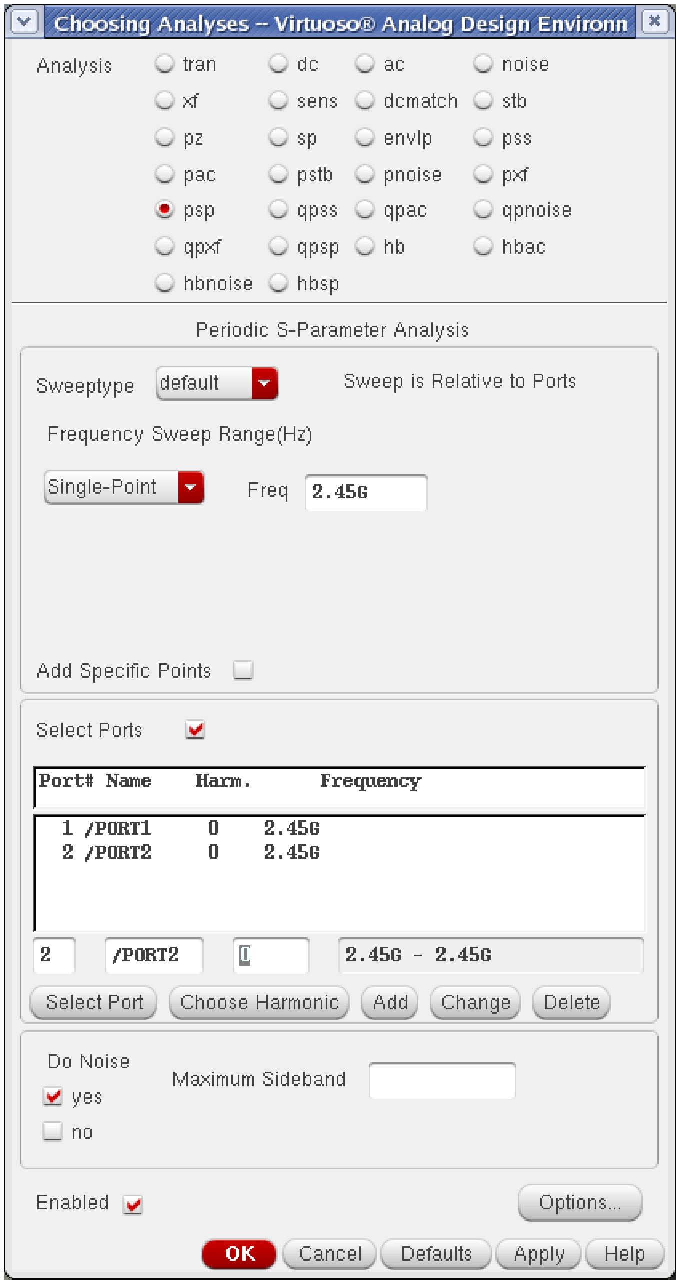
3. Click *Choose Harmonic*. In the window that appears, select the input frequency and click *OK*.



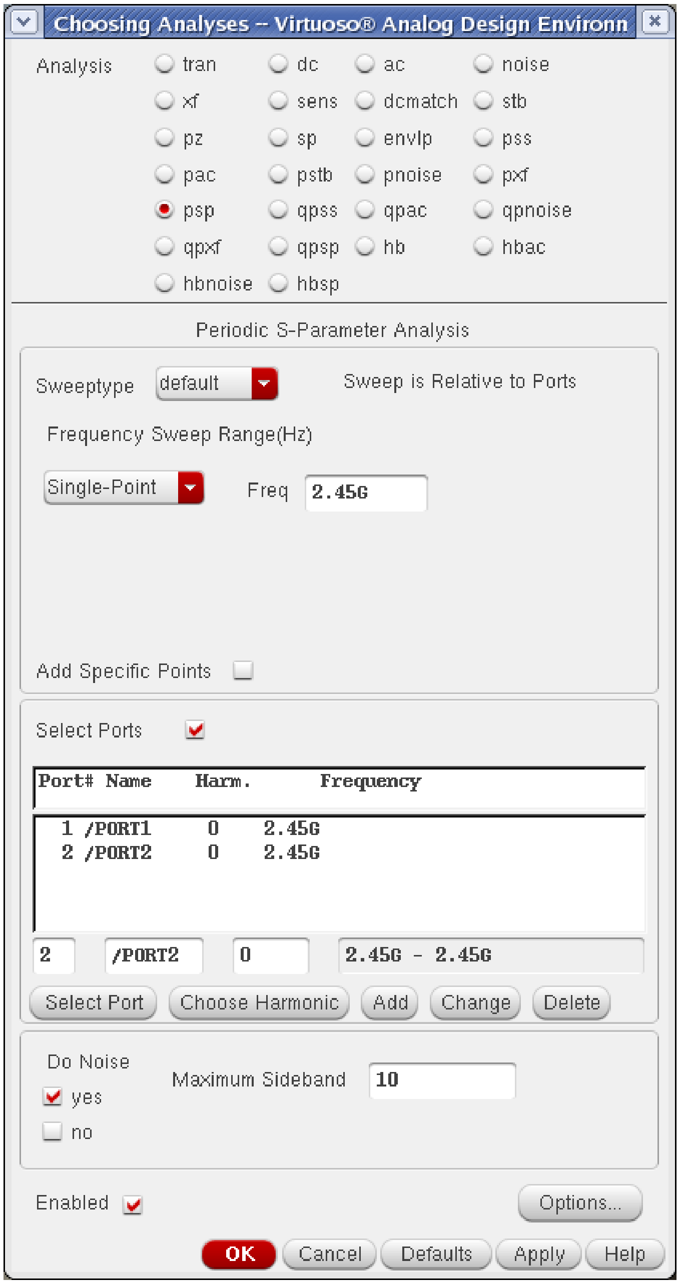
4. Click *Add*. The port/frequency combination is entered in the list.



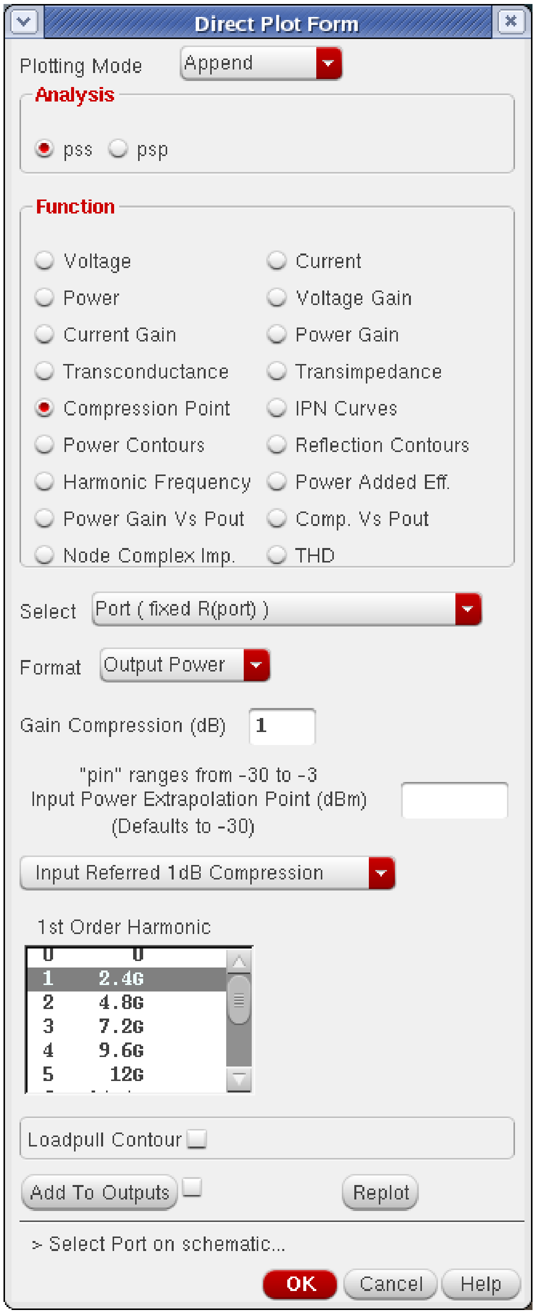
5. Repeat for the output port/frequency combination.



6. If you want a noise analysis, set *Maximum sideband* to the same value used for pss harmonics. This analysis is the same as pnoise. Please refer to the pnoise section earlier in this chapter for details.



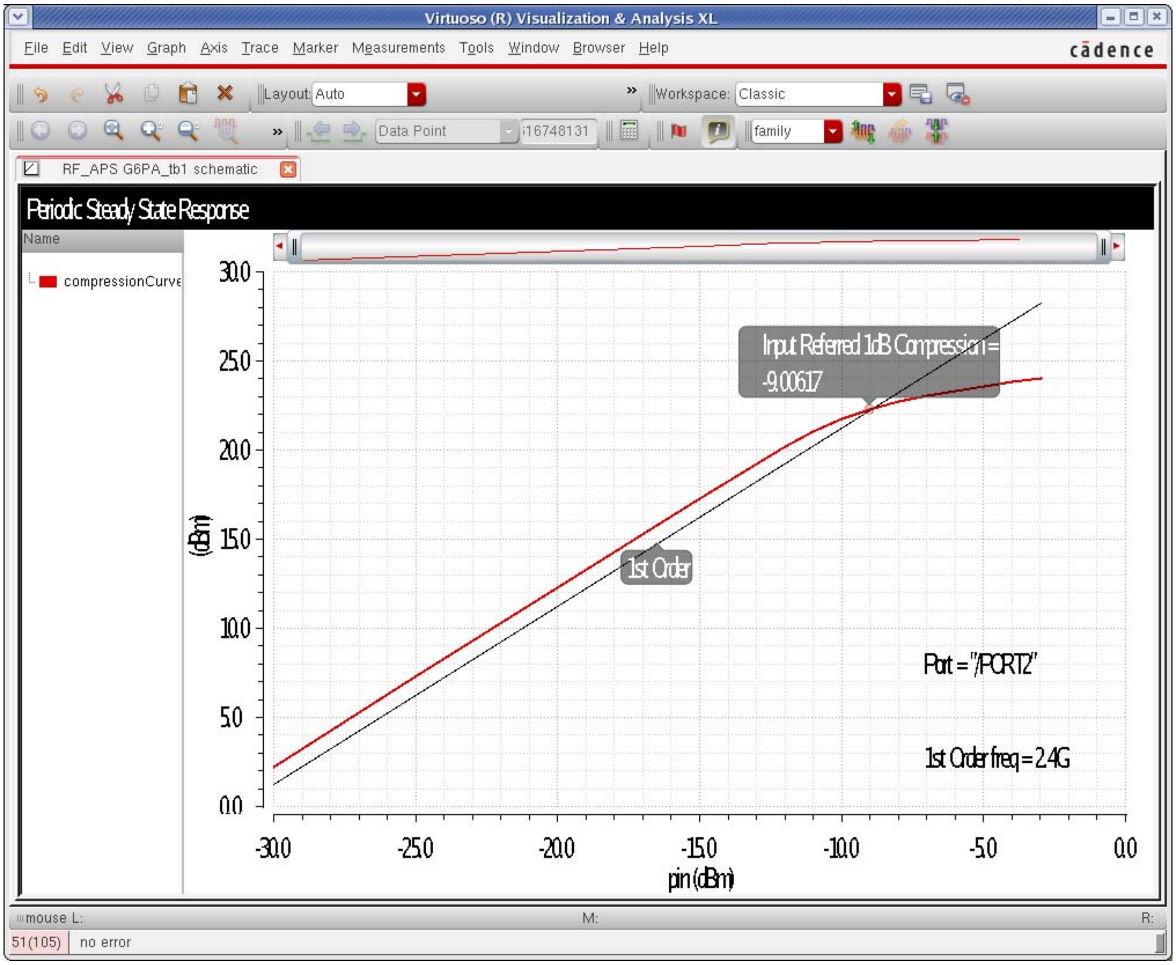
7. Click *OK*, and run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in ADE. The *Direct Plot Form* is displayed.



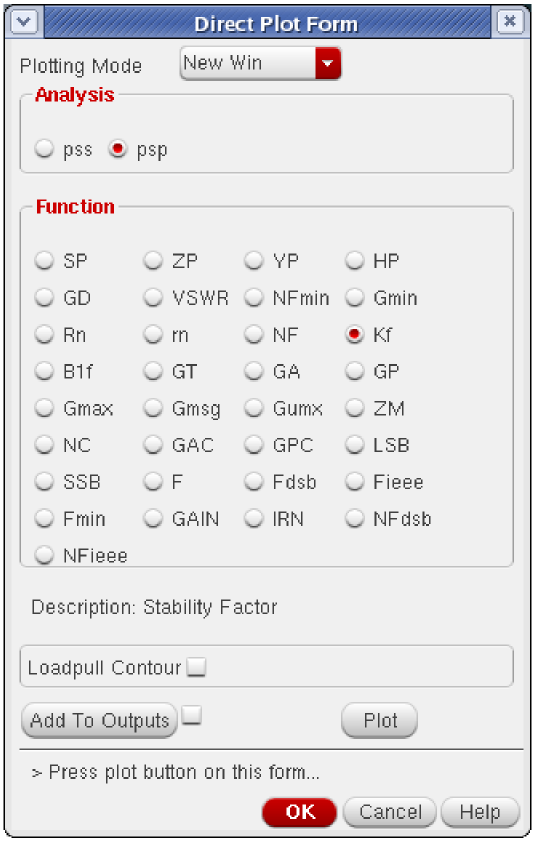
8. Select *pss* from the *Analysis* section.

9. Plot the compression curve for the amplifier.

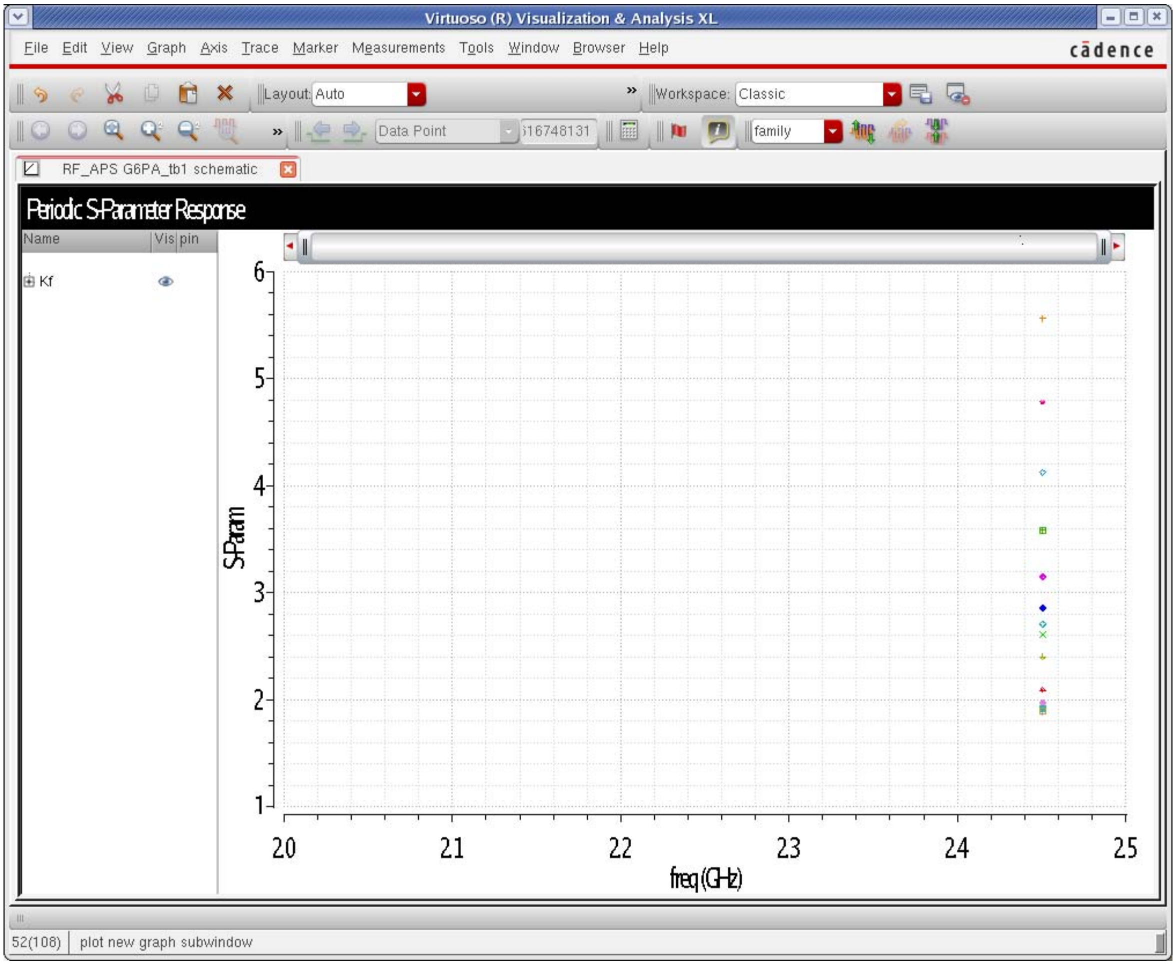
10. The compression point is shown below.



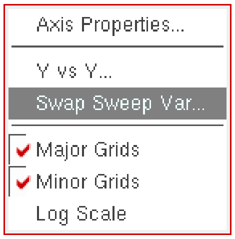
11. Now plot stability factor from the psp results (Kf).



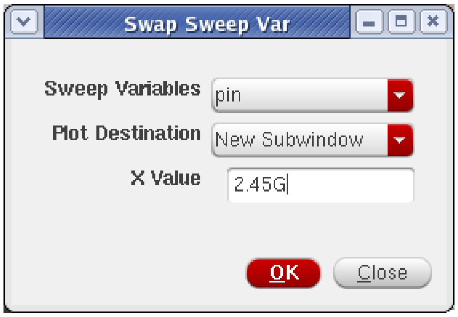
12. The plot displayed, as shown below.



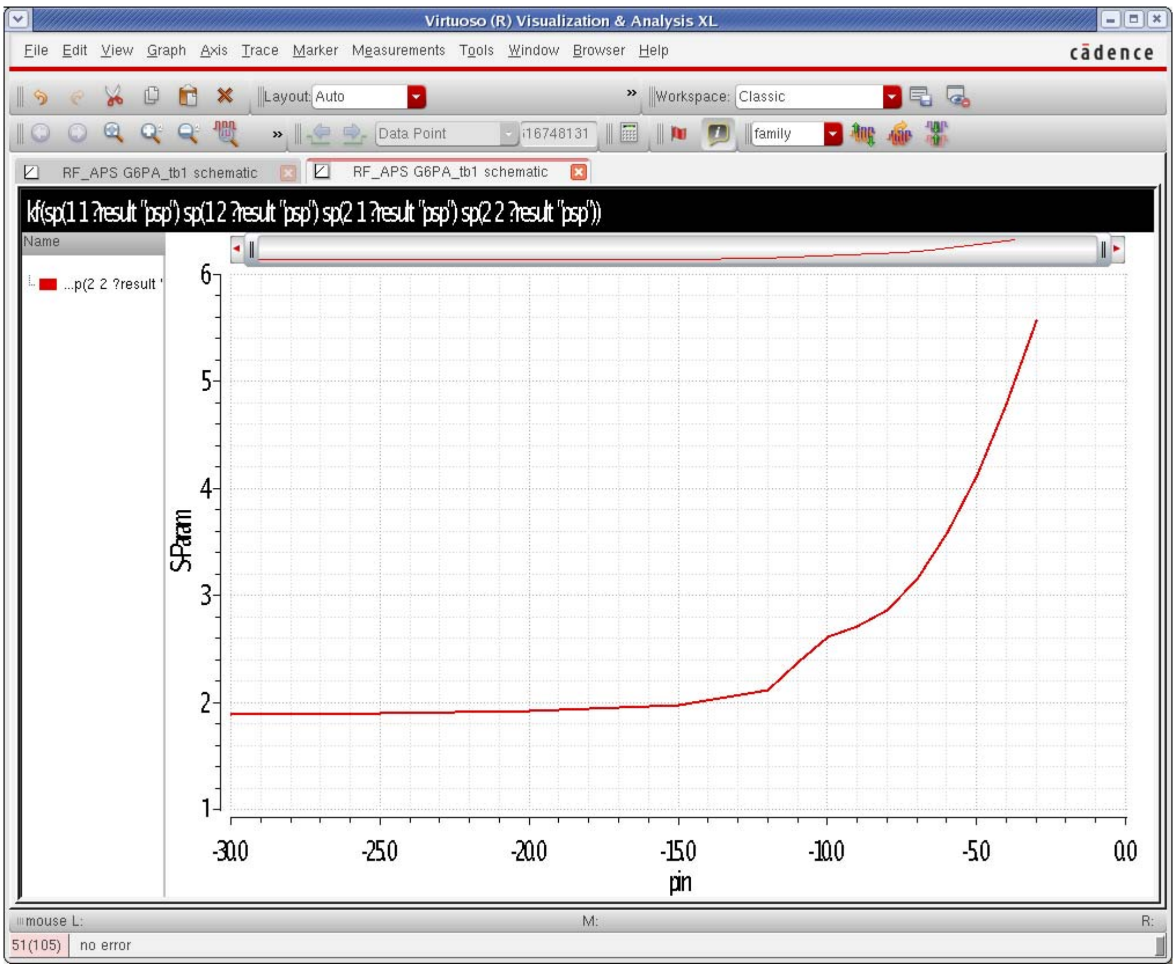
13. To make the plot more usable, move your mouse cursor over one of the numbers on the X Axis, and click the right mouse button. and select *Swap Sweep Var*, from the context menu.



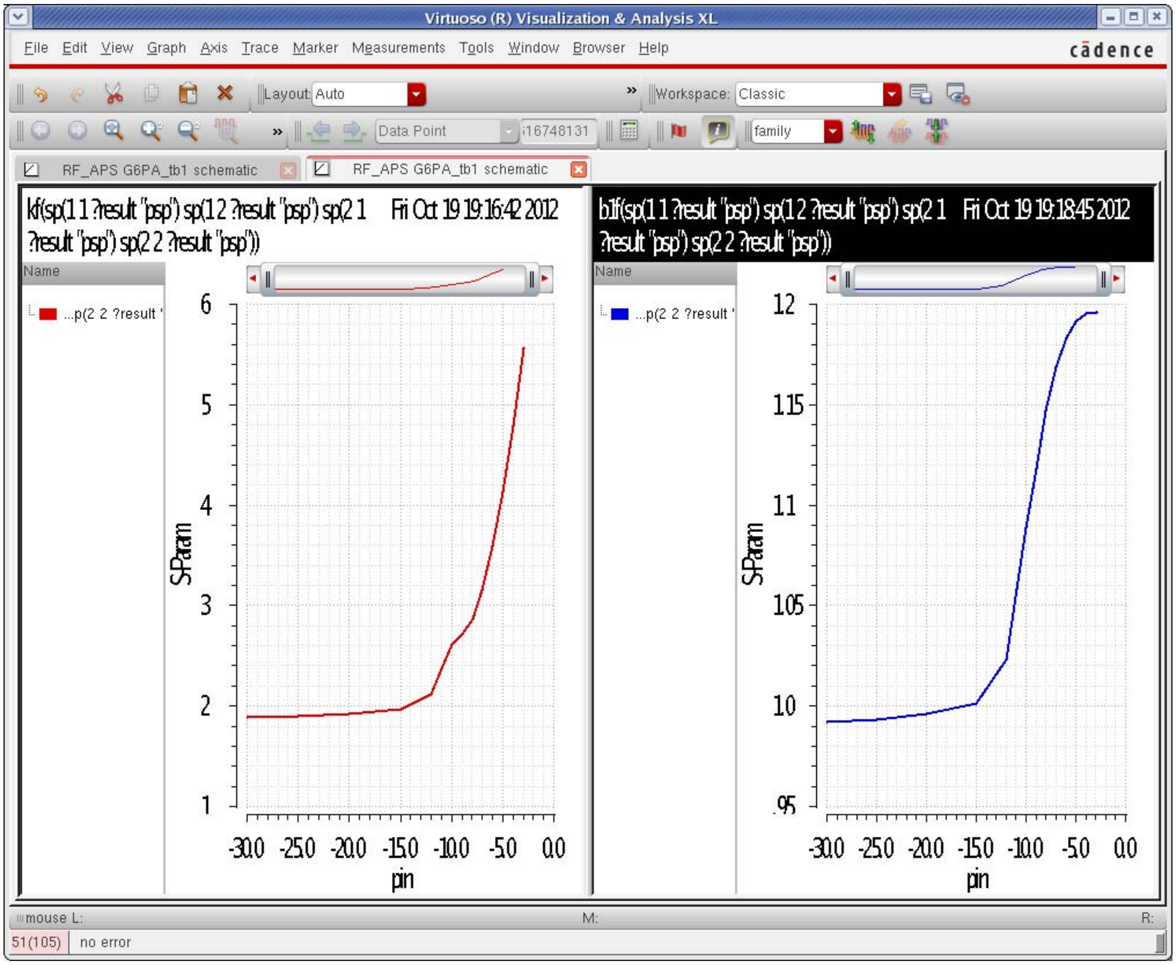
14. A new window appears. Type the frequency value in the X Value field and click OK.



Kf is plotted. The amplifier is stable when the gain is greater than one and the stability factor is greater than one.



B1f is also available. When B1f is greater than zero, and the gain is greater than one, the system is stable.



Both plots show increasing stability as the signal amplitude gets larger.

5

Multiple Input Large and Small-signal Analyses

Quasi-Periodic Steady-State Analysis (QPSS)

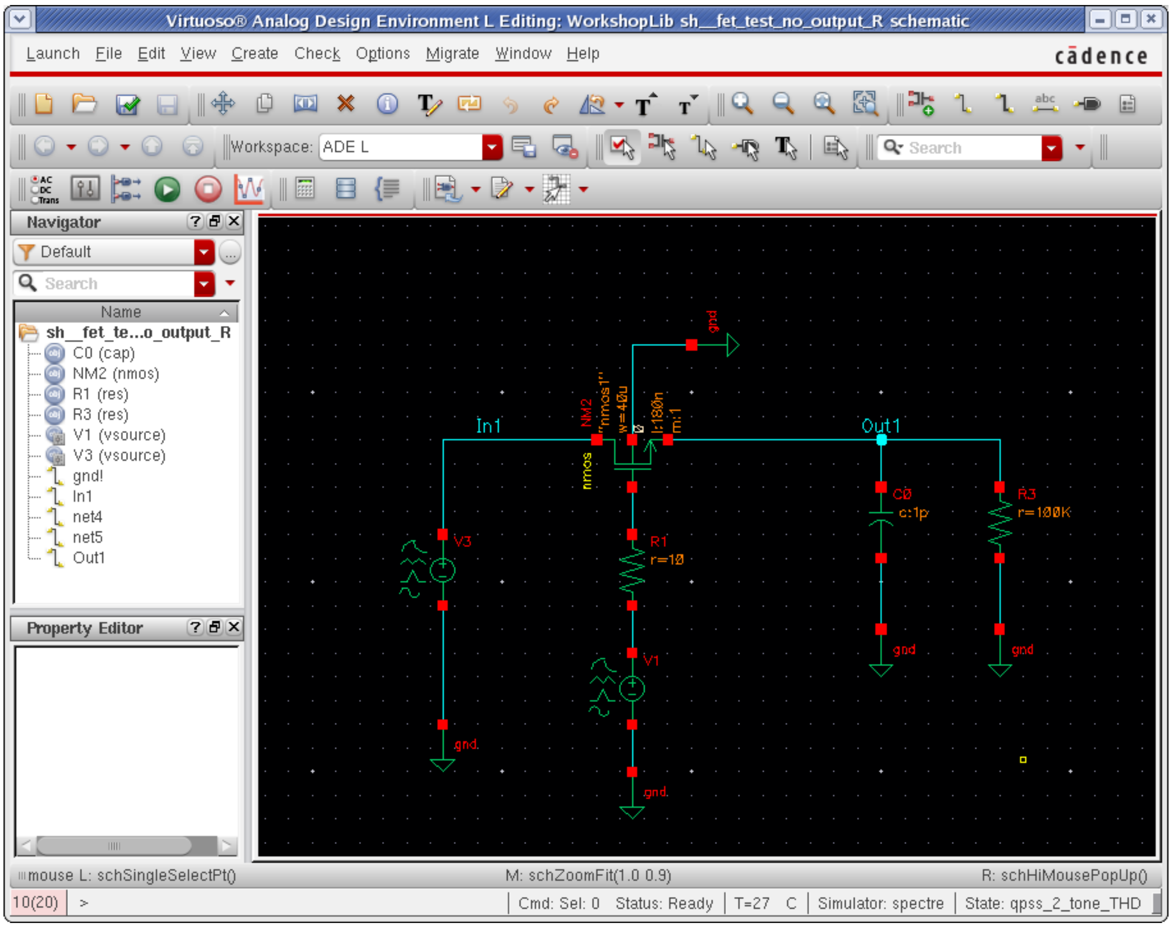
QPSS is a large-signal analysis that is intended for two or more input frequencies. Both shooting and harmonic balance engines are available in qpss. In general, harmonic balance is faster and uses less memory than shooting for most applications. Shooting qpss should be used for the cases where harmonic balance just does not work because the system is too nonlinear. Examples include sampling circuits with a sample clock and a second periodic input, and switched-capacitor circuits with a sample clock and a periodic input.

In general, when harmonic balance qpss is needed, it is better to use the hb analysis as described in Chapter 3 of this user guide. The reason for this suggestion is that all the harmonic balance improvements are being implemented in the hb analysis *Choosing Analyses* form, and generally are not added to the qpss *Choosing Analyses* form. An example is the ability to run an oscillator and a periodic input, which can be done in hb, but not in qpss. Another example is auto harmonic selection.

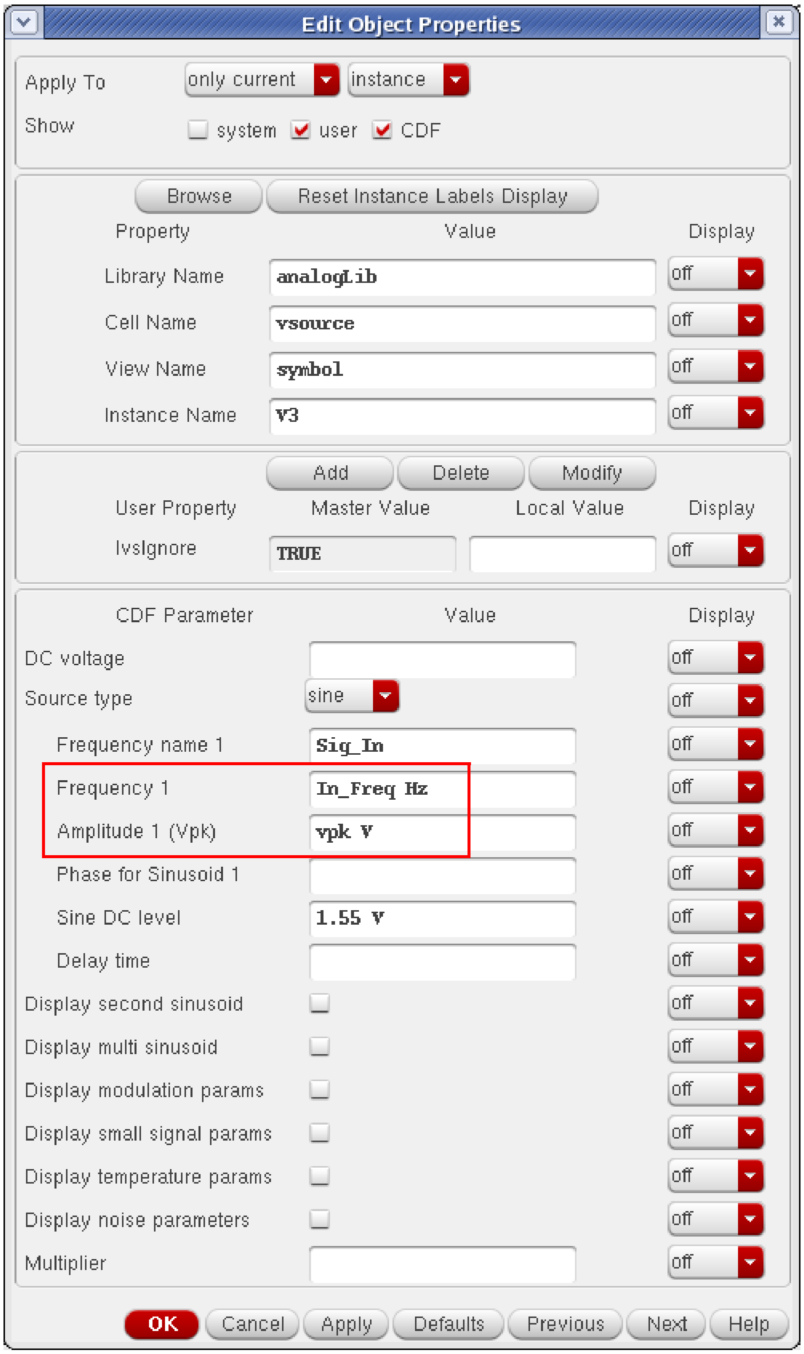
As a strong recommendation, when you need harmonic balance qpss, switch to the hb *Choosing Analyses* form instead.

Example

Consider a sampling circuit shown below.

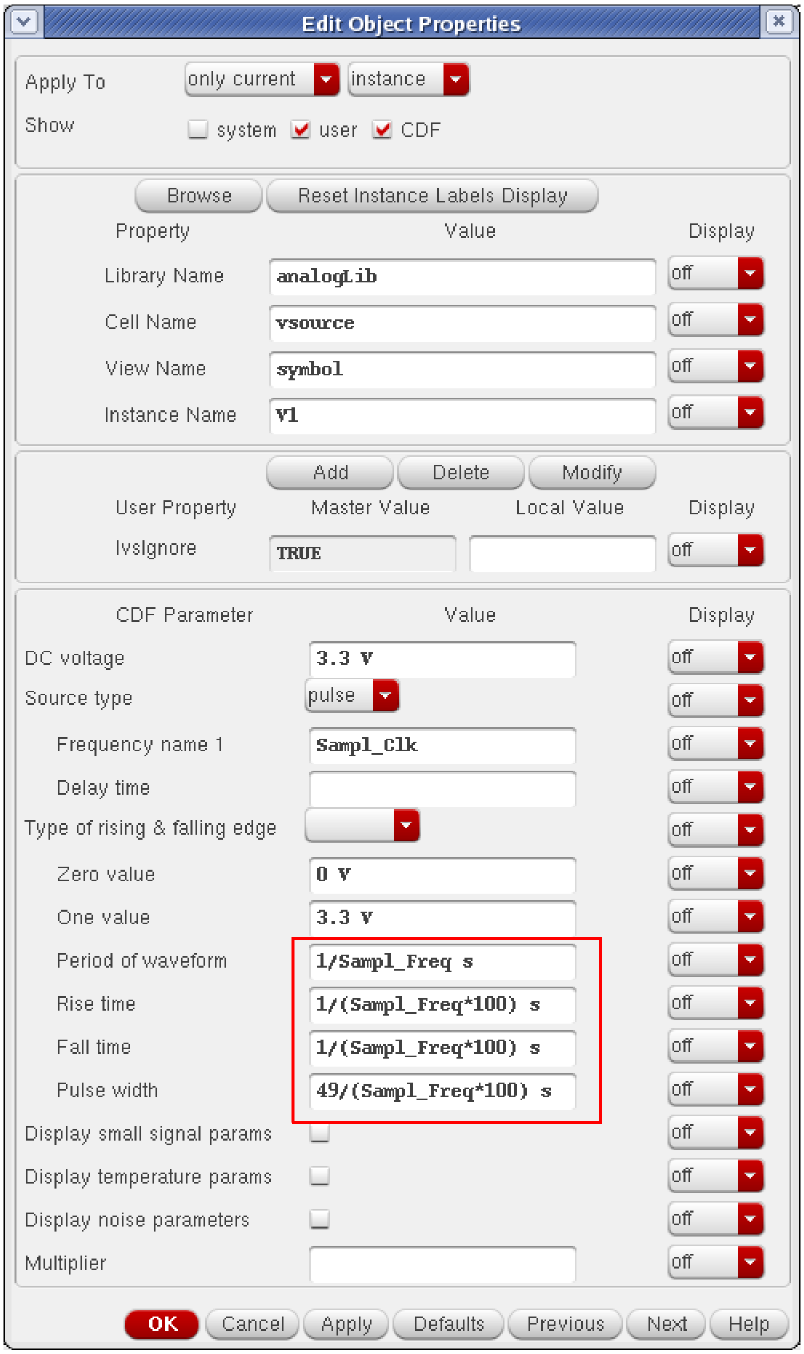


The input source is sinusoidal, and the frequency and amplitude have been set to variable names.

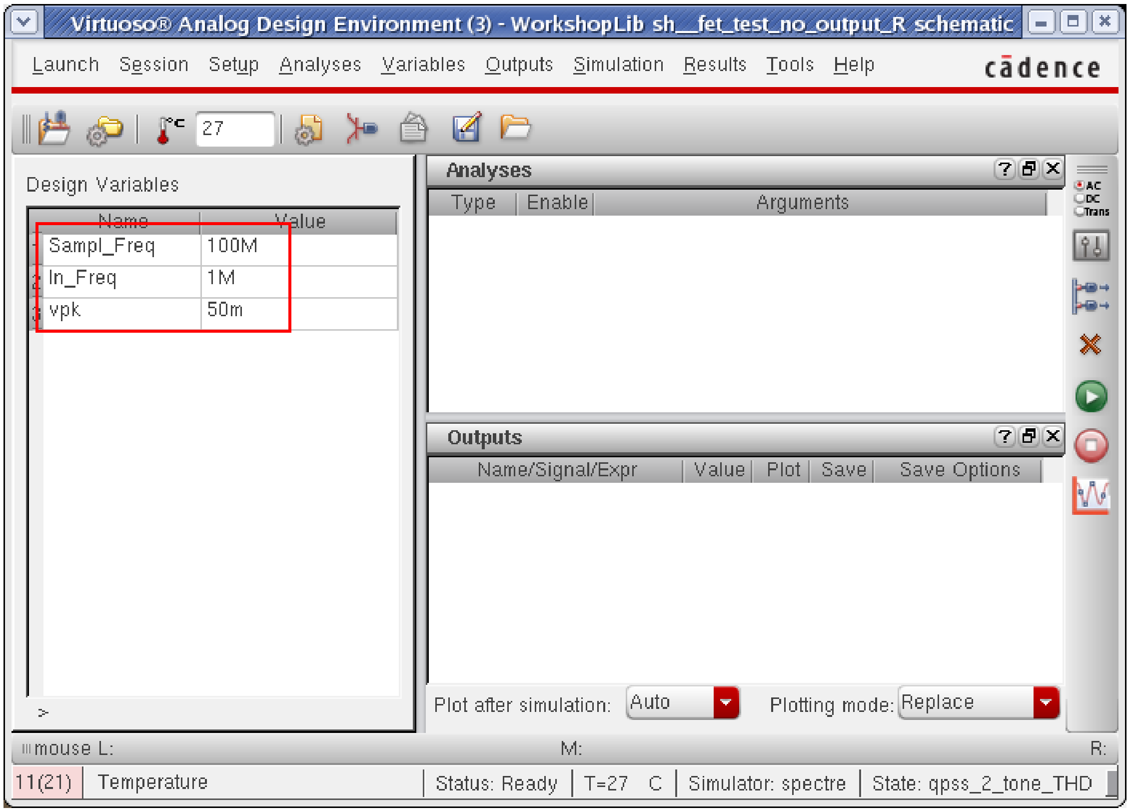


This allows the setting of the frequency and amplitude by changing those variables in the ADE environment without needing to change the circuit. Note also that there is a DC level for the sinusoid. This is because the circuit only includes an N FET switch. This places the signal in the middle of the range of the circuit.

Variables are also used to set the sample clock.

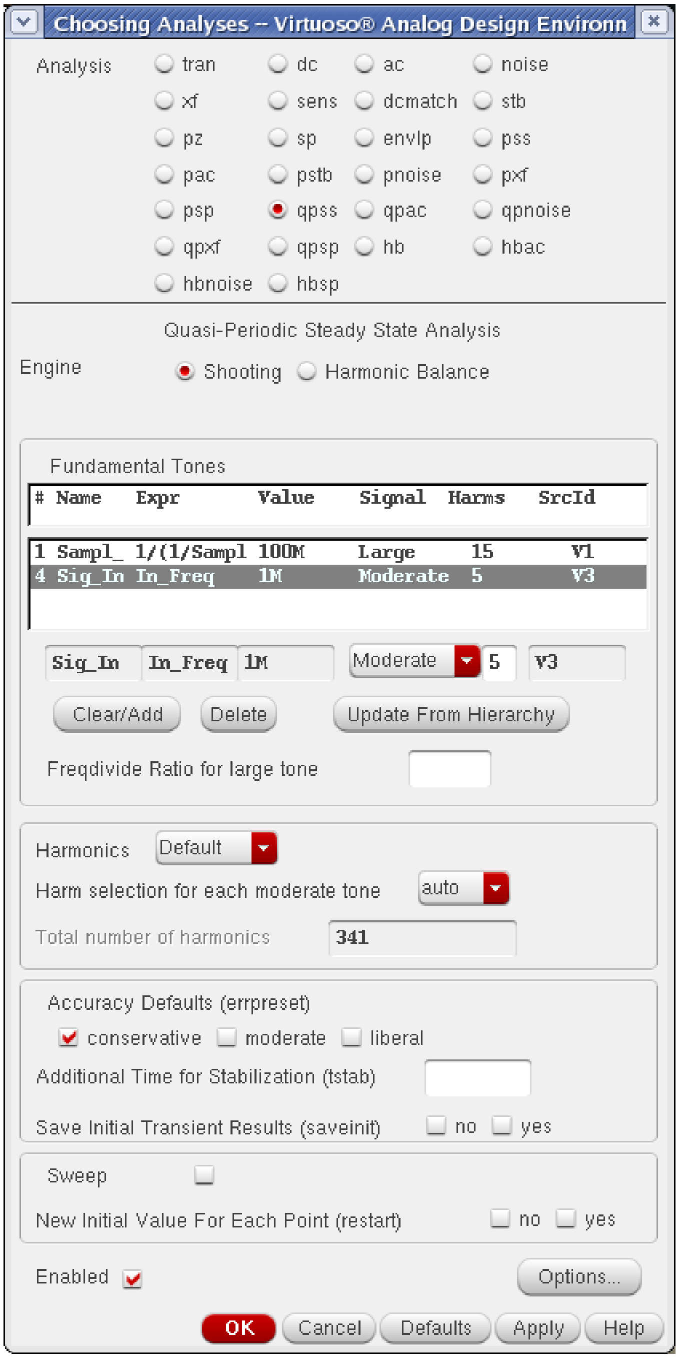


Those variables are set in ADE.



For this example, the input frequency is 1MHz with a 50mV peak amplitude. The sample clock is 100MHz.

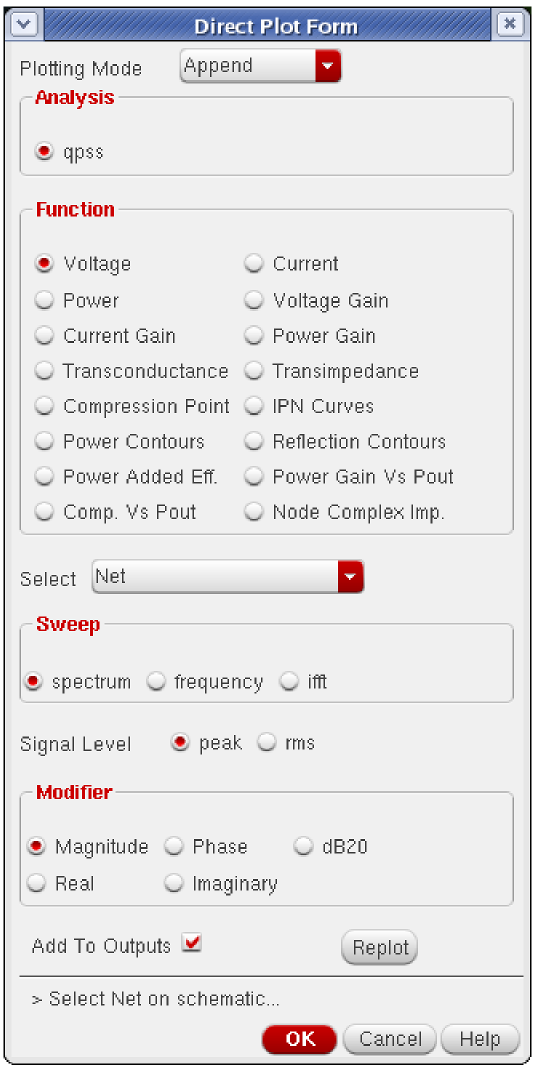
Now the qpss analysis needs to be set up.



The *Engine* is *shooting*. One signal must be set to *Large*, and the rest must be set to *Moderate*. The large signal should be the one that causes the most distortion in the circuit. In this case, that signal is the sample clock. *Large* and *Moderate* are just terminologies used by qpss. The qpss solution is a large-signal solution with all the signals applied to the circuit. More information will be provided later on in this chapter.

The number of harmonics to be calculated must also be set. For reasons described later, the number of harmonics on the Large signal are not critical. For the moderate signals, if the signal amplitude is well below the compression point, two or three harmonics are suggested. If a signal is at the compression point, five harmonics are suggested. If the signal is above the compression point, start with seven harmonics. You may need more harmonics to get an accurate solution.

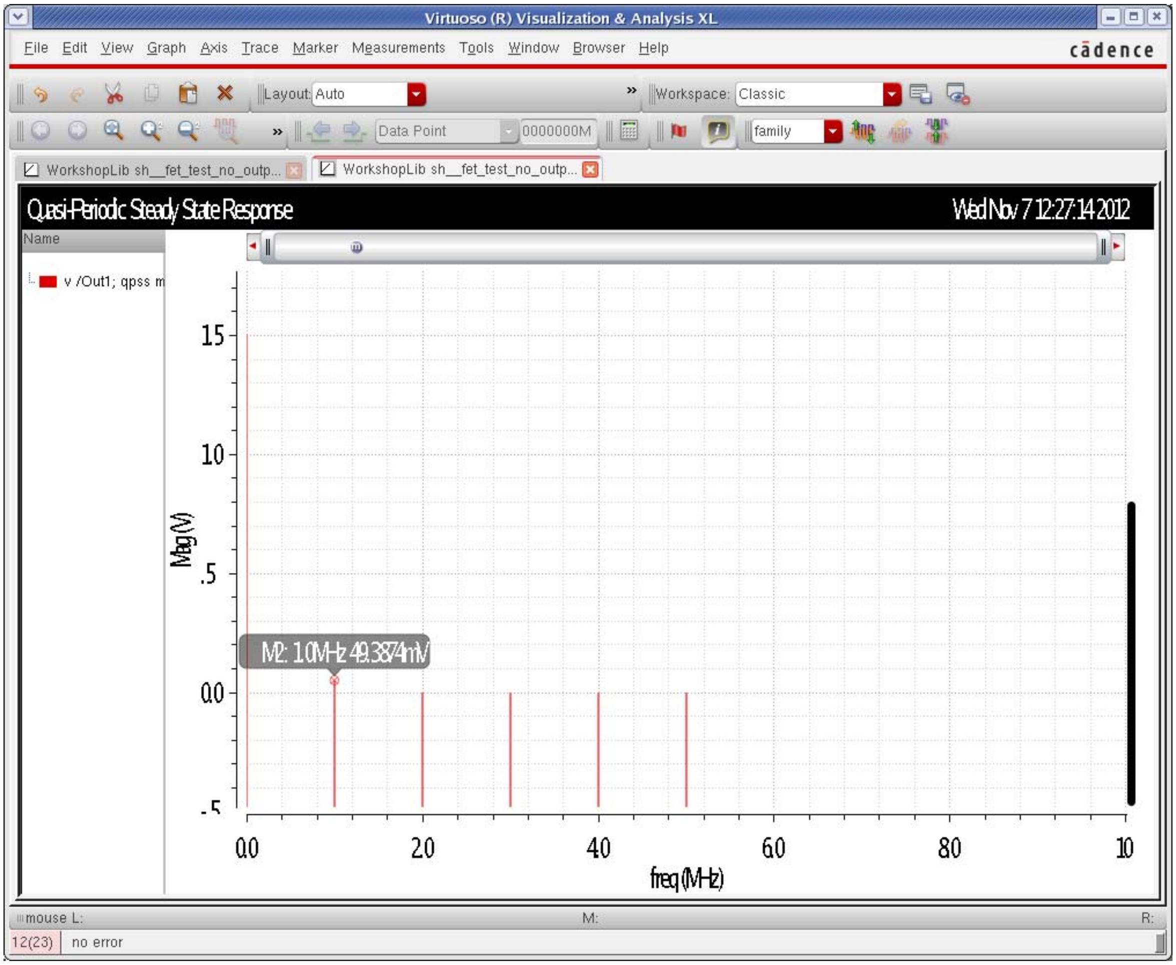
Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE. The *Direct Plot Form* is displayed.



Select qpss results. In this case, the output spectrum will be displayed. Peak is selected because the input is 50mV peak. This allows a direct comparison to be made. Magnitude and not dB is set to get a direct comparison.

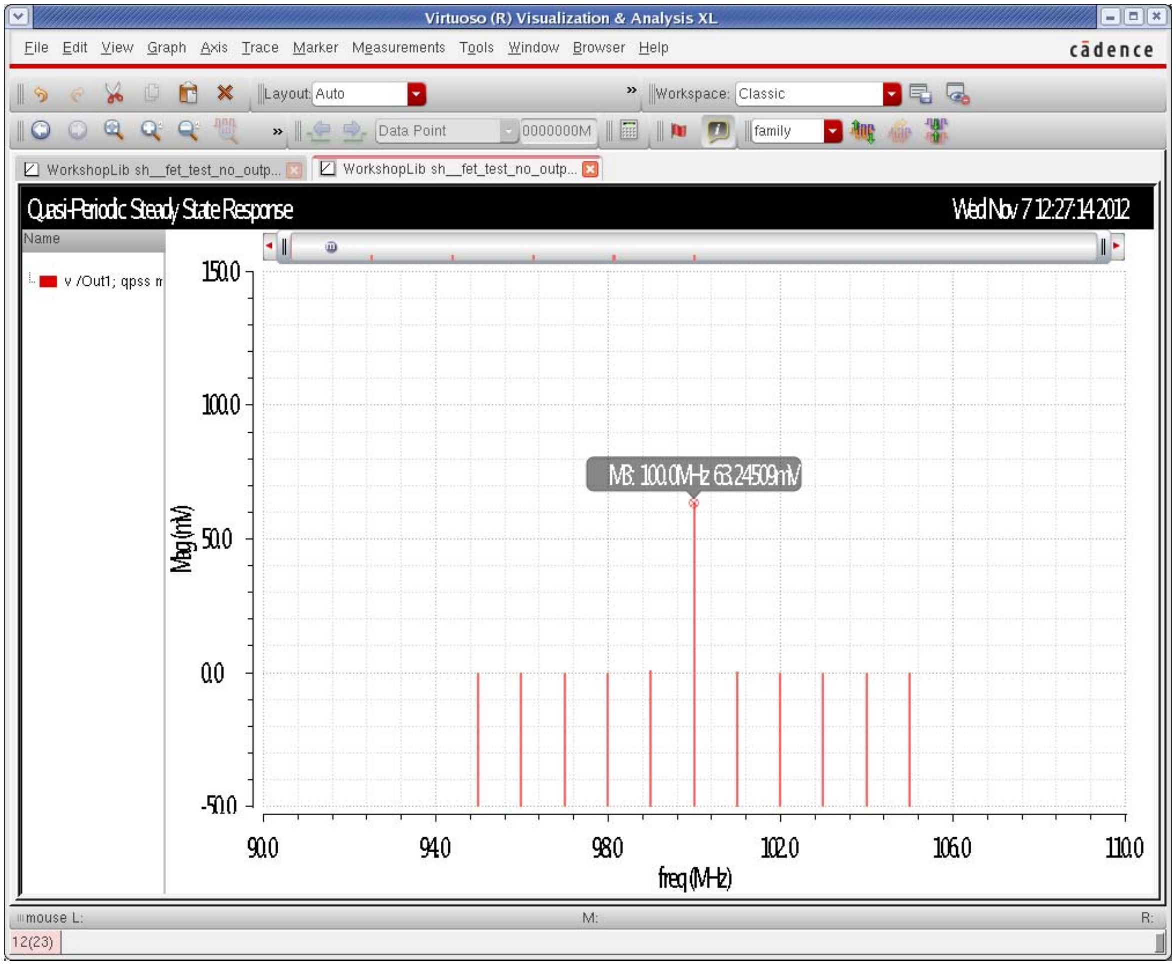
Now select the net in the circuit. The output net is selected for this example.

The waveform is displayed.



The waveform is zoomed in near DC. A marker has been placed to measure the output amplitude at 1MHz. As expected, the output amplitude is slightly less than 50mV peak.

The sample clock feedthrough is also measured. In the zoomed in view near 100MHz (the sample frequency), the amplitude is about 63 mV peak
.



QPSS Shooting Concepts

Think of an example where 1GHz and 1GHz plus 1MHz frequencies are applied to a circuit. If we try to use pss shooting on this case, then 1000 cycles of 1GHz and 1001 cycles of 1.001GHz need to be simulated by solving for the continuous waveform that is produced.

This is impractical for several reasons. First, shooting pss is known to have convergence problems when the total number of cycles of the highest frequency to be simulated exceeds somewhere between 50 and 250 cycles, depending on the nonlinearity of the circuit. Second, when 1000 cycles need to be simulated, at least 20K timepoints are required to get a reasonably accurate waveform. Because pss shooting saves not only the solution voltages and currents (if selected) but also the solution matrix at each timepoint, the amount of memory required for a simulation like this is very high.

If the pss simulation could be completed, all the harmonics of 1MHz would need to be calculated through at least the third harmonic of 1001 MHz, or 3003 harmonics. Not only does the time-domain simulation take a long time, but also the Fourier transform takes a long time.

If you have this case where the pss beat frequency is a small fraction of the highest frequency, use qpss or hb instead.

Qpss is similar to harmonic balance in that you need to specify the number of harmonics to calculate for each tone in the circuit. Qpss calculates the harmonics of the inputs and the mixing products that are specified by the number of harmonics that are set for each input tone. It does not calculate all the harmonics of 1 MHz. Because it does not calculate all the harmonics, it does not need all the data that the pss analysis needs, thus it runs faster than pss shooting for an equivalent set of input frequencies.

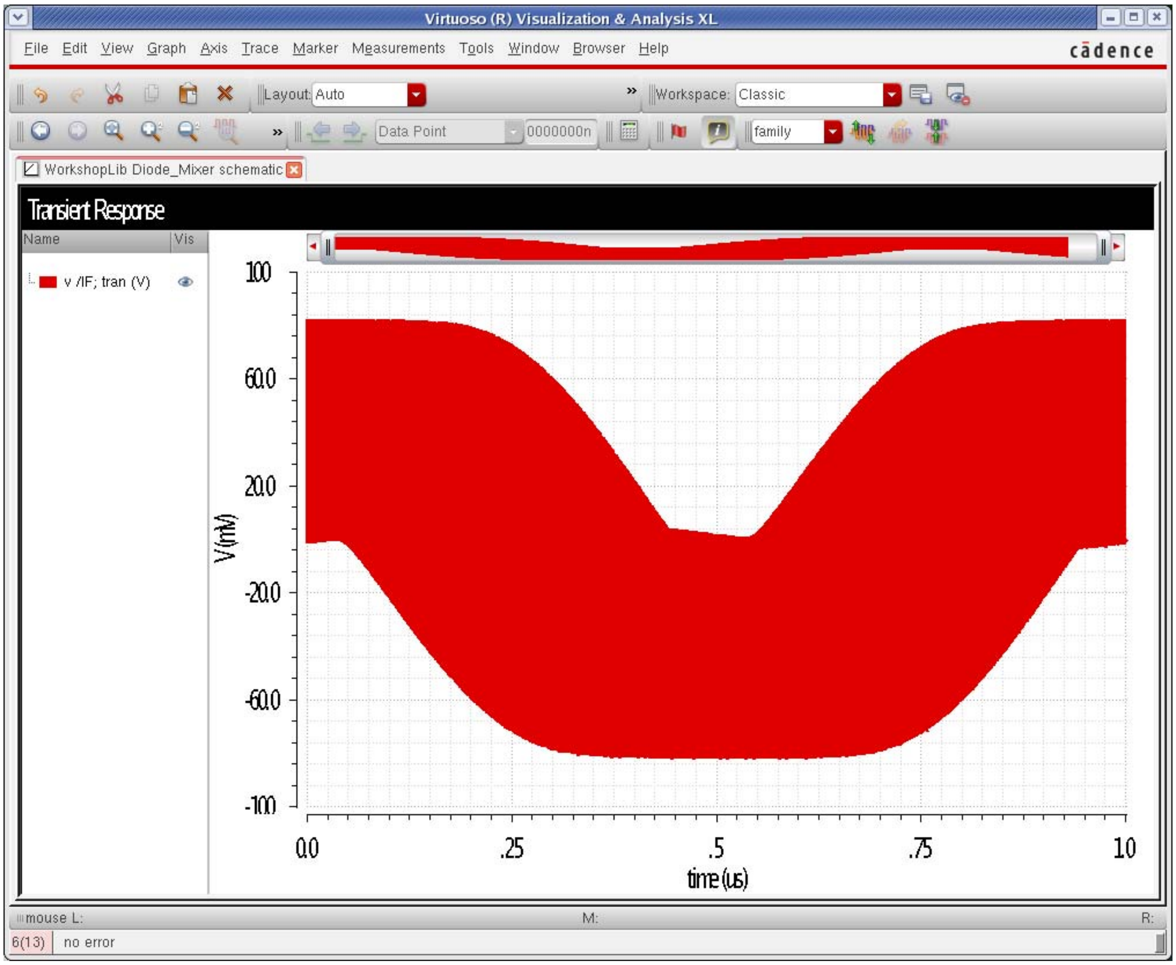
Qpss does not simulate the continuous waveform. Instead, it solves for the waveform in small slices of time during the overall period. The more harmonics you want to solve for requires that more slices of time need to be simulated. Just like in harmonic balance, calculating more harmonics takes more time and memory. All the input signals are applied in the circuit, and qpss solves for the steady-state waveforms in each slice of time. This time-domain output is never available for plotting. Instead, an ifft is offered to calculate the time-domain waveform.

Qpss shooting requires that you select one signal as large, and the rest of the signals are set to moderate. Large and moderate are just terminology used by the qpss analysis. Qpss shooting is a large-signal analysis with all the signals applied.

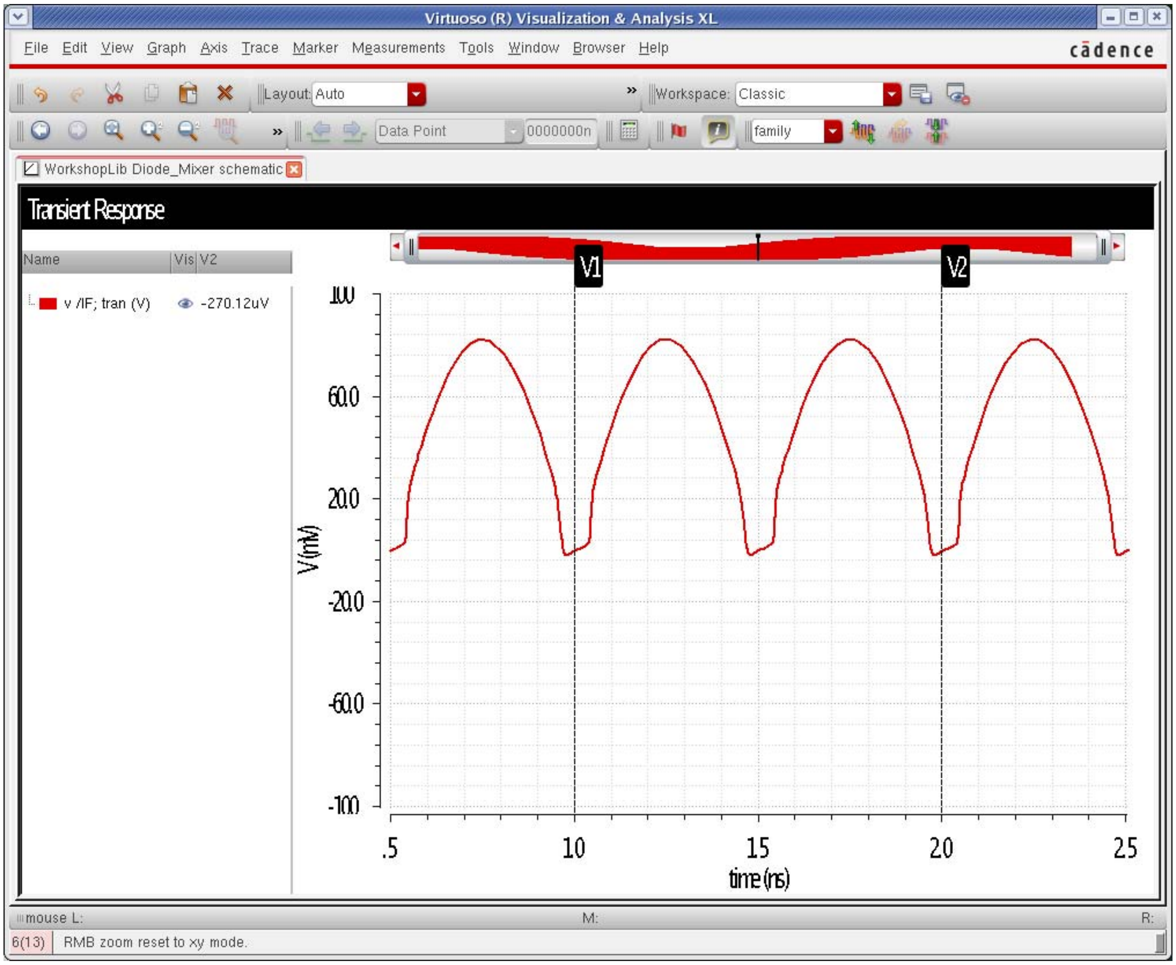
The large signal should be the signal that causes the largest amount of distortion in the circuit, or the non-sinusoidal input if there is one present. All the moderate tones must be sinusoidal.

The large signal specifies the period of each slice in time. In order for qpss shooting to be efficient, the large signal must be a high frequency so that the period of each slice is kept short. If you take an example of 1MHz and 1GHz being applied to the circuit, you must select the 1GHz signal as large, or the runtime of qpss will be really long. If you take this example, if you simulate slices of 1GHz, everything works. If you simulate slices of 1MHz, you have very long runtimes.

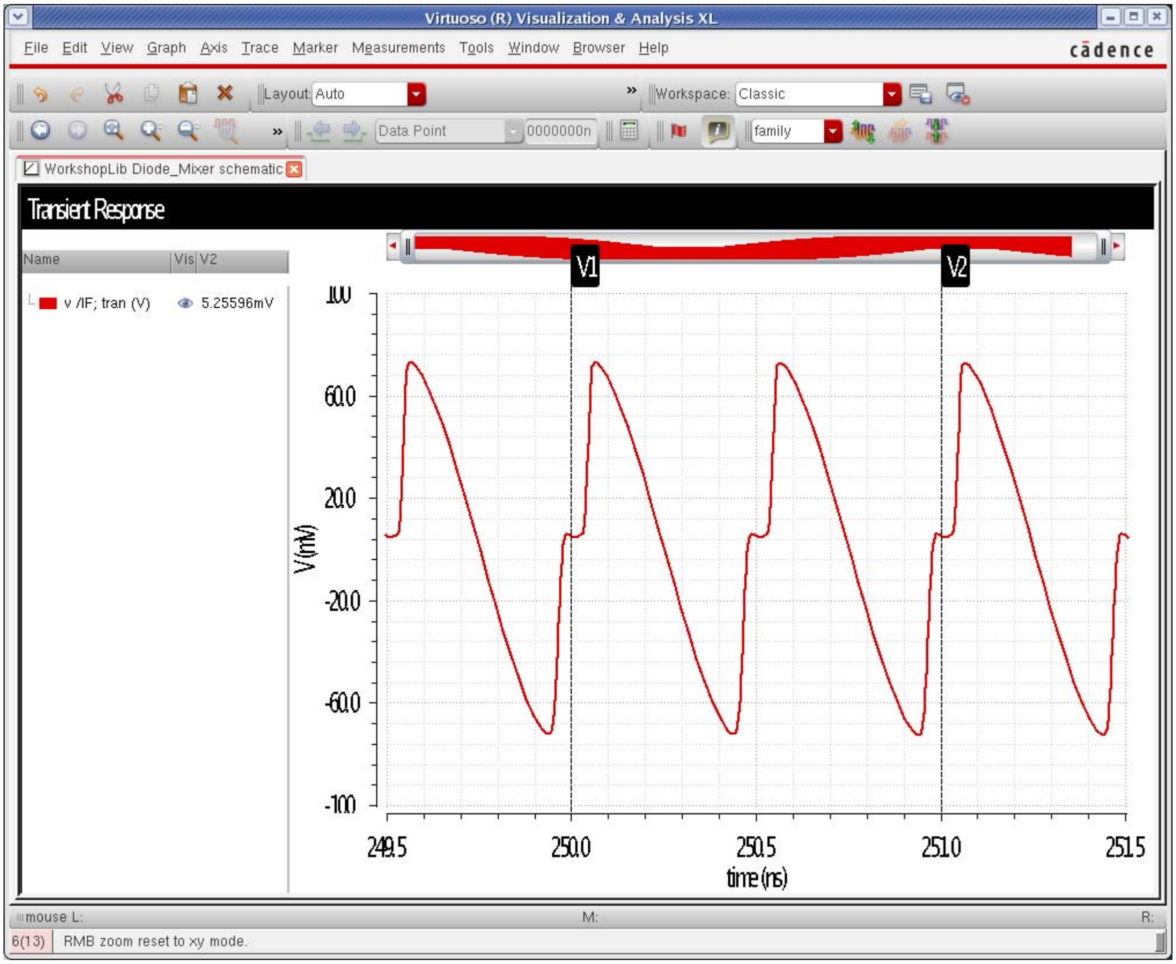
Here is an example for a system with 1GHz and 1.001GHz applied. The overall waveform contains 1000 cycles of the 1GHz signal and 1001 cycles of the 1.001GHz signal. This waveform comes from the transient analysis.



Below are several shots of a 1 nsec period at different times during the simulation. This is at the very beginning of the simulation.



This is 1/4th of the way through. The waveform in this slice is very different.


Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Layout: Auto
Workspace: Classic
Data Point
family
WorkshopLib Diode_Mixer schematic
Transient Response
Name
Vis
V2
v /IF; tran (V)
5.25596mV
V1
V2
V (mV)
100
60.0
20.0
-20.0
-60.0
-100
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250.0
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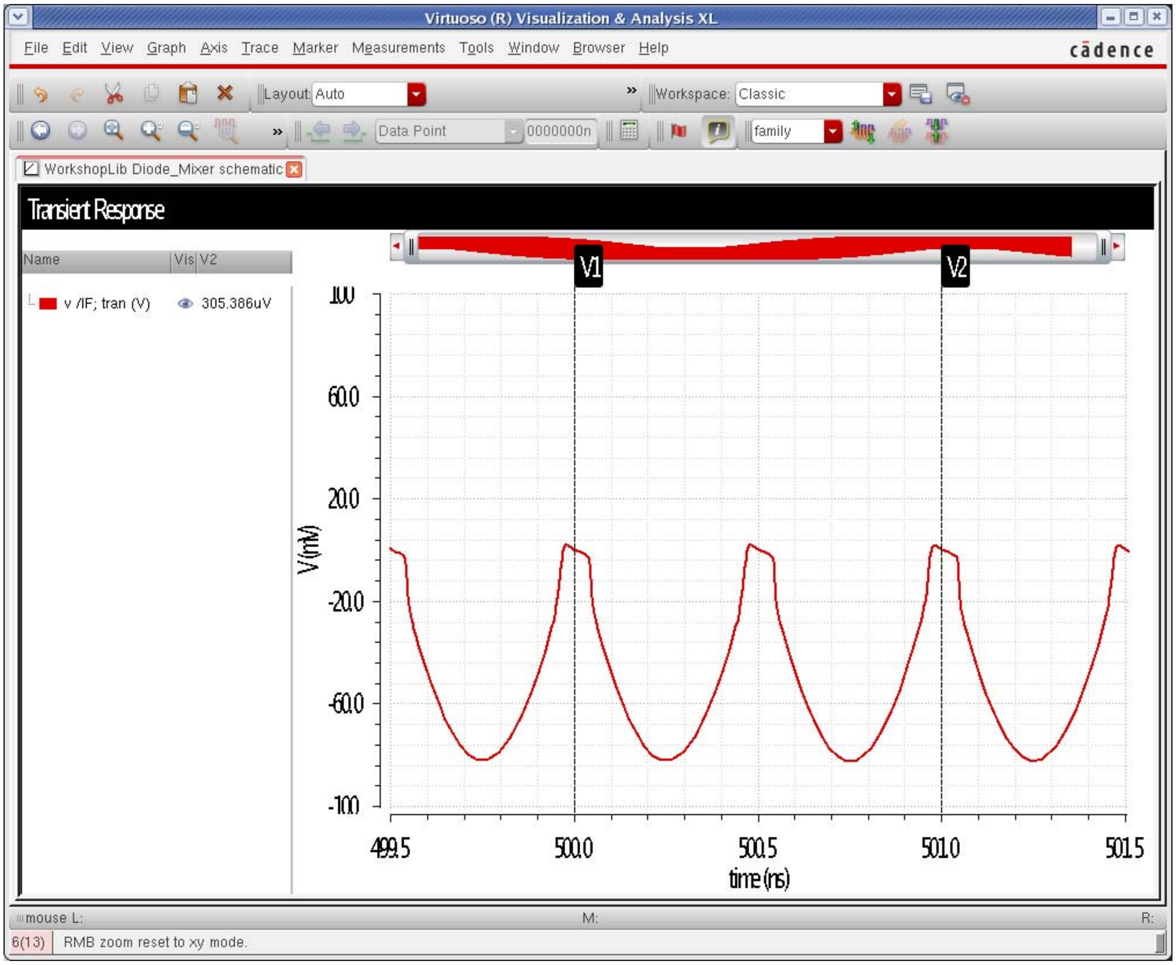
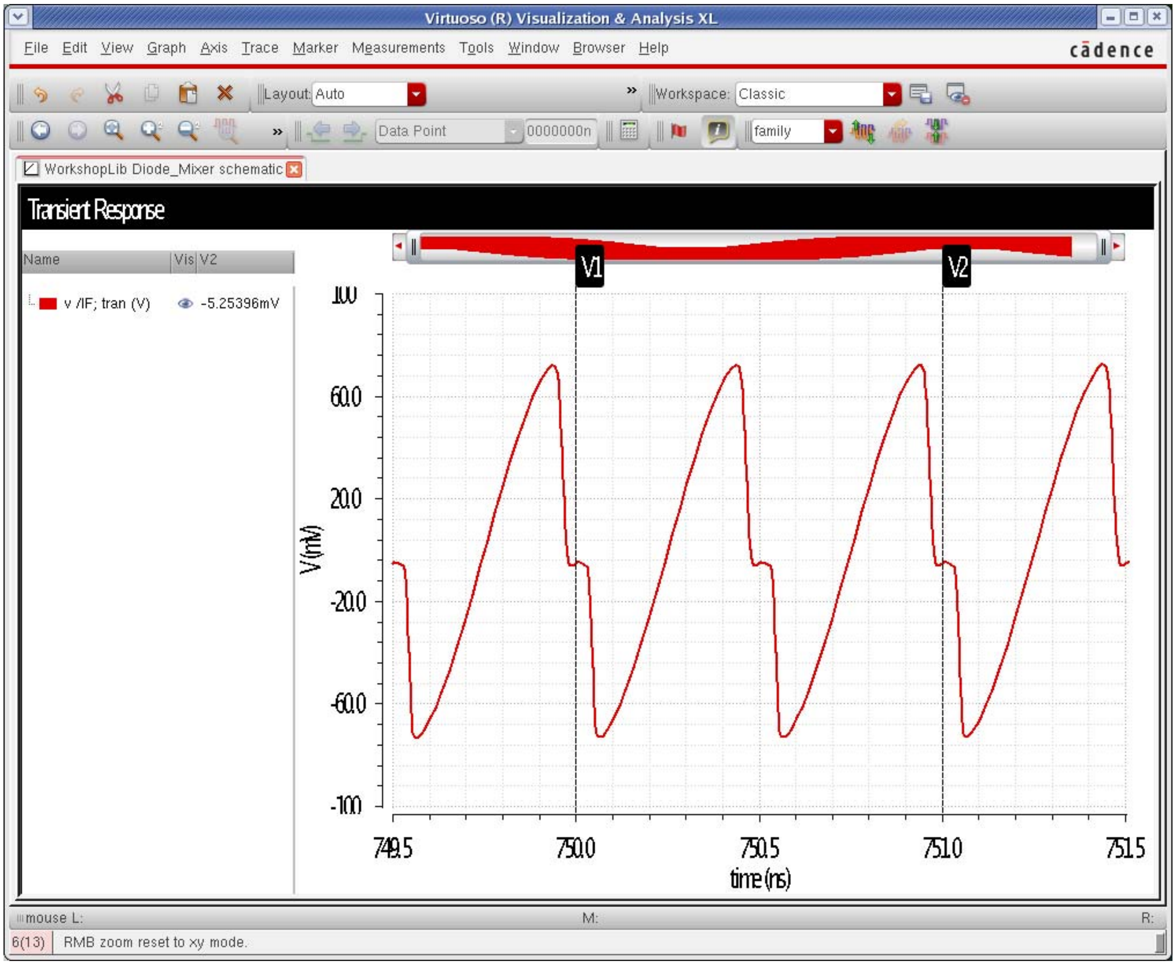
251.5
time (ns)
mouse L:
M:
R:
6(13)
RMB zoom reset to xy mode.

This is half way through.



This is 3/4ths of the way through.

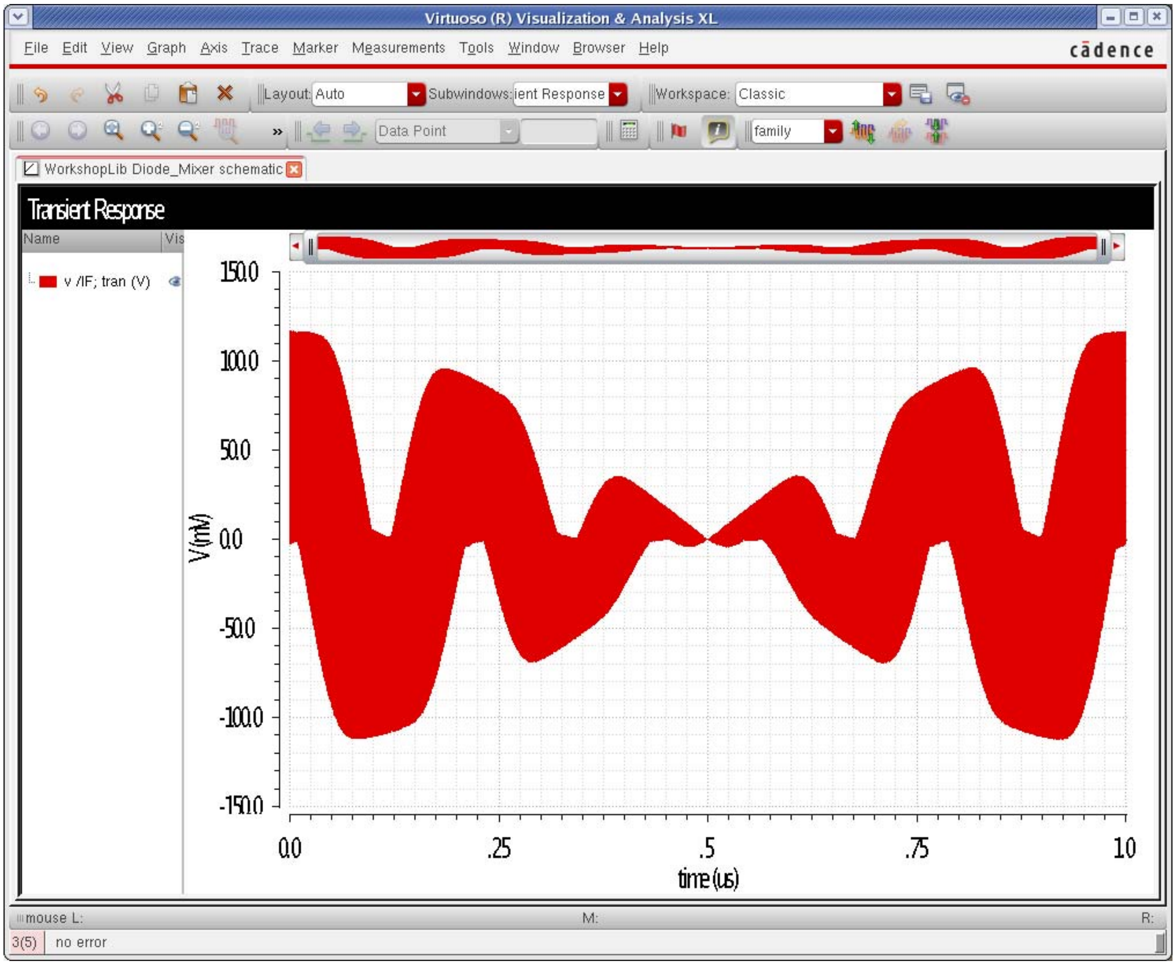


What qpss shooting does is solve for the steady-state waveform with all the signals applied in the slices that are needed in order to calculate the number of harmonics that are specified for each signal.

Calculating harmonics for the large signal in each slice and averaging the power at each harmonic gives the power in the large signal.

For one moderate signal, (2 * Number of harmonics) + 1 slices need to be simulated. For three harmonics on the moderate signal, then 7 slices need to be simulated. This gives enough information to solve for the positive frequencies, negative frequencies, and the DC term to be calculated.

If there are two moderate signals, then the number of slices is [(2 * Number of harmonics on moderate 1) + 1] * [(2 * Number of harmonics on moderate 2) + 1]. If there are three harmonics on each moderate tone, 7 * 7 or 49 slices need to be simulated. The waveform becomes more complex and the number of harmonics that are created by the system are larger when three tones are applied to the system. below is the transient waveform for 1GHz, 1.004GHz and a 1.005GHz signals applied to the system. This waveform is from the transient analysis.

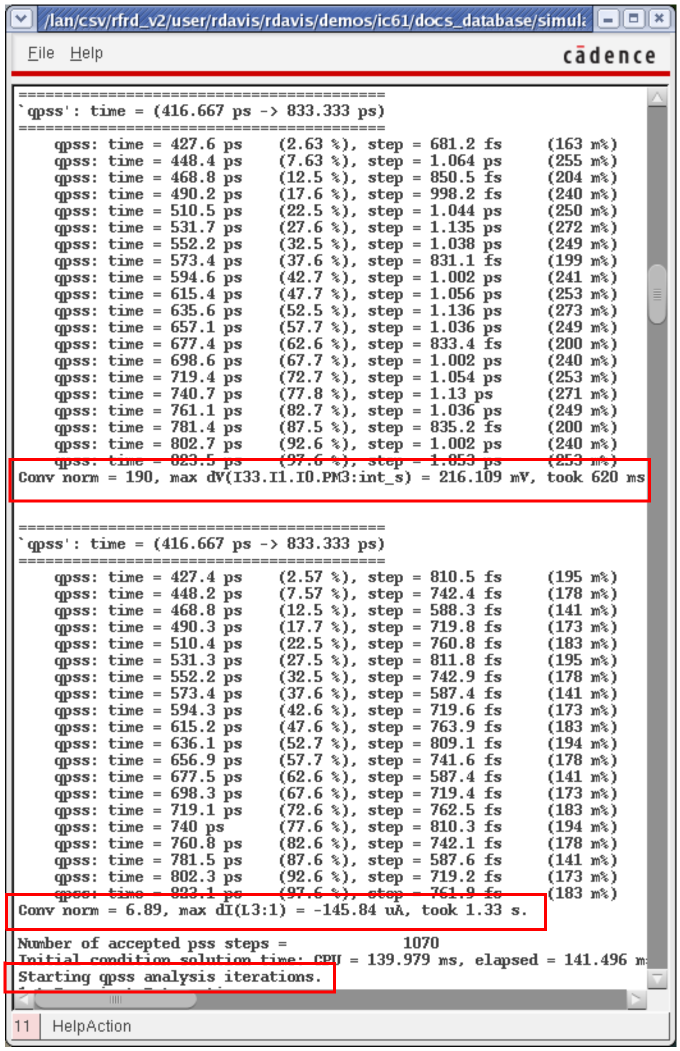


Each slice is a pss analysis where the solution voltages and currents (if specified) and the solution matrix need to be saved at every timepoint of every pss analysis. For more information about the pss analysis, see the beginning of Chapter 4.

At the end of the simulation, the harmonic content is calculated based on the waveforms in the slices. Since all the input signals are applied in every slice, the harmonics of all the mixing products can be calculated.

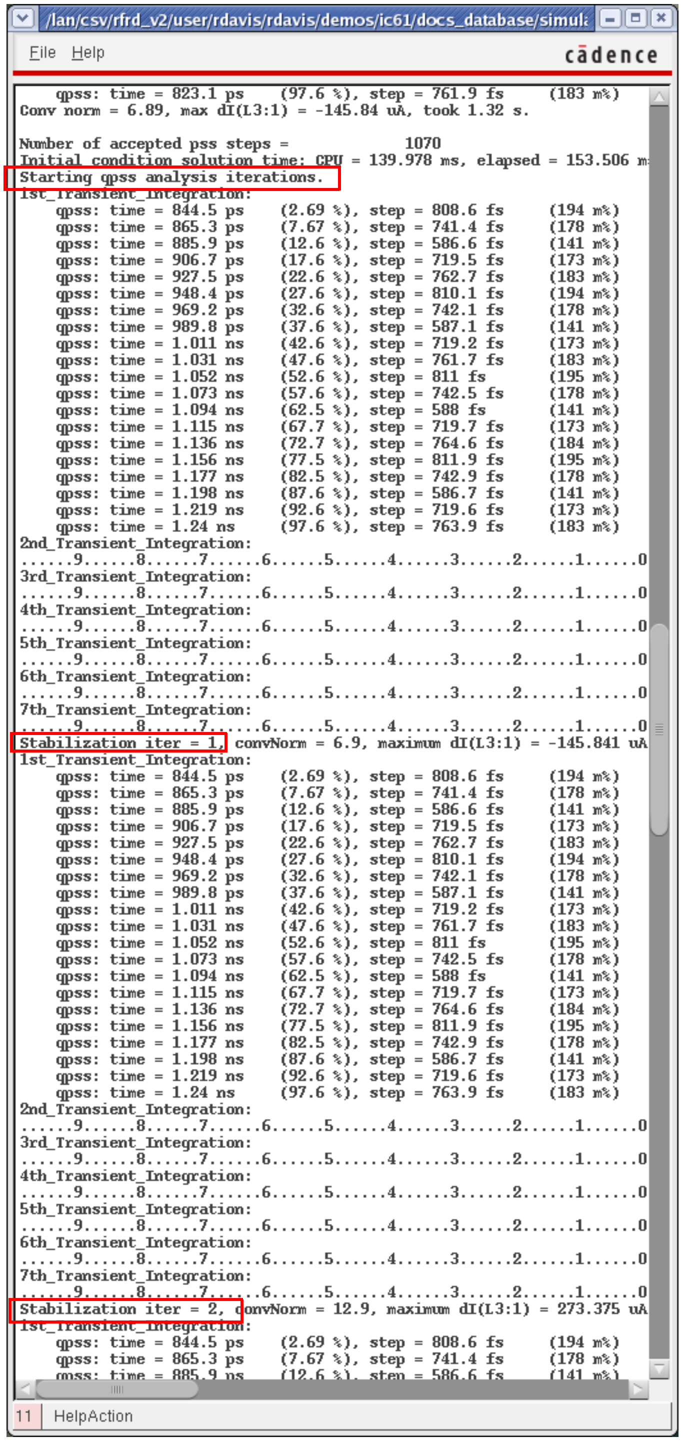
What Happens When you Start a QPSS Analysis?

First, two pss iterations are performed with only the large signal applied. This gives an approximate harmonic content with just the large signal applied. This is shown in the Spectre output window, as shown below.

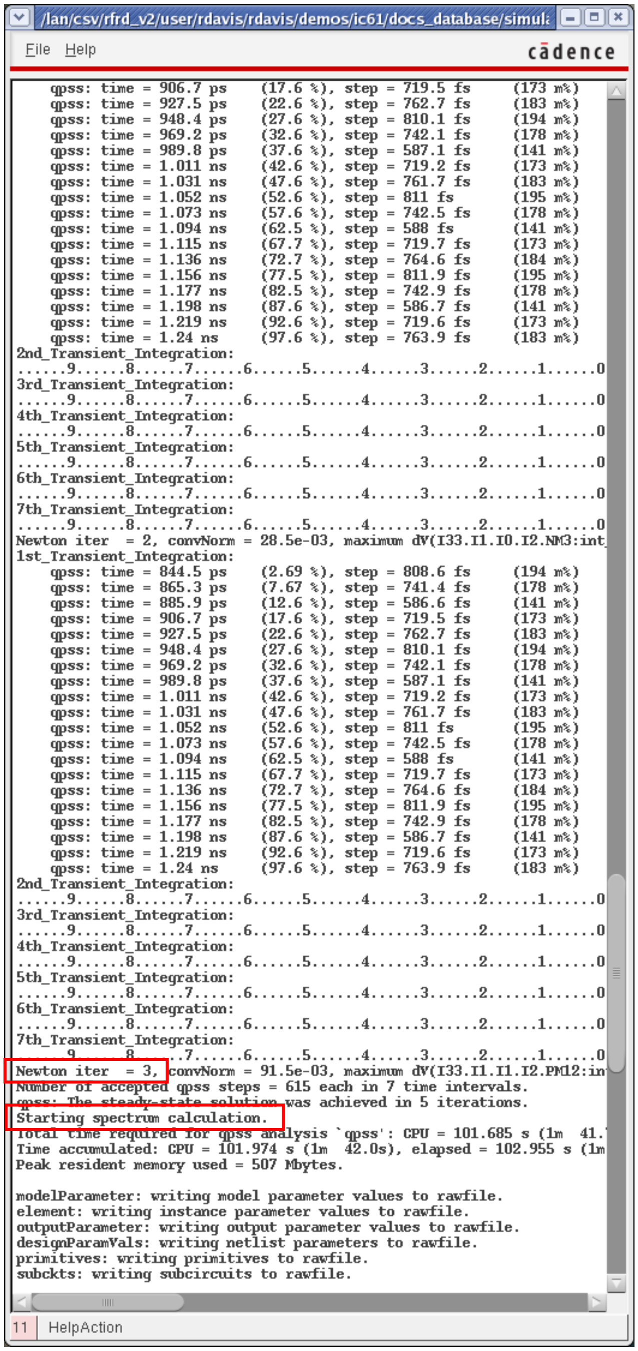


Next, two (by default) stabilization cycles are performed. All the signals are applied to the circuit, and a very simple-minded optimization to attempt to have all the currents sum to zero at all the harmonics at all the nodes is performed. In this phase you will see that all the slices are being simulated. There is only a very small time delay between the first set of slices and the second set of slices. These slices are reported as transient integrations in the qpss output window. In this case, there was a large signal and a single moderate signal with three harmonics specified. This results in seven slices that need to be simulated. Each slice is called a transient integration in the Spectre output window.

The number of stabilization cycles can be controlled by an option called stabcycles. This is shown in the Spectre output window, as shown below.



Finally, a a series of full Newton iterations to the solution of the system is performed. Again, all the slices are solved, and iterating continues until convergence is reached. After each set of slices is simulated, there is a delay while the next iteration is calculated in the Newton iteration. A maximum of 20 Newton iterations are allowed by default, and can be changed by setting the maxperiods option. This circuit converged in three Newton iterations.



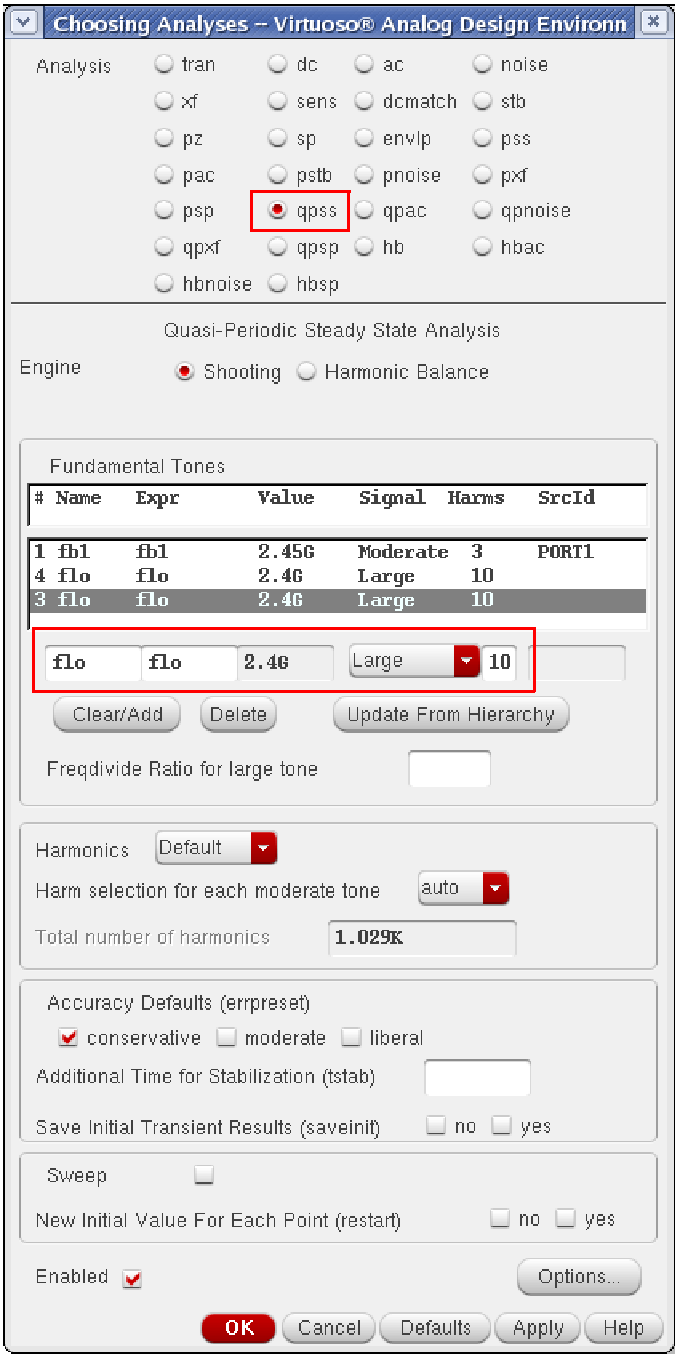
ADE Implementation

In this section, just the qpss analysis *Choosing Analyses* form and the options for qpss are shown. An examples section follows with all the steps needed for a simulation.

To get the *Choosing Analyses* form select *Analysis - Choose* in ADE, or select the *Choosing Analyses* icon 

located on the upper right of the ADE window.

The *Choosing Analyses* form is displayed as shown below.



In the *Choosing Analyses* form:

1. Select *qpss*.

2. The default engine is *Shooting*. If you prefer Harmonic Balance, use the hb form, as this form is most up to date.

3. All the sources are listed at the top of the form. To change the settings, select one of the lines, and change the values in the editing line just below the list.

4. One signal must be *Large*. This should be the signal that causes the most distortion in the circuit. The frequency of this signal must be relatively high compared to the highest input frequency, or the simulation will be quite slow and require a lot of memory. If there is a non-sinusoidal source in the circuit, that signal must be set as large. All the moderate tones must be sinusoids.

5. Select an accuracy level. Liberal is not suggested.

Setting errpreset changes the defaults of a number of qpss options. The options that are affected are shown in the table below.

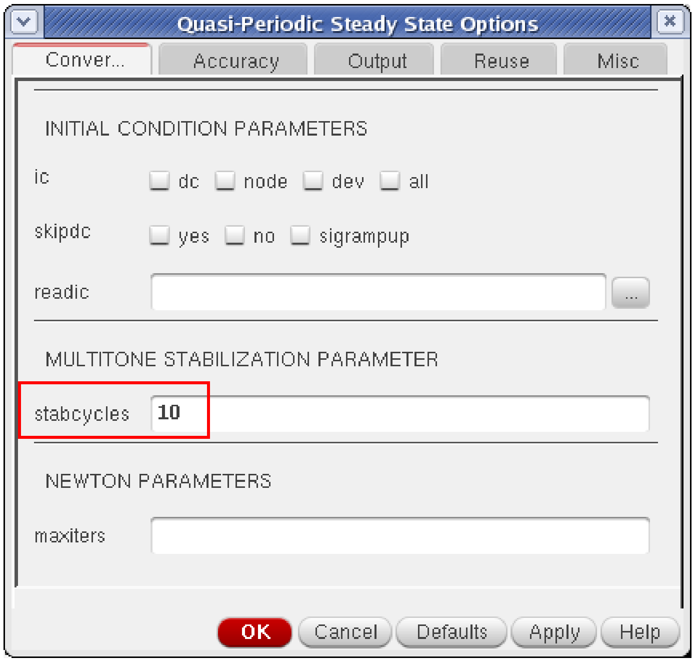
Errpreset	Reltol	Lteratio	Maxstep
`liberal`	`1e-3`	3.5	Period/80
`moderate`	`1e-4`	3.5	Period/100
`conservative`	`1e-5`	10; 3.5; if reltol is smaller than `2.85e-4`	Period/200

QPSS Options

Commonly Used Options

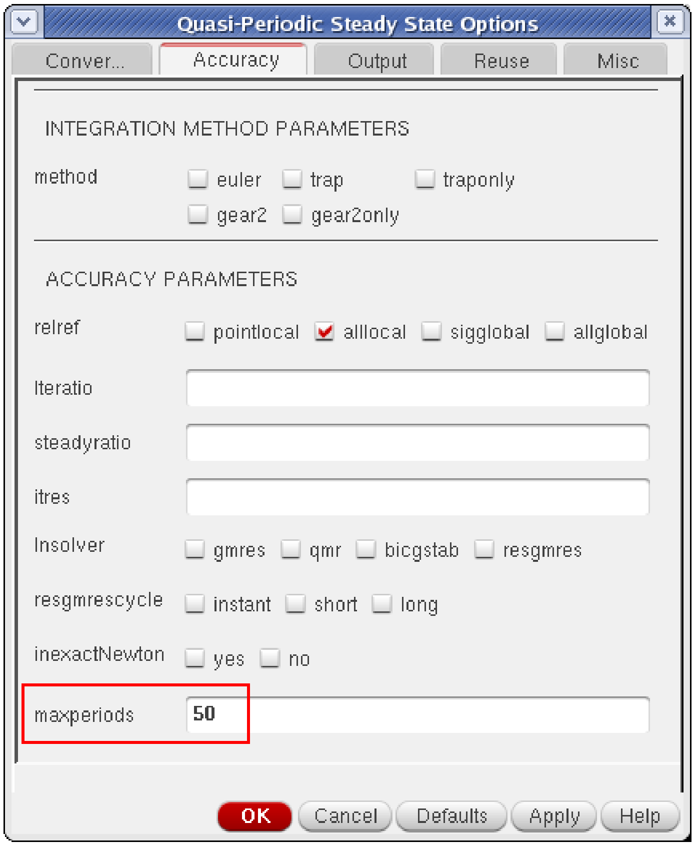
There are only a small number of options that are commonly used.

stabcycles



If you are having convergence problems, increasing stabcycles usually helps. Increase to about 10. If this does not work, look at the `conv_norm` number at the end of each stabilization cycle, and increase stabcycles until the `conv_norm` does not decrease much with more cycles.

maxperiods



Watch the *conv_norm* number that is reported at the end of each Newton iteration. If this number is generally trending down, then more newton iterations may allow convergence. Start with about 50.

Uncommonly Used QPSS Options

The following options are in the *Convergence* tab.

ic

Initial conditions can be specified graphically by selecting *Simulation - Convergence Aids - Initial Condition* in ADE. Initial conditions can also be specified from a file using the *readic* property. Capacitors and inductors have the initial condition properties in the property list for the component. For capacitors, this is an initial voltage across the capacitor, and for inductors, it is an initial current in the inductor. The default is to observe all the initial conditions in the DC analysis that is used as the time-zero timepoint. The initial conditions force a voltage or current to be present in the time-zero solution. The initial conditions are released for the rest of the transient simulation in the tstab interval. The *ic* option controls the initial conditions that should be observed in the time-zero timepoint. *all* is the default. *dev* means that only the initial conditions on capacitors and inductors are observed. *node* means that only the initial conditions on a node are observed. *dc* means that no initial conditions are observed.

skipdc

In some cases, the time-zero timepoint DC analysis does not converge. Instead of stopping the simulation, *skipdc* allows the tstab simulation to continue using an assumed solution for the time-zero timepoint in the tstab interval. The default is *no* and a DC analysis is run to get the initial timepoint. *yes* means skip the DC solution, and proceed directly to the tstab simulation. All the nodes with initial conditions specified start at the initial condition value. Nodes with batteries start at the battery voltage. Nodes with no initial conditions start at zero volts. For `skipdc=yes`, the signal sources start as specified immediately in the tstab simulation. *sigrampup* uses the same assumptions for the starting voltages as *yes* does, but the start time is set to negative one tenth of the stop time for the tstab interval. At this time, the signal source time-varying part starts at zero and linearly ramps up to the full value at time = zero. After time = zero, the sources have the full amplitude time-varying part.

readic

This specifies an ascii file that contains two columns to be read as initial conditions. The first column is the node name. The second column is the voltage value. If the entry does not start with / (slash), the file is located in the netlist directory. To find the netlist directory, select *Setup- Simulator/Directory/Host* in ADE. Look in the *Project Directory* field for the location of the simulation directory. Navigate to that directory and then to the `<Circuit Name>/spectre/<schematic or config>/netlist` directory. You can also click (...) and browse to the directory.

maxiters

The transient analysis for tstab and shooting interval iterate to a solution at each timepoint. *maxiters* specifies the maximum number of iterations before the timestep is cut for another try at convergence. In some cases, model parameters can cause discontinuities in the device current or capacitance. If this occurs, the change in the circuit condition when the discontinuity is crossed can be large enough to require more than the five iterations that are allowed by default. Specifying *maxiters* to the 40 to 100 range usually allows the simulator to converge in spite of the discontinuity.

The following options are in the Accuracy tab.

method

This option controls the integration method for tstab and the shooting windows.The default value is *gear2only*, which is recommended for all qpss simulations.

relref

In the order of most to least accurate, the *relref* settings are *pointlocal*, *alllocal*, *sigglobal*, and *allglobal*. The default for the *moderate* and *conservative* accuracy selections is *alllocal* which is preferred. Liberal is not suggested.

relref is used in the transient analysis in the tstab interval for both shooting and harmonic balance and in the shooting window for shooting. In some cases, the timestep can collapse to near zero when the default is used for relref. The symptom is that in the Spectre output window, no progress is made for many reporting intervals. By default, every 10 seconds, Spectre will show progress in the tstab and shooting intervals. For example, for every update in the Spectre output window, the percent complete might stay at 15%. The solution to this is to set *relref* to *sigglobal*, which is slightly less accurate. In the Fourier transform, the noise floor usually degrades by 3 to 6 dB. Another possibility is to try the *lteminstep* option as described in the *AdditionalParams* option of the *Misc* tab.

Refer to Relref on page 563 for a detailed discussion on *relref*.

lteratio

lteratio is a multiplier for the allowable numerical integration error in the tstab interval and the shooting windows. The default is 3.5. *lteratio* cannot be set smaller than 1.0 and is normally between 3.5 and 100. In some cases, the timestep collapses to near zero In this case, there might be a model discontinuity. To test this, disable numerical integration timestep control by setting *lteratio* to 1e9. Note that this is an extreme measure not to be used under normal simulations. If the timestep still collapses, there is a discontinuity in one of the models in the circuit. This could be a device model or a Verilog-A model. If it is necessary to set *lteratio* very large to get the simulation to complete, you must also set *maxstep* small enough to preserve the accuracy of the simulation.

itres

Use the default for this option.

Itres controls the precision (number of digits solved for) in the first Newton iteration. The default is 1e-4 which causes four digits of resolution in the first Newton iteration. Subsequent iterations are solved with more precision.

lnsolver

Leave this option at the default. At the end of each qpss iteration, a large matrix is formed and solved to calculate the new starting value in the shooting windows. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases with increasing iterations. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

You need to have a good understanding of the linear algebra theory to understand the differences between these methods.

- *Gmres* is the Generalized Minimum RESidual method.
- *Qmr* is the Quasi-Minimal Residual method.
- *Bicgstab* is the STABilized BI-Conjugate Gradient method.
- *Resgmres* is the REStarted Generalized Minimal Residual method.

resgmrescycle

Leave this option at the default. For the *resgmres* linear solver, there are several different options.

inexactnewton

In the simulation time between iterations, an estimate is made for the starting point for the next iteration. *inexactNewton* causes an inexact solution for the matrix that calculates the next starting point to be calculated. In some cases, solving the matrix exactly on the first iteration can cause the iterative solver to need many iterations to achieve convergence. Setting i*nexactNewton* causes a different series of iterations, which may speed up the simulation, or alternatively, allow the simulation to converge. The inexactness is reduced as the iterations progress. Try this option when you have convergence difficulties.

The following options are in the *Output* tab.

annotate

This option controls the level of detail in the output log. The default is *status*. No detail is provided when you select *no*, while more details are provided as you move towards right with steps option providing maximum detail.

save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved.

nestlvl

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* must be an integer.

compression

Normally, this should not be used for RF simulation. It is not digital compression. This option applies only in the tstab interval of the simulation. For RF simulation, where the input is sinusoidal, the size of the results file will normally double. It is useful only for circuits that are predominantly square wave.

oppoint

Normally, this should be left at the default of *rawfile*. It controls where the time-zero operating point solution should be saved to.

skipstart

Normally, this is not used for RF simulation unless there is a very long tstab. It is a way of reducing the amount of data in the output file from the tstab interval by not writing many of the timepoints to the output file. *skipstart* controls the simulation time where the skipping is to start. The default is 0 (zero).

If you want to save all the data for the first 5% of tstab, set skipstart to 0.05 * Tstab time. This will save all the data in the first part of the startup interval in tstab, and then begin reducing the amount of data sent to the output file.

skipstop

Normally, this is not used for RF simulation unless there is a very long tstab. This specifies the simulation time in the tstab interval where the skipping is to stop. The default is the stop time in the tstab interval.

If you want to save all the data for the last 5% of tstab, set *skipstart* to 0.95 * Tstab time. This will save all the data in the last part of the startup interval in tstab. This allows full resolution at the end of the startup interval.

skipcount

If *skipstart* and *skipstop* are set, either *skipcount* or *strobeperiod* should be set, but not both. *skipcount* saves one out of every skipcount points to the output data file. A *skipcount* of 3 saves one, then skips two points. Skipcount of 10 saves one in 10 timepoints to the output file. The default is 1, which saves every point.

strobeperiod

If *skipstart* and *skipstop* are set, either *skipcount* or *strobeperiod* should be set, but not both. *strobeperiod* forces the simulation datapoints between *skipstart* and *skipstop* at the interval of *strobeperiod*.

strobedelay

strobedelay specifies a time after the skipstart point for the beginning of the strobing. *strobedelay* must be between 0 (zero) and the time set in *strobeperiod*.

The following options are in the *Reuse* tab.

write

write specifies a filename in which to write the DC solution used as the first timepoint. If the name does not start with slash (/), the file will be written in the *netlist* directory. To determine the netlist directory location, select *Setup - Simulator/Directory/Host* in ADE. The *Project Directory* field lists the location of the simulation directory. From this directory, the netlist directory is in <`Circuit Name`>/<`simulator name`>/*<schematic or config>/netlist*.

writefinal

writefinal writes the final qpss timepoint from the shooting window (not the final timepoint of the tstab interval) to the filename specified. See the description for *write* for the location.

swapfile

During the shooting intervals, the solution matrices need to be saved in order to calculate the settled voltages of every node for the next iteration. The solution matrices are the internal solution for all the simultaneous equations for the circuit. The solution vector is also saved, and these are the solution voltages and currents. Normally, this data is kept in memory which makes the Spectre process get larger, starting with the first Newton iteration. On 32-bit systems, a maximum process-size of slightly less than 4GB limits the size of the circuit that can be simulated. Even on 64-bit systems, there is a limit set by the amount of memory that is installed on the system that is running Spectre. If the process gets larger than the installed memory, then the process starts swapping to the disk.

Swapping that is done by the operating system is inefficient because there is no logical relationship for the swapped memory pages. Setting swapfile to a name causes Spectre to write all the solution matrices to the disk in sequential cylinders so that the read time from the disk is faster than using swapping by the operating system. Disk accesses are inherently much slower than memory accesses, however, using swapfile is considerably faster than swapping in the OS. Using swapfile reduces the size of the Spectre process so that it fits in the memory installed in the machine, or allows larger circuits to be simulated. There is a large amount of data that typically is written to the disk. Because of this, the best choice is to write the data in a disk that is local to the machine. If the data is written over a high-latency network, it might actually be better to use the local swap space. Only a trial can determine which is better.

writeqpss

This option specifies that the full internal state of the qpss analysis be written out to the file specified in the option.

readqpss

This option specifies the file to be read in as a starting point for the next qpss analysis. If nothing is changed in the circuit, then only one Newton iteration is performed and small-signal analyses can be run with a much faster time for the qpss analysis. Small changes can be made to the circuit or to the analysis options as long as the topology of the circuit stays the same. Changes introduce a discontinuity at the beginning of the qpss analysis, which might be large enough to cause the qpss analysis to not converge.

saveclock

saveclock saves the state of the simulation in the tstab interval at the time interval in seconds specified by *saveclock*. When the clock time has passed in the tstab interval, the file specified by the option *savefile* is created. When subsequent clock time intervals have passed, the file is overwritten. Use only one of the save options at a time.

saveperiod

saveperiod saves the state of the simulation in the tstab interval at the simulation time interval in seconds specified by *saveperiod*. When the simulation time has passed in the tstab interval, the file specified by the option savefile is created. When subsequent simulation time intervals have passed, the file is overwritten. Use only one of the save options at a time.

savetime

savetime is a list of times in the tstab interval where the state of the simulation is written out. The list should be specified with spaces between the entries. The information is written out with the filename specified in the *savefile* option with extensions for the time after that. Use only one of the save options at a time.

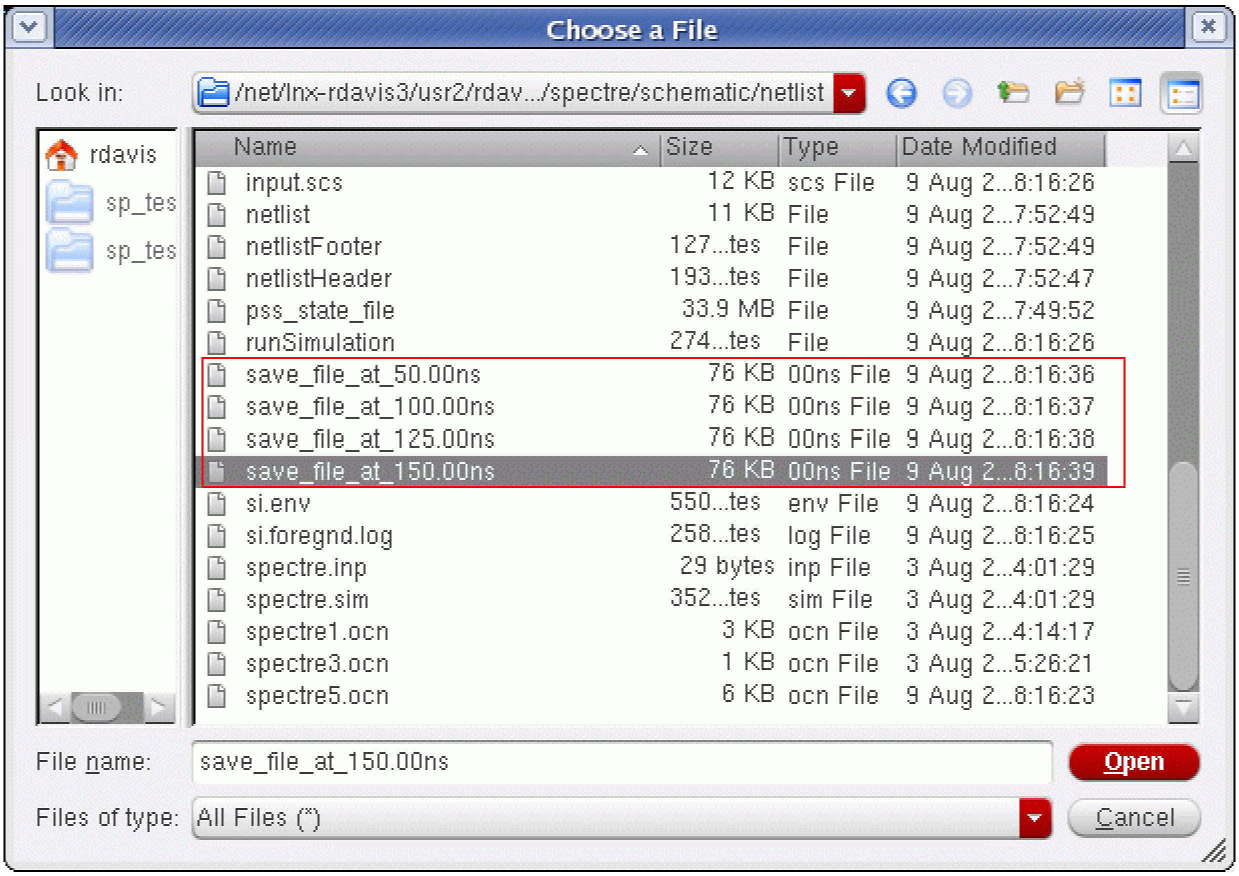
savefile

This is the file name to write out the tstab state information to. If you do not specify a filename that starts with the slash (/) character, the file is stored in the *netlist* directory. To locate the *netlist* directory, first select *Setup - Simulator/Directory/Host* menu in ADE, and read the path to the project directory. This is the path to the *simulation* directory. In the simulation directory, navigate to the <*circuit_name*>*/spectre/<schematic or config>/netlist* directory.

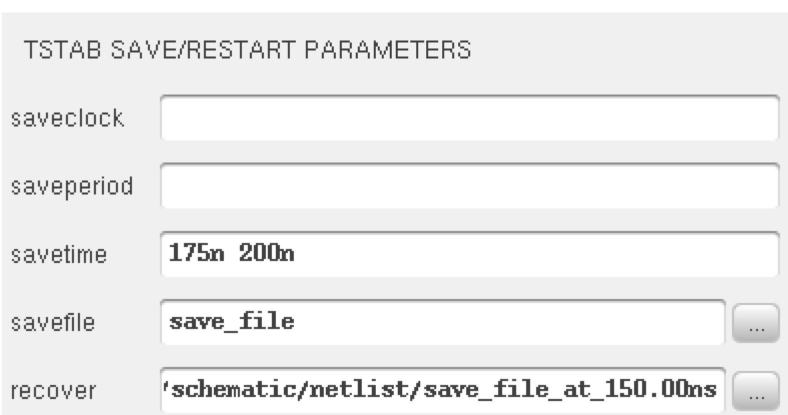
Do not specify a relative path name like *./save_file/run1*.

recover

recover specifies the file name to recover the tstab simulation from. If *saveperiod* or *saveclock* are used to make the savefile, then just the same name as specified in *savefile* is used. If *savetimes* has a list of times specified, then several files are created at the times specified in the list. Click (...) and browse to the netlist directory where the savefiles are shown.



Select one from the list, and click *Open*. This adds the full path to the *recover* option, as shown below. Also note that in the example below, the *savefile* option is still set, and times after the restart times are specified. This is specifically allowed. As long as tstab is 200n or larger, the files at 175n and 200n will be added and can be reused later.



The following options are in the *Misc* tab.

step

Leave this option at the default. All the nodes are checked for curvature in the local truncation error (Numerical integration error) check. If that node has a capacitor or inductor connected to it, the timestep should be controlled to reduce the numerical integration error. If the node does not have a capacitor or inductor connected to it and it has a lot of curvature, the timestep can only be reduced to the value specified in the *step* option. The default is the simulation interval divided by 1000.

maxstep

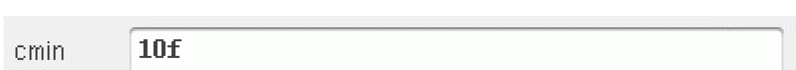
In qpss shooting, *maxstep* is observed in the tstab interval and in each of the shooting windows. Normally, *maxstep* should be left at the default and either setting *errpreset* or changing *reltol* is used to control the accuracy. In some cases, extreme accuracy is needed, and this is accomplished by setting a small maxstep value. A small constant timestep produces maximum accuracy at the cost of runtime and memory consumption.

readns

This specifies an ascii file with the same format as an ic file that is used as nodesets for the time-zero DC solution. Nodesets do not force a voltage to be held for the time-zero solution. Instead, they are a way of speeding up the time-zero calculation. As a suggestion, set the *write* option and the *readns* option to the same filename. The *write* option writes the time-zero solution to a file. When this is used as a starting point, many fewer iterations are needed for the time-zero point to converge.

cmin

cmin can be used to improve convergence in the tstab and shooting intervals. If a value is set for *cmin*, a capacitor with this value is added to every node with the other terminal of the capacitor connected to the global ground node. If a 10f to 50f capacitor is added, this prevents instantaneous changes from occurring from timepoint to timepoint, thus improving the convergence at the cost of adding non-physical capacitors to the circuit. An example with 10 femtoFarads is shown below. Note that if 10 is entered, a 10 Farad capacitor is added from every node to ground. Remember the multiplier in the entry. An example is shown below.



tstart

Normally, this is not used. *tstart* specifies the starting time for the tstab interval and defaults to 0 (zero). A negative, 0, or positive *tstart* is allowed.

additionalParams

This is a field for `<option_name>=value` statements. Multiple `<option_name>=value` statements are allowed with a space between them. Generally, this is used to unlock the beta features, however, there is one case where this field might be useful for an option that is not in the GUI.

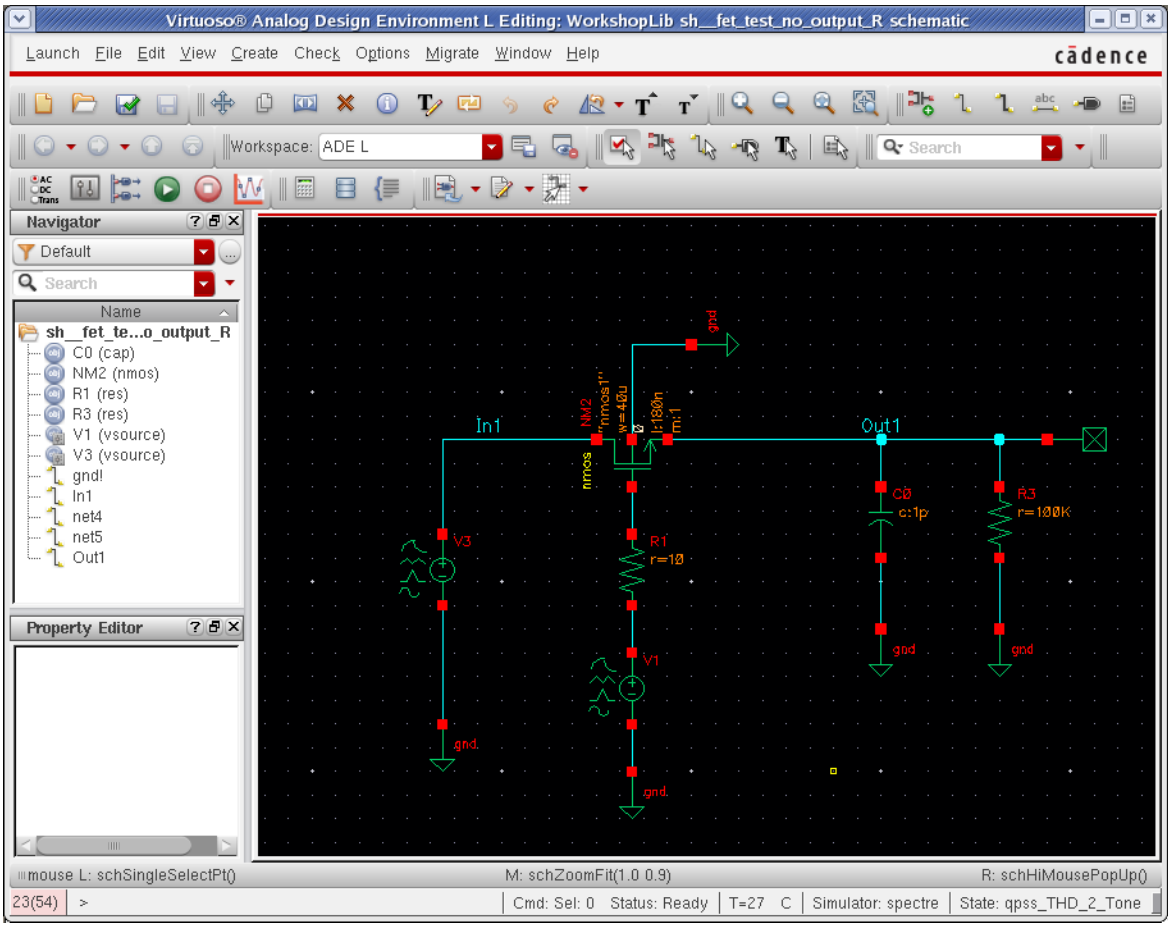
In some cases, either in the tstab interval or in one or more of the shooting windows, the timestep collapses to near zero. The symptom is that the percent complete stays at the same value for many reporting intervals. In this case, there might be a discontinuity in one of the models.

In transient simulation, two things can reduce the timestep; having too much numerical integration error or taking too many iterations at a single timestep can cause the timestep to be cut. Spectre does not report what is causing the timestep to become small. One way to eliminate the possibility of numerical integration error (which is called local truncation error in the simulator) as the cause of reducing the timestep is to set *lteratio* to 1e9. The disadvantage of this is that it disables the normal method of timestep control, therefore, *maxstep* must be set to maintain accuracy.

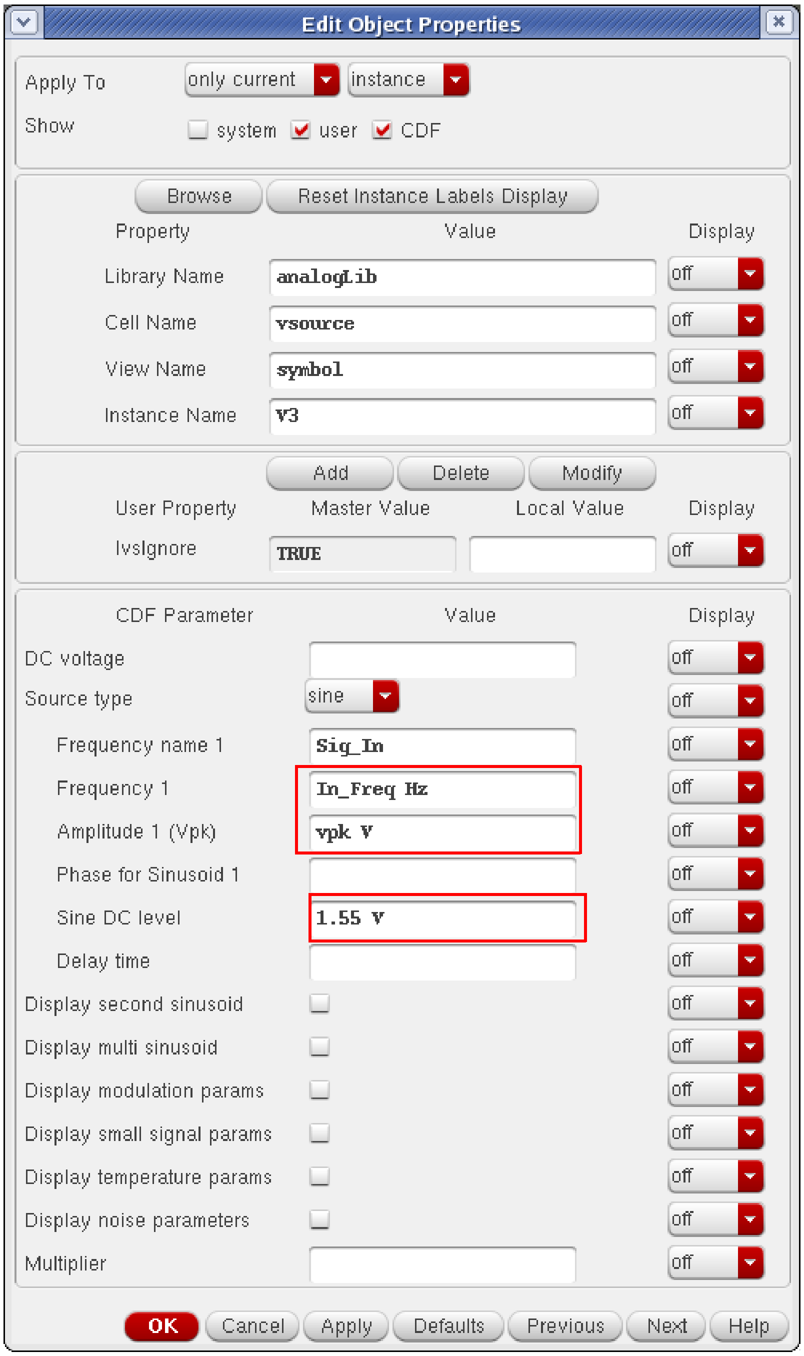
Another way to accomplish this is to set *lteminstep* to a value of about the stop time divided by 1e5. This allows the normal method of timestep control, but if for some reason the numerical integration error wants to cut the timestep too small, the error is ignored. *lteminstep* is the way to specify this. To do this in the ADE, type `lteminstep=<Stop_Time>/1e5`. If you use `lteratio=1e9`, or if you set *lteminstep* and you still have very small timesteps, try increasing *maxiters* in the *Convergence* tab to between 40 and 100. When *lteratio* or *lteminstep* is set and you still have timestep problems, there is likely a discontinuity in either a device model or in a Verilog-A model. An example is shown below.

Example

Consider a sample and hold circuit below.

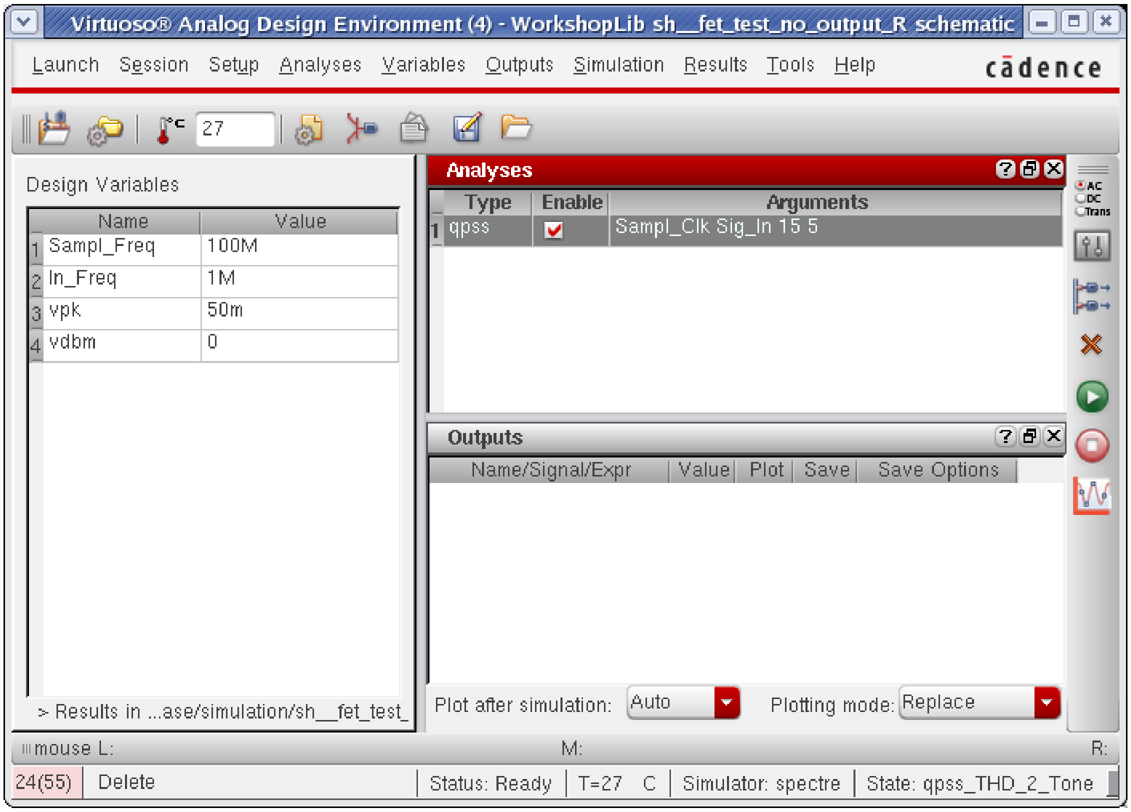


1. The input signal properties are shown below.

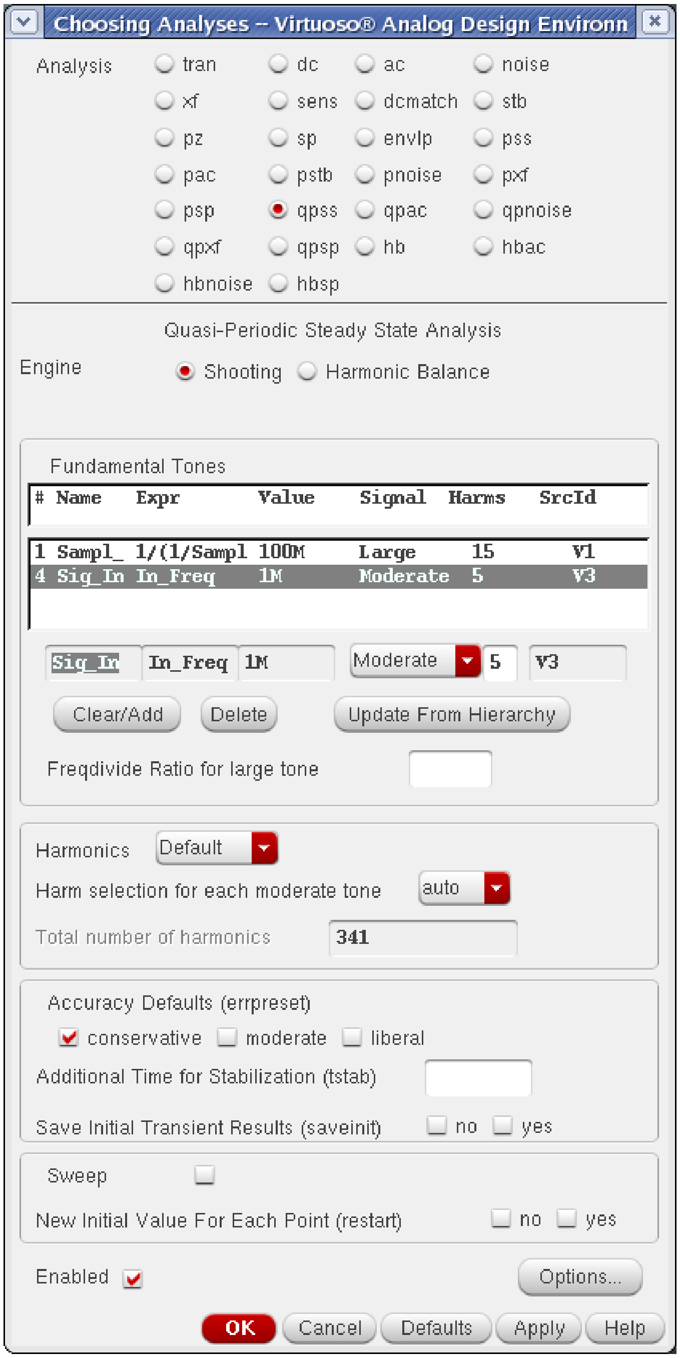


2. The input frequency and amplitude terms are set to variable names so that the amplitude and frequency can be easily set in ADE. Also, note that the sinusoid is centered at 1.55 volts. This is because the simple circuit above runs between 0 volts and 3.3 volts.

3. Next, the variable values are set in ADE. The sample clock is 100MHz, and the signal is at 1MHz.

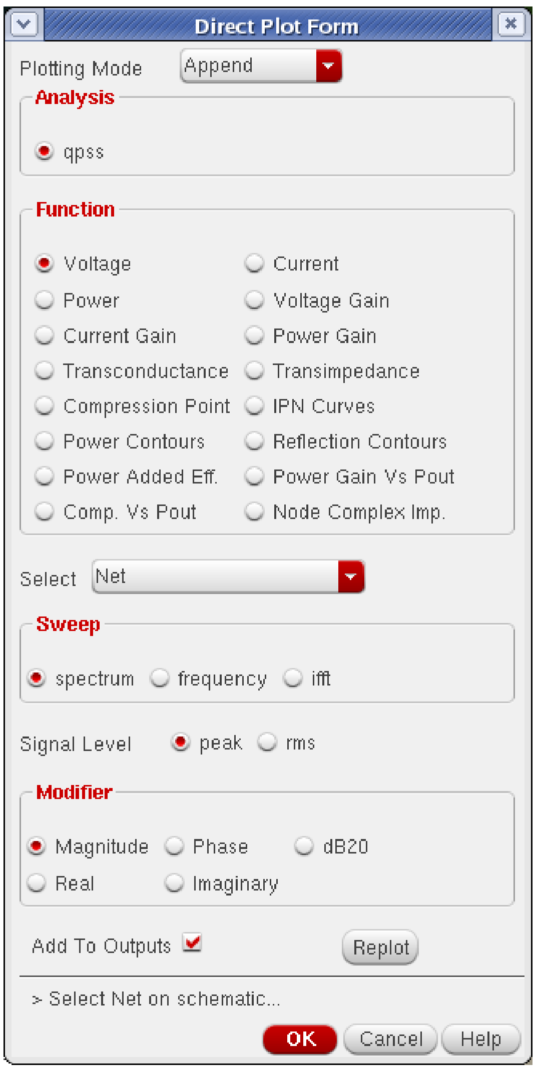


4. Now, the qpss analysis is set up, as shown below.



5. All the sources in the circuit are listed in the table at the top. The sample clock is set to large for two reasons. First, it causes the most distortion in the circuit. Second, it is the nonsinusoidal source. The input signal is set to moderate. 15 sample clock harmonics are set, and 5 harmonics for the input signal are set. Conservative is used because the distortion is likely to be small at low input amplitude.

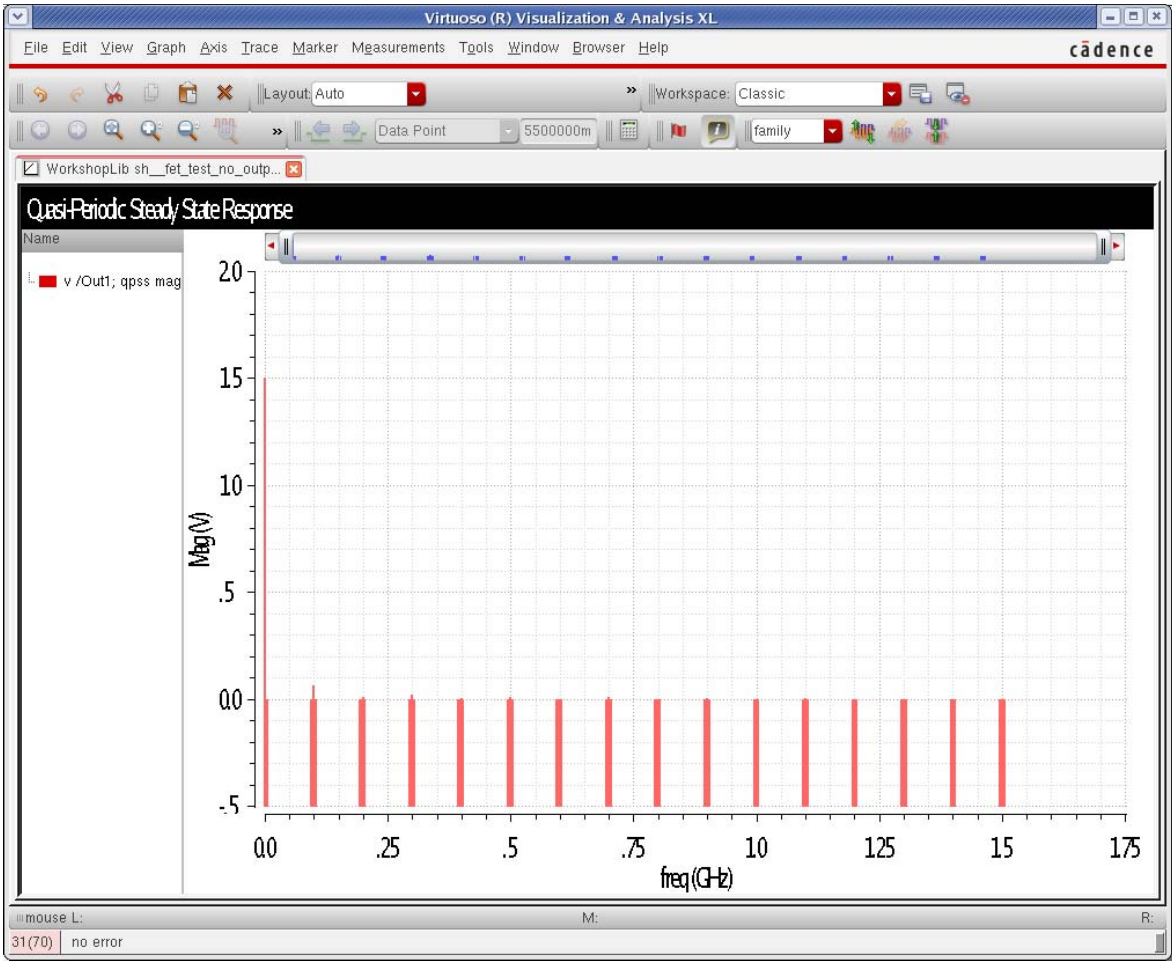
6. Now the simulation is started. When the run is complete, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* is displayed.



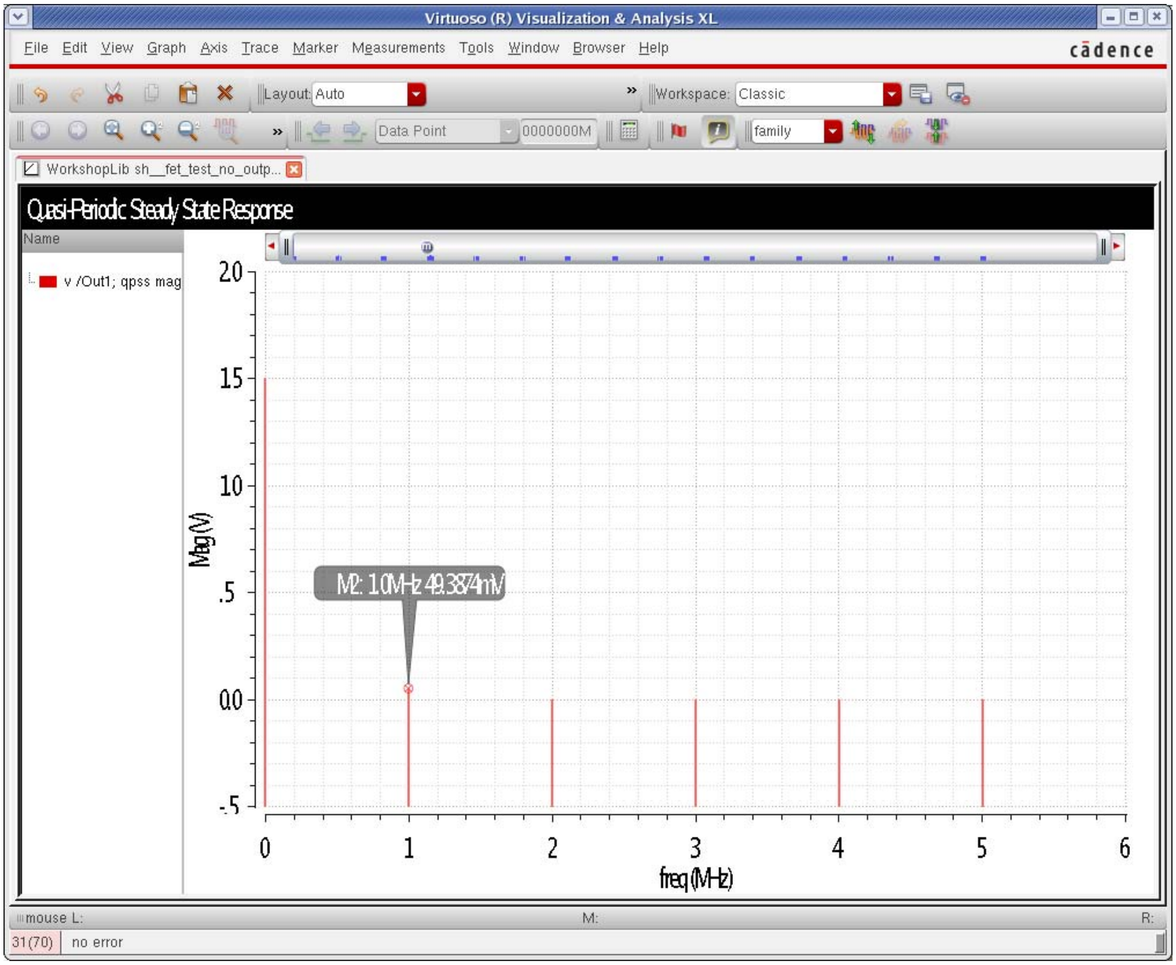
In the *Direct Plot Form*:

7. Select *qpss*. In this case, the voltage spectrum is plotted. Peak is used because it allows a direct comparison of input amplitude in volts peak to the output amplitude in volts peak. Add To Outputs is selected so that an expression gets added to the ADE outputs section.

8. The result is displayed in the waveform window.



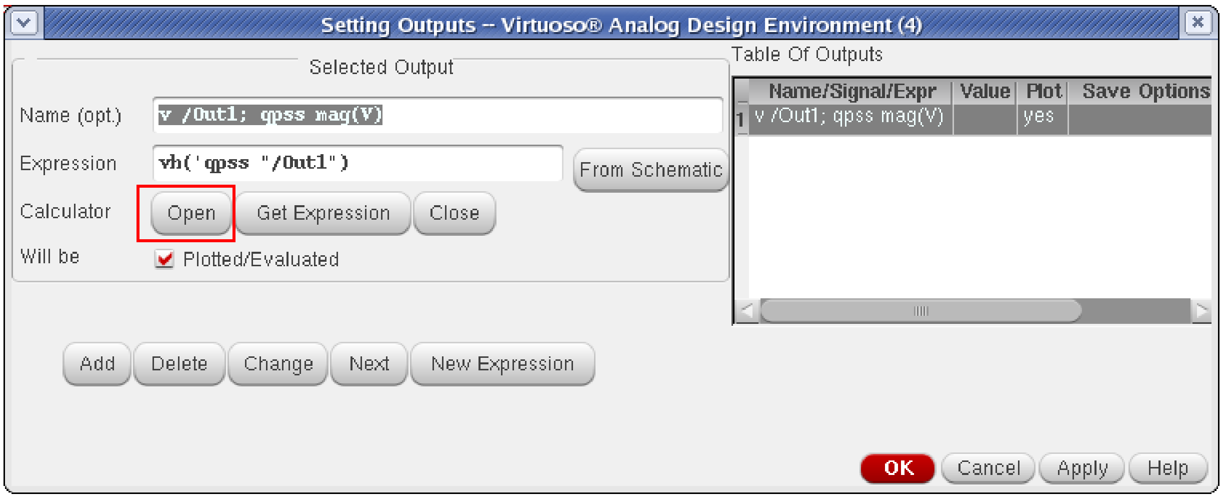
9. Since the output at low frequency is desired, the X Axis is scaled to show just the output terms and the harmonics. To place a marker, move your mouse cursor near the frequency point you want, and type m.



10. The output is near 50mv, as set in the input source.

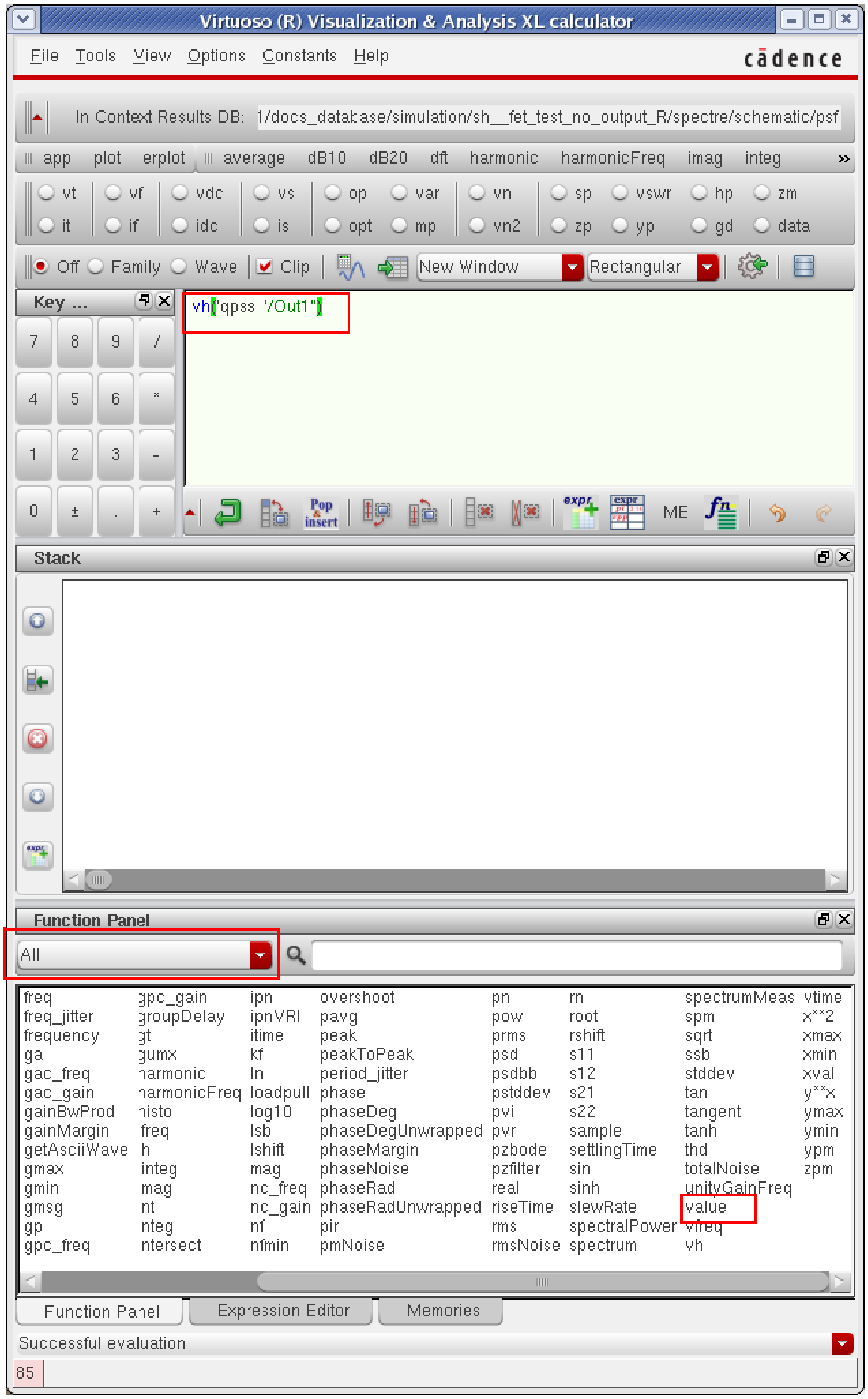
Add an ADE expression to calculate the amplitude of the output at the input frequency.

1. Double-click the expression in the outputs section. This displays the *Setting Outputs* window in ADE. The expression that was added from the *Direct Plot Form* is in the Expression line.



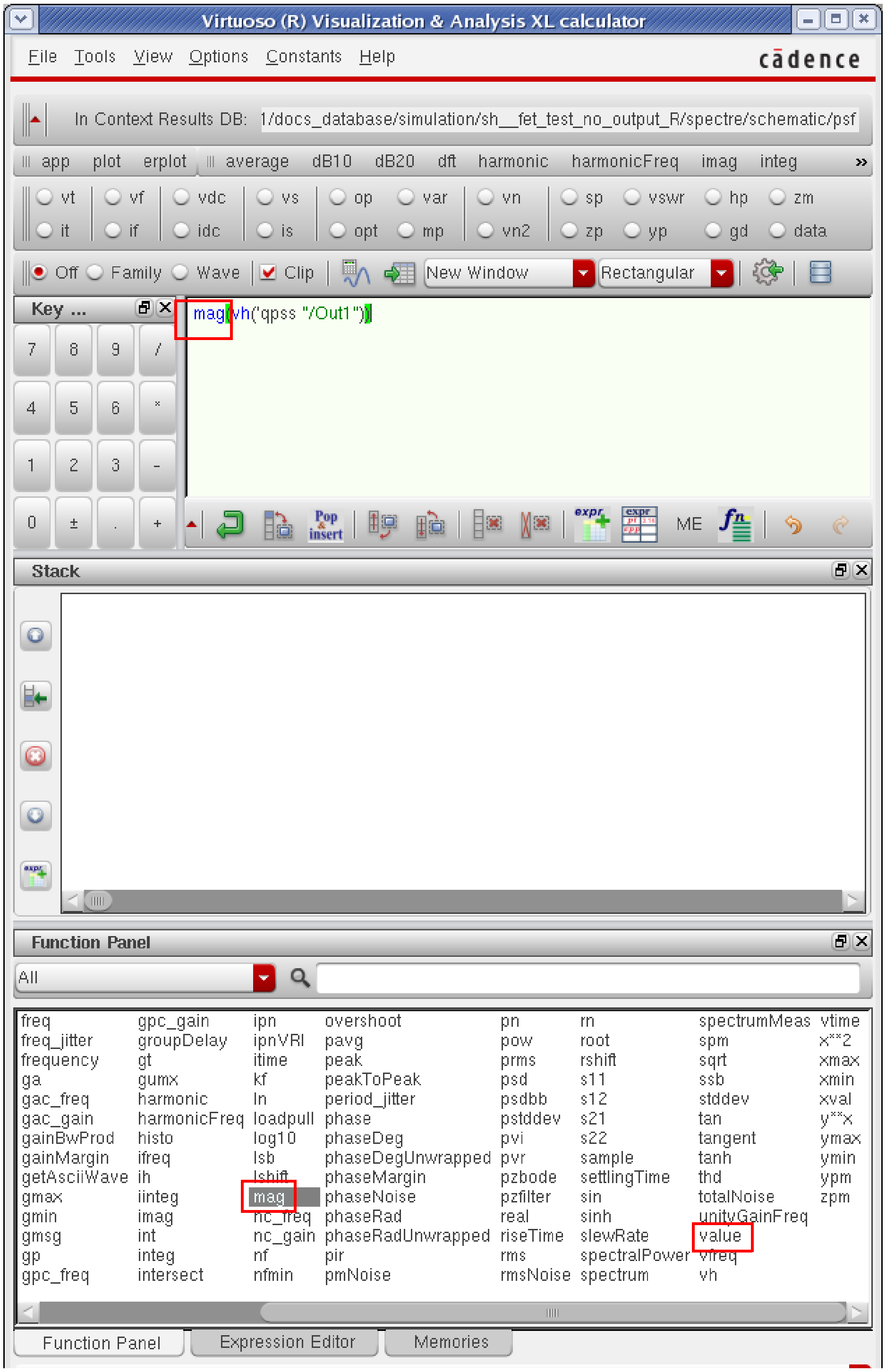
Note: Starting with the IC615 ISR12 release and later, the RF expressions are much simpler than they were in the past.

2. Click Open. This starts the calculator with the expression in the buffer. The calculator in the example below has been stretched vertically from the default size to show detail.

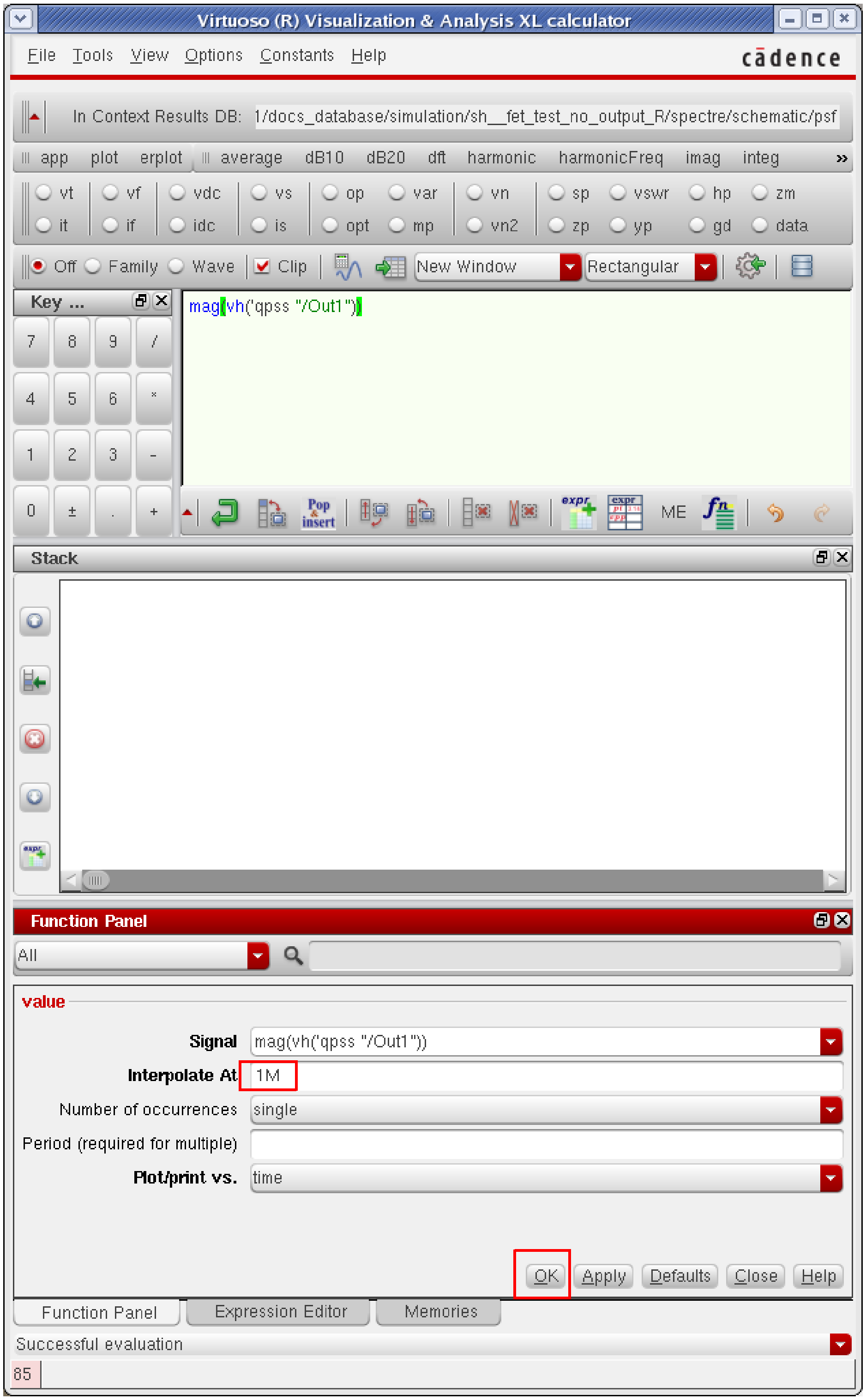


3. You may need to select *All* in the Function Panel to see all the functions.

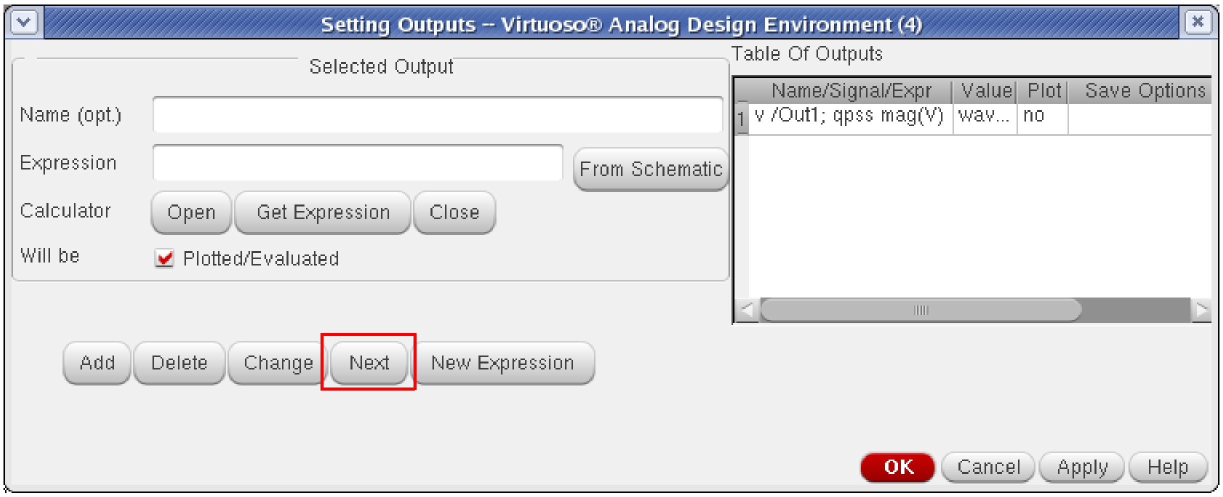
4. In the calculator, Use the *mag* function to convert the complex voltage to a scalar voltage. The assumption here is that the load is resistive, and any voltage regardless of phase will create power.
‘



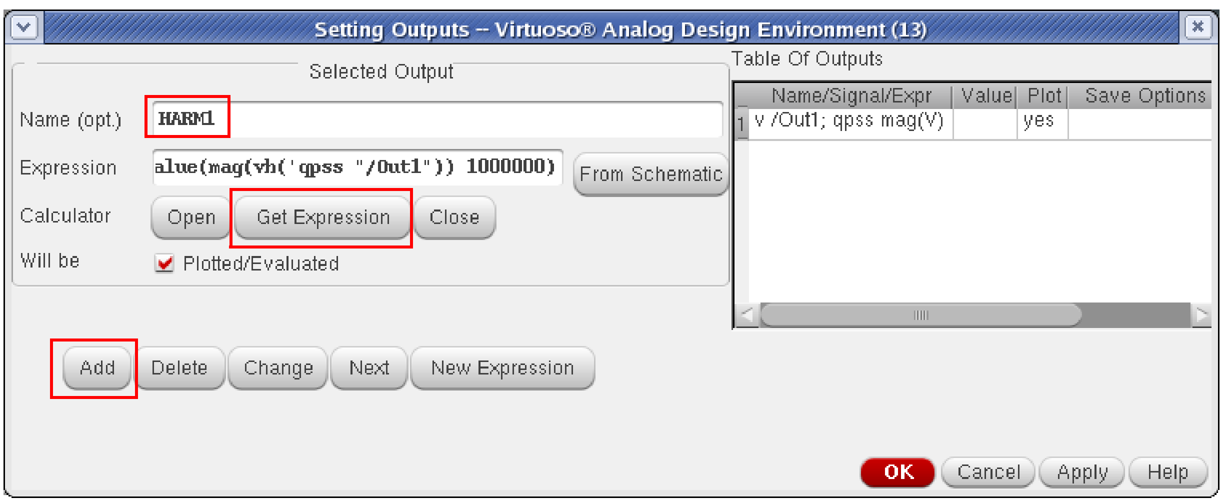
5. In the calculator, Use the *value* function to get the value at 1MHz and click OK.



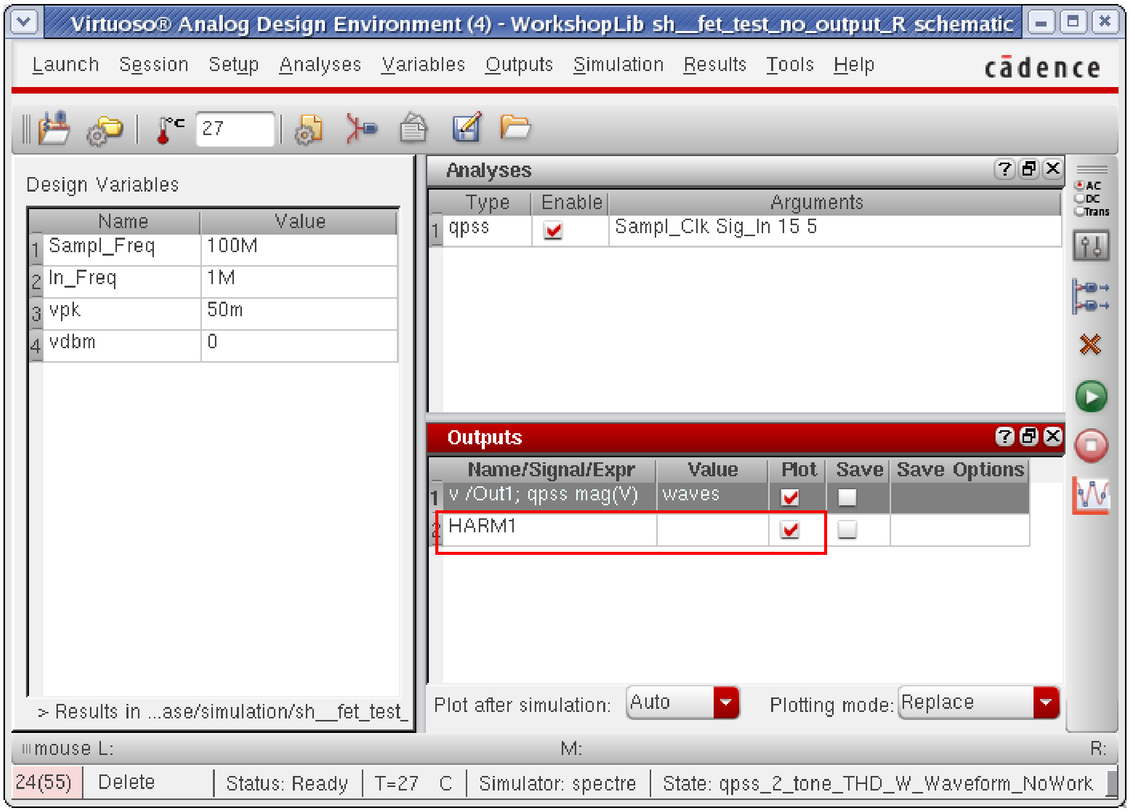
6. In the *Setting Outputs* window, click *Next*. The *Name* and *Expression* fields will be blank.



7. Click *Get Expression*. The *Expression* field will get the expression from the calculator. Type in a name in the *Name* field. This example uses *HARM1* as the name.

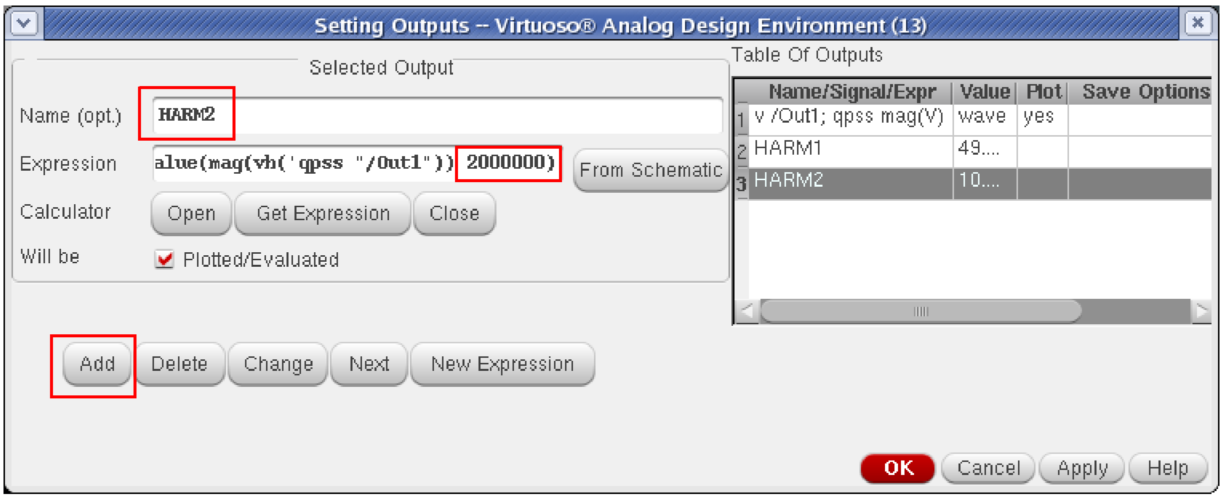


8. Click *Add*. This adds the expression to the Outputs section in ADE.



In a similar manner, add expressions for all 5 harmonics of the output

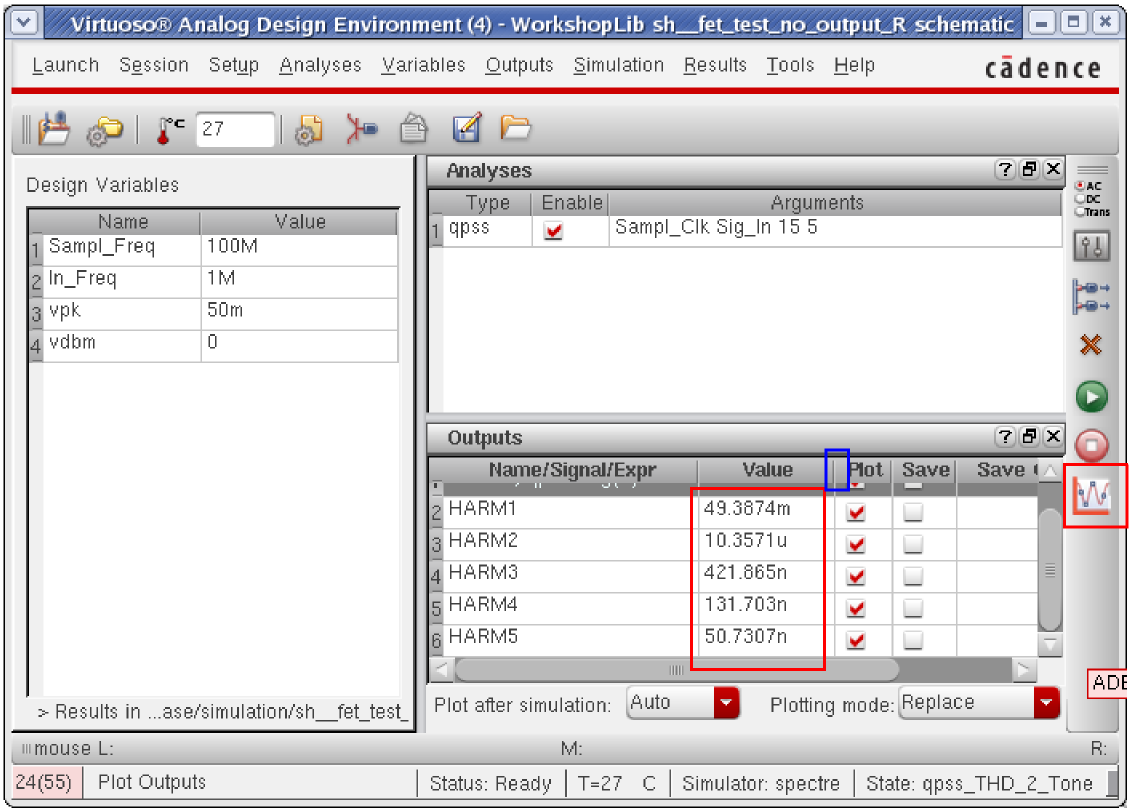
1. In the *Setting Outputs* window, the *Name* and *Expression* fields will be blank. Click *Get Expression*. Change 1M to 2M. Type *HARM2* in the *Name* field.



2. Click *Add*.

3. Continue until you have expressions for 1MHz through 5MHz. To verify that the expressions work, click the *plot Outputs* icon. This is the lower right icon in the ADE window. The value of each harmonic will appear in the ADE outputs section. You may need to adjust the size of the column. Use your mouse cursor on the vertical bar between

the legend *Value* and *Plot* as highlighted in the blue box below. Click and hold, drag to the desired place, and release the mouse button.

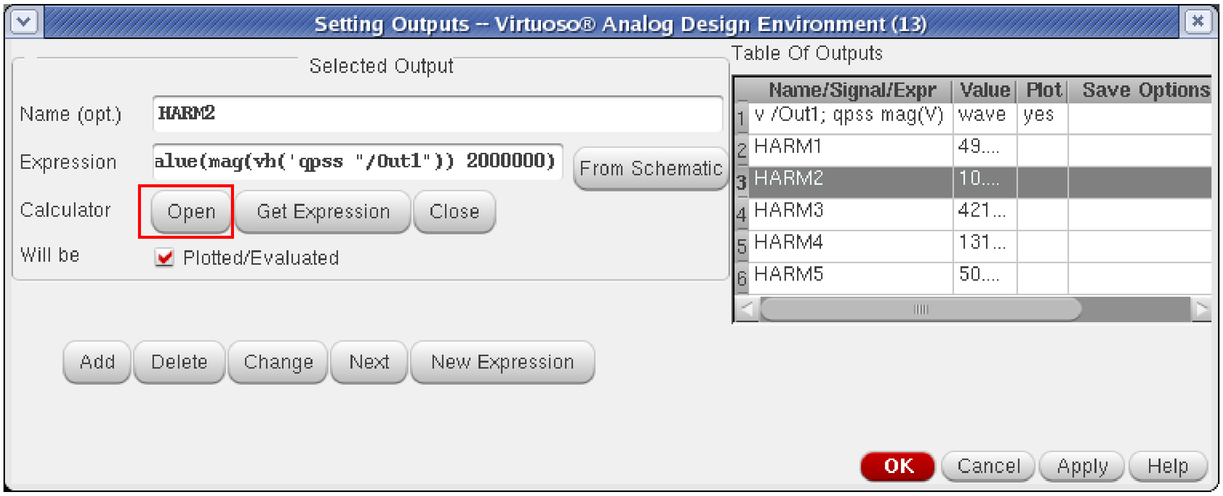


Now build a calculator expression to calculate THD.

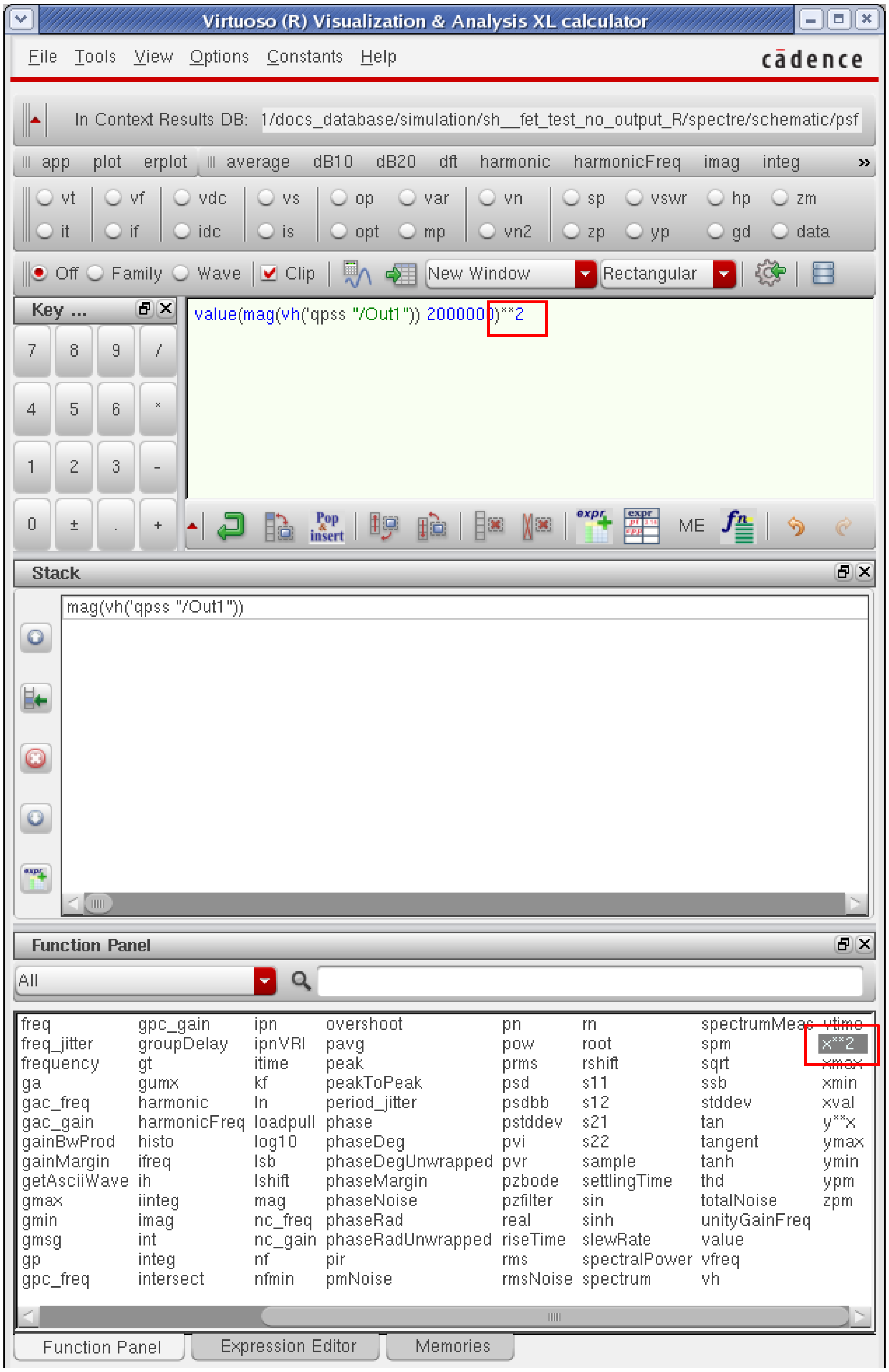
THD in percent is defined as: 100* square root (total power in the harmonics divided by the power in the fundamental).

First, build an expression to calculate the power in the harmonics.

1. Double-click the HARM2 line in the outputs section of the ADE. The *Setting Outputs* window is displayed.

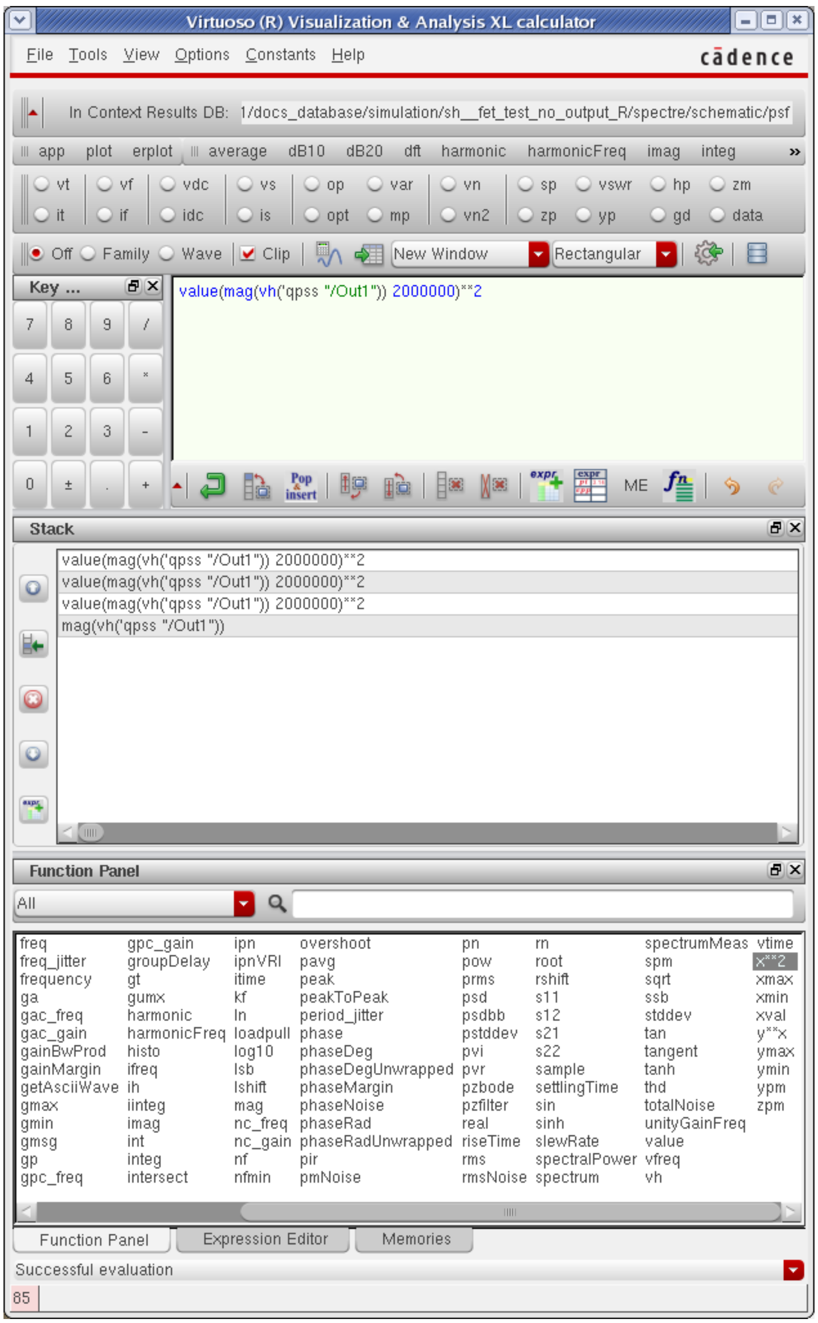


2. Click *Open* in the *Setting Outputs* window. This adds the expression into the calculator. In the calculator, select the *x**2* function. (Square the expression) This converts voltage to power.

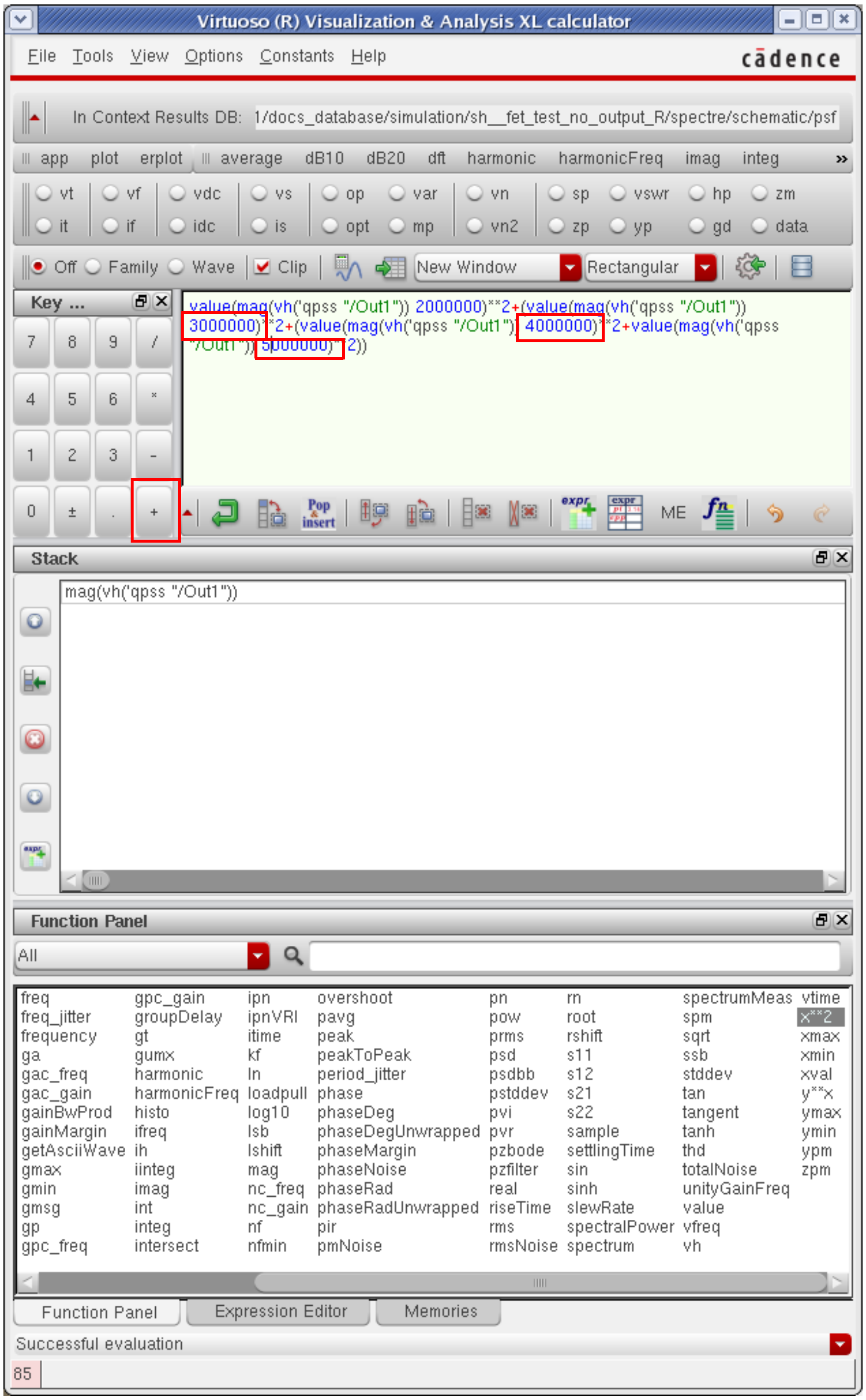


3. Press the <*Enter*> key three times. You may need to click *View - Show Stack* to view the stack. This is the beginning of adding the power in the higher harmonics.

Note: The number of times you press the <*Enter*> key will depend on the number of harmonics that are specified in the qpss analysis. Here, there are five total harmonics, the fundamental and four higher harmonics. The total number of higher harmonics is four. To get four total expressions, you press the <*Enter*> key three times.

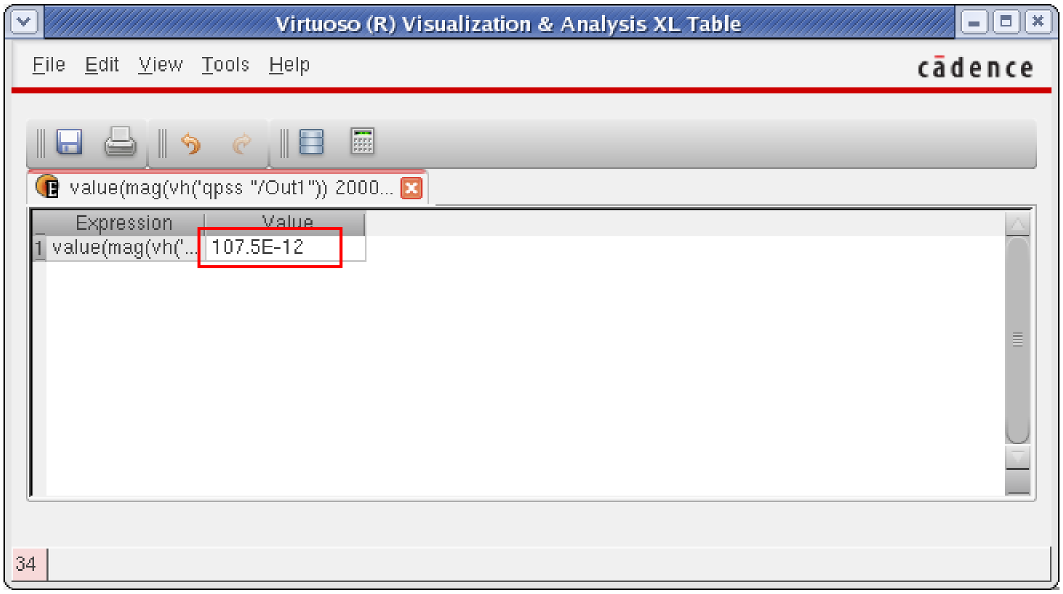


4. Click + (the plus sign) three times. Now change the frequencies in the buffer so you have 2M + 3M +4M +5M. This adds the power in the higher harmonics.

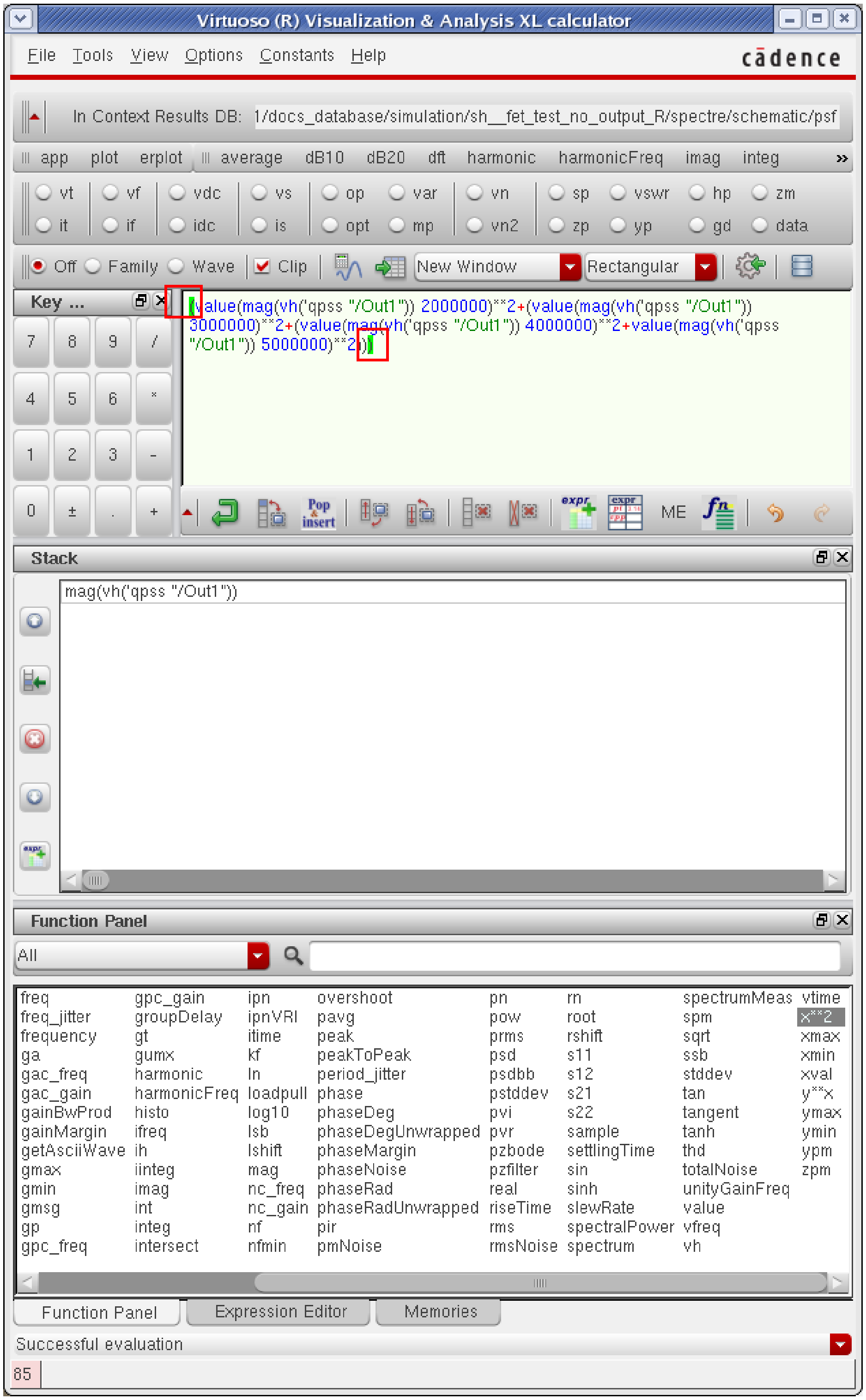


5. Verify that the expression is legal by clicking *Evaluate the buffer* and display the results in a table. This is an insurance step that is always recommended when you build calculator expressions. The icon for the table output is shown in the red box below.

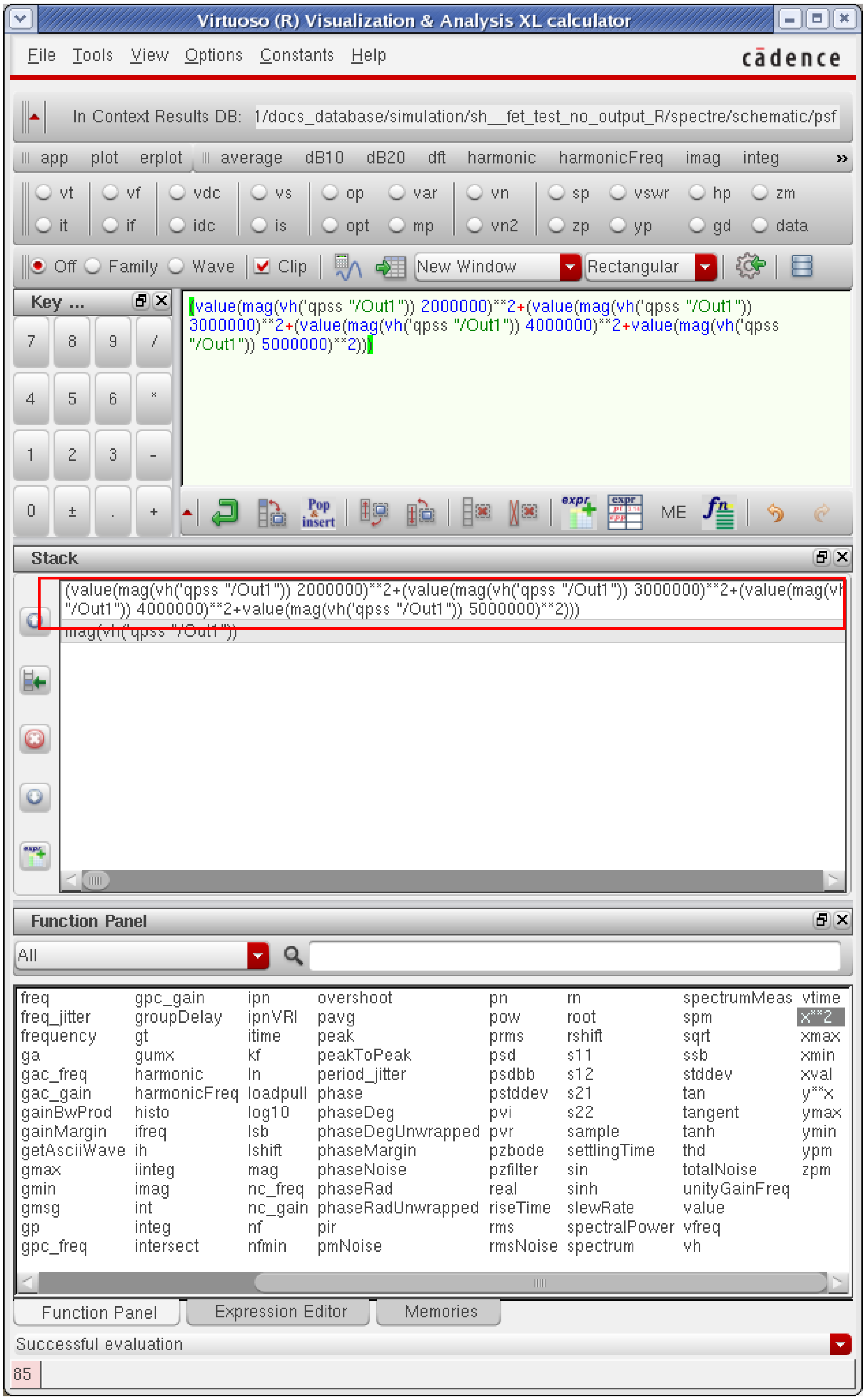
6. The value is shown below.



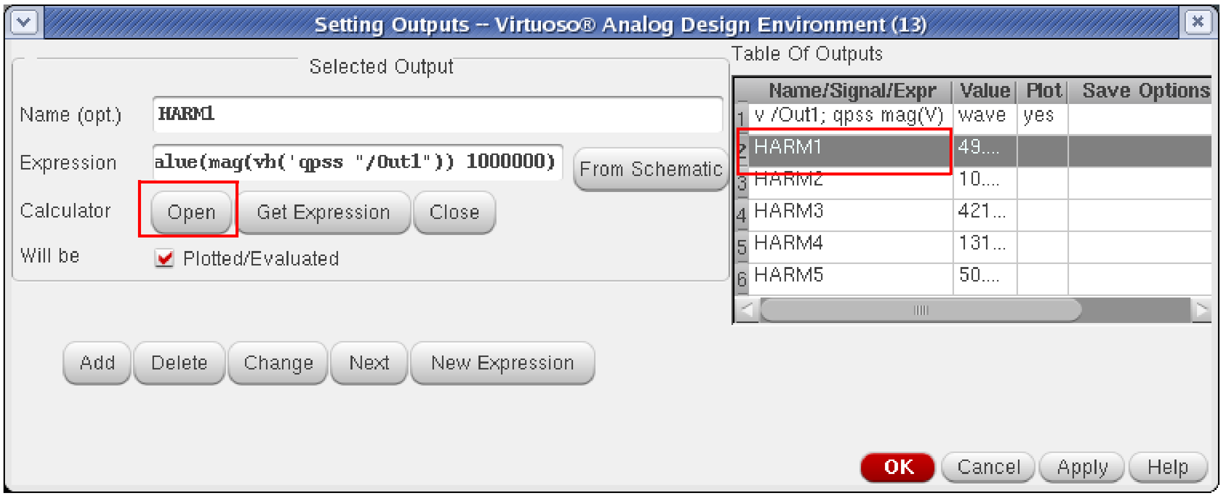
7. Add parentheses around the entire expression using the mouse and keyboard.



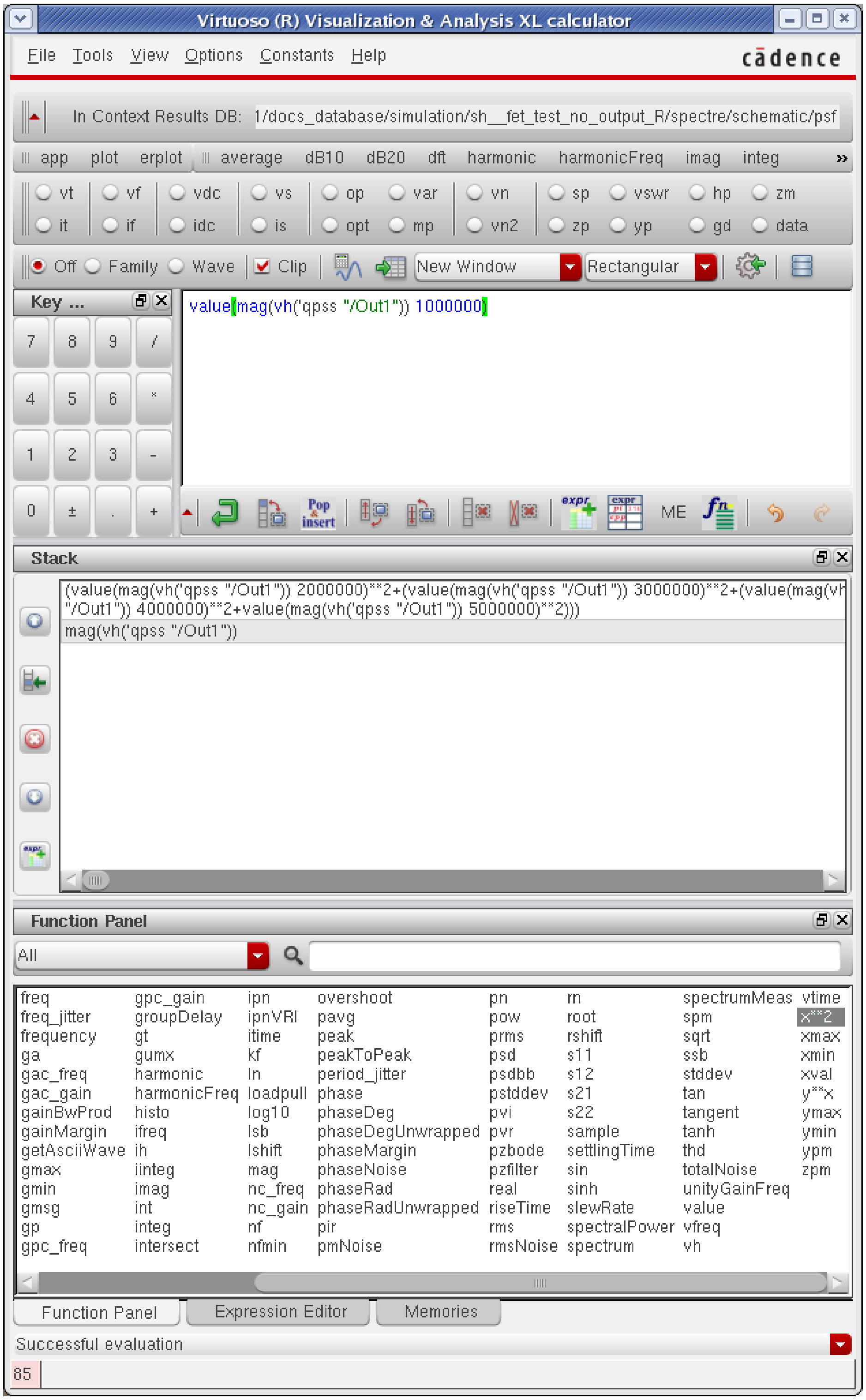
8. Press the <*Enter*> key on your keyboard. The expression is entered into the stack.



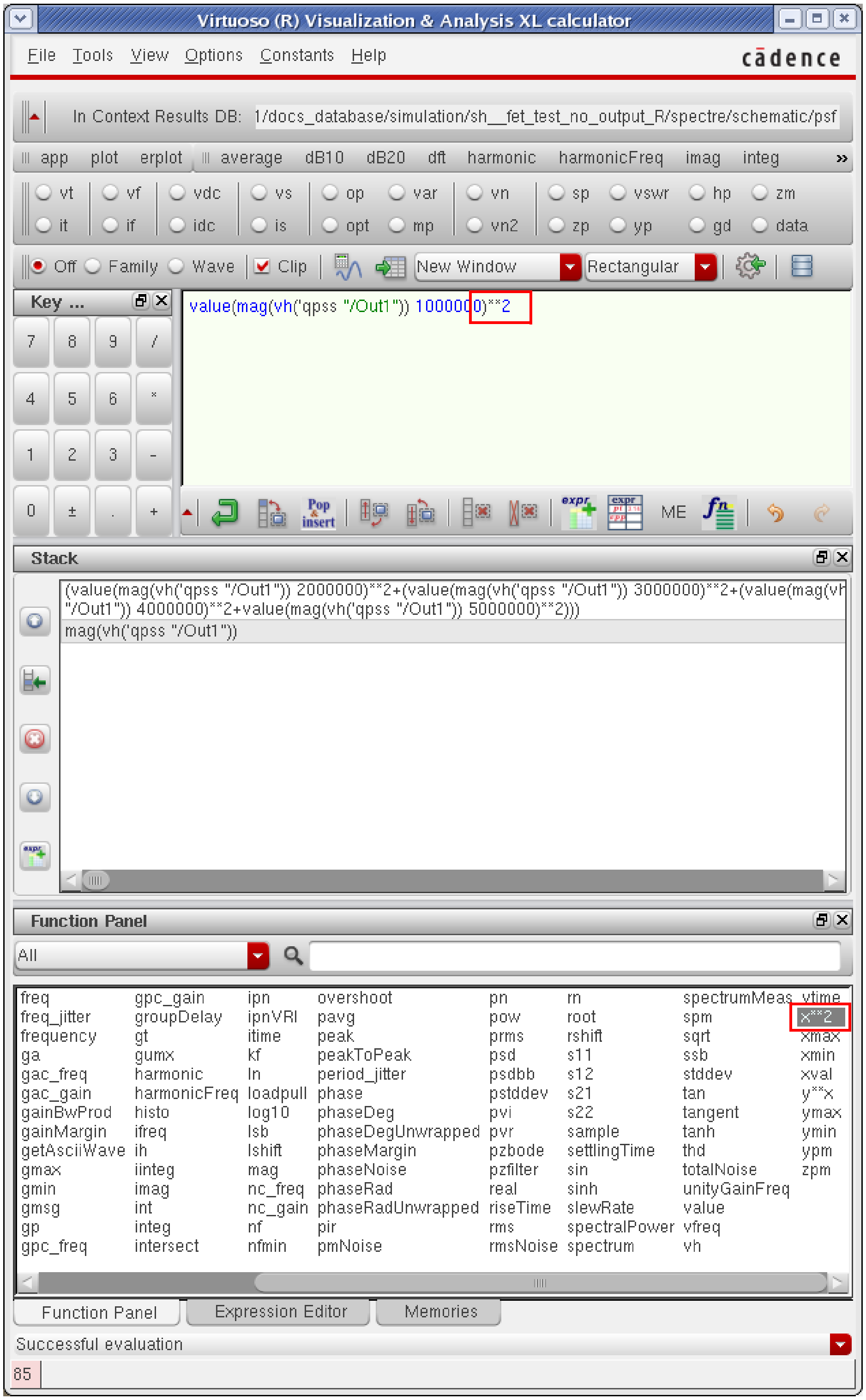
9. In the Setting outputs window, click *HARM1*. The expression for the first harmonic is shown.



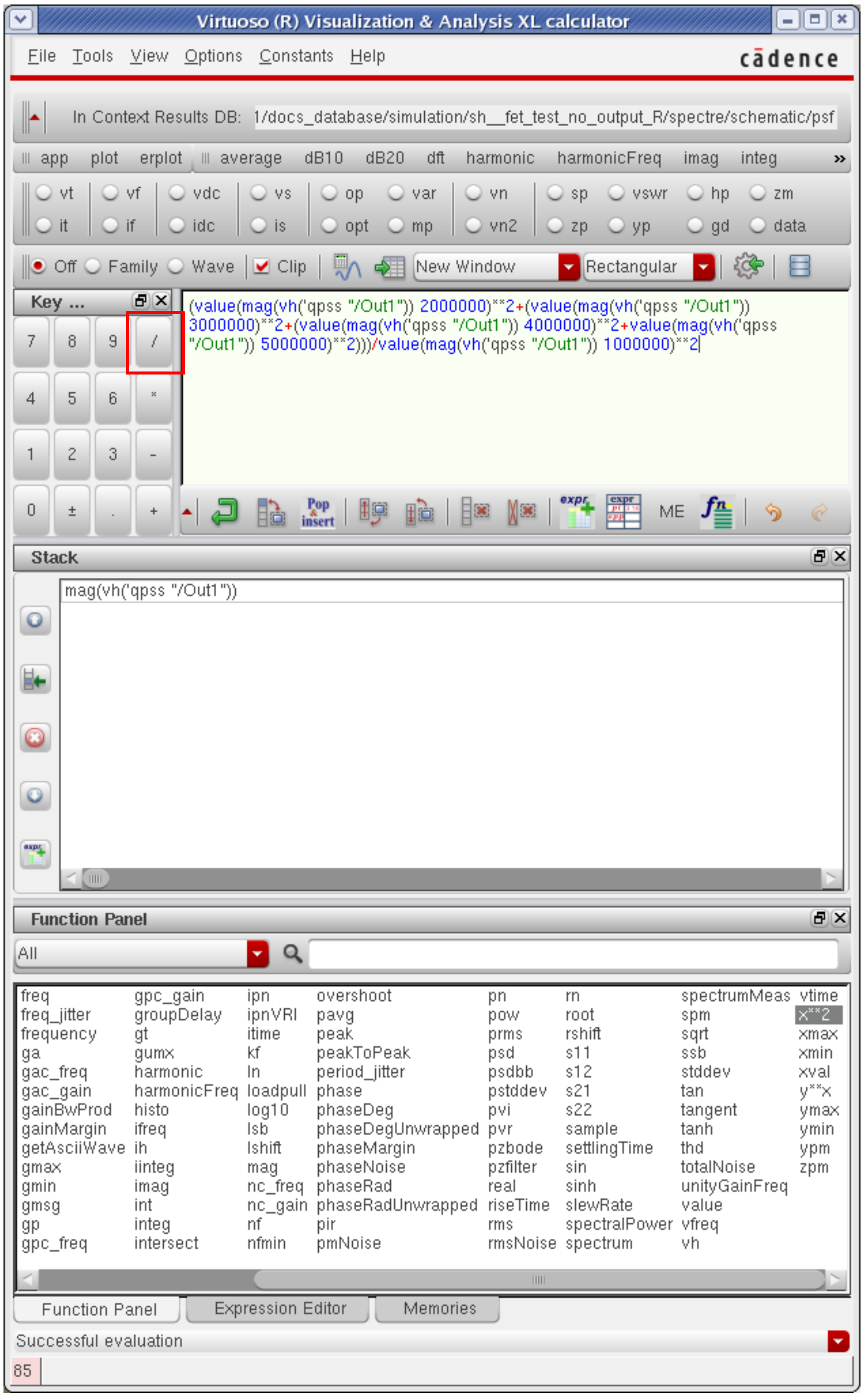
10. Click *Open*. This enters the expression into the calculator.



11. Click the X**2 special function. This calculates the power in the first harmonic.

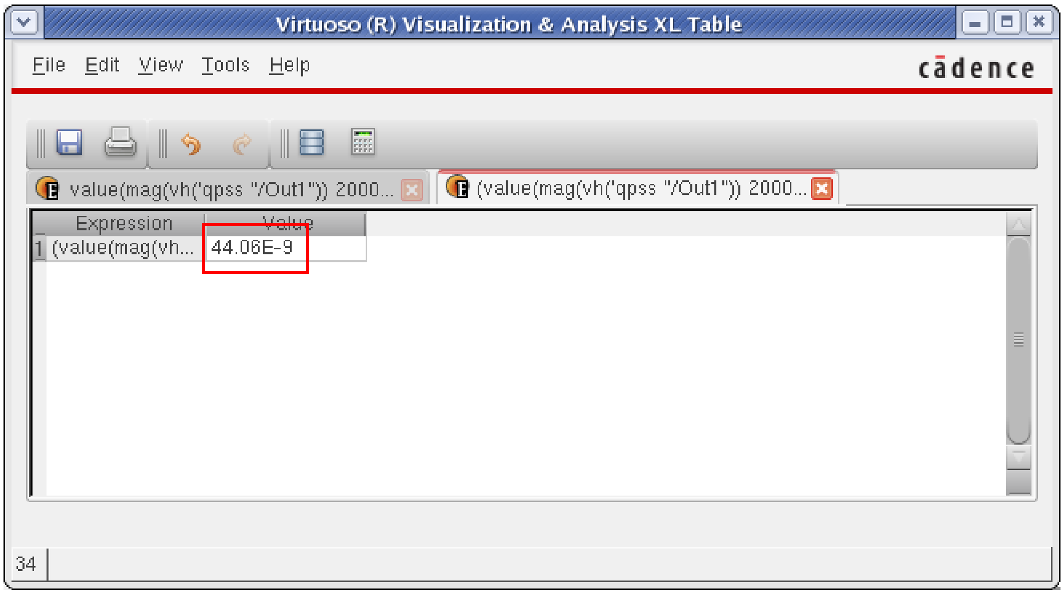


12. Select the / (divide by) key. This divides the power in the harmonics by the power in the fundamental.

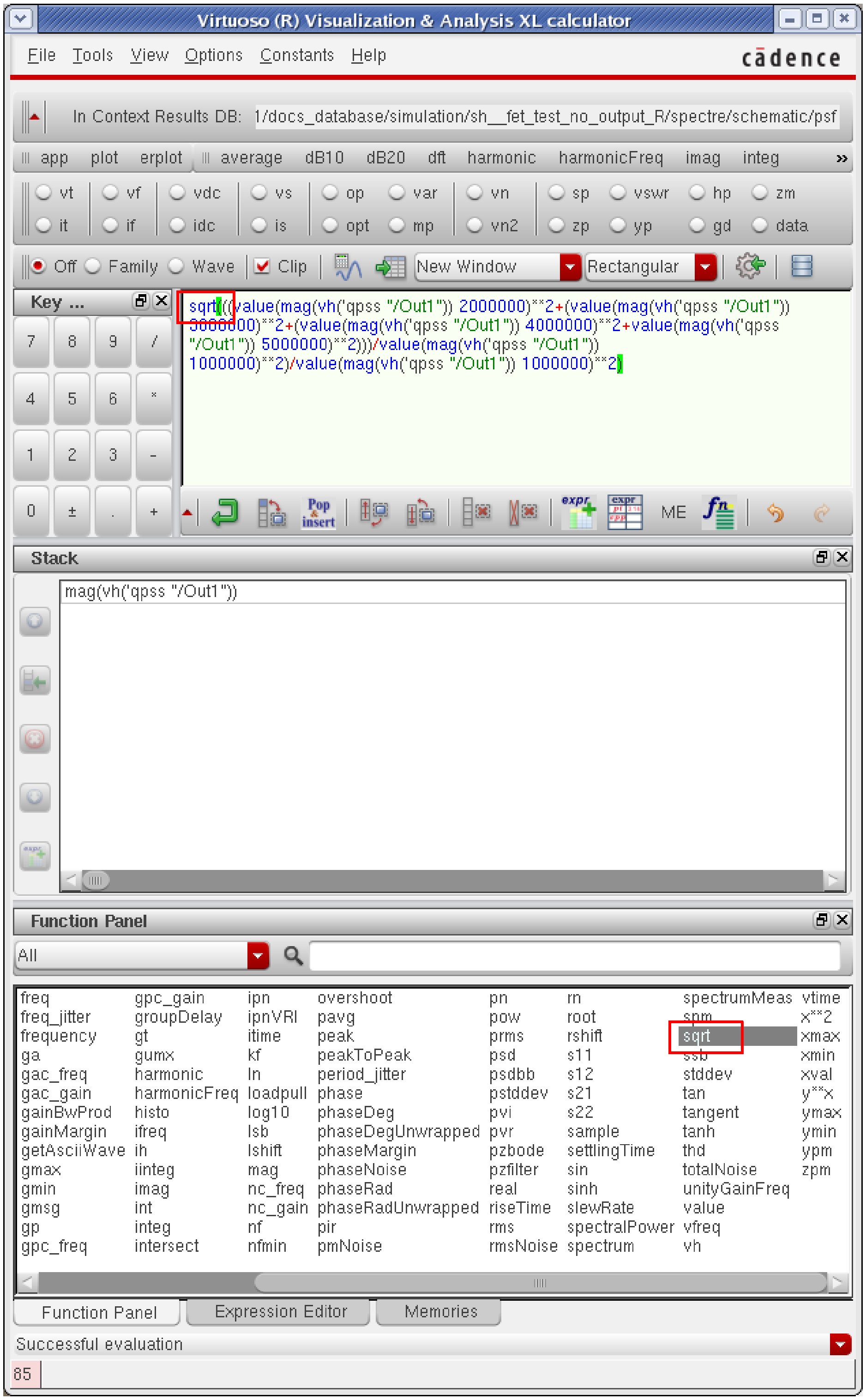


13. Verify that the expression is legal by clicking the *Evaluate the buffer and display the results in a table* button in the calculator.

14. The result is shown below.

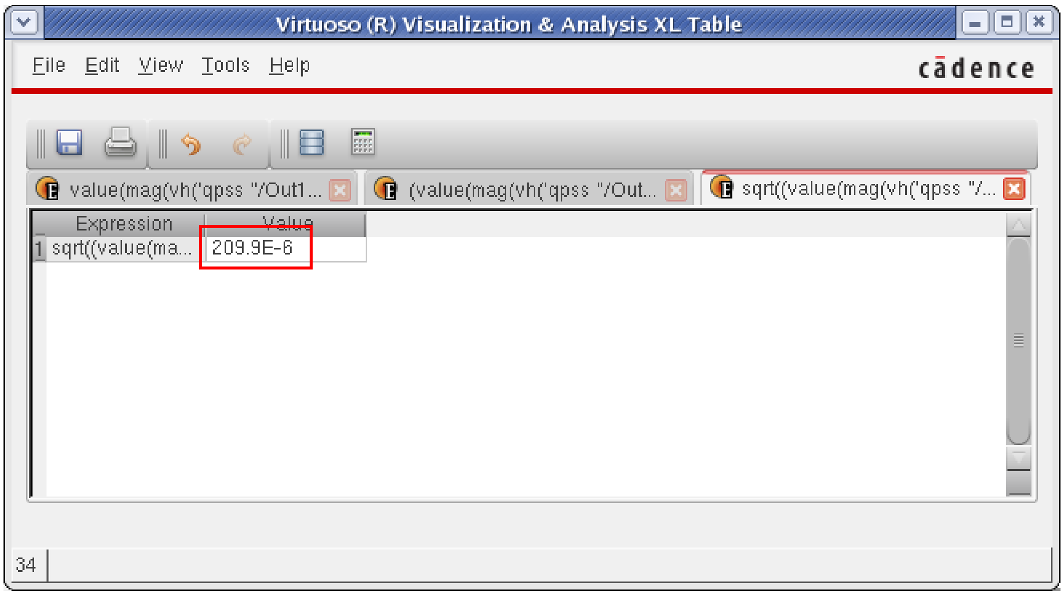


15. Select the sqrt (square root) special function.

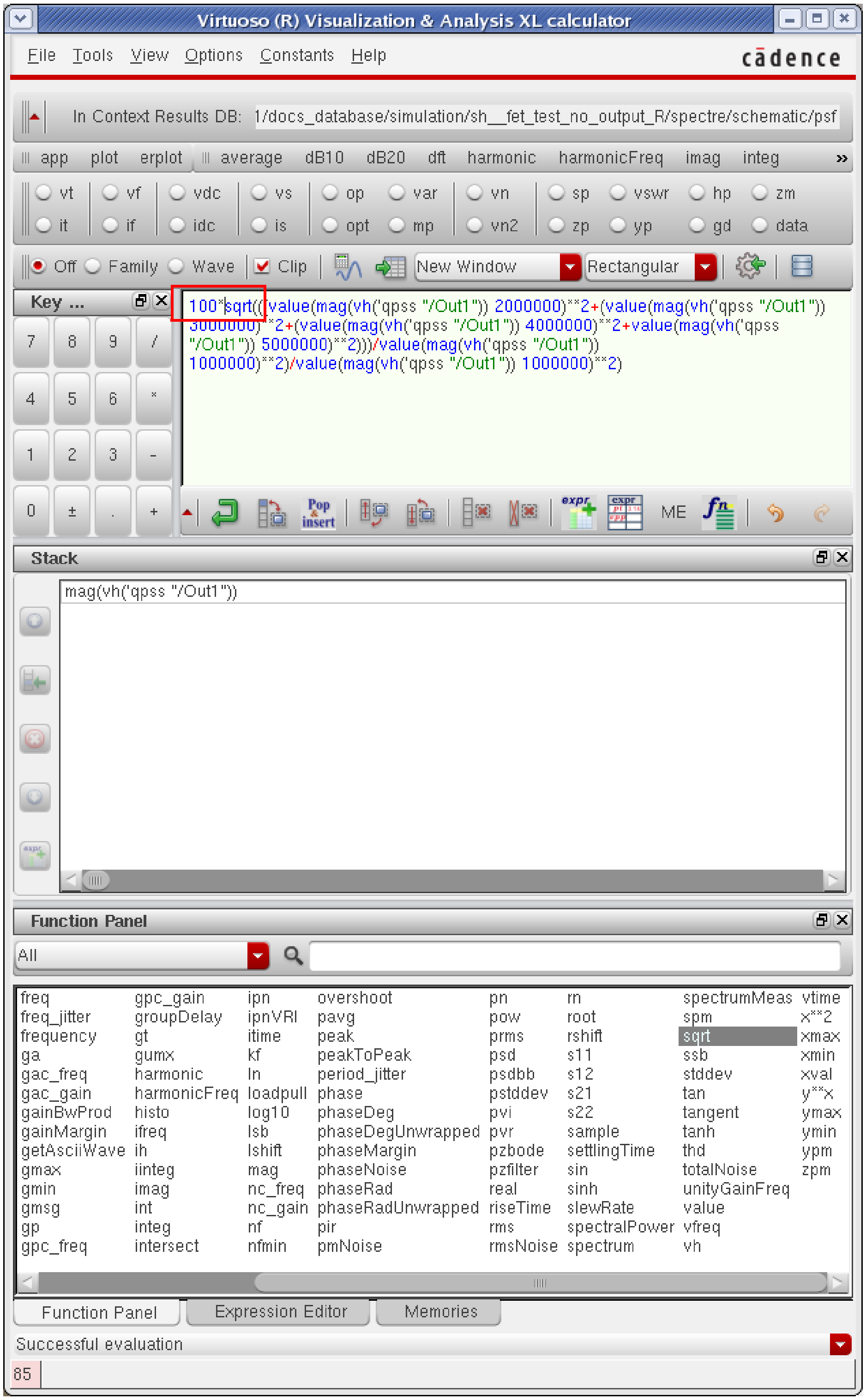


16. Verify that the expression is legal by clicking the *Evaluate the buffer and display the results in a table* button in the calculator.

17. The value is shown below.

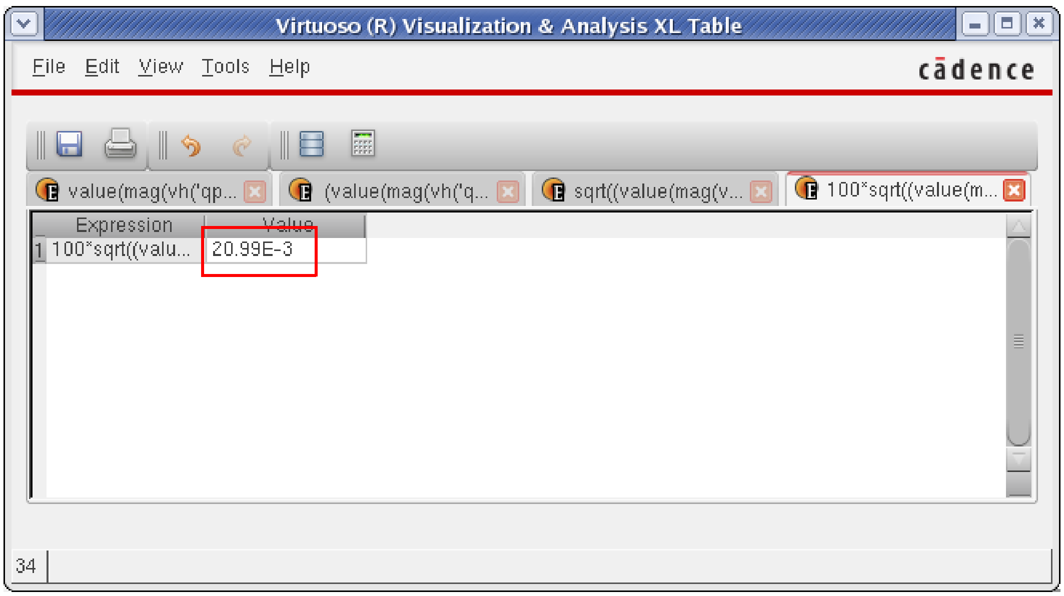


18. Click at the beginning of the expression, and type 100*.

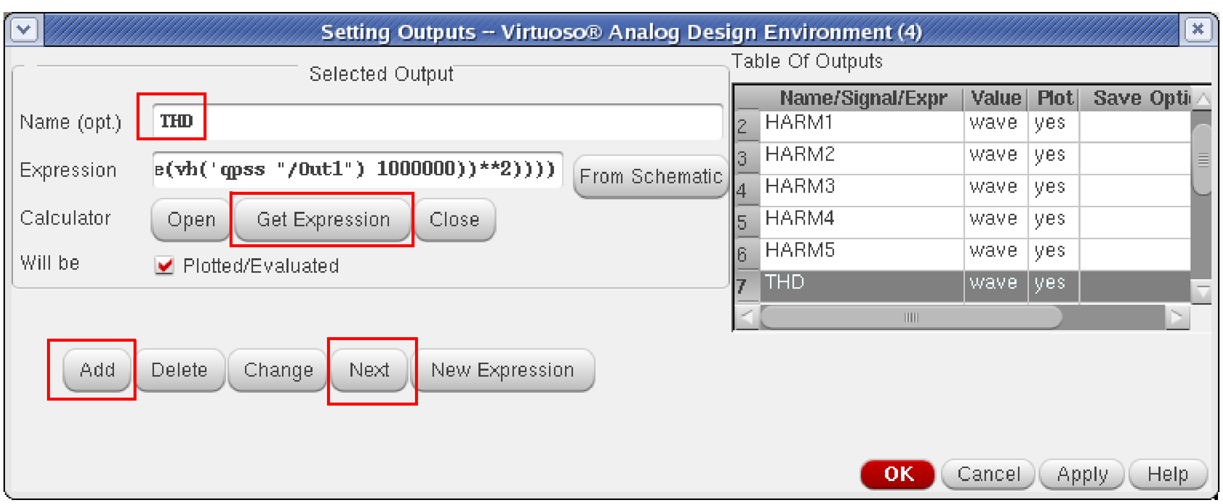


19. Verify that the expression is legal by clicking the *Evaluate the buffer and display the results in a table* button in the calculator.

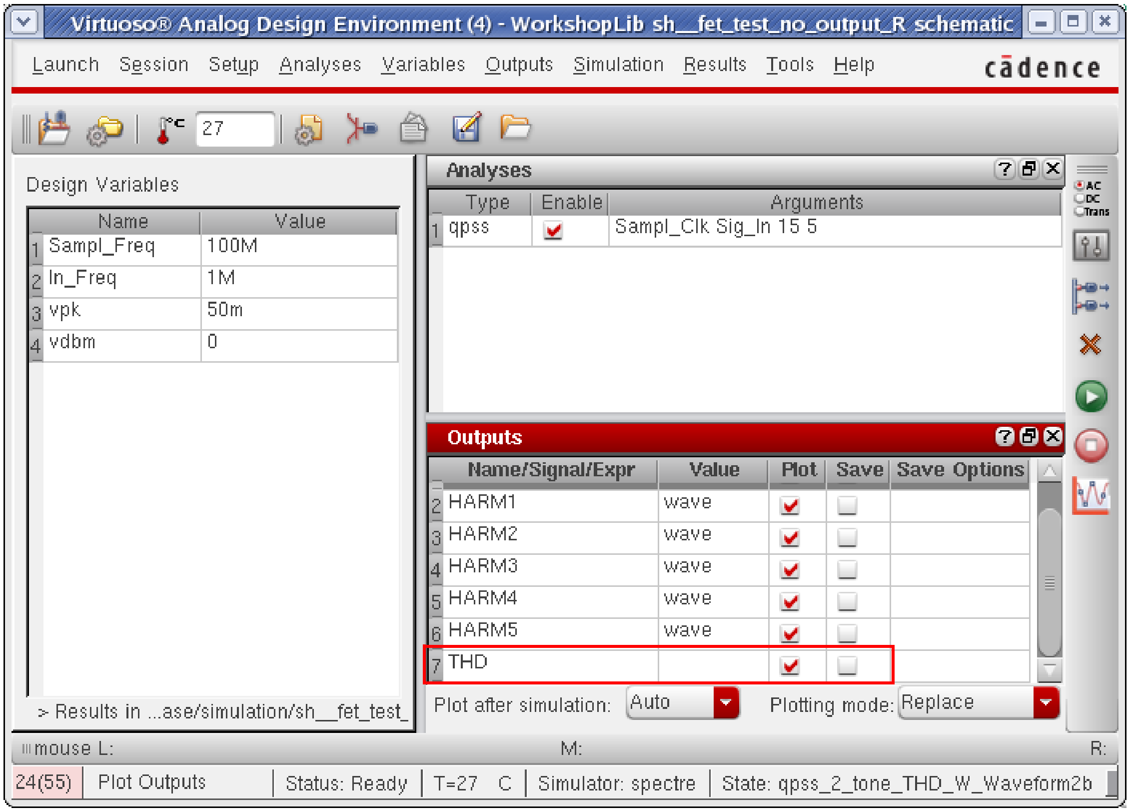
20. The value is shown below.



21. In the *Setting Outputs* window, select *Next* until you get a blank field for the *Name* and *Expression* fields. Click *Get Expression*. Type *THD* in the *Name* field.

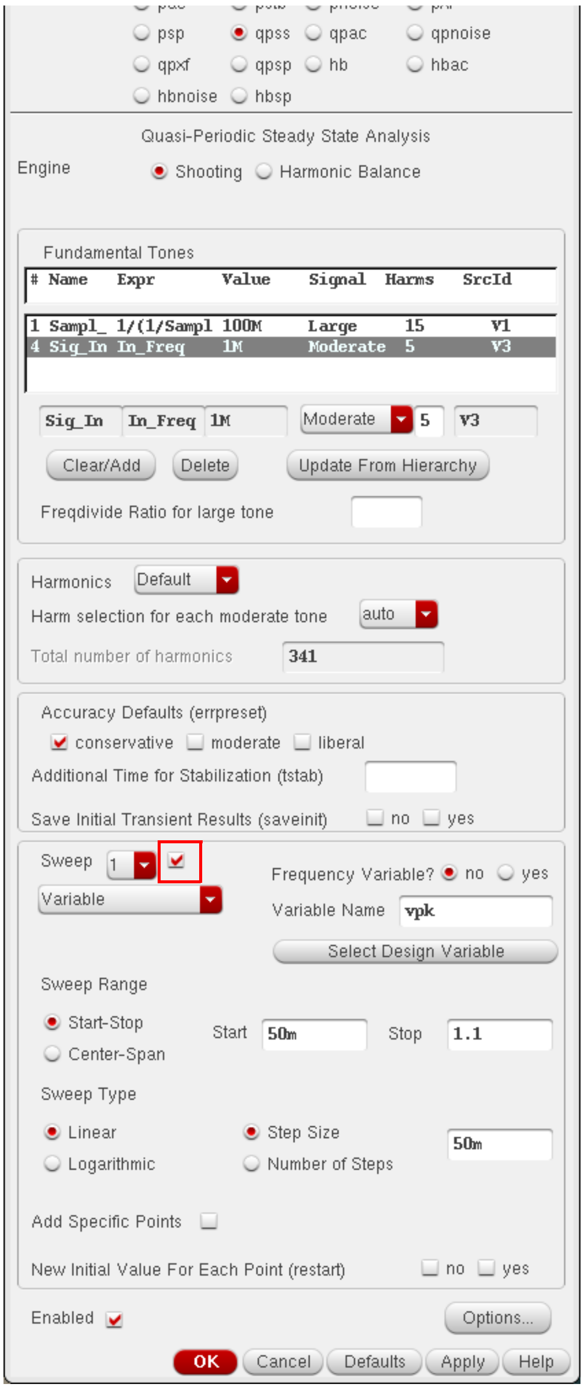


22. Click *Add*. The THD function is added to the ADE *Outputs* section.



Now sweep the input level, and run the simulation.

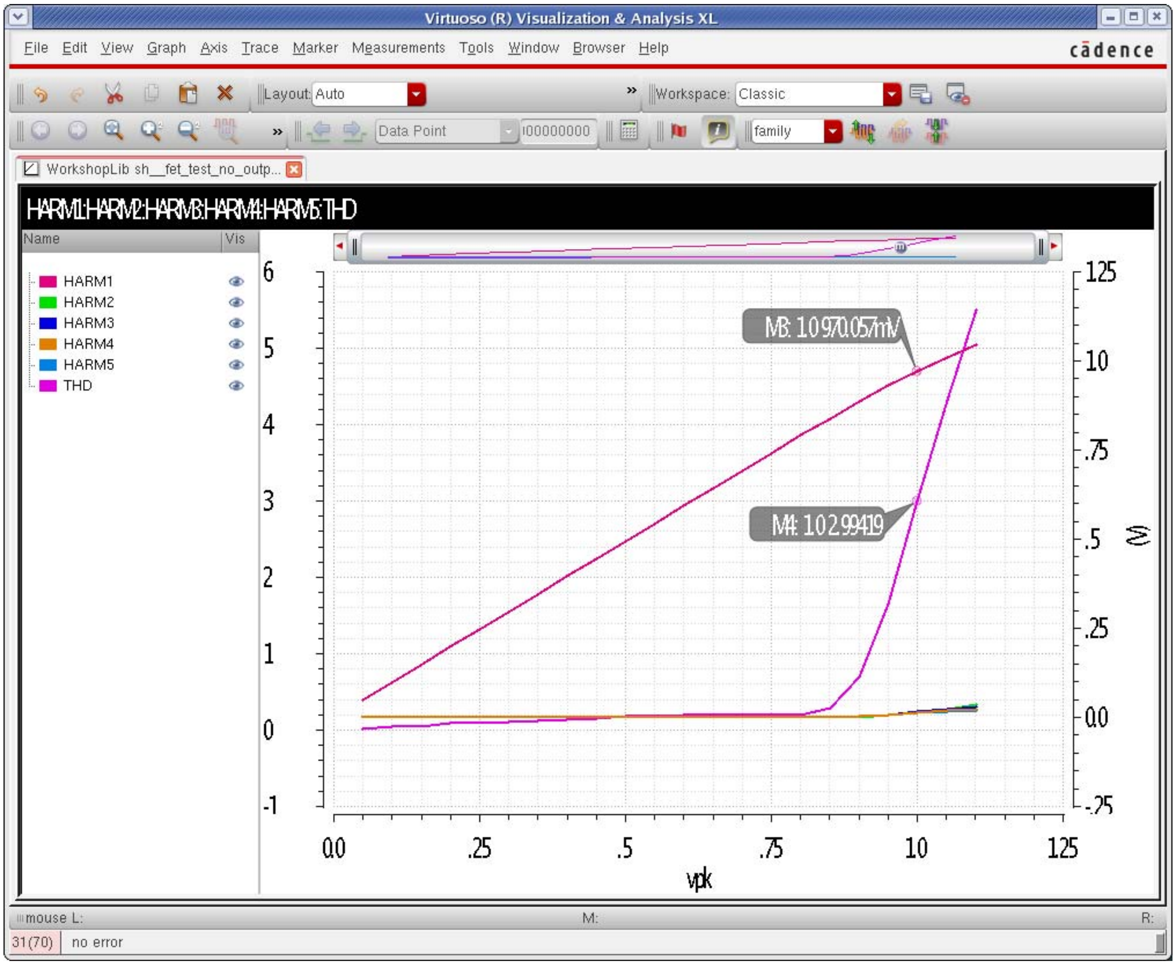
1. In the ADE window, double-click the qpss line. The qpss *Choosing Analyses* Form id displayed.



2. Select the check box to the right of the Sweep drop-down list.

3. Select the appropriate design variable that sets the input amplitude.

4. Specify an appropriate range and step size for your application. In this example, 50mv to 1.1v is the sweep range, with linear spacing. Click OK, and run the analysis. When the simulation completes, the amplitude of the first five harmonics, and the THD plot, as shown below.

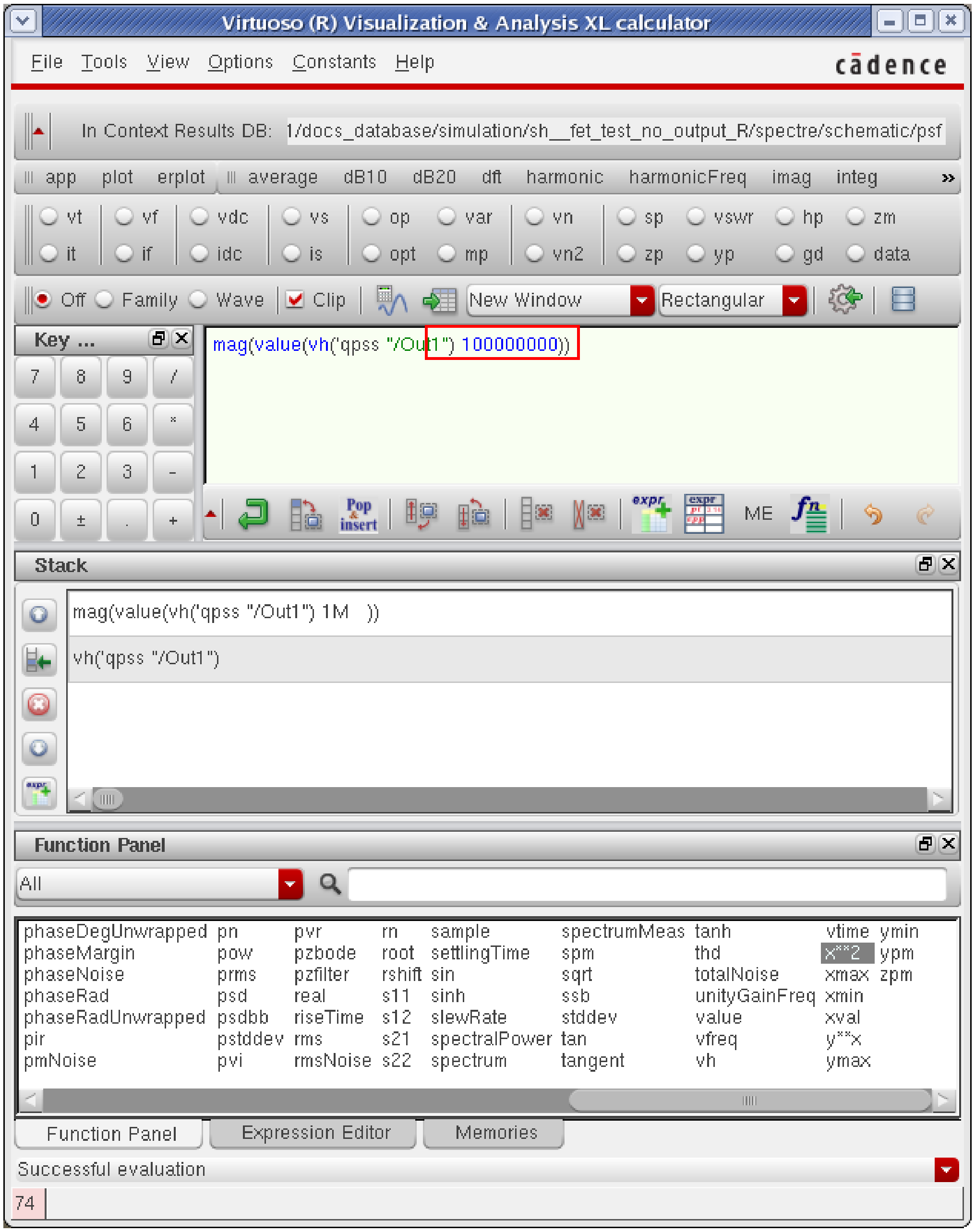


5. Markers are placed at the input amplitude of 1 volt peak. At that level, the output is 970mV peak, and the THD is 2.99%.

Add an expression to measure the clock feedthrough

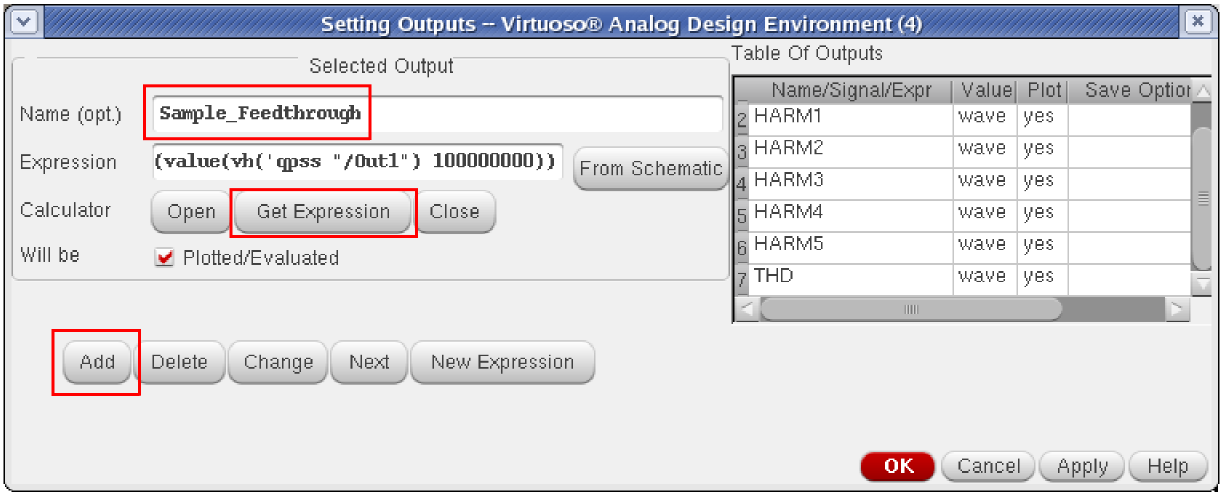
1. In the *Outputs* section of the ADE, double-click the *Harm5* line.

2. Click *Open*. This adds the expression for the 5MHz term into the calculator. In the calculator, change 5M to 100M.

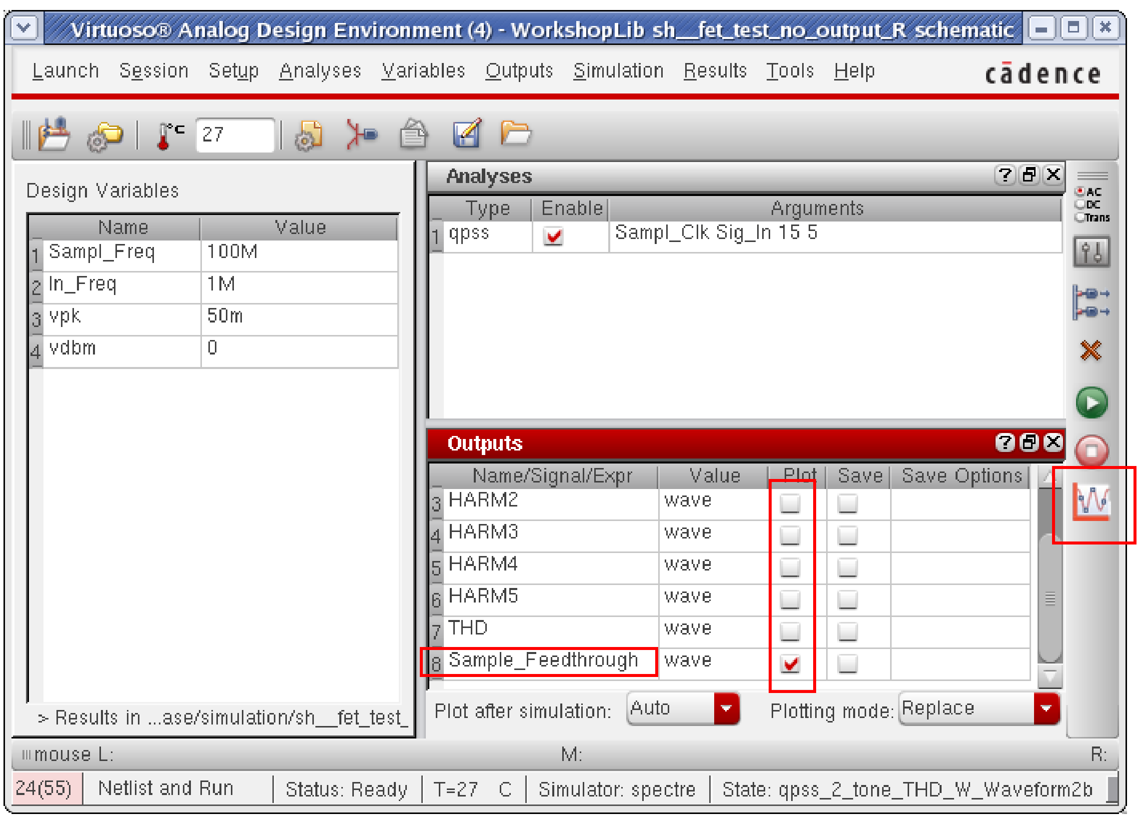


3. In the *Setting Outputs* window, click *Next* until the *Name* and *Expression* fields are blank,

4. Click *Get Expression*. Type in a name like *Sample_Feedthrough*.



5. Click *Add*. This adds the expression to the ADE *Outputs* section. Deselect all the check boxes under in the *Plot* column for all the expressions except for *Sample_Feedthrough*, and click the *Plot Outputs* icon in ADE.



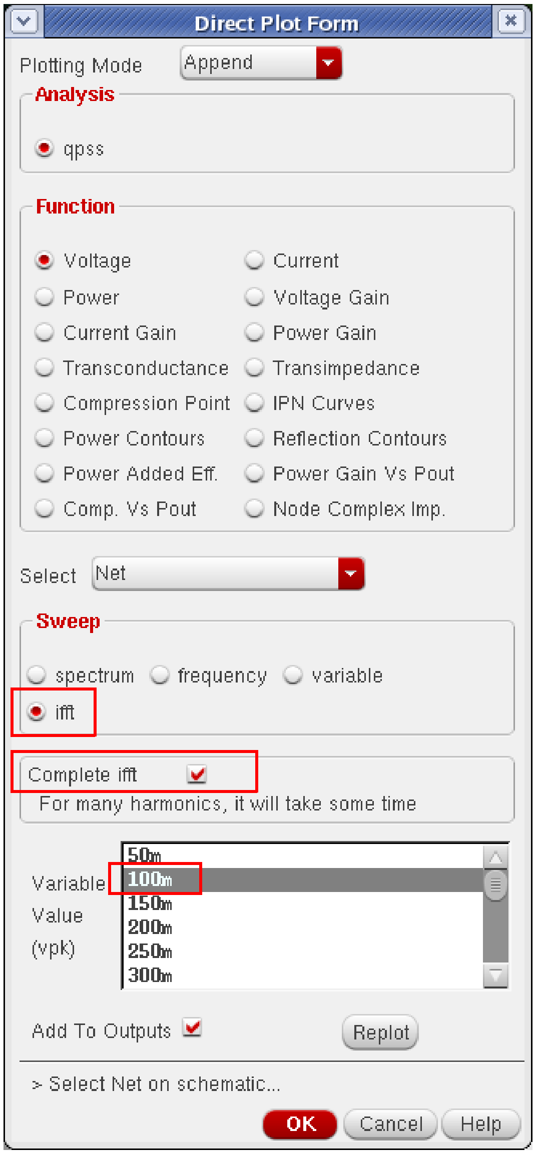
This plots the clock feedthrough.

Remember that this is a peak measurement.

Now plot the time-domain waveforms

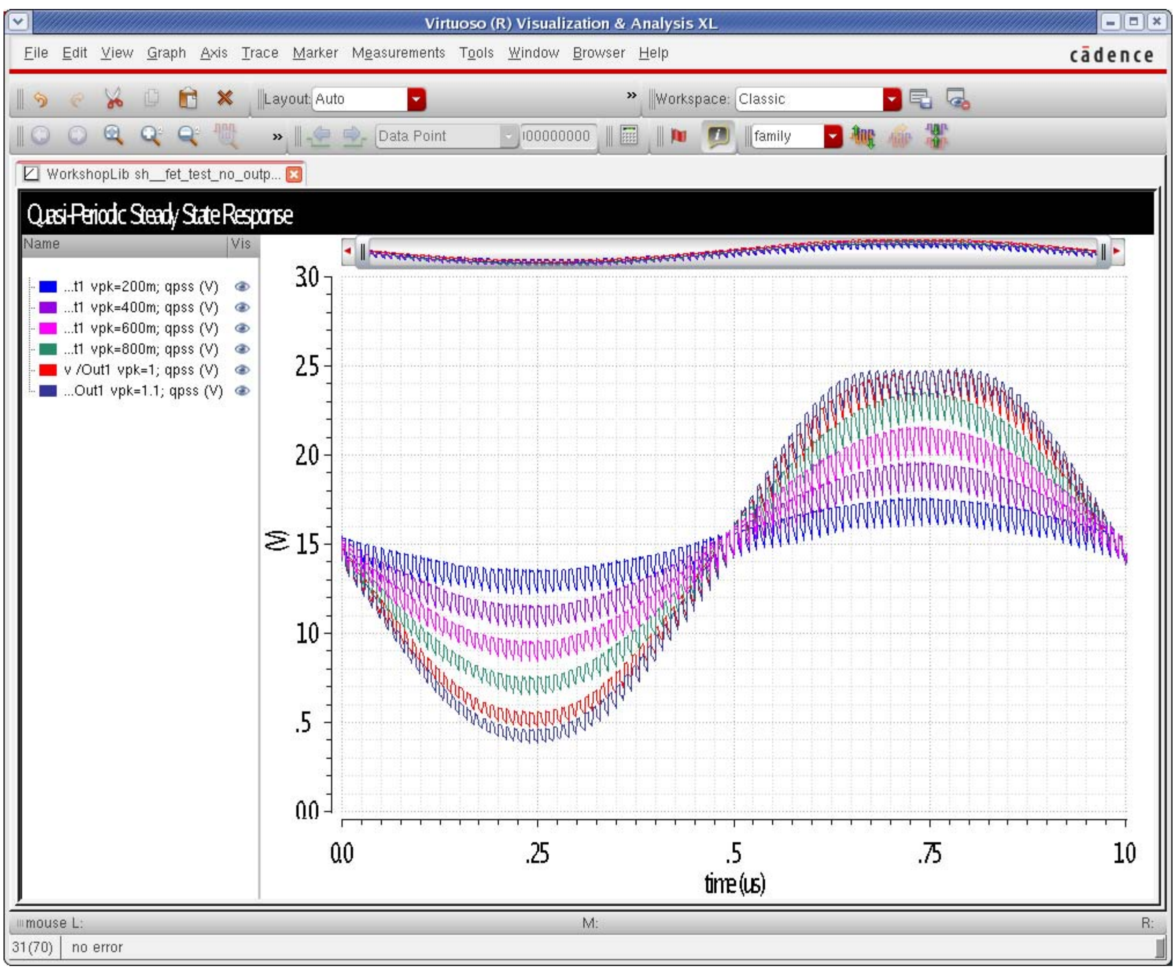
1. In the waveform tool, click *File - Close All Windows*.

2. In ADE, click *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed.



3. In the *Direct Plot Form*, select *ifft* from the *Sweep* section
4. Select *Complete ifft*.
5. Select *100m*.
6. Click *Replot*.
7. Select *300m*.
8. Click *Replot*.

9. Select *500m*.
10. Click *Replot.*
11. Select *700m*.
12. Click *Replot*.
13. Select *900m*.
14. Click *Replot*.
15. Select *1.1*.
16. Click *Replot*.



Although the distortion is not visible until the input amplitude becomes 1.1 volts, there is distortion in the output waveform at the lower levels.

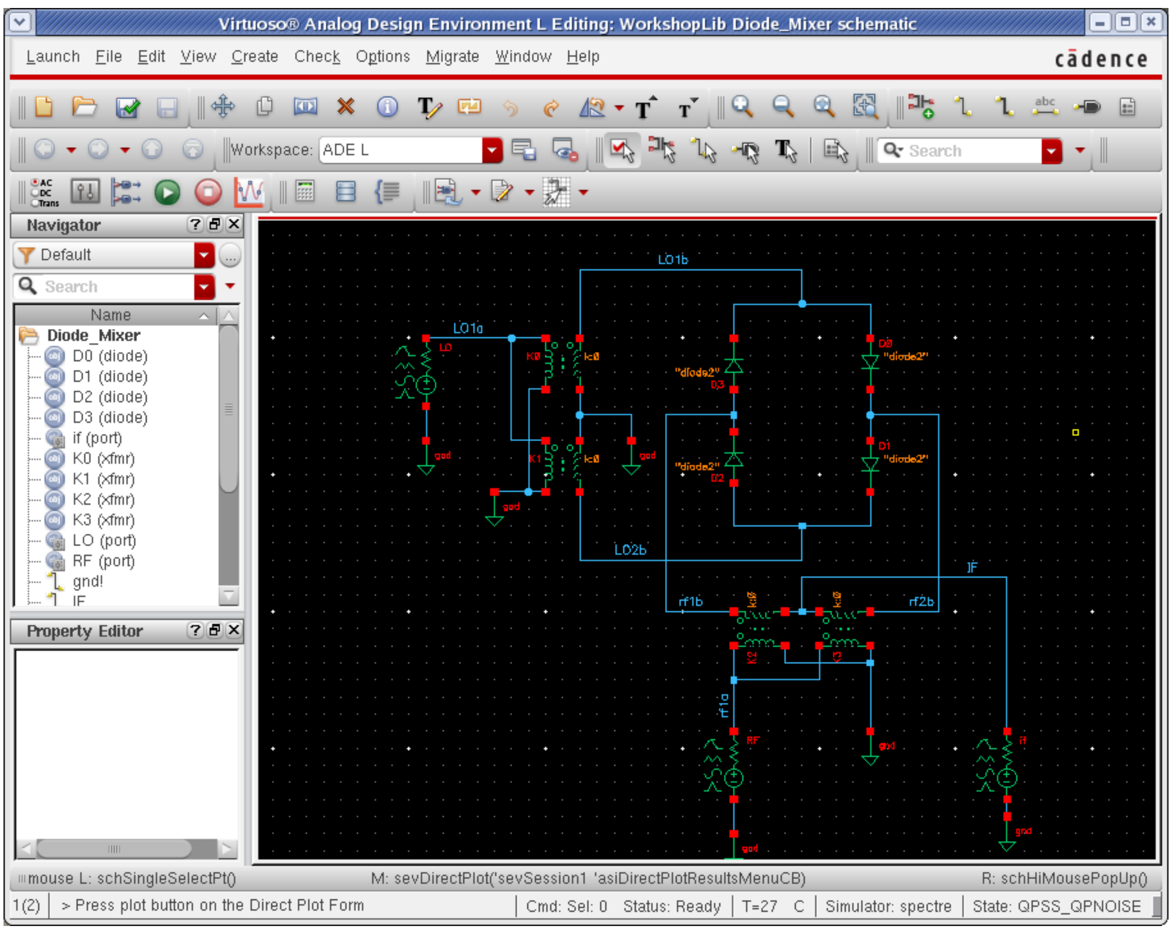
Quasi-Periodic AC Analysis (QPAC)

Refer to the pac analysis section in Chapter 4 for details about the pac analysis.

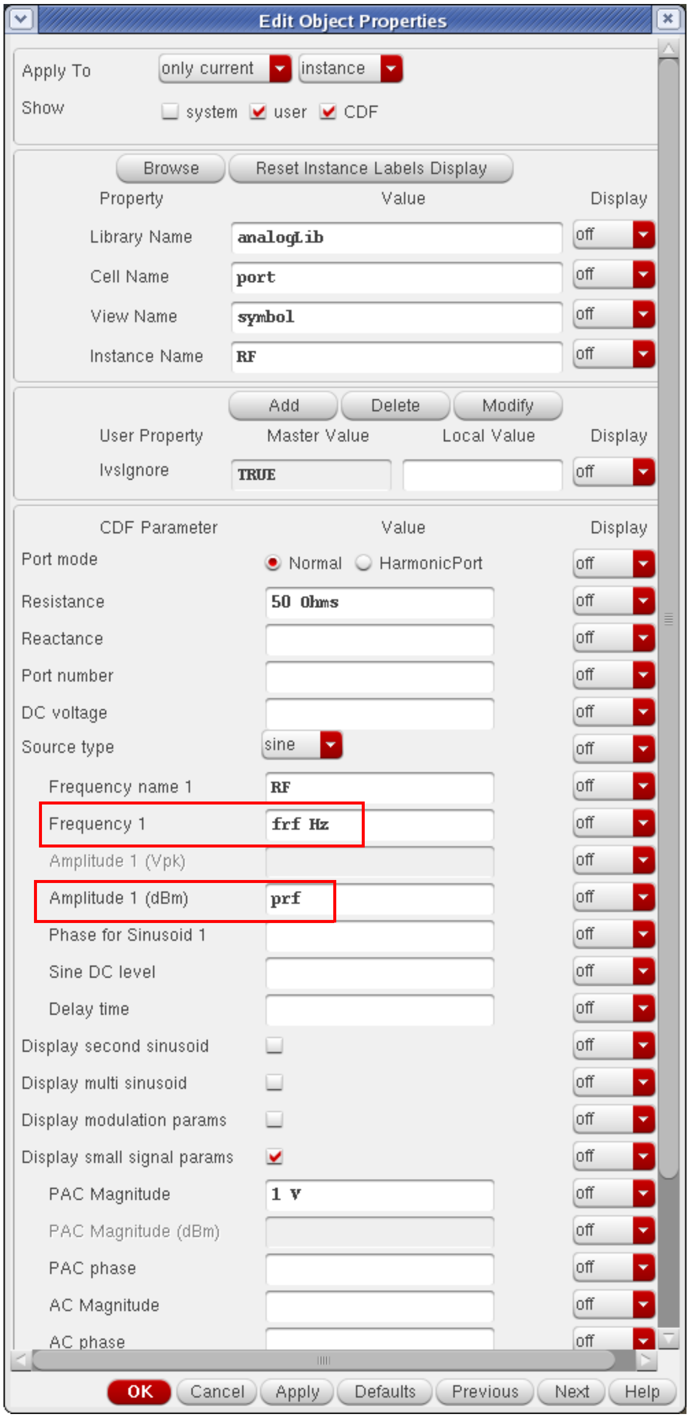
Qpac is just like pac, except it runs after a qpss analysis instead of after a pss analysis. Qpss captures the harmonics of the system, and qpac applies a single input frequency at a time, and calculates the output mixing products when the qpac input mixes with the harmonics of qpss. Conceptually, the only difference from pss-pac is that qpss has more signals and more harmonics in it to mix with, so there are more output mixing terms.

Example

Consider a diode mixer below.

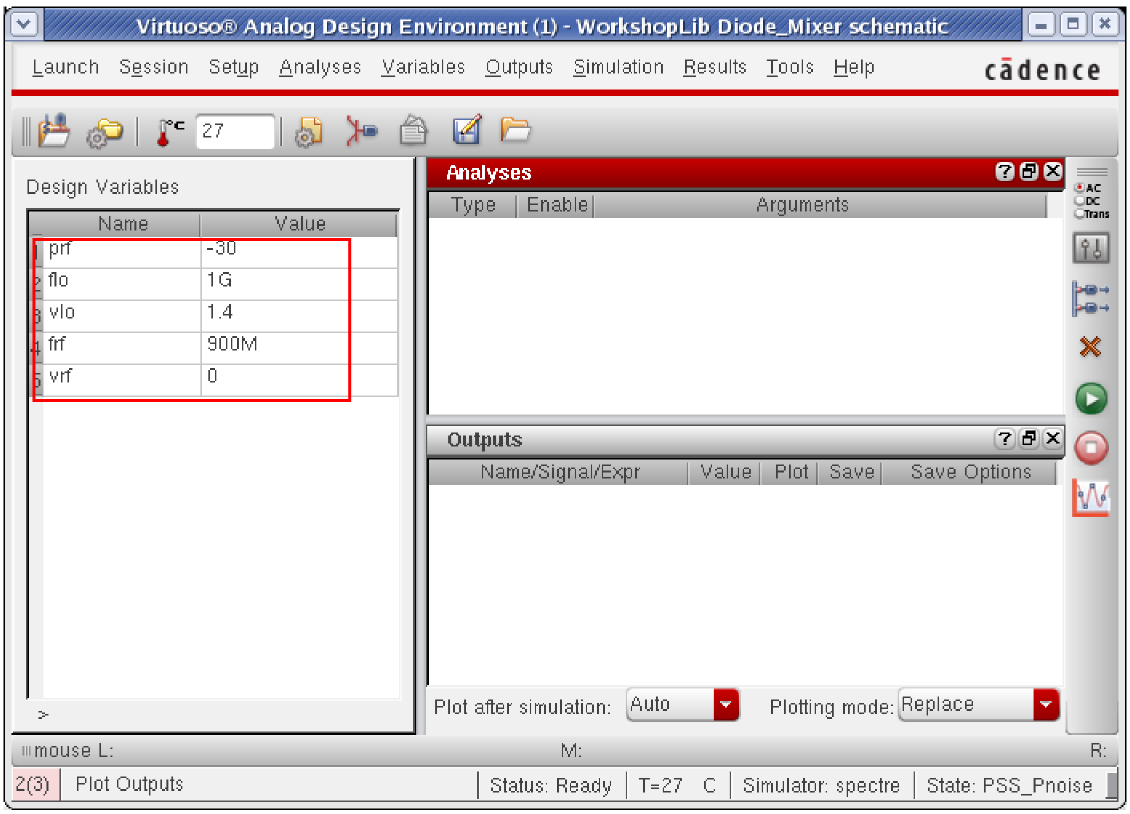


The input port has the frequency and amplitude set using variable names. This is done in order to allow the frequency and amplitude changes in ADE without the need for changing the schematic.

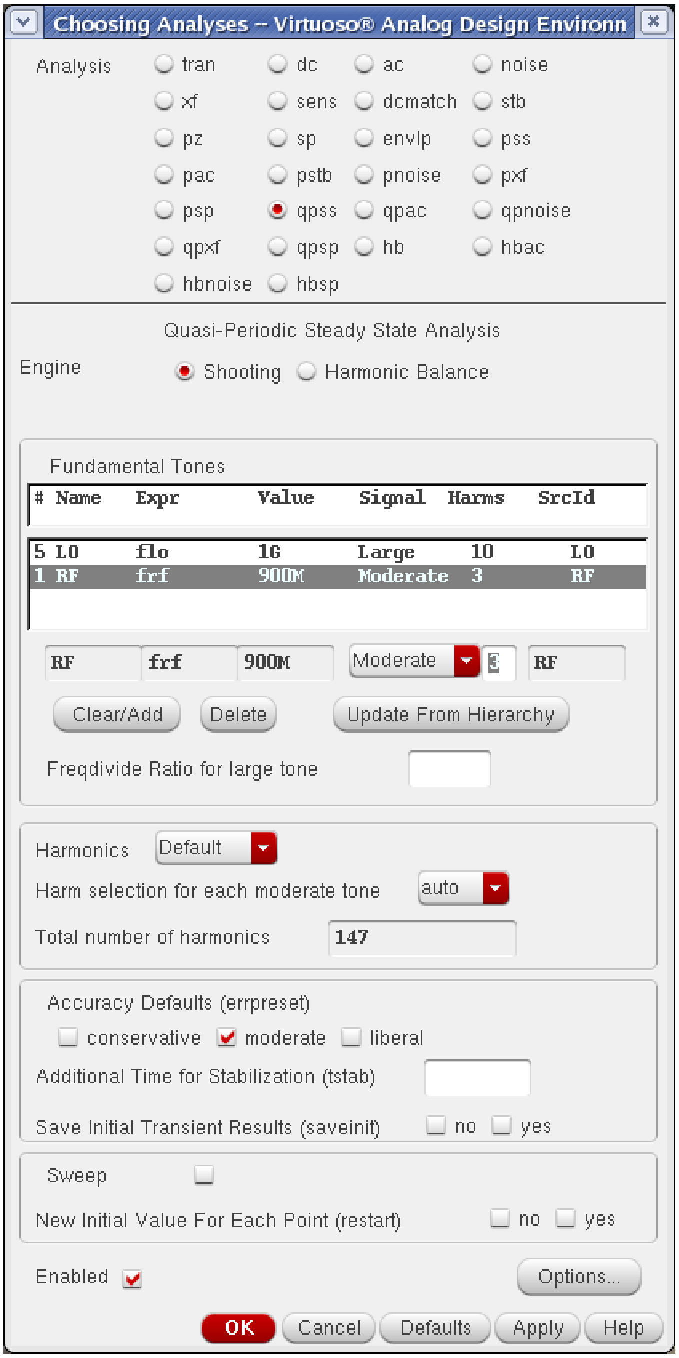


PAC Magnitude is set to *1 V* in the port. *PAC Magnitude* is used by pac and qpac to define the amplitude of the input signal for both analyses. This allows convenient measurement of conversion gain. Since the input is set to 0 dB, the conversion gain can be read directly just by probing the desired net.

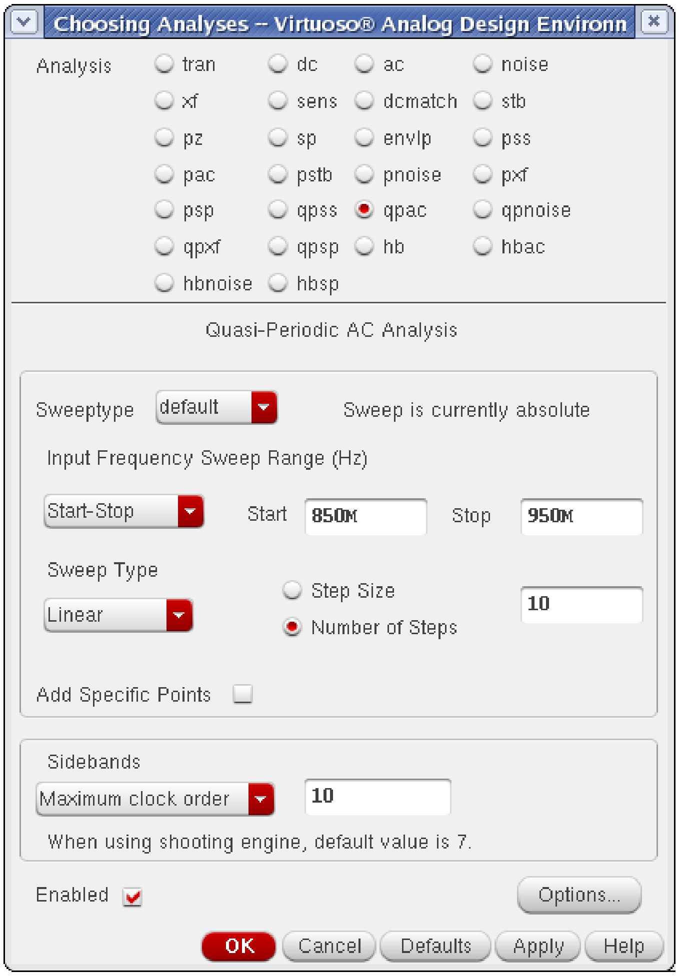
In a similar manner, the LO signal uses variable names. These variables are set in the ADE window.



Next, the qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter.

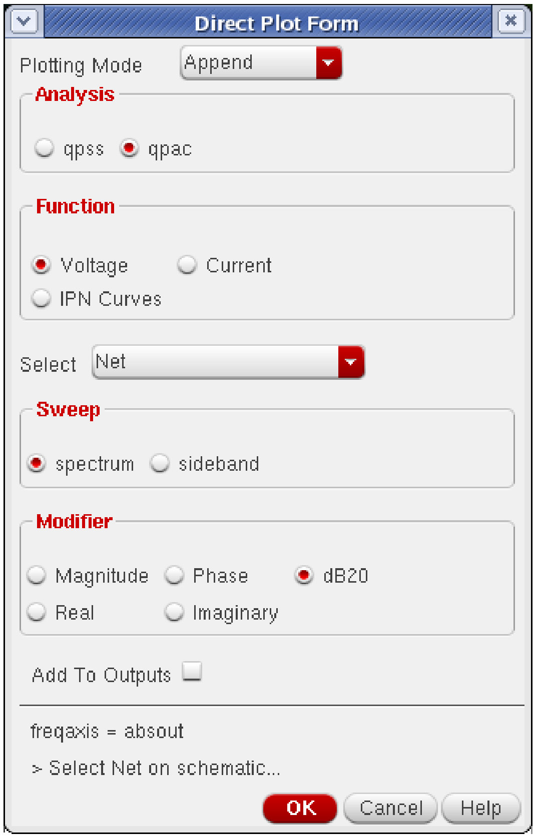


Next, the qpac analysis needs to be set up.



1. The input is from 850MHz to 950MHz, with a linear sweep.
2. Because 10 harmonics were set in the qpss analysis for the LO tone, 10 is set for *Maximum clock order*. These numbers should agree.
3. Now run the analysis.

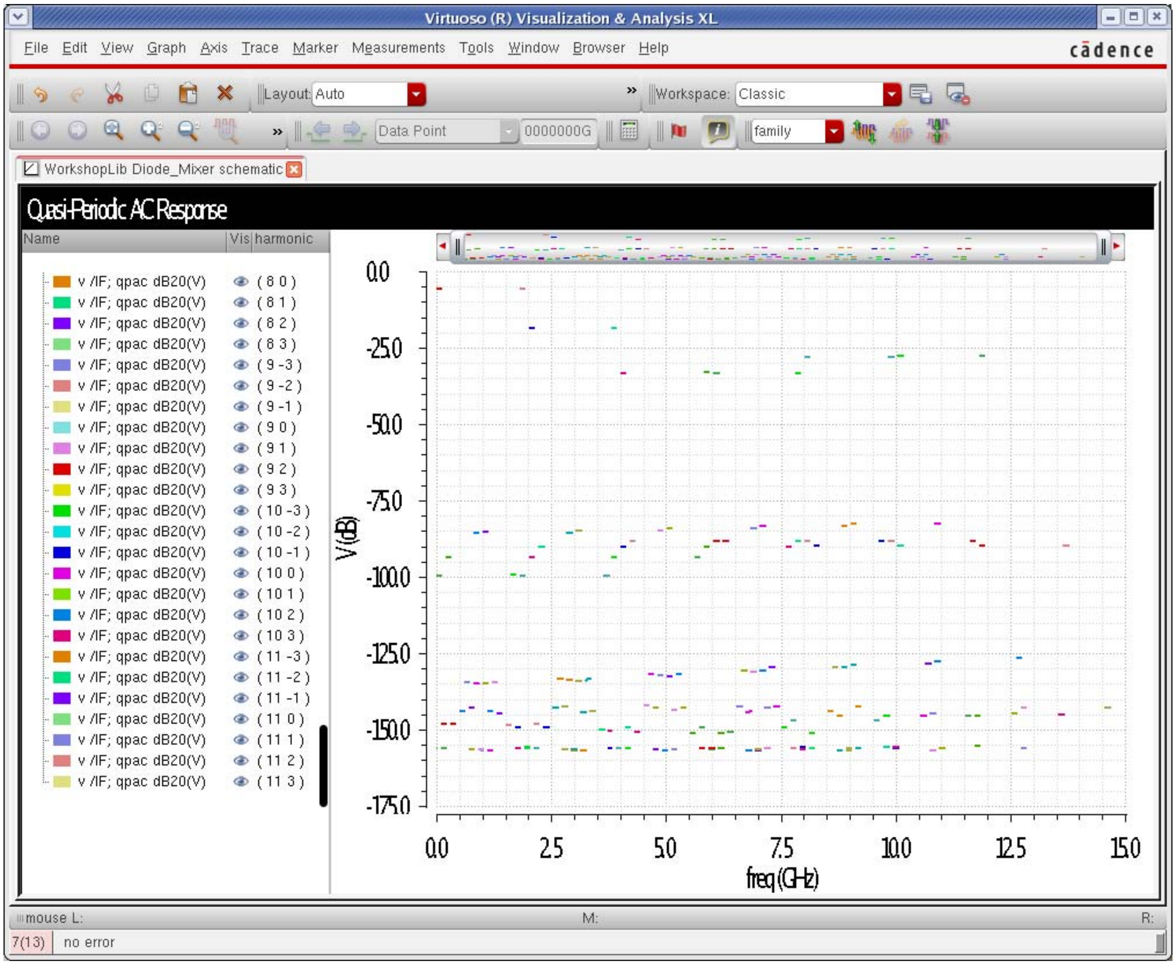
4. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* is displayed.



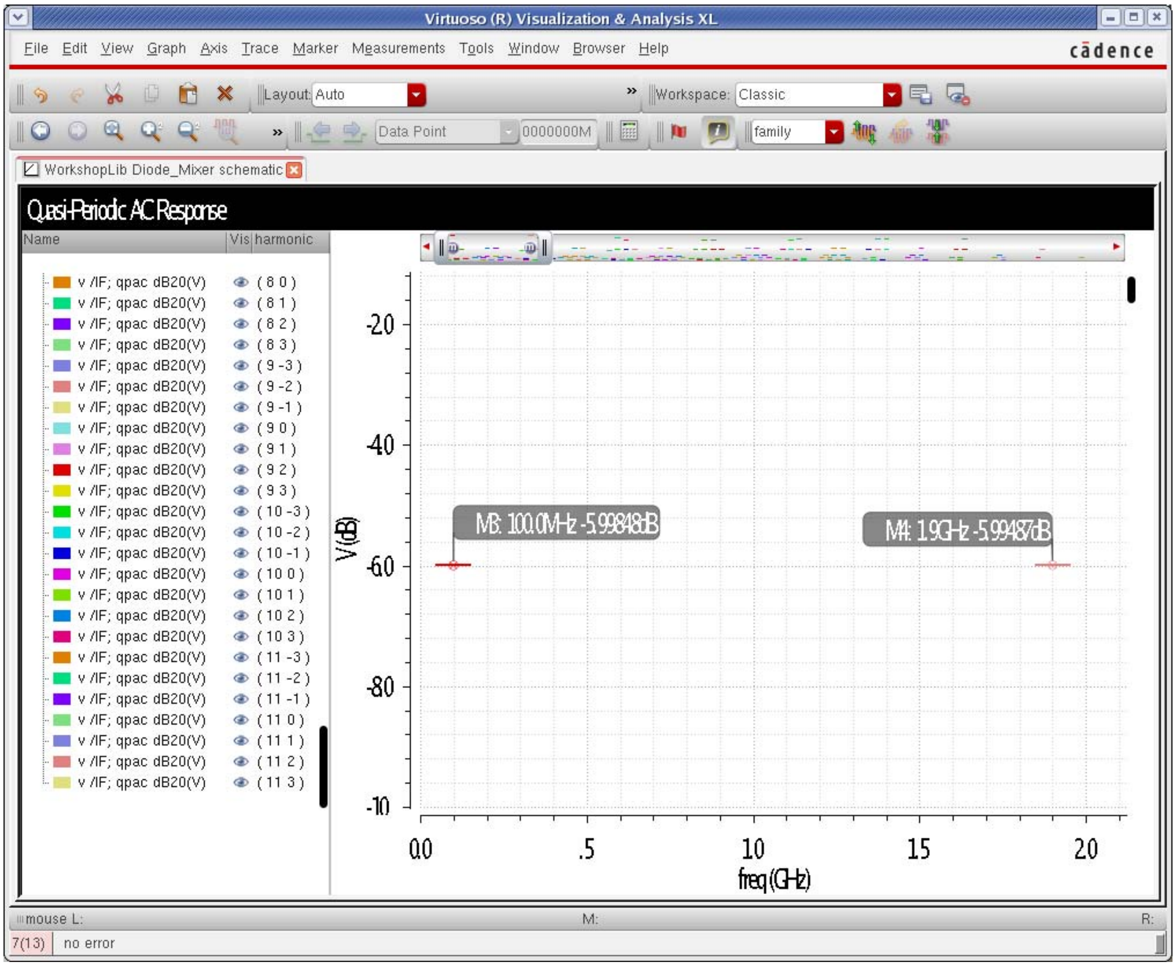
5. Select *qpac* from the Analysis section.

6. In many cases, *dB20* is useful.

7. The output net in the schematic is selected, The conversion gain terms are plotted, as shown below.



Because there are a large number of harmonics in the circuit, there are a large number of mixing products that are produced. In the view below, zooming is applied to show just the main mixing products. Markers have been placed for the center frequency of the sum and difference frequency mixing terms.



As is common for diode mixers, the insertion loss is about 6 dB.

Qpac Concepts

Generally, when qpac is needed, harmonic balance is appropriate for the simulation engine. In this case, the hb and hbac analyses are recommended because the hb and hb analysis forms always have the latest features.

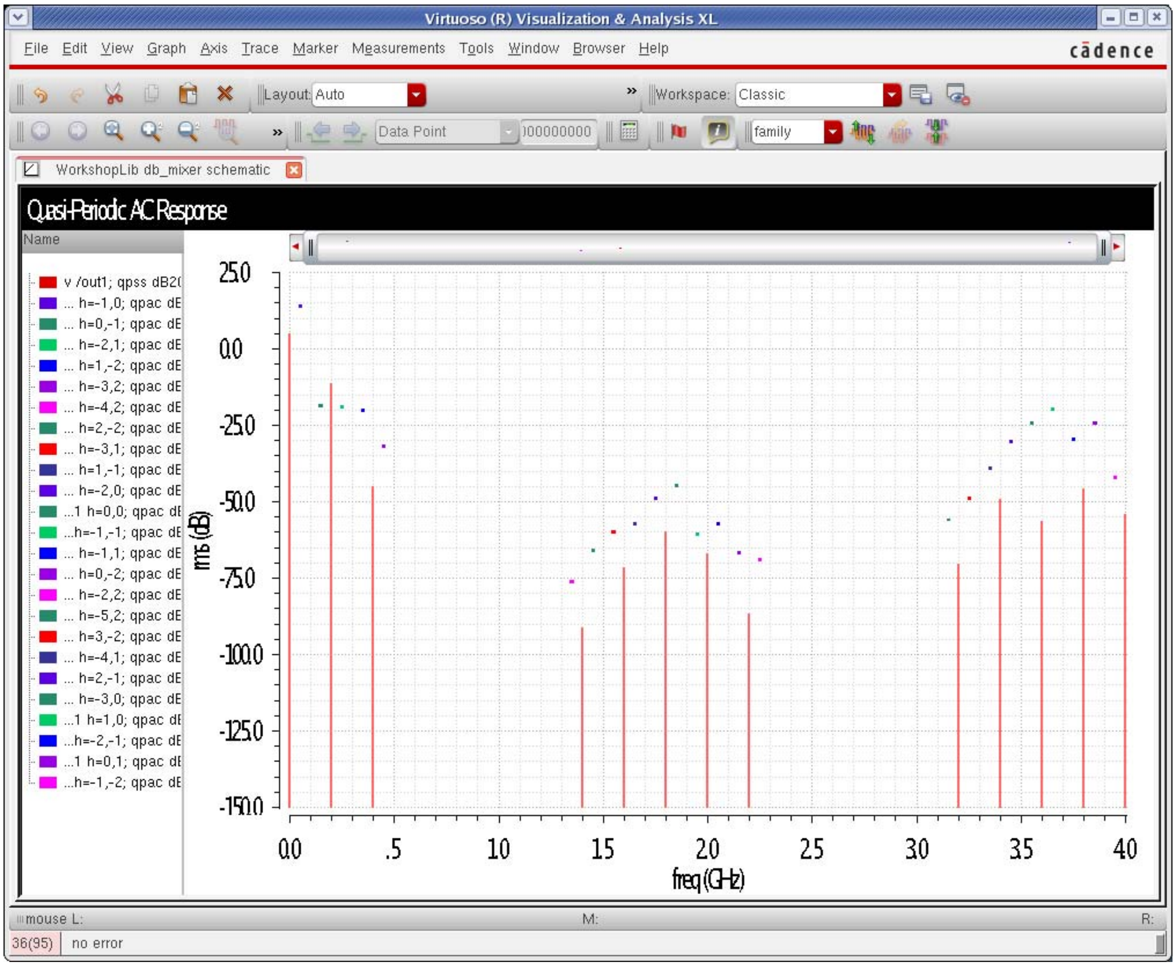
Qpac applies a small-signal input to the system that is simulated with the qpss analysis. Qpss solves for the harmonics of the system with two or more inputs to the circuit. Qpac calculates the small-signal outputs that are created by mixing with all the harmonics that are in the qpss solution.

One application of qpss-qpac is measuring conversion gain of an LNA-Mixer circuit when a large amplitude blocking signal is present. In this case, the LO and the blocker are applied in qpss, and the small-signal conversion gain is measured using qpac. Note that for this application, harmonic balance is likely to run faster than shooting, and as a strong recommendation, use the hb and hbac analysis for this application.

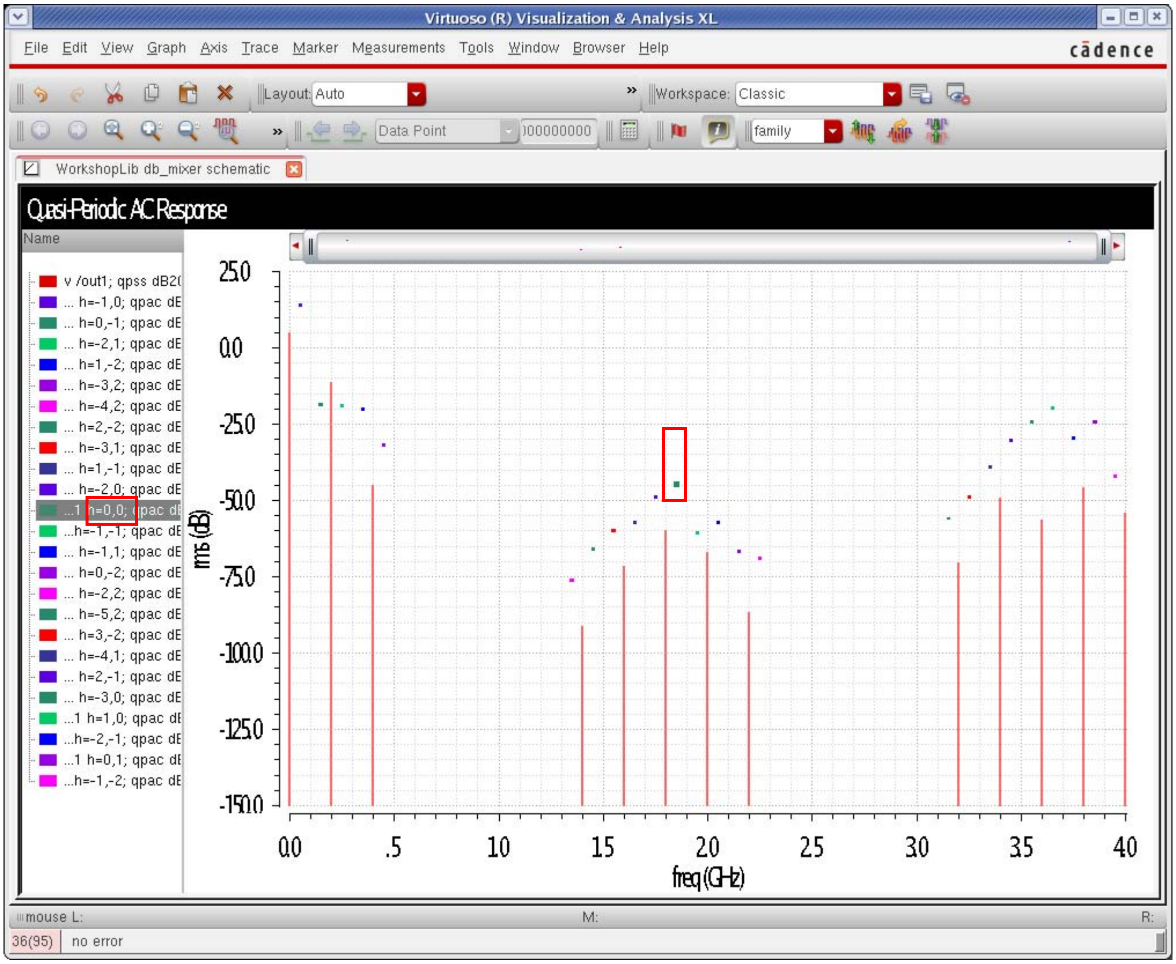
Unlike hbac, qpac has only one mode, which measures the small-signal mixing products. Modulated, sampled, rapid IP3 and IP2, and the distortion summaries from hbac are not available in qpac.

Down Conversion: Input Near the First harmonic of the LO

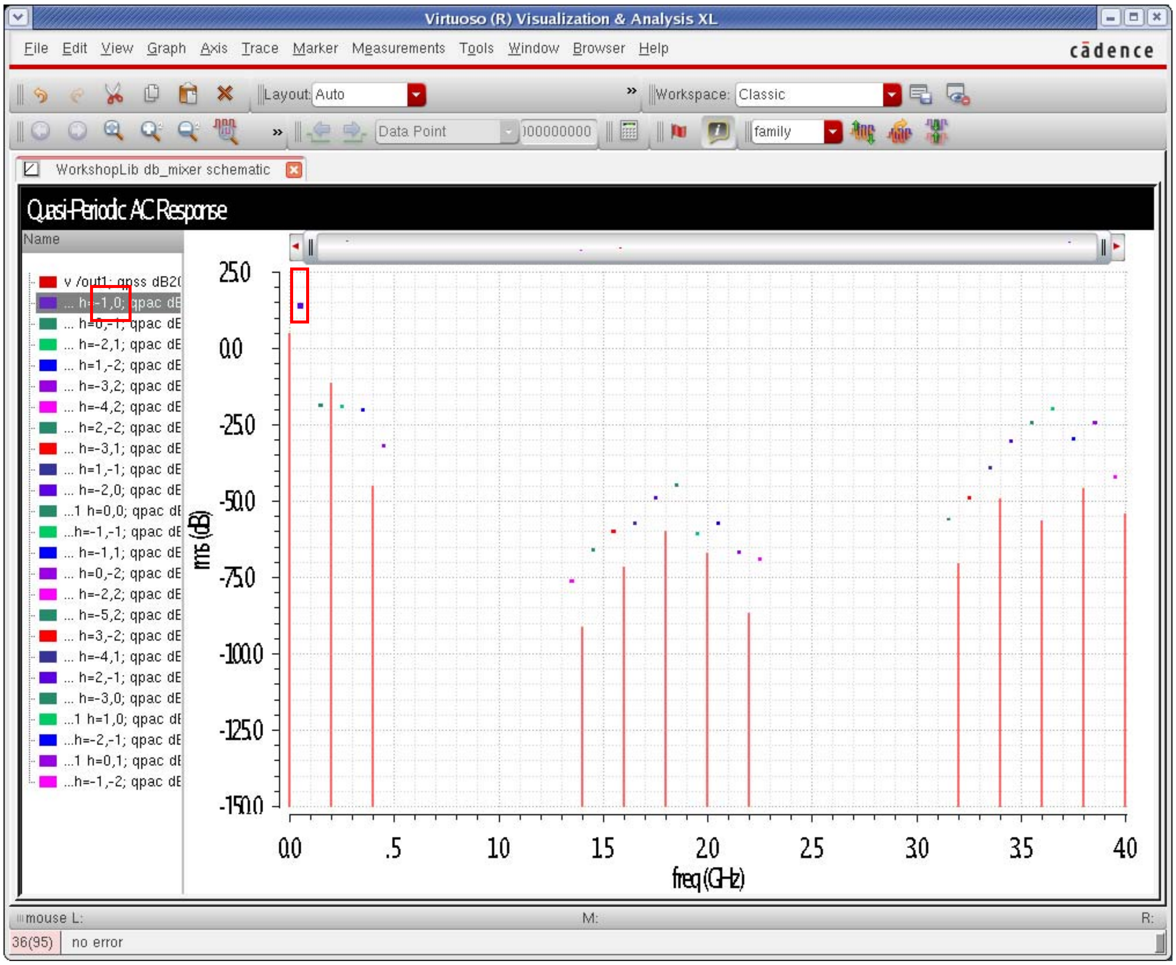
Qpac is for measuring the conversion gain that is the average conversion gain with all the signals applied to the circuit (usually the LO or the sample clock and a blocking signal) applied to the circuit. The mixing products that are produced are called sidebands. The sideband number has a value for each signal that is applied to the circuit, and is the harmonic number that is being mixed with to provide the output. A positive number means that the output mixing product is higher in frequency than the input frequency specified in the *Choosing Analyses* form. A negative number means that the output is lower in frequency than the input frequency. If there are two inputs to the circuit in the qpss analysis, there will be two numbers for each qpac sideband. In the example below, there is an LO at 1.8GHz and a blocker at 2GHz. The RF small-signal input is at 1.85GHz. A waveform plot is shown below. The red spectrum is the spectrum calculated by the qpss analysis, and the small dots are the qpac result. The amplitude of the qpac output is larger than the qpss output because the qpac outputs assume a 1 volt input, and the qpss has much smaller input levels. I will explain further.



All the sideband numbers are plotted on the left of the waveform tool. When 0 0 is selected, the output frequency is the input frequency that does not mix with either input frequency. Thus, the 0 0 sideband is at 1.85GHz (the input frequency) as shown below.

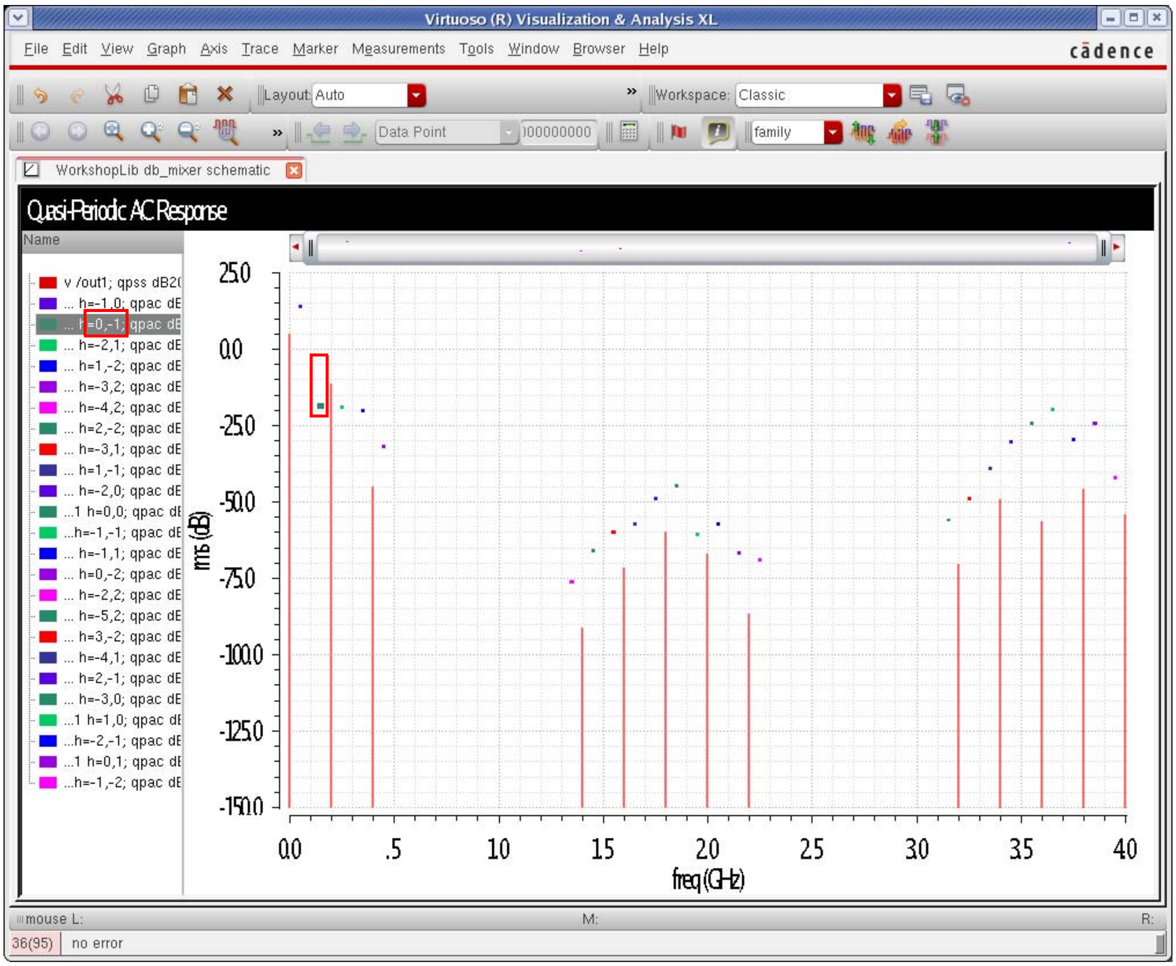


The primary output mixing product is the signal that mixes down with the LO, and does not mix with the blocker. It mixes down with the first harmonic of the LO, and does not mix with the RF blocker, thus the sideband number is -1 0.



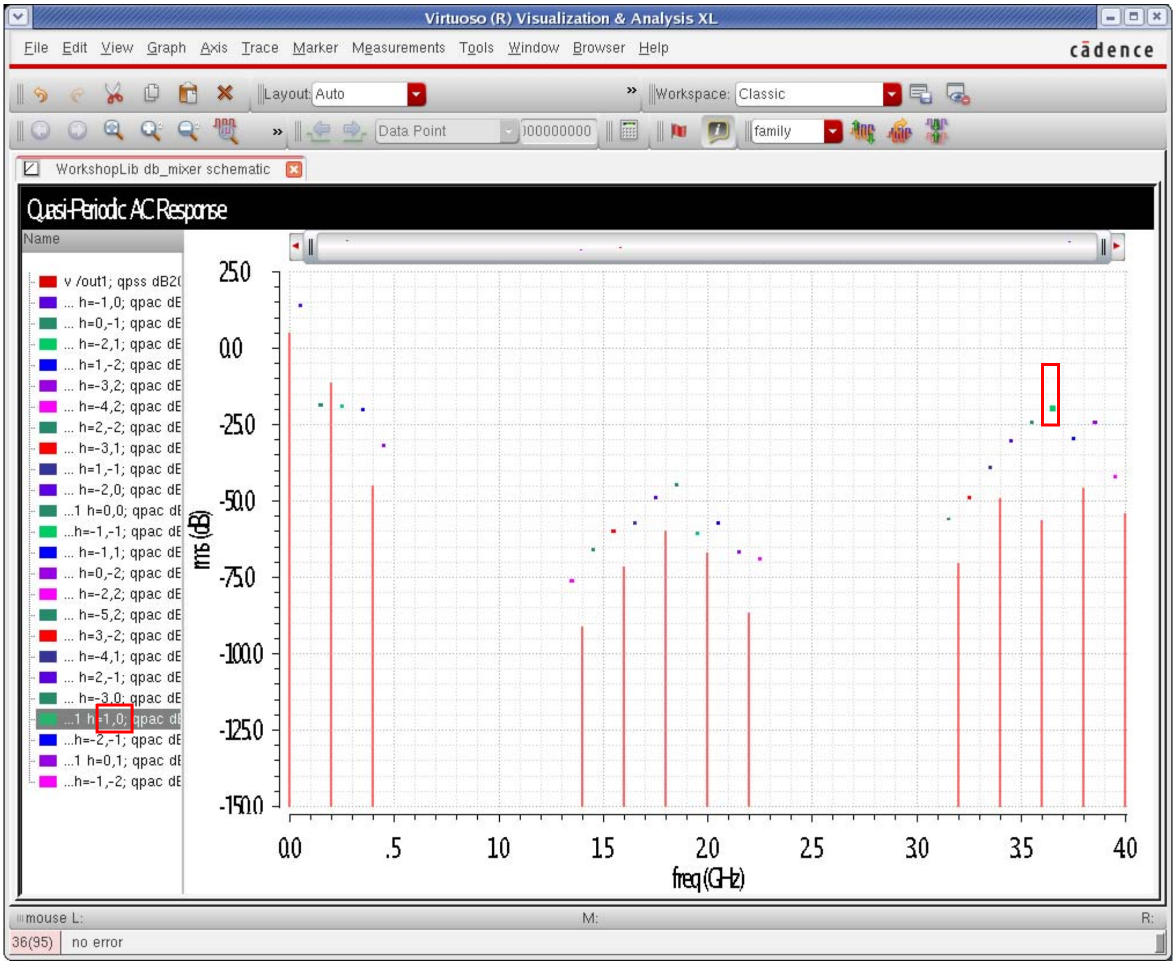
1.85GHz (the input frequency) minus 1*LO frequency (1.8GHz) is 50MHz.

1.85GHz can also mix down with the blocker frequency. In this case, the signal mixes down with the first harmonic of the blocker (2GHz) and not with the LO.



The sideband number here is 0 -1. 1.85GHz -1*2GHz = -150MHz. By default, this is mirrored up to the positive frequency domain in the qpac analysis.

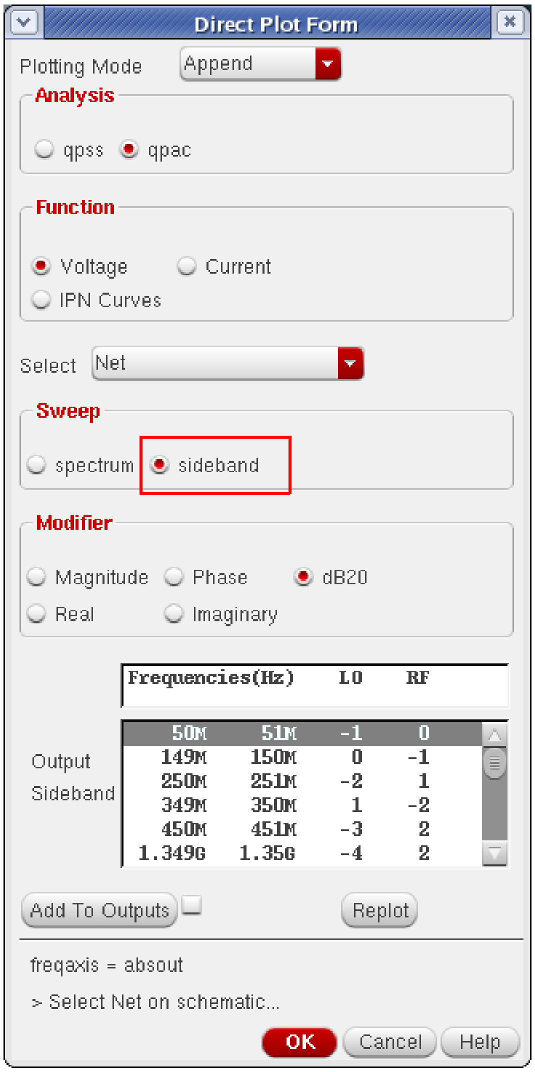
Next, the frequency can mix up with the LO. This is the 1 0 sideband.



To calculate the frequency, the input is 1.85GHz + 1* 1.8GHz = 3.65GHz.

The Easy Way to Calculate Sideband Numbers and Frequencies

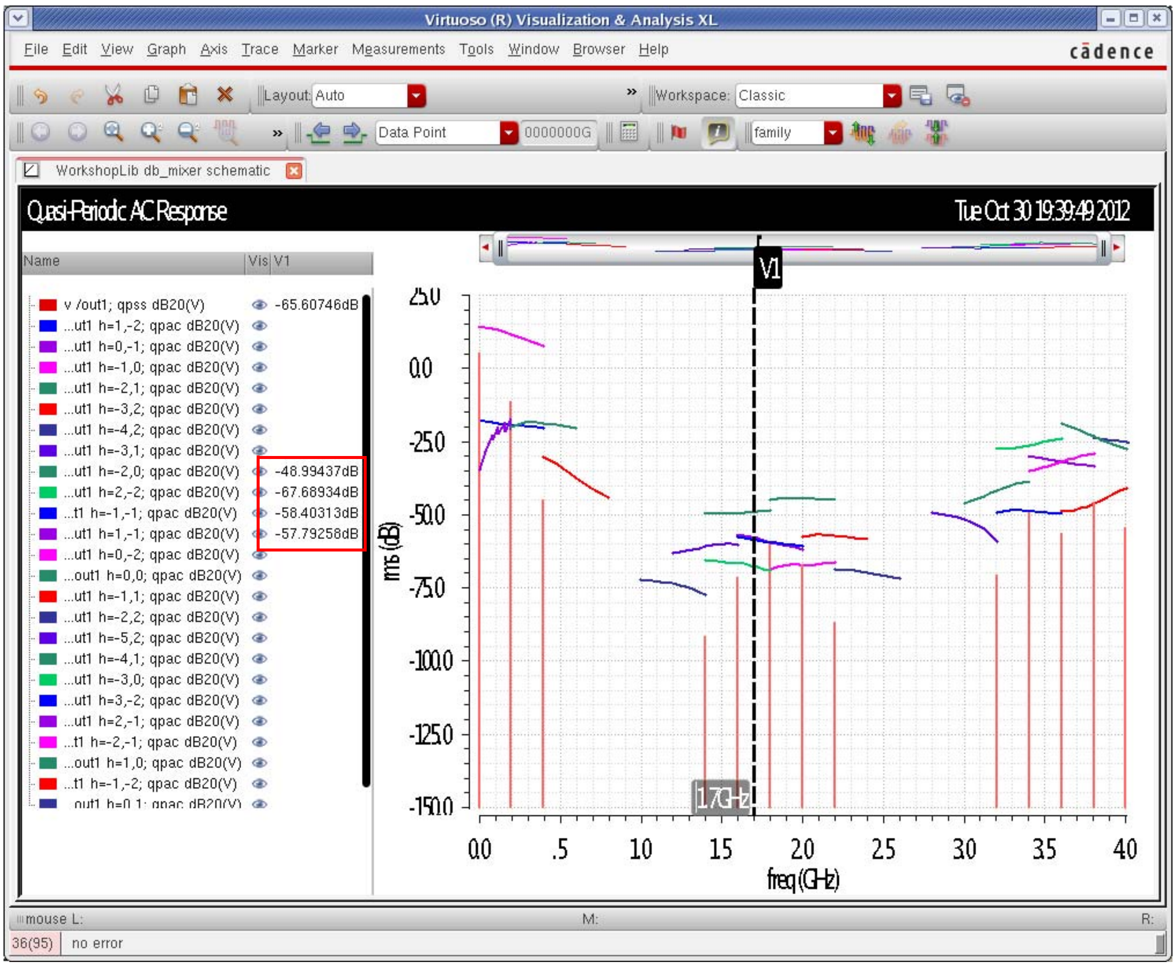
The easy way to calculate sideband numbers and frequencies is to use the *Direct Plot Form*. The *Direct Plot Form* displays the sideband numbers and the frequency ranges when you select *sideband*, as shown below.



In this example, the LO is at 1.8GHz, the RF Blocker is at 2GHz, and the input sweep range is 1850MHz to 1851MHz.

Avoid Wide Sweep Ranges

Wide sweep ranges cause very confusing results. This should be avoided. An example is shown below.



The marker is placed at 1.7GHz. The value from qpac at the marker value is shown in the legend area on the left side of the waveform display. The different traces result from mixing with different harmonics in the qpss analysis.

Relative Frequency Sweep

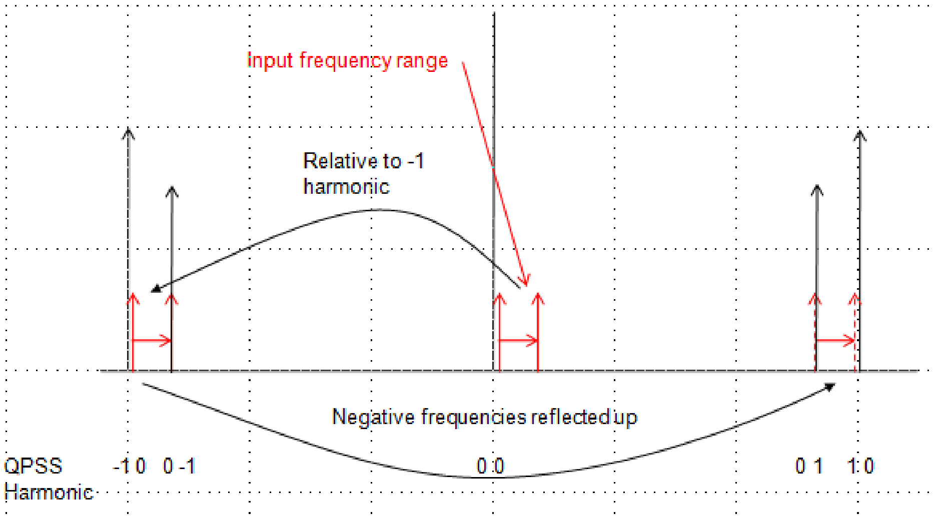
The qpac analysis is like AC, where the input frequency is swept. The input frequency range is specified in the qpac *Choosing Analyses* form.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonic numbers in the qpss analysis.

For example, assume that the qpss has a 1GHz large-signal input, and a 900MHz *Moderate* input. You want a log sweep above the first harmonic of the 1GHz signal in the qpss analysis. In this case, you need to select *relative* sweep, and specify 1 0 in the *Relative Harmonic* field. The first number is the frequency shift in multiples of the LO frequency, and the second number is the frequency shift in multiples of the RF input frequency. Next, type in 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade.

The sweep will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.

To use a log sweep below the first harmonic of the 1GHz input, use the same frequency range, and type in -1 0 as the relative harmonic. This is illustrated below.



Maximum Clock Order

In this case, clock is the terminology for the large signal in the qpss analysis. Sidebands (the output mixing products calculated by qpac) are calculated through the highest harmonic of the large signal in the qpss analysis. This is usually the LO signal.

Conceptually, an infinite number of mixing products are produced when a single input is applied to a nonlinear system. From the practical point of view, usually only a small number of mixing products need to be measured.

Consider an example. Assume that a 1GHz large signal and a 900MHz moderate signal is applied in qpss. 10 harmonics are set for the large signal at 1GHz, and three harmonics are specified for the moderate signal. Assume there is no need to measure the mixing products above the third harmonic of the LO because there is no spec at frequencies above that. In that case, you could set *Maximum clock order* to three, and only the mixing products that are produced by mixing with the first three harmonics of the LO are calculated in the qpac analysis.

Sideband is the name of the qpac output mixing product. The sideband number is the qpss harmonic numbers that is being mixed with. If the sideband number is negative, the output frequency is mixed down in frequency from that input. If the sideband is positive, the output is mixed up in frequency. A sideband contains the same number of numbers as there are tones in the circuit. If there are 2 tones in the circuit, each sideband has two numbers, like -1 0. If there are three inputs, each sideband has three numbers, like -1 0 0.

Setting Harmonics in QPSS to get Accurate QPAC Results

The qpss analysis, when shooting is selected, has a minimum of 200 timepoints in each slice, so the large signal in qpss inherently has frequency domain content through the 100th harmonic of the large signal input. Only the harmonics that are specified in the moderate tone are included in the solution. The impact of this is that the qpac mixing products that are produced from mixing with the large signal are almost always correct, but the mixing with the moderate signal might or might not be accurate, depending on the number of harmonics that are specified for the moderate signal, and the number of harmonics actually produced by the system. Like harmonic balance, you need to run the qpss and qpac with a number of harmonics specified for the moderate tone(s), and make the measurement in qpac. Then increase the number of moderate harmonics by about 50%, and run again. Then note the measurement. If the result changed significantly, then you need more moderate harmonics. If the result did not change, you might be able to reduce the number of moderate harmonics. Use the smallest number of moderate harmonics that produce a stable qpac result.

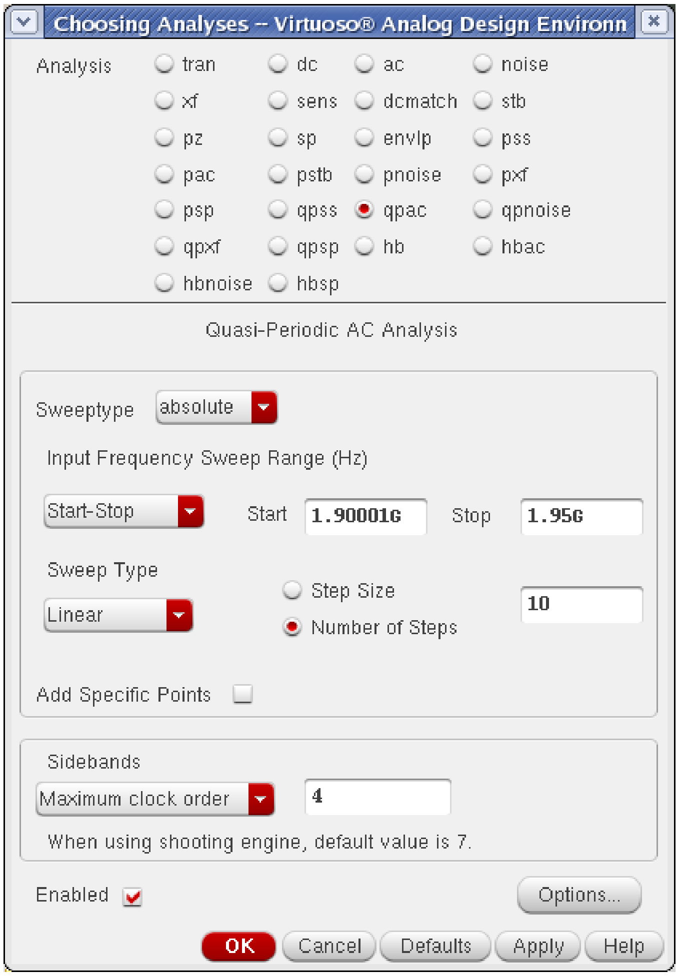
ADE Implementation

In this section, just the qpac analysis *Choosing Analyses* form and the options for qpac are shown. An examples section follows with all the steps needed for a simulation.

To open the *Choosing Analyses* form, select *Analysis - Choose* in ADE, or click he *Choosing Analyses* icon (

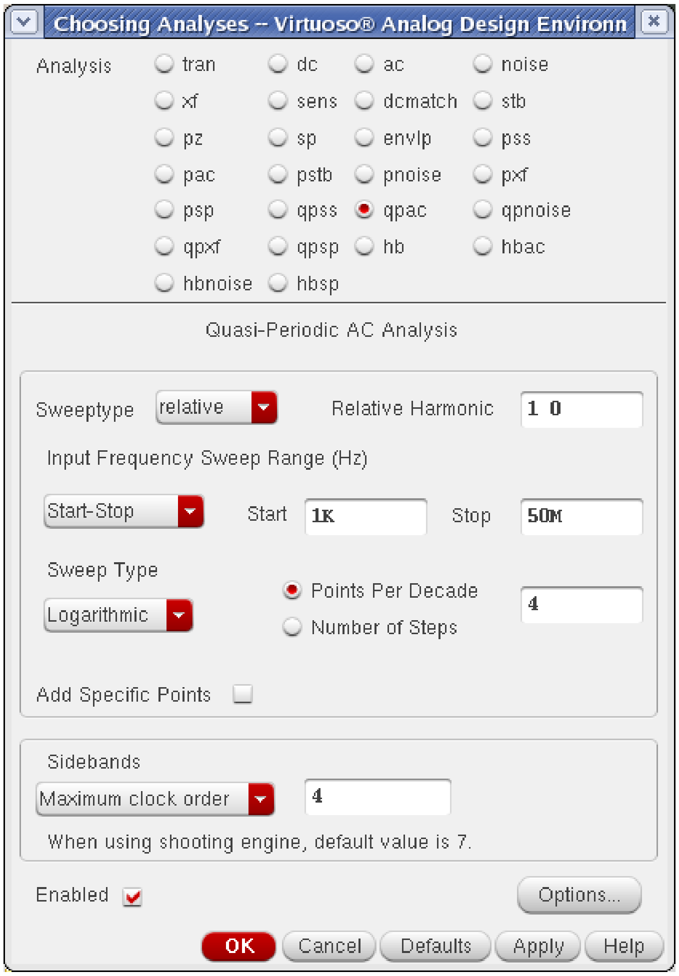
)on the upper right of the ADE window.

The qpac *Choosing Analyses* form appears.



The frequency range is the input frequency range. It is usually better to select Linear or Log spacing, and define your own sweep because auto is likely to take more points (and time) than you need. *Maximum clock order* specifies the highest qpss large signal harmonic to calculate mixing products for.

If you need a log sweep around a harmonic, set the Sweeptype to *Relative*.



The number of numbers in the relative harmonic must be equal to the number of qpss tones. If there are 2 qpss tones, there should be two numbers in the relative harmonic. The first number is the harmonic number and direction that the qpac tone mixes with for the large tone in qpss. The second number specifies the harmonic number of the moderate tone that the input frequency mixes with. Generally, the first tone is the LO tone, and the second tone is the RF blocker. Generally, we design to mix with he LO, and not mix with the blocker, thus the first number is usually 1 or -1 because generally we mix with the first harmonic of the LO, and generally the second number is zero because we don’t deliberately mix with the RF blocker.

QPAC Options

None of the qpac options are commonly used.

tolerance

Leave this option at the default value.

Qpac uses an iterative solver to calculate the output amplitudes. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough, and the tolerance option specifies that accuracy for the qpac analysis. For shooting, the default tolerance is 1e-9. When harmonic balance is the qpss engine, the default is 1e-6.

gear_order

Do not change this option.

solver

The default solver is the turbo solver.

When hb is used as the engine in qpss, leave this option at the default.

When shooting is used, sometimes when the qpac input frequency is very close to the frequency of one of the harmonics in the qpss, warning messages will appear in the qpac output warning that the accuracy might not be good enough. If you see these messages, select the std solver, which has better ability to handle frequencies that are very close to a harmonic in the qpss solution, but which takes longer to run than the turbo solver.

lnsolver

Leave this option at the default. Each qpac sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

resgmrescycle

Leave this option at the default. For the resgmres linear solver, there are several different options.

hbprecond_solver

This option is only available only when harmonic balance is selected in qpss.

The basic solver is the only solver available in standard Spectre. *autoset* is the default solver when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the *basic* solver, and prints a message in the Spectre output window.

annotate

Selections to the left produce less qpac output in the Spectre output window. No detail is provided when you select no, while more details are provided as you move towards right with steps option providing maximum detail.

freqaxis

freqaxis specifies whether you want to see the negative frequency axis or not.

The analysis calculates the output frequencies and amplitudes, so *out* and *absout* are reasonable choices. *out* displays the negative frequency axis. *absout* (absolute value of the output) displays positive output frequencies. This is the default.

If you select *in*, all the outputs at the different frequencies are plotted on the same input frequency range scale. This is not recommended.

save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished using the *Outputs - To Be Saved - Select On Schematic* and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.

nestlvl

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer.

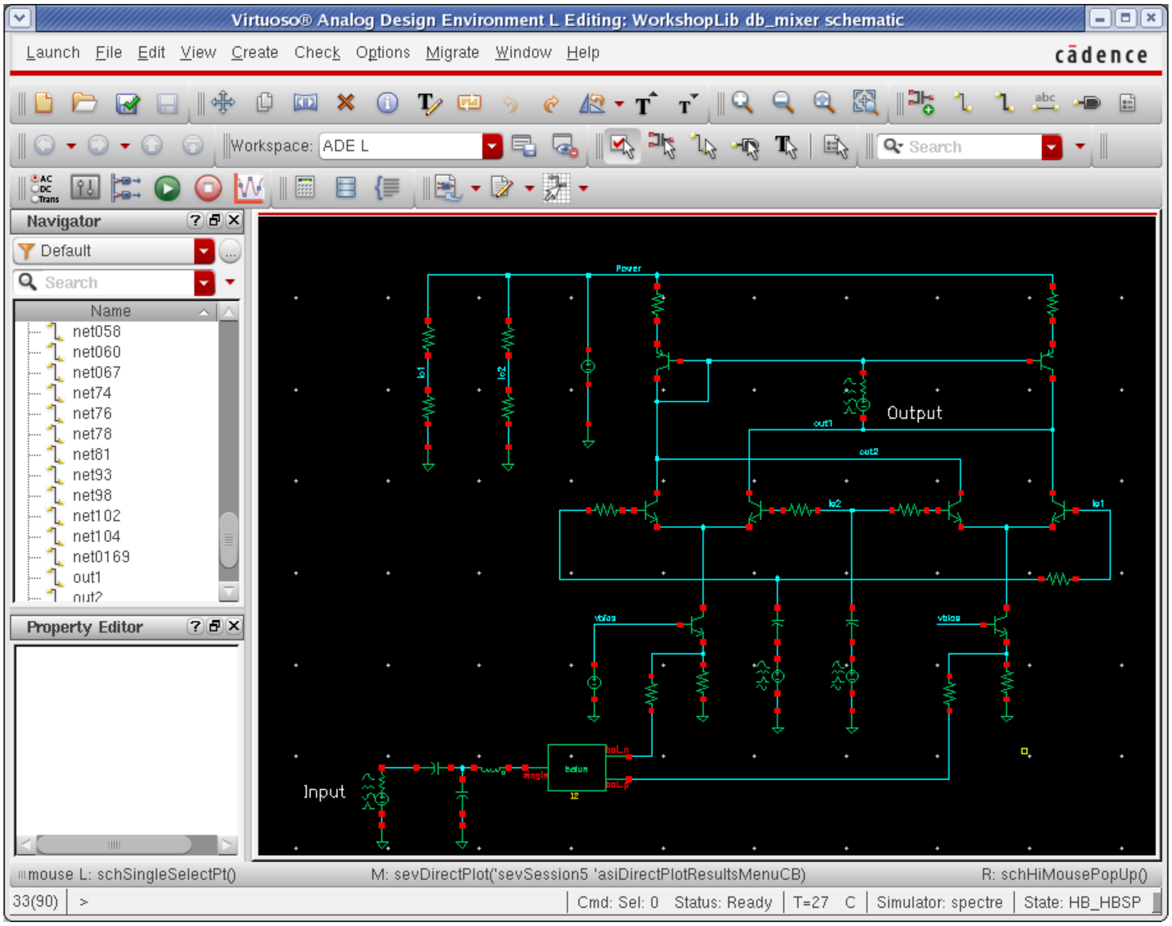
additionalParams

additionalParams is typically used for new features that are being beta tested. It is a field where `keyword=value` pairs are entered.

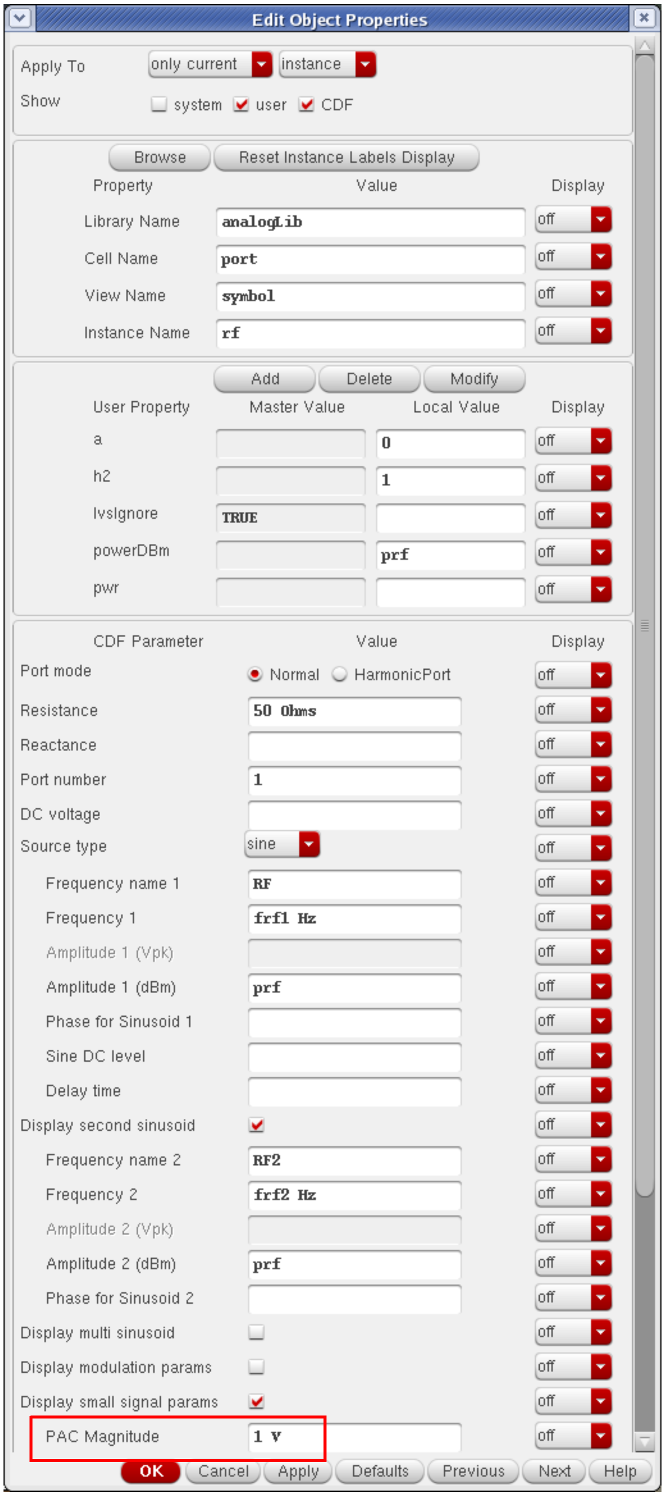
For more information about the other options, type `spectre -h qpac` at the command prompt in a Unix shell window

Example

Consider a mixer circuit below.

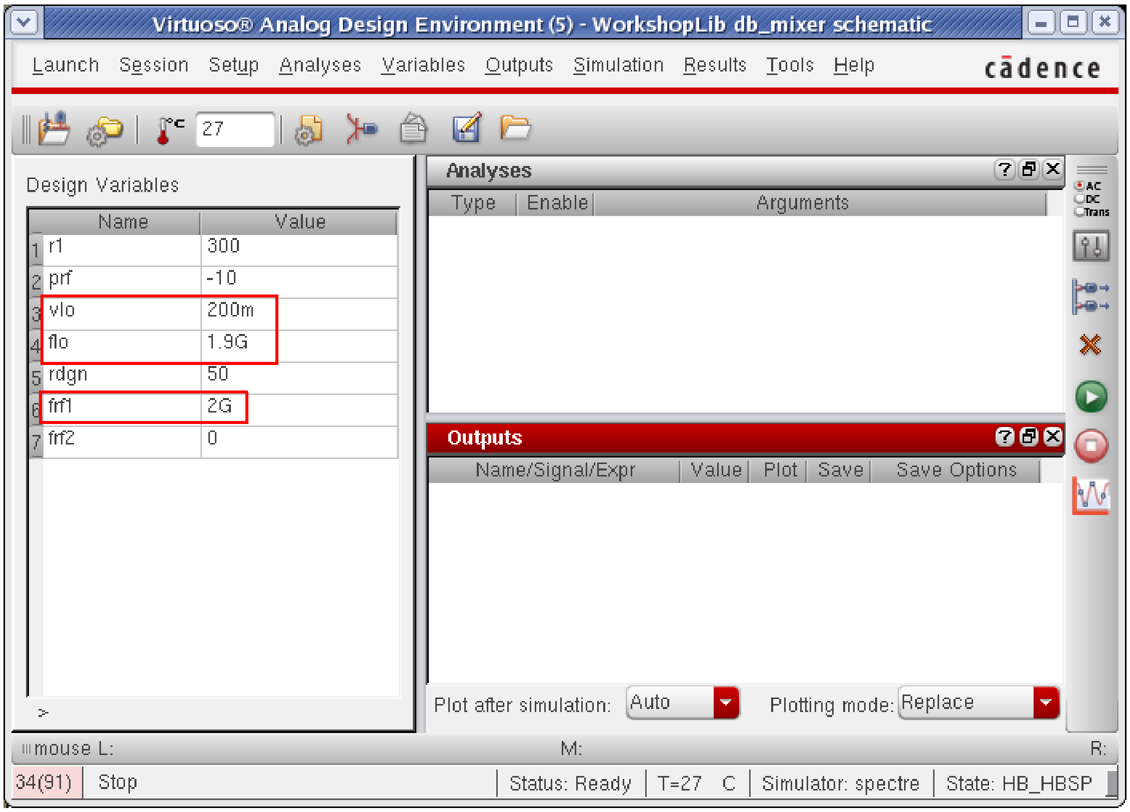


The RF source properties list is shown below.



PAC magnitude controls the amplitude of the qpac input signal. Setting the amplitude to 1 volt allows the conversion gain to be read directly from the output waveform.

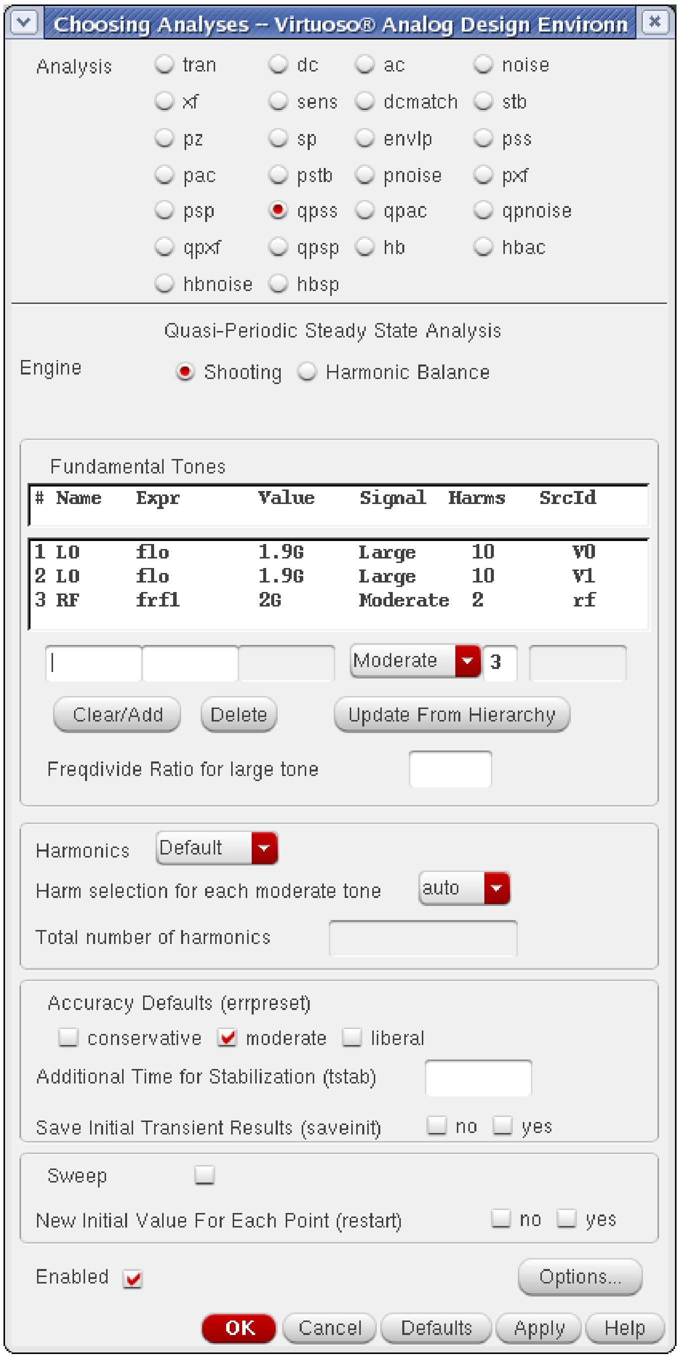
The large-signal properties are set to variable names so the frequencies and amplitude can be controlled using variables in ADE.



In this example, the LO is at 1.9GHz, and the RF input is at 2GHz. The second RF input is disabled for the qpss analysis by setting its frequency to zero.

1. Now select *Analysis - Choose*, or click the *Choosing Analyses* icon ()in the ADE window.

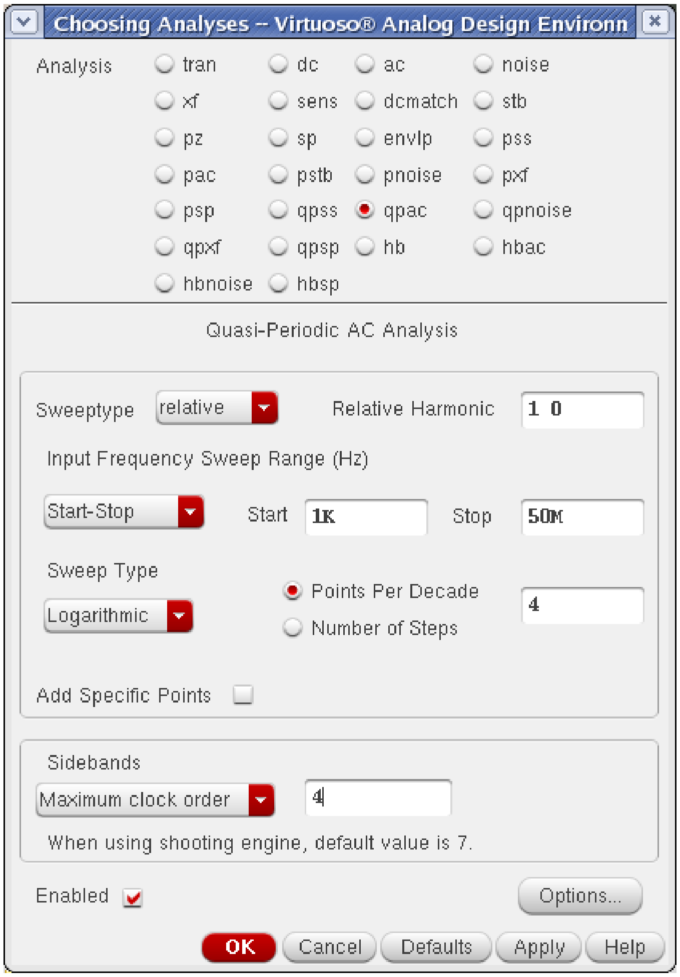
2. Select qpss. The qpss analysis must be run before a qpac can run. For more information on setting the qpss form, see the qpss section at the beginning of this chapter.



3. If there is a non-sinusoidal source, it must be set to large. This is a requirement even if the non-sinusoidal signal produces only a small amount of distortion. Otherwise, the large signal should be the signal that causes the most distortion in the system.

4. Set harmonics large enough to capture the actual harmonics that are produced by your circuit.

5. Now select *qpac*.



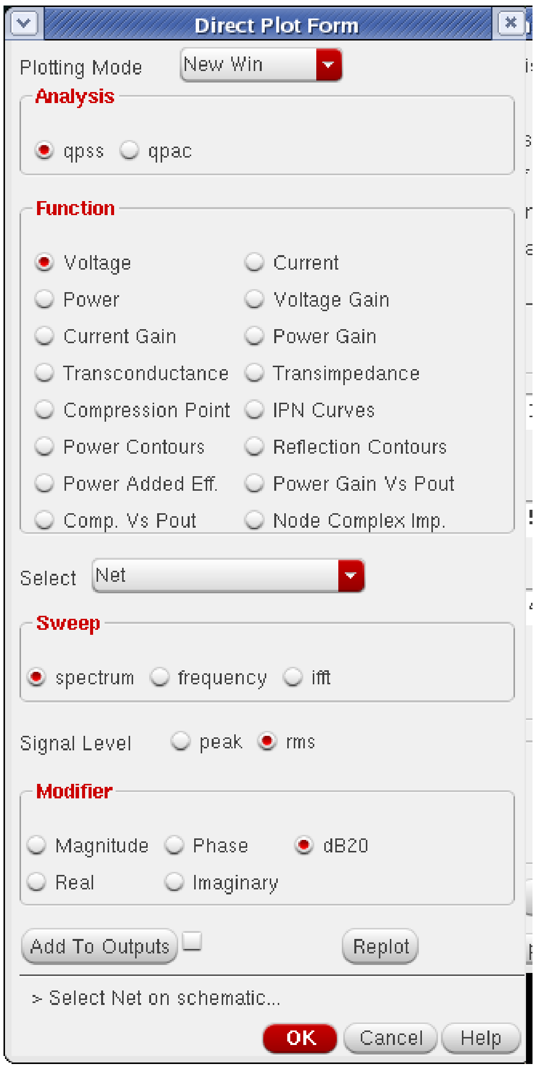
6. This setup causes a log sweep just above the first harmonic of the LO signal. The qpss analysis has 2 tones in it, so the *Relative Harmonic* field has two numbers in it. The first number is the harmonic number of the large signal that is being mixed with, and the second number is the harmonic number of the moderate tone that is being mixed with. 1 0 causes the input frequency to be shifted up by mixing with the first harmonic of the LO signal. This puts the input frequency just above the LO frequency.

7. It is usually better to select a Linear or Log sweep and specify your own frequency points because automatic will likely take more points (and runtime) than is needed.

8. *Maximum clock order* specifies the highest harmonic of the LO tone to calculate outputs for. If you do not need to measure high order mixing products, this can be set fairly

small. If you use the harmonic balance engine in qpss, it is suggested that you set *Maximum sideband* to the same value used for the number of LO harmonics in the qpss, or leave the field blank, which does the same thing.

9. Run the simulation. When the simulation completes, select *Results* - *Direct Plot* - *Main Form*. The *Direct Plot Form* is displayed.



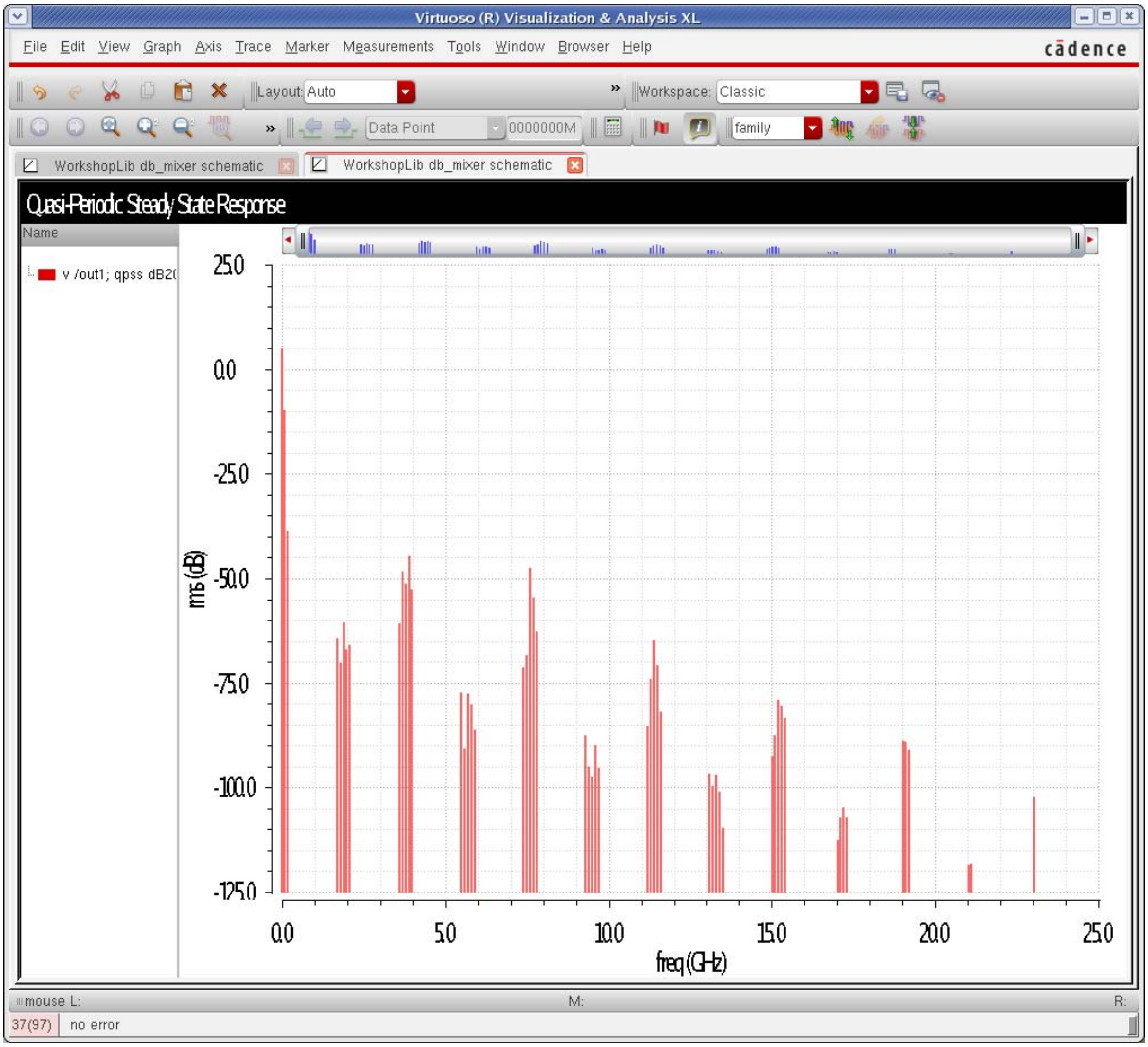
10. Select *qpss* from the *Analysis* section.

11. Select the output net in the circuit.

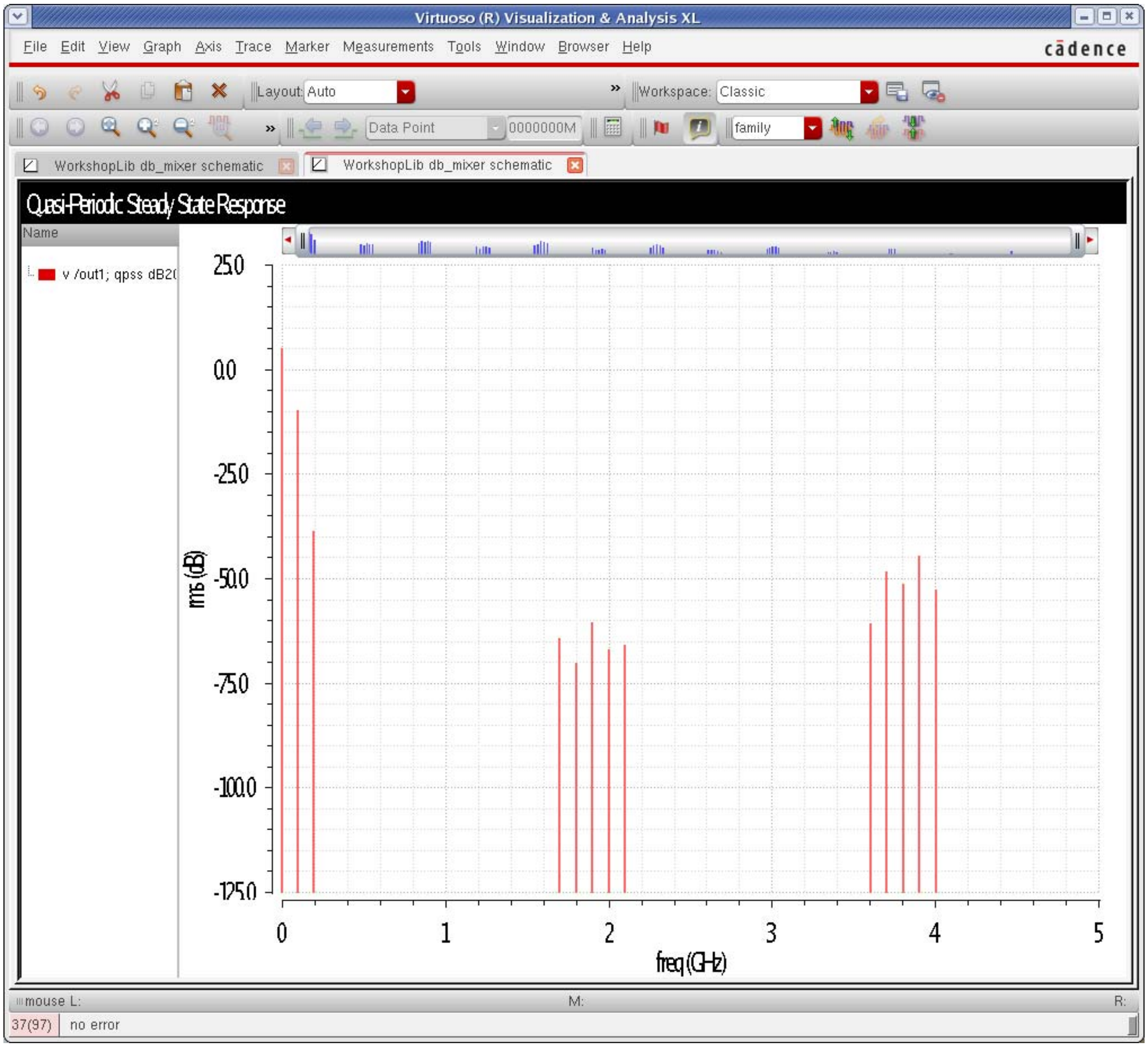
12. Click OK.

The spectral plot is shown below.

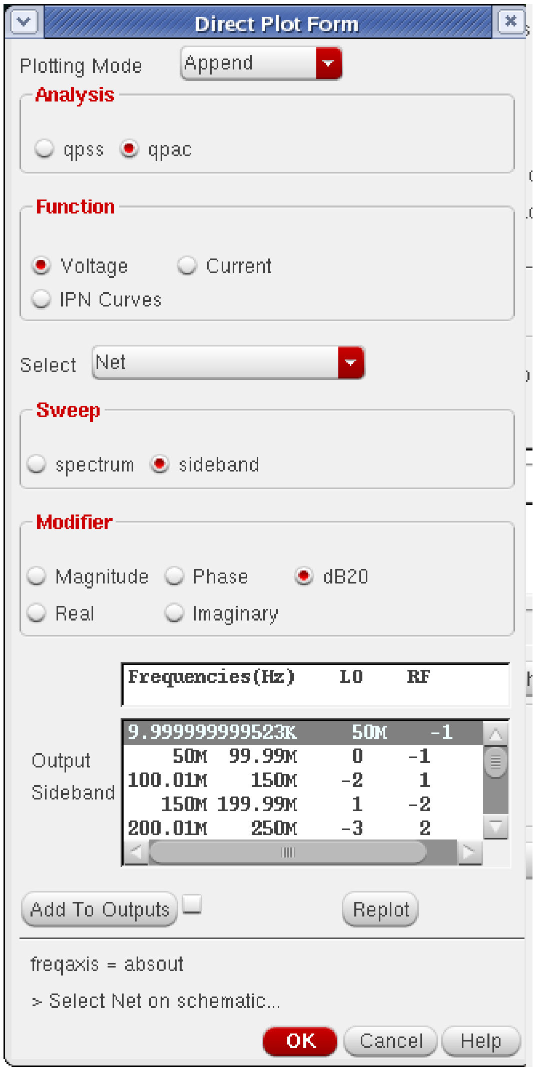
.



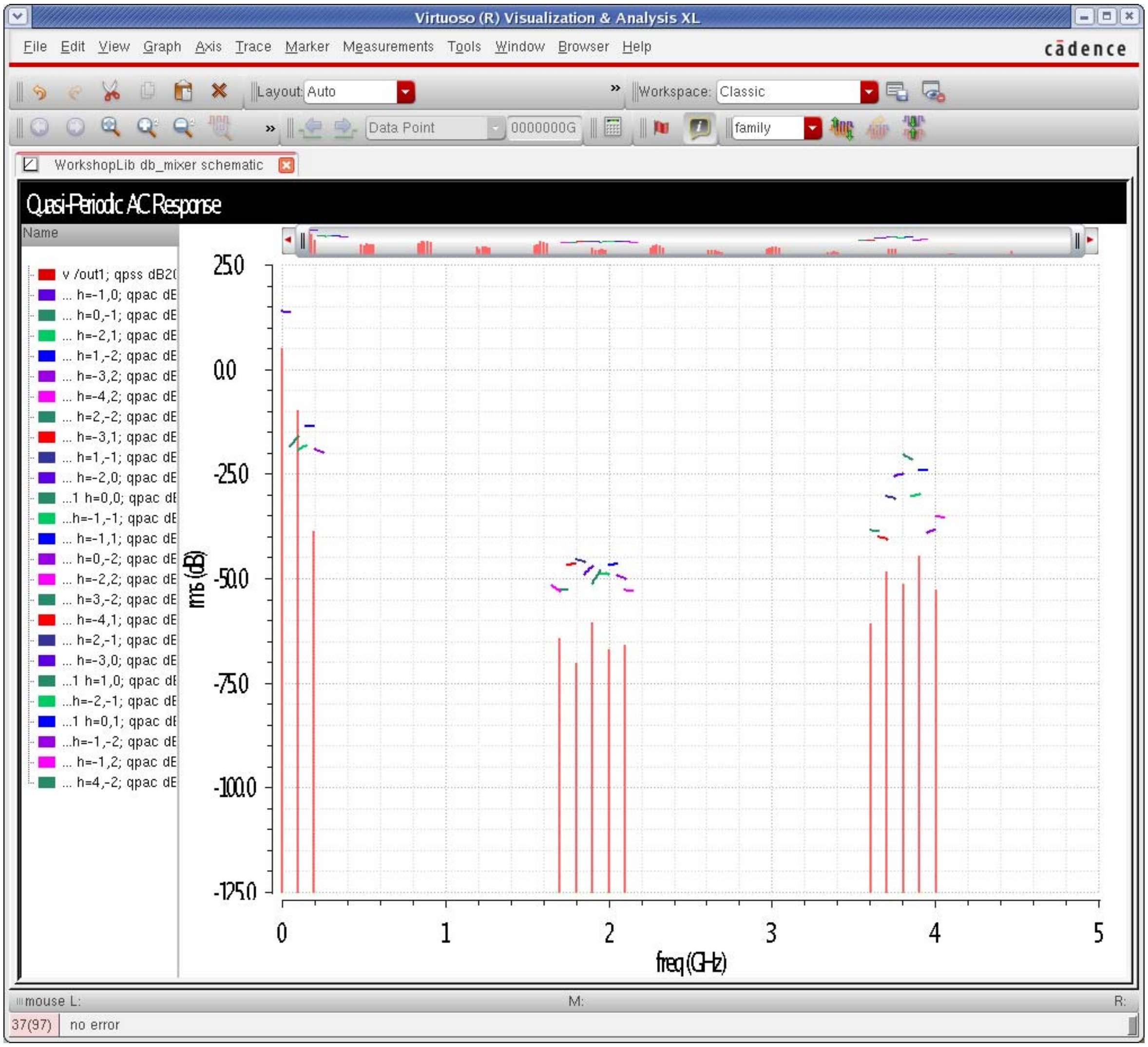
13. In this example, the display has been zoomed in for more detail, as shown below.



14. In the *Direct Plot Form*, select *qpac* from the *Analysis* section.

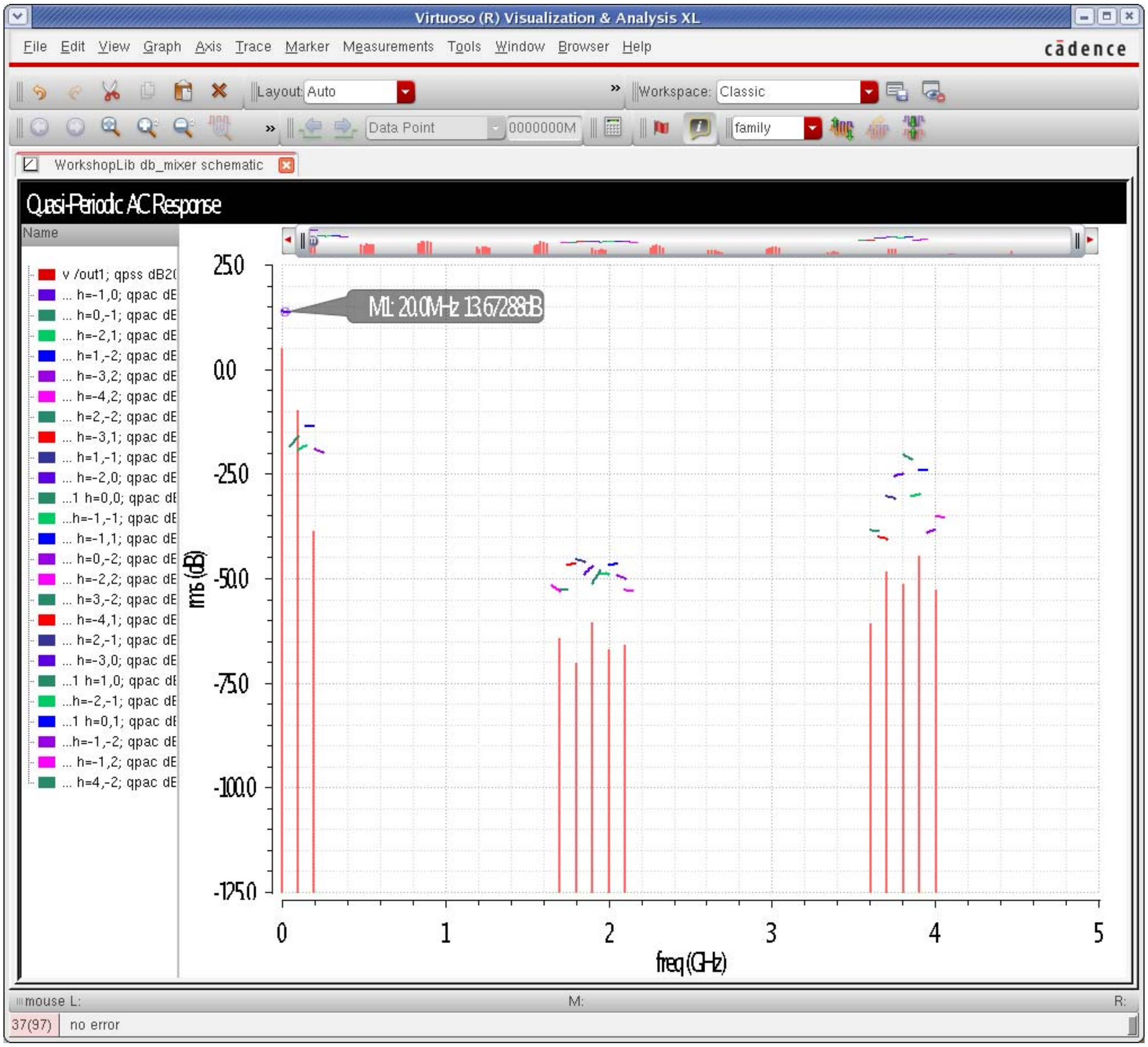


15. Select *sideband* from the *Sweep* section. This allows the plotting of the individual output mixing products. Select the term you want from the list, and select the desired net in your circuit. The output of all the sidebands in range of the X axis is shown below.



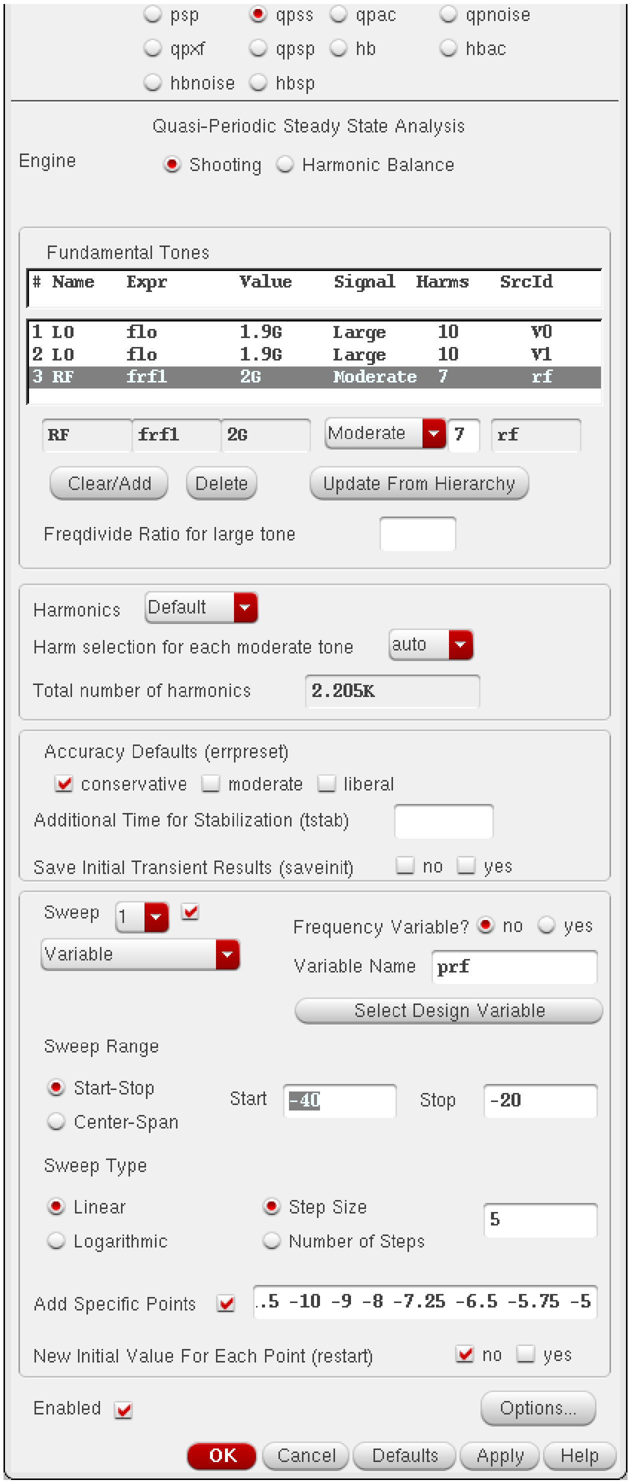
16. The qpac output has a larger amplitude than the qpss output because the qpac input amplitude is set to 1 volt, and the qpss amplitudes are much smaller.

17. To measure conversion gain, go to the desired output frequency, and position a marker as shown below, or use the tracking cursor in the waveform tool.

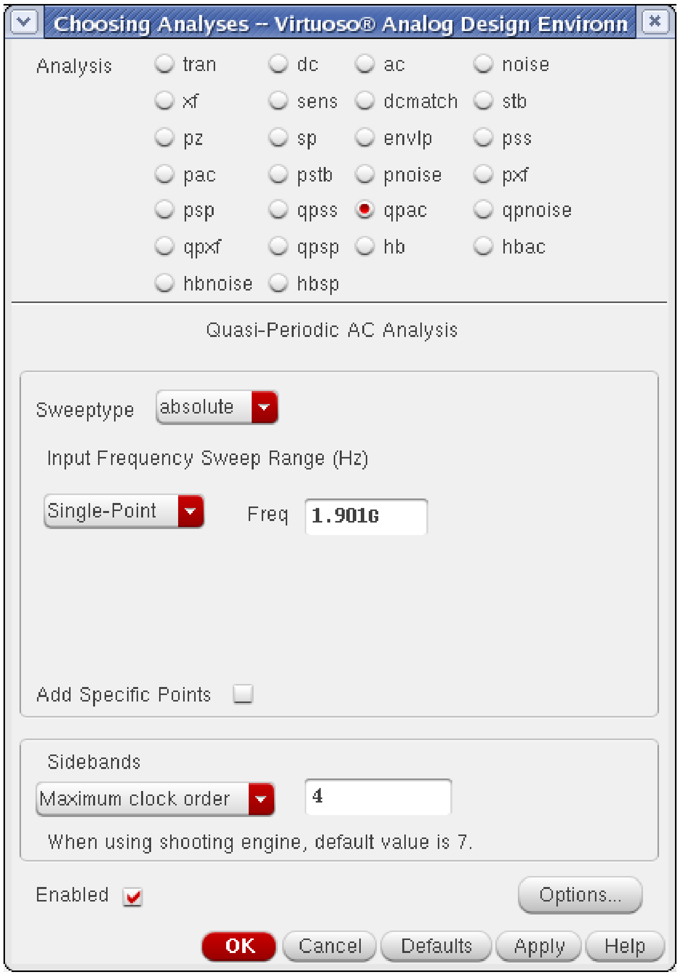


18. The conversion gain is about 13.7dB.

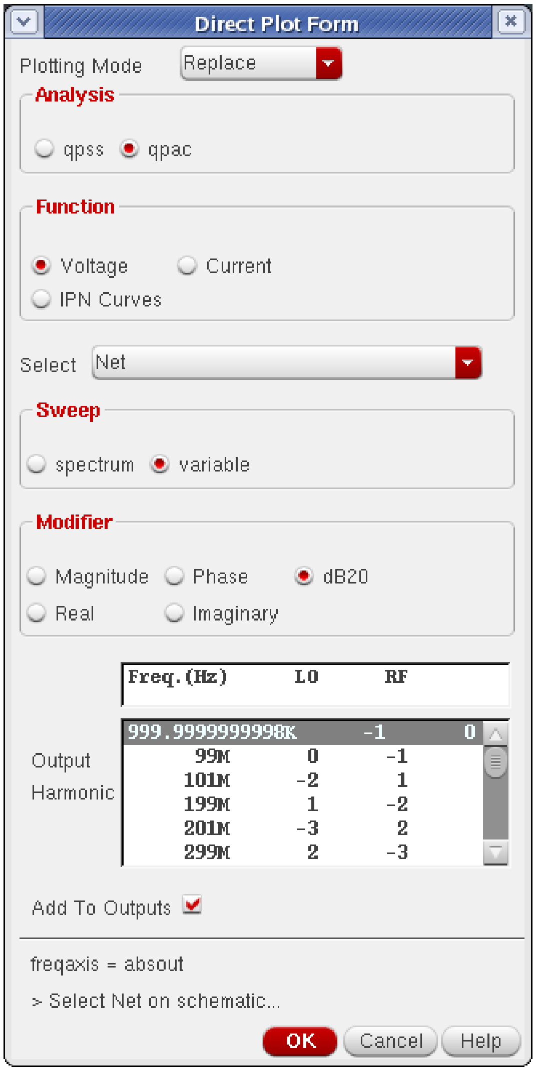
19. Now sweep the input power of the RF blocker in the qpss analysis. Select the check box next to *Sweep*, and specify a sweep range that is appropriate for your system.



20. Set the qpac analysis to a single frequency, and set the input frequency in the passband.

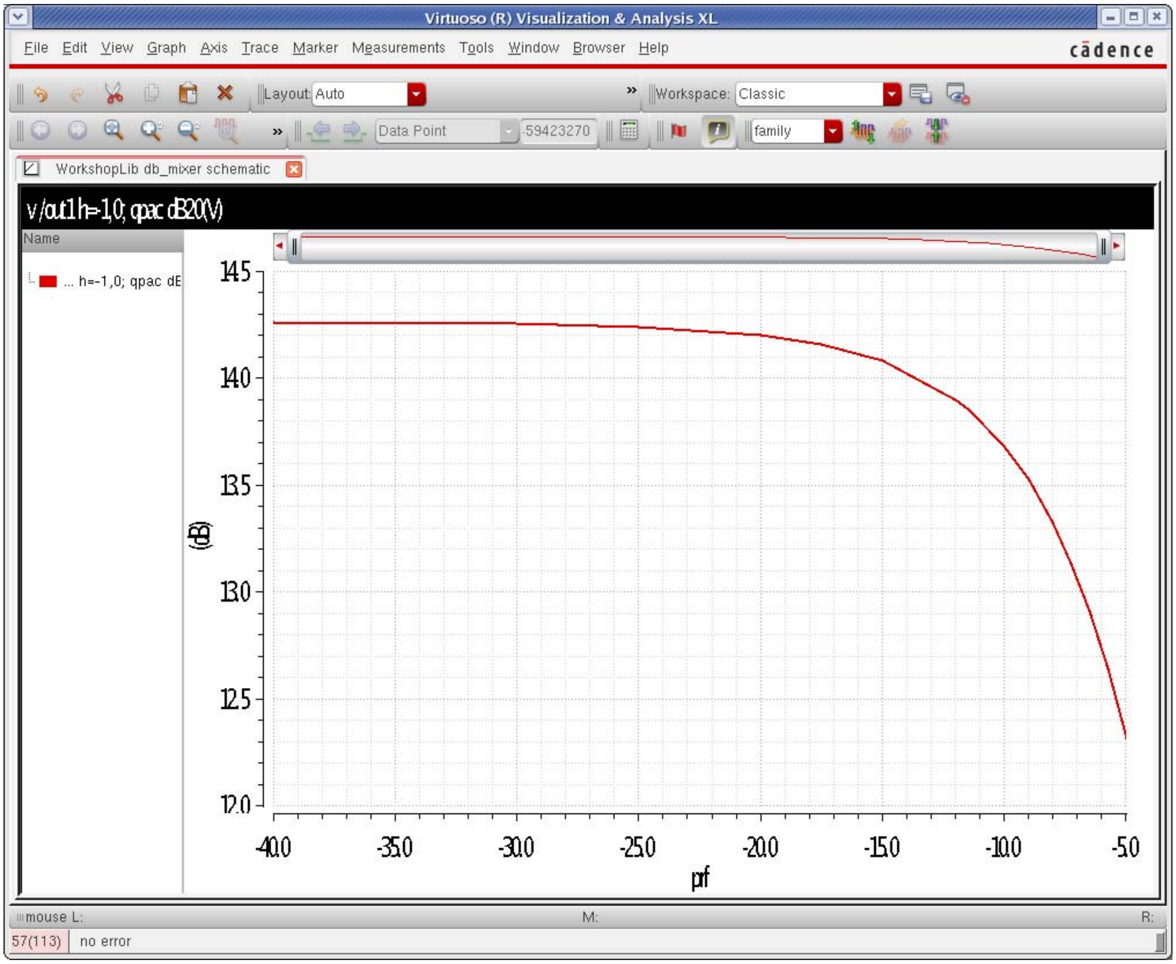


21. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main form* in the ADE window. The *Direct Plot Form* is displayed.



22. Select *variable* from the *Sweep* section, and select the output frequency. Usually, dB20 is appropriate. Now select the output net in the schematic.

23. The Waveform displays. As expected, the conversion gain goes down when the blocker amplitude gets large.



Quasi Periodic Noise Analysis

Overview

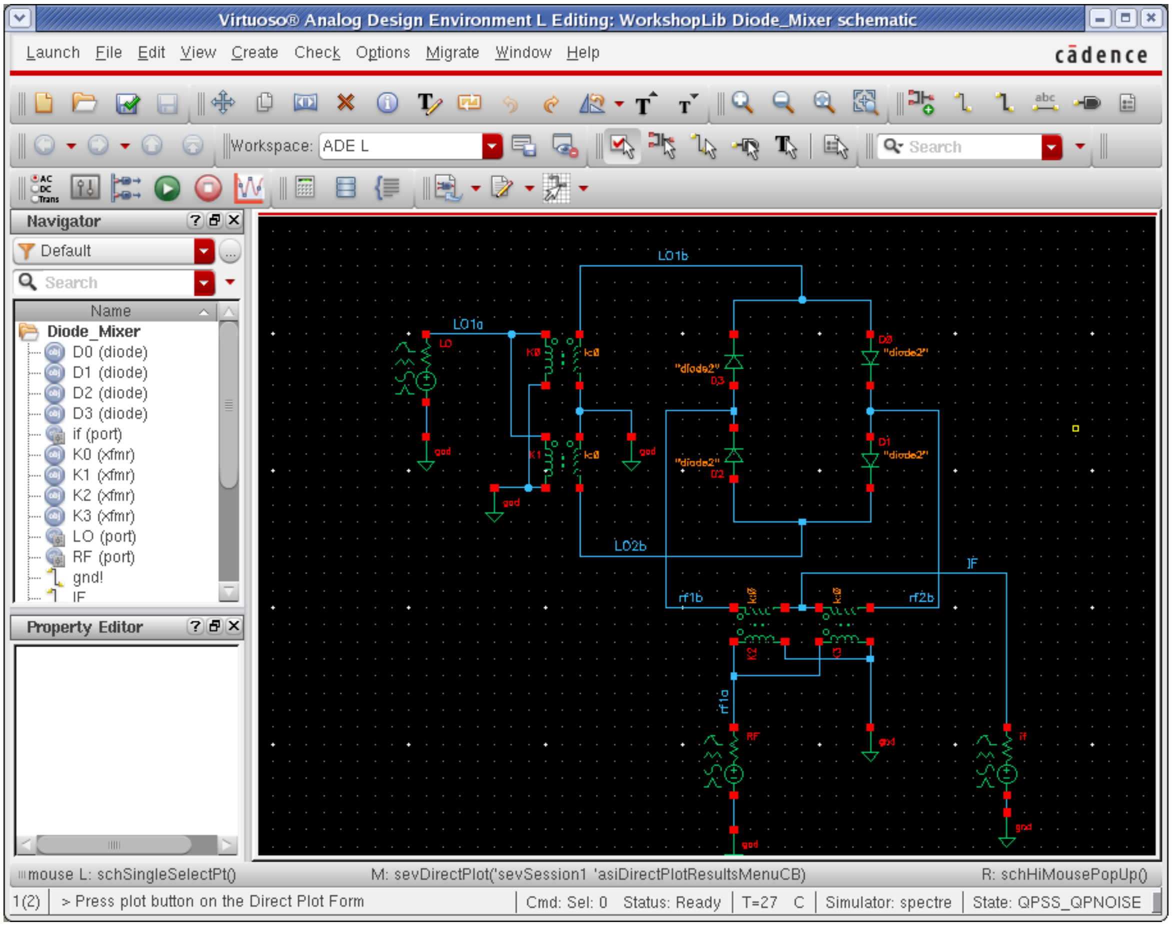
The concepts of qpnoise analysis are exactly the same as pnoise analysis, except that instead of running a pss analysis to capture the nonlinearity of the circuit, qpss captures the nonlinearity of the circuit. In qpss, there are multiple inputs which cause many harmonics to be produced by the circuit. All of these frequencies in the circuit can mix with noise frequencies to be converted to the output frequency of the qpnoise analysis. Qpnoise calculates all the frequency translations from all the harmonics in the qpss analysis, and sums the noise power at the output of the circuit.

The normal application of qpnoise is to measure the noise figure degradation of lna-mixer chains with a large amplitude RF blocking signal applied to the circuit.

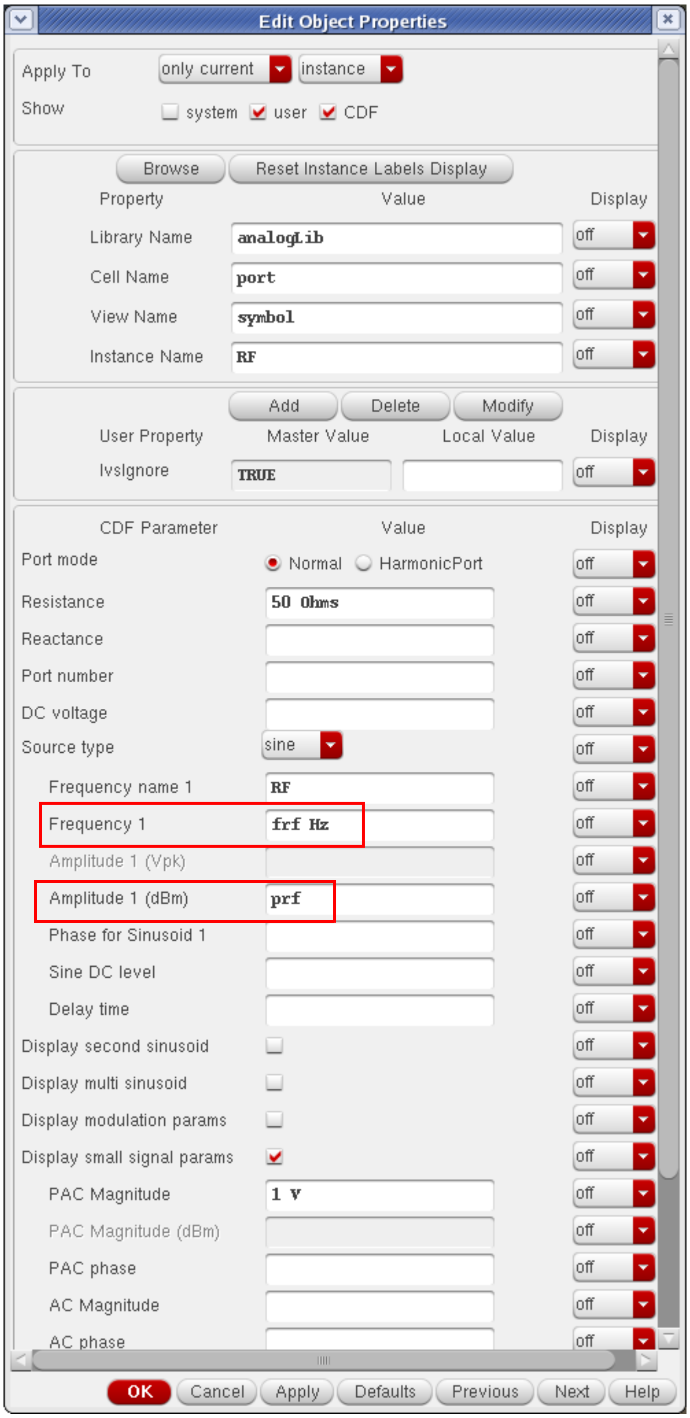
Note that for this application, harmonic balance is likely to run faster than shooting. If you need harmonic balance, the hb and hbnoise analyses are strongly recommended because these analyses always reflect the latest features that are available in the harmonic balance simulation.

Example

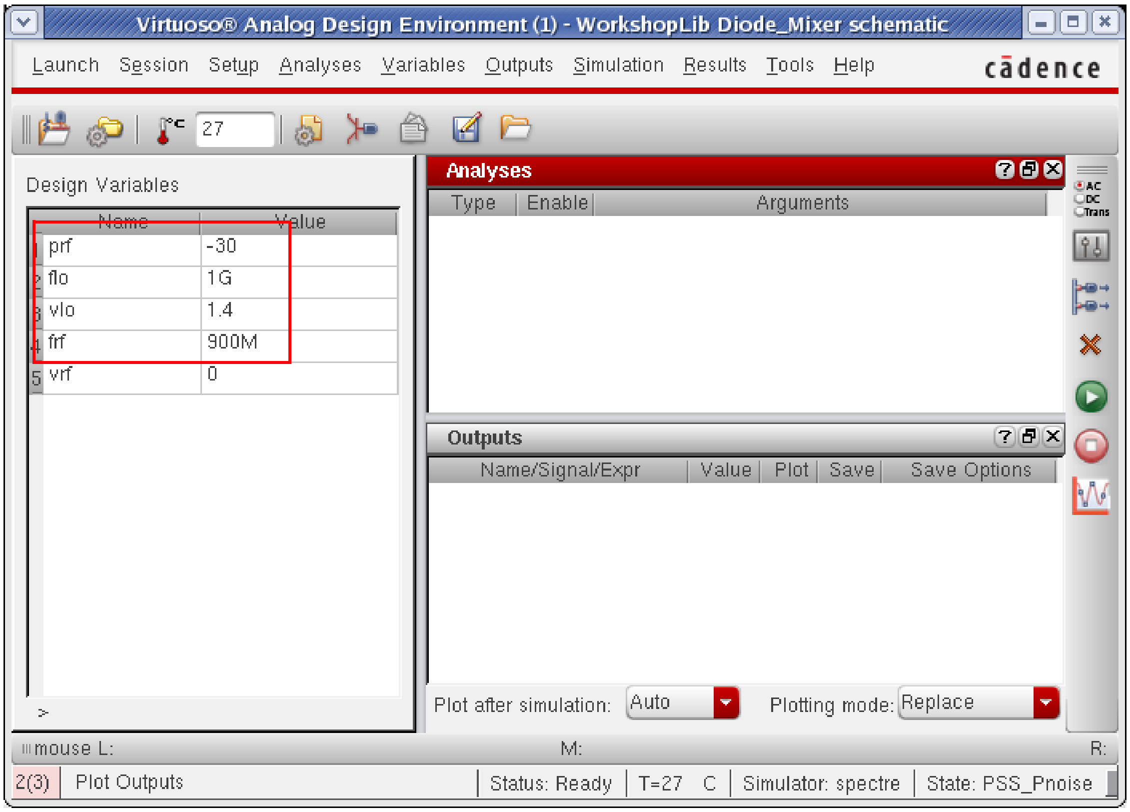
Consider a diode mixer below.



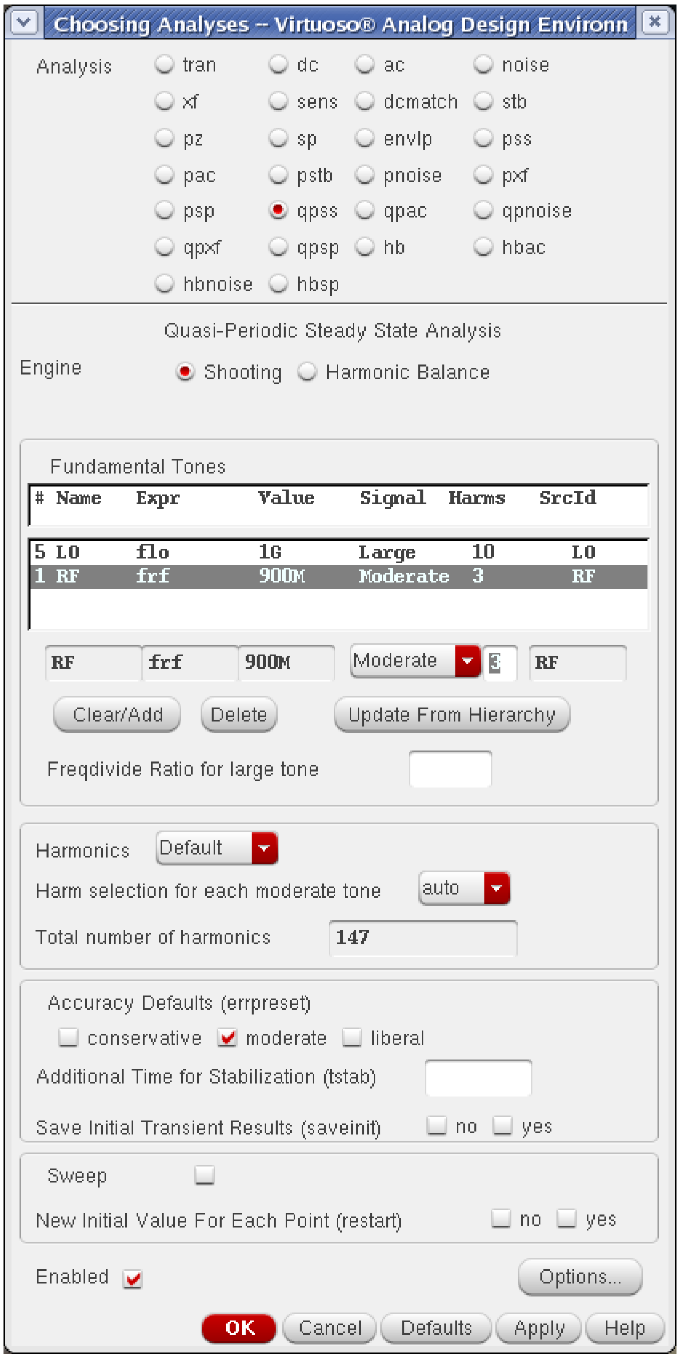
The input port has the frequency and amplitude set using variable names. This is done in order to allow frequency and amplitude changes in ADE without the need for changing the schematic.



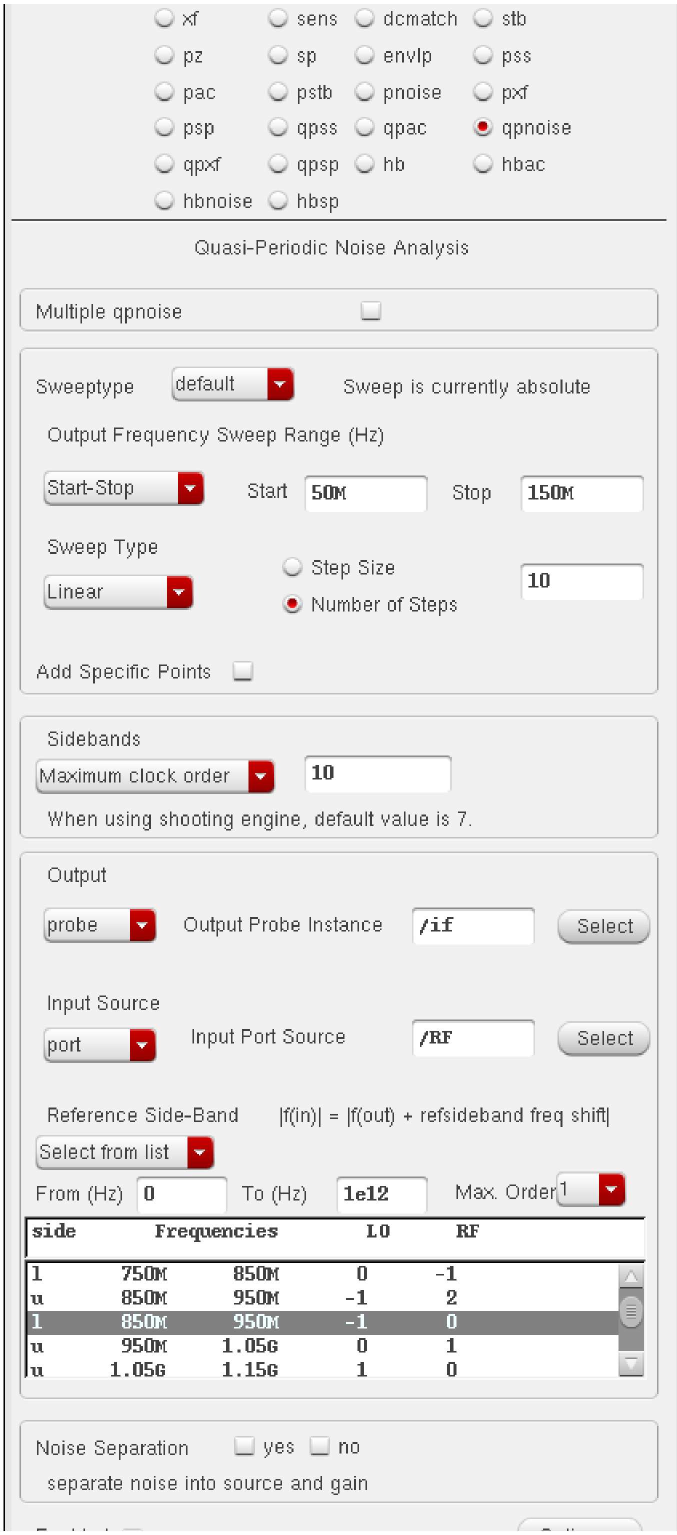
In a similar manner, the LO port uses variable names. These variables are set in the ADE window.



1. Next, the qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter. In this example, a 1GHz LO signal and a 900MHz blocker are applied.



2. Next, the qpnoise analysis needs to be set up.

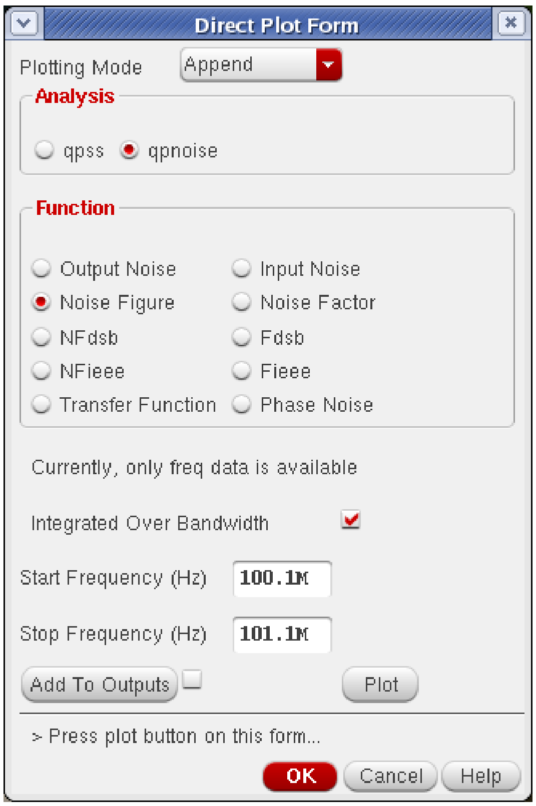


3. The output is from 50MHz to 150MHz with a linear sweep. Because 10 harmonics were set in the qpss analysis for the LO tone, 10 is set for *Maximum clock order*. These numbers should agree. If harmonic balance is the engine in qpss, this field can be left blank, which will cause all the harmonics in the qpss analysis to be considered in qpnoise.

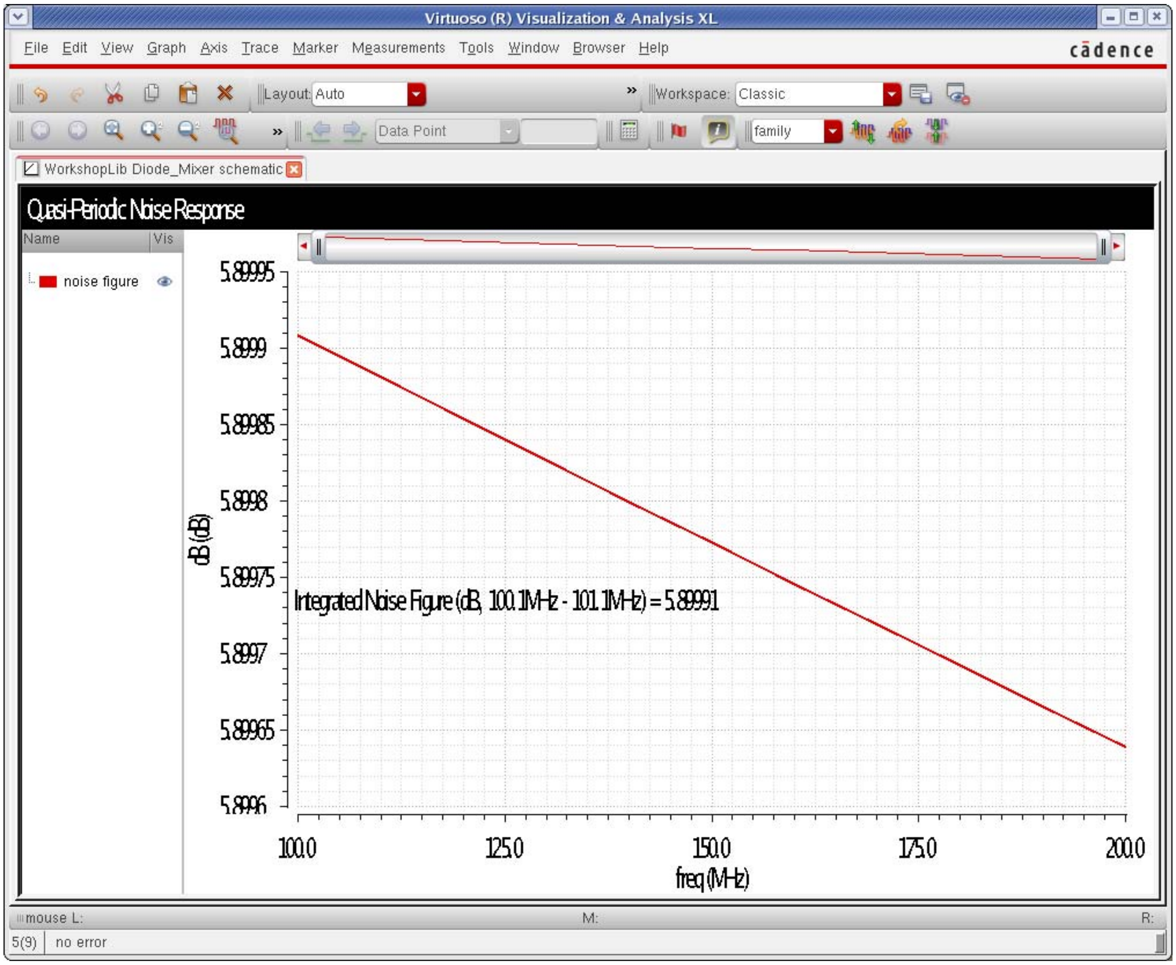
4. The output port is set. This subtracts the noise from the output port from the noise figure calculation. The input port is also set. This is required if you want a noise figure calculation.

5. The reference sideband defines the passband frequency for the noise figure calculations. The passband for this case is just under the LO input at 1GHz.

6. Now run the analysis. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* is displayed.



7. The single-sideband noise figure is selected. Click *Plot*.



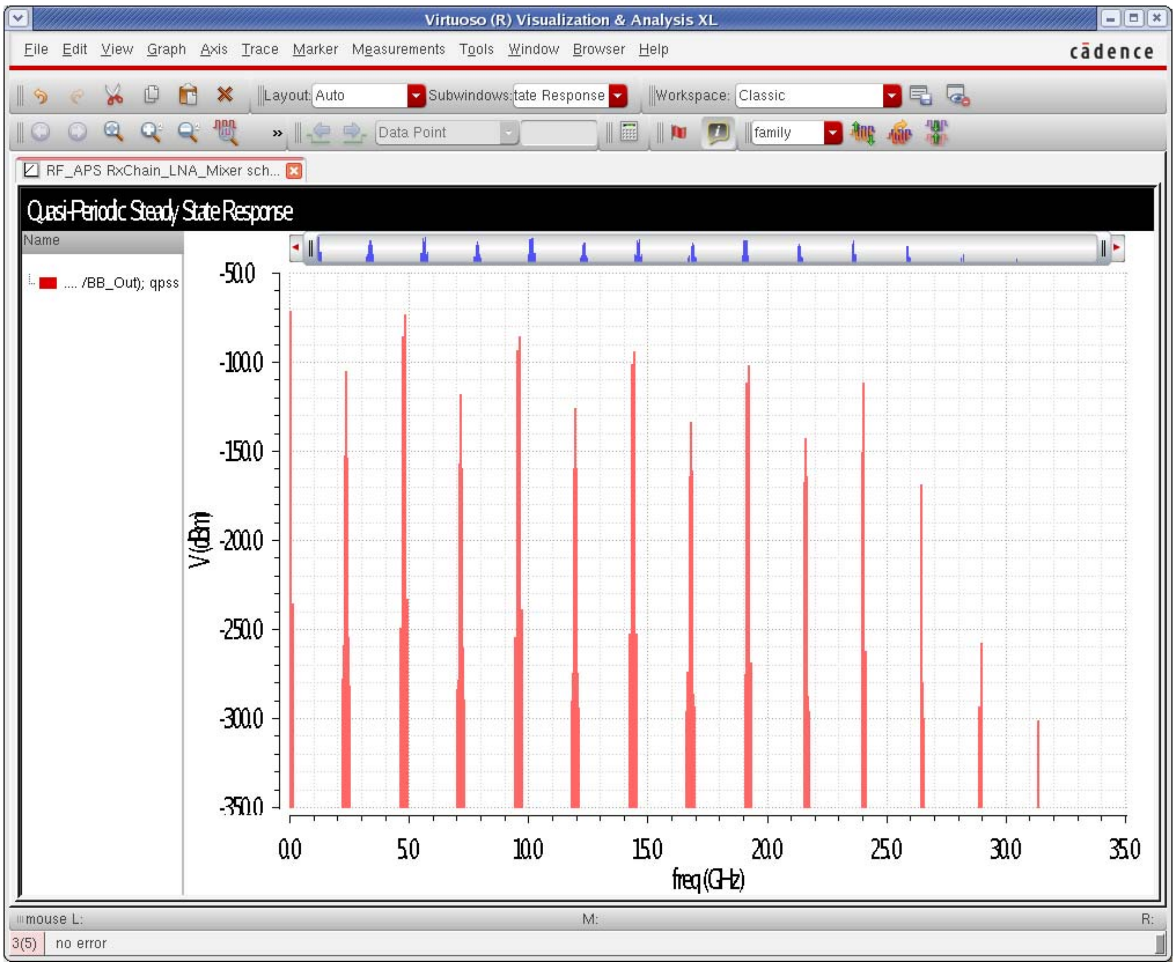
As expected, the noise figure is very near 6dB, which is the conversion loss for the mixer.

Qpnoise Concepts

Unlike pnoise and hbnoise, qpnoise has only one mode. Qpnoise can only sum the noise power at the output, and refer that noise back to the input. Modulated and sampled functions are not available. If you need those functions, use either pss-pnoise or hb-hbnoise.

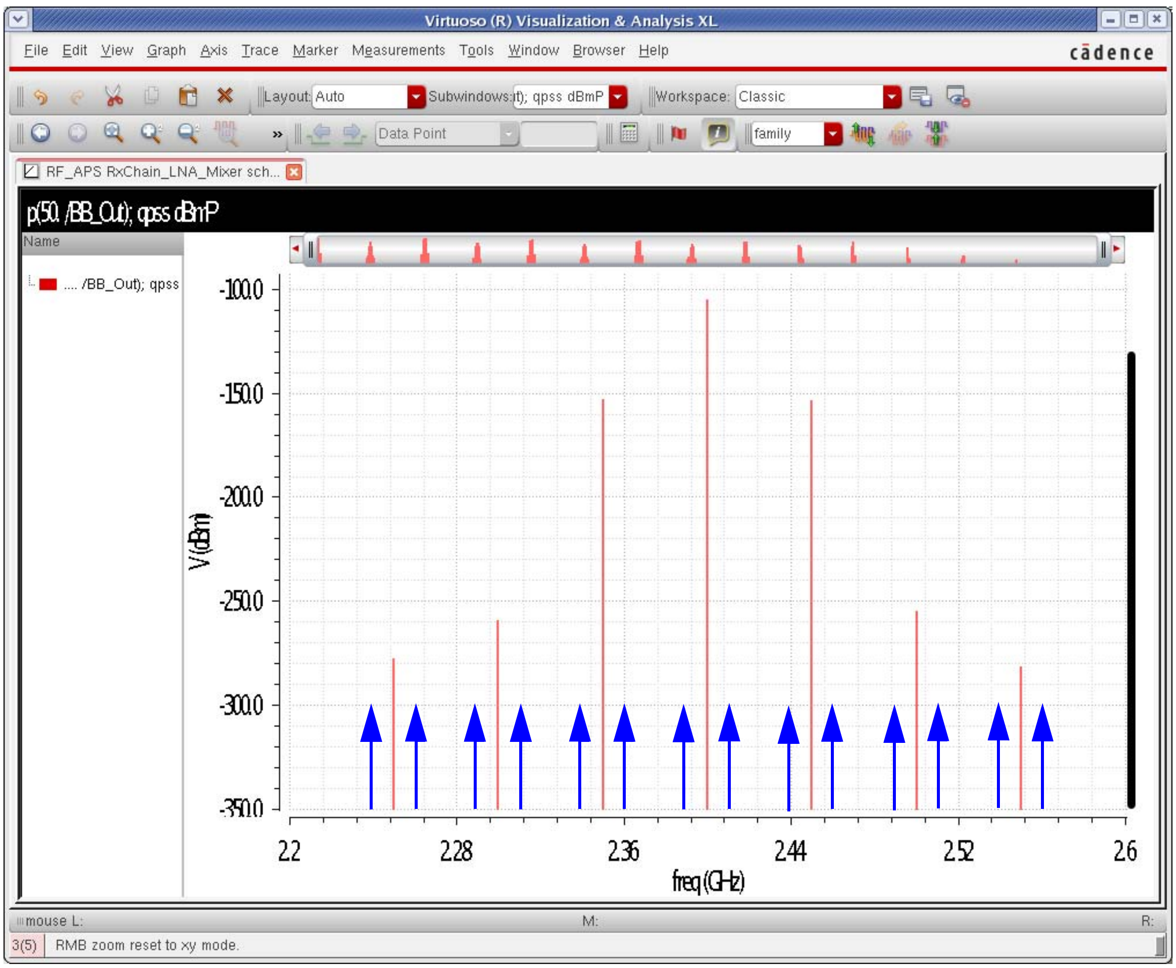
Because there are many harmonics created in the qpss analysis, many transfer functions need to be calculated from the different noise frequencies to the output.

Consider an LNA-Mixer with a 2.4GHz LO and an RF signal at 2.45GHz. The spectrum at the output of the circuit is shown below.



If the noise output frequency was 1MHz, then noise at 1MHz on either side of each harmonic in the qpss analysis mixes down to 1MHz. The frequency response of each noise source is determined by the qpss solution. This solution also supplies the nonlinearity to qpnoise so qpnoise can calculate the transfer function from each noise source to the output. The noise power at each noise frequency times the conversion gain gives the noise power at the output. All the different noise outputs caused by each noise frequency mixing down is summed, and then this same process is repeated for each noise source in the circuit.

A zoomed in view near the LO frequency is shown below.



The noise 1MHz on either side of these harmonics mixes with each of these harmonics to the output at 1MHz. These frequencies are shown with the blue arrows above. Note that this shows the spectral content at the output of the circuit only. The amplitudes of these signals are much larger in other parts of the circuit. Noise that is 1MHz above and below all the harmonics mix down to the output at 1MHz. Because the noise frequencies are slightly different for all the mixing products, the actual nose power and the conversion gain is slightly different for each mixing product.

Measuring Noise Figure

The formulas for measuring noise figure are the same as those presented in pnoise.

For:

`No` = total output noise
`Ns` = noise at the output due to the input probe at the passband frequency (the source)
`Nsi` = noise at the output due to the image harmonic at the source
`Nso` = noise at the output due to harmonics other than input at the source
`Nl` = noise at the output due to the output probe (the load)
`IRN` = input referred noise
`G` = gain of the circuit
`F` = noise factor
`NF` = noise figure
`Fdsb` = double sideband noise factor
`NFdsb` = double sideband noise figure
`Fieee` = IEEE single sideband noise factor
`NFieee` = IEEE single sideband noise figure

then,

```
IRN = sqrt(No^2/G^2)
F = (No^2 - Nl^2)/Ns^2
NF = 10*log10(F)
Fdsb = (No^2 - Nl^2)/(Ns^2+Nsi^2)
NFdsb = 10*log10(Fdsb)
Fieee = (No^2 - Nl^2 - Nso^2)/Ns^2
NFieee = 10*log10(Fieee)
```

Note that in order to measure noise figure, the passband frequency needs to be identified. Qpnoise calculates many transfer functions from the input to the output. The passband frequency is set in the Reference sideband parameter. The reference sideband is the sideband number that defines the passband frequency. If there are two inputs in qpss, there will be two numbers in the reference sideband. If there are three inputs, there will be three numbers. The easiest way to define the reference sideband is to use the *Select from List* capability, and then select the design input frequency from the list.

Frequency Sweep

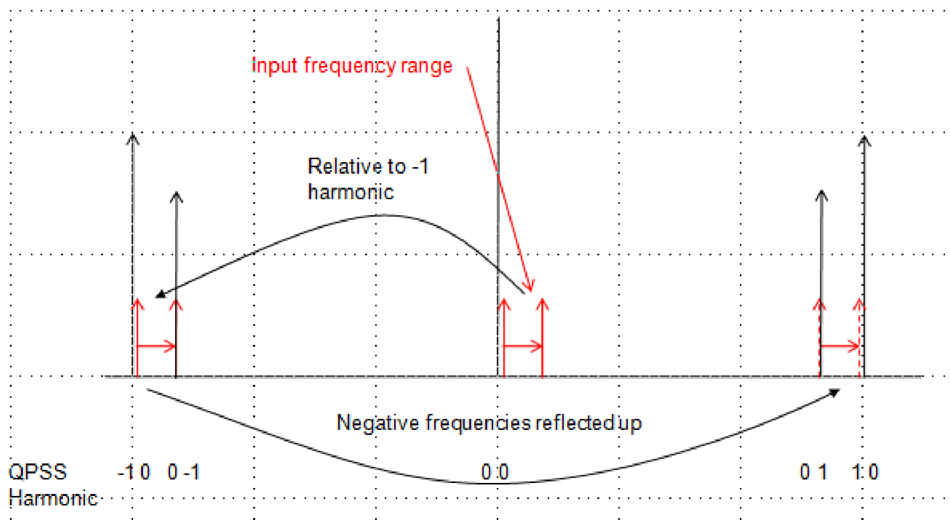
In qpnoise, the fundamental quantity that is calculated is the noise power at the output of the circuit. The Frequency Sweep Range is always the **output** frequency range for qpnoise. The output frequency range is specified in the qpnoise *Choosing Analyses* form.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonic numbers in the qpss analysis.

For example, assume that the qpss has a 1GHz LO input, and a 900MHz RF input. You want a log sweep above the first harmonic of the 1GHz signal in the qpss analysis. In this case, you need to select *relative* sweep, and specify 1 0 in the *Relative Harmonic* field. The first number is the frequency shift in multiples of the LO frequency, and the second number is the frequency shift in multiples of the RF input frequency. Next, type in 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade.

The noise analysis output frequency will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.

To use a log sweep below the first harmonic of the 1GHz input, use the same frequency range, and type in -1 0 as the relative harmonic. This is illustrated below.



Most of the time, the LO signal is the large signal in qpss. This is the first number in the shift. Generally, we do not design to deliberately mix with the RF blocking signal, so generally, the second number is zero. The first number is usually 1 or -1, because generally the RF signal mixes with the first harmonic of the LO. If the first number is -1, this is the signal just below the first harmonic of the mixer. If the first number is positive, it means the design input frequency is above the LO frequency.

Maximum Clock Order

In qpnoise terminology, the large signal from qpss is called the clock. Thus, *Maximum clock order* specifies the highest large signal harmonic number (The large signal is usually the LO signal) to calculate the conversion gain terms for. Usually, the conversion gain is needed for all the harmonics that are present in the qpss analysis, so usually *Maximum clock order* should be set to the number of LO harmonics in the QPSS analysis. Each conversion gain term that is calculated is called a sideband in a qpnoise analysis.

Sideband Numbers

The sideband number is the qpss harmonic number that is mixed with. Because qpss always has multiple input frequencies, qpss has multiple harmonic numbers that are being mixed with. If there are 2 tones in the qpss simulation, there are two numbers in qpnoise that describe which harmonics of which tone are mixing to produce that output frequency. If there are three tones in the qpss analysis, then there are three numbers for each qpnoise sideband.

If the sideband number is negative, the noise frequency is more negative than the output frequency specified in the qpnoise *Choosing Analyses* form. If the sideband is positive, the noise frequency is more positive than the output frequency.

Although this seems complicated, there are features in the *Direct Plot Form* to make this easier to understand.

Conceptually, an infinite number of conversion gain terms exist when a single output frequency is analyzed in a nonlinear system. From the practical point of view, usually only the noise frequencies near all the qpss harmonics need to be measured.

Setting Moderate Harmonics in QPSS to get Accurate QPnoise Results

The qpss analysis, when shooting is selected, has a minimum of 200 timepoints in each slice, so the large signal in qpss inherently has frequency domain content through the 100th harmonic of the large signal input frequency. Only the harmonics that are specified in the moderate tone affect the accuracy of the solution. The result of this is that the qpnoise mixing products that are produced from mixing with the large signal are almost always correct, (Usually the LO tone) but the mixing with the moderate signal might or might not be accurate, depending on the number of harmonics that are specified for the moderate signal, and the number of harmonics actually produced by the system. Like harmonic balance, you need to run the qpss and qpnoise analyses with a number of harmonics specified for the moderate tone(s), and make the measurement in qpnoise. Then increase the number of moderate harmonics, run again, and make the same measurement. If the result changed significantly, then you need more moderate harmonics. If the result did not change, you might be able to reduce the number of moderate harmonics. Use the smallest number of moderate harmonics that produce a stable qpnoise results.

When harmonic balance is selected for the qpss engine, the adjustment of harmonic numbers above also needs to include the LO harmonics. Unlike shooting, where inherently the 100th harmonic is present in the qpss data for the large signal tone, only the number of harmonics actually specified for the LO are present in the solution.

Noise Separation

In addition to the total output noise, the individual noise contributions can be plotted if noise separation is selected in the *Choosing Analyses* form. More information about the noise separation will be provided in the examples section. The things that can be plotted are:

1. Total noise at the output from each individual noise input frequency (sideband). This allows the identification of which noise frequencies are causing the noise problem.

2. The noise at the output from each instance name with all noise mechanisms included for mixing from an individual noise input frequency. Once the noise frequency is identified in point 1 above, this capability plots the total noise from each component at the troublesome noise frequency. This allows the identification of which component in the circuit is causing the problem.
3. The noise at the output from each instance name with all noise mechanisms broken out separately for mixing from an individual noise input frequency. This capability identifies the specific noise sources at the troublesome noise frequency. For example, if a parasitic resistor was found to be the problem, the component might be resized to reduce the noise.
4. The current noise at the noise instance for each instance name with all noise mechanisms included for mixing from an individual noise input frequency. This capability has limited value. For this plot, all the noise source currents for all the noise mechanisms within the component are added together.
5. The noise current in the instance from each instance name with all noise mechanisms broken out separately for mixing from an individual noise input frequency. This capability allows the identification of which noise sources are the largest in the circuit. The measurement is at the source, not at the output.
6. The gain to the output from each individual instance with all the individual noise sources broken out separately for mixing from an individual input frequency. This capability plots the gain from the noise sources in point 5 above to the output of the circuit.

Input-Referred Noise

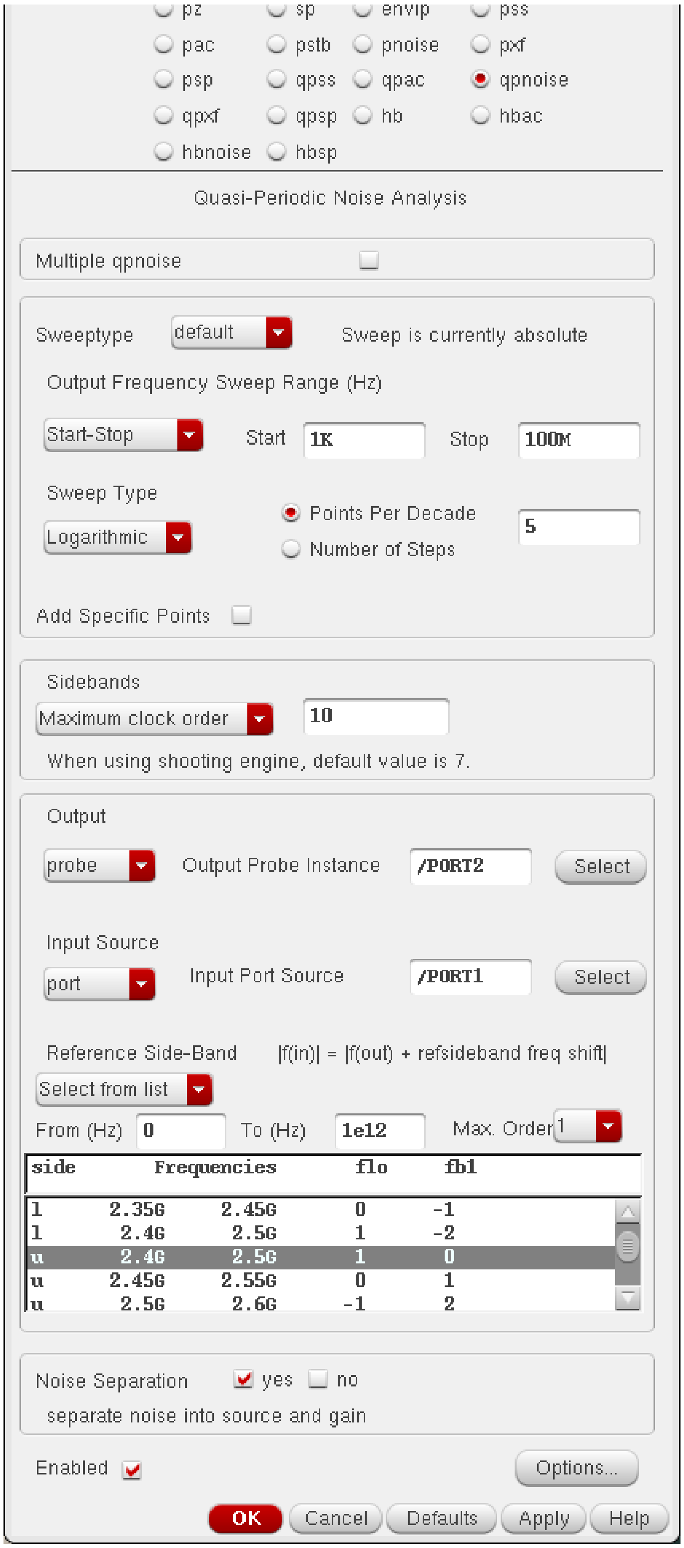
Important

When the qpnoise analysis is set up with the input source being a port, input-referred noise may not be exact. In this case, the input-referred noise is referred to the voltage source that is inside the port component. The amplitude of the noise is divided by two, which assumes a perfect match. If an accurate input-referred noise is desired, set the *Input Source* to *voltage*, and select the input source. This will disable the noise figure calculation.

ADE Implementation

The shift now is from theoretical to practical with the description of where you can find the settings talked about in the concepts section. This section describes the qpnoise *Choosing Analyses* form and the options form. For examples with all the steps shown, see the examples section that follows this section.

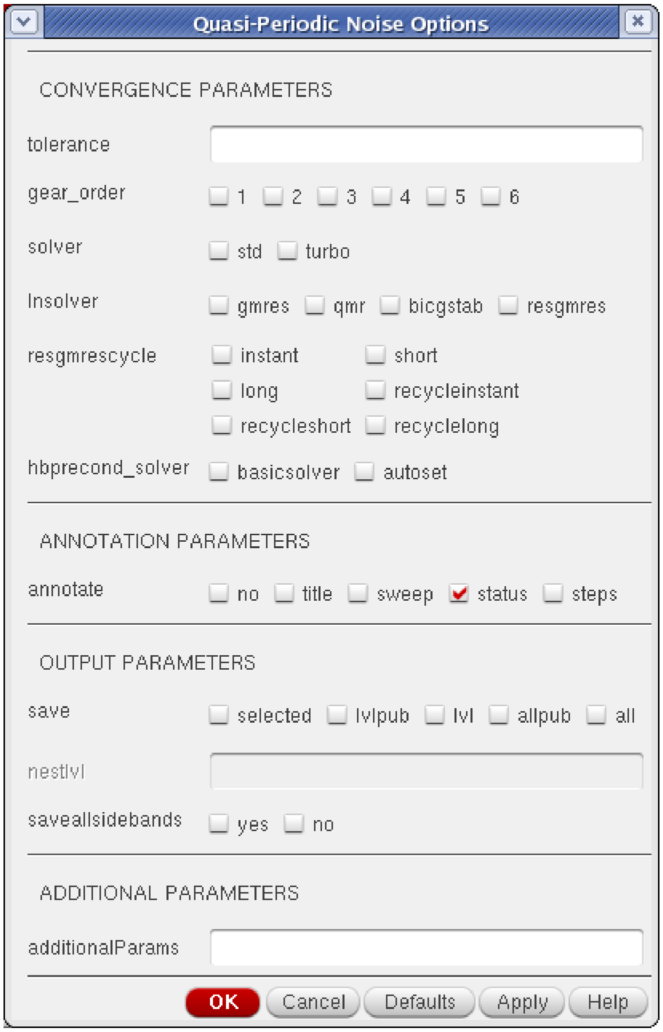
Below is the *Choosing Analyses* form.



1. The default *Sweeptype* is *absolute*, which means that the noise frequencies defined in the F*requency Sweep Range* fields will not be moved in frequency. The frequency sweep range is always an **output** frequency range.
2. Generally, it is better to select Linear or Log spacing and define your own sweep. Auto usually takes more points that are necessary, and that produces longer runtimes than needed.
3. *Maximum Clock Order* should be set to the same number that you have set the Large signal harmonics to in the qpss form. If you use harmonic balance as the engine in qpss, you can leave this field blank, which accomplishes the same thing.
4. When *Output* is set to *probe*, select a resistor or a port that serves as the load. The noise of this component will be subtracted for the noise figure calculation. That port or resistor also defines the output nets in the circuit. The nets that the port or resistor are connected to will be the output nets for the noise calculation.
5. When output is set to voltage, select a net or nets in the circuit as the output. No noise will be excluded from the noise figure calculation.
6. In order to make a noise figure calculation, the input must be a port.
7. When input-referred noise is needed, do not use a port as the input source. Instead, use a separate voltage source with a resistor in series. Although using a port does work, note that the input-referred noise is really referred to the voltage source that is in the port, and then that noise is divided by two in amplitude. This assumes a perfect match, which rarely occurs in practice.
8. The reference sideband defines the passband frequency for the noise figure calculations. The easiest way to define the passband frequency is to choose *Select from list*, and then select the input frequency range that defines the passband from the list. This also defines the input frequency for the input-referred noise calculation.
9. Noise separation is provided so that troublesome noise frequencies and components can be identified in the circuit. There is an example in the Example section below.

Qpnoise Options

None of the qpnoise options are commonly used.



tolerance

Leave this option at the default.

Qpnoise uses an iterative solver to calculate the noise, and any iterative solver needs a tolerance term to specify when the system has been solved accurately enough. The default for shooting qpnoise is 1e-9. The harmonic balance qpnoise tolerance is 1e-6.

gear_order

Do not set this option.

solver

The default is *turbo*.

When shooting is used as the engine in qpss, sometimes you will see warning messages in qpnoise that the residual is larger than the tolerance specified. This usually occurs at low offset frequencies from one of the harmonics in the qpss analysis. If you see this message, then select the *std* solver, which is more able to handle low offset frequencies. The runtime will increase when the *std* solver is selected.

When harmonic balance is the qpss engine, do not change this option.

lnsolver

Leave this option at the default, which is *gmres*. To calculate the conversion gain from the noise sources to the output, many matrices are formed and solved to calculate the noise at the output of the system. These matrices are solved using an iterative solver. Several different algorithms are provided for the iterative solver. Gmres is the default because the accuracy of each iteration inherently increases with increasing iterations. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, seek graduate-level textbooks that describe linear algebra theory.

- Gmres is the Generalized Minimum RESidual method.
- Qmr is the Quasi-Minimal Residual method.
- Bicgstab is the STABilized BI-Conjugate Gradient method.
- Resgmres is the REStarted Generalized Minimal Residual method.

resgmrescycle

Leave this option at the default, which is *short*. For the resgmres solver, there are several different options.

hbprecond_solver

Note that if you plan to use harmonic balance qpss, the hb form is suggested instead.

This option is only available when harmonic balance is set for the qpss engine. When APS is not used, only the basic solver is available. When APS is used, first a faster preconditioner is used. Occasionally, this preconditioner will stagnate, and when it does, it issues a warning that it is switching to the basic solver, and it reverts back to the basic solver. A preconditioner is a mathematical algorithm that makes the iterative matrix solving process faster.

annotate

This option controls the level of detail in the output log. The default is *status*. The leftmost selection provides the least output. As you move right, more detail is provided in the output log.

save

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes. *selected* saves only the nodes that are specifically saved.

nestlvl

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer.

saveallsidebands

Leave this option at the default value of *no*. If you want to see the noise contributors from the different noise input frequencies, select *yes* for noise separation in the qpnoise *Choosing Analyses* form.

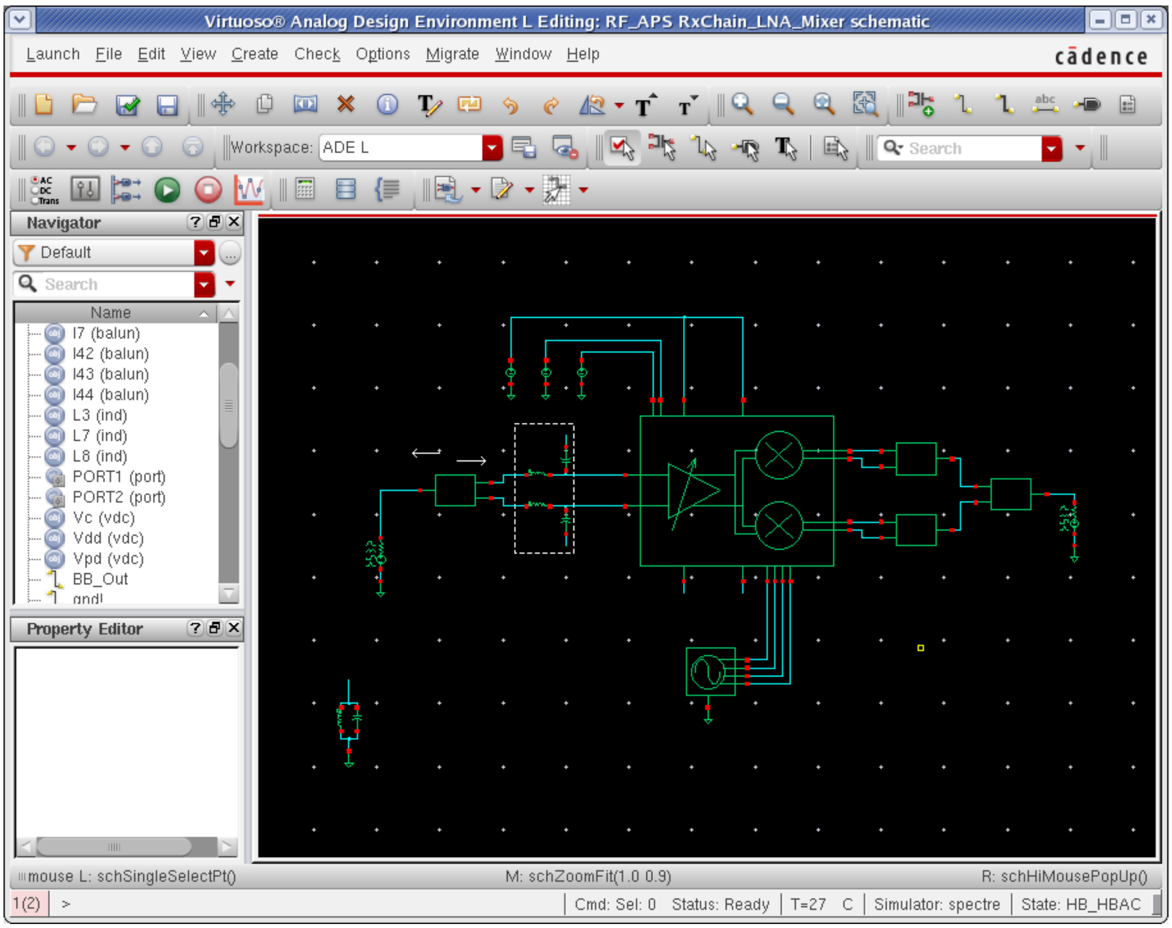
additionalParams

This is typically used for customers who are evaluating new features. If you know the option name and the value, you can type it here. For example, you can type *solver=std* in this field instead of using the check box in the options form. multiple *keyword=value* pairs can be entered in this field. Separate the entries with a space.

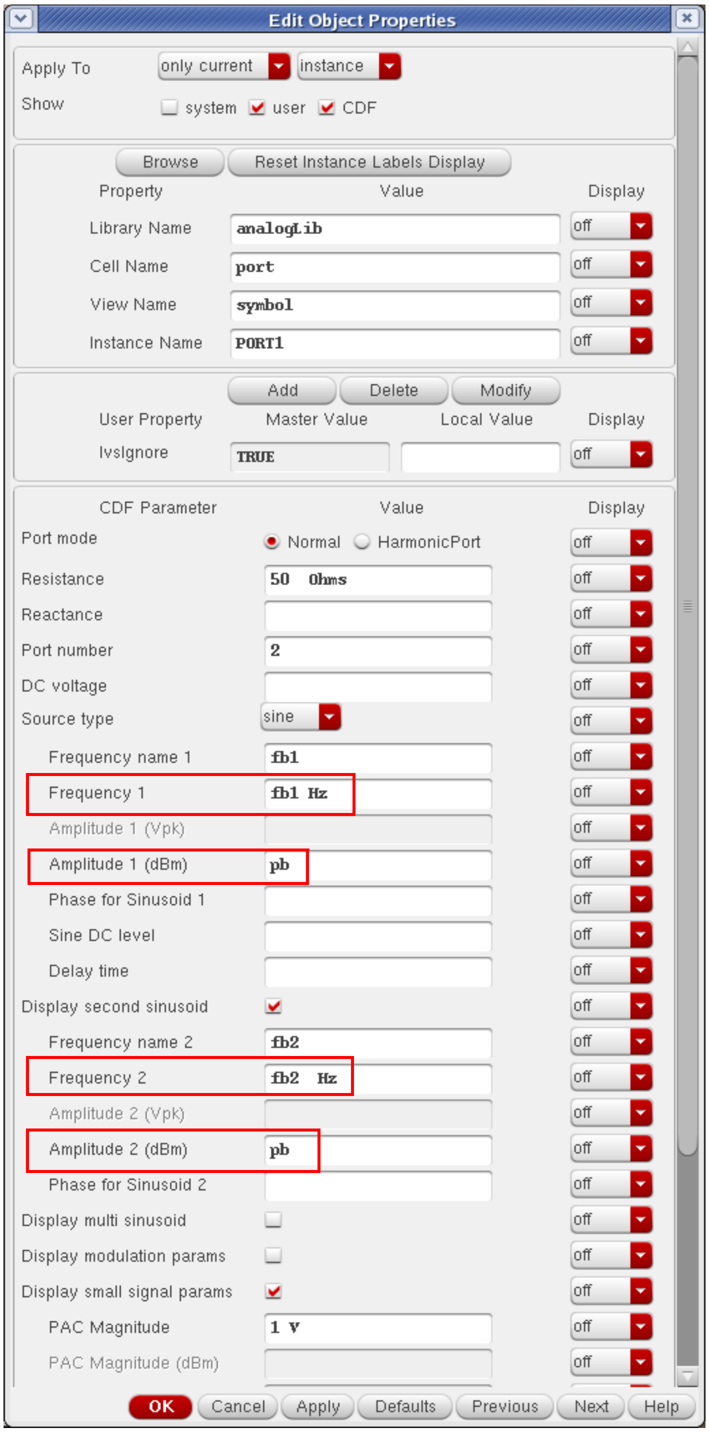
Example

Noise Figure and Noise Summary

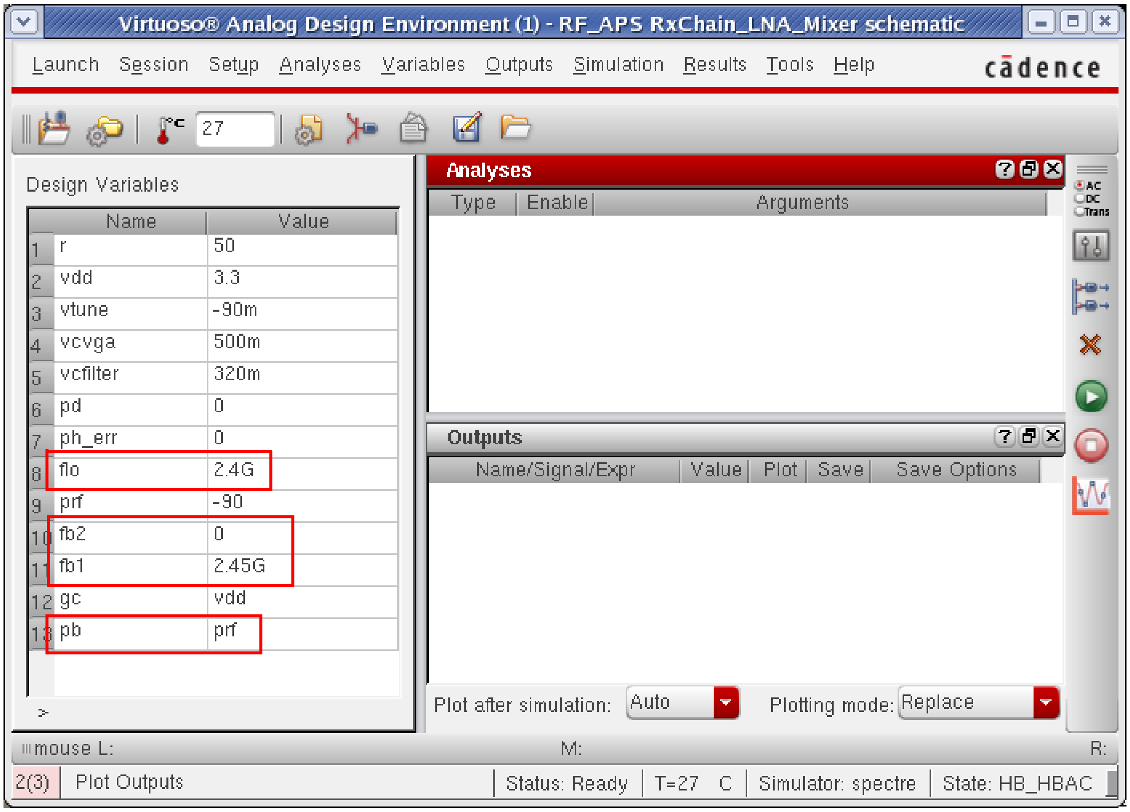
Consider a 2.4GHz LNA-Mixer.



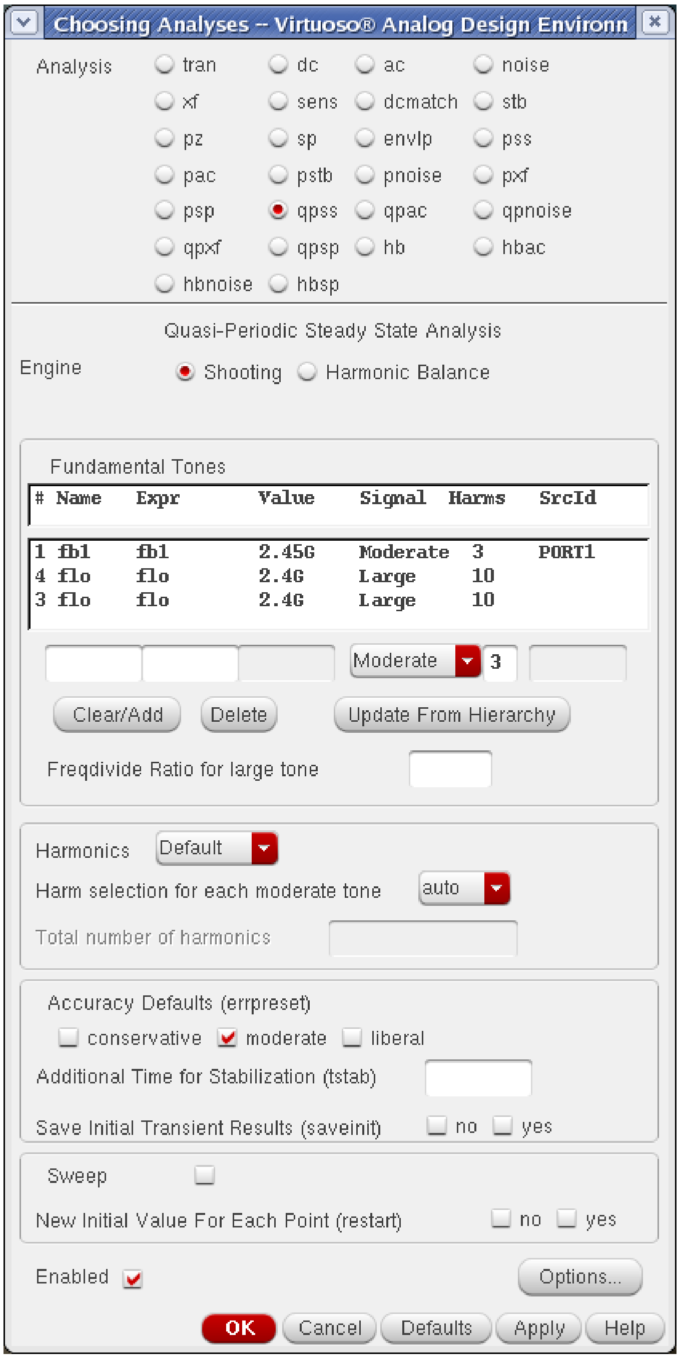
The input source has variables set to a variable name. This is done in order to allow easy frequency and amplitude changes from the ADE environment without needing to change the schematic. Here is the properties list for the input port.



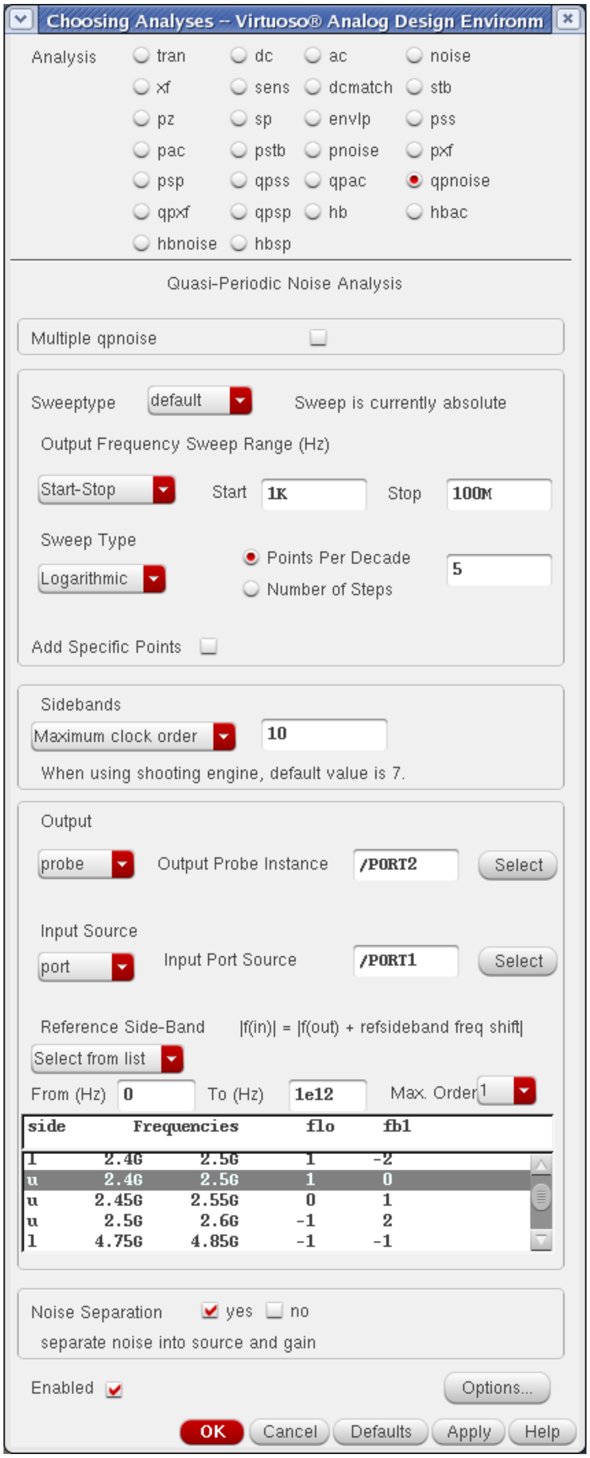
The variables are set in the ADE environment.



1. First, the qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter. Here, a 2.4GHz LO and a 2.45GHz blocker are set up.



2. Now set up the qpnoise analysis.



3. The output frequency range is set, and log sweep is used. It is usually better to set log or linear and define your own sweep because in general automatic takes more points than are usually needed.

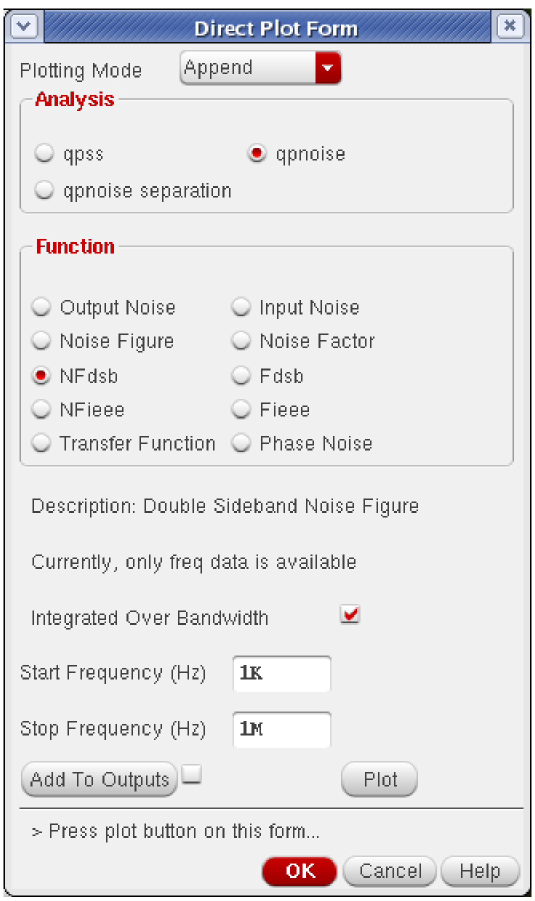
4. *Maximum clock order* should be set to the number of harmonics that are used for the LO in the qpss *Choosing Analyses* form.

5. The output port and the input port are defined. Specifying *probe* for the *Output* and selecting either a port or a resistor automatically excludes the noise of the port from the noise figure calculation.

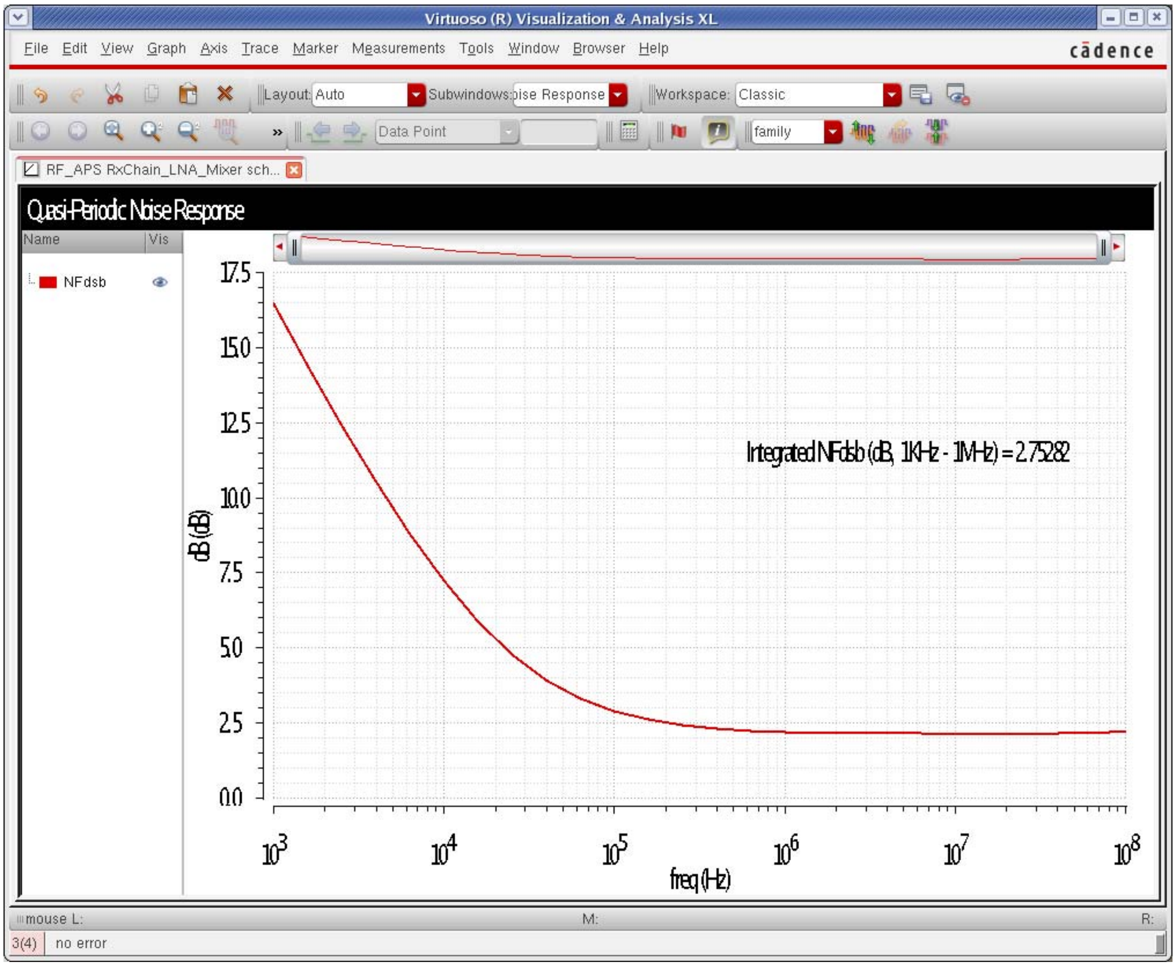
6. Using *Select from list* in the *Reference Side-Band* section is highly recommended. Select the passband frequency (the design input frequency) from the list.

7. If you want to see the different frequency and device contributions in the *Direct Plot Form*, select *yes* for *Noise Separation*.

8. Now run the analysis. When the analysis finishes, select *Results - Direct Plot - Main Form* from ADE. The *Direct Plot Form* is displayed.



9. In the *Direct Plot Form*, select qpnoise from the *Analysis* section.
10. Select the desired measurement, and click *Plot*.
11. If you have a noise figure spec over a defined bandwidth, select *Integrated Over Bandwidth*, type in the frequency range for the measurement, and click *Plot*. The noise figure curve, and the noise figure over the bandwidth are plotted.



Noise Summary

The noise summary provides information about the devices that contribute noise to the output. This information is available any time a qpnoise simulation completes.

In ADE, select *Results - Print - Noise Summary*. If noise separation is enabled, there will be two choices at the very top. `Qpnoise_src` has information about the noise currents at the individual noise sources in the circuit. `Qpnoise` has information about noise voltage at the output of the circuit.

If you want the noise contributors at a single frequency, select *spot noise*. If you want to have the noise integrated over a frequency range, select *integrated noise*.

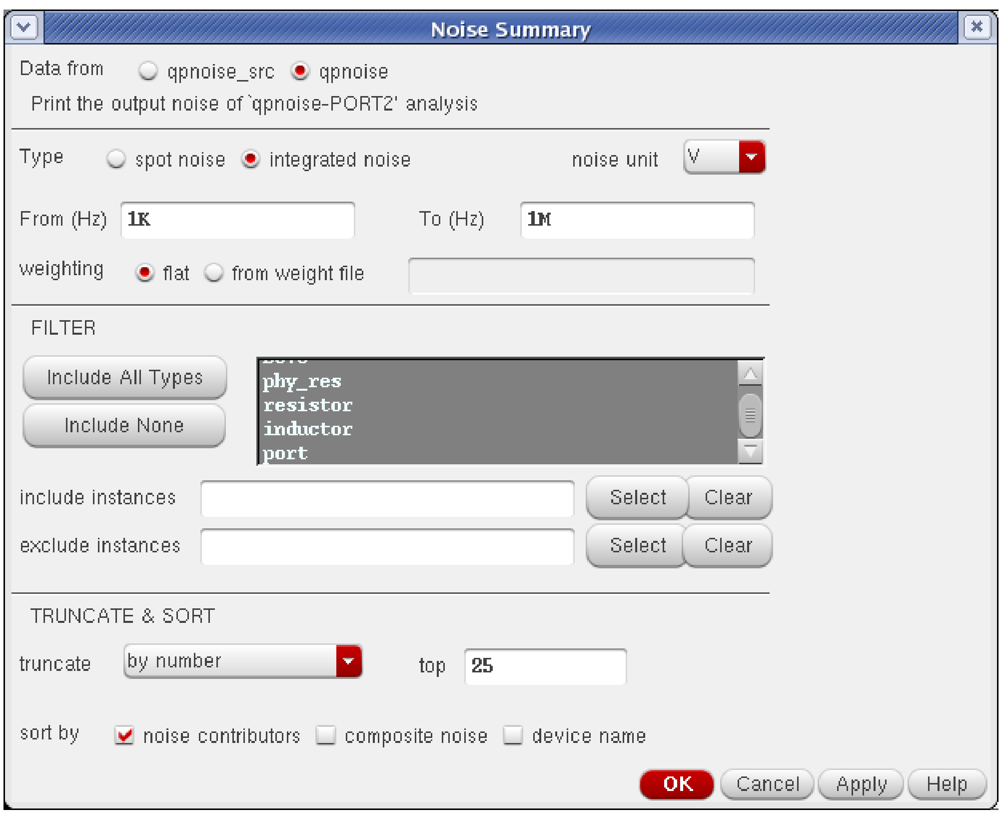
If you want noise in volts, set the *noise unit* to *V.*

Generally, *Include All Types* should be selected. If you just want the noise from specific types of noise generators, you can select them from the list.

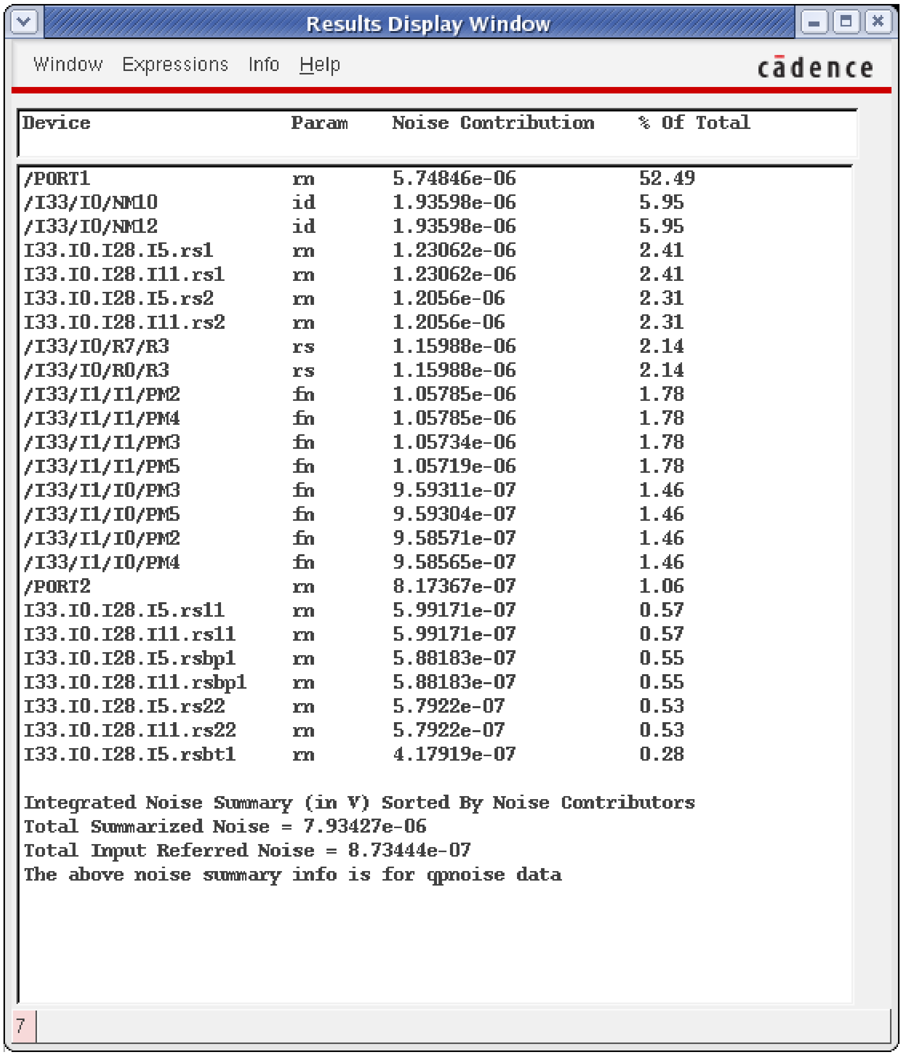
If you want to specifically include or exclude instances in the list, click the *Select* button to the right side of the *include instances* and *exclude instances* fields and then select the components in the schematic.

truncate just applies to the list. The total input and output-referred noise at the bottom of the noise summary output always includes all the noise from everything.

The figure below is the noise summary window with the sort function being set to noise contributors.



An example of the noise summary output sorted by noise contributors.



Device	Param	Noise Contribution	% Of Total
/PORT1	rn	5.74846e-06	52.49
/I33/I0/NM10	id	1.93598e-06	5.95
/I33/I0/NM12	id	1.93598e-06	5.95
I33.I0.I28.I5.rs1	rn	1.23062e-06	2.41
I33.I0.I28.I11.rs1	rn	1.23062e-06	2.41
I33.I0.I28.I5.rs2	rn	1.2056e-06	2.31
I33.I0.I28.I11.rs2	rn	1.2056e-06	2.31
/I33/I0/R7/R3	rs	1.15988e-06	2.14
/I33/I0/R0/R3	rs	1.15988e-06	2.14
/I33/I1/I1/PM2	fn	1.05785e-06	1.78
/I33/I1/I1/PM4	fn	1.05785e-06	1.78
/I33/I1/I1/PM3	fn	1.05734e-06	1.78
/I33/I1/I1/PM5	fn	1.05719e-06	1.78
/I33/I1/I0/PM3	fn	9.59311e-07	1.46
/I33/I1/I0/PM5	fn	9.59304e-07	1.46
/I33/I1/I0/PM2	fn	9.58571e-07	1.46
/I33/I1/I0/PM4	fn	9.58565e-07	1.46
/PORT2	rn	8.17367e-07	1.06
I33.I0.I28.I5.rs11	rn	5.99171e-07	0.57
I33.I0.I28.I11.rs11	rn	5.99171e-07	0.57
I33.I0.I28.I5.rsbp1	rn	5.88183e-07	0.55
I33.I0.I28.I11.rsbp1	rn	5.88183e-07	0.55
I33.I0.I28.I5.rs22	rn	5.7922e-07	0.53
I33.I0.I28.I11.rs22	rn	5.7922e-07	0.53
I33.I0.I28.I5.rsbt1	rn	4.17919e-07	0.28

```
Integrated Noise Summary (in V) Sorted By Noise Contributors
Total Summarized Noise = 7.93427e-06
Total Input Referred Noise = 8.73444e-07
The above noise summary info is for qpnoise data
```

For all of the noise summaries, here is a table of the abbreviations used in the *Param* column:

Abbreviations Used In the Param Column

Resistor and Inductor:

Parameter	**Meaning**
`rn`	resistor thermal noise

Parameter	Meaning
fn	Flicker noise

Diode:

Parameter	Meaning
id	Diode current shot noise
rs	Series parasitic resistor noise

Gummel-Poon BJT:

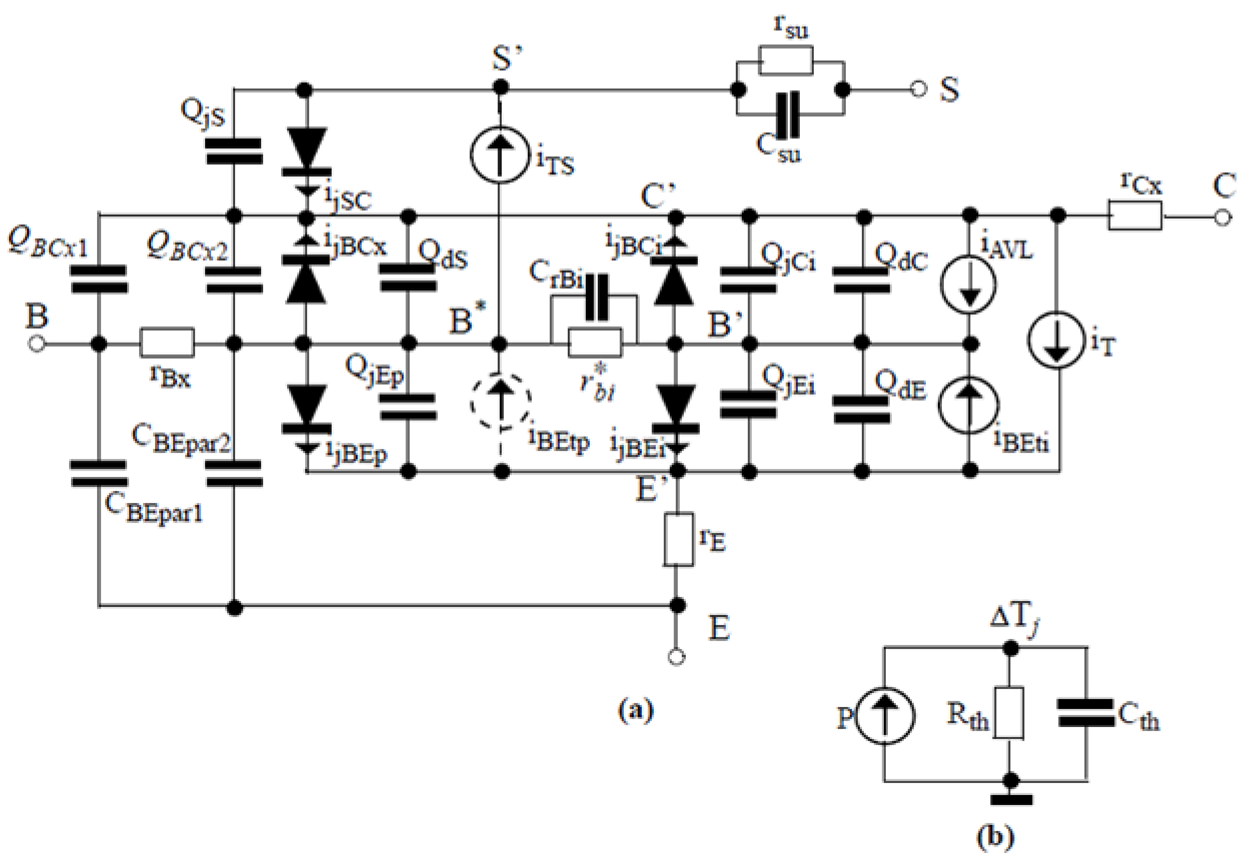
Parameter	Meaning
fn	Flicker noise
ic	Collector current shot noise
ib	Base current shot noise
rc	Thermal noise from the collector parasitic resistor
rb	Thermal noise from the base parasitic resistor
re	Thermal noise from the emitter parasitic resistor

HiCUM:

Parameter	Meaning
ib	Base current shot noise
ic	Collector current shot noise
rbi	Thermal noise from the internal base parasitic resistor (See diagram below)
rbx	Thermal noise from the external base parasitic resistor (See diagram below)
rcx	Collector parasitic resistor thermal noise (See diagram below)

Parameter	Meaning
`re`	Emitter parasitic resistor thermal noise (See diagram below)
`fn`	Flicker noise

Figure 5-1 HiCUM Schematic Diagram

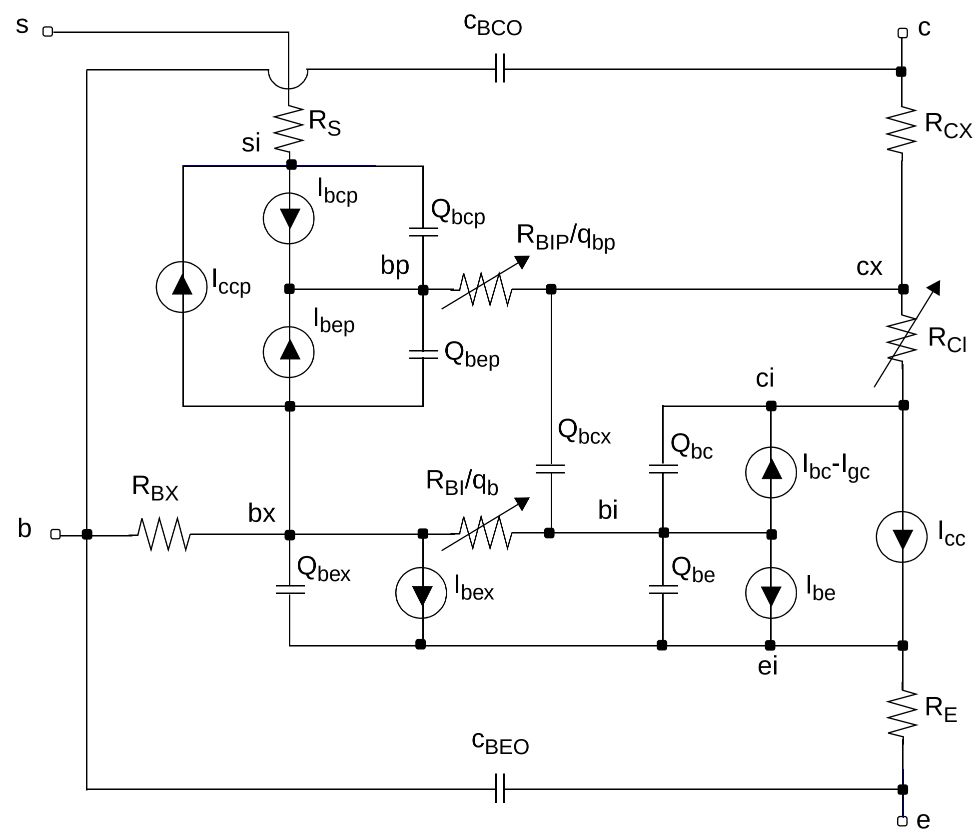


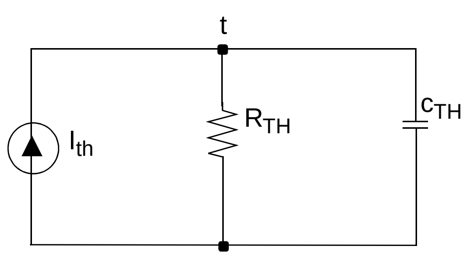
VBIC:

Parameter	Meaning
`rbi`	Thermal noise in resistor Rbi (See diagram below)
`rbx`	Thermal noise in resistor Rbx (See diagram below)
`rci`	Thermal noise in resistor Rci (See diagram below)

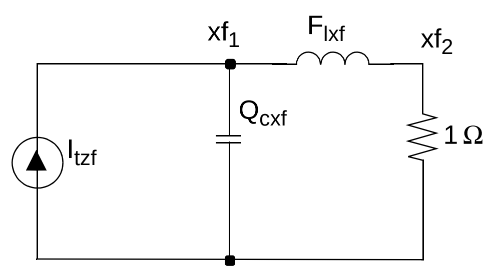
Parameter	Meaning
`re`	Thermal noise in resistor Re (See diagram below)
`rs`	Thermal noise in resistor Rs (See diagram below)
`itzf`	Collector current shot noise
`ibe`	Total base current shot noise
`ibex`	Node Bx to E base current shot noise (See diagram below)
`fn`	Ibe flicker noise (See diagram below)
`fnx`	Ibex flicker noise (See diagram below)
`iccp`	Parasitic transport collector to emitter current noise (See diagram below)
`ibep`	Parasitic transport base to emitter current noise (See diagram below)
`fnp`	Ibep flicker Noise (See diagram below)

Figure 5-2 VBIC Schematic Diagram





Thermal network



Excess phase network

BSIM3:

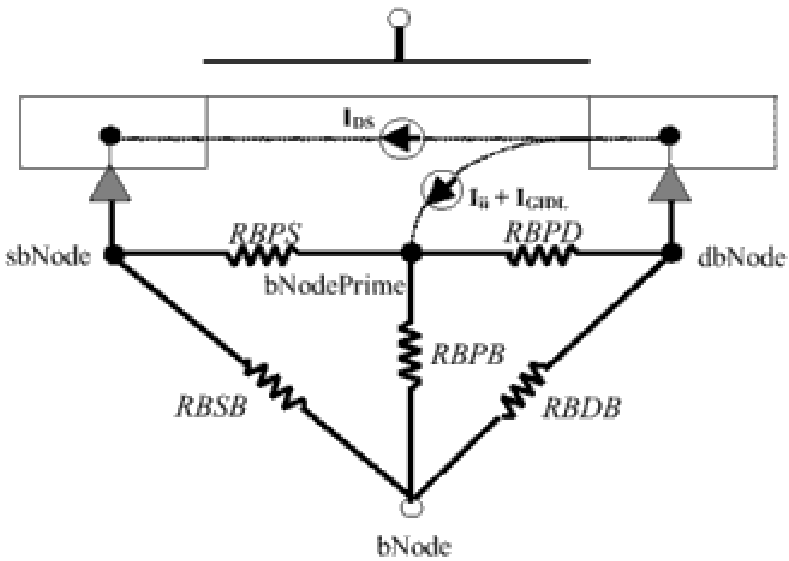
Parameter	Meaning
`fn`	Flicker Noise
`id`	Thermal noise from the resistive drain current

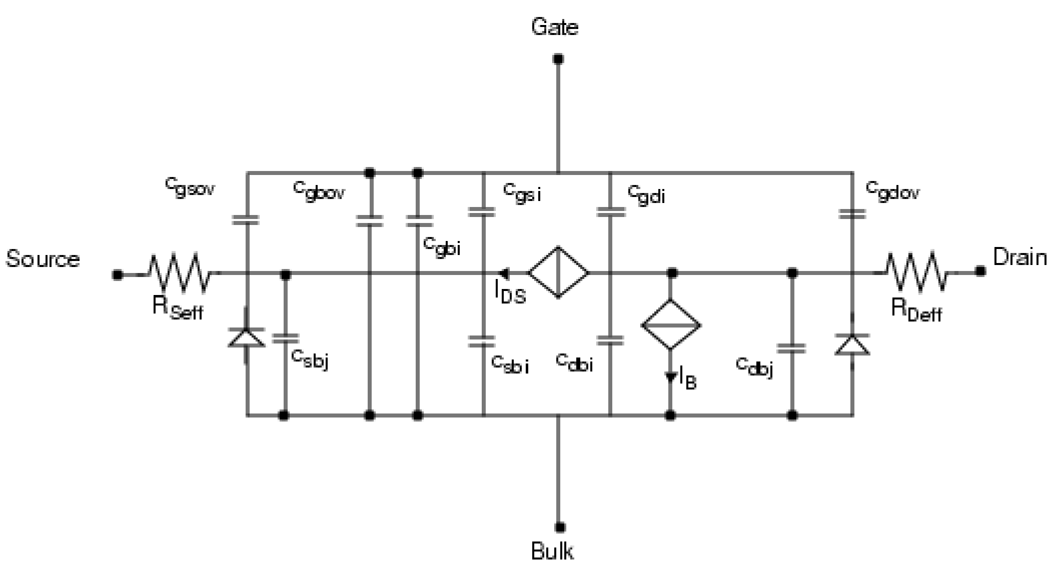
Parameter	Meaning
`rs`	Thermal noise from the source parasitic resistor
`rd`	Thermal noise from the drain parasitic resistor

BSIM4:

Parameter	Meaning
`fn`	Flicker noise
`id`	Thermal noise from the resistive drain current noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`igd`	Gate-to-drain tunneling current noise
`igs`	Gate-to-source tunneling current noise
`igb`	Gate-to-bulk tunneling current noise
`rbdb`	Resistor thermal noise between dbNode and bNode (See diagram below)
`rbpb`	Resistor thermal noise between bNode and bNodePrime (See diagram below)
`rbpd`	Resistor thermal noise between Bulk and dbNode (See diagram below)
`rbps`	Resistor thermal noise between bulk and bNode prime
`rbsb`	Resistor thermal noise between sbNode and bNode (See diagram below)
`rgbi`	Gate Bias-Independent Resistor thermal noise

Figure 5-3 BSIM4 Schematic Diagrams



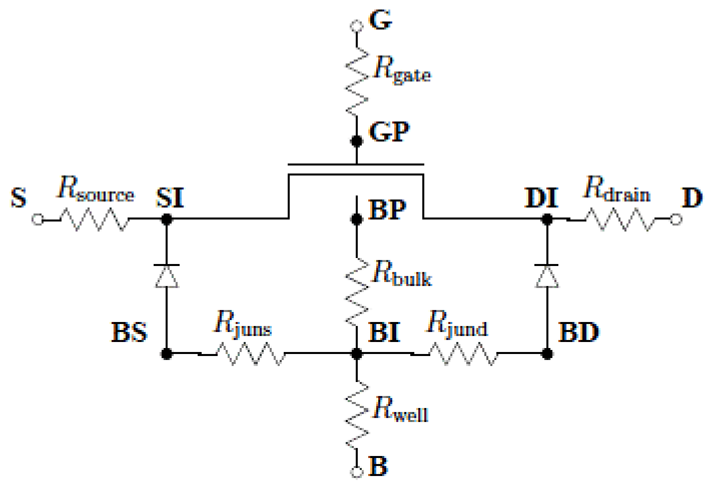


PSP-103

!

Parameter	Meaning
`sfl`	Channel current flicker noise
`sth`	Channel current thermal noise
`sig`	induced gate noise at 1 Hz
`sigth`	Induced gate noise
`shotgd`	Gate current shot noise from gate to drain
`shotgs`	Gate current shot noise from gate to source
`sjnoise`	Source to Bulk leakage current noise
`djnoise`	Drain to Bulk leakage and avalanche current noise
`rgatenoise`	Gate resistance thermal noise
`rbulknoise`	Bulk resistor thermal noise between node BP and BI (See diagram below)
`rjnoise`	Source side bulk resistor thermal noise between node BI and BS (See diagram below)
`rjundnoise`	Drain side bulk resistor thermal noise between node BI and BD (See diagram below)
`rwellnoise`	Well resistor thermal noise between node BI and B (See diagram below)
`rsourcenoise`	Source parasitic resistor thermal noise
`rdrainnoise`	Drain parasitic resistor thermal noise

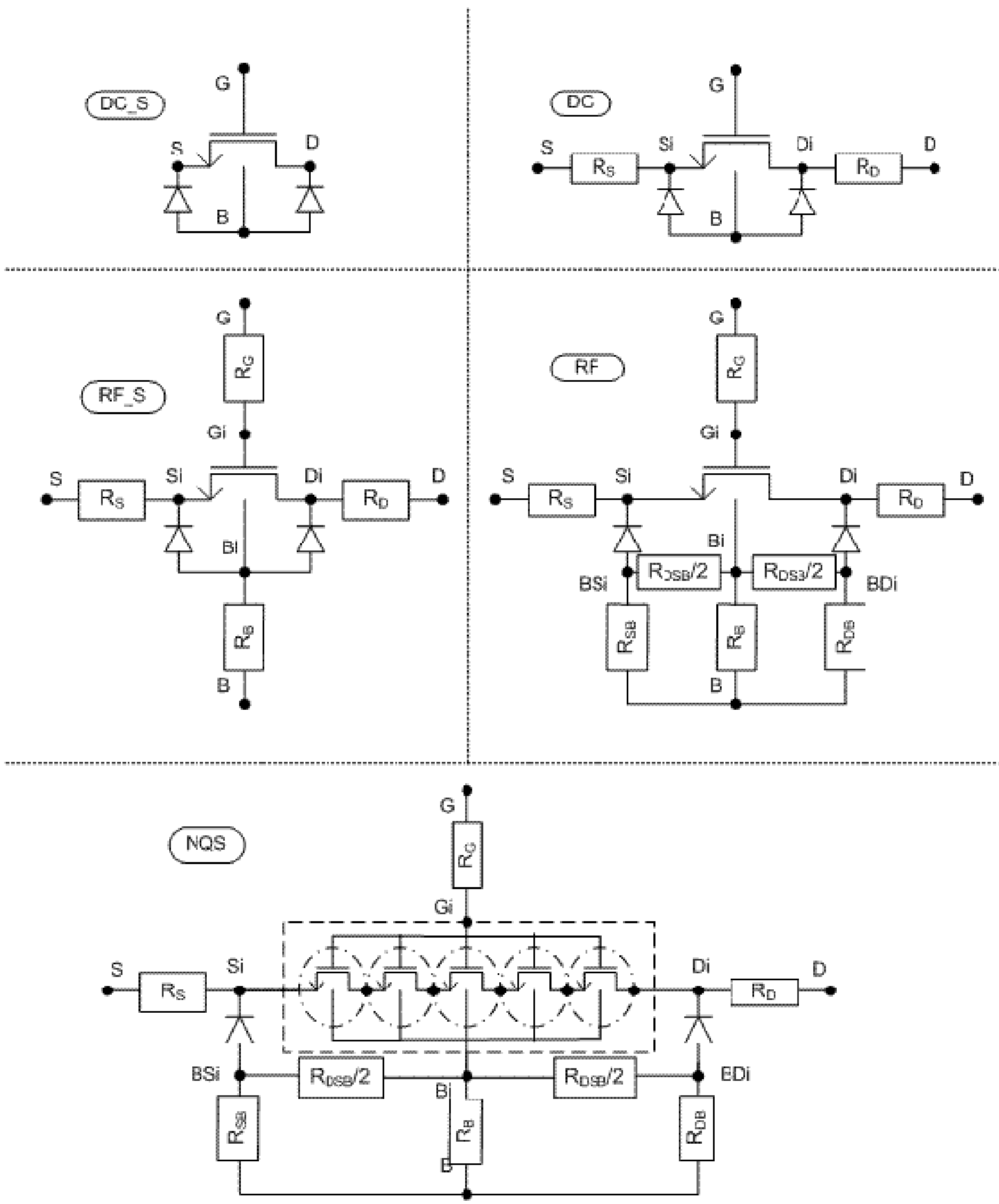
Figure 5-4 PSP-103 Schematic Diagram



EKV 3.0

Parameter	Meaning
`fn_id`	Channel current flicker noise
`fn_ig`	Gate current flicker noise
`rs`	Source parasitic resistor thermal noise
`rd`	Drain parasitic resistor thermal noise
`rg`	Gate parasitic resistor thermal noise
`rsb`	Thermal noise for the substrate resistor connected between BSi and B. (See diagram below)
`rdb`	Thermal noise for the substrate resistor connected between BDi and B. (See diagram below)
`rbbs`	Thermal noise for the substrate resistor connected between BSi and Bi. (See diagram below where it is labeled $R_{DSB}/2$)
`rbbd`	Thermal noise for the substrate resistor connected between BDi and Bi. (See diagram below where it is labeled $R_{DSB}/2$)
`id`	Thermal noise of the channel resistor
`cor`	Induced gate noise
`shot_ig`	Gate current shot noise

Figure 5-5 EKV3.0 Schematic Diagrams



For other model types:

1. In a unix/linux shell window type `which spectre`.

   This should return a path that ends in `tools/bin/spectre`.

2. Triple-click the line with the path, and then click the center mouse button to enter it as a command.

3. Use the backspace key, and delete the word `spectre`.

4. Type `cdnshelp` at the end of the expression, and press `Enter`.

   A browser window will be displayed.

5. Close the *Tip of the day* window.

6. At the top, you should see the release number of the simulator you are using.

7. Click the plus sign (+) to the left of the MMSim folder.

8. Click the plus sign (+) to the left of the *Virtuoso Simulator Components and Device Models Reference* folder.

9. Select the device type you are using.

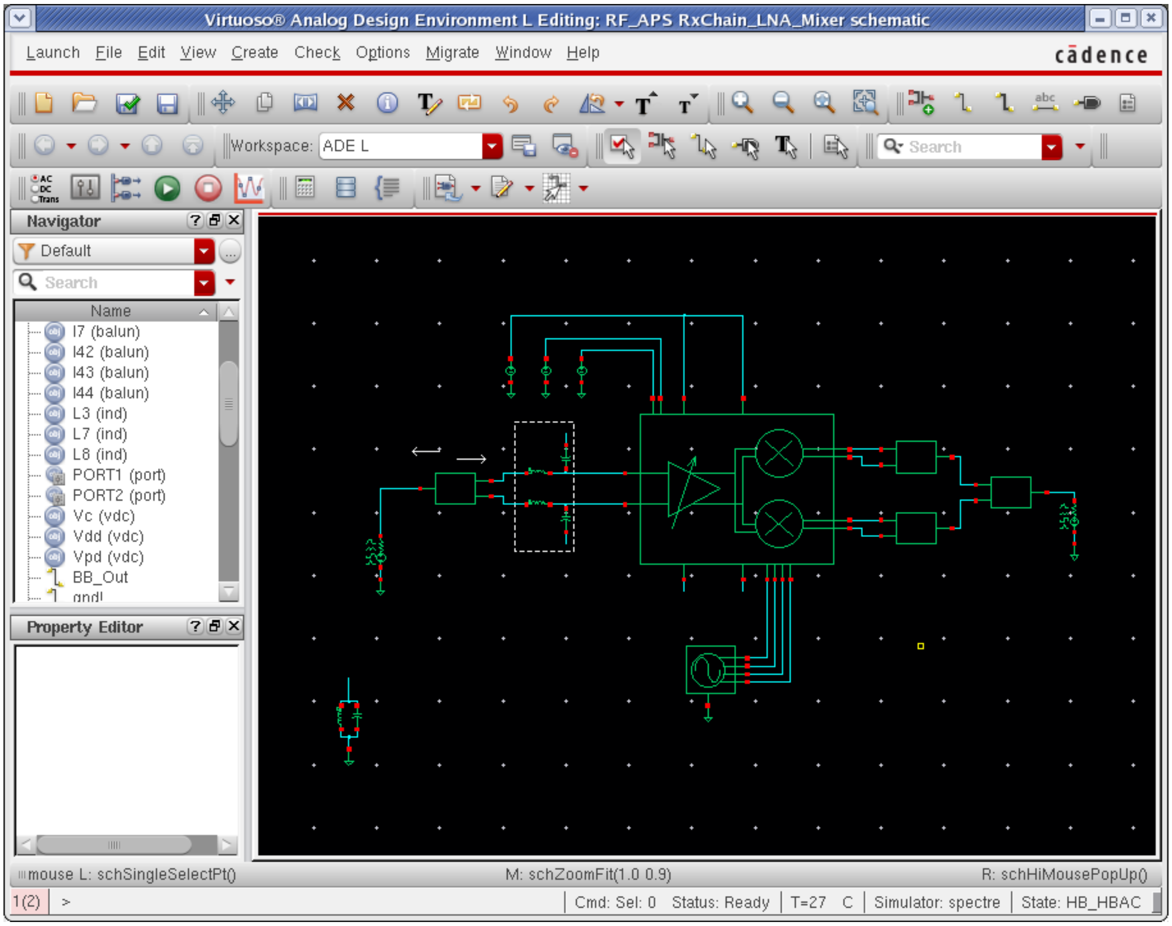
10. At the bottom of that device listing, you will see *Component Statements*. Double-click on this.

11. Enter the name of the parameter in the search box in the right pane of the cdnshelp window.

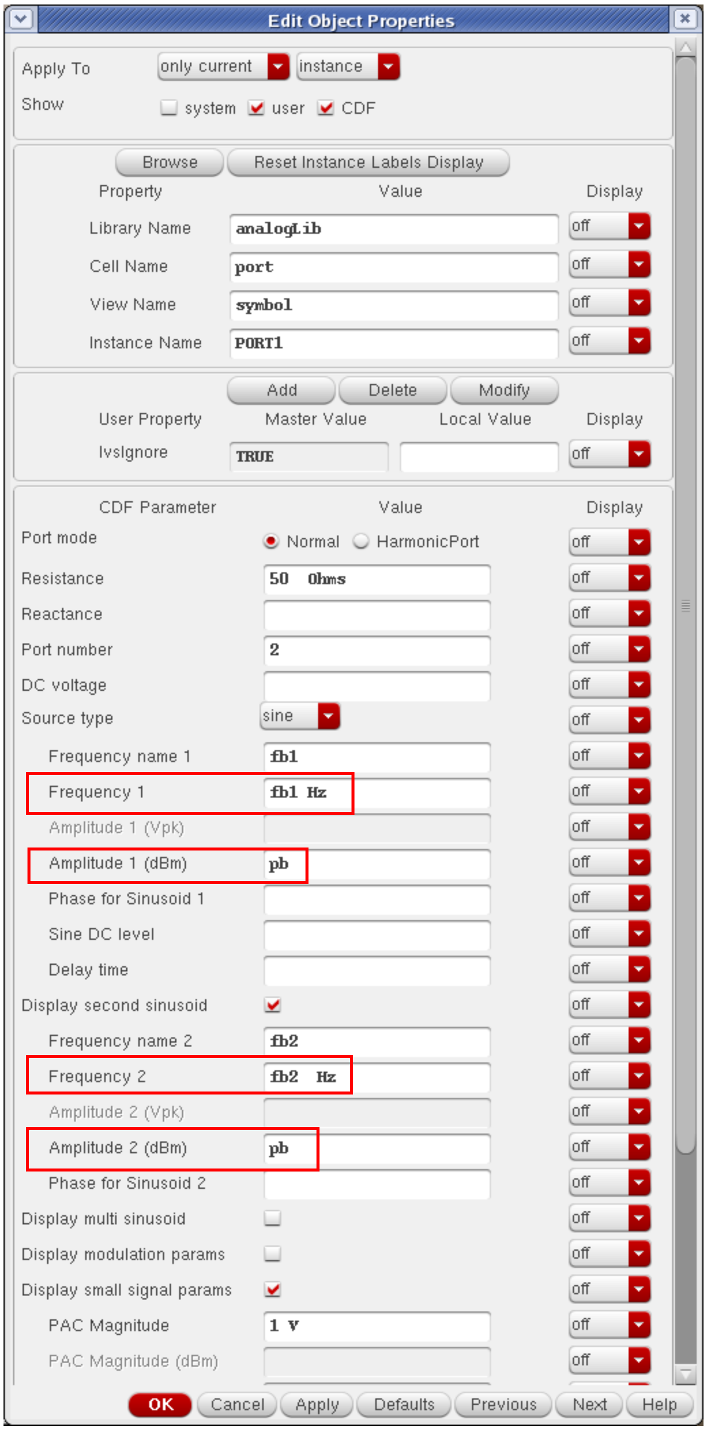
12. Click the down arrow to the right of the *Find* field until you see the parameter name in the pane on the right side, and then see the definition of that parameter.

Viewing Noise Separation Results

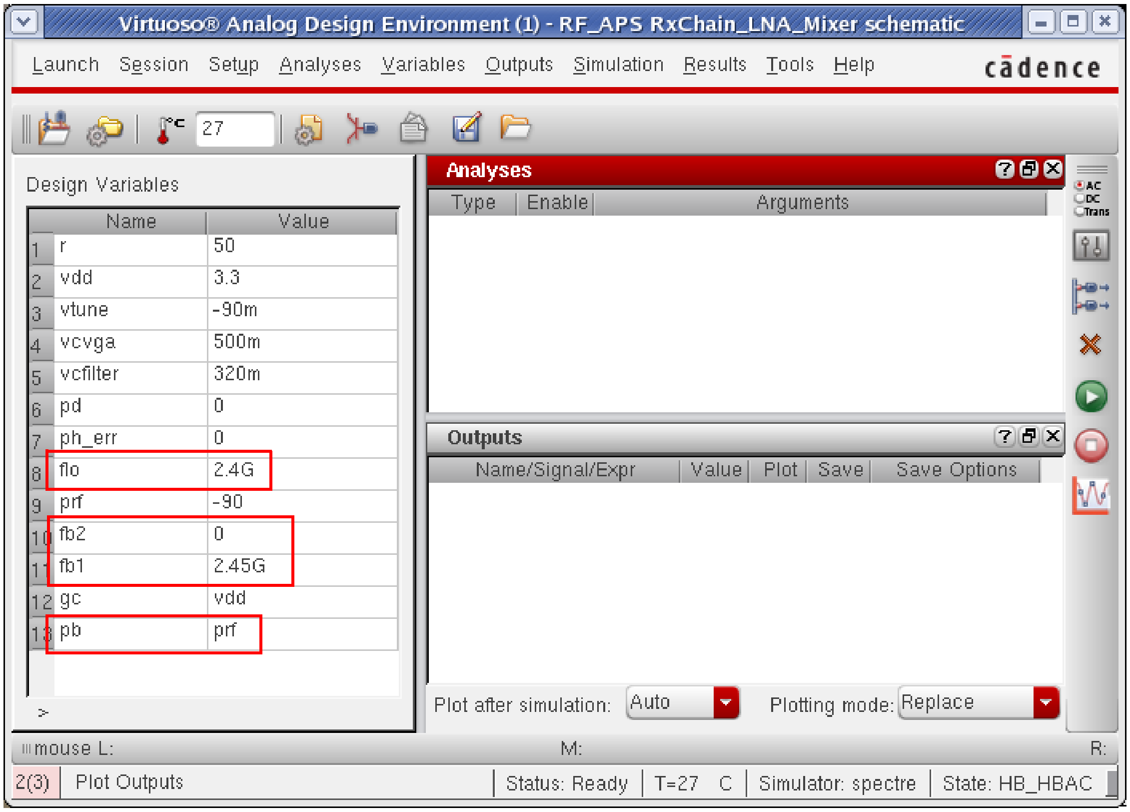
Consider a 2.4GHz LNA-Mixer.



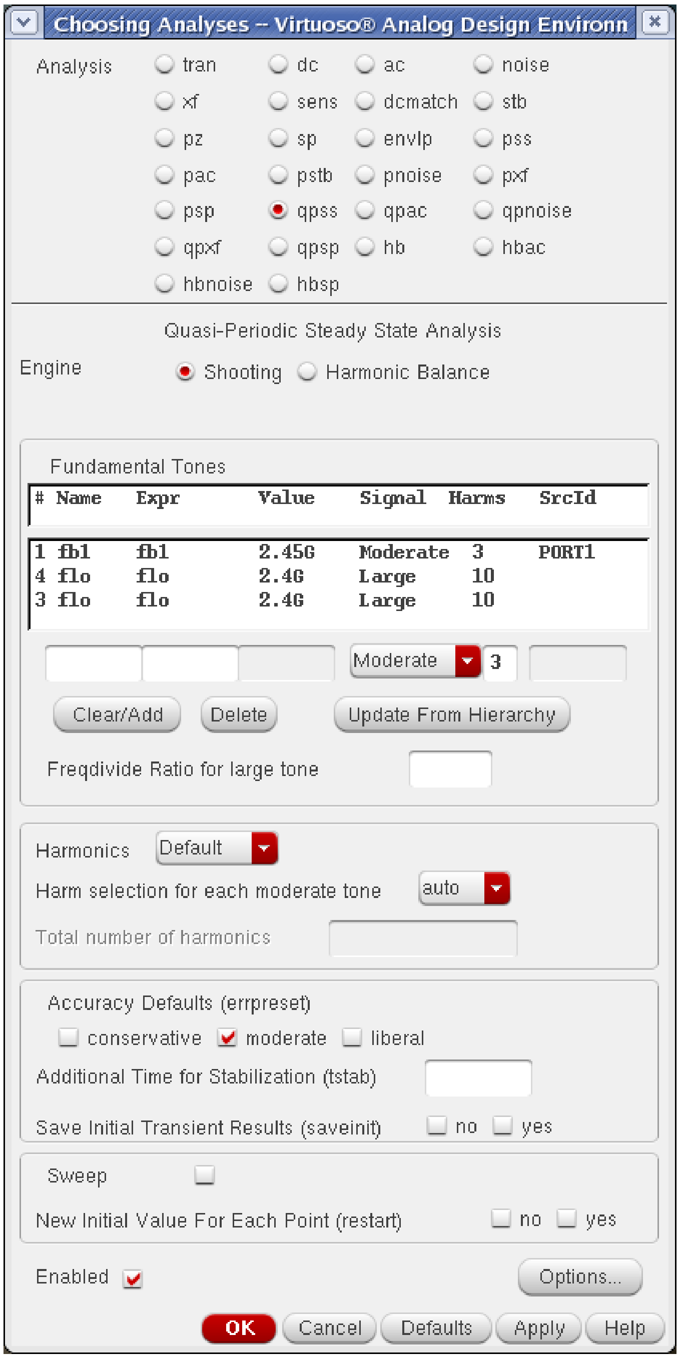
The input source has variables set to a variable name. This is done in order to allow easy frequency and amplitude changes from the ADE environment without needing to change the schematic. Here is the properties list for the input port.



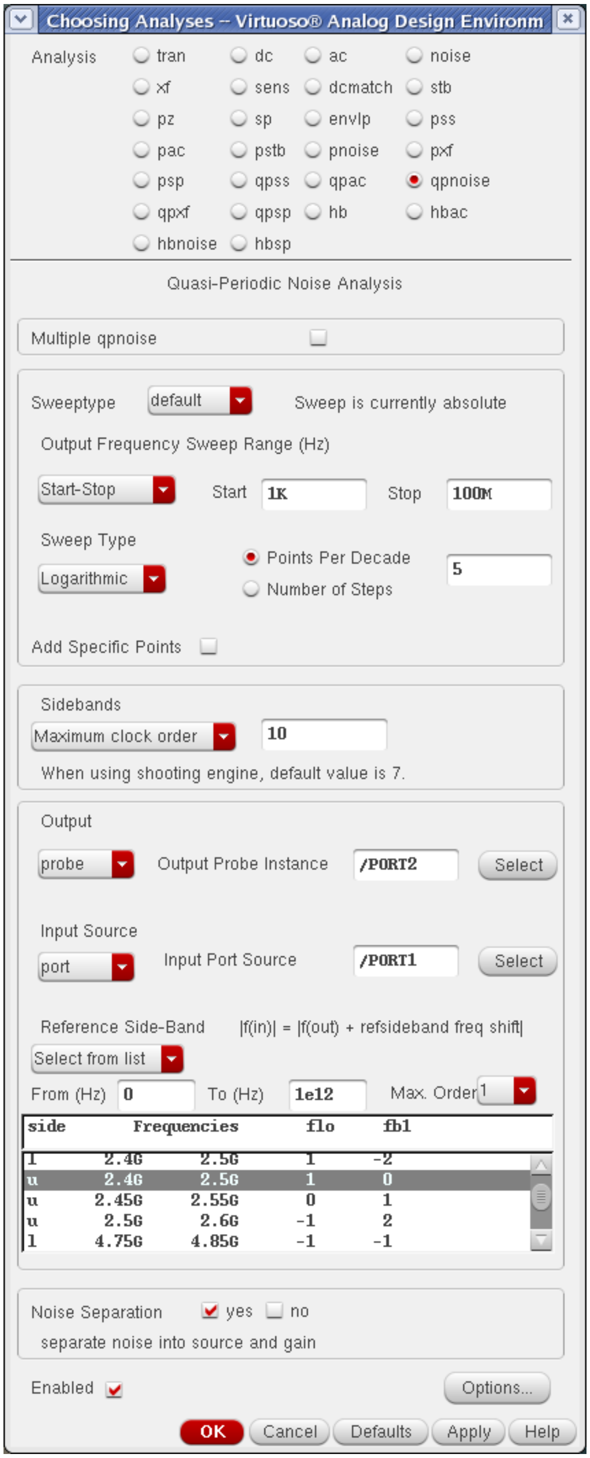
The variables are set in the ADE environment.



1. First, the qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter. Here, a 2.4GHz LO and a 2.45GHz blocker are set up.

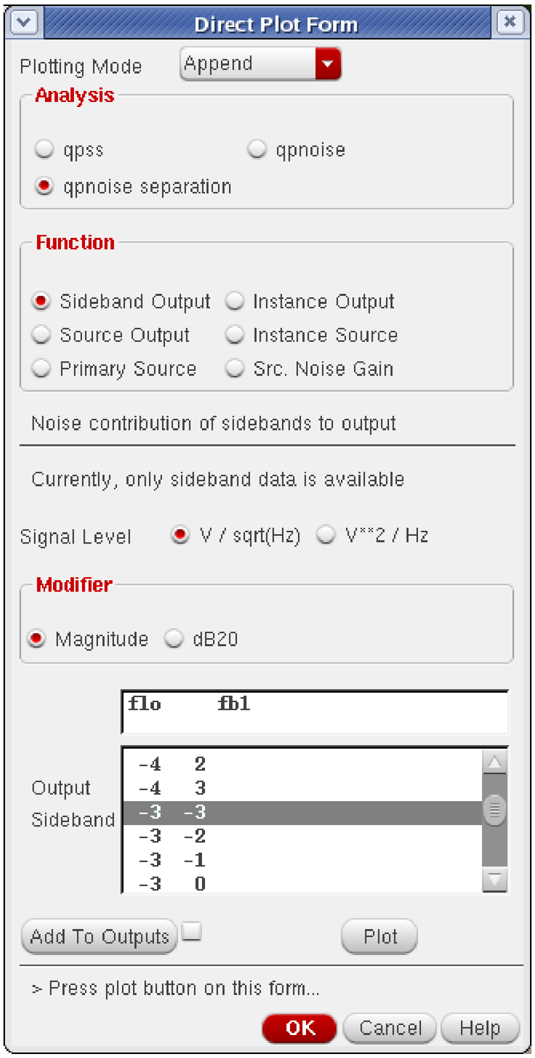


2. Now set up the qpnoise analysis.



3. The output frequency range is set, and log sweep is used. It is usually better to set log or linear and define your own sweep because in general automatic takes more points than are usually needed.
4. *Maximum clock order* should be set to the number of harmonics that are used for the LO in the qpss *Choosing Analyses* form. If you use the harmonic balance engine in qpss, this field can be left blank, which does the same thing.
5. The output port and the input port are defined. Specifying probe for the output and selecting either a port or a resistor automatically excludes the noise of the selected component from the noise figure calculation.
6. Using *Select from list* in the *Reference Side-Band* section is highly recommended. Select the passband frequency (the design input frequency) from the list.
7. If you want to see the different frequency and device contributions in the *Direct Plot Form*, select *yes* for *Noise Separation*.

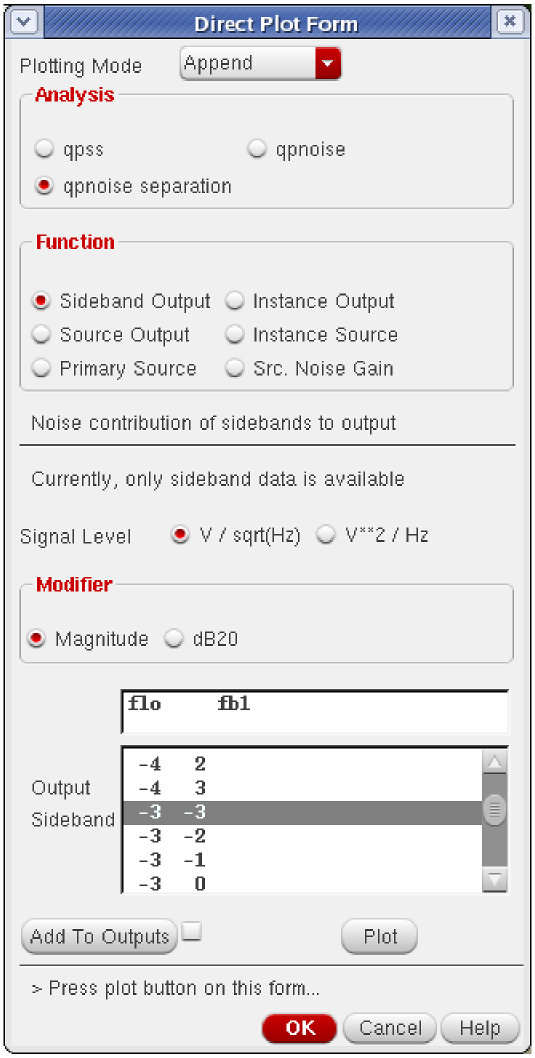
8. Now run the analysis. When the analysis finishes, select *Results - Direct Plot - Main Form* from ADE. The *Direct Plot Form* is displayed.



Viewing the Total Noise at the Output from a Single Noise Input Frequency

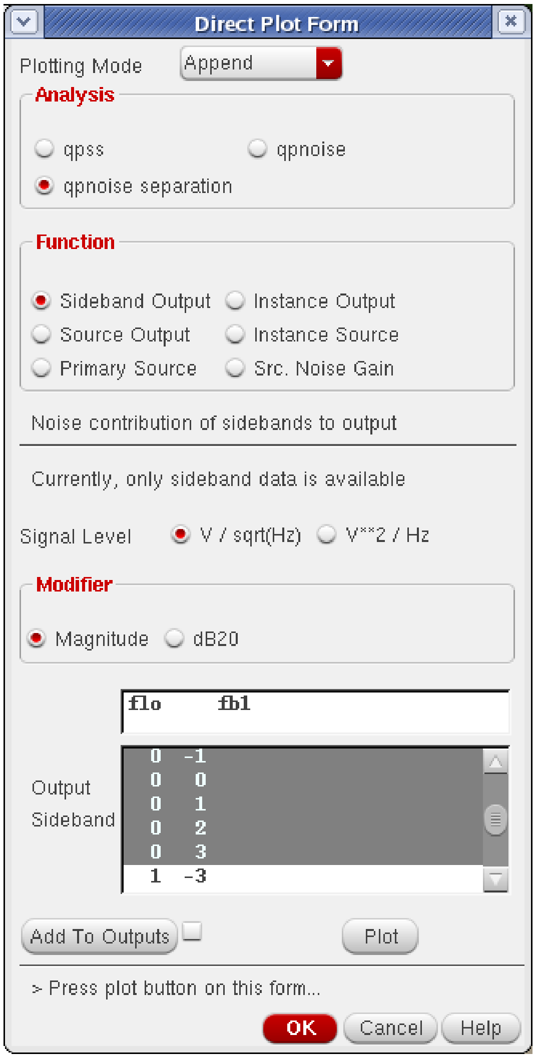
This capability allows the noise from all sources in the circuit at different noise input frequencies to be measured. For example, all the noise that does not change frequency (the zero, zero sideband) can be plotted, along with the other noise input frequencies (sidebands) This plotting capability allows problem frequencies to be identified.

On the Direct Plot Form:



1. Select q*pnoise separation* from the *Analysis* section.
2. To view the total noise at the output select *Sideband Output* from the *Function* section.
3. Select *V/sqrt(Hz)* or *V**2/Hz* from the *Signal Level* radio buttons.
4. Select the appropriate modifier from the *Modifier* section.
5. Select the sideband number form the *Output Sideband* list. Start with about the -3 X sideband that borders the -4 X sideband.

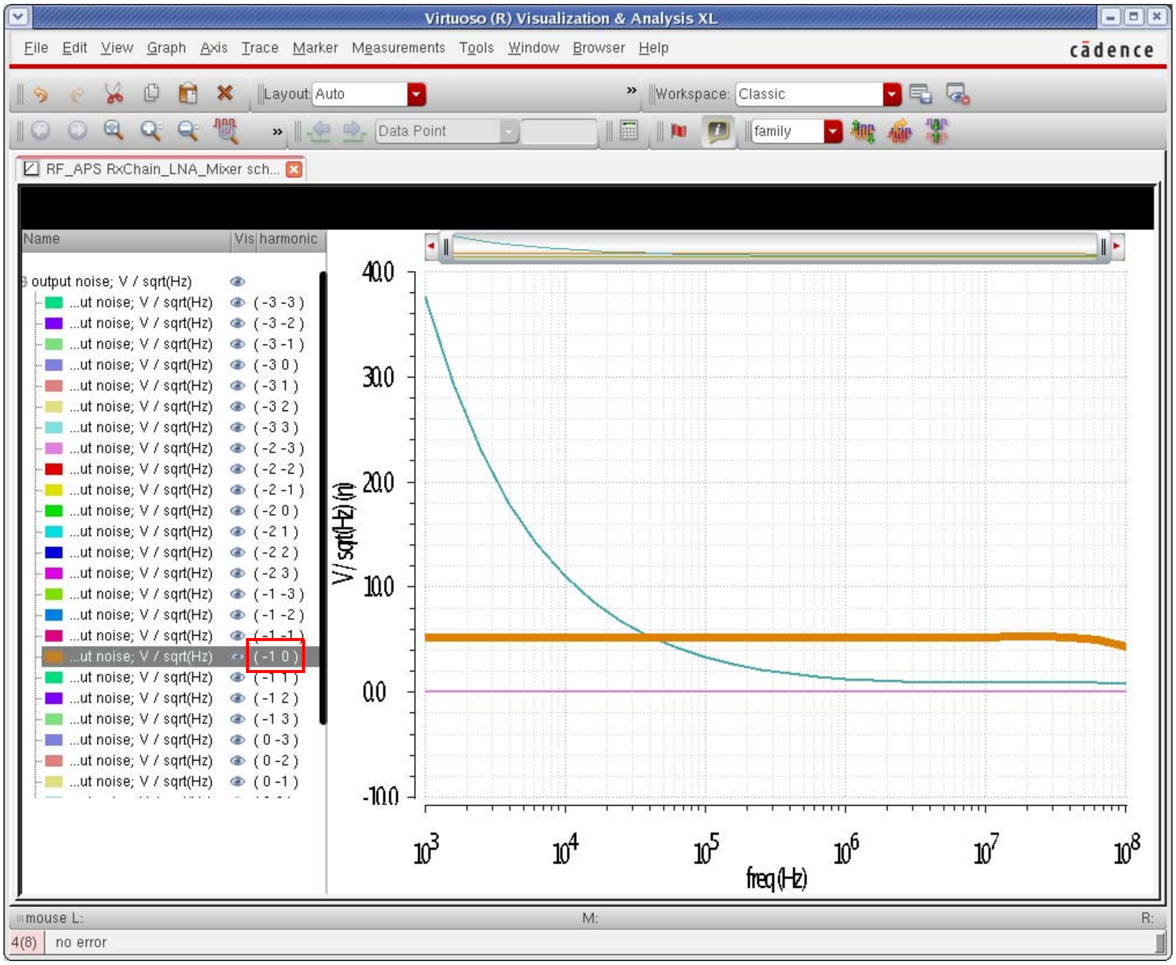
6. Use the slider to get to the boundary between the 0 X sideband and the -1 X sideband. Hold the Press and hold the <`Shift`> key, and click the last 0 X sideband. A lot of sidebands will highlight.



7. In some cases, you want to start with the 0 X sideband and go through the 3 X sideband.

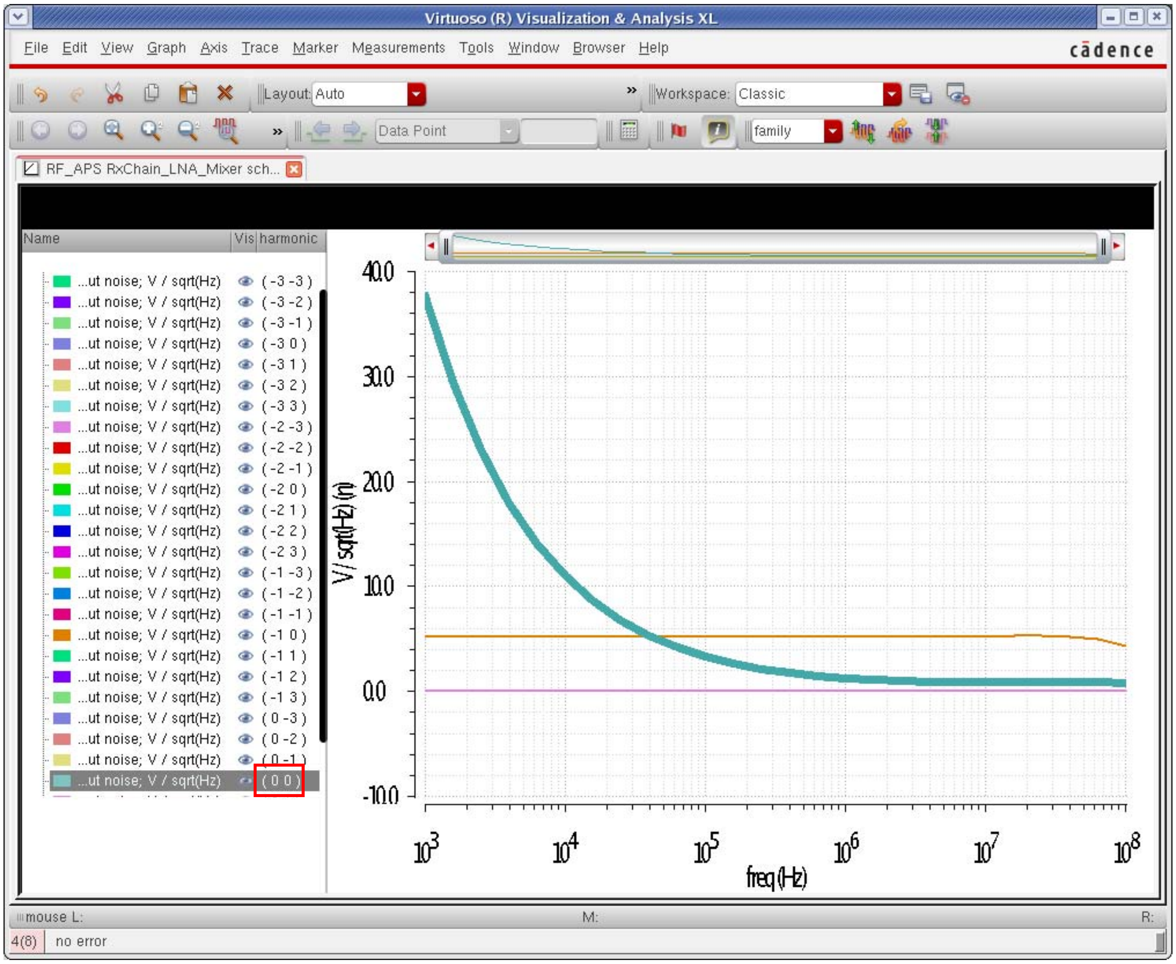
8. Usually, above the third harmonic, little noise is mixed to the output frequency.

9. Click *Plot*.



10. Select the trace with the highest amplitude at high frequency. Note that this is the -1 0 sideband in this case. -1 0 means the noise input frequency is mixing with the first harmonic of the LO and not with the RF input, and the noise input frequency is more negative than the output frequency. The noise frequency is 1KHz to 100MHz minus 1 * (LO Frequency of 2.4GHz) or -2.5GHz to nearly -2.4 GHz.

11. Select the trace with the largest 1/F noise.



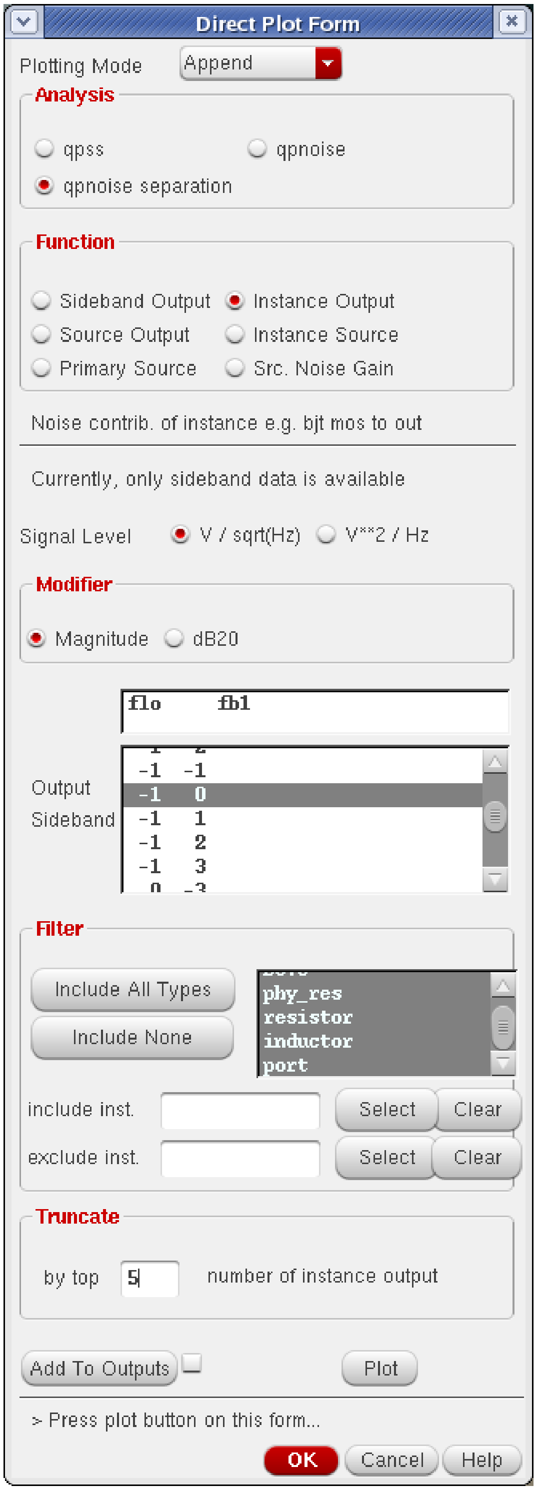
12. This is the 0 0 sideband. There is no mixing at all with either the LO or the RF, so the noise frequency range is from 1KHz to 100MHz.

13. For the rest of this example, the -1 0 sideband is used. For isolating 1/F noise contributors, the 0 0 sideband would be used. The steps are exactly the same for both cases.

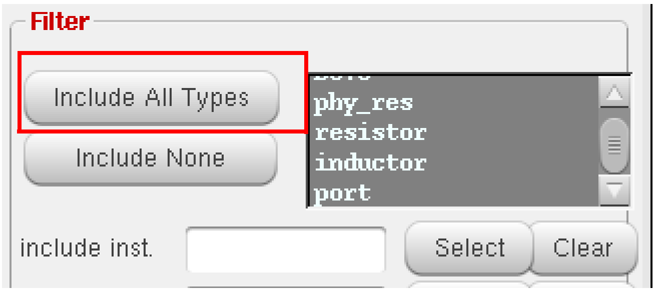
Viewing Total Noise at the Output from each Instance

Once the problem frequencies are identified in the plot of *Sideband Output*, the problem components need to be identified. From the previous plot, the -1 0 sideband contributes the largest amount of noise to the overall output noise. Using *Instance Output* allows the identification of which components with noise in the -1 0 sideband contribute the most noise. The total noise for all the noise mechanisms within the component is plotted.

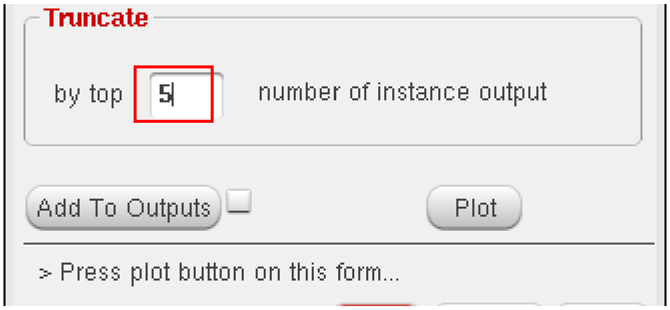
On the *Direct Plot Form*:



1. Select *Instance Output* from the *Function* section.
2. Select *V/sqrt(Hz)* or *V**2/Hz*.
3. Select the appropriate modifier.
4. Select the sideband number. In this example, the -1 0 sideband was determined to contribute the most noise at the output of the circuit.
5. Select the type of component to display. If you want all of the types, choose *Include All Types*.

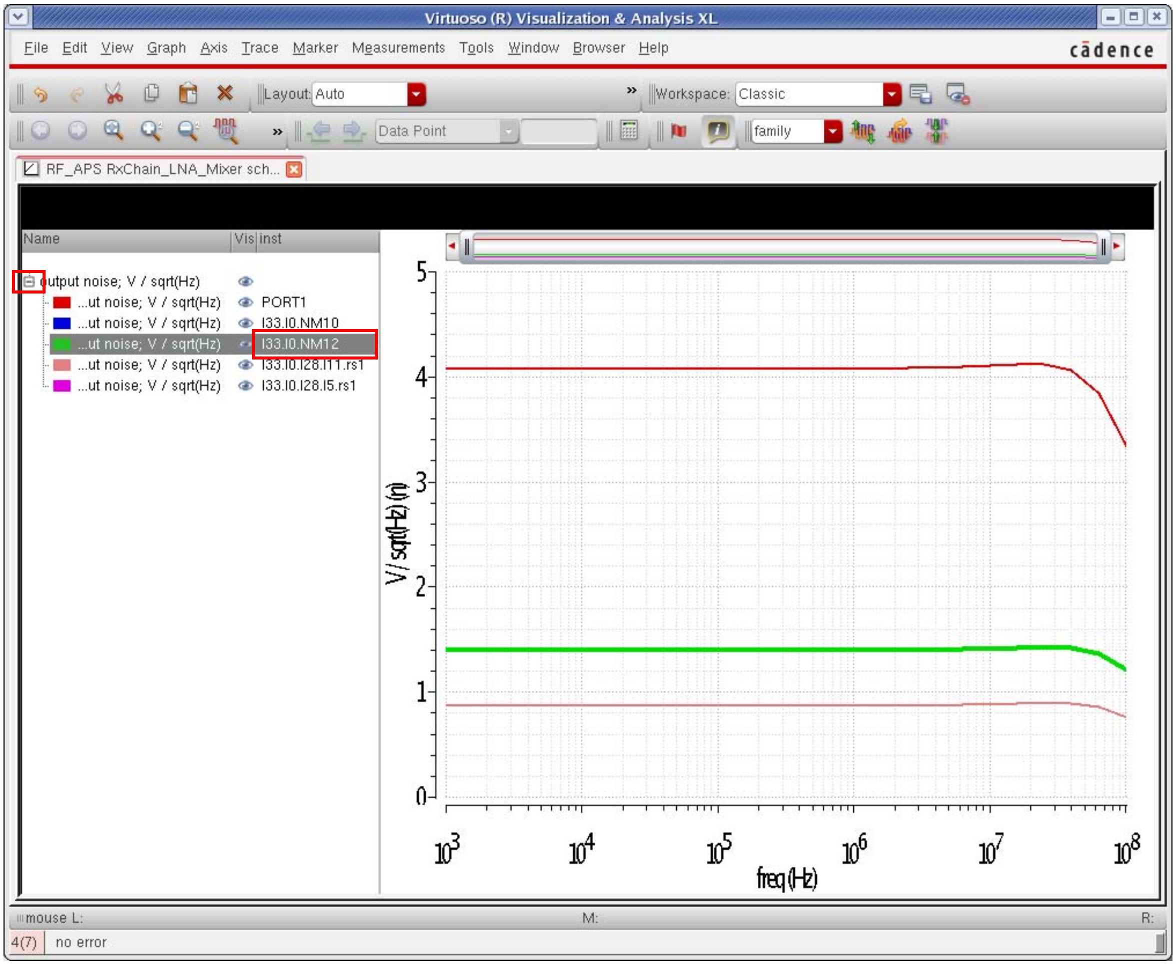


6. Specify the number of results you want to plot.



7. Click *Plot*.

The total noise at the output for the top five components is plotted.



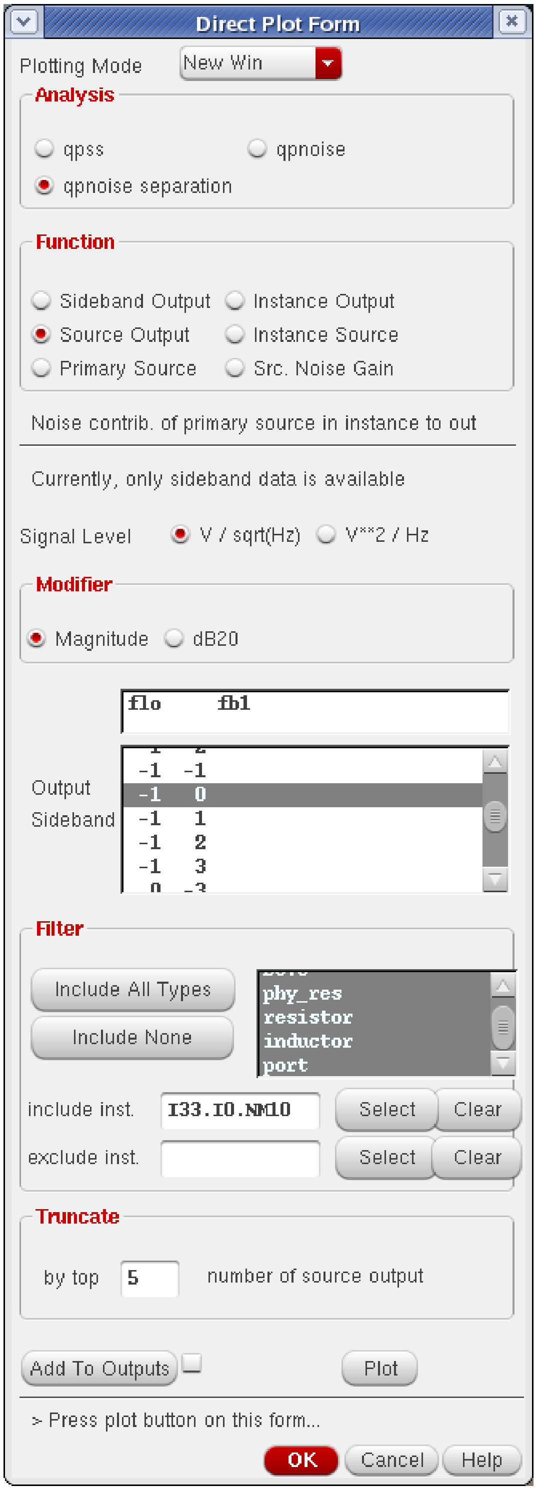
To see which component goes with which trace, select the plus sign (highlighted in the red box on the previous graphic) to the left or the text at the upper left of the display area. The legends expand, and you can see which trace is which in the *Name* column on the left side of the display tool.

The red trace above comes from the input port. The next largest contributor from the circuit is the green curve, which is selected. The instance name is shown in the legend. I33.I0.NM12 is the highest contributor. I33 is the top-level instance. I0 is the next level down instance. NM12 is the next level down instance.

Viewing the Exact Noise Mechanism Within the devices

The sideband with the largest contribution was determined by selecting *Sideband Output* and plotting. The specific component was identified by selecting *Instance Output*. The specific noise mechanisms can be identified using *Source output*. This plots the largest individual noise mechanisms for the sideband you select.

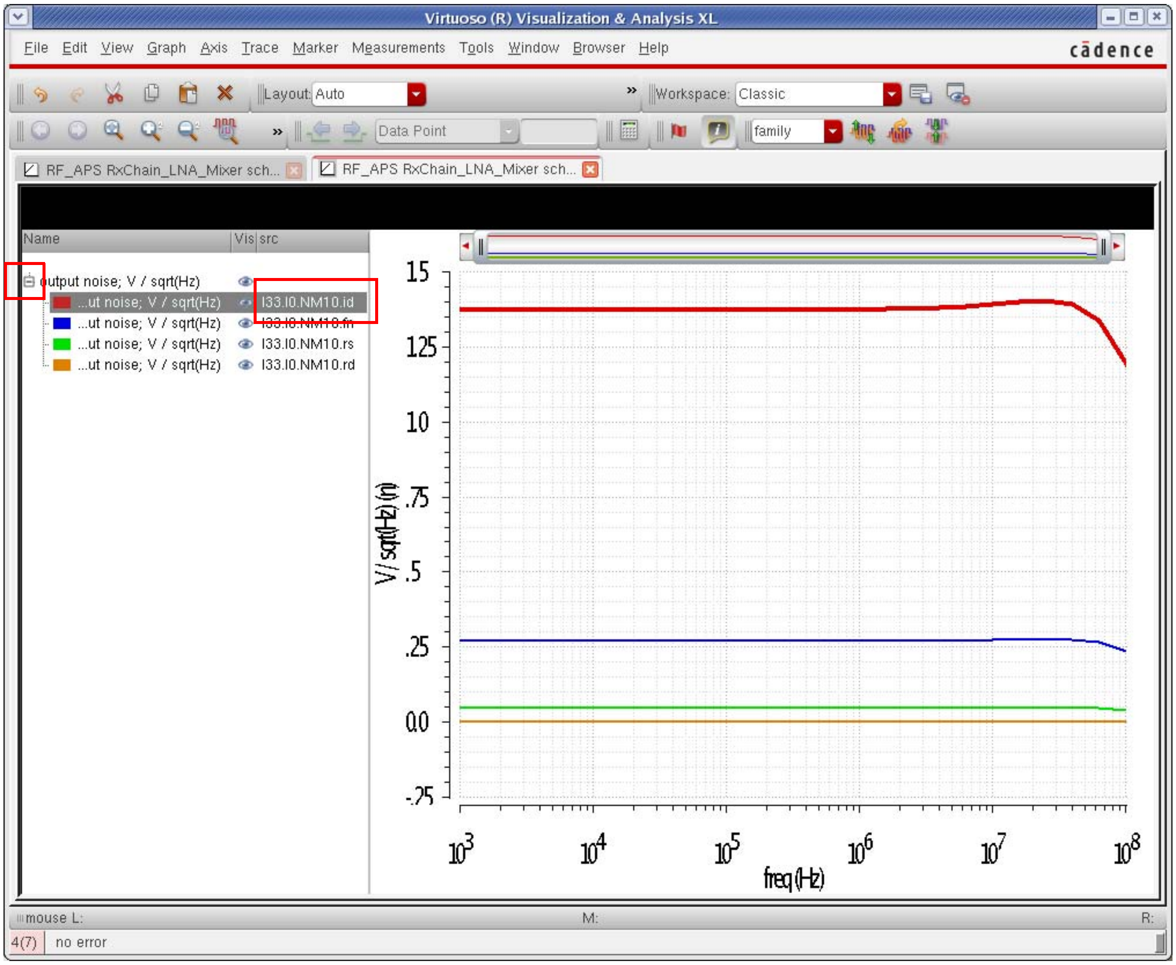
On the *Direct Plot Form:*



1. Select *qpnoise separation* from the *Analysis* section.
2. Select *Source Output* from the *Function* section.
3. Select *V/sqrt(Hz)* or *V**2/Hz*.
4. Select the appropriate modifier.
5. Select the sideband number.
6. Select the type of component to display. If you want all of the types, choose *Include All Types*.

7. If you want to restrict to one device, put its instance name in the *include inst* field. A list of devices separated by a space is also acceptable. In this case, we already know the instance name is I33.I0.NM12. This is entered into the Include Inst field above.
8. If you want to exclude a device (or devices) enter the instance name(s) in the *exclude inst* field.
9. Specify the number of results you want to plot in the *by top* field.
10. Click *Plot*.

The exact noise mechanism within the devices is plotted.

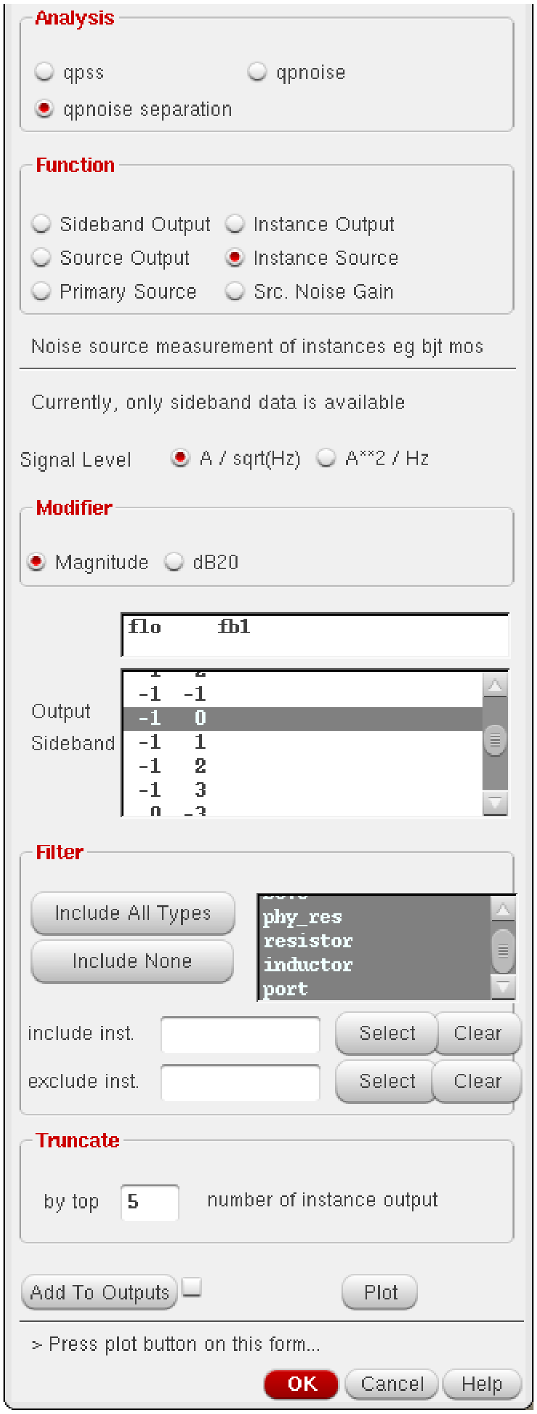


To see which noise source goes with which trace, select the plus sign to the left or the text at the upper left of the display area. The legends will be shown individually in the *Name* field of the waveform tool. The device name and the individual noise contributor will be shown. You can select either the individual legend or the trace and both the legend and the trace will be highlighted. The first part is the instance name, and the second part is the specific noise mechanism within that instance. For a list of the abbreviations used for the noise parameter, refer to Abbreviations Used In the Param Column.

Viewing the total noise currents within the devices

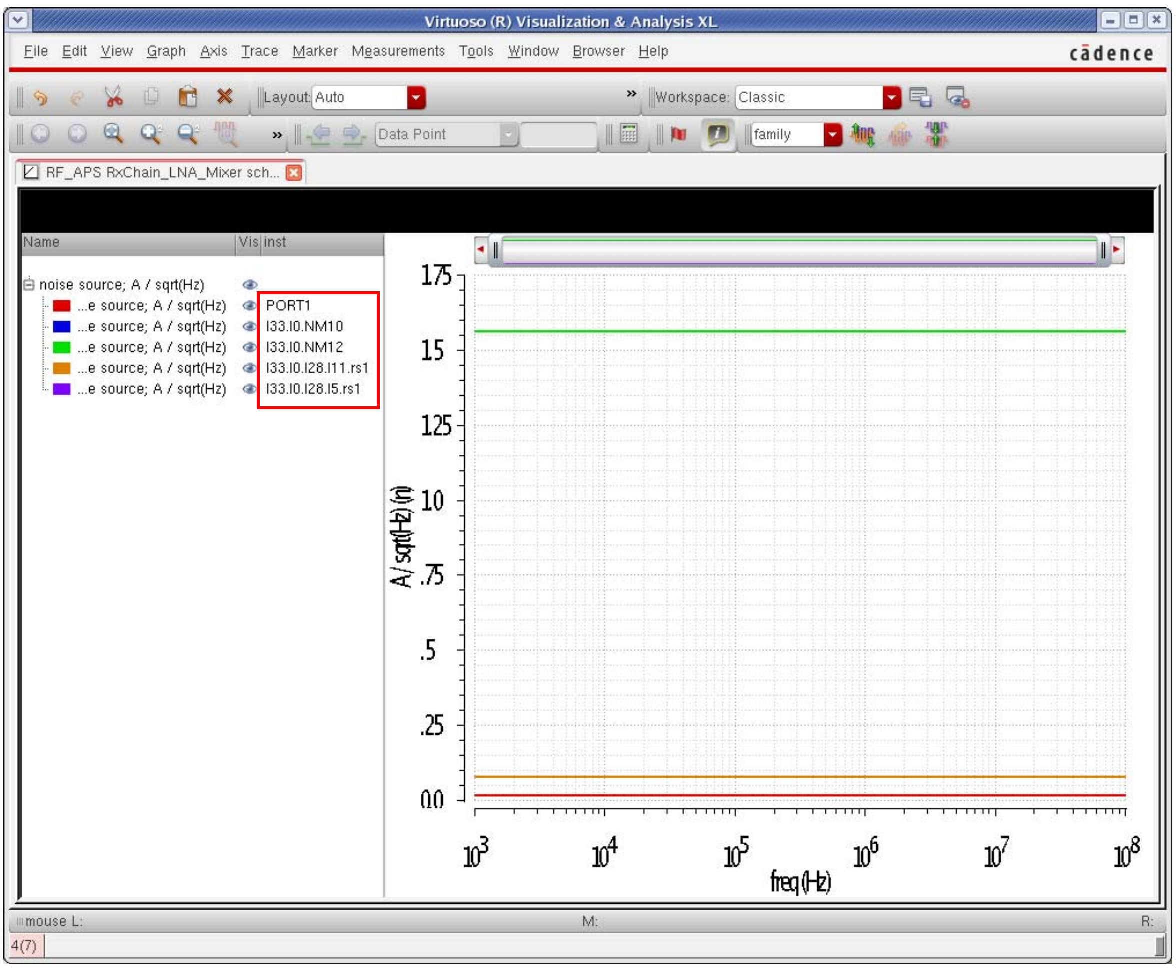
This capability has limited value. All the noise currents from all the noise mechanisms within the instance are added up without regard to the transfer functions to the output. No gain function is applied or are available for this measurement. It just adds up everything within that instance. No correlations are observed. It just adds up the total current.

In the *Direct Plot Form*:



1. Select *Instance source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the Signal Level selection.

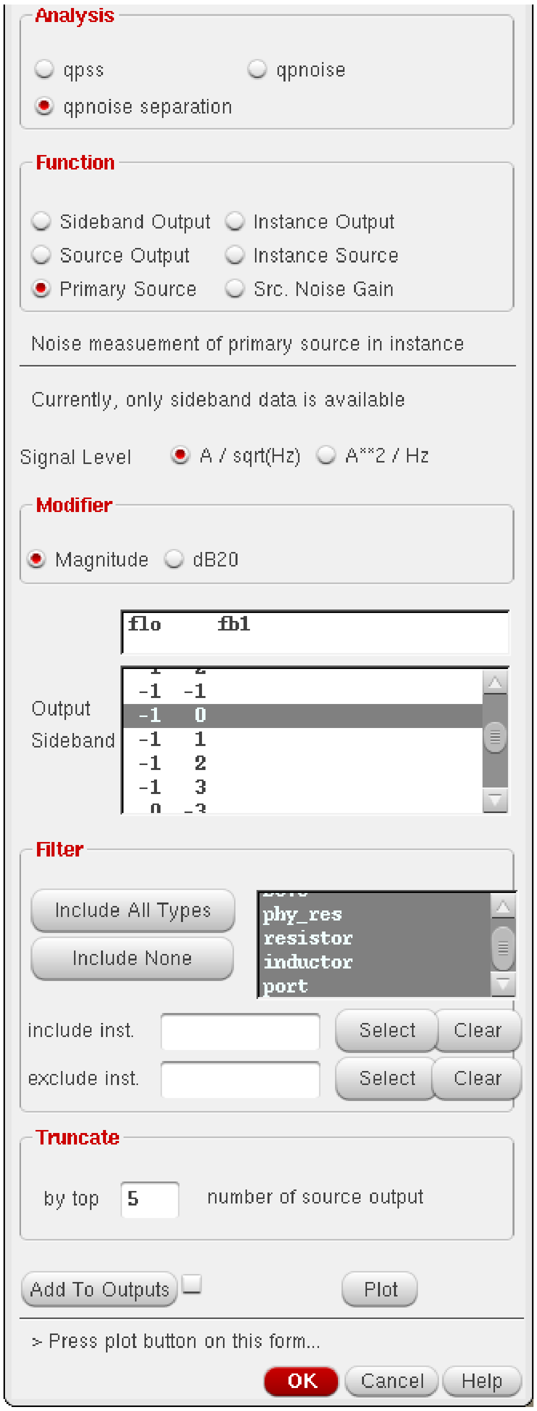
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.
10. The frequency response curves are displayed.



11. The instance names are shown in the legend.

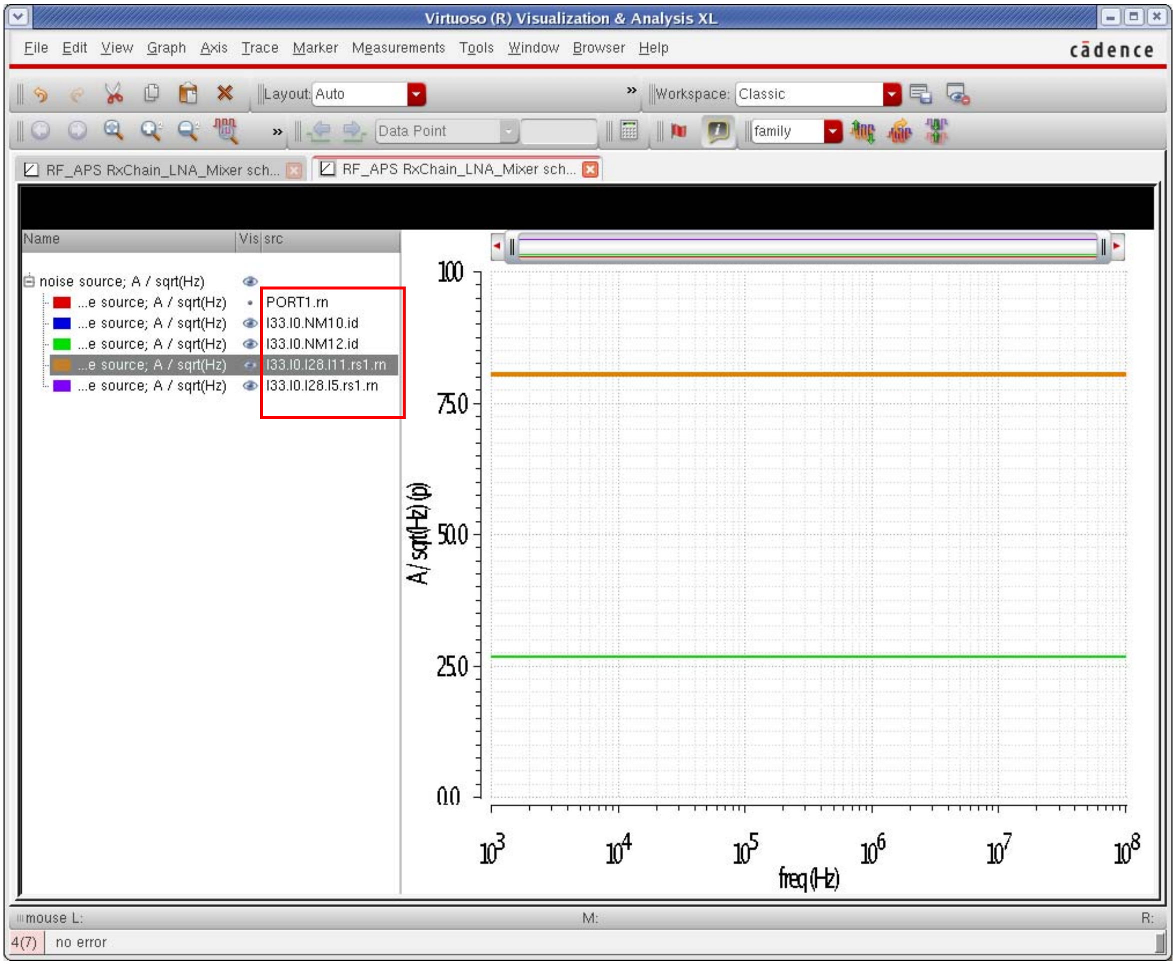
Plotting the individual noise source currents

This capability plots the individual noise currents within the device. This capability gives you the noise current at the source instead of at the noise at the output of the circuit. This capability allows you to identify the largest noise sources in the circuit.



1. Select *Primary Source*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* selection.
3. Select *Magnitude* or *dB20* in the Modifier section.
4. Select the sideband number.
5. If you want every noise source to be considered, select I*nclude All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic.
8. Set the number of curves to plot by setting the Truncate function.
9. Click *Plot*.
10. The frequency response curves are displayed

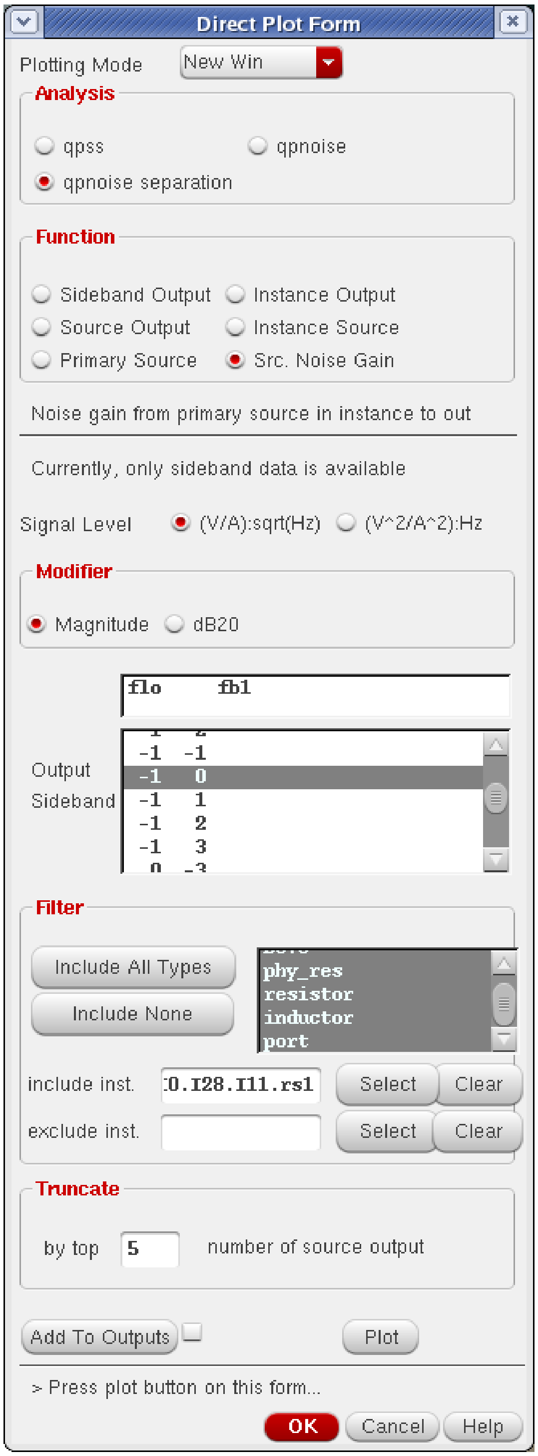
11. The specific noise sources are displayed in the legend.



For a list of the noise parameter names, Abbreviations Used In the Param Column. In this list, *rn* is a parasitic resistance, and *id* is the thermal noise from the channel.

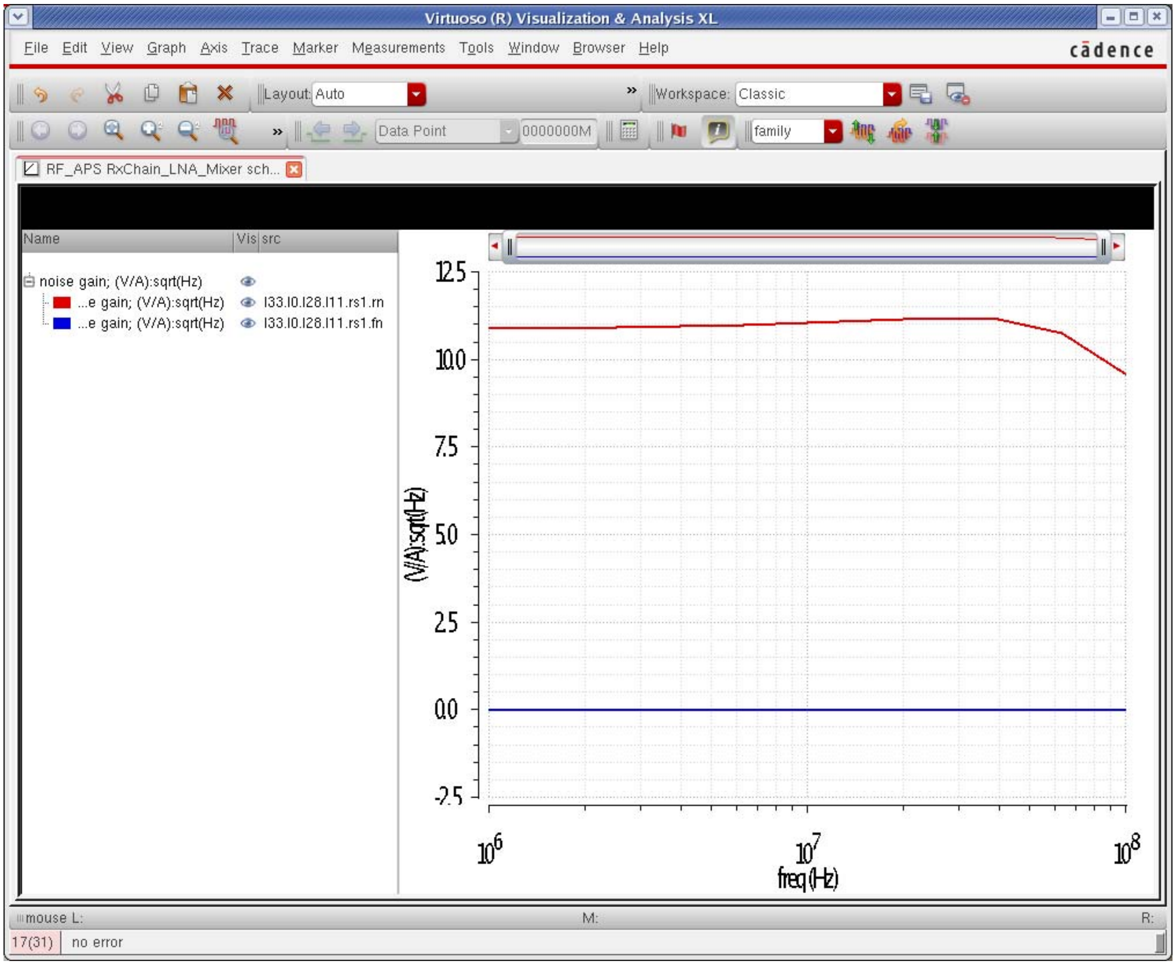
Plotting the gain from the individual noise sources to the output

This capability gives the gain from the individual noise currents to the output of the circuit.



1. Select *Source Noise Gain*.
2. Select *A/sqrt(Hz)* or *A**2/Hz* in the *Signal Level* selection.
3. Select *Magnitude* or *dB20* in the *Modifier* section.
4. Select the sideband number.
5. If you want every noise source to be considered, select *Include All Types*.
6. To exclude instances, click *Select* to the right of the *exclude inst* field, and select the device instances you want to exclude from the plot. Everything will be plotted except for those instances.
7. To include specific instances, click *Select* to the right of the *include inst.* field, and select the instances on the schematic. All other instances will not be in the results when you use this capability. this example, the instance name of the largest noise source from the previous section was entered in the include inst field.
8. Set the number of curves to plot by setting the *Truncate* function.

9. Click *Plot*.

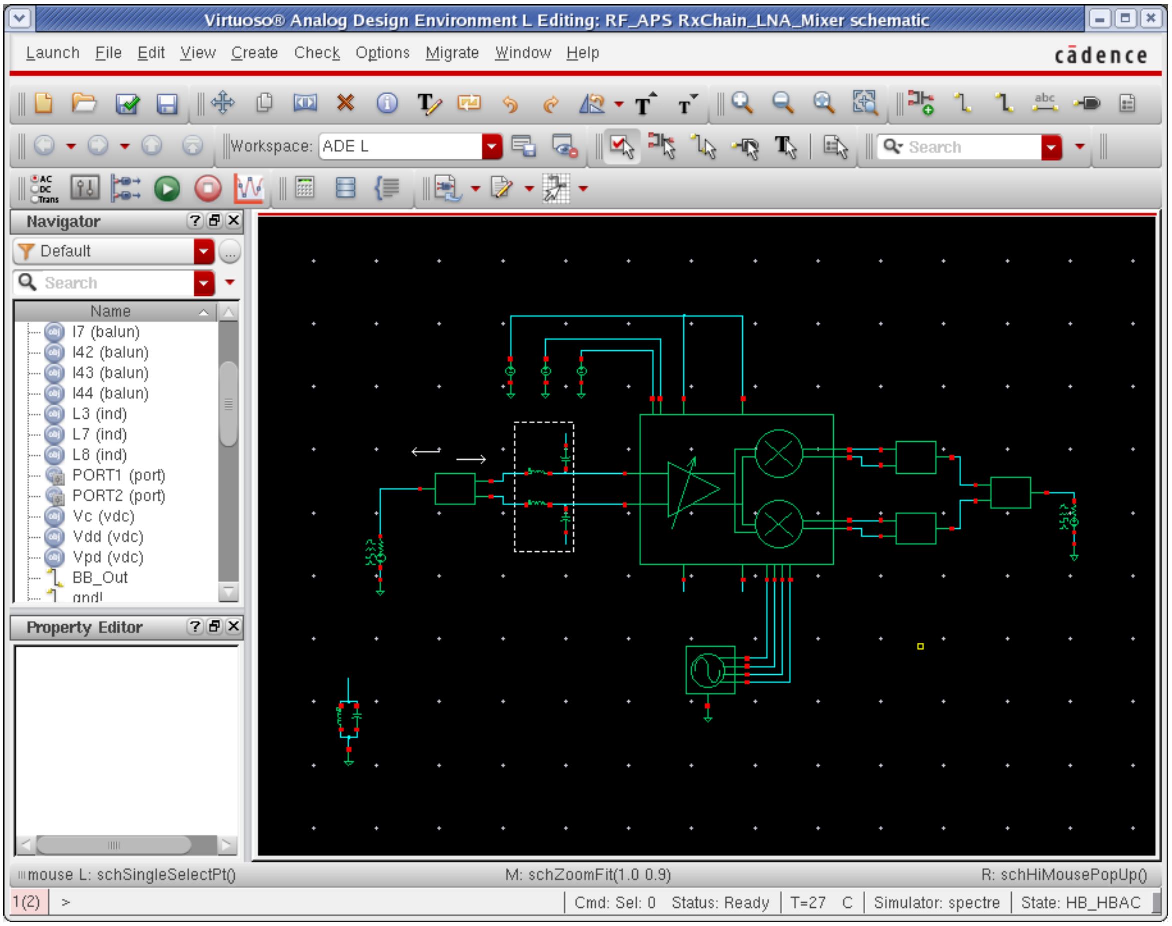


The frequency response curves are displayed. For a list of the abbreviations used for the noise parameters, Abbreviations Used In the Param Column. In this list, *fn* is flicker noise, and *rn* is resistor thermal noise.

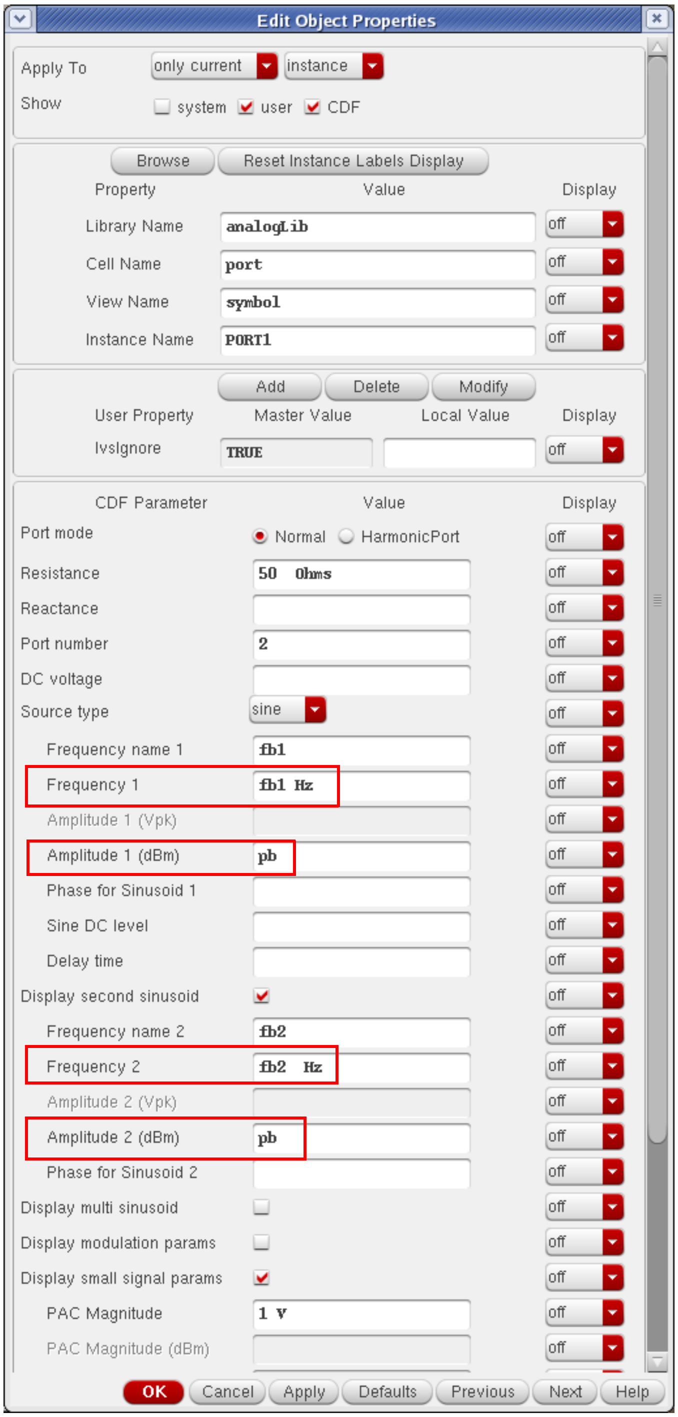
Noise as a Function of Blocker Power

This measurement is almost always faster when harmonic balance is used. Consider using hb and hbnoise analysis as described in Chapter 3, “Frequency Domain Analyses: Harmonic

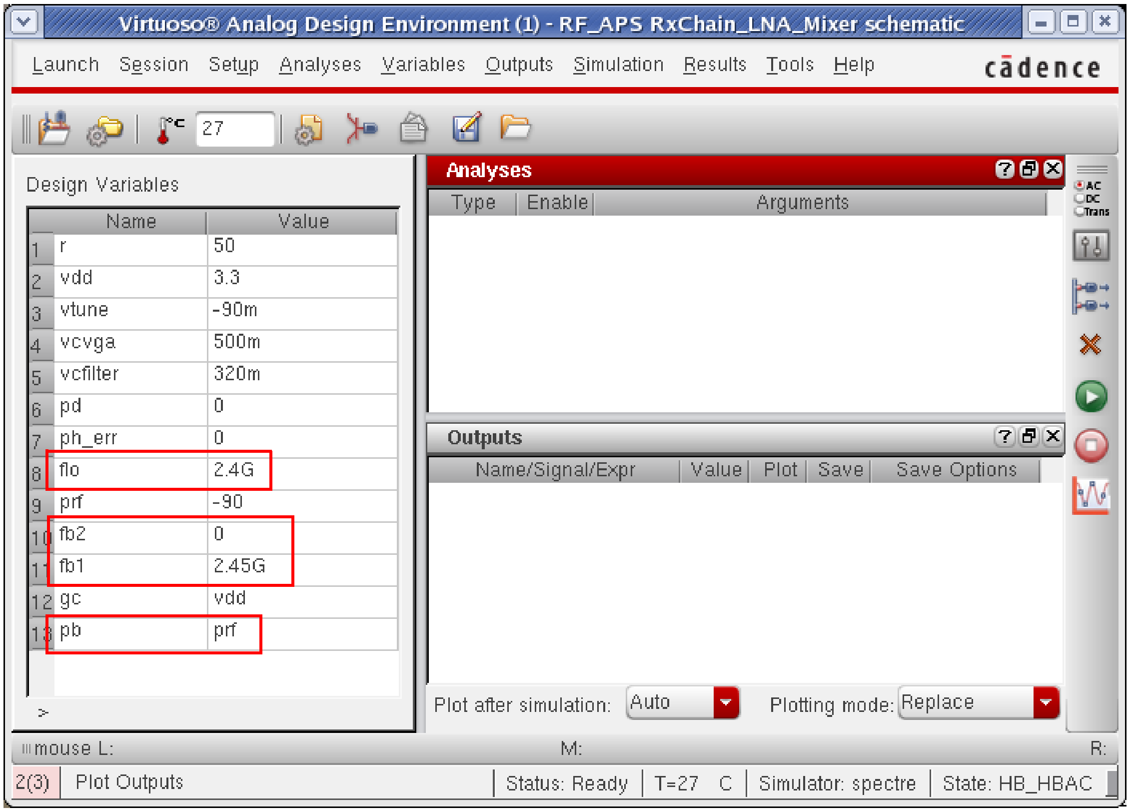
Balance," because the hb and hbnoise forms always include the latest improvements, and qpss and qpnoise generally are not improved. Consider a 2.4GHz LNA-Mixer.



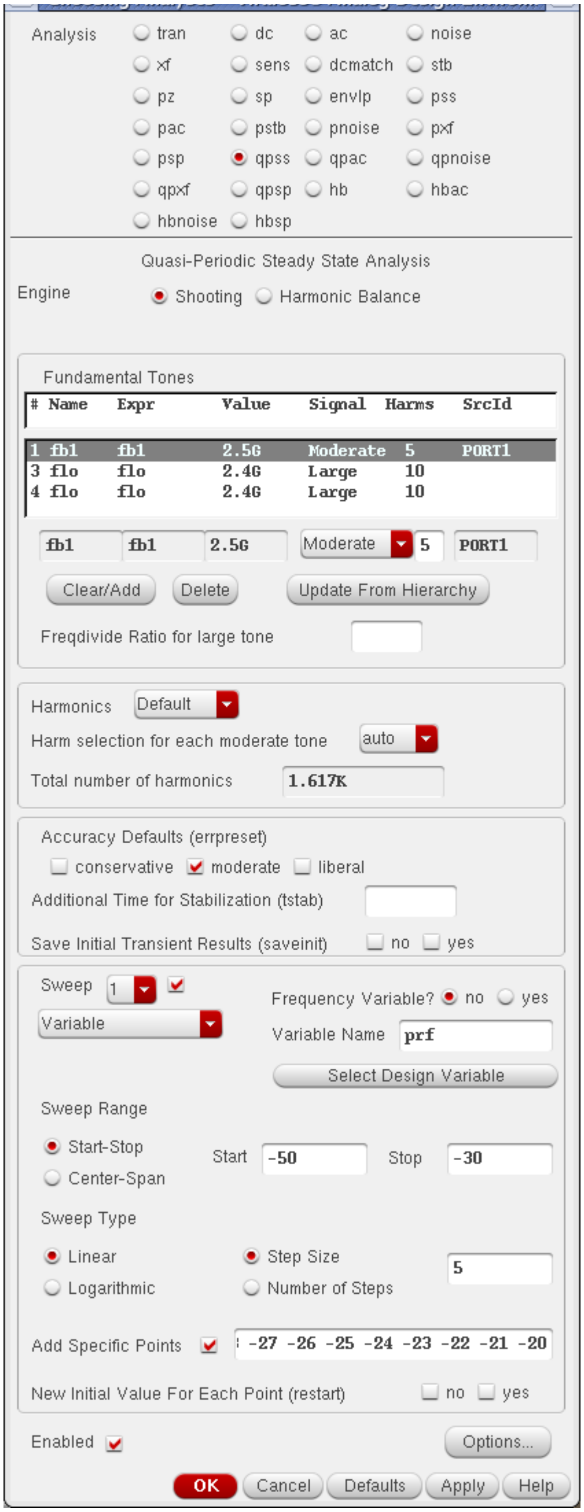
The input source has variables set to a variable name. This is done in order to allow easy frequency and amplitude changes from the ADE environment without needing to change the schematic. Here is the properties list for the input port.



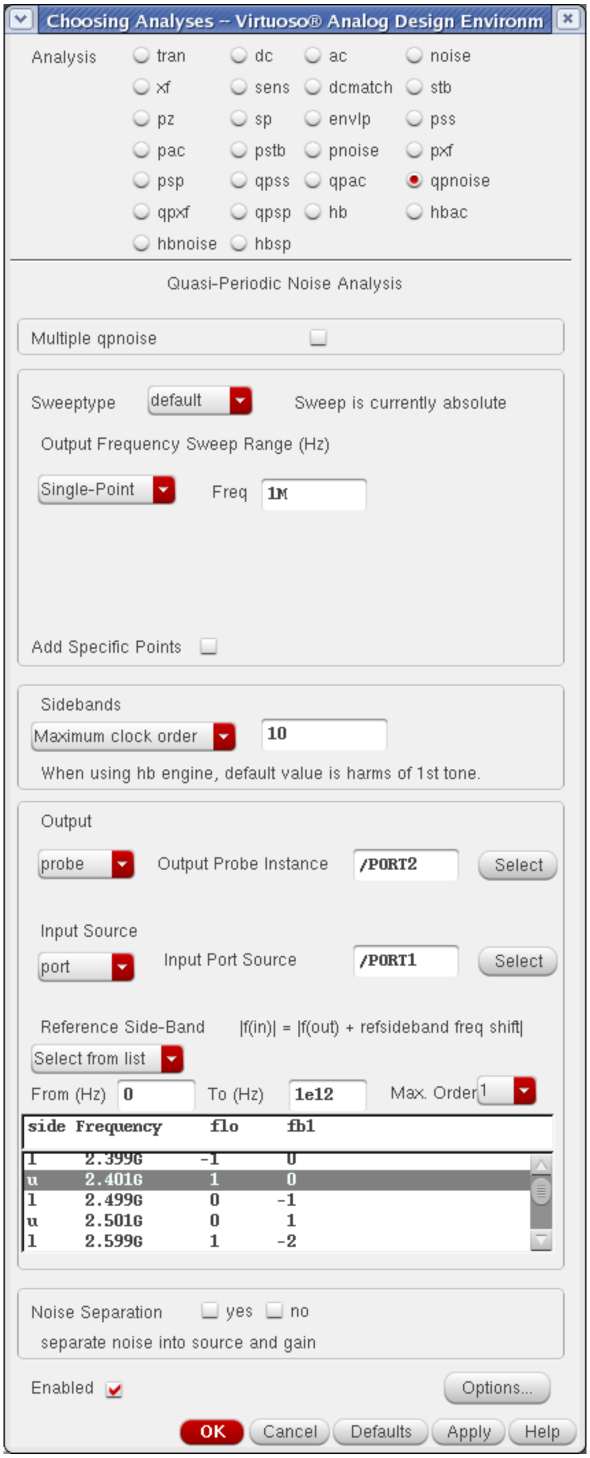
The variables are set in the ADE environment.



1. First, the qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter. Here, a 2.4GHz LO and a 2.45GHz blocker are set up. Set up a sweep of the blocker power that is appropriate for your circuit. In this example, the blocker power is swept from -50 to -20dBm. Because the power will be swept to a large amplitude, 5 harmonics are set on the moderate tone.

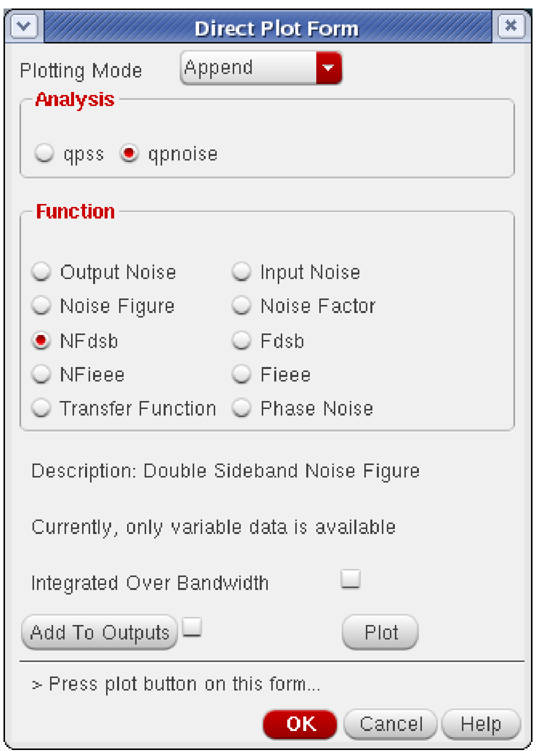


2. Now set up the qpnoise analysis.



3. When the power is swept in qpss, the *Output Frequency Sweep Range* is set to *Single Point* automatically. Set the frequency to an appropriate output frequency for your design. In this example, 1MHz is set.
4. *Maximum clock Order* should be set to the number of harmonics that are used for the LO in the qpss *Choosing Analyses* form. If harmonic balance is used as the qpss engine, this field can be left blank, which has the same effect.
5. The output port and the input port are defined. Specifying Probe for the output and selecting either a port or a resistor automatically excludes the noise of the selected component from the noise figure calculation.
6. Using *Select from list* from the *Reference Side-Band* drop-down list is highly recommended. Select the passband frequency (the design input frequency) from the list. In this example, the input frequency just above the 2.4GHz LO is selected.
7. Click *OK* and run the analysis. When the analysis finishes, select *Results - Direct Plot - Main Form* from ADE. The *Direct Plot Form* is displayed.

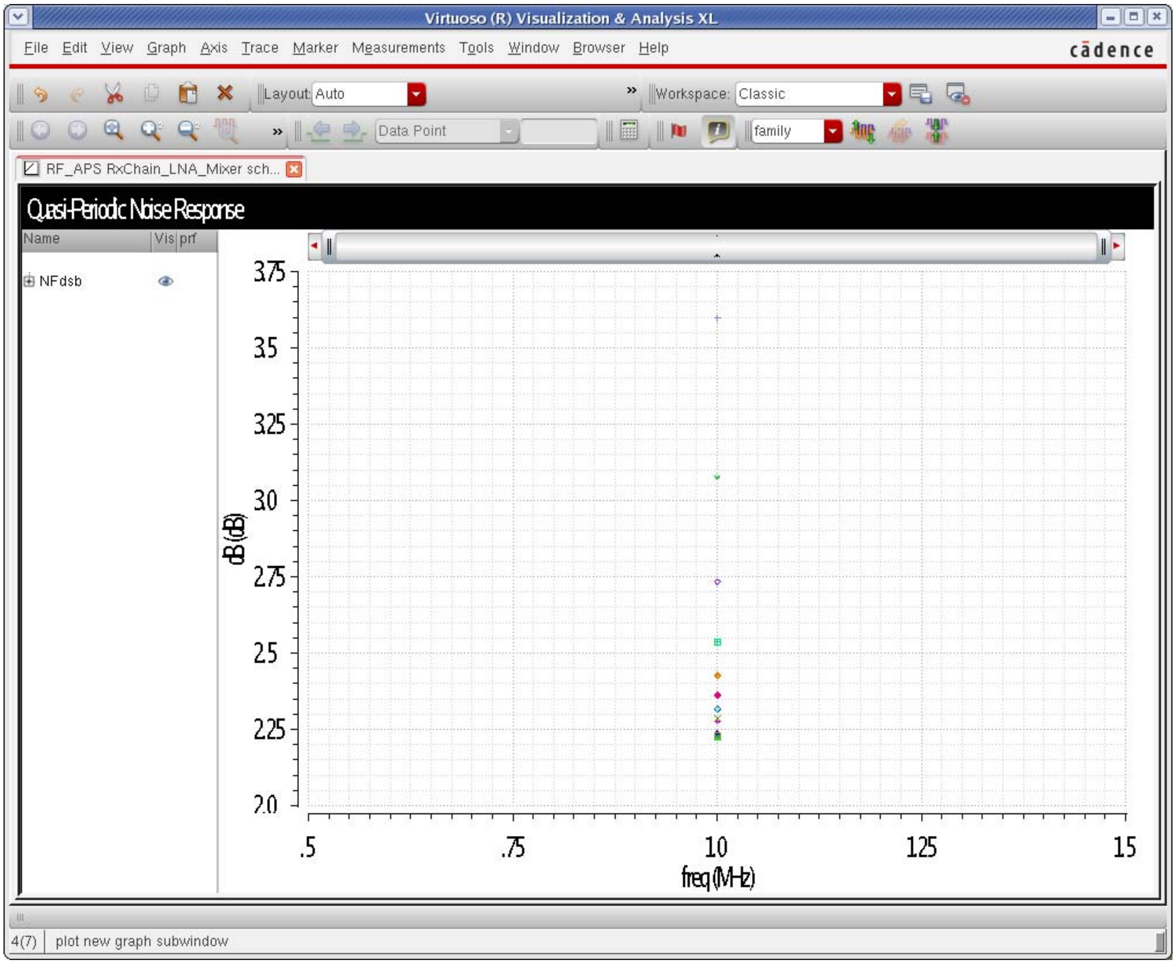
.



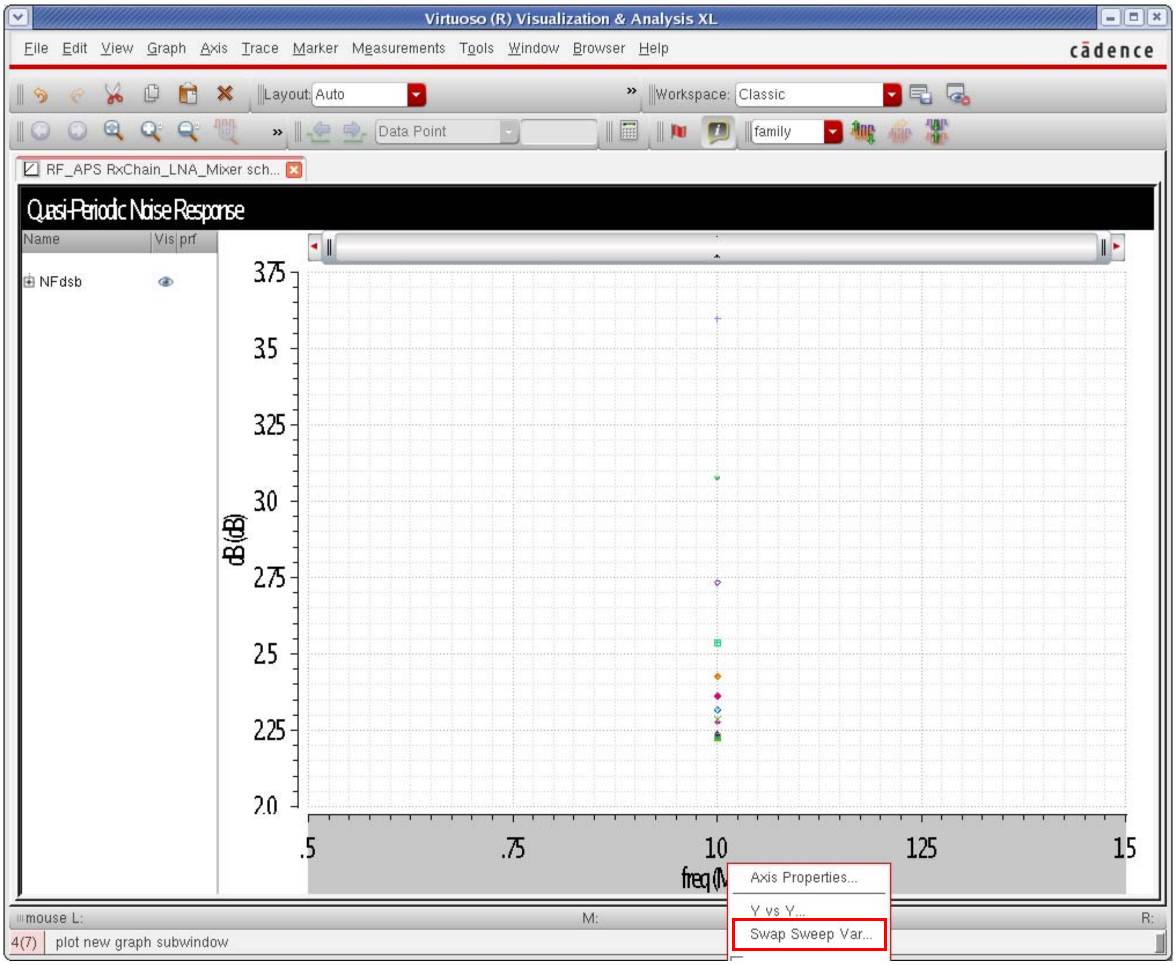
8. Select *qpnoise* in the *Analysis* section.
9. Select the desired measurement and click *Plot*.

   In this example, the double-sideband noise figure is plotted

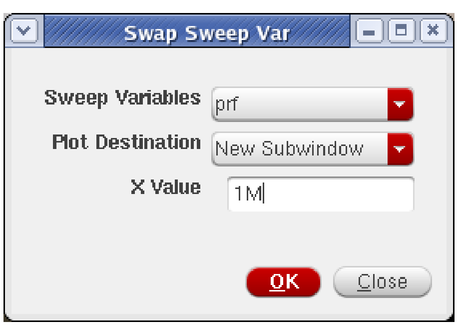
The output appears in the waveform tool.



10. Next, the display will be made easier to read. Move your mouse cursor over one of the numbers on the X Axis, click the right mouse button and select *Swap Sweep Var* from the context menu.

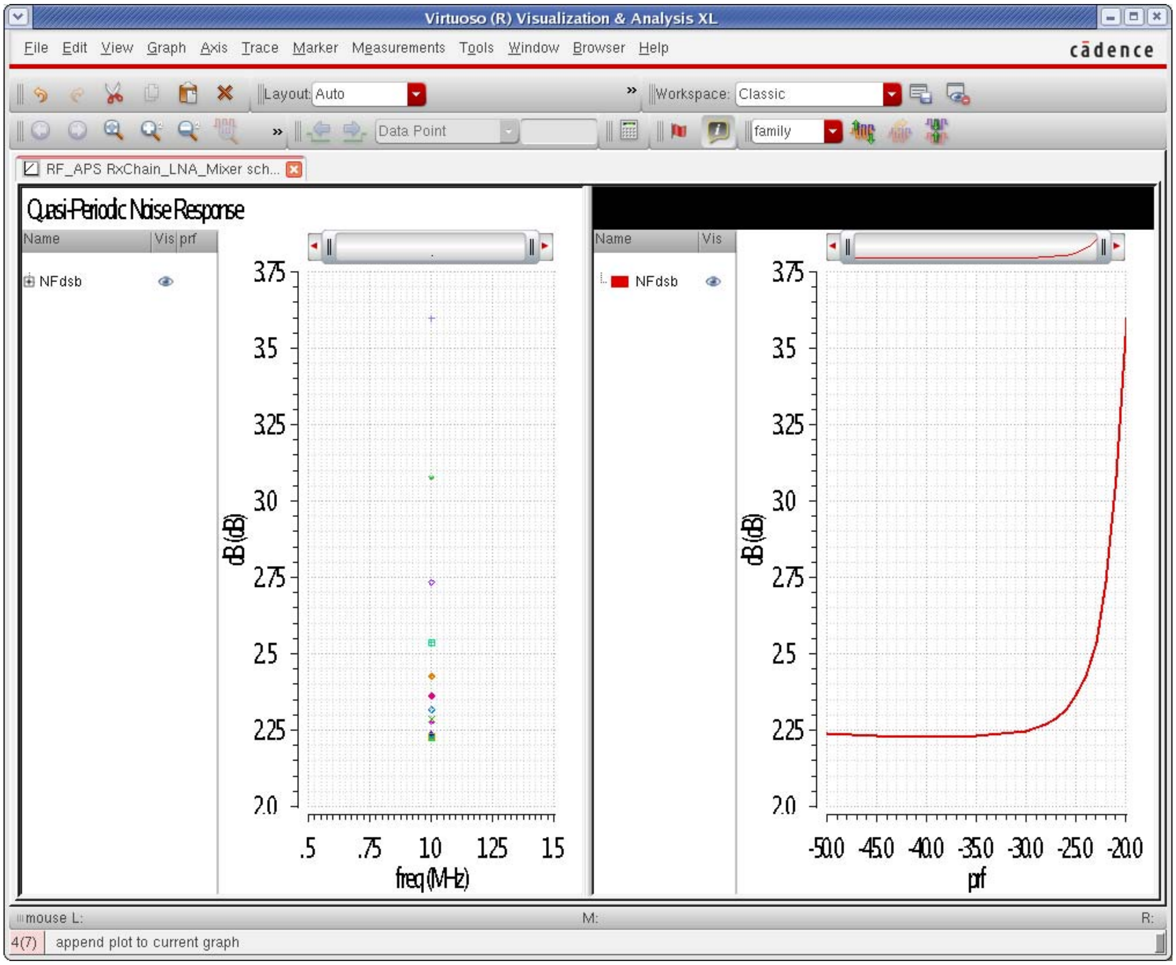


11. In the *Swap Sweep Var* window, the sweep variable is selected automatically. Type in the X-Axis value for the measurement. In this example, this is 1M.



12. Click *OK*.

The waveform window adds a subwindow with a more traditional presentation.



13. As expected, the noise figure increases with increasing blocker power.

QPXF Analysis

Usually, qpxf is used to measure forward conversion gain when an LO and a high amplitude blocking signal are applied to the circuit. In this application, hb with hbxf is likely to run faster than shooting qpss with qpxf. Although harmonic balance is available in qpss, it is strongly recommended that you use hb and hbxf because the latest capabilities in the simulator are always available there.

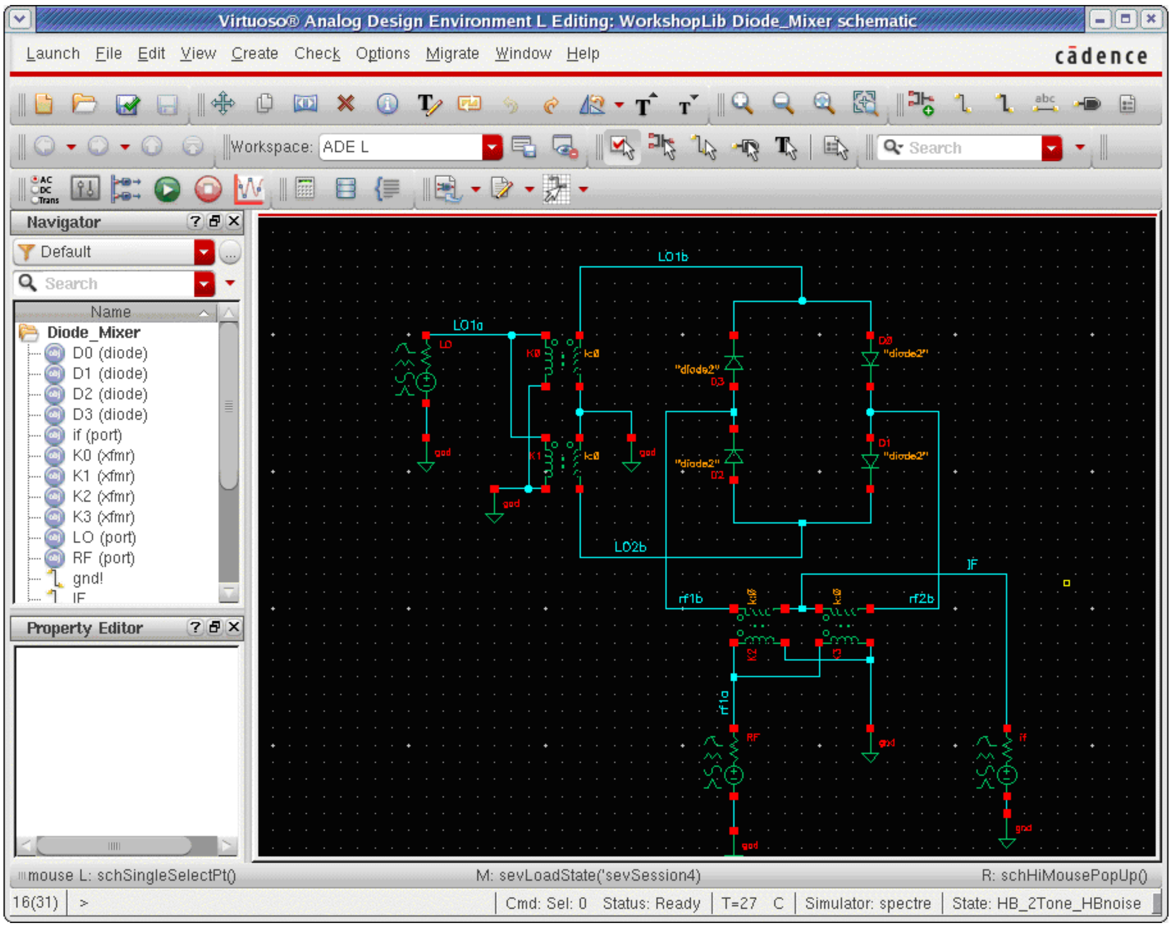
For more information, see Periodic Transfer Function Analysis (PXF) on page 1060.

Qpxf is just like pxf in that qpxf calculates forward conversion gain. The difference is that qpxf runs after a qpss analysis where there are more input frequencies than in pss, and as a result, the circuit produces many more harmonics. Because there are many more harmonics in the large-signal analysis, there are many more transfer functions to calculate. Unlike pxf, qpxf has only one mode which measures the conversion gain of the system. If you need to measure AM to PM conversion (using modulated) or an instantaneous transfer function (using sampled) you need to use pss-pxf which has these capabilities. Qpxf does not have these capabilities.

As engineers, we are accustomed to driving the input and measuring the output of the circuit. This is the idea of qpac. In qpac, we apply the LO signal (or the sample clock) and a blocking signal in the qpss analysis, and then we drive the input with the small-signal qpac analysis and measure the mixing products that come out of the circuit. While this is very intuitive, the output display can be very confusing to read when the conversion gain near several qpss harmonics is desired. Qpxf changes the focus to the output. One output frequency at a time is considered, and in qpxf, the input frequencies that cause the output frequency are calculated, and the forward conversion gain from all those input frequencies is also calculated. Using qpxf, you can measure the conversion gain near a number of LO harmonics (or clock harmonics) and blocker harmonics in the same simulation run.

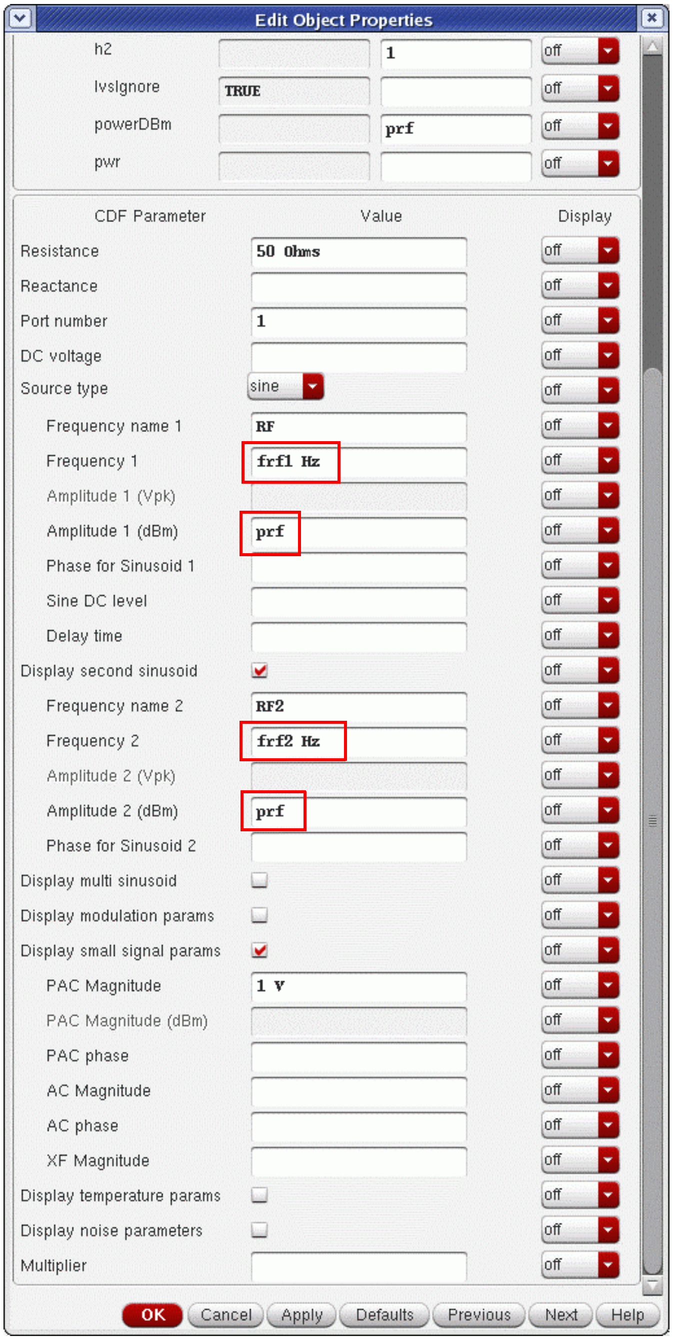
Example: Conversion Gain

Consider the double-balanced diode mixer shown below.



The RF input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. This allows changes in frequency or amplitude from the ADE environment without needing to change the schematic itself.

The property list is shown below.

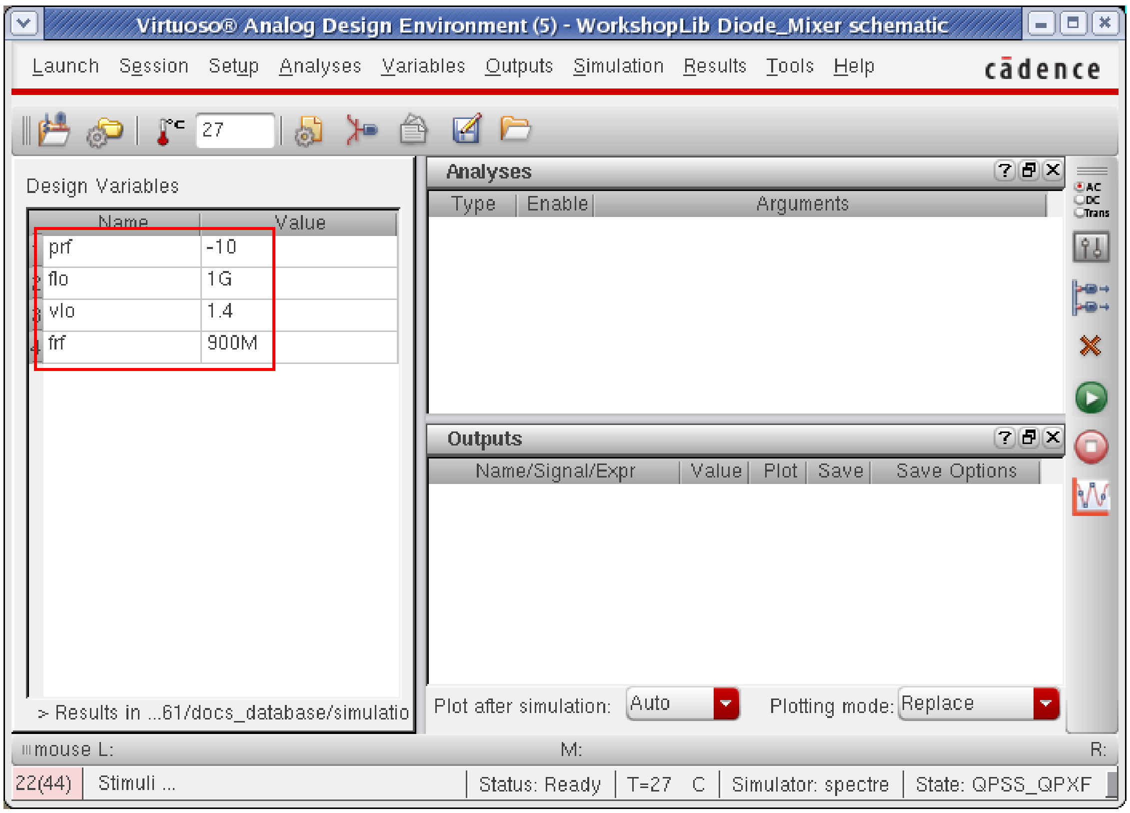


The LO signal also uses variable names.

The easiest way to get the variables list into the ADE is to select *Variables - Copy From Cellview*. To set the values, select the value field to the right of the variable, type in the desired value, and press *Enter.* When you have set the variables, if you want to add the

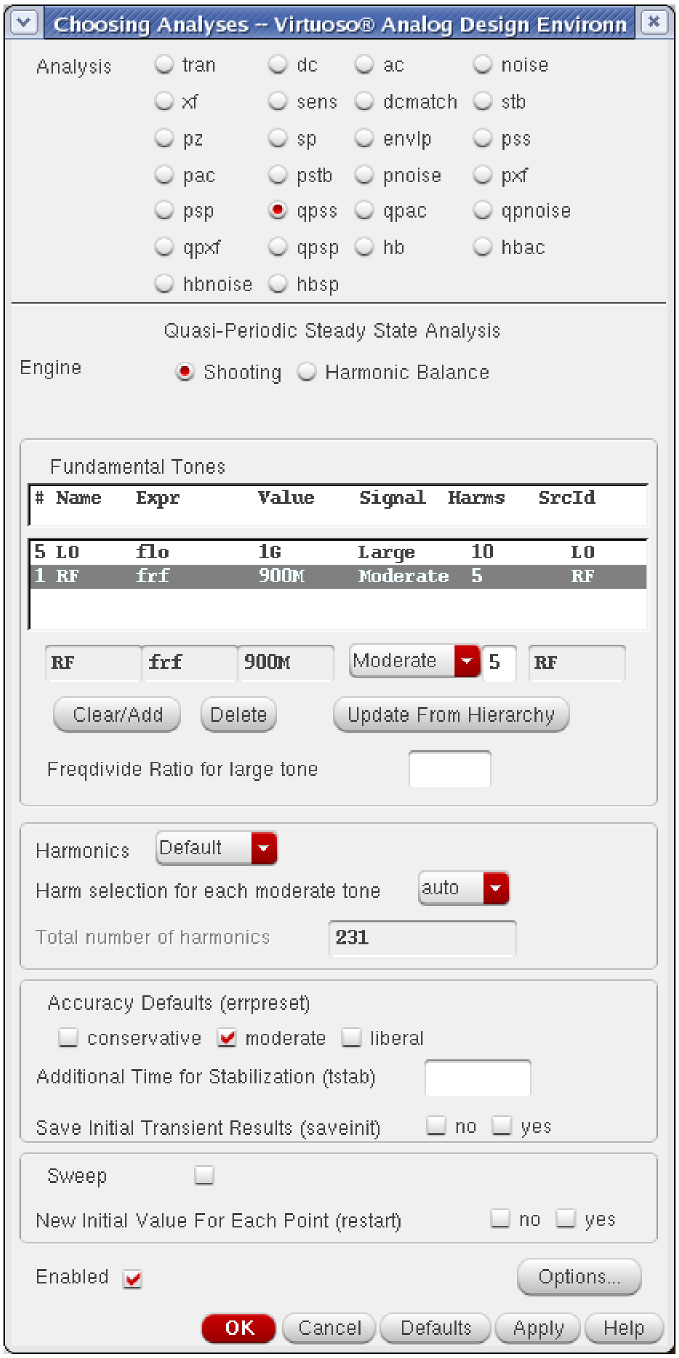
values back into the cellview, in the ADE window, select *Variables - Copy To Cellview*, and then perform a *Check and Save* in the schematic.

In this example, the LO frequency is 1GHz and its amplitude is 1.4 volts peak. This is +13 dBm, which is a common drive level for diode mixers. The RF frequency is 900MHz, and the amplitude is set to -10dBm.



The basic strategy is to apply the signals that cause the nonlinearity to occur in the qpss analysis, and then measure the forward conversion gain using qpxf analysis.

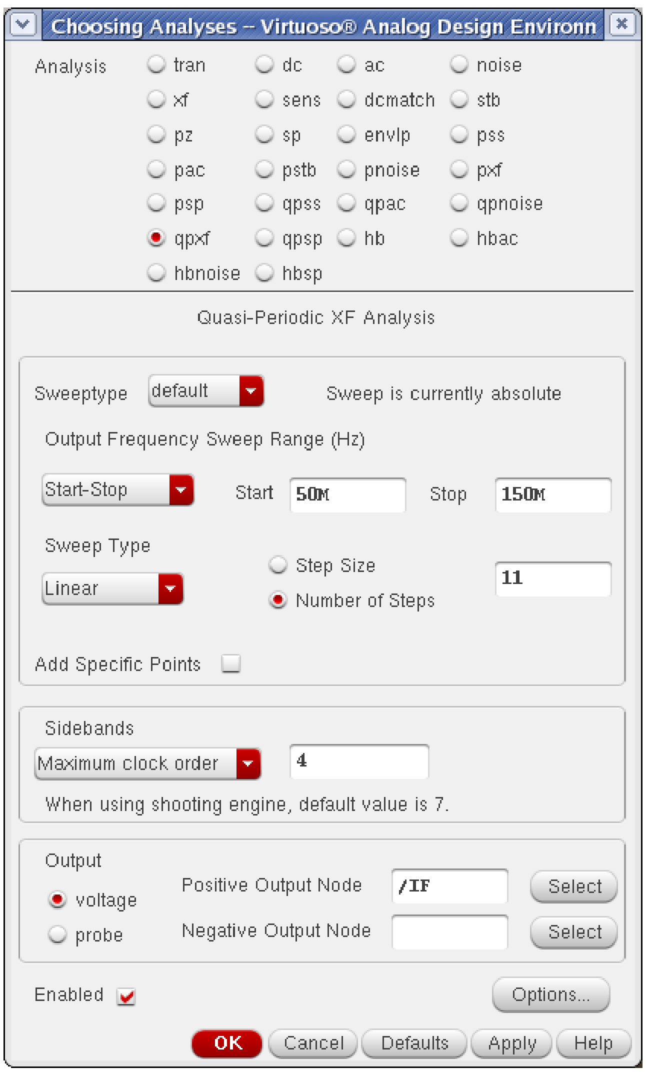
The qpss analysis needs to be set up first. The example below shows an LO signal at 1GHz with 10 harmonics, and the RF input at 900MHz with five harmonics. For more information, see the qpss section at the beginning of this chapter.



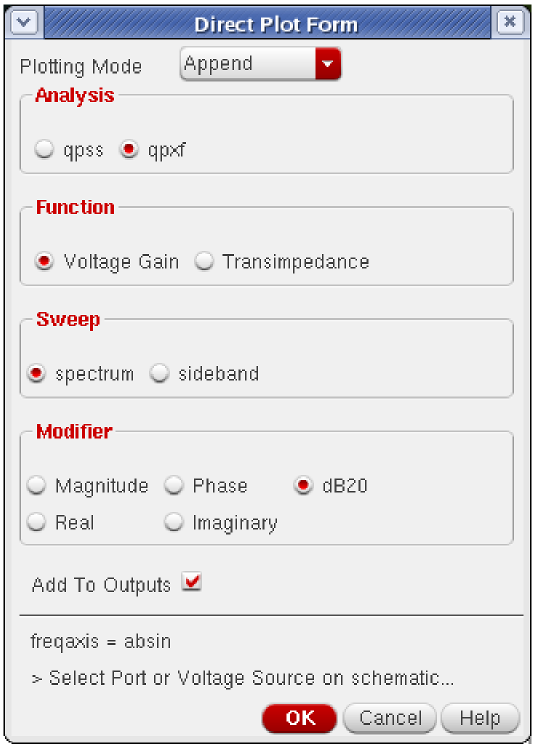
Next, the qpxf analysis needs to be set up. Like xf, pxf, and hbxf, the frequency range specified in the *Choosing Analyses* form is the **output** frequency range. Usually, it is better to select Linear or Log spacing and define the resolution of the sweep yourself. Most of the time, Automatic will take more frequency points than you need, and that extends the runtime.

When shooting is used as the qpss engine*, Maximum clock order* should be set to three or four. Usually, the higher order conversion gain products are not needed. If harmonic balance is used for the qpss engine, either leave this field blank, or set *Maximum clock order* to the same number you used for the LO harmonics. In either case, the *Direct Plot Form* can be used to select the result that you want.

Select the output net in the circuit. The conversion gain terms will be measured to this net. If you have an output current, add an iprobe from analogLib in series with the output, and select *probe* in the *Output* section. Then specify the instance name of the iprobe in the *Output Probe Instance* field.

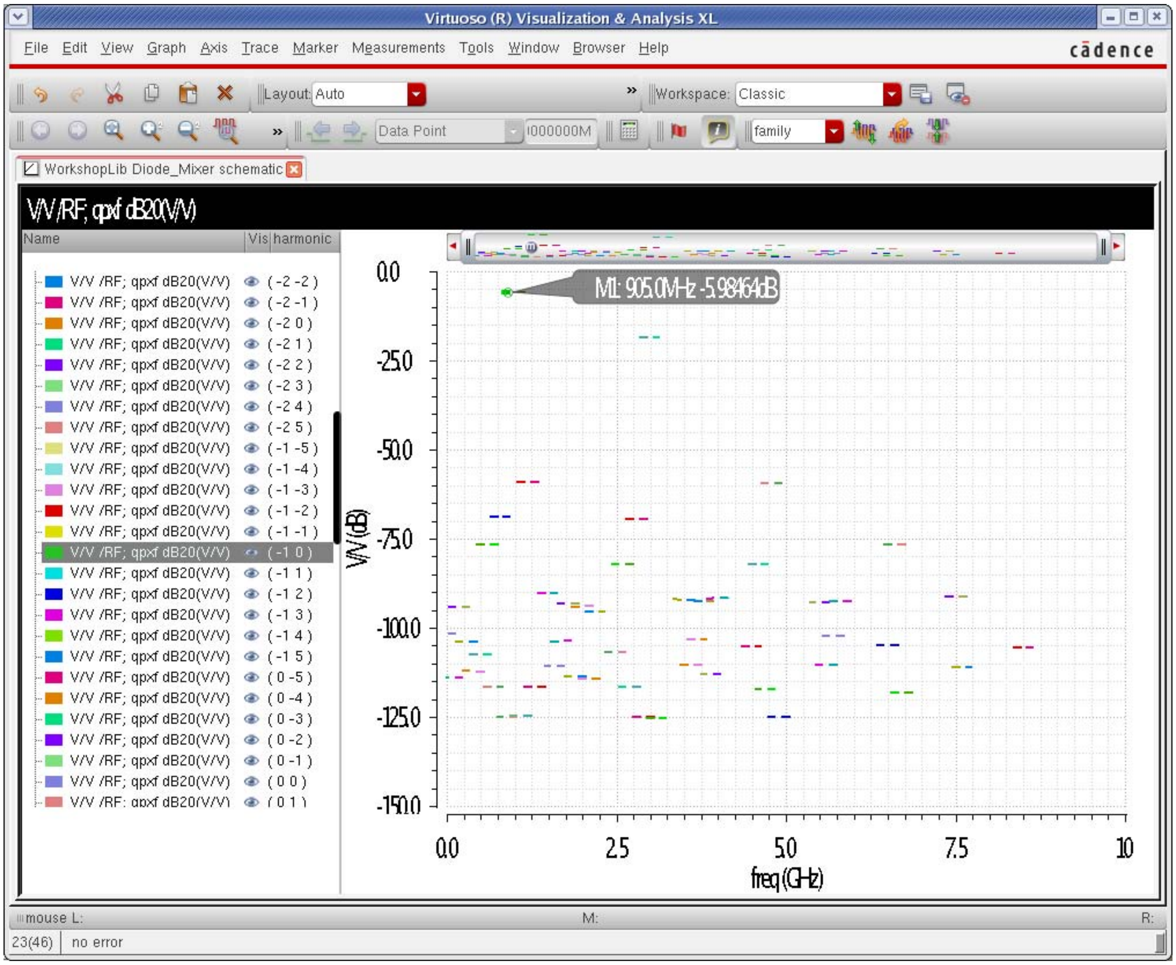


Once the analyses are set up, the simulation can be run. When the simulation finishes, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* is displayed.



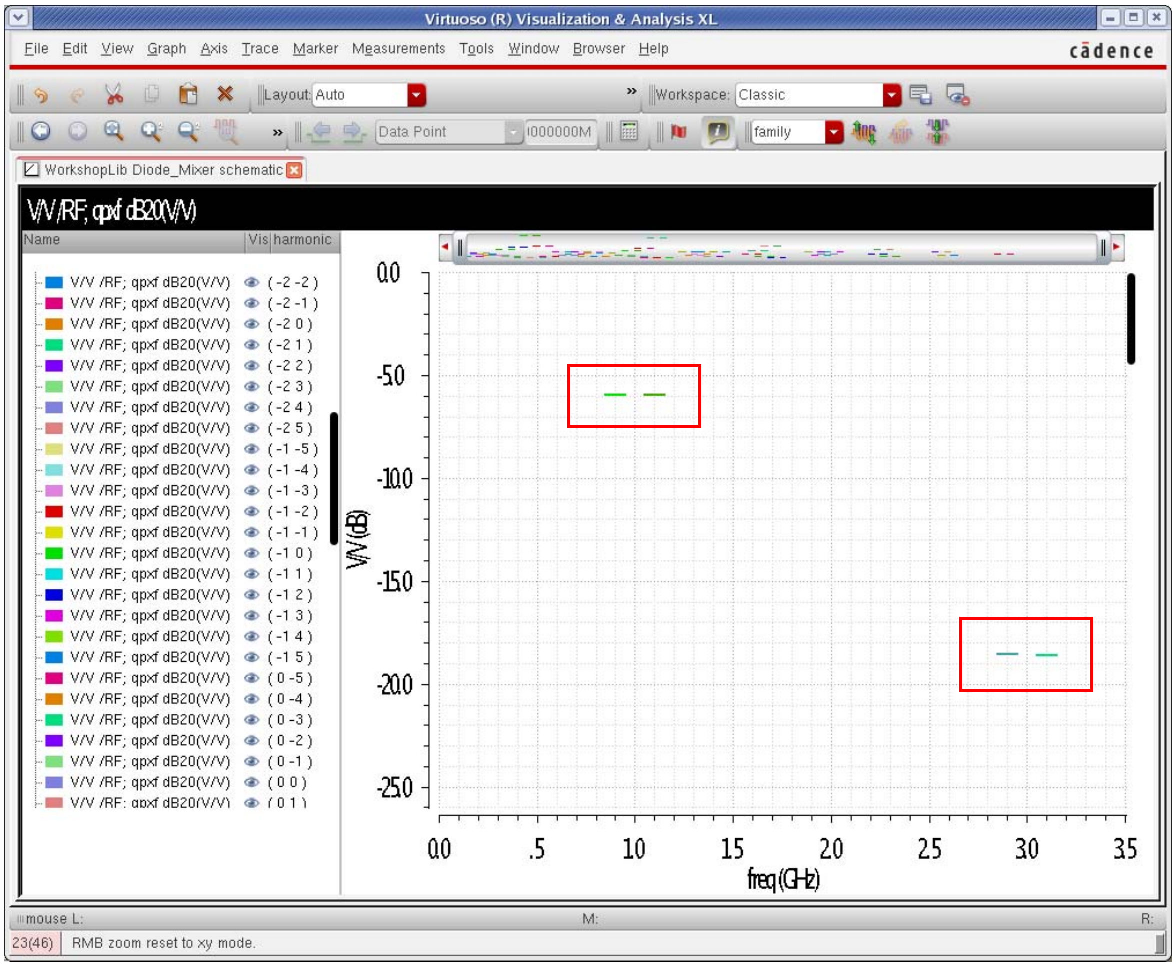
The example below shows a typical plot

.



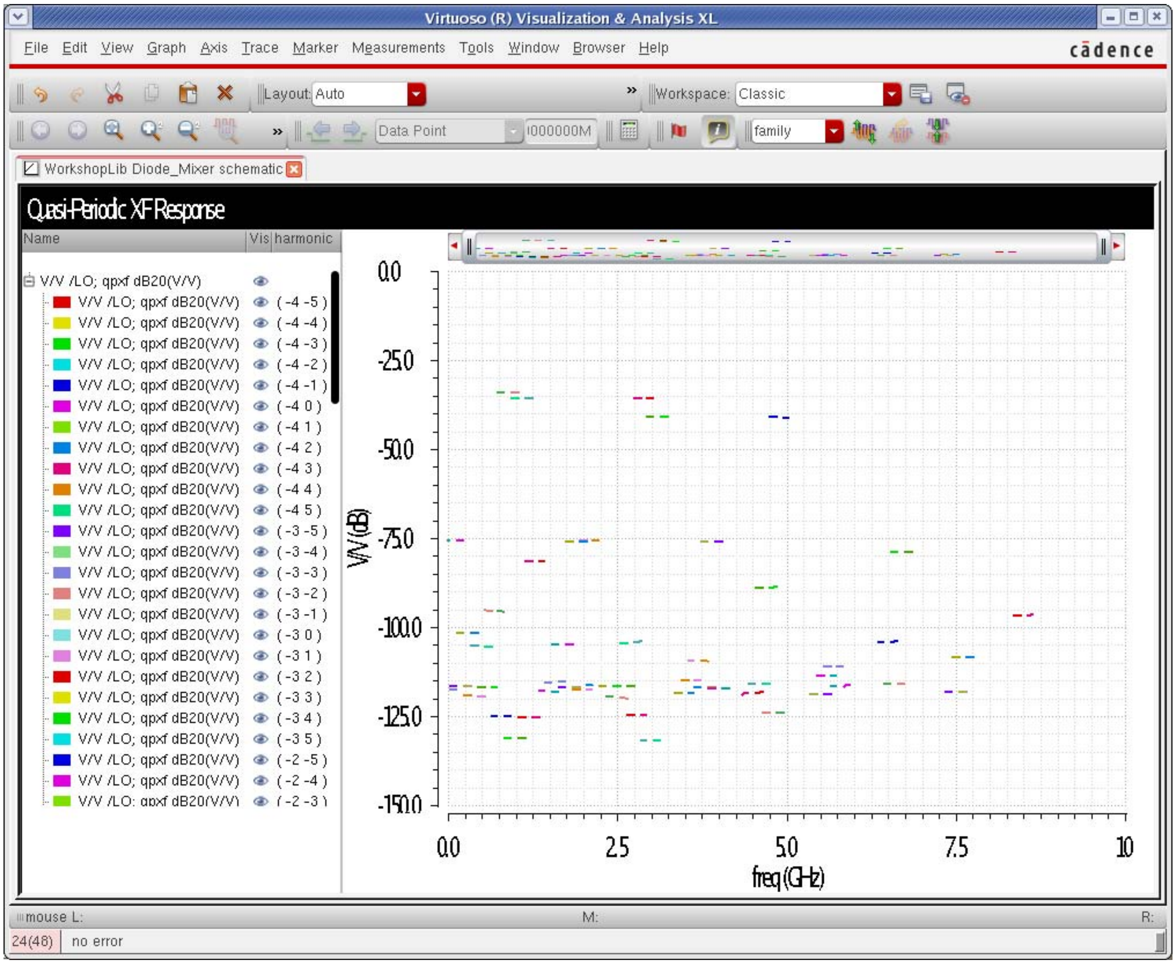
To measure the conversion gain, place a marker at the desired **input** frequency, and read the forward conversion gain. The conversion gain is about -6dB as expected.

In a zoomed-in view, the conversion gain near the first harmonic of the LO (1GHz) is about 6 dB. The conversion gain near the third harmonic of the LO is also quite high at about -18.5 dB.



The qpxf direct plot is a bit non-intuitive. The *Choosing Analyses* form specifies the output frequency range and the output node. Qpxf calculates the transfer functions from all the sources in the circuit to that output node. Therefore, to plot, you select a source that you want the forward conversion transfer functions to be displayed. When the source in the schematic is selected, the waveform is displayed.

Note that all the short line segments are the forward conversion gain from the selected source (the LO source in the case below) to the output. None of the conversion gain terms are very high from the LO port to the output.



Overview of Simulation Capabilities

The primary application of qpxf is to measure conversion gain. Qpxf is a small-signal analysis. It calculates the small-signal conversion gain. If you need a large-signal measurement of conversion gain, then use a large-signal analysis like harmonic balance (hb) or quasi-periodic steady-state (qpss) analysis. Qpxf measures gain in a way that requires some thought for most engineers because we are so accustomed to connecting an input, driving the system, and measuring what comes out. This is the idea of qpac, where we capture the large-signal operating point in qpss, and then drive a small-signal input, and measure the mixing products that come out. If you think of this methodology, to measure the conversion gain around several harmonics usually requires a separate simulation near each harmonic.

Qpxf was created to allow the measurement of the conversion gain transfer functions near all the qpss harmonics at the same time. In qpac, we had one input frequency at a time, and output mixing products at multiple output frequencies are calculated. In qpxf, we focus on one output mixing product at a time, and measure the input frequencies that will cause that output frequency, and then calculate the forward conversion gain from all those input frequencies to the single output frequency. Multiple output frequencies are usually analyzed in a single qpxf analysis. Unlike qpac where the output can be seen at every node in the circuit, qpxf calculates the gain only from all the independent sources in the circuit to the output. Thus, to measure a conversion gain, you select the source instance in the schematic from the Direct plot form instead of selecting a net.

For example, imagine a mixer with an LO frequency of 1GHz and an RF signal at 900MHz. (Think easy math.) Now, imagine the down conversion case where the output is 1MHz. Given the LO frequency, inputs at 1MHz, 999MHz, 1.001GHz, 1.999GHz, 2.001GHz, 2.999GHz, 3.01GHz, and so on all cause an output at 1MHz. Signals 1MHz on either side of the RF harmonics will also convert to 1MHz, so 899MHz, 901MHz, 1799MHz, 1801MHz and so on will have conversion gain terms. There are also intermodulation products that are created by the circuit, and because there are so many conversion gain terms, it can be a bit confusing. Depending on the circuit, all those different input to output paths will have different gains from each input frequency to the output at 1MHz. In qpxf, you specify the highest qpss large signal harmonic to calculate the transfer functions from in the Maximum clock order field in the Choosing Analyses form. All these different input frequencies are called sidebands in qpxf. There is the ability to plot just the desired conversion gain terms in the Direct Plot form.

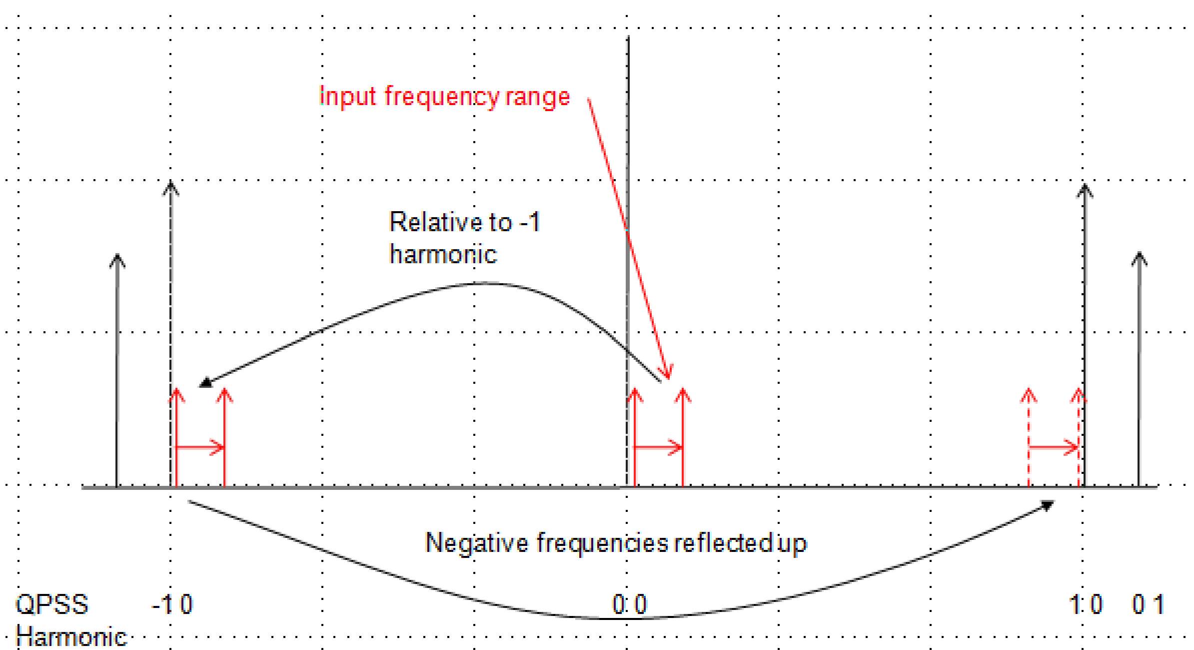
Frequency Sweep

This analysis is like xf and pxf, where the **output** frequency is varied. The output frequency range to be analyzed is specified in the qpxf *Choosing Analyses* form. Qpxf then calculates the conversion gain terms.

In addition to the frequency, there is a selection called *Sweep Type*. The choices are *relative* and *absolute*. When *absolute* is selected, the frequency range is used directly with no frequency conversion. When *relative* is selected, a *Relative Harmonic* field appears. The frequency sweep is shifted by the specified frequency of the harmonics in the qpss analysis. For example, assume that the qpss has signals at 1GHz and at 1.1GHz, and a log sweep is desired above the first harmonic of the 1GHz signal in the qpss analysis. In this case, you need to select *relative* sweep, and specify 1 0 in the *Relative Harmonic* field. Next, type in 1K to 100M for the frequency range with a log sweep and 3 to 5 points per decade. The first number is the frequency shift in integer multiples of the Large signal in qpss, which is usually the LO frequency. The second number is the frequency shift in integer multiples of the RF blocker frequency. Since usually we design to mix with the first harmonic of the LO and not with the RF signal, the first number is usually either 1 or -1, and the second number is usually zero.

Assuming the qpss analysis has inputs at 1GHz and 1.1GHz, and the qpxf frequency range is 1K to 100M, and the relative harmonic number is 1 0, the sweep will start at 1G + 1K, and sweep to 1G + 100M using log spacing from 1K to 100M. This is illustrated below.

To use a log sweep below the first harmonic, use the same frequency range, and type in -1 0 as the relative harmonic. This is illustrated below.



Maximum Clock Order

In qpxf terminology, the large signal from qpss is called the clock. Thus, *Maximum clock order* specifies the highest large signal harmonic number (The large signal is usually the LO signal) to calculate the conversion gain terms for. Usually, the conversion gain is only needed for a small number of LO harmonics, so when shooting is used as the qpss engine, usually *Maximum clock order* is fairly small, on the order of 3 to 5. Each conversion gain term that is calculated is called a sideband in a qpxf analysis. When harmonic balance is used for the qpss engine, the *Maximum clock order* field should be left blank, or the same as the number of LO harmonics, which does the same thing. The *Direct Plot Form* has the ability to select the individual terms that are needed.

The sideband number is the qpss harmonic number that is mixed with. Because qpss always has multiple input frequencies, qpss has multiple harmonic numbers that are being mixed with. If there are two input tones in the qpss simulation, there are two numbers in qpxf that describe which harmonics of which tone are mixing to produce that output frequency. If there are three tones in the qpss analysis, then there are three numbers for each qpxf sideband.

If the sideband number is negative, the input frequency is more negative than the output frequency specified in the qpxf *Choosing Analyses* form. If the sideband is positive, the input frequency is more positive than the output frequency.

Although this seems complicated, there are features in the Direct Plot form to make this easier to understand.

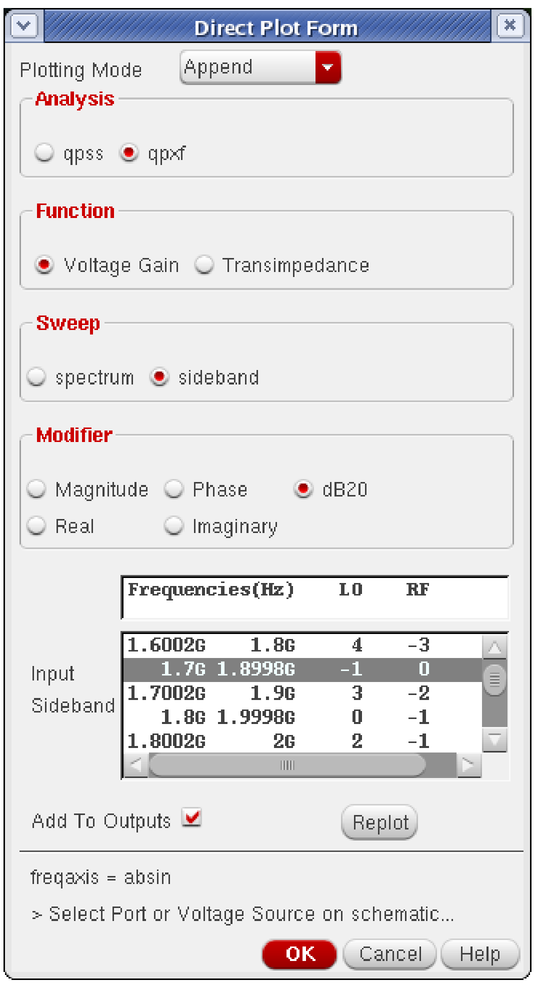
Setting Moderate Harmonics for Accurate QPXF Results

The qpss analysis, when shooting is selected, has a minimum of 200 timepoints in each slice, so the large signal in qpss inherently has frequency domain content through the 100th harmonic of the large signal input frequency. (Usually the LO signal) Only the harmonics that are specified in the moderate tone affect the accuracy of the solution. The result of this is that the qpnoise mixing products that are produced from mixing with the large signal are almost always correct, but the mixing with the moderate signal might or might not be accurate, depending on the number of harmonics that are specified for the moderate signal, and the number of harmonics actually produced by the system. Like harmonic balance, you need to run the qpss and qpxf analyses with a number of harmonics specified for the moderate tone(s), and make the measurement in qpxf. Then increase the number of moderate harmonics, run again, and make the same measurement. If the result changed significantly, then you need more moderate harmonics. If the result did not change, you might be able to reduce the number of moderate harmonics. Use the smallest number of moderate harmonics that produce a stable qpnoise results. If the moderate signal is below compression, start with two or three harmonics on the moderate tone. If the signal is at compression, start with five harmonics. If the signal is above compression, start with seven harmonics.

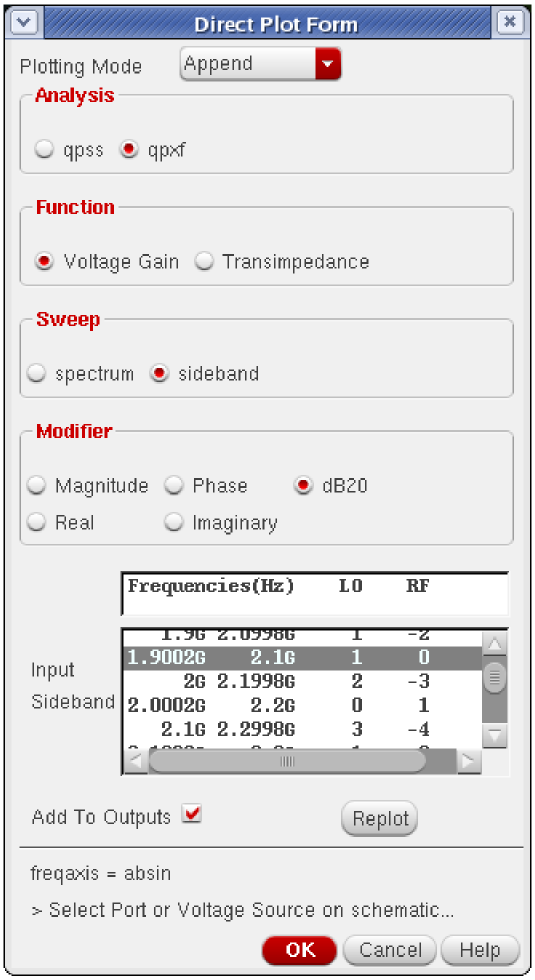
Qpxf is usually preferred to qpac when you need to measure the conversion gain near all the qpss harmonics.

Two methods are generally used to plot the conversion gain terms. One method is to plot the terms individually from the *Direct Plot Form*. Because there are so many harmonics to mix with and so many transfer functions, this method is preferred by many. A detailed example is in the examples section.

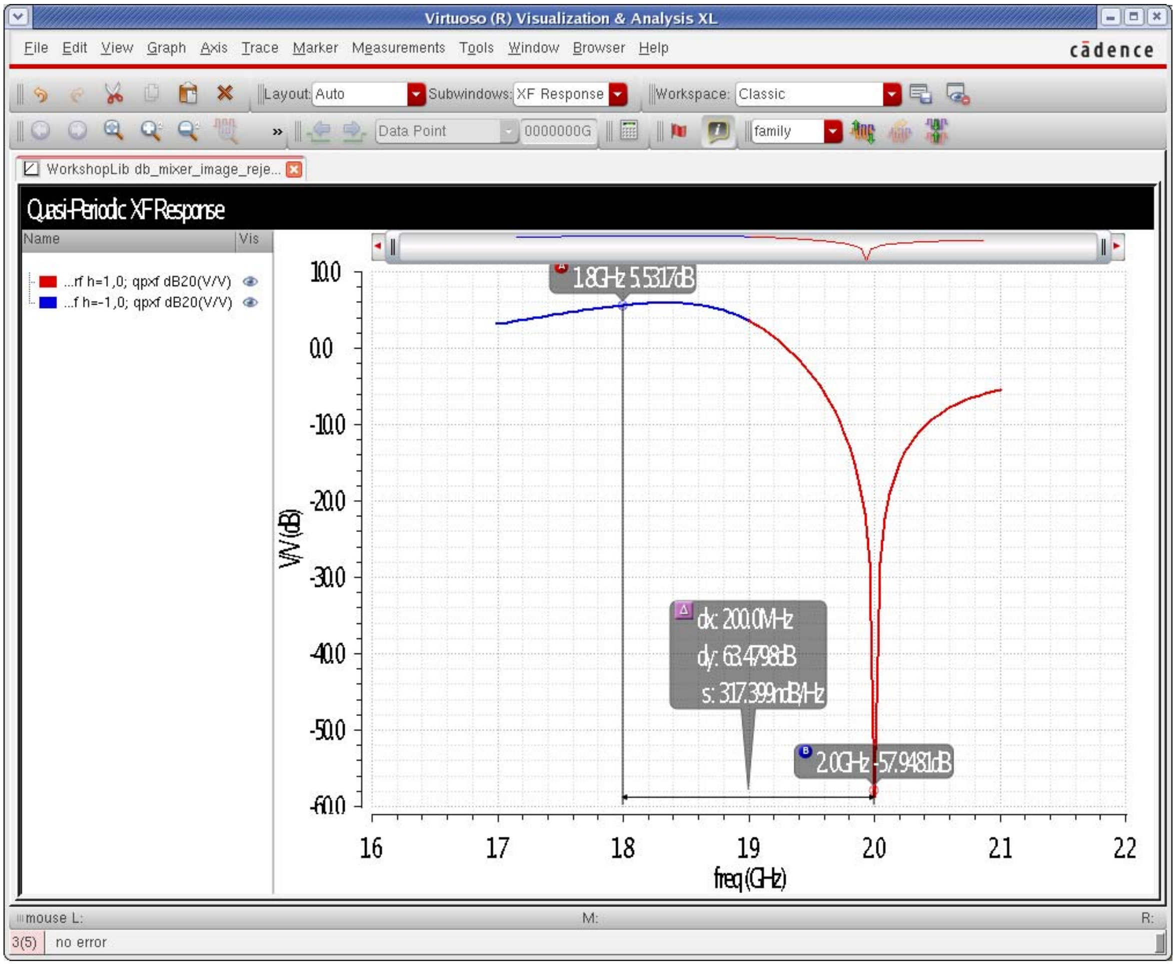
In the example below, the LO is at 1.9GHz, and the RF blocker is at 1.95GHz. The circuit is an image reject mixer with a 100MHz IF. Given this, the passband and image are at 1.8GHz and 2GHz. After the simulation has been run, in the *Direct Plot Form*, select *sideband* in the *Sweep* section. This allows individual conversion gain terms to be plotted. For the qpxf analysis, the output frequency range is 200KHz to 200MHz. Below is the *Direct Plot Form* for the low side injection transfer function. When the input frequency range is chosen, select the source in the circuit that you want the conversion gain terms for.



Below is the direct plot form for the high side injection transfer function. After the first selection, you can select *Replot*, which will take the same source as the choice for plotting.



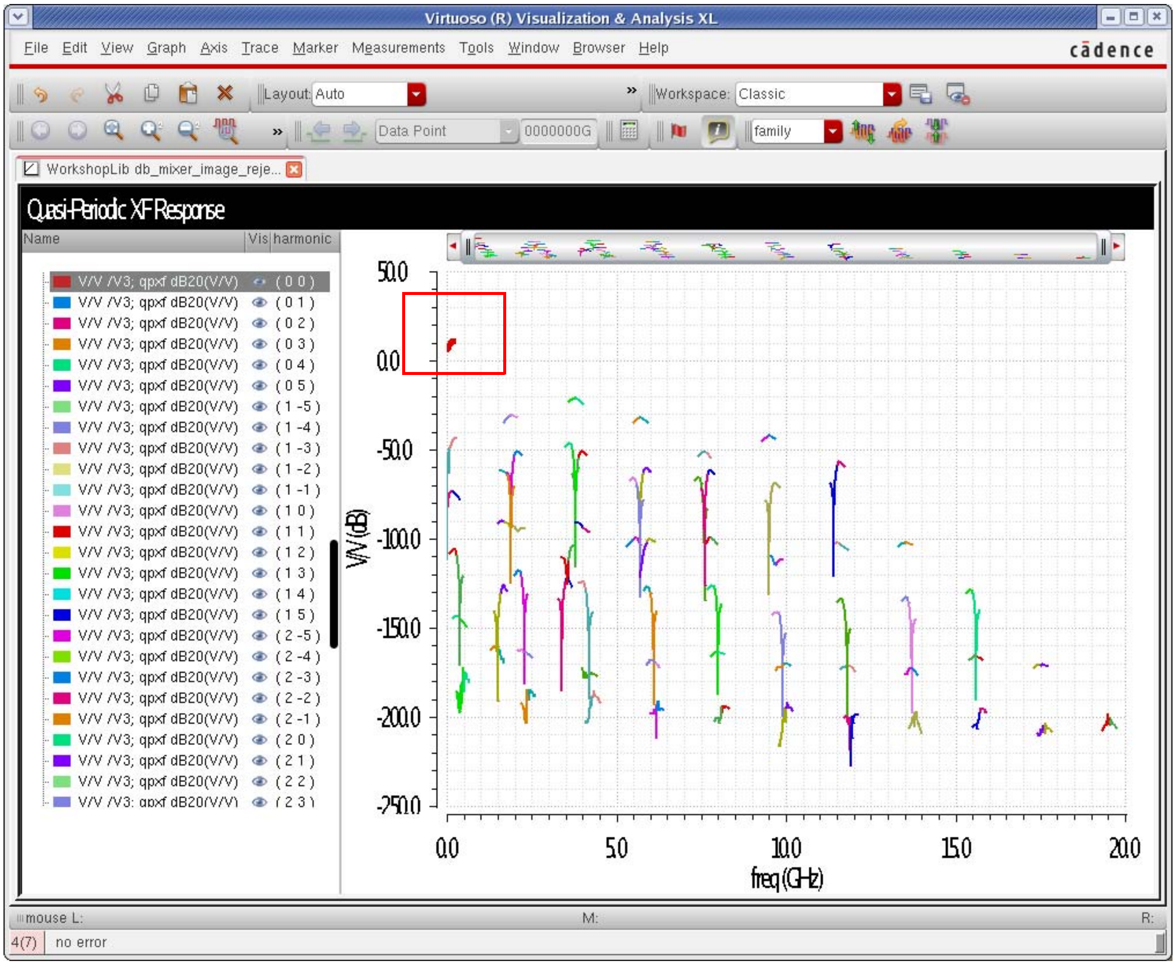
Below are the transfer functions. Markers A and B have been placed at the passband and image frequencies.



The passband gain is 5.53dB, and the image rejection is about 63.4dB.

The other method (plotting all the transfer functions at the same time) is given in the examples section.

Qpxf can also be used to measure the conversion gain from the battery that supplies power to the output. All the gain and conversion gain terms should be below unity, or a potential oscillator is created. In the plot below, the gain from the bias source to the output at the IF frequency of 100MHz is about 10dB. This circuit is a potential oscillator.

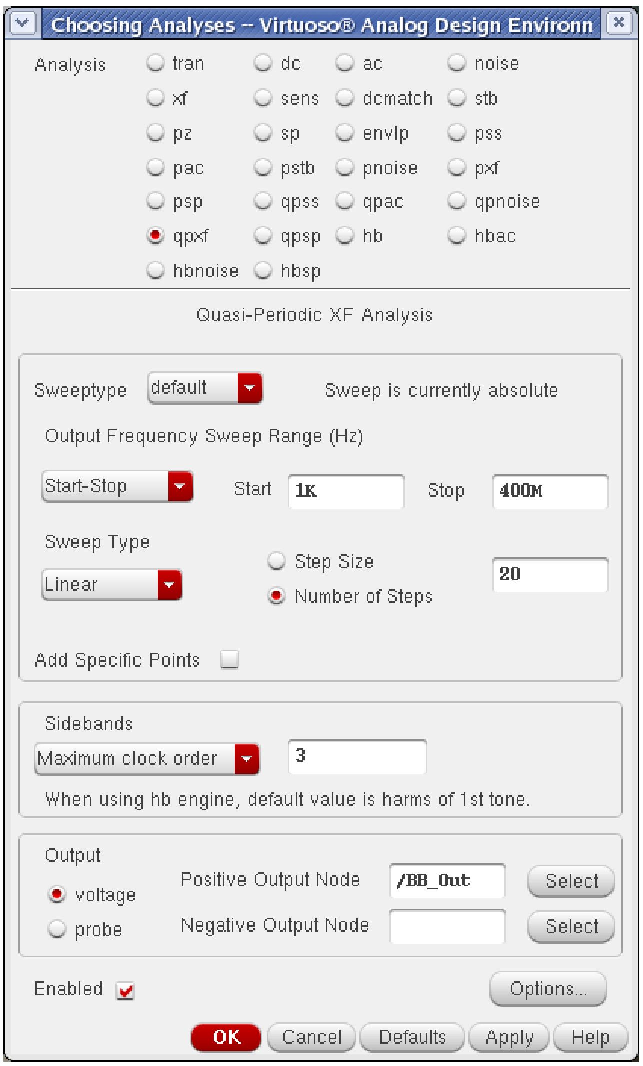


ADE Implementation

This section jumps from the theoretical to the practical with examples of how to use the qpxf analysis we just covered. The focus is only on the qpxf Choosing Analysis form so you can see where the individual settings are made, and the qpxf options form. For examples with all the steps shown, see the examples section that follows this section.

QPXF for a Normal Conversion Gain Measurement

Here is an example of the *Choosing Analyses* form for a normal conversion gain measurement.



Sweeptype can be relative or absolute. The default is absolute. The selection that is being used is shown to the right of the *Sweeptype* selection.

1. The frequency sweep range is always an **output** frequency range. It is not a sweep in the traditional sense of connecting an input, sweeping over a frequency range, and measuring the output. Instead, it is an output frequency range to be analyzed. At each output frequency in the sweep, the input frequencies that will mix with the qpss harmonics to cause that output frequency are calculated, and the forward conversion gain is also calculated.

2. Linear and log spacings are provided. Generally, Auto should be avoided because it will always calculate 50 frequency points. Usually, a much smaller number is needed, so it is recommended that you set Linear or Log, and control the resolution yourself to save time.

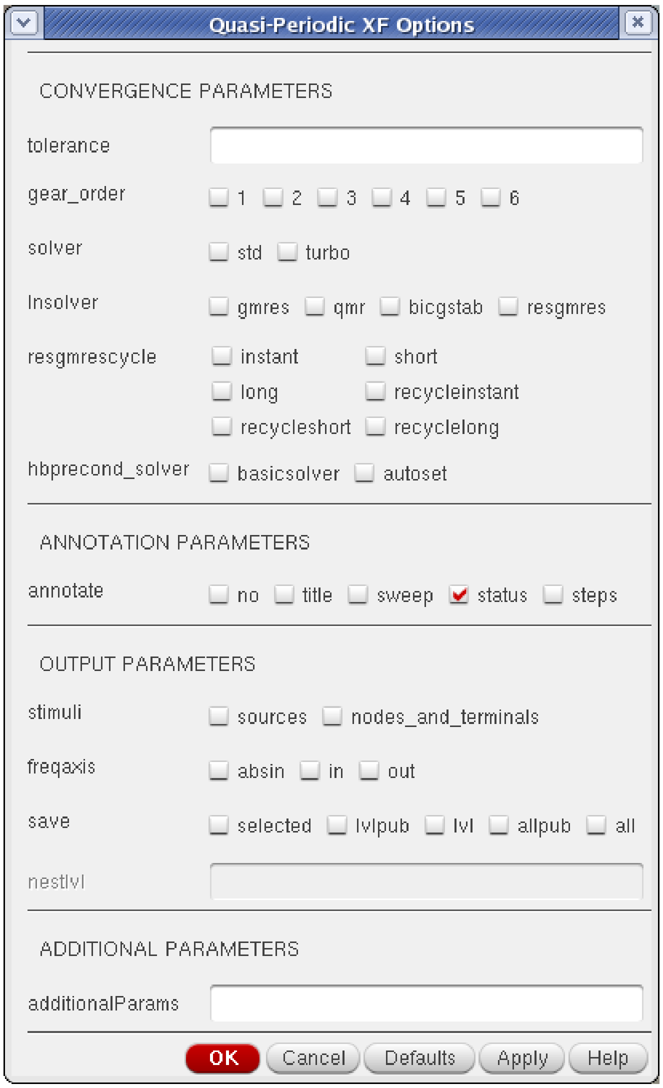
3. Although there are choices for the Sidebands selection, it is usually better to leave the setting at *Maximum clock order*. When you choose shooting as the qpss engine, and if you need a small subset of the output results, you can set a small number (like 3 or 5) in the *Maximum clock order* field. If you use harmonic balance, set *Maximum clock order* to the same number you used to set the LO harmonics in qpss, or leave the field blank, which accomplishes the same thing. In either case, you can use the *Direct Plot Form* to select just the results you want.

4. Run the simulation, and make the measurement in qpxf. Now increase the number of moderate harmonics by about 50% and run again. If the qpxf result changed significantly, then you need more harmonics on the moderate tone (or tones) in qpss. If it did not change, use the smallest number of harmonics that results in an stable measurement.

5. The output can measure either a voltage or a current. The voltage measurement is the default. Specify the output node or nodes in the circuit. If the output is a current select probe for the output, add an iprobe from analogLib (a current probe) in series with the output in the schematic, and specify the instance name in the *Output Probe Instance* field in the *Choosing Analyses* form.

QPXF Options

Below is the options form.



None of the qpxf options are commonly used.

tolerance

Leave this option at the default value.

Qpxf uses an iterative solver to calculate the conversion gain terms. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough. and The tolerance option specifies that accuracy for qpxf. For shooting, the default tolerance is 1e-9. When HB is the qpss engine, the default is 1e-6.

gear_order

Do not change this option.

solver

The default solver is the turbo solver.

When hb is used as the engine in qpss, leave this option at the default.

When shooting is used, sometimes when the qpxf input frequency is very close to the frequency of one of the harmonics in the qpss, warning messages will appear in the qpxf output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the qpss, but which takes longer to run than the *turbo* solver.

lnsolver

Leave this option at the default.

Each qpxf sideband is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases as the number of iterations increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

resgmrescycle

Leave this option at the default. For the resgmres linear solver, there are several different options.

hbprecond_solver

This option is only available only when harmonic balance is selected in qpss and APS is used.

The basic solver is the only solver available in standard Spectre. Autoset is the default solver when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window. If you have stagnation, it will save a small amount of time to set this option to *basicsolver*.

annotate

This option controls the level of detail in the Spectre output log. No detail is provided when you select *no*, while more details are provided as you move towards right with steps option providing maximum detail.

stimuli

This option is not implemented at the current time.

freqaxis

freqaxis specifies whether you want to see the negative frequency axis or not.

The qpxf analysis calculates the input frequencies and forward conversion gain, so *in* and *absin* are reasonable choices. *in* displays the negative frequency axis. *absin* (absolute value of the input) displays positive input frequencies. This is the default.

save

This option is not implemented at the current time.

nestlvl

This option is not implemented at the current time.

additionalParams

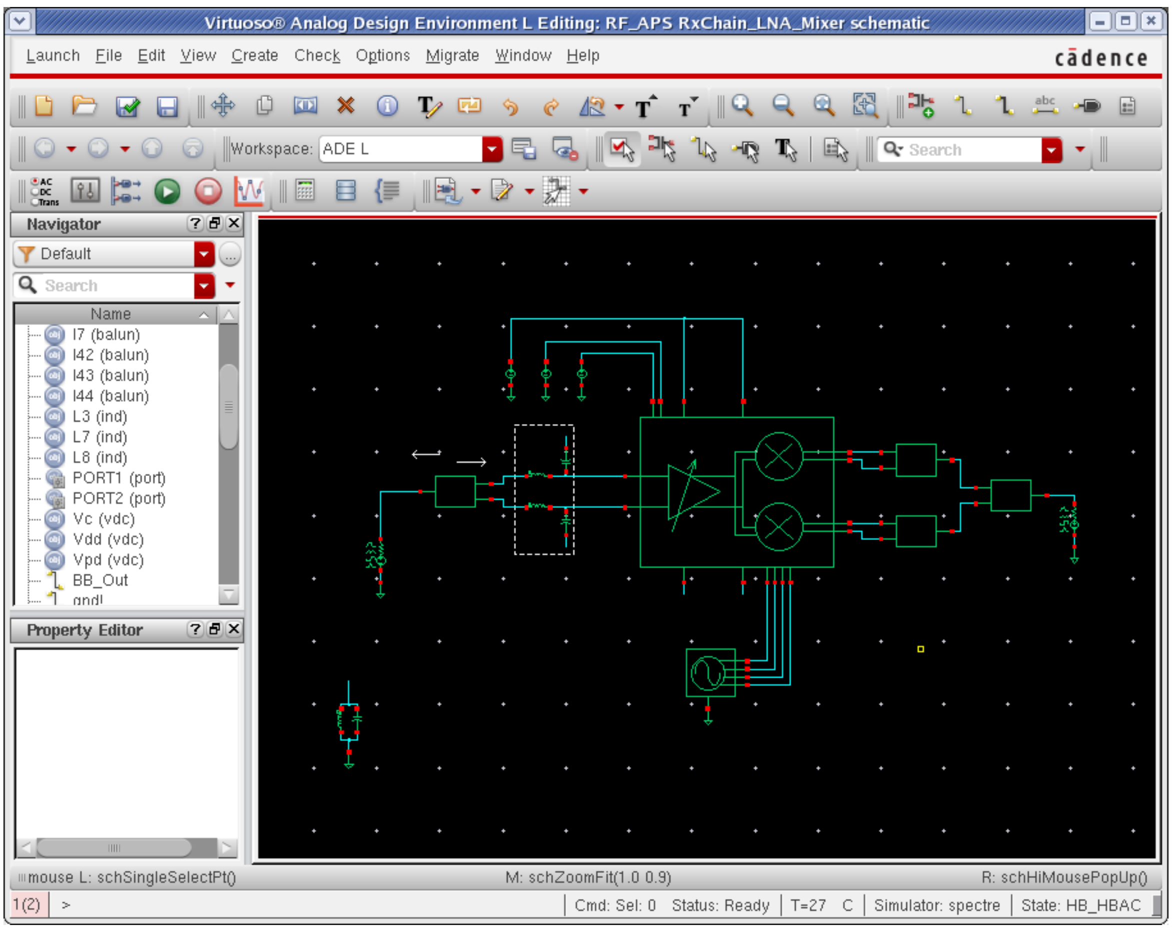
additionalParams is typically used for new features that are being beta tested. Keyword=value pairs are expected in this field. An example is `solver=std`.

For more information about the other options, type `spectre -h qpxf` at the command prompt in a Unix shell window.

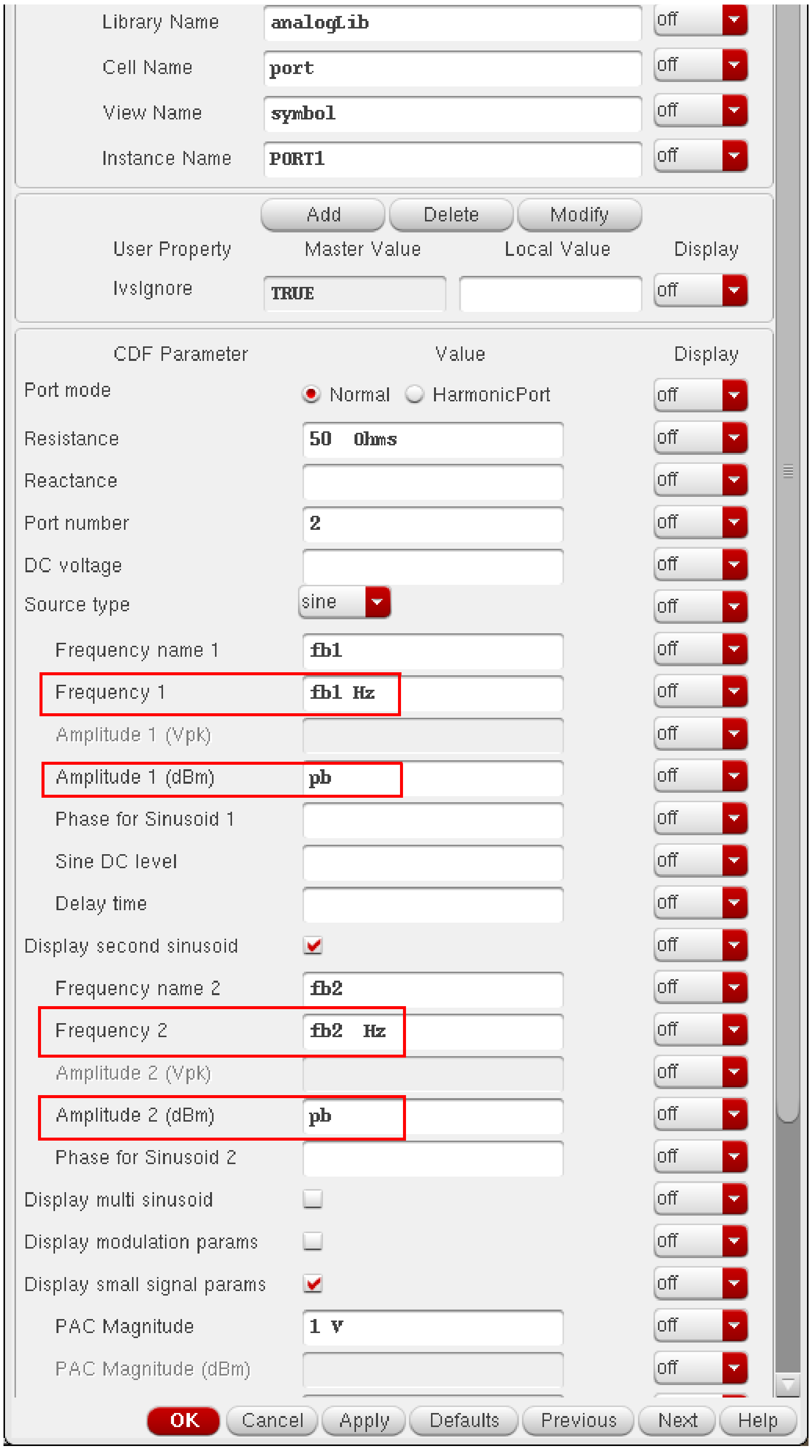
Example

Note that for this application, harmonic balance is likely faster than shooting. If you plan on using harmonic balance, please use the hb and hbnoise analyses to measure the transfer functions. Hbnoise has hbxf integrated into the noise measurement. The hb and hbnoise Choosing Analyses forms always have the latest harmonic balance features.

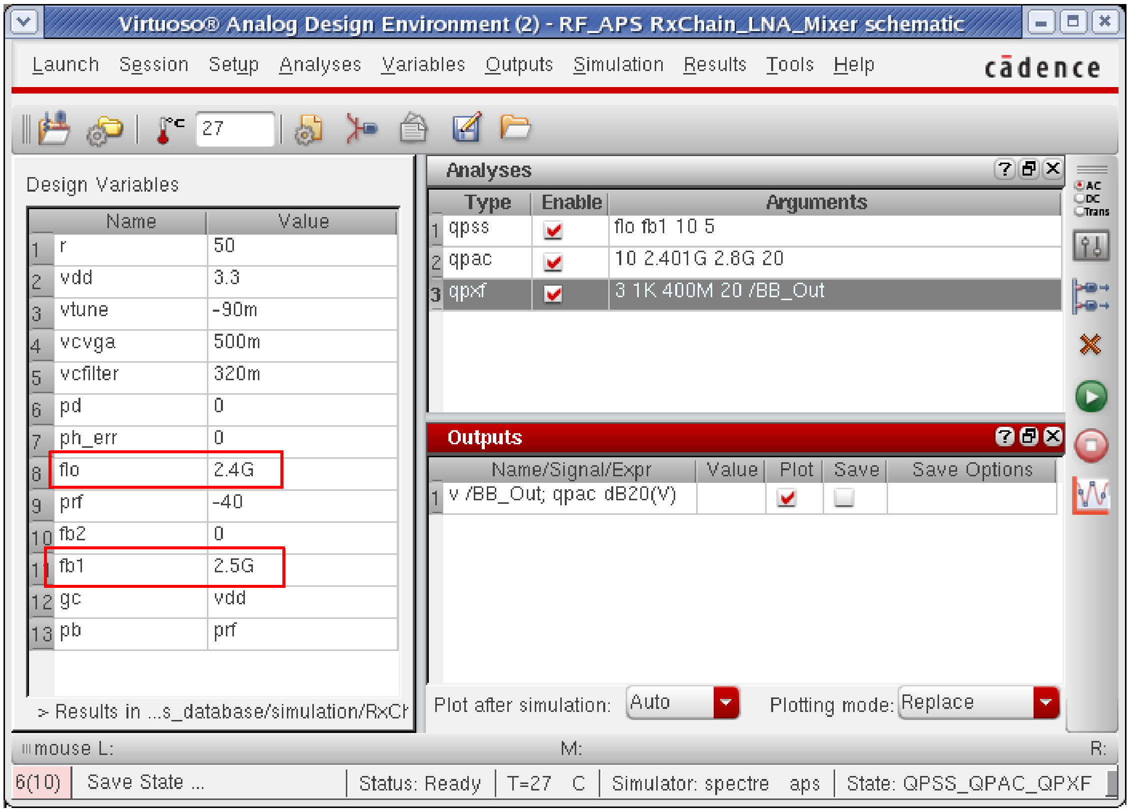
Consider the LNA-Mixer circuit shown below.



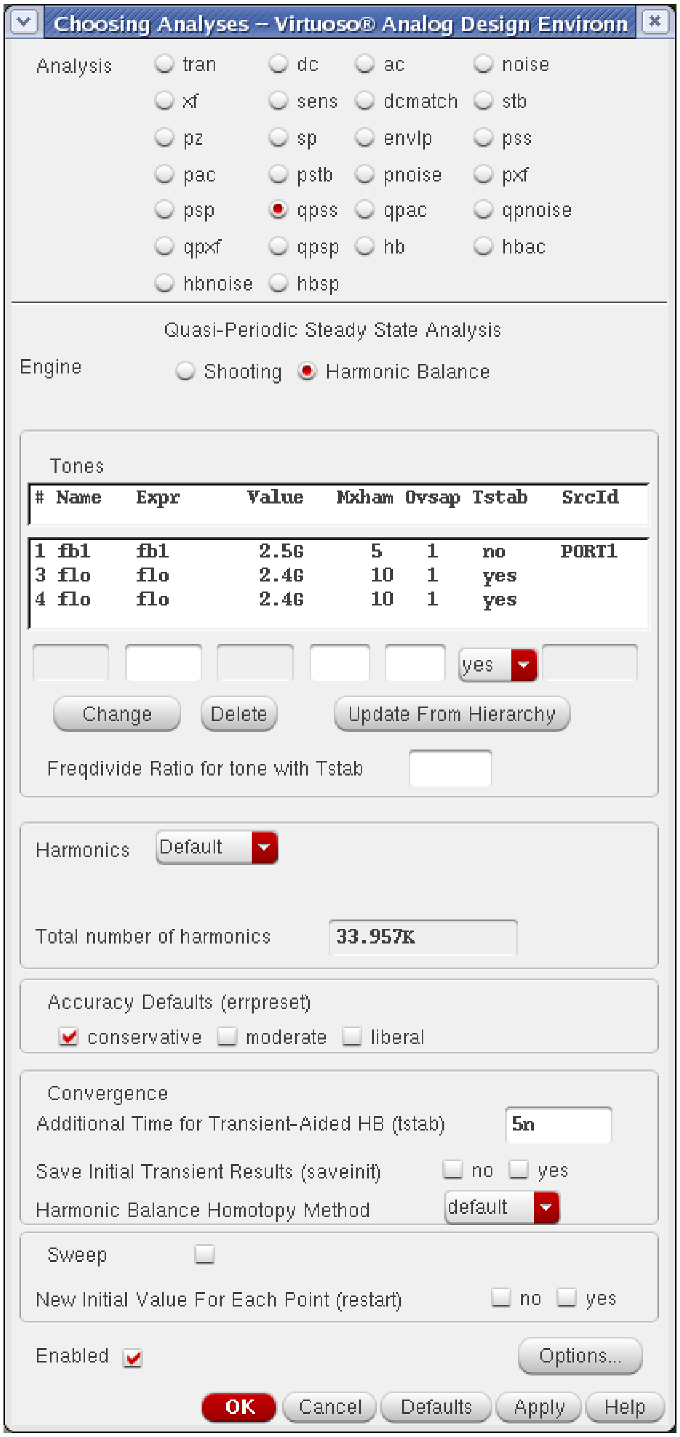
1. The input port has variable names to set the values of the frequency and amplitude. This is suggested so these terms can be easily changed in the ADE environment without needing to change the schematic.



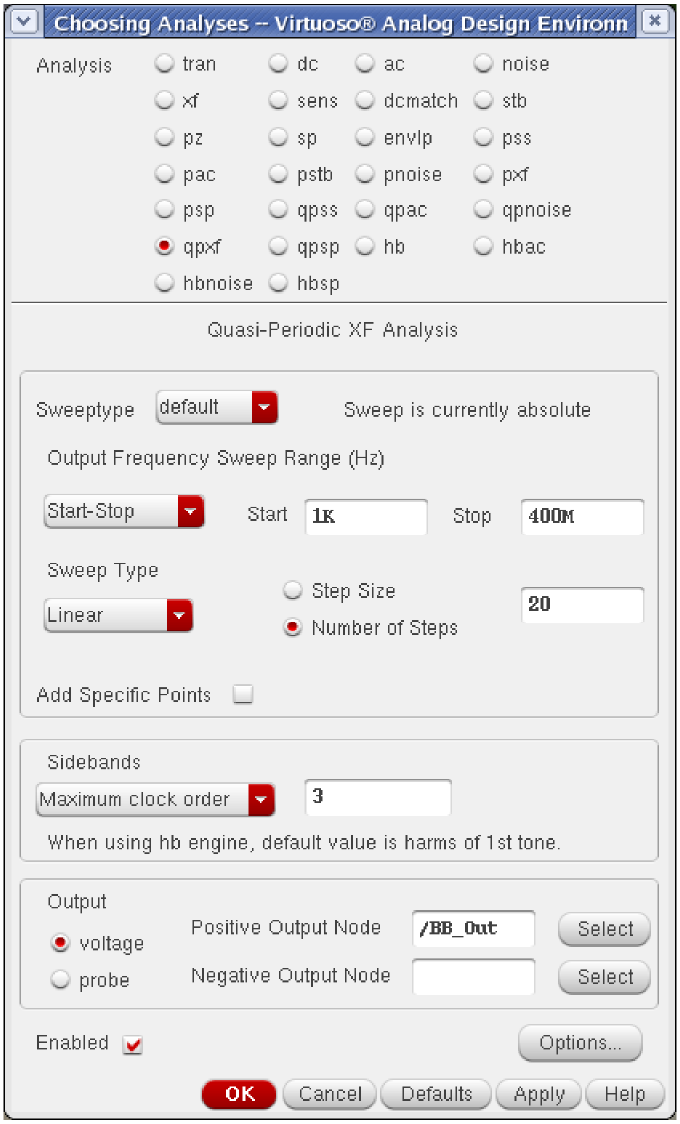
2. The variables are set in the ADE.
3. The LO is at 2.4GHz and the RF blocker is at 2.5GHz.



4. First, a qpss analysis needs to be set up. For more information, see the qpss section at the beginning of this chapter.



5. Now set up the qpxf form.



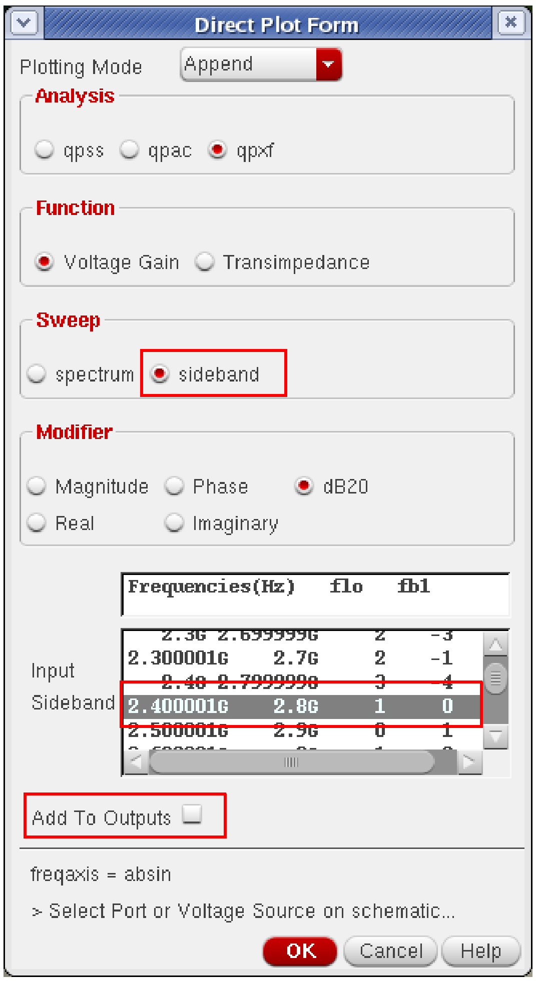
6. Choose an appropriate output frequency range to analyze for your circuit. This frequency range is wider than usual so the conversion gain curves can be easily seen in the waveform tool.

7. Usually, it is better to select Linear or Log spacing, and define your own sweep. Automatic usually takes more points than needed, and this extends the runtime.

8. Set *Maximum clock order* as appropriate for your needs. In this case, mixing product that mix with the third harmonic and lower of the LO are calculated. If you use the harmonic balance engine in qpss, the *Maximum clock order* field should either be left blank, or should be set to the same number of LO harmonics in the qpss analysis. Both methods accomplish the same thing.

9. Choose an output node in the circuit. If your circuit has an output current, add an iprobe (current probe) from analogLib in series with the output current, and specify it in the *Output* section.

10. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* appears.

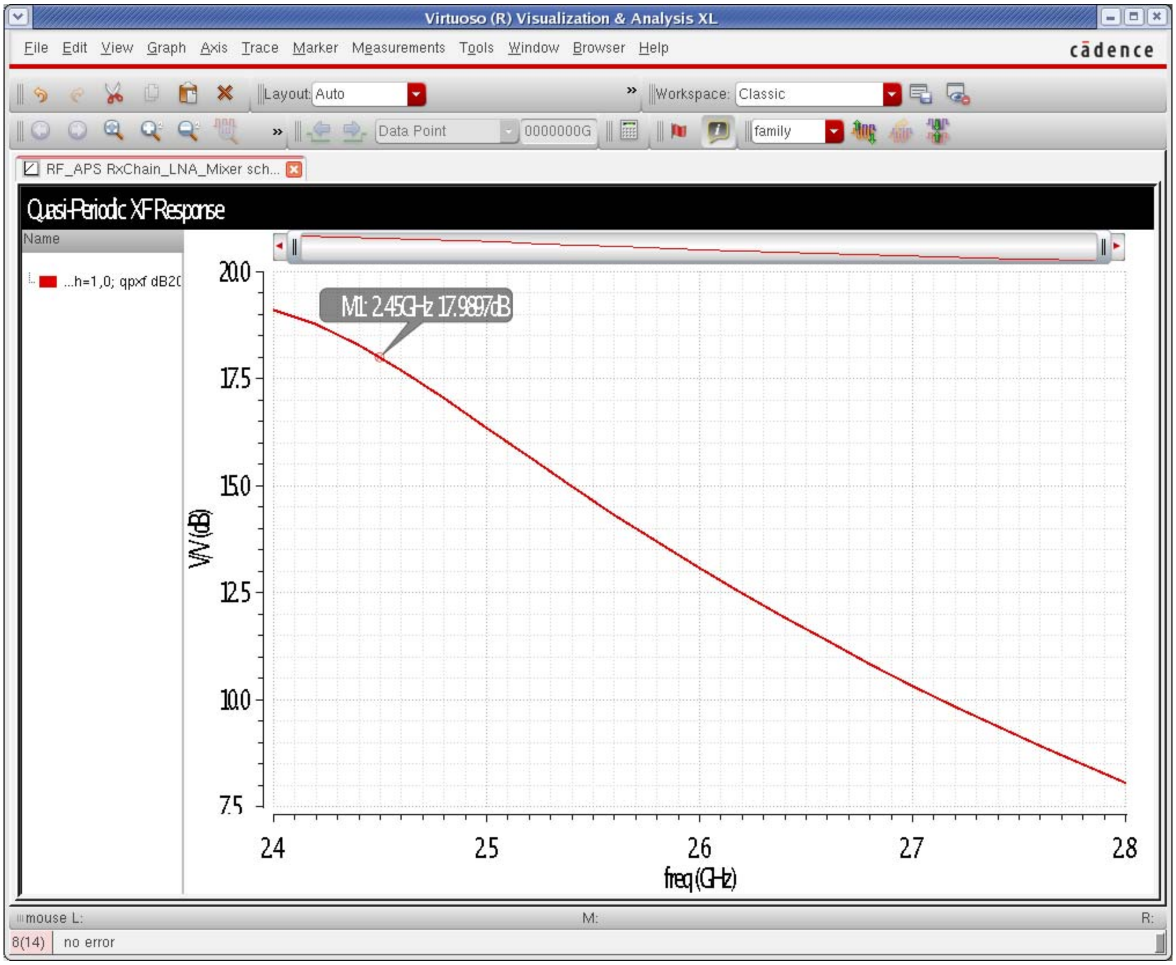


11. To plot individual conversion gain terms, select *sideband* in the *Sweep* section.

12. Select the **input** frequency range you want to plot in the *Input Sideband* section.

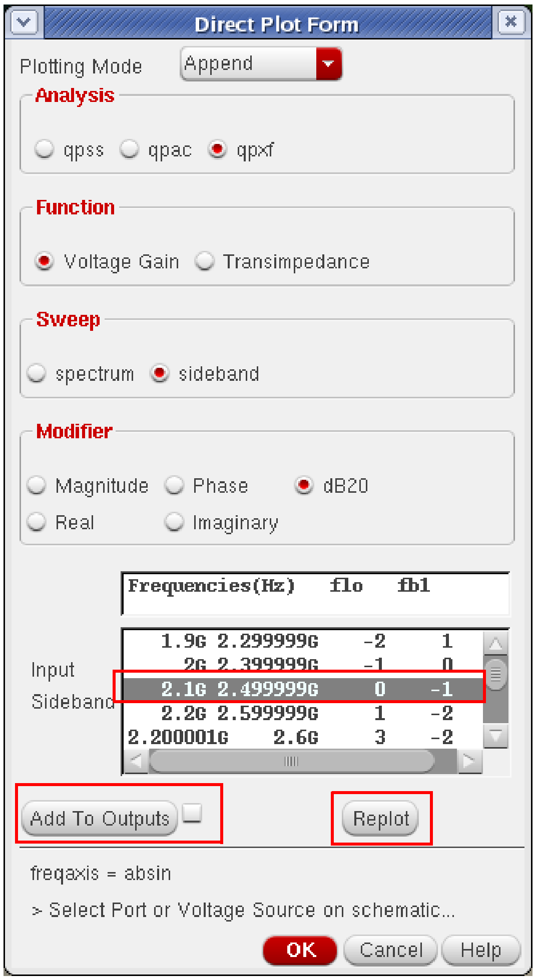
13. If you want this measurement to plot automatically after subsequent simulations, select *Add To Outputs* at the bottom of the *Direct Plot Form*. In this example, it was not selected.

14. Click the desired source in the schematic for the gain measurement. For the plot below, the input port was selected. To measure the conversion gain, move your mouse cursor to the input frequency, and type m to place a marker. The conversion gain at 2.45GHz input frequency is about 18dB.



15. There are also mixing terms from mixing with the RF blocker signal as well. When the output frequency is near zero to 400MHz, the input frequency is from 2.1GHz to nearly

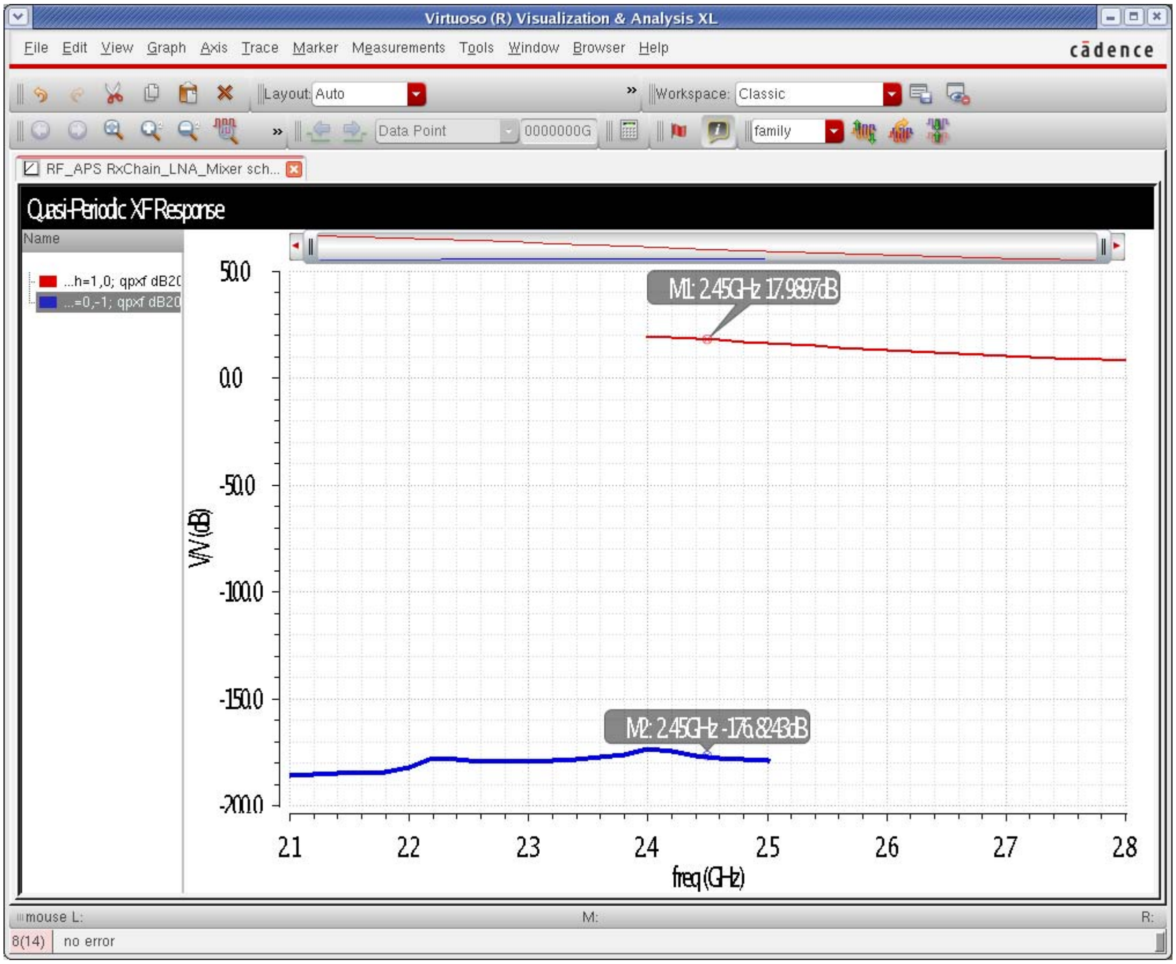
2.5GHz. Select that term from the *Direct Pot Form*, and if you still want the gain from the input source, select *Replot*.



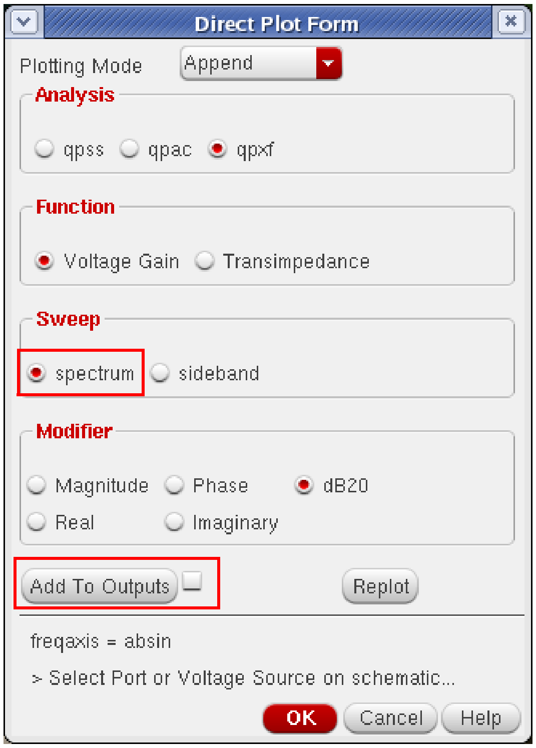
16. If you want this measurement to plot automatically after subsequent simulations, select *Add To Outputs* at the bottom of the *Direct Plot Form*. In this example, it was not selected.

17. The conversion gain from mixing with the RF input is quite small.

.

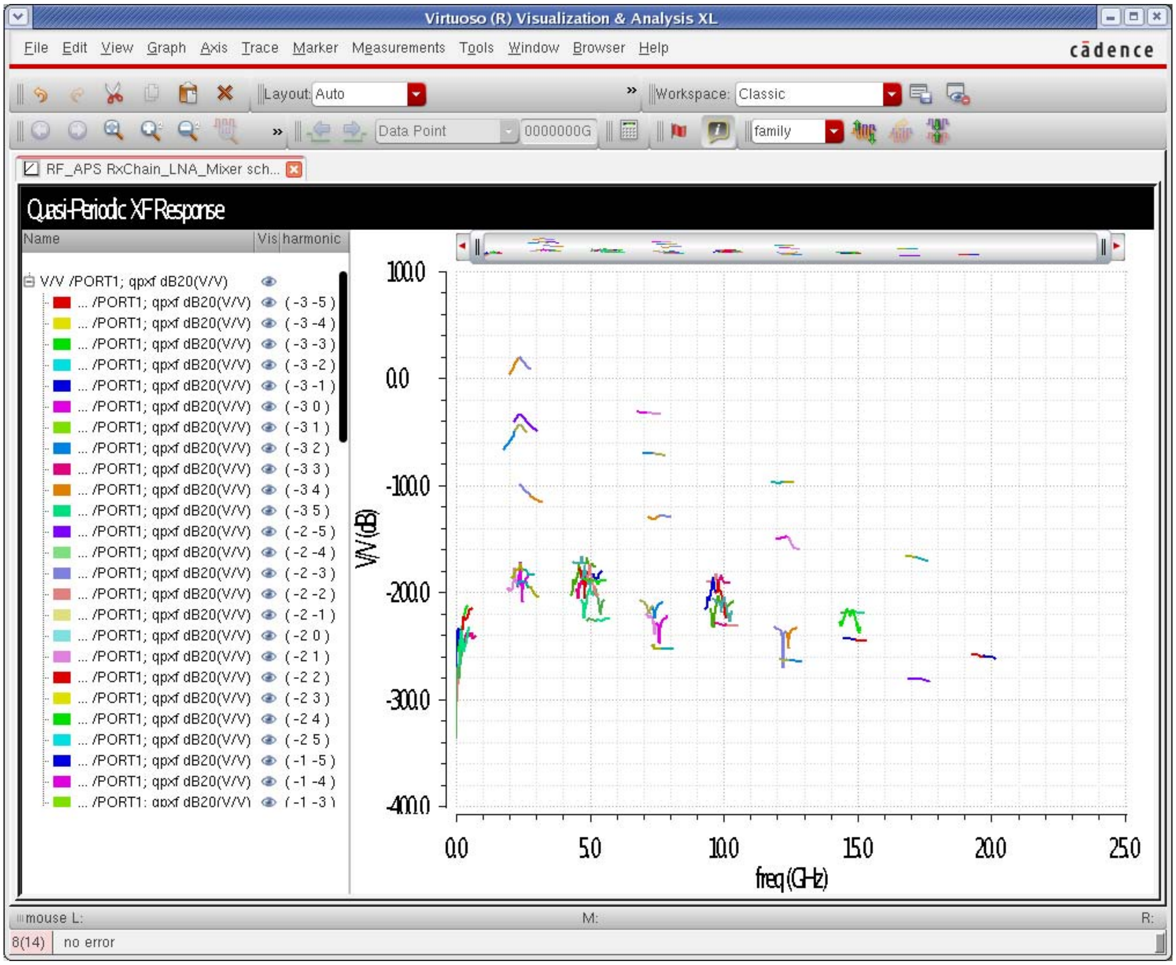


18. All of the conversion gain terms can also be plotted. To do this, select *spectrum* in the *Sweep* section.



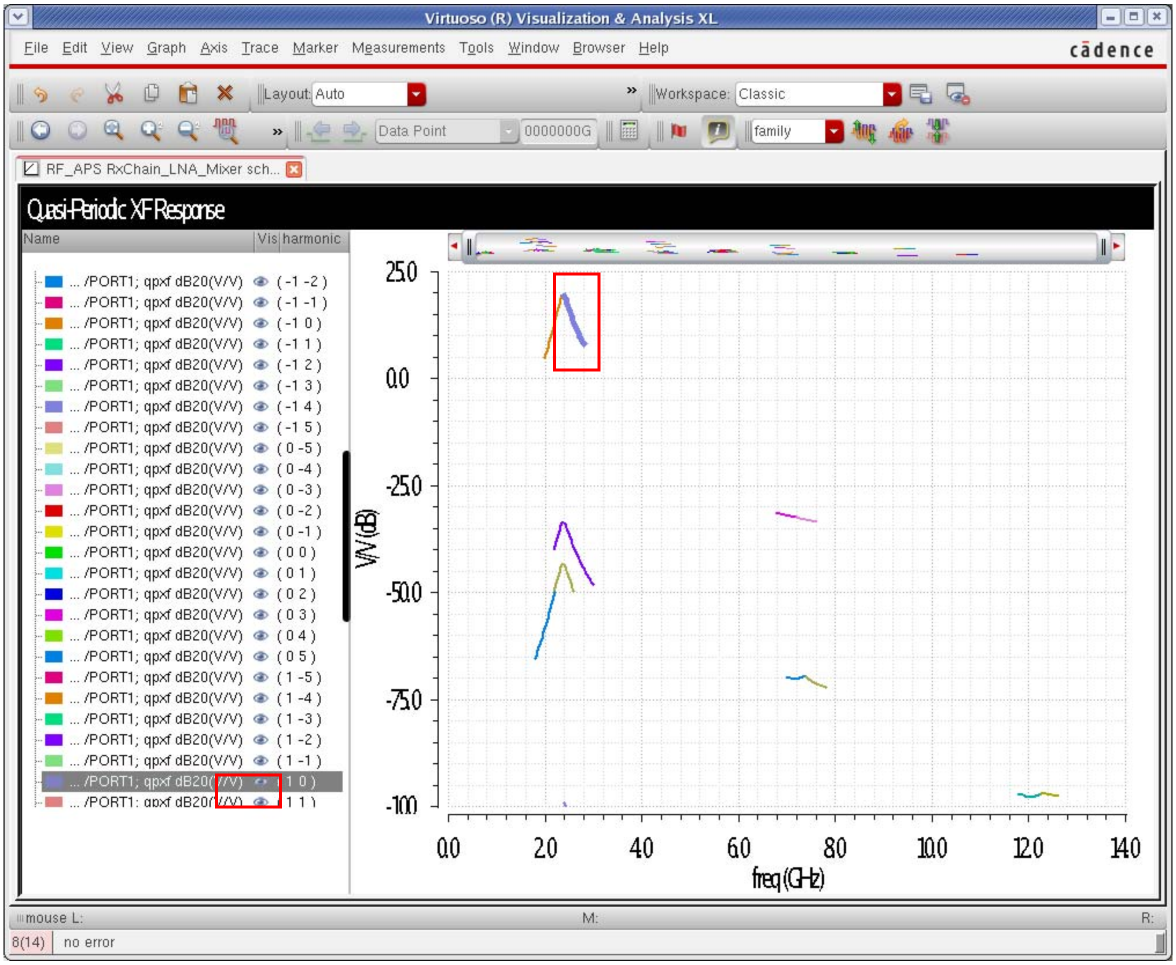
19. If you want this measurement to plot automatically after subsequent simulations, select *Add To Outputs* at the bottom of the *Direct Plot Form*. In this example, it was not selected.

20. All of the conversion gain terms are plotted.



21. There are a lot of conversion gain terms because there are a lot of harmonics to mix with.

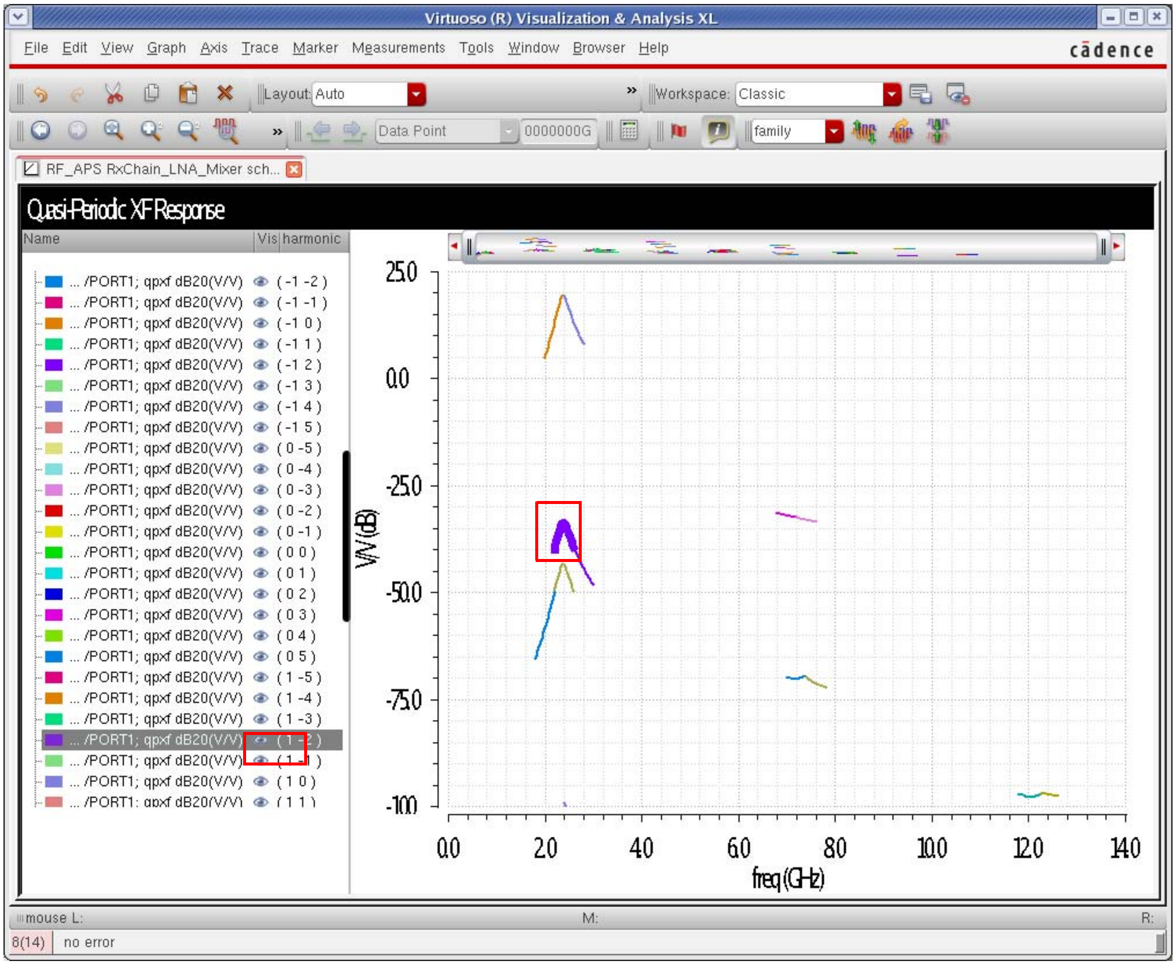
22. Now the waveform window is zoomed in to the part of the output where the gain is larger than -100dB. Select one of the curves. It highlights.



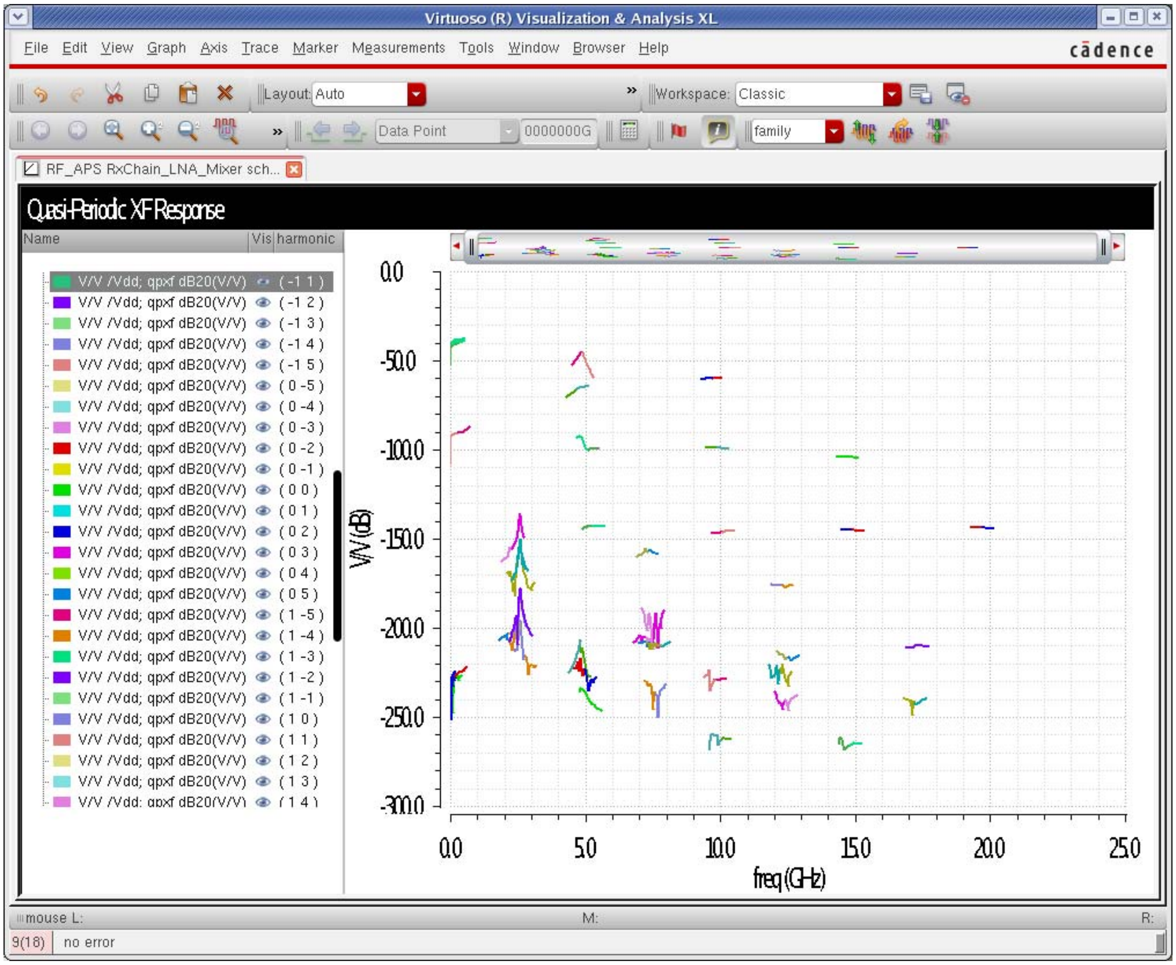
23. The sideband number is 1 0. The output frequency from the *Choosing Analyses* form is 1KHz to 400MHz. The first number is the frequency shift from output to input in multiples of the LO frequency. The second number is the frequency shift in multiples of the RF blocker frequency. The input signal can mix with any of the harmonics the system produces. 1 0 means that the input is mixing with the first harmonic of the LO and not with the RF input. Positive numbers mean that the input frequency is more positive than the output frequency. so, in this case, the input frequency is 1KHz to 400MHz + (1 * LO frequency of 2.4GHz) = 2.4+ GHz to 2.8GHz.

24. Below, the 1 -2 sideband is selected. The input mixes with the first harmonic of the LO and the second harmonic of the RF. 1KHz to 400MHz + (1 * 2.4GHz) + (-2 * 2.5GHz) = -

2.59999GHz to -2.2GHz. By default, this is reflected up into the positive frequency domain.



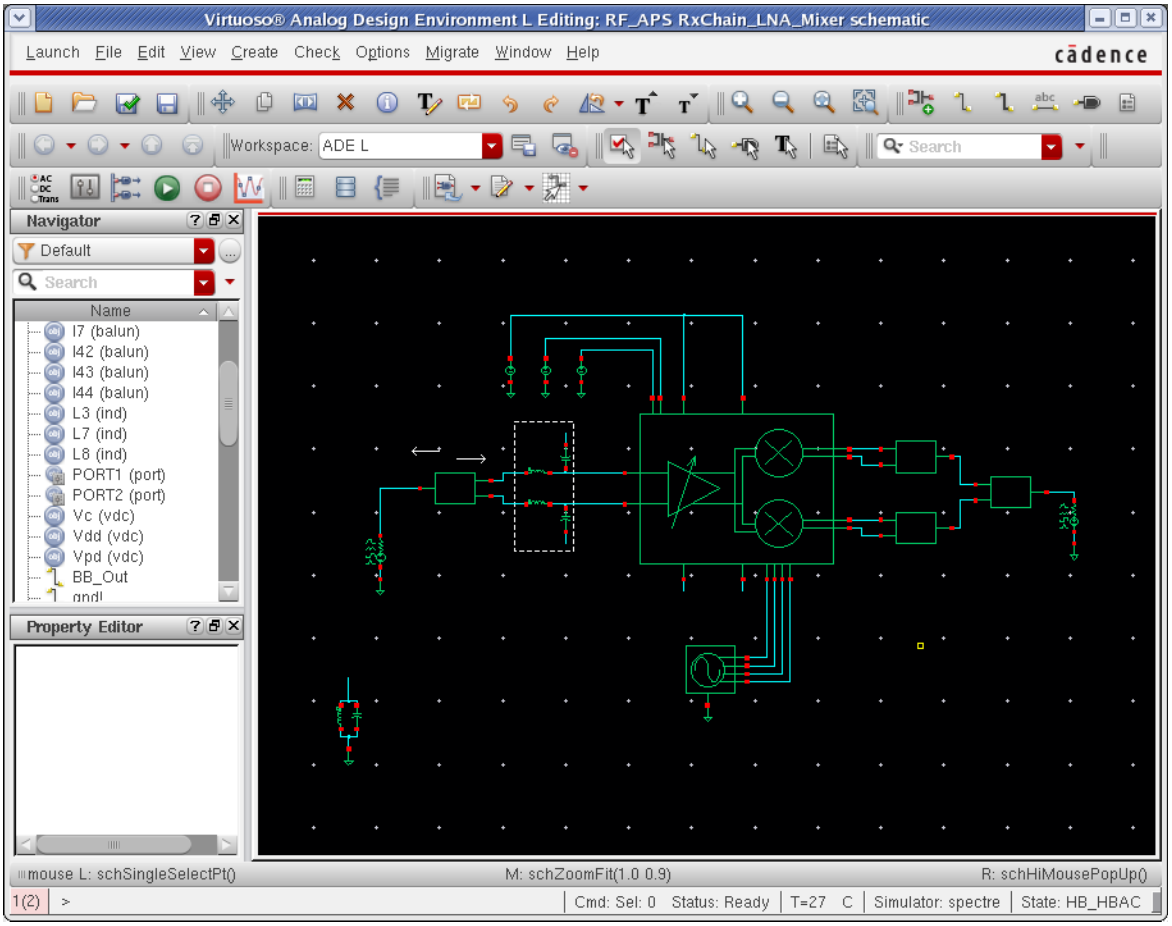
25. It is always suggested that you check the conversion gain from the power supply to the output to see if there are any mixing paths that have greater than unity gain. To measure these conversion gain terms, select the DC source in the schematic that supplies power to the circuit. In this case, all the conversion gain terms are less than unity.



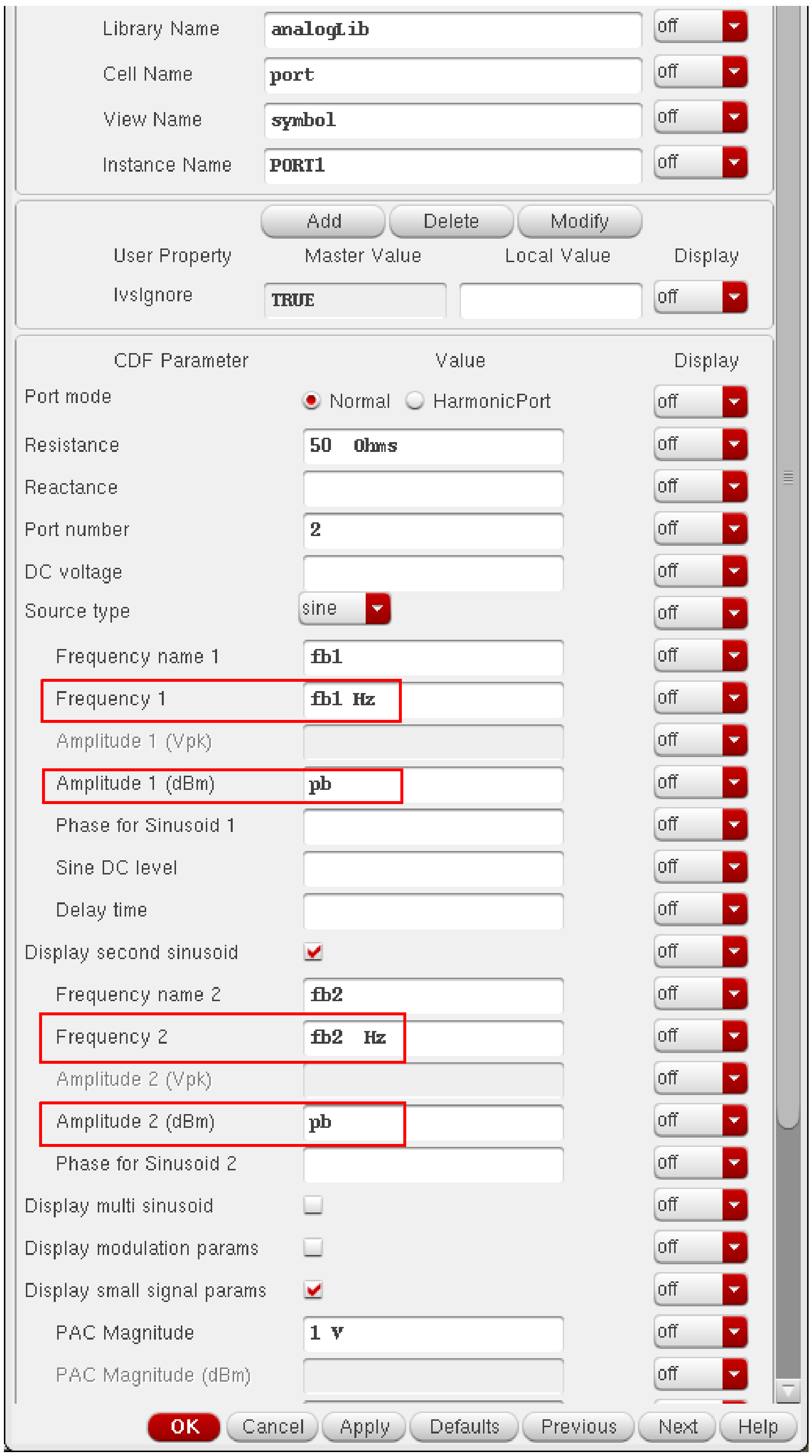
26. Note that finding a gain path like this in the lab is extremely difficult. This is one of the benefits of simulation.

Conversion Gain as a Function of Blocker Power

Consider the LNA-Mixer circuit shown below.

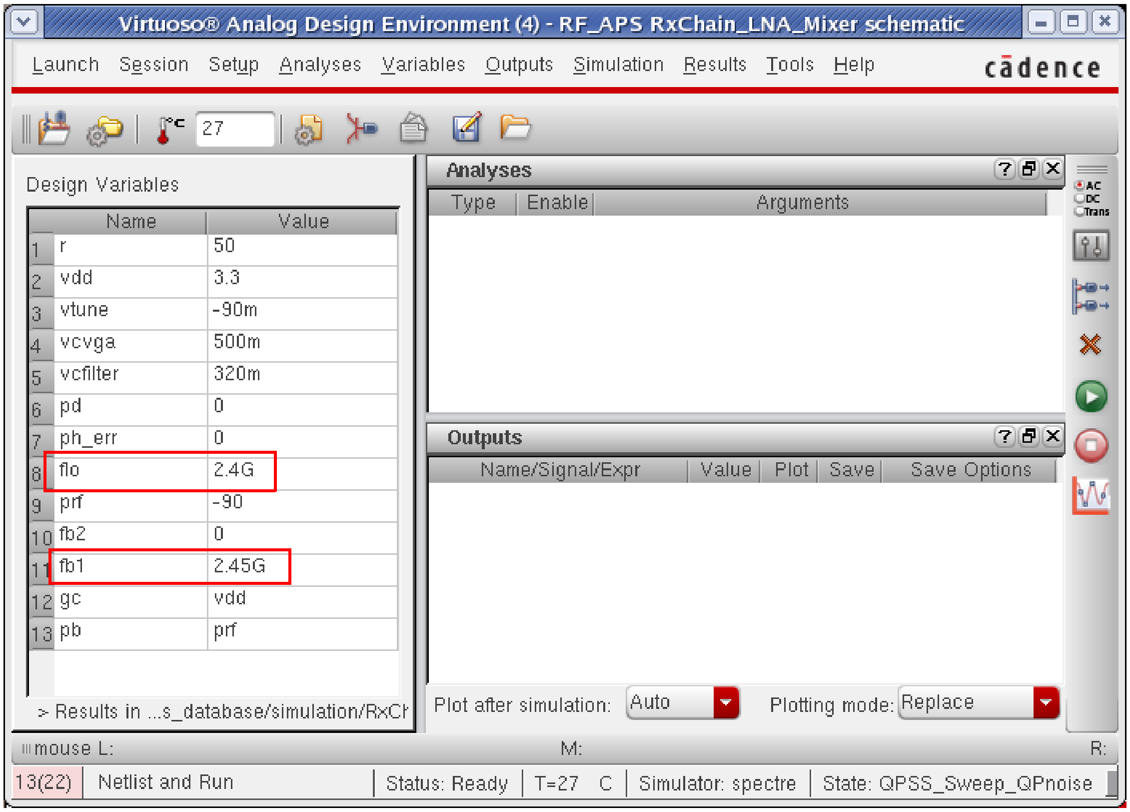


1. The input port has variable names to set the values of the frequency and amplitude. This is suggested so these terms can be easily changed in the ADE environment without needing to change the schematic.

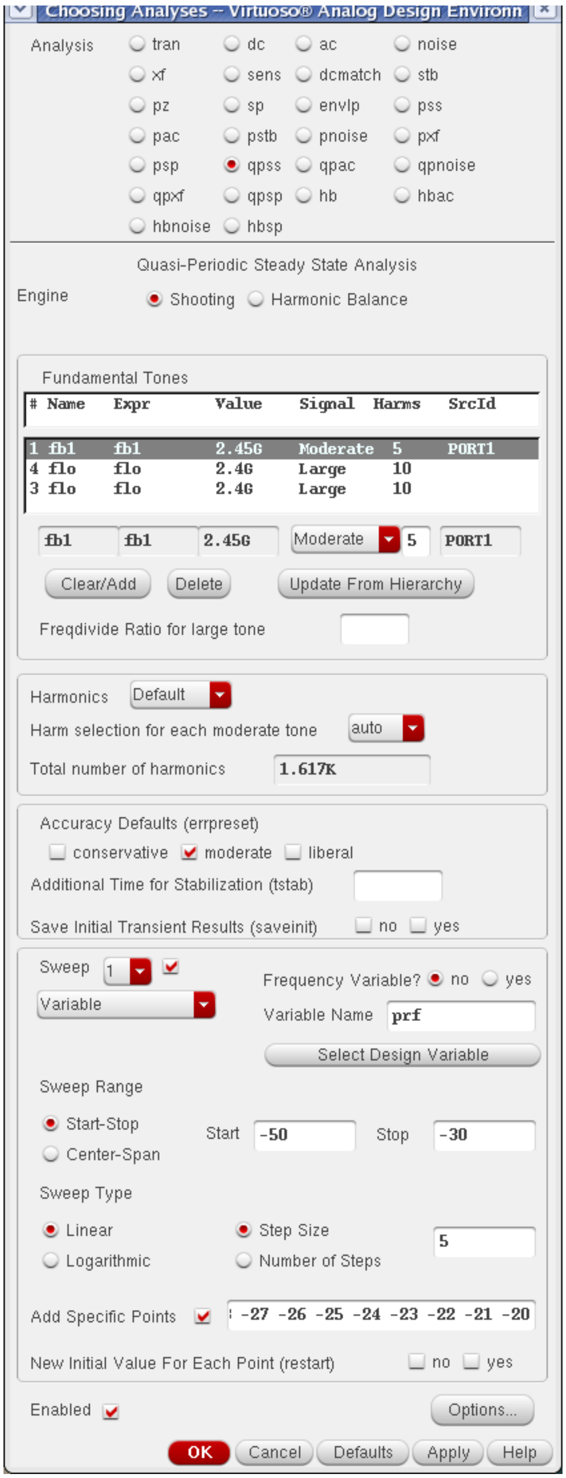


2. The variables are set in the ADE.

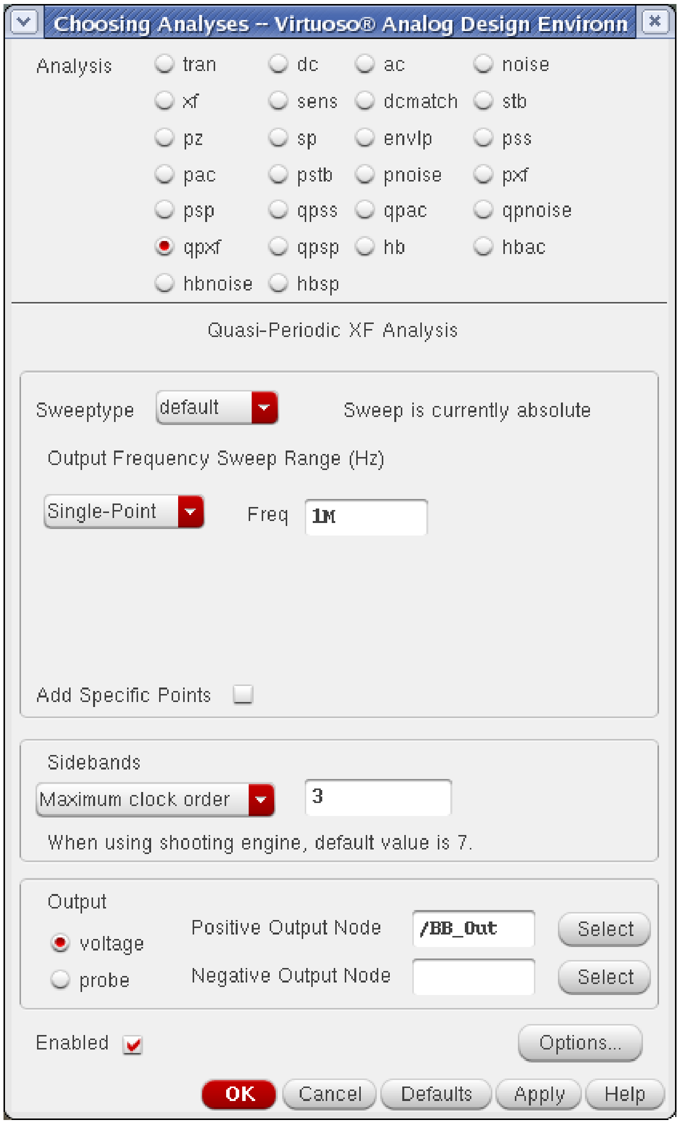
3. The LO is at 2.4GHz and the RF blocker is at 2.45GHz.



4. First, a qpss analysis needs to be set up. Specify a sweep of blocker power that is suitable for your system. For more information, see the qpss section at the beginning of this chapter.



5. Now set up the qpxf form, as show below.

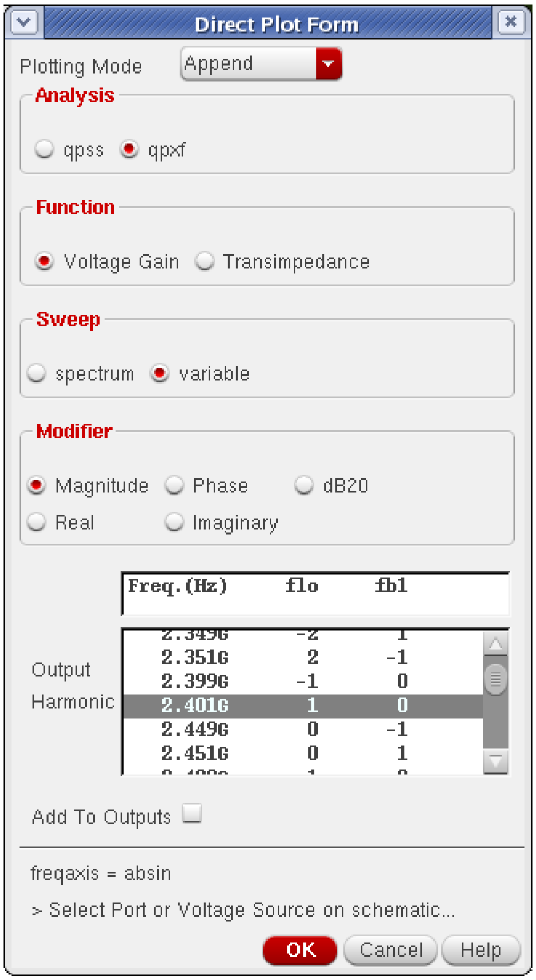


6. The *Output Frequency Sweep Range* is set to *Single-Point* when a sweep is performed in qpss. Set the desired output frequency in the *Freq* field.

7. Set *Maximum clock order* as appropriate for your needs. In this case, mixing product that mix with the third harmonic and lower of the LO are calculated. If you are using the harmonic balance engine in qpss, this should be set to the number of LO harmonics in the qpss analysis, or left blank, which does the same thing.

8. Choose an output node in the circuit. If your circuit has an output current, add an iprobe (current probe) from analogLib in series with the output current, and specify it in the *Output* section.

9. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main Form* in the ADE window. The *Direct Plot Form* is displayed.

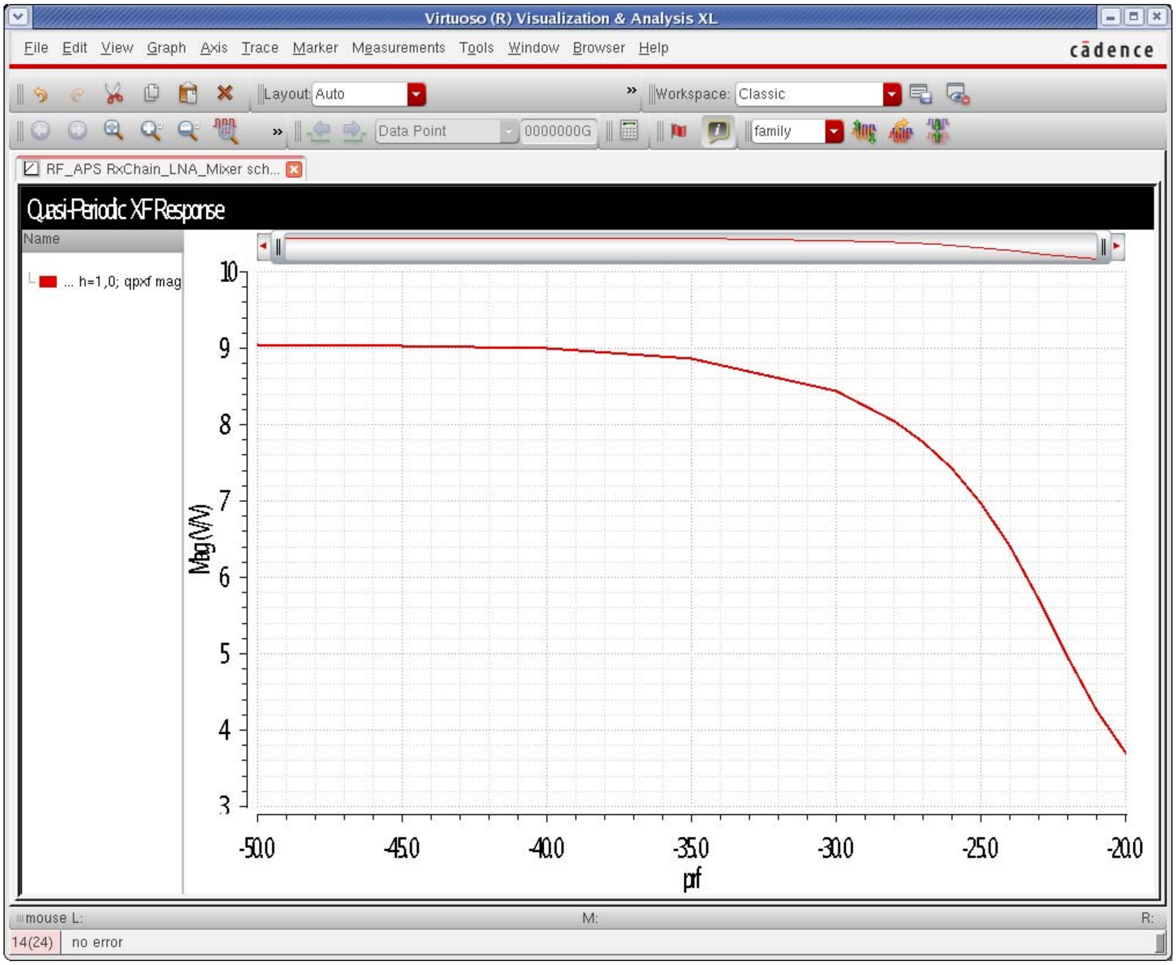


10. Select *variable* from the *Sweep* section.

11. Select the input frequency range you want to plot in the *Input Sideband* section. Note that the *Input sideband* field is labeled incorrectly as *Output Harmonic*.

12. If you want this measurement to plot automatically after subsequent simulations, click *Add To Outputs* at the bottom of the *Direct Plot Form*. In this example, it was not selected.

13. Click the desired source in the schematic for the gain measurement. For the plot below, the input port was selected.



14. As expected, the conversion gain drops with blocker power.

QPSP Analysis

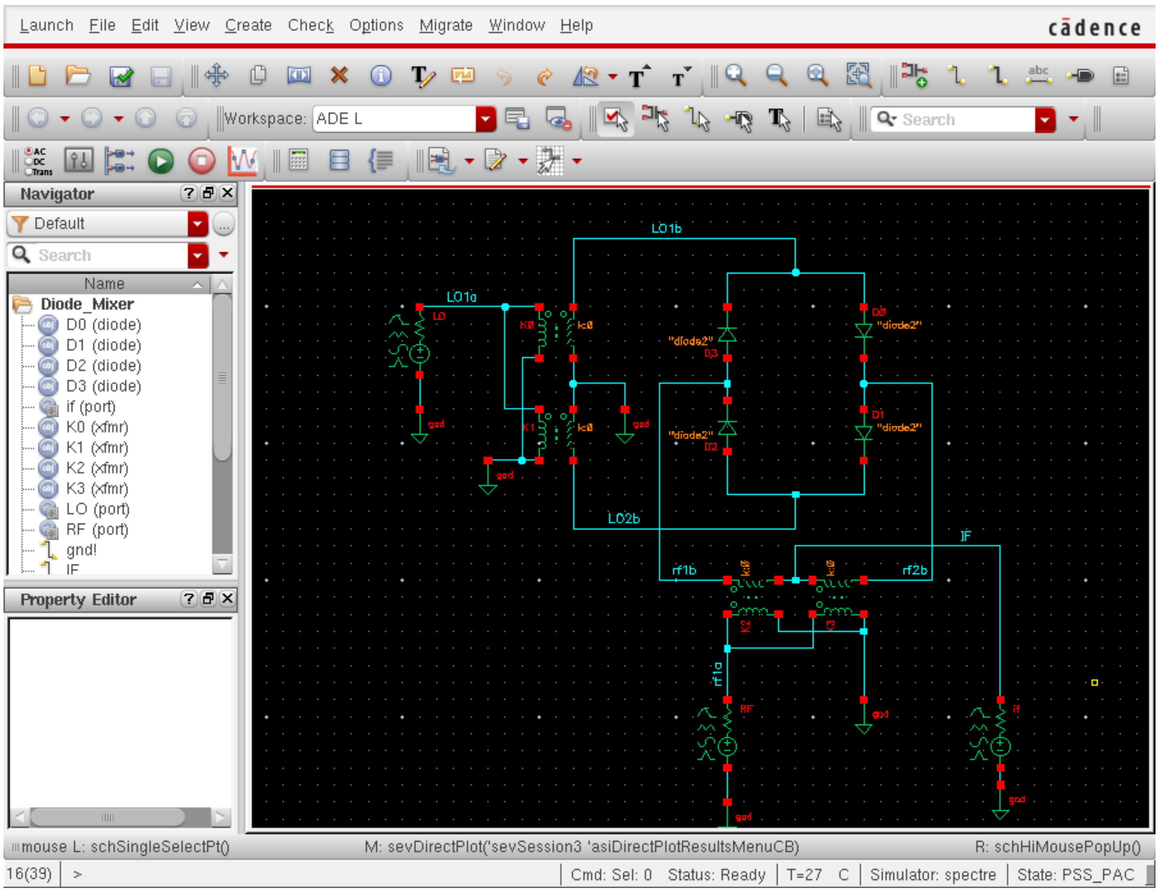
In order to run qpsp, a qpss analysis needs to be run first. For many applications, qpss will run much faster using the harmonic balance engine. If you want to use harmonic balance, instead of using the qpss and qpsp *Choosing Analyses* forms, use the hb and hbsp forms instead. The hb form always has the latest harmonic balance features in it, and the qpss form usually does not.

Please see the section in Chapter 4 on psp analysis. Qpsp is just like psp. The only difference is that instead of running after a pss analysis, it runs after a qpss analysis. Conceptually, the only difference is that in qpss, there are many more harmonics, and because of that there may be mixing paths from the input frequency to the output frequency by mixing with different harmonics. This will be shown in the example below.

Qpsp is provided to measure S-Parameters for circuits that translate frequencies. All the basic assumptions of S-Parameters apply to qpsp, with the addition of having different input and output frequencies. Qpsp can be used for up or down-conversion applications.

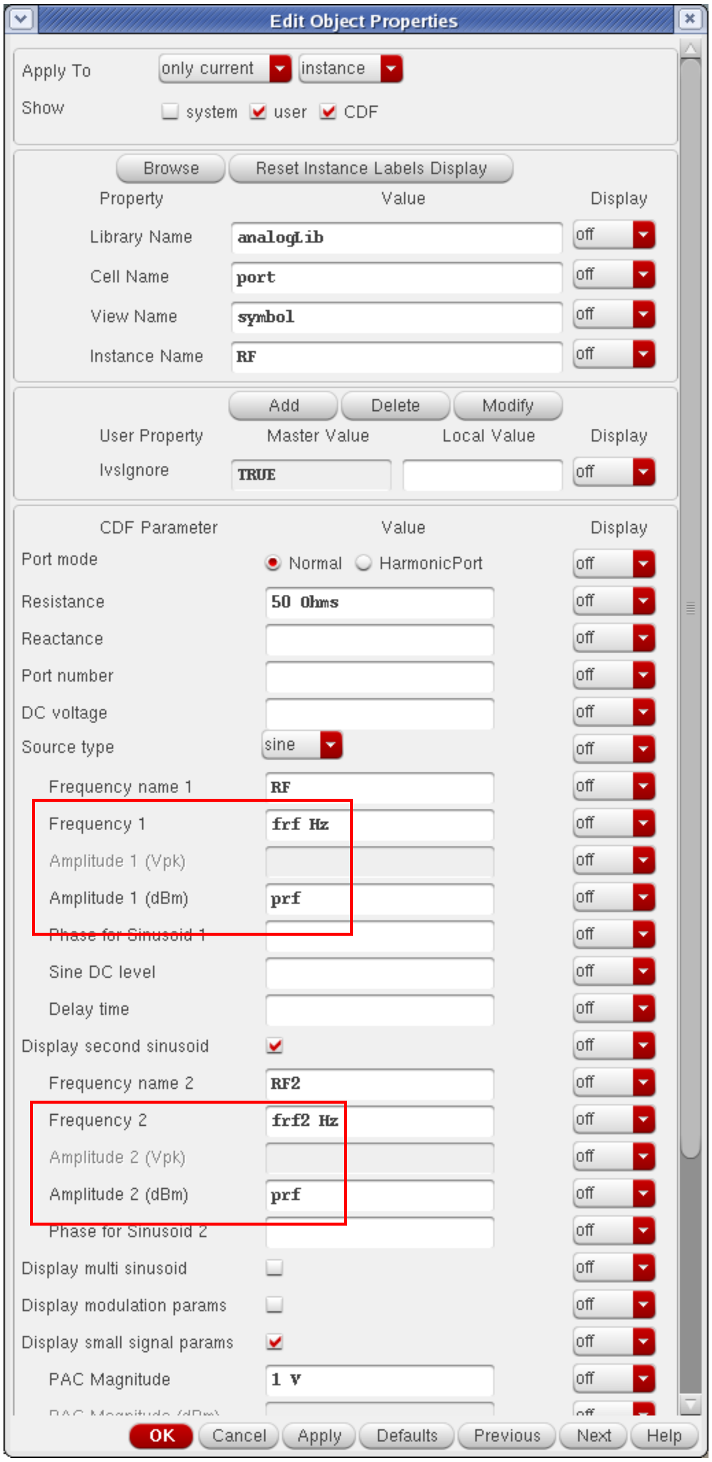
Example: Down-Conversion Diode Mixer

Consider the double-balanced diode mixer circuit below:

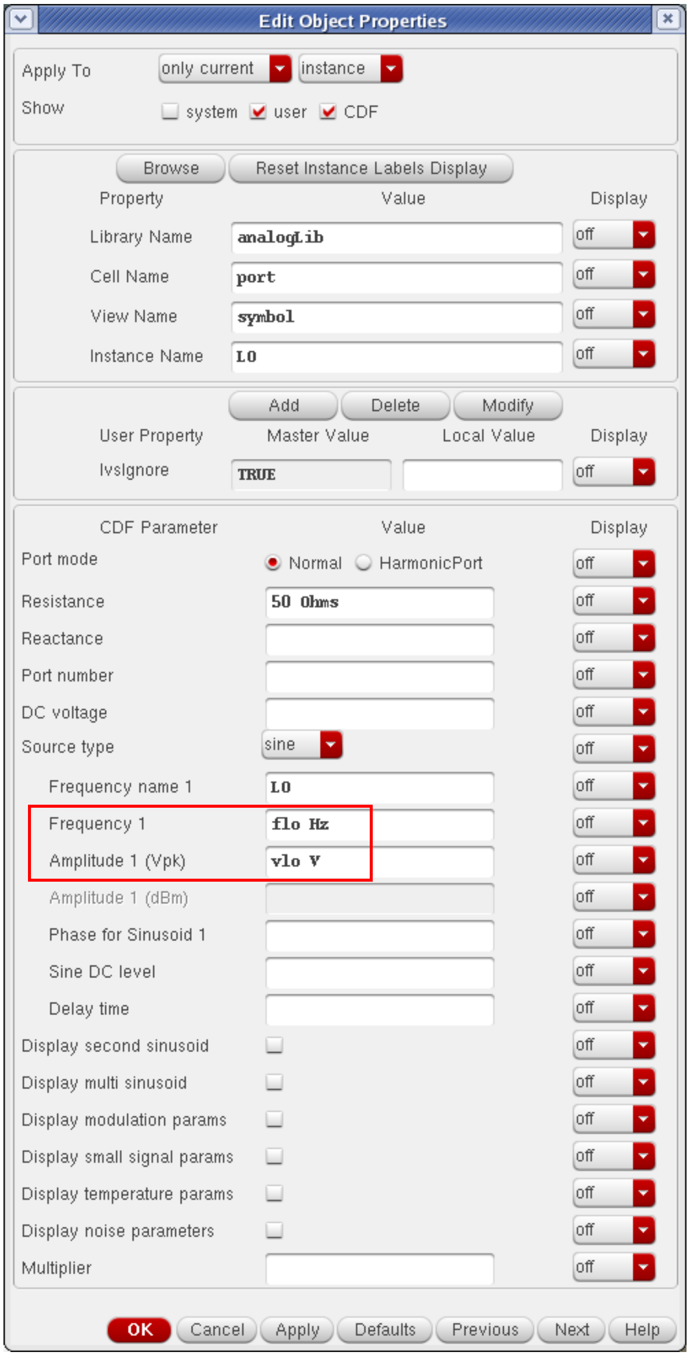


The input source has the frequency and amplitude set to variable names in order to allow changing the test setup without needing to change the circuit. This is suggested for all the sources used in the circuit that might need to be changed. This allows changes in frequency or amplitude from the ADE environment without needing to change the schematic itself.

The property list is shown below.

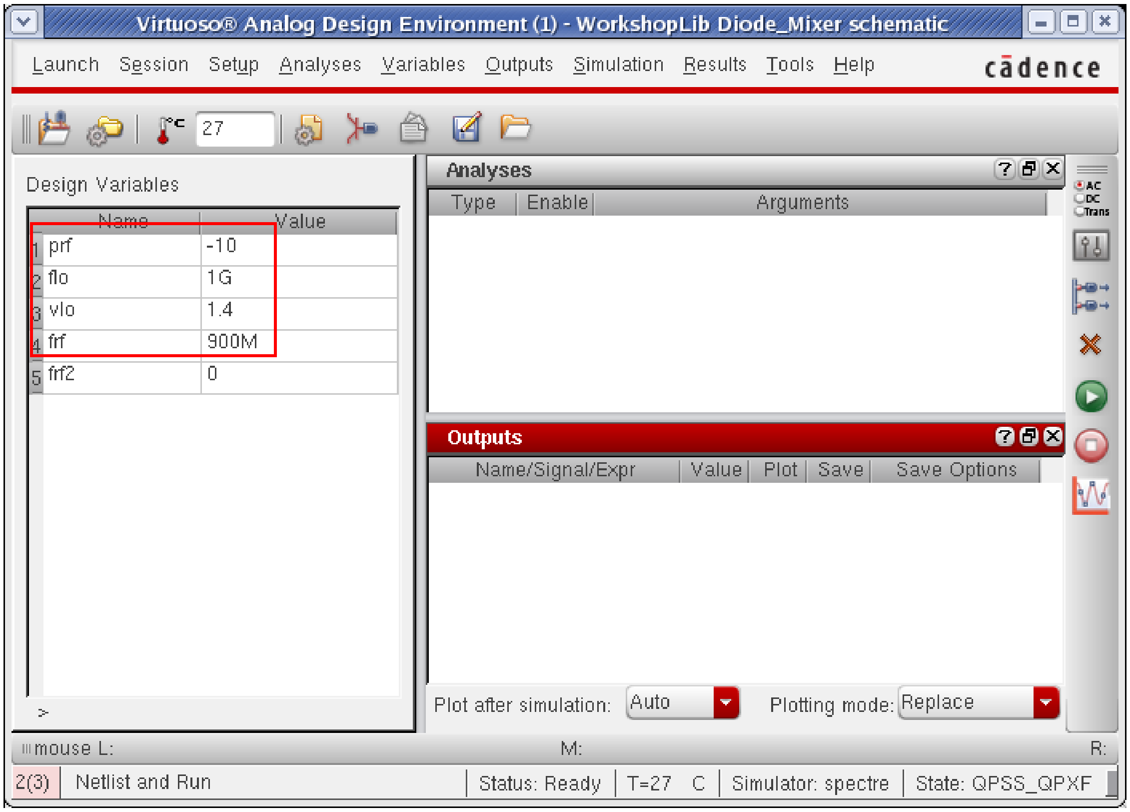


The same strategy is used for the LO port.



The easiest way to get the variables list into the ADE is to select *Variables -> Copy From Cellview*. To set the values, select the value field to the right of the variable, type in the desired value, and press *Enter.* When you have set the variables, if you want to add the

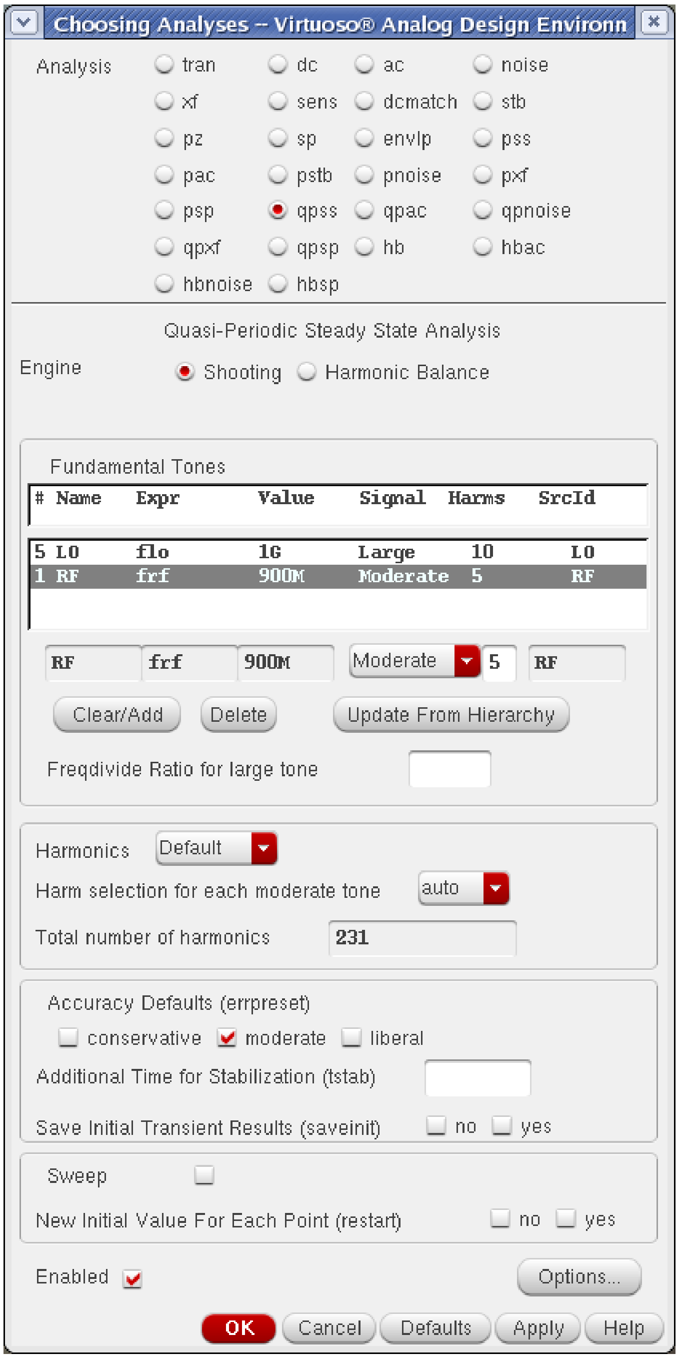
values back into the cellview, in the ADE window, select *Variables - Copy To Cellview*, and then perform a *Check and Save* in the schematic.



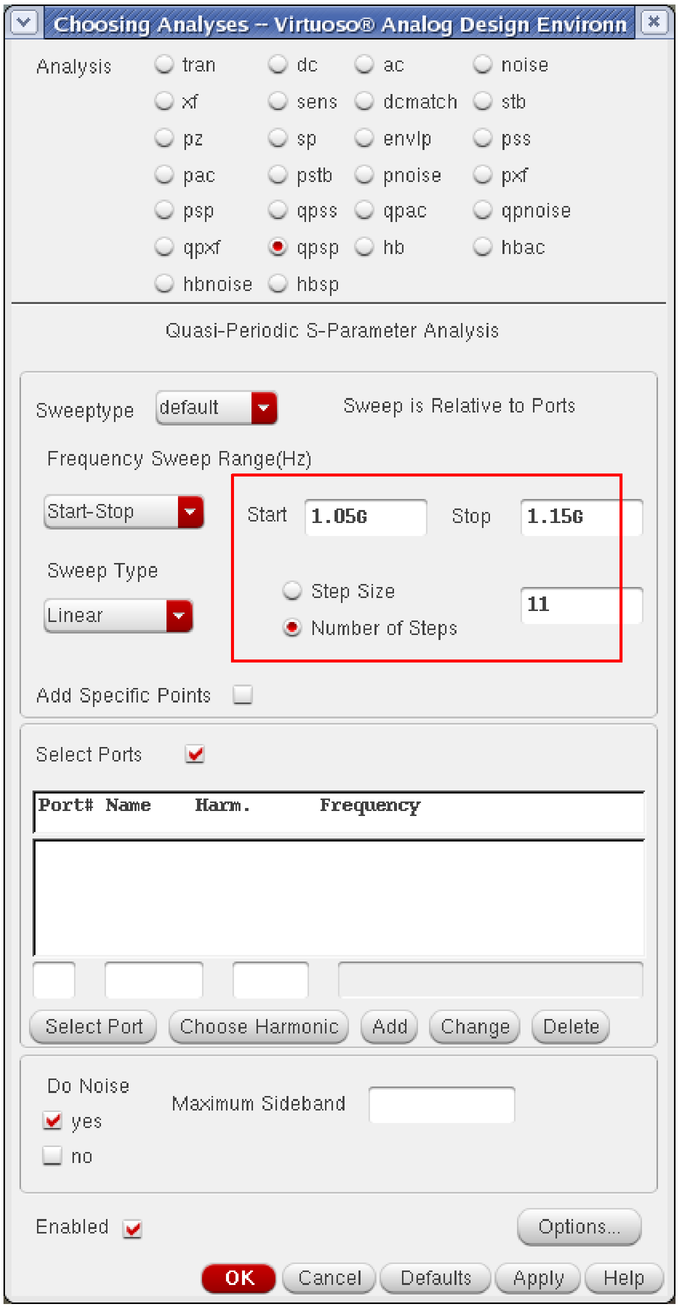
The first RF input is at 900MHz, the second RF input frequency is disabled (by setting the frequency to zero), and the LO is at 1GHz with a 1.4 volt peak signal. This is +13 dBm, which is a common drive level for a diode mixer.

The basic strategy is to apply the signals that cause the nonlinearity in the qpss analysis, and then measure the frequency-shifted S-Parameters using qpsp. Doing this allows measuring the input match at the input frequency, and the output match at the output frequency in the same analysis. For the forward and reverse gain terms, the specified frequency translation is taken into account.

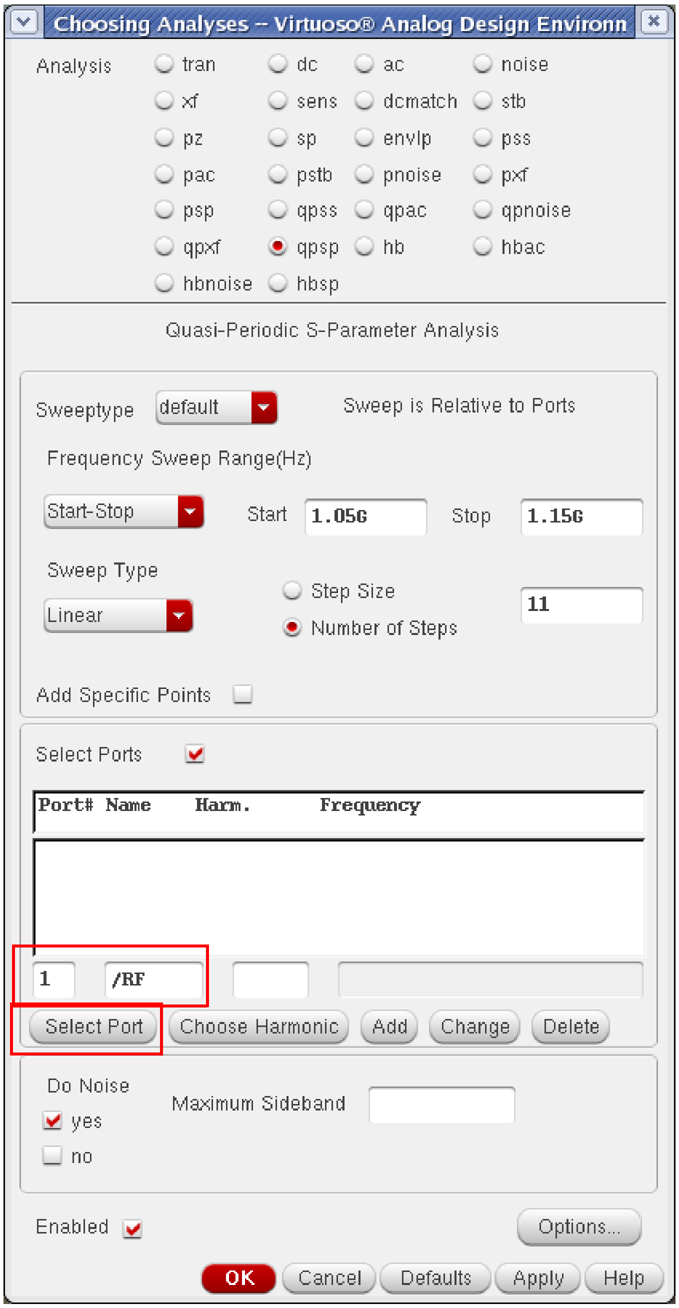
The qpss analysis needs to be set up first. The example below shows the LO input at 1GHz with 10 harmonics and the RF blocker input with 5 harmonics. For more information, see the qpss section at the beginning of this chapter.



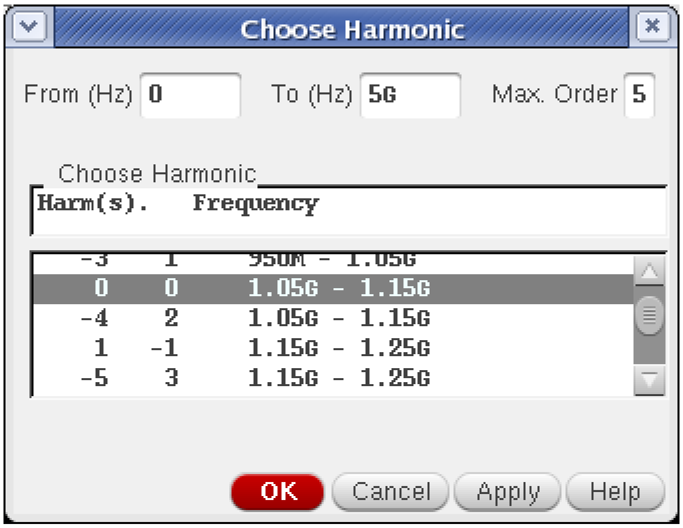
Next, the qpsp analysis needs to be set up. The frequency range depends on your application, and can be either an input frequency range, or an output frequency range. This is the input frequency range just above the LO frequency. The actual frequency for each port-frequency definition will be set up later.



Start with Port 1. In the *Port #* column field, type 1. In the *Name* column field, either type the instance name, or use the *Select Port* button just below the edit line to select the port in the circuit. In this case, Port 1 is the RF port.



Now click *Choose Harmonic*. In the *Choose Harmonic* window, select the frequency range you want associated with that port number. For the down-conversion application, The input is just above the LO frequency of 1GHz.



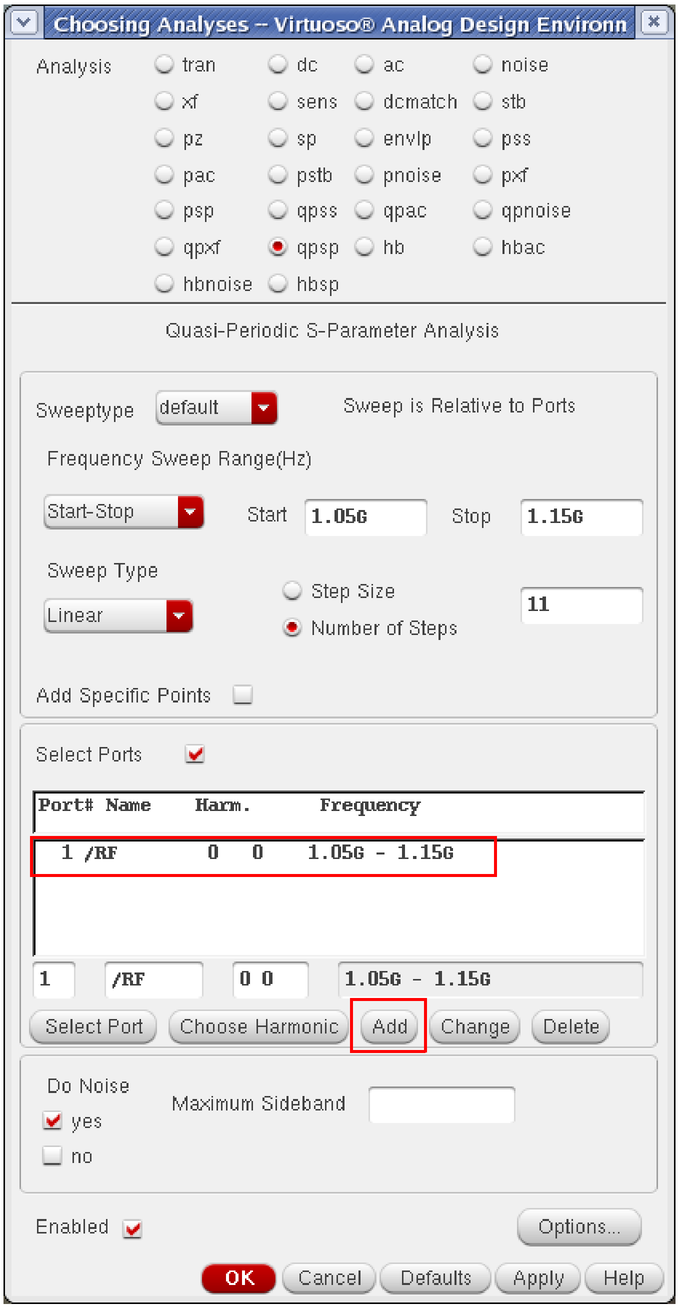
Note that there are multiple instances at the same input frequency range. This is because there are mixing paths using different harmonics that produce the same frequency range. Choose the entry where the sum of the absolute value of the indices is smallest. In this case, 0 + 0 = 0, and 4 + 2 = 6. Zero is less than six, so that is the entry that is chosen.

Like the other qpxx analyses, the sideband number displays the frequency shift for each input in multiples of the input frequency. In other words, a signal can mix with different harmonics of the input to provide an output.

In this case, 0 0, means do not mix with anything. The center frequency is 1.1GHz. For the other term, 1.1GHz -4*(1GHz) +2*900MHz = -1.1GHz. The default is to reflect this term to the positive frequency domain. The -4 2 term mixes down with the fourth harmonic of the LO and up with the second harmonic of the RF blocker.

Click *OK*.

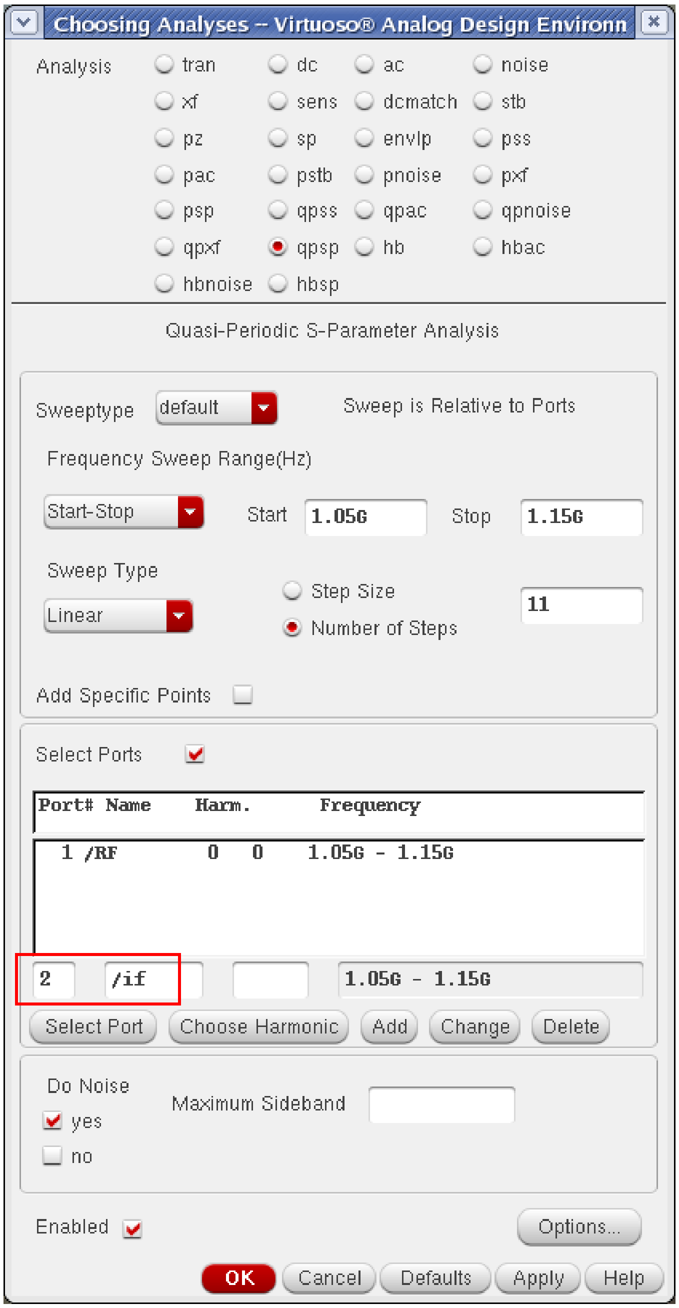
Click *Add* to the right of the *Choose Harmonic* button. This adds that port and frequency definition to the list for the analysis. By convention, port 1 defines the input frequency.



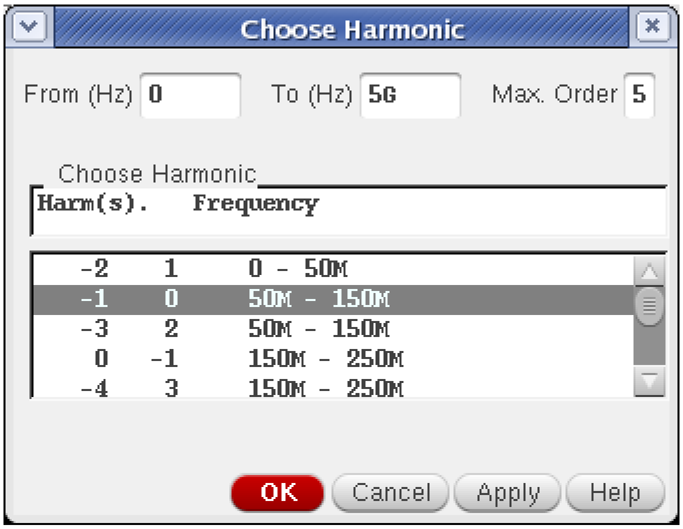
Note that port 1 is now the input port at the input frequency range of 1.05GHz to 1.15GHz. A single port definition in qpsp has both a port name and a frequency definition.

In qpsp, multiple port and frequency definitions can be run at one time.

The next step is to measure RF to IF conversion gain. Type *2* in the *Port #* column field. Click the *Select Port* button, and select the IF port.



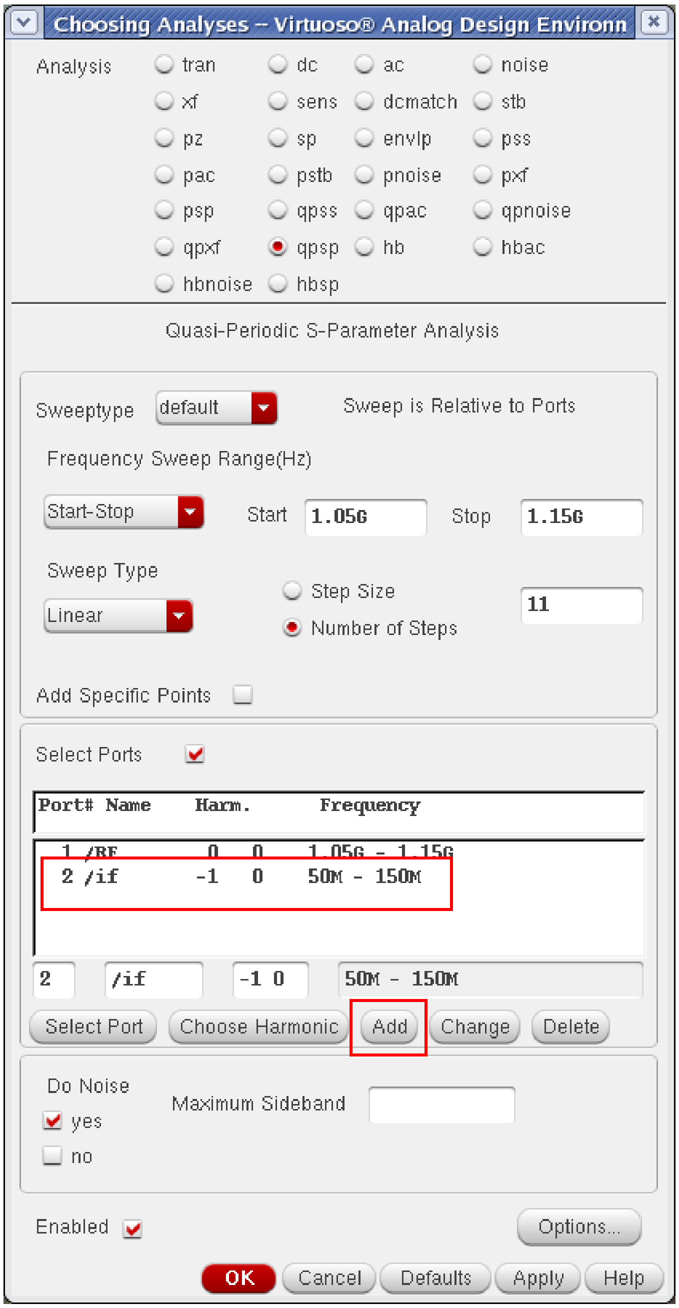
Now define the frequency range for that port. Click *Choose Harmonic*, and select the IF frequency from the list. Next, click *OK*.



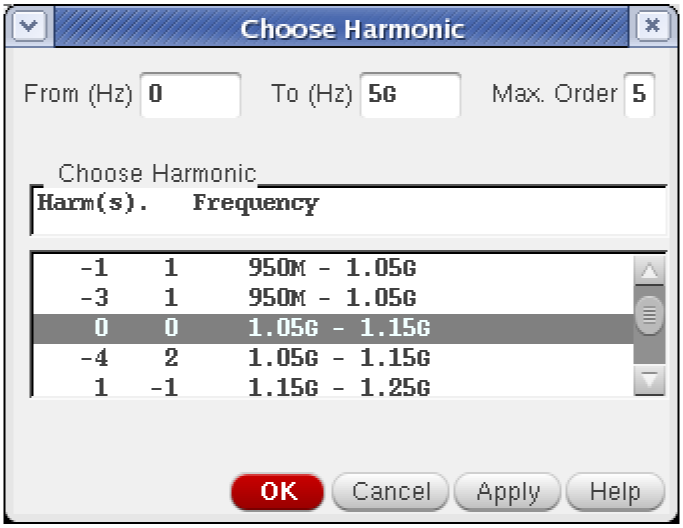
Again, there are multiple entries at the same frequency from multiple mixing paths. Taking the center frequency of 1.1GHz, then one term is 1.1G - 1 * 1G = 100M. The other term is 1.1G -3 * 1G + 2 * 900M = -100MHz

Take the entry where the sum of the absolute values of the indices is smaller.

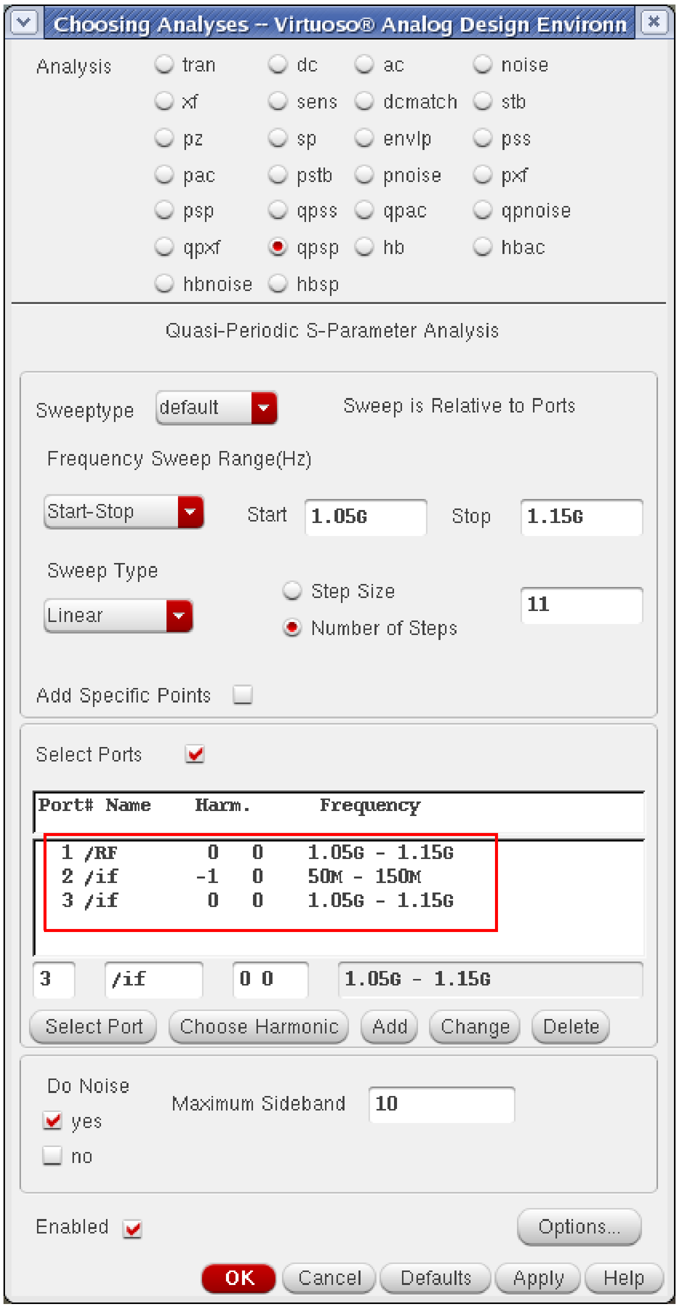
Click the *Add* button to the right of the *Choose Harmonic* button. The second port-frequency definition is added to the list. By convention, port 2 defines the output frequency range.



Now measure the RF to IF isolation. Type 3 in the *Port #* column field. Click *Select Port*, and again select the IF output port. Click *Choose Harmonic*, and select the input frequency range. Click *OK*. Once again, there are multiple entries at the same frequency from multiple mixing paths. Take the entry where the sum of the absolute values of the indices is smaller.



Click *Add* to the right of the *Choose Harmonic* button. The third port-frequency definition is added.

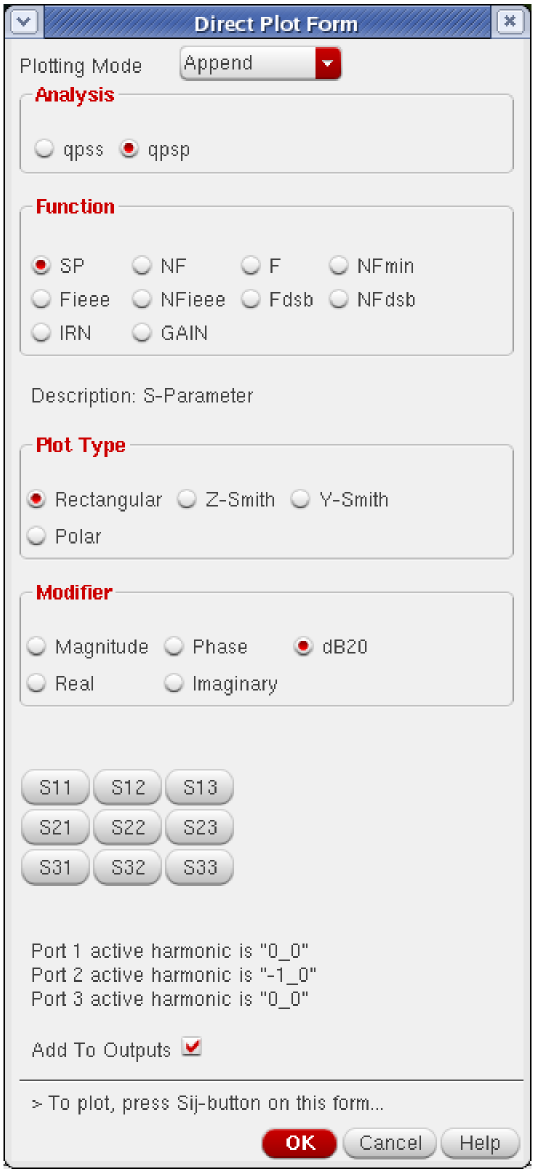


Make sure all the port definitions are in the list. It is easy to forget to add the last port.

With the above port-frequency definitions, S_{11} measures the input match at the RF frequency. S_{22} measures the output match at the IF frequency. S_{31} measures the RF to IF isolation, and S_{21} measures the RF to IF conversion gain.

Once the analyses are set up, the simulation can be run. When the simulation finishes, select *Results - Direct Plot - Main Form* in the ADE window.

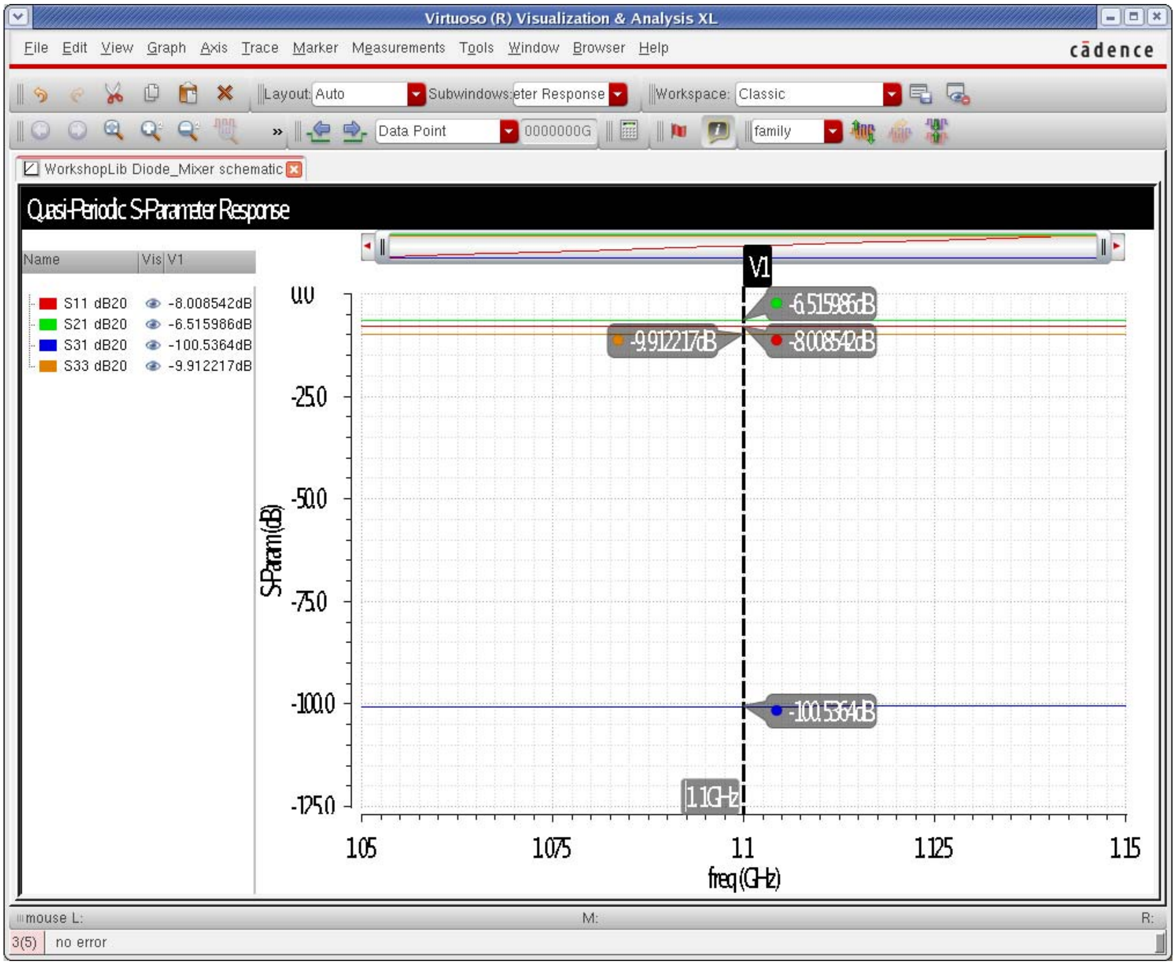
The example below shows a typical *Direct Plot Form*.



If you click *Add To Outputs*, an expression is entered in the ADE outputs section so that the next time you simulate your circuit, this measurement plots automatically.

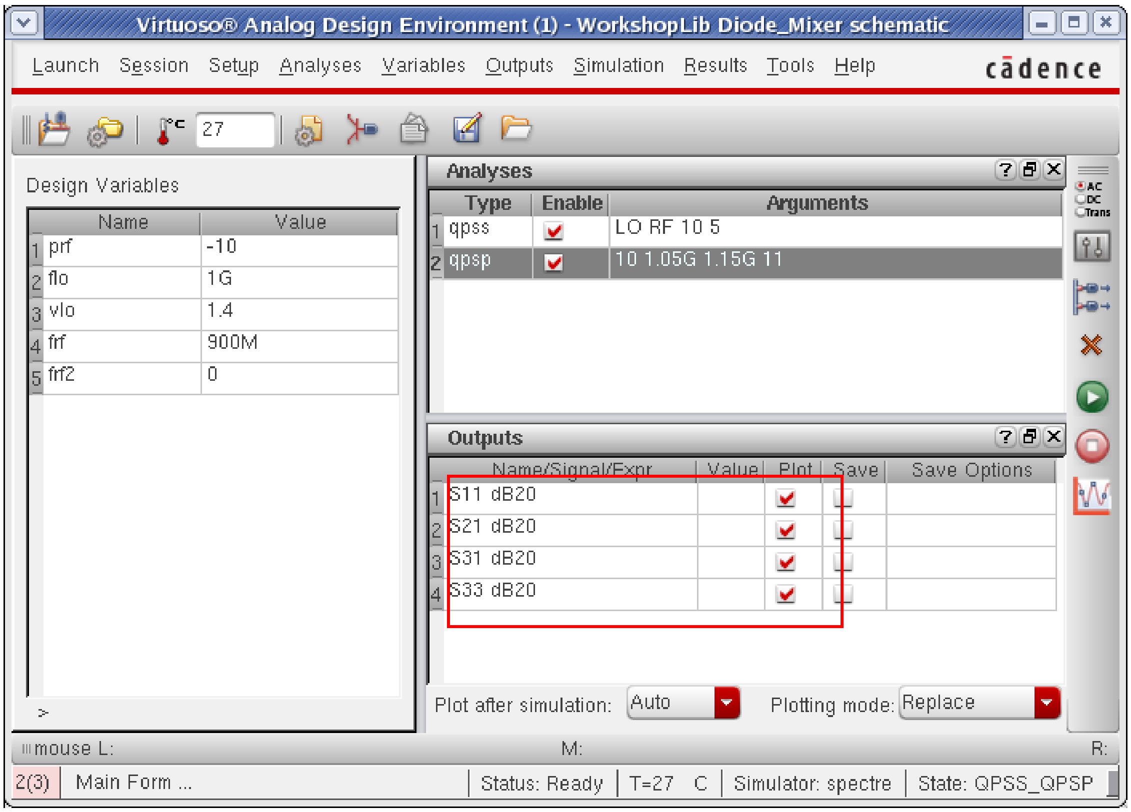
Here are the S_{11}, S_{21}, S_{31}, and S_{33} plots.

.



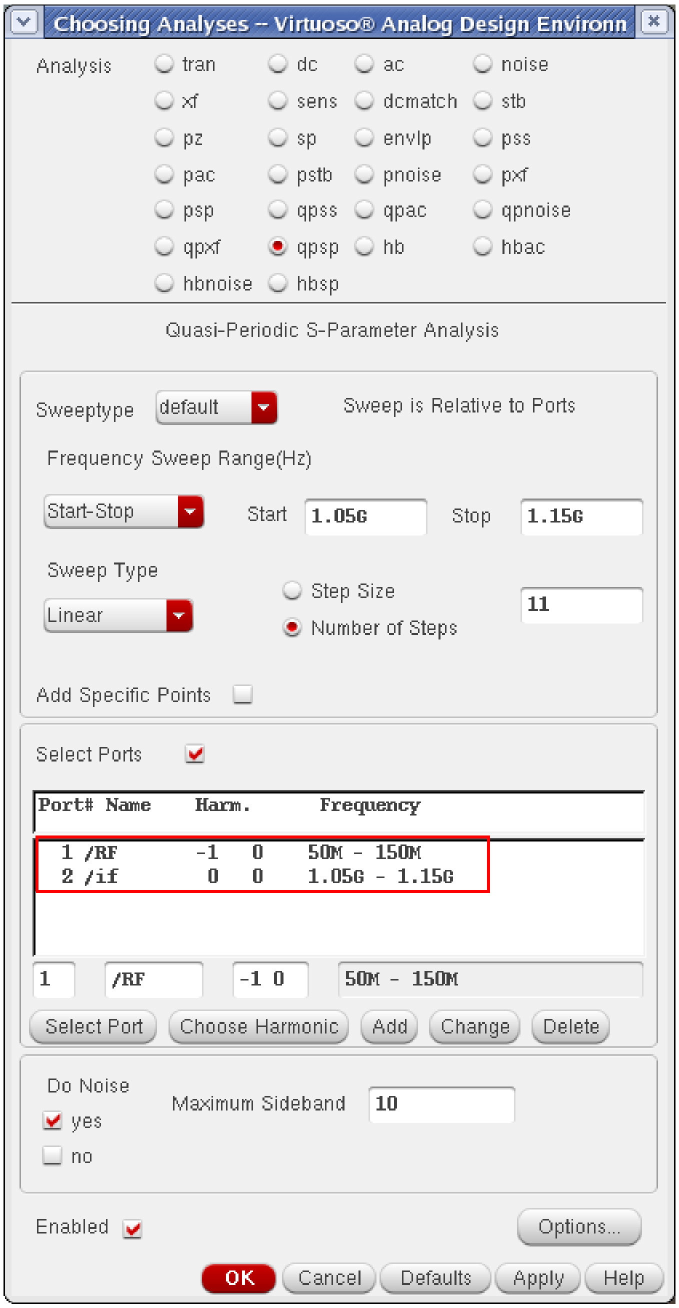
The conversion power loss is about 6.5dB. The input match is about -8dB. The output match is about -9.9dB. The RF to IF isolation is about 100dB.

Because *Add to outputs* was selected in the *Direct Plot Form* as shown previously, ADE now has expressions that will plot automatically on subsequent simulations.



Note that these can be plotted on a Smith Chart as well. The Smith Chart capability is only available from the *Direct Plot Form*. Expressions in the ADE outputs section will always plot on a rectangular plot. The Smith chart has not been shown because this circuit is inherently very wide band. The curve on the Smith Chart would be tiny. An example using a Smith Chart is shown later on in the Examples section.

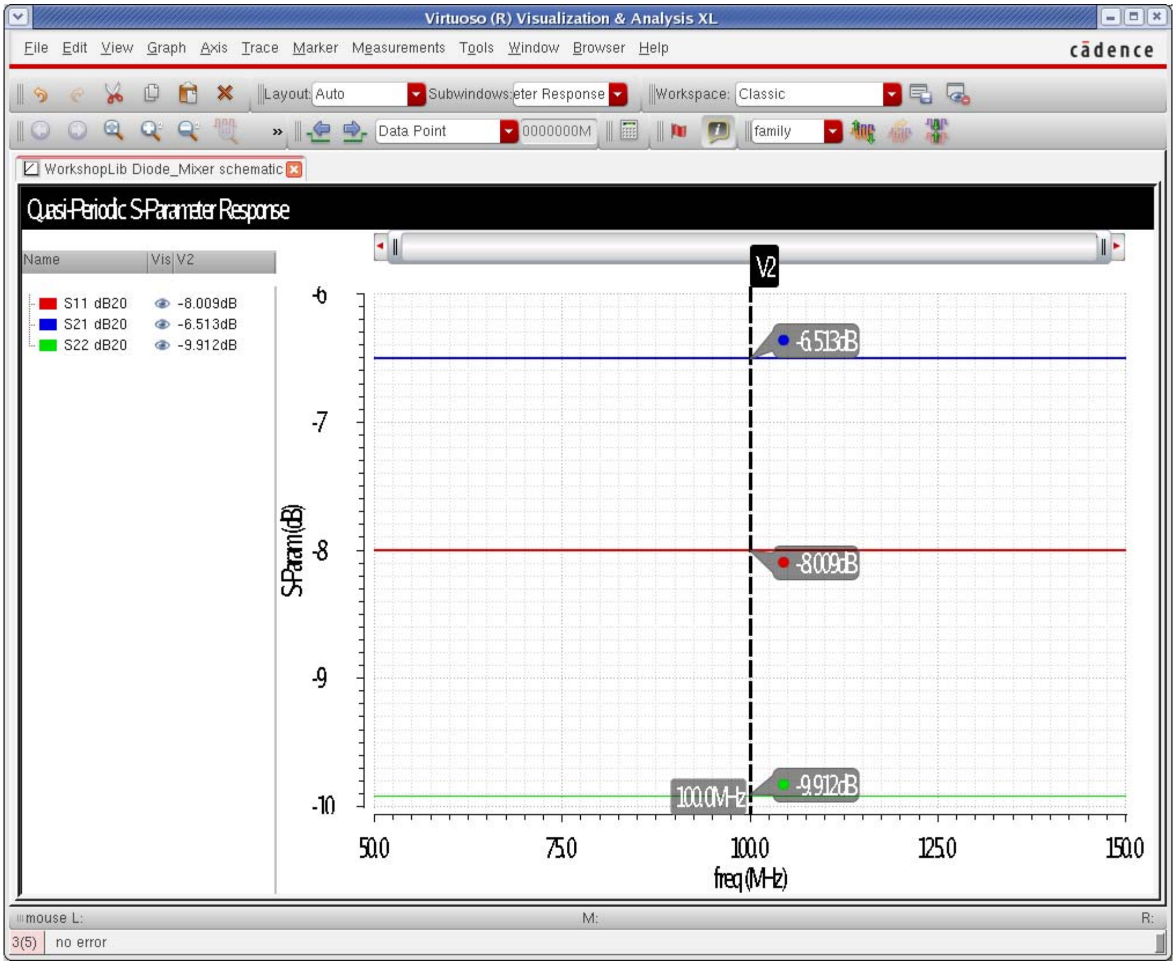
Qpsp can be used for up conversion as well. To do that, just change the frequency definitions for the ports. Since IF to RF gain is usually not needed, only a 2 port measurement has been set up. Note the frequency difference for Ports 1 and 2.



Note that the *Frequency Sweep Range* did not change. Only the frequency definitions for the two ports changed. With the above port-frequency definitions, S11 measures the input match at the low frequency.

S22 measures the output match at the upconverted frequency. S21 measures the small-signal conversion gain from low to high frequency.

Now run the analysis. S_{11}, S_{21}, and S_{22} are plotted below. The conversion power loss is about 6.5dB. The input match is about -8dB. The output match is about -9.9dB.



Note that the frequency range on the X Axis of the plot changed. This is because the input frequency changed, and by default, the input frequency is shown on the waveform plot X Axis.

Overview of Simulation Capabilities

Qpsp is similar to sp in that power gain measurements are made. Sp is based on the DC operating point, so it can only produce results that are linear. All the frequencies are the same for all the S-Parameter measurements.

Qpsp uses the qpss analysis results as the basis for making the S-Parameter measurement. Because the qpss result is the operating point solution for qpsp, the input can mix with any of the harmonics that are present in the qpss analysis for the frequency-translated terms. The term S_{XX} is not enough to describe the measurement. We need not only port numbers and names, but also the frequency needs to be defined for each port definition.

In the example in the preceding section, the LO was at 1GHz and the RF input was at 900MHz. The specified frequency for qpsp was just above the first harmonic of the LO. In the qpsp definition for each port, both the port name and the frequency that is associated with that measurement need to be defined.

Only the S-Parameters and the noise calculations are available for plotting in qpsp. Stability factors like Kf and the other calculations associated with sp like Gamma-Min are not available in qpsp.

Noise analysis is available in qpsp, and is the same as pnoise. Since very few people use qpsp for noise measurements, it is not covered in this section. Please see the pnoise or hbnoise chapter for noise information.

Setting Harmonics and moderate Sidebands

In the qpss analysis, when shooting is selected, each slice of the Large signal (usually the LO signal) has a minimum of 200 timepoints, so it inherently has frequency domain content through the 100th harmonic of the signal in each slice. Only the harmonics specified for the moderate tones affect the accuracy of the qpss solution. If you specify enough harmonics in the moderate tones in the qpss analysis to accurately capture the circuit performance, then the qpsp solution will be accurate.

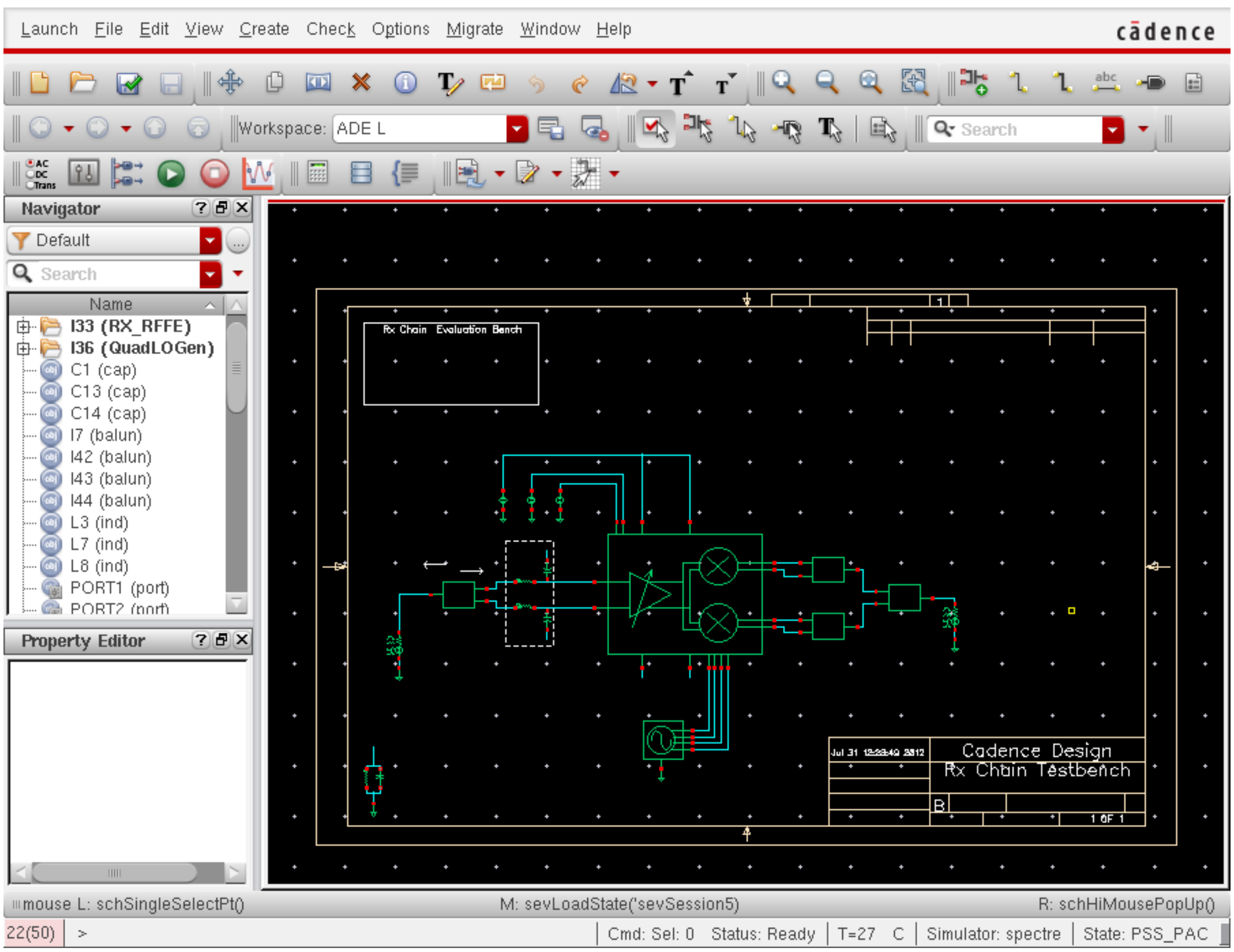
To see if you have enough moderate tones, set a number of moderate harmonics, and run the simulation. Make the measurement. Now increase the number of moderate harmonics by about 50%. Run again, and make the same measurement. If the measurement did not change significantly, you had enough moderate harmonics to begin with, and you might be able to reduce the number of moderate harmonics. If it did change, then you need more moderate harmonics. If your moderate signal is below compression, start with two or three harmonics. If the moderate signal is at compression, start with five harmonics. If the moderate signal is above compression, start with about seven harmonics.

ADE Implementation

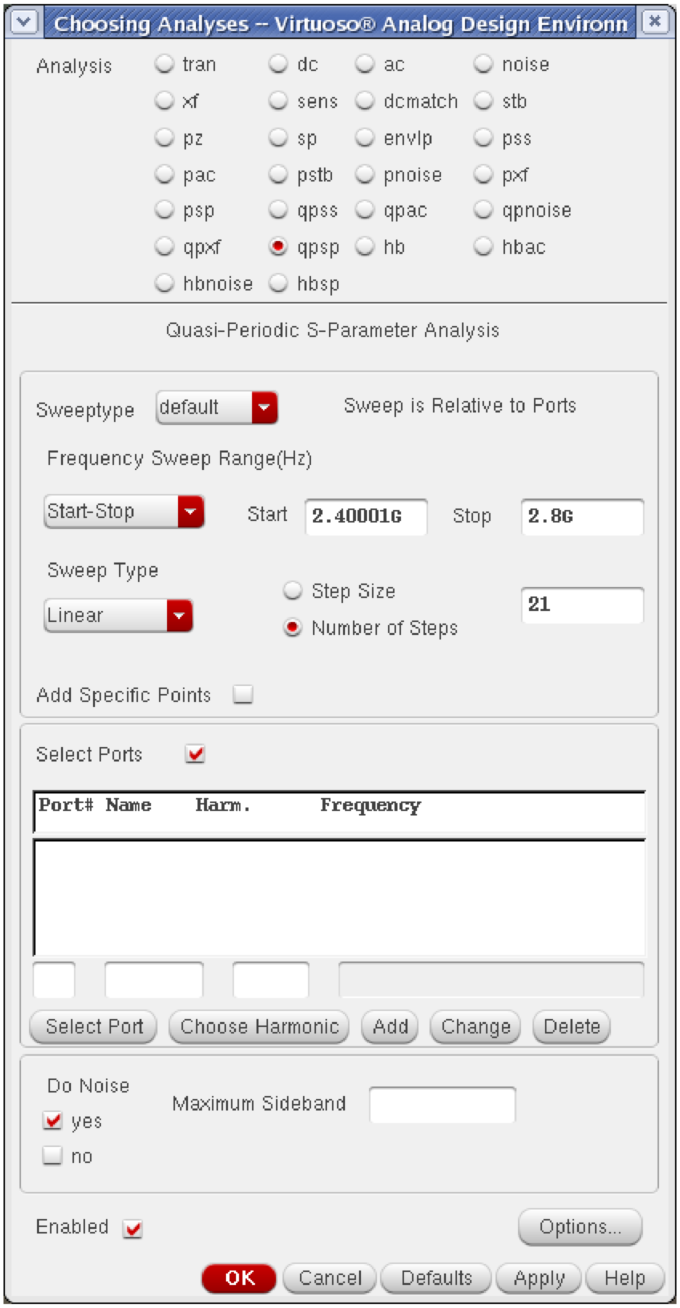
This section jumps from theoretical to the practical with examples of how to use qpsp. The focus is only on the qpsp *Choosing Analyses* form so you can see where the individual settings are made. To see examples with all the steps shown, see the examples section that follows this section.

QPSP Setup

For this example, consider an LNA-Mixer circuit shown below.

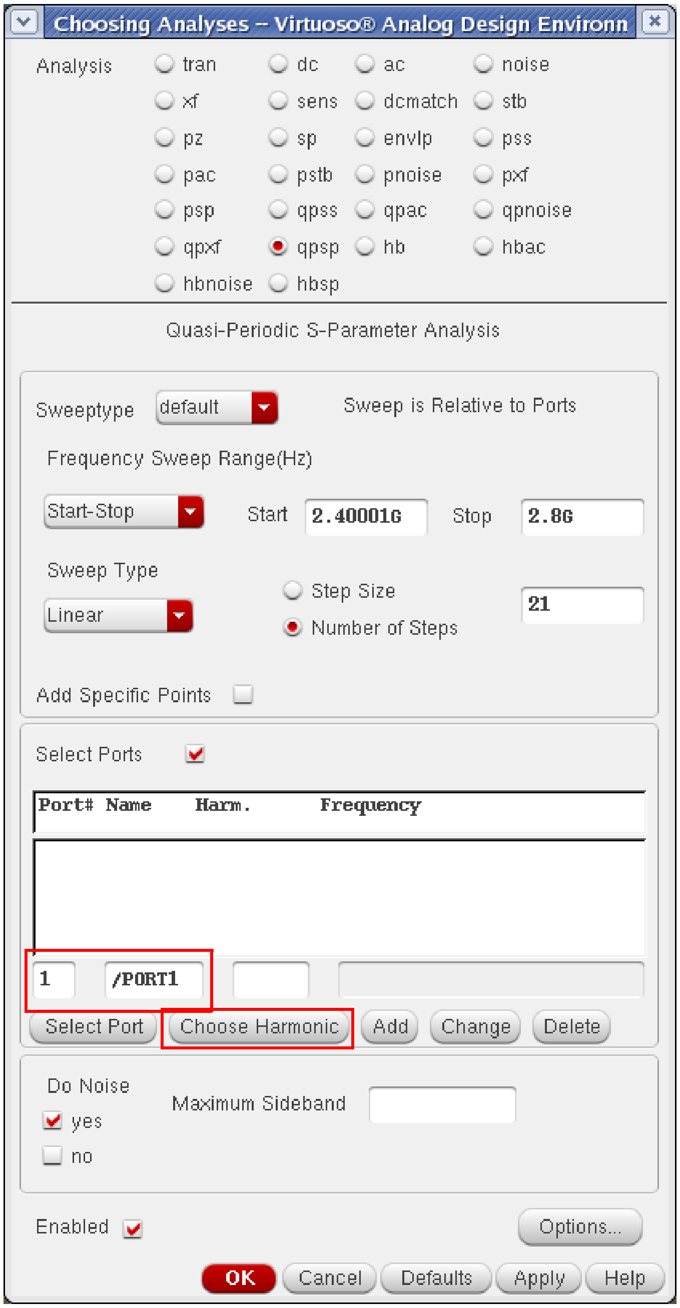


1. First, define either an input or output frequency range in the *Frequency Sweep Range* fields. It is usually easier to define an input frequency range, as shown below for a 2.4GHz application.

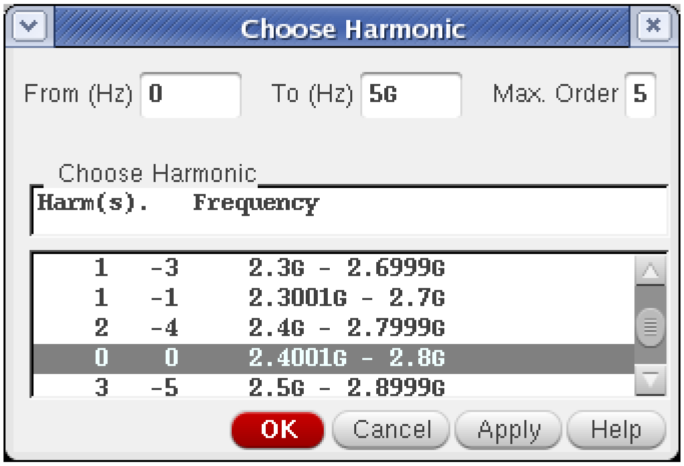


2. Linear and log sweeps are provided. It is usually better to select Linear or Log, and specify the number of steps instead of using automatic. Automatic will run 50 frequency points, which usually is not needed for most applications.

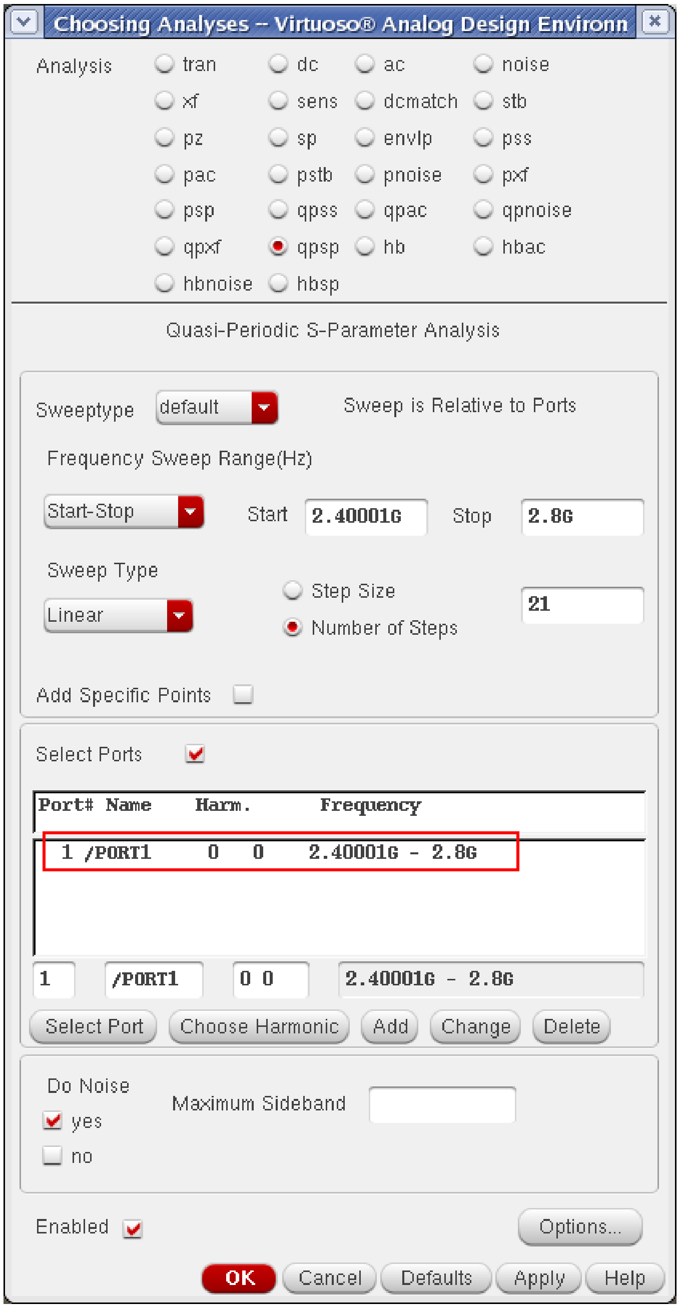
3. Now port and frequency definitions need to be made.
4. Type 1 in the Port # column field. Click *Select Port*, and select the input port.



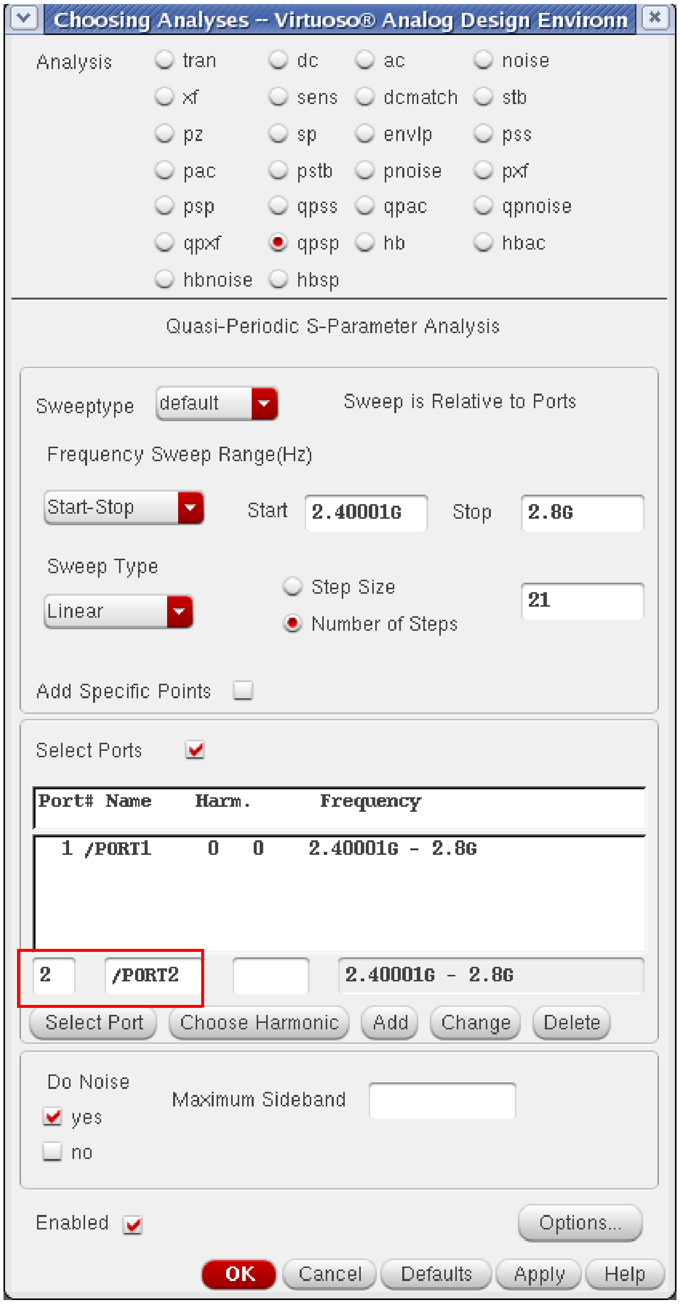
5. Click *Choose Harmonic*. In the window that appears, select the input frequency range. Now select *OK*.



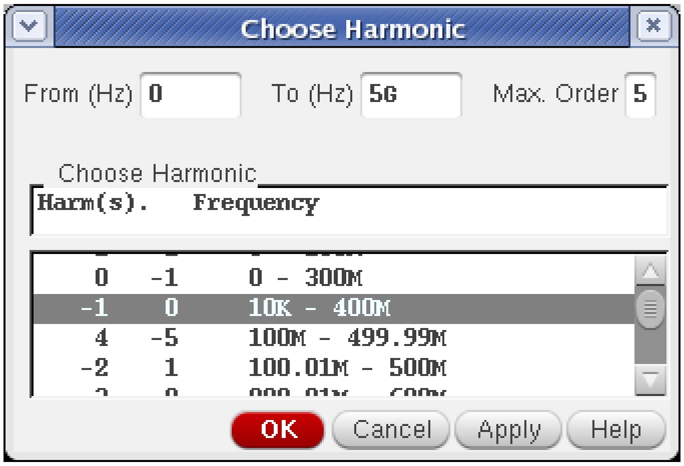
6. Click Add. The port-frequency combination is added to the list. By convention, port 1 sets the input frequency for qpsp.



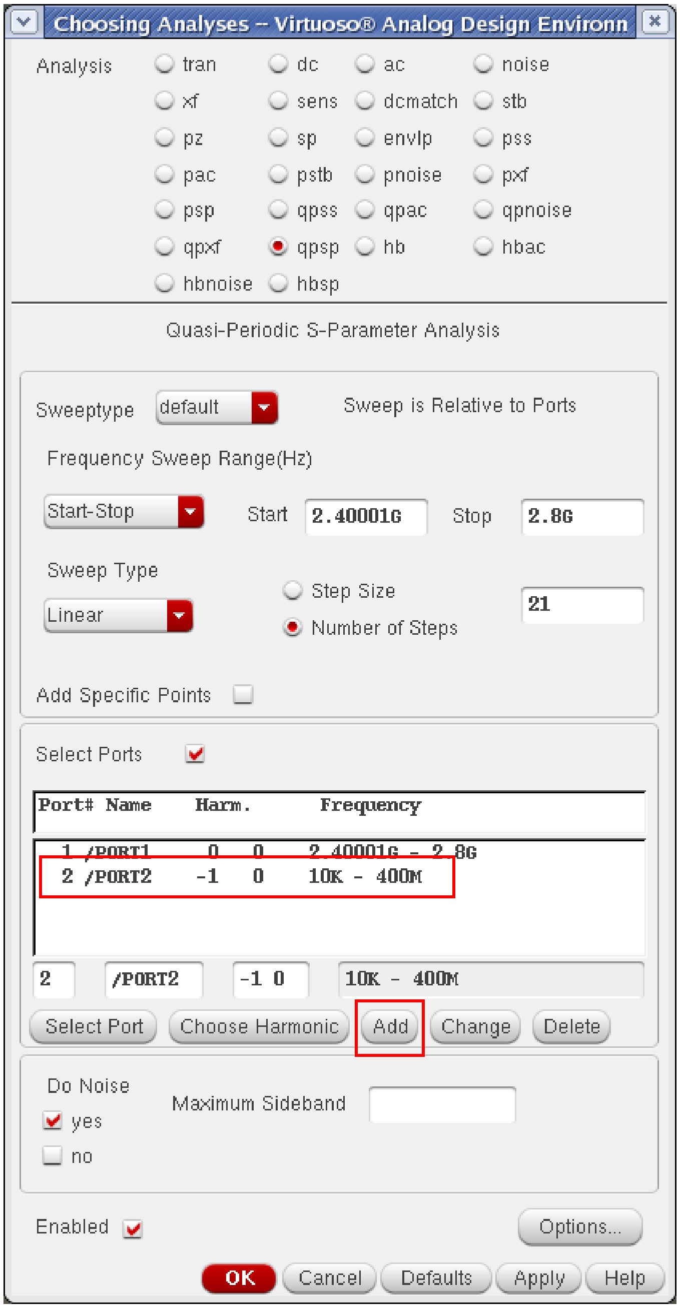
7. Now define the output port as port 2.



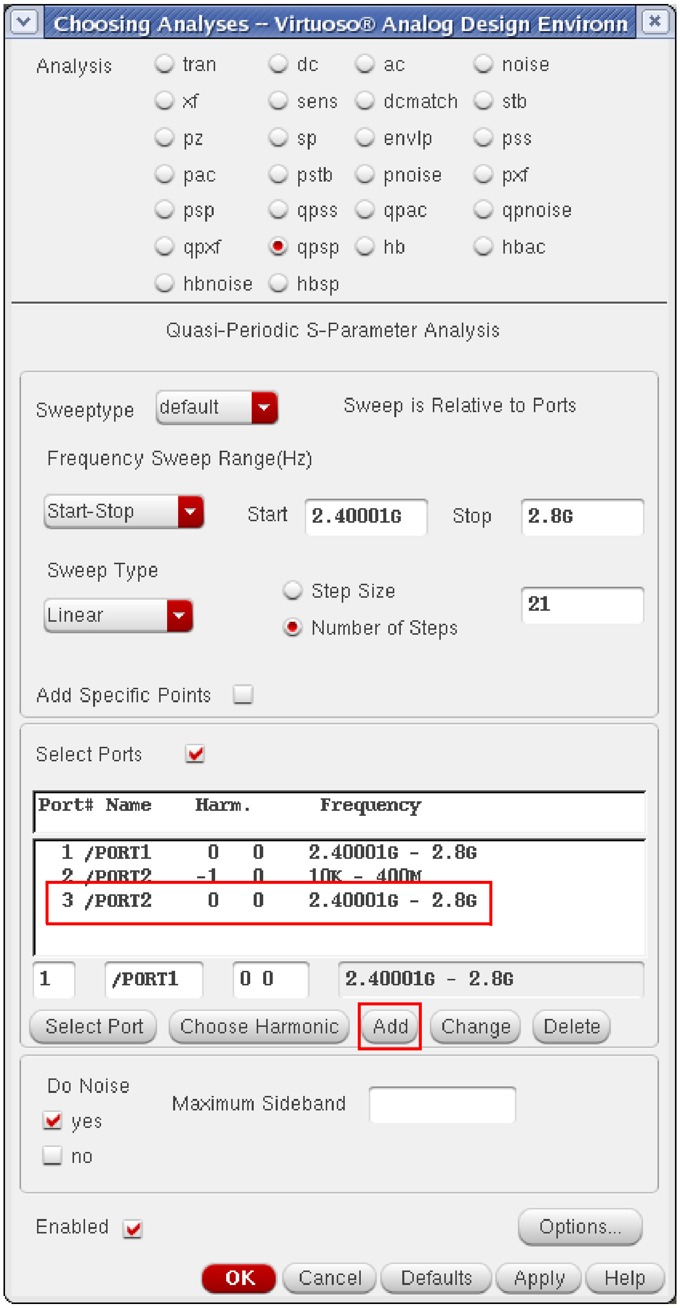
8. Select the output frequency range by clicking *Choose Harmonic*, and selecting the output frequency range as shown below.



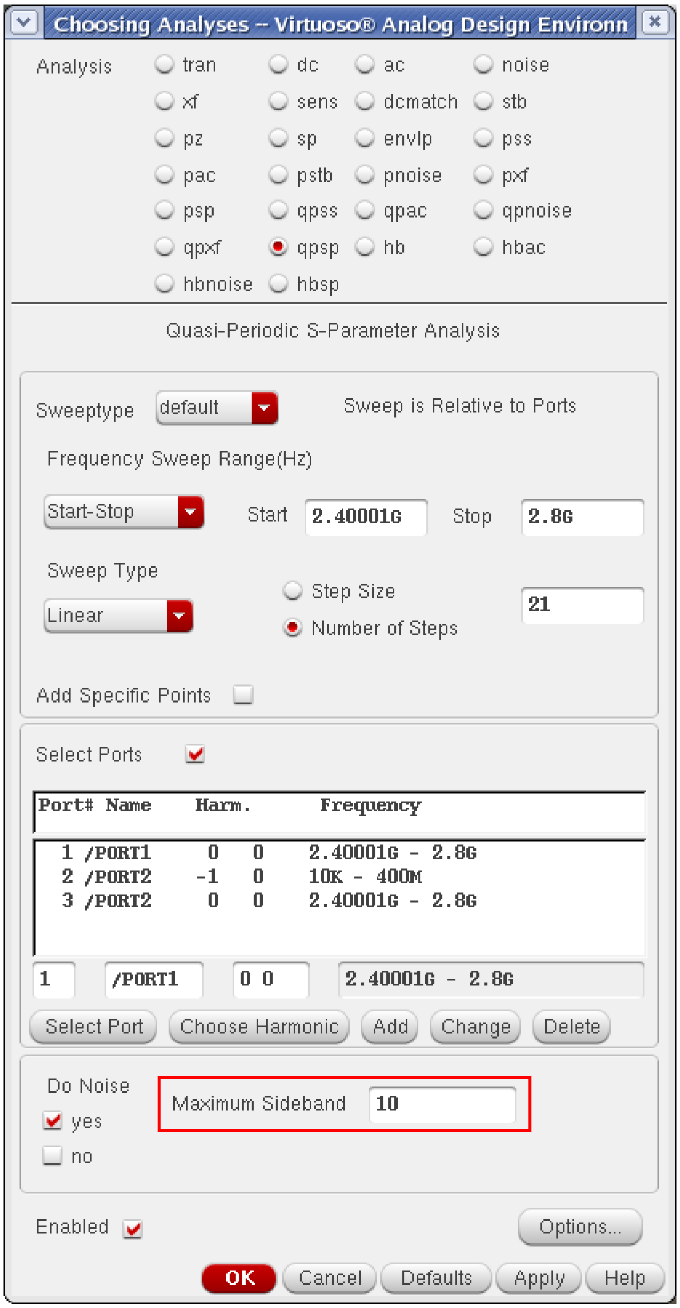
9. Next, click *Add*. This adds the port/frequency combination as port 2 in the list. By convention, port 2 sets the output frequency.



10. If necessary, repeat the process to define port 3 as the gain from RF to IF with no frequency translation. (The RF to IF isolation.)

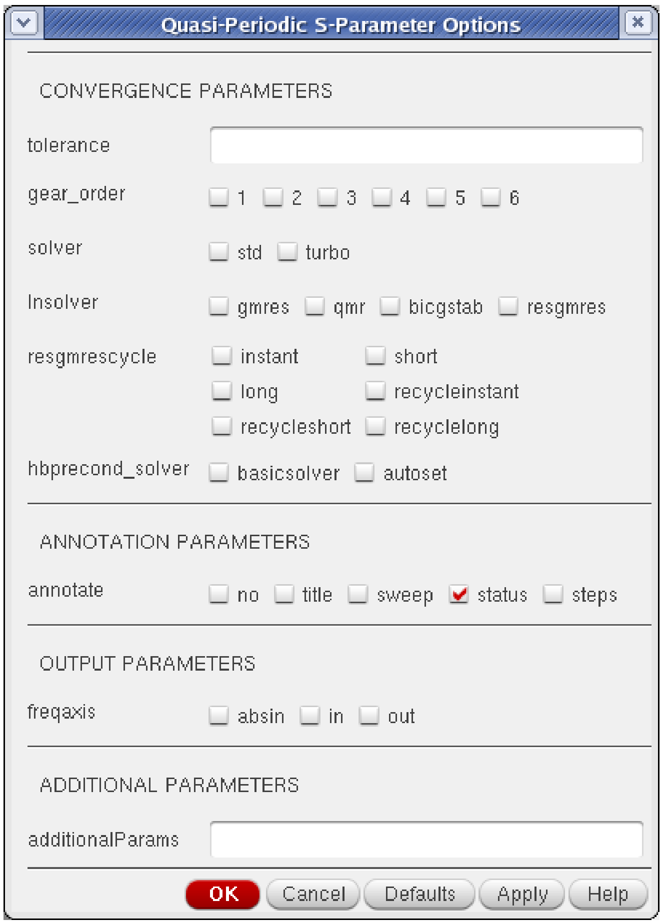


11. If you want a noise simulation, define the *Maximum Sideband* term above. For more information, see the pnoise section in this chapter.



QPSP Options

Below is the qpsp options form.



None of the options are commonly used.

tolerance

Leave this option at the default value.

Qpsp uses an iterative solver to calculate the S-Parameter measurements. Any iterative solver needs an error tolerance to specify when to stop iterating because the solution is accurate enough. The tolerance option specifies that accuracy for qpsp. For shooting, the default tolerance is 1e-9. For circuits where HB is the qpss engine, the default is 1e-6.

gear_order

Do not change this option.

solver

The default solver is the turbo solver.

When hb is used as the engine in qpss, leave this option at the default.

When shooting is used, sometimes when the qpsp analysis frequency is very close to the frequency of one of the harmonics in the qpss analysis, warning messages will appear in the qpsp output warning that the accuracy might not be good enough. If you see these messages, select the *std* solver, which has better ability to handle frequencies that are very close to a harmonic in the qpss, but which takes longer to run than the turbo solver.

lnsolver

Leave this option at the default.

Each qpsp frequency point solution is internally calculated from a matrix. This matrix is solved using an iterative solver. Several different algorithms are provided for the iterative solver. *gmres* is the default because the accuracy of each iteration inherently increases. Convergence is generally good as well. The other solvers may require less memory, but they are less robust for convergence and may suffer from false convergence.

Considerable knowledge is needed to understand the differences between these methods. For more information, refer to the books on linear algebra theory.

- **gmres** is the Generalized Minimum RESidual method.
- **qmr** is the Quasi-Minimal Residual method.
- **bicgstab** is the STABilized BI-Conjugate Gradient method.
- **resgmres** is the REStarted Generalized Minimal Residual method.

resgmrescycle

Leave this option at the default. For the resgmres linear solver, there are several different options.

hbprecond_solver

This option is available only when harmonic balance selected in qpss and APS is used.

The basic solver is the only solver available in standard Spectre when harmonic balance is selected. Autoset is the default solver in harmonic balance when APS is used. This solver is faster, but occasionally stagnates. When stagnation is detected, APS automatically switches to the basic solver, and prints a message in the Spectre output window. If you have stagnation, it will save a small amount of time to set this option to basicsolver.

annotate

This option controls the level of detail in the Spectre output log. No detail is provided when you select no, while more details are provided as you move towards right with steps option providing maximum detail.

freqaxis

Qpsp displays all the results on the same frequency scale. The default is absin (absolute value of the input frequency). If there is a negative input frequency, in will show that frequency on the positive frequency axis. The selection In also shows the input frequency, but if the input frequency is negative, it will be shown on a negative frequency axis. The selection Out shows the output frequency range. If the output frequency is negative, it will show on a negative frequency axis.

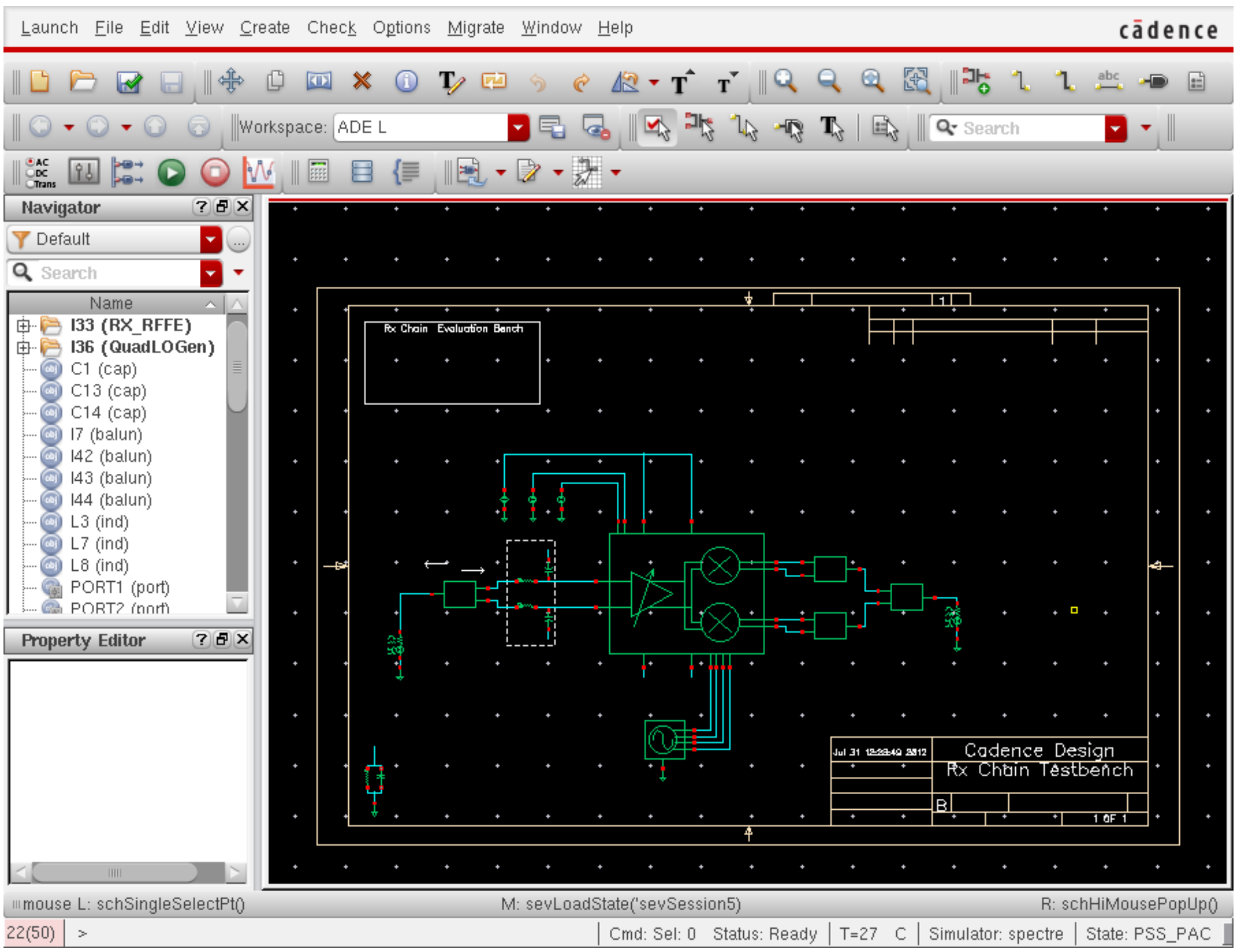
additionalParams

additionalParams is typically used for new features that are being beta tested. Keyword=value pairs are expected in this field.

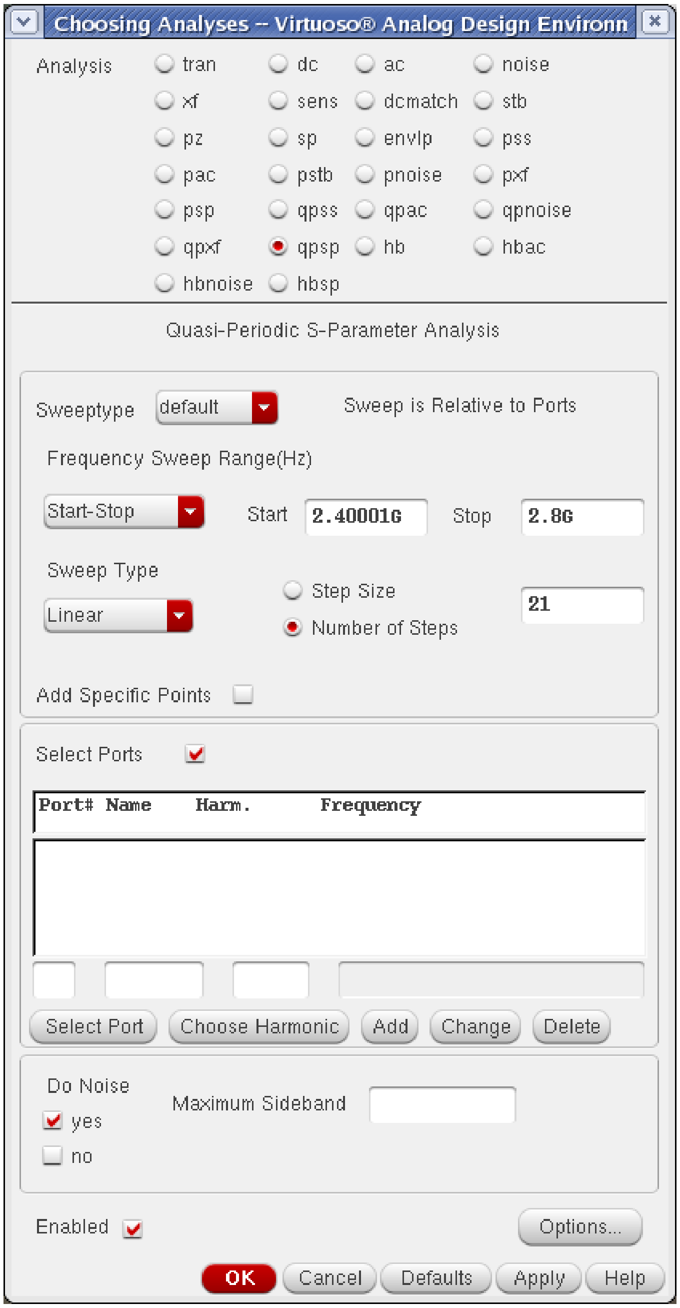
For more information about the other options, type `spectre -h qpsp` at the command prompt in a Unix shell window.

Examples

For this example, consider an LNA-Mixer circuit shown below.



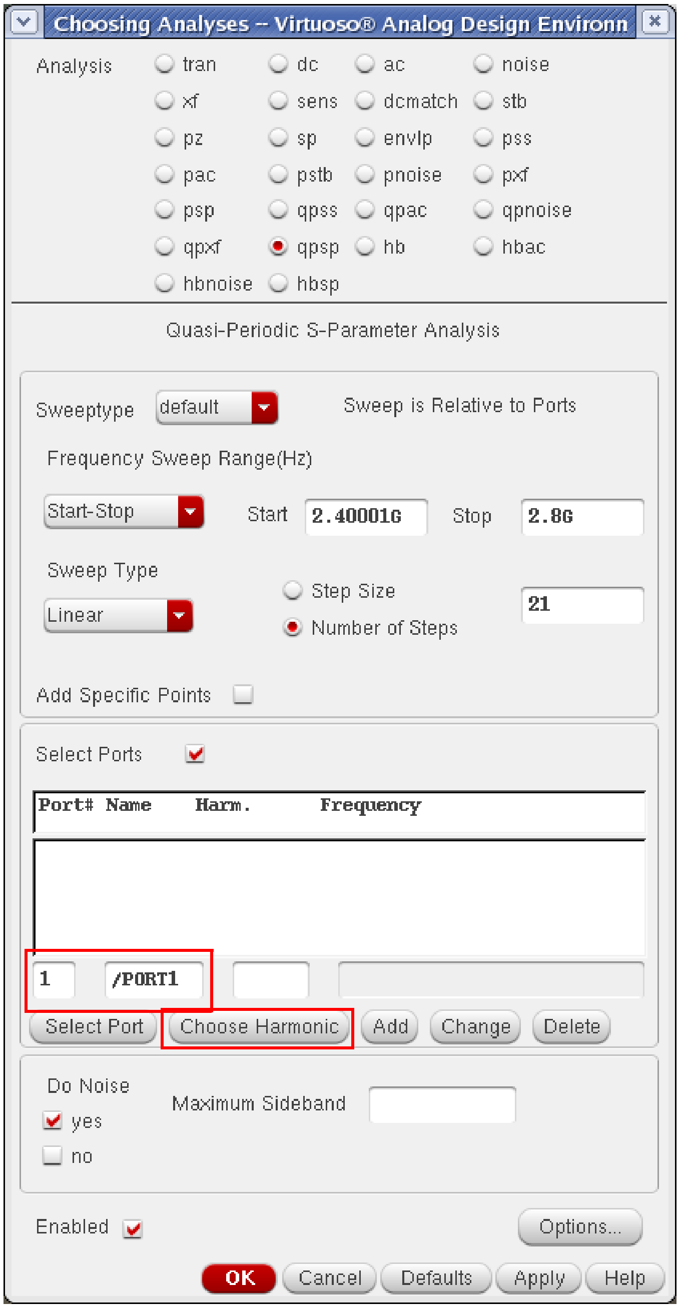
1. First, define either an input or output frequency range in the frequency sweep range fields. It is usually easier to define an input frequency range, as shown below for a 2.4GHz application.



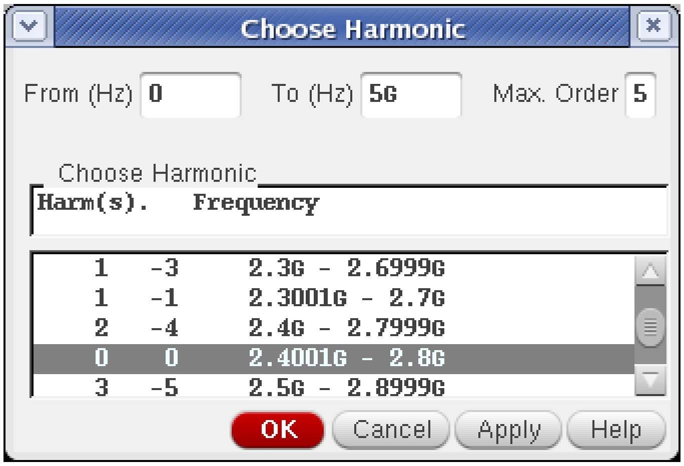
2. Linear and log sweeps are provided. It is usually better to select Linear or Log, and specify the number of steps instead of using automatic. Automatic will run 50 frequency points, which usually is not needed for most applications.

3. Now port and frequency definitions need to be made.

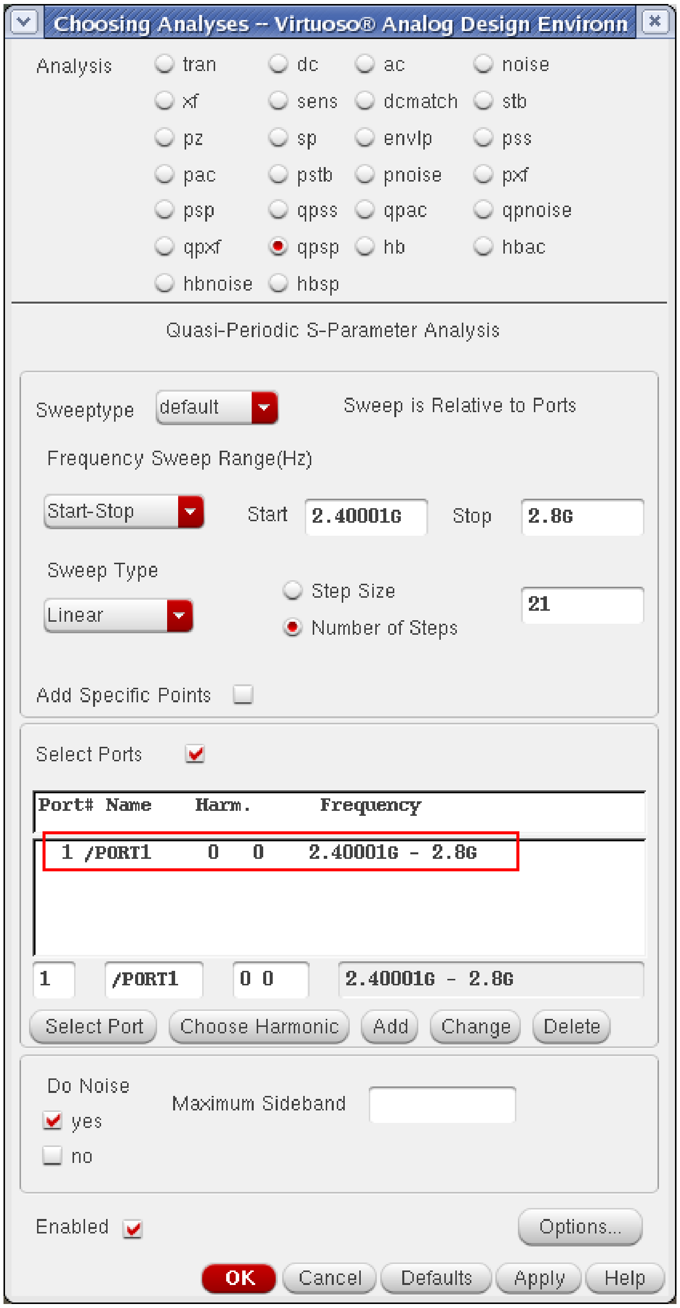
4. Type 1 in the *Port #* column field. Click *Select Port*, and select the input port.



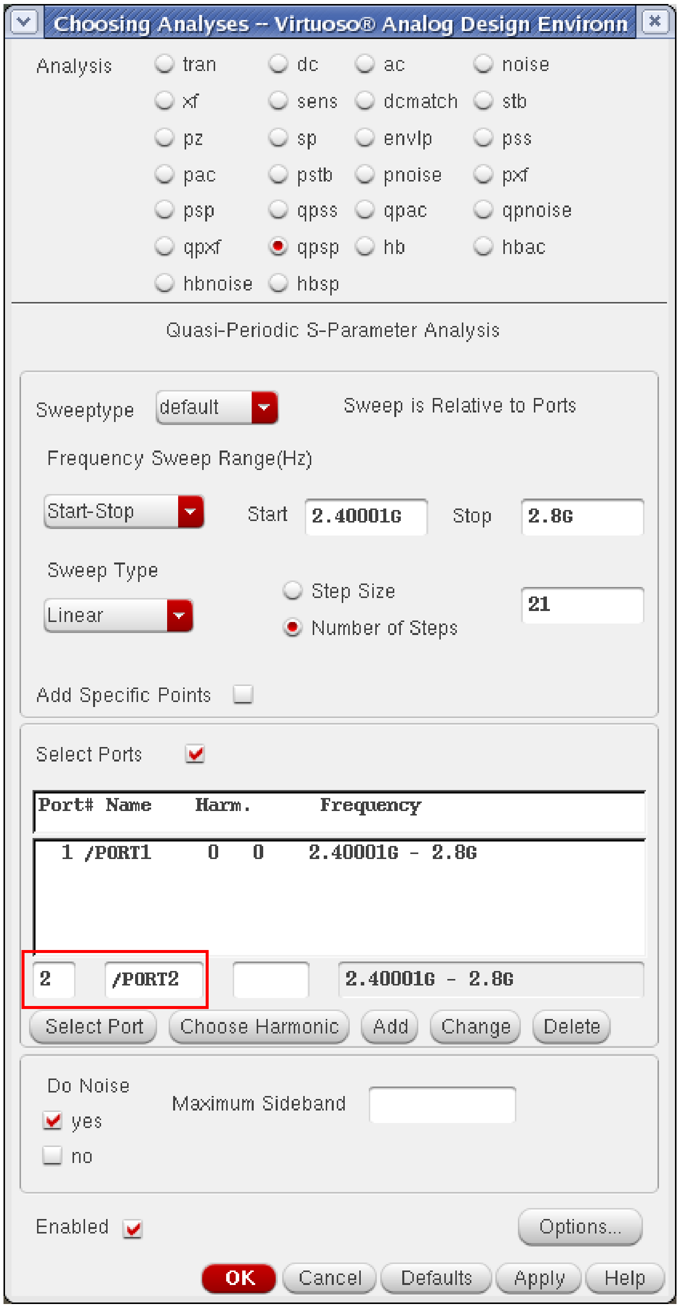
5. Click *Choose Harmonic*. In the window that appears, select the input frequency range. Next, click *OK*.



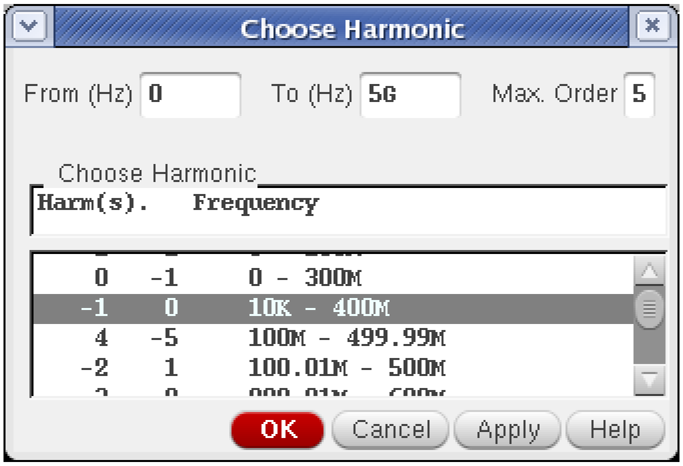
6. Click *Add*. The port-frequency combination is added to the list. By convention, port 1 sets the input frequency for qpsp.



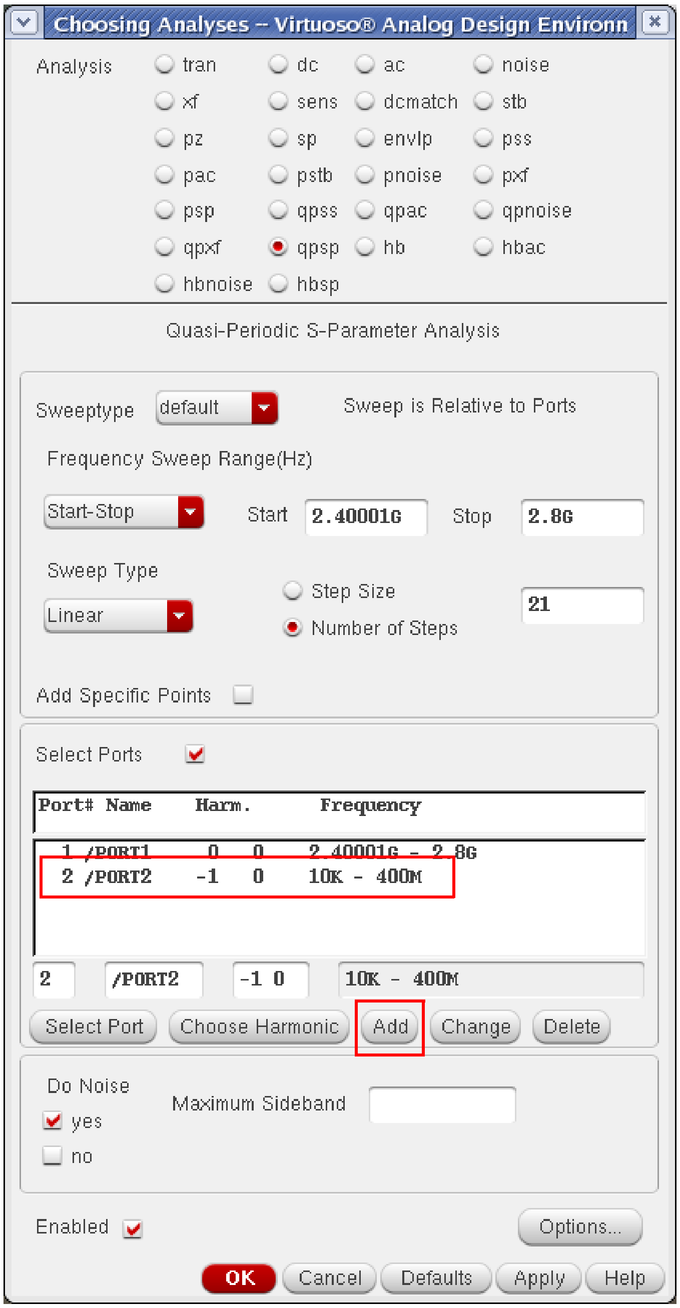
7. Now define the output port as port 2.



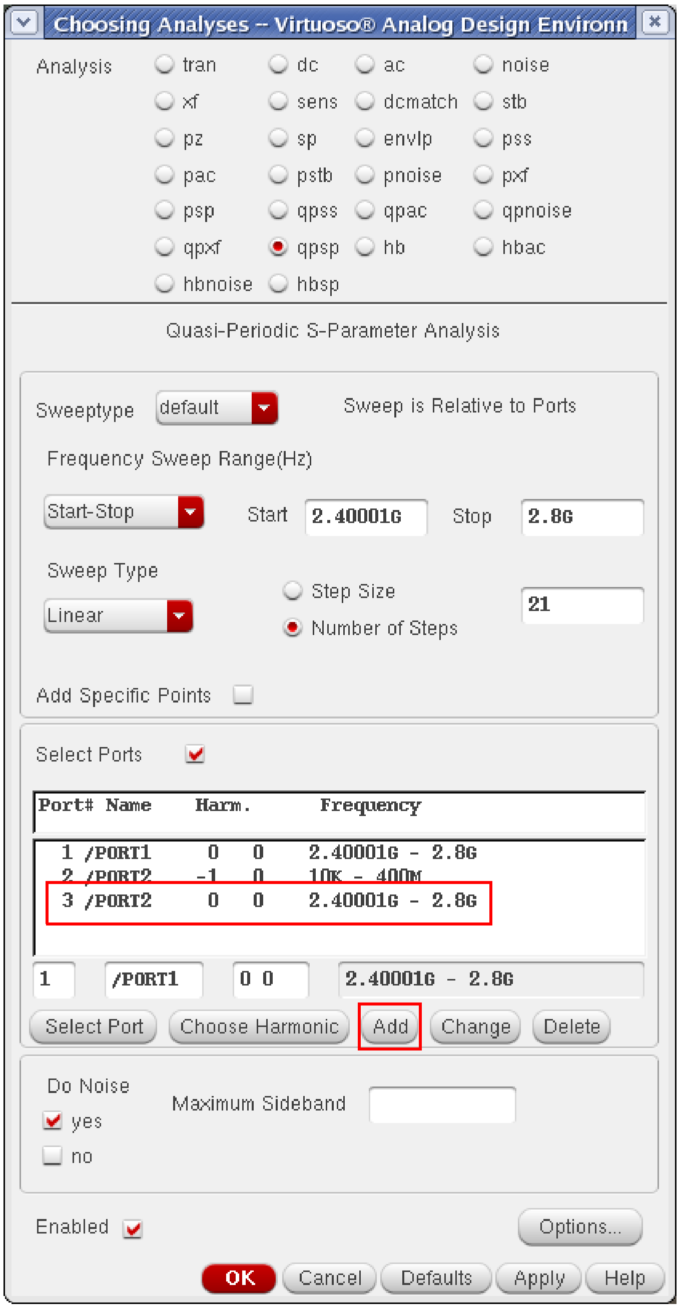
8. Select the output frequency range by clicking *Choose Harmonic*, and selecting the output frequency range as shown below.



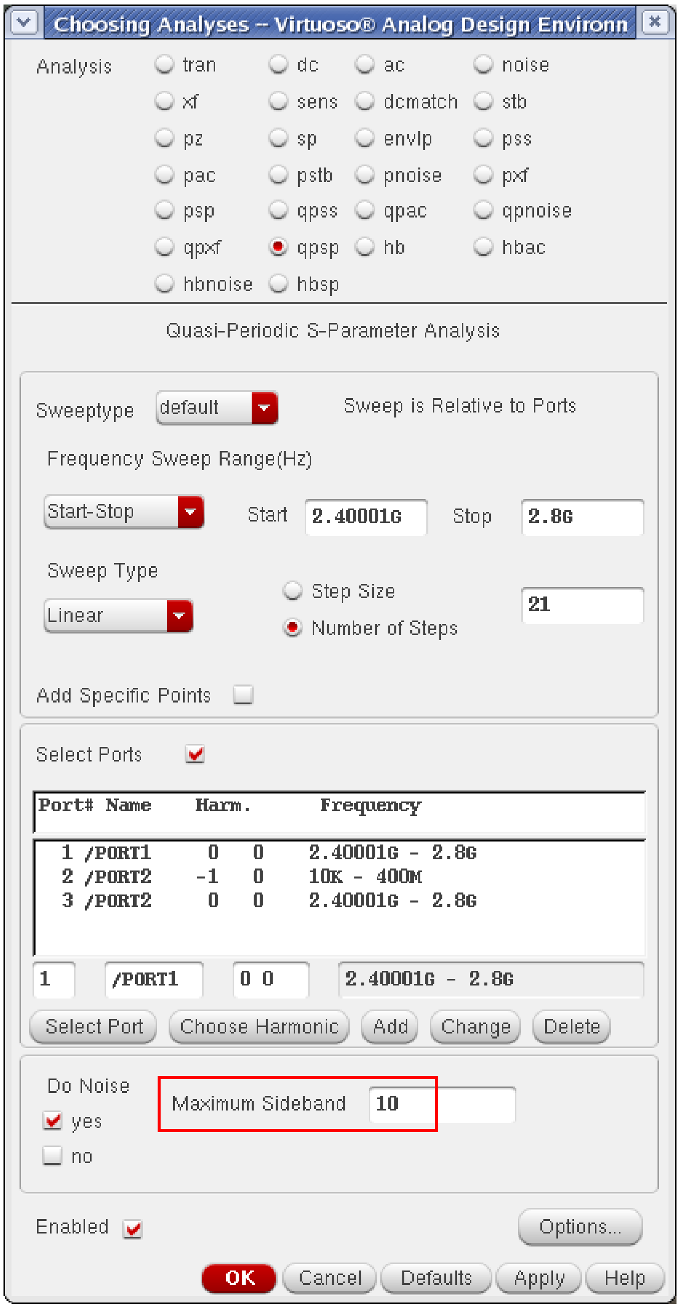
9. Next, click *Add*. This adds the port/frequency combination as port 2 in the list. By convention, port 2 sets the output frequency.



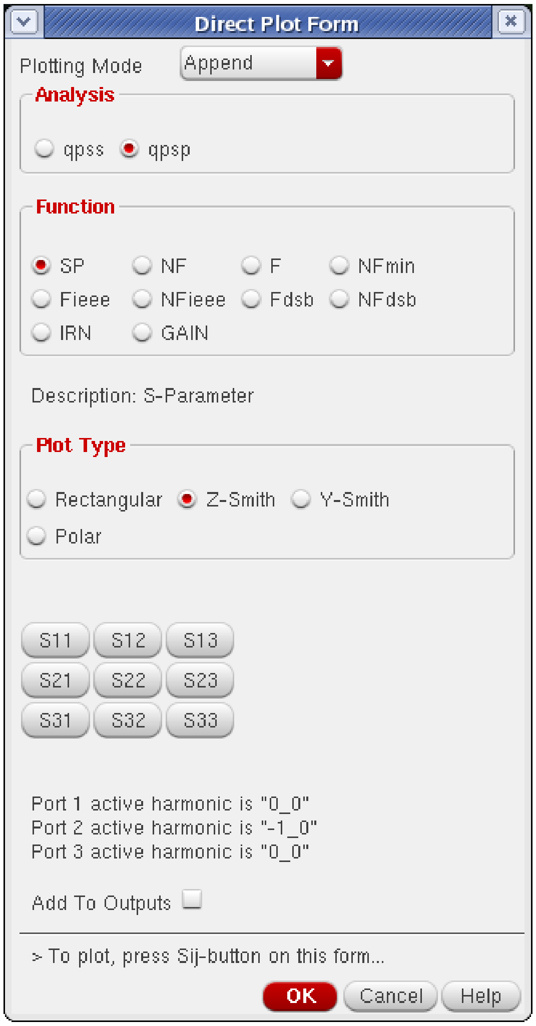
10. Repeat the process to define port 3 as the gain from RF to IF with no frequency translation. (The RF to IF isolation.)



11. If you want a noise simulation, define the maximum sideband term above. For more information, see the pnoise section in this chapter.



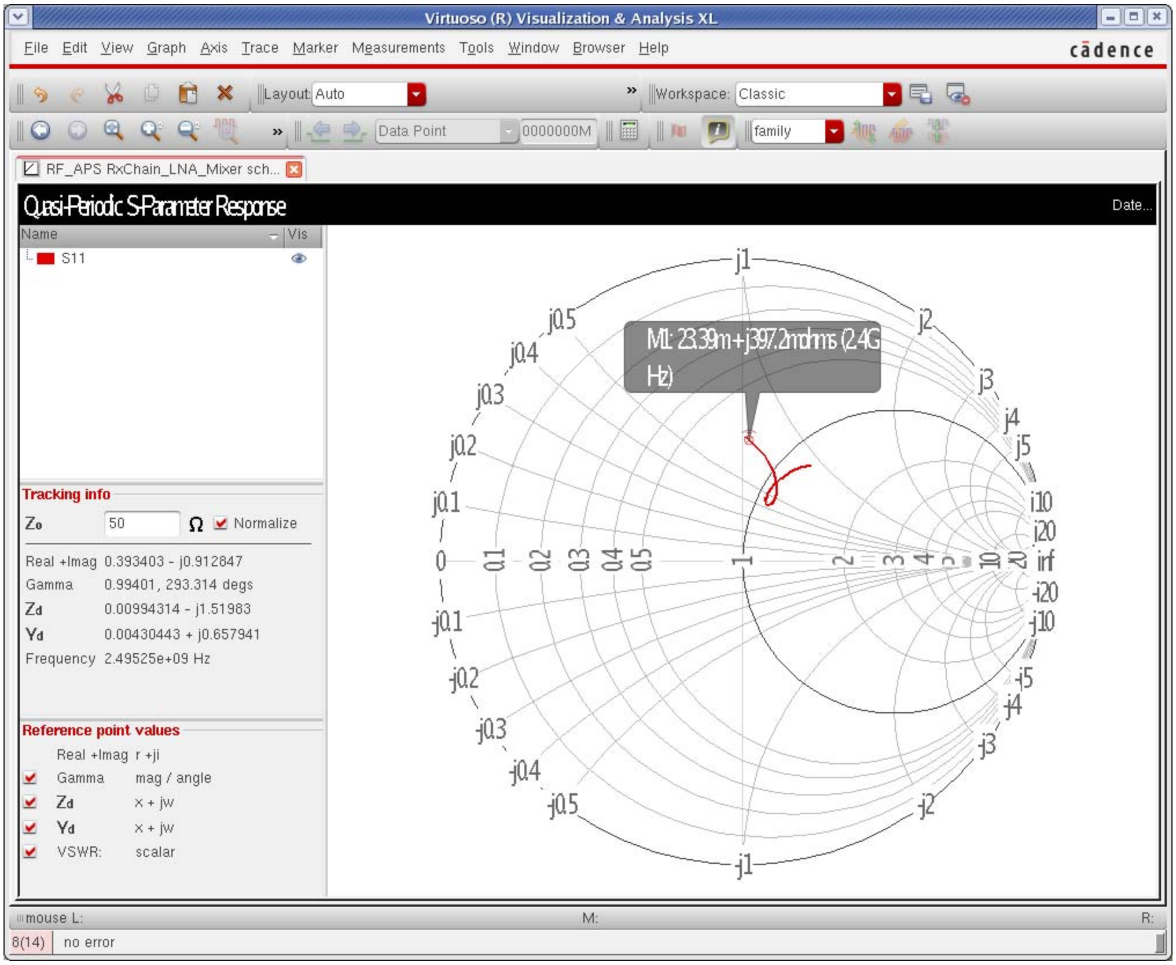
12. Now run the simulation. When the simulation completes, select *Results - Direct Plot - Main form* in the ADE window. The *Direct Plot Form* is displayed.



13. Note that only the S-Parameters and the noise calculations are available for plotting in the *Direct Plot Form*.

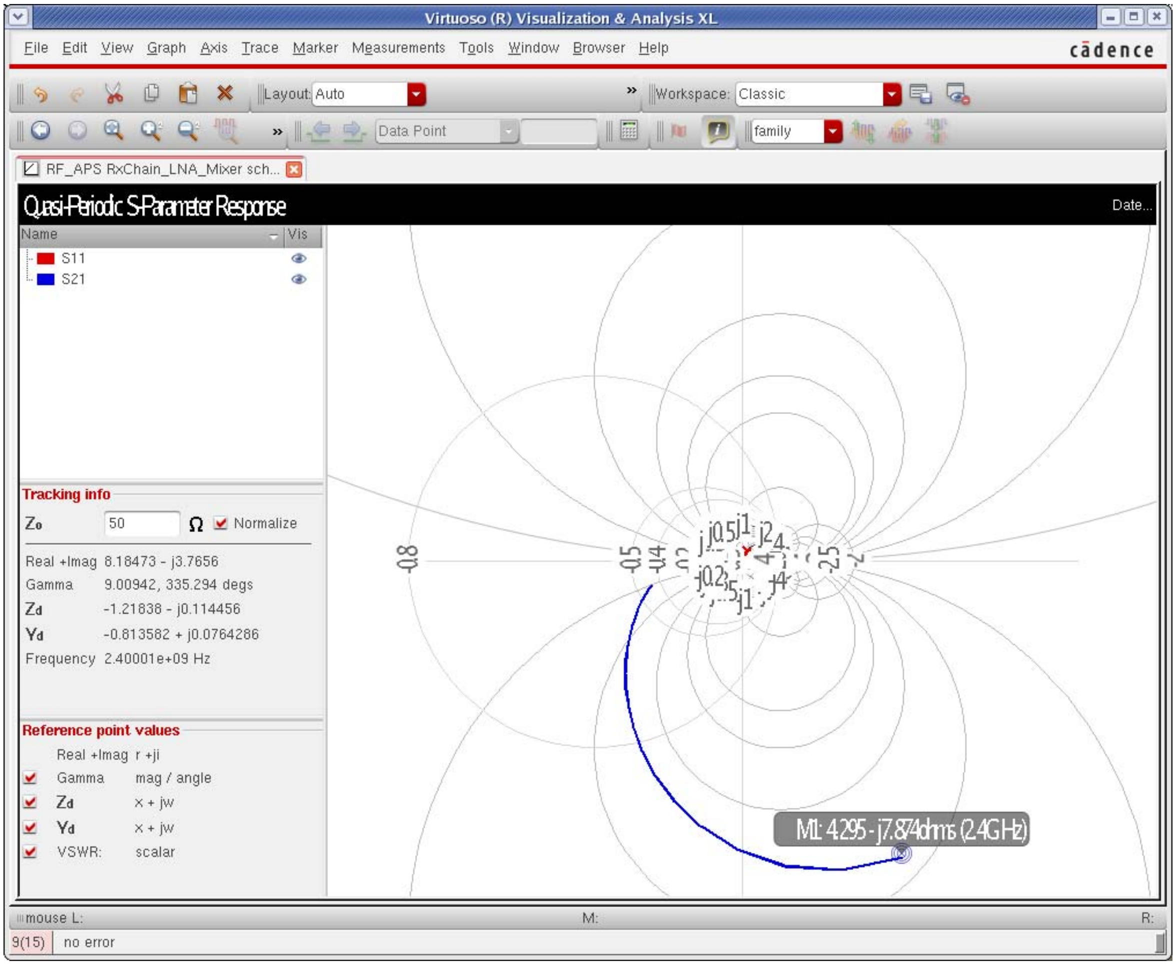
14. Select *Z-Smith* in the *Plot Type* section in the *Direct Plot Form*. Now plot S11 on the Smith Chart.

15. Z-Smith is shown for the S_{11} plot below. The marker is placed at 2.4GHz.

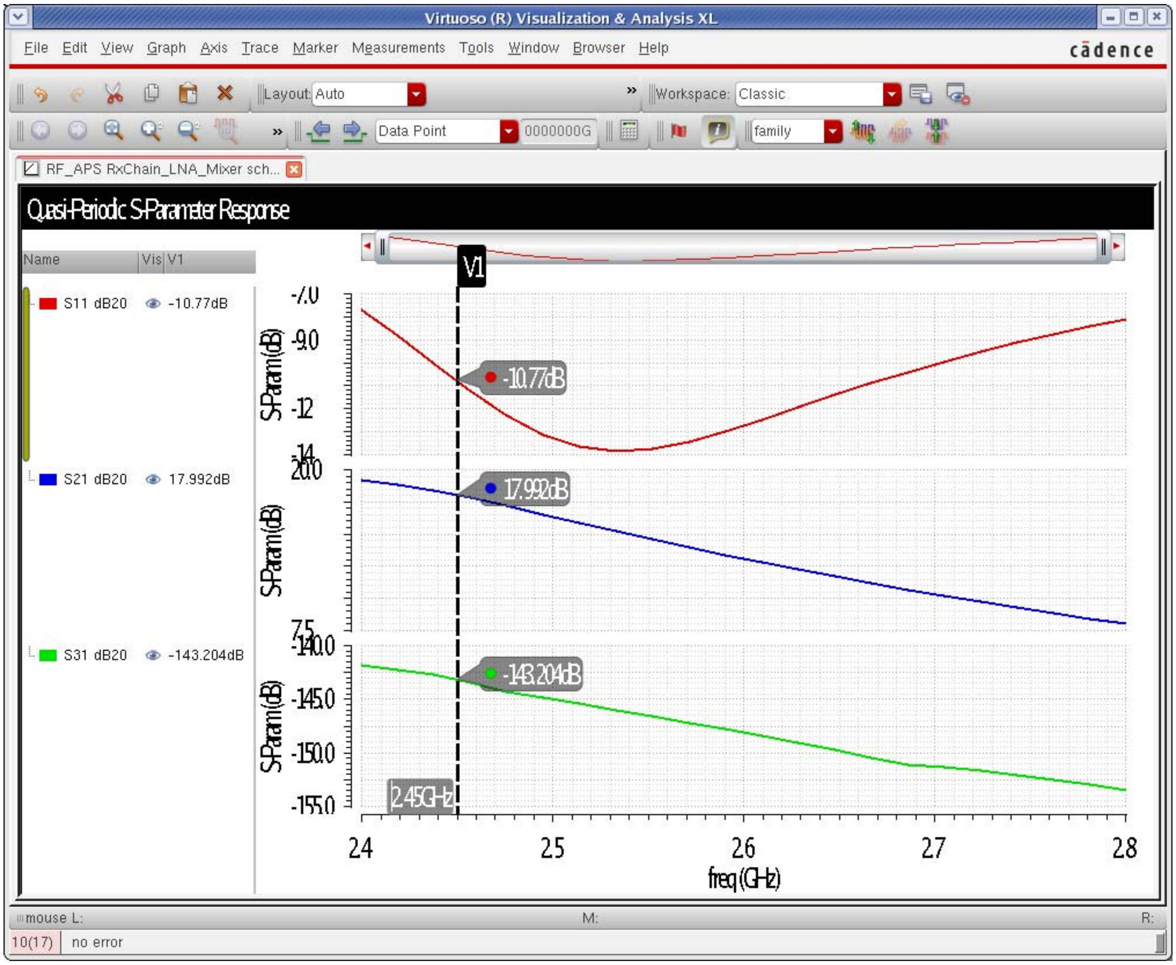


16. In the waveform tool, there are two zoom features that are helpful. Typing the f key fits the data to the screen. It will not necessarily display the unit circle, depending on the actual S-Parameters to be plotted. <Shift>-f fits the Smith Chart to the unit circle as shown above. This is useful for getting the big picture of the plot.

17. When S_{21} is plotted, the curve goes outside the unit circle. The small red trace in the center is the S_{11} plot from the previous graphic.



18. Rectangular plots are also available. Select *Rectangular* in the *Plot Type* section in the *Direct Plot Form*, as shown below.



6

Envelope (ENVLP) Analysis

Envelope analysis is used instead of a transient analysis when there are fast things and slow things in the same circuit. The simulation term for this condition is called multi-rate simulation. In the transient, a large number of cycles of the fast signal must be simulated in order to capture the slow behavior in the circuit. Instead of simulating every fast cycle as in the transient, envelope skips many cycles of the fast signal because the change is very slow. Although many cycles are skipped, the result is almost as accurate as transient analysis.

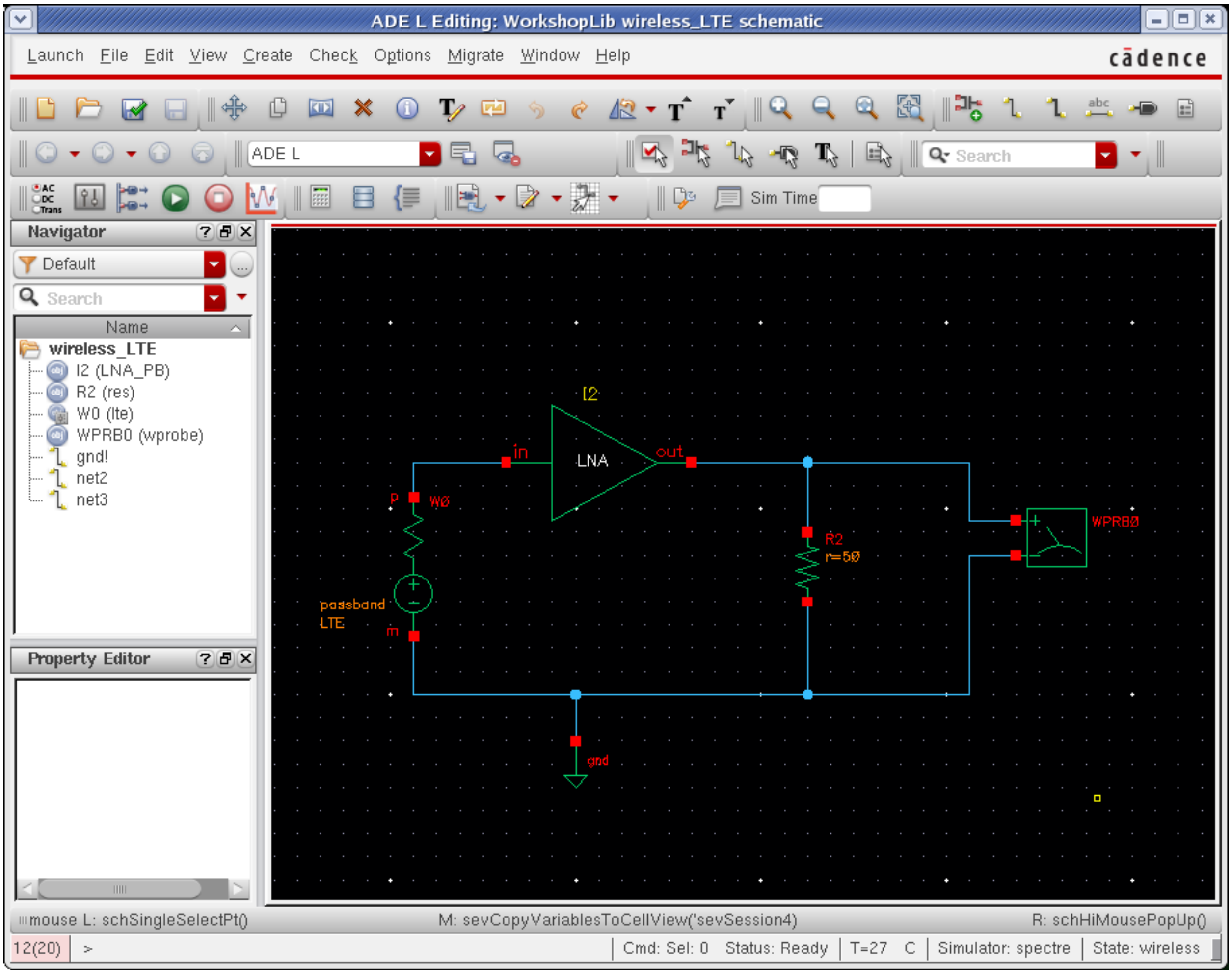
The primary application for envelope analysis is for digital modulation of a high frequency carrier, or calculating the startup waveform of a high-Q oscillator.

Envelope can solve for one cycle of the RF carrier either by solving for the waveform when shooting is selected, or by using harmonic balance when harmonic balance is selected. For very nonlinear systems like polar modulation, use shooting. For relatively more linear circuits like and I/Q modulator and or power amplifier, use harmonic balance. Harmonic balance has the advantage of having two fast envelope modes that can dramatically speed up simulation.

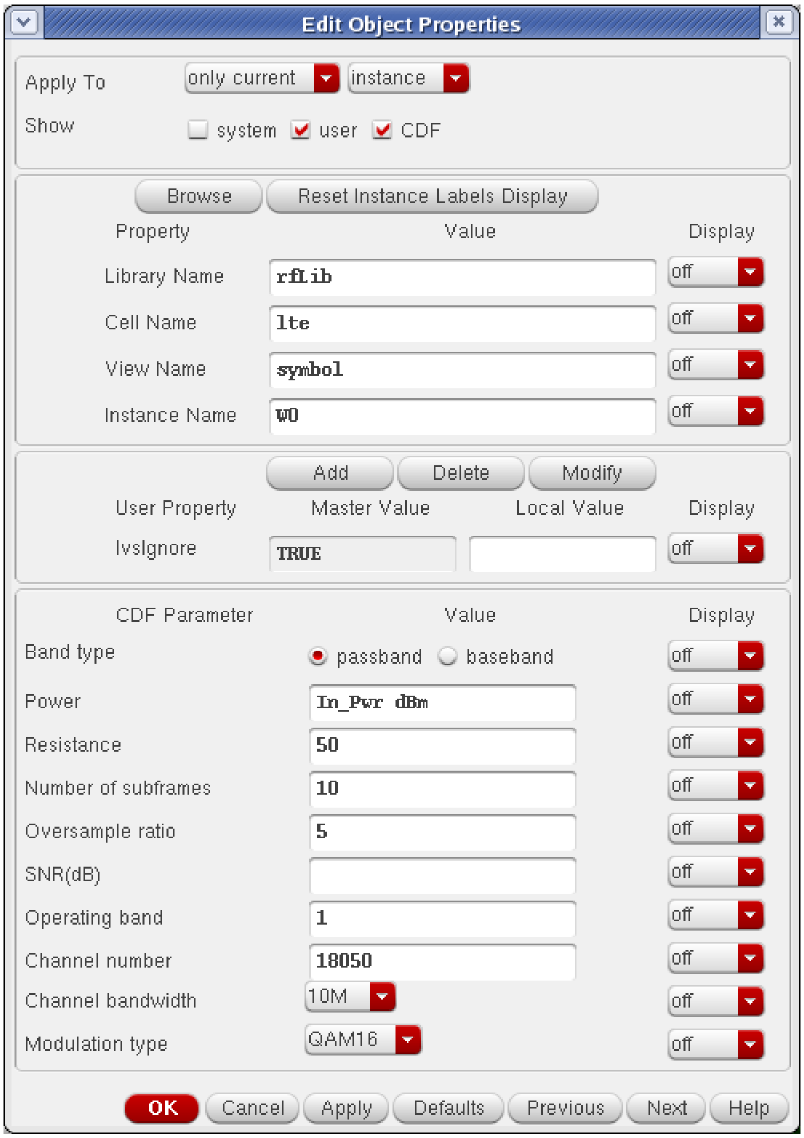
Harmonic balance envelope has two different algorithms. The first is to simulate everything at the transistor level. The second, Level1, characterizes the circuit at the start of the simulation, and builds a behavioral model that is then run instead of the circuit. Level1 is useful for circuits that do not have memory effects.

Example

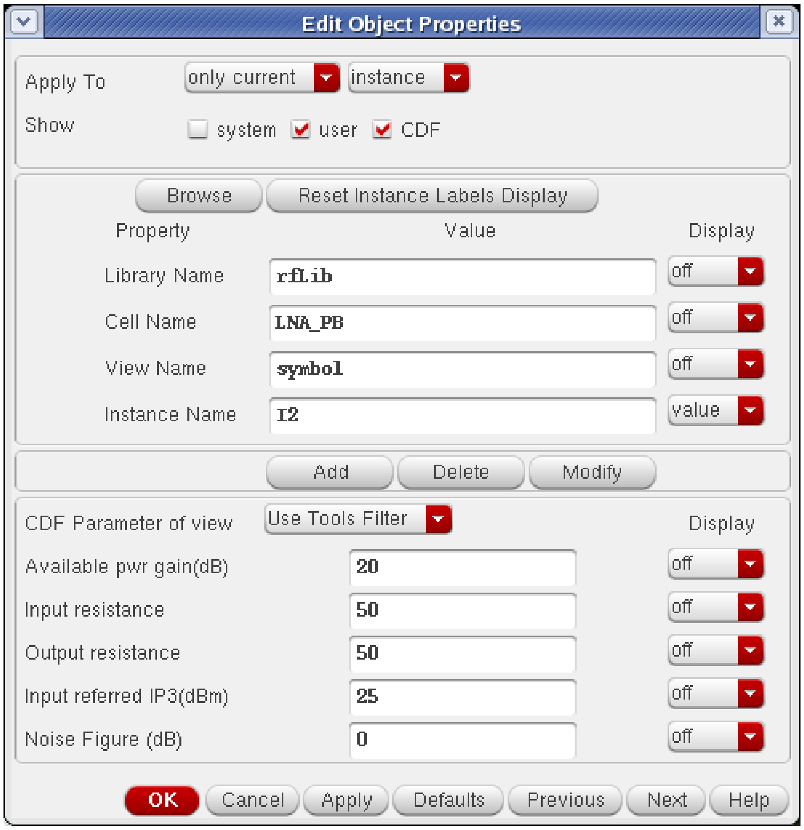
Consider a circuit with a wireless signal source, a behavioral PA, a wireless probe, and a 50 ohm resistor, as shown below. Multiple wireless probes are allowed. Connect a wireless probe at each point in the circuit you want to measure.



The wireless source from `rfLib` generates a fully compliant LTE signal..

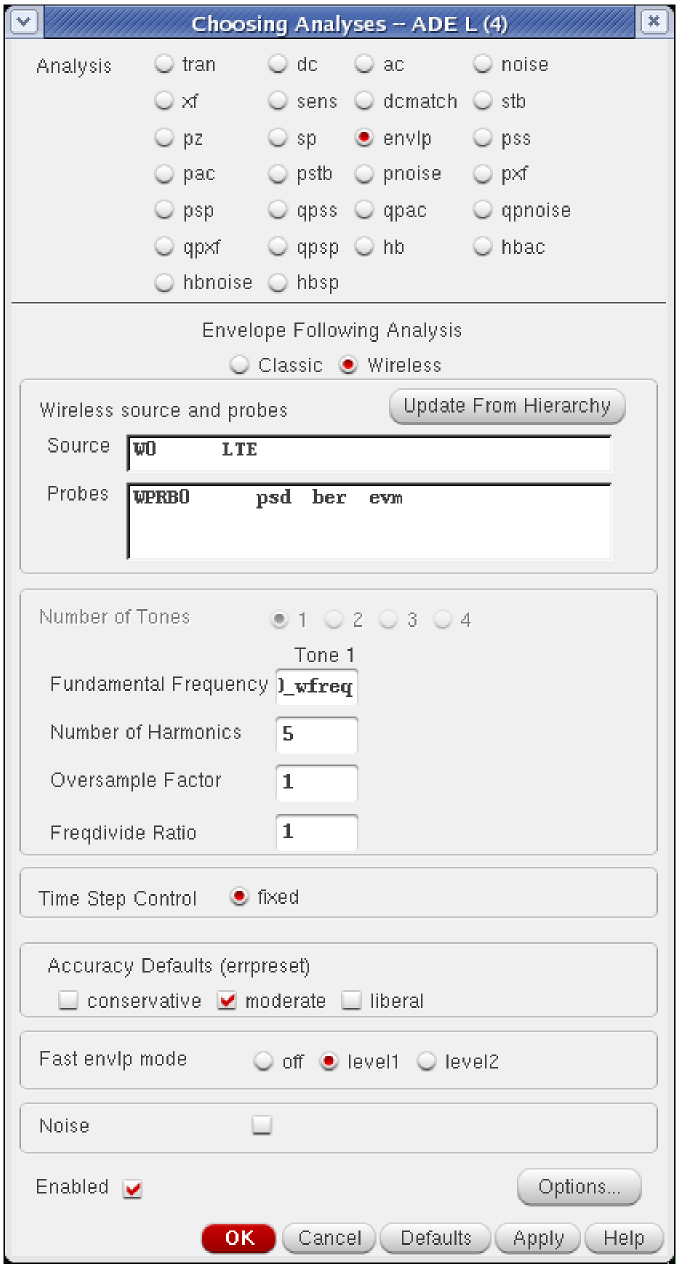


The behavioral PA has 20dB of gain, and is slightly nonlinear.

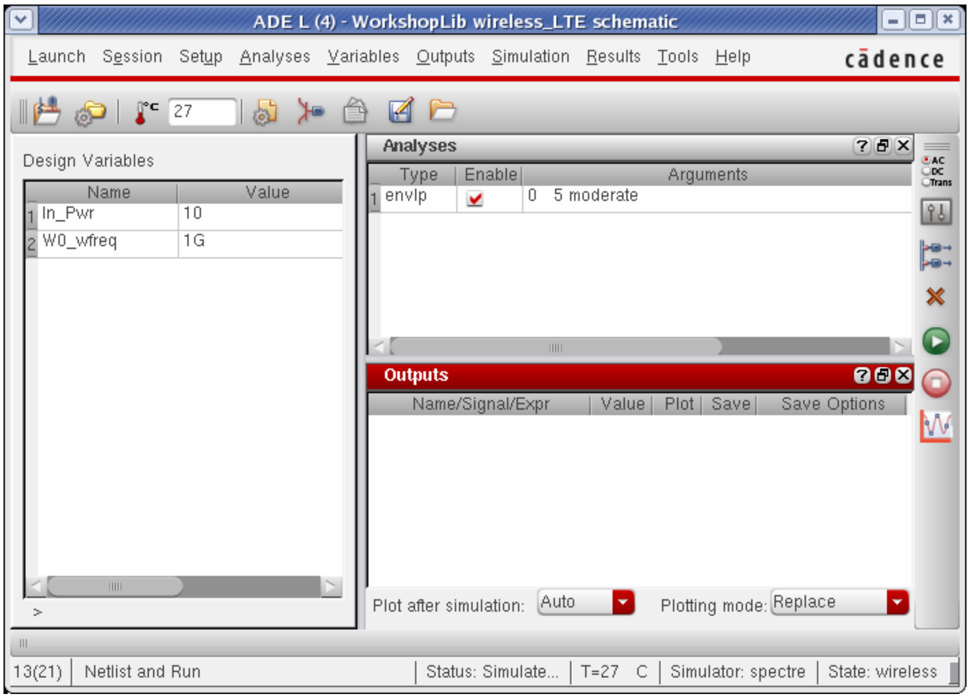


Now start ADE, and open the envelope *Choosing Analyses* form. The form is blank.

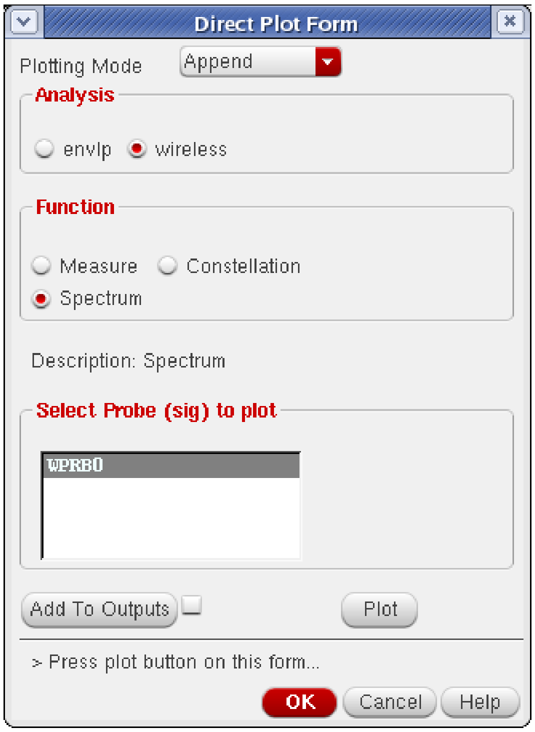
Next, set wireless mode, and type in the number of harmonics you want to simulate. Generally, `level1` fast envelope is used because of the speed advantage compared to transistor-level simulation. Both modes work with wireless.



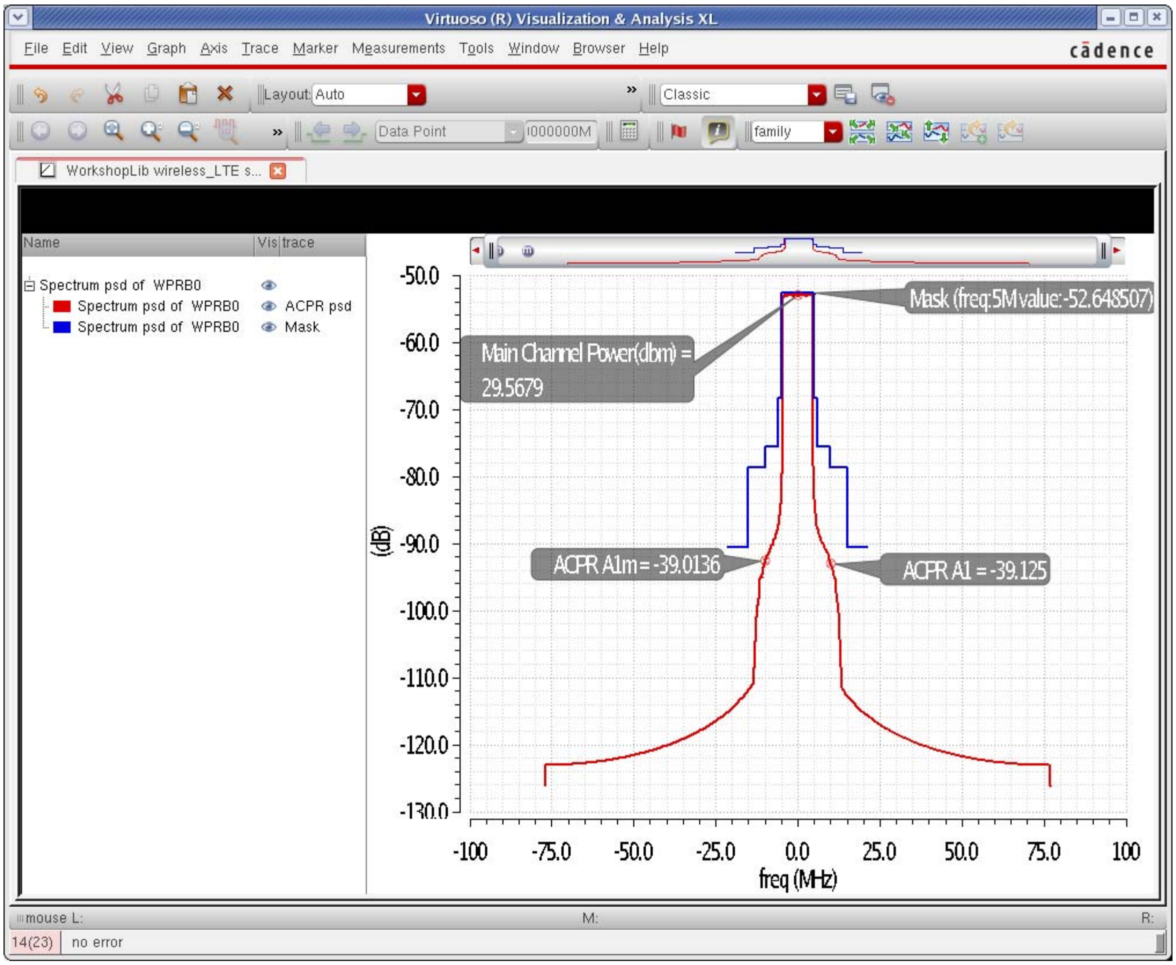
Now run the simulation from the ADE window. When you start the simulation for the first time, a variable `W0_wfreq` is introduced into the ADE variables list. This variable is used only when modulators are present in the design, and this variable should be used to set the LO frequency in the modulator. The actual frequency used in simulation is defined by the frequency range and channel number that is defined in the RF source.



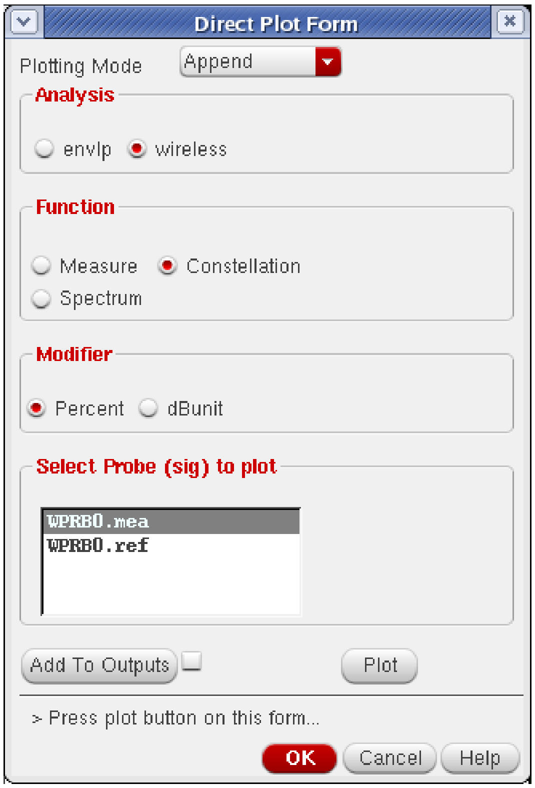
Now open the *Direct Plot Form*. All the common measurements are available from the *Direct Plot Form*. For all wireless plots, select the wireless analysis choice at the top of the form. To plot the spectrum, select *Spectrum*, and then select the probe you want to plot.



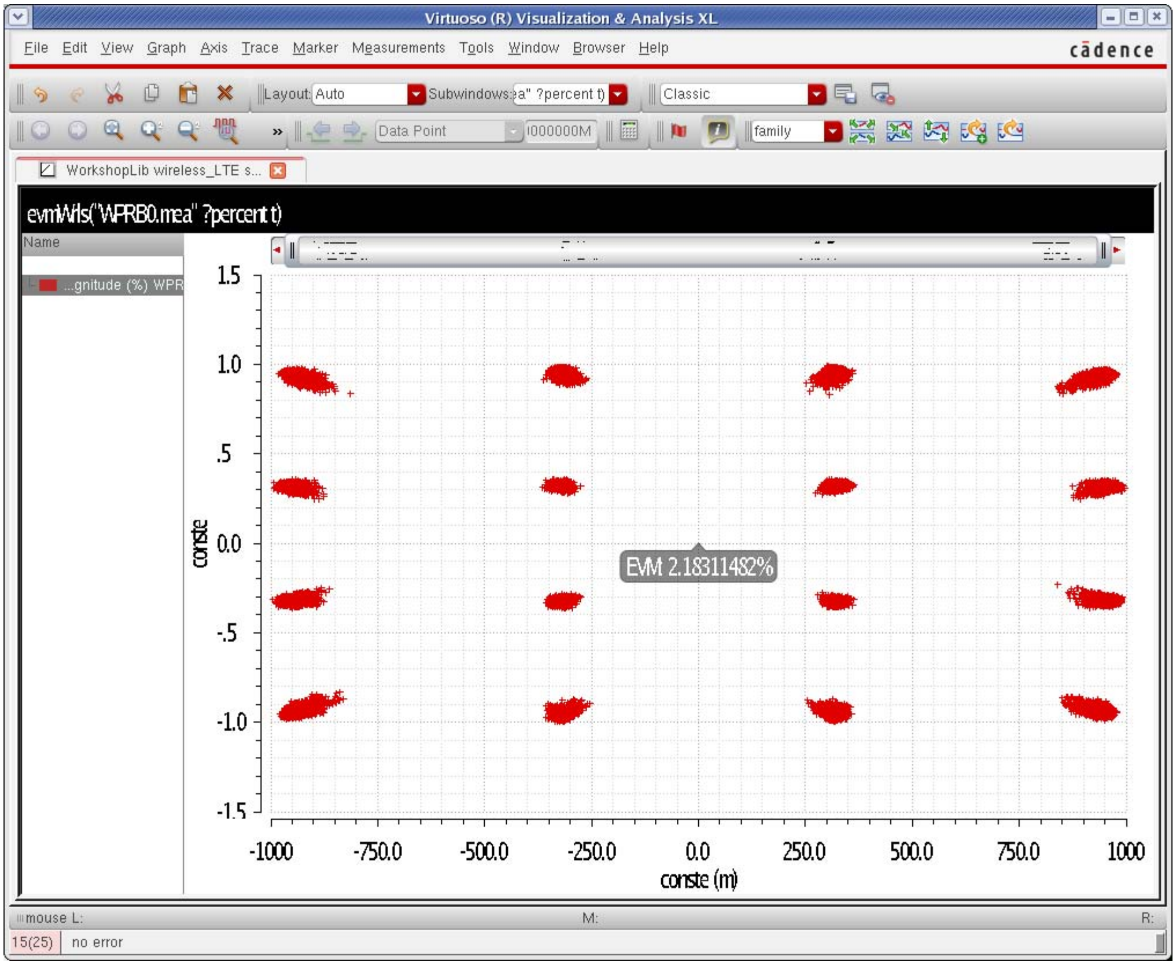
Click *Plot*, and the spectrum is plotted. The spectral mask for the appropriate standard is also plotted. The channel power and the high and low acpr measurements are displayed using markers.



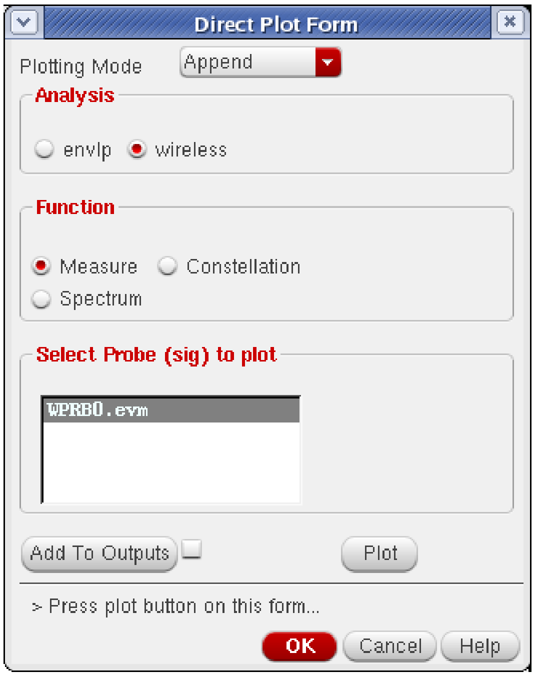
To plot the constellation diagram, select *Constellation*, and then either the *Percent* or dB value for the EVM measurement. Now select either the measured constellation, or the ideal constellation fitted to the actual datapoints, and click *Plot*.



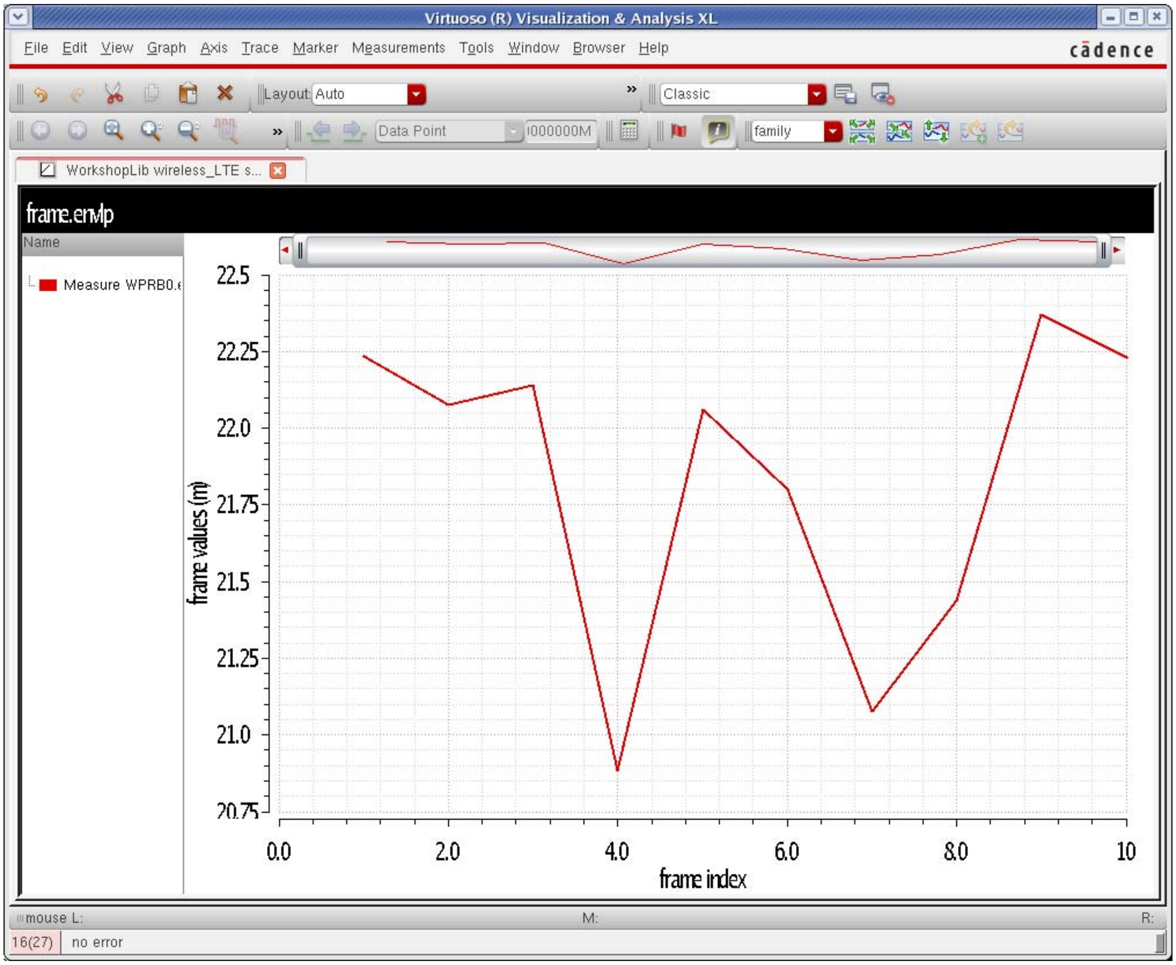
The constellation is displayed along with the EVM calculation. For the 802.11n standard, EVM for the data, pilot, or the weighted average of both is available.



To plot the subframe by subframe EVM, select *Measure*, and then select the <`probe name`>*.evm*.



The EVM subframe by subframe is displayed.



Envelope Principles

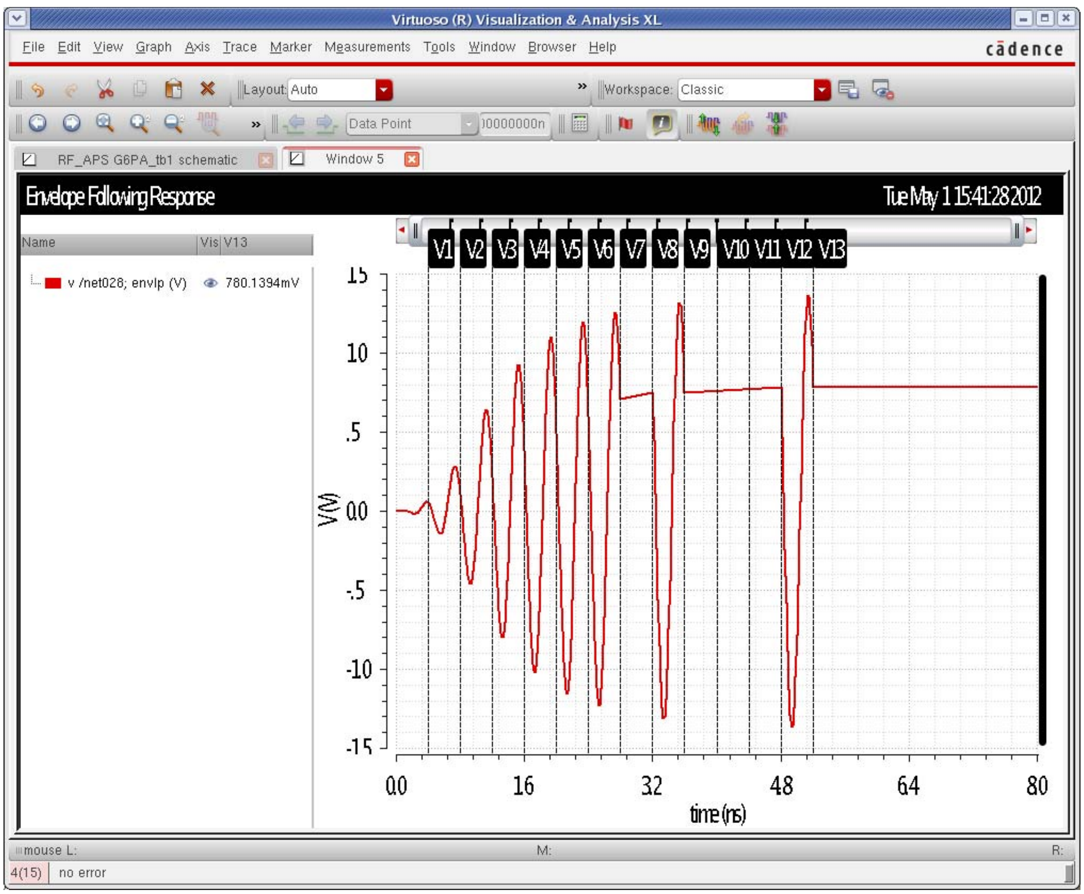
Harmonic Balance

Use harmonic balance if the circuit has sinusoidal waveforms and is fairly linear. This is true for circuits with sinusoidal waveforms operating below compression. If polar modulation is used where the circuit is operated in a very nonlinear mode, then shooting is suggested.

Basic Principles

Consider an amplifier that is being driven by an unmodulated source. Because of the finite bandwidth of the system, there is some startup behavior, and then the amplifier reaches steady-state.

The time-zero time point is the initial transient solution, which is a DC analysis that observes all the time-zero values for all the waveforms in the system.

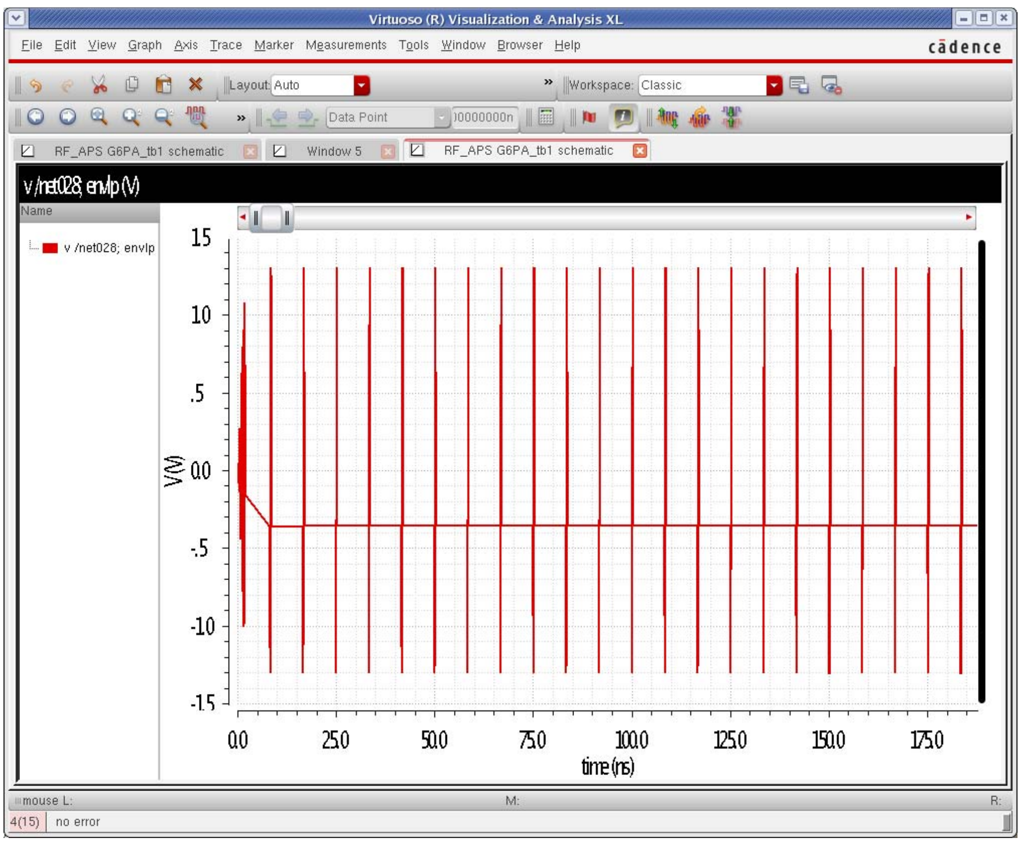


The waveform above is an actual starting waveform for harmonic balance. Vertical markers have been placed at 400nsec intervals, which correspond to one cycle of 2.5GHz.

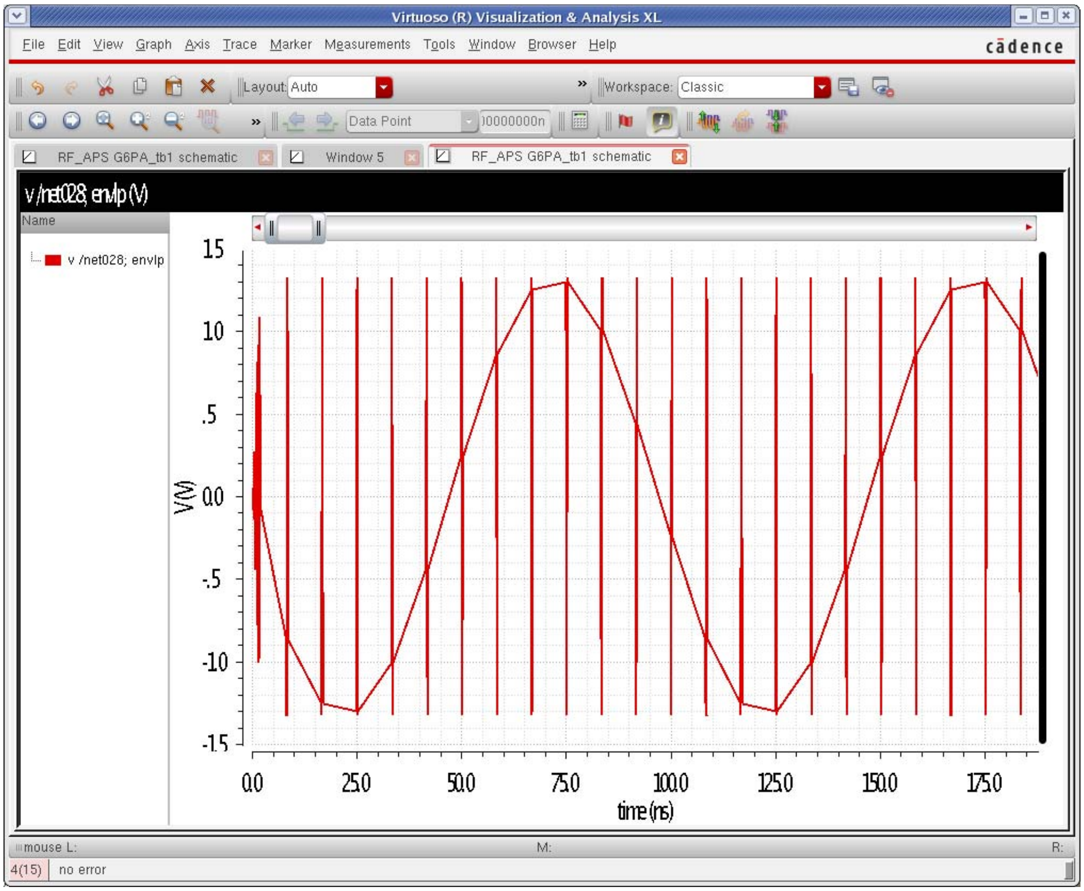
When the system becomes predictable with a low order polynomial, harmonic balance envelope begins to skip cycles. The number of cycles it skips depends on whether fixed or adaptive timestep control is selected. For fixed timestep control, the next strobing point is simulated. For adaptive timestep control, the number of cycles skipped depends on the curvature of the envelope. For an unmodulated sinusoid, conceptually, the cycle skipping could continue to grow to the end of the simulation because the same waveform repeats over and over again.

Note that for an accurate PSD and ACPR measurement, the data needs to be strobed at the time interval in the original I and Q modulation file. In harmonic balance, there does not need to be an integer number of skipped cycles.

If that delta-time was 8.3333333333nsec, to get an integer number of skipped cycles, the carrier frequency needs to be an integer multiple of 120MHz. Note that when the carrier is 2.4GHz, this is a multiple of 20. Because of the integer multiple, the envelope time-domain waveform shows a horizontal line between the cycles of simulation.

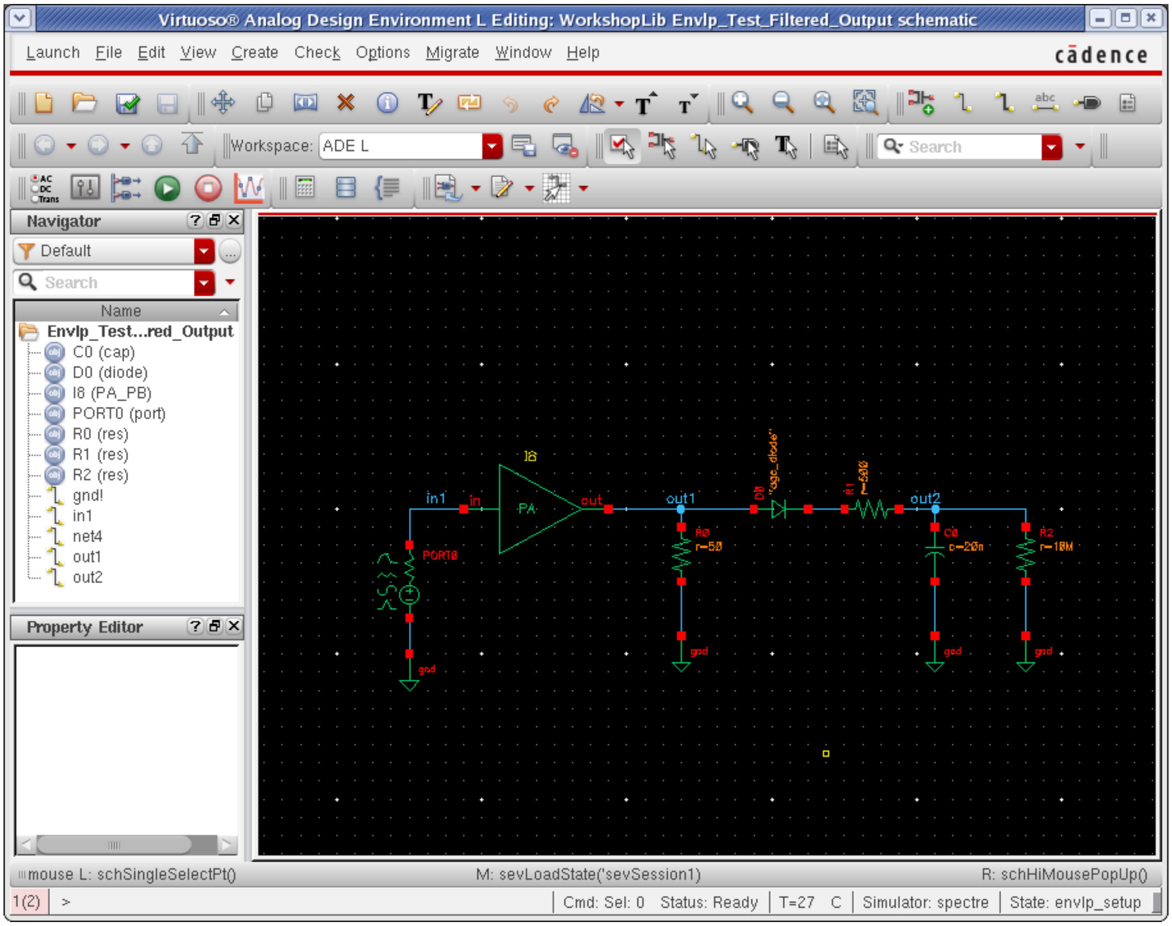


When the carrier is 2.41GHz, there is a small offset from the actual period of the carrier frequency. This causes a sinusoidal line to appear between the cycles that are simulated. This is because the periods do not exactly align.

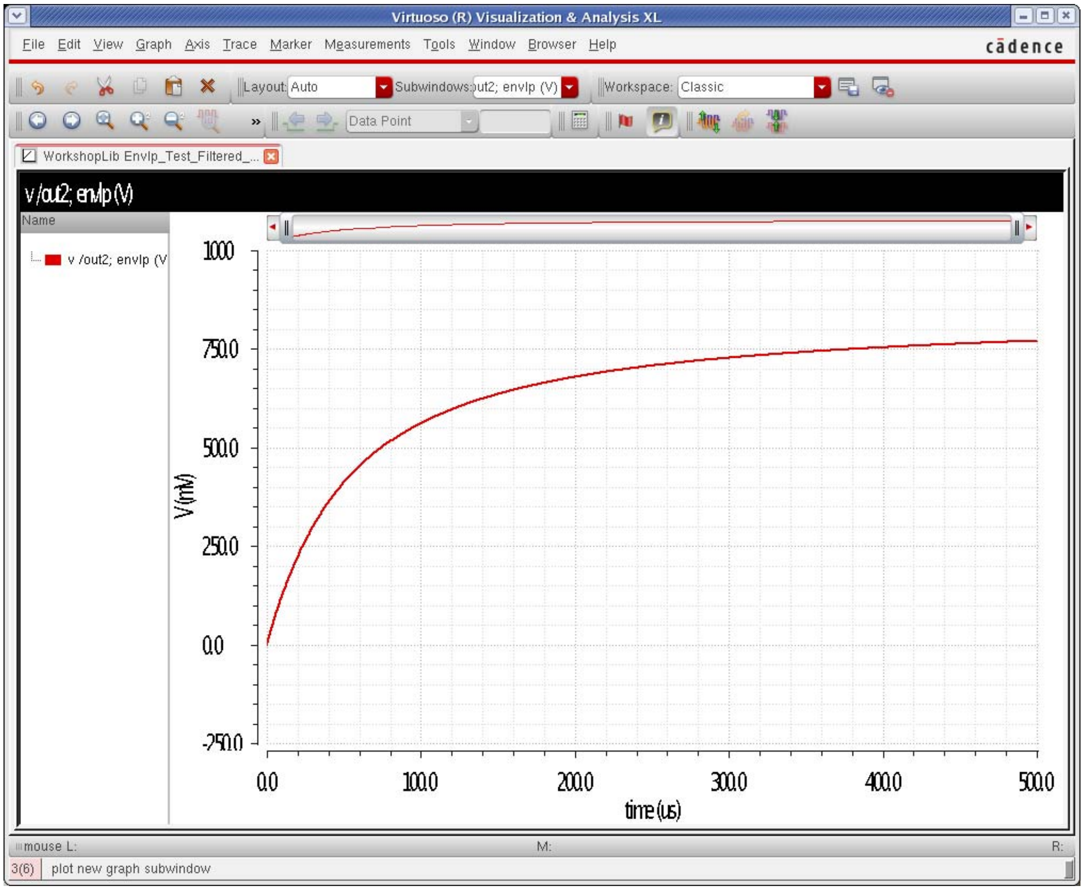


Note that harmonic balance envelope is not a series of steady-state harmonic balance simulations. It is a hybrid of transient and harmonic balance.

Consider an amplifier followed by a diode peak detector shown below.



At time zero, there is no signal, so the filtered output at the net *out2* starts at zero. The resistors and capacitors cause the signal to ramp up in an exponential fashion.



If the envelope simulation had been steady-state at each time point, the voltage at the net out2 would have been constant and non-zero from the start.

For systems that can have complex behavior, make sure that adaptive time step control is selected. This allows the number of skipped cycles to be dynamic along with the variations in the envelope signal. If this system had been an analog AGC circuit, the feedback might have resulted in overshooting and ringing. This behavior could accurately be determined by using adaptive timestep control.

Envelope-Transient Simulation

In envelope-transient analysis, a circuit is driven by one or more modulated carriers of the form:

$$x(t) = Re\left\{X(t)e^{jw_o t}\right\} = Re\left\{A(t)e^{j\theta(t)}e^{jw_o t}\right\} = Re\left\{[I(t) + jQ(t)]e^{jw_o t}\right\}$$

Subject to the modulated carrier excitation, circuit waveforms can be represented by sums of slowly modulated carriers harmonically related to w_0. Every voltage, charge, and current looks like:

$$v(t) = \sum_{n=-N}^{N} V(t)e^{jnw_0 t}$$

Envelope-transient analysis solves for the harmonic modulations (complex envelopes) $V_n(t)$ at each node. Because the envelope bandwidths are typically much lower than the carrier frequency, the sampling rates are correspondingly lower and the simulations more efficient than traditional transient analysis.

Under modulated carrier excitation, circuit equations take the form:

$$\frac{d}{dt}Q(V(t)) + jwQ(V(t)) + F(V(t)) = X(t)$$

where $V(t)$ is the vector of nodal harmonic envelopes. The remaining terms represent the quasi-static contributions to the envelope circuit equations, equivalent to a repeated solution of steady-state circuit equations by means of harmonic balance, one for each envelope time step t_i, where the sources are treated as sinusoids having a time-varying complex amplitude $X(t_i)$.

The dynamic term describes the contribution of reactive elements to the envelope circuit equations. Taking a linear capacitor as an example, it becomes:

$$C\frac{dV(t)}{dt}$$

This term vanishes unless the modulation is relatively wideband and/or the capacitance is relatively large.

Transistor Level V/s Fast Envelope

Transistor-level (envelope-transient) simulation solves for all of the circuit equations rigorously, taking dynamic effects into account. The cost of transistor-level envelope simulation is virtually the same as a sweep of M harmonic balance simulations, where M is the number of envelope time samples. Transistor-level envelope analysis is usually more efficient than transient simulation for modulated inputs, but it remains computationally very expensive when the number of the time samples is large, as is usually the case for today's communication standards.

The fast envelope method relies on two basic assumptions to accelerate simulation speed:

- The modulation is relatively slow and the contribution of the dynamic term mentioned earlier is small or negligible.
- There is a single-modulated carrier at the input (or, equivalently, a pair of baseband I/Q channels).

With those assumptions and considering the output signal as a modulated carrier of the form,

$$y(t) = Re\left\{Y(t)e^{jw0t}\right\}$$

the output envelope is a function of the input signal's amplitude and phase:

$$y(t) = f(A(t), \theta(t))$$

Two forms of f(A(t) , θ(t)) are observed in practice:

The Passband Model

In the passband model the nonlinear behavior is dependent on the magnitude of the input signal only.

$$f(A,\theta) = G(A)e^{j\theta}$$

`G(A)` describes the circuit's amplitude and phase compression characteristic. It is extracted efficiently by treating the input signal as a sinusoid and sweeping its amplitude over the expected range of the complex envelope's magnitude. At runtime, the output is obtained by table look-up and interpolation, which are very efficient operations. Since only a handful of points are required to build the look-up table, typically around 10, the runtime cost of the passband fast envelope model is equivalent to approximately 10 harmonic balance simulations. For a typical modulation source consisting of thousands, or tens of thousands of samples, the speed-up offered by fast envelope is much higher than the transistor-level methods.

The Baseband Model

The most frequent application of the passband model is in PA design where the input,

$$x(t) = Re\left\{X(t)e^{jw_o t}\right\} = Re\left\{A(t)e^{j\theta(t)}e^{jw_o t}\right\} = Re\left\{[I(t) + jQ(t)]e^{jw_o t}\right\}$$

may be viewed as a pair of I/Q channels upconverted by an ideal quadrature modulator. For most other situations, especially when the circuit contains transistor-level modulators, the output is a function of both the I and Q channels (or, equivalently, the modulated carrier's amplitude and phase), and therefore, requires a 2D sweep for proper characterization.

SpectreRF now includes a highly efficient and accurate baseband fast envelope model, which typically requires approximately 100 or fewer harmonic balance sweep points for proper characterization.

Fast envelope models are extracted from swept harmonic balance simulations performed over a range of the input envelope magnitude (for the passband model) and phase (for the baseband model). The number of sweep points has a significant impact on both the simulation accuracy and on speed of model extraction. In the present implementation, by default, the number of sweep points is set at 7 for the magnitude and (in baseband model only) 12 for the phase. This is a controllable parameter set by the options sweepmethod, Magnitude Points, and Phase Points in the envelope options form. Choosing Fine for sweepmethod will run 10 magnitude points and 12 phase points. Setting userdefined allows you to set both the number of magnitude and phase points for the characterization phase of the fast envelope extraction. Adaptive is a work in progress.

Power Scaling in Baseband Mode

In passband mode, the power specified by the `wsource` instance parameter *Power* represents the average power dissipated in a matched load. In baseband mode, *Power* represents the sum of the average power dissipated in matched load resistors in the I and Q channels. More precisely, assuming that the power available in the I and Q channels is the same, the sum of power in the I/Q loads is:

$$P = \frac{2}{RT}\int_0^T v^2(t)dt$$

Here, `v(t)` is the voltage across either of the load resistors. Then,

$$v_{RMS} = \sqrt{RP/2}$$

Using `P=10 mW` and `R=50` as examples, the rms voltage should be 500 mV.

WFreq

Notice that the fundamental frequency is specified as `W0_wfreq`. The variable, termed `wfreq`, is entered on your behalf directly from the wsource instance name. In general, its format is <*wsource instance name*>`_wfreq`.

`wfreq` is a reserved variable, which represents the carrier frequency of the source. Each standard has channel numbers and band numbers (if applicable) that define the carrier frequency that is currently in use.

Instead of calculating carrier frequencies manually, you can directly access the frequency computed by the source by means of `wfreq`.

`wfreq` is available in the analysis form but you can also use it to set values of the instance parameters.

When you click *OK*, ADE will automatically add a variable called <*wsource instance name*>`_wfreq` to the list of *Design Variables* and set it to `1G`. The wireless engine overrides it with the correct value internally and issues an info message in the log to report the actual value used during simulation. The actual value of the carrier frequency is reported in the Spectre output window.

When do I use Passband or Baseband?

Use the following simple rules to choose the model:

- If the input is a pair of baseband I/Q signals upconverted by a transistor-level (or otherwise non-ideal) modulator, use the baseband model.

- If the input is a modulated carrier (typically a case for PA modeling), use the passband model. An uncommon exception to this rule is the modeling of demodulators. Since the output of a demodulator depends both on the real and imaginary parts of the input signal, use the baseband method to model it.

Fast Envelope Noise

Optionally, in the presence of random noise, the output is modeled as:

$$y(t) = Re\left\{[Y(t) + N(t)]e^{jw0t}\right\}$$

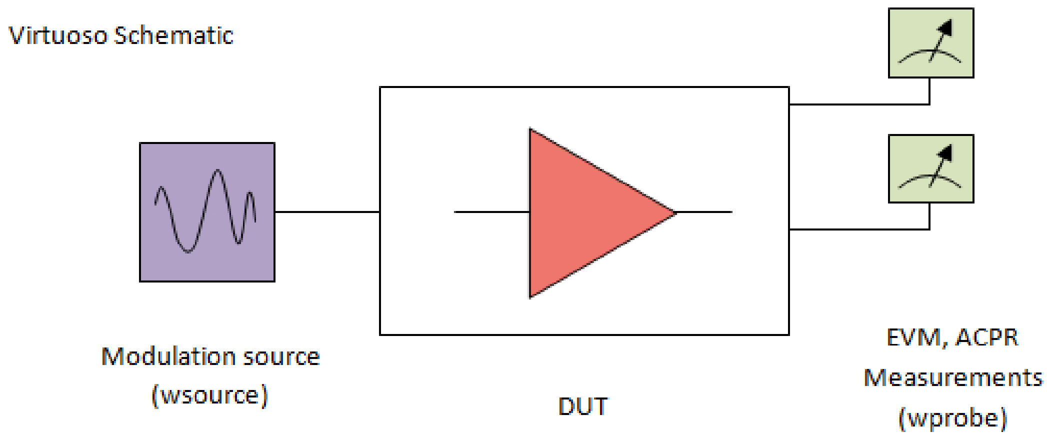
The random noise signal, `N(t)`, is extracted by one of the two methods: linear noise analysis around the DC operating point, or periodic noise analysis computed around a sinusoidal input (or the equivalent DC representation in the case of baseband input) given by:

$$\sqrt{\overline{A^2(t)}}e^{\overline{j\theta(t)}}$$

In both cases, the noise analysis is performed over the modulation bandwidth to extract the noise spectrum and calculate its complex-envelope representation. The periodic noise representation takes more time to extract than the linear model but is recommended as more rigorous and accurate than the linear approximation.

Wireless Mode in Envelope Analysis

Starting with the 13.1 release of SpectreRF, RFIC designers can evaluate system-level performance metrics in an integrated, automated, and easy-to-use simulation flow enabled by the new wireless analysis option.



In wireless analysis, standards-compliant modulation sources are applied to the DUT and measurement probes calculate system-level performance metrics. The sources and the probes are implemented by two new Spectre primitives: `wsource` and `wprobe`. Based on the familiar SpectreRF envelope-following algorithms, the wireless engine works with `wsource` and `wprobe` to automate the analysis setup and simulation parameters. The sources and the probes are standard-aware, resulting in a powerful and streamlined flow - though rich in standard support and complete in system-level simulation coverage, it is designed from the ground up for everyday use by RFIC designers.

Supported Standards

The 13.1 release supports LTE, ZigBee and 802.11n standards.

Wireless Analysis Vs. Traditional Envelope Following

Wireless analysis, based on traditional envelope-following analysis, has the advantage of automated set-up and post processing using the new `wsource` and `wprobe` components. Advanced modeling options, such as fast envelope, are supported as well. Wireless analysis produces, among other features, the traditional envlp dataset, making it compatible with all of the pre-existing Direct Plot capabilities and post-processing functionality.

Wireless analysis offers a number of additional benefits beyond the traditional envelope following:

- **Parametric modulation sources**. Traditional envelope-following relies on generic PWL sources of fixed frame structure, sampling rate, bit patterns, and modulation options. In contrast, wsources are fully parametrized according to the specified communications standard.
- **Automated simulation setup and control**. Wireless analysis works with standards-aware wsources, which help determine all of the relevant simulation parameters – sampling rates, stop times, strobe options, and carrier frequencies, as dictated by the communications standard. A key benefit of automated simulation control is that the designer is now less likely to make common setup mistakes which affect simulation and measurement accuracy. Wireless analysis is fully supported within ADE-L behind an easy-to-use and familiar-looking analysis setup form.
- **Efficient post-processing**. Most of the wireless analysis post processing is performed at runtime by wprobes. Much like the sources, wprobes are aware of the communications standard specifications. They output all of the standard-specific system-level metrics, such as EVM, BER, and spectrum masks directly to the output database. System-level metrics are available directly from the Direct Plot form, without manual post processing and in compliance with the modulation format and communications standard.

Limitation of Wireless Analysis

Wireless analysis works in conjunction with Cadence-supplied `wsources` and `wprobes`. Custom PWL sources are not supported in this mode. Traditional envelope-following can still be used with custom PWL sources as a fully supported simulation option.

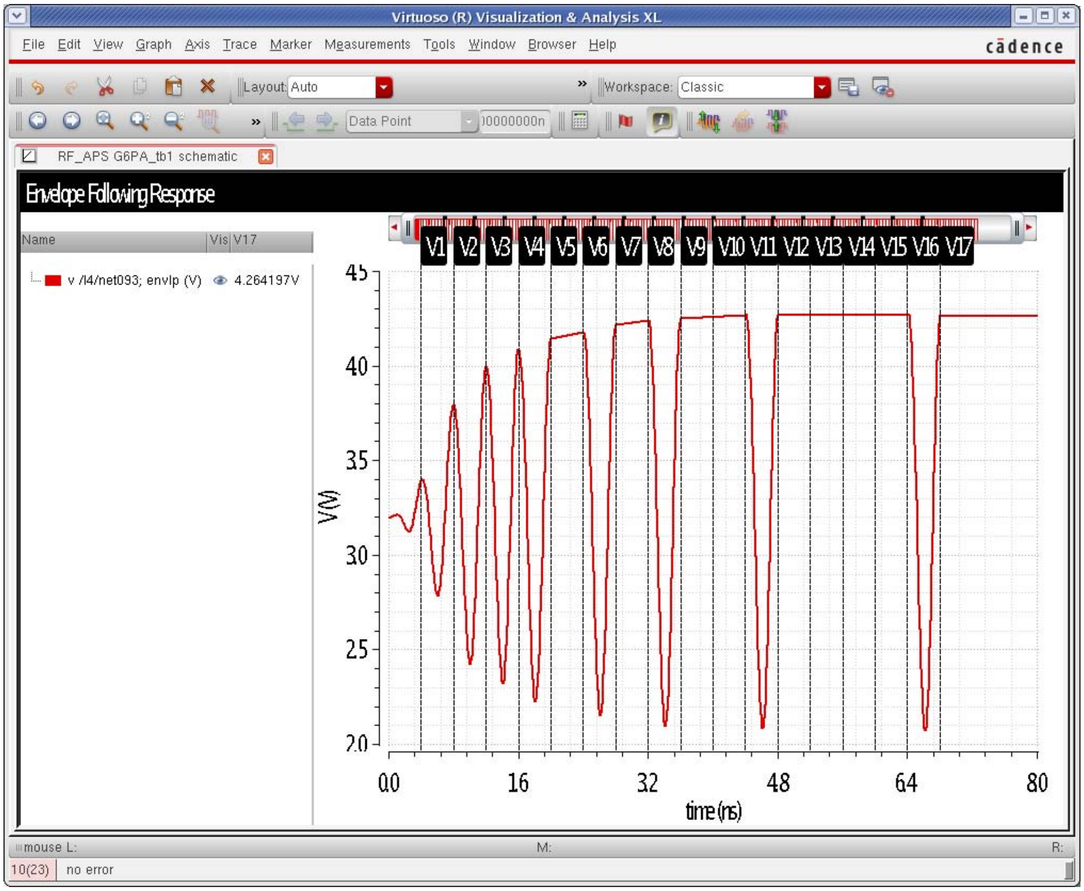
Shooting

Wireless mode is not available when shooting is selected.

Shooting is suggested if polar modulation is used where the circuit is operated in a nonlinear mode. Use harmonic balance if the circuit has sinusoidal waveforms and is fairly linear. This is true for circuits with sinusoidal waveforms operating below compression.

Consider an amplifier that is being driven by an unmodulated source. Because of the finite bandwidth of the system, there is some startup behavior, and then the amplifier reaches steady-state.

The time-zero time point is the initial transient solution, which is a DC analysis that observes all the time-zero values for all the waveforms in the system. At this point, the only thing that is known is the time-zero value, so the projected ending point of the first period is the starting point.



The waveform above is an actual starting waveform for shooting. Vertical markers have been placed at 400nsec intervals, which correspond to one cycle of 2.5GHz.

Note that at the end of the first period, the ending point misses the starting point by a lot. This error between the predicted and actual ending point is used to control the simulation. In this case, the error is large, so the transient continues for another period. The beginning and ending points of the first cycle allow a prediction of a ramp as the ending point for the second cycle.

When the system becomes predictable with a low order polynomial, shooting envelope begins to skip cycles. First, it will skip one period, which is done twice in a row in the above example. Then it skips 2 periods, and then it skips 4 periods. For an unmodulated sinusoid, conceptually, the cycle doubling could continue to the end of the simulation because the same waveform repeats over and over again.

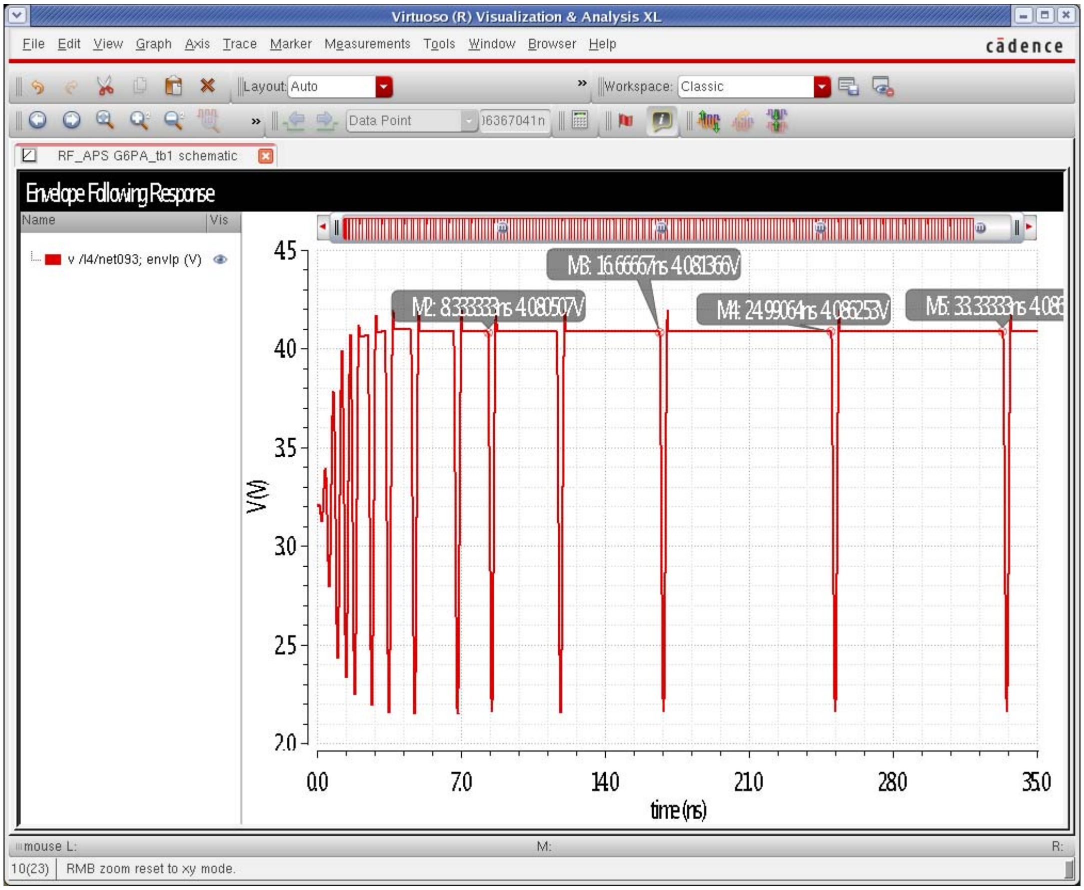
Shooting has the limitation that only an integer number of cycles can be skipped. This is because the starting and ending voltages need to be predictable, which happens only when an integer number of cycles is skipped.

When digital modulation is present, the number of skipped cycles depends on the curvature of the envelope. When the envelope has little curvature, many cycles can be skipped. When the envelope has a large curvature, not many cycles can be skipped.

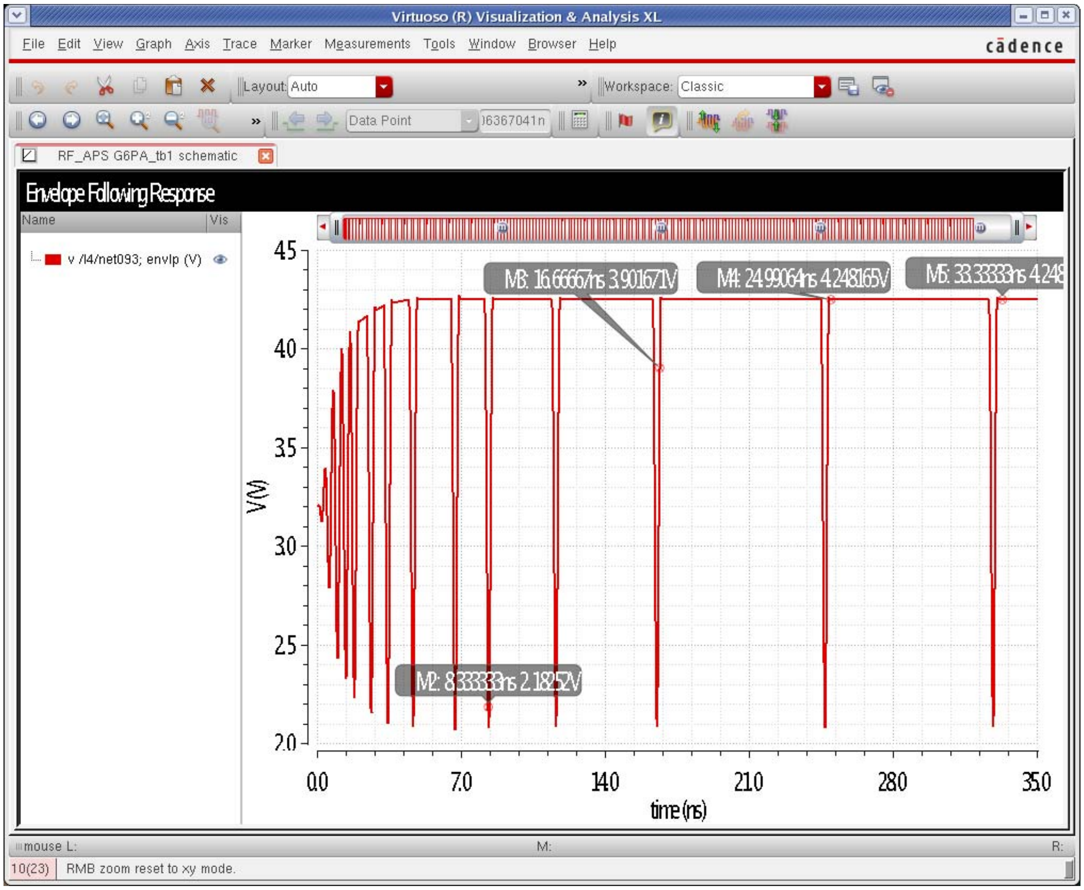
In order to maintain accuracy, adaptive timestep needs to be used with shooting to allow the envelope algorithm to maintain the accuracy of the simulation.

Note that for an accurate PSD and ACPR measurement, the data needs to be strobed at the time interval in the original I and Q modulation file. For maximum accuracy, the carrier frequency needs to be an integer multiple of the reciprocal of the delta-T in the I and Q modulation file.

If that delta-time was 8.3333333333nsec, the carrier needs to be an integer multiple of 120MHz. Note that when the carrier is 2.4GHz, this is a multiple of 20. The sampled carrier cycles are exactly aligned with the data in the I and Q modulation file.

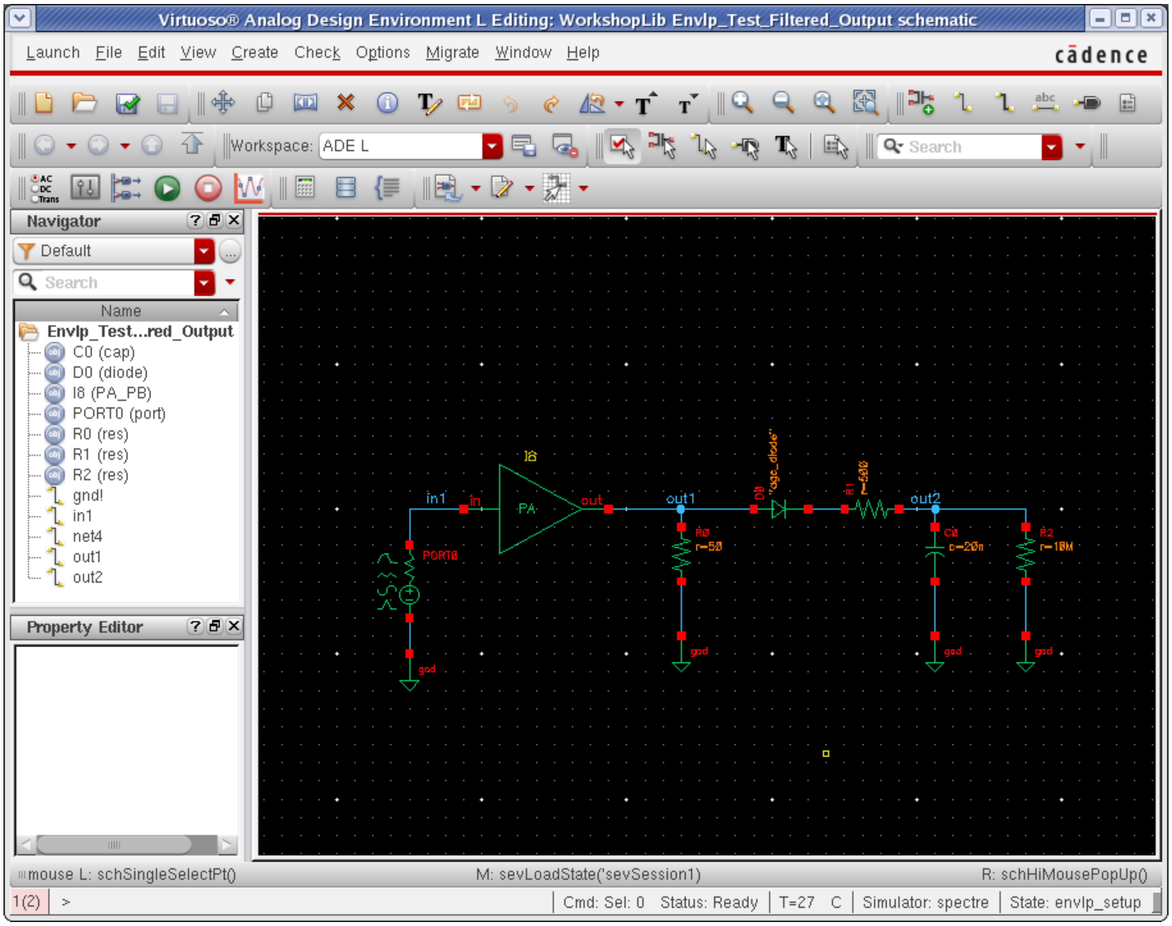


When the carrier is 2.45GHz, because of the limitation of skipping an integer number of cycles, the timing of the actual points in the envelope is not an integer multiple of 8.3333nsec. This causes small interpolation errors in the result due to the time mismatch.

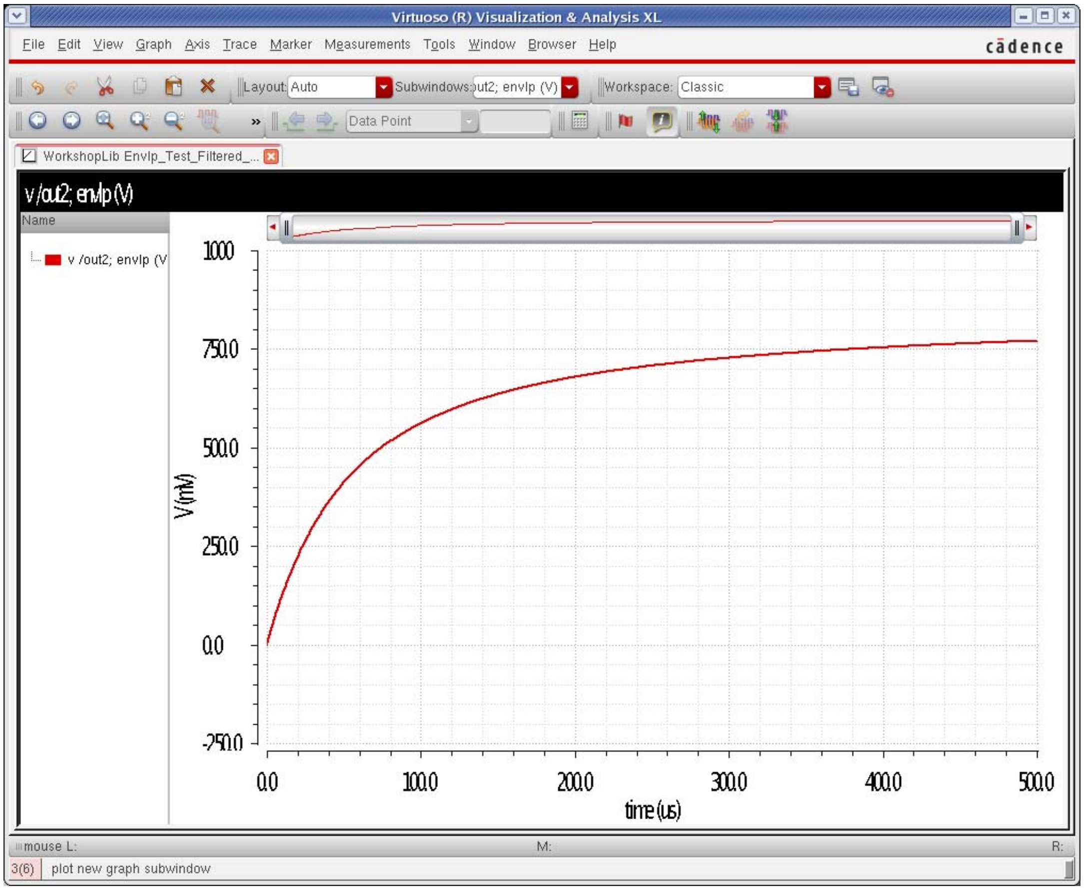


Shooting envelope is a general-purpose extension of the transient and is useful for calculating the startup behavior of many circuits that have very long time constants and very short time constants in the same circuit. An example is analog AGC (Automatic Gain Control).

As an example, consider an amplifier that has a diode peak detector after the amplifier.



The capacitor voltage starts at zero because there is no signal at time zero, and then has an exponential charging curve set by the resistors and capacitor in the peak detector.

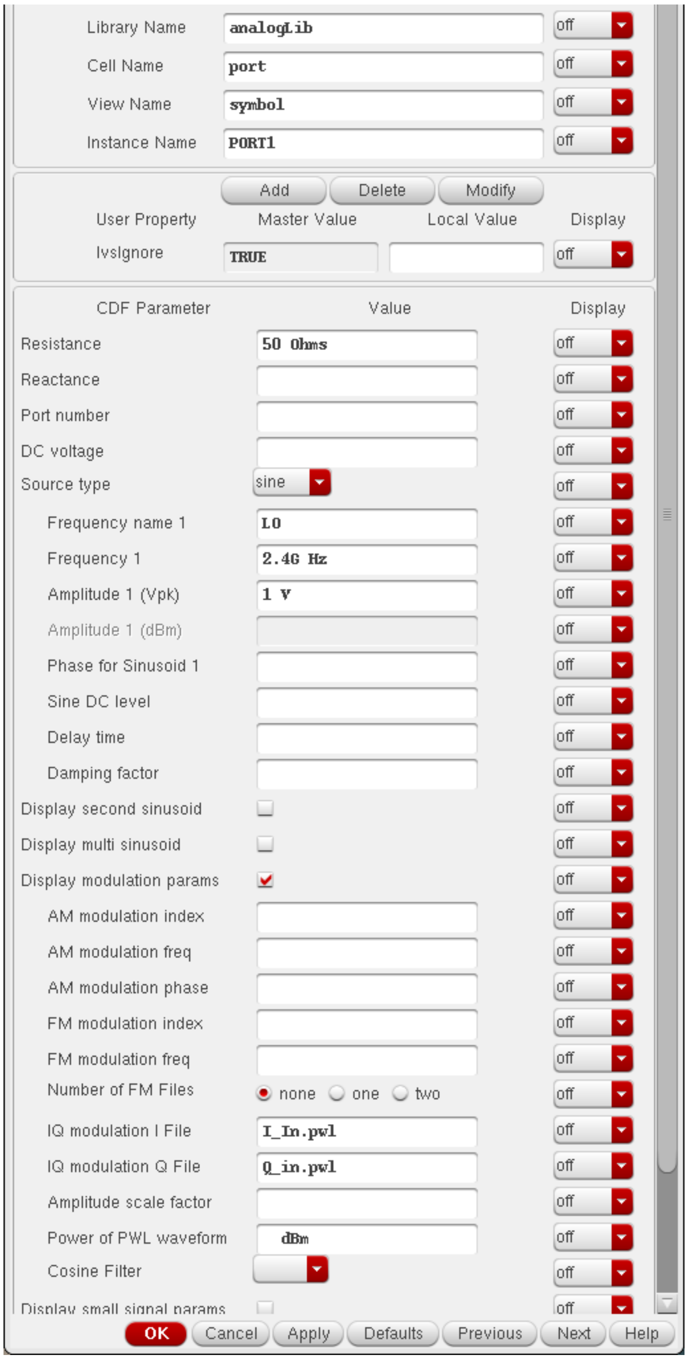


ADE Implementation and Detailed Envelope Settings for a Driven Circuit

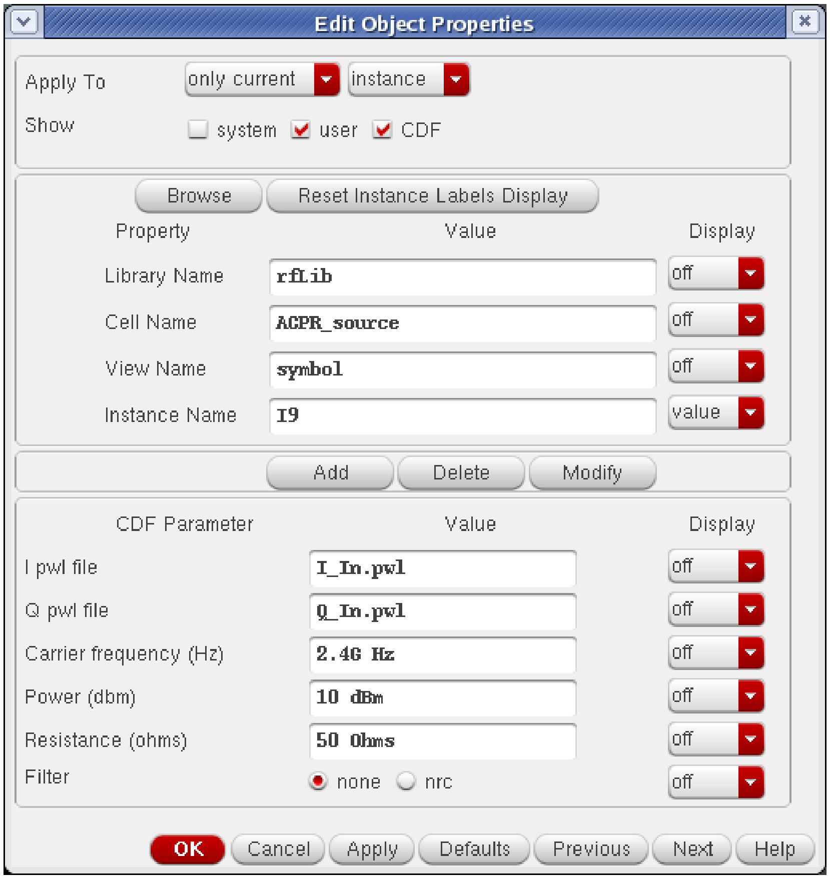
First a modulation source is needed. Your choice of modulation source depends on whether you will be using wireless mode or using your own I and Q modulation files. Cadence provides three components in *rfLib* called *wlan11n, lte, or ZigBee*, which have modulated signals for the three standards. If you use one of these sources, then add wprobes in your circuit to make the measurements.

If you are using your own I and Q modulation files,, you will need to get a piecewise linear I and Q modulation file from your system design group. These files will have a constant delta-time between the points. This delta-time should be 1/(channel bandwidth*6) or smaller if you need an ACPR measurement, and 1/(channel bandwidth*10) or smaller if you need a second adjacent channel power ratio calculation. This is required to meet the Nyquist criteria for the modulation.

To create the modulated RF signal from the I and Q modulation files, either a port component from *analogLib* or *ACPR_Source* from *rfLib* can be used. If the port is used, you need to scale the amplitude of the RF signal to achieve the correct power at the output of the port. If you use the *rfVsource*, it will read the I and Q data files, and the power in dBm can be set directly on the source. Note that this power will be achieved when you simulate to exactly the end time in the I and Q modulation files. The properties list for the port is shown below.



The properties list for the ACPR source is shown below.



The raised cosine filter should not be needed for most simulations.

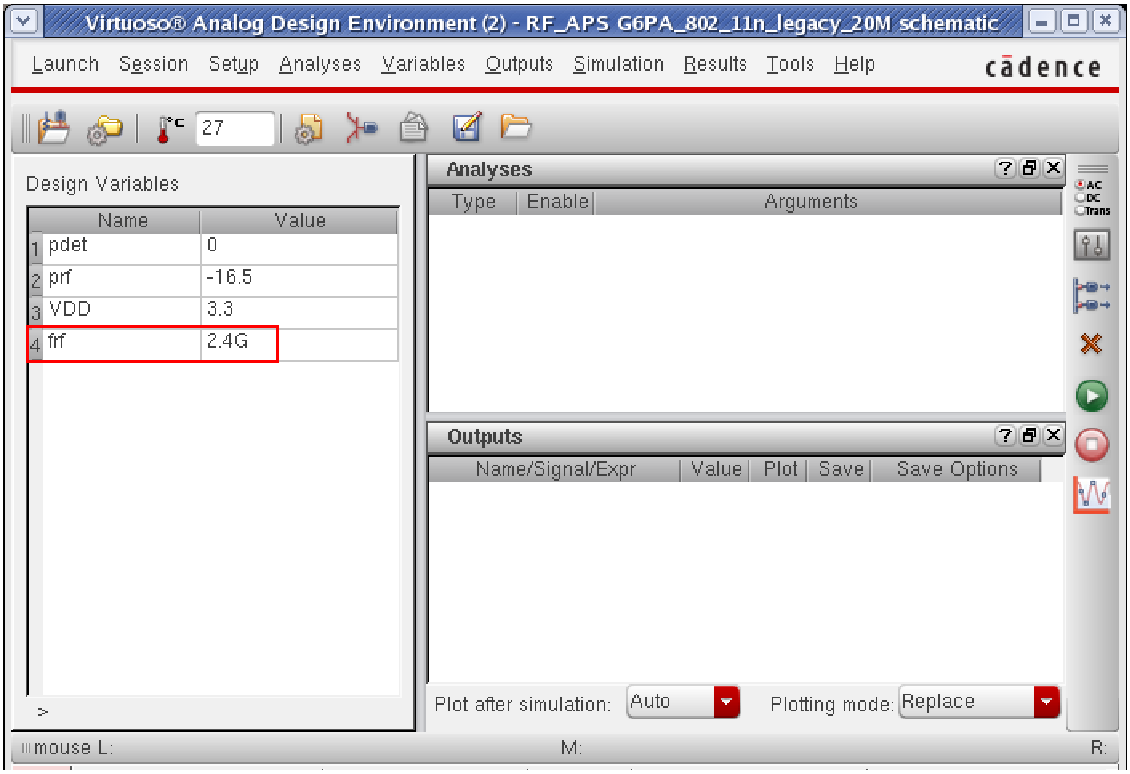
Setting up the Envelope Analysis in ADE: Wireless Mode

If you are using wireless mode, all you have to do is place a wireless source in the schematic, add wireless probes in the schematic at every point you want to make a measurement, select wireless mode in ADE, and define a reasonable number of harmonics for your system. After you run the envelope analysis, direct plot functions are available for all the common measurements. See the example at the beginning of this chapter for details.

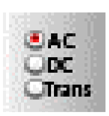
Setting up the Envelope Analysis in ADE: Using your own I and Q Files

1. Everything starts with the I and Q files you are working with.

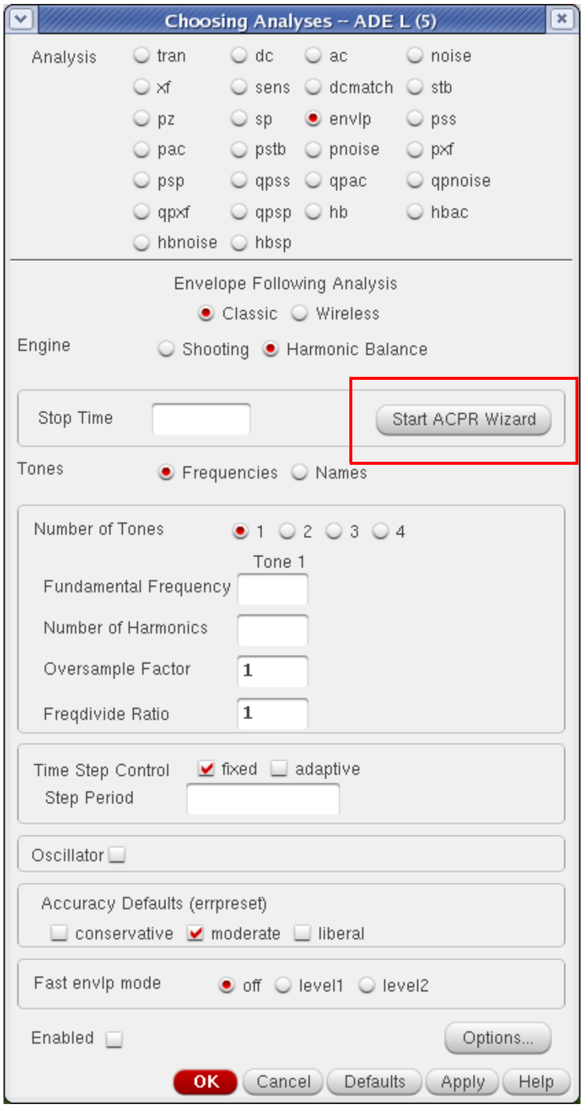
2. Calculate 1/time difference in these files. If you are using shooting, consider making the carrier frequency in the design an integer multiple of this frequency. This will reduce the numerical noise floor of the simulation. For harmonic balance, it is not needed unless you are trying to determine the phase of the signal based on the time-domain result. The carrier frequency has been set in the ADE variable *frf* shown below.



3. Now select *Analysis - Choose* or the *Choosing Analyses* icon on the right side of the ADE window.



4. Select *envlp*. The ENVLP *Choosing Analyses* form is displayed.

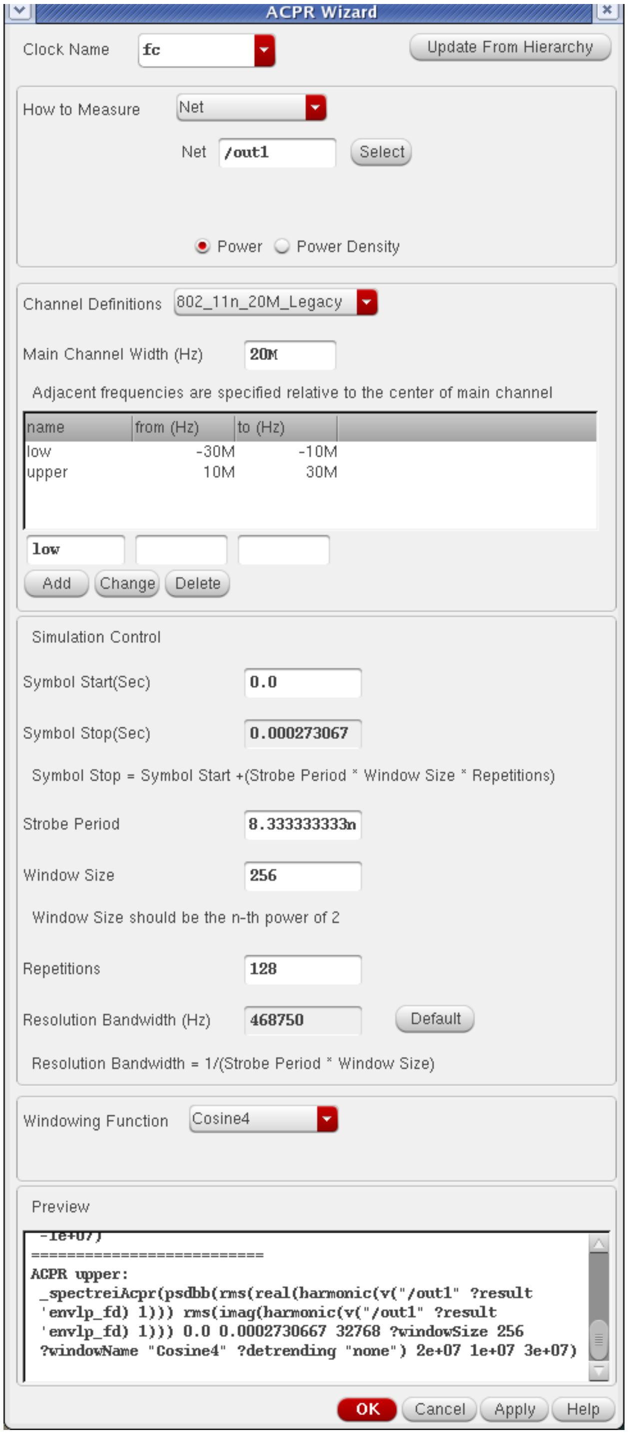


5. Now click the *Start ACPR Wizard* button.

The ACPR wizard is displayed.

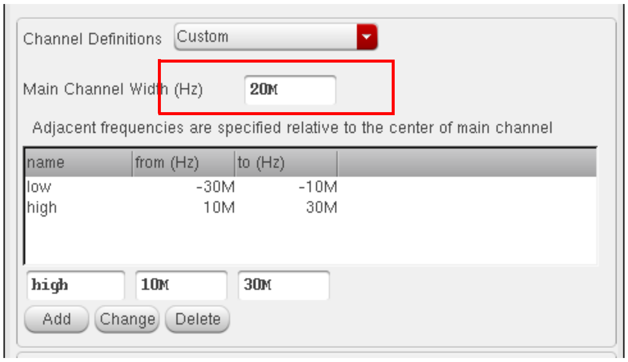
6. From the *Clock Name* drop-down list, select the name that is in the *Frequency name* 1 or 2 property that sets the carrier frequency in the circuit. If you are using *ACPR_source*, this name is *fc*.
7. Select *Net* or *Differential net*, depending on your design.
8. Click *Select* on the right of the *Net* field and then select the output net(s) in the schematic.
9. If you are working on a Qualcomm standard, select *Power Density*. Otherwise, select *Power*.

10. If you are working on an 802.11n standard, select the standard from the Channel Definitions drop-down list. The rest of the fields in the form will have starting values. Please see the discussion for Symbol Start, Window Size, and Repetitions below.



11. If you are working on another standard, select Custom.

12. Type the channel bandwidth in the *Main Channel Width* field.



13. In the Adjacent frequencies section select *low*.

The edit fields below this section become active.

14. Type the frequency limits for the first low adjacent channel for the ACPR calculation. Note that these frequencies are referenced to the center of the main channel. For frequencies on the low side of the main channel, the numbers will be negative.

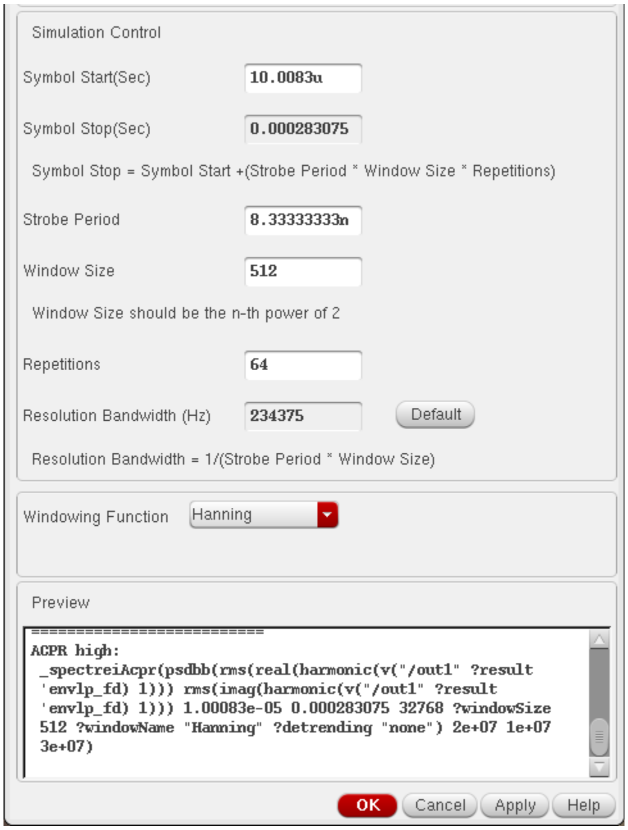
15. Select *Change*.

This moves the contents of the edit field up into the list.

16. In a similar manner, select *high*, enter the frequency range, and click *Change*.

17. If you have second adjacent specifications, add the name and frequency limits in the edit field above, and then click *Add*.

18. Set *Symbol Start(Sec)* field to the time when you want the Fourier transform to begin. In many I and Q files, there is a time delay caused by digital filtering. The start time for the frequency response calculation needs to be after that delay time. If you are unsure, simulate the I and Q file in a piecewise-linear source in the transient analysis, and watch for the time that the output becomes non-zero, and pick a time slightly later than that.



19. Set the strobe period to the delta-time in the I and Q modulation file. If you are unsure, view the file in a text editor, and look at the left entry of the second line. The first line should have a left entry of zero (seconds) The second entry left entry (in seconds) should be entered in the strobe period field. The right entry is the voltage at the time in the left column.

Note: When you enter a strobe period, the ACPR wizard will update the Symbol Start field so that it is an integer multiple of the strobe period.

The Fourier transform used to calculate the frequency response of the circuit is inherently noisy. To make the spectrum less noisy, the *psd* function is used. The *psd* function splits the time that is simulated into smaller pieces, which allows multiple shorter time sequences to be considered individually. A fourier transform is performed on all the pieces individually, and then the power of all the first harmonics is averaged. This is done for the rest of the harmonics also. Because of the averaging, the noise is reduced. Because a shorter time exists in each piece, the frequency of the first harmonic goes up. In the *psd*, this reduces the resolution (the number of frequency points analyzed in the spectrum). Generally, a trade-off is needed between the number of pieces, the length of the simulation, and the noise in the psd calculation.

20. Window size is the number of delta-times in the I and Q input files you want to include for the spectral calculation. This needs to be a power of two. Generally, this is in the range of 256 to 8192.

21. Specify repetitions as a power of 2. This will run as many windows sizes as you specify.

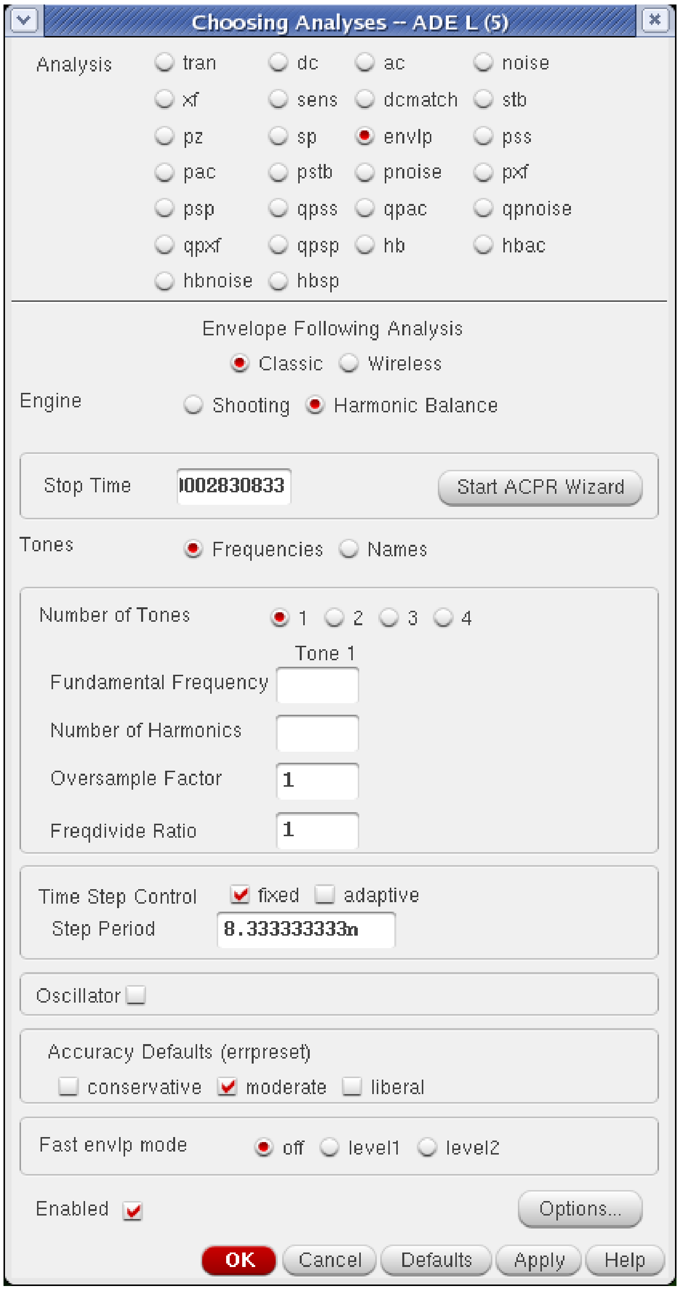
22. The *Symbol Stop(Sec)* is calculated based on the symbol start, window size, and repetitions fields. Make sure that the data in your I and Q modulation file has entries to at least this time. View the I and Q modulation file, and look at the left entry (the time) in the last line of the file to make sure that ending time is larger than the *Symbol Stop(Sec)* time in the ACPR wizard.

The resolution bandwidth is calculated based on the entries for strobe period and window size. It is the frequency resolution of the Fourier transform of the output signal.

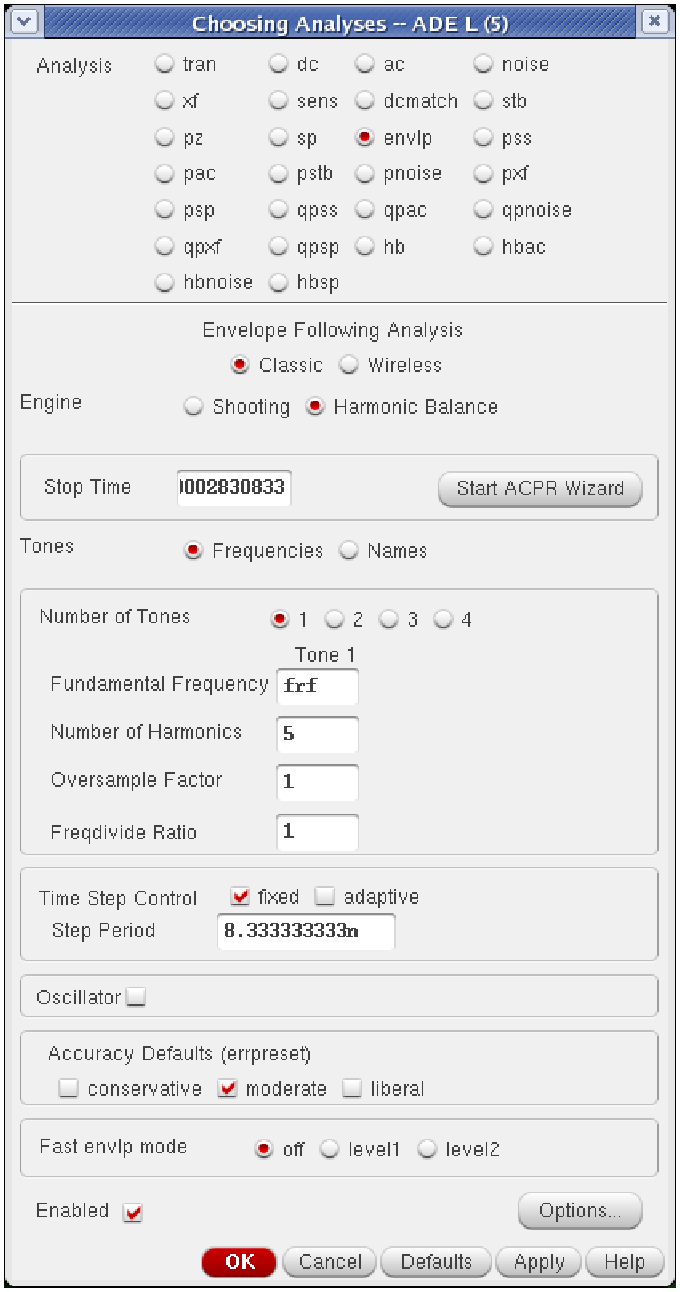
23. Because the data bits in the modulation are random, a Fourier transform window function needs to be selected. There are advantages and disadvantages to all window functions. The *Hanning* window is a reasonable window if the window function is not specified by the standard. Check with your system designer to find out his or her preference.

24. When you are done, click *OK*.

In the *Choosing Analyses* form for ENVLP, the *Stop Time* field is populated. The *Time Step Contro*l is also set to *fixed* and the *Step Period* field has the entry that you specified for the strobing interval in the ACPR wizard. A number of envelope options are also set to provide the output from the simulator at the times needed for the FFT-based frequency response calculation.



25. Either type the frequency of the carrier, the number of harms, and the oversample factor, or select names, and set the desired number of harmonics.



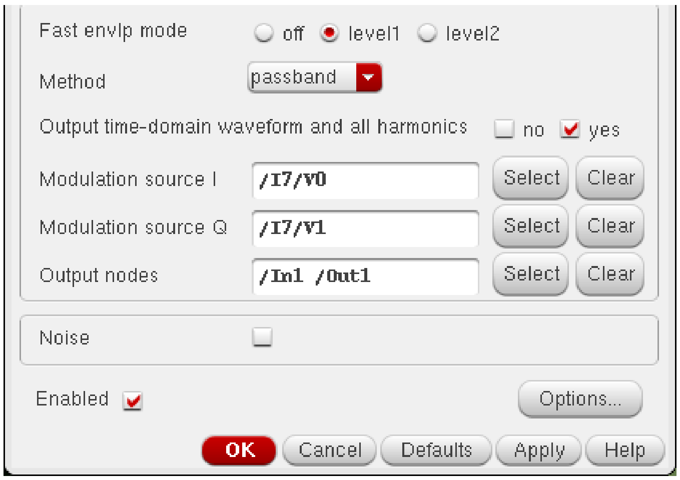
26. If you have square waves in your circuit, use an oversample factor if you choose harmonic balance, or select shooting. If you decide to use harmonic balance, you need relatively more harmonics. In this case, also set oversample factor to two to eight. The more non-sinusoidal the waveform is, the higher oversample factor should be.

27. If you have a frequency divider in your circuit, specify the divider ratio in the *Freqdivide Ratio* field.

28. For the *Time Step Control* field, select *fixed* or *adaptive* for Harmonic Balance and *adaptive* for Shooting.

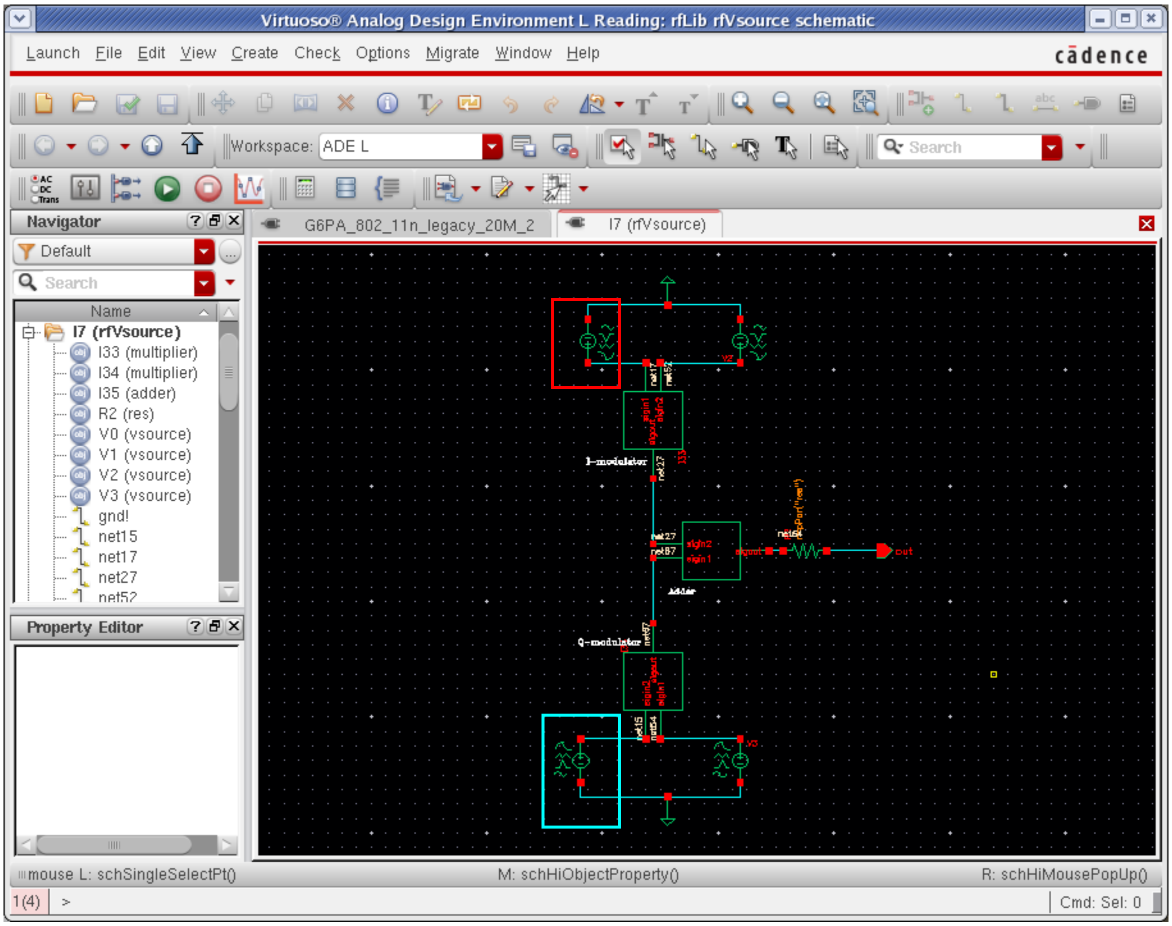
29. For most ACPR measurements, select *conservative*.

30. Select the fast envelope level you desire. This is available only when harmonic balance is chosen for the engine. When the fast envelope mode is set to off (default), normal envelope is performed at the transistor level. Harmonic Balance or Shooting is used to solve for a subset of the RF cycles. Level 1 is available for circuits where acceleration is desired, and the circuit has little memory effect Level1 requires defining the I and Q sources in the circuit, and the nodes to save data for in the circuit.



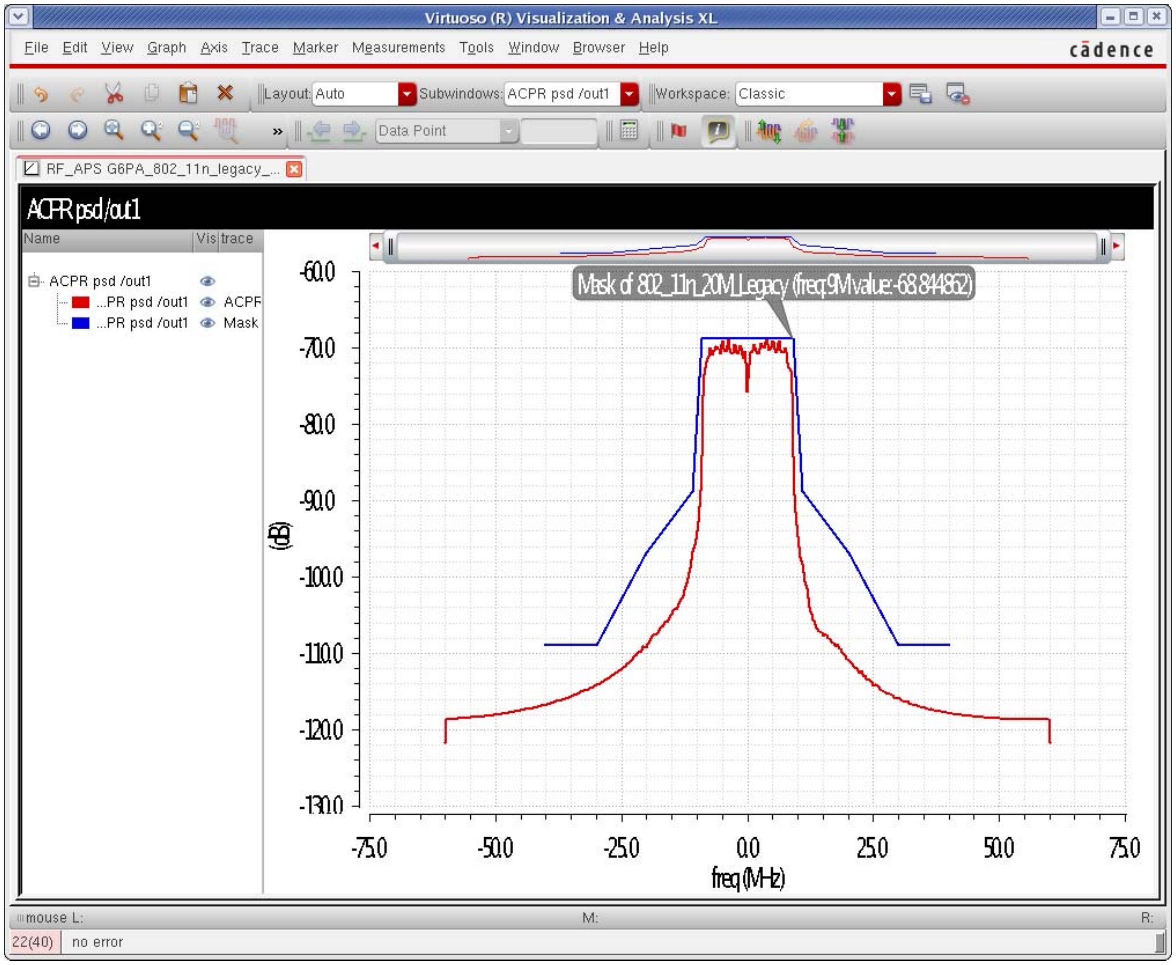
31. Select whether you want to have all the harmonics on the output along with the time-domain waveform or not. The default in fast envelope is to not calculate the time-domain output waveform and higher harmonics in order to save time. If you take the default, only the harmonic number you specify in the *Output harmonic* field is calculated. If you want the all the harmonics and/or the time-domain waveform, select the *yes* check box to the right of *Output time-domain waveform and all harmonics*.

If you are using the port component with an I and Q modulation file, that port should be selected for both the I and Q source. If you are using the *ACPR_source*, descend-read into the *ACPR_source* schematic. The I source is highlighted in red below, and the Q source is highlighted in blue below.



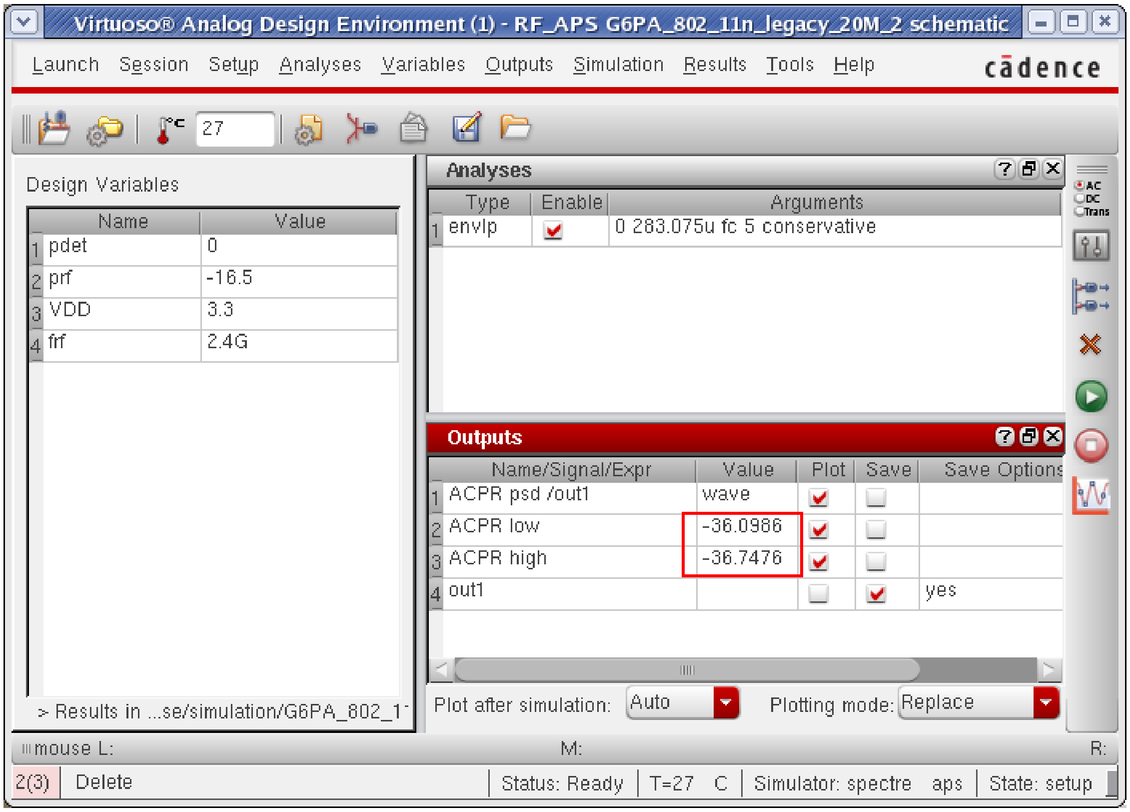
The mathematical model can be saved by setting the `writeenv` option to a file name. After this, the model file can be read in and used directly by setting the `readenv` option. You cannot set both options at the same time.

Now run the simulation. When the simulation completes, the psd (frequency response) plots in the waveform window.



The unit for the psd is dBW (dB with respect to 1 Watt).

The ACPR is shown in ADE outputs section.

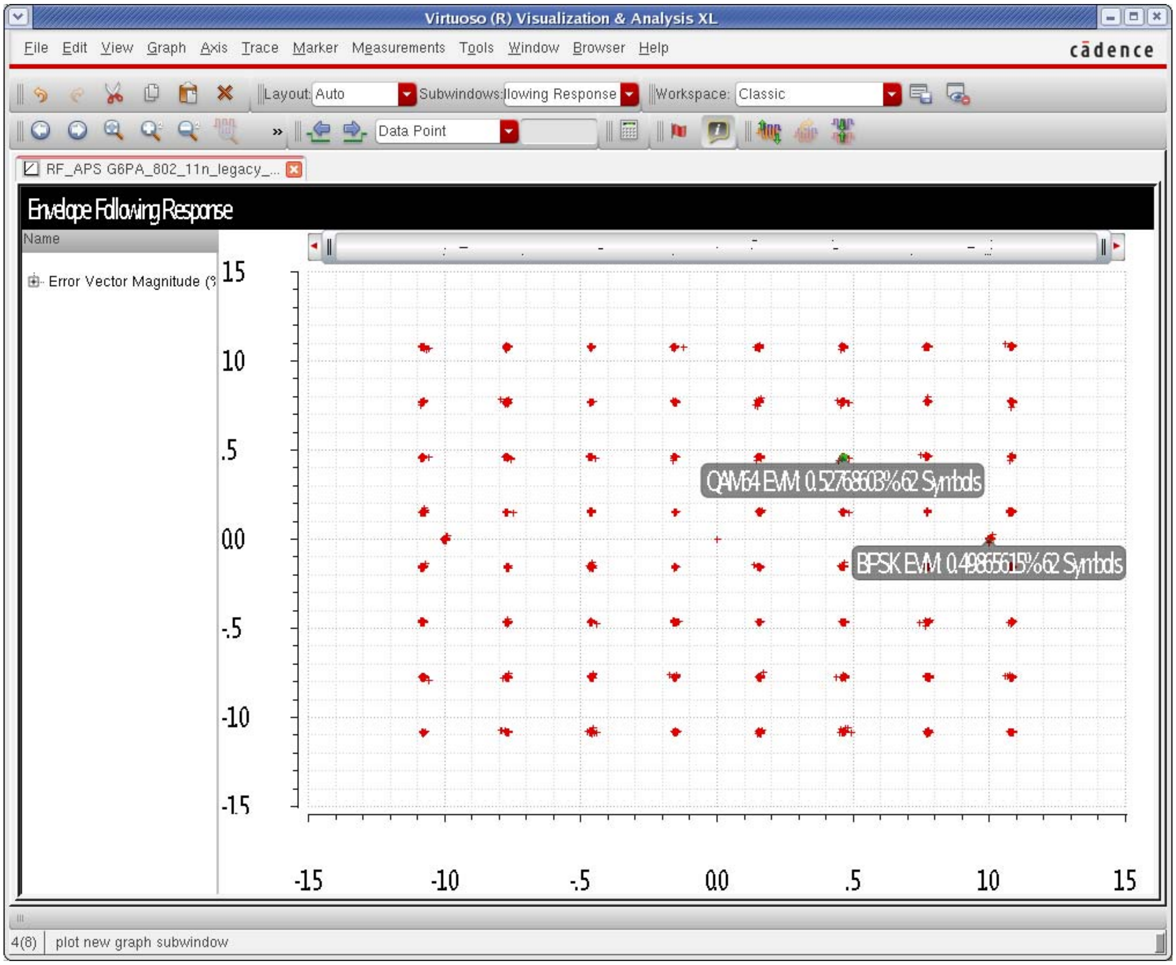


Plotting EVM

EVM is integrated into the Direct Plot Form for the 802.11 standards only.

1. In ADE, select *Results - Direct Plot - Main Form*.
2. Select *envlp* results.
3. Select *EVM*.
4. Select the first harmonic.
5. Select *OFDM*.
6. If you are using rfVsource, select the standard you are using in rfVsource.
7. Set the symbol start to 0 (zero) except for 802.11a, which is 4u.

8. Select the output net in the circuit.



The EVM for both the data channels and the control channels is displayed.

Plotting Main Channel Power

Plotting main channel power is a bit less straightforward than plotting EVM.

1. In ADE, select *Results - Direct Plot - Main Form*.

2. Select *envlp* results.

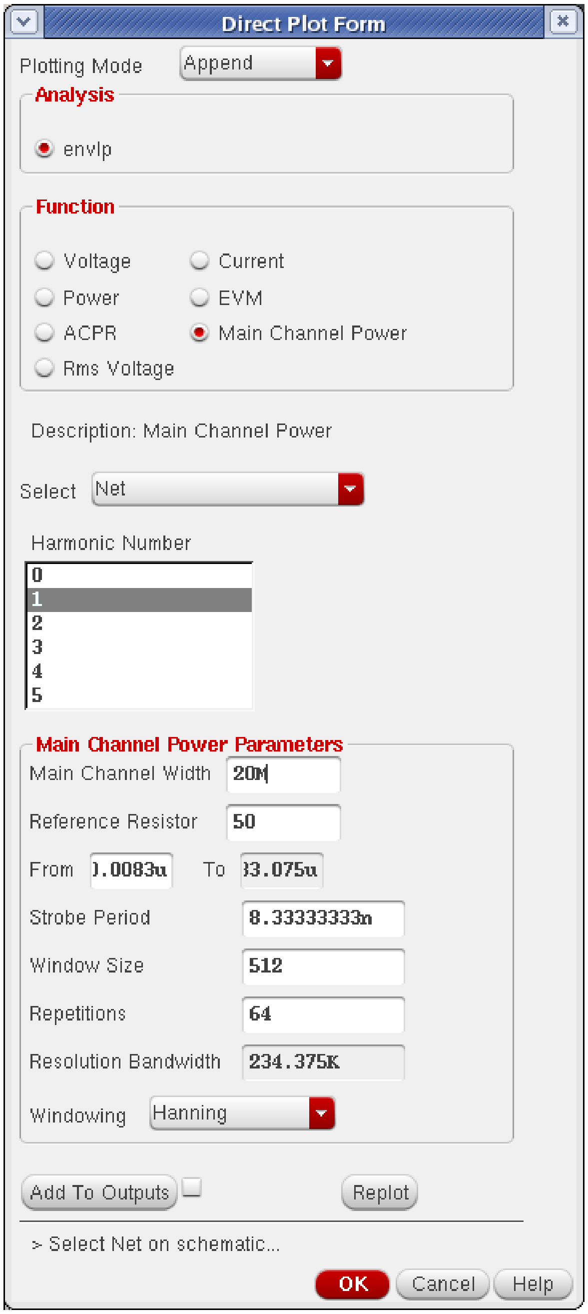
3. Select *Main Channel Power*.

4. Select the first harmonic.

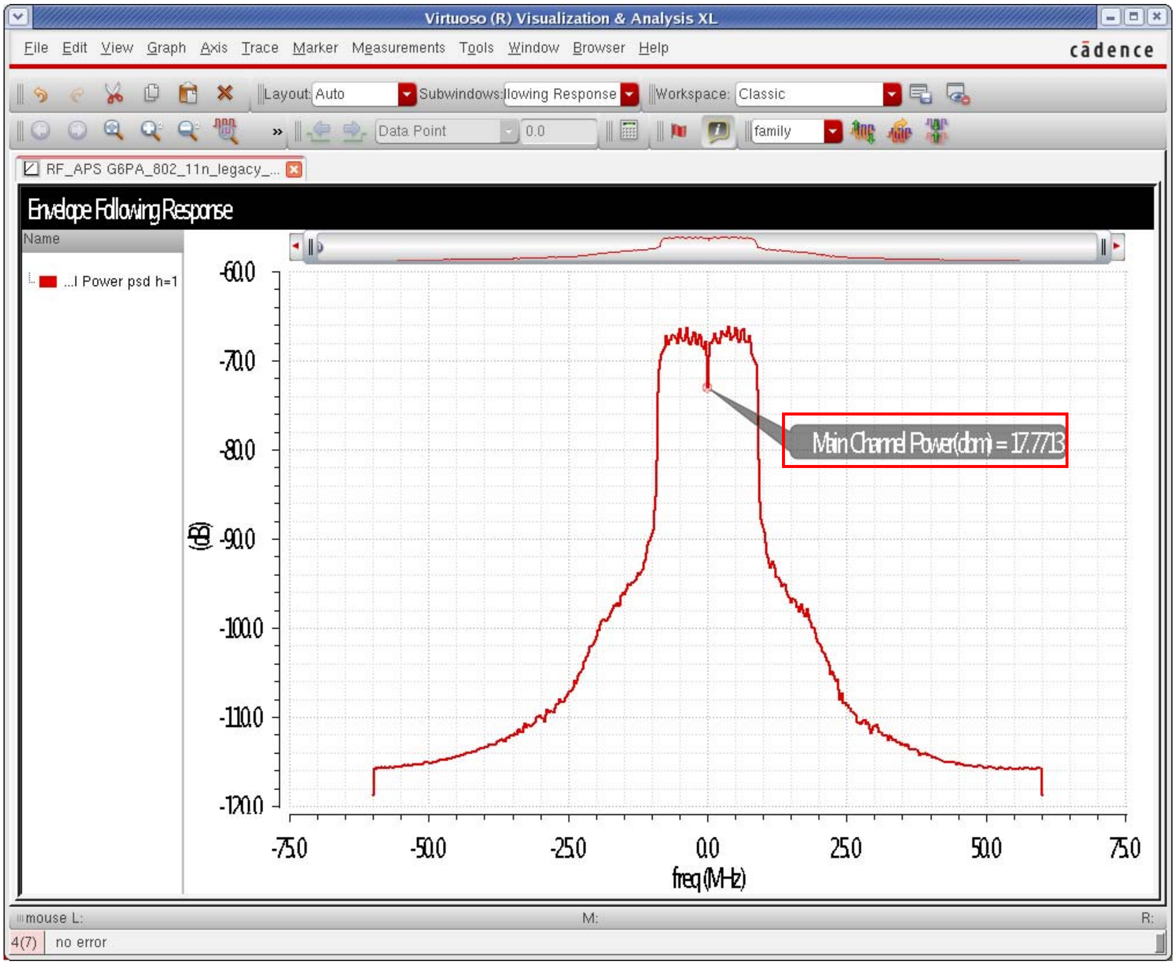
5. Type the bandwidth of the main channel.

6. Open the envelope *Choosing Analyses* form.
7. Open the ACPR wizard.
8. In the ACPR wizard, triple-click the entry in the *Symbol Start* field.
9. In the *Direct Plot Form*, click the middle mouse button in the *From* field. The entry from the ACPR wizard should transfer.
10. In a similar manner, transfer the Strobe Period, Window Size, and Repetitions entries.

11. Select the same window function that was used in the ACPR wizard.



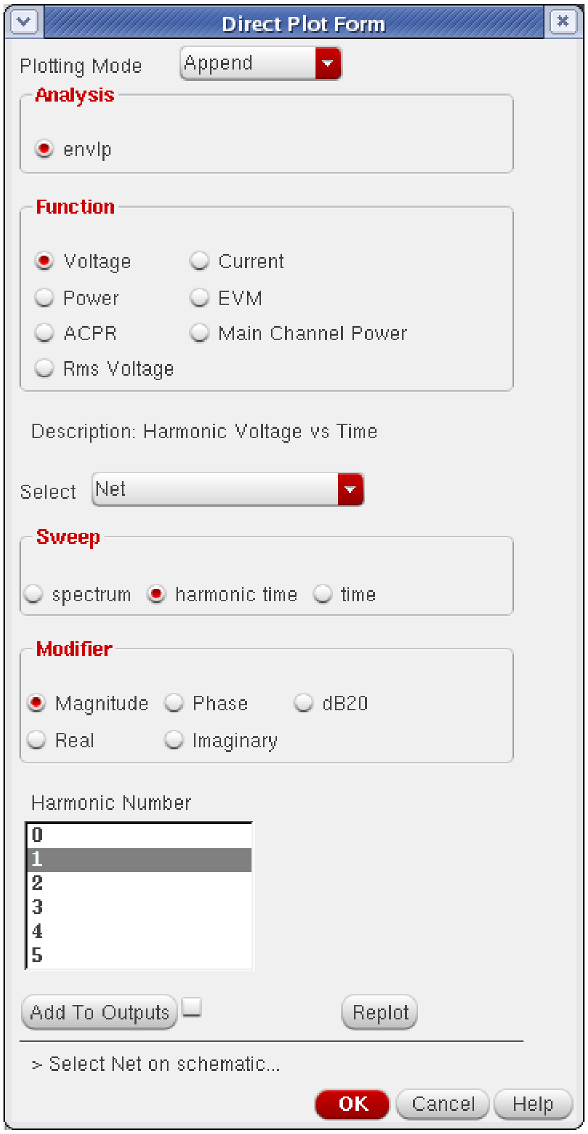
12. Select the output net in the schematic. The power calculation is displayed with a marker in the display tool.



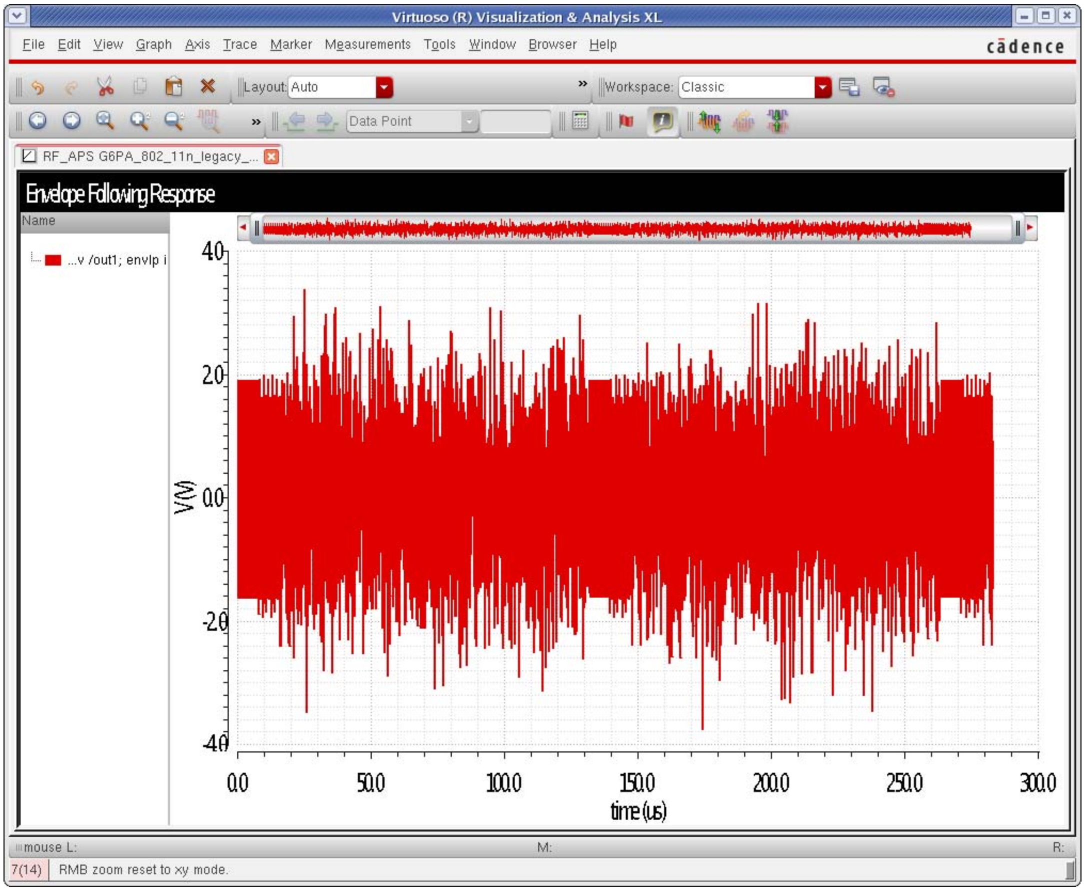
Plotting the Envelope Voltage Versus Time

1. In the ADE, select *Results - Direct Plot - Main Form*.
2. Select *envlp* results.
3. Select *Voltage*.
4. Select harmonic time.
5. Select the appropriate modifier.

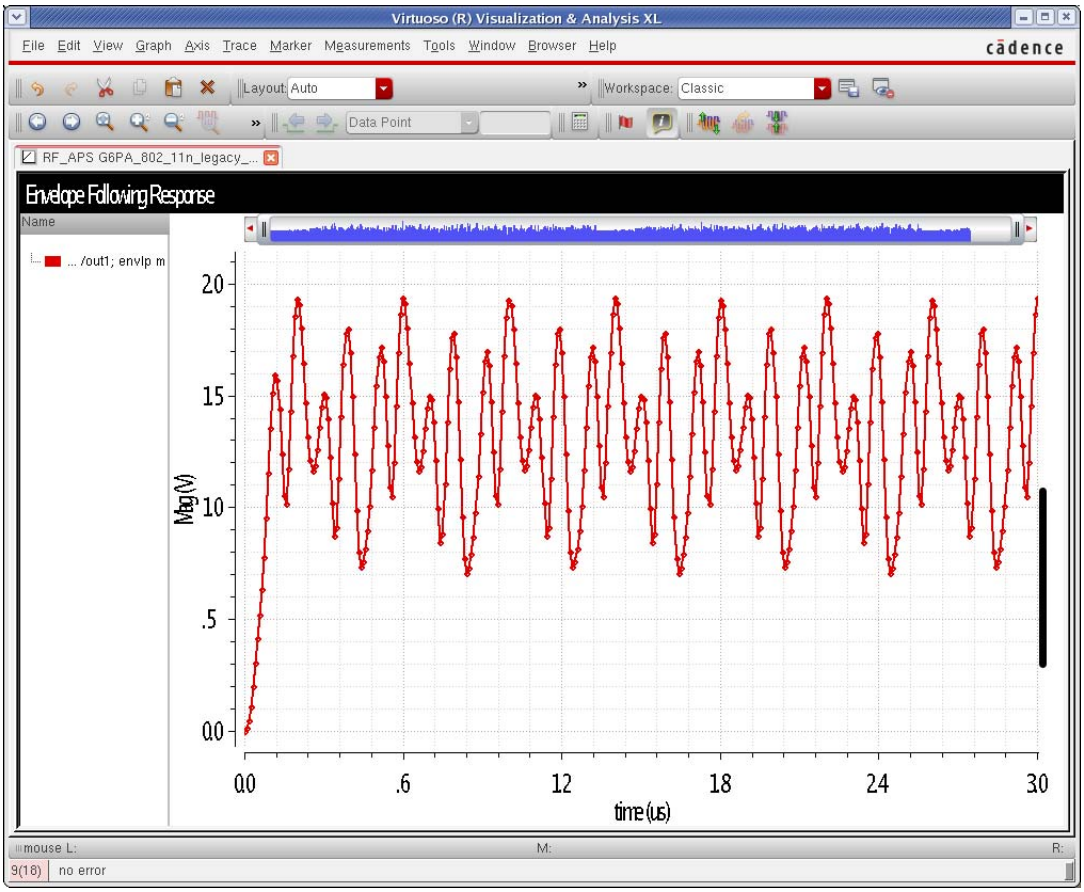
6. Select the first harmonic.



7. Select the net in the schematic.



8. Zoom in if you would like. The symbols are turned on in the display below so the strobe points are visible.



Plotting the Time-Domain Waveform

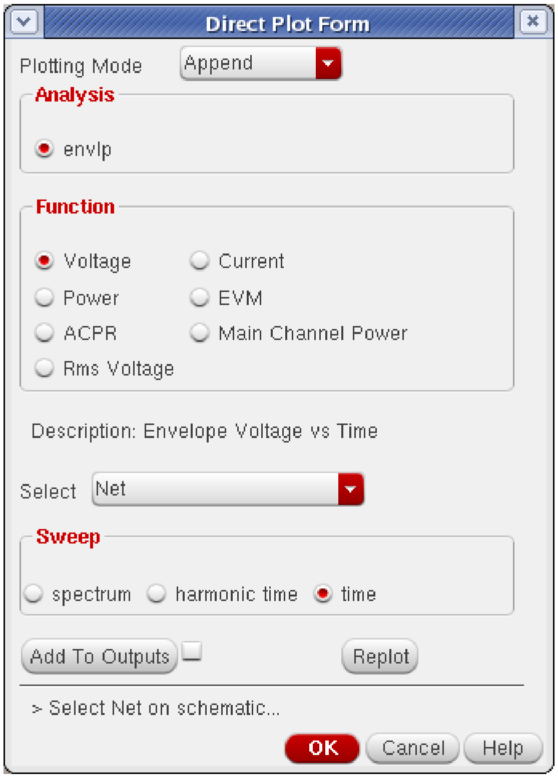
This is available in fast envelope *level1* and *level2* only when *yes* is selected for Output time-domain waveform and all harmonics.

1. In ADE, select *Results - Direct Plot - Main Form*.

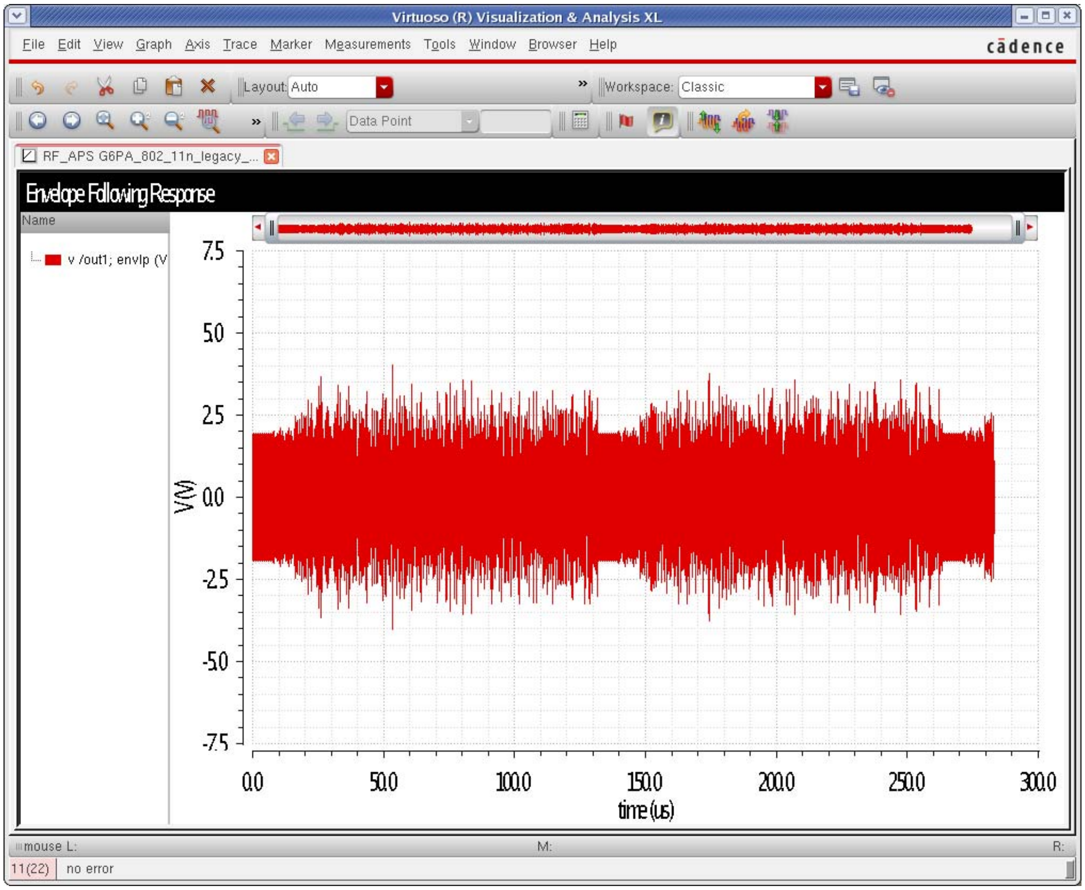
2. Select *envlp* results.

3. Select *Voltage*.

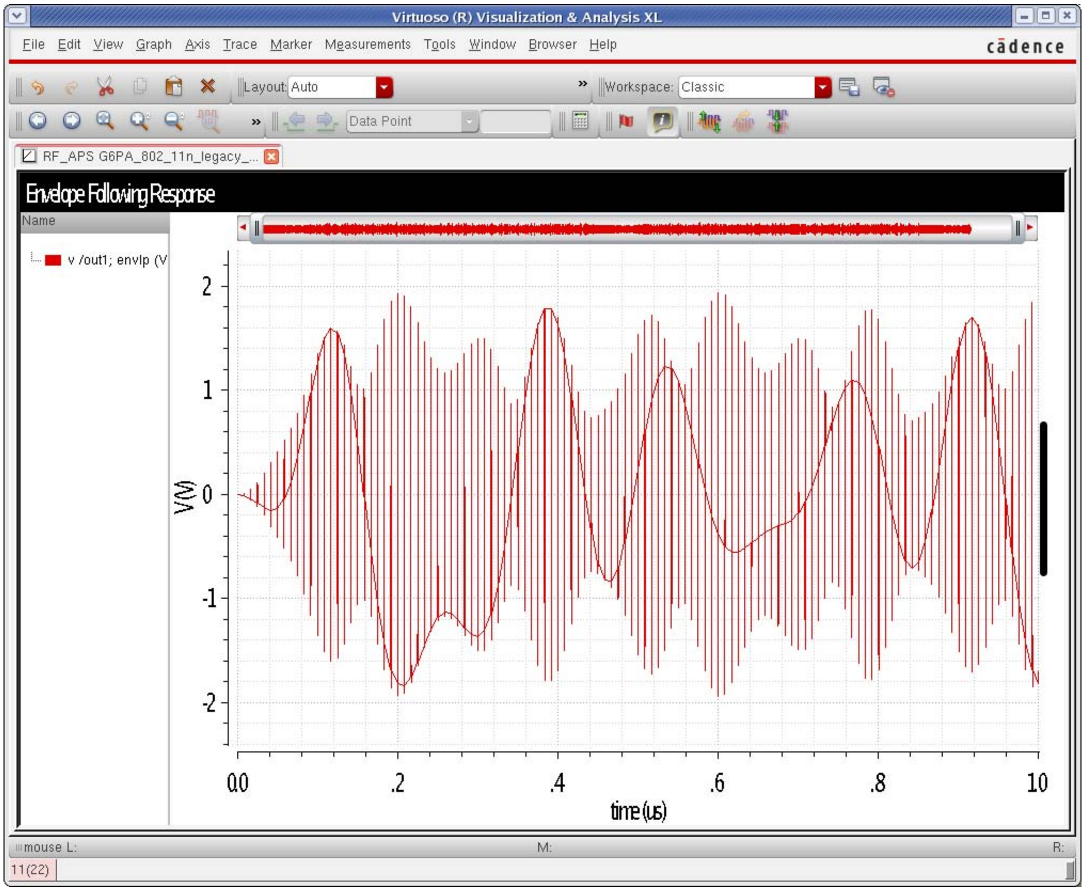
4. Select *Time*.



5. Select the net in the schematic.

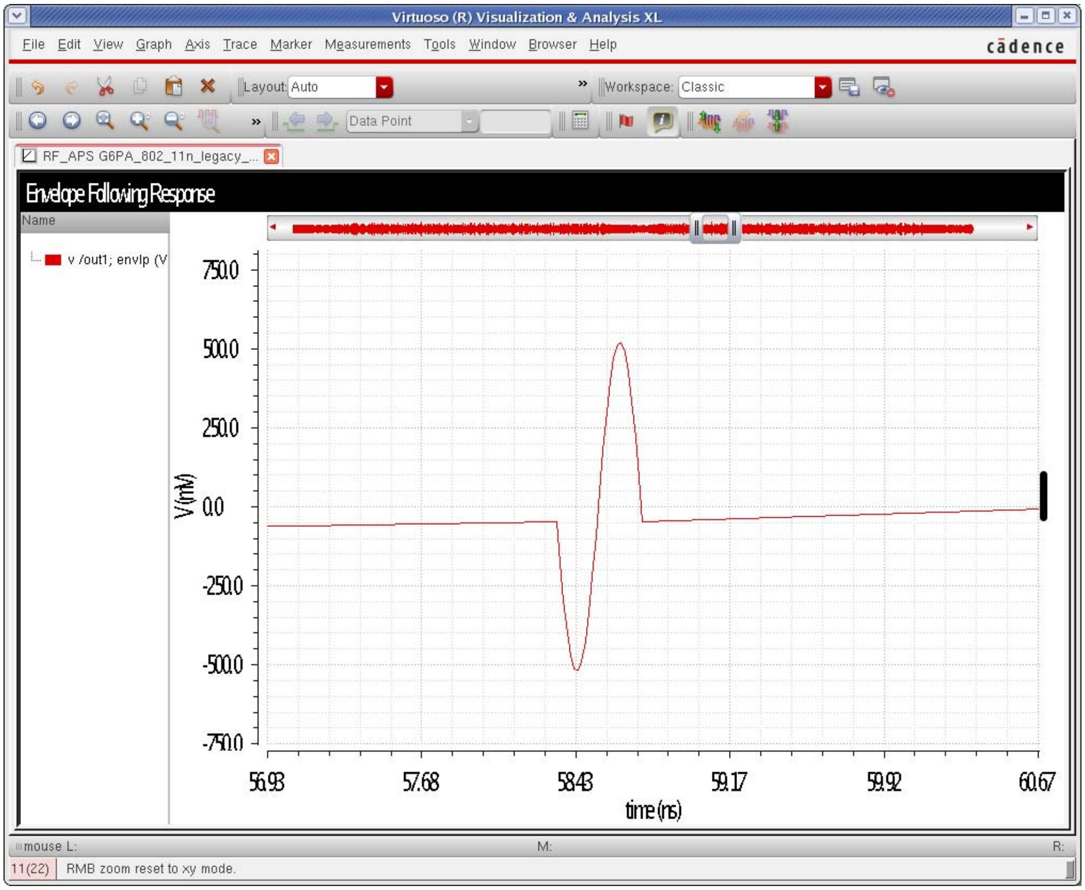


6. Zoom in, if desired.

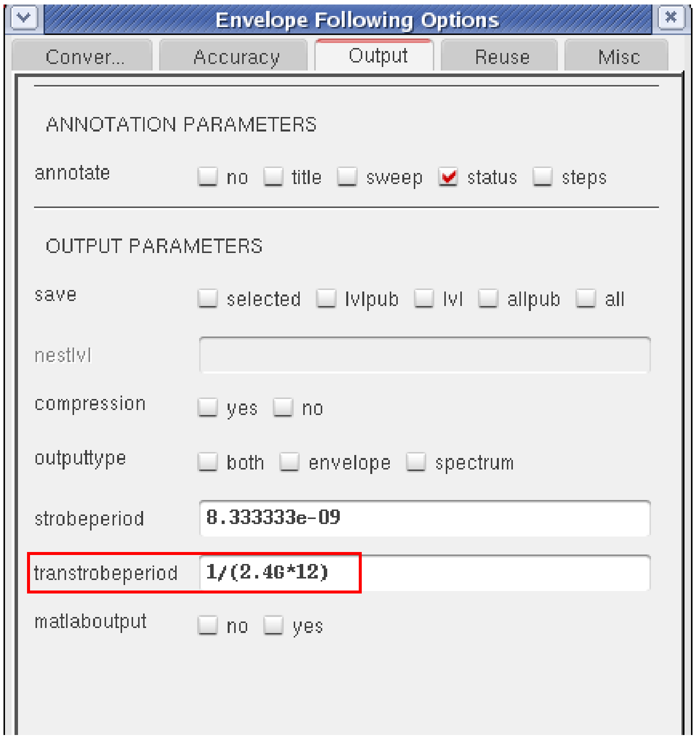


The line that is visible between the spikes can be used to determine the phase of the signal as time increases only if the carrier frequency is an integer multiple of 1/delta-T in the I and Q modulation file. When the line is near the middle, the phase is zero or 180 degrees. When the line is near the top, the phase is 90 degrees. When the line is near the bottom, the phase is 270 degrees.

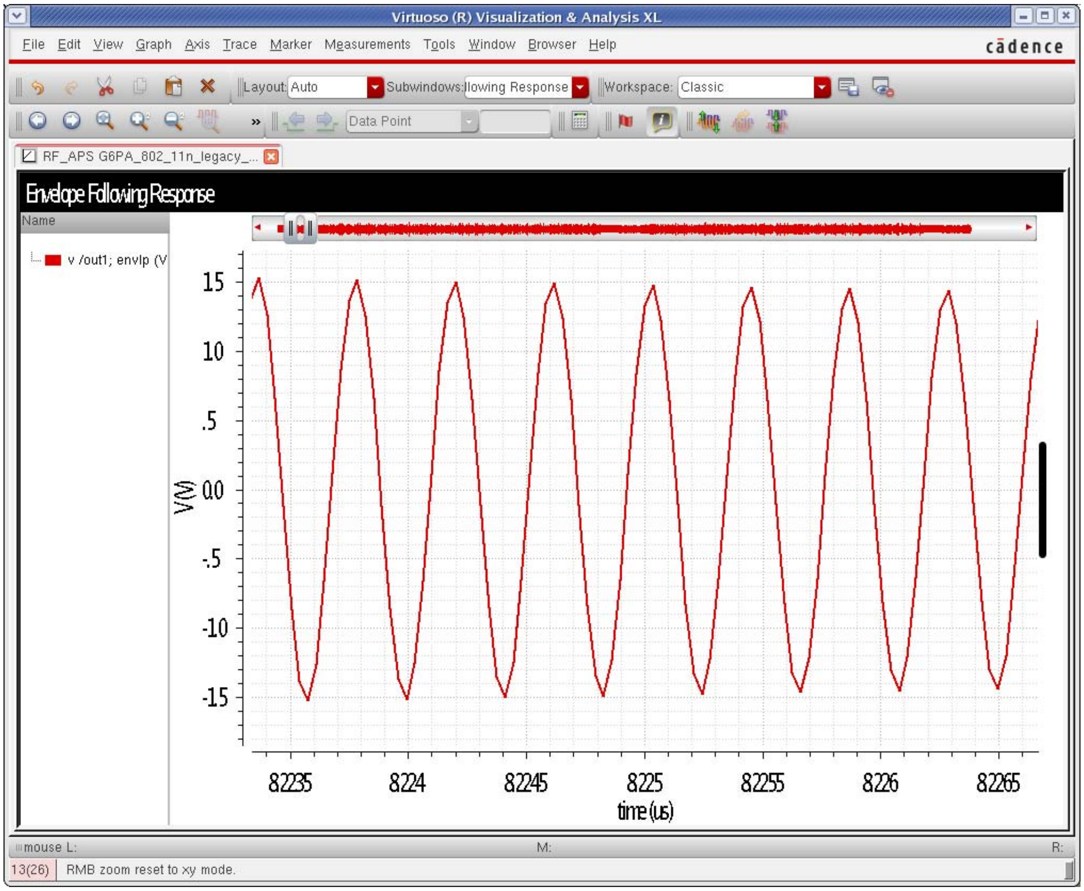
When you zoom in further, you can see the individual carrier cycles that envelope is calculating.



To see the full transient output, set the *transtrobeperiod* option to the period of the carrier divided by an integer. The choppiness of the waveform and the runtime is controlled by the number of time points in the waveform that gets calculated. This is available in harmonic balance transistor-level simulation only. An example is shown below.



When zoomed in, the waveform is continuous.



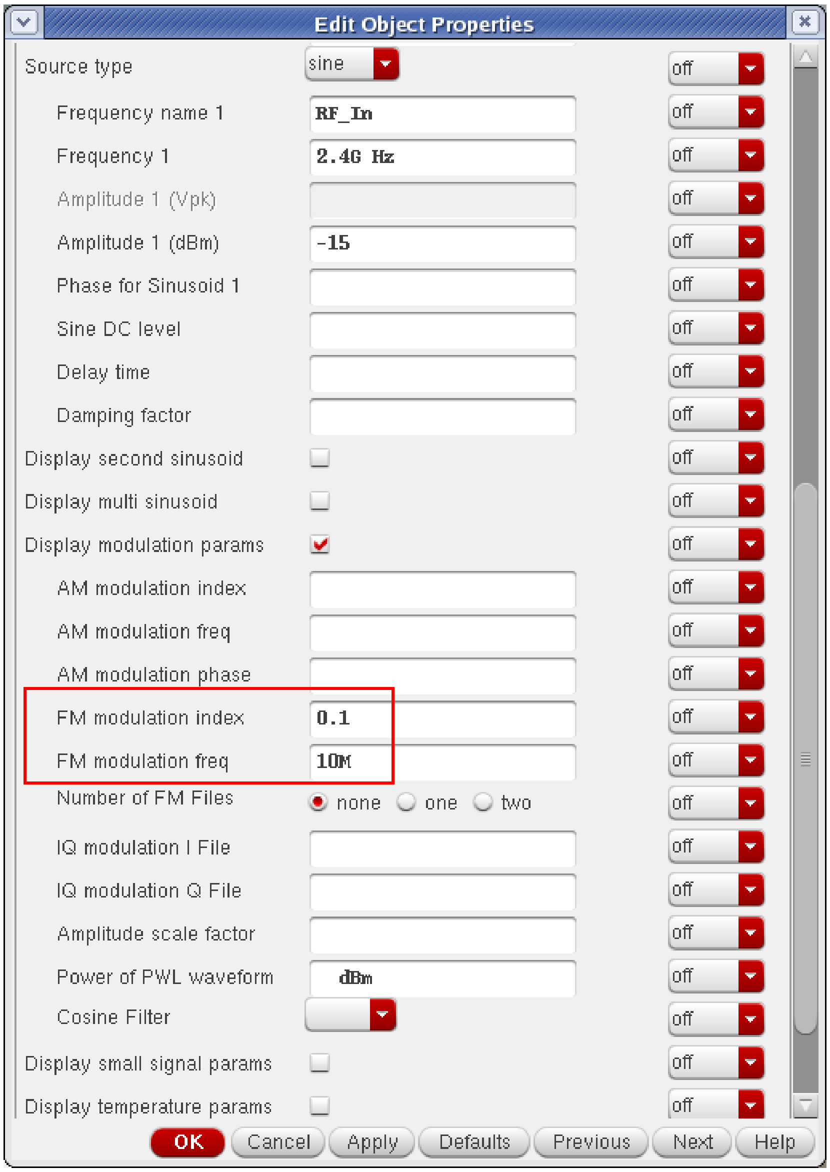
Frequency Modulated Input Signals

Frequency modulated signals can be simulated in transistor-level envelope simulations only. When frequency modulation is used, the period of the waveform varies as a function of the modulation that is applied. Conventional envelope simulation cannot be used because the fundamental assumption is that the period is constant and defined by the carrier signal.

To allow frequency modulation, a new option has been added that allows the period to be solved at each carrier cycle that is simulated.

To simulate a frequency modulated signal, follow these steps:

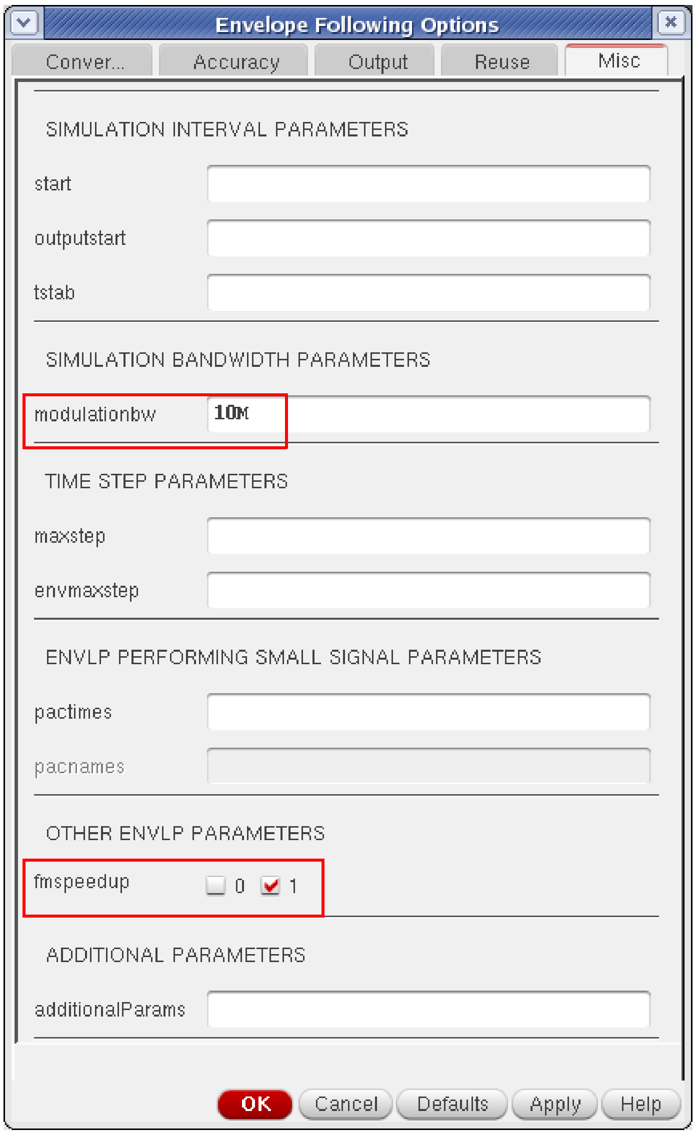
1. Set up the frequency modulated source. In this example, a simple sinusoidal modulated port is used. If you use this element, remember that the modulation index is in percent.



2. Set up for a simulation using the envlp *Choosing Analyses* form and the ACPR wizard as shown before. Choose a standard with a bandwidth that is slightly larger than the FM modulation frequency. This will set the simulation to produce data at the time interval needed for the psd calculation. See the previous section for the steps that are needed.

3. Click *Options* in the envelope *Choosing Analyses* form.

4. Select the *Misc* tab.

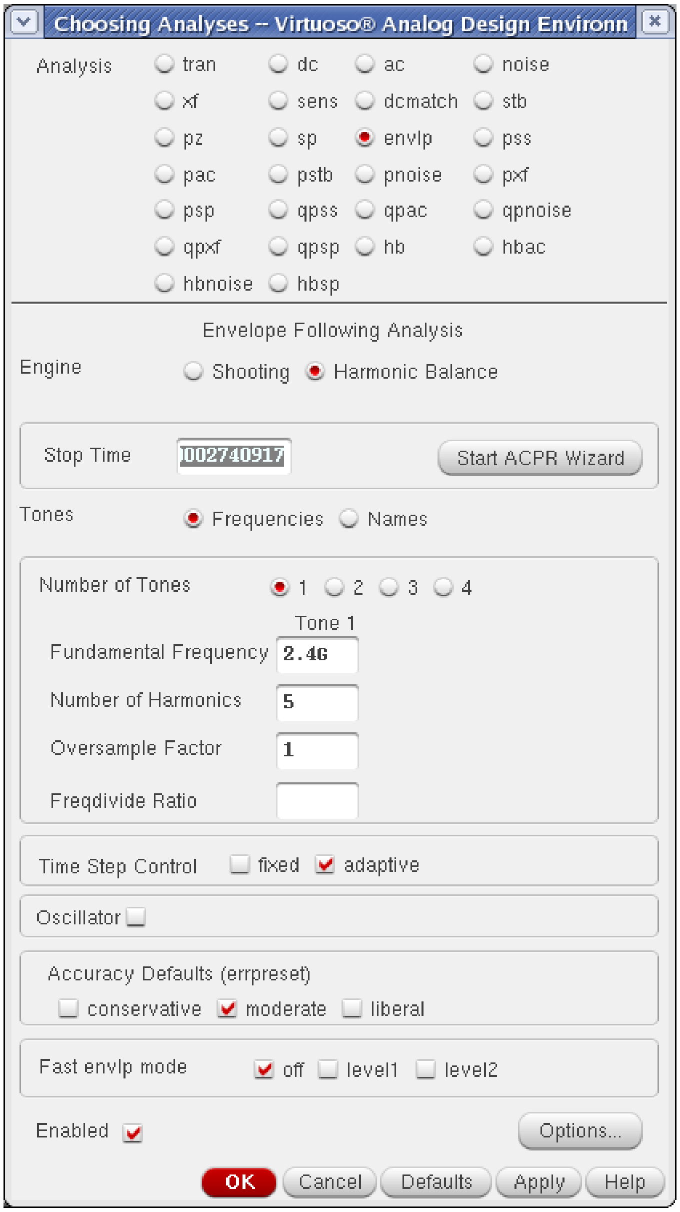


5. Select the check box to the left of the number 1 for the *fmspeedup* option.

6. Type the modulation frequency in the *modulationbw* field.

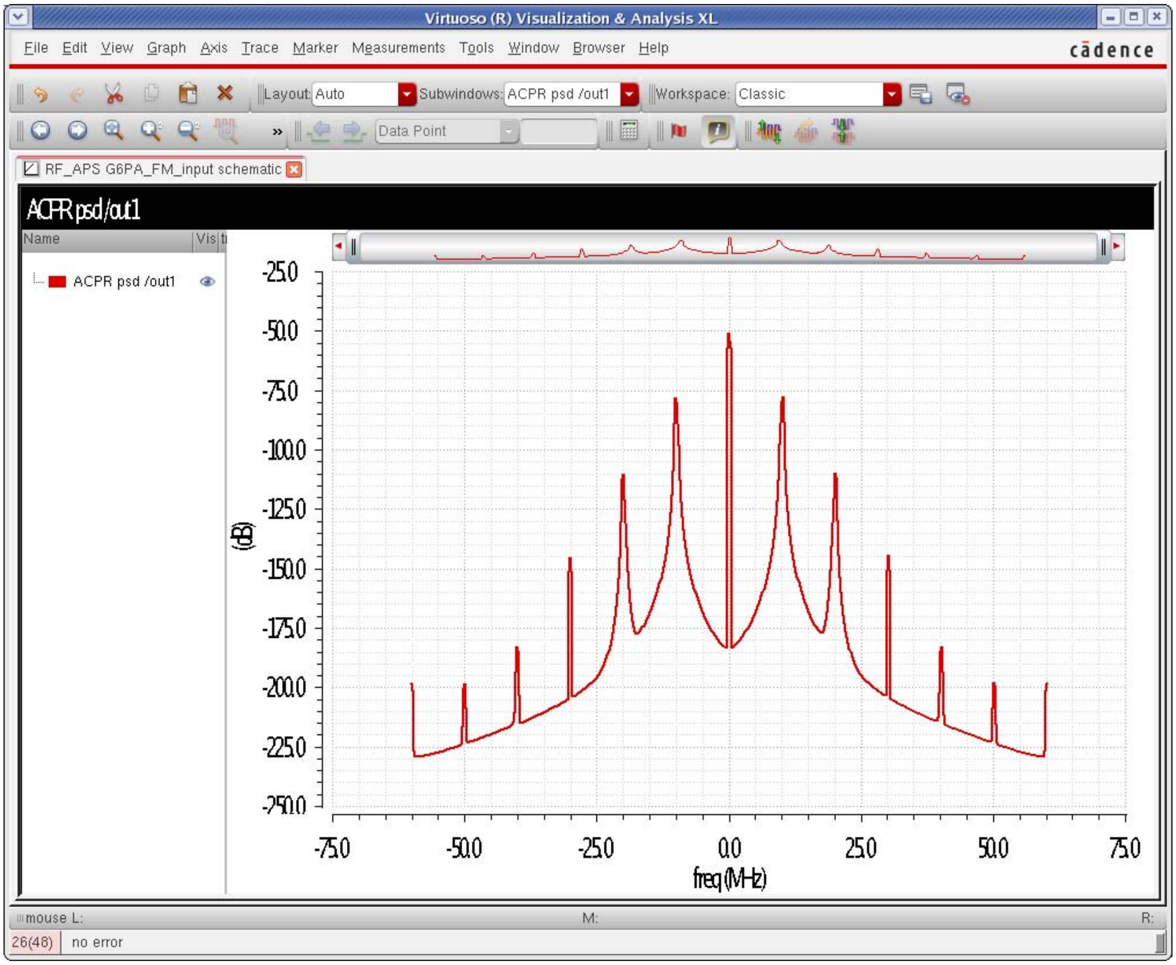
7. Click *OK*.

8. In the envelope *Choosing Analyses* form, type in the center frequency in the *Fundamental Frequency* field.

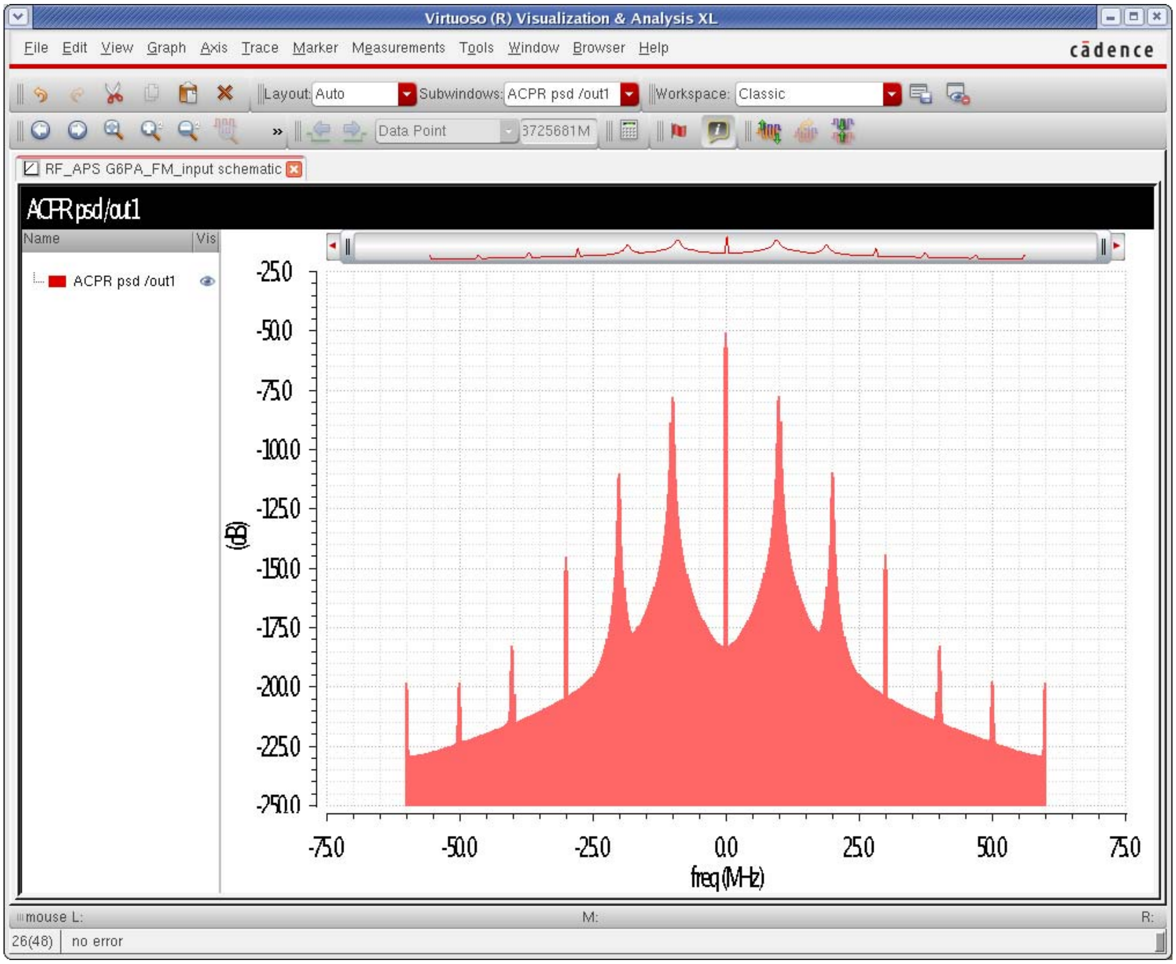


9. Set harmonics and oversample as required. For more information, see the harmonic balance section at the beginning of Chapter 3.
10. Choose an accuracy level.
11. Set the *Fast envlp* mode to *off*.
12. Click *OK* and run the simulation.

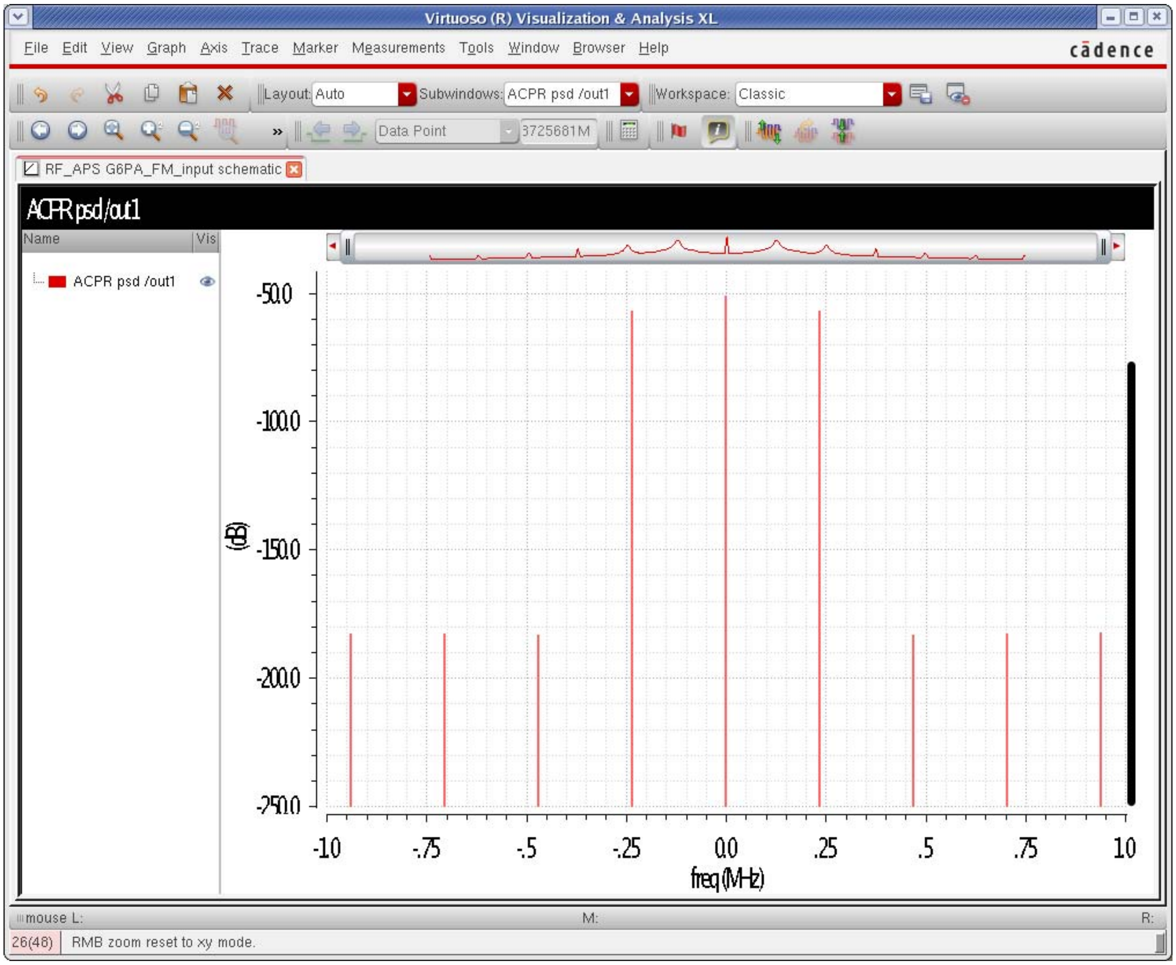
13. When the simulation completes, the psd will plot.



14. If you want to see a spectral plot, set the line style to spectral. To do this, select the legend at the upper left. Right-click and select *Type* from the context menu. In the popup window, select *Spectral*. The line style will change to a more familiar plot.



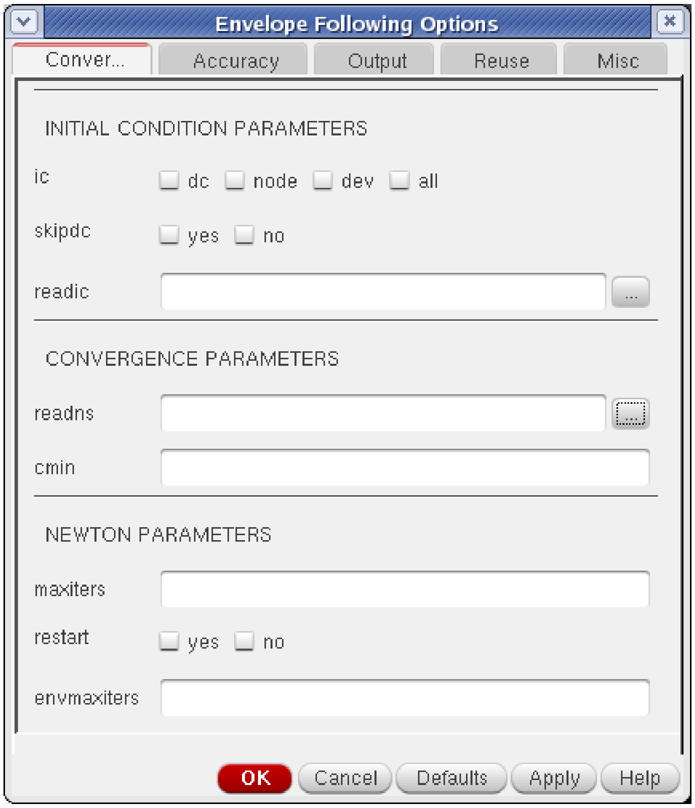
15. When you zoom in to the individual spikes, you will see multiple spikes. There will be either one or two phantom spikes on each side of the output sideband because of the DFT sample window used in the psd function. The number of spikes and the relative amplitude depend on the actual window function you selected. Remember that these spikes are not real. They are produced because of the window function itself.



Commonly Used Options

Options are provided to control the specific behavior of the simulation. This next section discusses the effect of the options in the options form for envelope. They are listed as they appear on the *Options* form.

Convergence Tab

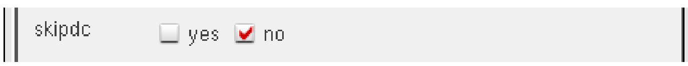


ic (Shooting and Harmonic Balance)

Initial conditions can be specified graphically by selecting *Simulation - Convergence Aids - Initial Condition* in the ADE. Initial conditions can also be specified from a file using the *readic* property. Capacitors and inductors have the initial condition properties in the property list for the component. For capacitors, this is an initial voltage across the capacitor, and for inductors, it is an initial current in the inductor. The default is to observe all the initial conditions in the DC analysis that is used as the time-zero time point. The initial conditions force a voltage or current to be present in the time-zero solution. The initial conditions are released for the rest of the transient simulation in the tstab interval. The *ic* option controls the initial conditions that should be observed in the time-zero time point. *all* is the default. *dev* means that only the initial conditions on capacitors and inductors are observed. *node* means that only the initial conditions on a node are observed. *dc* means that no initial conditions are observed. The example below shows *node*.

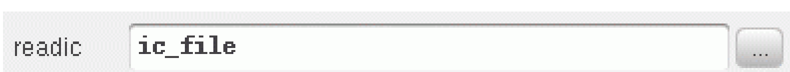
Skipdc (Shooting and Harmonic Balance)

In some cases, the time-zero time point DC analysis does not converge. Instead of stopping the simulation, *skipdc* allows the envelope simulation to continue using an assumed solution for the time-zero time point. The default is *no* and a DC analysis is run to get the initial time point. *yes* means skip the DC solution, and proceed directly to the envelope simulation. All the nodes with initial conditions specified start at the initial condition value. Nodes with batteries start at the battery voltage. Nodes with no initial conditions start at zero volts. For `skipdc=yes`, the signal sources start as specified immediately in the envelope simulation. The example below shows the *skipdc* option set to *no*. The default is *yes*.



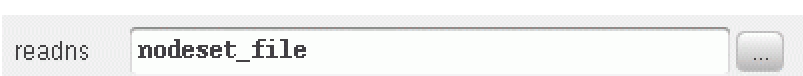
Readic (Shooting and Harmonic Balance)

This specifies an ascii file that contains two columns to be read as initial conditions. The left column is the node name. The right column is the voltage value. If the entry does not start with / (slash), the entry is located in the netlist directory. To find the netlist directory, select *Setup- Simulator/Directory/Host* in ADE. Look in the *Project Directory* field for the location of the simulation directory. Navigate to that directory and then to the *`<Circuit Name>`*`/spectre/`*`<schematic or config>`*`/netlist` directory. You can also click (...) and browse to the directory. An example is shown below. There is no default.



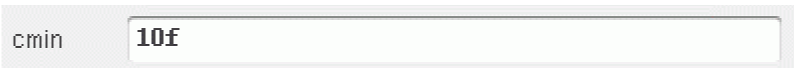
Readns (Shooting and Harmonic Balance)

This specifies an ascii file with the same format as an ic file that is used as nodesets for the time-zero DC solution. Nodesets do not force a voltage to be held for the time-zero solution. Instead, they are a way of speeding up the time-zero calculation. As a suggestion, set the *write* option and the *readns* option to the same filename. The *write* option writes the time-zero solution to a file. When this is used as a starting point, many fewer iterations are needed for the time-zero point to converge. An example is shown below. There is no default.



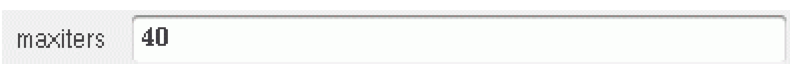
Cmin (Shooting and Harmonic Balance)

cmin adds a capacitor to ground from all nodes in the circuit to ground with the value specified in the *cmin* field. Note that if 1 is specified, a 1 farad capacitor is added from every node to ground in the circuit. Do not forget the multiplier suffix. Values from 1f to 100f are typically used to allow convergence. An example with *cmin* set to 10f is shown below. The default is 0.



Maxiters (Shooting and Harmonic Balance)

The transient analysis for tstab (for both shooting and hb) and shooting interval iterate to a solution at each time point. *maxiters* specifies the maximum number of iterations before the timestep is cut for another try at convergence. In some cases, model parameters can cause discontinuities in the device current or capacitance. If this occurs, the change in the circuit condition can be large enough to require more than the five iterations that are the default. Specifying *maxiters* in the 40 to 100 range usually allows the simulator to converge in spite of the discontinuity. An example with *maxiters* set to 40 is shown below.



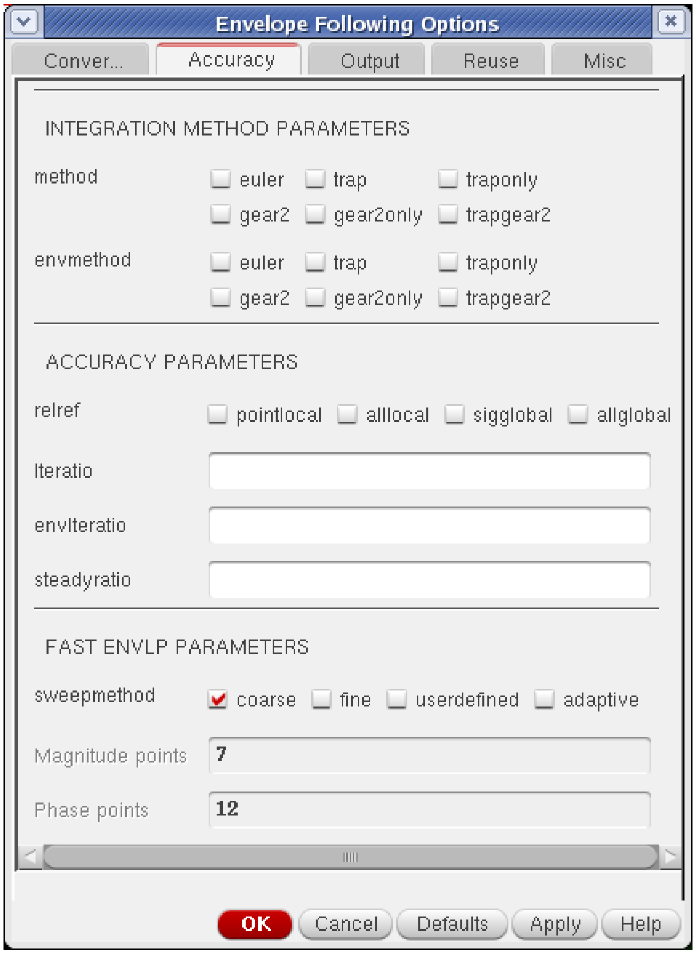
restart (Shooting and Harmonic Balance)

This option is used when a sweep is run. If restart is *no*, the time-zero time point does not restart from scratch. It restarts from the time-zero solution of the previous sweep value. *yes* forces a new time-zero solution to be re-run from scratch every time. The default is *no*.

envmaxiters (Shooting and Harmonic Balance)

The solution for each period of the carrier is an iterative solution. *envmaxiters* sets the maximum number of iterations that are allowed before the number of periods skipped is reduced. The default is three iterations for shooting and 40 for harmonic balance.

Accuracy Tab



method (tstab and shooting window for shooting, tstab only for Harmonic Balance)

This option controls the integration method for the shooting window. Note that in harmonic balance, the *method* option only applies in the tstab interval because the harmonic balance envelope solver works in the frequency domain. The default value for the shooting window is determined by the setting of *errpreset*. For *moderate* and *conservative*, the default is *gear2only*. For *liberal* it is *traponly*. Generally, *gear2only* is preferred because of the absence of trapezoidal ringing inherent in the *trap* (trapezoidal) method. Note that *liberal* is not recommended for RF simulation. When you simulate high Q oscillators, set *method* to *traponly*. This is needed because *gear2only* slightly numerically damps the oscillations.

envmethod (Shooting and Harmonic Balance)

This option controls the integration method for the part of the envelope analysis during the skipped cycles. In this part of the simulation, numerical integration still needs to be performed to get correct answers. For example, if there is a long time constant, the behavior on the node with the long time constant needs to be integrated even during the skipped cycles. *envmethod* controls this. The default is *gear2only* except for autonomous envelope (oscillators) where the numerical damping of *gear2only* is unacceptable. In this case, the default is *traponly*, which neither damps nor exaggerates the oscillations.

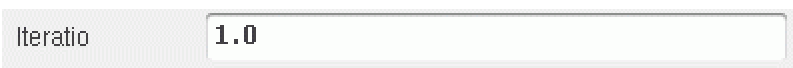
relref **(Shooting (tstab and shooting window) and Harmonic Balance (tstab only))**

In the order of most to least accurate, the *relref* settings are *pointlocal*, *alllocal*, *sigglobal*, *allglobal*. The default for *moderate* is *sigglobal* and for *conservative* it is *alllocal. relref* is used in the transient analysis in the tstab interval for both shooting and harmonic balance and in the shooting window for shooting. *relref* is used in hb only for the tstab interval because the solver works in the frequency domain. In some cases, the timestep can collapse to near zero. The symptom is that in the Spectre output window, the percent done number remains the same for many reporting intervals. The solution to this is to set *relref* to *sigglobal*, which is slightly less accurate. Another possibility is to use the *lteminstep* option as described in the *AdditionalParams* option of the *Misc* tab.

An example with *relref* set to *sigglobal* is shown below.

lteratio ((tstab and shooting window for shooting) and Harmonic Balance (tstab only))

lteratio is a multiplier for the allowable numerical integration error in the tstab interval and the shooting window. The default is 3.5 or 10 depending on the setting of *reltol* and *errpreset*. See Controlling Accuracy on page 571 for details. *lteratio* cannot be set smaller than 1.0 and is normally between 3.5 and 100. In some cases, the timestep collapses to near zero and setting *relref* to *sigglobal* does not fix the problem. In this case, there might be a model discontinuity. To test this, disable numerical integration timestep control by setting *lteratio* to 1e9. Note that this is an extreme measure not to be used under normal simulations. If the timestep still collapses, there is a discontinuity in one of the models in the circuit. This could be a device model or a Verilog-A model. If it is necessary to set *lteratio* very large to get the simulation to complete, you must also set *maxstep* small enough to preserve the accuracy of the simulation. An example with *lteratio* set to the smallest value of *1.0* is shown below. Note that *lteminstep* might also be used to treat this condition. See *AdditionalParams* in the *Misc* tab to see how to use this option.

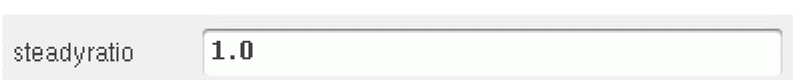


envlteratio (Shooting and Harmonic Balance)

envlteratio is a multiplier for the allowable numerical integration error in the numerical integration that occurs in the part of the waveform where cycles are skipped. The default is 0.35 for conservative and 3.5 for moderate. Liberal is not recommended.

steadyratio (shooting Only)

The maximum mismatch between the beginning and ending values for the voltages and currents in the shooting window is determined by the *convergence criteria * lteratio * steadyratio*. An example with *steadyratio* set to 1.0 is shown below. The default is 0.1 for moderate, and 1.0 for conservative.



sweepmethod

When fast envelope is used, a series of harmonic balance simulations are run. The amplitude is swept over a range of values present in the I and Q modulation files. When the baseband model is used in fast envelope, the phase also needs to be swept. The sweepmethod option controls the density of the sweep points. The default is coarse, where 7 amplitude and 12 phase points are used. This is sufficient for most simulations. Fine uses 10 amplitude points and 16 phase points. Userdefined allows you to specify the number for both magnitude and phase points. Adaptive is a work in progress.

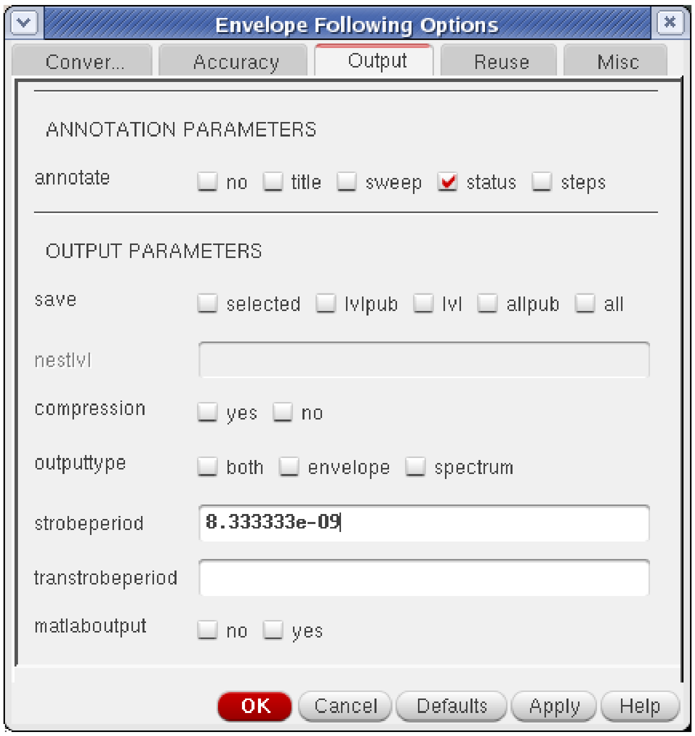
Magnitude points

When sweepmethod is set to userdefined, Magnitude points specifies the number of different amplitude points to use in the characterization phase of fast envelope.

Phase points

When sweepmethod is set to userdefined, and there is a baseband source in the circuit, this option controls the number of phase points to use in the characterization phase of fast envelope.

Output Tab



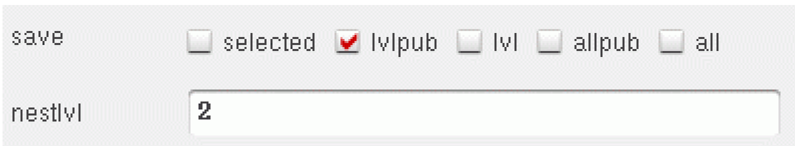
annotate

Annotate (Shooting and Harmonic Balance)

This option controls the level of detail in the output log. The default is *status.* The level of detail increases as you move to the right.

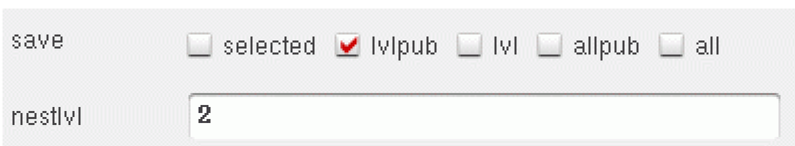
Save (Shooting and Harmonic Balance)

The default is *allpub*. This saves all the public voltages at all levels of the hierarchy in the schematic. It excludes the internal nodes of the device models. *all* adds the internal nodes of all the devices. *lvl* saves all the nodes including the internal nodes of the devices through the level of hierarchy set in the *nestlvl* option. *lvlpub* is like *lvl*, but it does not save the internal nodes of the devices. *selected* saves only the nodes that are specifically saved. In ADE, this is accomplished using the *Outputs - To Be Saved - Select On Schematic* menu option and then selecting the nodes and terminals specifically in the schematic. At the netlist level, this is accomplished by using a save statement with a list of names to be saved.



Nestlvl (Shooting and Harmonic Balance)

If *save* is set to *lvl* or *lvlpub*, this controls the maximum level of hierarchy to be saved. If *nestlvl* is *1*, only the top level is saved. If *nestlvl* is 2, the top level and the next level down are saved. The value for *nestlvl* can be any integer. The example below shows *lvlpub* selected that saves two levels of hierarchy. There is no default for *nestlvl*.



Compression (Shooting and Harmonic Balance)

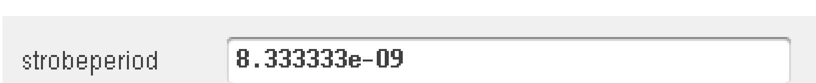
Do not use this option.

outputtype (Shooting and Harmonic Balance)

This option is rarely used. It controls which simulation results to save. Normally, both time and frequency domain data is saved. Envelope saves the time-domain data only. Spectrum saves just the frequency-domain data. The default is to save both time and frequency-domain data. Note that in fast envelope, you need to select the *yes* check box to the right of *Output time-domain waveform and all harmonics* in order to calculate time-domain output.

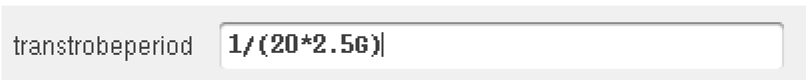
strobeperiod (Shooting and Harmonic Balance)

This option specifies the envelope time in seconds to force cycles of the carrier to be simulated. Shooting will round this time to an integer number of periods of the carrier. This is used to force data at the times needed for the psd calculation. There is no default value.



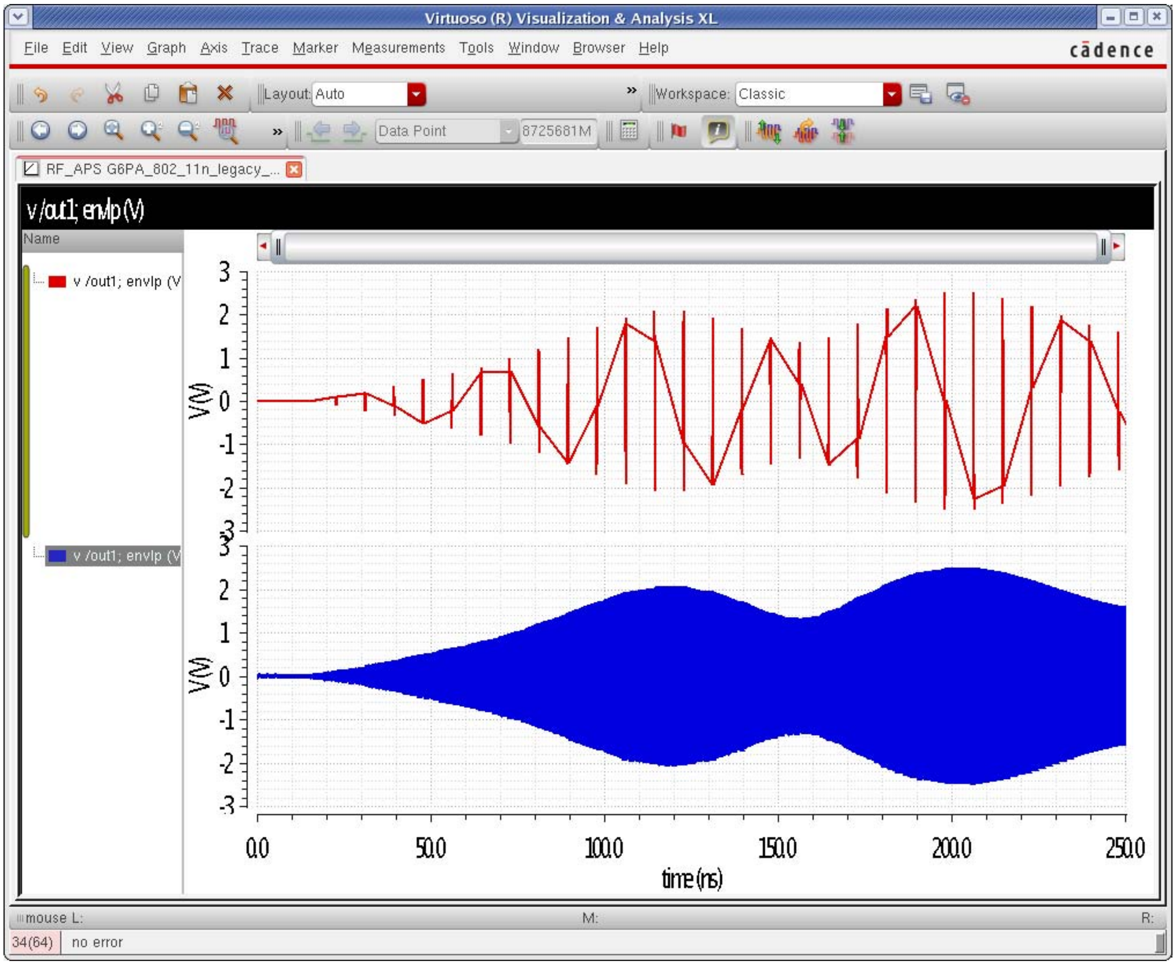
transtrobeperiod (Shooting and Harmonic Balance)

Normally, the time-domain waveform is a cycle of carrier followed by many skipped cycles, followed by another cycle of simulation. Since there is no waveform between the last point for the first cycle and the first point of the second cycle, the waveform tool just connects those points together with a straight line. The waveform can be made into a continuous waveform using the *transtrobeperiod* option. This should be set to an integer fraction of one period of the carrier frequency. When this is set, Spectre still skips cycles, but it also creates an interpolated, continuous waveform with time points at the *transtrobeperiod* interval. In the example below, 20 points are set for one period of the carrier at 2.5G.

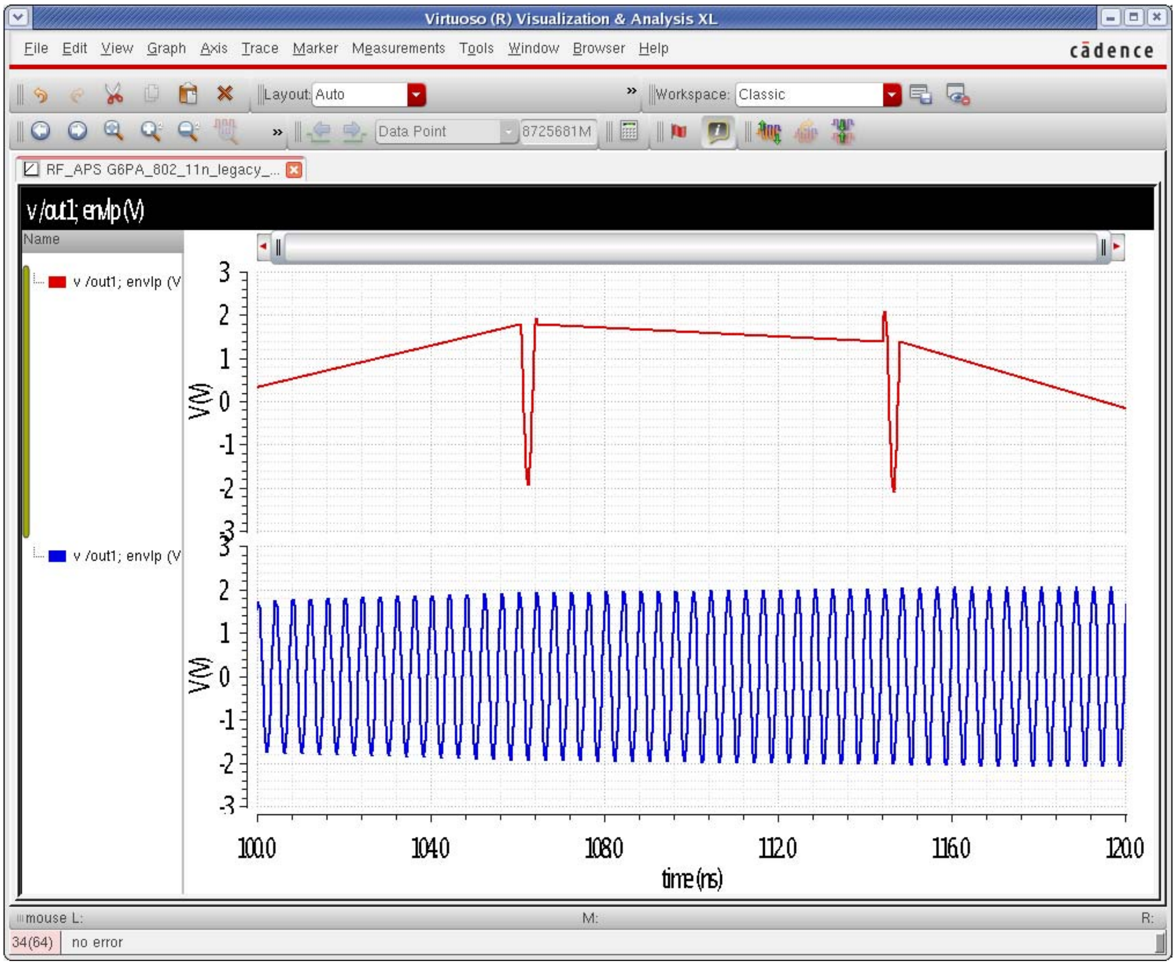


Spectre does take expressions, as shown above. There is no default value for *transtrobeperiod*.

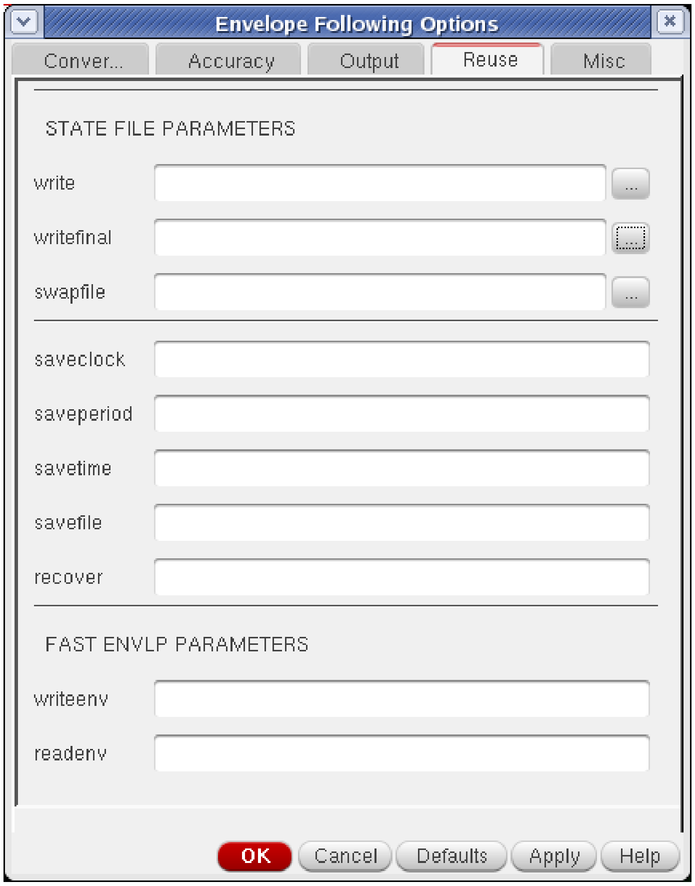
Here is a comparison of the 2 waveforms.



The red waveform is the normal waveform. The vertical spikes in the red waveform are the individual periods of the carrier that are normally calculated. When *transtrobeperiod* is set, the waveform becomes continuous, as shown in blue. A closer view is shown below.

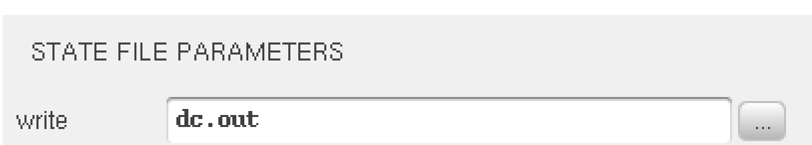


Reuse Tab



Write (Shooting and Harmonic Balance)

write specifies a filename to which the DC solution used as the first time point be written to. If the name does not start with slash (/), the file is written in the *netlist* directory. To determine the netlist directory location, select *Setup - Simulator/Directory/Host* in ADE. The *Project Directory* field lists the location of the simulation directory. From this directory, the netlist directory is in <`Circuit Name`>/<`simulator name`>/*<schematic or config>/netlist*. There is no default value.



Writefinal (Shooting Only)

writefinal writes the final envelope time point from the last period of the envelope analysis (not the final time point of the tstab interval) to the filename specified. See the description for *write* for the location. There is no default value.

Swapfile (Shooting Only)

This option is not currently implemented.

Saveclock (Shooting and Harmonic Balance)

This option is not currently implemented.

Saveperiod (Shooting and Harmonic Balance)

This option is not currently implemented.

Savetime (Shooting and Harmonic Balance)

This option is not currently implemented.

Savefile (Shooting and Harmonic Balance)

This option is not currently implemented.

Recover (Shooting and Harmonic Balance)

This option is not currently implemented.

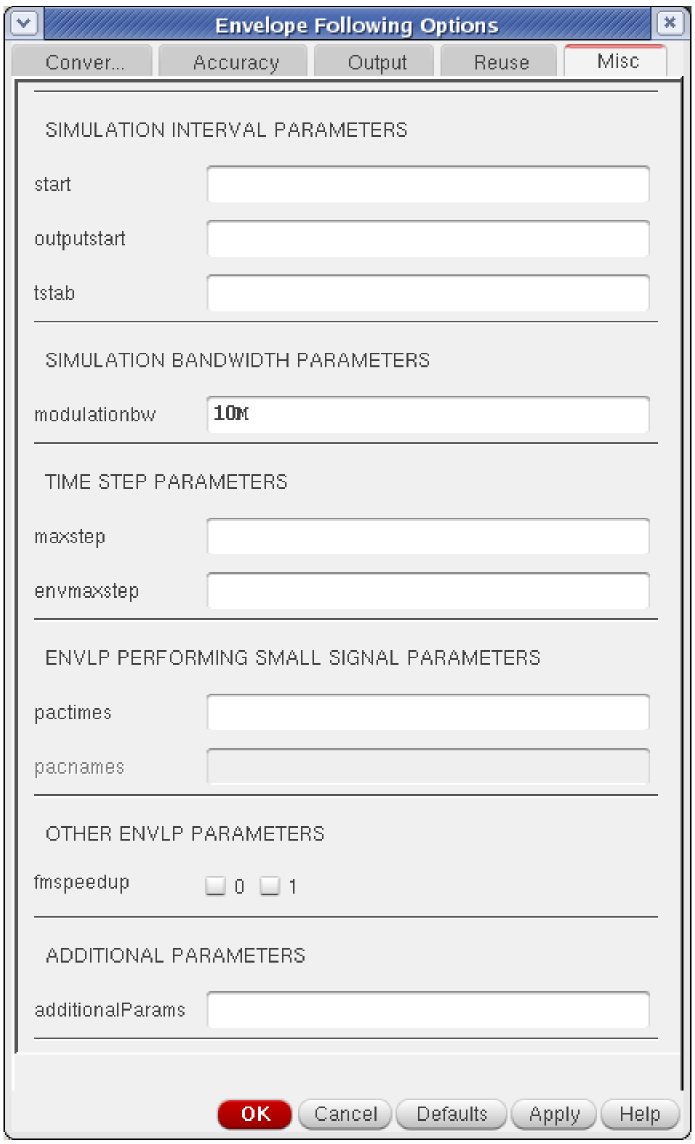
Writeenv

When fast envelope is used, the behavioral model can be written to a file and reused. Writeenv specifies the filename to save the behavioral model for fast envelope in.

Readenv

When fast envelope is used, the behavioral model can be written to a file and reused. Readenv specifies the file to read back in instead of running the characterization from scratch.

Misc Tab



start (Shooting and Harmonic Balance)

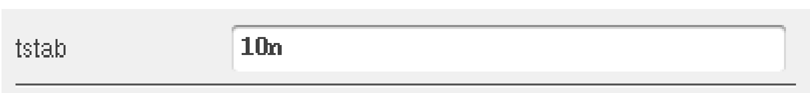
This option is rarely used. It defines the starting time for the envelope simulation. The default is 0.

outputstart (Shooting and Harmonic Balance)

This option is rarely used. This defines the simulation time to begin to save data to the output file. The default is to start writing data at time 0.

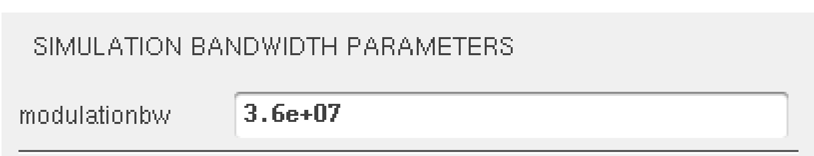
tstab (Shooting and Harmonic Balance)

This is seldom used in envelope simulation. This specifies a simulation time for the transient analysis to run before starting the envelope simulation. The default is to run 3 periods of the carrier signal.



modulationbw (Shooting and Harmonic Balance)

modulationbw and *envmaxstep* accomplish the same thing, which is to define a maximum time interval to skip between carrier cycles that are simulated. When *modulationbw* is set, at least eight periods of the carrier are simulated in the time period of one cycle of the frequency *modulationbw*. *modulationbw* does not force data at equally spaced intervals, like *envstrobeperiod* does. Instead, it sets a a maximum delta-time for skipping cycles. There is no default.

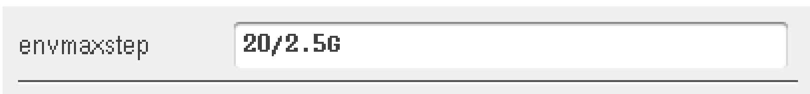


maxstep (Shooting: tstab and waveform calculation. Harmonic balance: tstab interval only)

This option is rarely used. This sets the maximum delta-time for the shooting waveform in each cycle of the carrier. The default forces at least 20 time points for moderate, and 50 time points for conservative. It also is used in the transient analysis in the tstab interval.

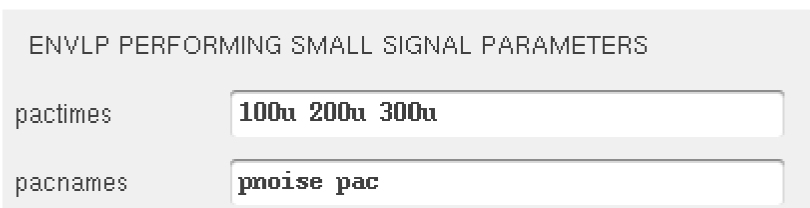
envmaxstep (Shooting and Harmonic Balance)

envmaxstep and *modulationbw* accomplish the same thing, which is to define a maximum time interval to skip between carrier cycles that are simulated. *envmaxstep* sets the maximum time between periods of the carrier that can be skipped. *envmaxstep* does not force datapoints at regularly spaced intervals. Instead, it sets a a maximum delta-time for skipping cycles. The default forces at least 25 periods to be simulated when *moderate* is selected, and 50 when *conservative* is selected.



pactimes (Shooting and Harmonic Balance)

During the transistor-level envelope simulation, one or more of the carrier periods that are simulated can be used as the nonlinear operating point for the small-signal pac, pxf, pnoise, or psp analyses. *pactimes* is a list of times separated by a space for the large-signal envelope solution that should be used for the small-signal analyses. The default is to not use any points for the small-signal analyses.



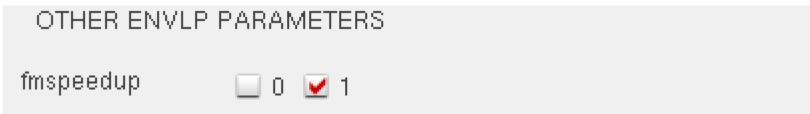
pacnames (Shooting and Harmonic Balance)

This option is used in combination with *pactimes*. Specify a list of the small-signal analyses pac, pxf, pnoise, and/or psp in this option field, and these analyses will be run at the times specified in *pactimes*. The default is to not run any small-signal analyses.

When this option has a list of small-signal analyses, the corresponding pac, pxf, pnoise, and psp analyses must also be set up and enabled.

fmspeedup (Shooting and Harmonic Balance)

This option allows the period of the carrier cycle to vary because of frequency modulation. When *fmspeedup* is set to the default of 0, the period of the carrier is not allowed to vary. When it is set to 1, the period is allowed to vary.



AdditionalParams (Shooting and Harmonic Balance)

This is a field for `<option_name>=value` statements. Multiple `<option_name>=value` statements are allowed with a space between them. Generally, this is used to unlock the beta features, however, there is one case where this field might be useful for an option that is not available in the GUI.

In some cases, in the tstab interval, the timestep collapses to near zero. The symptom is that the percent complete stays at the same value for many reporting intervals. In this case, there might be a discontinuity in one of the models.

In transient simulation, two things can reduce the timestep; having too much numerical integration error or taking too many iterations at a single timestep can cause the timestep to be cut. Spectre does not report what is causing the timestep to become small. One way to eliminate the possibility of numerical integration error (which is called local truncation error in the simulator) as the cause of reducing the timestep is to set *lteratio* to 1e9. The disadvantage of this is that it disables the normal method of timestep control, therefore, *maxstep* must be set to maintain accuracy.

Another way to accomplish this is to set *lteminstep* to a value of about the stop time divided by 1e5. This allows the normal method of timestep control, but if for some reason the numerical integration error wants to cut the timestep too small, the error is ignored. *lteminstep* is the way to specify this. To do this in the ADE, type `lteminstep=<Stop_Time>/1e5`. If you use `lteratio=1e9`, or if you set *lteminstep* and you still have very small timesteps, try increasing *maxiters* in the *Convergence* tab to between 40 and 100. When *lteratio* or *lteminstep* is set and you still have timestep problems, there is likely a discontinuity in either a device model or in a Verilog-A model. An example is shown below.

Autonomous ENVLP Analysis (Oscillators)

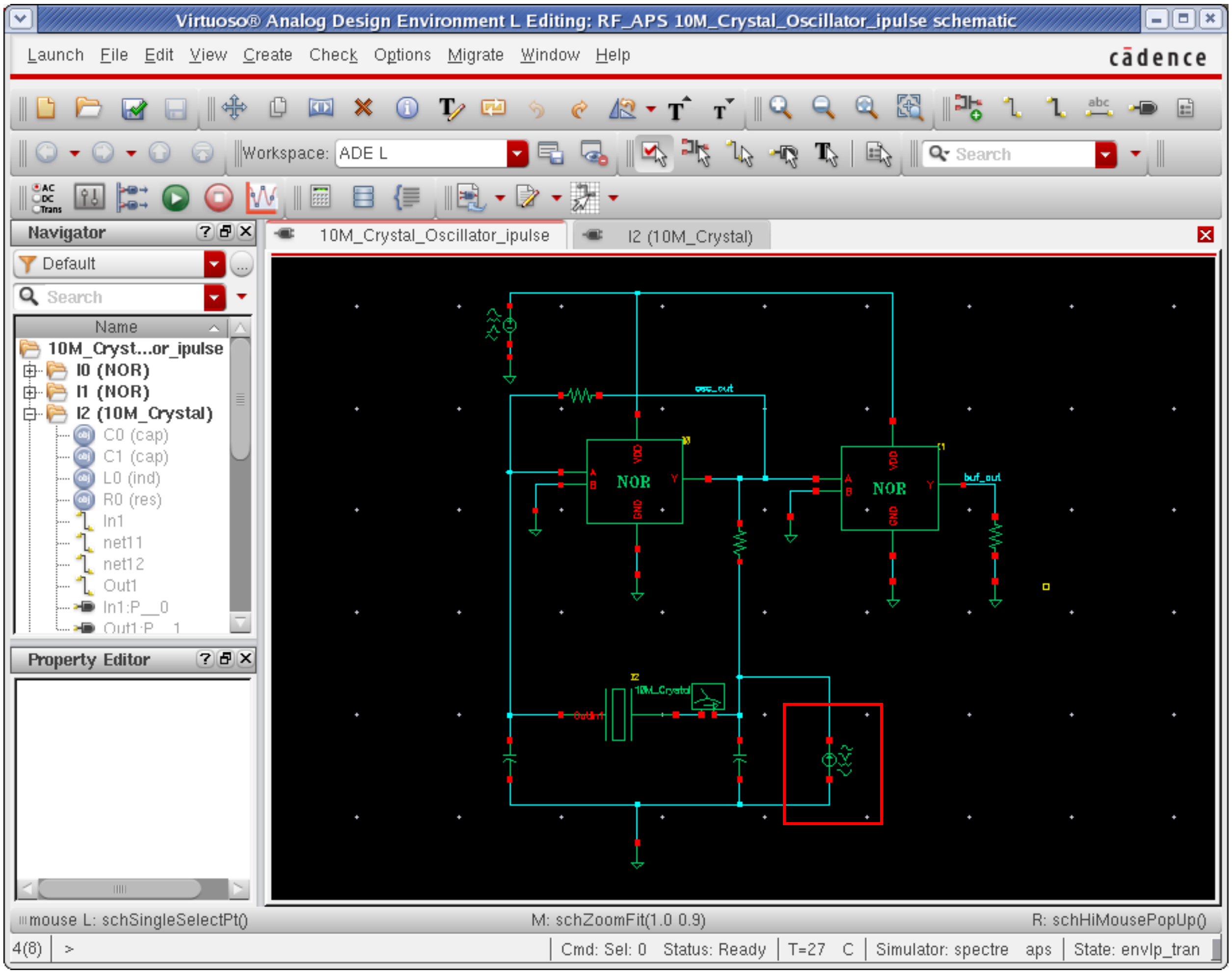
Autonomous envelope is useful for simulating the startup behavior of high Q oscillators, and for simulating the frequency pulling and/or pushing effects caused by random data being applied to a system. In autonomous envelope, the period and amplitude of each cycle of the oscillator that is simulated needs to be solved.

In general, harmonic balance is suggested for LC oscillators, and shooting should be used for ring oscillators. Crystal oscillators should use harmonic balance.

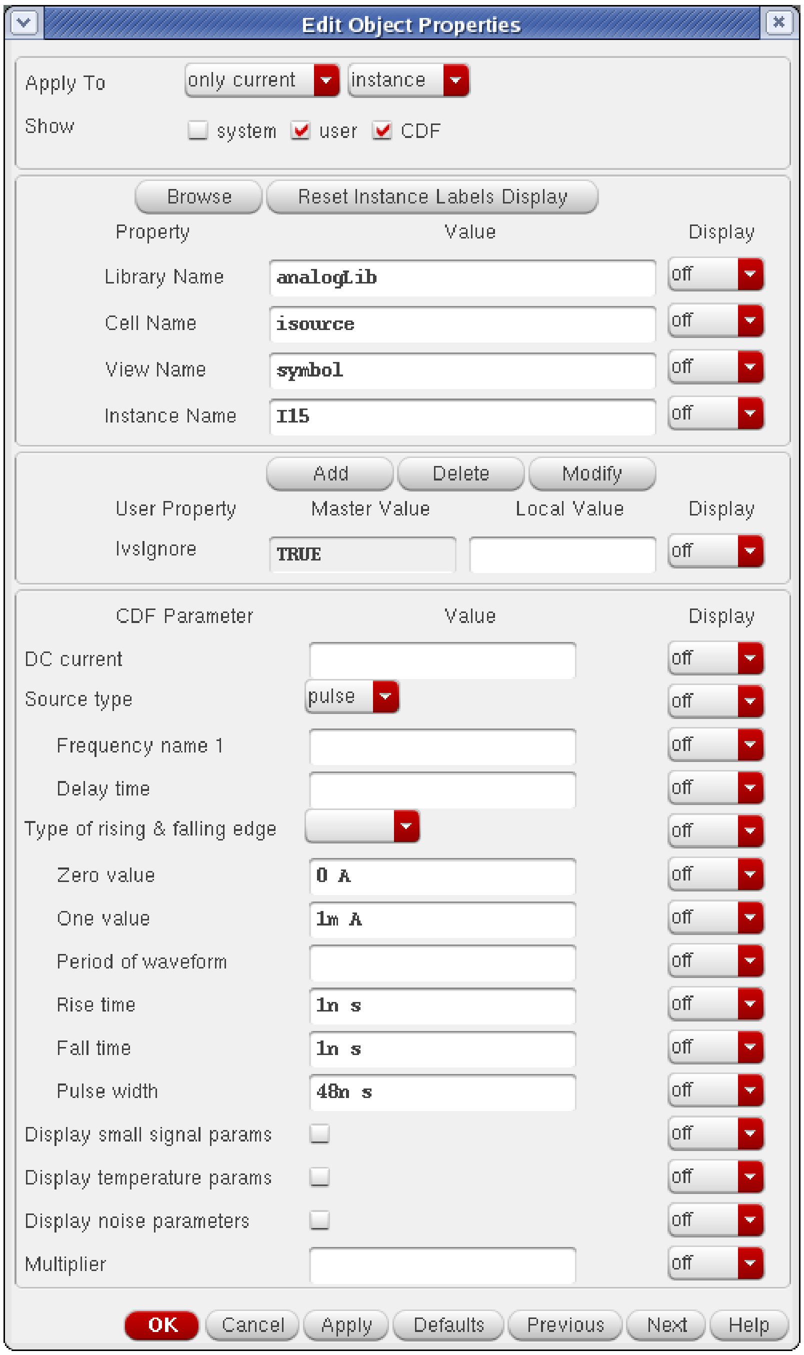
The only additional requirement in the *Choosing Analyses* form for oscillators is the requirement to select the check box for oscillators, and specify a node or pair of nodes that has the oscillator signal on it. If the *osc initial condition* check box is set to *linear*, the nodes must be in the resonator of the oscillator. Although not a requirement, it is recommended that you use adaptive timestep control, and set the *modulationbw* (modulation bandwidth) option to the frequency of the oscillator divided by the Q of the oscillator.

Example: Startup behavior of a crystal oscillator

Here is an example circuit.

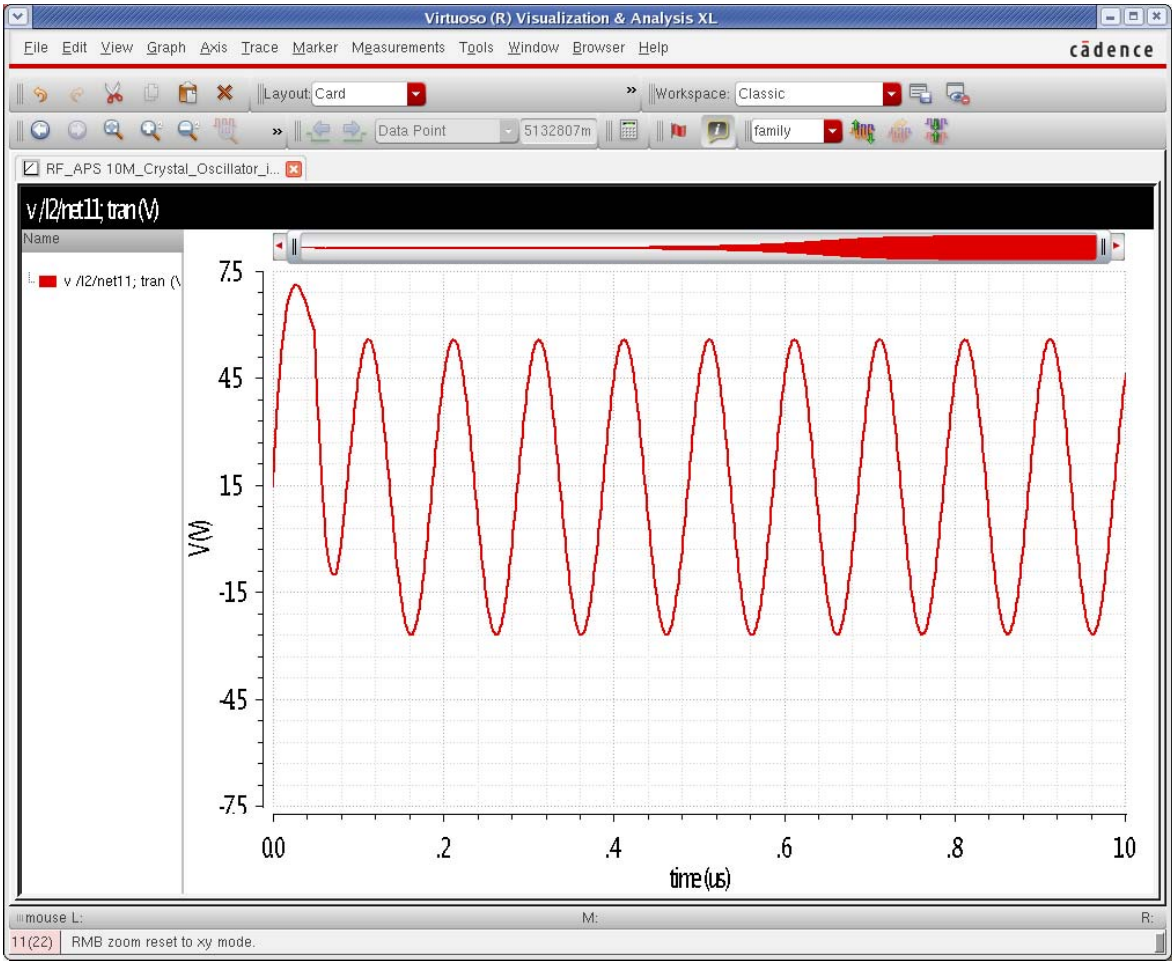


First, add an isource, set to *Source type* to *pulse*, and connect it to the resonator as highlighted above. The properties list is shown below.



Next, set the properties.

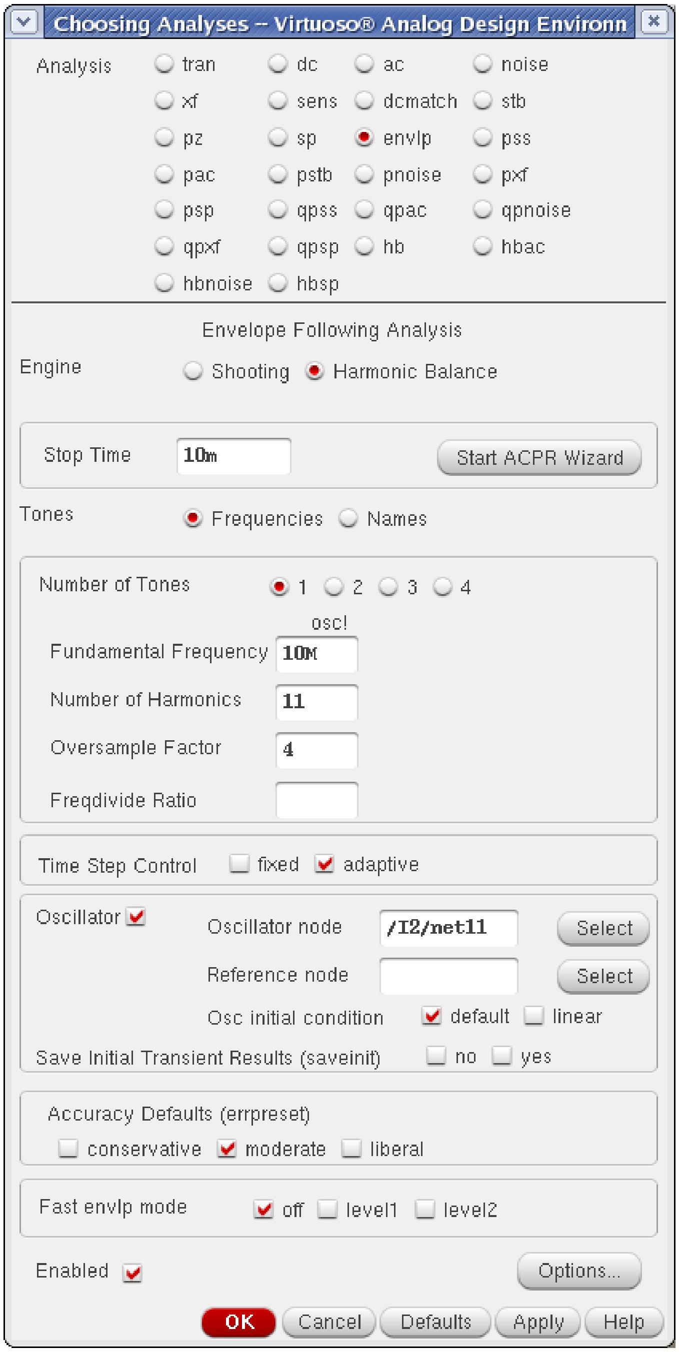
1. Set the *Zero value* to 0 (zero). The pulse starts at the zero value. With zero amps, the isource is a perfect open, so it does not load the circuit.
2. Set the *One value* to a current that starts the oscillator, but does not drive it too hard at the start. You will need to experiment with this value using the transient. In the waveform window below, the junction of the inductor and capacitor in the crystal motional equivalent circuit is plotted. This voltage will get quite large because of the high Q. Change the *One value* so that one to ten volts peak is generated at this node at the start of the transient simulation, and no huge transients are generated.



3. Set the *Rise time* and *Fall time* to about 1/10th to 1/100th of the period. This is a 10MHz crystal oscillator. In this case, 1/100th of the period was used.
4. Set the *Pulse width* so that half a period is generated by the pulse. Half the period of 10M is 50nsec, so the total of *Rise time*, *Fall time*, and *Pulse width* is 50nsec.
5. Do not make an entry in the *Period of waveform* property.

The effect of these entries is to set up a single current pulse to start the oscillator.

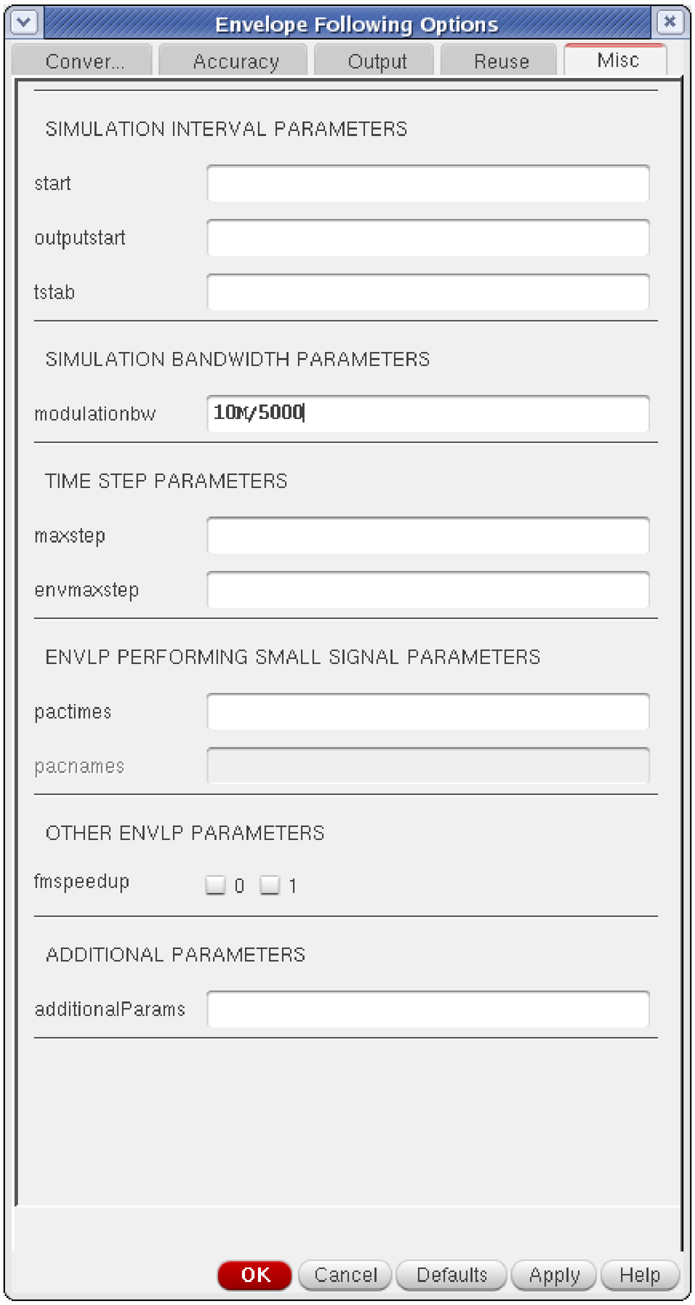
Now, set up the envelope *Choosing Analyses* form.



1. Select the appropriate engine. For LC and crystal oscillators, select *Harmonic Balance*. For ring oscillators select *Shooting*.

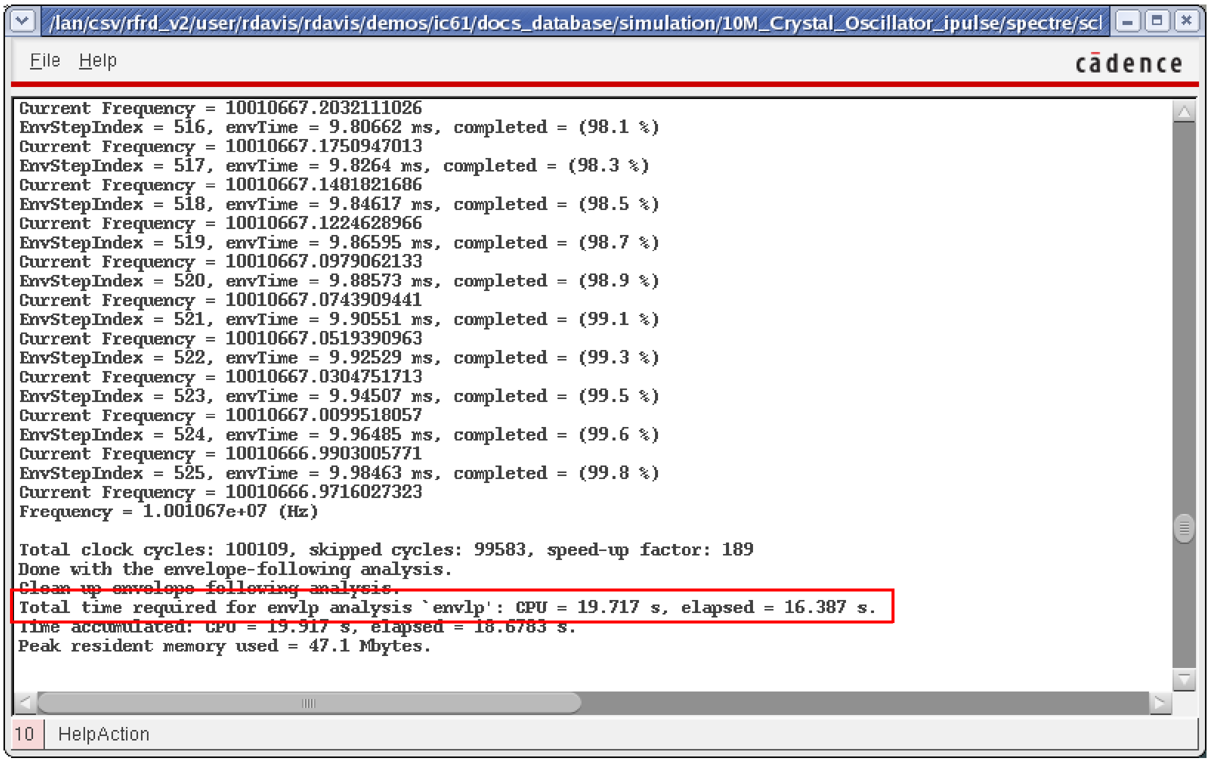
2. Set an appropriate *Stop Time* for your simulation. The time chosen will depend on the Q of the oscillator. About five times the Q periods need to be simulated.
3. Set the *Fundamental frequency*, *Number of Harmonics*, and *Oversample Factor* as needed for your simulation. See the discussion about setting harmonics and oversample factor in *Chapter 3: Harmonic Balance*.
4. Select *adaptive Time Step Control*. This allows the simulator to place the periods of oscillator simulation at the appropriate place.
5. Enable the check box next to the *Oscillator* Property, and specify a node inside the resonator for the oscillator node(s).
6. Set *Osc initial condition* to *default*. This ensures that the oscillator starts up as a result of the current pulse applied to the resonator.
7. Select *moderate* accuracy. In virtually every case, this is accurate enough.
8. *Fast envlp mode* must be set to *off*. Fast envelope is not available for oscillators.

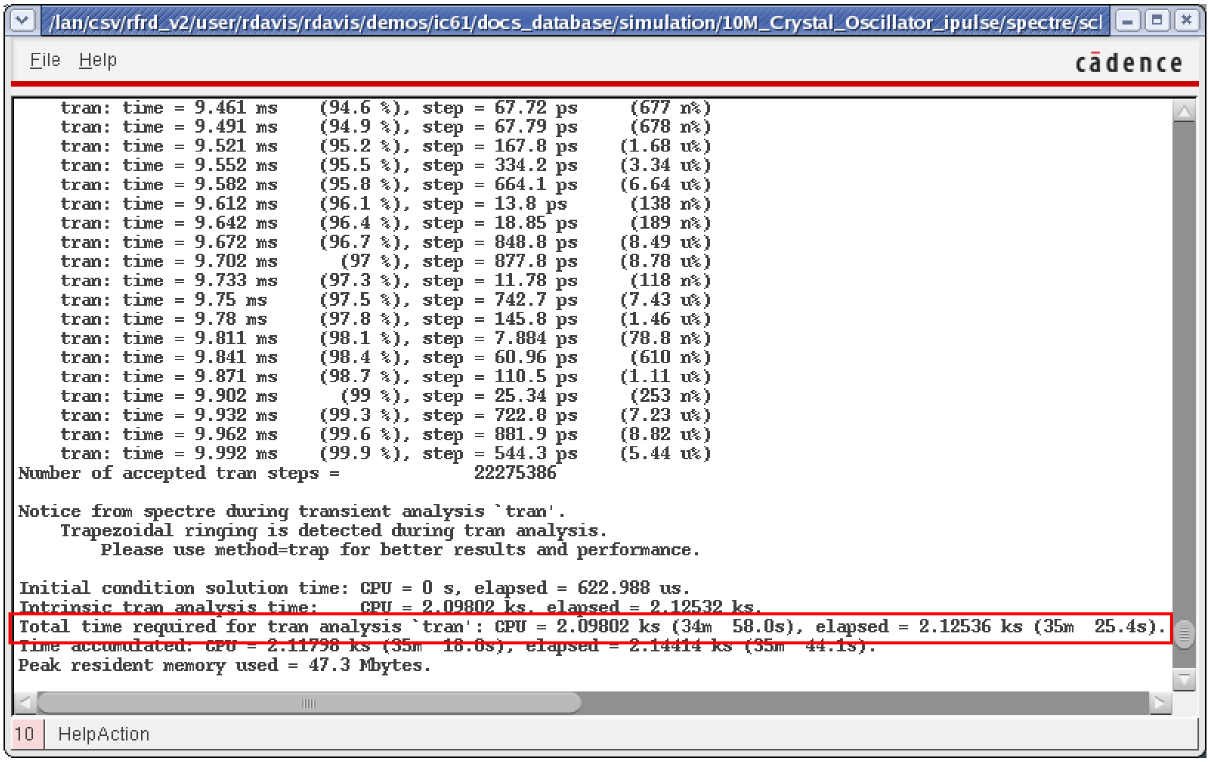
Options are not required, but are suggested. Click *Options* at the bottom of the *Choosing Analyses* form.



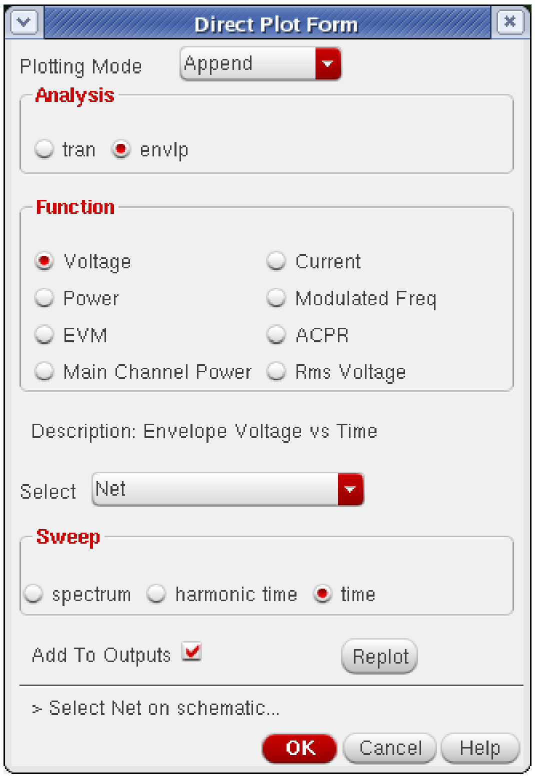
Set the *modulationbw* (modulation bandwidth) property to the period divided by the Q of the oscillator. Expressions are allowed by Spectre.

Now run the simulation. Below is a comparison of the runtimes for the envelope analysis and for the transient analysis. Note that the runtime is much shorter for the envelope analysis.



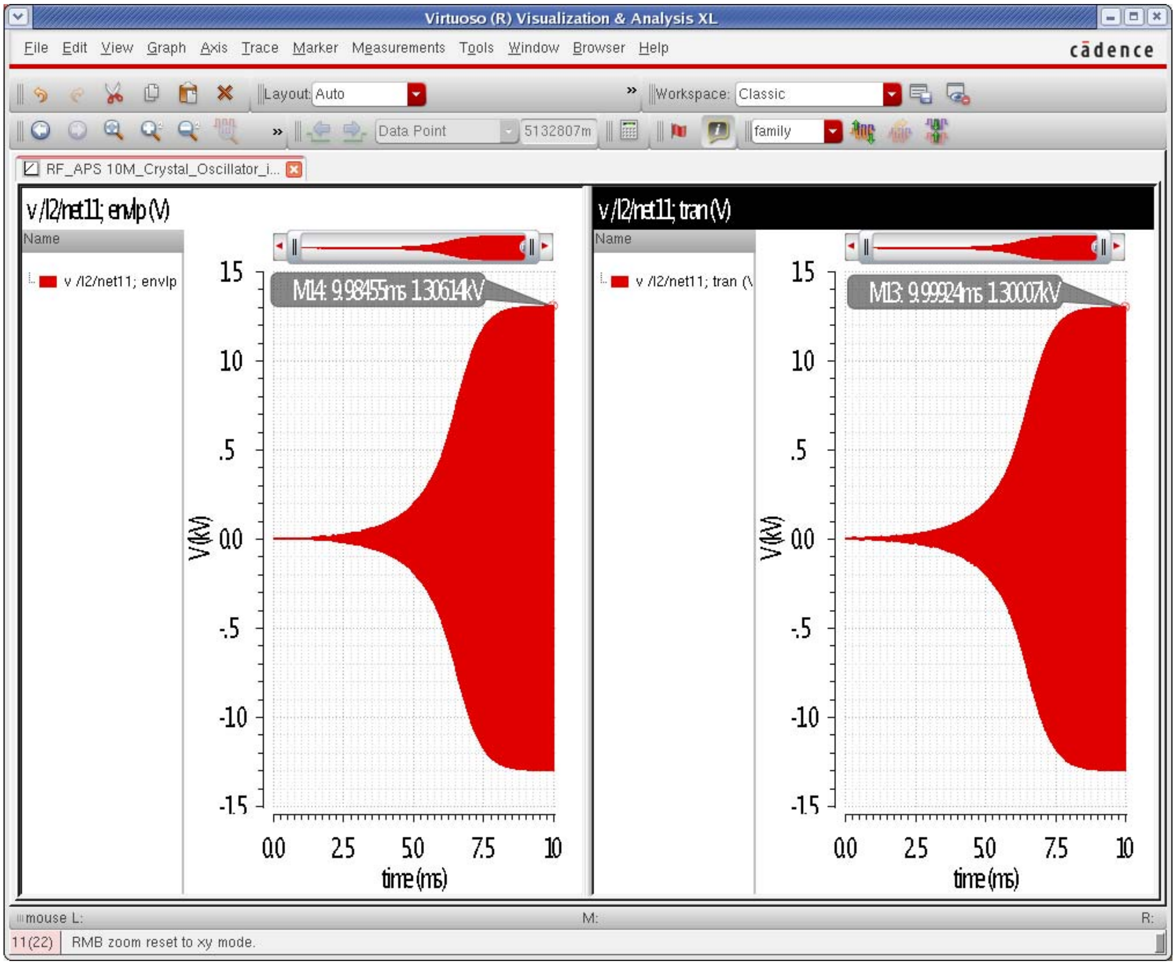

/lan/csv/rfrd_v2/user/rdavis/rdavis/demos/ic61/docs_database/simulation/10M_Crystal_Oscillator_ipulse/spectre/scl
File Help
cādence
tran: time = 9.461 ms (94.6 %), step = 67.72 ps (677 n%)
tran: time = 9.491 ms (94.9 %), step = 67.79 ps (678 n%)
tran: time = 9.521 ms (95.2 %), step = 167.8 ps (1.68 u%)
tran: time = 9.552 ms (95.5 %), step = 334.2 ps (3.34 u%)
tran: time = 9.582 ms (95.8 %), step = 664.1 ps (6.64 u%)
tran: time = 9.612 ms (96.1 %), step = 13.8 ps (138 n%)
tran: time = 9.642 ms (96.4 %), step = 18.85 ps (189 n%)
tran: time = 9.672 ms (96.7 %), step = 848.8 ps (8.49 u%)
tran: time = 9.702 ms (97 %), step = 877.8 ps (8.78 u%)
tran: time = 9.733 ms (97.3 %), step = 11.78 ps (118 n%)
tran: time = 9.75 ms (97.5 %), step = 742.7 ps (7.43 u%)
tran: time = 9.78 ms (97.8 %), step = 145.8 ps (1.46 u%)
tran: time = 9.811 ms (98.1 %), step = 7.884 ps (78.8 n%)
tran: time = 9.841 ms (98.4 %), step = 60.96 ps (610 n%)
tran: time = 9.871 ms (98.7 %), step = 110.5 ps (1.11 u%)
tran: time = 9.902 ms (99 %), step = 25.34 ps (253 n%)
tran: time = 9.932 ms (99.3 %), step = 722.8 ps (7.23 u%)
tran: time = 9.962 ms (99.6 %), step = 881.9 ps (8.82 u%)
tran: time = 9.992 ms (99.9 %), step = 544.3 ps (5.44 u%)
Number of accepted tran steps = 22275386
Notice from spectre during transient analysis `tran'.
Trapezoidal ringing is detected during tran analysis.
Please use method=trap for better results and performance.
Initial condition solution time: CPU = 0 s, elapsed = 622.988 us.
Intrinsic tran analysis time: CPU = 2.09802 ks, elapsed = 2.12532 ks.
Total time required for tran analysis `tran': CPU = 2.09802 ks (34m 58.0s), elapsed = 2.12536 ks (35m 25.4s).
Time accumulated: CPU = 2.11798 ks (35m 18.0s), elapsed = 2.14414 ks (35m 44.1s).
Peak resident memory used = 47.3 Mbytes.
10 HelpAction

Now use the *Direct Plot Form* to plot the waveform. In ADE, select *Results - Direct Pot - Main Form*.



1. Select *envlp* results.
2. Select *Voltage*.
3. Select *Net* or *Differential Nets*.
4. Select *time*.

5. Select a net in the schematic. The waveform is displayed.

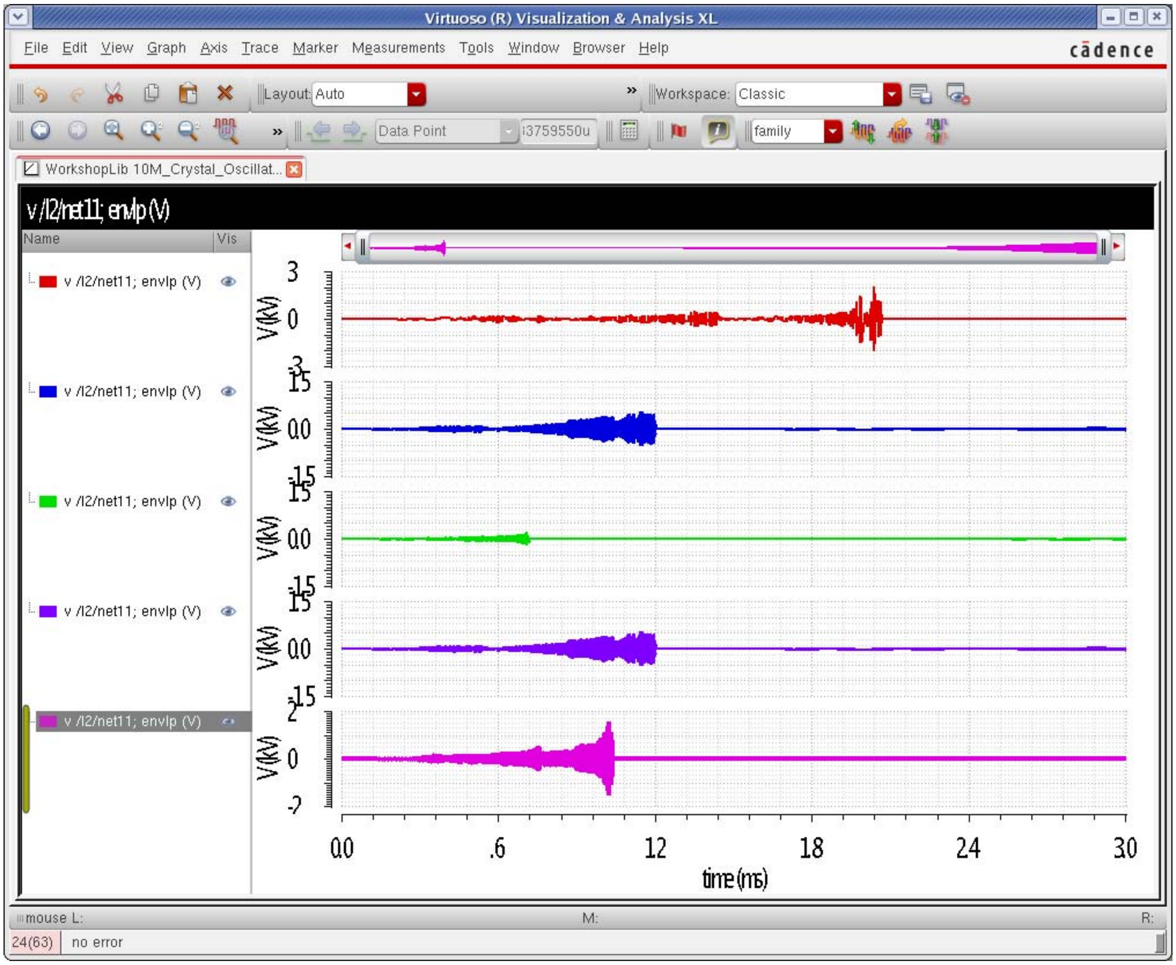


The waveforms above are the envelope results on the left, and the transient results on the right. The results are almost identical for both, and envelope takes 16.4 seconds versus 2125.4 seconds for the transient on the same machine.

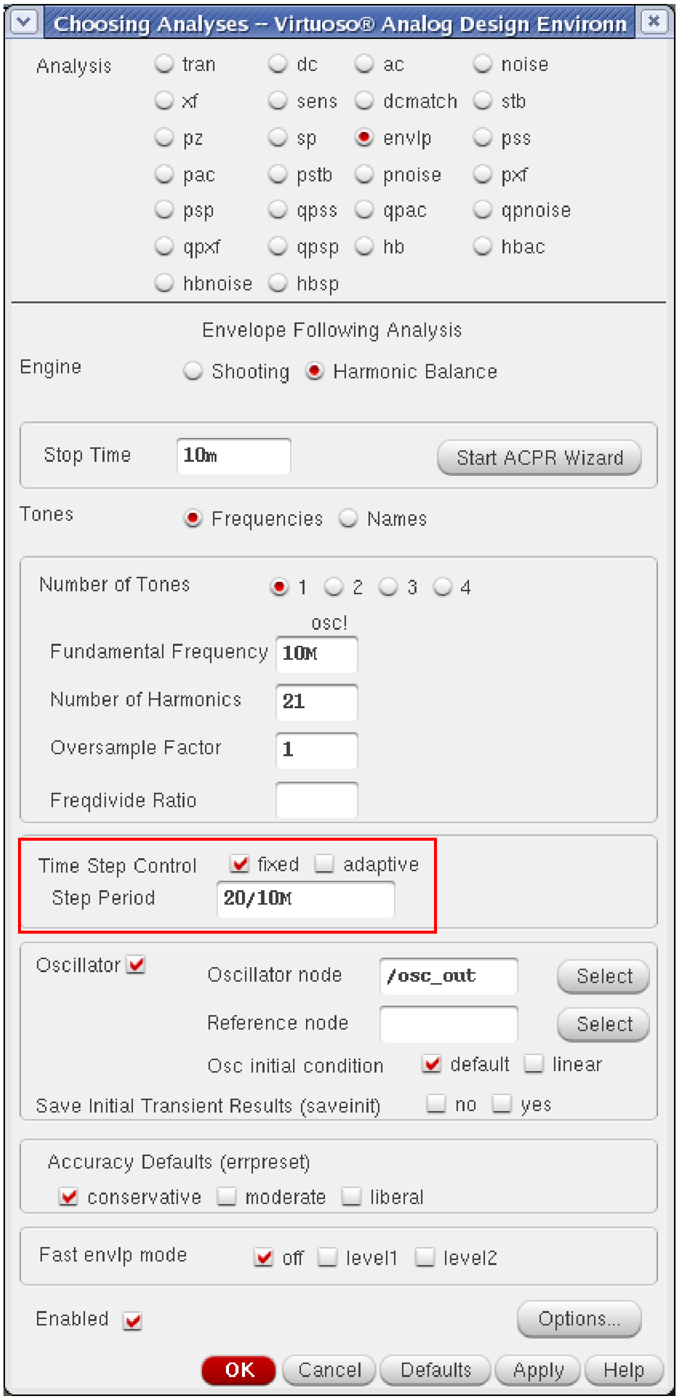
Common Problems With Oscillator Startup

In some cases, the oscillator will have a hard time starting up. You can recognize this by looking at the waveform in the resonator of the oscillator. Normally, the junction of the capacitor and the inductor in the crystal equivalent circuit will have voltages in the order of

KiloVolts peak at steady-state. When you see the ragged startup waveforms, as shown below, set a fixed timestep with about 15 to 100 periods as the step interval.



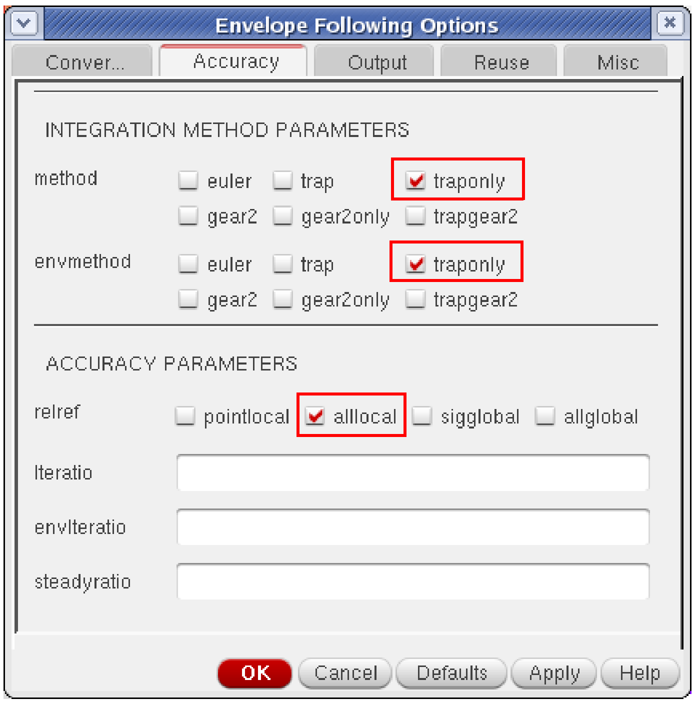
Spectre takes expressions in the entries, so it makes the period calculation easy.



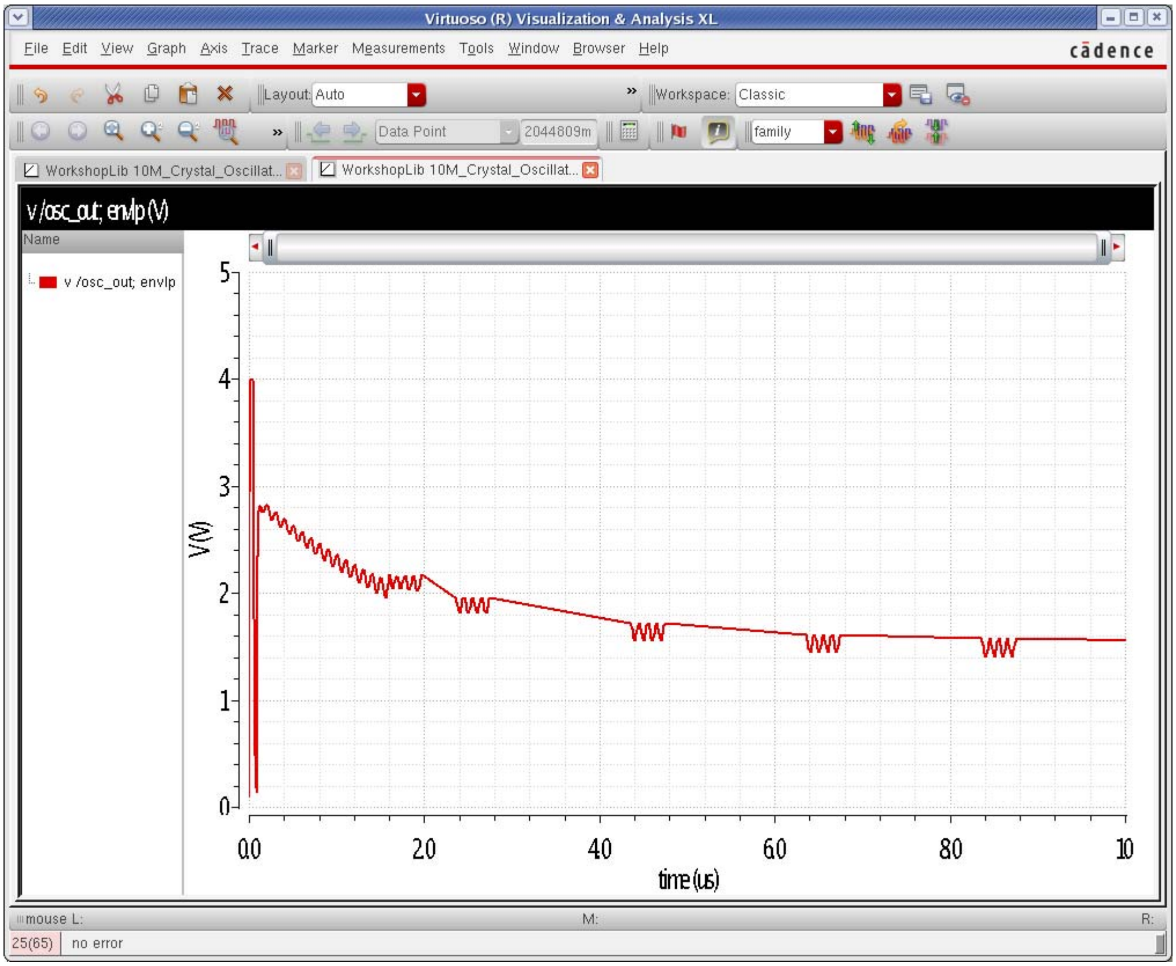
The example above sets 20 periods of 10MHz as the step interval.

Also, set the following analysis options:

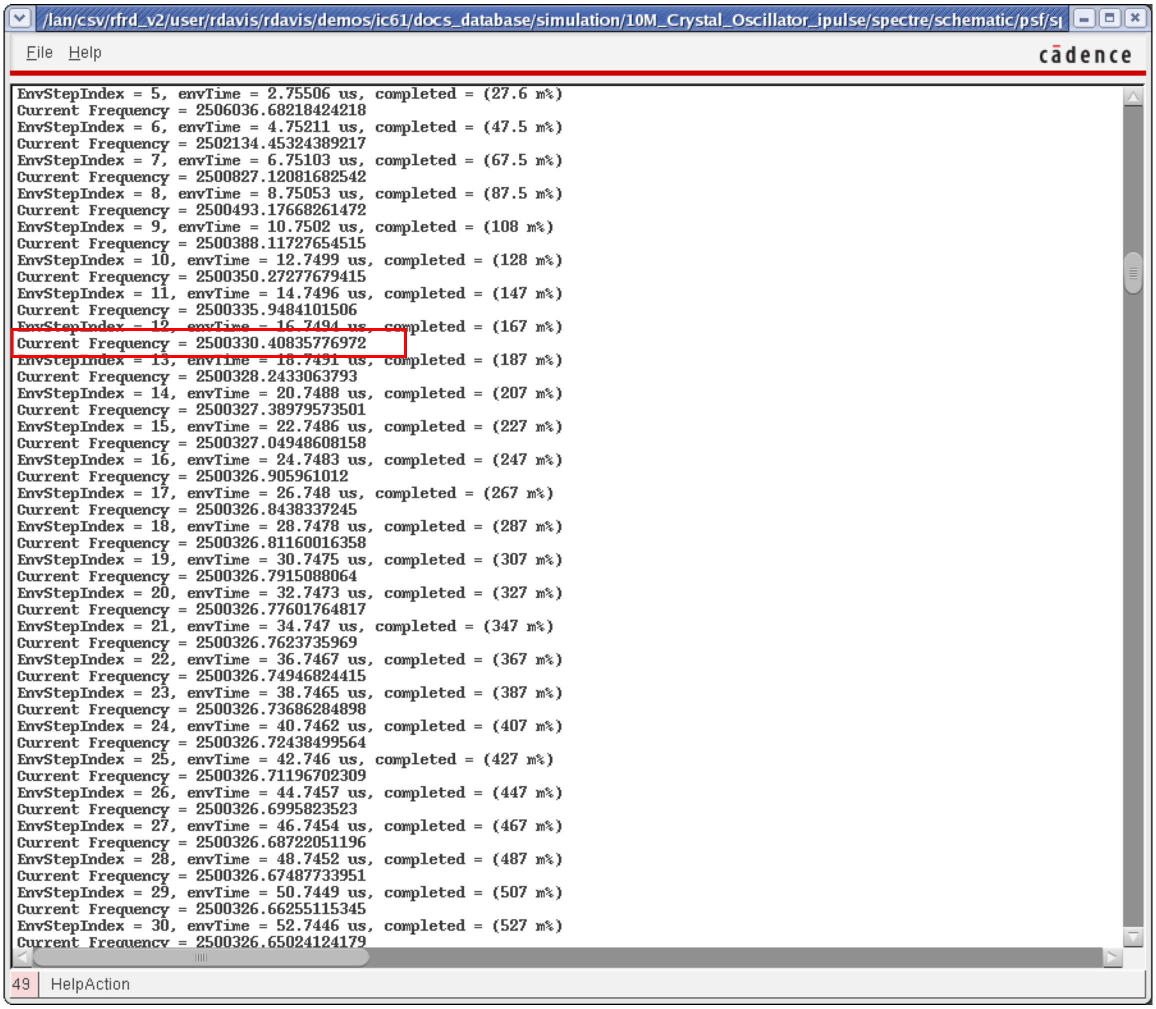
Set the *method* and *envmethod* options to *traponly*. *trap* does not damp or emphasize the oscillations of the oscillator. The default is *gear2only*, which slightly damps the oscillations. Also, set *relref* to *alllocal*. This is needed because of the mix of very large amplitude signals on the resonator with much smaller oscillations in the rest of the circuit.



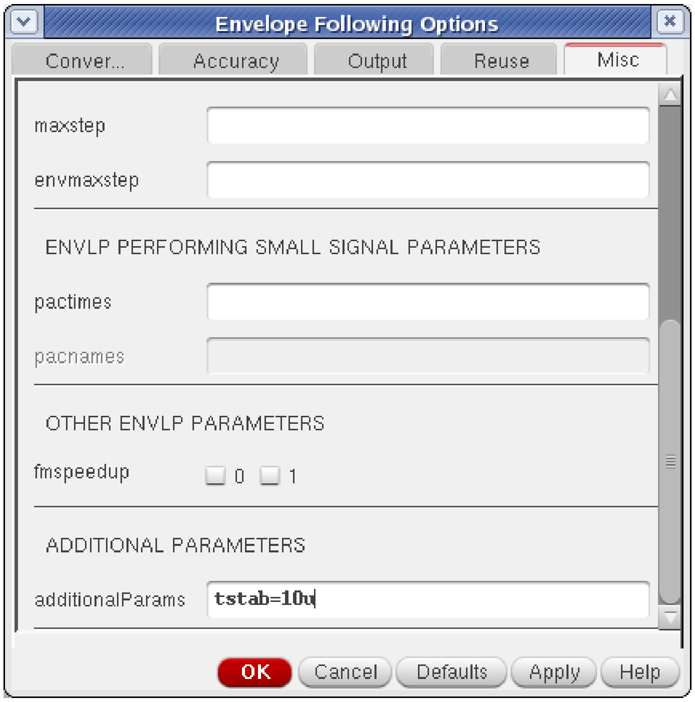
Also, in some circuits, the initial pulse that is used to start the oscillator can cause large transients at the beginning of the envelope analysis and cause problems with the determination of the frequency at the beginning of the simulation. To determine if this is the case, look at the startup waveform at the beginning of the simulation.



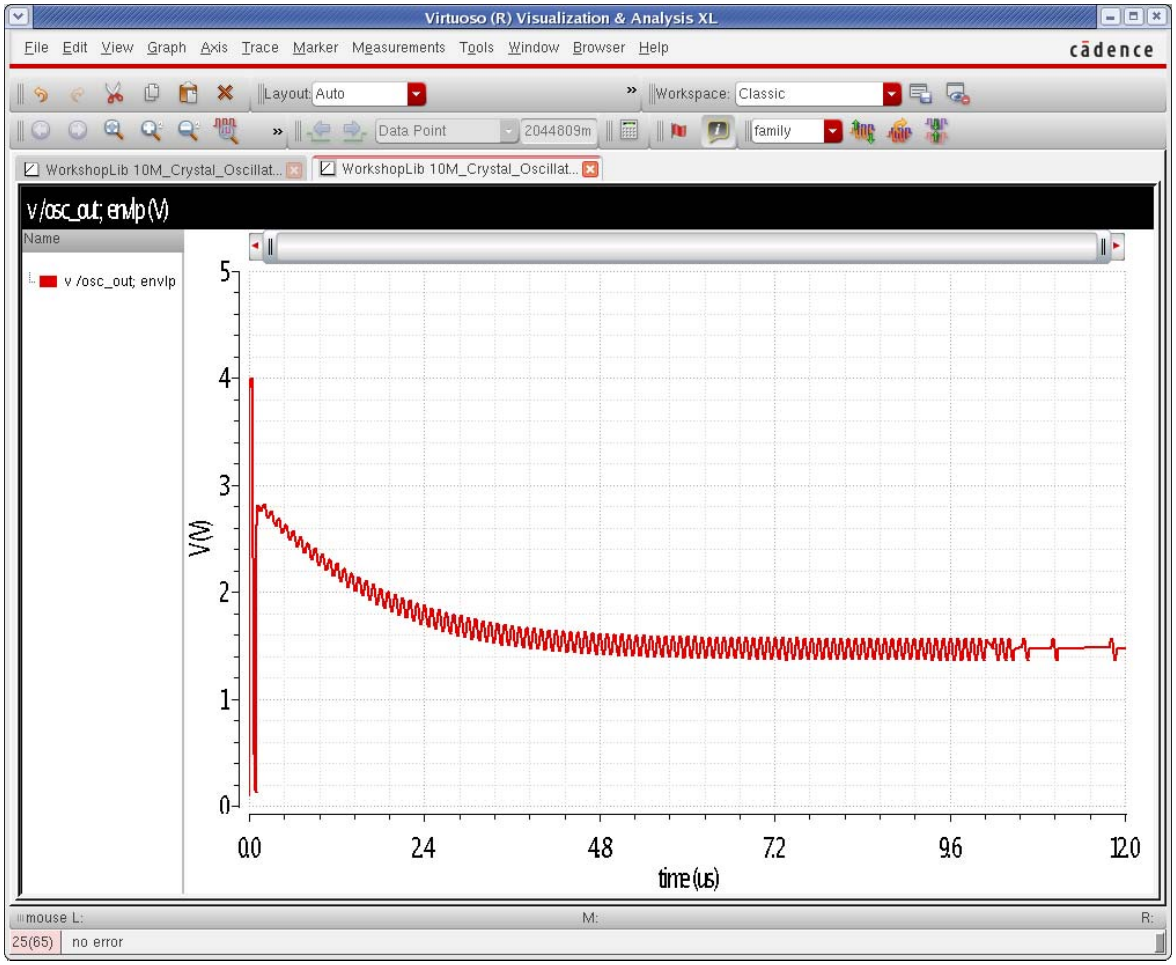
Note that the waveform is still settling at the beginning of the simulation. In this case, four periods are being simulated instead of one. You can see the frequency that is being used in the output log.



Setting *tstab* in the *Misc* tab can cause the simulation to start robustly.



The waveform is much smoother with tstab set.



Examples

There are wireless examples provided with the release. To access the examples:

1. Navigate to <mmsim_install_directory>/tools.<platform>/spectre/examples/SpectreRF_workshop.
2. Copy the rfworkshop.tar.Z file to a directory where you have write access, and extract the file.
3. Navigate to the directory /spectrerf_workshop/doc.

In this directory, you will find directions for the wireless workshop.

7

Large Signal S-parameter Simulation (LSSP)

Large Signal S Parameters

Characterizing RF circuits with small-signal S-parameters is a well established practice. However, small-signal S-parameters are not sufficient to describe nonlinear circuits with moderate to large amplitude signals.

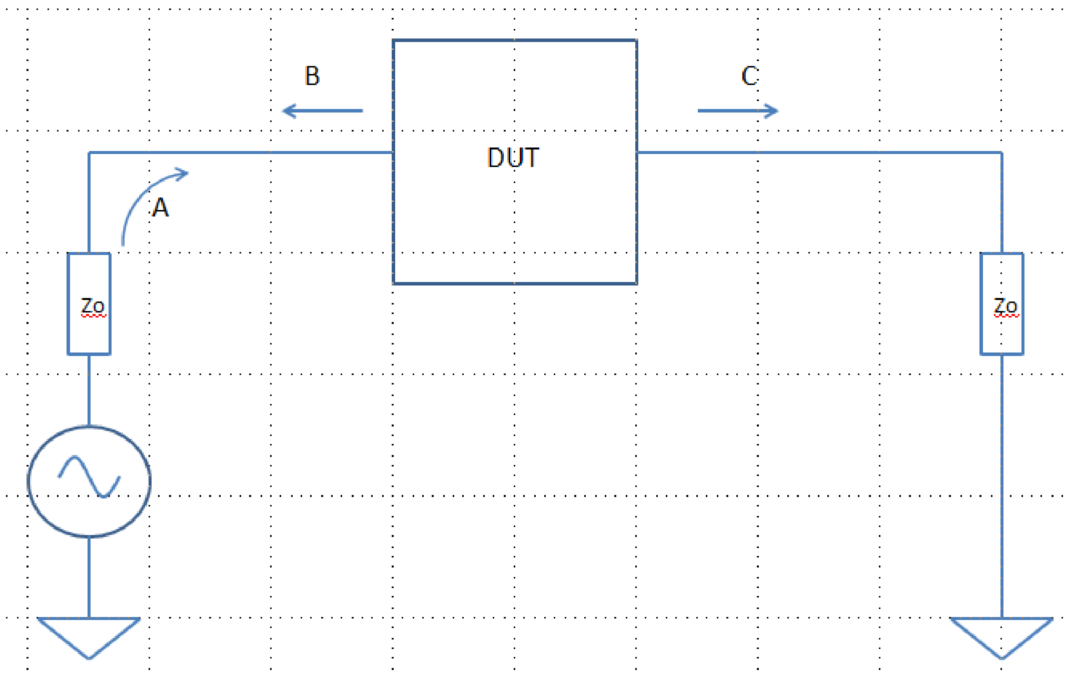
As a natural extension of small-signal S-parameters, large-signal S-parameters (LSSPs) are defined as the ratio of reflected (or transmitted) waves to incident waves. Since small-signal S-parameters are based on the simulation of a linearized circuit, they are independent of the input power. LSSPs include the nonlinear effects at different amplitudes.

The LSSP capability is implemented for single-tone harmonic balance and pss shooting or harmonic balance simulations.

Starting with the MMSIM13.1 release, the LSSP capability has been integrated into the hb and pss (shooting and hb) *Choosing Analyses* form. With this integration, the simulation is now much easier to set up, and allows the input and output frequencies to be different. Direct plot functions have also been implemented to allow plotting the large-signal S-Parameters on a Smith Chart. When these expressions are added to the Outputs section of the ADE window from the *Direct Plot Form*, ADE plots the output on the Smith Chart.

Large Signal S parameters for a Two Port Circuit

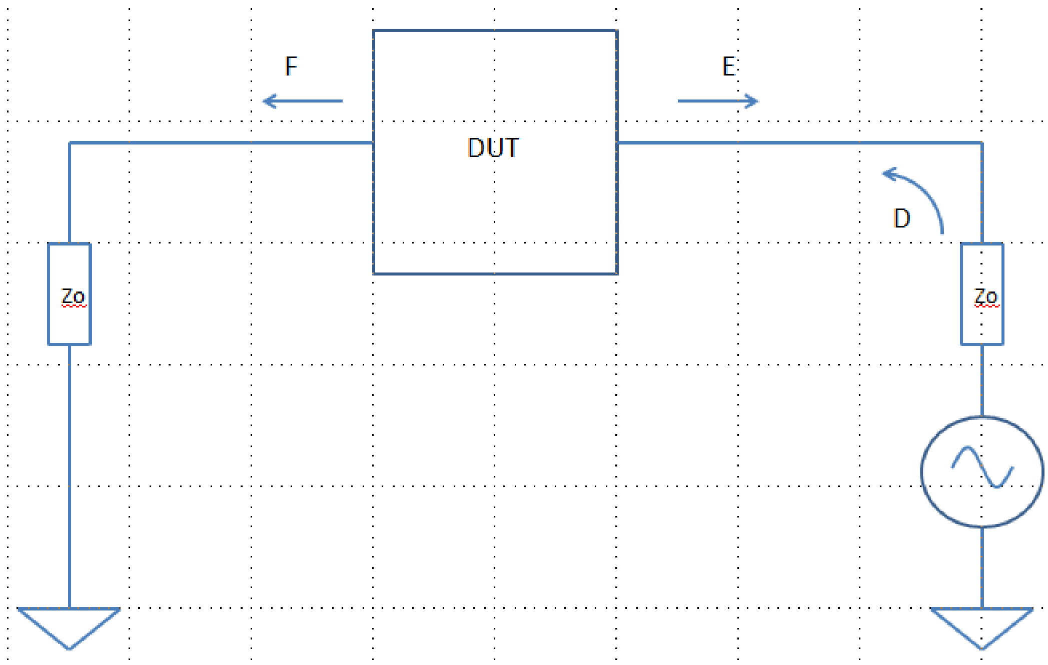
Consider calculating LSSPs for a two-port circuit. When you apply a signal at port 1 on the left and terminate port 2 on the right, as shown below, you can measure S_{11} and S_{21}.



For a wave A launched from port1 on the left, S_{11} is B/A, and S_{21} is C/A, where A, B, and C are all vectors. When this measurement is done, the power in the load resistor is calculated.

For a wave A launched from port1 on the left, S11 is B/A, and S21 is C/A, where A, B, and C are all vectors.

Similarly, when you launch wave D at port 2 on the right, and terminate port 1 on the left as shown below, you can measure S_{12} and S_{22}. The power measured above is set in port2 for this measurement.



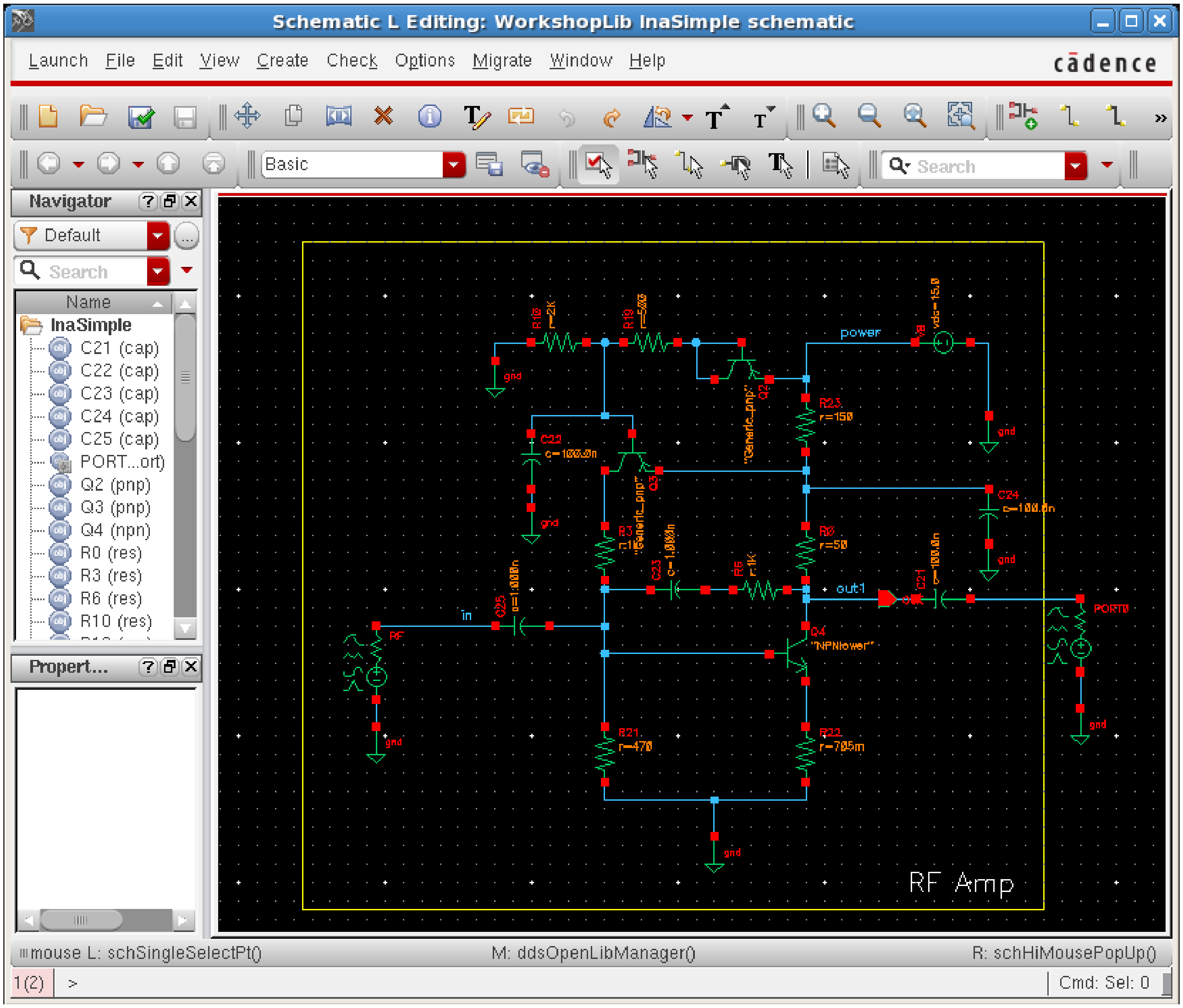
For a wave launched from port2 on the right, S_{22} is E/D, and S_{12} is F/D, where D, E, and F are all vectors.

For nonlinear circuits, the values of the S-Parameters can change as the amplitude of the signal changes. In addition, the frequency can be swept for a single input and output amplitude at a time

Circuit Setup

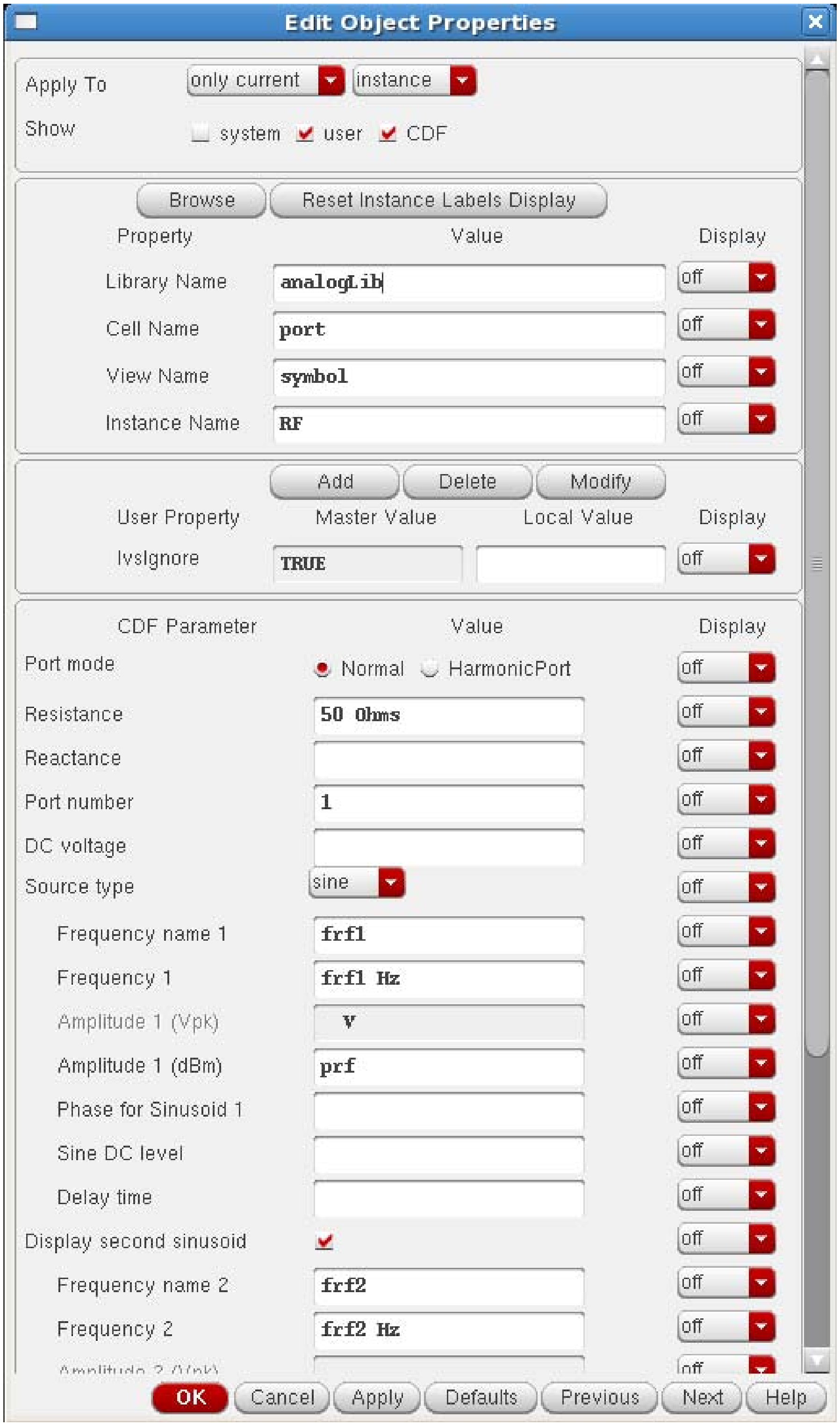
The normal setup for large-signal S-Parameter measurement is similar to the setup for a swept input power simulation, with the addition of setting the output *Source type* to *sine* and leaving all the properties for the sine wave blank. LSSP analysis causes port 1 to be driven first so the S_{11} and S_{21} measurements can be made, and then causes port 2 to be driven next so the S_{12} and S_{22} measurements can be made. The power in port 2 from the S_{21} measurement is set in port 2 for the S_{22} and S_{12} measurement.

A behavioral low noise amplifier is shown below.

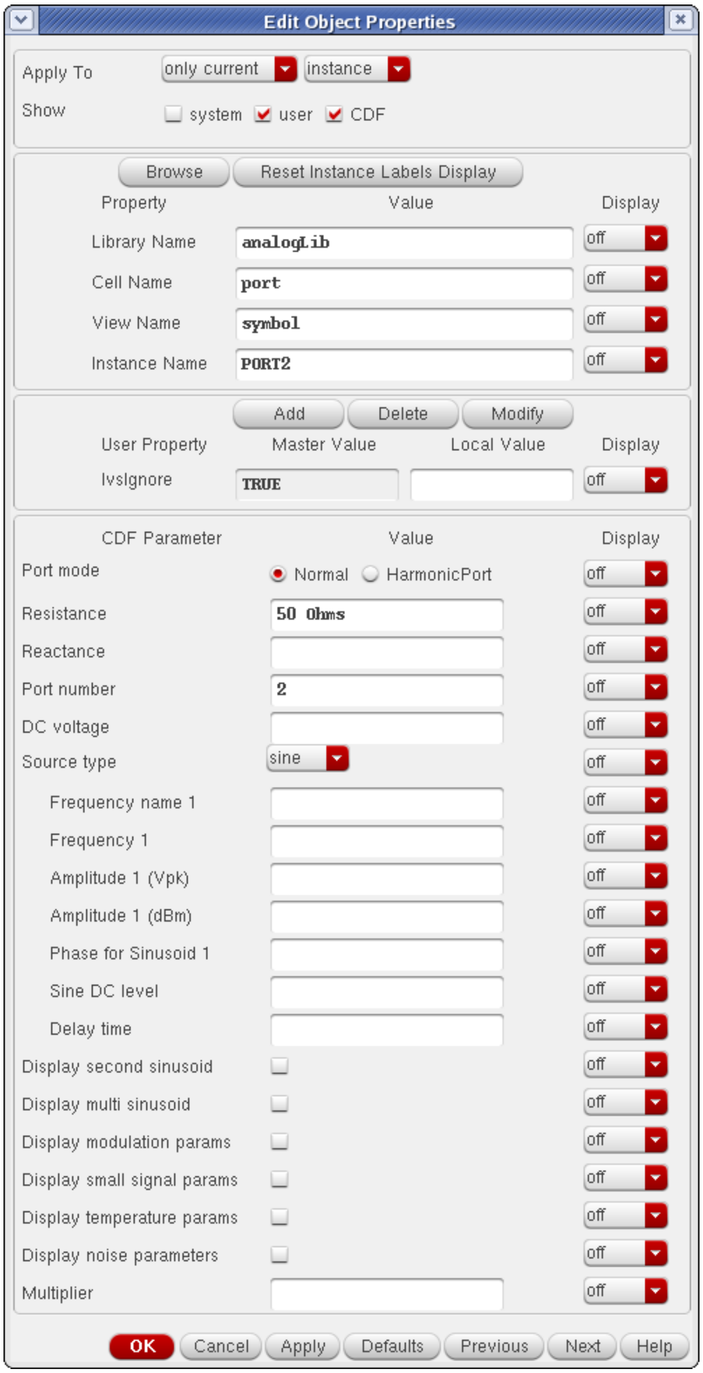


LSSP expects that a variable is used to set the input power. The normal setup is to specify the input power sweep as normal and then select the *LSSP* check box in the *Choosing Analyses* form. When this is chosen, you define the source and load ports in the *Ports* field, and you select the frequency of the output in the *Load Harmonic* field. The output frequency defines the frequency for the forward S_{21} measurement, and the frequency of port 2 for the S_{22} and S_{12} measurements. Set the output *Source type* to *sine* with no properties to define the frequency or amplitude.

The input port RF and output port *PORT2* setup are shown below.



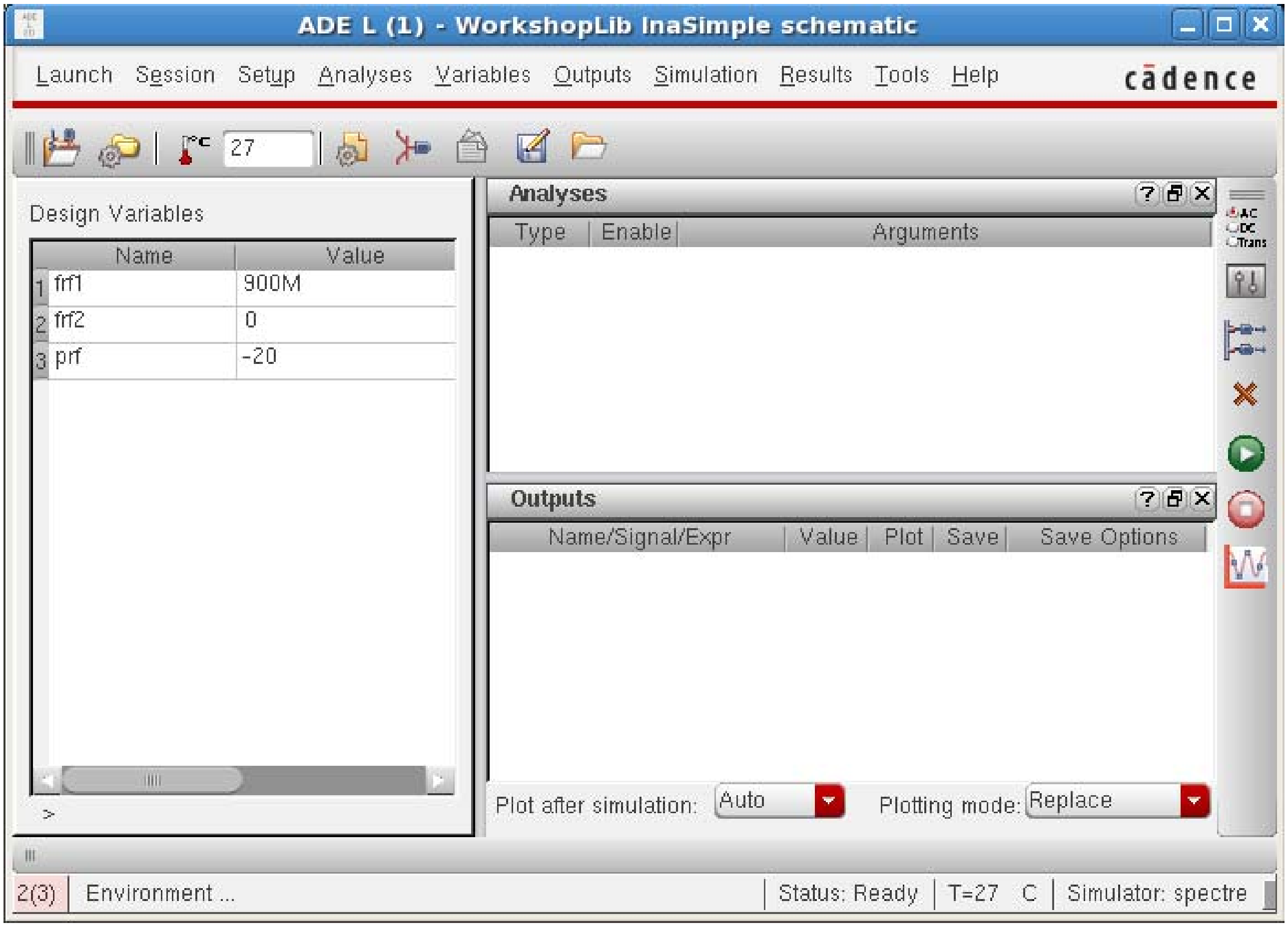
Note that the input source is set to produce two RF signals. Both signals have frequencies that are set by the variables *frf1* and *frf2*. The amplitude of both signals are set to the variable name *prf*. This is a common way to set sources because it allows changing the frequencies and amplitudes using variable names in ADE without needing to change the schematic.



The normal setting for the load port is to set the *Source type* to *sine*, and leave all the properties that define the sinusoid blank, as shown above.

ADE Setup

The ADE setup is shown below with the variables defined in the *Design Variables* section.



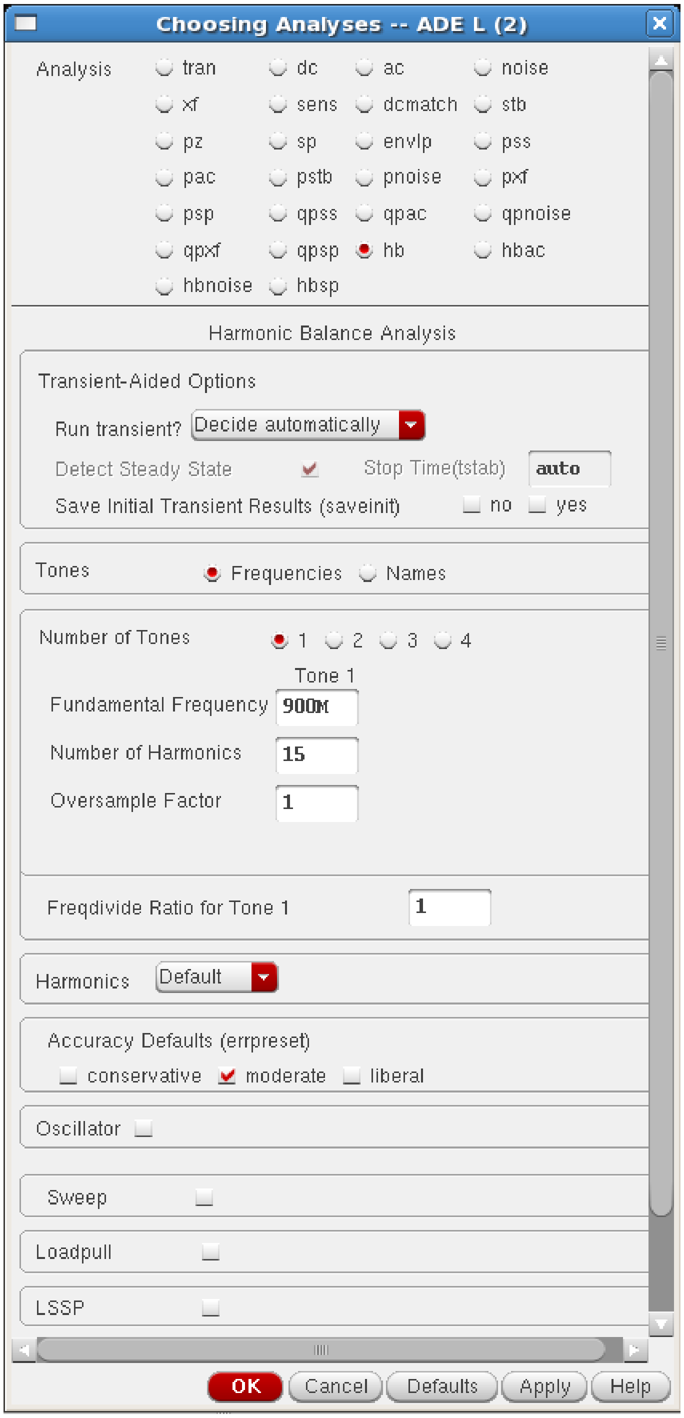
Note that the variable *frf2* has been set to zero to disable the second RF tone.

Analysis Setup

Next, set up the hb *Choosing Analyses* form, as shown below.

1. Select *Analyses - Choose* from the ADE window.

The *Choosing Analyses* form is displayed, as shown below.

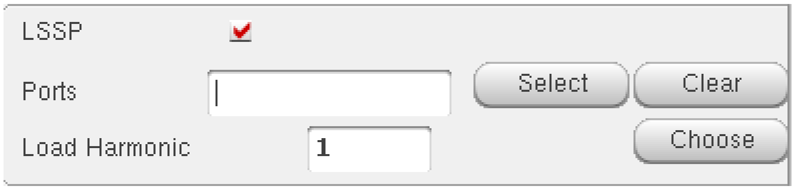


2. Specify the fundamental frequency in the *Fundamental Frequency* field. Here, 900M has been specified.

3. Specify the number of harmonics in the *Number of Harmonics* field. Here, 15 is specified.

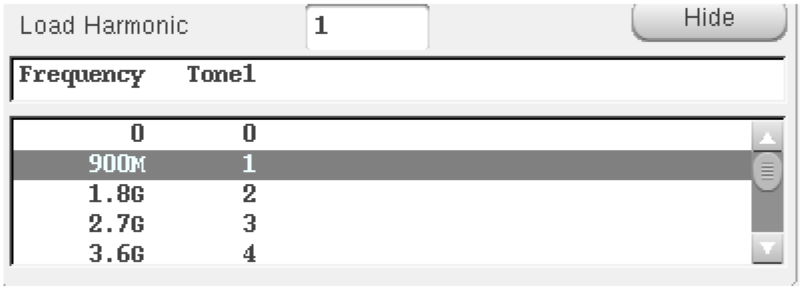
   It is recommended to set the number of harmonics manually and specify a value that is high enough to get accurate results at the highest input power level in the sweep.

4. Select the *LSSP* check box. The LSSP options are displayed, as shown below.



5. Click *Select* to the right of the *Ports* field and select the input source and output source from the schematic. The *Ports* field is populated with the names of the input and output ports, as shown below.

6. Click *Choose* to the right of the *Load Harmonic* field and select the frequency and tone, as shown below.

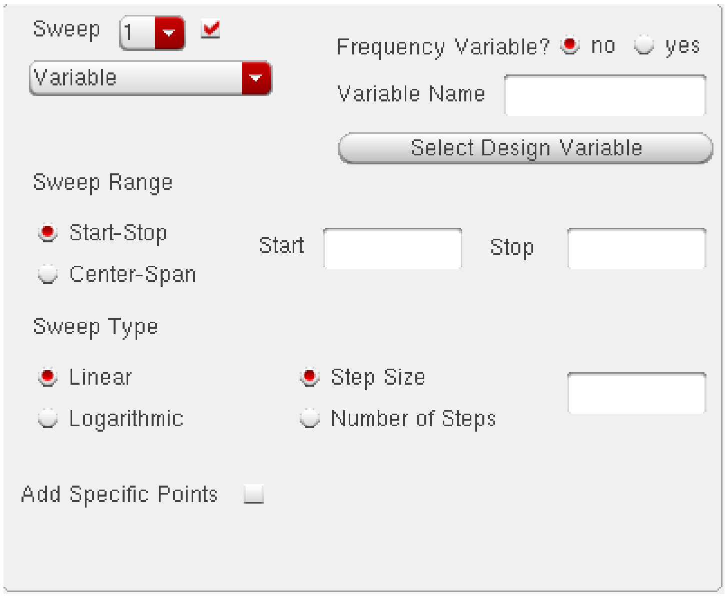


7. Click *Apply*.

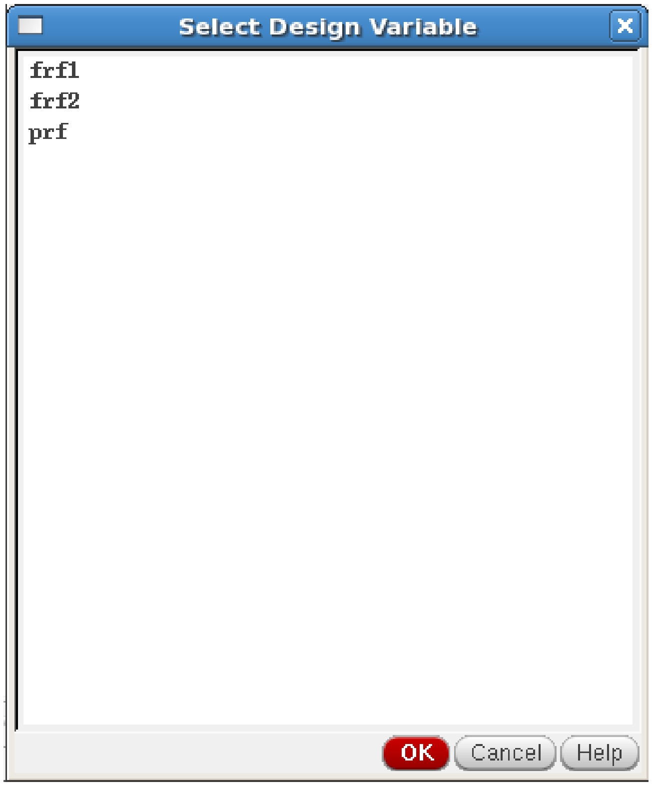
LSSP Amplitude Sweep

To set up the amplitude sweep:

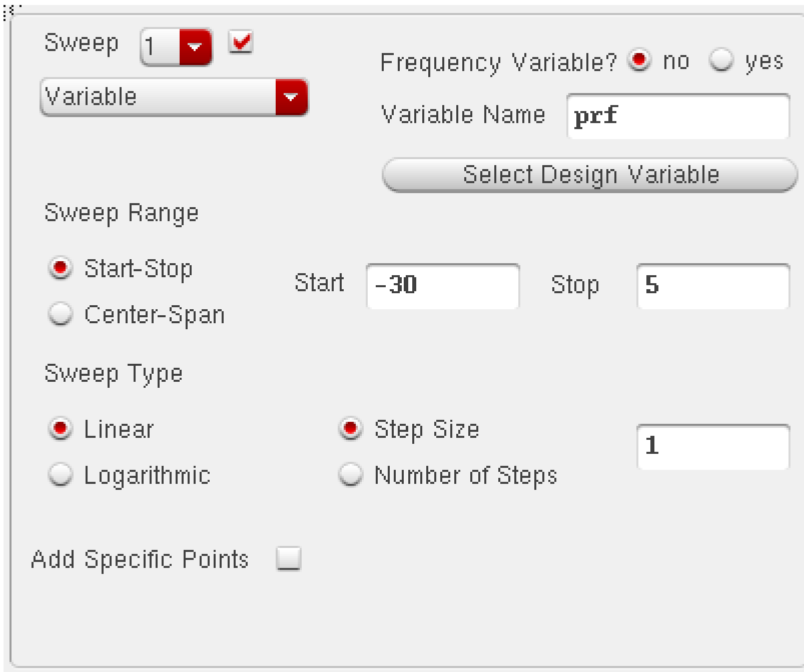
1. Select the *Sweep* check box. The Sweep options are displayed, as shown below.



2. Click *Select Design Variable*. The Select Design Variable dialog is displayed, as shown below.



3. Select a variable from the available list of variables. The *Variable Name* field is populated with the name of the variable, as shown below.

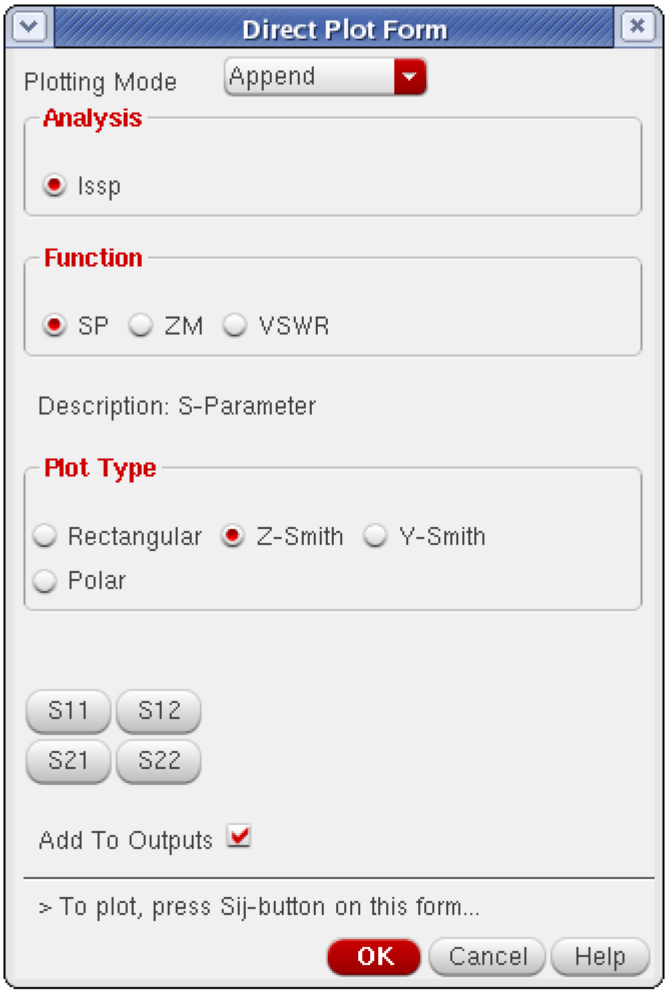


4. Specify the sweep range in the *Start* and *Stop* fields.
5. Specify the step size in the *Step Size* field.
6. Click *OK*.
7. Run the simulation.

Plotting the S11 Curve

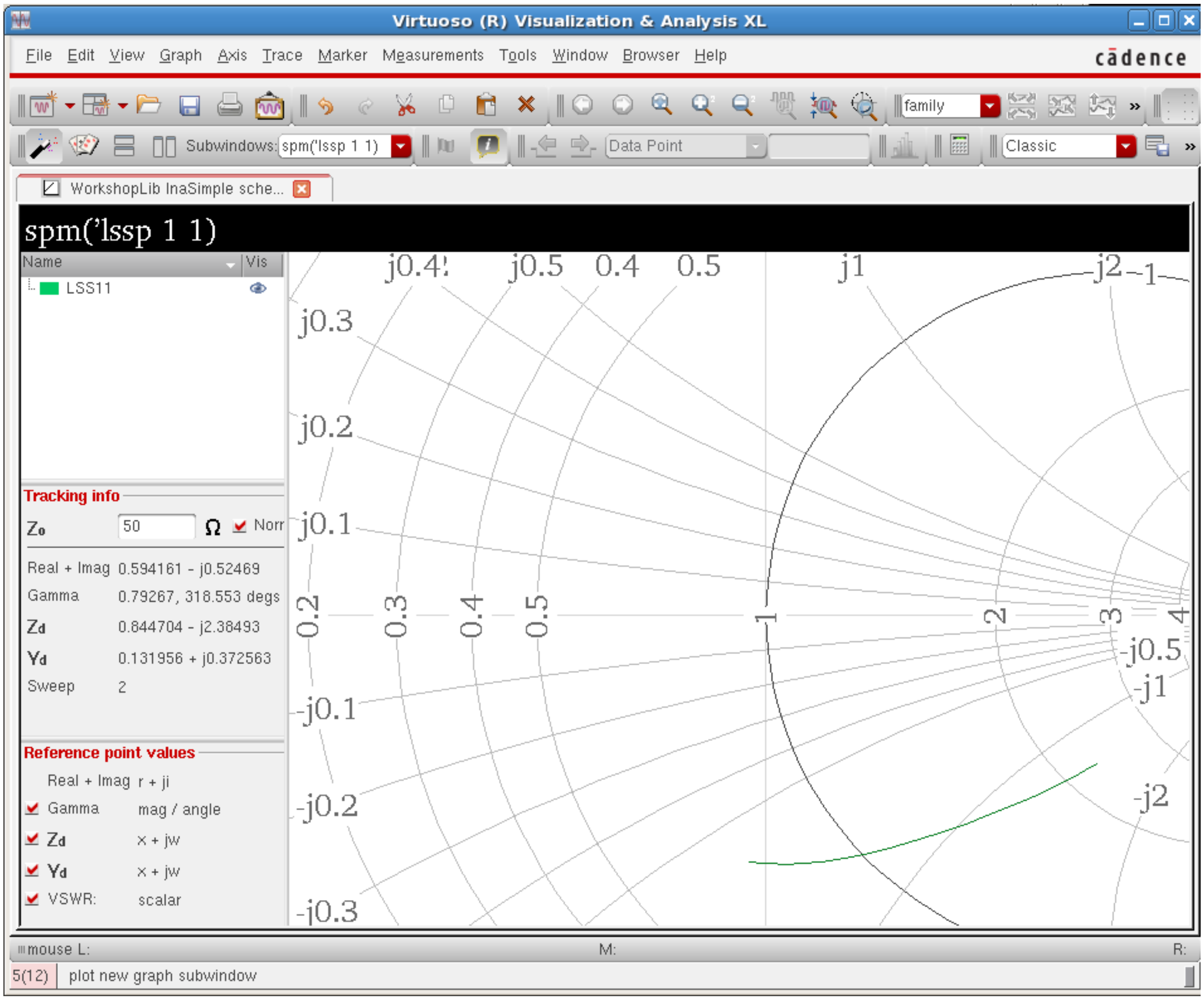
To plot the S_{11}curve:

1. In the ADE window, choose *Results - Direct Plot - Main Form*. The *Direct Plot Form* is displayed, as shown below.

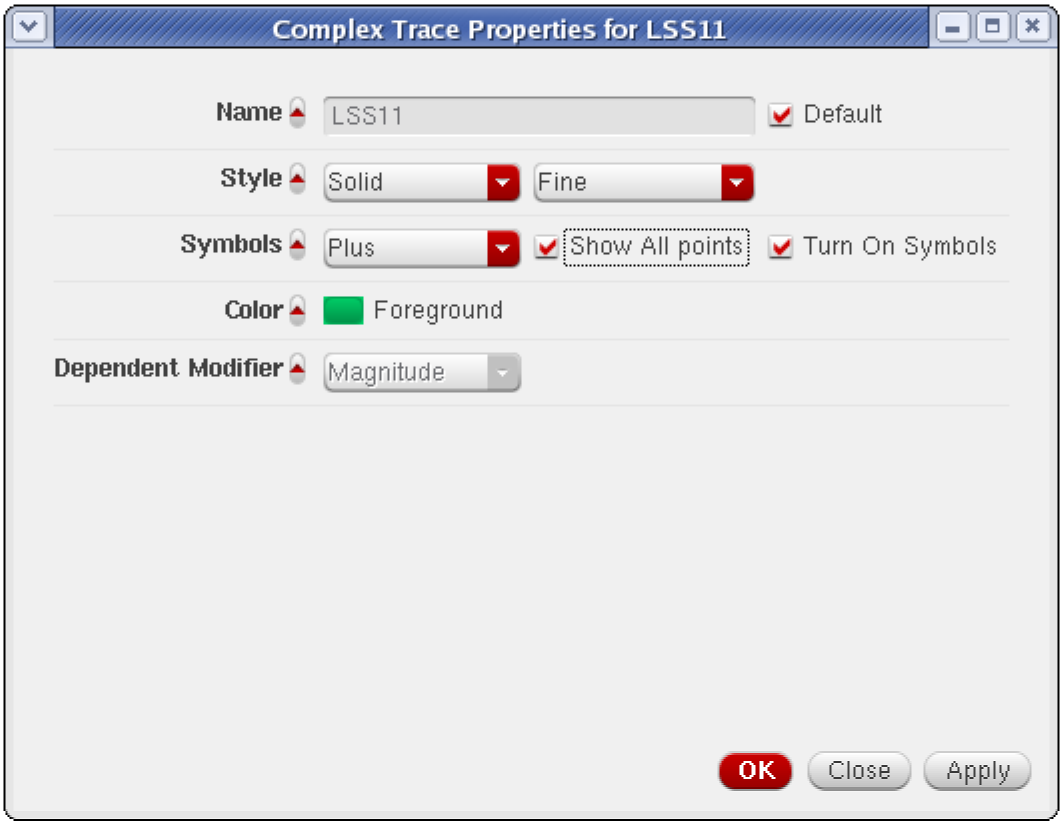


2. Choose *Z-Smith* from the *Plot Type* section.
3. Select the *Add to Outputs* check box to add the expression in the *Outputs* section of the ADE window.
4. Click the *S11* button.

   The input reflection coefficient is displayed on the Smith chart, as shown below.

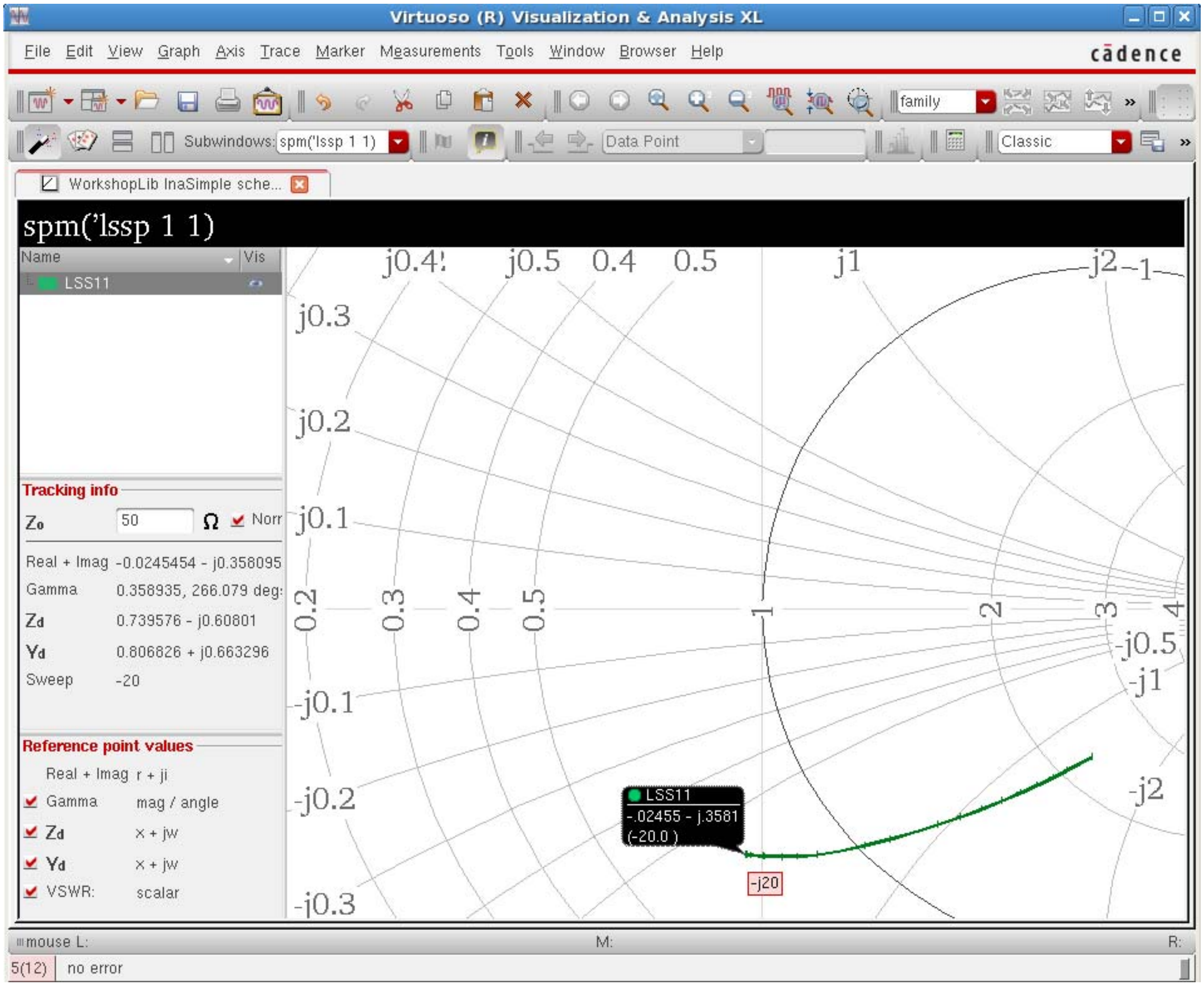

Virtuoso (R) Visualization & Analysis XL
File Edit View Graph Axis Trace Marker Measurements Tools Window Browser Help
cādence
Subwindows: spm('lssp 1 1)
Data Point
Classic
WorkshopLib lnaSimple sche...
spm('lssp 1 1)
Name
Vis
LSS11
Tracking info
Z0 50 Ω Norr
Real + Imag 0.594161 - j0.52469
Gamma 0.79267, 318.553 degs
Zd 0.844704 - j2.38493
Yd 0.131956 + j0.372563
Sweep 2
Reference point values
Real + Imag r + ji
Gamma mag / angle
Zd x + jw
Yd x + jw
VSWR: scalar
mouse L:
M:
R:
5(12) plot new graph subwindow

5. Double-click the trace legend. The *Complex Trace Properties for LSS11* window is displayed, as shown below.



6. Select the *Turn On Symbols* check box.
7. Select the *Show All points* check box.
8. Click *OK*.
9. In the waveform window, select *Marker - Snap Track Cursor* and move the mouse pointer over the trace legend.

The tracking cursor displays the input power level, as shown below.



Similarly, you can plot the S_{12} curve.

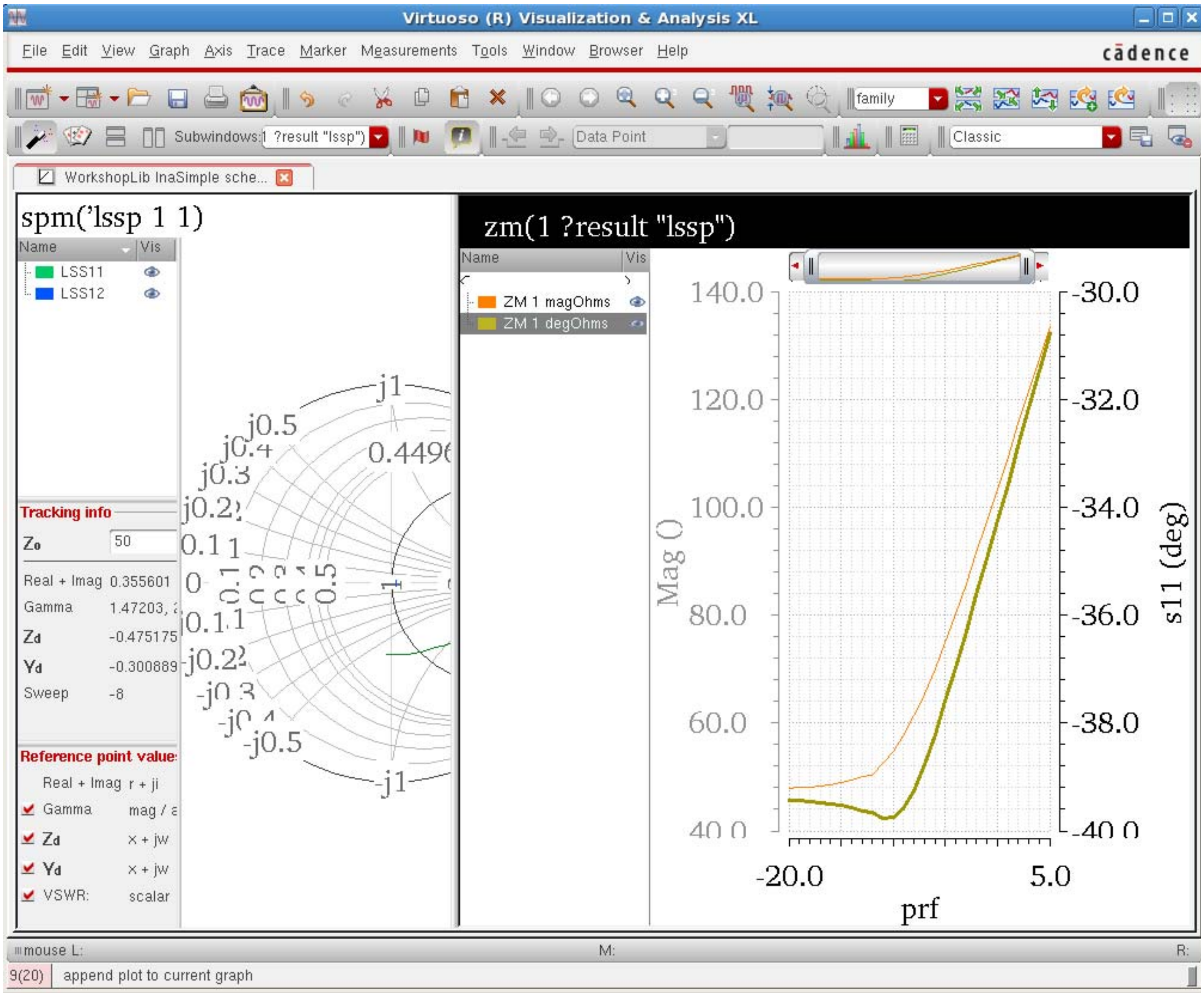
Plotting the Input Impedance Magnitude and Phase Curves

To plot the input impedance magnitude and phase curves:

1. In the *Direct Plot Form*, select *ZM1* from the *Function* section.
2. Select *Magnitude* from the *Modifier* section.
3. Click *ZM1*.
4. Select *Phase* from the *Modifier* section.

5. Click *ZM1* again.

The impedance and phase curves are displayed in the waveform tool, as shown below.



8

AnalogLib Components Used in RF Simulation

The Delayline Element

The delayline element is a lossless transmission line with a specified delay time T_d and characteristic impedance Z_0. It is intended for the small-signal ac, sp, and noise analyses, and the harmonic balance analysis and models ideal delays in these analyses.

The ABCD matrix of a lossless transmission line section is given by:

$$ABCD = \begin{bmatrix} \cos(\omega T_d) & jZ_d \sin(\omega T_d) \\ \dfrac{j\sin(\omega T_d)}{Z_0} & \cos(\omega T_d) \end{bmatrix}$$

The delayline symbol is shown below.

Command-line help

spectre -h mtline

CDF Parameters

CDF Parameter Label	CDF Parameter	Spectre	auCDL	auLVS	hspiceD	Ultrasim	Default
Characteristic Impedance	`z0`	X	-	-	X	X	50
Delay Time	`td`	X	-	-	X	X	1n

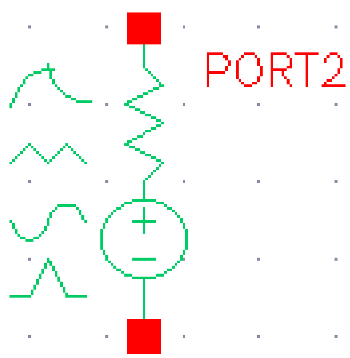
The properties list is shown below.

The PORT Element

You can use the *port* component, located in the *analogLib* library, in RF circuits for Virtuoso® Spectre® circuit simulator RF Analysis (SpectreRF) and Spectre S-parameter simulations.

The *port* component, located in the *analogLib* library, is a superset of the existing *psin* component. The *port* component supports all the source types of the Spectre *port* primitive: *pwl*, *pulse*, *sine*, *dc*, and *exp.* In addition, it includes the *bit* and *prbs* source types. If you are using the `psin` component, it is highly recommended that you switch to the port component because the properties are organized in a better way, and the levels in volts peak and in dBm both cannot be set at the same time. This is easily accomplished by obtaining the properties form for `psin`, and changing the cell name to `port`. All the properties that are set in the `psin` component will retain their values when you make the change.

Note: The existing `psin` is retained for legacy designs and will not be enhanced in any way moving forward.



The *port* component is an independent resistive source tied between positive and negative terminals. It is equivalent to a voltage source in series with a resistor, where the reference resistance of the *port* is the value of the resistor.

Capabilities of the port Component

While the *port* component is most useful as a stimulus in high-frequency circuits, it also has the following unique capabilities:

- It can define the ports of a circuit to the S-parameter analysis.
- It has an intrinsic noise source that lets the noise analysis directly compute the noise figure of the circuit.
- Is the only source for which you can specify the amplitude in terms of power.

- It generates sinusoidal, exponential, piecewise-linear, periodic piecewise-linear, pulse, bit, and prbs waveforms for all the large-signal analyses.
- It can have different impedances for each harmonic number in the harmonic balance large-signal analysis.

Terminating the Port

When you specify the voltage on a *port*, all Spectre simulations assume that the port is properly terminated in it's reference resistance. The specified voltage value is not the voltage on the internal voltage source, which is actually set to twice the value specified on the `port`. If you use a `port` source to drive an open circuit, the voltage (for DC, transient, AC, and PAC signals) is double its specified value. However, you can alternatively specify the amplitude of the sine wave in the transient and PAC analyses as the power in dBm delivered by the *port* when terminated with the reference resistance. For the `bit` and `prbs` source, the voltages specified for the zero and one states are doubled inside the port as usual. For the `prbs` source, you can add a matching resistor across the port to get the voltages specified in the zero and one states, or you can omit the resistor, and half the desired voltages. For the bit source, if you intend to use the z state, do not add a matching resistor across the source because this small resistor will totally swamp the Z-state effect. Thus, when you use the bit source in a port, the values for the zero and one states should be half the desired voltages. This is not an issue for the `vsource` element.

The *port* component *Edit Object Properties* form is shown in Figure 8-1 on page 1663 and Figure 8-2 on page 1664.

Figure 8-1 Top of the port Component Edit Object Properties Form

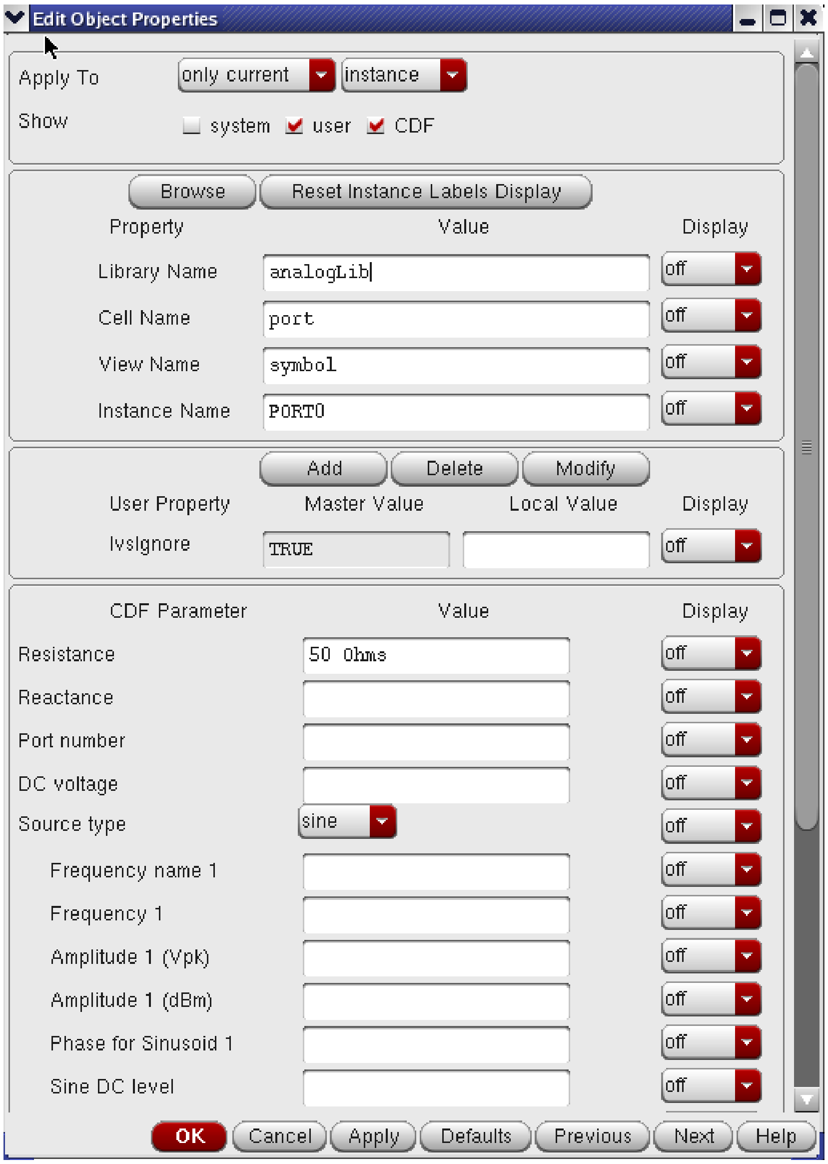
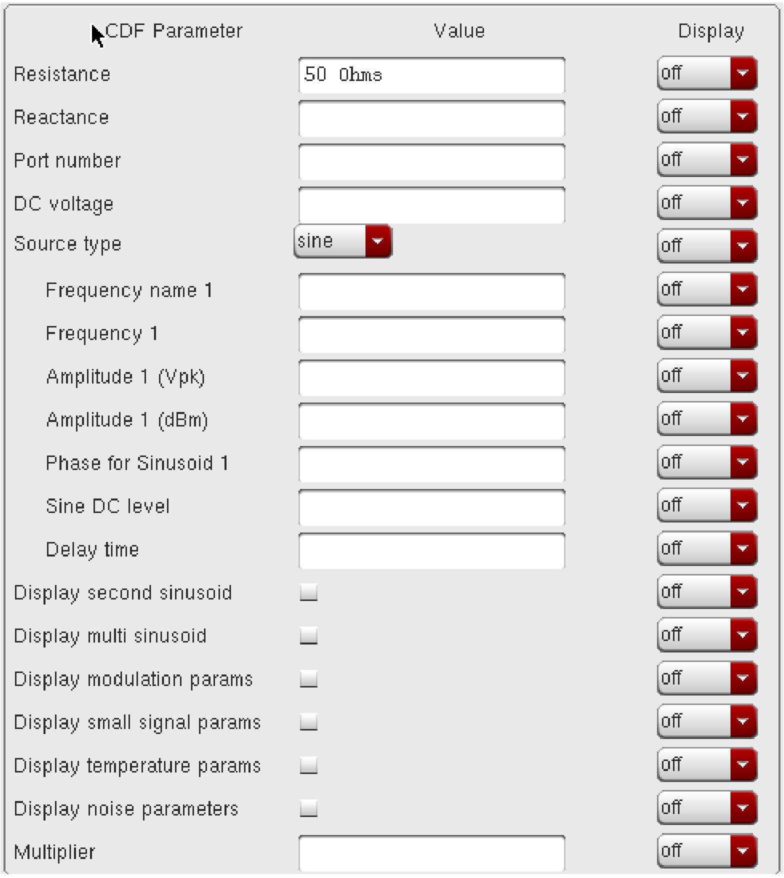


Figure 8-2 Bottom of the port Component Edit Object Properties Form



Parameters for the Port Component

Port Parameters

Port parameters include *Resistance*, *Reactance*, *Port Number*, and *Multiplier*.

Resistance

The reference resistance of the system. The value must be a real number, but not 0. The default value is 50 ohms. This applies for all analyses. Note that the resistance and reactance can be set individually for each harmonic in harmonic balance large-signal analyses only. For more information, refer to Using the Harmonic Port with Harmonic Balance on page 1665.

Reactance

The imaginary part of impedance is used for harmonic balance large and small-signal analyses, and for the ac, noise, xf, sens, stb, pz, and sp.analyses. The reactance is not used for the dc, transient, and shooting-based large and small-signal analyses. The value must be a real number. The default is zero ohms. Note that the resistance and reactance can be set individually for each harmonic in harmonic balance large-signal analyses only. For more information, refer to Using the Harmonic Port with Harmonic Balance on page 1665.

Port number

The number associated with the *port*. The value must be a non-zero integer and must be unique for each *port* in a schematic. The port number is not automatically indexed when you place a new *port* on your schematic. There is no need to assign this number because in the sp analysis, you just select the port instances you want from the schematic.

If you do assign port numbers, start with the number one, and number the ports sequentially. A gap in the port number sequence causes a simulation error if the port list is not defined in the S-parameter analyses.

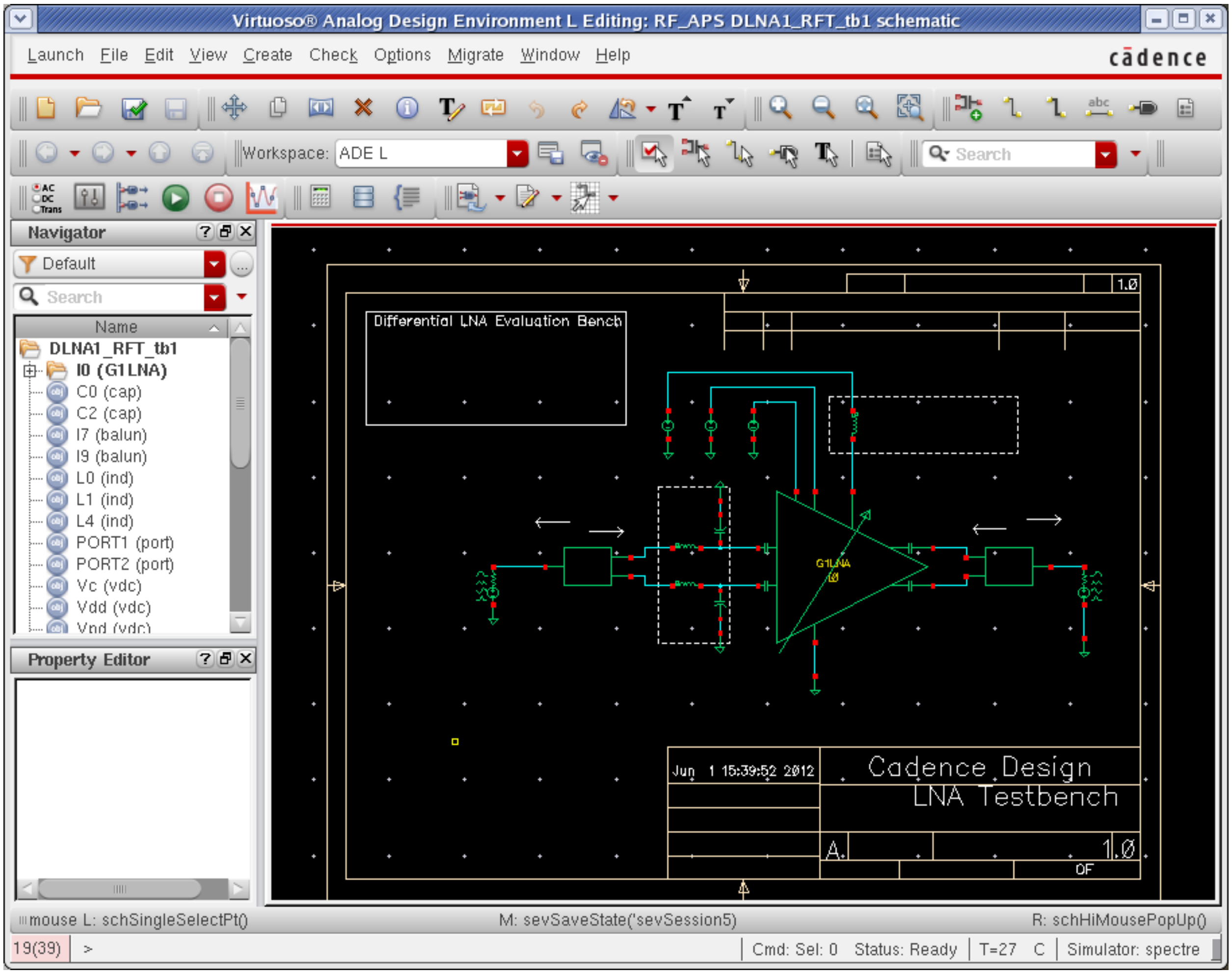
Multiplier

Leave this property at the default. The multiplicity factor specifies the number of *ports* in parallel. The value must be a non-zero real number and the default is 1. For example, if you set *Resistance* to 50 and *Multiplier* to 2, you specify two *ports* in parallel, with an effective reference resistance of 25 ohms.

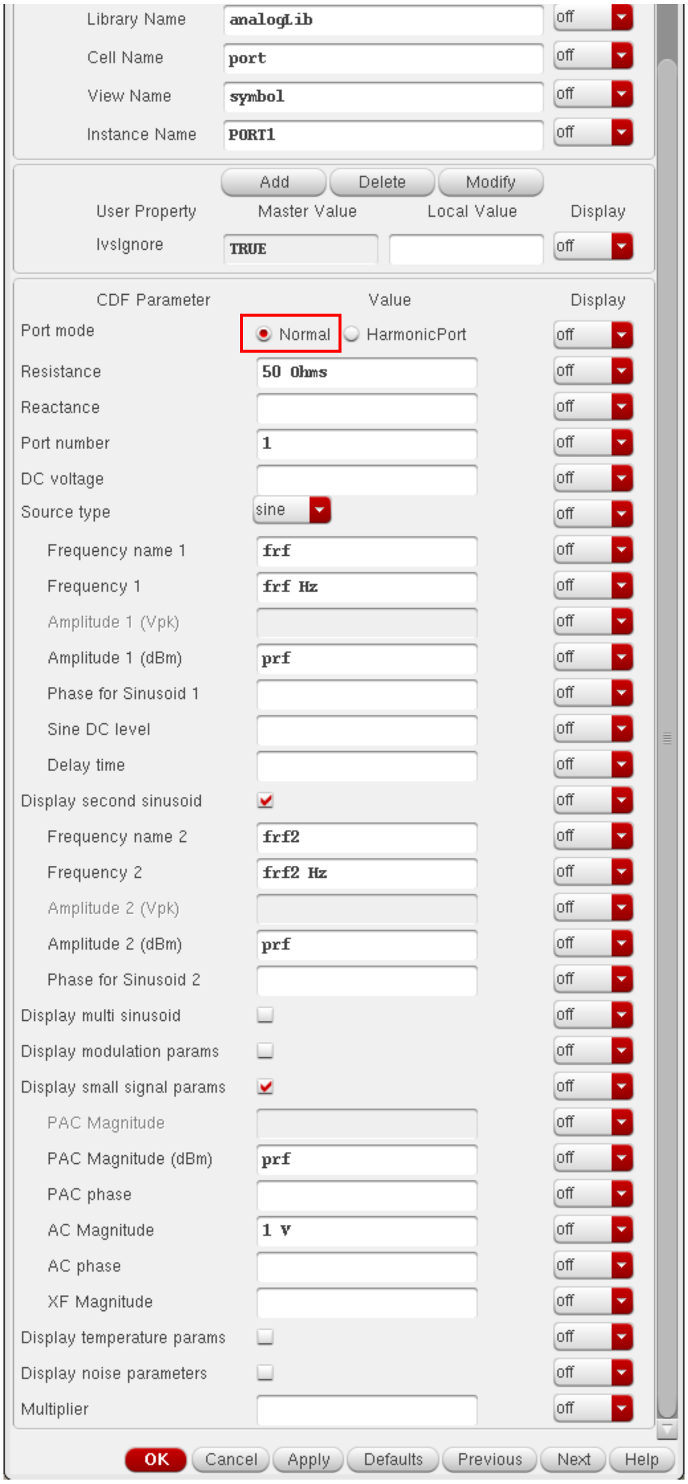
Using the Harmonic Port with Harmonic Balance

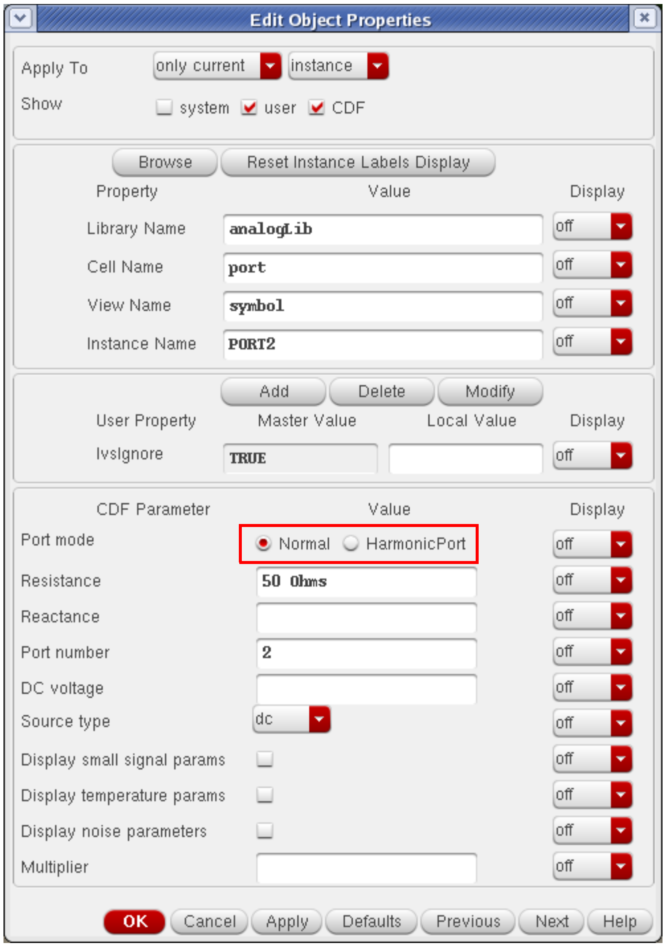
The port component has a feature called harmonic port that allows the specification of different impedances at different harmonic numbers. This is available for harmonic balance large-signal analyses only and does not work with any other analysis.

For this example, a 2.4GHz LNA is the example circuit.



The input and output ports are set up normally, as shown in the properties forms below.

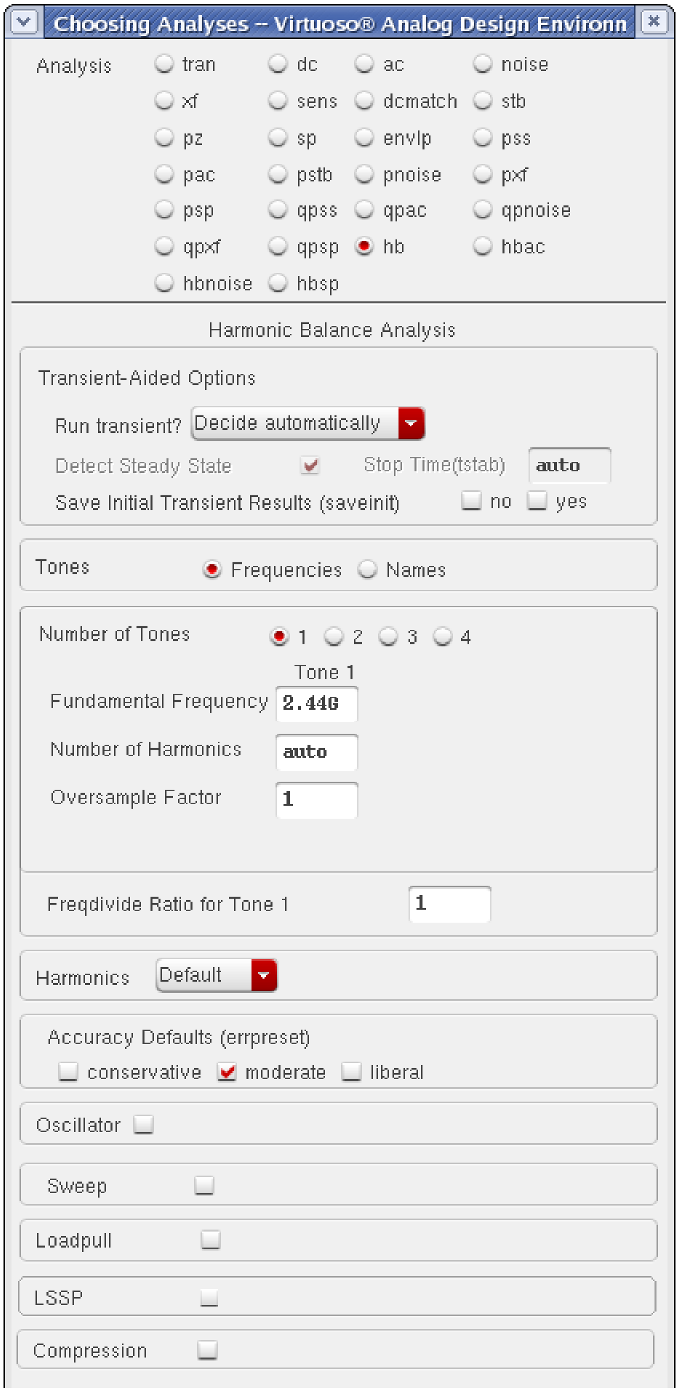




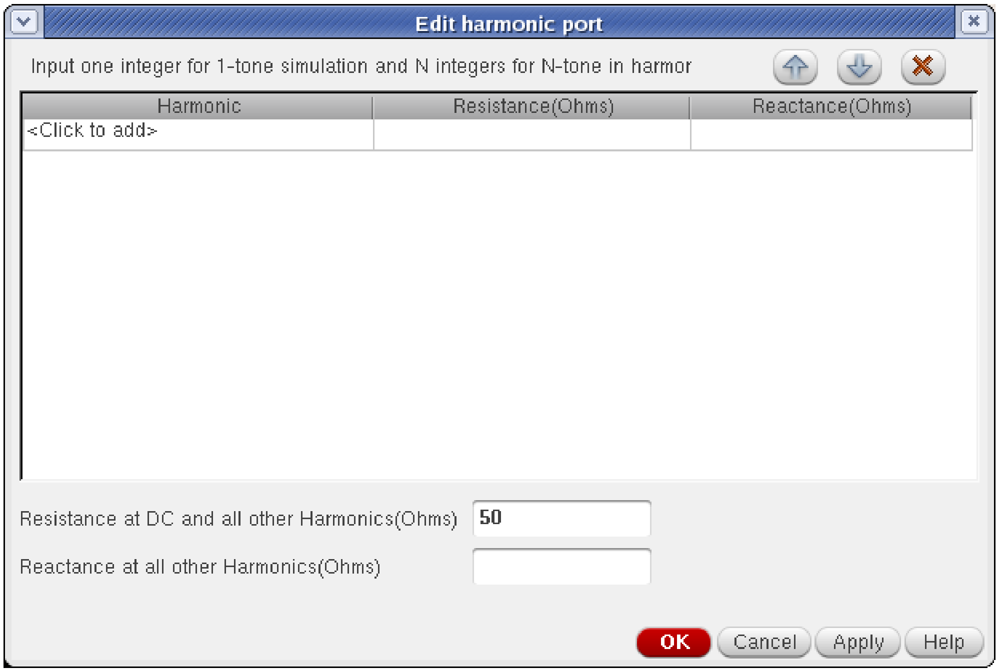
PORT1 is the input and *PORT2* is the output.

One Input Frequency

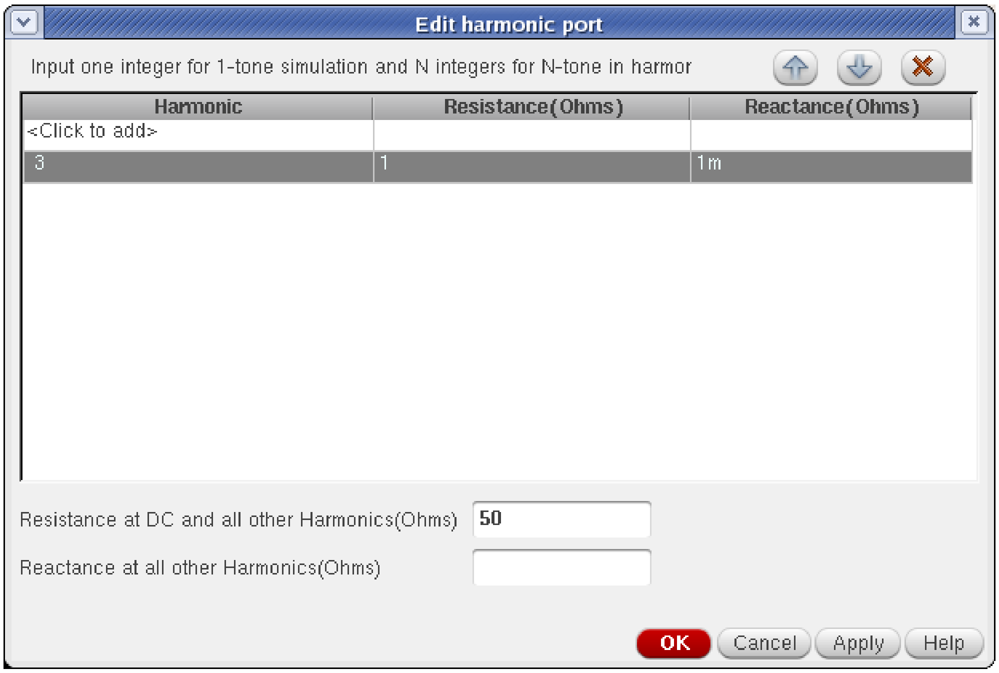
The harmonic balance *Choosing Analyses* form has a single input applied.



To use the harmonic port, select *Harmonic Port* on the properties form. In this case, the port being used as a load in the circuit is selected. The *Edit harmonic port* window is displayed.



For this setup, a single input frequency is applied to the circuit. In this case, harmonic numbers 1, 2, 3, 4, and 5 exist in the simulation result. In the *Harmonic* column, you enter the harmonic number. In the *Resistance* column, you enter the resistance. In the *Reactance* column, you enter the series reactance in ohms. Positive reactance adds inductance, and negative reactance adds capacitance. Multiple entries are allowed, so different impedances can be set for each harmonic. The resistance or reactance can be left blank, which is interpreted as zero resistance or reactance.



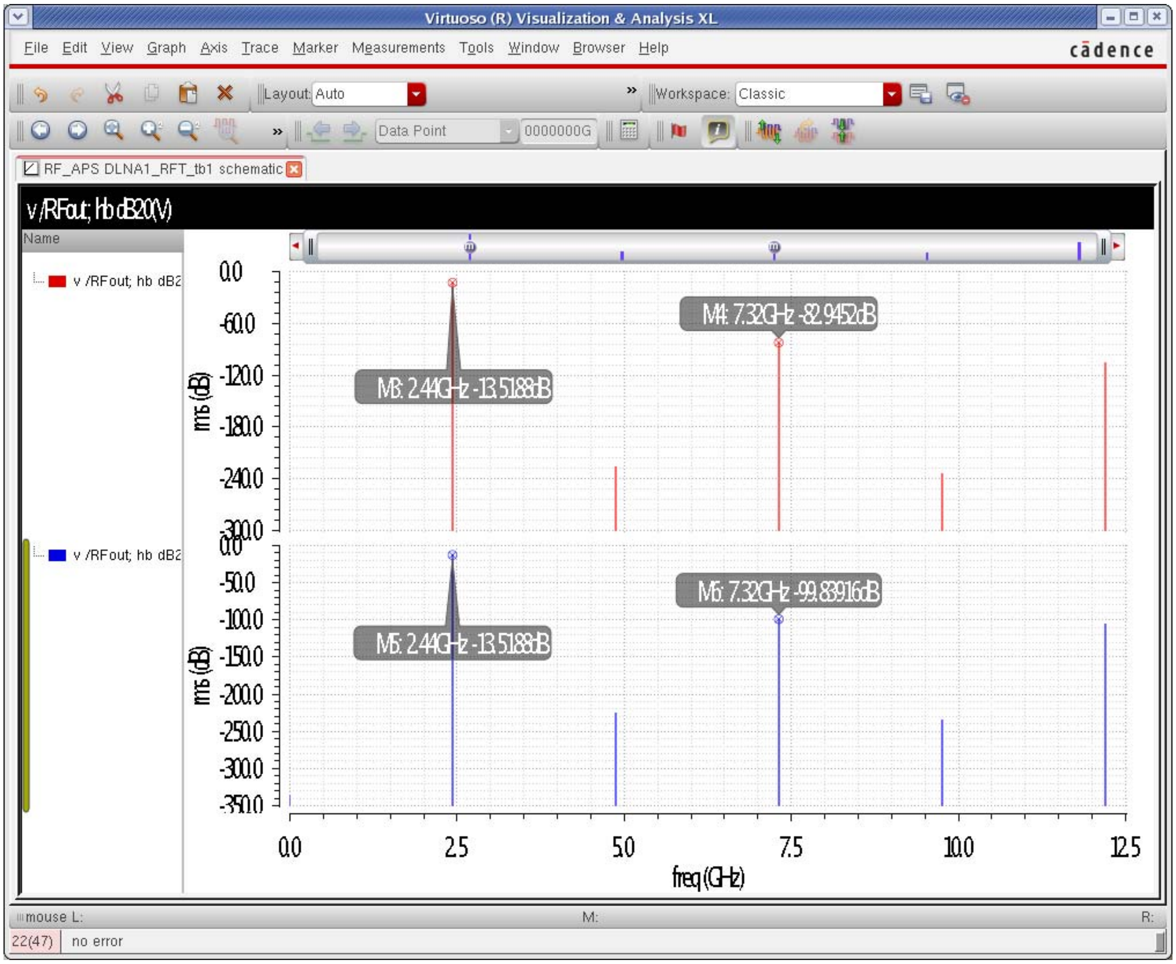
Note: When a variable name is used to set the value of resistance or reactance, that variable must be added to ADE manually. The *Variables - Copy From Cellview* menu option will not enter the variable into the variable table in ADE. To add a variable manually:

1. Double-click on the last variable in the variables list. The *Editing Design Variables* window is displayed.
2. In the *Editing Design Variables* window, click *Next*. The *Name* and *Value (expr)* fields become editable.
3. Enter the variable name in the *Name* field, and the starting value in the *Value (expr)* field.
4. Click *OK* or *Apply*. The variable and value are entered into the table.

Note: Whenever you have an expression for one of the values, you must enclose the expression in parentheses. If this is not done, a simulation error is generated:

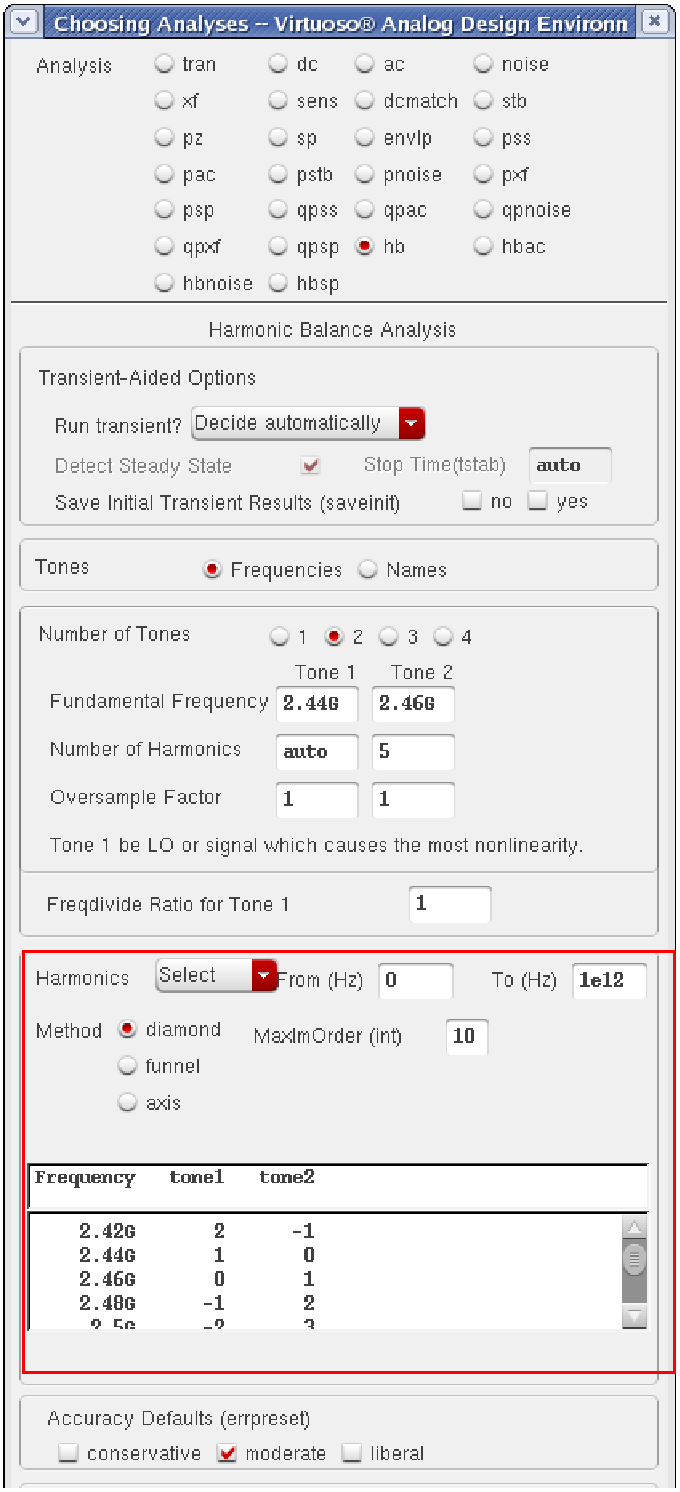
```
ERROR: Length of hvec is not equal to length of rvec multilies tones or length of
rvec is not equal to length of xvec.
```

Here is a comparison where the third harmonic is set normally in the top trace, and to 1 ohm in series with +j1mohm in the bottom trace. The lower impedance for the third harmonic in the bottom trace lowers the amplitude of the third harmonic.



Two Input Frequencies

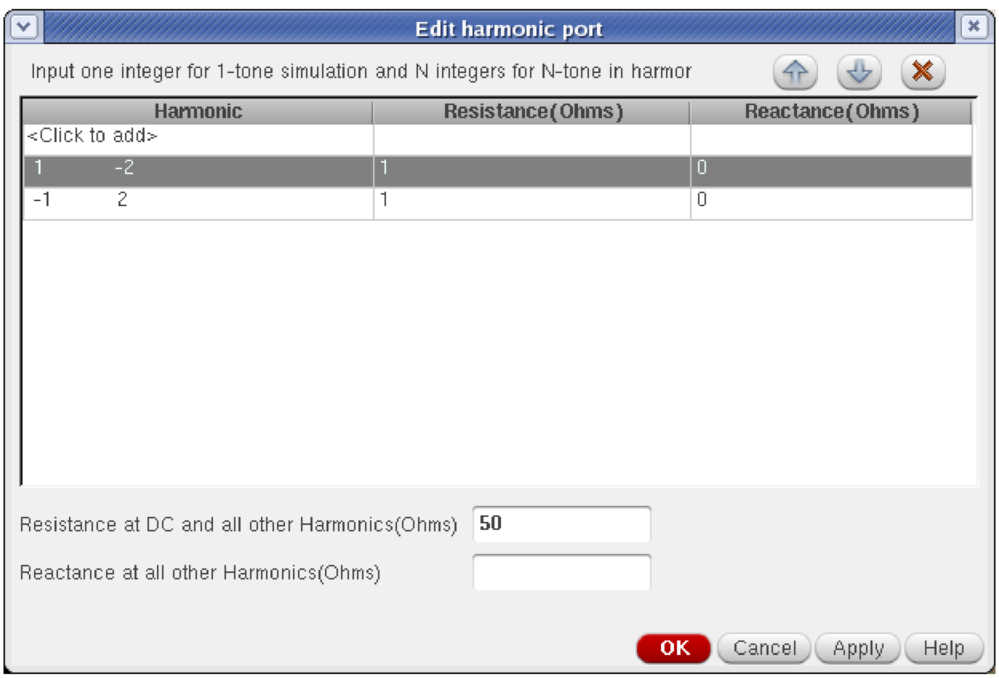
Now the harmonic balance *Choosing Analyses* form has two frequencies.



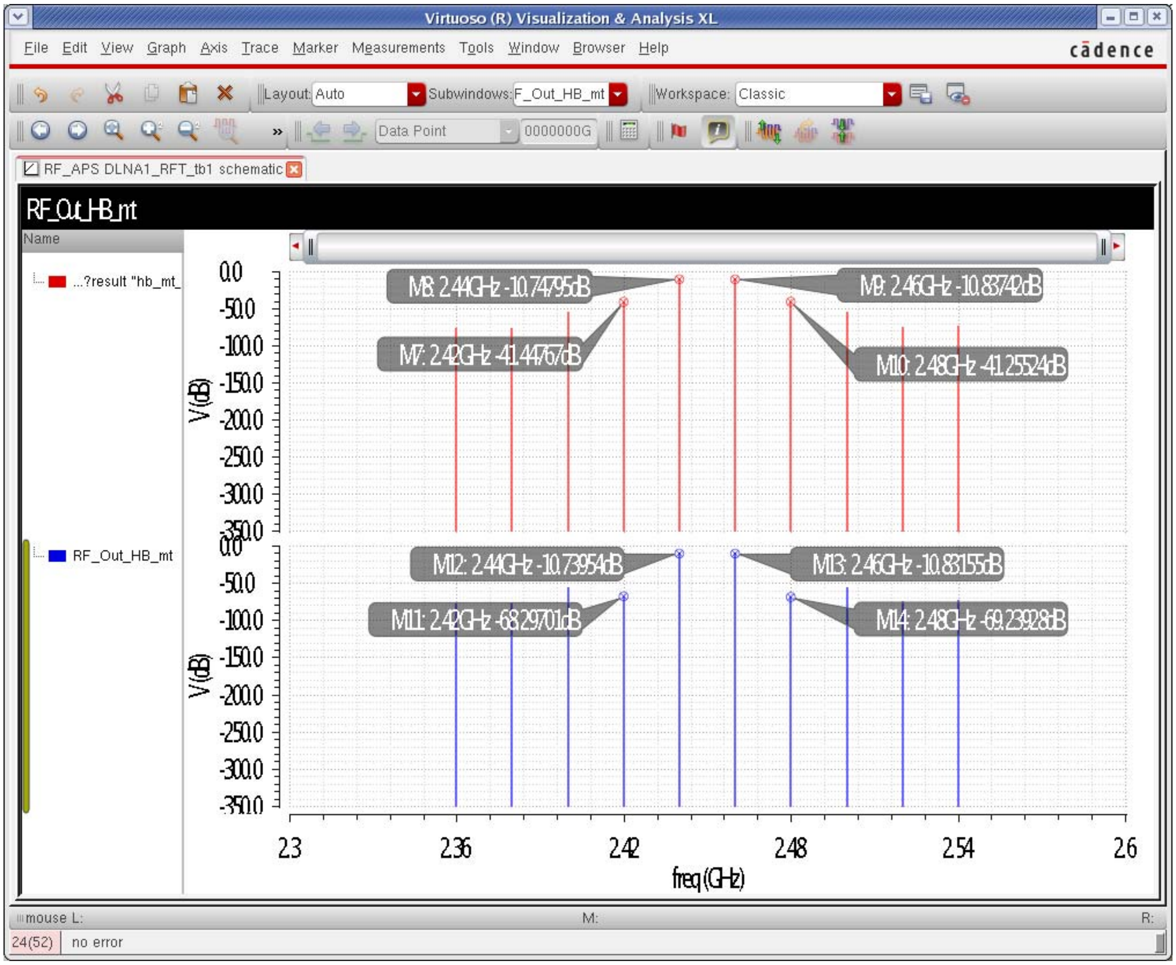
Also, the diamond cut is set with a very large *MaximOrder*. The reason for doing this is to see the harmonic numbers that correspond to the frequencies produced by the circuit. Each frequency has two harmonic numbers that indicate which harmonics are mixing with which to produce a specific output frequency. For example, take the 2.44GHz term. The harmonic number is 1 0. The frequency that is produced is 1*2.44G + 0*2.46G. Now look at the intermod at 2.42GHz. This index is 2 -1. The frequency is 2*2.44G - 1*2.46G. These are the harmonic numbers that can be set for the harmonic port. Wild cards (*) are not allowed in the harmonic number specification.

Note that if there were three input frequencies, there would be three numbers that specify every output frequency.

In this case, if the harmonic was specified as 3, there would be an error produced by Spectre because there is a single index in the harmonic, but there are 2 indices for all the harmonics of the existing simulation. The number of entries in the *Harmonic* field must match the number of input frequencies in the circuit.



The above example sets the impedance of the third-order intermods to be much lower than 50 ohms. If the source is in series, setting the resistance and/or reactance to a large number can block harmonics you don't want. If the port is in parallel like this one, setting small resistance and reactance values can short out an undesired harmonic. This can be used for debugging the circuit, or for setting values to maximize power-added efficiency. The resistance or reactance can be left blank, which is interpreted as zero resistance or reactance.



The top trace above is from the normal port, and the bottom trace is from the harmonic port. The third order harmonics are much smaller because they are essentially shorted out.

Source type

The *Source type* parameter lets you select a wave shape for the port from the *Source type* cyclic field: *dc*, *pulse*, *exp*, *pwl*, *sine, bit*, *prbs*, or blank. Each *Source type* has different parameter settings associated with it. You can define several different wave shapes and quickly switch between them without losing the wave shape settings. The wave shape settings are described in detail in the following sections:

- DC Parameters on page 1676

- Pulse Waveform Parameters on page 1677
- PWL Waveform Parameters on page 1691
- Sinusoidal Waveform Parameters on page 1716
- Exponential Waveform Parameters on page 1744,
- Bit Waveform Parameters on page 1746,
- PRBS Waveform Parameters on page 1755

The typical *Source types* used in SpectreRF analyses are: *dc*, *piecewise linear, pulsed*, and *sine*. For example, you can quickly switch from a sinusoid (for PSS analysis) to a DC level (for PAC analysis) by changing the *Source type* from *sine* to *dc*. The sinusoidal properties remain with the port, and can be changed when the *Source type* is returned to *sine*. This is true when switching between any of the sourcetypes. All the properties are retained for everything that you define.

When you set *Source type* to a blank value, the *port* acts as a resistive load.

DC Parameters

To generate a dc level from the *port* component, select *dc* in the *Source type* cyclic field. When the *Source type* is set to *dc*, the *dc* and *temperature effect* parameters are active.

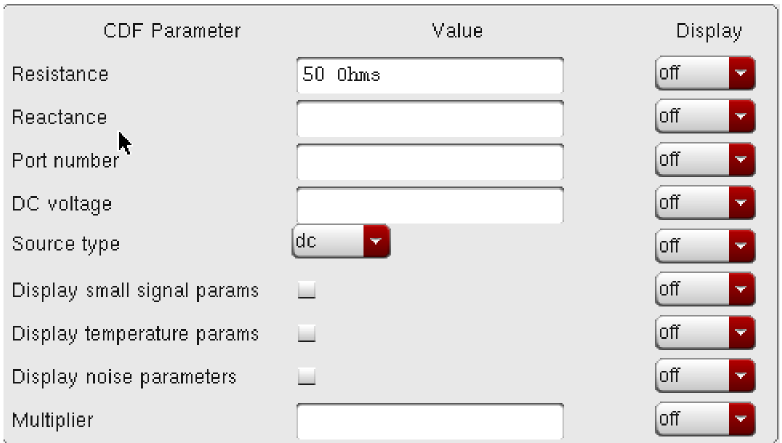
The *dc* setting sets the DC level for all analyses.

DC voltage

The *DC voltage* parameter sets the port’s DC level for all analysis. The value must be a real number. If you do not specify the DC value, the default value is 0 volts.

The *DC voltage* parameter specifies the DC voltage across the *port* when it is terminated in its reference resistance. In other words, the *DC voltage* of the internal voltage source is double the specified DC value, *dc*. Because all small signal analyses (AC, SP, XF, and Noise) use DC analysis results, the *DC voltage* level also affects the small-signal analyses.

Figure 8-3 Source type=dc in the Edit Object Properties form



The *Display small signal params, Display temperature params,* and *Display noise parameters* fields are discussed in Small-Signal Parameters on page 1802, "Temperature Effect Parameters" on page 1804, and "PRBS Waveform Parameters" on page 1755.

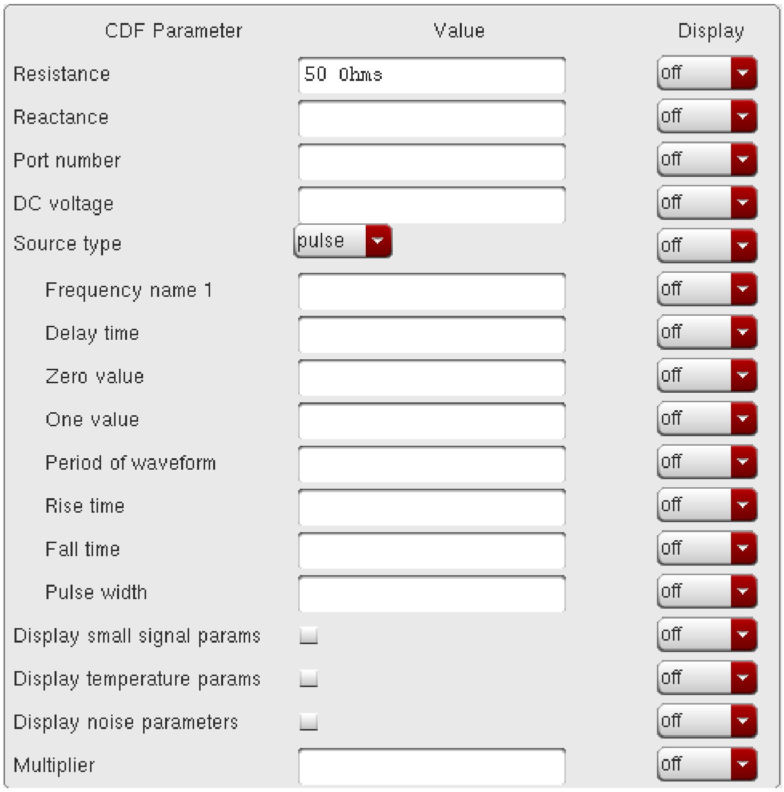
Pulse Waveform Parameters

The pulse waveform can generate a step, a single pulse, or a periodic pulse waveform.

To generate a pulse waveform from the *port* component, select *pulse* in the *Source type* cyclic field.

When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Therefore, the voltage on the internal source is set to twice the value specified on the *port*.

Figure 8-4 Source type=pulse in the Edit Object Properties form



CDF Parameter	Value	Display
Resistance	50 Ohms	off
Reactance		off
Port number		off
DC voltage		off
Source type	pulse	off
Frequency name 1		off
Delay time		off
Zero value		off
One value		off
Period of waveform		off
Rise time		off
Fall time		off
Pulse width		off
Display small signal params		off
Display temperature params		off
Display noise parameters		off
Multiplier		off

Frequency name 1

The *Frequency name 1* parameter is listed in the *Fundamental Tones* section of the *Choosing Analyses* form for pss, qpss, and hb when *Names* is selected.

Important

If you have two different sources at the same frequency or at integer multiple of the same frequency, make sure that you use the same *Frequency name* 1 or 2 property for both sources. An example might be RF1.

Delay time

The *Delay time* parameter is the simulation time that the port stays at the zero value without becoming periodic. Generally, this should be set less than the period of the waveform. Longer times are allowed, however, this forces a transient analysis until the longest delay time before the large-signal analyses pss, qpss, and hb can start. If a 1GHz signal and a delay time of 1 second is in your circuit, a transient is forced for 1 billion cycles of the input before the pss, qpss, or hb can start. This effectively prevents them from starting because of the long simulation time in the transient analysis. This must be a real number.

Zero value

The *Zero value* property is the time-zero value of the pulse in Volts. The default is zero volts.

One value

The *One value* property specifies the voltage that the signal transitions to at time equals zero plus. The default is one volt, which is unlikely to be what you want. Always set this property.

Period of waveform

The *period* property specifies the period of the pulse. If this property is left blank, a single pulse is generated at the beginning of the transient simulation.

Rise time

The *Rise time* property specifies the zero to 100% risetime for the pulse. This is the time for the complete transition from the zero value to the one value.

Fall time

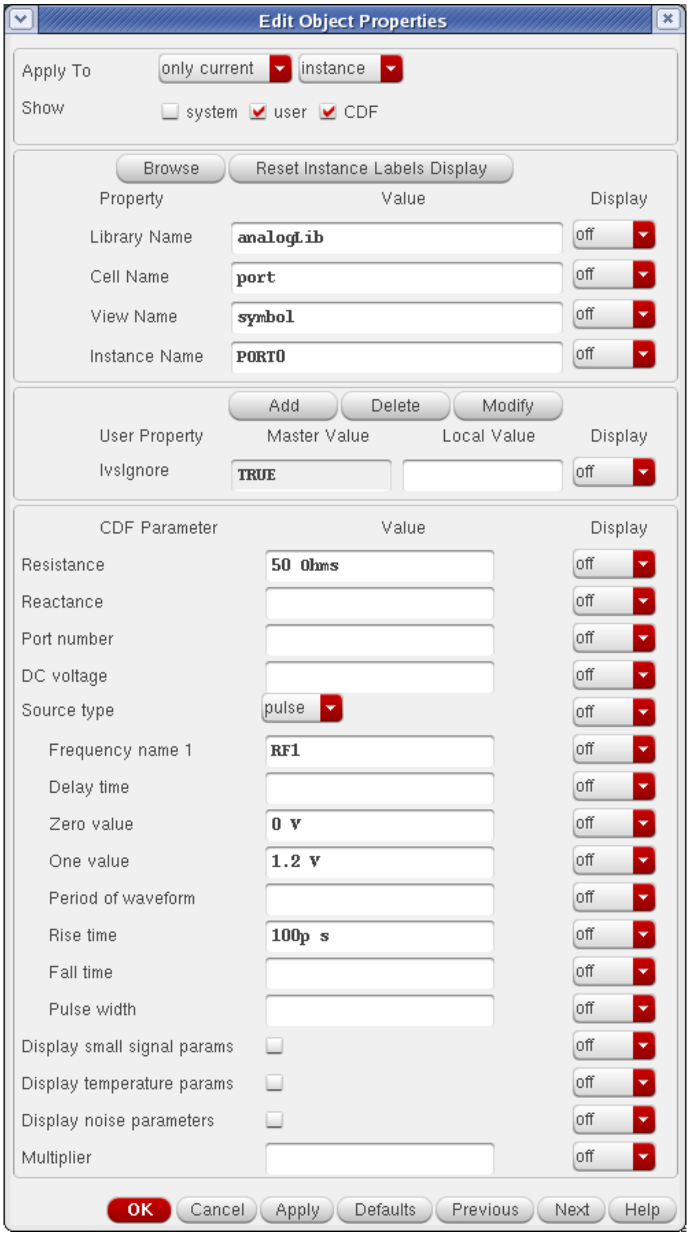
The *Fall time* property specifies the zero to 100% falltime for the pulse. This is the time for the complete transition from the one value to the zero value.

Pulse width

The *Pulse width* property specifies the time to remain at the one value. The default is infinity.

Pulse Waveform Examples

Generating a step from zero to 1.2 Volts

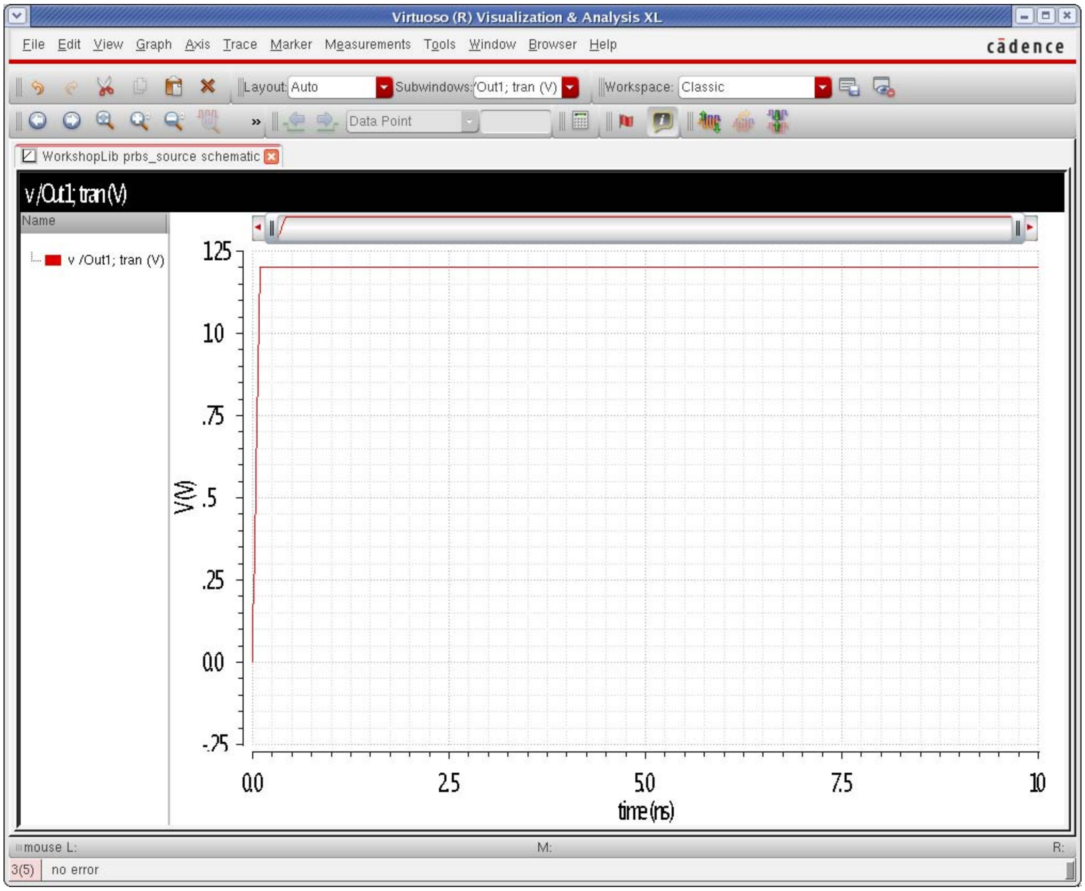


1. Set the *Zero value* field to 0 volts.
2. Set the *One value* field to 1.2 volts.

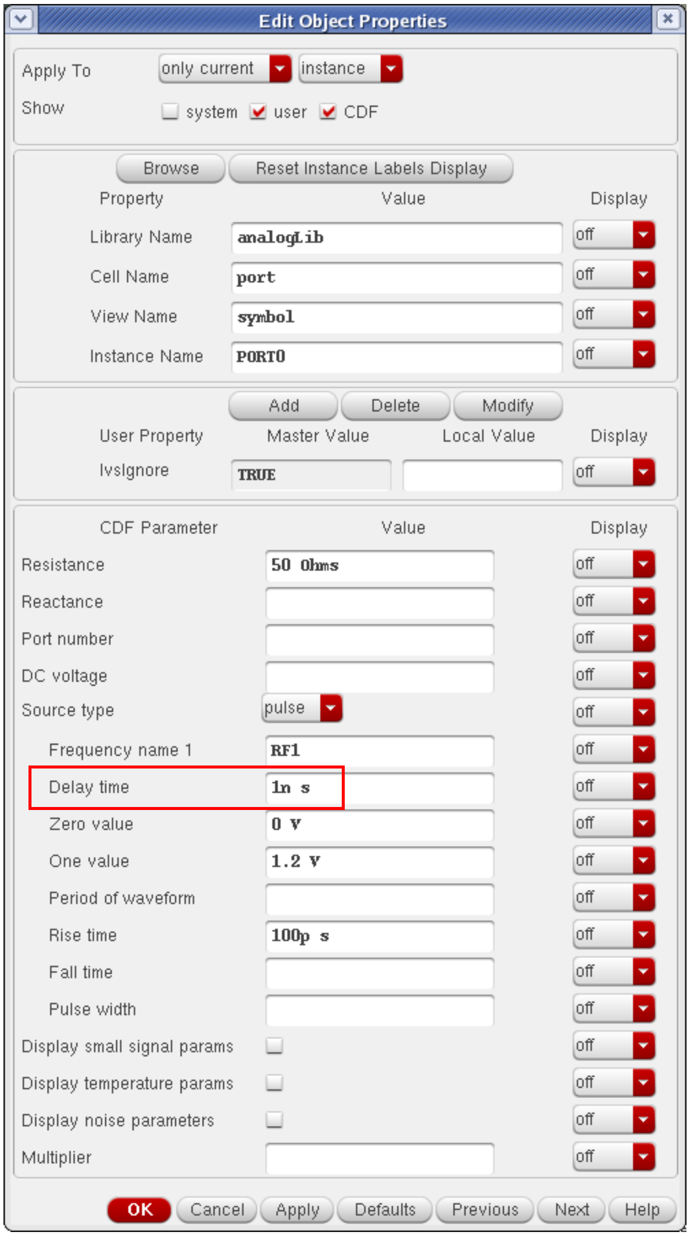
3. Set the risetime to a time that is appropriate for your circuit. A suggested starting value is about $\frac{1}{10 \times CircuitBandwidth}$.

The default is the falltime. If the falltime is not given, the default is $\frac{Period}{100}$. If the period is not given, then the default is $\frac{SimulationTime}{100}$.

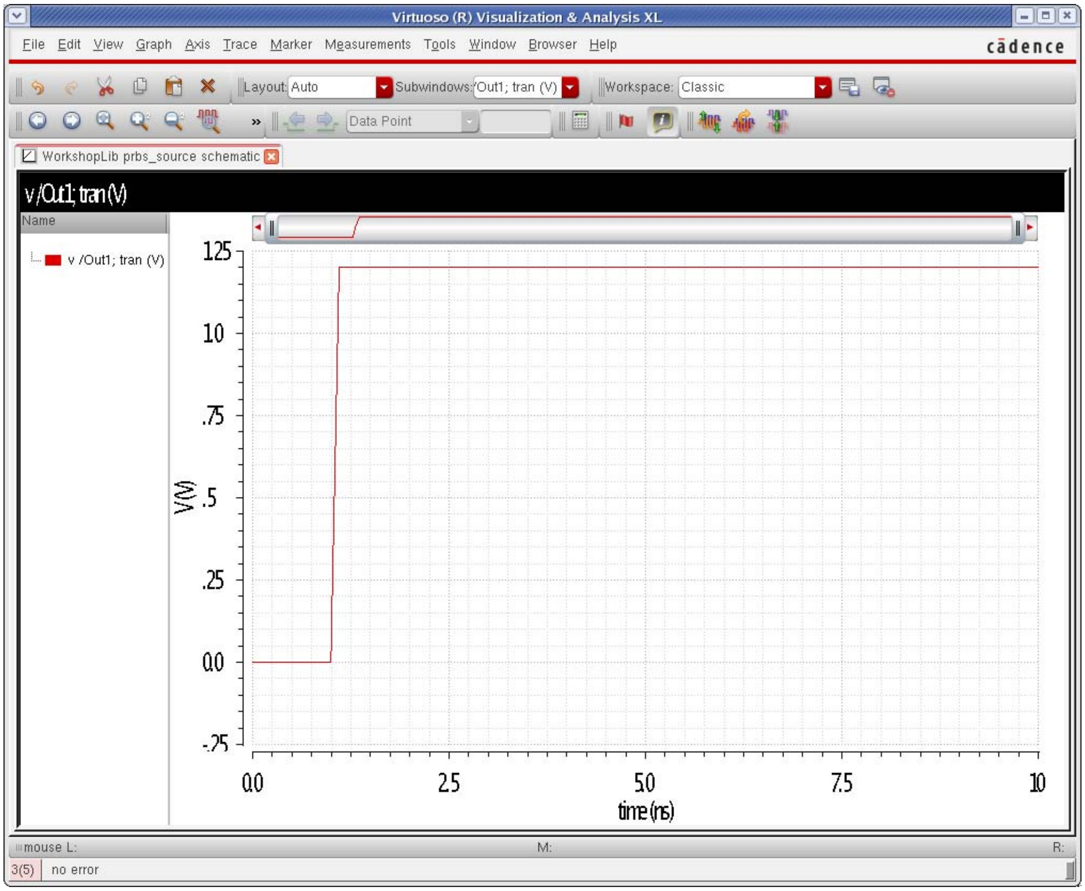
The waveform is shown below.



4. Setting a *Delay time* to 1n shifts the waveform to the right.



The waveform is shown below.

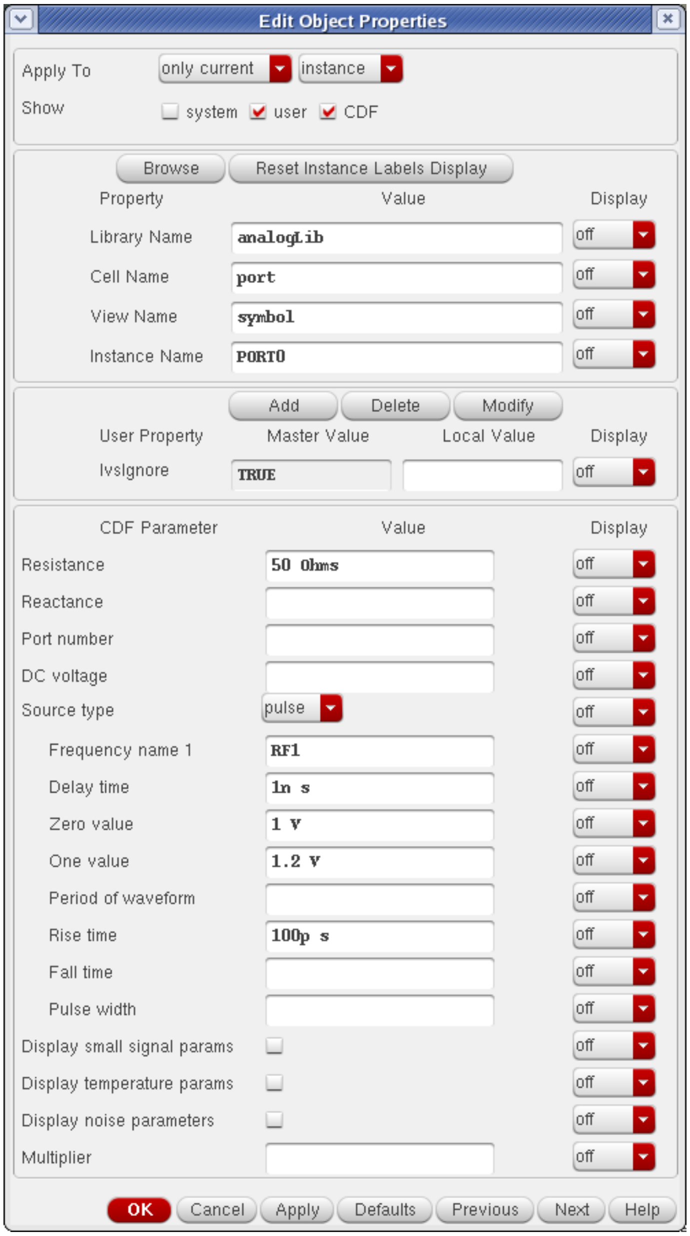


Specifying a startup bump in a supply voltage

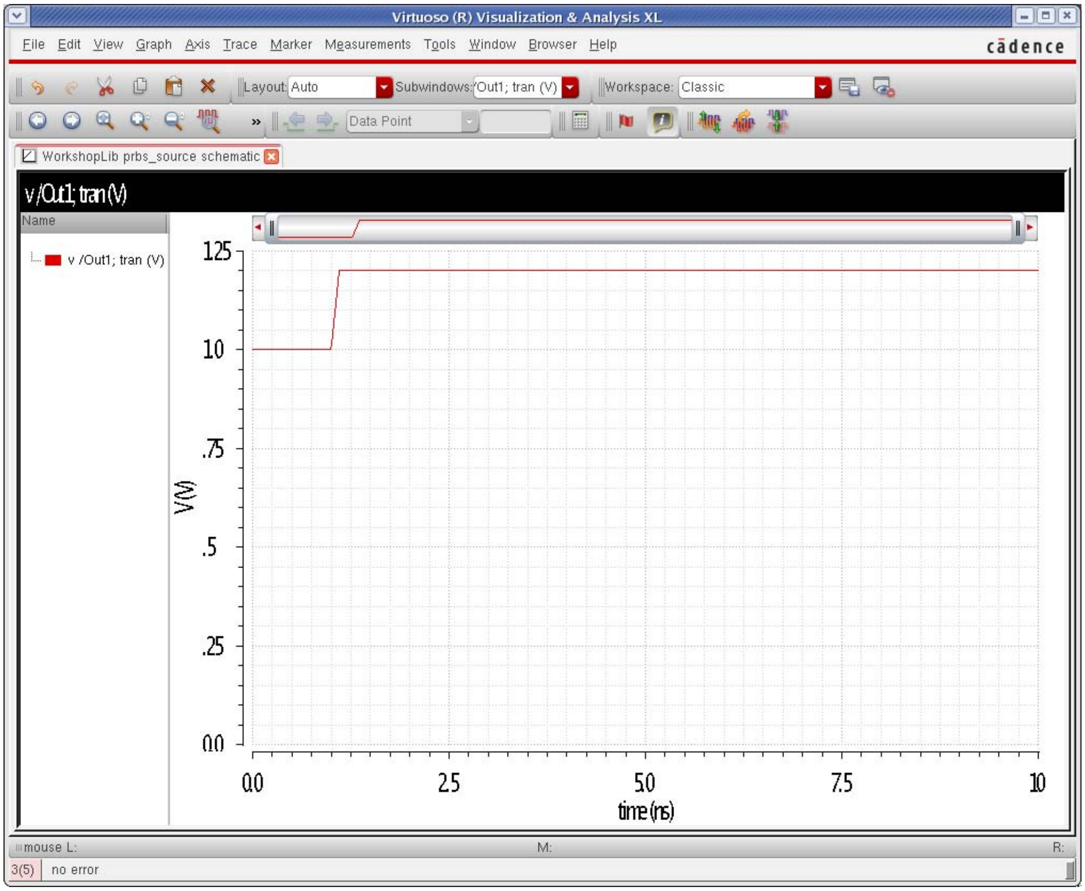
This is commonly used to start oscillators, and is usually done using a `vsource`.

1. Set the *Zero value* to about 80% of the supply voltage.
2. Set the *One value* to the supply voltage.

3. Set the *Rise time* as appropriate for the system you are simulating.

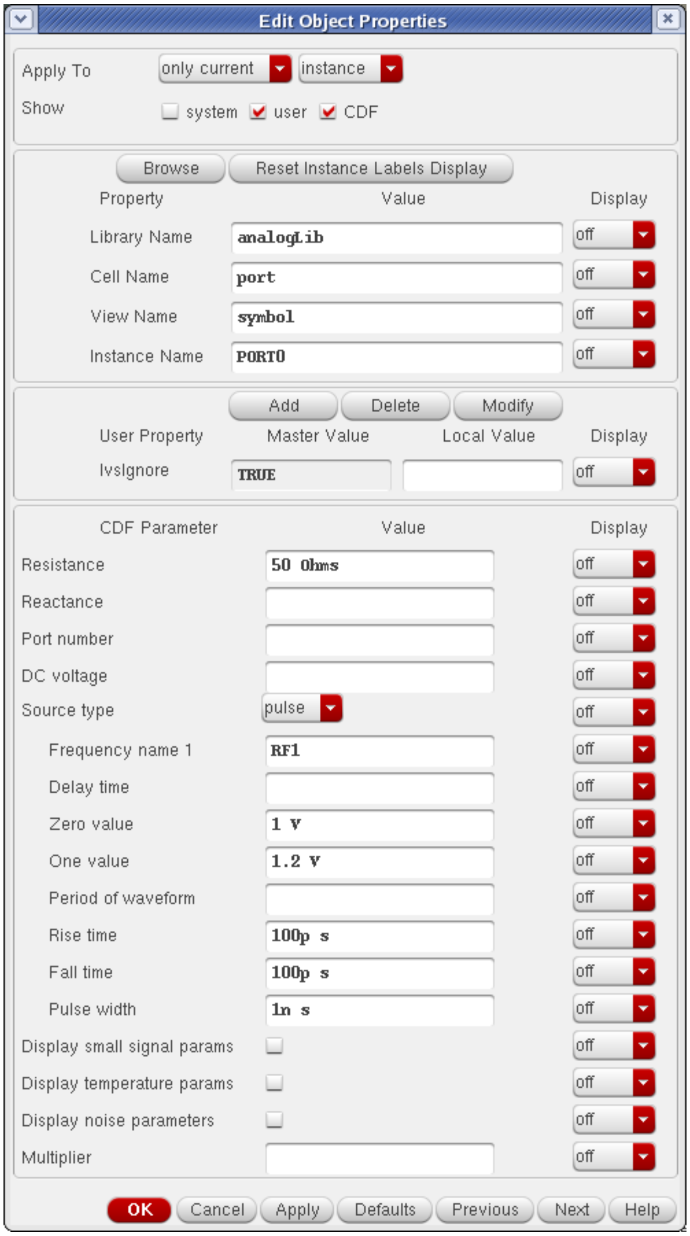


4. The waveform is shown below.



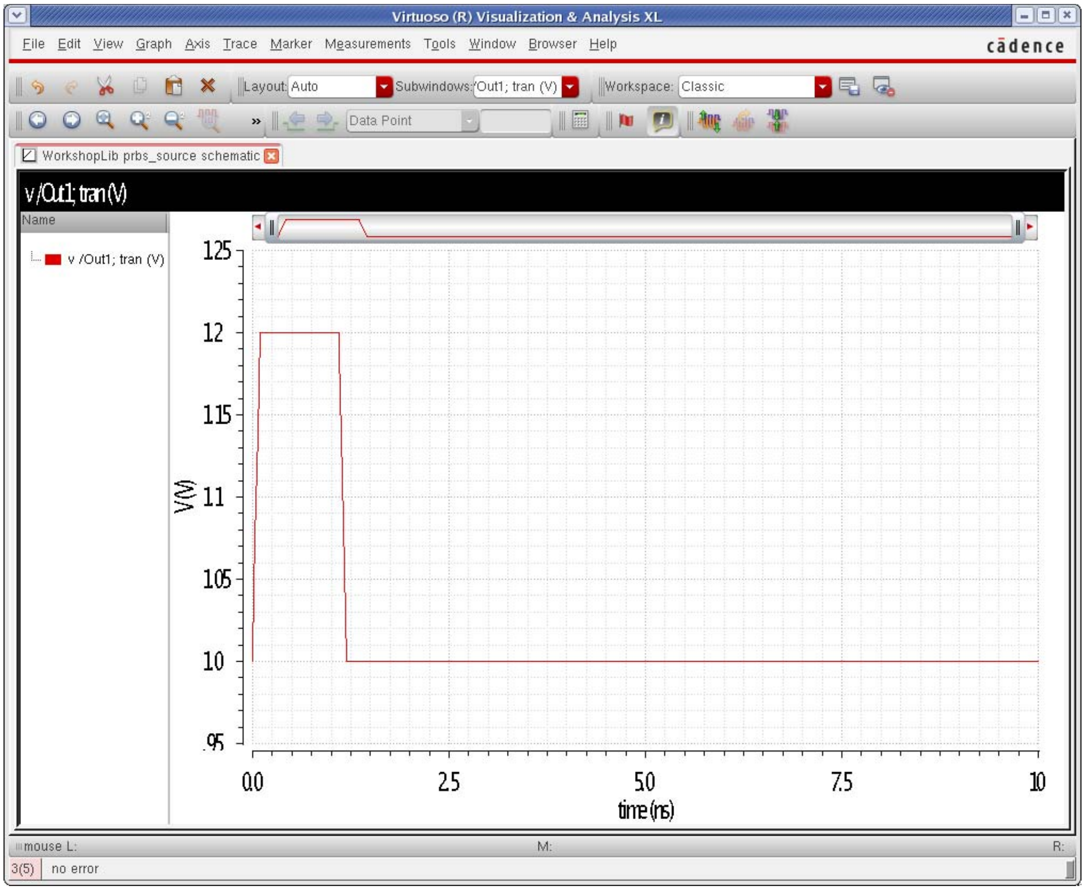
Specifying a single pulse at startup, then zero after that.

This is usually used with the ipulse or isource components to start an oscillator.



1. Set the *Zero value* to zero.
2. Set the *One value* large enough to get things started.
3. Set the rise and fall times to about one tenth of the period of the oscillator.

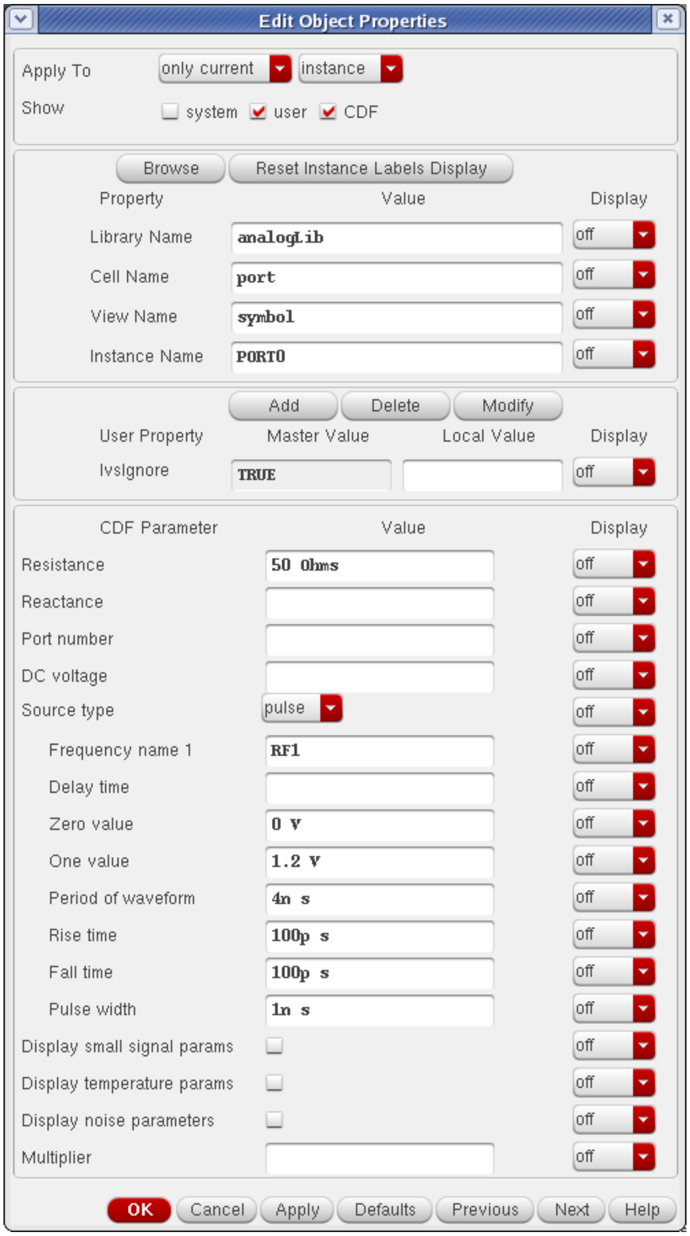
4. Set the *Pulse width* to about three tenths of the period.
5. Leave the *Period of waveform* field blank.
6. The waveform is shown below.



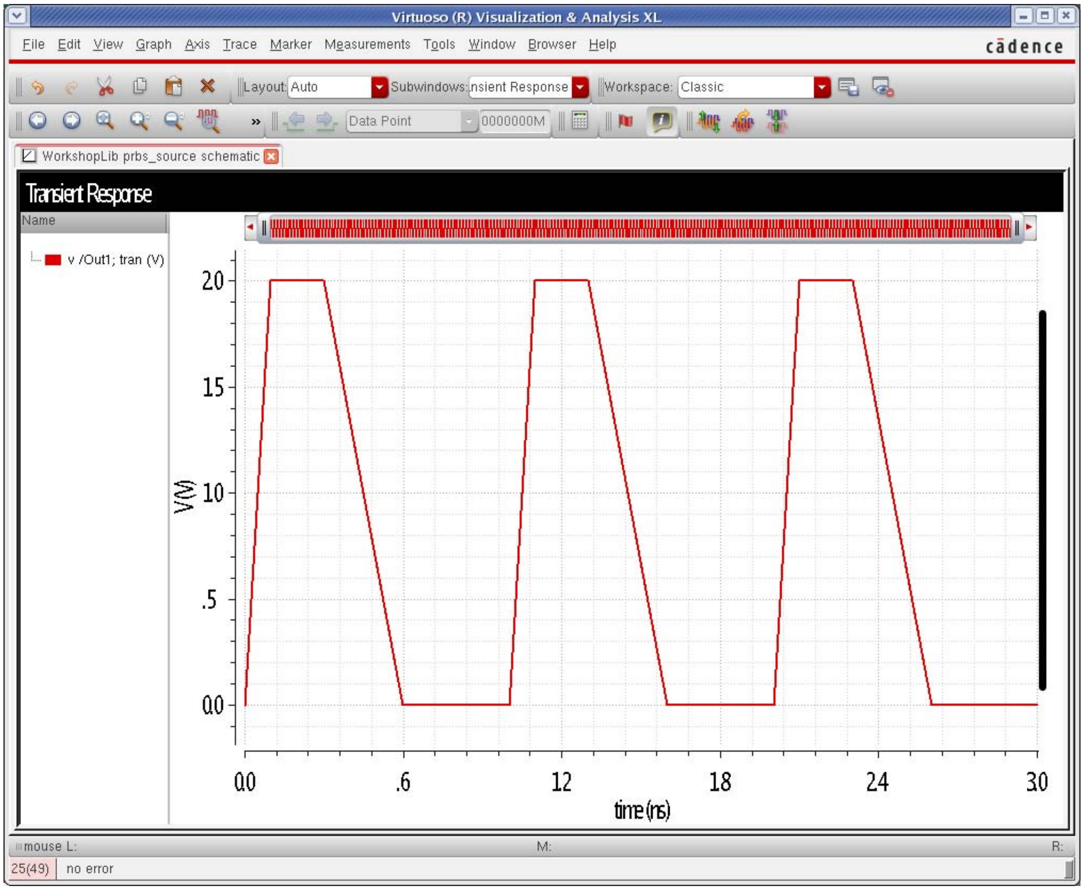
To generate a pulse with an arbitrary duty cycle and differing rise and fall times:

1. Set the *Zero value* and *One value* fields to the levels needed for your system.
2. Set the *Rise time* and *Fall time fields* to approximate the real input source. The values can be different.
3. Set the *Period of waveform* field to the value needed for your frequency.

4. Set the *Pulse width* between 0.05 to about 0.95 times the period.



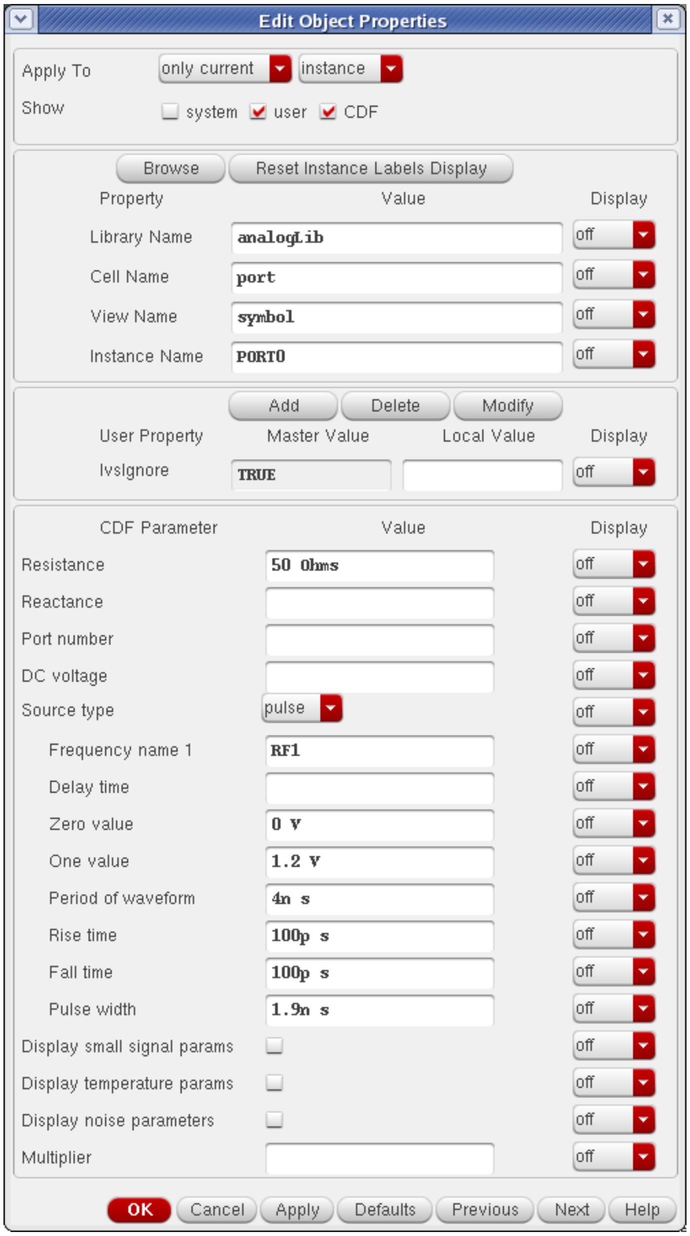
5. The waveform is shown below.



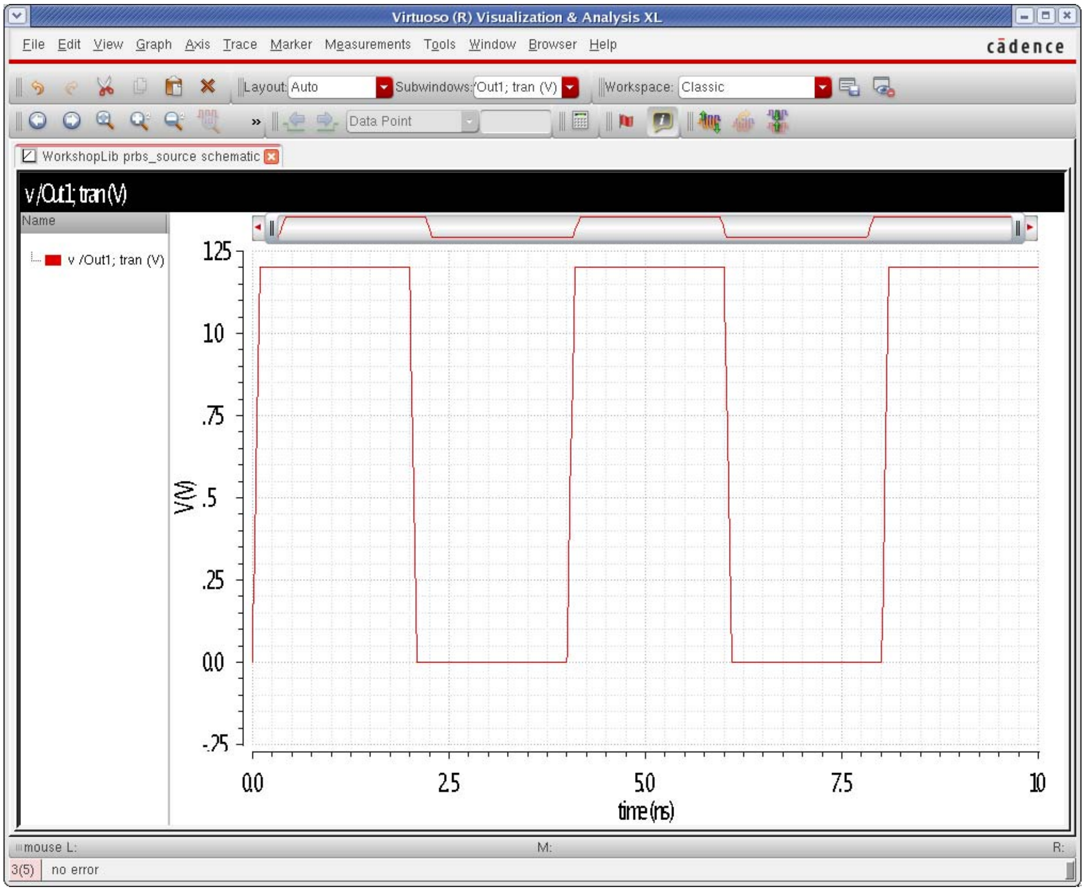
To generate a 50% duty cycle square wave:

1. Set the *Zero value* and *One value* fields as appropriate for your system.
2. Set the *Rise time* and *Fall time fields* as appropriate for your system and make them equal.
3. Set the *Period of waveform field* to that needed to generate your input frequency.

4. Set the *Pulse width* to half the period minus the risetime.



5. The waveform is shown below.



PWL Waveform Parameters

Piecewise linear waveforms allow an arbitrary input waveform to be generated. The input can either be a file that contains time and voltage pairs, or you can enter the time-voltage pairs directly in the PWL source properties form. Remember that the voltages you enter in the piecewise linear file assume that the port is properly terminated. The internal voltage source gets set to double the value specified in the piecewise linear voltage specifications.

Important

If the period property is set, the source repeats the waveform between zero and period over and over again. If the period property is not set and the transient analysis goes past the last point in the PWL file, the last voltage value in the file is held until the end of the simulation. If the period property is not set, the signal does not become periodic until after the time of the last point in the PWL file, where the source output becomes a DC level. If the file has a stopping point that has a large time, then the periodic analyses pss, qpss, and hb run a transient until the longest time in any input PWL file has been reached before they go into the steady-state analyses, and these analyses use the DC level that is present at the end of the PWL file as the input signal.

Important

If you want a periodic piecewise linear waveform for pss, qpss, and hb, make sure that the period property is set.

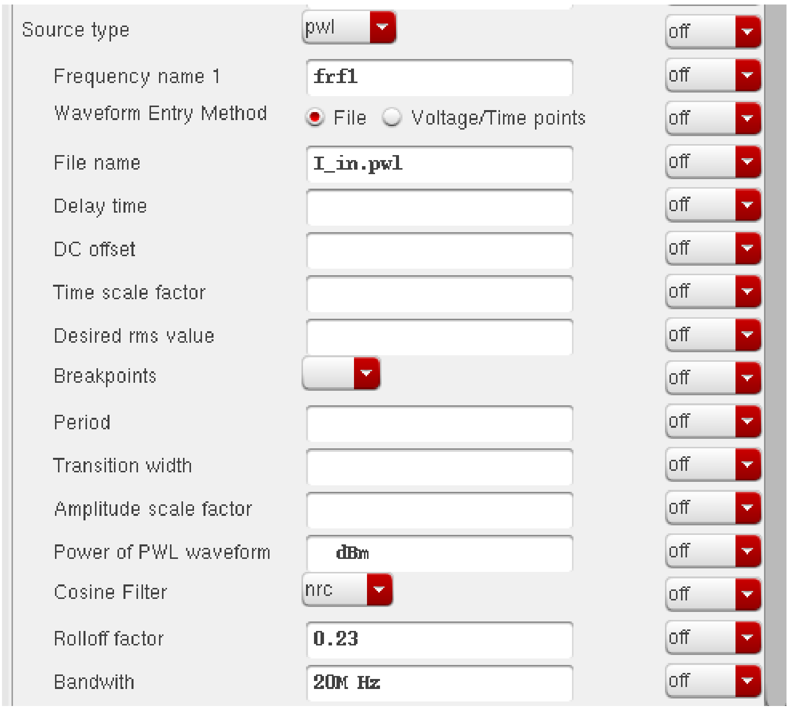
To generate a piecewise linear waveform from the *port* component, select *pwl* in the *Source type* cyclic field. This sets the *Source type* CDF parameter to *pwl* and displays additional fields for the PWL CDF parameter settings.

Important

The advantage of entering a square or pulse wave using the pwl source is that the starting value can be at the center of the waveform. In the pulse waveform, the signal starts at the zero value, and because of this, in some cases, the tstab time must be extended to allow the waveform to settle. Starting the waveform in the middle can drastically reduce tstab and/or dramatically improve convergence for systems that have square or pulse waves and relatively long settling times for the nodes with the square or pulse wave signal.

When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Therefore, the voltage on the internal source is set to twice the value specified on the *port*.

Figure 8-5 Source type=pwl in the Edit Object Properties form



Waveform Entry Method

With the *Waveform Entry Method* buttons, select how you enter piecewise-linear data,

- By specifying a *File name*.
- By entering a series of *Voltage/Time points* directly in the properties list for the port.

File name

When you select *File* as the *Waveform Entry Method*, type the name of the file containing your piece wise-linear data in the *File name* field. The file name must be a string. There is no default. In ADE, define the directory that contains the file by selecting *Setup - Simulation Files*, and entering either the rooted path (starts with slash (/)) or relative path (starts with dot (.)) in the *Include Paths* field. Multiple directories can be specified in the *Include Paths* fields.

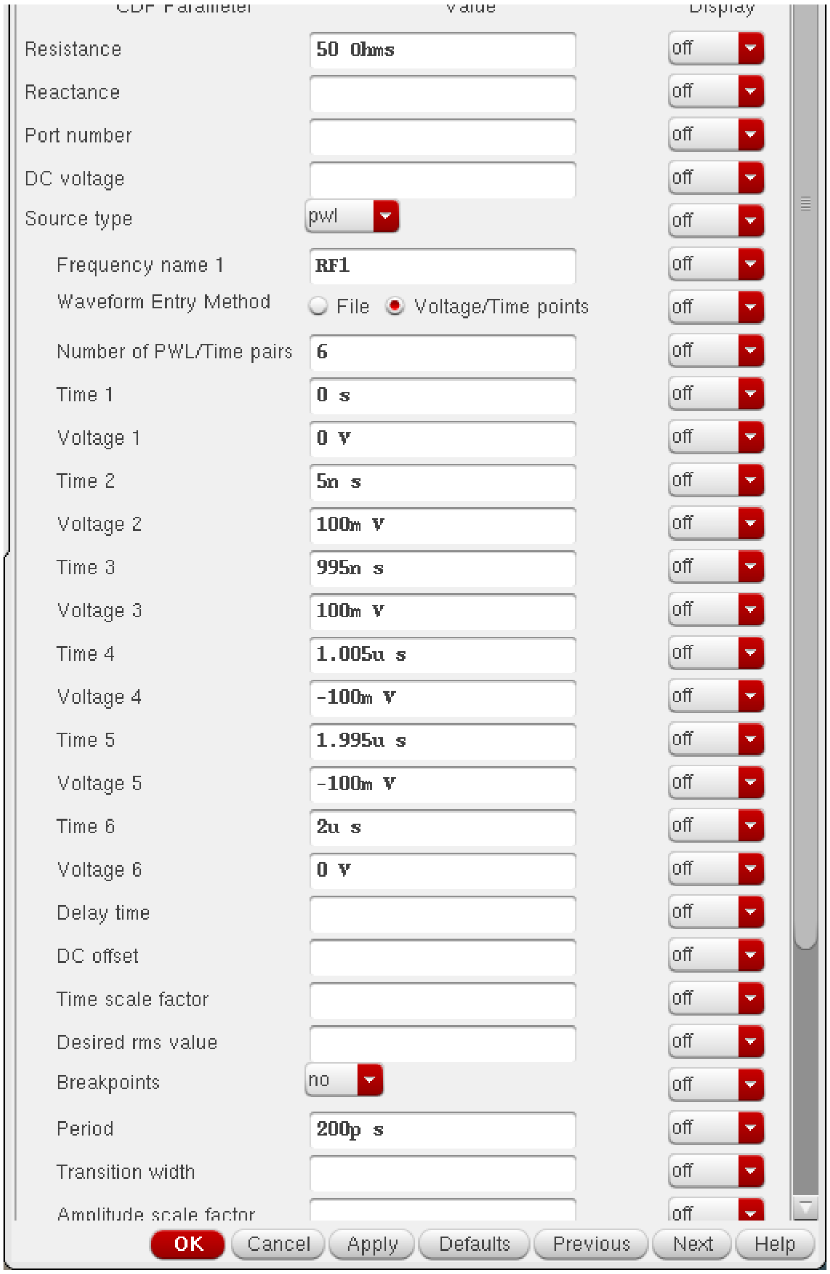
In your file, list the piecewise-linear data in the form of time-value pairs. Enter one pair per line with a space or tab between the time and voltage values. The numbers in the file must use exponential notation, for example, 1.04e-3. You can also use SI scale factors (p, n, u, m, k, M, G, etc.).

Number of PWL/Time pairs

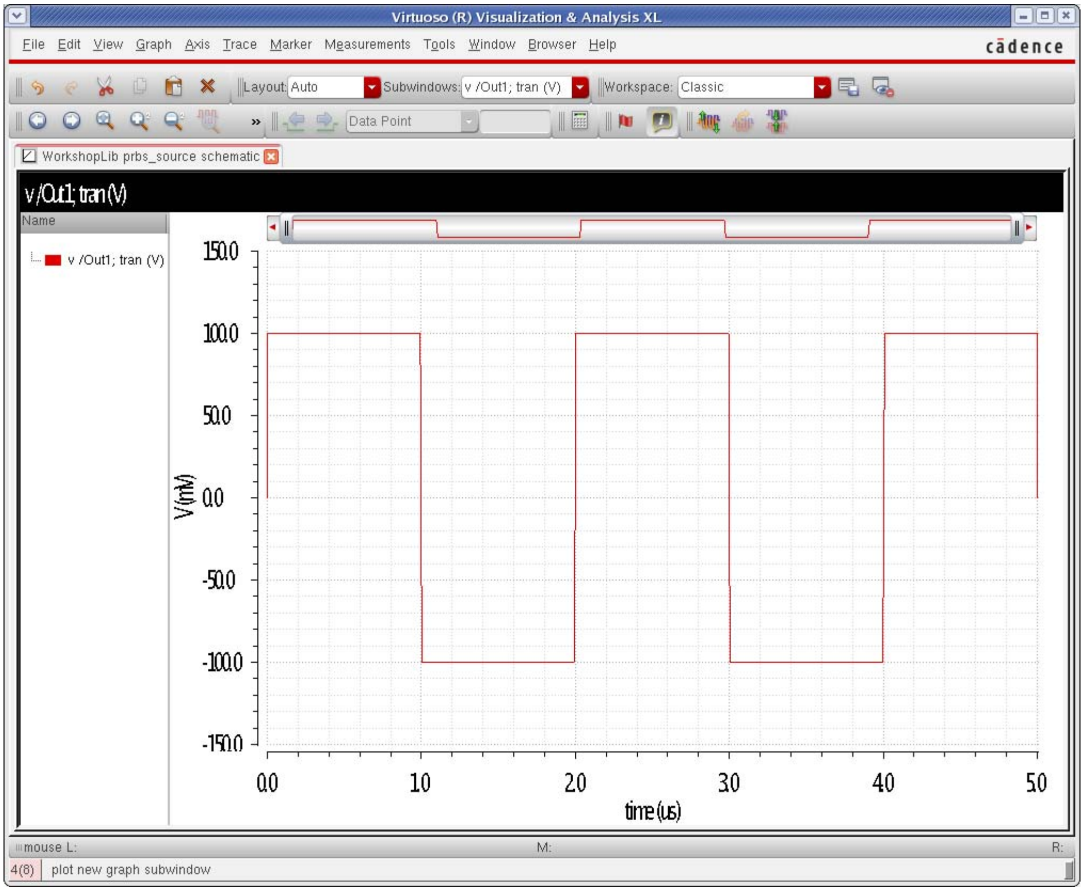
When you select *Voltage/Time points* as the *Waveform Entry Method*, the *Number of PWL-Time pairs* field (tvpairs) form opens. Enter the number of time-value pairs you plan to enter. The form expands to let you enter the designated number of *Time* and *Voltage* values. Units are seconds and volts. The default is 0 and the maximum value is 50.

In this example, the number of voltage-time pairs is 6.

Figure 8-6 Waveform Entry Method=Voltage/Time points



The waveform that is produced is shown below.



The *DC offset, Amplitude scale factor,* and *Time scale factor* parameter fields let you quickly adjust the amplitude, frequency, and offset of your piecewise-linear data pairs without editing each individual time-value pair in the *PWL* waveform.

DC offset

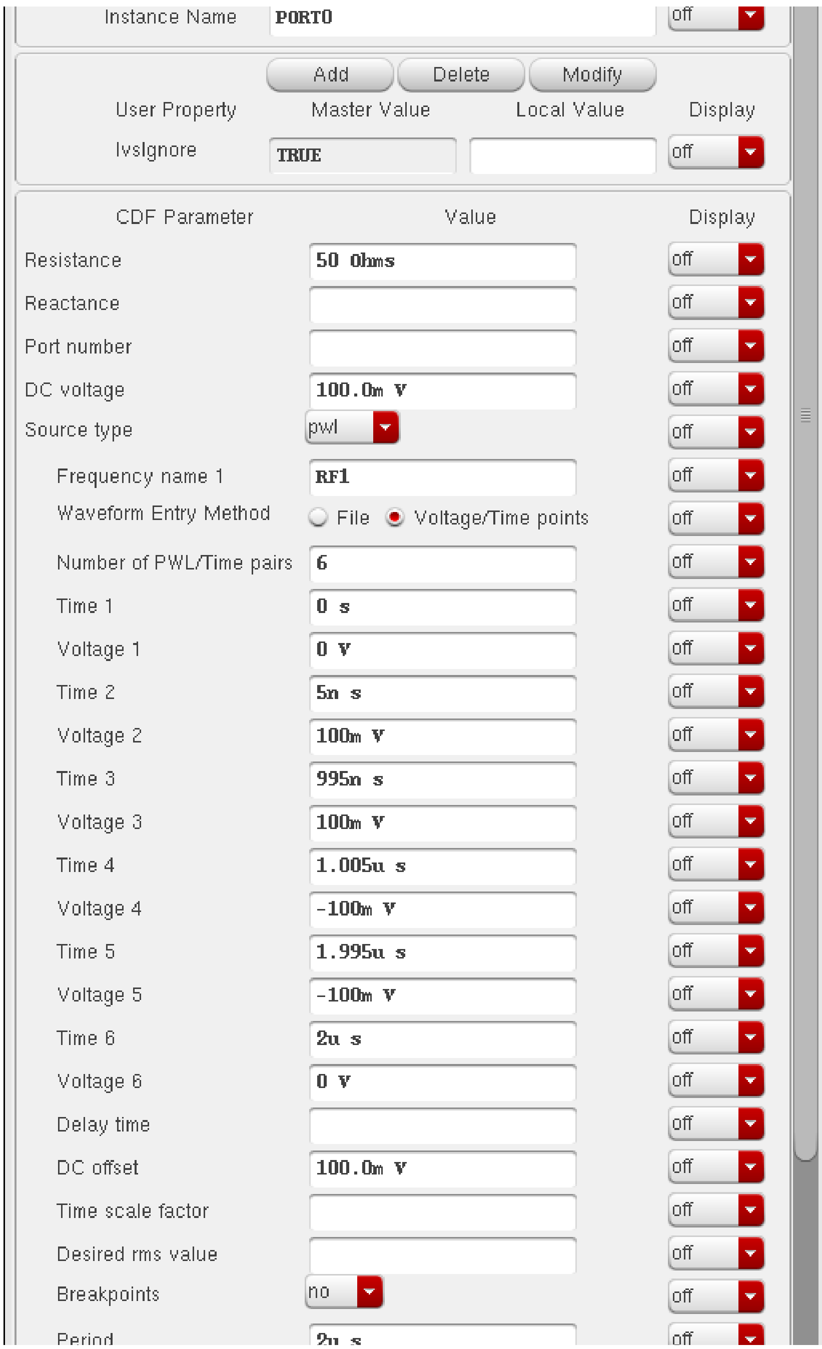
DC offset for the *PWL* waveform. The default is zero volts.

Important

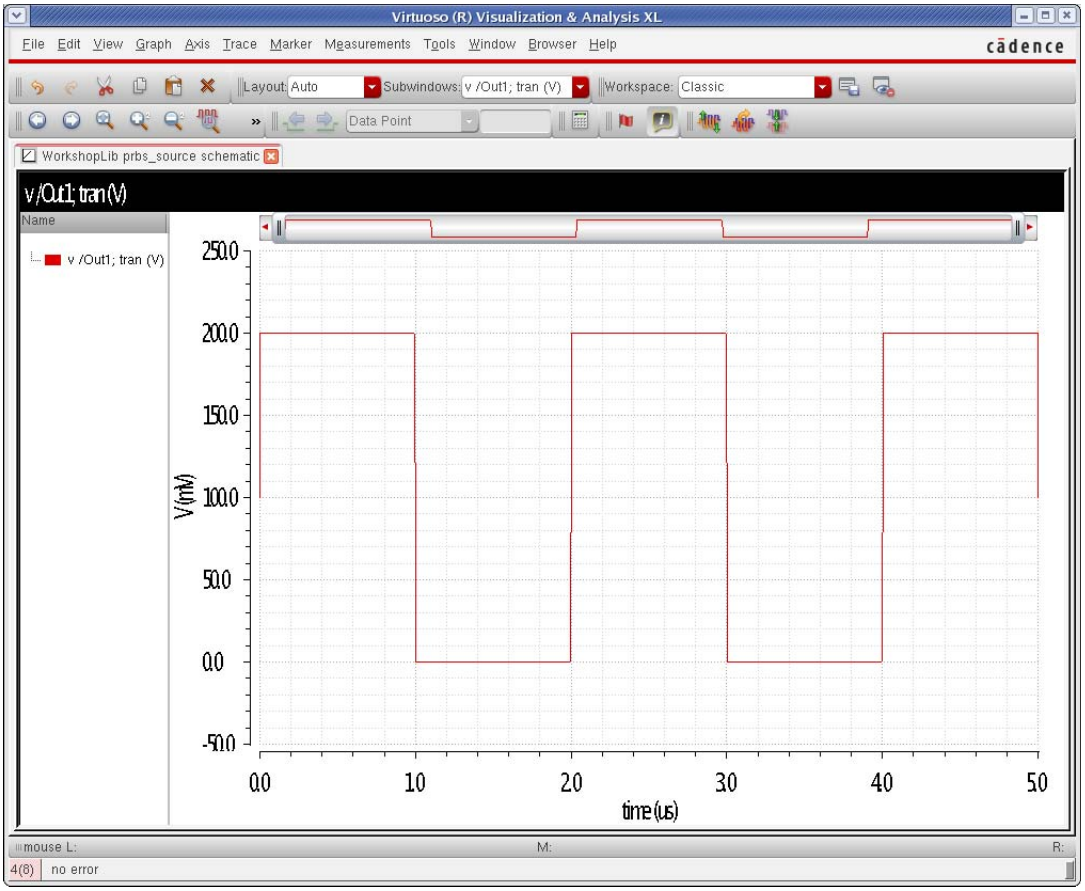
Note that for PWL waveforms, the DC Voltage term in the port only applies for the DC based analyses and not for the PWL waveform itself. To shift the DC level of the PWL waveform in the large-signal analyses (tran, envlp, pss, qpss, and hb) use the

DC offset property. When the DC voltage property and the DC offset property are both set, the actual offset value used in the large signal analyses is defined by the DC offset property by itself, and not the DC voltage plus the DC offset property.

To show the effect of setting both the DC voltage and DC offset properties, in the following example, the DC Voltage and the DC offset are both set to 0.1.

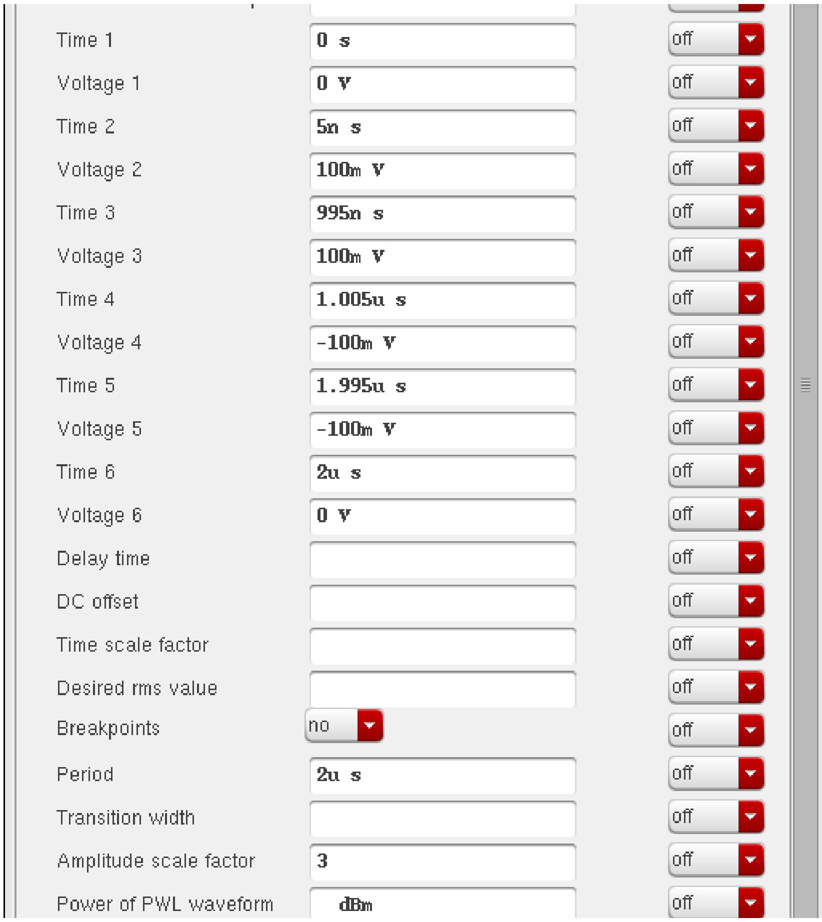


Note that the signal only observes the DC offset in the transient.

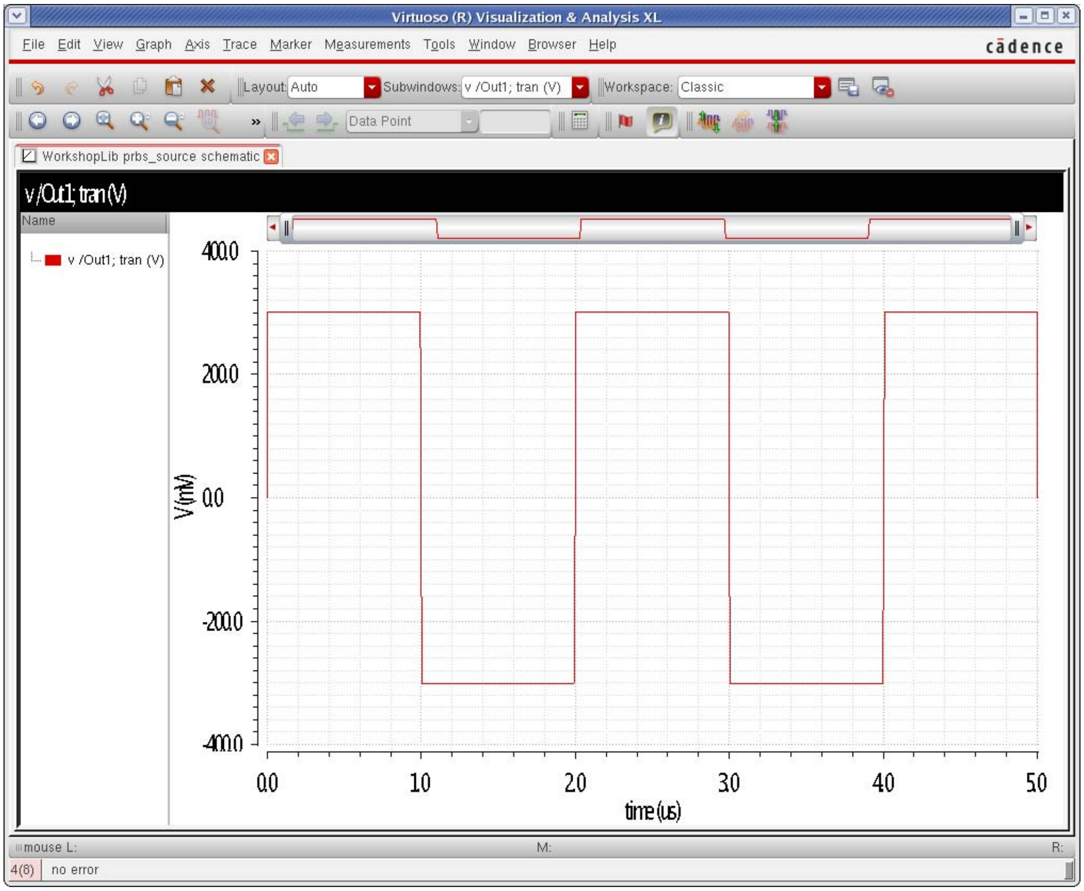


Amplitude scale factor

Amplitude scale factor (*scale*) for the *PWL* waveform. The default value is 1. The example below shows a scale factor of 3.

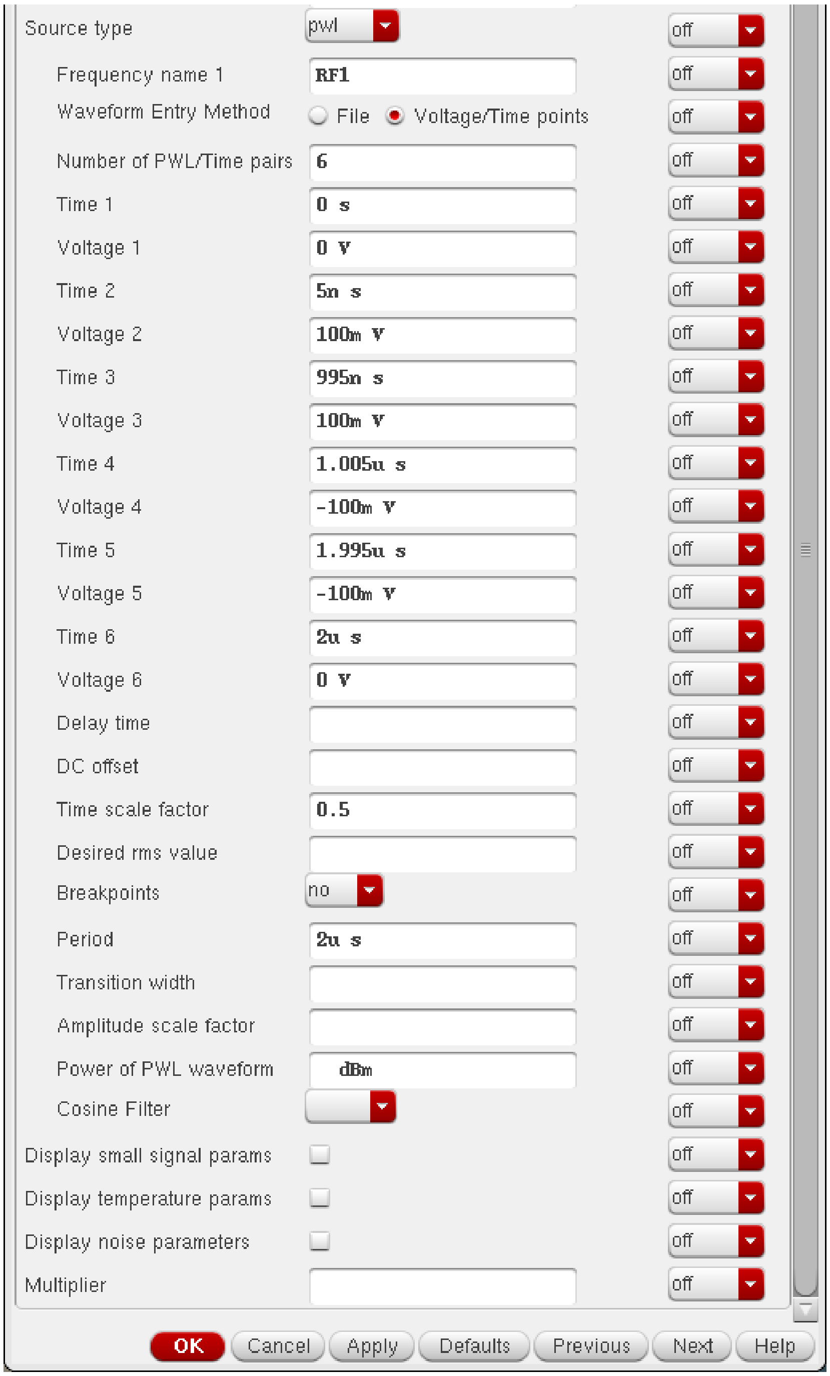


Note that the waveform is multiplied by three in amplitude.

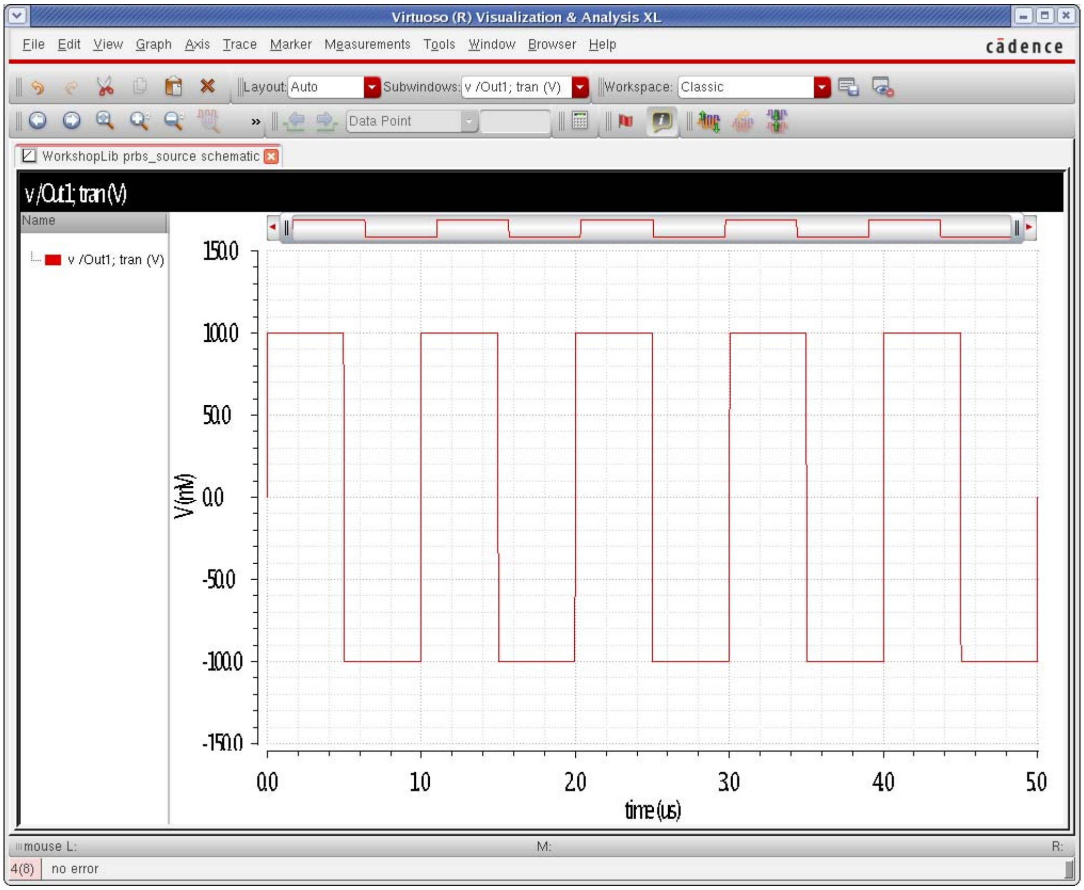


Time scale factor

Time scale factor is a multiplier for the times given for the *PWL* waveform. The default is 1. Setting the time scale factor greater than 1 increases the times in the file, and reduces the frequency.

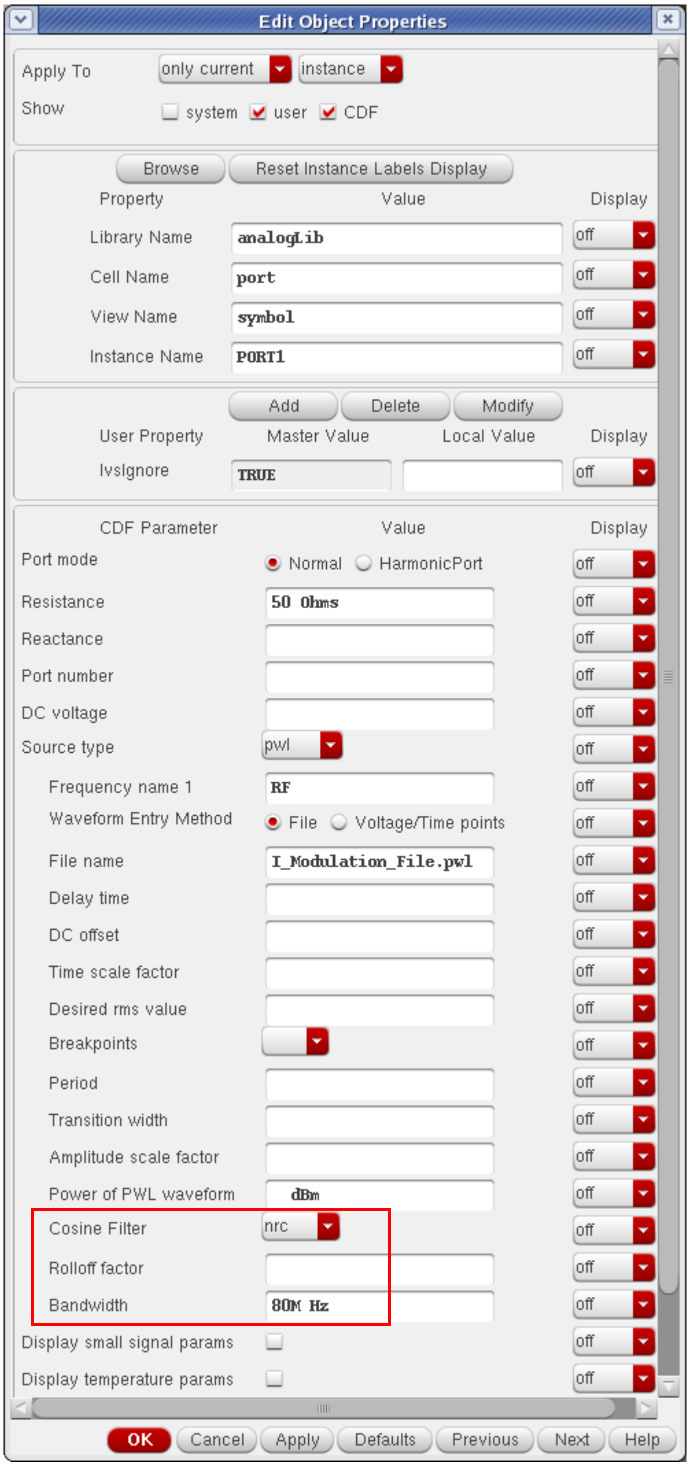


Note that setting the time scale factor to 0.5 multiplies all the times including the time in the period property.



Cosine Filter

The raised cosine filter is used when the PWL source is used to define digital modulation. Values are `none` (default), and `nrc`. `nrc` specifies a raised cosine filter to be used on the output waveform of the PWL source. This can be used to lower the numerical noise floor of the modulation specified in the I and Q modulating file. When `nrc` is selected, the PWL file must have evenly spaced points with timepoints at 1/(2*Bandwidth of the filter).



Rolloff factor

When `nrc` is selected, the *Rolloff factor* (*rolloff*) of the filter can be specified. The default value is `0.2`, which is reasonable for simulations. If you consider a continuous waveform, changing the rolloff factor has an effect on the ringing that is produced in the time domain. Smaller rolloff factors produce more ringing in the continuous waveform. Envelope does not simulate the continuous waveform. Instead, envelope samples the waveform at the time interval in the I and Q modulation file. This delta-time in the I and Q file is always related to the modulation frequency, and it also determines the bandwidth of the cosine filter as described below. The effect of this is that the sampling occurs at the times where the ringing of the filter is zero. Therefore, the cosine filter does not change the spectral content calculated in envelope. It just reduces the numerical noise floor of the input signal.

Bandwidth

The *Bandwidth* (*pwlbandwidth*) of the Cosine Filter. If you want to only allow in-band modulation, this is the channel bandwidth. If you want to allow an accurate ACPR measurement, set the bandwidth to three times the channel bandwidth. If you want to measure two adjacent channels for ACPR, set the bandwidth to five times the channel bandwidth. Note that the I and Q data must have properly spaced time intervals of $\frac{1}{2 \times Bandwidth}$ for this capability to be used.

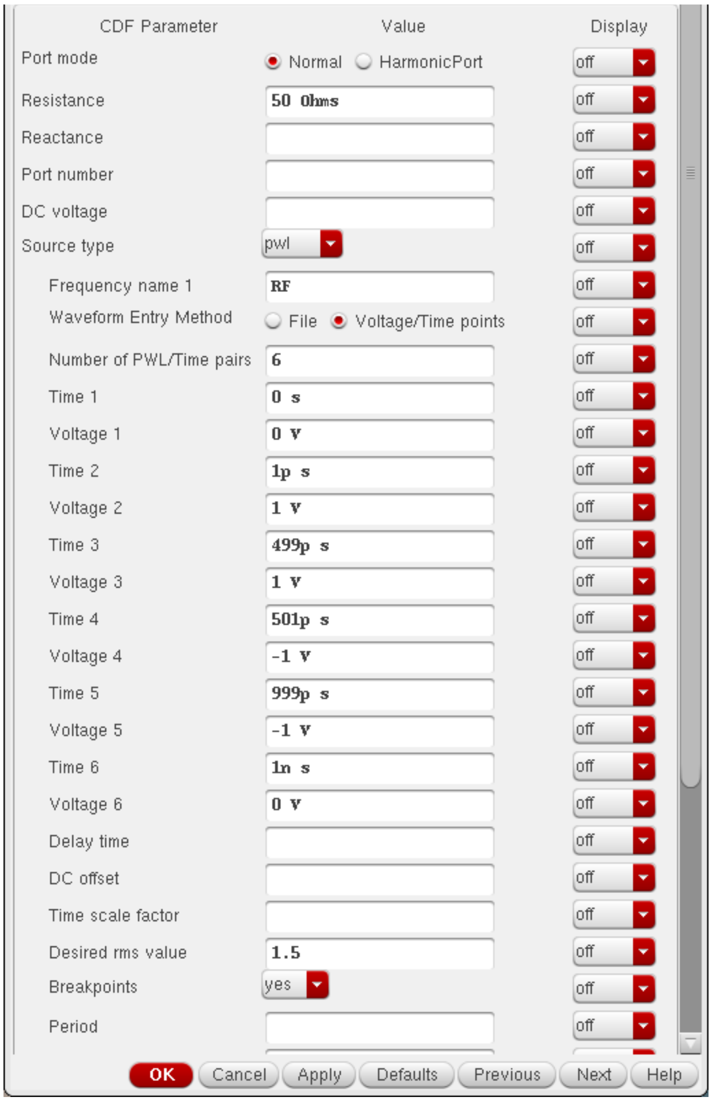
Note: The *Rolloff factor* and *Bandwidth* options are available only when `nrc` is selected in the *Cosine Filter* drop-down list.

Desired rms value

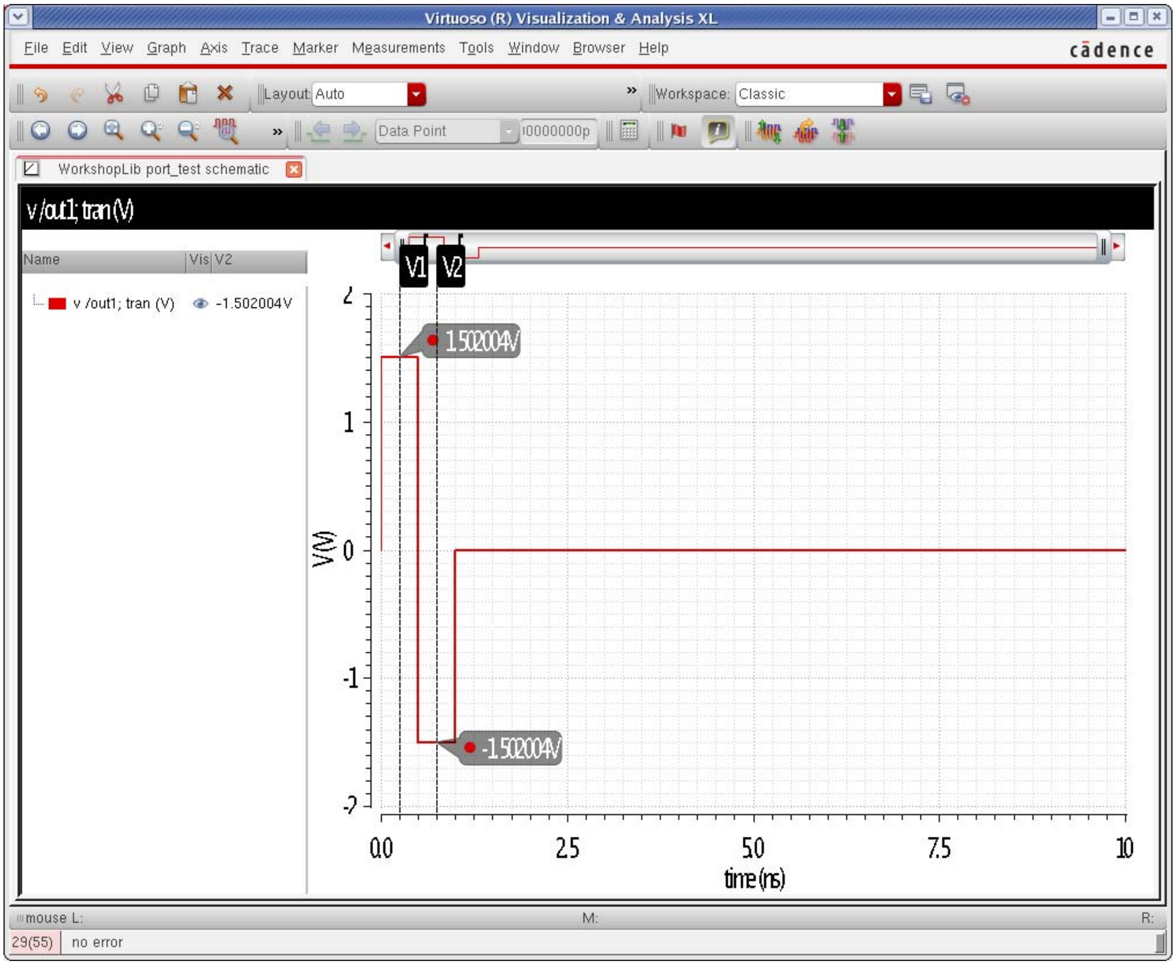
Desired rms voltage for the PWL waveform. Note that this will be set based on the datapoints in the PWL file or in the manually defined points entered directly into the properties form only. This has implications for periodic piecewise linear or if you go beyond the last point in the file in the transient analysis.

If the signal is defined for only part of the period or if you go beyond the last point in the pwl description, the rms calculation will not take into account the fact that the signal remains at the last voltage after the last timepoint in the pwl file. If you want this to be taken into account, define a point in the pwl description at the stop time of the transient or at the end of the period.

Consider an example in which a 1GHz signal is being generated, with points for the first period of 1GHz defined. No points after 1nsec exist in the PWL description. Also note that the desired RMS value is set to 1.5 (volts).

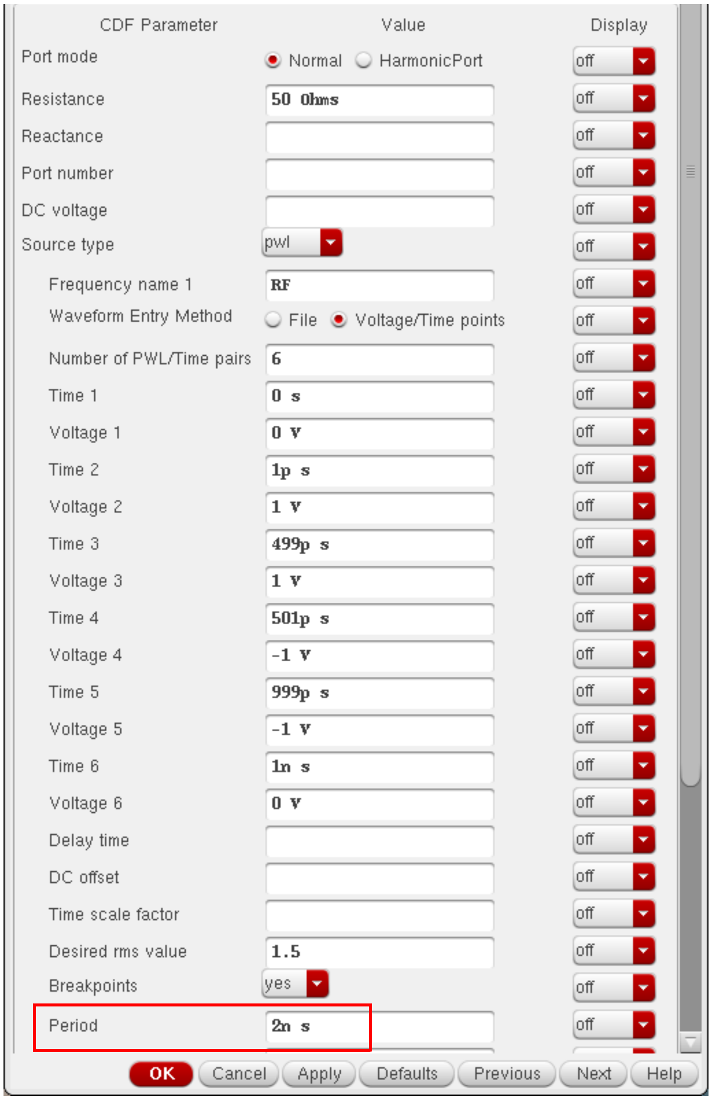


Now the transient result is plotted.

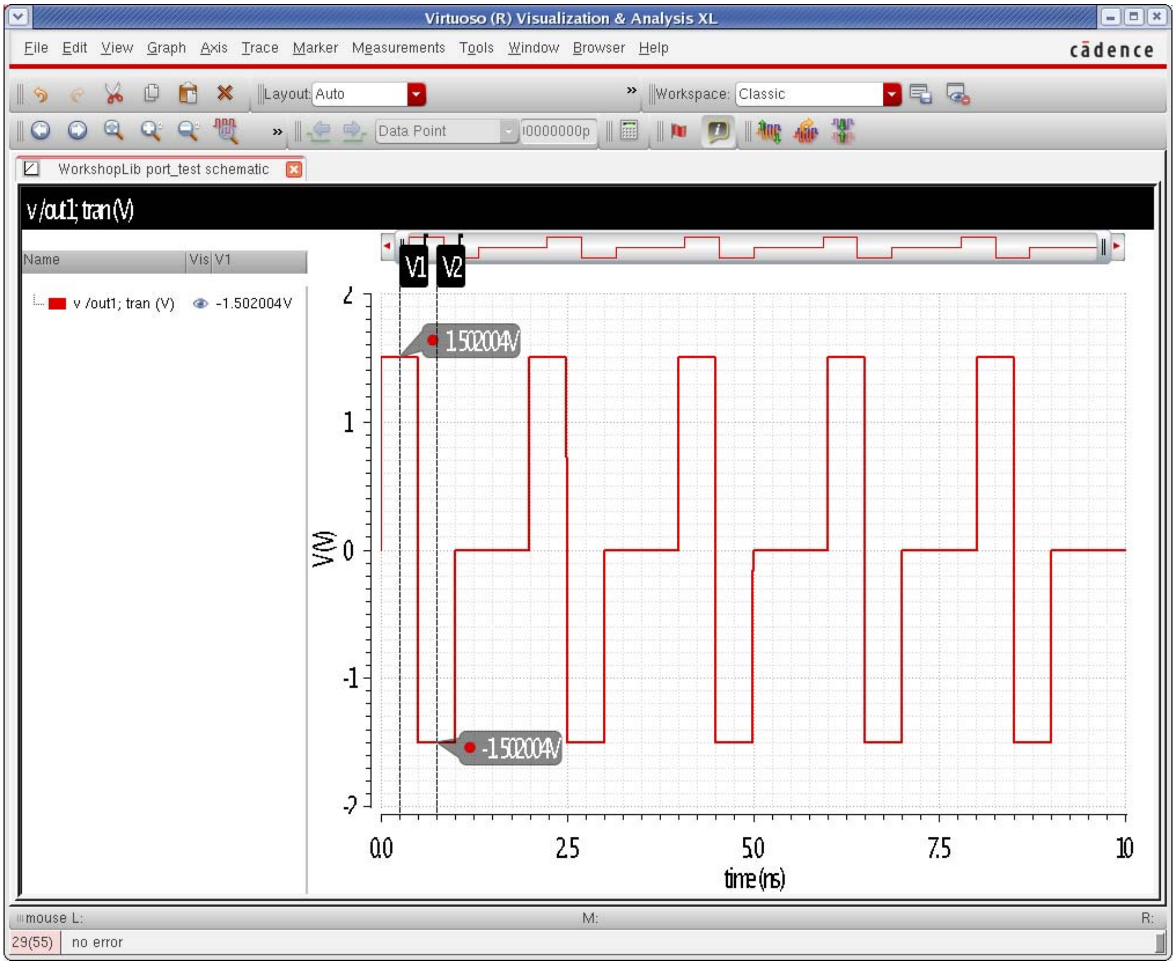


Vertical markers have been placed at the midpoint of the high and low states. The voltage is slightly greater than 1.5 because of the finite rise and fall times in the PWL description.

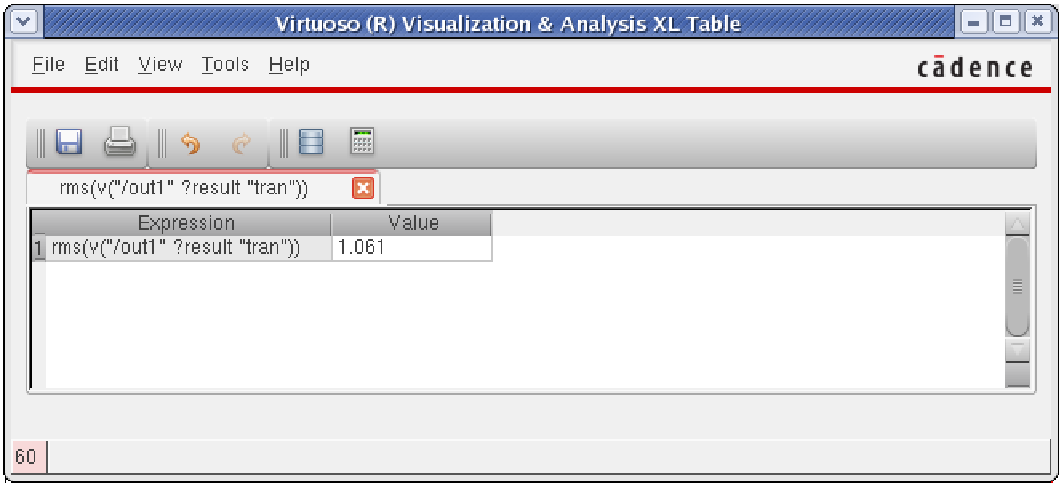
Now, the period is set to 2nsec. This will make the waveform periodic at 500MHz.



Now the transient result is plotted. Note that the high and low values are the same as before.

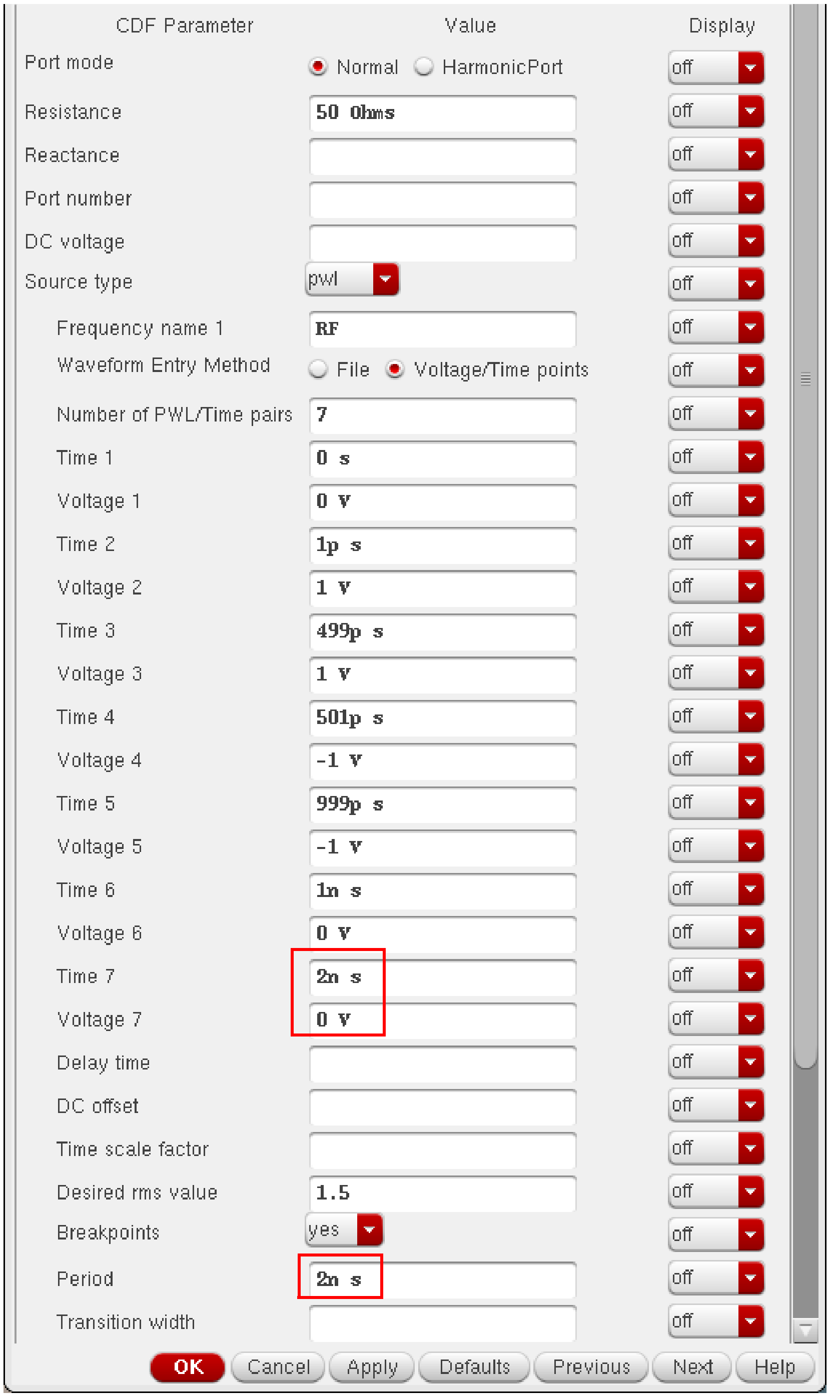


Now the transient waveform is sent to the calculator, and the actual rms value is calculated.

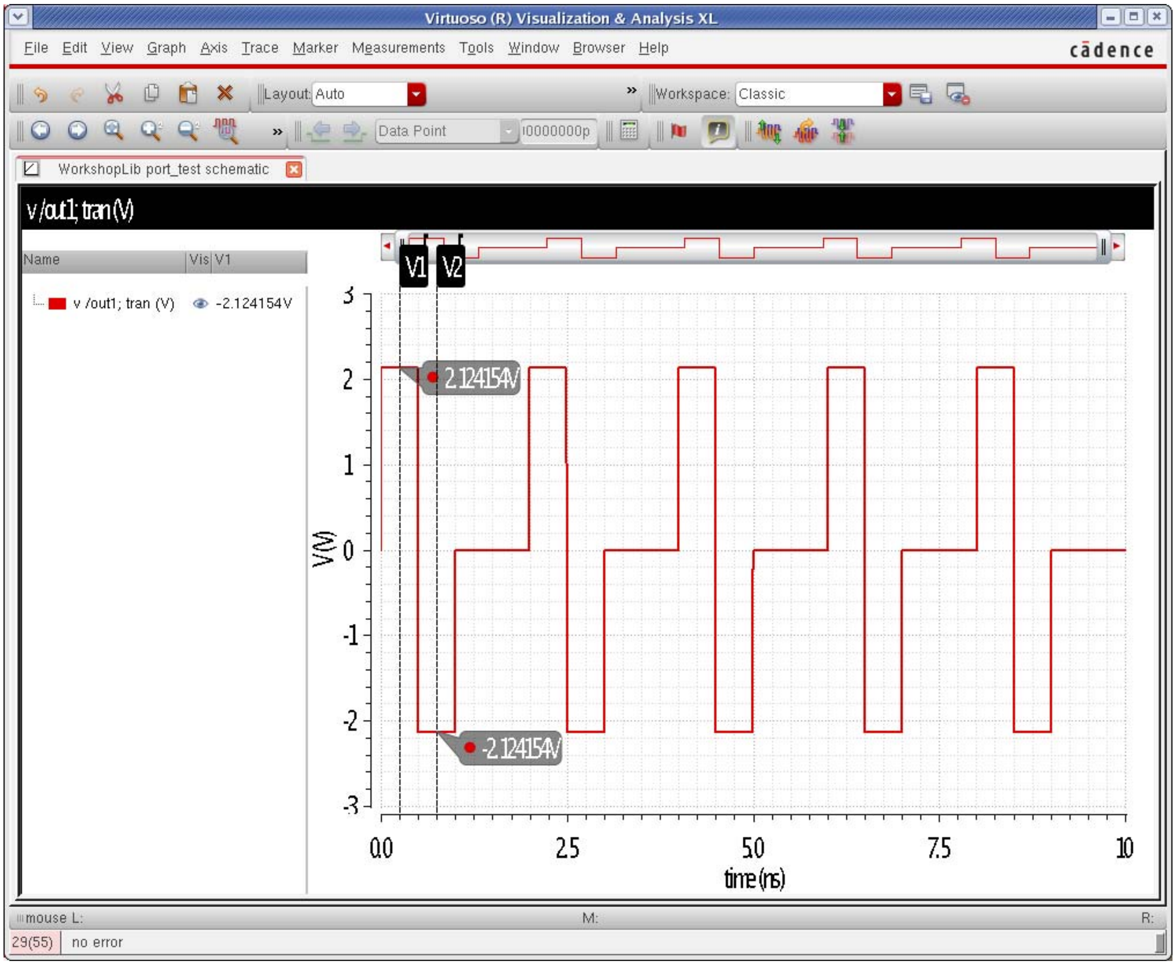


Instead of producing 1.5 volts as specified, only 1.061 volts is produced.

Now add another datapoint in the PWL description at 2nsec, and zero volts. Now the period and the time interval that defines the PWL points exactly agree.



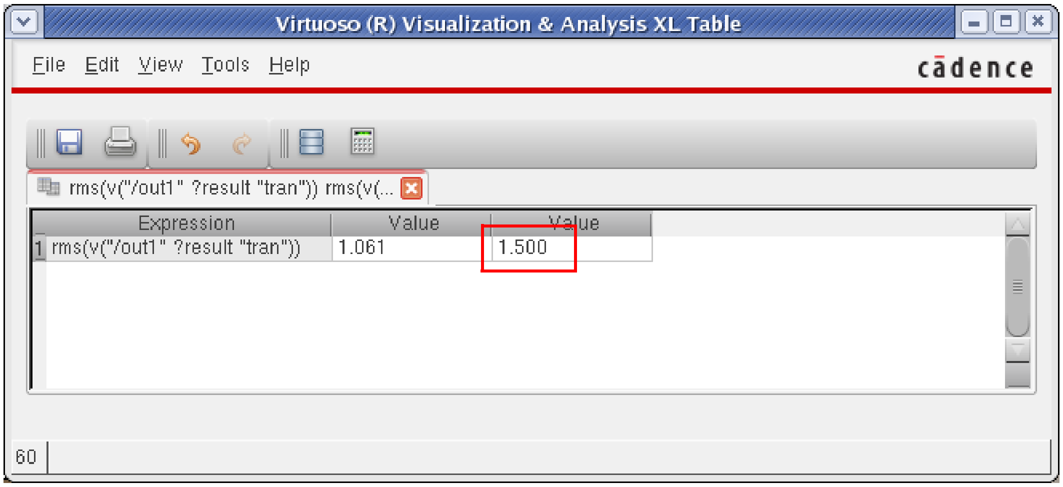
Now the transient result is plotted.



Note that the positive and negative voltages have changed. This is because the PWL description completely defines the waveform during the entire first period of the waveform.

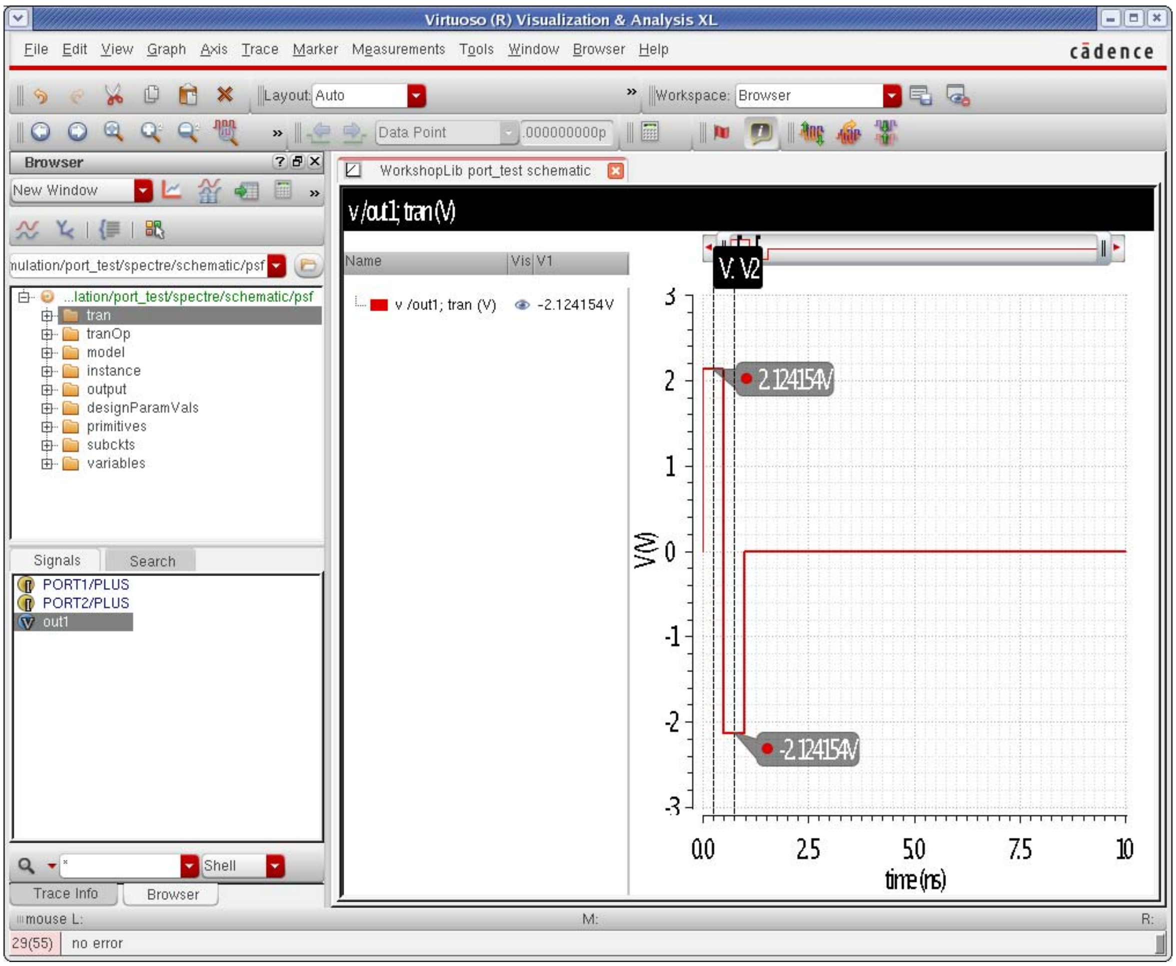
Now the signal is set to the calculator, and the rms is calculated. It is appended to the last table.

Now the RMS value is exactly 1.5 volts.



Now leave the last timepoint at 2nsec, and reduce the period to 1nsec.

Now, the transient result is plotted. The positive and negative voltages still reflect the rms value for the 2nsec interval defined by the timepoints themselves. It has not truncated the timepoints to just the first period. It is also not periodic. When the period is smaller than the time of the last timepoint in the PWL description, Spectre assumes that the period is in error, and ignores the property.



Tip

If you need to set an rms voltage for a periodic piecewise linear signal, make sure that the timepoints in the piecewise linear description define timepoints over exactly the entire first period of the waveform.

Do not set *Desired rms value* and *Power of PWL waveform* at the same time. If both are specified, first the amplitude is calculated based on the Power of PWL waveform property, and then that amplitude is multiplied by the *Desired rms value* setting.

Delay time

The *Delay time* parameter sets the time that the signal remains at the first point in the file before the pwl points begin to take effect. Generally, this should be set less than the period of the waveform. Longer times are allowed, however, this forces a transient analysis until the longest delay time before the periodic large-signal analyses pss, qpss, and hb can start. If a 1GHz signal and a delay time of 1 second is in your circuit, a transient is forced for 1 billion cycles of the input before the pss, qpss, or hb can start. This effectively prevents them from starting because of the long simulation time in the transient analysis.

Breakpoints

Possible values are *no*, *yes*, or blank. If you set *Breakpoints* to *yes*, you force SpectreRF to place time points at each point specified in a *PWL* waveform during a transient analysis. This can be very expensive for waveforms with many points. If you set *Breakpoints* to *no*, SpectreRF inspects the waveform, looking for abrupt changes, and forces time points only at those changes. If you set *Source type = pwl* and set *Breakpoints* to blank, the default is *yes* if the number of points you specify is less than 20.

Period

The *PWL* waveform is periodic if you specify *Period* in seconds.

If the value of the waveform you specify is not exactly the same at both its beginning and its end, then you must provide a non-zero value for *Transition Width*.

Power of PWL waveform

This is the power in dBm for a 50 ohm resistor for the output of the PWL source. See Desired rms value on page 1704 for important information about how to define timepoints in the PWL input points. Do not set *Desired rms value* and *Power of PWL waveform* at the same time. If both are specified, first the amplitude is calculated based on the *Power of PWL waveform* property, and then that amplitude is multiplied by the *Desired rms value* setting.

Transition Width

Transition width (*twidth*) is used when making *PWL* waveforms periodic and the ending value of the PWL file does not equal the beginning value. The default is the `PWL period/1000`.

Before repeating, the waveform changes linearly in an interval of *Transition Width* from its value at *(Period – Transition Width)* to its value at the beginning of the waveform. Thus, the *Transition Width* must always be much less than the *Period*.

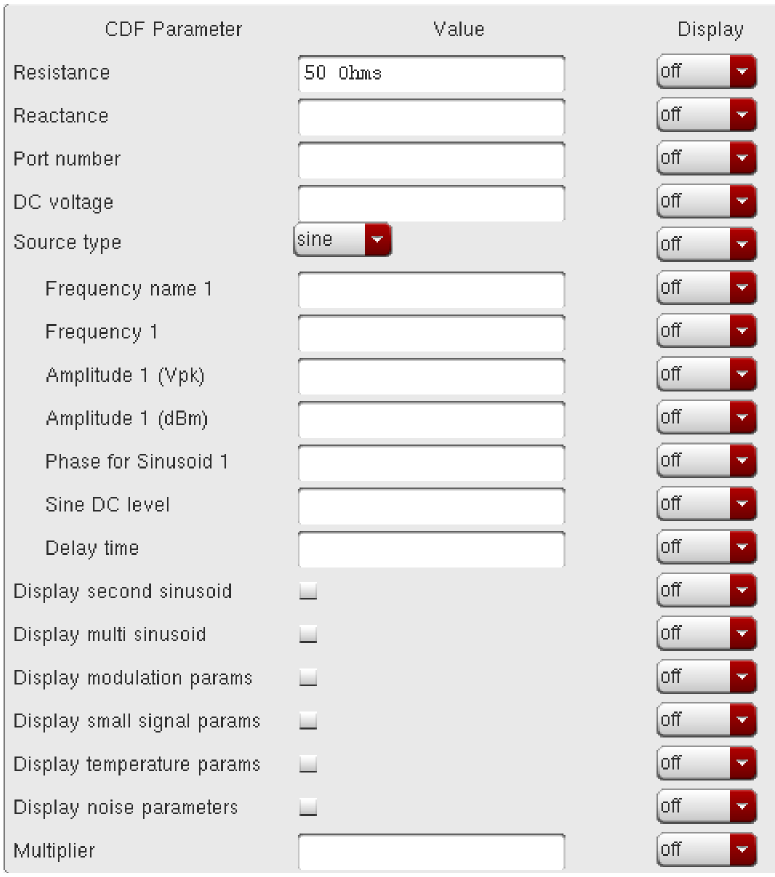
Sinusoidal Waveform Parameters

The *port* component can generate up to two sinusoids simultaneously. They are denoted as 1 and 2. You can set the amplitude, frequency, and phase for both individually. The amplitude can be set to either a voltage or a power level. When you set a power level, the assumption is that the port is perfectly matched. The source that is internal to the port gets double the amplitude specified by the power in dBm.You can also specify sinusoidal AM or FM modulation of sinusoid 1. Sinusoid 2 cannot be modulated.

To generate sinusoidal waveforms, set the *Source type* in the properties list to *sine*.

The first sinusoid is described by the parameters *Frequency name 1, Frequency 1, Amplitude 1 (Vpk)*, *Amplitude 1 (dBm)*, *Phase for Sinusoid 1*, *Sine DC level*, and by AM or FM modulation terms.

Figure 8-7 Source type=sine in the Edit Object Properties form



Frequency name 1

This property names the fundamental tones of sinusoid 1. After you save the schematic, the names you assign appear in the *Fundamental Tones* list box on the *Choosing Analyses* form.

Important

When you have multiple ports that are connected to the circuit at different places, but have the same frequency, or integer multiples of the same frequency, make sure that the *Frequency name 1* property is the same for every instance. An example might be RF1. For harmonic balance, when names is selected in the *Choosing Analyses* form, and for qpss, the frequency name property specifies that all the signals with the same name should be treated as one tone in the simulation. This can save a lot of time when you run those analyses.

Frequency 1 is the frequency of the first sinusoidal waveform (carrier frequency). You typically use unmodulated signals in SpectreRF analyses. The value must be a real number. The default frequency is zero, which causes no signal to be generated. Always set this property.

Amplitude 1 (Vpk)

The peak amplitude of the first sinusoidal waveform that you generate. The value specified is the voltage delivered into a matched load. You can select either *Amplitude 1 (Vpk)* or *Amplitude 1 (dBm)*, but not both. If *Amplitude 1 (Vpk)* has a value, the *Amplitude 1 (dBm)* field is grayed out. The value must be a real number. The default is 1 volt.

When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Thus, the voltage on the internal source is set to twice the value specified on the *port*.

Amplitude 1 (dBm)

Amplitude 1 (dBm) is the amplitude of the first sinusoidal waveform, in dBm.The resistance for the dBm calculation is taken from the resistance of the port. The value specified is the power delivered into a matched load. You can select either *Amplitude 1 (Vpk)* or *Amplitude 1 (dBm)*, but not both. If *Amplitude 1 (dBm)* has a value, the *Amplitude 1 (Vpk)* field is grayed out. The value must be a real number.

Phase for Sinusoid 1

The phase at the specified *Delay time*. To achieve a specified phase and still remain continuous, the sinusoidal waveform might start before the given *Delay time*. For example, if you want to generate a cosine wave, set this parameter to 90^{o}. The value must be a real number. The default is 0 degrees.

Important

For circuits that have long settling times for the nodes connected to the sinusoidal signal, it can save a lot of time and/or dramatically improve convergence if you set phases other than zero or 180 degrees by using the delay time parameter instead of the *Phase* parameter. Imagine a need to start at a 90 degree phase shift. If the *Phase for Sinusoid 1* property is set to 90, the waveform that is produced starts at the positive peak, If instead, the delay time was set to 3/4ths of the period and the *Phase for Sinusoid 1* property was set to zero, instead of starting at the peak and needing to settle to near the middle value in the tstab interval, the signal itself starts at the center value, so the circuit has to settle a much smaller amount.

Sine DC level

Sets the DC level in volts for sinusoidal waveforms in large-signal analyses. This parameter is used when the sinusoid has a different average level than the one specified for the DC analyses. If not specified, the average value of the sinusoid is the same as that of the DC level of the source.

Important

When this property is specified, it is used directly. The DC level of the sinusoid is not the DC voltage plus the Sine DC level. It is just the Sine DC level.

Modulation Parameters

Display Modulation Parameters

Important

The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

When selected, the form expands and the following modulation parameters are displayed: *FM modulation index, FM modulation freq, AM modulation index, AM modulation freq, AM modulation phase,* and *Damping factor*.

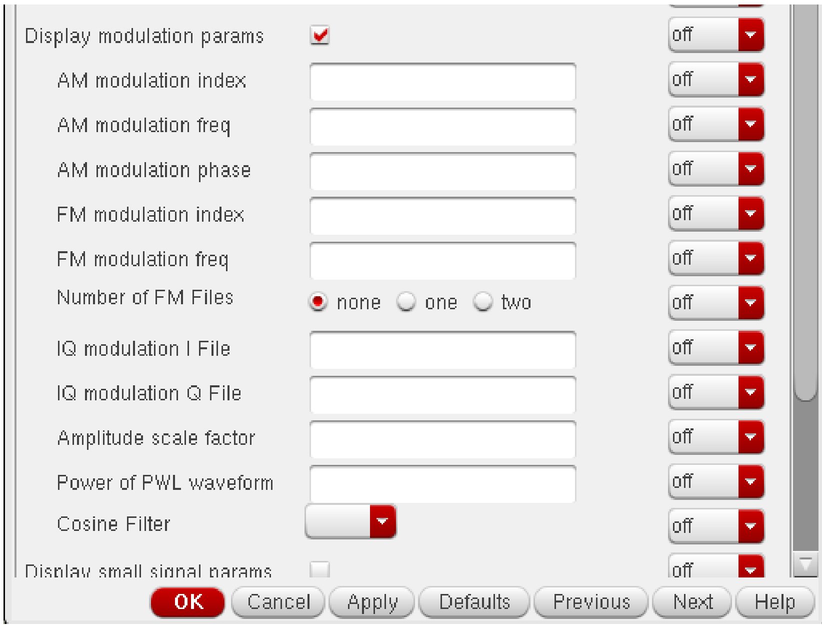
Important

These parameters only affect the sinusoidal waveform.

Important

Only the first `sinusoid` can be modulated.

Figure 8-8 Display modulation params



AM Modulation (Background Information)

The amplitude modulation for the sinusoidal case is defined as

$$V(t) = (A \bullet \sin(\omega_C t)) + \frac{AM}{2}[\sin((\omega_C + \omega_{Mod})t + \phi) + \sin((\omega_C - \omega_{Mod})t - \phi)]$$

where

- $V(t)$ is the amplitude modulation waveform
- ω_C is the carrier frequency
- AM is the AM modulation index
- ω_{Mod} is the modulation frequency
- ϕ is the *AM modulation phase*

The amplitude modulation parameters affect only the first sinusoid generated by *port*. They have no effect on the second sinusoid.

AM modulation Index

This property specifies the amplitude of the modulation signal. When the AM modulation index is 1.0, the signal is 100% modulated. When the AM modulation index is 0.5, the signal is 50% modulated.

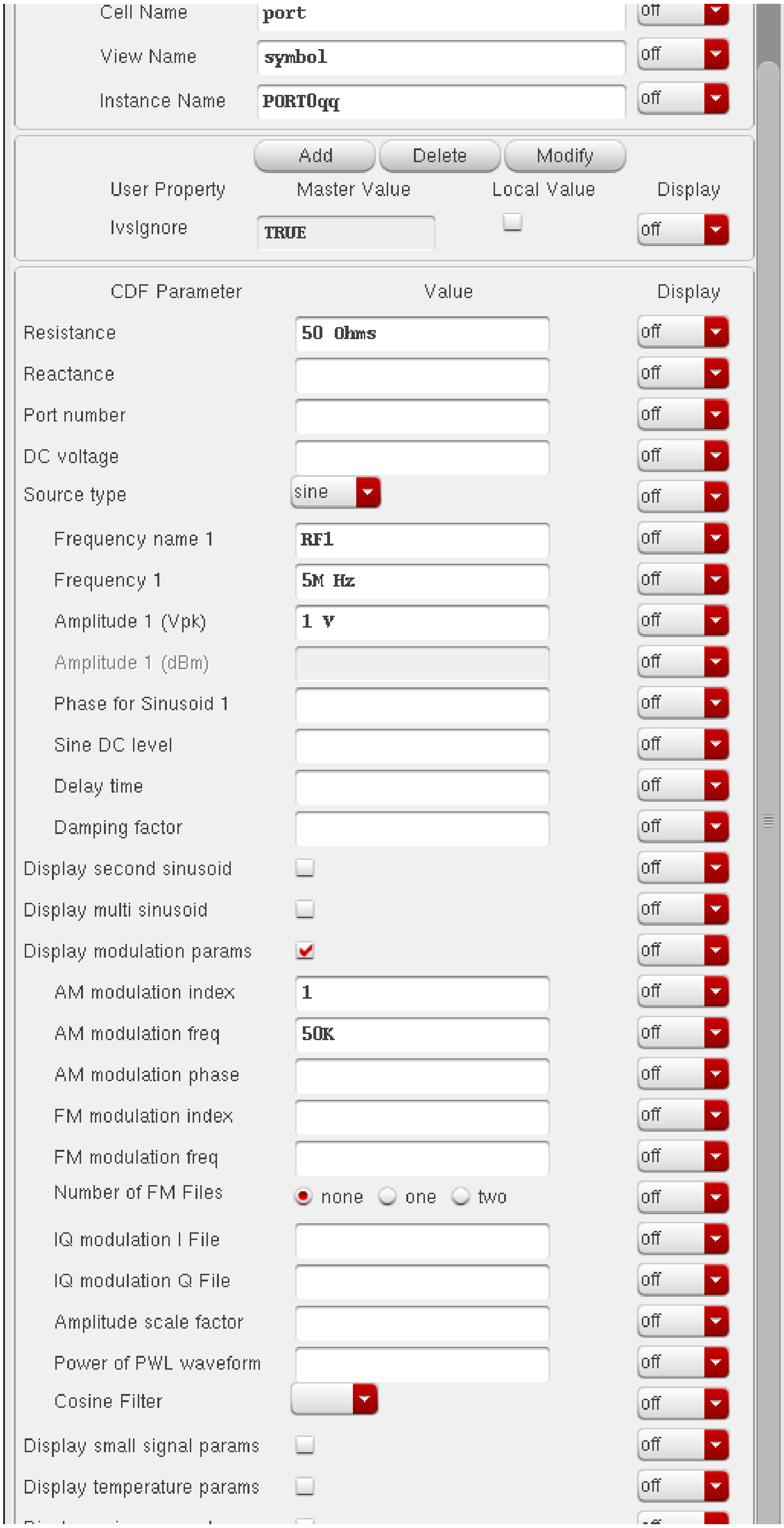
AM Modulation Frequency

This property specifies the frequency of the sine wave that is used to modulate the signal that is specified in the *Frequency 1* property. The AM modulation frequency should be much lower than the *Frequency1* property.

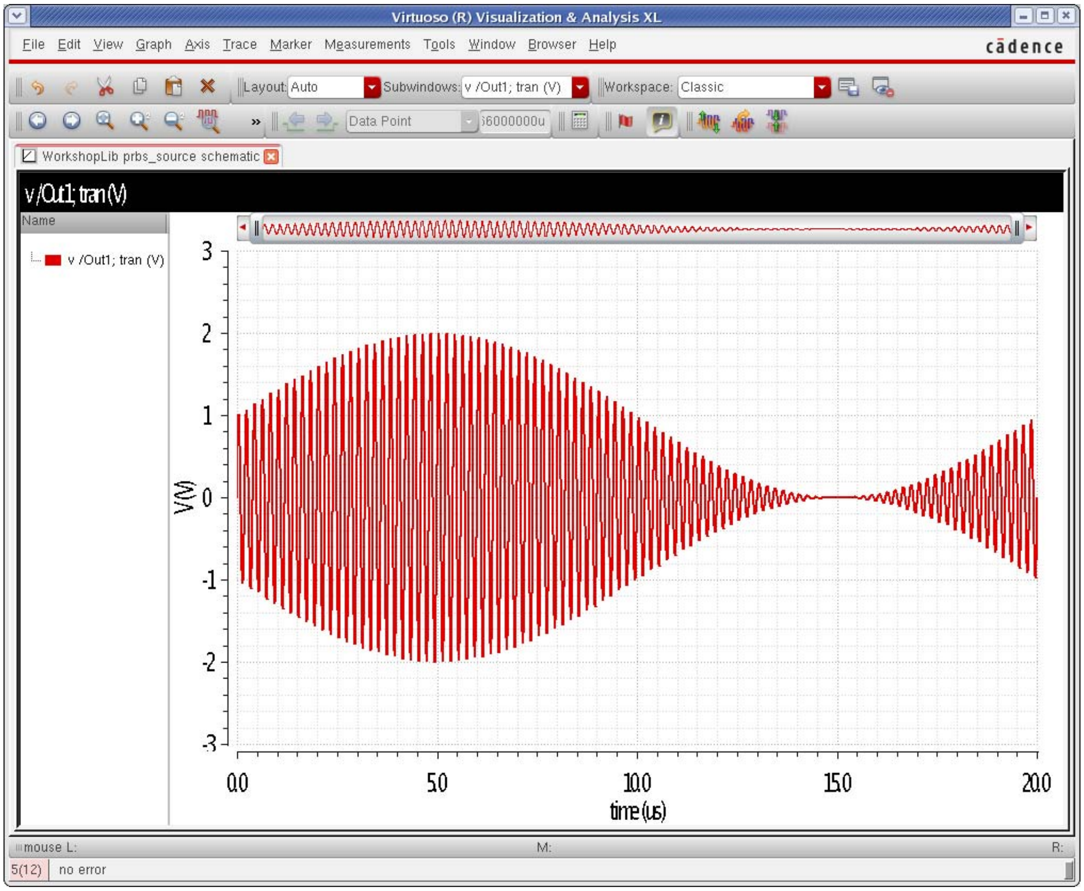
AM Modulation Phase

This specifies the initial phase of the AM modulation sinusoid. The default is zero degrees.

The *Edit Object Properties* form for a 100% modulated signal is shown below.


Cell Name
port
off
View Name
symbol
off
Instance Name
PORT0qq
off
Add
Delete
Modify
User Property
Master Value
Local Value
Display
lvsIgnore
TRUE
off
CDF Parameter
Value
Display
Resistance
50 Ohms
Reactance
Port number
DC voltage
Source type
sine
Frequency name 1
RF1
Frequency 1
5M Hz
Amplitude 1 (Vpk)
1 V
Amplitude 1 (dBm)
Phase for Sinusoid 1
Sine DC level
Delay time
Damping factor
Display second sinusoid
Display multi sinusoid
Display modulation params
AM modulation index
1
AM modulation freq
50K
AM modulation phase
FM modulation index
FM modulation freq
Number of FM Files
none
one
two
IQ modulation I File
IQ modulation Q File
Amplitude scale factor
Power of PWL waveform
Cosine Filter
Display small signal params
Display temperature params

The output waveform is shown below.



FM Modulation (Background Information)

The frequency modulation for the sinusoidal case is defined as:

$V_{FM}(t) = A \sin(2\pi f_c t + \beta \sin(2\pi f_m t) + \phi)$

where

- `A` is the amplitude of sinusoid 1
- β is the FM modulation index
- $\mathtt{sin(2\pi f_m t)}$ is the modulation signal
- $\mathtt{f_c}$ is the carrier frequency
- ϕ is the *phase for sinusoid 1*

The frequency modulation parameters affect only the first sinusoid generated by *port*. They have no effect on the second sinusoid.

FM modulation frequency

FM modulation frequency for the sinusoidal waveform (f_m in the previous equation).

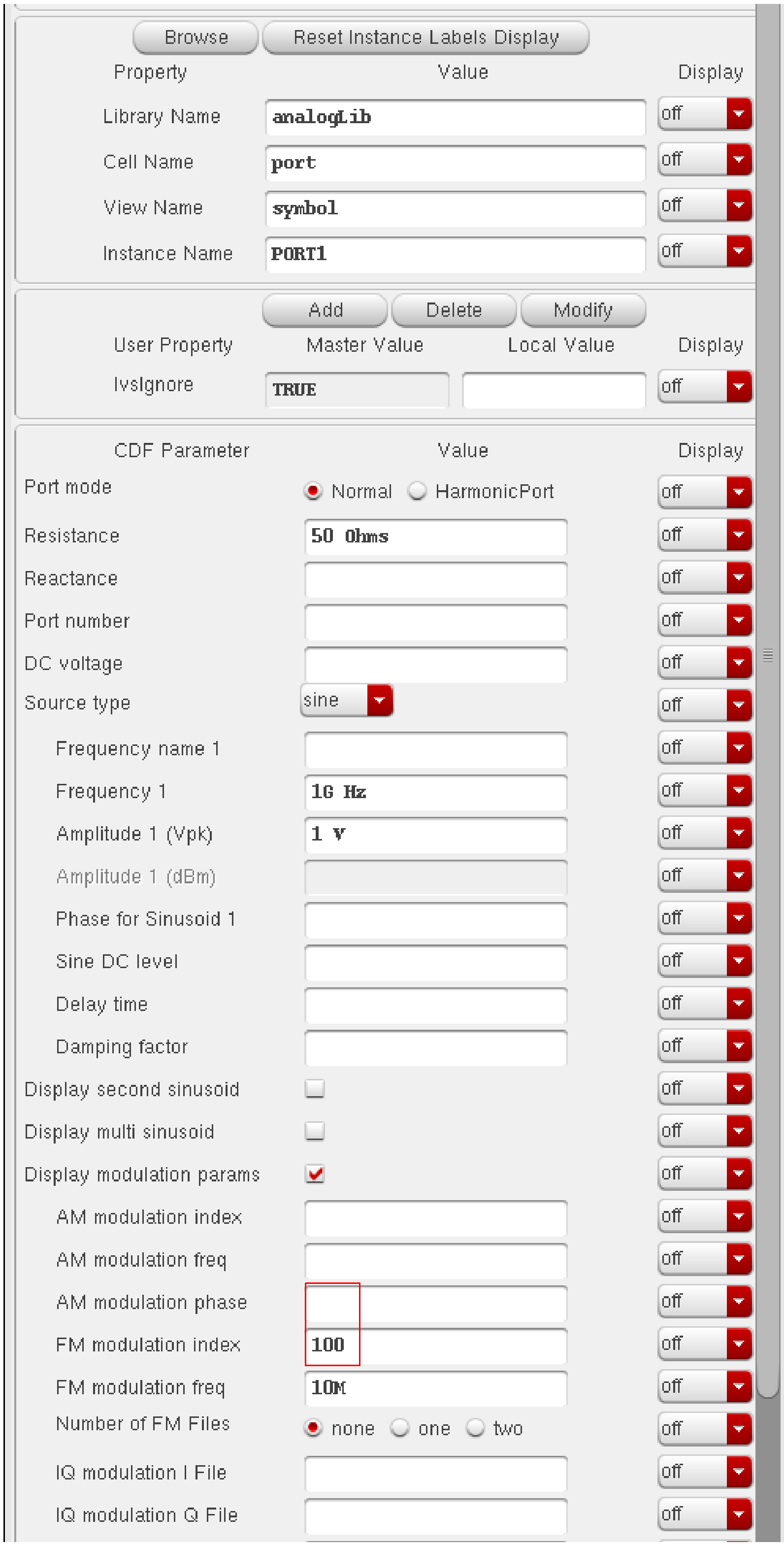
FM modulation index

FM index of modulation for the sinusoidal waveform. It is the ratio of the peak frequency deviation divided by the modulation frequency (β in the above equations).

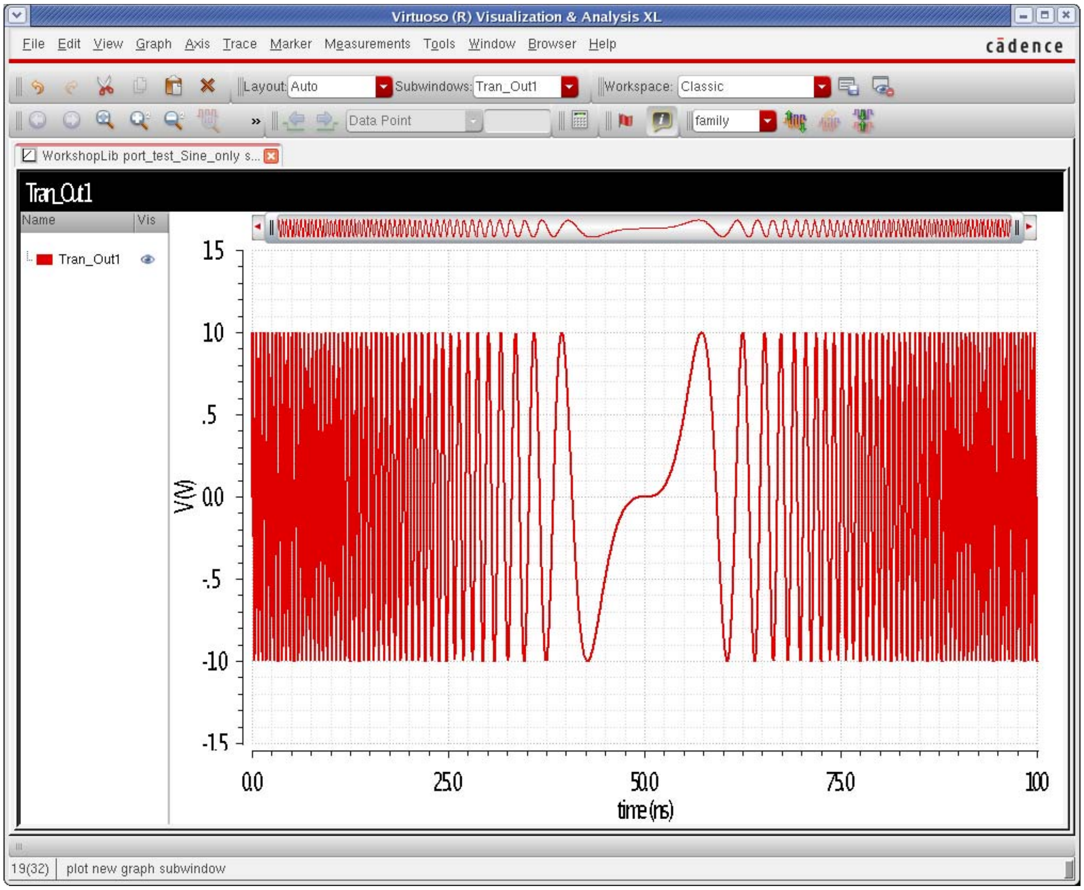
$\beta = \Delta f / f_m$

When the modulation index is 1.0, the deviation frequency from the carrier frequency is equal to the modulation frequency.

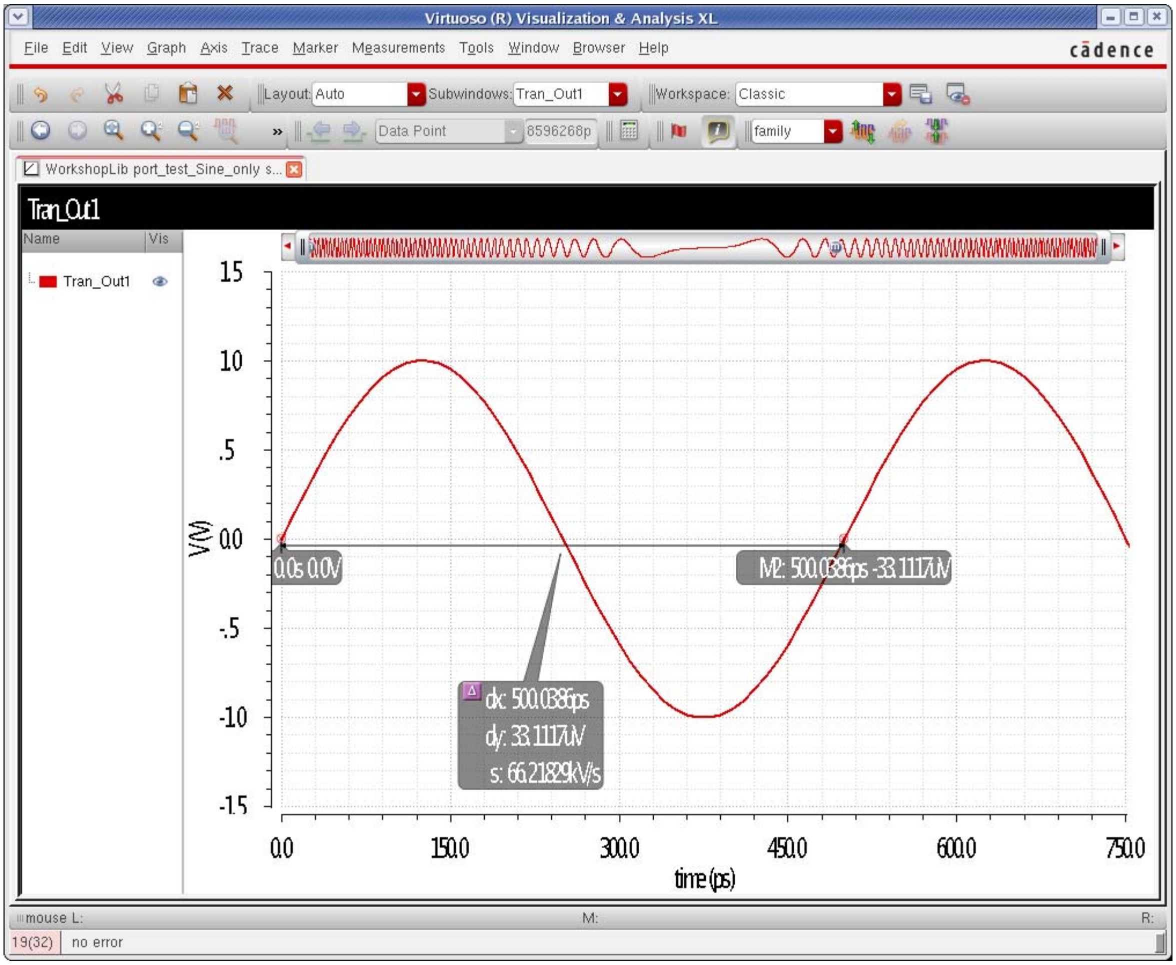
The property list for an FM signal with a modulation index of 100% is show below.


Browse
Reset Instance Labels Display
Property
Value
Display
Library Name
analogLib
Cell Name
port
View Name
symbol
Instance Name
PORT1
Add
Delete
Modify
User Property
Master Value
Local Value
lvsIgnore
TRUE
CDF Parameter
Port mode
Normal
HarmonicPort
Resistance
50 Ohms
Reactance
Port number
DC voltage
Source type
sine
Frequency name 1
Frequency 1
1G Hz
Amplitude 1 (Vpk)
1 V
Amplitude 1 (dBm)
Phase for Sinusoid 1
Sine DC level
Delay time
Damping factor
Display second sinusoid
Display multi sinusoid
Display modulation params
AM modulation index
AM modulation freq
AM modulation phase
FM modulation index
100
FM modulation freq
10M
Number of FM Files
none
one
two
IQ modulation I File
IQ modulation Q File
off

The transient waveform goes from zero frequency to twice the center frequency. This is an extremely large modulation index. Zero frequency is produced at the center of the below waveform.



When zoomed in near the beginning, the frequency is nearly double the carrier frequency.



Number of FM Files

The first sinusoid can also be phase modulated with a pair of PWL files that define the desired modulation.

- *none* specifies that the FM signal should be modulated with a sinusoid.

  In this case, the FM signal from the source will be represented as:

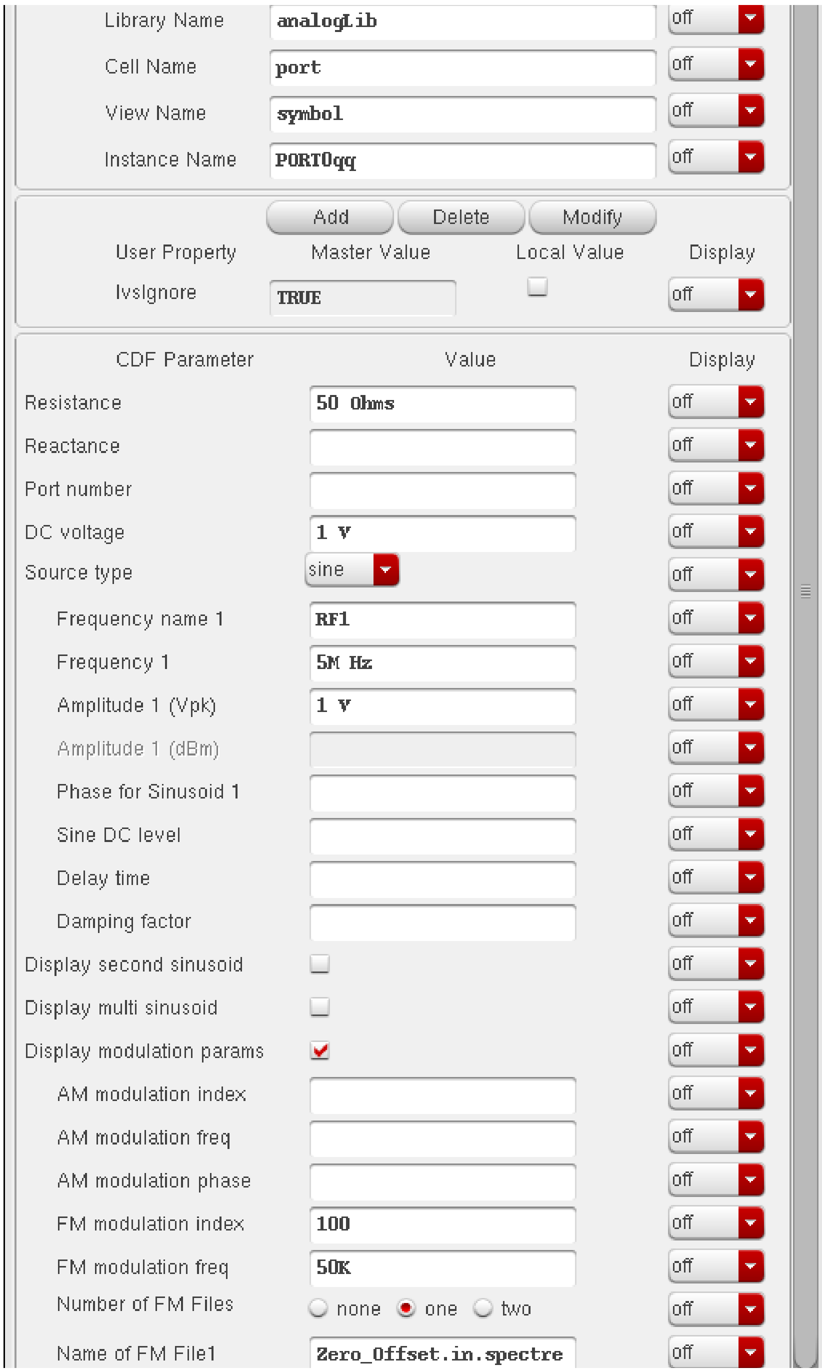
  $V_{FM}(t) = A \sin(2\pi f_c t + \beta \sin(2\pi f_m t))$

- When *one* is selected, a *Name of FM File1* field appears. Provide the filename of the PWL file that is to be used for the frequency modulation.

In this case, the FM signal from the source will be represented as:

$$V_{FM}(t) = A\sin(2\pi f_c t + \beta\int_0^t \delta\omega(\tau)d\tau)$$

This is shown in the *Edit Object Properties* form, as shown below.



- When *two* is selected, the *Name of FM File1 and Name of FM file2* fields appear. Provide the filename of the PWL file that is to be used for the phase modulation.

  In this case, the FM signal from the source will be represented as:

  $$V_{FM}(t) = A(t)\sin(2\pi f_c t + \beta\theta(t))$$

  Here,

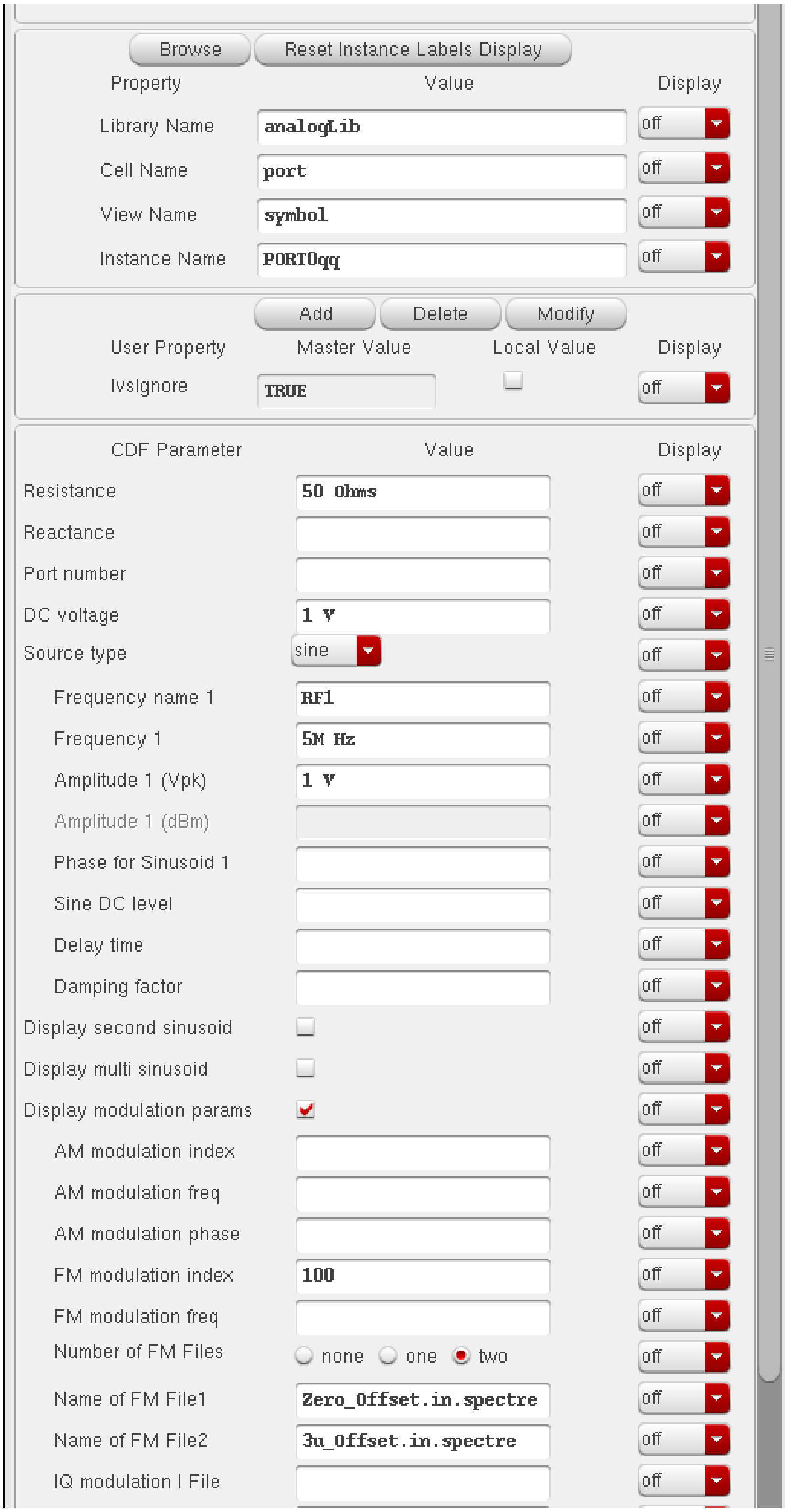
  β is the frequency modulation index

  $$A(t) = \sqrt{(I(t))^2 + (Q(t))^2}$$

  where `I(t)` is the waveform specified in the *Name of FM File1* pwl file, and `Q(t)` is

  $$\theta(t) = \arctan\left(\frac{Q(t)}{I(t)}\right)$$

waveform specified in the *Name of FM File2* pwl file. This is shown below.


Browse
Reset Instance Labels Display
Property
Value
Display
Library Name
analogLib
off
Cell Name
port
off
View Name
symbol
off
Instance Name
PORT0qq
off
Add
Delete
Modify
User Property
Master Value
Local Value
Display
lvsIgnore
TRUE
off
CDF Parameter
Value
Display
Resistance
50 Ohms
off
Reactance
off
Port number
off
DC voltage
1 V
off
Source type
sine
off
Frequency name 1
RF1
off
Frequency 1
5M Hz
off
Amplitude 1 (Vpk)
1 V
off
Amplitude 1 (dBm)
off
Phase for Sinusoid 1
off
Sine DC level
off
Delay time
off
Damping factor
off
Display second sinusoid
off
Display multi sinusoid
off
Display modulation params
off
AM modulation index
off
AM modulation freq
off
AM modulation phase
off
FM modulation index
100
off
FM modulation freq
off
Number of FM Files
none
one
two
off
Name of FM File1
Zero_Offset.in.spectre
off
Name of FM File2
3u_Offset.in.spectre
off
IQ modulation I File
off

Modulating a carrier with a digital modulation I and Q file

This capability provides exactly the same capability as the *ACPR_Source* in rfLib. The I and Q PWL files provide the information to be modulated, and the source acts as an ideal modulator.

IQ modulation I File

This parameter should be set to the filename to the PWL file that contains the digital modulation ‘I’ information. This is a file with two columns. The left column is the time, and the right column is the voltage. Usually, the file is generated with a constant delta-T from entry to entry in the file. Generally, this file is provided by the system designer.

IQ modulation Q File

This parameter should be set to the filename to the PWL file that contains the digital modulation ‘Q’ information.This is a file with two columns. The left column is the time, and the right column is the voltage. Usually, the file is generated with a constant delta-T from entry to entry in the file. Generally, this file is provided by the system designer.

Cosine Filter

The raised cosine filter is used when the PWL source is used to define digital modulation. Values are `none` (default), and `nrc`. `nrc` specifies a raised cosine filter to be used on the output waveform of the PWL source. This can be used to lower the numerical noise floor of the modulation specified in the I and Q modulating file. When `nrc` is selected, the PWL file must have evenly spaced points with timepoints at 1/(2*Bandwidth of the filter).

Rolloff factor

When `nrc` is selected, the *Rolloff factor* (*rolloff*) of the filter can be specified. The default value is `0.2`, which is reasonable for simulations. If you consider a continuous waveform, changing the rolloff factor has an effect on the ringing that is produced in the time domain. Smaller rolloff factors produce more ringing in the continuous waveform. Envelope does not simulate the continuous waveform. Instead, envelope samples the waveform at the time interval in the I and Q modulation file. This delta-Time in the I and Q file is always related to the modulation frequency, and it also determines the bandwidth of the cosine filter as described below. The effect of this is that the sampling occurs at the times where the ringing of the filter is zero, thus the cosine filter does not change the spectral content calculated in envelope. It just reduces the numerical noise floor of the input signal.

Bandwidth

The *Bandwidth* (*pwlbandwidth*) of the Cosine Filter. If you want to only allow in-band modulation, this is the channel bandwidth. If you want to allow an accurate ACPR measurement, set the bandwidth to three times the channel bandwidth. If you want to measure two adjacent channels for ACPR, set the bandwidth to five times the channel bandwidth. Note that the I and Q data must have properly spaced time intervals of $\frac{1}{2 \times Bandwidth}$ for this capability to be used.

Note: The *Rolloff factor* and *Bandwidth* options are available only when `nrc` is selected in the *Cosine Filter* drop-down list box.

The *Rolloff factor* and *Bandwidth* options are available only when `nrc` is selected in the Cosine Filter.

Damping factor

This property is available in the `vsin` and `isin` components only.

Damping factor for the sinusoidal waveform. 1/*Damping factor* specifies the time it takes to change the amplitude by a factor of e. For example, consider the following damped sinusoid:

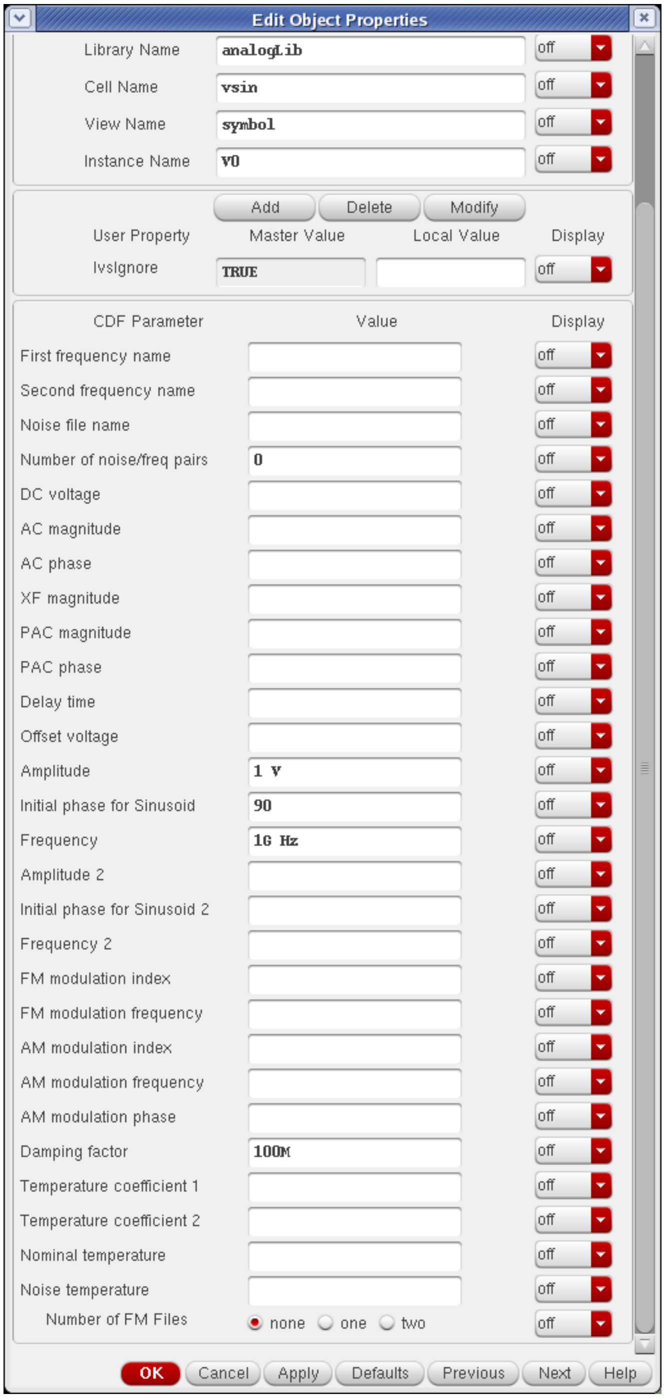
$v(t) = A\ e^{-\sigma t} \sin(2\pi ft + \phi)$

where

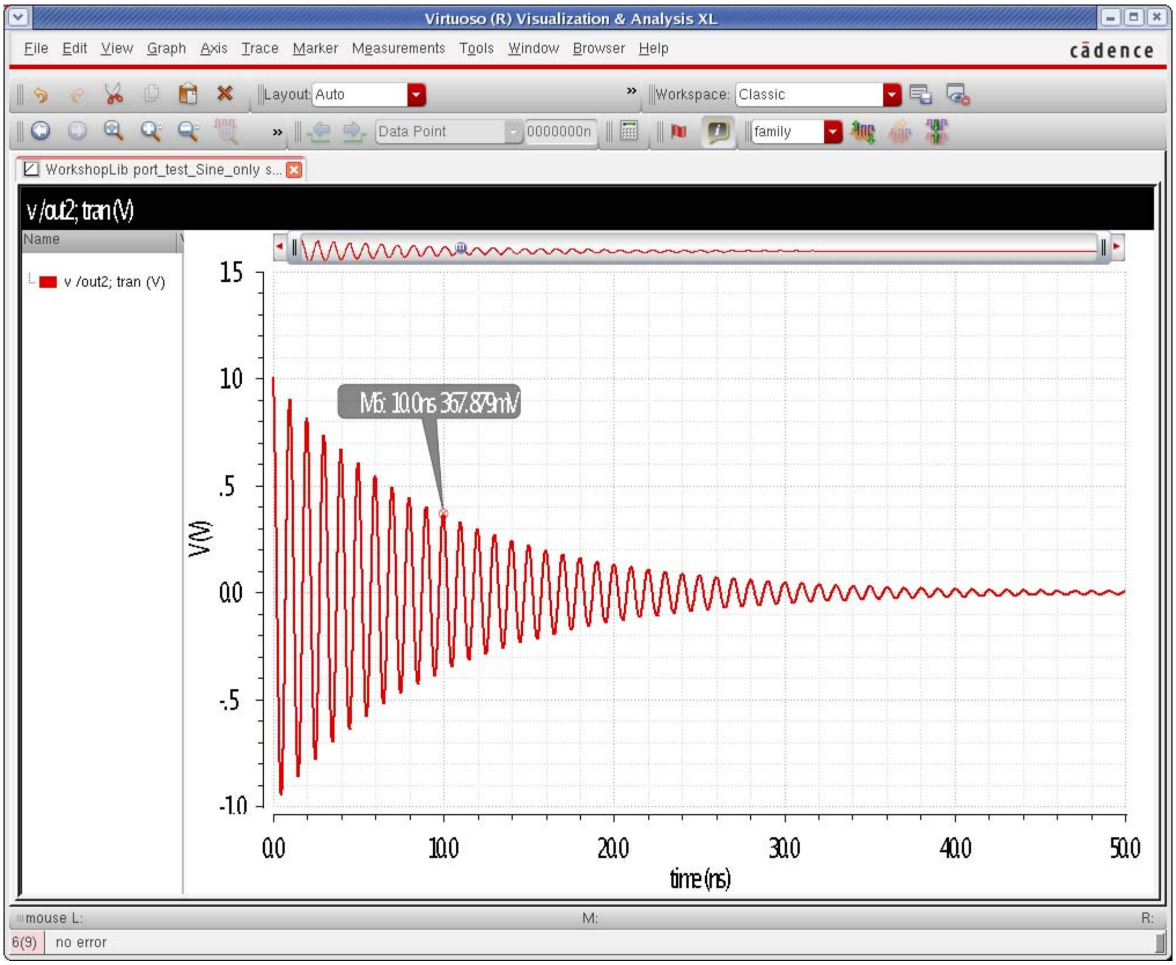
σ = *Damping factor*, `A` is the amplitude of sinusoid 1, and ϕ is the *Phase for sinusoid 1*

- If $\sigma = 0$, the waveform is a pure sinusoid (steady state).
- If $\sigma < 0$, the waveform exhibits growing amplitude.
- If $\sigma > 0$, the waveform exhibits decaying amplitude.
- The value must be a real number. The default is 0 and the unit is 1/seconds.

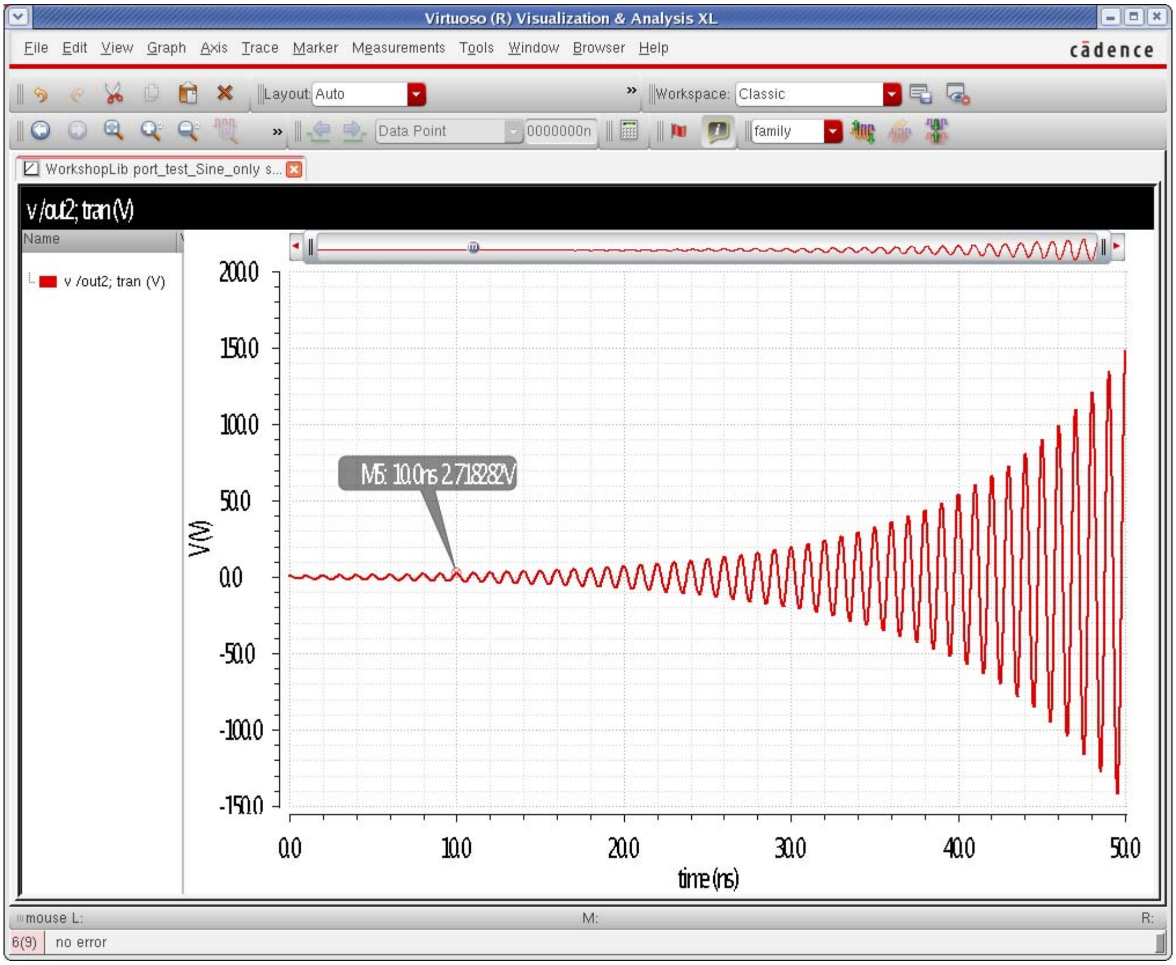
For the vsin component, *am* example property list is shown below.



The amplitude is 1 volt. The initial phase of 90 degrees starts the sine wave at a peak so that the amplitude at 1/100M seconds or 10nsec will be at a peak that can be measured with a marker in the waveform tool. The transient analysis has strobeperiod set so a datapoint at exactly 10nSec is produced in the output. With a positive damping factor, the waveform decays, and the amplitude is 1/e at 10nsec.



When the damping factor is negative, the amplitude is e (2.7182...) at 10 nsec.



Display second sinusoid

Important

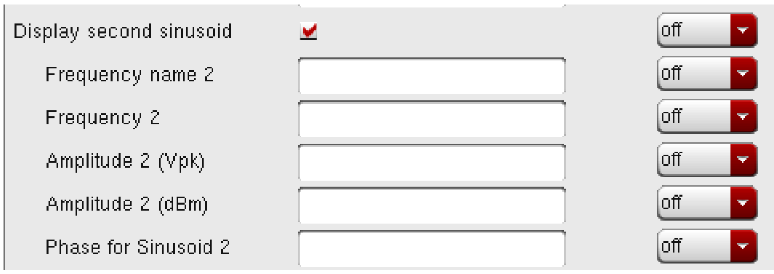
The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

Displays the CDF parameters for the second sinusoid in the *Edit Object Properties* and *Add Instance* forms. When selected, the form expands to show the following CDF parameters: *Frequency name 2*, *Frequency 2*, *Amplitude 2 (Vpk)*, *Amplitude 2 (dBm)*, and *Phase for Sinusoid 2*.

Important

The second sinusoid cannot be modulated.

Figure 8-9 Display second sinusoid



Frequency name 2

Name for the second sinusoid. After you save the schematic, the name you assign is listed in the *Fundamental Tones* section of the *Choosing Analyses* form for pss, qpss, and hb when names is selected.

Important

If you have two different sources at the same frequency or at integer multiple of the same frequency, make sure that you use the same frequency name 1 or 2 property for both sources. An example might be RF1.

Frequency 2 is the frequency of the second sinusoidal waveform in Hertz. There is no default value.

Amplitude 2 (Vpk)

Peak amplitude of the second sinusoidal waveform. The value specified is the voltage delivered into a matched load. You can select either *Amplitude 2 (Vpk)* or *Amplitude 2 (dBm)*, but not both. If *Amplitude 2 (Vpk)* has a value, the *Amplitude 2 (dBm)* field is grayed out. The value must be a real number. The default is 1 volt.

When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Thus, the voltage on the internal source is set to twice the value specified on the *port*.

Amplitude 2 (dBm)

Amplitude 2 (dBm) is the amplitude of the second sinusoidal waveform, in dBm. The resistance for the dBm calculation is taken from the resistance parameter of the port. the value specified is the power delivered into a matched load. You can select either *Amplitude 2 (Vpk)* or *Amplitude 2 (dBm),* but not both. If *Amplitude 2 (dBm)* has a value, the *Amplitude 2 (Vpk)* field is grayed out. The value must be a real number. Units: dBm

Phase for Sinusoid 2

The phase at the specified *Delay time* for the second sinusoid. To achieve specified phase while still remaining continuous, the sinusoidal waveform might start before the given *Delay time*. The value must be a real number. Default: 0 Units: degrees.

Display multi sinusoid

Important

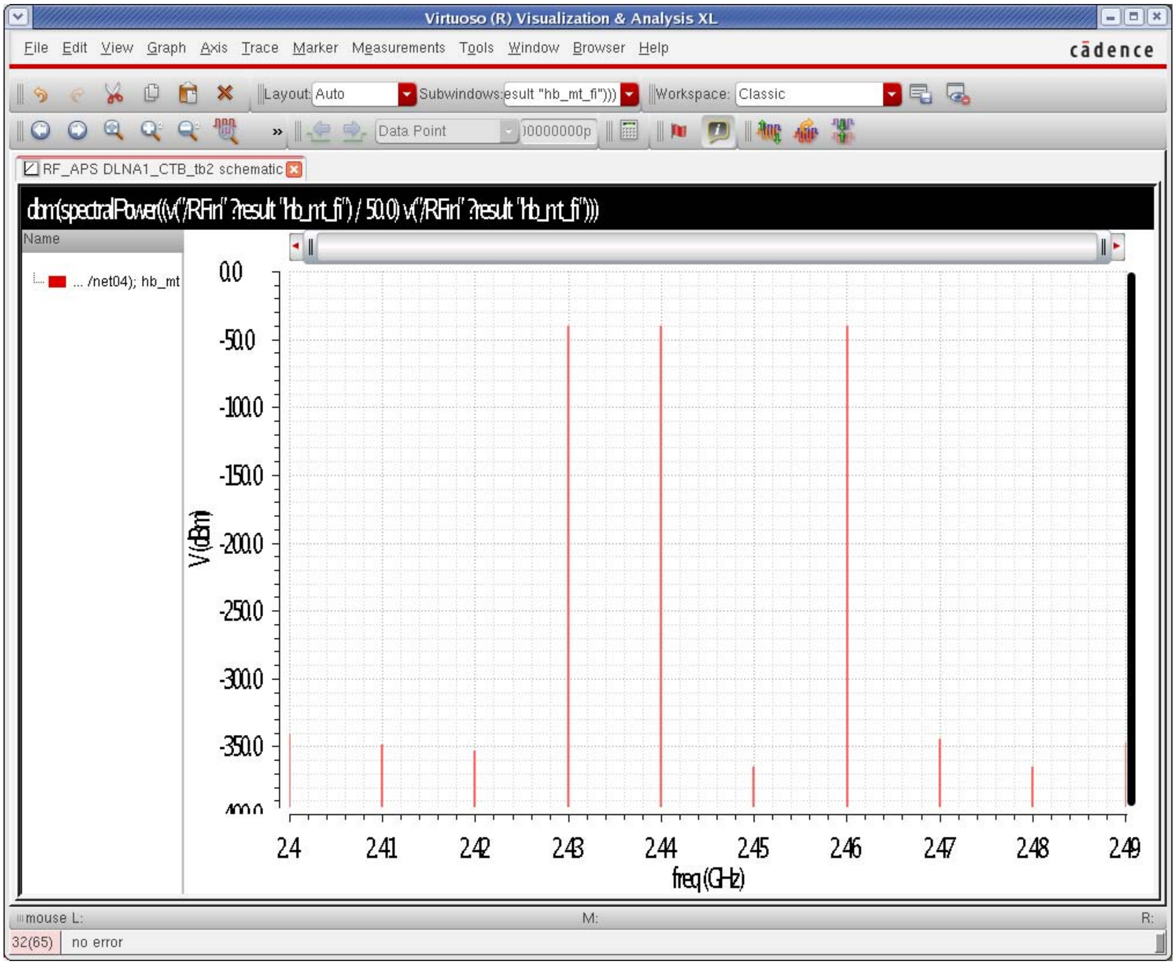
> The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

Multi sinusoid is a way of generating up to nine unmodulated signals at the same time from the port component. It works in all the large-signal analyses, and is usually used with harmonic balance or qpss-harmonic balance for the simulation of systems with multiple carriers.

When *Display multi sinusoid* is selected, the properties form expands to show more properties. The *Number of Frequencies* field specifies the number of carriers you want to generate. Up to nine carriers can be generated by the port component. For each one of the frequencies, a set of parameters patterned after: *Sinusoid Frequency 1, Sinusoid Ampl 1 (Vpk), Sinusoid Ampl 1 (dBm), Sinusoid Phase 1, Sinusoid Maxharm 1* is displayed for each tone you want to generate.

When *Display multi sinusoid* is selected, the frequency specified in the *Frequency1* parameter should be the middle frequency of the group of frequencies you want to generate. It must be one of the frequencies that could actually be generated by the port component. If four frequencies are to be generated, then the *Frequency1* property could be either the second or third frequency in the series.

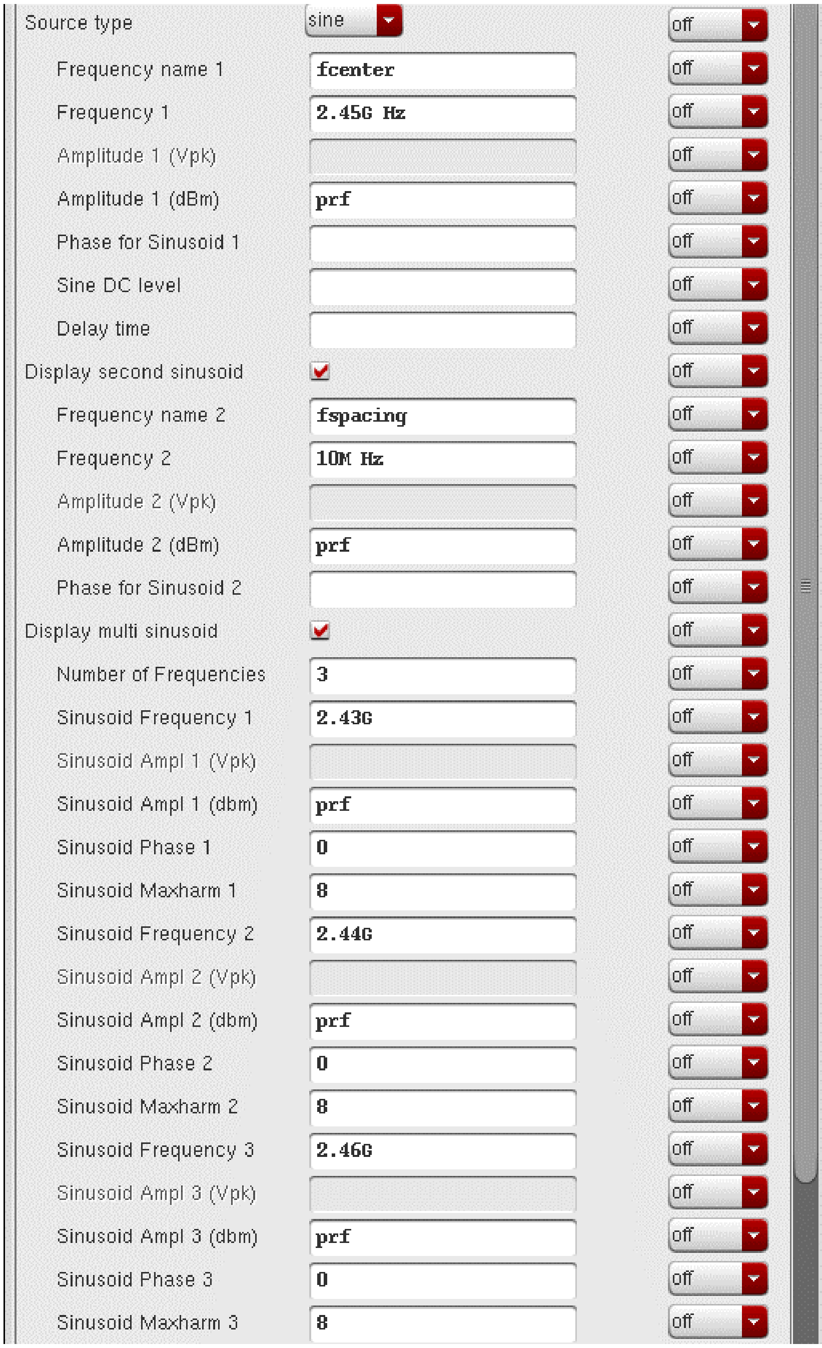
Consider an example where 2.43GHz, 2.44GHz, and 2.46GHz need to be generated for a composite triple beat measurement. The actual center frequency is 2.445GHz. Since this is not one of the frequencies in the series that is actually generated, it cannot be picked. Pick one of the frequencies that could be generated on either side of the actual center frequency. Here, the center frequency could be either 2.44GHz or 2.45GHz. Even though 2.45GHz is not being generated in this example, it is one of the frequencies that could be generated by the series, and thus it is eligible to be selected.



The frequency specified in the *Frequency 2* field should be the spacing frequency. All the frequencies specified in the multi-frequency section need to be separated by the spacing frequency. In the example above, the spacing frequency is 10MHz.

Up to nine frequencies can be generated in one port. When this feature is used with HB or QPSS, only specify the *Frequency1* and *Frequency2* frequencies in the frequencies section. The number of harmonics should be set to about two times the total number of tones that are generated in the multi-frequency section.

Figure 8-10 Display multi sinusoid



Number of Frequencies

Number of sinusoid frequencies to be specified.

Sinusoid Frequency 1

The frequency of the first sinusoidal waveform (carrier frequency) in hertz.

Sinusoid Ampl 1 (Vpk)

The peak amplitude of the sinusoidal waveform that you generate in volts. When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Thus, the voltage on the internal source is set to twice the value specified on the *port*.

Sinusoid Ampl 1 (dBm)

The amplitude of the first sinusoidal waveform when specified in dBm.

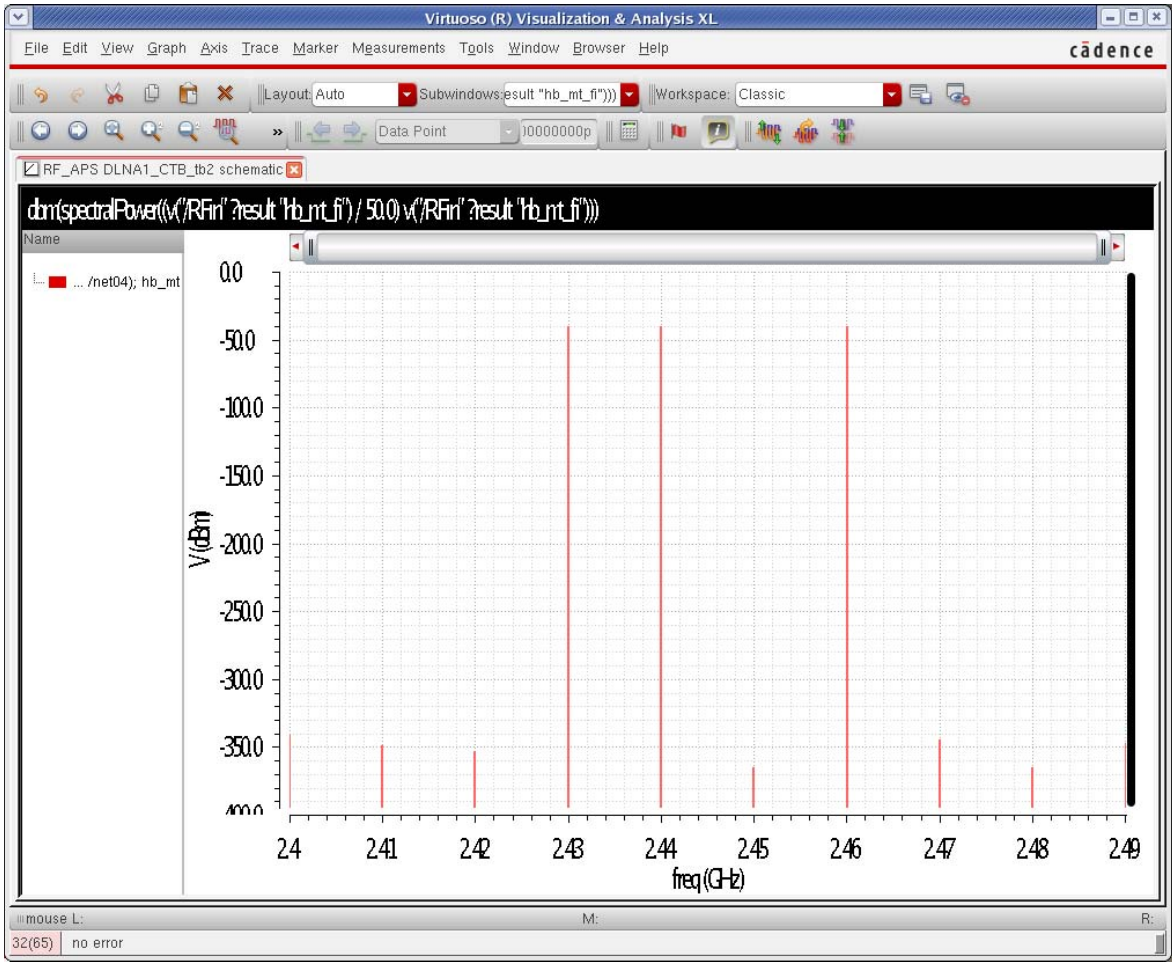
Sinusoid Phase 1

The phase at the specified delay time. To achieve a specified phase and still remain continuous, the sinusoidal waveform might start before the given delay time. For example, to generate a cosine wave, set this parameter to pi/2. The units are radians.

Sinusoid Maxharm 1

Do not set this parameter. Harmonics are usually set in the hb or qpss *Choosing Analyses* form.

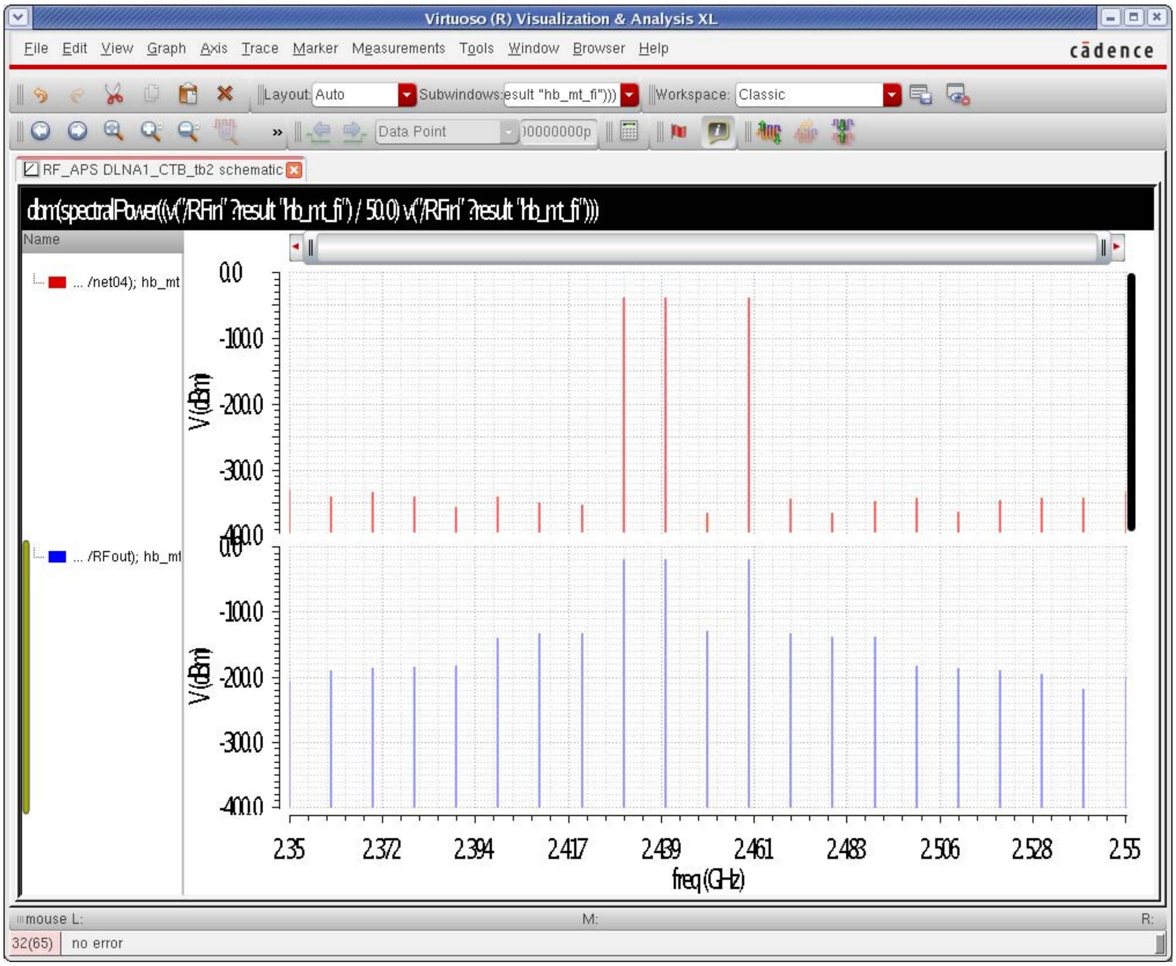
Below is a plot of the output when the variable `prf` is set to -40 (dBm).



The amplitudes of the tones at 2.4G, 2.41G, 2.42G and so on are at the numerical noise floor of the simulation.

When these three frequencies are applied to a real system, the nonlinearity of the system produces mixing products. For example, 2.43G mixes with 2.44G to produce 10M. 2.44G mixes with 2.46G to produce 20M. 2.43G mixes with 2.46G to produce 30M. 2.43G, 2.44G, and 2.46G are first order terms, so 10M, 20M, and 30M are second order terms.

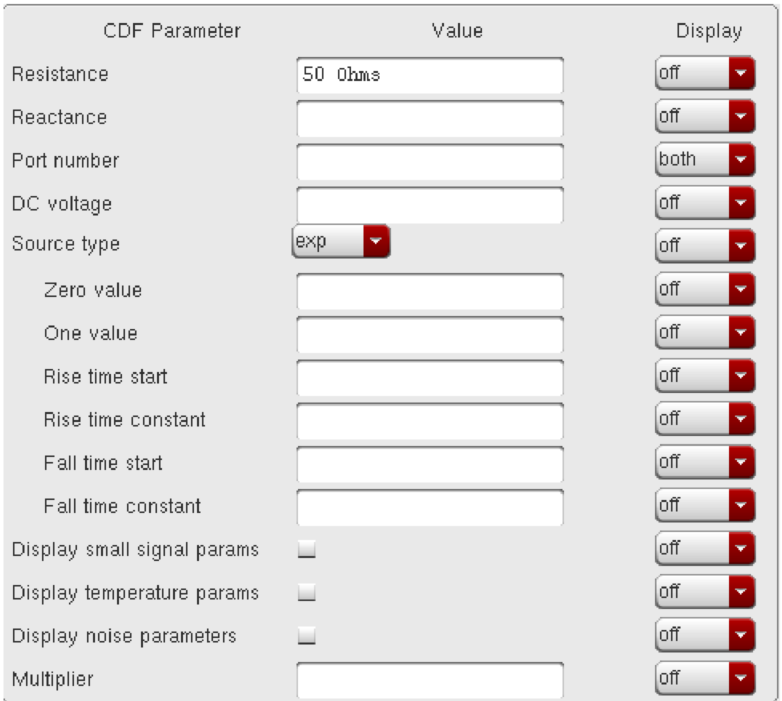
Now the 10M, 20M, and 30M terms can mix with 2.43G, 2.44G, and 2.46G. For example, 2.44G mixes with 10M to produce 2.45G. 2.44G is a first order term, and 10M is a second order term, so 2.45G is a third order term. 2.43G can mix with 20M to add an additional third order term at 2.45G. This leads to many harmonics that are actually produced, as shown below. The vertical axes have the same scale.



Exponential Waveform Parameters

To generate an exponential waveform from the `port` component, set the CDF parameter *Source type*=*exp*, as shown in the next figure.

Figure 8-11 Source type=exp in the Edit Object Properties form



The exponential waveform can generate one exponential pulse, and cannot generate a periodic signal. This it is not usable with SpectreRF.

When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Thus, the voltage on the internal source is set to twice the value specified on the *port*.

Zero value

The *Zero value* property is the time-zero value of the exponential waveform in volts.

One value

The *One value* property specifies the voltage that the signal transitions to at time equals zero plus. The default is one volt, which is unlikely to be what you want. Always set this property.

Rise time start

This property specifies the simulation time in seconds to begin the transition from the zero to the one value. The default is zero seconds.

Rise time constant

The *Rise time constant* specifies the time for the exponential waveform to reach 63 percent of the way from the zero value to the one value. The default is 0.075 percent of the stop time in the transient analysis.

Fall time start

The *Fall time start* specifies the simulation time to begin the transition from the one state to the zero state. The default is one percent of the transient analysis stop time.

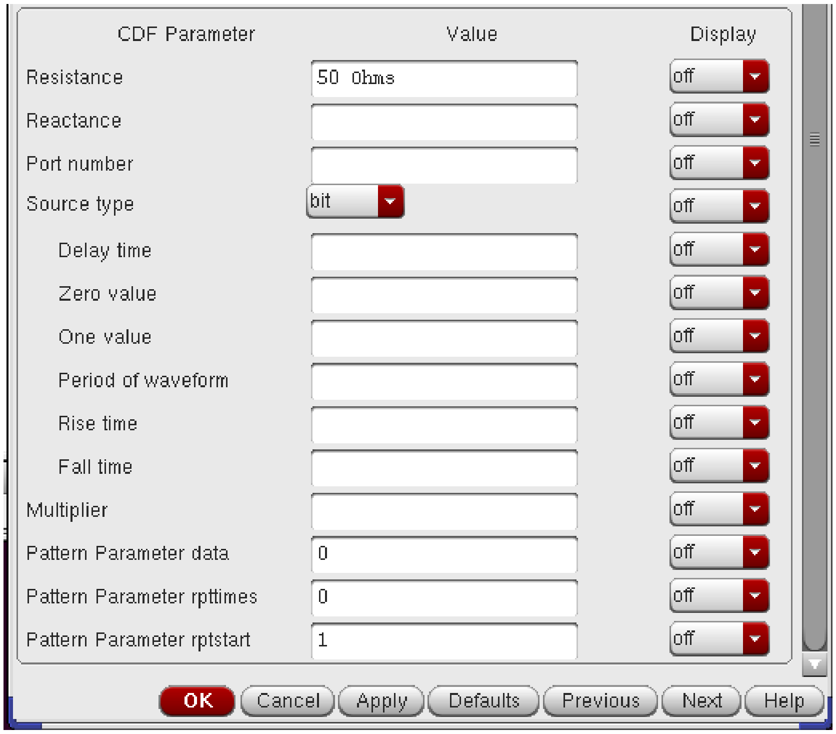
Fall time constant

The *Fall time constant* specifies the simulation time for the exponential waveform to reach 63 percent of the way from the one value to the zero value. The default is one percent of the stop time in the transient analysis.

Bit Waveform Parameters

To generate bit sequence/string from the port component, select bit in the *Source Type* cyclic field. The bit source has four states: `1`, `0`, `m`, and `z`, which represent the high, low, middle voltage/current and high impedance state respectively. It allows patterns defining a sequence of bits. When the *m* state is specified, the output voltage is set halfway between the zero and one voltage.

Figure 8-12 Source type=bit in the Edit Object Properties form



Delay Time

The *Delay time* property specifies the time the first bit in the string should be held before the first bit is generated.

Note: When the delay time property is set, the time of the first transition is at the delay time plus the period of the first bit, which is set by the *Period of waveform* property.

Zero Value

The *Zero value* property sets the voltage when the pattern specifies a zero state. The pattern will be defined shortly.

Important

For the zero and one value, the port is assumed to be terminated with a matching register. The internal voltage source is doubled.

If you intend to use the z state of the bit source, do not add a matching resistor across the port terminals, since this resistor will totally negate the high impedance state. Make sure that you halve the voltage values you want for the zero and one states when you use the port element to generate a bit pattern.

One Value

The *One value* property sets the voltage of the port when the pattern specifies a one.

Period of Waveform

The bit waveform type generates a bit stream where the period defines the period of one of the bits in the stream. For example, if the period property is set to 1n, then every nanosecond the bit switches to the next bit in the sequence.

Rise Time

The *Rise time* property specifies the transition time from the zero state to the one state.

Fall Time

The *Fall time* property specifies the transition time from the one state to the zero state.

Pattern Parameter data

This is the bit string, which is a series of the four states; `1`, `0`, `m`, and `z`. An example is 101100 (not separated by spaces) which would generate a 1, then a zero, then 2 ones in a row, and then two zeros in a row.

rptstart

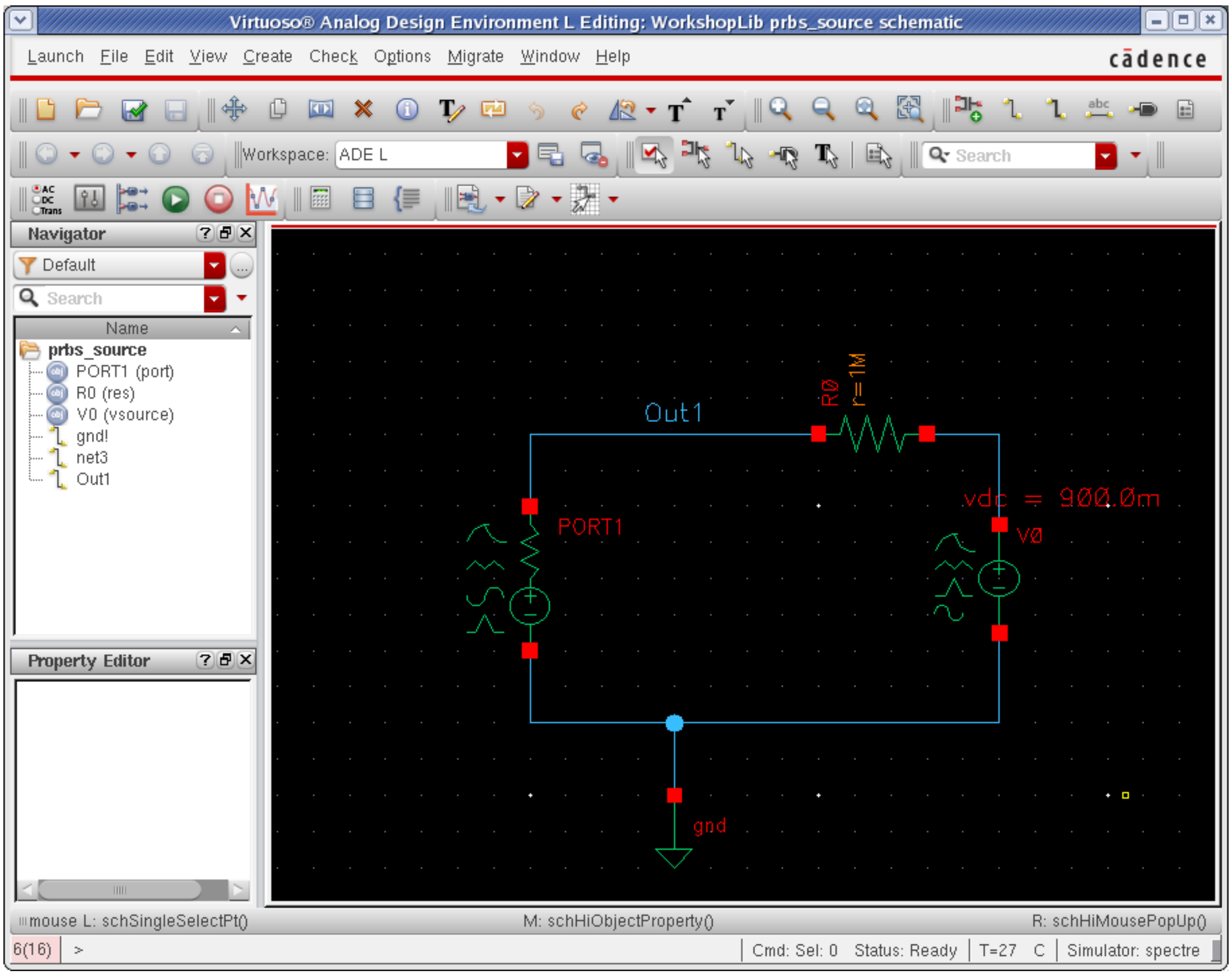
This specifies the starting bit number when repeating. Default value is `1`, which causes the first bit in the sequence to be the starting bit for the repeated pattern. The data repeats from the specified bit to the end of the bit string. The value of the parameter should be an integer from 1 to the length of the bit string.

rpttimes

This specifies the number of times the pattern should be repeated after the first sequence is generated. The default value is `0`. which specifies that the pattern should not be repeated. When its value is `negative`, the string repeats forever. If a finite number of repeats is specified and the simulation time increases to a time after the last bit is generated, the output will maintain the state of the last bit in the sequence.

Bit Waveform Examples

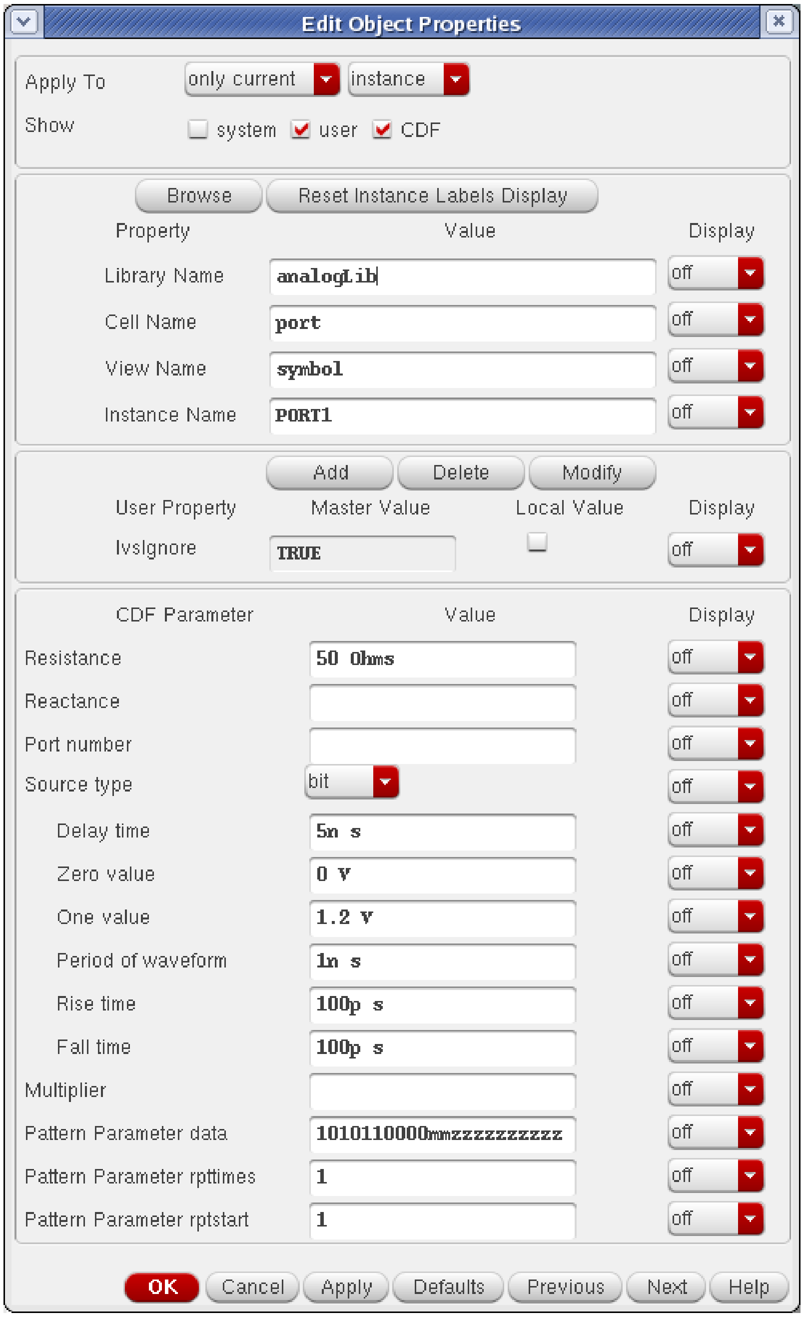
Here is the circuit that is used for all the examples.



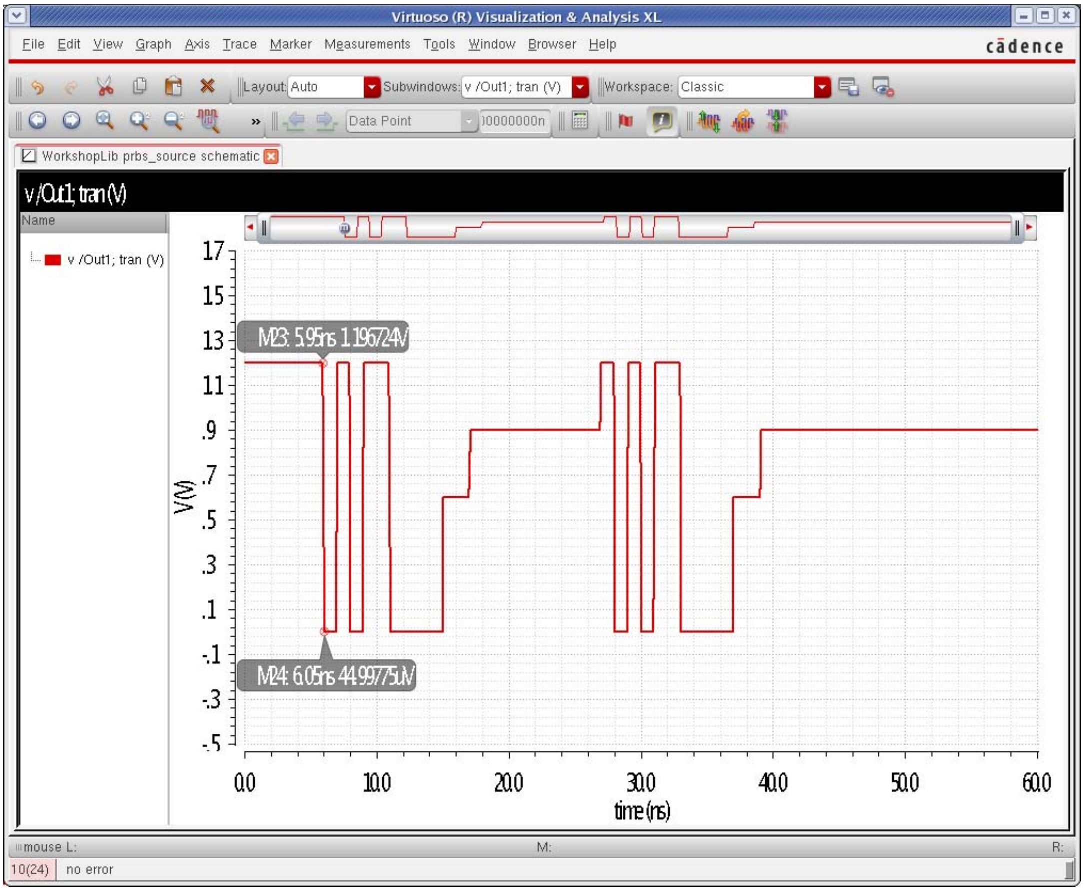
The port with the sourcetype set to bit is on the left. Then there is a 1MOhm resistor, and a DC voltage source set to 900mV.

Example 1

Here is a property list for the first example.



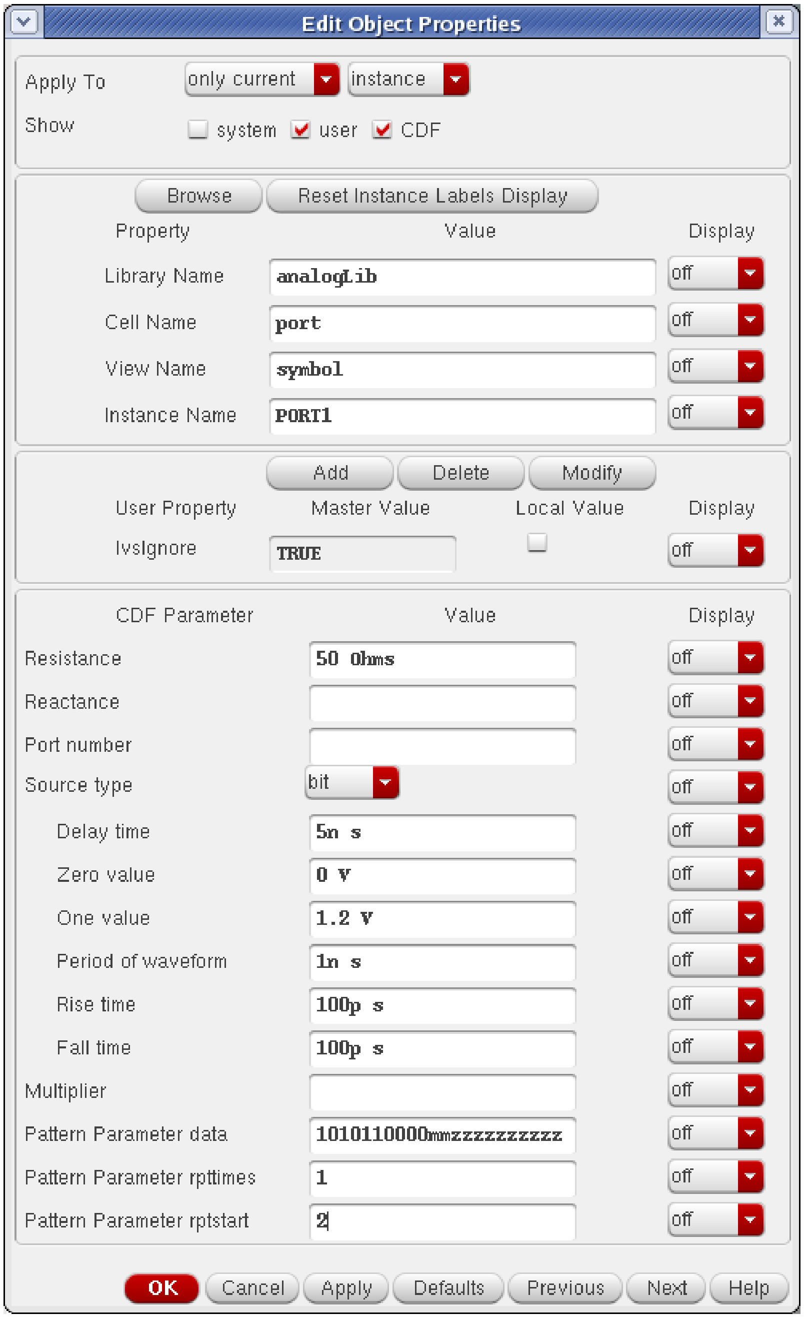
Here is the output signal which is *Out1* in the schematic.



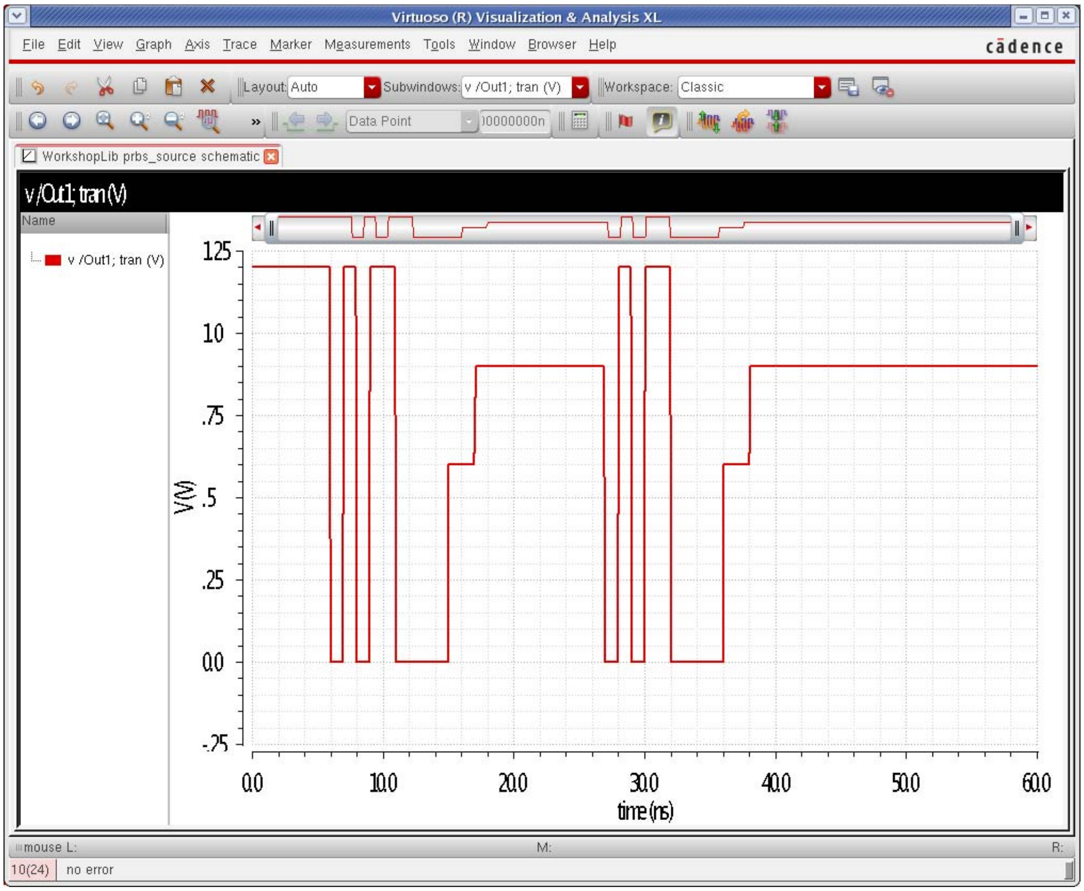
The delay time is 5n, and the bit period is 1n. This causes the signal to start at the first bit which is 1, and hold it for 5nsec. Then, it produces the first bit, which is 1 for the next nanosecond. If you look closely, you can see that the middle of the transition occurs at exactly 6nsec. The start of the transition is half the falltime before. The sequence is 1010110000mmzzzzzzzzzz. You can see the 0, 1, 0 transitions at 7nsec, 8nsec, and 9nsec. Then you see the two 1 states from 9nsec to 11 nsec, and the four 0 states from 11nsec to 15nsec. The next two states are middle states at 0.6v from 15nsec to 17 nsec. Then the output goes to high impedance, where the 900mv DC level at the output is seen. Rpttimes is set to 1, which repeats the pattern once, and the starting bit is the first bit in the sequence, which is a 1. At the end of the 900mv section, the signal transitions to the 1 state, and the entire pattern is repeated again. After the second pattern is done, the last state which is a z state is held after that.

Example 2

Here is the property list for the second example.

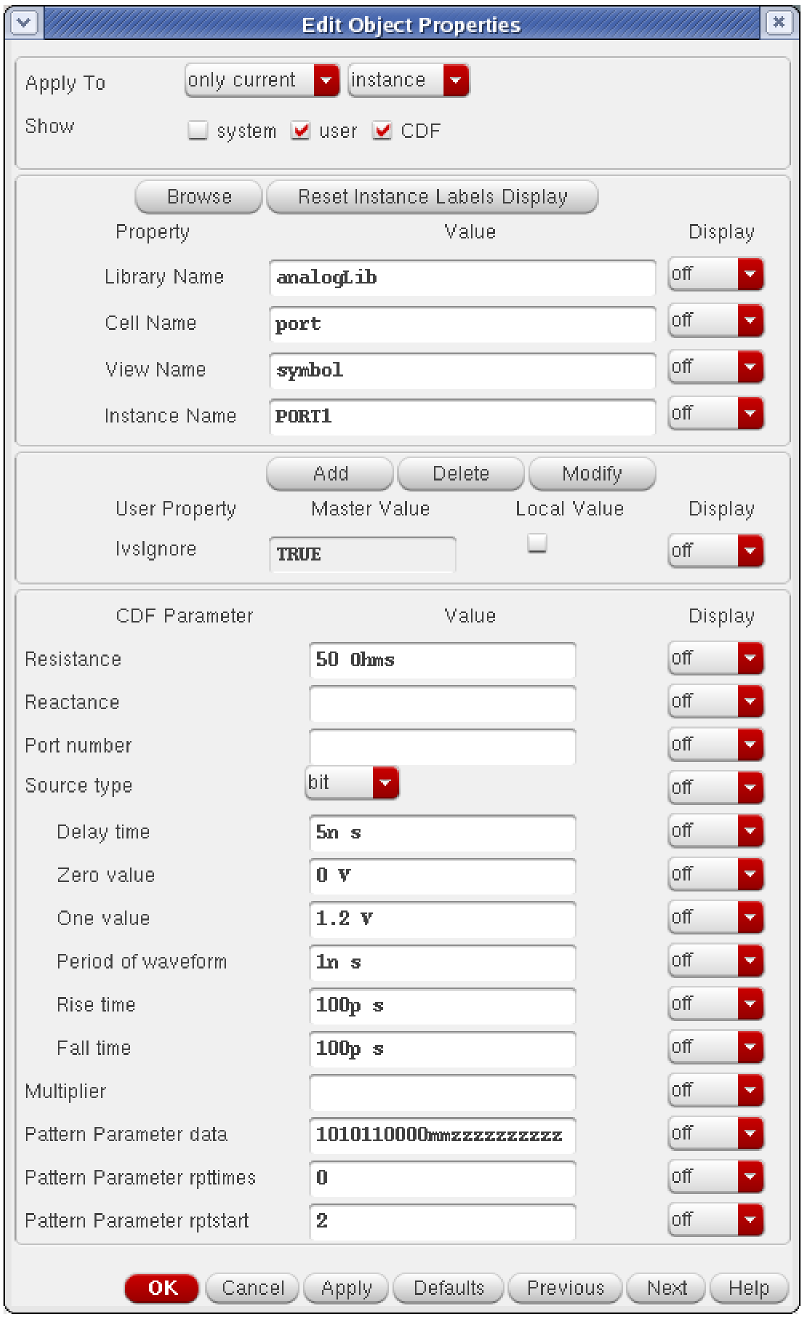


Everything is the same as the first example except for the repeat start, which is set to 2. This causes the second bit in the sequence, which is a 0, to be the starting bit. At the end of the first pattern, the signal repeats from the second bit in the sequence.

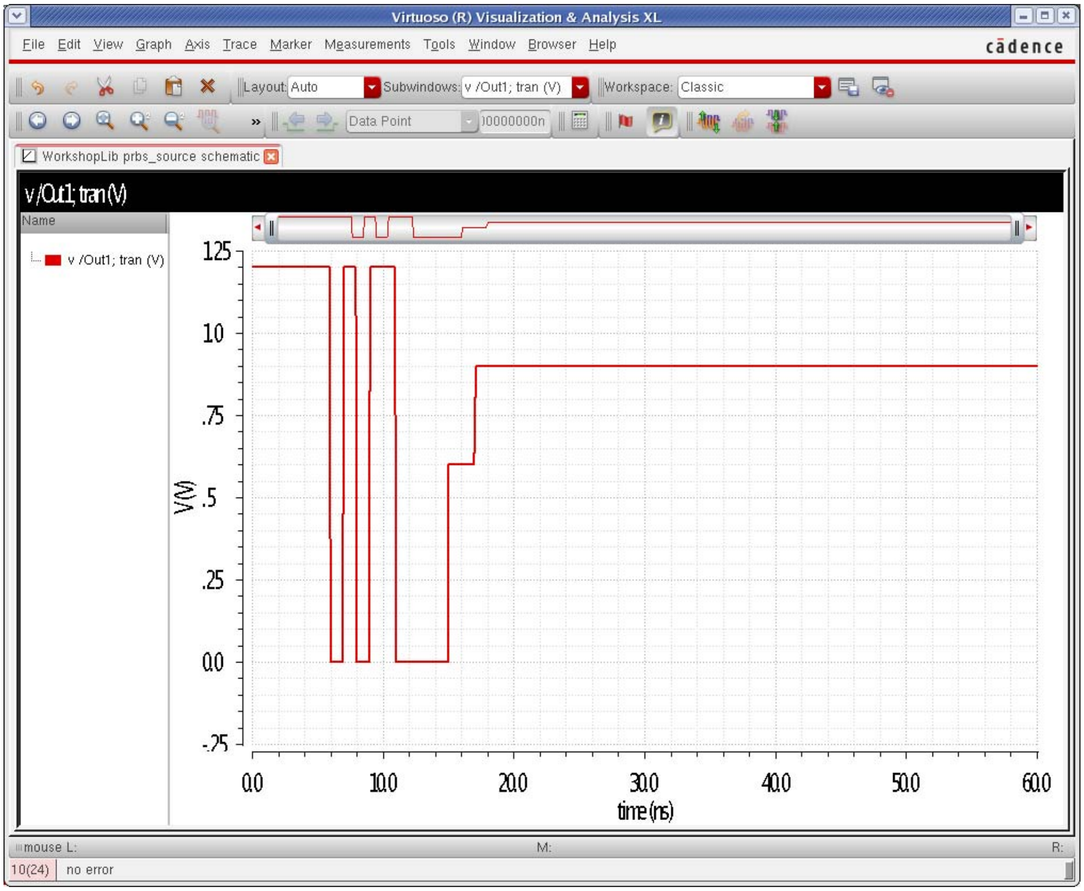


Example 3

Here is the property list for the third example.



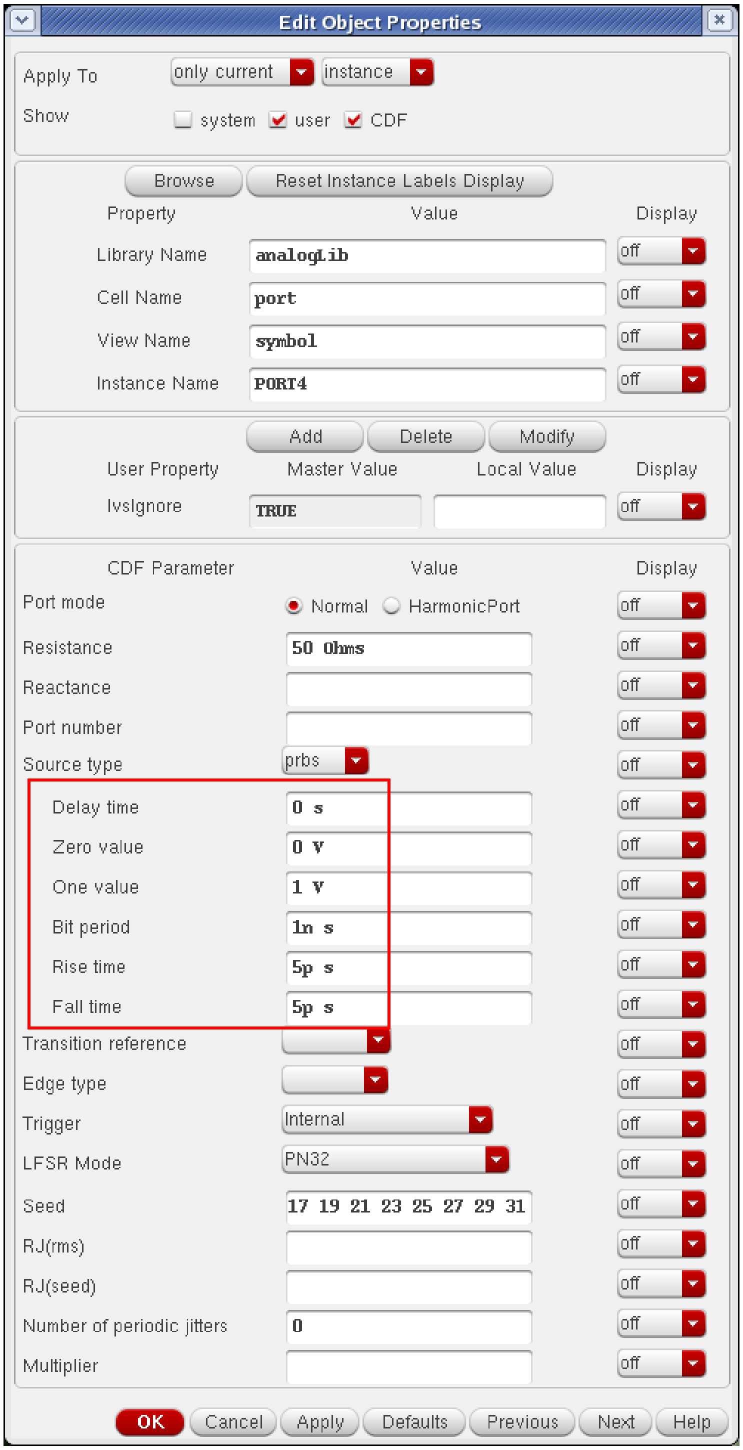
Here, there is no repeat.



PRBS Waveform Parameters

PRBS is an acronym for pseudo-random binary sequence. The PRBS source has three modes. It can be used to generate a maximum-length pseudo-random sequence. You can specify the beginning state and tap gains for a Fibonacci PRBS generator. A third mode allows reading an ascii file that describes the sequence of one and zero events to generate. To select this type of output, set the *Source Type* to *PRBS*.

For any mode, first you must define the output characteristics.



Delay Time

This specifies a time where the source stays at the zero value, and does nothing. The default is zero seconds. At the end of the delay time, the first bit will be produced.

Zero Value

This specifies the analog voltage to generate in the zero state. The default is zero volts. The port internally doubles the specified voltage as usual. To get the correct voltage when using the port, make sure that you add a matching resistor across the port terminals.

One Value

This specifies the analog voltage to generate in the one state. The default is one volt. The port internally doubles the specified voltage as usual. To get the correct voltage when using the port, make sure that you add a matching resistor across the port terminals.

Bit Period

This specifies the time of a single bit. This measures the time from the middle of the transition to the middle of the transition between the zero and one state for the duration of a single bit.

Rise Time

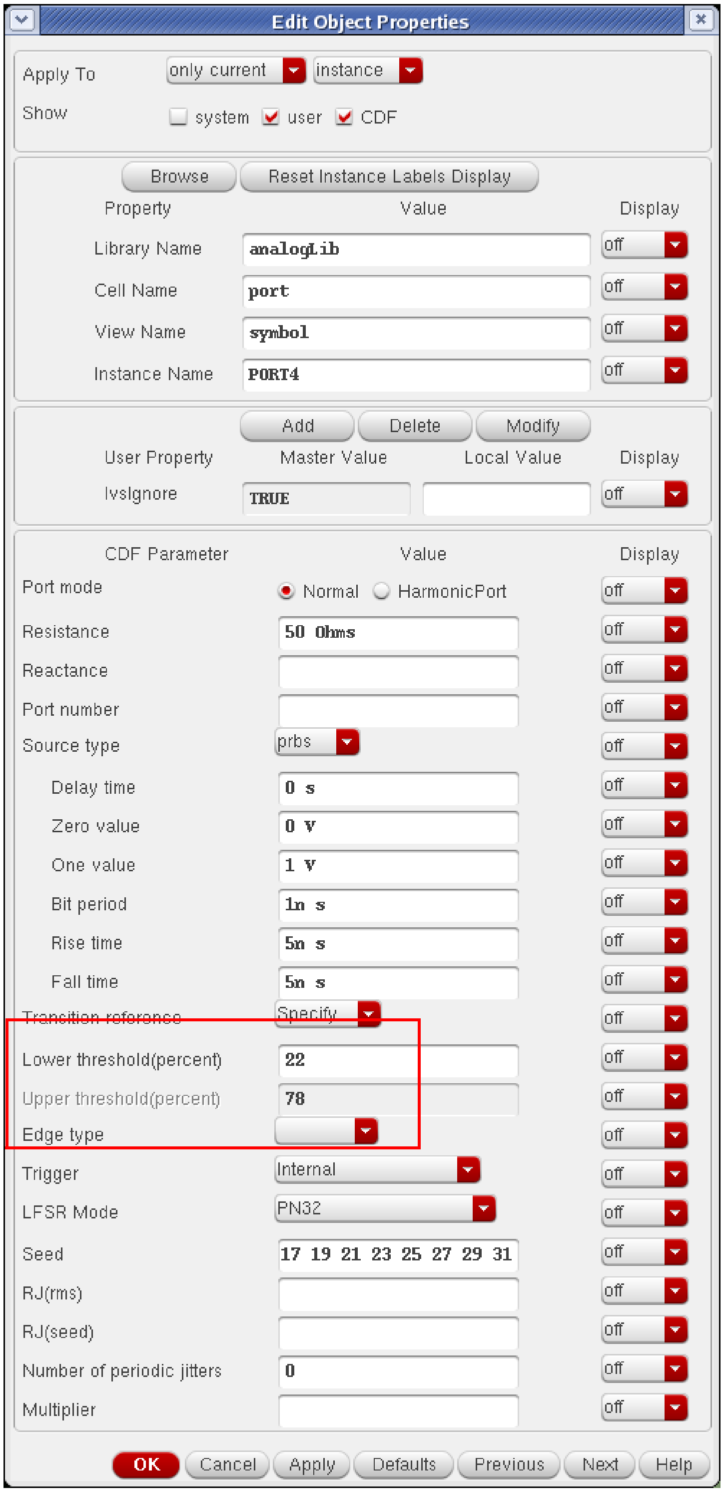
This is the zero percent to 100 percent (by default) risetime for the transition from the zero state to the one state. The waveshape of the transition and the definition of risetime are settable.

Fall Time

This is the zero percent to 100 percent (by default) falltime for the transition from the one state to the zero state. The waveshape of the transition and the definition of risetime are settable.

Transition Reference

This specifies the thresholds for the rise time and fall time specification. This can be set to 0 to 100% (default), 10 to 90%, 20 to 80%, 30 to 70%, 40 to 60%, and *Specify*. When *Specify* is selected, the lower threshold must be entered. The upper threshold is calculated and is equidistant from the midpoint as the lower threshold. For example, if 22% is set for the lower threshold, 78% will be the upper threshold.



Edge Type

The waveform shape of the transition defaults to linear, which generates a traditional pulse with straight edges. Selecting halfsine causes sinusoidal rounding of the edges.

LSFR Mode

This controls whether a maximal length sequence is generated, a file with a pattern of 1 and 0 states ia read and produced, or whether you specify the shift register, taps and starting the shift register state yourself.

Maximum Length Sequence LSFR Mode

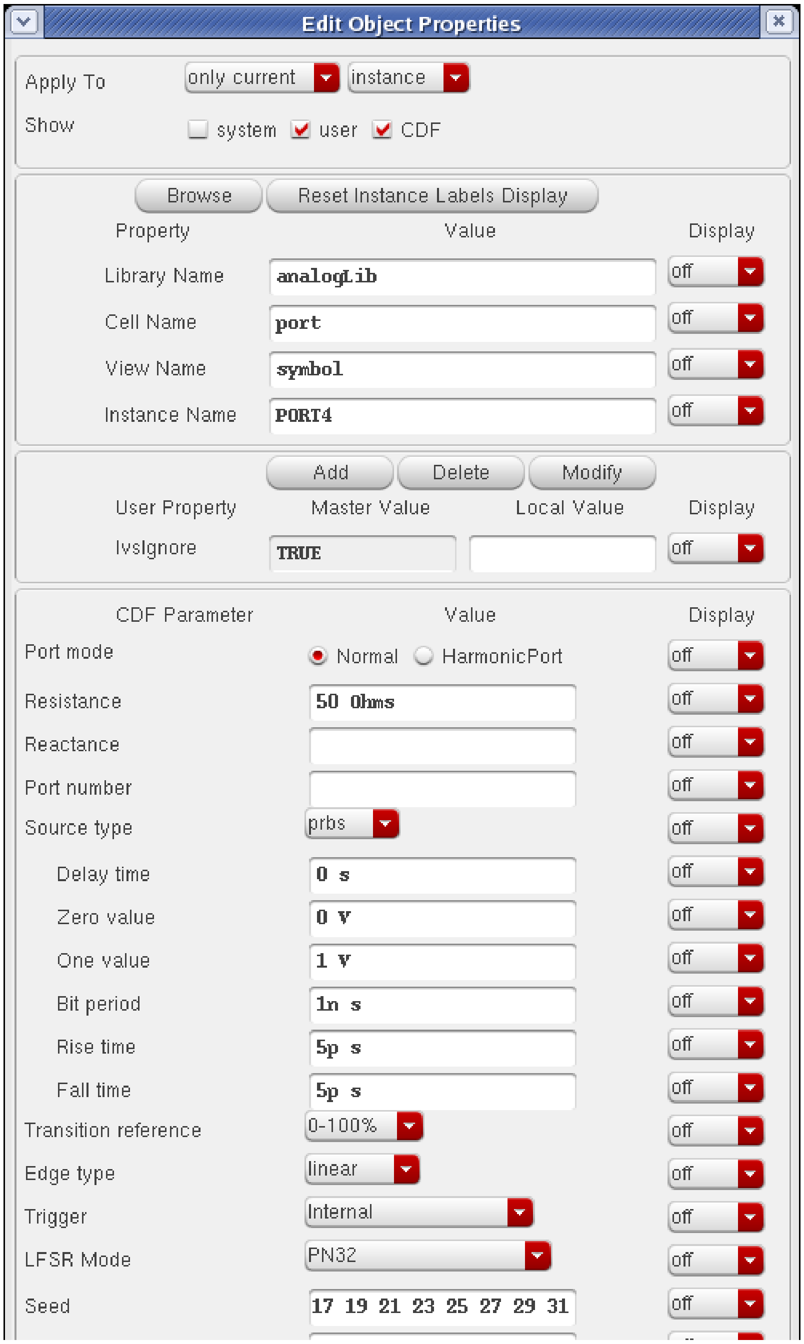
The default is PN32, which generates a maximum length sequence (MLS) with a 32-bit shift register, and alternating 1 and zero states in the shift register as the starting state. To generate a different Maximum Length Sequence, change the seed parameter to define a different starting state, or define your own register, taps, and starting the state manually. Maximum length sequences using 2 to 32 bit shift registers are available. The table below defines the taps for each length of the available shift registers.

Number of bits	Taps
2	2, 1, 0
3	3, 2, 0
4	4, 3, 0
5	5, 3, 0
6	6, 5, 0
7	7, 6, 0
8	8, 6, 5, 4, 0
9	9, 5, 0
10	10, 7, 0
11	11, 9, 0
12	12, 6, 4, 1, 0
13	13, 4, 3, 1, 0
14	14, 5, 3, 1, 0

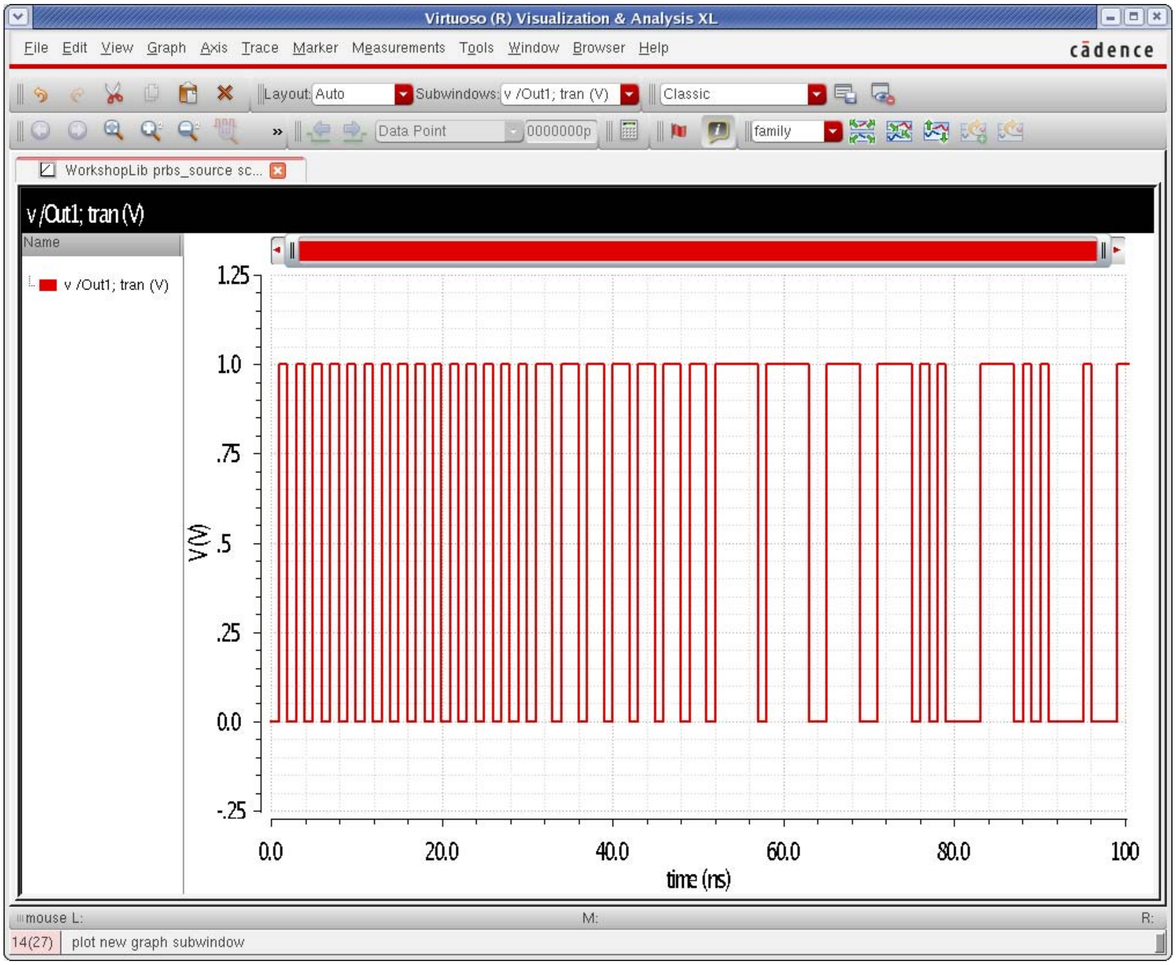
Number of bits	Taps
15	15, 14, 0
16	16, 15, 13, 4, 0
17	17, 14, 0
18	18, 11, 0
19	19, 6, 2, 1, 0
20	20, 17, 0
21	21, 19, 0
22	22, 21, 0
23	23, 18, 0
24	24, 23, 22, 17, 0
25	25, 22, 0
26	26, 6, 2, 1, 0
27	27, 5, 2, 1, 0
28	28, 25, 0
29	29, 27, 0
30	30, 6, 4, 1, 0
31	31, 28, 0
32	32, 22, 2, 1, 0

The taps are numbered from left to right, starting with zero and going through N where N is the number of bits in the shift register.

Below is an example with 0 to 100% risetime, linear edge type, with a 32 bit MLS set.

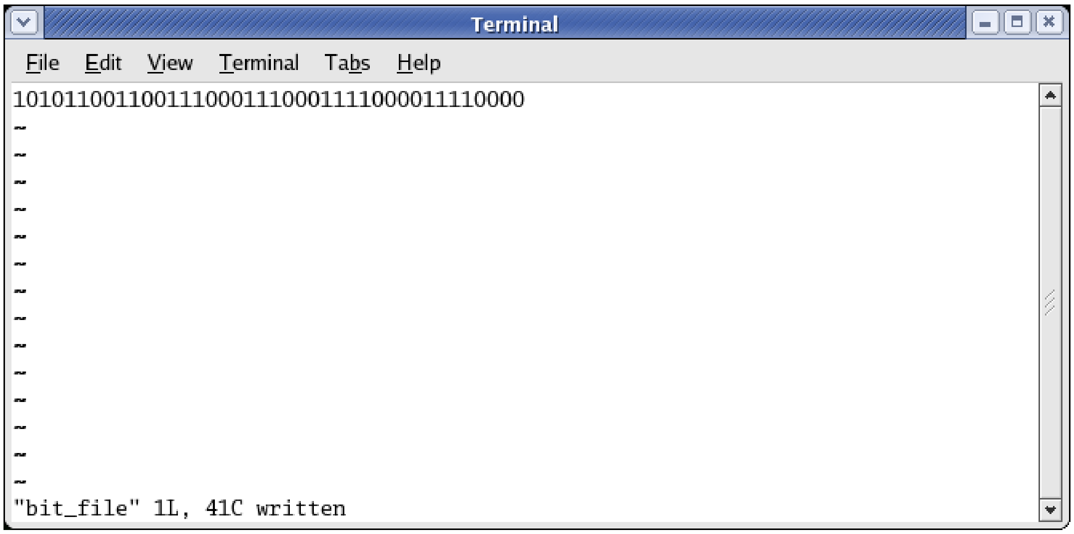


When this runs in the transient, first the alternating pattern of bits defined as the starting state of the shift register is produced, and then the sequence begins.



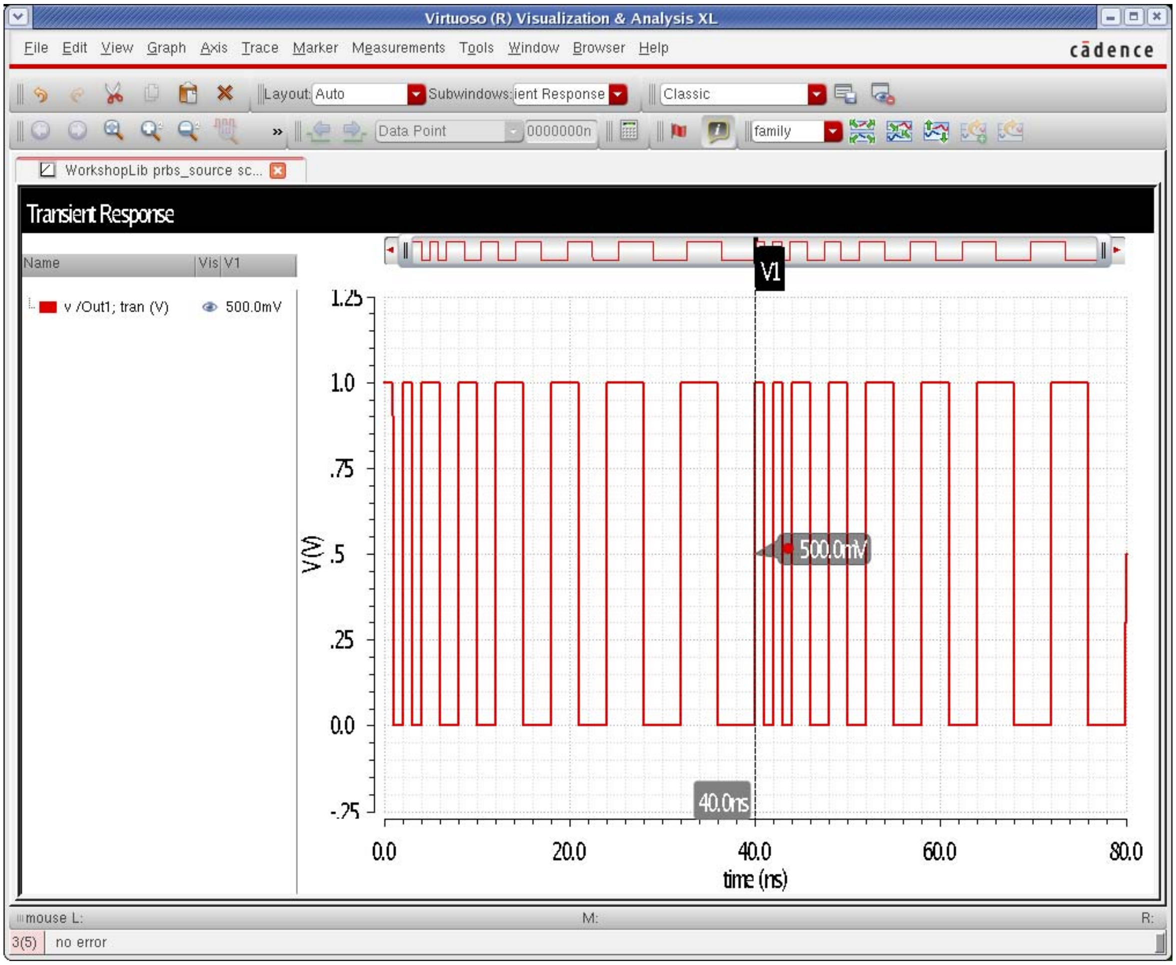
Specify Bit File LSFR Mode

In this mode, an ASCII file with a sequence of 1 and 0 states is read. In this way, any arbitrary sequence can be used in the PRBS source. The states in the file are read from left to right, and if the time of the simulation exceeds the number of bit states in the file, the pattern will repeat. An example of a bit file is shown below.



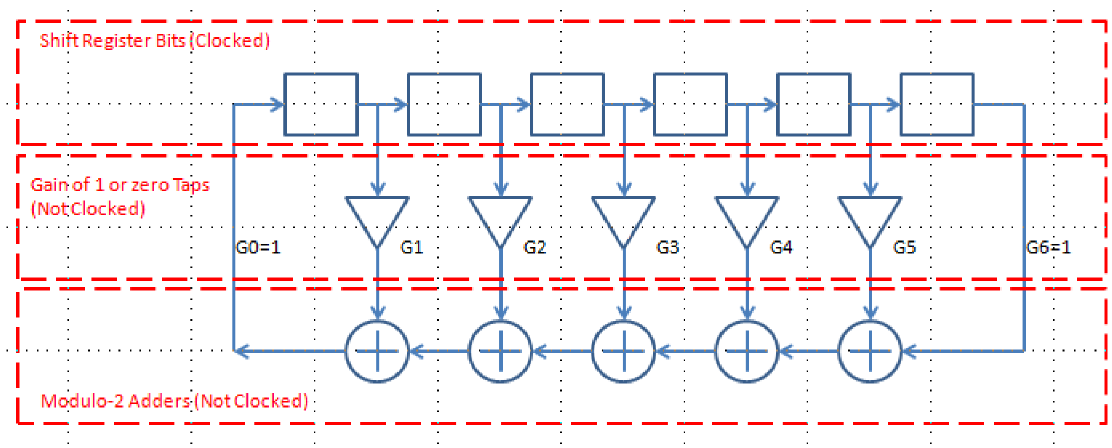
When the file with the bit pattern is available, set the LFSR mode to *Specify bit file*, and use the file browser to select the file. If you specify the filename manually, enter the complete path to the file for the file name. An example with the external bit file is shown below.

Note that the pattern in the file is reproduced, and repeats, as shown below.



Specify Seed and Taps LFSFR Mode

To use this feature, you must set the *Seed* and *Tap* information. The general configuration of a 6-bit Fibonacci shift register is as follows:



The boxes on the top are the shift register bits. The non-inverters in the middle can either be switched on or off, and are controlled by the taps vector. Taps zero and N (The taps at both ends of the shift register) are always on, which enables the circular connections. The shift register is clocked every period, and the bit shifts right. Depending on the setting of the feedback taps, different bits of the shift register can change the next bit that is loaded into the beginning (left) of the shift register. This logic is asynchronous. The adders at the bottom do modulo-2 addition, which is equivalent to an exclusive-or logic function.

Important

There are different conventions for numbering the taps. Please note that in Spectre, the taps are numbered from left to right, starting at zero. The bits in the shift register are numbered from left to right starting at one.

Seed

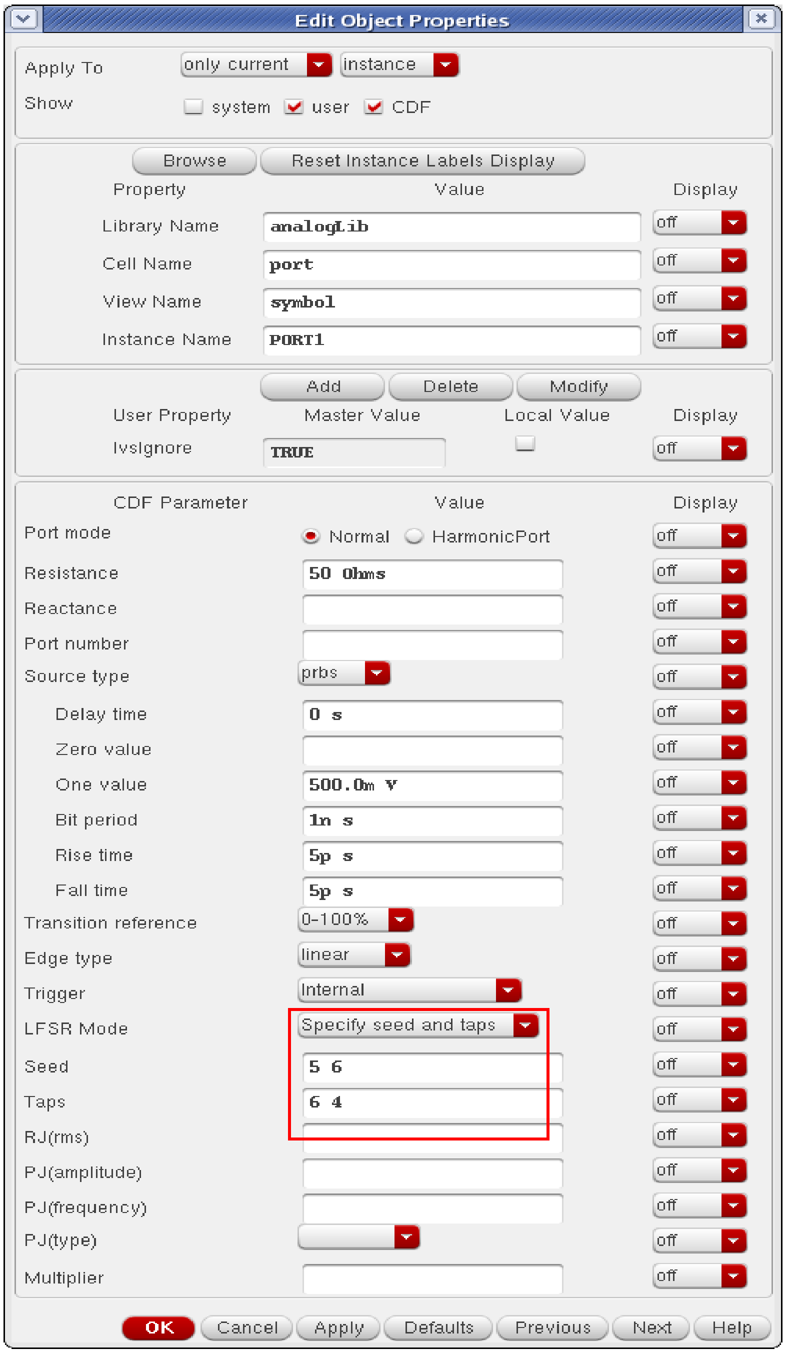
This is a list of bit numbers to set to the one state that are used to preload the shift register at time zero. An example is 1 3 5, which would load the first, third, and fifth bits with the one state, and the rest of the bits to the zero state. If the size of the shift register is smaller than the data, the higher bit numbers are simply discarded. The default is to set all the odd bits to the one state.

Taps

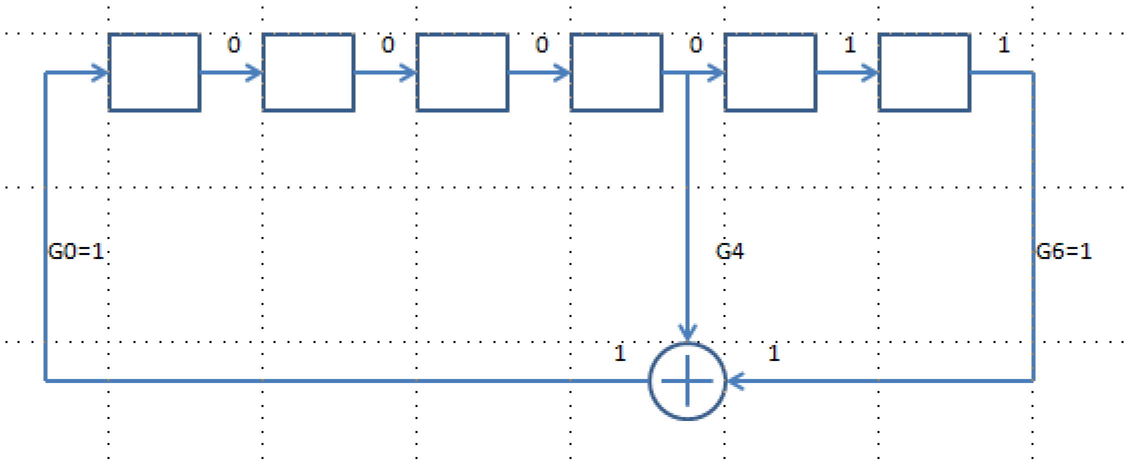
This is a list of the taps to set to one, and also defines the length of the shift register. In the above example, the numbers 1, 2, 3, 4, and 5 can be specified. Additionally, 6 must be set to define the length to be 6 bits. The zero tap and the tap with the highest value in the *Taps* field are always set to 1. By convention, the first number is the number of bits in the shift register, and the next series of numbers are in descending order and control the gain of the taps. In Spectre, the numbers can be specified in any order, and the largest number sets the size of the shift register. For example, 6 4 2 (separated by spaces) specifies a 6-bit shift register with G2 and G4 set to one. The rest of the gains are set to zero, or an open connection. Also note that 6 4 and 4 6 give the same sequence. The largest number in the string sets the size of the shift register, and the rest of the numbers define which taps are on and off. If the *Taps* field has a single number, this defines the size of the shift register, and the values in the *Seed* field will endlessly circle through the shift register.

Detailed Example of Setting the Taps

To manually specify the seed and taps, you must first set the LSFR Mode to Specify seed and taps. Next, set the size of the shift register, and define the taps. In this example, the size is set to six, and the fourth tap is enabled. The setting of the Taps property is 6 4 as shown below. To set the bits in the shift register to the digital pattern 000011, set 5 6 in the *Seed* property. This will load the one state into the rightmost 2 bits of the 6 bit shift register.

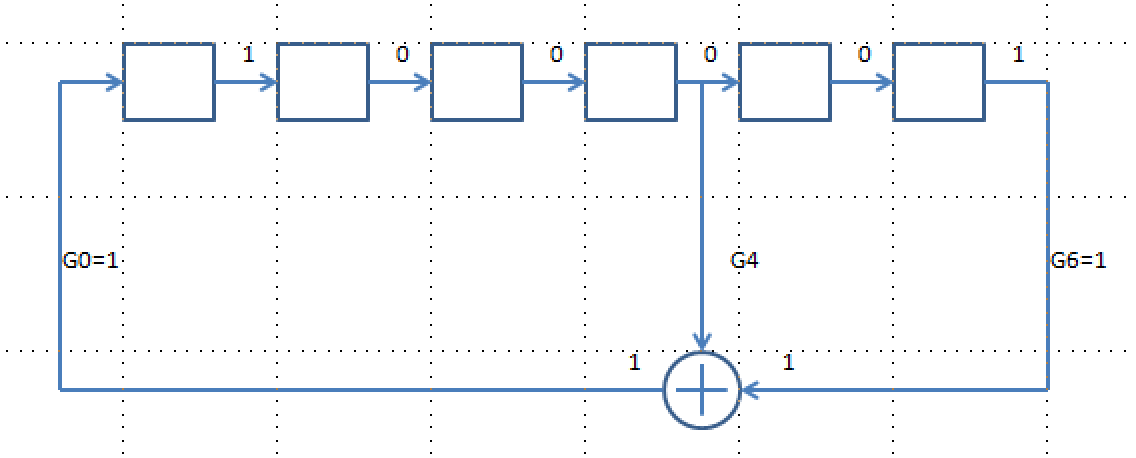


The internal bits get set, as shown below. When the tap is on, it is a connection. When it is off, it is an open. When the adders that are not shown have a zero on the top connection, they take the bit from the right, add zero, which just brings that bit to the left. The other adders just become wired connections.



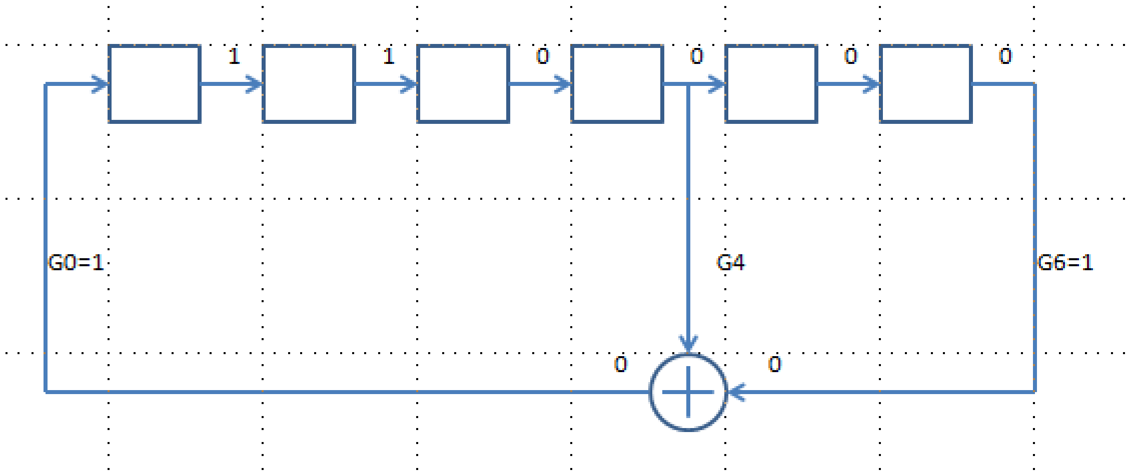
The taps vector enables G4, which brings the fourth bit from the left to the adder for that section. The adder sees a zero and a 1, so the output is 1. This 1 will be loaded in to the shift register at the next clock event.

At the next clock, the 1 from the output of the adder is loaded into the leftmost bit, and the contents shift right.

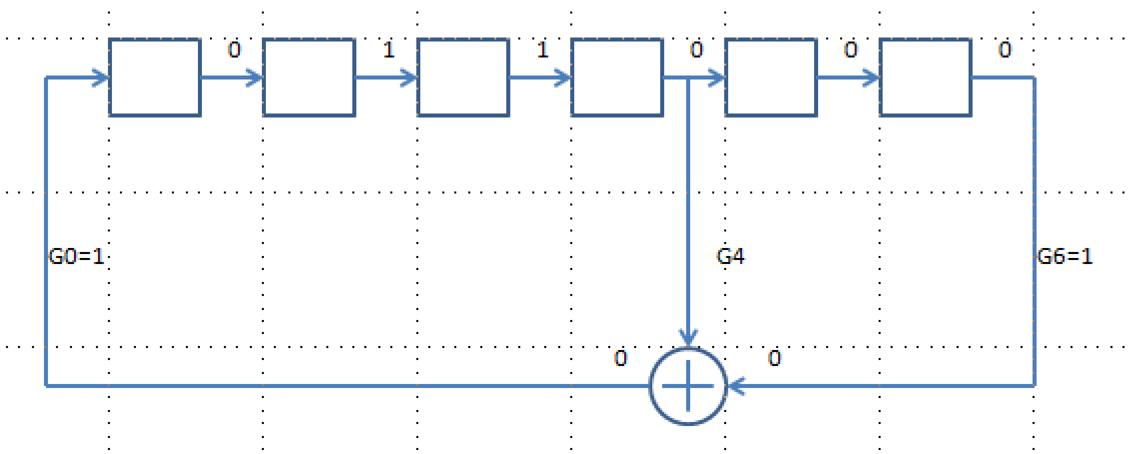


The output of the right bit is still 1, and the 4th bit from the left is still 0, so the output of the adder stays at 1.

At the next clock, the 1 is loaded in the leftmost bit, and the data shifts right.



Now, the adder sees zero on both inputs, so the output changes to zero. This is loaded in the next clock, and the data shifts right.

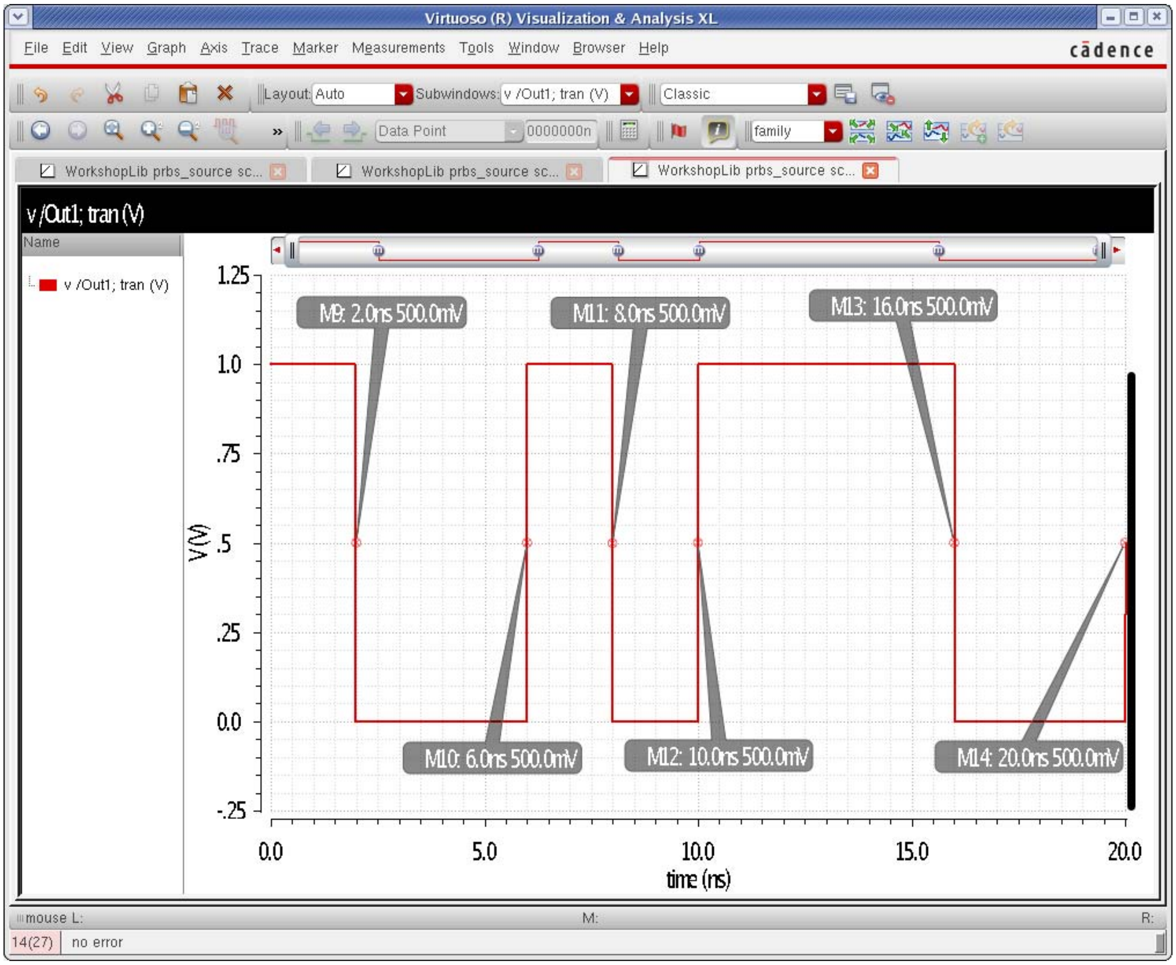


If you look at the sequence of the bits on the right side of the shift register so far, 1 1 0 0 has been generated. The sequence is from left to right. The first bit is a 1.

Continuing this exercise, it can be shown that the bits on the output should be 1 1 0 0 0 0 1 1 0 0 1 1 1 1 1 1 0 0 0 0 for the first 20 bits.

Now assume a bit period of 1nsec, The first bit, which is a 1, is generated from zero seconds to 1 nsec. The second bit, which is a 1, is generated from 1nsec to 2nsec. The third bit, which is a zero, is generated from 2nsec to 3nsec, The first transition from a 1 to a zero occurs at

2nsec. The bit sequence above would generate transitions at 2n, 6n, 8n, 10n, and 16n, which is shown in the result below.



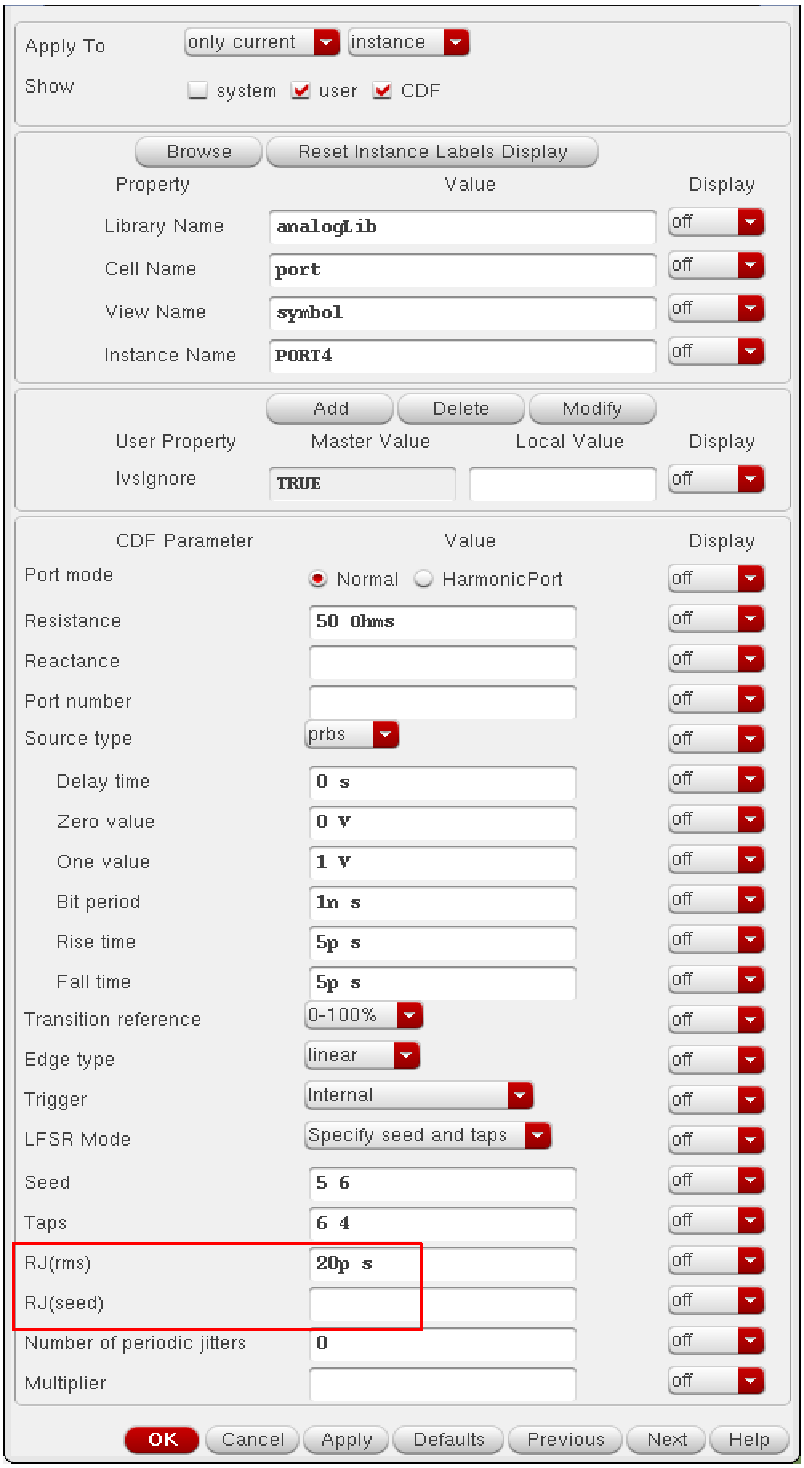
Jitter Generation

Random Jitter

The PRBS source can generate random jitter, periodic jitter, or a combination of both types of jitter. For periodic jitter, up to three periodic jitter sources can be specified. First, random jitter will be considered. The property for setting random jitter is *RJ(rms)*.

RJ(rms)

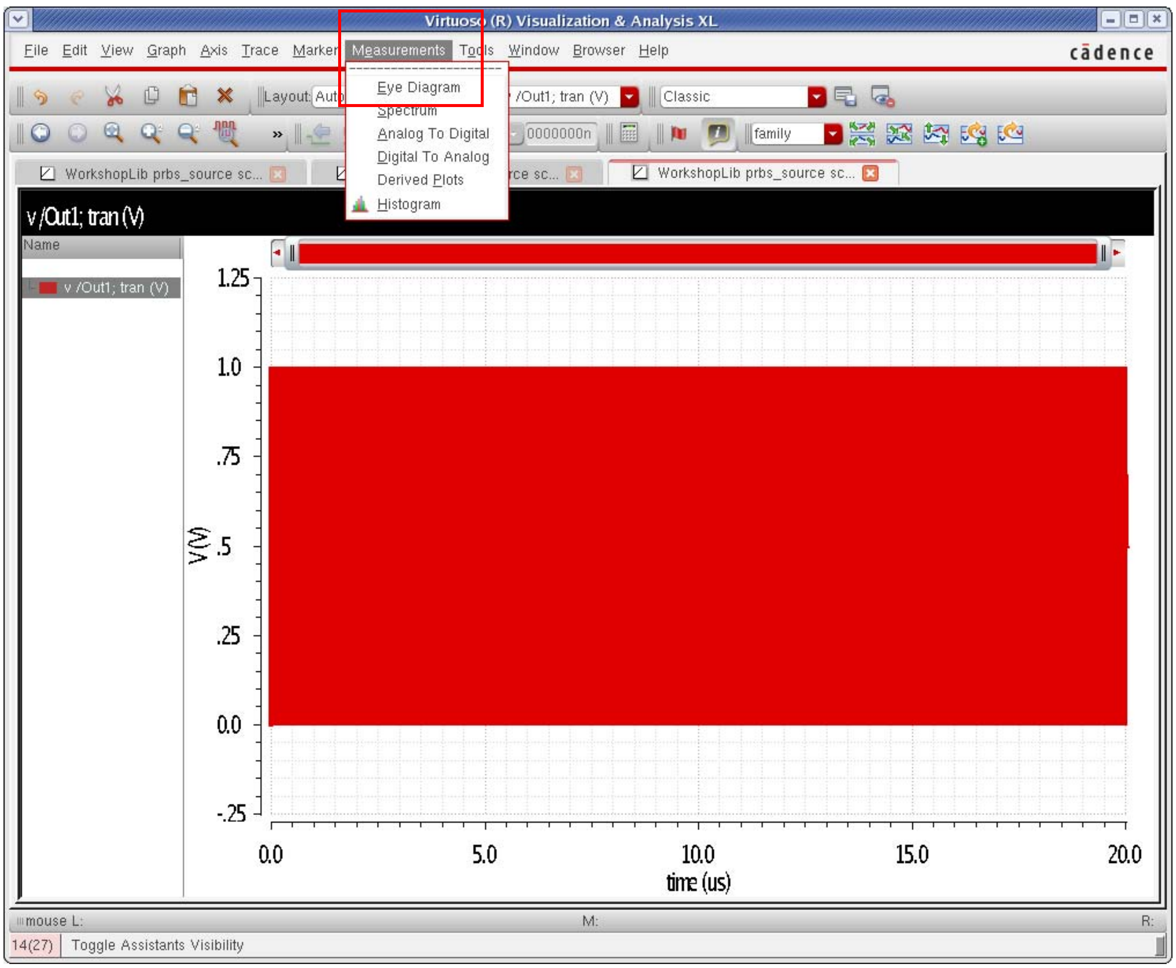
This property defines one standard deviation for random jitter to be generated in the PRBS output waveform. Random jitter with a gaussian distribution will be generated on the transitions of the output waveform. This example has 20 psec jitter added, as shown below.



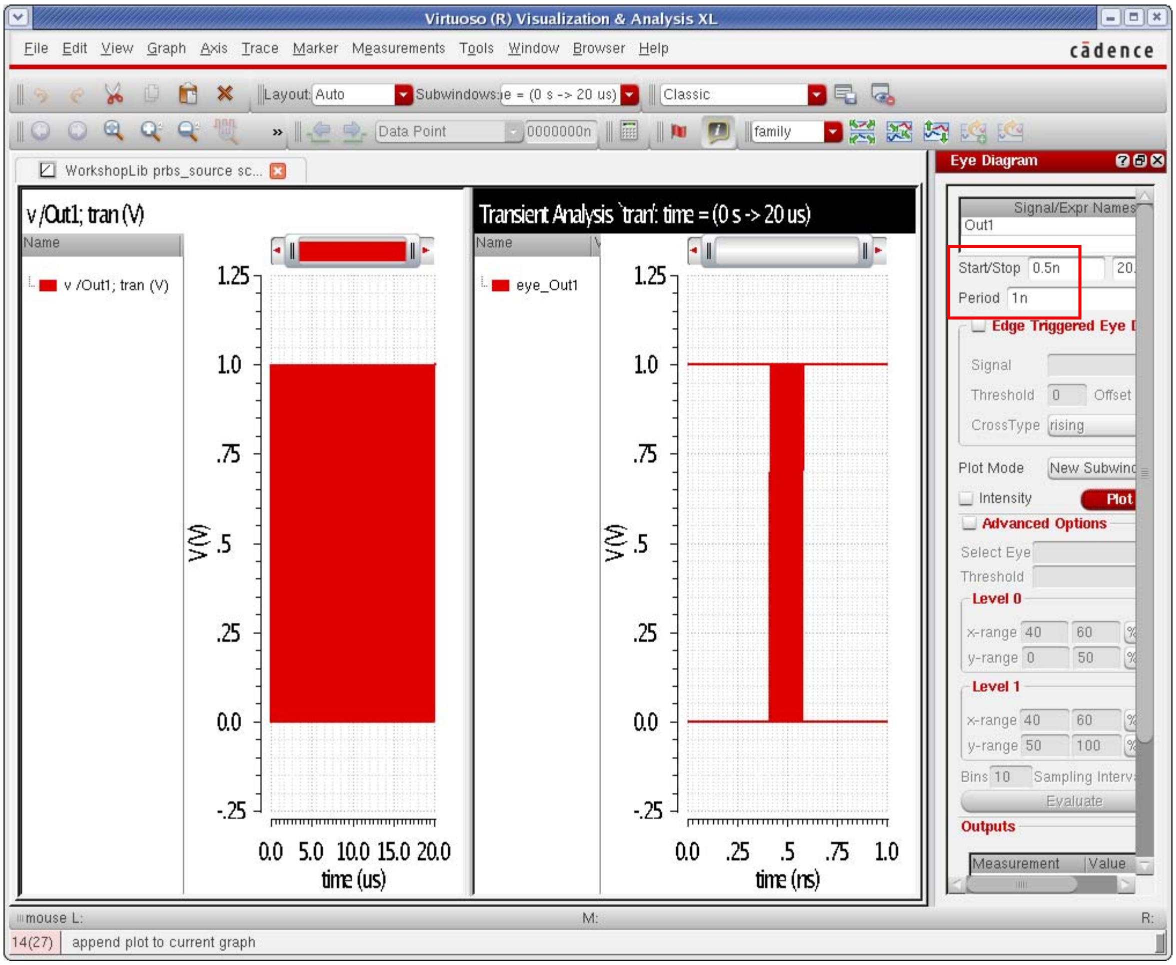
RJ(seed)

This is an optional property that can be used to generate different jitter sequences. Specify an integer in this field. The default value is 1.

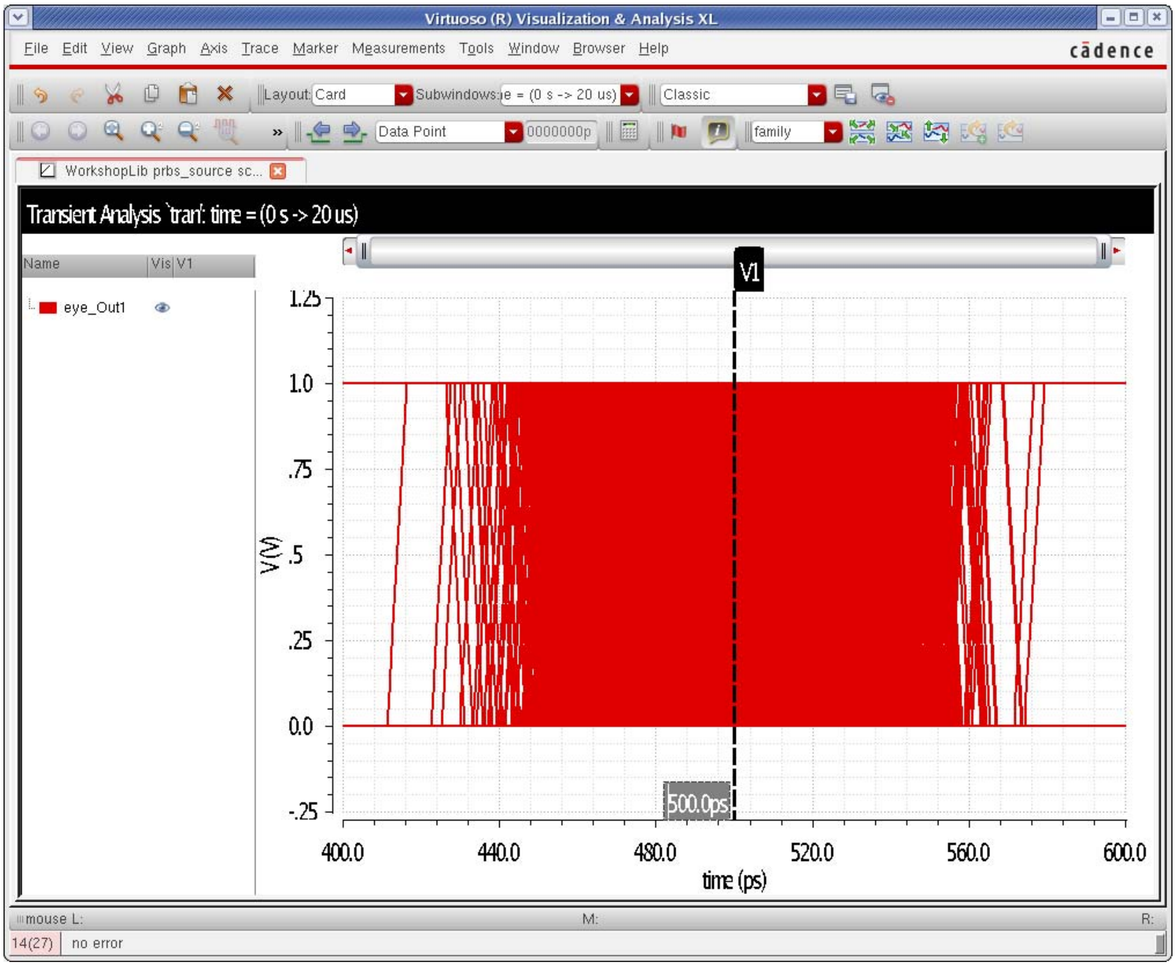
Next, this was run for 20 microseconds using the transient, and plotted in the waveform tool. To see the jitter, first select the trace, and then select *Measurements - Eye Diagram*.



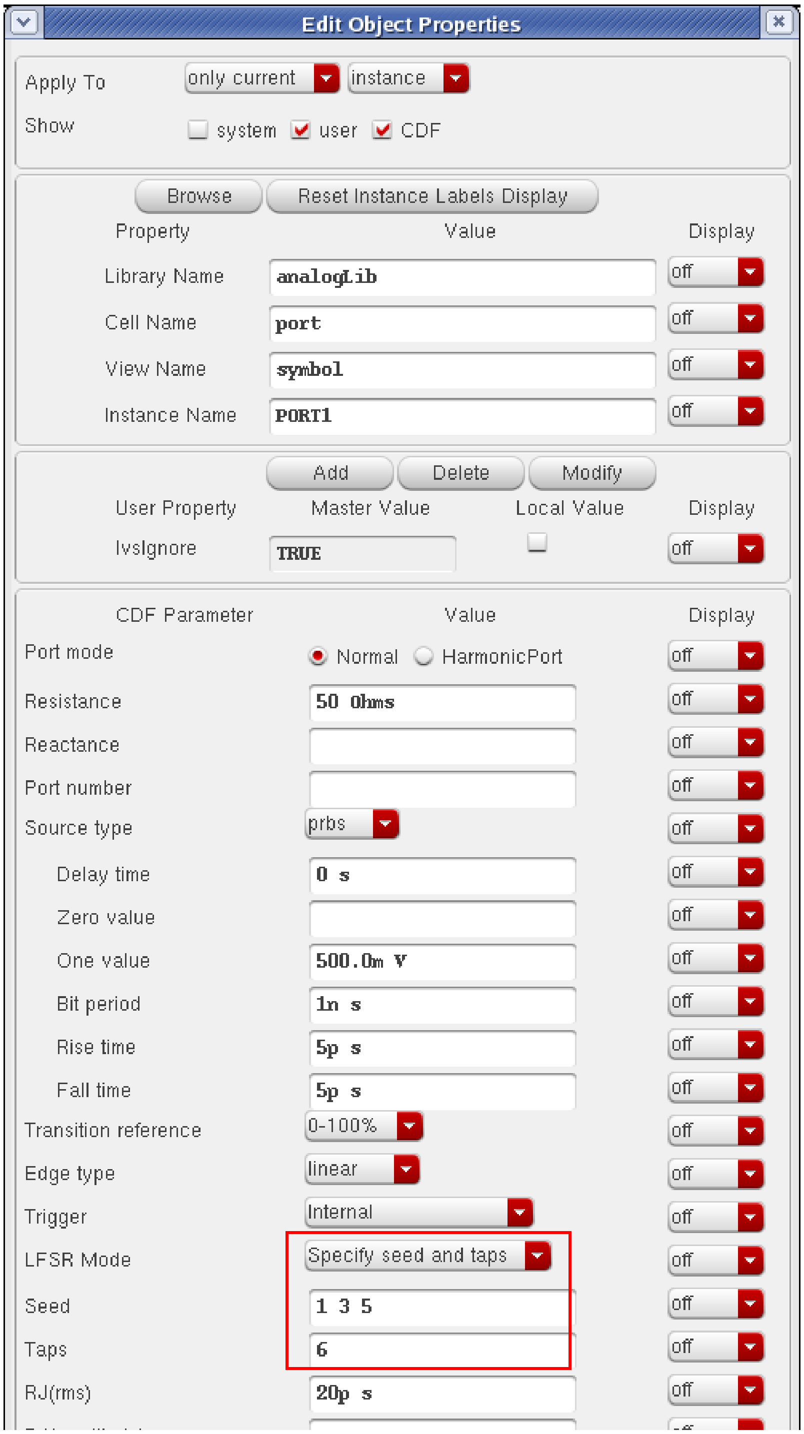
The eye diagram assistant starts. The data has a periodicity of 1 nsec, To put all the transitions in the center of the eye, a 0.5 nsec *Start Time* is set, along with the 1 nsec periodicity of the output signal. The timing variations blur the transitions in the center.



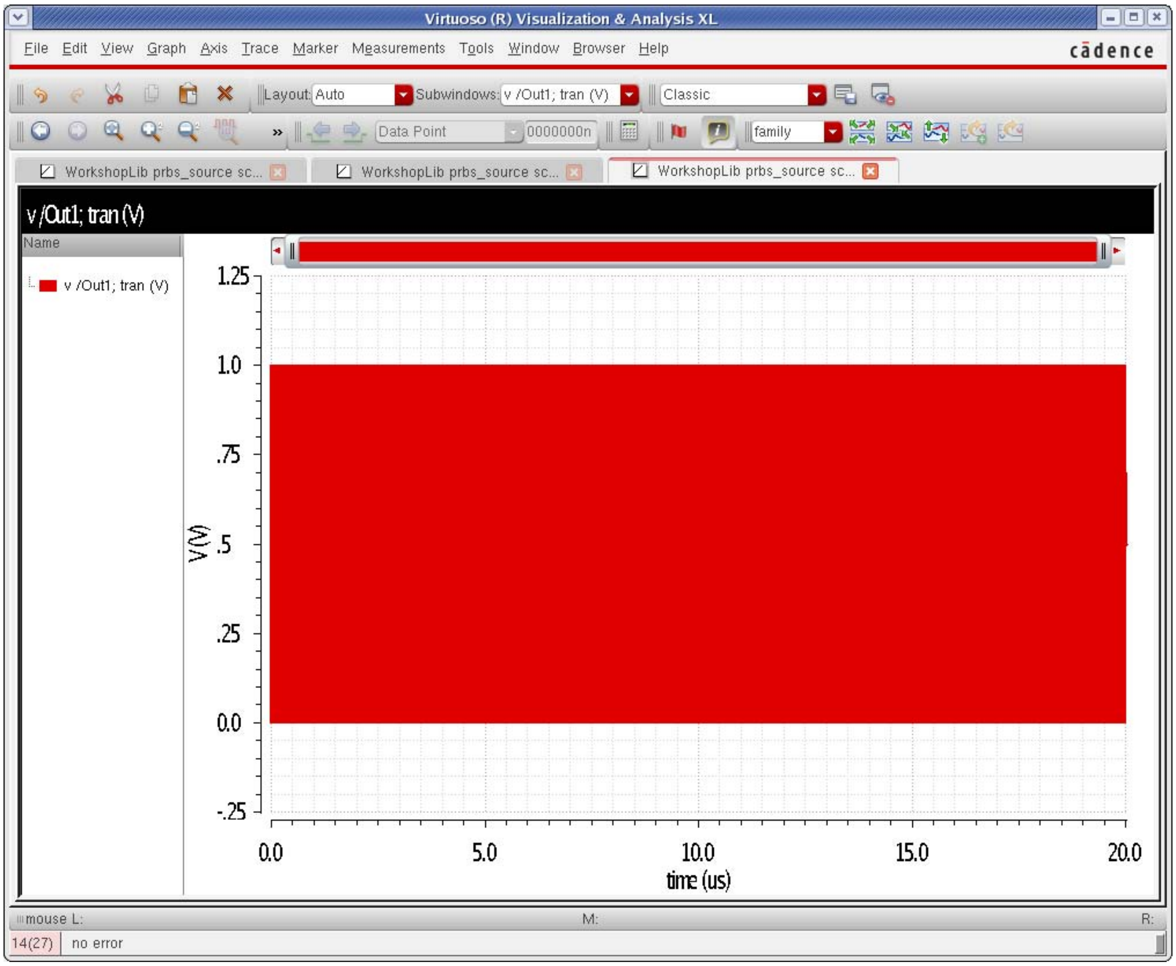
To see where the wave is centered, a zoom operation was performed, and a vertical marker has been placed at 0.5 nsec.



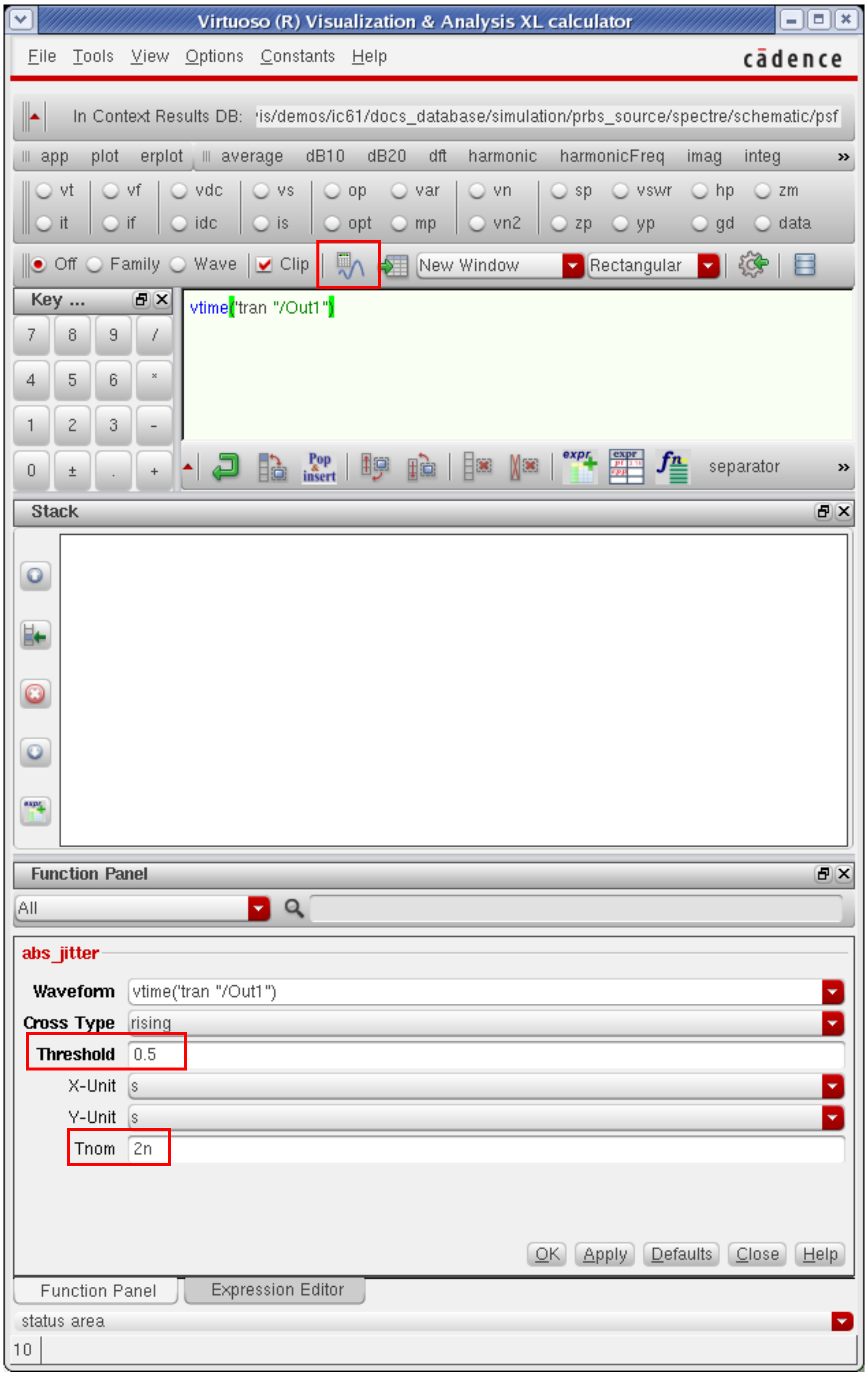
In order to get standard deviation calculation, we need to modify the shift register so that it provides a periodic output. This is done in the example below by setting the seed to 13 5, which sets a 1 state in the first, third, and fifth bit, and then taps is set to 6, which rotates the pattern through the shift register endlessly. In this way, a square wave is generated with a periodicity of 2 nsec. This periodicity is needed in order for the standard deviation functions to work.



Now the transient is run for a large number of cycles. in the example below, it is run for 20usec, which gives 10,000 cycles of shift register output.

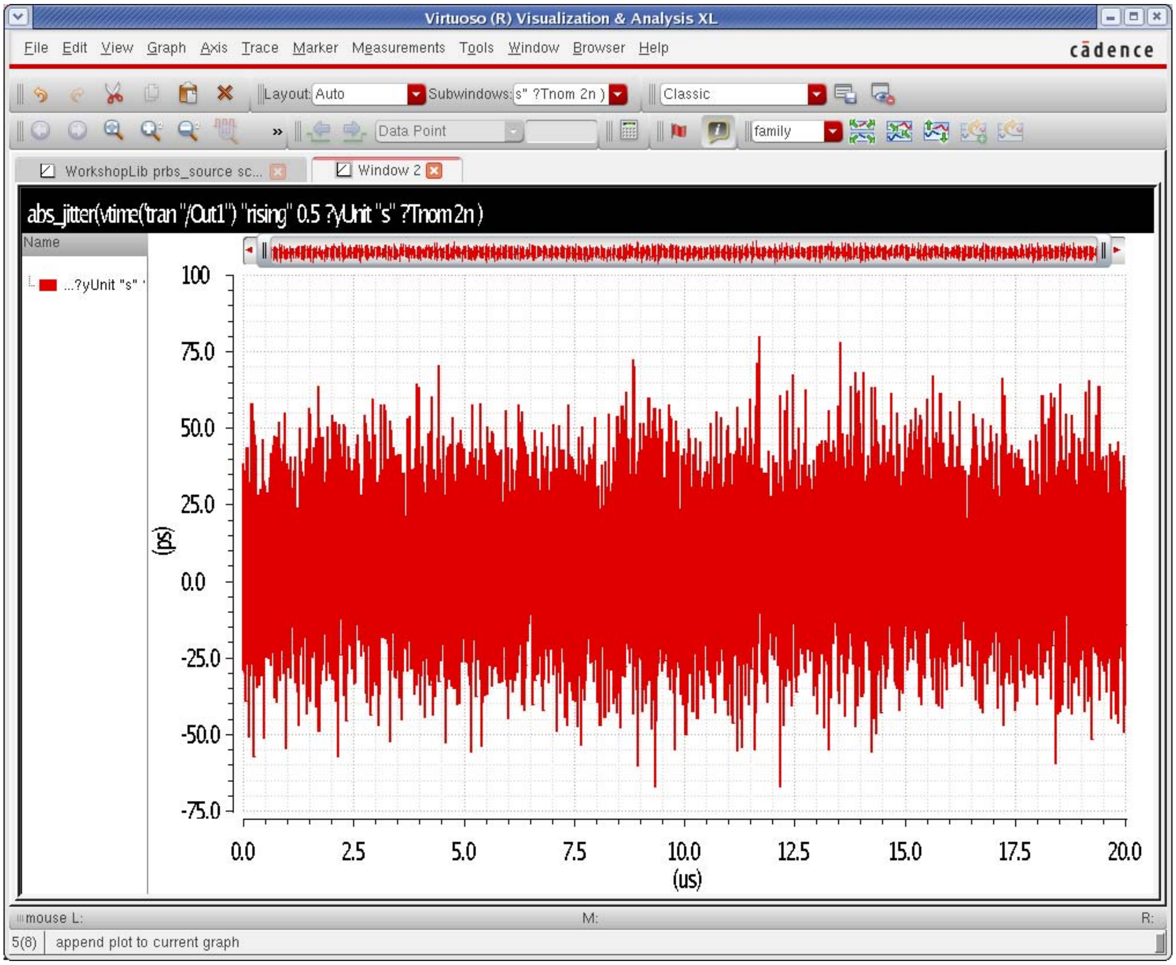


Select the trace, click the right mouse button and select *Send To Calculator*. This starts the calculator. Select the *abs_jitter* function. Set the threshold as appropriate for your circuit. 0.5 is appropriate for this example. Also set the period of the waveform in the *Tnom* field.

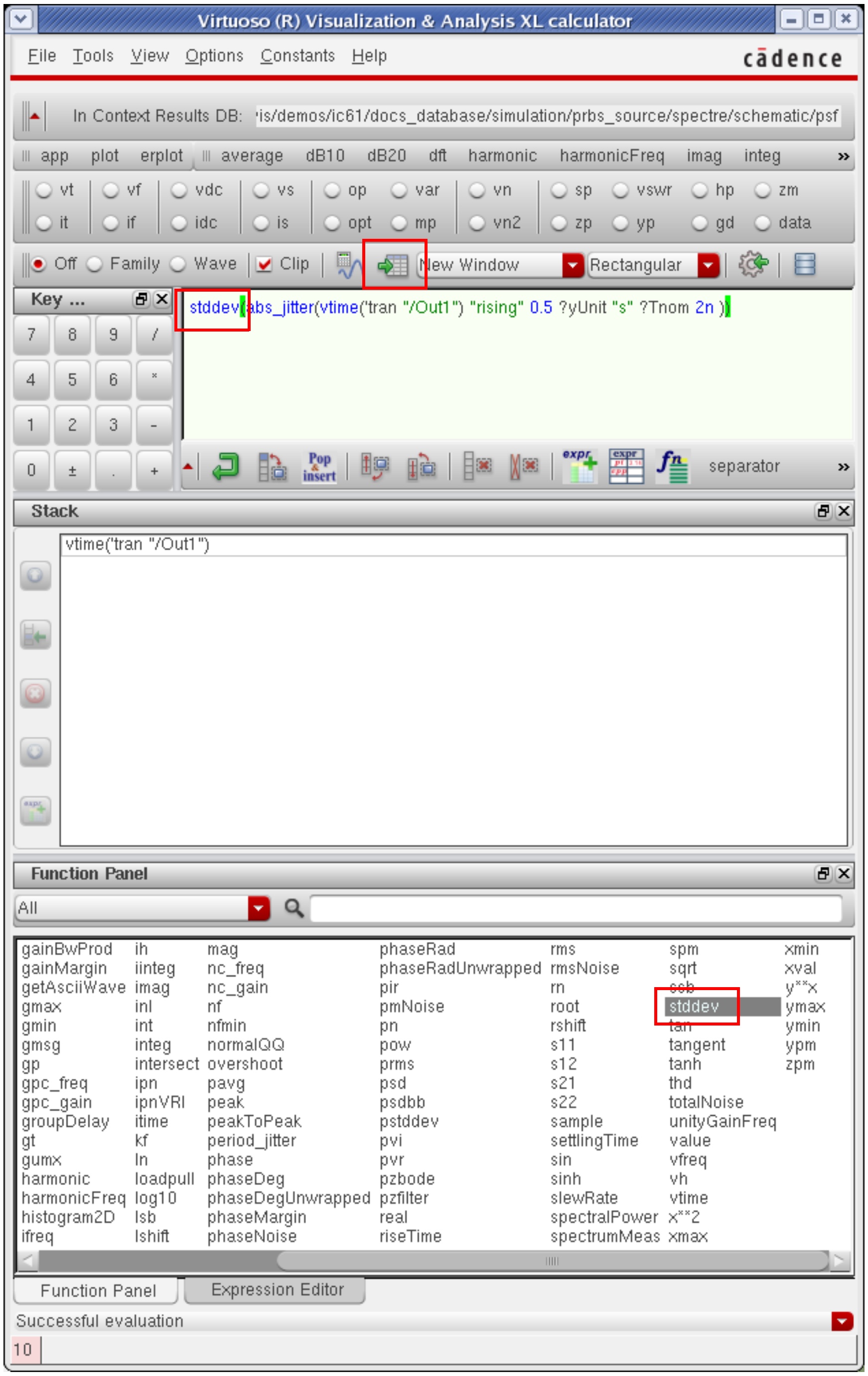


Click *OK*.

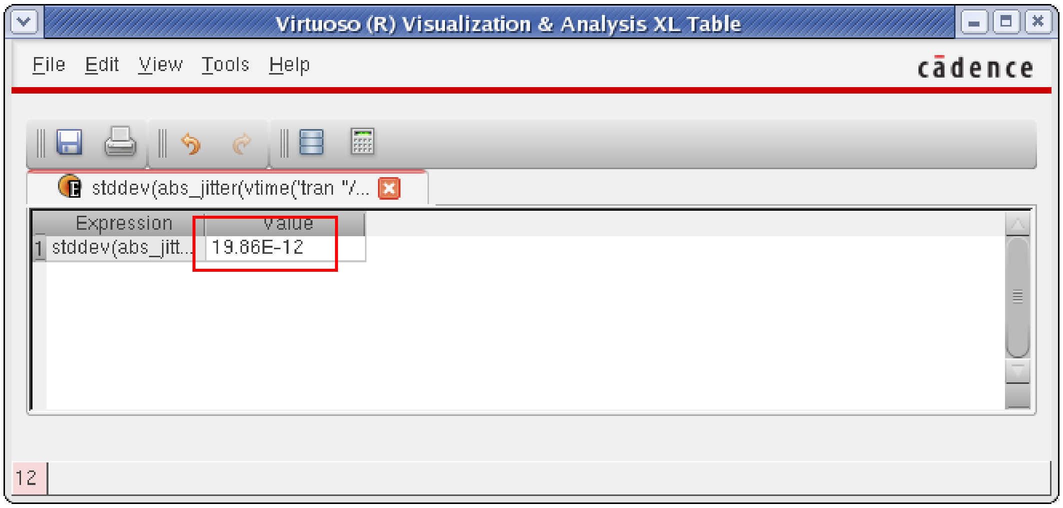
Now, plot the signal. The jitter plots, as shown below.



Now add the *stddev* function in the calculator. Send the result to a table.



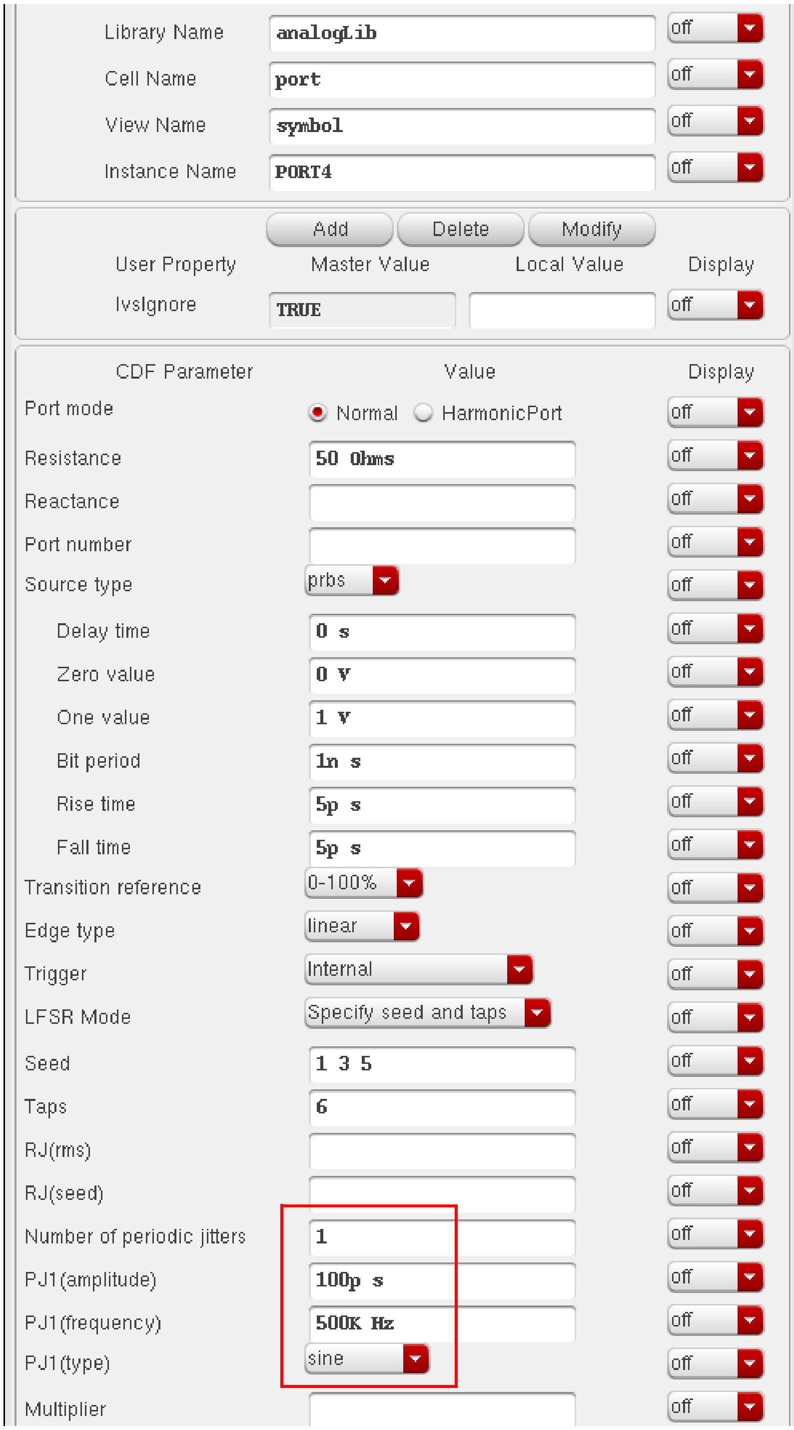
The result is very close to the 20psec setting in the `prbs` source.



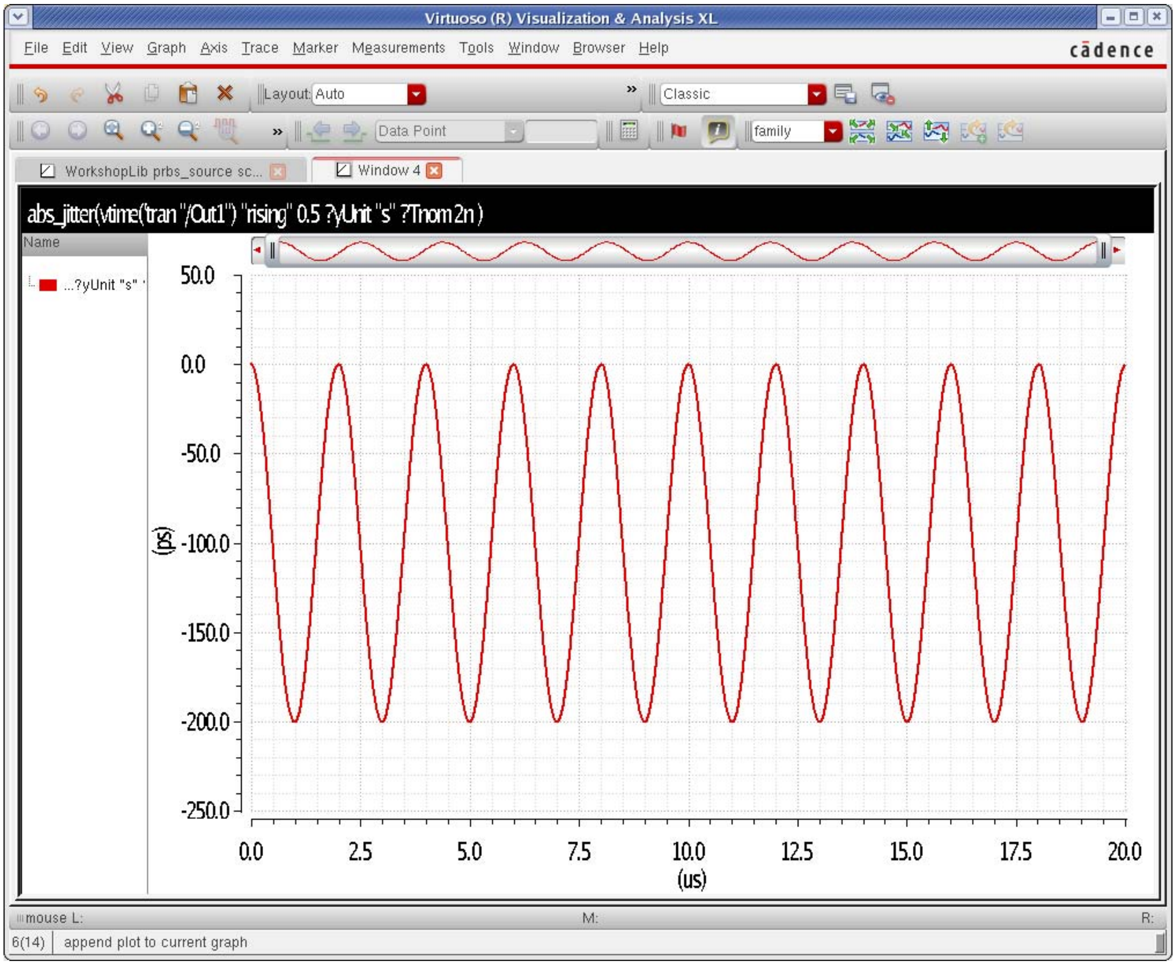
Periodic Jitter

Periodic jitter can be generated in the `prbs` source by setting the number of periodic jitter sources, choosing the waveshape for each jitter source, setting the frequency of each periodic jitter source, and by defining the delta-T (peak) for each jitter source in the

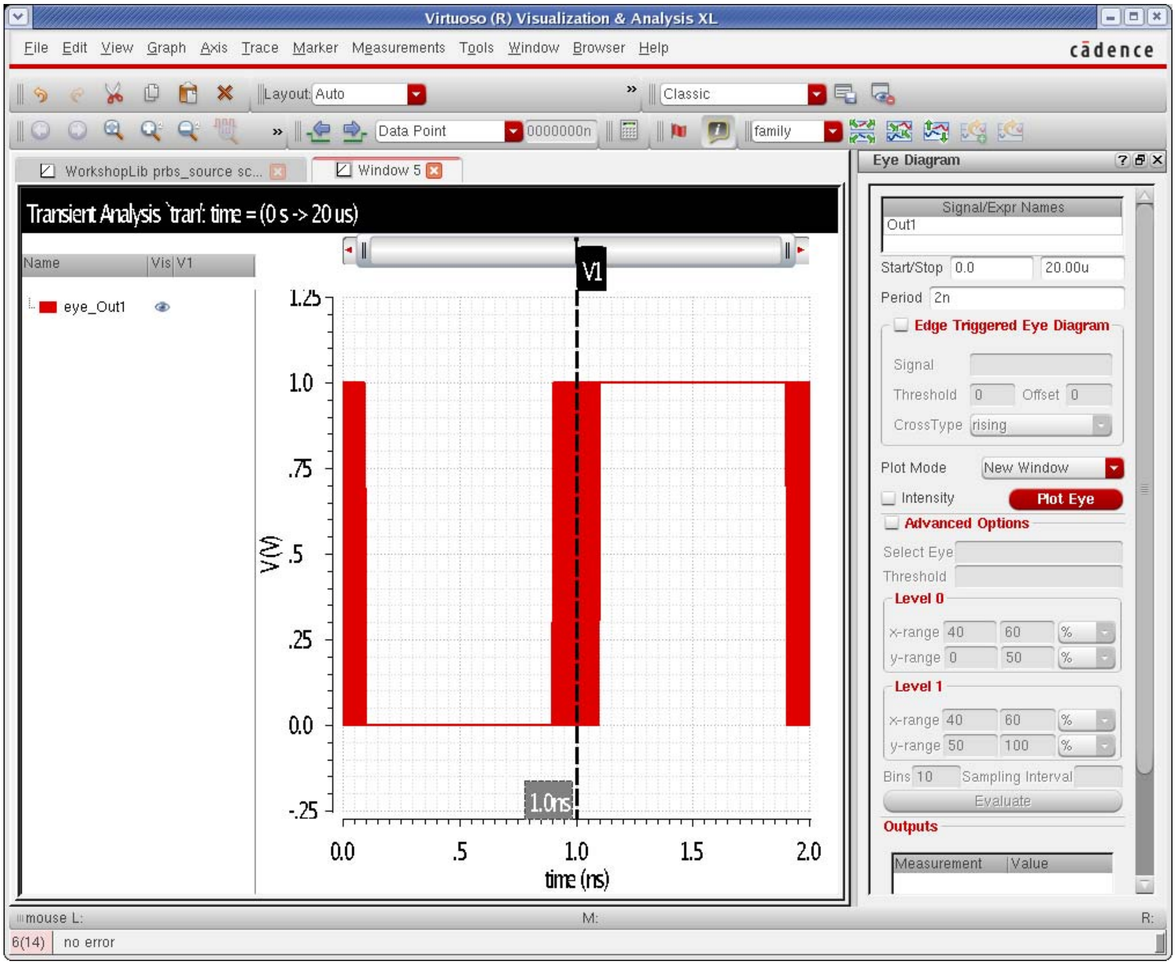
PJ(amplitude) field. Jitter waveforms can be sinusoidal, triangular, or square wave. Below is an example of a single periodic sinusoidal jitter source at 500KHz with 100psec peak jitter.



Run in the transient, and then send the waveform to the calculator. Apply the *abs_jitter* function as before. The jitter waveform is shown below.

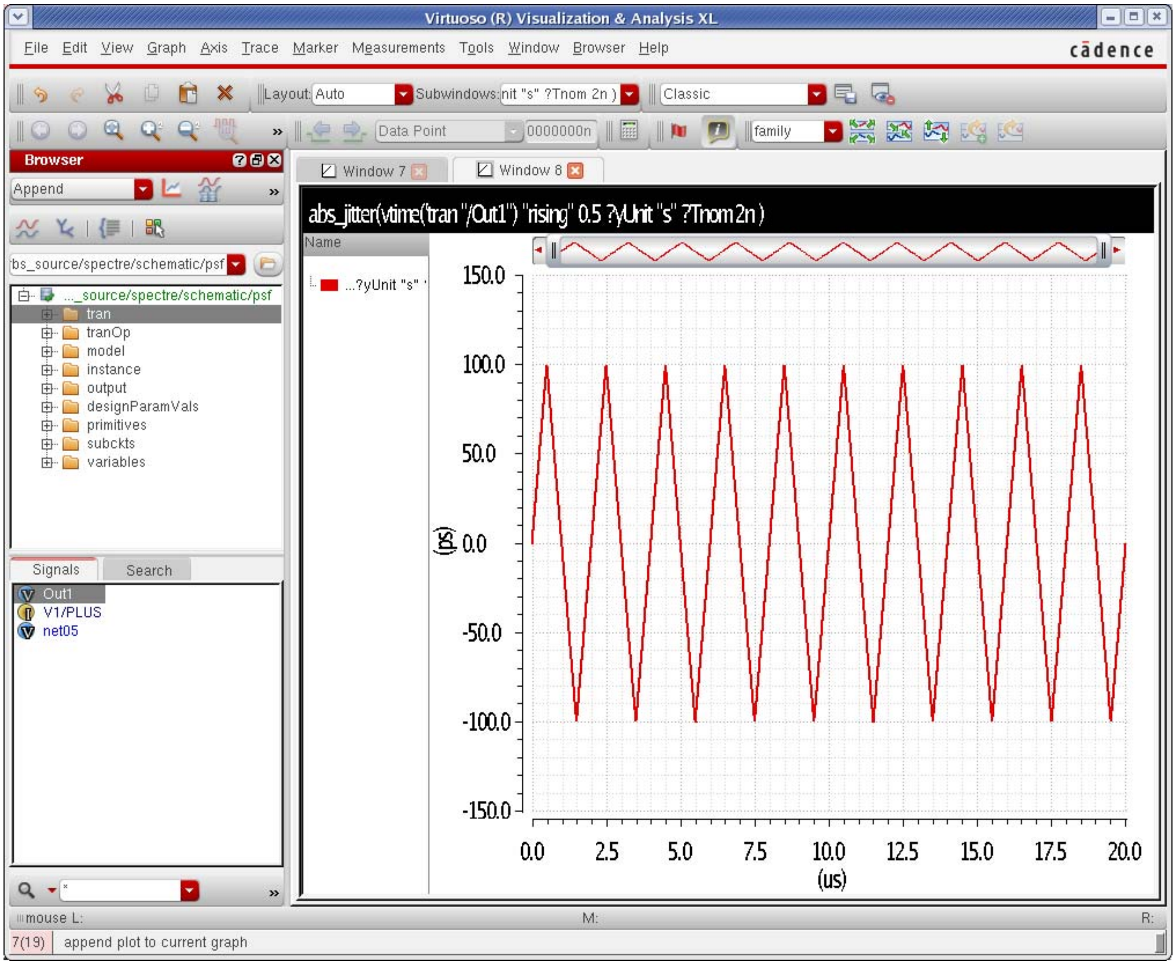


The curve seems to indicate only the negative jitter. An eye diagram was plotted to check the actual timing of the edges.

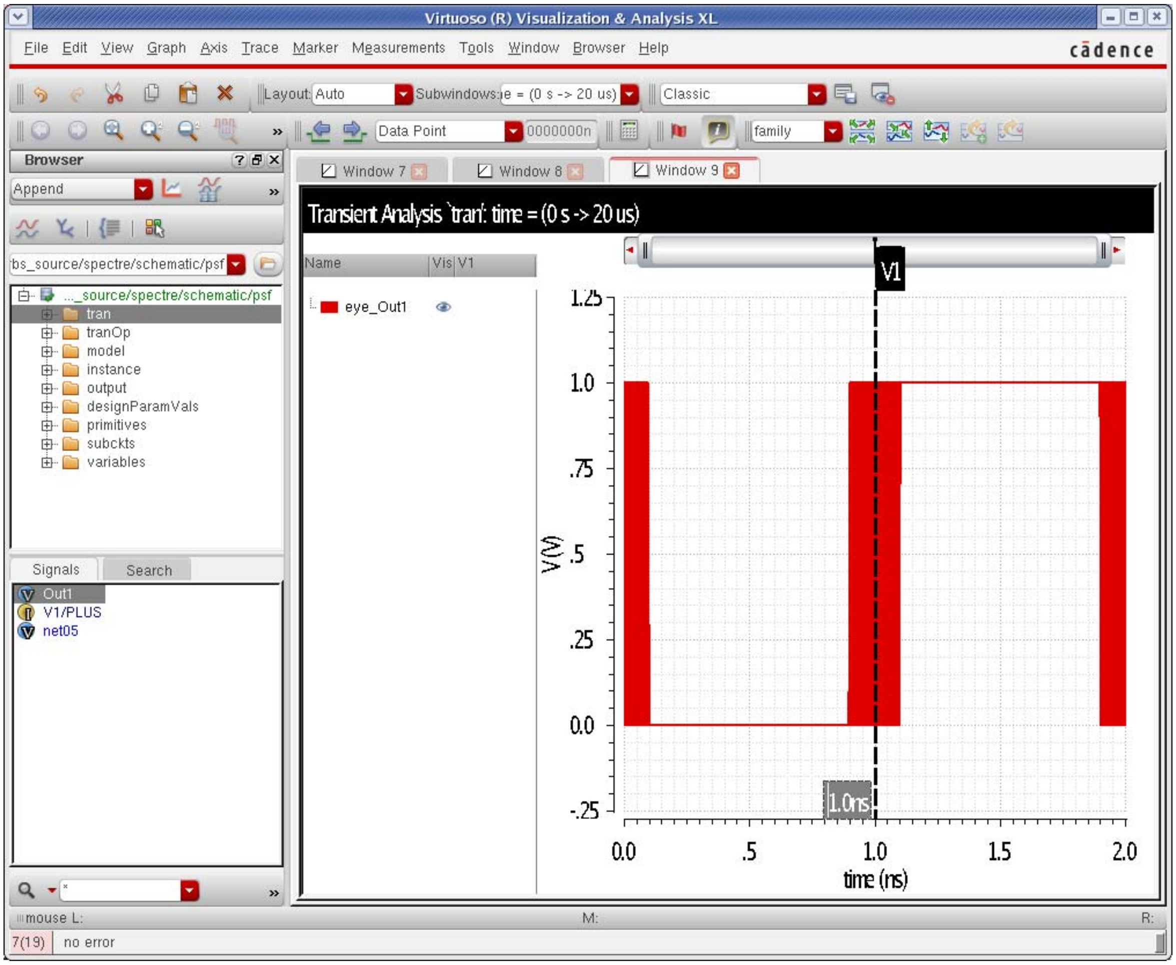


Note that the jitter is actually centered at 1nsec, which is the timing of the noise-free waveform. The *abs_jitter* function takes the time of the first transition of the time-domain waveform as the center time of the jitter calculation. As a result, the center value can be skewed if the timing of the first edge is not noise free.

Below is an example of a triangle wave with 100psec peak jitter.

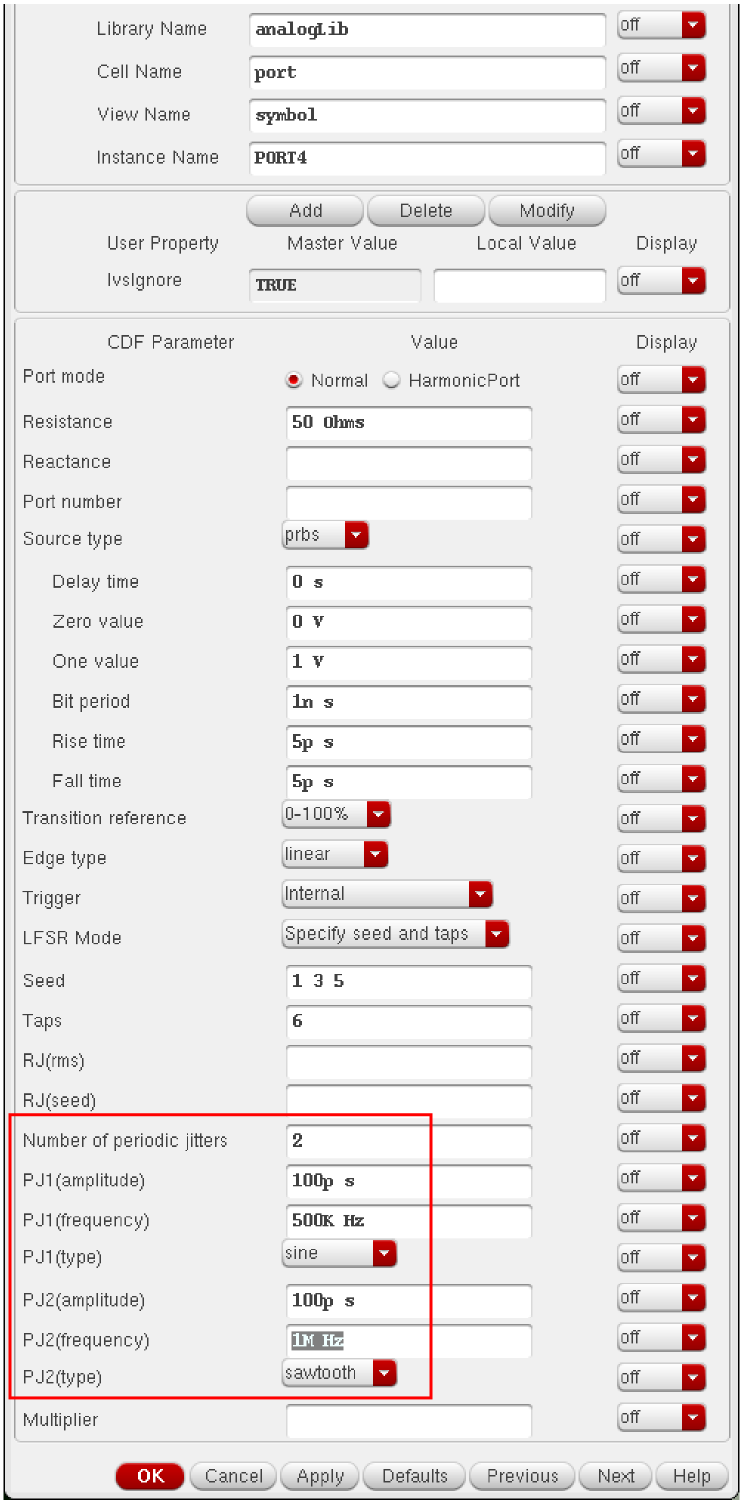


Again, with an eye diagram, the transitions are centered around 1nsec.

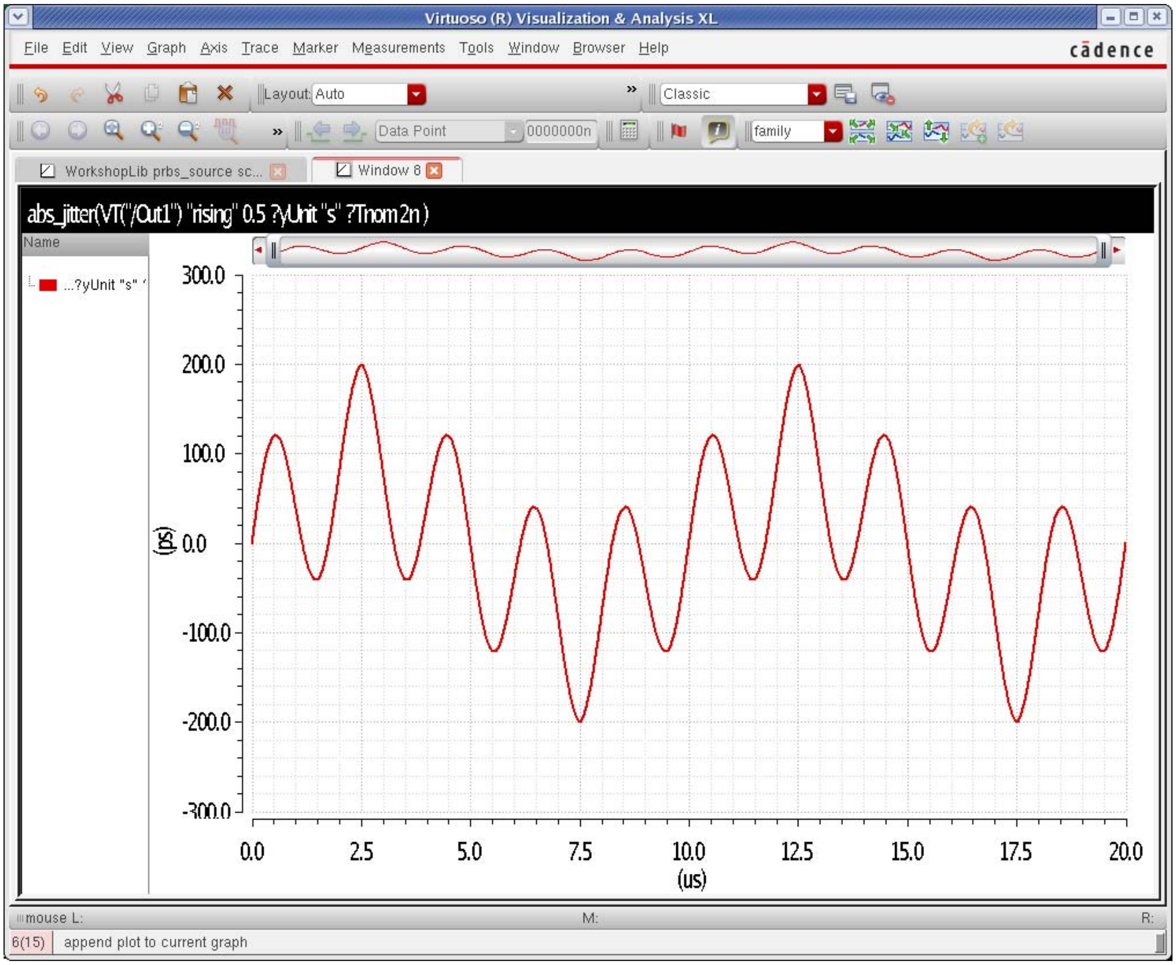


Two Periodic Jitter sources at the Same Time

Up to three periodic jitter sources can be set at the same time. Specify the number of periodic sources you want, and define the jitter parameters for each jitter source. An example with two jitter sources is shown below.

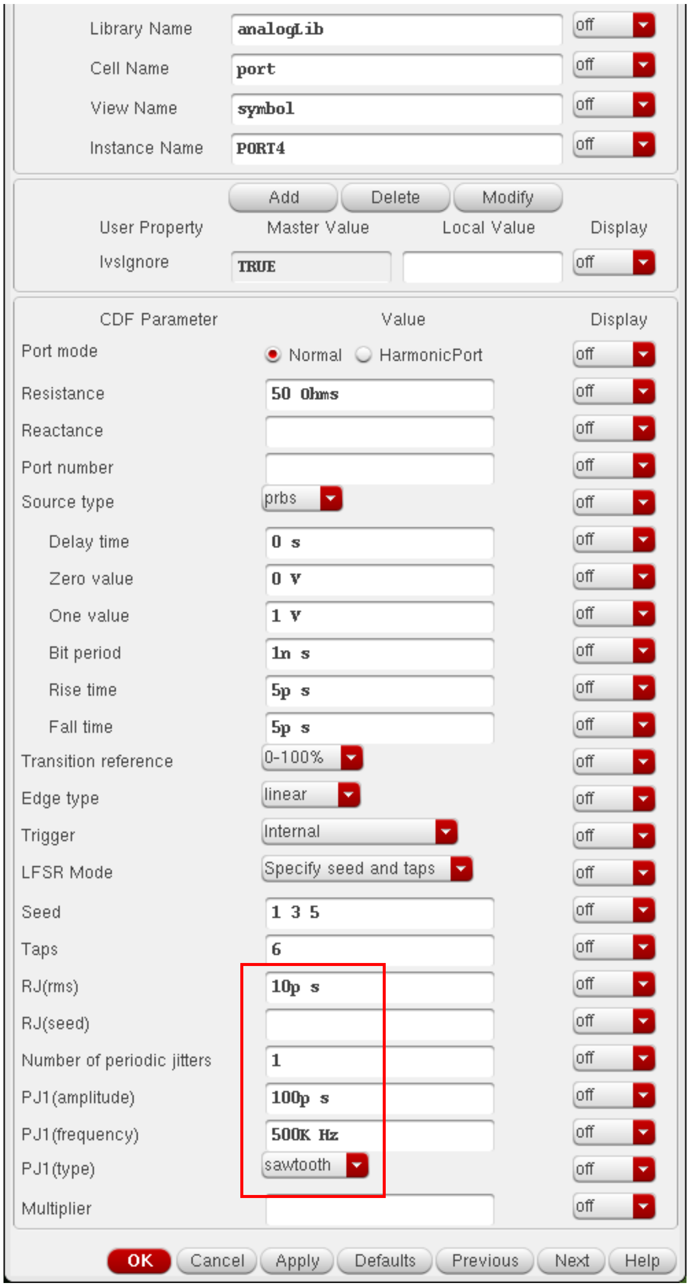

Library Name
analogLib
Cell Name
port
View Name
symbol
Instance Name
PORT4
Add
Delete
Modify
User Property
Master Value
Local Value
Display
lvsIgnore
TRUE
CDF Parameter
Value
Display
Port mode
Normal
HarmonicPort
Resistance
50 Ohms
Reactance
Port number
Source type
prbs
Delay time
0 s
Zero value
0 V
One value
1 V
Bit period
1n s
Rise time
5p s
Fall time
5p s
Transition reference
0-100%
Edge type
linear
Trigger
Internal
LFSR Mode
Specify seed and taps
Seed
1 3 5
Taps
6
RJ(rms)
RJ(seed)
Number of periodic jitters
2
PJ1(amplitude)
100p s
PJ1(frequency)
500K Hz
PJ1(type)
sine
PJ2(amplitude)
100p s
PJ2(frequency)
1M Hz
PJ2(type)
sawtooth
Multiplier
off
OK
Cancel
Apply
Defaults
Previous
Next
Help

The resulting abs_jitter plot is shown below. Both jitter sources are in the output.



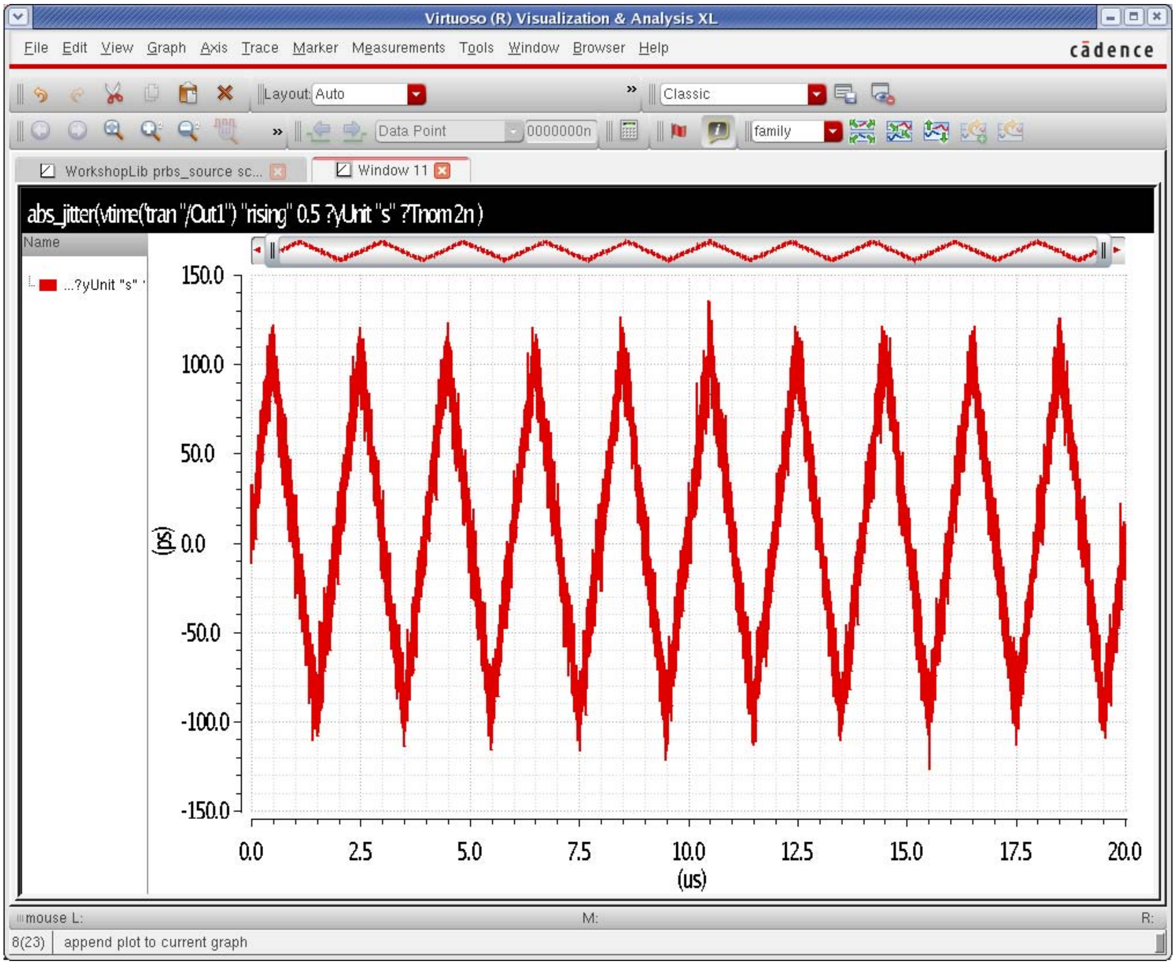
Random and Periodic Jitter at the Same Time

Both random and periodic jitter can be added at the same time. Below is an example of setting both in the `prbs` source.



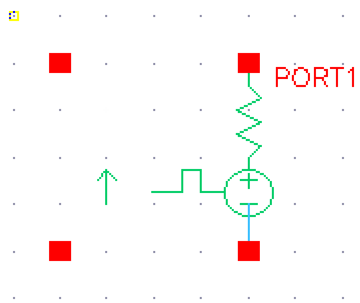
The *abs_jitter* plot shows both random and periodic jitter

.

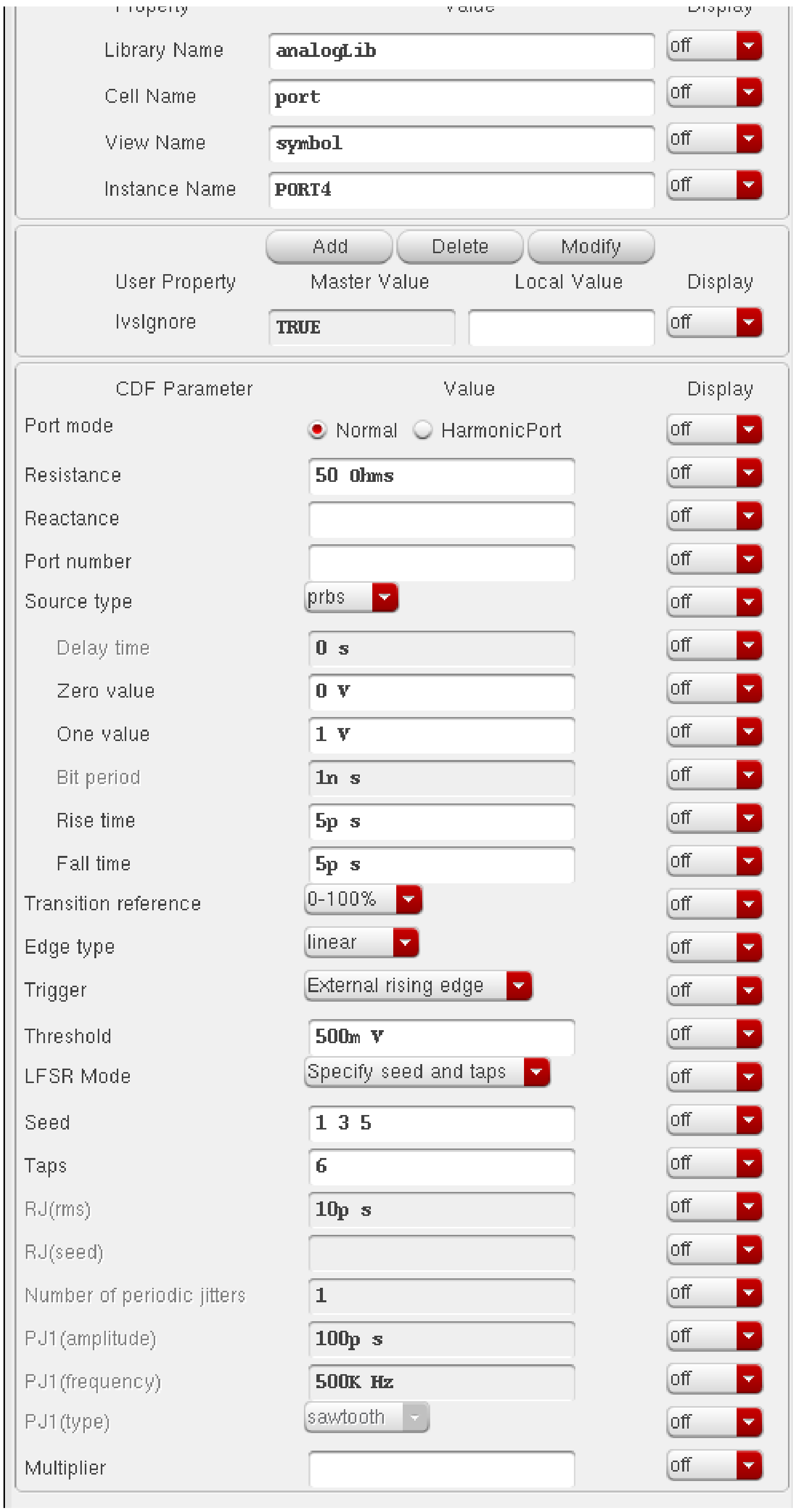


PRBS Mode External Triggering

When the `prbs` source is externally triggered, the schematic pcell redraws for the source, and two control terminals are added to the left of the source. This is shown below.

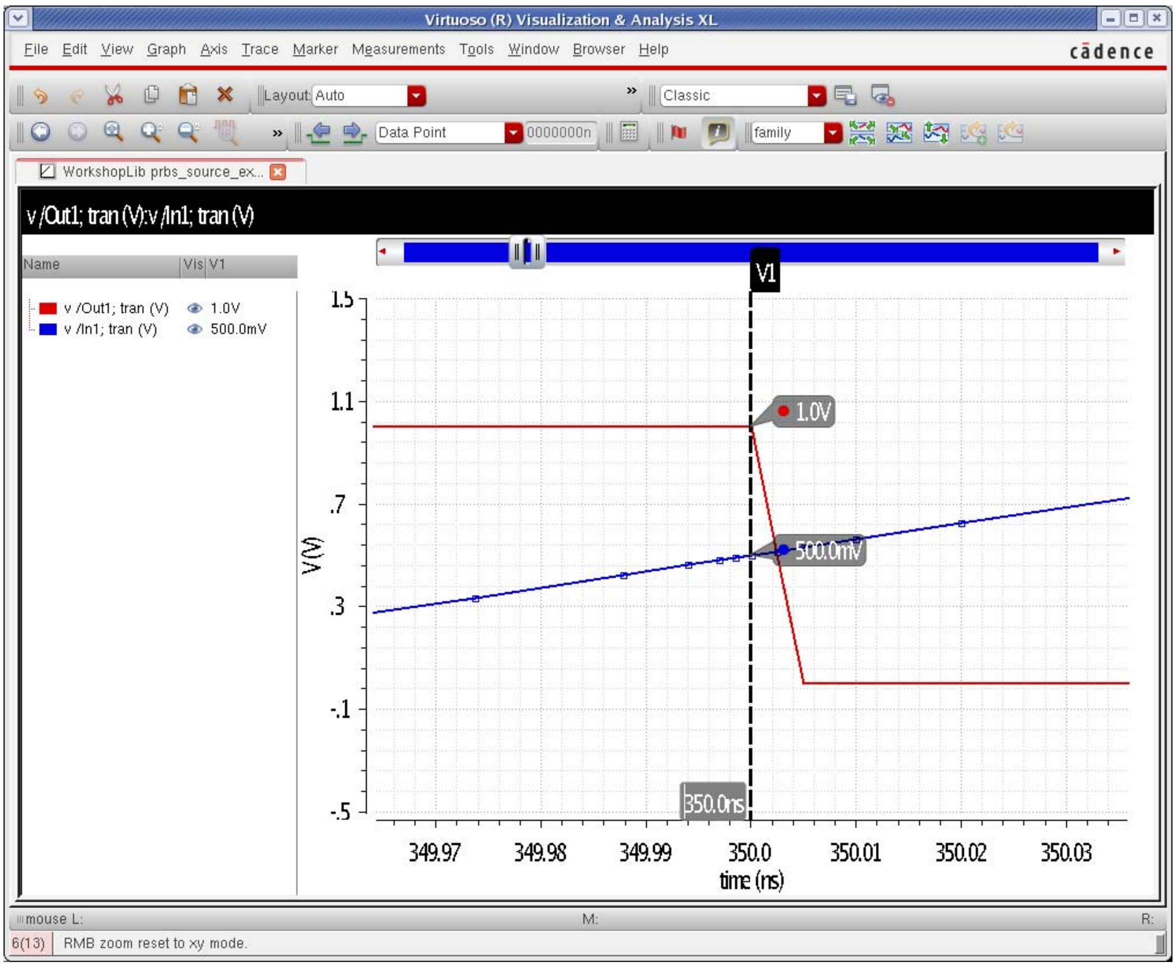


When the source is externally triggered, periodic and random noise generators are turned off. The properties form specifies the voltage level for the trigger and the edge mode for the trigger. In the example below, the trigger threshold is 0.5 Volts and the trigger is on the rising edge.

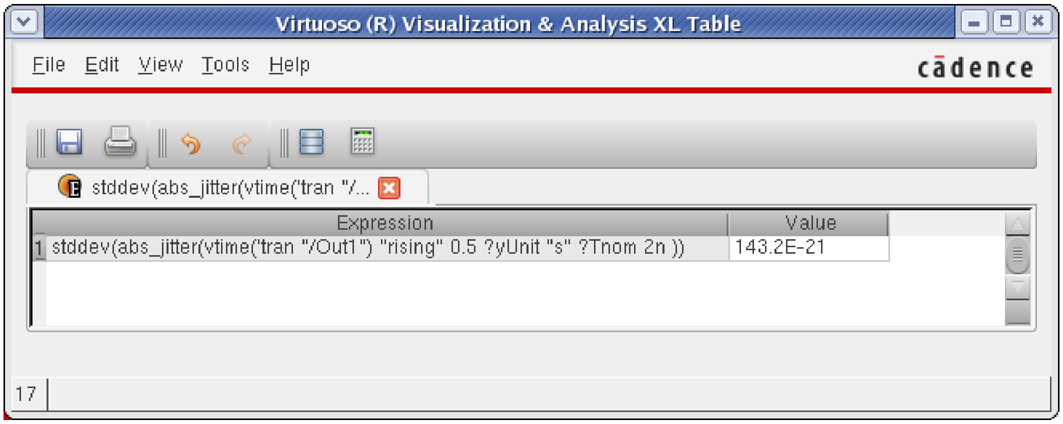


Note that the jitter terms are greyed out, indicating that the jitter is not available in the external trigger mode.

When the threshold is reached on the input waveform, the output begins the transition to the opposite state. The timestep is actively controlled in the region of the trigger threshold to make the signal trigger very accurately. In the waveform window below, the blue trace is the input, and the trace symbols are on and mark the timesteps actually used in the simulation. As the threshold is reached, the timestep is actively managed so that the crossing is accurately detected.



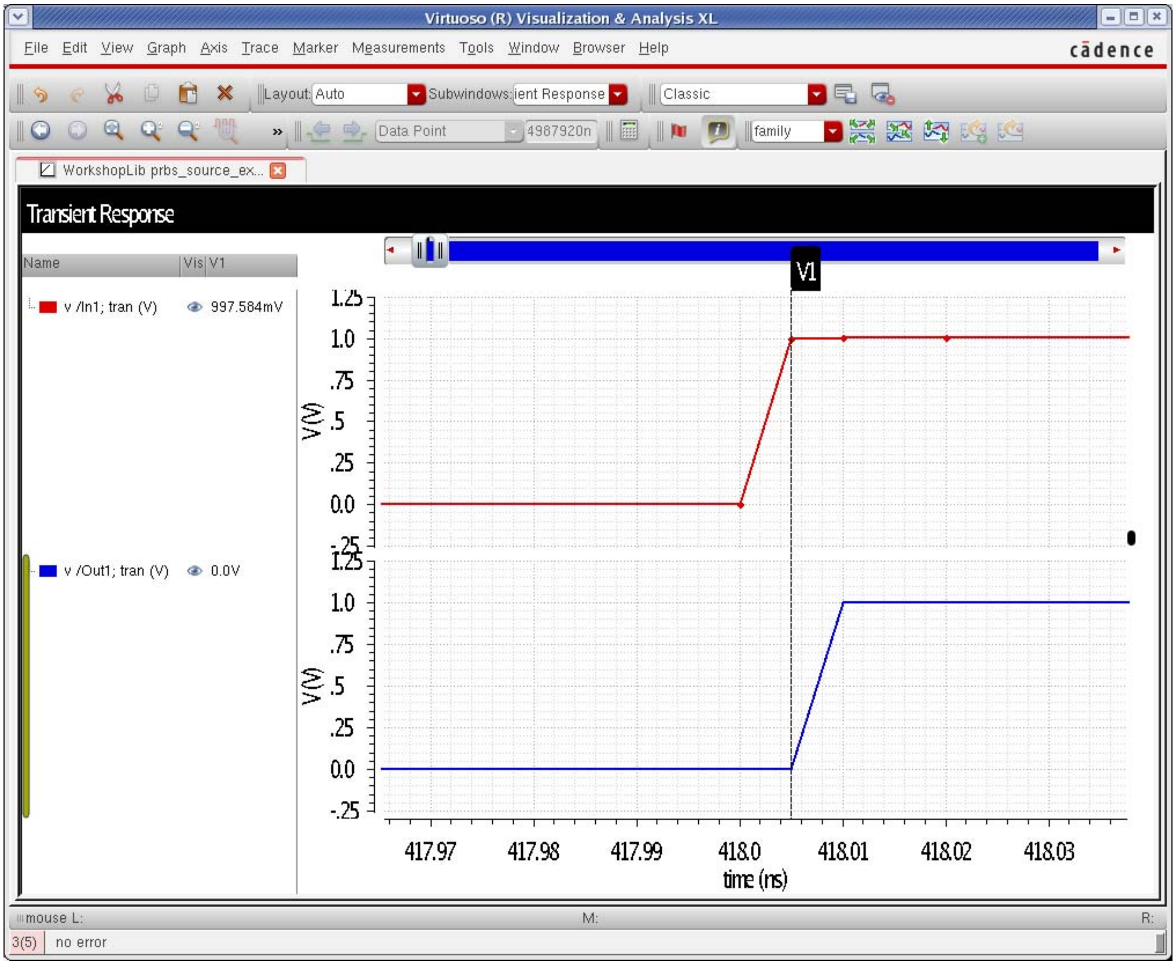
The jitter is also very small, as seen below.



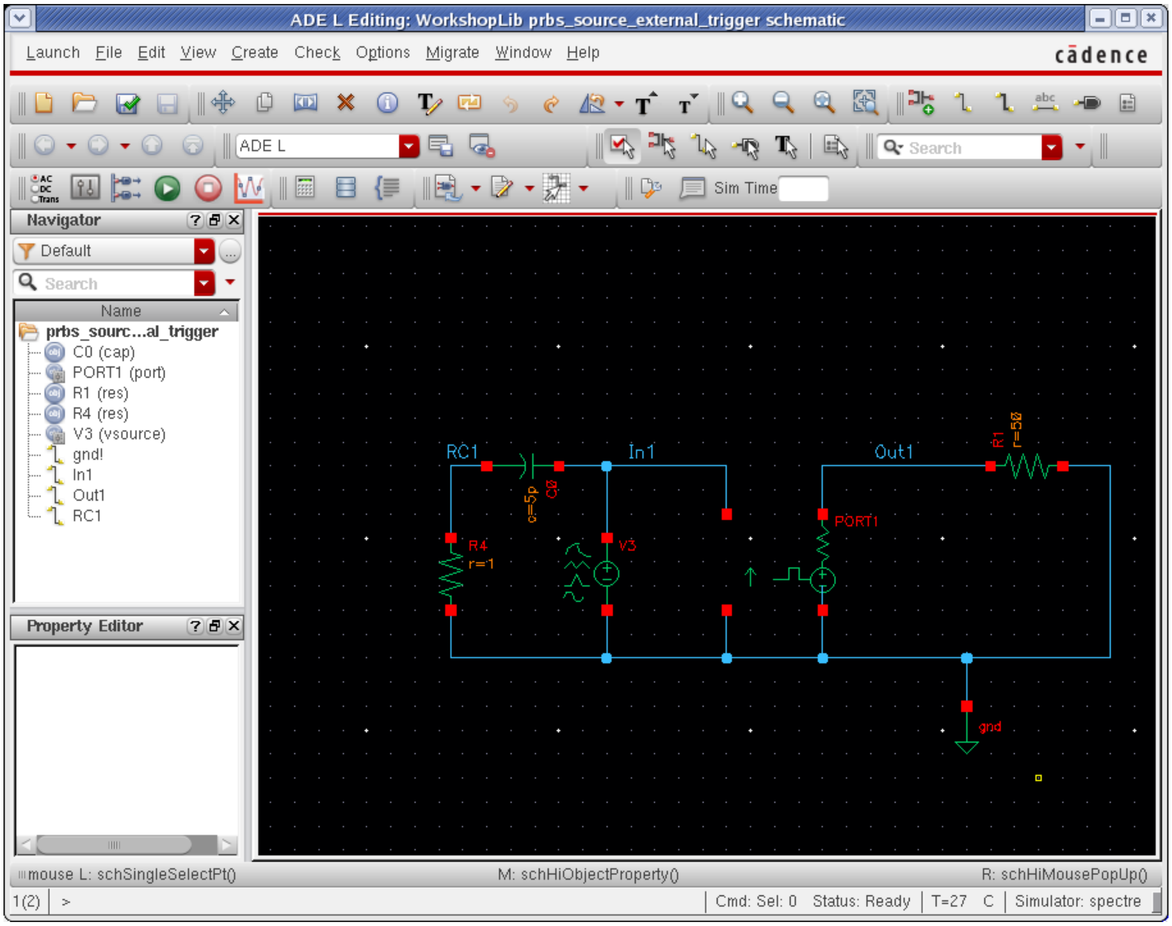
The calculator expression is shown on the left, and this is the same expression used before in this section to calculate the jitter.

PRBS Source with External Triggering Using a Pulse or PWL Signal

The external triggering capability of the `prbs` source depends on being able to calculate the slope of the input signal in order to accurately trigger at the specified voltage. The pulse and pwl sources guarantee timepoints at the corners of the waveforms, but they do not guarantee timepoints during the transition, especially if the transition is short. In that case, the prbs will trigger on the timepoint on the end of the transition as shown below.



The threshold for the trigger is 0.5 volts, and the input signal is the red trace. The actual timepoints of the input (red trace) are turned on, and the output starts its transition when the input is at 1 volt instead of at 0.5 Volts. This adds a slight delay in the prbs output, which in many cases is not significant. If you need to trigger on the actual threshold, then add a resistor and capacitor across the source as shown below. Set the R*C product to be the risetime of the input signal.



Adding these components adds numerical integration, and the error of this integration is controlled by Spectre by adding appropriate timepoints. This establishes a slope of the input signal so the trigger event can happen at the desired threshold of 0.5 volts in this example as shown below. The green trace below is the junction of the resistor and capacitor that was added to the circuit. Again, the timepoints are turned on for the input trace.

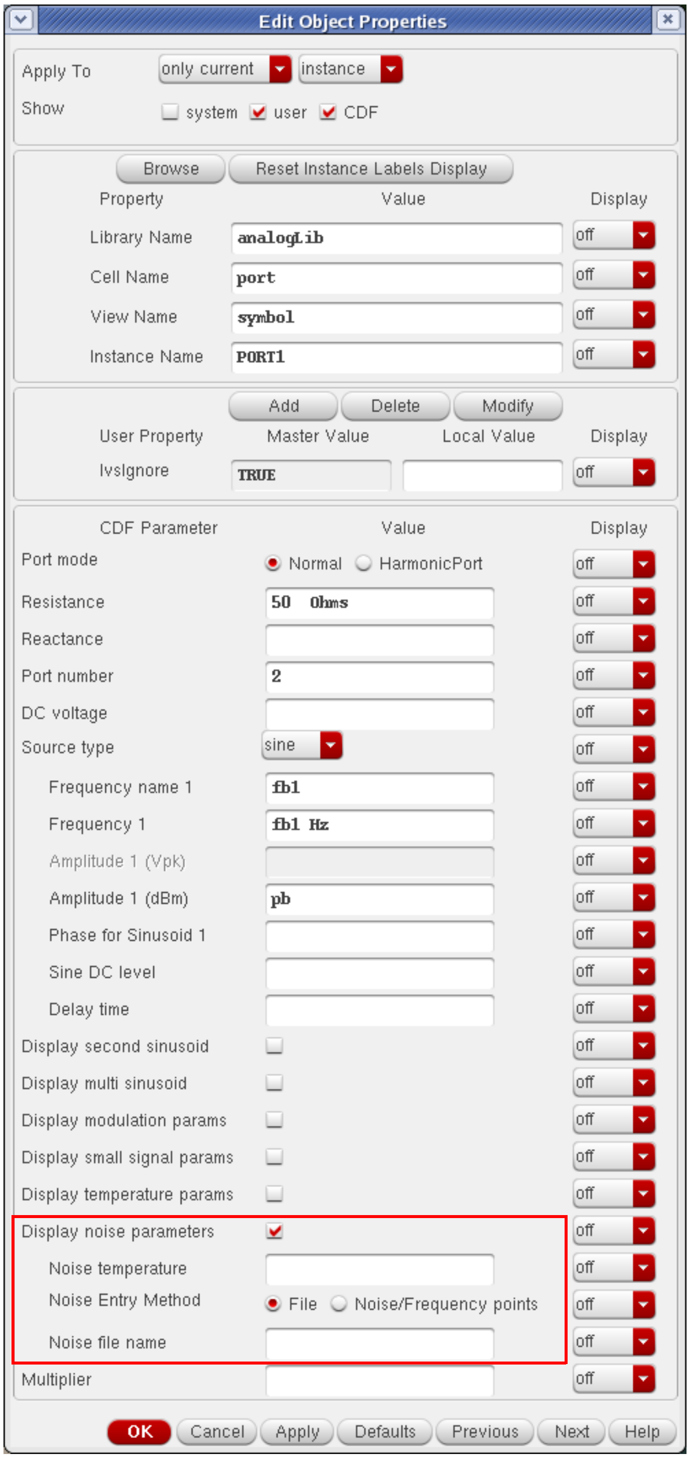
Keep the R*C product equal to the risetime of the pulse source or slightly less so the timestep is not excessively reduced for a long period of time.

Noise Parameters

Important

The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

The noise parameters include *Noise temperature*, *Noise Entry Method*, *Noise file name*, and *Noise/Frequency points*.



Noise temperature

The *Noise temperature* of the *port* resistor in degrees Celsius. If this is not specified, the *Noise temperature* is assumed to be the temperature that is set by the temperature option. When you compute the noise figure of a circuit driven at its input by a *port*, set the *Noise temperature* of the *port* to 16.85C (290K). This setting matches the standard IEEE definition of noise figure. In addition, disable all other sources of noise in the *port*, such as the Spectre parameters *Noise file name* and eliminate any entries you have in the fields that appear when you select *Noise/Frequency points*. If you want a noiseless *port*, set the *Noise temperature* to absolute zero or below, and do not specify a noise file or Noise/Frequency points. Absolute zero is -273C. The default noise temperature is 290C.

Noise Entry Method

You can select one of the two ways to enter noise data, either by specifying a *File* name or entering a series of *Noise/Frequency points*.

Important

Entering noise points define AM noise only. There is no mechanism in the port for including PM noise from a noise file.

Noise file name

If *Noise Entry Method = File* is selected, you enter the name of the file containing the noise data in the form of frequency-noise pairs. In your file, list the frequency-noise pairs as one pair per line with a space or tab between the frequency and noise values. The first entry is the frequency, and the second value is volts-squared per Hertz.

Important

Frequencies outside the range specified in the noise file are interpolated, and not zero. If you want zero, make sure that you specify zero over the range you want. Remember that in pnoise, a wide range of noise frequencies are typically evaluated, and this interpolation can cause more noise than expected in the analysis. If you want to see what the source is actually generating, make a circuit with a port and a terminating resistor, and run linear noise over a very wide frequency range. Then plot the noise at the output of the port.

Important

If you want just the noise specified in the file and nothing more, set the noise temperature of the port to -273. The default is to generate the noise from the file and add to it the noise from the resistor in the port. Setting the noise temperature to -273 makes the noise fro the resistor be zero.

Figure 8-13 Display noise parameters: Noise Entry Method=File

Important

The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

Important

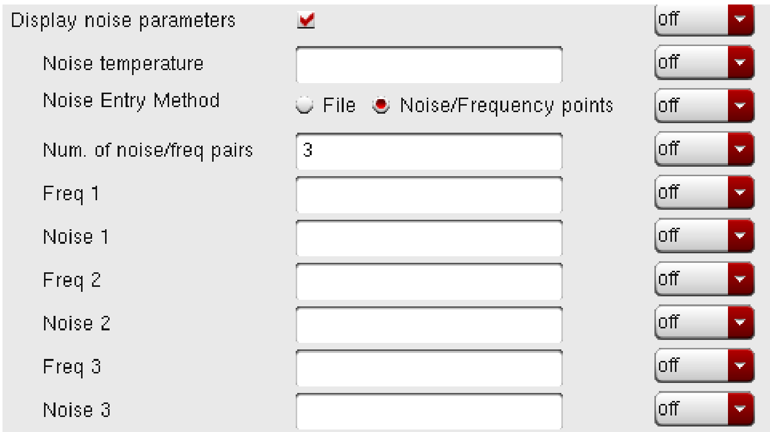
If you want just the noise specified in the noise file and nothing more, set the noise temperature of the port to -273. The default is to generate the noise from the file and add to it the noise from the resistor in the port. Setting the noise temperature to -273 makes the noise from the resistor be zero.

Num. of noise/freq pairs

If the *Noise Entry Method* is set to *Noise/Frequency points*, you specify the *Number of noise/freq pairs*. The form expands to let you type in the designated *Freq* and *Noise* values. The noise values must be in volts-squared per Hz, and frequency in Hz. The maximum number of points in the form is 10.

The example below has the *Number of noise/freq pairs* set to 3.

Figure 8-14 Display noise parameters: Noise Entry Method=Noise/Frequency Points



Important

If you want just the noise specified in the *Num. of noise/freq pairs* properties and nothing more, set the noise temperature of the port to -273. The default is to generate the noise from the noise properties and add to it the noise from the resistor in the port. Setting the noise temperature to -273 makes the noise from the resistor be zero.

Small-Signal Parameters

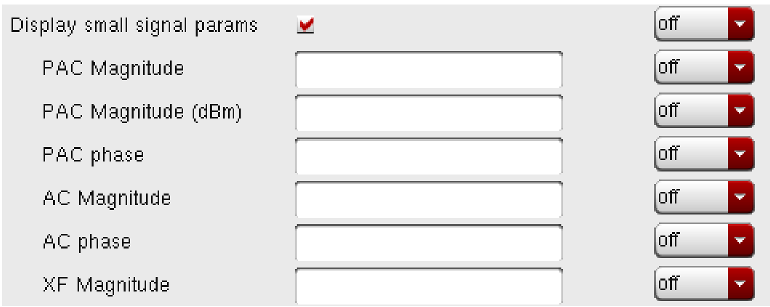
Display small signal params

Important

The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation.

If selected, the *Edit Object Properties/Add Instance* form expands to show the small-signal properties *PAC Magnitude*, *PAC Magnitude (dBm)*, *PAC phase*, *AC Magnitude*, *AC phase*, and *XF Magnitude*.

Figure 8-15 Display small signal params



When you specify the voltage on a *port*, you are specifying the voltage when the port is properly terminated, and not the voltage on the internal voltage source. Thus, the voltage on the internal source is set to twice the value specified on the *port*. The same is true for the values for the transient, AC, and PAC signals. However, the amplitude of the sine wave in the PAC and transient analysis can alternatively be specified as the power in dBm delivered by the `port` when terminated with the reference resistance. The *PAC magnitude* property is used by pac, hbac, and qpac. The *XF Magnitude* property is used by xf, pxf, hbxf, and qpxf.

PAC Magnitude

The peak periodic AC analysis magnitude. Setting this value to unity is a convenient way of computing the transfer function from this source to the output.

You can select either *PAC magnitude* or *PAC magnitude (dBm)*, but not both. If *PAC magnitude* has a value, the *PAC magnitude (dBm)* field is grayed out. The PAC magnitude property is used by pac, hbac, and qpac.

PAC Magnitude (dBm)

The periodic AC analysis magnitude in dBm (alternative to *PAC magnitude*). You can select either *PAC magnitude* or *PAC magnitude (dBm)*, but not both. If *PAC magnitude (dBm)* has a value, the *PAC magnitude* field is grayed out. The *PAC magnitude(dBm)* property is used by pac, hbac, and qpac.

PAC phase

The periodic AC analysis phase. The value must be a real number. The default value is 0 and the units is in degrees.

Typically, only one source in the circuit has a *PAC magnitude* set to a value other than zero, and many times it has a *PAC magnitude*=1 and *PAC phase*=0. However, there are situations where more than one source has a nonzero *PAC magnitude*. For example, applying a differential small-signal input could be done with two sources with the *PAC magnitude*s set to 0.5 and the *PAC phase*s set to 0 and 180. The *PAC Phase* property is used by pac, hbac, and qpac.

You do not specify the PAC frequency in the port *Edit Object Properties* form. Instead, you set the PAC frequency in the PAC *Choosing Analyses form*. For example, when making an IP3 measurement, you set the PAC frequency to a variable value in the *Choosing Analyses* form. Then, you enter the same variable in the *PAC Amplitude* (or *PAC Amplitude dBm*) field of the *port Edit Object Properties* form.

AC Magnitude

The peak small-signal voltage for the ac analysis in volts.

AC phase

The ac small-signal phase in degrees.

Typically, only one source in the circuit has *AC Magnitude* set to a value other than zero, and many times it has an *AC magnitude*=1 and *AC phase*=0. However, there are situations where more than one source has a non-zero *AC magnitude*. For example, you can apply a differential small-signal input with two sources with the *AC magnitude*s set to 0.5 and the *AC phase*s set to 0 and 180.

XF Magnitude

This property is seldom used. It specifies the transfer function analysis magnitude. This is used by xf, pxf, hbxf, and qpxf.

Temperature Effect Parameters

Temperature effect parameters include *Linear temp. coefficient*, *Quadratic temp. coefficient*, and *Nominal temperature*.

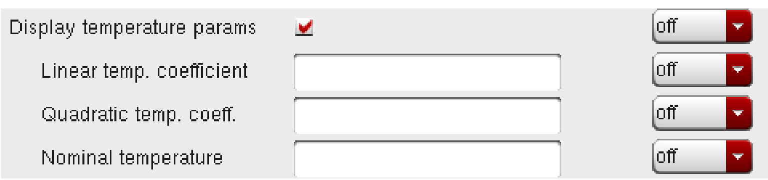
Display temperature params

Important

The check box is a display feature in the form only. If there are properties that are set and the display check box is not selected, although you will not see the parameters in the form, these parameters will still be netlisted, and will still have an effect on the simulation

If the check box is selected, the *Edit Object Properties/Add Instance* form expands and the following three parameters appear in the form: *Linear temp. coefficient, Quadratic temp. coeff.,* and *Nominal temperature*.

Figure 8-16 Display temperature params



Linear temp. coefficient

This property sets the first order (linear) temperature coefficient of all the voltages specified for any waveform in the port. 0.01 entered in the *Linear temp. coefficient* field is interpreted as a one percent change per degree Celsius. If the number is positive, the voltages increase with temperature. If the number is negative, the voltages decrease with temperature. If the temperature moves far enough, the values can become negative.

Quadratic temp. coeff.

This property sets the second order (quadratic) temperature coefficient of all the *voltages* the source generates. 0.01 causes a change in value of one percent per degree squared.

Nominal temperature

The *Nominal temperature* (tnom) for the linear and quadratic temperature coefficients.

Simulating Tabulated S-Parameters Using the Nport Component

Nport from analogLib provides a generic way to incorporate an S-Parameter file into a circuit simulation. The nport component allows you to set the number of ports, and when this is done, the schematic symbol re-draws with the number of ports shown. There is also n1port, n2port, n3port, and n4port in analogLib, which have a fixed number of ports. These components are provided for legacy purposes only, and should not be used in current simulations.

For the frequency-domain based analyses, such as ac, noise, xf, stb, pz, harmonic balance, hbac, hbxf, and hbnoise, the S-Parameter file is used directly in the frequency domain. For the DC and transient based simulations dc, tran, and shooting, a time-domain model must be created from the frequency-domain description in the S-Parameter file. This is a non-trivial process.

Note: Currently, the improvements to the nport component are restricted to transient analysis only.

In general, there are two different ways to create a time-domain model from an S-Parameter file. One way is to do a rational fit, which attempts to match the frequency domain description using a combination of poles, zeros, and gain blocks. The other way uses a convolution-based approach. This approach calculates the impulse response, and then creates a time-domain model based on the impulse response.

Regardless of the method, there are fundamental requirements for the S-Parameter file. These are:

- The file must contain a frequency point that is low enough so that when the DC point is extrapolated, the DC point correctly defines your circuit behavior.

  This is a fundamental requirement. To run a transient, first you need to run the DC analysis to capture the time-zero timepoint. If the first frequency in the file is not zero frequency, then a zero-frequency point must be extrapolated. If the first frequency in the S-Parameter file is not low enough to accurately represent the real DC characteristics of the system, the DC extrapolation will be incorrect.

- The phase of the DC point must be zero or 180 degrees.

- The frequency-domain data must be dense enough to accurately describe the transfer function.

  Smooth transfer functions without local complex behavior might need relatively few datapoints. Systems with complicated behavior need to have enough datapoints to capture the behavior well enough so that when the data is interpolated between the given points, and the interpolation error is small.

For bandpass filters, make sure that data from near zero Hz through at least the frequency of the highest harmonic in the simulation is present in the S-Parameter file.

- The frequency data must exist to a high enough frequency so that harmonics of the system are accurately simulated.

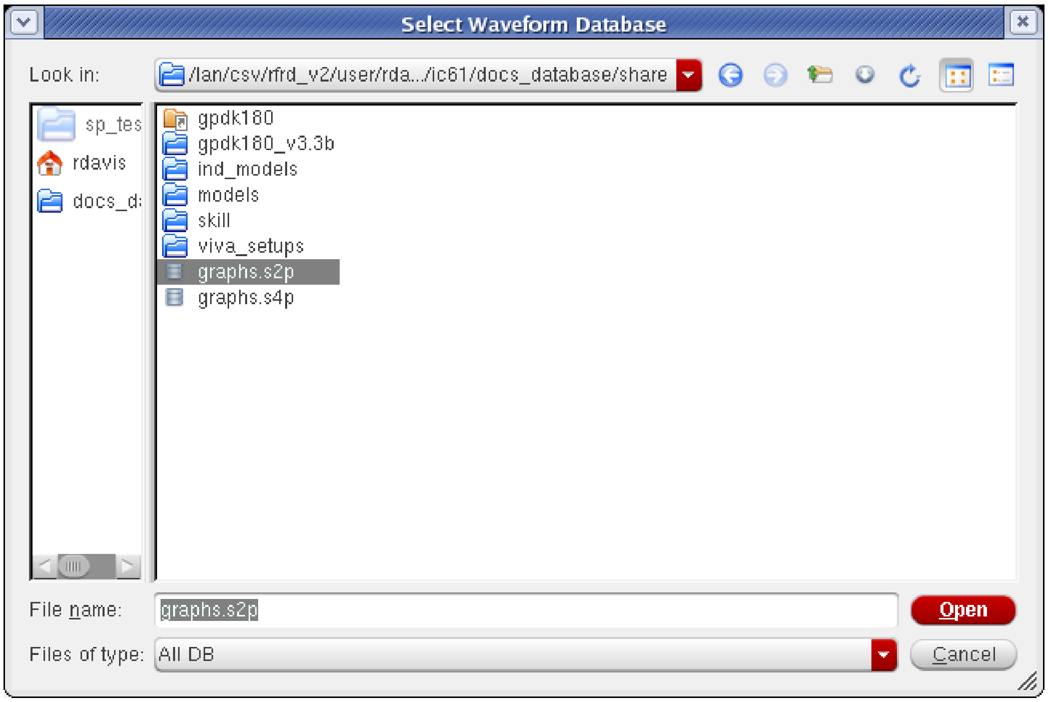
  If your S-Parameter file has data only through the second harmonic of the input signal and you are calculating six harmonics in a large-signal analysis, just note that the S-Parameter data above the second harmonic is extrapolated. The simulation is not based on real data, and as a result, it is different from the real system. If you want your simulation to be accurate for the harmonics you are simulating, make sure that the S-Parameters themselves go high enough in frequency.

- As a strong suggestion, regardless of the source of the data, plot the data from your S-Parameter file before you simulate. The plot for all the S-Parameters should plot in a clockwise arc with no sudden jumps. These discontinuities cause large inaccuracies in the transient model. Spectre now checks for discontinuities in the S-Parameter data.

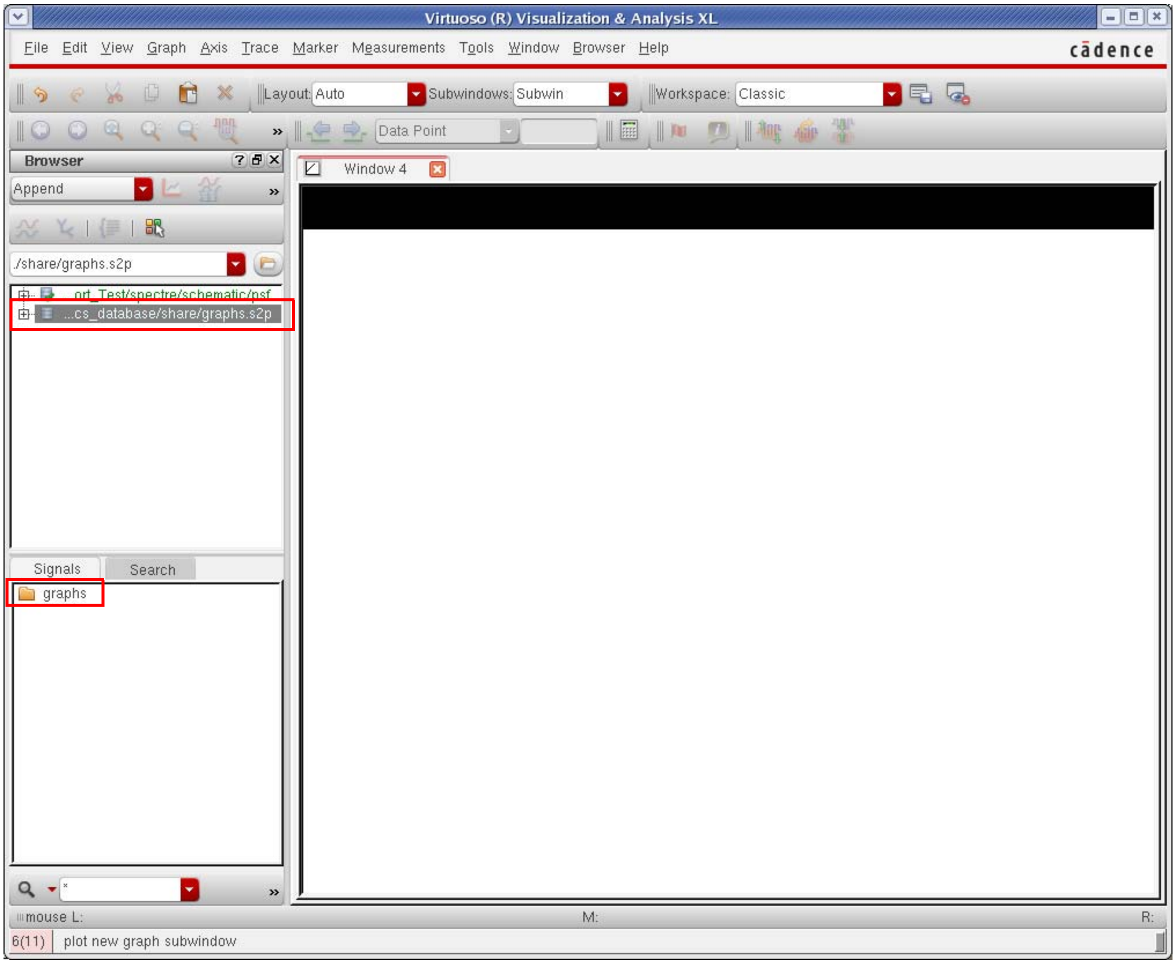
To plot an S-Parameter file directly in the waveform tool, the file format must be in Touchstone1 format. Here are the steps required to plot an S-Parameter form an S-Parameter file.

1. From the ADE window, select *Tools-Waveform*.

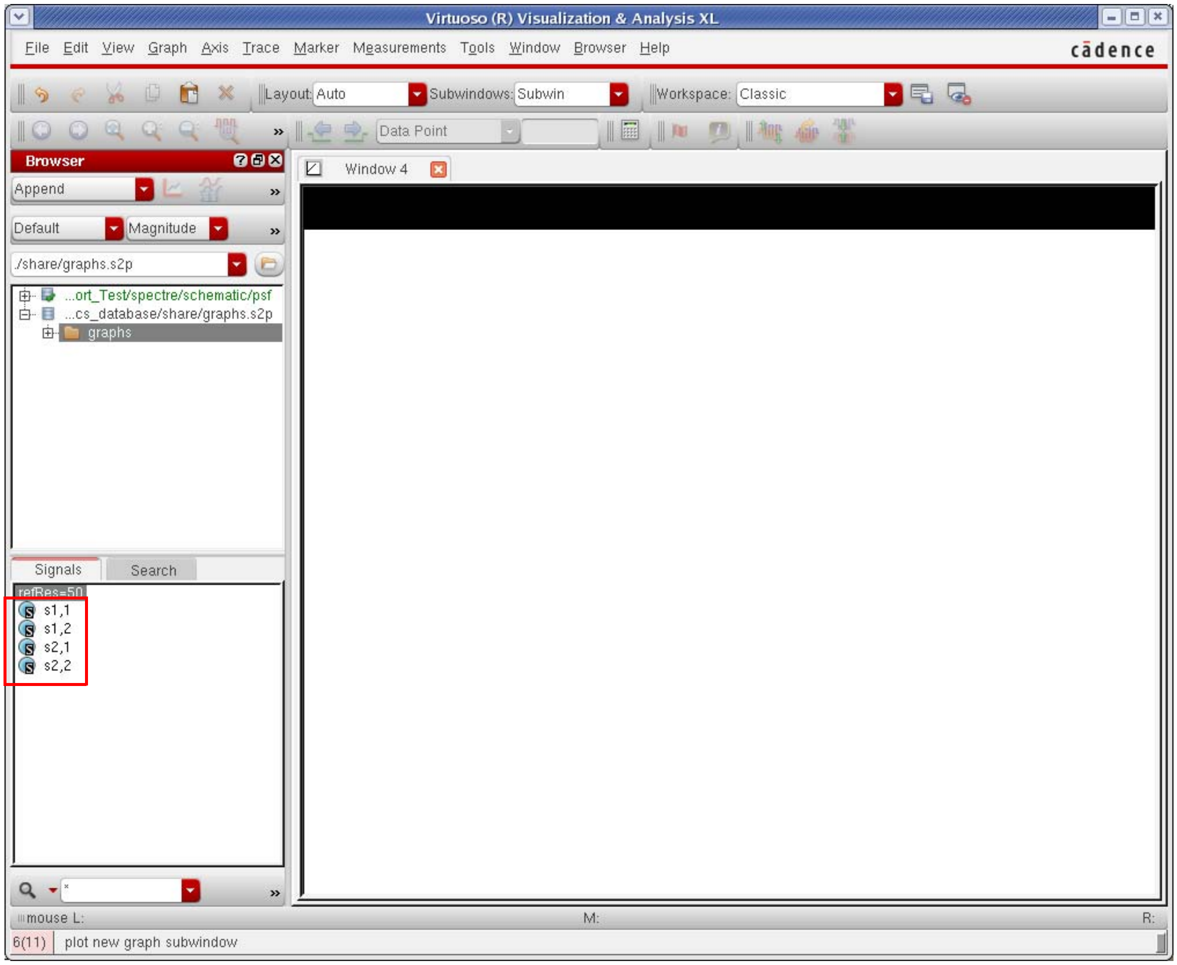
2. In the waveform tool, select *File-Open Results*, and browse to the location of the S-Parameter file.



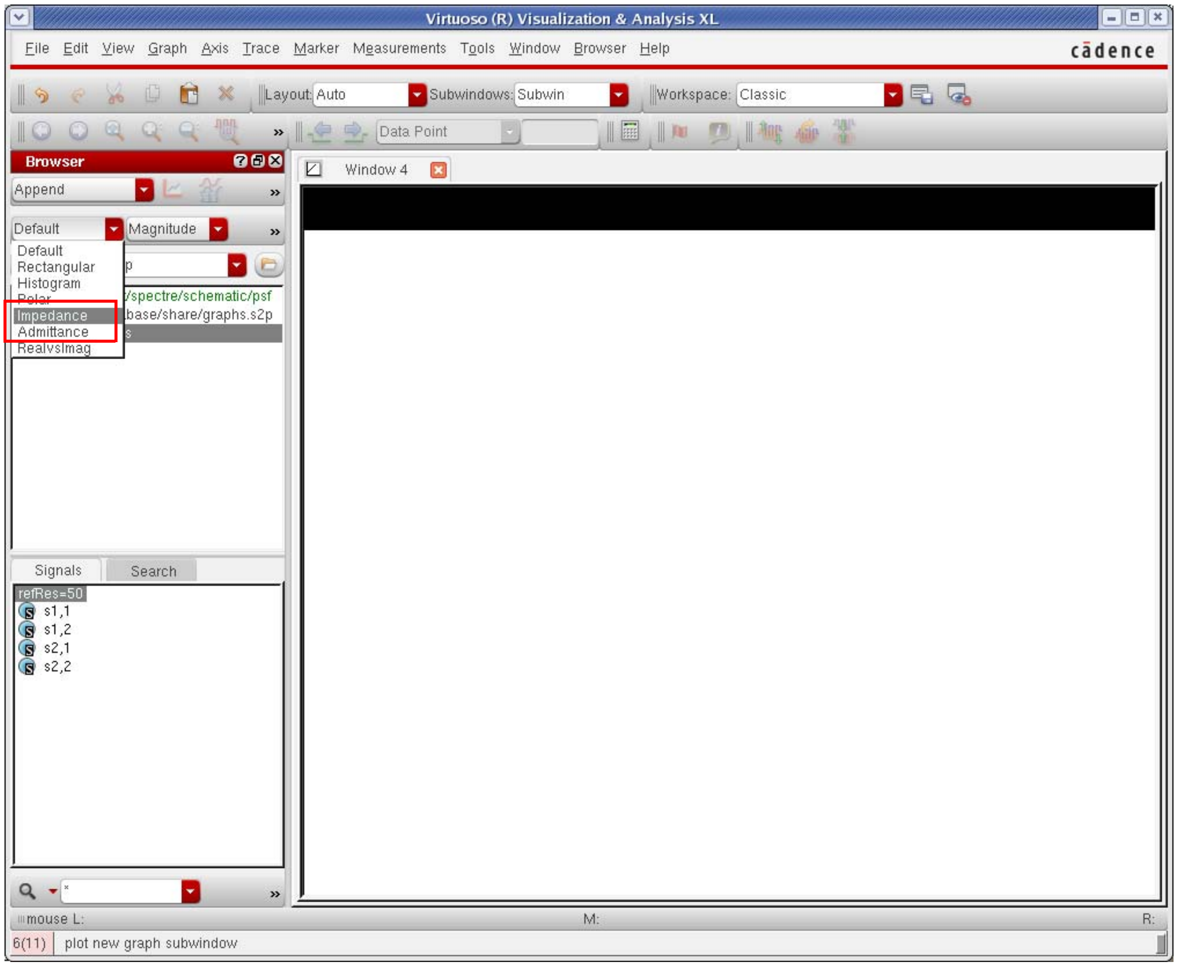
3. When the file is selected, the waveform tool has the path to the results at the top-left and the S-Parameter filename at the bottom. If you started from ADE, you may also have the path to the psf file on the upper left.



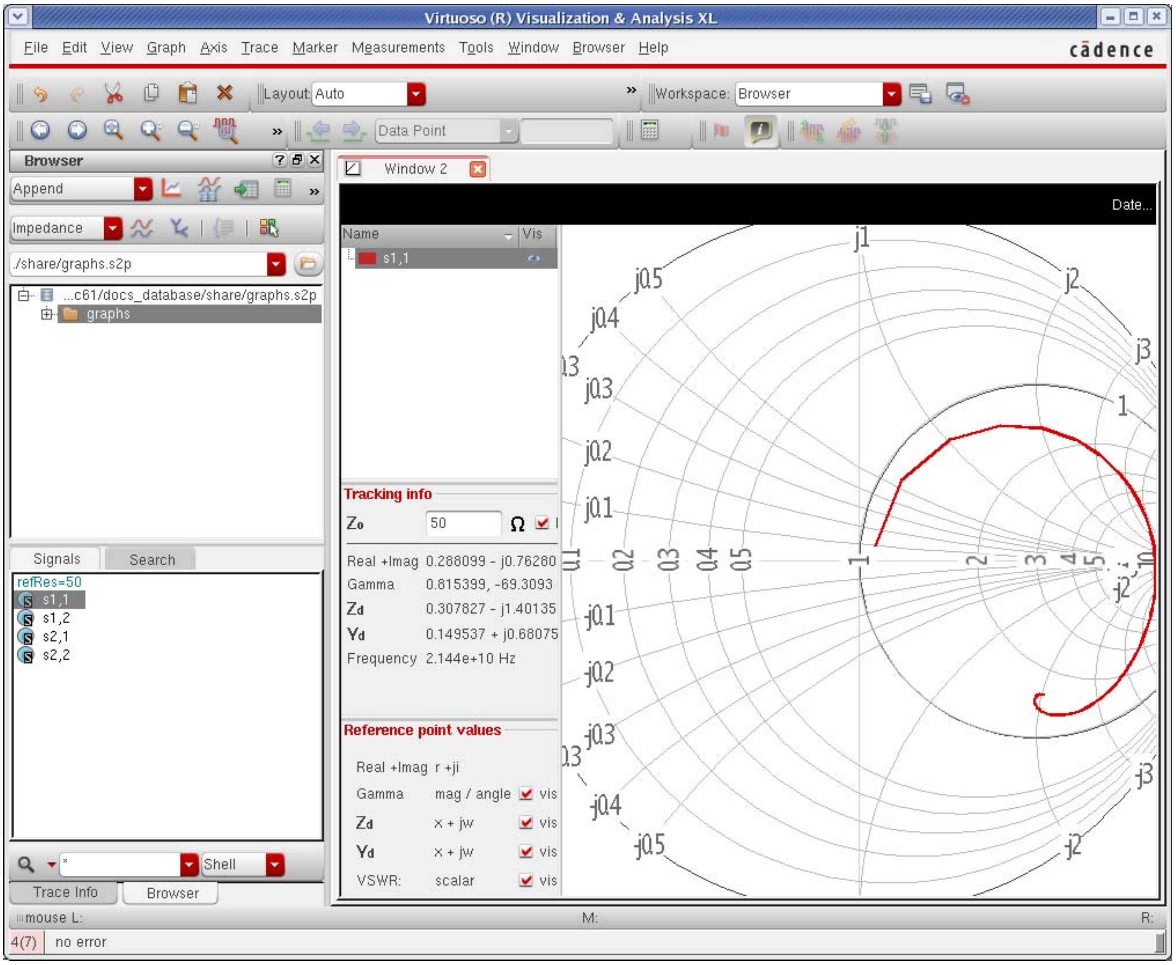
4. Double-click the filename. The S-Parameters in the file are displayed.



5. On the upper left, select *Impedance* or *Admittance*. This tells the waveform tool to plot on a Smith Chart.



6. Click one of the S-Parameters. Click the right mouse button and select *Move to Plot Signal*. The signal plots.



Important

S11 above has visible piecewise-linear behavior at low frequencies. This S-Parameter file will produce incorrect time-domain results at the frequencies between the inflection points in the S-Parameter plot.

Potential Problems with S-Parameter Files

Not all frequency domain descriptions are causal. This is a fundamental requirement for the time-domain model to be accurate. When Spectre reads in an S-Parameter description and causality is being checked, the transfer function from the highest frequency in the S-Parameter file through three times the highest frequency in the S-Parameter file is adjusted to enforce causality. Frequencies above three times the frequency in the S-Parameter file should be avoided.

Convolution-Based Method

The convolution-based method is strongly suggested for most simulations. This method is used when the interpolation method is set to linear or spline. The default interpolation method is linear, and the convolution-based method is the default.

The convolution-based method samples the frequency-domain data, and then calculates the impulse response for that frequency description. Then a Fourier transform is used to create the DC and time-domain model.

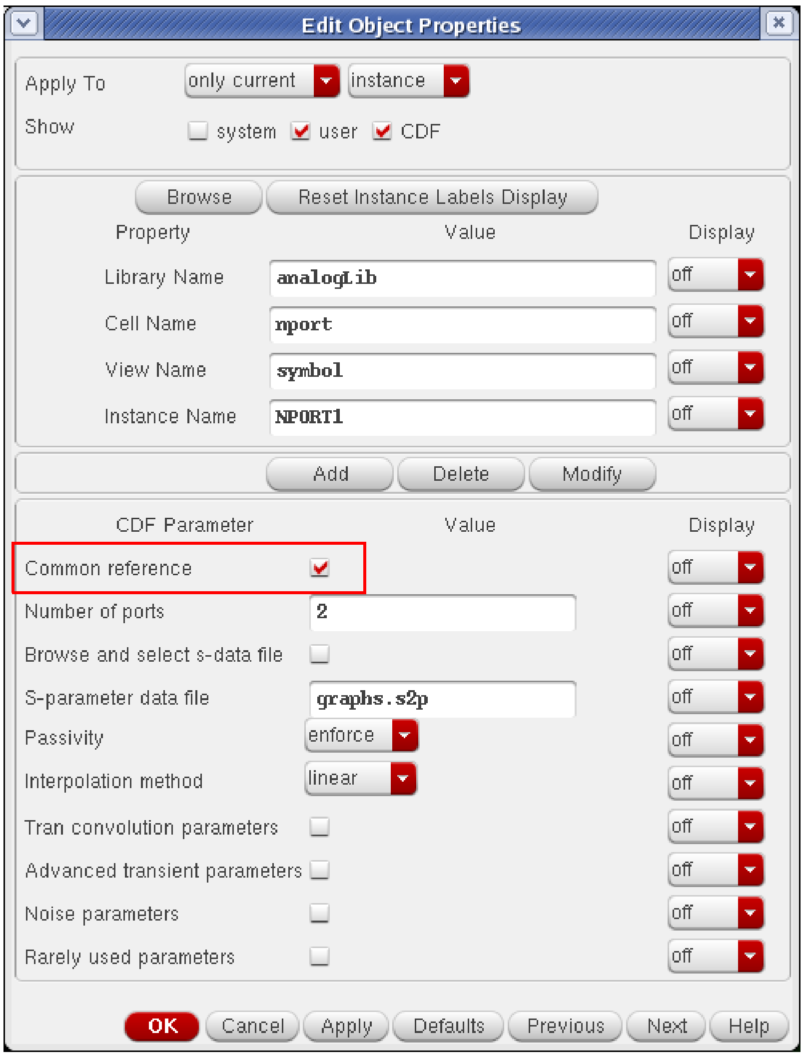
Considerable development has been made in MMSIM12.1 so that the only thing that should be required is to set the S-parameter file name, the interpolation method, and the number of ports. The default settings are strongly recommended for all other parameters. By default, the impulse responses are cached. This allows the reuse of previously computed impulse response files, thereby saving a lot of time in simulations after the first time the simulation is run. If multiple instances in the schematic use the same S-Parameter file, the impulse response is calculated only once. The impulse response may need to be recalculated if you specifically set some of the properties that are available in the nport. Except for the *Passivity* parameter and the *Accuracy* parameter, changing any other property is highly discouraged unless you are an expert in simulating S-Parameters. The old Impulse response will not be erased. If you switch back to the previous settings in the nport, those files will still be available, and will still be reused. The location of the files depends on the setting of the *nportfiledir* option which is documented later.

S-Parameter files with a large number of ports require lengthy calculations. In many cases, although there are a large number of ports in the S-Parameter file, only a small number of ports are actually connected in the circuit. This is common for package models. Starting with the MMSIM 12.1.1 release, the number of calculations can be reduced by eliminating the unused ports from the circuit, thus, improving the simulation time. This is done at the beginning of the simulation, and affects all the analyses that are specified to be run. If there is a component in the circuit connected to a port in the circuit, it is kept. If there is no connection, this port is removed. If you desire to monitor the output of a port that is not connected, add a large resistor to ground on this port so you can see what appears on this connection point. By default, if there are 10 ports or more, and 80% or less are actually connected to other components, the nport will be compressed. There are also global options in the *Simulation - Options - Analog form* (*Components* tab) to set compression explicitly.

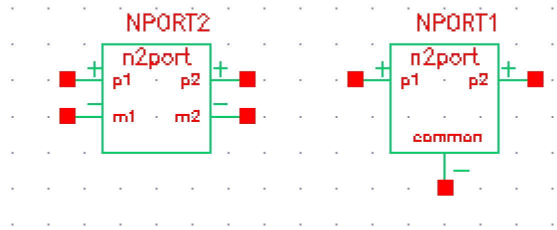
Using the nport Component

The nport symbol is a schematic pcell that is redrawn when you change the number of ports. The symbol has two different modes. At the top of the properties list, is a check box (highlighted below) called *Common reference*. If the item is not checked, plus and minus pins will be displayed for all the individual ports. If the check box is selected, then the symbol redraws with a single common ground reference pin at the bottom of the symbol. This eliminates the need to add ground connections to each port of the symbol.

Figure 8-17 nport Component Edit Object Properties Form

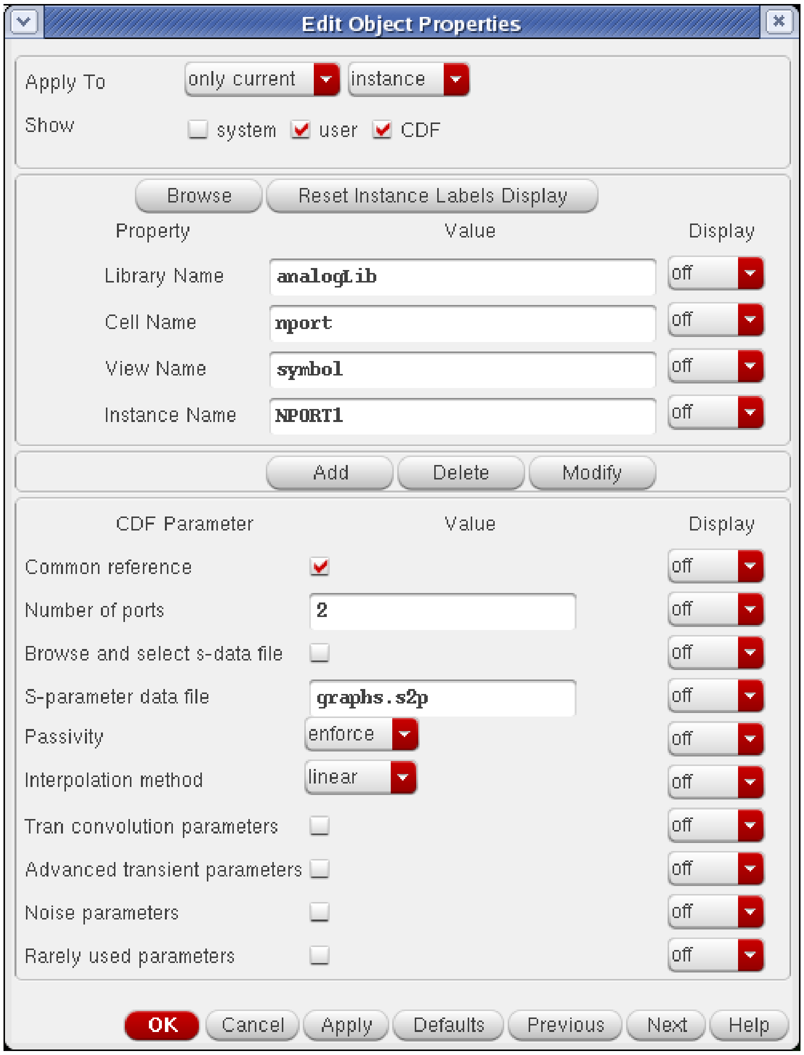


The symbol without the selection is shown on the left in the figure below, and the symbol with the selection is shown on the right.



Use the *nport* component to read in S-parameter data. Follow these steps to prepare the component for use in SpectreRF simulations.

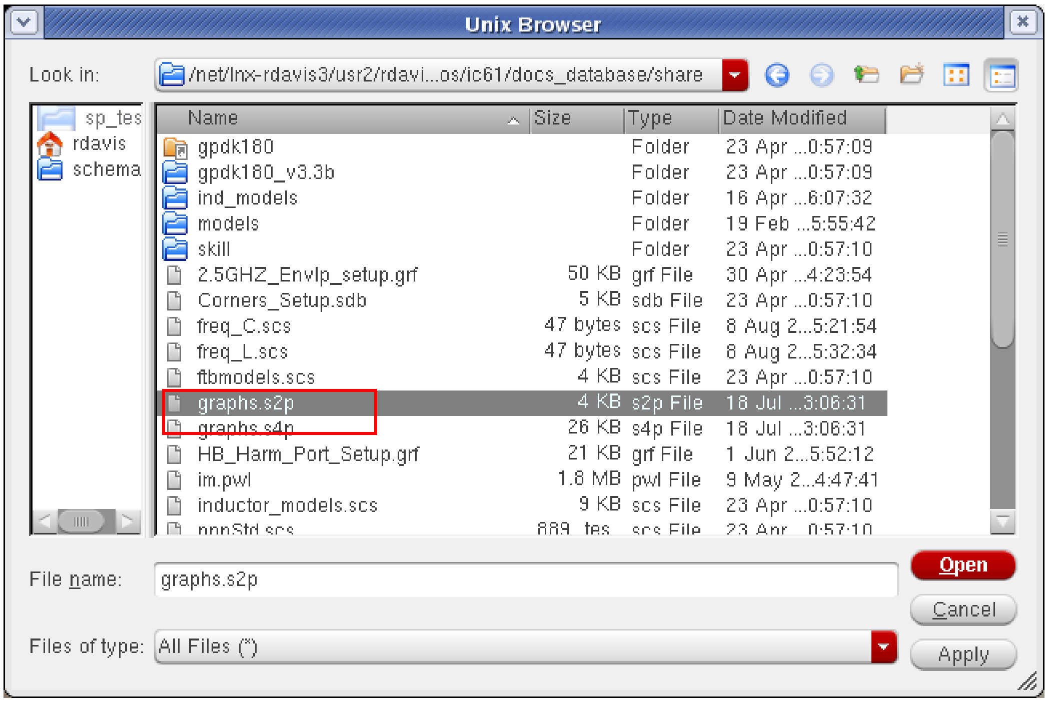
Figure 8-18 nport Component Edit Object Properties Form



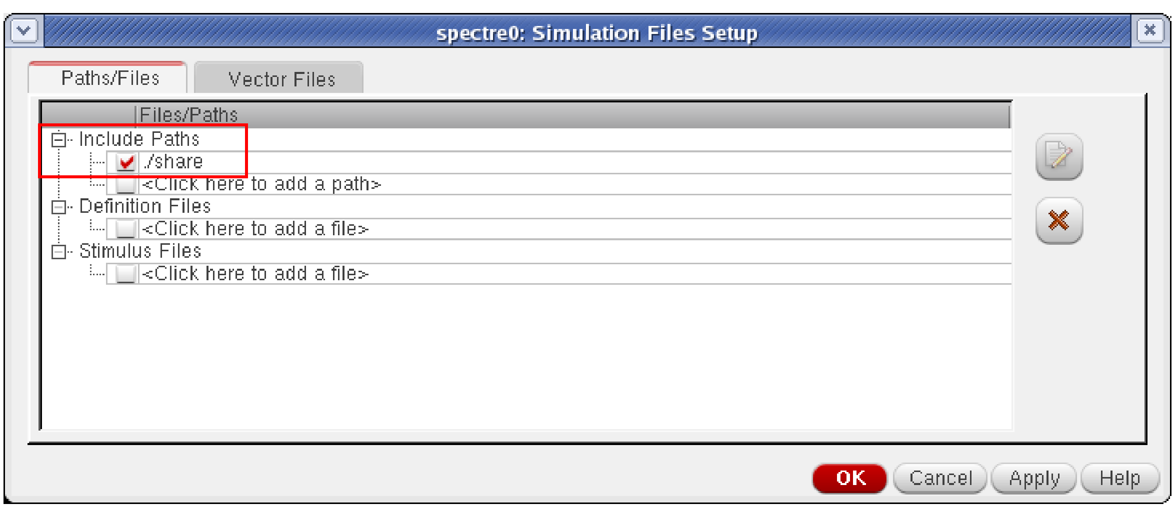
1. Select the *nport* component from *analogLib*.
2. Place the *nport* component in the schematic window.
3. Set whether you want a single ground terminal on the nport symbol or not.
4. Set the number of ports that are in the S-Parameter file.
5. In the Schematic window, highlight the new component and then choose *Edit – Properties – Objects, or use the bindkey* you have set up for setting properties.

   The *Edit Object Properties* form for the *nport* component is displayed.

6. Click the file browser button located to the right of *Browse* and select the S-data file in the properties form. From the browser, select the S-Parameter file.



Alternatively, type in the S-Parameter data filename. If you specify just the filename in the port component (not the complete filename starting from slash), then in ADE, select *Setup - Simulation Files*, and enter the path to the directory that contains the S-Parameter file in the *Include Paths* section. This is necessary so that the file can be found.



7. Although the properties form is labeled S-Parameter data file, it can also be a Y-Parameter or Z-Parameter file. Spectre recognizes all three formats. It is labeled this way because S-Parameter files are more common than Y or Z-Parameter files.

8. Specify whether you want the passivity check to run. Setting *Passivity* to *no* causes no passivity check to be run. Setting *Passivity* to *check*, which is the default, will run a passivity check, and the result of the check will be sent to the Spectre output window. The S-Parameter file will be used as specified. If *Passivity* is set to *enforce*, the passivity check will be run, and the S-Parameter file will be changed as necessary to ensure the passivity of the result. Note that this will change the S-Parameters as specified in the file if a non-passive entry is detected.

9. Set the *Interpolation method* cyclic field.

The *linear* and *spline* values are usually preferable to *rational.* In general, the recommended method is *linear.*

Linear and spline control the sampling of the S-Parameter data for the convolution-based method. In both methods, the S-parameter data is sampled using a linear frequency spacing from zero to three times the highest frequency in the S-Parameter data file in order to calculate the impulse response of the transfer function.

Important

When the S-Parameter file is for a physically large structure like a package model, because of the physical size of the structure, the inductance drops starting in the range of 100KHz to 1MHz. To correctly model this behavior, you require data in the S-parameter file that accurately describes this behavior, and it requires a huge number of frequency sample points in order to get a correct time-domain model.

When *linear* is selected as the interpolation method, linear interpolation is used to get a datapoint needed in the sample that is not directly in the S-Parameter file. Spline uses a cubic spline algorithm. Cubic spline can occasionally introduce errors when there are rapid changes in the transfer functions defined in the S-Parameter file near the sample point.

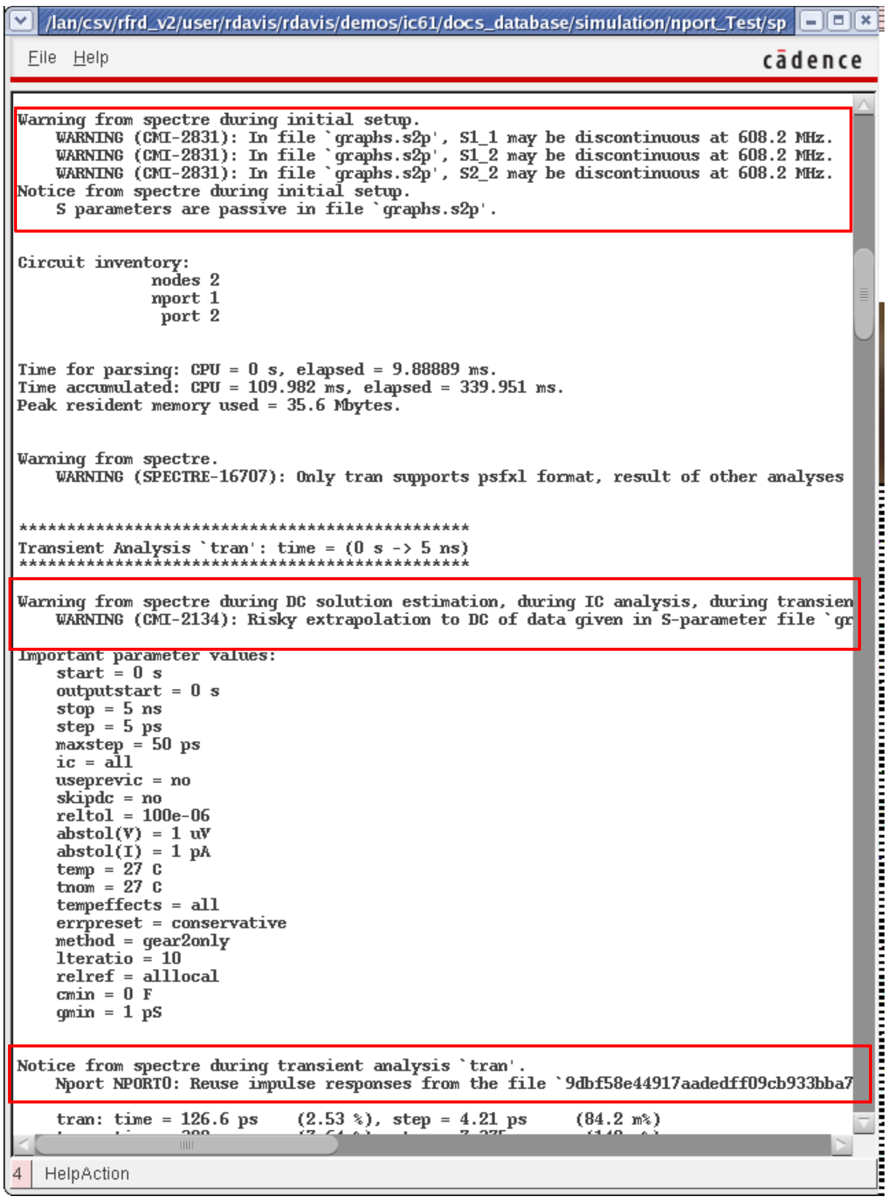
When the data in the S-Parameter file is dense, linear and spline produce comparable results. When the data in the S-Parameter file is sparse, non-physical results can be introduced because of the wiggles that can be introduced by a polynomial fit. Linear is usually a better choice when the data is sparse.

When you select *rational*, four new fields are displayed. For information about these fields, see Controlling Rational Fit Accuracy on page 1831.

You can also select *bbspice*, which uses the rational fit algorithm with passivity enforcement to create the time-domain model from an S-Parameter file.

10. If exceptional accuracy is desired, select *Conservative* for the Accuracy default. This will cause more frequency-domain sample points, and produce a more accurate impulse response at the cost of runtime. *Default* accuracy is set tight enough that *Conservative* should only be needed in rare instances.

When the DC or transient-based simulation is run, Spectre provides information about the S-Parameter file and the impulse response file in the output window.



Nport Compression

In cases where package models are used, many ports may be specified in the S-Parameter file, but only a small number of ports are actually connected to the circuit for simulation. Although the nport caches the data from the S-Parameter file, with a large number of ports, the extraction time can be an issue. To speed these extractions, the S-Parameter file can be compressed, which removes the unused ports from the extraction and from the simulation result, but causes the extraction to be considerably faster. The extraction time is reduced to the square of the connected port fraction. If 10% of the ports are connected, the extraction time goes to 0.01*old extraction time.

Unused ports are ports that are not connected to any circuit component, ports that are connected to ground, or ports that are shorted together.

By default, if there are 10 or more ports in the S-Parameter file, and less then 80% of the ports are connected, nport compression runs. First, the connections to ground and shorted ports are applied, then an s-parameter analysis runs to capture the performance of the connected ports. That file is written to the netlist directory by default, and has a global option to control the location. To find the netlist directory, first determine the location of the simulation directory by selecting *Setup - Simulator/Directory/Host* menus in ADE, and read the location of the *Project* Directory. Navigate to this directory, then type *cd <circuit_Name>/spectre/ <schematic_or_config>/netlist*. The compressed filename is the original S-Parameter filename with an extension. This is the file that is actually used for the simulation.

Note that this occurs before any of the analyses are run, so all the simulation results from all the analyses will have data for only the ports that have circuit connections. If you want to monitor the result on a port that you do not want to connect to the circuit, add a large resistor to ground that is connected to that port so that the data for that port is saved.

Controlling the Directory for the Compressed S-Parameter File

This directory can be set using the global options form by selecting *Simulation - Options - Analog*, and then selecting the *Component* tab. Specify a full pathname to the desired directory location in the *nportcompressfiledir* field.

Compression can also be controlled manually by selecting *yes* or *no* for the *nportcompress* option.

These options are shown below.

nportunusedportgmin

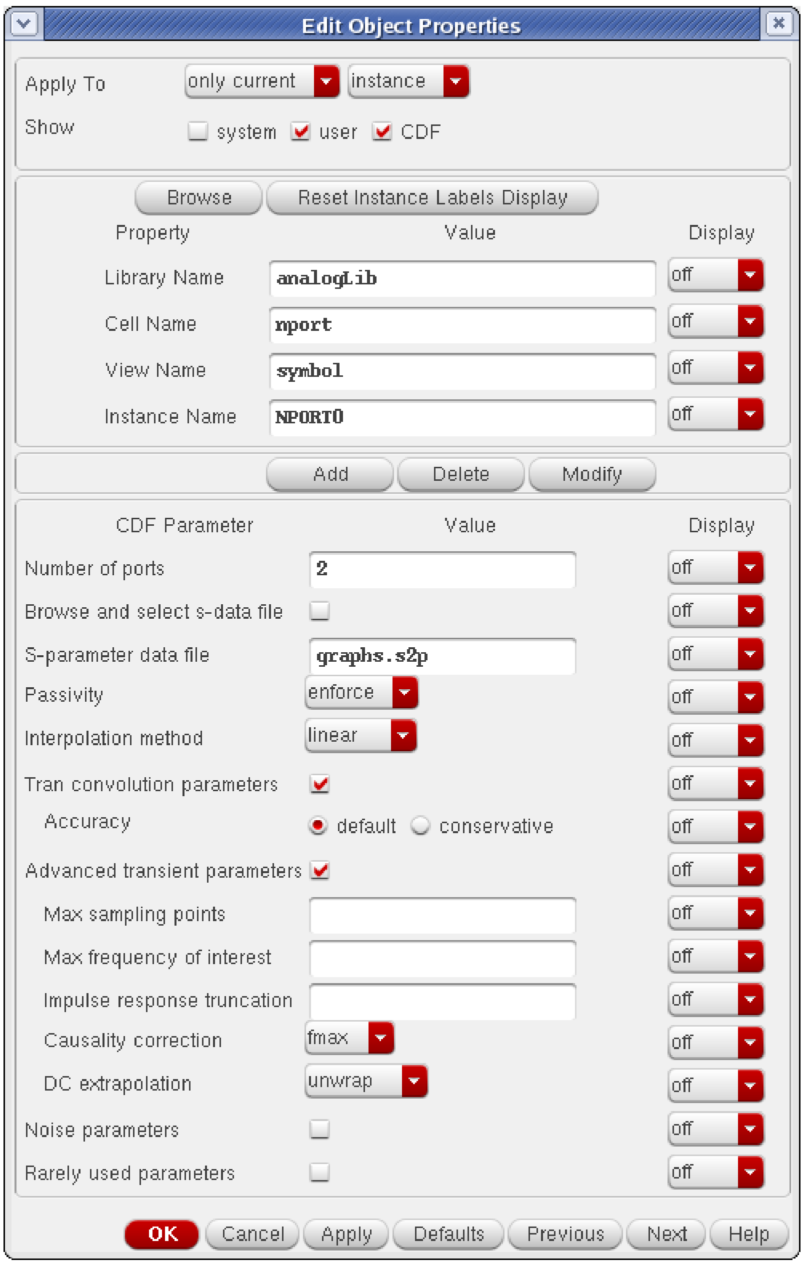
This option adds resistors to nport pins that are not connected to circuit components. Setting a small value (on the order of 1e-9 to 1e-3) adds a resistor with conductance set to this value from each unconnected pin to ground. This introduces a small error, but adds a small amount of damping to the nport response. This is sometimes useful for obtaining a passive response.

nportunusedportrmin

This option adds resistors in series with nport pins that are shorted together. Setting a small value (on the order of 1e-3 to 1 ohm) adds a resistor with resistance set to this value in series with the connection between pins. This introduces a small error, but adds a small amount of damping to the nport response. This is sometimes useful for obtaining a passive response.

Advanced transient convolution parameters

These properties are provided for S-Parameter simulation experts only.



Max sampling points

Max sampling points defines the maximum number of frequency points to be sampled in the adaptive algorithm. The default is 131072. In every case, encountered so far, the actual number of samples taken by the adaptive algorithm is much smaller than the default. In extremely unusual cases, it can be raised to 262144.

Max frequency of interest

The property *Max frequency of interest* should not be changed unless you are an expert in the simulation of S-Parameter files.

Max frequency of interest (fmax) controls the highest frequency for the frequency domain sampling of the S-Parameter file. Leave this property at the default. The default is three times the highest frequency in the S-Parameter file.

Impulse response truncation

Impulse response truncation is used to deliberately cut off the tail of the impulse response which might theoretically continue to infinite time. Leave this property at the default of 1e-4, which corresponds to a gain of -80 dB.

Causality Correction

The *Causality Correction* drop-down list contains three choices: *no*, *fmax*, or *auto*. *fmax* is the default and is highly recommended. Causality enforcement is required in order to have reasonable results from an nport in either the DC or transient-based analyses. Causality correction is performed by setting the transfer function between the highest frequency in the S-Parameter file and three times this frequency so that the data becomes causal. The data within the frequencies specified in the S-Parameter file is unchanged. Setting causality to `fmax` or `auto` overrides the setting of the high frequency extrapolation property.

- *no* does not add a causality check.

> *Important*
>
> Setting the causality check to *no* is incredibly risky unless you are absolutely sure that the S-Parameter file is causal as described.

- *fmax* retains the data in the frequency range of the S-parameter file, and then adds a transfer function above the frequency range in the S-parameter file to force the system to be causal. This transfer function extends to the setting of *Max frequency of interest*, which defaults to three times the highest frequency in the S-Parameter file. If you suspect that the maximum frequency of interest needs to be changed, use causality *Auto* instead, if you are not an expert.

- *auto* applies the causality correction in a similar manner to choosing *fmax*. *auto* can also vary the maximum frequency of interest if it needs to get a causal time-domain model.

DC extrapolation

The *DC extrapolation* can be set to *constant* or *unwrap*. *constant* projects the first point down to zero frequency at exactly the same level. *unwrap* does an estimation based on the first few frequency points in the S-Parameter file. The default is *constant*.

Noise Parameters

These properties should never be needed.

Thermal noise

Thermal noise defaults to *yes*. If you want to disable noise production, set to *no*.

Thermal noise model

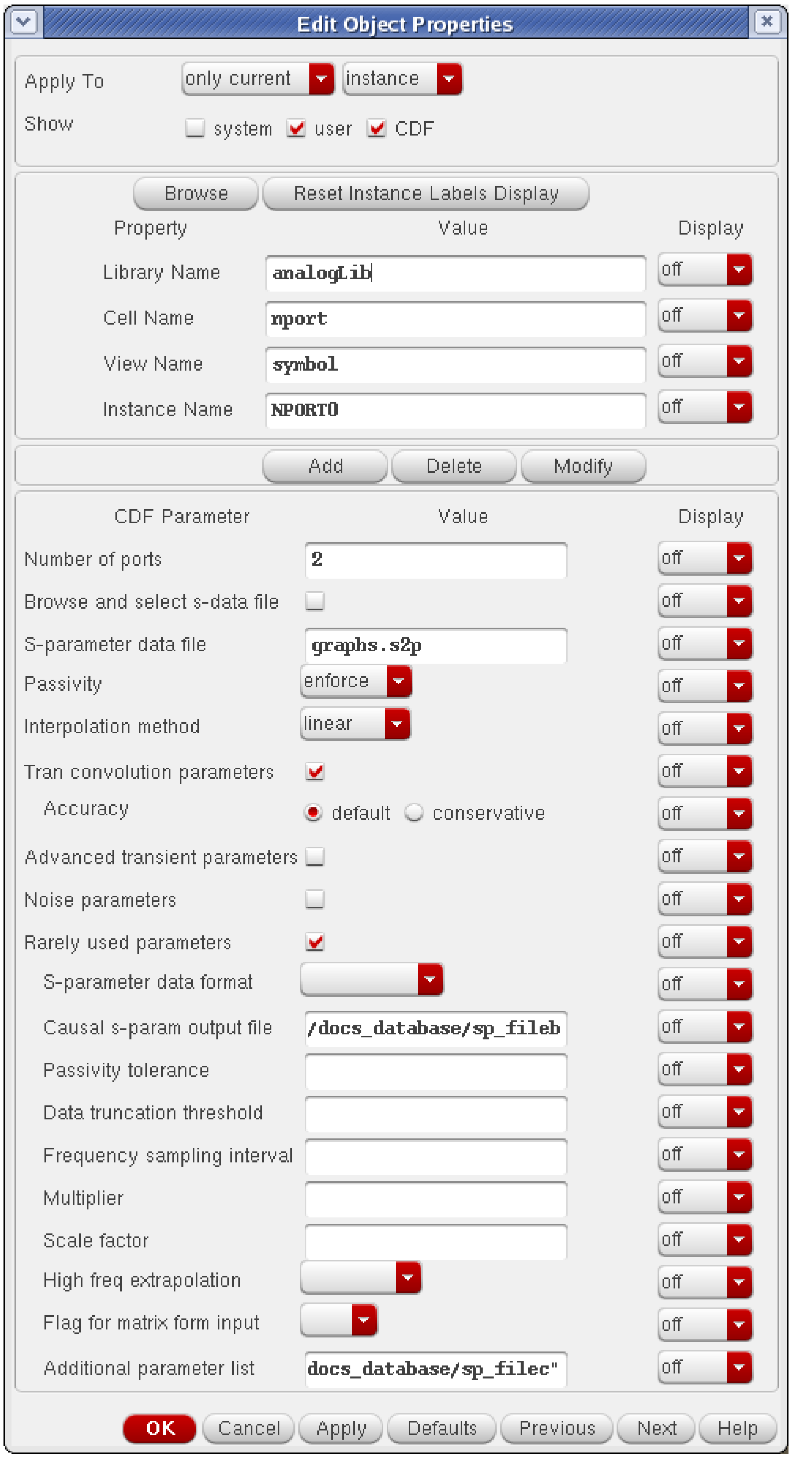
Thermal noise model defaults to external, which reads the noise parameters in the S-parameter file if it is available, and if not, it uses an internal noise model. Internal forces the internal noise model.

Noise correlation

The default is complex. The noise parameters in the S-parameter file might have real and imaginary parts. By default, if these parameters are complex, Spectre observes them. In some cases, passive components may have complex noise parameters in the noise file, which is theoretically not possible. In this case, set *Noise correlation* to *real*.

Rarely Used Parameters

These properties should never be needed.



S-Parameter data format

S-Parameter data format is not needed because Spectre can directly parse an S, Y, or Z-parameter file in Spectre, Touchstone1, Touchstone2, CITI, or rfm file format.

Note that the Touchstone1 file format now provides support for multiple impedances at each port. This is done in the header of the file. When R is listed, then multiple values can be entered up to the maximum number of ports in the file. If a number is omitted, then that number takes on the value of the first R entry.

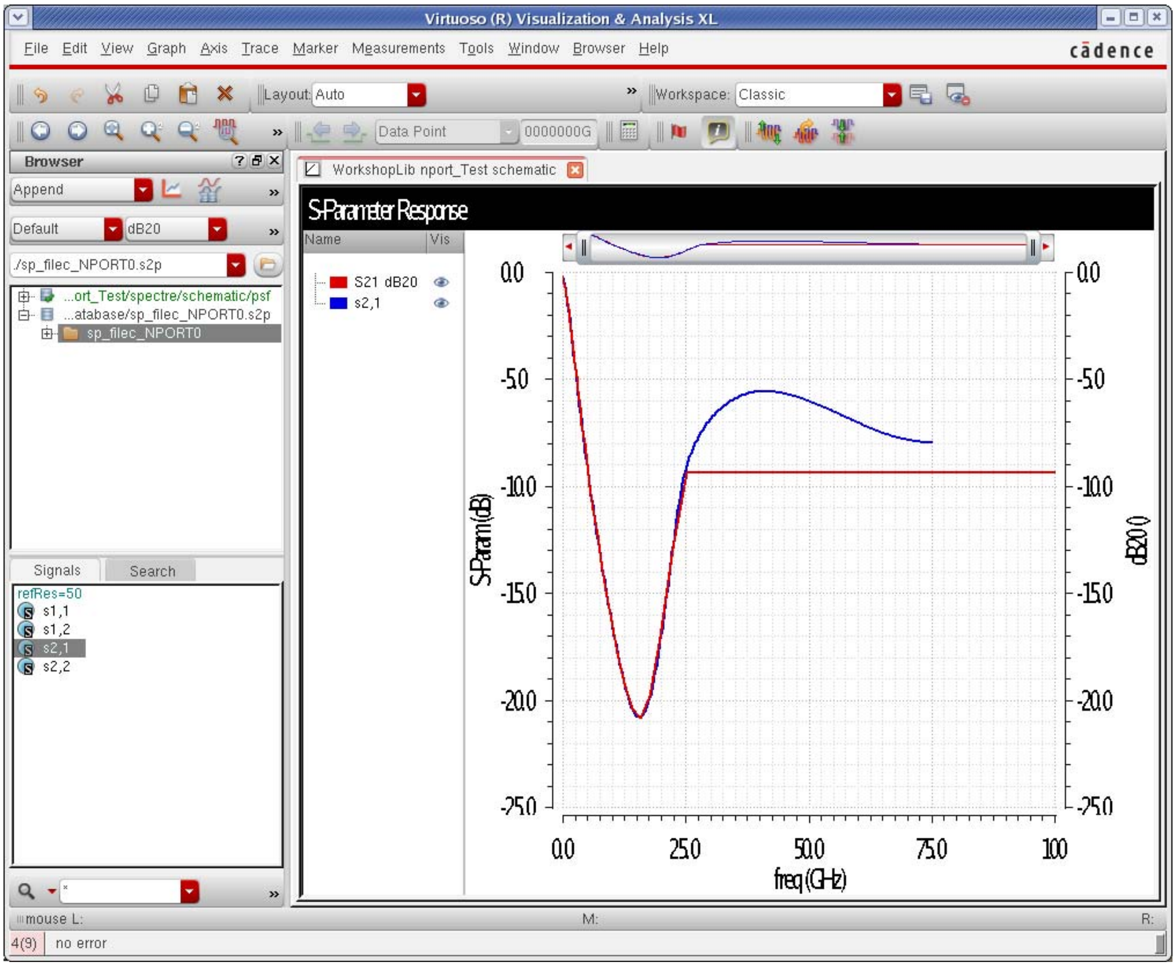
For example, `#GHz RI R 50 1 1 50` means the frequency numbers are in GHz. The data is in real-imaginary format. The resistance is 50 Ohms on port 1 and 4, and 1 Ohm on ports 2 and 3.

`#GHz RI R 50 10 20 30` means 50 ohms on port 1, 10 ohms on port 2, 20 ohms on port 3, and 30 ohms on port 4.

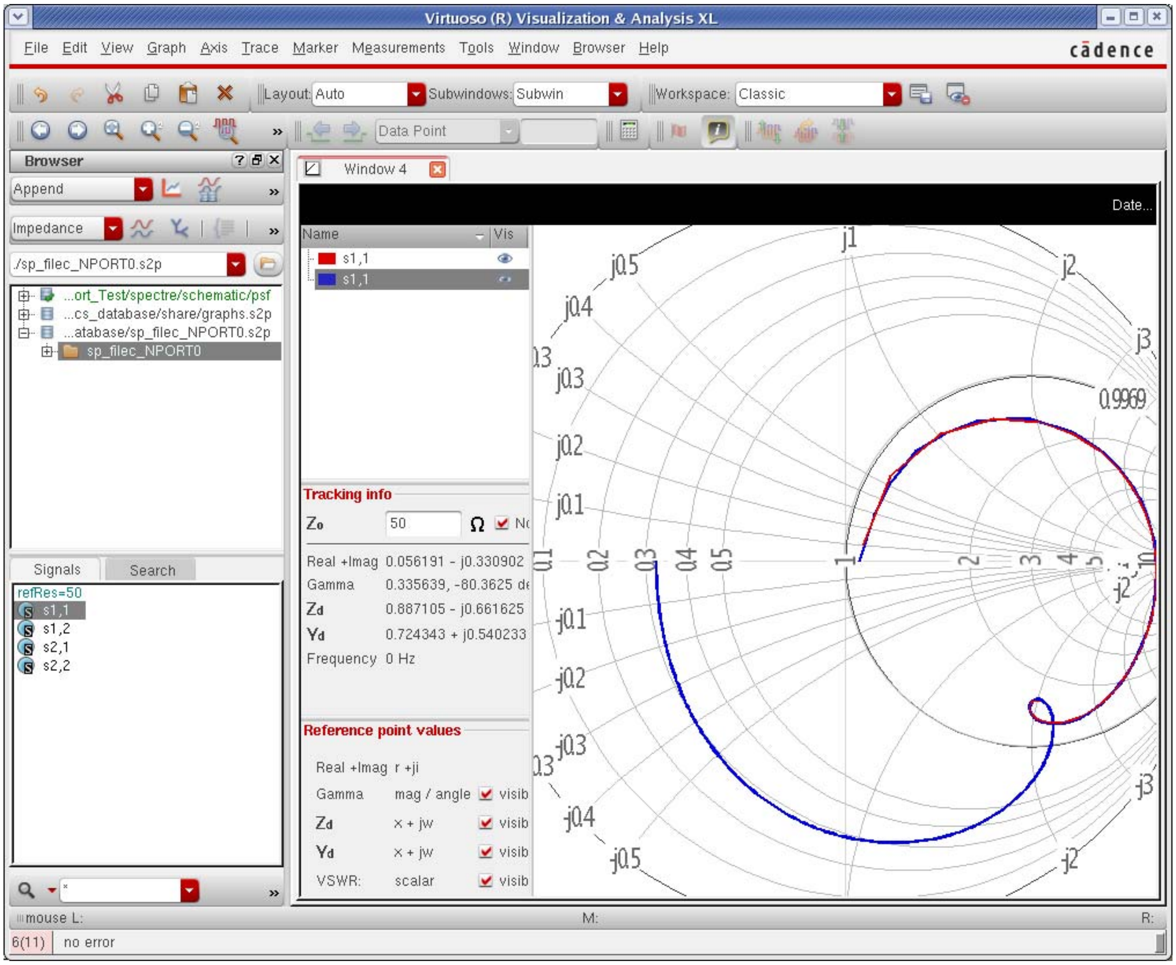
`#GHz RI R 50 10` means 50 ohms on ports 1, 3and 4, and 10 ohms on port 2.

Causal s-param output file

Enter a filename beginning with slash (/) in this field. Specifying a filename causes the S-Parameter data after causality correction to be placed in this file. This file can then be plotted directly in the waveform display tool. Most of the time, the data within the S-Parameter file frequencies is not disturbed, and the data between the highest frequency and three times the highest frequency is the data that is added to make the time-domain model causal. Below is a comparison of the raw data and the causality-corrected data. The red trace is the original data.

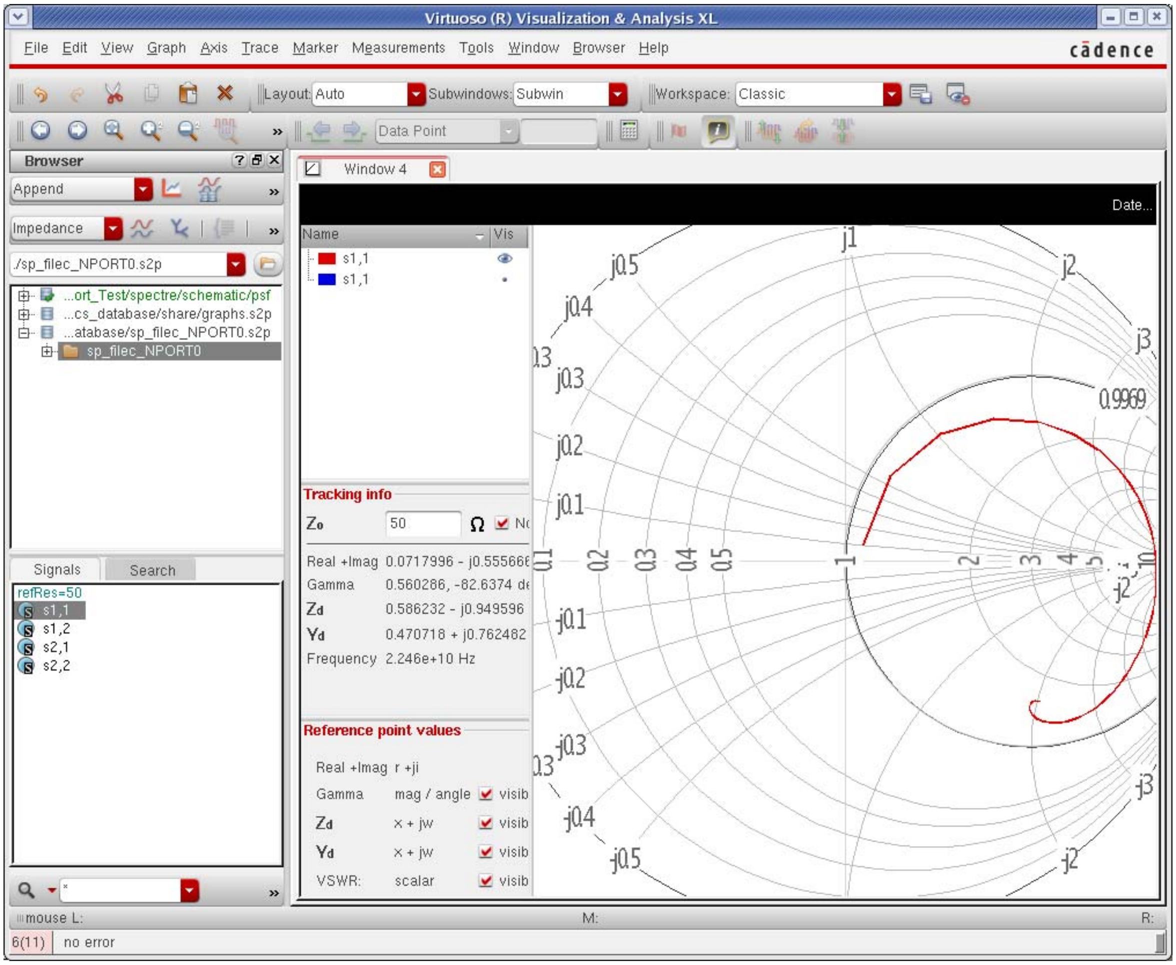


This has the data plotted on a linear X-Axis.



This has the data plotted on the Smith Chart. The original data is in red, and the causality-corrected data is in blue.

Here is the original data in the Smith Chart.



Passivity tolerance

This property is only used when the *Passivity* property is set to *check* or *enforce*. Passivity tolerance does not need to be set. The default is 1e-6. This defines how close to unity gain should be modified by the passivity check and enforcement. Passivity will be enforced and/or reported when the gain is (`1 - Passivity tolerance`) or greater.

Data truncation threshold

Leave this property at the default of 1e-3, which corresponds to -60dB gain. When the cross coupling terms become smaller than the data truncation threshold, they are ignored. Cross coupling is the coupling from one port to another port.

Frequency sampling interval

Frequency sampling interval sets the delta frequency for the sampling from zero to the maximum frequency of interest. Leave this property at the default value. With adaptive sampling, this should never be necessary. If used, this delta should be a power of two divisor of the maximum frequency of interest.

Multiplier

Multiplier specifies how many nport devices to put in parallel. This is rarely used.

Scale Factor

This scales the frequency of the S-Parameter file. For example, many S-Parameter files have the frequency in GHz. In this case, set the *Scale factor* to 1e9.

High freq extrapolation

This property should not be used. This property is ignored when causality correction is applied. The *High freq extrapolation* field can be set to *constant* or *linear*. If the causality check is not run, *constant* maintains the same amplitude and phase as the last point in the S-parameter file to infinite frequency. *linear* keeps the amplitude constant at the last frequency point, but the phase increases linearly with frequency.

Flag for matrix form input

Flag for matrix form of input should not be needed to be set. In the past, each time the simulation ran, the impulse response was calculated for every port of every instance of the nport every time the simulation was started. In some cases, especially with a large number of ports, this could take considerable time. This flag was provided so the step of calculating the impulse response could be skipped. Since the impulse response is cached and available for re-use at any time, this property should never be needed.

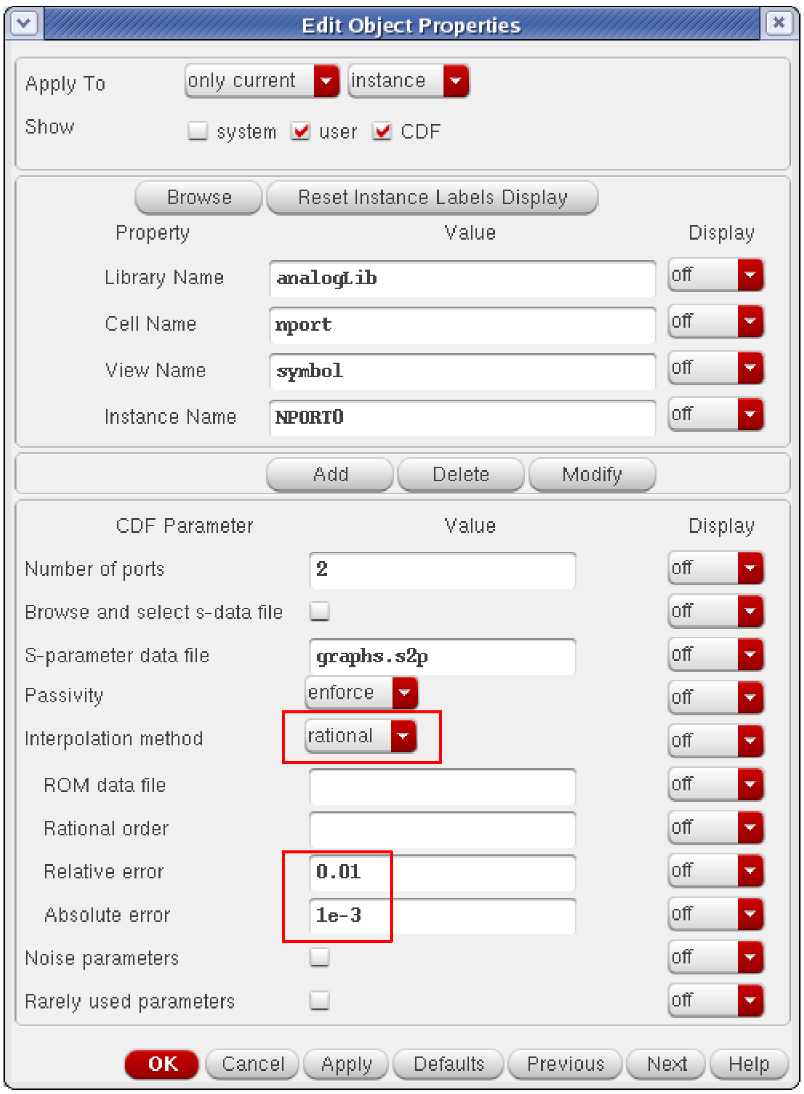
Additional parameter list

This field is typically used by beta customers to unlock new features. Note that when this feature is used, a warning message will be issued. This warning can be ignored.

Controlling Rational Fit Accuracy

Rational fitting is the process of adding multiple poles, zeros, and gain blocks to match the frequency response that is defined in the S-Parameter file. Note that the S-Parameter file may have distributed effects. Approximating these distributed effects with poles and zeros can require a system with a very high order. Matching an arbitrary frequency-domain transfer function with poles and zeros is a difficult and complicated process.

At the current time, it is usually better to use the convolution-based algorithms when the interpolation method is set to *linear* or *spline*.



If you do use the rational fit, always test the rational model that is produced by running sp analysis using a circuit composed of an input port, an output port, and an nport with the interpolation method set to *rational*. Run the sp analysis, and plot all the S-Parameters. Look for differences between the original S-Parameter file and the S-Parameters with the rational fit.

Also, the algorithm is known to take a very long time to run for S-Parameter files with a large number of ports.

The *nport* component has three parameters to control the accuracy of the rational interpolation process:

- *relerr* (The *Relative error* field)
- *abserr* (The *Absolute error* field)
- *ratorder* (The *Rational order* field)

You can use these parameters to trade off accuracy against model size and simulation time. In general, the more stringent the accuracy requirement, the higher the model order. Higher-order models require longer extraction and simulation time. SpectreRF can automatically generate a model that meets a specified accuracy requirement. You can also specify the model order directly to SpectreRF.

Using the relerr and abserr Parameters

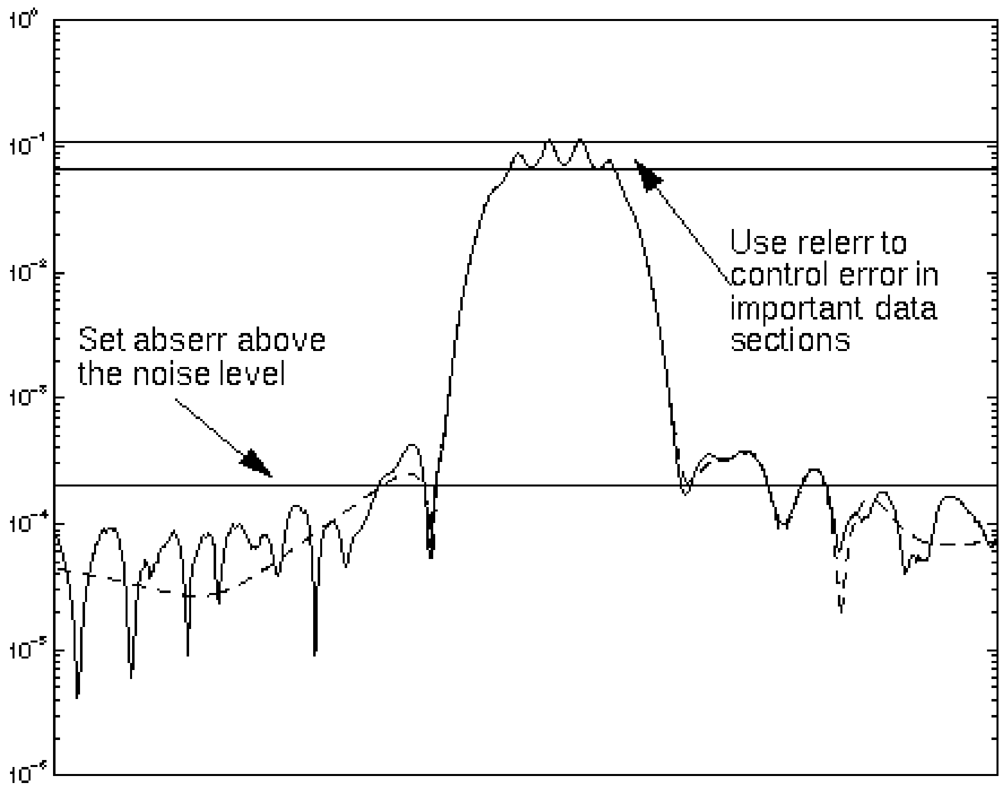
The *relerr* parameter is used to control the accuracy of the rational fit when the gain is larger than the gain specified in abserr. Usually, *relerr* is on the order of about 0.01, which sets a 1% error in the rational fit when the gain is large. Usually, *abserr* is in the order of 1e-3 to 1e-4, which specifies that when the gain is less than *abserr*, allow a lot of error in the rational fit. 1e-3 is -60dB gain, and 1e-4 is -80 dB gain. The *abserr* parameter lowers the order of the model by allowing fewer poles and zeros when the gain is small.

Consider the data shown by the solid line in Figure 8-19. In this example:

- *relerr* can control the error in the passband
- *abserr* can control the error in the stopband

If you are not interested in the details of the filter behavior in the stopband, you can set *abserr* to the level shown and these details are ignored in the interpolation.

Figure 8-19 Using relerr and abserr



This example has approximately a 10% ripple in the passband. To ensure adequate error control there, set *relerr* to much less than 0.1, for example, 0.01.

The dashed line in Figure 8-19 shows an interpolation function with the error control levels specified in Figure 8-19: *abserr*=2e^{-4}, and *relerr*=0.01. Details of the data below the *abserr* threshold are not resolved by the rational interpolation.

To recover the details in the stopband, set *abserr* to about 1e^{-5} for this example.

If there is noise in the data, set *relerr* and *abserr* above the respective noise levels. Otherwise, Spectre and SpectreRF attempt to interpolate the noise, resulting in very high order models and very long simulation times.

Remember that if you set *relerr* to zero, then, from the formula above, pure absolute error is used. Conversely, if you set *abserr* to zero, the error control is based solely on the errors relative to the magnitude of the input data.

Using the rational order (ratorder) Parameter

In general, it is usually best to specify only the accuracy parameters, *relerr* and *abserr*, and to let SpectreRF automatically select *rational order,* the order of the rational approximation. However, if you have information about your data set, you can direct SpectreRF to use a specific order of approximation in the rational interpolation. For example, if you know that your tabulated S-parameter data represents a sixth-order filter, you might instruct SpectreRF to use a seventh or eighth order fit. The slightly higher order gives SpectreRF flexibility to adjust for any noise or non-ideal behavior in the data.

When you specify the *ratorder* parameter, SpectreRF tries to meet the error requirements specified by *relerr* and *abserr*, but is limited to the number of poles and zeros it can use to achieve them. If the error specified cannot be achieved with the number of poles and zeros you have specified, a warning message is generated to let you know that the error requirements were not met.

ROM data file

If this file is specified, when the rational fit is performed, the results of the rational fit are stored in the specified file as a Spectre format reduced order model.

Troubleshooting

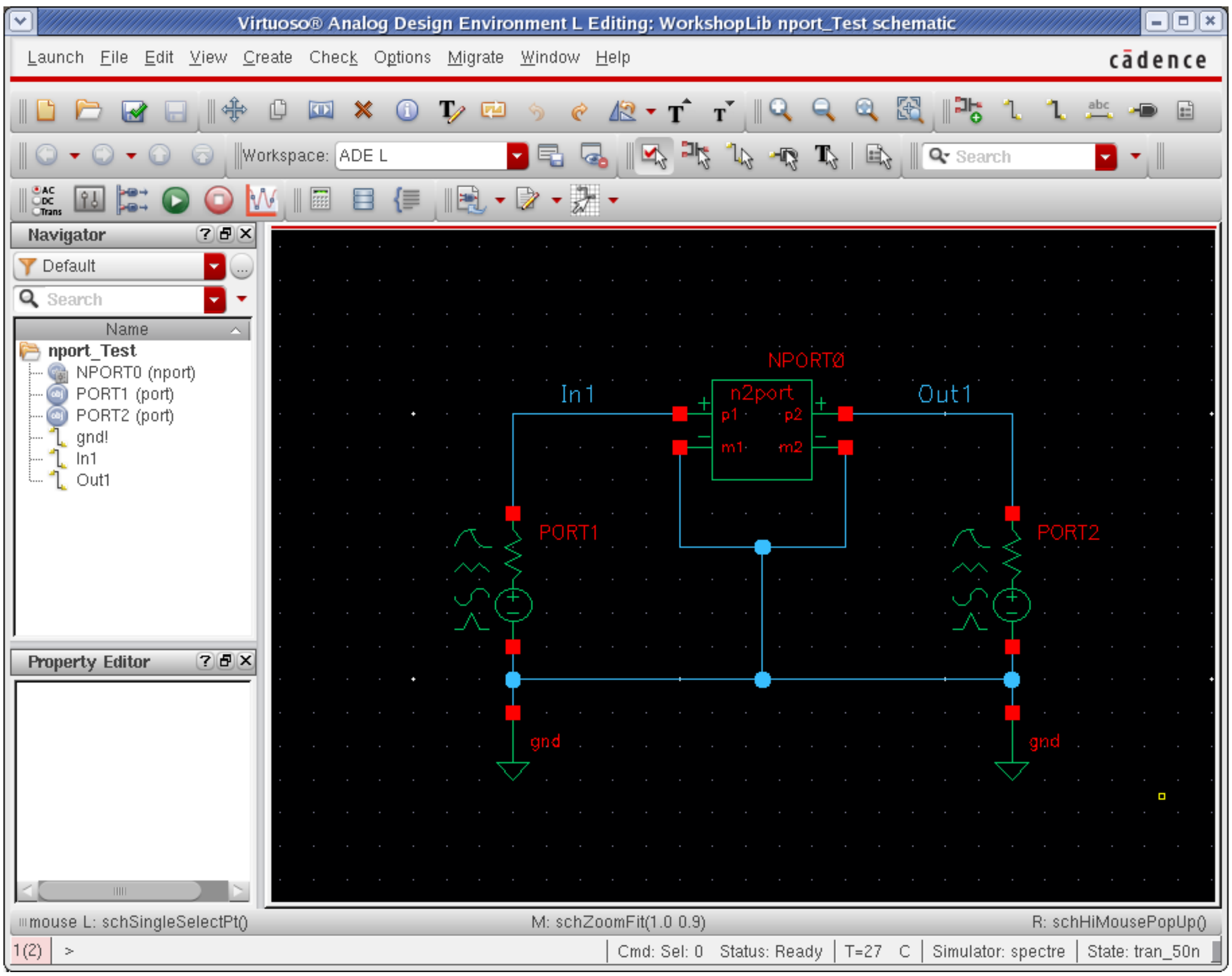
Certain S-parameter data sets might cause difficulty for the rational interpolation process. Types of data to avoid are:

- Data specified only over a very narrow frequency range. Time-domain simulation requires time-domain models with a wider range of values; values that lie outside the narrow frequency range. Data extrapolation might be difficult and is always risky.
- Very noisy data. Noisy data might lead to large, unreliable time-domain models.
- Data on a very sparse frequency grid. Accurate interpolation of such data might be impossible.
- Data with long ideal delays.
- Data representing idealized lossless elements, such as lossless transmission lines.

Assessing the Quality of the Rational Interpolation

If you suspect a problem with the rational interpolation process, you can investigate it using the *sp* analysis using the following steps:

1. Construct a test schematic consisting of an *nport* component with interpolation set to rational, as discussed in "Using the nport Component" on page 1814.
2. Next add the appropriate number of *port* components to the schematic. An example with two ports is shown below.



3. Perform an *sp* analysis on the *nport* component and look for anomalies.

   Large swings in interpolated values and S-parameter magnitudes greater than one both suggest a problem.

Large changes in the interpolated data result from an inaccurate fit. Usually, when this happens, there are local peaks or suckouts in the S-Parameter data.

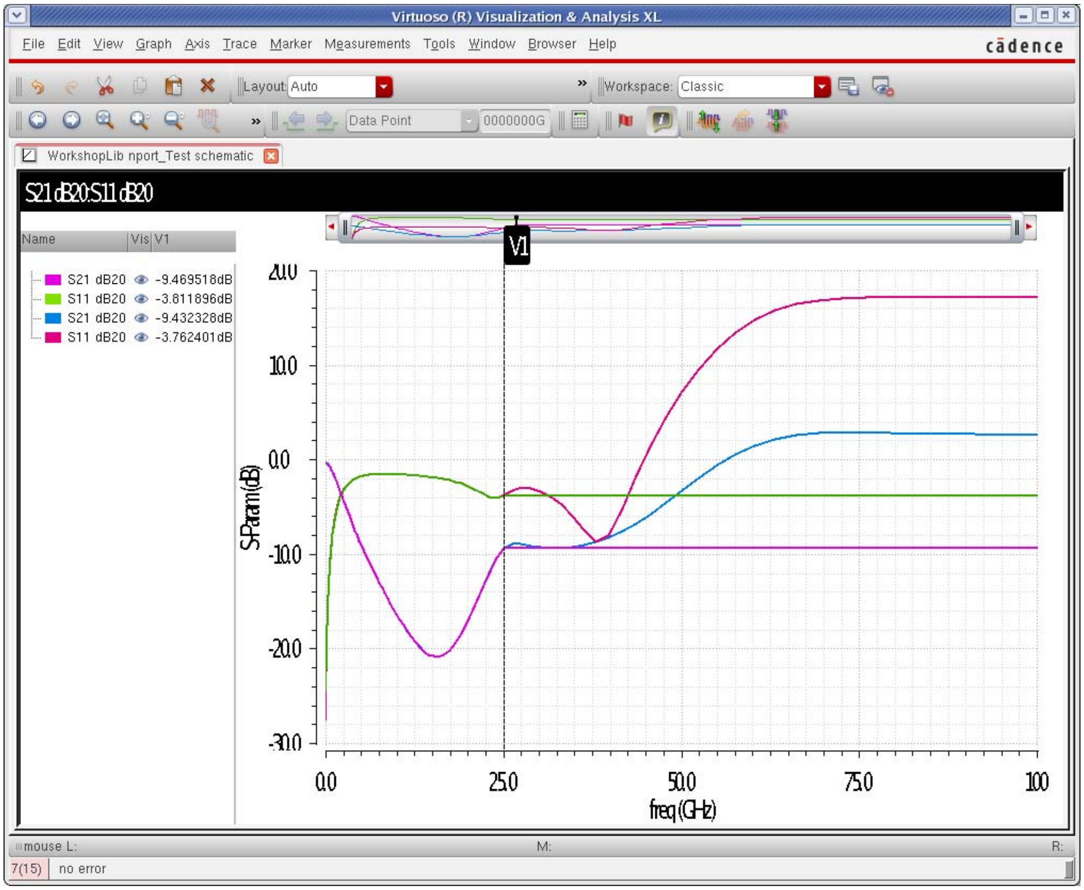
S-parameter values greater than one result from a non-passive (energy-generating) model that might create unstable time-domain solutions. Be particularly critical of anomalies near the zero frequency (DC). Most S-Parameter files do not have zero frequency data, so the DC point must be extrapolated. Sometimes in rational fit, the DC point can be inaccurate.

When the anomalous behavior occurs *within* the frequency range of the data in the S-parameter data file, it usually indicates an inaccurate rational interpolation. Anomalous behavior below the first frequency point and above the last frequency point are uncommon, but always needs to be checked. If you have this anomalous behavior, linear interpolation is suggested.

Possible Problems with Rational Fit

Because the rational fit can have unexpected behavior, it is suggested that you run an S-Parameter simulation over a very wide frequency range. Check for those behaviors in the sp analysis.

A comparison between rational and linear interpolation is shown below. Note that S21 and S11 both go considerably above unity gain (0dB) above about 60GHz for the rational fit. The data in this file stops at 25GHz. This data is from a passive structure.

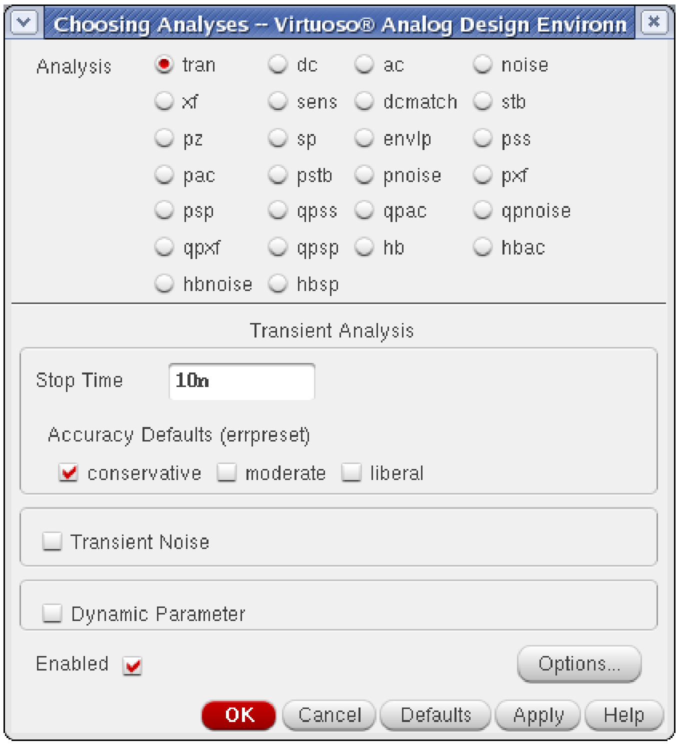


Magenta and green are the original data. Blue and red are the rational fit.

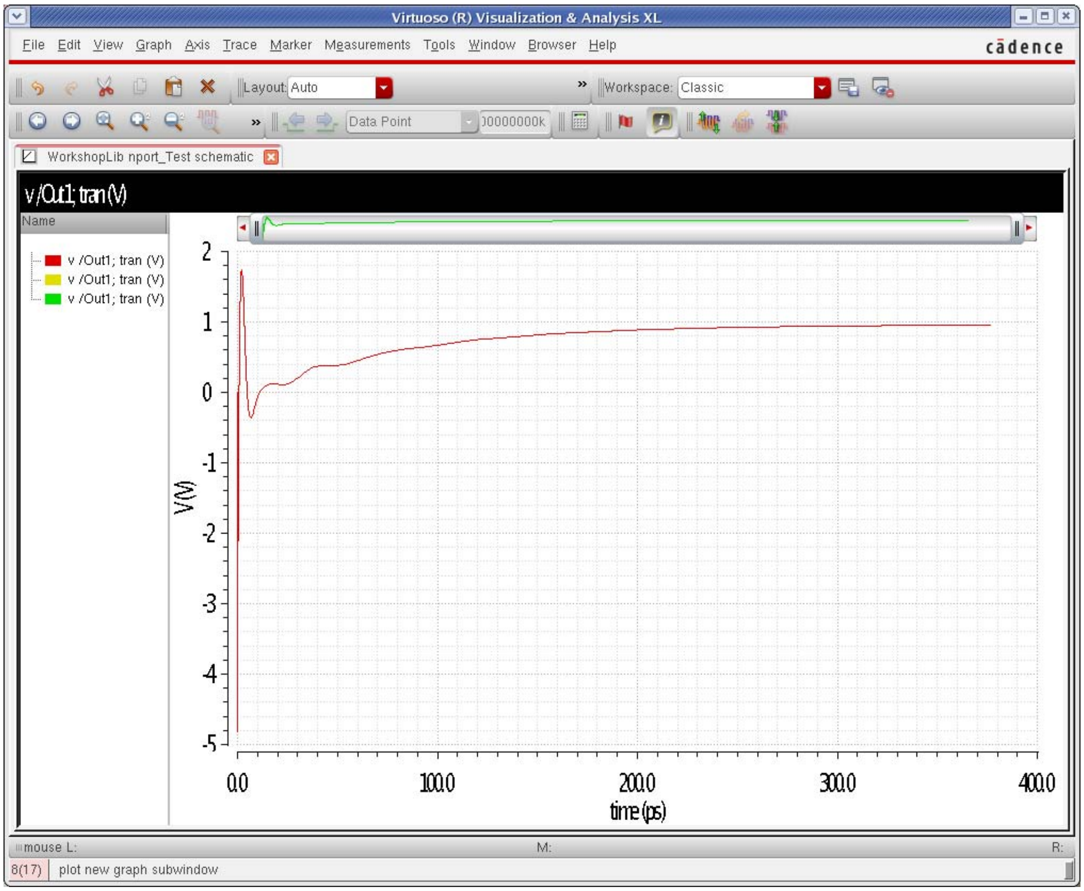
Also note the differences in the low frequency data shown below.

1. Verify that all the conditions listed at the beginning of the Troubleshooting section are met.
2. Try decreasing *relerr* or *abserr* or both.
3. Try specifying a higher-order interpolation with the *ratorder* parameter.
4. Remember that measured data can contain fine details that might require a higher order than you might expect from a casual inspection of the data.
5. If the anomalous behavior occurs *outside* the tabulated frequency range, try changing the *abserr*, *relerr*, or *ratorder* parameters. Sometimes anomalies can be removed by using a more accurate fit. However, there are limits to the ability to extrapolate outside the frequency interval you specify. You might need to specify additional data points in the S-Parameter file to fix the problem.

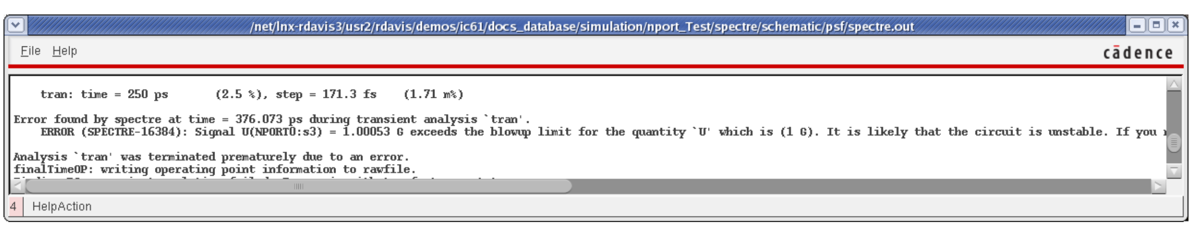
Difficulty can also sometimes be seen in running the transient analysis. Below is a *Choosing Analyses* form that shows a setup to run the transient to 10nsec.



The simulation runs, and the data plots, but note that only part of the waveform has been calculated.



When you look in the Spectre output log, the iterations have stopped because the solution goes above 1GigaVolt.



Model Reuse

Rational Fit

When *rational* is set for the interpolation method, Spectre caches the rational model in the simulation directory. As long as you do not change the S-Parameter file, the rational fit will be reused.

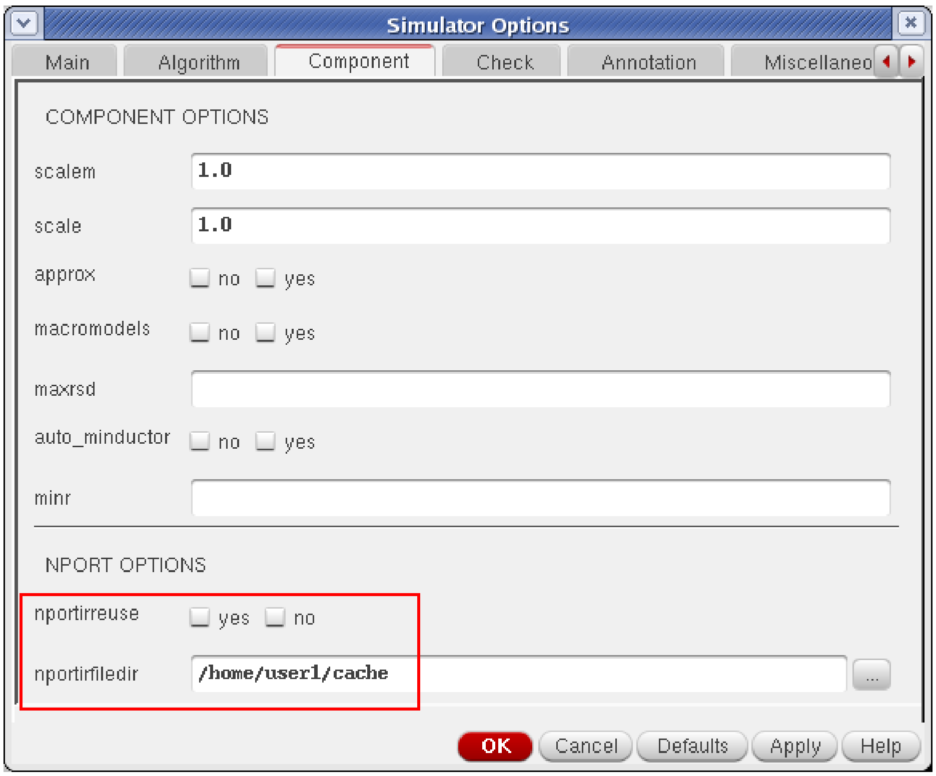
Linear and Spline Fit

Prior to MMSIM 12.1, every instance of every nport calculated the impulse response at the beginning of the simulation. In some cases, this added a lot of time in order to accomplish this. Starting with MMSIM 12.1, only the impulse responses that need to be calculated are performed. For example, if there are multiple instances of nport with the same S-Parameter file and the same nport settings, the impulse response is calculated only once.

The impulse response data is cached by default so that as long as the S-Parameter data and the properties on the nport do not change, the cached data is reused in subsequent simulations. When the properties on the nport instance are modified, then a new impulse response is needed. The impulse response is recalculated and cached without eliminating the original cached data file. This is done so that if you go back to the original settings, you do not need to re-run the impulse response.

Additionally, file locking has been implemented to allow multiple users to have access to a shared impulse response directory. This is controlled by the *nportfiledir* option described below.

There are two options that control caching. These options are located in the ADE options located in *Simulation-Options-Analog*.



Nportirreuse

This option has the default of *yes*. Keep this at the default to save time. This option enables caching of the impulse response.

nportirfiledir

This controls the location of the cached impulse response files. The default (blank) sets the location to `/home/<username>/.cadence/dfII/mmsim`. When a directory name is specified in the *nportirfiledir* option, the impulse response files are cached in the specified directory. This can be a shared directory between users, as long as all the users have write permissions in that directory.

Dcblock, dcfeed, indq, and capq

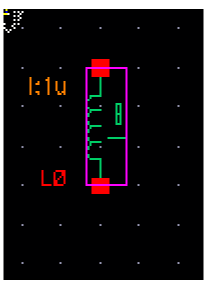
Ideal components are provided to allow what-if analysis without needing details of specific structures to allow the determination of what is needed to meet the specifications.

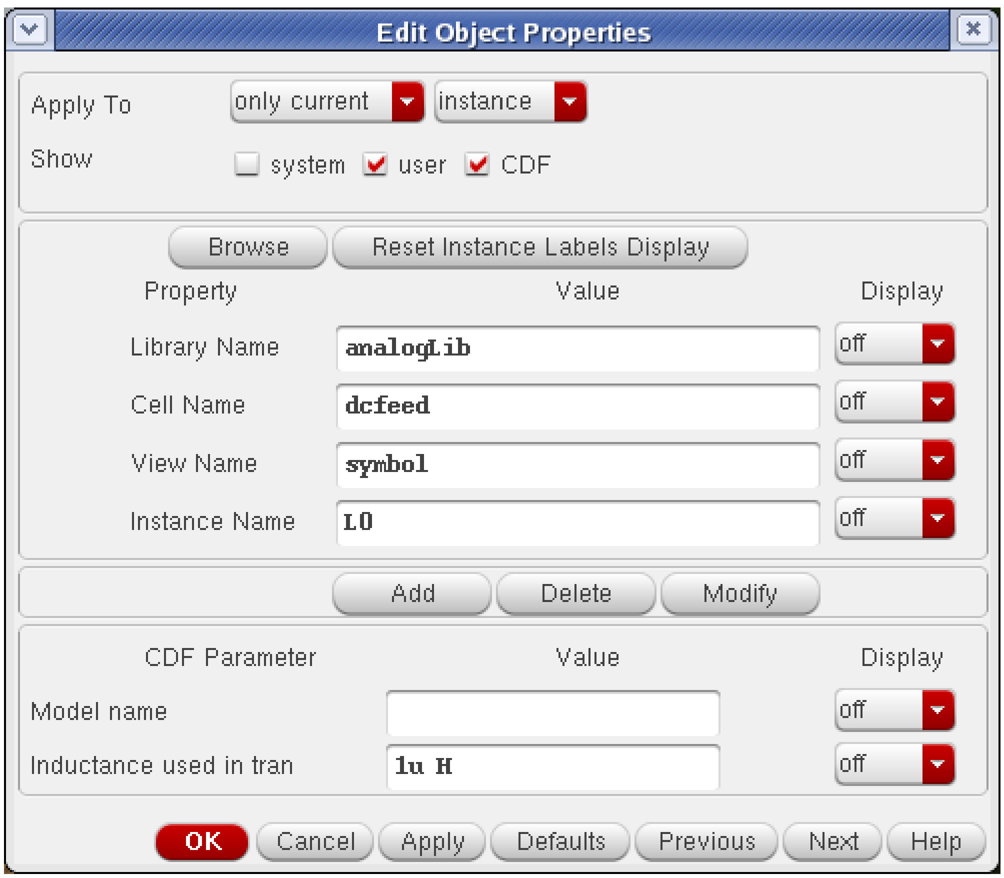
These components are provided for any analysis in Spectre except shooting. This allows the use of inductors and capacitors with Q in s-parameter, ac, and hbac analyses, along with the other small-signal Spectre and SpectreRF analyses.

Dcfeed

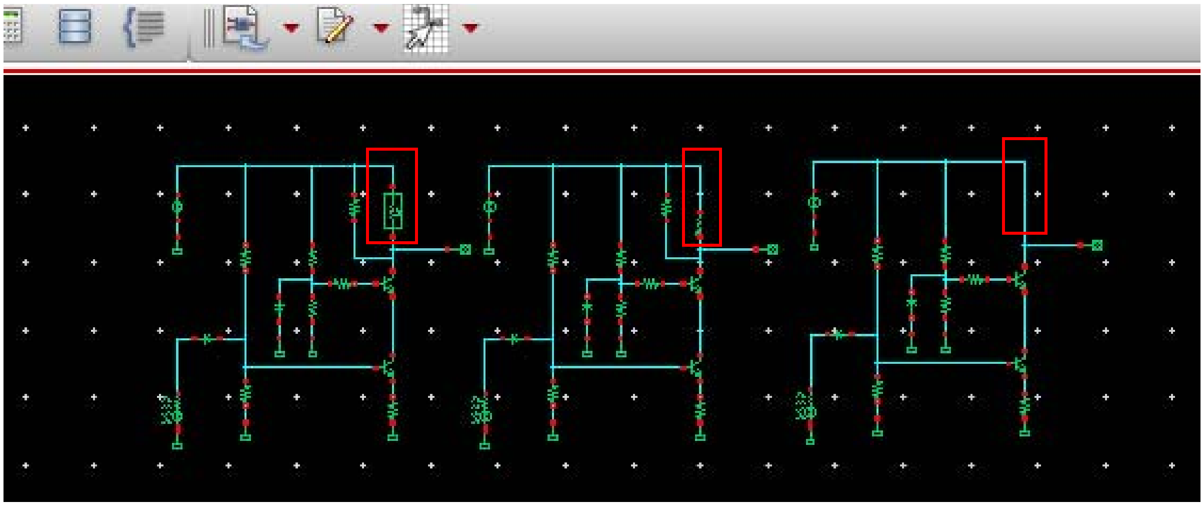
Dcfeed is an ideal short at DC and an ideal open at all other frequencies. For the transient analysis, dcfeed is an inductor with a default value of 1uh.

The symbol and the properties form are shown below. The *Model Name* field should be left blank.





An example comparing the dcfeed, and inductor, and a wire is shown below.



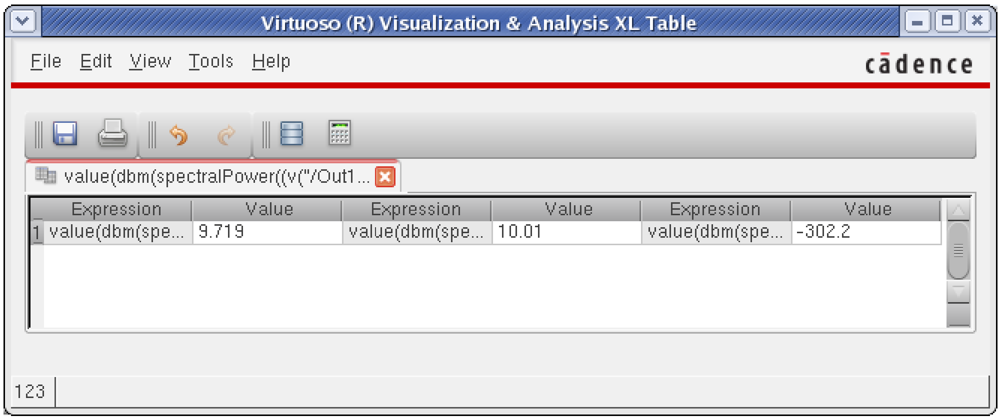
This example shows three cascode amplifiers with different connections to the power supply for the upper transistor. The left amplifier has a dcfeed. The center amplifier has an inductor. The right amplifier has a wire. In the amplifiers above, the resistor to the left has the output point set to 1K ohm.

When the inductor and the dcfeed are essentially open, the gain of the circuit is set by the resistor. When the impedance is less than 1Kohms, the load impedance is dominated by the impedance of the DCfeed or inductor. In this way, we can deduce what is happening at low frequency by running a simulation.

The amplitude of the input is deliberately large so the amplifiers tend toward the mixer behavior. The input signals are at 2.45GHz and 2.451GHz. The difference frequency is 1MHz. The 1 MHz signal will be analyzed so the behavior of the dcfeed, inductor, and wire can be evaluated.

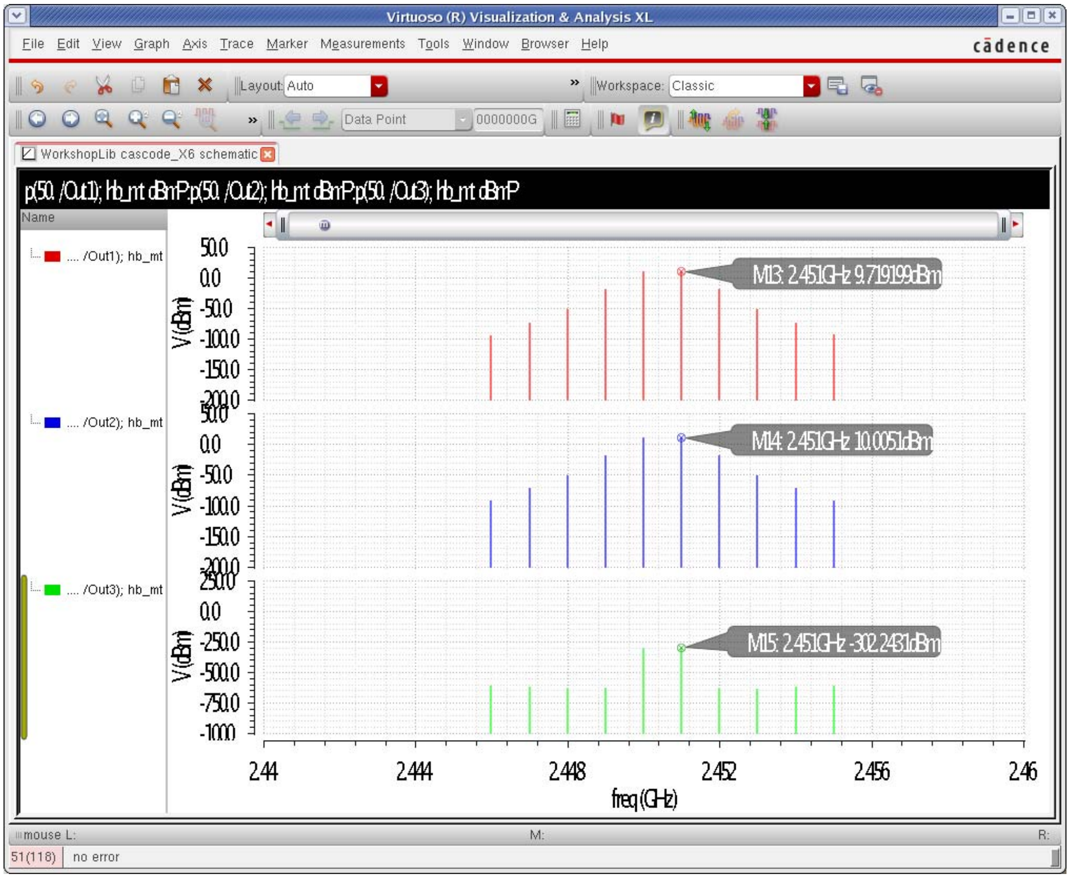
At all frequencies other than DC, the dcfeed is open. The inductor impedance goes up with frequency. At 1MHz, the impedance of the 1uh inductor is 6.28 ohms.

Below is the simulation result from harmonic balance near 2.45GHz. The result agrees for the dcfeed and inductor. With a wire, essentially no signal is present.

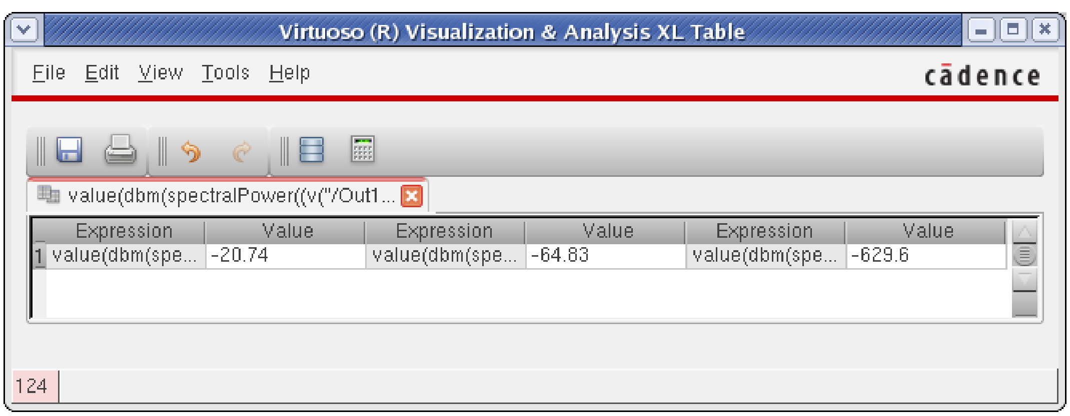


In the table above, the left result is the output signal level in dBm for the amplifier with the dcfeed, the middle is the signal level in dBm for the amplifier with the inductor, and the right is the output signal level in dBm for a wire connected to the power supply. The third output is at the numerical noise floor of the simulation.

Below is the spectral plot near 2.4GHz.

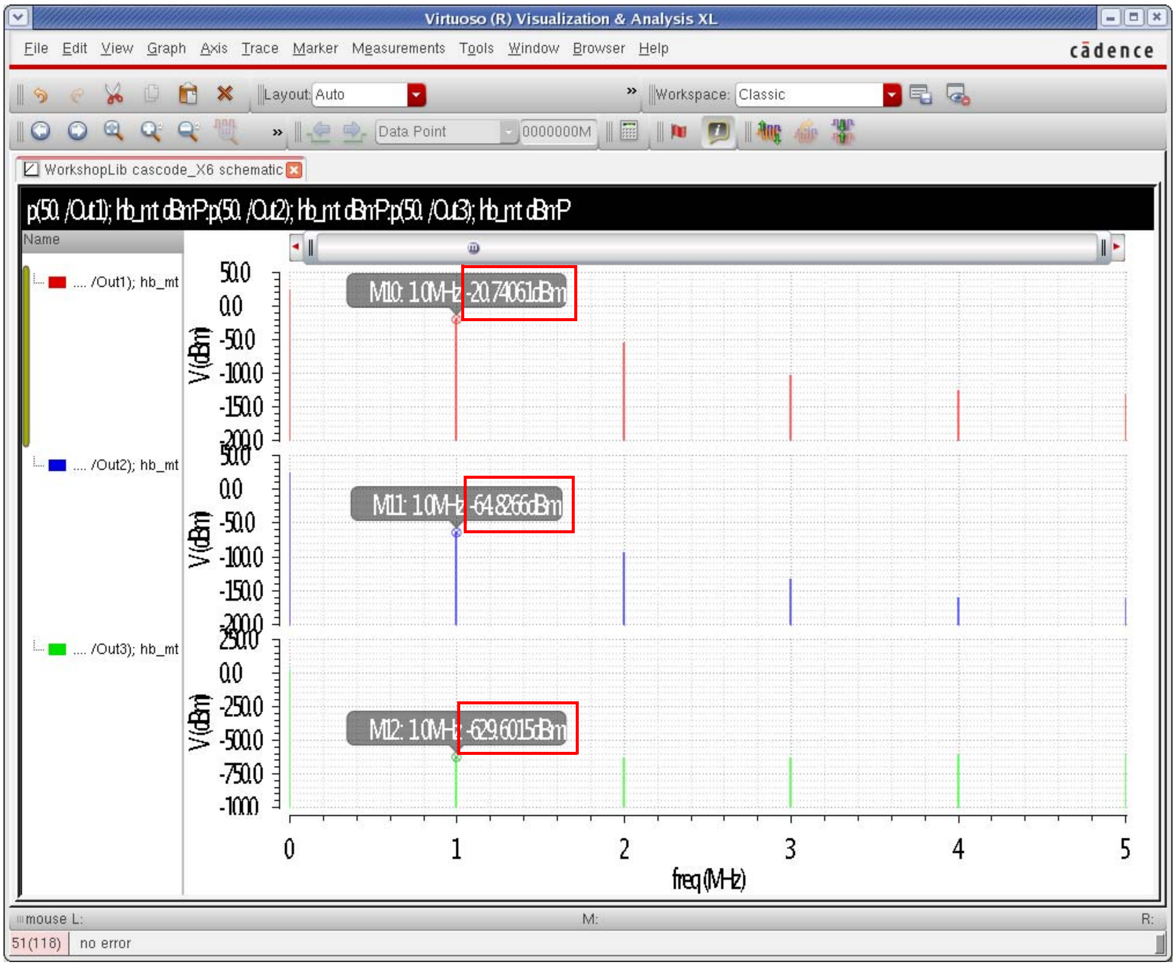


Below is the simulation result near 1MHz.



In the table above, the left result comes from the amplifier with the dcfeed. The center is with the inductor, and the right is with the wire. The dcfeed is open at 1MHz, so the signal level is set by the 1K resistor. Because the inductor has an impedance of only 6.28 ohms at 1MHz, the signal is much smaller than for the dcdfeed. With a wire, you see the numerical noise floor of the simulation.

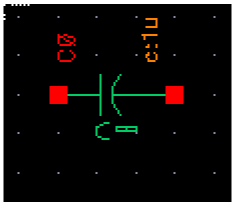
The spectral plot is shown below.

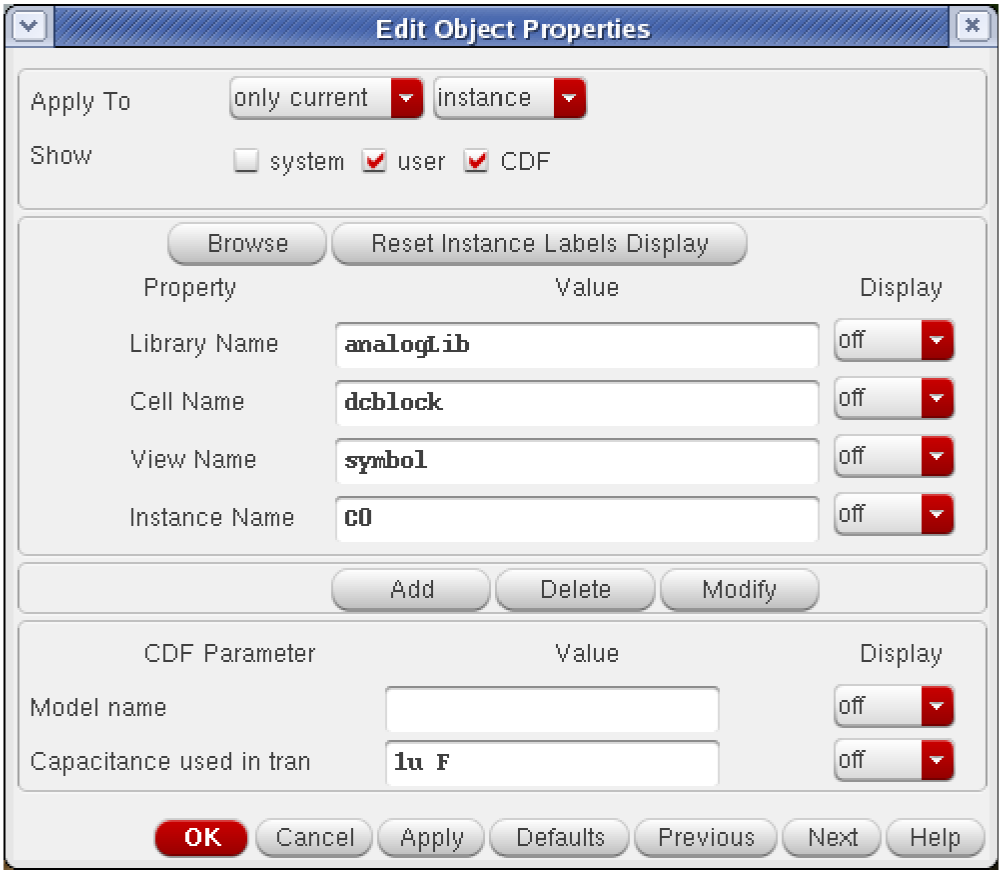


Dcblock

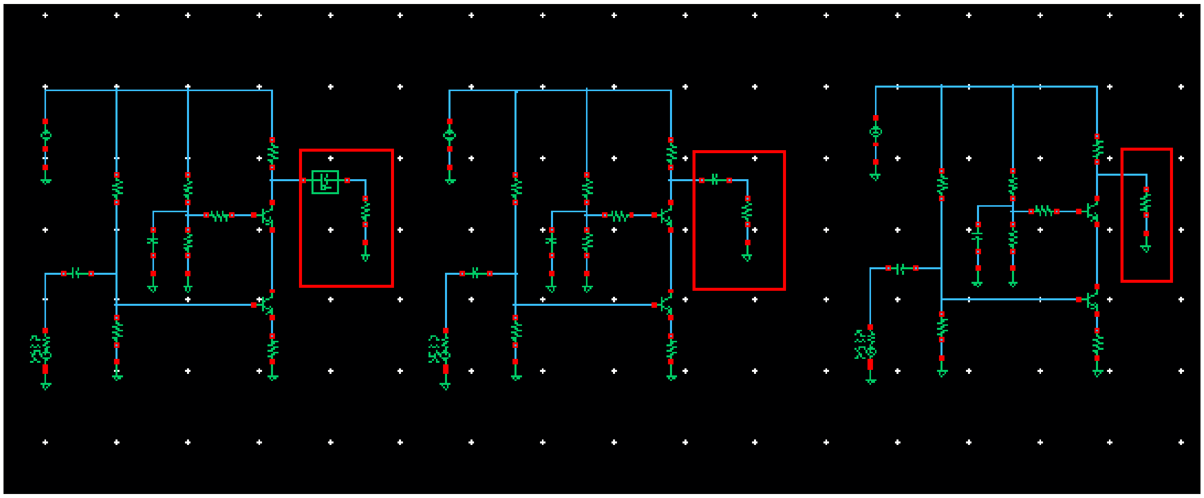
Dcblock is a near open (1Gohm) at DC and an ideal short at all other frequencies. For the transient analysis, dcfeed is a capacitor with a default value of 1uF.

The schematic symbol and properties list are shown below.

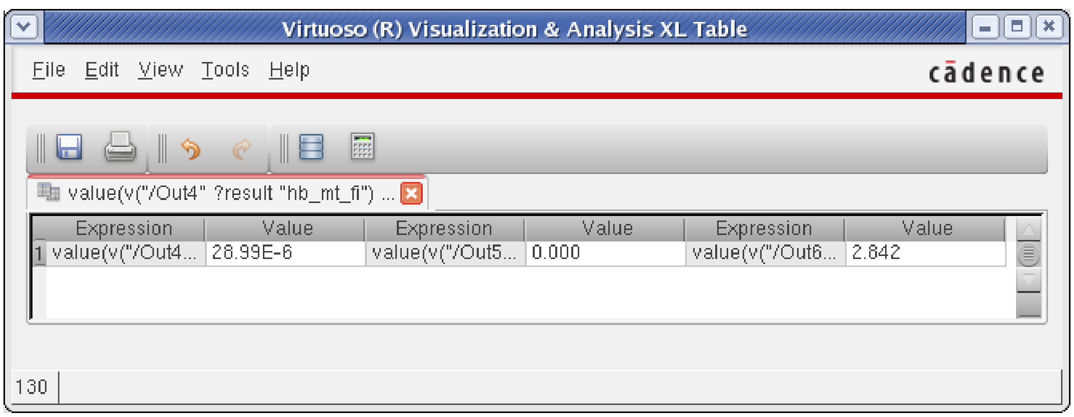




Below is an example with three cascode amplifiers that use different connections to the output load resistor. The left amplifier has a dcblock in series with the load resistor. The middle amplifier has a capacitor in series with the load resistor. The amplifier on the right has a wire connected to the load resistor. The output resistance is 10KOhms.



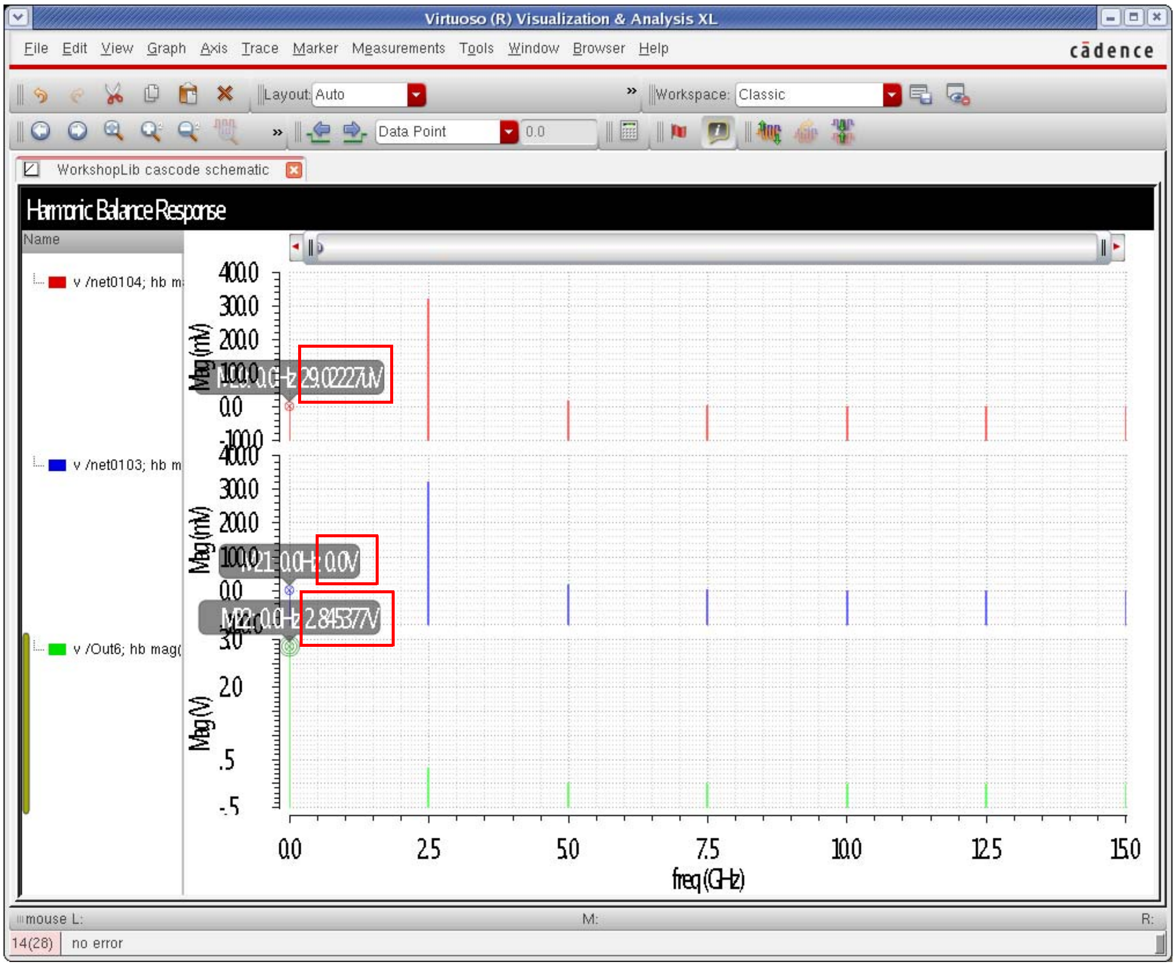
Here is a table with the DC values from the three cases. The left column is the DC voltage on the load resistor with the dcblock, the center column is the DC voltage with the capacitor, and the right column is the DC voltage with the wire.



The voltage with the dcblock is only a few microvolts. This is because it is a 1Gohm resistor at DC. The capacitor is a perfect open at DC, so the voltage is exactly zero. With a wire, the DC bias level appears at the load resistor.

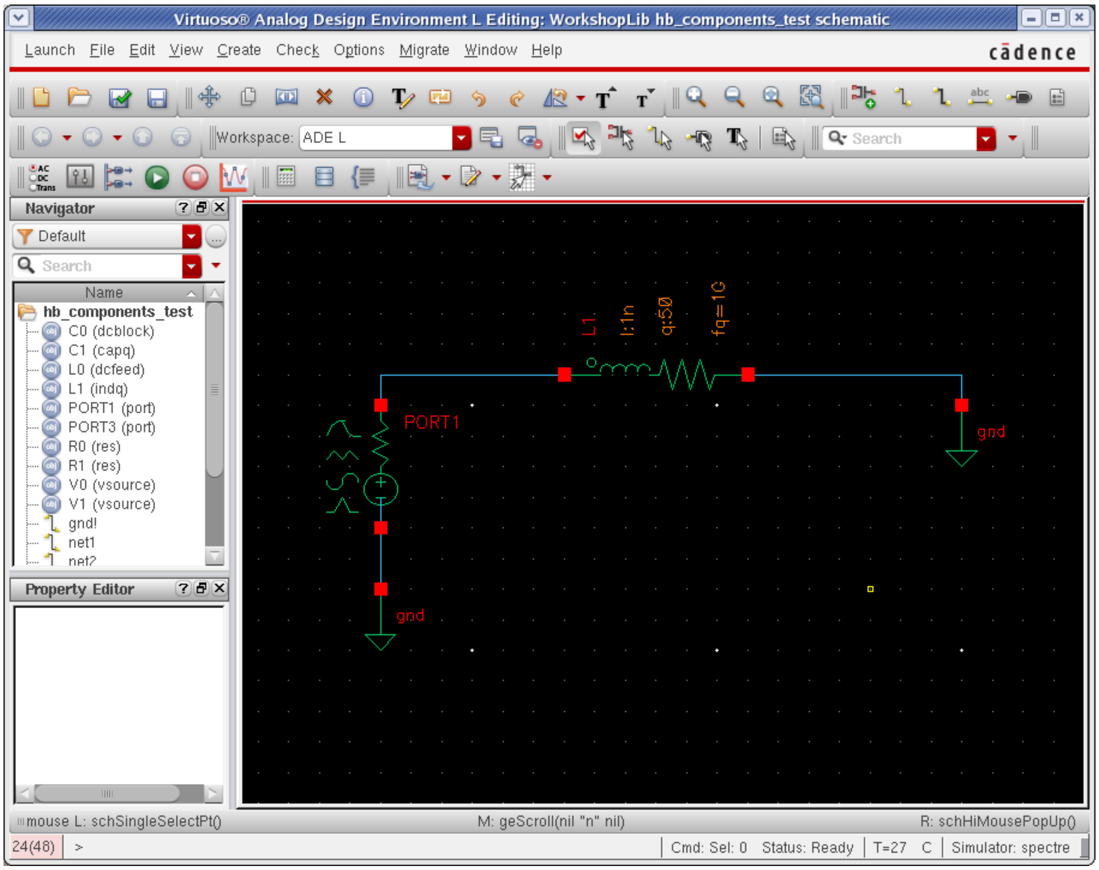
Here, is the simulation results from harmonic balance. Instead of dB, the vertical axis is linear volts. Note that the DC level is small for the dcblock. This is the top trace. The dcblock is a

1e9 ohm resistor at dc, and an ideal short at all non-zero frequencies. The center trace is from the amplifier with the capacitor, and the bottom trace is from the circuit with the wire.

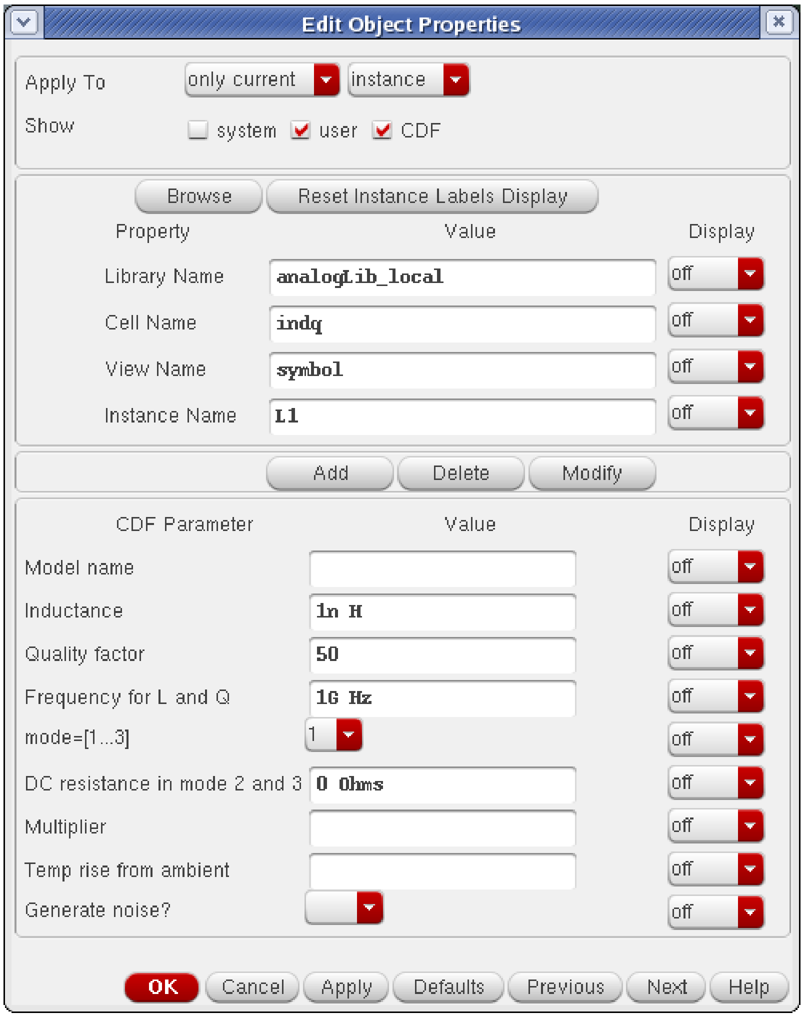


indq

Indq is a series inductor and resistor. There are three different modes with different frequency behavior in each mode. Below is the test circuit for the plots that will come later. PORT1 is defined as port1 in a single port S-parameter analysis so that the Z-Parameters for the inductor can be directly plotted.



The properties list for *mode 1* is shown below.



The inductor with Q works in all the analyses except for shooting.

The `indq` component has three modes, with different behavior in each mode. For all three modes, you need to specify the inductance, Q, and the frequency for the inductance and Q specification.

Mode 1

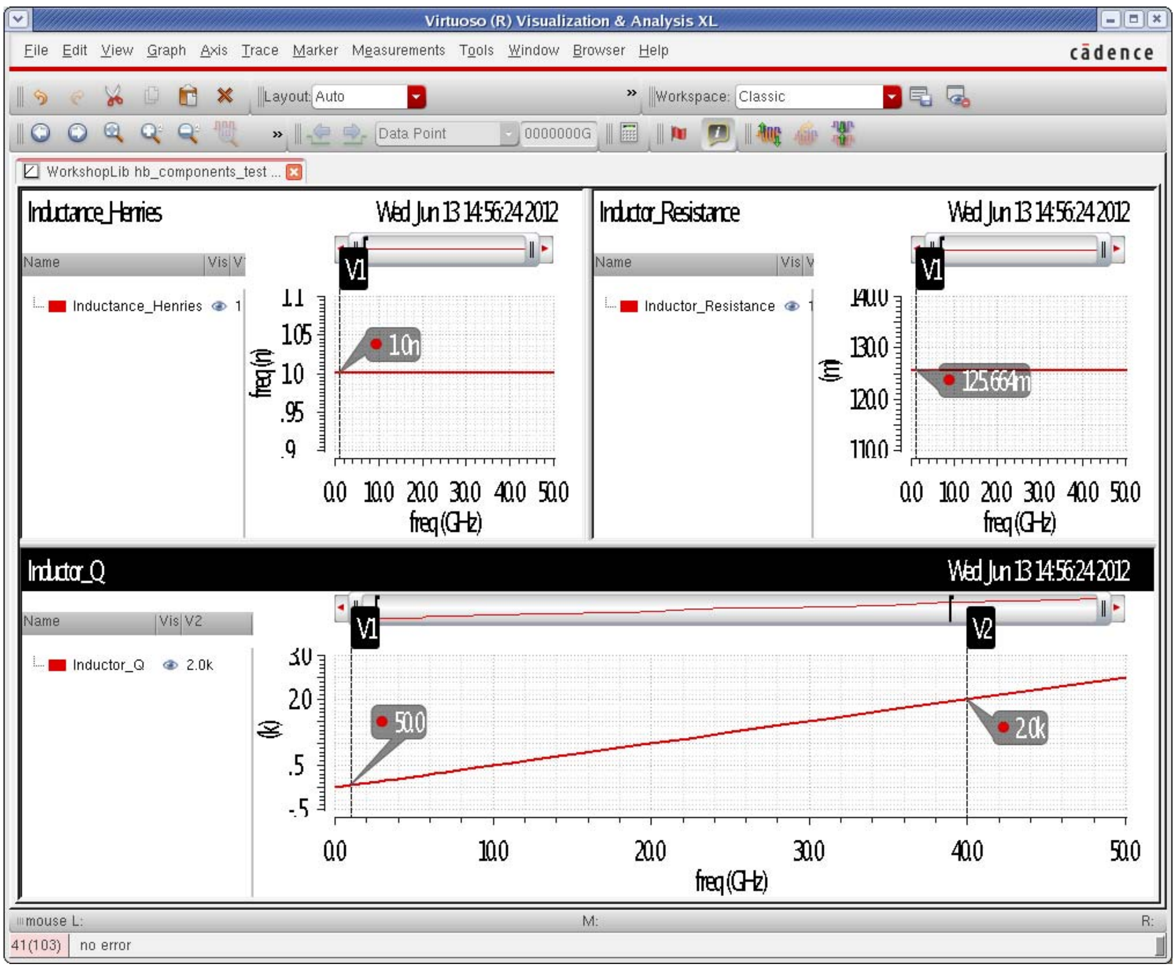
In this mode, the `indq` model is a series RL branch with R=constant and L=constant at the value given by the *Inductance* property. The value of the property *Rdc in mode2 and 3* is not used in this mode.

Special case: specifying Q=0 or F=0 makes the `indq` component just an inductor with the value L without any series resistance.

The resistance is calculated as follows:

$$R = \frac{\omega_Q L}{Q}$$

Below is a plot of the inductance, resistance, and Q with respect to frequency. A vertical marker has been placed at 1GHz, which is the frequency for L and Q. The inductance and resistance are constant, and the Q rises with frequency.



Mode 2

In this mode, both the resistance and the inductance are frequency dependent.

$$R(\omega) = R_{DC} + R''\sqrt{\omega}$$

$$L(\omega) = L\infty + R''\sqrt{\frac{1}{\omega}}$$

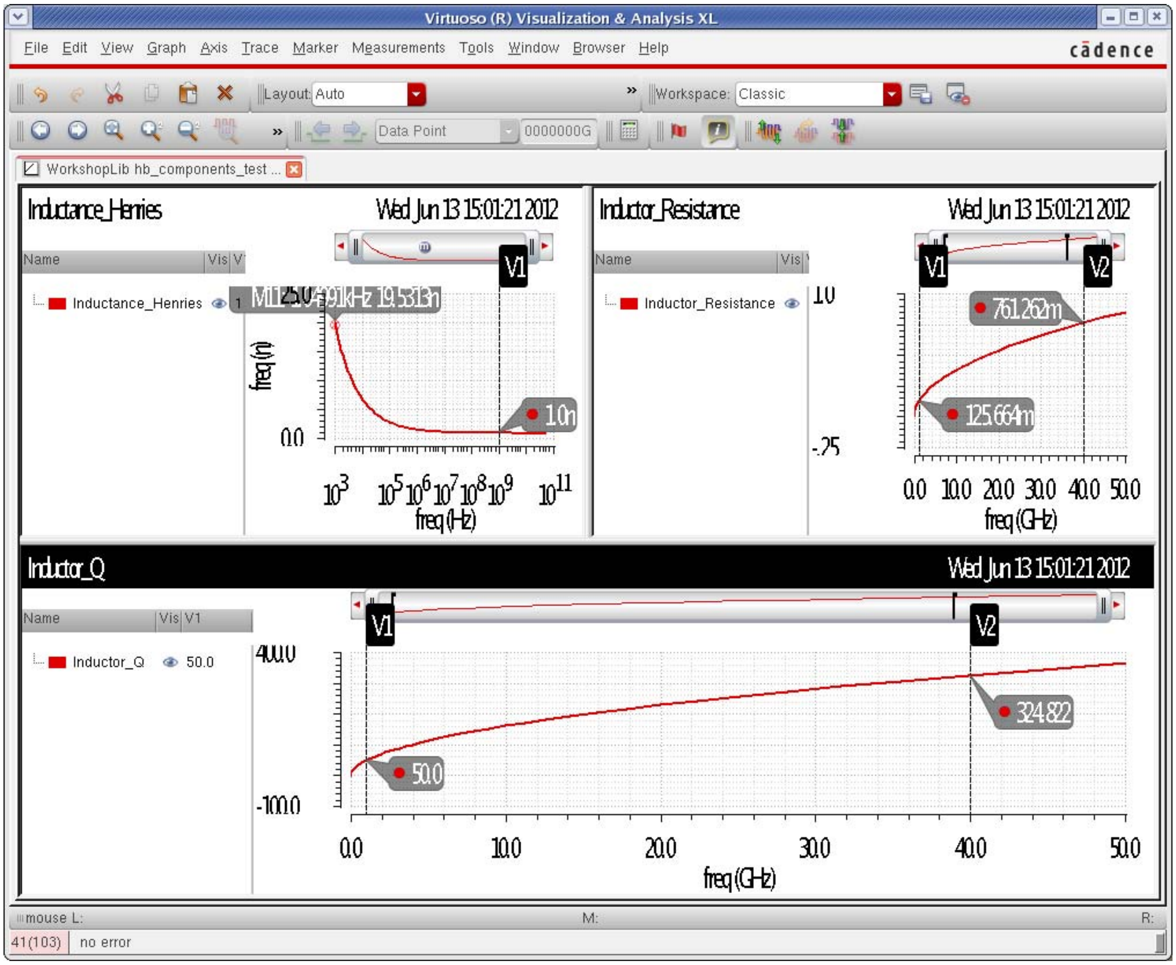
If R_{DC} in *mode 2 and 3* property is set to zero or is left blank, the DC resistance is 0.05*the resistor value that is set at the frequency defined by the *Frequency for L and Q* property.

R_{DC}, R", and $L(\infty)$ being constants are the classic forms of the skin-effect expressions for the resistance (grows as the square root of frequency) and inductance (the internal inductance decays inversely proportional to the square root of frequency). These expressions specify a causal frequency dependence of impedance. At the frequency specified in the *Frequency for L and Q property*, L and Q are equal to the values specified for *Inductance* and *Quality factor*.

The transient analysis cannot use frequency-dependent impedances, so it uses the *inductance* specified in the properties form, and the R_{DC} *in Mode 2 and 3* property for the resistance.

Below is the property form for mode 2.

Below is the inductance, resistance, and Q. Note the log frequency for inductance.



Mode 3

The advantage of mode 3 is that the non-physical frequency dependence of R near f=0 (DC) present with mode=2 is corrected. In this mode, R(f) has a reasonable frequency derivative near f=0. However, the expressions are non-causal as R varies as a function of frequency while L remains constant.

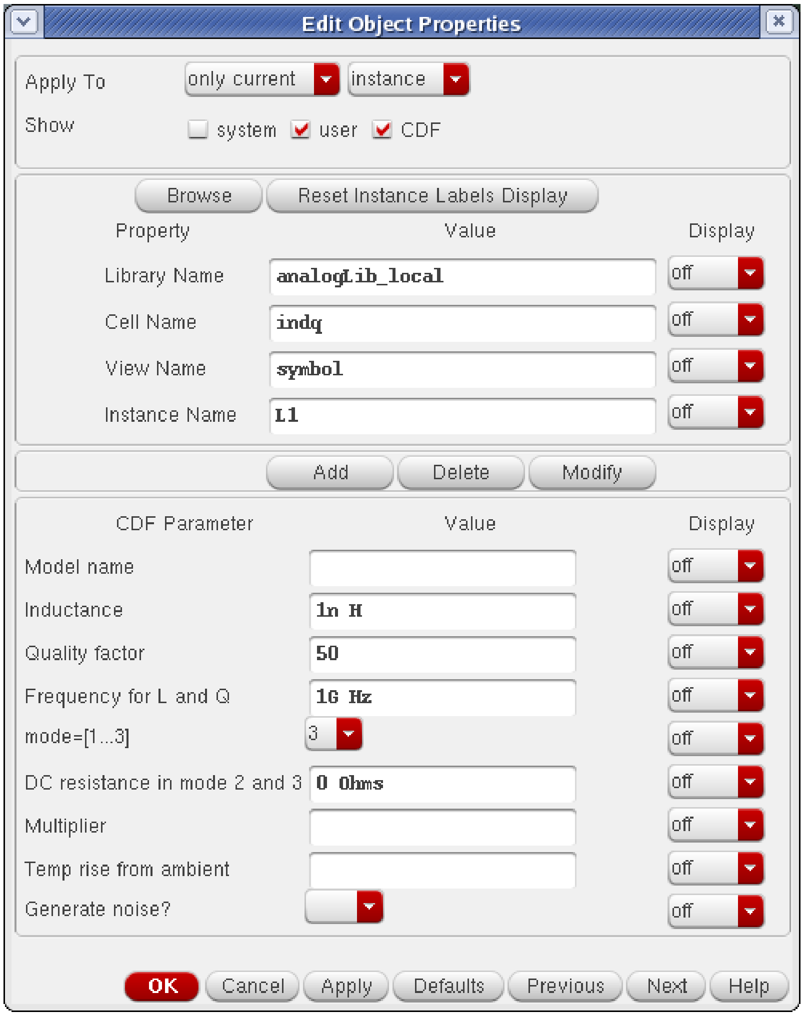
In this mode, L is constant at the value specified and

$$R = \sqrt{(R_{DC})^2 + \left(\frac{\omega L}{Q}\right)^2}$$

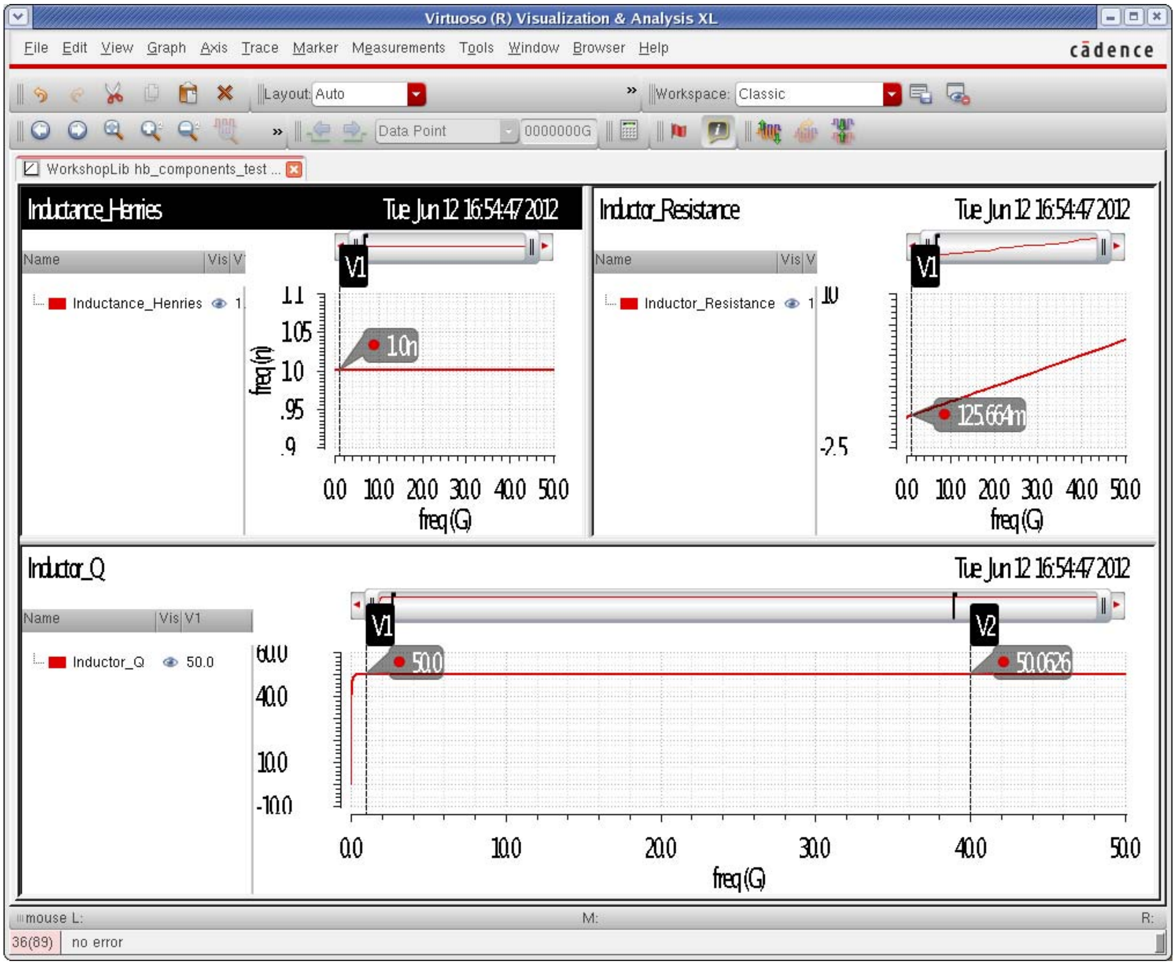
If the R_{DC} in *mode 2 and 3* property is set to zero, the DC resistance is 0.05*the resistor value that is set at the frequency defined by the *Frequency for L and Q* property.

Because the transient analysis cannot handle frequency-dependent resistance, the transient analysis uses the *inductance* specified in the properties form, and R_{DC} for the resistance.

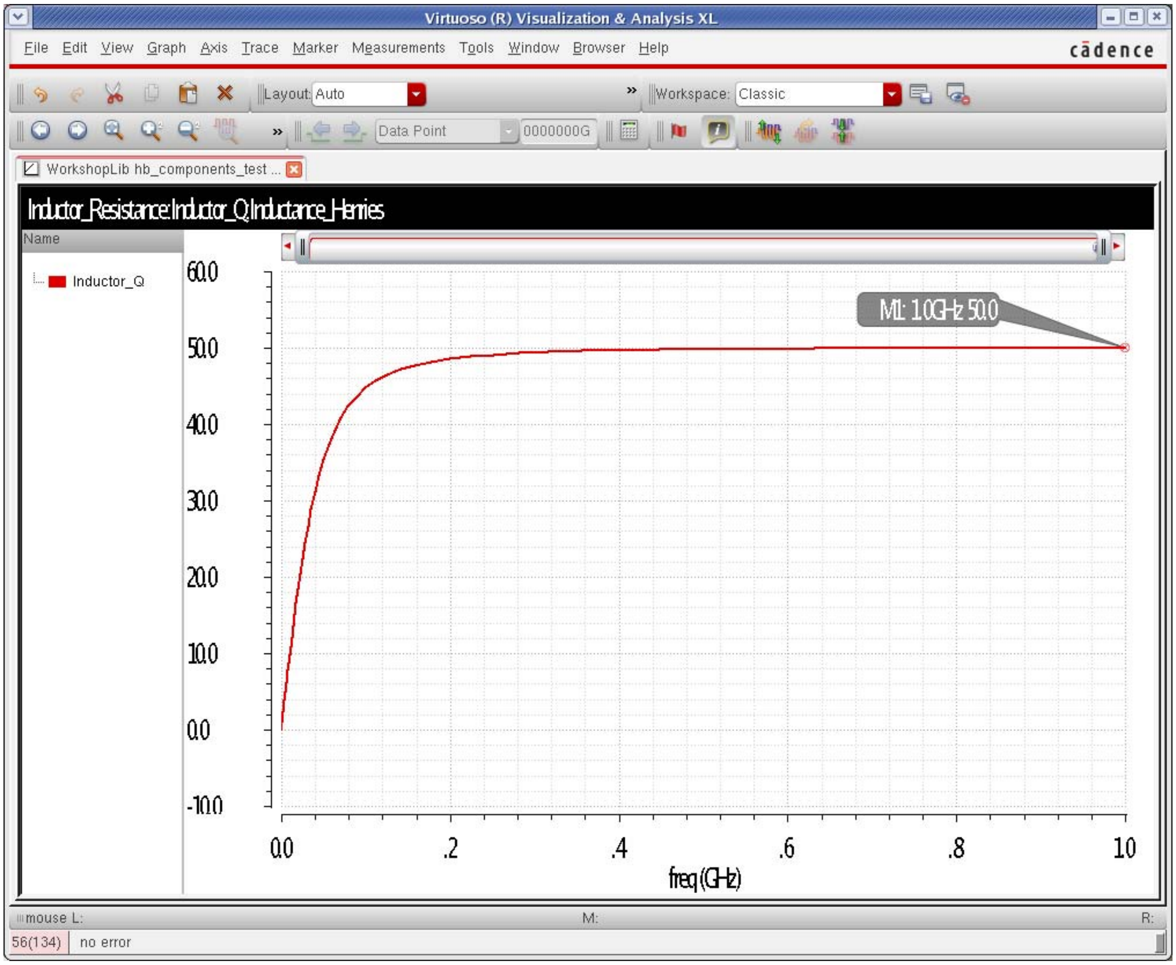
The properties list for mode 3 is shown below.



The inductance, resistance, and Q in mode 3 are plotted versus frequency, as shown below.

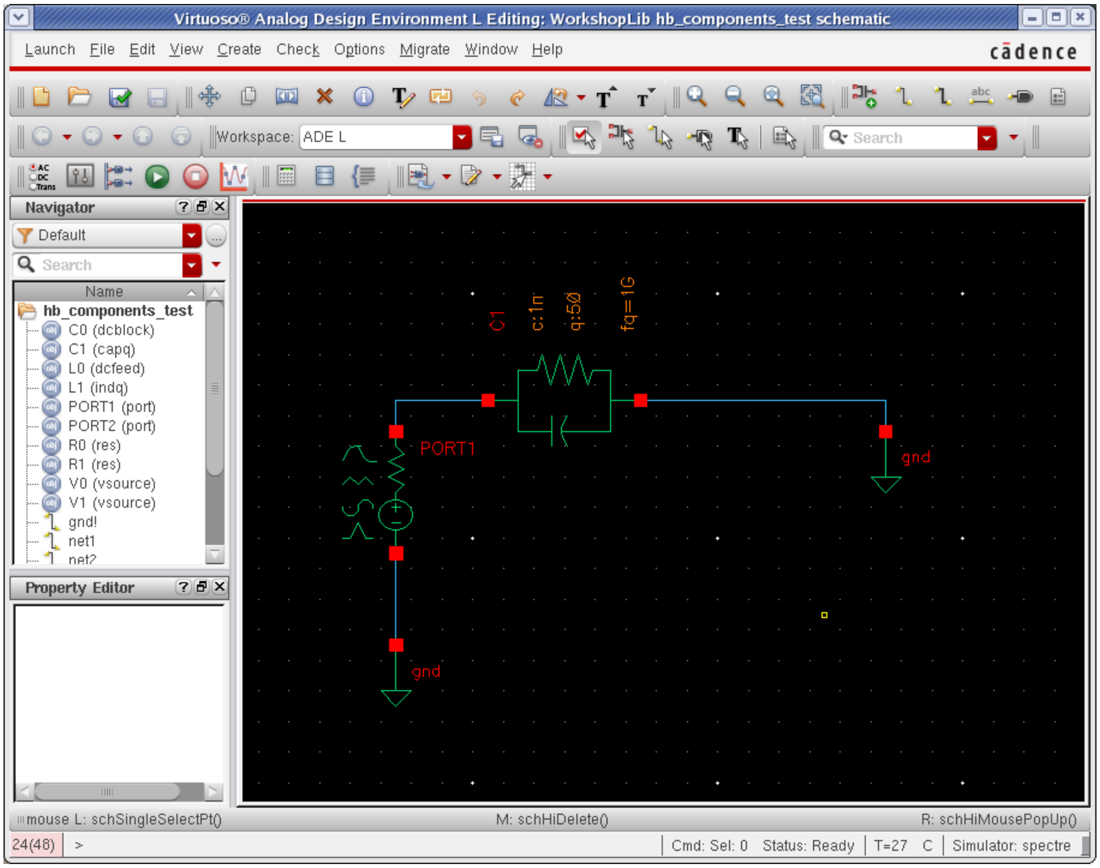


Here is an expanded view of Q at low frequency. Q drops off at low frequency.



capq

Capq is a parallel capacitor and resistor. There are three different modes with different behavior in each mode. In all three modes, the capacitor is constant at the value specified in the properties form. Below is the test circuit for the plots that will come later. PORT1 is defined as port1 in a single port S-parameter analysis so that the Y-Parameters for the capacitor can be directly plotted.



Mode 1

In this mode, the capq model is a parallel R C branch with R=constant and C=constant at the value given by the *Capacitance* property.

The conductance is constant as given below.

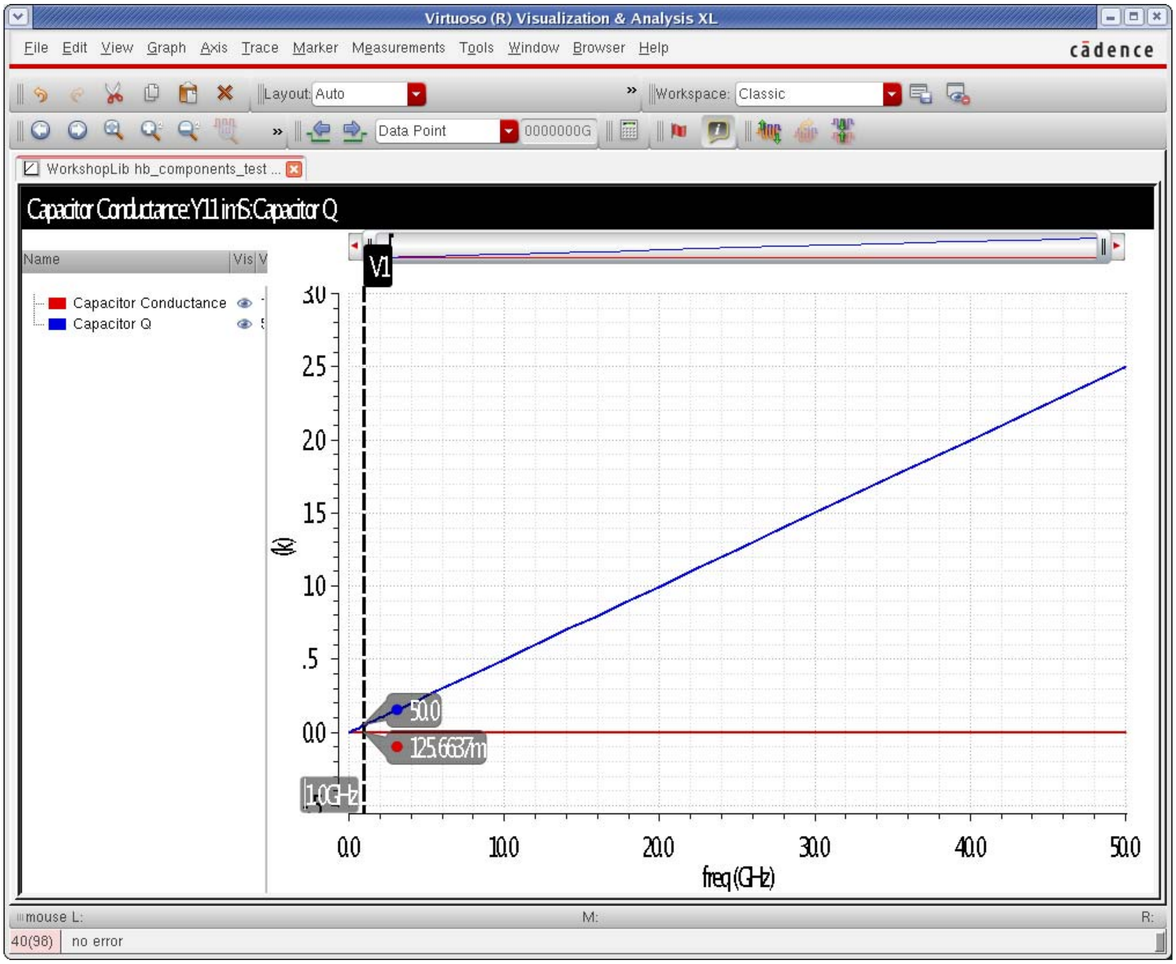
$$Real(Y) = \frac{C\omega_Q}{Q}$$

ω_Q above is the frequency for the Q value in radians per second.

This model is used in all the analyses, except shooting, where it is not supported.

Below is the properties list for the capq with mode 1 selected.

Below is a plot of the conductance, and Q with respect to frequency in mode 1. A vertical marker has been placed at 1GHz, which is the *frequency for Q*. The capacitance and resistance is constant, and the Q rises linearly with frequency.



Mode 2

In mode 2, the conductance is frequency dependent as shown in the equation below.

$$Real(Y) = \frac{C\sqrt{\omega \times \omega_Q}}{Q}$$

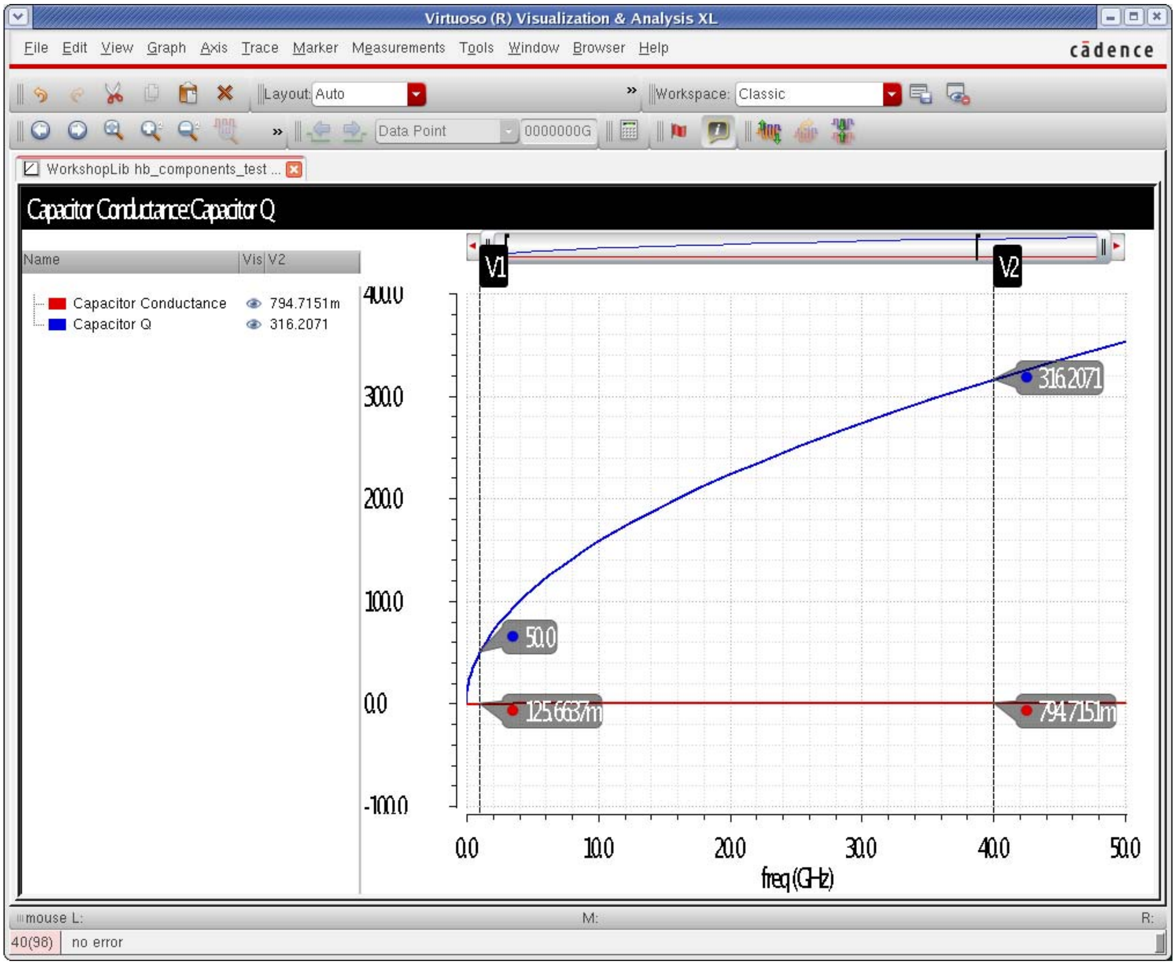
The capacitance is constant in all modes.

ω_Q above is the frequency where Q is specified in radians per second.

Mode 2 is non-causal. Because the resistance is frequency dependent, the transient analysis uses the capacitance specified in the properties list with no resistor in parallel.

Below is the properties form for mode 2.

Below is the conductance and Q versus frequency. Vertical markers have been placed at 1GHz and 40GHz. Mode 2 is a compromise between a linear Q increase with frequency in mode 1, and a constant Q in mode 3. The Q goes up with frequency, but not as fast as in mode 1.



Mode 3

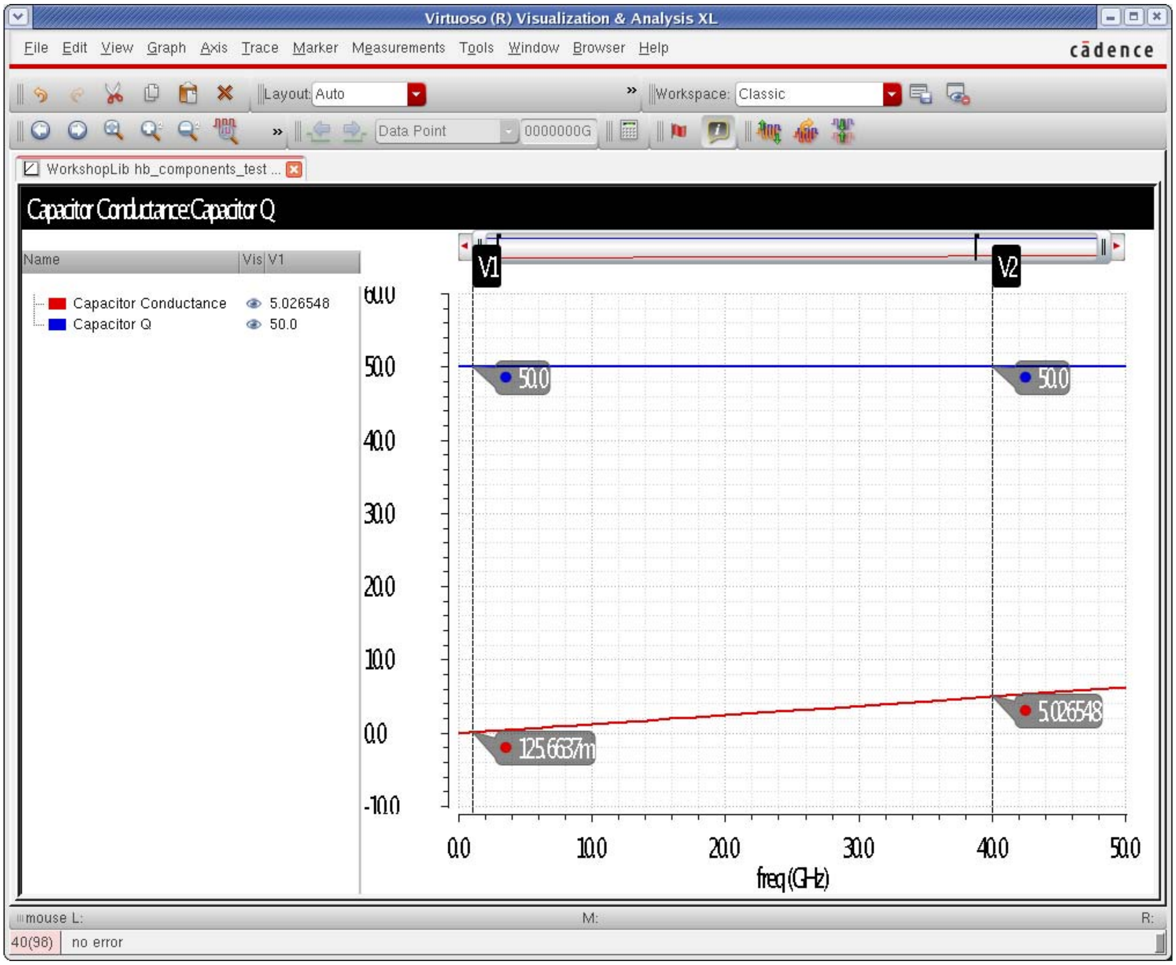
In mode 3, C is constant at the value specified in the properties form, and the conductance is

$$Real(Y) = \frac{\omega C}{Q}$$

Mode 3 is non-causal. Because the resistance is a function of frequency, the transient analysis uses just the capacitance term.

The properties list for mode 3 is shown below.

The conductance and Q versus frequency is plotted below.



In mode 3, the capacitance is constant, and the Q is constant. The resistance is a function of frequency.

Reference: S-Parameter Equations

This section describes the equations used by the VIVA Calculator to plot data generated by Spectre S-parameter simulations.

Using the VIVA Calculator in the Analog Design Environment, you can plot the following S-parameter data:

- Network Parameters
- Two-Port Scalar Quantities

- Two-Port Gain Quantities
- Two-Port Network Circles
- Equation for VSWR (Voltage Standing Wave Ratio)
- Equation for GD (group delay)

Use the buttons located in the *Function* area of the S-parameter *Direct Plot Form* to specify the type of data to plot.

Network Parameters

You can plot S, Y, Z, and H network parameters.

S-parameters, Y-parameters, and Z-parameters (denoted as SP, YP, and ZP on the *Direct Plot Form*) are defined for circuits with any number of ports. H-parameters (denoted as HP on the *Direct Plot Form*) are defined only for two-port circuits.

You can plot parameters on polar charts, Smith charts, or on rectangular plots after applying a *Modifier* option. The dB conversion uses $20\ \log_{10}X$ because the parameters represent scalar ratios (for example, voltage).

Equations for Network Parameters

For the ZP, YP, and HP parameters, Spectre returns S-parameters to the analog design environment. The environment converts them, as needed, to the equivalent Z, Y, and H matrixes using standard published methods. Spectre calculates S-parameter values.

SP (S-parameter) values

SP (S-parameter) values are calculated by Spectre.

ZP (Z-parameter) equation

The Z-parameter equation is as follows:

$$Z_m = [Z_{ref}][I + S_m][I - S_m]^{-1}[Z_{ref}]$$

Where

- S_m is the N-port S-parameter matrix

- I is the N x N identity matrix
- Z_{ref} is the characteristic impedance of the port
- Z_m is the resulting Z-parameter matrix

YP (Y-parameter) equations

The Y-parameter equations are as follows:

$$Y_m = [Y_{ref}][I - S_m][I + S_m]^{-1}[Y_{ref}]$$

Where

- S_m is the N-port S-parameter matrix
- I is the N x N identity matrix
- Y_{ref} is a diagonal matrix defined as

$$Y_{ref} = \frac{1}{\sqrt{\Re e\{Z_i\}}}$$

Where

  - Z_i is the terminating impedance at port i
  - Y_m is the resulting Y-parameter matrix

The HP (H-parameter) equations

The HP (H-parameter) equations only apply to two-port circuits.

D is

$$D = (1 - S_{11})(1 + S_{22}) + S_{21}S_{12}$$

H_{11} is

$$H_{11} = \frac{[(1+S_{11})(1+S_{22}) - S_{21}S_{12}]Z_{ref1}^{2}}{D}$$

H_{21} is

$$H_{21} = \left(\frac{-2S_{21}}{D}\right)\frac{Z_{ref1}}{Z_{ref2}}$$

H_{12} is

$$H_{12} = \left(\frac{2S_{21}}{D}\right)\frac{Z_{ref1}}{Z_{ref2}}$$

and H_{22} is

$$H_{22} = \frac{(1-S_{11})(1-S_{22}) - S_{21}S_{12}Z_{ref}^{2}}{D}$$

Where

- Z_{ref1} is the terminating impedance at port 1
- Z_{ref2} is the terminating impedance at port 2

Two-Port Scalar Quantities

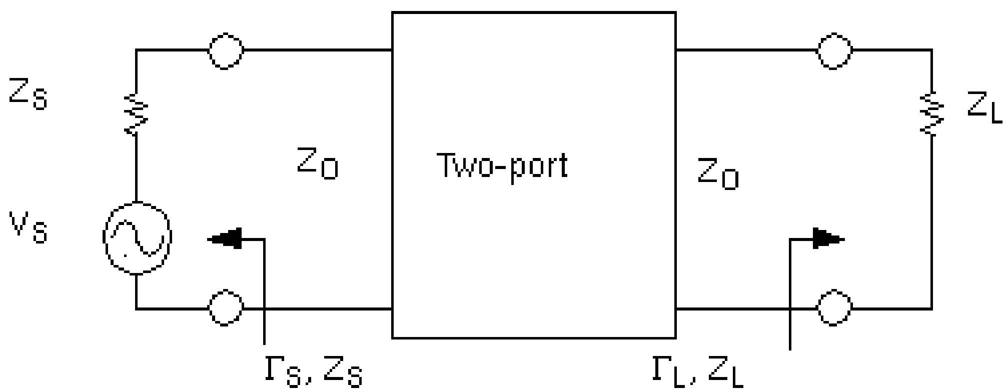
The two-port scalar quantities include

- NF_{min} is minimum noise figure
- NF is noise figure
- R_n is equivalent noise resistance
- G_{min} is Optimum Noise reflection coefficient

- B_{1f} is alternative stability factor
- K_f is stability factor

You can plot two-port scalar quantities only against frequency. In addition, you can plot them only on rectangular charts. Figure 8-20 on page 1870 illustrates a generic two-port circuit that defines impedances and reflection coefficients.

Figure 8-20 A Generic Two-Port Circuit



In Figure 8-20:

- Z_s is the source impedance
- Γ_s is the input reflection coefficient
- Z_L is the load impedance
- Γ_L is the load reflection coefficient
- Z_O is the characteristic impedance

Equations for Two-Port Scalar Quantities

G_{min} (Optimum Noise Reflection Coefficient) Equation

G_{min} (also known as Γ_{min} in the literature) is the reflection coefficient associated with minimum noise figure. You can plot G_{min} on a Smith chart or, by using the *Modifier* field, on a rectangular chart.

$$G_{min} = \Gamma_{min} = \frac{Z_{on} - Z_o}{Z_{on} + Z_o}$$

Where

- Z_o is the characteristic impedance
- Z_{on} is the source impedance associated with minimum noise figure (NF_{min})

NF_{min} (Minimum Noise Figure) and NF (Noise Figure) Equations

The NF_{min} and NF plots are controlled by the *Modifier* option in the *Direct Plot Form*.

Plot noise figure (NF) in dB by setting *Modifier* to *dB10*

- Plot noise factor (F) by setting *Magnitude*

Use NF_{min} and NF only for two-port circuits.

$$NF = 10\log_{10}F$$

Where

- F is the noise factor
- NF_{min} is the minimum noise figure

The NF_{min} values are calculated by Spectre

The NF (noise figure) equation is calculated by the Analog Design Environment from NF_{min}, G_{min} (Γ_{min}), and r_n. You can specify the optional source reflection coefficient Γ_S as an argument if you use the Analog Design Environment calculator. From the *Direct Plot Form*, the Analog Design Environment assumes Γ_S to be 0 (input matched to reference termination).

$$NF = NF_{min} + \frac{4r_n\left|r_s - r_{min}\right|^2}{\left(1 - \left|r_s\right|^2\left|1 + r_{min}\right|^2\right)}$$

Where

$$r_n = \frac{R_n}{Z_o}$$

Here r_n is the *normalized* equivalent noise resistance.

Rn (equivalent noise resistance)

R_n plots equivalent noise resistance. The R_n values are calculated by Spectre.

Stability Factors

K_f and B_{1f} plot the Rollet stability factor and its intermediate term. Use these parameters only for two-port circuits.

D is

$$D = S_{11}S_{22} - S_{21}S_{12}$$

K_f is

$$K_f = \frac{1 - |S_{11}|^2 - |S_{22}|^2 + |D|^2}{2|S_{21}||S_{12}|}$$

and B_{1f} is

$$B_{1f} = 1 + |S11|^2 - |S22|^2 - |D|^2$$

Two-Port Gain Quantities

The following gain quantities are valid only for two-port circuits:

- G_A (available gain) is the power gain obtained by optimally matching the output of the network.

- G_P (power gain) is the power gain obtained by optimally matching the input of the network.
- G_T (transducer gain) shows the insertion effect of a two-port circuit. This quantity is used in amplifier design.
- G_{umx} (maximum unilateral transducer power gain)
- G_{max} (maximum available gain) shows the transducer power gain when there exists a simultaneous conjugate match at both ports.
- G_{msg} (maximum stable gain) shows the gain that can be achieved by resistively loading the two-port such that $k = 1$ and then simultaneously conjugately matching the input and output ports. For conditionally stable two-ports, you can approach the maximum stable gain as you reduce the input and output mismatch. If you attempt a simultaneous conjugate match and $k < 1$, the two-port oscillates.

Equations for Two-Port Gain Calculations

G_A (Available Gain) Equations

Available gain equations for output conjugately matched.

$$G_A = \frac{|S_{21}|^2\left(1 - |\Gamma_S|^2\right)}{|1 - S_{11}\Gamma_S|^2\left(1 - |\Gamma_2|^2\right)}$$

Where

$$\Gamma_2 = S_{22} + \frac{S_{12}S_{21}\Gamma_S}{1 - S_{11}\Gamma_S}$$

Note: When you use the S-parameter *Direct Plot Form*, Γ_S is set to zero, and therefore available gain (G_A) is plotted as:

$$G_A = \frac{|S_{21}|^2}{1 - |S_{22}|^2}$$

To plot G_A for non-zero values of Γ_S, use the Analog Design Environment calculator.

G_P (Power Gain) Equations

Power gain equations for input conjugately matched.

$$G_P = \frac{|S_{21}|^2\left(1 - |\Gamma_L|^2\right)}{|1 - S_{22}\Gamma_L|^2\left(1 - |\Gamma_1|^2\right)}$$

Where

$$\Gamma_1 = S_{11} + \frac{S_{12}S_{21}\Gamma_L}{1 - S_{22}\Gamma_L}$$

Note: When you use the S-parameter *Direct Plot Form*, Γ_L is set to zero, and therefore the power gain, G_P is plotted as:

$$G_P = \frac{|S_{21}|^2}{1 - |S_{11}|^2}$$

To plot G_P for nonzero values of Γ_L, use the Analog Design Environment calculator.

G_T (Transducer Gain) Equations

$$G_T = \frac{\left(1 - \left|\Gamma_S\right|^2\right)\left|S_{21}\right|^2\left(1 - \left|\Gamma_L\right|^2\right)}{\left|(1 - S_{11}\Gamma_S)(1 - S_{22}\Gamma_L) - S_{12}S_{21}\Gamma_S\Gamma_L\right|^2}$$

Note: When using the S-parameter *Direct Plot Form*, the Analog Design Environment assumes that the source (Γ_S) and load (Γ_L) reflection coefficients are zero. G_T, therefore, plots the insertion gain.

$$G_T = \left|S_{21}\right|^2$$

Using the calculator, you can plot G_T and specify the source and load terminations.

G_{max} (Maximum Available Gain) Equations

For $K > 1$

$$G_{max} = \left|\frac{S_{21}}{S_{12}}\right|\left[K - \sqrt{K^2 - 1}\right]$$

For $K <= 1$

$$G_{max} = \frac{\left|S_{21}\right|}{\left|S_{12}\right|}$$

Where K is the stability factor, K_f

G_{umx} (Maximum Unilateral Transducer Power Gain) Equation

$$G_{umx} = \frac{|S_{21}|^2}{\left(1-|S_{11}|^2\right)\left(1-|S_{22}|^2\right)}$$

G_{msg} (Maximum Stable Power Gain) Equation

$$G_{msg} = \frac{|S_{21}|}{|S_{12}|}$$

Two-Port Network Circles

NC plots constant noise contours at the input of a two-port circuit. *GAC* plots constant gain contours at the input port, and *GPC* plots constant gain contours at the output port. Gain contour values reflect an optimum match at the opposing port.

Noise and Gain circles can be plotted at a single dB value for a range of frequencies or at a single frequency for a range of dB values. If you do not enter values for the frequency range, a circle is plotted for every simulated frequency for which a circle with the specified value exists.

SSB plots stability circles at the input port, and *LSB* plots stability circles at the output port. You can also specify a limited frequency range for these contours.

Equations for Two-Port Network Circle

NC (Noise Circle) Equations

$$N_i = \frac{(F_i - F_{min})|1+\Gamma_{min}|^2}{4r_n}$$

where

$$\Gamma_{min} = G_{min}$$

The center is calculated using

$$C_N = \frac{\Gamma_{min}}{1 + N_i}$$

The radius is calculated using

$$r_N = \sqrt{\frac{N_i^2 + N_i\left(1 - \left|\Gamma_{min}\right|^2\right)}{1 + N_i}}$$

Where i is the index number

GAC (Available Gain Circle) Equations

The center is calculated using

$$C_A = \frac{g_a(S_{11}{}^* - D^* S_{22})}{1 + g_a\left(\left|S_{11}\right|^2 - |D|^2\right)}$$

The radius is calculated using

$$r_A = \frac{\sqrt{1 - 2K_f\left|S_{21}S_{12}\right|g_a + \left|S_{12}S_{21}\right|^2 g_a^2}}{\left|1 + g_a\left(\left|S_{11}\right|^2 - |D|\right)^2\right|}$$

Where

$$G_a = \frac{G_A}{|S_{21}|^2}$$

And

$$D = S_{11}S_{22} - S_{12}S_{21}$$

GPC (Power Gain Circle) Equations

The center is calculated

$$C_P = \frac{g_p(S_{22}^* - D^*S_{11})}{1 + g_p\left(|S_{22}|^2 - |D|^2\right)}$$

The radius is calculated using

$$r_P = \frac{\sqrt{1 - 2K_f|S_{21}S_{12}|g_p + |S_{12}S_{21}|^2 g_p^2}}{1 + g_p\left(|S_{22}|^2 - |D|^2\right)}$$

Where

$$g_p = \frac{G_P}{|S_{21}|^2}$$

And

$$D = S_{11}S_{22} - S_{12}S_{21}$$

LSB (Load Stability Circle) Equations

The center is calculated using

$$C_L = \frac{S_{11}D^* - S_{22}{}^*}{|D|^2 - |S_{22}|^2}$$

The radius is calculated using

$$r_L = \left|\frac{S_{12}S_{21}}{|D|^2 - |S_{22}|^2}\right|$$

Where

$$D = S_{11}S_{22} - S_{12}S_{21}$$

SSB (Source Stability Circle) Equations

The center is calculated using

$$C_S = \frac{S_{22}D^* - S_{11}{}^*}{|D|^2 - |S_{11}|^2}$$

The radius is calculated using

$$R_S = \left|\frac{S_{12}S_{21}}{|D|^2 - |S_{11}|^2}\right|$$

Where

$$D = S_{11}S_{22} - S_{12}S_{21}$$

Equation for VSWR (Voltage Standing Wave Ratio)

VSWR is calculated from the S-parameters. You can plot the *VSWR* at any port in the circuit on a rectangular chart.

$$VSWR_i = \frac{1 + |S_{ii}|}{1 - |S_{ii}|}$$

Where `i` is the port number.

Equation for ZM (Input Impedance)

You can plot input impedance if all other ports are matched.

$$Z_m = \frac{1 + S_{ii}}{1 - S_{ii}}R$$

Where `R` the reference impedance of the port of interest and `i` is the port number.

Equation for GD (group delay)

GD (group delay) approximates the derivative of the phase with respect to frequency, normalized to 360 degrees. Units for group delay are in seconds.

$$G_d \cong \frac{-d\phi}{d\omega}$$

Group delay is calculated from the phase of the corresponding S-parameter (for example, GD_{21} corresponds to S_{21}).

References

To learn the technical details about how SpectreRF converts S-parameter data to a time-domain description, see the article “Robust rational function approximation algorithm for model generation,” by C. P. Coelho, J. R. Phillips, and L. M. Silveira. This article appeared in the proceedings of the 36th Design Automation Conference, New Orleans, LA, June 1999.

9

Cosimulation with MATLAB® and Simulink®

This chapter describes how to set up and use a cosimulation link between the MATLAB® and Simulink® system-level simulation environment and Virtuoso® Spectre® circuit simulator RF analysis (SpectreRF). The sections in this chapter are:

- Introduction to Cosimulation with MATLAB on page 1884
- Software Requirements on page 1884
- Setting Up and Running a Cosimulation on page 1884
- Connecting the Coupler Block Into the System-Level Simulink Schematic on page 1885
- Determining How You Want to Start and Run the Cosimulation on page 1889
- Generating a Netlist for the Lower-Level Block on page 1889
- Running the Cosimulation on page 1898
- MATLAB Support Matrix on page 1900

Introduction to Cosimulation with MATLAB

Cosimulation combines the best of system-level simulation with lower-level analog and RF simulation. Simulink provides large libraries of DSP algorithms for generating complicated signals and post processing while SpectreRF supports transient and envelope analysis of common RF and communication circuits such as mixers, oscillators, sample and holds, and switched capacitor filters at both the transistor and behavioral levels.

Cosimulation makes it easier

- To detect concept errors early
- To detect design flaws before tape-out
- To quickly correct issues and re-simulate

The system-level design in Simulink serves as a golden reference. System designers can use the Simulink system-level design as a testbench for implementing and simulating the design. Unfortunately, when system-level designs are simulated by themselves, the effects originating from subsystems are often not considered. With cosimulation, system designers can create low-level models of critical analog blocks and use these models one at a time to analyze the performance impact of individual blocks on the system-level simulation.

Software Requirements

Cadence recommends that you use Cadence software version MMSIM 7.1 or later.

Setting Up and Running a Cosimulation

To prepare for and run a cosimulation, you

1. Connect the coupler block into the system-level Simulink schematic.
2. Determine how you want to start and run the cosimulation.
3. Generate a netlist for the lower-level block that reflects how you want to start and run the cosimulation.
4. Run the cosimulation.

These steps are described in greater detail in the following sections.

Before you continue, however, be sure that the programs are ready to run.

➤ Add the SpectreRF and MATLAB/Simulink engine install paths to the `MATLABPATH` environment variable.

You can automate this step by adding the appropriate command to your `.cshrc` file. For example, if you are using a C shell, you can add the command

```
setenv MATLABPATH `cds_root spectre`/tools/spectre/simulink:${MATLABPATH}
```

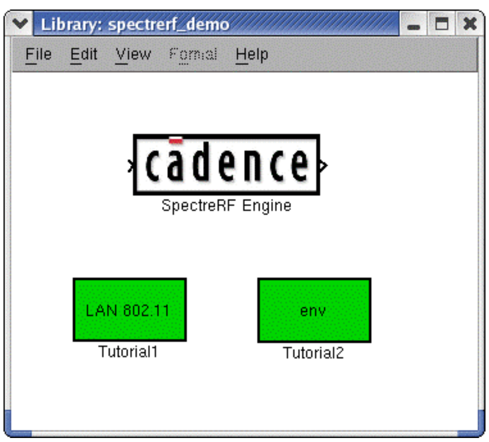
Connecting the Coupler Block Into the System-Level Simulink Schematic

To prepare the Simulink part of the design for cosimulation,

1. Start MATLAB, by typing

```
matlab &
```

The Simulink library opens.



This library contains the coupler module (distinguished by the Cadence logo and labeled `SpectreRF Engine`). You can insert the coupler module in any Simulink design by dragging and dropping it from this library.

2. Open your testbench or high-level design.

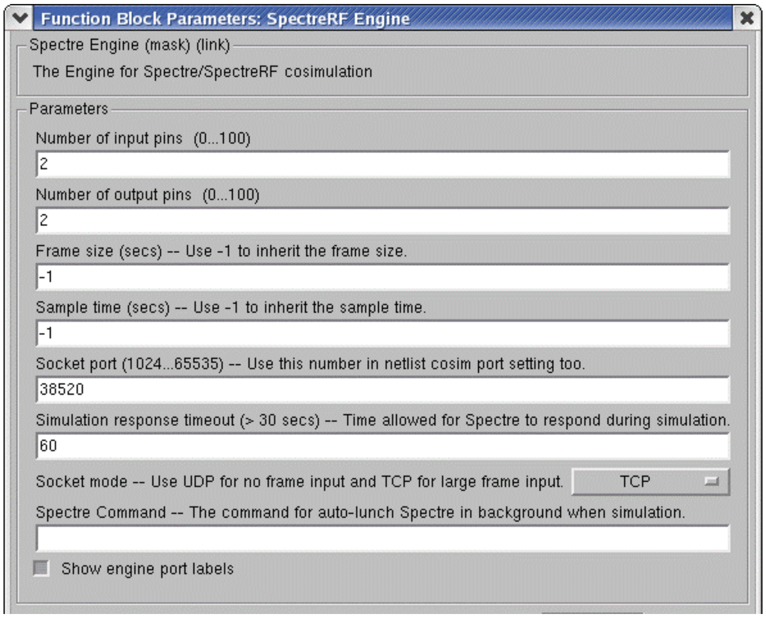
3. Drag and drop the `SpectreRF Engine` coupler block into the testbench and place it in the design.

4. Connect the `SpectreRF Engine` block into the design.

To make a signal connection, move the mouse pointer over the module pin. The pointer changes to a cross. Press the left mouse button, move the cursor to the destination pin and release the button.

5. Double-click the `SpectreRF Engine` block.

This opens the Function Block Parameters: SpectreRF Engine window where you can edit the coupler block's parameter values.



The fields have the following meanings:

Field	Meaning
Number of input pins	The number of input pins to the `SpectreRF Engine` block (0…100). The input to the block is the input signal from the Simulink design.
Number of output pins	The number of output pins from the `SpectreRF Engine` block (0…100). The output from the block is the signal generated by the SpectreRF simulator.
Frame size	The number of samples per frame (any positive integer or -1; default: -1). Allows you to use the coupler in a loop where frame size inheritance does not work. Simulink automatically detects conflicting frame sizes. Set this value to -1 to inherit the frame size from the input.

Sample time	Sample period of the block, in seconds (default: -1). The *Sample time* is the frame period, not the time between the samples inside the frame. Allows the coupler to have only outputs, where the Simulink coupler acts as a source and defines a *Sample time*. Also allows the coupler to have only inputs. Set this value to -1 to inherit the sample period of the connected blocks. See the MATLAB and Simulink *Help* menu for more information on *Sample time* propagation.
Socket port	The number of the service that is identified by TCP/IP. Normally, the system reverses port numbers less than 1024, so this value can range from 1024 to 65535 (default: 38520).
Simulation response timeout	Maximum time, in seconds, to wait for an answer from the SpectreRF simulator during simulation. Increase this time if the simulator requires a long calculation time for each sample or frame (default: *120*).
Socket mode	TCP or UDP. Choose UDP if frame size is less 50. TCP and UDP are two different translate protocols in TCP/IP. Without CRC, UDP is usually faster in a good net environment with small data packages and TCP is usually better for large data packages.
Spectre command	Used to run a SpectreRF simulation. This parameter enables a use mode where SpectreRF is called internally by the Simulink simulation.
Show engine port labels	If checked, the `SpectreRF Engine` shows label information.

6. Set the number of input pins and output pins as well as any other parameters you need to set.

7. Note the Spectre command field but do not change the value now.

Depending on how you choose to run the cosimulation, you might need to return to this field and type in a `spectre` command.

8. Click *OK*.

The form closes and the `SpectreRF Engine` block is updated with the correct number of pins.

9. (Optional) Make other necessary changes in the Simulink testbench window.

   For example,

   - ❑ Choose *Format – Port/Signal Displays – Signal Dimensions*.

     This switches the signal dimension display on so you can see details of framed signals.

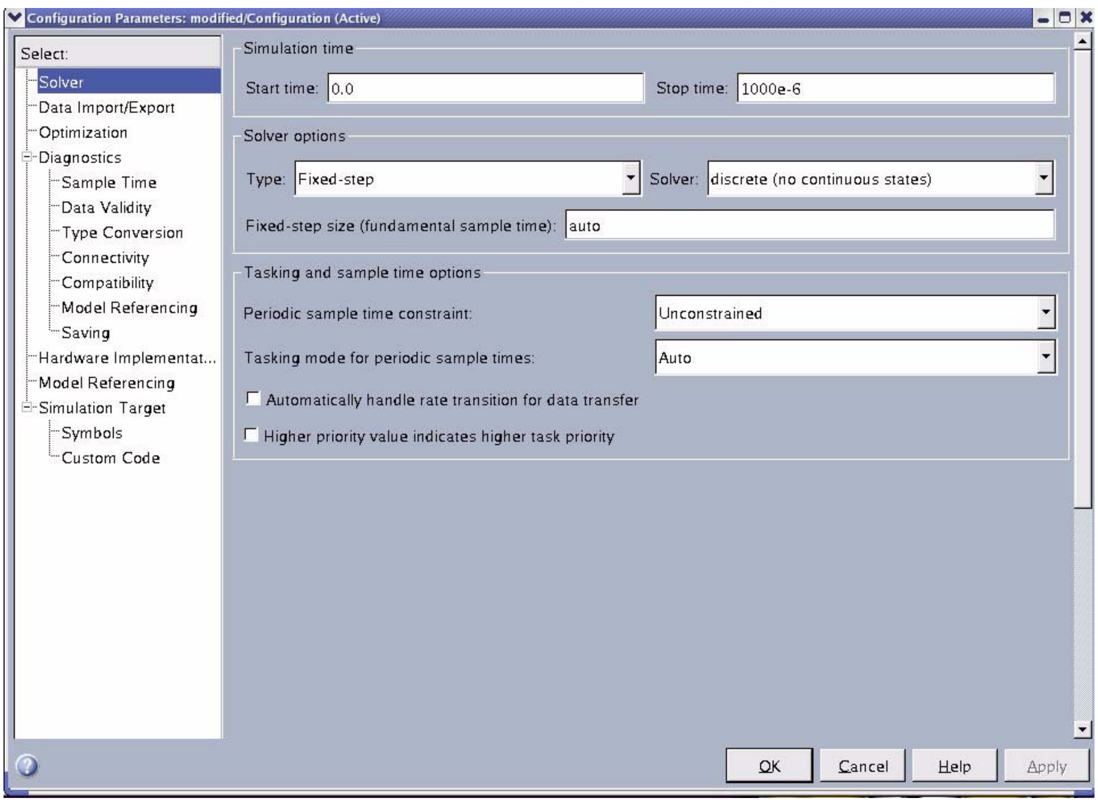
   - ❑ Choose *View – Simulink library* to display the library browser. Double-click *Math Operations*.

     The library appears. Here you can select Simulink converters, such as a `Complex to Real-Imag` block. Drag and drop blocks into the schematic and connect them as necessary.

   Double-click any placed blocks and set appropriate values.

10. Choose *Simulation – Configuration Parameters*.

    The Configuration Parameters form opens.



11. In the *Stop time* field, set the stop time.

12. Click *OK*.

The Configuration Parameters form closes.

13. Save the modified design by choosing *file – save as*.

This completes the necessary modifications to the Simulink testbench.

Determining How You Want to Start and Run the Cosimulation

There are three ways to run a cosimulation, after all the setup is finished.

1. You can start the two applications (SpectreRF and MATLAB) separately.

   This method is appropriate if you need to be able to modify both the system-level Simulink design and the analog circuit.

2. You can start ADE and arrange to have MATLAB start automatically.

   This method is appropriate if you are an analog design who needs to validate a circuit with system-level design input and output.

3. You can start MATLAB and arrange to have SpectreRF start automatically.

   This method is appropriate if you are a system-level designer, because, after set up, you can use the SpectreRF Engine just like another block in Simulink.

The setup differs for each of these approaches so it is useful to decide which is most appropriate for your design before continuing.

Generating a Netlist for the Lower-Level Block

The previous section describes how to insert a coupler block into the system-level design. That defines one end of the connection but you must still establish a connection with the lower-level analog block that is simulated by SpectreRF. To do that, you insert a `cosim` statement into the netlist, either by hand or by using the ADE environment.

Preparing the Netlist When Using ADE

This section describes how to prepare a netlist for cosimulation using the Analog Design Environment (ADE).

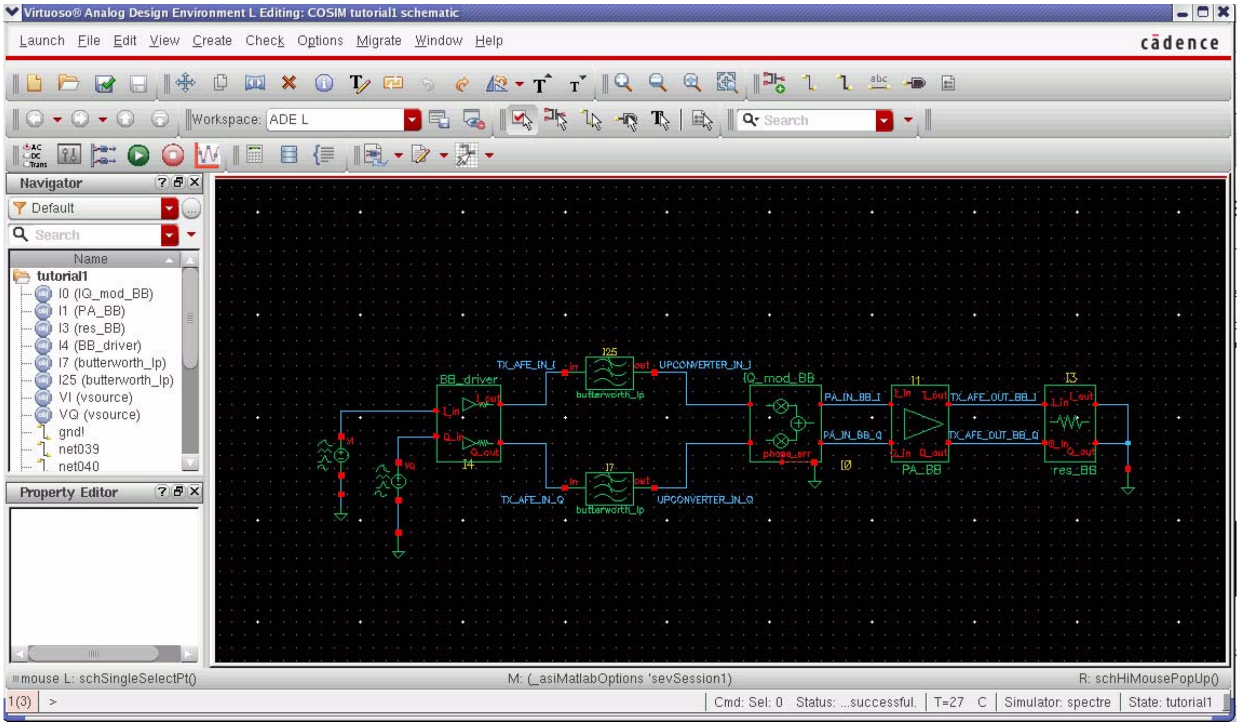
1. Start the virtuoso tool.

```
virtuoso &
```

The recommended version of `virtuoso` is release IC6.1.4 or later.

2. Open the schematic view of the cell.

   For example,



3. Open the Virtuoso Analog Design Environment from the schematic editor by choosing *Launch – ADE L*.

4. Choose *Setup – Matlab/Simulink – Setting*.

   The Cosimulation Options form appears.

5. In the Cosimulation Options form, click the *Select* button located beside *Cosimulation inputs*. Switch to the schematic viewer, where you see the following information below the schematic.

```
Select source instance as cosimulation inputs. Press Esc when done.
```

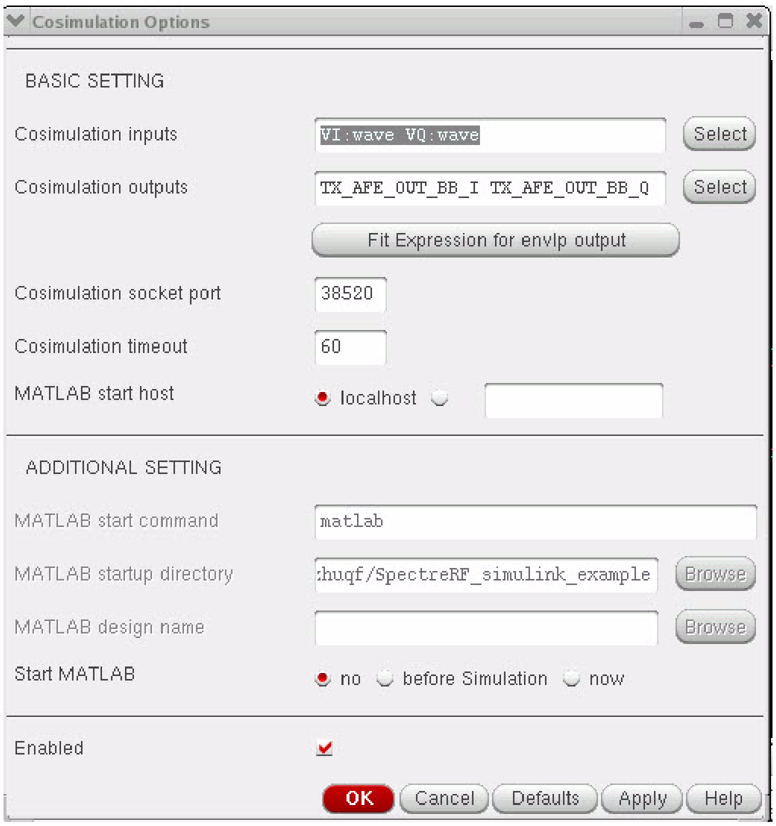
   Select the sources (which are connected to the outputs from the Simulink level of the design), then press the *Esc* key.

6. In the Cosimulation Options form, click the *Select* button beside *Cosimulation outputs*. Switch to the schematic viewer, where you see the following information below the schematic.

```
Select Net/Terminal as cosimulation outputs. Press Esc when done
```

Select the outputs (which are connected to the inputs at the Simulink level of the design), then press the *Esc* key.

The Cosimulation Options form looks something like this:



7. Ensure that the value of *Cosimulation socket port* is the same as the port value of the `SpectreRF Engine` block defined in "Connecting the Coupler Block Into the System-Level Simulink Schematic" on page 1885.

8. In the Cosimulation Options form, turn on *Enabled*.

9. Examine the possible values for the *Start MATLAB* field.

Value	Behavior
now	ADE launches a MATLAB session and opens a MATLAB desktop that is no different than it would be if you launched it manually. When you use the *now* value, you must start the simulation in the Simulink design before starting the SpectreRF simulation. Using the *now* value is a good way to test whether ADE can start MATLAB and open the design successfully.
before Simulation	ADE launches an internal MATLAB session and no MATLAB desktop appears. ADE opens the specified MATLAB design and initiates the MATLAB simulation before Spectre simulation starts. ADE starts MATLAB once but cannot start it again before you close the first session. Cadence suggests using the *now* value as a test before you run with the *before Simulation* value. The *now* and *before Simulation* values share one log file, which can be opened by choosing *Setup – Matlab/ Simulink – Log file*. You can use the log file to monitor the simulation when no MATLAB desktop is visible.
no	The *MATLAB start command*, *MATLAB startup directory*, and *MATLAB design name* fields become active. When you use the *no* value, you must start the simulation in the Simulink design before starting the SpectreRF simulation.

10. Continue the process of preparing the netlist according to how you want to start the applications that run the cosimulation.

To use this starting method...	Follow the guidance in this section...
Start the two applications (SpectreRF and MATLAB) separately.	“Preparing to Start the Two Applications Separately” on page 1893
Start ADE and arrange to have MATLAB start automatically.	“Preparing to Start ADE Manually and MATLAB Automatically” on page 1893

To use this starting method...	Follow the guidance in this section...
Start MATLAB and arrange to have SpectreRF start automatically.	“Preparing to Start MATLAB Manually and SpectreRF Automatically” on page 1895

Preparing to Start the Two Applications Separately

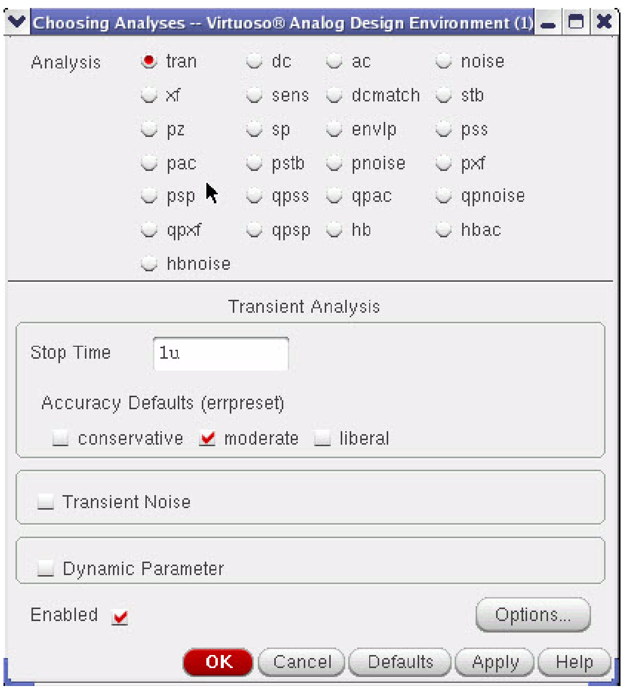
1. Set the *start MATLAB* value to *no*.

2. Click *OK* to close the Cosimulation Options form.

3. Use the Choosing Analyses form to set up the analysis.

The *Stop Time* can be any value. The SpectreRF simulator synchronizes the stop time with Simulink.

The Choosing Analyses window looks something like this:



4. Create a netlist and make sure the Simulink `cosim` statement appears in it.

Preparing to Start ADE Manually and MATLAB Automatically

1. Set the value of the *Start MATLAB* field to *before Simulation* in the Cosimulation Options form.

2. Type the name of the MATLAB design into the *MATLAB design name* field.

   Such designs have an extension of `.mdl`. For example, `env_d.mdl`. The MATLAB design must be in a location that is included in the `MATLABPATH` environment variable.

3. Set the value of the *Start MATLAB* field to *now*.

   A new MATLAB application starts and helps you open the Simulink design.

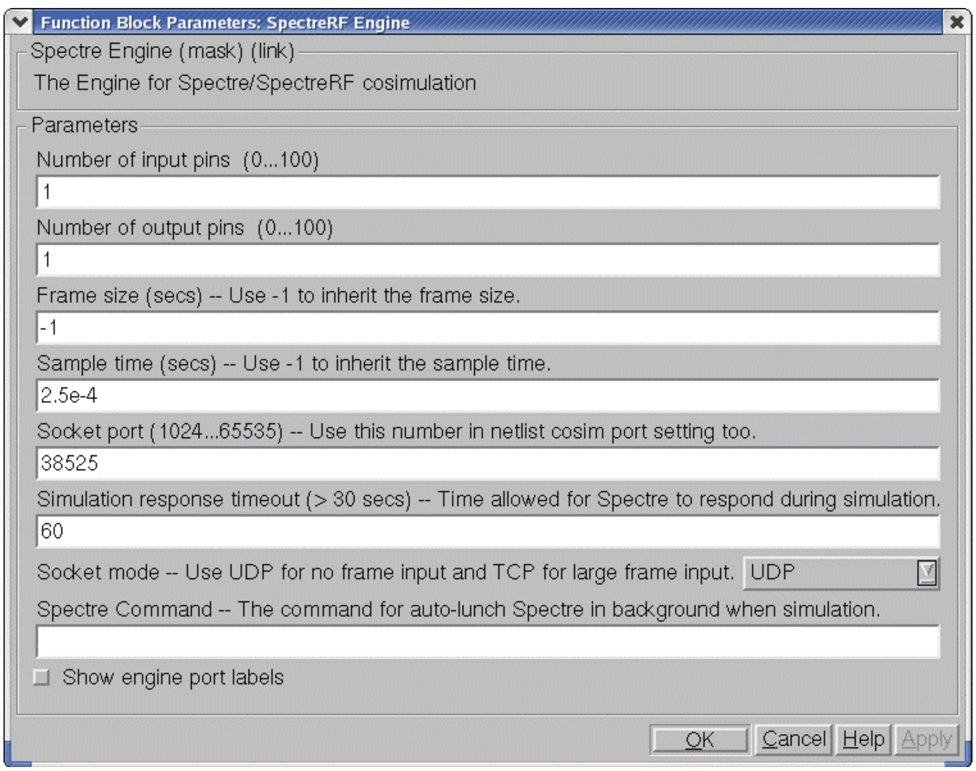
4. Double-click the `SpectreRF Engine` in the design and set the parameters.

   a. Set the *Sample time*. The Sample time can be set to -1, if you are uncertain about the appropriate time to use.

   b. Set *Socket Port* to the value you set in the Cosimulation Options form.

   c. Leave the *Spectre Command* field empty.

      This field is used when you wish to start SpectreRF automatically after starting MATLAB manually.



   d. Click *OK* to close the SpectreRF Engine form.

5. In the Cosimulation Options form, set *Start MATLAB* to *before Simulation*.

6. Click *OK* to close the Cosimulation Options form.

7. Use the Choosing Analyses form to set up the analysis.

8. Choose *File - Save* to save the design in MATLAB. Then exit from the MATLAB that was opened by ADE L.

Preparing to Start MATLAB Manually and SpectreRF Automatically

1. Set the value of the *Start MATLAB* field to *now*.

   A new MATLAB application starts and helps you open the Simulink design.

2. Double-click the `SpectreRF Engine` in the design and set the parameters.

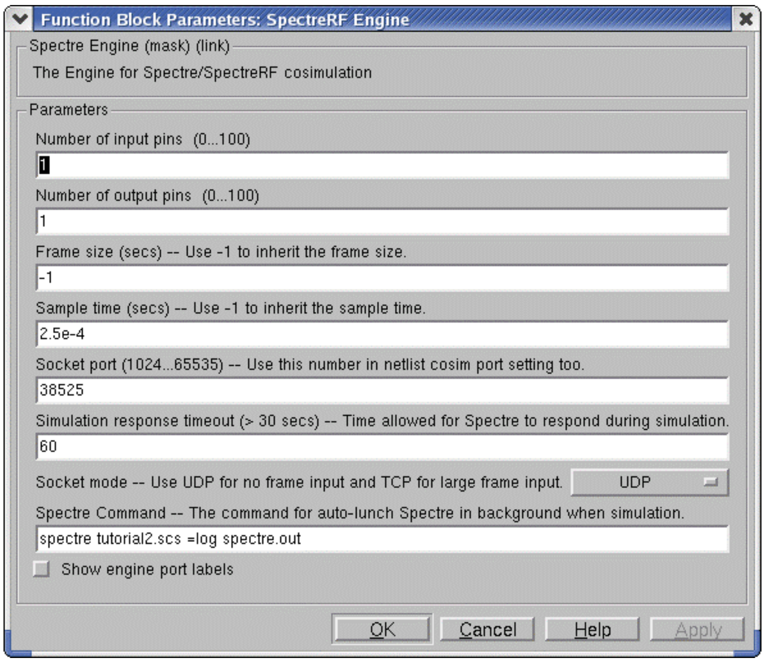
   a. Set the *Sample time*. The Sample time can be set to -1, if you are uncertain about the appropriate time to use.

   b. Set *Socket Port* to the value you set in the Cosimulation Options form.

   c. Type a command similar to the following into the *Spectre Command* field.

   `spectre netlist_file =log spectre.out`

   The Function Block Parameters: SpectreRF Engine window looks something like this:



   d. Click *OK* to close the Function Block Parameters: SpectreRF Engine form.

3. In the Cosimulation Options form, set *Start MATLAB* to *no*.

4. Click *OK* to close the Cosimulation Options form.
5. Use the Choosing Analyses form to set up the analysis.
6. Choose *File - Save* to save the design in MATLAB. Then exit from the MATLAB that was opened by ADE L.

Preparing the Netlist Without Using a Graphical User Interface

The steps in this section are necessary only if SpectreRF needs to start MATLAB automatically. If you are using the “Start MATLAB and arrange to have SpectreRF start automatically” starting method, ensure that there is no `cosim` statement in the netlist.

1. Edit the netlist file.

   For example,

   ```
   vi tutorial1/tutorial1.scs.
   ```

2. Add a `cosim` statement to the netlist.

   For example,

   ```
   matlab cosim server="bj2lnx20"  port=38525 inputs=["VI:wave" "VQ:wave"]
   outputs=[TX_AFE_OUT_BB_I TX_AFE_OUT_BB_Q]
   ```

   The parameters for the `cosim` statement are described below:

Table 9-1 Parameters of the cosim Statement

Parameter	Meaning
`design`	Refers to the file Simulink associates with the netlist. `design="env_d.mdl"` means the testbench whose file is `env_d.mdl`.

Table 9-1 Parameters of the cosim Statement, *continued*

`inputs`	Input vector to identify the flow from MATLAB to `SpectreRF Engine`. The format is `[“instance_name:wave” “instance_name1:wave” …]` where `wave`, is a key word. **Note:** The sequence of instances in square brackets follows the same one as the `SpectreRF Engine` port labels. This label information can be found by double-clicking the `SpectreRF Engine` and then by checking the *Show engine port labels* at the bottom in Function Block Parameters of the `SpectreRF Engine`. After the inputs are set in the netlist, the sources associated with the inputs are meaningless and their parameters do not affect the simulation.
`outputs`	Output vector from `SpectreRF Engine` to MATLAB. This vector can be the net name or terminal name of interest. If net name is used, only voltage is given; if terminal name is used, only current is output. To get current in an ENVLP analysis, you must add probes in the topology.
`port`	The port on the server running MATLAB. Its value should be the same as the value in the `SpectreRF Engine` in your current MATLAB design.

`server`	The name of the machine that MATLAB starts. It can be set with the machine name or with the IP address of that machine. The accepted form is
	`server = ”155.110.110.110”`
	and
	`server = “bj2lnx20”`
	`server=localhost` means that Spectre and MATLAB use the same machine.

`silent`	Tells MATLAB whether to open the window during simulation. If `silent=yes`, the testbench window is opened.
`timeout`	The period of time MATLAB spends waiting for a response from the Spectre simulator. `timeout=60` stops MATLAB if it does not receive any response.

3. Close and save the netlist.

   After the above steps, the netlist is ready for simulation.

Running the Cosimulation

With the coupler connected into the MATLAB design and with an appropriate netlist for the low-level design, you are ready to run the cosimulation. Note that these starting methods work only when the design is prepared as described earlier in this chapter.

To use this starting method...	Follow the guidance in this section...
Start the two applications (SpectreRF and MATLAB) separately.	"Starting the Two Applications Separately" on page 1898
Start SpectreRF and arrange to have MATLAB start automatically.	"Starting SpectreRF Manually and MATLAB Automatically" on page 1899
Start MATLAB and arrange to have SpectreRF start automatically.	"Starting MATLAB Manually and SpectreRF Automatically" on page 1899

Starting the Two Applications Separately

1. Open the high-level design or testbench in the MATLAB design window.
2. In the MATLAB design window, choose *Simulation – Start*.

   The MATLAB desktop issues the following message.

   ```
   block 'modified/SpectreRF Engine': (COSIM_OK) Waiting for incoming connection
   on port 38520, timeout: 60 sec ...
   ```

   Then quickly do step 3.

   Note: The time interval between step 2 and step 3 must be within the *Simulation response timeout* defined as Function Block Parameters of the `SpectreRF Engine` (by double-clicking the `SpectreRF Engine`).

3. In the ADE window, click *Netlist and Run* or enter a spectre command at the command line. For example,

   ```
   spectre tutorial1/tutorial1.scs
   ```

   The cosimulation begins. The MATLAB desktop issues a message similar to the following when simulation ends.

   ```
   block 'modified/SpectreRF Engine': (COSIM_OK) Simulation finished
   ```

Starting SpectreRF Manually and MATLAB Automatically

To run the cosimulation by starting SpectreRF,

- If you are using ADE, open the low-level design in the ADE window.

  a. Click *Netlist and Run*.

  The cosimulation begins. The MATLAB desktop issues a message similar to the following when simulation ends.

  ```
  block 'modified/SpectreRF Engine': (COSIM_OK) Simulation finished
  ```

- If you are using standalone SpectreRF,

  a. At the command line, enter a command similar to the following.

  ```
  spectre tutorial1/tutorial1.scs
  ```

  The cosimulation begins.

Starting MATLAB Manually and SpectreRF Automatically

a. Start MATLAB.

```
matlab&
```

b. Choose *Simulation – Start*.

The MATLAB desktop issues a message similar to the following.

```
block 'env_d/SpectreRF Engine': (COSIM_OK) Waiting for incoming connection
on port 38525, timeout: 60 sec ...
```

```
block 'env_d/SpectreRF Engine': (COSIM_OK) Launch Spectre with commad
'spectre tutorial2.scs =log spectre.out'”
```

The cosimulation runs.

c. After the cosimulation finishes, review the MATLAB/Simulink output, close the MATLAB desktop, and exit from MATLAB.

MATLAB Support Matrix

Table 9-2 MATLAB Toolbox

MMSIM	MATLAB Release	Supported Platform	
		Linux 32 / 64	***Solaris 64***
	R2007a	Supported	Supported
	R2007b	Supported	Supported
	R2008a	Supported	Supported
MMSIM12.1	R2008b	Supported	Supported
	R2009a	Supported	Supported
	R2009b	Supported	Supported
	R2010a	Supported	Not Supported [1]
	R2010b	Supported	Not Supported [1]
	R2011a	Supported	Not Supported [1]
	R2011b	Supported	Not Supported [1]
	R2012a	Supported	Not Supported [1]

1. From MATLAB versions 2010a onwards, Solaris Platform is no longer supported.

Table 9-3 MATLAB Cosimulation

MMSIM	MATLAB Release	Supported Platform	
		Linux 32 / 64	***Solaris 64***
	R2007a	Supported	Supported
	R2007b	Supported	Supported
	R2008a	Supported	Supported
MMSIM12.1	R2008b	Supported	Supported
	R2009a	Supported	Supported
	R2009b	Supported	Supported
	R2010a	Supported	Not Supported [1]

R2010b	Supported	Not Supported [1]
R2011a	Supported	Not Supported [1]
R2011b	Supported	Not Supported [1]
R2012a	Supported	Not Supported [1]

1. From MATLAB versions 2010a onwards, Solaris Platform is no longer supported.

A

Design Workshops

Introduction

This chapter introduces you to the basic building blocks of running successful and effective high frequency simulations using the Cadence® software. The following sections and screenshots explain how to set up your software and environment to run the example circuits in this chapter.

Before you perform the various SpectreRF analyses, you need to set up the component files and start the Cadence® software, as explained in the sections below.

Using SpectreRF from the MMSIM Hierarchy

The Virtuoso Spectre Circuit Simulator (Spectre) and SpectreRF are present in the MMSIM release stream.You must download and install the MMSIM simulators in a separate installation hierarchy than the IC hierarchy you use for the Cadence software. Documentation for new features and most bug fixes are provided exclusively with the MMSIM release stream.

The SpectreRF examples in this chapter use the `CDSHOME` environment variable to point to the installation hierarchy.

`CDSHOME`	Modify this path as necessary to point to the IC installation directory.

`CDSHOME` may already be set in your environment. If it is already set, then there is no need to reset it. Please check with your Cadence Tool System Administrator for more information.

Accessing the Most Current SpectreRF Documentation

The documentation for the latest features of SpectreRF is always found in the MMSIM hierarchy.

Important

Note that the help buttons on the forms in ADE lead you to the IC version of the documentation, which should not be used. Instead, use the SpectreRF documentation located in the MMSIM hierarchy.

Creating a Local Editable Copy of the ExampleLibRF Library

Perform the following steps to create a copy of the *ExampleLibRF* library and save it to your home or working directory:

1. Navigate to the directory where you want the workshop to be located.
2. Use the UNIX `cp` command to copy `RF_Doc_Database.tar.gz` from the hierarchy to your desired directory.

   ```
   cp <path to>/RF_Doc_Database.tar.gz
   ```
3. Type `tar xfz RF_Doc_Database.tar.gz.`

Work with your system administrator to locate the MMSIM installation directory at your site. This will be in the MMSIM hierarchy at `<MMSIM>/tools/spectre/examples/SpectreRF_workshop/RF_Doc_Database.tar.gz`

Downloading and Using GPDK180

PDK is an abbreviation for Process Design Kit. A PDK is a complete set of technology files to enable analog and mixed signal custom IC circuit design within the Cadence Design System's Custom IC Design Environment. These PDKs are available for download online.

1. Navigate to *<path to>/RF_Doc_Database.*
2. Use the UNIX `ls` command to list the contents in the directory.

   You will see the following files and directories:

   `cds.lib*` `doc`/ `libs`/ `models`/ `readme` `setup.csh*` `share`/ `simulation`/, and `skill`/.

   This directory structure is organized as follows:

   `cds.lib`: The Cadence library file for the project

   `simulation`: Simulation directory

   `libs`: Directory containing the libraries for the project `skill`: Any SKILL code needed

`models`: Spectre models that are not gpdk specific

`share`: Directory where gpdk180 is located after being downloaded from the `pdk.cadence.com` site.

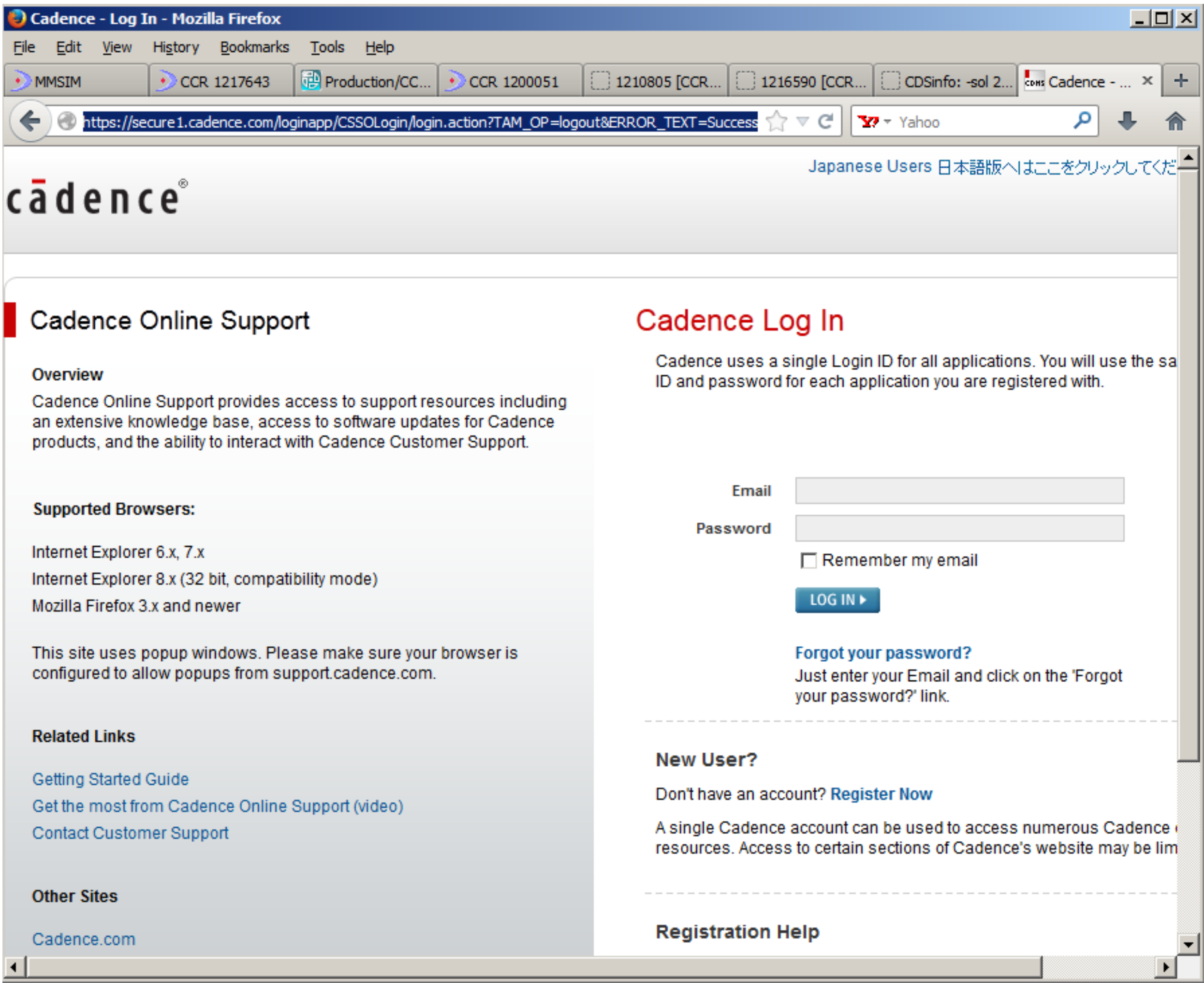
For example, the gpdk needs to be present in the `/share/gpdk180_v3.3` directory.

The models are present in the `/share/gpdk180_v3.3/models/spectre` directory.

To download gpdk180, follow these steps:

1. Open a browser and go to *http://support.cadence.com*. The Cadence Online Support website is displayed, as shown below.

Figure A-1 Cadence Online Support Website



2. Specify your email and password in the *Email* and *Password* fields and click *LOGIN.*

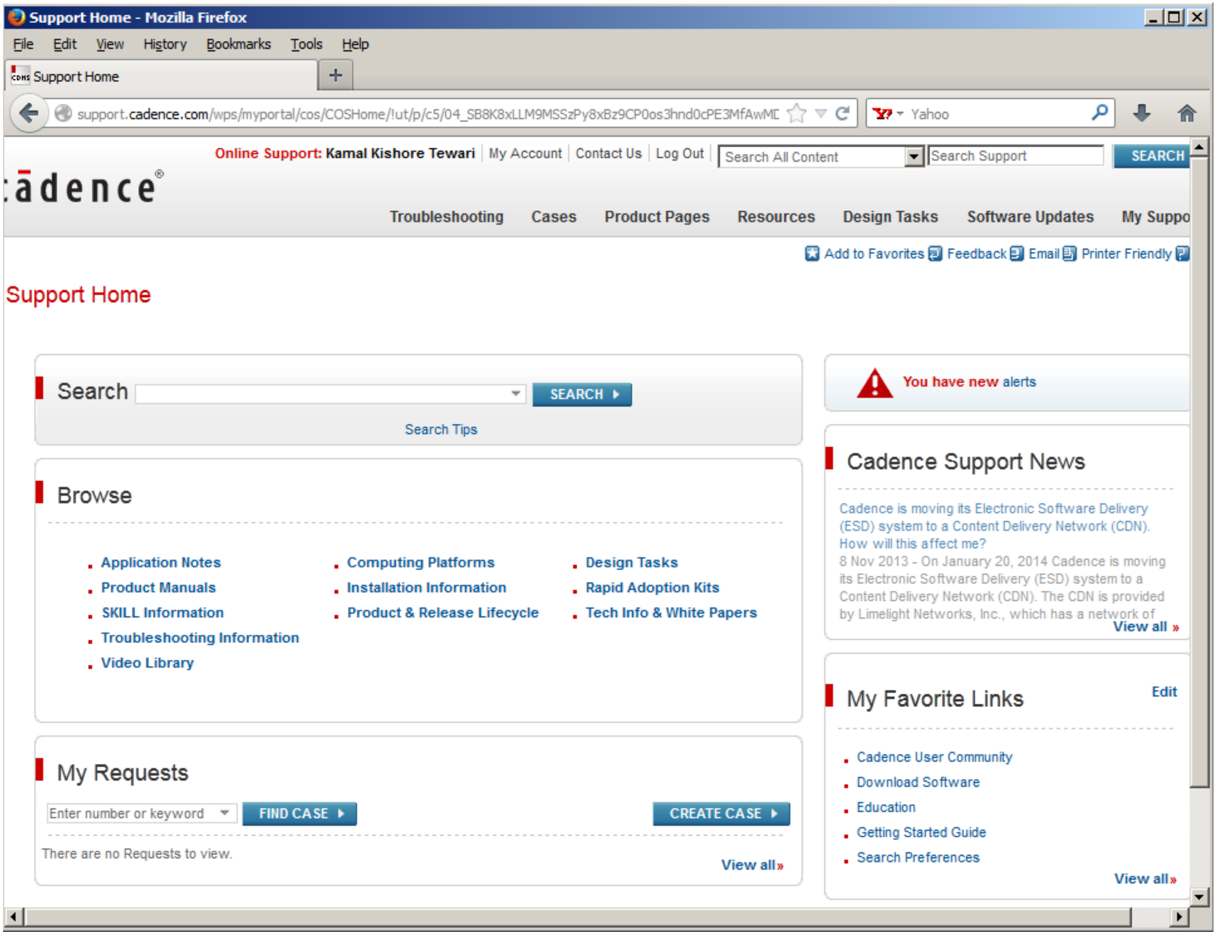
   Note: If you are new user, click *Register Now* and follow the steps to register yourself.

For the example designs in Appendix A, you need to use Cadence's GPDK180 which can be downloaded as follows.

1. Enter your email and password.

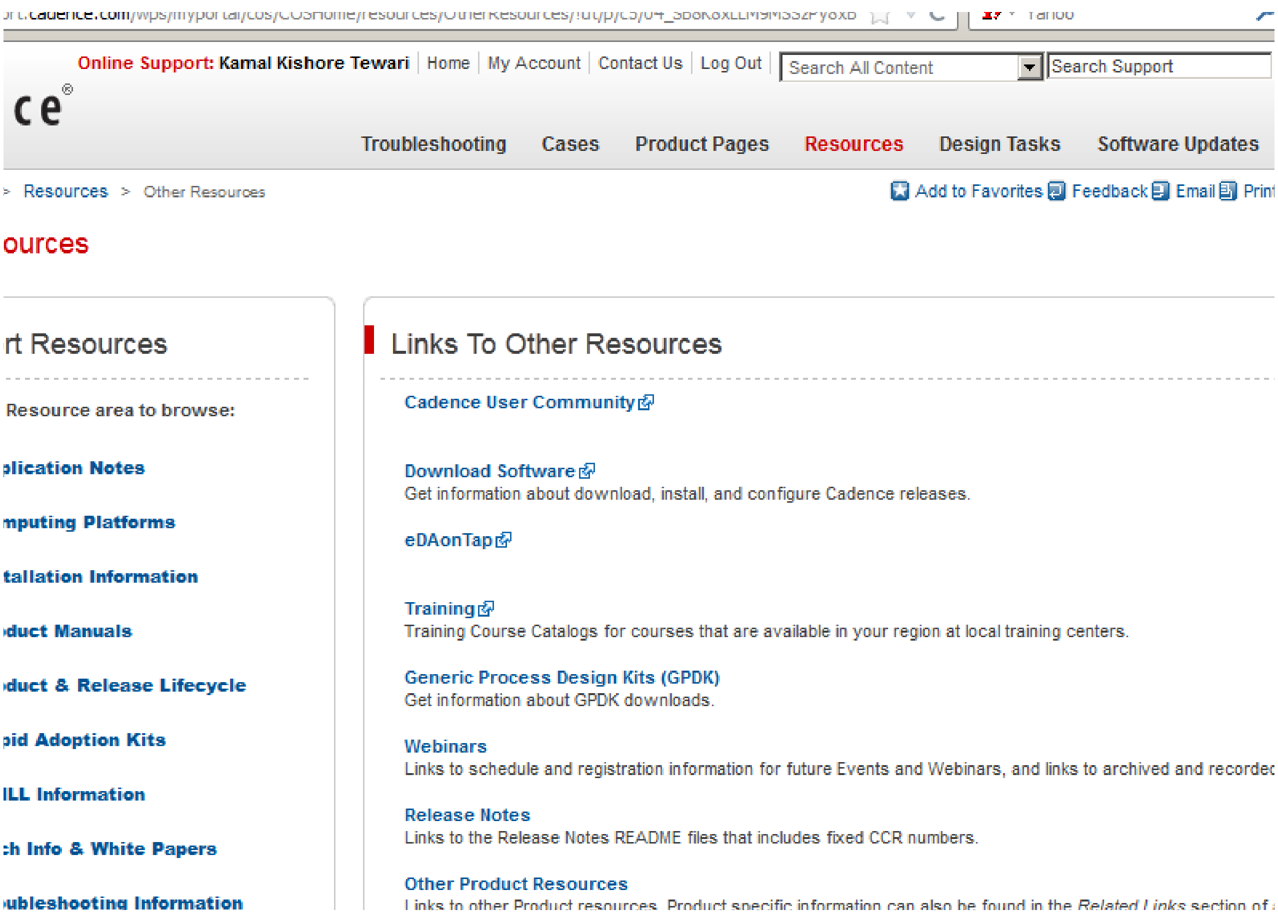
   The *Support Home* web page is displayed, as shown below.

Figure A-2 Cadence Support Home Page



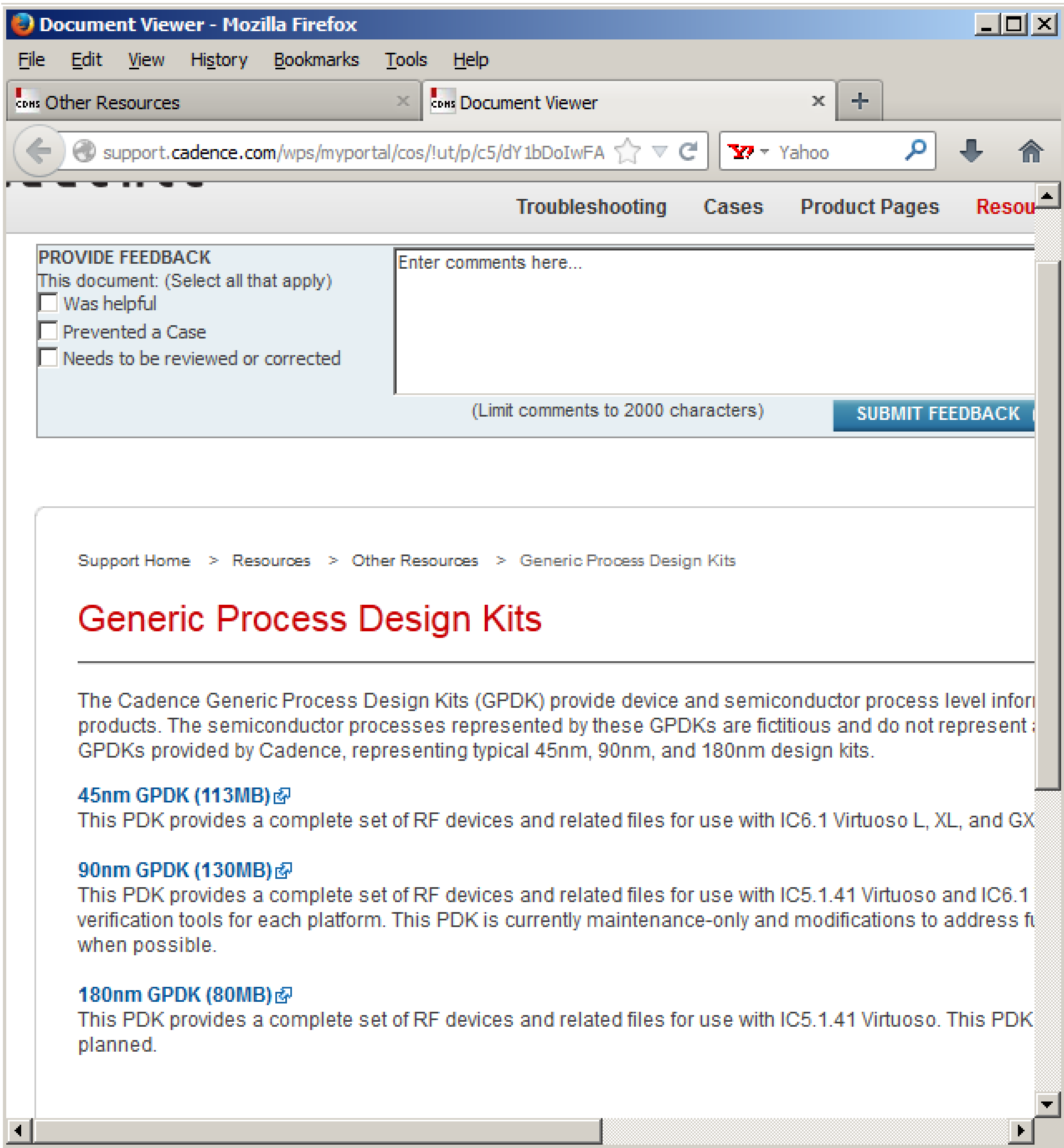
2. Select *Resources - Other Resources*. The *Other Resourecs* Web page is displayed, as shown below.

Figure A-3 Other Resources Page

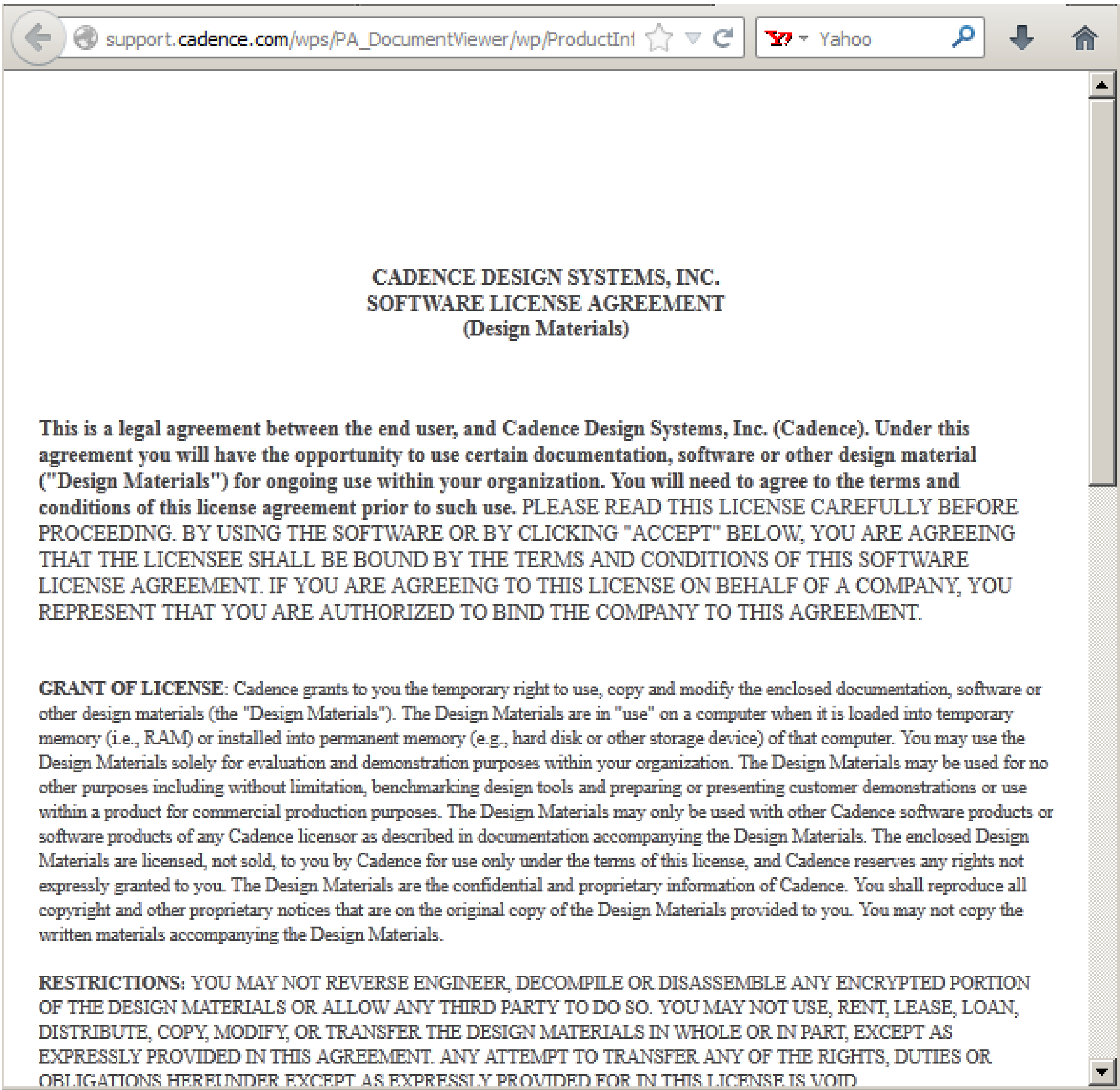


3. Click the *Generic Process Design Kits (GPDK)* link. The *Generic Process Design kits* Web page is displayed, as shown below.

Figure A-4 Generic Process Design Kits Web Page



4. Click the *180nm GPDK (80MB)* link. The Software License Agreement page is displayed, as shown below.



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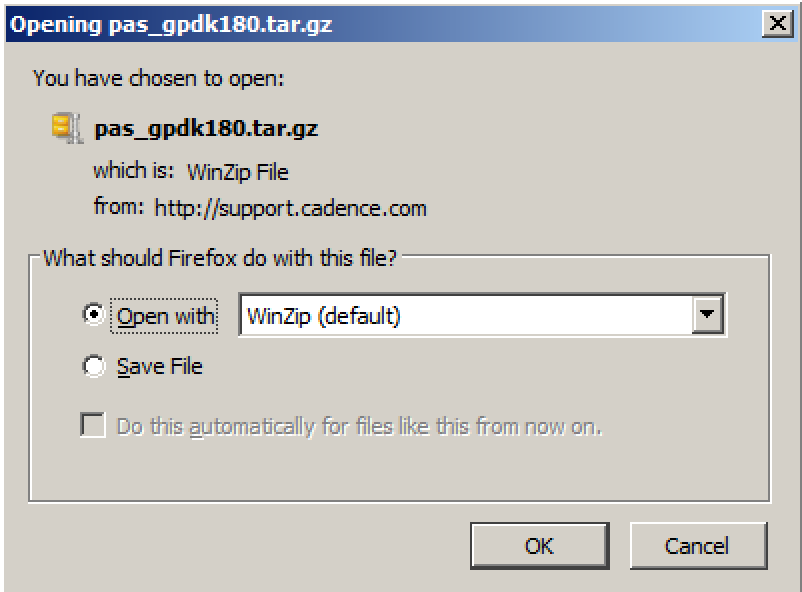
5. Scroll down the page and click *I ACCEPT*. The download dialog is displayed, as shown below.



6. Select *Save File* and click *OK*.

7. Select the location where you want to save the *pas_gpdk180.tar.gz* file. The file will start downloading. The time taken to download is about a minute, but that varies depending on the nature and speed of your internet connection.

8. Save the file `pas_gpdk180.tar.gz` to `<path to>/RF_Doc_Database/share`.

9. Navigate to the `RF_Doc_Database/share` directory and untar the `pas_gpdk180.tar.gz` file using the following command.

```
tar -xvzf pas_gpdk180.tar.gz
```

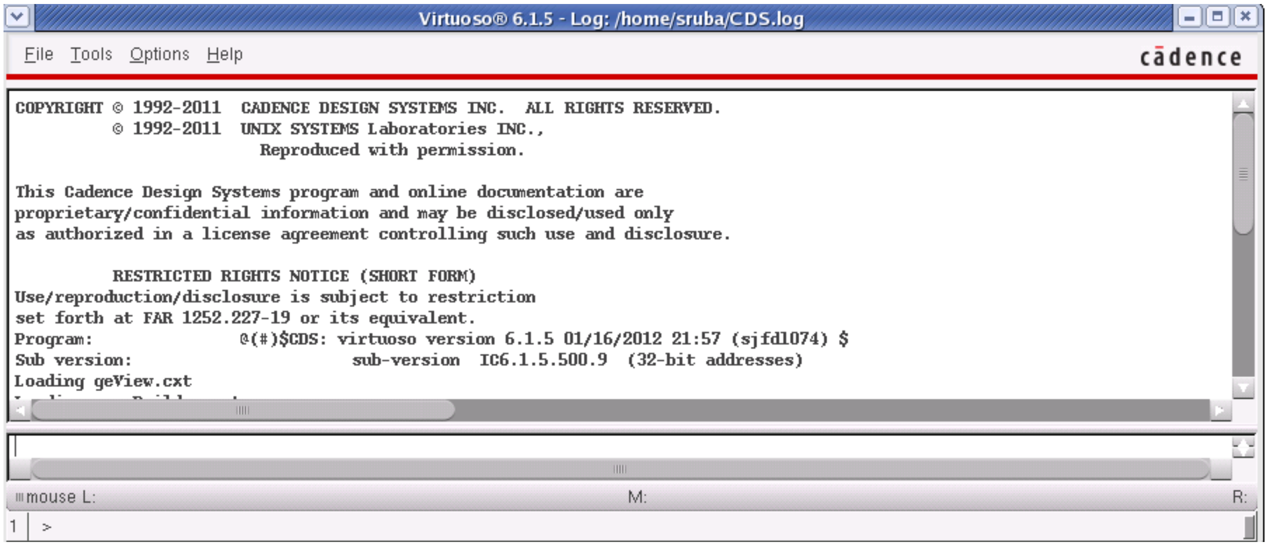
Starting Virtuoso

To access the Library Path Editor, perform the following steps:

1. In a UNIX window, type `virtuoso &` to start the Cadence software.

   The Command Interpreter Window (CIW) is displayed, as shown below.

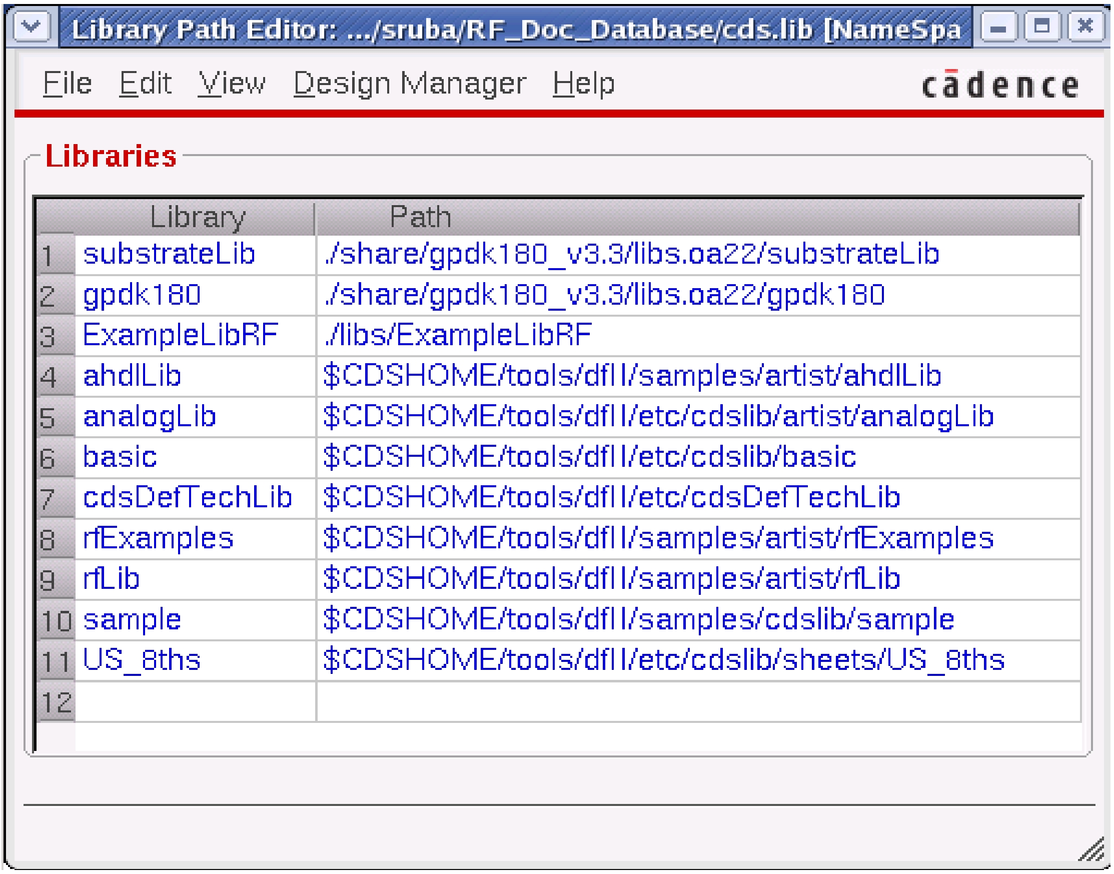
Figure A-5 Command Interpreter Window



2. In CIW, choose *Tools – Library Path Editor*.
3. The *Library Path Editor* is displayed.

   This next step is to check that all of the libraries necessary are accessible. You will not be changing anything.

Figure A-6 Library Path Editor Window



Make sure that all of the library names are displayed in blue or green. If any libraries are displayed in red text, there is an error and you need to work with your system administrator to fix the path.

4. Exit the *Library Path Editor*.

Example Circuits

In the next part of the documentation, the following basic circuit types will be explored:

- Simulating Low Noise Amplifiers on page 1914
- Simulating Oscillators on page 2000
- Simulating Mixers on page 2221

Simulating Low Noise Amplifiers

The SpectreRF simulator can simulate very linear circuits, such as Low Noise Amplifiers (LNA). This section uses the lnaSimple circuit to illustrate how the SpectreRF simulator can determine the characteristics of an LNA design.

The first stage of a receiver is typically an LNA, whose main function is to set the boundary as well as to provide enough gain to overcome the noise of subsequent stages (for example, in the mixer and IF Amplifier). Apart from providing enough gain in addition to introducing as little noise as possible, an LNA should accommodate large signals without distortions, offer a large dynamic range, and provide good matching to its input and output. Good matching is extremely important if a passive band-select and image-reject filter precedes and succeeds the LNA because the transfer characteristics of many filters are sensitive to the quality of the termination.

In the LNA example that follows, you will plot the following characteristics of the low noise amplifier.

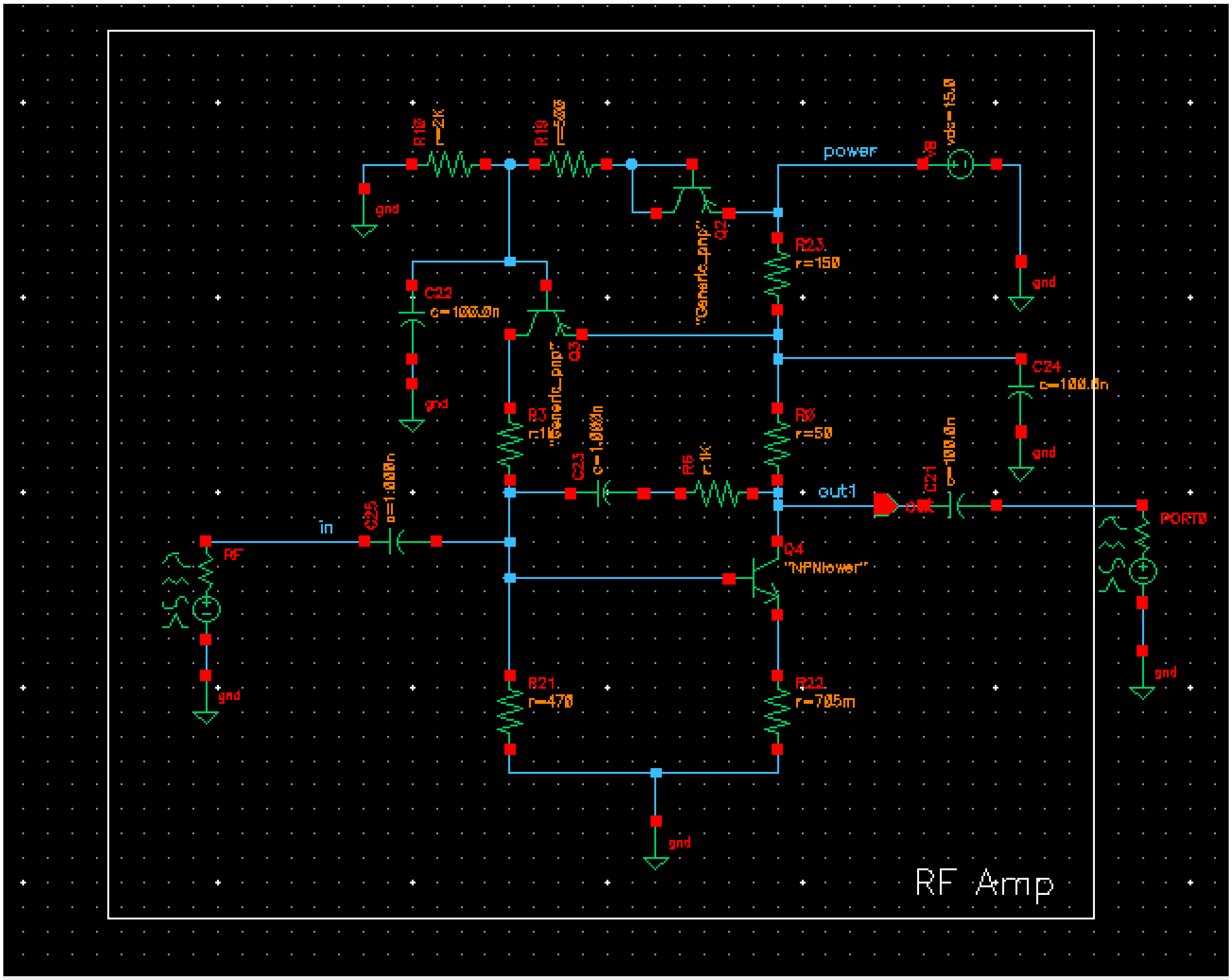
LNA Measurements (lnaSimple	**Analyses**
S-Parameter Analysis, Small Signal Gain	sp
Stability, Stability Circles	sp
Linear two-port Noise Frequency Measurements, Noise Circles	sp+noise
Third-Order Intercept Measurement	hb
Rapid IP3 Using Specialized AC Analysis	ac

To use this section, you must be familiar with the SpectreRF simulator analyses as well as know about LNA designs. For more information about the SpectreRF simulator analyses, refer to the various chapters in this user guide and also the *SpectreRF Simulation Option Theory.*

The lnaSimple Low Noise Amplifier Circuit

The lnaSimple circuit can be found in the *ExampleLibRF* library. Refer to the Introduction on page 1903 for the instructions on accessing the *ExampleLibRF* library. The schematic for the lnaSimple circuit is shown below. It is a differential low noise amplifier.

Figure A-7 Schematic for the lnaSimple Low Noise Amplifier



The following tables list some measured values for different aspects of the lnaSimple low noise amplifier.

Measurement	**Measured**
RF frequencies (Hz)	900M
Output frequency (Hz)	900M
RF voltage	200mV peak
RF power	-10 dBm
Gain	*measurement needed*
Stability Factor	*measurement needed*
Noise figure	*measurement needed*
1dB compression point	*measurement needed*
Input IP3 (from swept power)	*measurement needed*
Input IP3 (from AC analysis)	*measurement needed*

Design Variable	**Default Value**
prf (RF power)	-10 dBm
frf1 (RF frequency)	900M

Setting Up to Simulate the lnaSimple Low Noise Amplifier

Opening the lnaSimple Circuit in the Schematic Window

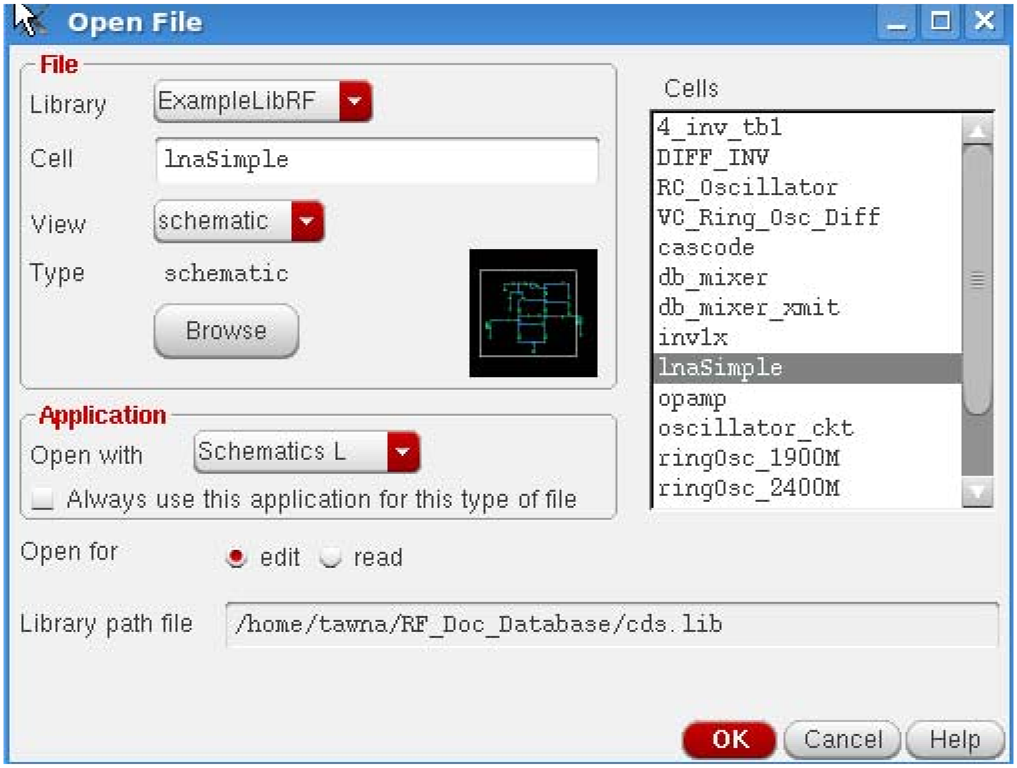
1. In CIW, choose *File – Open*.

   The *Open File* form is displayed.

2. Choose *ExampleLibRF from the Library* drop-down list.
3. Choose *lnaSimple* from the *Cells* list box.
4. Choose *schematic* from the *View* drop-down list.

The completed *Open File* form will look like the one below.

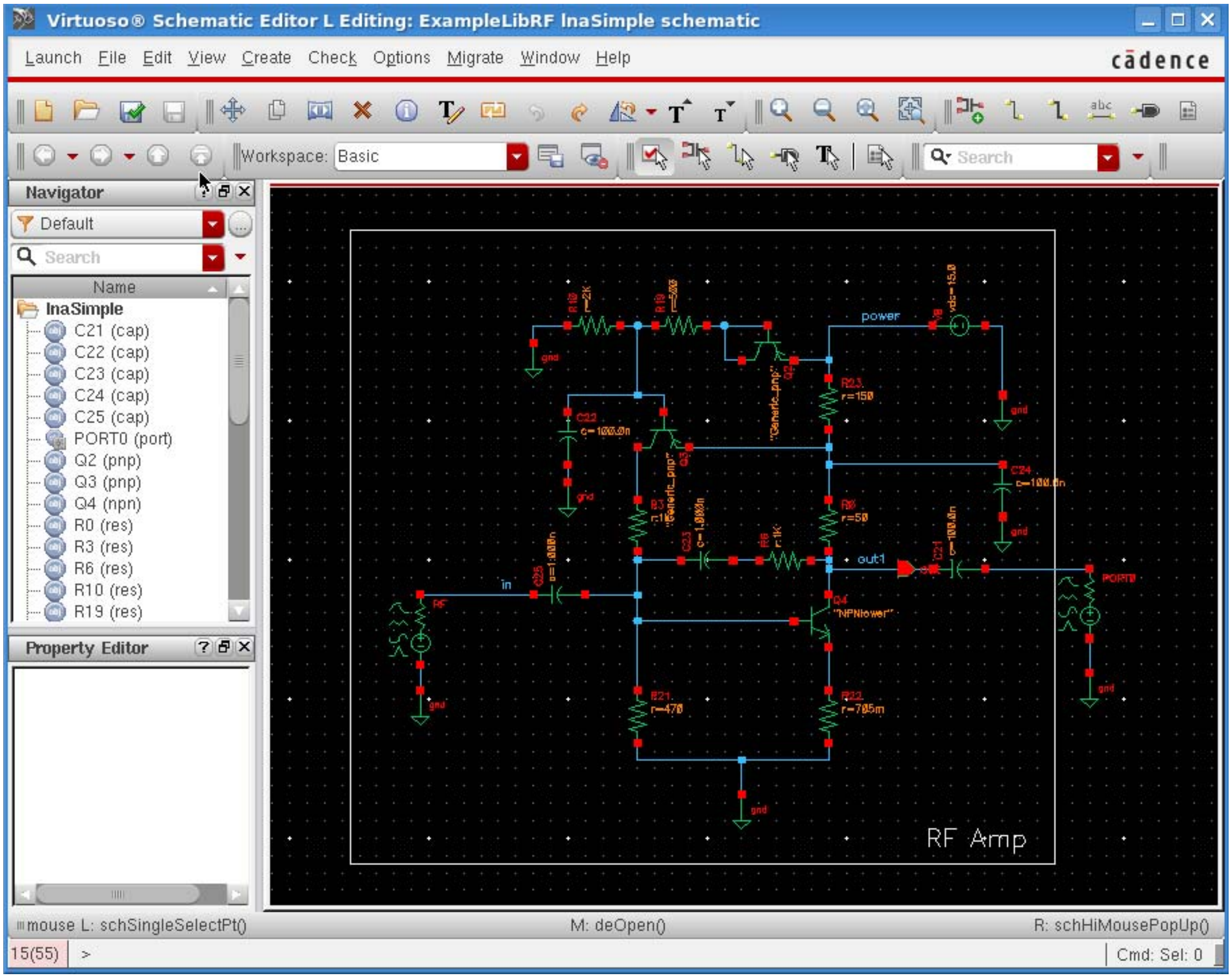
Figure A-8 Open File Form



5. Click *OK*.

The Schematic window for the *lnaSimple* circuit is displayed.

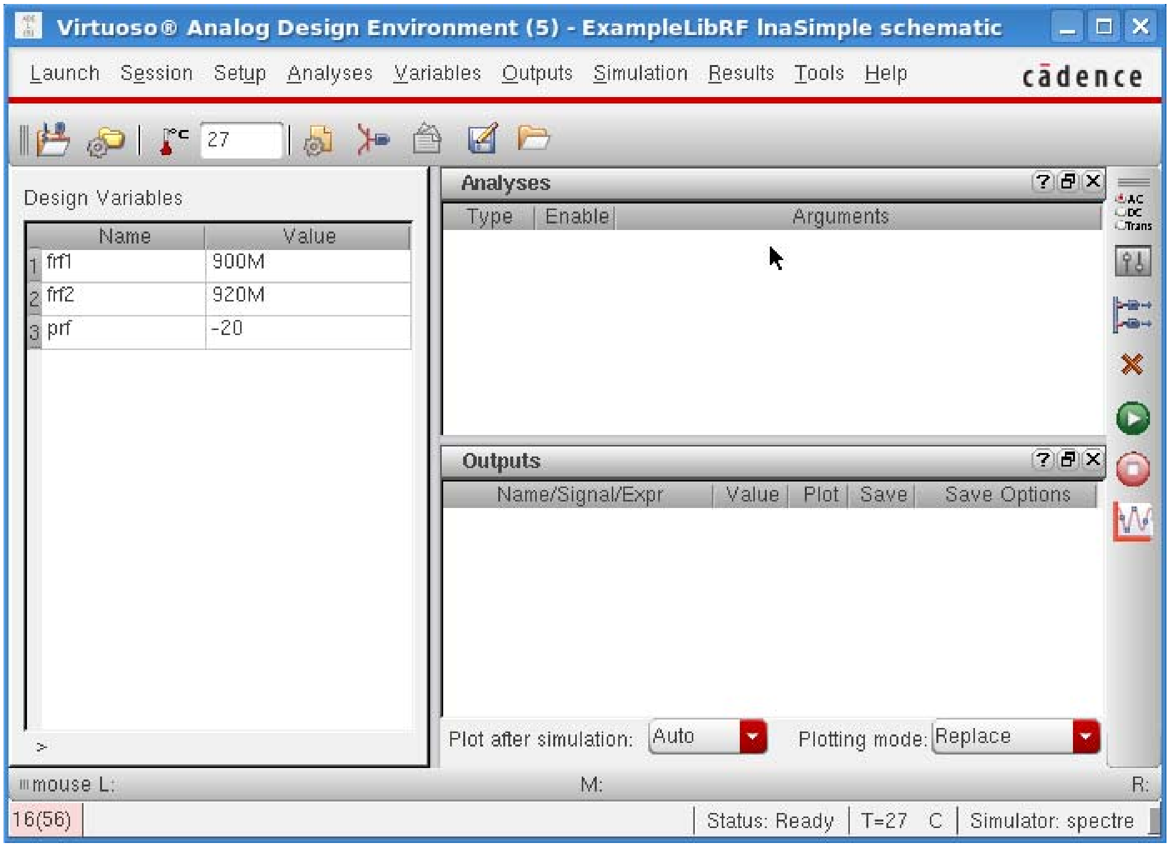
Figure A-9 lnaSimple Schematic



6. In the Schematic window, choose *Launch– ADE L*.

The *Virtuoso Analog Design Environment* window is displayed, as shown below.

Figure A-10 Analog Design Environment Window



Choosing Simulator Options

1. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

2. Specify the following:

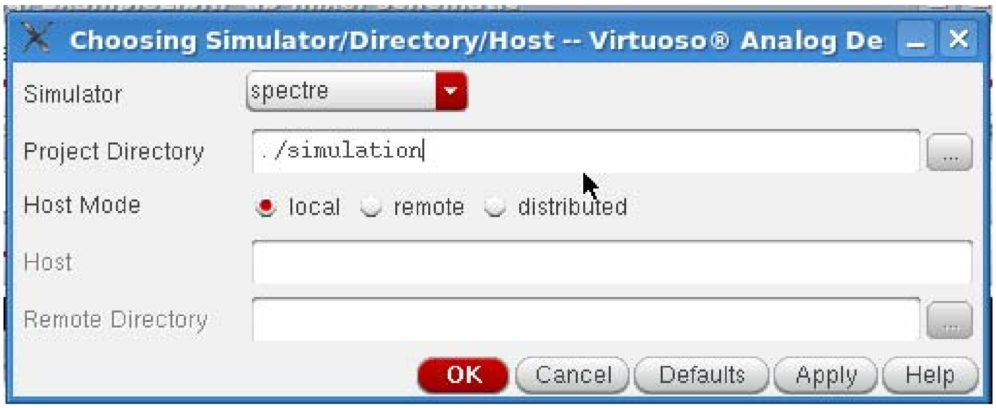
   a. Choose *spectre* from the *Simulator* drop-down list.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form will look similar to the one below.

Figure A-11 Choosing Simulator/Directory/Host Form

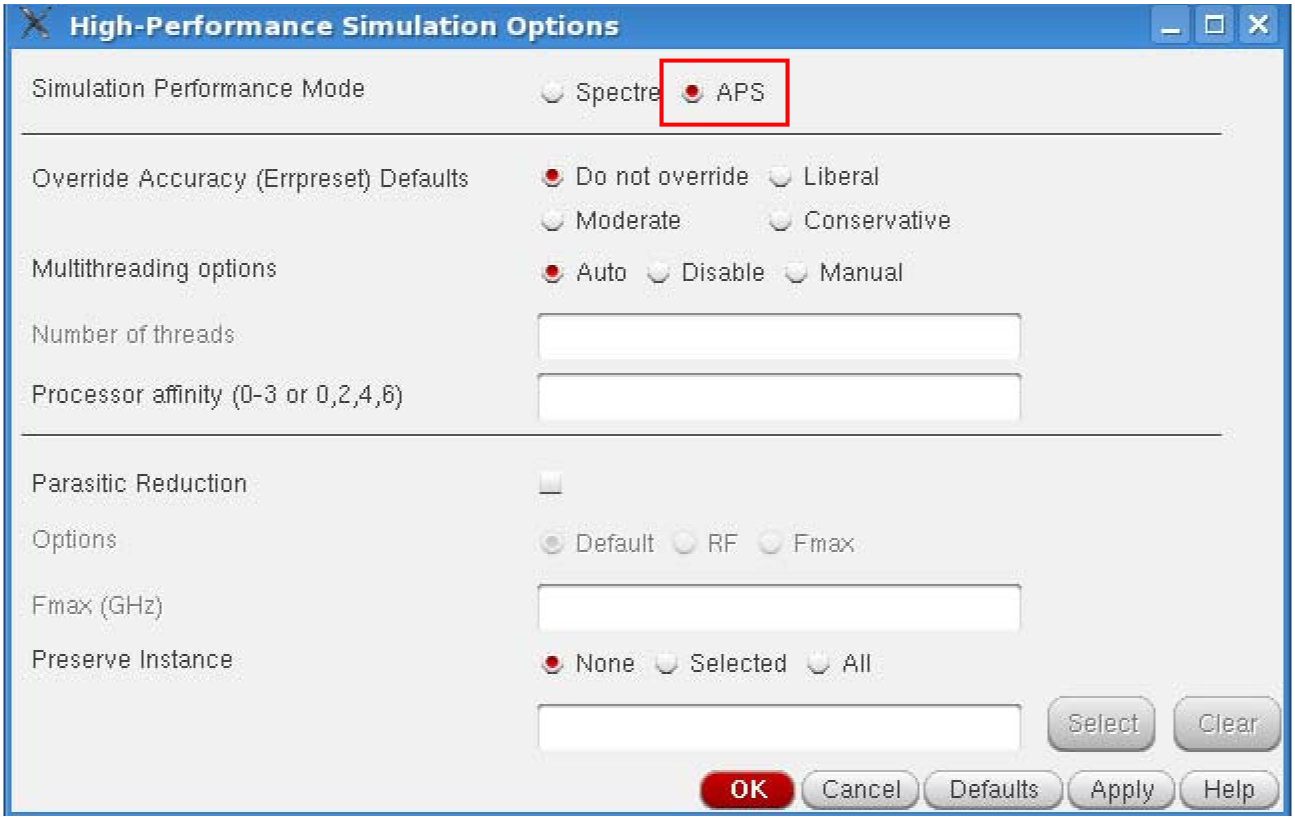


3. Click *OK*.

4. Now, set up the High-Performance Simulation Options, as follows:

   In the ADE window, choose *Setup-High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

Figure A-12 High Performance Simulation Options



In the *High Performance Simulation Options* window, select *APS* for *Simulation Performance Mode*. Note that *Auto* is selected for *Multithreading options.* The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores.

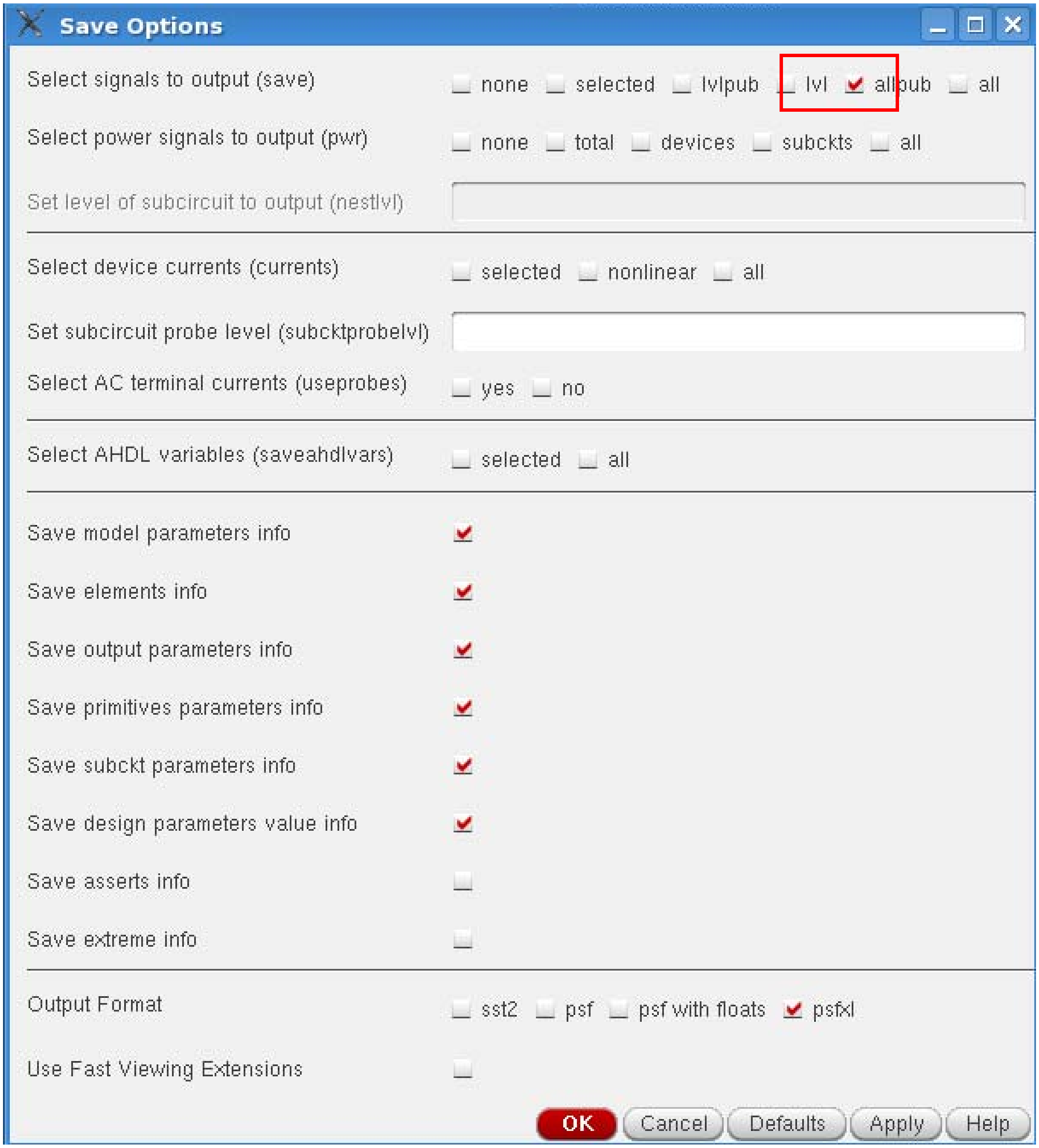
Note: The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading. For more information, refer to the Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator User Guide.

Click *OK*.

5. In the Virtuoso Analog Design Environment window, choose *Outputs – Save All*.

The *Save Options* form is displayed.

Figure A-13 Save Options Form



6. In the *Select signals to output(save)* section, ensure that *allpub* is selected.

This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

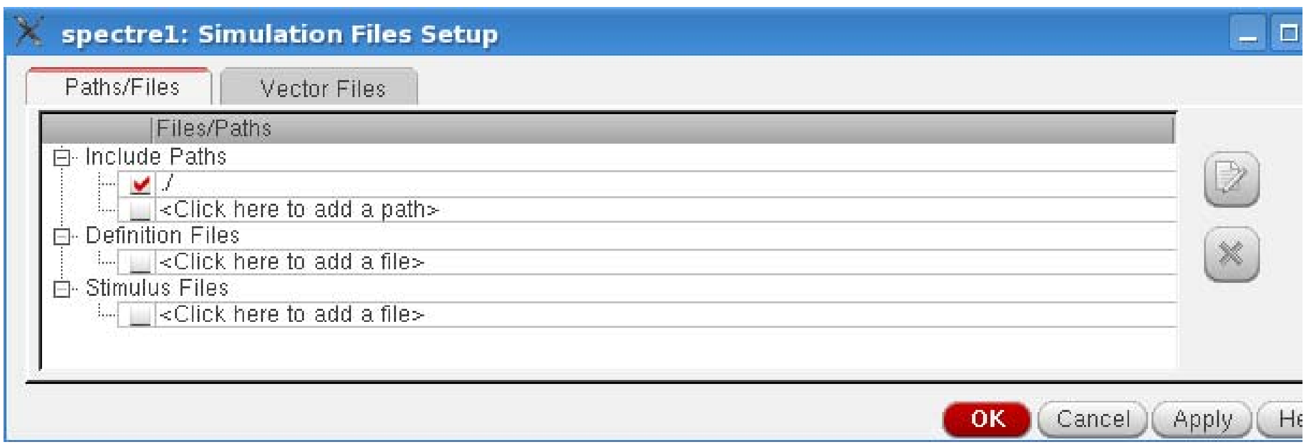
To save the currents, select *nonlinear* in the *Select device currents (currents)* section if you just want to save the device currents, or select *all* if you want to save all the currents in the circuit.

7. Click *OK* to close the *Save Options* form.

Setting Up Model Libraries

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files*. The *Simulation Files Setup* form is displayed.

Figure A-14 Simulation Files Setup Form



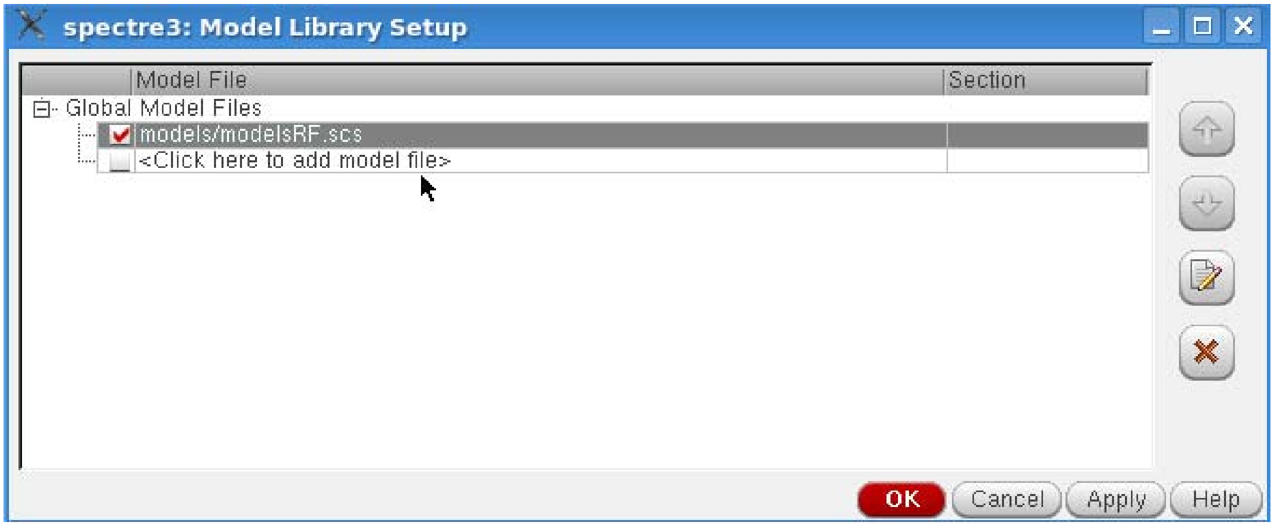
2. Verify that the *Include Path* is set, as shown above, and click *OK* to close the form.
3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

   The *Model Library Setup* form is displayed.
4. Type in the name of the model file, as follows:

   `models/modelsRF.scs`
5. Click *Add*.

   The *Model Library Setup* form will look like the following:

Figure A-15 Model Library Setup



6. Click *OK*.

SP Analysis and Small Signal Gain

The S-Parameter (sp) analysis is the most useful linear small signal analysis for low noise amplifiers. In this section, you will set up an *sp* analysis by specifying the input and output ports and the range of sweep frequencies.

The S-parameter analysis linearizes the circuit about the DC operating point and computes the S-parameters of the circuit taken as an N-port. In the netlist, the port statements define the ports of the circuit. Each active port is turned on sequentially, and a linear small-signal analysis is performed. Spectre converts the response of the circuit at each active port into S-parameters and outputs these parameters. There must be at least one active port statement in the circuit.

Three power gain definitions appear in the literature and are commonly used in LNA design.

The following gain quantities are valid only for two-port circuits:

- *GA* (available gain) is the power gain obtained by optimally matching the output of the network.
- *GP* (power gain) is the power gain obtained by optimally matching the input of the network.
- *GT* (transducer gain) shows the insertion effect of a two-port circuit. This quantity is used in amplifier design.

Besides these three gain definitions, there are three additional gain definitions you can use to evaluate the LNA design.

- *Gumx* (maximum unilateral transducer power gain)
- *Gmax* (maximum available gain) shows the transducer power gain when a simultaneous conjugate match exists at both ports.
- *Gmsg* (maximum stable gain) shows the gain that can be achieved by resistively loading the two-port such that `k = 1` and then simultaneously conjugately matching the input and output ports. For conditionally stable two-ports, you can approach the maximum stable gain as you reduce the input and output mismatch. If you attempt a simultaneous conjugate match and `k < 1`, the two-port oscillates.

There are also two gain circles that are helpful to the design of input and output matching networks.

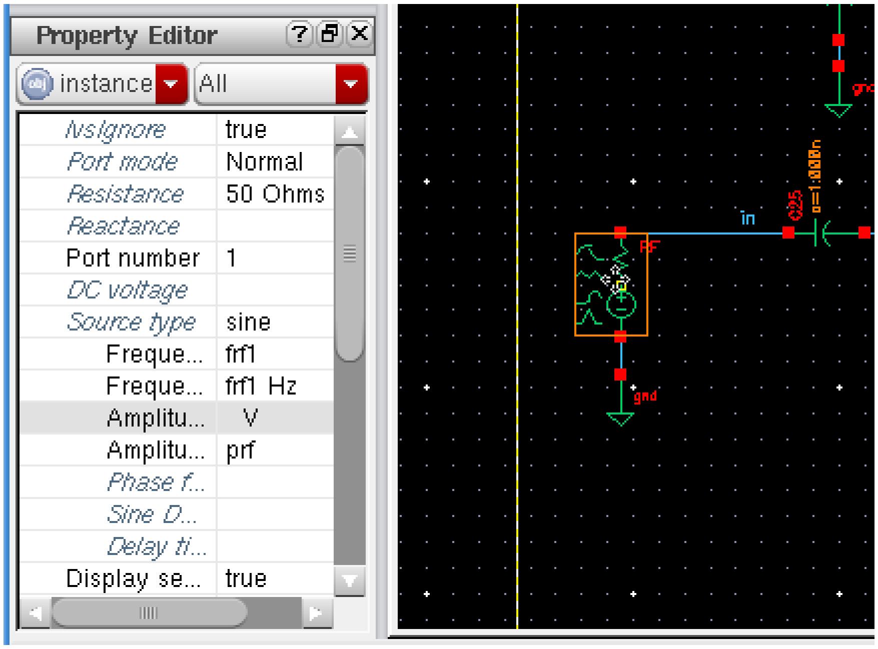
- *GPC*: power gain circle
- *GAC*: available gain circle

The next steps will walk you through these simulations and measurements.

Editing the Schematic

1. In the Schematic window, click the RF voltage source. This is the input port to lnaSimple.

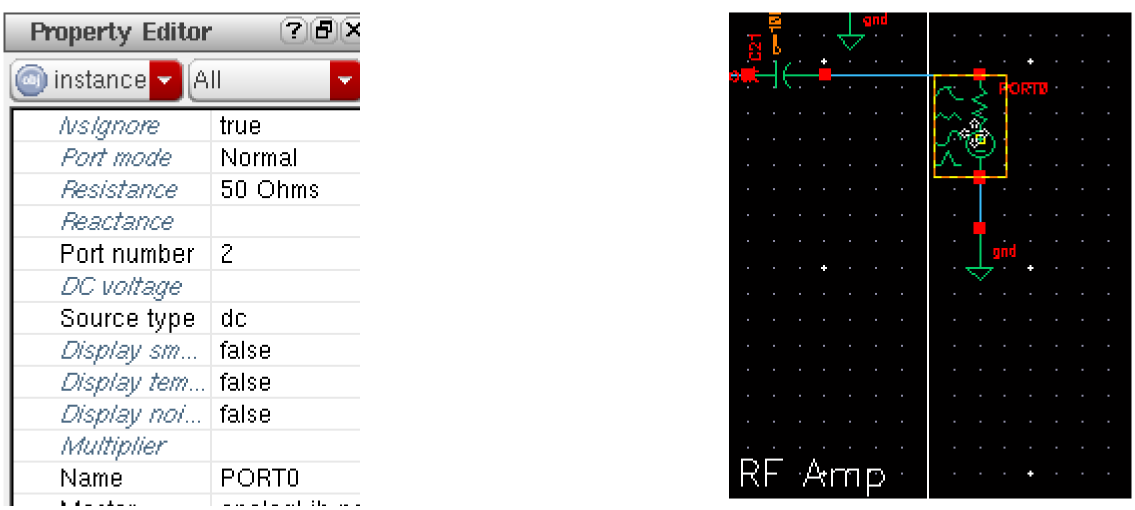
Figure A-16 RF Voltage Source in lnaSimple



Note that when you select the port, the *Property Editor* populates with the instance properties on the port. Note that the RF port has a *Port Number* of 1 and input *Resistance* of 50 Ohms. Leave the *Source type* parameter set to `sine` and disable the frequency source in the *Design Variables* section of the ADE L window.

2. Next, click the port on the right side of the schematic. This is the output port of lnaSimple. The *Property Editor* shows the instance properties on the output port.

Figure A-17 lnaSimple Output Port

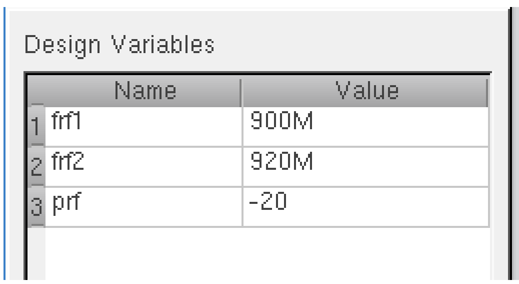


Note that the output *Port Number* is 2 and the reference *Resistance* (real part of the reference impedance) is 50 Ohms. The *Source type* is set to *dc*, as there is no large signal generated on this port.

Setting Design Variables

Perform the following steps to set the design variables to the values required for each simulation. The *Design Variables* section is located in the ADE L simulation window:

Figure A-18 Design Variables Section of ADE L Window

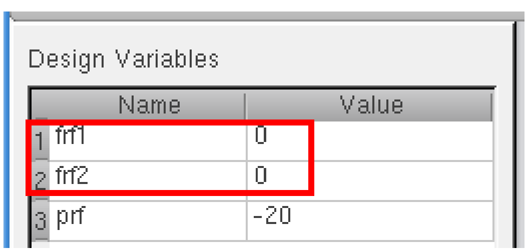


Design Variables

	Name	Value
1	frf1	900M
2	frf2	920M
3	prf	-20

1. Change the design variables *frf1* and *frf2* to `0`. To edit the value, simply click the value to the right of the variable name, and type in a value. Then press *Enter*. Setting the input frequency to 0 disables the production of waveforms for the large-signal analyses like tran, pss, and hb (harmonic balance.)

Figure A-19 Edited Design Variables Section of ADE-L Window



Design Variables

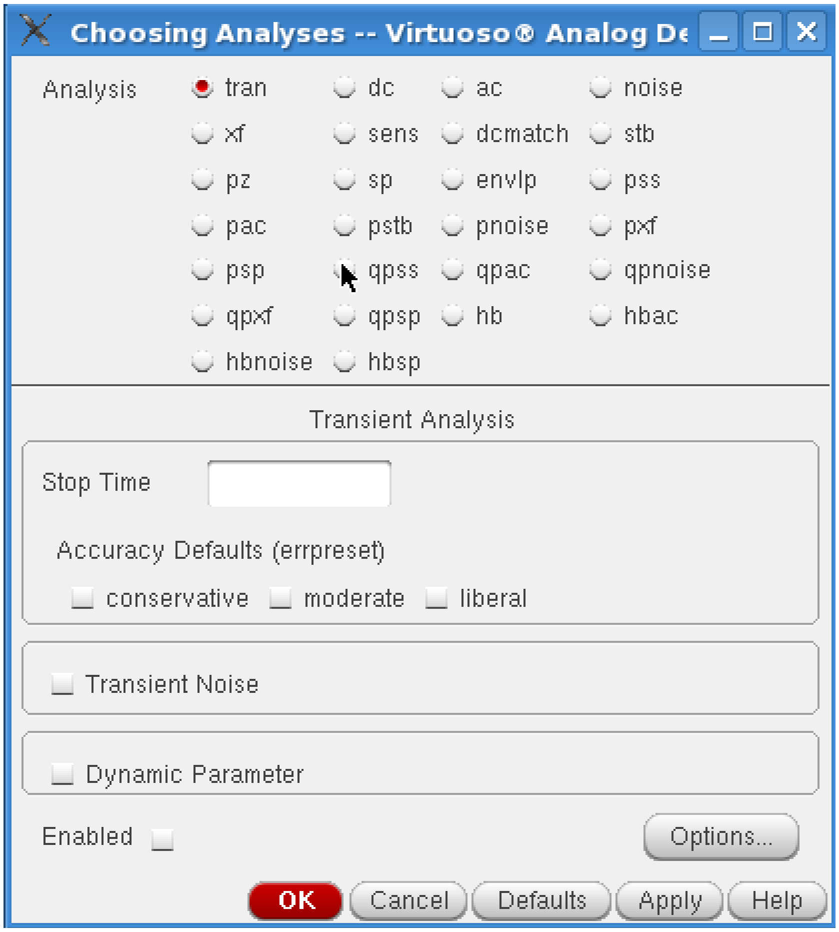
	Name	Value
1	frf1	0
2	frf2	0
3	prf	-20

The small-signal analysis begins by linearizing the circuit about an operating point. By default, this analysis computes the operating point, if it is not known, or recomputes it if any significant component or circuit parameter has changed.

2. In the ADE L window, choose *Analysis - Choose* or click the *Choosing Analyses* icon ()on the right side of the ADE L window.

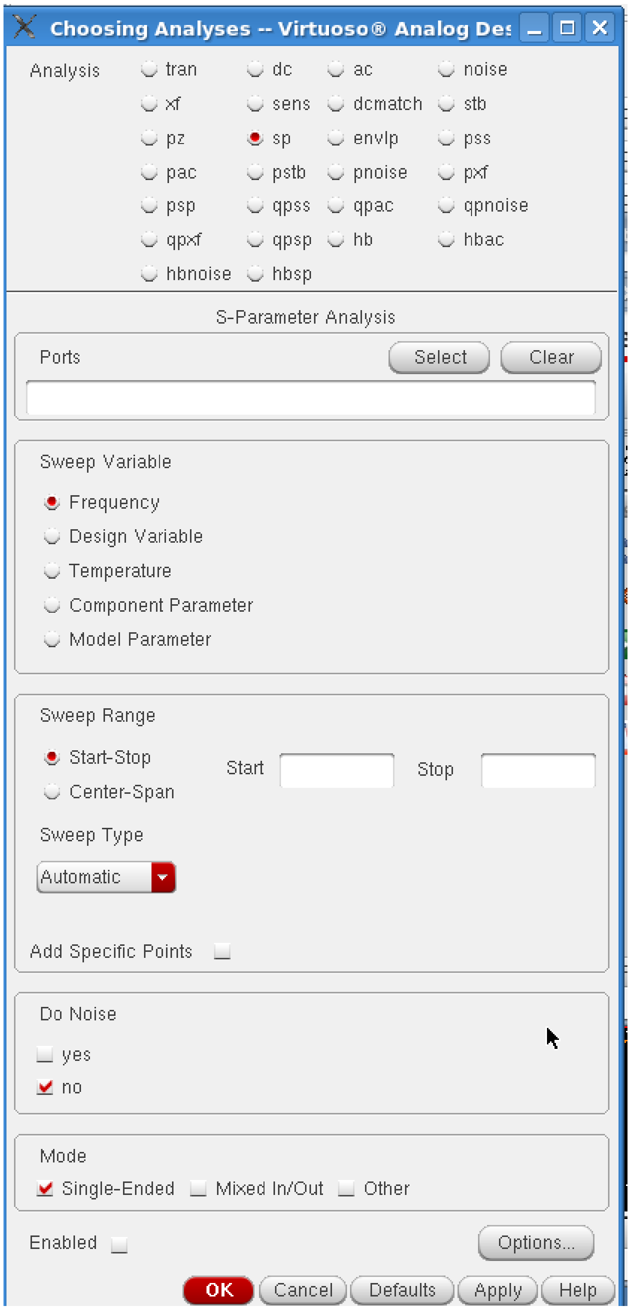
   The *Choosing Analyses* form is displayed.

Figure A-20 Choosing Analyses Form



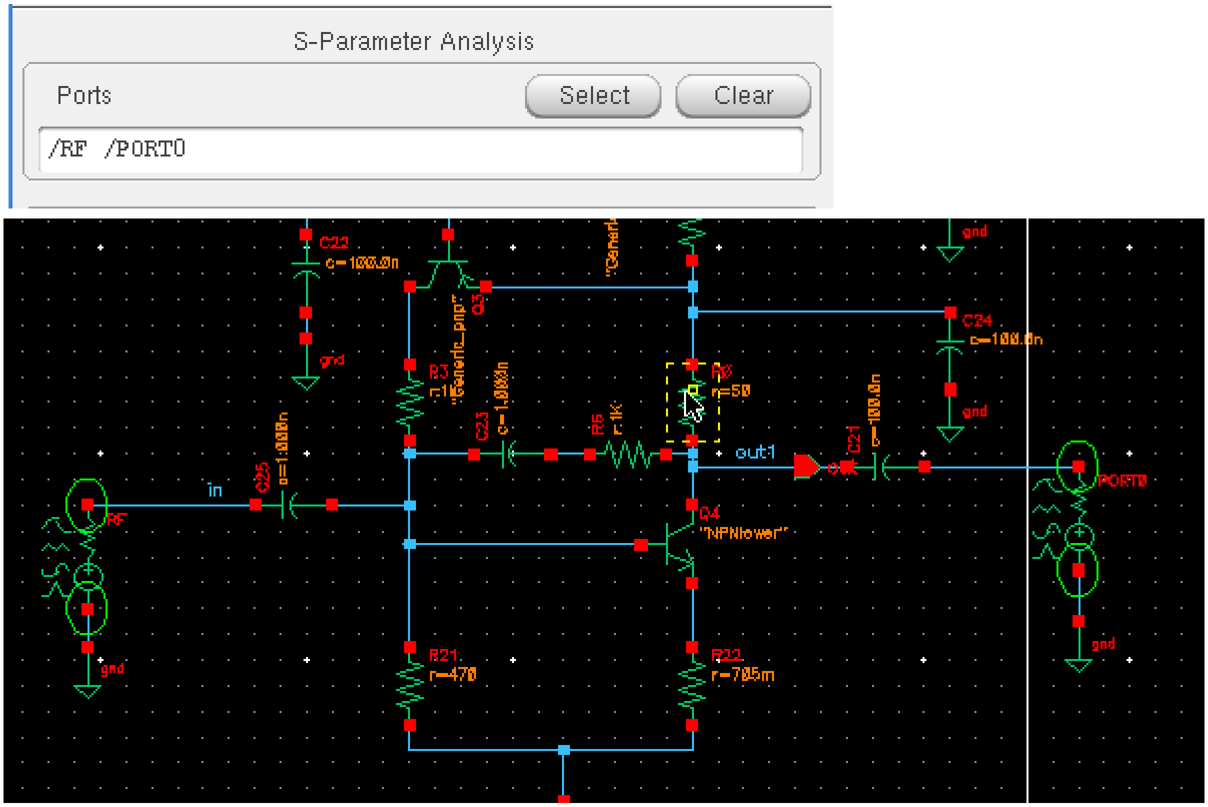
3. In the *Analysis* section, select the *sp* radio button. The form expands.

Figure A-21 sp Choosing Analyses Form



4. In the *Ports* section, click *Select*. Next, click the input (*RF*) port, followed by the output port (*PORT0*). When finished with the two selections, press the *Esc* key. The form and schematic will look like the following:

Figure A-22 Choosing Analyses Form with Ports Selected



If the list of active ports is specified with in the Ports field, the ports are numbered sequentially from one in the order given. Otherwise, all ports present in the circuit are active, and the port numbers used are those that were assigned on the port statements in the netlist (or in the *Edit Properties* form).

Spectre can perform AC/SP analysis while sweeping a parameter. The parameter can be frequency, temperature, component instance parameter, component model parameter, or netlist parameter. If changing a parameter affects the DC operating point, the operating point is recomputed at each step. After the analysis is complete, the modified parameter returns to its original value.

You can define sweep limits by specifying the end points or the center value and span of the sweep. Steps can be linear or logarithmic, and you can specify the number of steps or the size of each step. If you do not specify a step size parameter, the sweep is linear when the ratio of stop to start values is less than 10 and logarithmic when this ratio is 10 or greater. All frequencies are in Hertz.

5. You will be sweeping frequency in this simulation. In the *Sweep Variable* section of the *Choosing Analyses* form, select *Frequency* (this is the default value).

6. In the *Sweep Range* section, select *Start-Stop*. Enter 500M in the *Start* field and 4.0G in the *Stop* field.

7. Set *Sweep Type* to *Linear*. Select *Number of Steps* and set that to *50*.

8. In the *Do Noise* field, select *yes*. This sets up the small signal (linear 2 port) noise analysis. The small signal assumption is valid when the input power level is low (at least 10dB below the 1dB compression point) and the circuit is operating in the linear range.

   When `donoise=yes` is specified, the noise correlation matrix is computed. If in addition, the output is specified using Output probe (oprobe), the amount that each noise source contributes to the output is computed. Finally, if an input is also specified (iprobe), the two-port noise parameters are computed (F, Fmin, NF, NFmin, Gopt, Bopt, and Rn). When an input port is specified, the two-port noise parameters are computed (F, Fmin, NF, NFmin, Gopt, Bopt, and Rn).

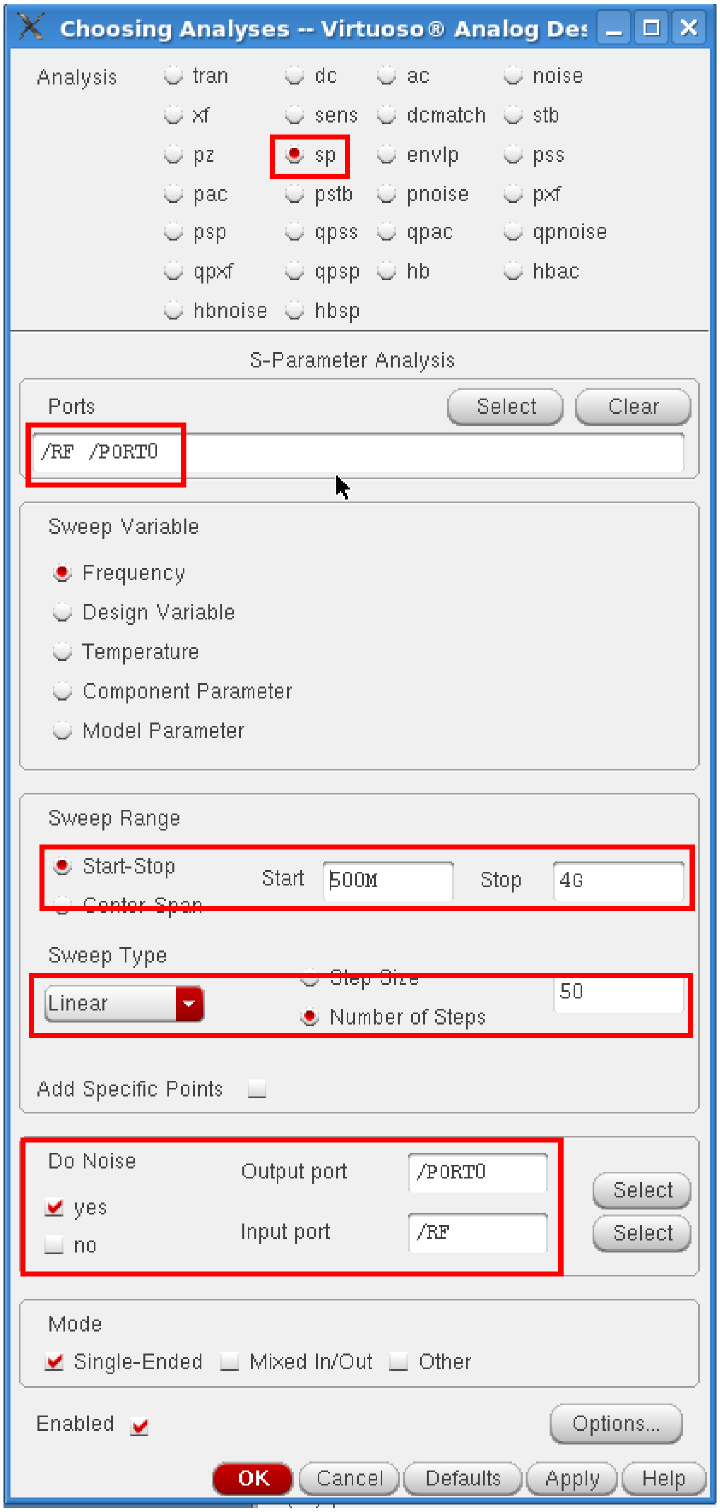
9. Click *Select* to the right of the *Output port* label and click the output port (PORT0) in the schematic.

10. Click *Select* to the right of the *Input port* label and click the input port in the schematic (RF). Alternately, you can type `/load` in the *Output port* field and `/rf` in the *Input port* field.

11. In the *Mode* section, select *Single-Ended*. If you are simulating mixed-mode parameters, select the *Mixed In/Out* option. For more information, see the Virtuoso Specter Circuit Simulator and Accelerated Parallel Simulator User Guide or type `spectre -h sp` at the command prompt.

12. Click *OK* at the bottom of the form. The *Choosing Analyses* form will look like the one below.

Figure A-23 sp Choosing Analyses Form



13. Click *OK* .

Your ADE L simulation window will look like the following:

Figure A-24 Analog Design Environment Simulation Window

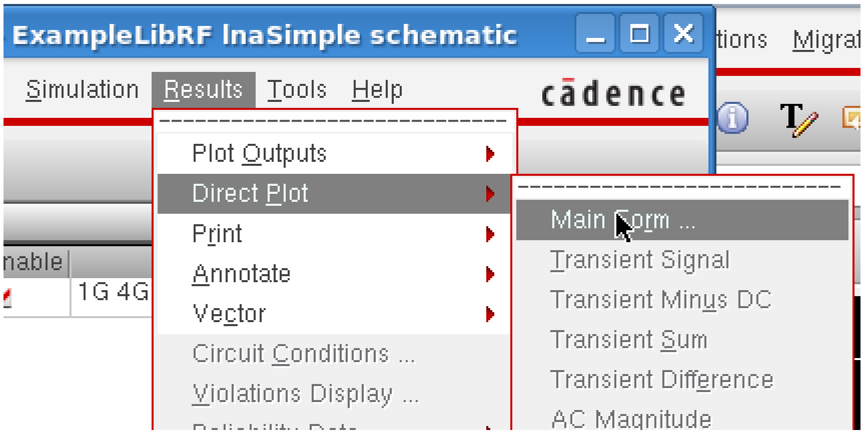
Running the Simulation and Plotting the Results

Start the analyses by clickng the green arrow icon  in ADE L or in the Schematic Editor.

This netlists the design and runs the simulation. A SpectreRF status window isdisplayed (`spectre.out` log file). When the analysis has completed, you may iconify the status window.

1. In the *Analog Design Environment* window, select *Results - Direct Plot - Main Form*.

Figure A-25 Invoking the Direct Plot Form



The *Direct Plot Form* is displayed. Alternately, you can click the *Direct Plot* icon in the Schematic window.

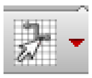
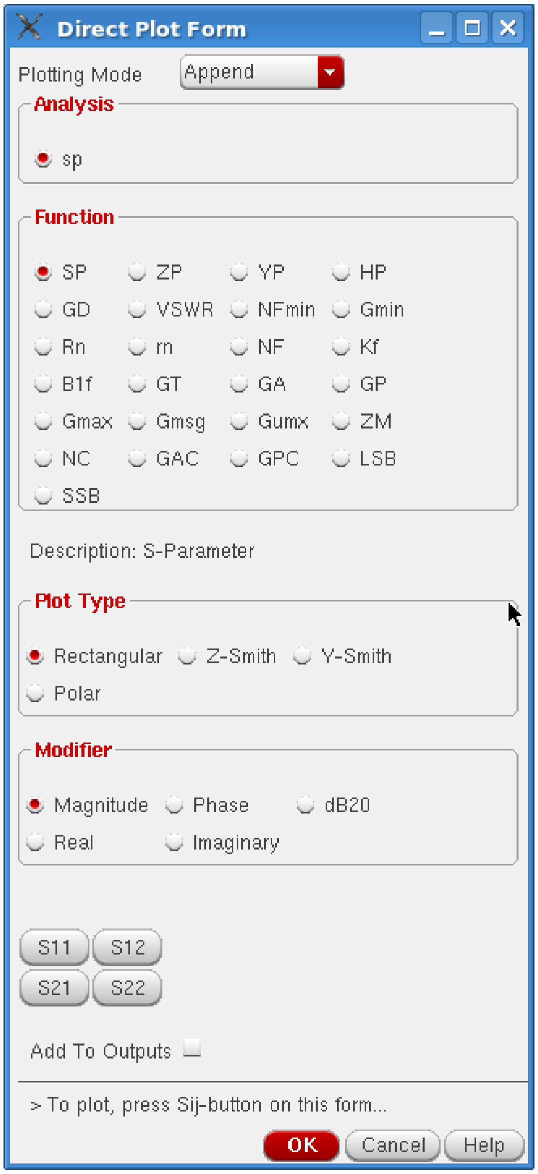


Figure A-26 sp Direct Plot Form



2. In the *Direct Plot Form*, leave the *Plotting Mode* set to *Append* (this is the default).

3. In the *Function* section, select *GT* (for Transducer Gain). Transducer power gain, *GT*, is defined as the ratio between the power delivered to the load and the power available from the source.

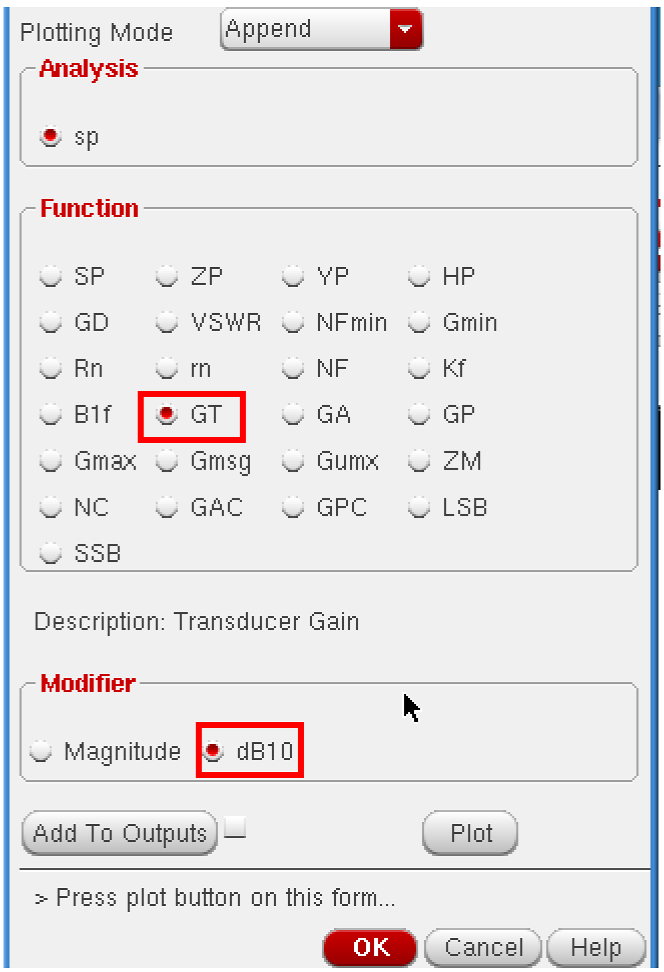
   Note: When using the S-parameter *Direct Plot Form*, the Analog Design Environment assumes that the source (Γ_S) and load (Γ_L) reflection coefficients are zero. *GT*, therefore, plots the insertion gain assuming source and load impedances are matched.

(A-1) Transducer Gain with Source and Load Impedances Matched

$$G_T = |S_{21}|^2$$

4. In the *Modifier* section, select *dB10* because you are plotting power.
5. The sp *Direct Plot Form* will look like the following:

Figure A-27 sp Direct Plot Form



6. Click *Plot*.
7. In the *Function* section, select *GA* (for Available Power Gain). Available power gain, *GA*, is defined as the ratio between the power available from the network and the power available from the source.

Note: When you use the S-parameter *Direct Plot Form*, Γ_S is set to zero, and therefore available gain (*GA*) is plotted as:

(A-2) Available Gain with T_s=0

$$G_A = \frac{|S_{21}|^2}{1 - |S_{22}|^2}$$

8. Click *Plot* again.

9. In the *Function* section, select *GP* (for Power Gain). Power Gain is defined as the ratio between the power delivered to the load and the power input to the network.

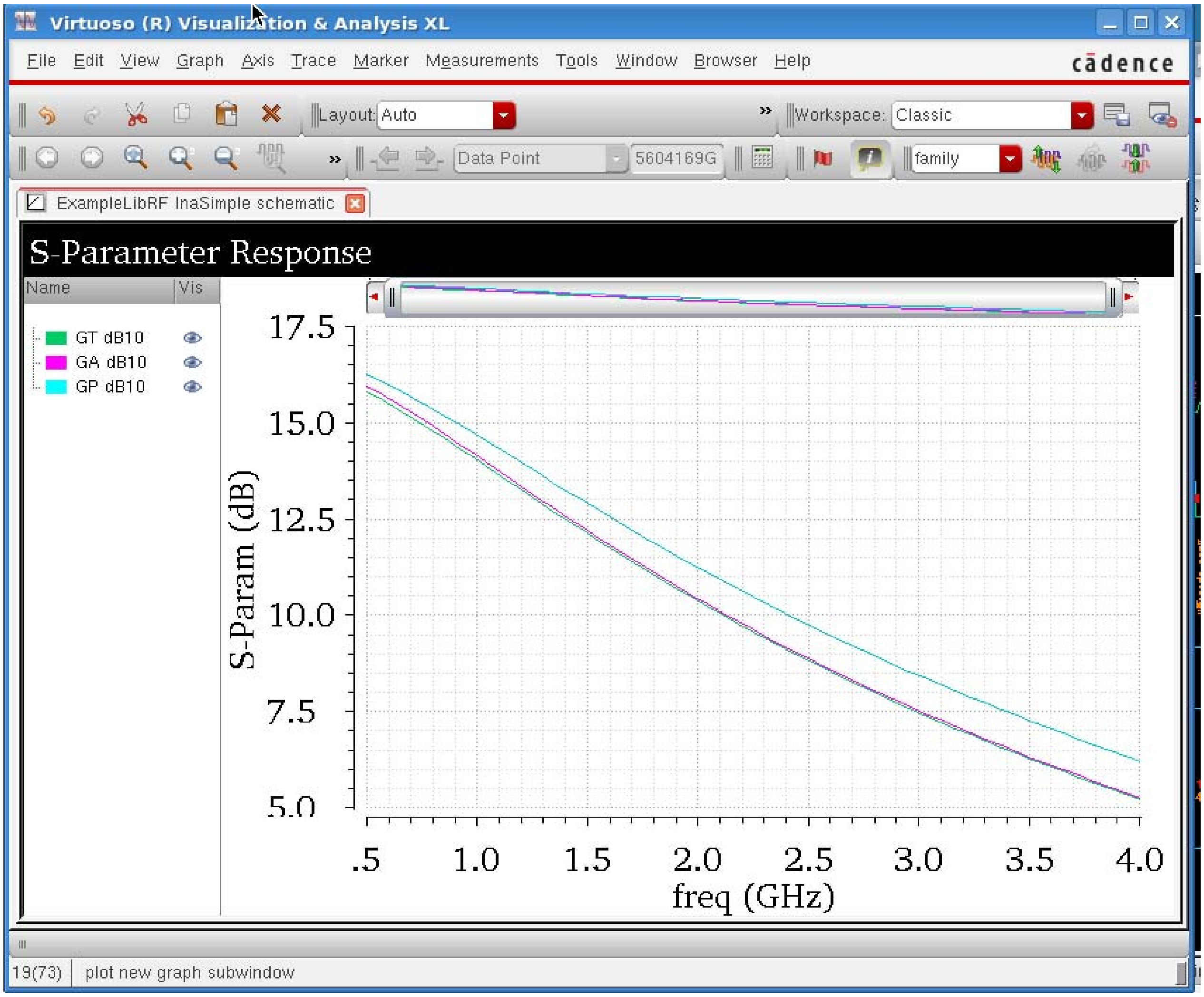
Note: The ADE-L environment assumes that Γ_{in}=0 (the input is matched) so the equation for power gain (*GP*) reduces to the equation below:

(A-3) Power Gain with T_L=0.

$$G_P = \frac{1}{1 - |S_{11}|^2} |S_{21}|^2$$

10. Click *Plot* once more. All three gains (*GT, GA, GP*) plots are displayed on one window, as shown below.

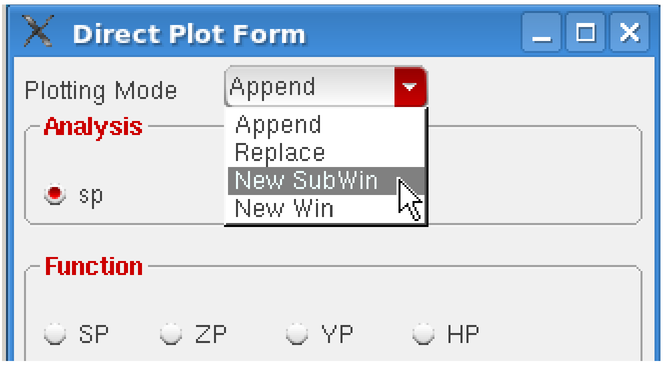
Figure A-28 Transducer, Available, and Power Gain



Note that *GT* is the smallest gain. Because the power available from the source is greater than the power input to the LNA network, the Power Gain is greater than the Transducer gain (*GP* > *GT*). The closer the two gains are, the better the input matching is. Similarly, because the power available from the LNA network is greater than the power delivered to the load , *GA* > *GP*. The closer these two gains are, the better the output matching is. The power gain *GP* is closer to the transducer gain *GT* than the available gain *GA* which means the input matching network is properly designed. That is, S11 is close to zero.

11. In the *Direct Plot Form*, change the *Plot Mode* to *New SubWin* (new subwindow).

Figure A-29 Changing Plot Mode to New SubWindow



12. In the *Function* section, select *Gmax* (for maximum Transducer Power Gain) and click *Plot*.

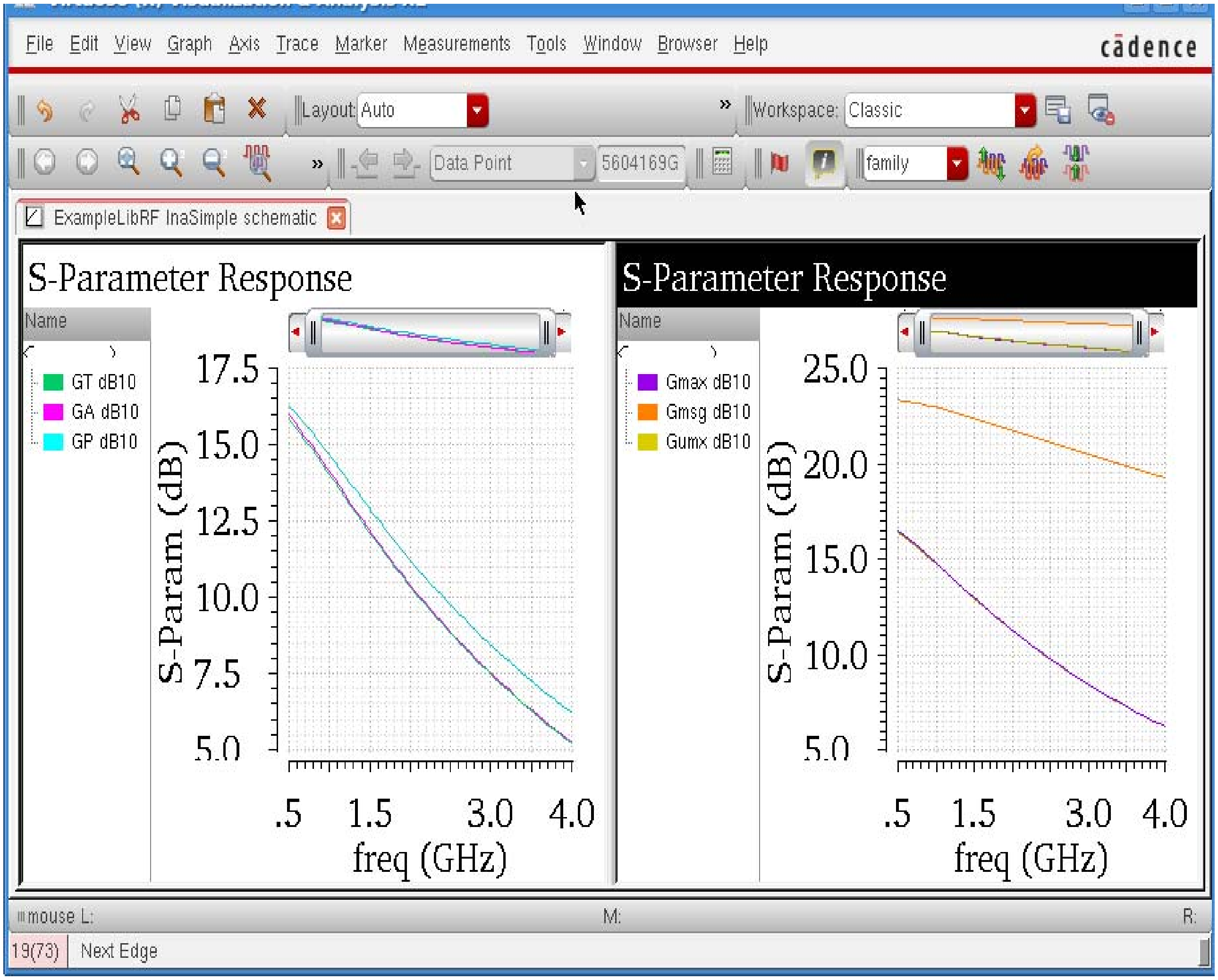
13. Change the *Plotting Mode* to *Append*.

14. In the *Function* section, select *Gmsg* (for Maximum Stability Gain), and Click *Plot*.

15. In the *Function* section, select *Gumx* (for maximum Unilateral Transducer Power Gain), and click *Plot* again.

The three waveforms are appended to the previous graph, as shown below.

Figure A-30 Adding Maximum Available, Maximum Stable, and Maximum Unilateral Gain Plots



- Maximum unilateral transducer power gain (*Gumx*) is the transducer power gain when you assume that the reverse coupling of the LNA S_{12}, is zero, and the source and load impedances are conjugately matched to the LNA. That is $S11=\Gamma_S$ and $S22=\Gamma_L$
- Maximum transducer power gain, *Gmax*, is the simultaneous conjugate matching power gain when both the input and output are conjugately matched.
- Maximum stable gain, *Gmsg*, is the maximum of *Gmax* when the stability condition, K > 1, is satisfied.

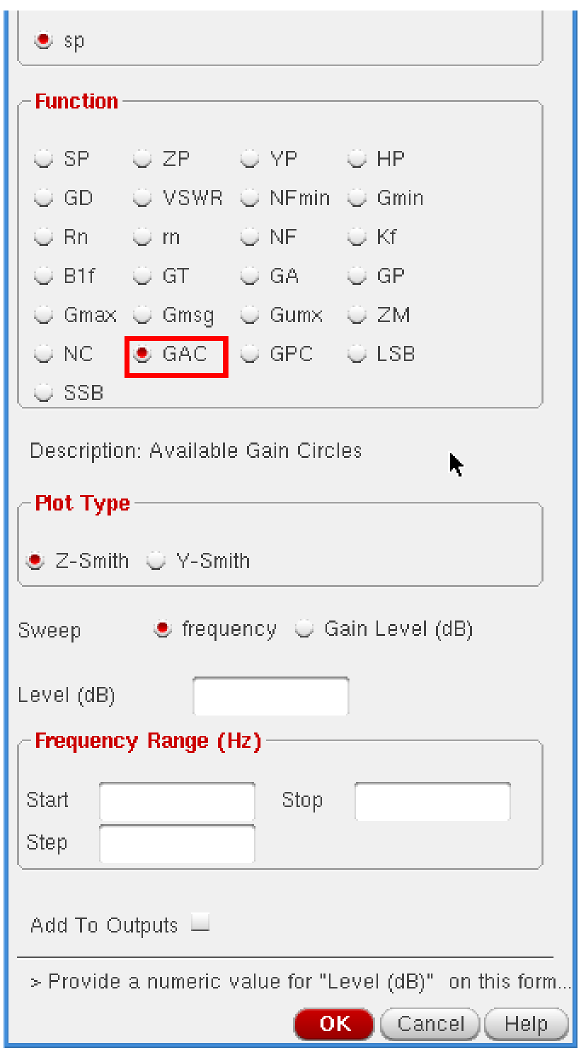
Note: Equations for *Gumx*, *Gmax*, and *Gmsg* and a discussion of the Stability factor *K* are discussed in Chapter 8, "AnalogLib Components Used in RF Simulation,"

In the plots above, the maximum unilateral transducer power gain (*Gumx)* is very close to the maximum transducer power gain (*Gmax)* which means the reverse coupling S12 is small. The maximum stable gain (*Gmsg*) is the largest of the six gains plotted.

1. In the ViVA waveform window, choose *File -Close All Windows* to close the waveform window.

2. Next, you plot the gain circles. There are two types of gain circles: Power Gain Circles and Available Gain Circles. In the *Function* section of the *Direct Plot Form*, select *GAC* (Available Gain Circles). The form expands.

Figure A-31 Direct Plot Form for Available Gain Circles



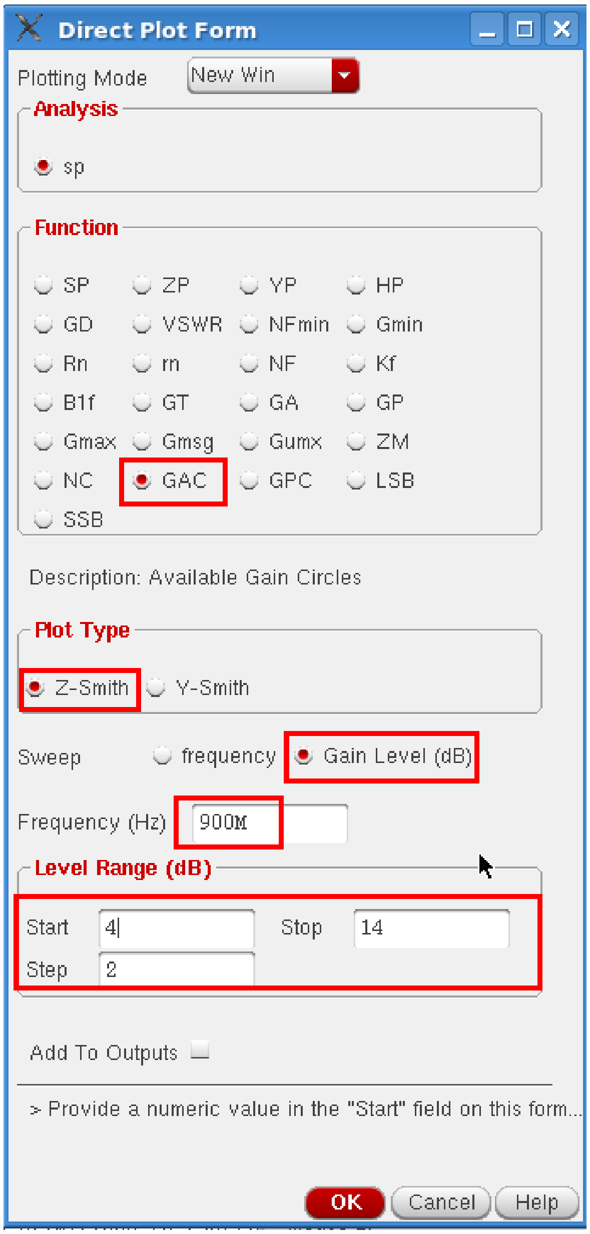
Available Gain Circle (GAC)

Available Gain (GA) is solely a function of the source reflection coefficient Γ_S . Thus, you can draw available gain contours on the Smith chart of Γ_S . The location for the peak of the contour corresponds to Γ_S producing the maximum available gain (*GA*). You can move the peak location by changing the design of the input matching network. The best location for the contour peak is at the center of the Smith chart, where Γ_S=0.

3. In the *Plot Type* section, choose *Z-Smith*. You will be plotting Gain Circles on the Impedance Smith Chart. (*Y-Smith* plots on the Admittance Smith Chart).
4. In the *Sweep* section, you can either choose *Gain Level (dB)* or *Frequency (Hz).* In this case, you will be sweeping Gain Level. The Frequency you specify depends on the operating frequency of your design. Since the lnaSimple circuit operates at 900MHz, enter that value in the *Frequency (Hz)* field.
5. In the *Level Range (dB)* section, set *Start* to 4, *Stop* to 14, and *Step* to 2.

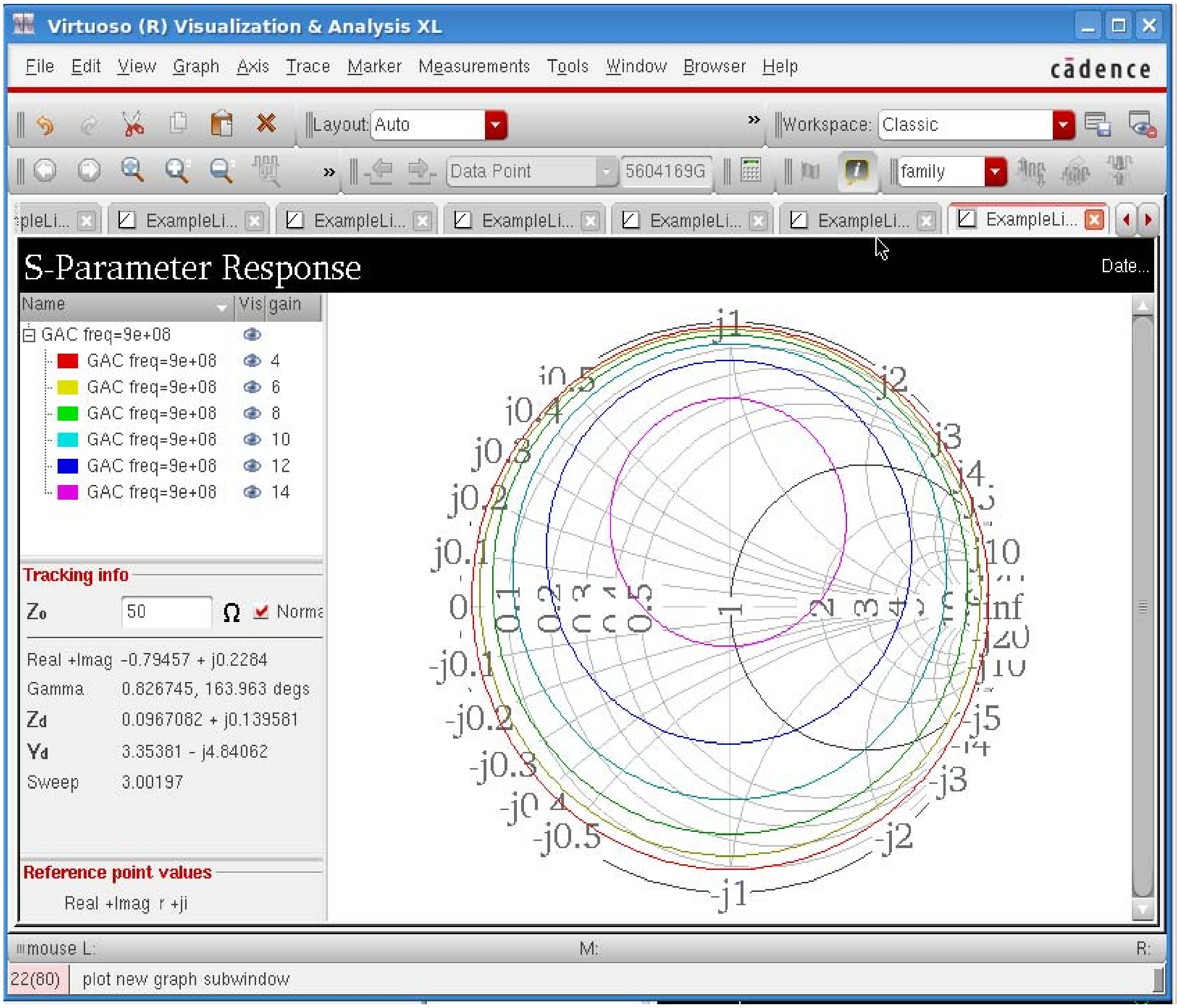
The *Direct Plot Form* should look like the following:

Figure A-32 Direct Plot Form for Plotting Available Gain Circles at a Constant Frequency



6. Click *Plot*. The available gain circles are plotted in the waveform window, as shown below.

Figure A-33 Available Gain Circles on Z-Smith Chart



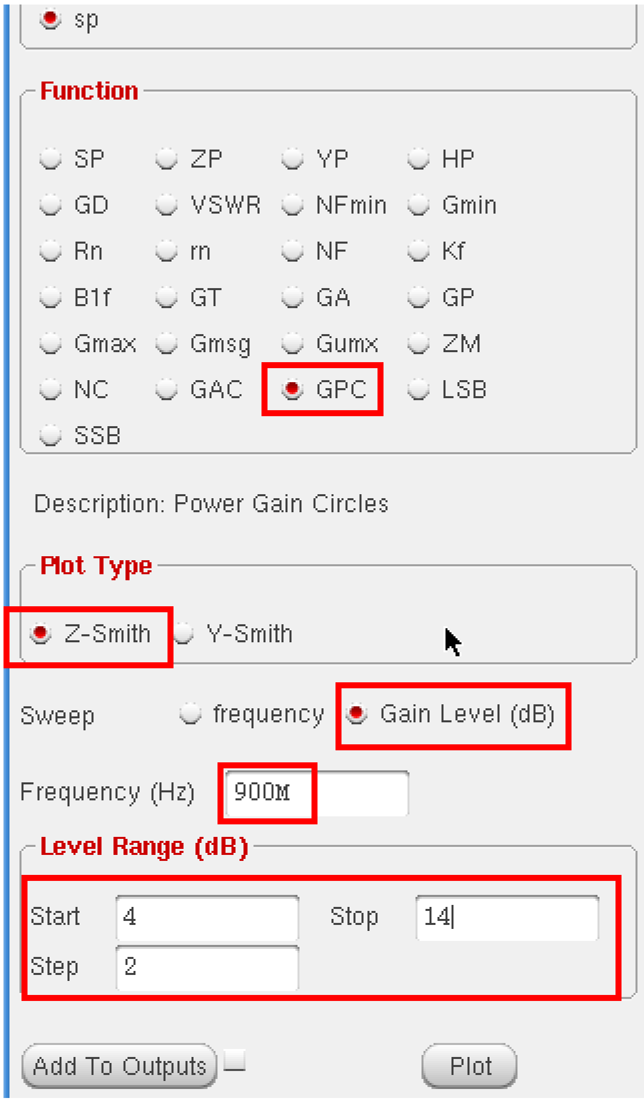
As you move your cursor around one of the Available Gain Circles, notice that the Tracking Cursor will read out both the Real and Imaginary part or the reflection coefficient directly from the Smith Chart in the *Tracking Info* section on the left side of the Smith Chart. The impedance or admittance at that point is also shown in the *Reference point values* section of the Legend.

7. In the *Direct Plot Form*, change the *Plot Mode* to *New Window.* Next, plot the Power gain circle (*GPC*).

Power Gain (*GP*) is solely a function of the load reflection coefficient Γ_L. Thus, you can draw the power gain contours on the Smith chart of Γ_L. The location for the peak of the contour corresponds to Γ_L producing the maximum power gain (*GP*). You can move the peak location by changing the design of the output matching network. The best location for the contour peak is at the center of the Smith chart, where $\Gamma_L = 0$.

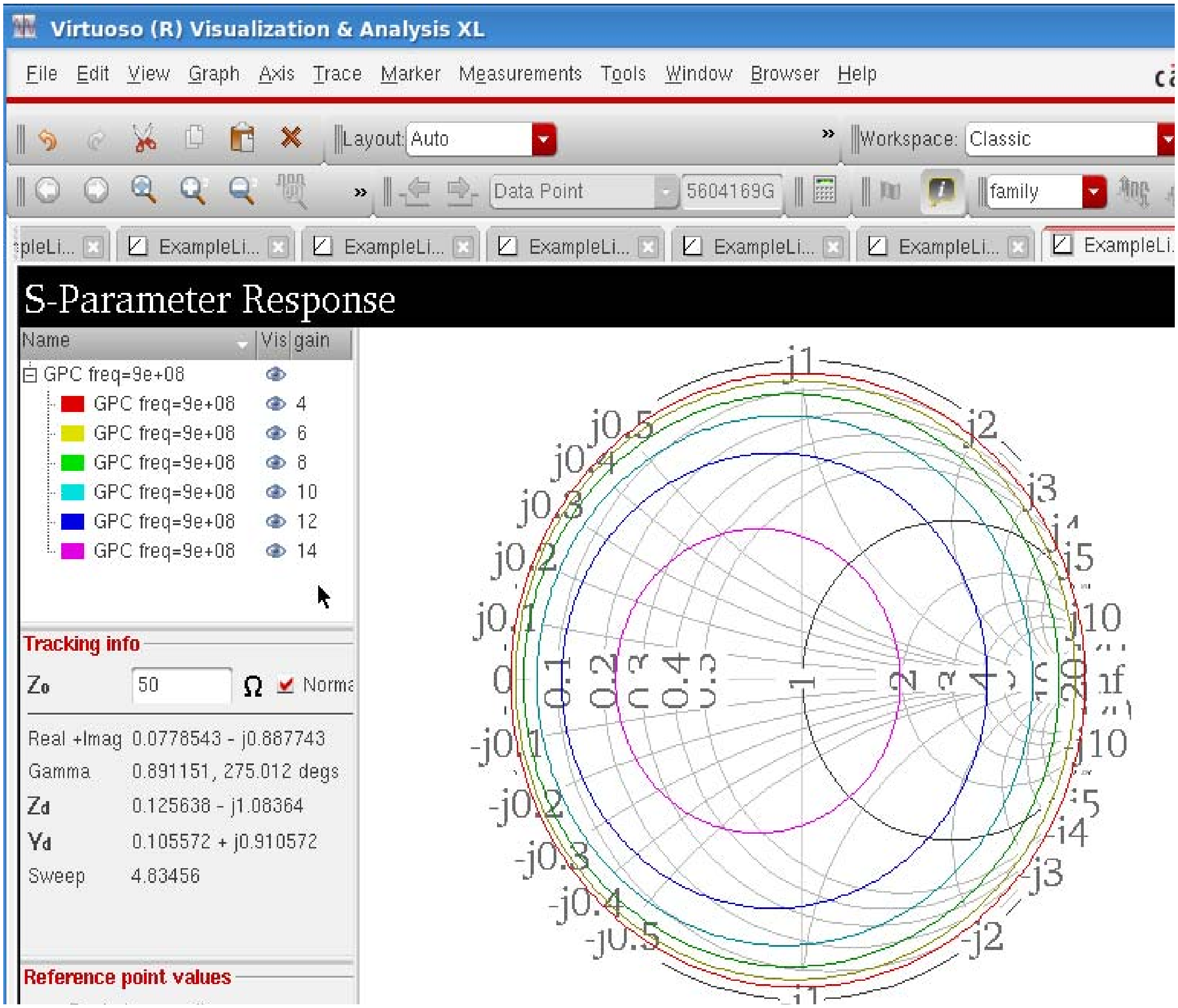
8. In the sp *Direct Plot Form*, select*GPC* in the *Function* section.
9. In the *Plot Type* section, select *Z-Smith* (it should be the default). You will be plotting Gain Circles on the Impedance Smith Chart.
10. In the *Sweep* section, your choices are *frequency* and *Gain Level (dB)*. Choose *Gain Level (dB)*. Since the lnaSimple operates at 900MHz, enter 900M in the *Frequency (Hz)* field.
11. In the *Gain Level (dB)* section, set *Start* to 4, set *Stop* to 14, and *Step* to 2. The *Direct Plot Form* should look like the following:

Figure A-34 Direct Plot Form for Power Gain Circles



12. Click *Plot*. The power gain circles are plotted on the Smith Chart, as shown below.

Figure A-35 Power Gain Circles



Next, you will look at circuit stability and plot Stability Circles.

Stability

After running an sp analysis, you can plot stability factor and stability circles.

Kf, the stability factor, is valid for two-port circuits only. Kf is defined as:

Figure A-36 Equation for Kf, stability factor

$$K_f = \frac{1 - |S_{11}|^2 - |S_{22}|^2 + |D|^2}{2|S_{21}||S_{12}|}$$

-

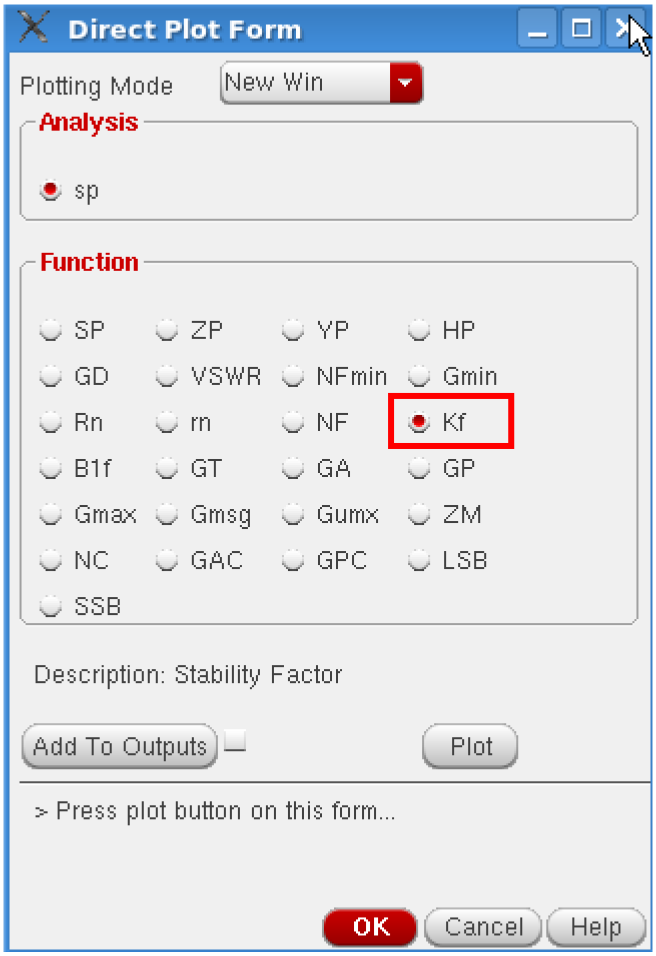
$$D = S_{11}S_{22} - S_{21}S_{12}$$

The above equation is valid for small-signal stability only. Under large signal conditions, the circuit is less likely to be stable. In the presence of feedback paths from the output to the input, the circuit might become unstable for certain combinations of source and load impedances. An LNA design that is normally stable might oscillate at the extremes of the manufacturing or voltage variations, and perhaps at unexpectedly high or low frequencies. When K > 1 and D < 1, the circuit is unconditionally stable. That is, the circuit does not oscillate with any combination of source and load impedances. You should perform the stability evaluation for the S parameters over a wide frequency range to ensure that K remains greater than one at all frequencies.

1. In the *Function* section of the sp Direct Plot form, select *Kf*.
2. Set the *Plotting Mode* to *New Window*.

   The sp *Direct Plot Form* should look like the following:

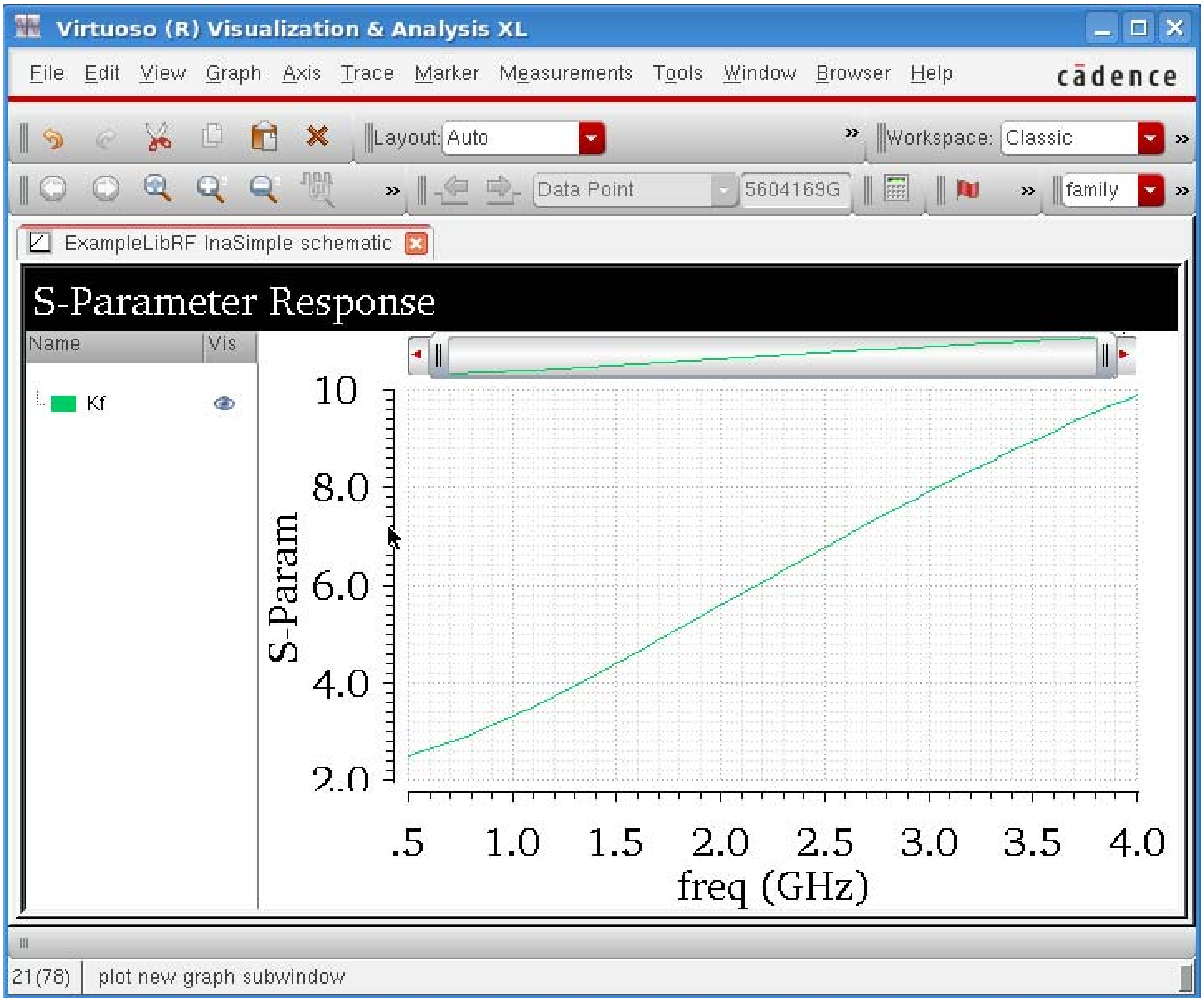
Figure A-37 Plotting Kf from the sp Direct Plot Form



3. Click *OK*.

The waveform window is displayed, as shown below.

Figure A-38 Plotting Stability Factor K_f vs Input Frequency



The stability factor *Kf i*s greater than 1 for all frequencies viewed, indicating that the circuit is stable at these frequencies. As the coupling (S12) decreases (reverse isolation increases), stability improves. You might use techniques, such as resistive loading and neutralization to improve stability for an LNA.

Next, plot the Stability Circles.

4. Close the waveform window by choosing *File - Close All Windows* and go back to the *Direct Plot Form*.

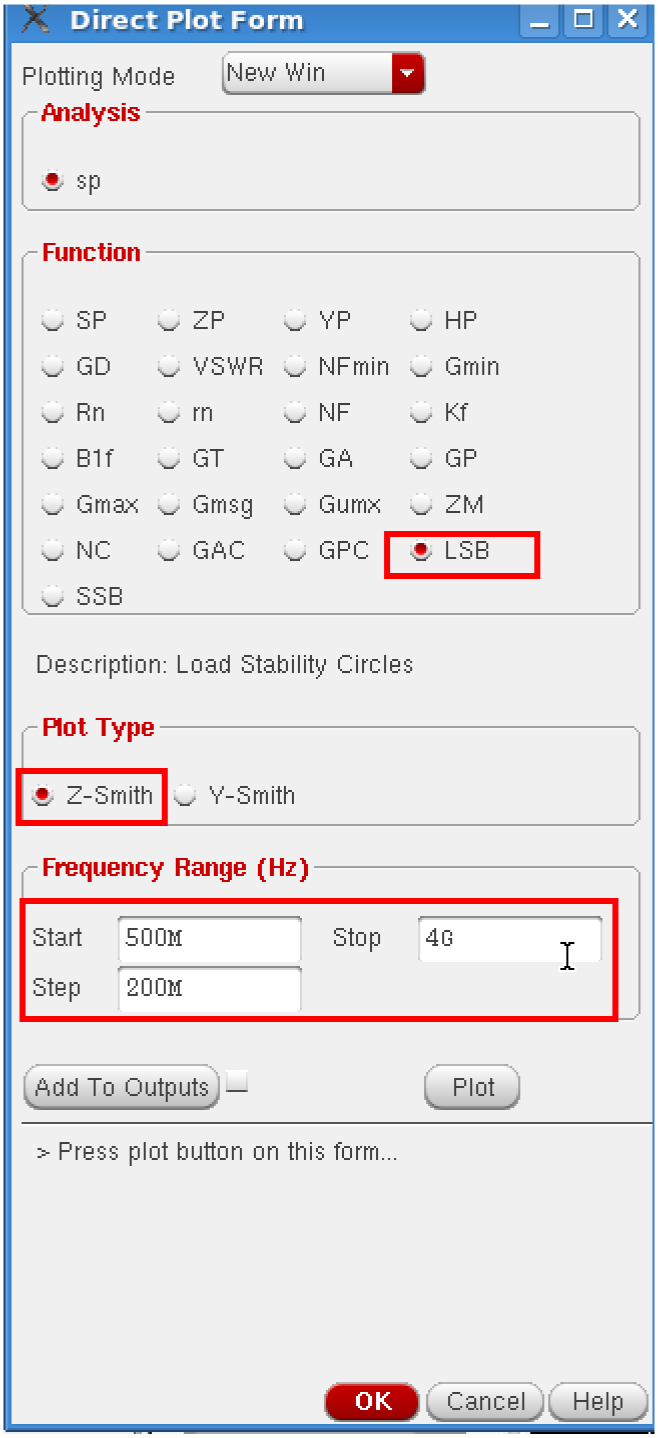
5. In the *Function* section, select *LSB* (Load Stability Circles). The form changes.

6. In the *Plot Type* section, choose *Z-Smith* (this is the default). You will be plotting Load Stability Circles (*LSB*) on the Impedance Smith Chart. (*Y-Smith* plots on the Admittance Smith Chart).

7. In the *Frequency Range (Hz)* section, enter *Start* 500M, *Stop* 4G, and *Step* 200M. The *Direct Plot* form should look like the following:

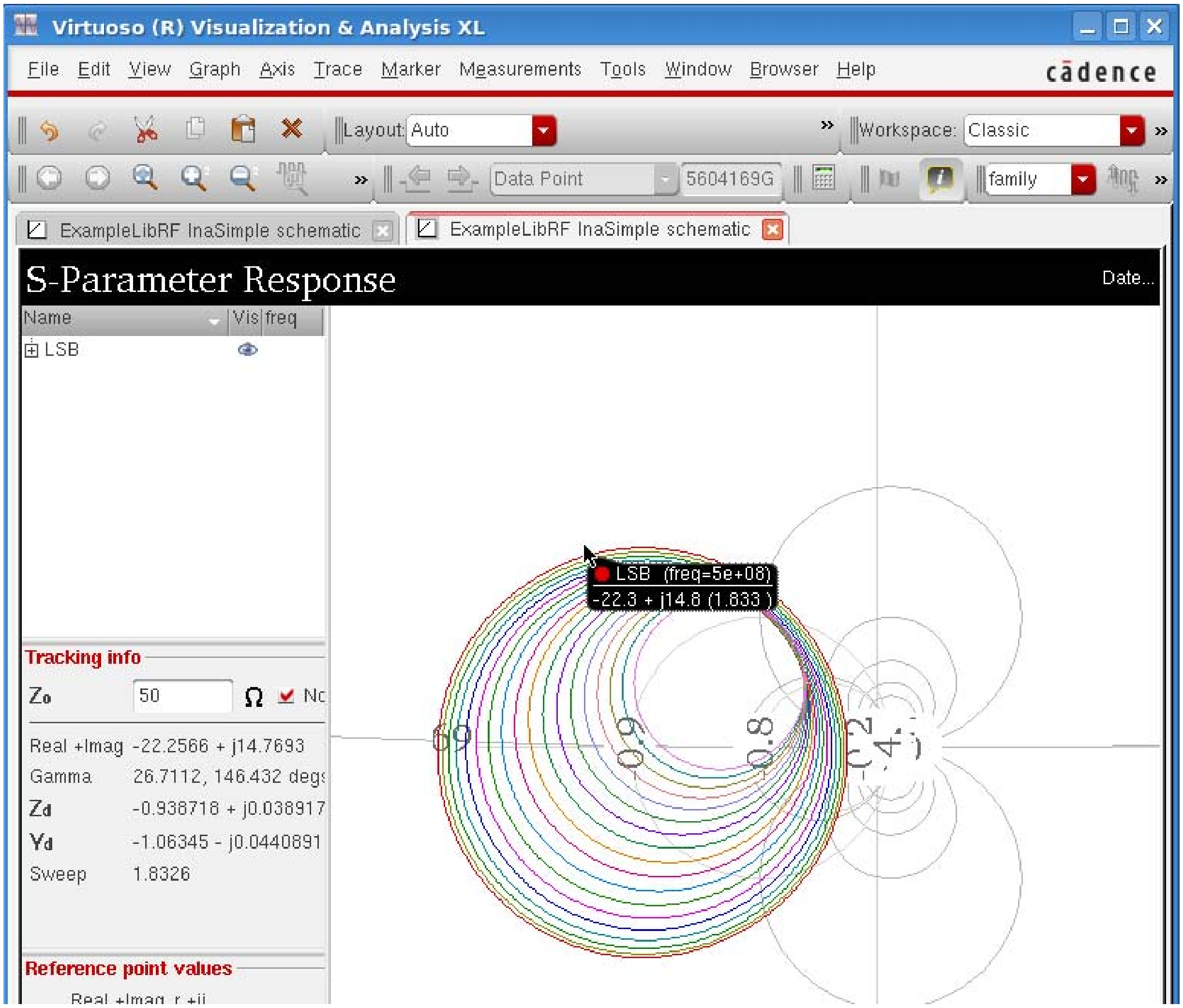
Figure A-39 Direct Plot Form for Plotting Load Stability Circles

.



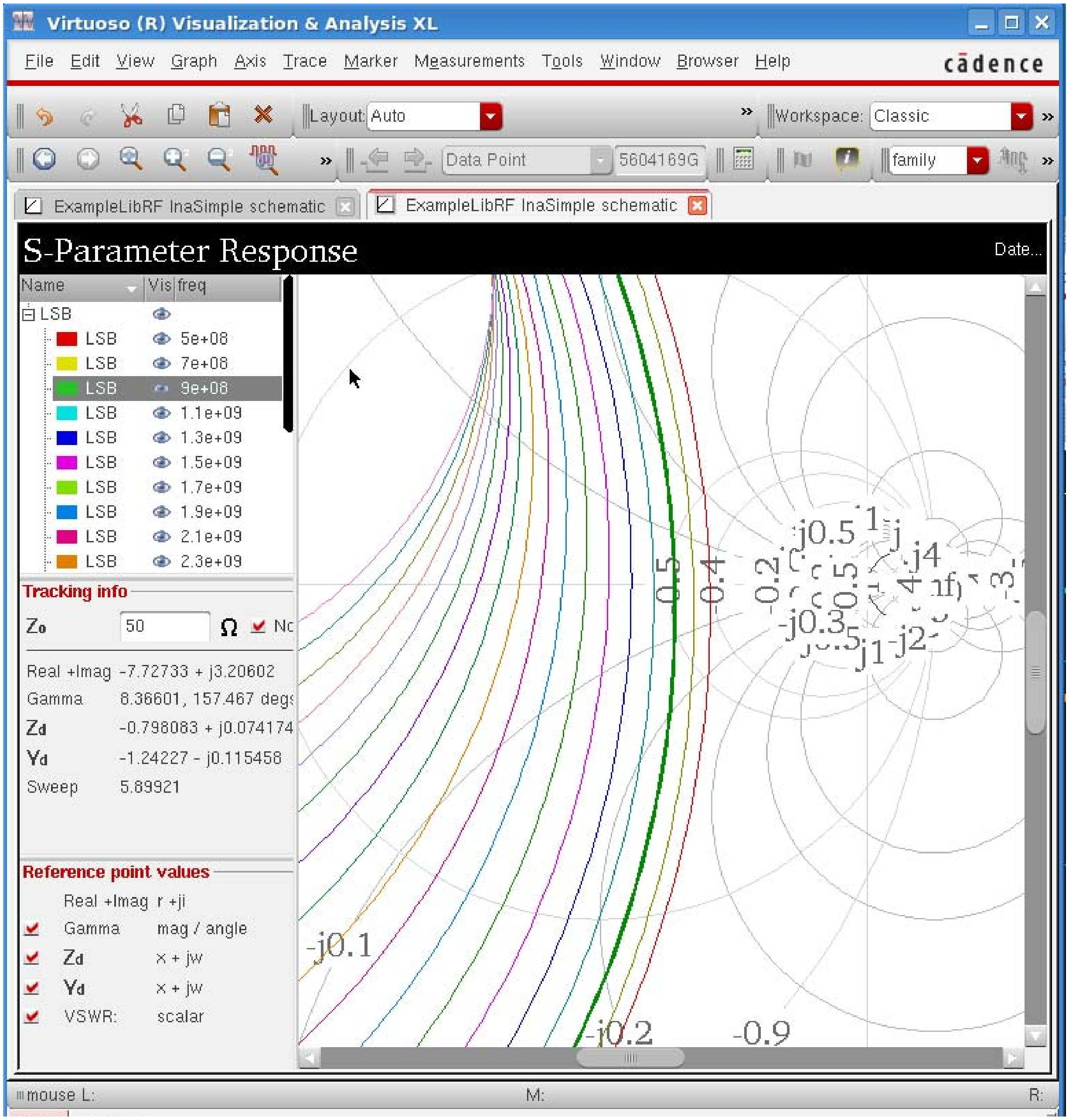
8. Click *Plot*. The Load Stability Circles are plotted, as shown below.

Figure A-40 Load Stability Circles



By default, the graph is zoomed out to show all traces. Zoom into the Smith Chart by holding down the right mouse button and dragging a square around the section of the Smith Chart you would like to view. When you release the button, the graph redraws. You can also determine which trace belongs to which frequency by clicking on the + button to the left of *LSB* in the upper left section of the graph legend. This is shown in the next figure.

Figure A-41 Zoomed in Load Stability Circles



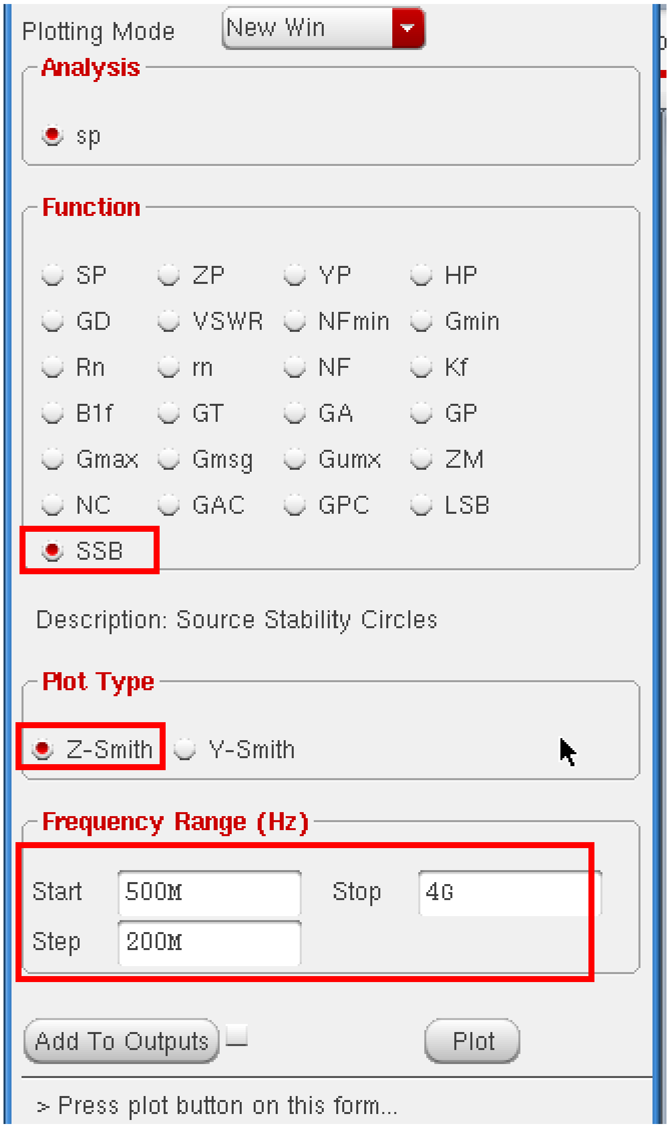
Next, you will be plotting the Source Stability Circles.

9. In the *Function* section, select *SSB* (Source Stability Circles). The form changes.

10. In the *Plot Type* section, choose *Z-Smith* (this is the default). You will be plotting Source Stability Circles on the Impedance Smith Chart. (*Y-Smith* plots on the Admittance Smith Chart).

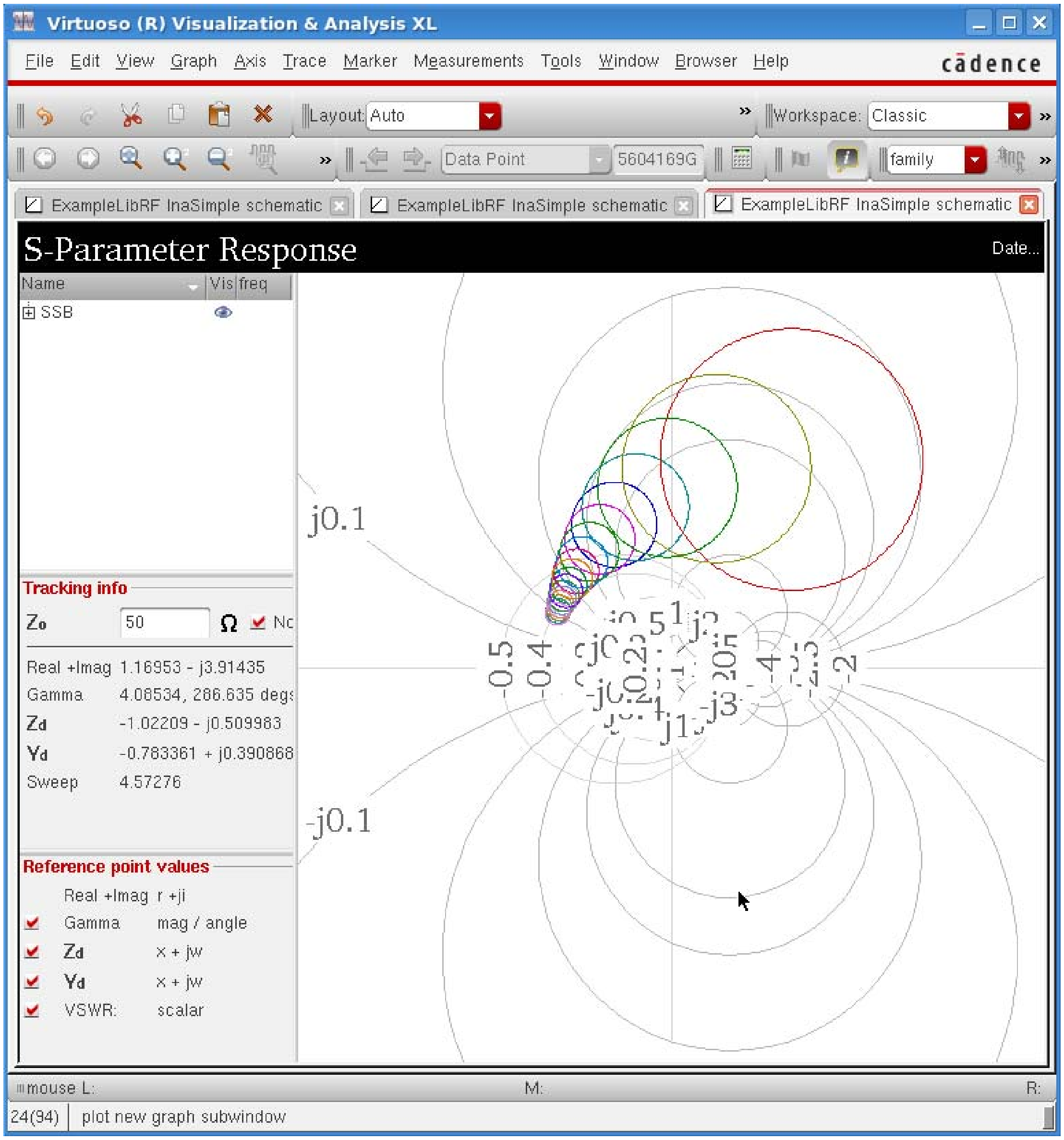
11. In the *Frequency Range (Hz)* section, enter *Start* 500M, *Stop* 4G, and *Step* 200M. The *Direct Plot Form* should look like the following:

Figure A-42 Direct Plot Form for Source Stability Circles



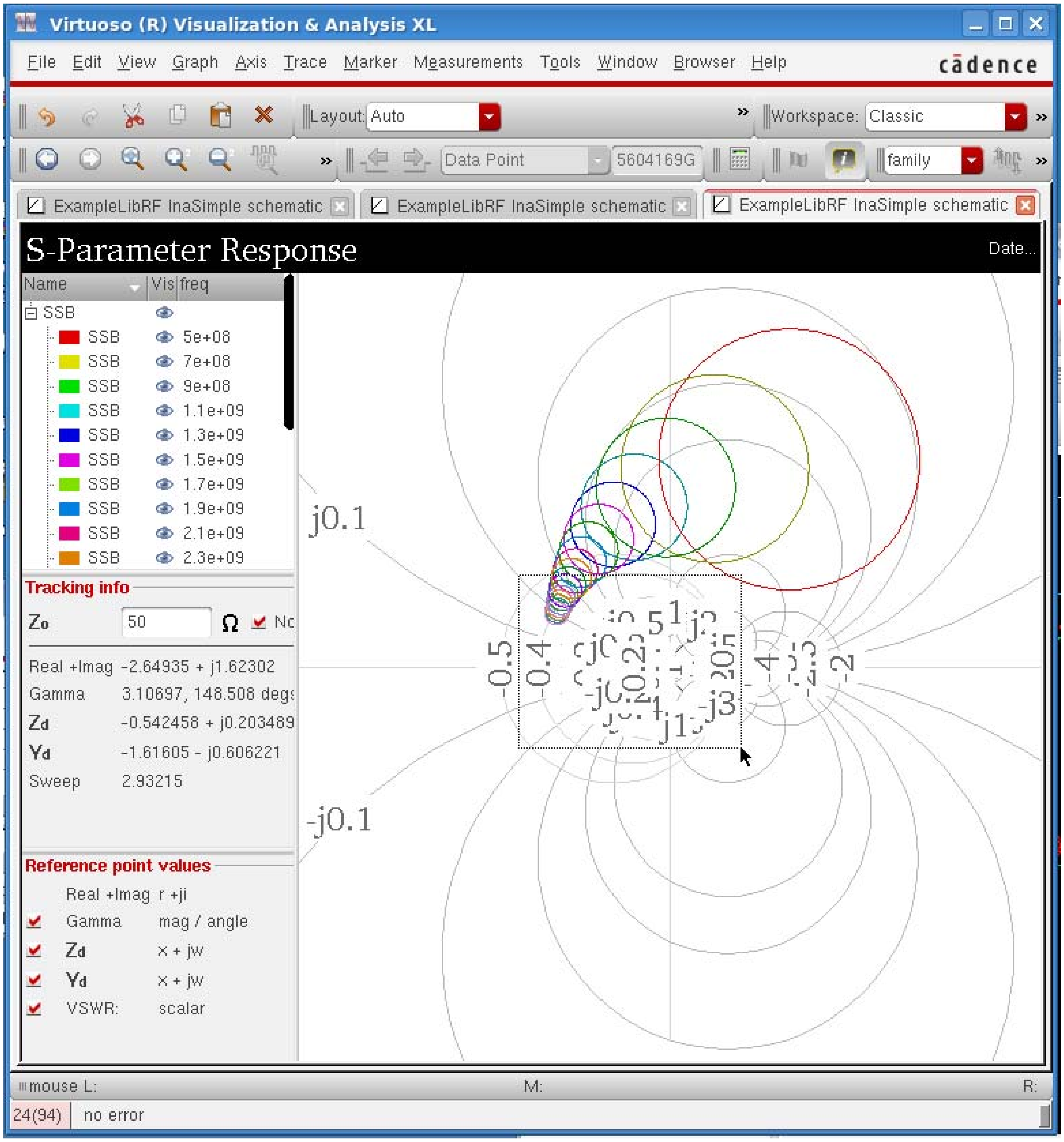
12. Click Plot. The Source Stability Circles are plotted,as shown below.

Figure A-43 Source Stability Circles for lnaSimple



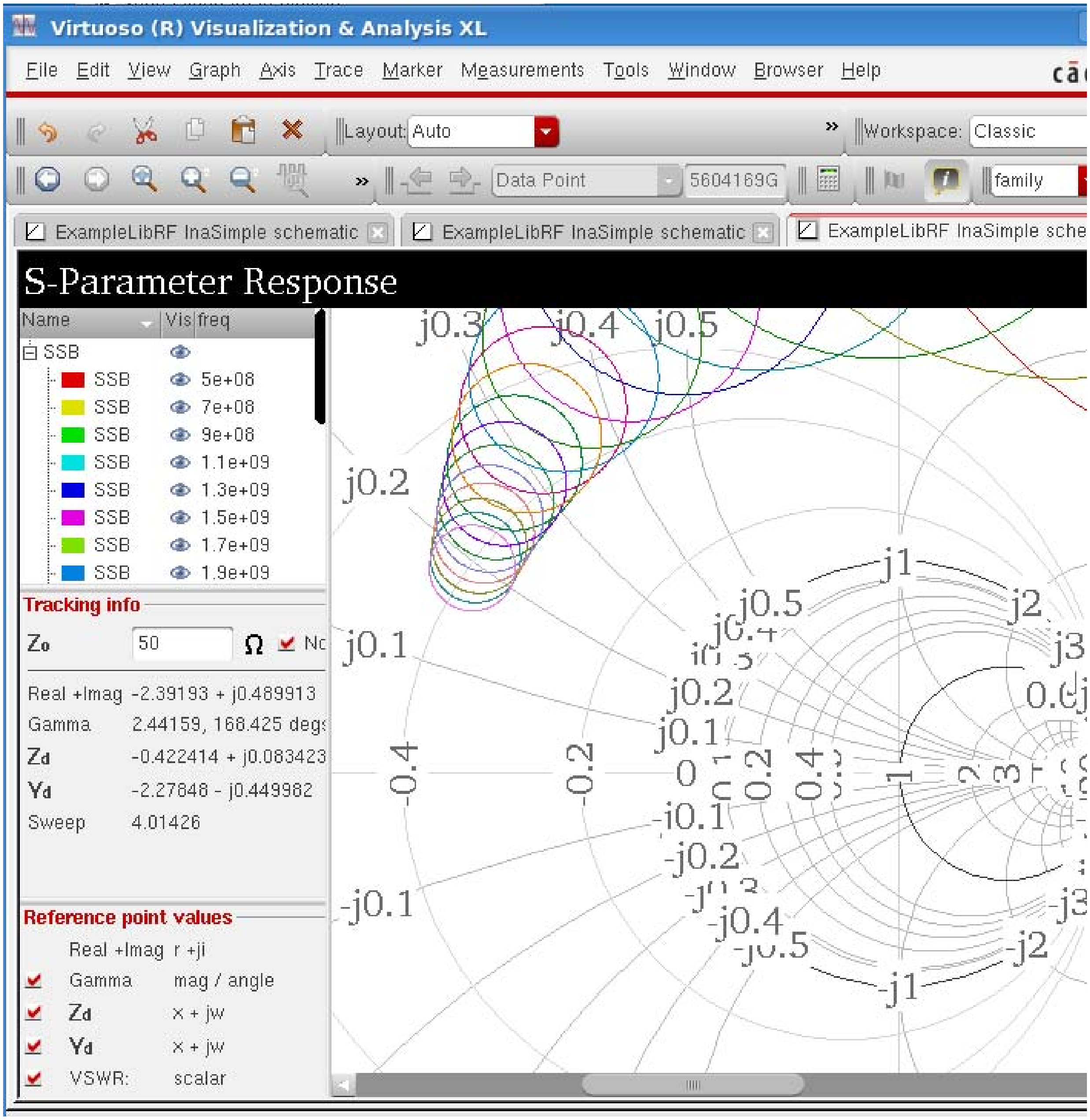
By default, the graph is zoomed out to show all traces. Zoom into the Smith Chart by holding down the right mouse button and dragging a square around the section of the Smith Chart you would like to view. When you release the button, the graph redraws. You can also determine which trace belongs to which frequency by clicking on the + button to the left of *SSB* in the upper left section of the graph legend. This is shown in the next figure.

Figure A-44 Zoom in Source Stability Circles



When you release the mouse button, the plot redraws, as shown in the next figure.

Figure A-45 Zoomed in View of Source Stability Circles



The source and load stability circles are also useful for checking for LNA stability. The input stability circle draws the circle Γ_{out} = 1 out on the Smith chart of Γ_S . The output stability circle draws the circle Γ_{in} = 1 on the Smith chart of Γ_L.

The non-stable regions of the two circles should be far away from the center of the Smith chart. In fact, it is better if the non-stable regions are located outside the Smith chart circles. This is the case for both Load and Source Stability circles.

13. In the ViVA waveform window, select *File - Close All Windows*. The waveform window closes.

The next measurement you will make is Noise Figure.

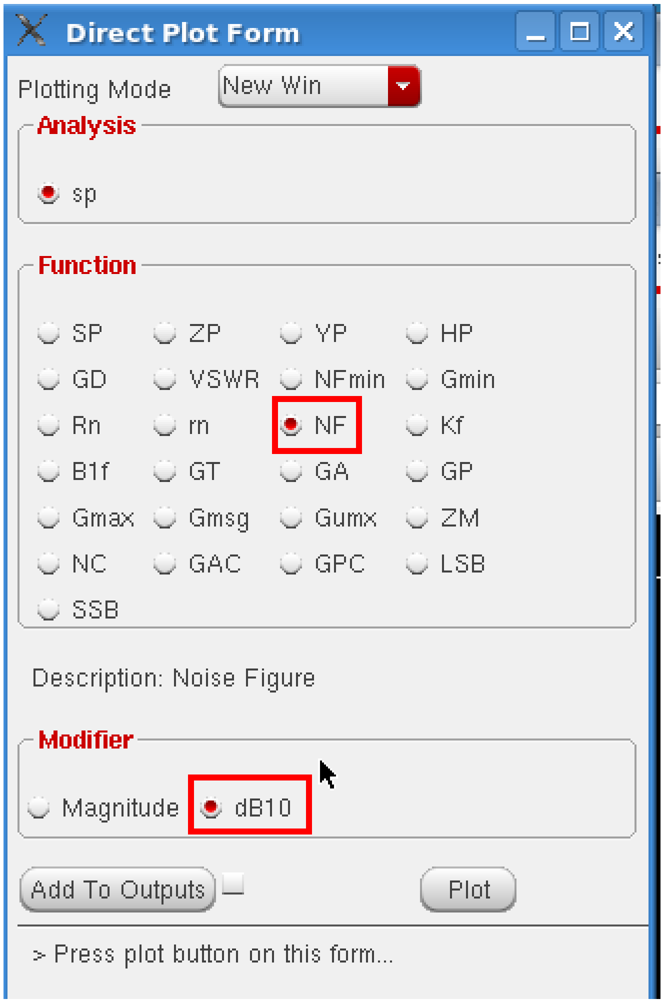
Linear 2-port noise analysis (NF, NFmin) and Noise circles

For cascaded stages, the overall noise figure is mainly determined by the first amplification stage, provided that it has sufficient gain. You achieve low noise performance by carefully selecting the low noise transistor, DC biasing point, and noise-matching at the input. The noise performance is characterized by noise factor, F, which is defined as the ratio between the input signal-to-noise ratio and the output signal-to-noise ratio. For equations to Noise Figure and minimum Noise Figure, see Noise Calculations in the Simulator on page 363.

You have already run the simulation for linear two-port noise as part of the sp analysis. Now, you will plot the results.

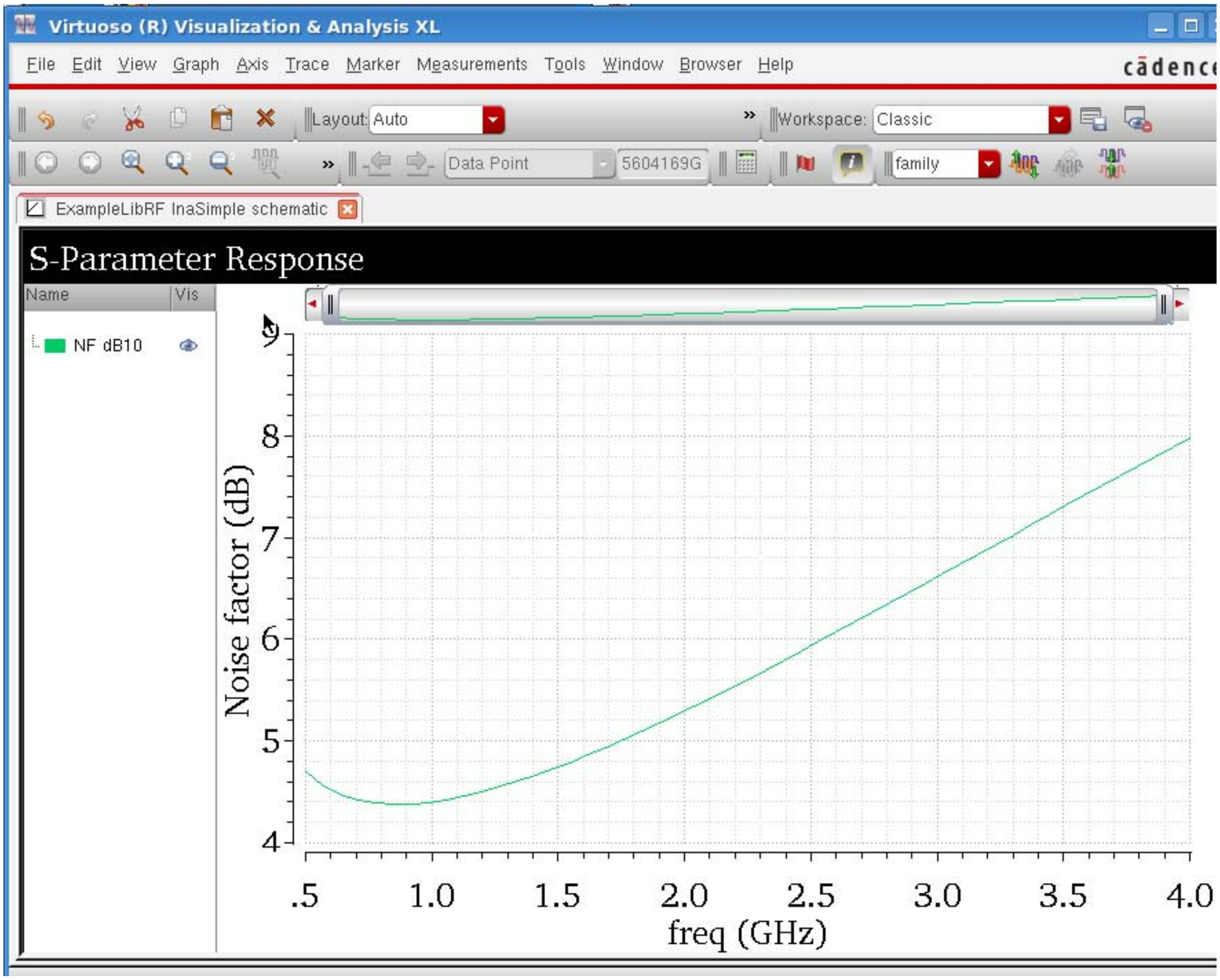
1. In the sp *Direct Plot Form,* select *NF*in the *Function* section. The form changes.
2. In the *Modifier* section, select *dB10* to plot Noise Figure. (If you wanted to plot Noise Factor instead, choose *Magnitude*). The sp *Direct Plot Form* should look like the figure below.

Figure A-46 sp Direct Plot Form for Plotting Noise Figure



3. Click *Plot*. The noise figure is plotted,as shown below.

Figure A-47 lnaSimple Noise Figure Plot



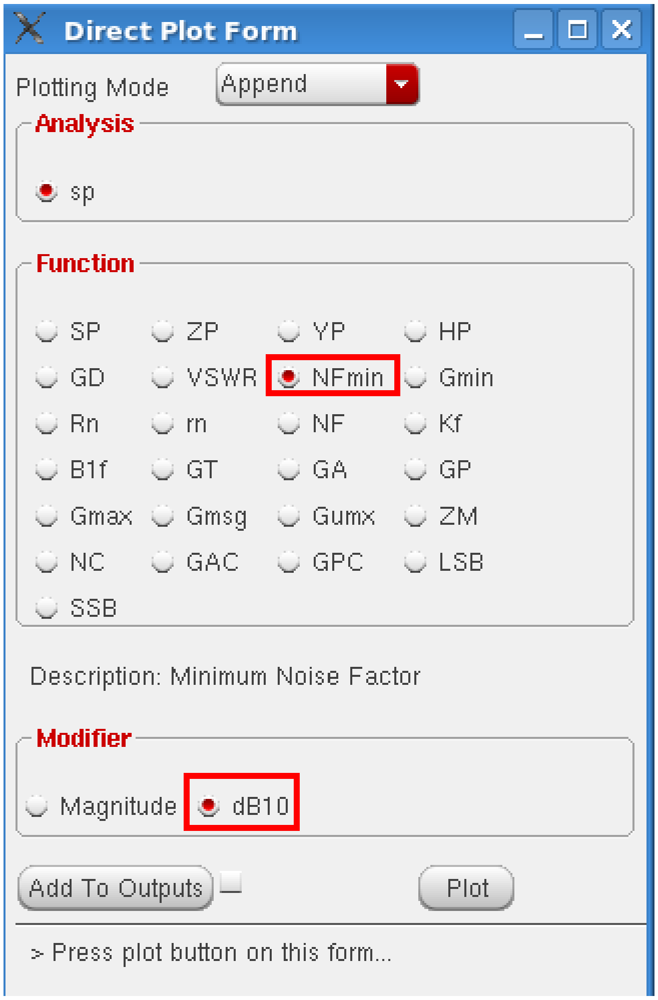
4. In the *Direct Plot Form*, change the *Plotting Mode* to *Append*.

5. In the *Function* section, select *NFmin*. You will be plotting minimum noise figure.

6. Set the *Modifier* to *dB10* to plot minimum noise figure. (Leave the modifier at the default value of magnitude to plot minimum noise factor).

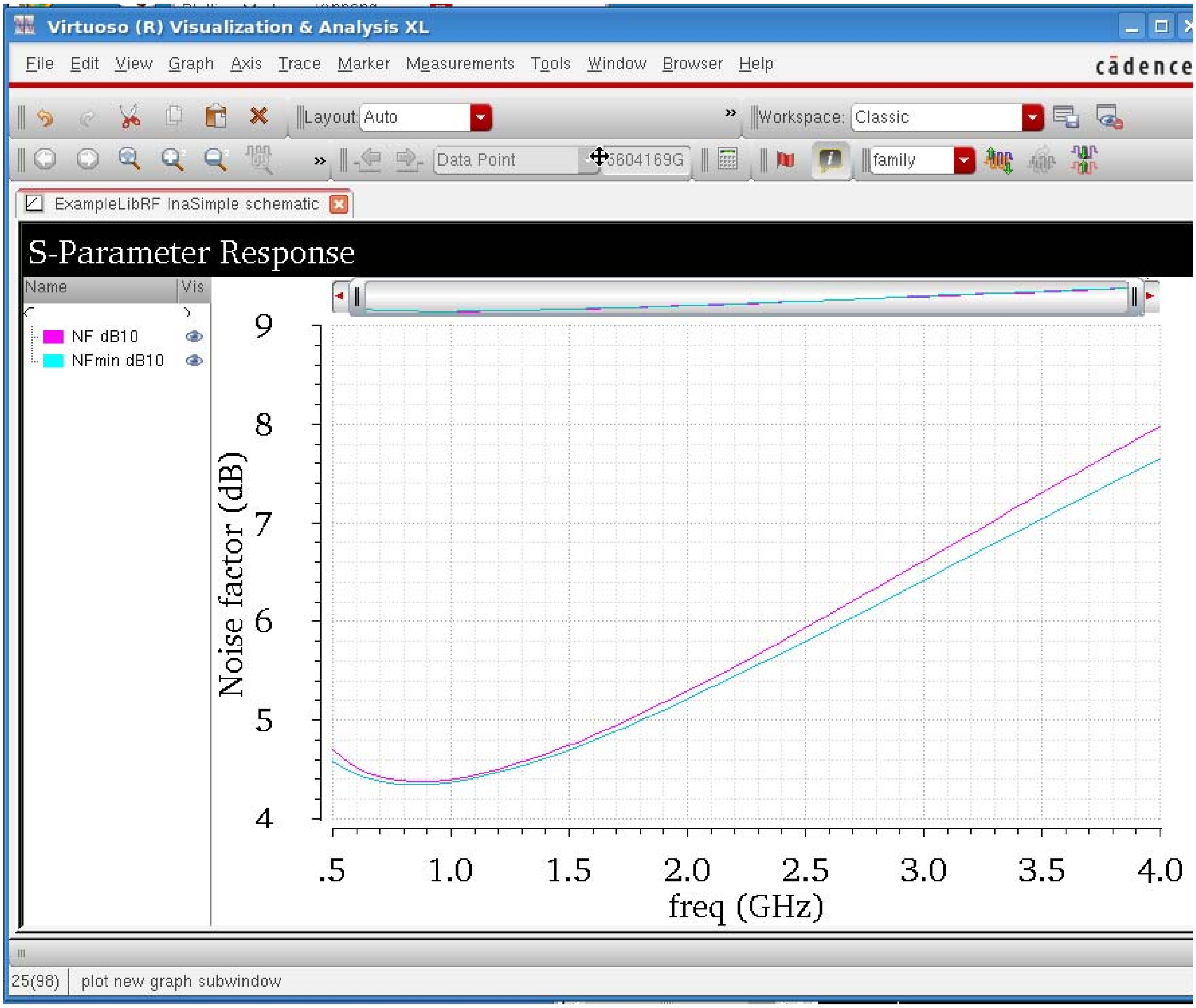
The *Direct Plot Form* should look like the following figure:

Figure A-48 sp Direct Plot Form for Minimum Noise Figure



7. Click *Plot* to plot minimum noise figure. The noise figure is plotted, as shown below.

Figure A-49 lnaSimple Minimum Noise Figure



Note: The y-axis label for Noise Figure and Minimum Noise Figure both show “Noise Factor (dB)”. View the legend at the upper left side of the plot. You will see NF dB10 and NFmin dB10. This shows that noise figure (rather than noise factor) is being plotted.

The Noise Figure plots are at a minimum at the frequency of operation.

Next you will plot Noise Circles.

8. In the *Direct Plot Form*, change the *Plotting Mode* to *New SubWindow.*

9. In the *Function* section, select *NC*. The form changes.

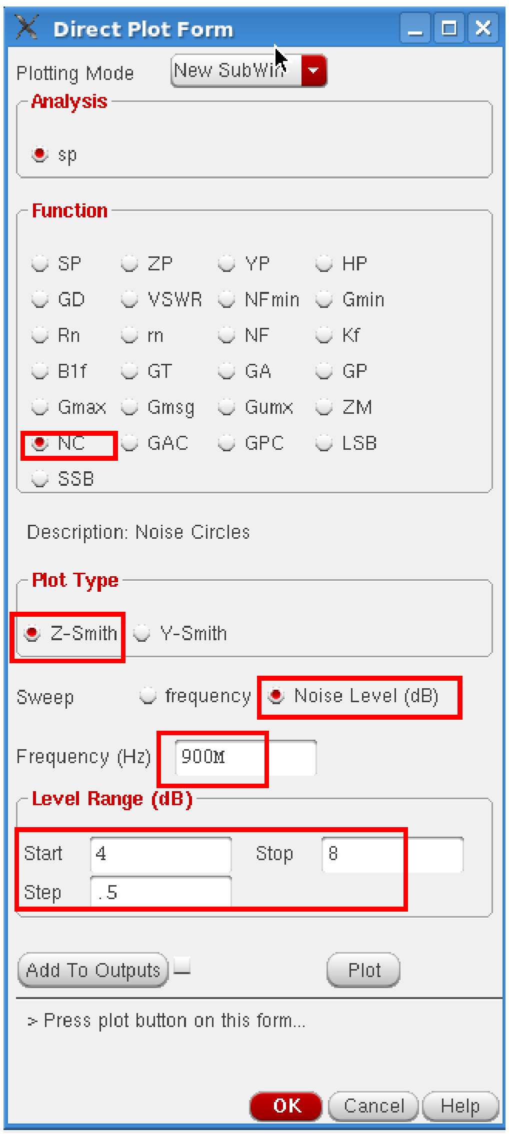
10. In the *Plot Type* section, select *Z-Smith*. This plots the noise circles on the Impedance Smith Chart (Choosing Y-Smith plots the noise circles on the Admittance Smith Chart.)

11. In the *Sweep* section, select *Noise Level (dB)*. You will be plotting circles of constant noise level at a single frequency.

12. In the *Frequency(Hz)* field, type 900M. This is the operating frequency of the lnaSimple circuit.

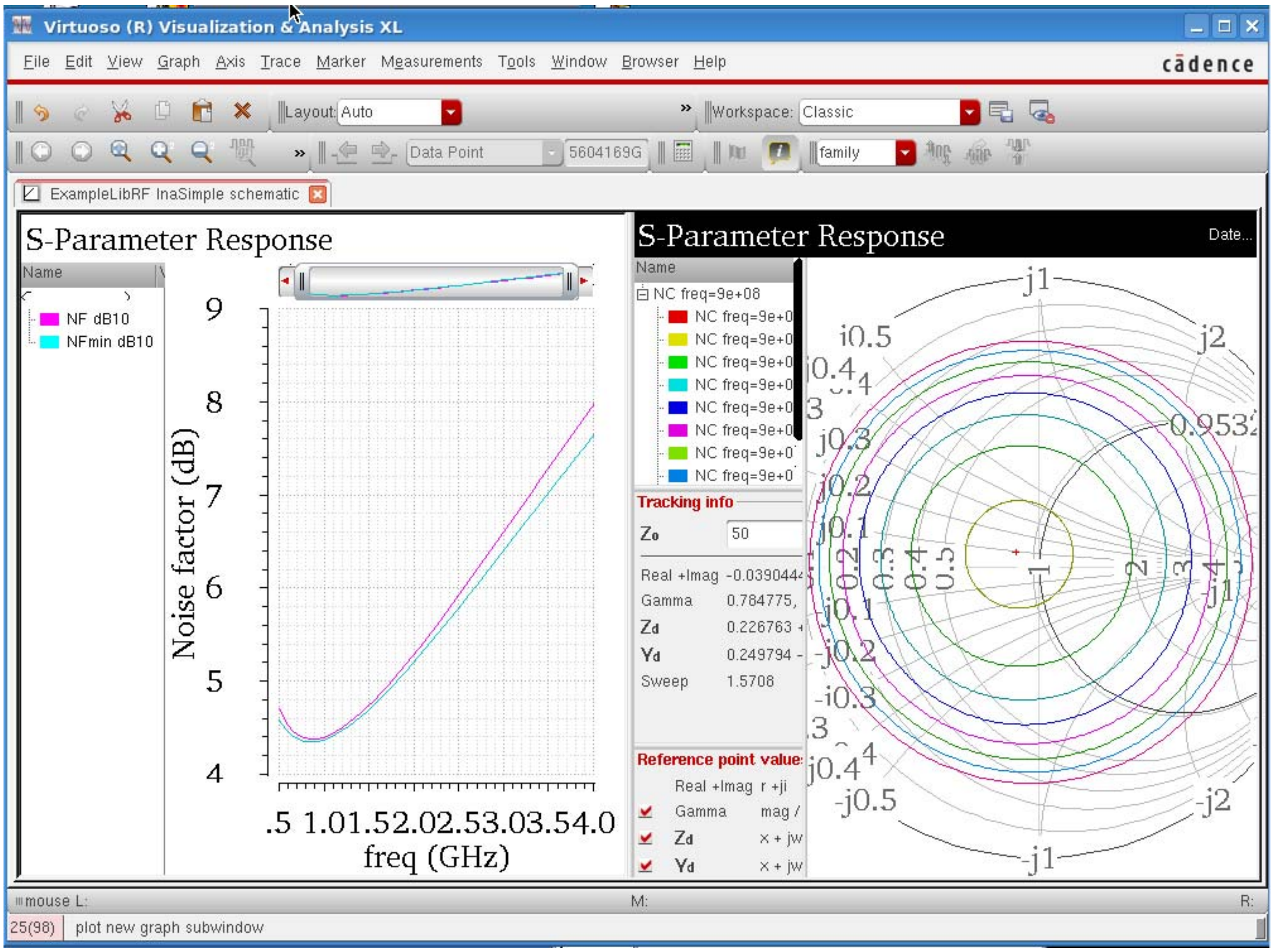
13. In the Level Range (dB) section, set *Start* to 4, *Stop* to 8, and *Step* to 0.5. The *Direct Plot Form* should look like the figure below.

Figure A-50 Direct Plot Form for Noise Circles



14. Click *Plot* at the bottom of the form. The Noise Circle plot is displayed, as shown below.

Figure A-51 Noise Circle Plot



The optimum location for the center of the noise circle is at the center of the Smith chart. However, it is hard to center both the available gain circle, GAC, and the noise circle, NC, in the Smith chart. When you design an LNA, plot NC, GAC, and the source stability circle, SSB, together in the same plot. Use this plot to trade-off the gain, noise, and stability for the input matching network design.

15. Clean up the screen for the next set of measurements.

   a. Close the *Results Display* window by choosing *Window - Close*.

   b. Close the *Analog Design Environment* window by choosing *Session - Quit.*

   c. Click *No* in the *Save State* Window

d. In the Schematic window, choose *File - Close.*

Summary

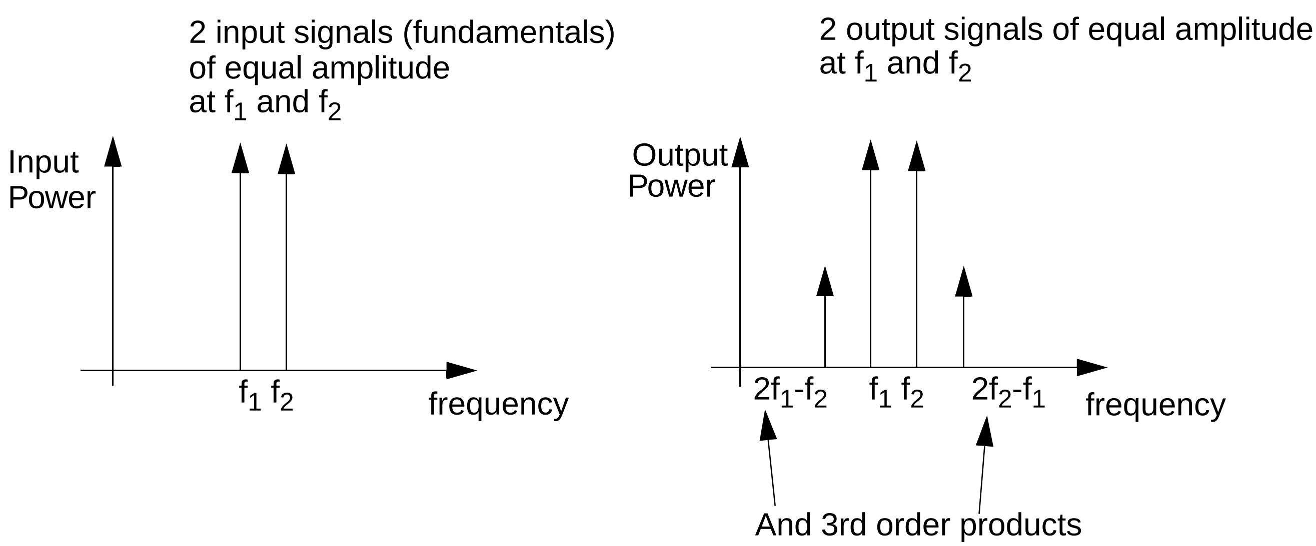
In this section, you have looked at common measurements made on Low Noise Amplifier circuits, specifically Gain, Stability, and Noise. For other examples of Measurements, see Chapter 3, "Frequency Domain Analyses: Harmonic Balance," and the LNA Workshop located in the MMSIM12.1.1 hierarchy.

In the next section, you will measure the 1dB Compression Point and IP3 using two different methods.

Third-Order Intercept measurement with HB (2 tone HB)

In narrowband circuits, distortion is commonly measured by applying two pure sinusoids with frequencies well within the bandwidth of the circuit (call these frequencies f1 and f1). The harmonics of these two frequencies would be outside the bandwidth of the circuit, however, there are distortion products that fall at the frequencies $2f_1$-f_2, $2f_2$-f_1, and so on. These frequencies are within the bandwidth of the circuit and can be used to measure the intermodulation distortion, or IMD, produced by the circuit.

IP3 is an important RF specification. The IP3 measurement is defined as the cross point of the power for the 1st order tones, f_1 and f_2, and the power for the 3rd order tones, $2f_1–f_2$ and $2f_2–f_1$, on the load.

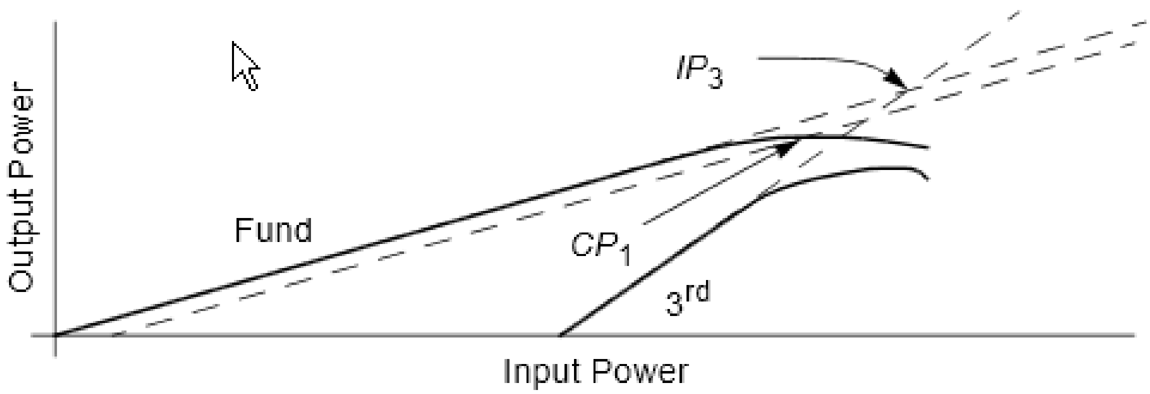


Assuming the input amplitudes are equal, the output first order terms will have the same amplitude and the output third order terms will also have the same amplitude.

As the first order components grow linearly and the third order components grow cubically, they eventually intersect as the input power level increases.

The third order intercept point is where the two output power curves intersect, as shown in the figure below.

Figure A-52 1dB Compression Point and IP3 Curves



CP_1 is the 1dB compression point
IP_3 is the third order intercept

SpectreRF provides several different ways of simulating IP3 for a low noise amplifier. You will measure IIP3 (input IP3) using two different ways.

The first way is to use hb analysis with two RF tones applied. Generally, the RF power is swept over a range in the small-signal region. Hb is a large-signal simulation that calculates all the harmonics. Because the third-order product is quite small at low-input power, the accuracy of the simulation result must be kept quite high. This usually requires lengthy simulations.

The fastest approach is to select Rapid IP3 from the AC analysis. This is the fastest approach but it is limited to small-signal (at least 10dB below the 1dB compression point) IP3 measurements. In a small-signal application, both techniques produce answers typically within 0.1dB of each other.

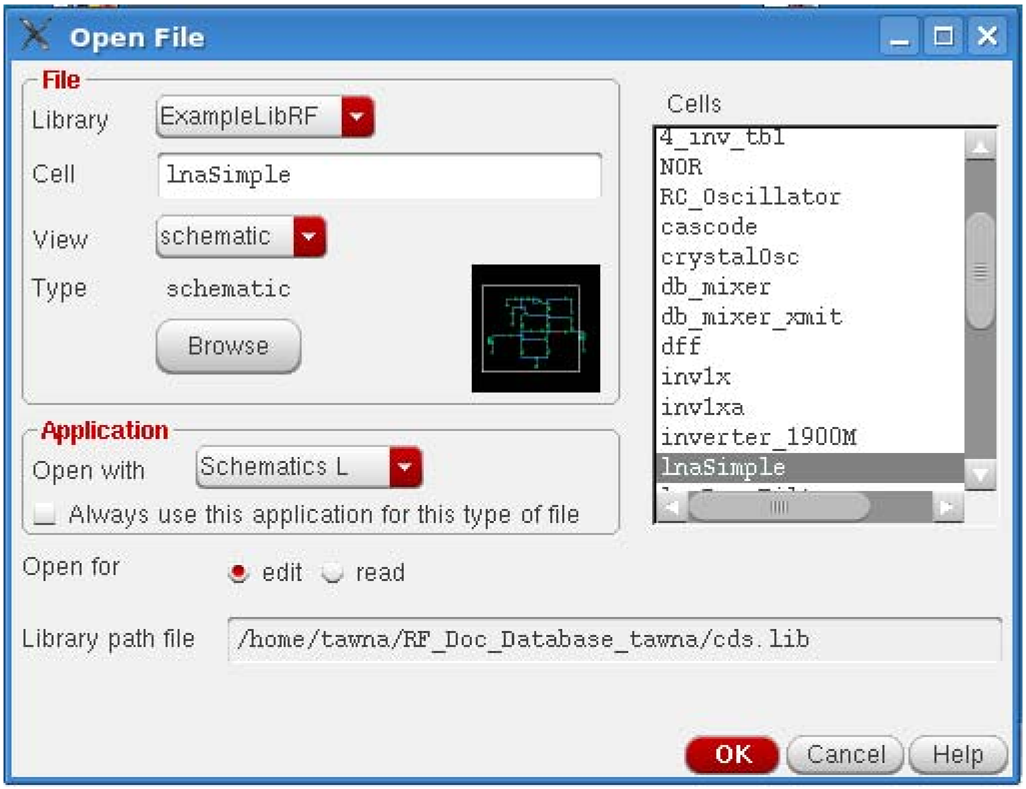
Opening the lnaSimple LNACircuit in the Schematic Window

1. In the CIW, choose *File – Open.*

   The *Open File* form is displayed.

2. Choose *ExampleLibRF* from the *Library* drop-down list.
3. Choose *lnaSimple* from the *Cell* list box. Leave the rest of the fields at their default values.
4. The completed *Open File* form appears like the one below.

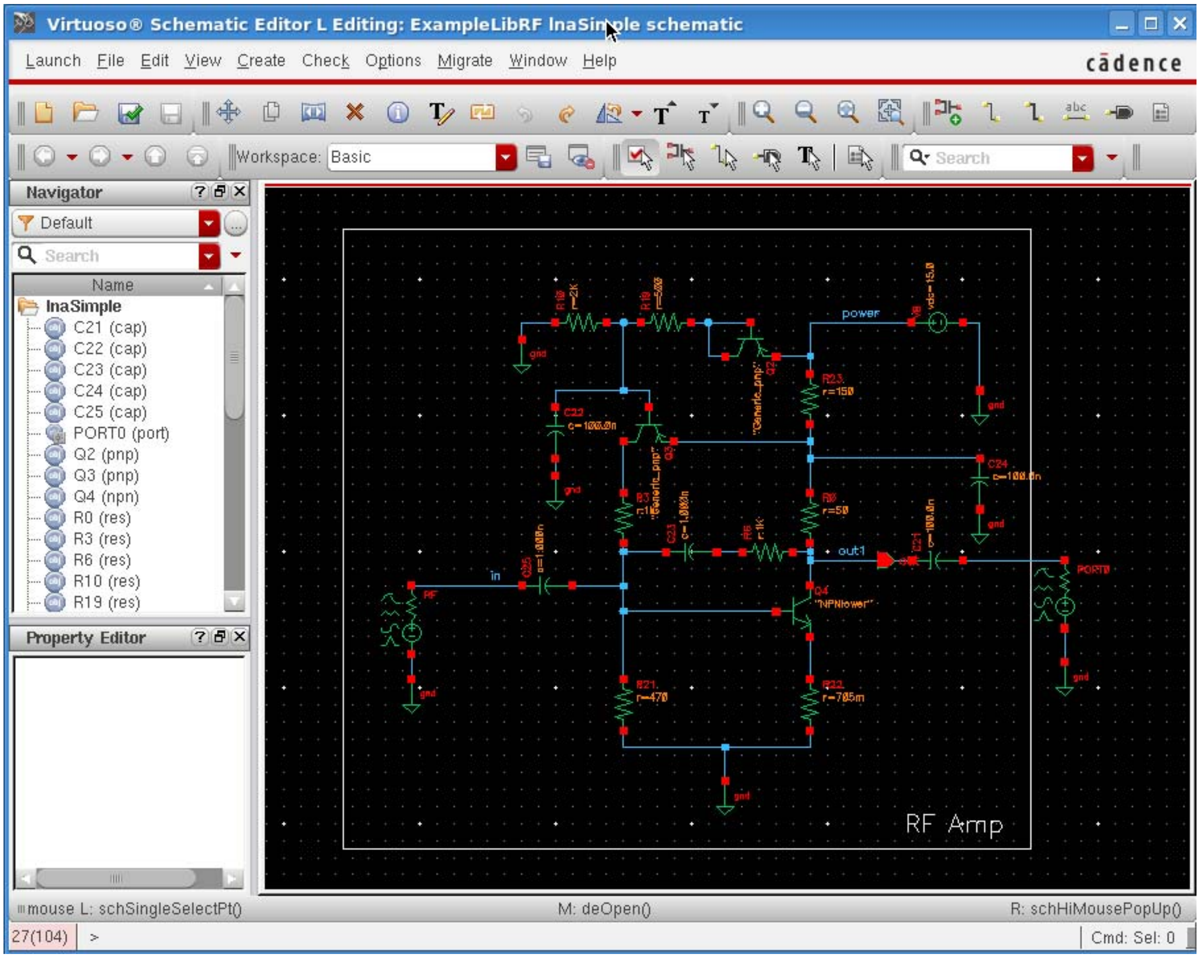
Figure A-53 Open File Form



5. Click *OK*.

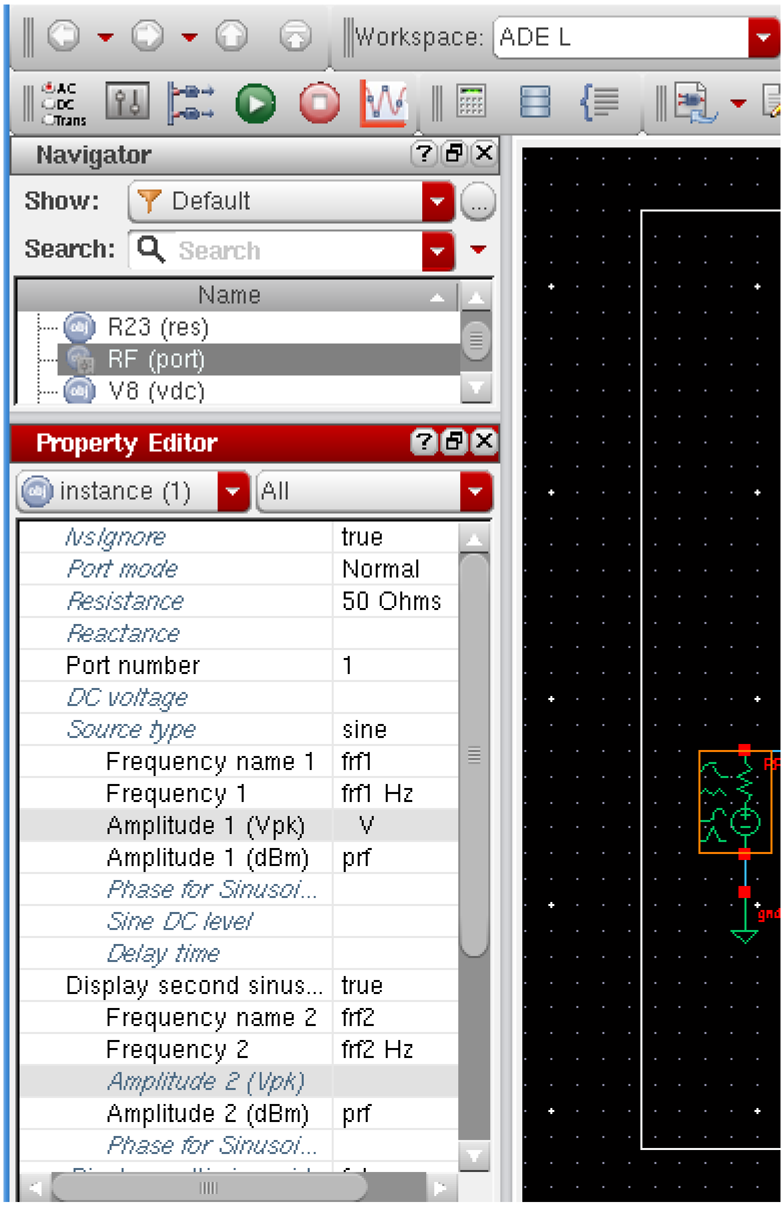
The Schematic window for the db_mixer mixer is displayed.

Figure A-54 lnaSimple Schematic



6. In the Schematic, click on the RF port on the left side of the schematic.

Figure A-55 RF Port Selected



Note that when you do this, the RF port properties are populated into the *Property Editor* on the left side of the schematic. Examine the input port settings.

a. The *Source type sine* means generate large signals for the time domain (transient, pss-shooting, qpss-shooting, envlp-shooting, and so on) or frequency domain (HB or envlp-HB) analyses.

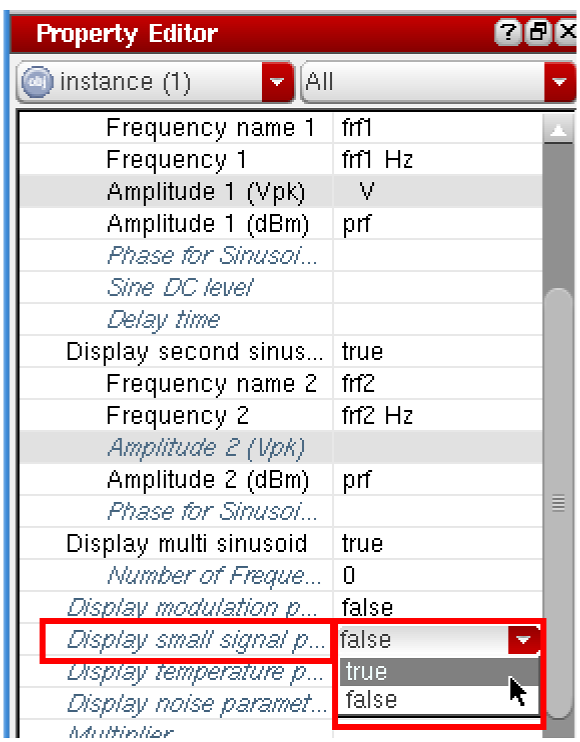
The port can generate one or two large signals. In this example, it is generating two signals. You need to specify a name for all of the large signals. This is done in the

Frequency name 1 and *Frequency name 2* property fields. The frequencies are set to a variable name. The value of the variables is set in the ADE window. This allows sweeps to be done in the Analog Design Environment. The amplitudes can be set in volts peak or in dBm. In this case, the amplitudes are set to a variable named *prf*. When the amplitude is set in dBm, the amplitude in volts is not selectable.

The Display second sinusoid option is a display function only. If there is an entry there and the display option is off, the second waveform will still be generated. To switch to a single input, remove all the entries, or set the amplitude or frequency to zero in the *Amplitude* field orthe *Design Variables* Section of the ADE L window.

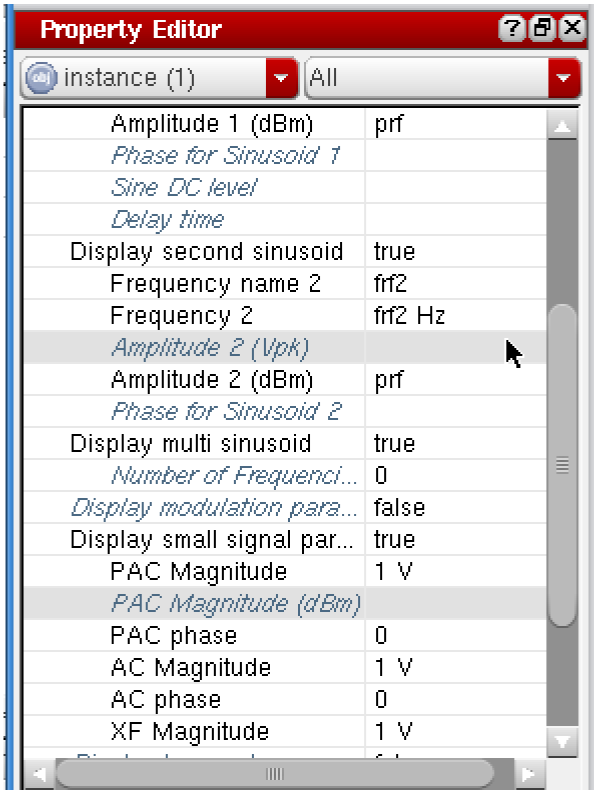
The amplitude for the small-signal analyses AC and PAC are set in the source. To view the small signal values, scroll down the form to the *Display small signal parameter* drop-down list and select *true* from the drop-down list. The amplitude for AC and PAC can be either in Volts peak or dBm, but not both at the same time.

Figure A-56 RF Port Display small signal parameters.



The form expands, as shown below.

Figure A-57 Expanded RF Port Display Small Signal Parameters

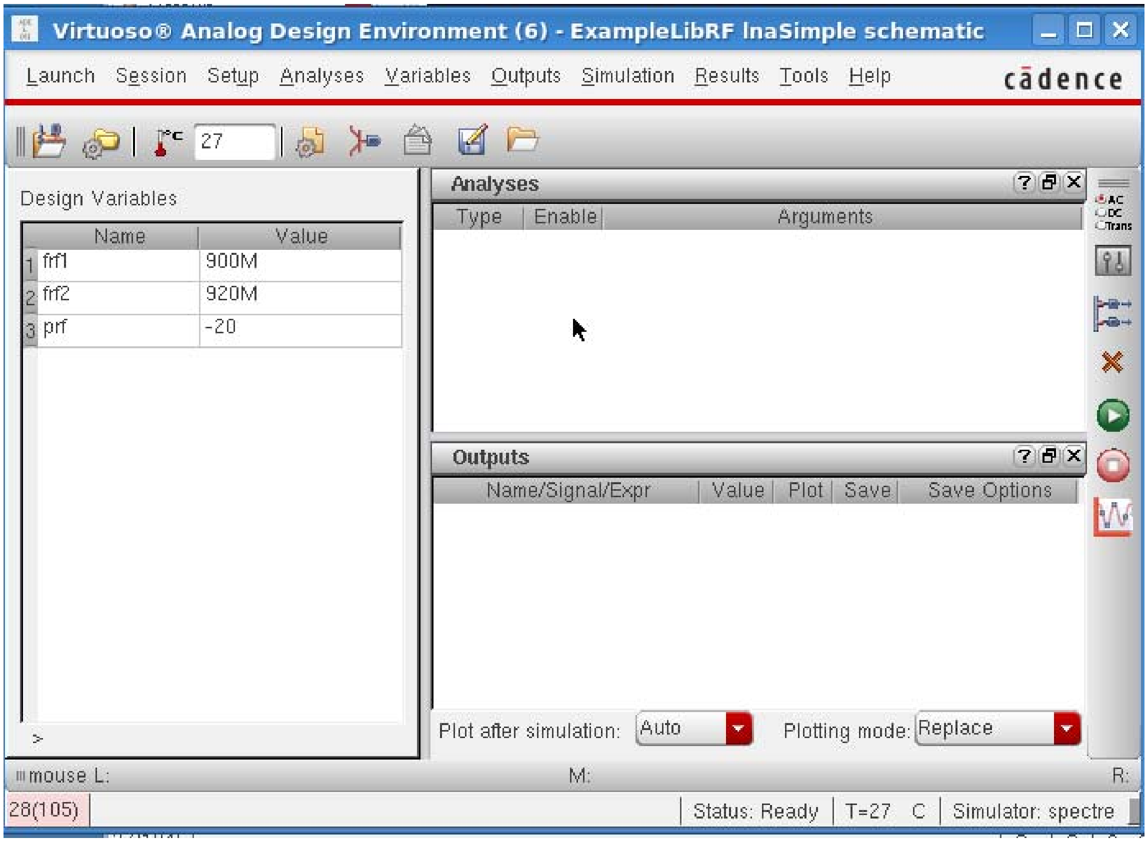


The frequency of these signals will be set in the small-signal setup form in ADE. The amplitude for PAC can be either in volts peak or in dBm, again with the form not allowing both to be set at the same time.

7. In the Schematic window, choose *Launch – ADE L*.

The Virtuoso Analog Design Environment window opens.

Figure A-58 Analog Design Environment Window



Choosing Simulator Options

1. Choose *Setup – Simulator/Directory/Host* in the Virtuoso Analog Design Environment window.

   The *Choosing Simulator/Directory/Host* form is displayed.

2. Specify the following:

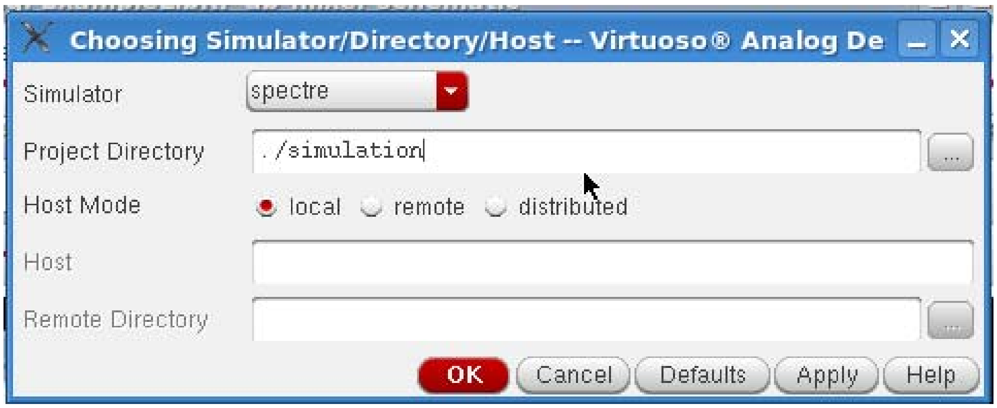
   a. Choose *spectre* from the *Simulator* drop-down list.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is the *~/simulation* directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, please contact your Cadence tools System Administrator for specific set-up instructions.

   The completed form similar to the one below is displayed.

Figure A-59 Choosing Simulator/Directory/Host Form

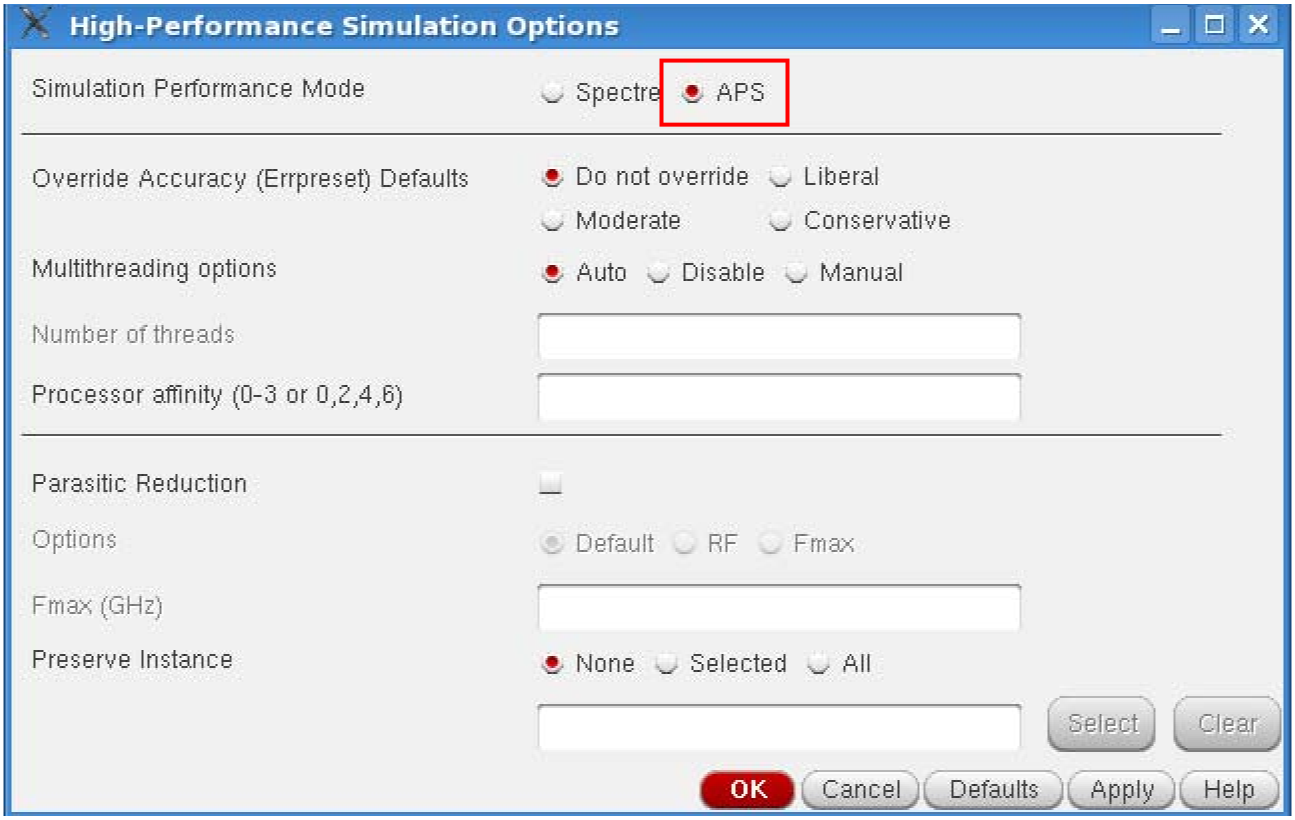


3. Click *OK*.

4. Set up the High Performance Simulation Options

   In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

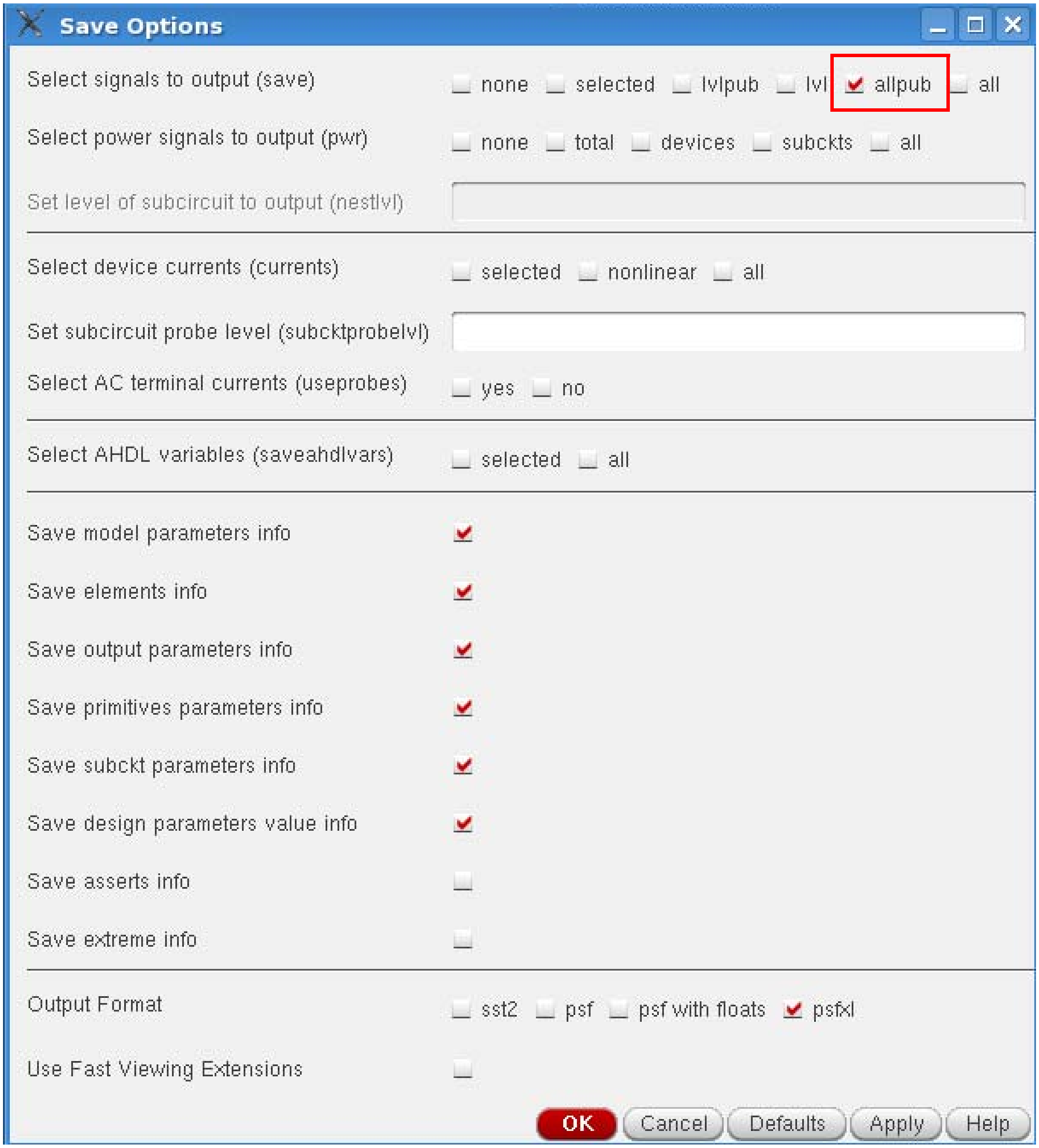
Figure A-60 High Performance Simulation Options



5. In the *High Performance Simulation Options* window, select *APS* as the *Simulation Perfomance Mode.* Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores.
6. Click *OK*.
7. In the Virtuoso Analog Design Environment window, choose *Outputs – Save All*.

   The *Save Options* form is displayed.
8. In the *Select signals to output* section, ensure that *allpub* is selected.

Figure A-61 Save Options Form



This is the default selection. This saves all the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

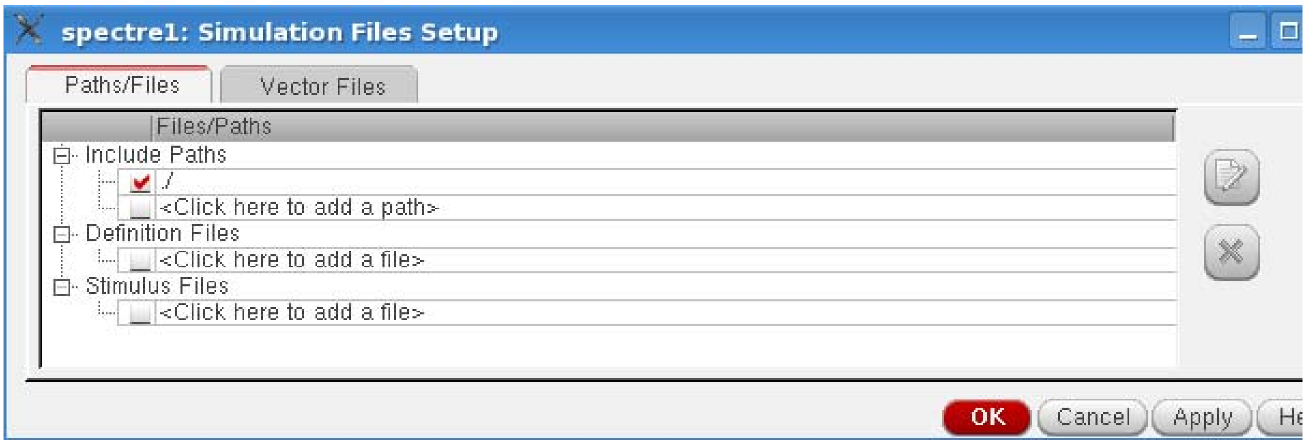
9. Click *OK*.

Setting Up Model Libraries

In the Virtuoso Analog Design Environment window, choose *Setup - Simulation Files*.

The *Simulation Files Setup* form opens.

Figure A-62 Simulation Files Setup Form



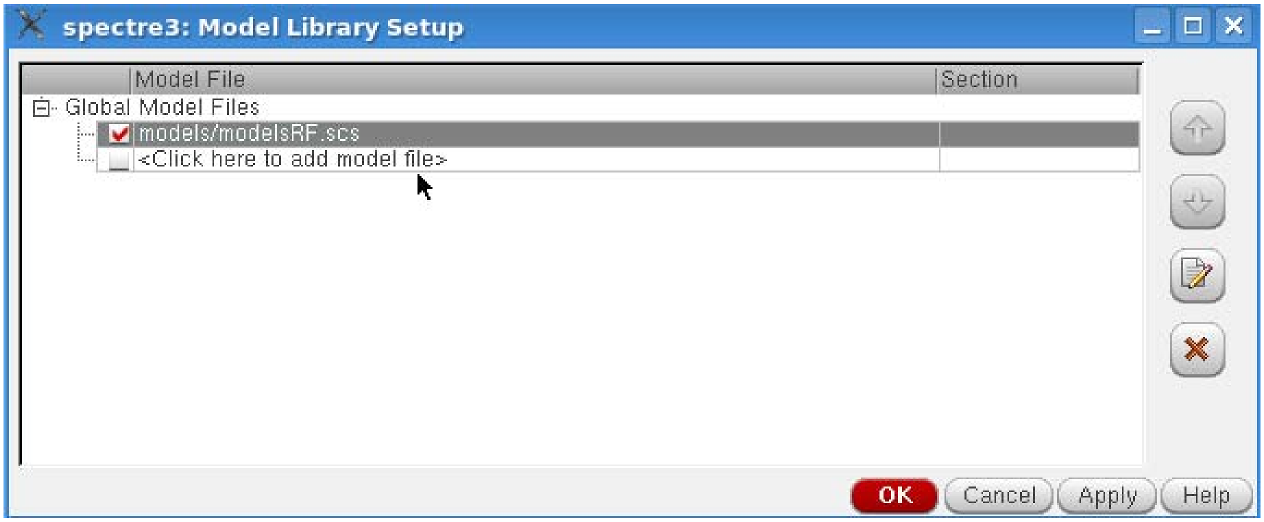
1. Verify that the *Include Path* is set, as shown above.
2. In the Virtuoso Analog Design Environment window, choose *Setup – Model Libraries.*

   The *Model Library Setup* form is displayed.
3. In the *Model Library File* field, type in the name of the model file, as shown below.

   `models/modelsRF.scs`
4. Click *Add*.

   The *Model Library Setup* form looks like the following:

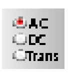
Figure A-63 Model Library Setup



5. Click *OK*.

   Alternately, you can click on the *Browse* button and browse to the `modelsRF.scs` model file.

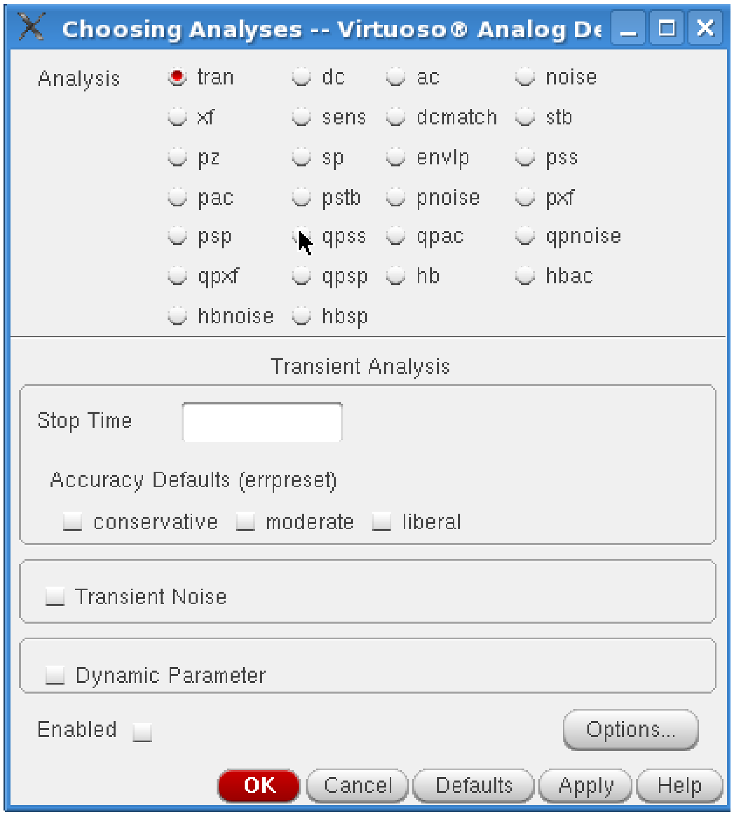
6. In the ADE L window, select *Analyses - Choose*. Alternately, you can click the *Choose Analyses* icon, as shown below.

   The *Choosing Analyses* form is displayed.

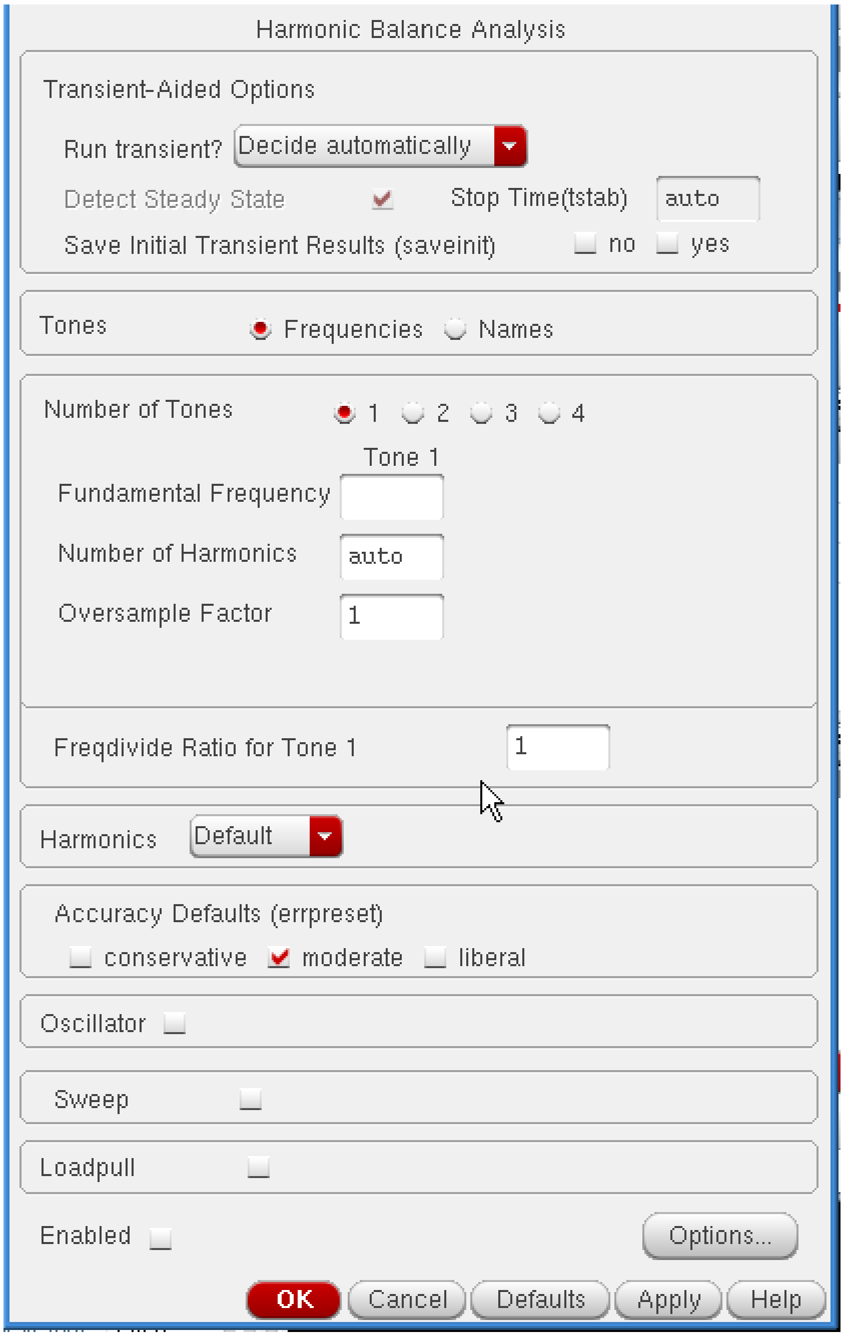
Figure A-64 Choosing Analyses Form

.



7. In the Analyses section, select *hb*. The form expands, as shown below.

Figure A-65 hb Choosing Analyses Form



Harmonic balance can set the harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* from the *Run Transient?* drop-down list in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected, and then from the transient analysis, the number of harmonics for

Tone1 (when *Frequencies* is selected) or for the tone that has tstab enabled (when *Names* is selected).

8. In the Transient-Aided Options section of the form, select the following

   a. For *Run transient?* select *Decide automatically* . (this is the default)

      Run transient? will run the large signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

   b. For *Stop time (tstab), auto* is automatically populated in the field

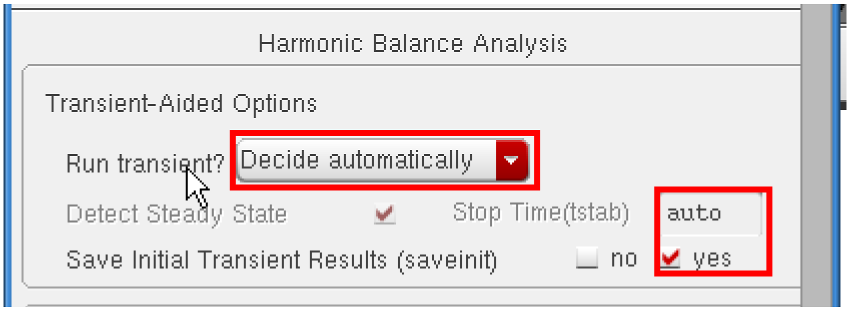
      When *auto* is selected for *Stop Time*, a small number of periods of the large signal is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

      When *Run transient?* is set to *Decide automatically*, the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

   c. For *Save Initial Transient Results (saveinit),* select `yes`.

      During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. The large signal in *Tone1* is enabled for this measurement. At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations

      All the signals are applied and the simulation is done in the frequency domain.Only the signal and its harmonics are calculated.Transient Assisted Harmonic Balance.



9. In the *Tones* section, select *Names*. When *Names* is selected, the *Tones* portion of the form expands. All the sources in the top-level schematic are read into the form automatically.

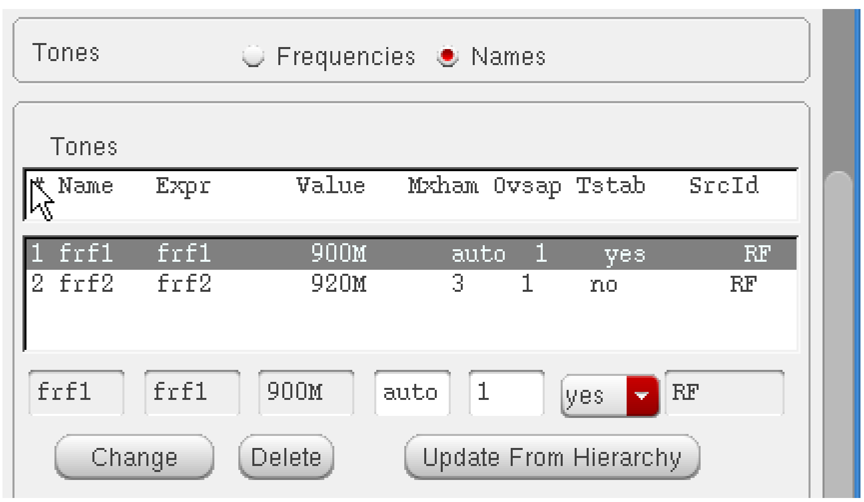
Note that there are two large tones. The frequency names for all large signal tones are automatically populated from the schematic. You viewed the tones earlier when viewing the Property Editor for the RF port.

The frequency values for the frf1 and frf2 tones are set in the *Design Variables* section of the ADE-L window, which will be shown later.

10. Select the *frf1* tone in the *Tones* field.

11. Set the *Mxham* value to *auto* and click *Change*. The form updates. Spectre will choose the appropriate number of harmonics for you.

Figure A-66 Tones Section of hb Choosing Analyses Form.



12. When you choose *auto* harms, leave *Oversample Factor* set to the default value of 1.

13. Leave *Tstab* set to the default value of *yes*.

Because you are using auto-tstab, you do not need to set *Tstab* to `yes` in the *Tones* section for one of the large signal tones. The signal with `tstab=yes` is the signal that is used for transient-assisted harmonic balance. Only one signal can have transient assist, that being the signal with *Tstab* set to `yes`.

Note: If for some reason you are not using auto-tstab, set *tstab* to `yes` on the signal that causes the largest amount of distortion in the system.

During the tstab interval, a transient analysis is run before the frequency domain iteration of harmonic balance. At the end of the tstab, an FFT is run and its result is used as the starting point for the frequency domain iterations. All the signals are applied and the simulation is performed in the frequency domain.

14. Typically, you want to use the auto-harmonic feature (enter *auto* into this field). Leave *Ovsap* (oversample) set to the default value of 1.

15. Leave the *Harmonics* field set to *Default*.

16. In the *Accuracy Defaults* section, verify that *moderate* is selected. For most normal measurements *errpreset* should be set to *moderate*. When you need to measure really small distortions, use *conservative*.

17. To set up a sweep analysis, select the *Sweep* check box and setthe value for *Sweep* to 1 (this is the default value).

18. For *Frequency Variable?* select *no*. You will be sweeping input power rather than frequency.

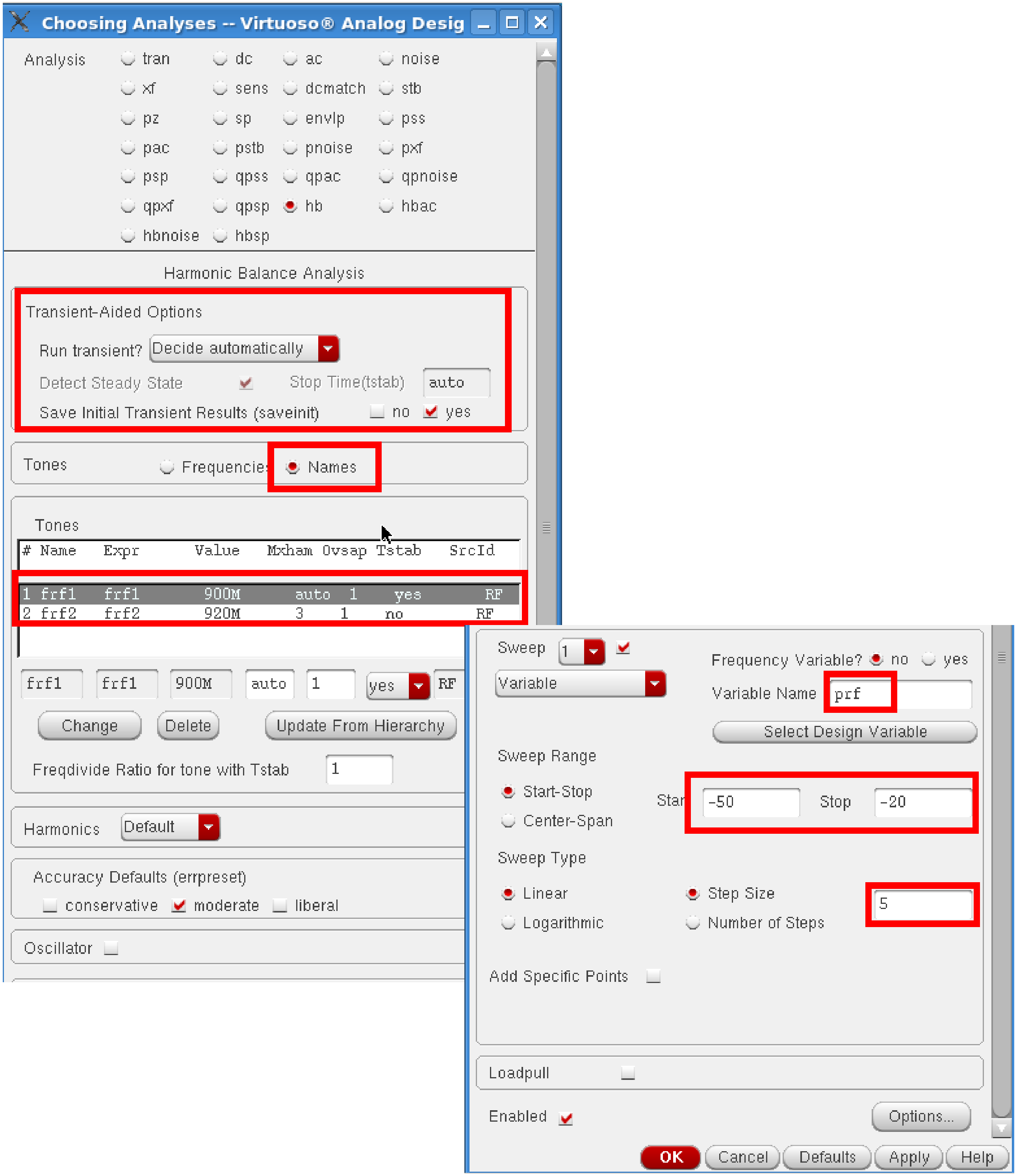
19. Type *prf* in the *Variable Name* field.

20. In the *Sweep Range*, type -50 in the *Start* field and -20 in the *Stop* field. Typically, you want to choose an input power that is at between 20-40 dB below the 1dB CP.

21. Select *Linear*from the *Sweep Type section* and enter 5 in the *Step Size* field.

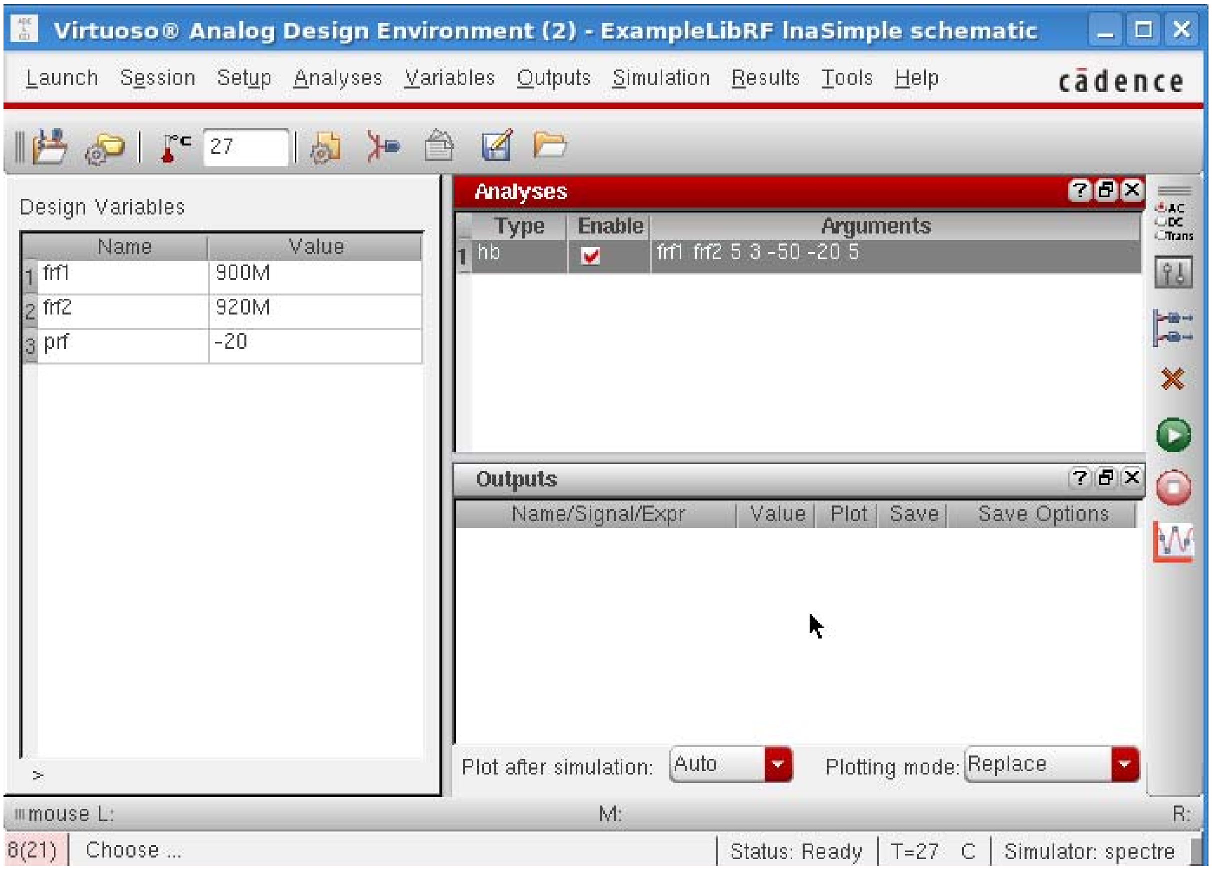
22. Leave the rest of the form set to the default values. The hb *Choosing Analyses* Form should look like the following figure:

Figure A-67 Choosing Analyses Form for Two Tone Swept HB Analysis

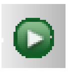


23. View the *Design Variables* section in the ADE Simulation window. Verify that your *frf1* and *frf2* values are *900M* and *920M* respectively. Your ADE window should look like the following:

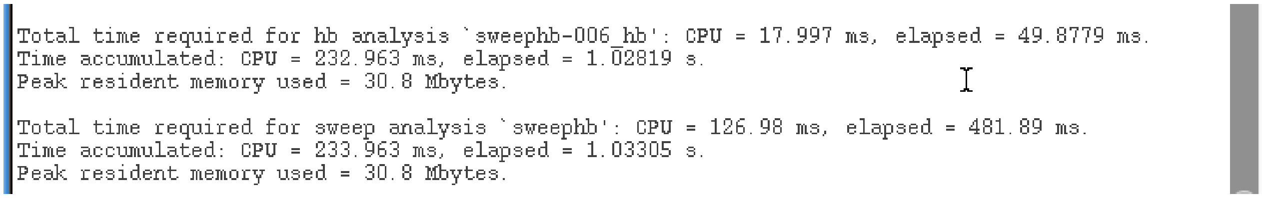
Figure A-68 Analog Design Environment Window



24. Start the simulation by choosing *Simulation - Netlist and Run* or by clicking the green arrow icon on the right side of the simulation window.



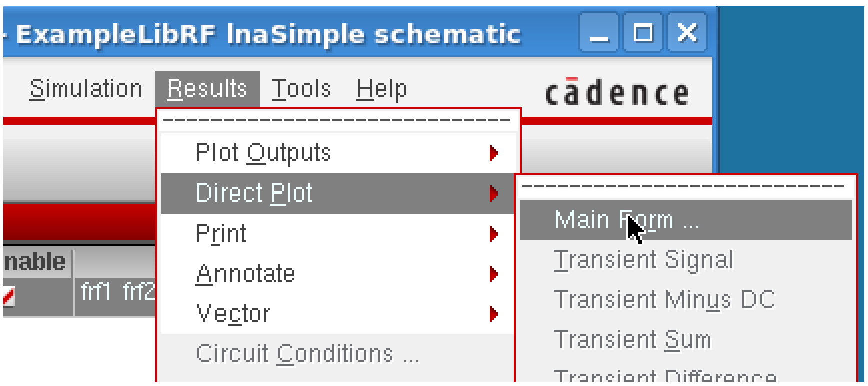
This netlists the design and runs the simulation. A SpectreRF status window appears (spectre.out logfile). Note the simulation time in the Spectre output logfile.



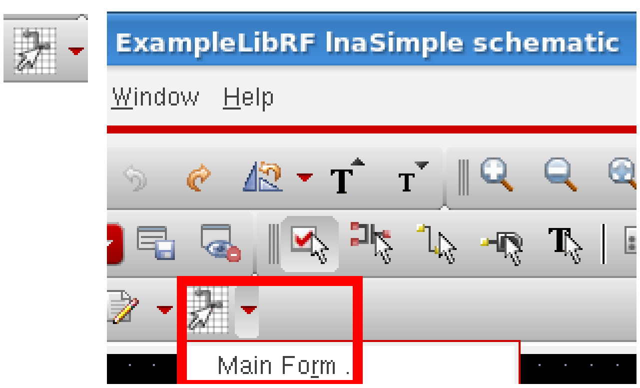
```
Total time required for hb analysis `sweephb-006_hb': CPU = 17.997 ms, elapsed = 49.8779 ms.
Time accumulated: CPU = 232.963 ms, elapsed = 1.02819 s.
Peak resident memory used = 30.8 Mbytes.

Total time required for sweep analysis `sweephb': CPU = 126.98 ms, elapsed = 481.89 ms.
Time accumulated: CPU = 233.963 ms, elapsed = 1.03305 s.
Peak resident memory used = 30.8 Mbytes.
```

25. When the analysis has completed, you may iconify the the status window.

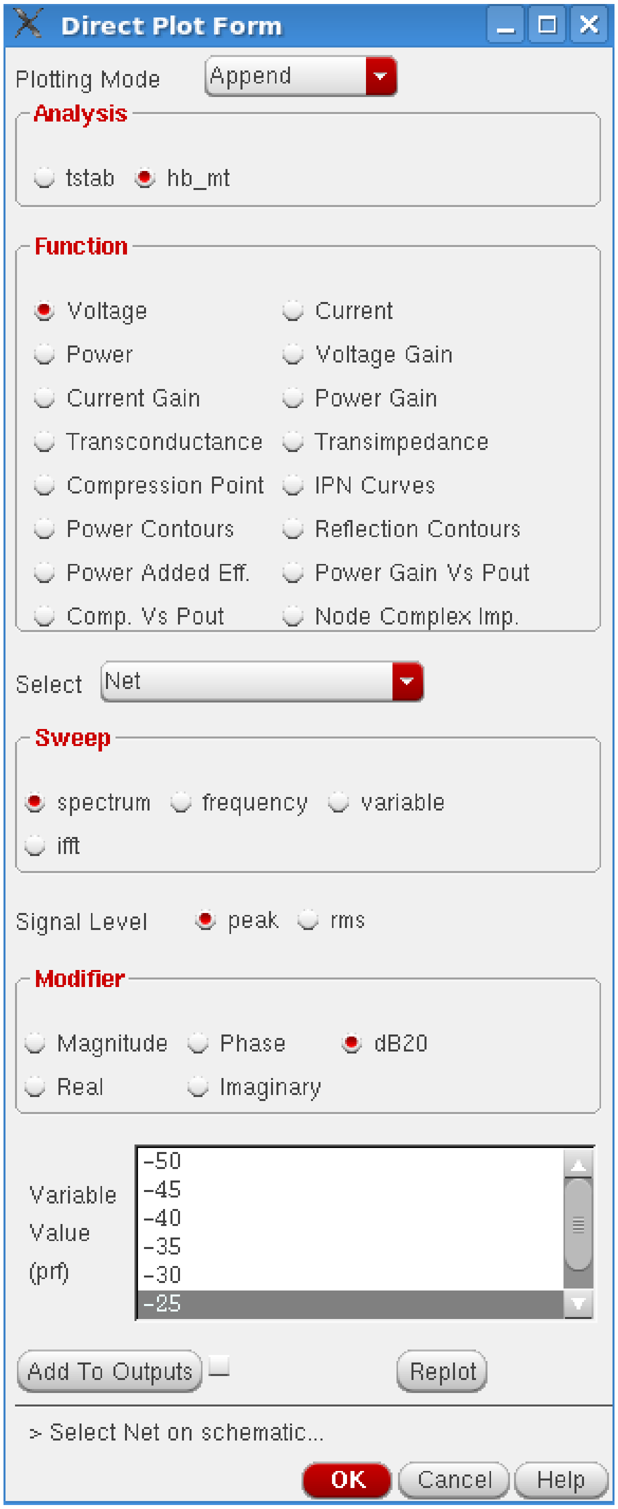
26. In the Analog Design Environment, select *Results - Direct Plot - Main Form*.



Alternately, you can press the *Direct Plot* icon from the schematic window.



27. The Direct Plot Form is displayed.



28. In the *Analysis* Section, select *hb_mt* . *hb_mt* refers to multitone harmonic balance.

29. In the *Function* Section, select *IPN Curves.*

30. Select *Net (Specify R)*. Leave the *Resistance* set to the default value of 50.

31. Select *Variable Sweep (“prf”)* for *Circuit Input Power*.

32. Leave *Input Power Extrapolation Point (dBm)* set to the default (first point in the sweep which is -50). The extrapolation point is where the ideal curves and the data are drawn through the same point. Pick an area where the third order curves have a slope of 3.

33. Choose *Input Referred IP3, Order* `3rd`

34. For the *3rd Order Harmonic*, choose `940M`.

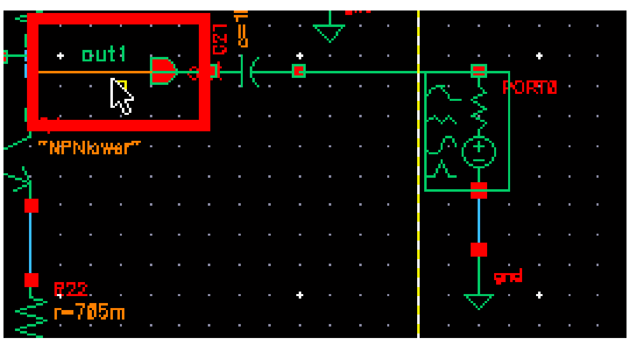
35. For the *1st Order Harmonic,* choose `900M`.

Alternately, you could have chosen 880M and 920M for the third and first order harmonics, respectively.

36. Select the *out1* node on the right side of the schematic

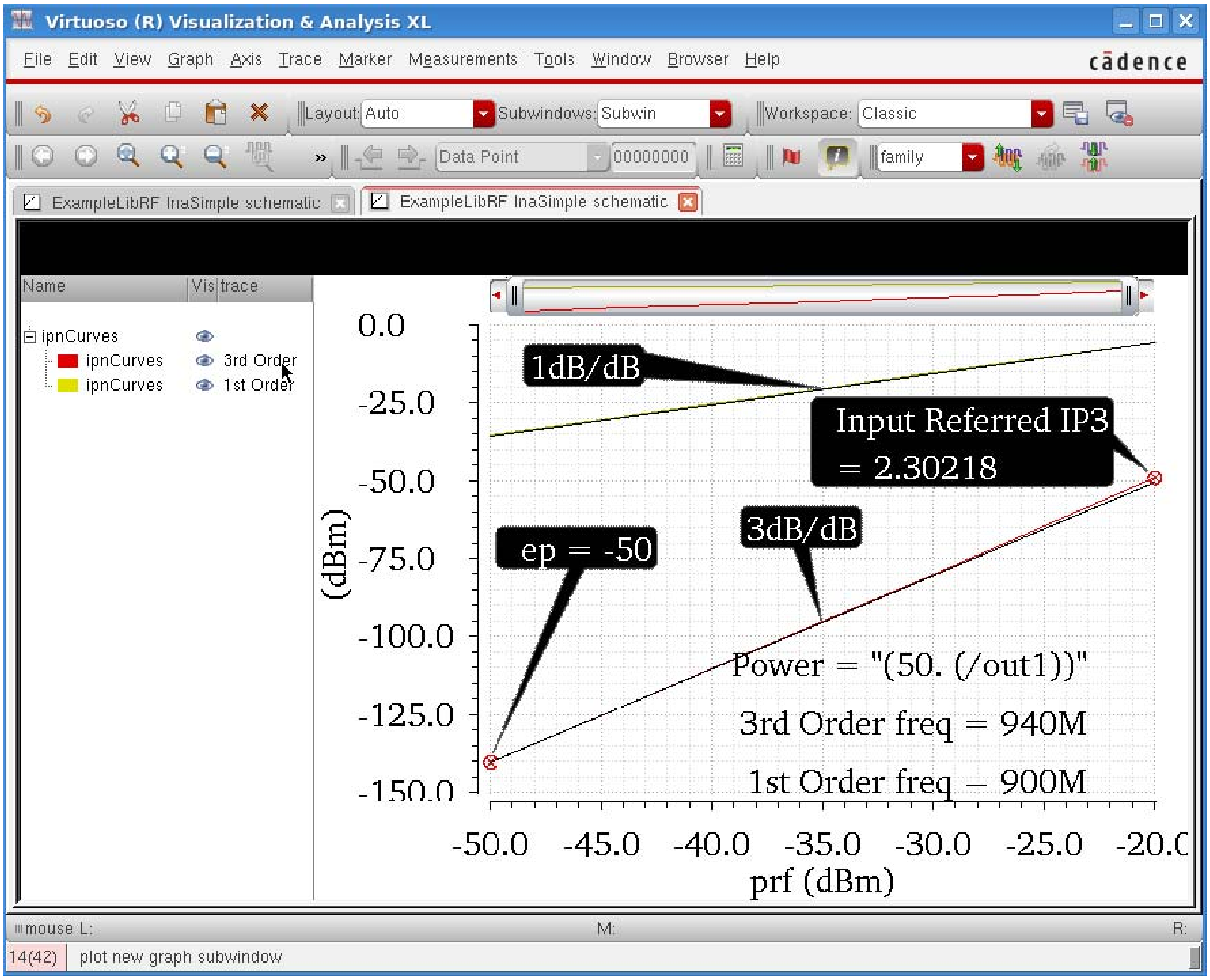
Figure A-69 Selecting the out1 Net on the lnaSimple Schematic

.



The IP3 plot is displayed. Note the IP3 readout.

Figure A-70 The IP3 Plot



Note: If you do not see the IP3 readout, click in the graphics area to deselect the marker, then select and move the *Input Referred IP3* readout so that it is positioned in the visible area of the graph.

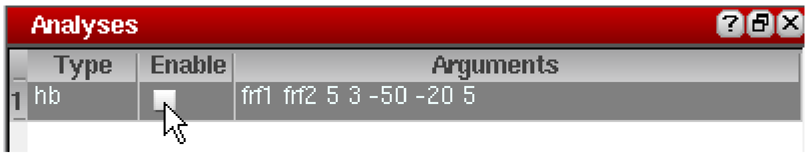
Intermodulation products increase at rates that are multiples of the fundamentals. In the small-signal region, third order terms increase 3dB per dB.

In the above plot, you can see that the circuit is operating within the small signal region. The third order curve is following a 3dB/dB slope. Note the IP3 measurement. In the next section, you will compare this to the IP3 measurement using the Rapid IP3 methodology.

37. In the Waveform tool, choose *File - Close All Windows*.

38. In the Analog Design Environment window, deselect the check box located under the *Enable* column to disable the hb analysis in the *Analyses* pane.

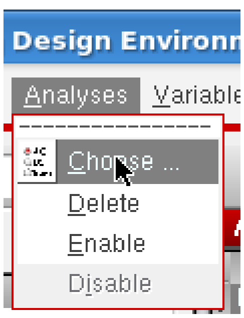
Figure A-71 ADE Simulation Analysis Pane



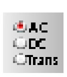
Measuring IP3 with Rapid IP3

1. In the Analog Design Environment window, select *Analyses - Choose*.

Figure A-72 Open Choosing Analyses Form

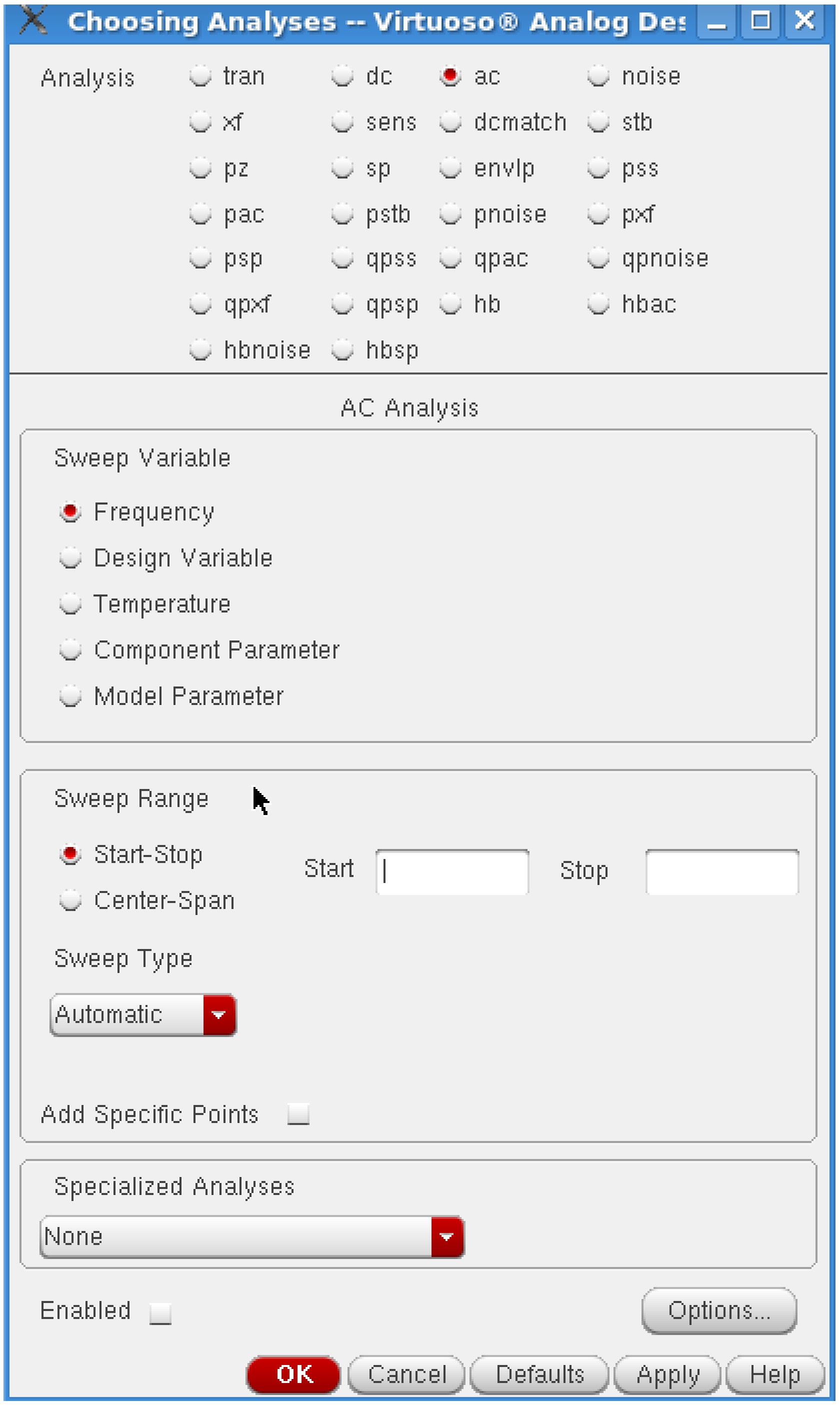


Alternately, you can click on the *Choose Analyses* icon on the right side of the ADE Window.



The *Choosing Analyses* form is displayed.

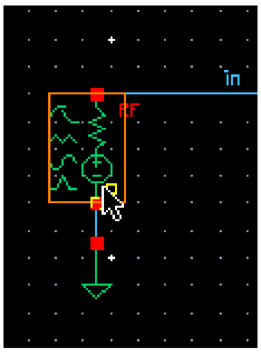
Figure A-73 Choosing Analyses Form



2. Select *ac* from the *Analysis* section.
3. At the bottom of the form, set *Specialized Analysis* to *Rapid IP3*. The form expands.

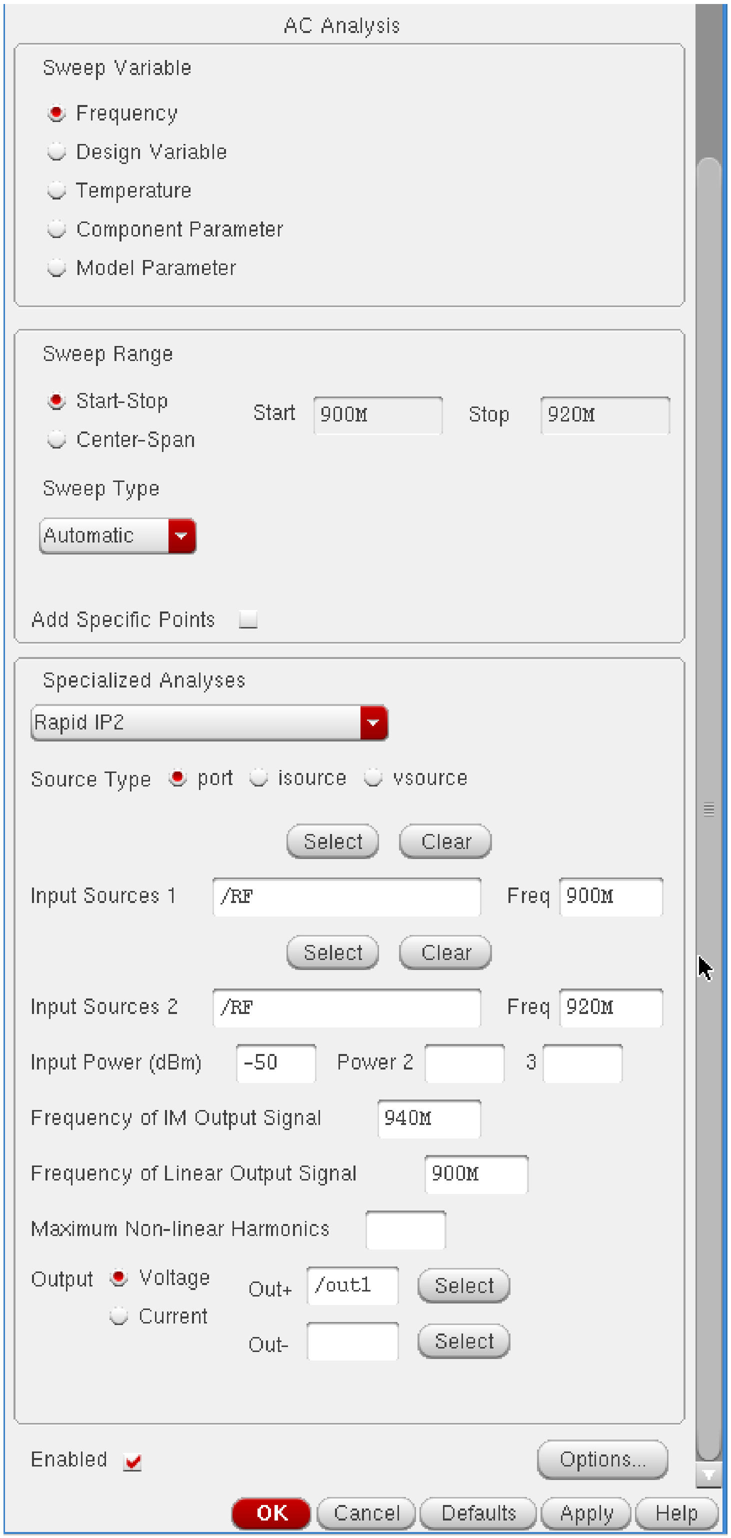
4. Select *port* for *Source Type*.
5. Select the *Input Sources 1* field.
6. Click the *Select* button just above that field.
7. Select the Port source /RF at the left of the schematic.

Figure A-74 Select Input Port



8. Press the <Esc> key. The entry for the source name will not be displayed in the *Choosing Analyses* form until the <Esc> key is pressed.
9. Type 900M in the *Freq* field to the right of *Input Sources 1*.
10. Type */RF* in the *Input Sources 2* field. (Alternately, you could select the RF port in the schematic by first clicking the *Select* button and then selecting the RF source in the schematic).
11. Type 920M in the *Freq* field the to the right of *Input Sources 2*.
12. Type -50 in the I*nput Power (dBm)* field.
13. Type 940M in the *Frequency of IM Output Signal* field.
14. Type 900M in the *Frequency of Linear Output Signal* field.
15. Type */out1* in the *Out+* field. Because you are leaving the *Out-* field blank, it will default to ground.
16. Leave the rest of the form at their default values. The *Choosing Analyses* form should look like the following:

Figure A-75 Rapid IP3 AC Analysis Form



17. Click *OK*.

18. Start the simulation by choosing *Simulation - Netlist and Run* or by clicking the green arrow icon  on the right side of the simulation window.

This netlists the design and runs the simulation. A SpectreRF status window appears (spectre.out logfile). Note the simulation time in the Spectre output logfile.

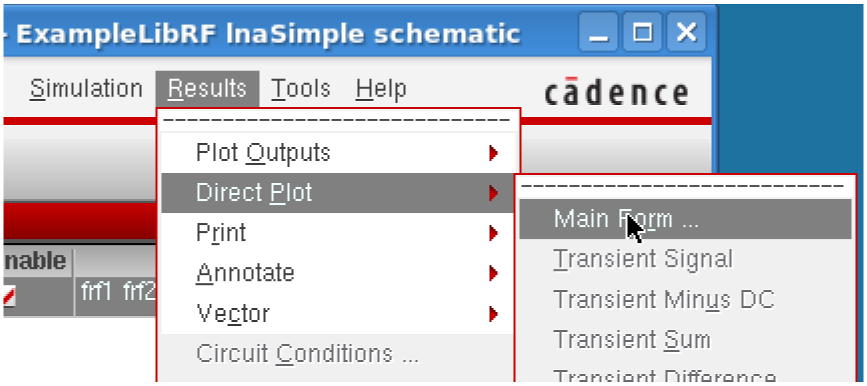
```
********************
IP3 measurement `ac'
********************
Input RF1 freq = 900 MHz
Input RF2 freq = 920 MHz
Output IM1 freq = 900 MHz
Output IM3 freq = 940 MHz

DC simulation time: CPU = 0 s, elapsed = 380.993 us.
Linear output:
f_out = f_in_1

IM3 output:
f_IM3 = 2 * f_in_2 - f_in_1
Accumulated DC solution time = 0 s.
Intrinsic ac analysis time =              20 ms.
Total time required for ac analysis `ac': CPU = 18.998 ms, elapsed = 203.026 ms.
Time accumulated: CPU = 118.981 ms, elapsed = 677.548 ms.
Peak resident memory used = 30 Mbytes.
```

Note that the simulation time is faster than using two tone hb analysis to simulate rapid IP3.

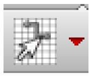
19. When the analysis has completed, you may iconify the the status window.

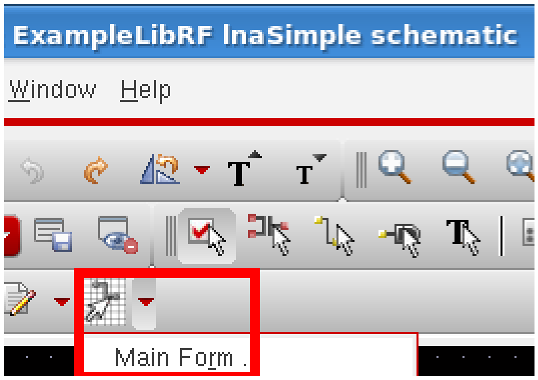
20. In the Analog Design Environment, select *Results - Direct Plot - Main Form*.

Figure A-76 Invoking Direct Plot Form



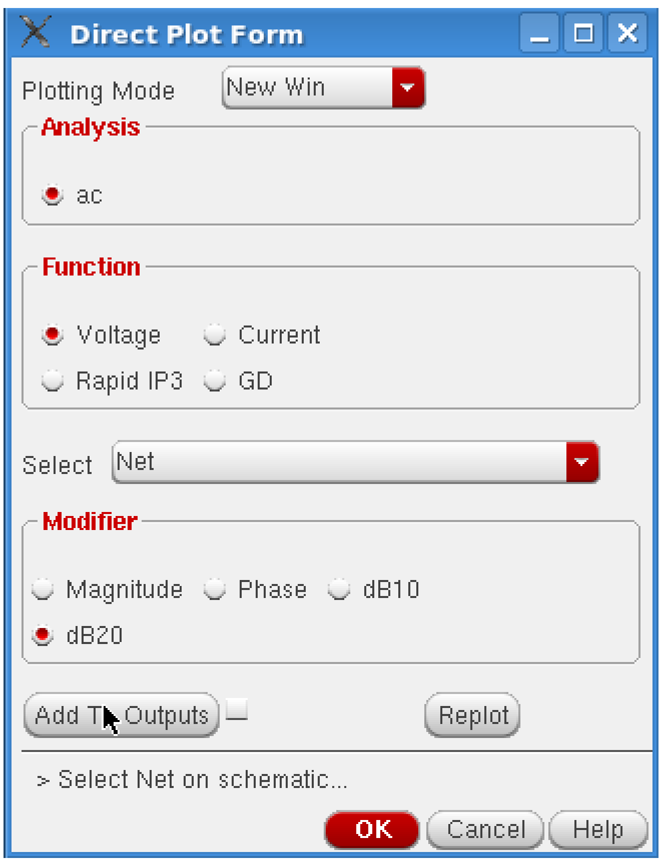
Alternately, you can press the *Direct Plot* icon from the schematic window.





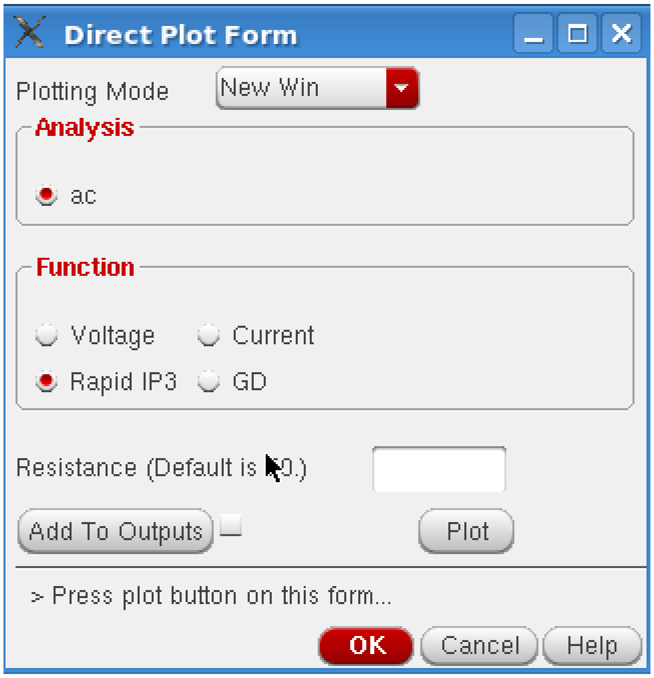
The *Direct Plot Form* is displayed.

Figure A-77 AC Direct Plot Form



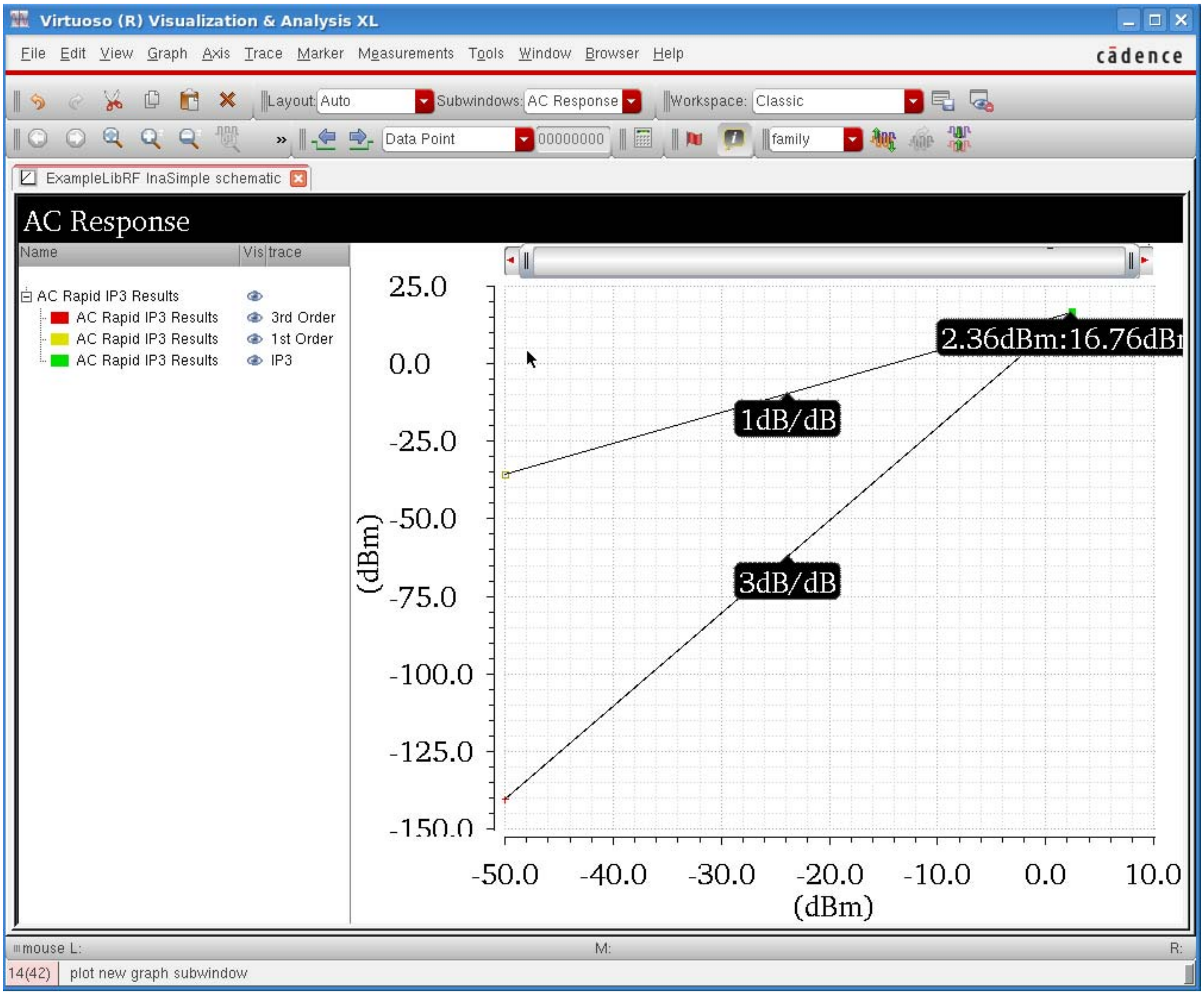
21. In the *Direct Plot Form*, select *Rapid IP3*. The form changes.

Figure A-78 AC Direct Plot Form - Rapid IP3



22. Leave the rest of the form at the default values. Click *Plot*. The IP3 plot is displayed.

Figure A-79 IP3 Plot Using Rapid IP3



Note that the value of Rapid IP3 (2.36dBm) is very close to that measured by two toned hb analysis (2.30dBm). Although simulating using both methods was relatively quick, rapid IP3 was faster.

In this section you measured IP3 on a low noise amplifier using two methods: two tone harmonic balance and rapid IP3 using AC analysis. Both gave good accuracy but rapid IP3 was faster.

Summary

The LNA section of Appendix A shows how to simulate and make typical measurements on a low noise amplifier

In the LNA section, the following measurements were shown:

- S-Parameter Analysis, Gain
- Stability, Stability Circles
- Linear 2 port Noise Figure measurements, Noise Circles
- Third Order Intercept measurement using 2 tone harmonic balance
- Rapid IP3 using specialized ac analysis

For more information on simulating low noise amplifiers, please refer to the chapters in the *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator RF Analysis User Guide* and the *Virtuoso Spectre Circuit Simulator RF Analysis Theory* guide.

Simulating Oscillators

Oscillators, which are autonomous circuits, are time-invariant circuits with time-varying responses. Thus, autonomous circuits generate non-constant waveforms even though they are not driven by a time-varying stimulus.

You cannot specify the analysis period for autonomous circuits because you do not know the precise oscillation period in advance of the simulation. Instead, you specify an estimate of the oscillation period for the simulation. The PSS analysis uses your estimate to compute the precise period and the periodic solution waveforms.

In the first section of this topic we will concentrate on simulating oscillators as autonomous circuits using mainly Harmonic Balance (HB) method. In the second section we will simulate and do measurements on Ring Oscillator using Shooting method.

Note that there are oscillators like injection locked ring oscillators which are simulated as driven circuits instead of autonomous circuits. However, this class of oscillators will not be discussed.

You will do the following measurements while doing the Oscillator simulation -

Measurements	**Analysis**
Starting and Stabilizing Feedback Oscillators	PSS - HB
Oscillator Loop Gain Measurement	PSS - Shooting, Pstb and stb
Phase Noise Measurement and Noise Summary Table	HB/HBnoise
Oscillator Swept Tuning Range and Phase Noise Measurement	HB/HBnoise
Ring Oscillator Measurements	
Starting and Stabilization of Ring Oscillators	PSS - Shooting/tstab
FM Jitter Measurement using PSS Shooting and Pnoise Jitter Analyses	PSS - Shooting/Pnoise
Calculating the Swept Tuning Range and Phase Noise for the Ring Oscillator	PSS - Shooting/Pnoise

Simulation Methods

There are two methods you can use to simulate oscillators using SpectreRF:

- Harmonic Balance (HB)
- Shooting Newton

The Harmonic Balance method is recommended for simulating mildly-nonlinear oscillators with resonators, such as LC oscillators and crystal oscillators. Shooting PSS method is recommended for simulating strongly nonlinear resonatorless oscillators, such as ring oscillators or relaxation oscillators.

Phases of Autonomous PSS/HB Analysis

A PSS/HB analysis has two or three phases.

1. If the *Calculate initial conditions (ic) automatically* check box is selected, then after the DC analysis for the time-zero timepoint has been calculated, the oscillator frequency and amplitude are estimated. The beginning of the transient analysis in the tstab interval includes this estimate.

2. A transient analysis phase to initialize the circuit.

   The transient analysis phase is divided into three intervals:

   a. A beginning interval that starts at *tstart*, which is normally 0, and continues through the onset of periodicity for the independent sources and continues through the longest delay time of the periodic sources, or the last point in a PWL waveform, whichever is the longest.

   b. A stabilization interval of length *tstab*

      In the hb *Choosing Analyses* form, you have the option to have Spectre automatically calculate the length needed for tstab.

      For driven circuits, the stabilization interval is optional. For autonomous circuits, *tstab* needs to be set to an interval that provides the circuit a good initial condition.

   c. A final interval that is four times the estimated oscillation period specified in the PSS/HB *Choosing Analyses* form. During the final interval, the PSS/HB analysis monitors the waveforms in the circuit and improves the estimate of the oscillation period. During this final interval, Spectre looks for frequency divider outputs in the circuit.

3. A Shooting or Harmonic Balance phase to compute the periodic steady state solution

   During this phase, the circuit is iterated repeatedly over one period. The length of the period and the initial conditions are modified to find the steady state solution.

See *Oscillators and Autonomous PSS Analysis* in *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more information on the autonomous PSS analysis algorithm. You may also refer to *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator RF Analysis User Guide*.

Phase Noise and Oscillators

Oscillators tend to amplify any noise present near the oscillation frequency. The closer the noise frequency is to the oscillation frequency, the greater the noise amplification. Noise amplified by the oscillator in this manner is called *phase noise*. Phase noise is the most significant source of noise in oscillators, and because phase noise is centered about the oscillation frequency, filtering can never completely remove it.

You can understand phase noise if you recognize that the phase of an oscillator is arbitrary because there is no drive signal to lock to. Any waveform that is a solution to an oscillator can be shifted in time and still be a solution. If a perturbation disturbs the phase, nothing restores the phase, so it drifts without bound. If the perturbation is random noise, the drift is a random walk. In addition, the closer the perturbation frequency is to the oscillation frequency, the better it couples to the phase and the greater the drift. The perturbation need not come from random noise. Noise might also couple into the oscillator from other sources, such as the power supplies.

Starting and Stabilizing Feedback Oscillators

To simulate an oscillator using either HB or Shooting PSS analysis, you must first start the oscillator by supplying either of the following:

1. A brief impulse stimulus

   The stimulus should couple strongly into the oscillatory mode of the circuit and poorly into other long-lasting modes such as bias circuitry. This is usually done by adding a single pulse of current that is connected somewhere in the circuit to start the oscillations.

2. A set of initial conditions (ICs) for the components of the oscillator’s resonator.

3. For LC and High Q oscillators, you can select the *Calculate initial conditions (ic) automatically* checkbox in the *Choosing Analyses* form instead of providing any stimulus or initial condition (*ic*).

   This setting tells Spectre to do a variation of the linear stability analysis just before the beginning of the tstab interval. It provides a good estimate of the oscillation frequency amplitude and phase at the beginning of the tstab interval. With a start that is very close, convergence is much easier.

Note that when you set this option then ICs (initial conditions) are automatically ignored.

Regardless of which technique you use to start the oscillator, allow the oscillator to run long enough to stabilize before you start the HB or Shooting phase and compute the steady state solution. Adjust the *tstab* parameter to supply the additional stabilization period. High Q oscillators often need a longer tstab. For extremely high Q circuits, you may need to set tstab to about 50K to 200K periods of oscillation with the default method. However, when setting *Calculate initial conditions (ic) automatically* option you may reduce it to just 10 periods.

In HB, if you set *Run transient* to *Decide automatically* then you donot need to provide any tstab value and the simulator automatically decides how long to run the transient analysis to compute the steady state solution. Also, in HB, while setting the tstab value, you can select the *Detect Steady State* checkbox if you want the transient analysis to stop automatically as soon as the steady state is reached rather than runningthe transient analysis for the whole tstab value.

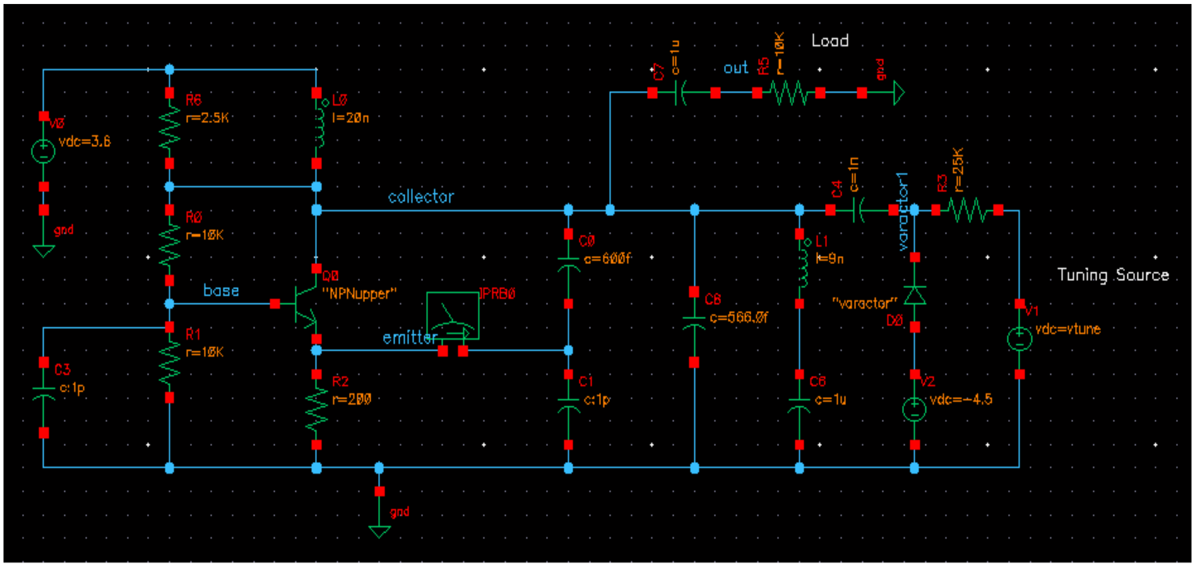
It is recommended toto save the initial transient simulation results. This can be done by settingthe *Save Initial Transient Results (saveinit)* option to *yes*. You can use this to verify that the oscillator has started up and is stable.

See *Oscillators and Autonomous PSS Analysis* in the *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more information on oscillator simulation. In addition to this you may also refer to *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator RF Analysis User Guide*.

The Oscillator Circuit

This example computes the periodic steady state solution and the phase noise for the *RF oscillator* circuit, as shown in Figure A-80.

Figure A-80 Schematic for the Oscillator Circuit oscillator_ckt

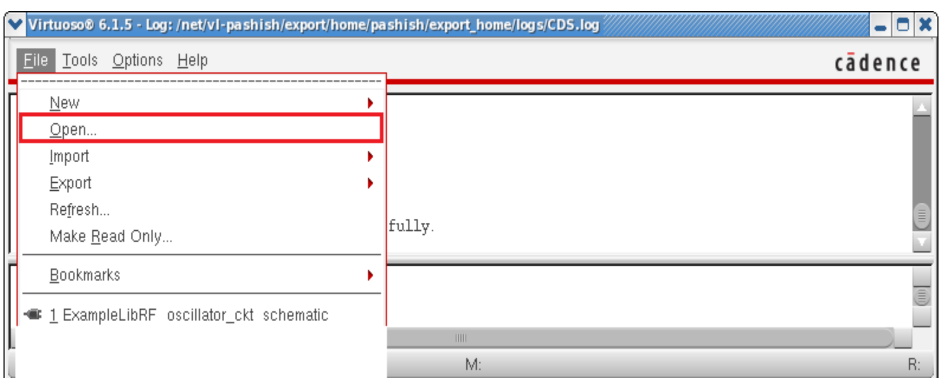


Setting Up to Simulate the Oscillator Circuit

Opening the Oscillator Circuit in the Schematic Window

1. In the Command Interpreter Window (CIW), choose *File – Open*.

Figure A-81 Virtuoso CIW Window - Opening Cellview

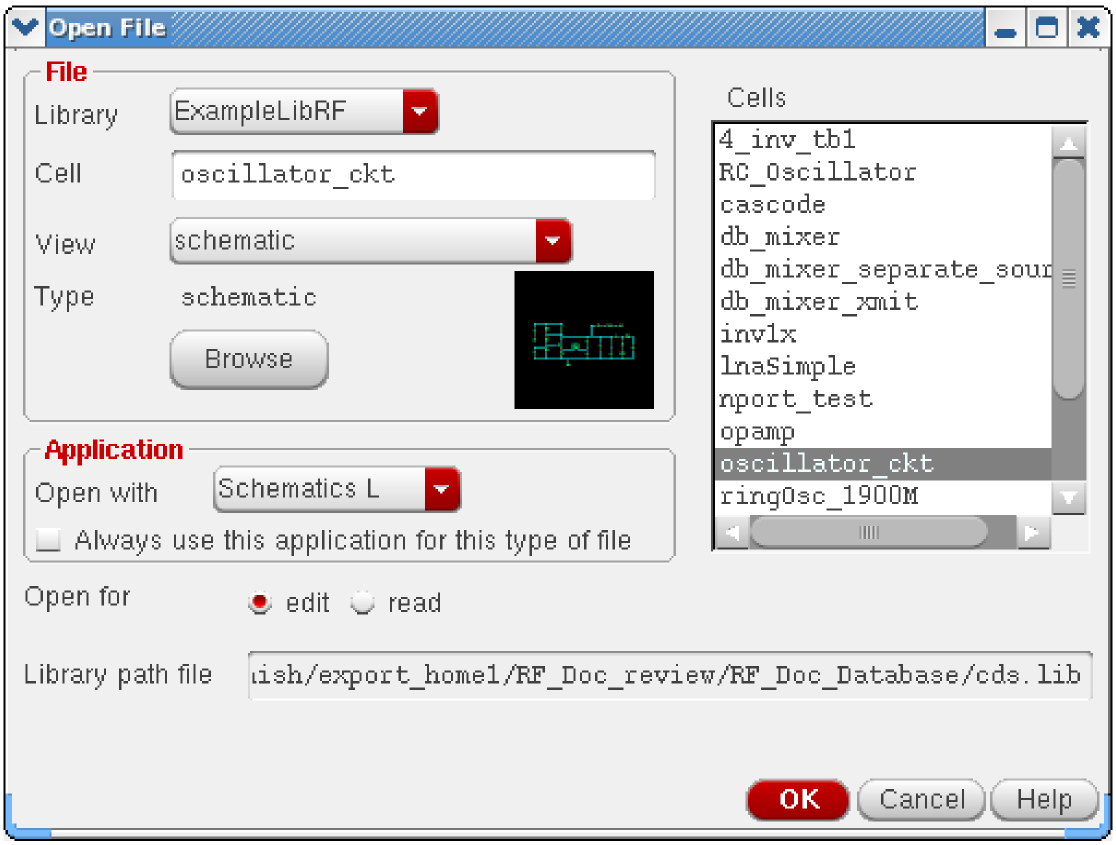


The *Open File* form is displayed.

2. Select *ExampleLibRF* from the *Library* drop-down list.

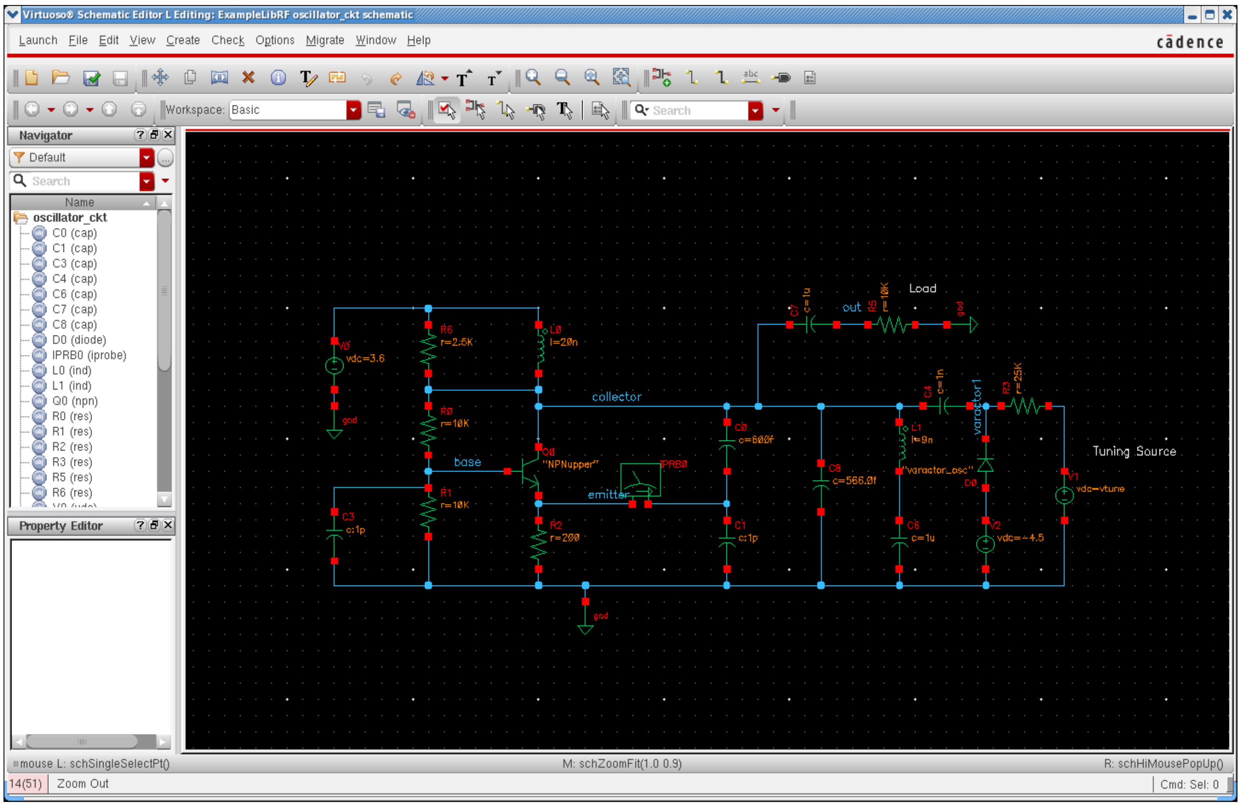
3. In the *Cells* list box, choose *oscillator_ckt*.
4. Select *schematic* from the *View drop-down list*.
5. In the *Application* section, select *Schematic-L* from the *Open with* drop-down list.
6. Leave *Open For* to *edit* (which is set by default).

Figure A-82 Open File Form to open the oscillator_ckt cell's Schematic View



7. Click *OK* to close the *Open File* form.
8. This will open *oscillator_ckt* schematic in Virtuoso Schematic Editor L window, as shown below.

Figure A-83 oscillator_ckt schematic in VSE-L Window

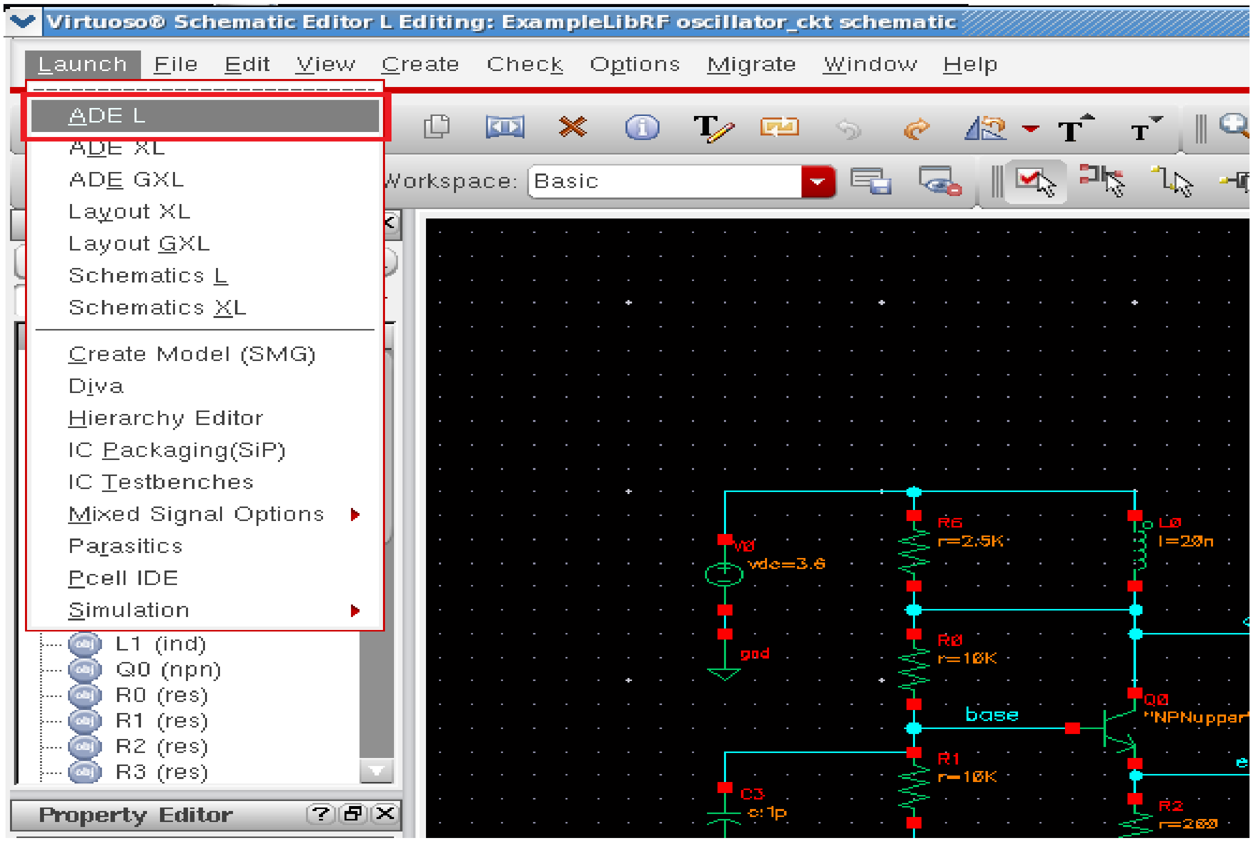


This example computes the periodic steady state solution for the *oscillator_ckt RF* oscillator circuit. You perform a PSS-HB or HB analysis first because the periodic steady state solution must be determined before you can perform any other periodic small-signal analysis like *pstb* or *hbnoise* etc. to determine the stability or phase noise.

Setting up ADE-L for Oscillator Simulation

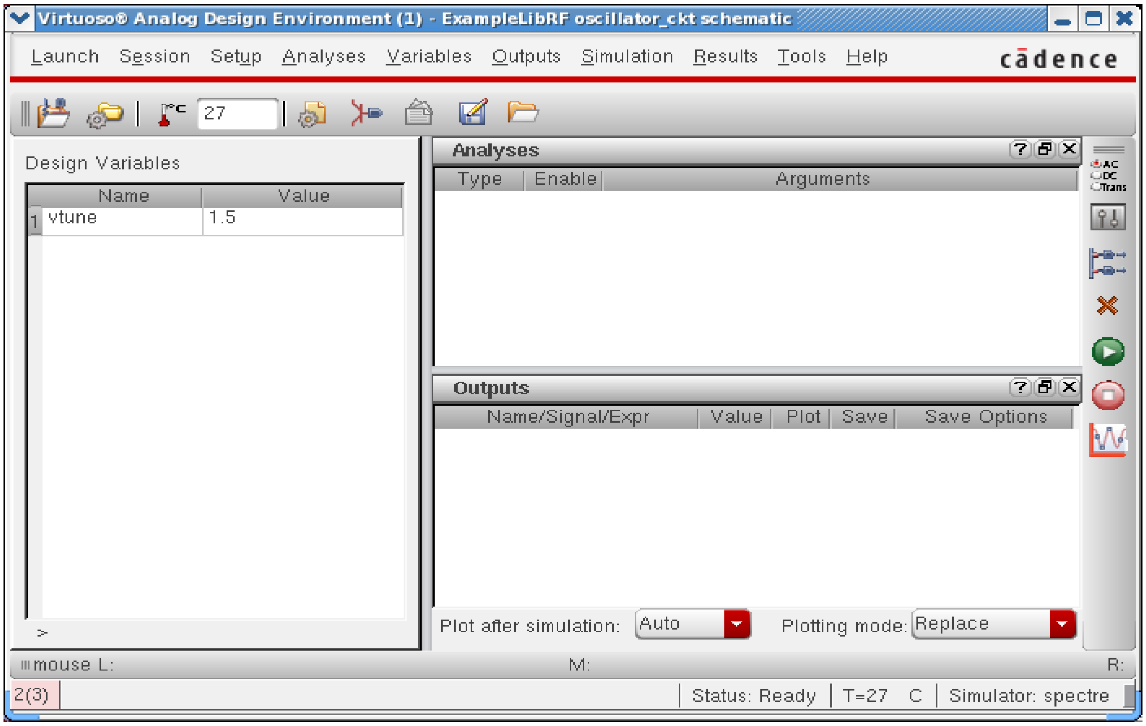
1. In the Schematic Window, choose *Launch - ADE L.*

Figure A-84 Opening ADE L window from VSE L window



The Virtuoso Analog Design Environment Window is displayed, as shown below:

Figure A-85 Virtuoso Analog Design Environment Window



2. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

The *Choosing Simulator/Directory/Host* form appears.

3. Specify the following:

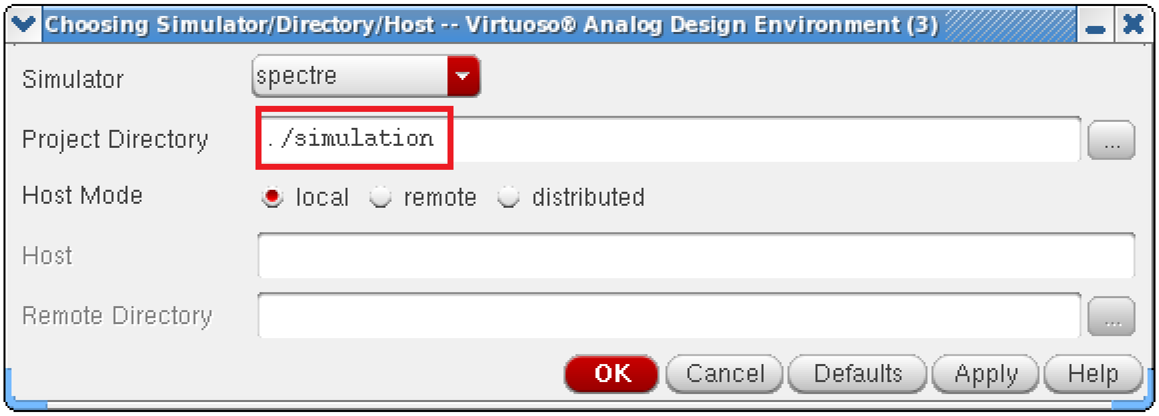
a. Choose *spectre* for the *Simulator*.

b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your ~/simulation directory. You may change the default location by editing the settings in your .cdsinit or .cdsenv file. In this workshop, the simulation directory is set to ./simulation.

c. Select the *Host Mode* that corresponds to your situation. For remote or distributed mode, please contact your Cadence tools System Administrator for specific set-up instructions.

The completed form is displayed.

Figure A-86 Choosing Simulator/Director/Host Form

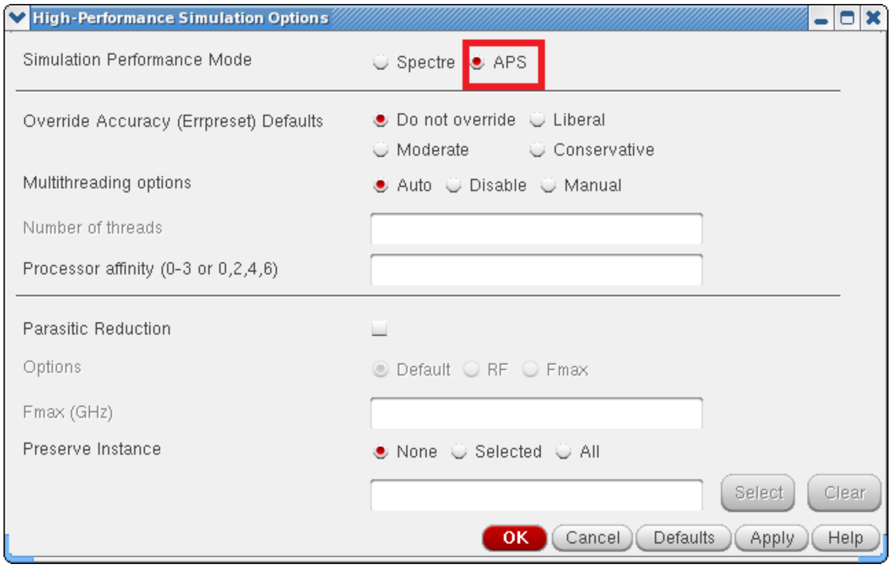


4. Click *OK*.
5. Set up the High Performance Simulation Options.

   In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

   In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually, it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

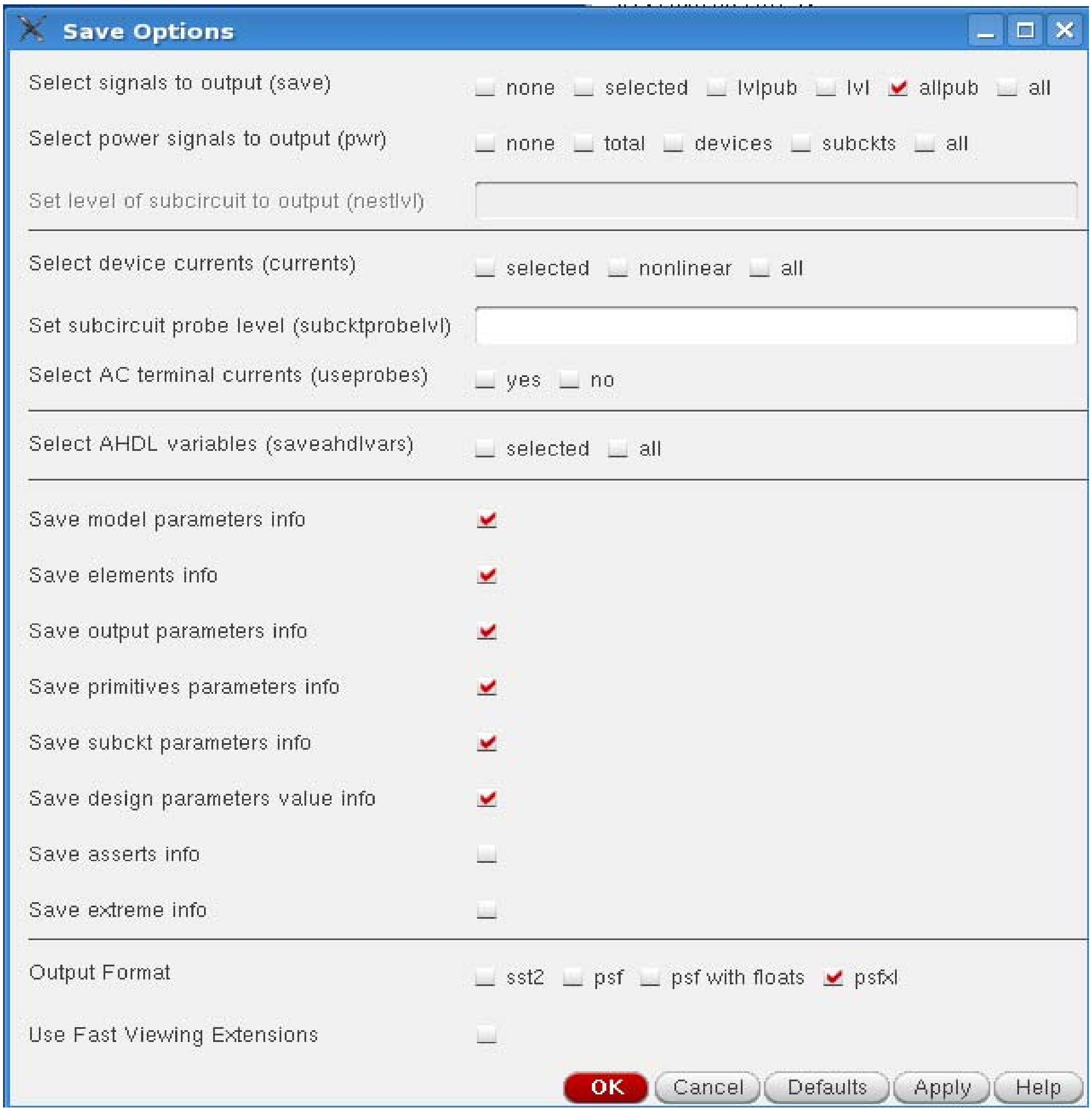
Figure A-87 High Performance Simulation Options Form



6. Click *OK*.
7. In the *Virtuoso Analog Design Environment window*, choose *Outputs – Save All*.

   The *Save Options* form is displayed, as shown below.

Figure A-88 Save Options Form



8. In the *Select signals to output* section, be sure *allpub* is highlighted.

   This is the default which saves all node voltages at all levels of hierarchy, but it does not include the node voltages inside the device models.

9. To save the currents, choose *nonlinear* for the *Select device currents (currents)* option, if you just want to save the device currents, or choose *all* if you want to save all

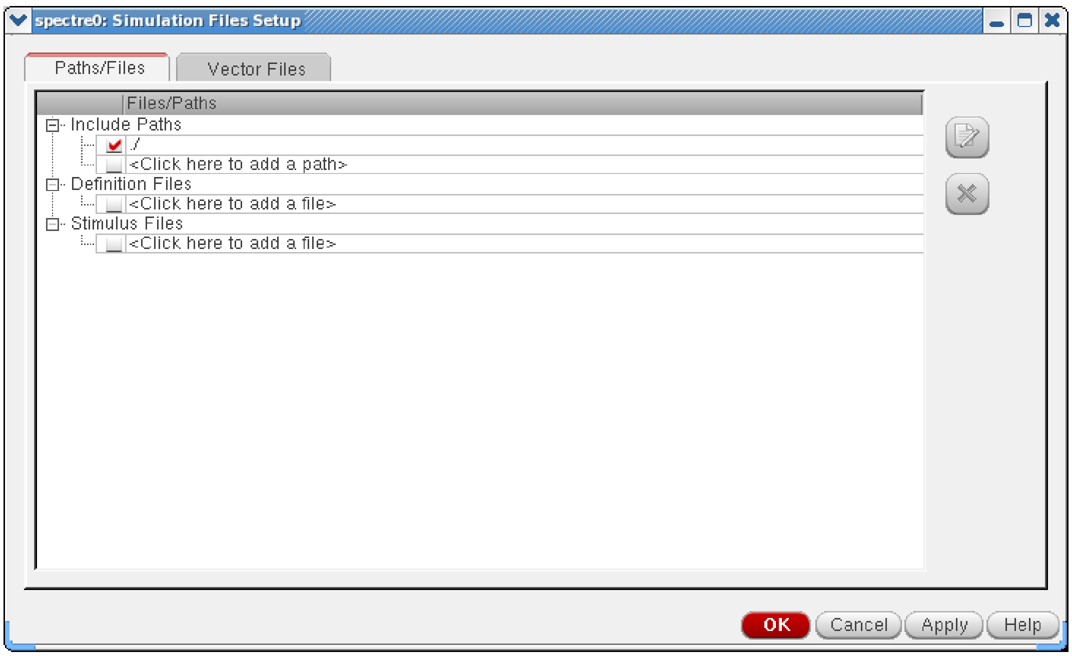
the currents in the circuit. When you save the currents, more disk space is required for the results file.

10. Click *OK*.

11. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

The *Simulation Files Setup* form is displayed.

Figure A-89 ***Simulation Files Setup*** **Form**



12. Select *./* by clicking in the *Include Paths* section.

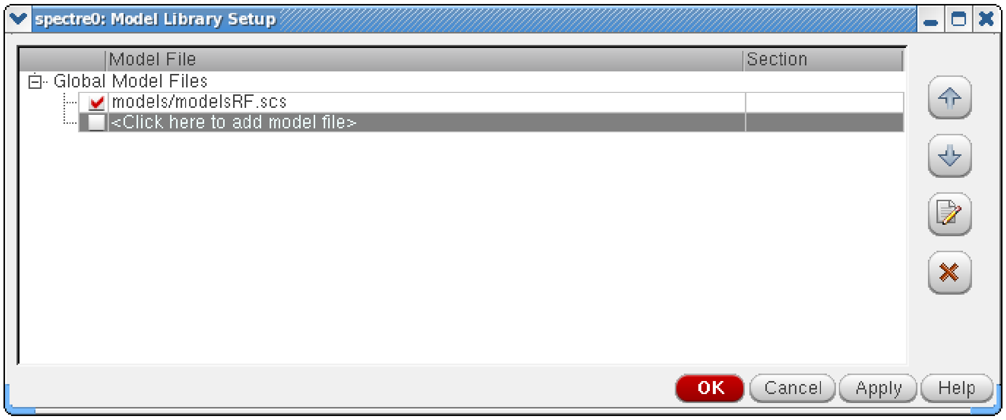
13. Click *OK* to close the *Simulation Files Setup* form.

14. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries*

The *Model Library Setup* form is displayed.

15. In the *Model File* field, type the path to the model file including the file name, such as *models/modelsRF.scs*.

Figure A-90 ***Model Library Setup* Form**



You can also browse to *modelsRF.scs* file.

16. Click *OK.*

Calculating the Steady-State Solution using PSS Harmonic Balance

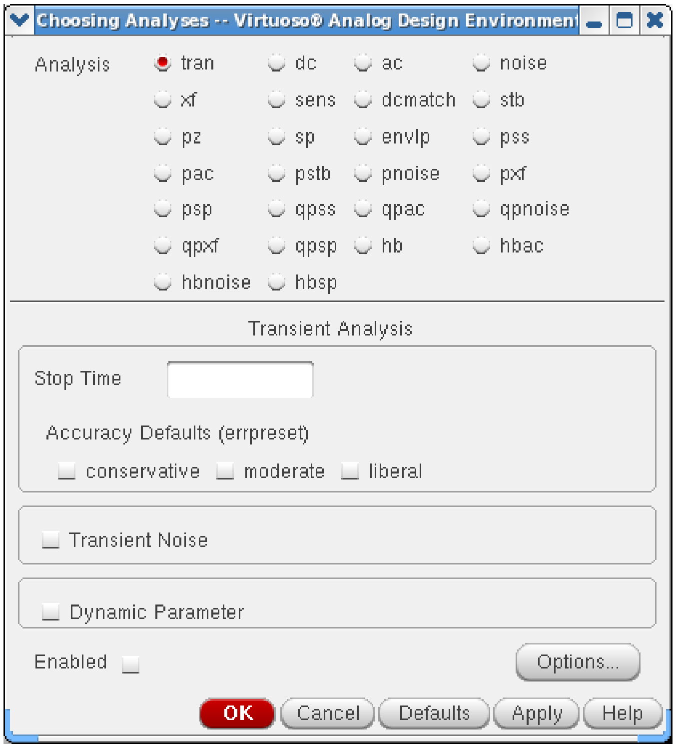
In this very first measurement, you will set up the PSS analysis and then determine the oscillation frequency of the oscillator. You will use the *Harmonic Balance* Engine in the PSS Analysis. You can also use *hb* analysis directly. Here, we have chosen PSS analysis with *Harmonic Balance* Engine because you will perform *pstb* analysis in the next measurement. Since there is no corresponding hbstb analysis available for hb analysis, this setup is done.

Setting up the PSS Analysis

1. Select *Analyses –> Choose* in the Virtuoso Analog Design Environment window.

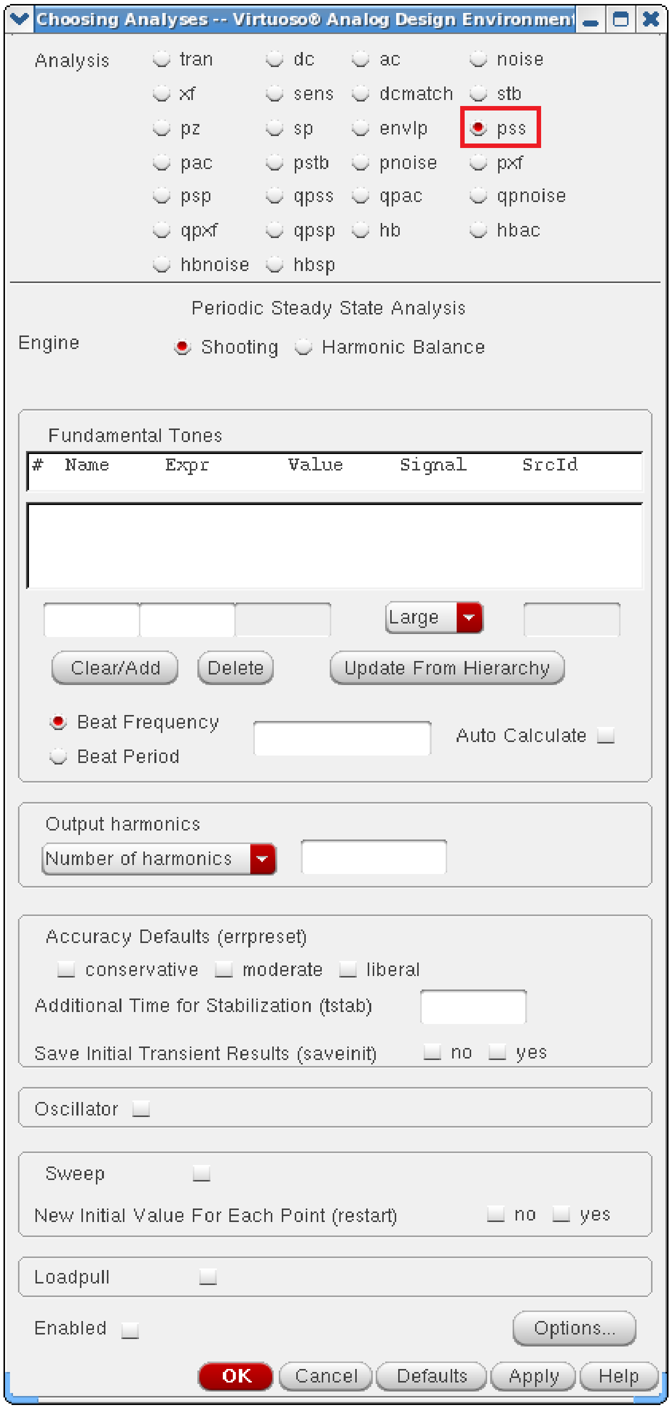
The *Choosing Analyses* form is displayed, as shown below.

Figure A-91 The *Choosing Analyses* Form



a. Select *pss* as Analysis. The form expands.

Figure A-92 The Choosing Analyses Form- Setting PSS Analysis



b. Select *Harmonic Balance* as Engine.

c. In the *Beat Frequency* field, type 1.9G. The frequency entered here is an approximate frequency of oscillation.

d. Leave Oversample Factor (*oversamplefactor*) as default (that is blank) which meansthat the Oversample Factor is equal to 1. Since the oscillator has sinusoidal waveforms, an oversample of 1 is appropriate.

e. In the *Number of harmonics* field, type 15.

   In general, you want to choose a number that is high enough to capture the nonlinearity of the circuit. Start with 10, and run the simulation. Increase by about 50% to 15 and re-run the simulation. If the harmonics donot change appreciably, then 10 is enough. If they change, increase the number again by about 50%. Use the smallest number of harmonics for the answer to be stable.

f. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. Conservative is recommended for all the oscillators.

g. Type 5n in the *Additional Time for Stabilization (tstab)* field. *tstab* is typically set to about 10 to 20 periods of the oscillation frequency when the *Calculate initial conditions (ic) automatically* checkbox is selected.

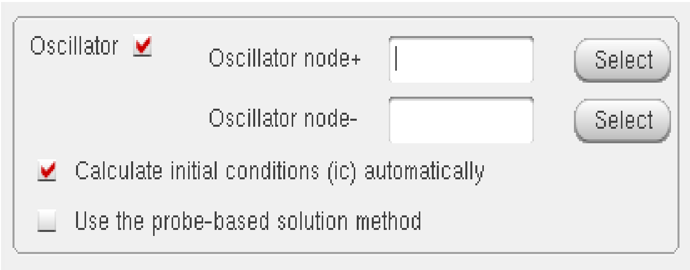
h. Select *yes* for *Save Initial Transient Results*. This will help in visualizing the buildup of the oscillation waveform.

i. Leave the *Harmonic Balance Homotopy Method* as *default* which is *tone*. HbHomotopy (*hbhomotopy*) is one of the HB Convergence Options which helps in the convergence of the simulation of circuit.

   You can refer to the earlier chapters for more details.

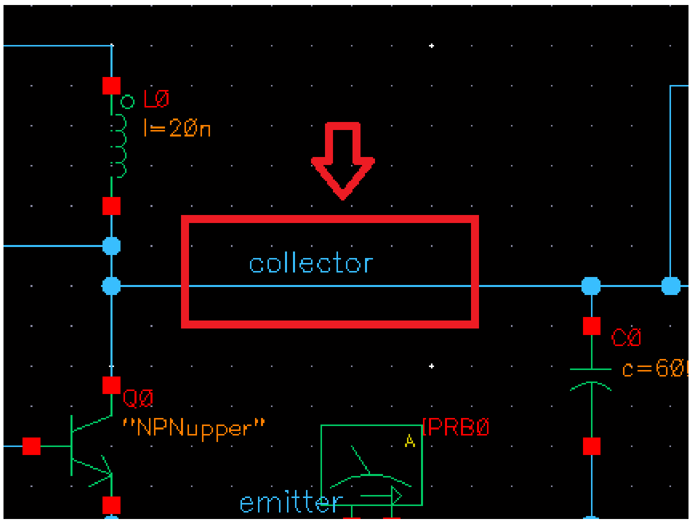
j. Select *Oscillator* . This is required for simulating an autonomous circuit.

Figure A-93 The Choosing Analyses Form - Oscillator Section



k. In the *Oscillator node* field, click *Select* just to the right. In the schematic, select the *collector* node. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it.

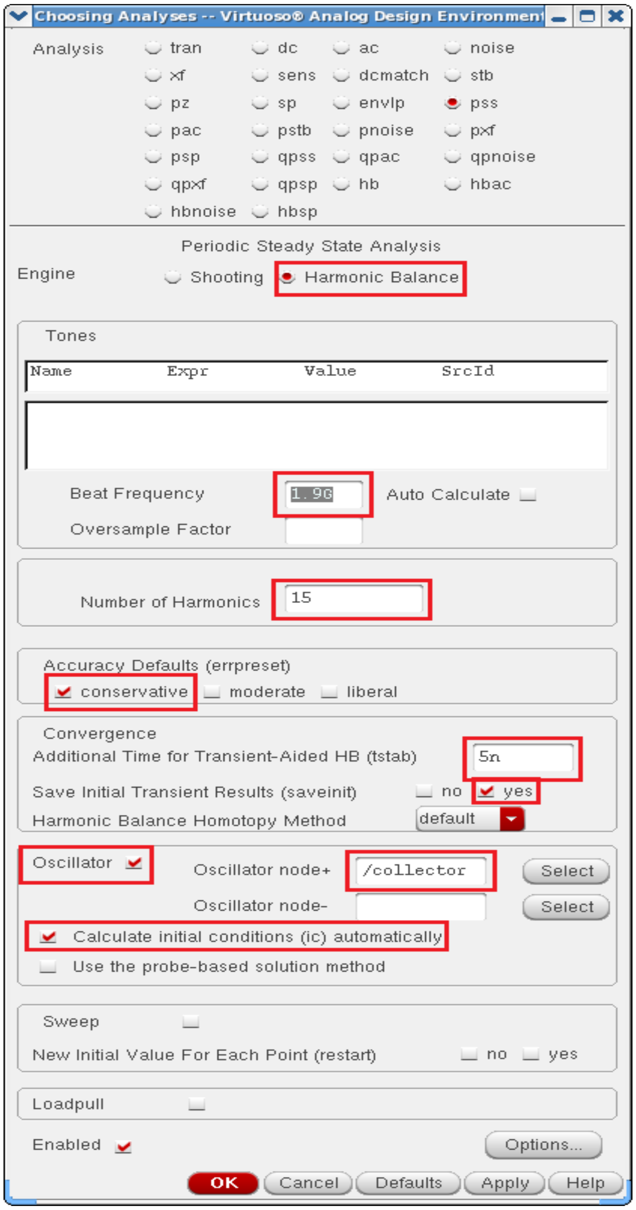
Figure A-94 Selecting *collector* net on oscillator_ckt schematic



l. If you have an LC oscillator, leave the *Calculate initial conditions (ic) automatically* checkbox selected (this is the default).

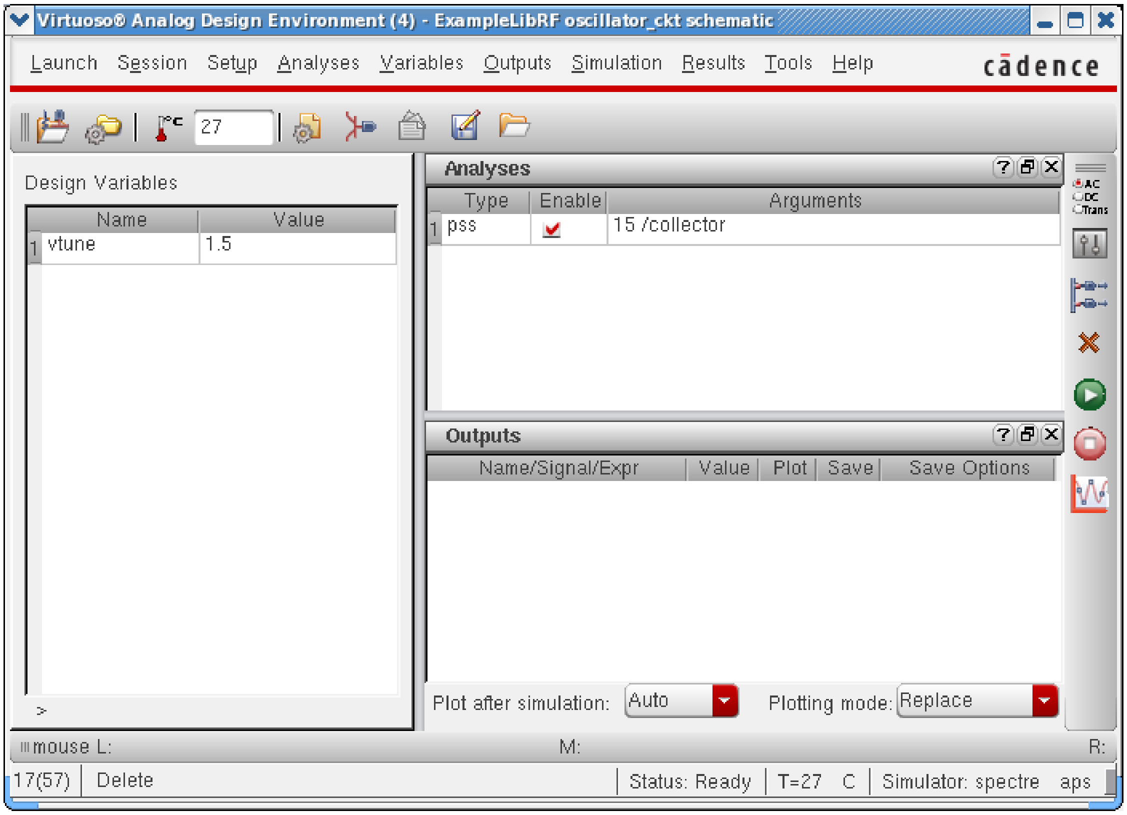
Note that *Calculate initial conditions (ic) automatically* is used to start the oscillator. Other methods to start the oscillator include putting a single current pulse into the resonator, setting initial conditions(ic), or ramping up the power at time = zero plus.

Figure A-95 Choosing Analyses Form - PSS-Harmonic Balance Setup

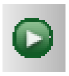


Now click *Ok* at the bottom. This will close the Choosing Analyses Form and add the *pss* analysis in the *Analyses* section of ADE window as shown below:

Figure A-96 ADE Simulation Window - PSS Analysis



Running the PSS analysis

Once finished setting up the PSS analysis click the green icon  on the right hand side of the Analog Design Environment window or on the Schematic window to run the simulation.

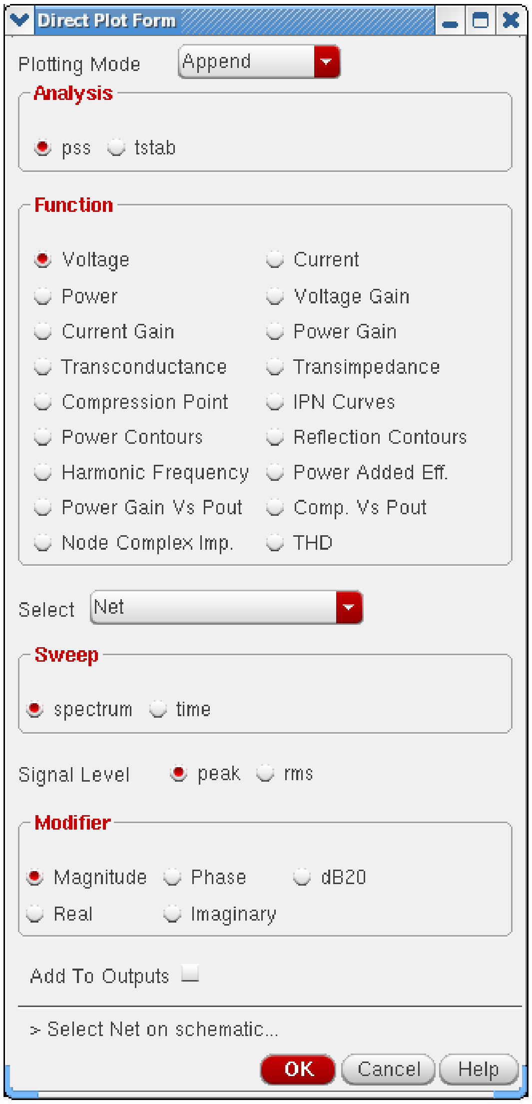
This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

Next, you will plot the results.

Plotting the PSS Analysis Results

In the *Virtuoso Analog Design Environment* window, choose *Results - Direct Plot - Main Form*. The *Direct Plot Form* window is displayed.

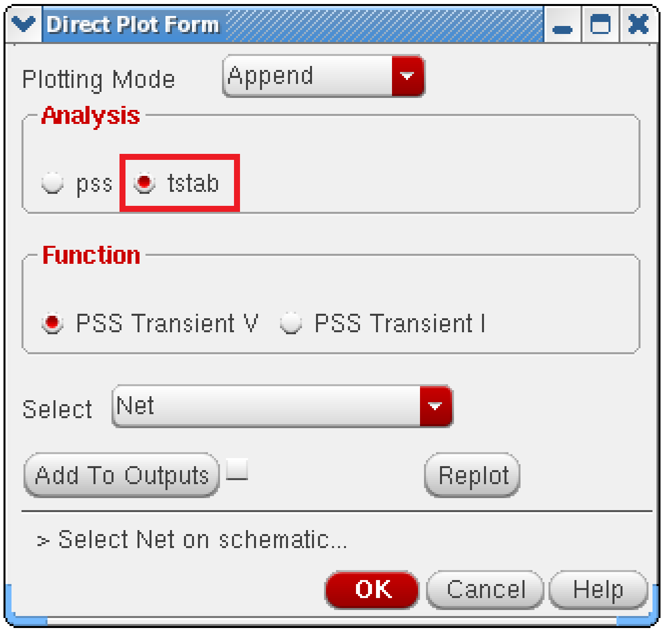
Figure A-97 The Direct Plot Form Window



Plot the oscillator startup waveform from tstab run.

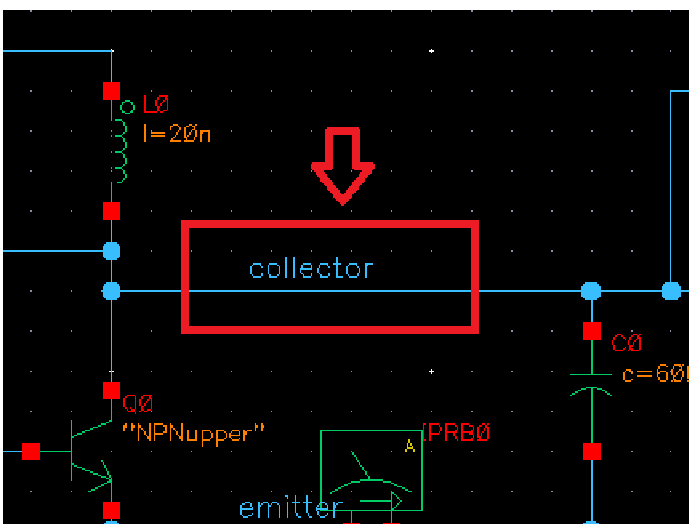
1. In the *Direct Plot Form* window, select *tstab* in the *Analysis* section.
2. Leave *Function* as *PSS Transient V* which is set by default.
3. Select *Net* in the center of the form. (This is the default. You can also select differential nets).
4. The *Direct Plot Form* window should like the following:

Figure A-98 PSS Analysis Direct Plot Setup - Initial Transient



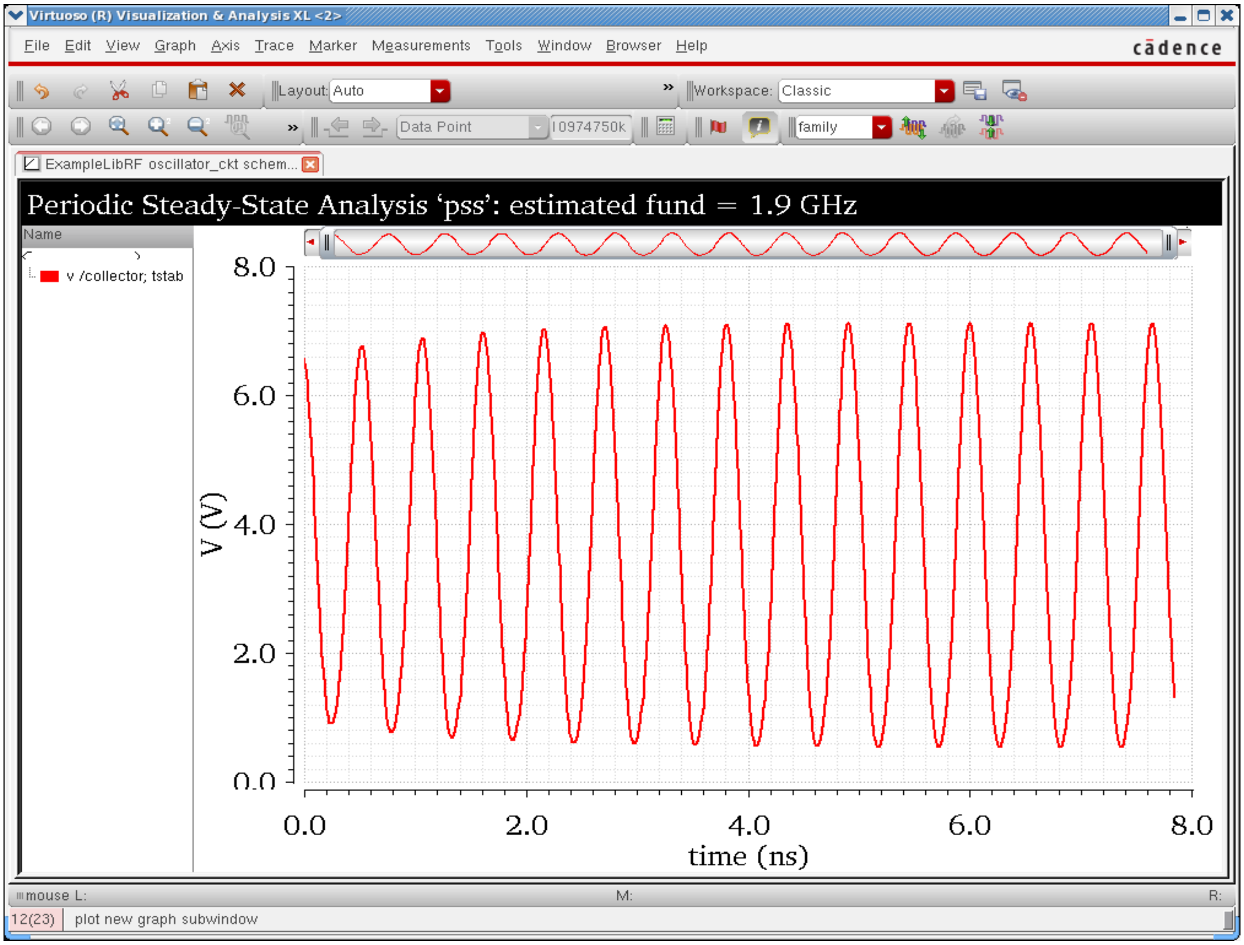
5. Select the *collector* net in the schematic. It is located just below the *collector* label.

Figure A-99 Selecting *collector* net on oscillator_ckt schematic



The waveform window is displayed.

Figure A-100 PSS Analysis - Initial Startup Waveform during tstab interval



This plot shows how the initial startup waveform for oscillator gets build up during the tstab interval. Note that the oscillator starts up immediately after time zero. This is because the calculate initial conditions was automatically set in the *Choosing Analyses* form.

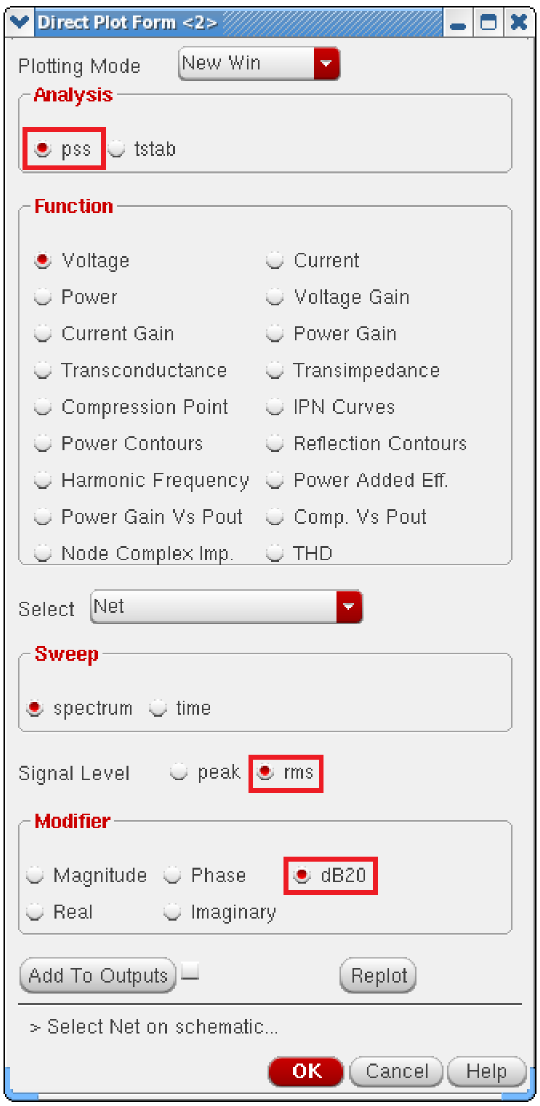
Next you will plot the oscillator output spectrum

Plot the oscillator spectral content, as follows:

1. In the *Direct Plot Form* window, set the *Plotting Mode* to *New Win*.
2. Select *pss* as Analysis. (This is the default)
3. Leave *Function* as *Voltage* which is set by default.
4. Leave the *Select* section of the form set to the default value which is *Net* (You can also select differential nets).

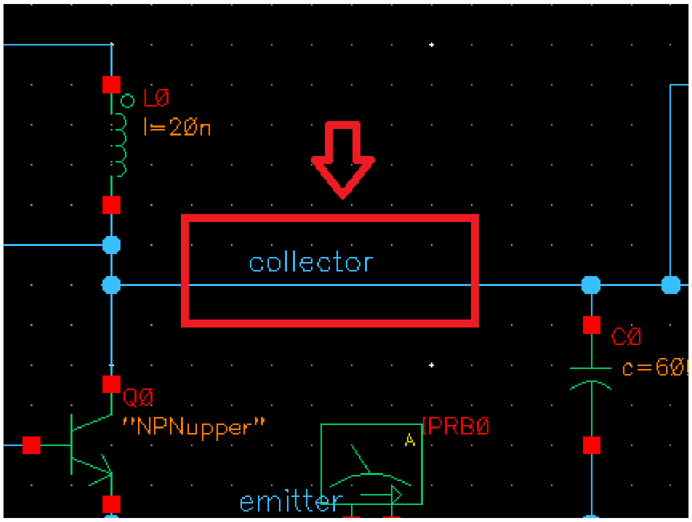
5. Select *Sweep* as *spectrum*. (This is the default)
6. Select *rms* as Signal Level (the default is *peak*).
7. Select *dB20* as Modifier.
8. Your *Direct Plot Form* window should like the following:

Figure A-101 PSS Analysis Direct Plot Form Setup



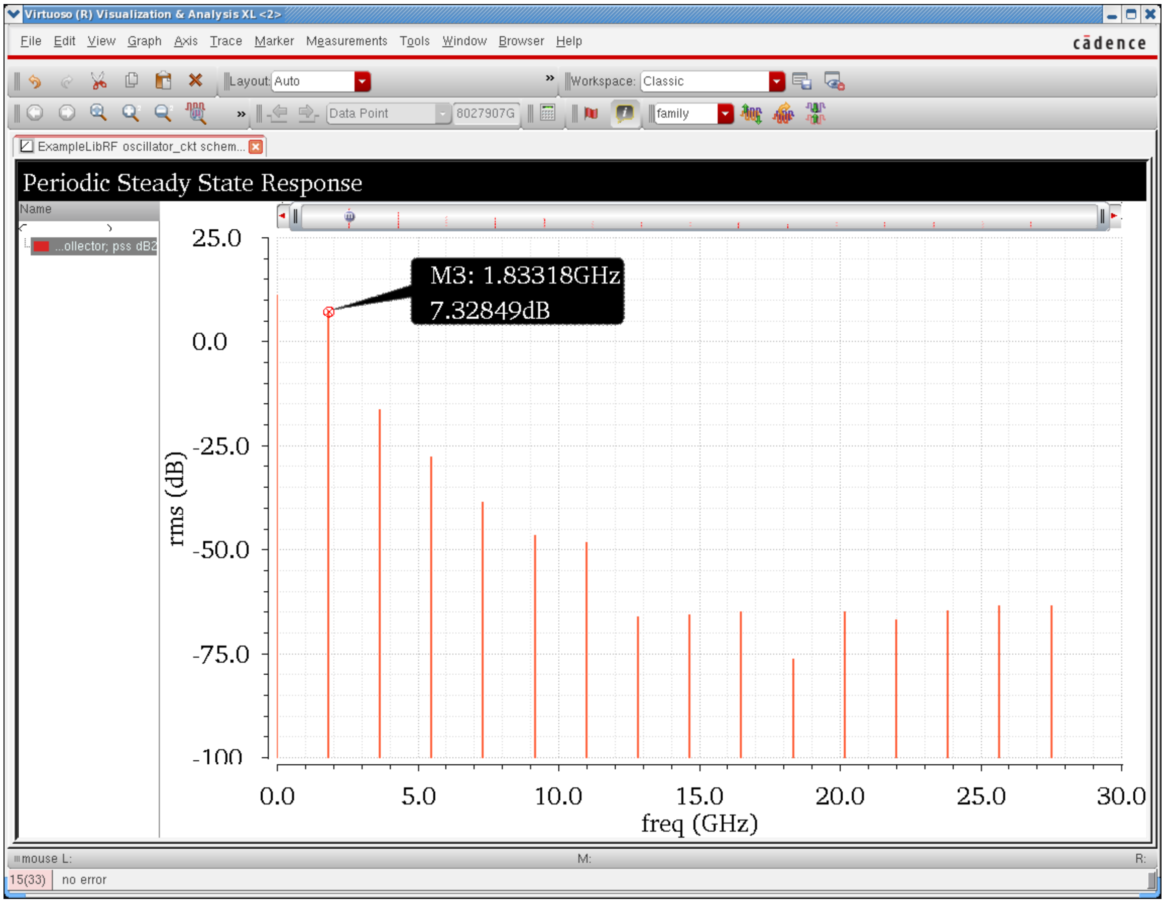
9. Select the *collector* net in the schematic. It is located just below the *collector* label.

Figure A-102 Selecting *collector* net on oscillator_ckt schematic



The waveform window is displayed, as shown below.

Figure A-103 PSS Analysis output Graph Window - Voltage Spectrum Plot



10. In the waveform window, position your cursor near the first harmonic, and press the 'm' key. Here 'm' is the bindkey to place a trace marker on the graph. The first harmonic is chosen as this is the frequency oscillator is designed for.

Note that this frequency is 1.833GHz. This is the frequency of oscillation.

11. In the *Direct Plot Form*, click *Cancel*.

12. In the ViVA window, choose *File - Close All Windows*.

13. Clean up the screen for the next set of measurements.

14. Close the Analog Design Environment windowby choosing *Session - Quit*.

15. Click *No* in the *Save State* window.

16. In the Schematic window, choose *File - Close*.

To summarize, a PSS analysis using Harmonic Balance was setup and simulation was run to determine the oscillation frequency of the oscillator. Next an Oscillator Loop Gain measurement will be performed.

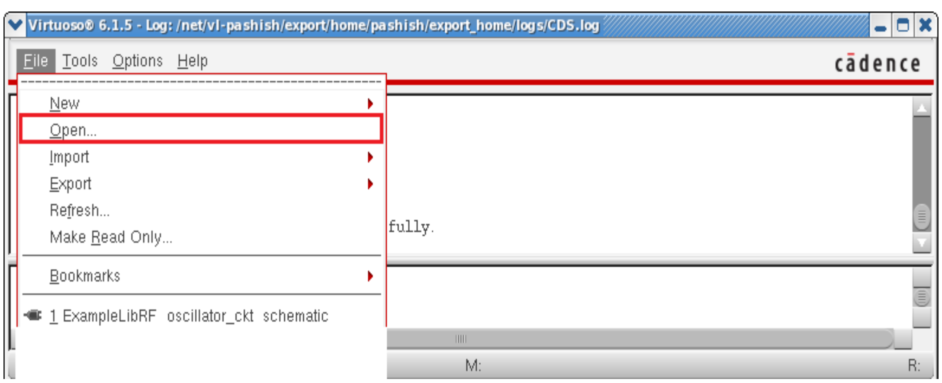
Oscillator Loop Gain Measurement

Oscillator loop gain measurements are typically made with a combination of linear stability (*stb*), PSS (*pss*), and Pstb (*pstb*) analyses. The reason we need *pss* analysis instead of *hb* here is because there is no corresponding small signal *hbstb* analysis available. The *pss* analysis solves for one period of the settled time-domain waveform. Stability Analysis (*stb*) allows the measurement of the loop gain and phase. This is quite useful for the design of the feedback network. Note that, *stb* analysis is a linear analysis and therefore does not take non-linearities of the circuit into account. However, it gives an approximate value of loop gain magnitude and phase and is faster to run. In order to get more accurate values of loop gain and phase of periodically time-varying non-linear circuits, you need to run *pstb* analysis. The periodic stability analysis (*pstb*) evaluates the local stability of a periodically time-varying feedback circuit. It is a small-signal analysis, like *stb* analysis, except that the circuit is first linearized about a periodically varying operating point (determined using *pss* analysis) as opposed to a simple DC operating point (which is used in *stb* analysis). Linearizing about a periodically time-varying operating point allows the stability evaluation to include the effect of the time-varying operating point.

Opening the Oscillator Circuit in the Schematic Window

1. In the Command Interpreter Window (CIW), choose *File – Open*.

Figure A-104 Virtuoso CIW Window - Opening Cellview

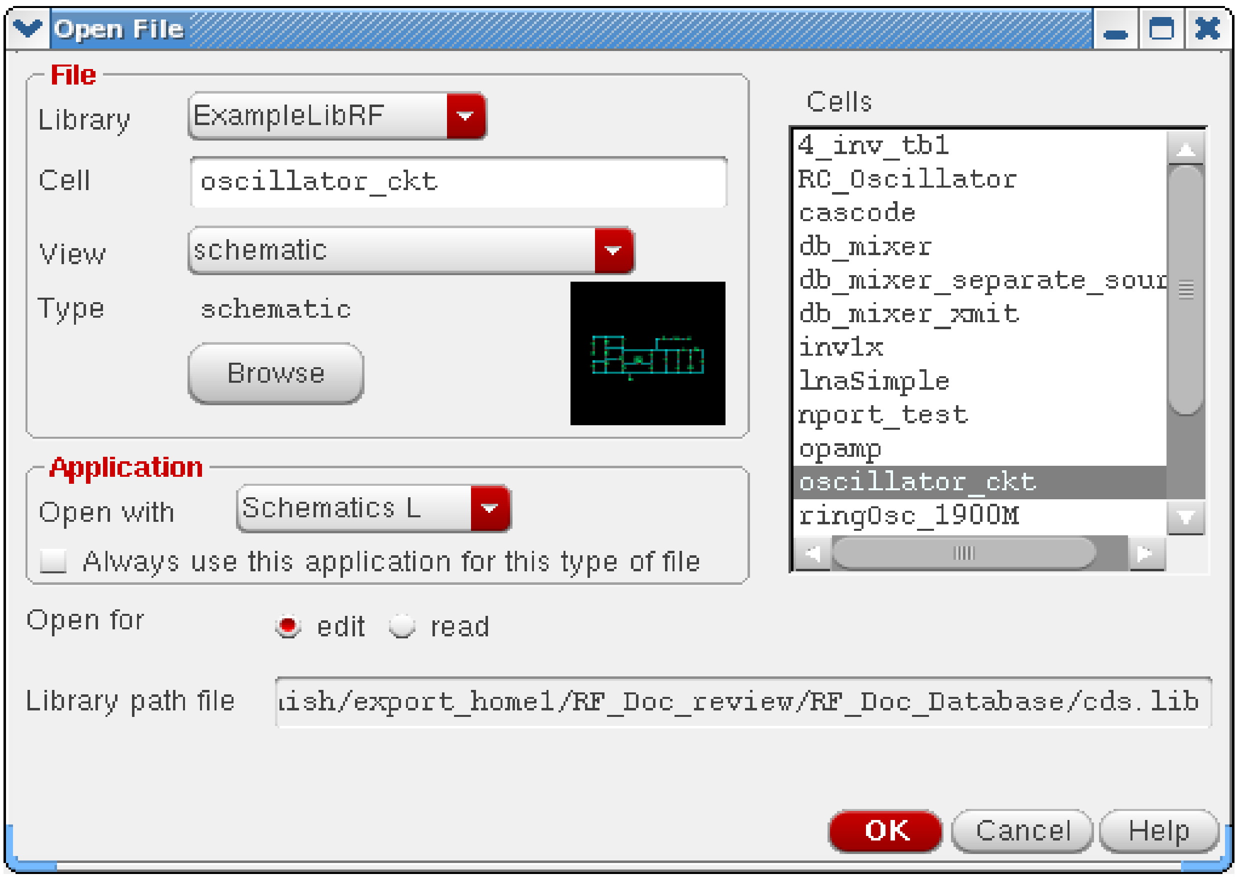


The *Open File* form is displayed.

2. Selec t*ExampleLibRF* from the *Library* drop-down list.

3. In the *Cells* field, type *oscillator_ckt*.
4. Choose *schematic* from the *View* drop-down list.
5. In the *Application* section, select*Schematics L* from the *Open With drop-down list.*
6. Leave *Open For* to *Edit* (which is set by default).

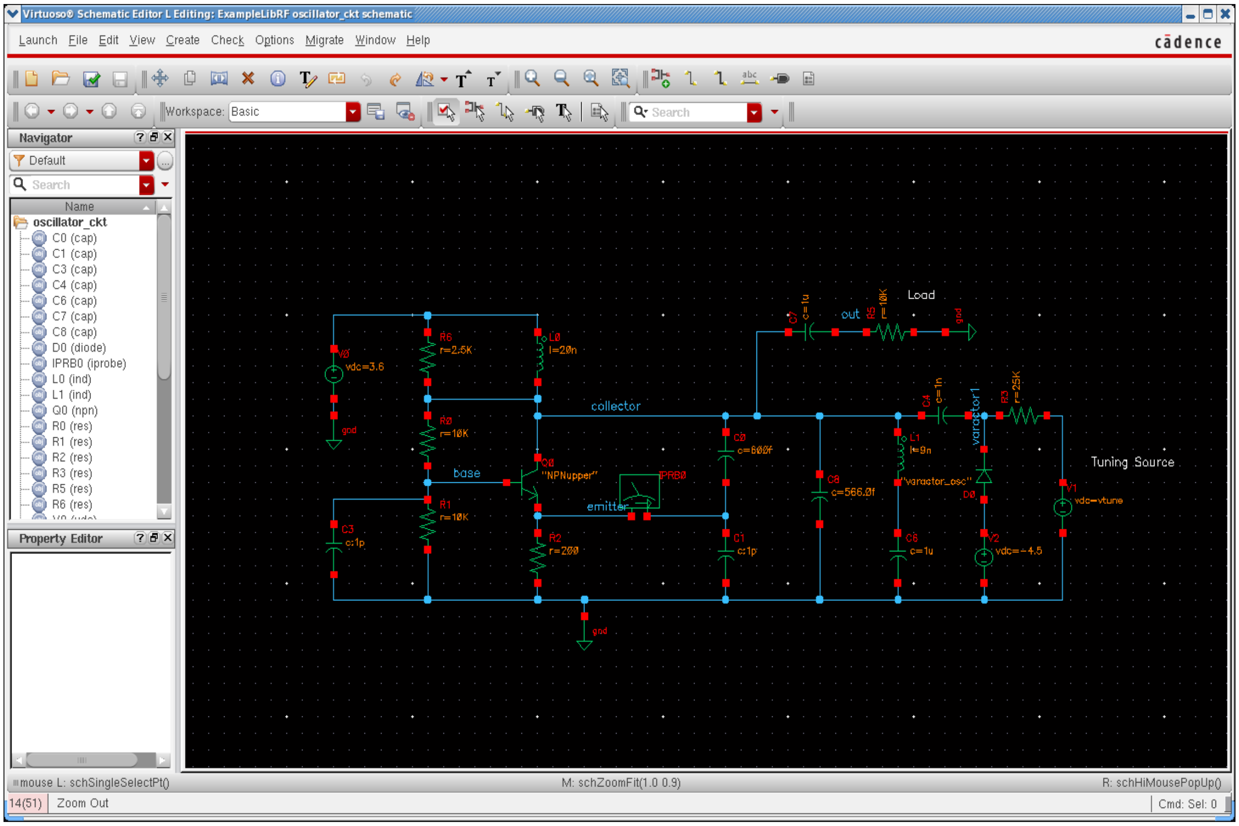
Figure A-105 Open File Form to open the oscillator_ckt cell's Schematic View



7. Once all the setup is done. Click *OK*.

   This will open the oscillator_ckt schematic in Virtuoso Schematic Editor L window, as shown below:

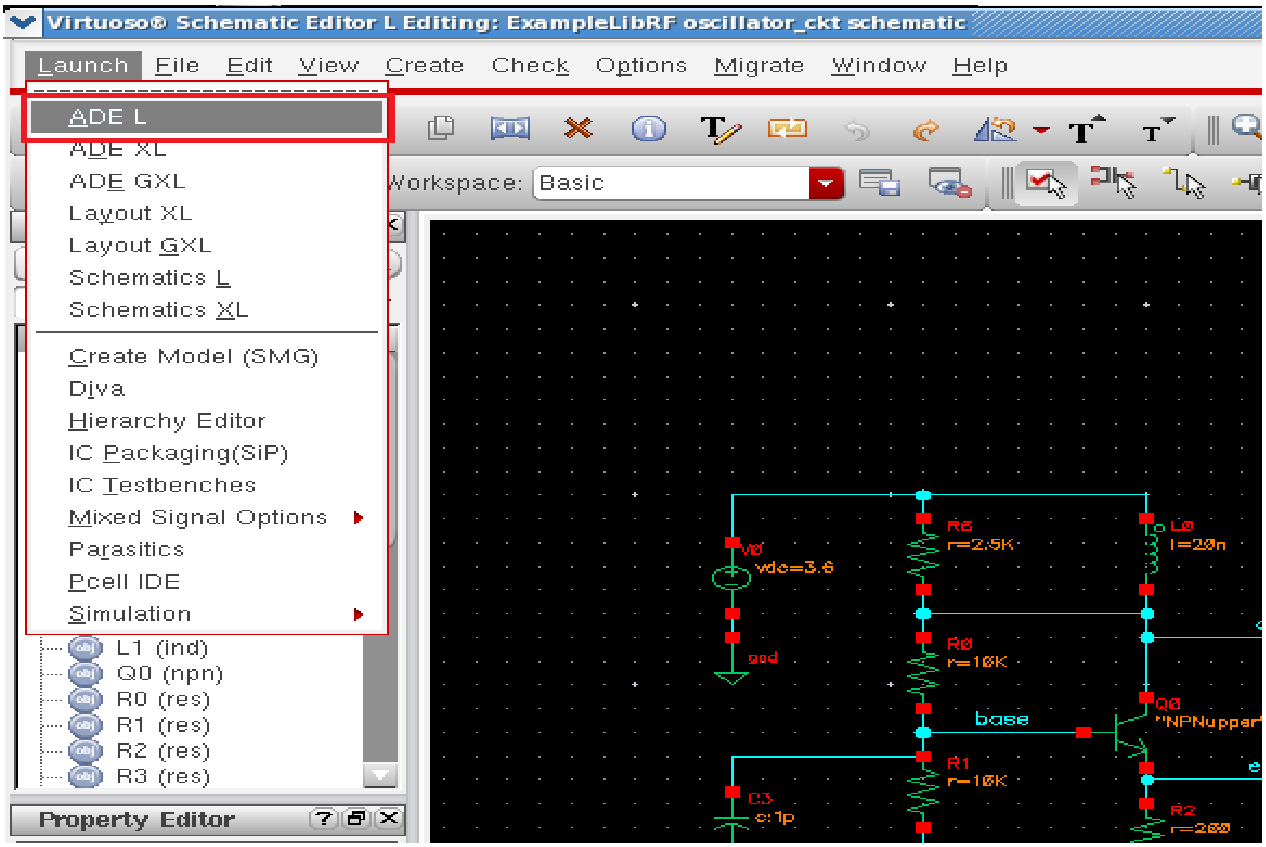
Figure A-106 oscillator_ckt schematic in VSE L Window



Setting up the stb Analysis

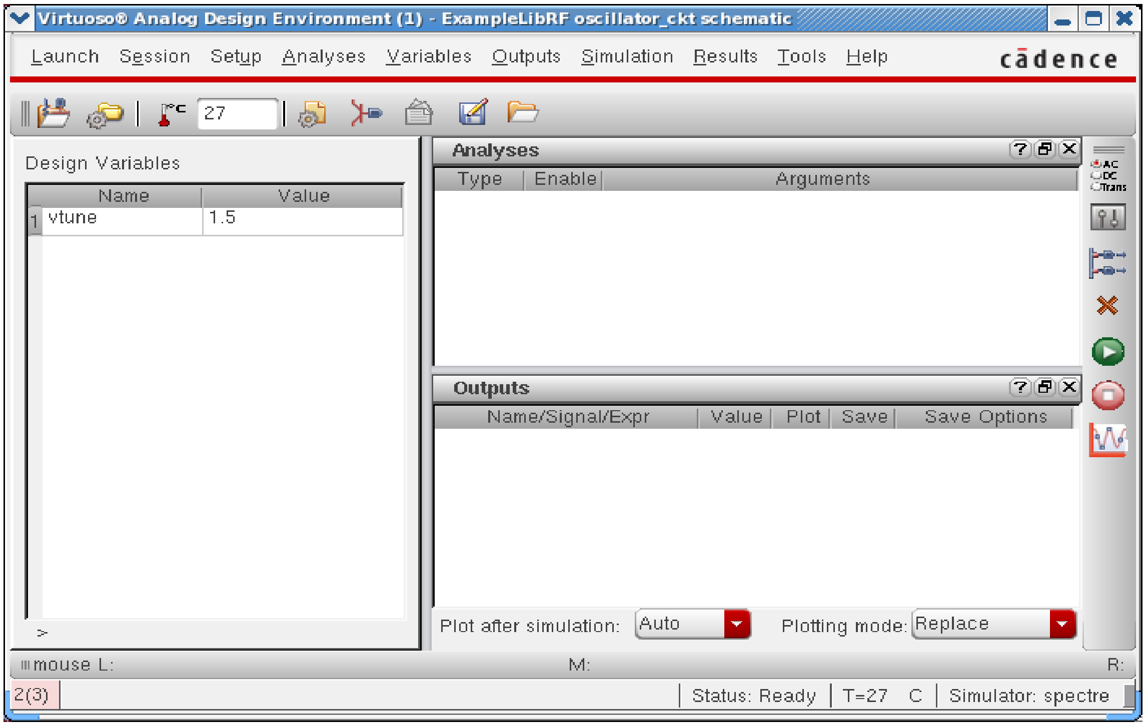
1. In the Schematic Window, choose *Launch - ADE L.*

Figure A-107 Opening ADEL window from VSE window



2. The *Virtuoso Analog Design Environment* Window is displayed, as shown below.

Figure A-108 Virtuoso Analog Design Environment Window



3. Choose *Setup – Simulator/Directory/Host* in the Virtuoso Analog Design Environment window.

   The *Choosing Simulator/Directory/Host* form is displayed.

4. Specify the following in the*Choosing Simulator/Directory/Host* form:

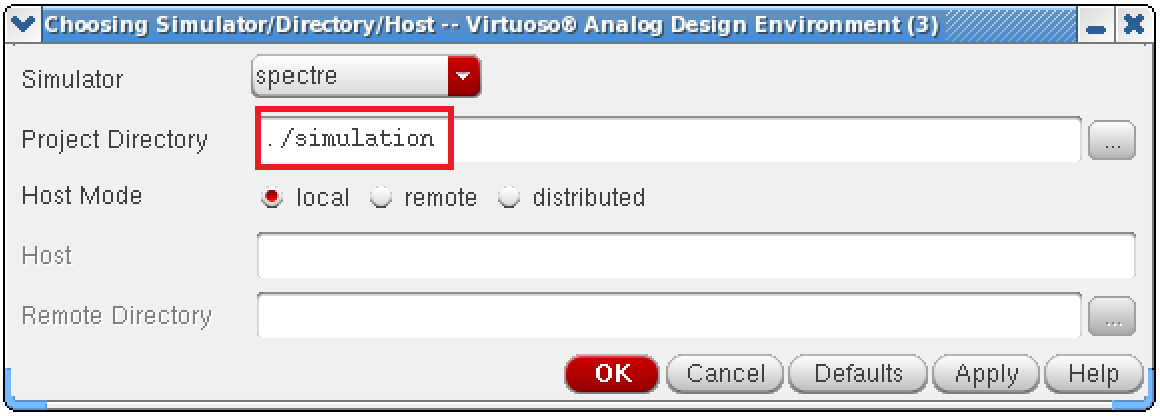
   a. Choose *spectre* for the *Simulator*.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your ~/simulation directory. You may change the default location by editing the settings in your .cdsinit or .cdsenv file. In this workshop, the simulation directory is set to ./simulation.

   c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, please contact your Cadence tools System Administrator for specific setup instructions.

   The completed form appears similar to the one below.

Figure A-109 Choosing Simulator/Director/Host Form

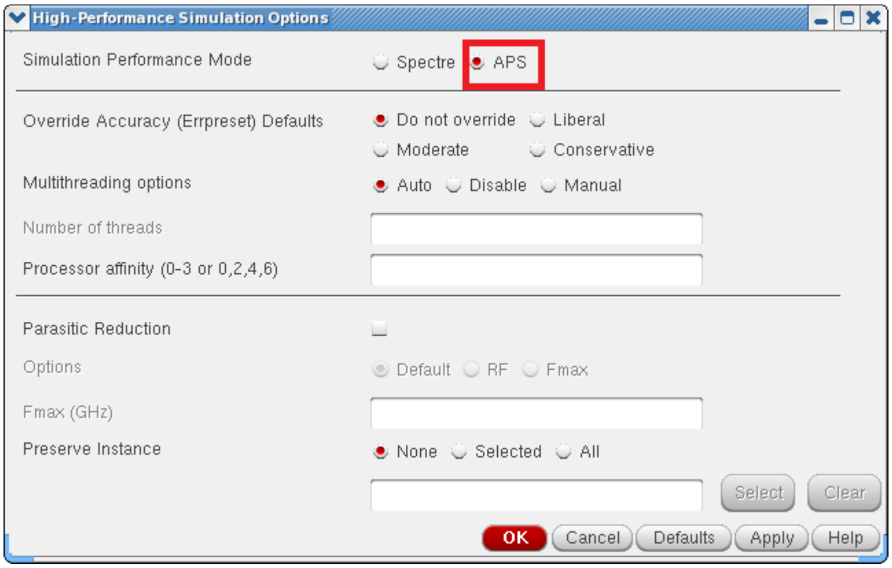


5. Click *OK*.
6. Set up the High Performance Simulation Options.

   In the ADE window, select *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

   In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

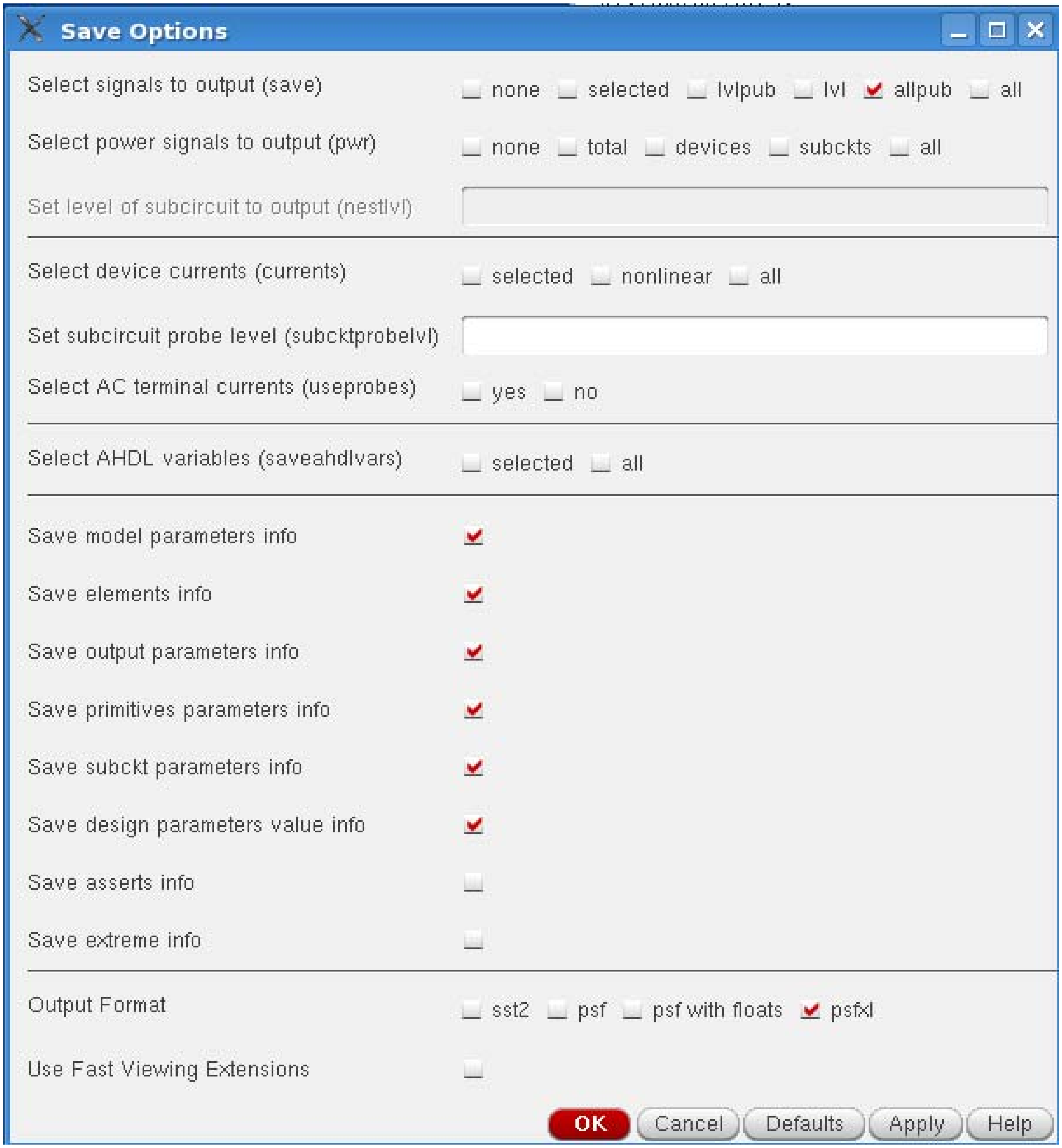
Figure A-110 High Performance Simulation Options Form



7. Click *OK*.
8. In the *Virtuoso Analog Design Environment* window, choose *Outputs – Save All*.

   The *Save Options* form is displayed.
9. In the *Select signals to output(save)* section, make sure that *allpub* is selected.

Figure A-111 Save Options Form



This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

To save the currents, use the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or all if you want to save all the currents in the circuit. When you save currents, more disk space is required for the results file.

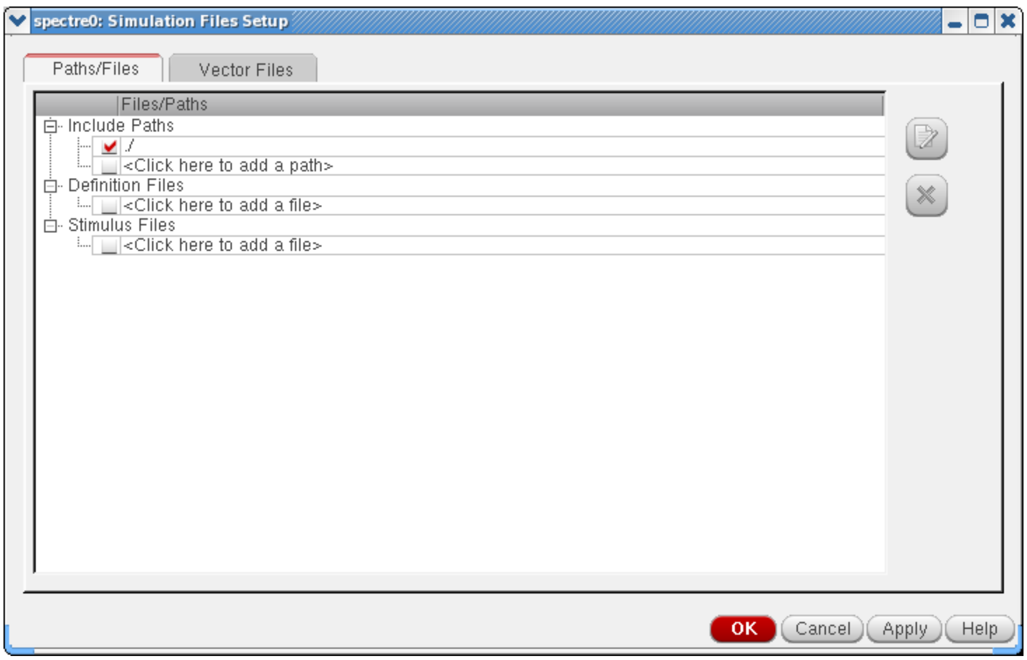
10. Click *OK*.

11. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

The *Simulation Files Setup* form is displayed.

12. In the *Simulation Files Setup* form, enter ./ by clicking in the *Include Paths* section. It should look like the following:

Figure A-112 *Simulation Files Setup* Form



13. Click *OK* to close the *Simulation Files Setup* form.

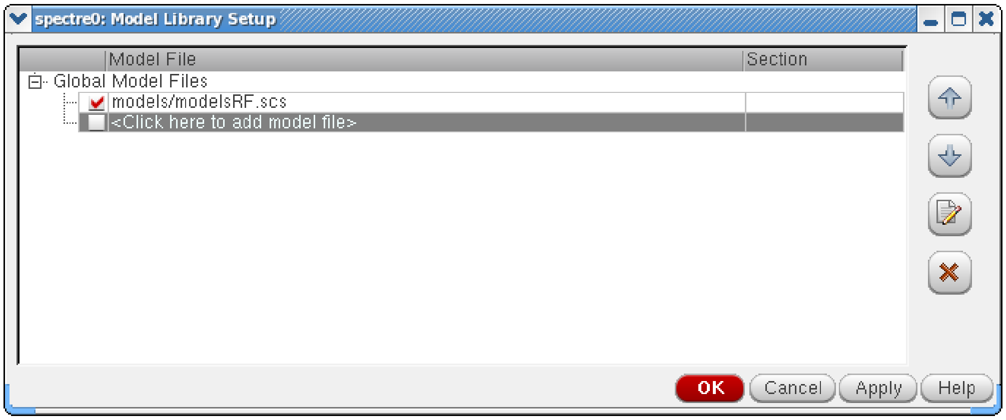
14. In the Virtuoso Analog Design Environment window, choose *Setup – Model Libraries*.

The *Model Library Setup* form is displayed.

15. In the *Model File* field, type the path to the model file including the file name, as shown below.

models/modelsRF.scs

Figure A-113 ***Model Library Setup*** **Form**



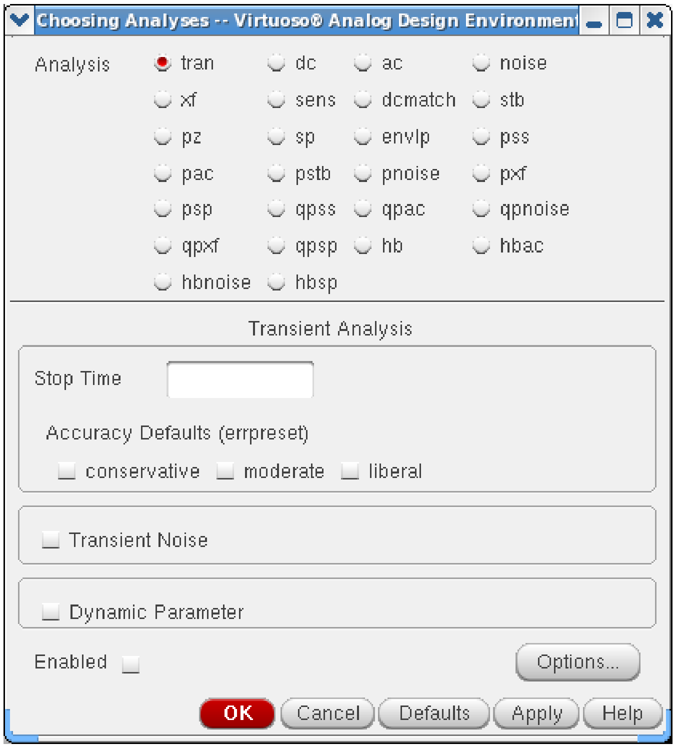
You can also browse to *modelsRF.scs* file.

16. Click *OK*.

17. In the ADE Simulation Window, select *Analyses - Choose.*

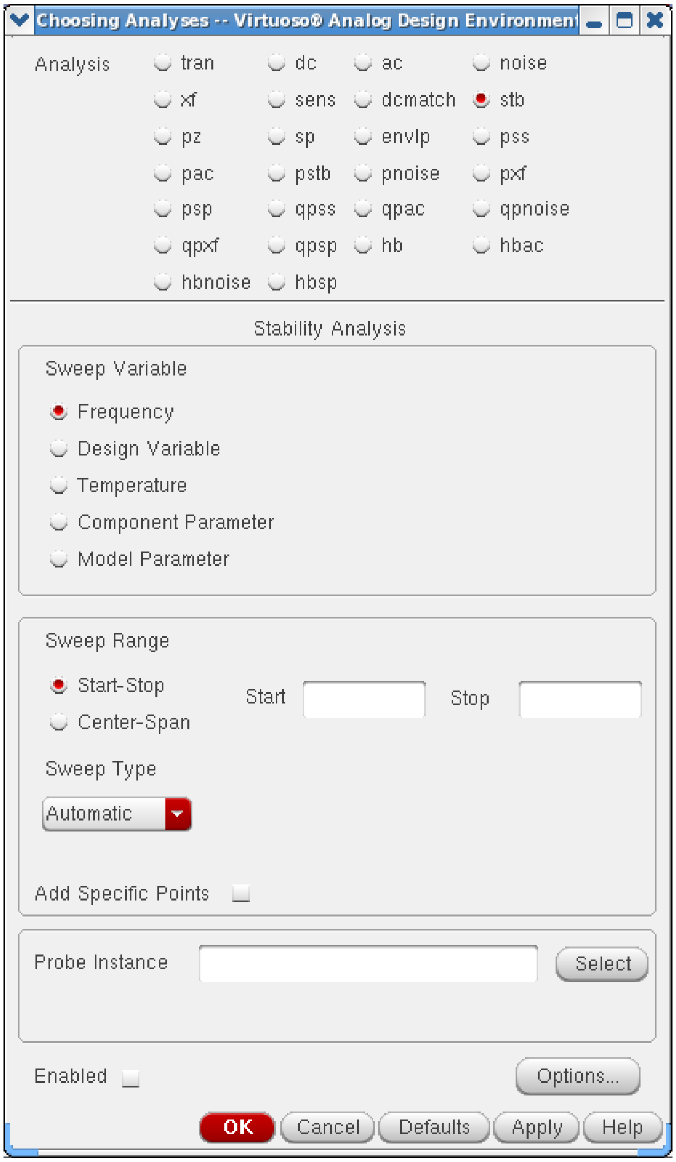
The *Choosing Analyses* form is displayed, as shown below.

Figure A-114 The Choosing Analyses Form



18. In the *Analysis* section, select *stb*.

Figure A-115 The Choosing Analyses Form - stb Analysis Setup



19. Type 1m in the *Start* Field. Note the lower case m. This means the start frequency is 0.001 Hz.

20. Type 1T in the *Stop* field. This will set the stop frequency to 1 THz.

The frequency range is deliberately set very wide in order to detect potential parasitic oscillation modes.

21. Select *Logarithmic* sweep. Type 100 points per decade. The frequency coverage is set to be very dense as this analysis runs quickly.

22. Press the *Tab* key. This moves the cursor to the next field.

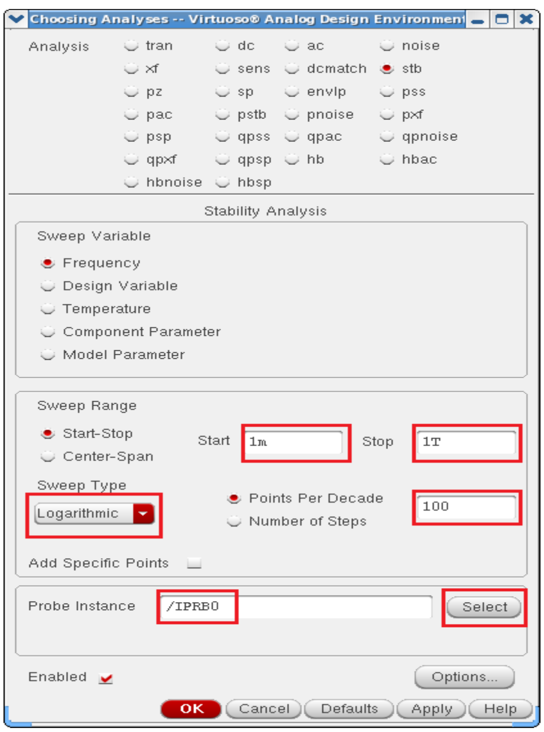
23. Click *Select* to the right of the cursor.

24. In the schematic, click the instance that looks like an analog meter in the center of the circuit. This is an iprobe from analogLib.

Note: To use stability analysis, either an iprobe or a vdc set to 0 Volts needs to be added in series with the feedback path. In this case, an iprobe is used. The current probe and the vdc source both have zero resistance. Because of this the loading in the loop is maintained. In the past, the AC part of the loading had to be broken in order to get a loop gain measurement. For a differential circuit, use diffstbprobe from analogLib.

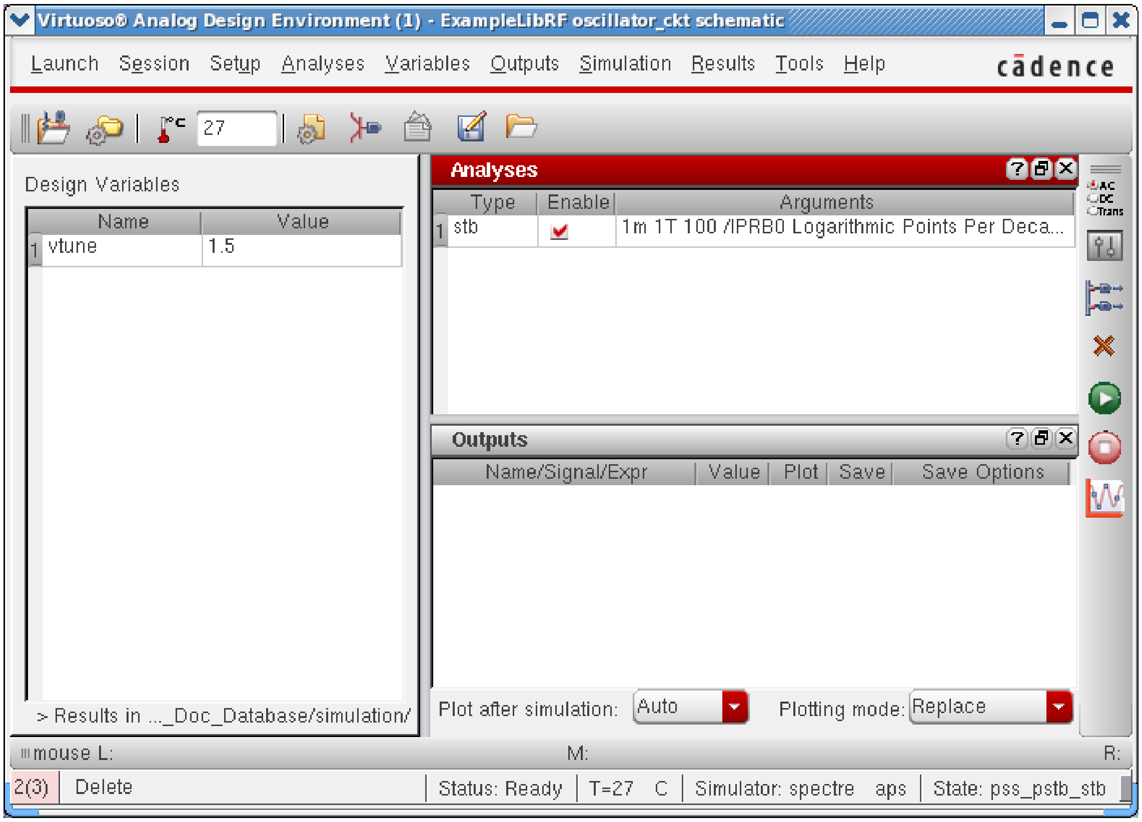
Your Choosing Analyses form should look like this:

Figure A-116 Choosing Analyses Form - *stb* Analysis Setup



Click *Apply* at the bottom of the *Choosing Analyses* form. This will add the *stb* analysis in the *Analyses* section of ADE window, as shown below:

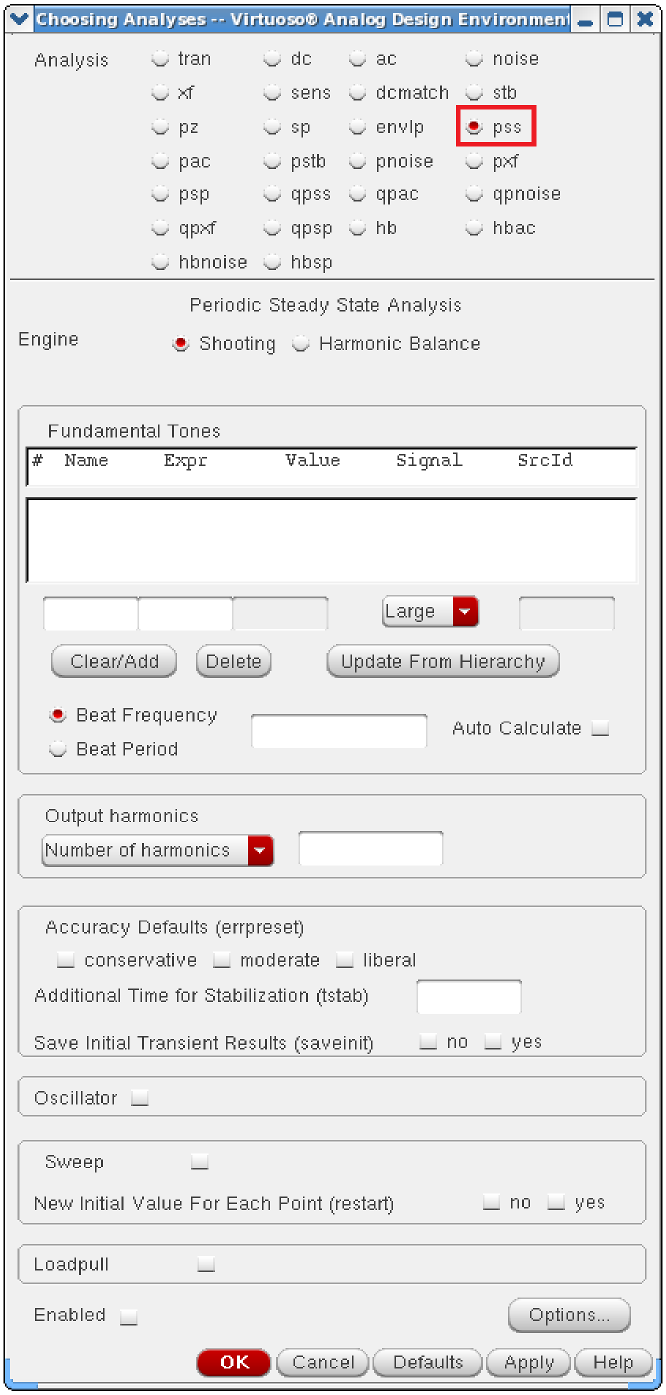
Figure A-117 ADE Simulation Window - *stb* analysis setup



Setting up the PSS Analysis

1. Select *pss* as *Analysis*. The form expands.

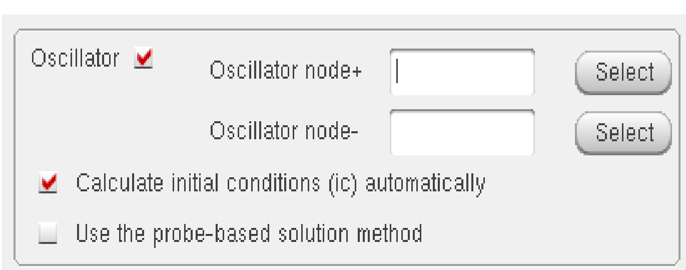
Figure A-118 The Choosing Analyses Form - Setting PSSAnalysis



2. Change the *Engine* from *Shooting* to *Harmonic Balance*.

3. In the *Beat Frequency* field, type 1.9G. The frequency entered here is an approximate frequency of oscillation.
4. Leave Oversample Factor (*oversamplefactor*) as default (that is, blank) which means that Oversample Factor is equal to 1. Since the oscillator has sinusoidal waveforms, an oversample of 1 is appropriate.
5. In the *Number of harmonics* field, type 15.
6. In general, you want to choose a number that is high enough to capture the nonlinearity of the circuit. Start with 10, and run the simulation. Increase by about 50% to 15 and re-run the simulation. If the harmonics donot change appreciably, then 10 is enough. If they change, increase the number by about 50%. Use the smallest number of harmonics for the answer to be stable.
7. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. Conservative is recommended for all the oscillators.
8. Type 5n in the *Additional Time for Stabilization (tstab)* field. *tstab* is typically set to about 10 to 20 periods of the oscillation frequency when the *Calculate initial conditions (ic) automatically* checkbox is selected.
9. Select *yes* for *Save Initial Transient Results*. This will help in visualizing the buildup of the oscillation waveform.
10. Leave the *Harmonic Balance Homotopy Method* as *default* which is *tone*. HbHomotopy (*hbhomotopy*) is one of the HB Convergence Options which helps in the convergence of the simulation.of circuit,for example, in this case it is oscillator.
11. You may refer to Chapter 3, “Frequency Domain Analyses: Harmonic Balance,”for more details on *hbhomotopy*.
12. Select *Oscillator*. This is required for simulating an autonomous circuit.

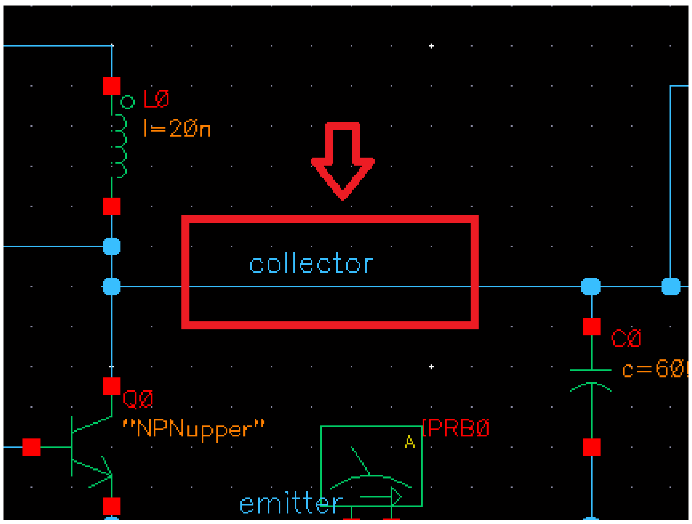
Figure A-119 The Choosing Analyses Form - Oscillator Section



13. In the *Oscillator node* field, click *Select* just to the right of the field. In the schematic, select the *collector* node. This is the net just below the collector label. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it. Leave the *Reference node* blank.

Note that if you have a single-ended oscillator, only specify one node. If the second node, that is, the *Reference node* is left blank, it will be connected to the global ground node automatically. However, if you have a differential oscillator, you need to specify both the nodes.

Figure A-120 Selecting *collector* net on oscillator_ckt schematic



14. If you have an LC oscillator, leave the *Calculate initial conditions (ic) automatically* checkbox selected (this is the default).

Note that *Calculate initial conditions (ic) automatically* is used to start the oscillator. Other methods to start the oscillator include putting a single current pulse into the resonator, setting initial conditions(ic), or ramping up the power at time = zero plus.

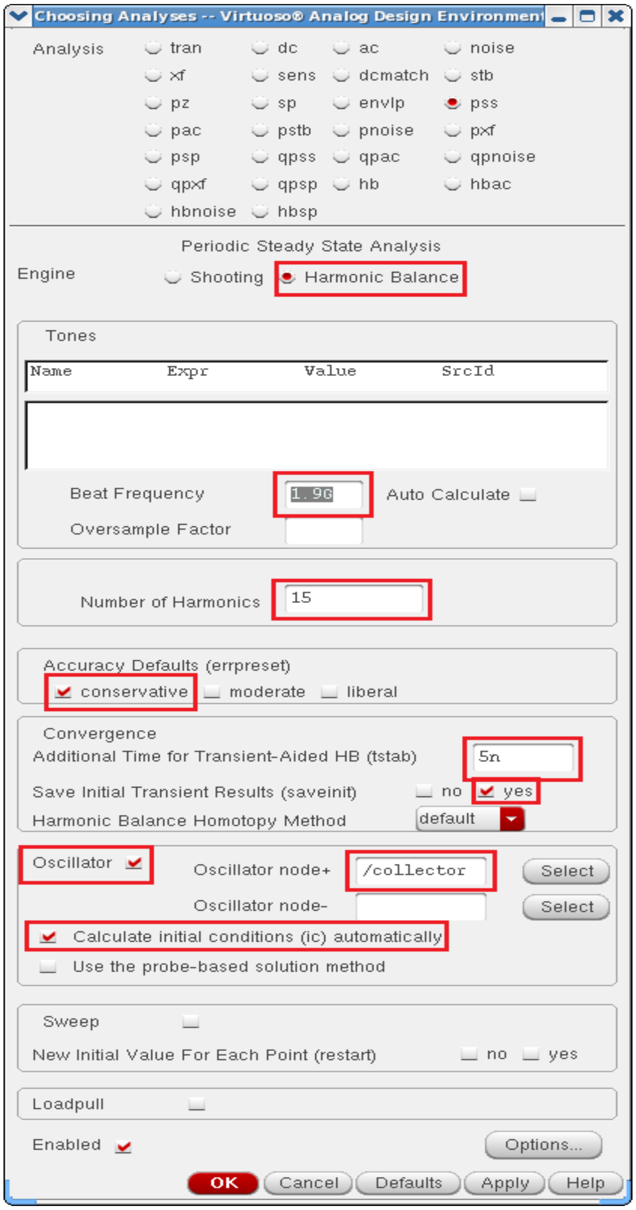
15. Note that by default *Use the probe-based solution method* (*oscmethod*) option is deselected (or unselected). Spectre will use the *onetier* method.

16. When *Setting Use the probe-based solution method* (*oscmethod*) option is selected, it iterates for the frequency solution in the outer loop and the amplitude and phase solution in the inner loop. The probe-based method has better convergence but is computationally intensive.

17. Please refer to *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more details.

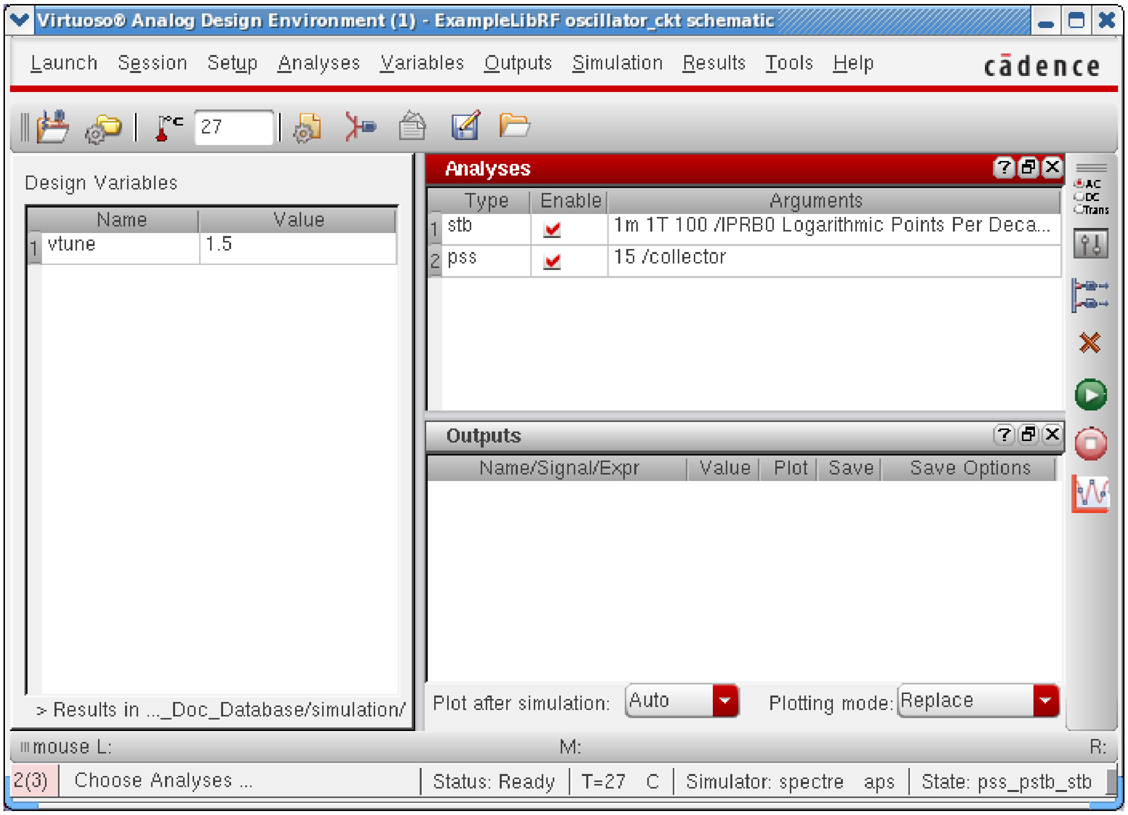
The Choosing Analyses form will look like the following:

Figure A-121 Choosing Analyses Form - PSS-Harmonic Balance Setup



18. Click *Apply* located at the bottom of the form*.* This will add the *pss* analysis in the Analyses section of ADE window along with *stb* analysis, as shown below:

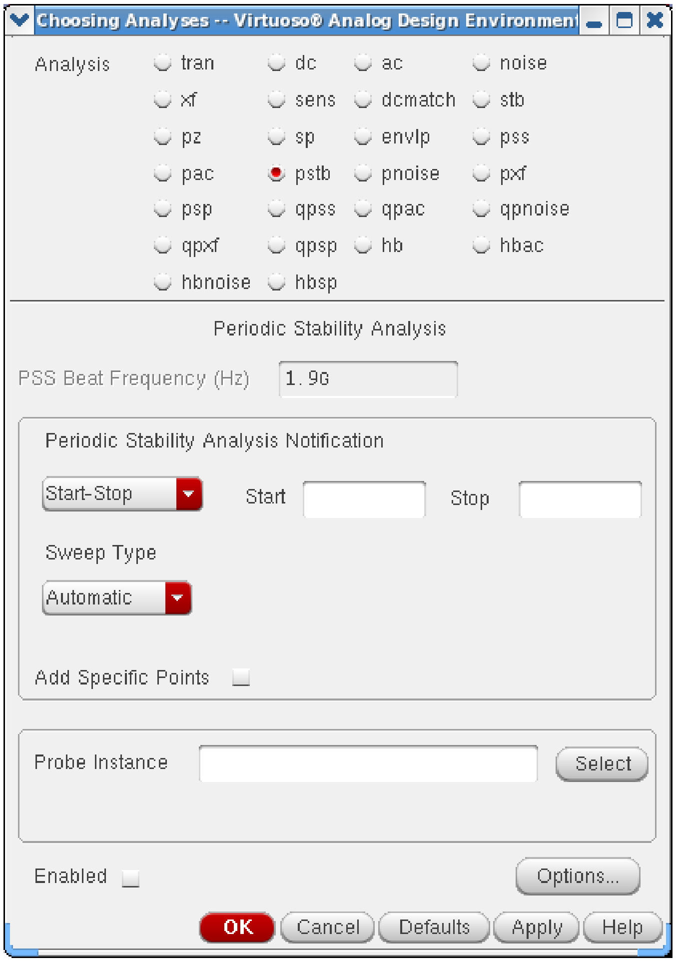
Figure A-122 ADE Simulation Window - *stb* and *pss* Analysis setup



Setting up the Pstb Analysis

1. In the *Choosing Analyses* form, select *pstb*.

Figure A-123 The Choosing Analysis Form - *pstb* Analysis Setup



Set your sweep frequency such that the expected oscillation falls within the sweep range.

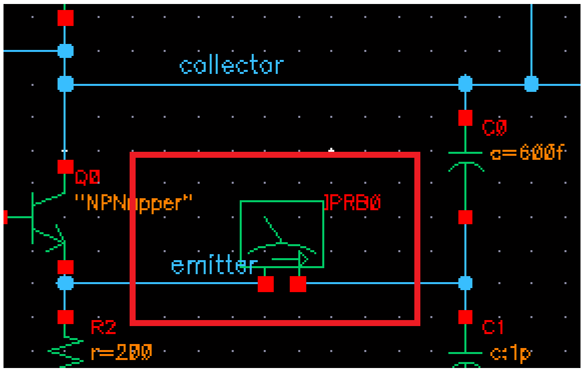
2. Type 1.7G in the *Start* field.
3. Type 2G in the *Stop* field.
4. Set the *Sweep Type* to *Linear*.
5. Select *Number of Steps*.
6. Type 100 in the *Number of Steps* field.

   You need to provide number of steps such that you have enough resolution in frequency. In this case, you have 300MHz/100 = 3MHz step resolution.

7. Click your mouse cursor in the *Probe Instance* field and click *Select* which is just to the right of this field. The *pstb* requires that you place the probe on the feedback loop to identify and characterize the particular loop of interest.

8. In the schematic window, select the *iprobe* in the center of the circuit. It looks like an analog meter. 'iprobe' is required to determine the feedback without breaking the feedback loop.

Figure A-124 Selecting iprobe instance IPRB0 on the schematic

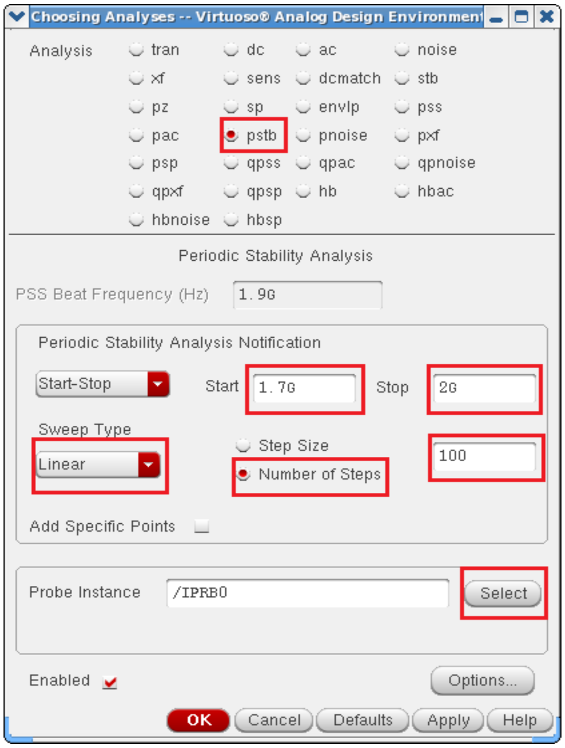


Note: To use pstb analysis, either an iprobe or a vdc set to 0 Volts needs to be added in series with the feedback path. In this case, an iprobe is used. The current probe and the vdc source both have zero resistance. Because of this the loading in the loop is maintained. In the past, the AC part of the loading had to be broken in order to get a loop gain measurement.

Note: For a differential circuit, use the diffstbprobe component from analogLib.

The *Choosing Analyses* form should look like the following.

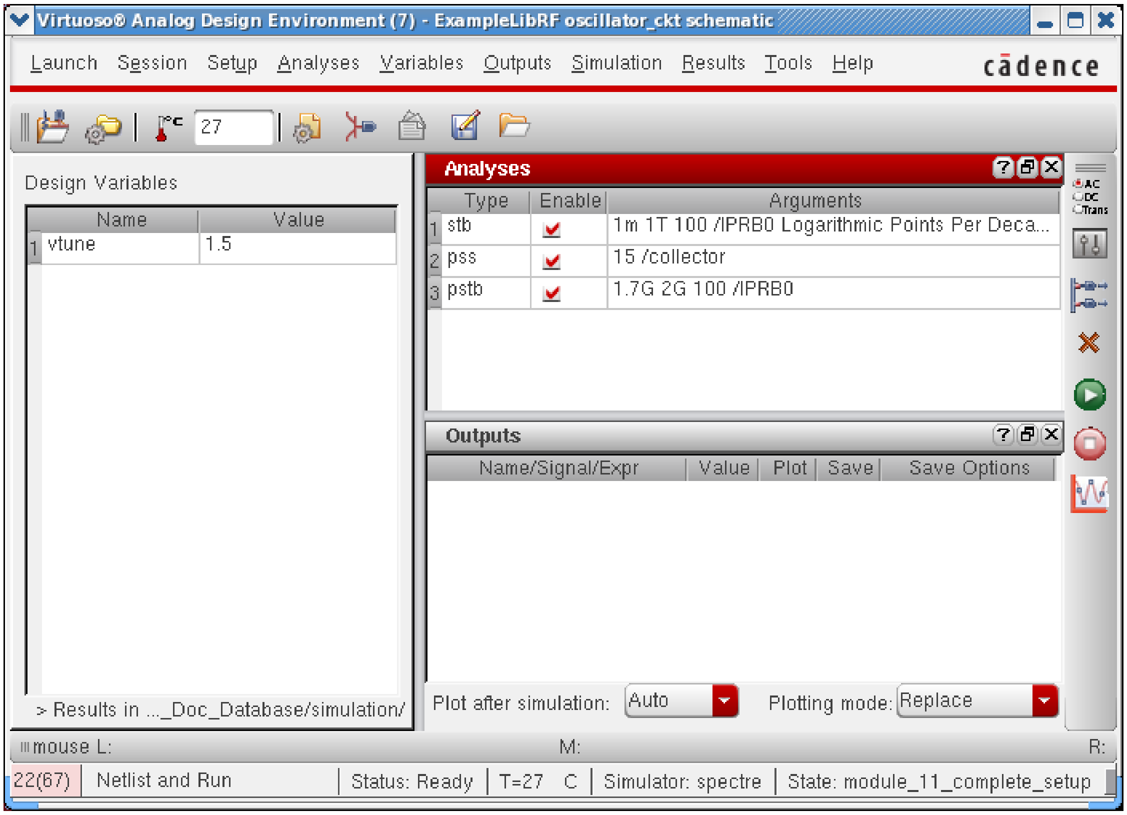
Figure A-125 Choosing Analyses Form - *pstb* Analysis Setup



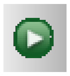
9. Click *OK* button at the bottom of the form to close the *Choosing Analyses* form.

The *pstb* analysis gets added along with *stb* and *pss* analysis in the Analyses section of ADE window, as shown below.

Figure A-126 ADE Simulation Window - *stb*, *pss* and *pstb* setup



Running the PSS, Pstb and stb analysis

Once finished setting up the PSS, Pstb and stb analysis click the green icon  located on the right hand side of the Analog Design Environment window or on the Schematic window to run the simulation.

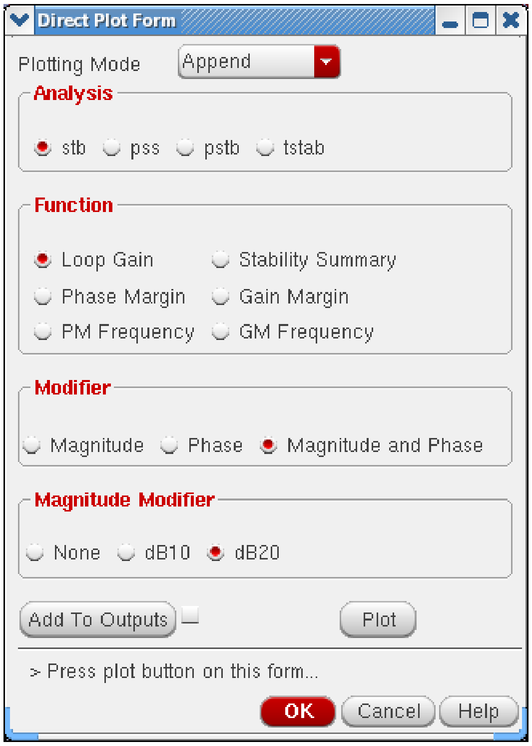
Plotting the results

Plotting the stb analysis results

First plot the Loop Gain magnitude and phase from the stb analysis.

1. In the *Virtuoso Analog Design Environment* window, select *Results - Direct Plot - Main Form*.

Figure A-127 The Direct Plot Form - stb, pss and pstb Analysis

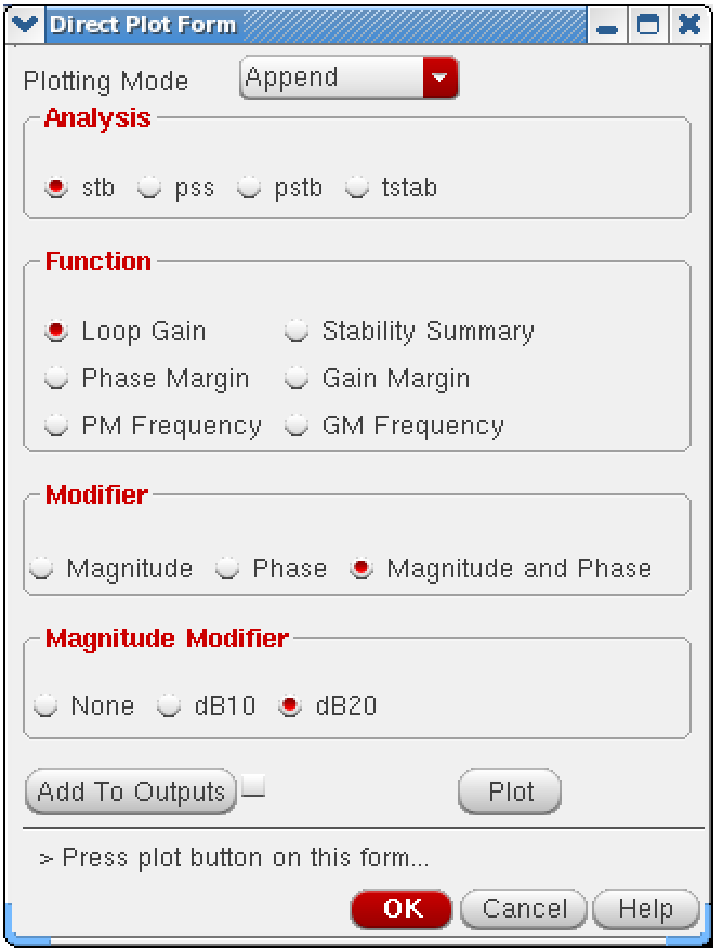


2. Select *stb* as Analysis.

3. Keep the rest of the settings at their default values, that is *Loop Gain* as *Function*, *Magnitude* and *Phase* as *Modifier* and *dB20* as *Magnitude Modifier*. Remember that we are trying to plotthe Loop Gain magnitude and phase plot.

   The Direct Plot Form looks like the following:

Figure A-128 *stb* Analysis Direct Plot Form Setup



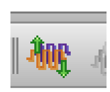
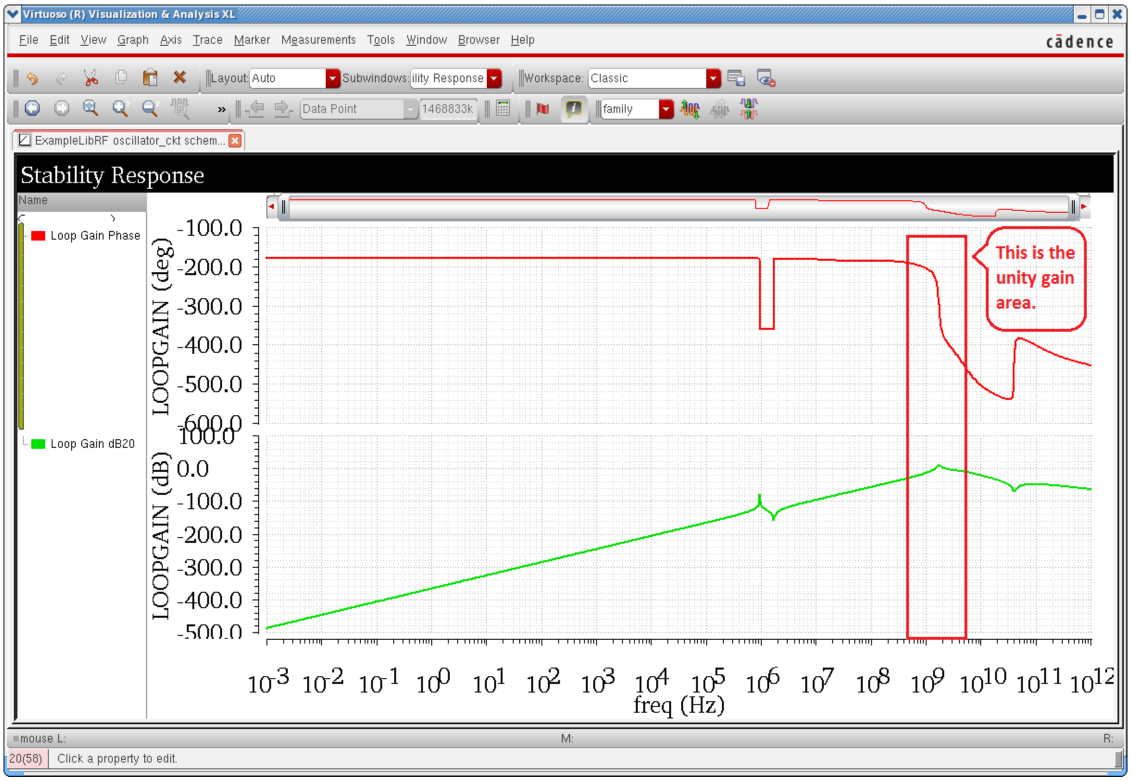
4. Click *Plot*.
5. In the waveform window, click the *Split Current Strip* icon  . This will add the Loop Gain and Phase plots in two different strips in the same graph window, as shown below.

Figure A-129 stb Analysis Ouput Graph Window - Loop Gain Magnitude and Phase Plot



On the top of the graph is a plot of loop gain phase in degrees. On the bottom of the graph is a plot of loop gain magnitude in dB. The area in the red box is the unity gain area.

Note: There is only one place just above 1 GHz where the loop gain approaches unity.

6. Now Zoom in to the unity gain area of the Loop Gain Plot. To do this:

a. Move your mouse cursor over one of the numbers on the X axis..

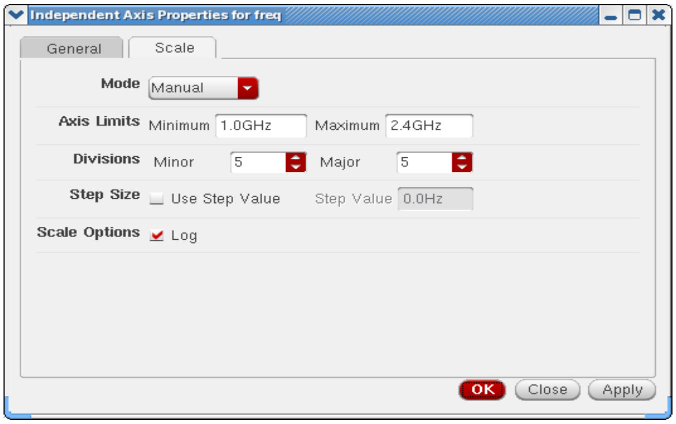
b. Click the right mouse button and select *Axis Properties*.

c. Click the *Scale* tab and set *Scaling* to *Manual*.

d. Type 1G in the *Min* field. This is near to the frequency before the unity gain area starts.

e. Type 2.4G in the *Max* field. This is frequency near to which the unity gain area ends

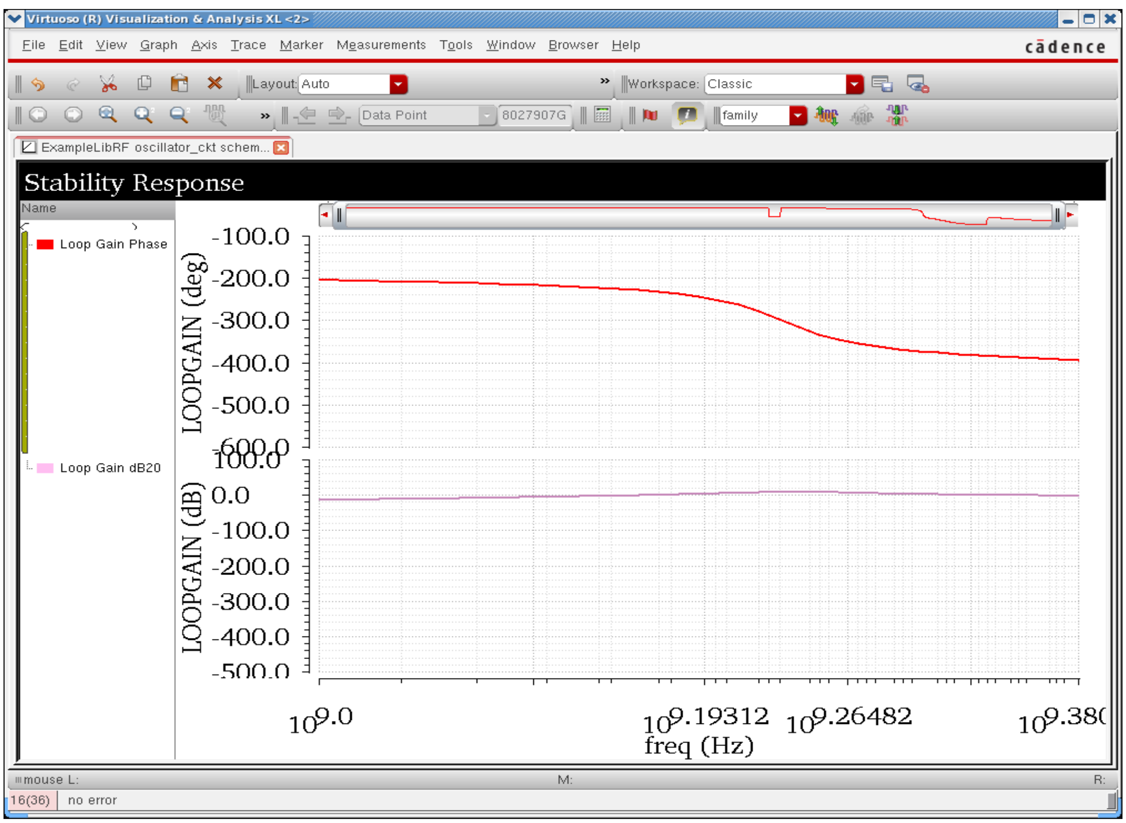
Figure A-130 Graph Window X-Axis Setup



f. Click *OK*.

This will provide the zoomed in plot of Loop Gain's Magnitude and Phase from 1GHz to 2.4GHz frequency range.

Figure A-131 *stb* Analysis Loop Gain Magnitude and Phase Zoomed-in Plot



In traditional control system analysis, an oscillator will oscillate when the loop gain is greater than unity and the phase is 180 degrees. This assumes that there is an inverting summing junction in the feedback loop. In the circuit, we cannot separate out the summing junction, and so the loop gain measurement contains it. In the phase plot, what one looks for is zero or multiples of 360 degrees, not 180 degrees.

1. Place a vertical marker at 360 degrees phase,as follows:

    a. Move the mouse cursor near the 360 degree phase point and type v. This will place a vertical marker.

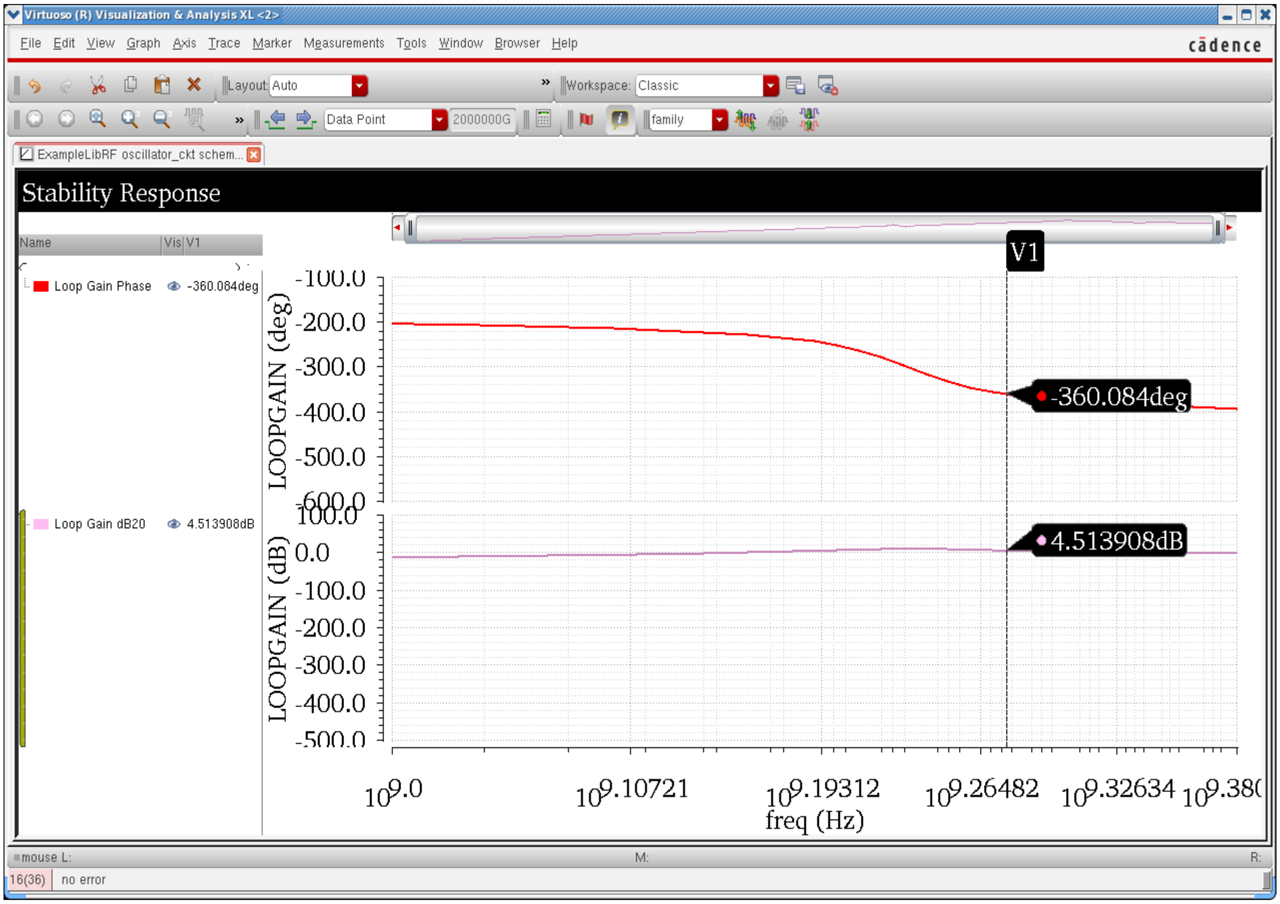
    b. Move your mouse cursor over the vertical marker. The cursor will change shape.

    c. Right-click and select *Intercepts - On*.

    d. Select the vertical marker, click and hold the mouse button, and place it at 360 degrees.

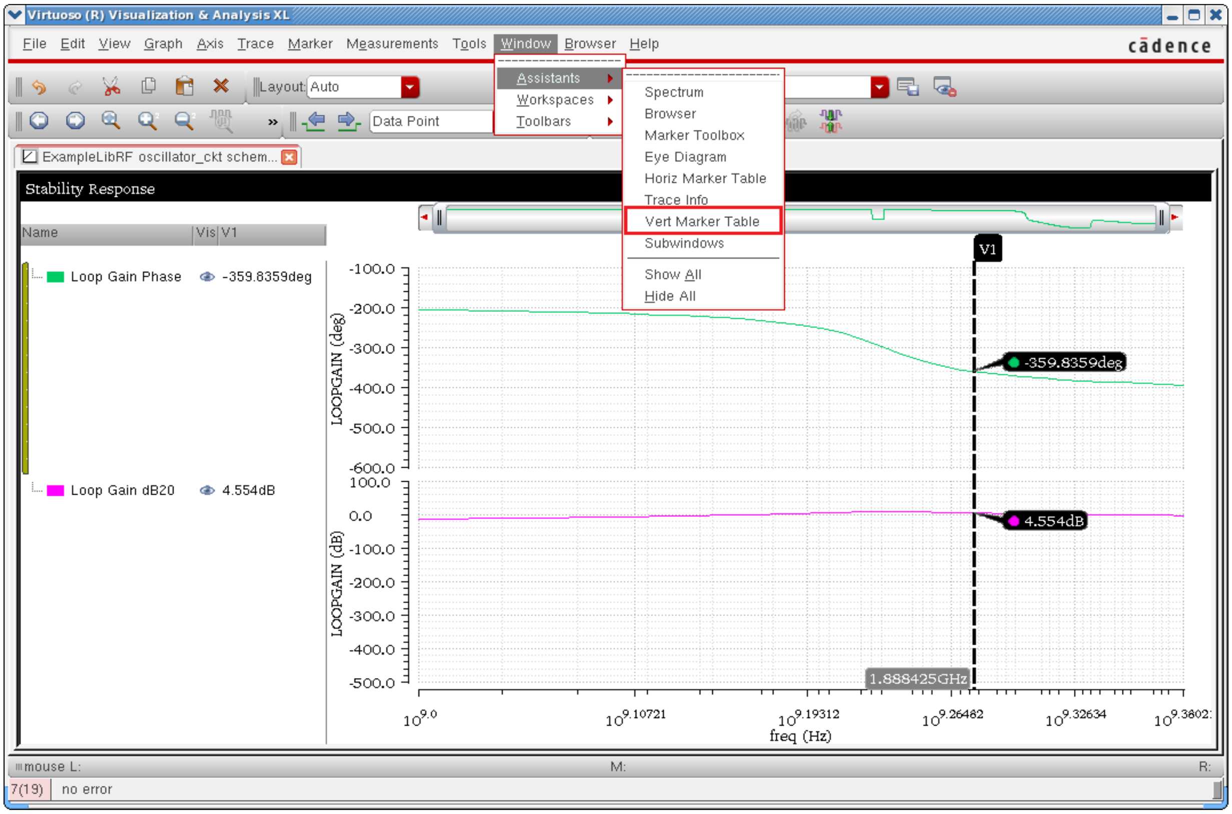
    The Graph Window will look like the following:

Figure A-132 stb Analysis - 360 degree Marker Placement



2. Select *Window - Assistants - Vert Marker Table*.

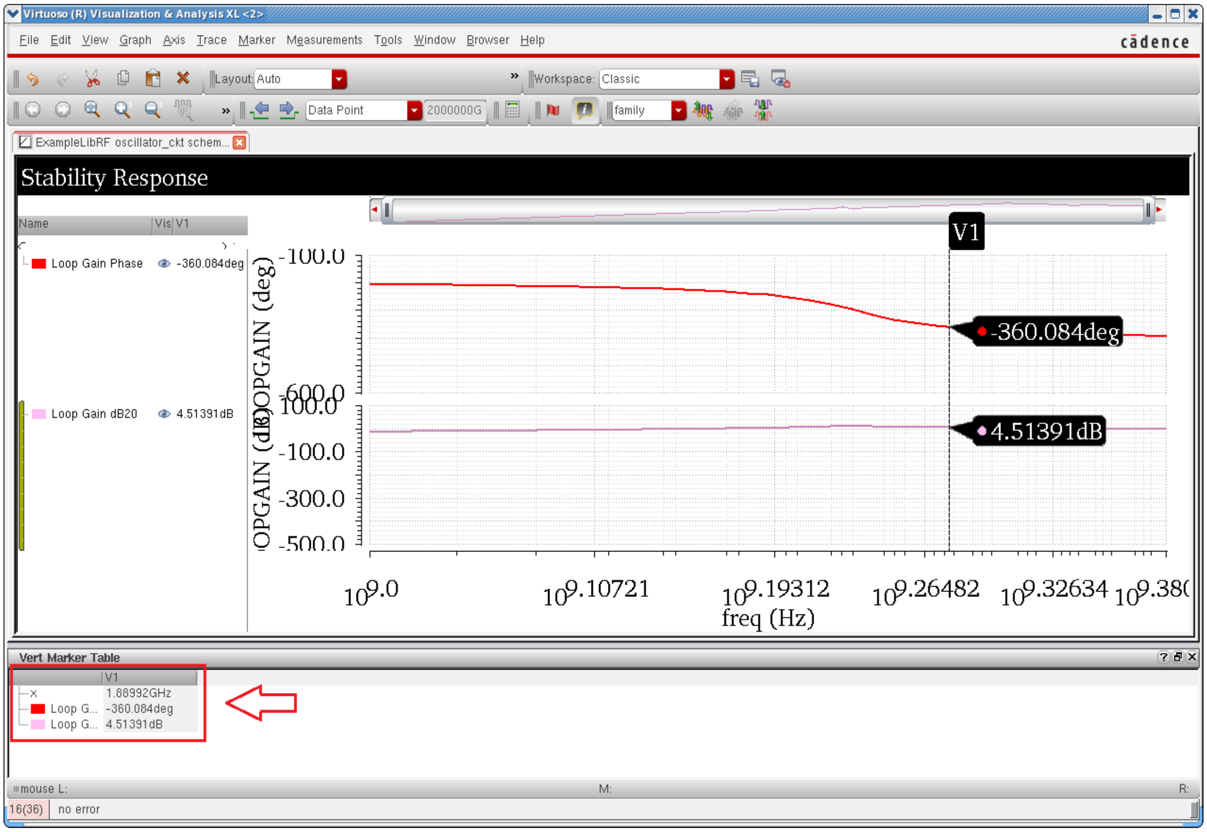
Figure A-133 ViVA XL - Selecting Marker Toolbox



The Marker Table window appears with the data for the phase and gain curve intercepts along with the frequency.

The Graph Window will look like the following:

Figure A-134 Oscillation frequency determination from *stb* Loop Gain Plot



Note that the loop gain is just over 4.5 dB on the Loop Gain dB20 curve. (Greater than unity gain.) The oscillator should oscillate.

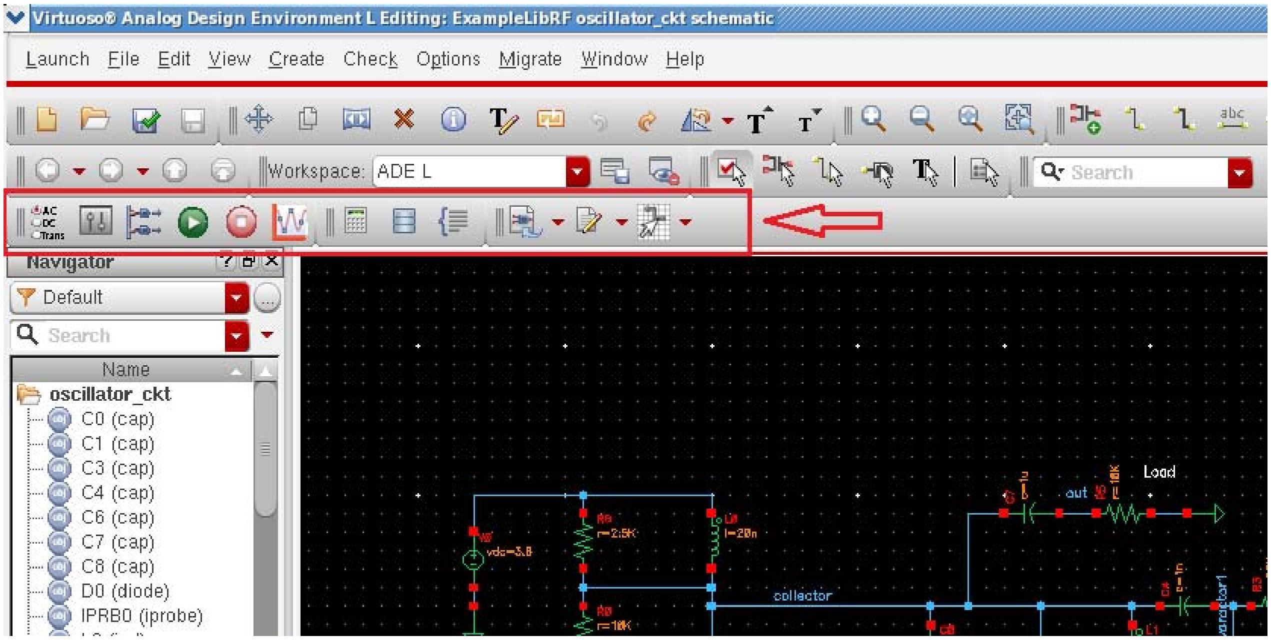
Note that the frequency of oscillation based on linear *stb* analysis is 1.889GHz. You'll refer to this later.

3. In the *Direct Plot Form*, click *Cancel*. In the ViVA window, select *File - Close All Windows*.

Plotting up the Pstb Analysis Results

Next plot the *pstb* Loop Gain magnitude and phase. This time you will plot the results from the Schematic Window. Once you open an ADE L session/window from Virtuoso Schematic Editor L Window, the simulation menu icons bar gets added to this Schematic Editor Window, as shown in the figure below:

Figure A-135 ADE L menu in VSE L Window



You may refer to *Virtuoso Analog Design Environment L User Guide* for more information regarding these simulation menu icons.

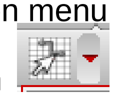
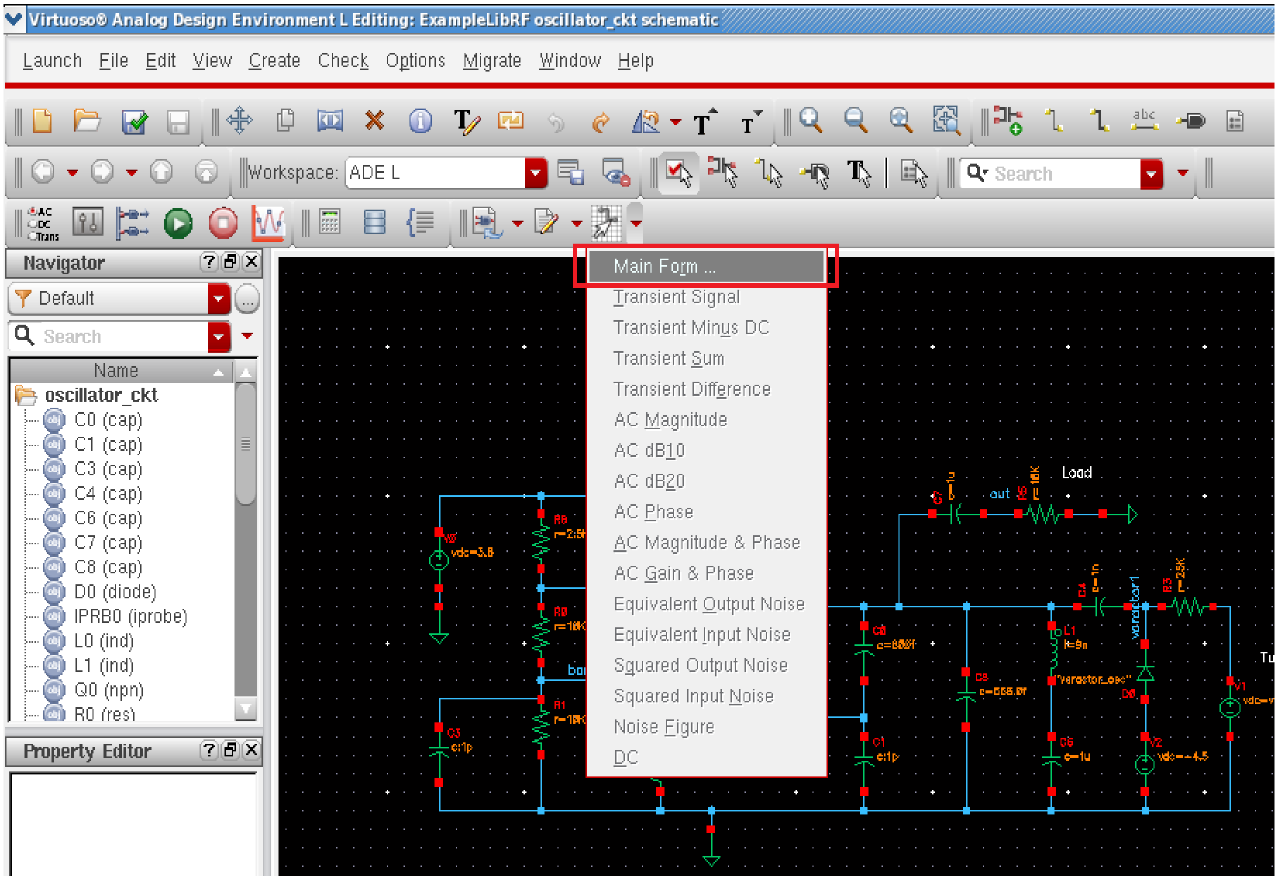
Click the red arrow  in the icon  located at the right. This will open the Direct Plot menu options. Select *Main Menu*, as shown below:

Figure A-136 Running Simulation from VSE L Window

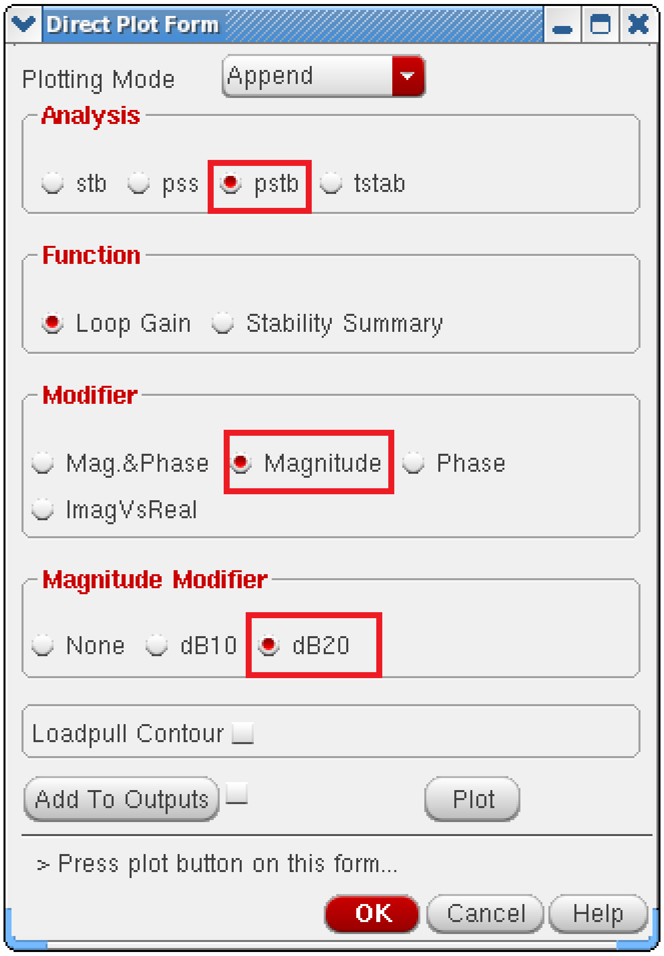


The *Direct Plot Form* is displayed.

1. In the *Direct Plot Form*, select *pstb* in the *Analysis* section.
2. Select *Loop Gain* in the *Function* section.
3. Select *Magnitude* in the *Modifier* section.
4. Select *dB20* in the *Magnitude Modifier* section.

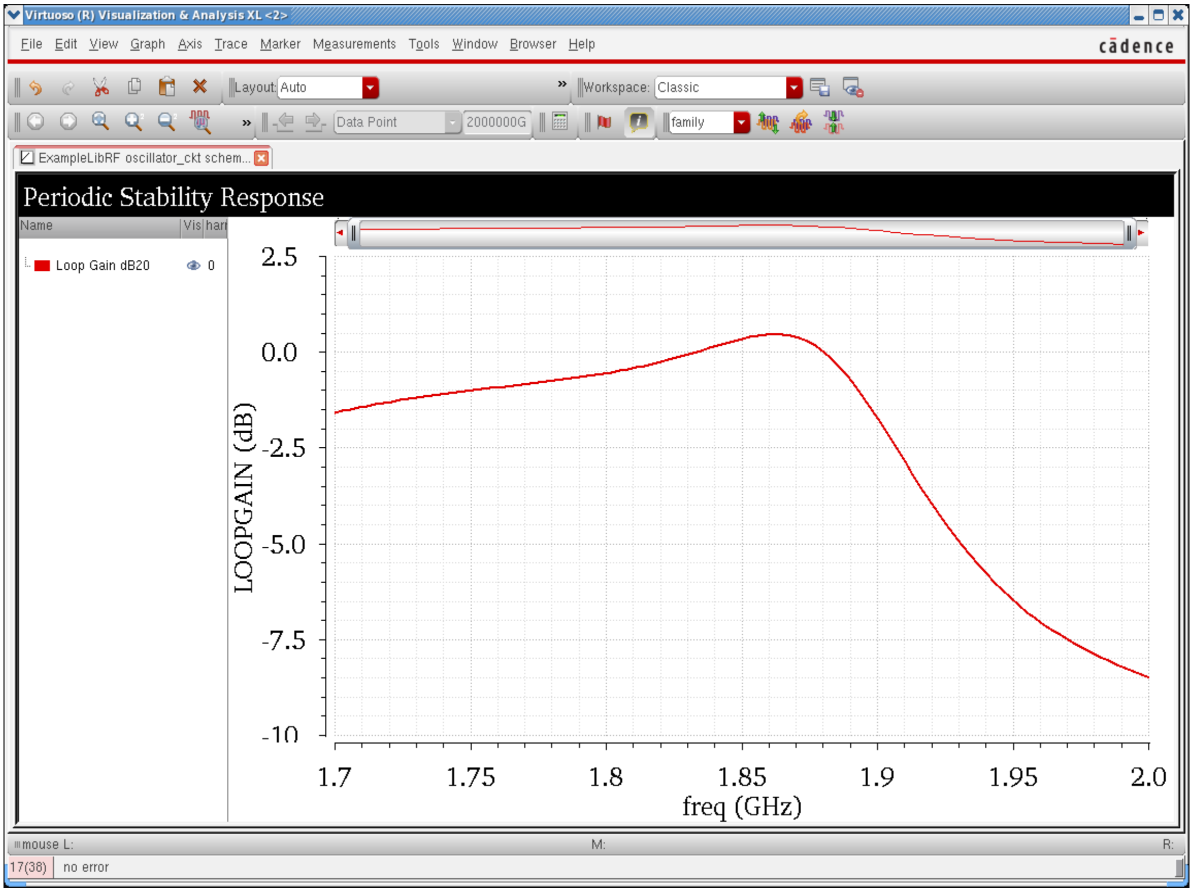
   The *Direct Plot Form* looks like the following:

Figure A-137 Pstb Direct Plot Loop Gain Magnitude Setup



5. Click *Plot* . The Plot Window will be displayed, as shown below.

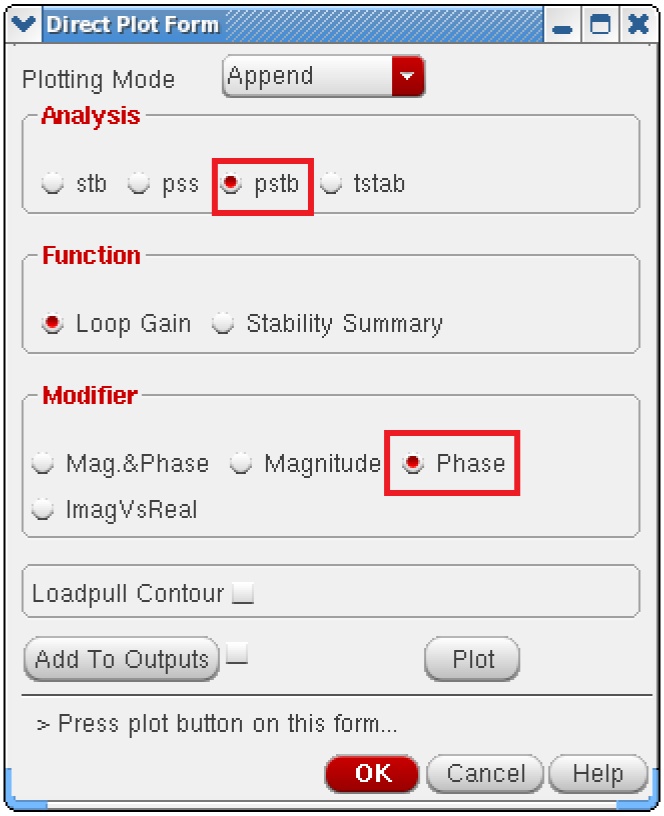
Figure A-138 Pstb Loop Gain Magnitude Plot



6. Next, plot the pstb Loog Gain phase. Select *Phase* inthe *Modifier* section with *Loop Gain* selected inthe *Function* section.

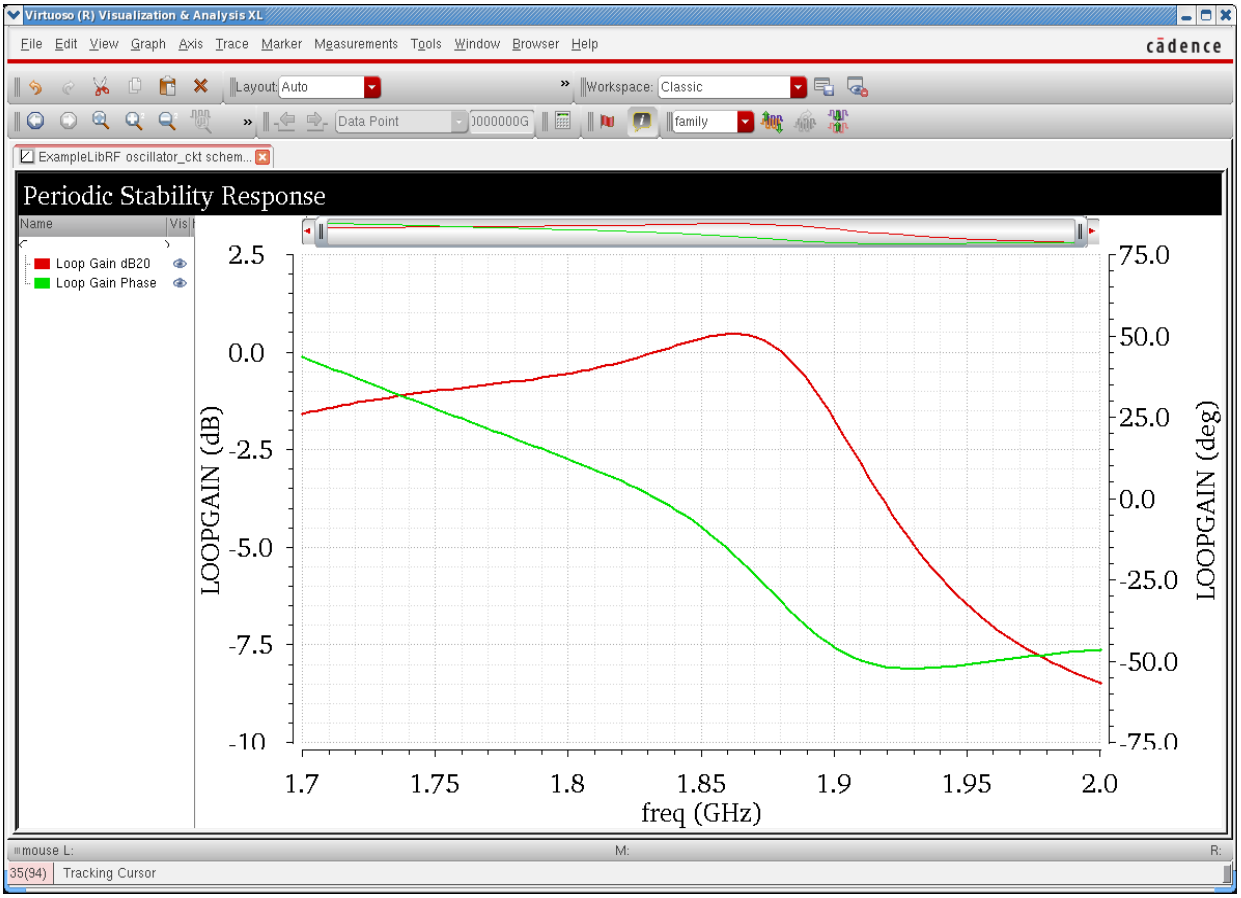
The *Direct Plot Form* looks like the following:

Figure A-139 Pstb Direct Plot Loop Gain Phase Setup



7. Click *Plot* . The Plot Window will look like the following:

Figure A-140 ***pstb*** **Analysis Loop Gain Magnitude and Phase Plot**



Note: Loop Gain's Magnitude and Phase are plotted separately as when *Mag&Phase* is selected, there is no dB20 modifier for the loop gain.

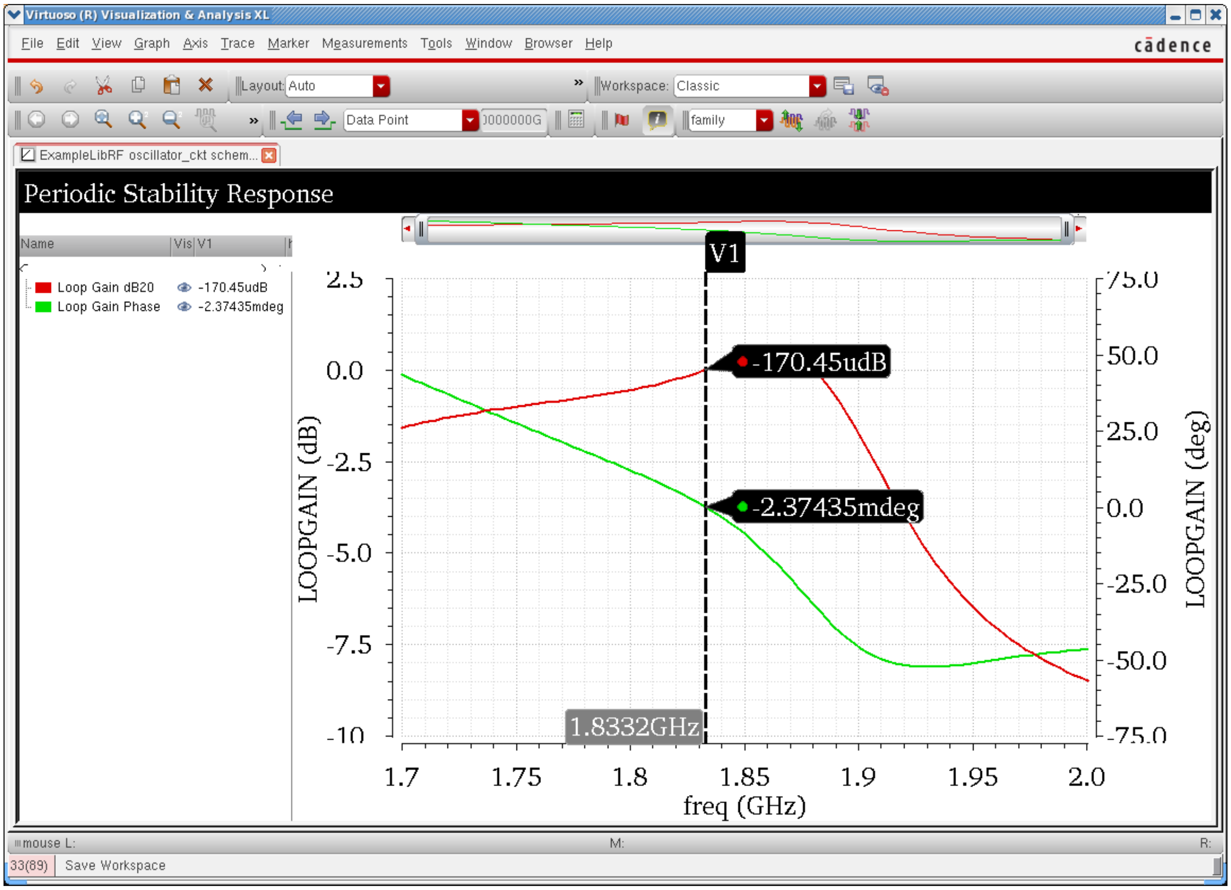
8. Next, you need to place a vertical marker at zero degrees on the plot and get the marker table. To do this:

 a. Type v to place a vertical marker.

 b. Select the vertical marker, click and hold the left mouse button, and place it at near zero degrees.

The Plot will look like the following:

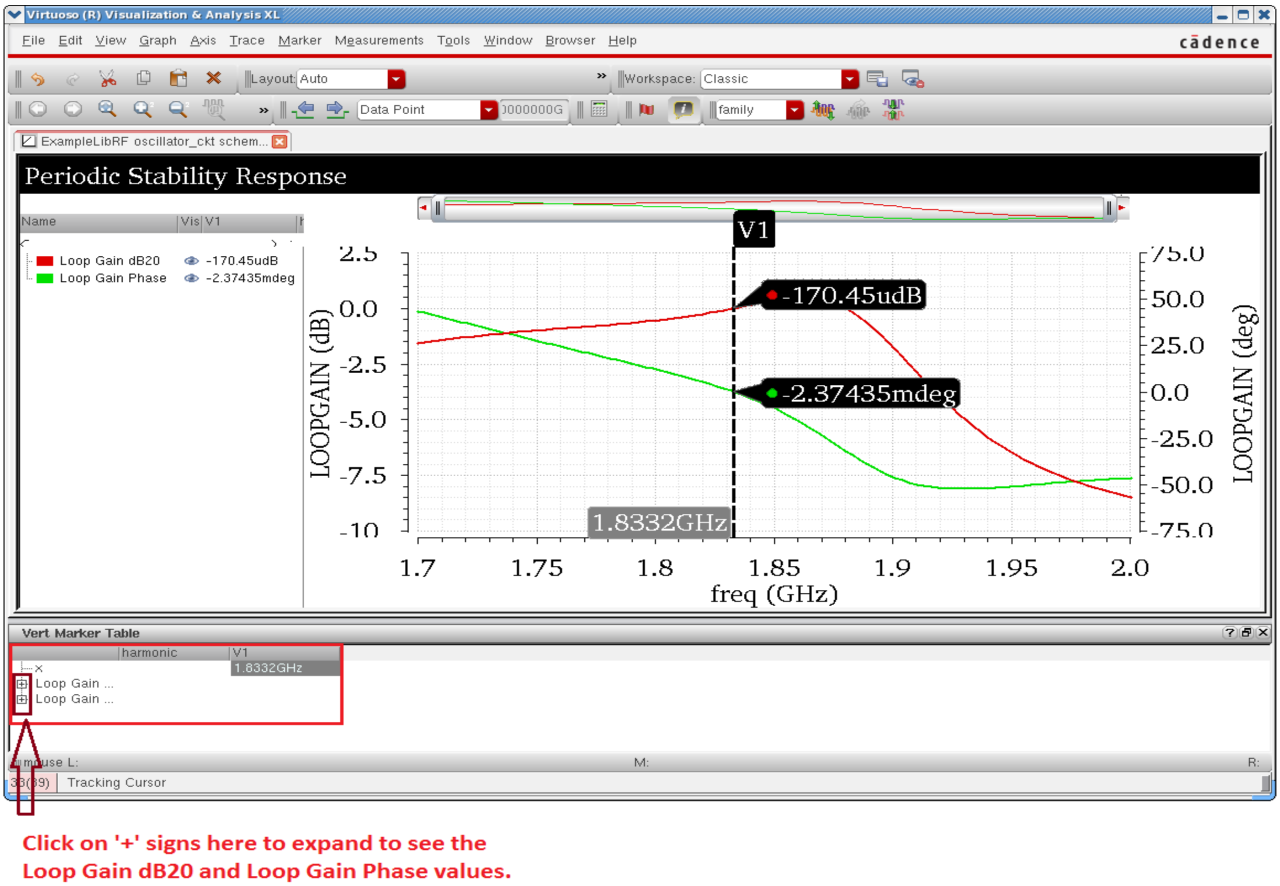
Figure A-141 pstb Loop Gain magnitude and phase plot with vertical marker



9. Select *Window - Assistants - Vert Marker Table*.

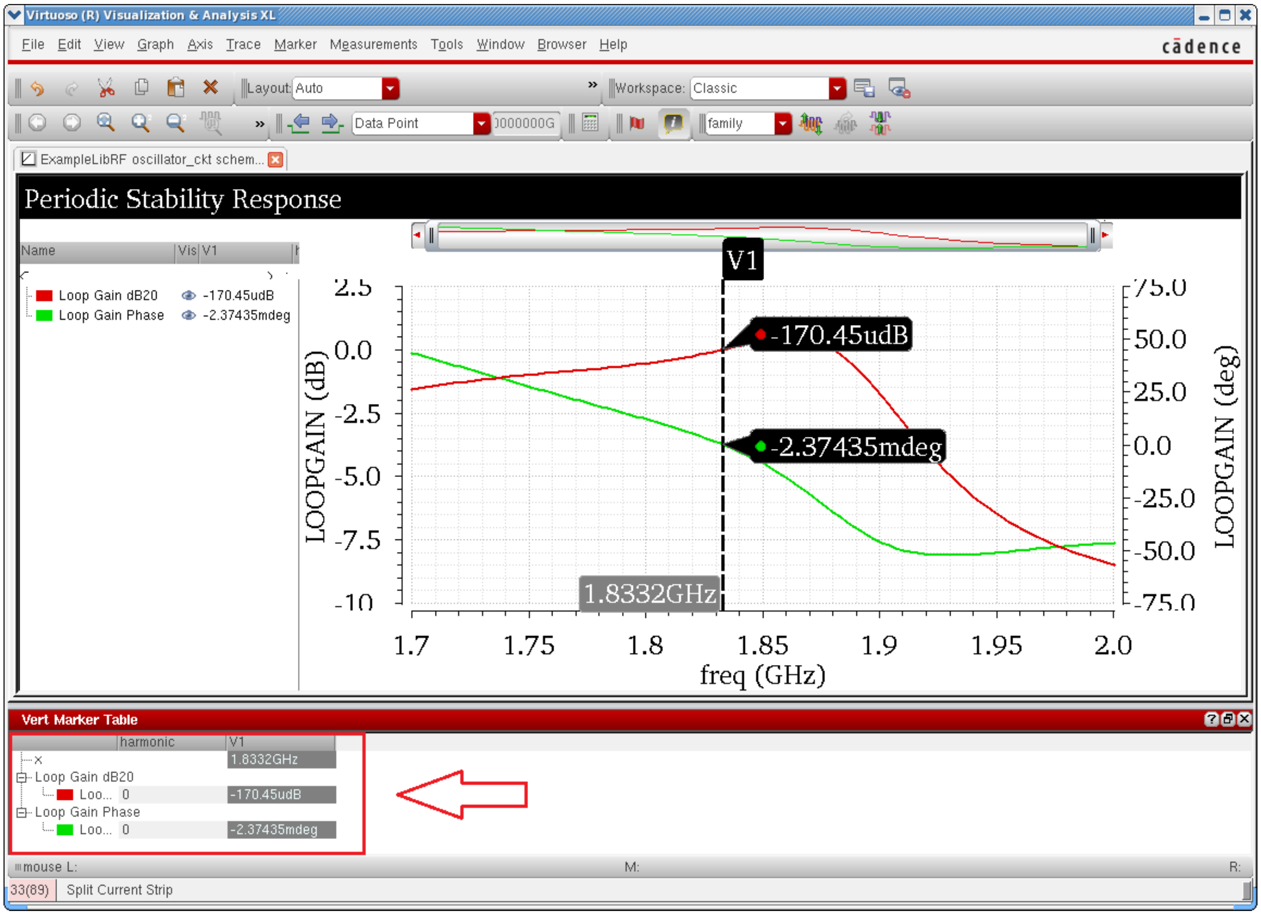
   The Marker Table window isdisplayed with the data for the phase and gain curve intercepts along with the frequency.

Figure A-142 pstb Analysis Loop Gain Plot with unexapnded Verticle Marker Table



10. Note that in the plot window, initially only frequency value is visible and the Loop Gain and Phase values are blank. Click the + sign, locatdat the left of the Loop Gain dB20 & Loop Gain Phase entry to expand them and see their respective values, as shown in the figure below.

Figure A-143 pstb Analysis Loop Gain Plot with Vertical Marker Table



Note that the Loop Gain magnitude is very near unity gain.

Also note that the oscillation frequency determined from *pstb* analysis is 1.833GHz, slightly different than the oscillation frequency 1.889GHz determined from linear stability analysis, that is,*stb*. Linearizing about a periodically time-varying operating point in *pstb* analysis allowed the stability evaluation to include the effect of the time-varying operating point as mentioned earlier.This has changed the oscillating frequency a little bit compared to the one determined using *stb*.

Next, PSS Analysis Results will be plotted. You would note that how well pss results agree with the pstb results.

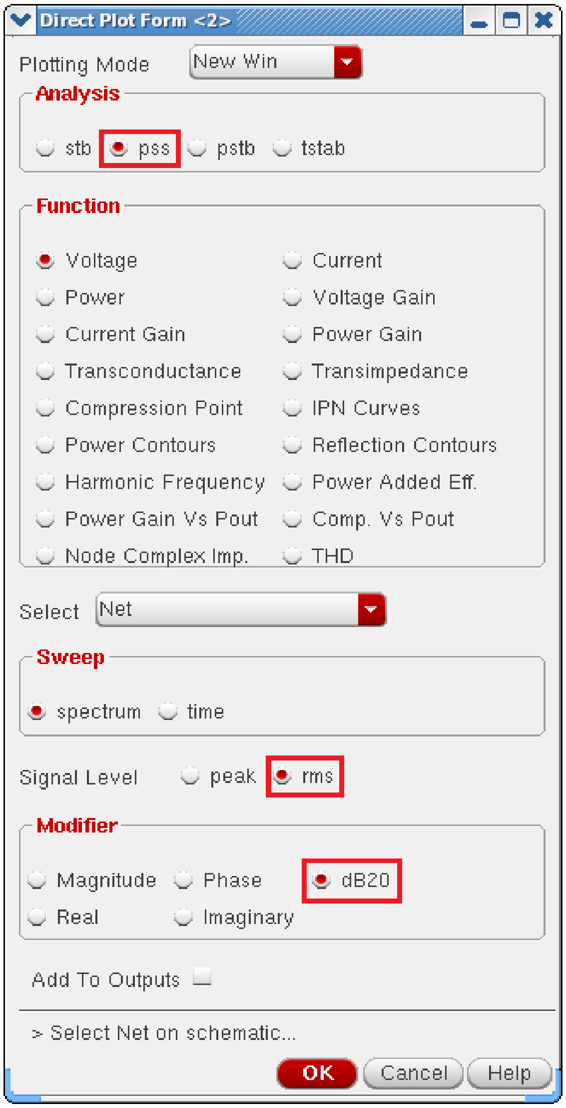
Plotting up the PSS Analysis Results

Next plot the oscillator spectral content -

1. In the Direct Plot Form window, set the *Plotting Mode* to *New Win*.

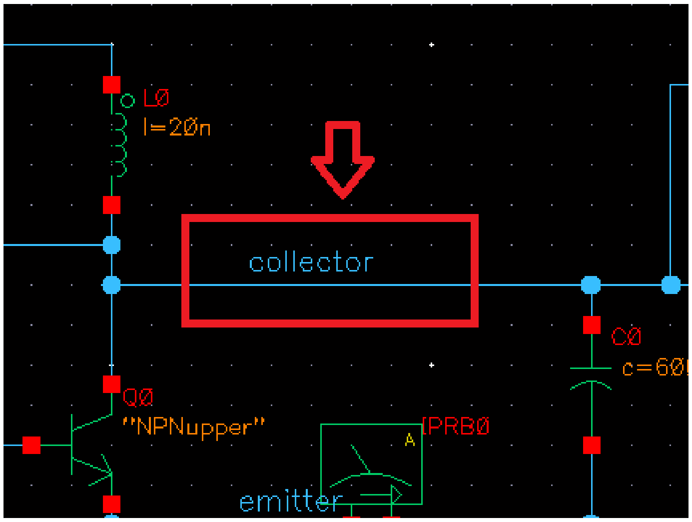
2. Select *pss* as *Analysis*.
3. Leave *Function* as *Voltage* which is set by default.
4. Select *Net* in the center of the form. (This is the default. You can also select differential nets).
5. Select Sweep as spectrum. (This is the default)
6. Select *rms* as Signal Level (the default is *peak*).
7. Select *dB20* as Modifier (the default is *Magnitude*).
8. Your *Direct Plot Form* should like the following:

Figure A-144 PSS Analysis Direct Plot Form Setup



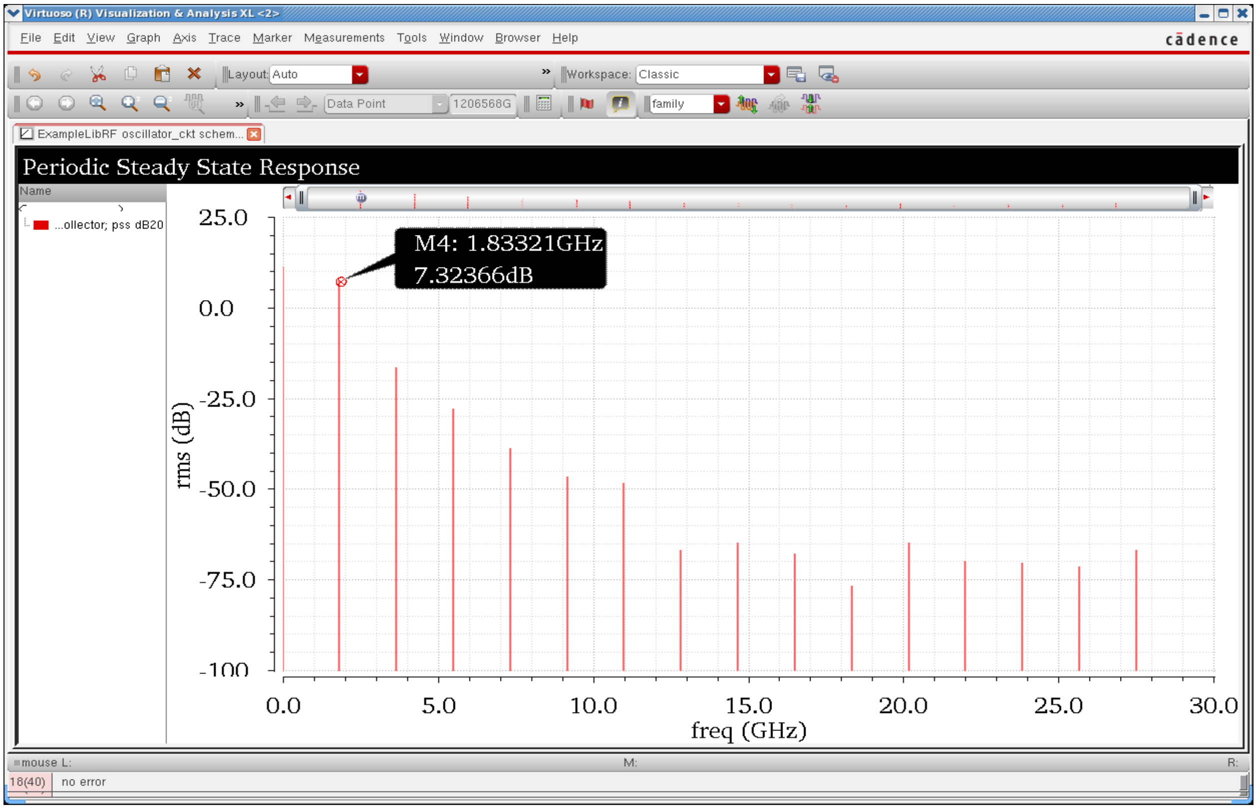
9. Select the *collector* net in the Schematic. It is located just below the *collector* label.

Figure A-145 Selecting *collector* net on oscillator_ckt schematic



The waveform window is displayed, as shown below.

Figure A-146 PSS Analysis output Graph Window - Voltage Spectrum Plot



10. In the waveform window, position your cursor near the first harmonic, and press the *m* key. Here *m* is the bindkey to place a trace marker on the graph. The first harmonic is chosen as this is the frequency oscillator is designed for.

Note that this frequency is 1.833G. This is the frequency of oscillation.

Note that the oscillation frequency determined from pstb analysis agrees very well with the pss-hb analysis.

11. In the *Direct Plot Form*, click *Cancel*. In the ViVA window, choose *File - Close All Windows*.

12. Close the Analog Design Environment windowby selecting *Session - Quit*.

13. Click *No* in the *Save State* Window.

14. In the Schematic window, choose *File - Close*.

In this section, the Oscillator Loop Gain magnitude and phase measurements were done using *stb*, *pss*-hb, and *pstb* analysis. In addition, the oscillation frequency was determined using these measurements.

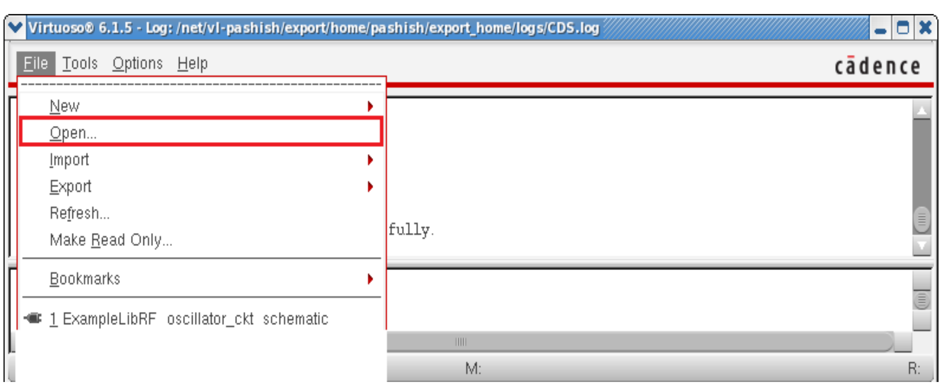
Next,the phase noise measurement which is one of the key measurement in oscillator design will be done. Also, the Noise Summary Table will be obtained.

Phase Noise Measurement and Noise Summary Table

Opening the Oscillator Circuit in the Schematic Window

1. In the Command Interpreter Window (CIW), choose *File - Open*.

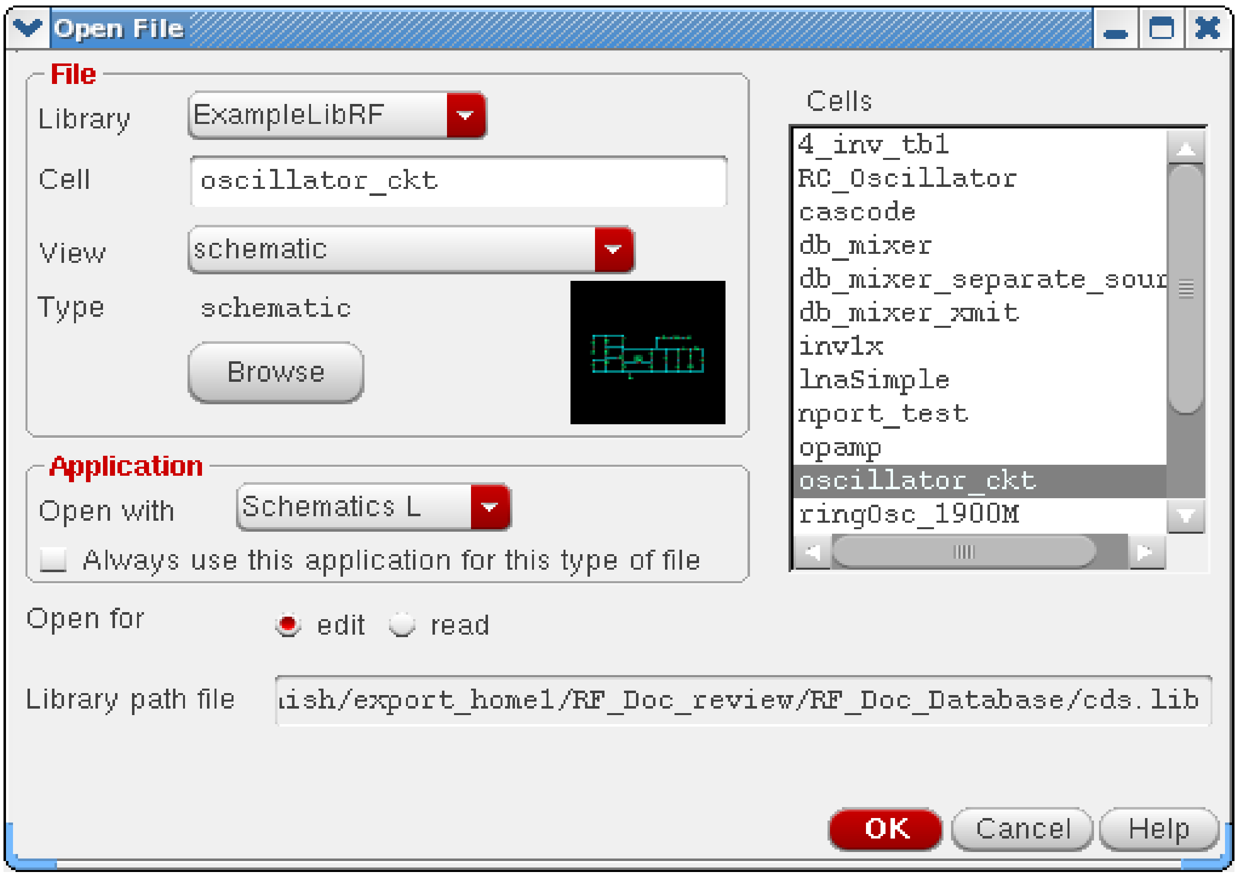
Figure A-147 Virtuoso CIW Window - Opening Cellview



The *Open File* form is displayed.

2. Select *ExampleLibRF* from the *Library* drop-down list.
3. In the *Cells* field, type *oscillator_ckt*.
4. Select *schematic* fromthe *View* drop-down list.
5. In the *Application* section, select *Schematic-L* from the *Open With* drop-downlist.
6. Leave *Open For* to *Edit* (which is set by default).

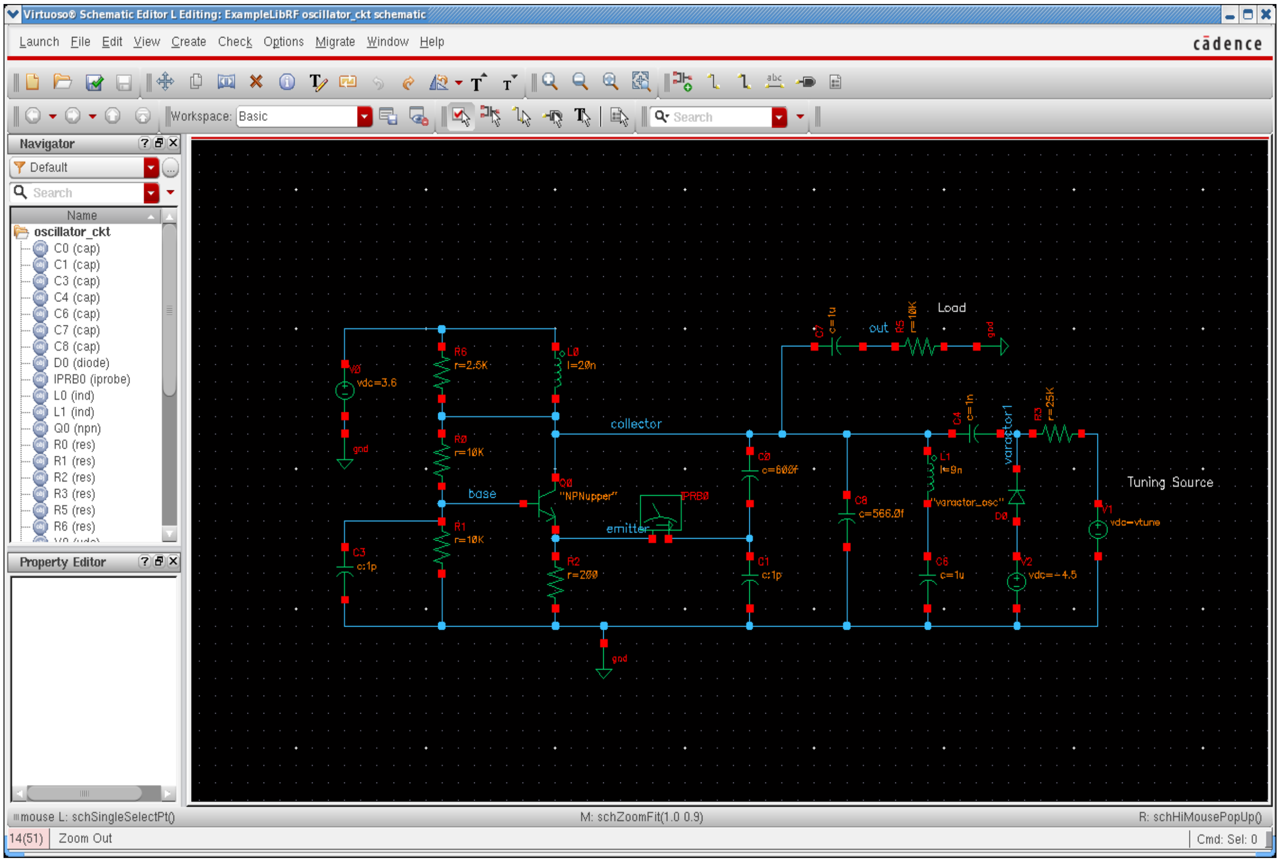
Figure A-148 Open File Form to open the oscillator_ckt cell's Schematic View



7. Once all the setup is done, click *OK* to close the *Open File* form.

   This will open the oscillator_ckt schematic in Virtuoso Schematic Editor L window, as shown below.

Figure A-149 Oscillator Schematic opened in Virtuoso Schematic Editor L Window



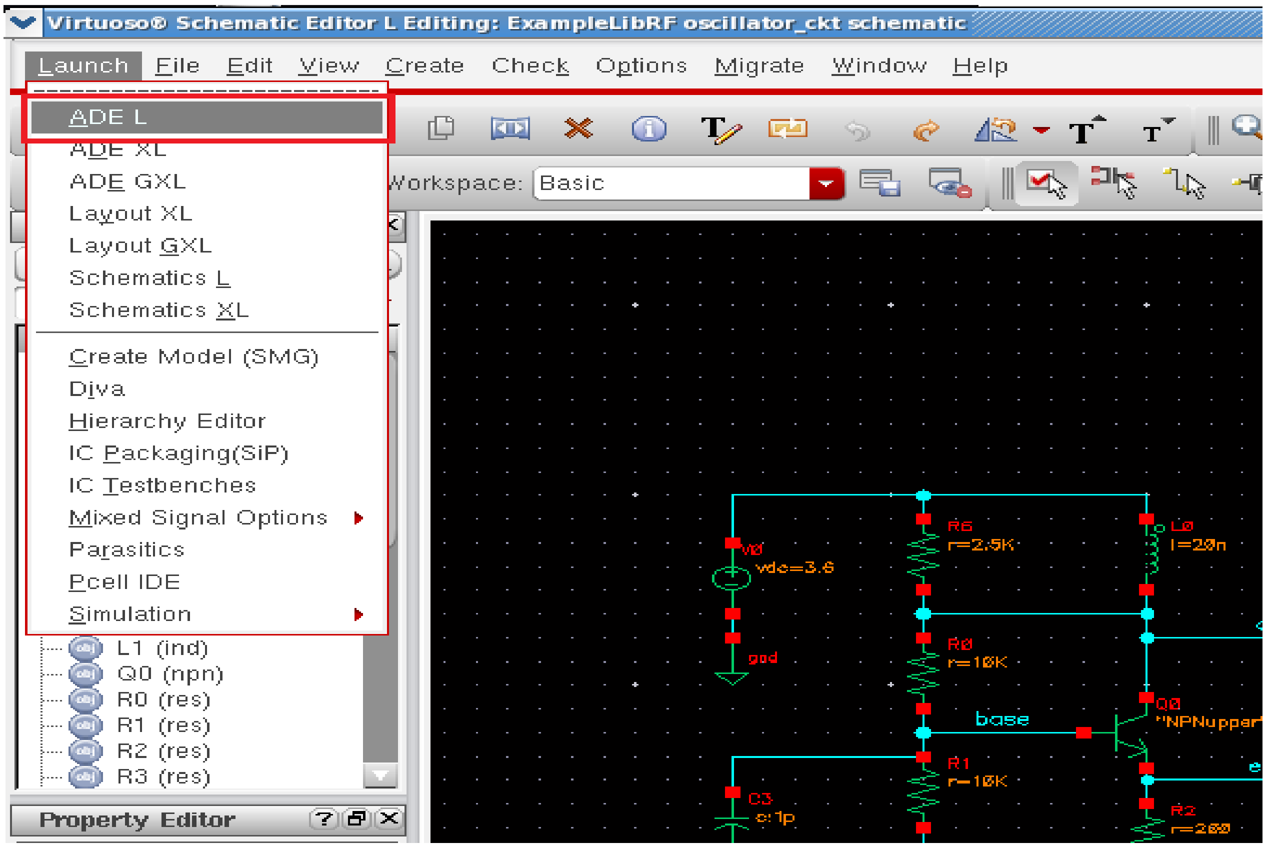
Setting up the HB and HBnoise Analysis

In this section, for performing phase noise measurement, you would be using the *hb* and *hbnoise* analyses.

Setting up the HB Analysis

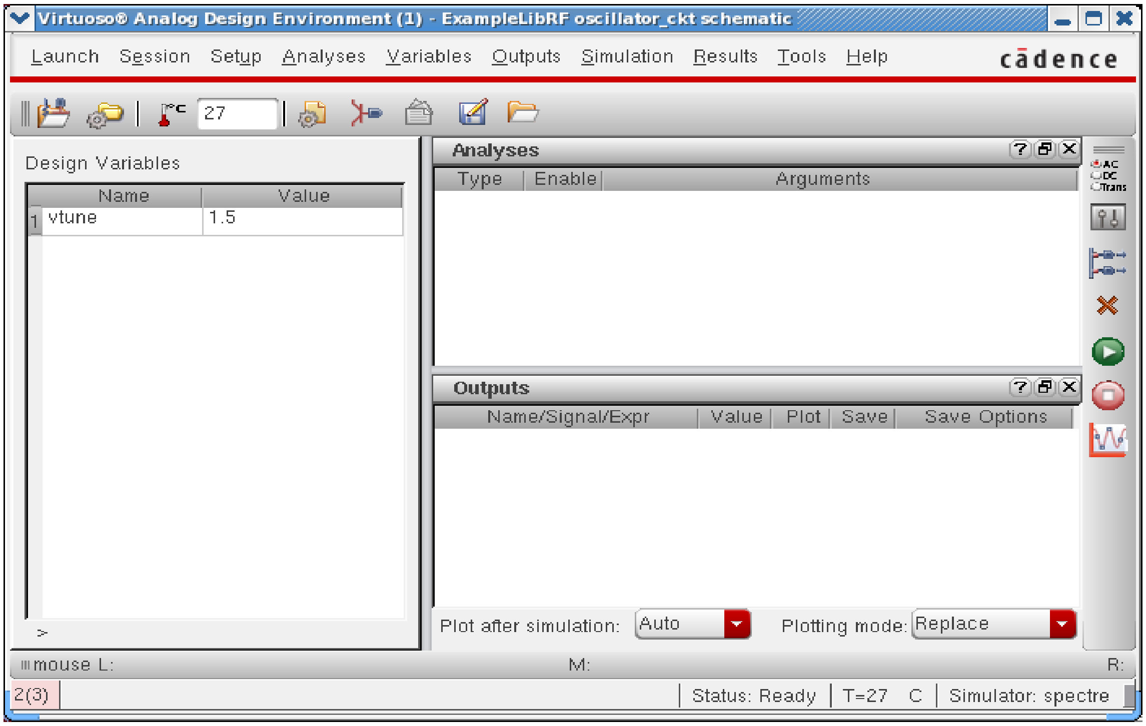
1. In the Schematic Window, choose *Launch - ADE-L.*

Figure A-150 Opening ADEL window from VSE window



2. The *Virtuoso Analog Design Environment* Window is displayed, as shown below.

Figure A-151 Virtuoso Analog Design Environment Window



3. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

4. Specify the following:

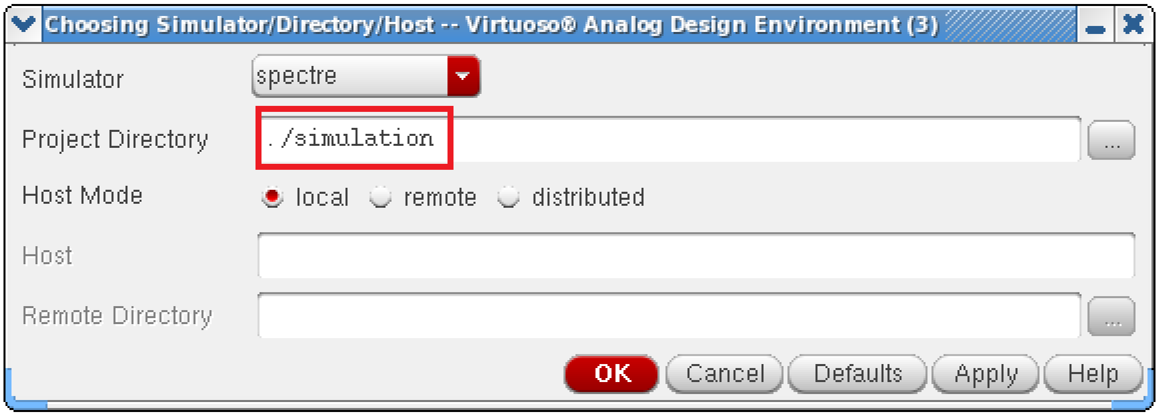
   a. Choose *spectre* for the *Simulator*.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your ~/simulation directory. You may change the default location by editing the settings in your .cdsinit or .cdsenv file. In this workshop, the simulation directory is set to ./simulation.

   c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, please contact your Cadence tools System Administrator for specific set-up instructions.

   The completed form is displayed similar to the one below:

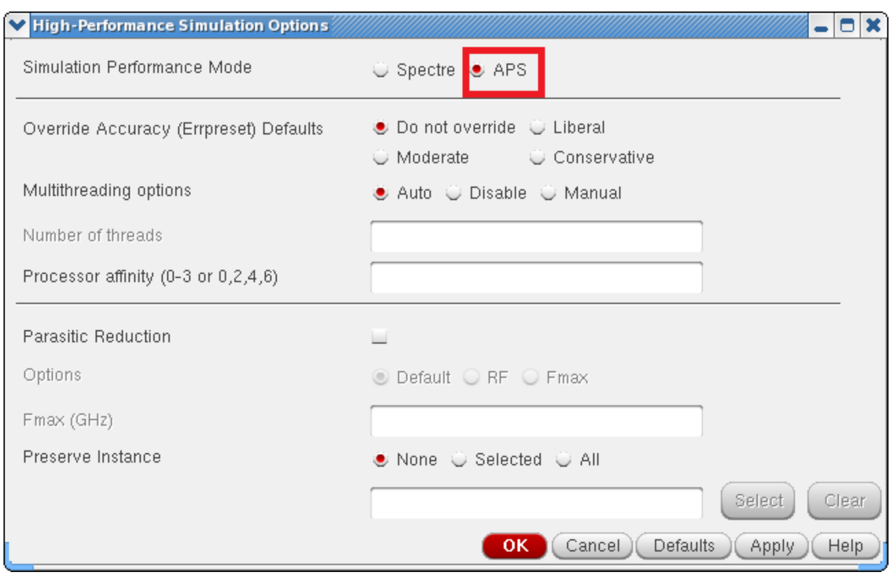
Figure A-152 Choosing Simulator/Director/Host Form



5. Click *OK*.
6. Set up the High Performance Simulation Options.

   In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

Figure A-153 High Performance Simulation Options Form



7. In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually, it is better to specify the number of threads yourself. Small circuits should use a small

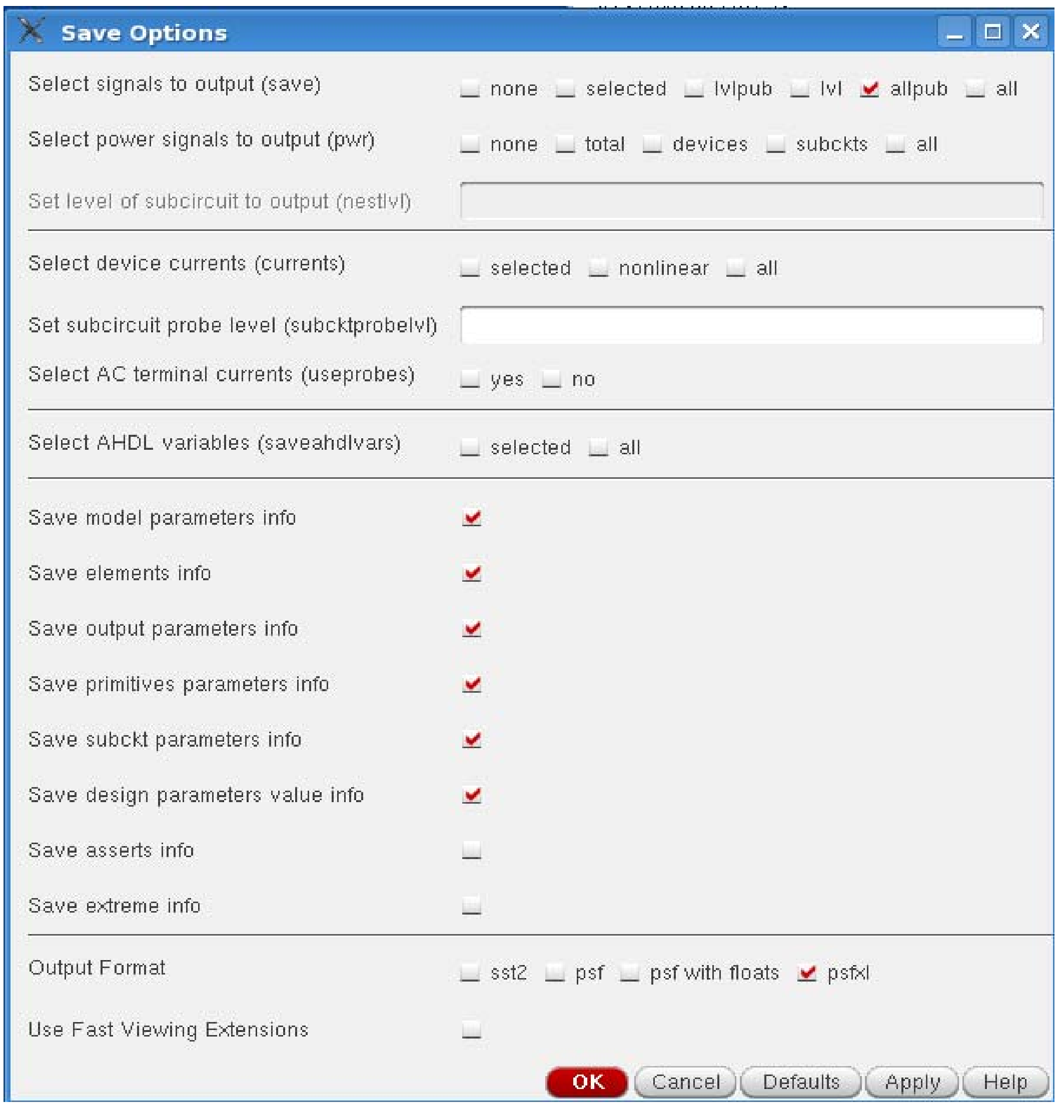
number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

8. Click *OK* to close the *High Performance Simulation Options* form.

9. In the *Virtuoso Analog Design Environment* window, choose *Outputs - Save All*.

The *Save Options* form is displayed.

Figure A-154 Save Options Form



10. In the *Select signals to output section (save)*, make sure that *allpub* is selected.

This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

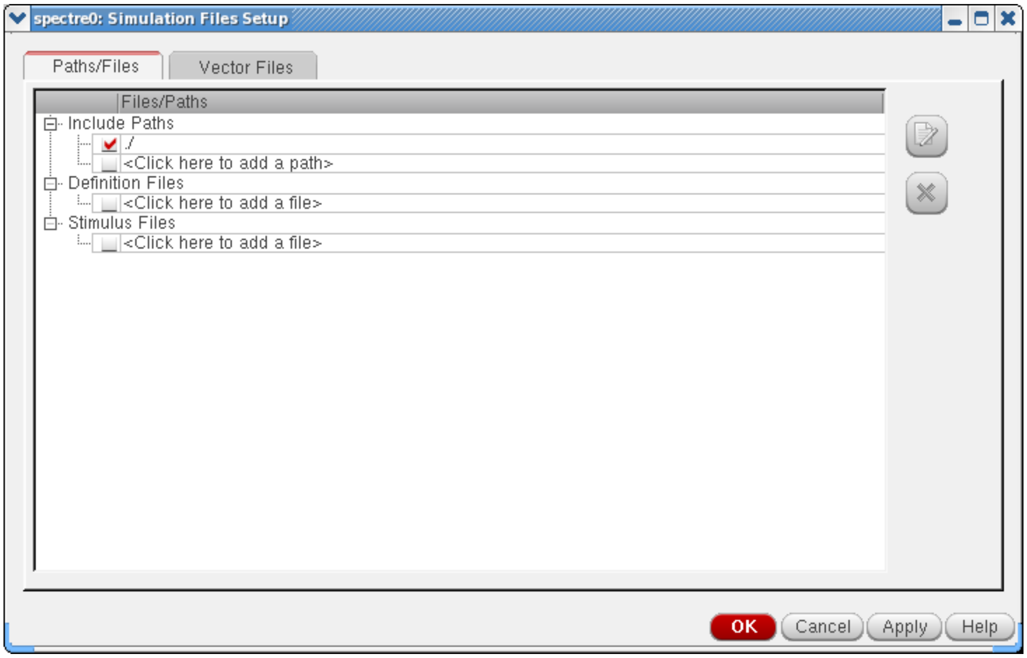
To save the currents, use the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or all if you want to save all the currents in the circuit. When you save currents, more disk space is required for the results file.

11. Click OK.

12. In the *Virtuoso Analog Design Environment* Window, choose *Setup - Simulation Files.*

The *Simulation Files Setup* form is displayed, as shown below.

Figure A-155 ***Simulation Files Setup*** **Form**



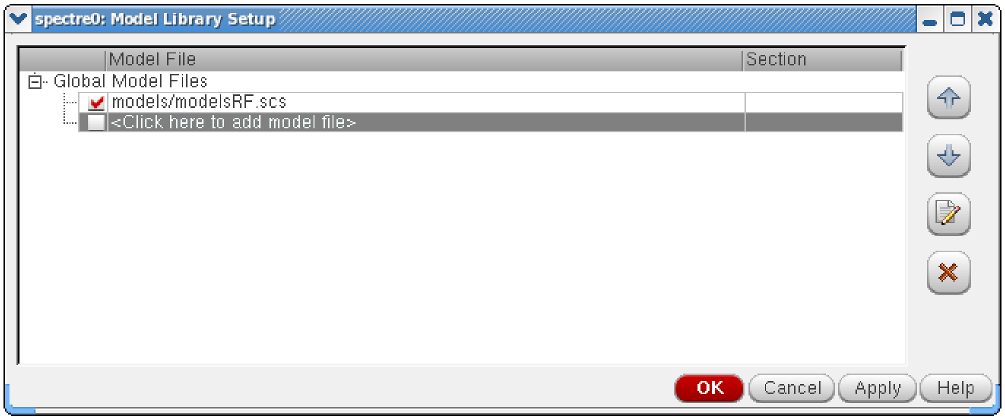
13. In the *Simulation Files Setup* form, type ./ by clicking in the *Include Paths* section.

14. Click *OK* to close the *Simulation Files Setup* form.

15. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries*.

The *Model Library Setup* form is displayed.

Figure A-156 *Model Library Setup* Form



16. In the *Model File* field, type the path to the model file including the file name, as shown below:

    models/modelsRF.scs

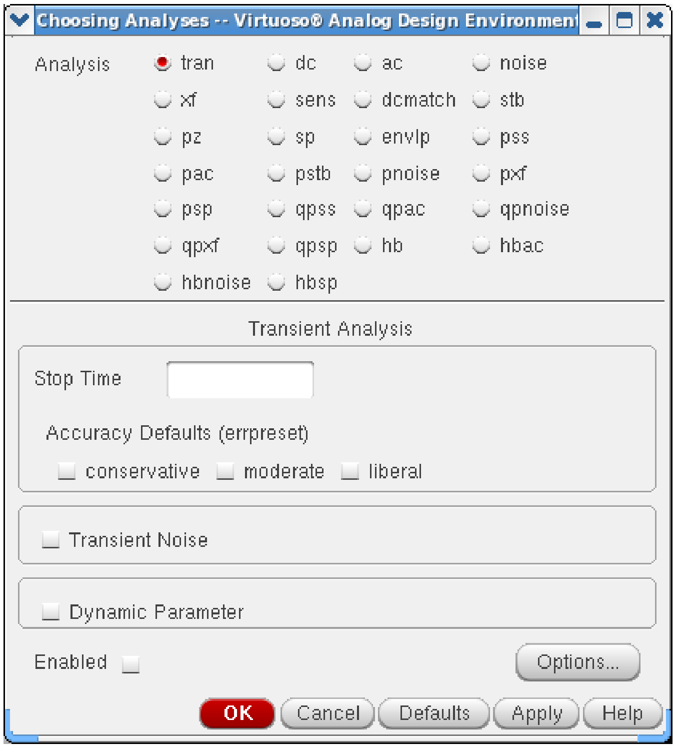
    You can also browse to the *modelsRF.scs* file.

17. Click *OK* to close the *Model Library Setup* form.

18. Choose *Analyses - Choose* in the *Virtuoso Analog Design Environment* window.

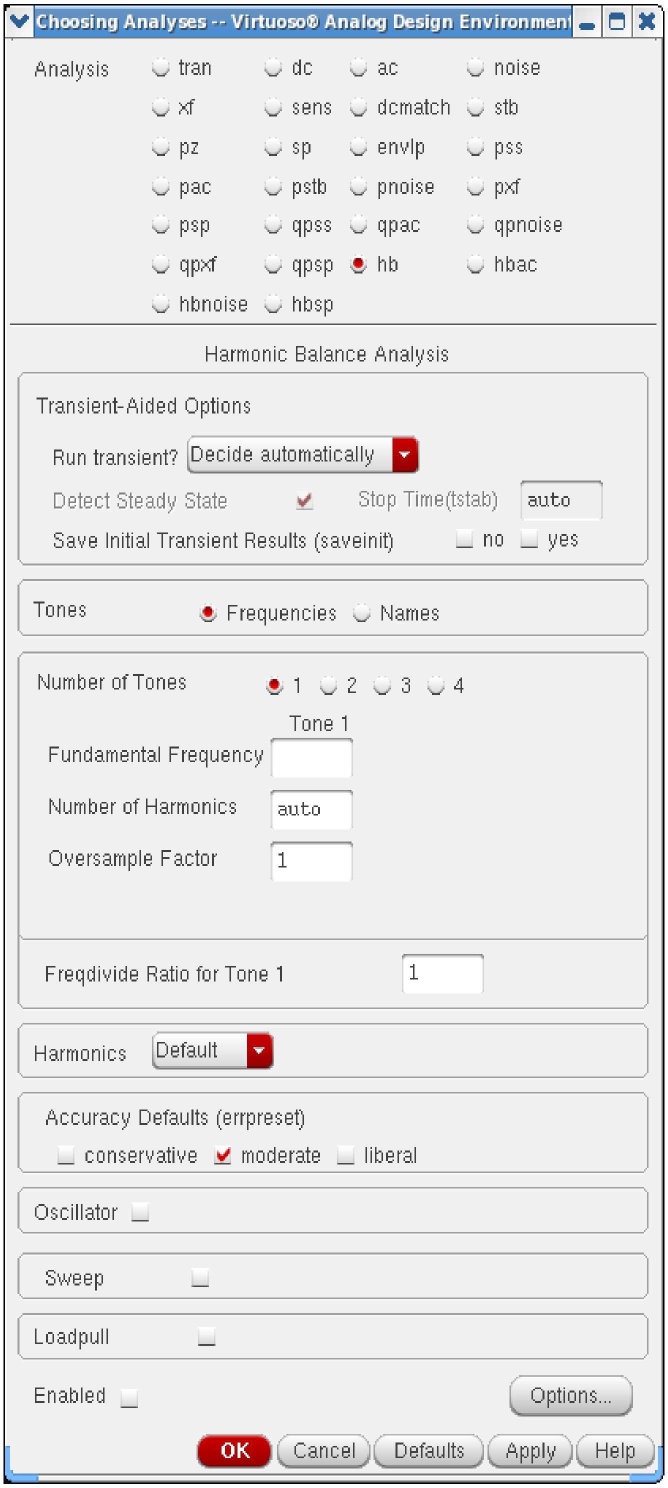
    The *Choosing Analyses* form is displayed, as shown below..

Figure A-157 The Choosing Analyses Form



19. Select *hb* in the *Analysis* section. The form expands, as shown below.

Figure A-158 The Choosing Analyses Form - Setting *hb* analysis

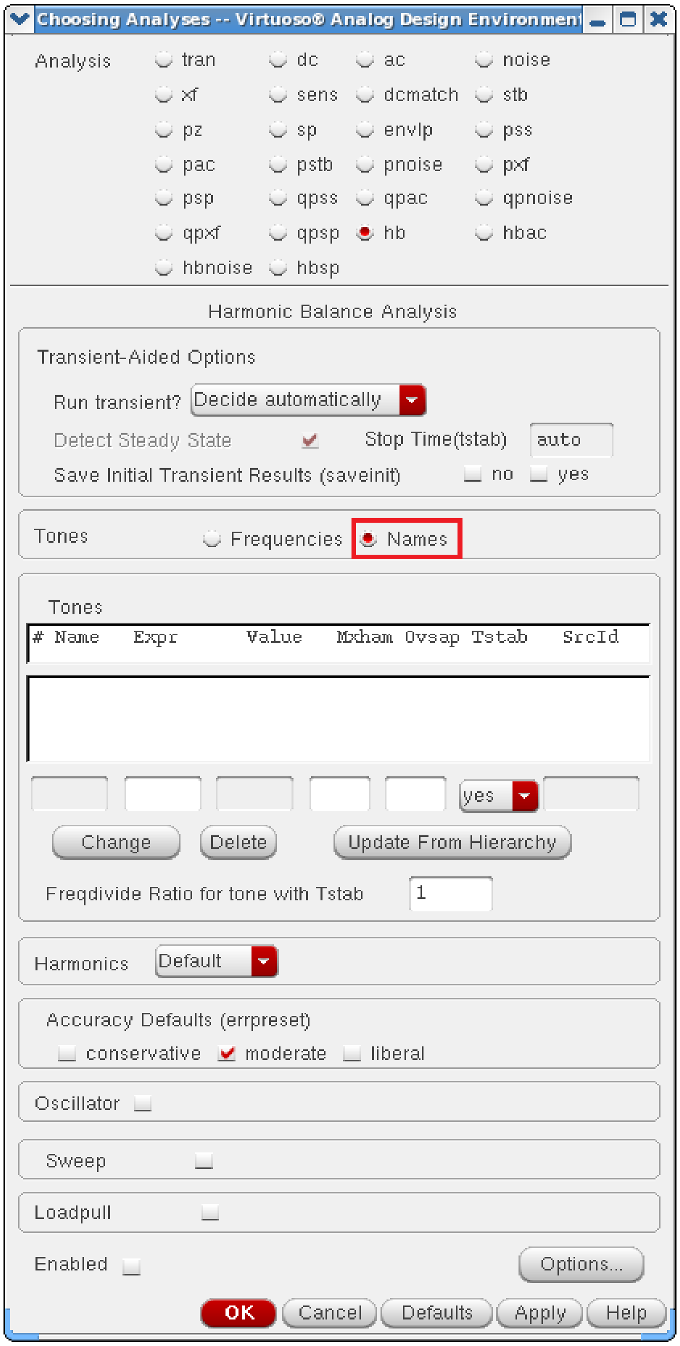


20. In the *Transient-Aided Options* leave *Run Transient?* as *Decide automatically* which is set by default. The other choices for this option are *Yes* and *No*. If *Yes* is used then the *Detect Steady State* option becomes active and the *Stop Time(tstab)* field is also activated. This means that you can decide whether you would like the Steady State to be detected automatically during the transient run or not and also specify a *Stop time(tstab)* for that transient run. When you select the checkbox for *Detect Steady State* option, this will run *the*transient analysis until steady-state is detected andthen switches to *hb*.

Select *yes* for *Save Initial Transient Results* (*saveinit*) option. This will help in visualizing the buildup of the oscillation waveform.

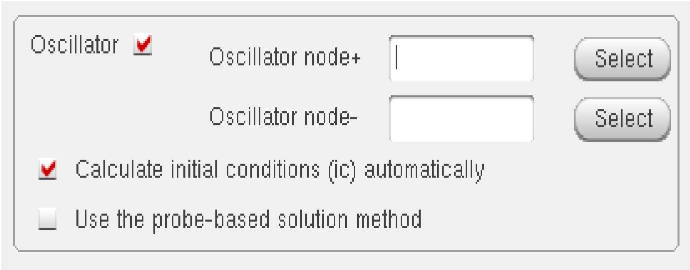
21. Select *Names* in the *Tones* section. The other option is *Frequencies* which is selected by default. *Names* is chosen here as it is similar to *pss* Harmonic Balance analysis setup. The form changes,as shown below.

Figure A-159 The Choosing Analyses Form - Setting *hb* analysis using *Names*



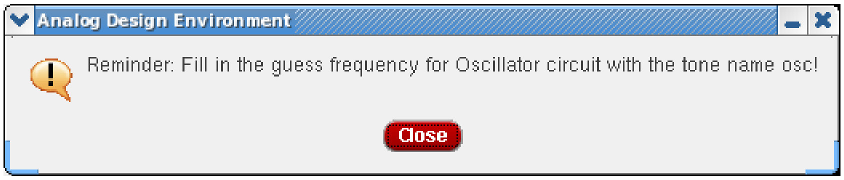
22. Select the *Oscillator* option. This is required for simulating an autonomous circuit.

Figure A-160 The Choosing Analyses Form - Oscillator Section



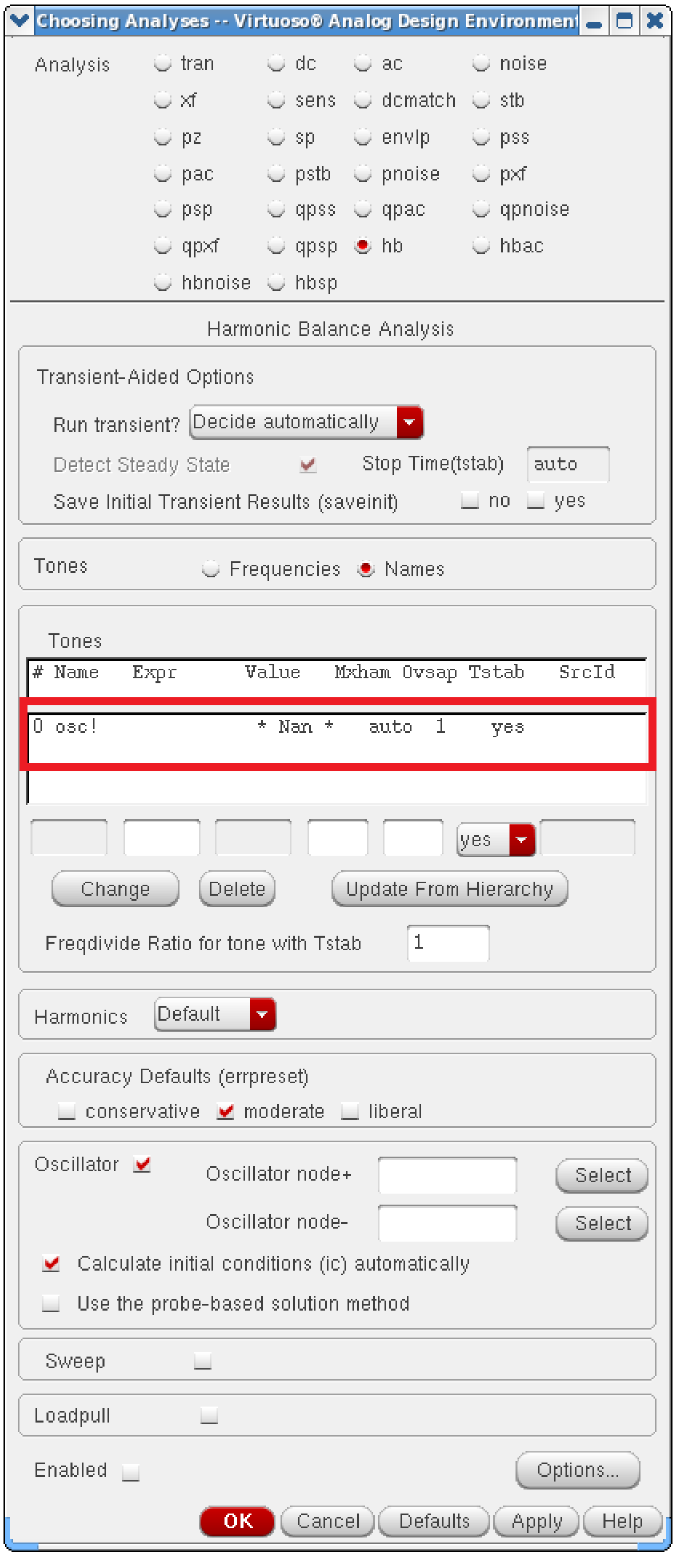
23. A popup window, as shown below appears. This informs about creation of an osc! frequency line entry in the *Tones* field. Click *Close* to close the window.

Figure A-161 Popup Window - osc! entry creation message



Closing the above popup window will result in creation of the osc! frequency line in the *Tones* section, as shown below.

Figure A-162 Choosing Analyses Form - *hb* autonomous setup



24. Select the osc! line in the *Tones* section. Enter 1.9G in the *Expr* field. Press the Tab key to move to next field and for the changes to take affect. Leave the *Number of Harmonics* option set to *auto*. When *Number of Harmonics* is set to *auto,* the simulator calculates harmonics automatically. The calculation is based on Fourier analysis of transient steady-state waveforms.

25. Leave the *Oversample Factor* option as set to 1 by default. Since the oscillator has sinusoidal waveforms, an oversample of 1 is appropriate.

26. Enter 1 for the *Freqdivide Ratio for Tone with tstab* option (which is set by default) as there is no frequency divider in the circuit. If there is a frequency divider in the circuit, then you need to set the *Freqdivide Ratio for Tone with tstab* to the divide ratio of the divider. For example, if the divider is divide-by-two, then the divide ratio is 2. Therefore, you will set *Freqdivide Ratio for Tone with tstab* to 2.

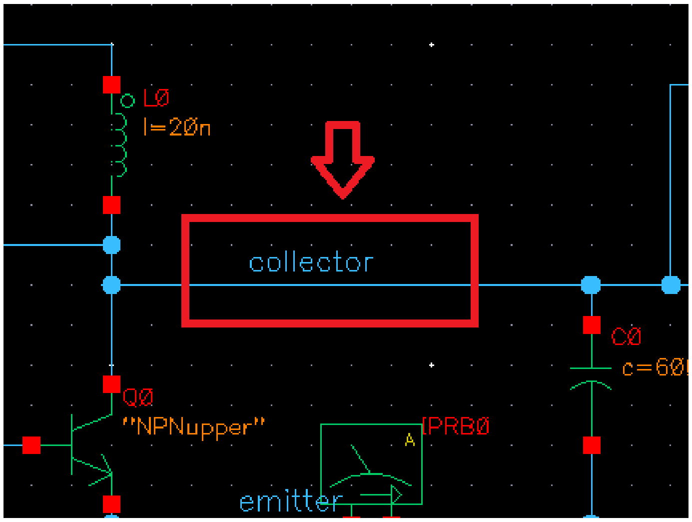
27. *Leave the Harmonics* option as it is which is set to *auto* by default.

28. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. Conservative is recommended for all the oscillators.

29. In the Oscillator section, in *Oscillator node* field, click *Select* just to the right of this field. In the schematic, select the *out* node. Instead of selecting the node from the schematic you can also type */out* in the *Oscillator node* field. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it. Leave the *Reference node* blank.

Note that if you have a single-ended oscillator, only specify one node. If the second node that is, the *Reference node* is left blank, it will be connected to the global ground node automatically. However, if you have differential oscillator, you need to specify both the nodes.

Figure A-163 Selecting *collector* net on oscillator_ckt schematic



30. If you are simulating an LC oscillator, leave the *Calculate initial conditions (ic) automatically* checkbox selected (this is the default).

Note that *Calculate initial conditions (ic) automatically* is used to start the oscillator. Other methods to start the oscillator include putting a single current pulse into the resonator, setting initial conditions(ic), or ramping up the power at time = zero plus.

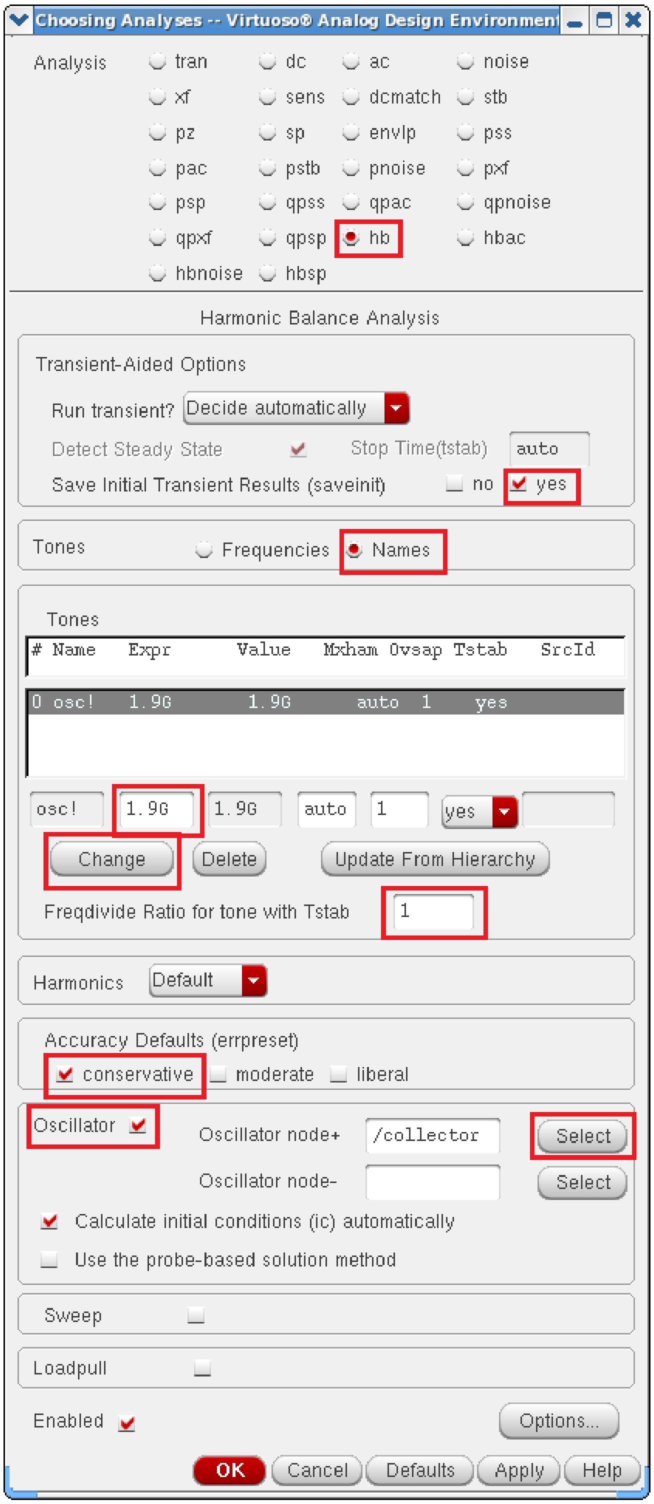
31. Note that, by default, the *Use the probe-based solution method* (*oscmethod*) option is deselcted (or unselected). Spectre will use the *onetier* method.

When the *Setting Use the probe-based solution method* (*oscmethod*) option is selected, it iterates for the frequency solution in the outer loop and the amplitude and phase solution in the inner loop. The probe-based method has better convergence but is computationally intensive.

Please refer to *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more details.

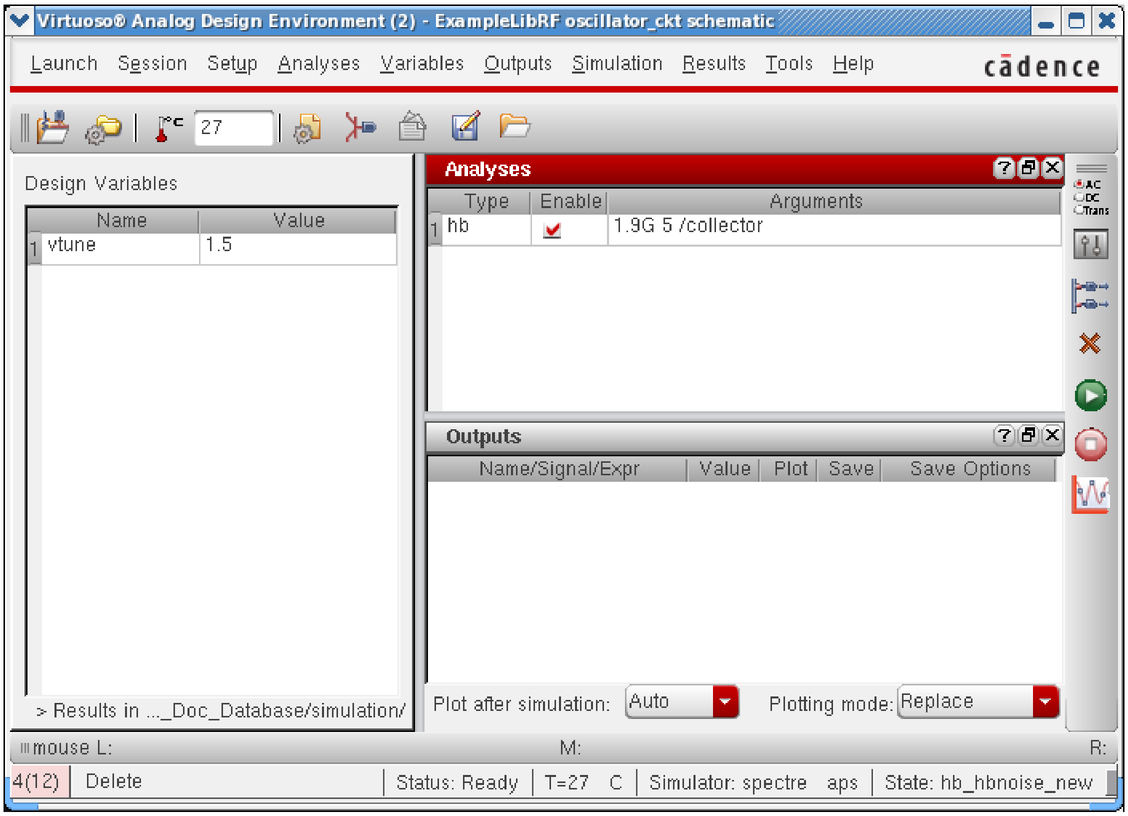
32. The *Choosing Analyses* form will look like the following:

Figure A-164 *Choosing Analyses* Form - HB Analysis Setup



33. Now click *Apply* at the bottom of the form. This will add hb analysis to the ADE Analyses Section, as shown below:

Figure A-165 ADE Simulation Window - hb analysis setup



Setting up the *hbnoise* Analysis

1. In the *Choosing Analyses* form select *hbnoise* analysis. The form expands, as shown below.

Figure A-166 *hbnoise* Choosing Analyses Form

2. Leave the *Sweeptype* to *default*.

   For oscillators, the *hbnoise/pnoise* frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the hb/pss form, you should specify the approximate frequency of oscillation for the frequency-divided signal. In the *hbnoise/pnoise* form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of *relative* is to take the frequency of the harmonic number specified and add to it the frequencies specified in the *Choosing Analyses* form. If the oscillator has a 1GHz output, and the pnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

a. Type 1 in the *Relative Harmonic* field. You are trying to determine the noise associated with the fundamental frequency of the oscillator.

Next, you will set the output frequency sweep range. Frequency sweep is set for a noise simulation as the noise is spread over the frequency range and it can affect the adjacent frequency channels. Therefore, it is critical to determine its behavior over a frequency range.

b. Type 10 in the *Start* Field in the *Ouput Frequency Sweep Range (Hz)* section.

c. Type 100M in the *Stop* Field in the *Ouput Frequency Sweep Range (Hz)* section.

Set the frequency sweep range as appropriate for your circuit (or application).

d. Set the *Sweep Type* to *Logarithmic*.

e. Type 3 in the *Points Per Decade* field. Typically, 3 to 5 points per decade are a reasonable number to capture the noise behavior of the circuit.

3. Leave the *Maximum Sideband* field blank. In general, Maximum sideband needs to be set high enough to include all the frequencies that could mix down to the oscillator output frequency. By default, when this field is left blank, all the mixing with all the *hb* harmonics are present in the *hbnoise* result.

4. Set the *Output* to *voltage*.

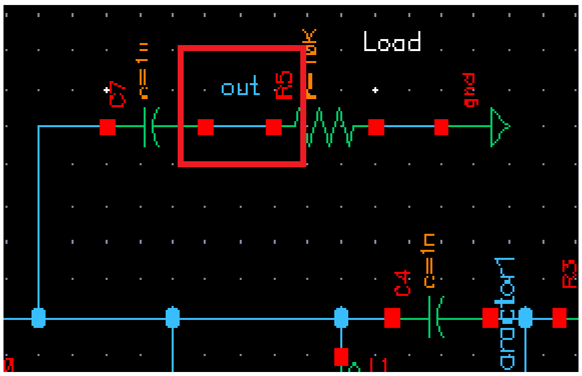
a. Type /out in the *Positive Output Node* field. You can also select the *out* net from the schematic by clicking the *Select* button on the right of the *Positive Output Node* field and then selecting the net just below the *out* label in schematic.

b. Leave the *Negative Output Node* field blank. If the second node that is the *Negative Output Node* is left blank, it will be connected to the global ground node automatically. However, if you have a differential oscillator, you need to specify both the nodes.

c. Set the *Input Source* to *none*.

In this case, because we are calculating output Total Noise, the Input Source is not needed (using an Input Source generates NF, IRN, and other parameters). Input Source is not usually used in oscillator simulations.

Figure A-167 Selecting *out* net from the schematic

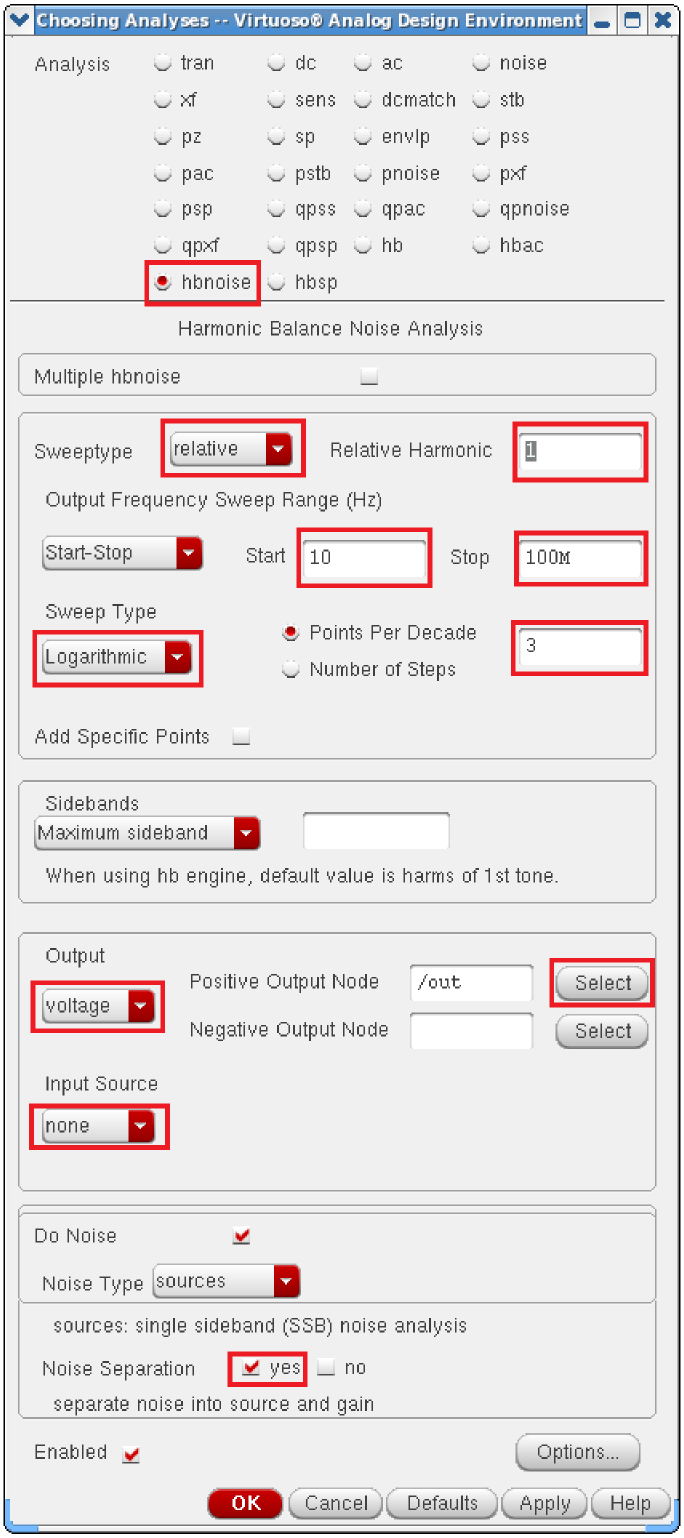


5. The *Do Noise* option is selected by default. Next, you will set the *Noise Type*.
6. Set the *Noise Type* to *sources*. Here, the available options are *sources, jitter, timedomain, and modulated*.

   When the *Noise Type* is *sources*, the total time averaged noise (AM + PM) is calculated. This is used most of the time when determining Single Sideband phase noise of the circuit.
7. Set *Noise Separation* to *yes*.

   Select *yes* for *Noise Separation*, if you want to plot the individual noise contributors.
8. The Choosing Analyses form should look like the following:

Figure A-168 Choosing Analyses Form - *hbnoise* Analysis Setup



9. You now need to set up the *augmented* and *lorentzian* options for the *hbnoise* analysis.

HBnoise/Pnoise is a small-signal analysis and is not limited by large-signal effects, such as clipping or slew-rate limits. As a result, at low offset frequency, the phase noise might be significantly greater than 0dBc/Hz. This indicates that the noise is larger than the oscillations, which is not physically possible. If you want to see the phase noise curve level off at low frequency, set the *lorentzian* option to yes.

The *Augmented* option which applies only to oscillators is an algorithm which provides better accuracy when the circuit has low frequency poles. It is strongly recommended when you have long time constants in the oscillator circuit.

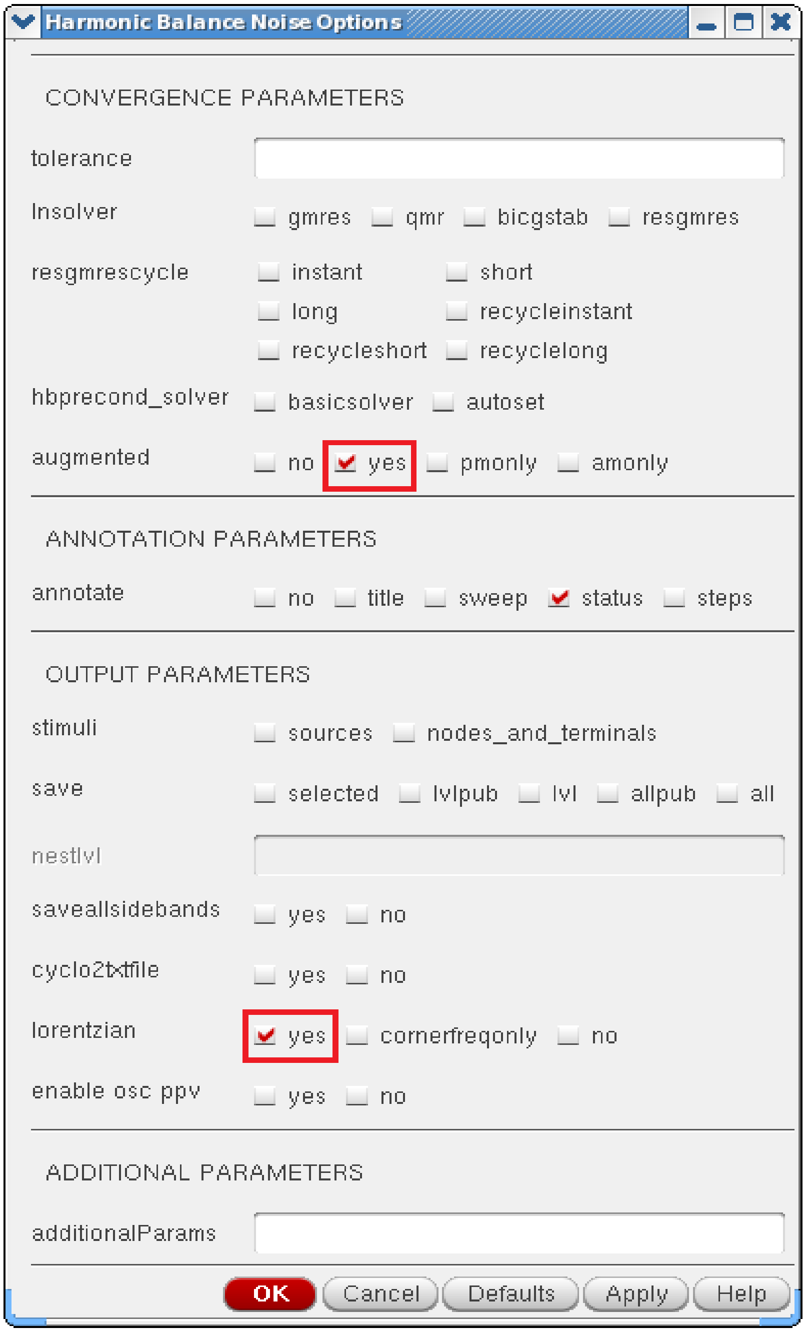
a. At the bottom of the *Choosing Analyses* form, select *Options*. This will open *Harmonic Balance Noise Options* form.

b. In the *Harmonic Balance Noise Options* form, select *yes* for the augmented option. Augmented is set to *yes* by default.

c. Select *yes* for *lorentzian* option in the *OUTPUT PARAMETERS* section of this form.

The *Harmonic Balance Noise Options* Form should look like the following:

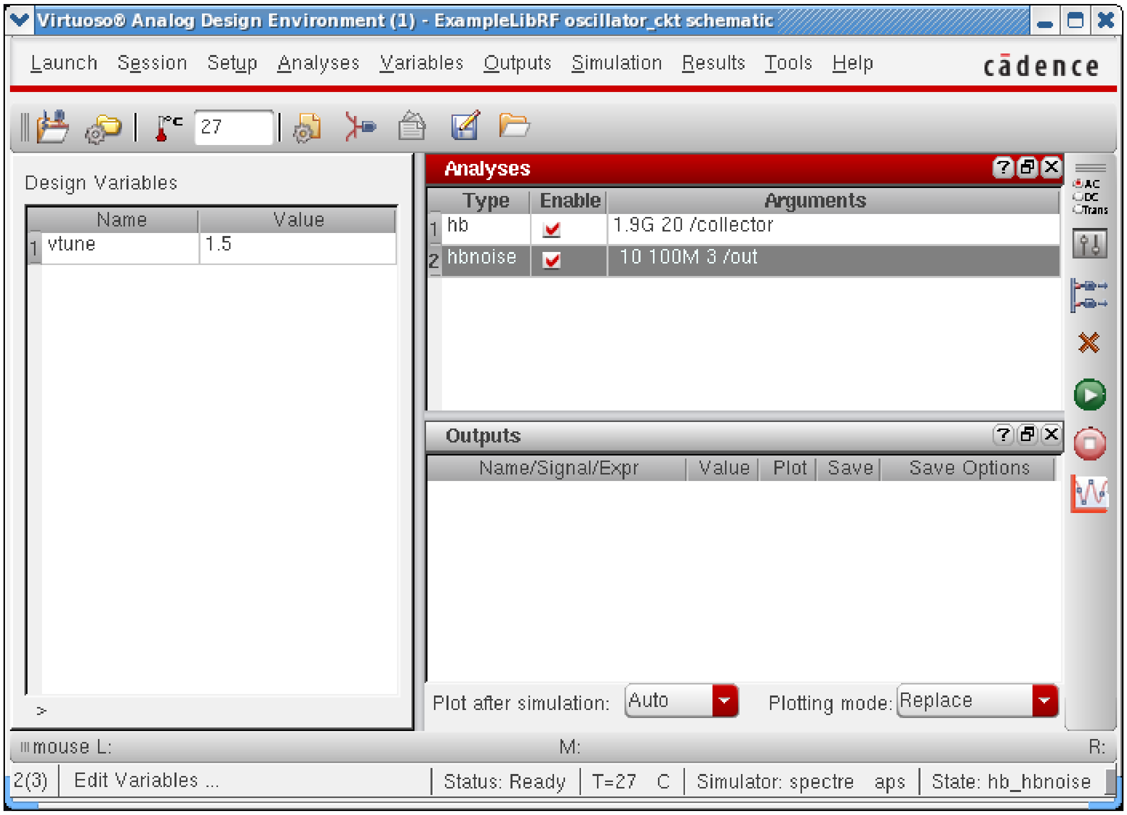
Figure A-169 *hbnoise* Options Setup Form



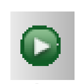
10. Select *OK* to close the *Harmonic Balance Noise Options* form.

11. Next click *OK* to close the *Choosing Analyses* form. This will also add the *hbnoise* analysis in addition to the *hb* analysis in the ADE *Analyses* Section, as shown below.

Figure A-170 ADE Simulation Window - *hb* and *hbnoise* analysis setup



Running the simulation

Once finished setting up the *hb* and *hbnoise* analyses. Click the green icon  located on the right side of the *Virtuoso Analog Design Environment* window or on the Schematic window to run the simulation.

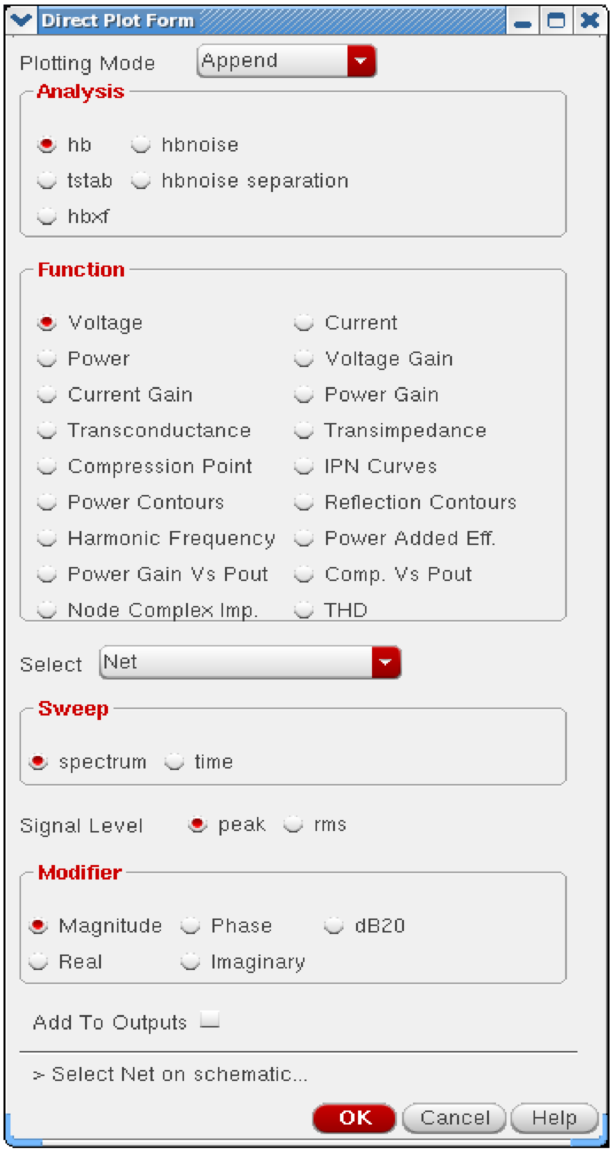
Next, plot the phase noise results and obtain the *Noise Summary* Table once the simulation is finished.

Plotting the results from *hb and hbnoise* analysis

First plot the SSB (single sideband) phase noise,as follows:

1. In the *Virtuoso Analog Design Environment* window, choose*Results - Direct Plot - Main Form*.

Figure A-171 ***hb*** **and** ***hbnoise*** **Direct Plot Form**



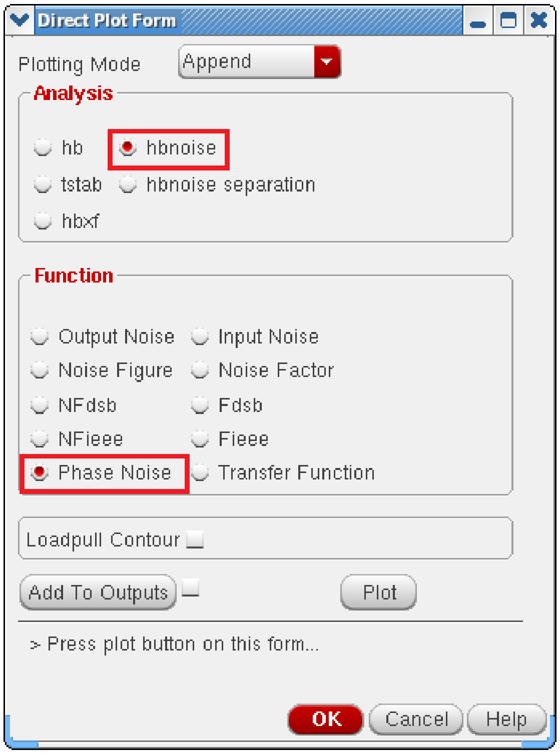
Note that even though you only selected *hb* and *hbnoise* analyses, you also have a *hbxf* analyses listed in the *Direct Plot Form*. *hbnoise* automatically runs a *hbxf* (harmonic balance transfer function) simulation as part of running a *hbnoise* simulation. *hbnoise separation* analysis gets added as you have set *Noise Separation* to *yes* in the *hbnoise* analysis setup form and *tstab* gets added as you have set *Save Initial Transient Resuts (saveinit)* to yes in the *hb* analysis setup form.

2. Select *hbnoise* as the Analysis.

3. Select *Phase Noise* as the Function.

The *Direct Plot Form* Setup should look like the following:

Figure A-172 *hbnoise* Analysis Direct Plot Form Setup



4. Click *Plot*.

   The phase noise is plotted, as shown below:

Figure A-173 Phase Noise Plot with CorFreq Label

The label *CorFreq* shows an estimate of the point on the phase noise where the phase noise curve in the physical world levels off instead of continuing its rise.

5. Select *File - Close All Windows* to close the Graph Window in ViVA.

Next, you will plot the hbnoise Separation.

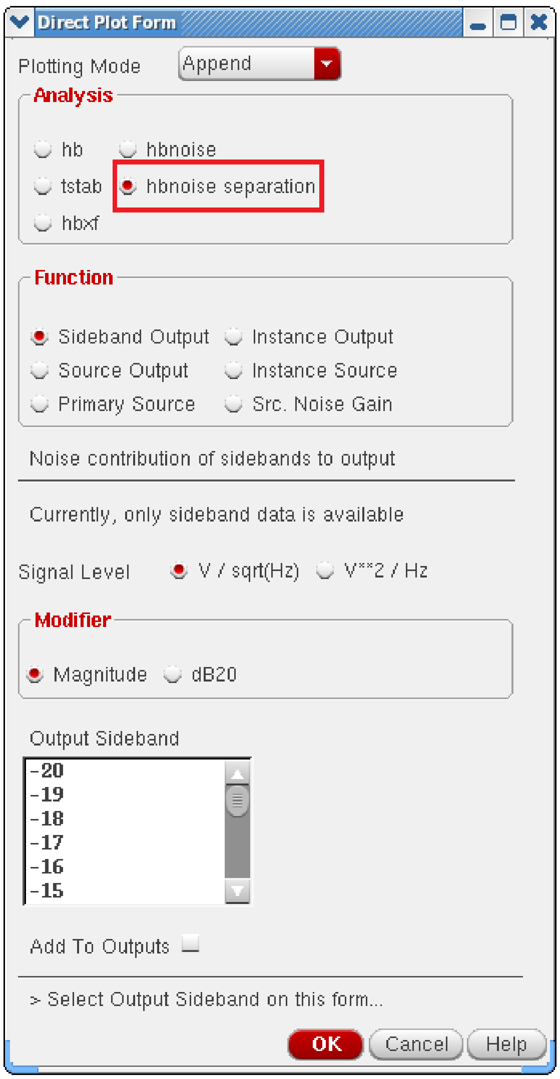
The key aspect in noise analysis is finding out what is causing the noise problem. Hbnoise provides the total noise from all noise frequencies, and in the noise summary, it provides the noise contributors with all the noise frequencies taken into account. The idea of noise separation is several-fold. First, you can identify the noise frequencies that cause the most noise at the output. Once you know the troublesome frequencies, you can identify the troublesome components. Once you know the troublesome components, you can identify the specific mechanisms within the component that are causing the problem. You can then work upon to find the solution to this problem which may include using different device dimensions or alternate circuit architectures and so on.

Therefore, to sumarize, noise separation is a way of extending standard hbnoise to find out more information about what is causing the problem related to noise. Once the problem is identified, then a solution can be worked upon to fix it.

When *Noise type* is set to *sources* and *Noise Separation* is set to *yes* in the *Choosing Analyses* form while setting *hbnoise*, the *hbnoise separation* feature is included during the simulation and the corresponding results are saved.

6. In the *Direct Plot Form*, select *hbnoise separation* in the *Analysis* section. The *Direct Plot Form* changes, as shown below.

Figure A-174 Direct Plot Form for hbnoise Separation



7. In the *Function* section, select *Sideband Output*. *Sideband Output* plots the noise contribution of selected sidebands to the output.

8. In the *Signal Level* section, select *V/sqrt(Hz)*.

9. Choose *dB20* as the *Modifier*.

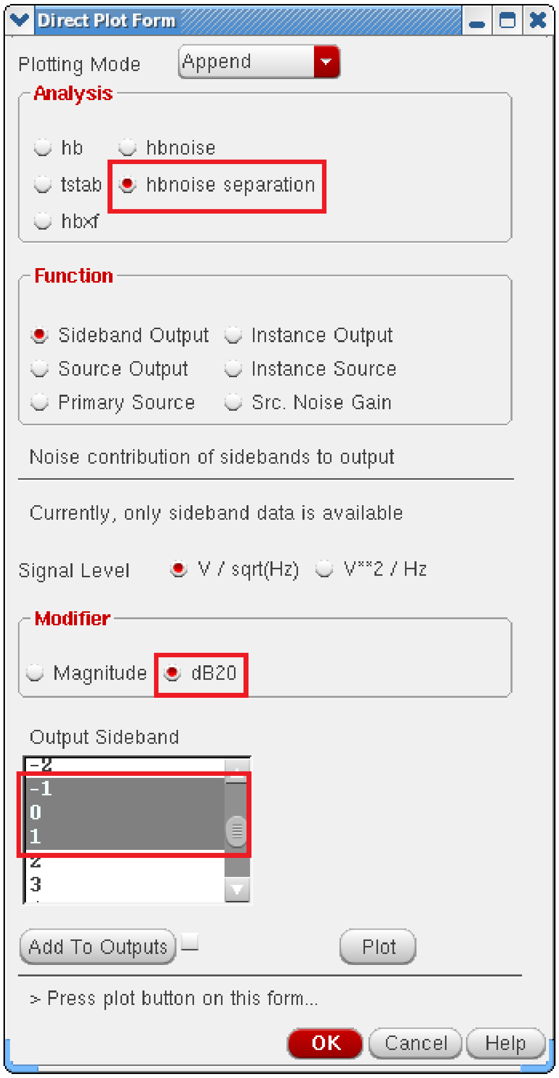
10. Select the -1, 0, and 1 *Output Sidebands*. This will plot the noise contribution of the -1, 0, and 1 sidebands to the output. This allows the identification of which noise frequencies are causing the largest noise contribution at he output of the circuit.

Tip

You can click and hold themouse over the -1 output sideband and slide the cursor to include the 1 output sideband.

The *Direct Plot Form* should look like the following:

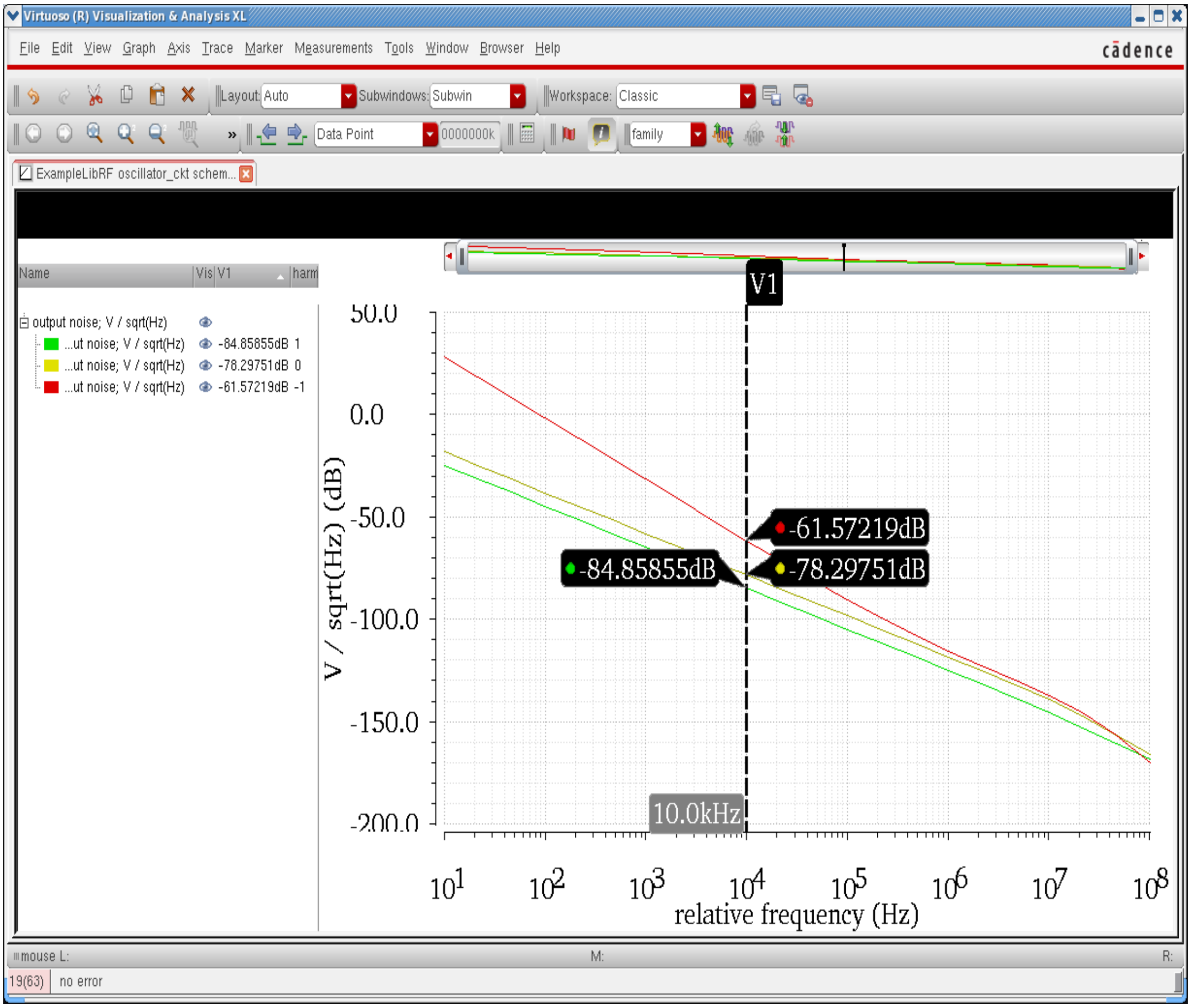
Figure A-175 Direct Plot Form setup for hbnoise Separation - Sideband Output



11. Click *Plot* .

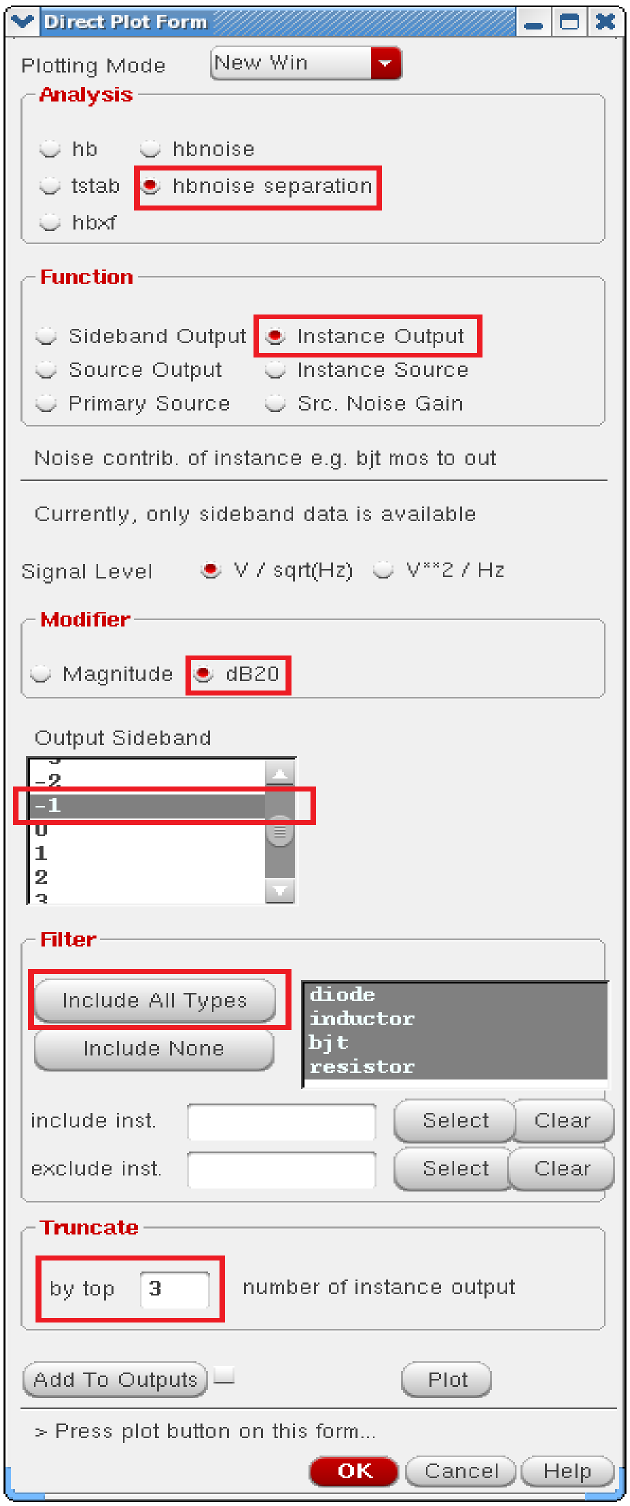
The hbnoise separation plot is shown below. Note that the harmonic numbers plotted are in the legend on the left side of the screen.

Figure A-176 HBnoise Separation, plot of -1, 0, 1 Sideband



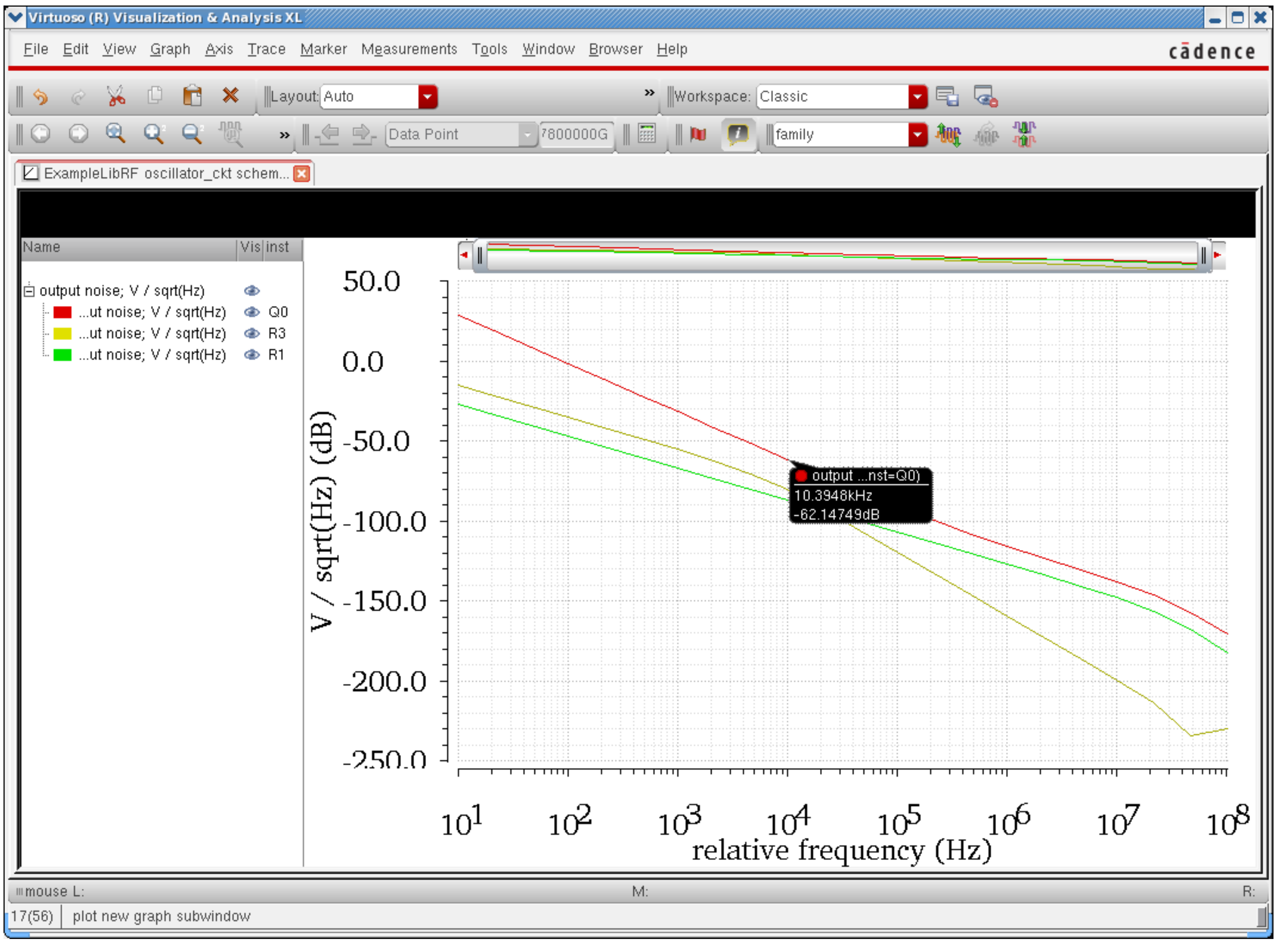
12. Next, plot theInstance output for the largest noise contributor above. (It is the -1 sideband in this case) This will show which components contribute the most noise at the output. In the *Direct Plot Form*, first set the *Plotting Mode* to *New Win* then select *Instance Output*. Verify that the Signal Level is set to *V/sqrt(Hz)*. Set the Modifier to *dB20*. Select the *-1* Output sideband. In the *Filter* section, select *Include All Types*. In the *Truncate* section, enter *3* in *by top number of instance output* . The *Direct Plot Form* should look like the one below:

Figure A-177 Direct Plot Form for Plotting Instance Output for Noise Contribution of instances



13. Click *Plot* . The plot is displayed. There are three plots for the top three noise contributors.

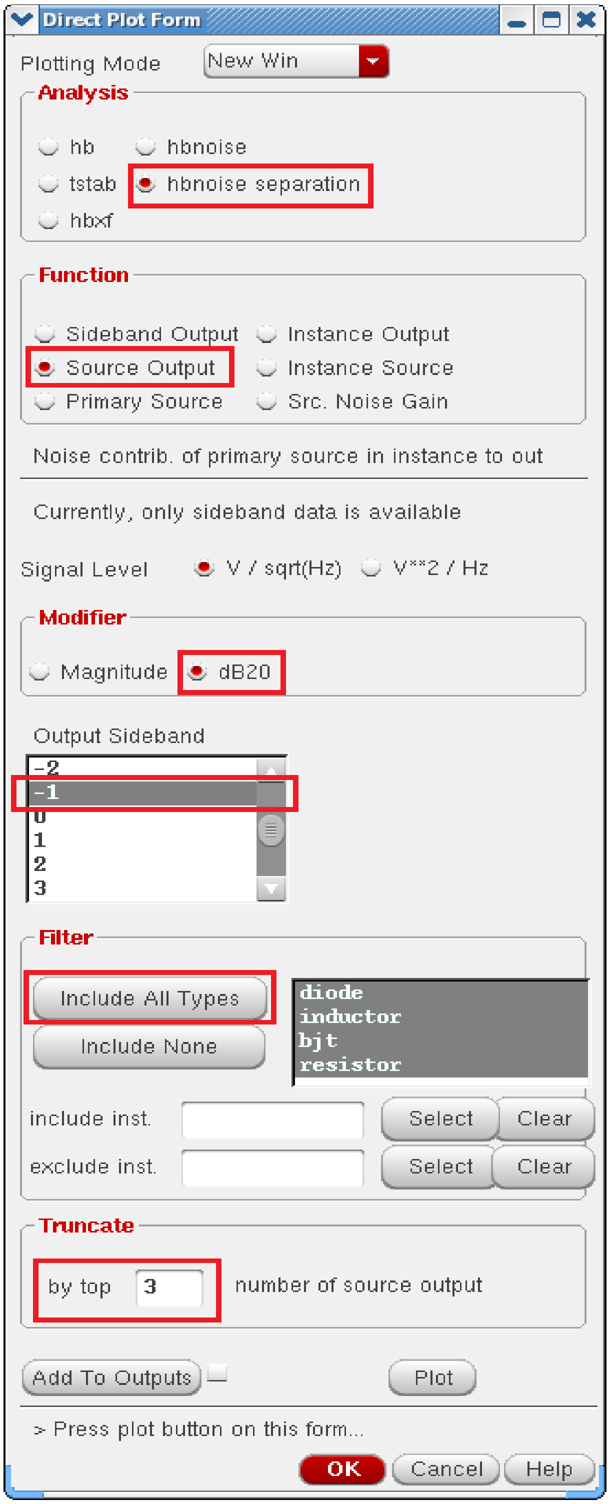
Figure A-178 Plotting Output Noise Top Instance Contributors



14. If you hover your mouse over a particular trace, it will show which instance contributes the most amount of noise. Above, the red line represents the noise contribution from the instance Q0 (legend shown), the yellow line below represents the noise contribution from instance R2, and the bottom green line represents the noise contribution from instance R6.

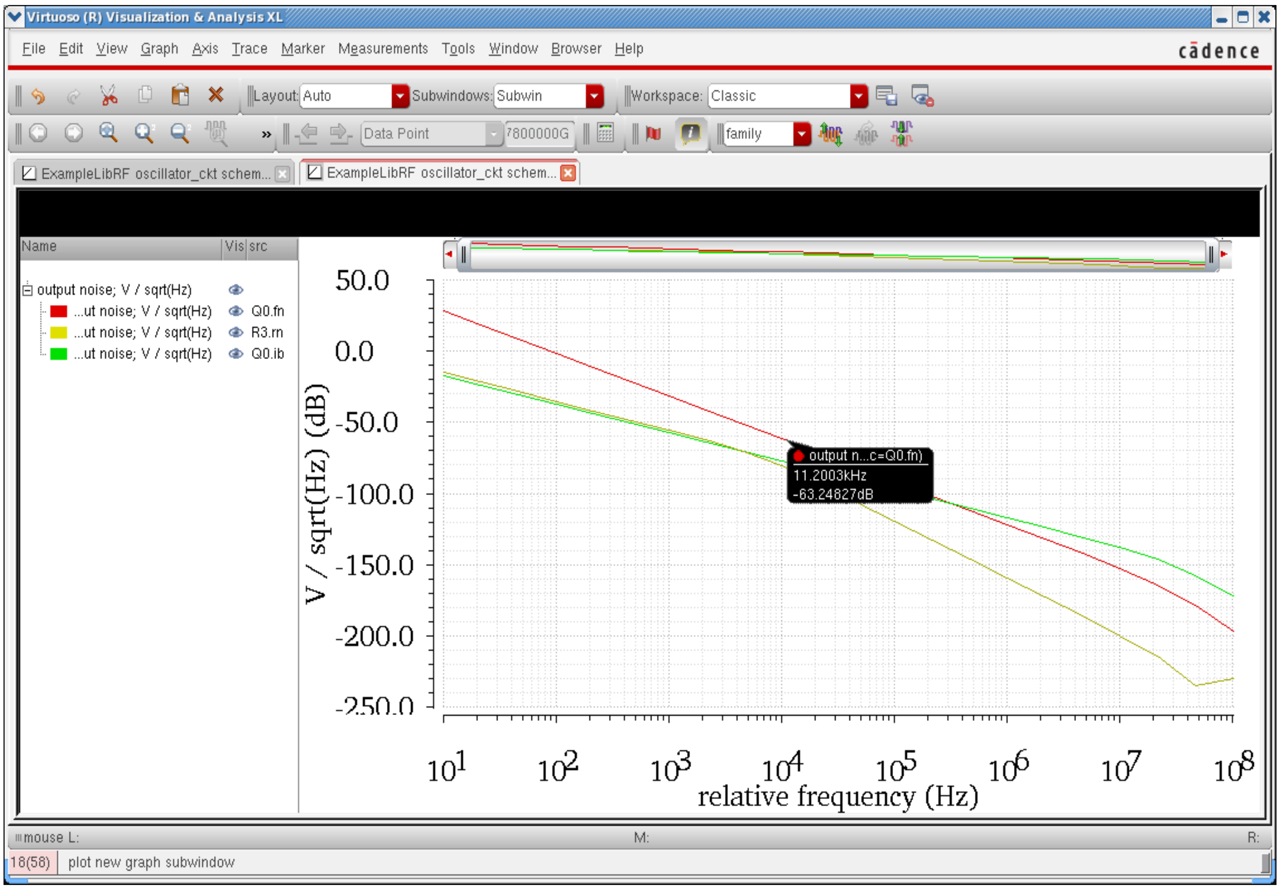
15. Next, plot the *Source Output* so you can see which individual noise sources within the circuit are contributing the most noise. In the *Direct Plot Form*, change the *Function* to *Source Output*. Select *Signal Level* to be *V/sqrt(Hz)*. Set the *Modifier* to *Magnitude*. Leave the *Output Sideband* at *0*. In the *Filter* section, select the *Include All Types* option. In the *Truncate* section, type 3 in the *by top* field. The *Direct Plot Form* should look like the one below.

Figure A-179 Direct Plot Form for Plotting by Top 3 Source Noise Contributors



16. Plot the source output so you can see which individual noise sources within the circuit that are making the most noise.

Figure A-180 Top 3 Source Noise Contributors



17. Note that the top three Source Output contributors are displayed in the legend on the left side of the graph: Q0:fn, R3:rn and Q0:ib. Here “fn” is the flicker noise, “ib” is the base current shot noise of BJT and “rn” is the resistor thermal noise. See Chapter 3 in this guide for more information on the various noise parameters.

18. In the *Direct Plot Form*, click *Cancel*.

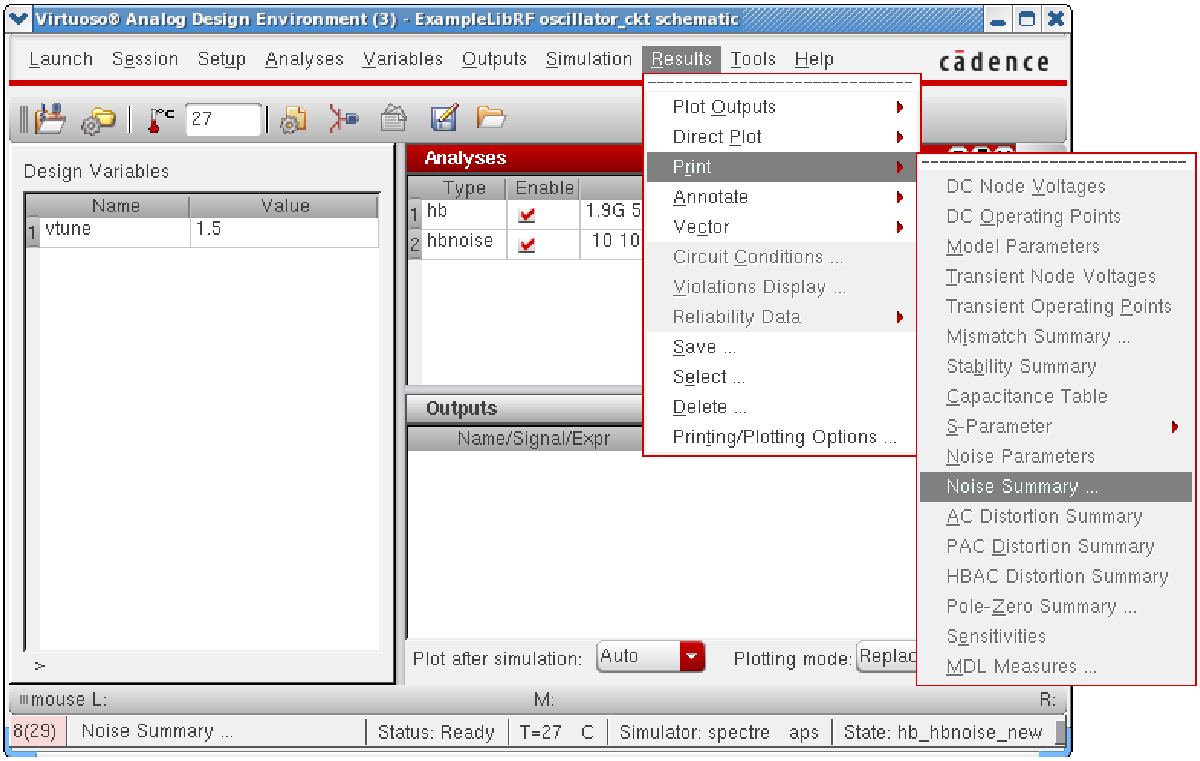
19. In the ViVA window, choose *File - Close All Windows*.

Noise Summary Table

This table provides the list of noise contributors based on the selections made on the Noise Summary Table form.

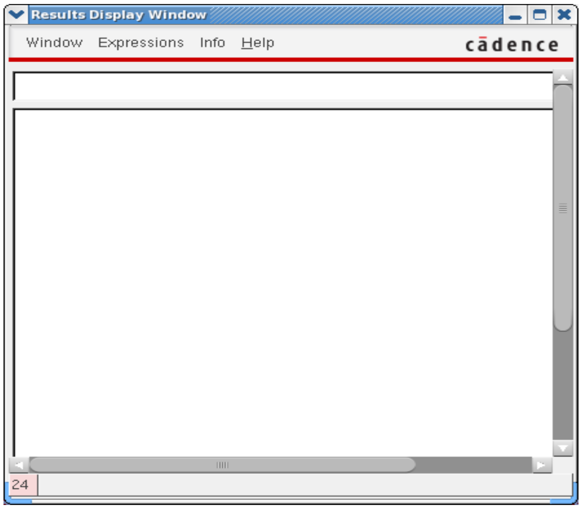
1. Choose *Results -Print - Noise Summary* in the *Virtuoso Analog Design Environment* window.

Figure A-181 Print Noise Summary in ADE L Window



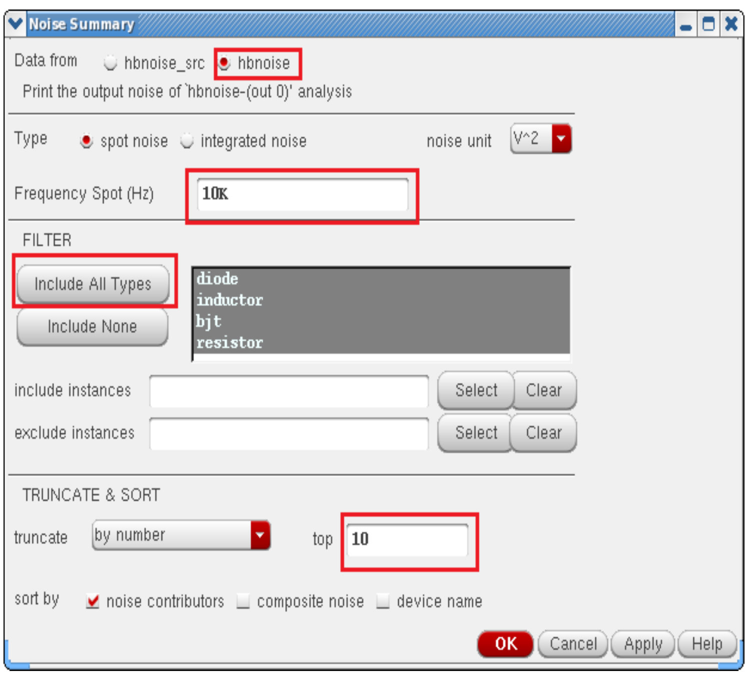
The *Results Display Window* and *Noise Summary* windows are displayed. The Results Display Window will currently blank, as shown below:

Figure A-182 Results Display Window



The *Noise Summary* window is shown below.

Figure A-183 Noise Summary Setup Form



2. In the *Noise Summary* window, set *Data from* as *hbnoise*.
3. Set *Frequency Spot (Hz)* to 10K. In spot noise, you will be looking at the noise at one frequency.

4. Click *Include All types* in the *Filter* Section. This will includethe noise contributions from all devices including resistor, inductor and so on.

5. In the *Truncate and Sort* section, enter 10 inthe *top* field. The top 10 noise contributors in the *Results Display* window are displayed.

   Note: You can shorten your noise summary either:

   - *by number* - that is, by specifying the highest contributors to include in the summary,

     or

   - *by relative threshold* - that is, by specifying the percentage of noise a device must contribute to be included in the summary,

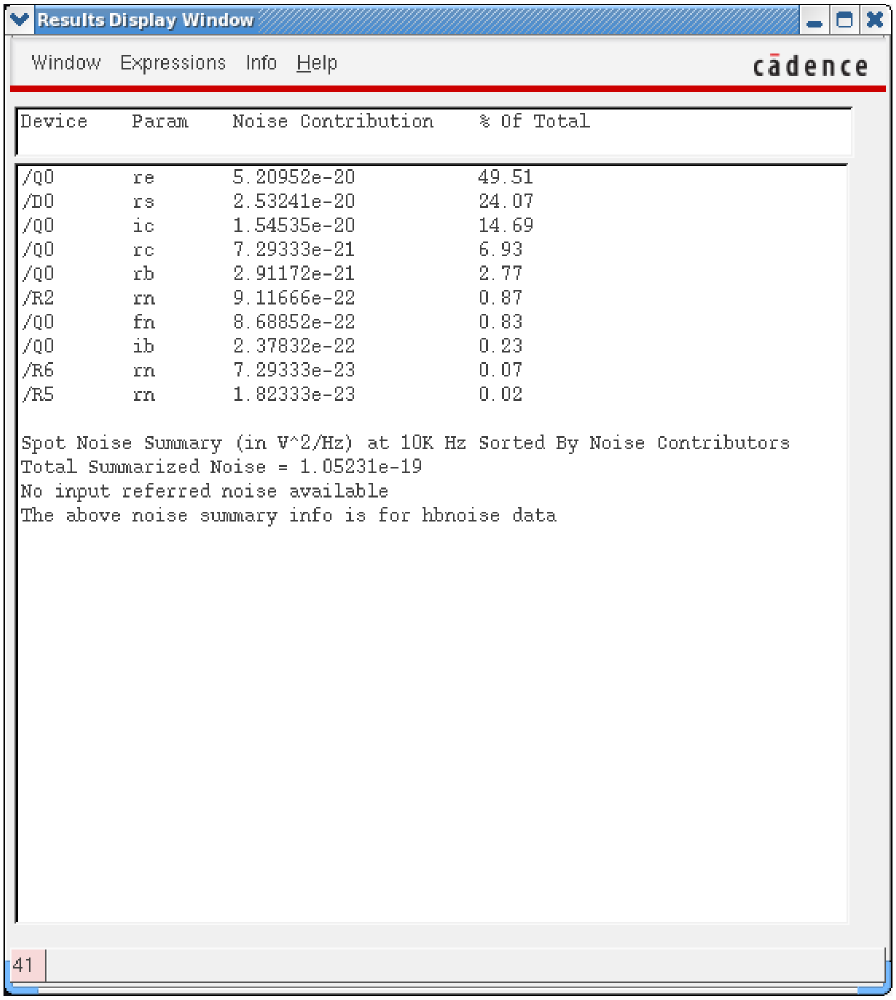
     or

   - *by absolute threshold* - that is by specifying the level of noise a device must contribute to be included in the summary.

6. Click *OK*.

   The noise summary resultsare displayed in the *Results Display Window,* as shown below:

Figure A-184 Noise Summary Results Display Window



The *Results Display Window* lists the individual contributors, the specific noise mechanism within the semiconductors causing the noise, and the noise contribution. The Total Summarized Noise is shown at the bottom of the Results Display window. Since there is no Input Source present, so there is no input referred noise available. This is usually the case for oscillators.

Note that the output noise include the noise from all the noise contributors, and not just the contributors in the form.

7. In the *Results Display Window,* choose *Window - Close* to close the window.

8. Close the Analog Design Environment window by choosing *Session - Quit*.

9. Click *No* in the *Save State* Window.

10. In the Schematic window, choose *File - Close*.

In this section, one of the key measurements of Oscillator design, that is, phase noise measurement was explained along with obtainingthe noise summary.

Next, you will perform Swept HB measurements,that is Tuning Voltage vs. Oscillation Frequency and Phase Noise vs. Tuning Voltage Measurements.

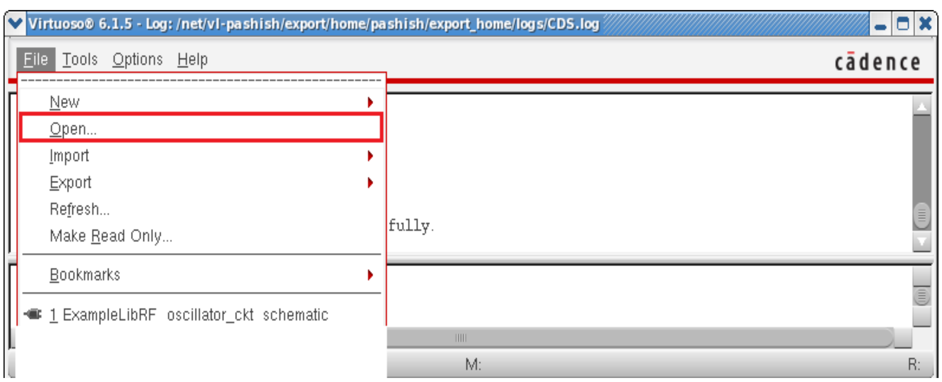
Oscillator Swept Tuning Range and Phase Noise Measurement

In this section, you will sweep the oscillator tuning range while running multiple HB/HBnoise analyses. This will give the information about phase noise vs. tuning range.

Opening the Oscillator Circuit in the Schematic Window

1. In the Command Interpreter Window (CIW), choose *File - Open*.

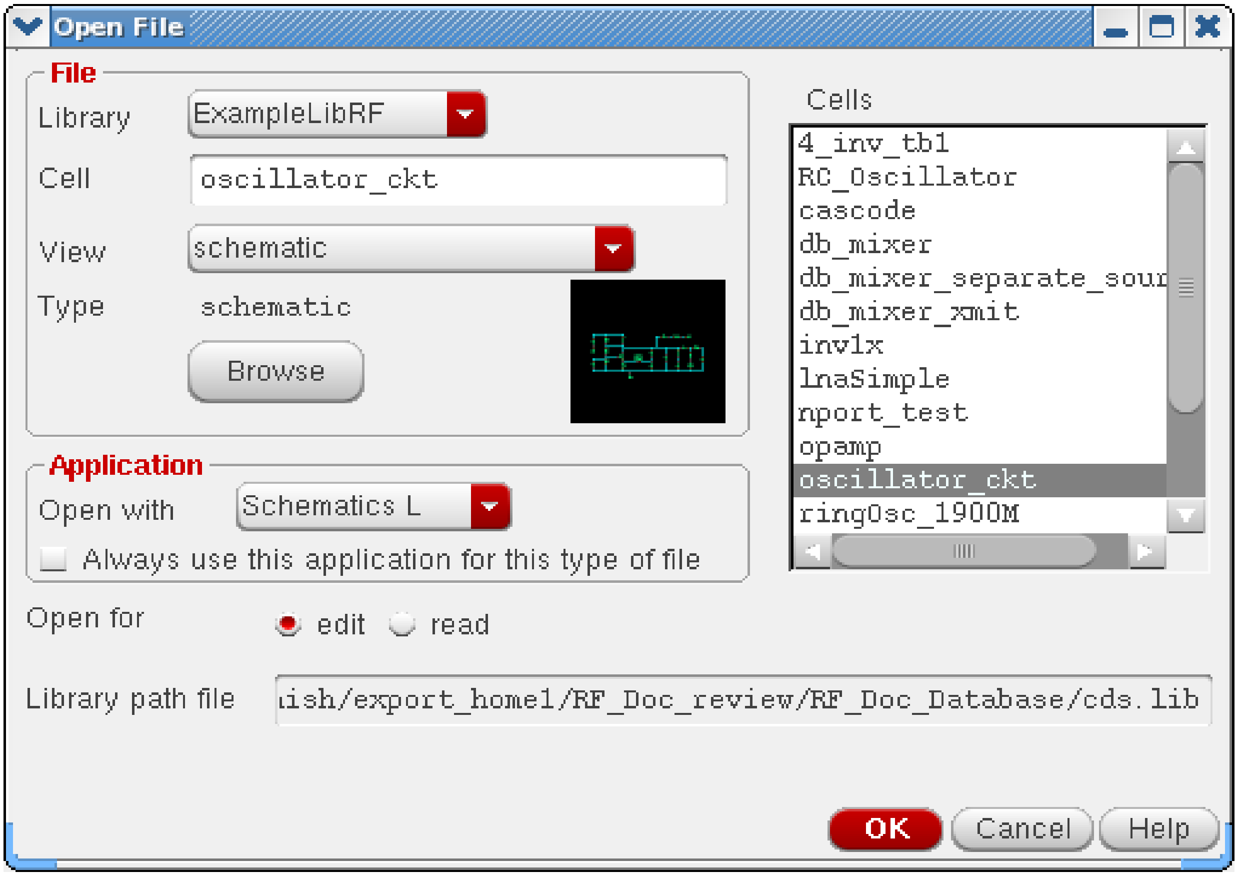
Figure A-185 Virtuoso CIW Window - Opening Cellview



The Open File form is displayed.

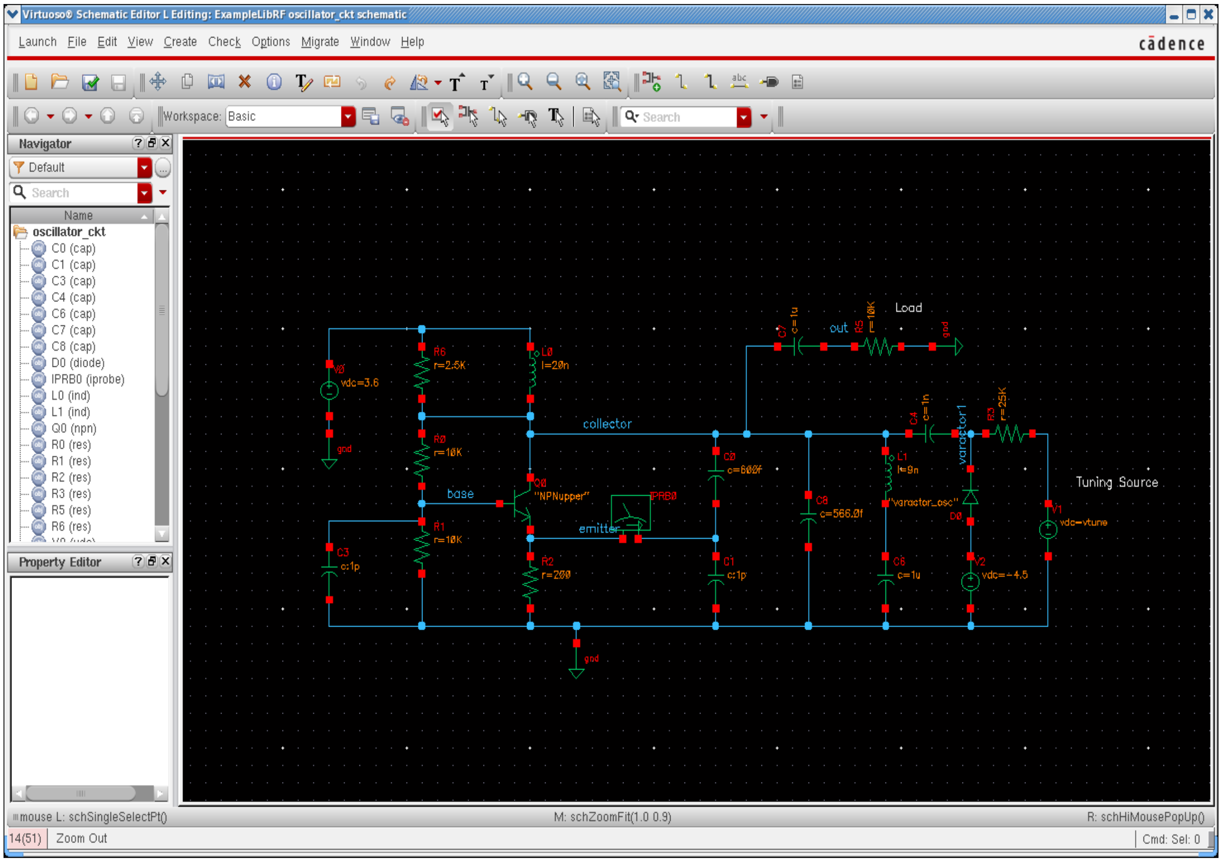
Figure A-186 Open File Form to open the oscillator_ckt cell's Schematic View

.



2. Select *ExampleLibRF* from the *Library* drop-down list.
3. In the *Cells* field, type *oscillator_ckt*.
4. Choose *schematic* from the *View* drop-down list.
5. In the *Application* section, select *Schematic-L from the Open With* drop-down list.
6. Leave *Open For* to *Edit* (which is set by default)
7. Once all the setup is done. Click *OK* to close the *Open File* form.
8. This will open the oscillator_ckt schematic in Virtuoso Schematic Editor L window.

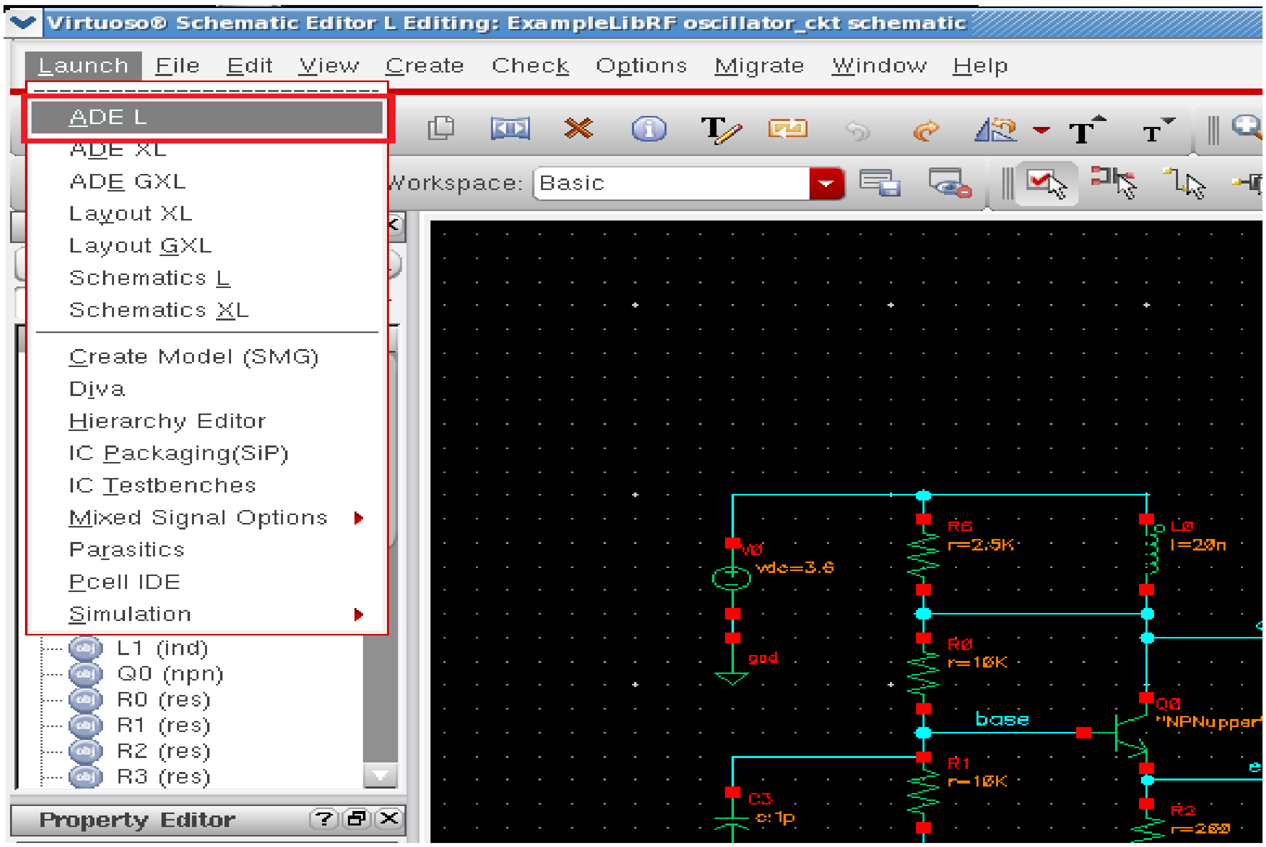
Figure A-187 Oscillator Schematic in VSE-L Window



Setting up the HB and HBnoise Analysis

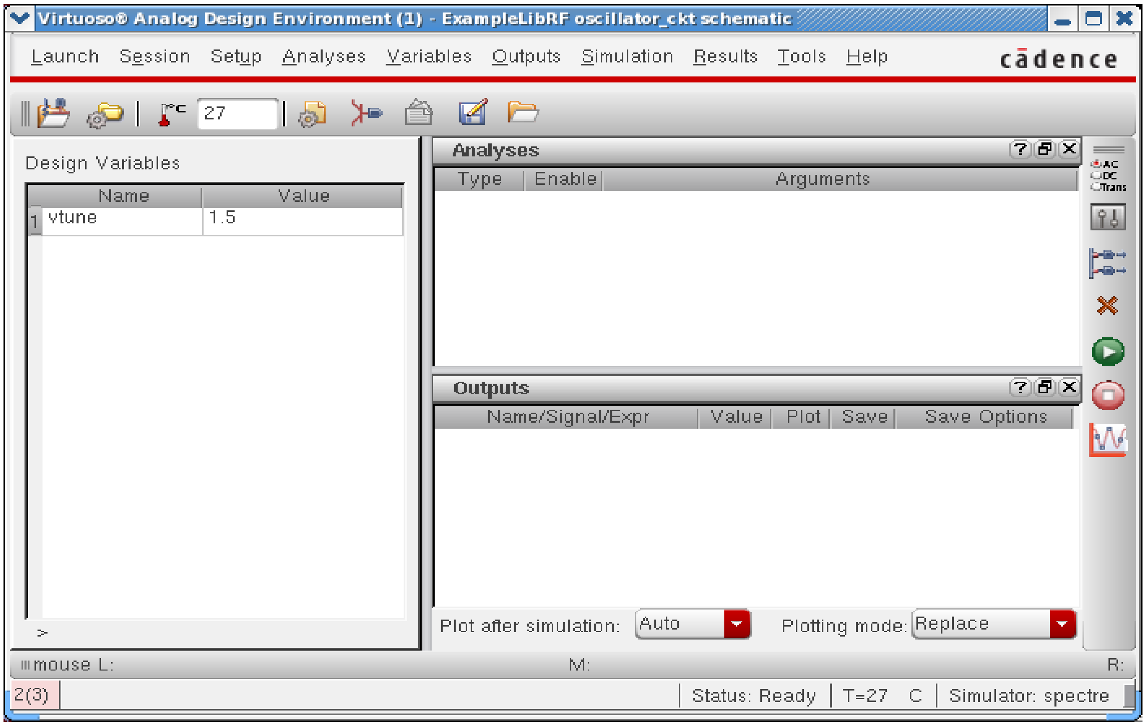
1. In the Schematic Window, choose *Launch - ADE-L.*

Figure A-188 Opening ADEL window from VSE window



2. The *Virtuoso Analog Design Environment* Window is displayed, as shown below.

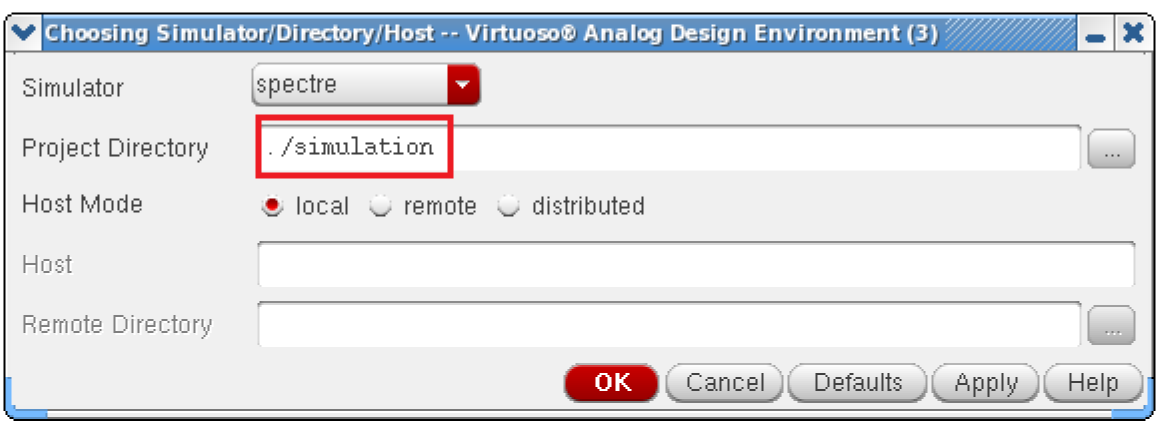
Figure A-189 Virtuoso Analog Design Environment Window



3. Choose *Setup - Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

Figure A-190 Choosing Simulator/Director/Host Form



4. Specify the following:

a. Choose *spectre* for the *Simulator*.

b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

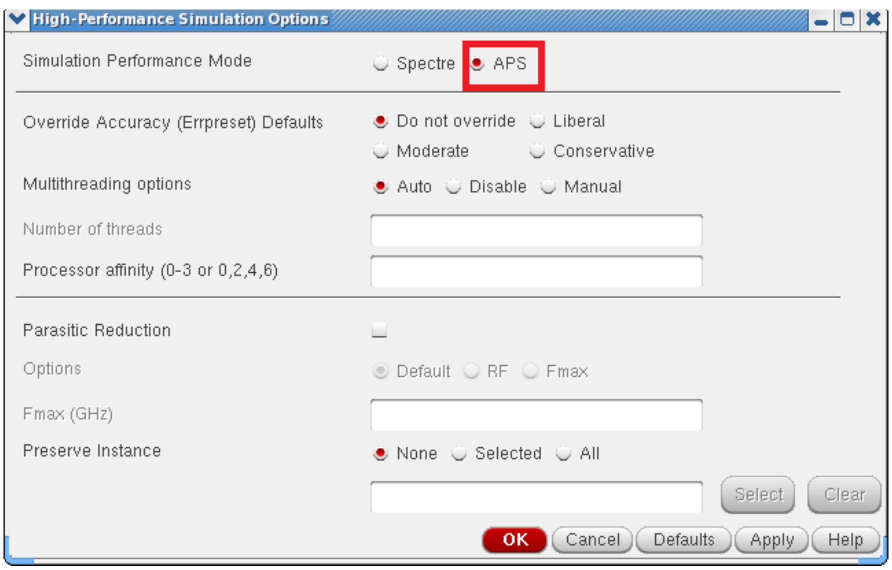
c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, please contact your Cadence tools System Administrator for specific set-up instructions.

5. Click *OK* to close the *Choosing Simulator/Directory/Host* form.

6. Set up the High Performance Simulation Options.

   In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* form is displayed.

Figure A-191 High Performance Simulation Options Form



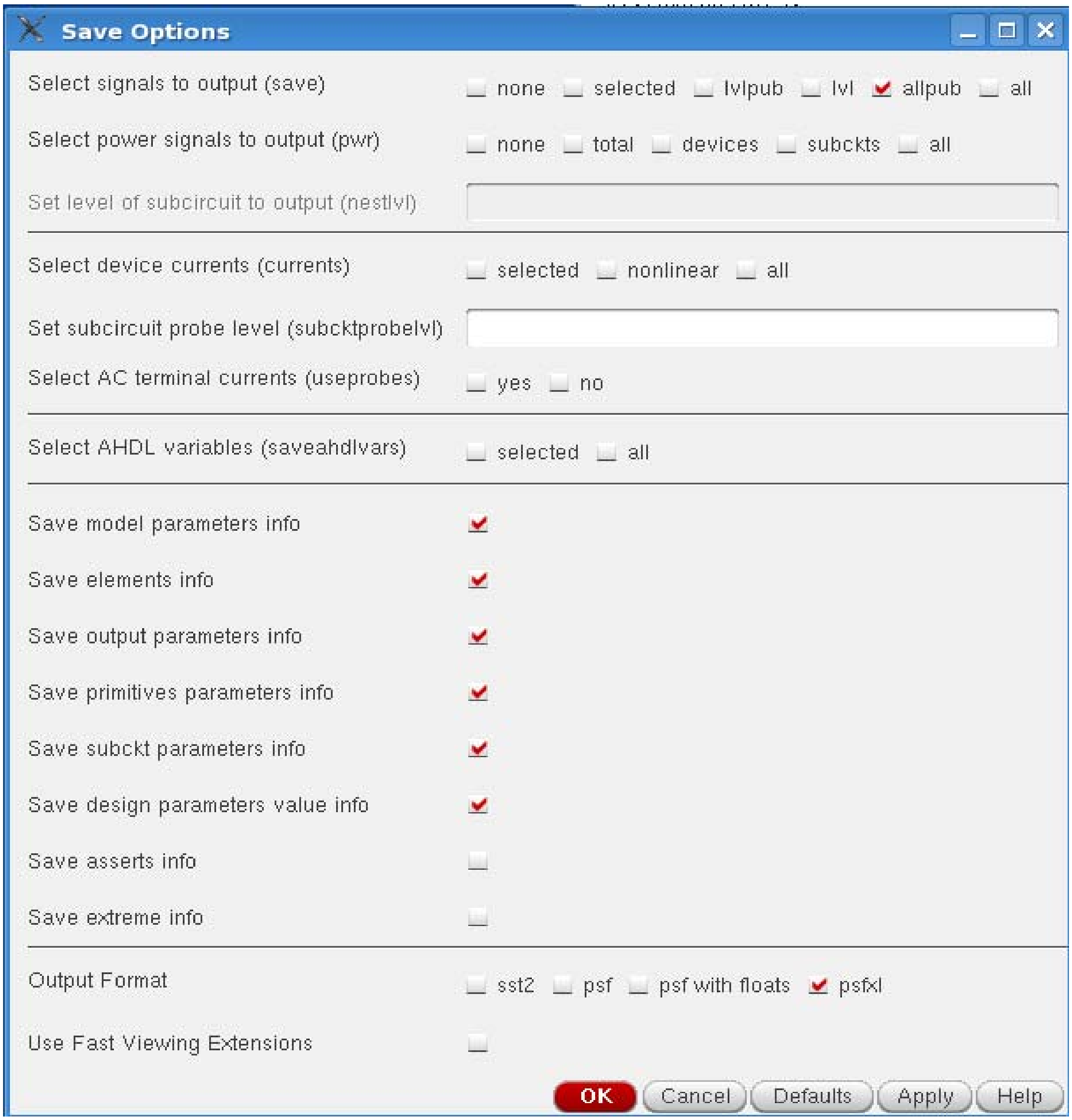
7. In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually, it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

8. Click *OK* to close the *High Performance Simulation Options* form.

9. In the *Virtuoso Analog Design Environment* window, choose *Outputs - Save All*.

The *Save Options* form is displayed.

Figure A-192 Save Options Form



10. In the *Select signals to output (save)* section, make sure that *allpub* is selected.

This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

To save the currents, use the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or all if you want to save all the currents in the circuit. When you save currents, more disk space is required for the results file.

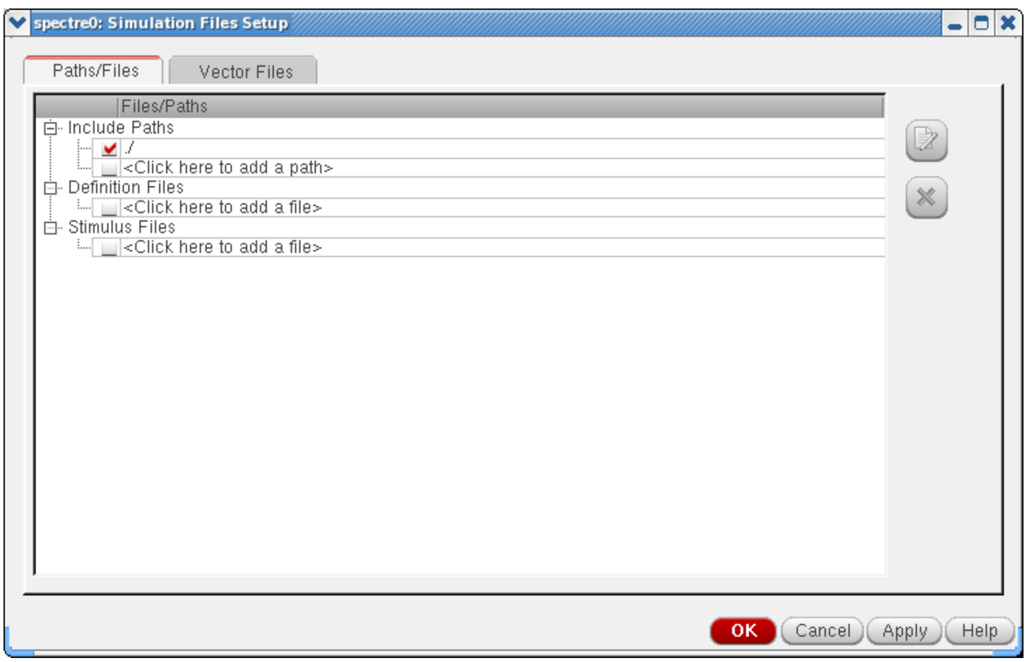
11. Click *OK*.

The Save Options form closes.

12. In the Virtuoso Analog Design Environment window, choose *Setup - Simulation Files.*.

The *Simulation Files Setup* form is displayed, as shown below.

Figure A-193 ***Simulation Files Setup*** **Form**

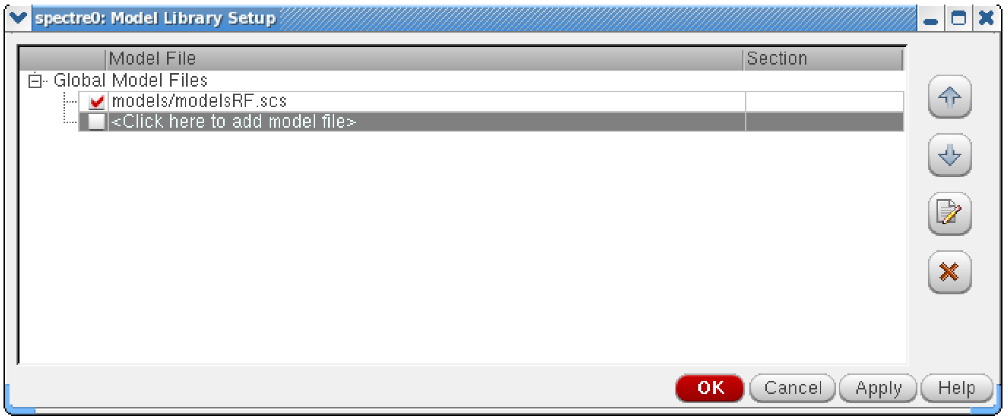


13. In the *Simulation Files Setup* form, enter ./ by clicking in the *Include Paths* section.

14. Click *OK* to close the *Simulation Files Setup* form.

15. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*The *Model Library Setup* form is displayed.

Figure A-194 *Model Library Setup* Form



16. In the *Model File* field, type the path to the model file including the file name, as follows:

models/modelsRF.scs

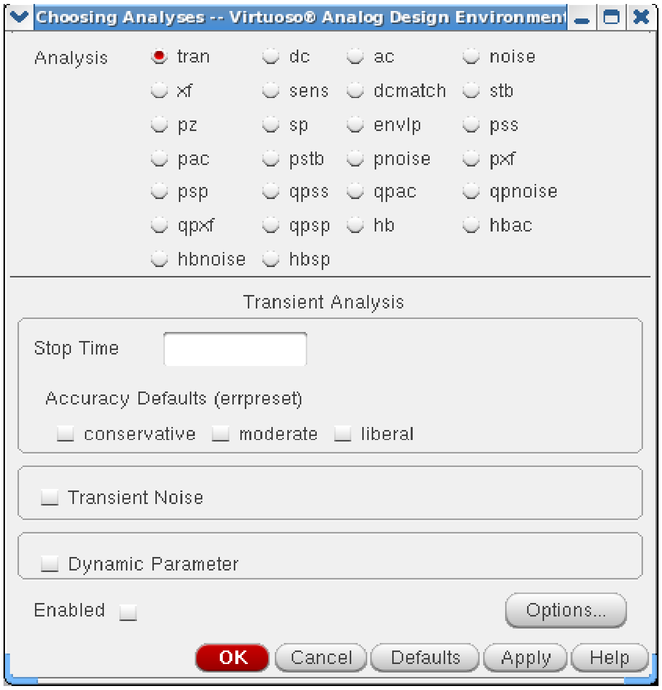
You can also browse to the *modelsRF.scs* file.

17. Click *OK* to close the *Model Library Setup* form.

18. Choose *Analyses - Choose.* in the *Virtuoso Analog Design Environment* window.

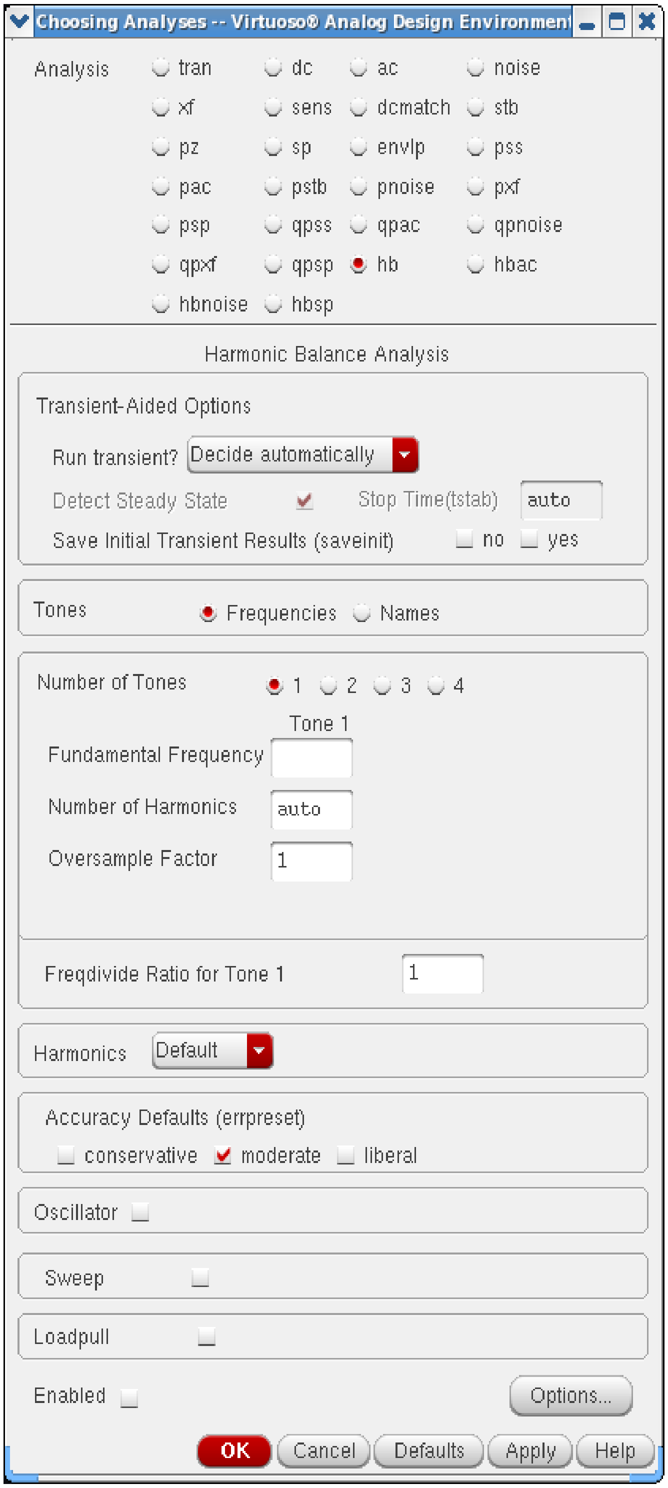
The *Choosing Analyses* form is displayed.

Figure A-195 The Choosing Analyses Form



19. Select *hb* in the *Analysis* section. The form expands.

Figure A-196 The Choosing Analyses Form - Setting *hb* analysis



20. In the *Transient-Aided Options,* leave *Run Transient?* as *Decide automatically* which is set by default. The other choices for this option are *Yes* and *No*. If *Yes* is selected, then the *Detect Steady State* option becomes active and the *Stop Time(tstab)* field is also activated. This means that you can decide whether you would like the Steady State to be detected automatically during the transient run or not and also specify a *Stop time(tstab)* for that transient run. When you select the checkbox for *Detect Steady State* option, this will run *tran* until steady-state is detected and then switches to *hb*.

21. Select *yes* for the *Save Initial Transient Results* (*saveinit*) option. This will help in visualizing the buildup of the oscillation waveform.

22. Select *Frequencies* in *Tones* section (This is selected by default). The other option is *Names*. The *Names* option when selected, is similar to the *pss* Harmonic Balance analysis setup.

23. Enter 1.7G in *Fundamental Frequency* field. Set the *Number of Harmonics* option to *20*. When *Number of Harmonics* is set to *auto,* the simulator calculates tone-1 harmonics automatically. The calculation is based on Fourier analysis of transient steady-state waveforms.

24. Leave the *Oversample Factor* option to 1 by default. Since the oscillator has sinusoidal waveforms, an oversample of 1 is appropriate.

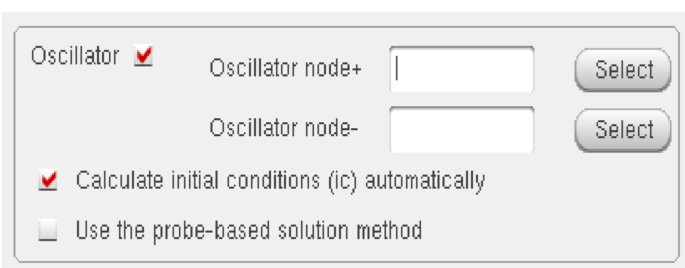
25. Set the *Freqdivide Ratio for the Tone 1* option to 1 as there is no frequency divider in the circuit. If there is a frequency divider in the circuit then you need to set the *Freqdivide Ratio for Tone 1* to the divide ratio of the divider. For example, if the divider is divide-by-two then the divide ratio is 2. Therefore, you will set *Freqdivide Ratio for Tone 1* to 2.

26. *Leave the Harmonics* option as is which is set to *Default*.

27. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. Conservative is recommended for all the oscillators.

28. Select *Oscillator*. This is required for simulating an autonomous circuit.

Figure A-197 The *Choosing Analyses* Form - Oscillator Section

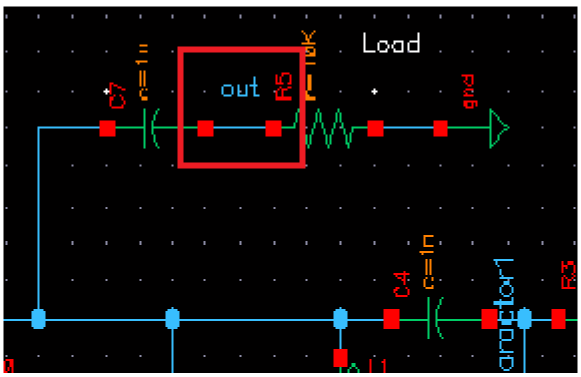


Selecting the Oscillator button changes the Tone1 name in the Tones section to osc!.

29. In the Oscillator section, in *Oscillator node* field, click *Select* just to the right of this field. In the schematic, select the *out* node. Instead of selecting the node from the schematic you can also type */out* in the *Oscillator node* field. This oscillator node will be used by Spectre for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it. Leave the *Reference node* blank.

Note: If you have a single-ended oscillator, only specify one node. If the second node, that is, the *Reference node* is left blank, it will be connected to the global ground node automatically. However, if you have differential oscillator, you need to specify both the nodes.

Figure A-198 Selecting *out* net from the schematic



30. If you have an LC oscillator, leave the *Calculate initial conditions (ic) automatically* checkbox selected (this is the default).

31. Note that *Calculate initial conditions (ic) automatically* is used to start the oscillator. Other methods to start the oscillator include putting a single current pulse into the resonator, setting initial conditions(ic), or ramping up the power at time = zero plus.

32. Note that by default *Use the probe-based solution method* (*oscmethod*) option is deselected (or unselected). Spectre will use the *onetier* method. In *onetier* method, the frequency and voltage spectrum are solved simultaneously in one single set of nonlinear equations.

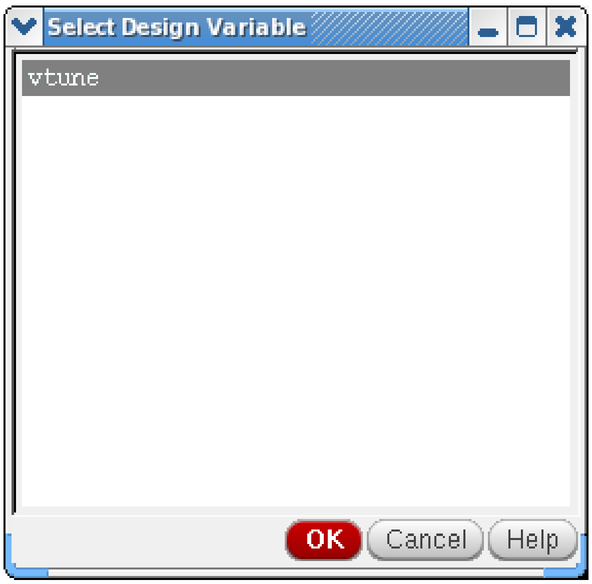
Note: When the*Setting Use the probe-based solution method* (*oscmethod*) option is selected/enabled, it iterates for the frequency solution in the outer loop and the amplitude and phase solution in the inner loop. The probe-based method has better convergence but is computationally intensive.

Refer to *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more details.

33. Click *Sweep*. This will enable the sweeping of tuning voltage (in this case it is named as vtune) which is explained below.

34. Click *Select Design Variable* . The *Select Design Variable* window is displayed, as shown below.

Figure A-199 Choosing vtune in Select Design Variable Form during HB Analysis setup



35. In the Select Design Variable window, select *vtune*.

a. Click *OK* to close the *Select Design Variable* form.

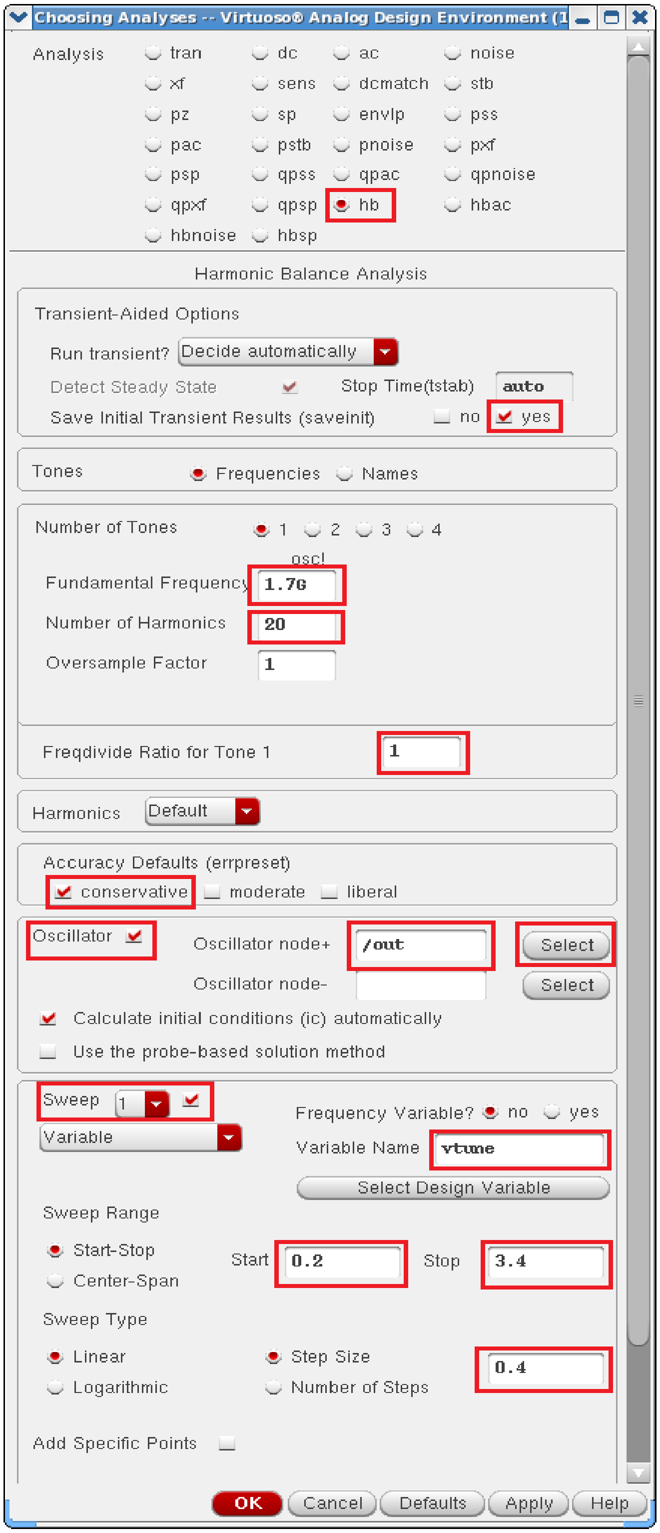
b. In the *Sweep Range* section, type *0.2* in the *Start* field.

c. Type *3.4* in the *Stop* field.

d. By default, the *Sweep Type* is set to *Linear*. Type *0.4* in the *Step Size* field.

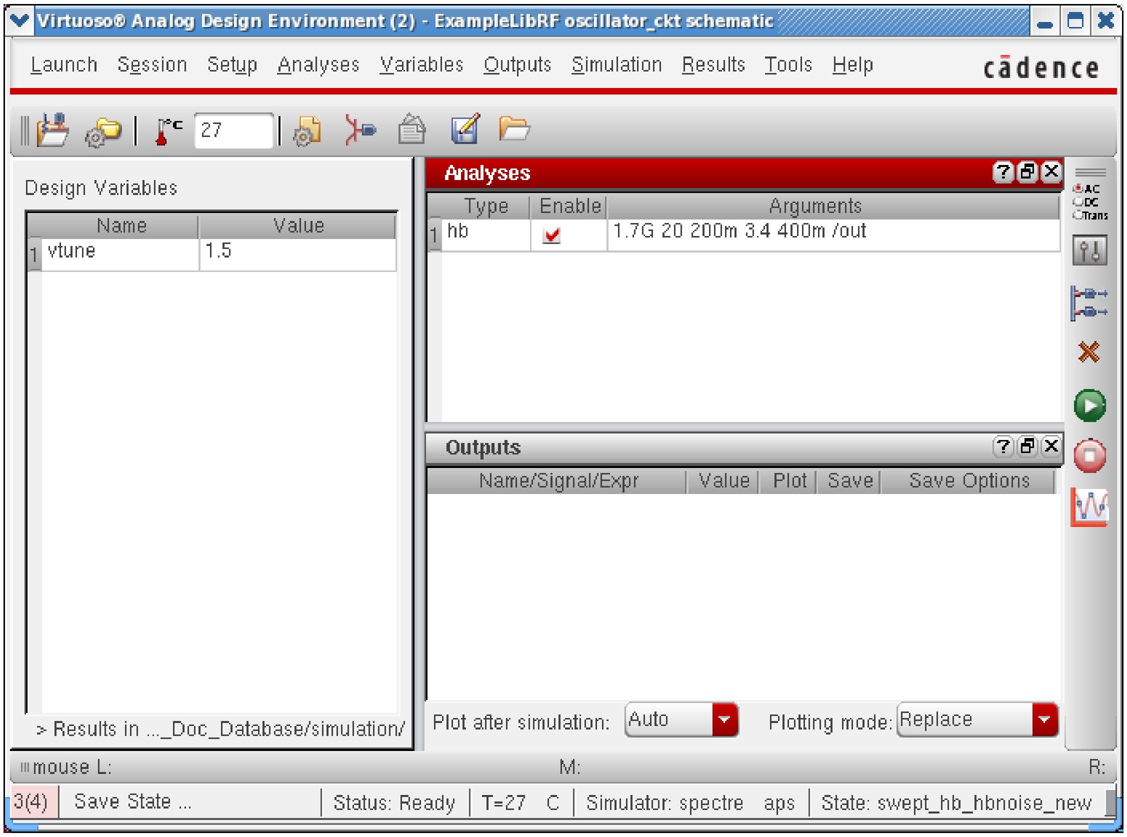
The *Choosing Analyses* form should like the figure below:

Figure A-200 Choosing Analysis Form - swept *hb* Analysis Setup



Click *Apply* at the bottom of the *Choosing Analyses* form. This will add *hb* analysis with sweep setup in the *Analyses* section of ADE window, as shown below.

Figure A-201 ADE Simulation Window - *hb* analysis setup

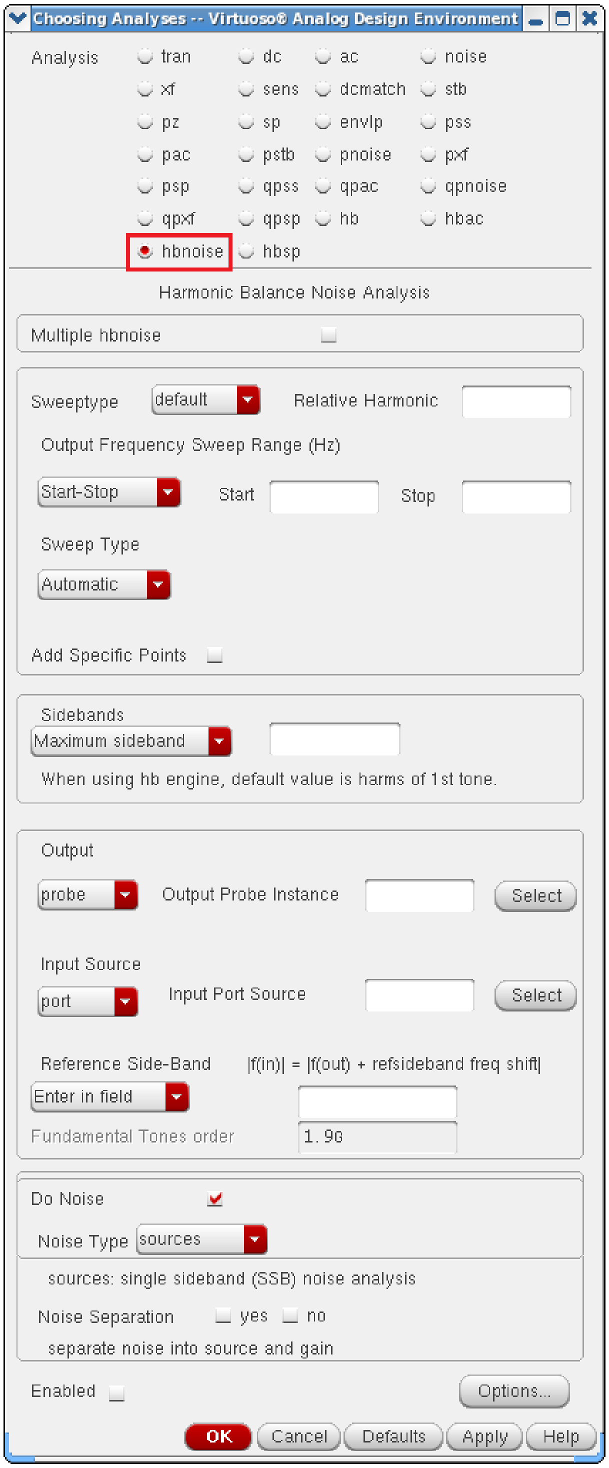


Setting up the HBnoise analysis

This analysis is set to do the phase noise measurement. *hbnoise* analysis is a small signal analyses run after *hb* analysis.

1. In the *Choosing Analyses* form, select *hbnoise*. The form expands.

Figure A-202 The Choosing Analyses Form - *hbnoise* Analysis Setup



2. Leave the *Sweept Type* to *default*.

For oscillators, the *hbnoise/pnoise* frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the hb/pss form, you should specify the approximate frequency of oscillation for the frequency-divided signal. In the *hbnoise/pnoise* form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of *relative* is to take the frequency of the harmonic number specified and add to it the frequencies specified in the *Choosing Analyses* form. If the oscillator had a 1GHz output, and the pnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

  a. Type 1 in the *Relative Harmonic* field as you are simulating an oscillator by itself.

  b. Select *Single Point* for *Output Frequency Sweep Range.* Type 1M in the *Freq* Field. In general, the hbnoise frequency would be set as appropriate for your application.

3. Leave the *Maximum Sideband* field blank. In general, *Maximum sideband* needs to be set high enough to include all the frequencies that could mix down to the oscillator output frequency. By default, it should match the number of harmonics set in HB analysis.

4. Set the *Output* to *voltage*.

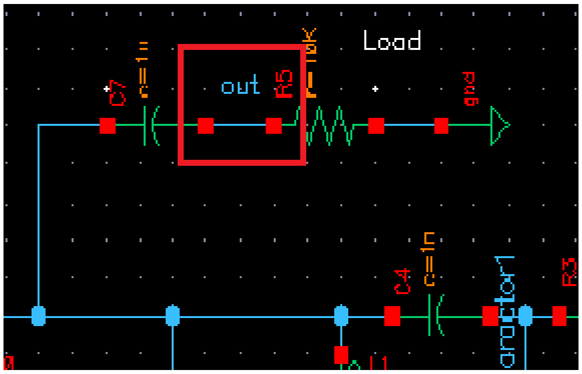
  a. Type /out in the *Positive Output Node* field. You can also select *out* net from schematic by clicking the *Select* button on the right of the *Positive Output Node* field and then selecting the net just below the *out* label in schematic.

  b. Leave the *Negative Output Node* field blank. If the second node, that is, the *Negative Output Node* is left blank, it will be connected to the global ground node automatically. However, if you have differential oscillator, you need to specify both the nodes.

  c. Set the *Input Source* to *none*.

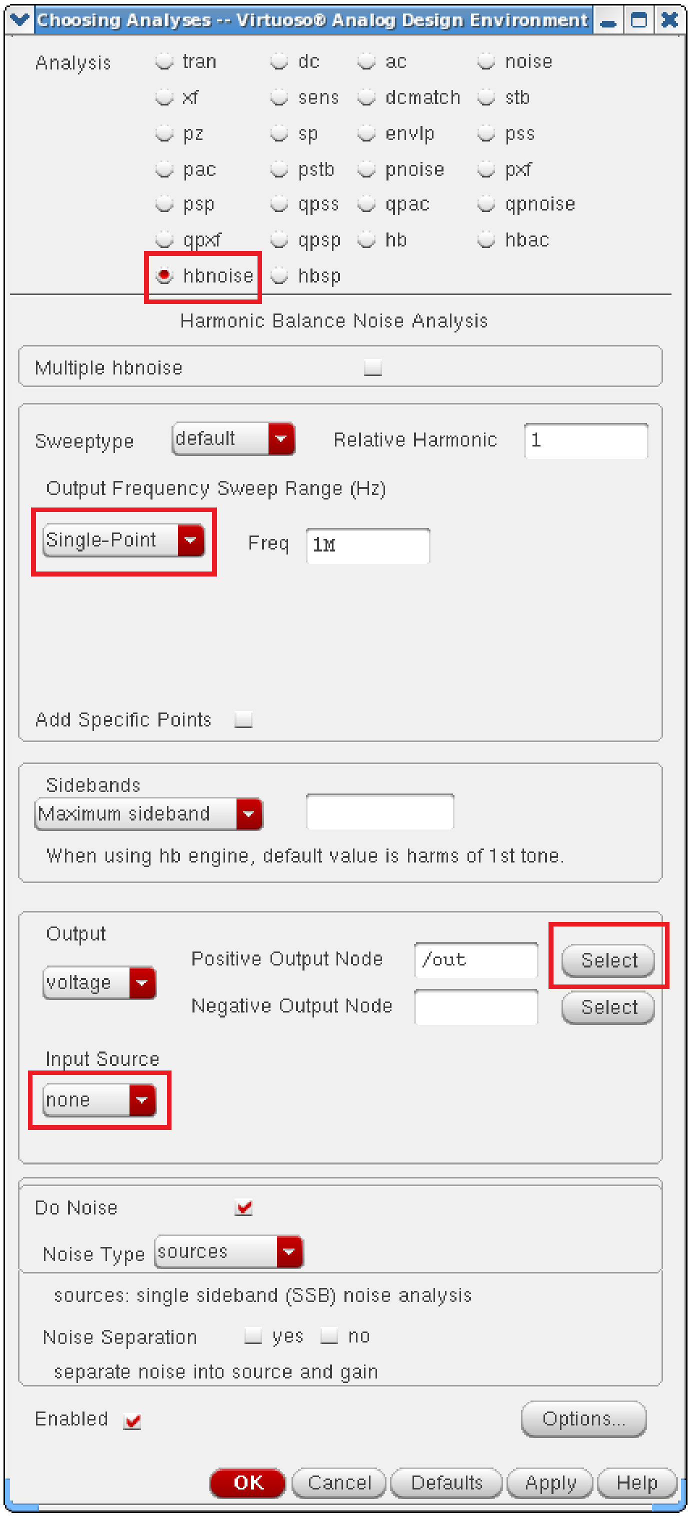
  Using an Input Source generates NF, IRN, and other parameters which are not needed here. Input Source is not usually used in oscillator simulations.

Figure A-203 Selecting *out* net from the schematic



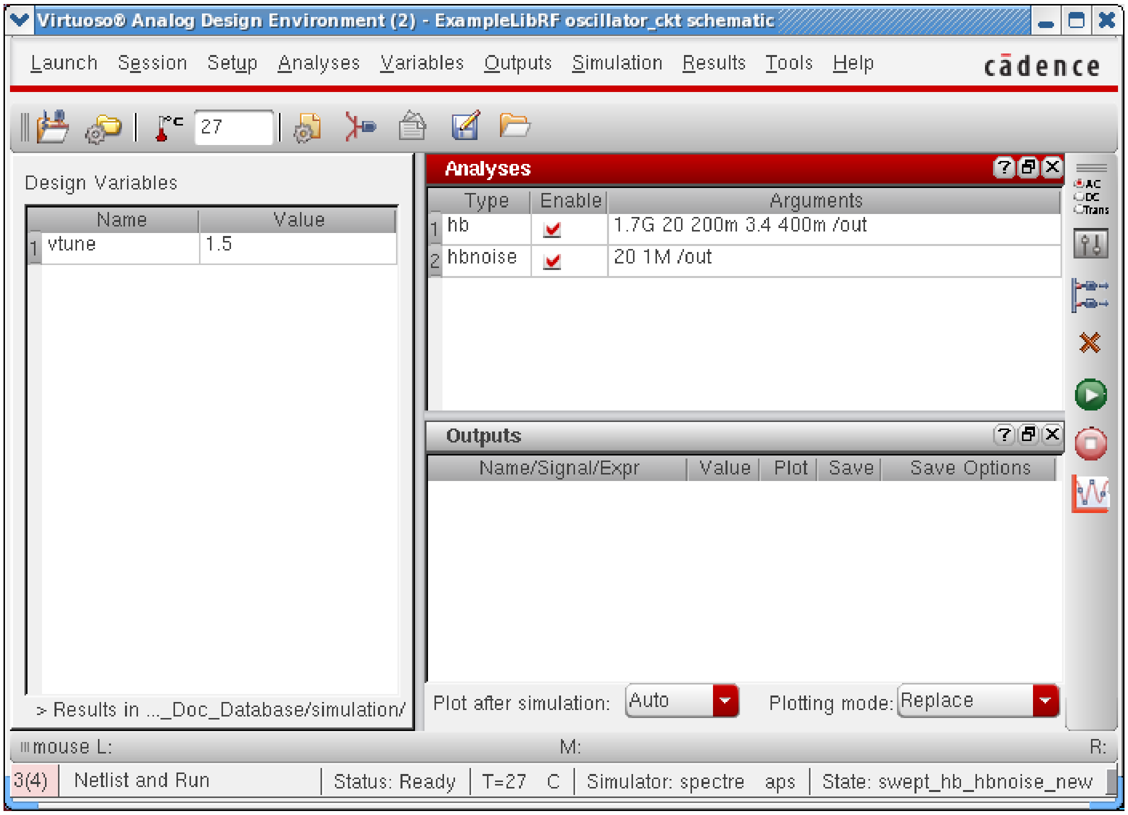
The *Choosing Analyses* Form should look like the following:

Figure A-204 Choosing Analysis Form - *hbnoise* Analysis Setup

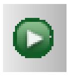


5. Click *OK* to close the *Choosing Analyses* form. This will add *hbnoise* analysis along with *hb* analysis in the *Analyses* section of ADE window, as shown below:

Figure A-205 ADE Simulation Window - *hb* and *hbnoise* analysis setup



Running the simulation

Once finished setting up the *hb* and *hbnoise* analyses click the gree icon  on the right of the *Virtuoso Analog Design Environment* window or on the Schematic window to run the simulation.

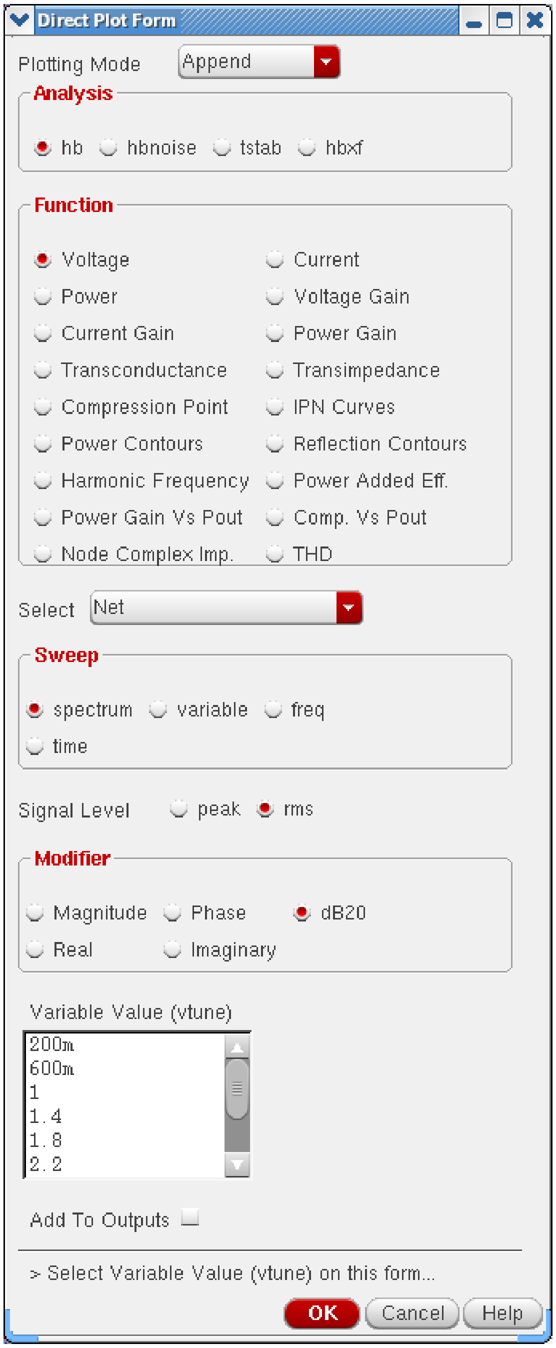
Next, plot the results when the simulation is finished.

Plotting the results

First Plot the oscillator output frequency -

1. In the *Virtuoso Analog Design Environment* window, choose *Results -- Direct Plot - Main Form*.

Figure A-206 Swept *hb* and *hbnoise* Analysis Direct Plot Form



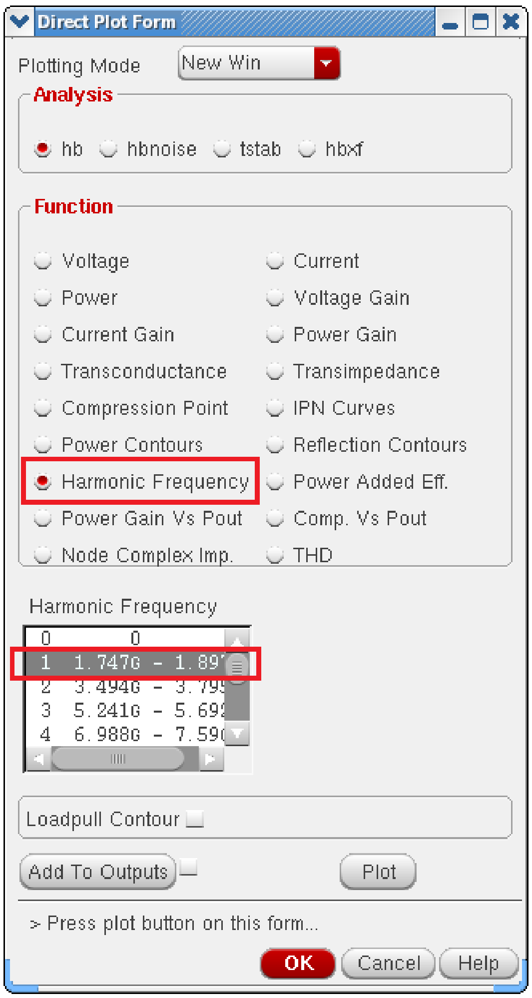
2. In the *Direct Plot Form*, select *hb* in the *Analysis* section. This is selected by default.

3. Select *Harmonic Frequency* in the *Function* section.

4. In the *Harmonic Frequency* section, select the first harmonic. This will plot only the change in first harmonic of the oscillation frequency vs. change in oscillator's tune voltage.

The *Direct Plot Form* setup should look like the following:

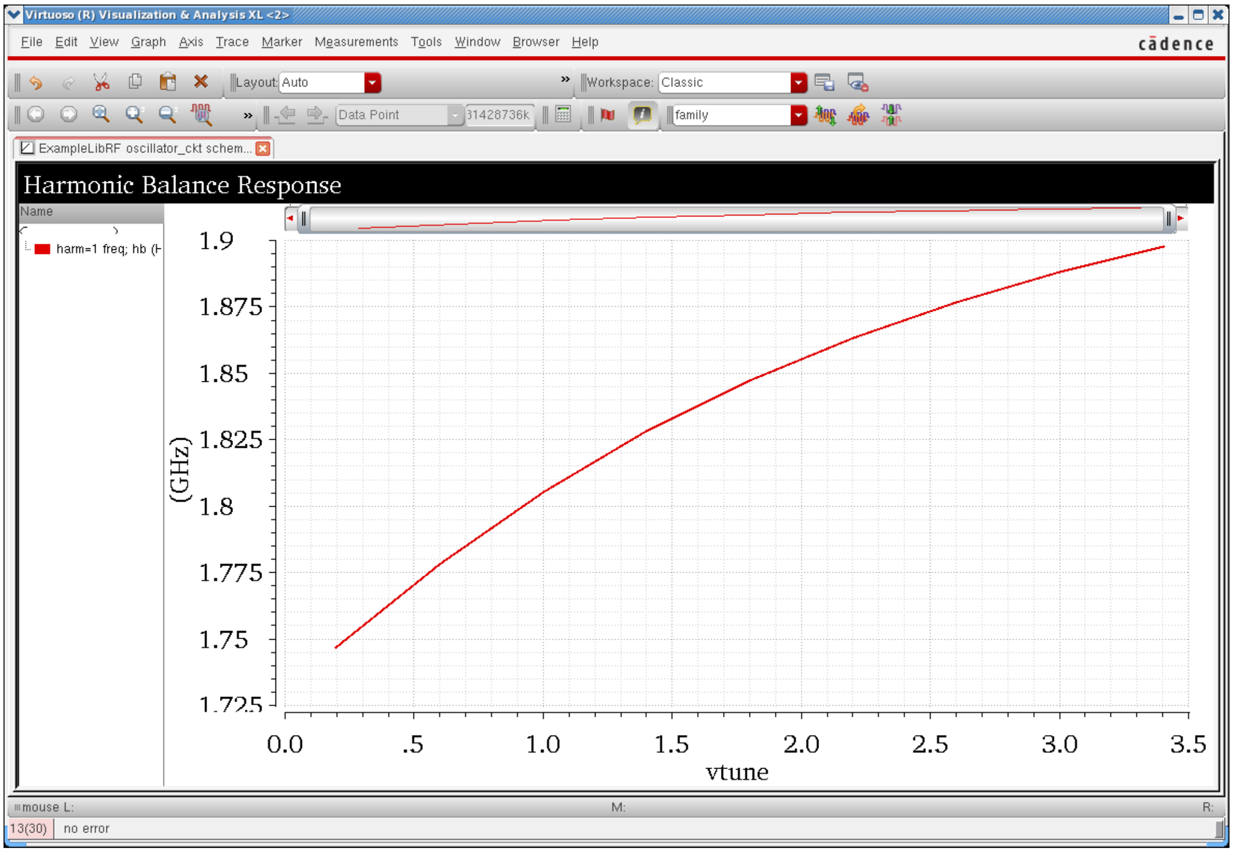
Figure A-207 Swept *hb* Analysis Direct Plot Form Setup



5. Click *Plot*.

The oscillator tuning range is plotted. You can see from the plot that as you increase the tuning voltage, the oscillator frequency increases.

Figure A-208 Swept HB Measurement Plot - vtune vs. osc frequency variation plot



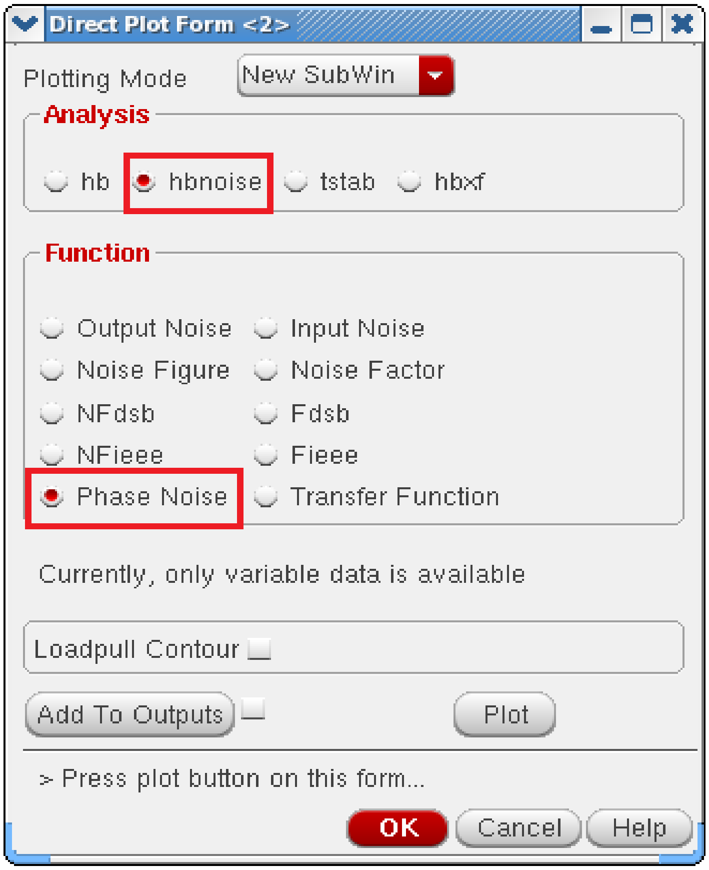
6. Choose *File - Close All Windows* to close ViVA.

Next plot the phase noise.

1. In the *Direct Plot Form*, select *New SubWin*.
2. Select *hbnoise* in the *Analysis* section.
3. Select *Phase Noise* in the *Function* section.

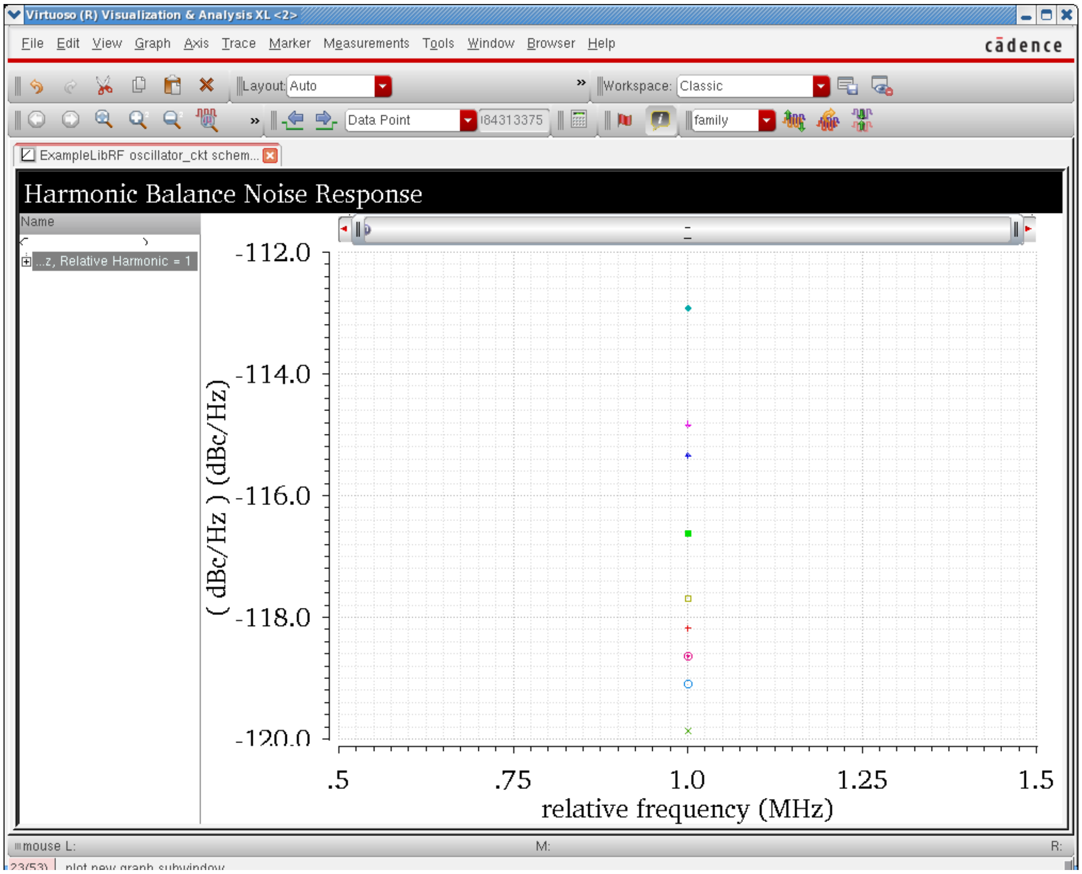
   The *Direct Plot Form* should look like the following:

Figure A-209 ***hbnoise*** **Direct Plot Setup**



4. Click *Plot*. This will plot the Single Sideband Phase Noise vs. Relative Frequency graph, as shown below.

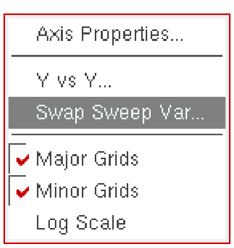
Figure A-210 Phase Noise vs. Relative Frequency Plot



In the above plot, you have a graph of phase noise vs. relative frequency. Next, you will modify the phase noise plot to provide more useful information. Here are the steps to plot phase noise vs. tuning voltage:

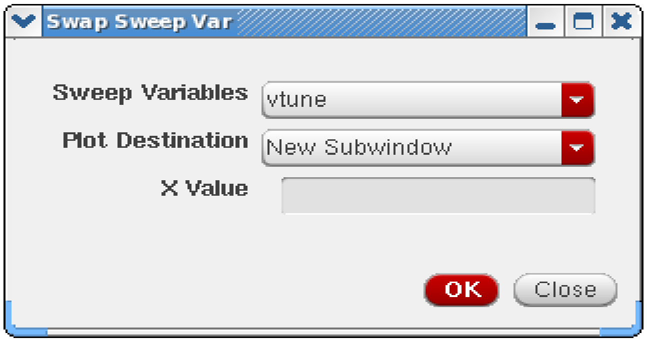
5. In the plot window, click the right mouse button on one of the numbers on the X Axis, and select *Swap Sweep Var* from the context menu.

Figure A-211 X-Axis - Right Mouse Button (RMB) - Object Menu



6. In *Select Sweep Var* dialog box which opens, select *vtune* from the *Sweep Variables* drop-down list. Keep *Plot Destination* as *New SubWindow*.

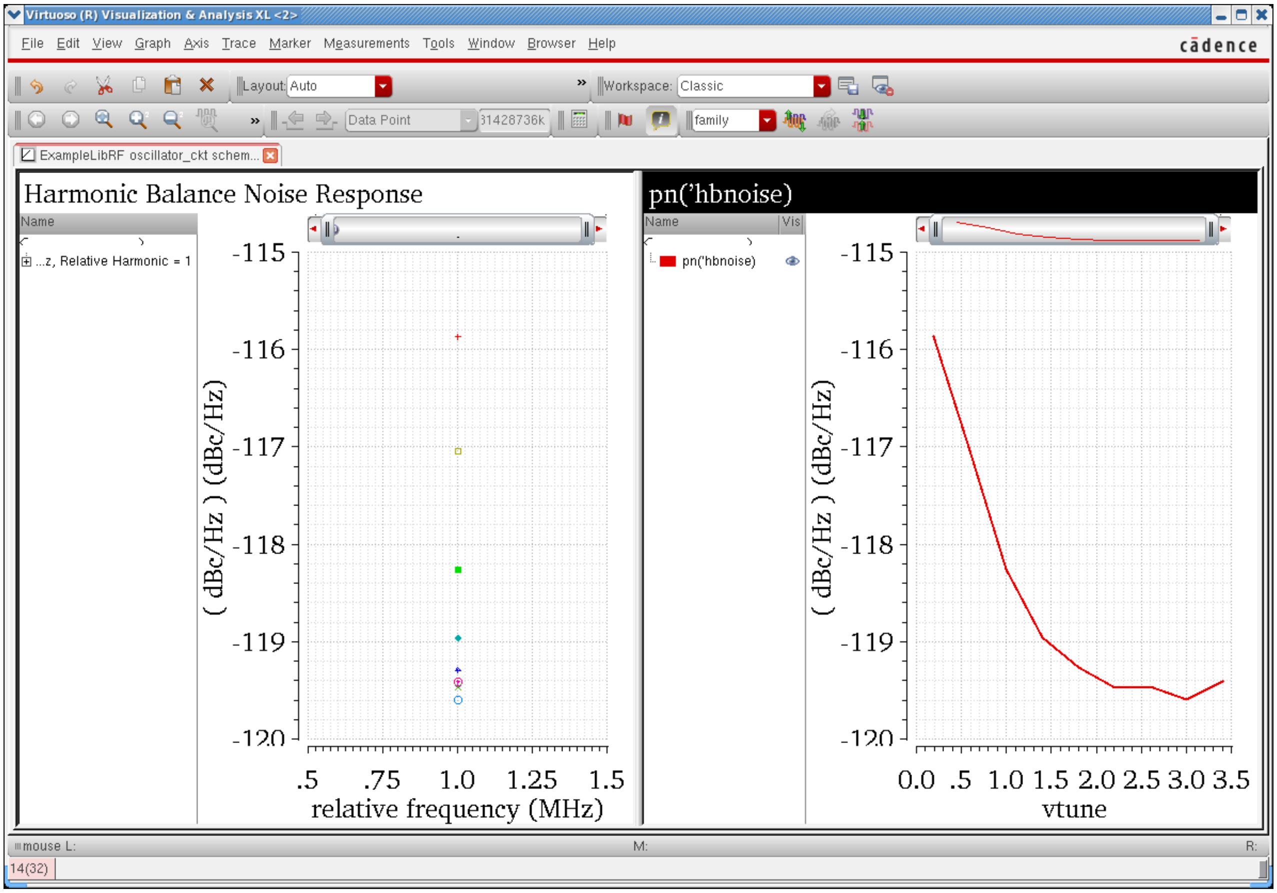
Figure A-212 Swap Sweep Var Dialog Box Window



7. Click *OK* to close the Swap Sweep Var dialog box..

   The waveform tool draws a new subwindow.

Figure A-213 vtune vs. Phase Noise Plot



If you look at the graph on the right, you will see that the phase noise is worse at the lower end of the tuning range.

8. In the *Direct Plot Form*, click *Cancel*.

9. In the waveform window, choose *File - Close All Windows*.

10. Clean up the screen for the next set of measurements.

a. Close the Analog Design Environment window by selecting *Session - Quit*.

b. Click *No* in the *Save State* window.

c. In the Schematic window, select *File - Close*.

Summary

In this section,Tuning Voltage vs. Oscillation Frequency and Phase Noise vs. Tuning Voltage Measurements were done.

Ring Oscillator Measurements

In this section, you will peform measurements on Ring Oscillator.

Starting and Stabilization of Ring Oscillators

To simulate a Ring Oscillator, it is recommended to use Shooting PSS analysis as they are highly non-linear in nature. Recent improvements in harmonic balance allows ring oscillators to be simulated in hb as well. In this case, set the oversample factor to 4. In the tstab interval, or in the transient analysis, measure the fastest slew rate for the rising and falling edges, and calculate the zero to 100% transition time based on this slew rate measurement. Set the number of harmonics to at least the period of the oscillations divided by the transition time. You can first start the oscillator by supplying initial conditions(ICs). You should choose initial conditions such that a transient analysis displays the correct oscillation frequency (and amplitude) at the output of the ring oscillator. This is the actual check that your initial conditions are working to start the oscillator. As a suggestion, in a ring oscillator, you set the output of one stage high or low. If the oscillator configuration is differential, then set the output of 1 stage high on one differential net, and low on the other differential net for the same stage.

Regardless of which technique you use to start the oscillator, allow the oscillator to run long enough to stabilize before you start the Shooting phase and compute the steady state solution. Adjust the *tstab* parameter to supply the additional stabilization time, a value of 2-3 periods is recommended.

It is recommended to savethe initial tansient simulation results. This can be done by setting *saveinit=yes*. You can use this to verify that the oscillator has started up and is stable.

In the case of Injection Locked Ring Oscillator where you apply a signal (vsource) to your ring oscillator which causes the oscillator to oscillate at the injection frequency makes it a driven circuit - as it is driven by the injection source. Thus, in this case, you do not use an autonomous PSS analysis. In order to do that, you do not click the *Oscillator* option on the PSS *Choosing Analyses* form for such oscillators. SpectreRF considers this type of circuit a driven circuit and will error out if you click the *Oscillator* option on the PSS *Choosing Analyses* form.

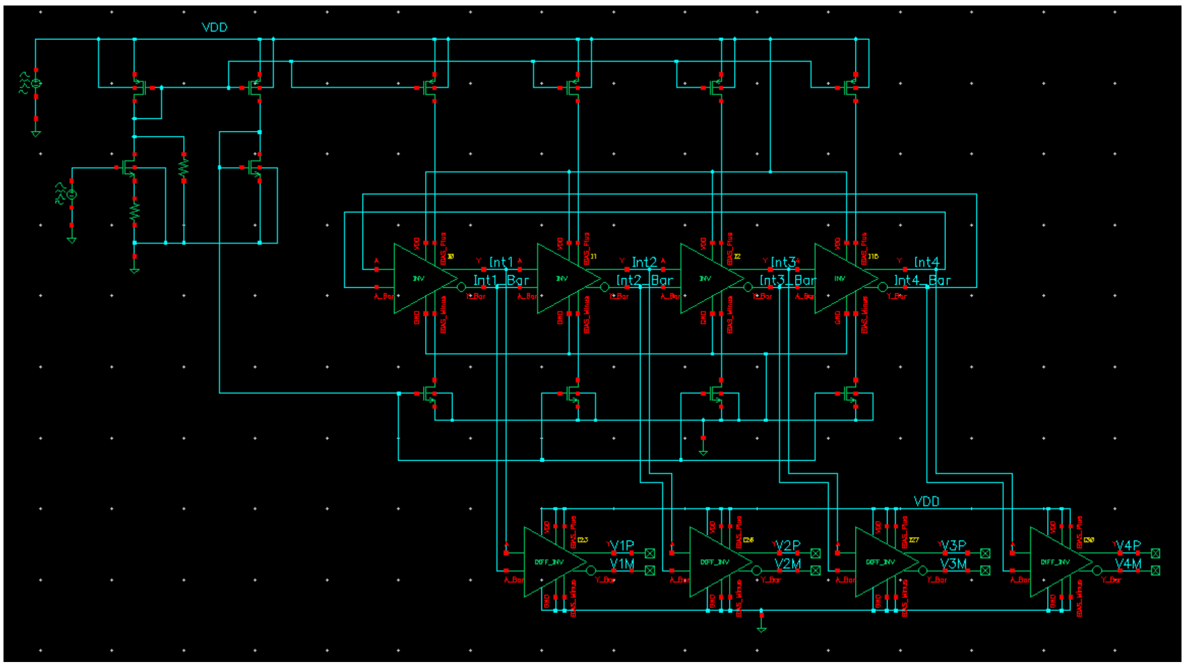
See *Oscillators and Autonomous PSS Analysis* in the *Virtuoso Spectre Circuit Simulator RF Analysis Theory* for more information on oscillator simulation.

The Oscillator Circuit

This example computes the periodic steady state solution and the phase noise for the *Ring oscillator* circuit shown in Figure A-80.

This is a four stage fully differential ring oscilaltor designed at 1.9GHz.

Figure A-214 Schematic for the Ring Oscillator Circuit ringOsc_1900M



Simulating the Oscillator Circuit

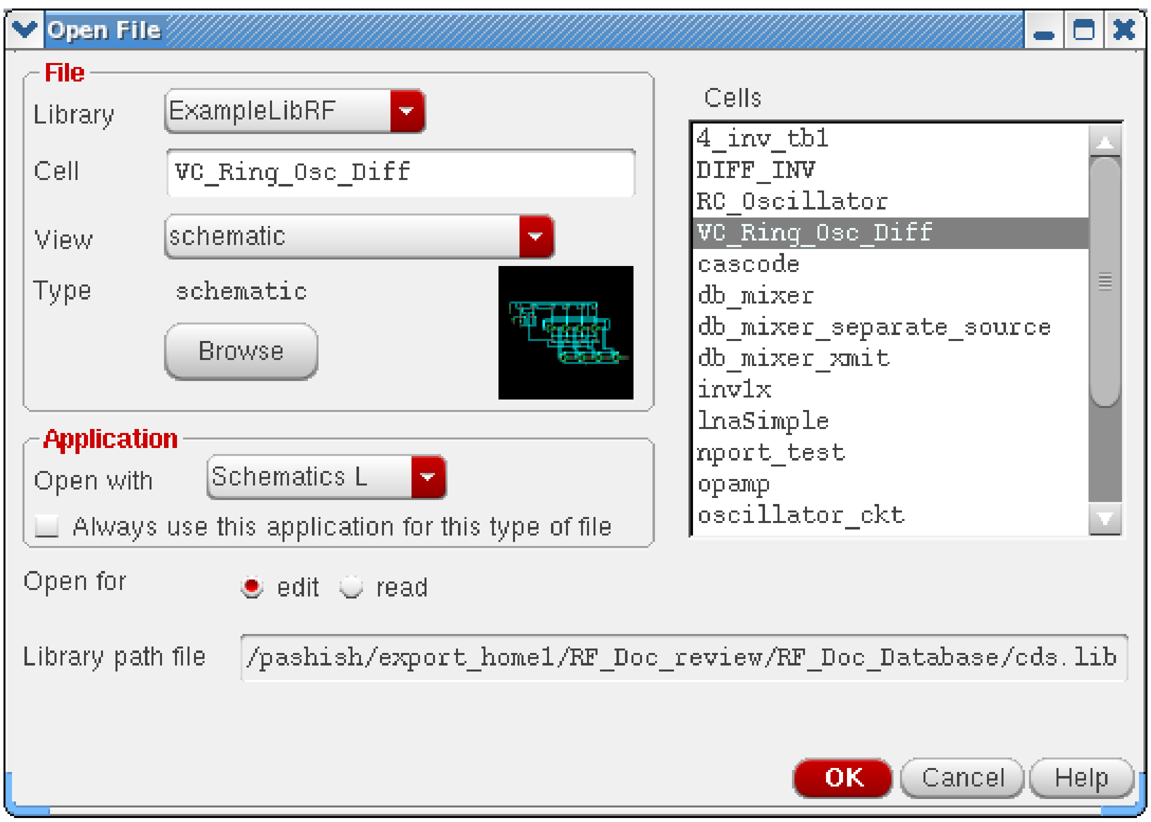
Opening the Oscillator Circuit in the Schematic Window

1. In the Command Interpreter Window (CIW), choose *File –> Open*.

Figure A-215 Virtuoso CIW Window - Opening Cellview

The *Open File* form is displayed.

Figure A-216 Open File Form to open the testbench_1900M cell's Schematic View

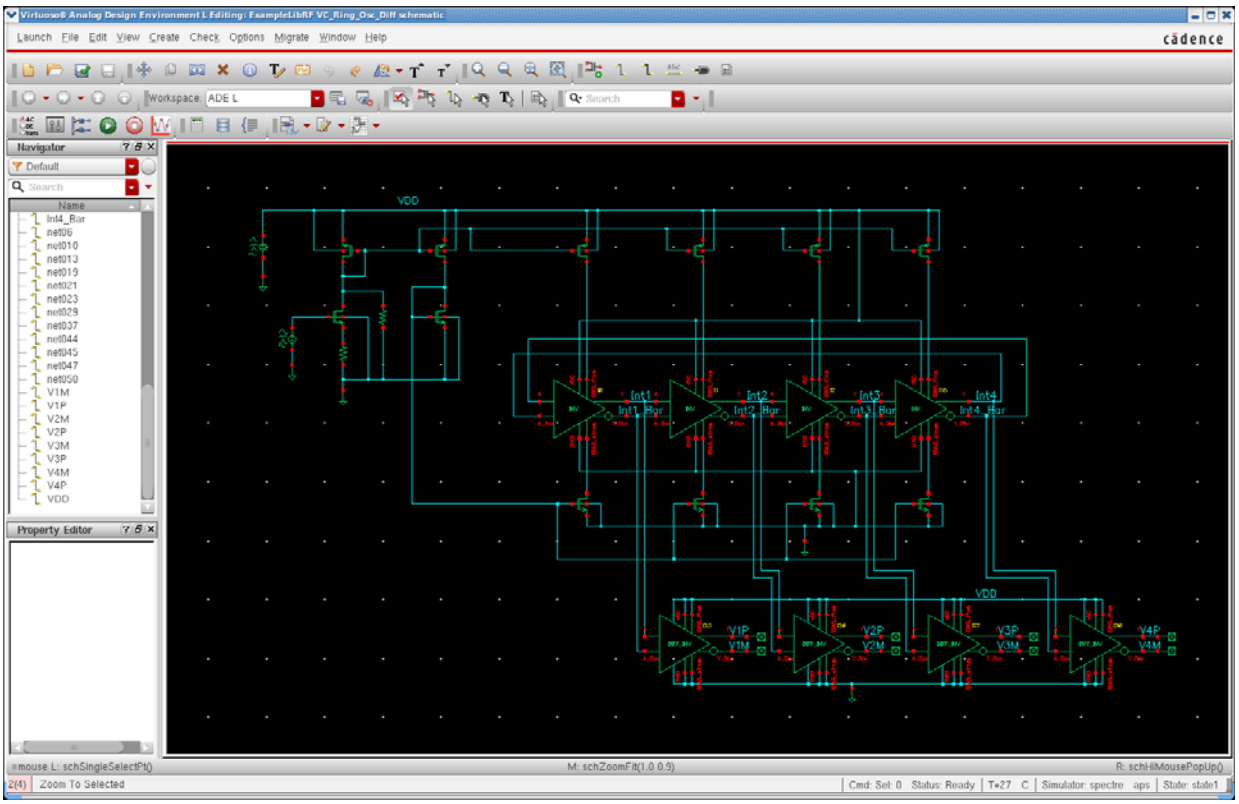


2. Select *ExampleLibRF* from the *Library* drop-down list..

3. In the *Cells* field, type *VC_Ring_Osc_Diff*.

4. Choose *schematic* from the View drop-down list.
5. In the *Application* field, select *Schematic L* from the Open *With* drop-down list.
6. Leave *Open For* to *Edit* (which is set by default).
7. Once all the setup is done. Click *OK*.
8. This will open the *VC_Ring_Osc_Diff* schematic in Virtuoso Schematic Editor L window, as shown below:

Figure A-217 *VC_Ring_Osc_Diff* schematic in VSE-L Window



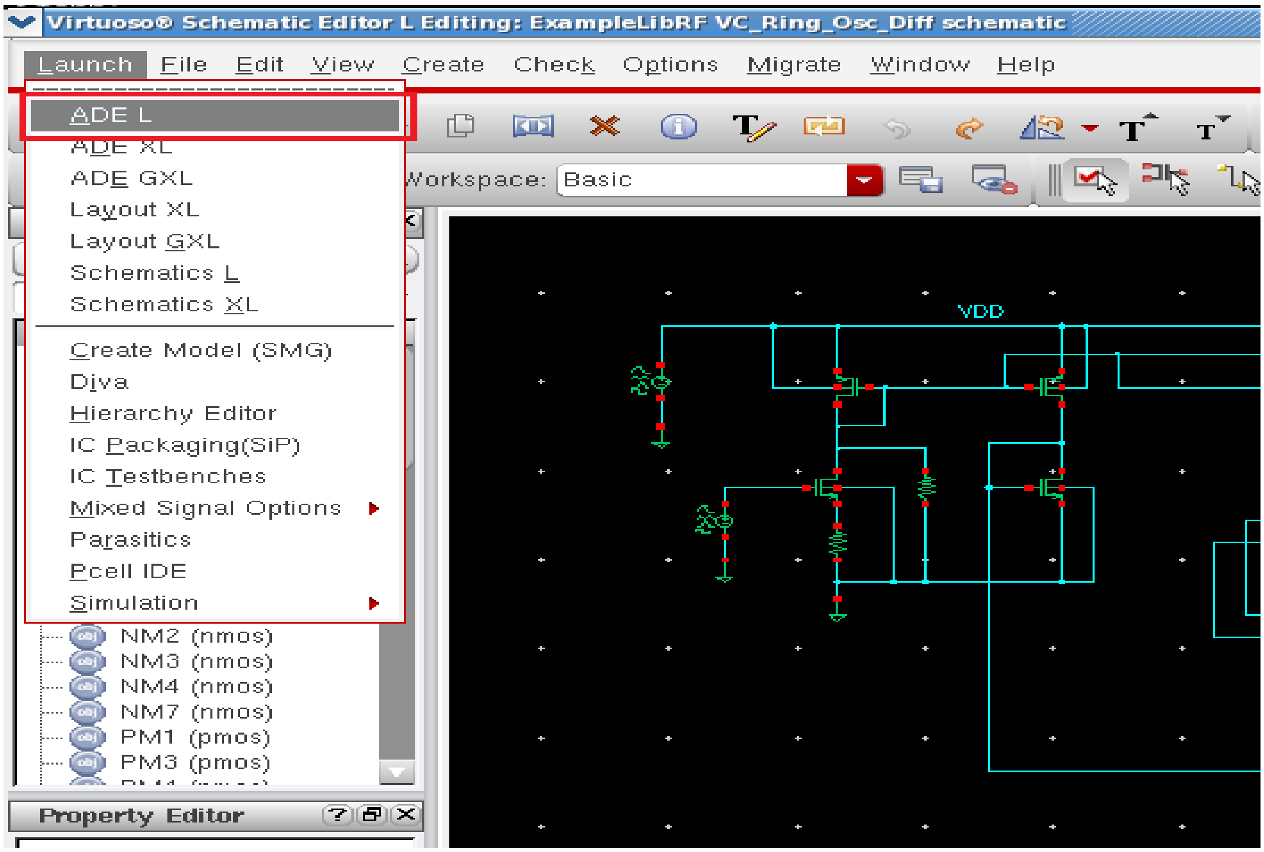
Calculating the Steady-State Solution using PSS Shooting Analysis

This example computes the periodic steady state solution for the *VC_Ring_Osc_Diff* oscillator circuit. You perform a PSS-Shooting analysis first because the periodic steady state solution must be determined before you can perform any other periodic small-signal analysis, such as pnoise to determine phase noise.

Setting up the PSS Analysis

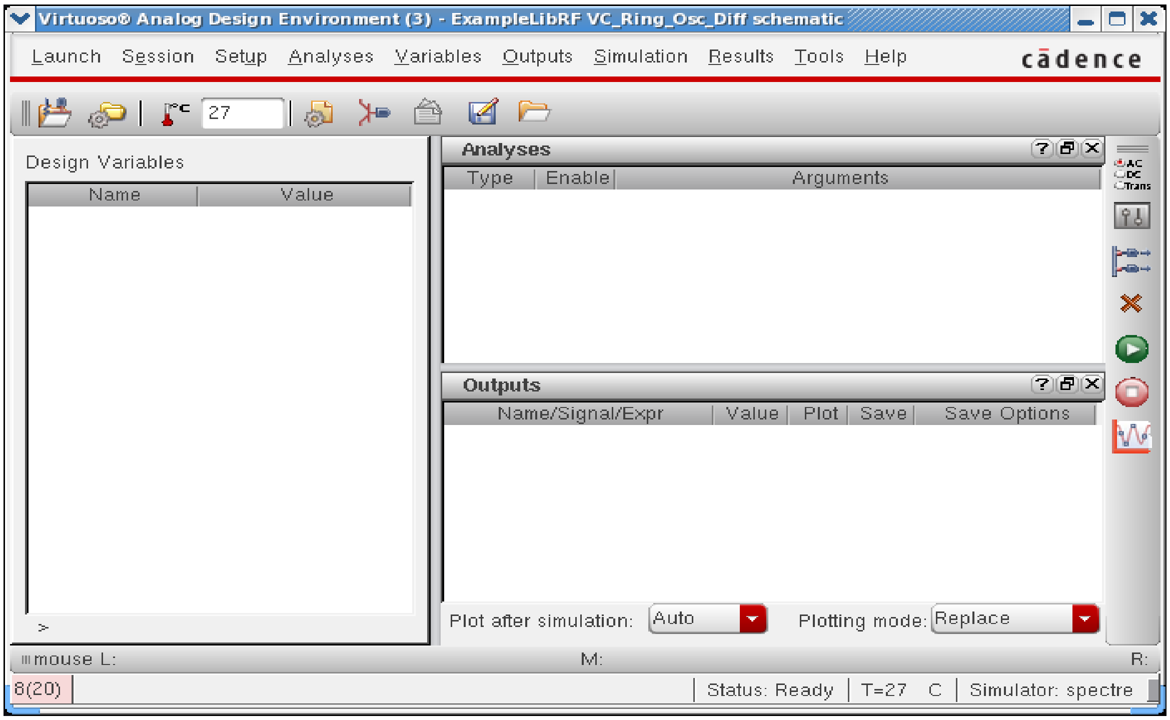
1. In the Schematic Window, choose *Launch - ADE L.*

Figure A-218 Opening ADEL window from VSE window



2. The Virtuoso Analog Design Environment Window opens, as shown below.

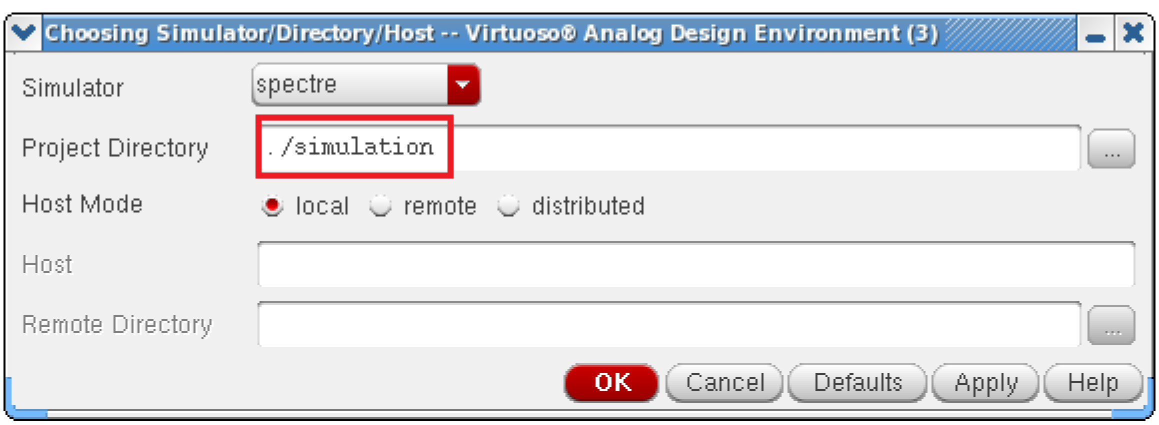
Figure A-219 Virtuoso Analog Design Environment Window



3. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

Figure A-220 Choosing Simulator/Director/Host Form



4. Specify the following in the *Choosing Simulator/Directory/Host* form:

a. Choose *spectre* fromthe *Simulator* drop-down list.

b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your ~/simulation directory. You may change the default location by editing the settings in your .cdsinit or .cdsenv file. In this workshop, the simulation directory is set to ./simulation.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific set-up instructions.

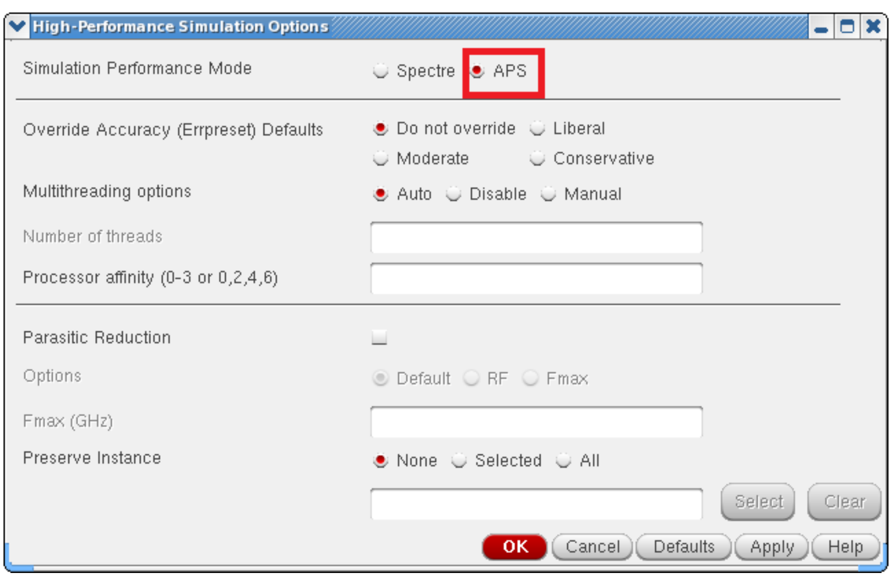
5. Click *OK* to close the *Choosing Simulator/Directory/Host* form.

6. Set up the High Performance Simulation options.

   In the ADE window, select *Setup - High Performance Simulation*. The High Performance Simulation Options form is displayed.

   In the *High Performance Simulation Options* form, select *APS*. Note that *Auto* is selected for Multithreading options. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

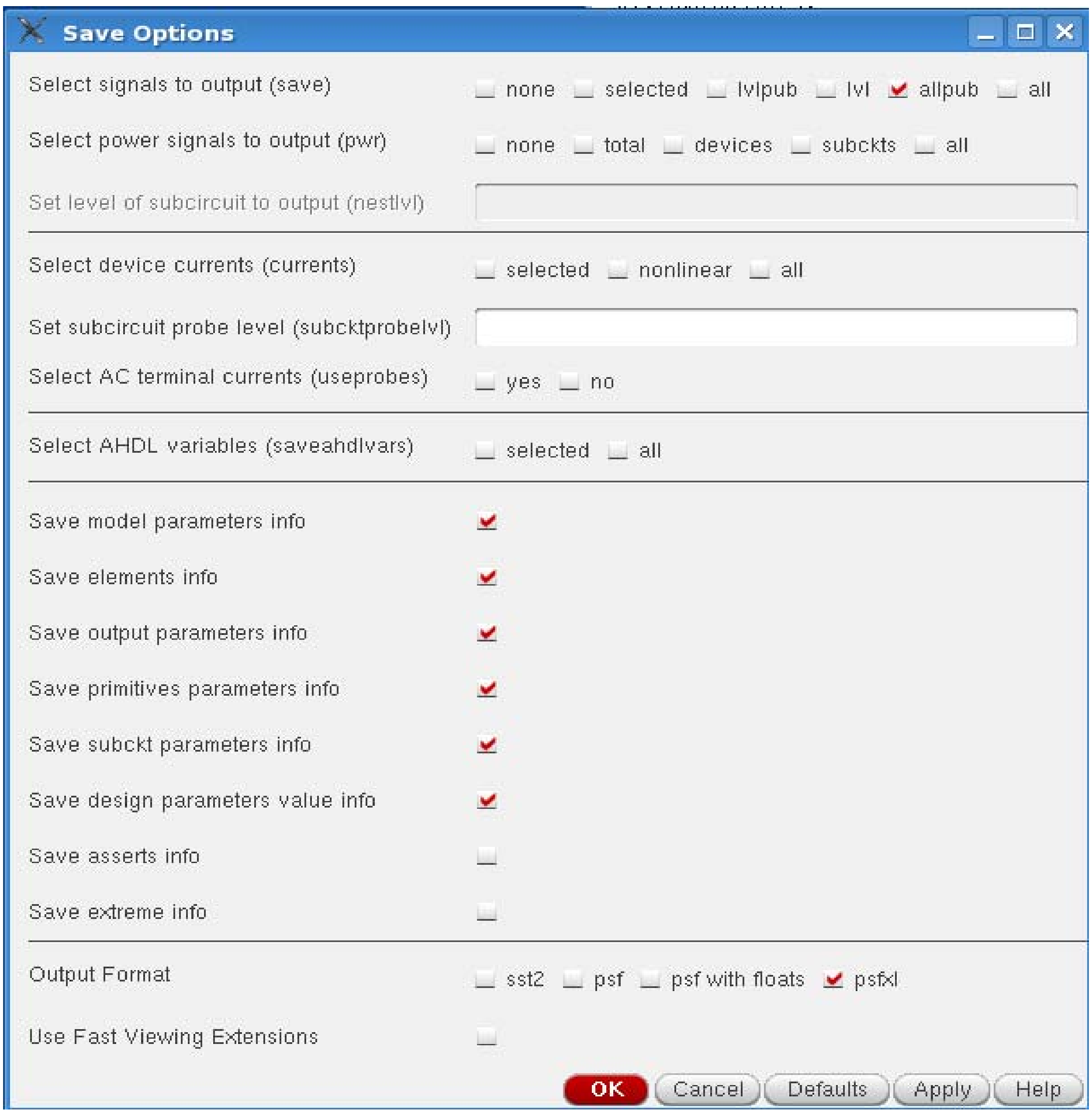
Figure A-221 High Performance Simulation Options Form



7. Click *OK* to close the *High Performance Simulation Options* form.
8. In the *Virtuoso Analog Design Environment* window, choose *Outputs - Save All*.

   The *Save Options* form is displayed.
9. In the *Select signals to output(save)* section, make sure that *allpub* is highlighted.

Figure A-222 Save Options Form



This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

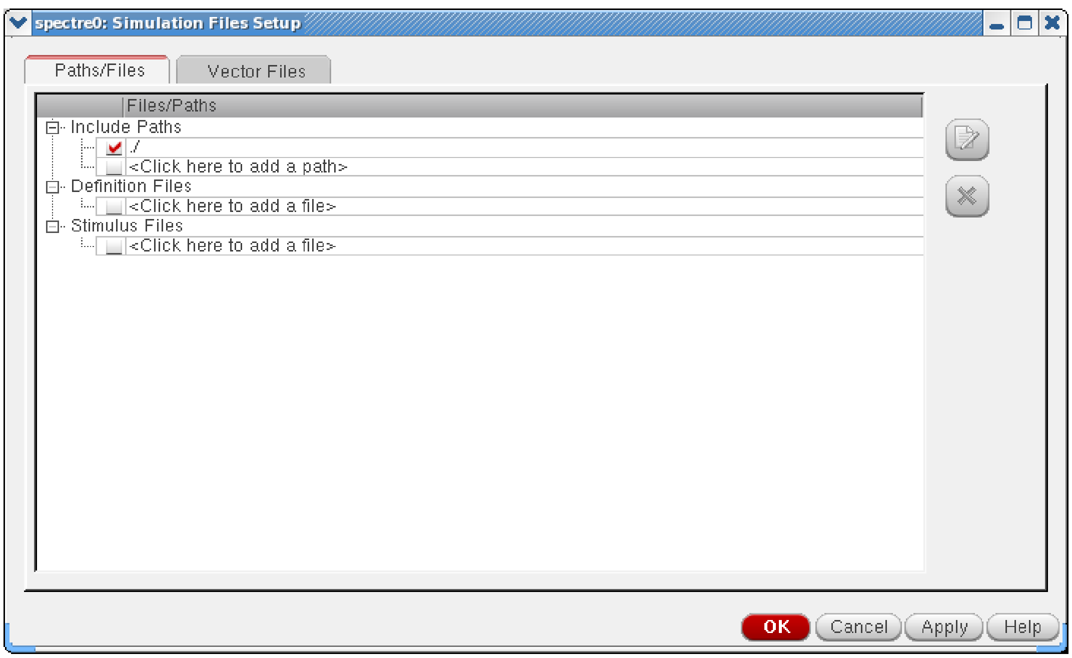
To save the currents, select the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or all if you want to save all the currents in the circuit. When you save currents, more disk space is required for the results file.

10. Click *OK*.

11. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

12. In the *Simulation Files Setup* form which gets opened, enter ./ by clicking in the *Include Paths* section. The form should look like the following:

Figure A-223 *Simulation Files Setup* Form



13. Click *OK* to close the *Simulation Files Setup* form.

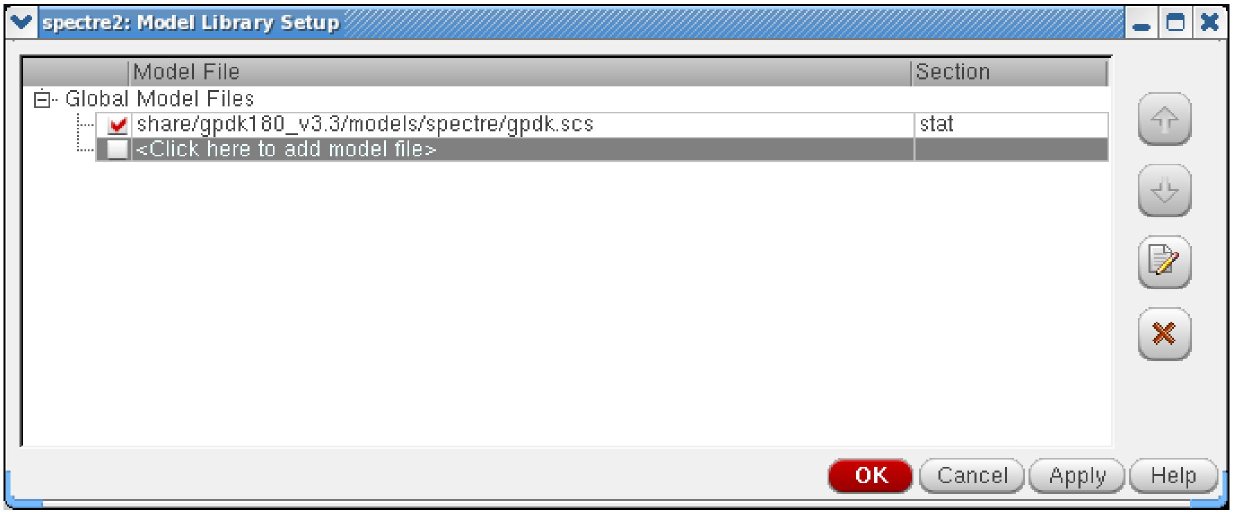
14. In the *Virtuoso Analog Design Environment* window, choose *Setup - Model Libraries* .

The *Model Library Setup* form is displayed.

15. In the *Model File* field, type the path to the model file including the file name, as shown below.

share/gpdk180_v3.3/models/spectre/gpdk.scs.

Figure A-224 ***Model Library Setup*** **Form**

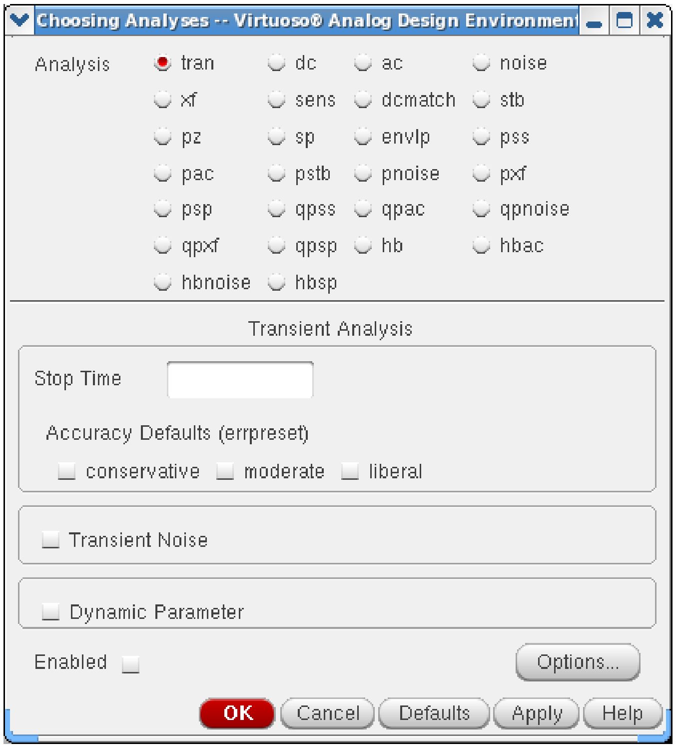


You can also browse to *gpdk.scs* file and then set the *Section* to *stat*.

16. Click *OK* to close the *Model Library Setup* form.
17. Choose *Analyses - Choose...* in the Virtuoso Analog Design Environment window.

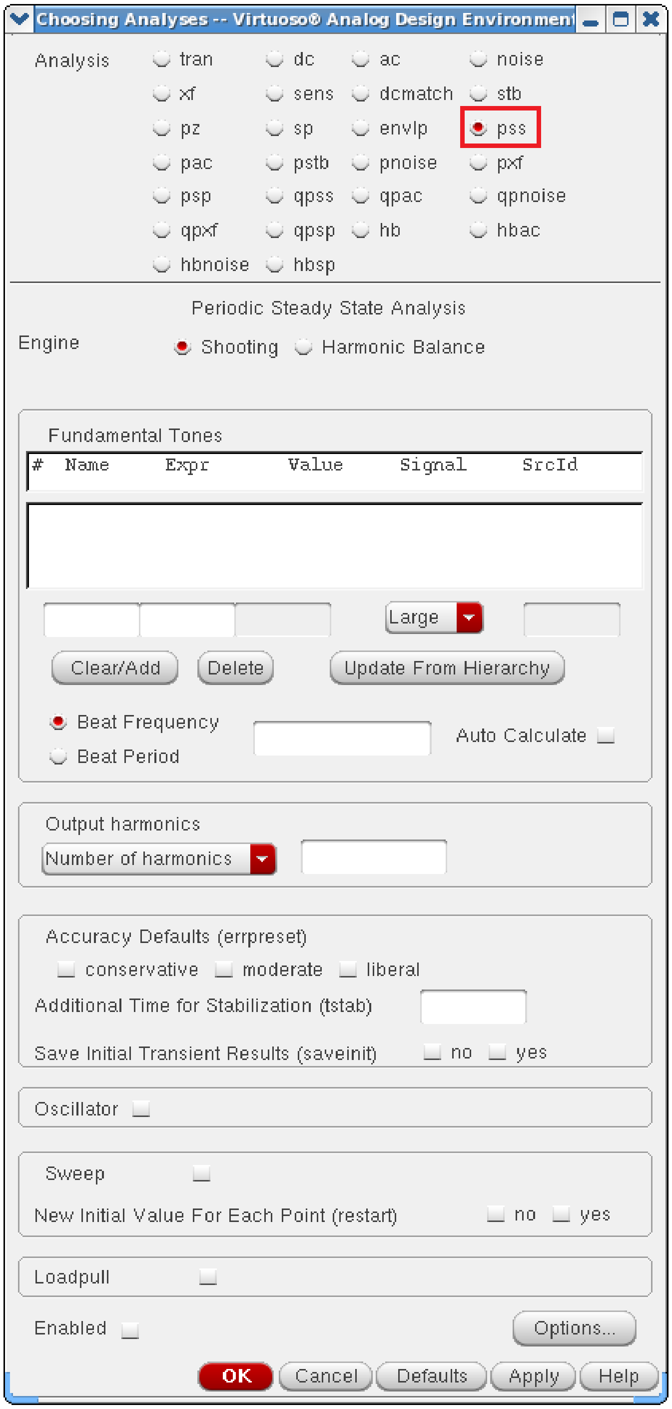
    The *Choosing Analyses* form is displayed.

Figure A-225 The Choosing Analyses Form



18. In the *Analysis* section, select *pss*. The form expands, as shown below.

Figure A-226 The Choosing Analyses Form- Setting PSS Analysis



19. In the *Engine* section, verify that *Shooting* is selected (this is the default).

20. In the *Beat Frequency* field, type 1.9G. The frequency entered here is an approximate frequency of oscillation.

21. In the *Number of harmonics* field, type 20.

In general, you want to choose a number that is high enough to capture the nonlinearity of the circuit.

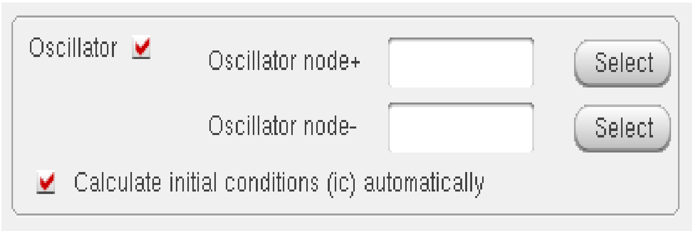
22. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. *conservative* is recommended for all oscillators.

23. Type 2n in the *Additional Time for Stabilization (tstab)* field. *tstab* is typically set to about 2-3 periods of the oscillation frequency for ring oscillator circuits. The tstab should be long enough so that the oscillator must reach near the steady-state behavior during this phase of simulation. This would result in better convergence.

24. Select *yes* for Save Initial Transient Results. This will help in visualizing the buildup of the oscillation waveform.

25. Select the *Oscillator* option. This is required for simulating an autonomous circuit. The oscillator section expands, as shown below.

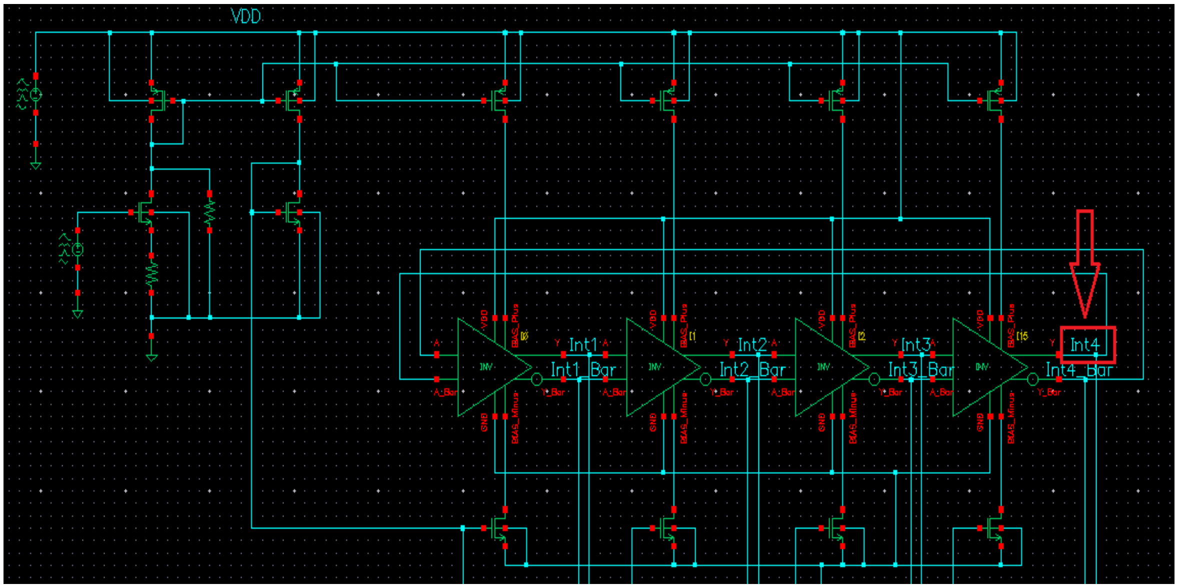
Figure A-227 The Choosing Analyses Form - Oscillator Section



26. In the *Oscillator node+* field, click *Select* to the right. In the schematic, select the *Int4* node. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it.

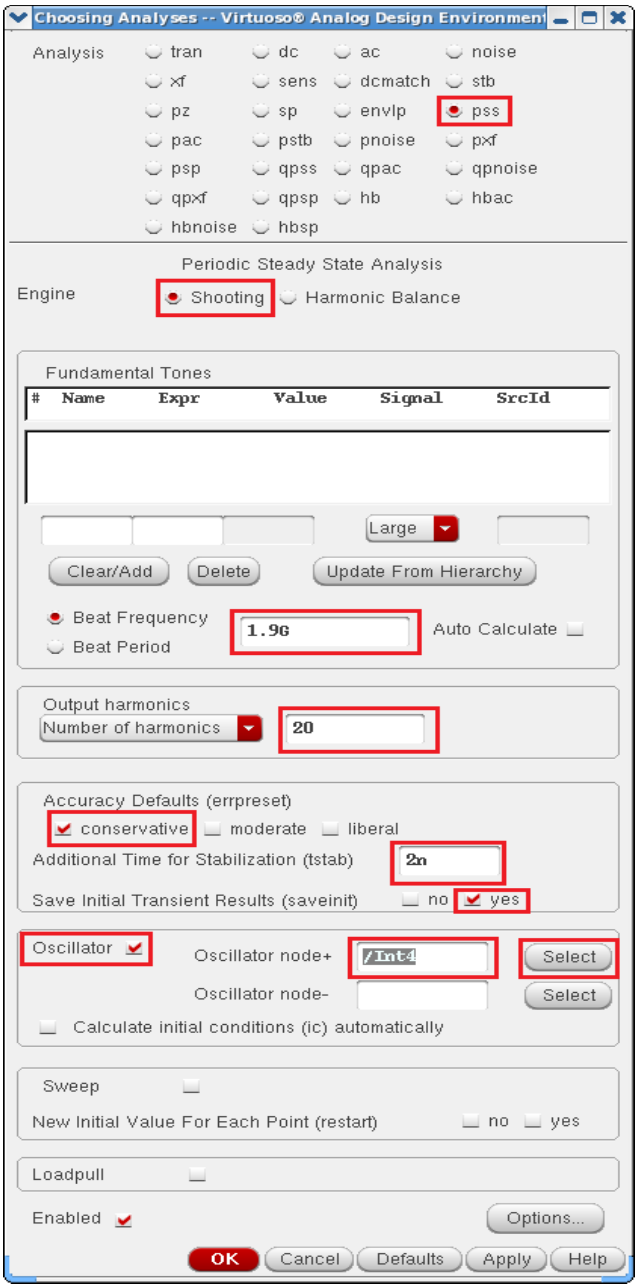
Also, deselect the *Calculate initial conditions (ic) automatically* checkbox. This is because selecting this checkbox only works for feedback oscillators while ring oscillator is not a feedback oscillator. Therefore, selecting this checkbox will not be able to find an oscillatory state for ring oscillator.

Figure A-228 Selecting *Int4* net on schematic



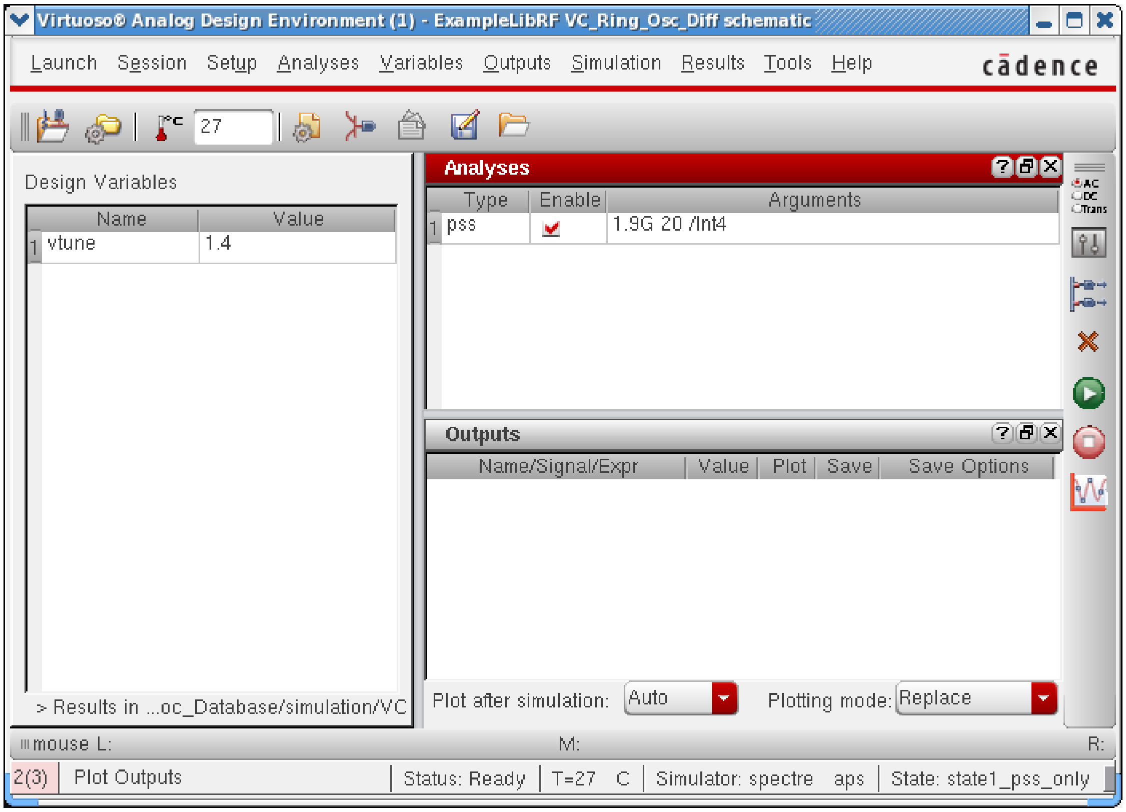
The *Choosing Analysis* form should look like the following:

Figure A-229 Choosing Analyses Form - PSS-Shooting Setup



27. Click *OK* . This will close the *Choosing Analyses* form. In addition, this will add the *pss* analysis in the *Analyses* section of ADE window, as shown below:

Figure A-230 ADE Simulation Window - PSS Analysis



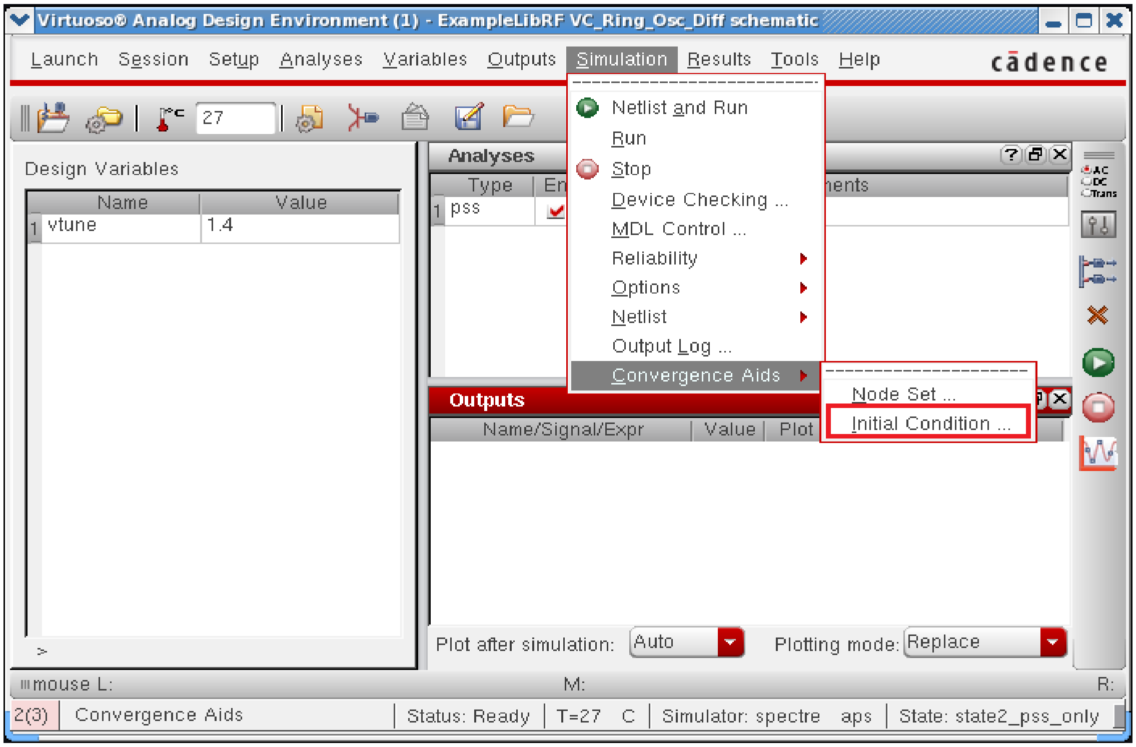
Setting initial conditions forces the circuit to a specific state at the time zero timepoint, and then removes that force after the time zero point is calculated.

In this case, we will set one stage of the oscillator so that one differential output is high, and one differential output (in the same stage) is low.

This defines all the nodes in the circuit, and the nodes that are connected to the nodes that are forced are pulling as hard as they can to get the forced nodes to change to the other state. When the forcing condition is removed, this action starts the oscillations.

Set the intial conditions by choosing *Simulation - Convergence Aids - Initial Condition* in ADE window, as shown below -

Figure A-231 PSS Analysis - Setting Initial Condition in ADE



This *Select Initial Condition Set* dialog box is displayed.

28. Type 0 (zero) in the *Node Voltage* field.

29. Select *Int4_Bar* node in the schematic. Note that the node highlights in the schematic.

30. Click *Apply*.

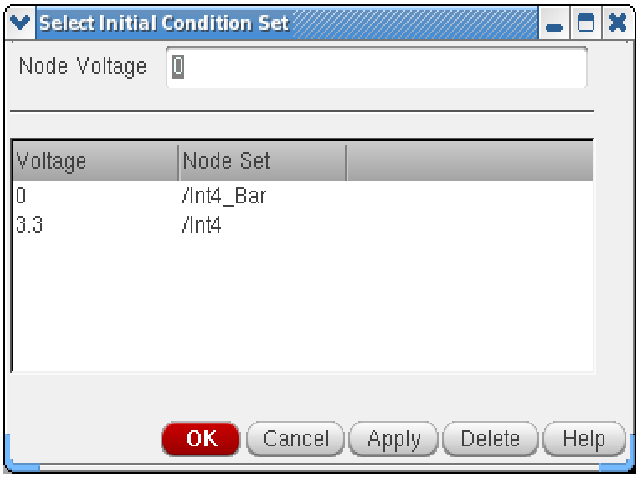
31. Type 3.3 in the *Node Voltage* field.

32. Select the *Int4* node in the schematic. Note that the node highlights in the schematic.

33. Click OK.

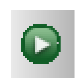
The populated *Select Initial Condition Set* dialog box would look like the following:

Figure A-232 Setting Initial Condition Form



This finishes the setting of PSS Analysis with setting up of Initial Conditions.

Running the PSS analysis

Once finished setting up the PSS analysis, click the gree icon  on the right side of the Analog Design Environment window or on the Schematic window to run the simulation.

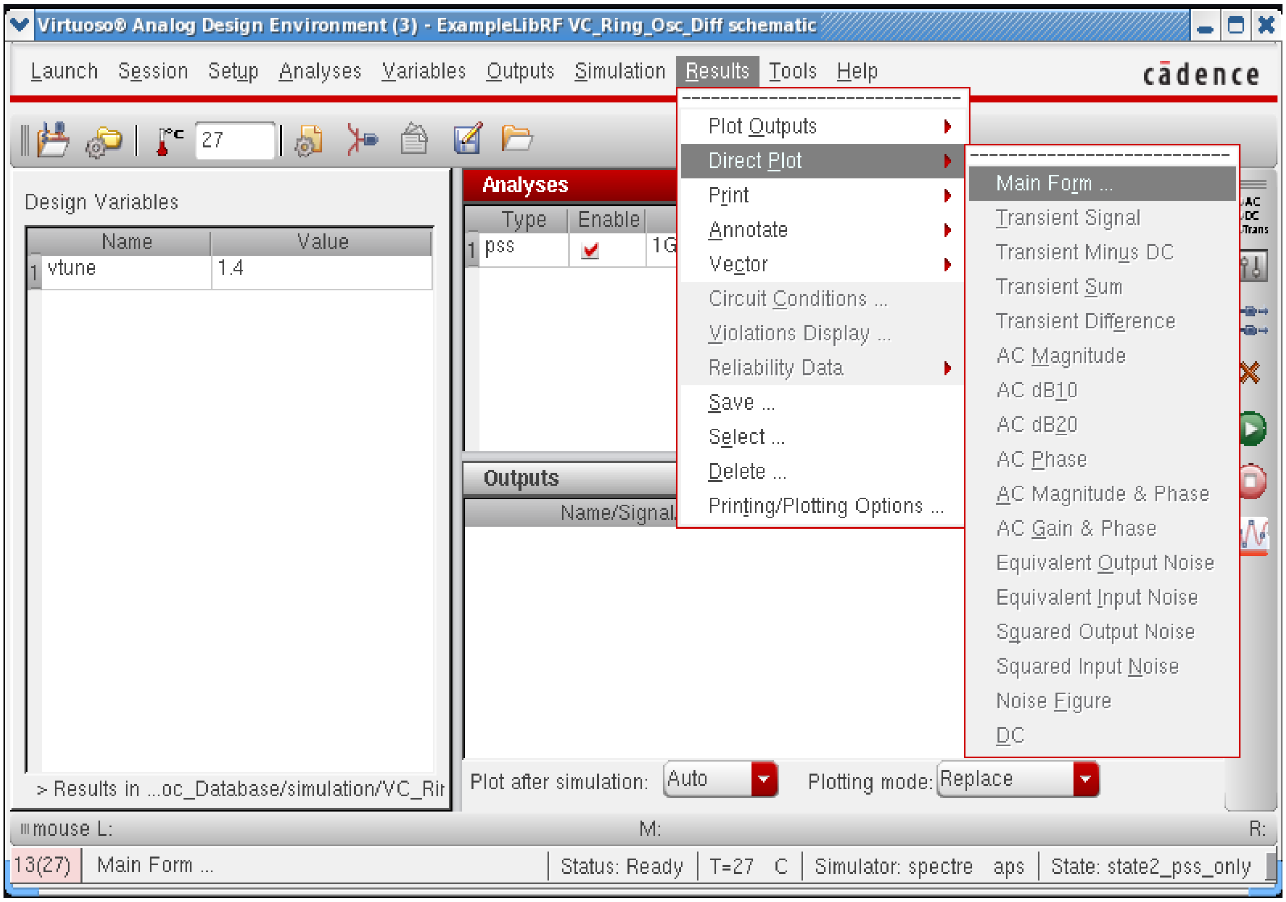
This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

Next, you will plot the results.

Plotting the PSS Analysis Results

1. In the *Virtuoso Analog Design Environment* window, choose *Results - Direct Plot - Main Form*.

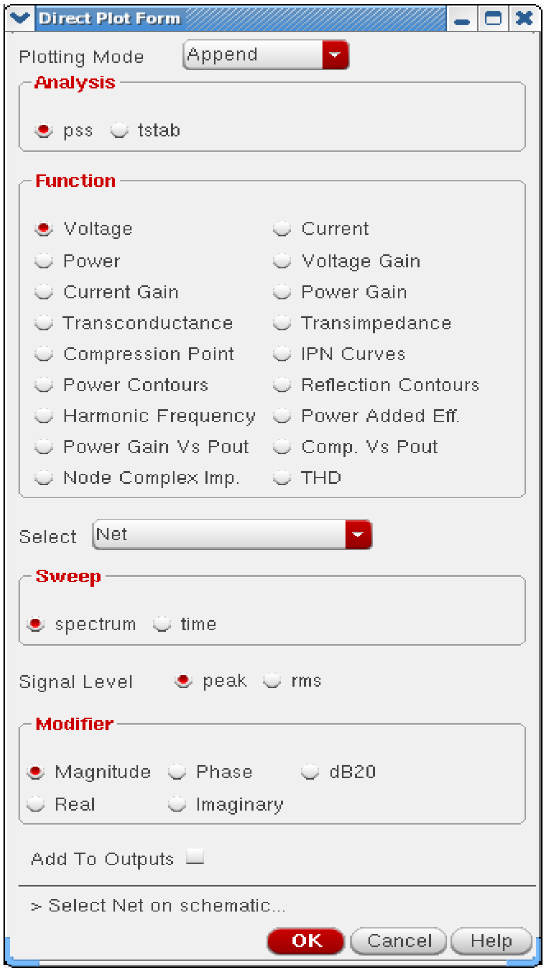
Figure A-233 Invoking Direct Plot Form



The *Direct Plot Form* is displayed.

You will see that there is pss analysis and tstab analysis in the *Analysis* section although you have only run pss analysis. The tstab analysis gets added as part of pss analysis run and is used to plot the initial transient waveforms.

Figure A-234 The Direct Plot Form

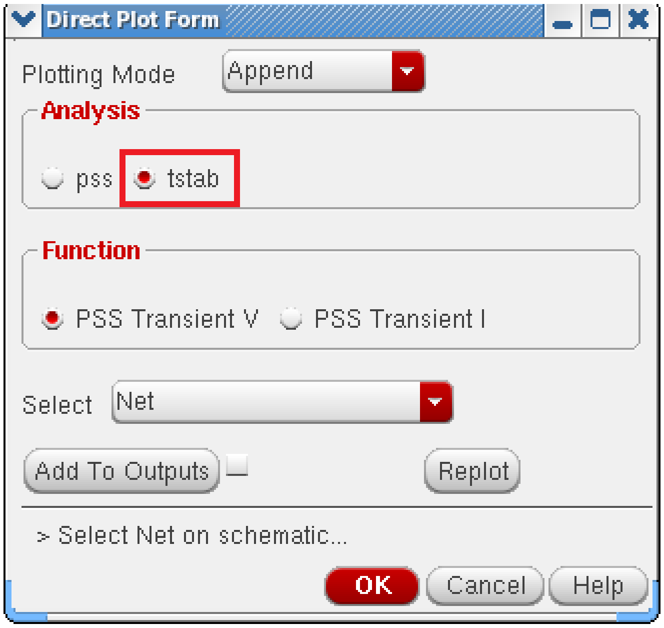


Plot the oscillator startup waveform from the tstab run -

2. In the *Direct Plot Form*, select *tstab* in the *Analysis* section.
3. Leave *Function* as *PSS Transient V* which is set by default.
4. Select *Net* in the center of the form. (This is the default. You can also select differential nets).

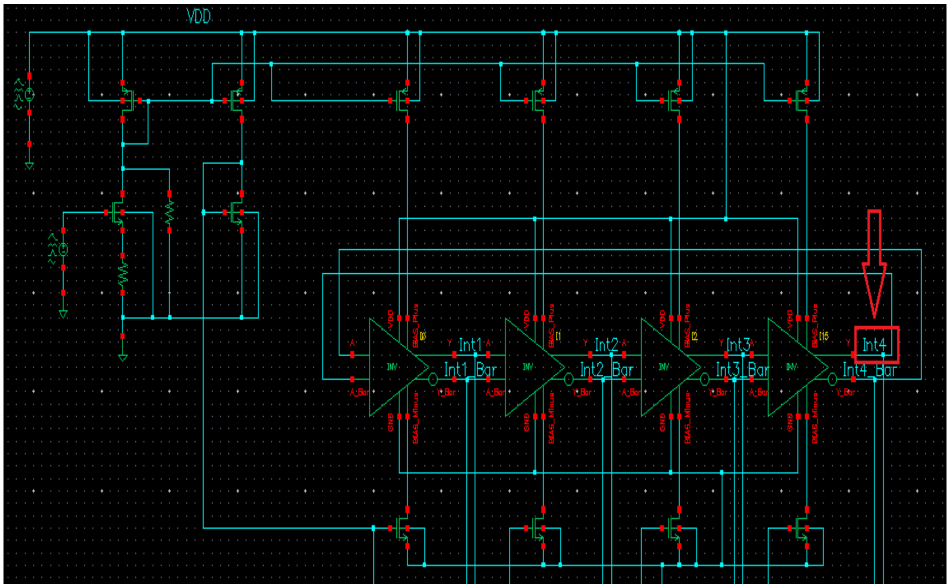
The *Direct Plot Form* window should like the following:

Figure A-235 PSS Analysis Direct Plot Setup - Initial Transient



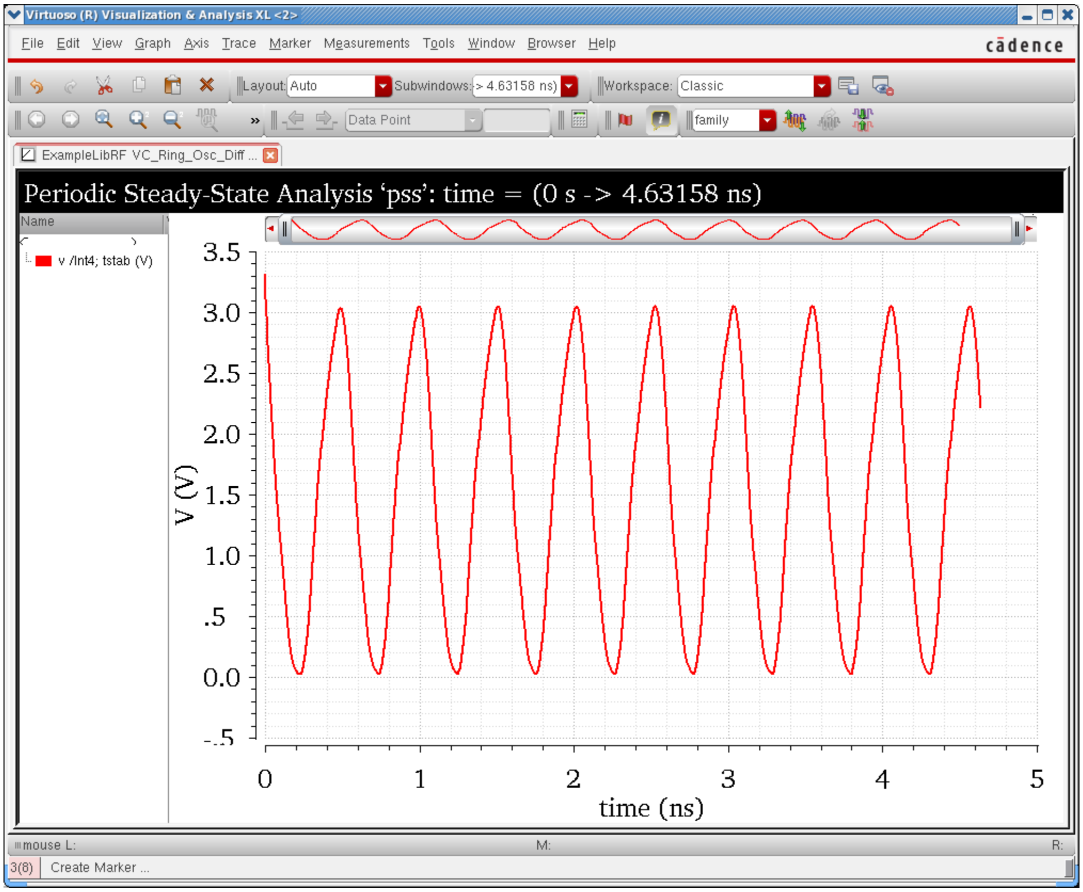
5. Select the *Int4* net in the schematic. It is located just below the *Int4* label.

Figure A-236 Selecting *Int4* net on schematic



The waveform window is displayed, as shown below.

Figure A-237 PSS Analysis Initial Transient Voltage Waveform



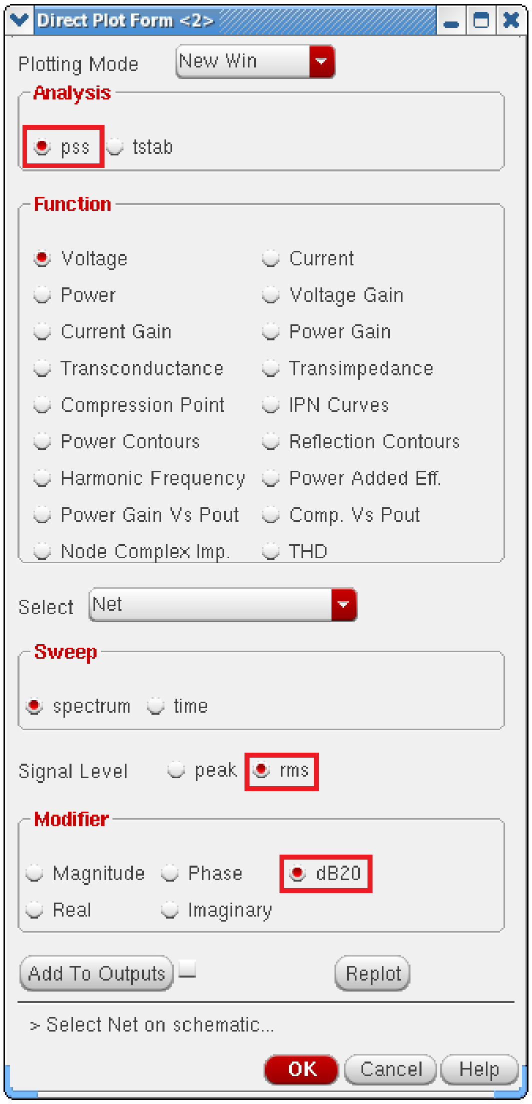
You can see from the plot that the initial transient waveform is for 4.63158ns as mentioned in the Output log window. The oscillator is in steady stae.

Next, you will plot the oscillator output spectrum.

1. In the *Direct Plot Form*, set the *Plotting Mode* to *New Win*.
2. Select *pss* inthe *Analysis* section.
3. Leave *Function* as *Voltage* which is set by default.
4. Select *Net* in the center of the form. (This is the default. You can also select differential nets).
5. Select *spectrum* in the Sweep section. (This is the default)
6. Select *rms* inthe *Signal Level* section (the default is *peak*).
7. Select *dB20* inthe *Modifier* section.

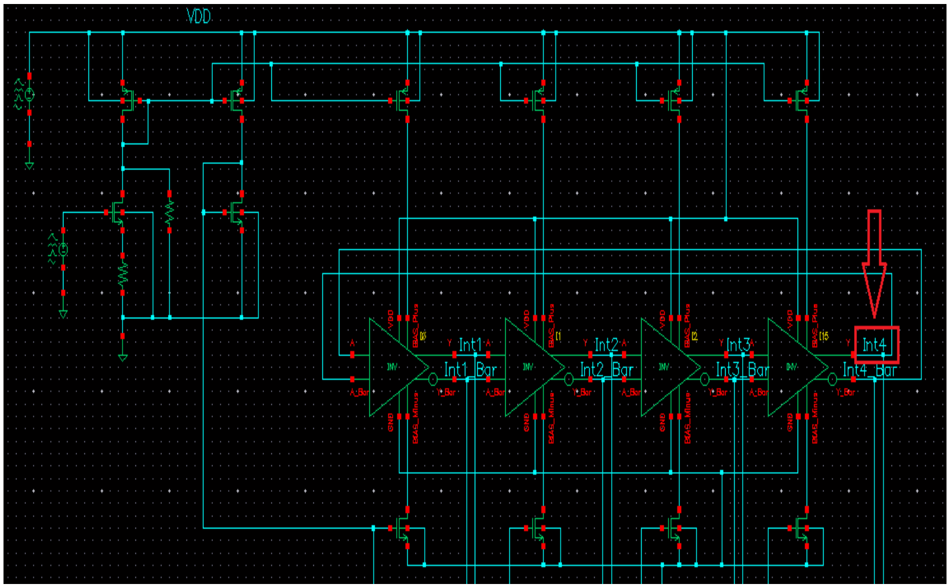
The *Direct Plot Form* window should like the following:

Figure A-238 PSS Analysis Direct Plot Form Setup



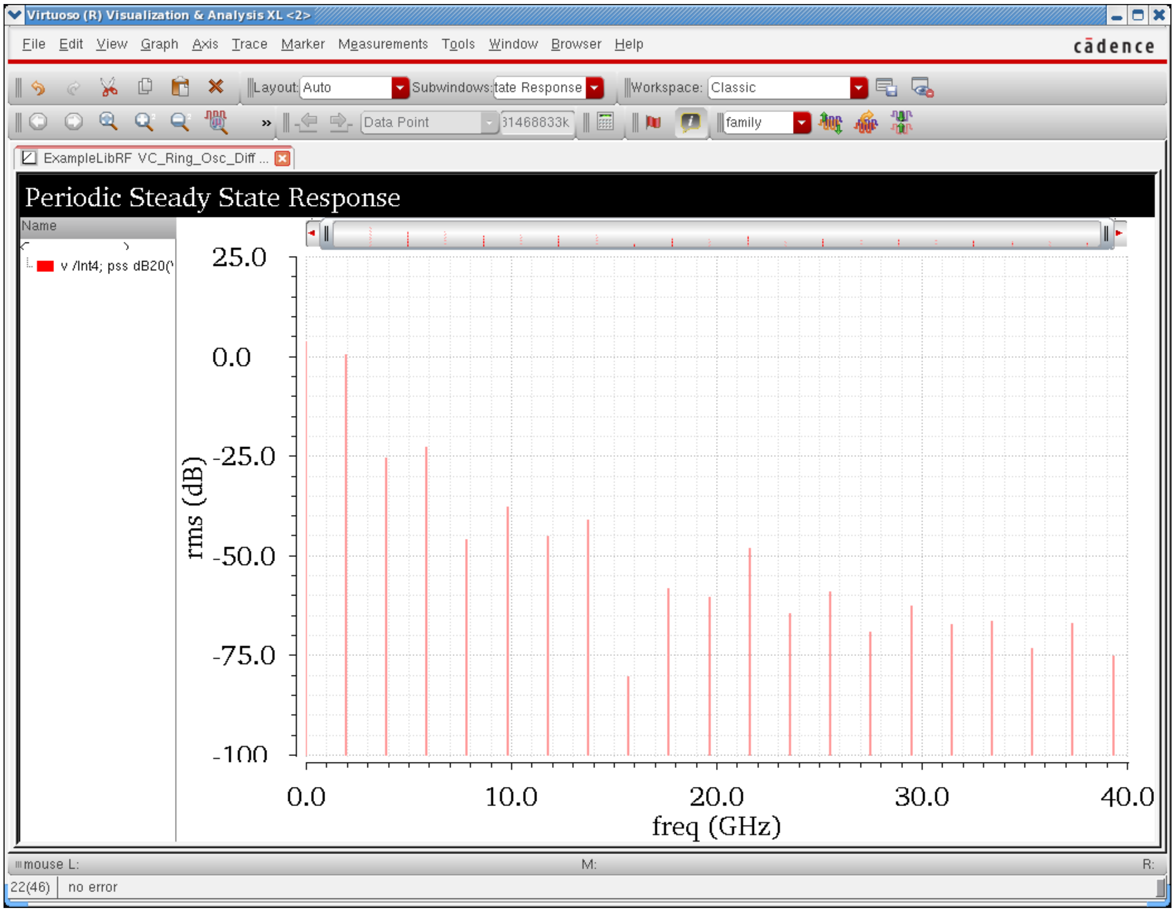
8. Select *Int4* net in the schematic. It is located just below the *Int4* label.

Figure A-239 Selecting *Int4* net on schematic



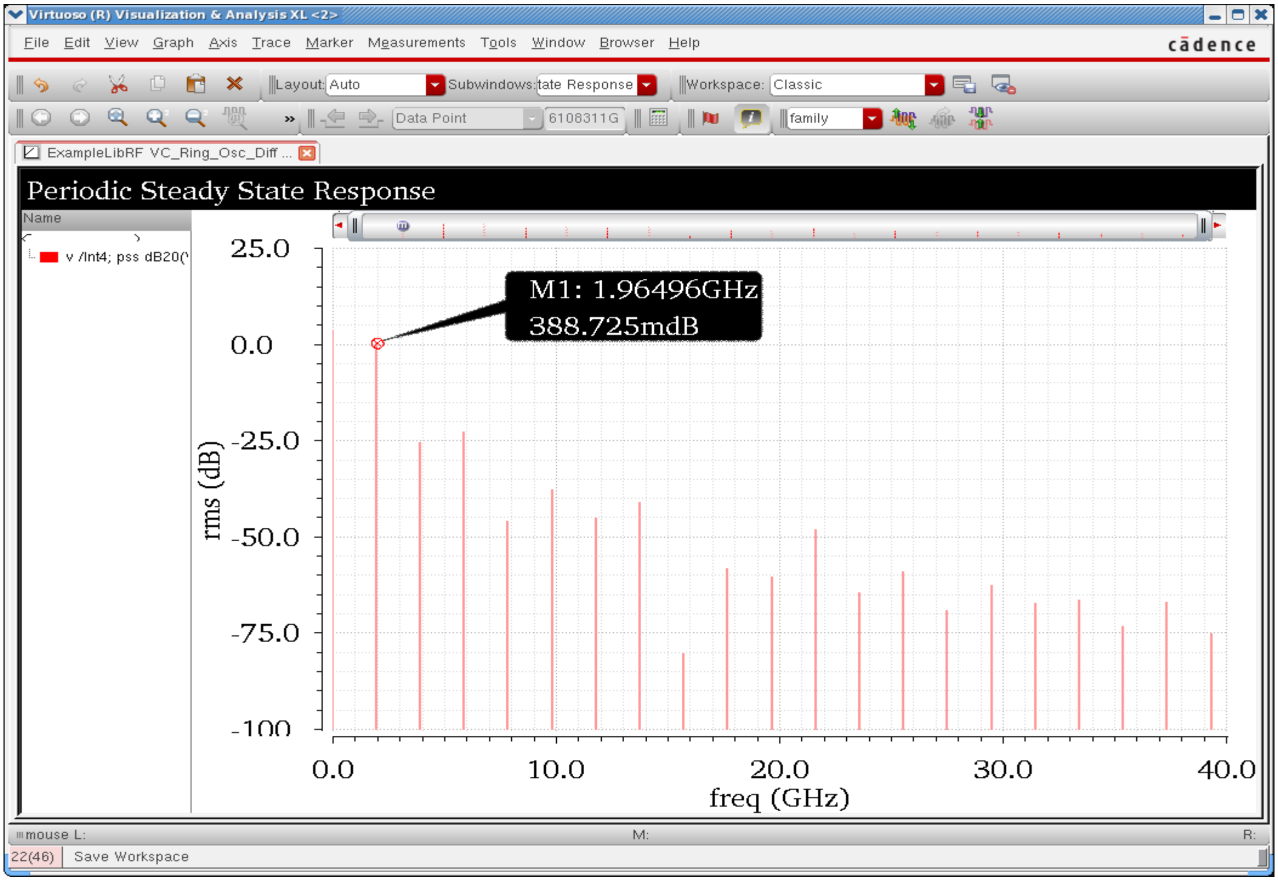
The waveform window is displayed, as shown below.

Figure A-240 PSS Analysis output Graph Window - Voltage Spectrum Plot



9. In the waveform window, position your cursor near the first harmonic, and press the *m* key. Here *m* is the bindkey to place a trace marker on the graph. The first harmonic is chosen as this is the frequency oscillator is designed for.

Figure A-241 PSS Analysis output Graph Window - Voltage Spectrum Plot (with marker placed)



10. Note that this frequency is 1.96496GHz. This is the frequency of oscillation.

11. In the *Direct Plot Form*, click *Cancel*. In the waveform window, choose *File - Close All Windows*.

12. Clean up the screen for the next set of measurements.

a. Close the Analog Design Environment windowby selecting *Session - Quit*.

b. Click *No* in the *Save State* window.

To summarize, a PSS analysis was set up using the Shooting Method and a simulation was run to determine the oscillation frequency of the oscillator.

Next, you will perform the FM jitter value measurements.

FM Jitter Measurement using PSS Shooting and Pnoise Jitter Analyses

This example computes the periodic steady state solution using the shooting method for the *VC_Ring_Osc_Diff* oscillator circuit. It then runs a periodic small-signal analysis pnoise to determine the FM jitter values of the oscillator. You perform a pss-shooting analysis first because the periodic steady state solution must be determined before you can perform any other periodic small-signal analysis like pnoise, pxf etc. to determine phase noise or transfer function and so on.

Determining FM Jitter

FM jitter calculates a standard averaged phase noise measurement, and also the AM and PM components from the modulated analysis, and adds the ability to integrate the phase noise curve to calculate the cycle jitter or the cycle-to-cycle jitter. The jitter calculations are integrated into the *Direct Plot Form* in ADE. All the measurements are averaged over the oscillator cycle.

PM Jitter is like the timedomain jitter measurement. You specify a threshold voltage and the timing jitter is calculated. In PM Jitter, a quantitative jitter measurement can be made from the direct plot form.

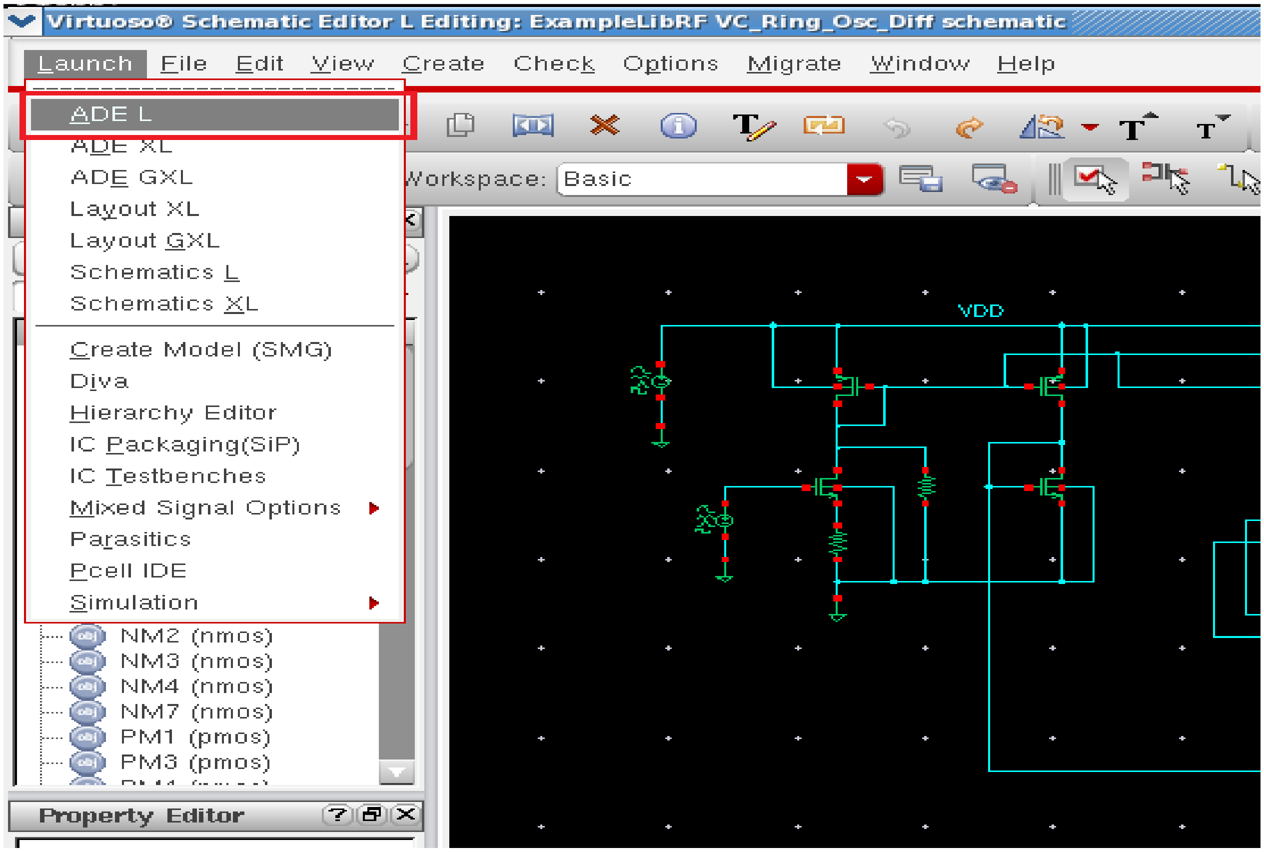
You may also refer to Chapter 3, “Frequency Domain Analyses: Harmonic Balance,” for more details on AM and PM Jitter.

In this measurement, you will perform the FM jitter measurement and plot the cycle jitter Jc.

Setting up the PSS Analysis

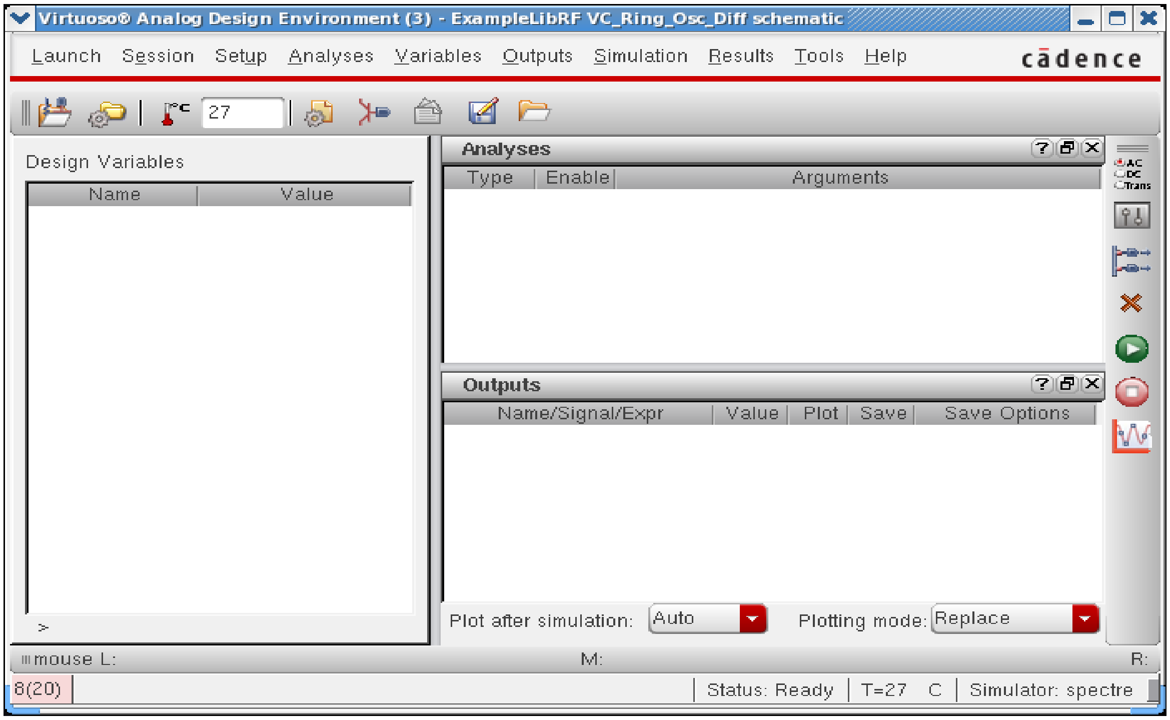
1. In the Schematic Window, choose *Launch - ADE L.*

Figure A-242 Opening ADEL window from VSE window



2. The Virtuoso Analog Design Environment Window is displayed, as shown below.

Figure A-243 Virtuoso Analog Design Environment Window



3. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

4. Specify the following in the *Choosing Simulator/Directory/Host* form:

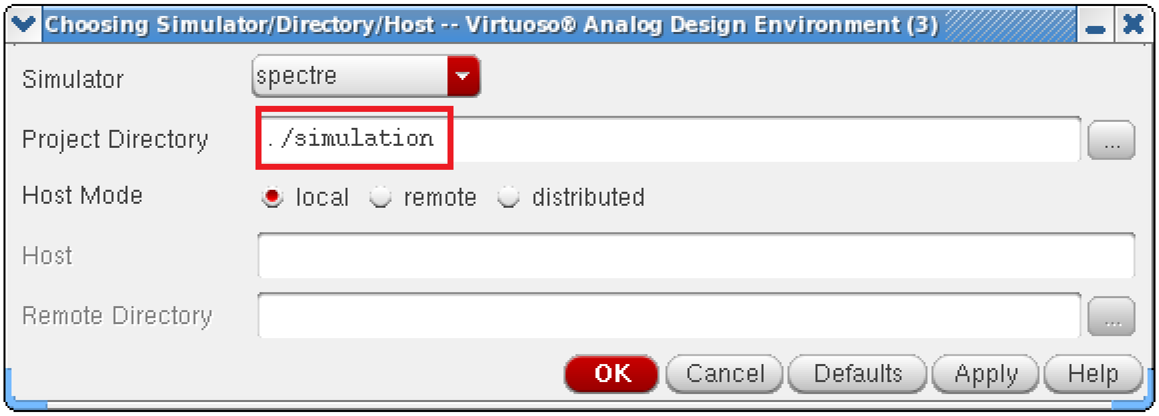
   a. Choose *spectre* for the *Simulator*.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

   c. Select the *Host Mode* that corresponds to your situation. For remote or distributed mode, please contact your Cadence tools System Administrator for specific set-up instructions.

   The completed form appears like the following:

Figure A-244 Choosing Simulator/Director/Host Form

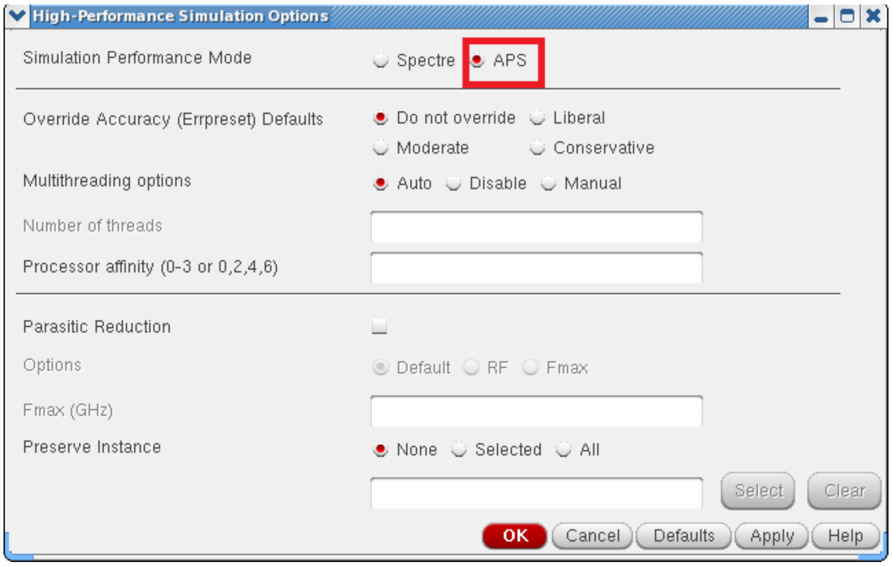


5. Click *OK* to close the *Choosing Simulator/Directory/Host* form.
6. Set up the High Performance Simulation Options.

   In the ADE window, choose*Setup - High Performance Simulation*. The High Performance Simulation Options window is displayed.

   In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

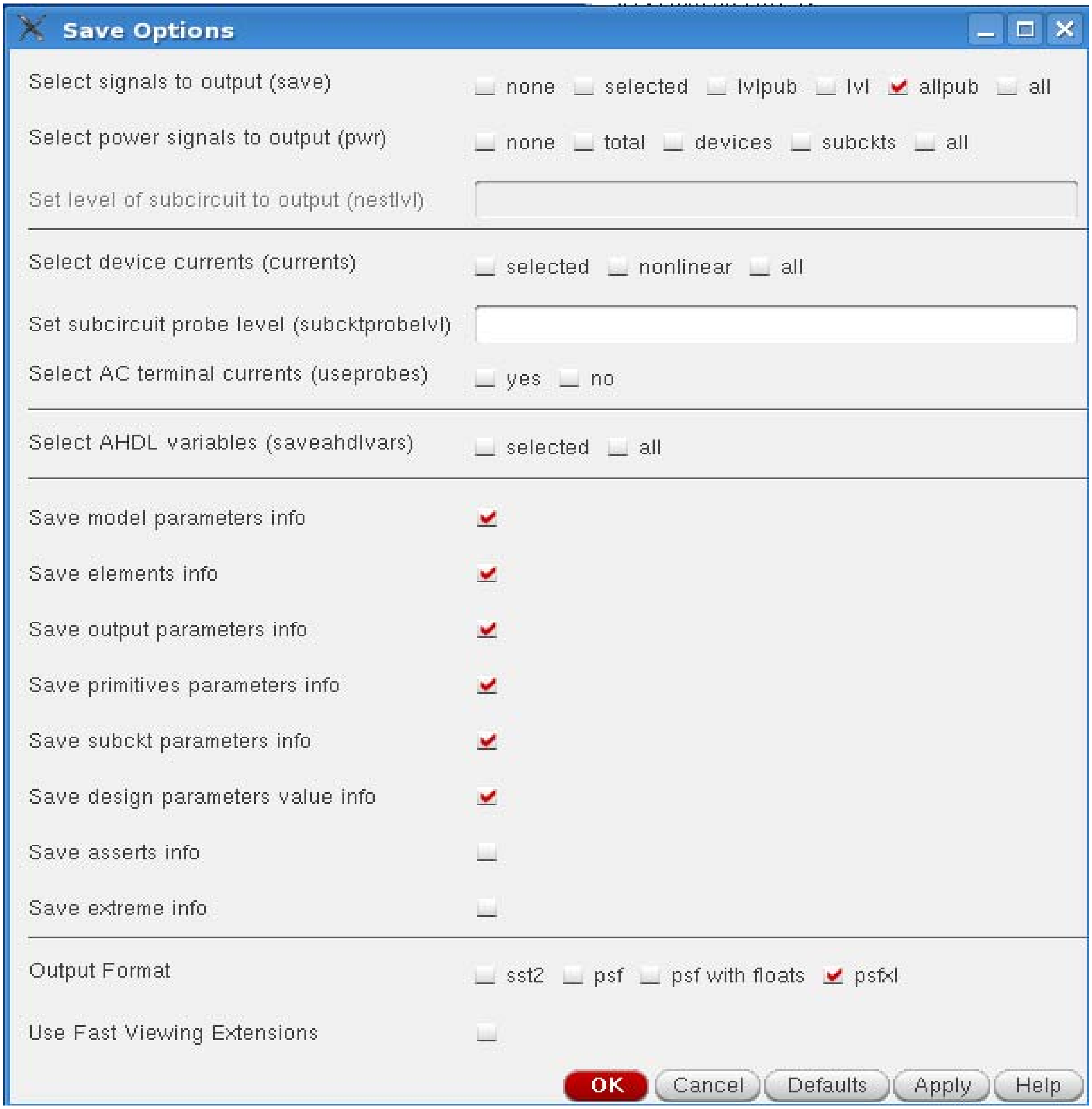
Figure A-245 High Performance Simulation Options Form



7. Click *OK* to close the *High Performance Simulation Options* form.
8. In the *Virtuoso Analog Design Environment* window, choose *Outputs - Save All*.

   The *Save Options* form is displayed.
9. In the *Select signals to output(save)* section, make sure that *allpub* is selected.

Figure A-246 Save Options Form



This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

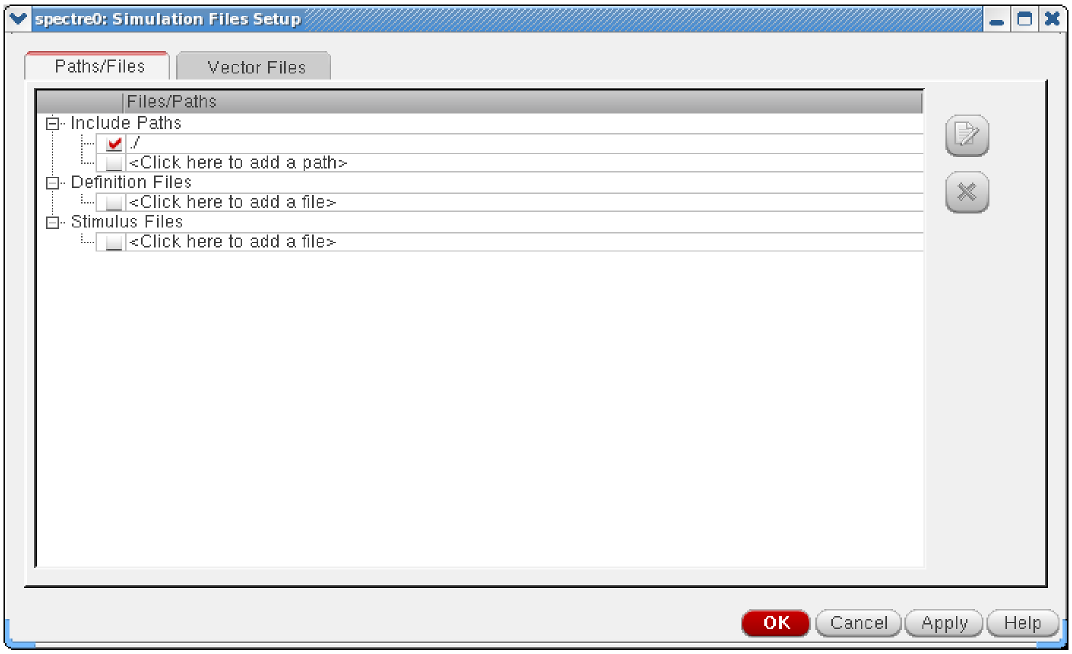
To save the currents, use the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or all if you want to save all the currents in the circuit. When you save currents, more disk space is required for the results file.

10. Click *OK*.

11. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

12. In the *Simulation Files Setup* form which gets opened, enter ./ by clicking in the *Include Paths* section. It should look like as shown below:

Figure A-247 ***Simulation Files Setup*** **Form**



13. Click *OK* to close the *Simulation Files Setup* form.

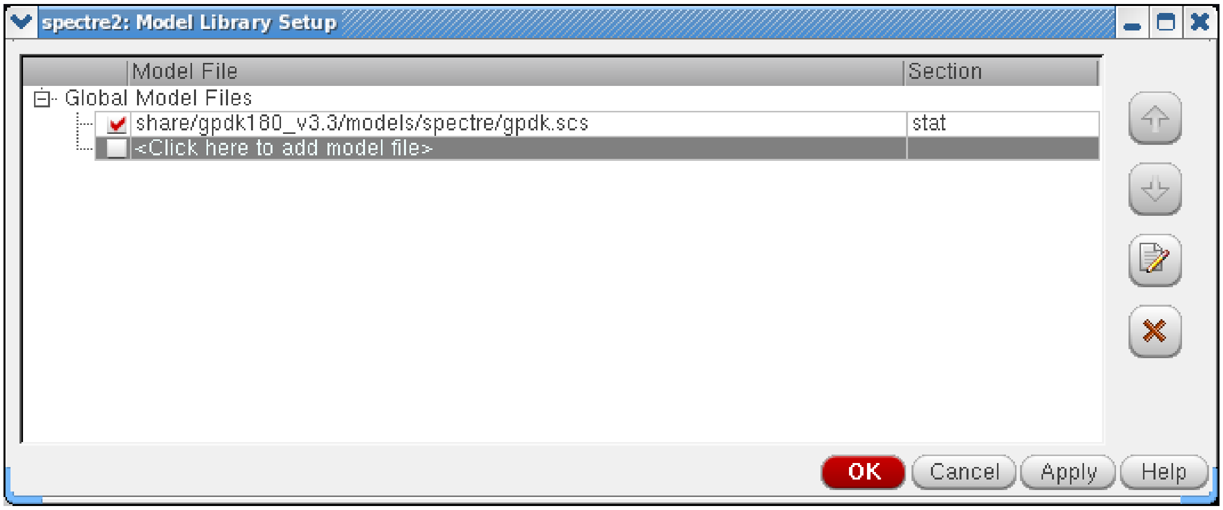
14. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

The *Model Library Setup* form is displayed.

15. In the *Model File* field, type the path to the model file including the file name, as follows:

share/gpdk180_v3.3/models/spectre/gpdk.scs.

Figure A-248 *Model Library Setup* Form



You can also browse to *gpdk.scs* file.

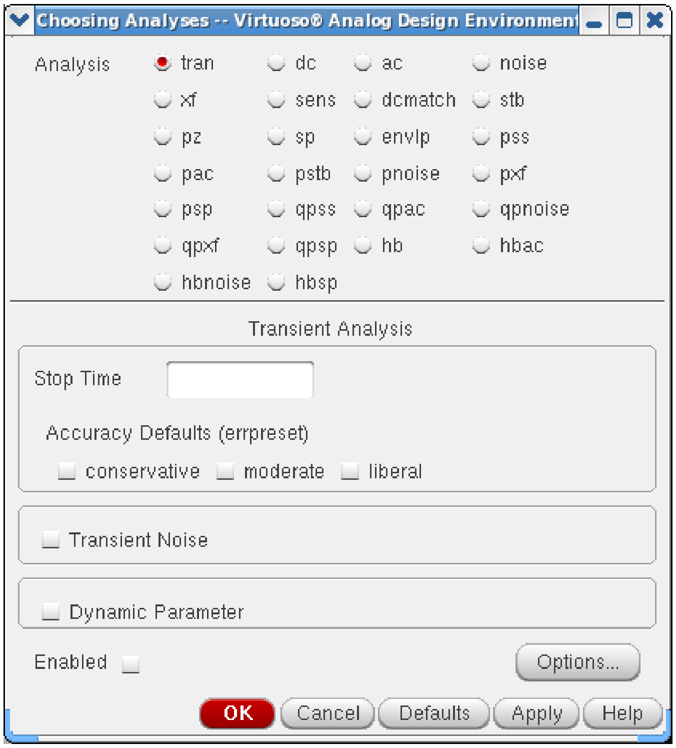
16. Set the *Section* to *stat*.

17. Click *OK* to close the form.

18. Choose *Analyses - Choose* in the Virtuoso Analog Design Environment window.

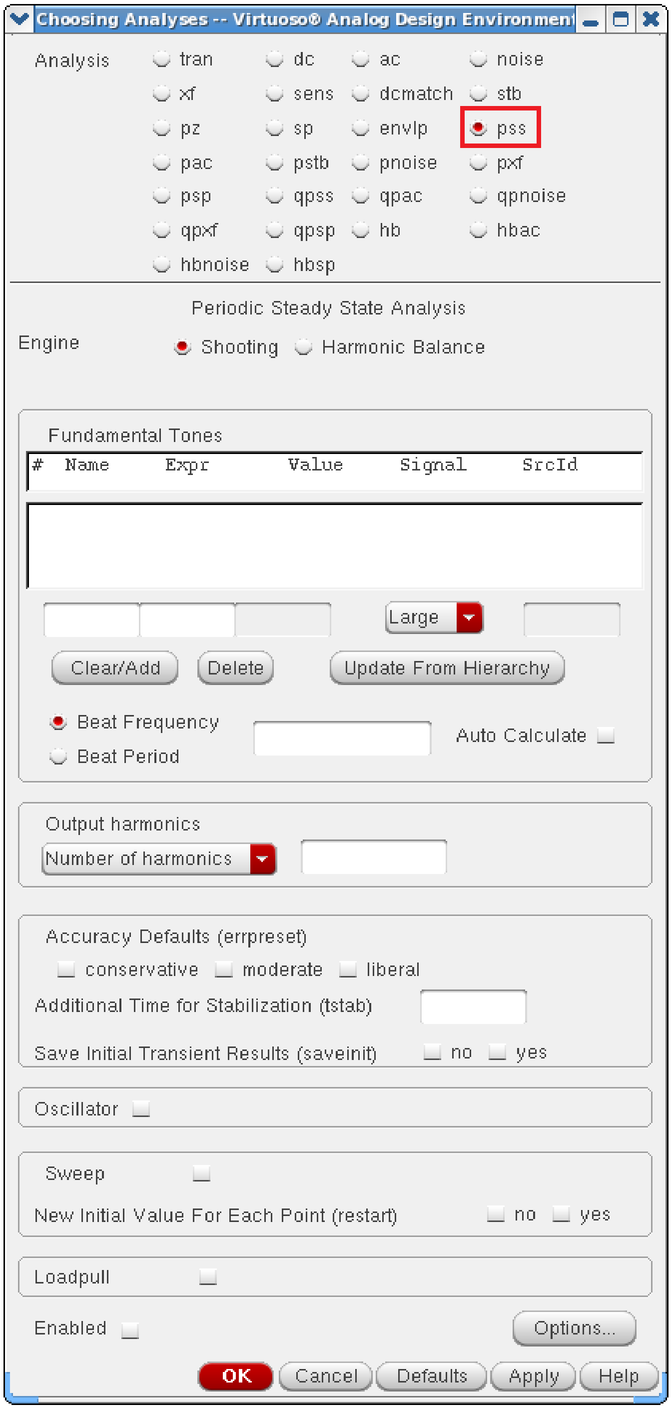
The *Choosing Analyses* form is displayed, as shown below.

Figure A-249 The Choosing Analyses Form



19. Select *pss* as *Analysis*. The form expands, as shown below.

Figure A-250 The Choosing Analyses Form- Setting PSS Analysis



20. In the *Engine* section, verify that *Shooting* is selected (this is the default).

21. In the *Beat Frequency* field, type 1.9G. The frequency entered here is an approximate frequency of oscillation.

22. In the *Number of harmonics* field, type 20.

23. In general, you want to choose a number that is high enough to capture the nonlinearity of the circuit. Start with 10, and run the simulation. Increase by about 50% to 15 and re-run the simulation. If the harmonics do not change appreciably, then 10 is enough. If they change, raise the number again by about 50%. Use the smallest number of harmonics for the answer to be stable.

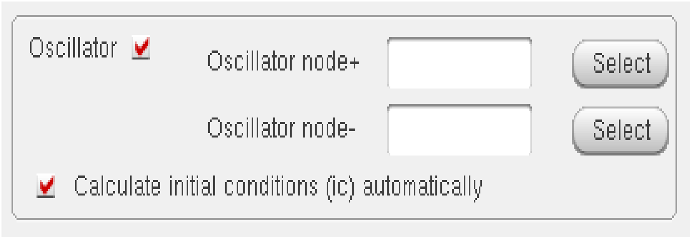
24. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. *conservative* is recommended for all the oscillators.

25. Type 2n in the *Additional Time for Stabilization (tstab)* field. *tstab* is typically set to about 2-3 periods of the oscillation frequency for ring oscillator circuits.

26. Select *yes* for *Save Initial Transient Results*. This will help in visualizing the buildup of the oscillation waveform.

27. Select the *Oscillator* option. This is required for simulating an autonomous circuit. The oscillator section expands, as shown below.

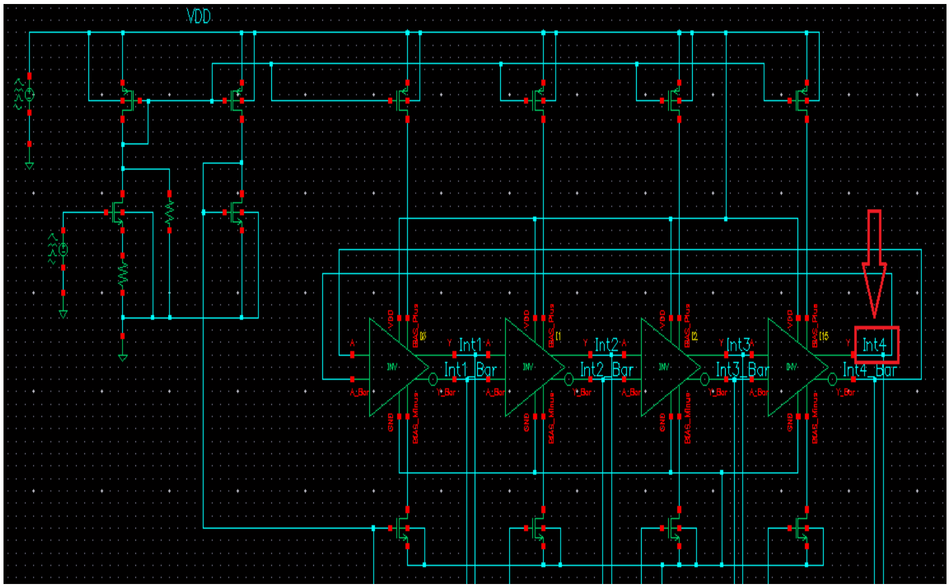
Figure A-251 The Choosing Analyses Form - Oscillator Section



28. In the *Oscillator node+* field, click *Select* just to the right. In the schematic, select the *Int4* node. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it.

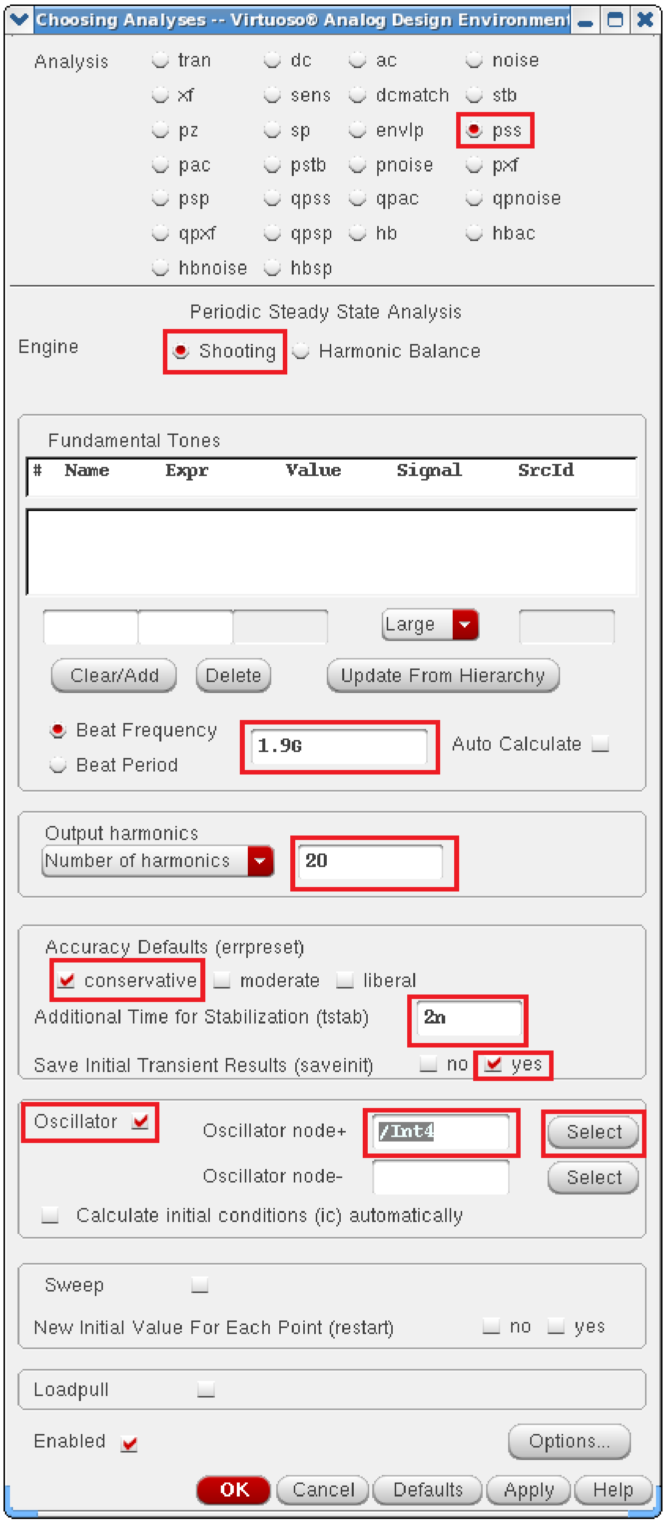
29. Deselect the *Calculate initial conditions (ic) automatically* checkbox. This is because selecting this checkbox only works for feedback oscillators while ring oscillator is not a feedback oscillator. Therefore, selecting this checkbox will not be able to find an oscillatory state for ring oscillator.

Figure A-252 Selecting *Int4* net on schematic



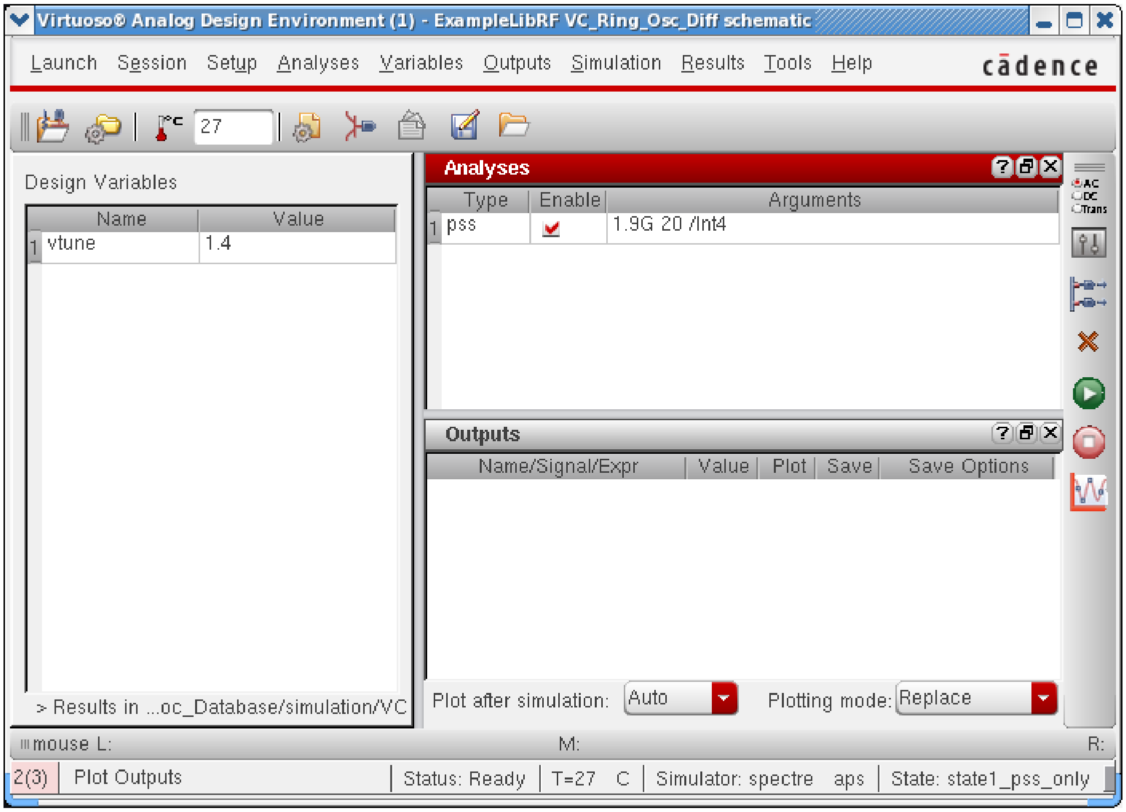
The *Choosing Analyses* Form should looks like the following:

Figure A-253 Choosing Analyses Form - PSS-Shooting Method Setup



30. Click *Ok*. This will close the *Choosing Analyses* form. In addition, this will add the *pss* analysis in the *Analyses* section of ADE window, as shown below.

Figure A-254 ADE Simulation Window - PSS Analysis



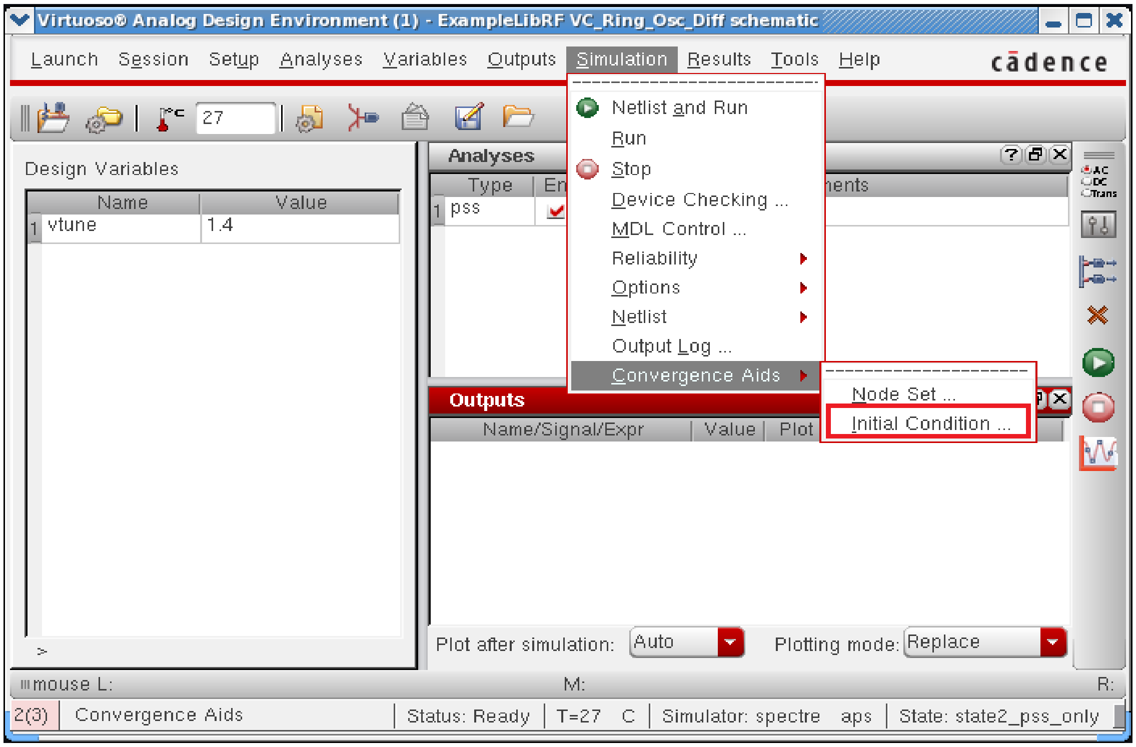
Next, set the initial conditions, which forces the circuit to a specific state at the time zero timepoint, and then removes that force after the time zero point is calculated.

In this case, we will set one stage of the oscillator so that one differential output is high, and one differential output (in the same stage) is low.

This defines all the nodes in the circuit, and the nodes that are connected to the nodes that are forced are pulling as hard as they can to get the forced nodes to change to the other state. When the forcing condition is removed, this action starts the oscillations.

Set the intial conditions by choosing *Simulation - Convergence Aids* - I*nitial Condition* in ADE window, as shown below.

Figure A-255 PSS Analysis - Setting Initial Condition in ADE



This will open the *Select Initial Condition Set* dialog box.

31. Type 0 (zero) in the *Node Voltage* field.

32. Select the *Int4_Bar* node in the schematic. Note that the node highlights in the schematic.

33. Click *Apply*.

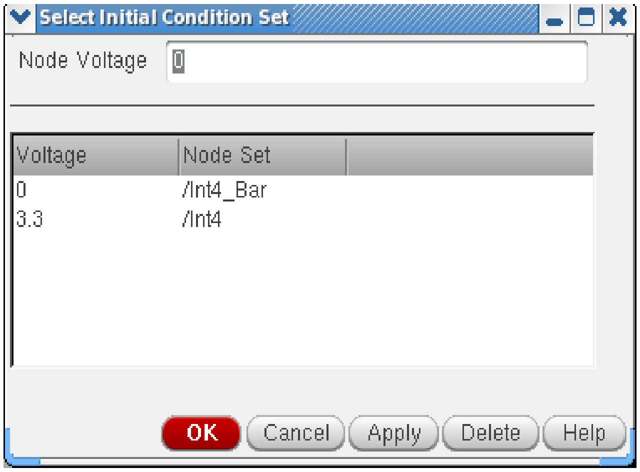
34. Type 3.3 in the *Node Voltage* field.

35. Select the *Int4* node in the schematic. Note that the node highlights in the schematic.

36. Click *OK*.

The populated *Select Initial Condition Set* dialog box would look like the following:

Figure A-256 Setting Initial Condition Form



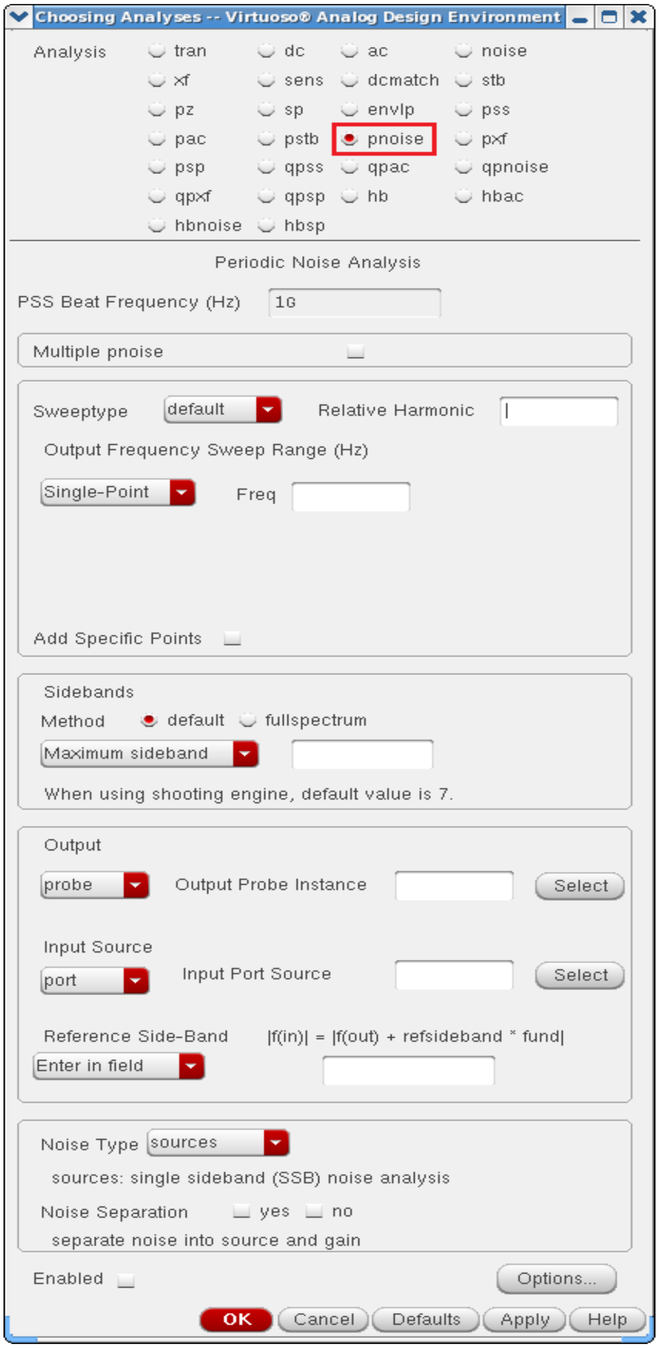
This finishes the setting of PSS Analysis with setting up of Initial Conditions.

Setting up the Pnoise analysis

This analysis is set to do the phase noise measurement. It is run after *pss* analysis.

1. In the *Choosing Analyses* form, select *pnoise*. The form expands, as shown below.

Figure A-257 The Choosing Analyses Form - *pnoise* Analysis Setup



2. Leave the *Sweep Type* to *default*.

   For oscillators, the *hbnoise/pnoise* frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the hb/pss *Choosing Anlayses* form, you should specify the approximate frequency of oscillation for the frequency-divided signal. In the *hbnoise/pnoise Choosing Anlayses* form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of *relative* is to take the frequency of the harmonic number specified and add to it the frequencies specified in the *Choosing Analyses* form. If the oscillator had a 1GHz output, and the pnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

a. Type 1 in the *Relative Harmonic* field as you are simulating an oscillator by itself.

Next, you will set the output frequency sweep range. Frequency sweep is set for a noise simulation as the noise is spread over the frequency range. Oscillator phase noise adds phase uncertainty for phase modulated signals. Note that BPSK, QPSK, and all QAM signals have phase information in the constellation. If the LO signal has phase noise on it, it makes it harder to demodulate the signals because there is more spread in the phase of the received signal.Therefore, it is critical to determine its behavior over a frequency range.

b. In the *Ouput Frequency Sweep Range (Hz)* section, type 10K in the *Start* Field .

c. Type 1G in the *Stop* Field.

Set the frequency sweep range as appropriate for your circuit (or application).

d. Set the *Sweep Type* to *Logarithmic*.

e. Type 4 in the *Points Per Decade* field. Typically, 3 to 5 points per decade are a reasonable number to capture the noise behavior of the circuit.

3. In the Sidebands section, choose*fullspectrum* as the *Method* .

4. Leave the *Maximum Sideband* field blank. Since *fullspectrum* is set, and the oscillator frequency is well above 100KHz, the *Maximum sideband* field should be left blank.

5. Set the *Output* to *voltage*.

a. Type /V4P in the *Positive Output Node* field. You can also select *V4P* net from schematic by clicking *Select* on the right of the *Positive Output Node* field and then selecting the net just below the *out* label in schematic.

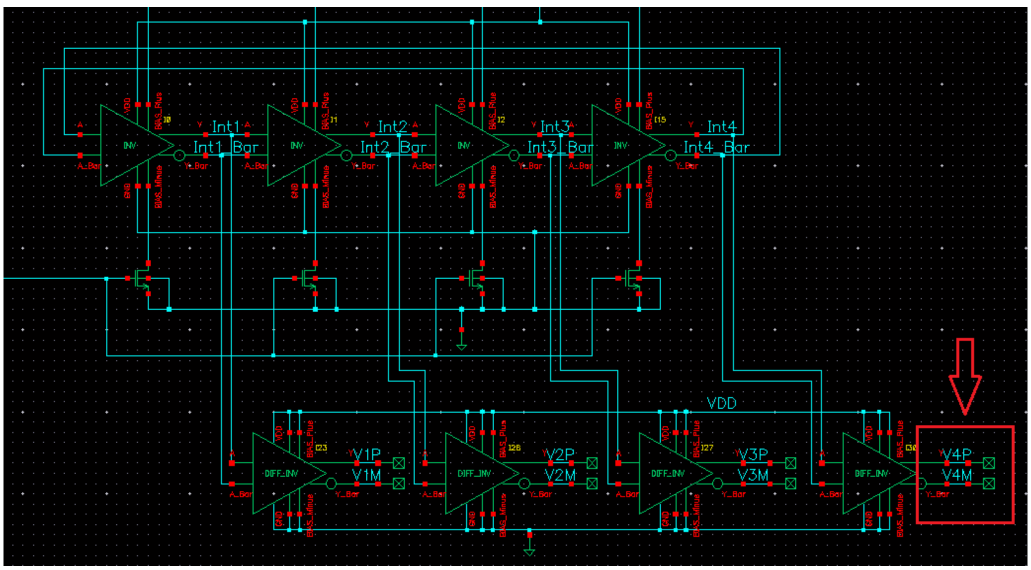
b. Type /V4M in the *Negative Output Node* field. If the second node, that is, the *Negative Output Node* is left blank, it will be connected to the global ground node automatically. However, if you have differential oscillator, you need to specify both the nodes.

c. Set the *Input Source* to *none*.

Using an Input Source generates NF, IRN, and other parameters which are not needed here. Input Source is not usually used in oscillator simulations.

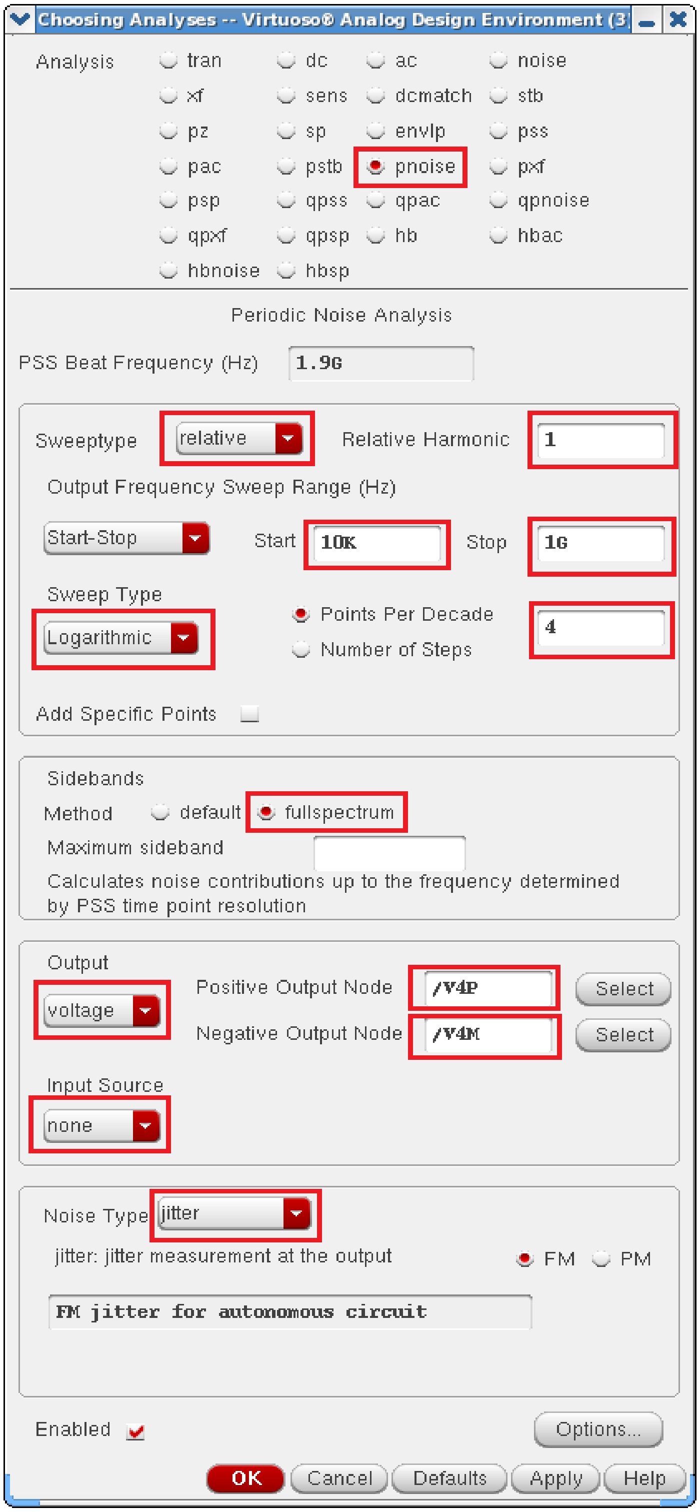
6. Choose noise type as jitter from the drop down box.

Figure A-258 Selecting *V4P and V4M* net from the schematic



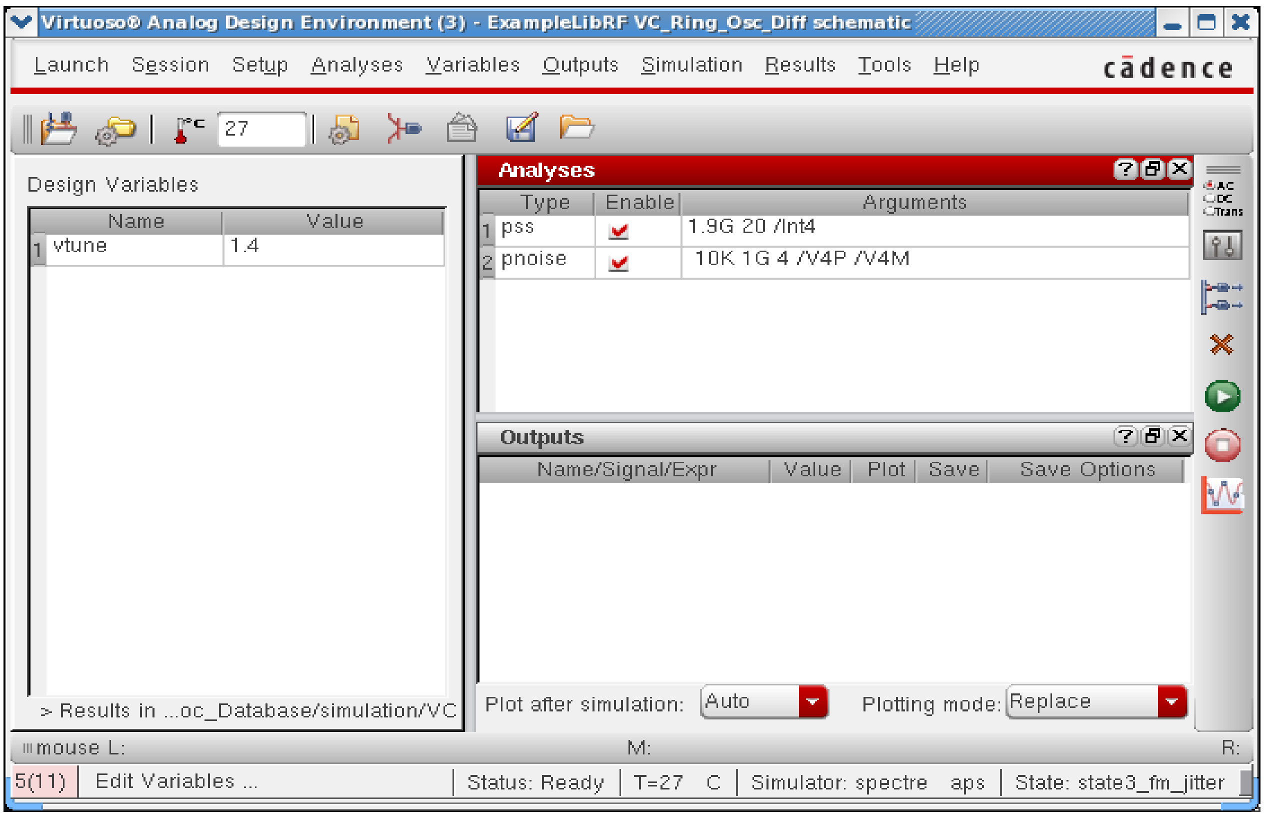
The *Choosing Analyses* form should look like the following:

Figure A-259 Choosing Analysis Form - *pnoise* Analysis Setup



7. Click *OK* to close the *Choosing Analyses* form. This will add the *pnoise* analysis along with *pss* analysis in the *Analyses* section of ADE window, as shown below:

Figure A-260 ADE Simulation Window - *pss* and *pnoise-jitter* analysis setup



Running the PSS and Pnoise analysis

Once finished setting up the PSS and Pnoise Analyses clickthe green icon  on the right of the Analog Design Environment window or on the Schematic window to run the simulation.

This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

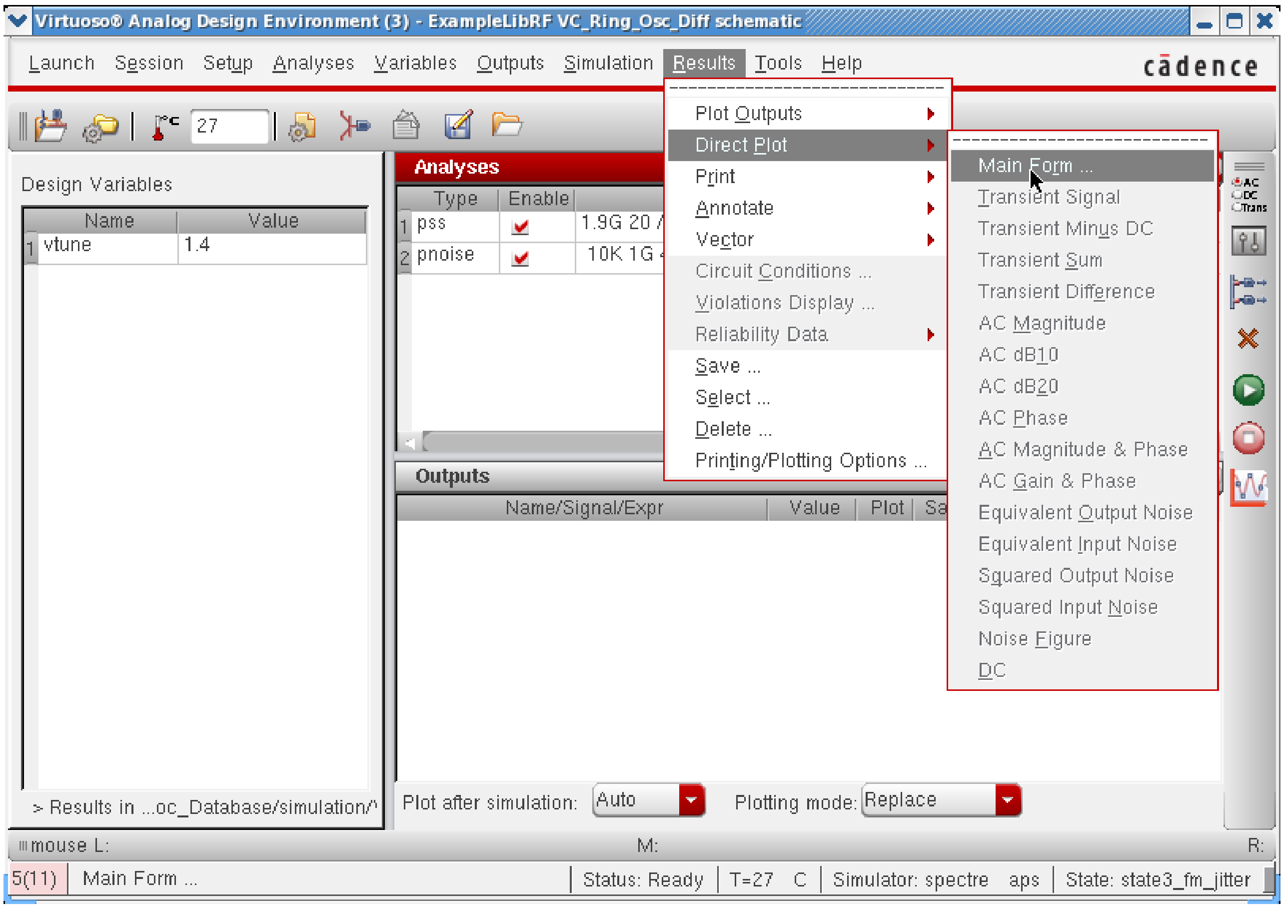
Next, you will plot the results.

Plotting the results

First Plot the oscillator phase noise based on *pnoise* analysis -

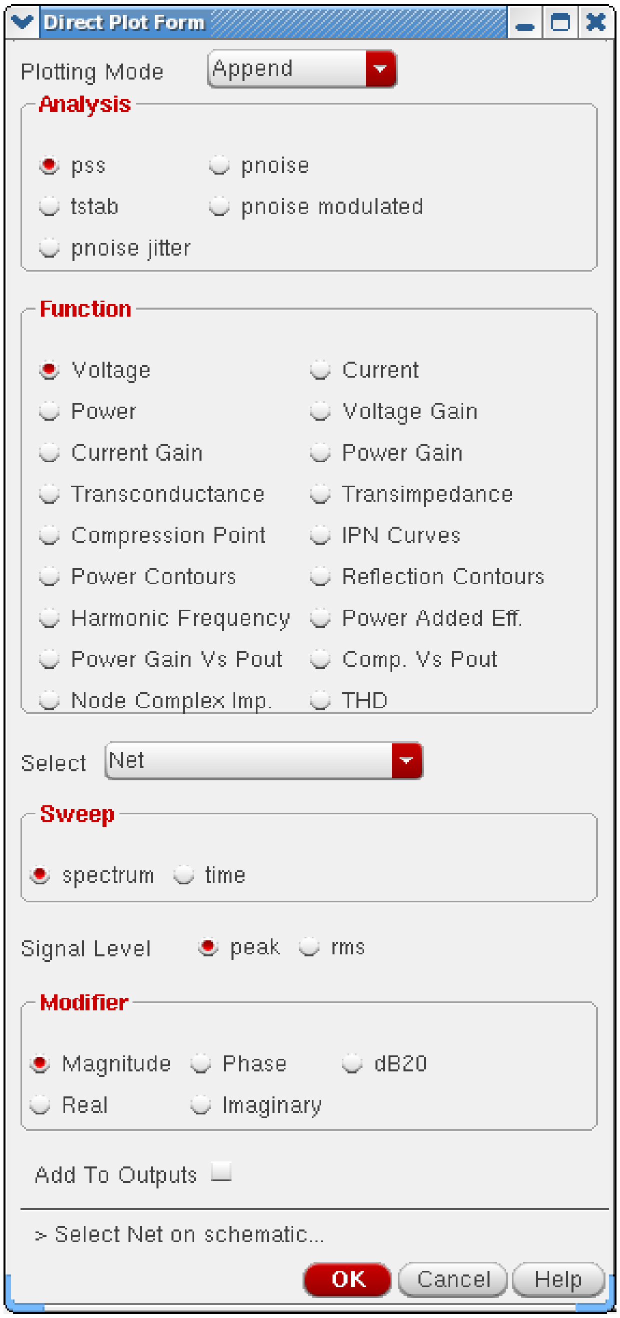
1. In the *Virtuoso Analog Design Environment* window, choose *Results - Direct Plot - Main Form*.

Figure A-261 Invoking Direct Plot Form



The *Direct Plot Form* is displayed, as shown below.

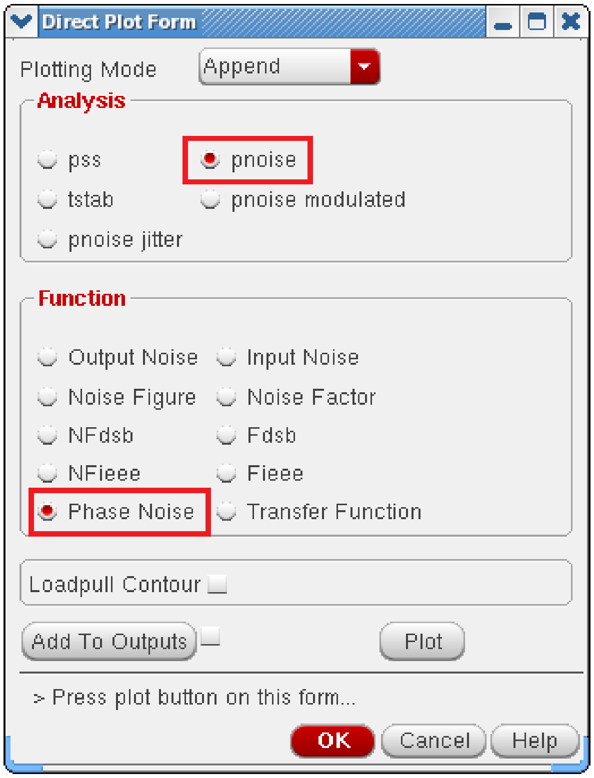
Figure A-262 *pss* and *pnoise-jitter* Analysis Direct Plot Form



1. In the *Direct Plot Form*, select *pnoise* in the *Analysis* section. The form changes.
2. Select *Phase Noise* in the *Function* section.

   The *Direct Plot Form* Setup should look like the following:

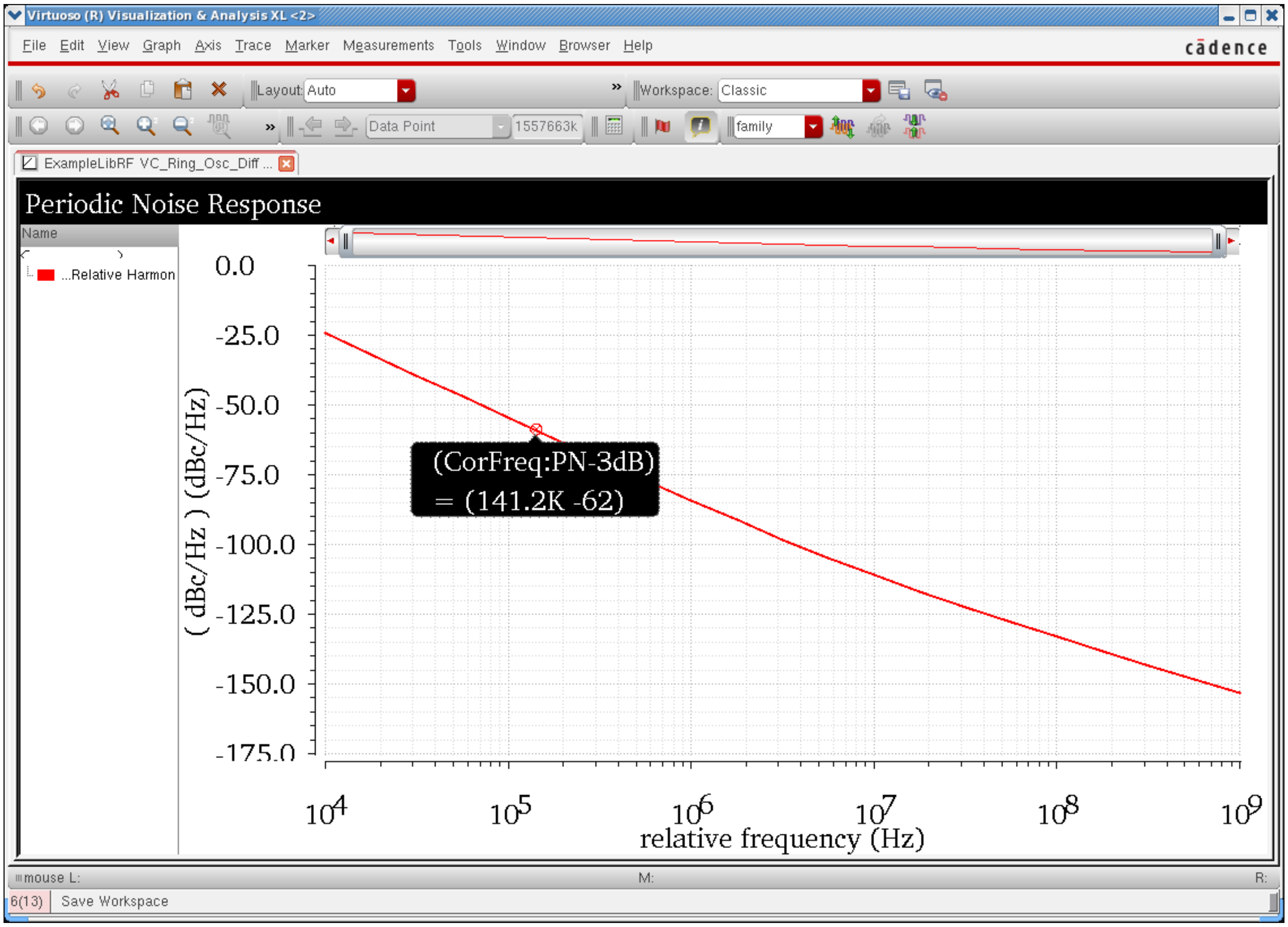
Figure A-263 ***pss*** **and pnoise FM jitter Analysis Direct Plot Form Setup**



3. Click *Plot*.

The oscillator phase noise curve is plotted. This is a Single Sideband (SSB) Plot.

Figure A-264 Phase Noise (SSB) plot from pnoise Analysis



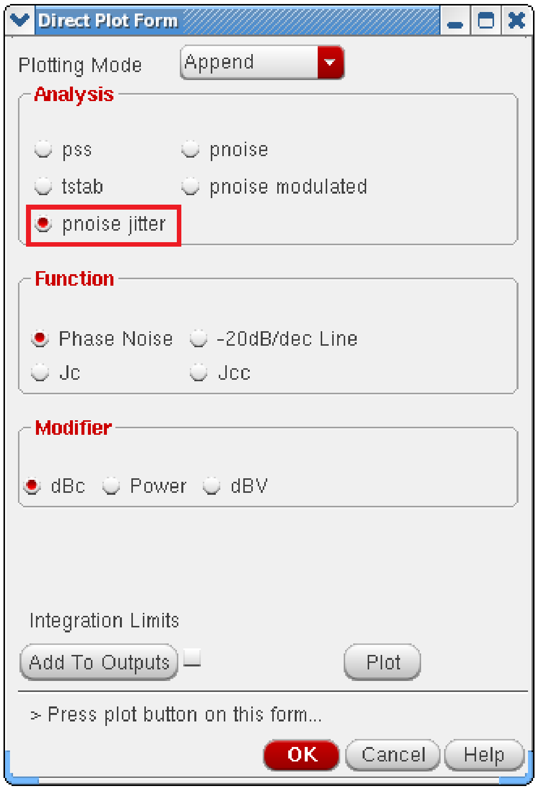
Next, you will plot the phase noise plot from *pnoise jitter* analysis. This phase noise plot will be a Double Sideband (DSB) plot which is 3dB higher than the SSB phase noise plot.

Plotting phase noise from pnoise jitter analysis

1. In the *Direct Plot Form*, select *pnoise jitter* in the *Analysis* section.
2. Select *Phase Noise* in the *Function* section. This is selected by default.
3. Set *dBc* in the *Modifier* section. This is selected by default

   The *Direct Plot Form* Setup should look like the following:

Figure A-265 *pss* and pnoise FM jitter Analysis Direct Plot Form Setup



4. Click *Plot*.

The oscillator phase noise curve is plotted.

Figure A-266 Phase Noise (Double Sideband(DSB) & Single Sideband(SSB)) plot

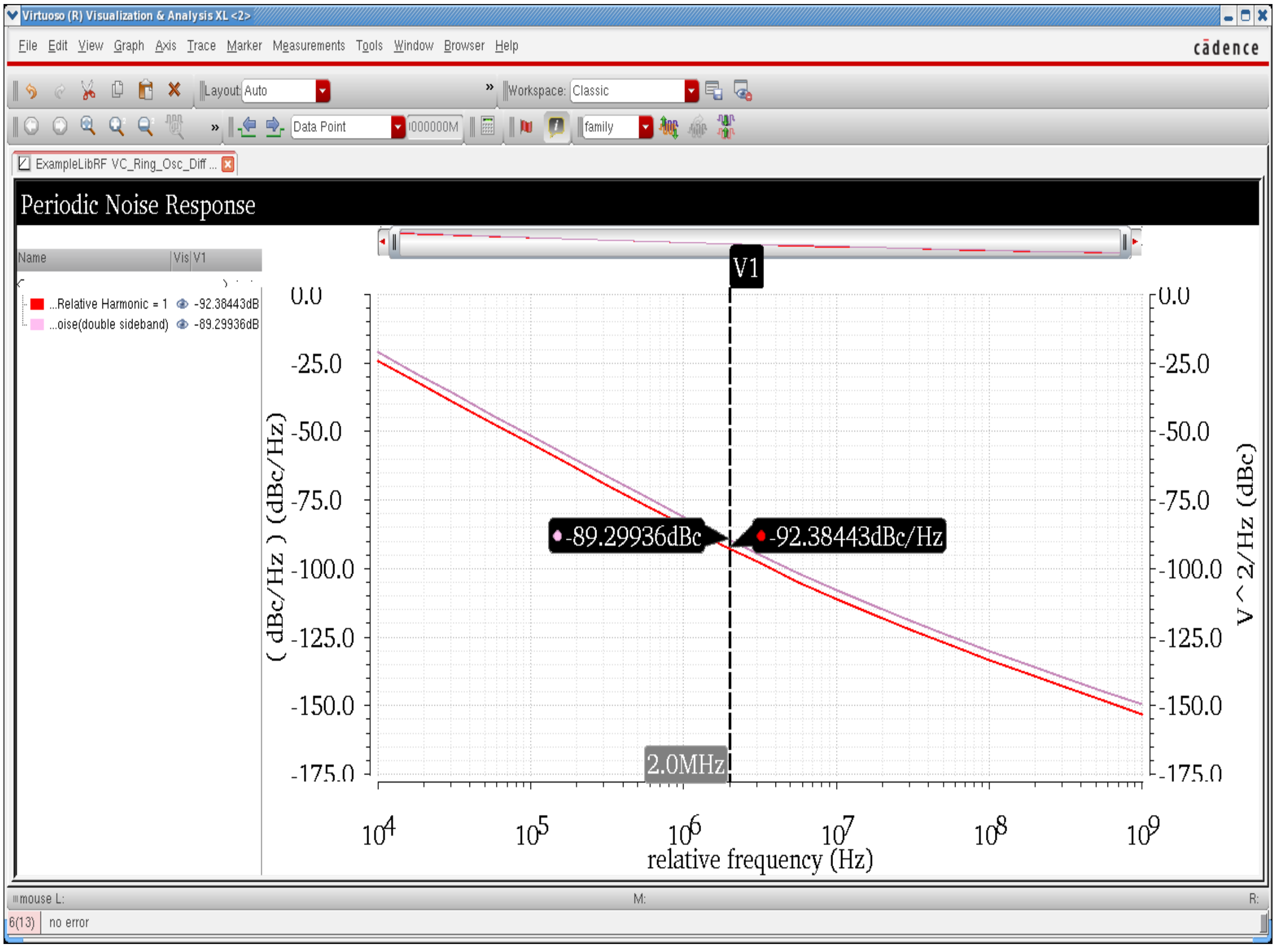
As mentioned above this is a Double Sideband (DSB) phase noise plot.

Next you need to place a vertical marker to see that the DSB phase noise plot is 3dB higher than the SSB phase noise plot. In order to do this -

5. Type v to place a vertical marker anywhere in the plot window.
6. Move your mouse cursor over the vertical marker. The cursor will change shape.
7. Clickthe right mouse button, select *Intercepts,* and set it to *On*.
8. You will see that the difference between the DSB and SSB phase noise is around 3dB.

The graph window will look like the following:

Figure A-267 DSB and SSB phase noise plot showing 3dB difference



As can be seen from the graph, in this case, a vertical marker is placed at 2MHz frequency and the SSB phase noise value is 92.38443 dBc/Hz while the DSB phase noise value is 89.29936dBc/Hz. The difference between the two values is approximately. 3dB.

Next, plot the Jc (cycle or period) jitter value.

1. In the *Direct Plot Form*, select *Jc* in the *Function* section. Jc is cycle or period jitter while Jcc is cycle-to-cycle or period-to-period jitter. The form expands.
2. Leave *Number of Cycles (k)* as 1 (which is set by default). Number of cycles determines whether one period or k-periods jitter will be computed. In this example, you are going to determine only one cycle Jc (cycle) jitter.
3. Leave *Signal Level* as *rms* (this is selected by default). *rms* jitter is typically used.

4. In the *Modifier* section, leave the default selection as *Second*. Here, the other options are *UI* which is Unit interval, and *ppm* which is parts per million.

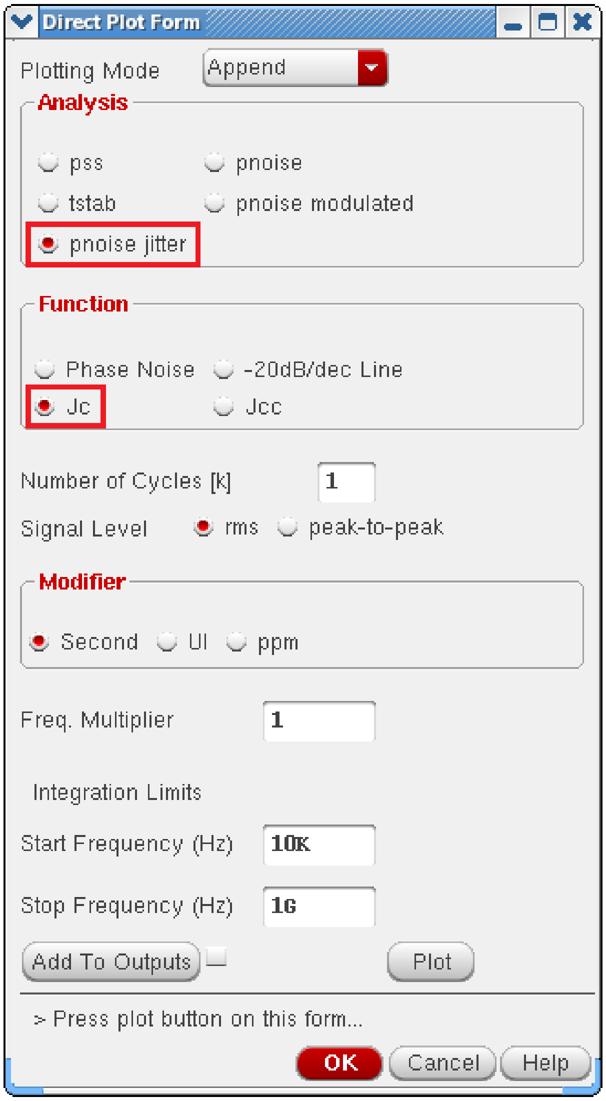
   Select the *Modifier* value based on the units you are using in your design specifications.

5. Leave the *Freq. Multiplier* set to 1 (which is by default). You use the *Freq. Multiplier* parameter when the selected output is not at the PSS fundamental/beat frequency.

6. Fill in the *Start Frequency (Hz)* and *Stop Frequency (Hz)* fields. These are the limits of integration to determine jitter. The appropriate values are included either in the jitter specifications or in the verification methodology. The best values to use also depend on the circuit properties. For a small *Number of Cycles [k]*, such as a short observation time of one or a few periods, the lower frequency noise does not contribute much to either Jc or Jcc jitter. For a larger *Number of Cycles [k]*, the lower frequency noise is important.

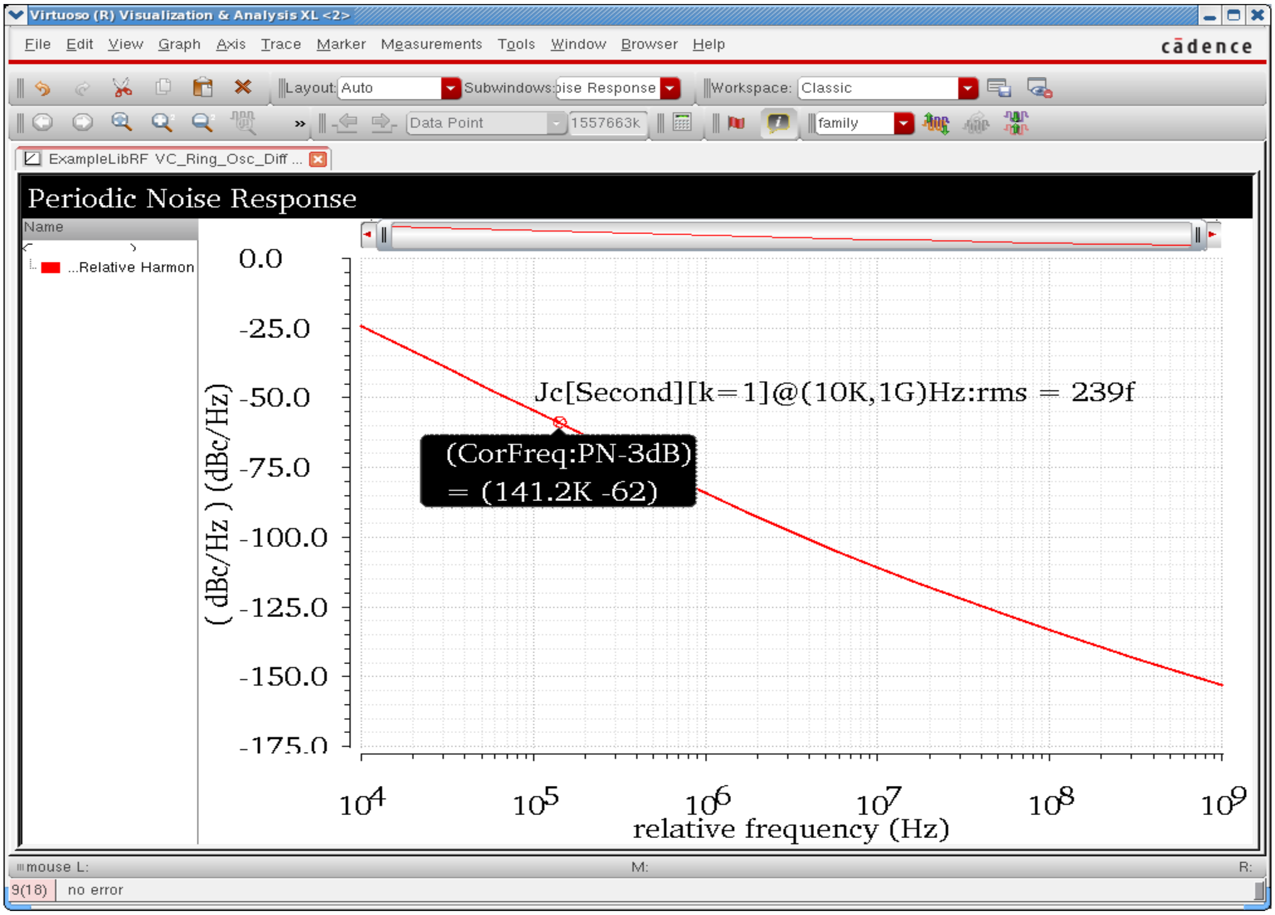
   Refer to the *Jitter Measurements Using SpectreRF* Application Note for more information on how to set up the lower limit of the integration.

Figure A-268 Direct Plot pnoise jitter Jc Setup



a. Click *Plot*. This will calculate the Jc (cycle jitter) value and the value of Jc will appear on the plot as a label, as shown below.

Figure A-269 Jc value as label on Phase Noise Plot



Since the Jc value appears as a lable on the plot, it is important that there is a plot window already open when the Jc value is plotted/calculated.

To summarize, PSS analysis using Shooting Engine and Pnoise was setup and simulation was run to determine the FM jitter of the oscillator.

Next you will determine the tuning range and phase noise of the Ring Oscillator.

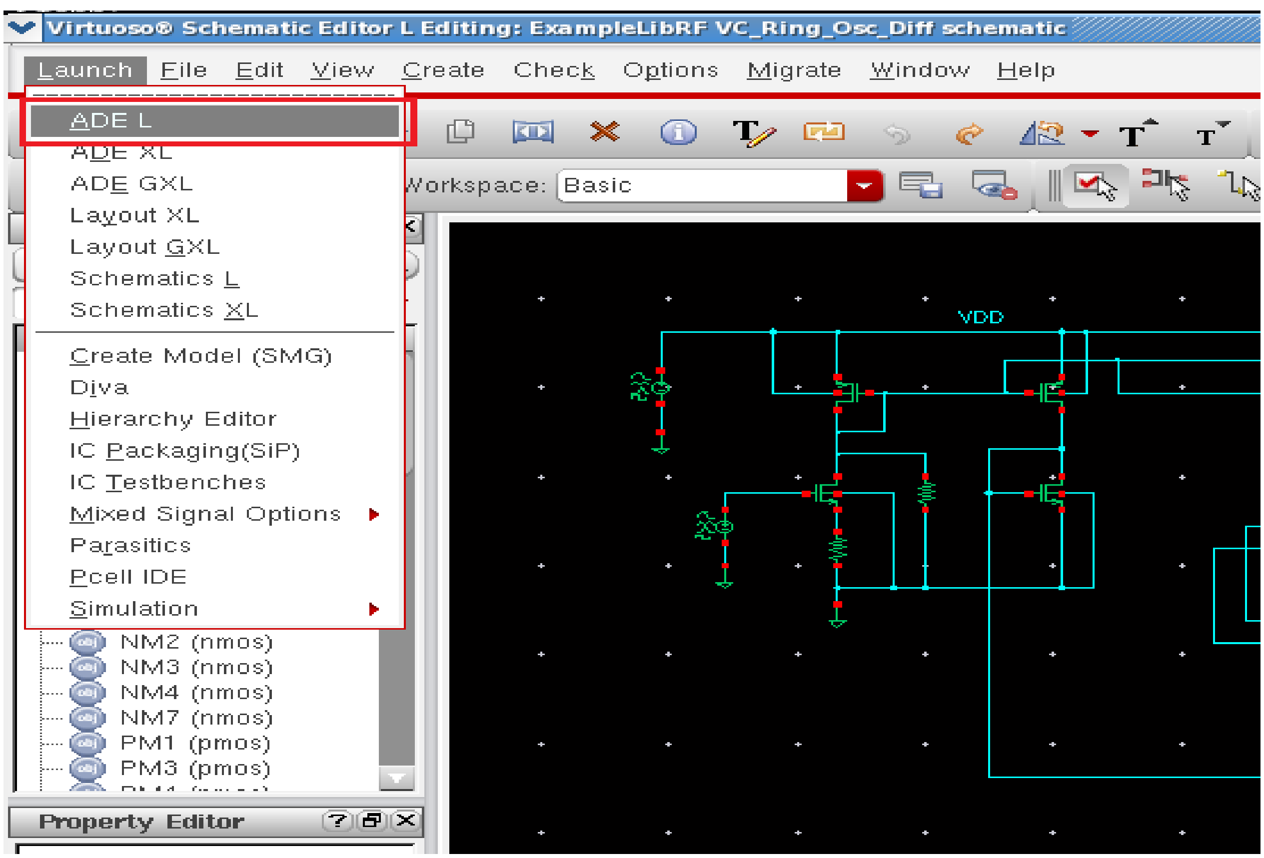
Calculating the Swept Tuning Range and Phase Noise for the Ring Oscillator

This example computes the swept periodic steady state solution using the shooting method for the *VC_Ring_Osc_Diff* oscillator circuit. You will sweep the tuning voltage to determine its effect on oscillation frequency. It then runs a periodic small-signal analysis pnoise to determine the phase noise of the oscillator. You perform a PSS-Shooting analysis first because the periodic steady state solution must be determined before you can perform any other periodic small-signal analysis,such as pnoise pxf to determine phase noise or transfer function and so on.

Setting up the PSS Analysis

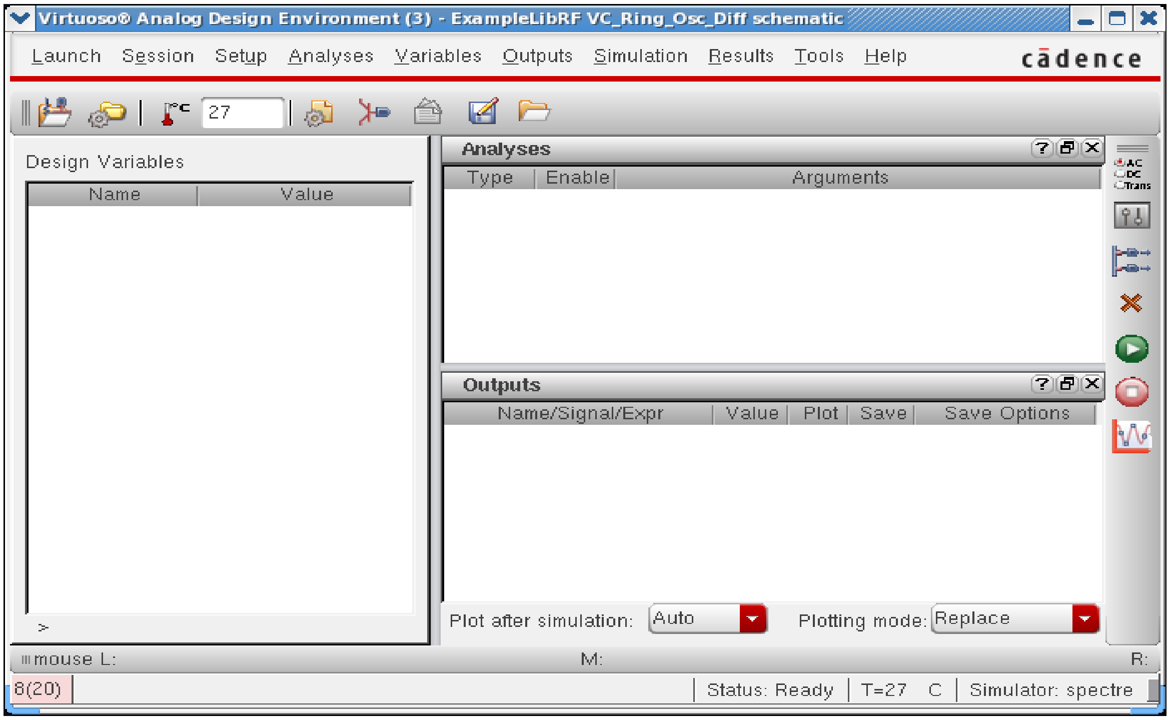
1. In the Schematic Window, choose *Launch - ADE L.*

Figure A-270 Opening ADEL window from VSE window



2. The Virtuoso Analog Design Environment Window is displayed, as shown below.

Figure A-271 Virtuoso Analog Design Environment Window



3. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

The *Choosing Simulator/Directory/Host* form is displayed.

4. Specify the following in the *Choosing Simulator/Directory/Host* form:

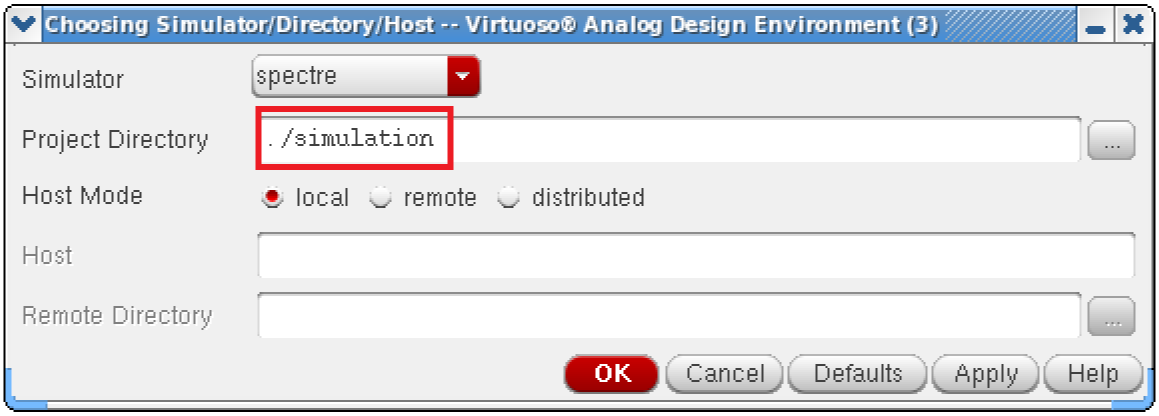
a. Choose *spectre* as the *Simulator*.

b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form isdisplayed below:

Figure A-272 Choosing Simulator/Director/Host Form

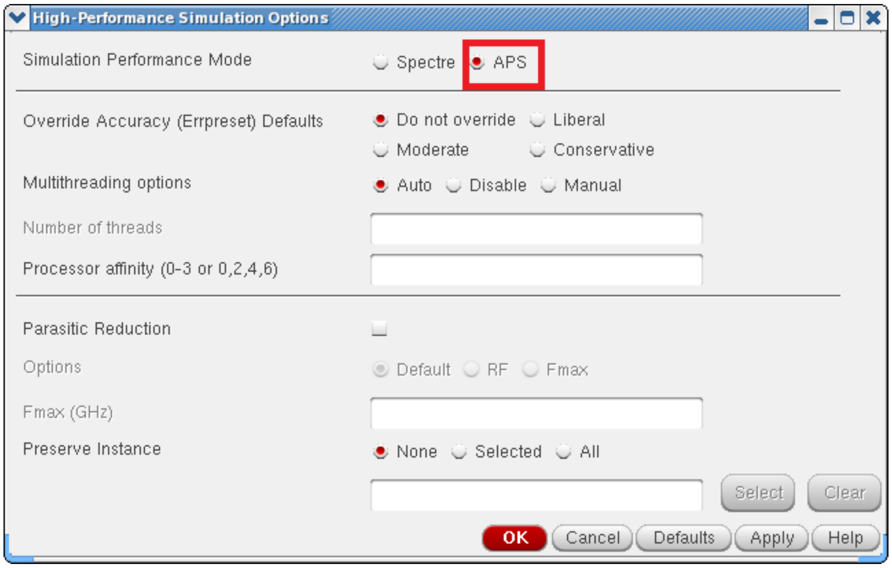


5. Click *OK* to close the *Choosing Simulator/Directory/Host* form.
6. Set up the High Performance Simulation Options,as follows:

   In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed.

   In the *High Performance Simulation Options* window, select *APS*. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. Usually it is better to specify the number of threads yourself. Small circuits should use a small number of threads, and larger circuits can use more threads. The overhead of managing 16 threads on a smaller circuit may actually slow the simulation down, compared to two or four threads. Use the number of threads that maximizes the performance.

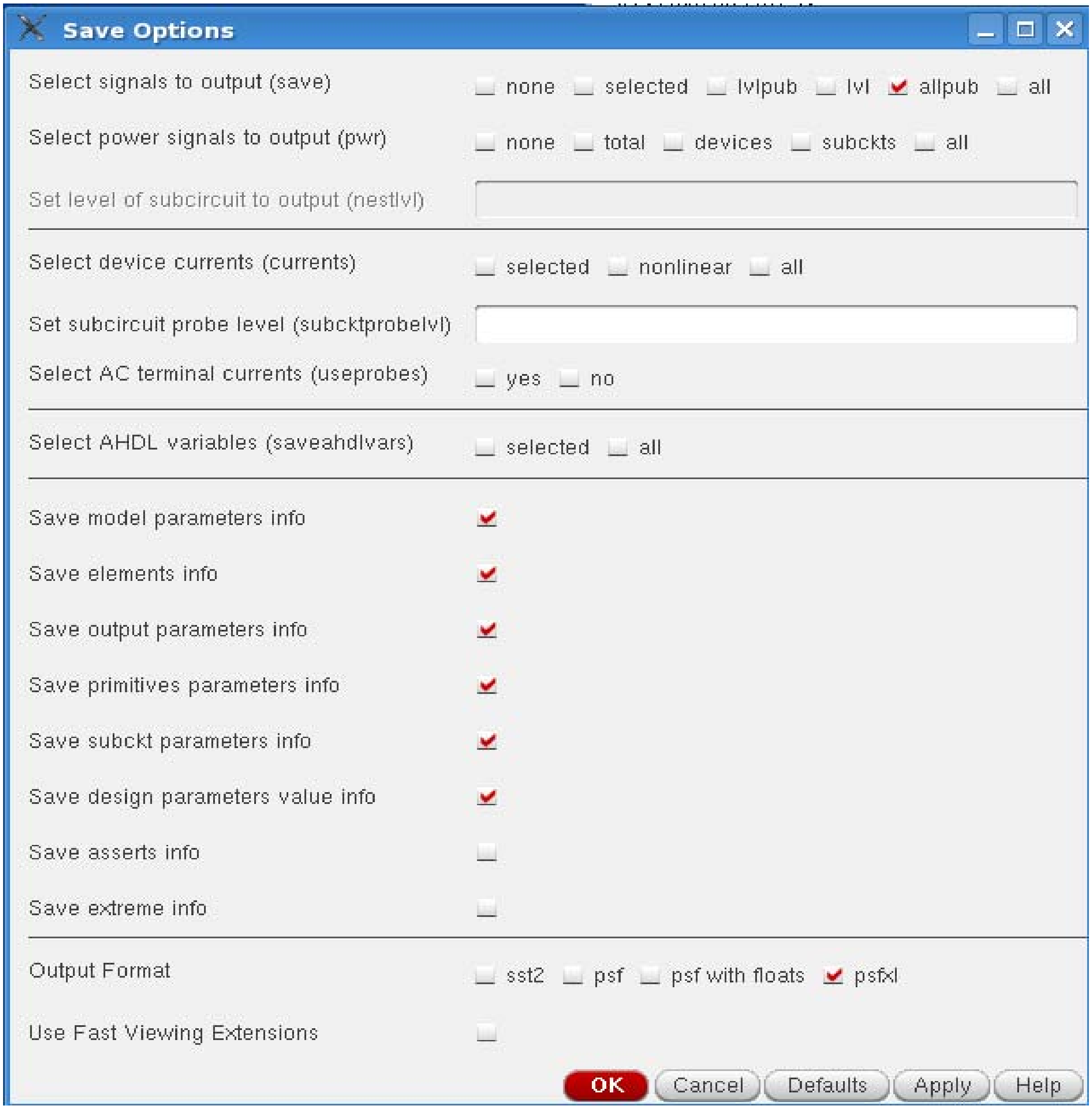
Figure A-273 High Performance Simulation Options Form



7. Click *OK* to close the *High Performance Simulation Options* form.
8. In the *Virtuoso Analog Design Environment* window, choose *Outputs - Save All*.

   The *Save Options* form is displayed.
9. In the *Select signals to output(save)* section, make sure that *allpub* is selected.

Figure A-274 Save Options Form



This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

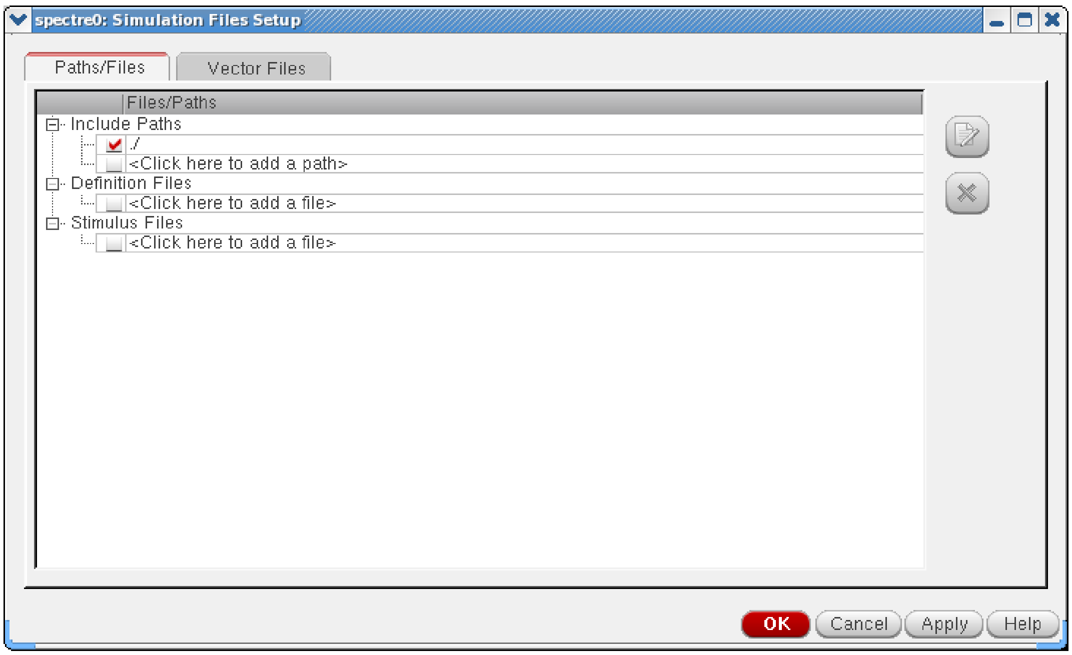
To save the currents, use the *Select device currents (currents)* option, and select *nonlinear* if you just want to save the device currents, or select *all* if you want to save all the currents in the circuit. When you save currents, you require more disk space for the results file.

10. Click *OK*.

11. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

12. In the *Simulation Files Setup* form which opens, enter `./` by clicking in the *Include Paths* section. The *Simulation Files Setup* form should look like the following:

Figure A-275 *Simulation Files Setup* Form



13. Click *OK* to close the *Simulation Files Setup* form.

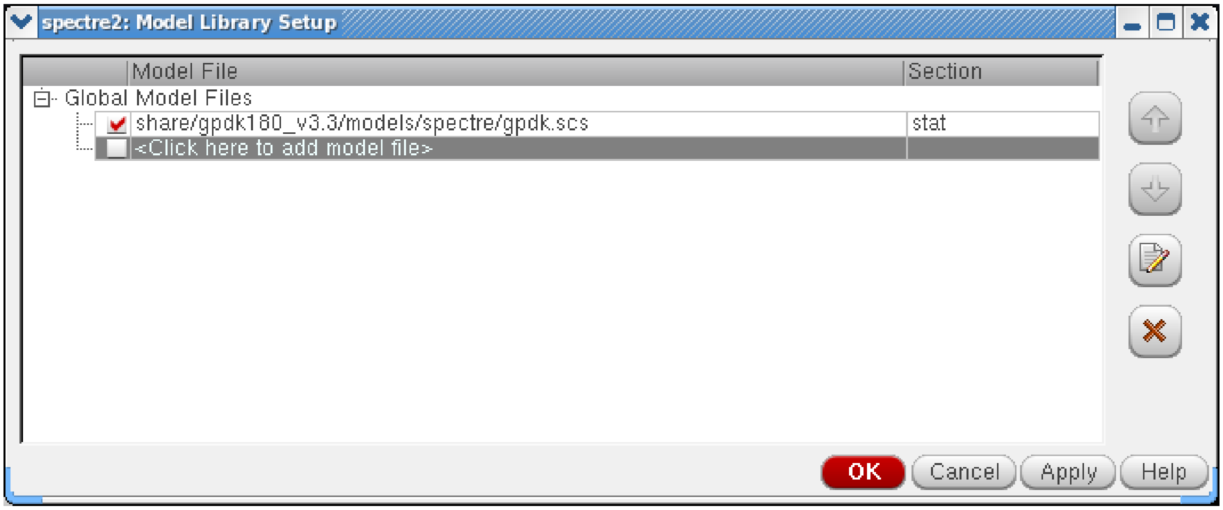
14. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries*.

The *Model Library Setup* form is displayed.

15. In the *Model File* field, type the following path to the model file including the file name:

share/gpdk180_v3.3/models/spectre/gpdk.scs.

Figure A-276 Model Library Setup Form



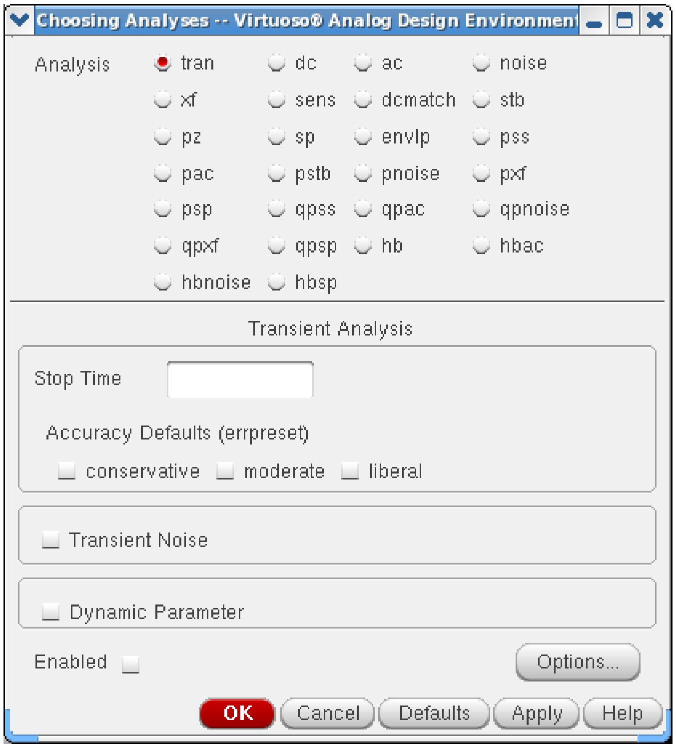
You can also browse to *gpdk.scs* file. Set the Section to *stat*.

16. Click *OK* to close the *Model Library Setup* form.

17. Choose *Analyses - Choose* in the *Virtuoso Analog Design Environment* window.

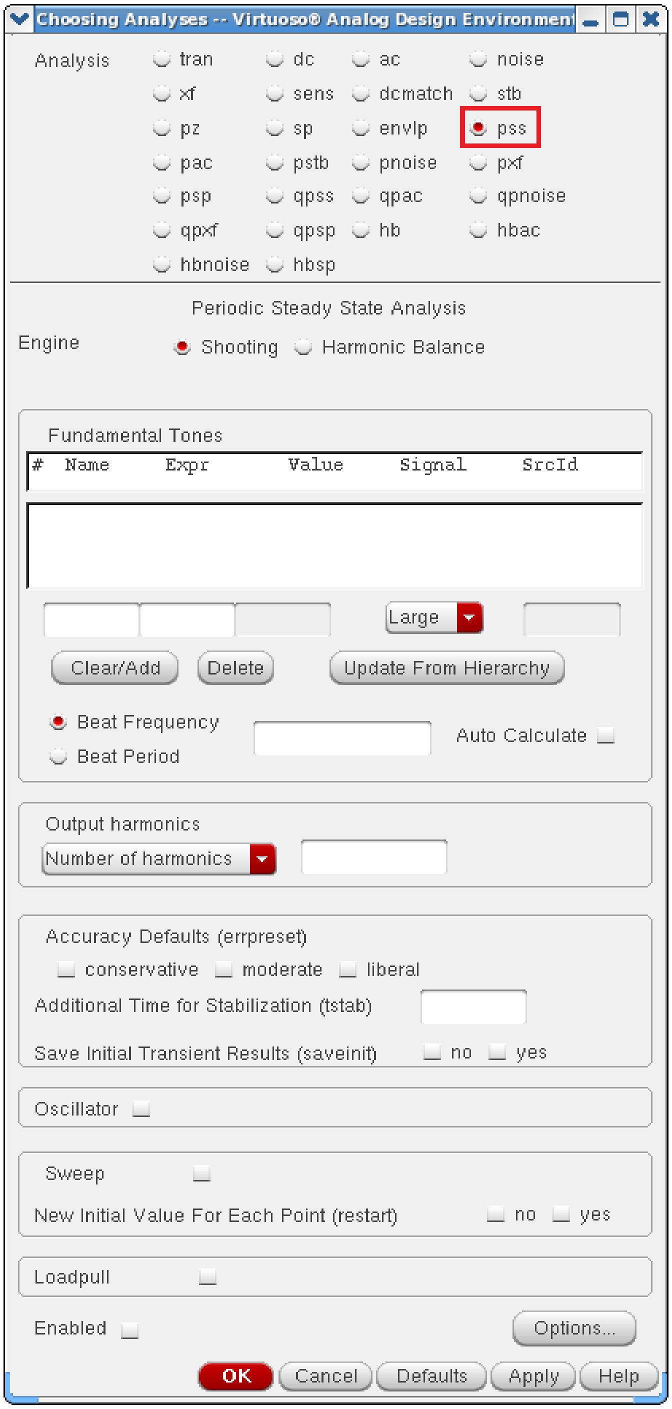
The *Choosing Analyses* form is displayed.

Figure A-277 The Choosing Analyses Form



18. In the *Analysis* section, select *pss*. The form expands, as shown below.

Figure A-278 The Choosing Analyses Form- Setting PSS Analysis



19. In the *Engine* section, ensure that *Shooting* is selected (this is the default).

20. In the *Beat Frequency* field, type 1G. The frequency entered here is an approximate frequency of oscillation. Normally, the oscillation frequency is set near the actual oscillation frequency. In this case, we have a very wide tuning range for this oscillator. A value near the middle frequency is chosen so that pss can converge on all the frequencies that are produced in the sweep.

21. In the *Number of harmonics* field, type 20.

22. In general, you want to choose a number that is high enough to capture the nonlinearity of the circuit.

For example, in this measurement you would be running fullspectrum pnoise. For fullspectrum pnoise, start with 10 pss harmonics, select *fullspectrum*, and select *APS*. Run the simulation, and plot the noise result. Now increase the number of harmonics which forces more pss timepoints, and run the simulation again. If the pnoise result did not change, then you had enough harmonics to begin with. If the noise result did change, then increase the number of harmonics and run the simulation again till your pnoise result do not change.

In this measurement 20 number of harmonics gives stable pnoise results.

23. In the *Accuracy Defaults* (*errpreset*) section, select *conservative*. *conservative* is typically used because very small amplitude phase noise measurements are normally desired. *conservative* is recommended for all the oscillators.

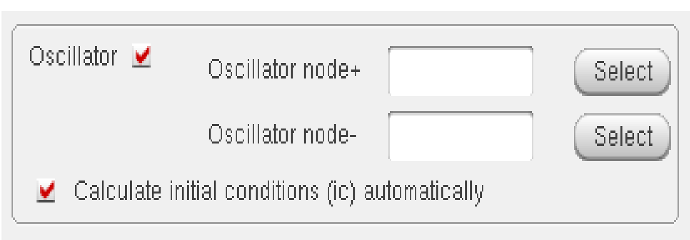
24. Type 10n in the *Additional Time for Stabilization (tstab)* field. *tstab* is typically set to about 2-3 periods of the oscillation frequency for ring oscillator circuits. Note that this needs to apply for the lowest frequency that is produced. For that reason, 10n is set.

25. Select *yes* for *Save Initial Transient Results*. This will help in visualizing the buildup of the oscillation waveform.

26. Select the *Oscillator* option. This is required for simulating an autonomous circuit.

27. The oscillator section expands, as shown below.

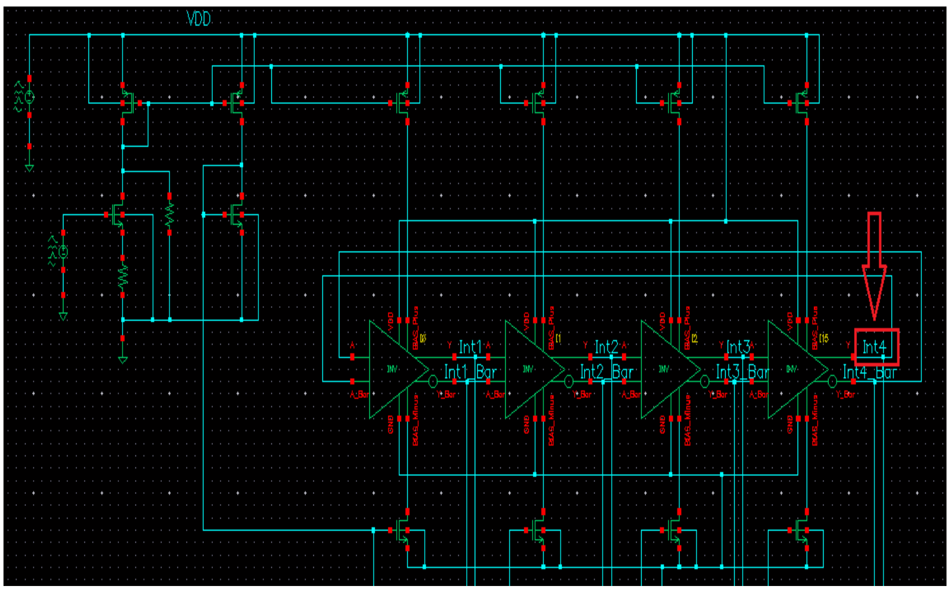
Figure A-279 The Choosing Analyses Form - Oscillator Section



28. In the *Oscillator node+* field, click *Select* on the right. In the schematic, select the *Int4* node. This oscillator node will be used by the simulator for the period calculation of the oscillations. It just needs to be a node that has the oscillator signal on it.

Also, deselect the *Calculate initial conditions (ic) automatically* checkbox. This is because selecting this checkbox only works for feedback oscillators while the ring oscillator is not a feedback oscillator. Therefore, selecting this checkbox will not be able to find an oscillatory state for ring oscillator.

Figure A-280 Selecting *Int4* net on schematic

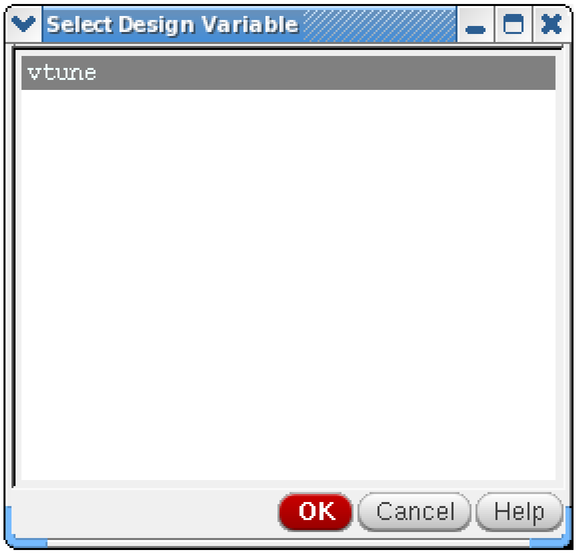


29. Select the *Sweep* option. This allows you to sweep the tuning voltage (vtune), as explained below.

30. Click the *Select Design Variable* button. The *Select Design Variable* window is displayed.

a. In the *Select Design Variable* window, select *vtune*.

Figure A-281 Choosing vtune in Select Design Variable Form during PSS Analysis setup



b. Click *OK* to close the *Select Design Variable* Form.

c. In the *Sweep Range* section of the form, type *0.0* in the *Start* field.

d. Type *3.2* in the Stop field.

e. By default, the *Sweep Type* is set to *Linear*.

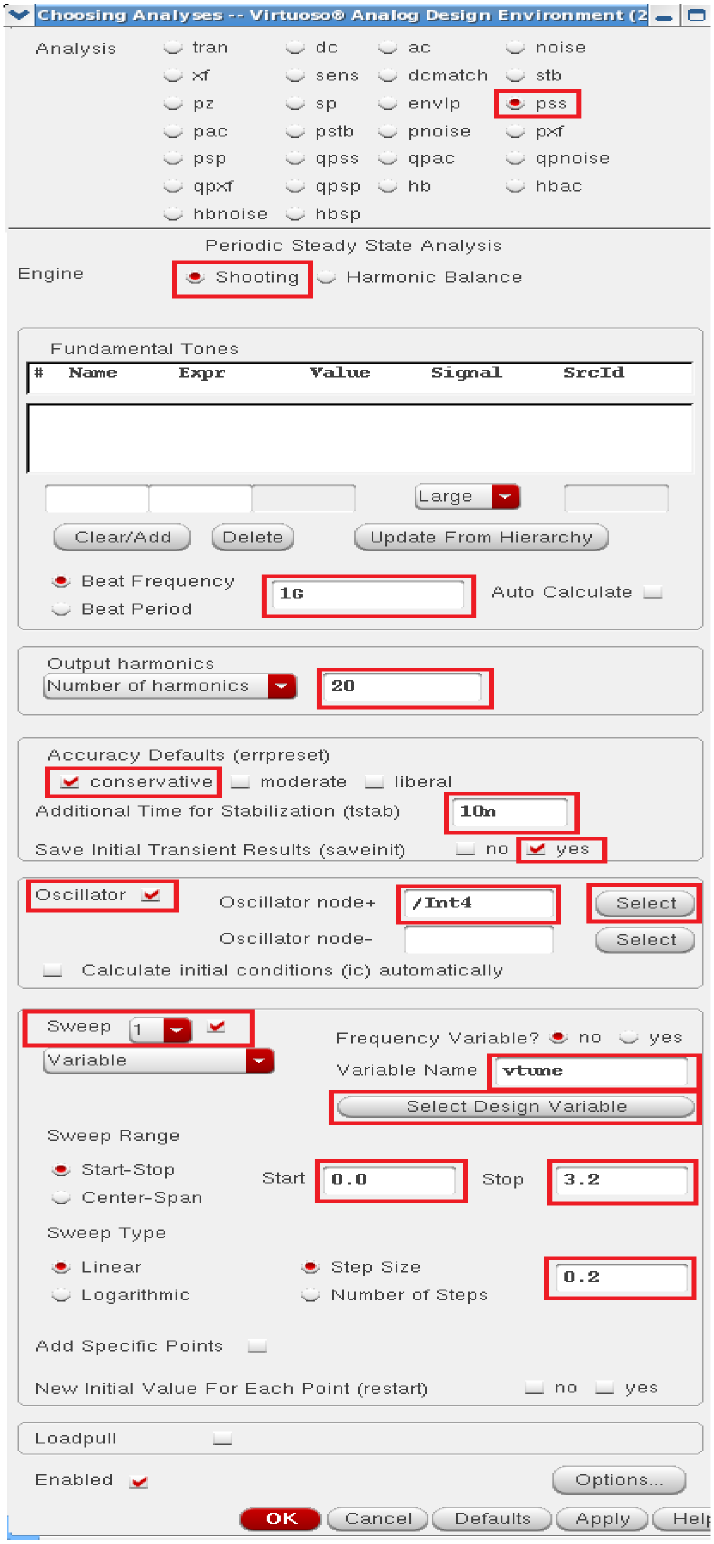
f. Type *0.2* in the *Step Size* field.

The sweep range for *vtune* is set based on for what tuning range your oscillator is designed for. This would be based on your oscillator design specifications.

31. Click *Apply* in *Choosing Analyses* Form. It checks the entries in the form for legality.

The completed *Choosing Analyses* form should like the figure below.

Figure A-282 Choosing Analyses Form - PSS-Shooting Method Setup - Part1

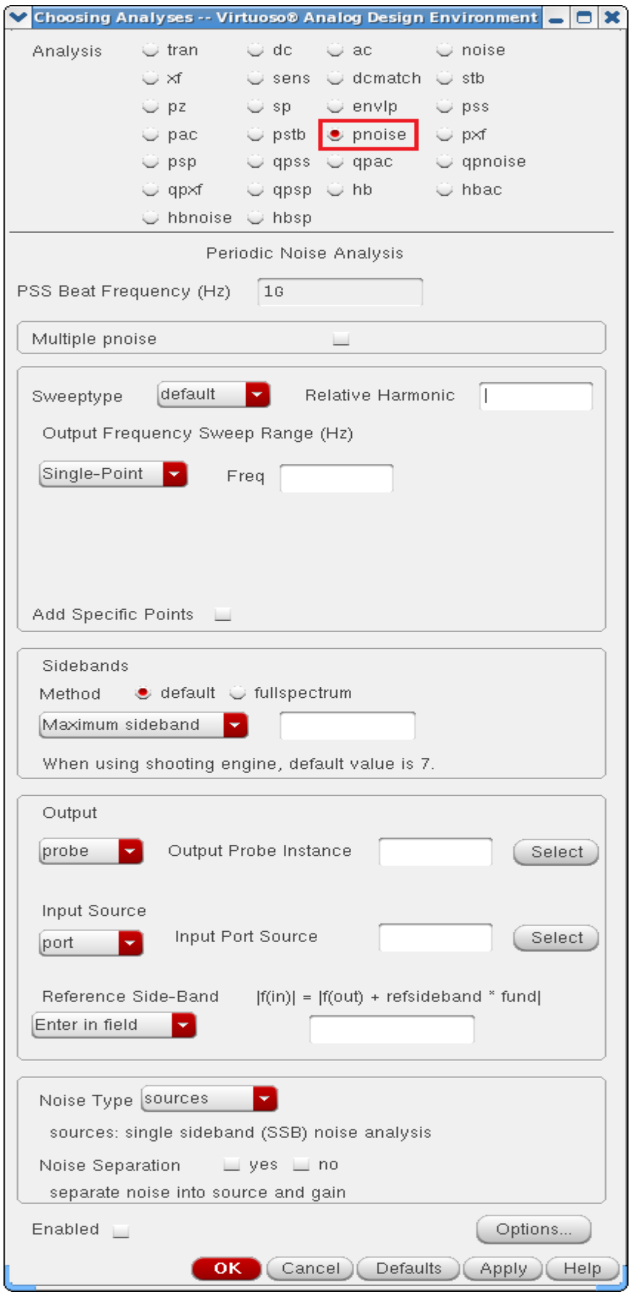


Setting up the Pnoise analysis

This analysis is set to do the phase noise measurement. It is run after *pss* analysis.

1. In the *Choosing Analyses* form, select *pnoise*. The form expands, as shown below.

Figure A-283 The Choosing Analyses Form - *pnoise* Analysis Setup



2. Leave the *Sweep Type* to *default*.

   For oscillators, the *hbnoise/pnoise* frequency range defaults to *relative*. Specify the harmonic number as appropriate for the system you are simulating. If you are simulating an oscillator by itself, then the harmonic number is likely to be 1. If you have an oscillator and a diode frequency doubler, then the harmonic number is likely to be 2. If you have an oscillator with a frequency divider, in the hb/pss form, you should specify the approximate frequency of oscillation for the frequency-divided signal. In the *hbnoise/pnoise* form, if the noise is desired on the frequency divided output, then the relative harmonic is 1. If the noise is desired at the output of the oscillator, the relative harmonic number is the divide ratio. The meaning of *relative* is to take the frequency of the harmonic number specified and add to it the frequencies specified in the *Choosing Analyses* form. If the oscillator had a 1GHz output, and the pnoise had 1M relative to the first harmonic specified, the actual output frequency is 1G + 1M, or 1001M.

   a. Type 1 in the *Relative Harmonic* field as you are simulating an oscillator by itself.

   b. Select *Single Point* for *Output Frequency Sweep Range.* Type 1M in the *Freq* Field. In general, the pnoise frequency would be set as appropriate for your application.

3. Selectthe *Method* as *fullspectrum* in the *Sidebands* section.

   Full-Spectrum pnoise is useful for circuits like switched-capacitor filters or sampling circuits where aliasing occurs through very high harmonics of the clock. The runtime advantages are large with no loss in accuracy of the result. Full-Spectrum pnoise is available when *Shooting* is selected for the pss *Engine* and *APS* is selected in the *Setup - High Performance Simulation* menu in Virtuoso Analog Design Environment (ADE) window. Selecting *fullspectrum* in the pnoise form forces *APS* to be selected in Virtuoso Analog Design Environment (ADE) window. If you are running from the command line without using *+aps*, *fullspectrum* will not be used. In normal pnoise when *Shooting* is selected as *Engine*, you need to set the maximum sideband term. In full-spectrum pnoise you do not do this except in cases where the pss beat frequency is 100KHz or less. In this case, set maximum sidebands to the 1/f noise corner frequency divided by the pss beat frequency. Pnoise calculates all the noise translations it can, based on the maximum timestep in the pss analysis. The easiest way to change the maximum timestep in pss is to increase the number of harmonics above 10 in the pss *Choosing Analyses* form. Full-spectrum pnoise runs faster than normal pnoise when the maximum number of sidebands in pnoise is about 50 or greater. The larger the number of sidebands, the larger the speedup for *fullspectrum* pnoise analysis.

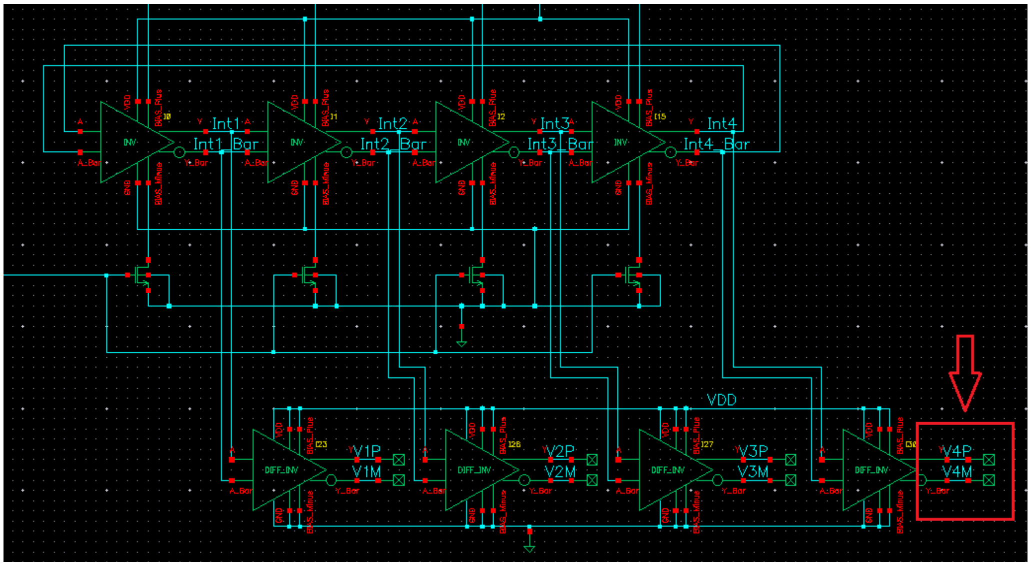
   Fullspectrum pnoise is available for all available noisetypes. Noise separation is not supported when fullspectrum is selected.

   Note: When the default method is selected, specify the number of desired frequency translations in the *Maximum sideband* field. This will always be specific to the design. For a rough estimate, plot the frequency domain content, and find the harmonic number where the amplitude has dropped about 40dB. Try this number in maximum sidebands. To verify that you have enough sidebands, raise the number by about 50% and run again. If the noise result did not change, then you had enough sidebands to begin with. Up to 40 sidebands can be specified by default. If you need more than 40 sidebands, set the number of pss harmonics to the number of sidebands divided by two to four or set the pss option maxacfreq to the beat frequency times the number of pnoise sidebands.

4. Leave *Maximum sideband* field blank as explained above since the *Method* selected is *fullspectrum*.
5. Set the *Output* to *voltage*.
   a. Type /V4P in the *Positive Output Node* field. You can also select *V4P* net from schematic by clicking *Select* button on the right of the *Positive Output Node* field and then selecting the net just below the *out* label in schematic.
   b. Type /V4M in the *Negative Output Node* field. If the second node, that is, the *Negative Output Node* is left blank, it will be connected to the global ground node automatically. However, if you have differential oscillator, you need to specify both the nodes.
   c. Set the *Input Source* to *none*.

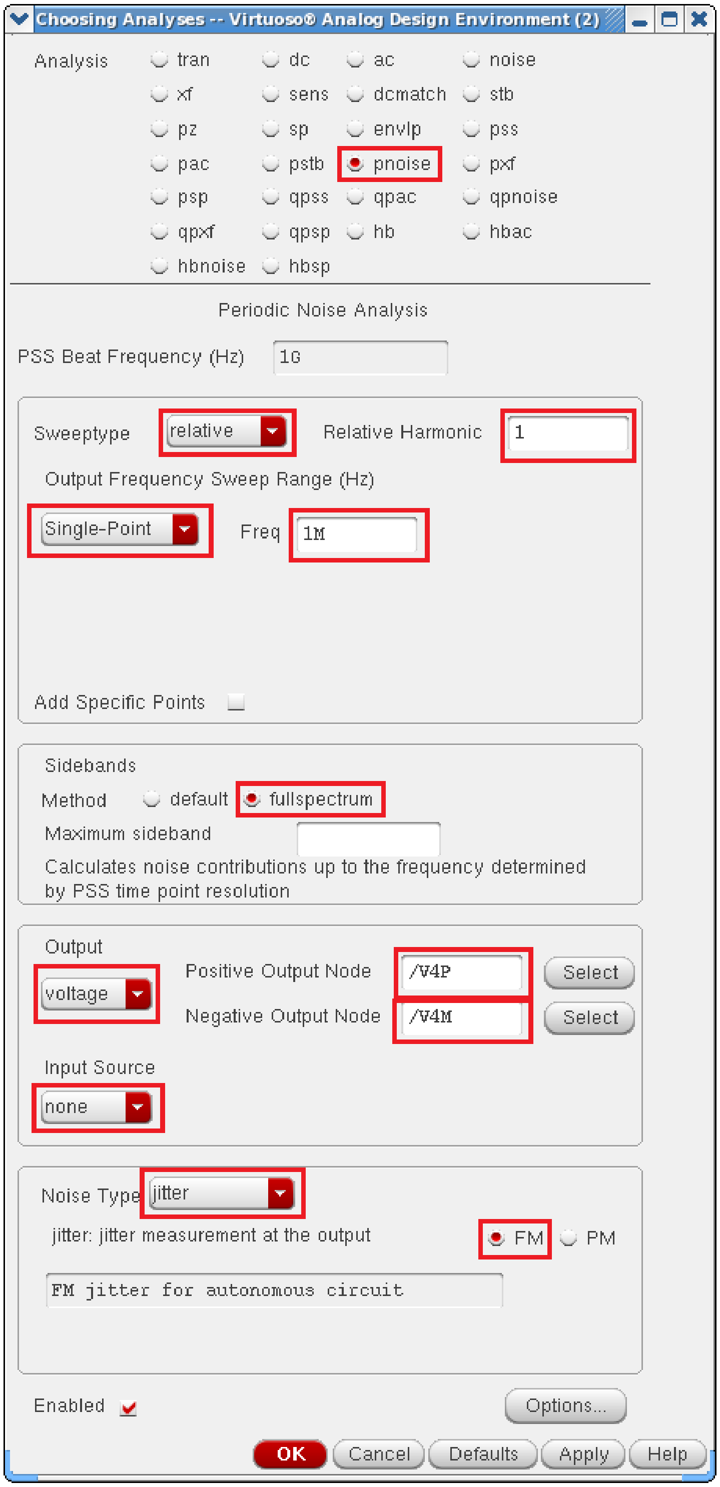
      Using an Input Source generates NF, IRN, and other parameters which are not needed here. Input Source is not usually used in oscillator simulations.
6. Choose noise type as *jitter* from the drop-down list.

Figure A-284 Selecting *V4P and V4M* net from the schematic



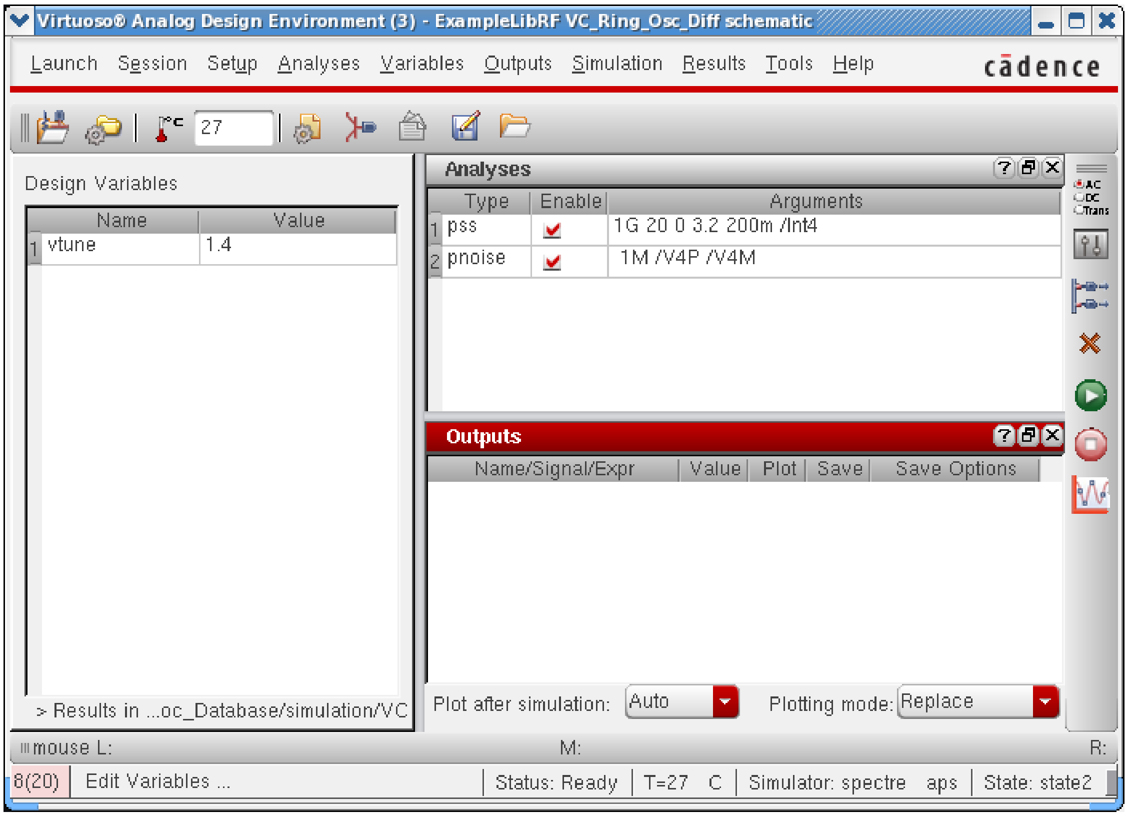
The *Choosing Analyses* form should look like the following:

Figure A-285 Choosing Analysis Form - *pnoise* Analysis Setup



7. Click *OK* to close the *Choosing Analyses* form. This will add the *pnoise* analysis along with *pss* analysis in the *Analyses* section of ADE window, as shown below.

Figure A-286 ADE Simulation Window - *pss* and *pnoise-jitter* analysis setup



Running the PSS and Pnoise analysis

Once finished setting up the PSS and Pnoise Analyses click the green icon  on the right of the Analog Design Environment window or on the Schematic window to run the simulation.

This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

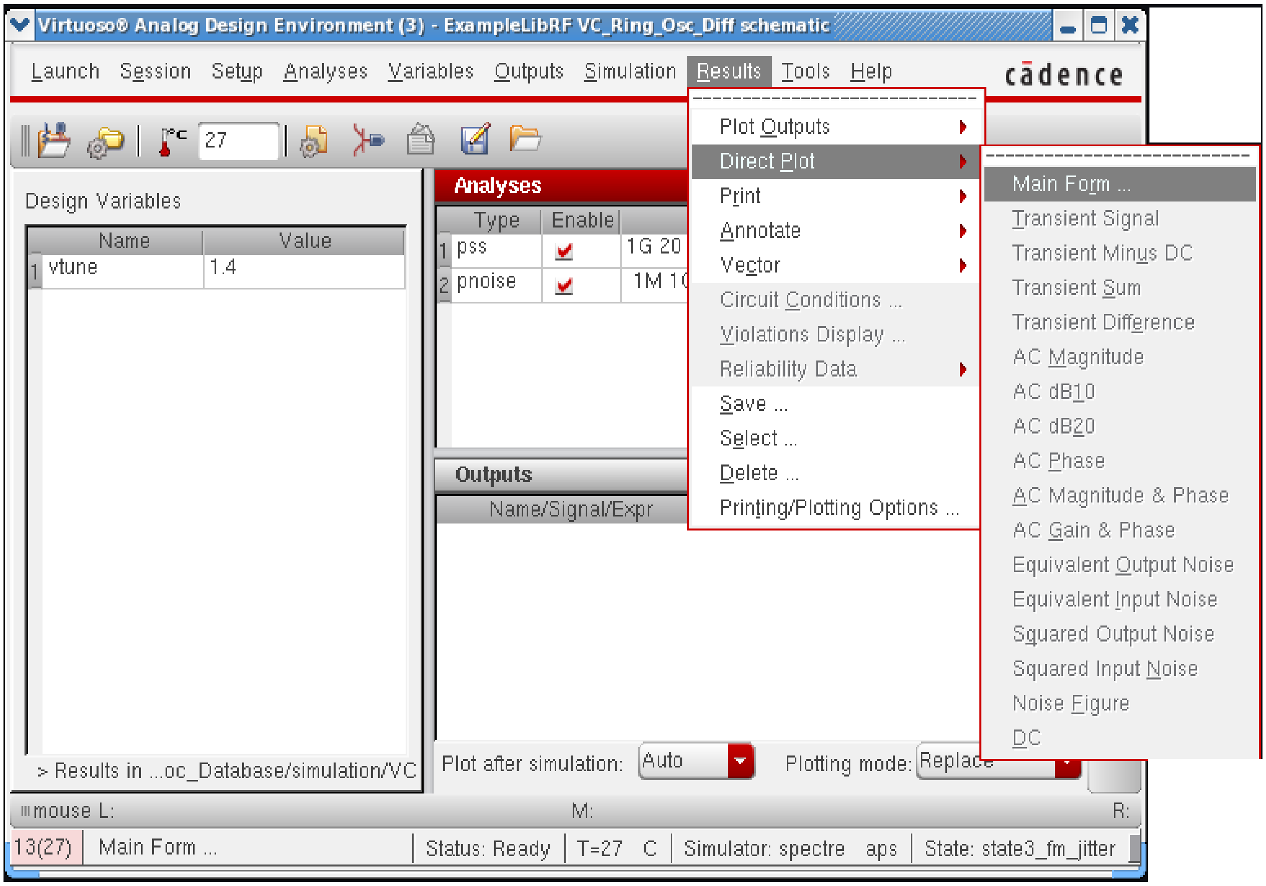
Next, you will plot the results.

Plotting the results

First Plot the oscillator output frequency, as follows:

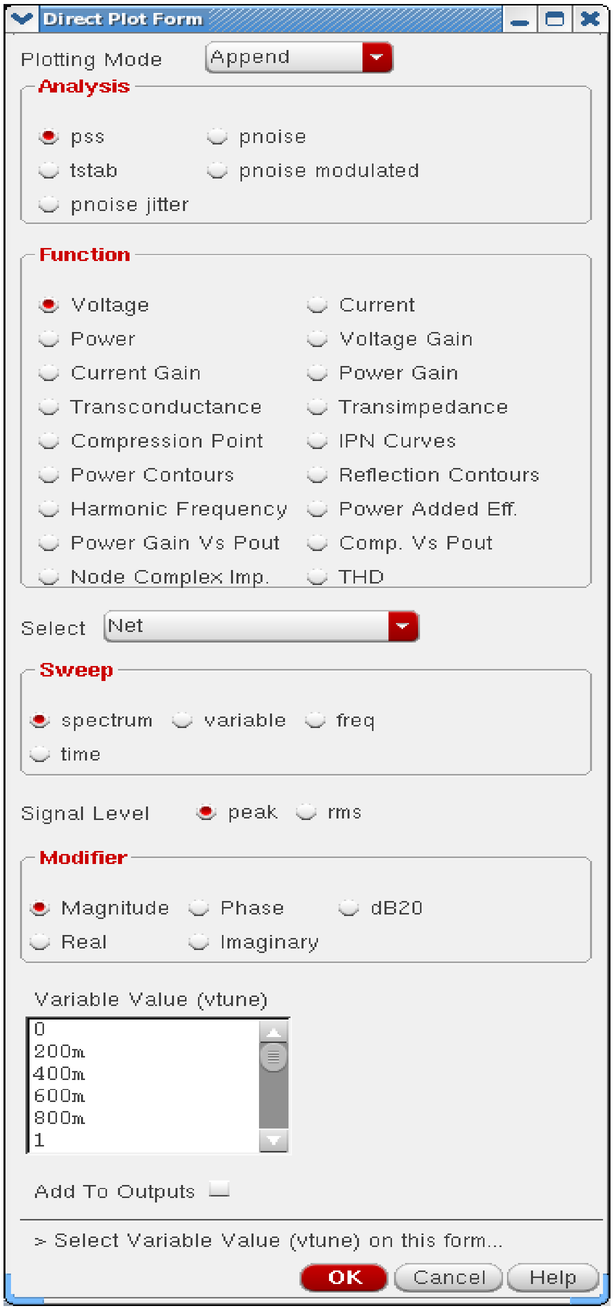
1. In the *Virtuoso Analog Design Environment* window, choose *Results - Direct Plot - Main Form*.

Figure A-287 Invoking Direct Plot Form



The *Direct Plot Form* is displayed, as shown below.

Figure A-288 Swept *pss* and *pnoise-jitter* Analysis Direct Plot Form

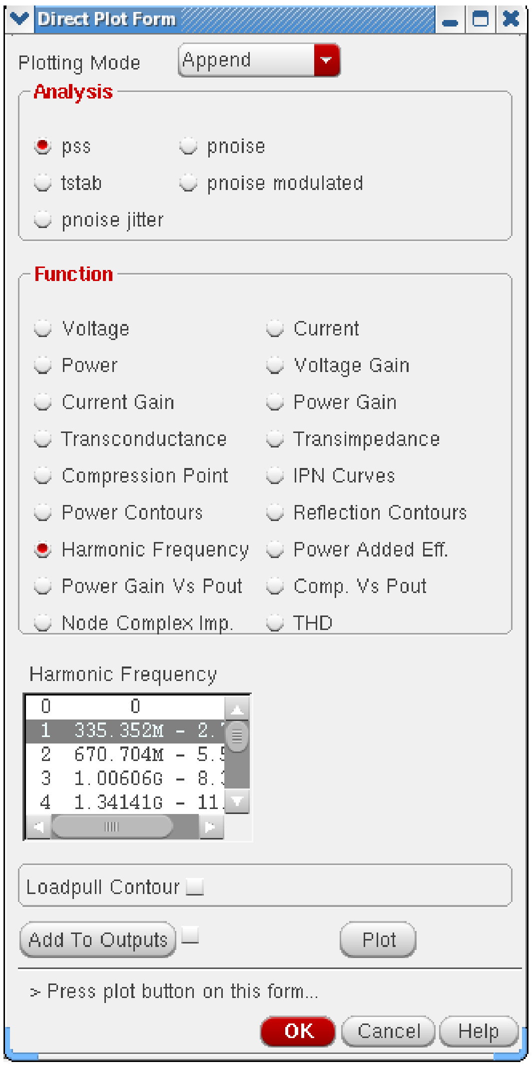


2. In the *Direct Plot Form*, select *pss* in the *Analysis* section. This is selected by default.
3. Select *Harmonic Frequency* inthe *Function* section.

4. In the *Harmonic Frequency* section , select the first harmonic. This will plot only the change in first harmonic of the oscillation frequency vs. change in oscillator's tuning voltage.

The*Direct Plot Form* should look like the following:

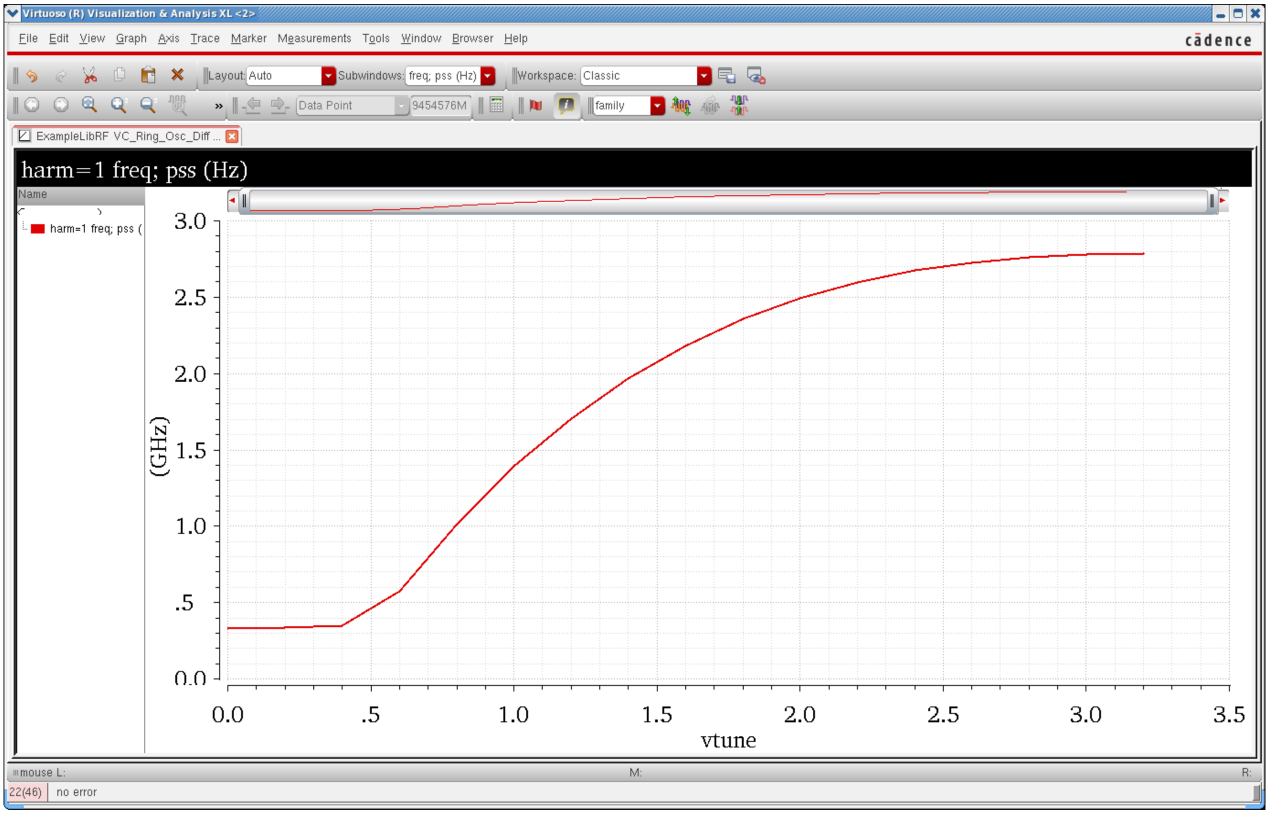
Figure A-289 Swept *pss* Analysis Direct Plot Form Setup



5. Click *Plot*.

The oscillator tuning range is plotted. From the plot you can observe that as you increase the tuning voltage, the oscillator frequency increases.

Figure A-290 Swept PSS Measurement Plot - Harmonic frequency variation plot



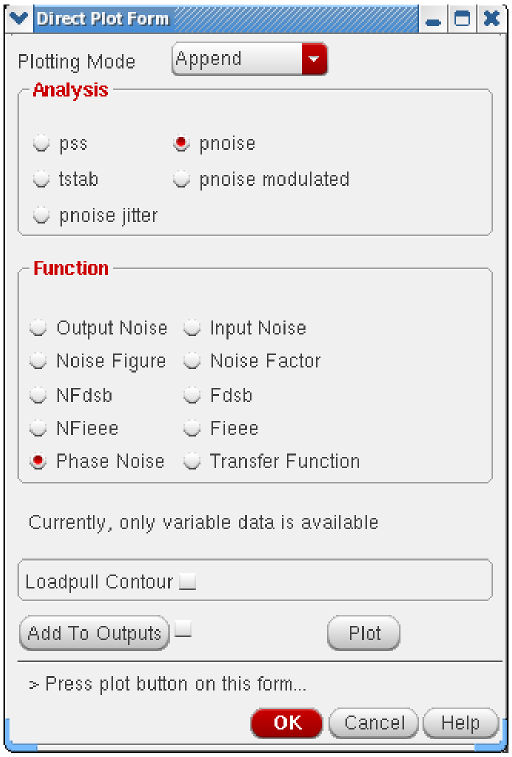
6. Choose *File - Close All Windows* to close the waveform window.

Next plot the phase noise.

1. In the *Direct Plot Form*, select *pnoise* in the *Analysis* section.
2. Select *Phase Noise* in the *Function* section.

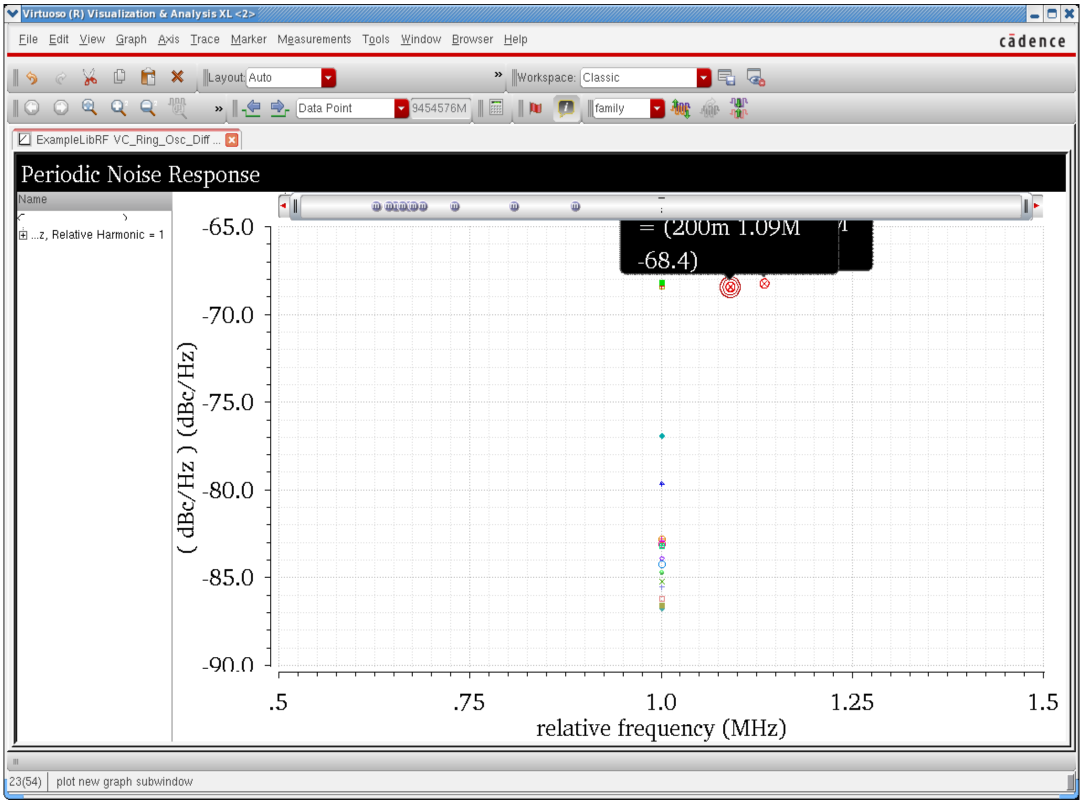
   The *Direct Plot Form* should look like the following:

Figure A-291 Direct Plot Phase Noise Setup



3. Click *Plot*. This will plot the Single Sideband Phase Noise vs. Relative Frequency graph, as shown below:

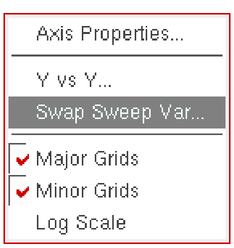
Figure A-292 Phase Noise vs. Relative Frequency Plot



In the above plot, you have a graph of phase noise vs. relative frequency for each vtune sweep value. Next, you will modify the phase noise plot to provide more useful information.

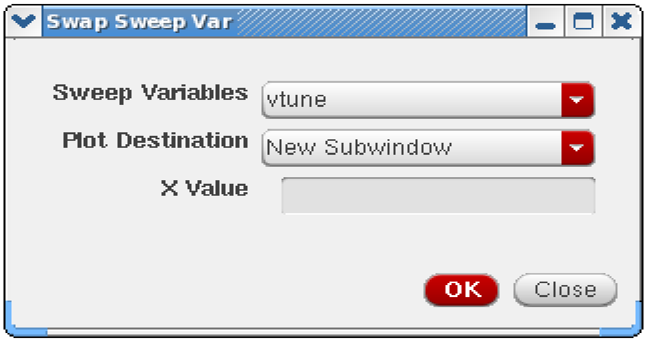
1. In the plot window, point the cursor to one of the numbers on the X Axis, click the right mouse button, and select *Swap Sweep Var* from the context menu.

Figure A-293 X-Axis - Right Mouse Button (RMB) - Object Menu



2. In *Select Sweep Var* dialog box which opens, select *vtune* from the Sweep Variables drop down list. Keep *Plot Destination* as *New SubWindow*.

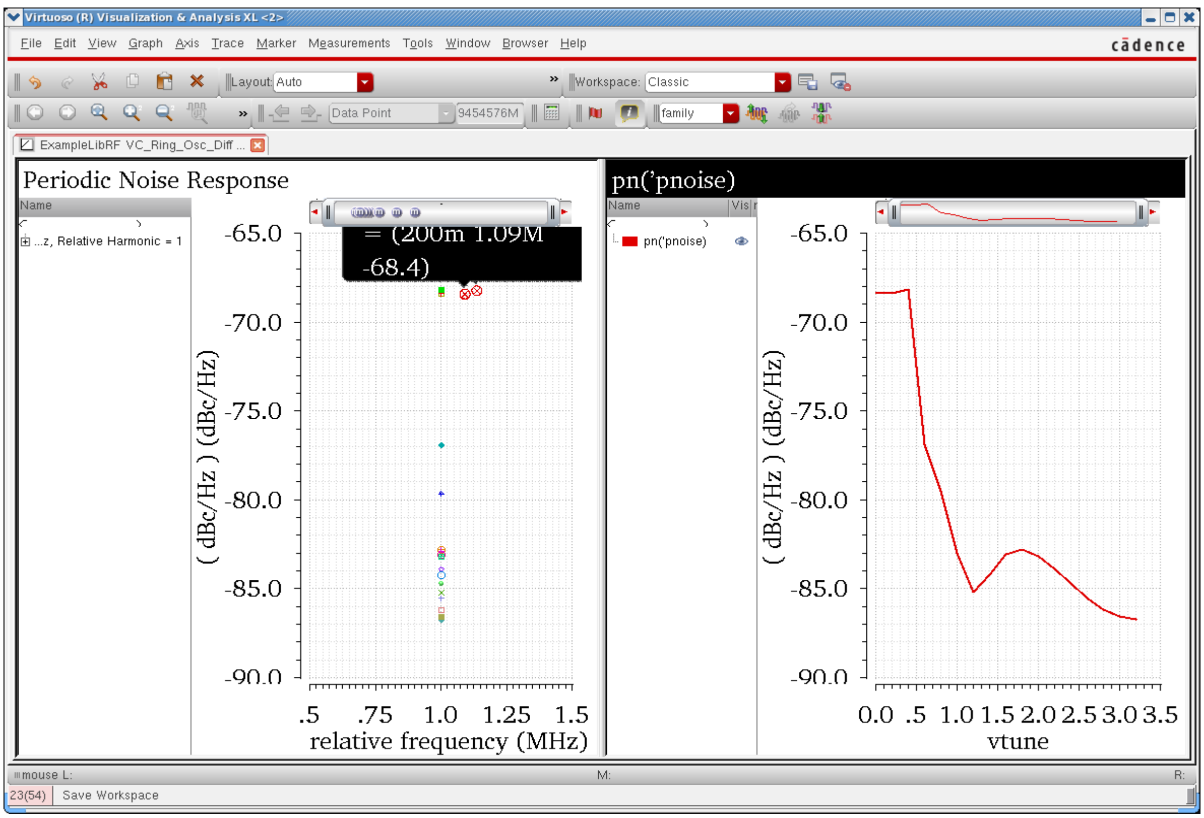
Figure A-294 Swap Sweep Var Dialog Box Window



3. Click *OK*.

   The waveform tool draws a new subwindow.

Figure A-295 Tuning voltage vs phase noise plot



If you look at the graph on the right, you will see that the phase noise is worse at the lower end of the tuning range.

4. Choose *File - Close All Windows* to close the waveform window.
5. Clean up the screen for the next set of measurements.
   a. Close the Analog Design Environment window by selecting *Session - Quit*.
   b. Click *No* in the *Save State* window.
   c. Exit the Virtuoso session by choosing *File - Exit* in CIW.

Summary

The Simulating Oscillator section shows how to simulate and make typical measurements on an oscillator.

In this Oscillaltor section, the following simulation setups/measurements were shown:

- Starting and Stabilizing of Feedback Oscillators
- Loop Gain Measurement of Feedback Oscillator
- Swept Tuning Range and Phase Noise Measurement
- Starting and Stabilization of Ring Oscillators
- FM Jitter Measurement of Ring Oscillator
- Swept Tuning Range and Phase Noise for the Ring Oscillator

For more information on simulating oscillators, please refer to the chapters in the *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator RF Analysis User Guide* and the *Virtuoso Spectre Circuit Simulator RF Analysis Theory* guide.

Simulating Mixers

The SpectreRF simulator can simulate circuits, such as mixers, that show frequency conversion effects.

This section uses two double balanced mixer circuits, `db_mixer` and `db_mixer_xmit`, to illustrate how the SpectreRF simulator can determine the characteristics of a mixer design.

In the mixer examples that follow, you will plot the following nonlinear characteristics of the `db_mixer` and `db_mixer_xmit mixer` circuits.

Receive Mixer Measurements (db_mixer)	**Analyses**
Mixer Conversion Gain and RF to IF Isolation	HB and HBAC
LO Leakage (LO to IF Leakage)	HB
Noise Figure Measurement, Noise Summary Table	HB and HBnoise
1dB Compression Point, Desensitization and Blocking	HB, HBAC, and HBnoise
Third-Order Intercept measurement with HB	HB
Rapid IP2, Rapid IP3	Specialized HBAC
Compression Distortion Summary	HB, HBAC
Transmit Mixer Measurements (db_mixer_xmit)	**Analyses**
Image Rejection	HB
Three tone Swept IP3 (large signal)	HB
Noise Figure	HB and HBnoise

To use the examples in this section, you must be familiar with the SpectreRF simulator analyses as well as know about mixer design. For more information about the SpectreRF simulator analyses, see *SpectreRF Simulation Option Theory*.

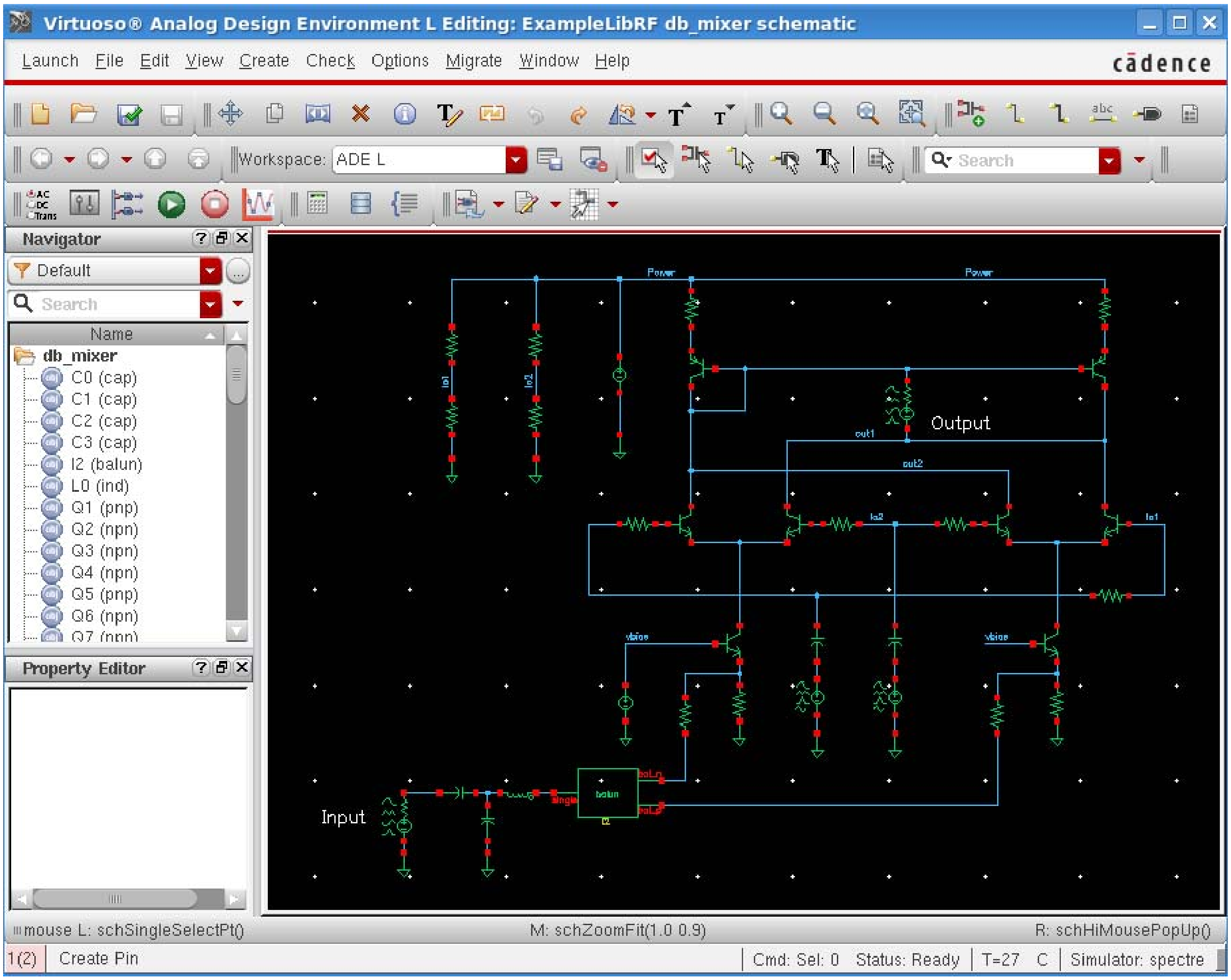
The db_mixer and db_mixer_xmit Mixer Circuits

The `db_mixer` and `db_mixer_xmit` circuits canbe found in the *ExampleLibRF* library. See Introduction on page 1903 for the instructions on accessing the*ExampleLibRF* library.

The `db_mixer` integrated circuit is a Gilbert cell (down-converting double balanced) mixer.

The schematic for the `db_mixer` circuit is shown below.

Figure A-296 Schematic for the db_mixer Mixer Circuit



On the left side of the schematic there is a port labeled *Input* which generates the input signal. To the right of that is a matching network and a behavioral balun from rfLib. This feeds the input to a double-balanced mixer. There are two LO sources in the circuit (in the middle of the schematic) between the bottom devices. The LO operates at 1.9GHz. Next to the label *Output*, is the output port of the mixer.

The `db_mixer_xmit` circuit is a similar circuit, but an up-converting double-balanced mixer with image rejection.

The following tables lists some measured values for different aspects of the `db_mixer` down-converting mixer.

Measurement	Measured
LO frequency (Hz)	1.9 GHz
RF frequencies (Hz)	1.904 GHz, 1.905 GHz
IF frequency (Hz)	4 MHz, 5 MHz
LO voltage	200mV peak
RF power	-30 dBm
Conversion gain	*measurement needed*
Double Sideband Noise figure	*measurement needed*
Input 1dB compression point	*measurement needed*
Input IP3 (from swept power)	*measurement needed*
Input IP3 (from HBAC analysis)	*measurement needed*

Design Variable	Default Value
prf (RF power)	-30 dBm
vlo (LO magnitude)	200 m
frf1 and frf2 (RF frequencies)	1.904 GHz, 1.905 GHz
flo (LO frequency)	1.9 GHz

Setting Up to Simulate the db_mixer Mixer

In a Unix window, type `virtuoso &` to start the Cadence software. (For more information, see Introduction on page 1903).

Opening the db_mixer Mixer Circuit in the Schematic Window

1. In the CIW (Command Interpreter Window), choose *File – Open*.

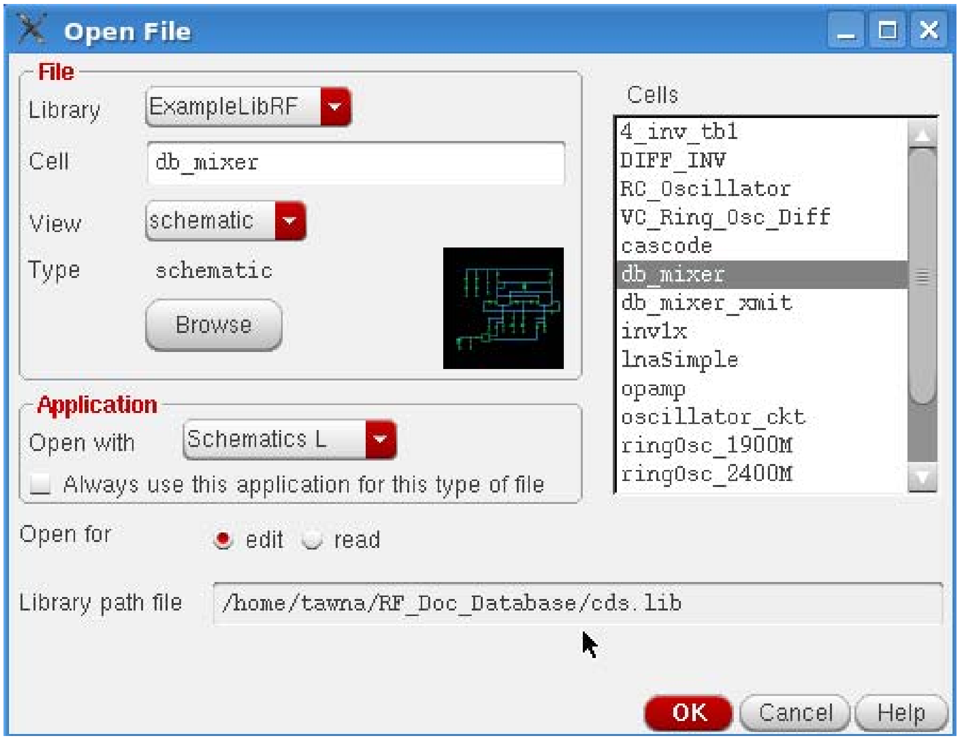
   The Open File form is displayed.

2. In the Open File form, choose *ExampleLibRF* from the Library drop-down list.

3. Type *db_mixer* in the Cell field or select the cell from the *Cells* list box.

   The completed *Open File* form appears like the one below.

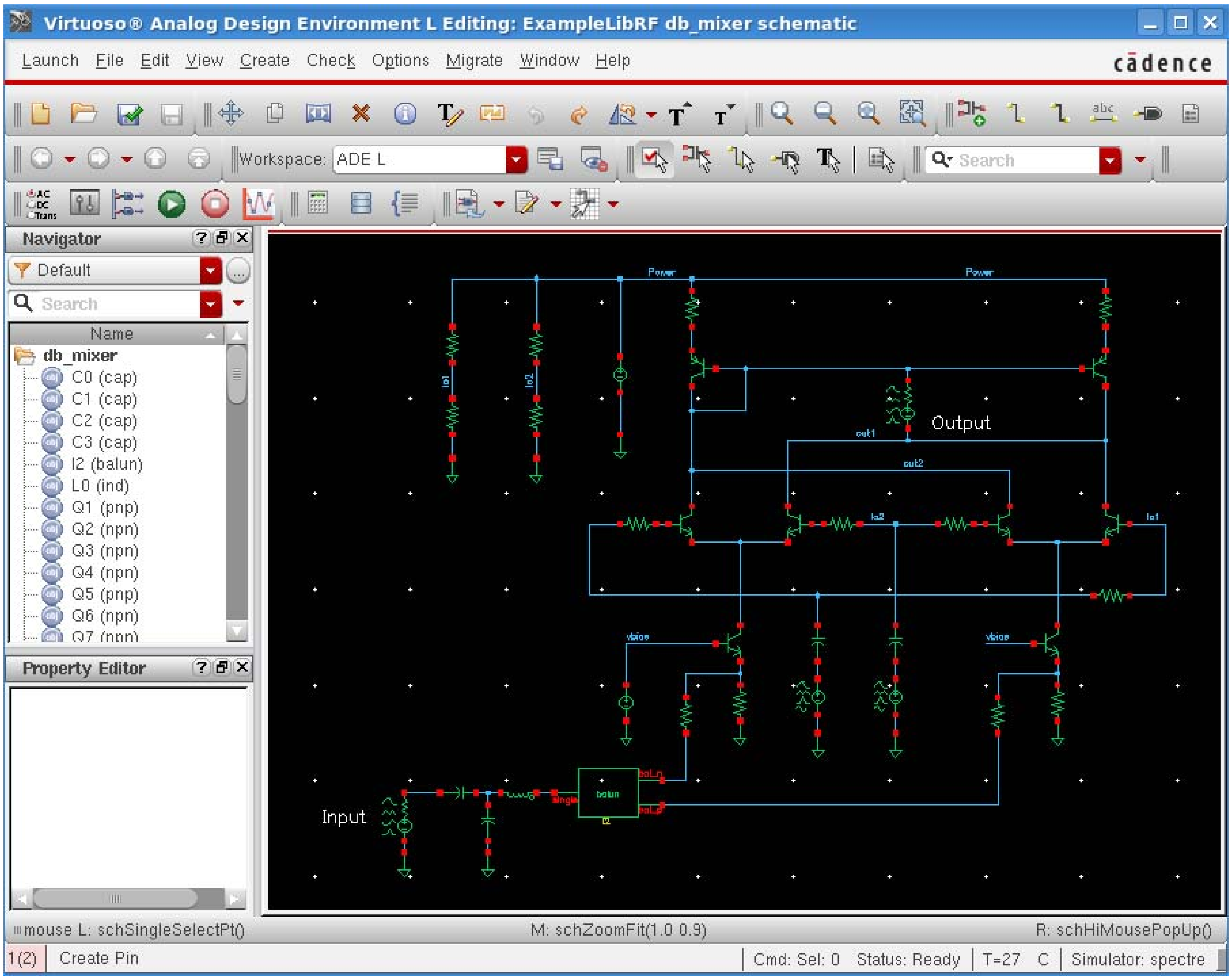
Figure A-297 Open File Form



4. Click *OK*.

   The Schematic window for the *db_mixer* mixer is displayed.

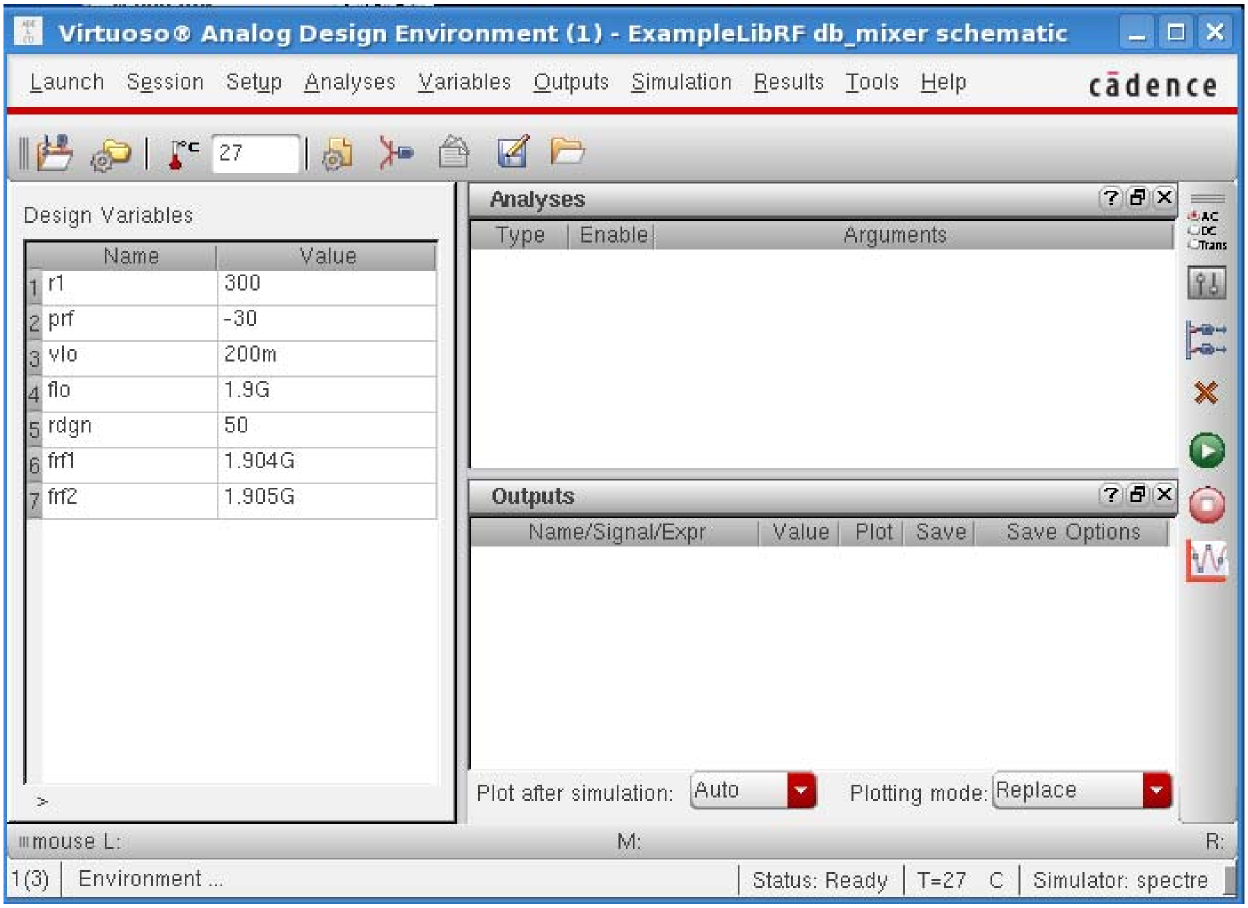
Figure A-298 db_mixer Schematic



5. In the Schematic window, choose *Launch – ADE L*.

   The V*irtuoso Analog Design Environment* window is displayed.

Figure A-299 Analog Design Environment Window



Choosing Simulator Options

1. Choose *Setup – Simulator/Directory/Host* in the Virtuoso Analog Design Environment window.

   The *Choosing Simulator/Directory/Host* form is displayed.

2. Specify the following in the *Choosing Simulator/Directory/Host* form:

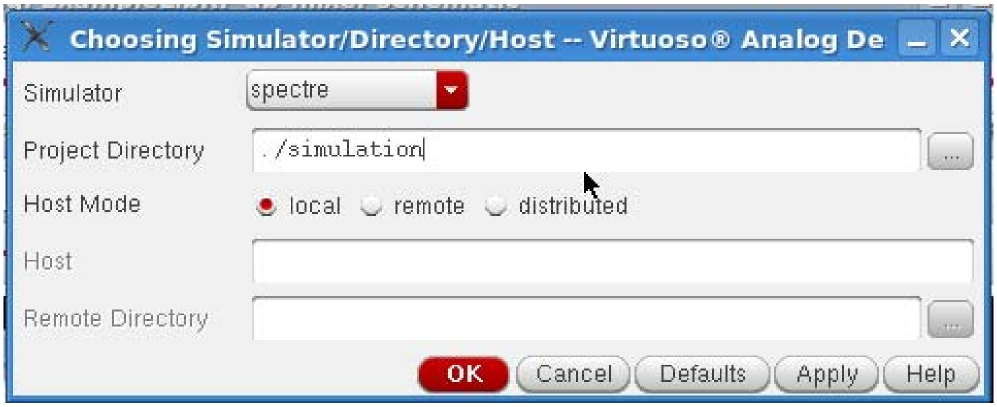
   a. Choose *spectre* from the *Simulator* drop-down list.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory. You may change the default location by editing the settings in your `.cdsinit` or `.cdsenv` file. In this workshop, the simulation directory is set to `./simulation`.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form appears similar to the one below.

Figure A-300 Choosing Simulator/Directory/Host Form - Completed

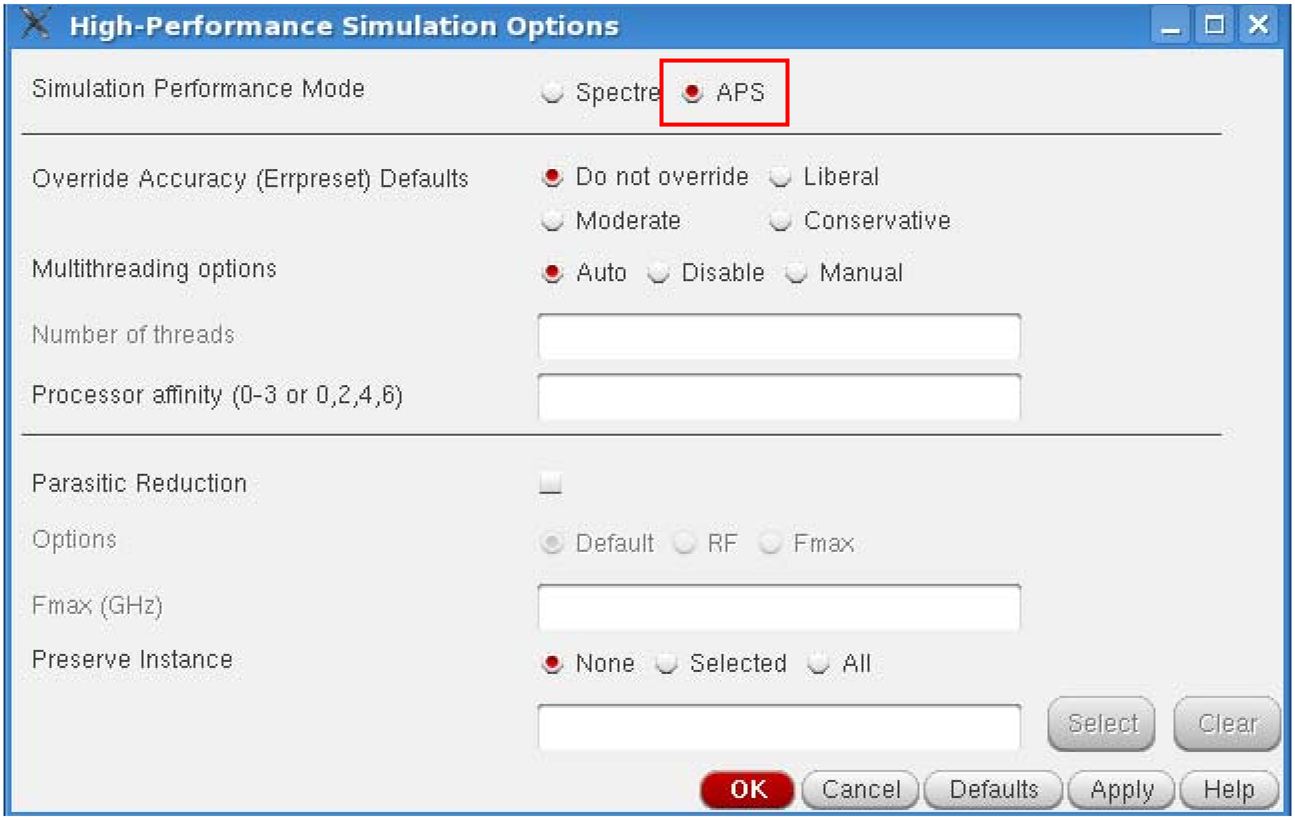


3. Click *OK to close the Choosing Simulator/Directory/Host* form.

4. Set up the High Performance Simulation Options,as follows:

In the ADE window, choose *Setup - High Performance Simulation*. The *High Performance Simulation Options* window is displayed, as shown below.

Figure A-301 High Performance Simulation Options



Select *APS* as the simulation performance mode. Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores.

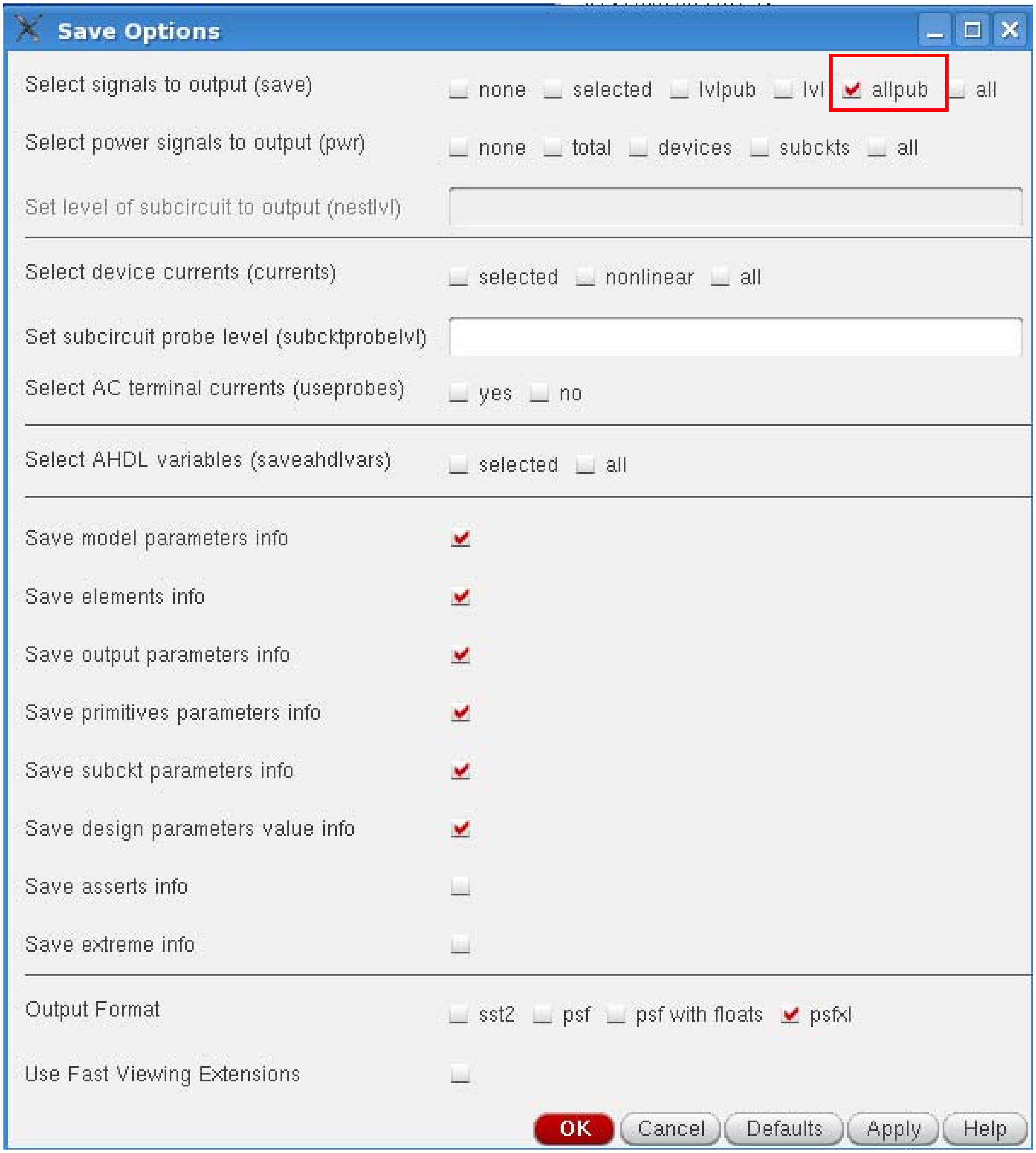
Note: The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading. For more information, see the Virtuoso Spectre User Guide.

a. Click *OK*.

5. In the Virtuoso Analog Design Environment window, choose *Outputs – Save All*.

The *Save Options* form is displayed, as shown below.

Figure A-302 Save Options Form



6. In the *Select signals to output* section, make sure that *allpub* is selected.

This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

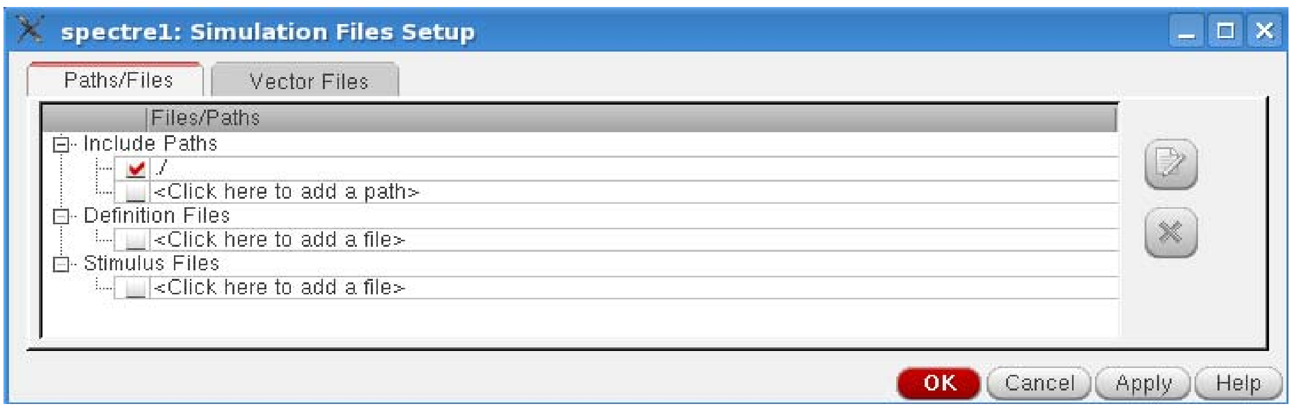
To save the currents, use the *Select device currents (currents)* option, and select `nonlinear` if you just want to save the device currents, or `all` if you want to save all the currents in the circuit.

7. Click *OK*.

Setting Up Model Libraries

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files*. The *Simulation Files Setup* form is displayed, as shown below.

Figure A-303 Simulation Files Setup Form



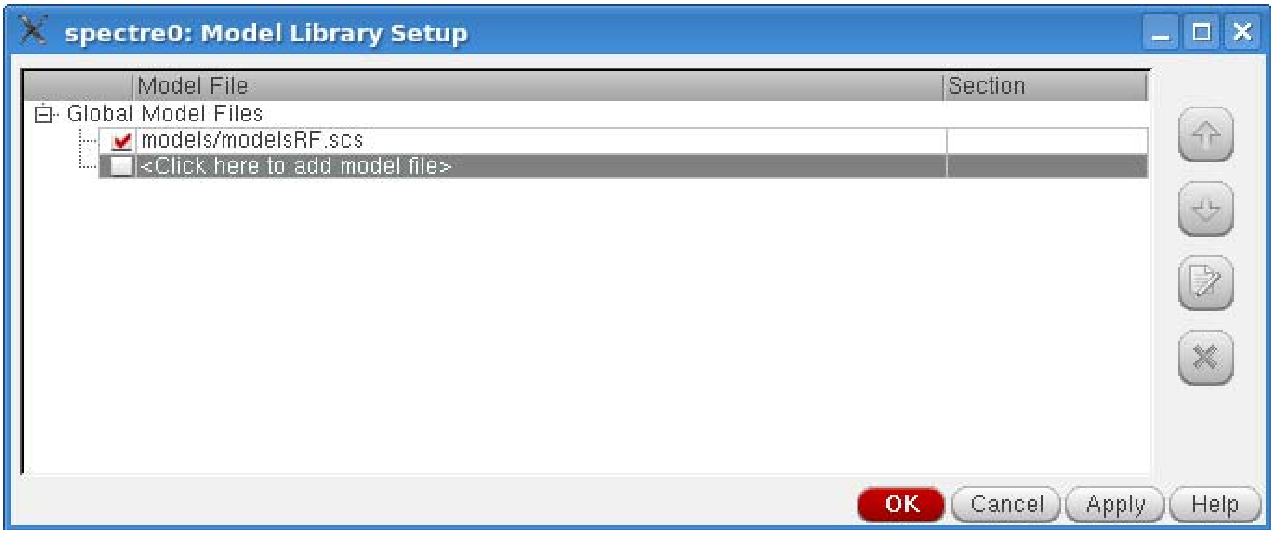
2. Verify that the *Include Path* is set as shown above.
3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

   The *Model Library Setup* form is displayed.
4. In the *Model Library File* field, typethe following for the name of the model file:

   `models/modelsRF.scs`
5. Click *Add*.

   The *Model Library Setup* form looks like the following:

Figure A-304 Model Library Setup



6. Click *OK*.

Mixer Conversion Gain, RF to IF Isolation, LO to IF Leakage, and Noise Figure

You can measure the conversion gain using hb (Harmonic Balance) by applying the signal which causes the frequency conversion (the LO) in a hb analysis and follow this with hbac (Harmonic Balance AC) to measure the small-signal conversion gain. This will also be used to measure RF to IF isolation. Because hbac is based on the hb result with all the harmonics that hb solved for, the mixing products produced by mixing the input frequency with any or all of those harmonics can be calculated with hbac. In addition, hbnoise (Harmonic Balance noise) will be used to calculate the output noise and noise figure with all the frequency translations inherent in the mixer.

Hb solves for the steady-state solution produced by the LO and captures the nonlinearity of the mixer. Hb calculates the nonlinearity in the frequency domain as a series of harmonics.

The hbac analysis calculates the conversion gain based on the nonlinearity created by the LO. Hbac is also used to calculate the RF to IF isolation. This measurement is not a simple AC frequency response. It is the frequency response from RF to IF with the LO applied to the mixer.

Hbnoise also uses the nonlinearity from the hb large-signal analysis in order to calculate how noise is folded to the desired output frequency. Both of these analyses are quite fast and accurate.

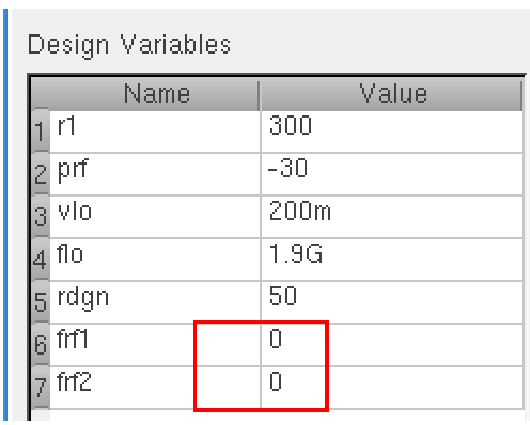
Hbac and hbnoise are both small-signal analyses. Both are nonlinear in the sense of being able to take into account the frequency conversions created by the LO. Noise is almost certainly a small-signal problem. In practice, hbac calculates the correct conversion gain for the cases where the RF input causes little compression (that is, the input is 10dB or more below the 1dB compression point.)

Setting Up the Simulation - Setting Design Variables

To set the design variables to the values required for each simulation, perform the following steps:

1. In the Design Variables section of the ADE L window, change the design variables *frf1* and *frf2* to *0*. *flo* should already be set to *1.9G*. To edit the values, simply click on the value to the right of the variable name, and type in a value. Then, press *Enter*. Setting the input frequencies to 0 disables the production of waveforms for the large-signal analyses like tran, pss, and hb (harmonic balance).

Figure A-305 Design Variables Section of ADE-L Window



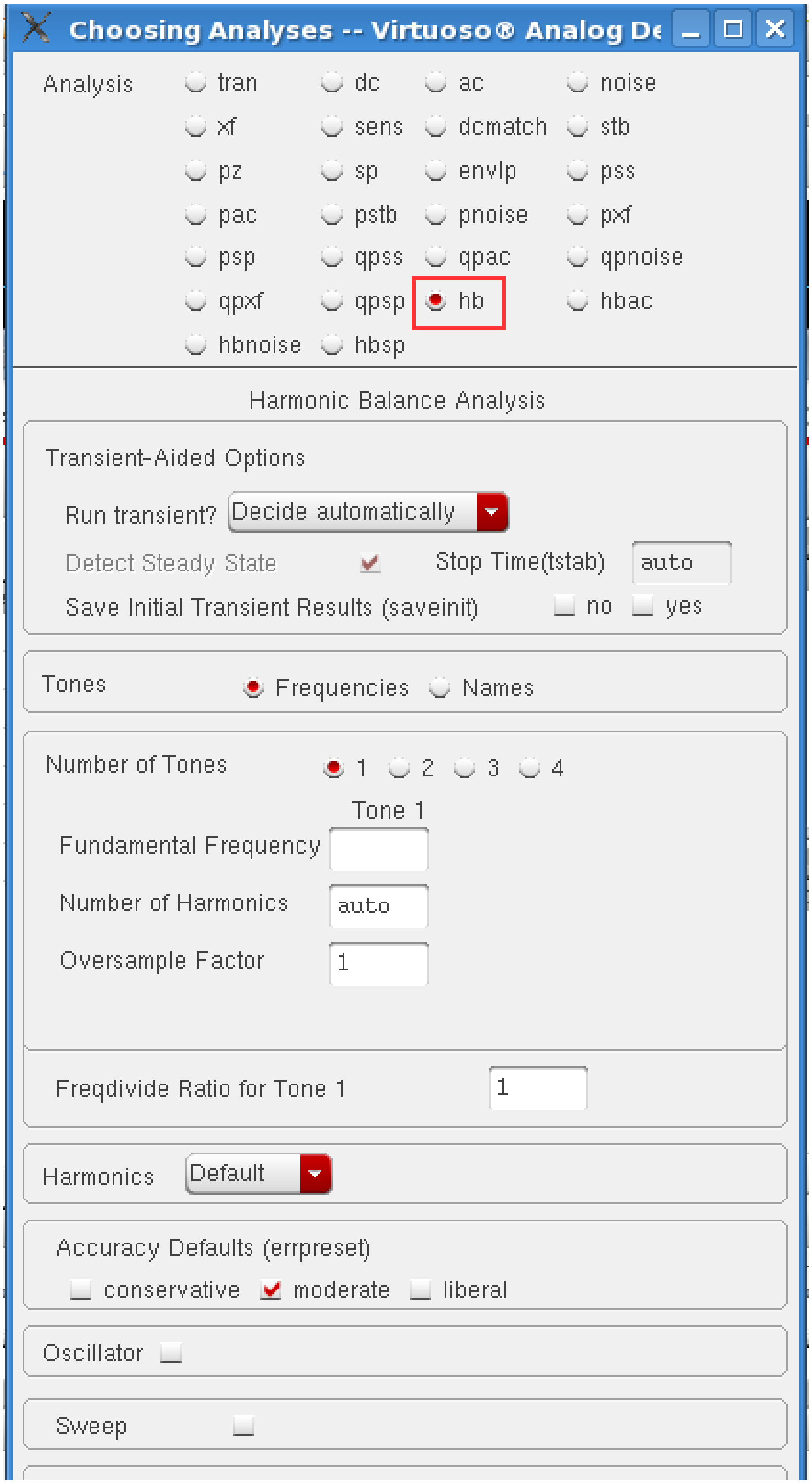
Setting Up the Harmonic Balance Analysis

1. Choose *Analyses – Choose* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Analyses* form is displayed.

2. In the *Choosing Analyses* form, select *hb*. The form expands, as shown below.

Figure A-306 The hb Choosing Analyses Form



3. In the *Transient-Aided Options section*, leave the settings at their default value:

a. Leave *Run Transient?* to the default value of *Decide Automatically*.

Run transient will run the LO signal using the transient (In SpectreRF, this is called the tstab interval) for a short period of time. At the end of tstab, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.*Stop Time (tstab) auto*.

When *auto* is selected for *Stop Time (tstab)*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the starting point in the hb iterations.

b. When *Run transient* is set to *Decide automatically*, The *Detect Steady State* option is selected automatically. When this is set, when steady-state is detected in the tstab interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

c. Leave *Save Initial Transient Results (saveinit) as* blank.

4. In the *Tones* section, ensure that *Frequencies* is selected. This is the default.

Harmonic balance can now set the harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. A transient analysis runs until steady-state is detected, and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has tstab enabled (when *Names* is selected). If you want to manually set transient-aided hb, select *Yes* from the *Run Transient?* drop-down list and set a time for the transient in the *Stop Time (tstab)* field. In this mode, the stop time of the transient analysis in the tstab interval cannot be automatically extended.

If you want to see the startup waveform, select *yes* for *Save Initial Transient Results (saveinit)*.

5. In the *Number of Tones* section, note that the number of tones is set to 1. (This is the default in hb). Since the RF tones were disabled, only the LO tone remains.

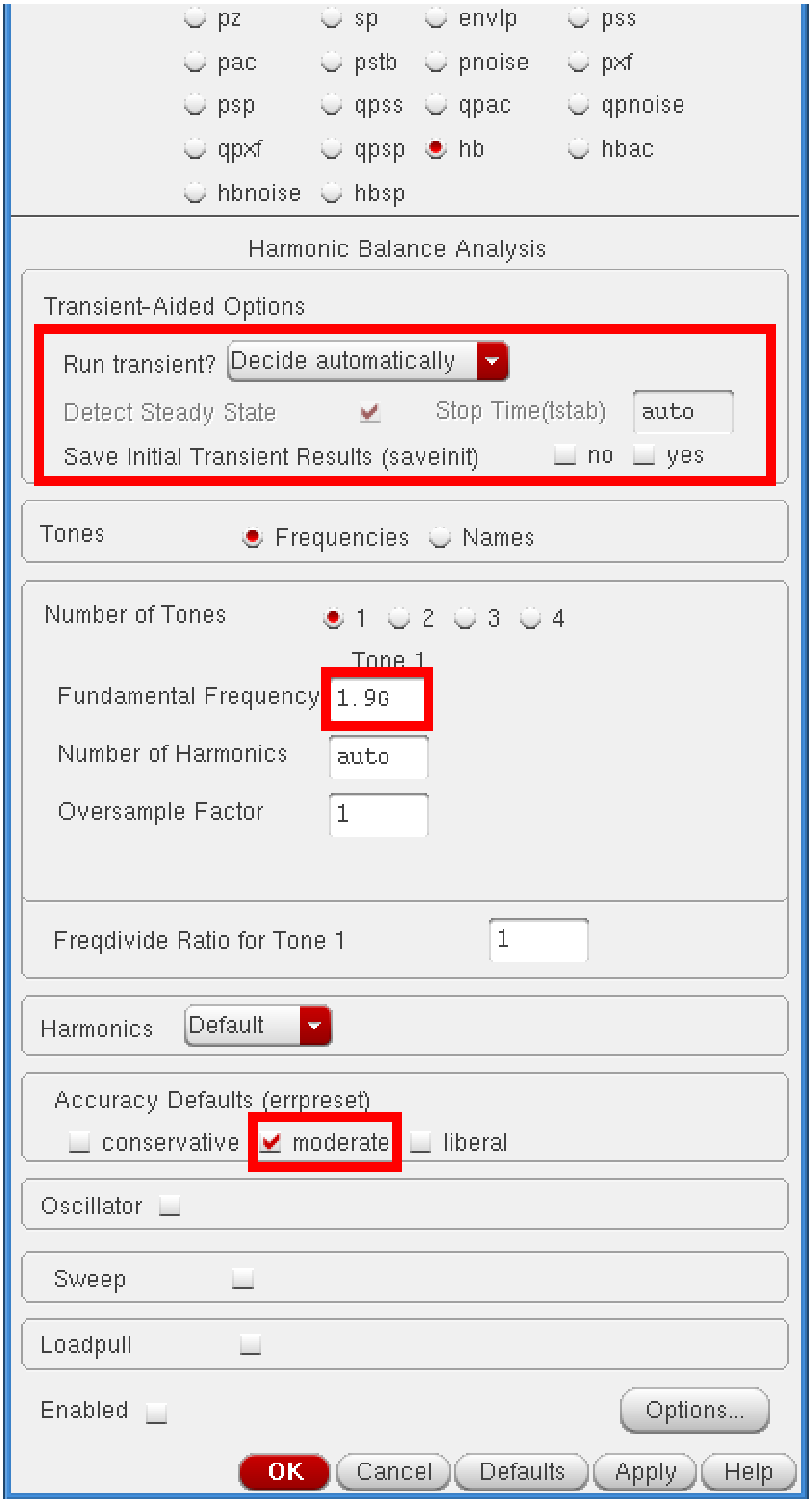
6. Enter 1.9G as the *Fundamental Frequency.*

7. Leave *Number of Harmonics* set to *auto*. This is the default. Spectre will choose the appropriate number of harmonics for you. *auto* is allowed if *Decide automatically* or *Yes* are selected from the *Run transient?* drop-down list.

When you choose *auto* for the number of harmonics, leave *Oversample Factor* set to the default value of 1. When all the signals in the system (including currents) are nearly sinusoidal, then, *Oversample Factor* should also be set to 1. Set *Accuracy Defaults (errpreset)* to *moderate*. Exceptional accuracy is not needed because only the high amplitude LO signal needs to be solved for, so *moderate* (the default) is selected.

8. Leave the rest of the form set to the default values. The hb *Choosing Analyses* form should look like the following figure:

Figure A-307 hb Choosing Analyses Form



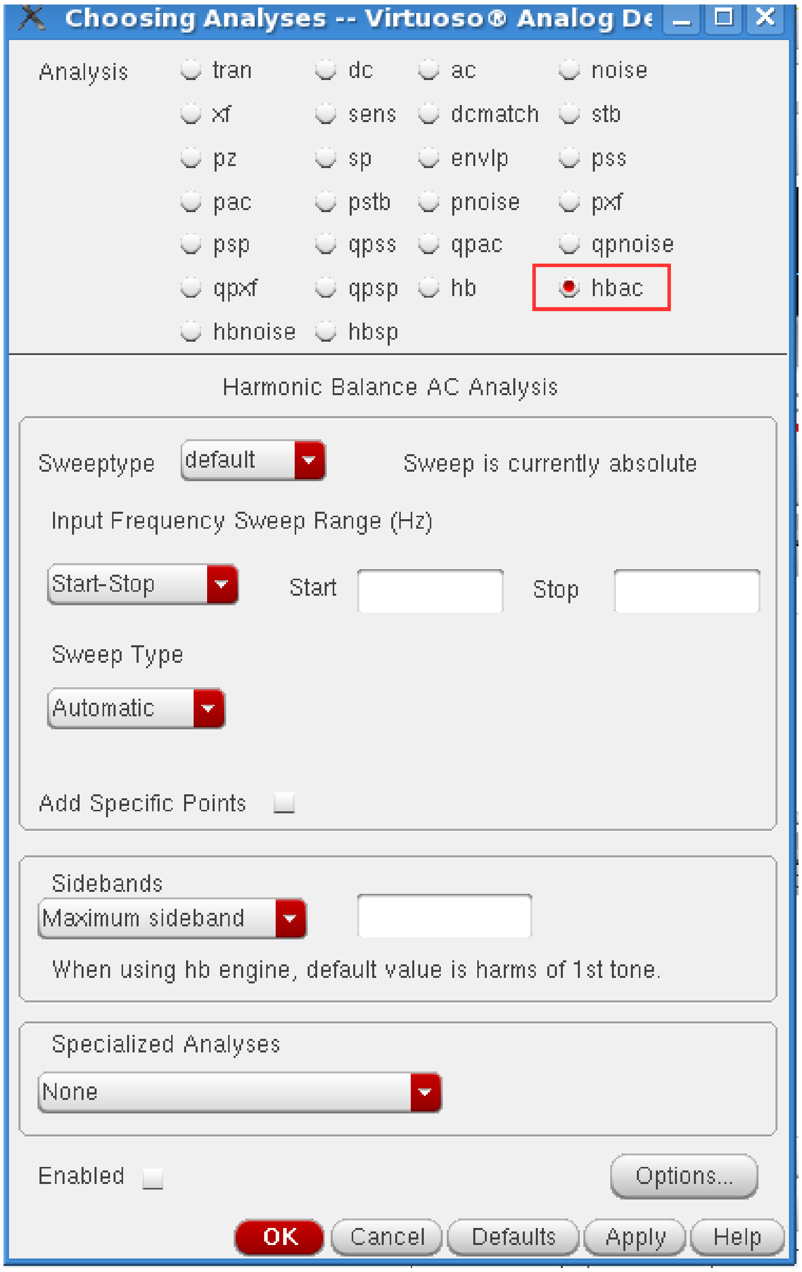
For more information on setting up the *Choosing Analyses* form, see Chapter 3, “Frequency Domain Analyses: Harmonic Balance,”

9. Click *Apply*.

Set up the HBAC Choosing Analysis form.

1. In the Choosing Analyses form, select *hbac*. The form changes, as shown below.

Figure A-308 hbac Choosing Analysis Form



2. Leave the *Sweeptype* set to the default value (absolute).
3. Set the *Input Frequency Sweep Range*.

   Absolute takes the frequency range as specified with no frequency translation. Relative is also available where the input frequency can be shifted up or down in multiples of the PSS frequency. This is useful for having log sweeps above or below a PSS harmonic.

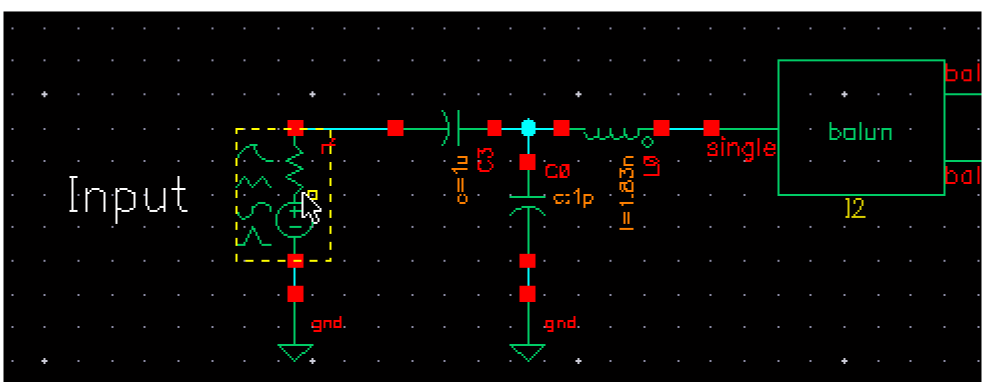
The frequency sweep range is always the input frequency range in hbac. Because hbac has the ability to calculate the outputs at different frequencies based on the nonlinearity caused by the LO, we choose which output frequencies to calculate in the Sidebands area of the Choosing Analyses form. Sideband is the name of the different output signals that are produced when mixing the input with harmonics of the LO signal.

4. Type 1.9001G in the *Start* field.
5. Type 2G in the *Stop* field.
6. Set the *Sweep Type* to *Linear*.
7. Select *Number of Steps* and type `10` just to the right of *Number of Steps*.
8. The amplitude for the hbac analysis is set in the input source (port *rf* with *PAC magnitude* `=1` in this case) in the schematic. 1 is convenient because it is 0 dBV. This allows direct conversion gain measurement by using the dB20 function when the output signal is plotted.

   Below are the steps showing how to set the *PAC Magnitude* in the schematic

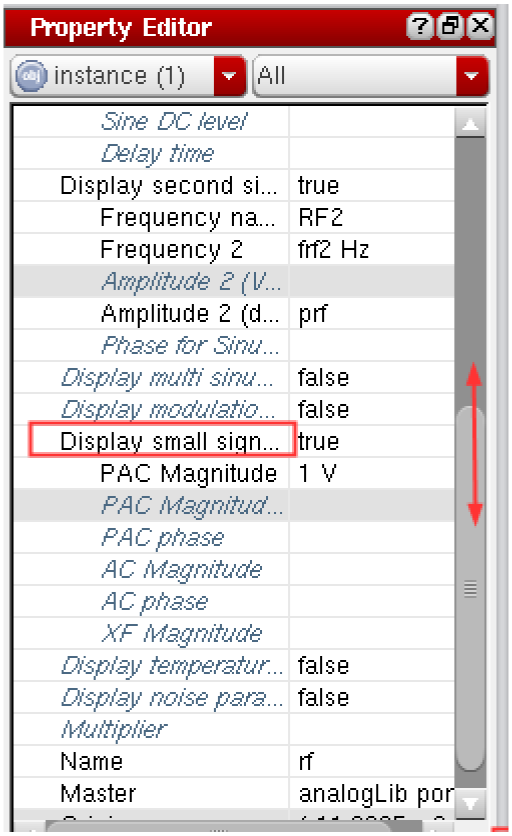
   a. To set the amplitude for hbac, select the *Input* port in the schematic.

Figure A-309 Input Port on Schematic



   b. After you click on the port, refer to the *Property Editor* on the left side of the schematic. You can scroll through the *Property Editor* by moving the scroll bar on the right side of the *Property Editor*.

Figure A-310 Property Editor



c. Select the *Display Small Signal Params* option near the bottom of the *Property Editor*,and set its value to *true*. This expands the *Property Editor* form.

   The value for *PAC Magnitude* should be set to 1 V. If it is not, set it to 1 V.

d. From the Schematic, choose *Check - Current Cellview*. You do not have to save the design to simulate it. This allows “what if” analysis. If you like the results, you can then save the design. If you do not like the results, you donot have to save it.

   In the hbac *Choosing Analyses* form, you will be selecting the direct conversion IF sideband and the output with no frequency translation from the *Sidebands* section under *Select from Range.*

9. In the *Sidebands* section, choose *Select from range*.

   Sidebands define the output frequencies to be calculated. In this setup, the direct conversion IF (1.9001G~2G-1*1.9G=100K~100M) is selected. In addition, hbac also calculates the outputs with no frequency translation (1.9001G~2G-0*1.9G=1.9001G~2G). You will be choosing these two sidebands.

The levels that are produced at each node at all the selected frequencies are also calculated by the hbac analysis.

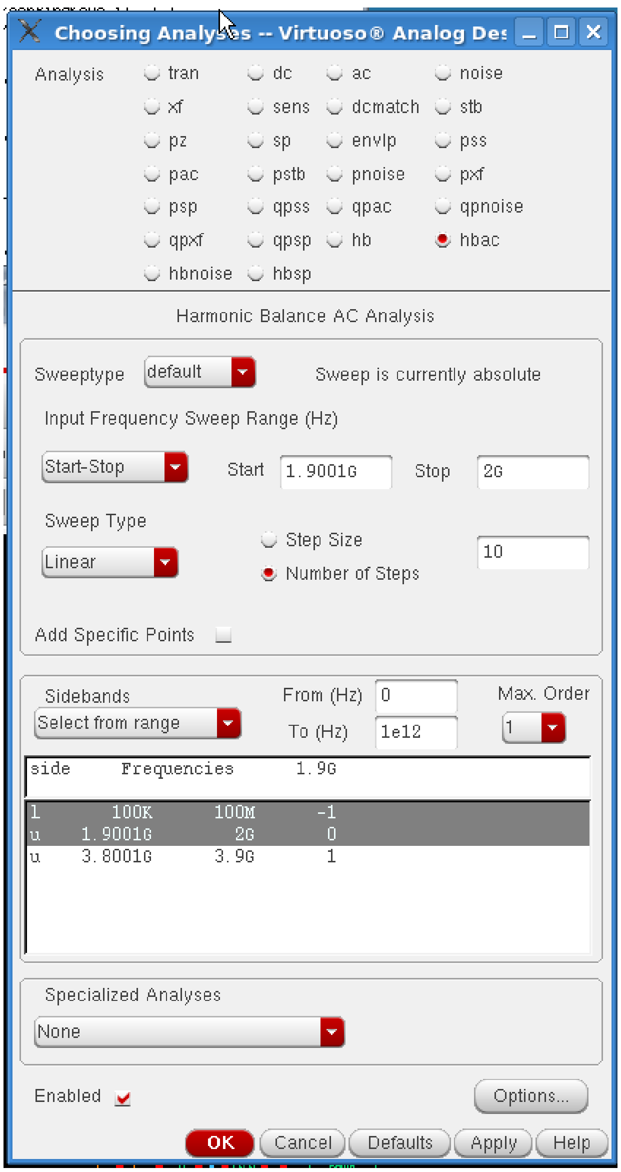
a. Highlight the *l 100K 100M -1* entry. This is the direct conversion IF sideband.

b. To select another sidebands, press and hold the *Ctrl* key, and select the *u 1.9001G 2G 0* entry. This is the output with no frequency translation.

10. Click *Apply*.

The hbac *Choosing Analyses* form looks like the figure below:

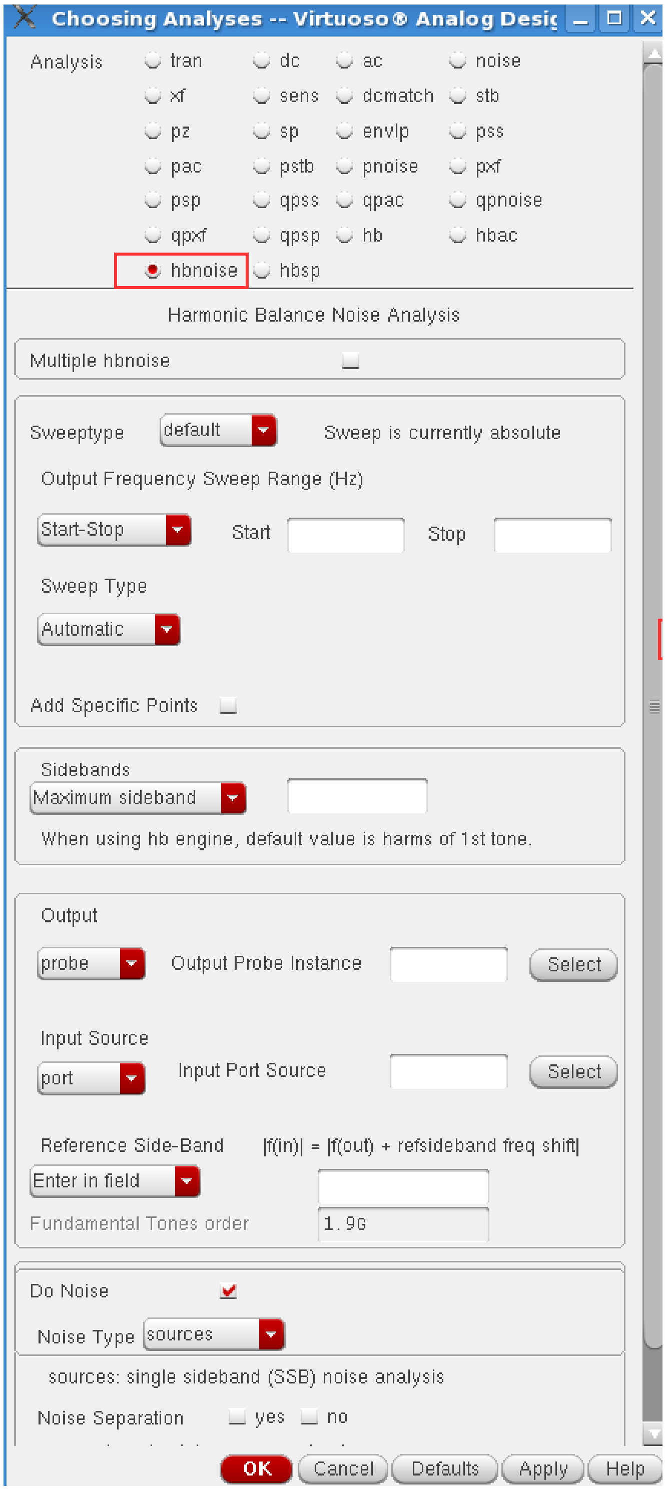
Figure A-311 hbac Choosing Analyses Form -Set up the hbnoise Choosing Analyses form



Set up the hbac *Choosing Analyses* form, as follows:

1. In the *Choosing Analyses* form, select *hbnoise.* The form changes, as shown below.

Figure A-312 hbnoise Choosing Analyses Form



2. Leave the *Sweeptype* set to default (*absolute*).

3. Set the *Output Frequency Sweep Range*.

   The *Frequency Sweep Range* fields are always the output frequency range in hbnoise. *Maximum sideband* specifies the highest harmonic of the LO we want to calculate the down converted noise from. In this case, noise through the 10th harmonic of the LO will be calculated.

4. Type *1K* in the *Start* field.

5. Type *100M* in the *Stop* field.

6. Select *Logarithmic* sweep.

7. Type *3* in the *Points Per Decade* field.

8. Leave the *Maximum sideband* field blank.

   Note: When left blank, the form uses the *number of harmonics* from the hb analysis.

9. In the *Output* section of the form, by default, the *Output* is set to *probe*. Click *Select* to the right of the *Output Probe Instance* field and select the source to the right of the label *Output* in the schematic window.

   When a port or a resistor is selected in this way, the noise of this component is excluded for the noise figure calculation. A port is a voltage source in series with a resistor as a single component and it is located in analogLib.

10. In the *Output* section, by default, *Input Source* is set to type *port*. If you want a noise figure calculation, you must select a port as the input. Click *Select* to the right of the *Input Port Source* field and select the source to the right of the label *Input* in the schematic window.

    In hbnoise, only output-referred and input-referred noise measurements are made, so both the input and output frequencies are declared in the Choosing Analyses form. In linear noise, the input-referred noise is the output-referred noise divided by the transfer function from input to output.

    In hbnoise, because we have the ability to calculate frequency conversion, the design input frequency range is chosen in the reference sideband section in order to get the correct input-referred noise and noise figure at the design input frequency. This can also be referred to as the passband frequency.

    In the *select from range* field, you will specify the input frequency range that goes with the output frequency range specified at the top of the form.

11. In the *Reference side-band label* section, choose *Select from list* from the drop-down list. .

12. Select the *u 1.900001G 2G 1* line.

Here is an example of how the reference sideband field is calculated:

suppose the reference sideband=k,

`|fin|=|fout|+k*fundamental frequency`.

Here, `fout=1K~100M` and the fundamental frequency is 1.9G.

`fin` should be `(1K~100M)+1.9G`,

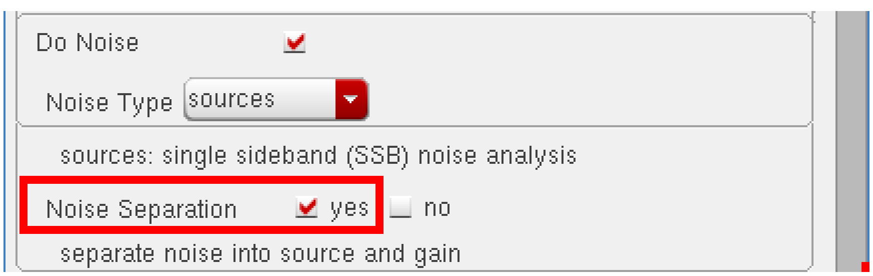
thus the reference sideband `k` is `1`.

13. Select the *Do Noise* check box (this is the default). Leave *Noise Type* set to *sources*.

Sources is a Single Side Band (Upper or Lower depending on the sweep frequency) noise. The results are time averaged over the period of the periodic steady state.

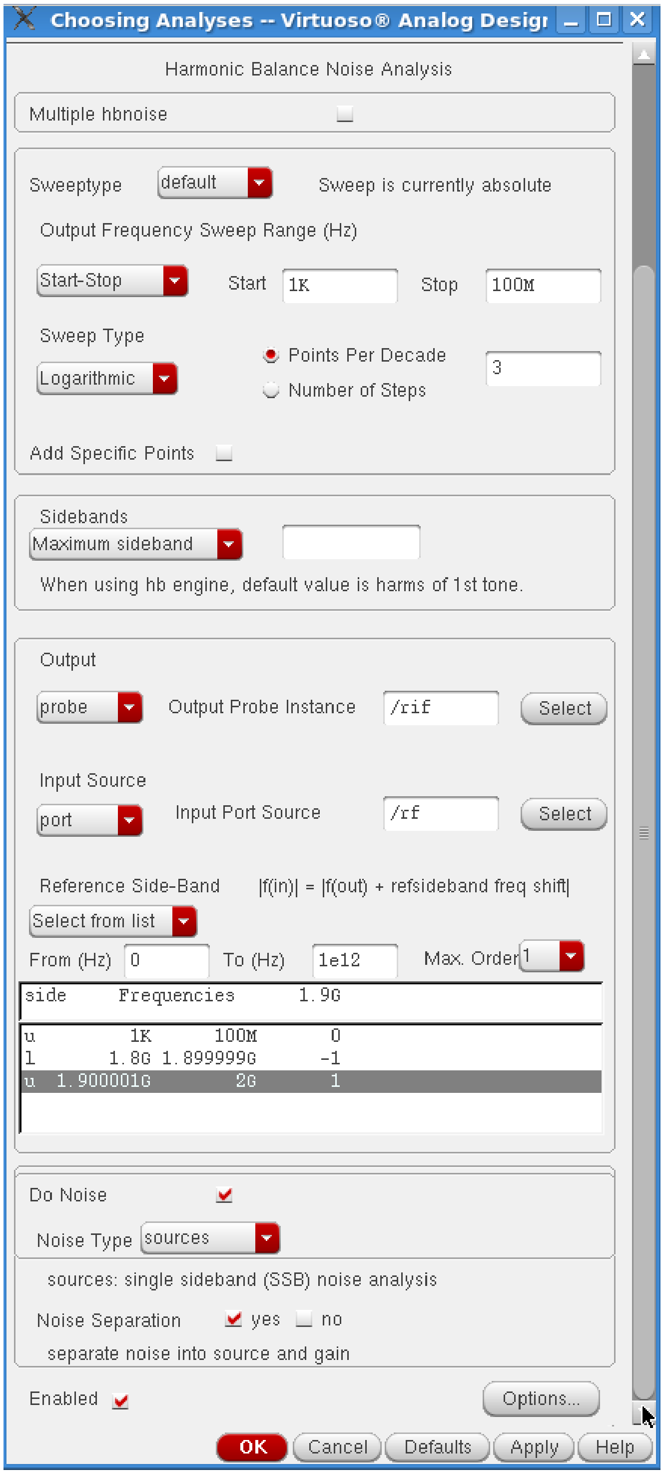
14. Select the *Noise Separation* check box.In addition to the total output noise, the individual noise contributions can be plotted when noise separation is selected. For more information on Noise Separation, see Chapter 3, “Frequency Domain Analyses: Harmonic Balance,”.

Figure A-313 hbnoise Noise Separation Check Box.



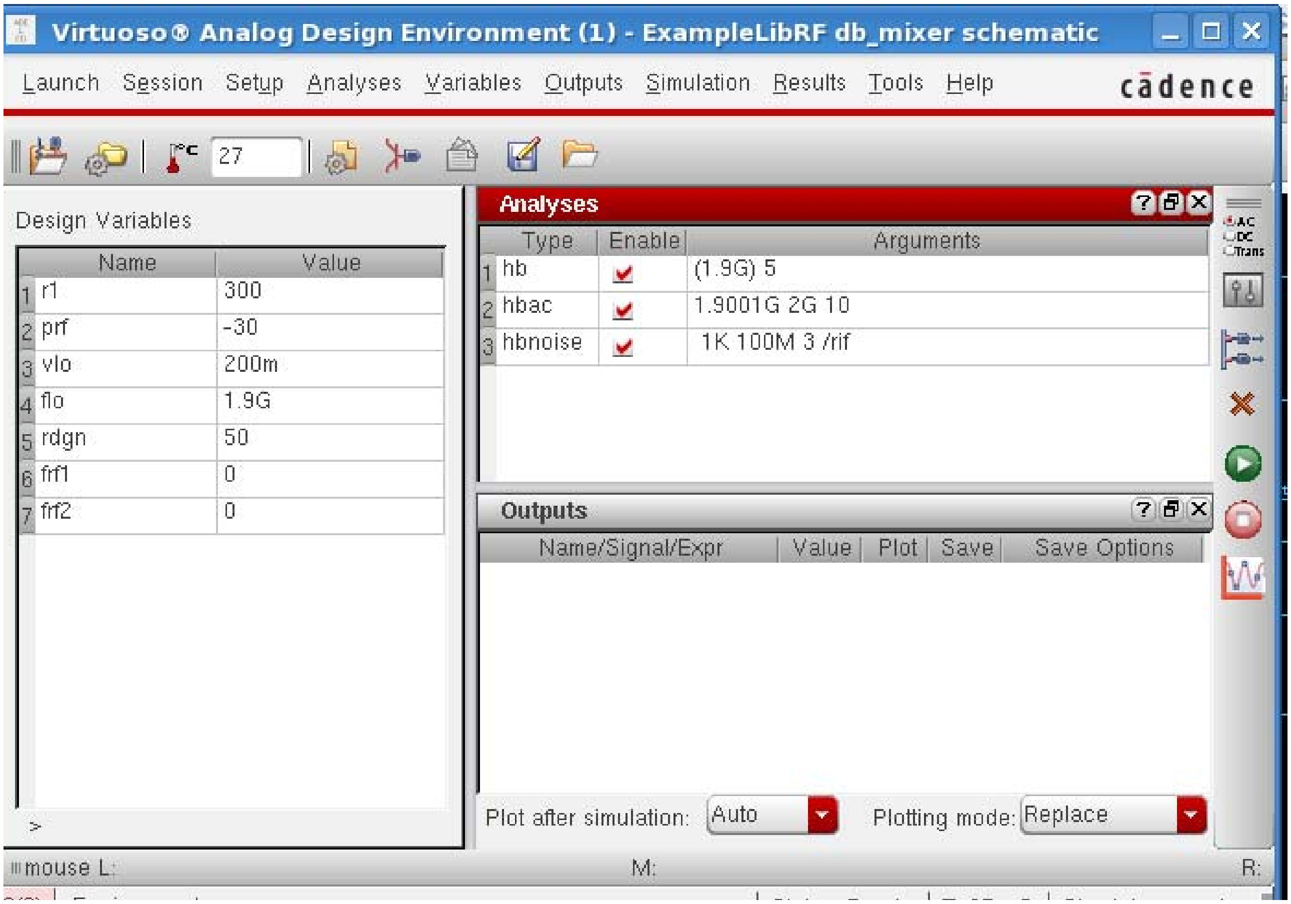
Your completed hbnoise *Choosing Analyses* form should look like the following figure.

Figure A-314 hbnoise Choosing Analyses Form



15. Click *OK* at the bottom of the form.

Figure A-315 ADE Simulation Window



Running the Simulation and Plotting the Results

Start the analyses by clicking the green arrow icon.  in ADE or in the Schematic Editor.

This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

Next, you will plot the following:

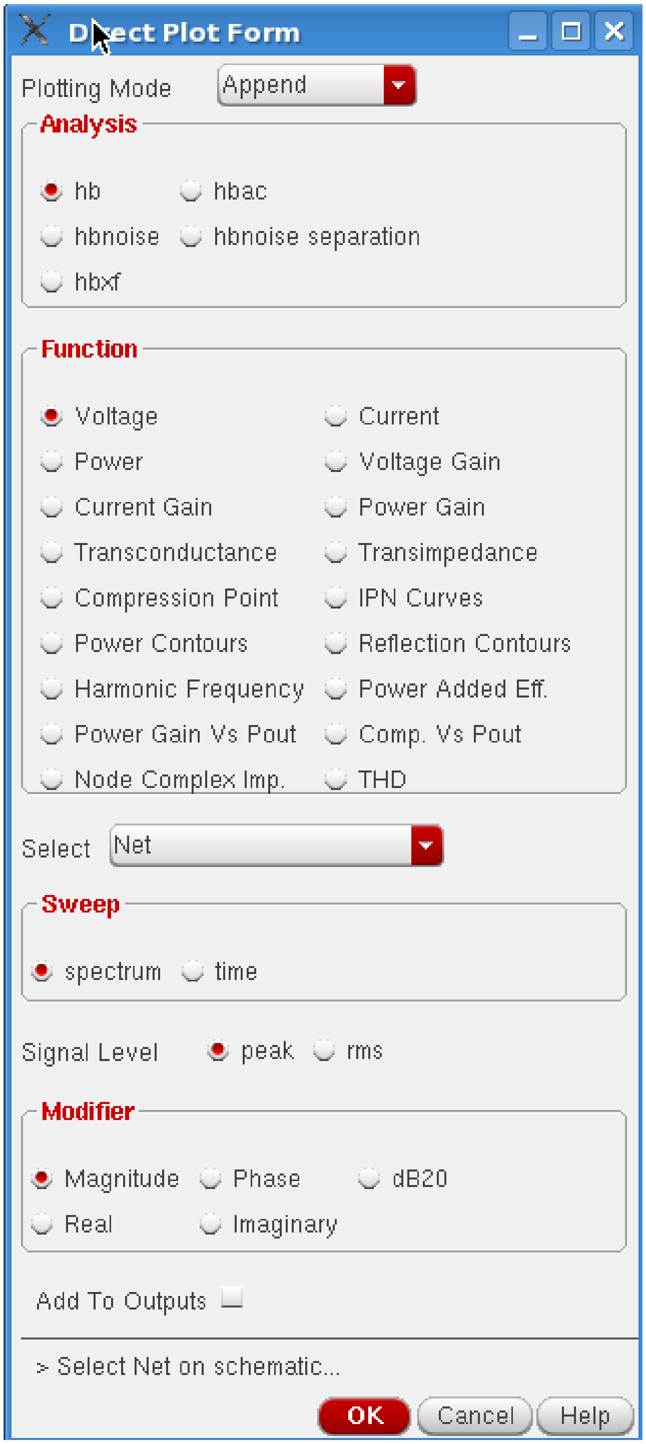
- Mixer Conversion Gain,
- RF to IF Isolation, LO to IF Leakage
- Noise Figure

Measuring Mixer LO to IF Leakage

1. In the Analog Design Environment window, choose *Results - Direct Plot -Main Form*.

   The *Direct Plot Form* is displayed.

Figure A-316 hb Direct Plot Form



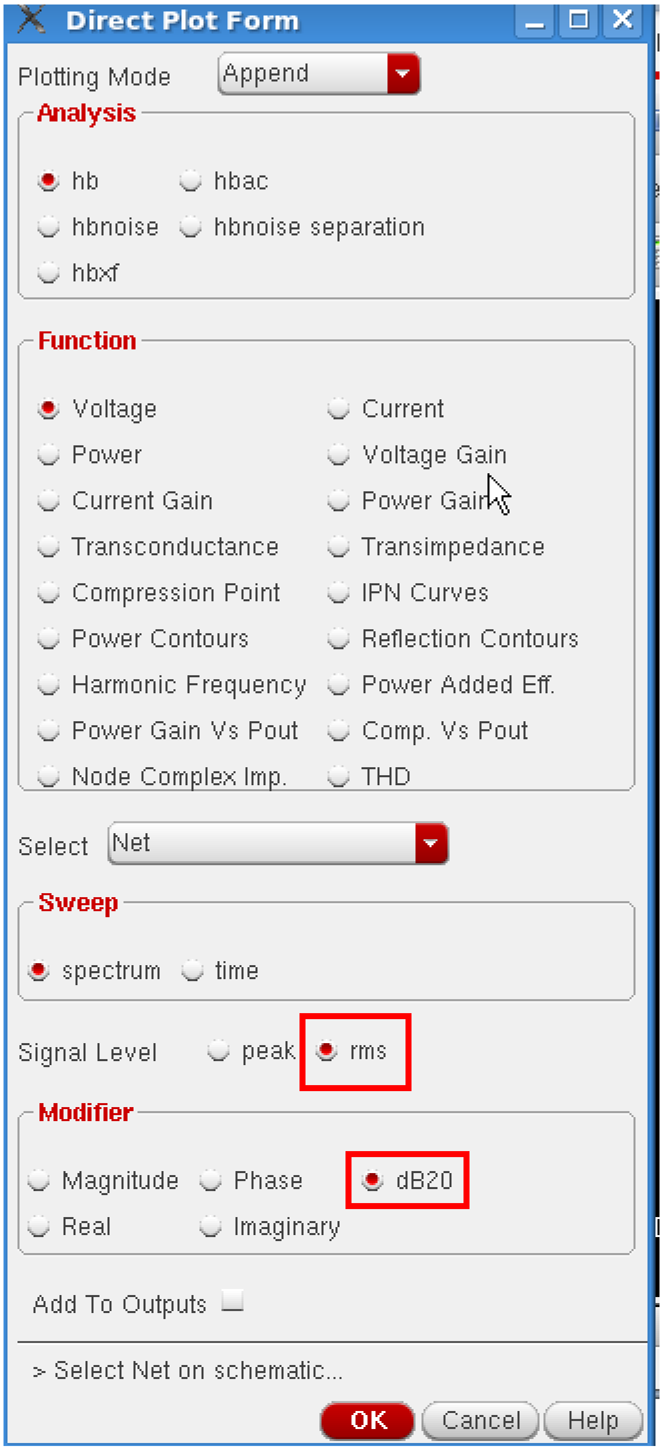
Note that even though you only selected hb, hbac, and hbnoise analyses, you also have an hbxf analysis listed in the *Direct Plot Form*. Hbnoise automatically runs an hbxf (harmonic balance transfer function) simulation as part of running a hbnoise simulation. In addition, there is an entry for hb noise separation.

First, plot the output spectrum

1. In the *Analysis* section, select *hb* .
2. In the *Function s*ection, select *Voltage*.
3. *Select* `Net` in the center of the form. (This is the default. You can also select *differential nets*).
4. In the *Sweep* section, select *spectrum* (this is the default).
5. In the *Signal Level* section*,* select *rms* (the default is *peak*).
6. In the *Modifier* section, select *dB20* .

   The *Direct Plot Form* should look like the following:

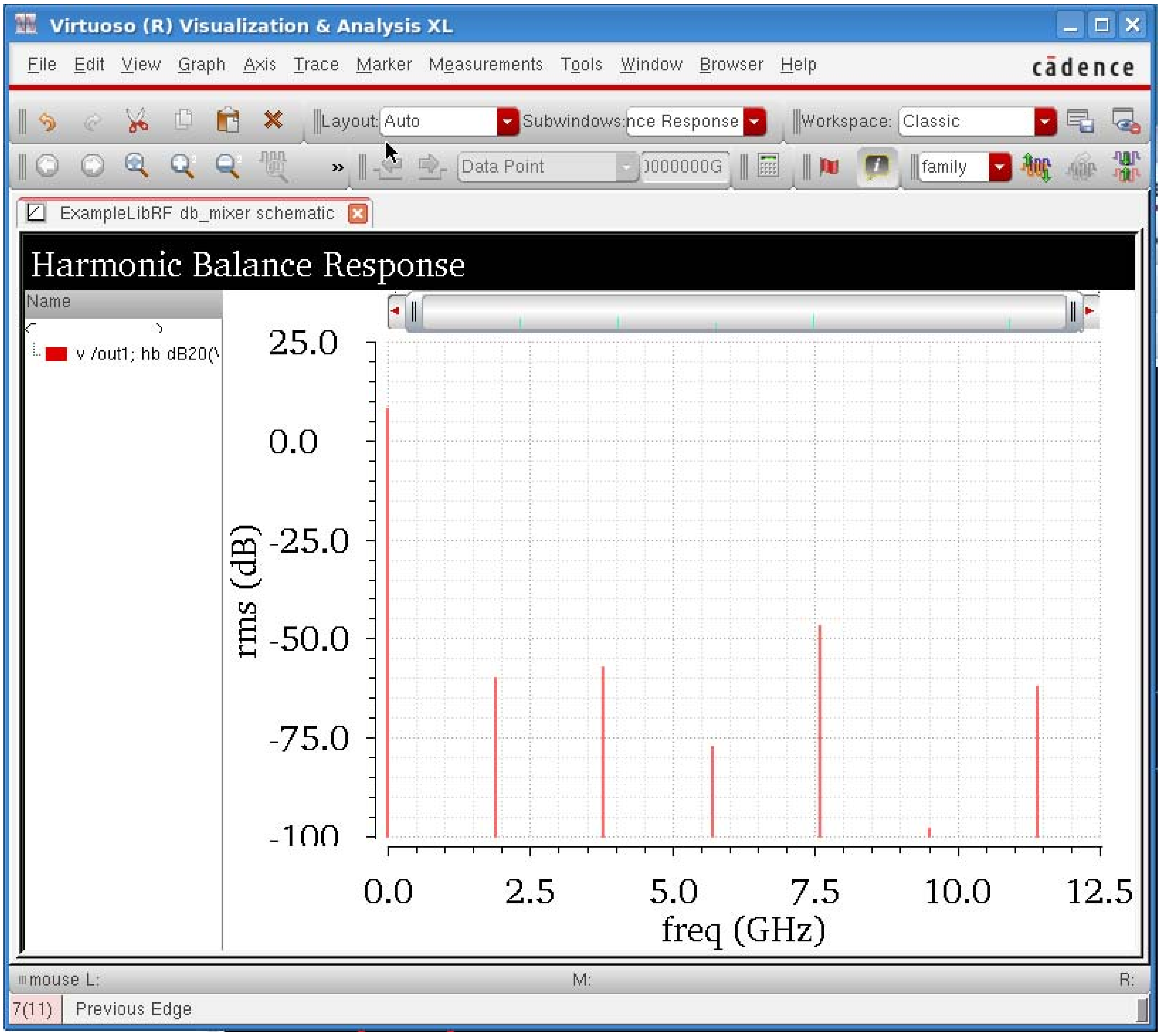
Figure A-317 Completed Direct Plot Form - Measuring Mixer LO Leakage



7. Select the *out1* net in the schematic. It is located just below the *Output* label.

The waveform window is displayed.

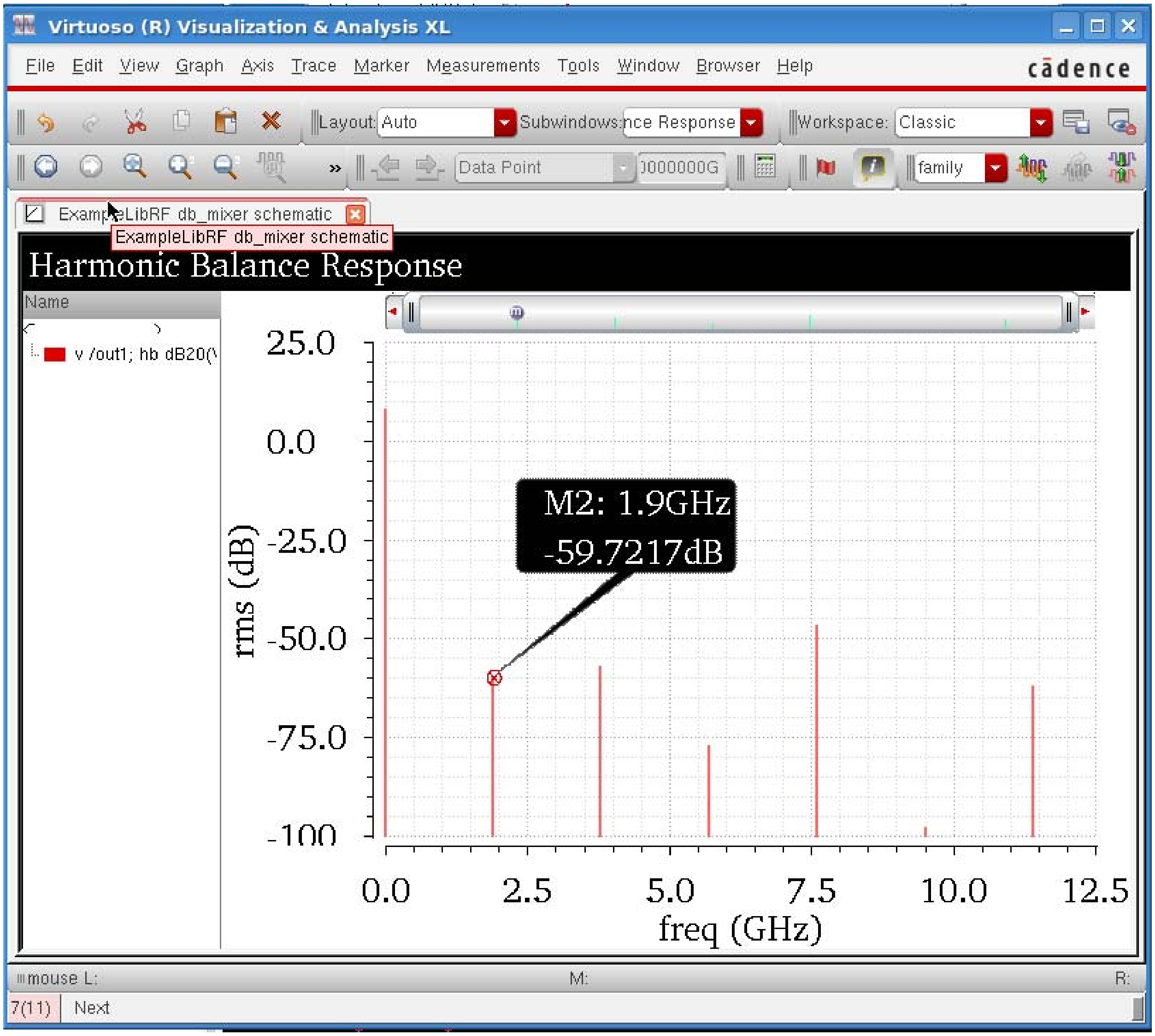
Figure A-318 Harmonic Balance Response of /out1 net



Measure LO leakage on the IF Output

8. Move the mouse cursor to the tip of the spike at *1.9GHz* and type *m*. A marker appears on the plot. Click and hold on the marker readout and move it to a place where you can see all the harmonics, then release the mouse button. The LO amplitude is about *-59.72 dB*.

Figure A-319 Measuring LO leakage on the IF Output



Plot the conversion gain and RF to IF isolation

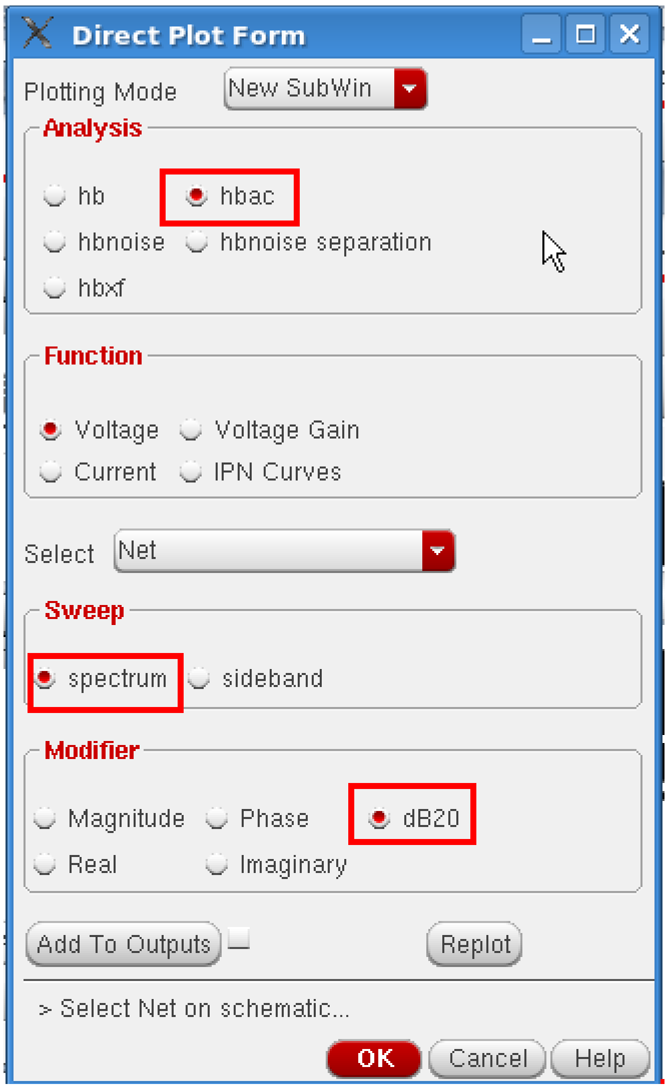
In the *Direct Plot Form*:

1. Change *Plotting Mode* to *New SubWin*.
2. In the *Analysis* section, select *hbac*.
3. Inthe *Function* section, select *Voltage* (this is the default).
4. In the *Sweep* section, select *spectrum* (this is the default).

5. In the *Modifier* section, select *dB20*.

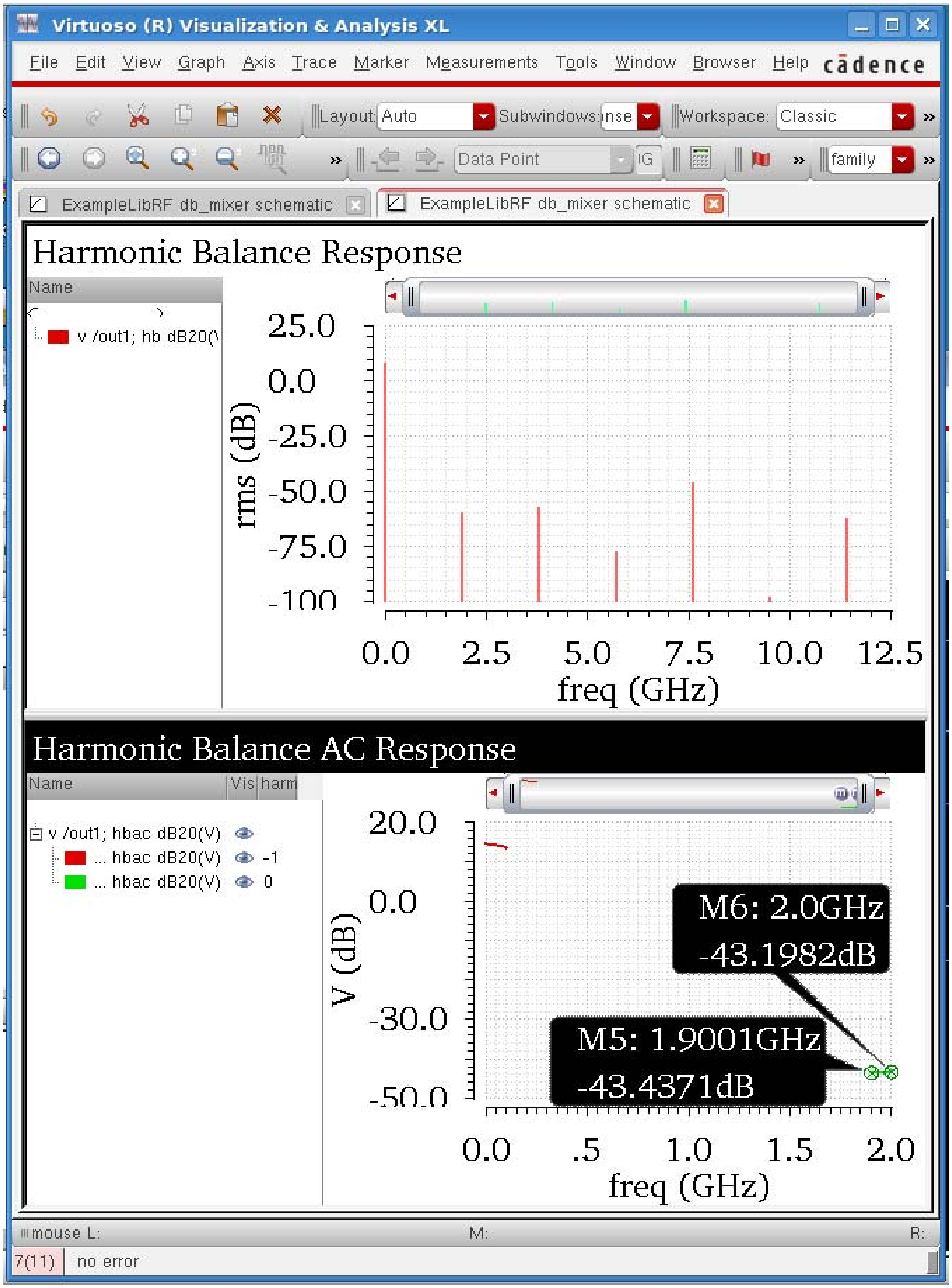
The *Direct Plot Form* looks like this.

Figure A-320 Direct Plot Form - Plotting Conversion Gain and RF to IF isolation



6. Click *Replot*. The waveform window updates, as shown below.

Figure A-321 Conversion Gain and RF to IF Leakage



Look at the bottom subwindow for the HBAC results (Note that in the software, the default window position will be displayed to the right). The mixer circuit is deliberately very slightly unbalanced. In the HBAC *Choosing Analyses* form, you defined an input frequency range to be swept. HBAC calculates the amplitudes of the output signals at the frequencies you chose in the sidebands section of the *Choosing Analyses* form.

7. Note that the trace near 2GHz is about *-43.4dB*, and measures the gain from input to output with no frequency conversion. This is due to RF to IF leakage.

If you want, place a marker at that location by moving the mouse cursor to the tip of the spike at 2GHz and typing *m*. Click and hold on the marker readout and move it to a place where you can see all the sidebands, then release the mouse button.

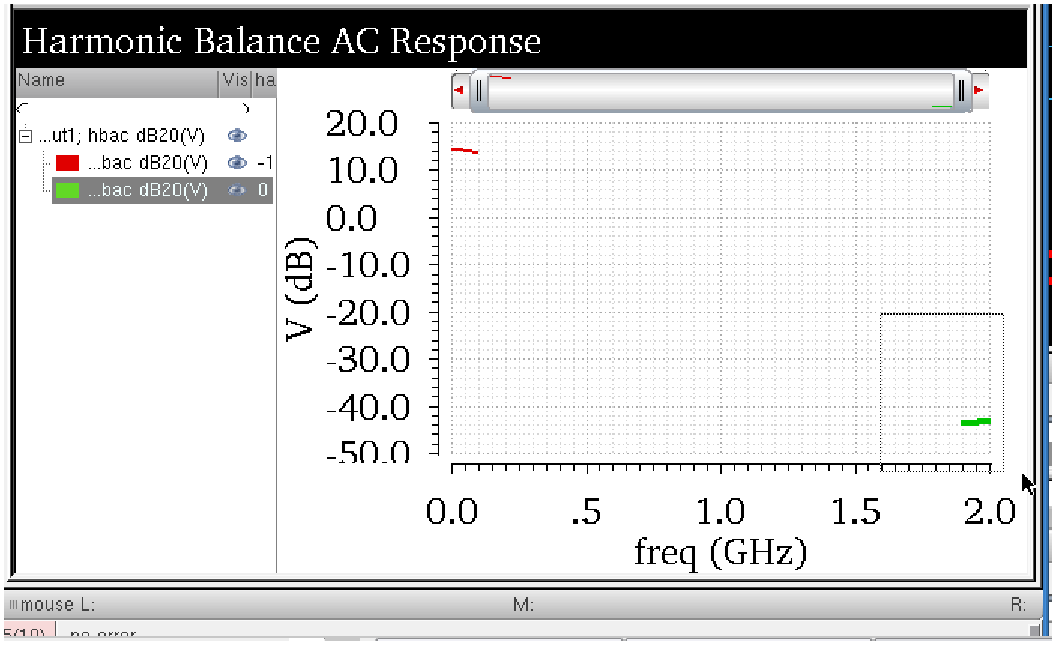
Next, move the cursor to 1.9001GHz and just below -40dB, read the value of the cursor marker. In this case, the RF to IF isolation is about *43.4 dB*. If you want, place a marker at that location as well.

Measure Voltage Gain vs Output Frequency

1. Optional: Remove the markers from the Waveform window by choosing *Marker - Delete All* or by pressing the *Ctrl+E* bindKey.

You may need to zoom in close to the waveform. To do that, click and hold the right mouse button and drag a square outline around the waveform, as shown in the next figure.

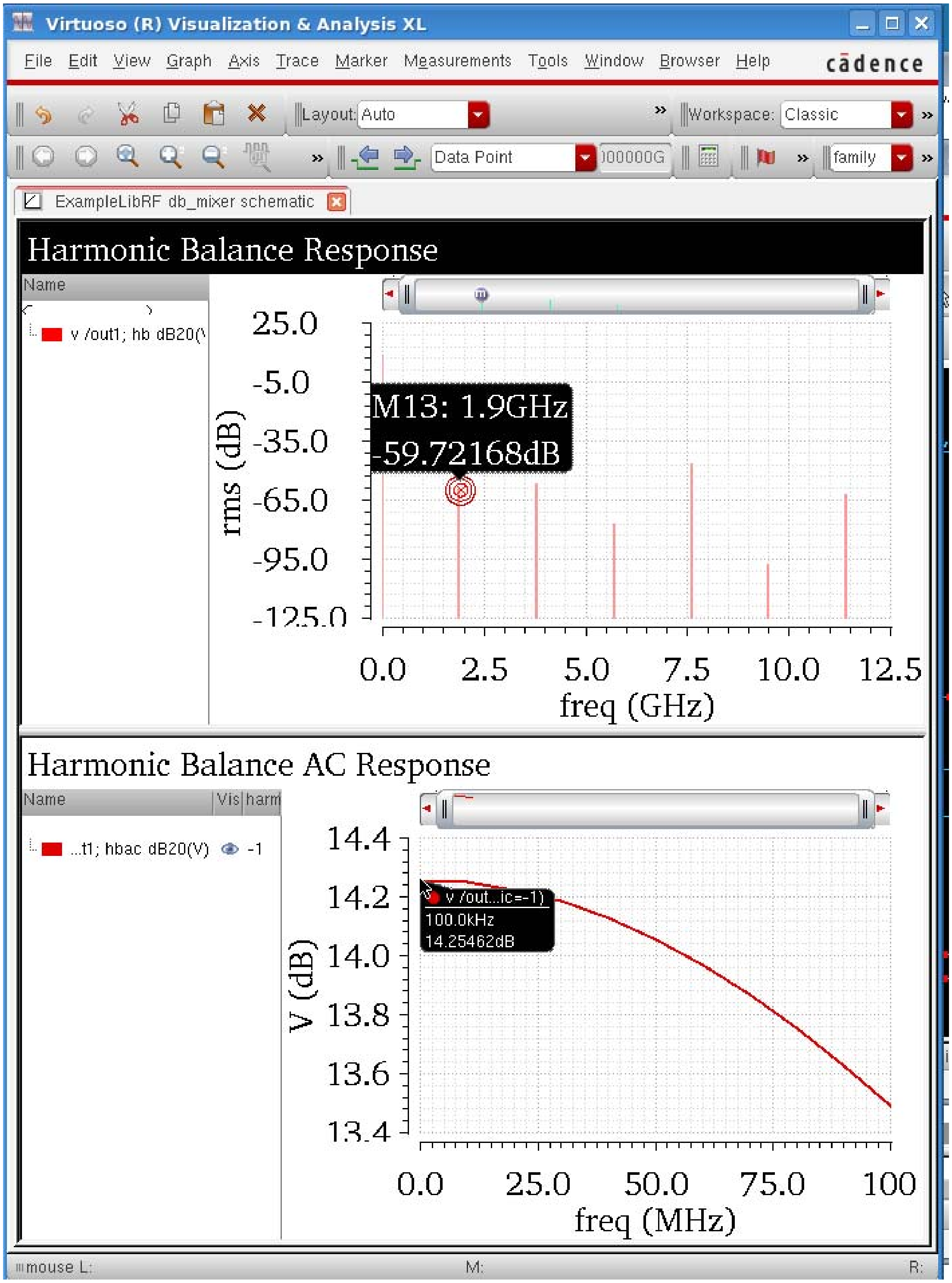
Figure A-322 : Choosing Sideband of Interest



2. Delete the result near 2GHz. To do this, in the waveform window, click the trace near 2GHz and just below -40 dB. Press the *Delete* key. To see the remaining trace, you may need use the bindkey *f* to fit (resize) the window so that the entire plot is shown.

   The waveform window redraws, as shown below.

Figure A-323 Voltage Conversion Gain vs. Output Frequency



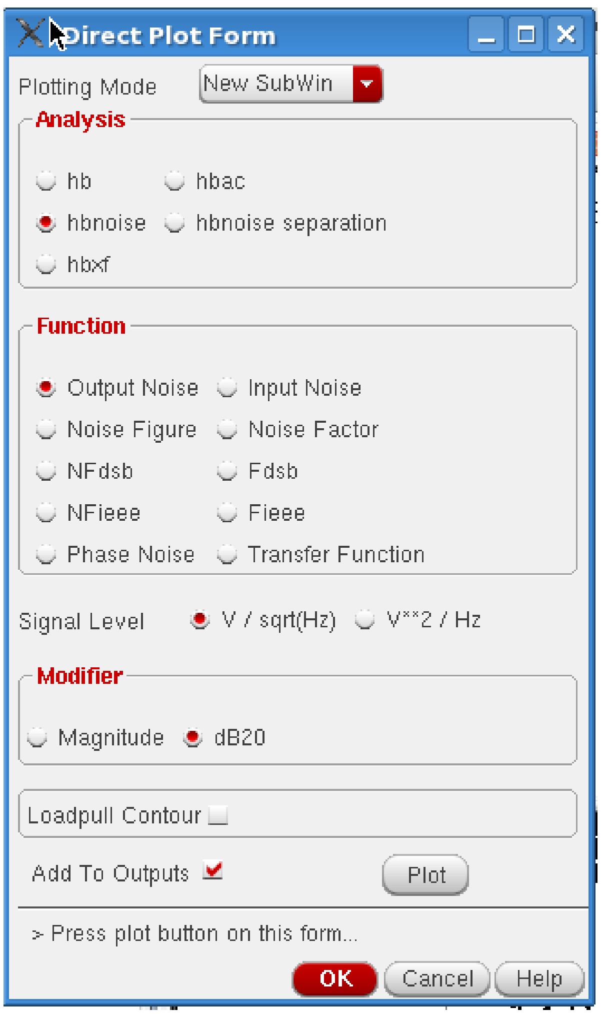
The lower subwindow displays the plot of the Voltage Conversion Gain versus the Output frequency. It rolls off at higher frequencies due to the degraded input match as the input moves above 1.9GHz. Since in the HBAC *Choosing Analyses* form,you set the *Sweep Type* to be *linear*, the plot has linear spacing on X-axis.

If you hold the cursor over the curve at 100KHz, you can read the low frequency conversion gain off the plot as shown in the above figure. It is *14.25dB* .

Plotting Mixer Noise Figure

1. In the *Direct Plot Form*, select *hbnoise* .

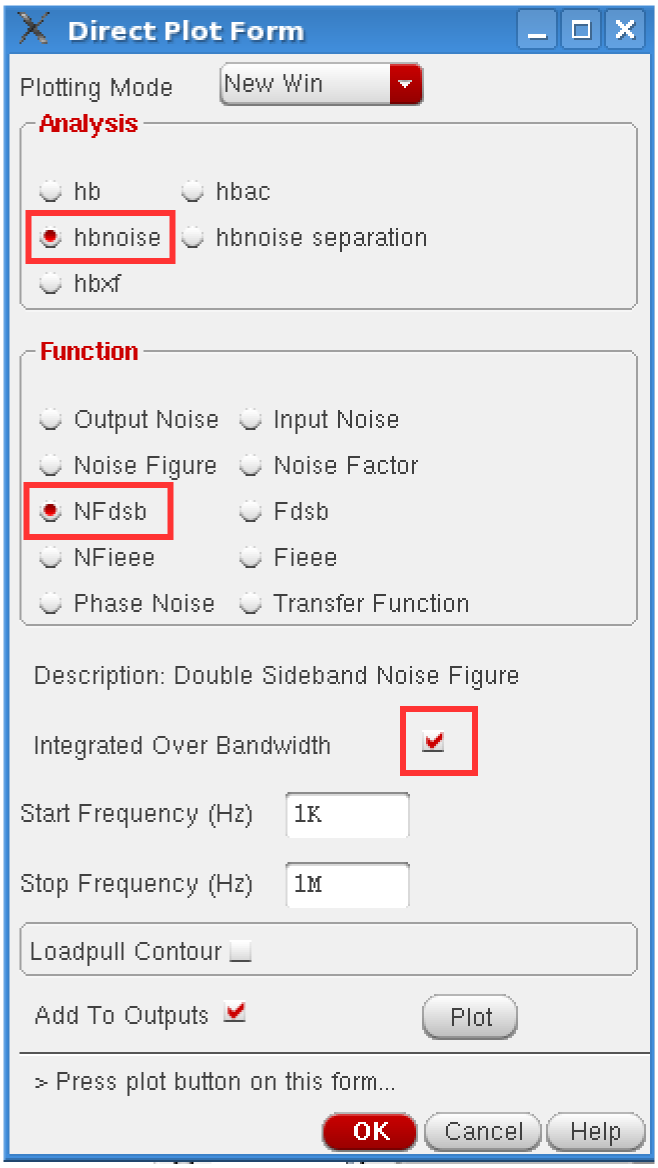
Figure A-324 Direct Plot Form - Plotting Mixer Noise Figure



2. Change the *Plotting Mode* to *New Win.*

3. Select *NFdsb,* in the *Function* section. NFdsb is double sideband noise figure. The form will change, as shown in the figure below

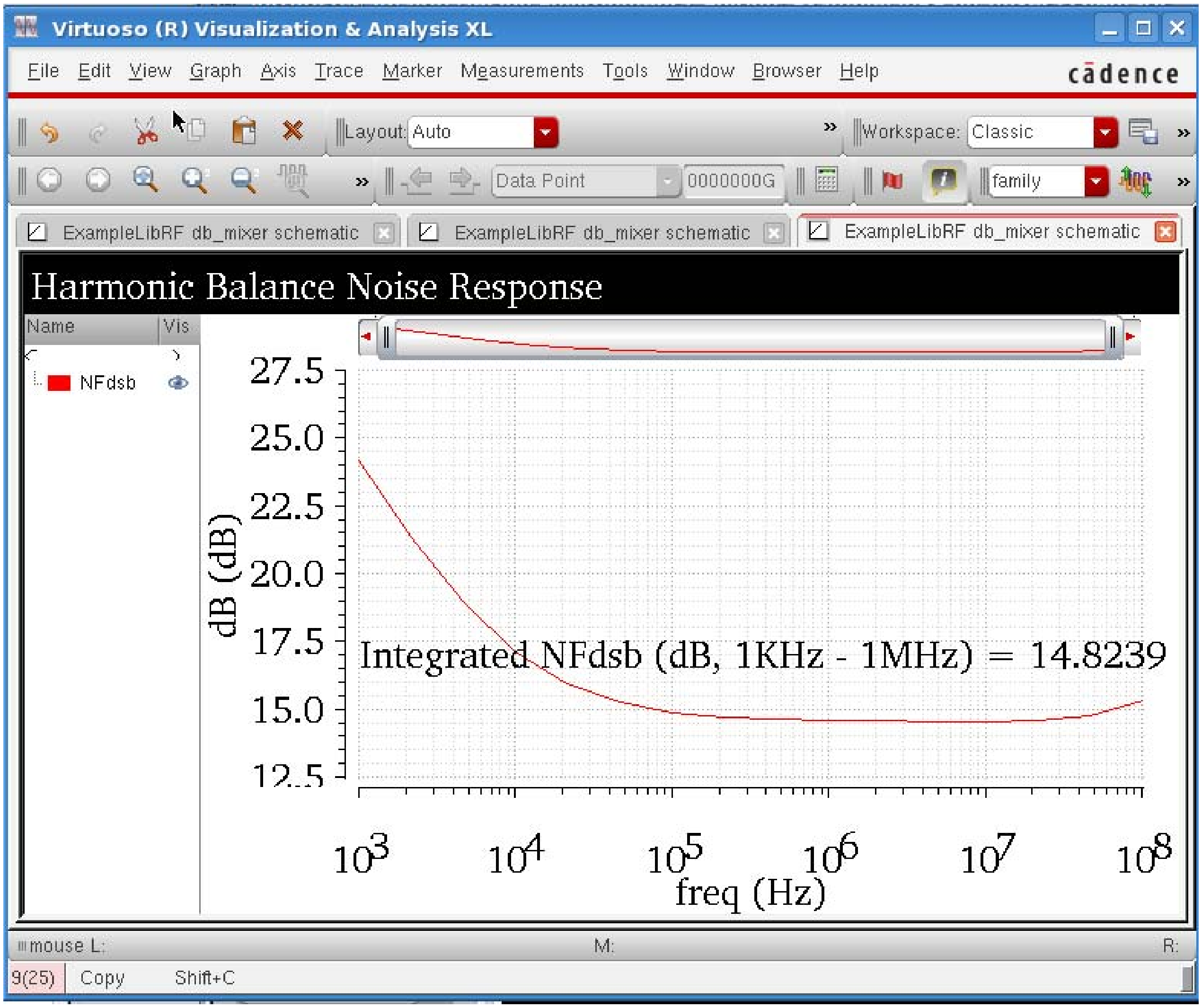
Figure A-325 Direct Plot Form DSB Noise Figure Setup



4. Select *Integrated Over Bandwidth*.
5. Type *1K* in the *Start Frequency (Hz)* field and *1M* in the *Stop Frequency (Hz)* field.
6. Click *Plot*.

   The Integrated Noise Figure Plot is shown below:

Figure A-326 Integrated Noise Figure Plot

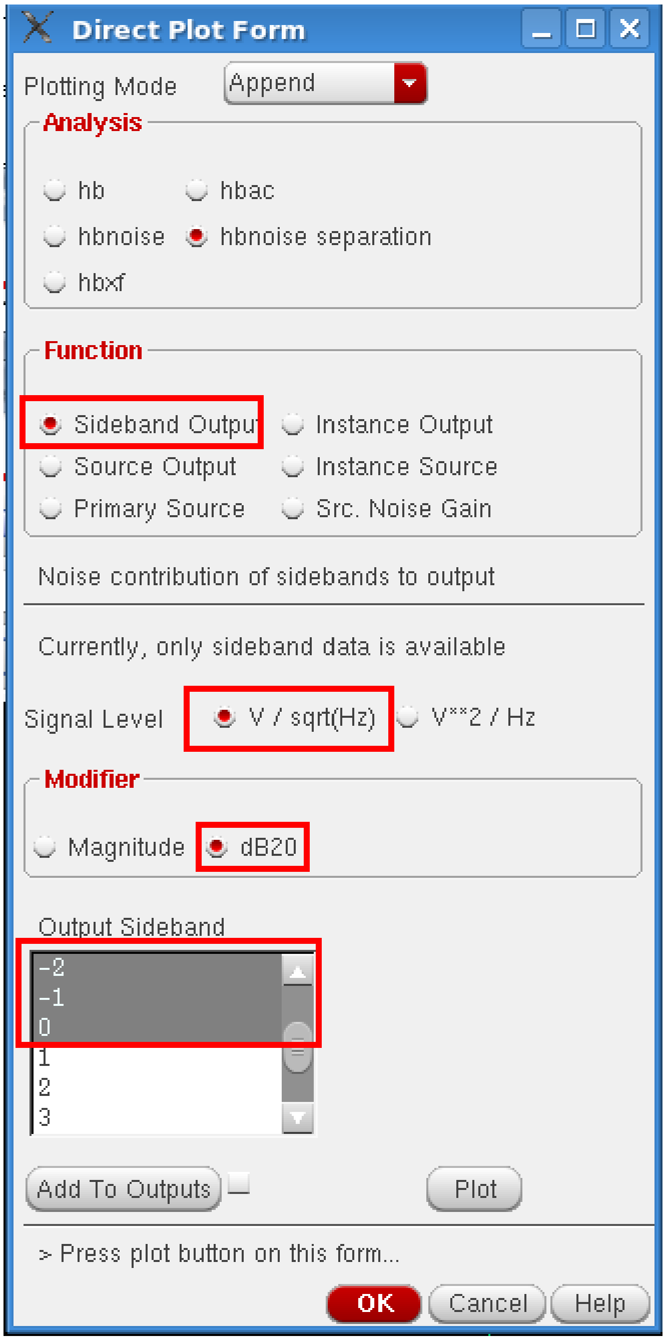


To determine the noise figure at different frequencies, move the cursor along the noise figure curve in the waveform window.

Next, you will plot the hbnoise separation. When you want to understand the specifics about your noise problem, you can use hbnoise separation. The Noise Summary is not able to differentiate how much noise is coming from the noise source and how much from the transfer function. With noise separation, you can attempt to decrease the output noise by decreasing the noise sources with different device dimensions or by decreasing the transfer function by choosing alternative circuit architectures. When *Noise separation* is set to `yes` in the *Choosing Analyses* form, the hbnoise separation feature is included during the simulation and the corresponding results are saved.

7. In the *Direct Plot Form*, select *hbnoise separation* in the *Analysis* section. The *Direct Plot Form* changes, as shown below.

Figure A-327 Direct Plot Form for hbnoise Separation



8. Change the *Plotting Mode* to *New Window*.

9. In the *Function* section, select *Sideband Output. Sideband Output* plots the noise contribution of selected sidebands to the output. Different sideband numbers measure the noise from different noise frequencies when mixed with different harmonics of the hb analysis. In this example, noise frequencies near zero Hz and the noise that mixes from just below the first and second harmonics of the LO are selected.

10. In the *Signal Level* section, select *V/sqrt(Hz).*

11. Choose *dB20* in the *Modifier* section.

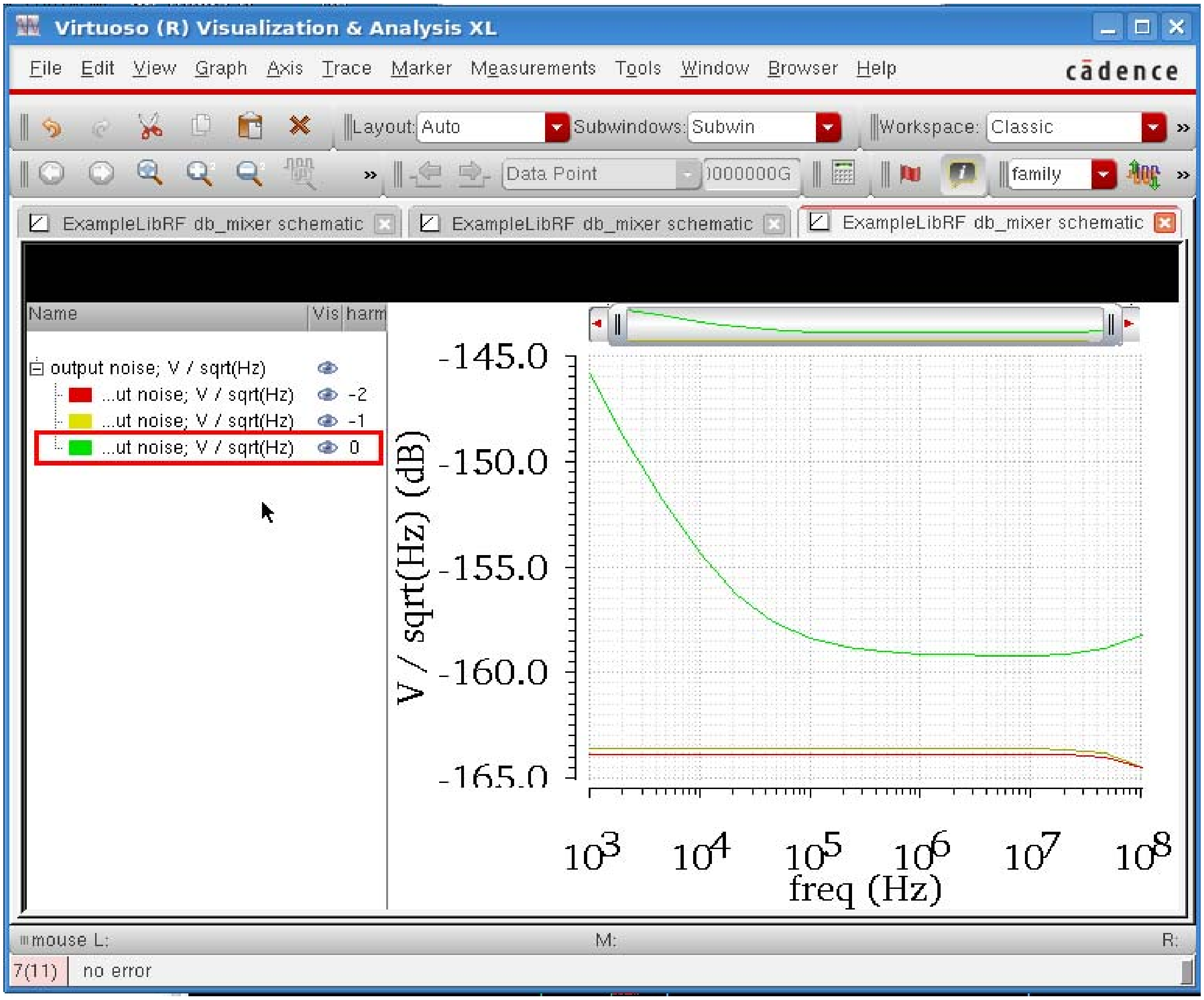
12. Select the *0, -1, and -2* in the *Output Sideband* section. This will plot the noise contribution of the *0, -1, and -2* sidebands to the output. This allows the identification of which noise frequencies are causing the largest noise contribution at he output of the circuit.

Tip

You can hold cursor over the *0* output sideband and slide the cursor to include the *-2* output sideband.

13. Click *Plot*. The hbnoise separation is plotted, as shown in the figure below. Note that the harmonic numbers plotted are in the legend on the left side of the screen. HBnoise Separation, 0, -1, -2 Sideband.

Figure A-328 HBnoise Separation, 0, -1, -2 Sideband



14. Click the + sign to the left of *output noise* in the legend on the left side of the waveform window. The legend expands. You can see that the *0 Output sideband* corresponds to the green trace. The *0 Output sideband* contributes the most noise and is the worst noise offender. Next, plot *Instance Output* for the largest noise contributor above. (The zero sideband) This will show which components contribute the most noise at the output.

a. In the *Direct Plot Form*, select *Instance Output* in the *Function* section.

b. Set the *Signal Level* to *V/sqrt(Hz)* and the *Modifier* to *Magnitude* .

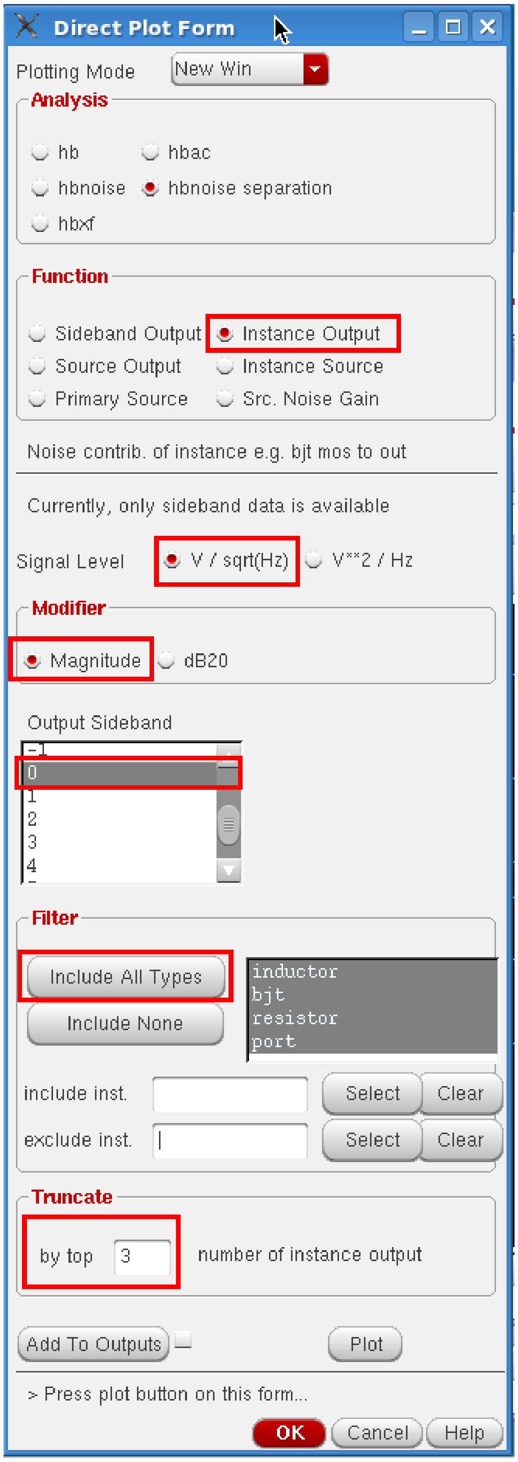
c. Select the *0 Output Sideband*.

d. In the *Filter* section, select *Include All Types.*

e. In the *Truncate* section, *type* `3` in the by top text field.

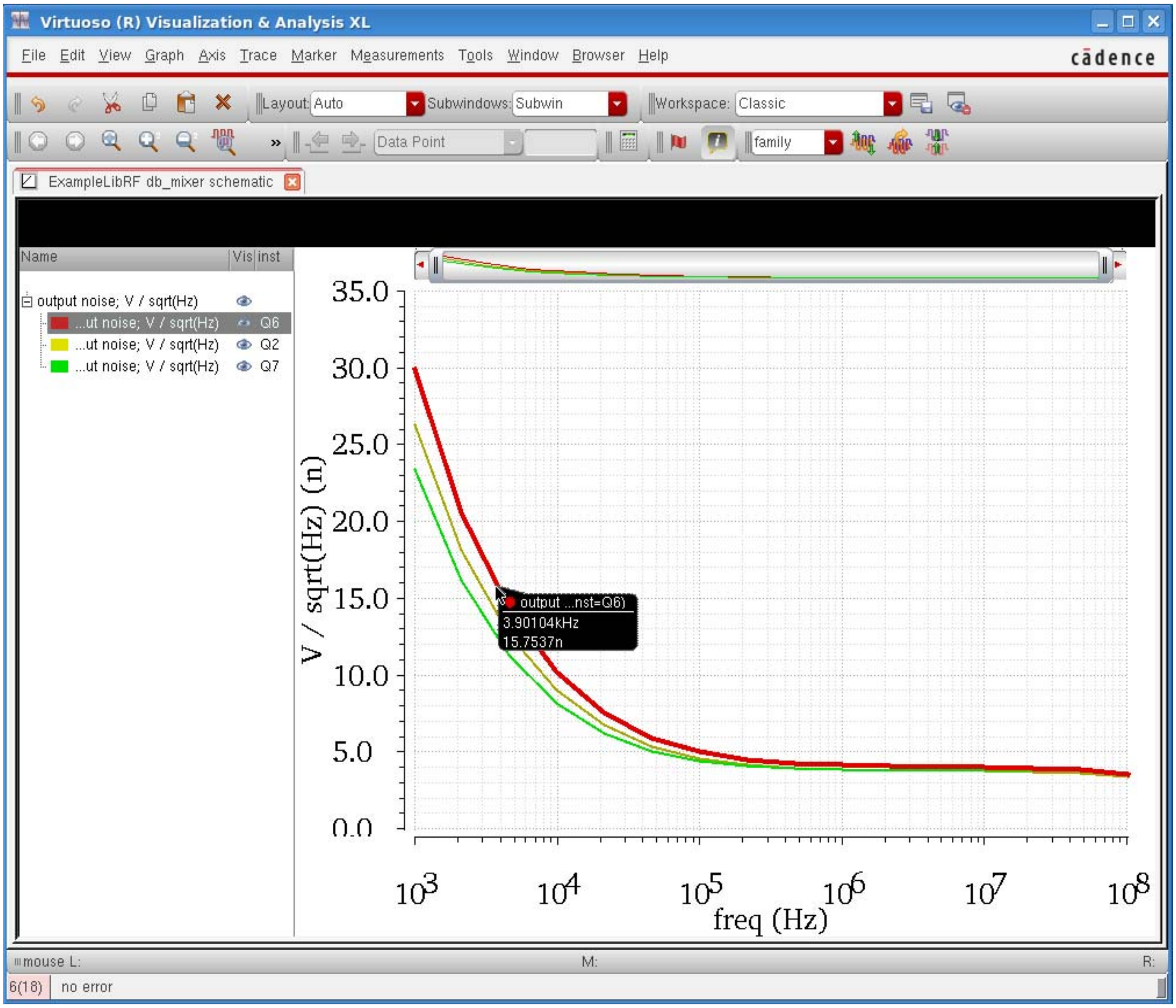
The *Direct Plot Form* should look like the one below.

Figure A-329 Direct Plot Form for Plotting Instance Output for Noise Contribution of instances



15. Click *Plot*. The plot showing the top three noise contributors is displayed.

Figure A-330 Output Noise Top Instance Contributors



Click the + sign to the left of *output noise* in the legend on the left side of the waveform window. The legend expands to show the top three instance noise contributors.

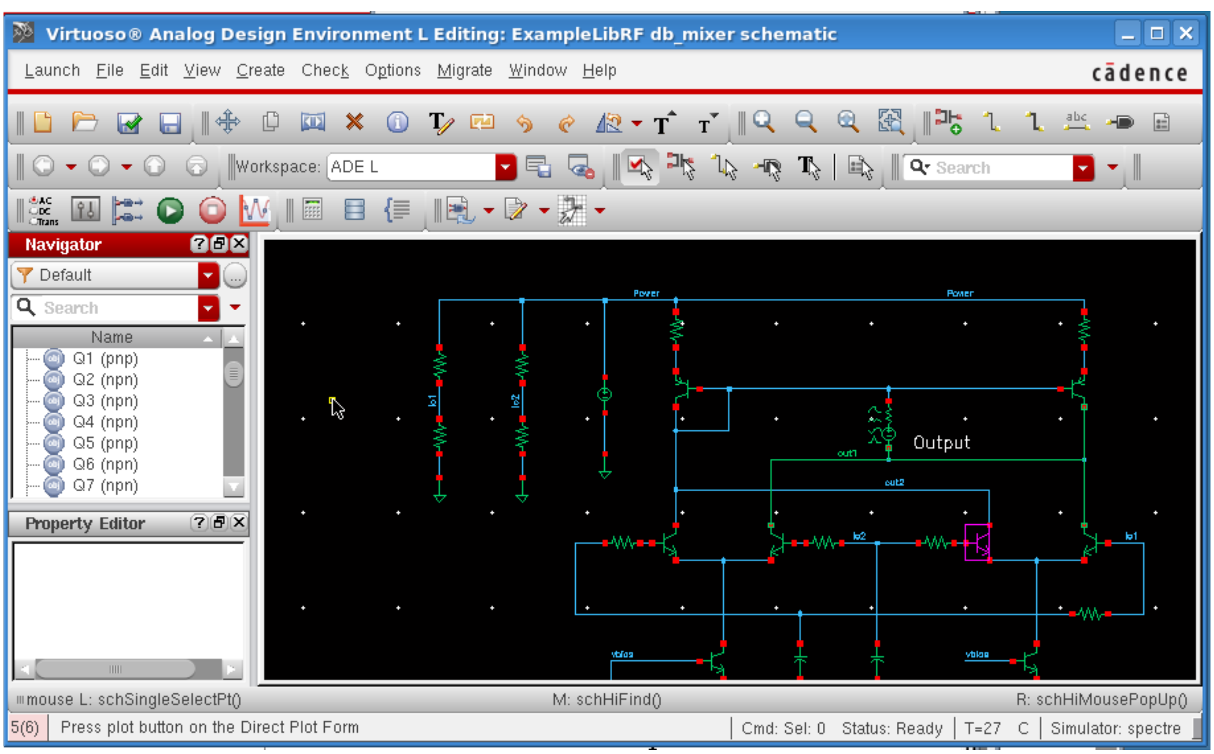
1. If you move your mouse cursor over one of the traces, you can read the instance from the cursor readout and identify which instance contributes the most amount of noise. Above, the red line represents the noise contribution from instance Q6 (legend shown), the yellow line below represents the noise contribution from instance Q2, and the bottom green line represents the noise contribution from instance Q7.

2. To find Q6 in the schematic, choose *Edit - Find* in the schematic. The *Schematic Find* form is displayed. Search for *instanceName == Q6* in the *cellview* and click *Apply*.

Figure A-331 Schematic Find Form

Q6 is highlighted in the db_mixer schematic below in pink, as shown below.

Figure A-332 Q6 Highlighted in dbmixer Schematic



3. Next, plot the Source Output so you can see which individual noise sources within the circuit are contributing the most noise. In the *Direct Plot Form*:

a. In the *Function* section, select *Source Output.*

b. In the *Signal Level* section,select *V/sqrt(Hz).*

c. In the *Modifier* section, select *Magnitude*.

d. Leave the *Output Sideband* at `0`.

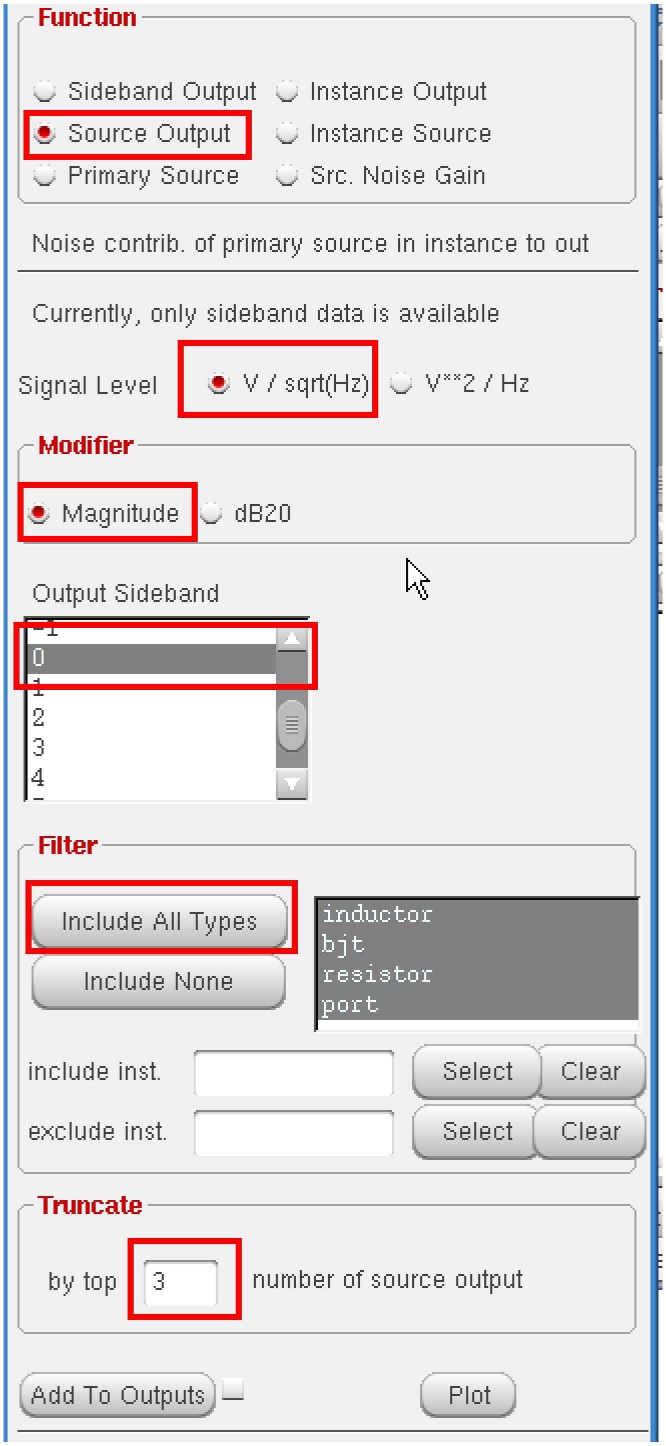
e. In the *Filter* section, select *Include All Types*.

f. In the *Truncate* section, *type 3* in the *by top number of source output* text field

g. The *Plotting Mode* should still be set to *New Window*.

The *Direct Plot Form* should look like the one below.

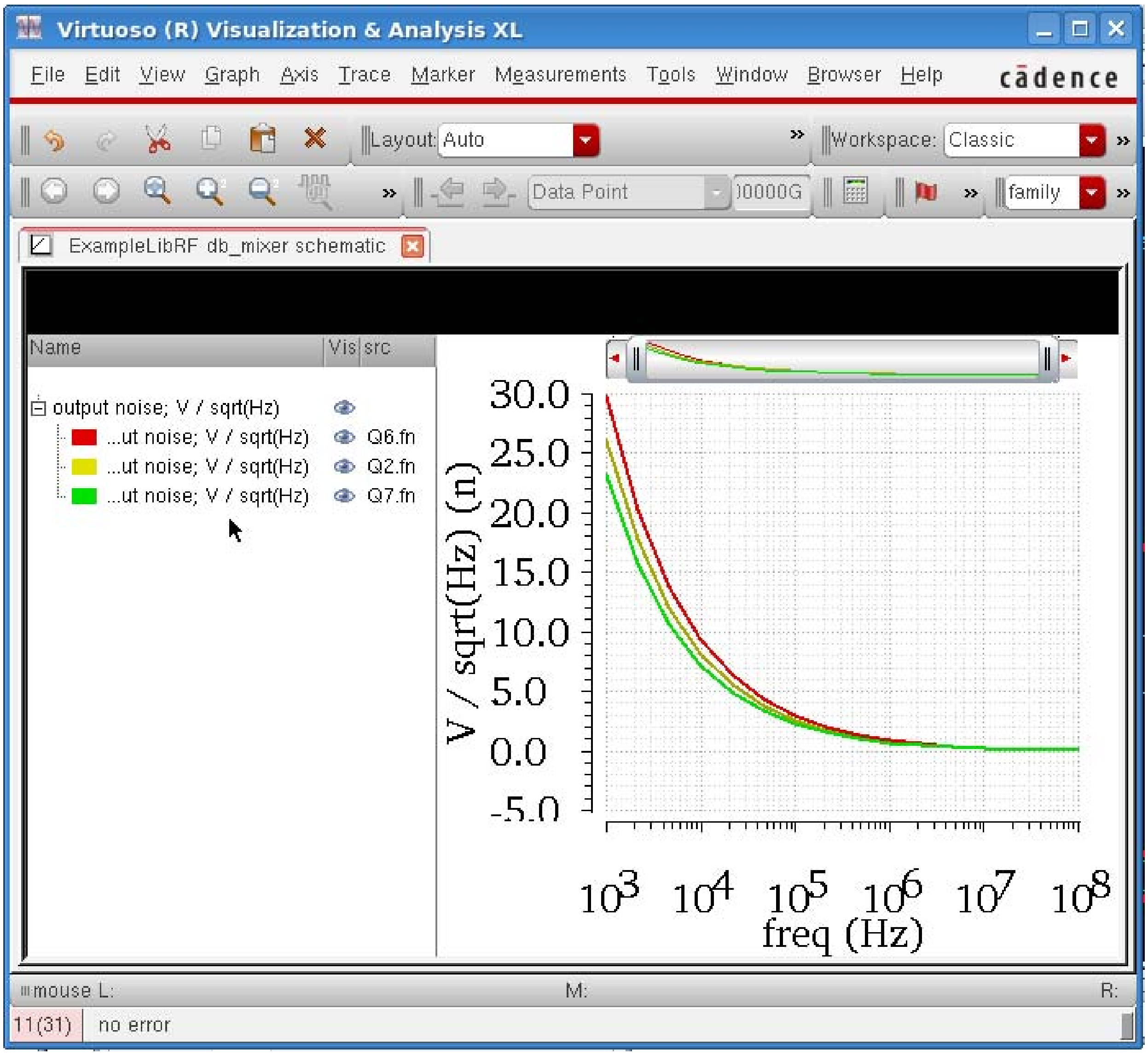
Figure A-333 Direct Plot Form for Plotting by Top 3 Source Noise Contributors



4. Click *Plot* to see which individual sources within the circuit are making the most noise.

5. In the waveform window, click the + sign to the left of *output noise* in the legend on the left side of the waveform window. The legend expands to show the top three source noise contributors.

Figure A-334 Top 3 Source Noise Contributors

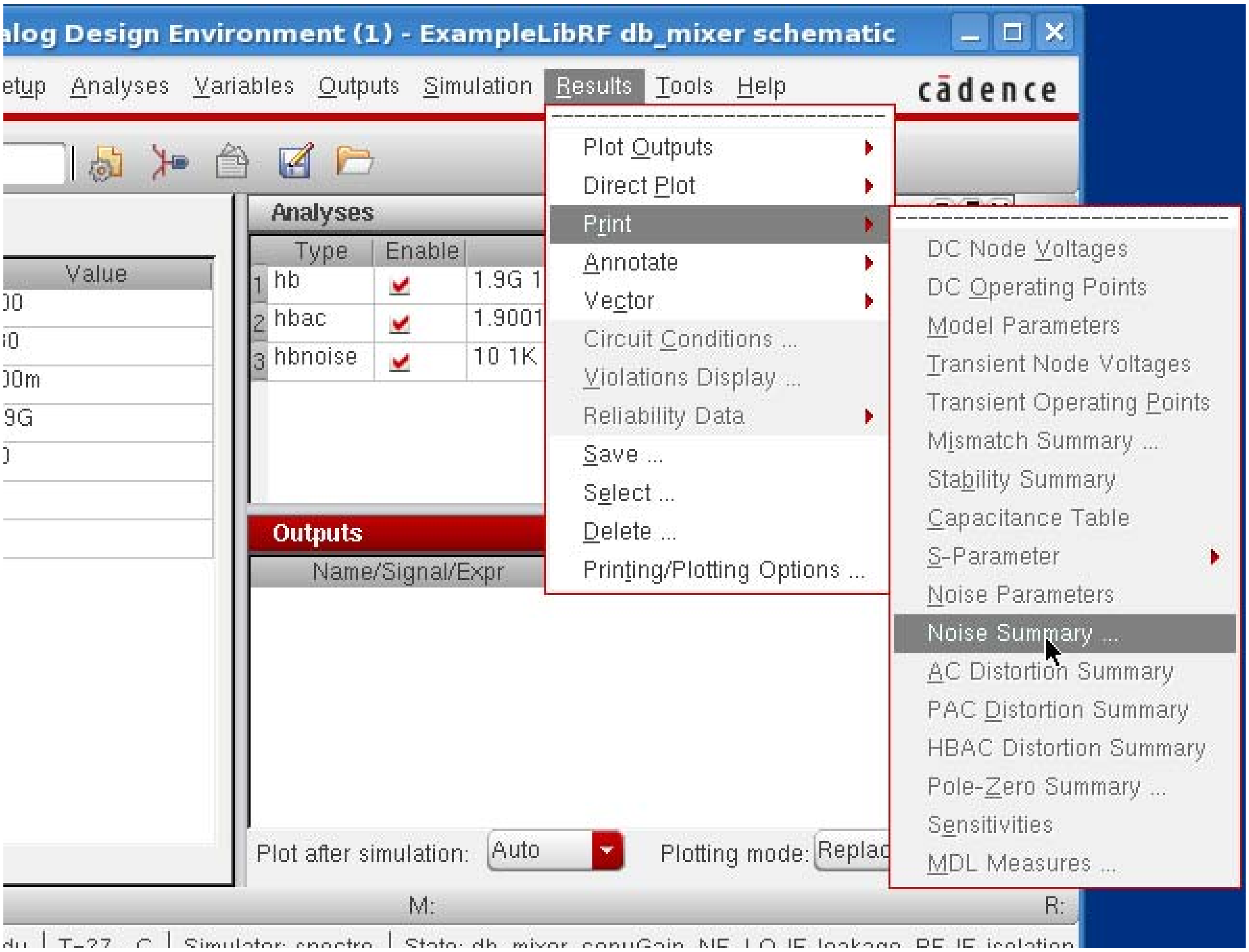


Note that the top three Source Output contributors are displayed in the legend on the left side of the graph: *Q6: fn*, *Q2: fn*, and *Q7:fn*. *fn* refers to flicker noise. Refer to the Noise

Summay section in Chapter 3, "Frequency Domain Analyses: Harmonic Balance," to get a description of the parameter names for the different device types.

6. In the *Direct Plot Form*, click *Cancel*.
7. In the waveform window, choose *File - Close All Windows.*
8. Next, look at the top noise contributors using the *Noise Summary Form*. In the Analog Design Environment window, choose *Results - Print - Noise Summary.*

Figure A-335 Invoking Noise Summary Form



The Noise Summary window is displayed, as shown below.

Figure A-336 Noise Summary form

9. In the *Noise Summary* window, select *hbnoise_src* from the *Data from section*. This prints the output noise of the hbnoise-rif analysis. Note that the selection *hbnoise_src* lists the noise at the source, not at the output of the circuit.

10. Select *integrated noise* from the *Type* section.and *flat* from the *weighting section*.

Integrated noise produces a noise summary integrated over a frequency range using the specified weighting. If you choose integrated noise, you have the option of using a weighting factor. The flat weighting factor specifies that the integration be performed on the original unweighted waveform. The *from weight file selection* option specifies that, before the integration is performed, the noise contributions of particular frequencies in the original waveform be weighted by factors supplied from an input file.

11. Select*V* from the the *noise unit drop-down list*. This will be the noise units used in the hbnoise summary.

12. Type *1K* in the *From (Hz)* text field.

13. Type *1M* in the *To (Hz)* text field.

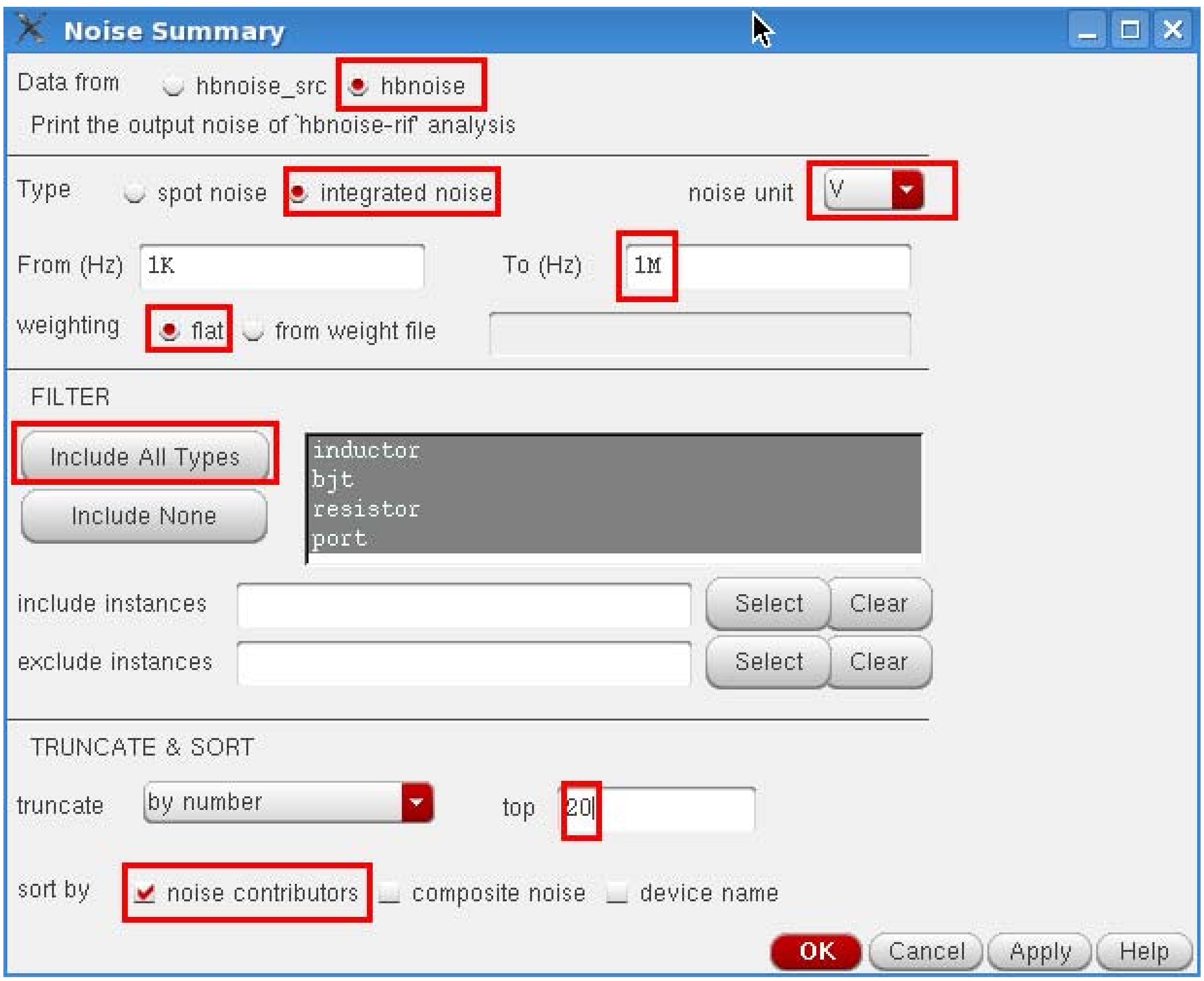
14. You can choose filtering details to include or exclude particular instances in your hbnoise summary. In the *FILTER* section, click *Include All Types*.

15. You can shorten your summary by specifying how many of the highest contributors to include in the summary, by specifying the percentage of noise a device must contribute to be included in the summary, or by specifying the level of noise a device must contribute to be included in the summary. Selectby number in the *truncate* section and type *20* in the *top* field. You will be printing the top 20 noise contributors.

16. Select *noise contributors* from the *sortby section*.

The*Noise Summary* form should look like the following:

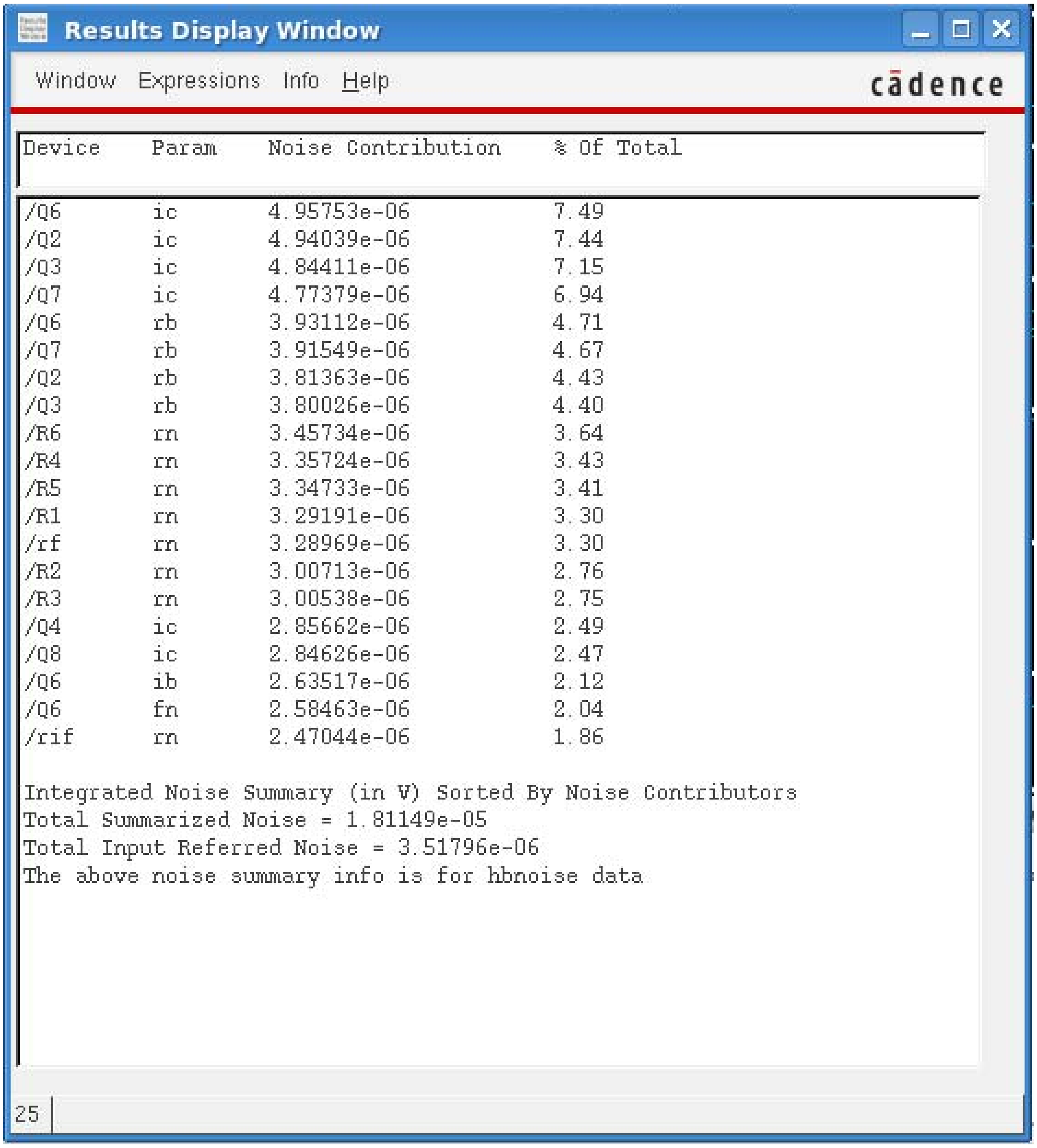
Figure A-337 HBnoise Summary Form



17. Click *OK*.

The *Results Display Window*is displayed containing the nosie summary results, as shown below.

Figure A-338 Noise Summary Results



The *Results Display Window* lists the individual contributors, the specific noise mechanism within the semiconductors causing the noise, and the noise contribution. The total input-referred noise voltage and output-referred noise voltage is shown at the bottom of the *Results Display Window*. You may need to use the scroll bar at the right side of the *Results Display Window* to see the bottom part of the results.

Note that the output and input noise include the noise from all the noise contributors, and not just the contributors in the form. The *Param* column uses the same abbreviations as seen in the legend on the left side of the waveform window. For example, *fn* refers to flicker noise. Refer to the*Noise Summary* section in Chapter 3, "Frequency Domain Analyses: Harmonic Balance," to get a description of the parameter names for the different device types.

18. Clean up the screen for the next set of measurements.

   a. Close the *Results Display Window* by choosing *Window - Close*.

   b. Close the *Analog Design Environment* window by choosing *Session -Quit.*

   c. Click *No* in the *Save State* window.

   d. In the Schematic window, choose *File - Close.*

Summary

Harmonic Balance (hb), hbac, and hbnoise analyses were setup and simulations were run to determine the Mixer Conversion Gain, RF to IF Isolation, LO to IF Leakage, and Noise Figure for a receive mixer. In addition, you identified different ways to view noise in the receiver using the Noise Separation feature and Noise Summary form

In the next section, you will measure 1dB Compression Point and Desensitization with an RF blocking signal present.

1dB Compression Point, Desensitization, and Blocking

All circuits reach a compression point. The large signal effects at this point cause the gain to stop increasing in a linear fashion. Since this is a large-signal effect, a large-signal analysis is required. When measuring 1dB Gain compression point, the idea is to sweep the input power over a range of power levels, and then plot the output power versus the input power. From the plot, the 1dB compression point can be determined (the 1dB compression point is the input signal level at with the receiver gain drops 1dB when compared to the ideal linear gain).

Receiver desensitization interference occurs when a strong off-channel signal overloads a receiver front end and thus reduces the sensitivity to weaker on-channel signals.

The setup for 1dB Compression Point is the same as that required by a desensitization measurement, except for the frequency of the RF input. For the compression point measurement, the RF signal is in the passband. For the desensitization measurement, the RF frequency is at the blocker frequency. Here, we show both the 1dB compression point and the desensitization measurement using the passband frequency. Usually, they are separate simulations with different frequencies for the RF signal. (This is done to save time in the workshop.)

For 1dB gain compression point measurement, the frequency component set in the RF input port is considered as the RF signal frequency.

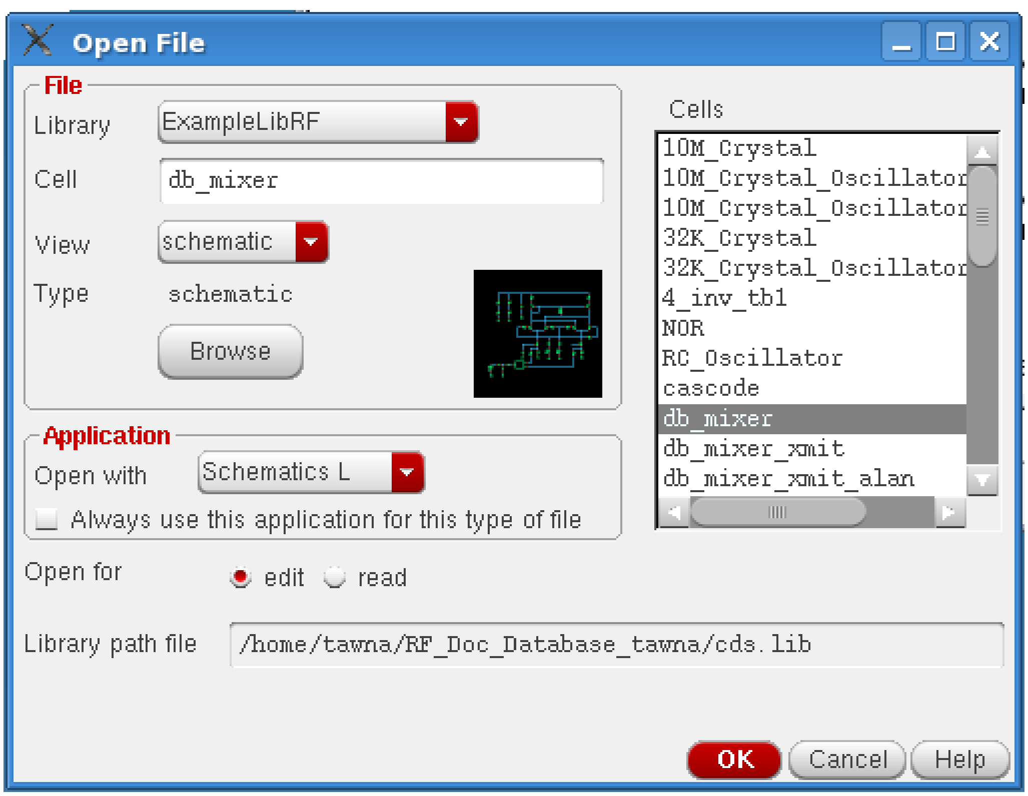
For desensitization measurement, the frequency component set in the RF input port is considered the RF blocker frequency.

You can set up the simulation to measure the conversion gain and noise figure as a function of RF blocker power.

Setting Up to Simulate 1dB CP and Desensitization

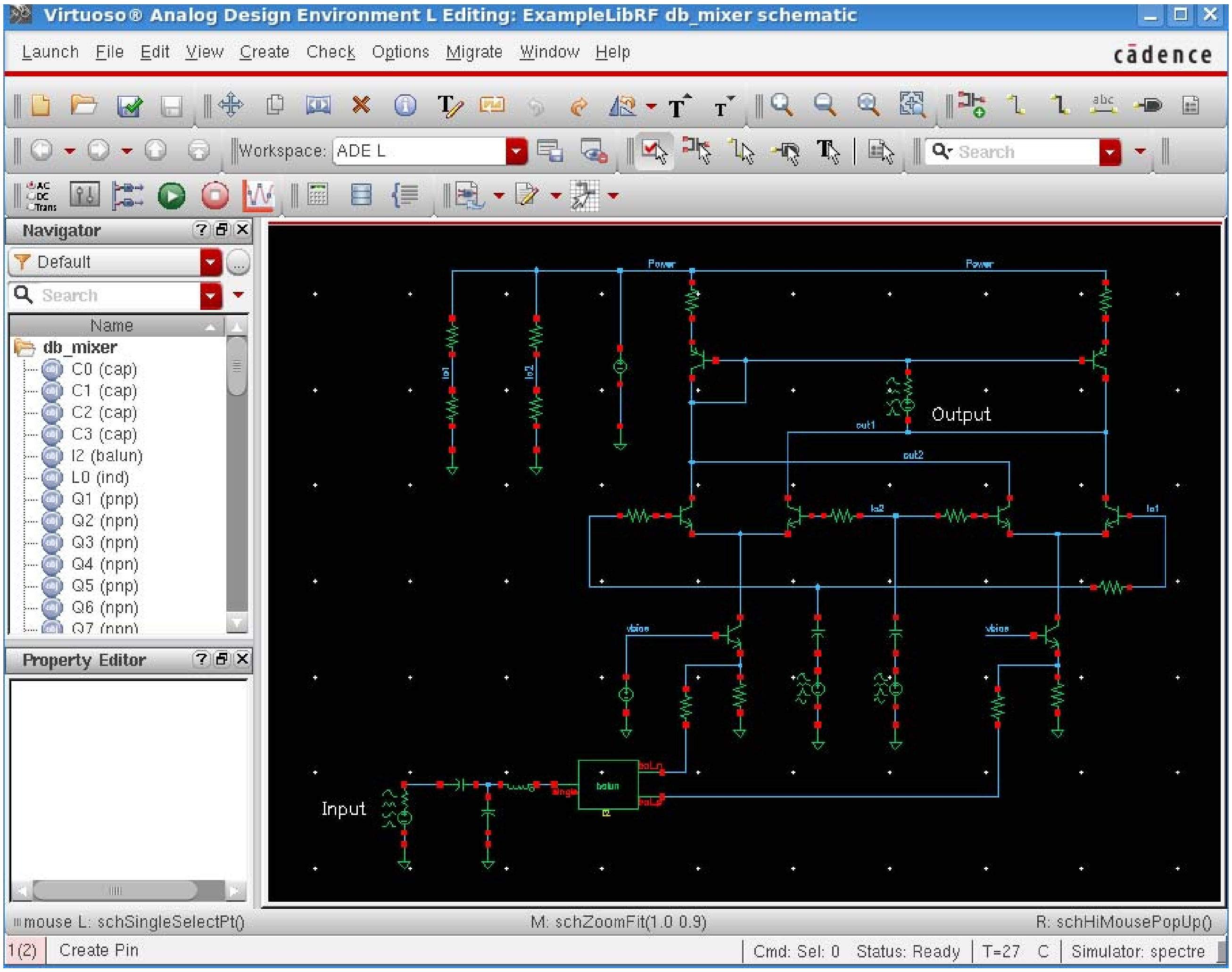
1. In the CIW, choose *File - Open - CellView*. The Open File form is displayed. Choose the *db_mixer schematic* from *ExampleLibRF*, as shown below.

Figure A-339 Open File Form



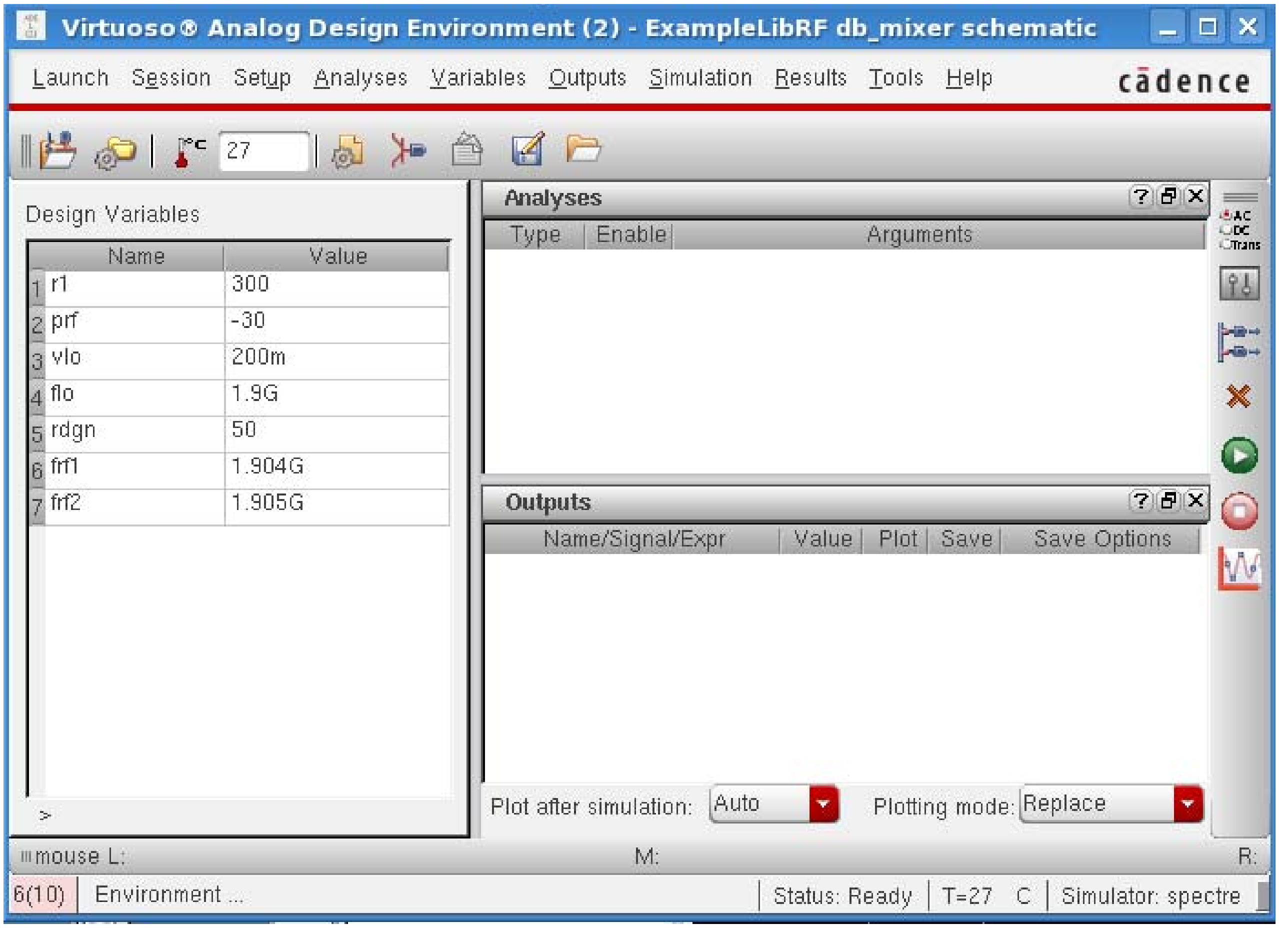
2. Click *OK*. The `db_mixer` schematic appears, as shown below.

Figure A-340 db_mixer Schematic



3. Start the *Analog Design Environment* from the schematic by choosing *Launch - ADE L*. The Simulation Window is displayed, as shown below.

Figure A-341 Analog Design Environment Simulation Window



4. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

The *Choosing Simulator/Directory/Host* form is displayed.

5. Specify the following in the *Choosing Simulator/Directory/Host* form:

a. Choose *spectre* as the *Simulator*.

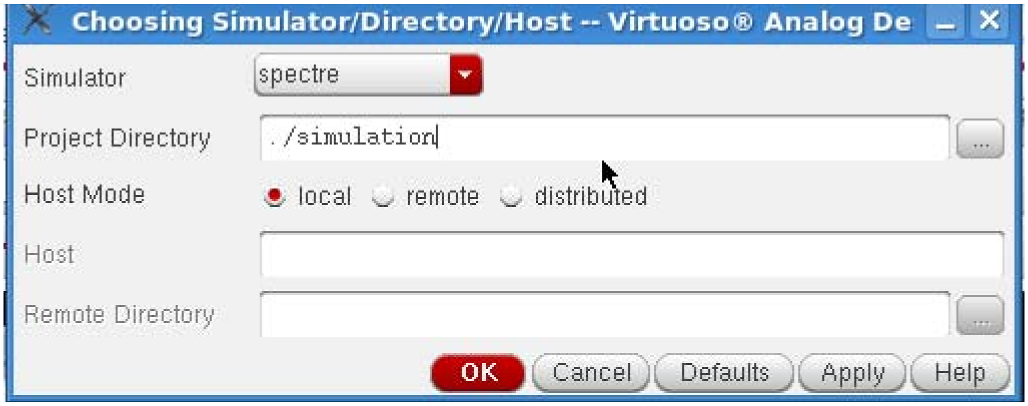
b. Type the name of the project directory, if necessary, in the *Project Directory* field.

c. Select the *Host Mode* that corresponds to your situation.

If you choose *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form appears similar to the following figure:

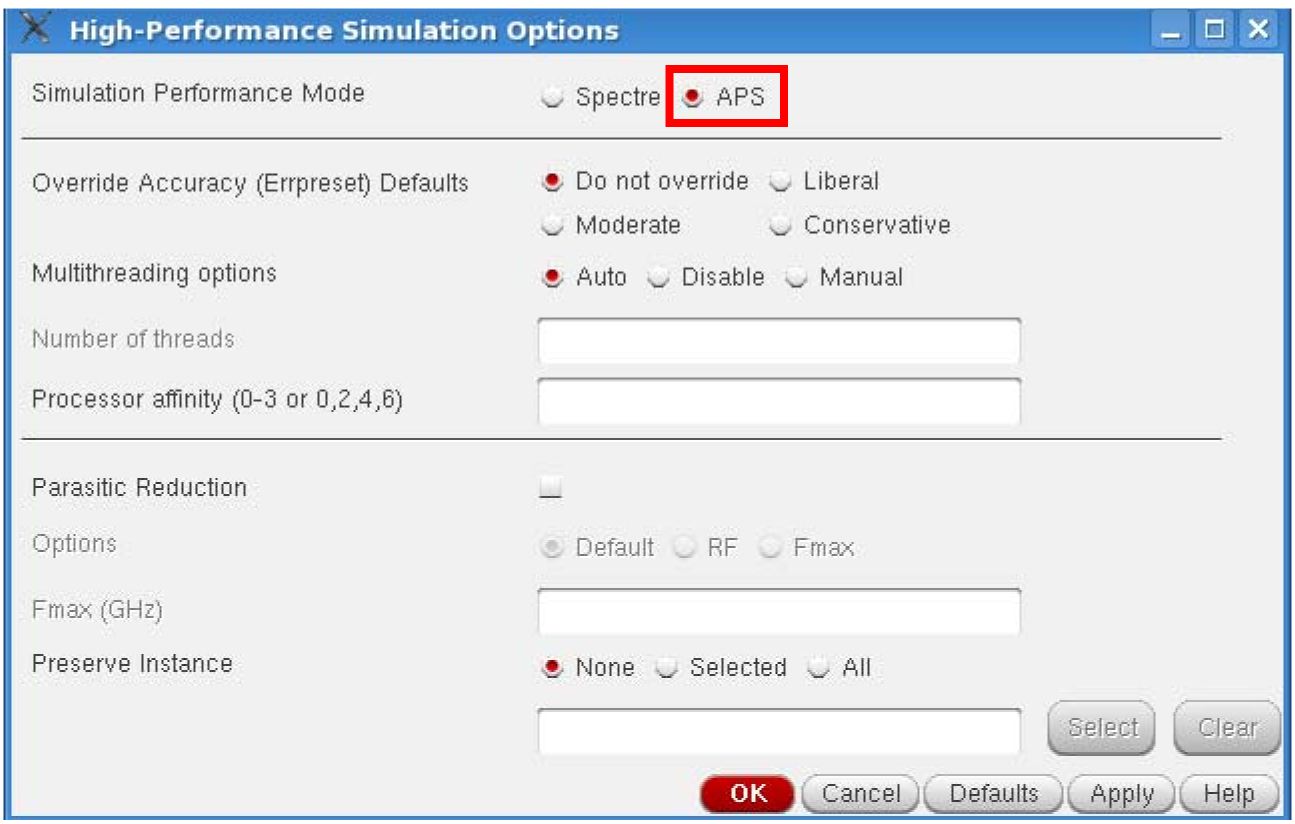
Figure A-342 Choosing Simulator/Director/Host Form



6. Click *OK* to close the *Choosing Simulator/Directory/Host* form.
7. Set up the High Performance Simulation Options,as follows:

   In the ADE window, choose *Setup - High Performance Simulation*. The High Performance Simulation Options window is displayed, as shown below.

Figure A-343 High Performance Simulation Options Form



a. In the High Performance Simulation window, select *APS*. Note that *auto* is selected for multithreading. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores.

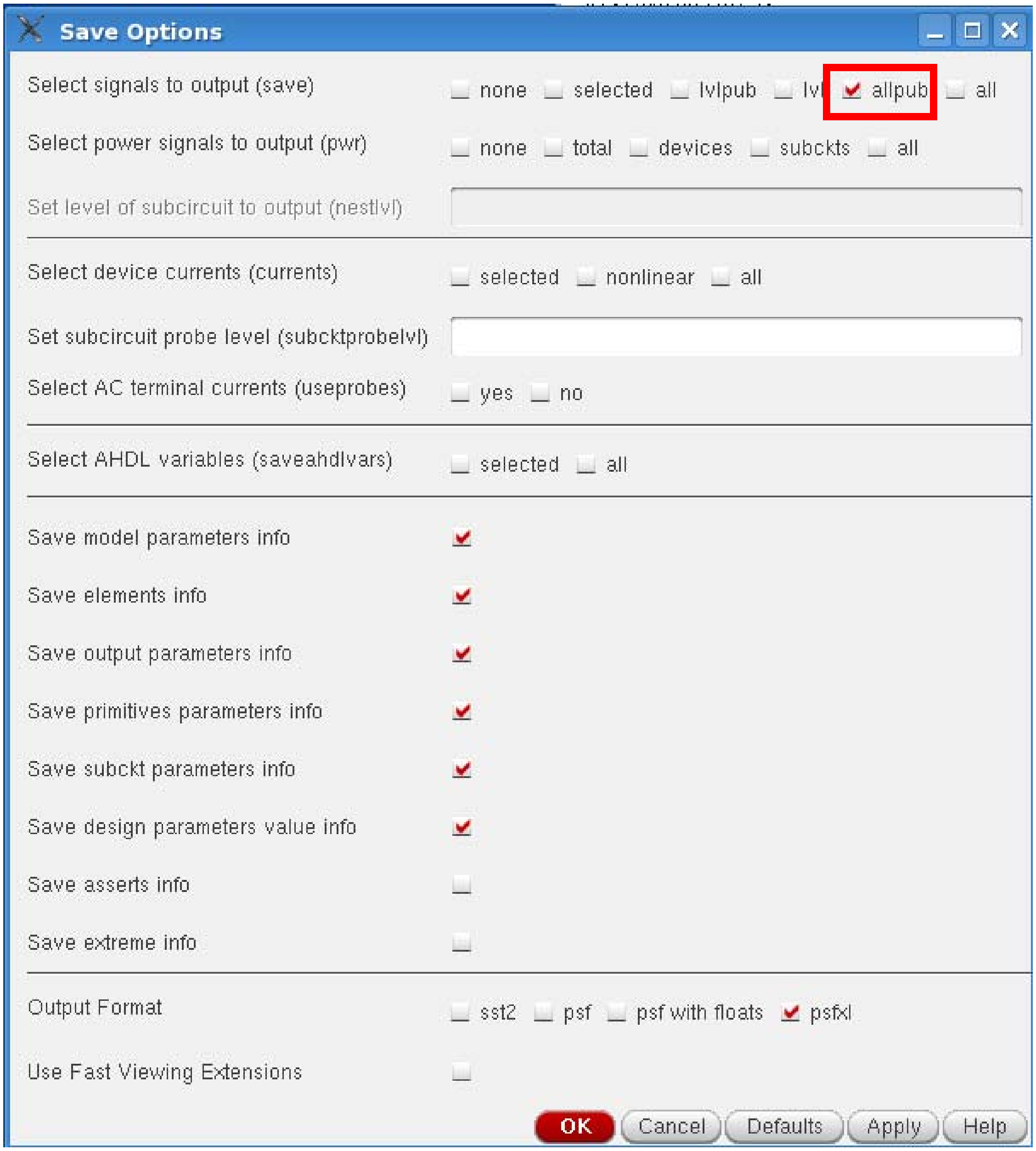
Note: The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading. For more information, see the *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator User guide*.

b. Click *OK.*

8. In the *Virtuoso Analog Design Environment* window, choose *Outputs – Save All*.

The *Save Options* form is displayed.

Figure A-344 Save Options Form



9. In the *Select signals to output* section, make sure that *allpub* is selected. This is the default which saves all node voltages at all levels of hierarchy, but it does not include the node voltages inside the device models.

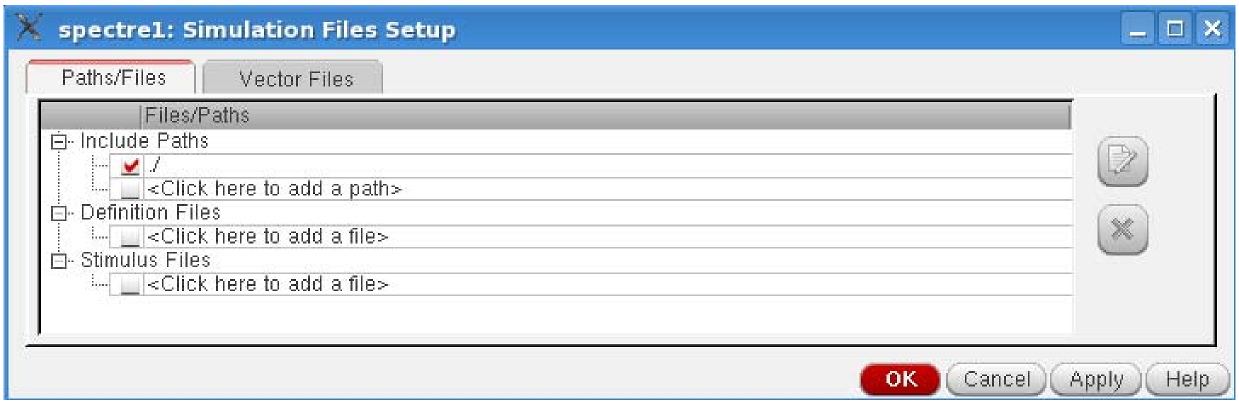
10. Click *OK* to close the *Save Options* form.

Setting Up Model Libraries

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.*

   The *Simulation Files Setup* form is displayed, as shown below.

   Figure A-345 Simulation Files Setup Form

2. Verify that *Include Paths* is set as shown above.
3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries*.

   The Model Library Setup form is displayed.

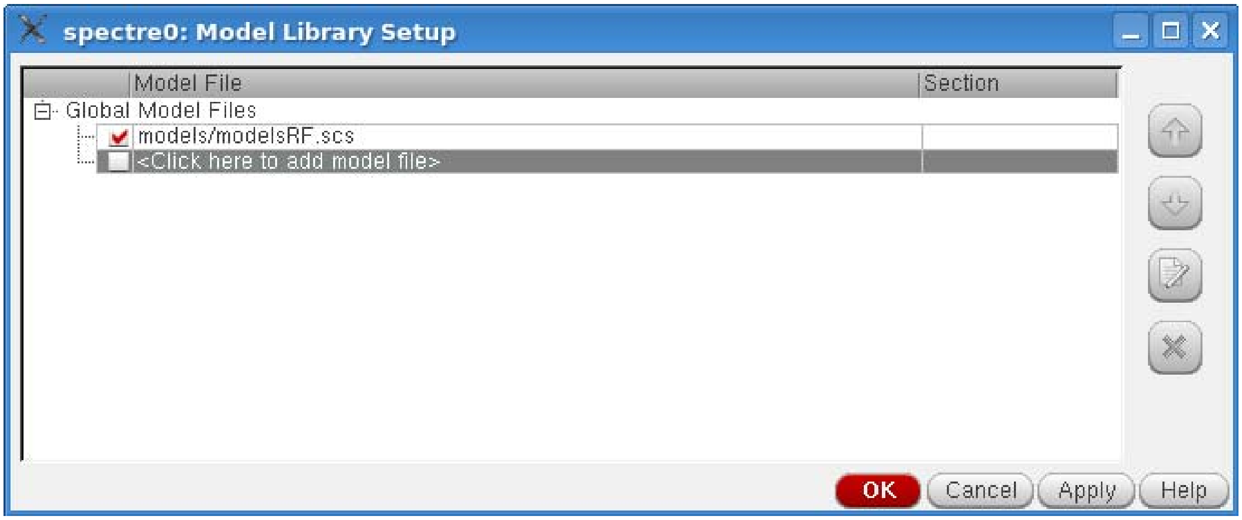
4. In the *Model Library File* field, type the following path to the model file including the file name:

   ```
   models/modelsRF.scs
   ```

   Alternately, you can browse to the location of your model files.

   The *Model Library Setup* form looks like the following:

Figure A-346 Model Library Setup Form



5. When you are finished adding model files, click OK to close the *Model Library Setup* form.

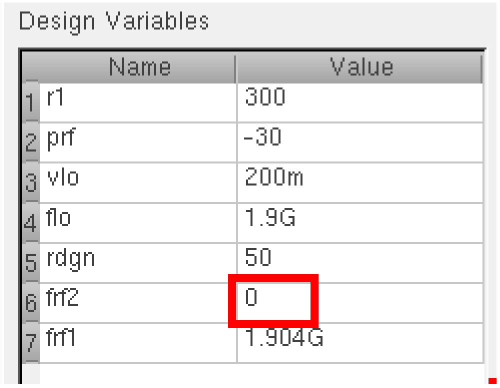
Setting Design Variables

In this simulation, the RF input frequency is set to 1.904G, and prf is the RF input power.

1. In the *Virtuoso Analog Design Environment* window, change the value of the *frf2* to *0* in the *Design Variables* section.
2. Ensure that the *frf1* variable is set to *1.904G*. If not, changeits value to 1.904G.
3. Press *Enter*.

   Your *Design Variables* section looks like the following figure:

Figure A-347 Design Variables Section of ADE

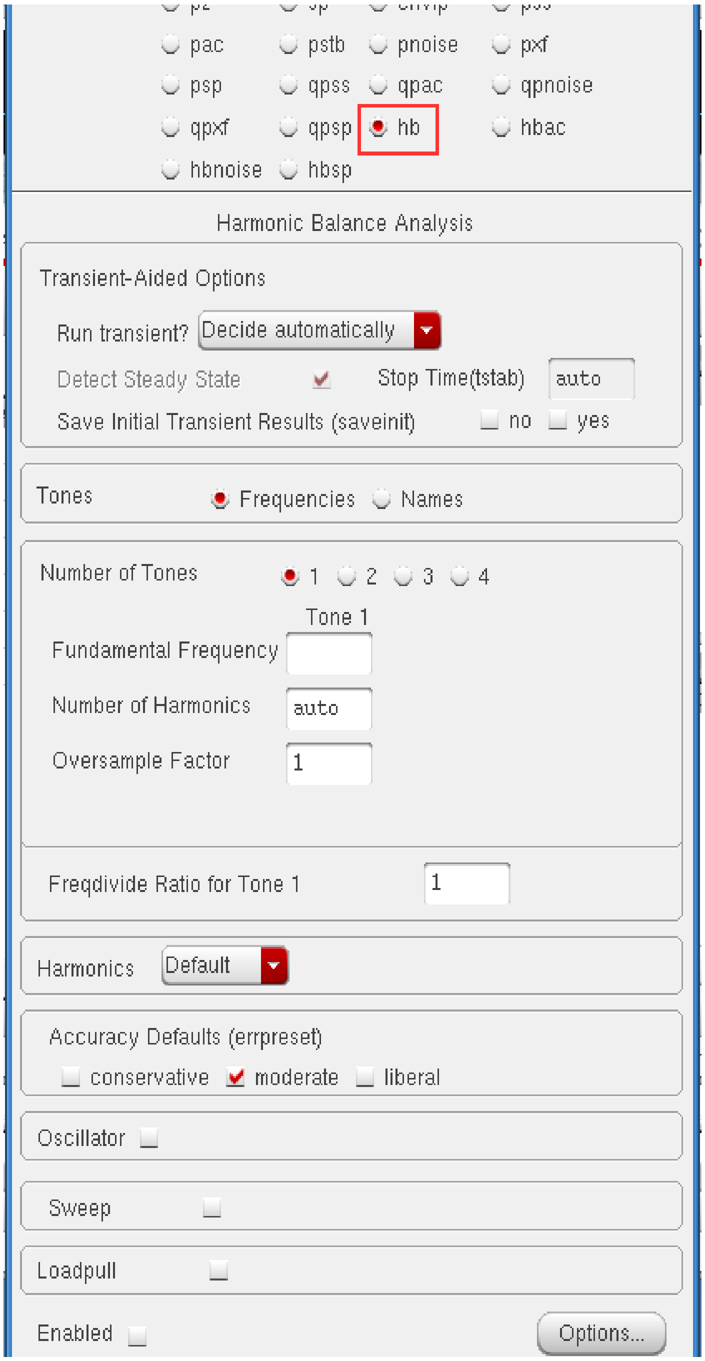


Set up the HB analysis

1. Click the *Choosing Analyses* icon on the right side of the ADE L window. 

   The *Choosing Analyses* form is displayed, as shown below.

Figure A-348 hb Choosing Analyses Form



Fill in the form as follows:

2. Select *hb* analysis.

3. Harmonic balance can set the harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient* selection in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected, and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

a. In the *Transient Aided Options* section, leave *Run transient?* at the default value of *Decide automatically*.

Run transient? will run the LO signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

b. Leave *Stop Time (tstab)* set to the default value of *auto*.

When *auto* is selected for *Stop time*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

c. When *Run transient?* is set to *Decide automatically*,the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

d. If you want to see the startup waveform, set *Save Initial Transient Results (saveinit)* to *yes*.

e. In the *Tones* section, choose *Frequencies* (this is the default value).

Harmonic balance can now set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. A transient analysis runs until steady-state is detected, and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected). If you want to manually set transient-aided hb, select *Yes* from the *Run Transient?* drop-down list and set a time for the transient in the *Stop Time (tstab)* field. In this mode, the stop time of the transient analysis in the *tstab* interval cannot be automatically extended.

f. Select *2* for the *Number of Tones*.

g. Type *1.9G* and *1.904G* in the *Fundamental Frequency* fields. *1.9 GHz* is the LO frequency and *1.904GHz* is the RF (frf1) frequency. *Tone 1* should be the LO or signal which causes the most nonlinearity.

   The *Number of Harmonics* field for *Tone 1* is automatically set to *auto*. For *Tone 2*, choose *5* harmonics. Your circuit is operating near the compression point, so a higher number of harmonics has been chosen.

h. Type *1* and *1* in the *Oversample Factor* fields. When all the signals in the system (including currents) are nearly sinusoidal, then an *Oversample* of *1* is appropriate.

i. Leave *Harmonics* set to *Default*.

j. In the *Accuracy Defaults* section, ensure that *moderate* is selected. For most normal measurements *errpreset* should be set to *moderate*. When you need to measure really small distortions, then use *conservative*.

k. To set up a sweep analysis, click *Sweep* and set *Sweep* to *1* (this is the default value).

l. For *Frequency Variable?* select *no*. You will be sweeping input power rather than frequency.

m. Type *prf* in the *Variable Name* field.

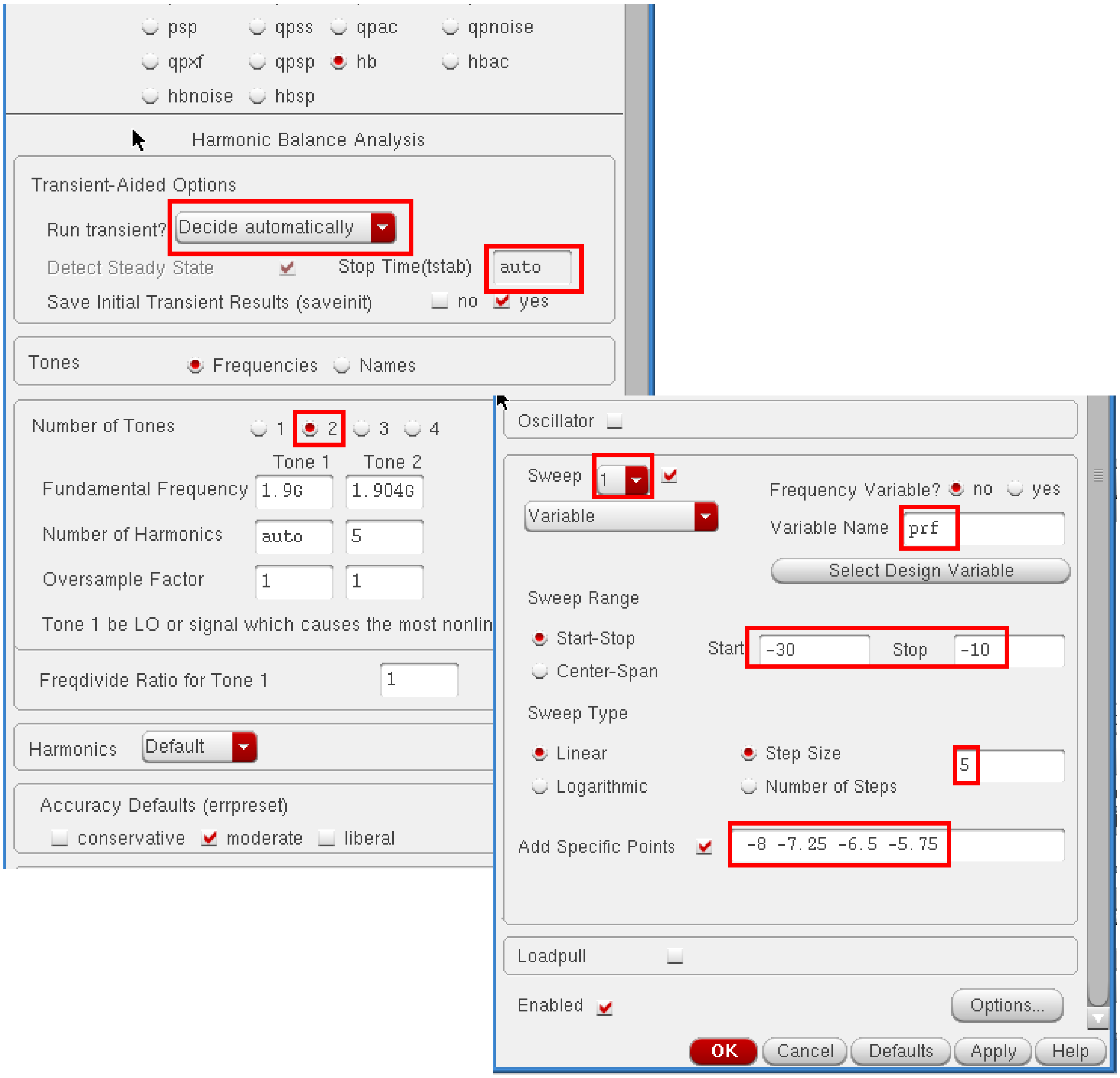
n. In the *Sweep Range* section, type *-30* in the *Start* field and *-10* in the *Stop* field.

o. Setthe *Sweep Typeto Linear* and type *5* in the *Step Size* field.

p. Click *Add Specific Points*.

   Type *-8 -7.25 -6.5 -5.75* in the *Additional Points* field. Note the spaces between the entries. You may need to use smaller spacings in the power sweep as the input power gets large and you approach the compression point. Using smaller spacings in the power sweep near the compression point helps with convergence.The *Choosing Analyses* form should look like the following:

Figure A-349 hb Choosing Analyses Form



For more information on setting up the Choosing Analyses form, see Chapter 3, “Frequency Domain Analyses: Harmonic Balance,”.

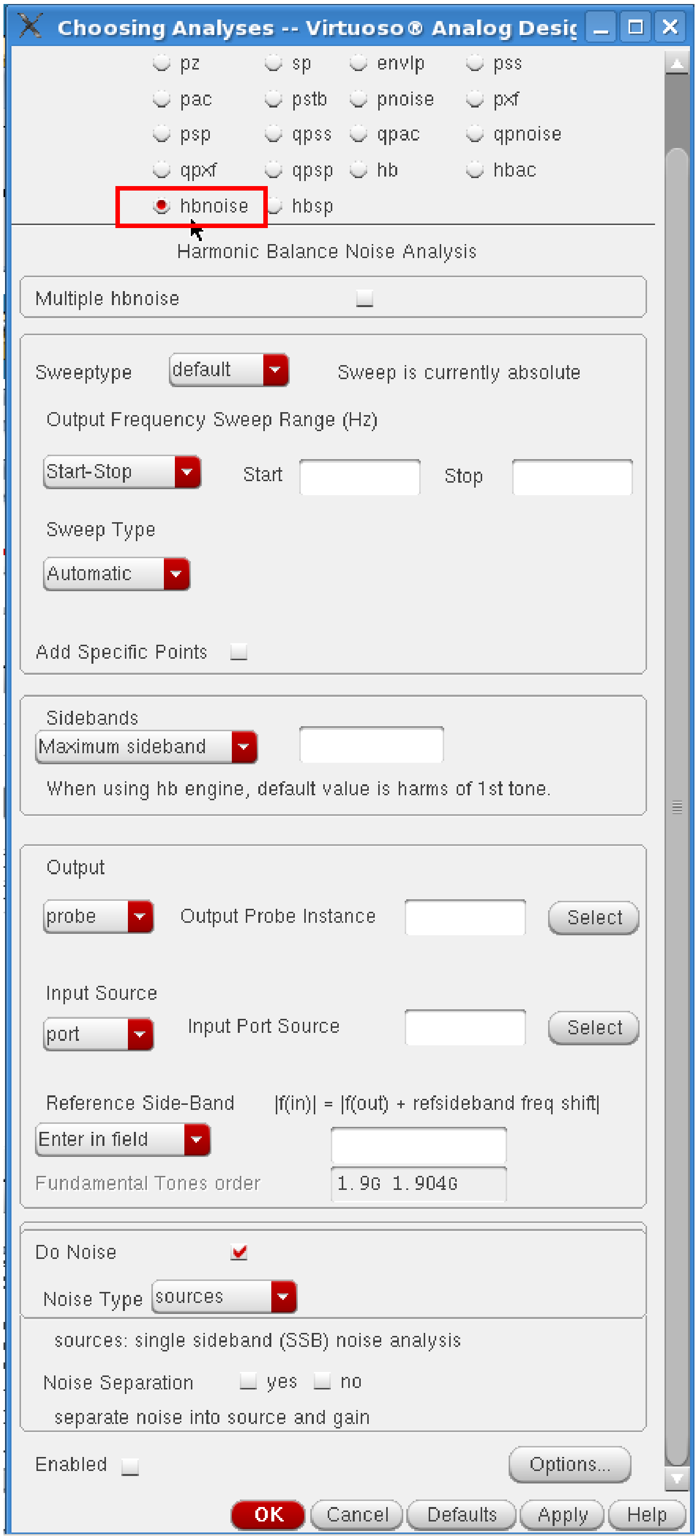
4. Click *Apply*.

Setting up hbnoise to measure noise figure

Hb is used to capture the large signal behavior. Hbnoise analysis follows, and is used to measure the noise performance. Because prf is being swept in hb, the noise figure can be measured as a function of input power (prf).

1. In the *Analysis* section of the Choosing Analyses form, select *hbnoise,* as shown below.

Figure A-350 hbnoise Choosing Analyses Form



Discussion

In hbnoise, the output frequency is `fout=200K`. The LO is at `1.9GHz`. The RF signal frequency is `1.9002G`, and calculated as follows:

`fin=fout+1*flo +0*frf1=200K+1*1.9G+0*1.904G=1.9002G`

The LO and one RF signal are applied in HB. The second RF signal is set to zero. The blocker frequency is set to `1.904G` at the input port.

When the RF signal gets large, it introduces nonlinearity to the system, and that causes mixing of additional noise frequencies with the RF input, thus adding more noise at the output of the system. In this case, noise `200KHz` above and below the LO harmonics of `1.9GHz` will mix to `200KHz` at the output, and noise `200KHz` above and below the RF harmonics of 1.`904HGz` will also mix to `200KHz` at the output.

The noise figure can be measured as a function of the blocker power `prf`.

Note: The number of significant digits is set to `4`. This is the default netlisting resolution for all numerical values from the ADE choosing analyses form. This is sufficient for this design.

However, in your particular design, you may need to increase it to a higher number (6 for example) to see all of the significant digits in the *Reference Sideband* field. If you do not set this, the frequency value will be truncated.

Type the following in CIW:

```
aelPushSignifDigits(6)
```

If you consistently need more than `4` significant digits in your forms, this can also be added to your `.cdsinitfile` so that it is set automatically every time you start ADE. In the workshop database, you can see in the `.cdsinit` file that this value has been set to `10`.

Fill the hbnoise form as follows:

1. Leave the *Sweeptype* set to default (*absolute*).
2. Set the *Output Frequency Sweep Range (Hz)* to *Single-Point*. Click the button just below *Output Frequency Sweep Range (Hz)* and select *Single Point*.
3. Type *200K* in the *Freq* field. This the output frequency (*fout*) from the mixer.
4. In the *Sidebands* section, leave the *Maximum sideband* field blank. When the *Maximum sideband* field is left blank, (strongly recommended for noise analysis) all the noise folding from all the harmonics in the hb analysis are included in the noise result.

5. In the *Output* section, leave the *Output* set to the default value of *probe*. Type in */rif* in the field to the right *Output Probe Instance*. Alternately, you can click the *Select* button and select the output port in the schematic.

   Since *probe* was selected as the output measurement technique for the pnoise analysis, Spectre will subtract any noise contribution by the load resistance from the noise figure calculation. The load resistor noise is still present in the output noise.

6. Ensure that the *Input Source* is set to type *port* (this is the default). Type */rf* for the *Input Port Source.* Alternately, click *Select* to the right of *Input Port Source* and select the input port in the schematic. If you want a noise figure calculation, you must select a port as the input source.

   In the *Reference Sideband* section, choose *Select From List*. The form will expand to show the available frequencies. The reference sideband specifies the RF passband frequency for the noise figure calculation.

7. Under *Select from list*, in the *From (Hz)* and *To (Hz)* fields, you should see *0* to *1e12* by default.

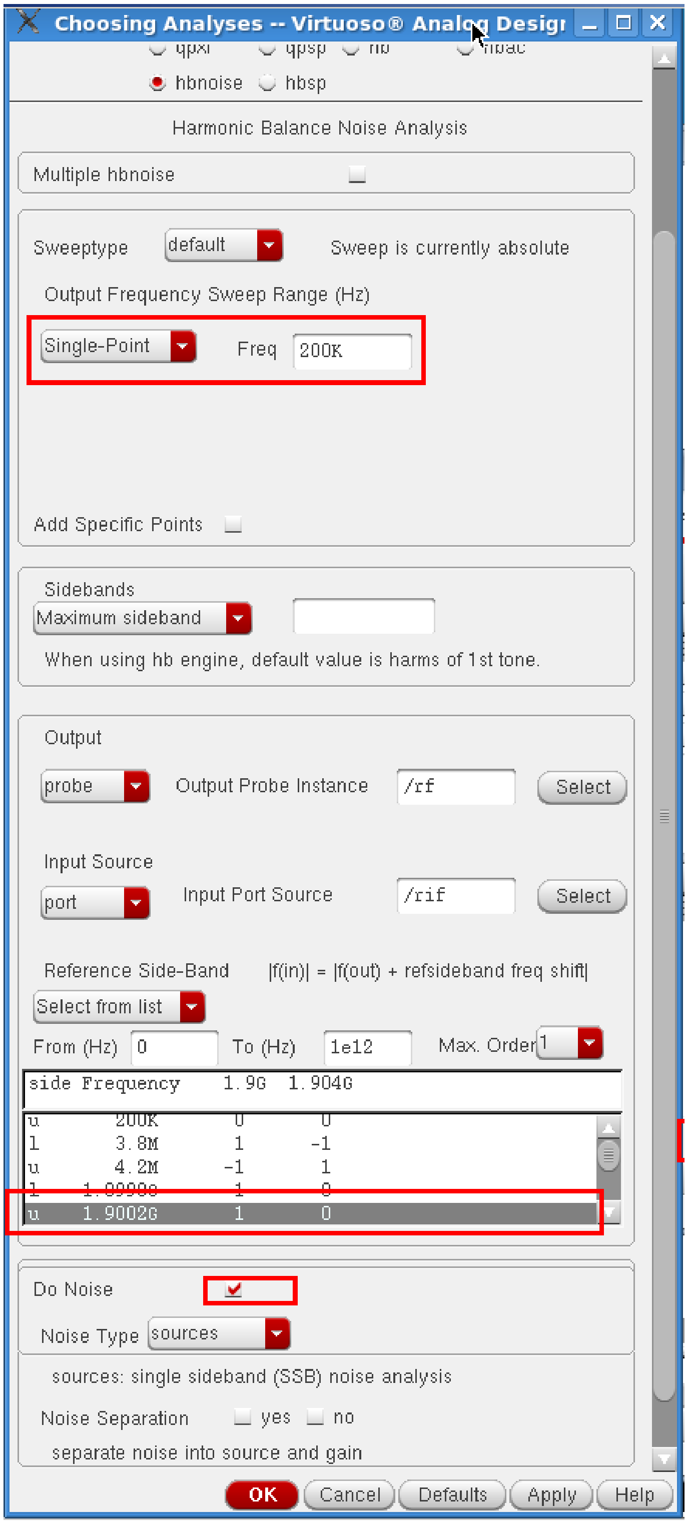
8. Select the *u 1.9002G 1 0* line. This corresponds to the input rf frequency.

9. Ensure that *Do Noise* is selected and that *Noise Type* is set to *sources*. When *Noise Type* is set to *sources*, the output noise calculated by hbnoise is the average noise power that is present with both the LO and RF signals applied in the circuit. Average noise power is also called RMS noise power. It is the true heating power that would be present in the load resistor.

10. Leave the rest of the form set to the default values.

    The *Choosing Analyses* form should look like the following:

Figure A-351 hbnoise Choosing Analyses form

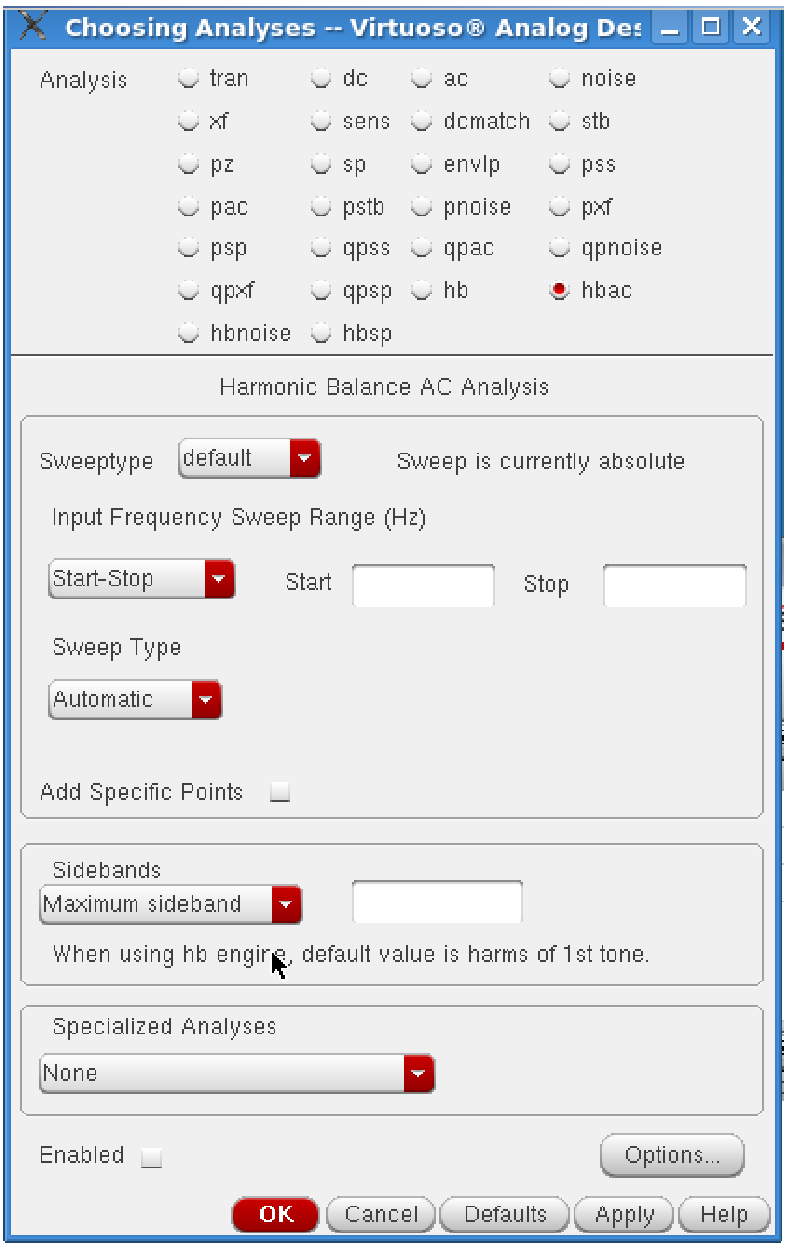


11. Click *Apply*.

Set up HBAC to measure conversion gain.

1. In the *Choose Analyses* form, select *hbac*. The hbac *Choosing Analyses* form is displayed, as shown below.

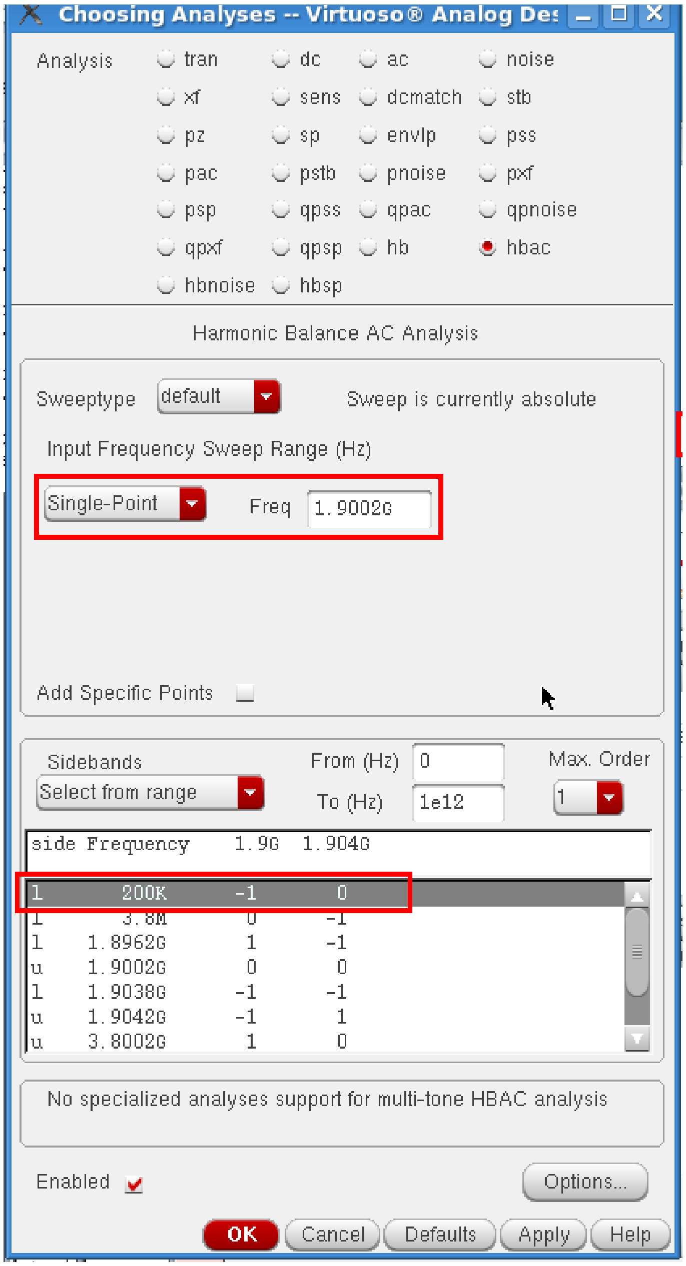
Figure A-352 hbac Choosing Analyses Form



Fill in the hbac Choosing Analyses form as follows:

2. Set the *Input Frequency Sweep Range (Hz)* selection to *Single-Point.*
3. Type *1.9002G* in the *Freq* field. This is the input frequency.
4. In the *Sidebands* section, choose *Select from range*.
5. Under *Select from range*, in the *From (Hz)* and *To (Hz)* fields, you should see *0* to *1e12,* by default.
6. Select the *l 200K -1 0* line. *200K* is the output frequency from the mixer. This will measure the conversion gain.
7. The *Choosing Analyses* form looks like the following:

Figure A-353 The hbac Choosing Analyses form



8. Click *OK*.

Run the simulation

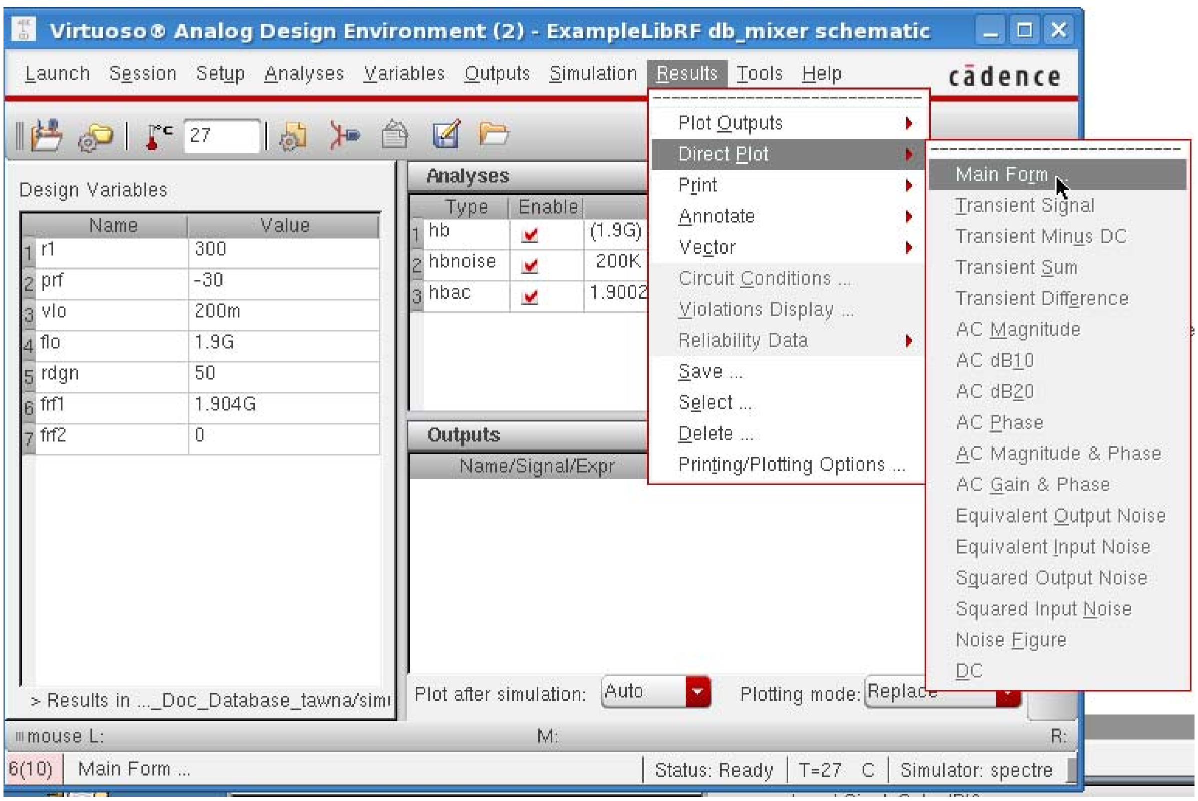
1. Click the green arrow icon on the right side of the ADE L window or in the Schematic window. . The simulation runs.

This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the the status window.

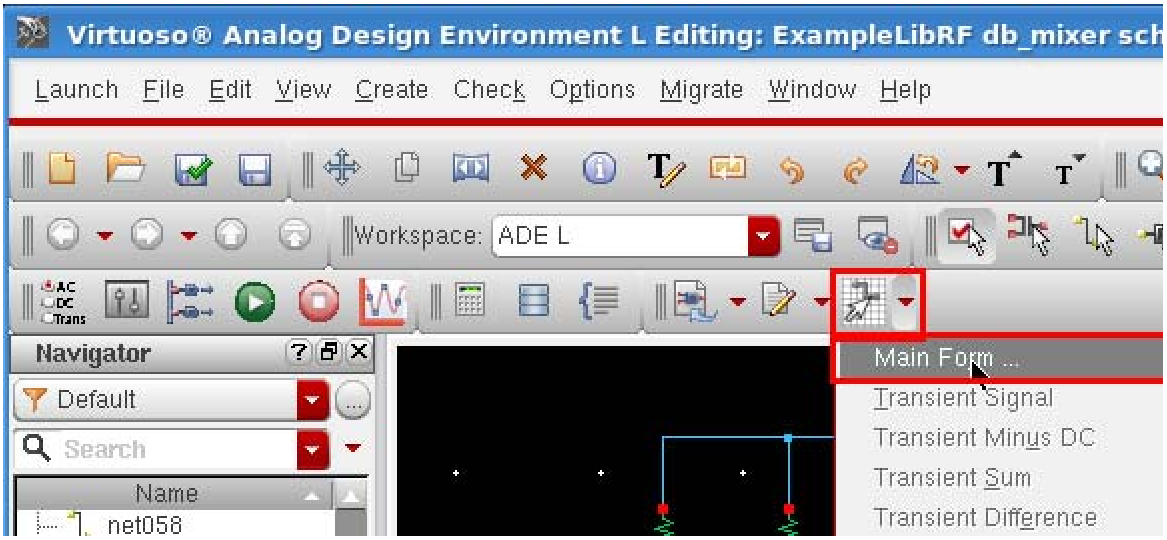
Plot the 1dB Gain Compression point

1. In the Analog Design Environment, choose *Results - Direct Plot - Main Form*.

Figure A-354 Calling up Direct Plot Form

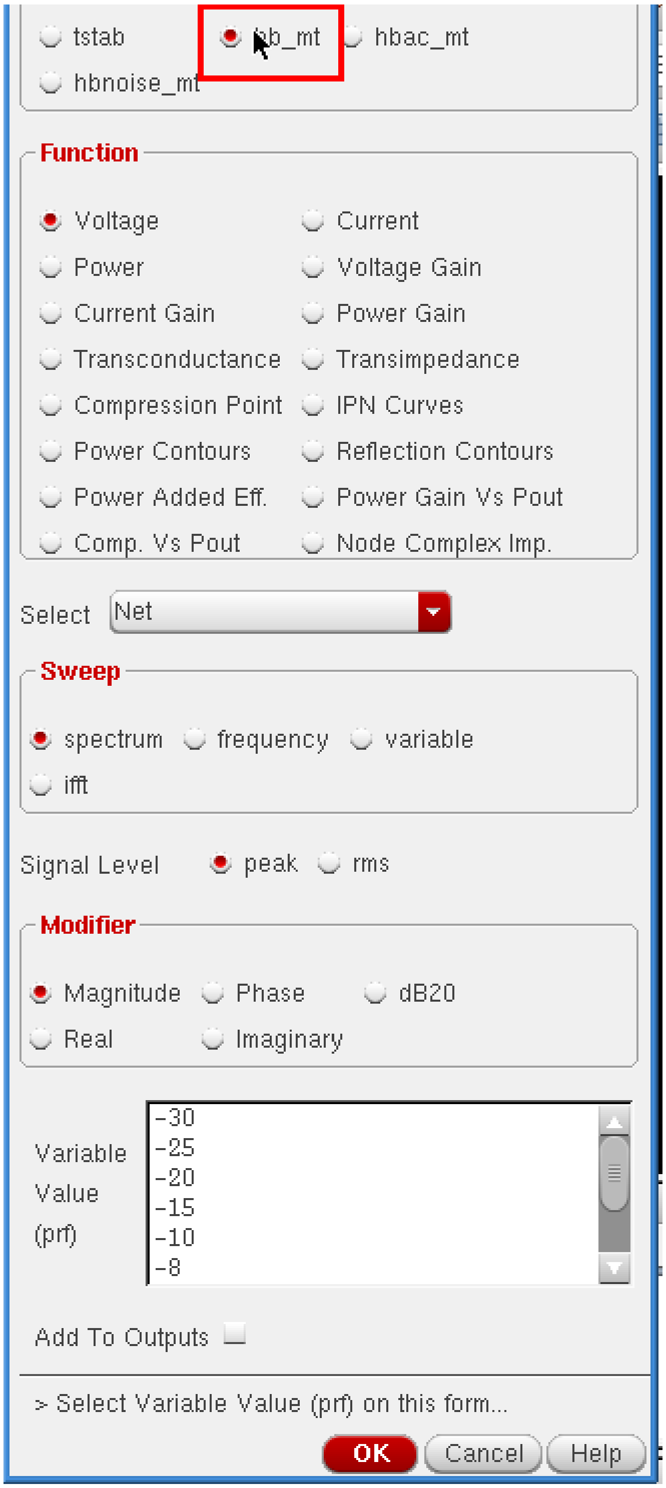


The *Direct Plot Form* is displayed. Alternately, you can press the *Direct Plot* icon in the schematic window,as shown below.



2. In the *Direct Plot Form*, select *hb_mt*. *hb_mt* refers to multitone harmonic balance.

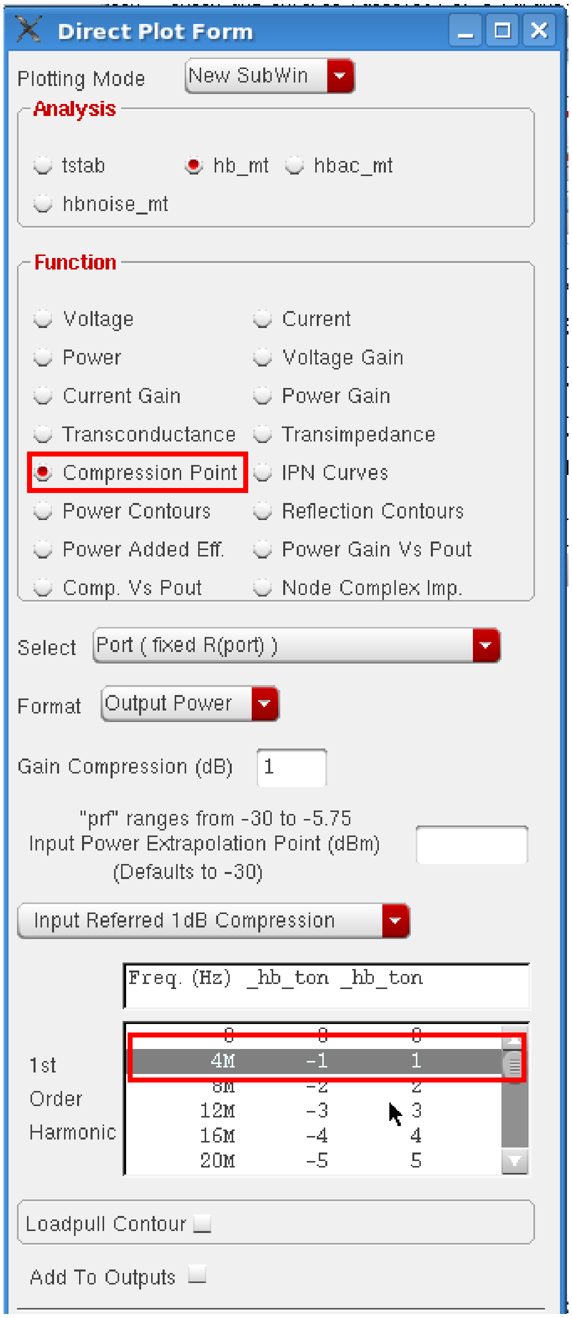
Figure A-355 Direct Plot Form



3. Select *Compression Point* in the *Funciton* section.
4. Ensure that *Port(fixed R(port))* from the *Select* drop-down list and *Output Power* is selected from the *Format* drop-down list.
5. Select the *4M -1 1* term in the *1st Order Harmonic* section.

   The *Direct Plot Form* should look like the following:

Figure A-356 Direct Plot Form hb_mt



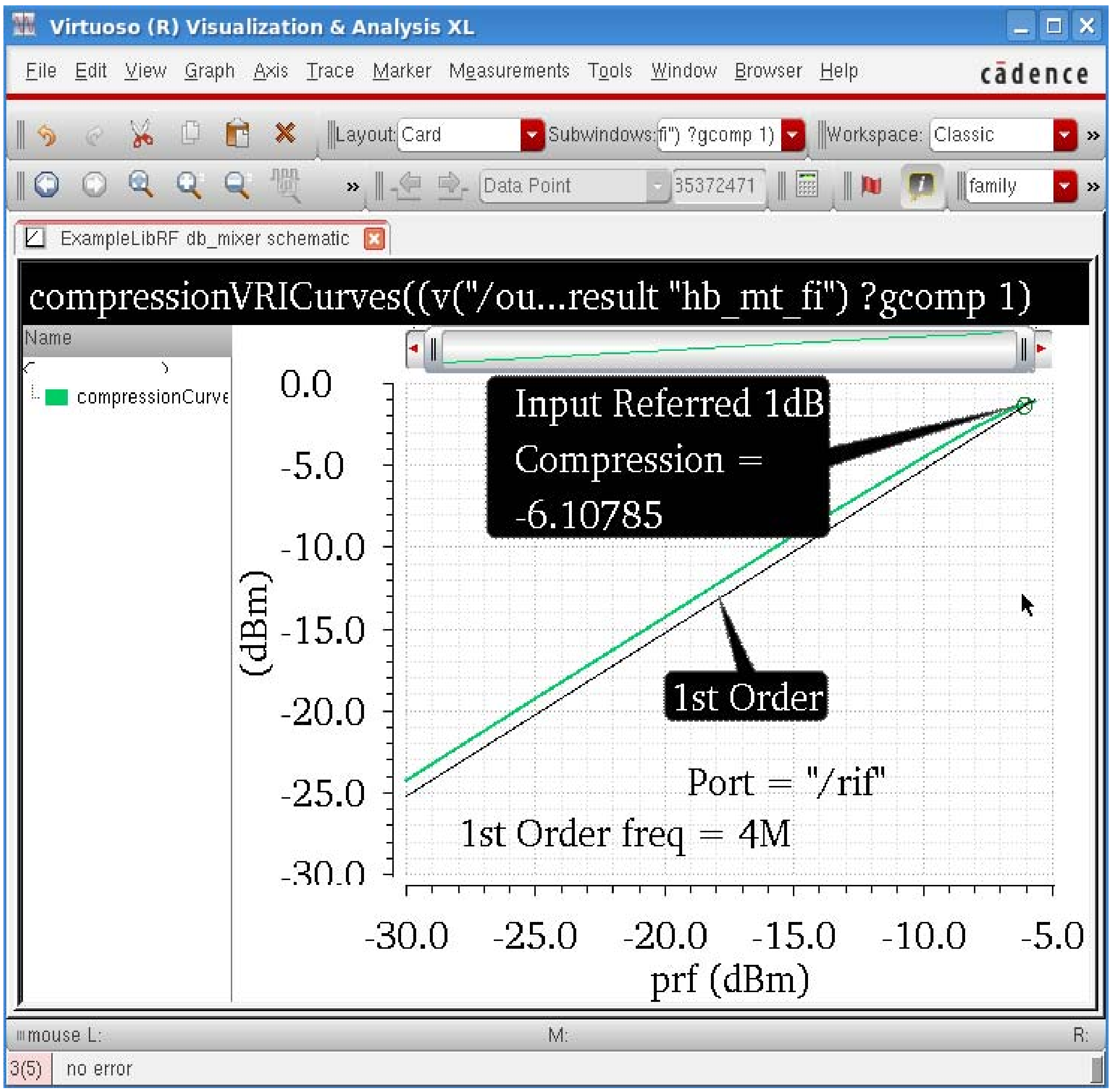
6. In the schematic, select the signal source just to the left of the *Output* label.

   The plot of Output Power vs. prf is displayed.

To move the label, click to select the 1dB Compression point label, , and hold and drag the Compression Point label to the desired location. (First, select a random point in the graphics area to deselect the marker intercept point. Otherwise, the label will not move.)

Note the compression point. The input-referred compression point is about *-6.1dB.*

Figure A-357 Input Referred 1dB Compression Point



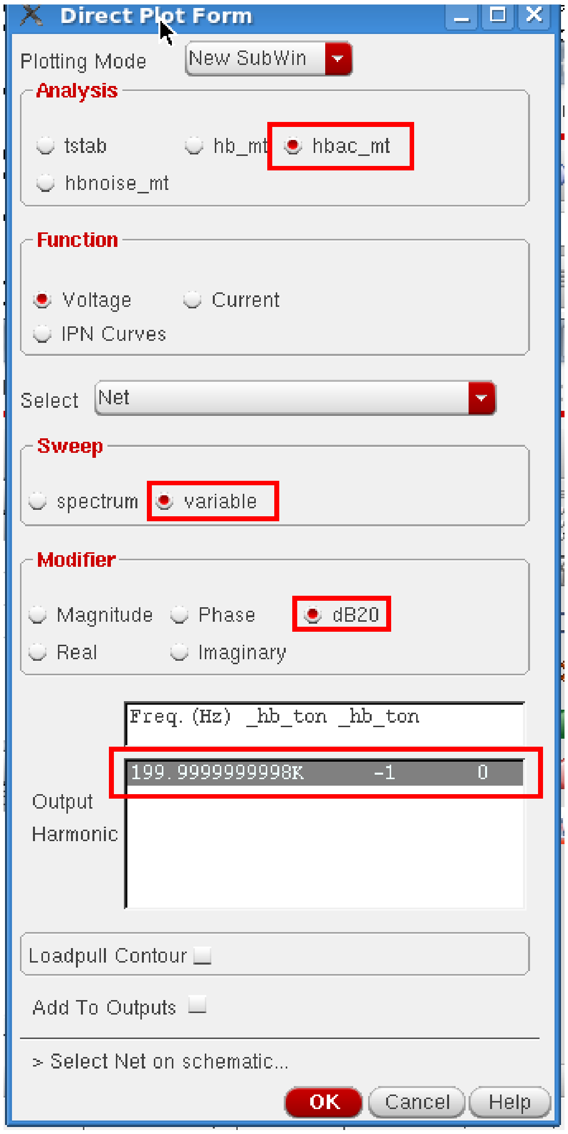
The expression used to plot the Compression curves is shown above in the waveform window.

Plot the RF-IF conversion gain as a function of RF (Blocker) power

7. In the *Direct Plot Form*, set *Plotting Mode* to *New SubWin*.
8. Inthe *Analysis* section, select *hbac_mt*.
9. In the *Function* section, select *Voltage.*
10. *Select Net* (this is the default).
11. In the *Sweep* section, select *variable*.
12. In the *Modifier* section select *dB20*.
13. In the *Output Harmonic* section, select the *199.9999999998K -1 0* line.

    The *Direct Plot Form* should look like the following:

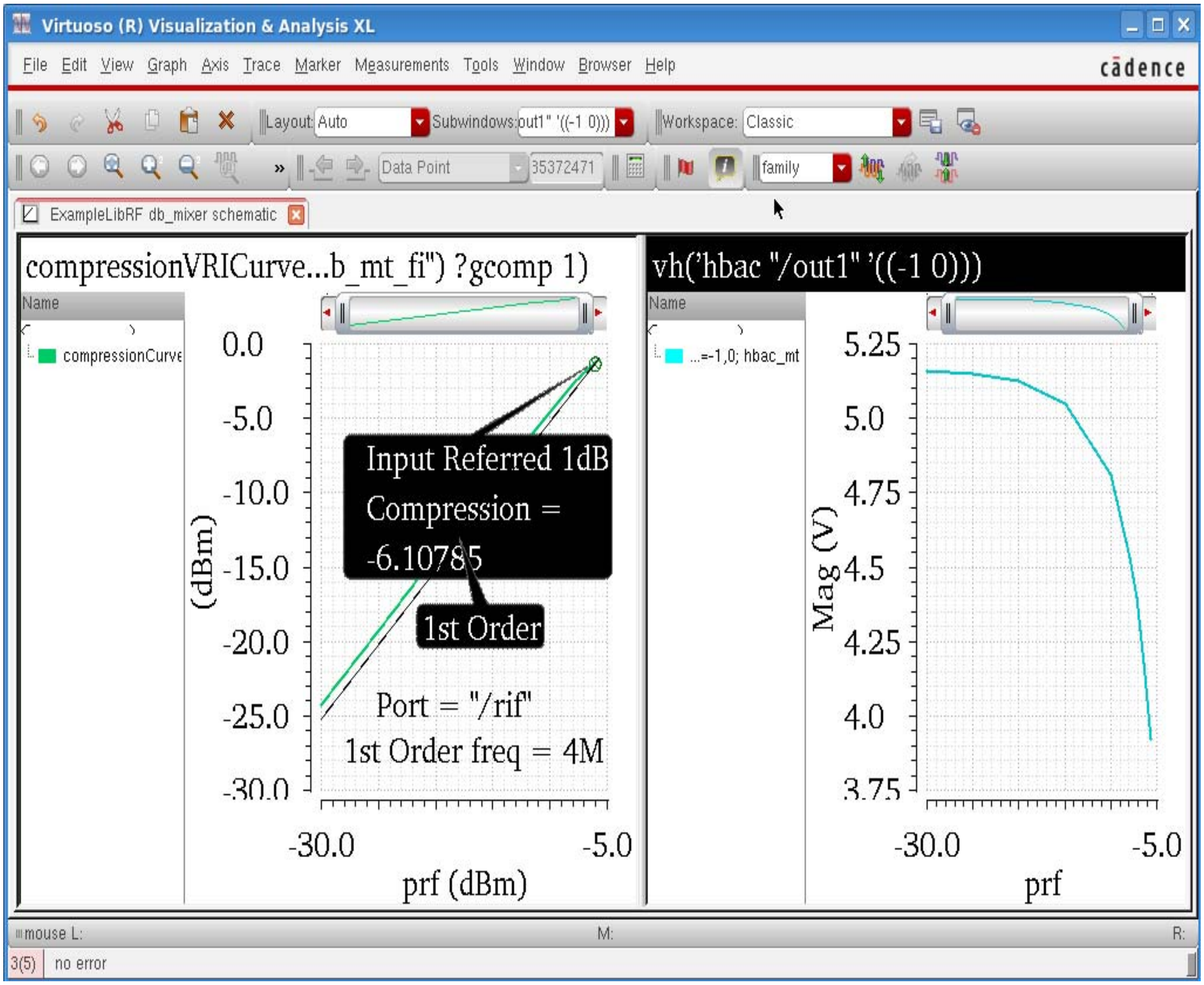
Figure A-358 Direct Plot Form hbac_mt



14. In the schematic, select the *out1* net near the *Output* label.

The conversion gain is plotted. As the blocker power goes up, the conversion gain drops.

Figure A-359 1dB Compression Point and Conversion Gain



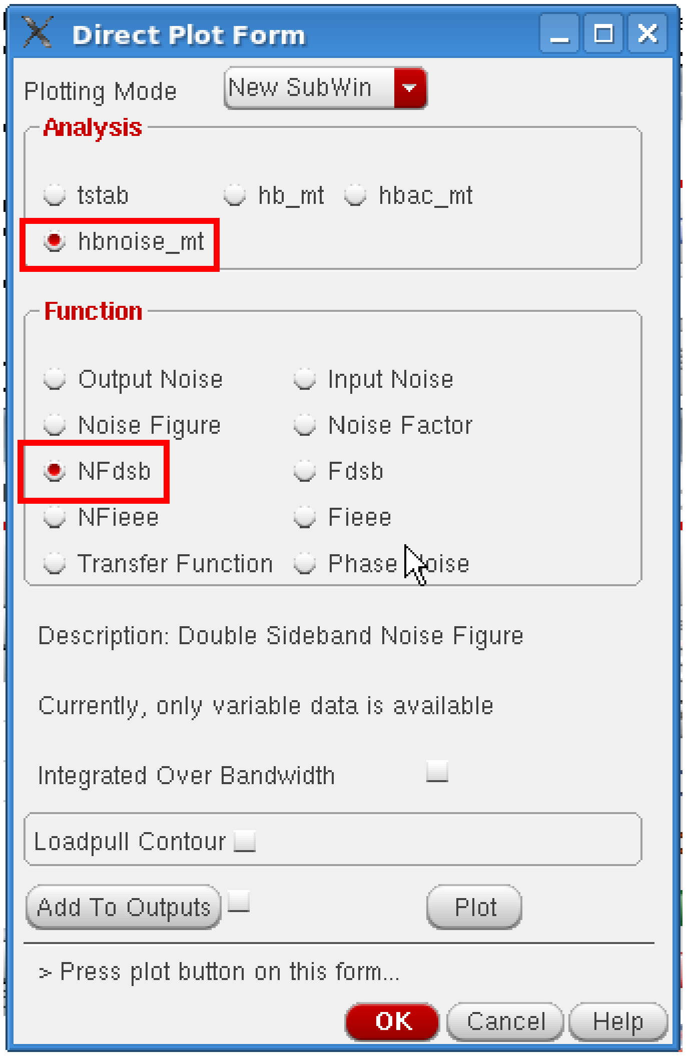
Plot the noise figure

15. In the *Direct Plot Form*, select *hbnoise_mt* in the *Analysis* section.

16. In the *Function* section, select *NFdsb*.

The *Direct Plot Form* should look like the following:

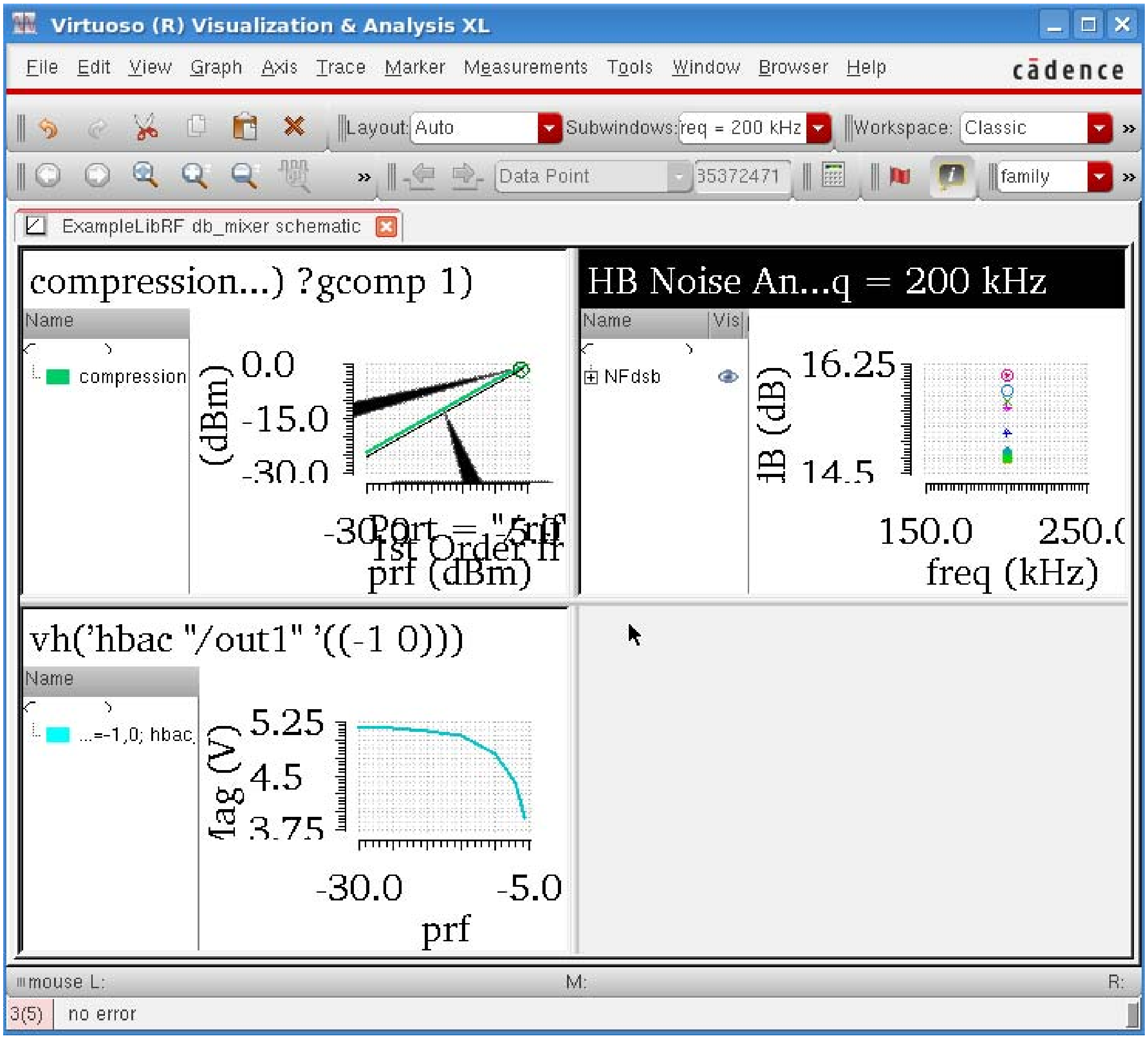
Figure A-360 Direct Plot Form Plotting NFdsb



17. Click *Plot*.

The noise figure plot is added to the previous plot in a new sub window, as shown below.

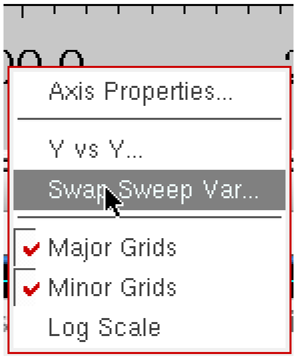
Figure A-361 1dB Compression Point, Conversion Gain, and Double Sided Noise Figure



Make the noise figure result more readable.

18. In the most recent sub window, move your mouse cursor over one of the numbers on the X Axis, click the right mouse button, and select *Swap Sweep Var* from the context menu, as shown below.

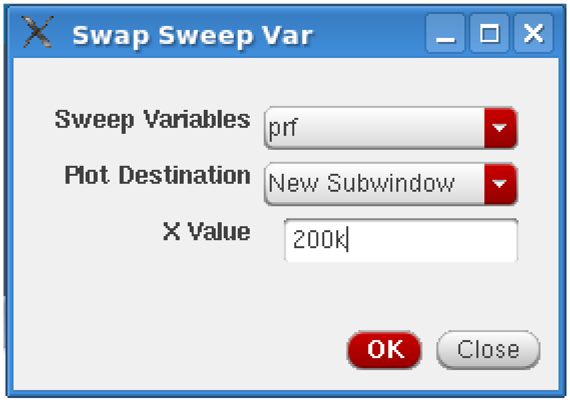
Figure A-362 X-Axis Properties Form



The Swap Sweep Var form is displayed.

19. In the *Swap Sweep Var* form, select *prf* from the *Sweep Variable* drop-down list. Leave *Plot Destination* set to *New Subwindow.* Enter *200k* in the *X Value* field. Note that for this menu, you must enter *k* , not *K*.

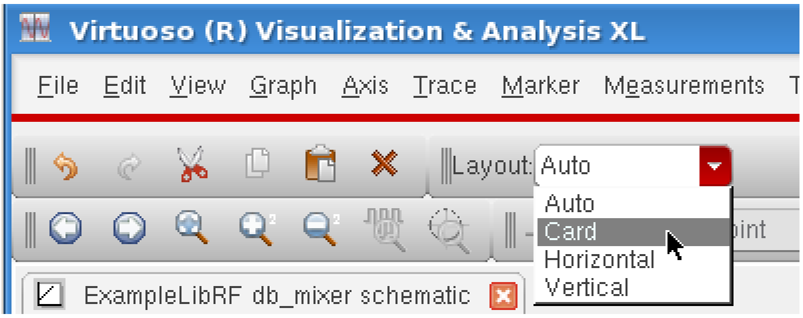
Figure A-363 Swap Sweep Variable Form



20. Click *OK.*

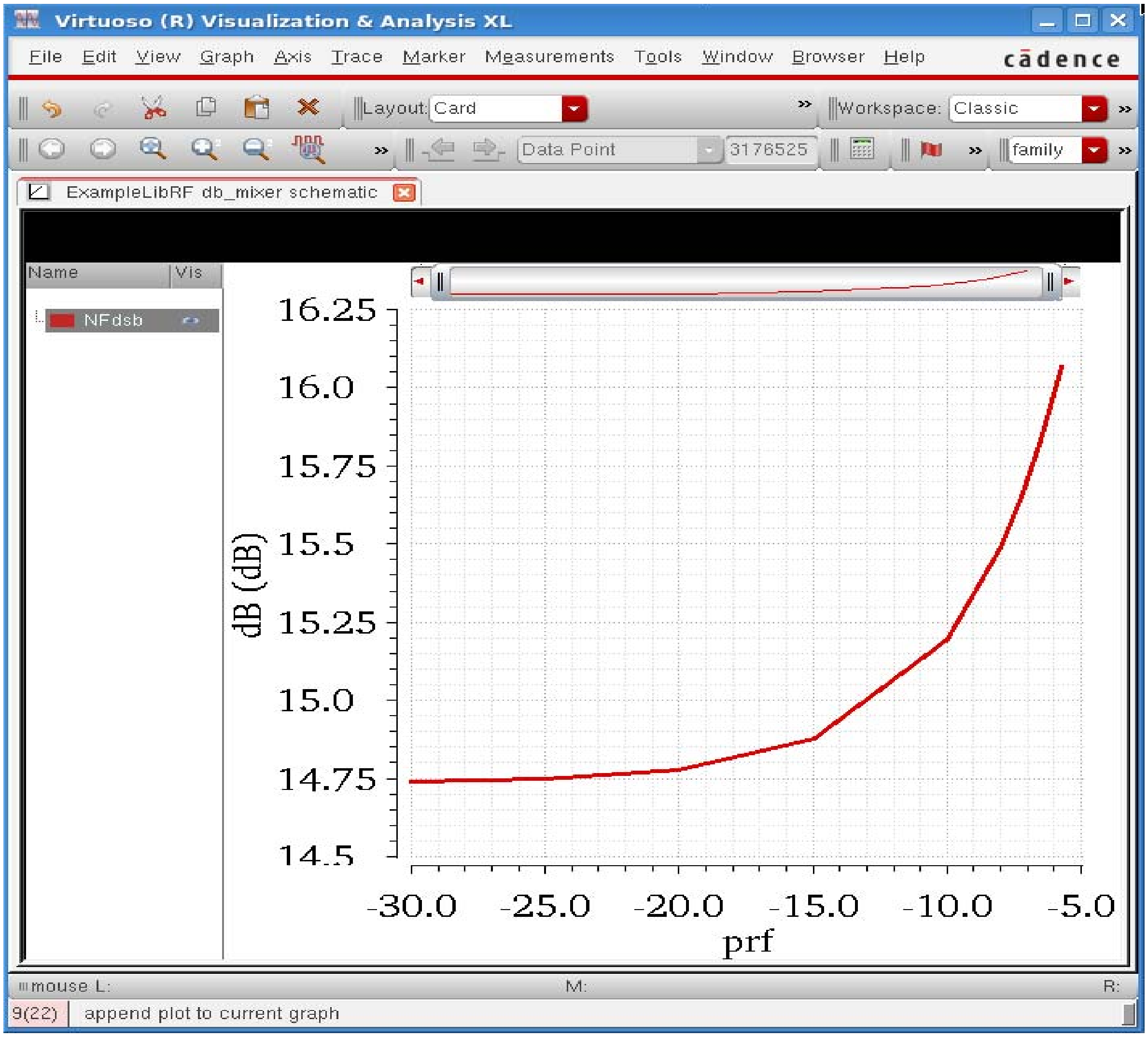
21. The waveform window is now fairly difficult to read. In the waveform window, select *Card* from the *Layout* drop-down list. The *Card* mode displays one subwindow at a time so that the display area becomes larger. You can switch which subwindow is displayed at any time.

Figure A-364 Selecting Card Mode Layout



The waveform window changes to Card mode and the Noise Figure plot is much easier to read, as shown below.

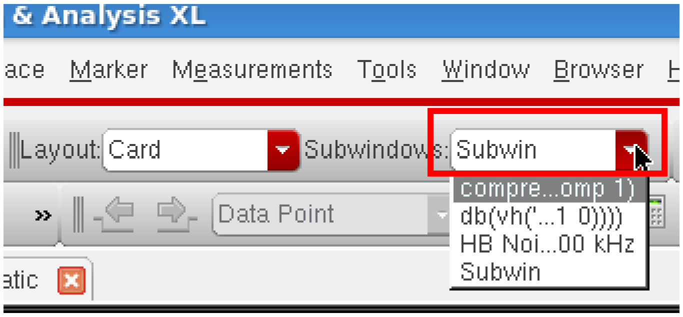
Figure A-365 Noise Figure



Note the trend in conversion gain and noise figure. As expected, the noise figure goes up and the conversion gain goes down with increasing RF blocker power (prf).

22. To view other subwindows, select another plot to view from the *Subwindows* drop-down list.

Figure A-366 Selecting Subwindow to View a Different Plot in Card Mode



Clean up the screen for the next set of measurements.

23. Close the Analog Design Environment window by choosing *Session - Quit.*

24. Click *No* in the *Save State* window.

25. In the Schematic window, choose *File - Close All.*

Summary:

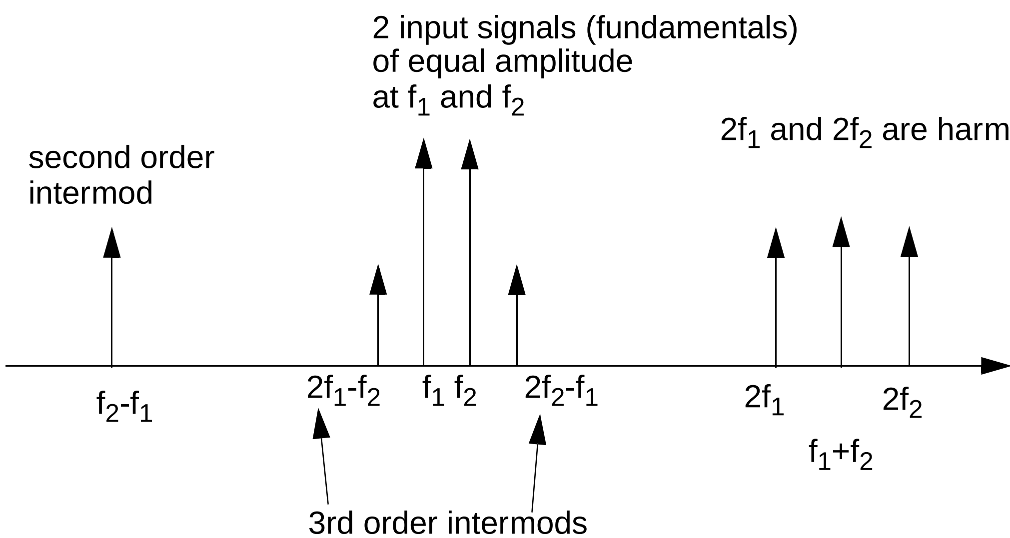
In this section, you measured 1dB compression point, conversion gain, and desensitization with an RF blocking signal present using hb, hbnoise, and hbac analyses.

In the next section, you will measurethe Third-Order Intercept using Harmonic balance.

Third-Order Intercept measurement with HB

In the frequency domain, third-order products are the intermodulation distortion products between one of the fundamental signals and the second harmonic of the other signal.

Figure A-367 Intermodulation Distortion Products



The presence of two or more tones in a nonlinear circuit generates intermodulation products. Many of the spurious tones are out-of-band and cause no problems. However, the third-order tones are nearest to the fundamental and are likely to fall in-band. They add distortion to the output signal. IP3 is a metric or figure of merit for linearity that is used to describe the intermodulation performance of a mixer.

To do the Third-Order Intercept measurement, in this section, both the RF and the LO signals are applied to the circuit. Using the transient analysis is impractical due to lengthy runtimes and spectral calculation times. The harmonic balance engine is used to overcome these disadvantages. *This setup has the advantage of being able to see all of the harmonics on the IF, but it takes longer to run than rapid IP3 for a small-signal measurement. If you want a small-signal IP3, a faster way to approach the problem is shown in the next section.* Using HB by itself for an IP3 measurement is typically only required for transmit mixers and power amplifiers. It is shown here so you can compare two different methods of obtaining IP3: Using three tone hb vs. rapid IP3.

Opening the db_mixer Mixer Circuit in the Schematic Window

1. In the CIW, choose *File – Open*.

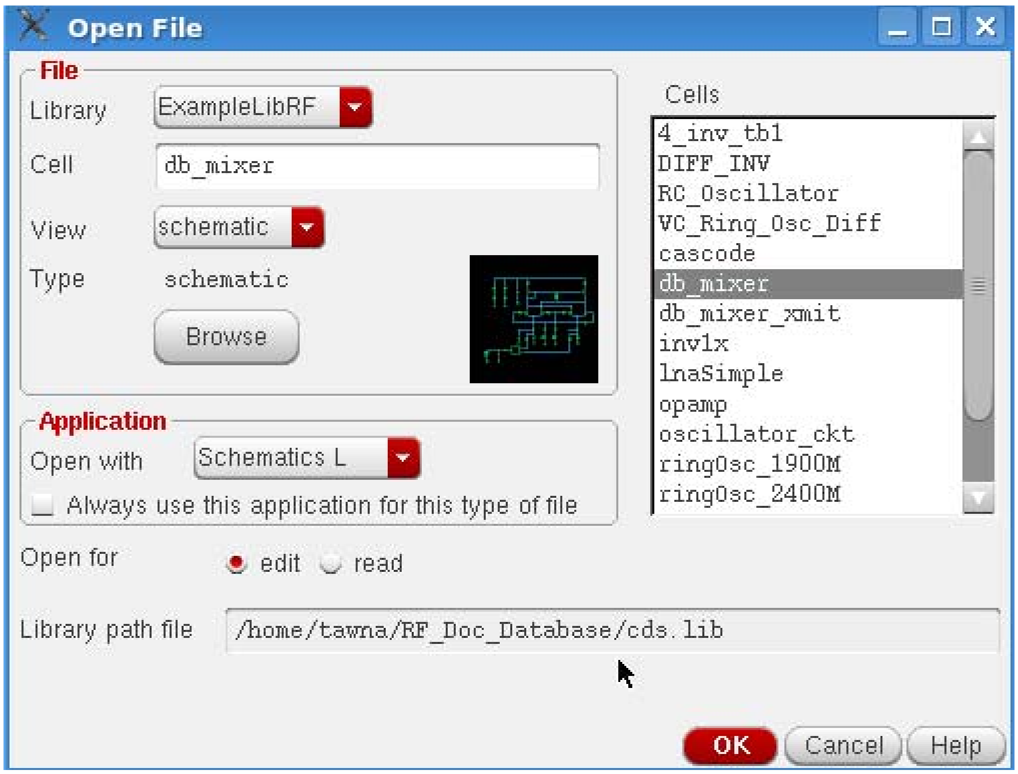
   The *Open File* form is displayed.

2. In the *Open File* form, choose *ExampleLibRF* from the *Library* drop-down list. Choose the editable copy of the *ExampleLibRF* that you created as described earlier in this manual.

3. Choose *db_mixer* in the *Cells* list box.

   The completed Open File form appears like the one below.

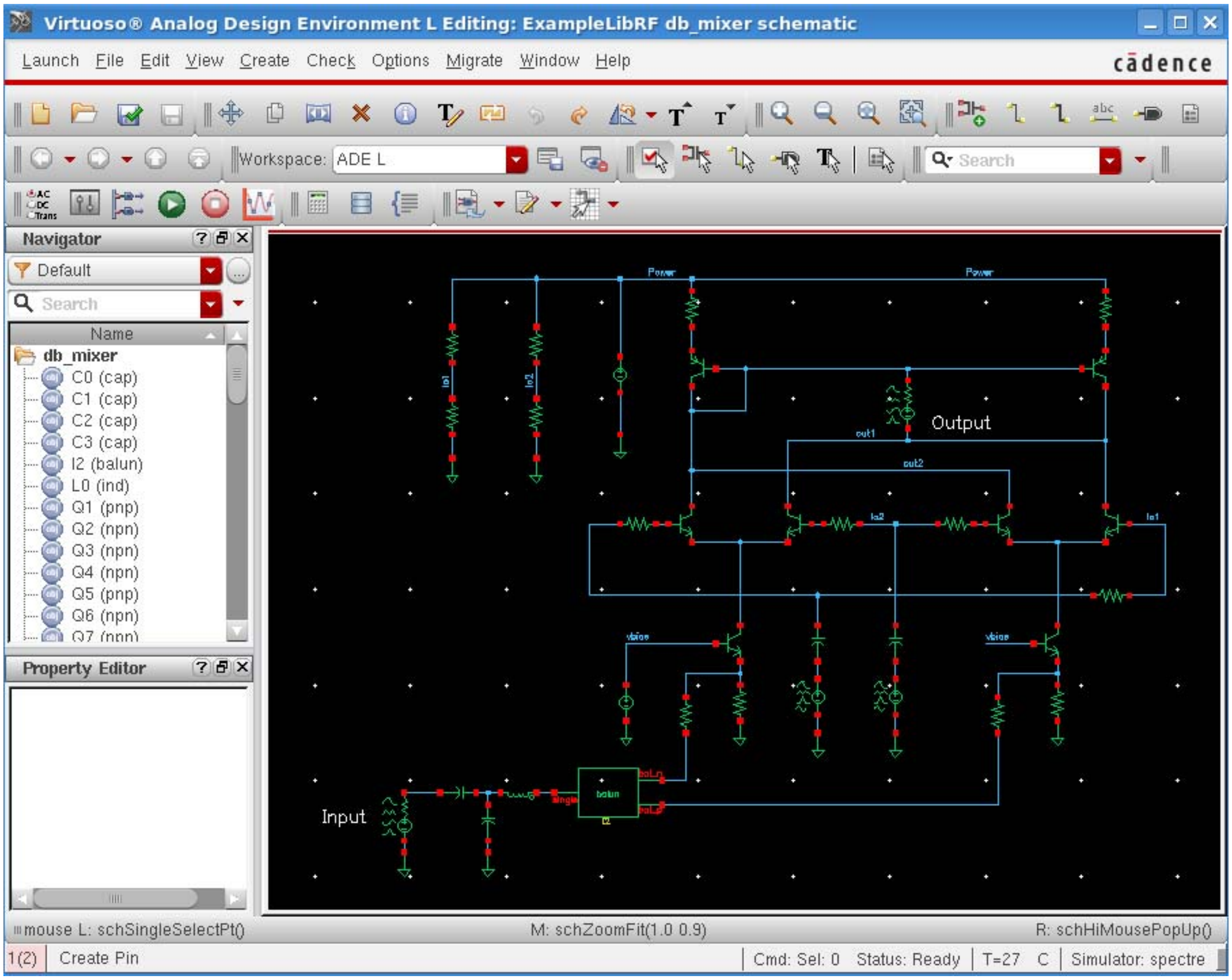
Figure A-368 Open File Form



4. Click *OK*.

   The Schematic window for the *db_mixer* mixer is displayed.

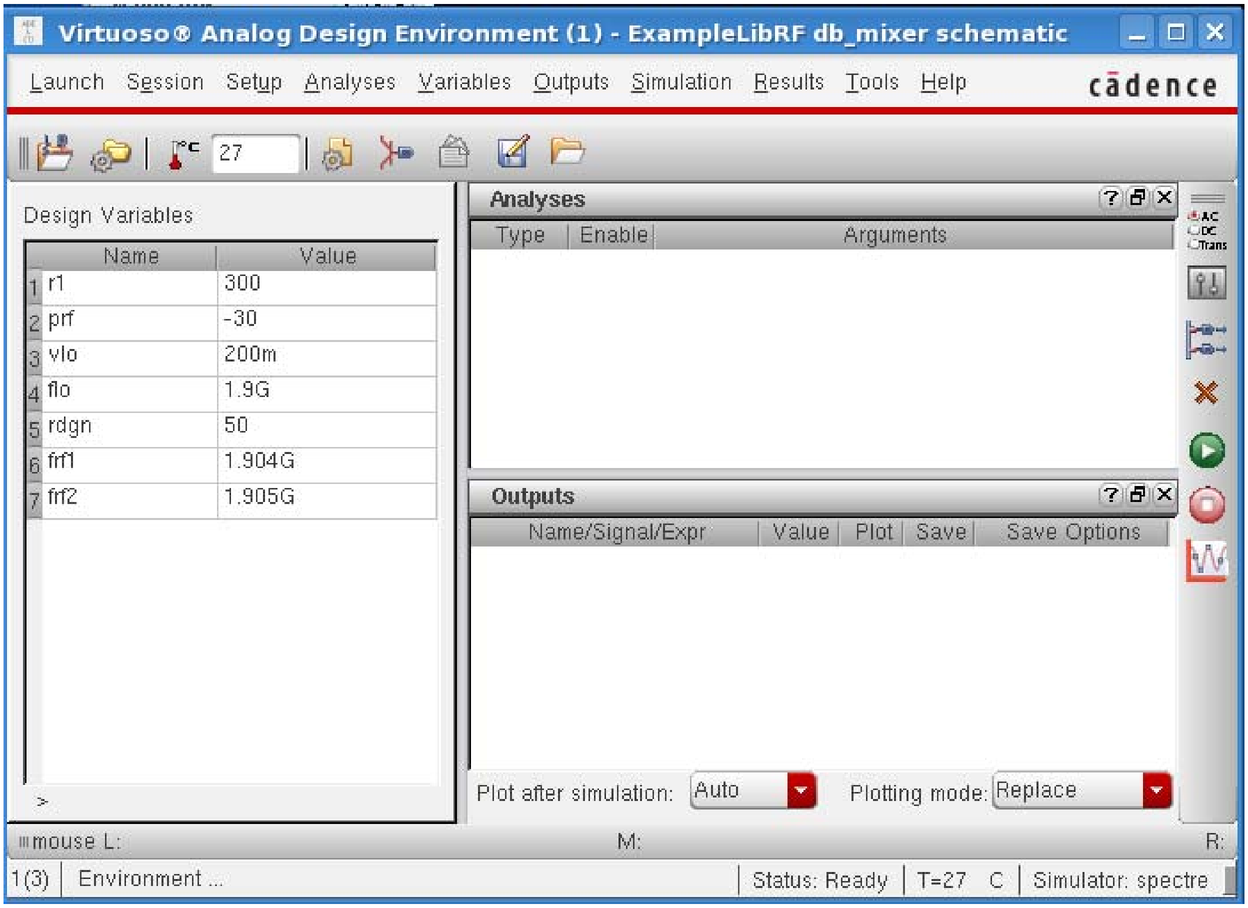
Figure A-369 db_mixer Schematic



5. In the Schematic window, choose *Launch – ADE L*.

The Virtuoso Analog Design Environment window is displayed, as shown below.

Figure A-370 Virtuoso Analog Design Enviroment Window



Choosing Simulator Options

6. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

The Choosing Simulator/Directory/Host form is displayed.

7. Specify the following in the Choosing Simulator/Directory/Host form:

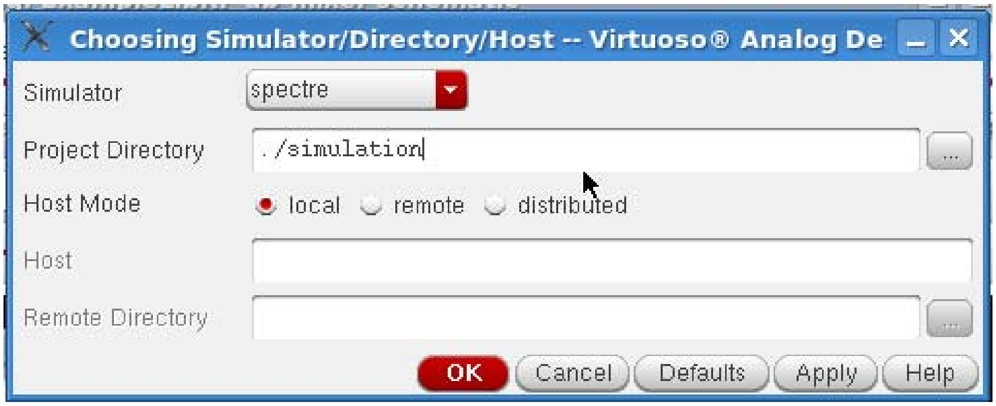
a. Choose *spectre* from the *Simulator* drop-down list.

b. Type the name of the project directory, if necessary, in the *Project Directory* field.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form appears similar to the one below.

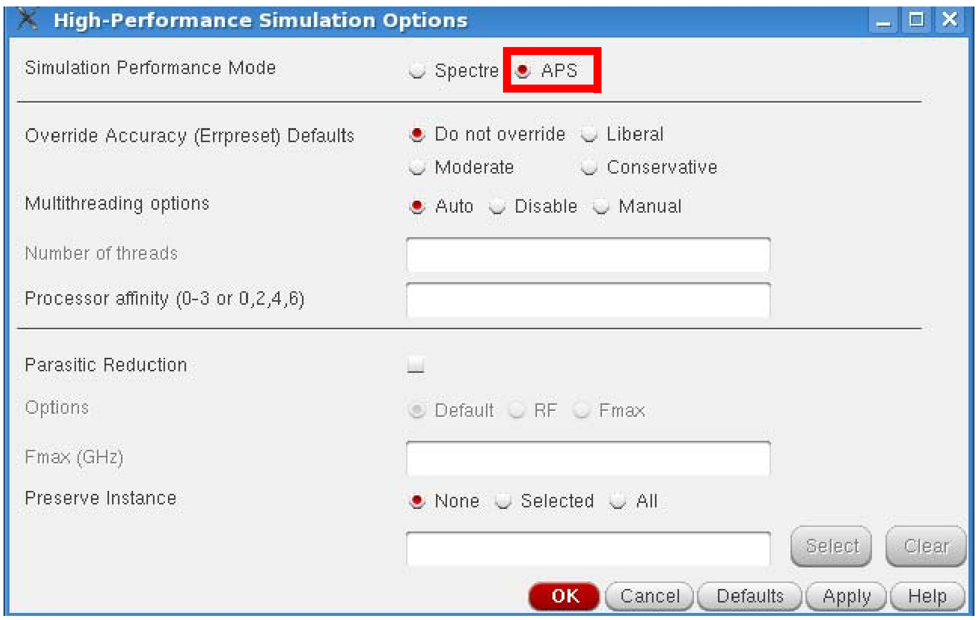
Figure A-371 Choosing Simulator/Directory/Host Form



8. Click *OK* to close the *Choosing Simulator/Directory/Host* form.
9. Set up the High Performance Simulation Options

   In the ADE window, choose *Setup - High Performance Simulation.* The *High Performance Simulation Options* window is displayed, as shown below.

Figure A-372 High Performance Simulation Options Window



In the *High Performance Simulation Options* window, select *APS*. Note that *auto* is selected for *multithreading*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading. For more information, see the *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator User Guide*.

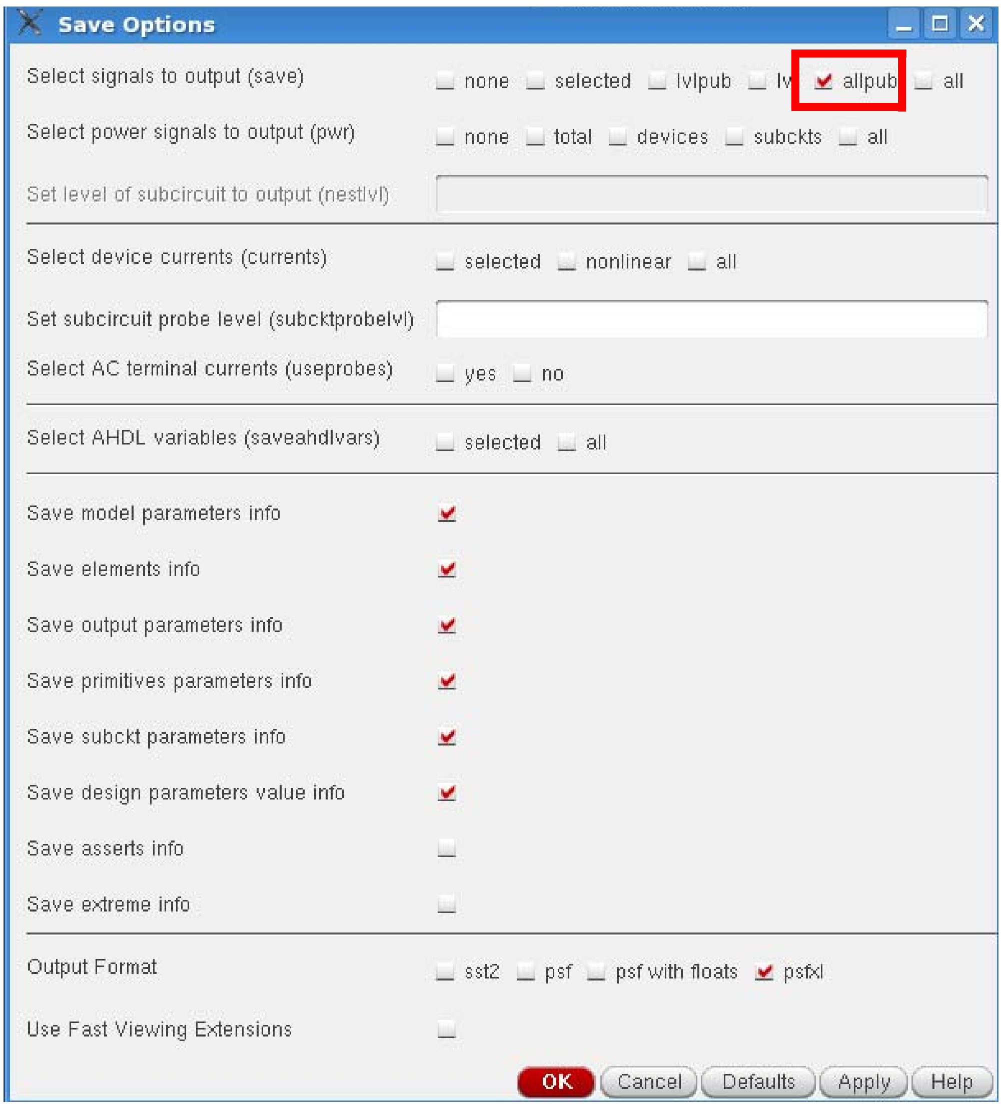
Click *OK*.

10. In the *Virtuoso Analog Design Environment* window, choose *Outputs – Save All*.

    The *Save Options* form is displayed.

11. In the *Select signals to output* section, make sure that *allpub* is selected.

Figure A-373 Save Options Form



This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

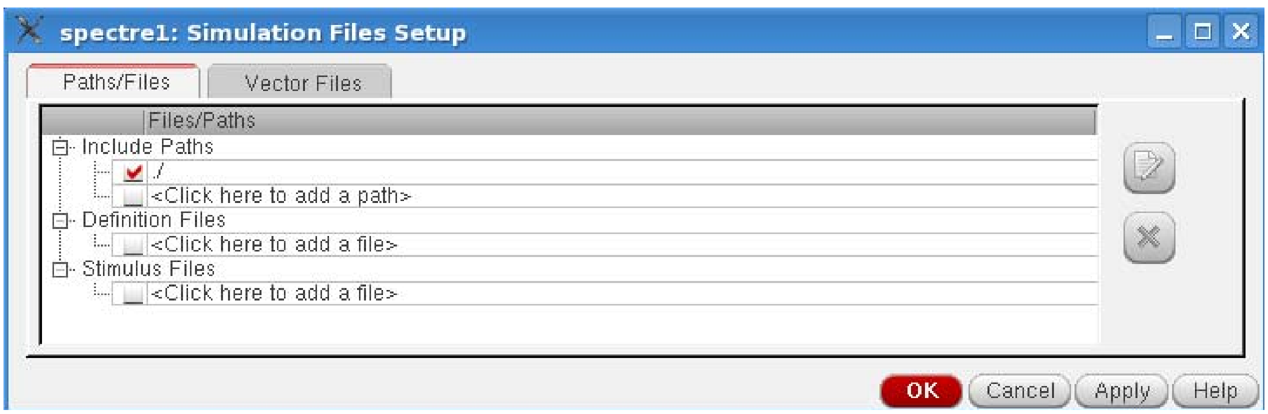
In the *Select signals to output* section, make sure that *allpub* is highlighted. This is the default which saves all node voltages at all levels of hierarchy, but it does not include the node voltages inside the device models.

12. Click *OK* to close the *Save Options* form.

Setting Up Model Libraries

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.* The Simulation Files Setup form is displayed, as shown below.

Figure A-374 Simulation Files Setup Form



2. Ensure that the Include Path is set as shown above.

3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries*.

The *Model Library Setup* form is displayed.

4. In the *Model Library File* field, type the following in the name of the model file:

```
models/modelsRF.scs
```

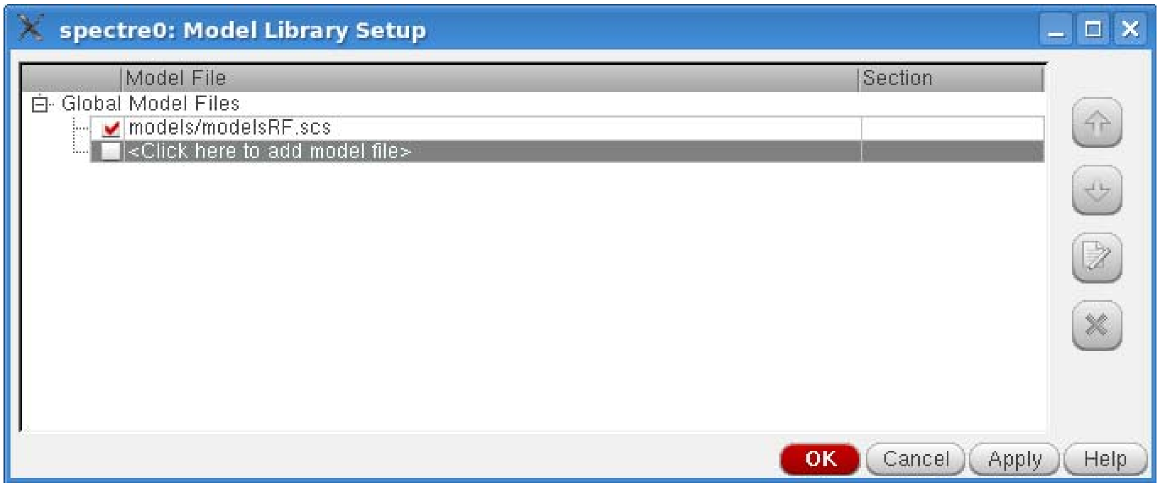
Alternately, you can click the *Browse* button and browse to the `modelsRF.scs` model file.

5. Ensure that the Model File name is selected.

6. Click *Apply*.

The *Model Library Setup* form looks like the following:

Figure A-375 Model Library Setup Form



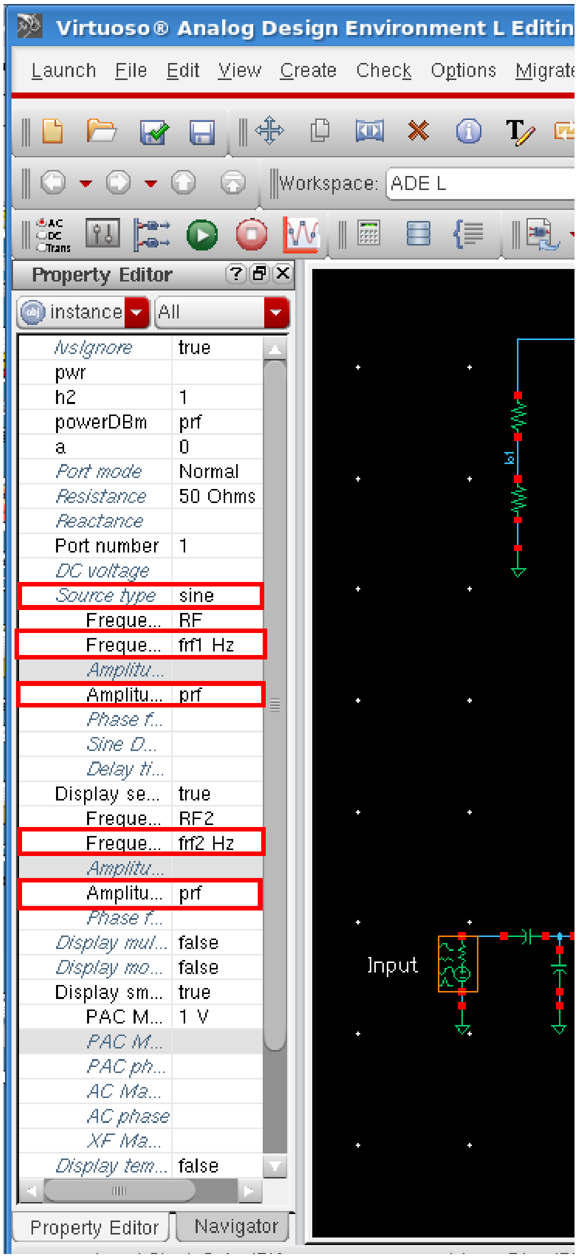
7. Click *OK* to close the *Model Library Setup* form.

Setting Up the Simulation

First view the input port settings.

1. In the schematic, select the RF source (port) just to the right of the *Input* label .
2. Note that the *Property Editor* (left side of schematic) displays the properties of the selected port. For instructions on how to re-size the *Property Editor* window, see the *Virtuoso SchematicEditor L User Guide*.

Figure A-376 Property Editor Showing Properties of Selected Port



3. Use the scrollbar on the right of the form, if necessary, so that you can view all of the properties.

   Note that the *Source type* is set to *sine*. This enables the RF tones.

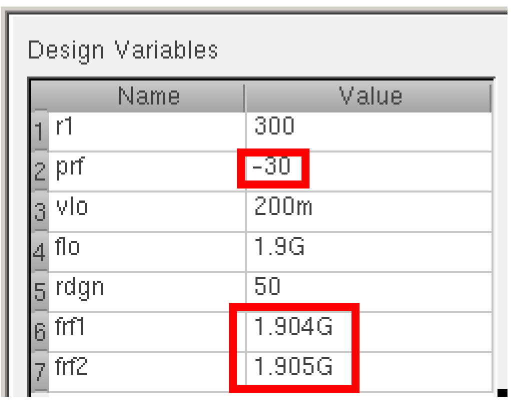
   Note that the first sinusoid's frequency is set to the variable *frf1* and the second sinusoid's frequency is set to the variable *frf2*.

Both sinusoids have the amplitude set to the variable *prf*.

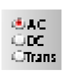
4. In the *Design Variables* section of the ADE window, verify that variables are set as follows:

- ❑ *frf1* to *1.904G*. This is the first RF frequency.
- ❑ *frf2* to *1.905G*. This is the second RF frequency.
- ❑ *prf* to *-30*. This is the input power level for both RF tones.

Figure A-377 Design Variables section

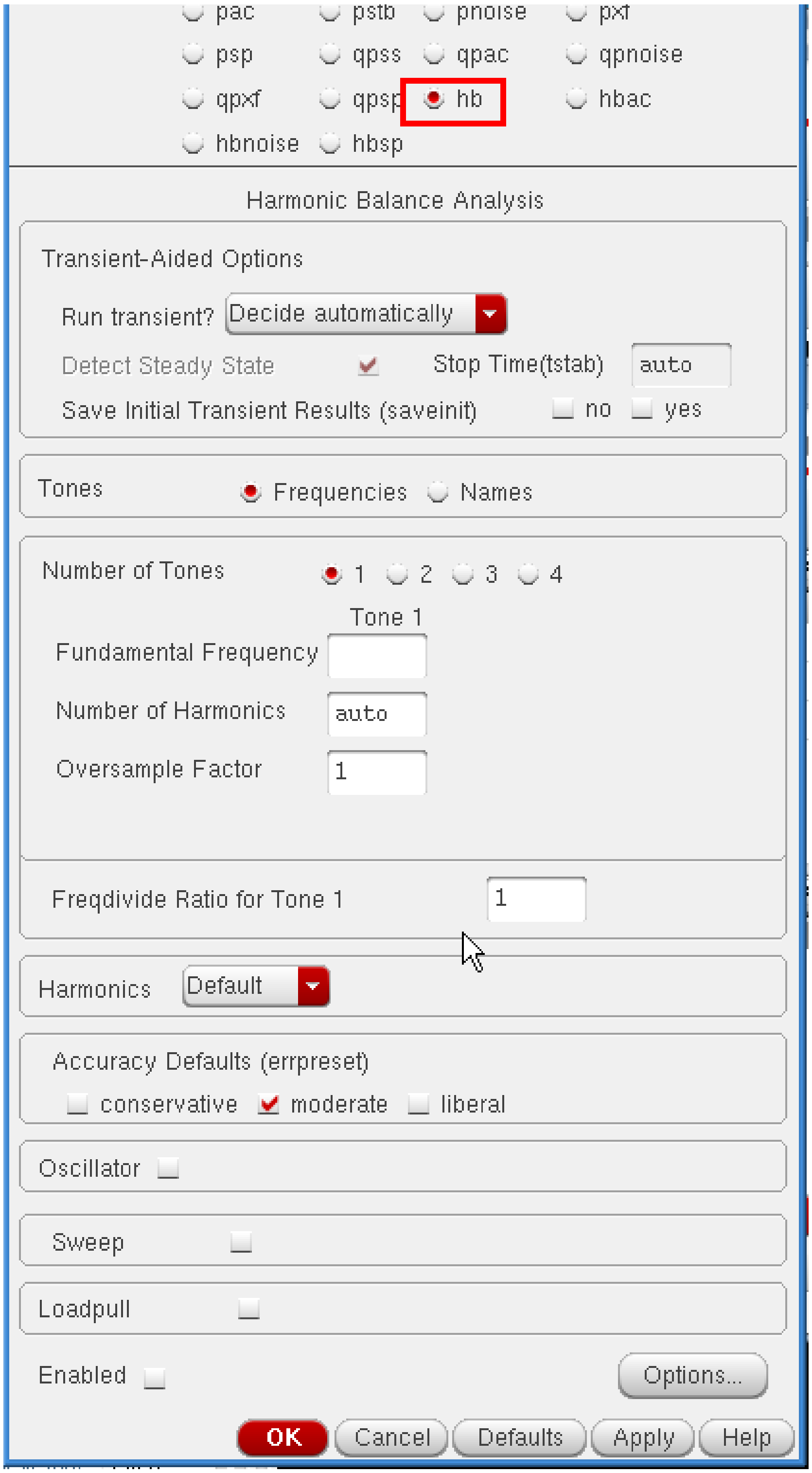


5. Click the *Choosing Analyses* icon on the right side of the ADE L window to open the *Choosing Analyses* form.



The *Choosing Analyses* form is displayed, as shown below.

Figure A-378 hb Choosing Analyses Form



6. In the *Analysis* section, choose *hb*.

   Harmonic balance can set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient* selection in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected, and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

7. In the *Transient-Aided Options* section of the form, leave the settings at their default values, unless otherwise noted.

   a. For *Run transient?* select *Decide automatically.*

      Run transient? will run the LO signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

   b. For *Stop time (tstab), auto* is automatically populated in the field.

      When *auto* is selected for *Stop time*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

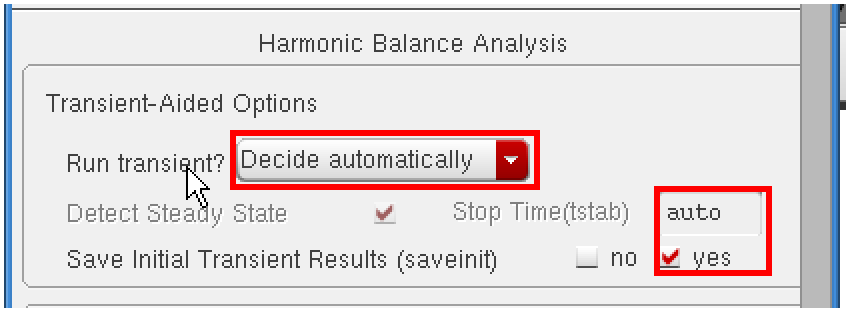
      When *Run transient?* is set to *Decide automatically*, the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

   c. For *Save Initial Transient Results (saveinit)*, select yes.

   During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. Only the signal in *Tone1* is enabled for this. Make sure you set the signal that causes the largest amount of distortion in the circuit as *Tone1*. This is usually the LO signal. At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations.

   All the signals are applied and the simulation is done in the frequency domain. Only the signals,harmonics, and the mixing products are calculated by hb.

Figure A-379 Transient Assisted Harmonic Balance



8. In the *Tones* section, select *Names*. When *Names* is selected, the *Tones* portion of the form expands. All the sources in the top-level schematic are read into the form automatically. Only one signal can have transient assist. When *Tones* is set to *Names*, the signal with *tstab* set to *yes* can have transient assist. Choose the signal that produces the largest amount of distortion for transient assist, in this case, it is LO.

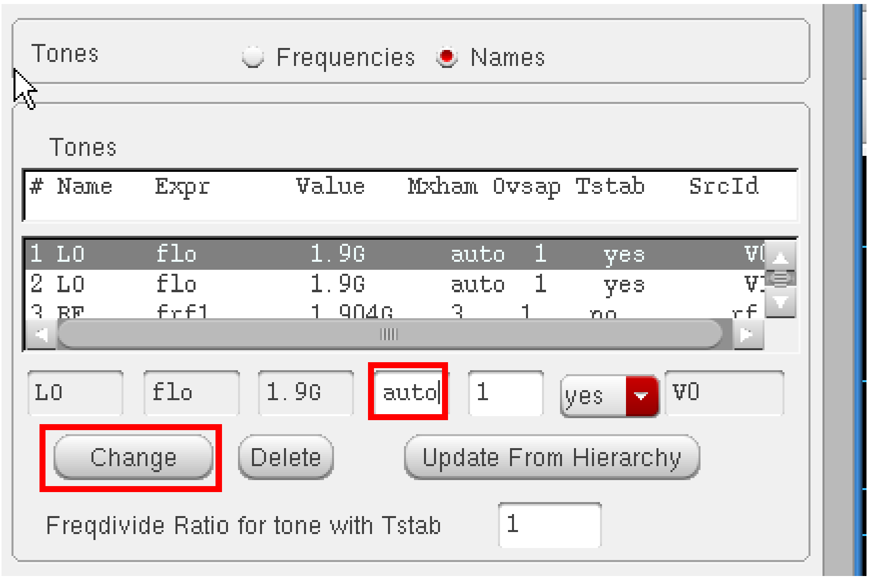
9. Select *LO* in the *Tones* section.

You can use hb with up to four signals present in the circuit. In this circuit, there are three tones, the LO and two RF tones. The two LO signals have the same name, the same frequency, and are considered a single tone. Whenever you have two signals at the *same* frequency, make sure you set the *Frequency Name 1* or *Frequency Name 2* property on the source to the *same* name as it was done in this example. When *Tones* is set to *Names*, the simulator considers both of the sources as a single input signal.

You viewed the names (*frf1* and *frf2*) in the input port sources in an earlier step.

Figure A-380 Tones Section of the Choosing Analyses Form with Tones Set to Names

.



10. Ensure that *Tstab* is set to *yes* in the *Tones* section for the LO tone. Set *tstab* to *yes* on the signal that causes the largest amount of distortion in the system. (In this example, that is the *LO* tone).

11. On the LO tone, ensure that *Mxham* is set to *auto* (type *auto* in the field, if necessary). Since the circuit is mildly nonlinear (thatis, the signals have no sharp edges), leave *Ovsap* (oversample) set to the default value of *1*.

12. Click Change. The form updates. Both LO tones now have *Mxham=auto* and *Tstab=yes*.

Note: If not using *Mxham* as *auto* on the LO tone, change the *Maximum harmonics* on the LO tone from *3* to *10*. Place the cursor in the field under *Mxham* and type *10*. Since the LO signal is well above compression, a higher number of harmonics is advised. When you set harmonics manually, if a signal is below compression, start with 3 harmonics. If a signal is at compression, start with 5 harmonics. If a signal is above compression, start with 7 harmonics. Run the simulation. Now increase the number of harmonics on each tone separately by about a factor of 1.5, and run again. If the result changes, increasethe harmonics again. If the result does not change, the original number of harmonics was enough, and you might be able to decrease the number of harmonics. For fastest runtime with full accuracy, use the smallest number of harmonics for each tone that is applied to the circuit.

13. Select the RF tone in the *Tones* box. Ensure that *Mxham* is set to *3*. Do the same for *RF2*.

14. Click *Change* to update the form.

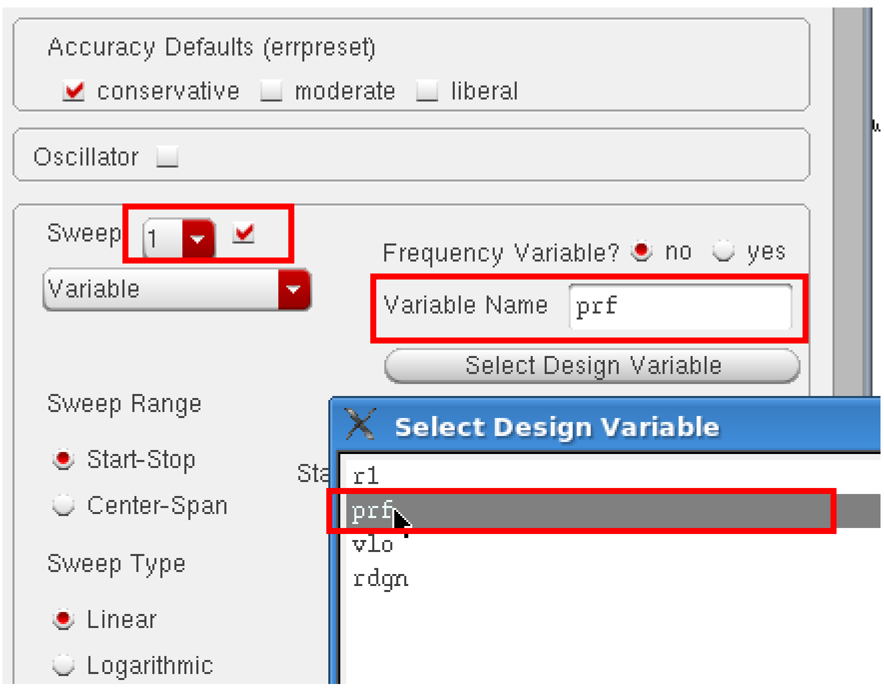
15. For the Harmonics selection (just below the list of inputs), leave it at the default value of *Default*. If you click *Select*, you would have the option of setting the method to *diamond*, *funnel*, or *axis* cut. (For more information on diamond, funnel, and axis cut, see Chapter 3, "Frequency Domain Analyses: Harmonic Balance.")

16. Set *Accuracy (errpreset)* to *conservative*. The third and fifth order intermodulation distortion is calculated with this setup.These amplitudes are small, and thus high accuracy is needed.

17. In the *Sweep* section, click *Sweep* . Leave it set it to the default value of *1.*

18. For *Frequency Variable?* select *no*. You will be sweeping input power rather than frequency.

Figure A-381 Sweep Section in hb Choosing Analyses Form



19. Click *Select Design Variable* and select the *prf* variable for the power sweep in the *Select Design Variable* form.

20. *Click OK* in the *Select Design Variable* form.

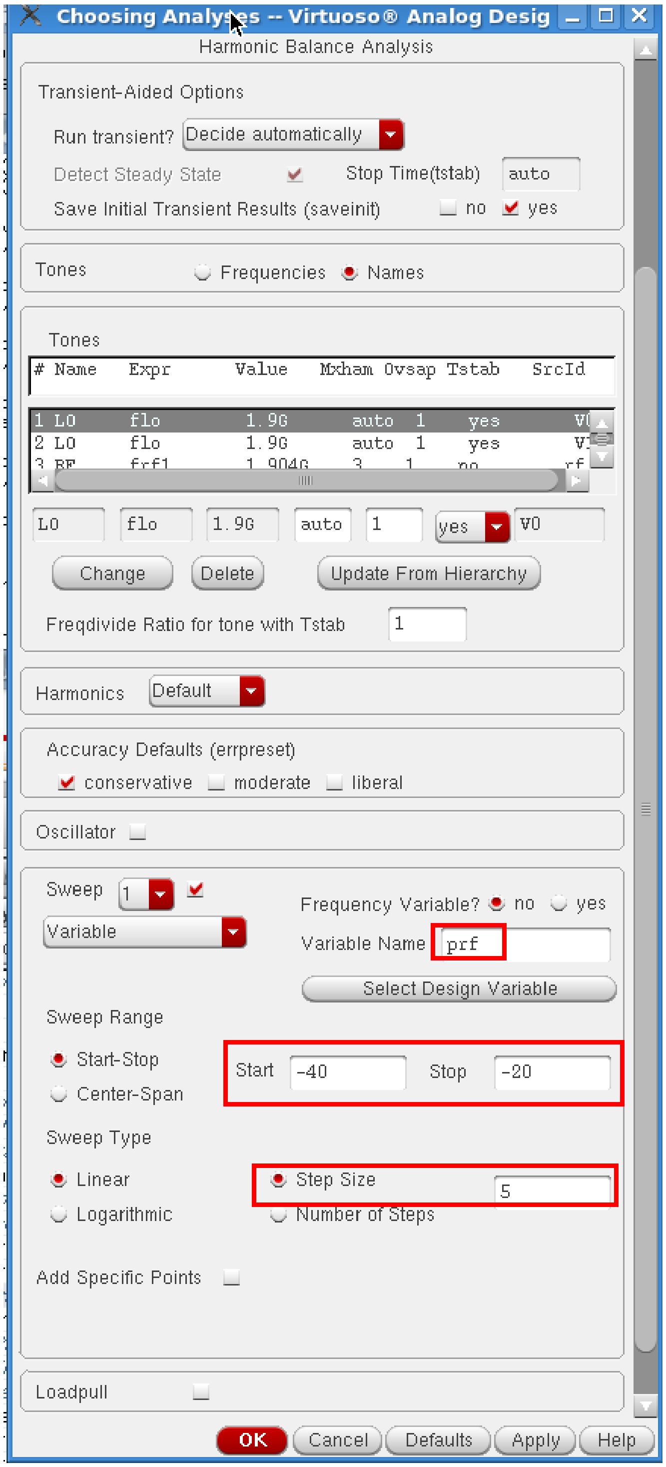
21. In the *Sweep Range* section, type *-40* in the *Start* field and *-20* in the *Stop* field.

22. Select *Sweep Type* as *Linear* and type *5* in the *Step Size* field.

Note: Because this is a small-signal measurement, you want to make sure your power sweep range does not go too high (stay operating at least 10dB below the 1dB compression point).

The completed *Choosing Analyses* form should look like the following:

Figure A-382 Choosing Analyses Form Setup for IP3

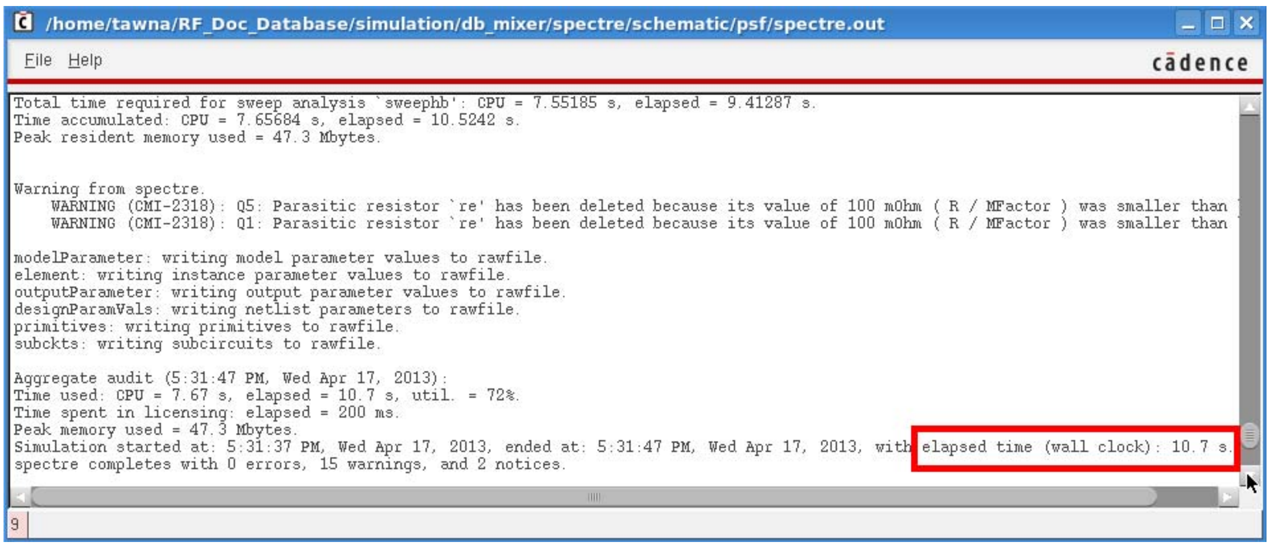


23. Click *OK*.

24. Start the simulation. From the ADE L main window or the Schematic, click the green arrow icon ( )on the right side of the window.

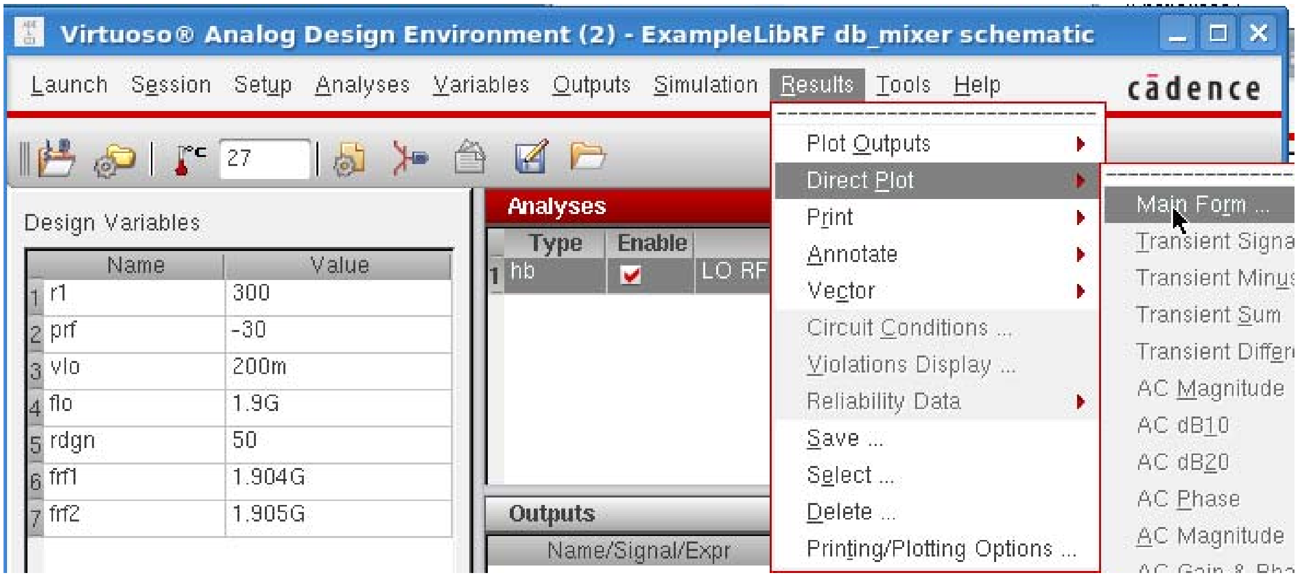
The Spectre output window is displayed with the simulator status information. Note the time it takes to run the simulation.

Figure A-383 Spectre.out File Showing Elapsed Time



25. After the simulation finishes, plot the output spectrum. In the *Analog Design Environment window*, choose *Results - Direct Plot - Main Form.* Alternately, you can click the *Direct Plot* icon ( )in the Schematic window.

Figure A-384 Accessing the Direct Plot Form



The Direct Plot Form appears.

26. In the *Analysis* Section, select *hb_mt*.

27. In the *Function* Section, select *IPN Curves.*

28. Select *Net (Specify R)*. Leave the *Resistance* set to the default value of *50*.

29. Select *Variable Sweep (“prf”)* for *Circuit Input Power*.

30. Leave *Input Power Extrapolation Point (dBm)* set to the default (first point in the sweep).

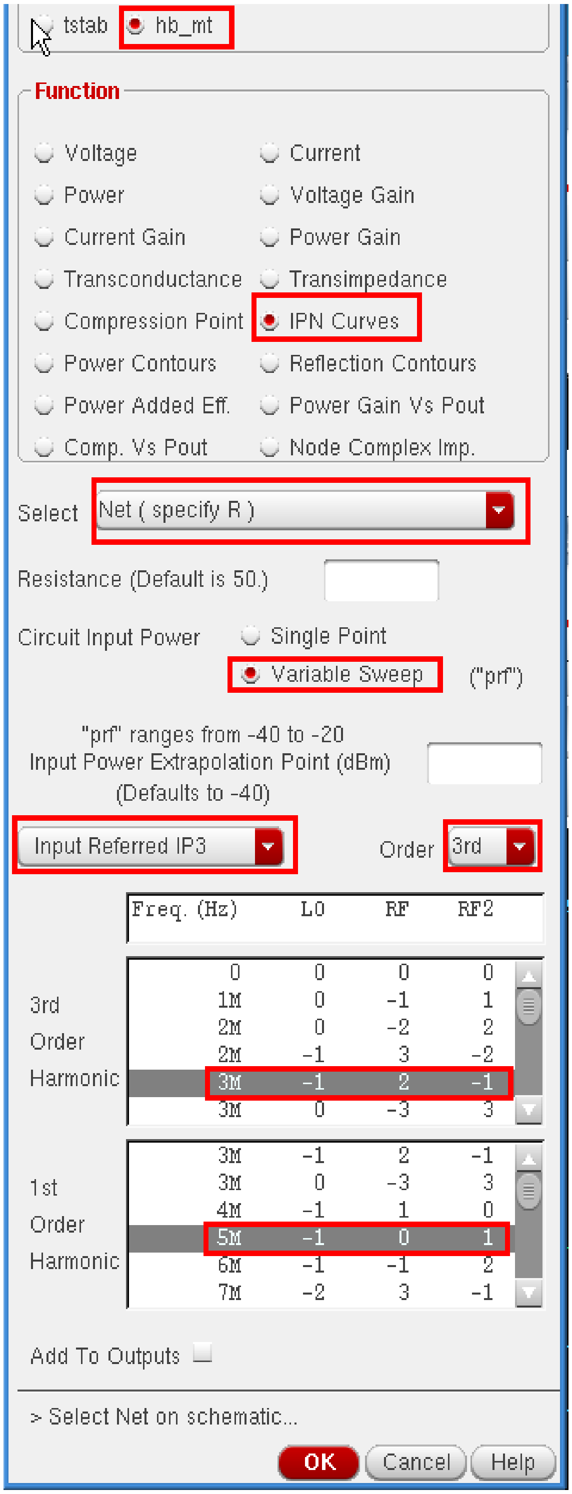
31. Choose *Input Referred IP3, 3rd Order.*

32. For the *3rd Order Harmonic*, choose the *3M -1 2 -1* entry.

Depending on the input frequencies chosen, hb may calculate multiple outputs at the same frequency. To determine which entry to select, add up the absolute value of all of the terms (for example, `|-1| + 2 + |-1| = 4` and `0 + |-3| + 3 = 6`) and choose the one with the lowest value. This is the lowest order term, which is usually the term that is desired.

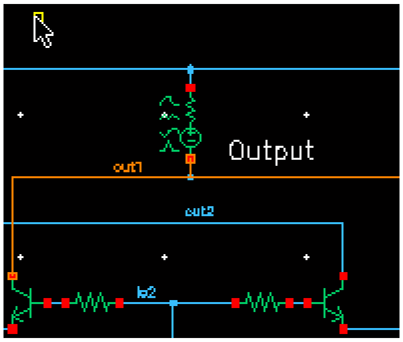
33. For the *1st Order Harmonic,* choose the *5M -1 0 1* entry.

Figure A-385 Direct Plot Form - IP3 from HB simulation



34. Click the *out1* net on the schematic just below the *Output* label, as shown below. Then press the *Esc* key, with the mouse cursor in the schematic window. This closes the *Direct Plot Form*.

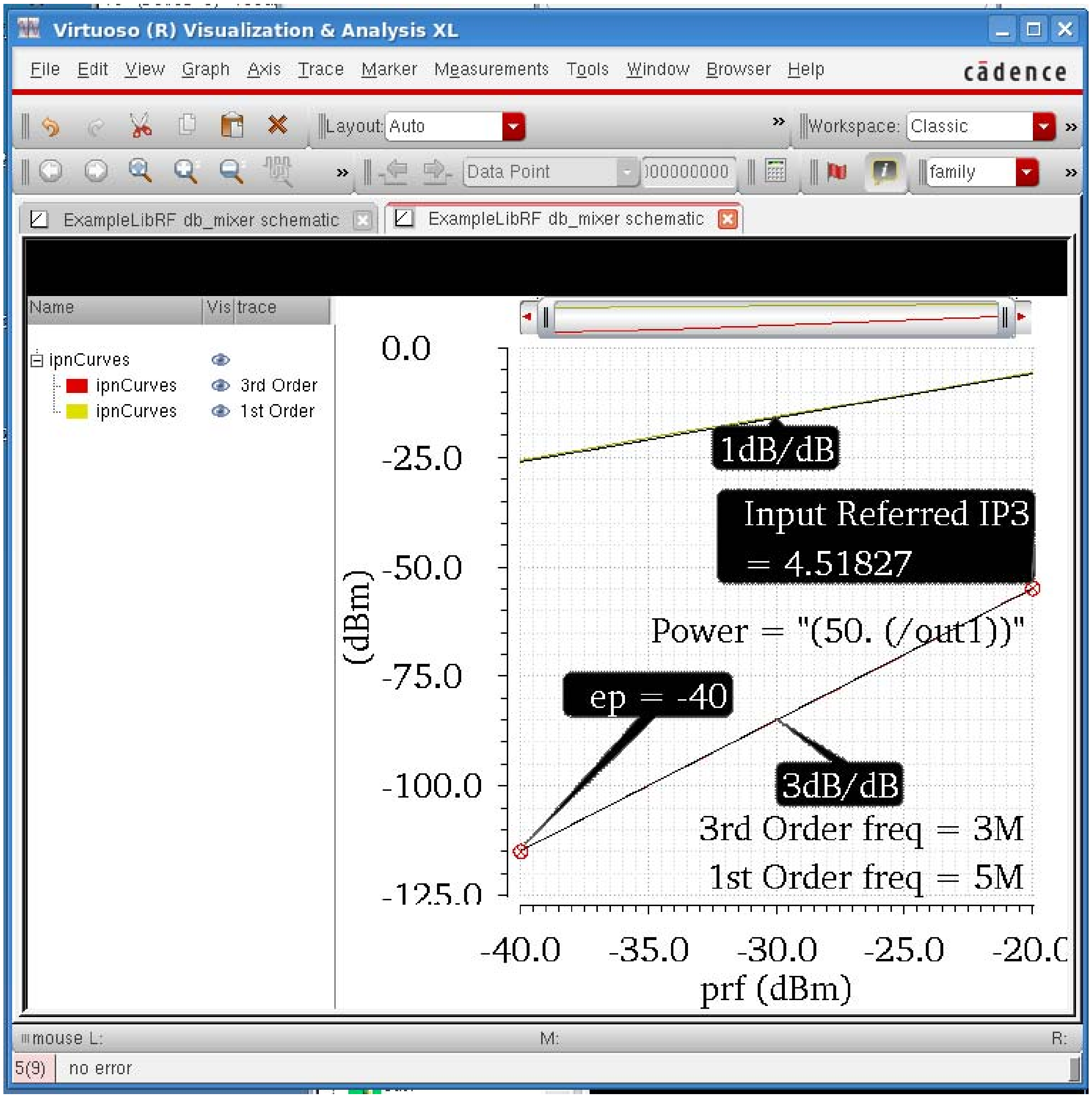
Figure A-386 Select Output Net on Schematic



The IP3 Plot is displayed in the waveform window.

Figure A-387 IP3 Plot from 3-tone Harmonic Balance simulation

.



Intermodulation products increase at rates that are multiples of the fundamentals. In the small-signal region, third-order terms increase 3dB per dB and the second-order terms increase 2dB per dB.

Note: If you do not see the IP3 readout, you may need to click in the graphics area to deselect the marker, then select and move the Input Referred IP3 readout so that it is positioned in the visible area of the graph.

In the previous plot, you can see that the circuit is operating within the small signal region. The third-order curve is following a 3dB/dB slope. Note the IP3 measurement. You will compare this to the IP3 measurement using the Rapid IP3 methodology.

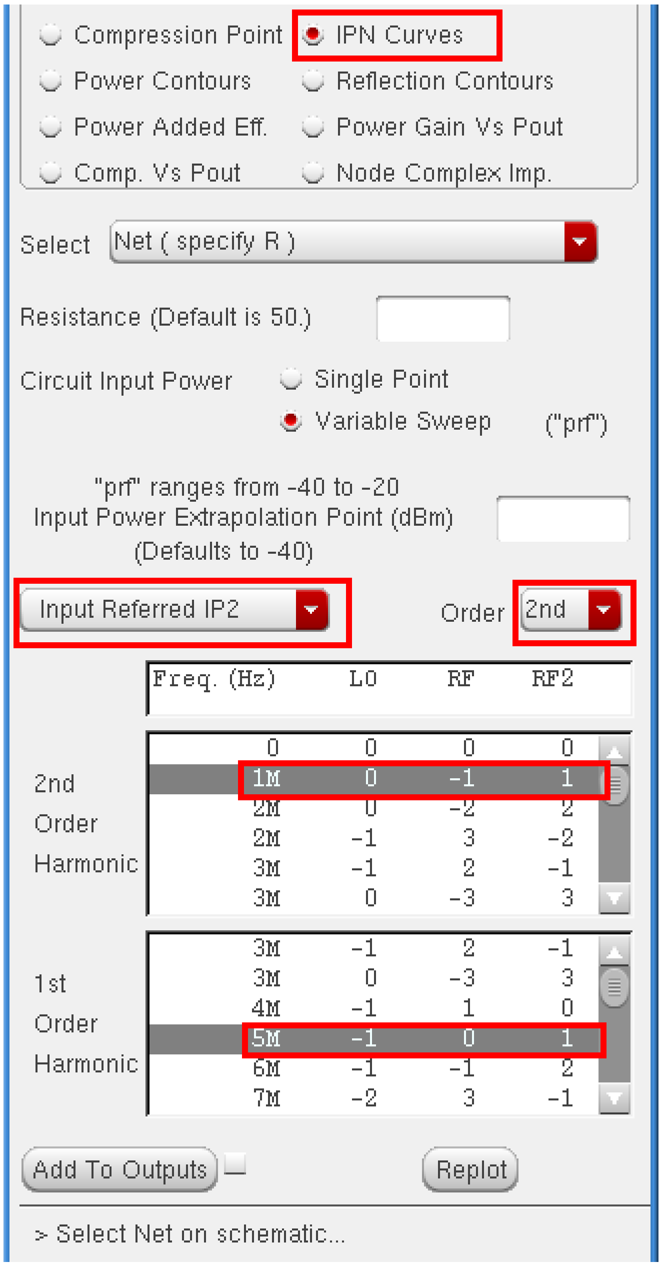
In the next section, you will plot the IP2 curves.

Plotting the IP2 Curves

1. In the *Direct Plot Form*, change the *Plotting Mode* to *Replace*.
2. Change the *Order* to *2nd*.
3. Choose *Input Referred IP2.*
4. For the *2nd Order Harmonic*, choose the *1M 0 -1 1* entry.
5. For the *1st Order Harmonic*, choose the *5M -1 0 1* entry.

   The*Direct Plot Form* should look like the following:

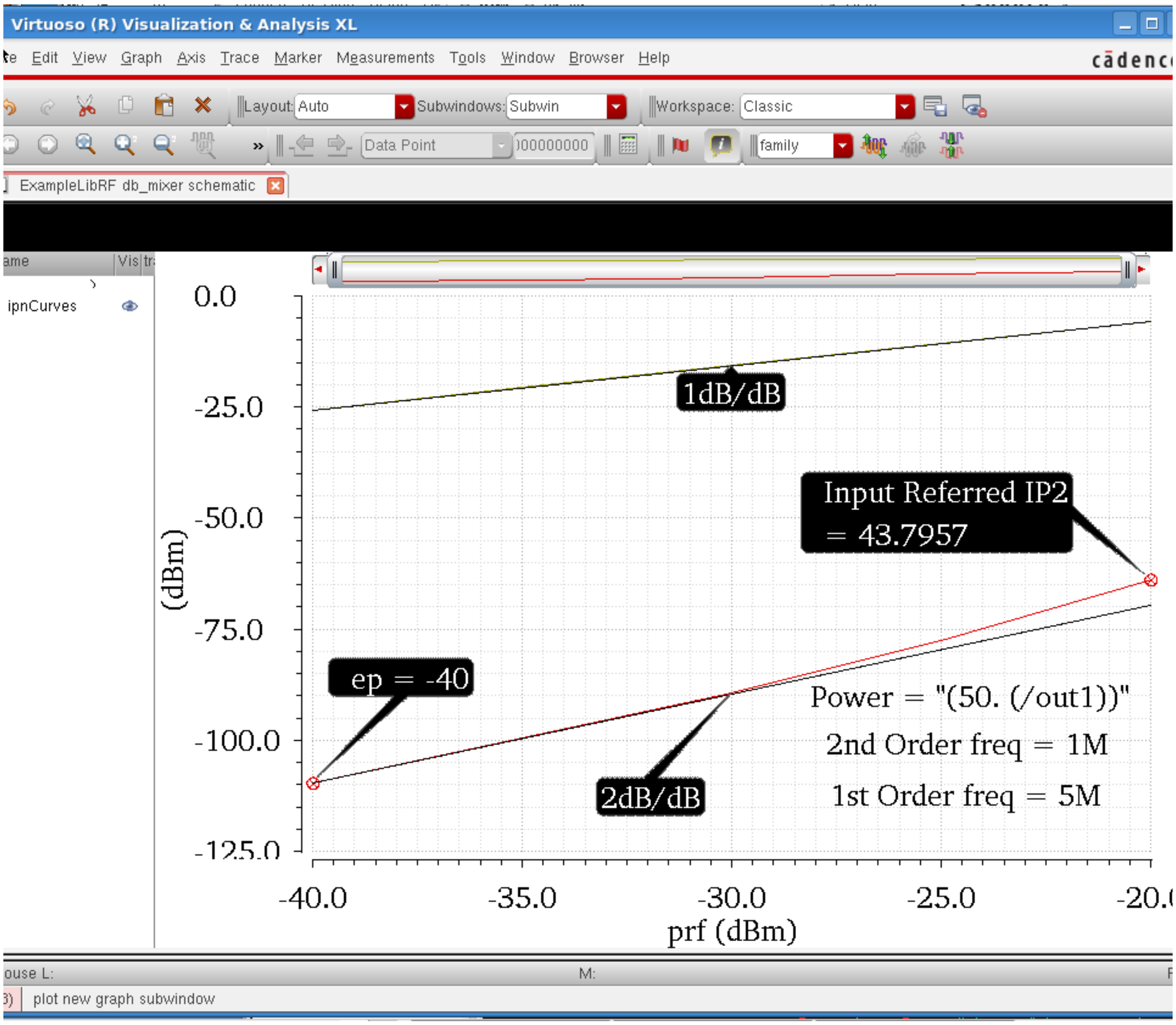
Figure A-388 Direct Plot Form for Input Referred IP2 Plot



6. Click Replot.

The IP2 Plot is shown in the following figure. Although it appears that the IP2 point is not at the intercept point of the first and second order curves, it does actually calculate that point. It is attached to the last result of the second order curve so the X axis can display just the simulation result. Also, note that above -30 dBm, the input power is high enough to cause deviations from the small-signal 2dB/dB curve. Since the extrapolation point at the first value of the sweep is used to do the IP2 calculation, the IP2 result is correct because the data follows the 2dB/dB curve below -30 dBm input power.

Figure A-389 IP2 Plot from 3-tone HB Analysis



Note: If you do not see the IP2 readout, you may need to click in the graphics area to deselect the marker, then select and move the Input Referred IP2 readout so that it is positioned in the visible area of the graph.

Note the IP2 measurement for comparison later when measuring IP2 using the Rapid IP2 methodology.

Leave the db_ mixer schematic and Analog Design Environment up. You will use them in the next section.

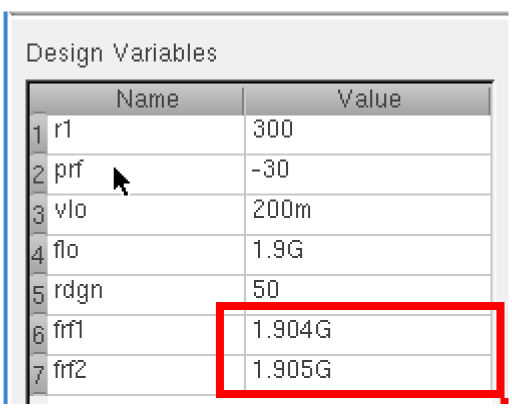
Rapid IP2 and IP3 Measurement

Another way to calculate IP3 and IP2 is to apply the LO only in the hb analysis and select Rapid IP2/IP3 from the hbac analysis. This is the fastest approach but it is limited to a small-signal IP3 measurement. It is also important to note that Rapid IP3 is not recommended for passive FET mixers where bsim3 or bsim4 models are used. In a small-signal application, this technique and the 3 tone hb method produce answers typically within 0.1dB of each other for IP3. For IP2, if extreme accuracy measures like very small reltol and vabstol, and a large number of harmonics are set in the 3-tone HB analysis, then closer agreement can be expected. These measures cause the 3-Tone HB analysis runtime to increase dramatically.

IP2/3 is calculated from a first and second/third order term. Because of the small-signal projection, hbac is used to measure IP2 and IP3. hb analysis applied to the LO tone captures the nonlinearity of the circuit created by the LO and the resultant frequency translation. The hbac analysis is used to calculate the amplitude of the first, second, and third order terms that are downshifted by the LO. The *Analog Design Environment* has a *Direct Plot* function to automate the IP2 and IP3 calculations.

1. In the *Analog Design Environment* simulation window, set the *frf1* and *frf2* design variables to *0*.

Figure A-390 Design Variables Section of ADE Simulation Window



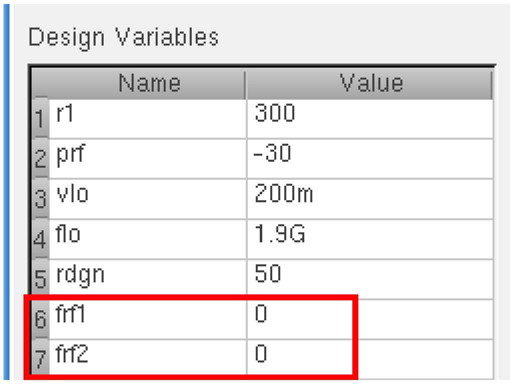
2. In the *Design Variables* section, click *1.904G* to the right of the variable *frf1* and type *0*. Next,press *Enter*.

3. Select *1.095G* to the right of the variable *frf2* and type *0*. Next, press *Enter*.

   Setting the input frequencies to 0 disables the production of waveforms for the large-signal analyses like tran, pss, and hb (harmonic balance). Theupdated *Design Variables* form looks like the following:

Figure A-391 Design Variables Section of ADE Simulation Window with frf1 and frf2 set to zero

.



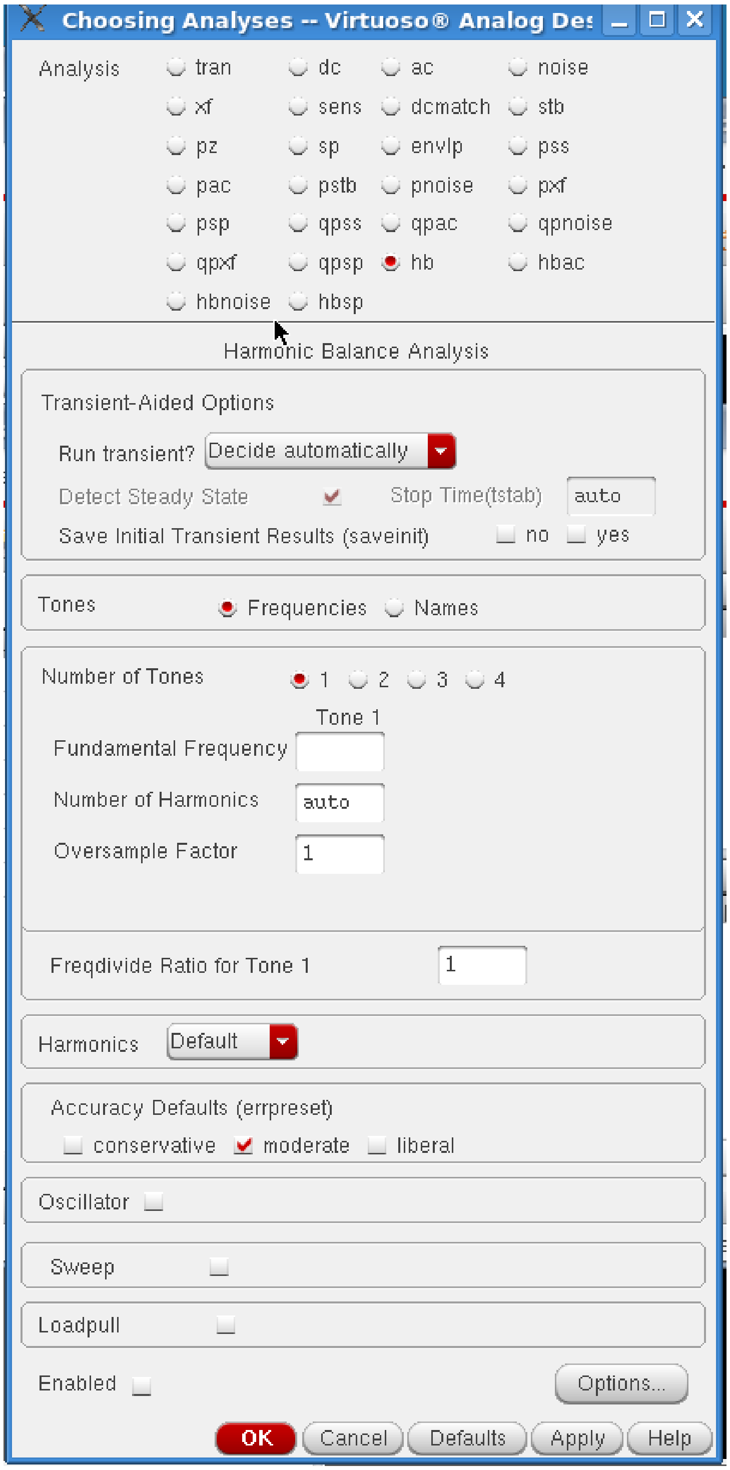
Design Variables

	Name	Value
1	r1	300
2	prf	-30
3	vlo	200m
4	flo	1.9G
5	rdgn	50
6	frf1	0
7	frf2	0

4. In the *Analog Design Environment* simulation window, choose *Analyses - Choose.*

The *Choosing Analyses* form is displayed. Select *hb* for the *Analysis* type. The form expands, as shown in the following figure:

Figure A-392 Choosing Analyses Form for hb Analysis



5. . Harmonic balance can set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient s*election in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected,and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

In the *Transient-Aided Options* section of the form, leave the settings at their default values unless otherwise noted.

a. For *Run transient?* select *Decide automatically*. (this is the default)

Run transient? will run the LO signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

b. For *Stop time (tstab), auto* is automatically populated in the field.

When *auto* is selected for *Stop time*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

When *Run transient?* is set to *Decide automatically*, the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

c. For *Save Initial Transient Results (saveinit),* select *yes*.

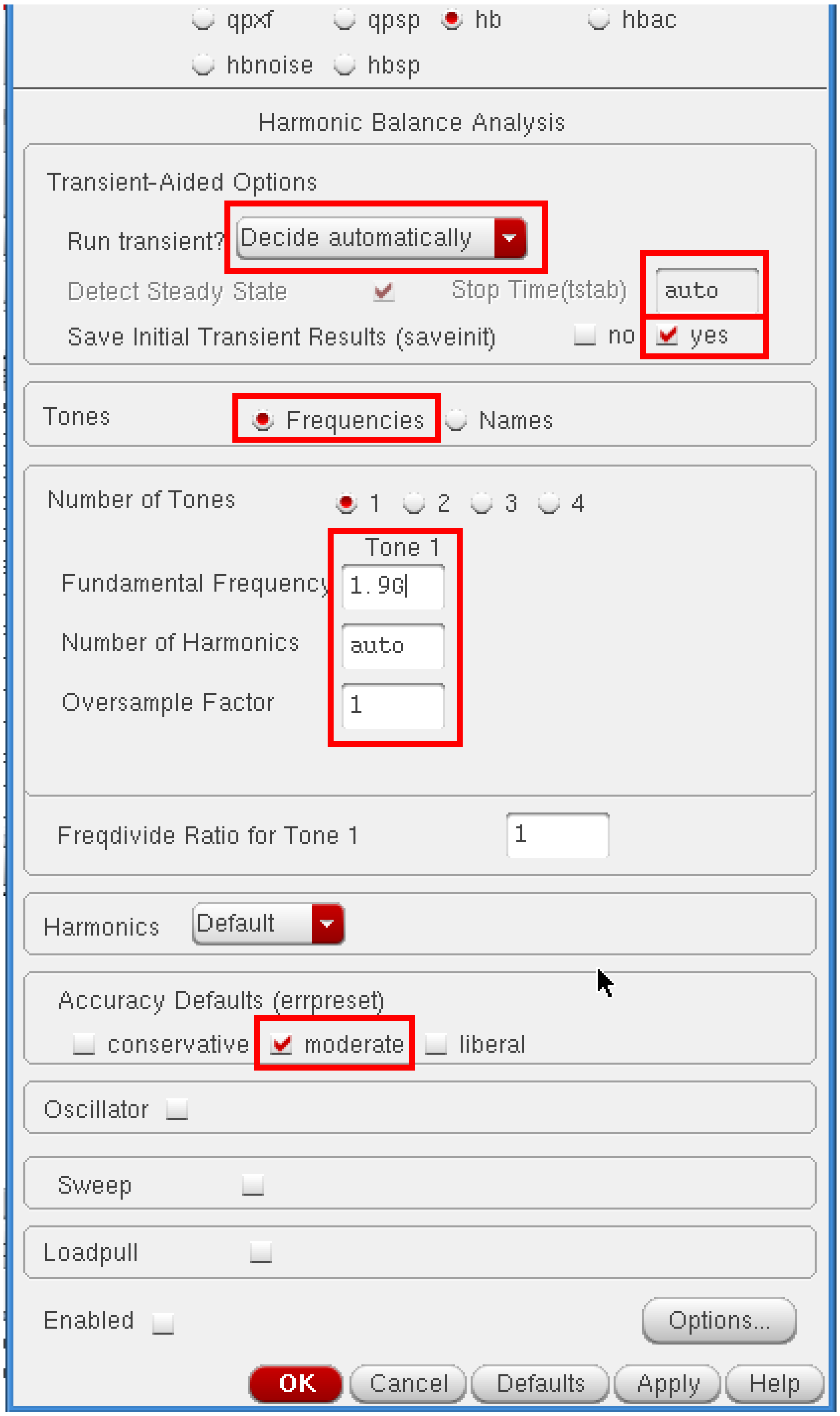
During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. Only the LO signal in *Tone1* is enabled for this measurement. At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations

All the signals are applied and the simulation is done in the frequency domain. Only the harmonics of the LO are calculated by hb.

Figure A-393 Transient Assisted Harmonic Balance

6. Ensure that *Tones* is set tothe default value of *Frequencies*.
7. The *Number of Tones* defaults to *1*.
8. Enter *1.9G* for the *Fundamental Frequency.*
9. Note that *Number of Harmonics* is set to *auto* by default.
10. Leave *Oversample factor* set to the default value of *1*. With auto harmonics, you do not need to set *oversample*. This circuit has sinusoidal-like waveforms (voltage and current), and an *oversample* of *1* is appropriate.
11. Because the first and third order mixing terms are calculated by hbac, high accuracy is not necessary in the hb analysis, so *moderate* is selected for *errpreset* and providesreasonable accuracy. The *Choosing Analyses* form should look like the following:

Figure A-394 Choosing Analyses Form - hb Setup for Rapid IP3

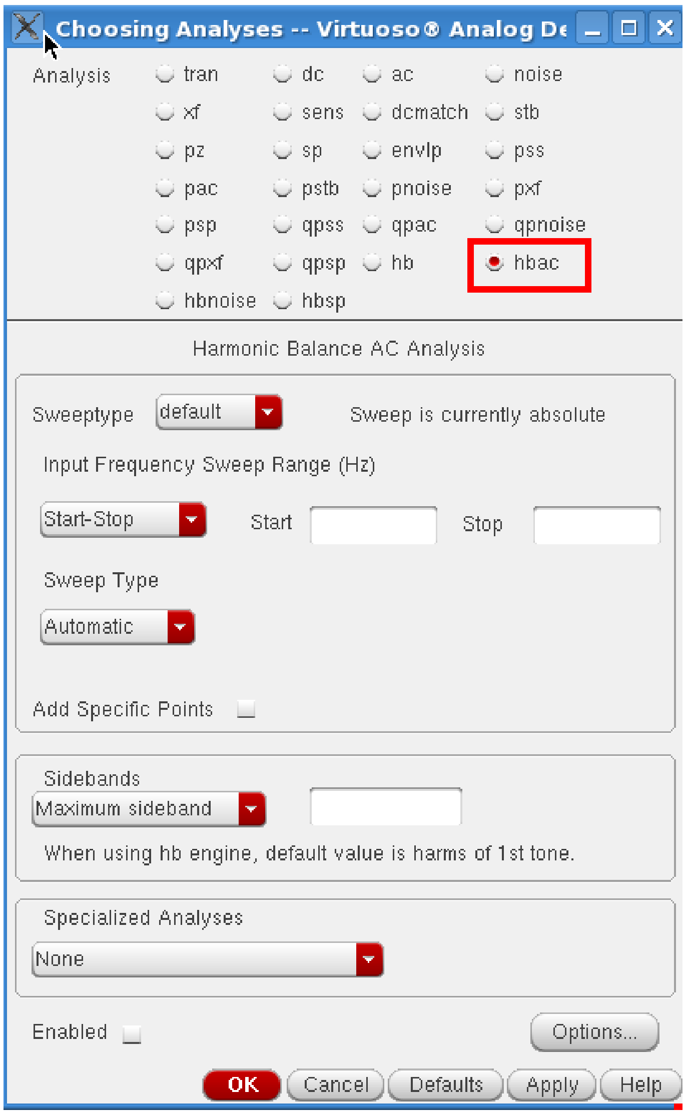


12. Click *Apply*.

Set-up HBAC Choosing Analyses Form

1. In the *Choosing Analyses* form, select *hbac* in the *Analysis* section. The *Choosing Analyses* form changes, as shown below.

Figure A-395 Choosing Analyses Form for hbac



2. At the bottom of the form, in the *Specialized Analyses* section, select *RapidIP3*. The form expands. Just below the *Rapid IP3* selection, for *Source Type*, leave it at the default setting of *port*.

3. Type */rf* for *Input Sources 1* and *2.* Alternately, you can click *Select* in the form and choose the *Input* rf port in the schematic.

4. Type *1.904G* in the *Frequency* Field to the right of *Input Sources 1*.
5. Type *1.905G* in the *Frequency* Field to the right of *Input Sources 2*.

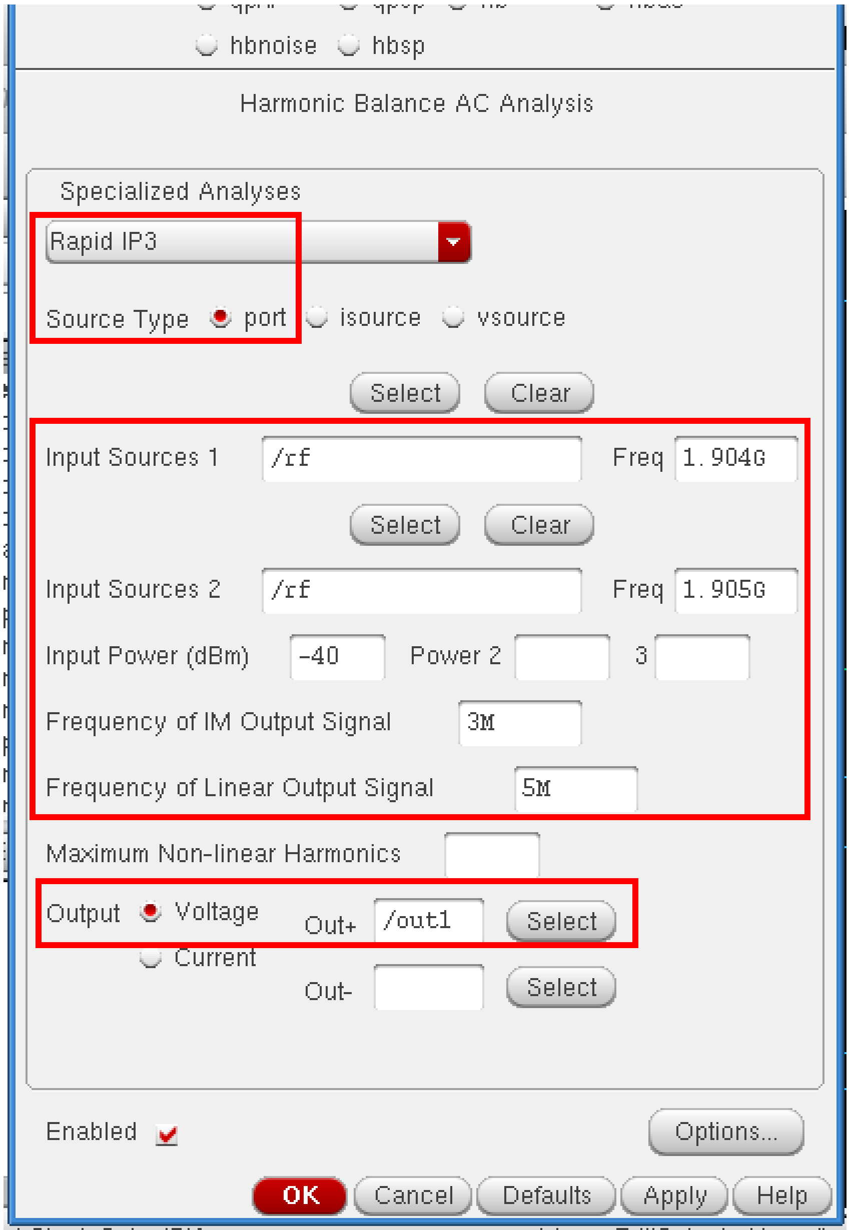
   1.904G and *1.905G* are the RF input frequencies.
6. Type *-40* for *Input Power (dBm).*
7. In the *Frequency of IM Output Signal* field, type *3M*.
8. In the *Frequency of Linear Output Signal* field, type *5M*.

   These are the linear and third-order output frequencies
9. Leave the *Maximum Non-linear harmonics* field blank. The default is fine for IP2 and IP3 measurments.
10. Select *Output Voltage* and in the *Out+* field, type */out1*.

    The*Choosing Analyses* form should look like the following

Figure A-396 HBAC Rapid IP3 Choosing Analyses Form

:



The selection of input power is very important for Rapid IP3. It should be set so that the system is in the small-signal range or linear region (more than 10dB below the 1dB compression point). If it is set too large, the system is in the large-signal range.

Three input power levels can be set in the Rapid IP3 *Choosing Analyses* form. To confirm if the IP3 result is accurate or not, you can set two different power levels around the chosen input power. If you get the same IP3 result, then the IP3 result is accurate, and the chosen input power level is in the suitable region.

The frequencies to be calculated depend on the choice of frequencies at the input and the LO frequency. Individual choices are provided for the frequencies which allows the selection of the smallest first order and largest second/third order product for your application.

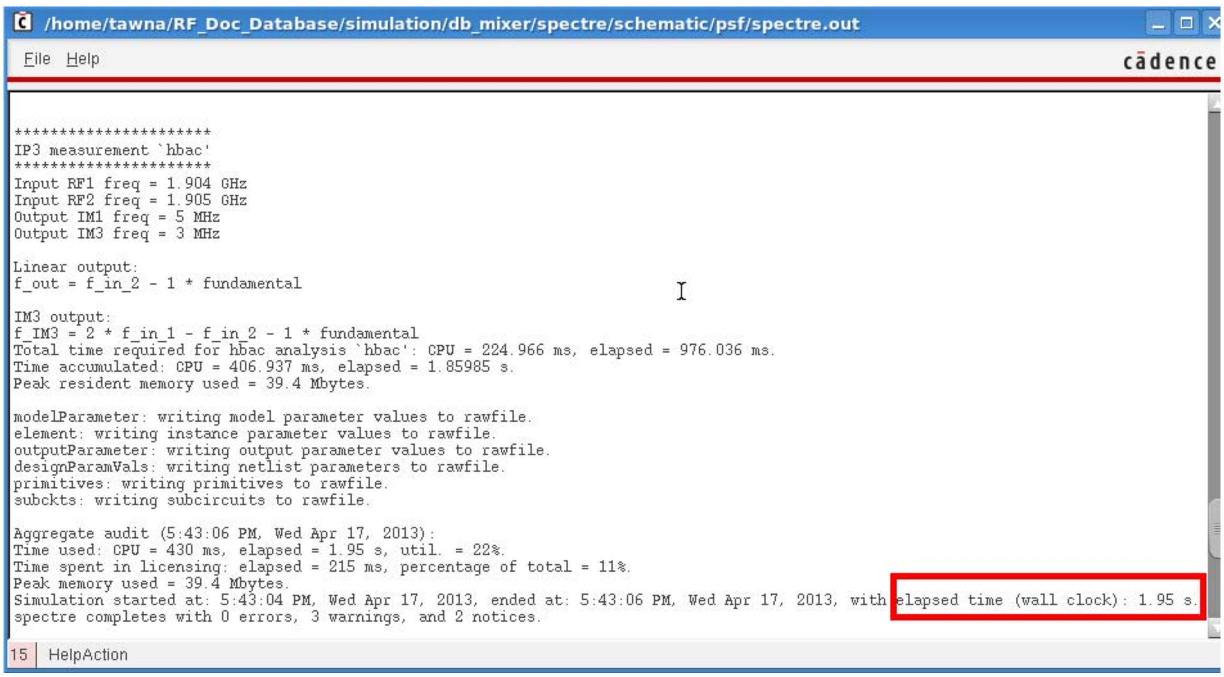
11. Click *OK* at the bottom of the *Choosing Analyses* form.

12. Start the simulation.

From the ADE-L main window or the Schematic, click the green arrow icon( )on the right side of the window.

13. The Spectre output window will appear with the simulator status information. Details of the IP3 measurement are also printed in the Spectre simulation log file. Note how much quicker the simulation finishes, compared to the three-tone hb simulation used previously to calculate IP3.

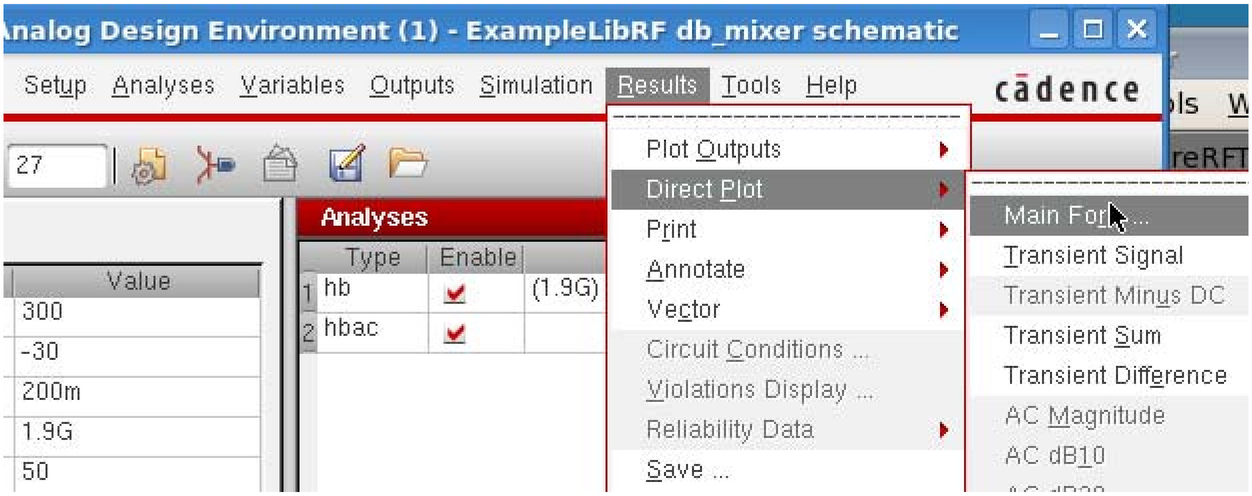
Figure A-397 Rapid IP3 spectre.out logfile



After the simulation finishes, plot the Rapid IP3.

14. In the *Analog Design Environment* window, select *Results - Direct Plot - Main Form.* You may also invoke the *Direct Plot Form* by clicking the *Direct Plot* icon ( )in the schematic.

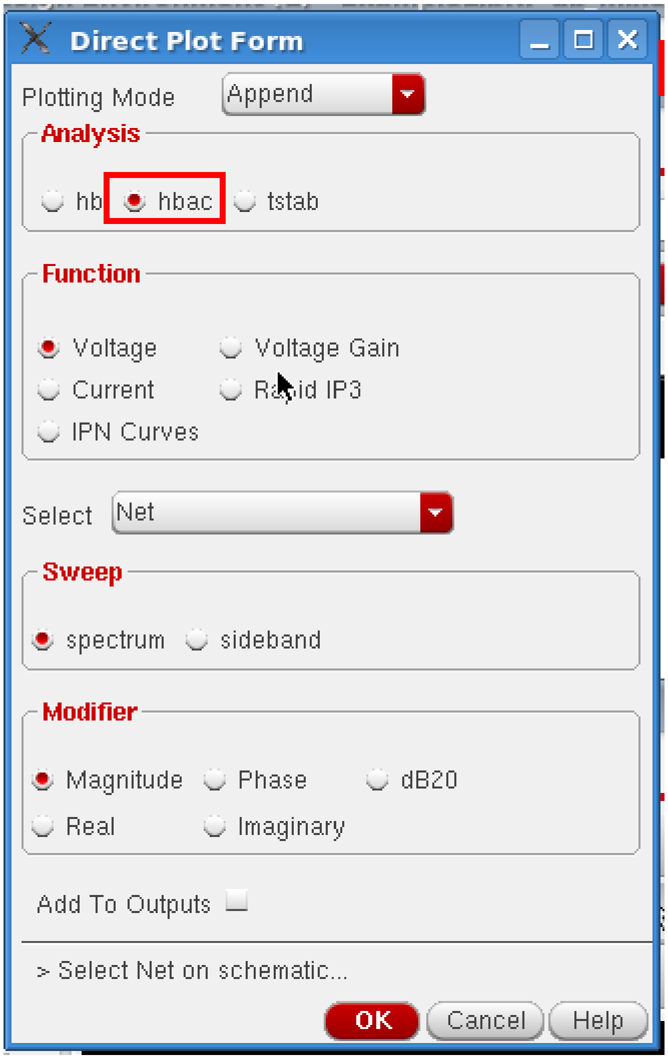
Figure A-398 Accessing the Direct Plot Form from ADE



The *Direct Plot Form* is displayed.

15. In the *Direct Plot Form*, select *hbac*. The form changes, as shown below.

Figure A-399 Direct Plot Form after Rapid IP3 Simulation

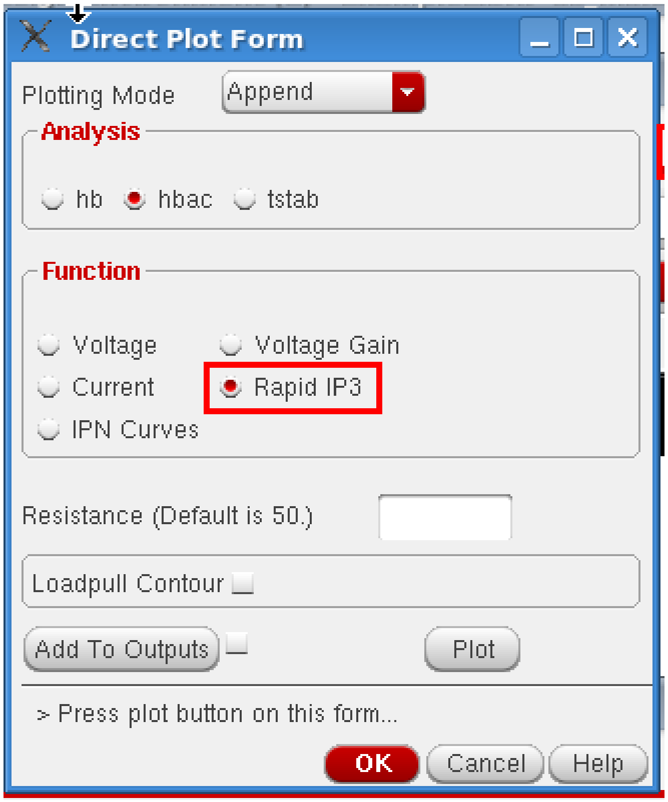


16. In the *Function* section, select *Rapid IP3.*

Leave the *Resistance* set to the default.

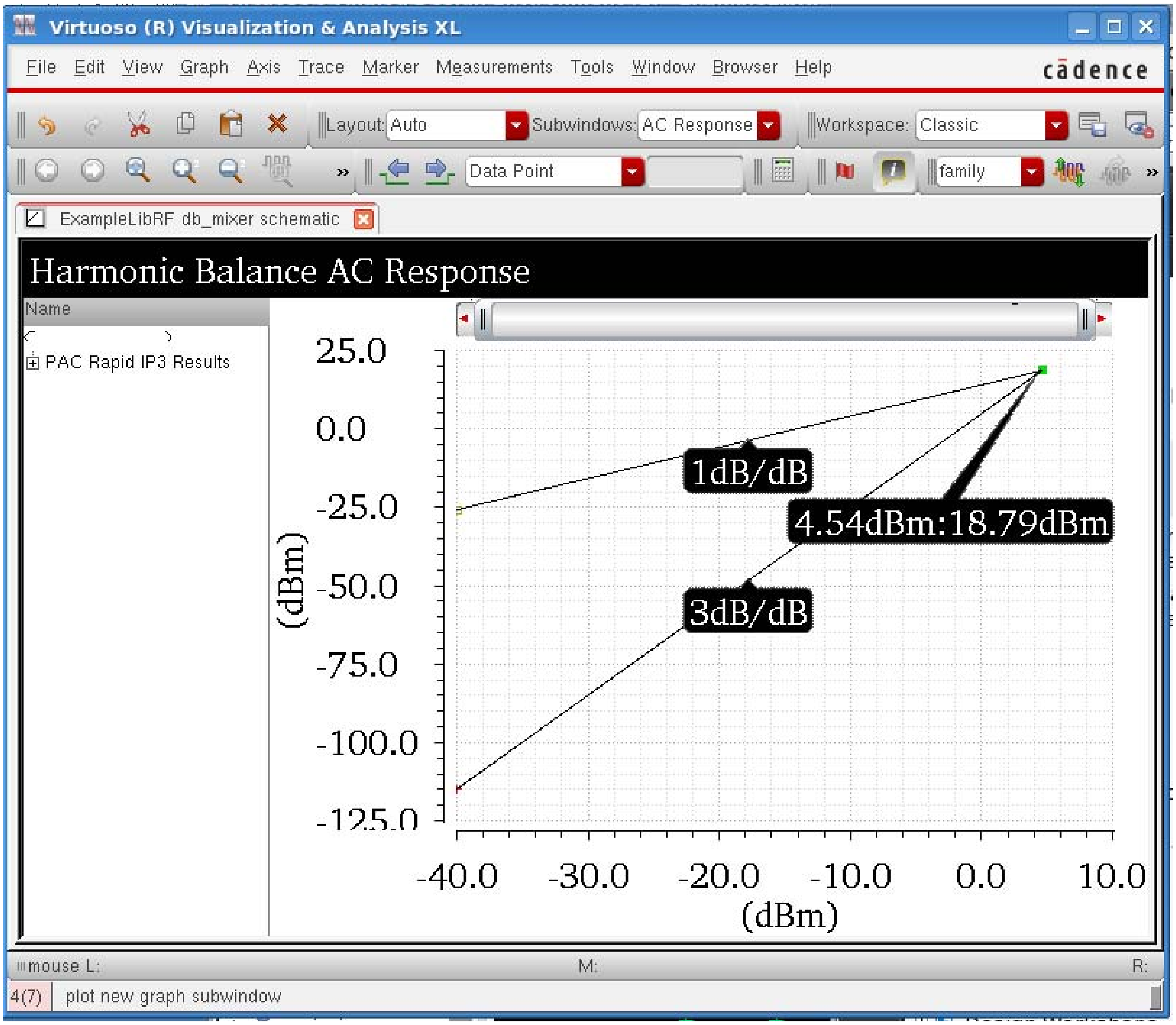
The *Direct Plot Form* should look like the following:

Figure A-400 hbac Direct Plot Form for Rapid IP3



17. Click *Plot*. The Rapid IP3 response is plotted, as shown below.

Figure A-401 Rapid IP3 Plot for db_mixer circuit



The input and output referred IP3 are shown in the waveform tool at the intercept point. The X axis of the plot is input power and the Y axis of the plot is output power, thus, the X-Y readout at the intercept point are the input-referred and output-referred IP3.

Note that the IP3 value from the 3 tone hb simulation (4.52) and rapid IP3 (4.54) are very close.

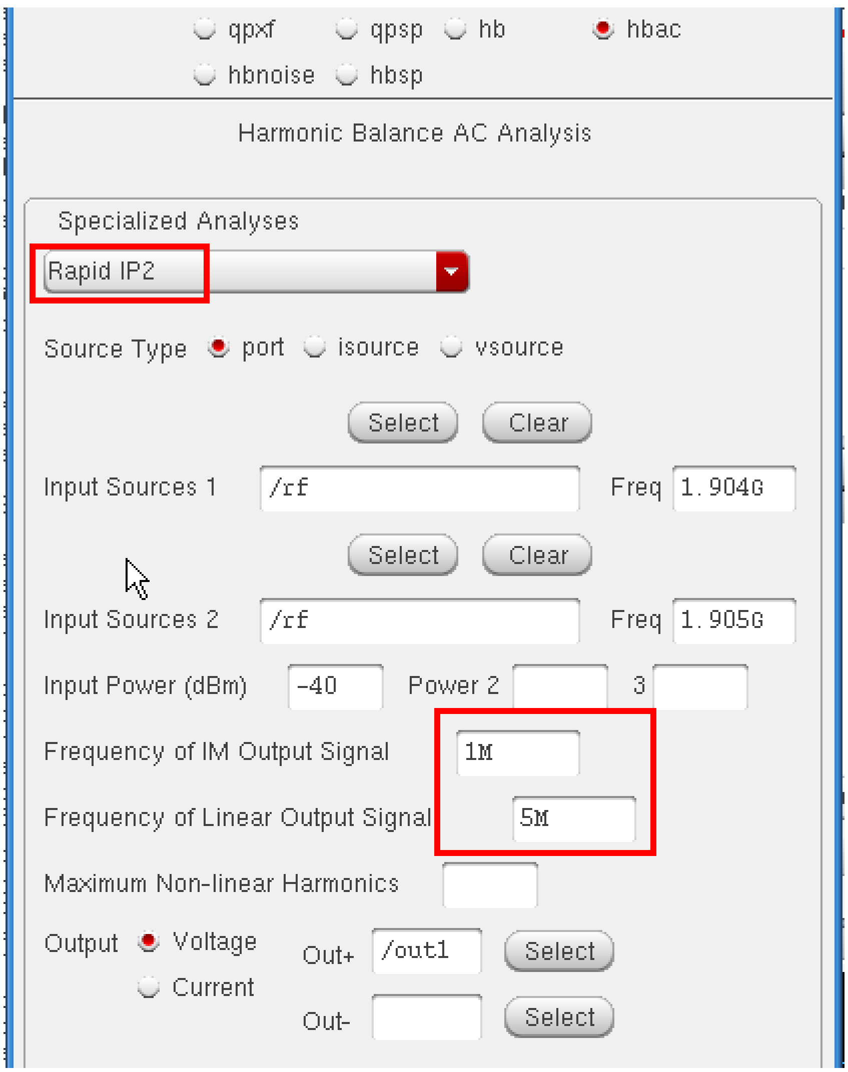
In the *Direct Plot Form*, click *Cancel*. In the next step, you will simulate and plot rapid IP2.

Simulate and Plot Rapid IP2

Next, set up the HBAC *Choosing Analyses* form, as shown below.

1. In the *Choosing Analyses* form, in the *Analysis* section, select *hbac*. Alternatively, in the ADE window, double click the hbac line in the *Analysis* section of the form.
2. At the bottom of the form, in the *Specialized Analyses* section, select *RapidIP2.* The form changes, as shown in the following figure.

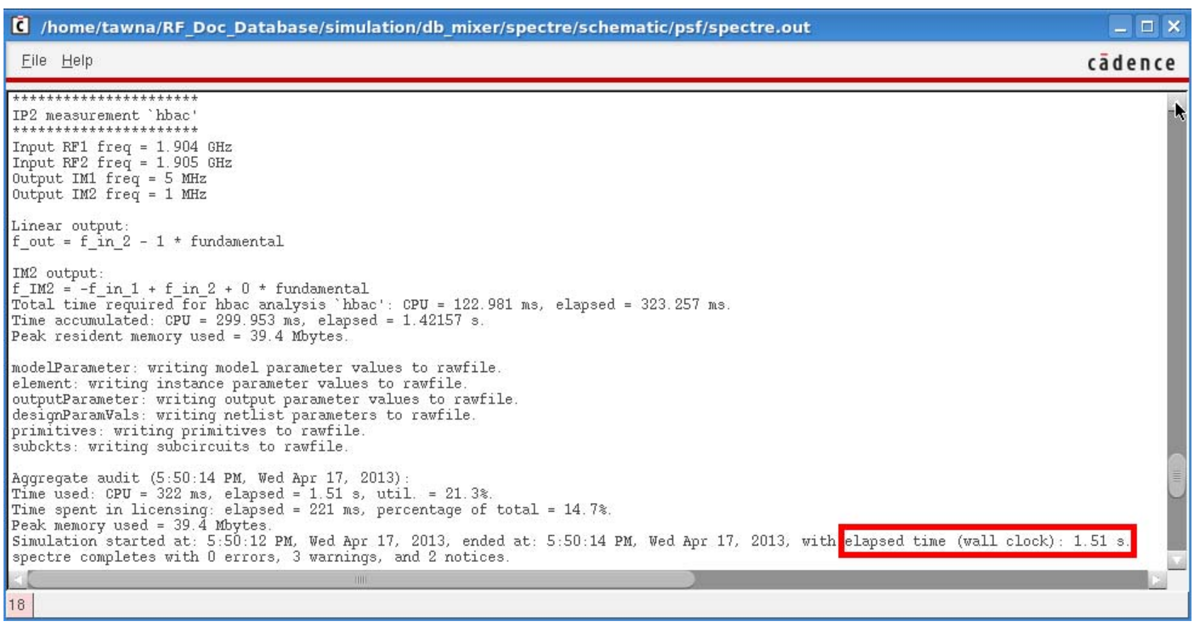
Figure A-402 Rapid IP2 Choosing Analyses Form



The form retains all the settings that were set for the rapid IP3. The only thing that needs to be changed is the frequency of the intermodulation product. The main downconversion frequencies are at 4MHz and 5MHz. The second order intermodulation product is at 1MHz . The frequencies to be calculated depend on the choice of frequencies at the input and the LO frequency. Individual choices are provided for the frequencies which allows the selection of the smallest first order and largest second/third order product for your application.

3. In the *Frequency of IM Output Signal* field, type *1M*.
4. In the *Frequency of Linear Output signal* field, leave it set to *5M*.
5. Click *OK* at the bottom of the *Choosing Analyses* form.
6. Start the simulation. In the ADE L main window or the schematic, click the greearrow icon ()on the right side of the window.
7. The Spectre output window isdisplayed with the simulator status information. Details of the IP2 measurement are also printed in the Spectre simulation log file. Note how much quicker the simulation finishes, compared to three-tone hb simulation used previously to calculate IP2.

Figure A-403 Rapid IP2 spectre.out Logfile



8. After the simulation finishes, plot the output spectrum.

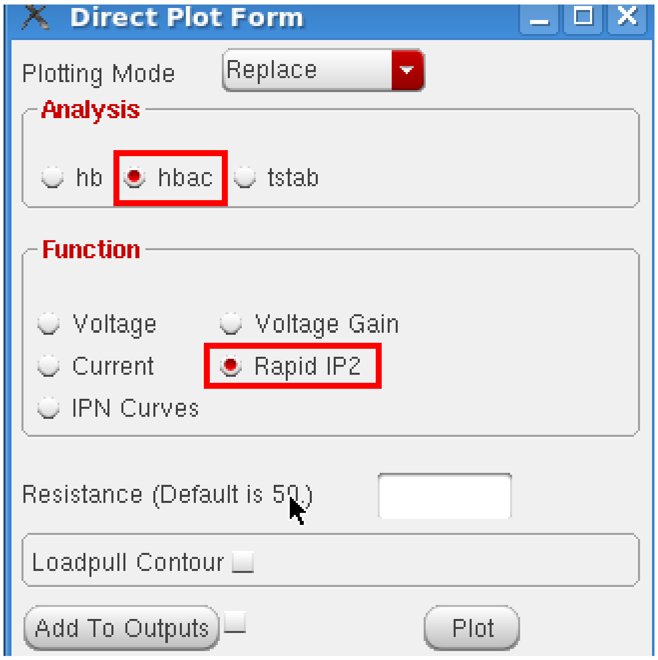
   In the Analog Design Environment window, choose *Results - Direct Plot - Main Form*.

   You may also invoke the Direct Plot Form by clicking the *Direct Plot* icon () in the schematic.

9. In the *Direct Plot Form*, select *hbac* in the *Analysis* section and *Rapid IP2* in the *Function* section. Leave the *Resistance* set to the default value *(50)*.

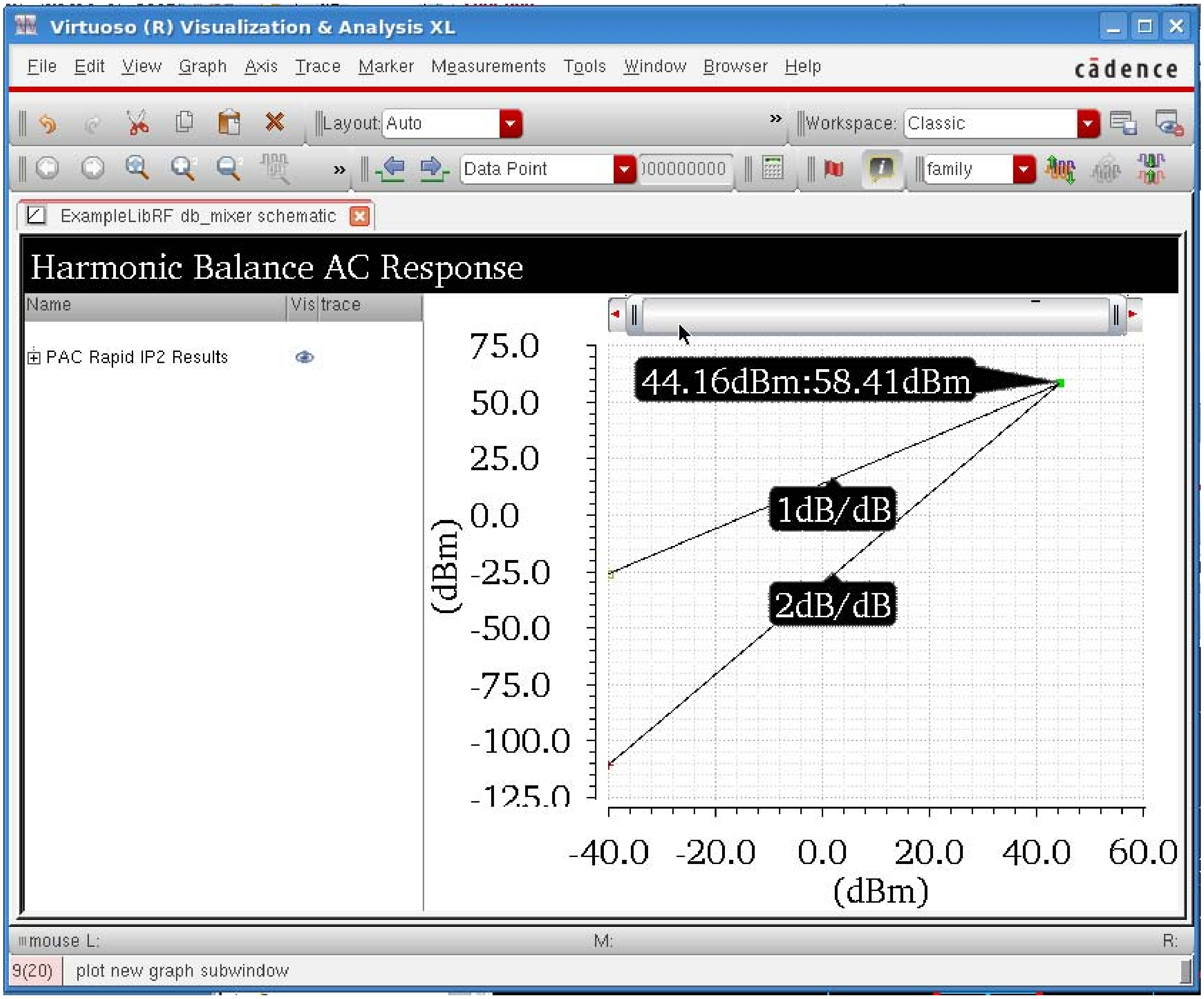
The *Direct Plot Form* is shown below.

Figure A-404 Rapid IP2 Direct Plot Form



10. Click *Plot*. The Rapid IP2 is plotted in the waveform window.

Figure A-405 Rapid IP2 Plot Results



The input and output referred IP2 are shown in the waveform tool at the intercept point. The X axis of the plot is input power and the Y axis of the plot is output power, thus the Xoutput-Y readout at the intercept point are the input referred IP2.

Note that the IP2 value from the three-tone hb simulation and rapid IP2 are within 1dB.

11. In the *Direct Plot Form*, click *Cancel*.
12. Close all waveform windowsby choosing *File - Close all windows*.
13. In the ADE L window, choose*Session - Quit.*
14. Select *No* in the *Save State* form.

15. In the Schematic Window, choose*File - Close All*.

Summary

In this section, you have measured the third-order intercept (IP3) using three-tone hb analysis and Rapid IP3 (hb and hbac specialized analysis). Rapid IP3 is a much faster way to measure IP3 and is just as accurate. In addition, you measured IP2 using the Rapid IP2 methodology.

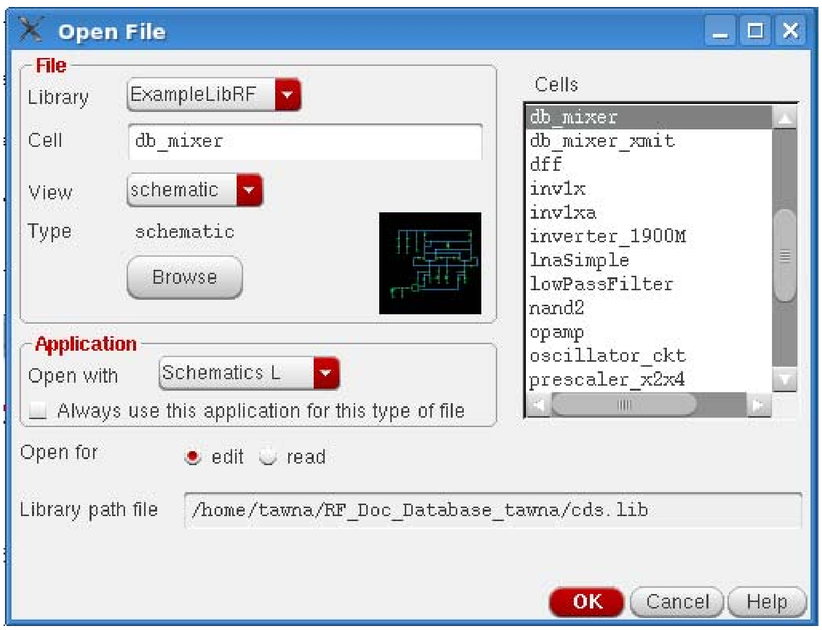
In the next section, you will make distortion measurements for the receive mixer.

Mixer Distortion Measurement

In the previous section, you measured IP3 and IP2. If you need to improve IP3 or IP2, it is helpful to know where the distortion is coming from, so you can fix the problem. Compression and IP3 are numerically related, so identifying components that cause compression also identify components that cause IP3 problems. In this next section, you will identify which components are causing compression in the circuit. For circuits that convert frequencies, hb and hbac compression distortion analyses are used. Similar capability for IP2 is provided in the IP2 distortion summary.

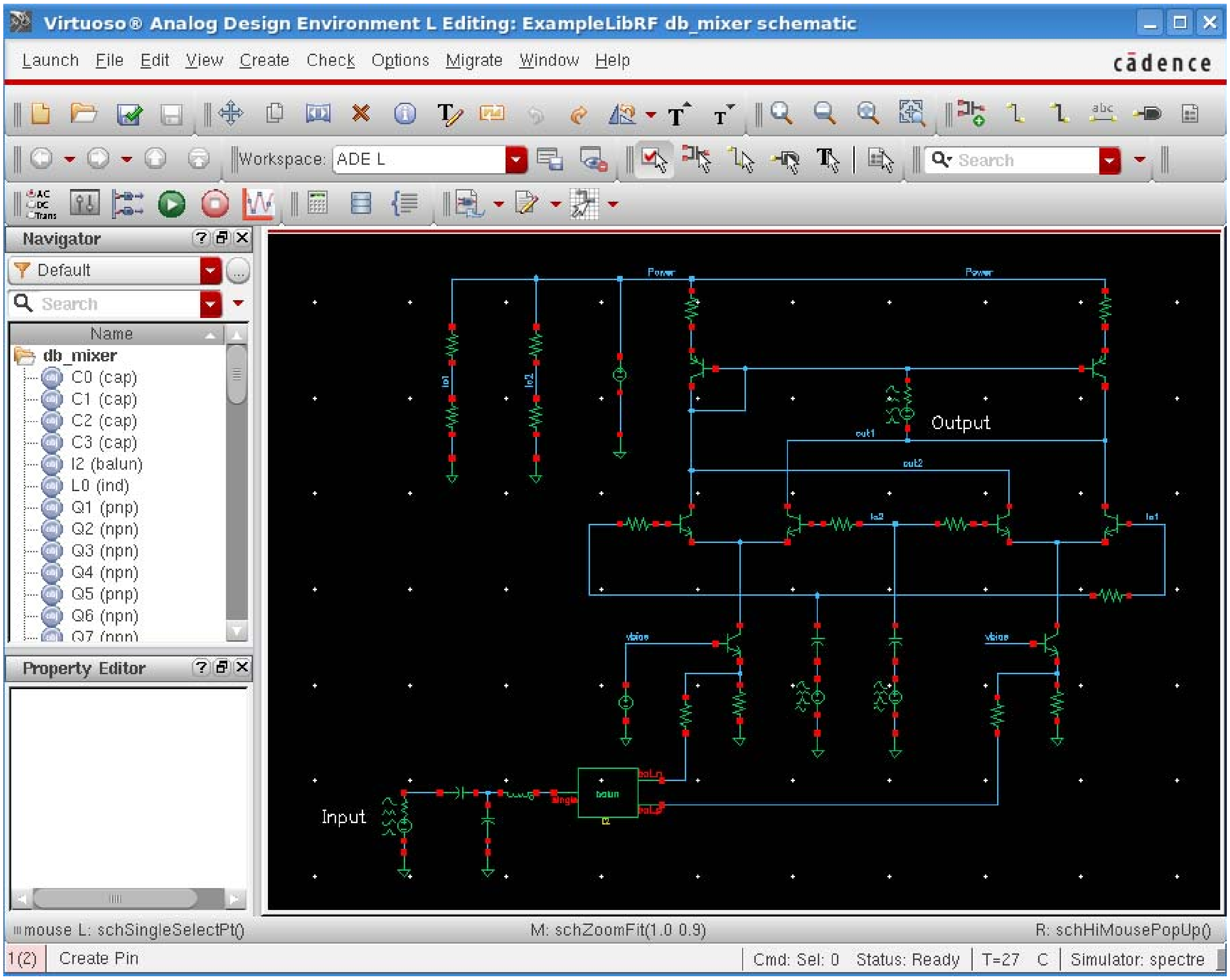
1. . In the CIW, choose *File - Open - CellView.* The *Open File* form is displayed. Choose the *db_mixer* schematic from *ExampleLibRF*.

Figure A-406 Open File Form.



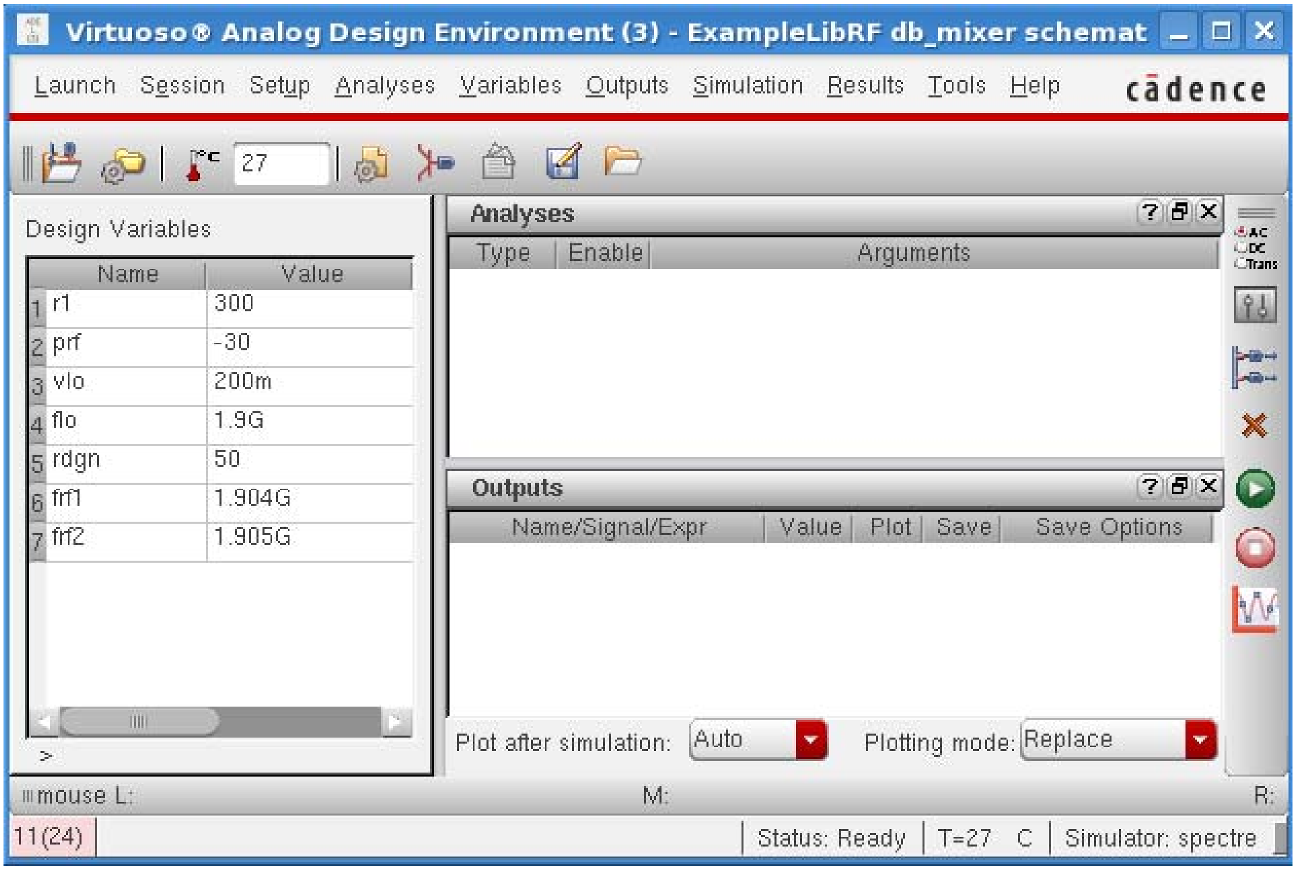
2. Click *OK*. The *db_mixer* schematic is displayed, as shown below.

Figure A-407 db_mixer Schematic.



3. Open the Analog Design Environment from the schematic by choosing *Launch - ADE L*. The Simulation Window is displayed, as shown below.

Figure A-408 Analog Design Environment Simulation Window



4. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

5. Specify the following in the *Choosing Simulator/Directory/Host* form:

   a. Choose *spectre* from the *Simulator* drop-down list.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field.

   c. Select the *Host Mode* that corresponds to your situation.

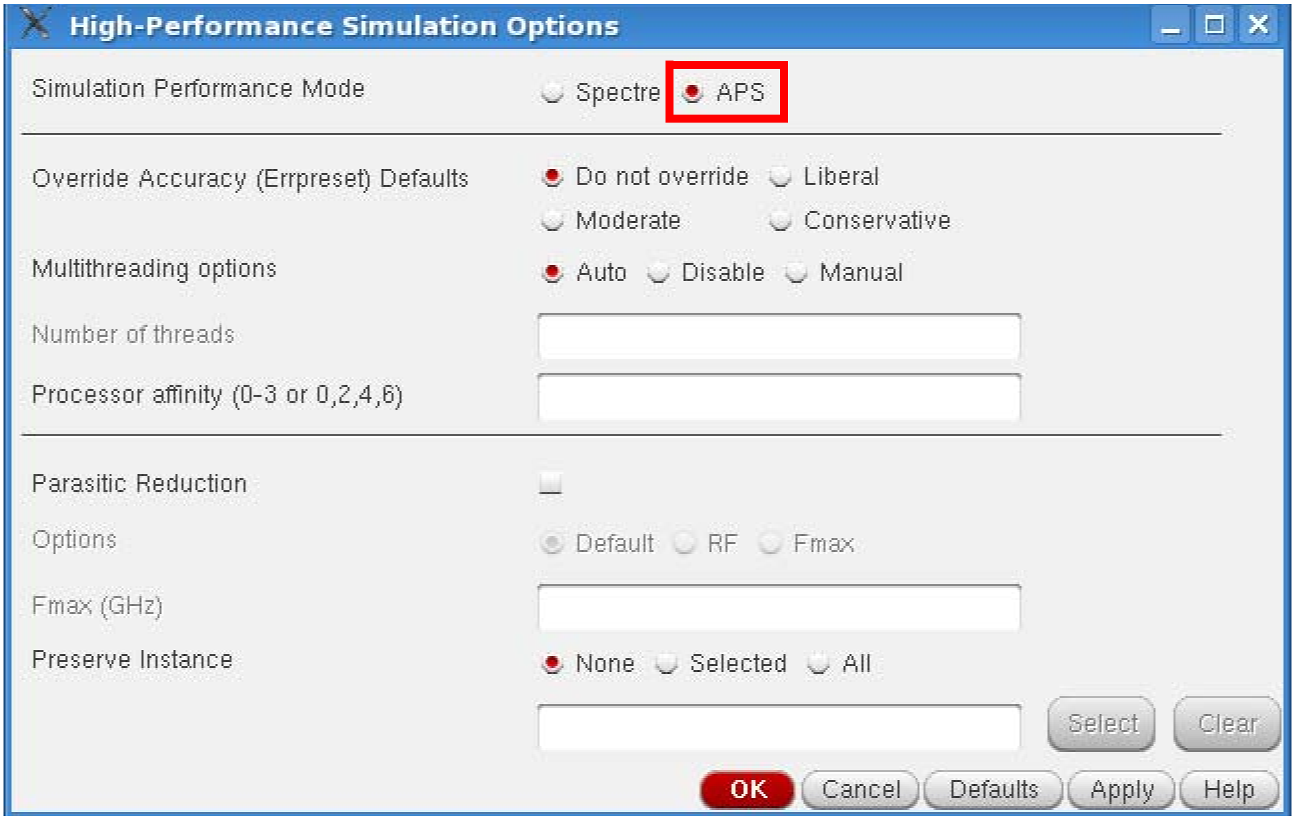
      If you choose *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

      The completed form appears similar to the following:

Figure A-409 Choosing Simulator/Director/Host Form

6. Click *OK* to close the *Choosing Simulator/Directory/Host* form.
7. Set up the High Performance Simulation Options, as follows:
   d. In the ADE window, choose *Setup - High Performance Simulation.* The *High Performance Simulation Options* window is displayed, as shown below.

Figure A-410 High Performance Simulation Options Form



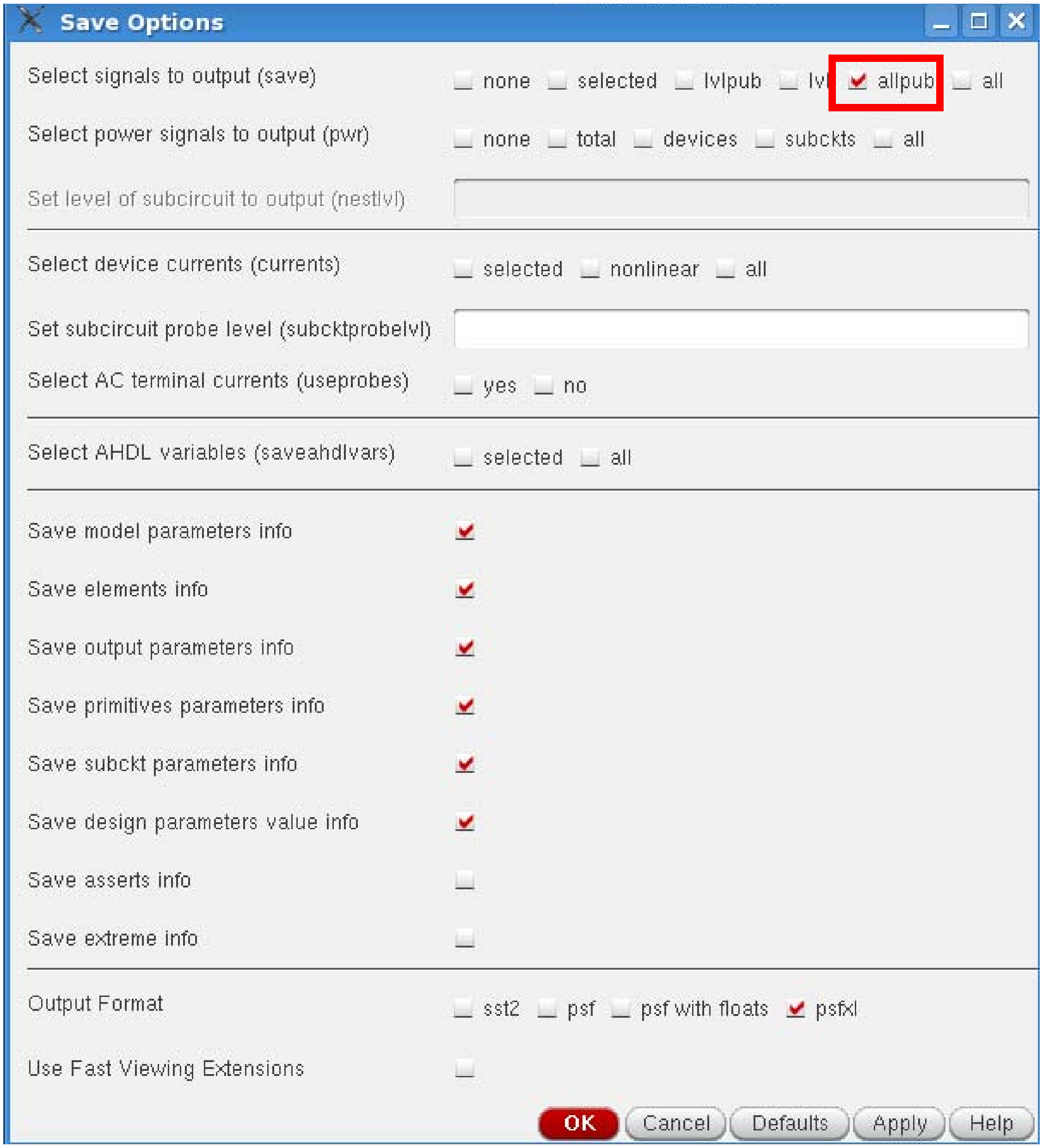
e. In the High Performance Simulation window, select *APS*. Note that *auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading. For more information, referto the the *Virtuoso Spectre Circuit Simulator and Accelerated Parallel Simulator User Guide*.

f. Click *OK*.

8. In the Virtuoso Analog Design Environment window, choose *Outputs – Save All.*

The *Save Options* form is displayed, as shown below.

Figure A-411 Save Options Form



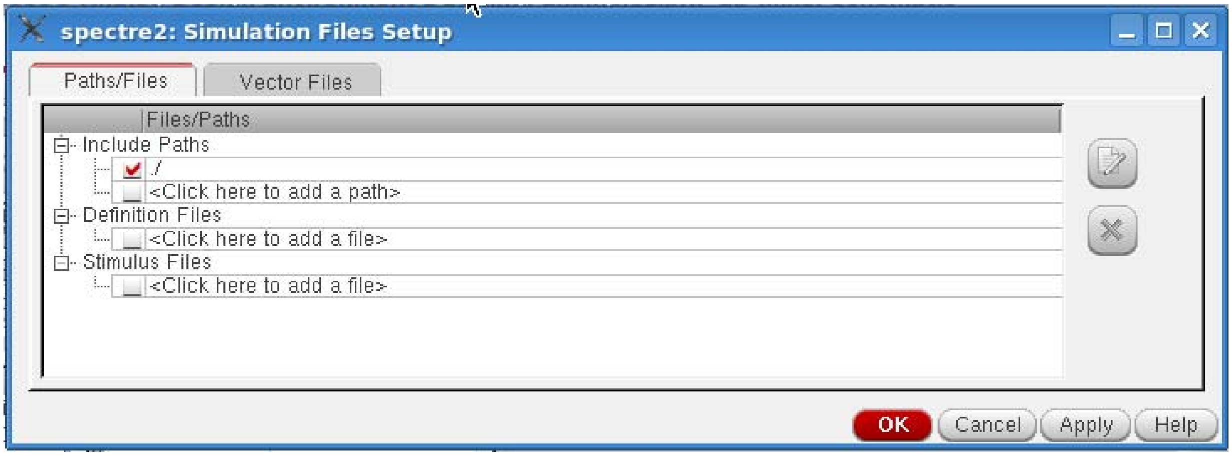
9. In the *Select signals to output* section, make sure that *allpub* is selected. This is the default which saves all node voltages at all levels of hierarchy, but it does not include the node voltages inside the device models.

10. Click *OK* to close the *Save Options* form.

Setting Up Model Libraries

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files*. The *Simulation Files Setup* form is displayed, as shown below.

Figure A-412 Simulation Files Setup Form



2. Ensure that the Include Path is set as shown above.
3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

   The *Model Library Setup* form is displayed.

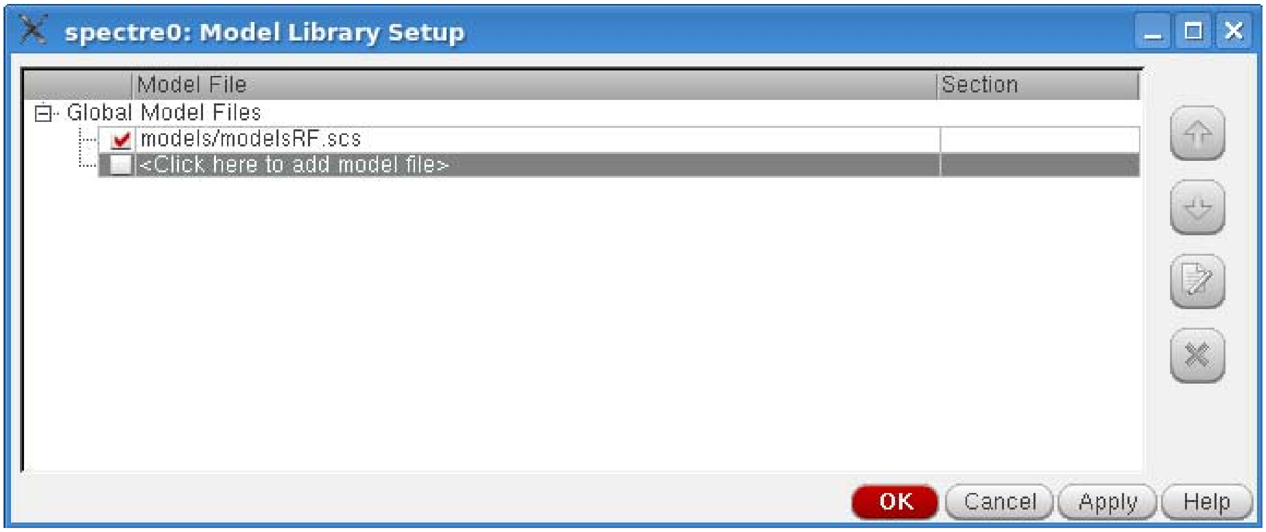
   In the *Model Library File* field, type the following path to the model file including the file name:

   ```
   models/modelsRF.scs
   ```

   Alternately, you can click *Browse* button and browse to the `modelsRF.scs` model file.
4. Ensurethat the Model File name is selected.
5. Click *Apply*.

   The *Model Library Setup* form looks like the following:

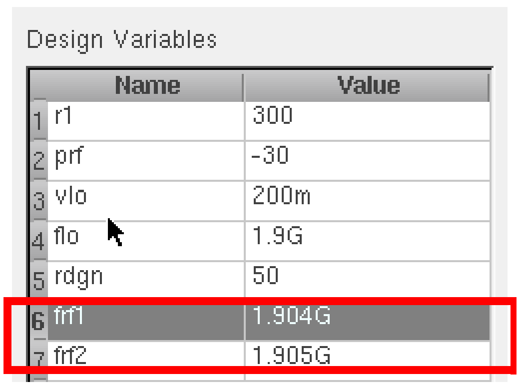
Figure A-413 Model Library Setup Form



6. Click *OK* to close the Model Library Setup form.

Setting the Design Variables

Figure A-414 Design Variables Section

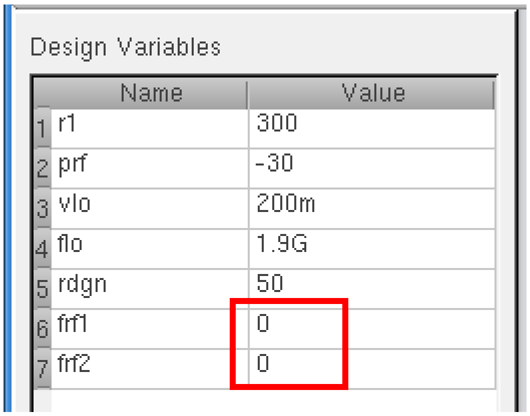


1. In the *Design Variables* section in the Virtuoso Analog Design Environment window, click *frf1*.

2. Select the value in the box to the right of the variable *frf1*. Type *0* (zero) and press *Enter*. Do the same for the variable *frf2*. Setting the input frequencies to 0 disables the production of waveforms for the large-signal analyses like tran, pss, and hb (harmonic balance).

   The *Design Variables* section looks like the following:

Figure A-415 Design Variables Section of ADE



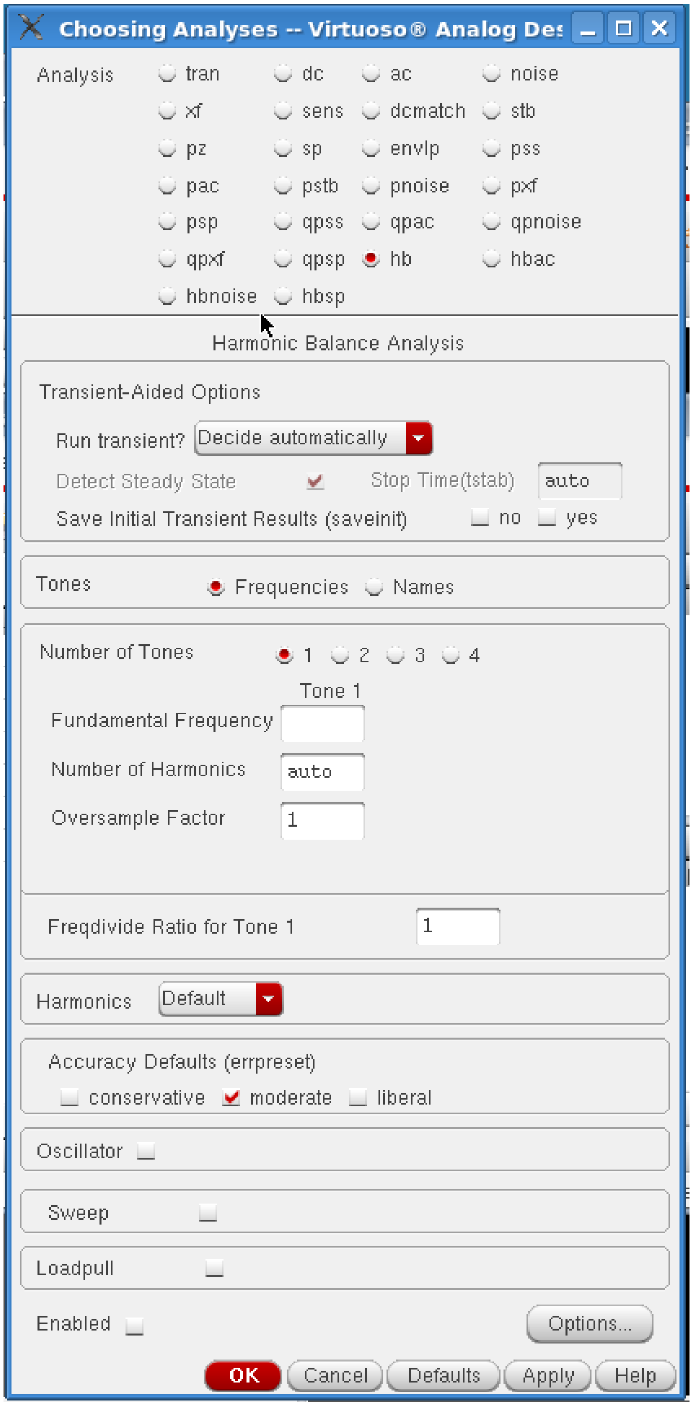
Seting up the HB analysis

1. Click the *Choosing Analyses* icon ()on the right side of the ADE L window.

   The Choosing Analyses form is displayed.

2. Select *hb* in the *Analysissection*. The form expands, as shown in the following figure:

Figure A-416 hb Choosing Analyses Form



3. Fill in the form as follows:

Harmonic balance can set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient* selection in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected,and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

In the Transient-Aided Options section of the form, select the following:

a. For *Run transient?* select *Decide automatically*. (this is the default)

Run transient? will run the LO signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

b. For *Stop time (tstab), auto* is automatically populated in the field.

When *auto* is selected for *Stop time*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

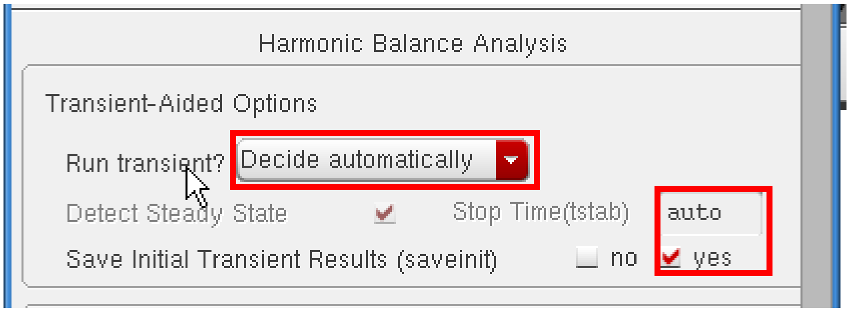
When *Run transient?* is set to *Decide automatically*, the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

c. For *Save Initial Transient Results (saveinit),* select *yes*.

During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. The LO signal in *Tone1* is enabled for this measurement. .At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations

All the signals are applied and the simulation is done in the frequency domain.Only the signal and its harmonics are calculated.

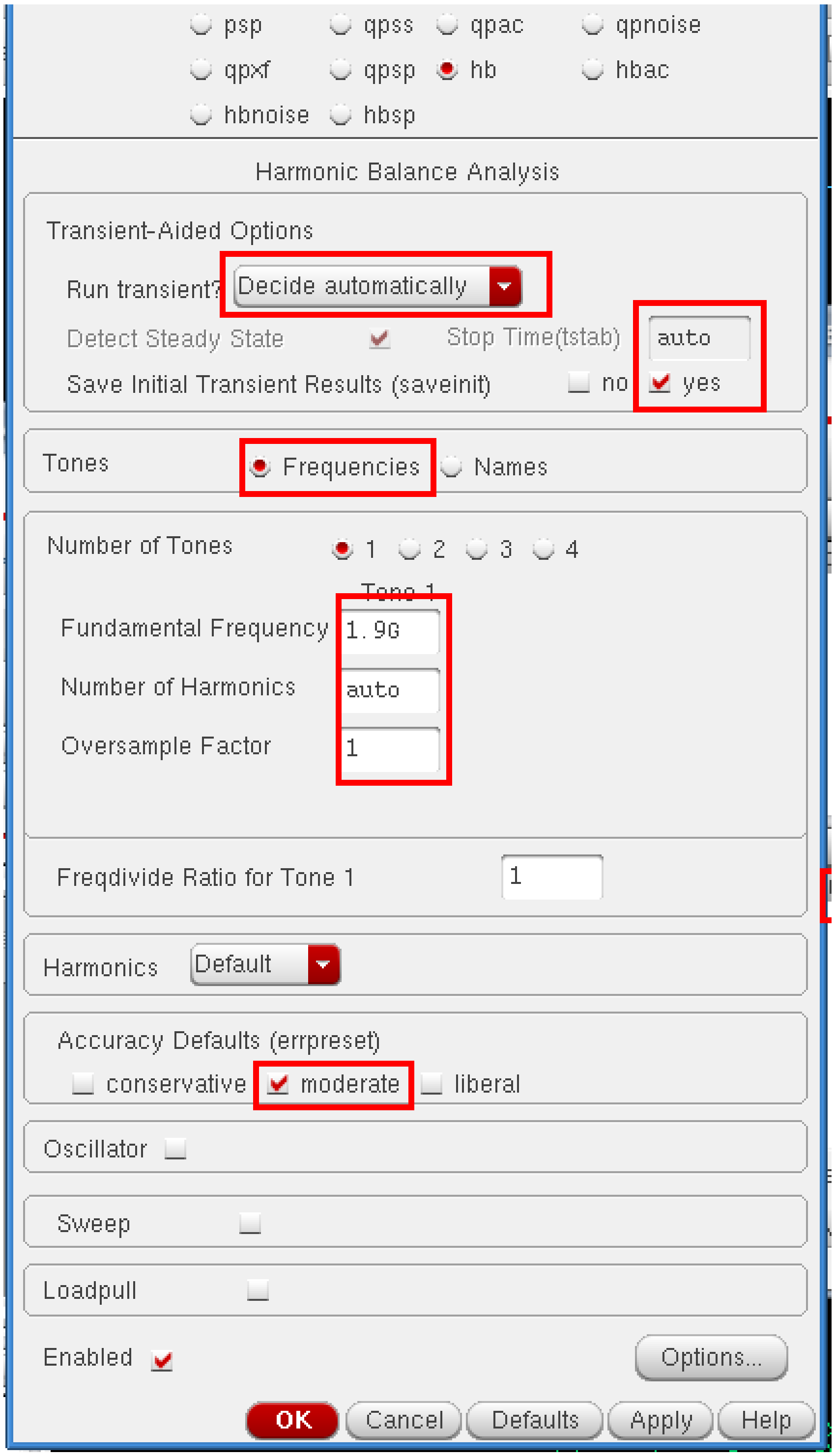
Figure A-417 Transient Assisted Harmonic Balance



4. In the *Tones* section, choose *Frequencies* (this is the default).
5. *Number of Tones* is set to *1* by default. The only large signal tone in this simulation is the LO tone at 1.9GHz.
6. Set the *Fundamental Frequency* to *1.9G*.
7. Leave the *Number of Harmonics* set to the default value of *auto*.
8. Leave *Oversample Factor* set to the default value of *1*. When using the autoharmonics feature, you do not have to set *Oversample Factor.*
9. In the *Accuracy Defaults (errpreset)* section, select *moderate*. For most typical measurements *errpreset* should be set to *moderate*. When you need to measure really small distortions, then *conservative* would be used.

   The *Choosing Analyses* form should look like the following:

Figure A-418 hb Choosing Analyses Form Set-up for Distortion Measurement

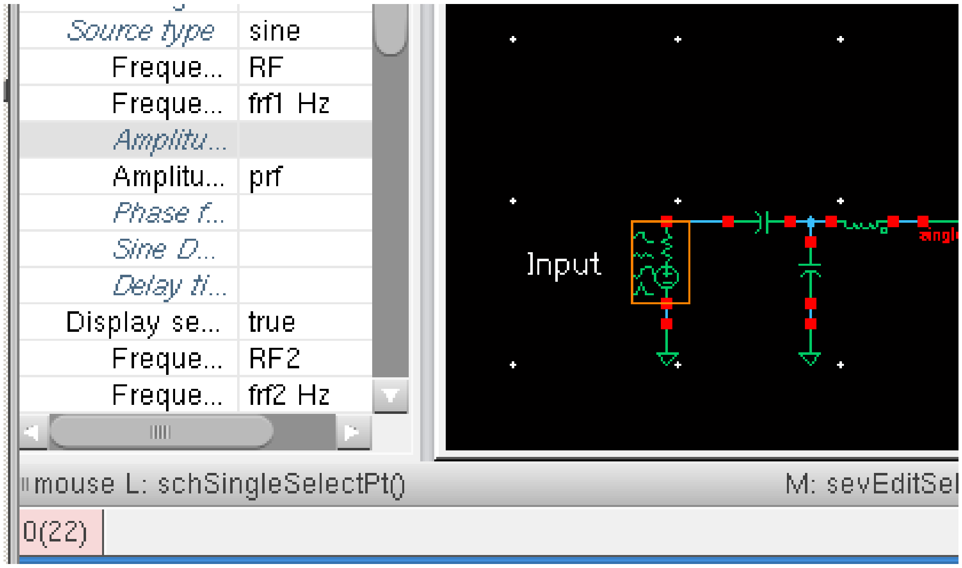


10. Click *Apply*.

11. In the Schematic window, set up the input port to perform a compression measurement.

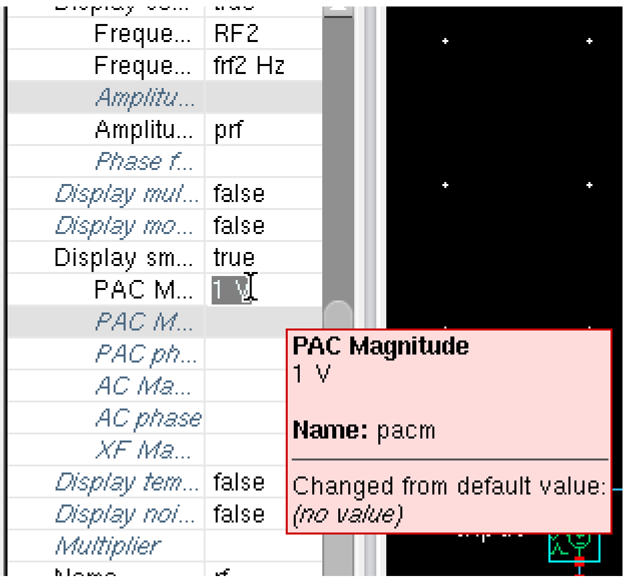
In the schematic, select the *Input* port. Once the input port is selected, the *Property Editor* on the left side of the schematic is populated with the properties of the selected instance, as shown below.

Figure A-419 Setting Up Input Port for Compression Distortion Measurement



12. Search the *Property Editor* (scroll down) for the *PAC Magnitude* entry.

Figure A-420 Editing Properties on Input Port for Compression Distortion Measurement

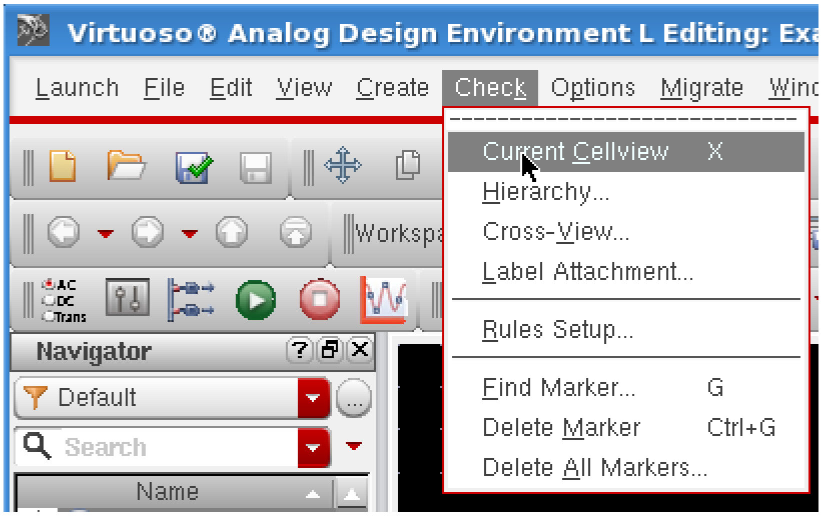


13. View the entry for *PAC Magnitude*. If set to *prf*, do not change anything. (If set to *1* volt, select the entry, press the *Tab* key, and then set *PAC Magnitude (dBm)* to *prf*. The port needs to be set up to supply a signal in the small-signal region because the HBAC analysis which is a small-signal analysis will be used to calculate the distortion.

The amplitude of the input signal is specified on the input port. For the distortion summary, the *PAC Magnitude (dBm)* property is typically used on the input port. This is the amplitude that is used as the input for the distortion measurement. The *prf* value is set in the variables section of the Analog Design Environment and is *-30* in this example.

14. Check the schematic without saving it. In the schematic window, choose *Check - Current Cellview.*

Figure A-421 Check Current Cellview Menu

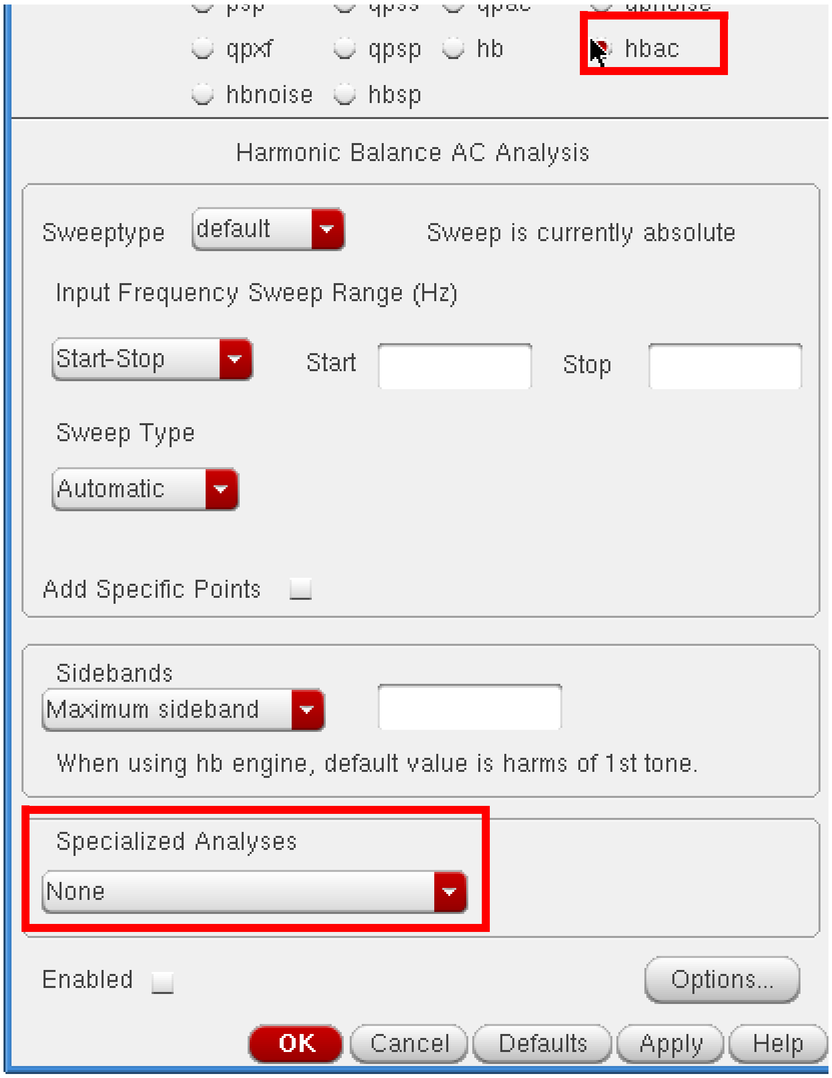


This allows ADE L to netlist from the schematic without saving a copy of the newly modified circuit to the disk. This allows "what-if analysis" without changing the original schematic. When the design gets to the desired performance, then a *Check and Save* can be done. If you want to revert back to the original schematic, quit the schematic without saving it, and then recall it.

Setting up the HBAC form

1. In the *Choosing Analyses* form, select *hbac*. The form expands, as shown below.

Figure A-422 hbac Choosing Analyses Form



2. At the bottom of the form in the *Specialized Analysis* section, select *Compression Distortion Summary*.

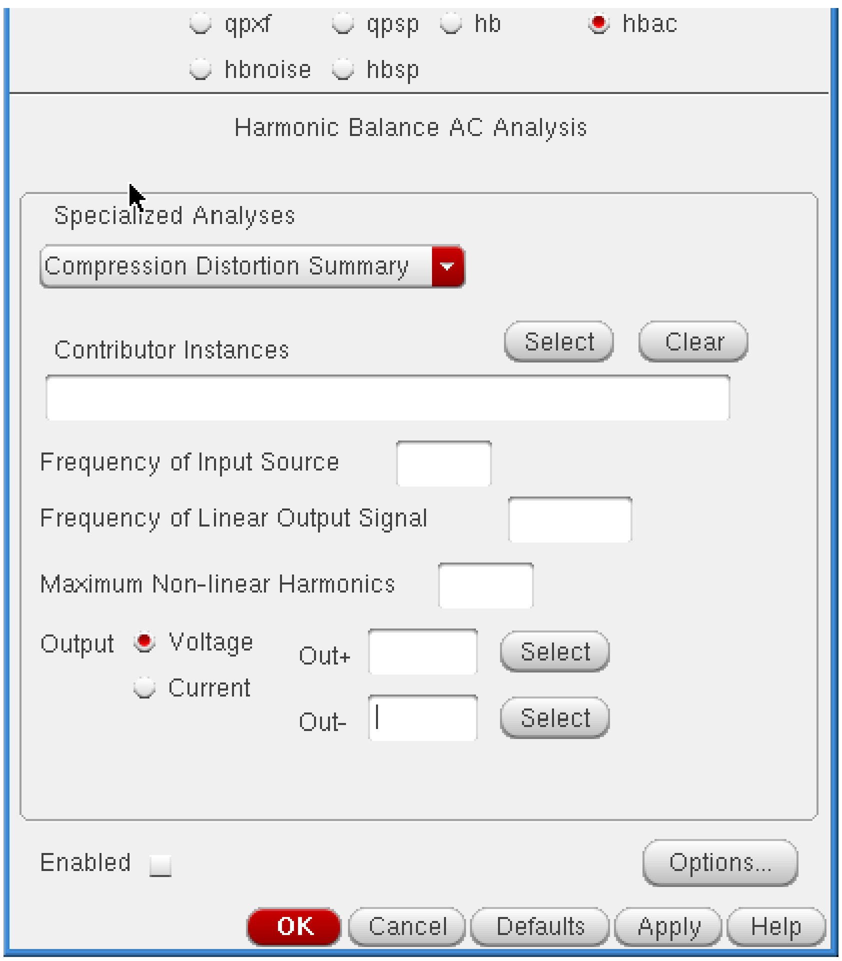
   Adialog box is displayed that reminds you to set the *pacmag* parameter on the RF source.

Figure A-423 Compression Distortion Summary Dialog Box

Since you have already set this parameter in the preceding step, click *Close* to close the dialog box.

3. Set up the *hbac Choosing Analyses* form.

Figure A-424 Compression Distortion Summary Choosing Analyses Form



4. You provide a list of the devices you want the distortion calculation for. If you leave this field blank, all of the nonlinear elements will be run. Since hbac will be run for each device specified, the computation can be time-consuming if the device list is large. You should select only those devices that are important to output. It is not recommended to run distortion summary for all nonlinear devices in the circuit if your circuit is large.

In the *Contributor Instances* field, type */Q1 /Q2 /Q3 /Q4 /Q5 /Q6 /Q7 /Q8*. Note the spaces between the entries. Alternately, you may click *Select* in the and select the components in the schematic.

5. Type *1.905G* in the *Frequency of Input Source* field. This is the RF input frequency.

6. Type *5M* in the *Frequency of Linear Output Signal* field. This is the mixer IF output frequency.

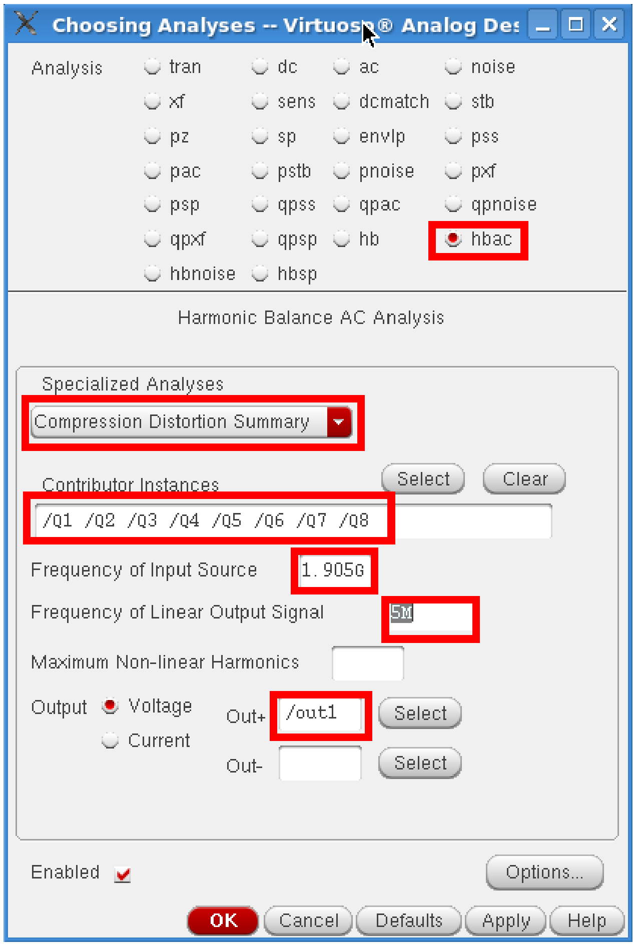
Input and output frequencies are specified as usual. Only a single input frequency is allowed at a time.

7. Leave the *Maximum Non-linear harmonics* field blank. The default value is optimum for the compression distortion measurement.

8. In the *Out+* field, type */out1*. You can also click *Select* to the right of the *Out+* field and choose a net in the schematic. When the *Out-* field is left blank, ADE automatically assigns this to the global ground node.

The *Choosing Analyses* form should look like the following:

Figure A-425 Choosing Analyses Compression Distortion Summary



9. Click *OK* at the bottom of the *Choosing Analyses* form.

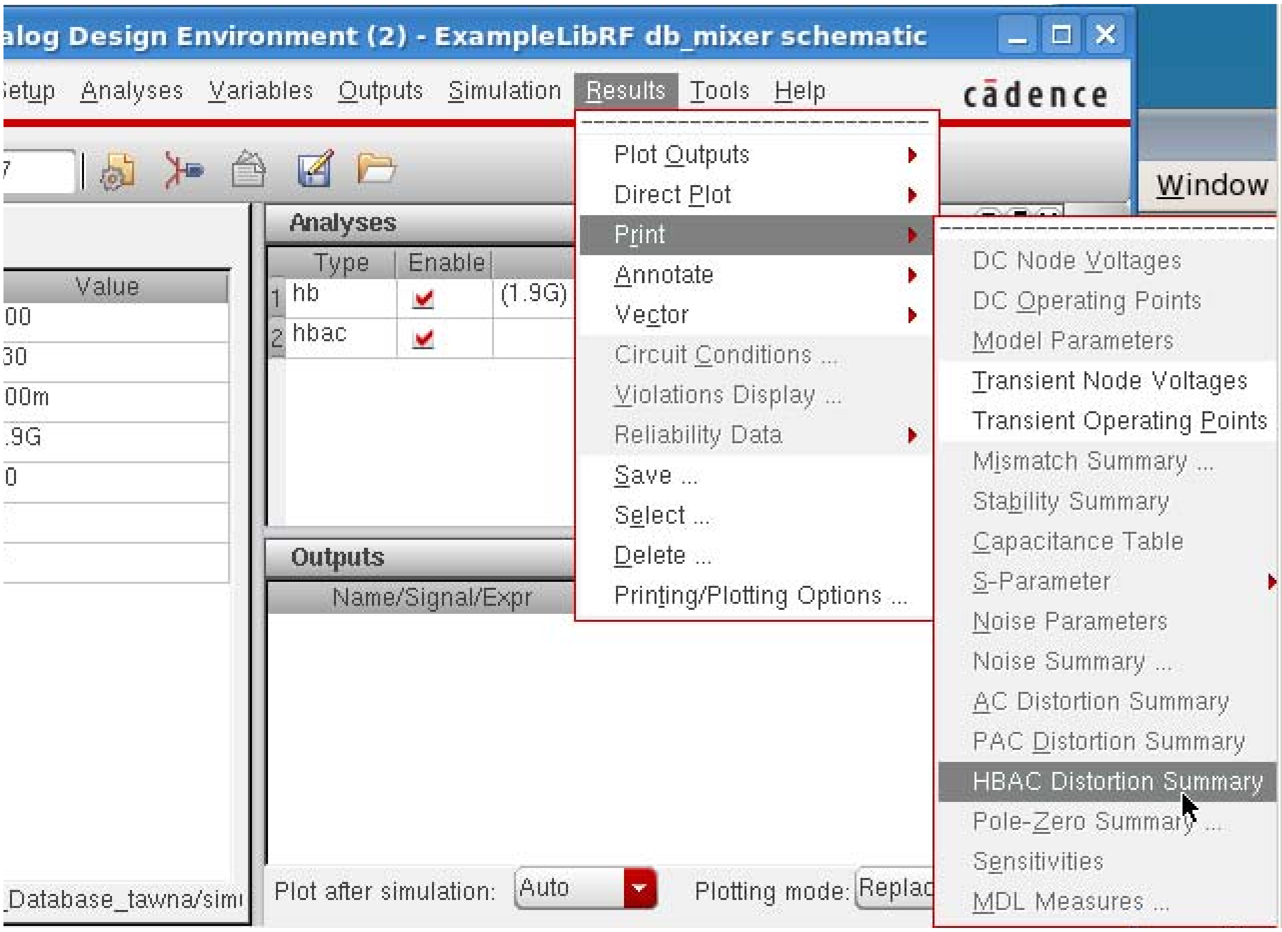
   The distortion measurement is performed along the entire signal path, which includes frequency translation and requires a declaration of the output node in the *Choosing Analyses* form.

   The compression distortion summary provides a relative measure of which components in a gain path contribute more or less compression. Because compression and IP3 are related mathematically, the compression distortion provides information about which components contribute to IP3.Start the simulation. In the ADE L main window or the Schematic, click the green arrow icon ( ) on the right side of the window.

10. The Spectre output window will appear with the simulator status information.

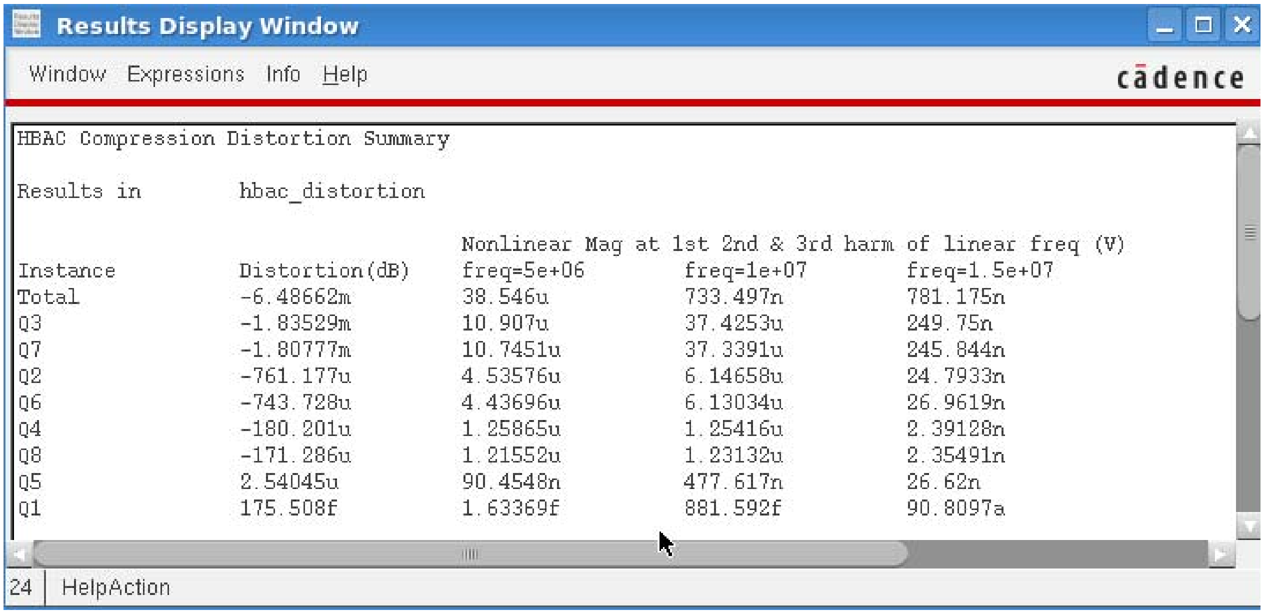
11. After the simulation finishes, choose *Results - Print-HBAC Distortion Summary.*

Figure A-426 Select HBAC Distortion Summary



The *Results Display Window* appears with the distortion contributions from the selected devices in the circuit. This summary gives the list of the various distortion contributors and how much distortion they contribute to the output. (If no devices are selected, all the nonlinear devices will be calculated). It helps identify the distortion sources. You can quickly see what your top distortion contributors are, and you can look up the distortion contribution for a particular device. The measurement in the output is the output level with the distortion included divided by the output level of the ideal system.

Figure A-427 Compression Distortion Summary Results Display Window



HBAC Compression Distortion Summary

Results in hbac_distortion

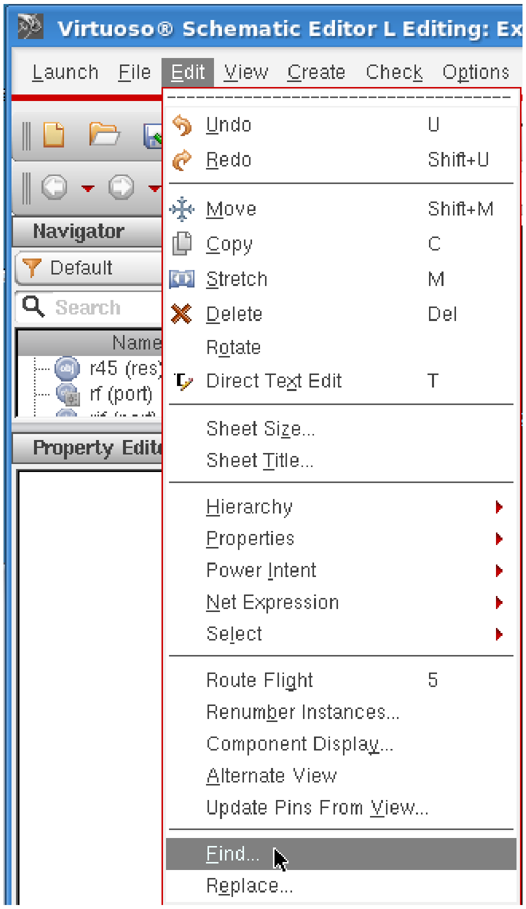
		Nonlinear Mag at 1st 2nd & 3rd harm of linear freq (V)		
Instance	Distortion(dB)	freq=5e+06	freq=1e+07	freq=1.5e+07
Total	-6.48662m	38.546u	733.497n	781.175n
Q3	-1.83529m	10.907u	37.4253u	249.75n
Q7	-1.80777m	10.7451u	37.3391u	245.844n
Q2	-761.177u	4.53576u	6.14658u	24.7933n
Q6	-743.728u	4.43696u	6.13034u	26.9619n
Q4	-180.201u	1.25865u	1.25416u	2.39128n
Q8	-171.286u	1.21552u	1.23132u	2.35491n
Q5	2.54045u	90.4548n	477.617n	26.62n
Q1	175.508f	1.63369f	881.592f	90.8097a

12. Note that *Q3*, *Q7*, Q2, and Q6 have the largest distortion contributors.

13. Find Q3 in the schematic.

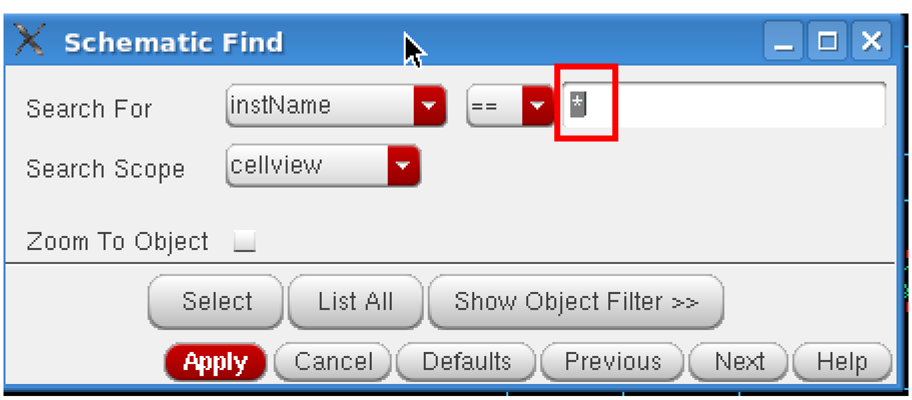
In the schematic window, choose *Edit - Find.*

Figure A-428 Edit-Find to bring up Schematic Find Form



The *Schematic Find* form is displayed, as shown below.

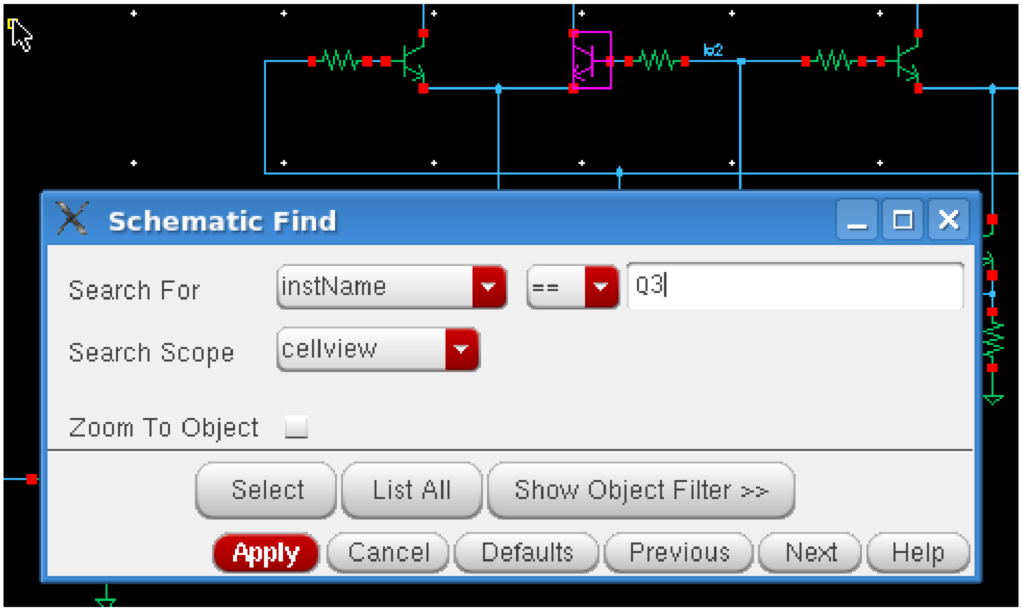
Figure A-429 Schematic Find Form



14. Delete the asterisk, type *Q3*., and click *Apply*.

Q3 is highlighted in the Schematic window, as shown below.

Figure A-430 Finding Sources of Distortion, in the Schematic Design.

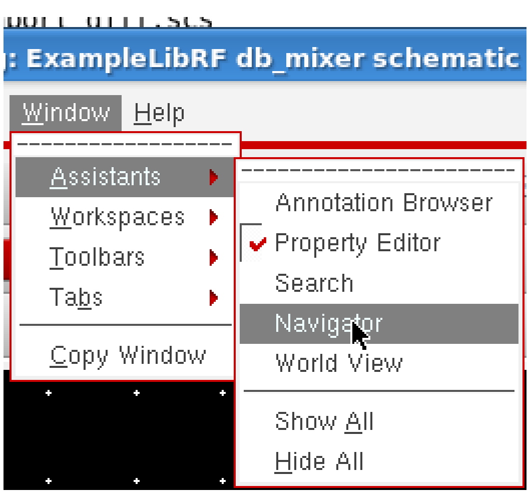


15. In a similar manner, find *Q2*, *Q6*, and *Q7*.

In the *Schematic Find* window, click *Cancel*.

16. Click in the graphics area away from any component so the input port is deselected.

17. If you deleted the Navigator in an earlier step, recall the Window Navigator by choosing *Window - Assistants - Navigator*.



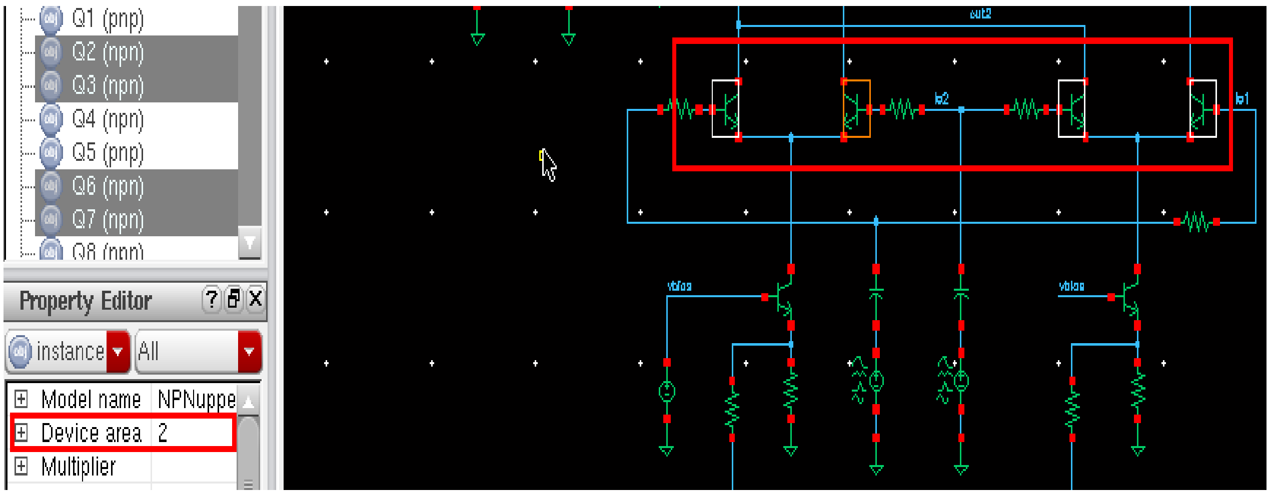
18. While holding the *Shift* key, select the four instances you found in the search. You should see *Q3*, *Q2*, *Q6*, and *Q7* highlighted in the *Navigator Assistant* and in the schematic.

19. In the *Property Editor* assistant, type *2* in the *Device area* field.

20. Press *Enter.*

The *Navigator Assistant* and the *Property Editor* on the right side of the schematic should look like the following:

Figure A-431 Navigator Assistant Showing Four Selected Transistors



21. Inthe schematic window, choose*Check - Current Cellview.* (This is to do a “what-if analysis” and not overwrite the Schematic)

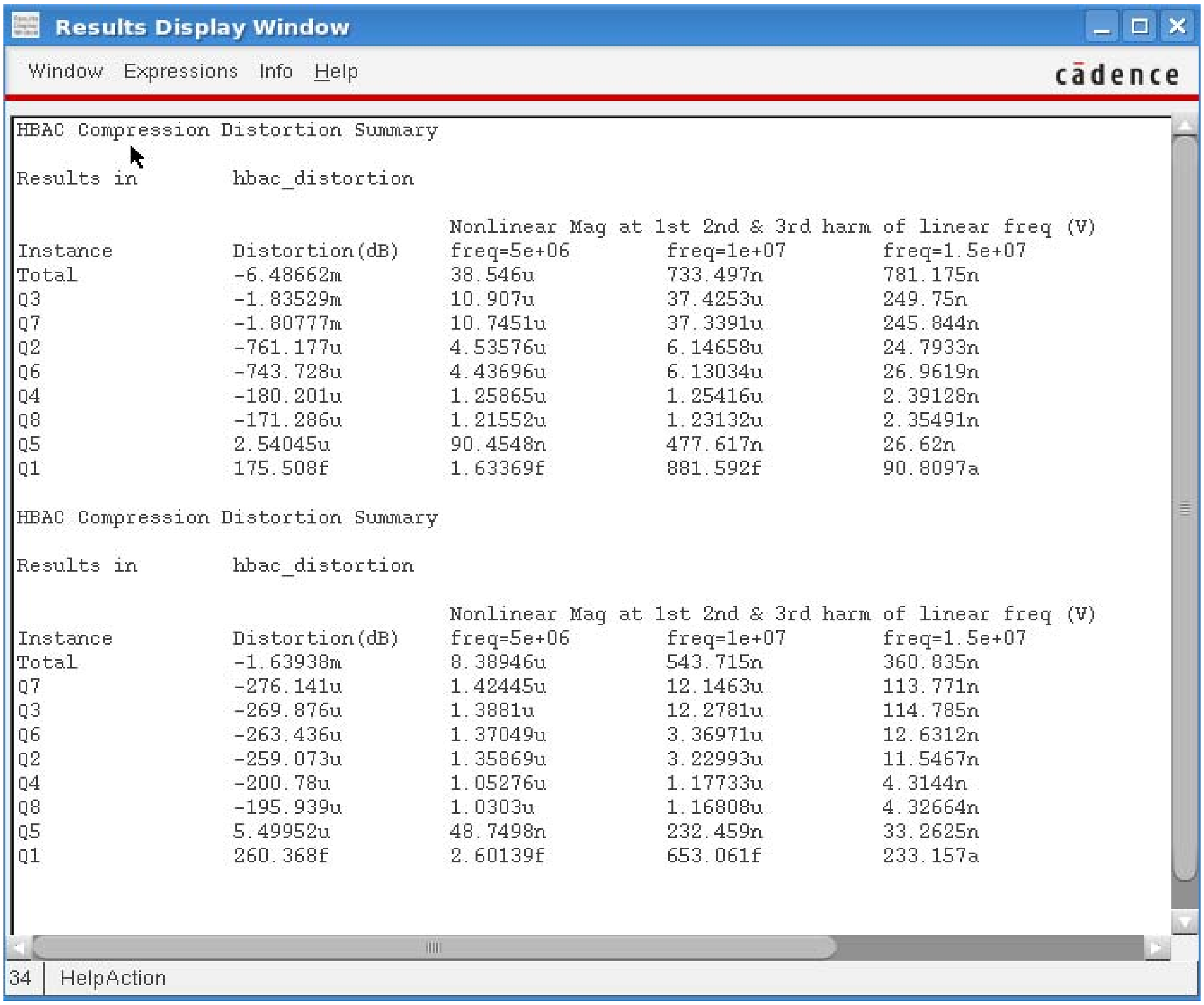
22. Start the simulation. From the ADE L main window or the Schematic, click the green arrow icon ( ) on the right side of the window.

23. The Spectre output window will appear with the simulator status information.

24. After the simulation finishes, choose*Results- Print- HBAC Distortion Summary.*

25. The Results are appended in the *Results Display Window* at the bottom of the list.

Figure A-432 Results Display Window -Two HBAC Compression Distortion Summary Simulations



Note that *Q3*, *Q7*, *Q6*, and *Q2* are still the largest distortion contributors. The distortion is lower with the Device area = `2`, as is the Total Distortion. This demonstrates the ability to do a “what-if” analysis changing the size of the NPN differential pairs.

Close the Analog Design Environment and the Schematic Window

1. In the ADE-L window, choose *Session - Quit.*
2. Click *No* in the *Save State* form.
3. In the schematic window, choose *File - Close All.*
4. Click *No* in the *Save Changes* window.

Summary

You have used hb and hbac analyses to perform Mixer Distortion Measurements. You used Compression Distortion Summary to locate the distortion contributions from various devices in the design.

This concludes the section on Receive Mixers. For more information on simulating receive mixers, please referto the chapters in this user guide. Inaddition, see *Virtuoso Spectre Circuit Simulator RF Analysis Theory Guide*.

The remaining sections focus on simulating Transmit Mixers.

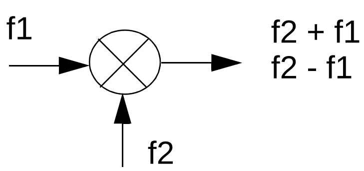
Setting Up to Simulate the db_mixer_xmit Mixer

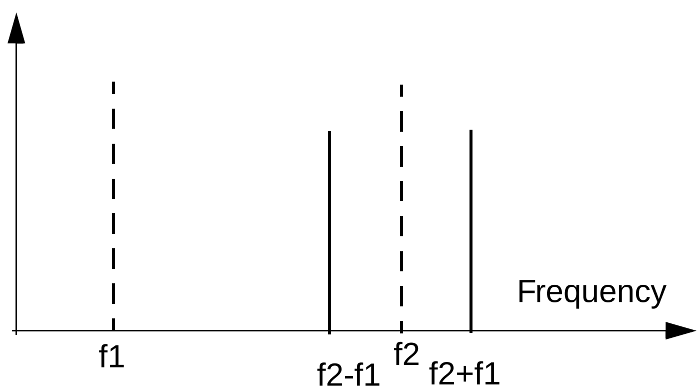
In this workshop, you will make common measurements for an up-converting mixer.

The `db_mixer_xmit` circuit is found in the *ExampleLibRF* library. The `db_mixer_xmit` integrated circuit consists of two Gilbert cell (up-converting double-balanced) mixers. Looking at the very left side of the schematic are two ports labeled *Input_top* and *Input_bottom* which generate the input signals. Both inputs feed the input to two double-balanced mixers through two voltage controlled voltage sources. There are four LO sources in the circuit between the bottom devices of the respective mixers. The LO operates at 1.9GHz. Next to the label *Output*, is the output port of the mixer.

Image reject mixers use phase cancelling techniques and remove one of the two major mixer products from the output of the mixing or multiplication process.

When two RF signals are mixed (*f1* and *f2*), the sum and difference signals are produced, as shown below.





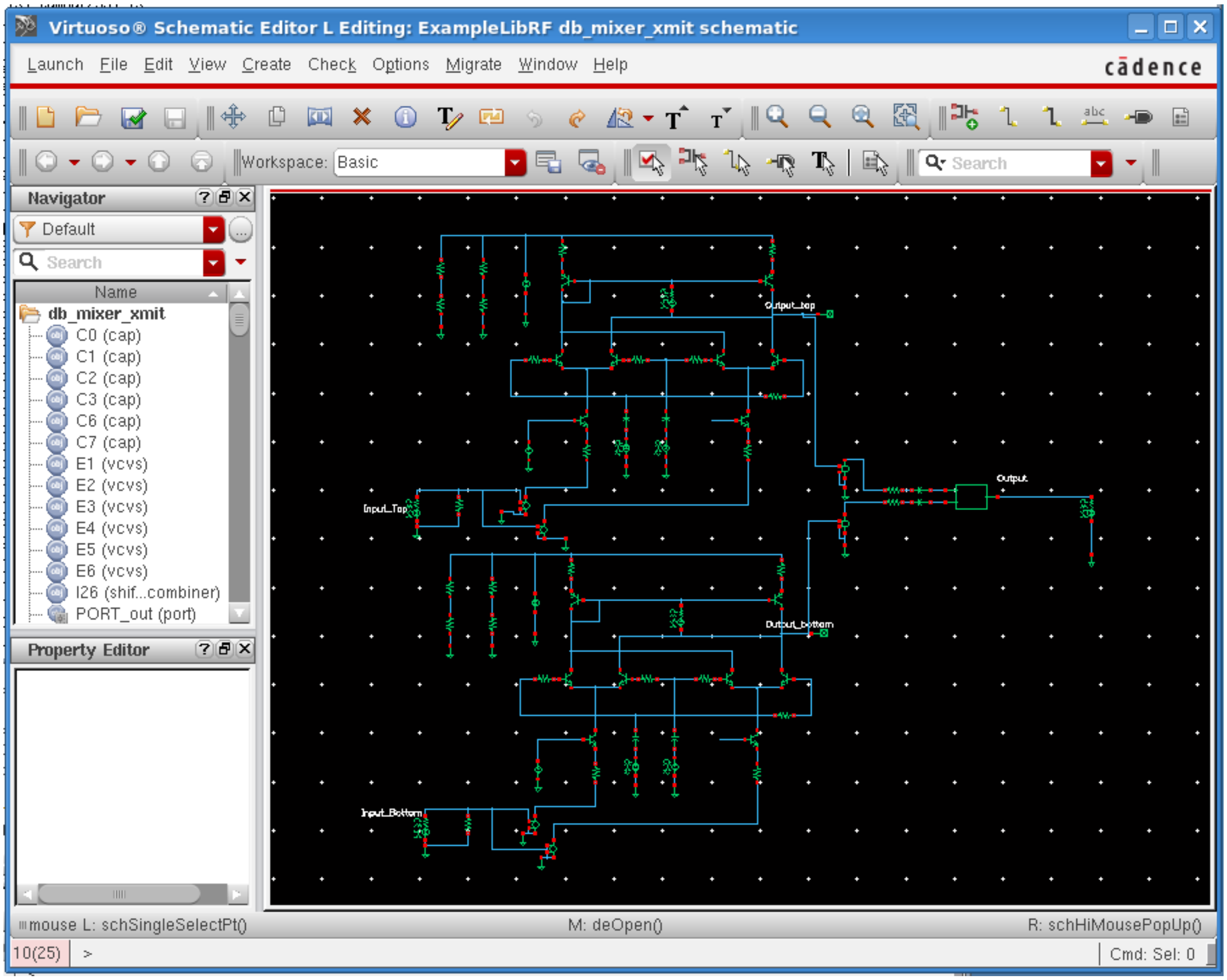
The output signals are (`f1 + f2`) and (`f1-f2`).

Of the two products from a mixer, normally only one is required. Often, the unwanted one will fall outside the required bandwidths and can be removed very easily. If the unwanted or image product is close to the desired signal, it can require complicated filtering to remove it sufficiently.

Image rejection mixers utilize phasing techniques to cancel out the unwanted mix products.

In the figure below, is the schematic of the upconverting mixer `db_mixer_xmit`. Following that is the list of measurements you will make in this module.

Figure A-433 db_mixer_xmit Schematic



Transmit Mixer Measurements (db_mixer_xmit)	**Analyses**
Three Tone Spectral Content and Image Rejection	HB
Three tone Swept IP3 (large signal)	HB
Noise Figure/Signal to Noise Ratio	HB and HBnoise

Measurement	**Measured**
LO frequency (Hz)	1.9 GHz
Desired RF frequencies(Hz)	1.904 GHz, 1.905 GHz
Image frequencies	1.895 GHz, 1.896 GHz
IF frequencies (Hz)	4 MHz, 5 MHz
LO voltage	200mV peak
IF power	-9 dBm
Image rejection	*measurement needed*
Input IP3 (from swept power)	*measurement needed*
Noise figure/SNR	*measurement needed*

Three Tone Spectral Content and Image Rejection

Transmit mixers usually operate at power levels high enough that small-signal analyses should not be used. Therefore, HB is used to capture the actual, large-signal behavior. When measuring three-tone spectral content, all signals (LO, IF1, and IF2) are applied to the circuit. Harmonic balance is used, rather than transient analysis, because it is quicker.

A conventional transmit mixer has two output responses at points above and below the LO frequency at `f1+f2` and |`f1-f2`|. The unused response, known as the image frequency, can be suppressed by an image-reject mixer. For the first measurement, you will check the output spectrum and view the frequency response to determine the image rejection. You will also look at IP3 measurements, which is a common figure of merit for mixers. For this circuit, the desired response is at 1.904G and 1.905G.

Setting Up the Simulation

Open the *db_mixer_xmit* Mixer Circuit in the Schematic Window, as follows:

1. In the CIW, choose *File – Open.*

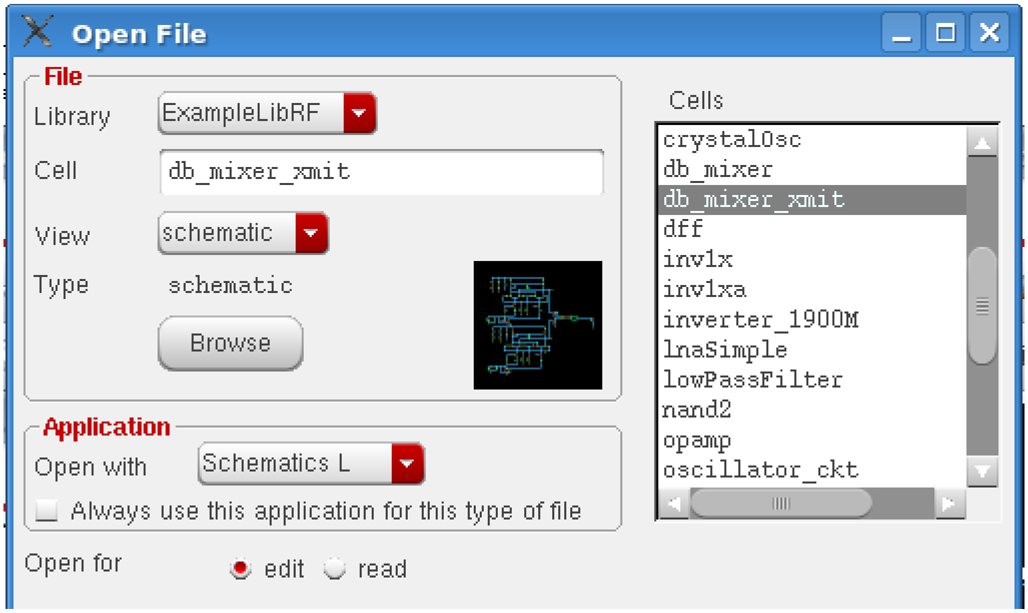
The Open File form is displayed.

2. In the *Open File* form, choose *ExampleLibRF* from the *Library* drop-down list.

3. Choose *db_mixer_xmit* from the *Cells* list box.

The completed *Open File* form looks like the following:

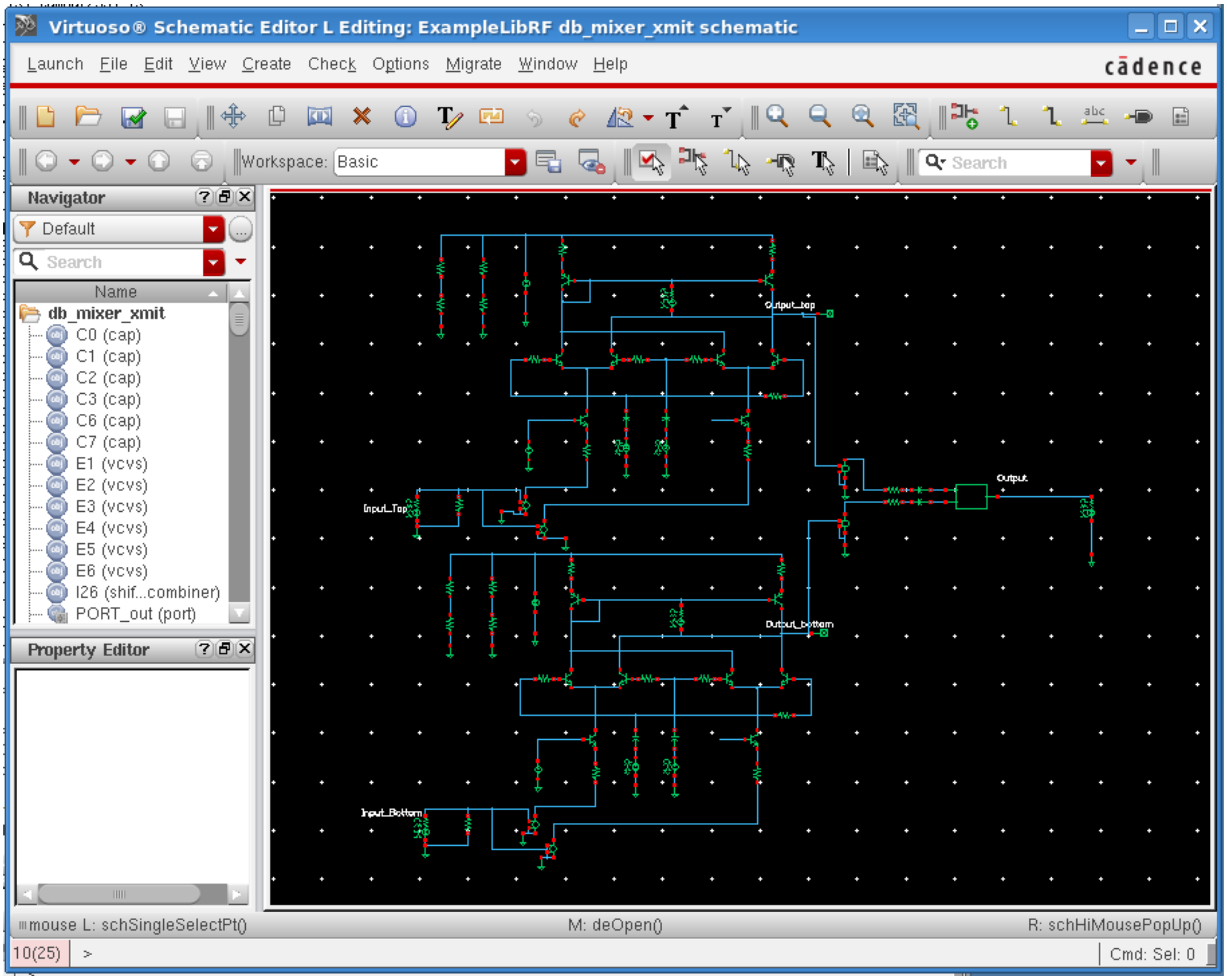
Figure A-434 Open File Form



4. Click *OK*.

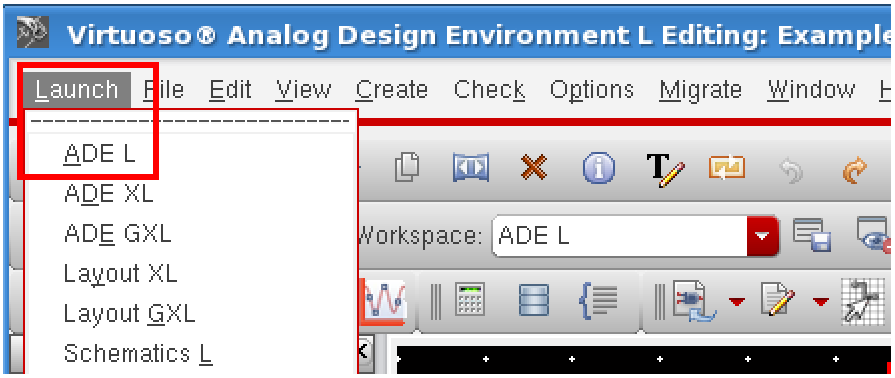
   The Schematic window for the `db_mixer_xmit` mixer is displayed, as shown below.

Figure A-435 db_mixer_xmit Schematic



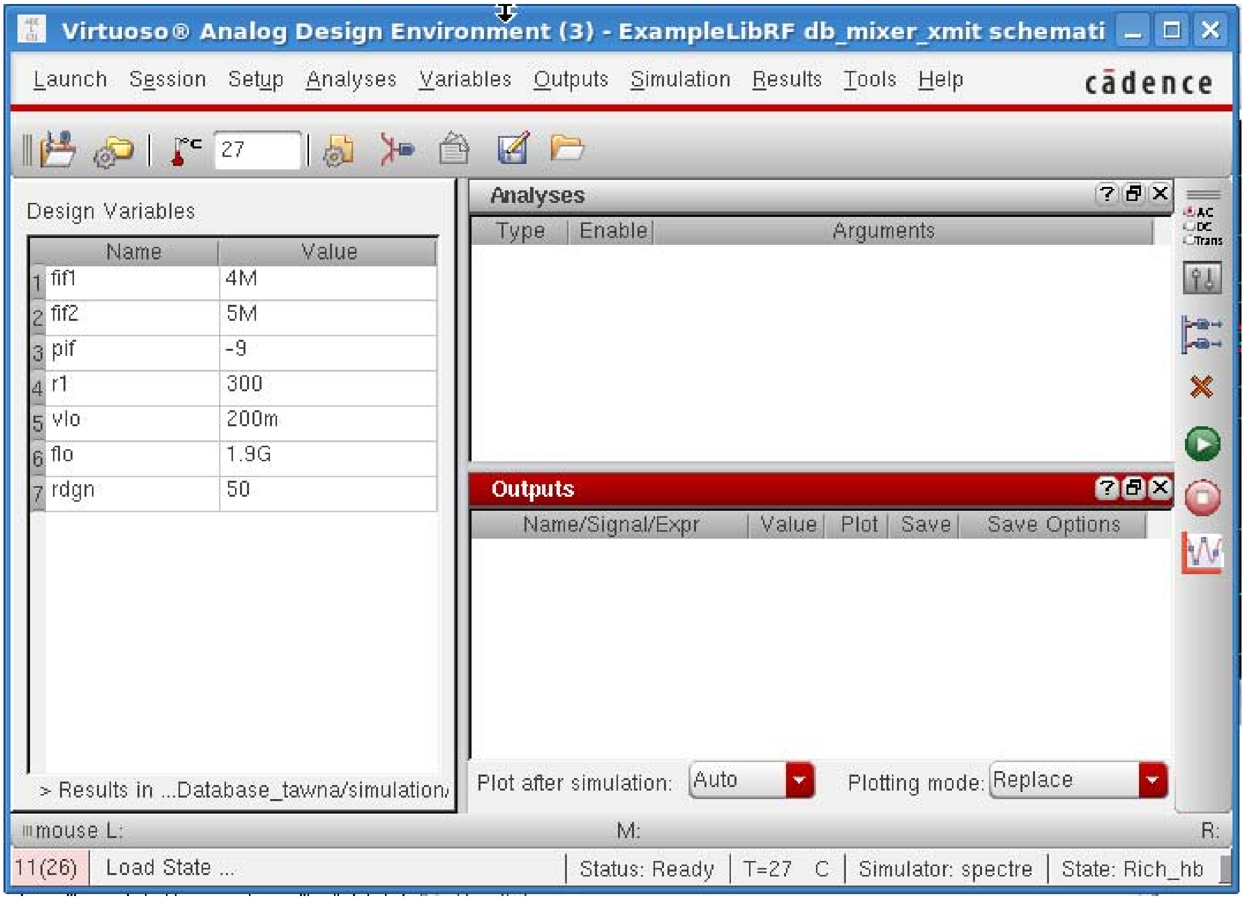
5. In the Schematic window, choose *Launch – ADE L.*

Figure A-436 Launching ADE-L from Schematic Window



The Virtuoso Analog Design Environment window is displayed, as shown below.

Figure A-437 Analog Design Environment Window



Choose the simulator options, as follows:

6. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

The *Choosing Simulator/Directory/Host* form is displayed.

7. Specify the following in the *Choosing Simulator/Directory/Host* form:

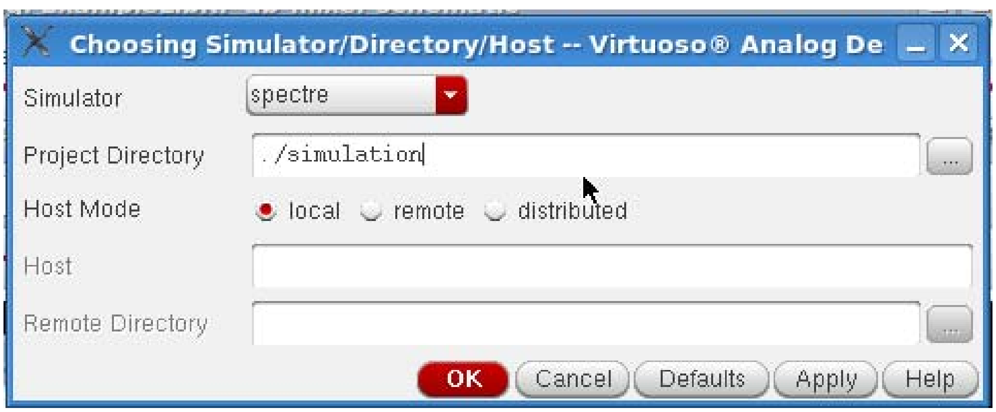
a. Choose *spectre* from the *Simulator* drop-down list.

b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory.

c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

The completed form looks like the following:

Figure A-438 Choosing Simulator/Directory/Host Form

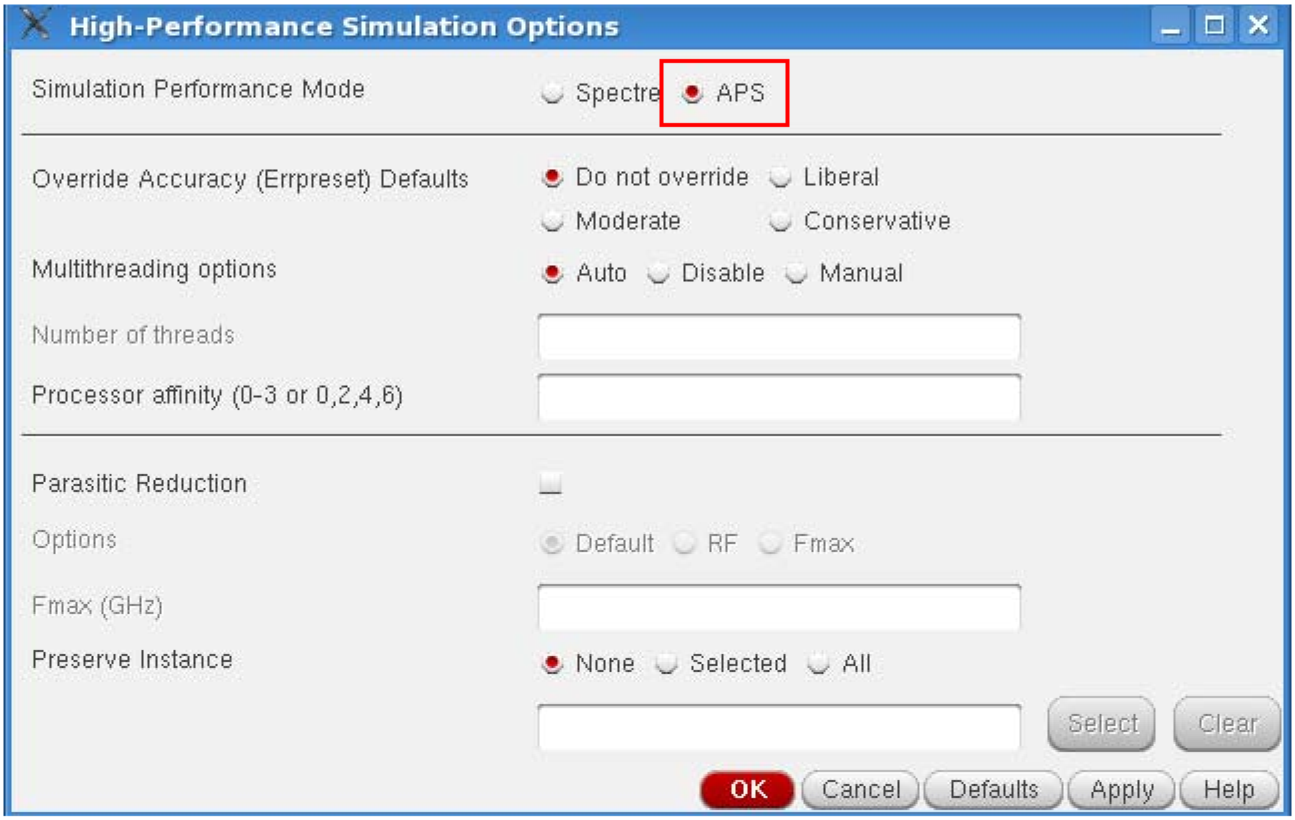


8. Click *OK* to close the *Choosing Simulator/Directory/Host* form.

9. Set up the High Performance Simulation Options, as follows:

d. In the ADE window, choose *Setup -> High Performance Simulation*. The *High Performance Simulation Options* window is displayed, as shown below.

Figure A-439 High Performance Simulation Options



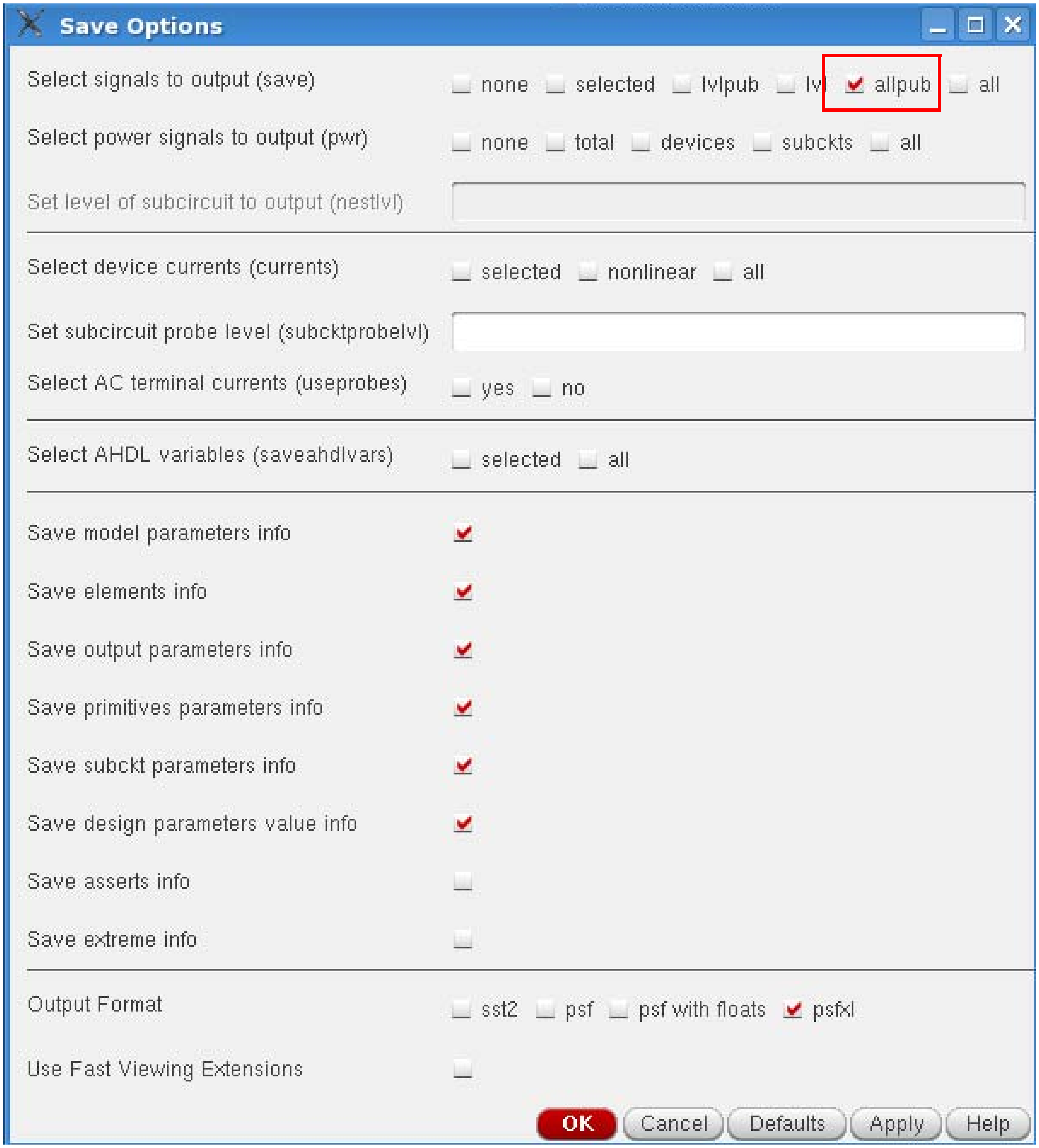
e. In the *High Performance Simulation* window, select *APS* as the *Simulation Perfomance Mode.* Note that *Auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores.

f. Click *OK*.

10. In the *Virtuoso Analog Design Environment* window, choose *Outputs – Save All*.

The Save Options form is displayed, as shown below.

Figure A-440 Save Options Form



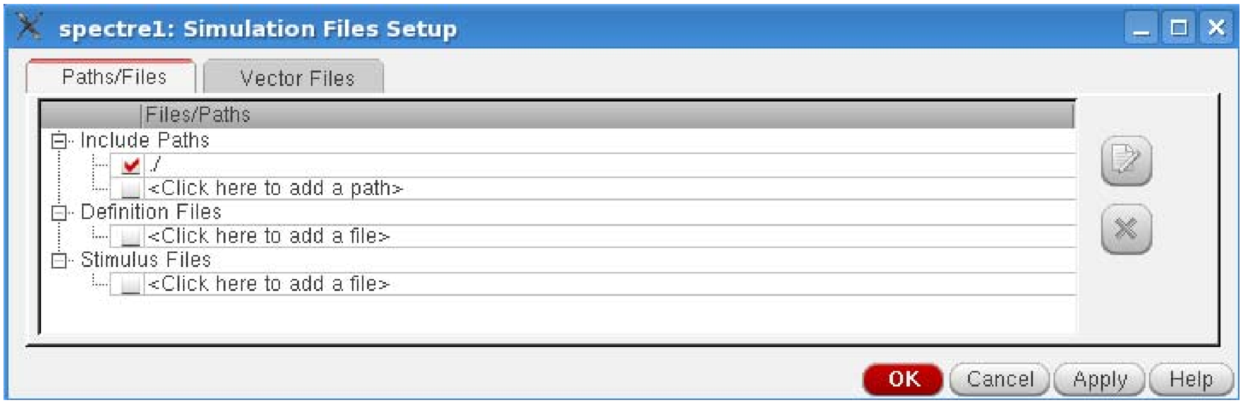
a. In the *Select signals to output(save)* section, make surethat *allpub* is selected. This is the default which saves all node voltages at all levels of hierarchy, but it does not include the node voltages inside the device models.

b. Click *OK* to close the *Save Options* form.

Setup the model libraries, as follows:

11. In the Virtuoso Analog Design Environment window, choose *Setup - Simulation Files*. The *Simulation Files Setup* form is displayed, as shown below.

Figure A-441 Simulation Files Setup Form



12. Ensure that the *Include Path* is set, as shown above.

13. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

The *Model Library Setup* form is displayed.

14. In the *Model Library File* field, typethe following as the name of the model file:

```
models/modelsRF.scs
```

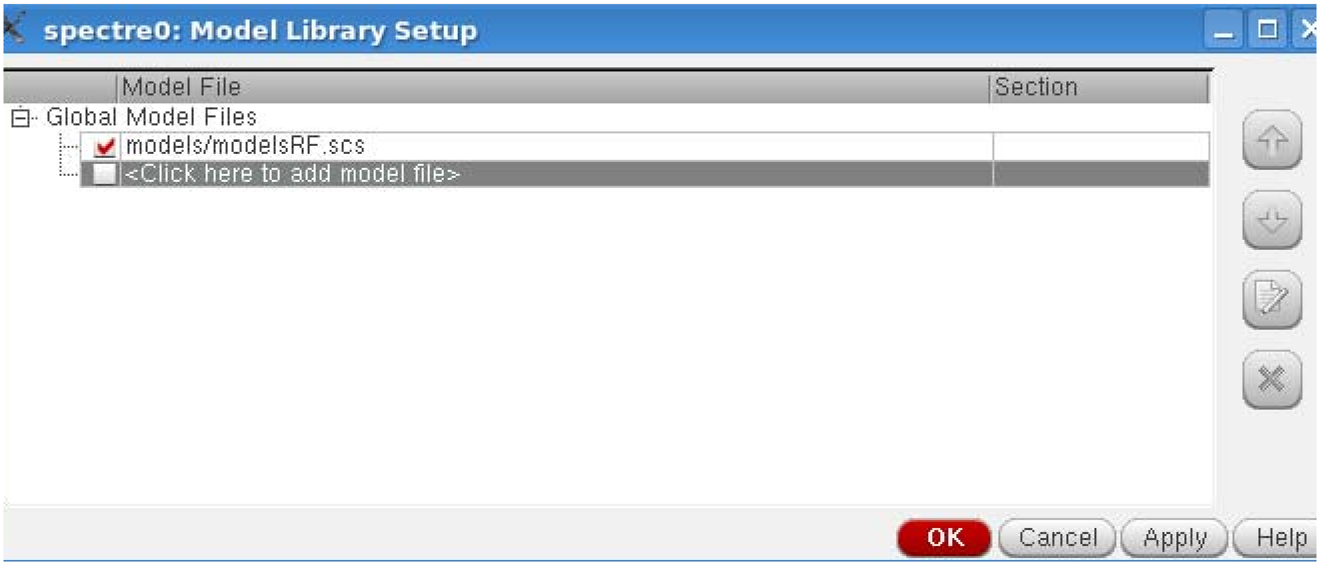
Alternately, you can click *Browse* and browse to the `modelsRF.scs` model file.

15. Make sure thatthe Model File name is selected.

16. Click *Apply*.

The *Model Library Setup* form looks like the following.

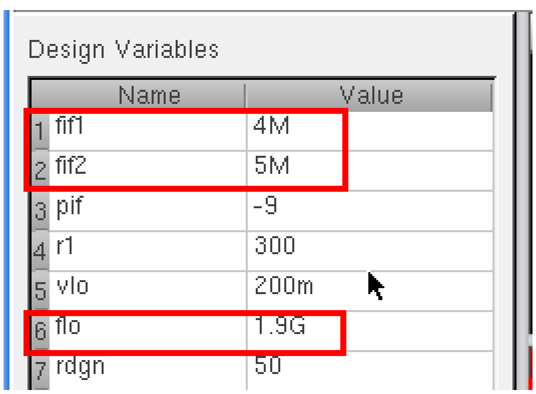
Figure A-442 Model Library Setup



17. Click *OK* to close the *Model Library Setup* form.

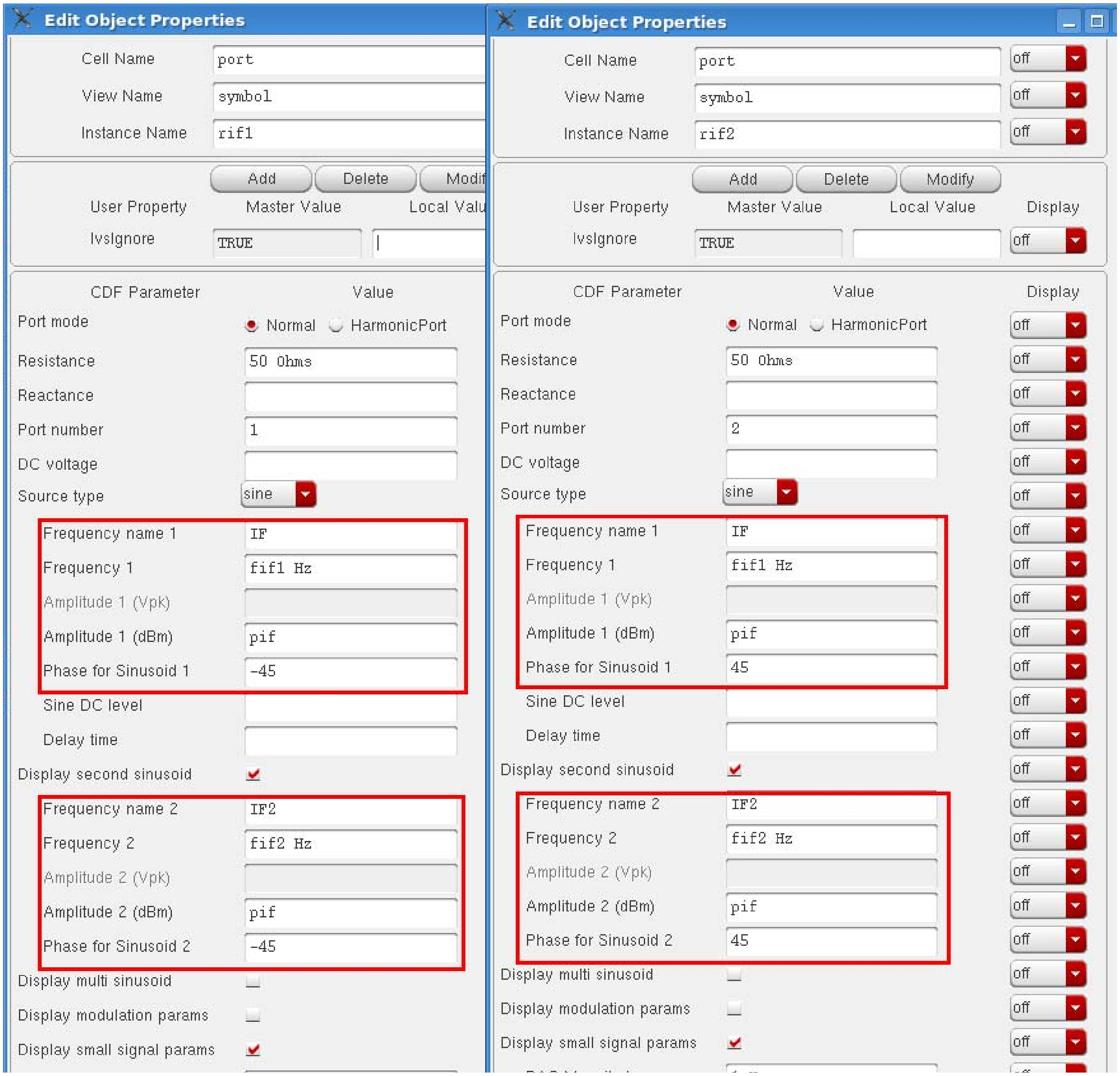
Set the design variables, as follows:

Figure A-443 Design Variables Section of ADE-L Window



1. In the Design Variables section in the ADE L window, verify that the design variables *fif1* and *fif2* are *4M* and *5M*, and *flo* is *1.9G*.If you view the *Edit Object Properties* form for the Input ports (select either the *Input_Top* or *Input_Bottom* port on the left side of the schematic and select the bindkey *q*), you will see how the *rif* tones are specified on the two *rif* ports, as shown below.

Figure A-444 Edit Properties Forms for IF ports



Set up the HB analysis, as follows:

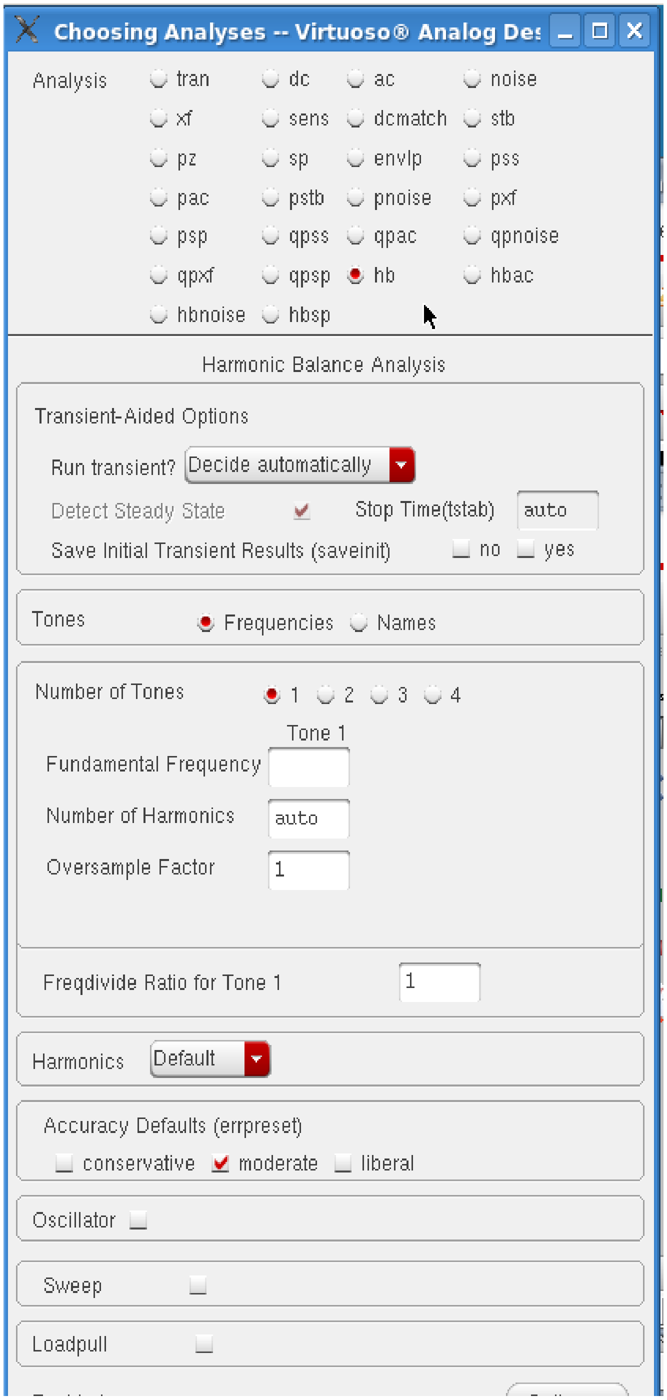
1. Choose *Analyses – Choose* in the *Virtuoso Analog Design Environment* window.

The *Choosing Analyses* form is displayed.

2. In the *Choosing Analyses* form, select *hb.* The form expands. Since the circuit is mostly sinusoidal (not strongly nonlinear) Harmonic Balance is the appropriate analysis to choose.

Figure A-445 The HB Choosing Analyses Form

.



Harmonic balance can set harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient* selection in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected,and then from the transient analysis, the number of harmonics for *Tone1* (when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

3. In the Transient-Aided Options section of the form, select the following

   a. For *Run transient?* select *Decide automatically* (this is the default).

   Run transient? will run the LO signal using the transient (In SpectreRF, this is called the *tstab* interval) for a short period of time. At the end of *tstab*, an FFT is performed, and this is used as the starting point in the harmonic balance analysis. Doing this improves the convergence of hb by giving it a better starting point at the cost of a short transient analysis.

   b. For *Stop time (tstab), auto* is automatically populated in the field.

   When *auto* is selected for *Stop time*, a small number of periods of the LO is run using the transient analysis. During this time, the signal is checked for steady-state conditions. If steady-state is not reached in the initial number of periods, more periods can be added automatically by the simulator. Using this feature allows an accurate FFT for the the starting point in the hb iterations.

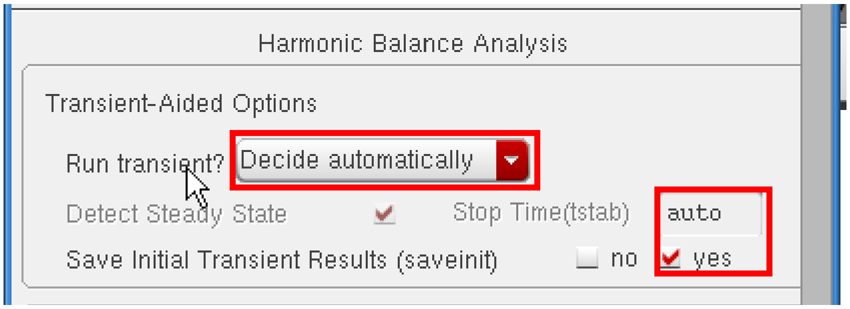
   When *Run transient?* is set to *Decide automatically*, the *Detect Steady State* option is checked automatically. When this is set, when steady-state is detected in the *tstab* interval, the simulator stops the transient analysis, runs the FFT, and starts iterating in the the frequency domain. Using *Decide automatically* simplifies the setup for harmonic balance, and produces correct answers from hb without needing to know how to set hb up manually.

   c. For *Save Initial Transient Results (saveinit),* select *yes*.

   During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. The LO signal in *Tone1* is enabled for this measurement. .At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations

   All the signals are applied and the simulation is done in the frequency domain. Only the signal and its harmonics are calculated.Transient Assisted Harmonic Balance.

Figure A-446 Transient-Assisted Harmonic Balance



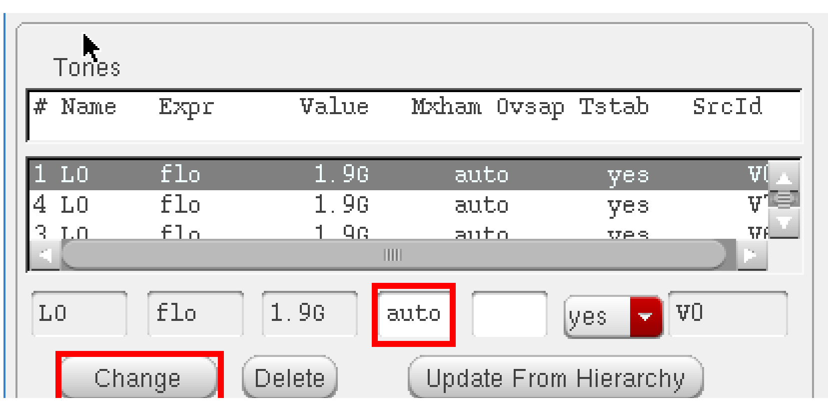
4. In the *Tones* section, select *Names*. When *Names* is selected, the *Tones* portion of the form expands. All the sources in the top-level schematic are read into the form automatically.

5. Select one of the LO sources in the *Tones* section. In the *Mxham* field, type in *auto*. Spectre will automatically detect the number of harmonics required. Note that all four LO sources are now updated with a *Mxham* of *auto*.

   You can use hb with up to four signals present in the circuit. In this circuit, there are three tones, the LO and two RF tones. The LO signals have the same name, the same frequency, and are considered a single tone. Whenever you have two signals at the same frequency, make sure you set the *Frequency Name 1* or *Frequency Name 2* property on the source to the same name as it was done in this example. When *Tones* is set to *Names*, the simulator considers both of the sources as a single frequency.

   You viewed the names (*frf1* and *frf2*) in the input port sources in an earlier step.

   Figure A-447 Tones Section of hb Choosing Analyses Form



   When you set the number of harmonics to *auto*, this identifies the signal on which to run *tstab*. This tone should be the LO tone. Only one signal can have transient assist, that being the signal with *tstab* set to *yes*; the signal with auto harmonics set.

Note: If you set the harmonics manually for all of the tones, set *tstab* to *yes* for the LO tone. Because you are using the auto-harmonics feature, you do not need to set the number of harmonics on the LO tone, nor do you need to set *Ovsap (*oversample*).* By default, *Ovsap* is set to *1.*If for some reason, you are not using the auto-harmonics feature, follow these guidelines:

If not using *Mxham* set to *auto* on the LO tone, change the *Maximum harmonics* on the LO tone from *3* to *10*. Place the cursor in the field under *Mxham* and type *10*.

Since the circuit is operating well above the compression point, a higher number of harmonics is advised.

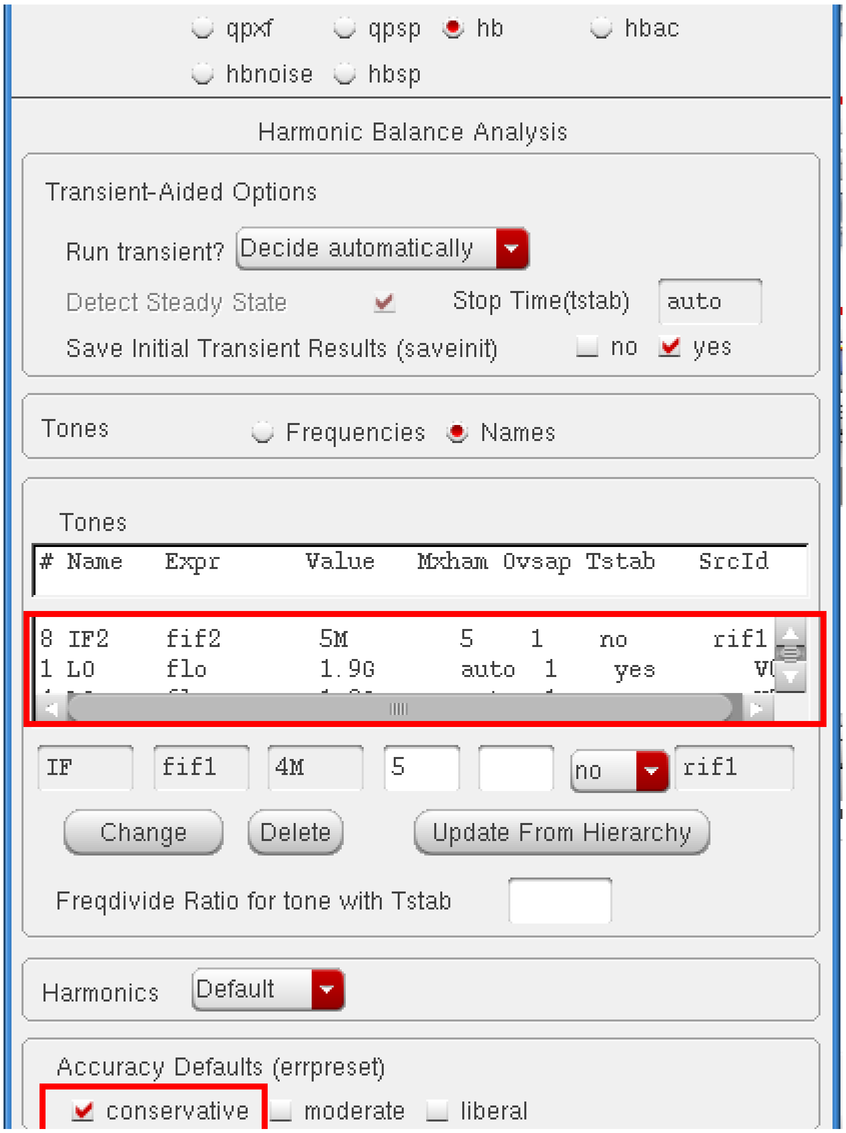
If the circuit is mostly sinusoidal (voltages *and currents*), leave *oversample* set to *1*. For strongly nonlinear circuits, you may need to increase oversample.

In general, when you set harmonics manually, if a signal is below compression, start with three harmonics. If a signal is at compression, start with five harmonics. If a signal is above compression, start with seven harmonics. Run the simulation. Now increase the number of harmonics on each tone separately by about a factor of 1.5, and run the simulation again. If the result changes, increase harmonics again. If the result did not change, the original number of harmonics was enough, and you might be able to decrease the number of harmonics. For fastest runtime with full accuracy, use the smallest number of harmonics for each tone that is applied to the circuit

6. Once finished, click *Change*. The form updates. Make sure that all of the LO tones have *Mxham=auto* and *Tstab=yes*.
7. Select the IF tone in the *Tones* box. For this measurement, three harmonics on the IF tones are not enough. You need to increase the number of IF tones for greater accuracy. Set *Mxham* to *5* . Do the same for IF2. Five (5) was found to be the optimium number of harmonics for the two RF tones, which are deliberately matched in amplitude.
8. Set *Accuracy Defaults (errpreset)* to *conservative*. *conservative* is used because the third and fifth order intermodulation distortion will be calculated with this setup. Since these amplitudes are small, high accuracy is needed.
9. Leave the rest of the form set to the default values.

The *Choosing Analyses* form should look like the following:

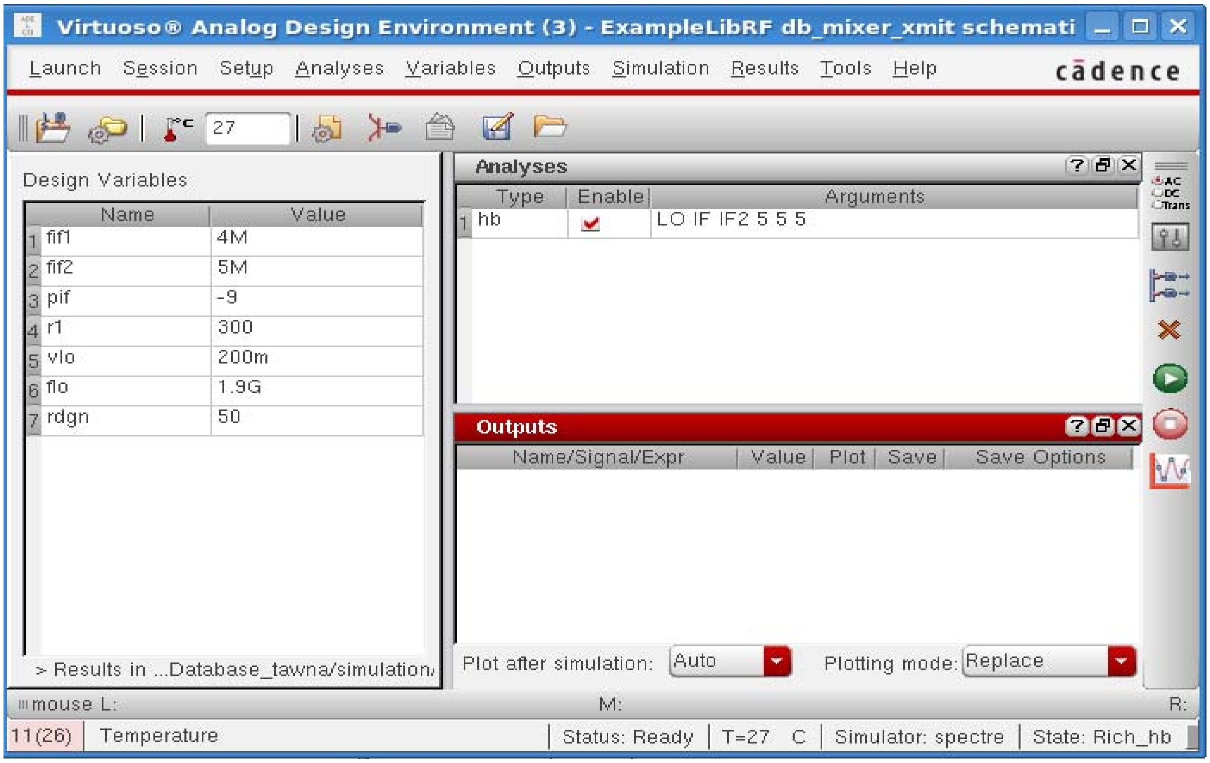
Figure A-448 HB Choosing Analyses Form



10. Click *OK* at the bottom of the *Choosing Analyses* form.

The *Analog Design Environment* window will look like the following:

Figure A-449 ADE Simulation Window



Run the simulation and plot the results, as follows:

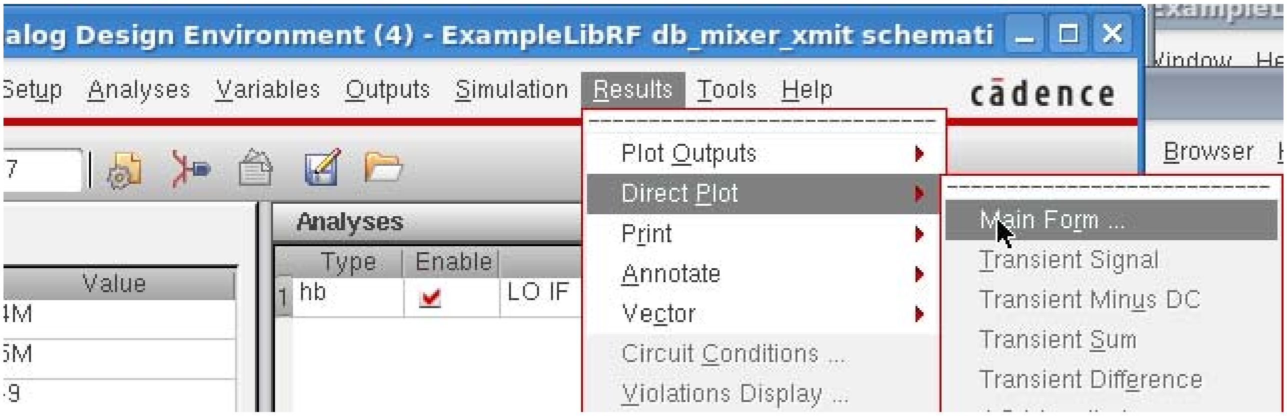
1. Start the simulation by clicking the green arrow icon. in ADE or in the Schematic Editor.

   This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

2. Next, you will plot the Voltage Spectrum.

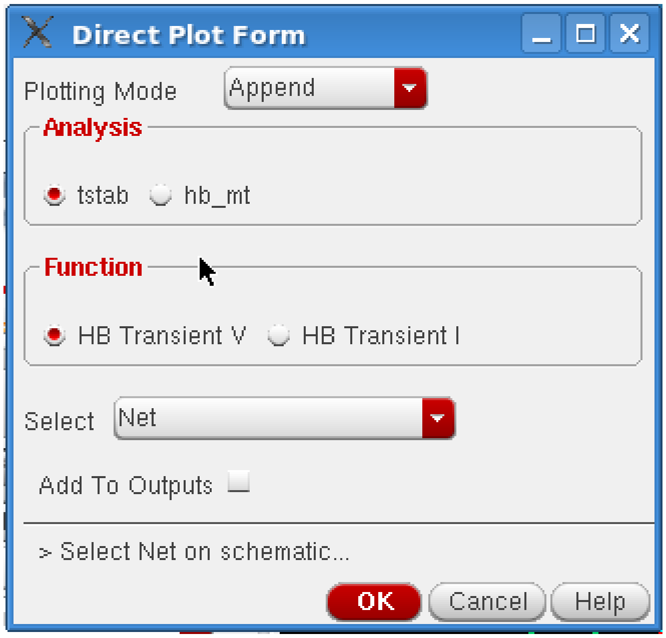
   In the Analog Design Environment window, choose *Results - Direct Plot - Main Form*. Alternately, you can click the *Direct Plot* icon ( )in the schematic window.

Figure A-450 Invoking Direct Plot Form After Simulation



The Direct Plot Form is displayed.

Figure A-451 Direct Plot Form



3. In the *Direct Plot Form*, you will notice that there are two available analyses, *tstab* and *hb_mt*. The tstab results appear because you chose *Save Initial Transient Results" (saveinit)* in the *Choosing Analyses* form. When there is more than one tone in the circuit, you will see *hb_mt*. A single tone simulation will show *hb*.

a. Select *Voltage* in the *Function* section.

b. Select *Net* in the *Select* drop-down list.

c. Select *spectrum* in the *Sweep* section.

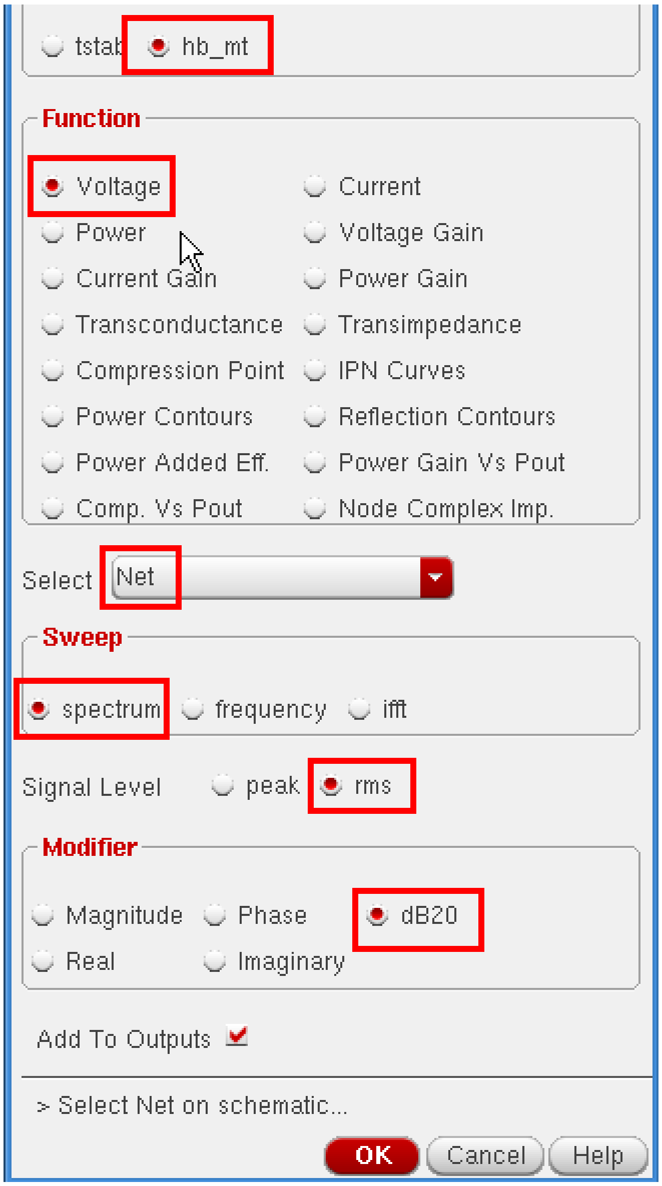
d. Select*rms* in the *Signal level* section.

e. Select*dB20* in the *Modifier* section.

f. If you want to plot this setup in subsequent simulation, Select the *Add to Outputs* option.

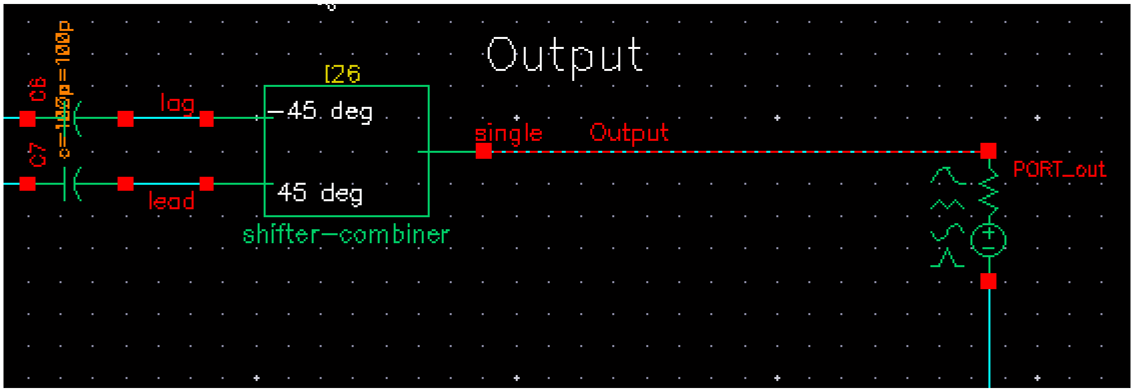
The *Direct Plot Form* is shown below.

Figure A-452 Harmonic Balance Direct Plot Form



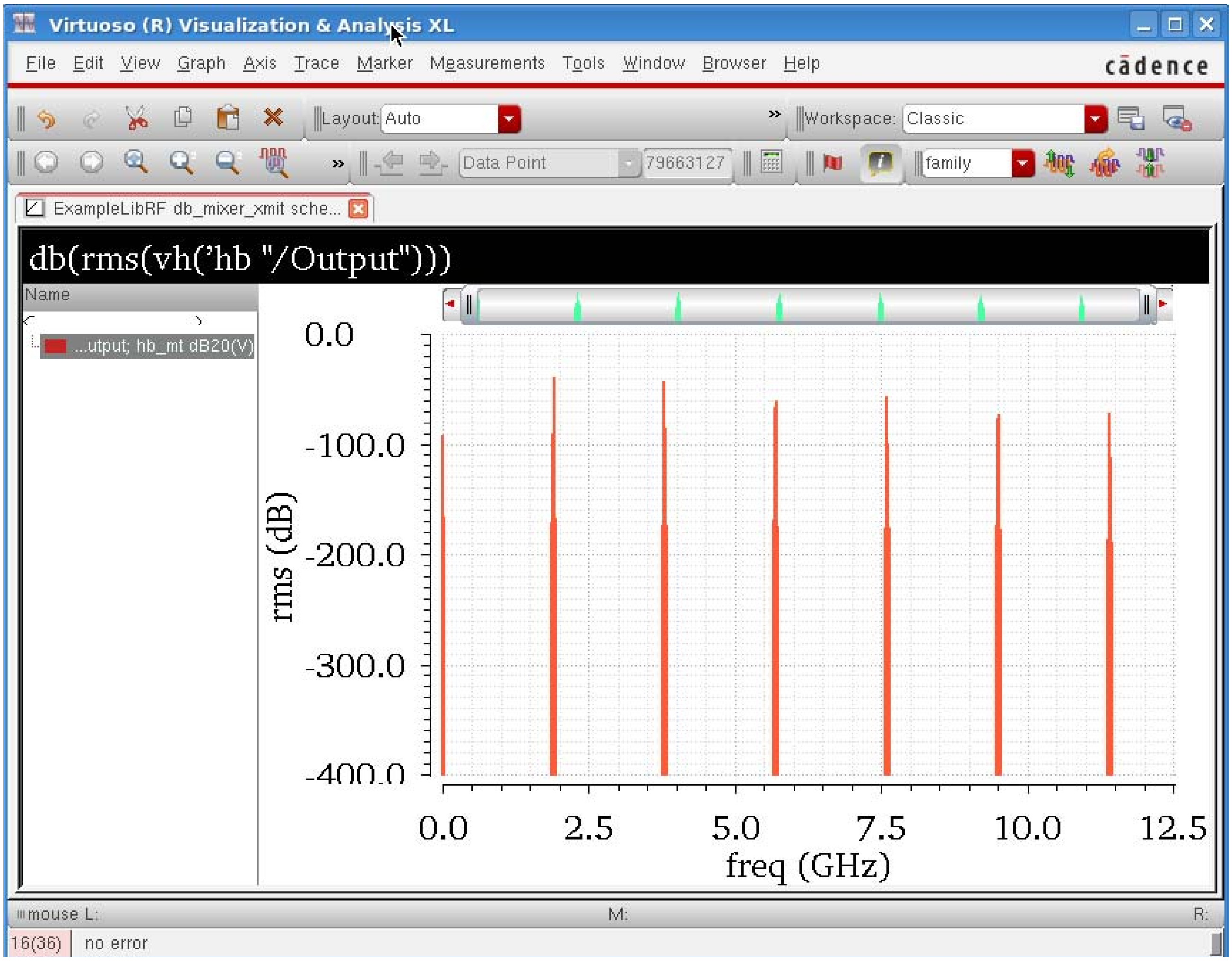
4. Click on the net labeled *Output* in the schematic.

Figure A-453 Output Net in Schematic



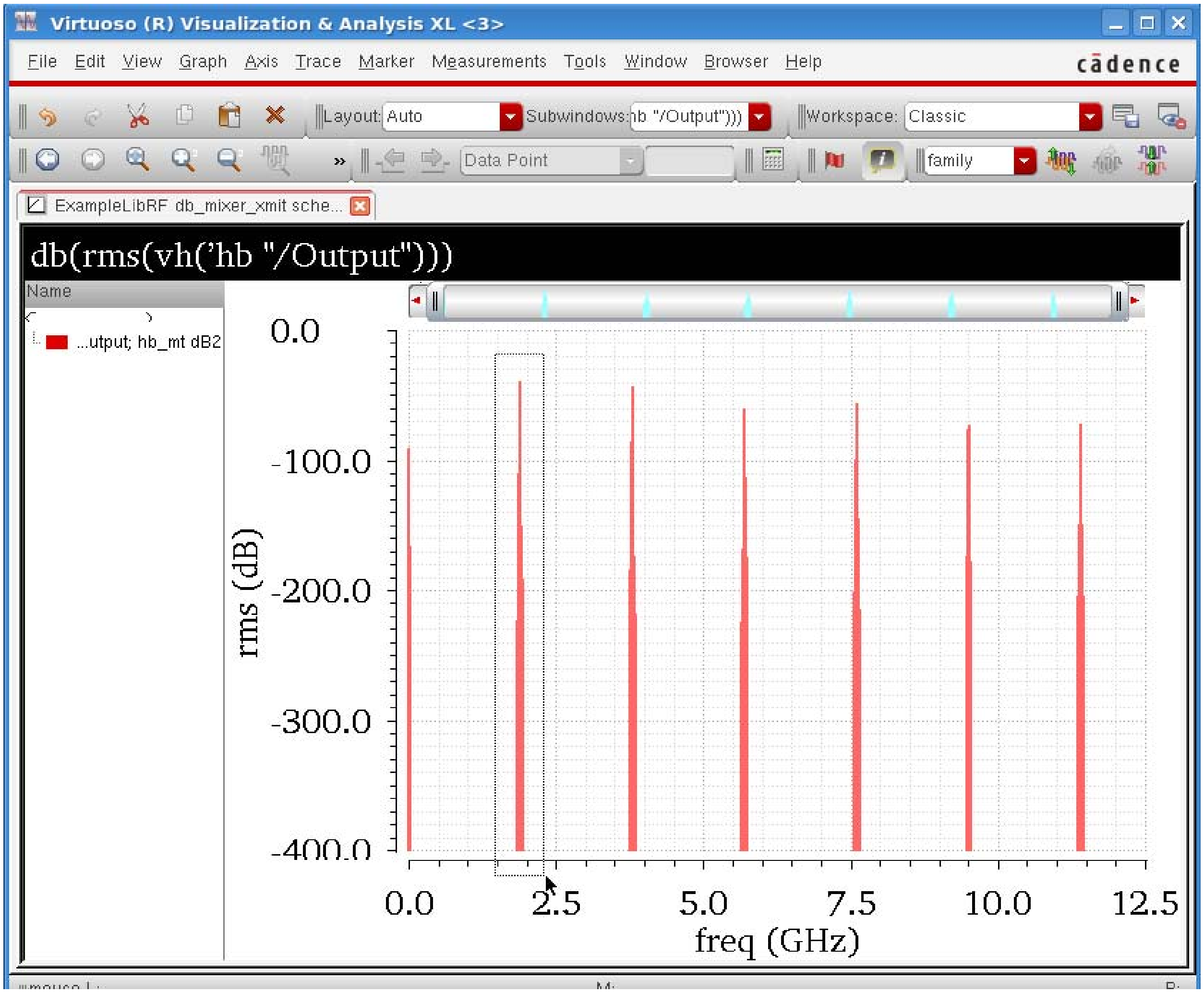
The plot is displayed.

Figure A-454 Voltage Spectrum of Transmit Mixer



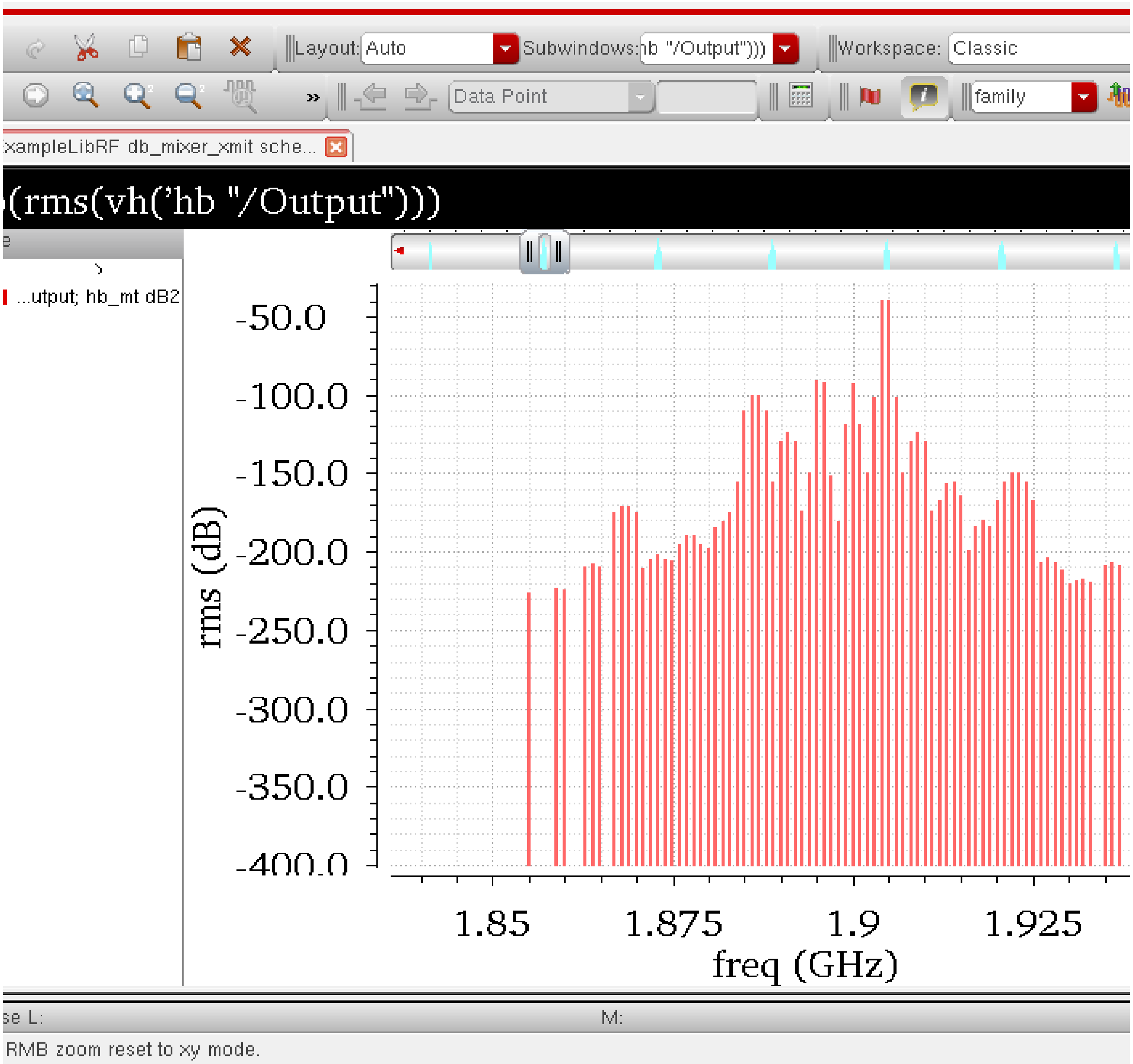
5. Zoom in on the area around 1.9GHz. Place the cursor in the waveform window, click the right mouse button and draw a box around the signals surrounding 1.9GHz. You may need to do this several times to zoomin specifically to the area around 1.9GHz.

Figure A-455 Zooming in Around 1.9GHz



6. The waveform window should look similar to the following:

Figure A-456 Zoomed in Voltage Spectrum

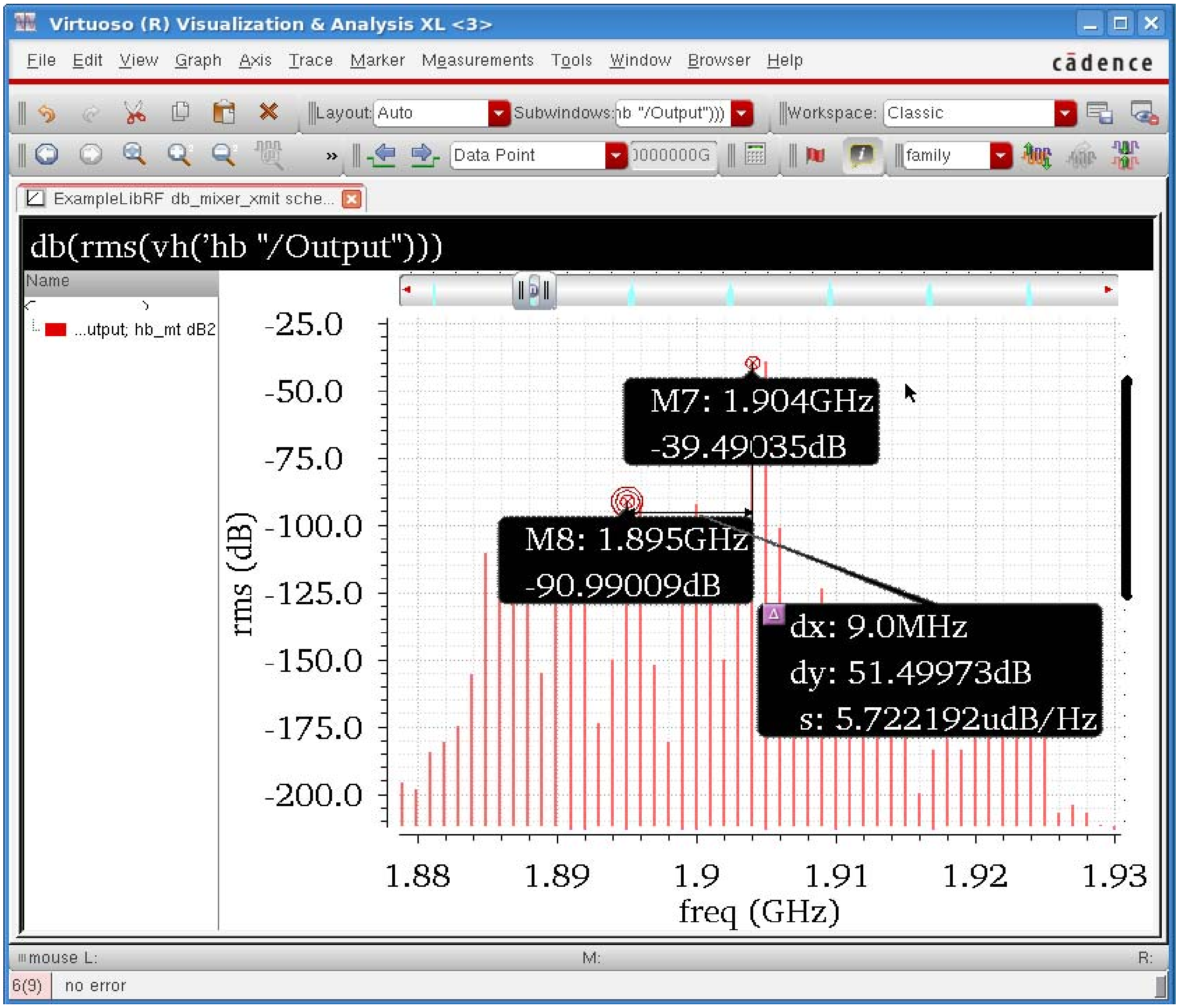


7. Move the mouse cursor to the tip of the harmonic at frequencies *1.895G*, *1.896G*, *1.9G*, *1.904G*, and *1.905G*. The tracking cursor values read the following:

	Frequency (GHz)	Amplitude (peak, dB)
Low side RF freq 2	1.8995	-90.99
Low side RF freq 1	1.896	-91.90
LO Frequency	1.9	-92.43
Hi side RF Freq 1	1.904	-39.49
Hi side RF Freq 2	1.905	-39.49

Image rejection is the ratio of the power in the passband divided by the power in the image. If you have dB, this is dB at the passband frequency minus dB at the image frequency. Place a marker on the RF1 frequency (1.904G) by placing the cursor on the trace and pressing the *m* bindkey Place a delta marker on the Image frequency (1.895GHz) by placing the cursor on the trace and pressing *d*.

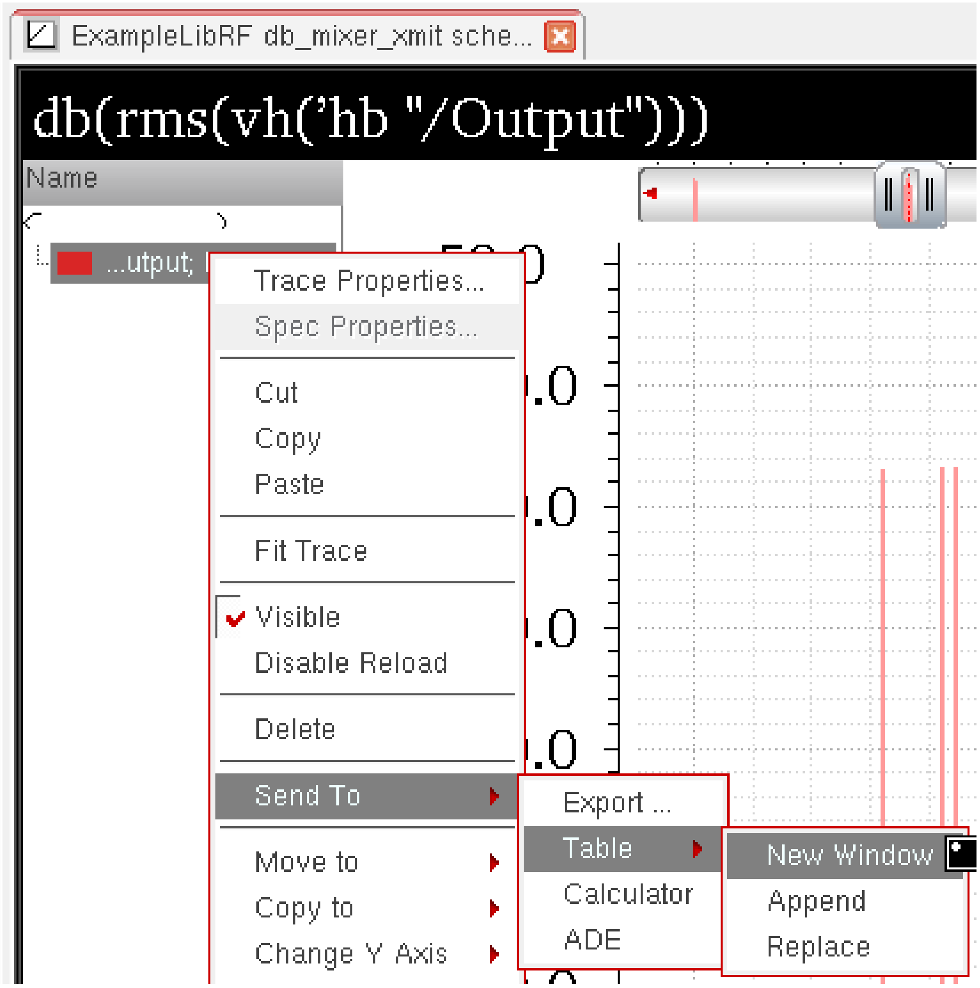
Figure A-457 Image Rejection



The image rejection is 51.5dB.

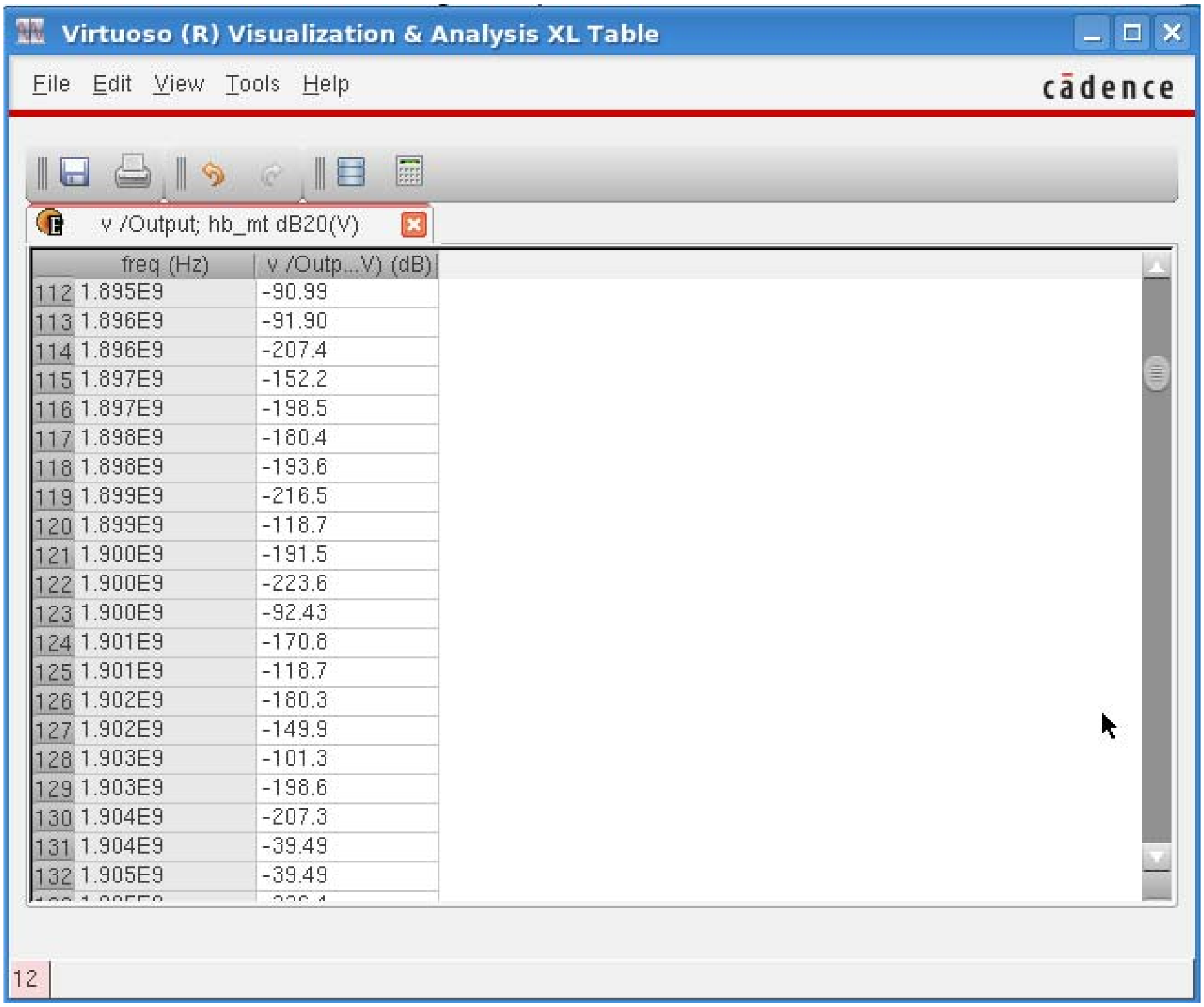
8. You can create tabular data from the results. In the waveform window, select the trace by clicking the legend on the left side of the graph area, clicking the right mouse button, and selecting *Send to - Table - New Window.*

Figure A-458 Sending Waveform Data to Table



A tabular table of data is displayed in a Virtuoso Visualization and Analysis XL Table window. You can scroll down to the frequencies of interest using the scroll bar on the right side of the table.

Figure A-459 Tabular Data



	freq (Hz)	v /Outp...V) (dB)
112	1.895E9	-90.99
113	1.896E9	-91.90
114	1.896E9	-207.4
115	1.897E9	-152.2
116	1.897E9	-198.5
117	1.898E9	-180.4
118	1.898E9	-193.6
119	1.899E9	-216.5
120	1.899E9	-118.7
121	1.900E9	-191.5
122	1.900E9	-223.6
123	1.900E9	-92.43
124	1.901E9	-170.8
125	1.901E9	-118.7
126	1.902E9	-180.3
127	1.902E9	-149.9
128	1.903E9	-101.3
129	1.903E9	-198.6
130	1.904E9	-207.3
131	1.904E9	-39.49
132	1.905E9	-39.49

Close the *Direct Plot Form* and the waveform window.

Summary

In this section, you measured the voltage spectrum and image rejection on an up-converting double-balanced (transmit) mixer using hb analysis. In the next section, you will measure the third-order intercept.

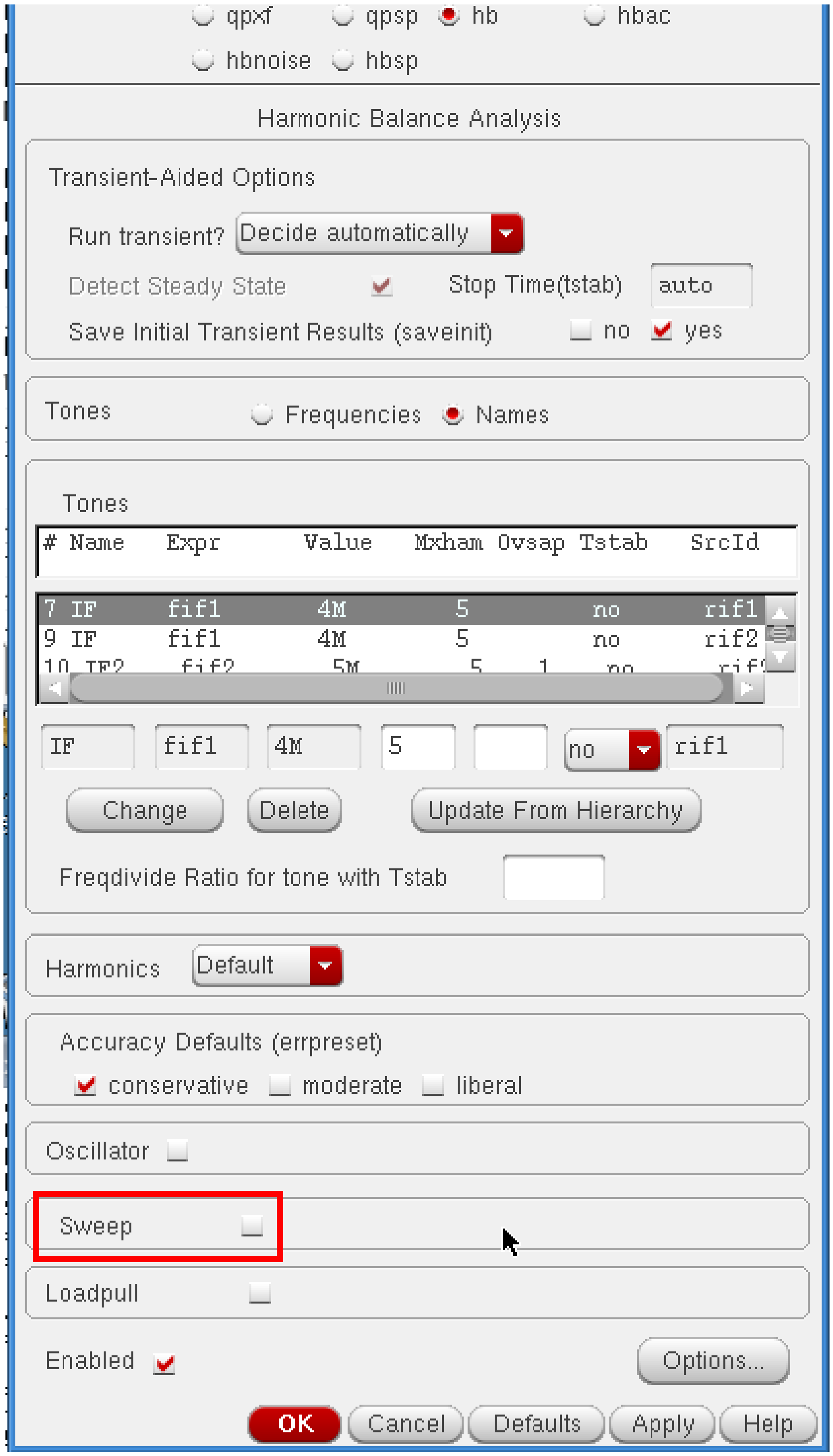
Three Tone IP3

For transmitters, there might be an IP3 spec at a signal in the large-signal range. This will not be a small-signal projection, so the traditional HB/HBAC IP3 or rapid IP3 cannot be used for this measurement. Instead, multi tone HB or QPSS-HB would be appropriate.

The simulation setup is nearly the same as the previous example. You only need to make a couple of modifications.

1. Open the hb *Choosing Analyses* form by double-clicking the *hb* line in the ADE window. The Choosing Analyses form is displayed, as shown below.

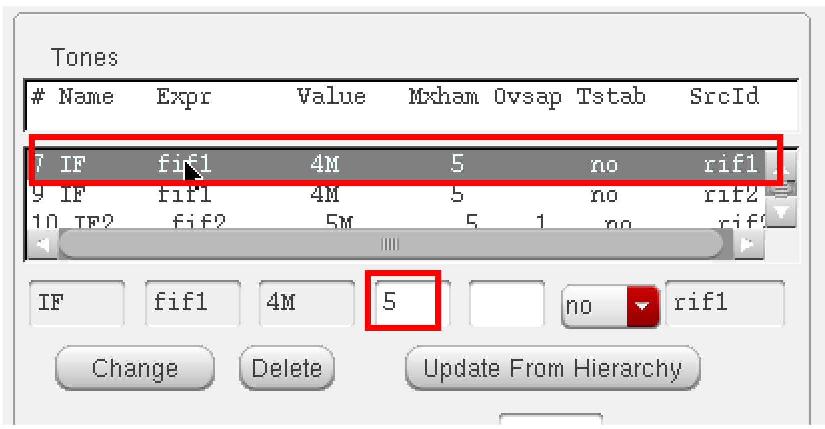
Figure A-460 hb Choosing Analyses Form



2. Ensure that *Accuracy Defaults (errpreset)* is set to to *conservative*. *conservative* is set because third and fifth-order intermodulation distortion will be calculated with this setup. Because these amplitudes are small, high accuracy is needed.

3. In the *Tones* section, select the line containing one of the IF tones. For this measurement, you will again use *5* harmonics for the IF and IF2 tones.

Figure A-461 Choosing Analyses Tones Section - Setting Mxham for IF Tones



4. When making an IP3 measurement using three large tones, you will need to do a power sweep. Click the *Sweep* button in the *Choosing Analyses* form. The form expands.

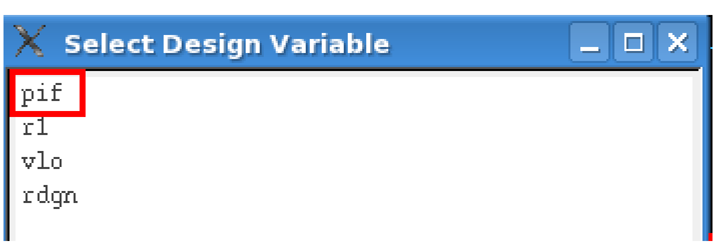
5. In the *Sweep* section:

   a. Set *Sweep* to *Variable*.

   b. For *Frequency Variable?* select *no*. You will be sweeping input power rather than frequency.

   c. Click the *Select Design Variable* button and choose *pif*. You will be sweeping the input power on the IF ports.

Figure A-462 Select Design Variable Form



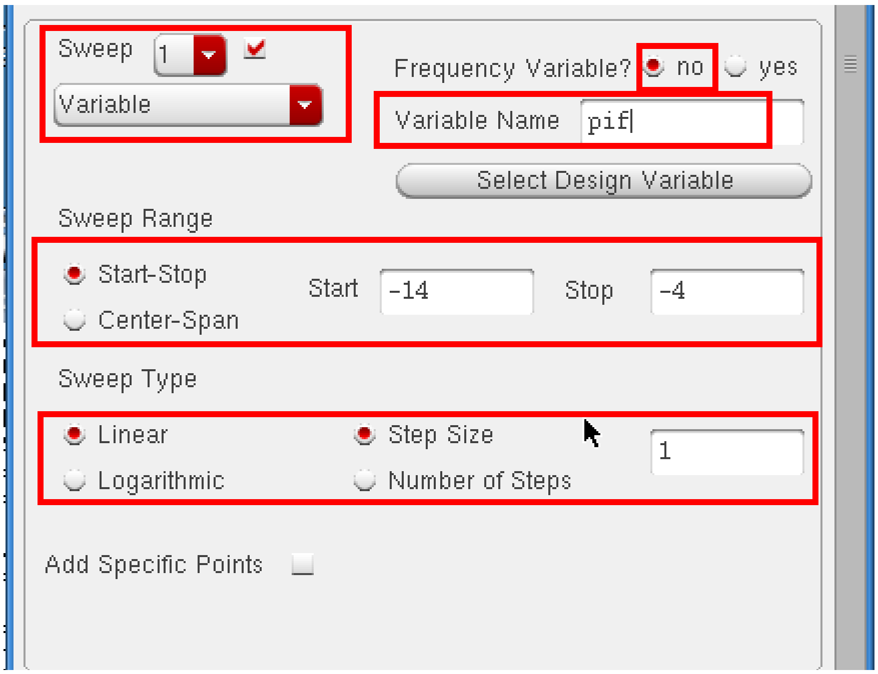
   d. *Click OK*.

e. In the *Sweep Range* section, choose *Start-Stop* and sweep from *-14* to *-4*.

f. Set the *Sweep Type* to *Linear* and *Step Size* to *1*.

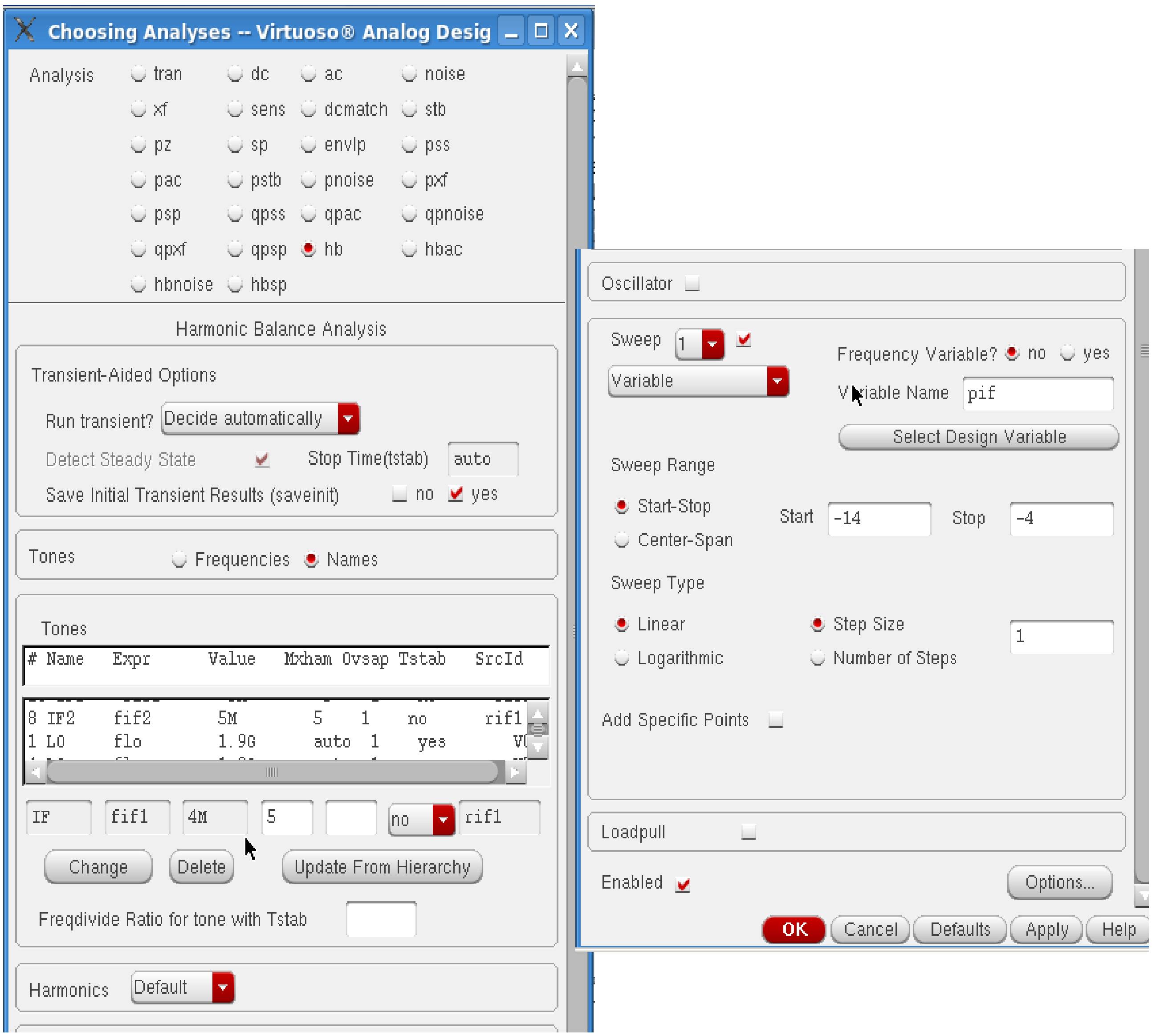
The Sweep section of the form should look like the following:

Figure A-463 hb Choosing Analyses Form Sweep Section for 3 Tone IP3 Measurement.



6. The complete hb *Choosing Analyses* form should look like the following:

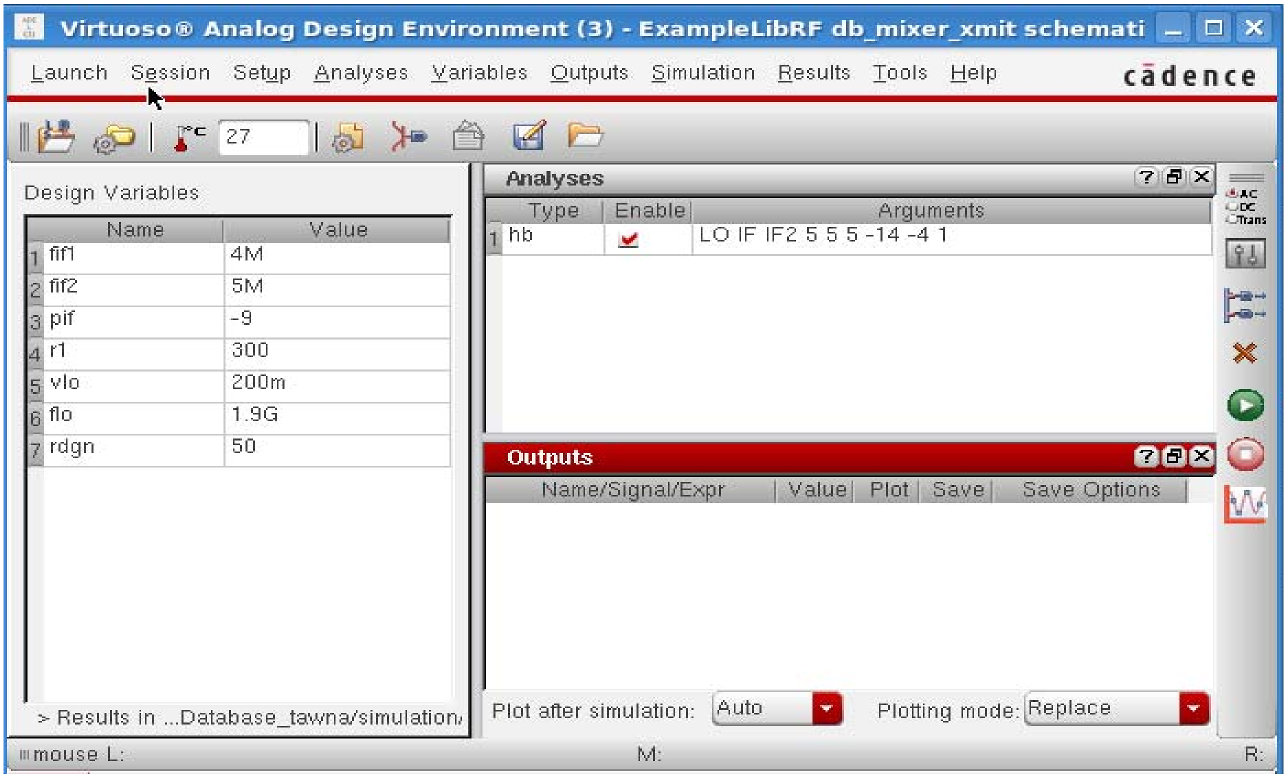
Figure A-464 hb Choosing Analyses Form for 3 Tone IP3



7. Click *OK* at the bottom of the *Choosing Analyses* form.

   The Analog Design Environment window is displayed.

Figure A-465 ADE Simulation Window



Note: If you selected the *Add to Outputs* button in the *Direct Plot Form* in one of the earlier simulations, you will see the *Outputs* in the *Outputs* pane of the simulation window. *Deselect* the plot expressions in the *Outputs* pane*.* If you do not, a family of spectral plots will bedisplayed when you plot the results.

Run the simulation and plot the results, as follows:

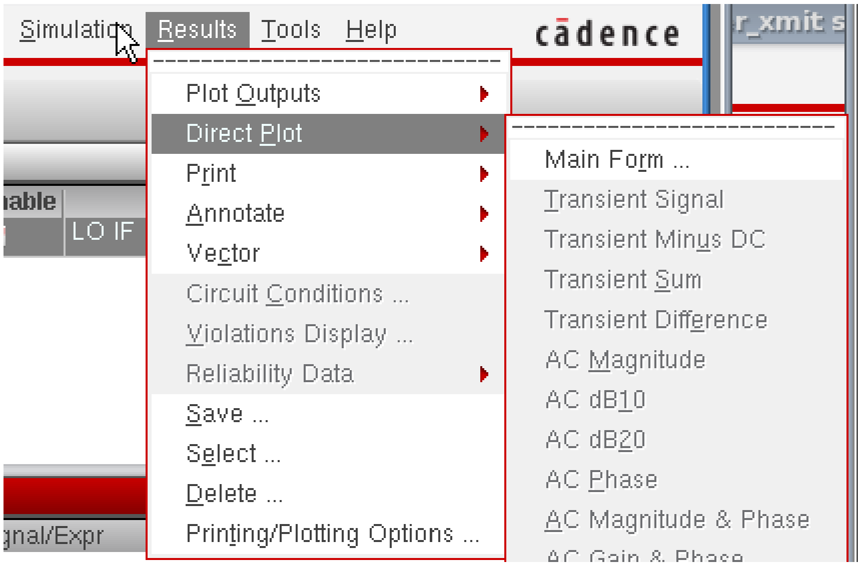
1. Start the analyses by clicking the green arrow icon.  in ADE or in the Schematic Editor.

   This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` logfile). When the analysis has completed, you may iconify the status window.

   Next, you will plot the IP3 results.

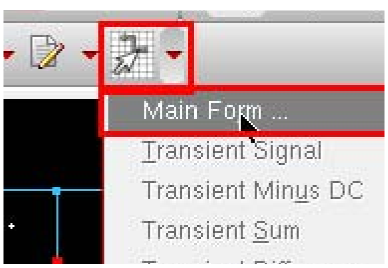
2. When the simulation has finished, in the ADE window,choose *Results - Direct Plot - Main Form.*

Figure A-466 Invoking Direct Plot Form After Simulation



You can also open the *Direct Plot Form* by clicking the *Direct Plot* icon in the Schematic window and choosing *Main Form,* as shown below.

Figure A-467 Direct Plot Icon



The *Direct Plot Form* is displayed, as shown below.

Figure A-468 Harmonic Balance Direct Plot Form

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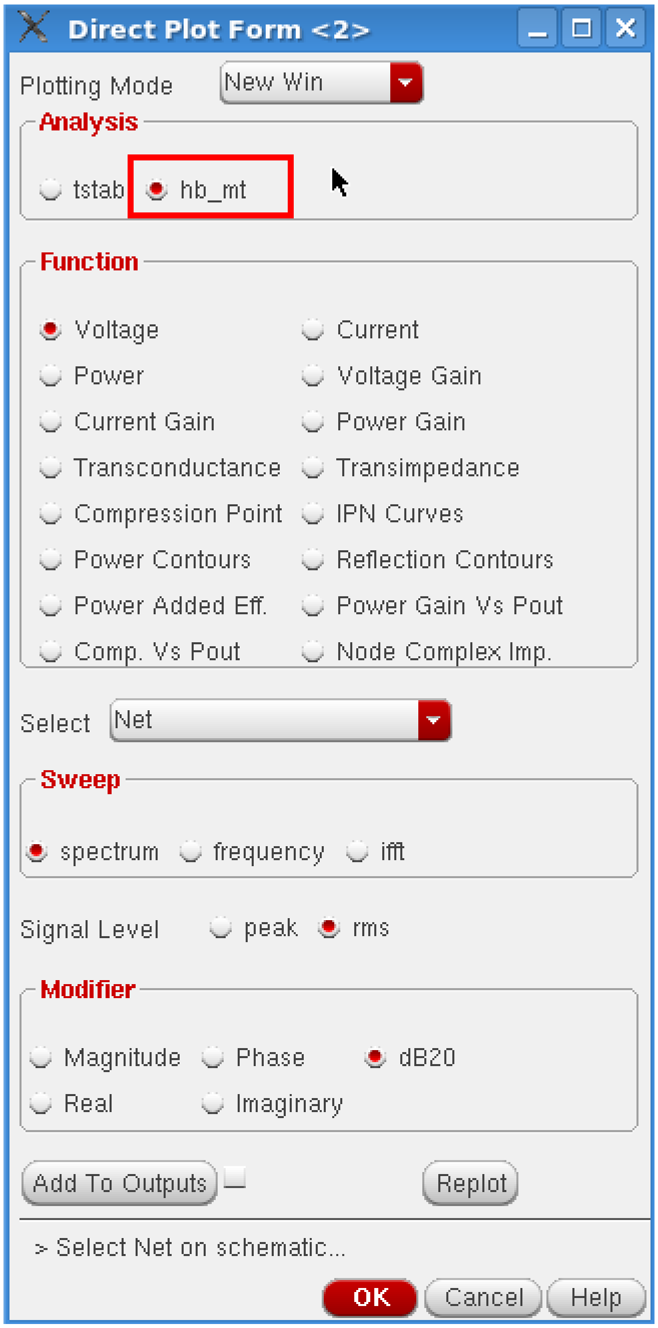


3. In the *Direct Plot Form*, you will notice that there are two available analyses, *tstab* and *hb_mt*. The transient-assisted results are accessible by selecting the *tstab* analysis. The hb results are accessible by selecting the *hb_mt* analysis. Note that when there is more than one tone, you will see *hb_mt* (signifying harmonic balance multi-tone). A single tone simulation will show *hb*. In the *Analysis* Section, make sure that `hb_mt` is selected.

a. In the *Function* section select *IPN Curves*.

Note: You will be plotting Input IP3. Note that the N in IPN Curves is selectable an can be 2,3, 4, 5, 6, or 7.

b. Select *Port(Fixed R(port))*.

c. Set *Circuit Input Power* to *Variable Sweep ("pif")*. You will be plotting output power vs. swept input power. *pif* is the swept power specified on the port

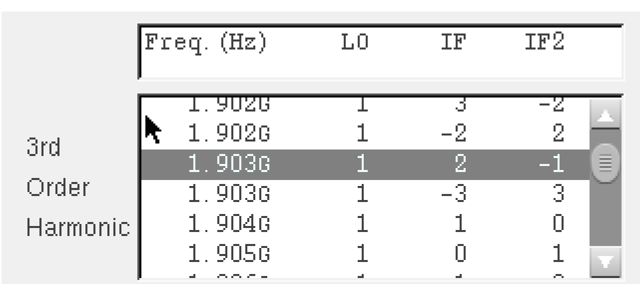
d. Set the *Input Power Extrapolation Point* to *-9.* This is the power at which the circuit operates.

e. Select *Output Referred IP3* and *Order 3rd* to plot input referred IP3.

f. The main upconversion frequencies are at 1.904G and 1.905G. The third order intermodulation products are at 1.903G and 1.906G. Select the lowest amplitude first order term and the largest amplitude of the third order term.In this case, since both first order terms and both third order terms are almost identical in amplitude, it doesnot matter which first and third order terms are selected. Select *1.906G* for the third order term and *1.904G* for the first order term. Alternately, you can select *1.905G* for the first order term and *1.903G* for the third order term.

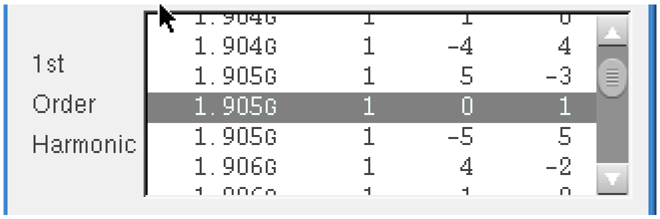
g. Note that if you have more than one selection for any of the frequency terms (for example, the third-order term), you will want to select the correct one. To select the correct frequency, add the absolute value of the indices displayed for the desired frequency and choose the frequency with the lowest value. For example, for the two 1.903G, frequencies, you have `|1|+|2|+|-1|=4` and `|1|+|-3|+|3|=7`. You want to choose the first *1.903G* entry because adding the absolute value of the indices produces the lowest number.

Figure A-469 Selecting the Correct 3rd Order Term When Plotting IP3



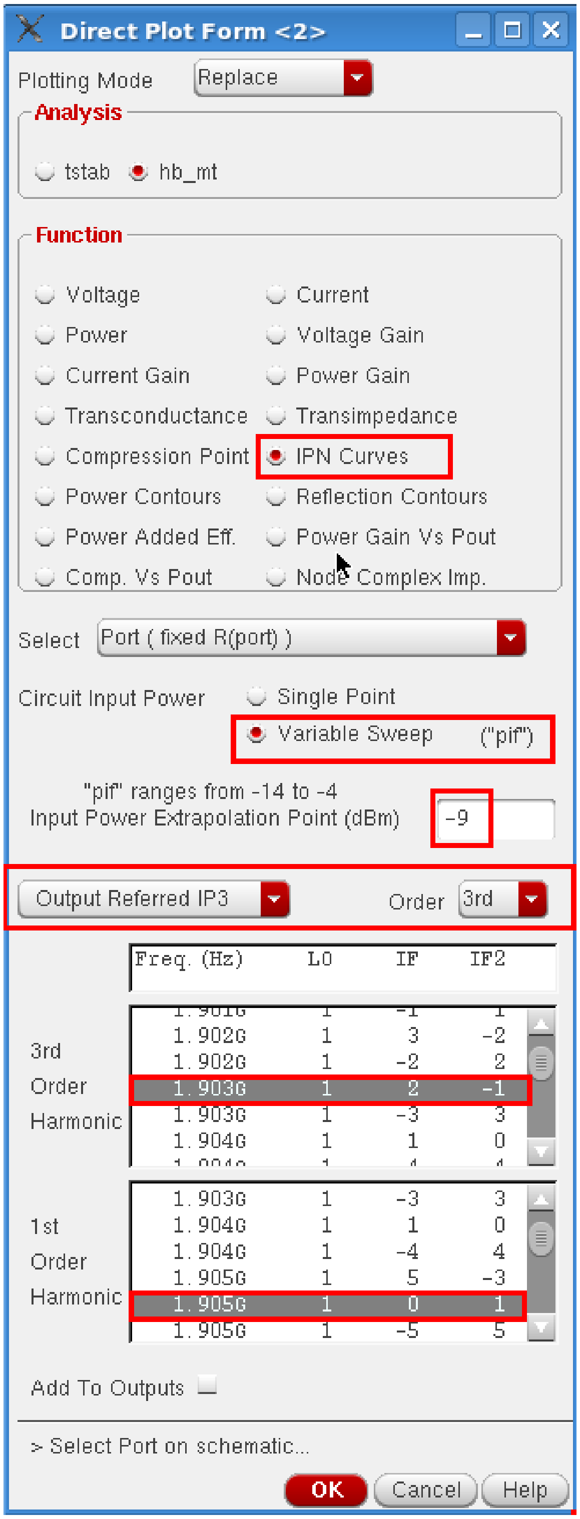
h. Similarly, you will need to choose the appropriate first order term. Note the selected *1.905G* entry. When adding the absolute value of the indices, it produces the lowest number.

Figure A-470 Setting the Correct 1st Order Term when Plotting IP3



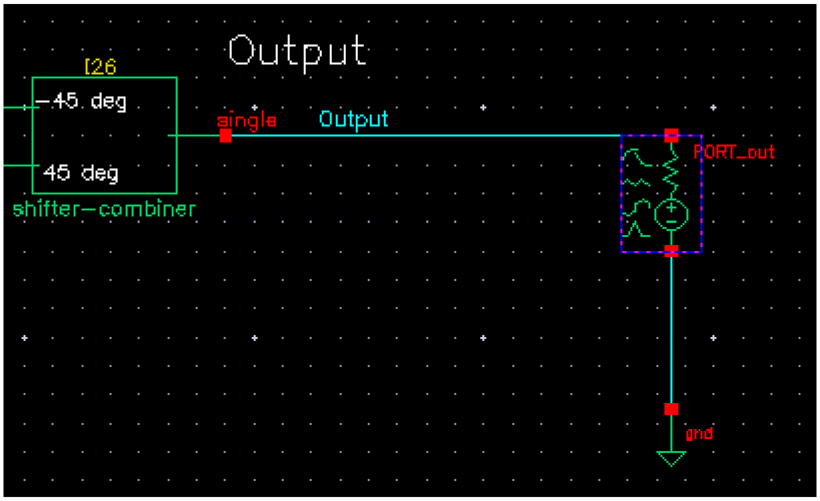
The *Direct Plot Form* should look like the following:

Figure A-471 Direct Plot Form for Measuring IP3



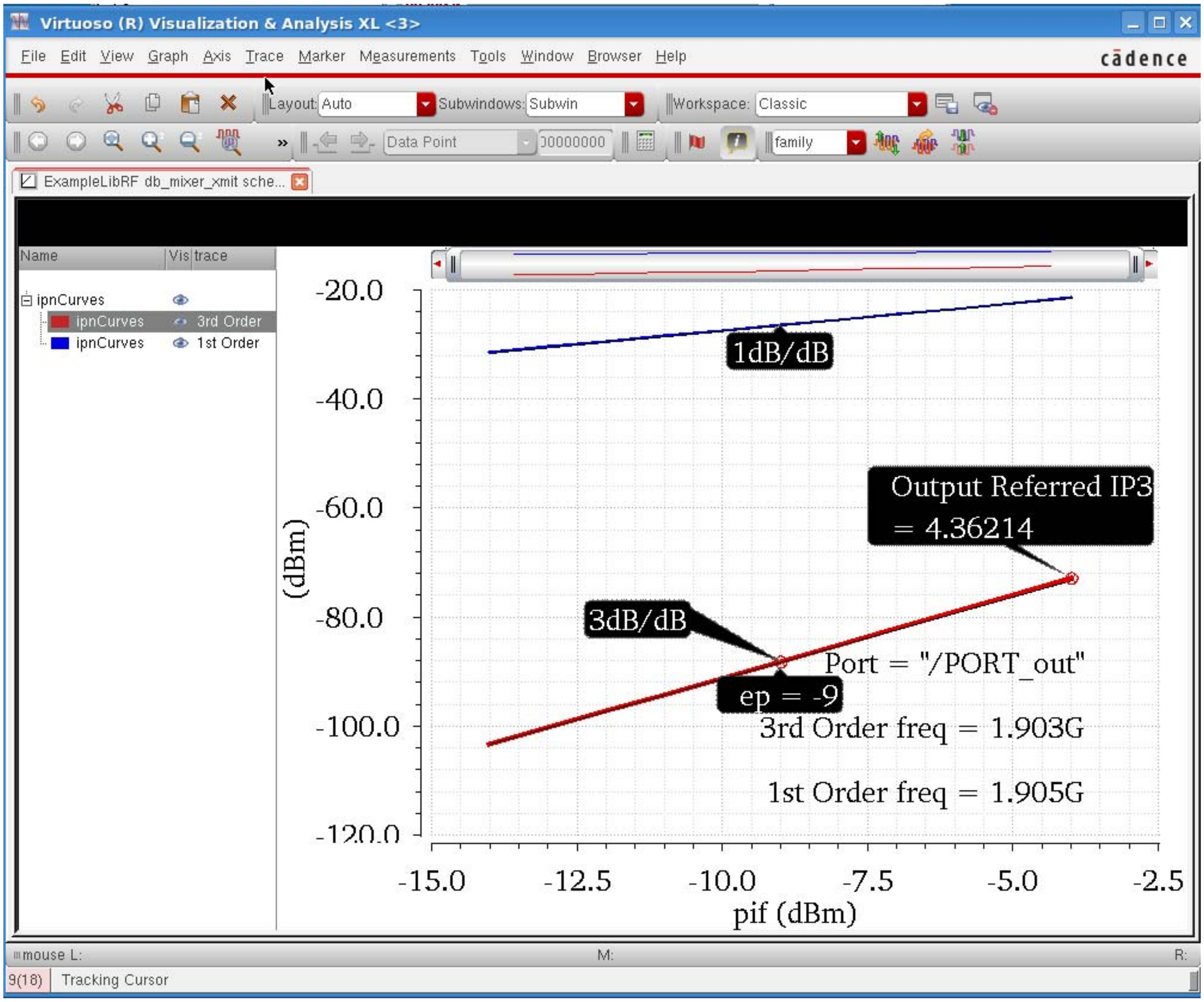
4. Select the *Output* port on the schematic.

Figure A-472 output port



The IP3 plot appears. Note that you can move the Input Referred IP3 label by clicking and holding the right mouse button and dragging the label to another part of the screen. You may need to first click in the graphics to deselect the IP3 marker.

Figure A-473 Output Referred IP3



The output referred IP3 is shown on the plot as *4.36dBm*. You can also plot input referred IP3 from the *Direct Plot Form* or read it directly off the x-axis.

Close the Analog Design Environment and Exit the Schematic Window, as follows**:**

1. In the ADE L window, choose *Session - Quit.*
2. Click *No* in the *Save State* form.
3. In the schematic window, choose *File - Close All.*

Summary

In this section, you measured the three tone spectral content, image rejection, and output IP3 of a Transmit Mixer. In the next section, you will measure the Signal-to-Noise ratio of a Transmit Mixer.

Signal-to-Noise Ratio

The image reject mixer makes it possible to transmit one of the two sidebands (for example, `f2+f1` or `f2-f1`). The only way to achieve image rejection in both legs is to have a +/- 45 degree phase difference on the inputs. Typically, the DSP (digital signal processor) produces the 45 degree phase difference. Since we are not able to measure the noise figure directly on this circuit, we calculate the signal to noise ratio (SNR) instead. The inputs are set in order to mimic a balun. We cannot use a shifter combiner on the input (it is an RC circuit) because it would be too narrowband.

Note: That there will be no image rejection for noise because noise is random and has uncorrelated phase. The only image rejection is on the signals 4MHz and 5MHz. The bandwidth of the signal that will be transmitted is approx 10MHz.

Open the `db_mixer_xmit` mixer circuit in the Schematic Window, as follows:

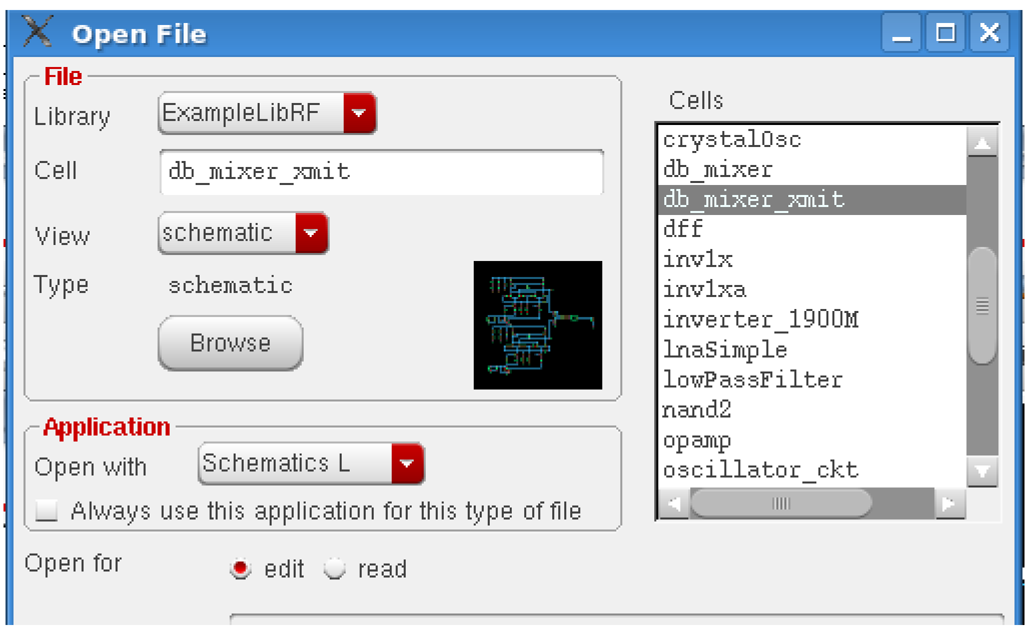
1. In the CIW, choose *File – Open*.

   The *Open File* form is displayed.

2. In the Open File form, choose *ExampleLibRF* from the *Library* drop-down list.
3. Choose *db_mixer_xmit* from the *Cells* list box.

   The completed *Open File* form looks like the following:

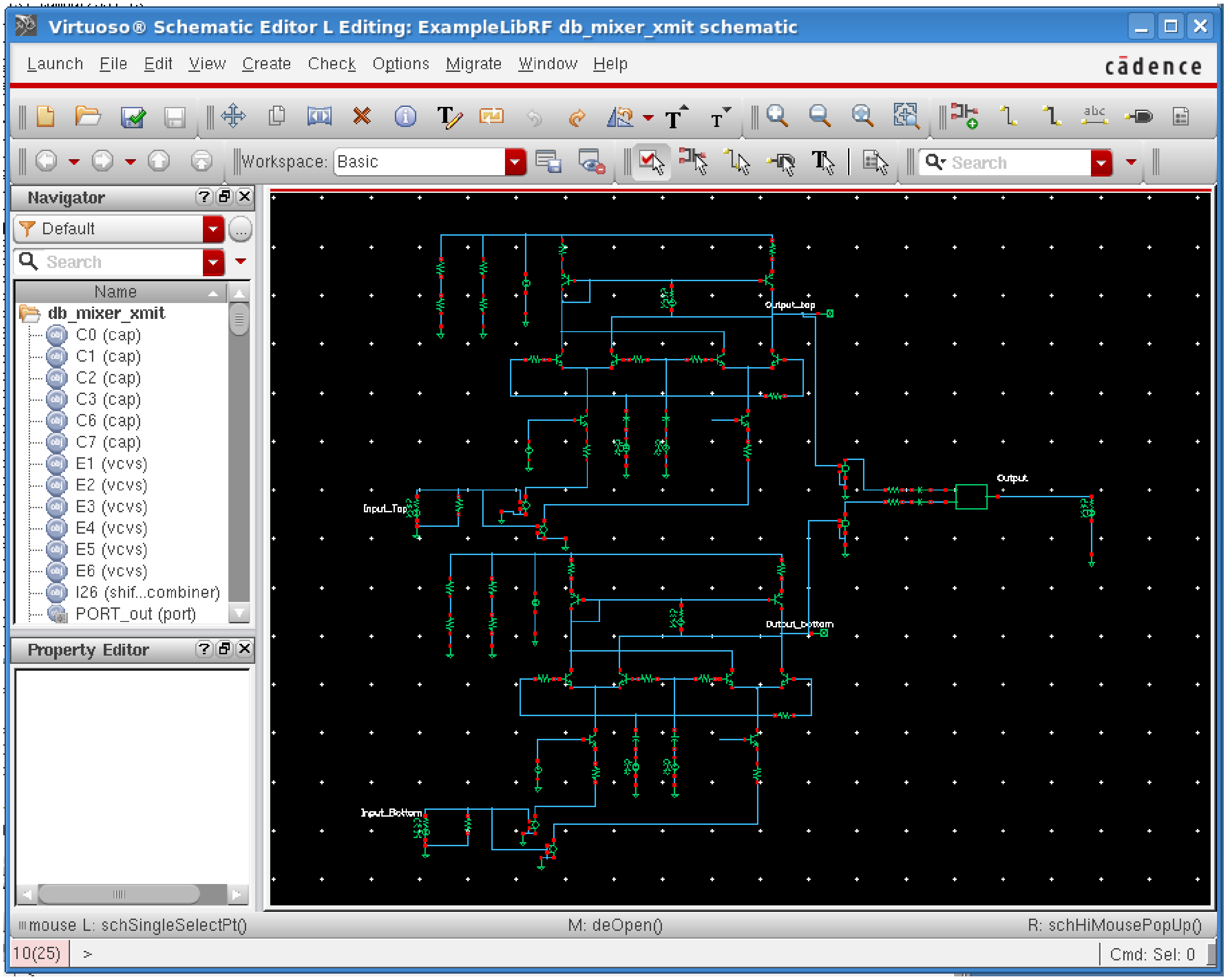
Figure A-474 Open File Form



4. Click *OK*.

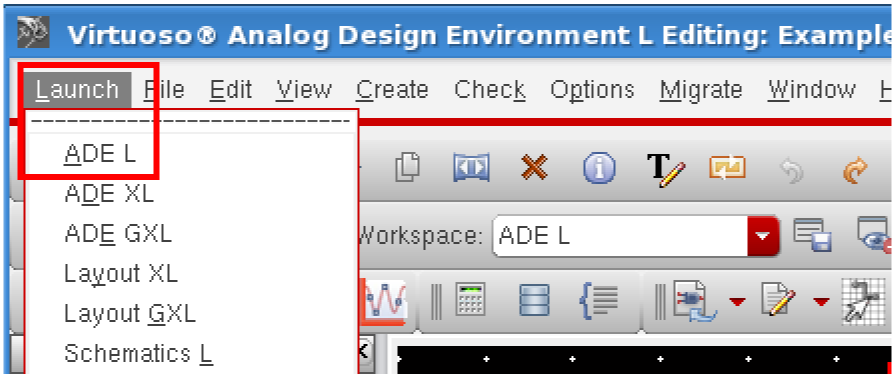
The Schematic window for the *db_mixer_xmit* mixer is displayed.

Figure A-475 db_mixer_xmit Schematic



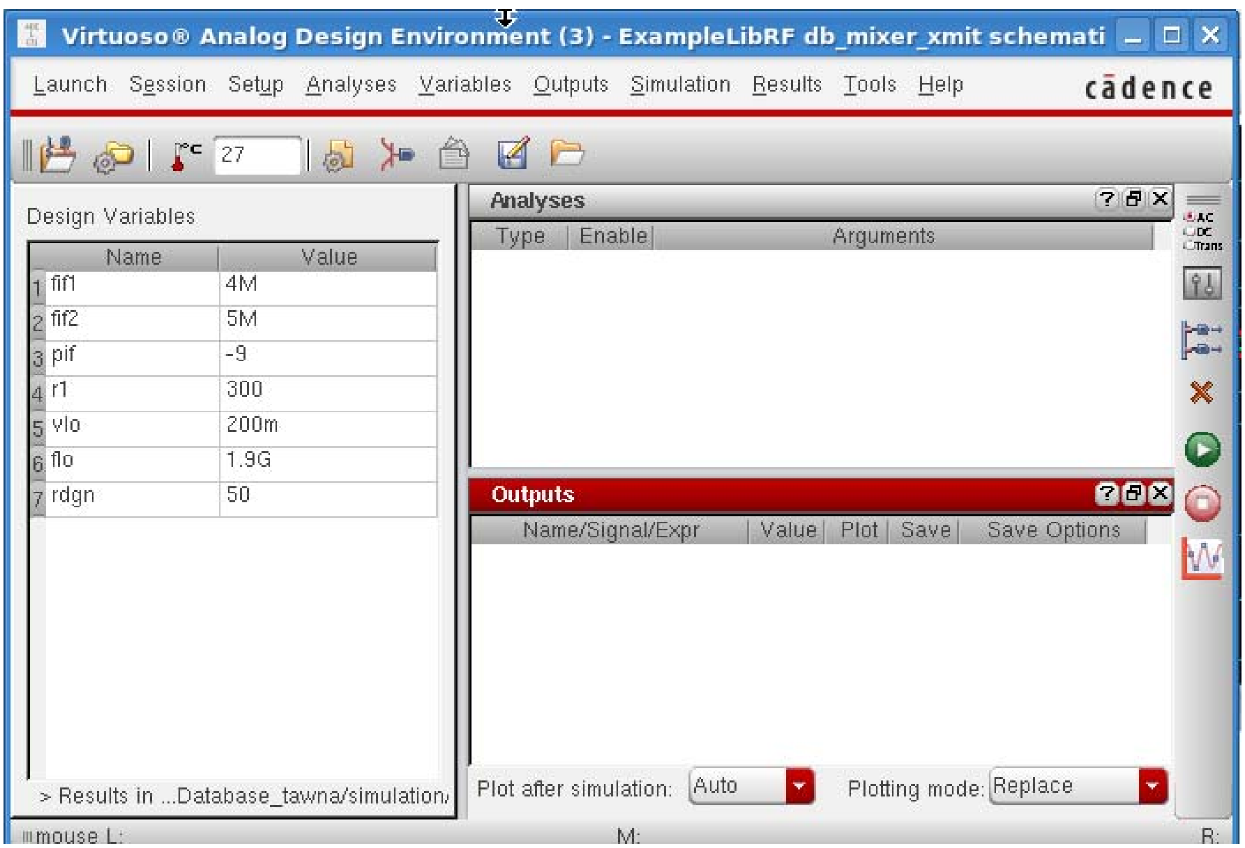
5. In the Schematic window, choose *Launch – ADE L*.

Figure A-476 Launching ADE-L From Schematic



The Virtuoso Analog Design Environment window is displayed, as shown below.

Figure A-477 Analog Design Environment Window



Choosing the simulator options, as follows:

1. Choose *Setup – Simulator/Directory/Host* in the *Virtuoso Analog Design Environment* window.

   The *Choosing Simulator/Directory/Host* form is displayed.

2. Specify the following in the *Choosing Simulator/Directory/Host*:

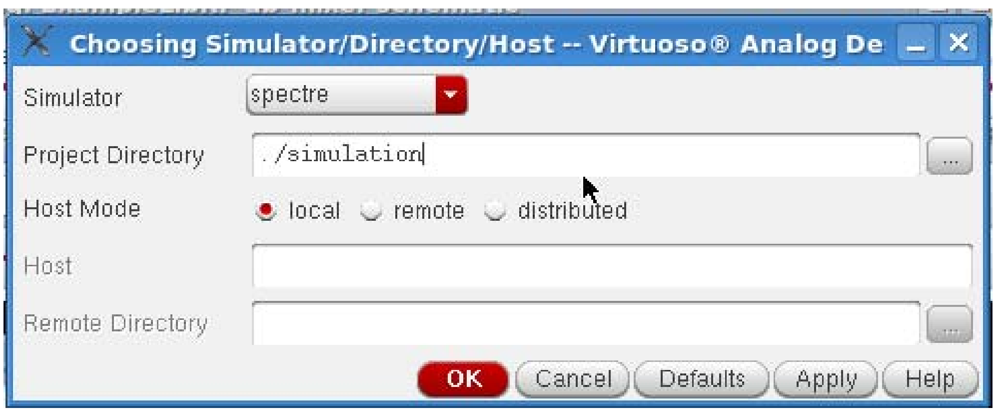
   a. Choose *spectre* as the *Simulator*.

   b. Type the name of the project directory, if necessary, in the *Project Directory* field. The project directory defines the location of the simulation directory, which stores the simulation results. The default location is your `~/simulation` directory.

   c. Select the *Host Mode* that corresponds to your situation. For *remote* or *distributed* mode, contact your Cadence tools System Administrator for specific setup instructions.

      The completed form looks like the following.

Figure A-478 Choosing Simulator/Director/Host Form

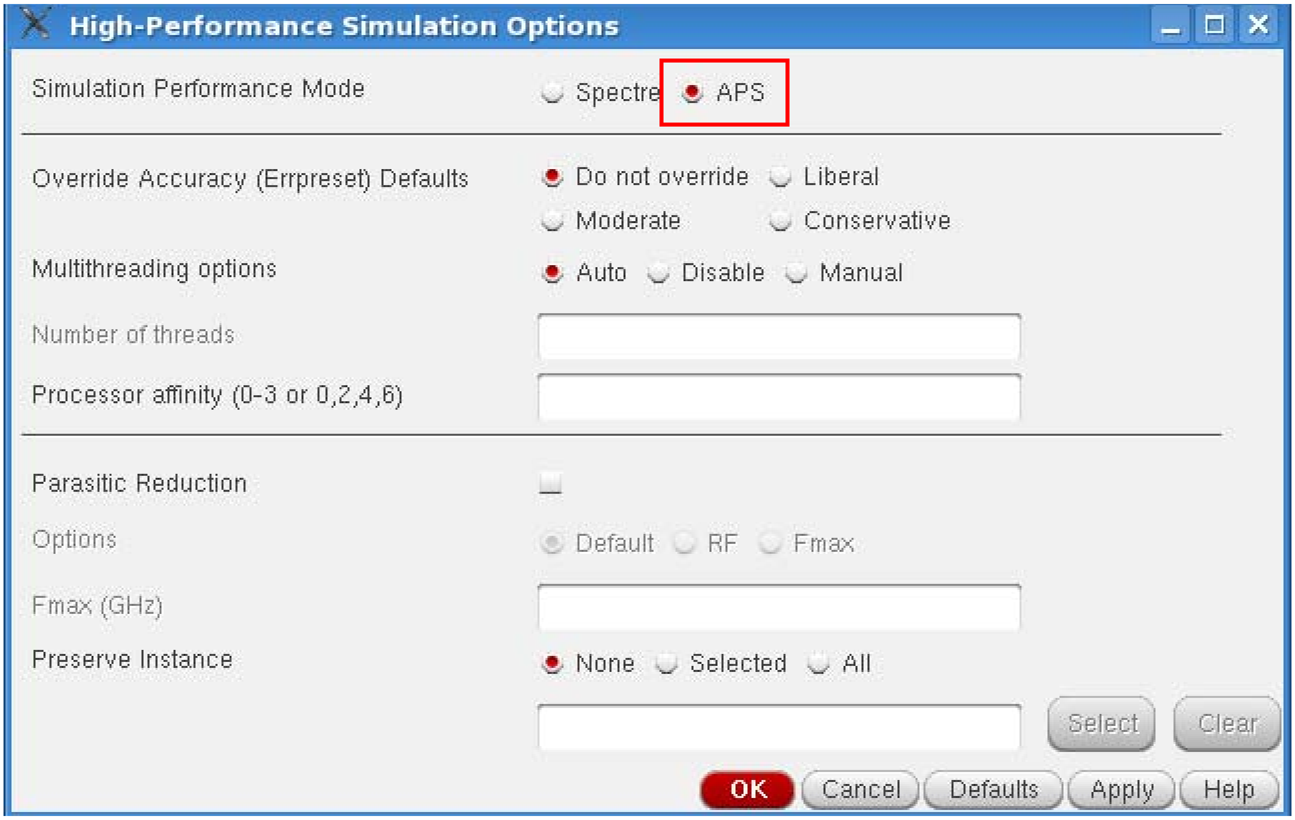


   d. Click *OK*.

3. Set up the *High Performance Simulation Options*, as follows:

   In the ADE window, choose *Setup - High Performance Simulation.* The *High Performance Simulation Options* window is displayed, as shown below.

Figure A-479 High Performance Simulation Options



In the *High Performance Simulation* window, select *APS*. Note that *auto* is selected for *Multithreading options*. The effect of this is to detect the number of cores on the system (up to 16) and then multi-thread on all the available cores. The bigger the circuit, the more threads you should use. For a small circuit such as this, you may want to set the number of threads to 2. Using 16 threads on a small circuit might actually slow things down because of the overhead associated with multithreading.

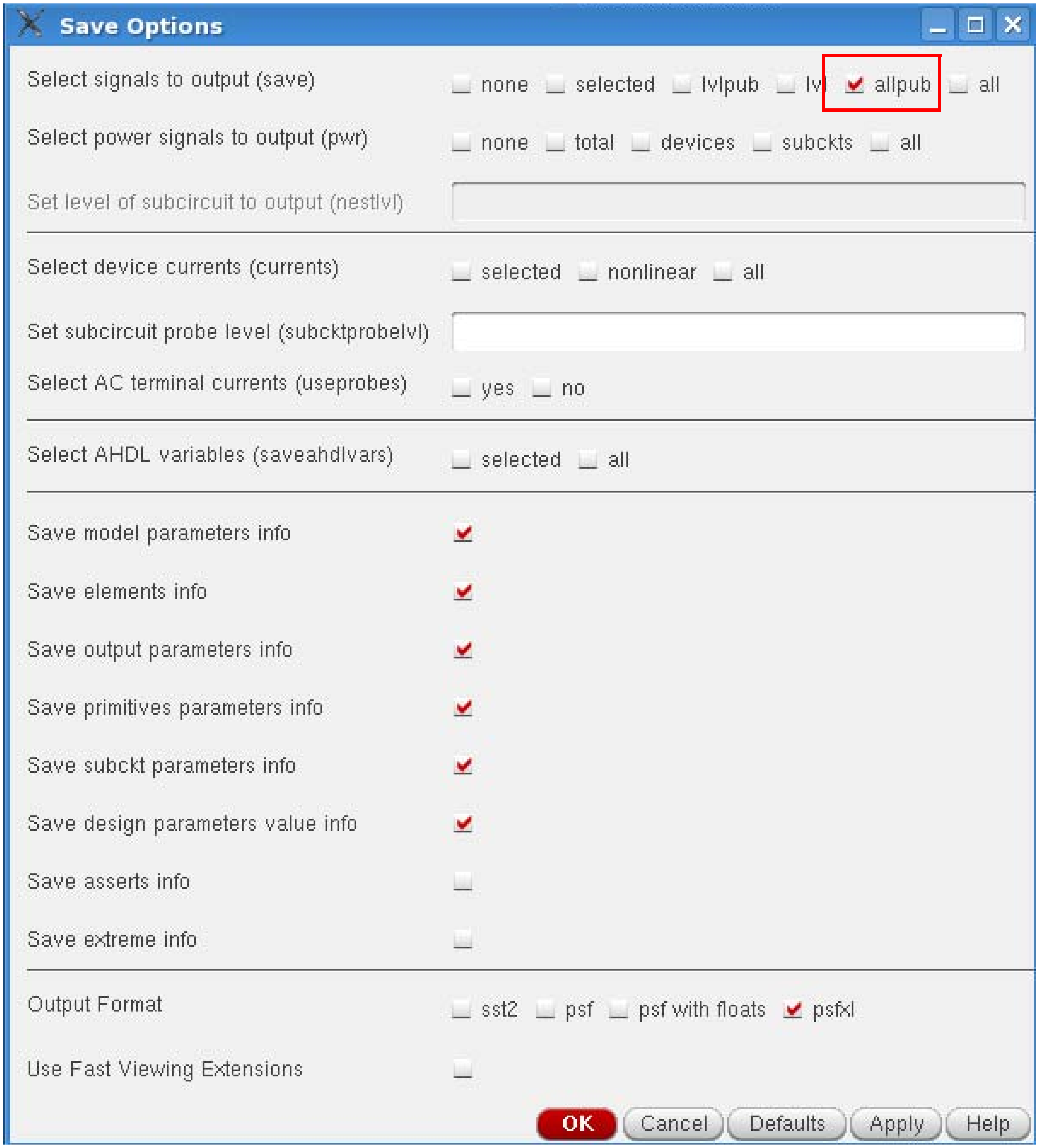
Click *OK* .

4. In the *Virtuoso Analog Design Environment* window, choose *Outputs – Save All.*

The *Save Options* form is displayed.

5. In the *Select signals to output* section, make surethat allpub is selected.

Figure A-480 Save Options Form



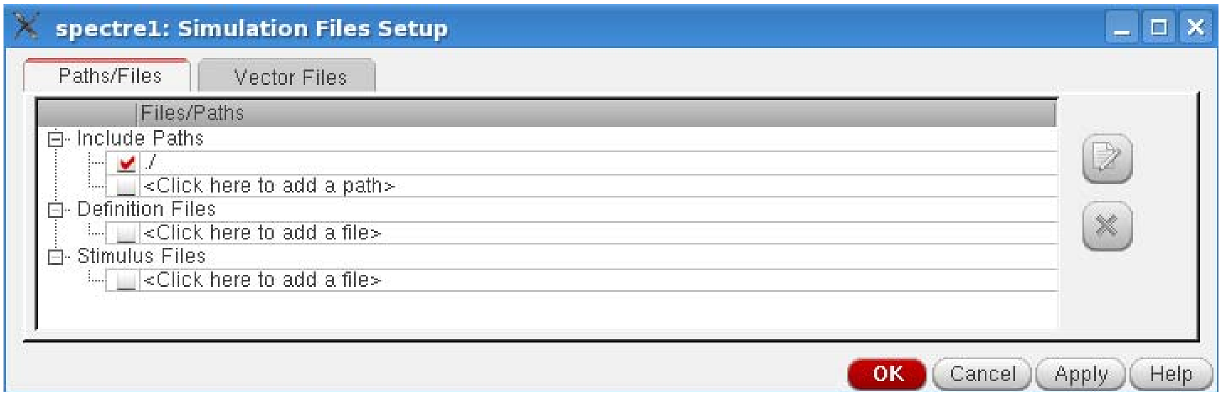
This is the default selection. This saves all of the node voltages at all levels of the hierarchy, but it does not include the node voltages inside the device models.

6. Click *OK*.

Set up the model libraries, as follows:

1. In the *Virtuoso Analog Design Environment* window, choose *Setup - Simulation Files.* The Simulation Files Setup form is displayed, as shown below.

Figure A-481 Simulation Files Setup Form



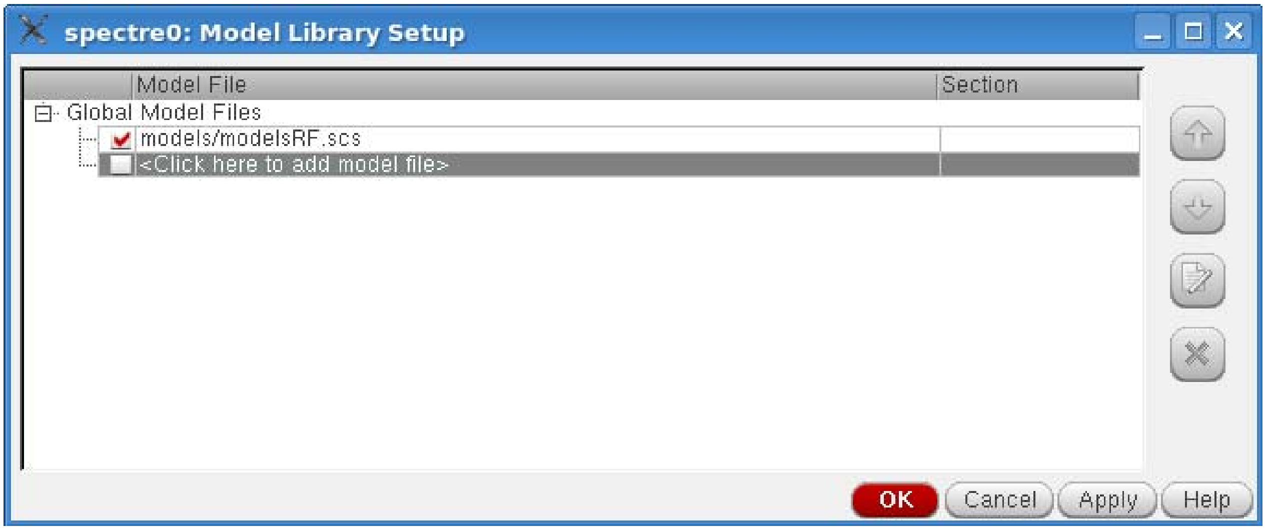
2. Ensure that the *Include Path* is set as shown above.
3. In the *Virtuoso Analog Design Environment* window, choose *Setup – Model Libraries.*

   The Model Library Setup form is displayed.
4. In the *Model Library File* field, type the following for the name of the model file:

   ```
   models/modelsRF.scs
   ```
5. Click OK..

   The *Model Library Setup* form looks like the following.

Figure A-482 Model Library Setup



Alternately, you can click on the *Browse* button and browse to the `modelsRF.scs` model file.

6. Make sure that the *Model File* name is selected.
7. Click *OK*.

Setting Up For Calculating Signal to Noise Ratio

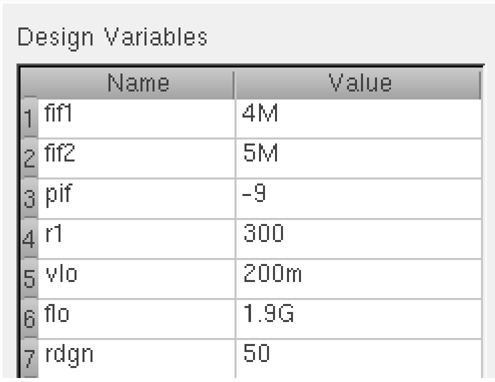
To calculate signal to noise ratio, you first need to calculate the signal using a three-tone hb analysis. You will take the signal value at 1.904G (alternately, you could look at 1.905G) and note that value for later. Then, you will set up a pnoise analysis.

Three Tone HB Analysis Setup

Setting the Design Variables

1. In the Design Variables section of the ADE L Simulation Window, ensure that your settings look like the following:

Figure A-483 Design Variables Section of ADE-L Window



Design Variables

	Name	Value
1	fif1	4M
2	fif2	5M
3	pif	-9
4	r1	300
5	vlo	200m
6	flo	1.9G
7	rdgn	50

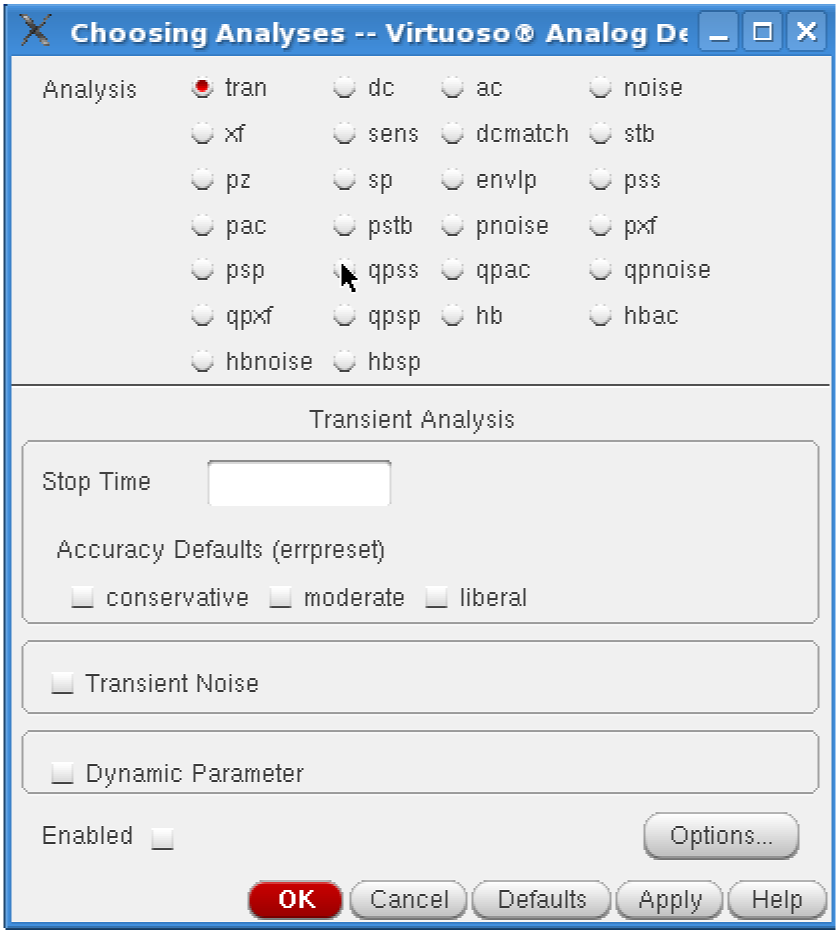
Setting Up the HB Analysis

1. Choose *Analyses – Choose* in the *Virtuoso Analog Design Environment* window. You can also select the Choosing Analyses icon ( )at the right of the Analog Design Environment window.

   The *Choosing Analyses* form is displayed, as shown blow.

Figure A-484 The Choosing Analyses Form

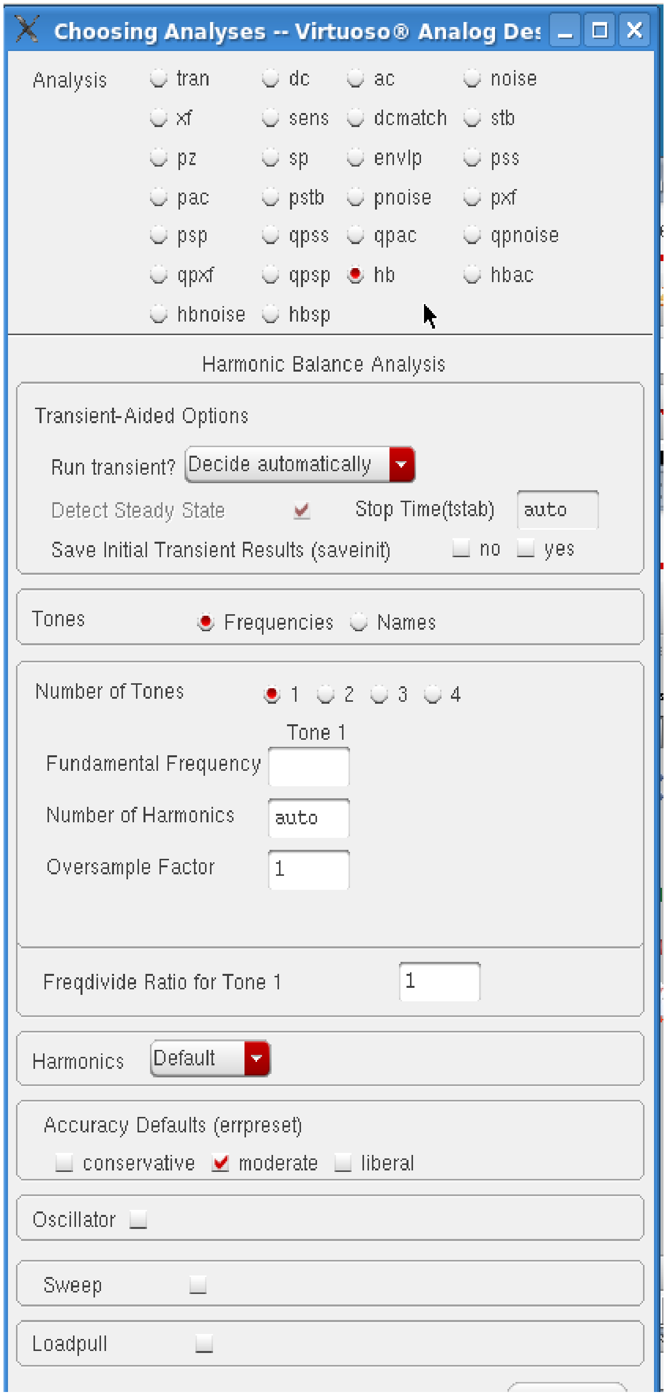
.



2. In the *Choosing Analyses form*, select *hb*. The form expands. Since the circuit is mostly sinusoidal (not strongly nonlinear) Harmonic Balance is the appropriate analysis to choose.

Figure A-485 The HB Choosing Analyses Form

.



Harmonic balance can set the harmonics automatically for the signal that causes the most distortion. This is recommended in the general case. To enable this, select *Decide automatically* or *Yes* for the *Run Transient* selection in the *Transient-Aided Options* section. This single action will cause a transient analysis to be run until steady-state is detected,and then from the transient analysis, the number of harmonics for *Tone1 (*when *Frequencies* is selected) or for the tone that has *tstab* enabled (when *Names* is selected).

3. In the Transient-Aided Options section of the form, select the following:

   a. For *Run transient?* select *Decide automatically*.

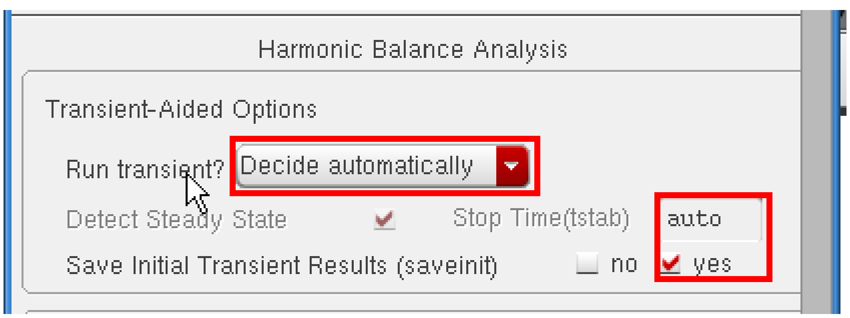
   b. For *Stop time (tstab)*, *auto* is automatically populated in the field.

   c. For *Save Initial Transient Results (saveinit),* select *yes*.

During the transient-assisted HB simulation, a transient simulation runs before the frequency domain iteration of harmonic balance. Only the LO signal is “on” during the *tstab* transient. At the end of the *tstab*, an FFT is run and its result is used as the starting point for the frequency domain iterations

All the signals are applied and the simulation is done in the frequency domain. Only the signals and the mixing products are calculated by hb.

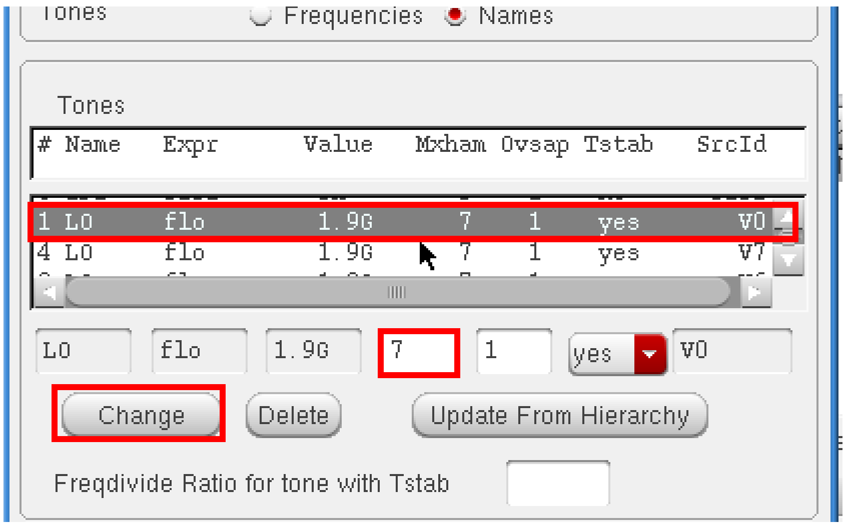
Figure A-486 Transient Assisted Harmonic Balance



4. In the *Tones* section, select *Names*. When *Names* is selected, the *Tones* portion of the form expands. All the sources in the top-level schematic are read into the form automatically.

5. Select one of the LO sources in the Tones section. You can use hb with up to four signals present in the circuit. In this circuit, there are three tones, the LO and two RF tones. The four LO signals have the same name, the same frequency, and are considered a single tone. Whenever you have two signals at the same frequency, make sure you set the *Frequency Name 1* or *Frequency Name 2* property on the source to the same name as it was done in this example. When *Tones* is set to *Names*, the simulator considers both of the sources as a single frequency.

Figure A-487 Tones Section of hb Choosing Analyses Form



Because you are using auto-tstab, *Tstab* is set to *yes* by default in the *Tones* section for the LO tone. The signal with `tstab=yes` is the signal that is used for transient-assisted harmonic balance. Only one signal can have transient assist, that being the signal with *tstab* set to *yes*.

Note: *tstab* is set to *yes* on the signal that causes the largest amount of distortion in the system. (In this example, that is the LO tone).

During the tstab interval, a transient analysis is run before the frequency domain iteration of harmonic balance. At the end of the tstab, an FFT is run and its result is used as the starting point for the frequency domain iterations.

Then all the signals are applied and the simulation is done in the frequency domain. Just the signals and the mixing products are calculated by hb.

6. Because you are not using the auto-harmonics feature, you need to change the *Number of harmonics* on the LO tone to *7*. Place the cursor in the field under *Mxham* and type *7*. Verify that *tstab=yes* on the LO tone. Click the *Change* button. Note that all four LO sources are now updated with a *`Mxham`* of *7*.

   Typically, you want to use the auto-harmonic feature (type *auto* into this field).You will be using the manual method in this example to show how this is done. Leave *Ovsap* (oversample) set to the default value of *1*.

7. Click the *Change* button when finished. The form updates. Note that the LO tones now have *Mxham=7* and *Tstab=yes*.

8. Select the IF tone in the *Tones* list box. Change the *Mxham* setting on both *IF1* and *IF2* tones from the default value of *3* to *5*. In the *Tones* section, select the line containing one

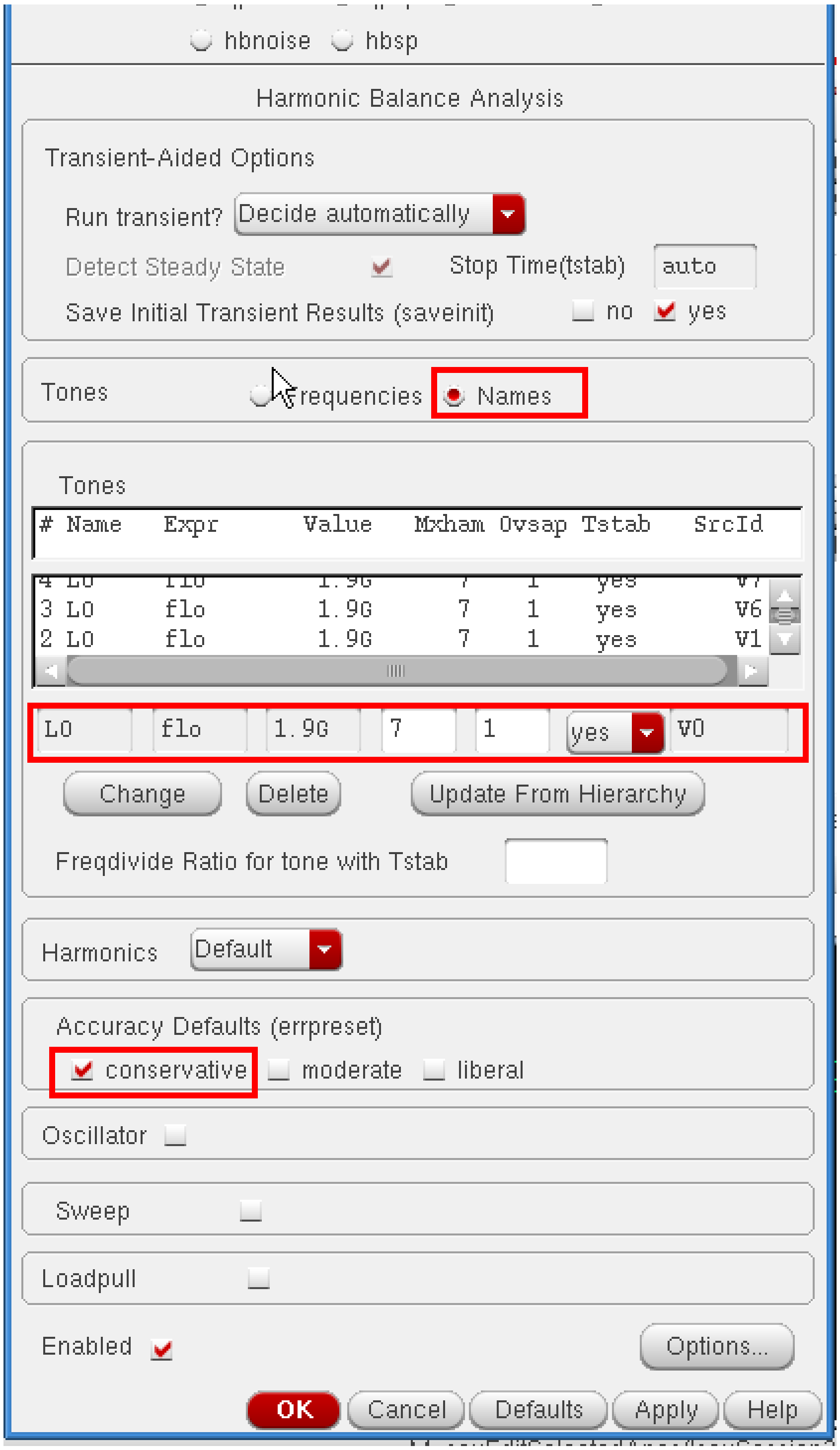
of the IF tones. For this measurement three IF harmonics is not enough. You will need to increase the number of IF tones for greater accuracy.

9. Set *Accuracy Defaults (errpreset)* to *conservative*.

10. Leave the rest of the form set to the default values.

    The *Choosing Analyses* form should look like the following:

Figure A-488 HB Choosing Analyses Form



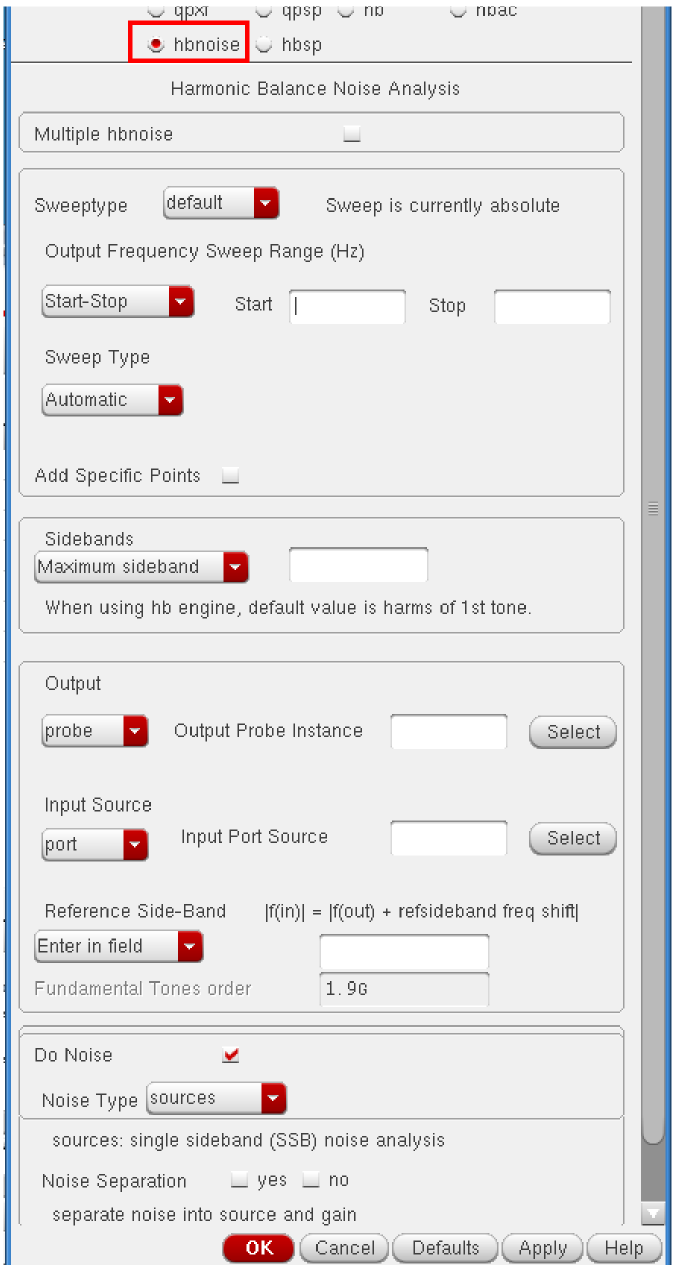
11. Click *Apply* at the bottom of the *Choosing Analyses* form.

Setting Up the HBnoise Analysis

Set up the HB Noise simulation. You will be running an hbnoise simulation with the rf tones turned on.

12. In the *Analysis* section select *hbnoise*. The form changes, as shown below.

Figure A-489 hbnoise Choosing Analyses Form



13. The default *Sweeptype* should be left at *absolute*.

14. In the *Output Frequency Sweep Range (Hz)* section, type *1.9G+1K* in the *Start* field and type *1.9G+1K+10M* in the *Stop* field.

15. In the *Sweep Type* section, select *Linear*. Type *9* in the *Number of Steps* field. Choose a number which does not allow the simulator to take a step at any of hb harmonics to avoid getting *infinite flicker noise* warnings.

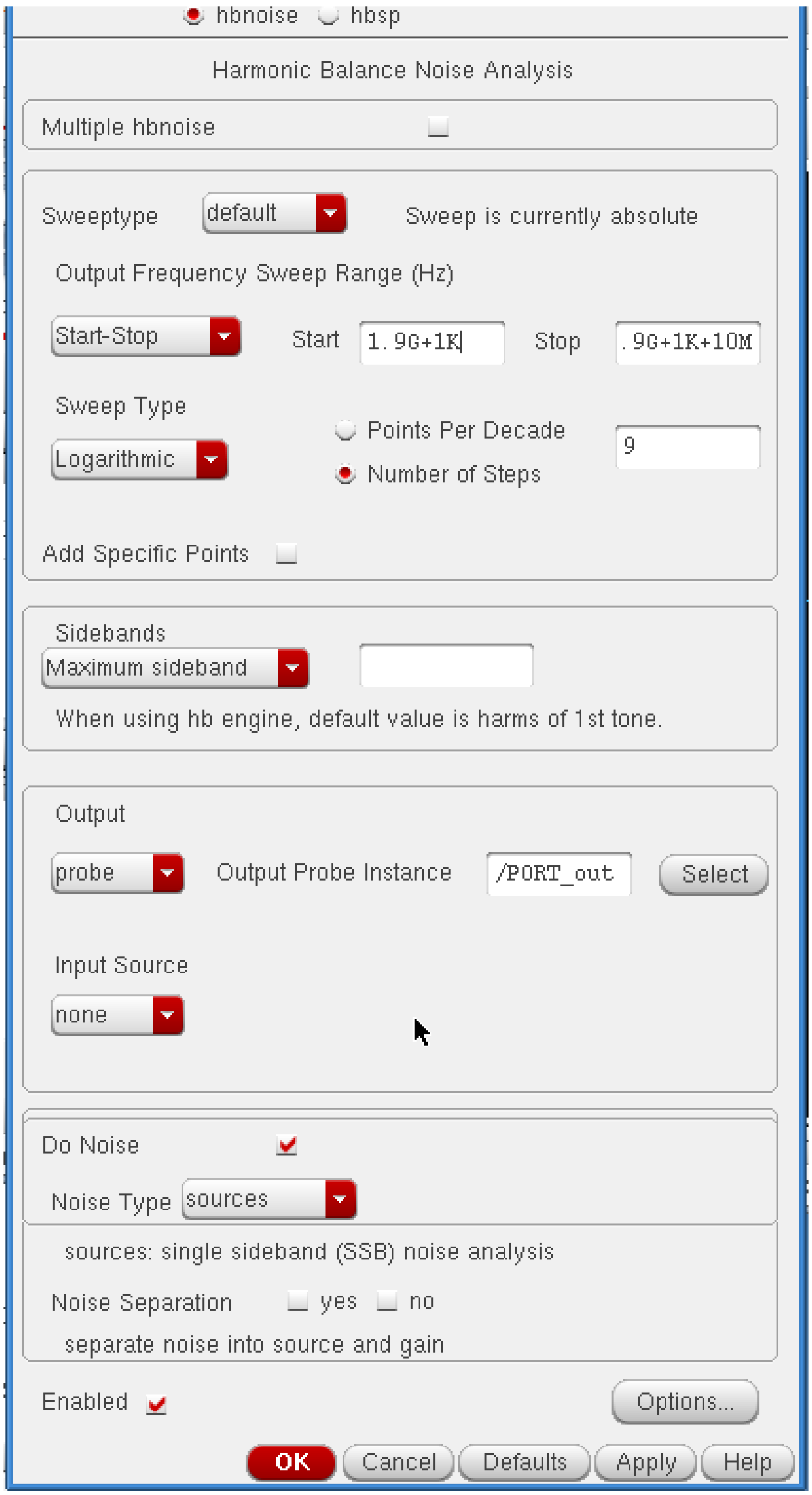
16. Leave the *Maximum sideband* field blank. In HBnoise, this value should either match the number of *harmonics* in HB analysis, or be left blank, in which case it uses the number of *harmonics* in HB.

17. In the *Output* section, select *probe*. Click *Select* to the right of the *Output Probe Instance* field In the schematic, click the source to the left of the *Output* label. Alternately, you can type */PORT_out* in the *Output Probe Instance* field.

18. For *Input Source*, select *none*. You only need to measure output noise and in this setup, you cannot measure noise figure.

Hbnoise analysis fundamentally calculates the output-referred noise, and it is the output frequency range that is specified in the hbnoise *Choosing Analyses* form. Mixing can occur with harmonics of all the input frequencies, and to get this, leave the *Maximum sideband* field blank. The *Choosing Analyses* form should look like the following:

Figure A-490 hbnoise Choosing Analyses form for Output Noise Measurement



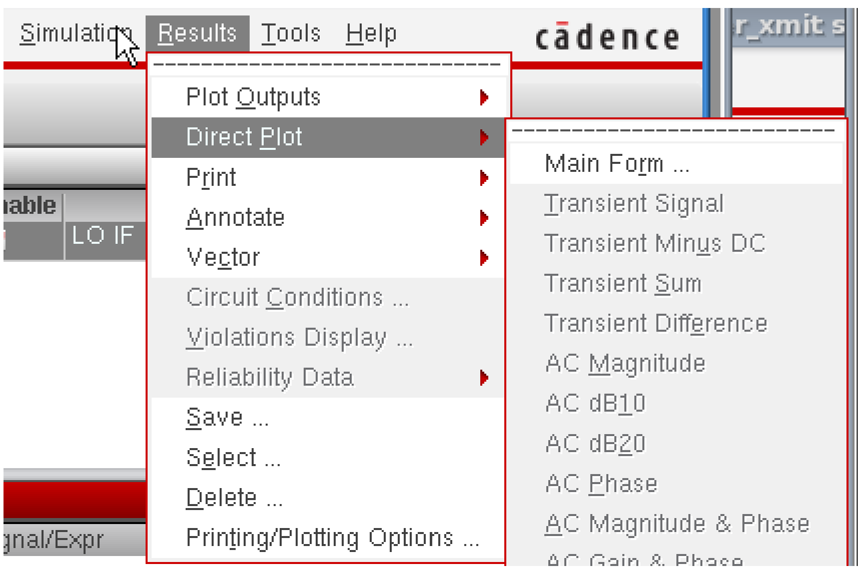
Running the Simulation and Plotting the Results

1. Start the analyses by clicking the green arrow icon.  in ADE or in the Schematic Editor.

   This netlists the design and runs the simulation. A SpectreRF status window appears (`spectre.out` log file). When the analysis has completed, you may iconify the status window.

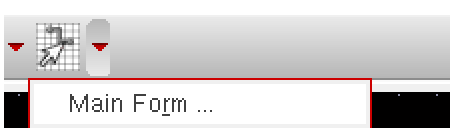
2. When the simulation has finished, choose *Results - Direct Plot - Main Form* in the ADE window.

Figure A-491 Invoking Direct Plot Form After Simulation



You can also open the *Direct Plot Form* by clicking the *Direct Plot* button in the Schematic window and choosing *Main Form*, as shown below.

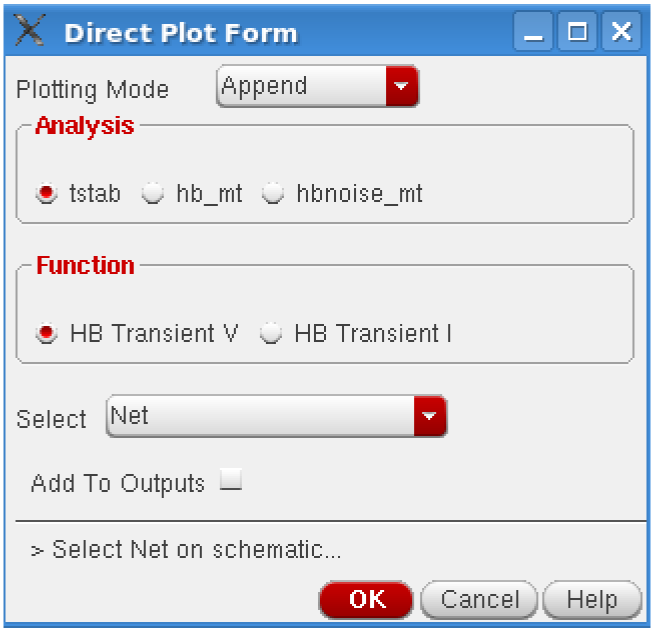
Figure A-492 Direct Plot Icon



The *Direct Plot Form* is displayed, as shown below.

Figure A-493 Harmonic Balance Direct Plot Form

.

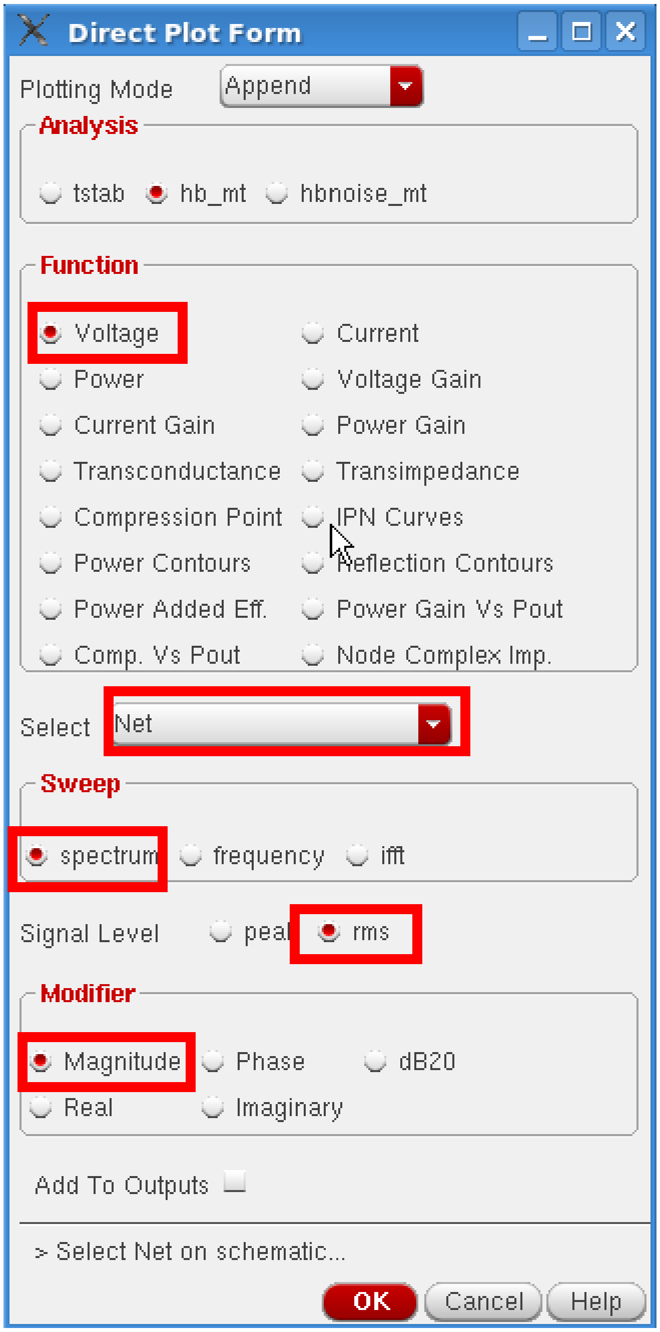


In the *Direct Plot Form*, you will notice that there are three available analyses: *tstab*, *hb_mt*, and *hbnoise*. The transient-assisted results are accessible by selecting the *tstab* analysis. The hb results are accessible by selecting the *hb_mt* analysis. Note that when there is more than one tone, you will see *hb_mt* (signifying harmonic balance multi-tone). A single tone simulation will show *hb*. The hb results are accessed by selecting the *hbnoise* analysis.

3. In the *Analysis* section, select *hb_mt*. The *Direct Plot Form* expands.
4. In the *Function* section, verify that the default value *of Voltage* is selected.
5. Select *Net.*
6. In the *Sweep* Section, select *spectrum*.
7. In the *Signal Level* section, select *rms* .
8. Inthe *Modifier* section, select *Magnitude*.

The *Direct Plot Form* should look like the following:

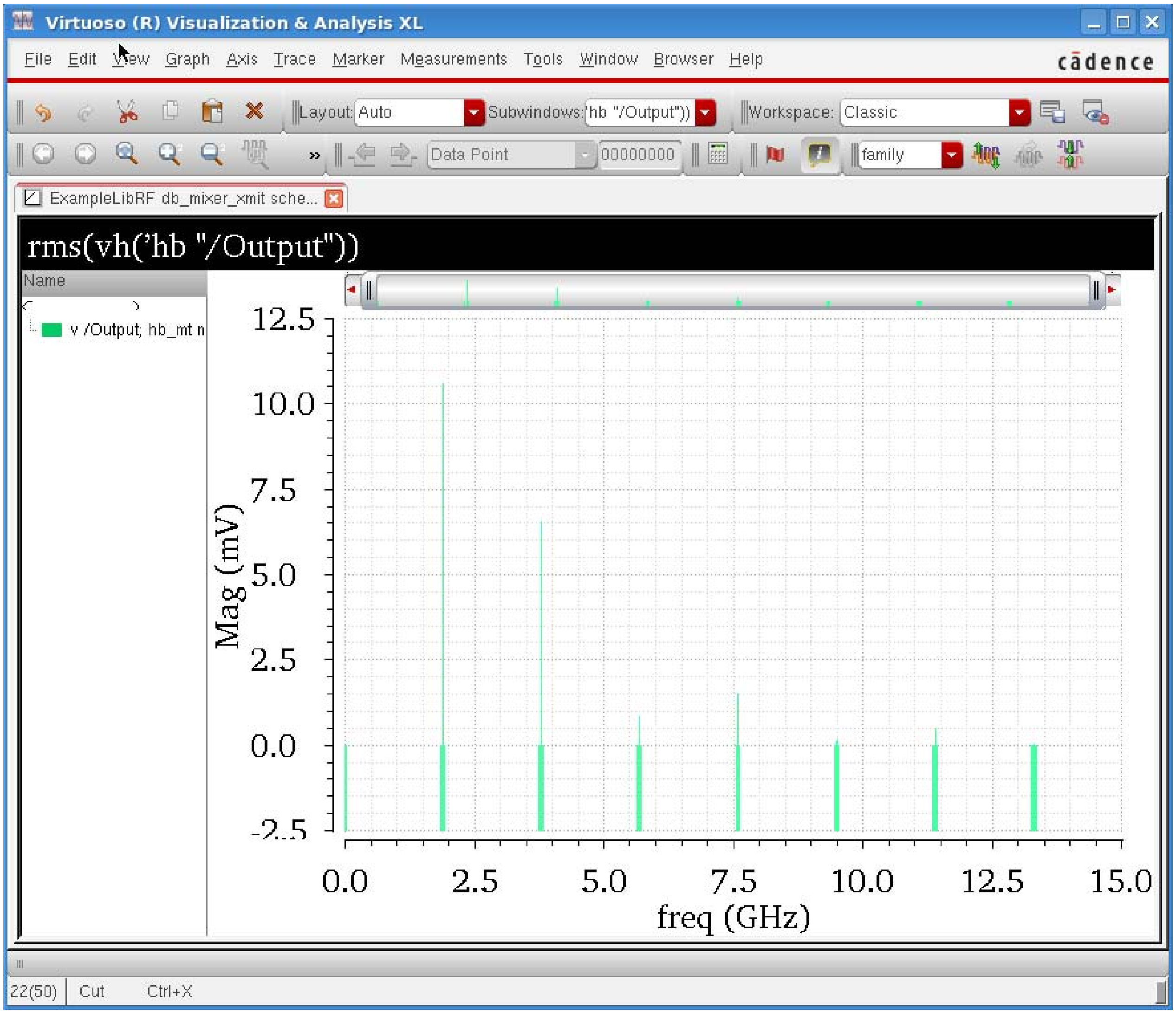
Figure A-494 Direct Plot Form



9. Click on the net labeled *Output* on the right side of the schematic.

10. The plot of Voltage Spectrum vs Frequency is displayed in the waveform window, as shown below.

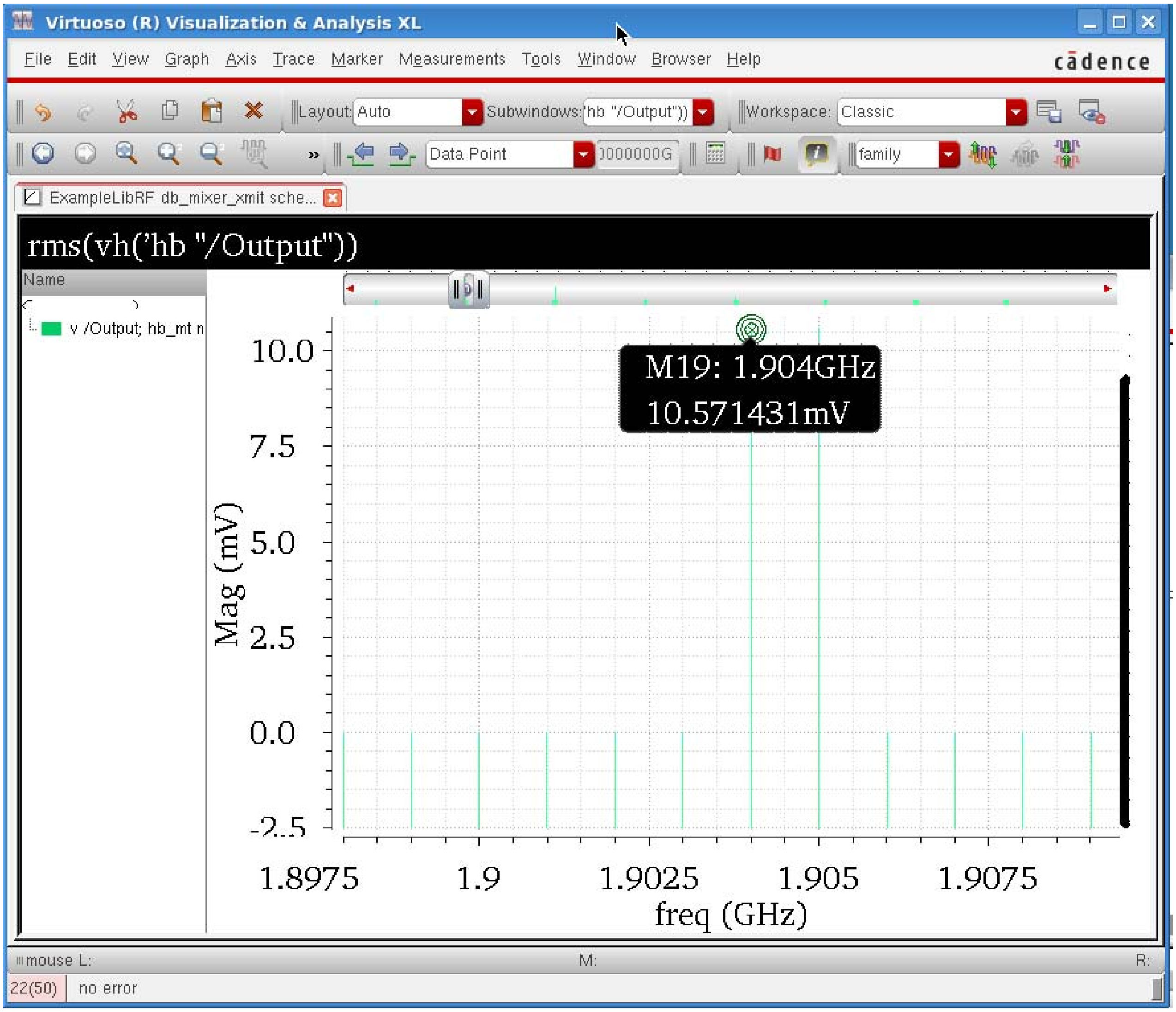
Figure A-495 RMS Voltage vs Frequency



11. Zoom into the region of the graph near 1.9GHz. Press and hold the right mouse button while moving the cursor to draw a box around the signals near 1.9GHz. When you release the mouse, you are zoomed in. You may need to do this several times to get zoomed in to just the area around 1.9GHz.

12. Place a marker at 1.904GHz (or 1.905GHz). Place the mouse over the 1.904G trace at the top and select the bindKey m to place the marker. The rms voltage at 9.104G is 10.57134mV, as shown below. This is the *signal* part of the Signal-to-Noise ratio you will be calculating in the next step.

Figure A-496 Zoomed in RMS Voltage vs Frequency

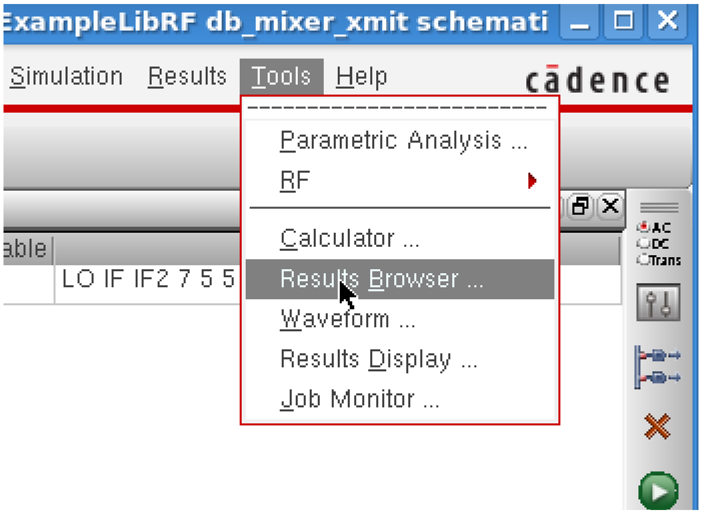


Plotting the Noise Results

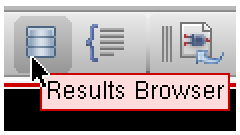
1. When the simulation has finished, plot the *Voltage Spectrum* in *V/sqrt(Hz)* from the Results Browser.

   In the *Analog Design Environment* window, choose *Tools - Results Browser.*

Figure A-497 Invoking the Results Browser After Simulation

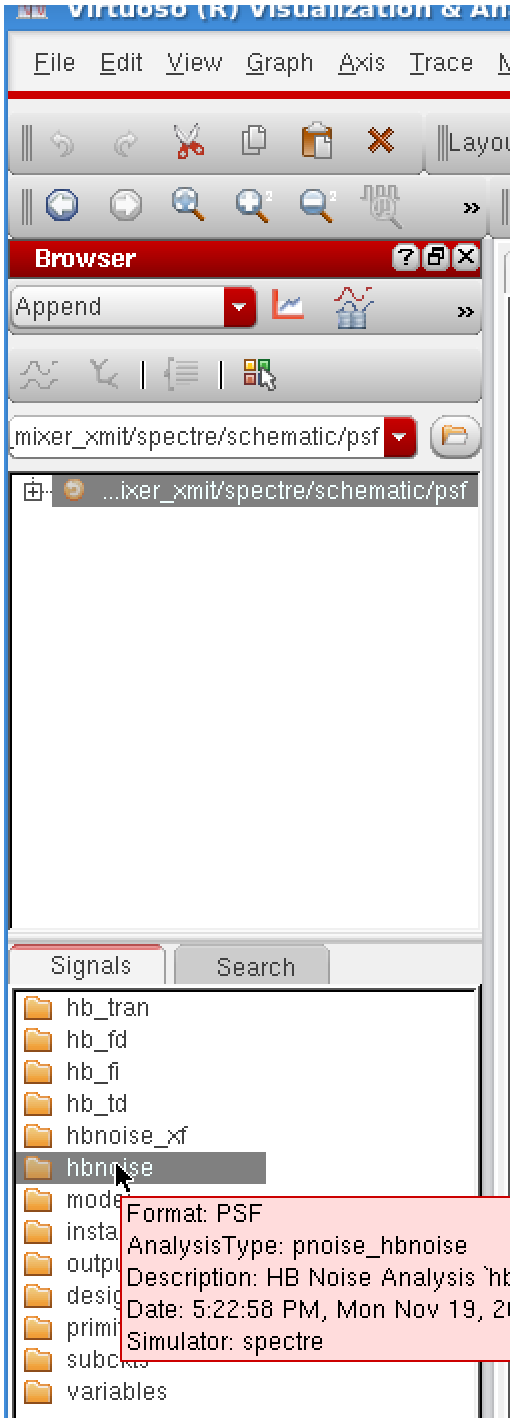


Alternately, you can click the Results Browser icon in the Schematic, as shown below.



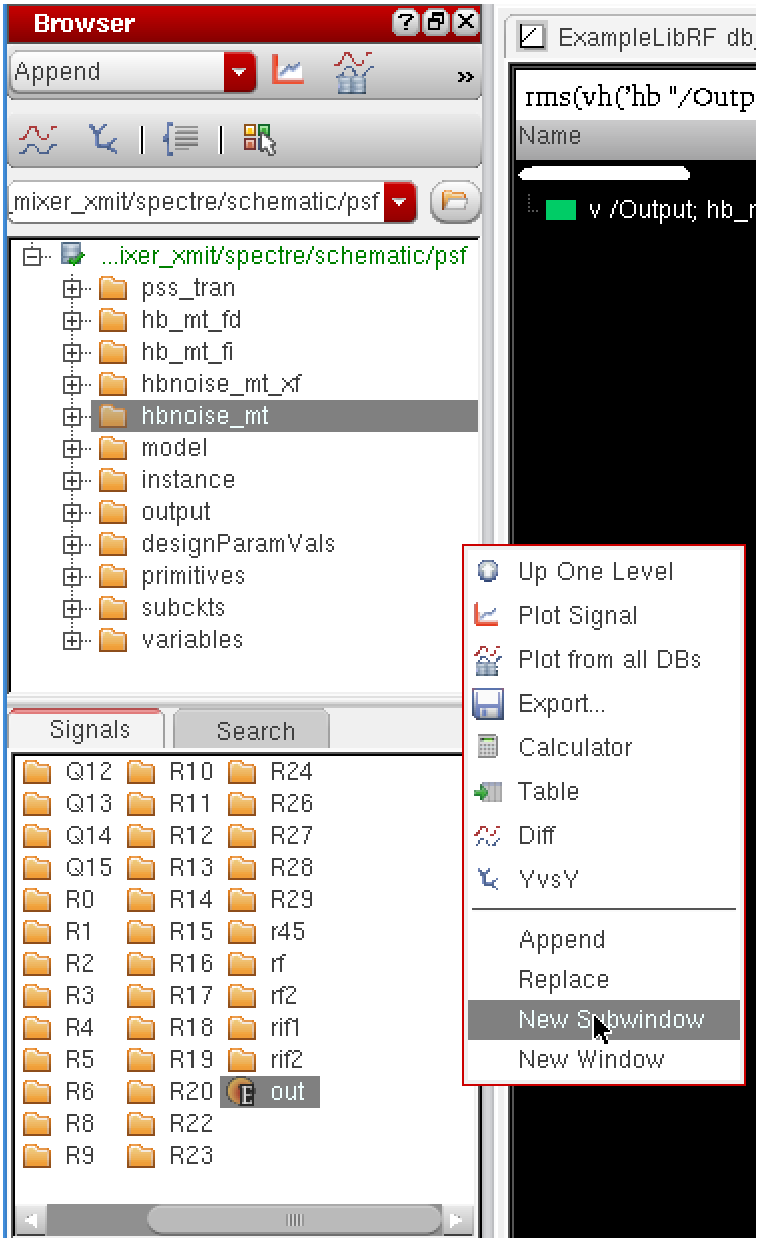
2. In the Results Browser, select *hbnoise* .

Figure A-498 Selecting hbnoise results using the Results Browser



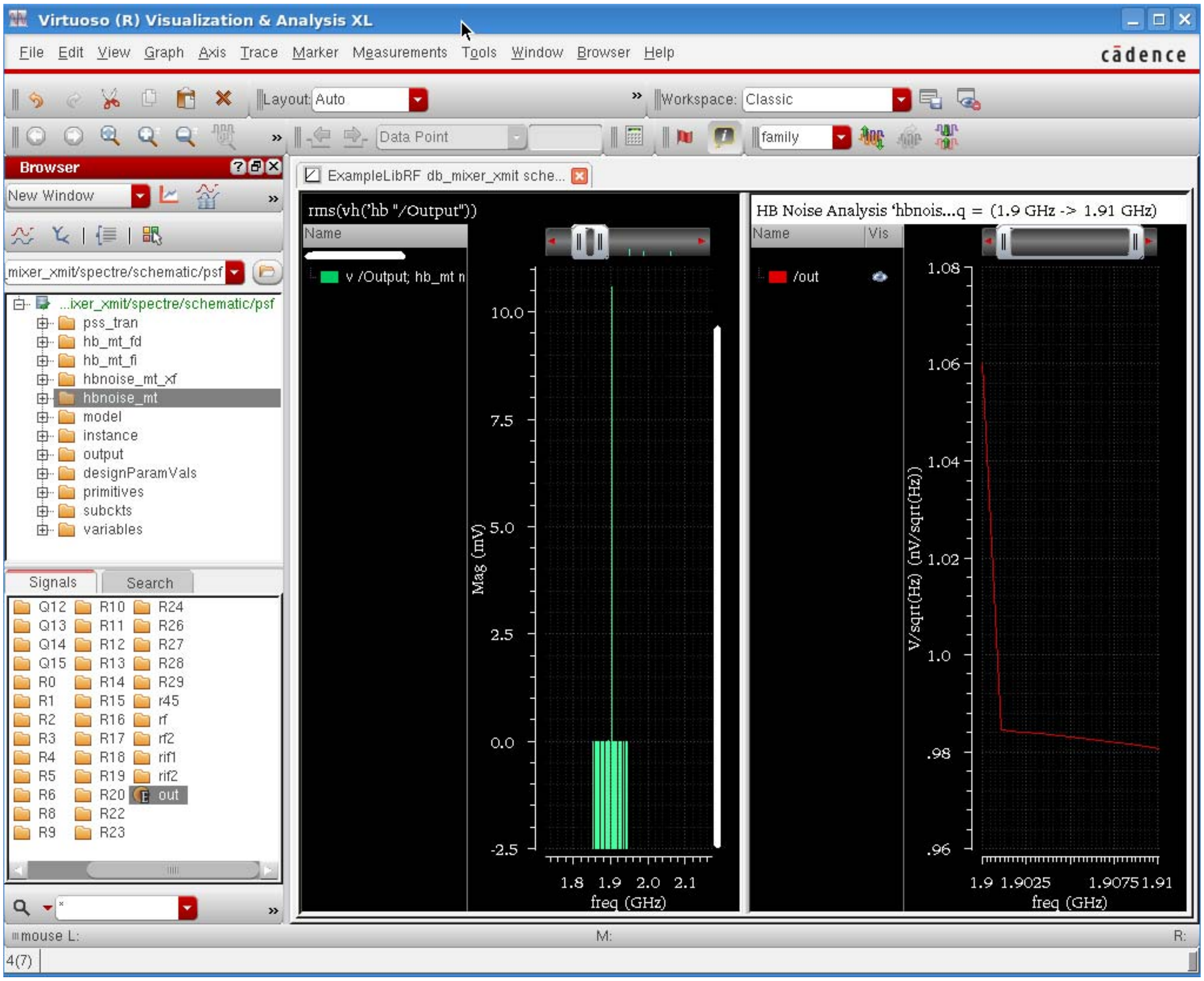
3. Double-click on the *hbnoise* folder shown highlighted in the above figure. The folder expands. Clickthe right mouse button on the *out* signal, and select *Plot Signal* from the context menu, as shown below.

Figure A-499 Selecting out Net in the Results Browser



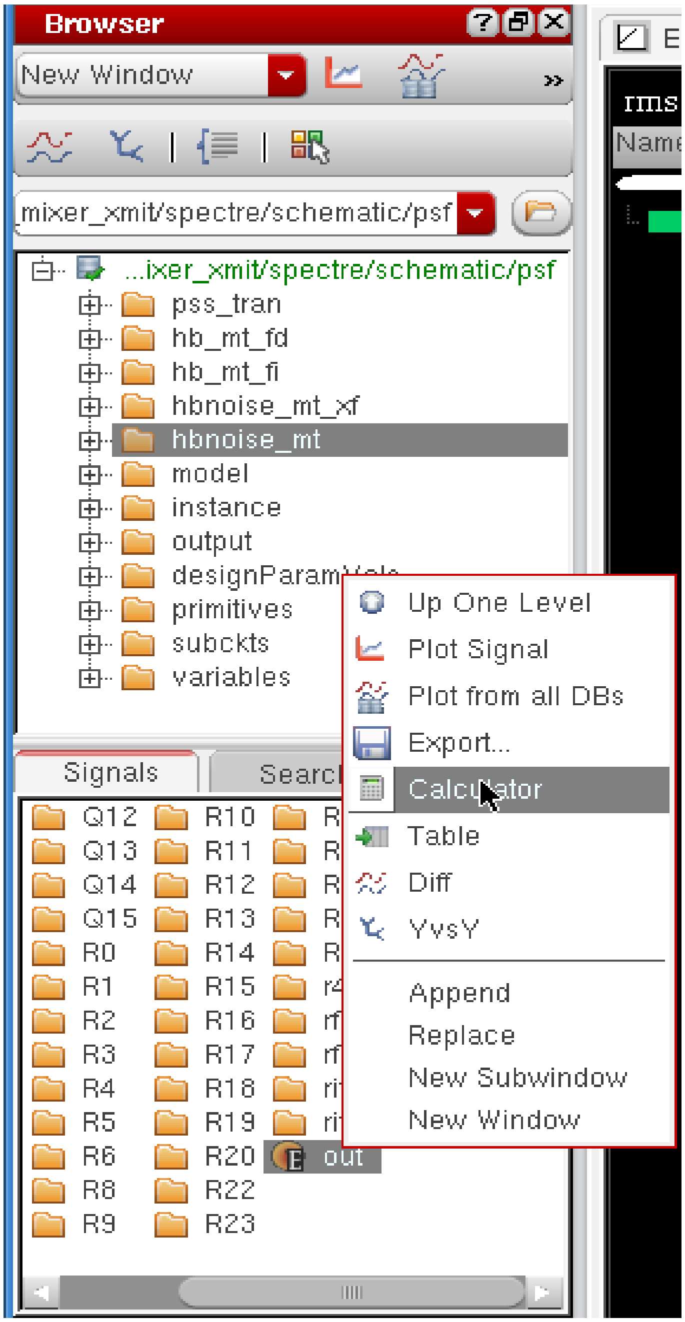
4. Select and release the mouse over the *New Subwindow*. The Noise is plotted in *V/sqrt(Hz)*, as shown below.

Figure A-500 Noise Results



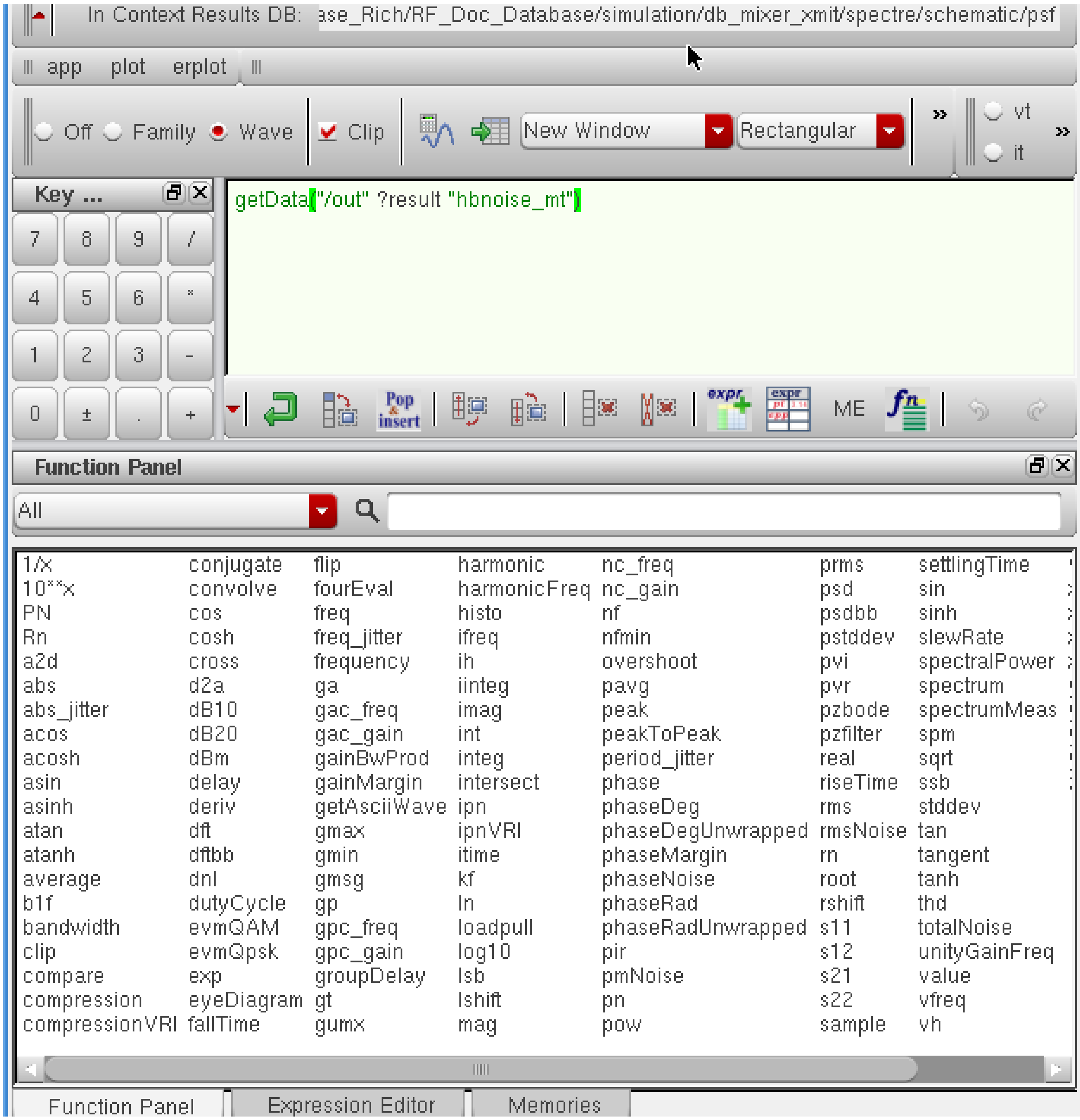
5. Note that the units are in *V/sqrt(Hz)*. You want to calculate the noise voltage in a 1Hz chunk of the spectrum. To do this, open the Calculator by clickingthe right mouse button on the *out* signal in the Results Browser and selecting *Calculator*.

Figure A-501 Sending Output Waveform to Calculator



The information is added in the Calculator Buffer, as shown below

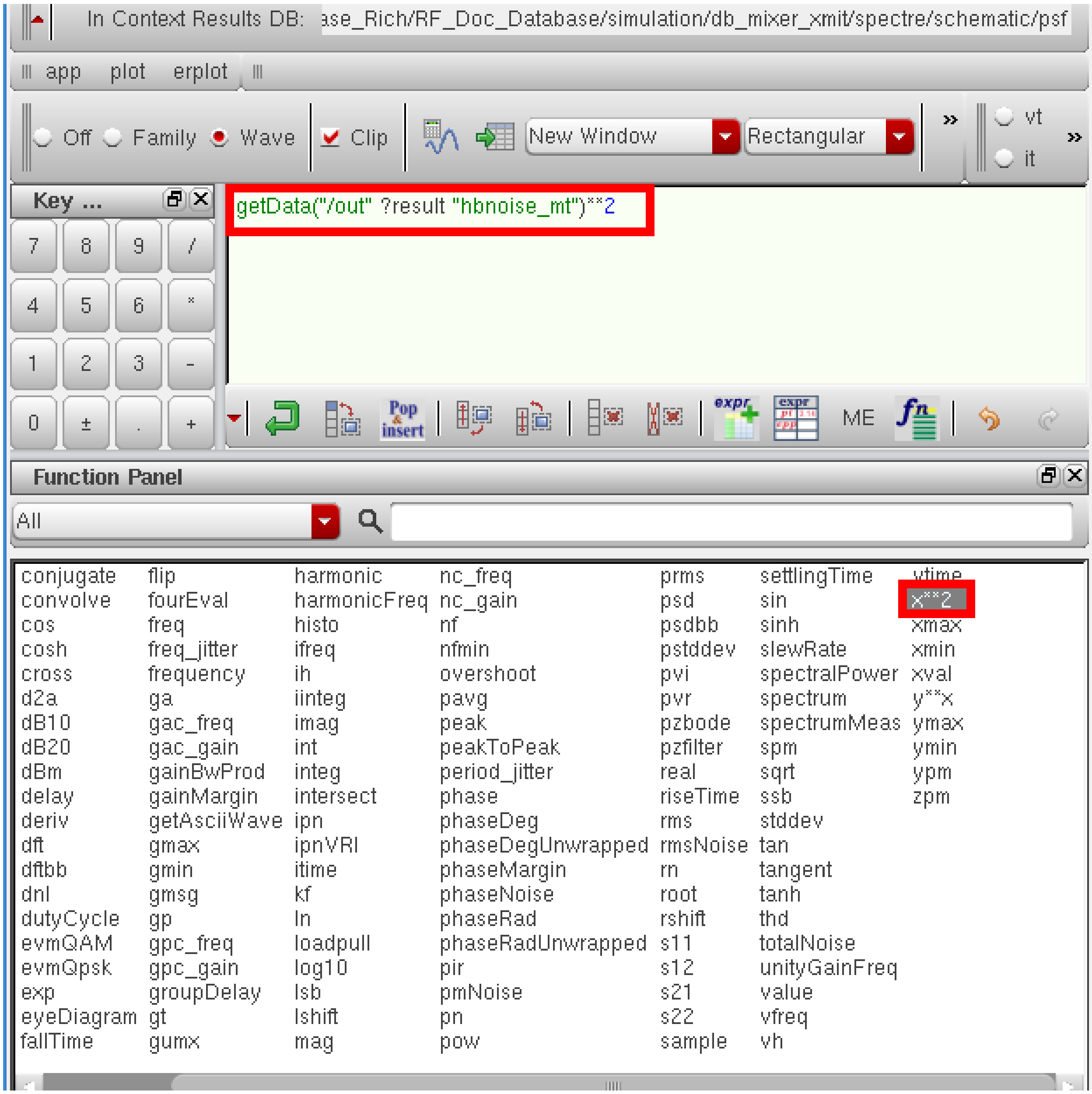
Figure A-502 Output Noise Data Entered into Calculator Buffer



6. You want to look at the Noise Voltage over a 1 Hz bandwidth.

   In the Calculator you need to square the voltage to get it in units of power. $P=V^2/R$, where R is some arbitrary value. In the lower part of the Calculator, in the *Function Panel* section, select *x**2*. The Calculator Buffer updates, as shown below.

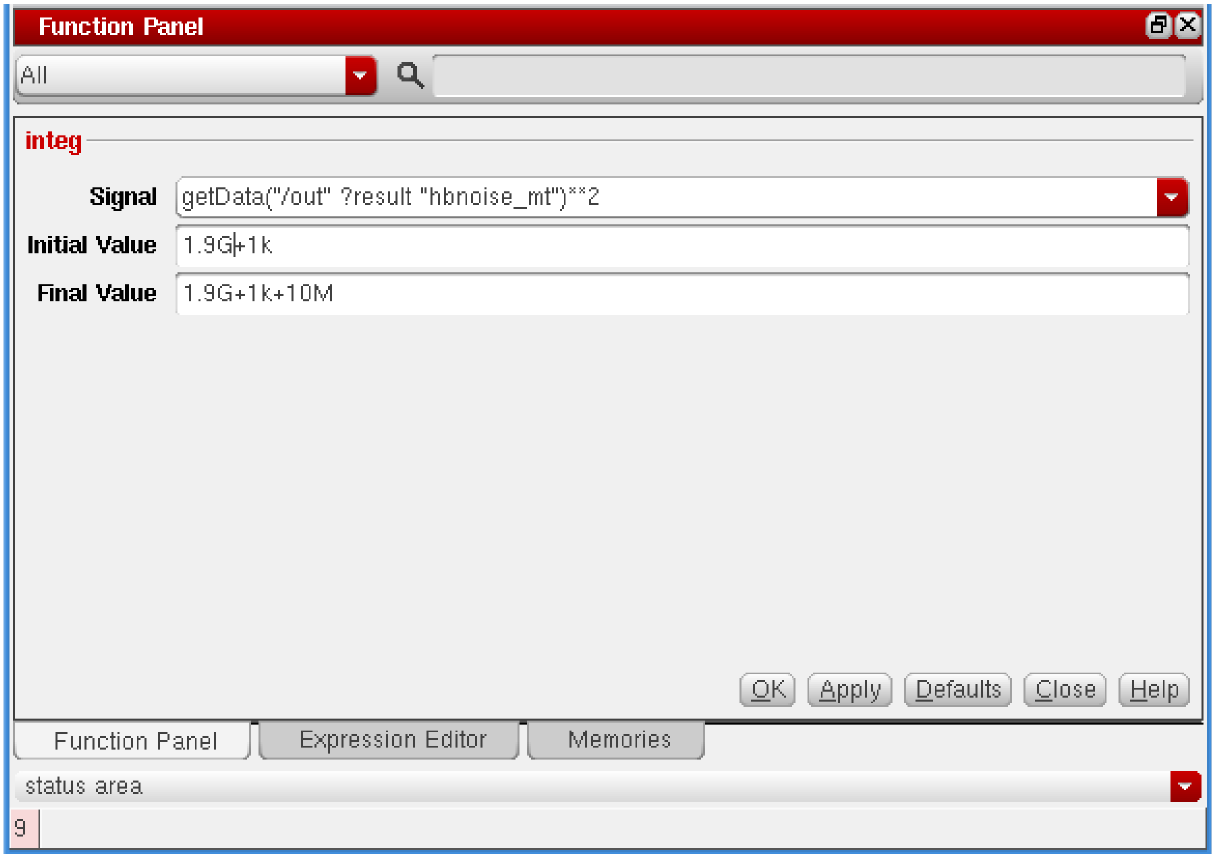
Figure A-503 Squaring the Noise Voltage



7. Next, you need to integrate the power over bandwidth. Select the *integ* function from the *Function Panel*. The form changes. You are integrating the power from `1K` to `10M` (these are the same values that were in the hbnoise *Choosing Analyses* form).

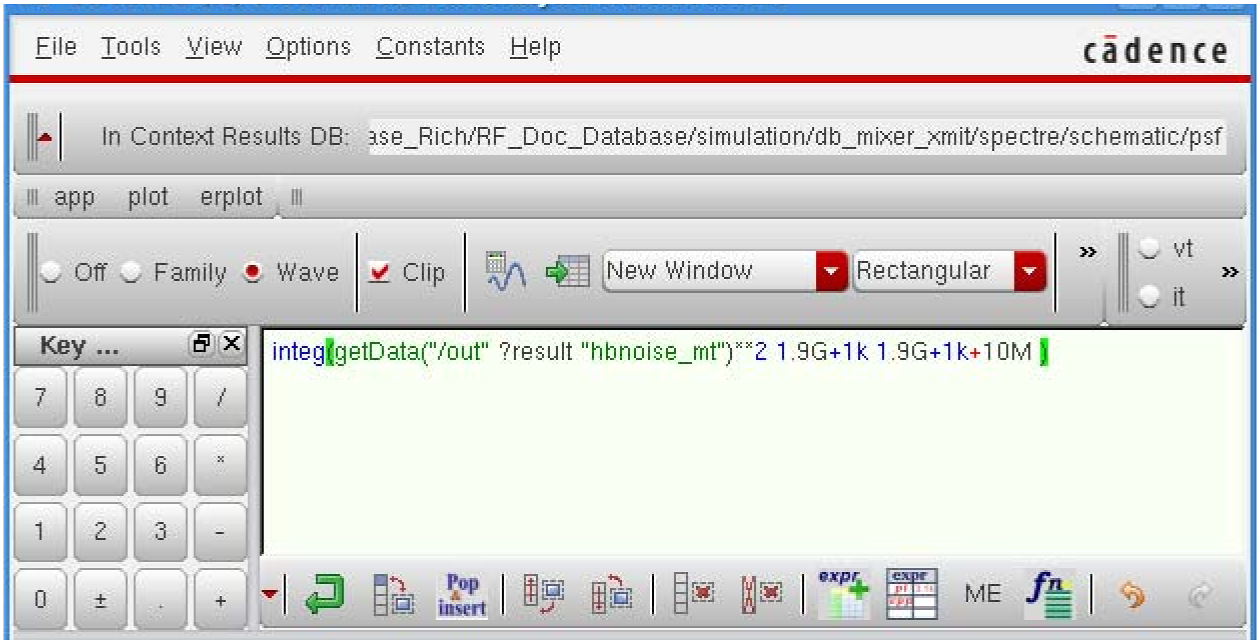
8. Verify that the signal is automatically populated in the *Signal* field. If it is not auto-populated, select *1.9G+1K* as the *Initial Value* and *1.G+1K+10M* as the *Final Value*. Note that this is the same range used in the hbnoise *Choosing Analyses* form.

Figure A-504 Integrating the Squared Noise Voltage over a 1Hz Bandwidth.



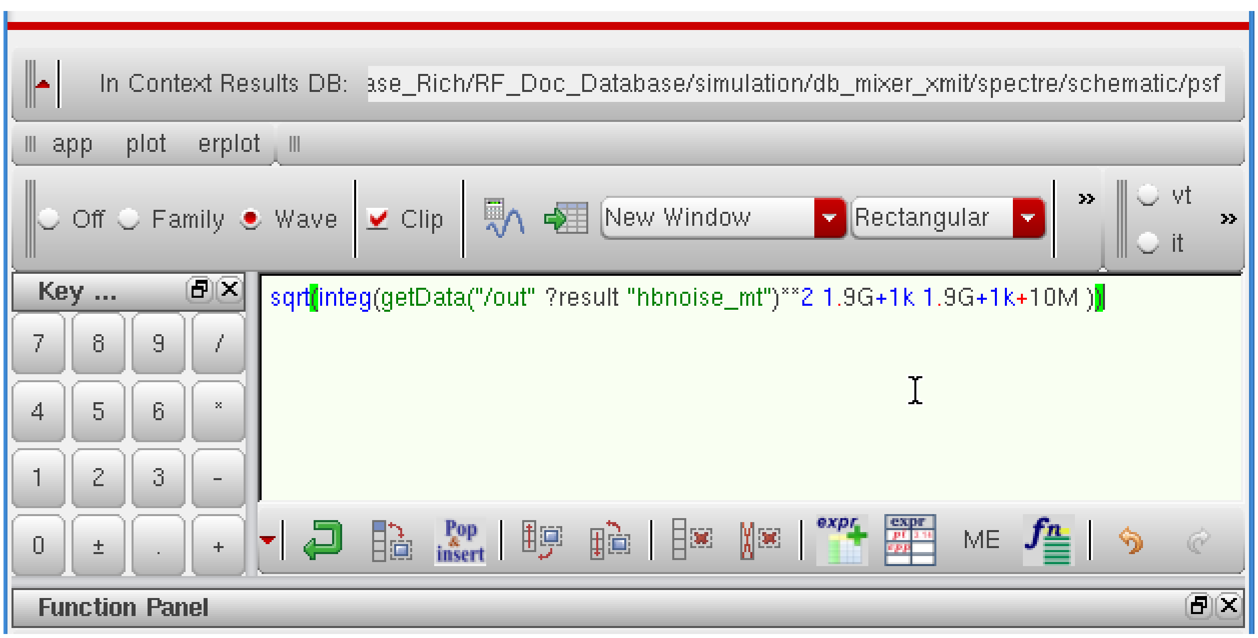
9. Click *OK*. The Calculator buffer is updated, as shown in the figure below.

Figure A-505 Integrating the Noise Power over a 1Hz Bandwidth



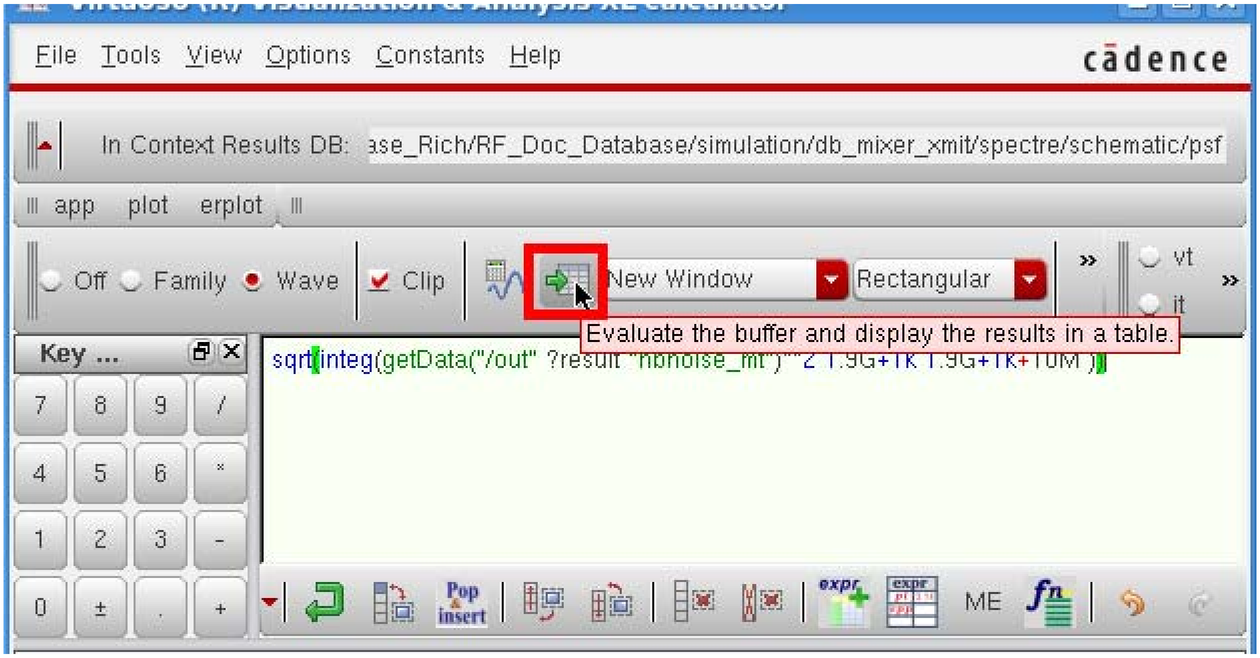
10. Finally, you need to take the square root of the expression in the buffer to put it back into terms of voltage. Click on the *sqrt* function in the *Function Panel.* The calculator buffer looks like the figure below:

Figure A-506 Integrated Noise Voltage



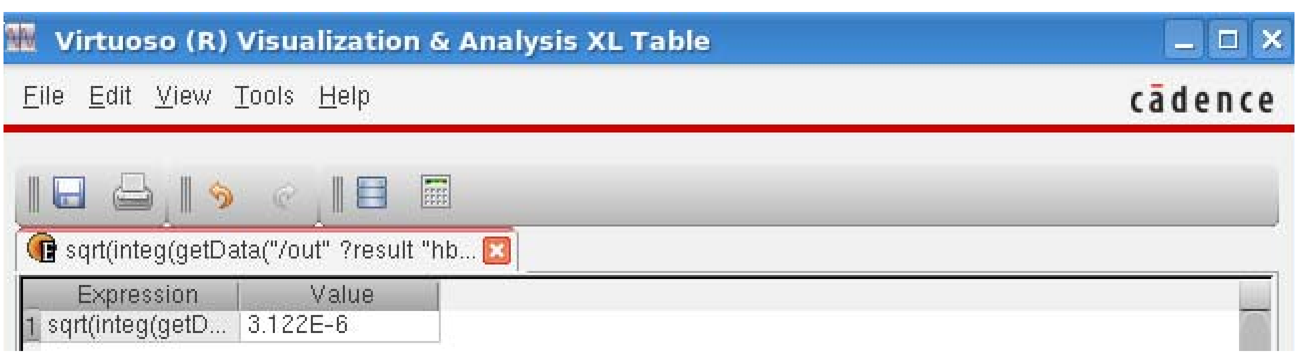
11. Click the Table icon ( )to evaluate the Calculator buffer contents and display the results in a table, as shown below:

Figure A-507 Evaluating the Calculator Buffer



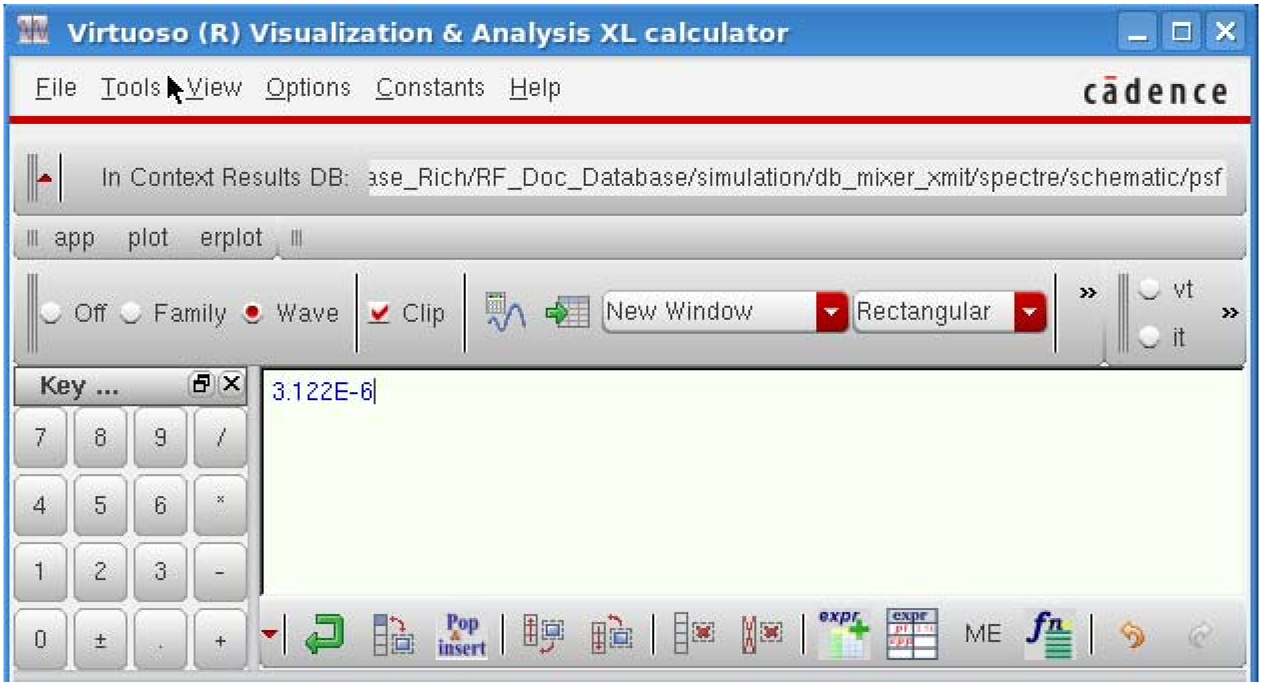
12. The result is sent to the Table, as shown below.

Figure A-508 Calculating Integrated Noise Voltage Using the Table Icon.



Alternately, if you click the *Plot* icon, the value for noise voltage appears in the Calculator buffer, as shown below.

Figure A-509 Calculating Integrated Noise Voltage Using the Plot icon



The Noise voltage is 3.122E-6 Volts.

Now you are ready to calculate the Signal to Noise ratio (SNR). You first plotted the rms Voltage (from a three-tone harmonic balance simulation) at 1.904G to be 10.57134mV. Next, you divide the Signal by the Noise to get the SNR.

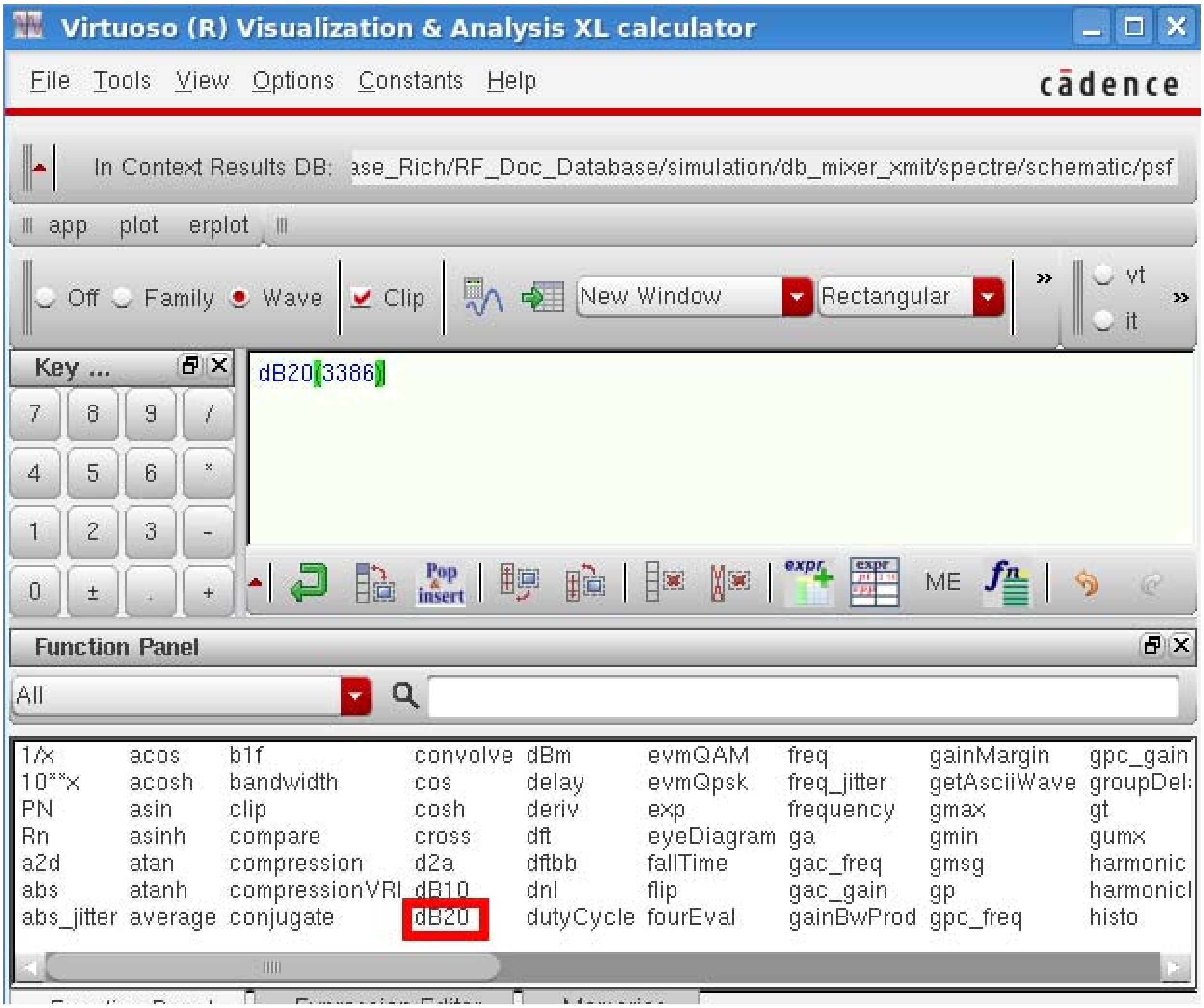
$$SNR = \frac{Signal}{Noise} = \frac{10.57134e^{-3}}{3.122e^{-6}} = 3386$$

To get the SNR in dB at 1.904GHz, take the log of the SNR value and multiple by 20.

$$SNR = 20 \bullet \log(3386) = 70.59dB$$

You can easily do this in the Calculator using the *dB20* Math function, as shown below.

Figure A-510 Using the dB20 Math Function



Click the *Plot* button ( ). This gives 70.59dB.

Note that the SNR is fairly high.

Summary

The Mixer section showed how to simulate and make typical measurements on a receiver mixer and a transmit mixer.

In the Receive Mixer Measurement section, the following measurements were shown:

- Mixer Conversion Gain using hb and hbac analyses

- LO to IF leakage using hb analysis
- Noise Figure measurements using hb and hbnoise analyses
- 1dB compression point, desensitization, and blocking using hb, hbac, and hbnoise
- Third-Order Intercept measurement using three-tone harmonic balance
- Rapid IP2 and Rapid IP3 using specialized hbac analysis
- Compression Distortion Summary using hb and specialized hbac analysis.

In the Transmit Mixer Measurement section, the following measurements were shown:

- Image rejection using hb analysis.
- Three-tone swept IP3 (large signal) using multi-tone hb analysis.
- Signal-to-Noise Ratio using the hb and hbnoise analyses.

For more information on simulating Mixers, please refer to the chapters in this user guide and the *Virtuoso Spectre Circuit Simulator RF Analysis Theory Guide*.

B

Top-down RF Design Methodology

This section describes a methodology for designing analog RF subsystems that fit into larger DSP systems. In particular, this section describes how to use a canonical set of top-down behavioral baseband models for exploring RF architectures in the analog design environment. These models come from the following categories in *rfLib*

- Category *top_dwnBB* contains models of common RF function blocks.
  - The default view of each model is the baseband view (called *veriloga*).
  - All models in this category also have a differential passband view (called *veriloga_PB*).

  The only exceptions are the BB_loss and VGA_BB models. They are meant only for baseband analysis and have no passband view.
- Category *top_dwnPB* contains single-ended passband versions of the baseband models.
- Category *measurement* contains the instrumentation block and baseband signal generator models used to make RF measurements. These elements are not part of an RF architecture. They simply facilitate RF measurements and diagnostics.
- Category *testbenches* contains the test circuits used in this chapter to define the model specifications in the *rfLib*. Where possible, the models are specified in terms of standard RF measurements. The most precise way to describe a measurement is with a test circuit, set up instructions, and sample measurements. The circuits in the testbenches category serve that purpose

These models provide RF designers with a fast method to map RF system specifications into detailed RF designs. The baseband models facilitate fast evaluation of candidate RF architectures specified with DSP metrics. The passband views of the baseband models provide a behavioral system testbench for checking detailed designs of individual RF system components.

Baseband models are behavioral models and all behavioral models sacrifice some accuracy for increased simulation speed. Such sacrifices are usually acceptable in architectural studies because many implementation-dependent details do not affect high level decisions.

The modeling approach taken in top-down design is to simulate only those effects that drive the decisions at hand.

Baseband modeling in no way replaces passband modeling. Some effects missed by equivalent baseband models can affect high level decisions. However, the application of baseband models early and passband models later minimizes the number of slow simulations needed at the lower levels of design abstraction. Baseband models help you to quickly weed out designs that would surely fail tests simulated with passband models.

The success of a modeling approach to top-down design hinges on knowing how the models fit into the design flow and knowing exactly what each modeling parameter means. This chapter has two goals:

- To describe the top-down design flow, from a modeling perspective, for baseband modeling
- To define, as clearly and concisely as possible, the parameters that specify the models

Top-Down Design of RF Systems

Ideally, the digital signal processing, or DSP, team specifies an RF subsystem that fits snugly into the DSP system. A *snug fit* means that

- The specified RF subsystem does exactly what it needs to do at the lowest possible cost
- A functional specification exists that describes requirements for the RF subsystem

In a top down design flow like the one shown in Figure B-1 on page 2468, the DSP team writes a functional specification for an RF subsystem that has not yet been designed. The functional specification describes what the RF subsystem should and should not do without describing how to build the RF subsystem.

The functional specification supplied by the DSP team describes the RF subsystem at the highest possible level of abstraction. At this point behavioral models can be specified rather than measured. This early in the design cycle, the functional specification might well be incomplete or inconsistent. A good top-down design flow can detect problems, such as omissions and inconsistencies in the design, early in the design cycle when they are easier and less expensive to fix. Problems detected later in the design cycle can be much more costly and very difficult to resolve.

Using the functional specification supplied by the DSP team and the behavioral baseband models from *rfLib*, the RF system designers can easily explore RF architectures in the analog design environment. The baseband models facilitate fast evaluation of candidate RF architectures specified with DSP metrics. By switching to the passband views of the baseband models, the RF design team maps DSP measurements to RF measurements. The passband views of the baseband models provide a behavioral system testbench for checking detailed designs for individual system components.

Using the functional specification and exploring and testing with the baseband and passband models, the RF team can efficiently create a detailed design specification that fully describes the RF subsystem. The design specification can include detailed instructions for building the RF subsystem. At this stage of the design cycle, everything that is known about the design is described at the lowest level of abstraction.

You can now extract behavioral models of a detailed design from simulated measurements. The problem remains that detailed designs usually do not exist until the project is complete. To jump directly to a detailed design implies that the design flow is bottom up. Bottom up flows are important in many projects, but not in all.

DSP and RF designers sometimes have trouble communicating through specifications because the two groups deal with different metrics. For example, DSP designers deal with *bit error rates* and *error vector magnitude statistics* whereas RF designers deal with *intercept points* and *noise figures*.

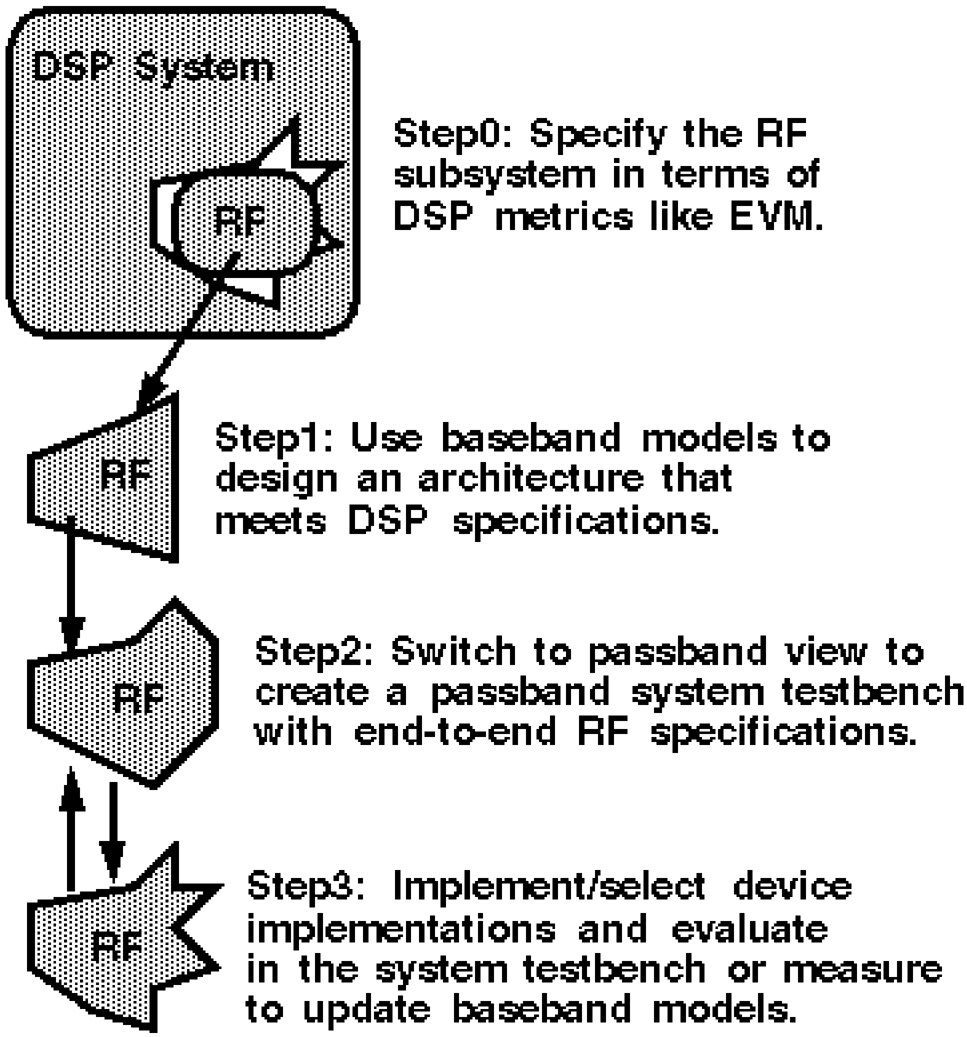
The new models described here are designed to help RF system designers in two ways.

- First, the baseband models enable RF system designers to quickly explore the RF architectural space, as specified by the DSP metrics, while letting the RF engineers specify the RF system components with RF metrics. The circuit implementations of the RF system components are easier to design and test when the components are specified with standard RF metrics.
- Second, the baseband models can be switched quickly to a passband views where the RF system model can generate end-to-end RF metrics. With end-to-end metrics, the new view can quickly simulate how the detailed design of a particular RF system component affects end-to-end performance.

Use Model for Top Down Design

The following steps outline the RF design process with focus on the early phases of the design as illustrated in Figure B-1 on page 2468.

Figure B-1 The Top Down Design Flow and Use Model



Specify the RF Subsystem in Terms of DSP Metrics

Before you begin the RF subsystem top-down design flow, the DSP design team should completely specify the RF subsystem in terms of DSP metrics. This preliminary step distinguishes the end of the DSP design flow from the beginning of the RF top down design flow and formally hands-off the RF subsystem design specifications to the RF design team.

Explore Candidate Architectures with Baseband Models

The first step in top down RF design is to select a candidate RF architecture. An RF architecture is a set of interconnected RF function blocks that, taken together, describe how a receiver or transmitter operates. You specify each function block in terms of standard RF metrics such as IP3, gain, bandwidth, and noise figure.

The models you use early in the design cycle as you explore candidate RF architectures must run fast. Each simulations can span hundreds of symbols and each symbol can easily span thousands of RF carrier cycles. The space defined by the function block specifications in each candidate architecture is far too vast to explore with slow, highly precise models. Models used for architectural exploration must quickly reduce the design space down to a size that can be explored with more precision.

The most efficient models for architectural exploration suppress the RF (IF) carrier and are called *baseband* models. In contrast, the *passband* models (introduced in the next step) do not suppress carriers.

You can use the Circuit Optimizer during architectural exploration to help balance the function block specifications for a candidate architecture. For example, you can use the Circuit Optimizer to minimize RMS EVM while ensuring that other measurements stay within acceptable limits.

When you have determined the nominal specifications for each function block, you must put tolerances around them. In the analog world *specifications without tolerances are meaningless*. The tolerance space is usually explored with some mix of experience, feasibility, a variety of analyses, and outright arbitrary decisions.

There are several ways that you can use the baseband models to test candidate tolerances as well as to determine some tolerances analytically.

One way to test a candidate set of tolerances is to run a Monte Carlo analysis on the metric of interest, like RMS, EVM, or signal-to-noise ratio (SNR).

Another approach is to use the Circuit Optimizer *in reverse*, as a de-optimizer, to determine worst case performance.

Yet another approach is to compute each tolerance separately from a parametric plot. When you have determined all but 2 or 3 tolerances, you can use a multidimensional parametric analysis to map out the performance space and easily identify the remaining tolerances.

Switch to Passband Models and Create an RF System Testbench

The second step in top down RF design is to create a passband view of the system model.

The passband system model performs two functions:

- Confirms that the filters perform as expected.

- Creates an end-to-end testbench that you can use to design the individual function blocks.

For computational efficiency during system passband testing, at any one time, model one or two selected function blocks at the device level. Model all other blocks in the system behaviorally using passband models.

Derive the tolerances by performing the same Monte Carlo analysis or Circuit Optimizer analysis you used to test the function block tolerances in the first step, but this time replace the DSP metrics with end-to-end RF metrics. After you know how far the end-to-end RF metrics can vary, you can insert a device-level model of a function block into the testbench to see how close it drives the system toward violating a derived end-to-end RF specification.

Implement the Function Blocks with Active and Passive Devices

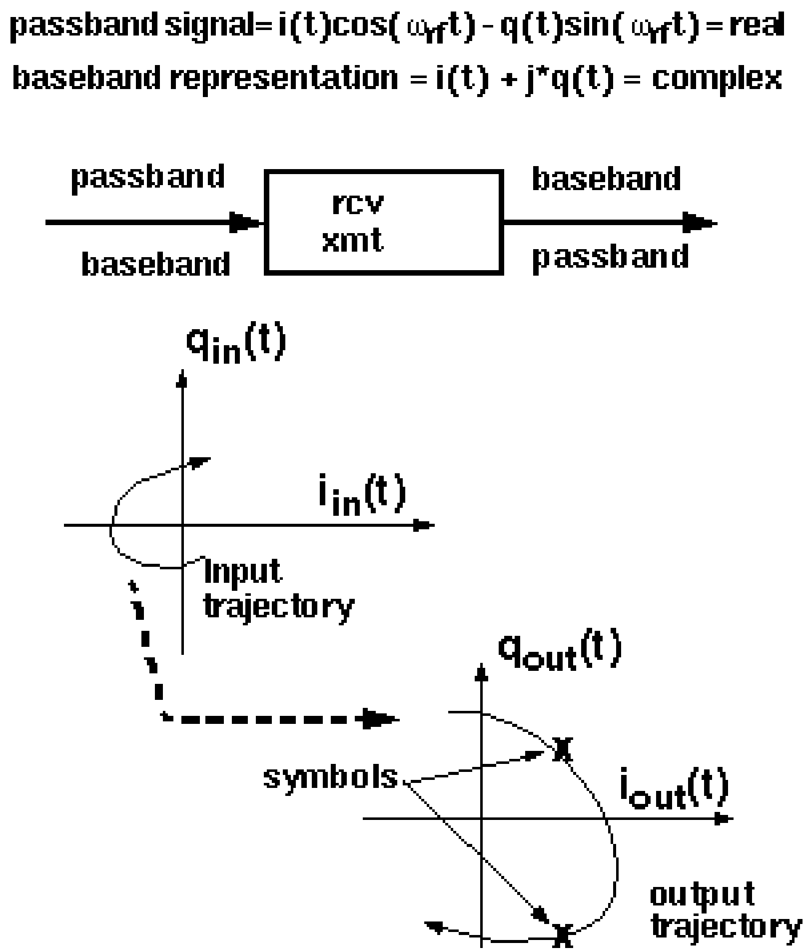
The last step in the top down design process is to implement the function blocks with device models. Because the function blocks are specified in terms of standard RF metrics, you can easily measure the modeling parameters to make sure they fall within the specified tolerances. You can also insert the measured parameters back into the baseband model of the system to check the DSP metrics, or insert the device-level model directly into the passband testbench to check the derived end-to-end RF specifications.

Baseband Modeling

A baseband model for an RF function block simulates what happens to the baseband representation of a signal as it passes through the block. A baseband model maps input baseband signal trajectories into output baseband signal trajectories. If you sample a baseband signal periodically in time and plot the samples in the complex plane, the resulting scatter plot shows the symbol constellation.

Figure B-2 on page 2471 mathematically defines a baseband representation of a passband signal. The *i* and *q* signals are the real and imaginary parts of a complex signal that rides on the two phases of an RF carrier.

Figure B-2 Baseband Representation of a Passband Signal



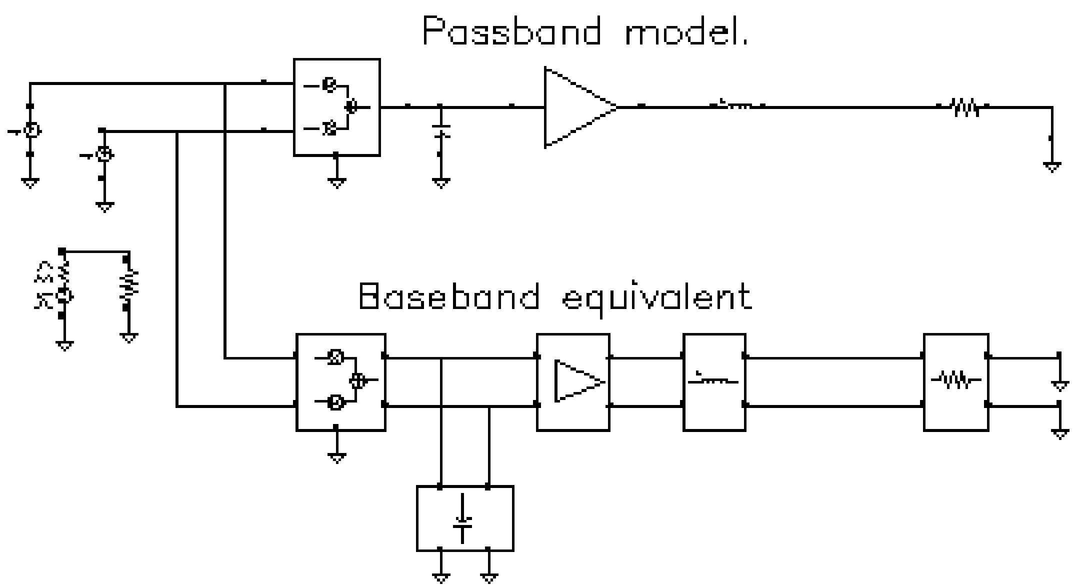
Baseband models simulate only what happens to the carrier fundamental. Consequently, they only account for non-linearities with odd symmetry. Non-linearities with even symmetry produce no output at the carrier fundamental; they affect the carrier fundamental only when cascaded. For example, a second order non-linearity in one block can create a DC offset at it's output. Upon passing through a subsequent block with another second order non-linearity, the DC offset can mix with the carrier to affect the output carrier fundamental. You should model cascaded blocks producing unfiltered even harmonics as a single baseband model rather than as separate baseband models cascaded together. The non-linearities that most often dominate performance have odd symmetry.

Example Comparing Baseband and Passband Models

The example in this section walks you through an Envelope analysis that illustrates the relationship between baseband and passband models. Following the simulation, you plot the baseband equivalent output signals as computed by the baseband and passband circuits.

The *BB_test_bench* schematic shown in Figure B-3 on page 2472 illustrates the difference between passband and baseband modeling. This circuit is located in the *rfExamples* library.

Figure B-3 The BB_test_bench Schematic



The *BB_test_bench* circuit shows a passband circuit (across the top of the schematic) and its baseband equivalent circuit (across the bottom of the schematic). The same baseband signals drive both circuits but only the passband circuit mixes the baseband signals up to RF. The power amplifier is not matched to either input or output impedances and both impedances are reactive.

Before you start, perform the setup procedures described in Chapter 3.

Opening the Baseband Test Bench Circuit

1. In the CIW, choose *File – Open*.

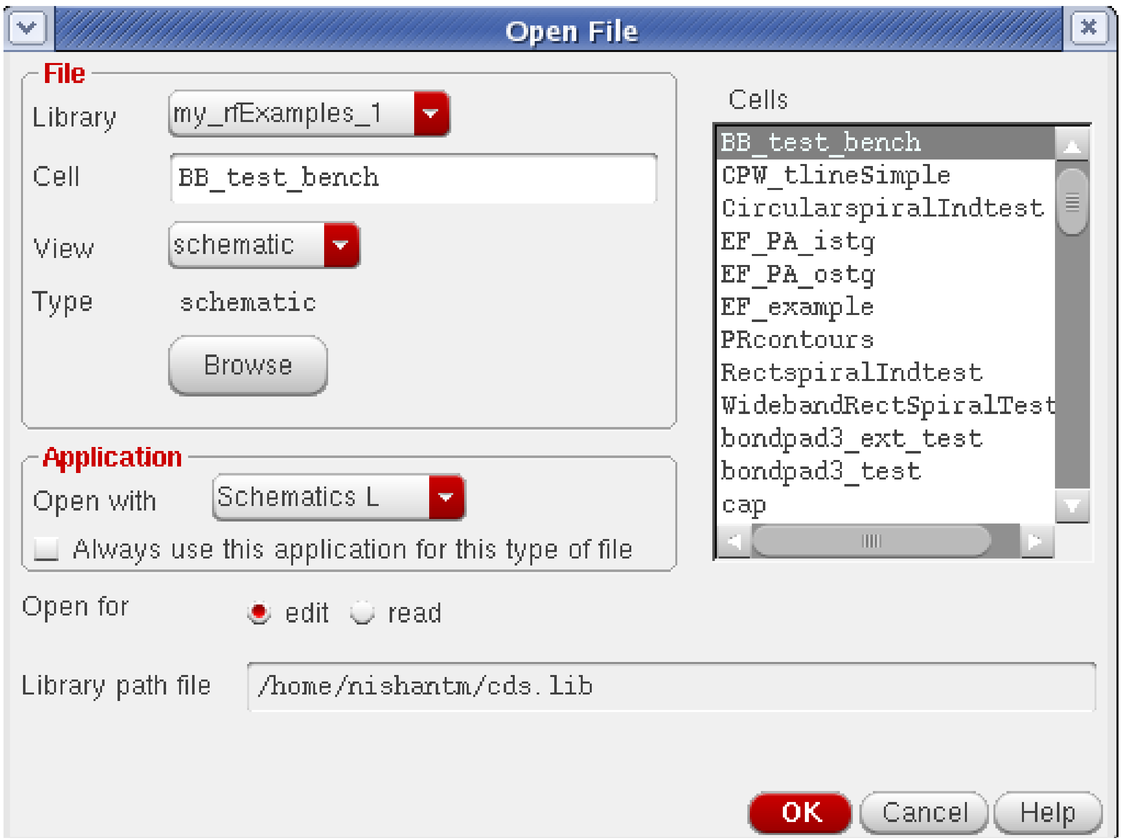
   The Open File form appears.

2. In the Open File form,

a. Choose *rfExamples* in the *Library Name* cyclic field. (Choose the editable copy of *rfExamples* you created as described in Chapter 3.)

b. Choose *BB_test_bench* in the *Cell Names* list box. Note that the *View Name* cyclic field displays *Schematic*.

c. The completed Open File form appears like the one below.

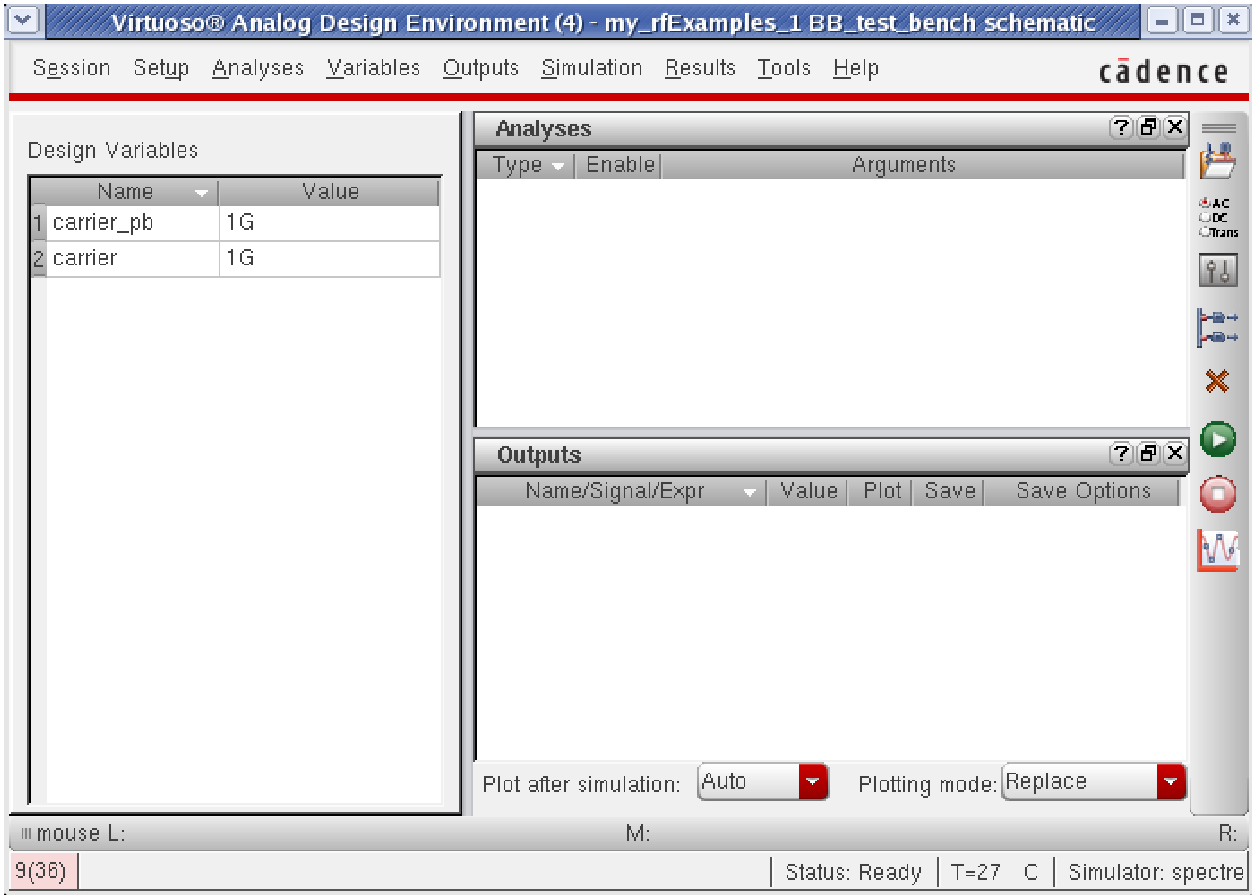


3. Click *OK*.

The Schematic window for the *BB_test_bench* appears.

4. In the Schematic window, choose *Tools – Analog Environment*.

The ADE window opens.



You can also choose *Tools – Analog Environment – Simulation* in the CIW to open the ADE window without opening the design. You can open the design later by choosing *Setup – Design* in the ADE window and choosing the *BB_test_bench* in the Choosing Design form.

Choosing Simulator Options

1. Choose *Setup – Simulator/Directory/Host* in the ADE window.

   The Simulator/Directory/Host form appears.

2. In the Simulator/Directory/Host form, specify the following:

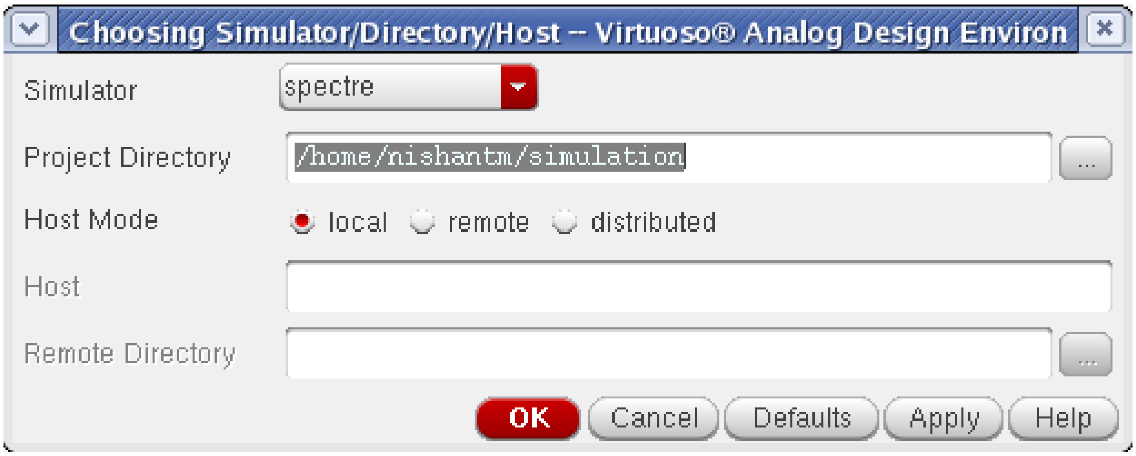
   a. Choose *spectre* for the *Simulator*.

   b. Type the name of the project directory, if necessary.

   c. Highlight the *local*, *remote*, or *distributed* button to specify the *Host Mode*.

For remote simulation, type the name of the host machine and the remote directory in the appropriate fields. For distributed simulation, fill out the other fields that appear.

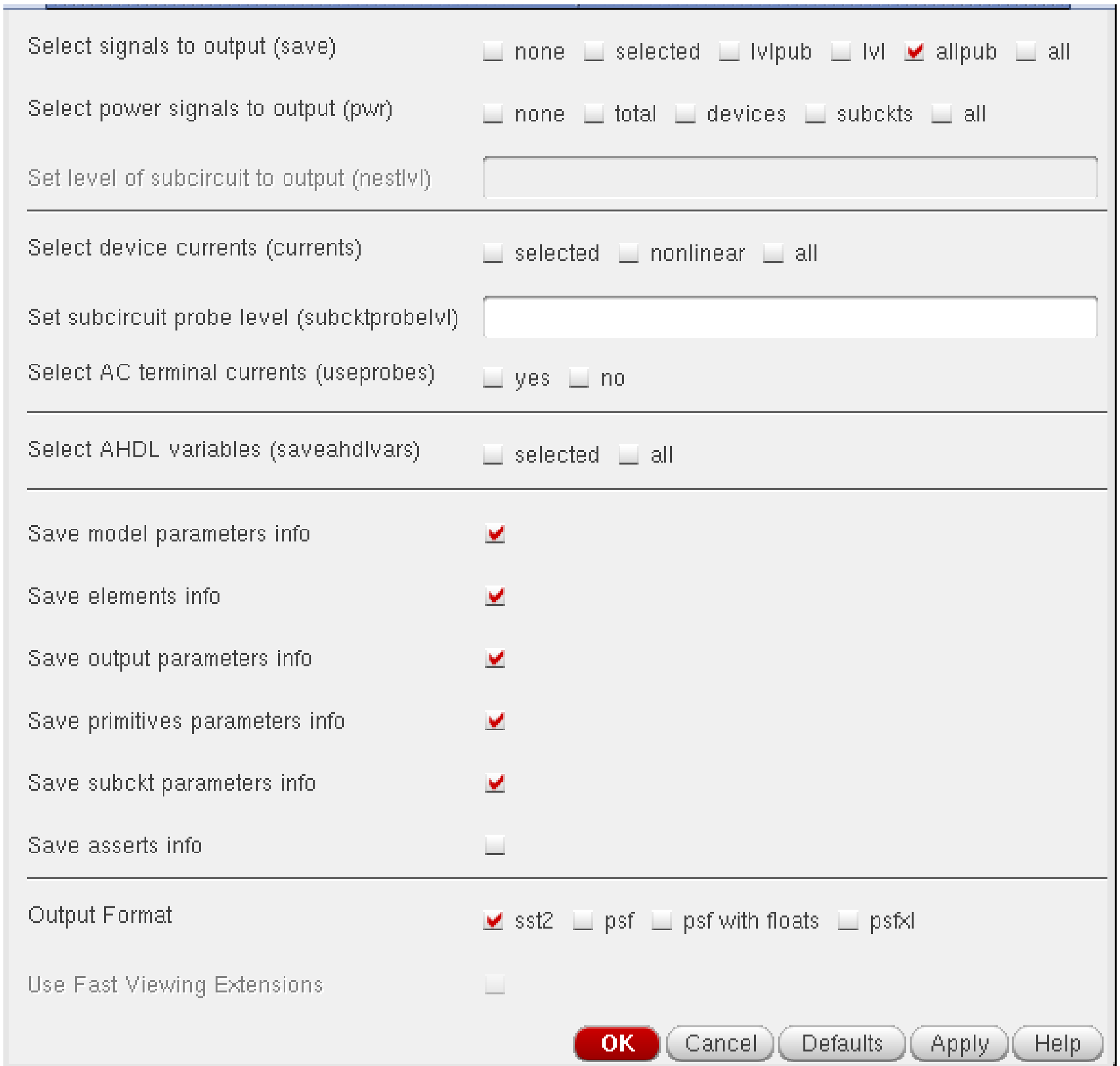
The completed form appears like the one below.



3. In the Simulator/Directory/Host form, click *OK*.
4. In the ADE window, choose *Outputs – Save All.*

The Save Options form appears.

5. In the *Select signals to output (save)* section, be sure *allpub* is highlighted.



6. In the Save Options form, click *OK*.

Setting Up Model Libraries

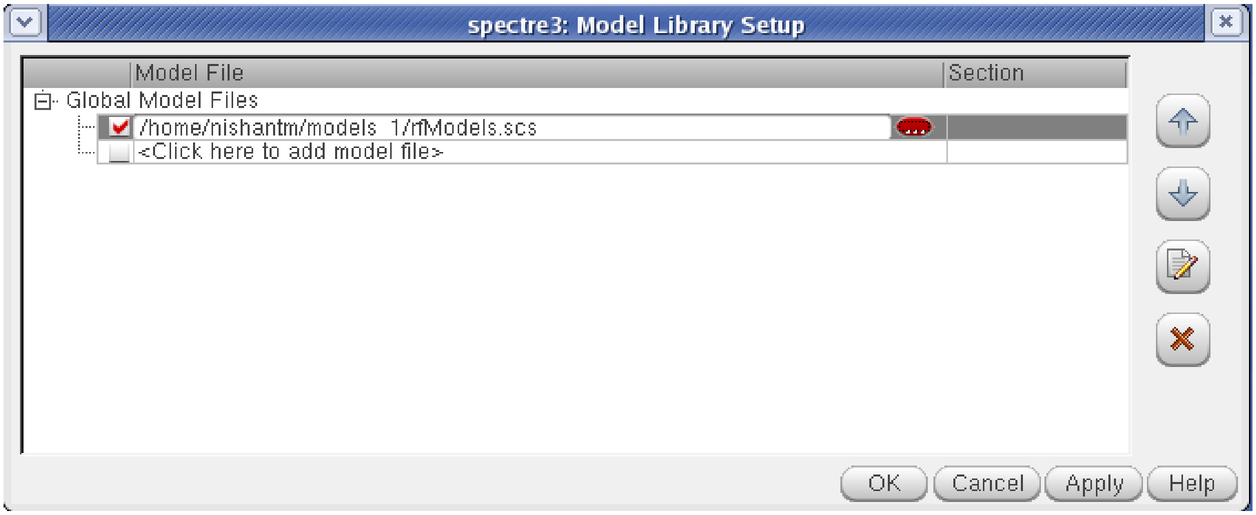
1. In the ADE window, choose *Setup – Model Libraries*.

   The Model Library Setup form appears.

2. In the *Model Library File* field, type the full path to the model file including the file name, `rfModels.scs`.

3. In the Model Library Setup form, click *Add*.

The completed form appears like the one below.



4. In the Model Library Setup form, click *OK*.
5. In the ADE window, use *Analysis - Disable* to disable any analyses you ran previously. (Check the ADE window to verify whether or not an analysis is enabled.)

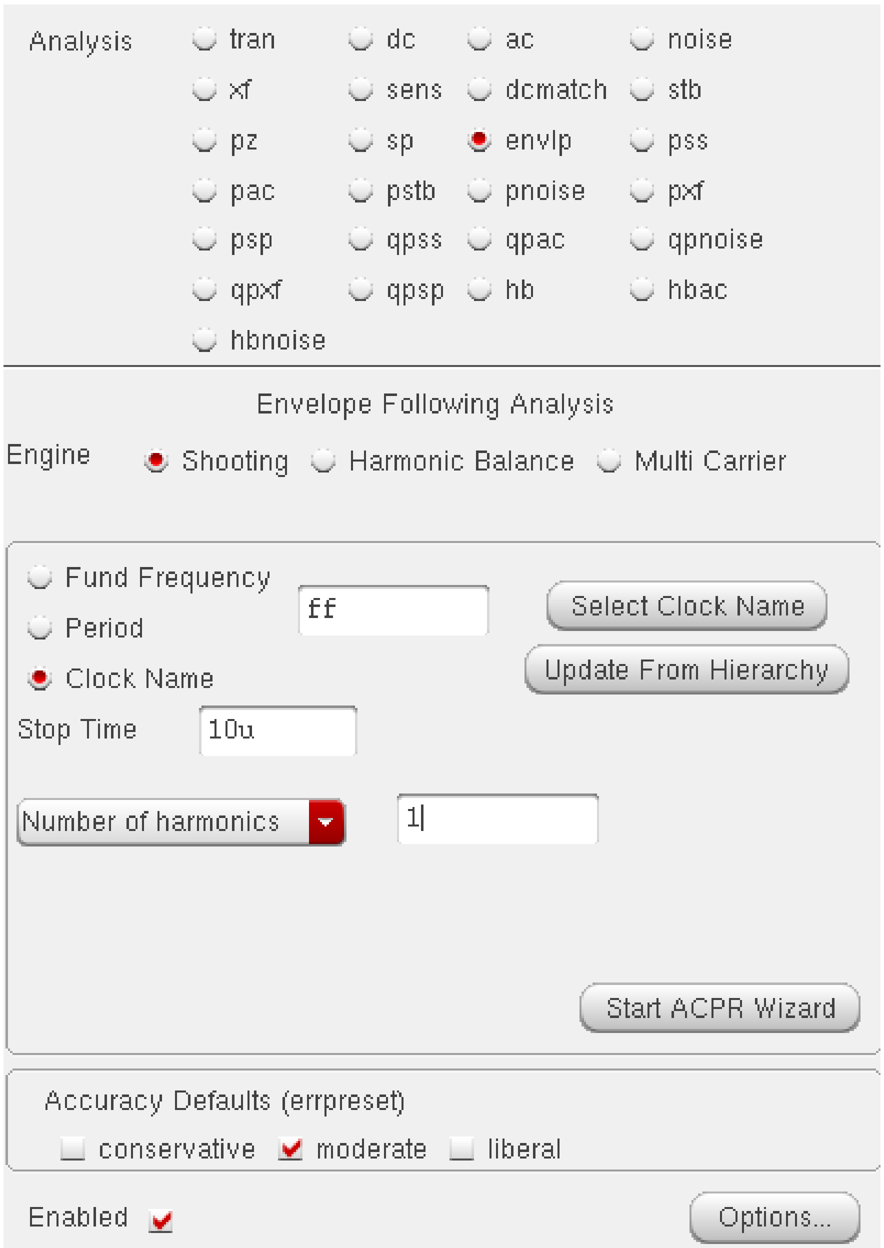
Setting Up the Envelope Analysis

1. Choose *Analyses – Choose* in the ADE window.

   The Choosing Analyses form appears.

2. In the Choosing Analyses form, click *envlp.*
   a. Enter *ff* in the *Clock Name* field.
   b. Enter `10u` in the *Stop Time* field.
   c. In the *Output Harmonics* cyclic field, select *Number of harmonics*.
   d. Enter `1` in the *Number of harmonics* field.
   e. Select *moderate* for the Accuracy Defaults.

The correctly filled out form appears below.



3. In the Choosing Analyses form, click *OK*.

Running the Simulation

1. In the ADE window, choose *Simulation – Netlist and Run*.

The output log file appears and displays information about the simulation as it runs.

Look in the CIW for a message that says the simulation completed successfully.

Plotting the Baseband Equivalent Output Signals

1. In the ADE window, choose *Results-Direct Plot-Main Form*.

   The Direct Plot form appears.

2. In the Direct Plot form, do the following:

   a. Highlight *Replace* for *Plotting Mode*.

   b. Highlight *envlp* for *Analysis*.

   c. Highlight *Voltage* for *Function*.

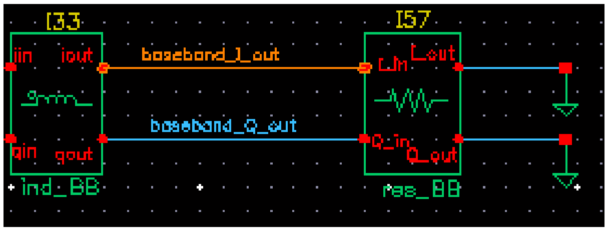
   d. Highlight *time* for *Sweep*.

      The *Net* selection appears in the *Select* cyclic field and the label *Description: Envelope Voltage vs Time* appears.

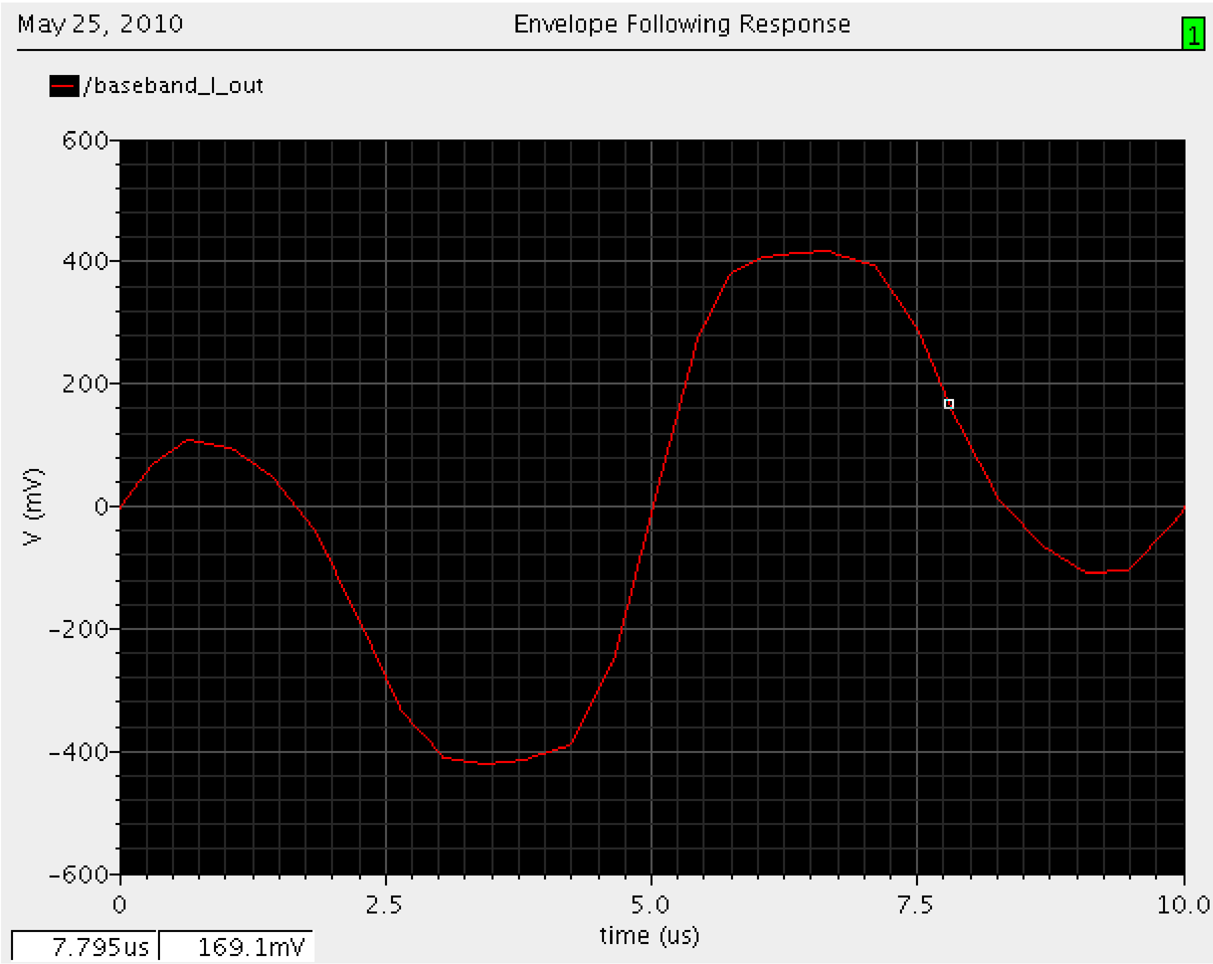
   e. Following the message at the bottom of the Direct Plot form

      Select Net on schematic...

      Click the `baseband_I_out` net.



The first trace appears in the waveform window.

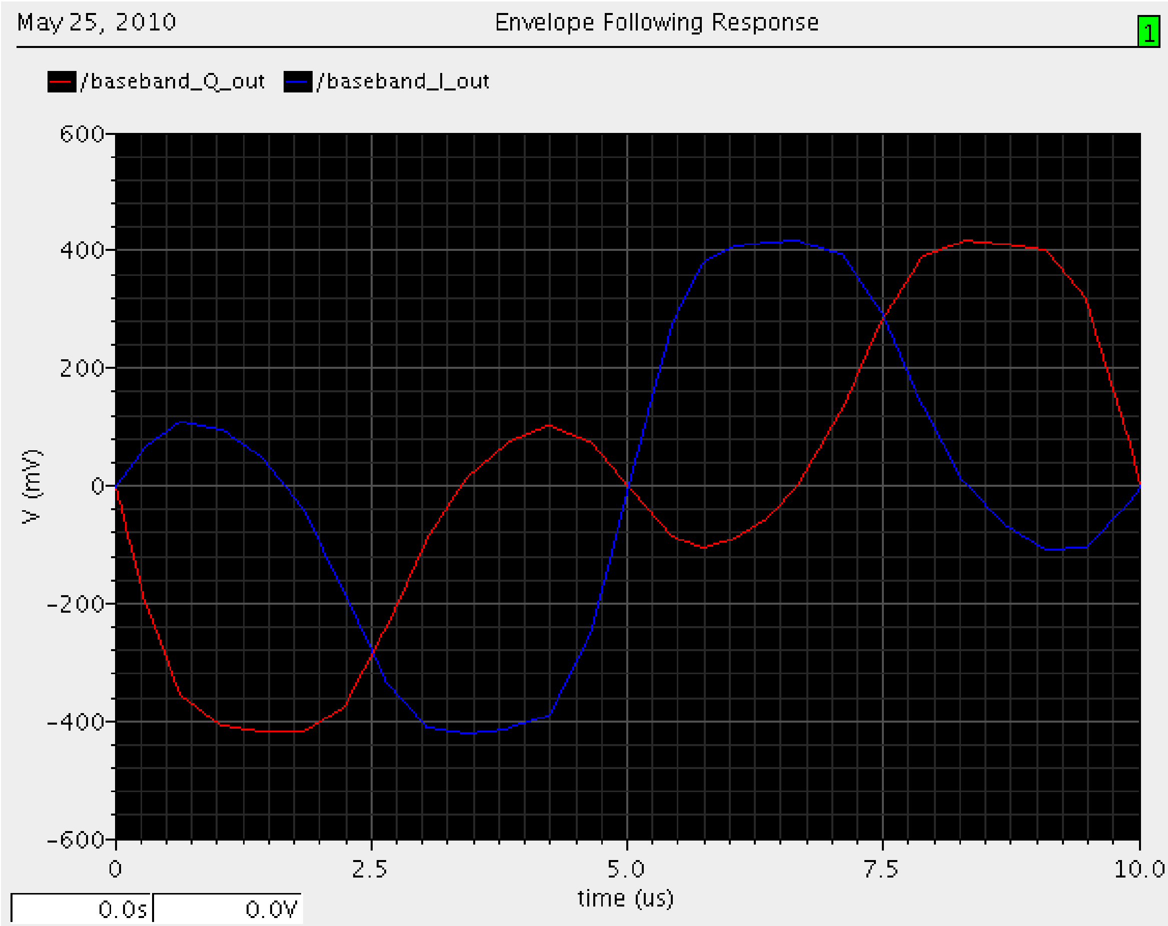


3. In the Direct Plot form, do the following:
   a. Highlight *Append* for *Plotting Mode*.
   b. Leave *Voltage* set for *Function* and *time* set for *Sweep*.
   c. Following the message at the bottom of the Direct Plot form

      Select Net on schematic...

      Click the `baseband_Q_out` net.

The second trace is added to the waveform window. Both baseband equivalent output signals for the baseband model are plotted.



4. In the Direct Plot form, do the following:

a. Leave *Append* for *Plotting Mode* and *Voltage* for *Function.*

b. Highlight *harmonic time* for *Sweep*.

c. Highlight *Real* for *Modifier*.

d. Following the message at the bottom of the form,

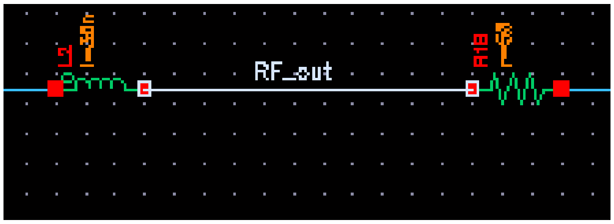
`Select Harmonic Number on this form...`

Select 1 for harmonic number.

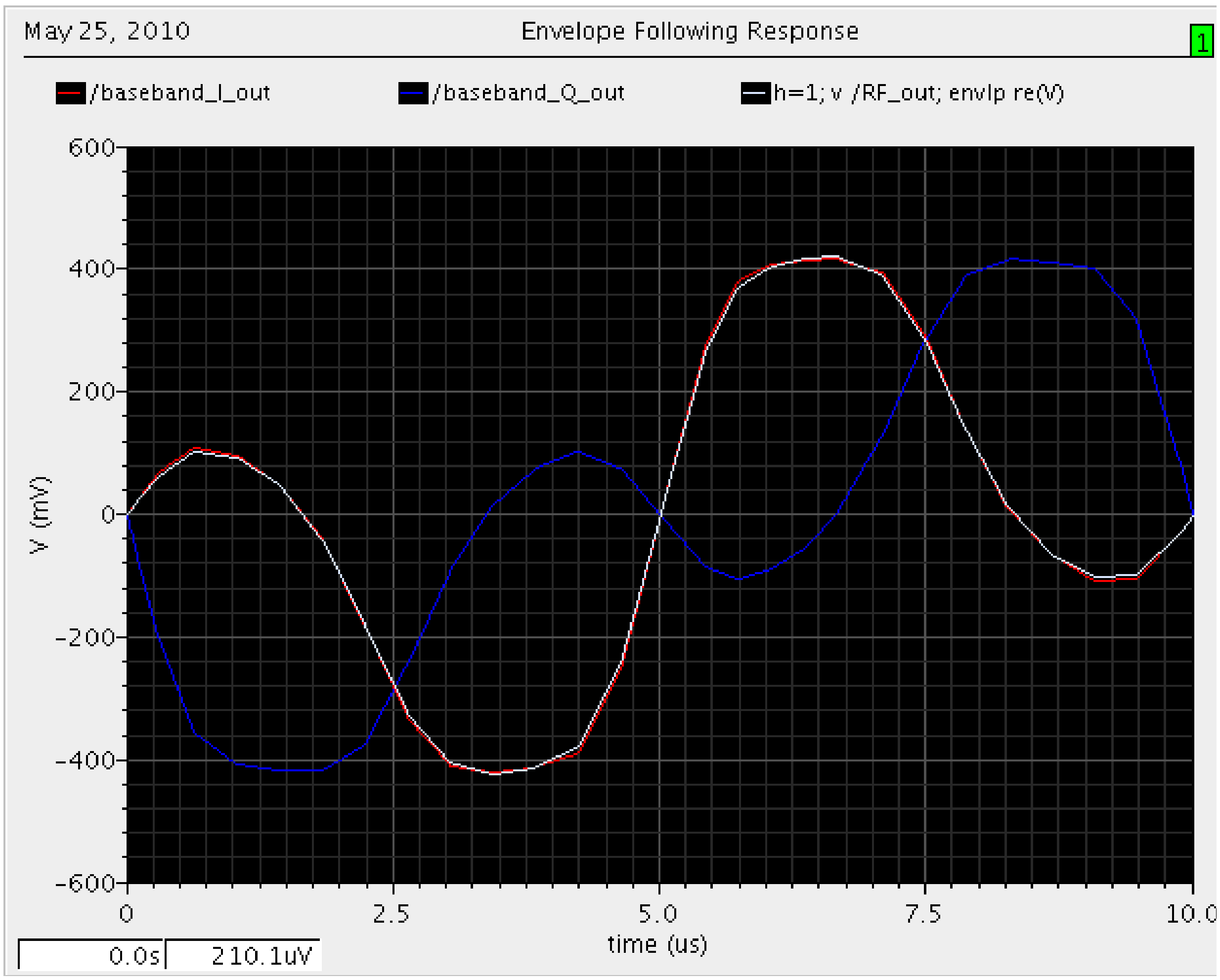
e. Following the next message at the bottom of the form

`Select Net on schematic...`

Click the `RF_out` net.



A third trace is added to the waveform window.



5. In the Direct Plot form, do the following:

a. Leave *Append* for *Plot Mode*, *Voltage* for *Function*, *harmonic time* for *Sweep*, and 1 for *Harmonic Number*.

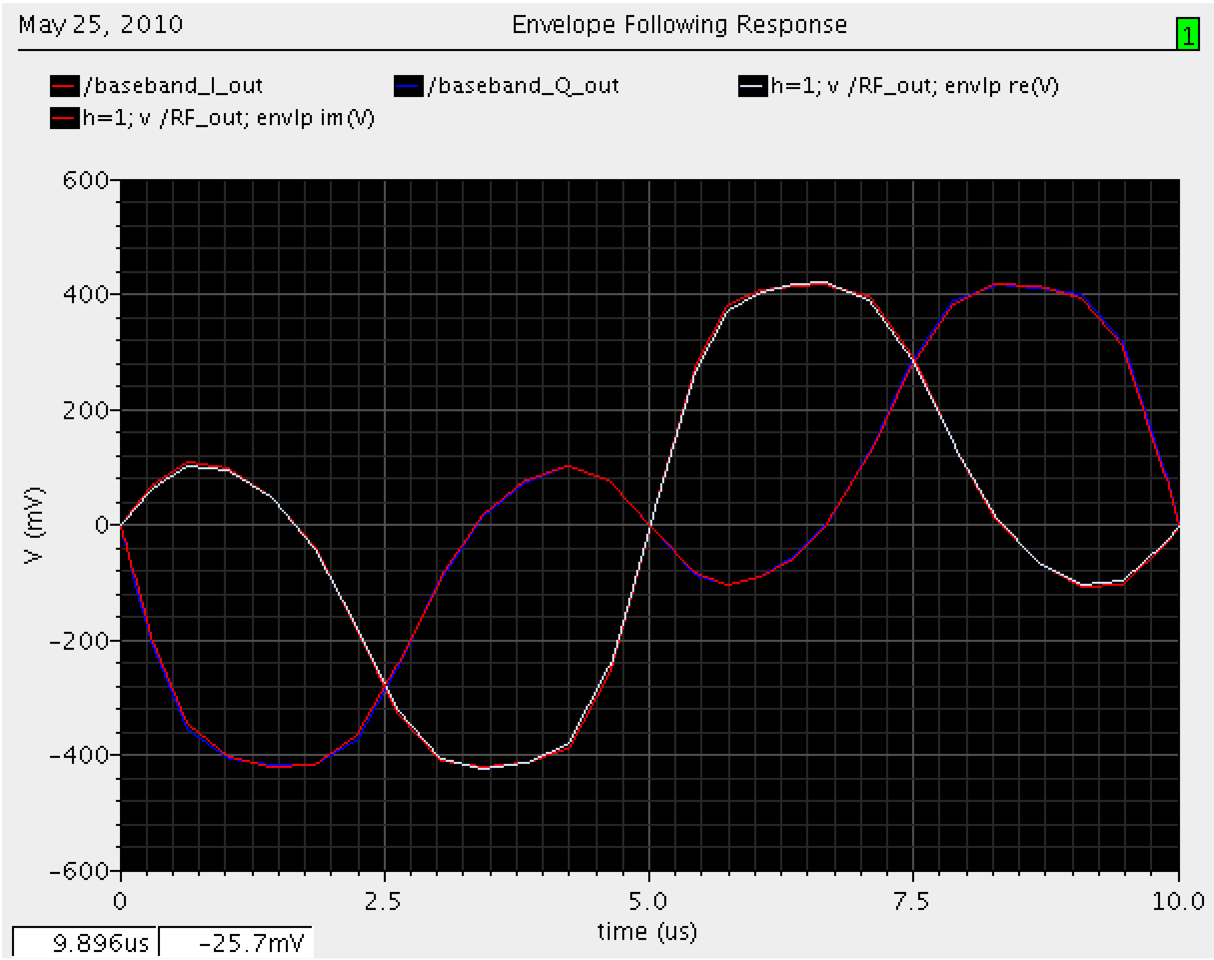
b. Highlight *Imaginary* for *Modifier*.

6. Following the message at the bottom of the form

   `Select Net on schematic...`

   Click the `RF_out` net again.

   A fourth trace is added to the waveform window. Both baseband equivalent output signals for the passband model are added to the plot.



In the waveform display window you should now see what at first appears to be two traces. When you look more closely, you should see that each trace is actually two traces, one nearly on top of the other, making a total of four traces.

The plot resulting from this example illustrates how well baseband modeling corresponds to the time-varying fundamental Fourier component computed by Envelope analysis and raises two questions:

- Why use baseband models when Envelope analysis gives the same results?

- Why not use baseband models all the time?

Running a transient analysis with only the baseband models answers the first question. If from the ADE window you deactivate the passband circuit by setting the *carrier_pb* variable to zero, disable the Envelope analysis, and set up and run a 10µs transient analysis, you observe the same baseband results, but the transient simulation runs over 100 times faster.

Examining the Envelope results answers the second question. If you look closely at both waveforms you notice that the baseband waveforms clip at a slightly lower level than the Envelope waveforms. This is because hard limiting of the carrier generates higher-than-third-order harmonics and the behavioral baseband model only simulates third order non-linearities.

rfLib Library Overview

The *rfLib* include three kinds of models to support baseband modeling:

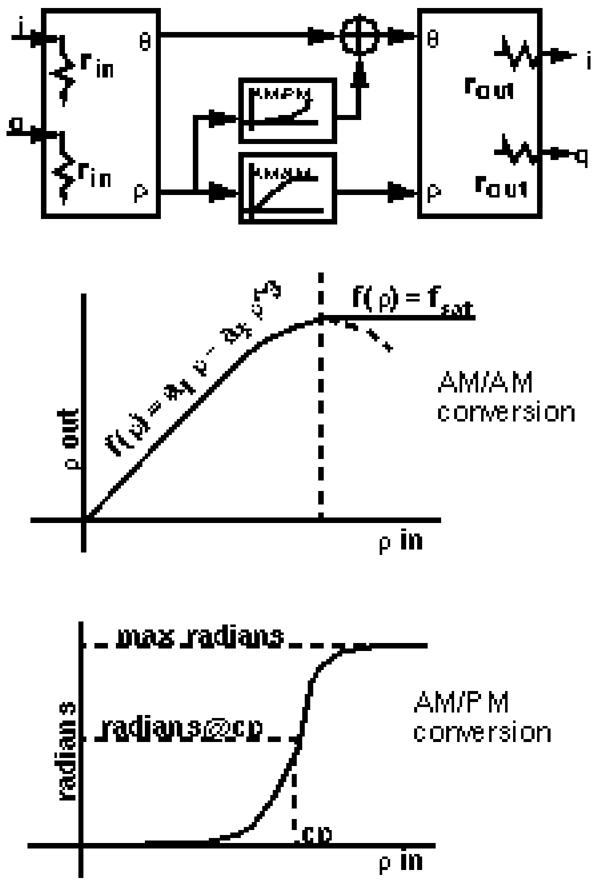
- Instrumentation models
- Non-linear memoryless models
- Linear models with memory

The instrumentation models provide stimuli, diagnostics, and performance metrics relevant to the DSP system.

Both the linear models with memory and the non-linear memoryless models simulate the function blocks in an RF architecture and are specified in terms of common RF metrics. The RF function block models include input referenced white Gaussian noise as specified by noise figure. The *rfLib* includes models for the following RF function blocks—amplifiers, mixers, filters, and phase shifters; where *filters* includes single resistors, capacitors, and inductors.

The non-linear models simulate AM/AM conversion [1] with a third-order polynomial that saturates at the peak of the transfer curve. The polynomial is specified by the gain and either the input-referred IP3 or the output-referred 1 dB compression point. Only the non-linear baseband models simulate AM/PM conversion. AM/PM conversion [1] is an important effect that is hard, if not impossible, to simulate with passband behavioral models. Figure B-4 on page 2485 shows the basic baseband non-linearity.

Figure B-4 Basic, Baseband Non-Linearity



The linear models are the key to simulating loading effects at baseband. In RF integrated circuits, loading effects are important because it is often hard to integrate impedance matching networks. The baseband models of reactive elements differentiate our approach from the spreadsheet-based approaches to RF system design. The baseband capacitor and inductor models (*cap_BB* and *ind_BB* in *top_dwnBB*) let you simulate reactive loading effects in the time domain, where non-linearities are more naturally modeled.

The baseband models of reactive elements also play a key role in modeling filters. Most digital communications text books [1,2] explain that you can model a passband transfer function at baseband by simply frequency-shifting the transfer function. What these books do not describe is how to implement the resulting transfer function in a general circuit simulator such as Virtuoso® Spectre® circuit simulator RF analysis (SpectreRF). The shifted transfer function usually lacks complex conjugate symmetry about zero frequency and therefore has a complex impulse response.

The first consequence of modeling RF function blocks at baseband is that all equivalent baseband models have four terminals instead of two:

- One set of terminals represents the in-phase signals, $i_{in}(t)$ and $i_{out}(t)$
- The other set of terminals represents the quadrature signals, $q_{in}(t)$ and $q_{out}(t)$

  Both sets of terminals are illustrated in Figure B-2 on page 2471.

The mathematics illustrated in Figure B-5 on page 2487 and Figure B-6 on page 2488 summarize the ideas behind a time-varying coordinate transformation that models reactive elements at baseband. The mathematics apply to capacitors as well as inductors.

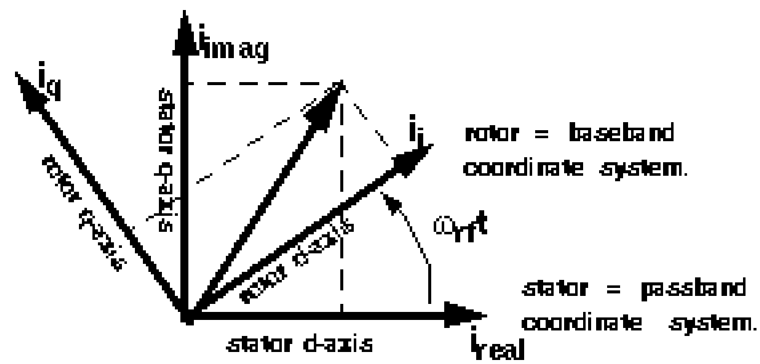
There is a well-documented but little-known electro-mechanical analogy for the derivation of the inductor baseband equivalent model. The four inductor terminals resemble the stator windings of a two-phase rotating machine with shaft speed equal to the RF carrier frequency. Modulation is mathematically analogous to the flux linking a stator winding due to currents in orthogonal rotor windings. The flux depends on the shaft angle just as a modulated signal depends on the carrier phase. Transforming the vectorial equation for $v=Ldi/dt$ to the rotor reference frame suppresses the RF carrier and introduces a *speed voltage* [3,4,5,6,7,8,9], or back electro-motive force (back EMF), that couples the differential equations.

An expression for the real current (i.e. the passband current) appears in Figure B-6 on page 2488. The real current is modeled as the projection of a two-dimensional rotating vector onto a stationary axis, the *real* axis. The vector rotates with an angular velocity equal to the RF carrier frequency.

Figure B-5 Passband Current for an Inductor

$$i_{real} = i_i(t)\cos(\omega_{rf}t) - i_q(t)\sin(\omega_{rf}t) = Real[(i_i+j^*i_q)^*e^{j\omega_{rf}t}]$$

$$i_{imag} = i_i(t)\sin(\omega_{rf}t) + i_q(t)\cos(\omega_{rf}t) = Imag[(i_i+j^*i_q)^*e^{j\omega_{rf}t}]$$



$$i=(i_i+j^*i_q)^*e^{j\omega_{rf}t}$$

$$v=(v_i+j^*v_q)^*e^{j\omega_{rf}t}$$

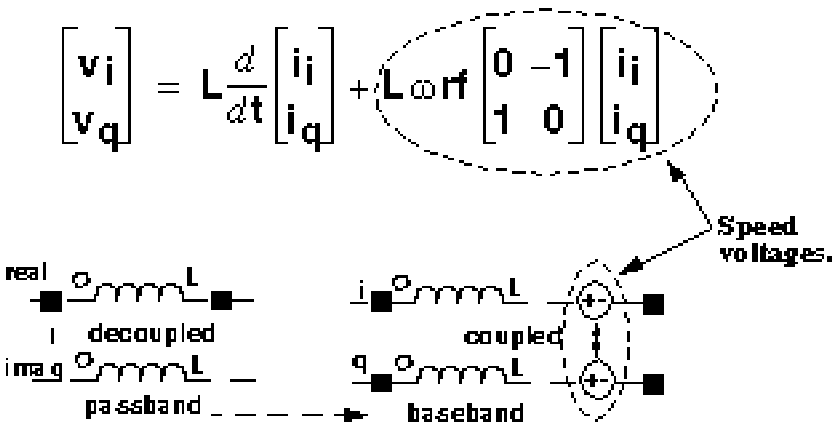
$$v=L\,di/dt$$

$$(v_i+j^*v_q)\cancel{e^{j\omega_{rf}t}}=L^*[d(i_i+j^*i_q)/dt + j^*\omega_{rf}{}^*(i_i+j^*i_q)]\cancel{e^{j\omega_{rf}t}}$$

The rotating vector also has a projection onto another stationary axis orthogonal to the real axis. In the baseband literature, the orthogonal projection is the Hilbert transform of the real signal. The constitutive relationship of the inductor, *v=Ldi/dt*, is expressed in terms of coordinates in a reference frame that rotates with the vector.

Figure B-6 on page 2488 shows the constitutive inductor relationship between voltage and current in the rotating reference frame. Note that the trigonometric terms, the terms that slow simulation speed, are gone and the two projections are now coupled through *speed voltages*. The term speed voltage comes from the fact that the voltages depend on the angular speed of the rotating reference frame. In motor theory, that speed is the shaft speed. Speed voltage is similar to the back EMF in a motor. Because of speed voltages, baseband models of filters and reactive elements must have their carrier frequency specified. The carrier frequency is the frequency for which the baseband signals are referenced. For example, the carrier frequency for an RF filter would be the RF frequency while the carrier frequency for an IF filter would be the IF frequency.

Figure B-6 Relationship Between Voltage and Current for an Inductor



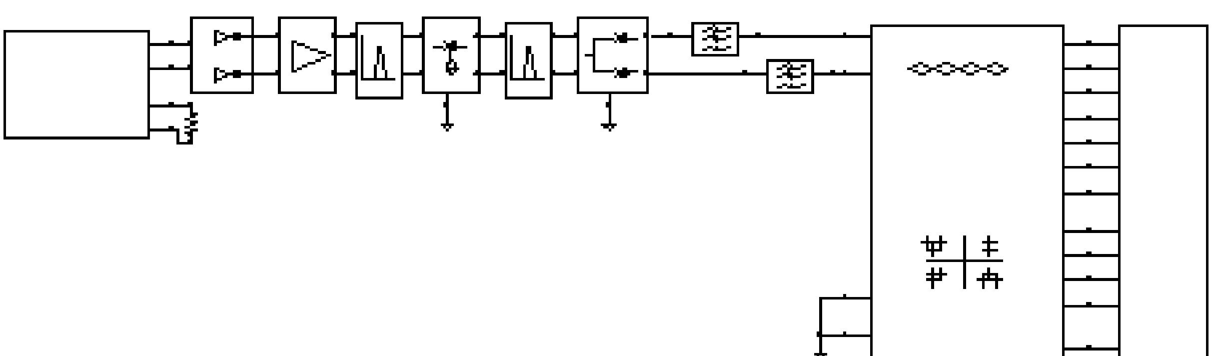
The baseband counterparts of the passband filter models are built up from inductors and capacitors modeled in the rotating reference frame.

In the complex expression for *v=Ldi/dt,* if you replace *d/dt* with $j\omega$, you find that the impedance of the inductor changes from $jL\omega$ to $jL(\omega+\omega_{rf})$. The same holds for capacitors, which means a filter transfer function, $H(\omega)$, has a baseband equivalent equal to $H(\omega+\omega_{rf})$. This is simply the original passband transfer function shifted to the left by an amount equal to the carrier frequency. Our time domain baseband models are consistent with the text book frequency domain explanation of baseband modeling.

Use Model and Design Example

This section describes how to use the baseband models during the architectural design phase. The following examples show you how to

- Construct a baseband model for a simple receiver



- Use the Circuit Optimizer to balance specifications among the function blocks
- Create a passband testbench for the receiver

The design goals were chosen arbitrarily. The example is meant simply to illustrate how to use the library and is a derivative of the design found in [10]. If you find that some parameters are not specified, leave them as default values. You construct the receiver from left to right, from input to output.

Opening a New Schematic Window

1. In the CIW, choose *File – New – Cellview*.

   The Create New File form appears.

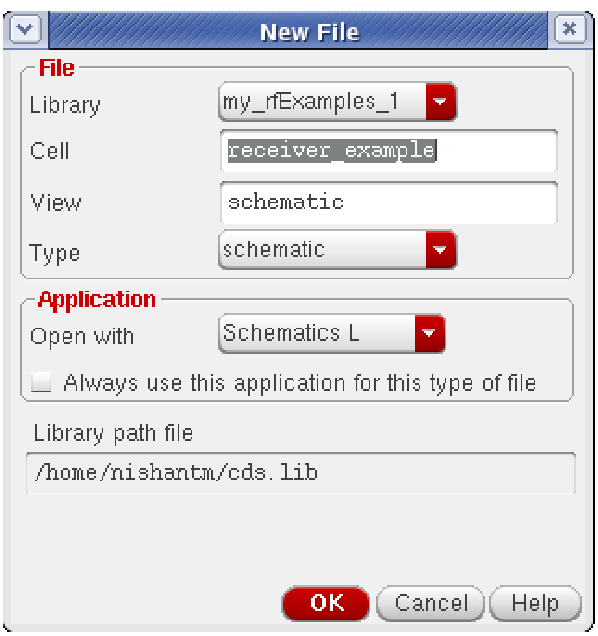
2. In the Create New File form,

   a. Choose *my_rfExamples* in the *Library Name* cyclic field. (Choose the editable copy of *rfExamples* you created as described in Chapter 3.)

   b. Enter `receiver_example` in the *Cell Name* field.

   c. Select *Composer-Schematic* in the *Tool* cyclic field. `schematic` appears in the *View Name* field.

   d. The completed form appears like the one below.



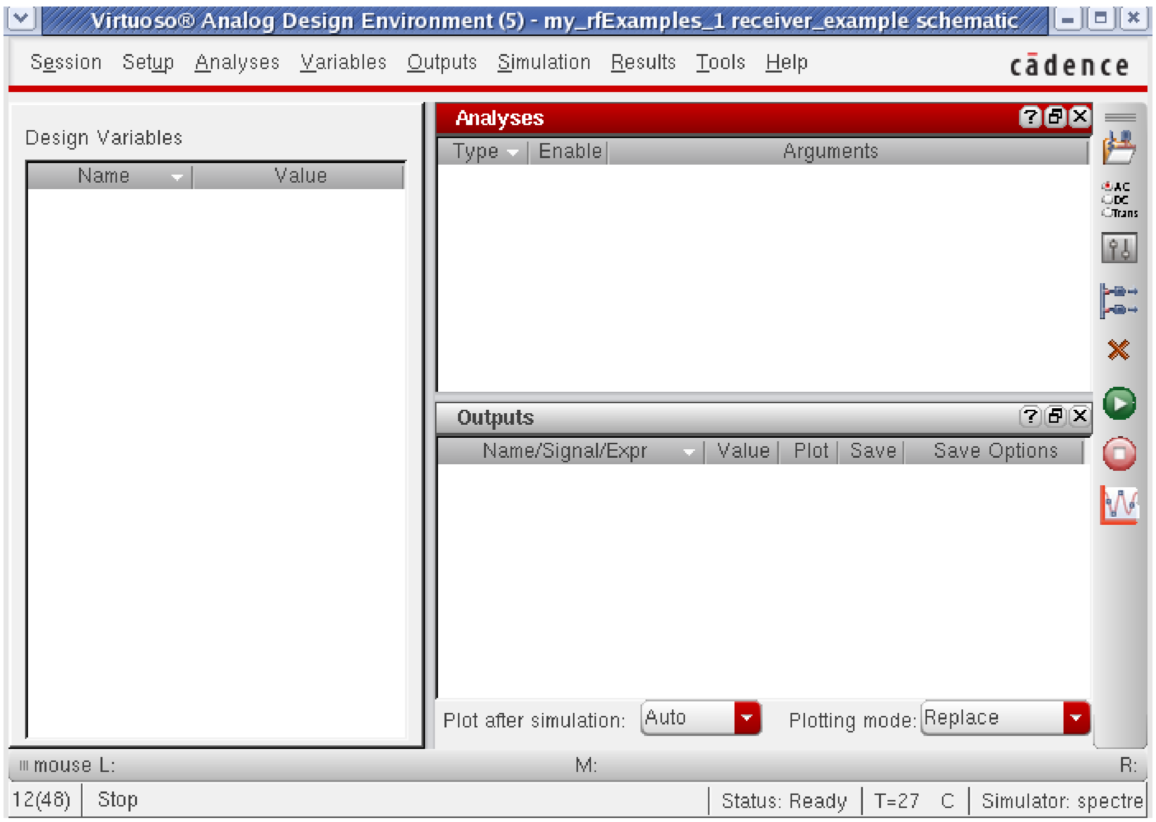
3. Click *OK*.

An empty Schematic window for the *receiver_example* appears.

Opening the Analog Environment

1. In the Schematic window, choose *Tools – Analog Environment*.

   The ADE window opens.



   The *Library*, *Cell*, and *View* names appear in the *Design* section of the ADE window.

2. Set the simulator options from the Simulator window as described in "Choosing Simulator Options" on page 2474.

3. Set up the model libraries from the Simulator window as described in "Setting Up Model Libraries" on page 2476.

Constructing the Baseband Model for the Receiver

Construct the receiver in the Schematic window by adding blocks from left to right, from input to output, as listed in Table B-1.

Except for the resistor, ground, and port models (which come from the *analogLib*), all blocks come from the *rfLib*. Unless otherwise instructed, leave the port resistances at their default value of 50 Ohms.

Table B-1 Blocks Used to Create the Receiver

Block Name and Reference	Element Name	Library and Category
CDMA signal source — See Adding the CDMA Signal Source	`CDMA_reverse_xmit`	From the *measurement* category in *rfLib*.
Resistor—attach to CDMA signal source	`res`	From *analogLib*
Driver — See Adding the Driver	`BB_driver`	From the *measurement* category in *rfLib*.
Low noise amplifier — See Adding the Low Noise Amplifier	`LNA_BB`	From the *top_dwnBB* category in *rfLib*.
Butterworth bandpass filter — See Adding a Butterworth Band Pass Filter	`BB_butterworth_bp`	From the *top_dwnBB* category in *rfLib*
RF-to-IF mixer — See Adding an RF-to-IF Mixer	`dwn_cnvrt`	From the *top_dwnBB* category in *rfLib*
Butterworth bandpass filter — See Adding Another Butterworth Bandpass Filter	`BB_butterworth_bp`	From the *top_dwnBB* category in *rfLib*
IQ demodulator — See Adding an IQ Demodulator	`IQ_demod_BB`	From the *top_dwnBB* category in *rfLib*
Butterworth lowpass filters (create two) — See Adding Two Butterworth Lowpass Filters	`butterworth_lp`	From the *top_dwnPB* category in *rfLib*
Instrumentation model — See Adding an Instrumentation Block	`offset_comms_instr`	From the *measurement* category in *rfLib*.
Terminator — See Adding an Instrumentation Terminator	`instr_term`	From the *measurement* category in *rfLib*
Grounds—attach to RF-to-IF mixer, IQ demodulator, and Instrumentation model	`gnd`	From *analogLib*

Adding the CDMA Signal Source

Add the first receiver block, a CDMA signal source (`CDMA_reverse_xmit`), to the schematic.

1. In the Schematic window, choose *Add – Instance*.

   The Add Instance form appears. It may be empty or it may display information for a previously added element. The default for the *View* field is `symbol`.

2. In the Add Instance form, click *Browse*.

   The Library Browser - Add Instance form appears.

3. In the Library Browser - Add Instance form,

   a. If necessary, click *Show Categories* to display the *Category* column so you can view the elements (or cells) in the *rfLib* by category.

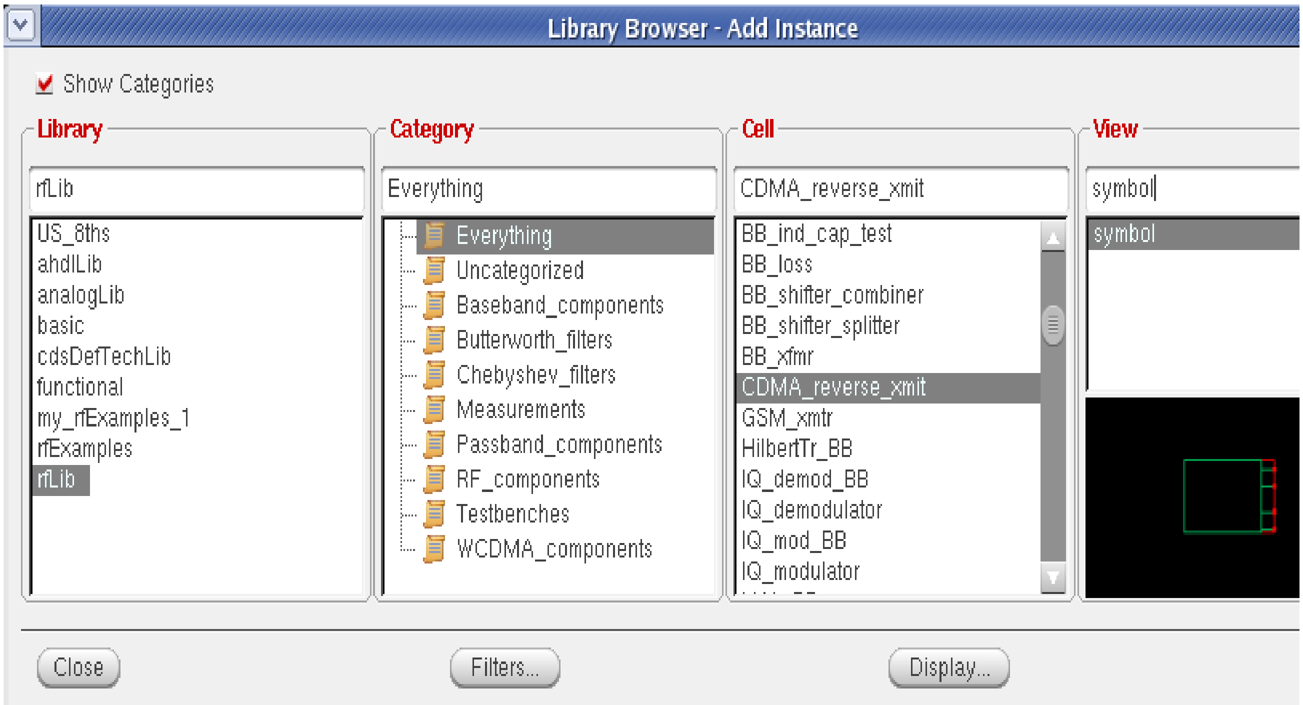
   b. In the *Libraries* column, click *rfLib* to display categories of elements in *rfLib*.

      The *Everything* category is displayed by default and all cells in *rfLib* are listed in the *Cells* column. (In the Add Instance form, *rfLib* displays in the *Library* field.)

   c. In the *Category* column, click *measurement* to list only the cells in the *measurement* category.

   d. In the *Cell* column, click *CDMA_reverse_xmit*.

In the Library Browser, cell *CDMA_reverse_xmit* and it's default view *symbol* are both selected.



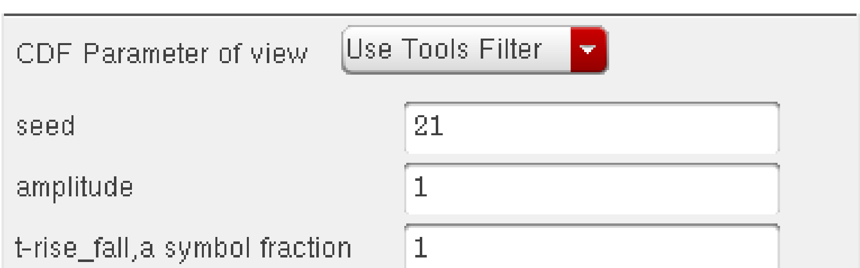
In the Add Instance form,

❑ `rfLib` appears in the *Library* field

❑ `CDMA_reverse_xmit` displays in the *Cell* field

❑ `symbol` displays in the *View* field

The CDF parameters for the element and their default values appear at the bottom of the form.



4. To place a *CDMA_reverse_xmit* block in the schematic,

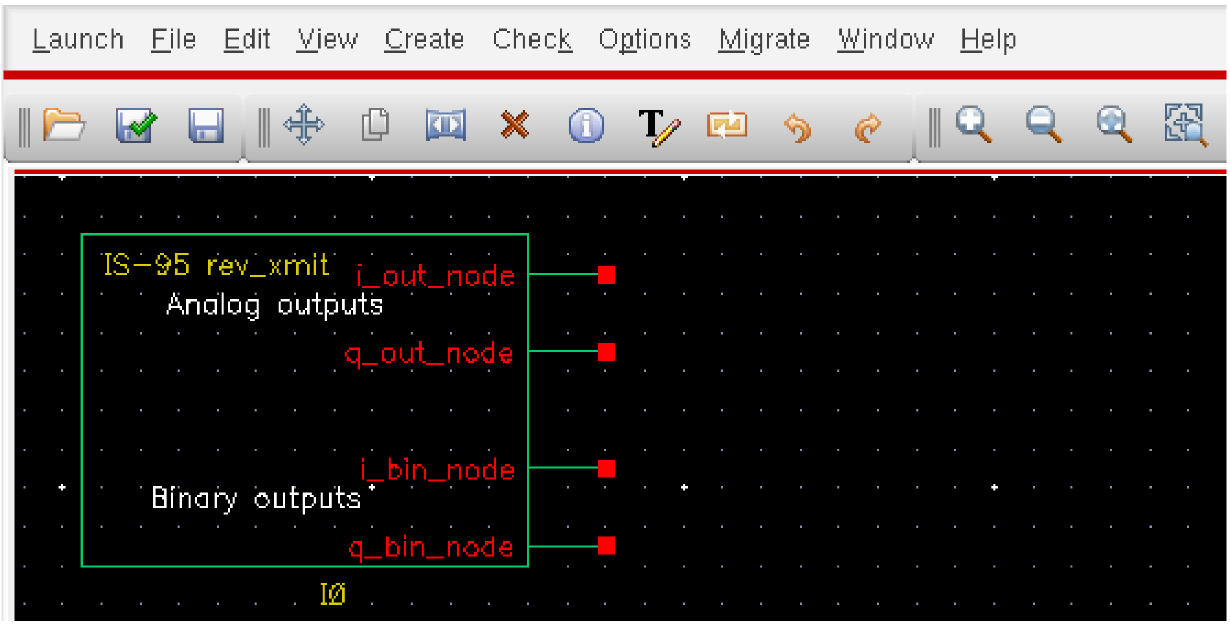
a. Move the cursor over the Schematic window.

The outline for the *CDMA_reverse_xmit* symbol is attached to the cursor.

b. Move the cursor near the top left corner of the schematic and click to place the *CDMA_reverse_xmit* block.

This block models a CDMA signal source.

c. Click *Esc* to remove the symbol from the cursor.



Adding a Resistor to the CDMA Signal Source

Because this example does not use the binary output nodes (`i_bin_node` and `q_bin_node`) on the CDMA signal source, connect a resistor between these nodes to avoid unused pin warnings.

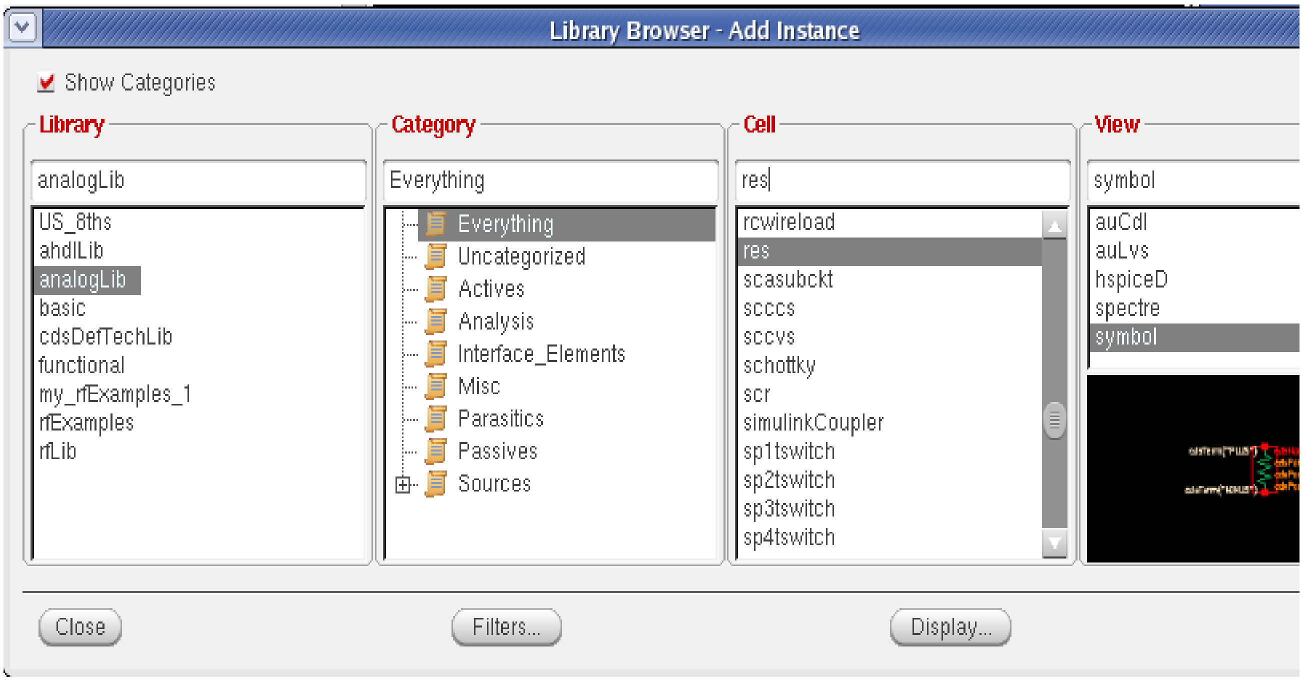
1. In the *Libraries* column of the Library Browser - Add Instance form, click *analogLib* to display elements in *analogLib*.

   If *Show Categories* is selected, the *Everything* category is displayed by default and all cells in *analogLib* are listed in the *Cells* column.

2. Scroll through the list of cells in *analogLib* to locate the resistor cell, *res*.

3. Click *res* in the *Cell* column.

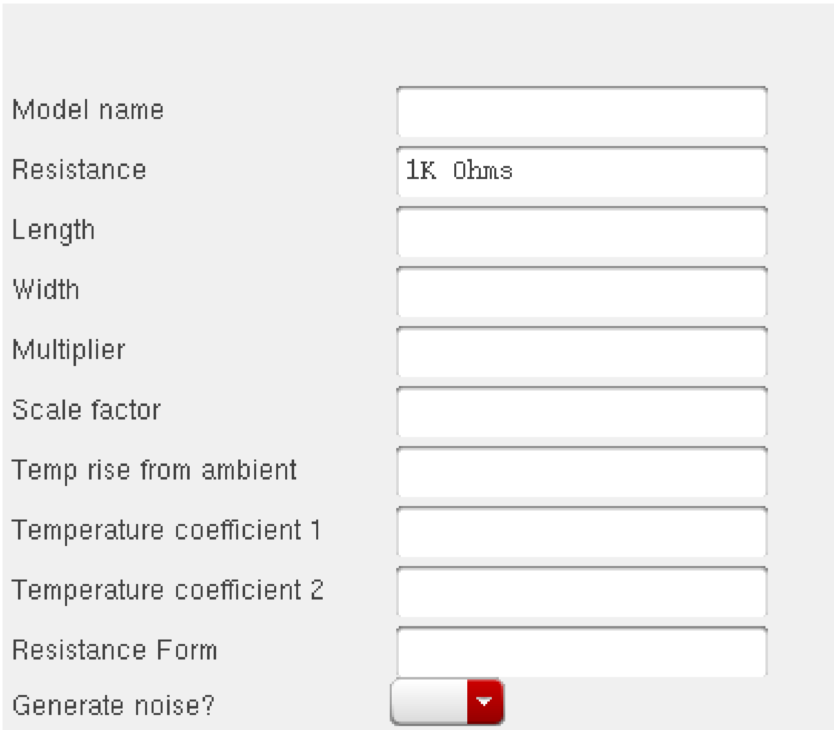
The cell *res* and it’s default view *symbol* are both selected.



In the Add Instance form,

- `analogLib` displays in the *Library* field
- `res` displays in the *Cell* field
- and `symbol` displays in the *View* field

The CDF parameters for the element and their default values appear at the bottom of the form.



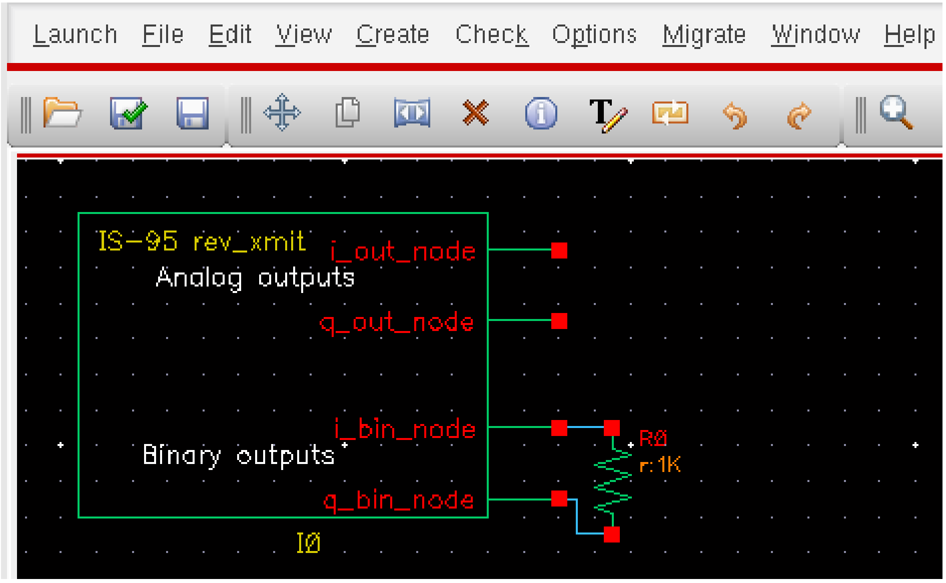
4. Move the cursor over the Schematic window.
5. Click to place the top resistor terminal in line with the top binary output node (`i_bin_out`) on the lower right side of the *CDMA_reverse_xmit* block
6. Click *Esc* to remove the symbol from the cursor.

Wiring the Resistor to the CDMA Signal Source

Wire the resistor to the binary outputs, `i_bin_node` and `q_bin_node`, of the CDMA signal source.

1. To wire the resistor to the *CDMA_reverse_xmit* block, in the Schematic window choose *Add - Wire (narrow)*.
2. Click `i_bin_node` on the `CDMA_reverse_xmit` block then move the cursor and click the top node of the resistor.
3. Click `q_bin_node` on the `CDMA_reverse_xmit` block then move the cursor and click the bottom node of the resistor.
4. Click *Esc* to stop wiring.

The CDMA signal source and resistor wired together appear as follows.



Adding the Driver

Add a driver block to the right of the CDMA signal source block.

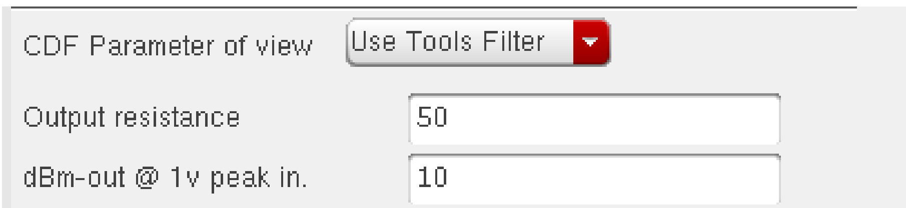
1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.
2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.
3. In the Library Browser - Add Instance form, make the following selections.

Library	Category	Cell	View
rfLib	*measurement*	*BB_driver*	*symbol*

At the top of the Add Instance form,

- `rfLib` displays in the *Library* field,
- `BB_driver` displays in the *Cell* field and
- `symbol` displays in the *View* field.

- ❑ The CDF parameters and their default values appear at the bottom of the Add Instance form.

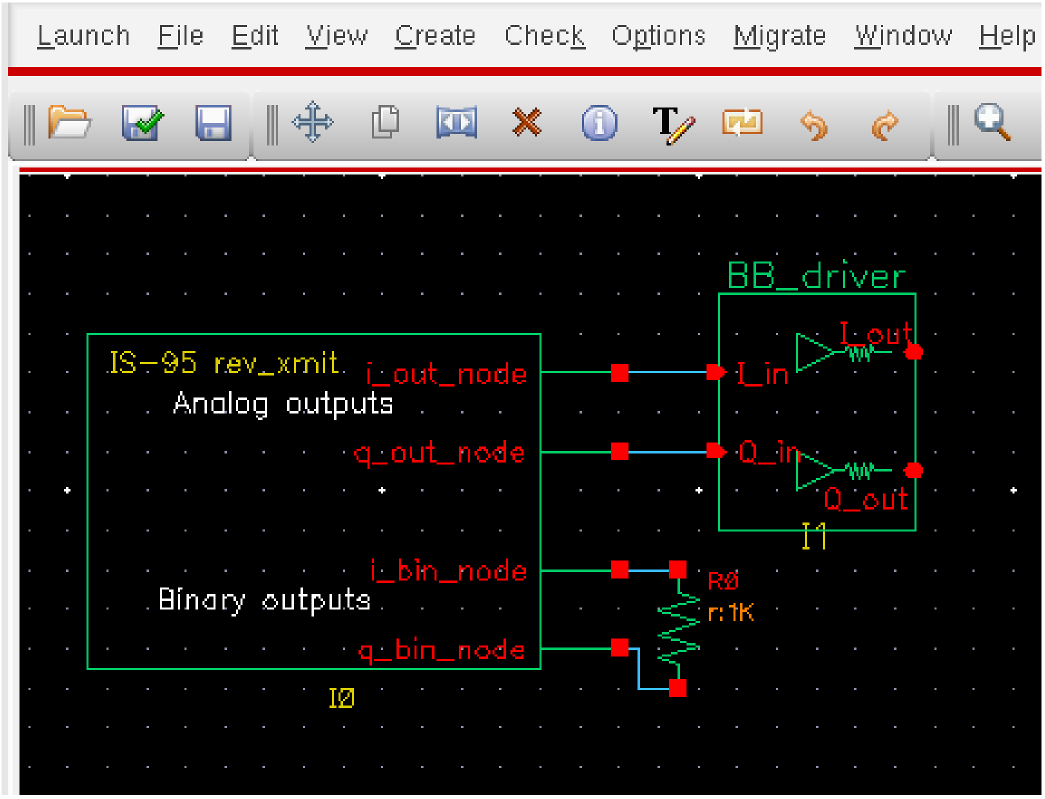


4. Move the cursor over the Schematic window.
5. Click to place the *BB_driver* to the right of the *CDMA_reverse_xmit* block. Align the input pins of the driver with the analog output pins of the CDMA signal source.
6. Click *Esc* to remove the symbol from the cursor.

Wiring the Signal Source to the Driver

1. To wire the *BB_driver* block to the *CDMA_reverse_xmit* block, in the Schematic window choose *Add – Wire (narrow)*.
2. Click `i_out_node` on the *CDMA_reverse_xmit* block then click `I_in` on the *BB_driver* block.
3. Click `q_out_node` on the *CDMA_reverse_xmit* block then click `Q_in` on the *BB_driver block*.
4. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the Driver

Edit the value of the *BB_driver* CDF parameter *dBm-out@1v peak in driver* as follows.

1. Choose *Edit – Properties – Objects* in the Schematic window.

   The Edit Object Properties form appears. You use this form to change the values of CDF (component description format) properties for the driver and modify the schematic for this simulation.

2. In the Schematic window, click the *BB_driver* block.
3. The Edit Object Properties form changes to display information for the *BB_driver* block
4. Change the *dBm-out@1v peak in* parameter value as follows.

Parameter Name	Value
dBm-out@1v peak in	*-16*

The driver converts 1 peak volt from the CDMA signal source to –16 dBm referenced to the output resistance of the driver.

5. Click *OK* in the Edit Object Properties form.

   The form closes.

Adding the Low Noise Amplifier

Add a low noise amplifier to the right of the driver.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

3. In the Library Browser - Add Instance form, make the following selections. (If necessary, click *Show Categories* at the top of the Library Browser, to display the *Category* column.)

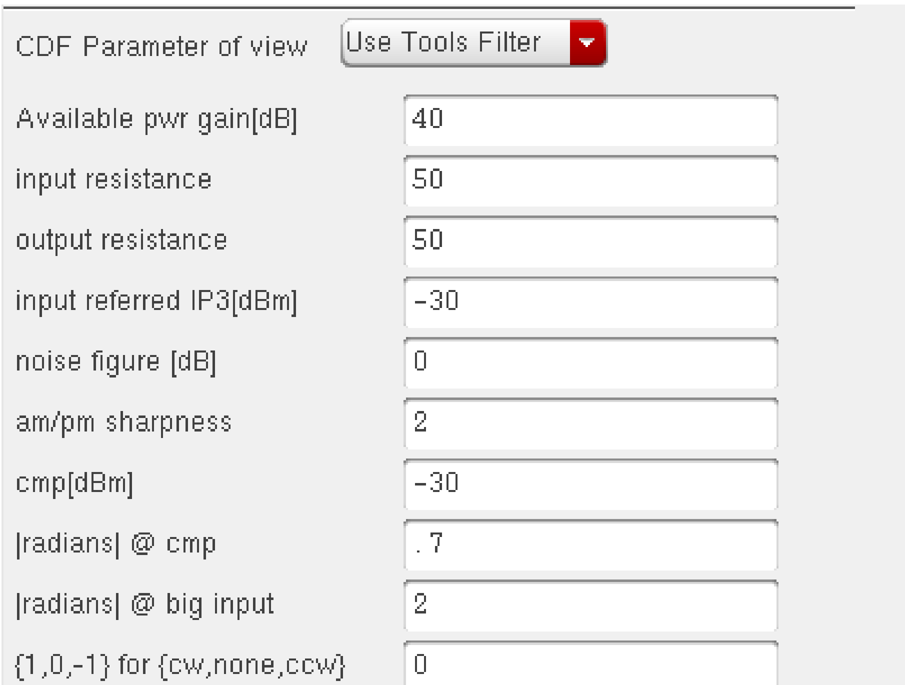
Library	Category	Cell	View
rfLib	top_dwnBB	LNA_BB	symbol

In the Library Browser, cell *LNA_BB* and it's default view *symbol* are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library* field
- ❑ `LNA_BB` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field

The CDF parameters for the element and their default values display at the bottom of the LNA_BB Add Instance form.

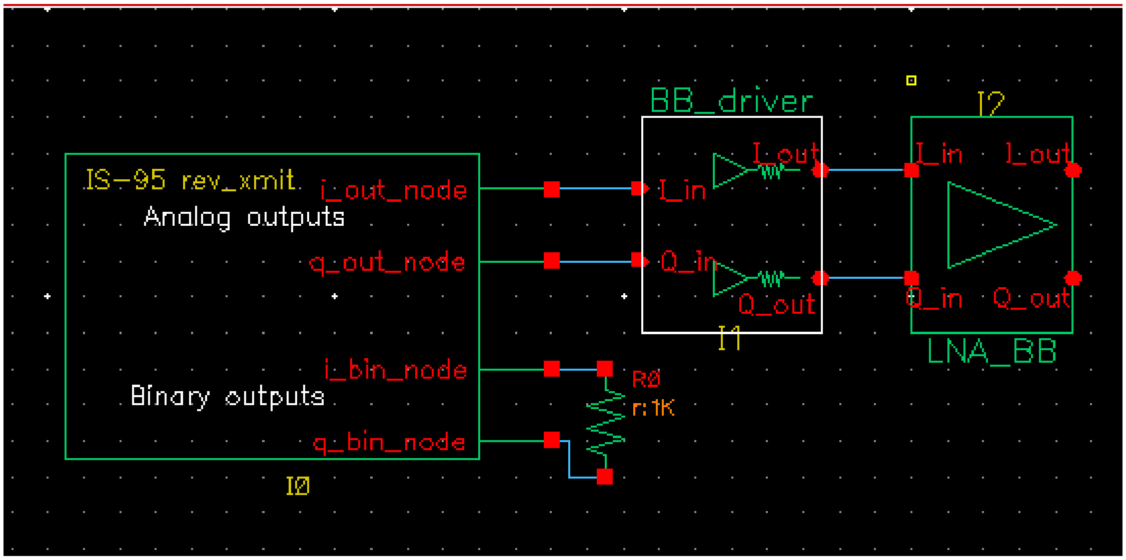


4. Move the cursor over the Schematic window. The outline for the LNA symbol is attached to the cursor. Align the input pins of the LNA with the output pins of the driver.
5. Click to place the *LNA_BB* block to the right of the *BB_driver* block.
6. Click *Esc* to remove the symbol from the cursor.

Wiring the Driver to the LNA

1. To wire the *LNA_BB* block to the *BB_driver* block, in the Schematic window choose *Add - Wire (narrow)*.
2. Click `I_out` on the *BB_driver* block then click `I_in` on the *LNA_BB* block.
3. Click `Q_out` on the *BB_driver* block then click `Q_in` on the *LNA_BB* block.
4. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the LNA

1. Edit the CDF parameter values for the LNA.

a. Choose *Edit – Properties – Objects* in the Schematic window.

The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for the LNA and modify the schematic for this simulation.

b. In the Schematic window, click the *LNA*.

The Edit Object Properties form changes to display information for the LNA.

c. Change the CDF parameter values for the LNA as follows.

Parameter Name	Value
Available pwr gain [dB]	`lna_gain`
Input resistance	`50`
Output resistance	`300`
Input referred IP3 [dBm]	`lna_ip3`
Noise figure [dB]	`10`

Parameter Name	Value
am/pm sharpness	`2`
cmp [dBm]	`lna_ip3`
|*radians*| *@ cmp*	`.05`
|*radians*| *@ big input*	`.7`
{1, 0, -1} for {cw, none, ccw}	`1`

2. Click *OK* in the Edit Object Properties form.

The form closes.

Adding a Butterworth Band Pass Filter

Add a Butterworth band pass filter to the right of the low noise amplifier.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

Information for the LNA is still displayed in the form.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

3. In the Library Browser - Add Instance form, make the following selection.

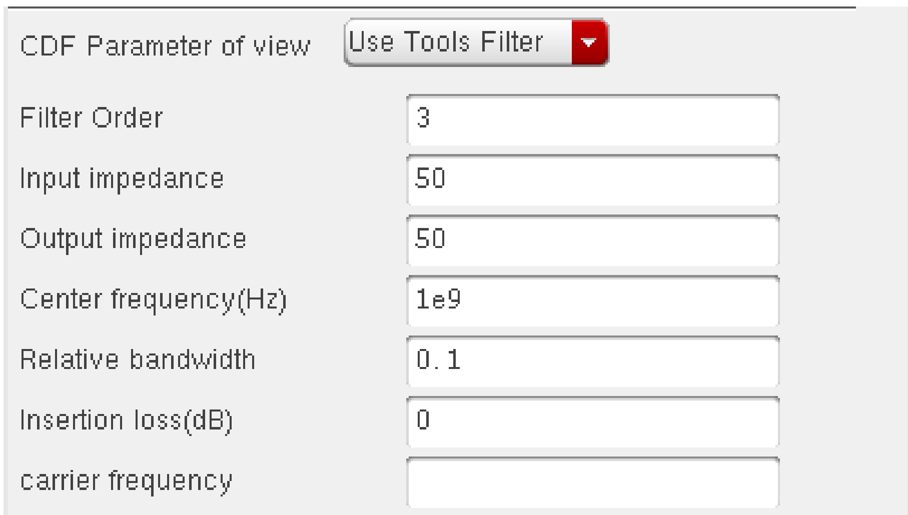
Library	Category	Cell	View
rfLib	*top_dwnBB*	*BB_butterworth_bp*	*symbol*

In the Library Browser, cell *BB_butterworth_bp* and it's default view are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library* field
- ❑ `BB_butterworth_bp` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field.

- ❑ The CDF parameters for the Butterworth band pass filter and their default values display at the bottom of the Add Instance form.

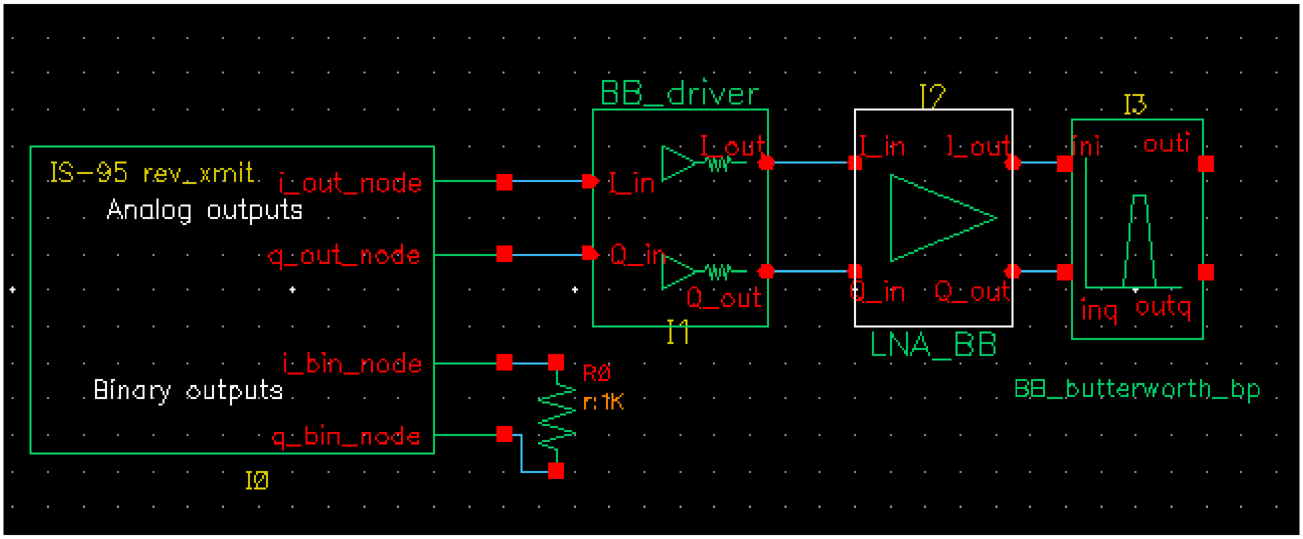


4. Move the cursor over the Schematic window. The outline for the filter symbol is attached to the cursor. Align the input pins of the filter with the output pins of the LNA.
5. Click to place the *BB_butterworth_bp* block to the right of the *LNA_BB* block.
6. Click *Esc* to remove the symbol from the cursor.

Wiring the LNA to the Filter

1. To wire the *BB_butterworth_bp* block to the *LNA_BB* block, in the Schematic window choose *Add – Wire (narrow)*.
2. Click `I_out` on the *LNA_BB* block then click `ini` on the *BB_butterworth_bp* block.
3. Click `Q_out` on the *LNA_BB* block then click `inq` on the *BB_butterworth_bp* block.
4. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the Band Pass Filter

Edit the CDF parameter values for the Butterworth band pass filter.

1. Choose *Edit – Properties – Objects* in the Schematic window.

   The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for the filter and modify the schematic for this simulation.

2. In the Schematic window, click the filter.

   The Edit Object Properties form changes to display information for the filter.

3. Edit the parameter values to match those in Table 2-2.

4. Click *OK*.

Table 2-2 CDF Parameter Values for the Butterworth Filter

Parameter Name	Value
Filter order	`3`
Input impedance	`50`
Output impedance	`50`
Center frequency (Hz)	`frf`

Table 2-2 CDF Parameter Values for the Butterworth Filter

Parameter Name	Value
Relative bandwidth	`rf_rbw`
Insertion loss (dB)	`3`
Carrier frequency	`frf`

Specify the *Carrier frequency* parameter value for the baseband equivalent model of the Butterworth band pass filter, just as you do for any reactive element. As shown in Figure B-6 on page 2488, the carrier frequency is used to compute speed voltages. Because filters are built up from inductors and capacitors which have speed voltages, you must specify the carrier frequency for filters.

When a filter follows an RF-to-IF mixer, its *Carrier frequency* parameter value is the IF frequency.

- The *Carrier frequency* is the frequency value to which the baseband signals are referenced.
- The *Center frequency* is the frequency for which a filter is designed. The *Center frequency* parameter value for a bandpass filter does not have to equal the *Carrier frequency* parameter value.

Adding an RF-to-IF Mixer

Add an RF-to-IF mixer (*dwn_cnvrt*) block to the right of the bandpass filter block.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.
3. In the Library Browser - Add Instance form, make the selections indicated in Table 2-3.

Table 2-3 Library Browser selections for the RF to IF Mixer

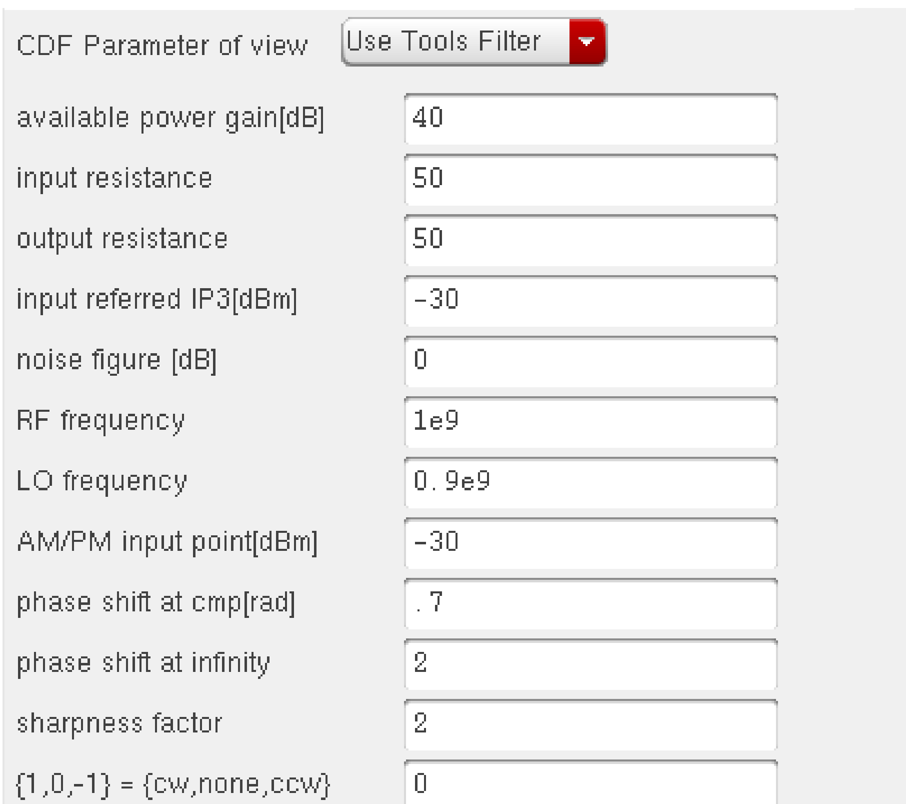
Library	Category	Cell	View
rfLib	*top_dwnBB*	*dwn_cnvrt*	*symbol*

In the Library Browser, cell *dwn_cnvrt* (the RF-to-IF mixer) and it's default view *symbol* are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library* field
- ❑ `dwn_cnvrt` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field.

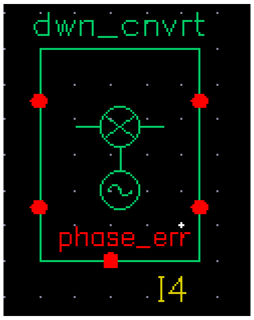
The CDF parameters for the down converter and their default values display at the bottom of the form.



4. Move the cursor over the Schematic window. The outline for the RF-to-IF Mixer symbol is attached to the cursor. Align the input pins of the mixer with the output pins of the filter.
5. Click to place the *dwn_cnvrt* block to the right of the *BB_butterworth_bp* block.
6. Click *Esc* to remove the symbol from the cursor.

Grounding the phase_err Pin on the Mixer

It is necessary to ground the phase error (*phase_err*) pin on the bottom of the RF-to-IF mixer.



1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

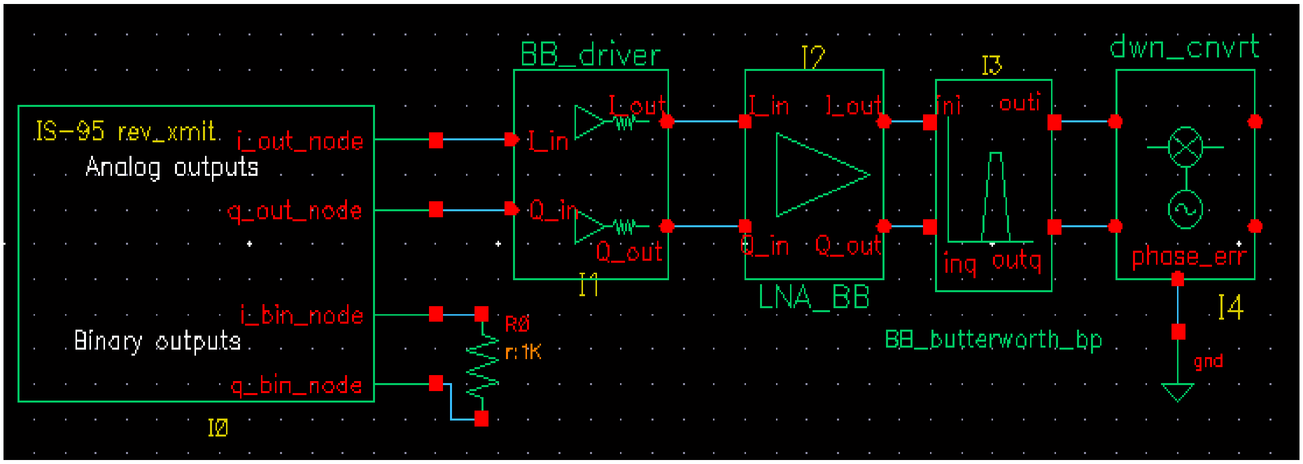
   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, type
   - `analogLib` in the *Library* field
   - `gnd` in the *Cell* field
   - `symbol` in the *View* field.
3. Move the cursor over the Schematic window.
4. Click to place the ground terminal in line with the phase error node (`phase_err`) on the bottom of the *dwn_cnvt* block.
5. Click *Esc* to remove the symbol from the cursor.

Wiring the Filter and Ground to the Mixer

1. To wire the *dwn_cnvrt* block to the *BB_butterworth_bp* block and the ground, in the Schematic window choose *Add - Wire (narrow)*.
2. Click `outi` on the *BB_butterworth_bp* block then click `I_in` on the *dwn_cnvrt* block.
3. Click `outq` on the *BB_butterworth_bp* block then click `Q_in` on the *dwn_cnvrt* block.
4. Click the port on the *gnd* block then click `phase_err` on the *dwn_cnvrt* block.
5. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the RF-to-IF Mixer

1. Edit the CDF parameter values for the RF-to-IF mixer (*dwn_cnvrt*) as listed in Table 2-4 on page 2509.

   a. Choose *Edit – Properties – Objects* in the Schematic window.

      The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for *dwn_cnvrt* and modify the schematic for this simulation.

   b. In the Schematic window, click *dwn_cnvrt*.

      The Edit Object Properties form changes to display information for *dwn_cnvrt*.

   c. Change the parameter values to match those in Table 2-4.

   d. Click OK.

Table 2-4 CDF Parameter Values for the RF-to-IF Mixer

Parameter Name	Value
available power gain[dB]	`if_mx_gain`
Input resistance	`50`
output resistance	`50`
input referred ip3[dBm]	`if_mx_ip`
noise figure [dB]	`10`

Table 2-4 CDF Parameter Values for the RF-to-IF Mixer

Parameter Name	Value
RF frequency	`frf`
LO frequency	`flo1`
AM/PM input point[dBm]	`-30`
phase shift at cmp[rad]	`.7`
phase shift at infinity	`2`
sharpness factor	`2`
{1,0,–1} = {cw,none,ccw}	`0`

Adding Another Butterworth Bandpass Filter

Add another Butterworth band pass filter block to the right of the RF-to-IF Mixer block.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.
2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.
3. In the Library Browser - Add Instance form, make the following selection.

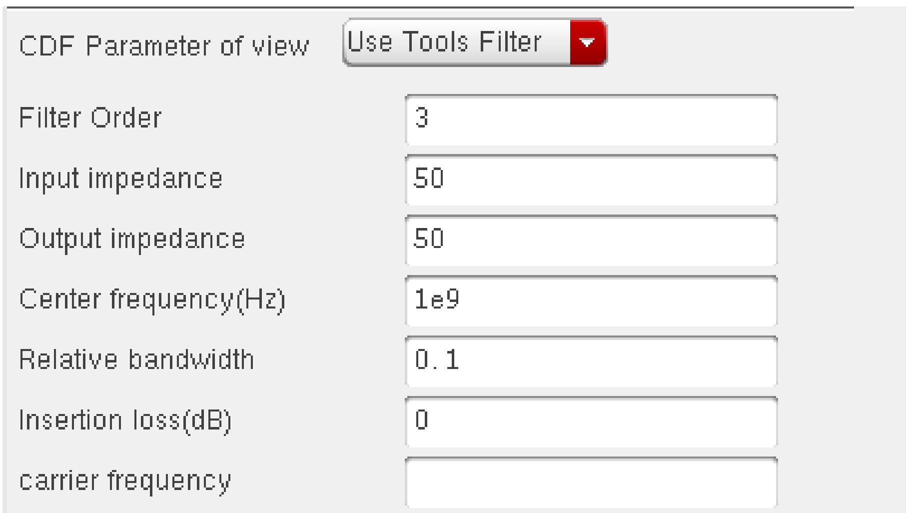
Library	Category	Cell	View
rfLib	*top_dwnBB*	*BB_butterworth_bp*	*symbol*

In the Library Browser, cell *BB_butterworth_bp* and it's default view are both selected.

At the top of the Add Instance form,

- `rfLib` displays in the *Library* field
- `BB_butterworth_bp` displays in the *Cell* field
- `symbol` displays in the *View* field.

The CDF parameters for the Butterworth band pass filter and their default values display at the bottom of the Add Instance form.



CDF Parameter of view	Use Tools Filter
Filter Order	3
Input impedance	50
Output impedance	50
Center frequency(Hz)	1e9
Relative bandwidth	0.1
Insertion loss(dB)	0
carrier frequency	

4. Move the cursor over the Schematic window. The outline for the filter symbol is attached to the cursor. Align the input pins of the filter with the output pins of the LNA.

5. Click to place the *BB_butterworth_bp* block to the right of the *LNA_BB* block.

6. Click *Esc* to remove the symbol from the cursor.

Wiring the Mixer to the Filter

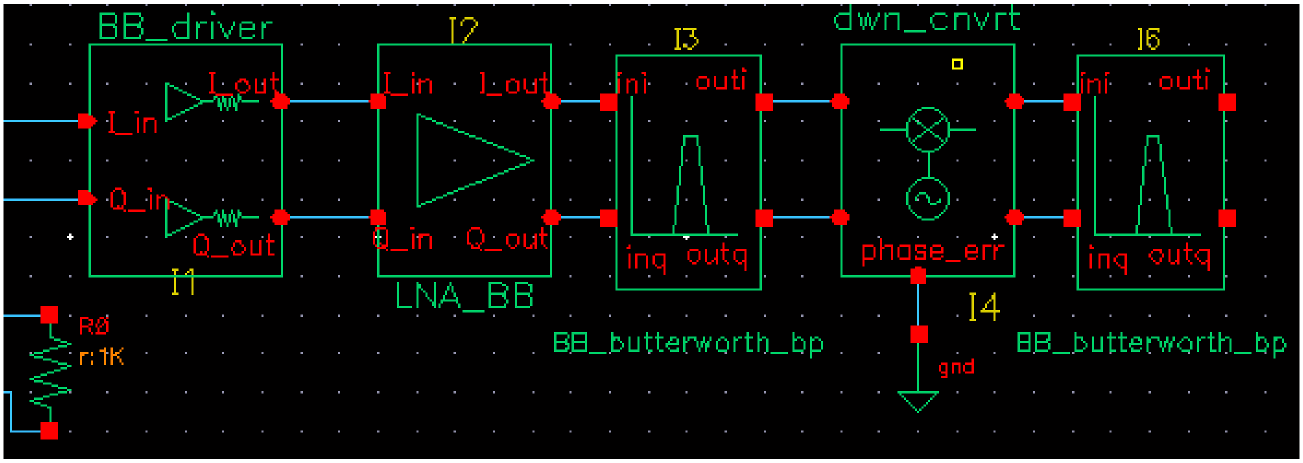
1. To wire the *BB_butterworth_bp* block to the *dwn_cnvrt* block, in the Schematic window choose *Add - Wire (narrow)*.

2. Click `I_out` on the *dwn_cnvrt* block then click `ini` on the *BB_butterworth_bp* block.

3. Click `Q_out` on the *dwn_cnvrt* block then click `inq` on the *BB_butterworth_bp* block.

4. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the Band Pass Filter

1. Edit the CDF parameter values for the Butterworth band pass filter as listed in Table 2-5 on page 2512.

a. Choose *Edit – Properties – Objects* in the Schematic window.

The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for the filter and modify the schematic for this simulation.

b. In the Schematic window, click the filter.

The Edit Object Properties form changes to display information for the filter.

c. Change the parameter values to match those in Table 2-5.

d. Click OK.

Table 2-5 CDF Parameter Values for the Second Butterworth Filter

Parameter Name	Value
Filter Order	`3`
Input impedance	`50`
Output impedance	`50`
Center frequency (Hz)	`-frf+flo1`
Relative bandwidth	`if_rbw`

Table 2-5 CDF Parameter Values for the Second Butterworth Filter

Parameter Name	Value
Insertion loss (dB)	`1`
Carrier frequency	`−frf+flo1`

As for the first band pass filter, specify the carrier frequency for the baseband equivalent model of the Butterworth band pass filter, just as you do for any reactive element. As shown in Figure B-6 on page 2488, the carrier frequency is used to compute speed voltages. Because filters are built up from inductors and capacitors which have speed voltages, you must specify the carrier frequency for filters.

When a filter follows an RF-to-IF mixer, its carrier frequency is the IF frequency. The carrier frequency is the frequency to which the baseband signals are referenced. The center frequency of the bandpass filter does not have to equal the carrier frequency. The center frequency is the frequency for which a filter is designed.

Adding an IQ Demodulator

Add an IQ Demodulator (*IQ_demod_BB*) block to the right of the bandpass filter block.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

3. In the Library Browser - Add Instance form,

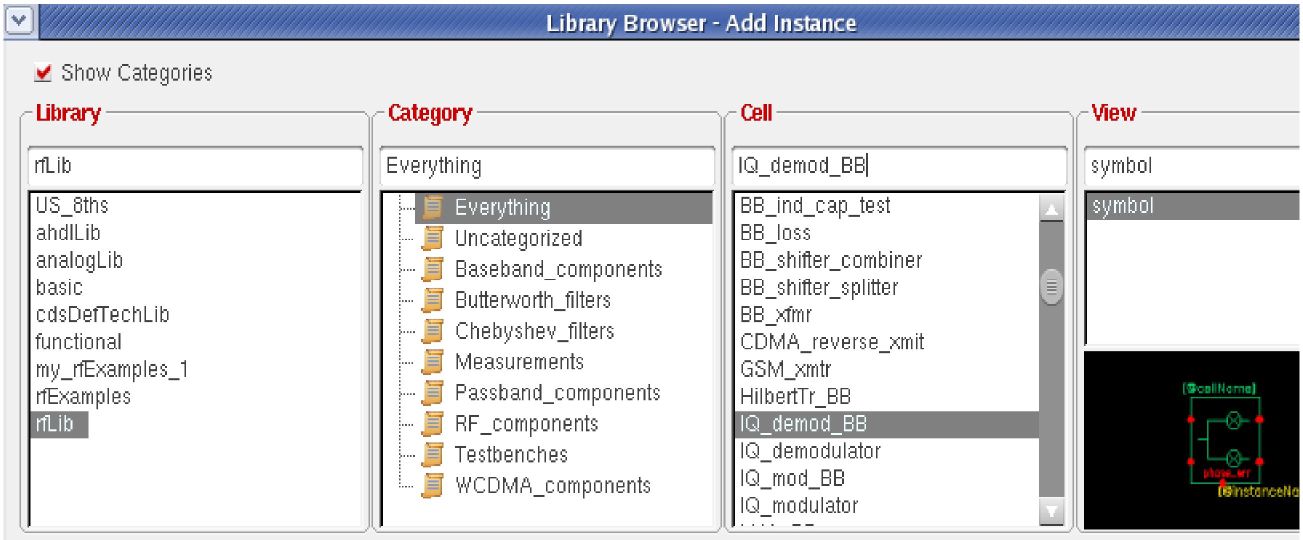
   a. If necessary, click *Show Categories* to display the *Category* column.

   b. Click *rfLib* to display elements in *rfLib*. The *Everything* category is displayed by default and all cells in *rfLib* are listed in the *Cells* column.

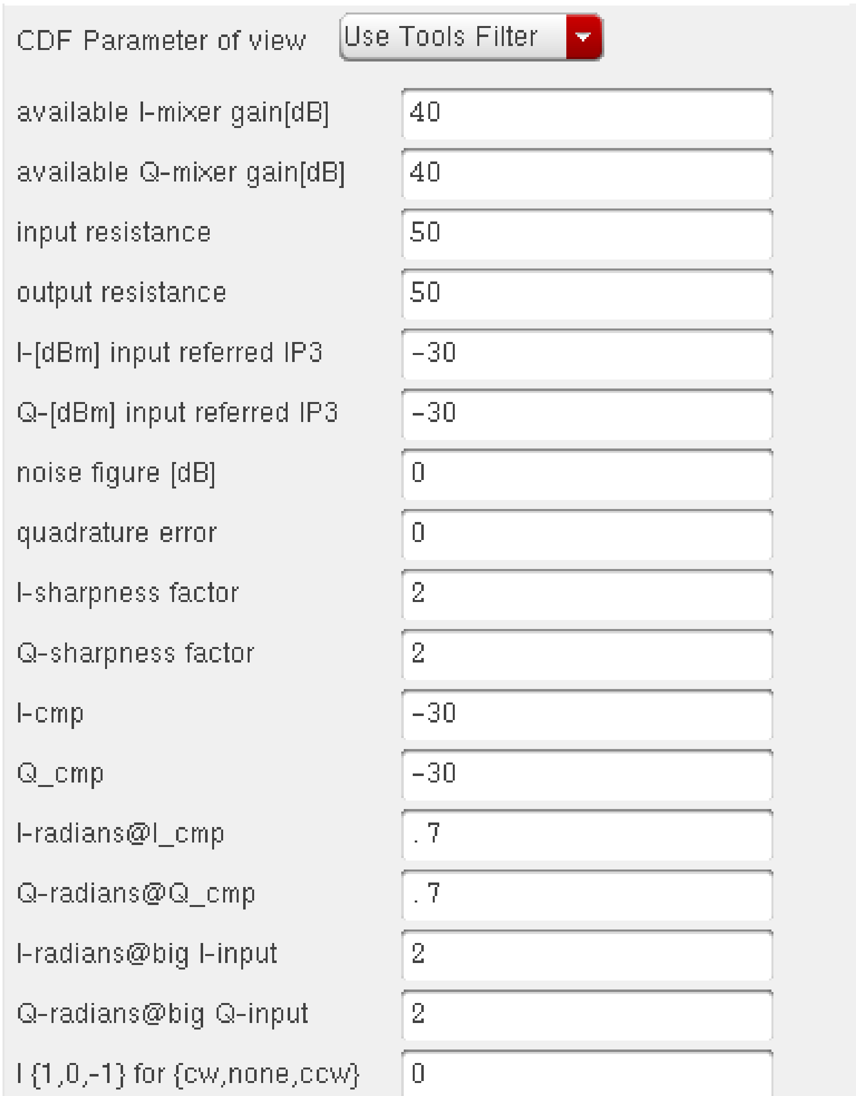
   c. In the *Category* column, click *top_dwnBB* to display cells in the *top_dwnBB* category.

   d. In the *Cell* column, click *IQ_demod_BB*.

In the Library Browser, cell *IQ_demod_BB* (the IQ demodulator) and it's default view *symbol* are both selected.



At the top of the Add Instance form, `rfLib` displays in the *Library* field, `IQ_demod_BB` displays in the *Cell* field and `symbol` displays in the *View* field. The CDF parameters for the IQ demodulator and their default values display at the bottom of the form.

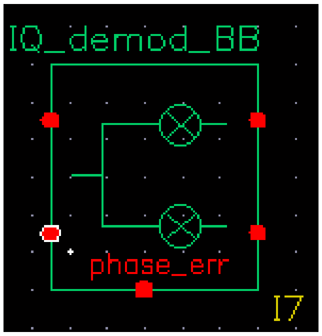


4. Move the cursor over the Schematic window. The outline for the IQ demodulator symbol is attached to the cursor. Align the input pins of the IQ demodulator with the output pins of the filter and click to place the *IQ_demod_BB* block to the right of the second *BB_butterworth_bp* block.

5. Click *Esc* to remove the symbol from the cursor.

Grounding the phase_err Pin on the IQ Demodulator

It is necessary to ground the phase error (*phase_err*) pin on the bottom of the IQ demodulator.



1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

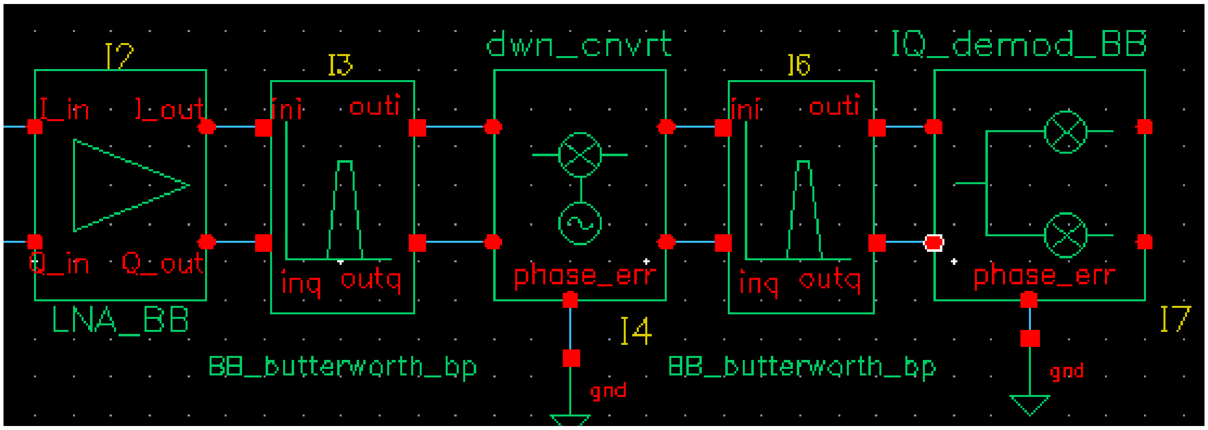
2. In the Add Instance form, type
   - ❑ `analogLib` in the *Library* field
   - ❑ `gnd` in the *Cell* field
   - ❑ `symbol` in the *View* field.
3. Move the cursor over the Schematic window.
4. Click to place the ground terminal in line with the phase error node (`phase_err`) on the bottom of the *IQ_demod_BB* block.
5. Click *Esc* to remove the symbol from the cursor.

Wiring the Filter and Ground to the IQ Demodulator

1. To wire the *IQ_demod_BB* block to the *BB_butterworth_bp* block and the ground, in the Schematic window choose *Add - Wire (narrow)*.
2. Click `outi` on the *BB_butterworth_bp* block then click `I_in` on the *IQ_demod_BB* block.
3. Click `outq` on the *BB_butterworth_bp* block then click `Q_in` on the *IQ_demod_BB* block.
4. Click the port on the *gnd* block then click `phase_err` on the *IQ_demod_BB* block.

5. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the IQ Demodulator

1. Edit the CDF parameter values for the IQ Demodulator (*IQ_demod_BB*) as listed in Table 2-6 on page 2517.

a. Choose *Edit – Properties – Objects* in the Schematic window.

The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for *IQ_demod_BB* and modify the schematic for this simulation.

b. In the Schematic window, click *IQ_demod_BB*.

The Edit Object Properties form changes to display information for *IQ_demod_BB*.

c. Change the parameter values to match those in Table 2-6.

d. Click OK.

Table 2-6 CDF Parameter Values for the IQ Demodulator

Parameter Name	Value
available I-mixer gain[dB]	0
available Q-mixer gain[dB]	0
Input resistance	50
output resistance	50

Table 2-6 CDF Parameter Values for the IQ Demodulator

Parameter Name	Value
I–[dBm] input referred IP3	`40`
Q–[dBm] input referred IP3	`40`
noise figure [dB]	`2`
quadrature error	`0`
I–sharpness factor	`2`
Q–sharpness factor	`2`
I_cmp	`–30`
Q_cmp	`–30`
I–radians@I_cmp	`.7`
Q–radians@Q_cmp	`.7`
I–radians@big I-input	`2`
Q–radians@big Q-input	`2`
I {1,0,–1} for {cw,none,ccw}	`0`
Q {1,0,–1} for {cw,none,ccw}	`0`

Adding Two Butterworth Lowpass Filters

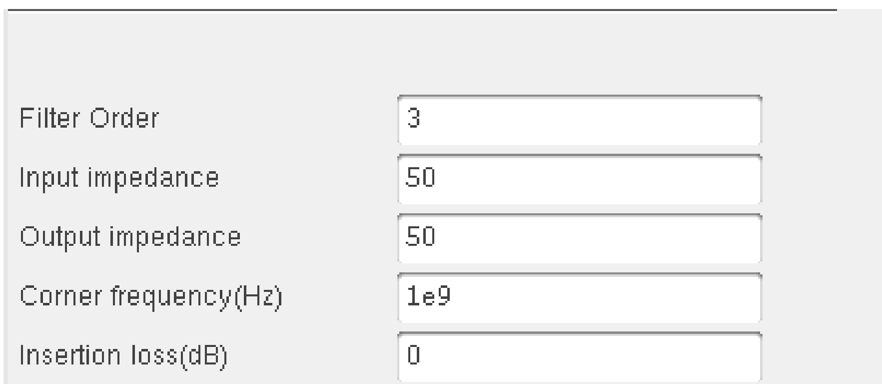
1. Add two Butterworth low pass filters to the right of the IQ demodulator block.
   - ❑ Align one filter with the demodulator's `i_out` pin.
   - ❑ Align the other filter with its `q_out` pin.
2. In the Schematic window, choose *Add – Instance* to display the Add Instance form.
3. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.
4. In the Library Browser - Add Instance form, make the following selections.

Library	Category	Cell	View
rfLib	*top_dwnPB*	*butterworth_lp*	*symbol*

In the Library Browser, cell *butterworth_lp* and it's default view are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library* field
- ❑ `butterworth_lp` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field.
- ❑ The CDF parameters for the Butterworth low pass filter and their default values display at the bottom of the Add Instance form.



5. Move the cursor over the Schematic window. The outline for the butterworth low pass filter symbol is attached to the cursor.

Align the *in* pin of the first butterworth low pass filter with the *I_out* pin (the top pin) of the IQ demodulator and click to place the filter close to the demodulator. Align the *in* pin of the second butterworth low pass filter with the *Q_out* pin (the bottom pin) of the IQ demodulator. You have to place it further from the demodulator to align it with the *Q_out* pin.

6. Click *Esc* to remove the symbol from the cursor.

Wiring the IQ Demodulator to the Filters

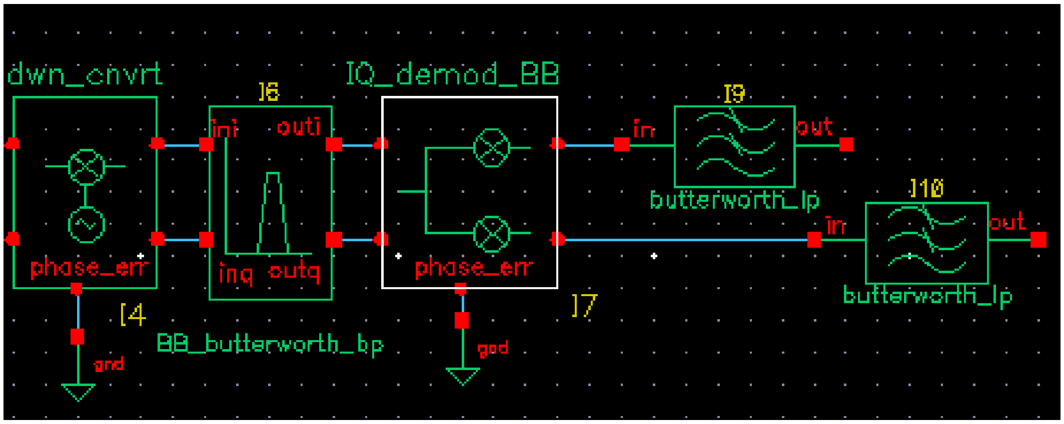
1. To wire the *IQ_demod_BB* block to the *butterworth_lp* blocks, in the Schematic window choose *Add - Wire (narrow)*.

2. Click `I_out` on the *IQ_demod_BB* block then click `in` on the first *butterworth_lp* block.

3. Click `Q_out` on the *IQ_demod_BB* block then click `in` on the second *butterworth_lp* block.

4. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for Both Low Pass Filters

1. Edit the CDF parameter values for the Butterworth low pass filters as listed in Table 2-7 on page 2520.

a. Choose *Edit – Properties – Objects* in the Schematic window.

The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for each filter and modify the schematic for this simulation.

b. In the Schematic window, click the first low pass filter.

The Edit Object Properties form changes to display information for the filter.

c. Change the parameter values to match those in Table 2-7.

Table 2-7 CDF Parameter Values for the Butterworth Low Pass Filters

Parameter Name	Value
Filter Order	3
Input impedance	50
Output impedance	50
Corner frequency (Hz)	10M
Insertion loss (dB)	0

d. Click Apply.

e. In the Schematic window, click the second low pass filter.

The Edit Object Properties form displays the information you entered for the filter as shown in Table 2-7.

f. Click OK.

Adding an Instrumentation Block

Add an instrumentation block (*offset_comms_instr*) block to the right of the low pass filter blocks.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

3. In the Library Browser - Add Instance form, make the following selections. (If necessary, click Show Categories at the top of the Library Browser, to display the Category column.)

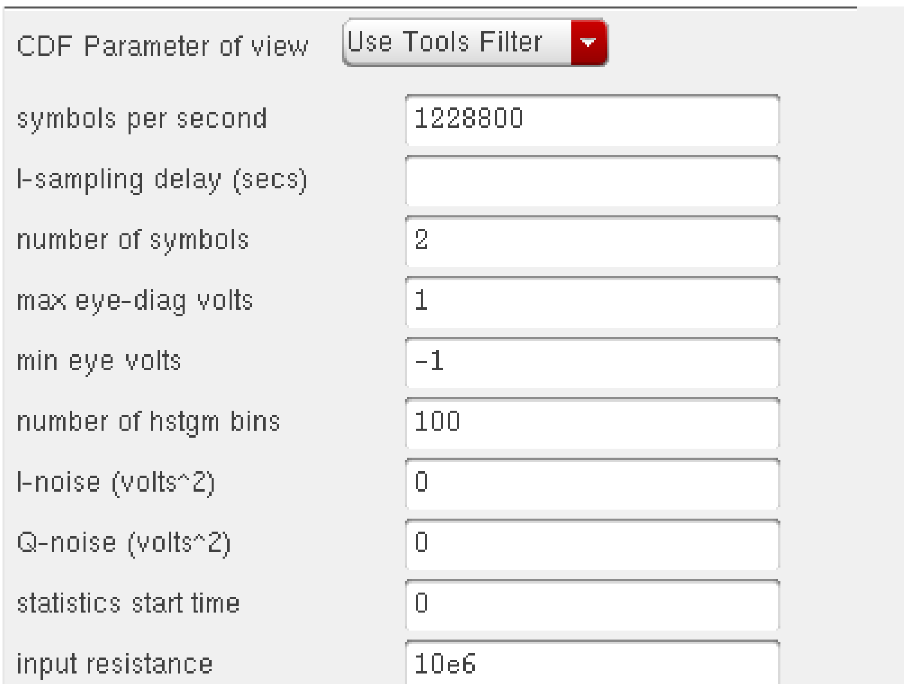
Library	Category	Cell	View
rfLib	*measurement*	*offset_comms_instr*	*symbol*

In the Library Browser, cell *offset_comms_instr* and it's default view *symbol* are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library*
- ❑ `offset_comms_instr` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field

The CDF parameters for the *offset_comms_instr* and their default values display at the bottom of the form.

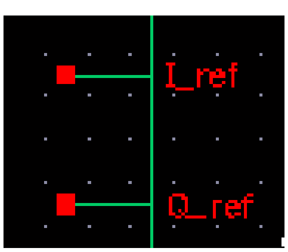


4. Move the cursor over the Schematic window. The outline for the *offset_comms_instr* block symbol is attached to the cursor. Align the *I_in* and *Q_in* pins of the *offset_comms_instr* block with the *out* pins of the butterworth low pass filters and click to place the *offset_comms_instr* block to the right of the low pass filter blocks.

5. Click *Esc* to remove the symbol from the cursor.

Grounding the Reference Pins on the Instrumentation Block

It is necessary to ground the reference pins (*I_ref* and *Q_ref*) pin near the lower left corner of the instrumentation block.



Grounding the phase_err Pin on the Mixer

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, type
   - ❑ `analogLib` in the *Library* field
   - ❑ `gnd` in the *Cell* field
   - ❑ `symbol` in the *View* field.

3. Move the cursor over the Schematic window.

4. Click to place the ground terminal in line with the `Q_ref` node on the bottom of the *offset_comms_instr* block.

5. Click *Esc* to remove the symbol from the cursor.

Wiring the Filter and Ground to the Instrumentation Block

1. To wire the low pass filters to the *offset_comms_instr* block and the ground, in the Schematic window choose *Add - Wire (narrow)*.

2. Click `out` on the upper *butterworth_lp* block (the low pass filter connected to the `I_out` node on the IQ demodulator) then click `I_in` on the *offset_comms_instr* block.

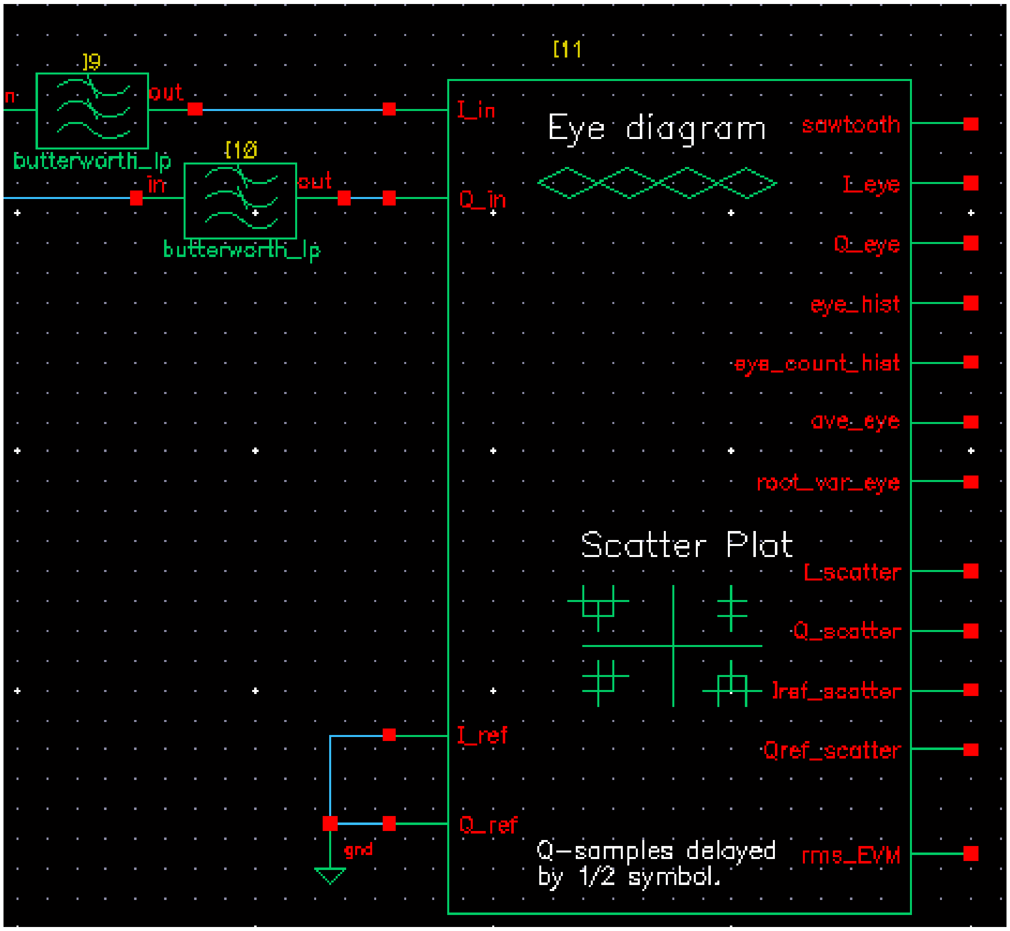
3. Click `out` on the lower *butterworth_lp* block (the low pass filter connected to the `Q_out` node on the IQ demodulator) then click `Q_in` on the *offset_comms_instr* block.

4. Click the port on the *gnd* block then click the `Q_ref` node on the *offset_comms_instr* block.

5. Click the port on the *gnd* block then click the `I_ref` node on the *offset_comms_instr* block.

6. Click *Esc* to stop wiring.

The schematic now appears as follows.



Modifying Parameter Values for the Instrumentation Block

1. Edit the CDF parameter values for the *offset_comms_instr* as listed in Table 2-8 on page 2525.

   a. Choose *Edit – Properties – Objects* in the Schematic window.

      The Edit Object Properties form appears. You use this form to change the list of CDF (component description format) properties for *offset_comms_instr* and modify the schematic for this simulation.

   b. In the Schematic window, click the *offset_comms_instr* block.

      The Edit Object Properties form changes to display information for *offset_comms_instr* block.

   c. Change the parameter values to match those in Table 2-8.

d. Click OK.

Table 2-8 CDF Parameter Values for the Instrumentation Block

Parameter Name	Value
symbols per second	`1228800`
I-sampling delay (secs)	`134n`
number of symbols	`2`
max eye-diag volts	`1`
min eye volts	`–1`
number of hstgm bins	`100`
I-noise (volts^2)	`0`
Q-noise (volts^2)	`0`
statistics start time	`30u`
input resistance	`50`

Adding an Instrumentation Terminator

Add an instrumentation termination (*instr_term*) block to the right of the instrumentation block. The *instr_term* block terminates the outputs on the instrumentation block and prevents unused pin warnings.

1. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

   The Add Instance form appears. It may be empty or it may display information for a previously added element.

2. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

3. In the Library Browser - Add Instance form, make the following selections. (If necessary, click Show Categories at the top of the Library Browser, to display the Category column.)

Library	Category	Cell	View
rfLib	*measurement*	*instr_term*	*symbol*

In the Library Browser, cell *instr_term* and it’s default view *symbol* are both selected.

At the top of the Add Instance form,

- ❑ `rfLib` displays in the *Library*
- ❑ `instr_term` displays in the *Cell* field
- ❑ `symbol` displays in the *View* field

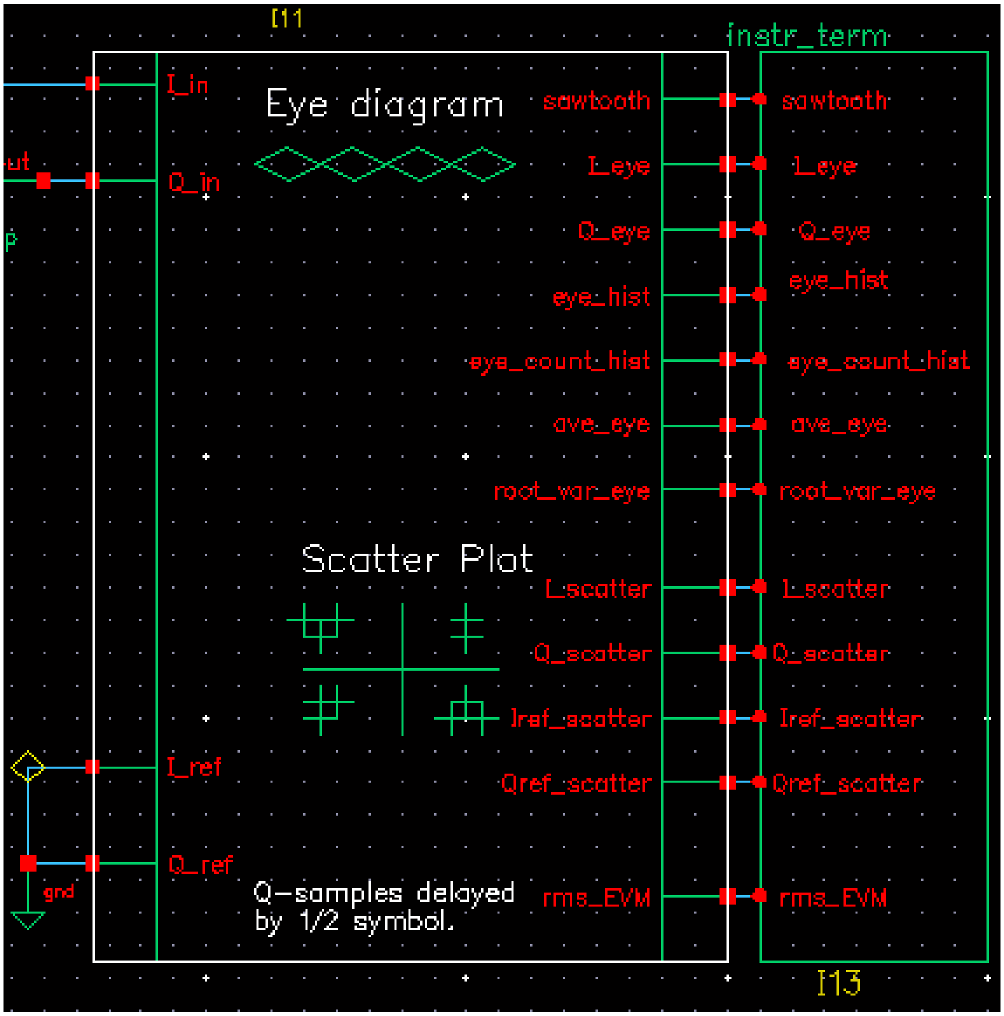
There are no CDF parameters for the *instr_term* cell.

4. Move the cursor over the Schematic window. The outline for the *instr_term* symbol is attached to the cursor. Move the *instr_term* block to the right of the instrumentation block and align the input pins of the instrumentation termination block with the output pins of the instrumentation block.
5. Click to place the *instr_term* block.
6. Click *Esc* to remove the symbol from the cursor.

Wiring the Termination Block to the Instrumentation Block

1. To wire the instrumentation (*offset_comms_instr)* block to the Instrumentation terminator (*instr_term)* block, in the Schematic window choose *Add - Wire (narrow)*.
2. Wire the aligned pins straight across.
3. Click *Esc* to stop wiring.

The schematic should look as follows.

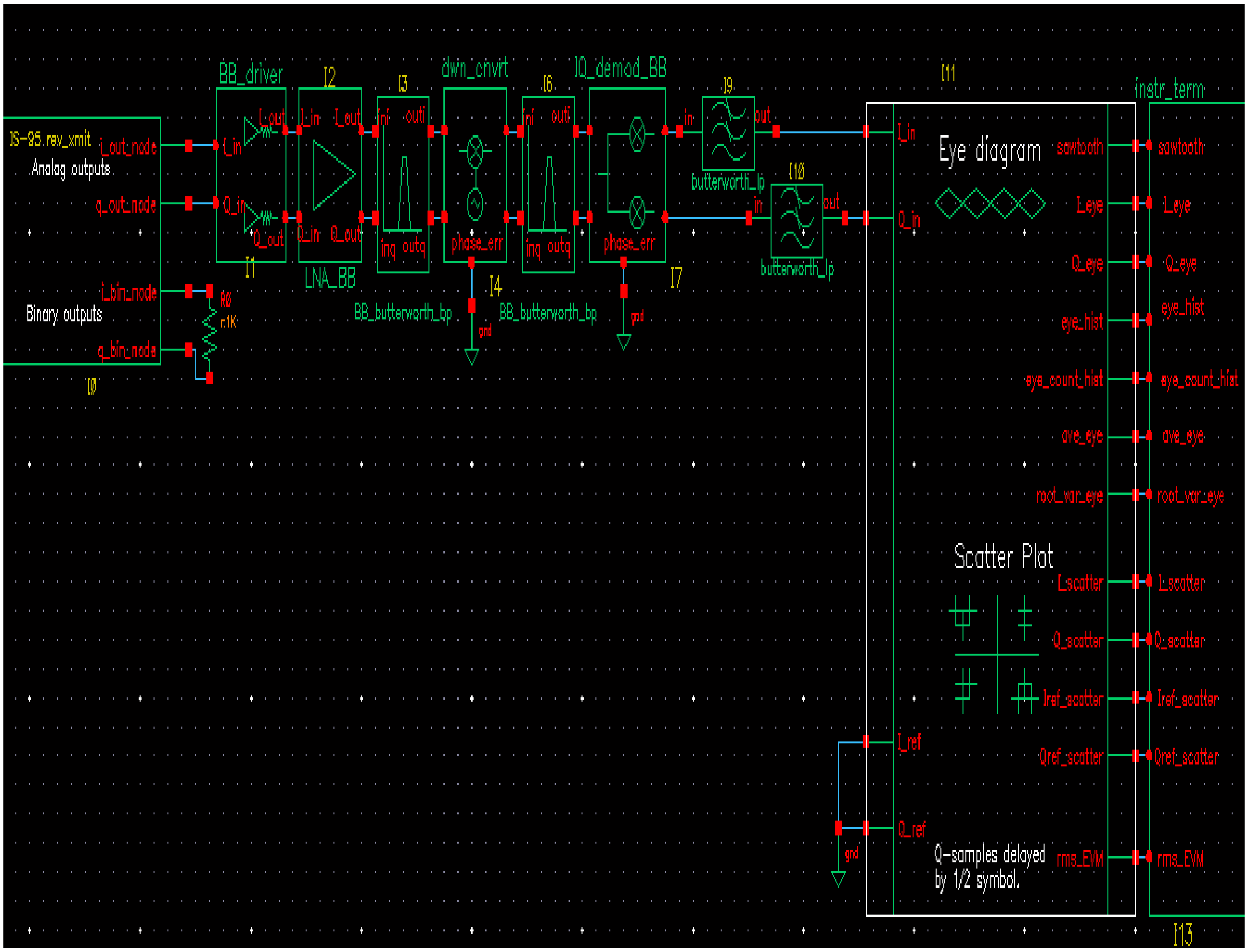


4. In the Schematic window, choose *Design – Design Check and Save*.

   The completed schematic is verified and saved.

The schematic for the complete receiver model should look like the one in Figure B-7 on page 2528.

Figure B-7 Completed receiver model

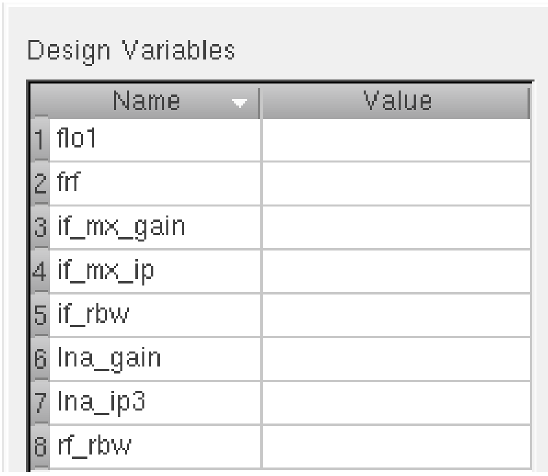


Setting Variable Values for the Receiver Schematic

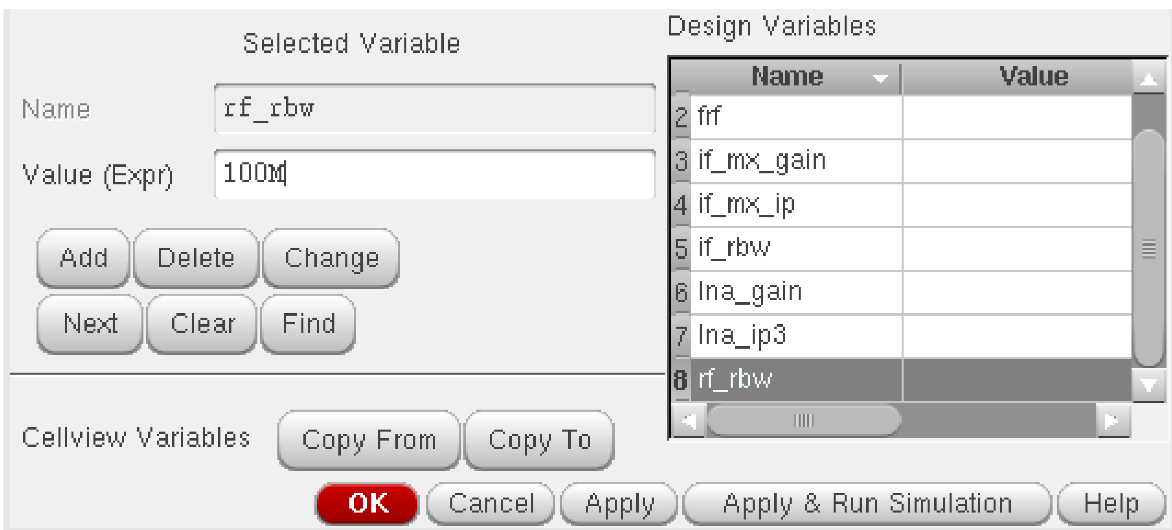
Copy the variables you entered as CDF parameters for the individual blocks from the receiver schematic to the ADE window. Then edit each variable to give it the value specified in Table B-9 on page 2529.

1. In the ADE window, choose *Variables – Copy From Cellview* to copy the variables from the receiver schematic to the Design Variables area on the ADE window.

The copied variables display in the *Design Variables* area in the ADE window.



2. In the ADE window, choose *Variables – Edit* to open the Editing Design Variables form.



Adding the Values to the Copied Variables

In the Editing Design Variables form, one by one, select each variable in the *Table of Design Variables* and associate with each one, the value listed in Table B-9.

Table B-9 Values for Receiver Variables

Variable	Value
`lna_gain`	15

Table B-9 Values for Receiver Variables

Variable	Value
`lna_ip3`	–5
`if_mx_gain`	10
`if_mx_ip`	35
`frf`	2.14G
`flo1`	2.354G
`if_rbw`	200m
`rf_rbw`	100m

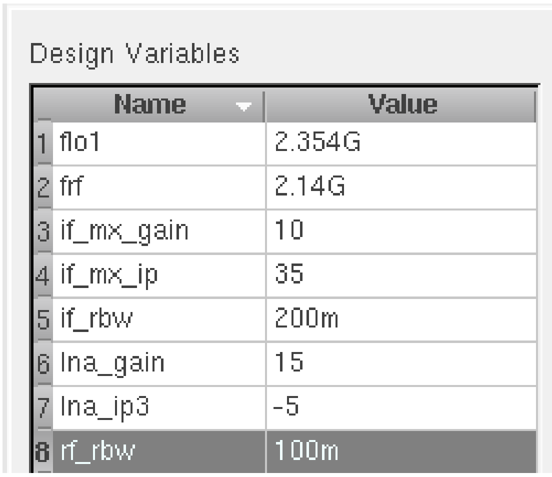
To associate a value with a design variable

1. In the *Table of Design Variables*, click `lna_gain`.

   `lna_gain` displays in the *Name* field.

2. In the *Value (Expr)* field, enter the number `15,` the value from Table B-9 on page 2529.
3. Click *Change* to list the variable name and its value from the *Table of Design Variables.*
4. Repeat these steps for the remaining variables listed in the *Table of Design Variables* to associate the values from Table B-9 on page 2529 with the variable names.
5. Click *OK* in the Editing Design Variables form after you have added all the variable values.

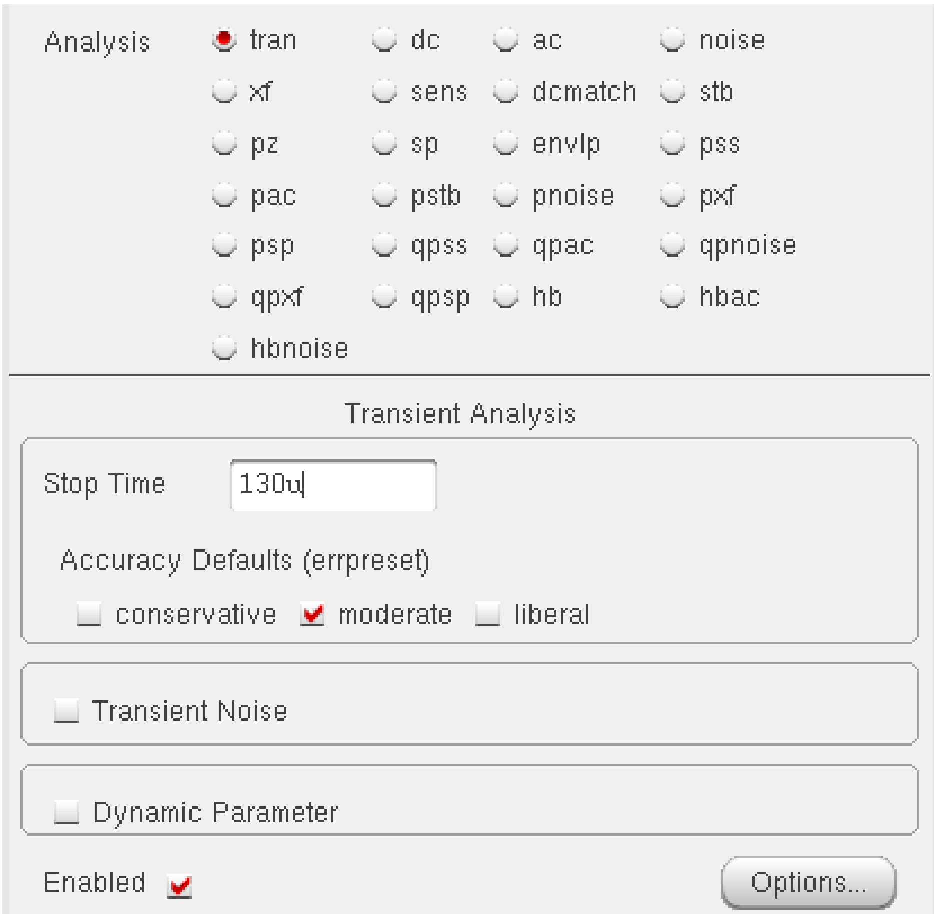
The table of *Design Variables* in the ADE window is updated and the Editing Design Variables form is closed.



Setting Up and Running a Transient Analysis

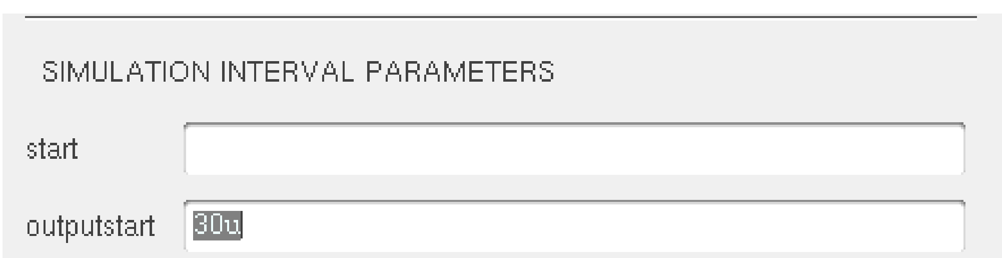
1. In the ADE window, choose *Analyses – Choose* to display the Choosing Analyses form.
2. In the Choosing Analyses form, if necessary, click *tran* to select a transient analysis.
3. In the Choosing Analyses form, enter `130u` in the *Stop Time* field.

4. Highlight *moderate* for *Accuracy Defaults (errpreset)*.



5. In the Choosing Analyses form, click *Options* to display the Transient Options form.

6. In the Transient Options form, enter `30u` in the *outputstart* field.



By delaying the output start, you remove start-up transients from the eye-diagrams and scatter plots.

7. Click *OK* in the Transient Options form.

8. Click *OK* in the Choosing Analyses form.
9. If you have not already done so, set up the simulator and model libraries with the following steps.
   a. Set the simulator options from the Simulator window as described in "Choosing Simulator Options" on page 2474.
   b. Set up the model libraries from the Simulator window as described in "Setting Up Model Libraries" on page 2476.
10. In the ADE window, choose *Simulation – Netlist and Run*.

    Messages display in the CIW. The simulation log window opens. Watch for messages stating that the simulation has completed successfully.

    Watch the CIW for messages stating the simulation is running and that it has completed successfully.

Examining the Results: Eye Diagram, Histogram, and Scatter Plot

In this section we examine the results of the transient analysis of the receiver.

Plotting the Eye Diagram (and Transient Response)

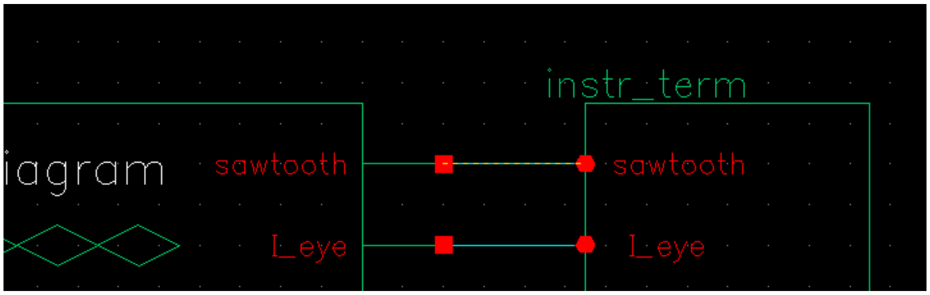
First plot an eye diagram.

1. In the ADE window, choose *Results – Direct Plot – Transient Signal*.

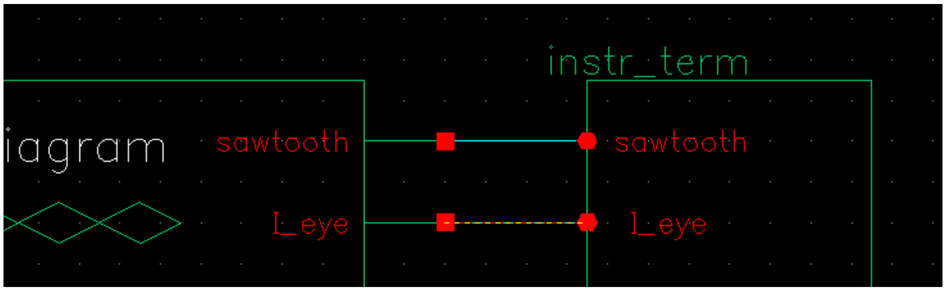
   The Virtuoso Schematic Editing window appears.
2. Following the prompts at the bottom of the window,

   > *Select nodes or terminals, press <esc> to finish selection*

   a. In the Schematic window, click the *sawtooth* net from the instrumentation (*offset_comms_instr*) block.

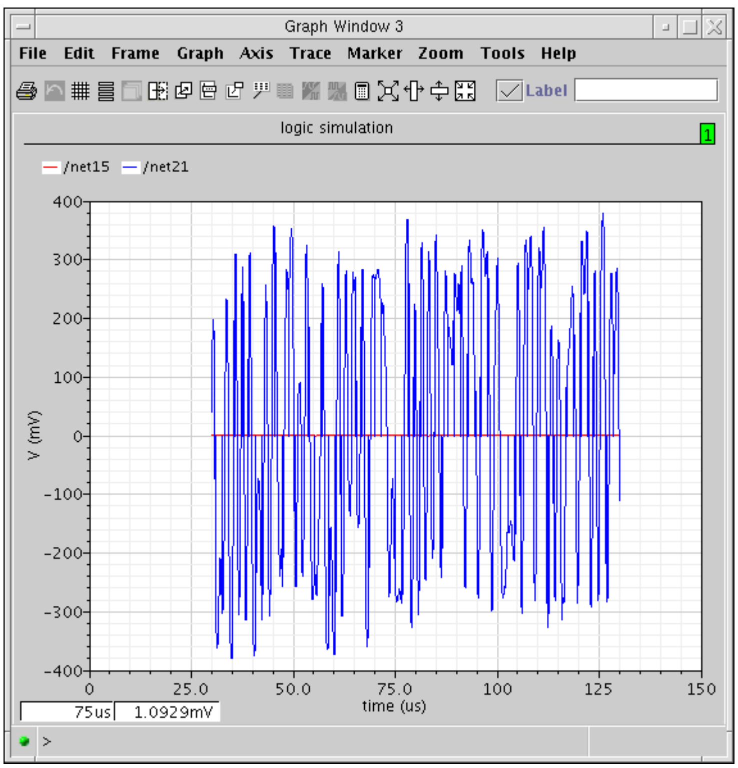


b. Click the *I_eye* net from the instrumentation (*offset_comms_instr*) block.



c. Press *Esc* to indicate that you have finished selecting outputs.

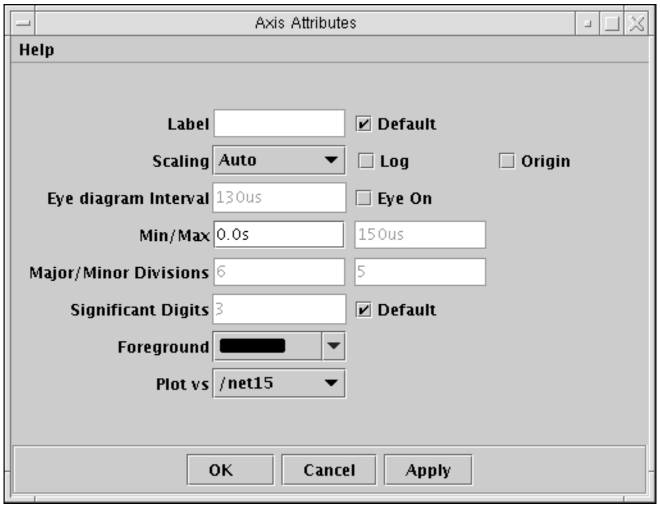
This opens a waveform window and creates a plot of the Transient Response of the net.



3. In the waveform window, double-click the X axis.

The Axis Attributes form displays:

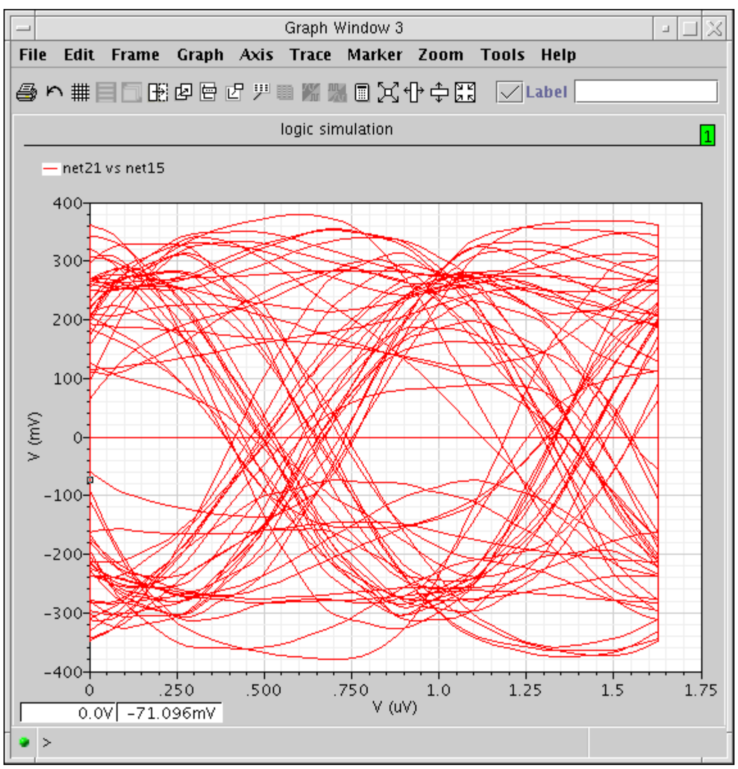
a. In the *Plot vs* field, select the `sawtooth` net, such as `/net15`.



b. Click *OK*.

You see the eye-diagram shown in Figure B-8 on page 2535.

Figure B-8 Eye-diagram



The I-sampling delay parameter in the instrumentation block (*I_del*) is chosen with respect to this eye diagram. The delay is the time when the eye opens the widest.

The instrumentation block samples the input waveforms with this delay to compute all statistics and to produce scatter plots.

4. In the waveform window, choose *File – Close*.

The waveform window closes.

Generate the Histogram

Now generate a histogram of the I-voltage at the sampling times.

1. In the ADE window, choose *Results – Direct Plot – Transient Signal*.

The Virtuoso Schematic Editing window appears.

2. Following the prompts at the bottom of the window,

  a. Click the *eye_hist* net from the instrumentation (*offset_comms_instr)* block.

  b. Click the *eye_count_hist* net from *offset_comms_instr*.

  c. Press *Esc* to indicate that you have finished selecting outputs.

  This creates a plot in the waveform window.

3. In the waveform window, double-click the X axis.

The Axis Attributes form displays:

  a. In the *Plot vs* field, select the *eye_hist* output, such as `/net17`.

  b. Click *OK*.

This creates an unintelligible intermediate plot in the waveform window.

4. In the waveform window, double-click the trace in the graph.

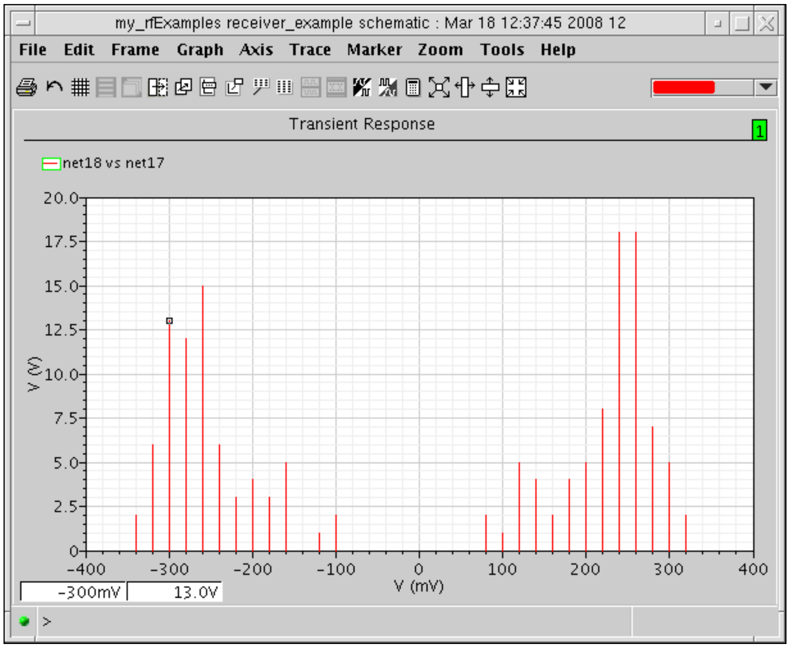
The Trace Attributes form displays.

  a. In the *Type/Style* cyclic field, select *Bars*.

  b. Click *OK*.

You see a plot like Figure B-9 on page 2537.

Figure B-9 Histogram



5. In the waveform window, choose *File – Close*.

   The waveform window closes.

Generating the Scatter Plot

Generate a scatter plot of the received symbols.

1. In the ADE window, choose *Results – Direct Plot – Transient Signal*.

   This displays the waveform window.

2. Following the prompts at the bottom of the waveform window,

   In the Schematic window:

   a. Click the *I_scatter* output from the instrumentation (*offset_comms_instr)* block.

   b. Click the *Q_scatter* output from *offset_comms_instr*.

   c. Click *Esc* to indicate that you have finished selecting outputs.

      This creates a plot in the waveform window.

3. In the waveform window, double-click the X axis.

The Axis Attributes form displays:

a. In the *Plot vs* field, select the *I_scatter* output, such as `/net22`.

b. Click *OK*.

4. In the waveform window, double-click the trace in the graph.

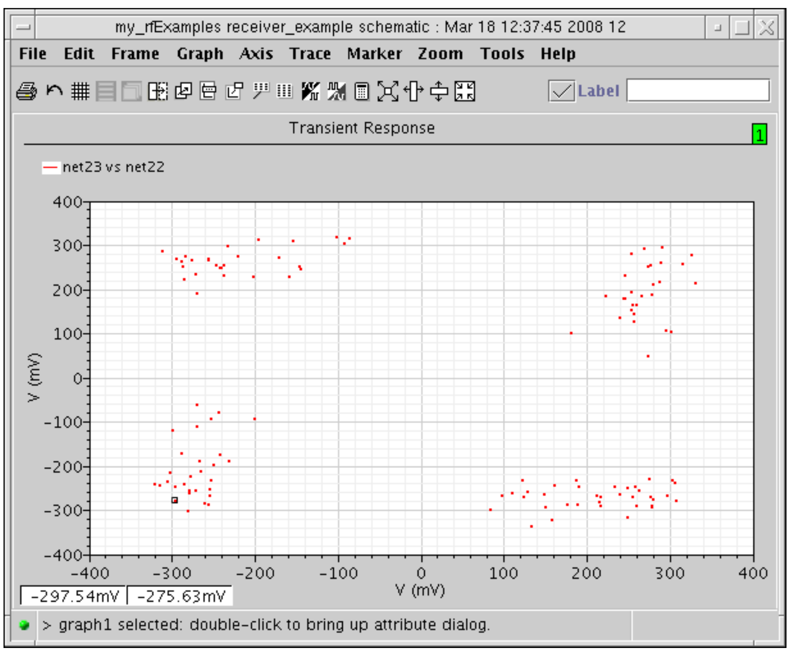
The Trace Attributes form displays.

a. In the *Type/Style* cyclic field, select *Points*.

b. Click *OK*.

You see a scatter plot like Figure B-10 on page 2538.

Figure B-10 Scatter Plot



5. In the waveform window, choose *File – Close*.

The waveform window closes.

The Various Instrumentation Blocks

The CDMA source (*CDMA_reverse_xmit)* produced offset QPSK symbols. Offset QPSK modulation avoids traversing the origin by staggering the digital changes in the I and Q signals. Running the baseband trajectory through the origin increases spectral regrowth in the transmitters.

The instrumentation block (*offset_comms_instr*) samples the I and Q signals at different times then plots the two staggered samples against each other. The resulting scatter plot shows the received symbols. A scatter plot of the unstaggered samples reveals only what is happening in one dimension, either the I or Q dimension.

For non-offset QPSK and QAM modulation schemes, use the *comms_instr* instrumentation block instead of the *offset_comms_instr* block.

Measuring RMS EVM

You can use the same instrumentation block (*offset_comms_instr*) to compute root-mean-squared error vector magnitude (RMS EVM). The error vector is the vectorial difference between the ideal received symbol and the actual received symbol.

- EVM (error vector magnitude) is the magnitude of the error vector.
- RMS EVM is the root-mean-squared value of a sequence of EVMs.

RMS EVM is one measure of a receiver's quality. RMS EVM can account for as much or as little distortion and noise as you like. The trick is to figure out where the ideal received symbol lies. You can do this using the *I_ref* and *Q_ref* inputs to the *offset_comms_instr* instrumentation block.

To calculate RMS EVM, you

- Create a duplicate copy of the receiver chain from the *BB_driver* to the *IQ_demod_BB* including these two blocks
- Place the duplicate copy below the original receiver chain in the Schematic window
- Modify the duplicate receiver chain to make it as *ideal* as you like by changing parameter values for the individual function blocks.

  For example, to see the effect of just the LNA's IP3 value on RMS EVM, in the duplicate receiver chain make the LNA's IP3 absurdly large.

Constructing the Ideal Receiver Chain

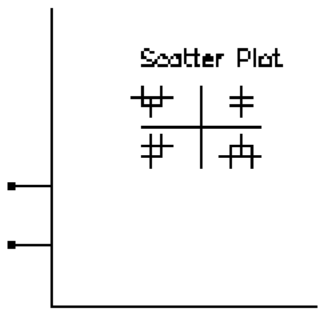
The *ideal* receiver chain (the duplicated and modified receiver chain) is driven from the same input, the *CDMA_reverse_xmit* block, as the original receiver chain. The output of the ideal receiver chain drives the instrumentation block's *I_ref* and *Q_ref* inputs.

In the ideal receiver chain, you copy the first receiver chain and make every block ideal.

1. Remove the *gnd* from the *I_ref* and *Q_ref* pins on the instrumentation block.

   a. In the Schematic window, choose *Edit - Delete*.

   b. In the Schematic window, click each wire and the *Gnd* symbol attached to the *I_ref* and *Q_ref* pins.



   c. Click *esc* to stop deleting.

2. Duplicate the receiver chain from the *BB_driver* to the *IQ_demod_BB* inclusive. Do not duplicate the filters. Follow the prompts at the bottom of the Schematic window.

   a. In the Schematic window, choose *Edit – Copy*.

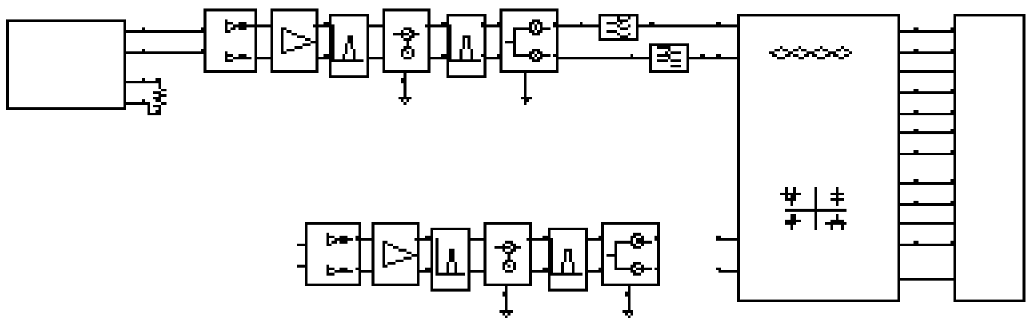
   b. In the Schematic window, draw a box around the blocks to copy by clicking to the left of and above the *BB_driver* block and dragging the cursor to a point below and to the right of the *IQ_demod_BB* block. Click again to complete the box.

      The blocks within the box are highlighted.

   c. Click within the highlighted area.

      A copy of the highlighted blocks in the receiver chain now moves with the cursor.

d. Place the duplicate receiver chain so that the output pins on the *IQ_demod_BB* are in line with the *I_ref* and *Q_ref* pins on the instrumentation block.



3. Wire the duplicate receiver chain to the CDMA signal source (*CDMA_reverse_xmit)* and the instrumentation block (*offset_comms_instr*).

a. In the Schematic window, choose *Add - Wire (narrow).*

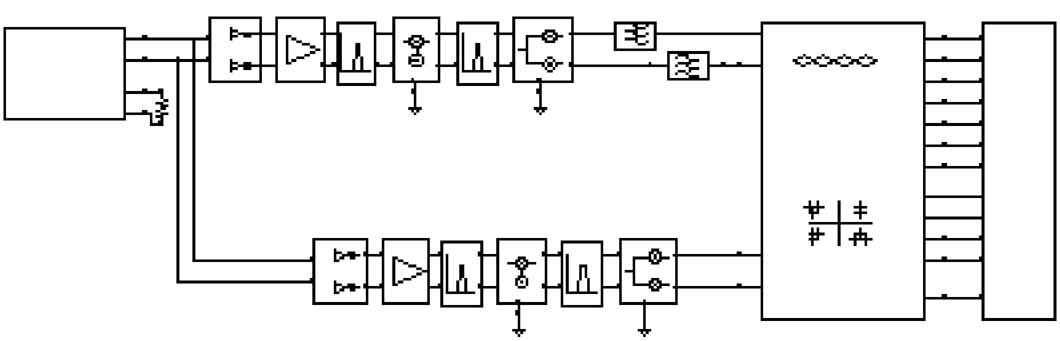
b. Connect the *IQ_demod_BB* outputs on the duplicate receiver to the *I_ref* and *Q_ref* pins on the instrumentation block.

c. Connect the output pins on *CDMA_reverse_xmit* to the input pins of the duplicate *BB_driver*. This drives the duplicate receiver from the CDMA signal source.

d. Click *esc* to stop wiring.

The schematic with the duplicate receiver chain wired up looks like Figure B-11.

Figure B-11 Receiver Model with Duplicated Receiver Chain



Modifying Parameter Values to Make the Blocks Ideal

Now modify the parameter values for each block in the duplicate receiver chain to create ideal blocks. Block names, parameter names, and parameter values are given in Table B-10.

Table B-10 Parameter Values to Create an Ideal Receiver

Block Names	Parameter Names	New Parameter Values
LNA_BB	*Input referred IP3 [dBm]* *{1,0,-1} for {cw, none, ccw}*	100 0
BB_butterworth_bp	*Cell Name*	BB_loss
dwn_cnvrt	*Input referred IP3 [dBm]*	100
BB_butterworth_bp	*Cell Name*	BB_loss
IQ_demod_BB	*I-[dBm] Input referred IP3* *Q-[dBm] Input referred IP3*	100 100

1. In the Schematic window, choose *Edit – Properties – Objects* to open the Edit Object Properties form.
2. In the Schematic window, select the *LNA_BB* block.

   The Edit Object Properties form changes to display properties for the *LNA_BB* block.

   a. Set *Input referred IP3 [dBm]* to 100.

   b. Set *{1, 0,-1} for {cw, none, ccw}* to 0. (This eliminates AM/PM conversion.)

   c. Click *Apply*.
3. In the Schematic window, select the first RF *BB_butterworth_bp* block.

   The Edit Object Properties form changes to display properties for the *BB_butterworth_bp* block.

   a. Change the *Cell Name* to *BB_loss*.

   b. Click *Apply*.

   The properties and symbol change to those for the *BB_loss* block. The sole purpose of the *BB_loss* model is to replace a filter in an RMS EVM analysis.

The *Reference impedance* for the *BB_loss* block should equal the *Output impedance* of the *BB_butterworth_bp* bandpass filter block it replaces. The value should be 50 ohms for both blocks and you should not have to change it.

The *BB_loss* model retains the filter's loss but eliminates the filter's dynamics so you can see what, if any, affect the filter has on EVM through inter-symbol interference. To eliminate the loss as well as the dynamics, you might even replace the filter with straight wires. This example uses the *BB_loss* block instead.

4. In the Schematic window, select the *dwn_cnvrt* block.

   The Edit Object Properties form changes to display properties for the *dwn_cnvrt* block

   a. Set *Input referred IP3 [dBm]* to 100.

   b. Click *Apply*.

5. In the Schematic window, select the second *BB_butterworth_bp* block.

   The Edit Object Properties form changes to display properties for the *BB_butterworth_bp* block

   a. Change the *Cell Name* to *BB_loss*.

   b. Click *Apply*.

6. In the Schematic window, select the *IQ_demod_BB* block.

   The Edit Object Properties form changes to display properties for the *IQ_demod_BB* block

   a. Set *I-[dBm] input referred IP3* to 100.

   b. Set *Q-[dBm] input referred IP3* to 100.

   c. Click *Apply*.

7. In the Edit Object Properties form, click *OK* to close the form.

8. In the Schematic window, choose *Design – Check and Save* to check and save your modifications to the circuit.

Set Up and Run a Transient Analysis

Set up and run a transient analysis as described in "Setting Up and Running a Transient Analysis" on page 2531. Set the *Stop Time* to `130u` and the *outputstart* option to `30u`. Click *OK* in both the Transient Options and Choosing Analyses forms. Choose *Simulation – Netlist and Run* to run the transient analysis.

Look for messages in the CIW stating that the simulation is starting. Watch the simulation log window for messages that the simulation has completed successfully.

Plot the RMS EVM Output

After the simulation, plot the RMS_EVM output of the instrumentation block.

1. In the ADE window, choose *Results – Direct Plot – Transient Signal*.

   This displays the waveform window.

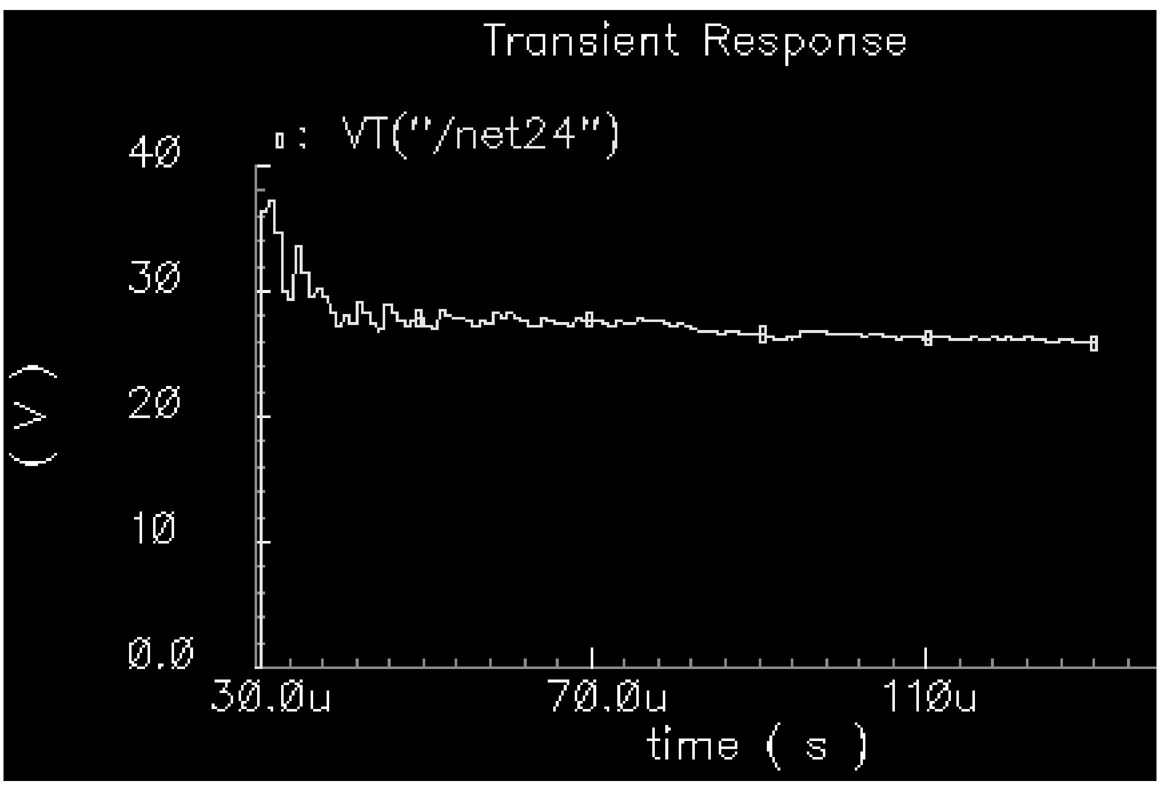
2. Following the prompts at the bottom of the waveform window,

   a. In the schematic, click the *rms_EVM* output net from the instrumentation (*offset_comms_instr)* block.

   b. Click *Esc* to indicate that you have finished selecting outputs.

   This creates the RMS EVM plot in the waveform window as shown in Figure B-12 on page 2544.

Figure B-12 RMS EVM



The RMS EVM trace starts at 30us, which is the `statistics start time` parameter of the instrumentation block. The `statistics start time` parameter keeps start-up transients out of the statistics.

The trace settles out at 25.84 Volts. This means that after 130us of data is collected, and ignoring the first 30us, the RMS EVM is 25.84%. The EVM measurement is normalized to the RMS magnitude of the ideal symbol then multiplied by 100 to express the measurement as a percentage.

3. In the waveform window, choose *File – Close*.

Computing Minimized RMS Noise Using the Optimizer

There is one more construction step before proceeding to the Circuit Optimizer application. You set up the Circuit Optimizer to minimize RMS noise subject to performance constraints. This step replicates the receiver chain yet one more time to generate the noise measurement.

1. Duplicate the original receiver chain from the *BB_driver* up to and including both low pass filters (*butterworth_lp*). Follow the prompts at the bottom of the Schematic window.

   a. In the Schematic window, choose *Edit – Copy*.

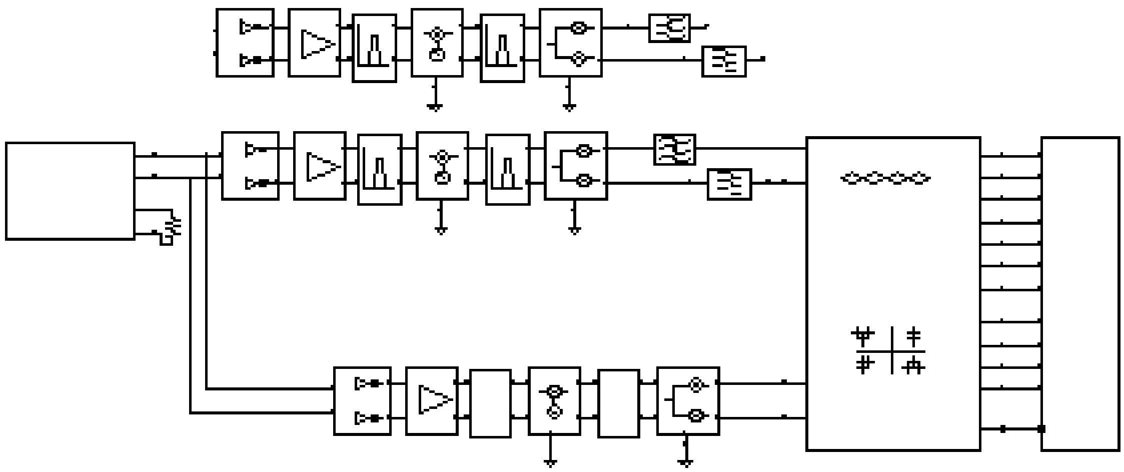
   b. In the Schematic window, draw a box around the blocks to copy by clicking to the left of and above the *BB_driver* block and dragging the cursor to a point below and to the right of the *butterworth_lp* filter blocks. Click again to complete the box.

      The blocks within the box are highlighted.

   c. Click within the highlighted area.

      A copy of the highlighted blocks in the receiver chain now moves with the cursor.

d. Place the duplicate receiver chain above the original receiver chain.



2. In the duplicate receiver chain, ground the *Q_in* pin on the *BB_driver* block.

a. In the Schematic window, choose *Add – Wire.*

b. Click the *Q_in* pin and run the wire to the *Gnd* symbol below the *dwn_cnvrt* block.

c. Click the *Gnd* symbol below the *dwn_cnvrt* block.

d. Click *esc* to stop wiring.

3. Add a 50mV DC voltage source to the left of the *BB_driver* block to drive the *I_in* pin on the *BB_driver* block. At the same time add a *gnd* symbol below the *port* in the schematic.

a. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

b. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

c. In the Library Browser - Add Instance form, click *analogLib*.

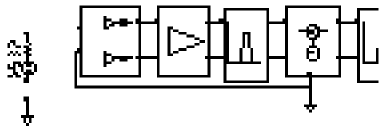
d. Scroll the elements in the *Cell* column and click *port*.

e. The outline for the *port* symbol is attached to the cursor. Move the *port* symbol to the left of the *BB_driver* block and click to place the *port* symbol.

f. Return to the Library Browser - Add Instance form and scroll the elements in the *Cell* column and click *gnd*.

g. The outline for the *gnd* symbol is attached to the cursor. Move the *gnd* symbol below the *port* symbol and click to place it there.

h. Click *Esc* to remove the *gnd* symbol from the cursor.



4. In the Schematic window, choose *Edit – Object – Properties* to modify the *port* symbol using the Edit Object Properties form.

a. In the Schematic window, click the *port* symbol.

The Edit Object Properties form changes to display information for the *port* symbol.

b. In the *Source Type* cyclic field, select *dc*.

c. In the *DC Voltage* field, enter 50m.

d. Highlight *Display small signal params* to display small signal parameters.

e. In the *AC Magnitude* field type `1V`.

f. In the *AC Phase* field type `0`.

g. Click *OK* in the Edit Object Properties form.

5. Load the low pass filters with a *res_BB* model from the *top_dwnBB* category of *rfLib*. Use the default parameters and ground the output pins.

a. In the Schematic window, choose *Add – Instance* to display the Add Instance form.

b. In the Add Instance form, click *Browse* to display the Library Browser - Add Instance form.

c. In the Library Browser - Add Instance form, click *rfLib*.

d. Scroll the elements in the *Cell* column and click *res_BB*.

The outline for the *res_BB* symbol is attached to the cursor.

e. Move the *res_BB* symbol to the right of the two low pass filters (*butterworth_lp*) and click to place the *res_BB* symbol.

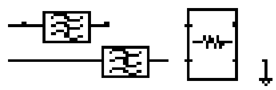
f. Return to the Library Browser - Add Instance form and click *analogLib.*

g. Scroll the elements in the *Cell* column and click *gnd*.

The outline for the *gnd* symbol is attached to the cursor.

h. Move the *gnd* symbol to the right of the *res_BB* symbol and click to place it there.

i. Click *Esc* to remove the symbol from the cursor.



6. In the duplicate receiver chain, wire the *port*, the *res_BB* block, and their *gnd* blocks.

a. In the Schematic window, choose *Add – Wire.*

b. Click the *I_in* pin on the *BB_driver*, then click the top pin on the *port*.

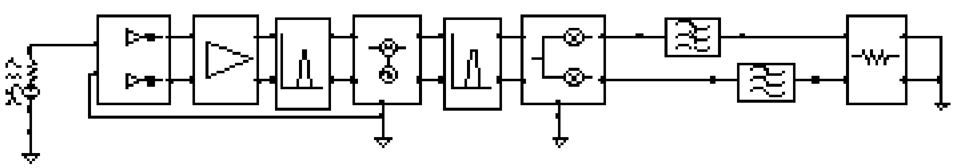
c. Click the bottom pin on the *port*, then click the *gnd* pin just below it.

d. Click the *out* pin on the top *butterworth_lp* filter, then click the *I_in* pin on the *res_BB* block.

e. Click the *out* pin on the lower *butterworth_lp* filter, then click the Q_*in* pin on the *res_BB* block.

f. Click the *I_out* pin on the *res_BB* block. Then click the top pin on the *gnd* located to it's right.

g. Click the Q_*out* pin on the *res_BB* block. Then click the top pin on the same *gnd*.

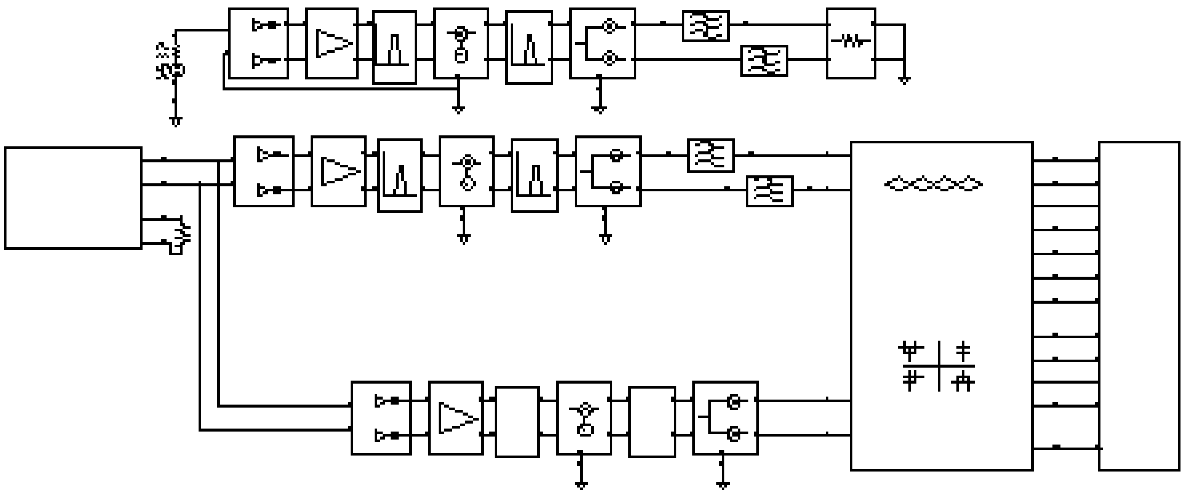


h. Click *esc* to stop wiring.

7. In the Schematic window, choose *Design – Check and Save* to check and save the schematic.

The schematic with the third receiver chain is shown in Figure B-13.

Figure B-13 Schematic with Noise Generating Receiver



Set Up and Run Transient and Noise Analyses

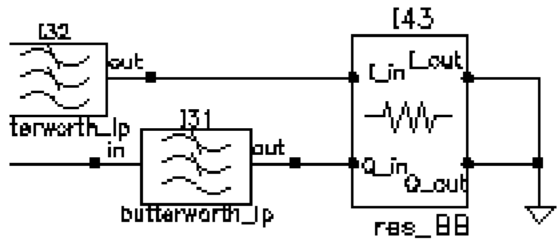
Set up a transient analysis as described in "Setting Up and Running a Transient Analysis" on page 2531. Set the *Stop Time* to `130u` and the *outputstart* option to `30u` and make sure that the transient analysis is enabled.

Set up a noise analysis as follows:

1. In the Choosing Analyses form, click *noise* to select a noise analysis.
2. For *Sweep Variable*, click *Frequency*.
3. For *Sweep Range*, click *Start-Stop*.
4. Set up the analysis to sweep frequency from 0 to100 MHz.
   a. For the starting frequency, in the *Start* field enter `0`.
   b. For the stop frequency, in the *Stop* field enter `100M`.
5. Set up the *Output Noise* source.

a. In the *Output Noise* cyclic field, select *voltage.*

b. To select the *Positive Output Node*, click *Select* next to the *Positive Output Node* field. Then, in the Schematic window, click the net next to the *I_in* pin on the *res_BB* block.

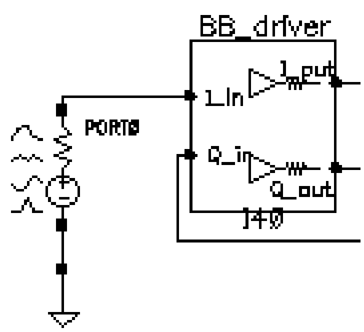


c. To select the *Negative Output Node*, click *Select* next to the *Negative Output Node* field. Then, in the Schematic window, click the net next to the *I_out* pin on the *res_BB* block.

6. Set up the *Input Noise* source.

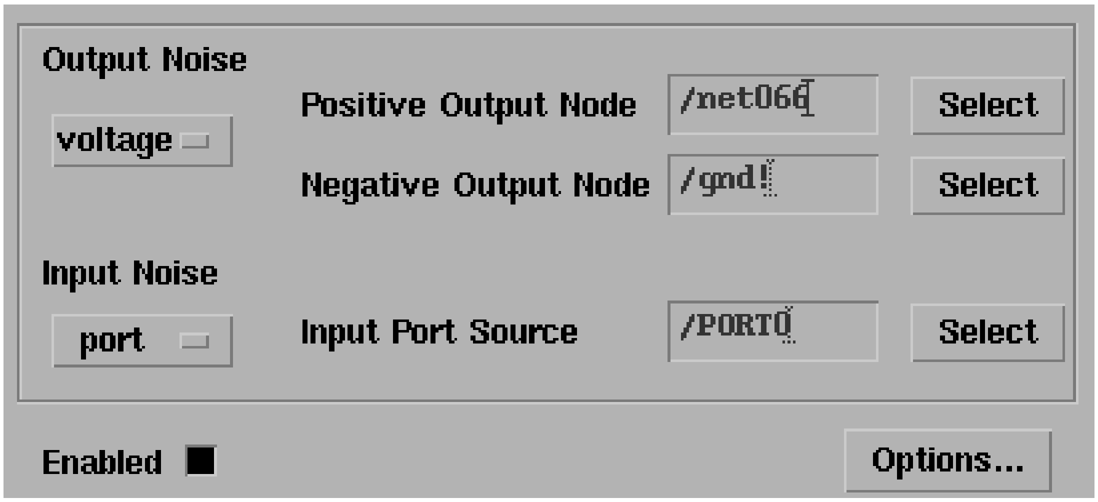
a. In the *Input Noise* cyclic field, select *port.*

b. To select the *Input Port Source*, click *Select* next to the *Input Port Source* field. Then, in the Schematic window, click the *DC input port* model.



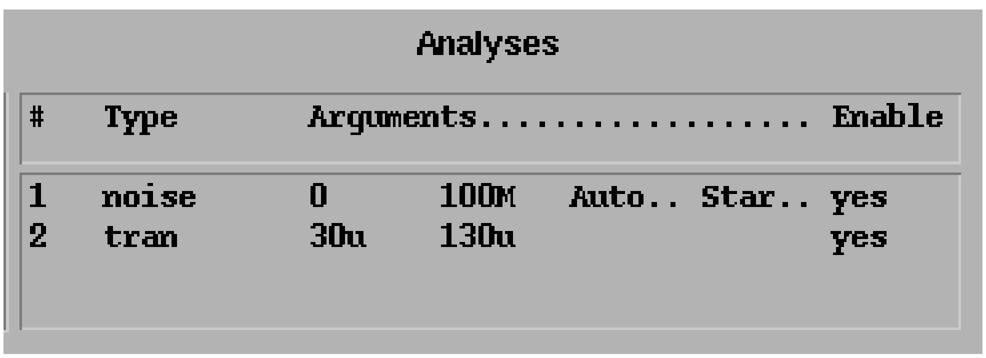
7. Click *Enabled*.

The *Output Noise* and *Input Noise* sections of the Noise analysis form look as follows. (The net instance numbers might be different, however.)



8. Verify that *Enabled* is highlighted and click *OK* in the Choosing Analyses form.

9. In the ADE window, check the Analysis area to verify that both the transient and noise analyses are set up properly and that they are both enabled.

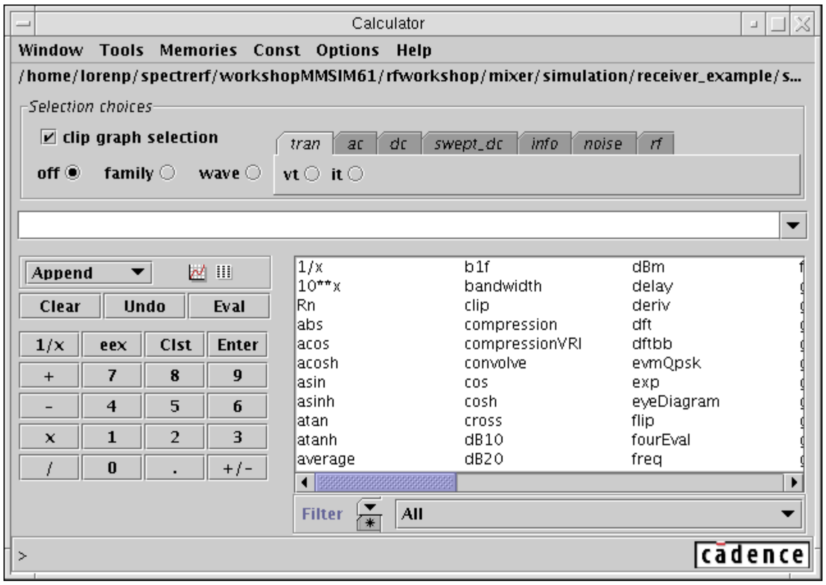


10. In the ADE window, choose *Simulation – Netlist and Run* to start the simulations.

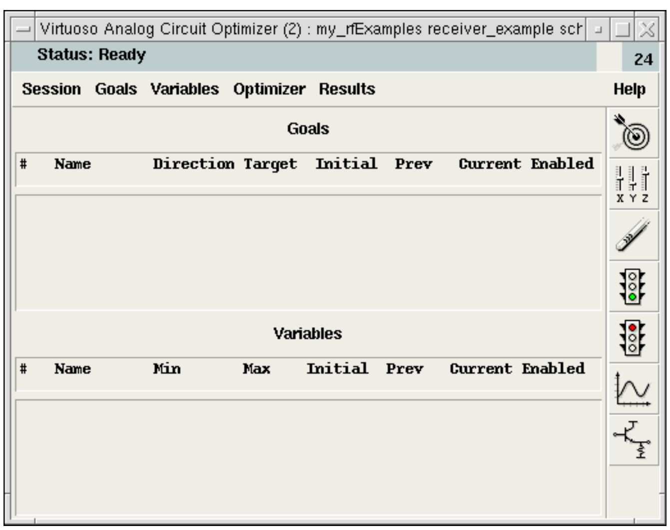
Watch the messages in the CIW to verify that everything is set up properly and that the simulations start. Check the simulation log window to see that the simulations run and complete properly.

Set Up to Run the Circuit Optimizer

1. In the ADE window, choose *Tools – Calculator* to open the calculator window.



2. In the ADE window, choose *Tools – Optimization* to open the Circuit Optimizer window.



Add the First Goal

1. In the calculator window's function panel, select *rmsNoise*.

2. When the RMS Noise panel opens,

a. In the *From* field, enter `0`.

b. In the *To* field, enter `100M`.

c. Click *OK*.

The expression

`rmsNoise(0,100M)`

displays in the buffer of the calculator window.

3. In the Circuit Optimizer window, choose *Goals – Add* to add the first goal.

The Adding Goals form opens.

4. Fill in the Adding Goals form.

a. In the *Name* field, type `rmsNoise`.

b. Click the *Get Expression* button to the right of the *Calculator* label.

The expression `rmsNoise(0 100000000)` displays in the *Expression* field.

c. Ensure that the *Direction* is minimize.

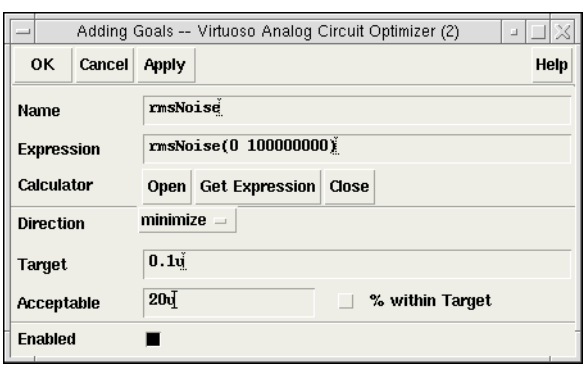
The default for the Optimizer is to minimize goals.

d. In the *Target* field, enter `0.1u`.

e. In the *Acceptable* field, enter `20u`.

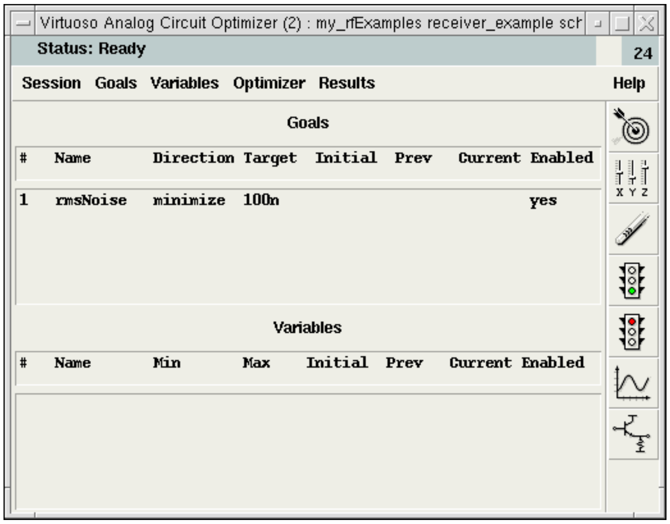
f. Verify that the form is enabled.

The Adding Goals form looks like this.



g. Click *OK*.

The first goal is added to the Circuit Optimizer window.



Add the Second Goal

1. In the calculator window, under the *tran* tab, click the *vt* button.

   The Schematic Editing window becomes active.

2. In the Schematic window, click the *rms_EVM* output net of the instrumentation block.

   An expression similar to `VT("/net24")` displays in the calculator buffer.

3. In the calculator window's function panel, select *value*.

   When the Value panel appears

   a. Enter `130u` in the *Interpolate At* field.

   b. Click *OK*.

      The expanded expression `value(VT("/net24") 130u )` displays in the buffer of the calculator window.

4. In the Circuit Optimizer, choose *Goals – Add*.

   When the Adding Goals form opens

   a. In the *Name* field, enter `evm`.

   b. Click the *Get Expression* button to the right of the *Calculator* label.

      An expression similar to

`value(VT("/net24") 0.00013)`

displays in the *Expression* field.

c. In the *Direction* cyclic field, select <=

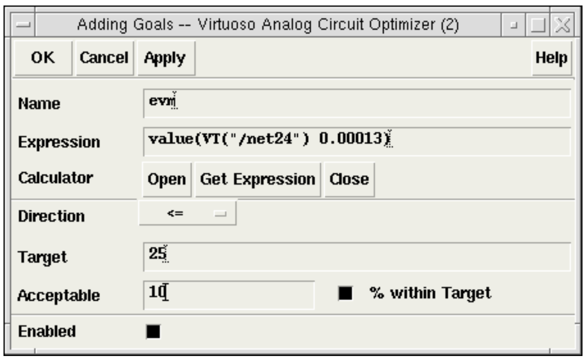
d. In the *Target* field, enter `25`.

e. In the *Acceptable* field, enter `10`.

f. Click the *% within Target* button.

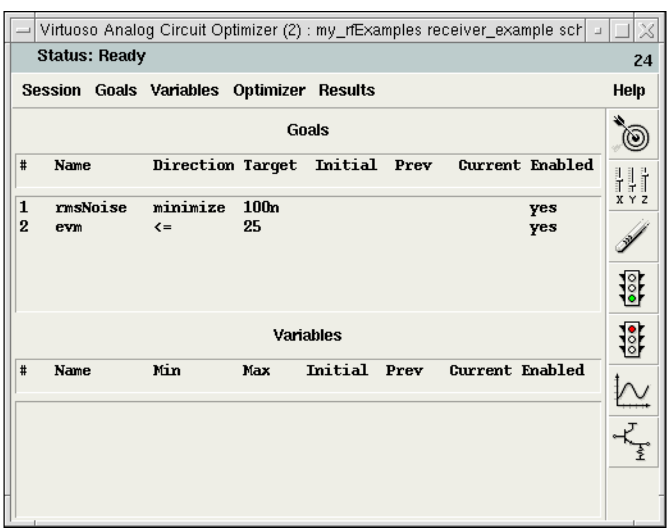
g. Verify that *Enabled* is active.

The Adding Goals form looks like this.



h. Click *OK*.

The second goal is added to the Circuit Optimizer window.



Add the Third Goal

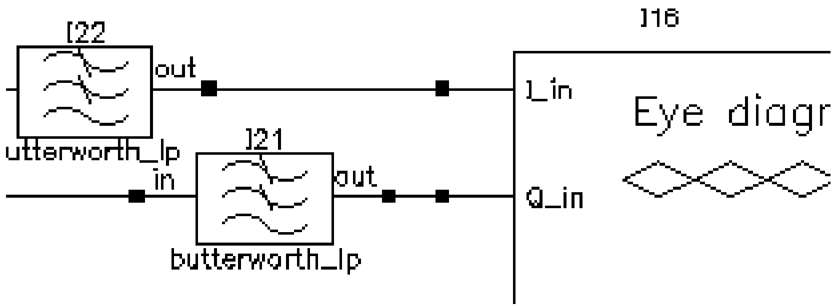
1. In the calculator window, click *Clear*.

   The buffer is cleared.

2. Under the *tran* tab, click the *vt* button.

   The Schematic Editing window becomes active.

3. In the Schematic window, click the *I_in* pin of the instrumentation block.



   An expression similar to `VT("/net14")` displays in the buffer of the calculator window.

4. In the calculator window's function panel, select *rms*.

   An expanded expression similar to

   `rms(VT("/net14"))`

   displays in the buffer of the calculator window.

   The objective is to keep the *rms* value of this signal level above 300 mV. Note that all goals must be scalars.

5. In the Circuit Optimizer, choose *Goals - Add*.

   The Adding Goals form opens.

6. Fill in the fields of the Adding Goals form.

   a. In the *Name* field, enter `sig_level`.

   b. Click the *Get Expression* button to the right of the *Calculator* label.

      And expression similar to

      `rms(VT("/net14"))`

      displays in the *Expression* field.

c. In the *Direction* cyclic field, select >=

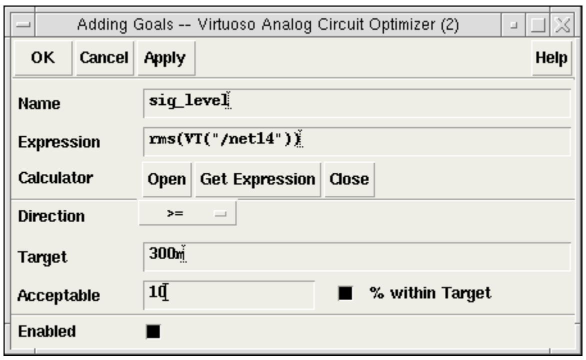
d. In the *Target* field, enter `300m`.

e. In the *Acceptable* field, enter `10`.

f. Click the *% within Target* button.

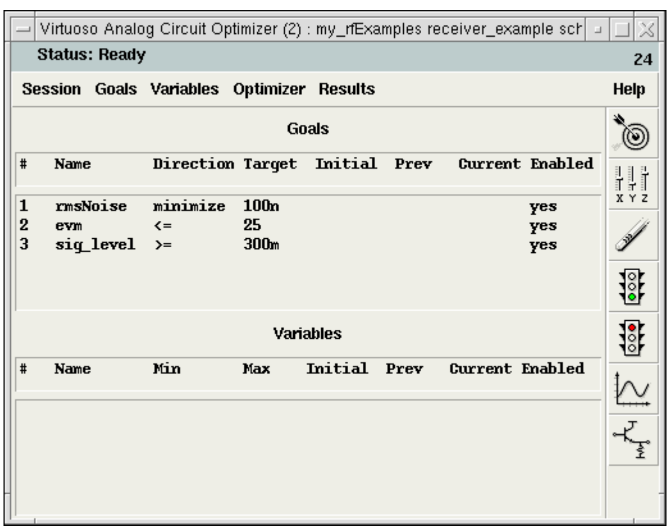
g. Verify that the *Enabled* button is active.

The Adding Goals form looks like this.



h. Click *OK*.

The third goal is added to the Circuit Optimizer window.



Add the Circuit Variables to the Optimizer

Add the variables to the Circuit Optimizer window.

1. In the Circuit Optimizer window, choose *Variables – Add/Edit*.

2. When the Editing Variables form opens

a. In the *Name* list box, click the *lna_ip3* variable.

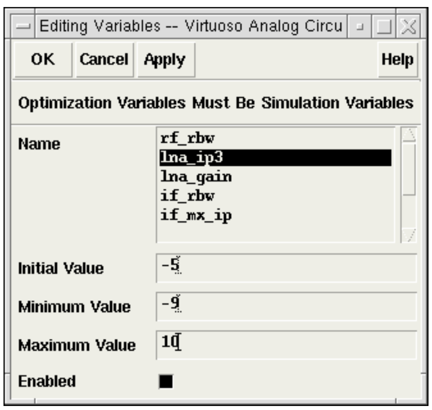
The *lna_ip3* variable is highlighted in the list box and it's current value `-5` displays in the *Initial Value* field.

b. In the *Minimum Value* field, enter `-9`.

c. In the *Maximum Value* field, enter `10`.

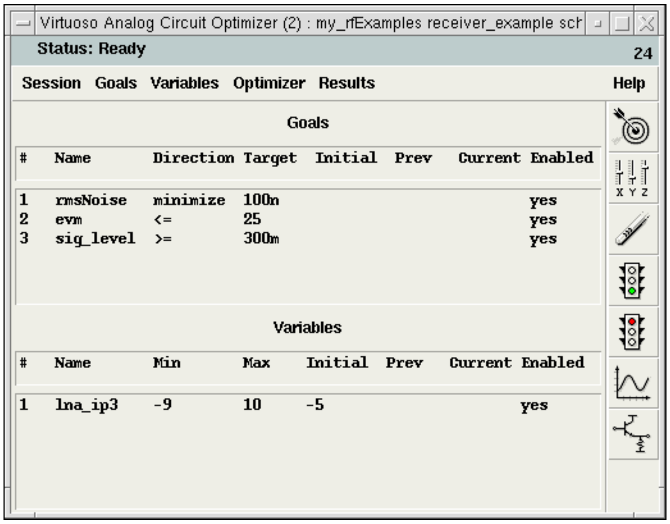
d. If necessary, click *Enabled*.

The Editing Variables form appears as follows.



e. Click *Apply*.

Information for the *lna_ip3* variable displays in the *Variables* section of the Circuit Optimizer.



3. Repeat this procedure to add all the variables and values listed in Table B-11

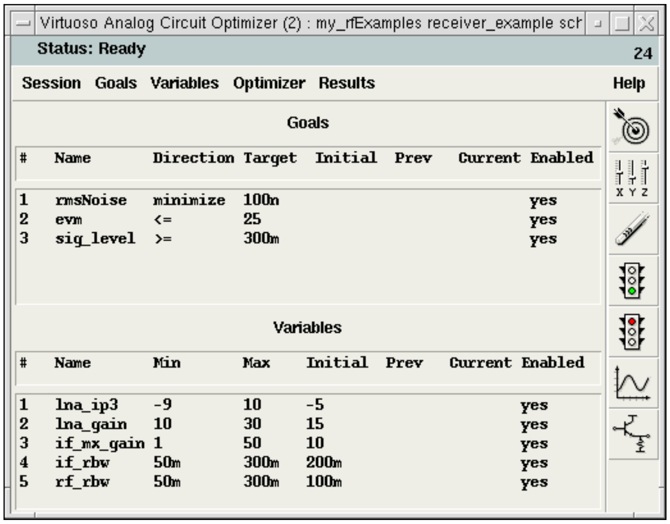
Table B-11 Variables and Values for the Optimizer

Variable Name	Initial Value	Minimum Value	Maximum Value
`lna_ip3`	-5	-9	10
`lna_gain`	15	10	30
`if_mx_gain`	10	1	50
`if_rbw`	200m	50m	300m
`rf_rbw`	100m	50m	300m

4. Click *OK* to close the Editing Variables form.

In the Circuit Optimizer window, the variables and optimization goals appear as shown in Figure B-14 on page 2560.

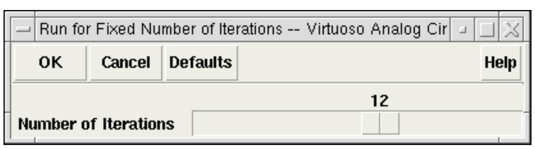
Figure B-14 Circuit Optimizer Setup



Run the Circuit Optimizer

Note: This Circuit Optimizer analysis might take several hours to complete.

1. In the Circuit Optimizer, choose *Optimizer – Run n* to display the *Run for Fixed Number of Iterations* form.
2. In the *Number of Iterations* field, move the slider to the right until *12* displays.



3. Click *OK.*

   The *Run for Fixed Number of Iterations* form closes and the Circuit Optimizer starts running.

   Watch for simulator startup messages in the CIW. Monitor the progress of the analysis in the log window.

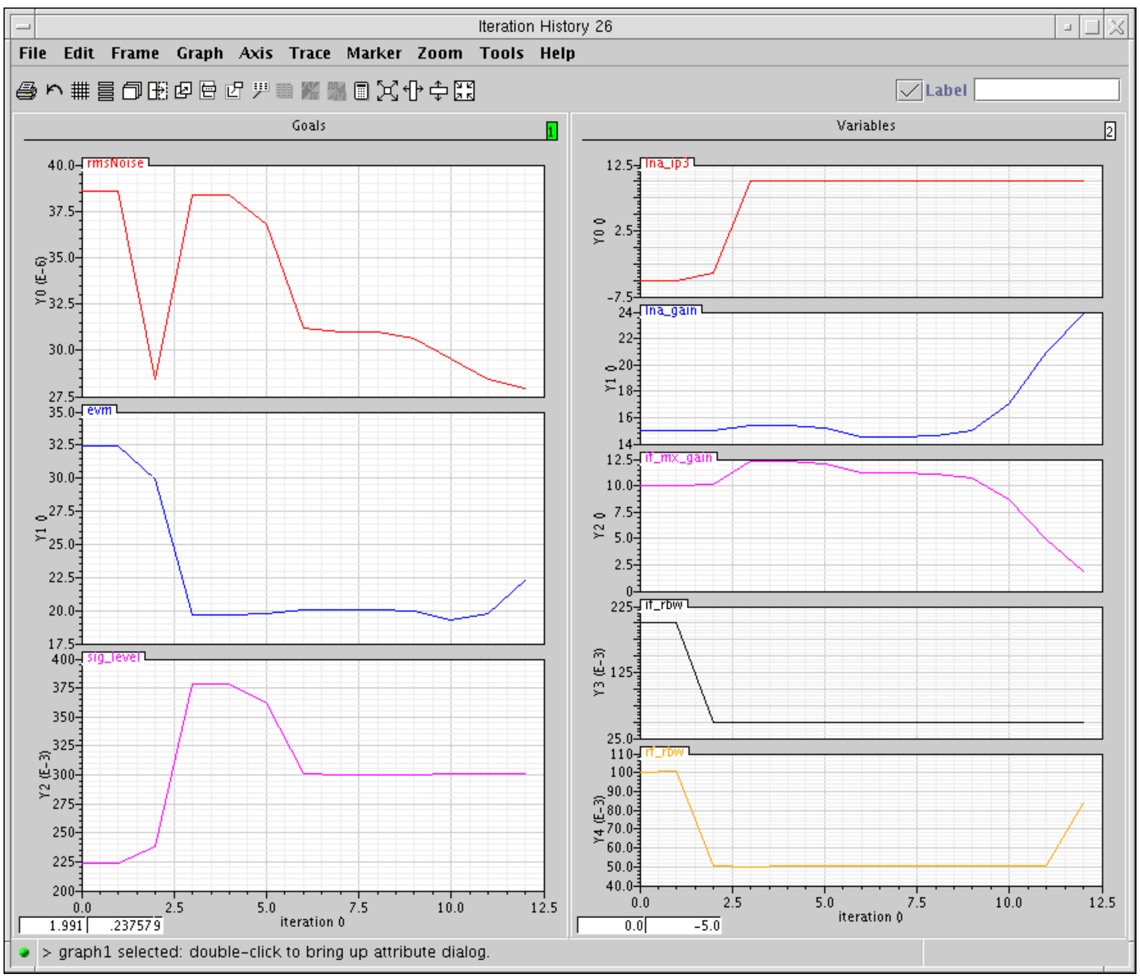
   The waveform window opens when the first simulation completes. As simulations complete, the results are added to the plots in the waveform window.

Viewing the Circuit Optimizer Output

The Circuit Optimizer results displayed in the waveform window are shown in Figure B-15 on page 2561.

- The traces on the left show the EVM and RMS output signal level, and the RMS noise at each Circuit Optimizer iteration.
- The traces on the right show the variable function block specifications at each Circuit Optimizer iteration.

Figure B-15 Circuit Optimizer Results After 12 Iterations



Although the example is contrived, it illustrates the use model. After the Circuit Optimizer met the constraints it tried to minimize RMS noise.

1. Save the initial state of the Analog Design Environment in case you want to start over.

2. Then in the Circuit Optimizer window, click *Results – Update Design*. The last click updates the variables in the Analog Design Environment window with the last set of variables found by the Circuit Optimizer. You use these states in the passband view.

Summarizing the Design Procedure

To summarize, the semi-automated design procedure consists of

1. Setting up the measurements
2. Placing tolerances on the block parameters
3. Constraining the system performance
4. Identifying a quantity to minimize (or maximize)
5. Running the Optimizer
6. Evaluating the results

This is why the process is called *semi*-automated. After evaluating results for the first or second time you probably need to

1. Refine tolerances
2. Refine goals
3. Add or delete constraints
4. Add or delete variables

Each simulation covers 100s, or about 80 CDMA symbols. The suppressed carriers are an RF carrier at 2.14 GHz, an LO carrier at 2.354GHz, and an IF carrier at 214 MHz. The symbol rate is 1.2288 Mega-symbols per second.

Creating a Passband View of the Architectural Model

After you have designed an architecture, you can quickly create a passband view of the architectural model. (Currently, the passband behavioral models in the *top_dwnPB* category and in the passband view do not introduce any specifications that are not in the baseband models.)

The passband view checks for problems that might have escaped detection in the baseband view. For example, although the baseband view quickly assesses what filters do to the baseband signal, baseband models do not indicate whether the filters are indeed removing undesired carrier harmonics.

Baseband modeling is also not the best way to evaluate image rejection. Although the baseband model accurately simulates how the desired signal propagates through an image rejection receiver, it does not accurately simulate how much of the image signal propagates to the receiver output.

The passband view also creates a system testbench as mentioned in “Top-Down Design of RF Systems” on page 2466.

Procedures for Creating the Passband Model of the Receiver

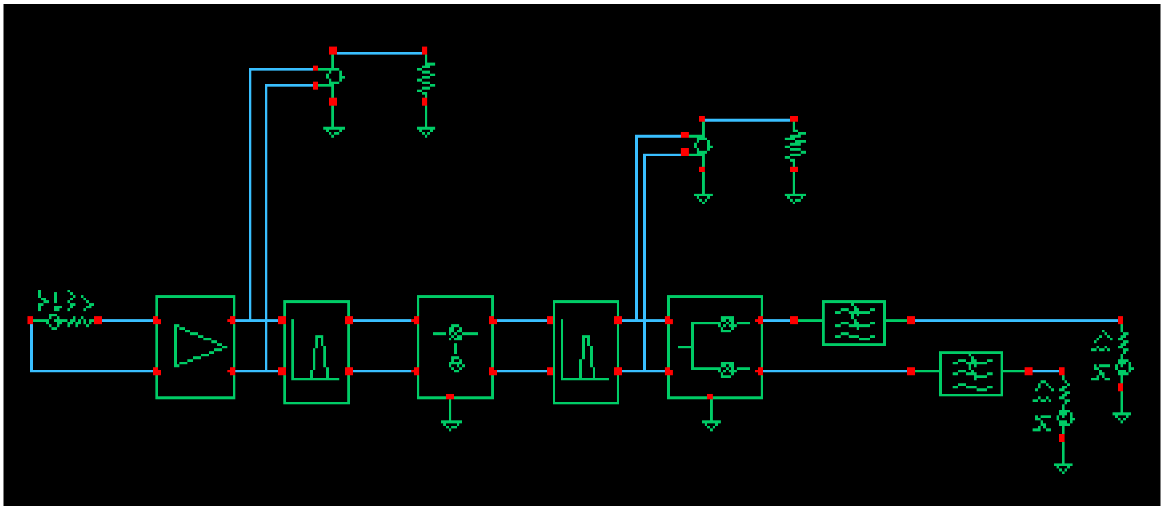
The procedures described in this section illustrate how to

- Switch from a baseband to a passband view
- Make an end-to-end RF measurement
- Measure the one dB compression point

  The one dB compression point is usually a transmitter specification but it is used to demonstrate this flow because it is easier to set up.

1. Copy the original receiver model from the Circuit Optimizer analysis to a new schematic window. Copy everything from the LNA to the low pass filters.
2. Edit the properties of the IQ_demodulator (`IQ_demod_BB`) to set the last parameter, *flo*, to *-frf+flo1*. The baseband view does not need the local oscillator frequency but the passband view does.
3. Load the low pass filters with ports.
4. Connect a port across the LNA_BB inputs. Set the Frequency name to `fin`, the frequency to `frf`, and the amplitude to `power`. (Do not abbreviate `power` to `pwr` because `pwr` is a reserved variable and you do not get any warning. SpectreRF may complain about a mysterious indexed undefined variable that increments from run to run.)
5. Add loaded voltage-controlled-voltage sources as shown in Figure B-16 on page 2564 to observe intermediate differential voltages.

Figure B-16 Passband View



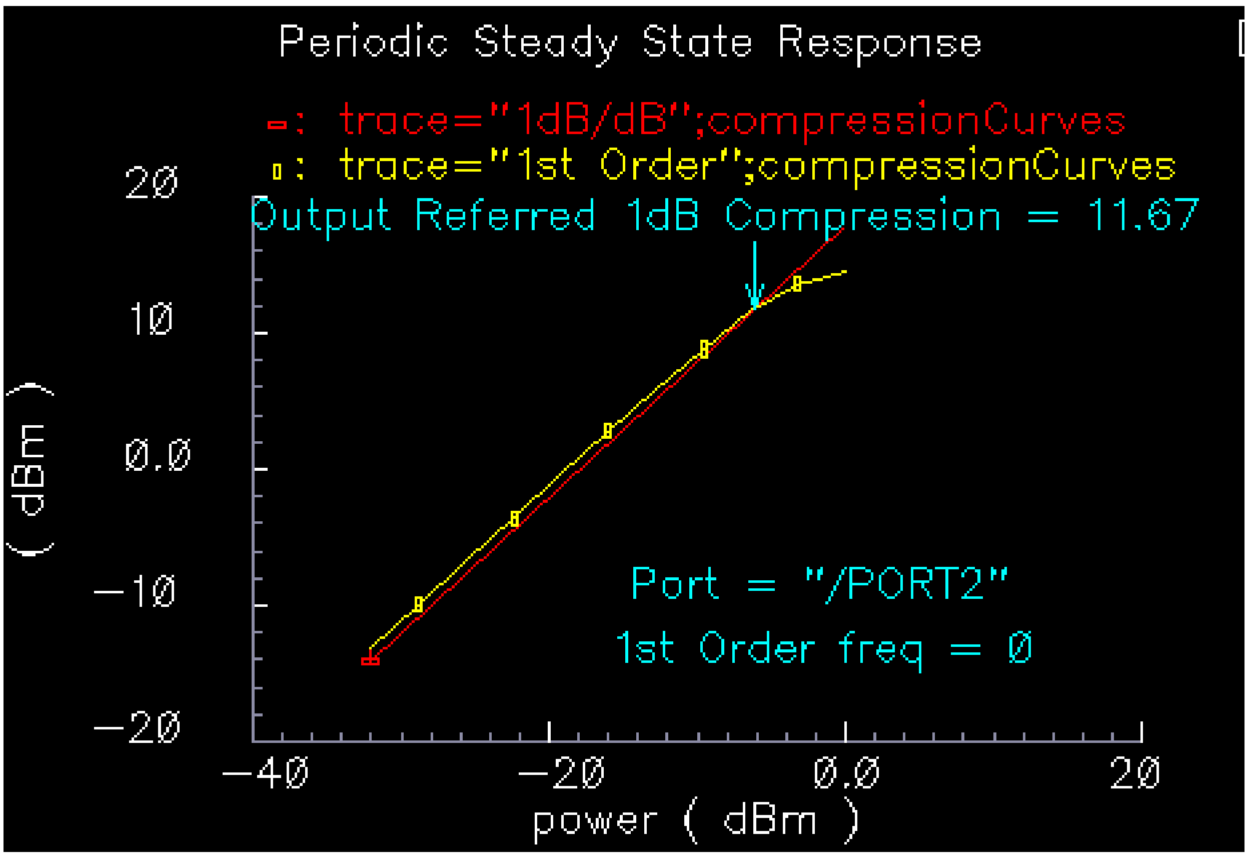
6. Check and save the schematic
7. Close the schematic window.
8. Bring up a Library Manager and select but do not open the schematic you just created.
9. In the Library Manager window click *File – New – Cell View*.
10. In the window that comes up, select Hierarchy Editor for the Tool. Click *OK*.
11. Type in `schematic` for the View then click *Use Template*.
12. Set the *Name* to `Spectre`.
13. Enter `veriloga_PB` as the first item in the View List, then click *OK*.
14. In the Hierarchy Editor window, click *File – Save*. This is important.
15. Click the *Open* button to bring up the schematic.
16. Bring up an Analog Design Environment tool and use the states from the last Circuit Optimizer iteration.
17. Recall the states you saved from the last Circuit Optimizer iteration. Add the `power` variable and set it to -16.
18. Delete the previous analyses and set up a PSS analysis. Enter a new Fundamental Tone named `LO` and make it 2.354GHz. *Auto Calculate* the *Beat Frequency* and type `1` for the *Number of harmonics*.
19. Run the analysis and observe the intermediate differential voltages. The model is indeed now a passband model. At the higher power levels the LNA output contains odd

harmonics of the RF carrier. The filter reflects the odd harmonics back to the LNA and does not let them propagate forward. The baseband model does not simulate the odd harmonics but it does simulate the intermodulation term between the second harmonic and fundamental that falls at the fundamental. One reason to simulate the passband view is to check for peak voltage levels that might exceed voltage ratings. The baseband models only simulate peak voltage at the carrier fundamental, not the absolute peak.

20. Set up a swept PSS analysis. Sweep *power* from -32 to 0 in 10 steps.

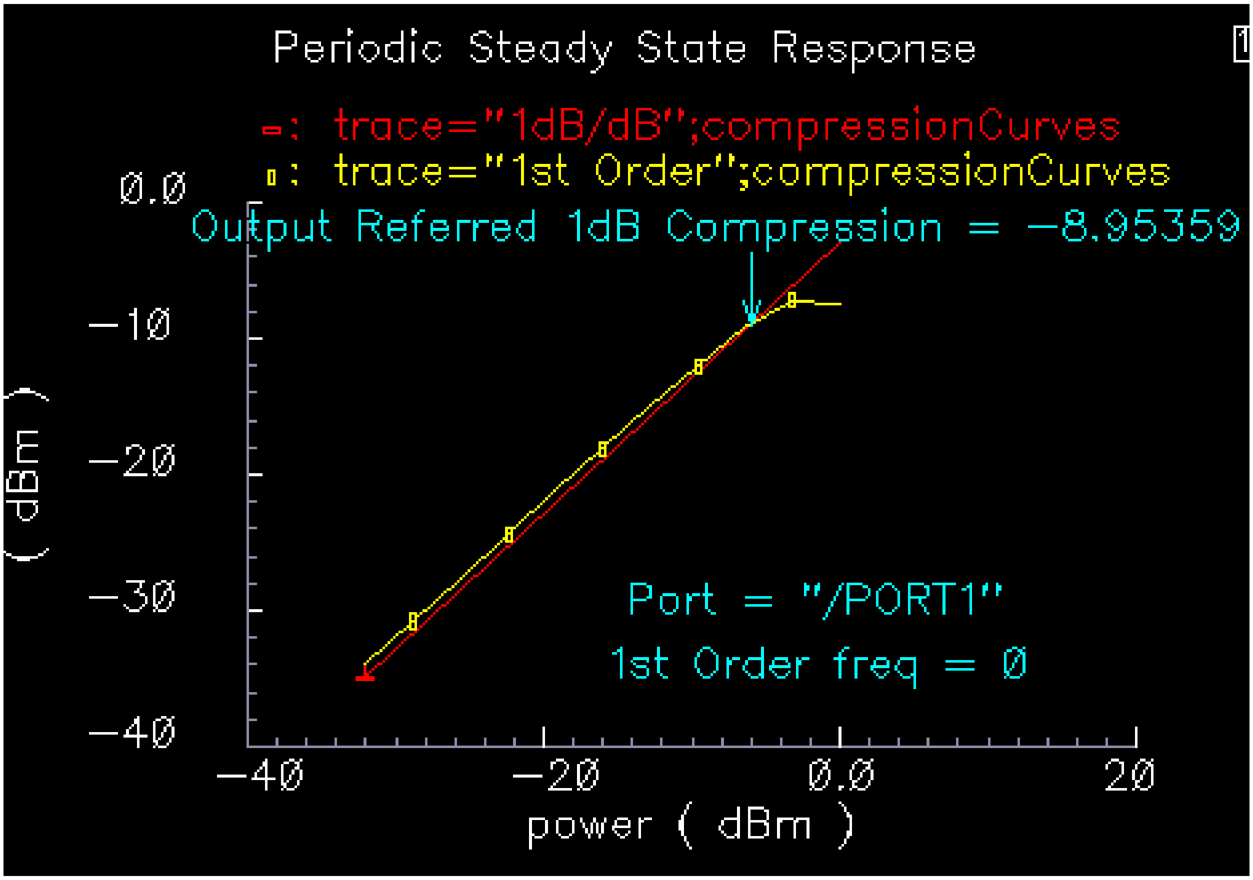
21. After the sweep finishes, click *Results – Direct Plot* in the Environment window, select *Compression point*, *1dB*, *Output Referred*. Select the 0 harmonic because the end-to-end system produces a baseband output. Then click the port loading the top output low pass filter. You should see the compression point plot shown in Figure B-17 on page 2565.

Figure B-17 End-to-end RF measurement, one dB compression point at the I-baseband output.



22. Repeat the last step but this time click the lower output port. You should see the compression plot in Figure B-18 on page 2566.

Figure B-18 End-to-end RF measurement, one dB compression point at the Q-baseband output.



Comparing Baseband and Passband Models

This section illustrates how to compare baseband and passband models by:

1. Setting up a Transient analysis with the passband view
2. Setting up a Transient analysis with the baseband view
3. Directly comparing the baseband and passband models.

You run one analysis of the baseband view and two analyses of the passband view. You perform the second passband analysis with tightened tolerances.

1. Save the passband schematic under a different name. You use the new copy.
2. Repeat steps 9 through 17 from the last recipe for the new copy but do not enter the `veriloga_PB` view in the *View List* yet. You do a baseband analysis first.

3. Delete the port driving the LNA.
4. Delete the loaded voltage-controlled-voltage sources.
5. You need to synthesize an antenna signal. Add an `IQ_mod_BB` from the `top_dwnBB` category. Set the I and Q gains to 0 dB. Set the 1dB compression points to 1000 so that the modulator is ideal. Instantiate it in front of the LNA with the pins aligned, then wire the pins straight across. Ground the `phase_err` pin.
6. Drive the `I_in` pin of the IQ modulator with a port. Set the port frequency to 2MHz and name the frequency BB1. Set the amplitude to -16dBm.
7. Do the same for the `Q_in` modulator input.
8. Load or duplicate the states from the 12-iteration Circuit Optimizer analysis but delete the Noise and Transient analyses.
9. Remove AM/PM conversion from the LNA by setting the last parameter in the properties list to zero.

   It is not fair to compare passband and baseband views with AM/PM conversion because the passband view does not capture it.
10. Set up a 1us Transient analysis with default options.
11. Run the analysis and plot the filtered baseband outputs, the outputs of the low pass filters.

    Note how fast the simulation runs. Save the results so you can plot them again later. [1]
12. Switch to the passband view by entering “veriloga_PB” in the View List in the Hierarchy Editor. Click the update button in the hierarchy editor.
13. After you switch to the `veriloga_PB` view, edit the IQ modulator properties to set *flo* to `frf`. Edit the demodulator properties and set its *flo* to `flo1-frf`.
14. Click *Results – Printing Plotting Options* then click the *Overlay Plots* button.
15. Overlay the passband results with the baseband results. You see the waveforms in Figure B-19 on page 2568. The comparison is not very good.
16. Rerun the analysis with conservative options and set reltol to 1e-6. This run takes longer.

---

1. Note that the baseband outputs are out of phase with each other, even though the baseband inputs are in phase. In the baseband model, changing the RF-IF mixer LO from “flo1” to “-flo1” fixes the sign problem. In the passband model, the IQ_demodulator flo should be frf-flo1. To maintain the convention, in the baseband model the IF filter’s carrier frequency should be frf-flo1.

17. Plot the results.

18. Recall the saved baseband results and overlay them with those from the last simulation. You see the waveforms in Figure B-20 on page 2569. The passband results now lay right on top of the baseband results but took much longer to compute! It was not obvious without the baseband results that the first passband simulation did not run with tight enough numerical tolerances.

Figure B-19 Passband and baseband results with default options in the passband analysis

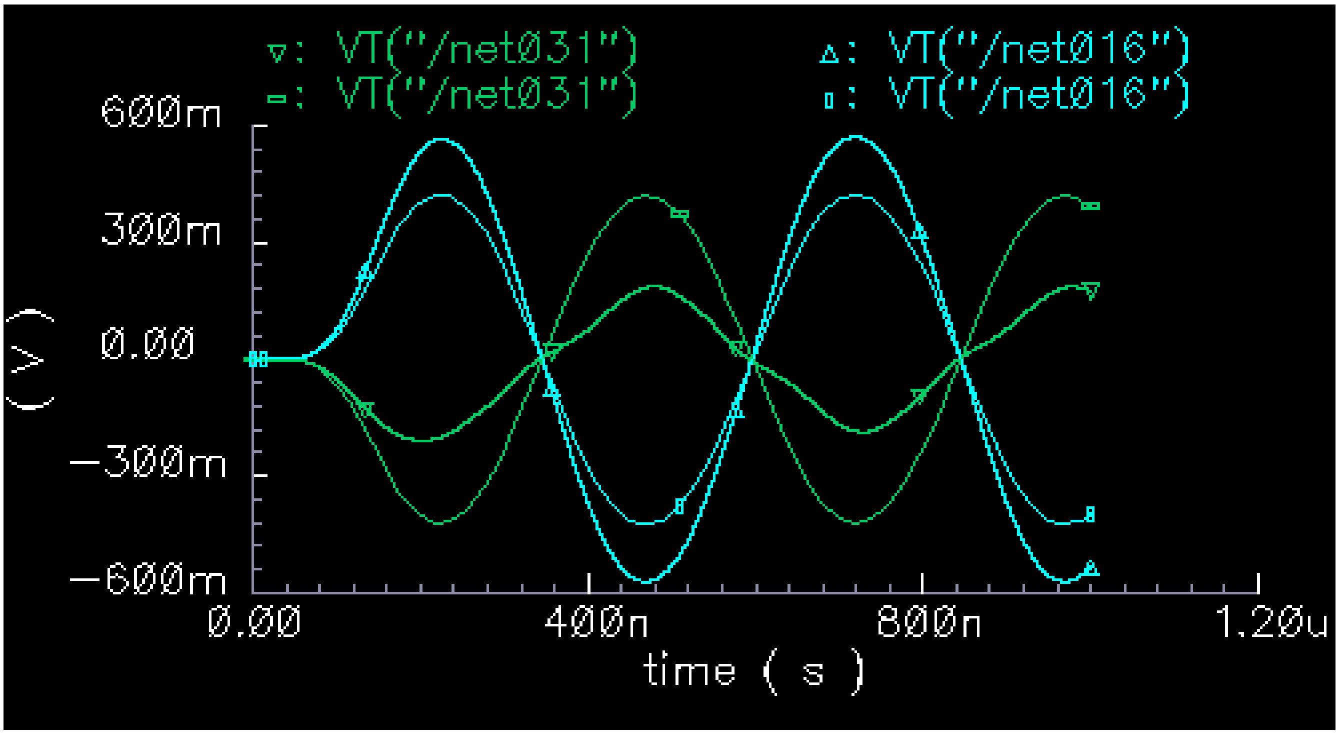
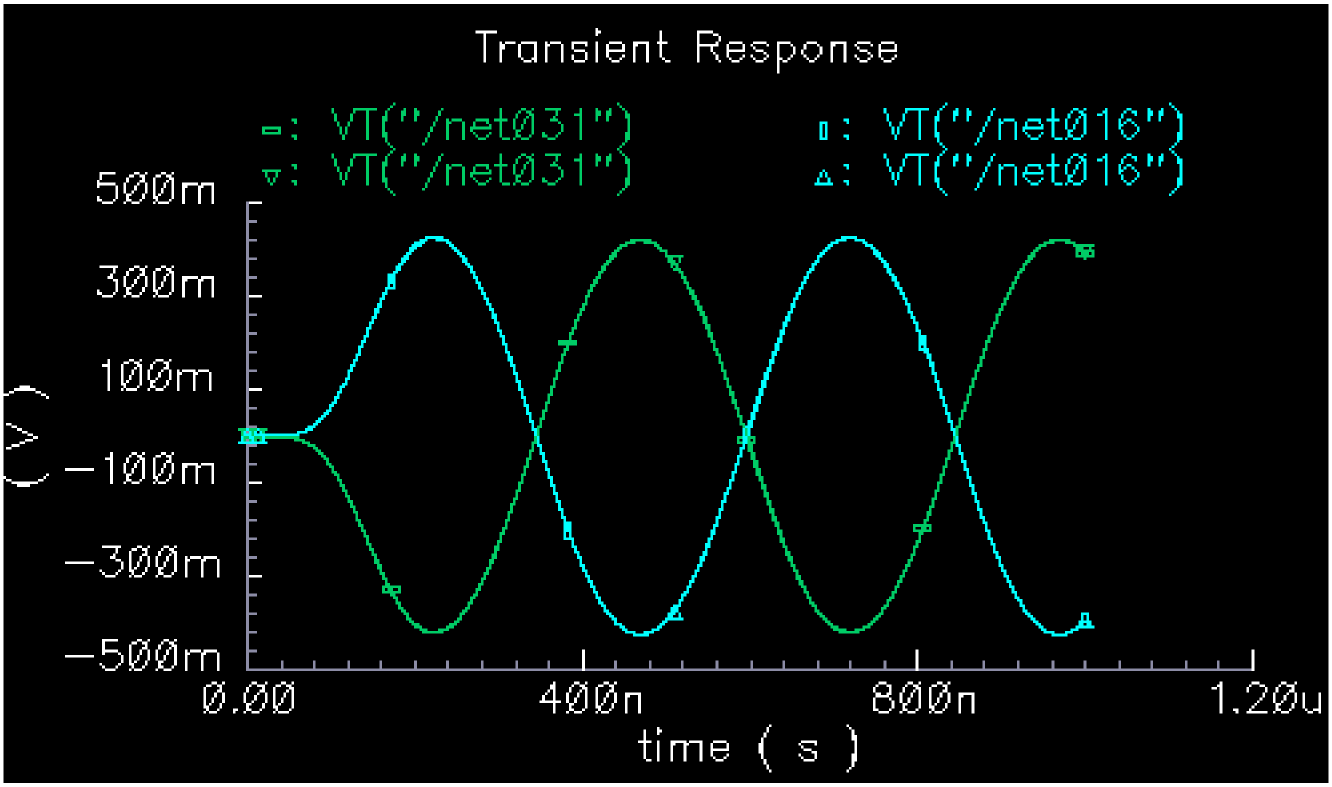


Figure B-20 Passband and baseband results with tighter options in the passband analysis



Relationship Between Baseband and Passband Noise

Noise analysis at baseband can be confusing because factors of two appear in a number of places throughout the calculations. For example:

- Each passband node becomes two equivalent baseband nodes.
  - Does noise injected at a passband node split between the two equivalent baseband nodes?
  - If so, does the noise split evenly?
- As shown in the `BB_test_bench` example, baseband models simulate peak in-phase and peak quadrature components of the carrier.

  When analyzing signal-to-noise ratios, does that mean you have to use half the square of the baseband signals?
- The analog design environment displays single sided power spectral densities.

Because the baseband power spectral density is the two-sided passband density shifted down, is there another factor of two because we can only see the baseband density for positive frequencies?

- White noise at the input of a mixer mixes up to the carrier from DC but there is also noise at twice the carrier that mixers down to the carrier.

  Does the baseband model account for this?

These questions are answered in this section.

Before sorting out the factors of two, note that baseband noise analysis is valid only for small signals. If any element in the architecture operates in a non-linear fashion, the noise analysis might be inaccurate. This is due to the fact that a baseband noise analysis follows a DC operating point analysis, rather than a PSS analysis.

Instantaneous incremental gain in a passband static non-linear model dithers at the carrier frequency.

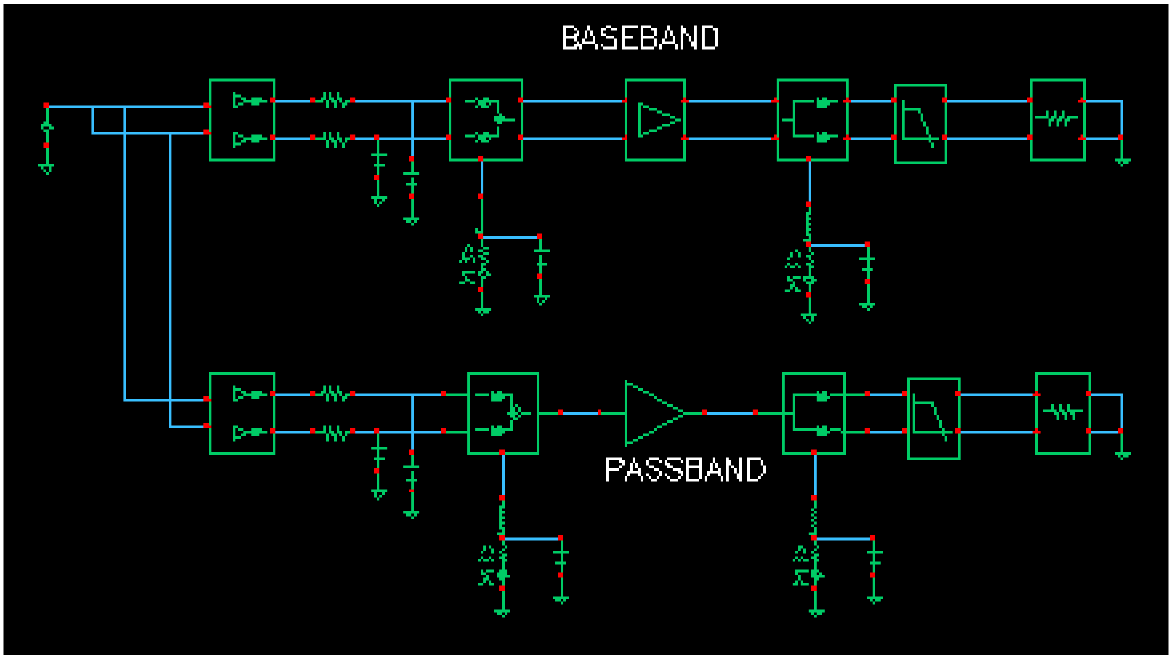
- When the carrier swings through zero, the incremental gain is large and noise at the input is amplified.
- When the carrier reaches its peak and drives the circuit into saturation, the incremental gain is smaller.

The average gain is greater than the minimum gain. The baseband model remains in the non-linear region because it only simulates peak voltages. Consequently, the incremental gain is always a minimum and the baseband model under-estimates the amount of noise propagating to the output. If the peak input signal drives the model into saturation, be sure to scale the baseband noise results accordingly.

Introduction to Analysis

The circuit discussed here is called `noise_test_circuit` and you can find it in the `rfExamples` library. The circuit looks as shown in Figure B-21 on page 2571.

Figure B-21 The noise_test_circuit in the rfExamples Library



The `noise_test_circuit` shows the relationship between baseband and passband noise. One branch consists of passband models. The other branch is a baseband equivalent of the first. You can assess noise at each of three observation points located in each branch of the circuit. At each observation point, you can examine both the noise and pnoise summaries.

The I and Q inputs are both driven by the same DC source so that you only have to view one baseband output, the other baseband output is identical by symmetry. Noise parameters in the passband and baseband models are identical. Aside from the behavioral blocks at the end of each branch, each behavioral block has noise injected at its input.

Preparation Steps for Analyses

1. Set up a PSS analysis.

   Because the local oscillator is inside the passband mixer models you have to manually enter the frequency (1GHz) into the PSS analysis form. Let the beat frequency be *autocalculated* and use 1 harmonic.

2. Set up a pnoise analysis.

   Set the start frequency equal to 0 Hz, the stop frequency equal to 100MHz, use a linear sweep with 100 steps. Set the Maximum sideband to 1. For the input source select *none* and for the output use *voltage*. For the positive output node, select the I-input of the IQ_modulator in the passband branch. Use ground for the negative node.

3. Set up a noise analysis.

   Sweep the frequency from 0 to 100MHz linearly in 100 steps. Use voltage for the output noise and select the I-input of the *IQ_mod_BB* component in the baseband branch for the positive node. Select ground for the negative node. Set the input noise port to any one of the ports in the circuit. Noise from that port does not affect either passband or baseband branches.

4. Run the analysis.

5. When the analysis finishes, go to the analog design environment simulation window and click *results – print – pss noise summary*.

6. In the form that appears, include All types, set *Type* to *integrated noise*, look at noise from 0 to 100MHz, select *truncate by number*, and view the top 2 noise contributors.

   There are only 2 noise contributors at this point, noise at the I/Q inputs and noise from the low pass resistors.

7. Print the noise summary (which is different from the pnoise summary).

   Again, print integrated noise (from 0 to 100MHz). Select All types and print noise from the top 2 noise contributors.

   Figure B-22 on page 2573 shows the noise summaries. The two summaries agree because at this point in the circuit, both nodes are really baseband nodes.

Figure B-22 Noise Summaries for the First PSS pnoise and Noise Analyses

```
Device    Param             Noise Contribution    % Of Total

/I2       IQ_modulator_i    1.56676e-12           70.06
/R20      rn                6.69678e-13           29.94

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 2.23644e-12
No input referred noise available


Device    Param          Noise Contribution    % Of Total
/I22      IQ_mod_BB_i    1.56676e-12           70.06
/R22      rn             6.69679e-13           29.94

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 2.23644e-12
Total Input Referred Noise = Inf
```

8. Repeat the analyses.

This time select the `outi` pins on the low pass filters as outputs in the appropriate noise analyses. Again print the same noise summaries but this time look at the top 12 noise contributors in each summary.

Figure B-23 on page 2574 shows the noise summaries for the second analysis. The top 7 noise contributors in the baseband and passband branches agree. The remaining noise contributors are negligible, and should be negligible for the circuit because it has no AM/PM conversion.

Figure B-23 Noise Summaries for the Second PSS pnoise Analyses

```
Device      Param                        Noise Contribution    % Of Total

/I77        PA_BB                        7.49879e-10           82.29
/I2         IQ_modulator_i               9.69448e-11           10.64
/R20        rn                           4.1437e-11            4.55
/I87        butterworth_lp_i_noise       1.52928e-11           1.68
/I84        IQ_demodulator               7.64648e-12           0.84
/PORT13     rn                           3.77571e-14           0.00
/PORT14     rn                           3.77571e-14           0.00
/I2         IQ_modulator_q               2.33139e-15           0.00
/R21        rn                           9.96506e-16           0.00
/R22        rn                           0                     0.00
/R23        fn                           0                     0.00
/R23        rn                           0                     0.00

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 9.11279e-10
No input referred noise available


Device      Param                        Noise Contribution    % Of Total
/I76        PA_BB_i                      7.49861e-10           82.29
/I22        IQ_mod_BB_i                  9.69448e-11           10.64
/R22        rn                           4.14371e-11           4.55
/I86        butterworth_lp_i_noise       1.52920e-11           1.68
/I82        IQ_demod_BB_i                7.64642e-12           0.84
/PORT15     rn                           3.77571e-14           0.00
/PORT9      rn                           3.77571e-14           0.00
/I76        PA_BB_q                      1.00329e-14           0.00
/I22        IQ_mod_BB_q                  2.33136e-15           0.00
/R23        rn                           9.96493e-16           0.00
/I82        IQ_demod_BB_q                1.54184e-31           0.00
/R23        fn                           0                     0.00

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 9.11279e-10
Total Input Referred Noise = 0.0151541
```

9. Repeat the analysis again.

This time change the Pnoise sweep to run from 900MHz to 1.1GHz in 200 linear steps and select the power amplifier output as the noise output node. For the noise analysis, leave the sweep at zero to 100MHz, use 100 steps as before, and change the output node to be the I-output of the power amplifier.

Look at the top 7 noise contributors in each analysis. This time, integrate noise from 900MHz to 1.1GHz for the pnoise run and integrate noise from 0 to 100MHz for the noise run.

Figure B-24 on page 2575 shows the new summaries. Although the noise analyses agree at the ends of the branches, the noise analyses appear to disagree at a point where the baseband node is only a baseband equivalent, not a true baseband node.

Figure B-24 Noise Summaries for the Third PSS pnoise Analyses

Device	Param	Noise Contribution	% Of Total
/I77	PA_PB	2.3769e-09	83.70
/I2	IQ_modulator_i	1.53663e-10	5.41
/I2	IQ_modulator_q	1.53663e-10	5.41
/R20	rn	6.568e-11	2.31
/R21	rn	6.568e-11	2.31
/I84	IQ_demodulator	2.42376e-11	0.85
/PORT13	rn	1.19682e-13	0.00

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 2.83995e-09
No input referred noise available

Device	Param	Noise Contribution	% Of Total
/I76	PA_BB_i	2.3769e-09	83.69
/I22	IQ_mod_BB_i	3.07294e-10	10.82
/R22	rn	1.31347e-10	4.62
/I82	IQ_demod_BB_i	2.42376e-11	0.85
/PORT9	rn	1.19682e-13	0.00
/I76	PA_BB_q	5.71606e-14	0.00
/I22	IQ_mod_BB_q	7.38993e-15	0.00

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 2.83997e-09
Total Input Referred Noise = 0.0148587

The *apparent* disagreement shown in Figure B-24 on page 2575 requires an explanation. Let us examine the noise contributors and try to answer some the questions we posed earlier.

Noise at the power amplifier output due to noise injected at a passband node

- Passband model contributors: *PA_PB* and *IQ_demodulator*, port 9.
- Baseband model contributors: *PA_BB_i*, *IQ_demod_BB_i*, port 13.

Passband and baseband counterparts contribute the same amount of noise. However, in the baseband model, from symmetry you see the same numbers if you look at the Q-node. This means the baseband model predicts twice as much total noise due to noise injected between the modulators.

This factor of two is intentionally introduced to maintain the correct signal-to-noise ratio. The baseband model simulates peak signals; the carrier is suppressed. Without the carrier, signal power equals the square of the peak rather than one half of the square. This factor of two is not as arbitrary as it seems. The baseband model predicts the correct noise after demodulation because the passband demodulator model includes an extra factor of two to offset the factor of two inherent in the demodulation process.

Let the modulated carrier be `i(t)*cos(wc*t) -q(t)*sin(wc*t) + noise(t)`.

Now consider the I-output. To generate the I-output, the demodulator multiplies the signal by `cos(wc*t)`. The only part that propagates through the subsequent filter is `(1/2)i(t) + noise(t)*cos(wc*t)`. To recover `i(t)`, the passband demodulator model must scale this sum by two. (The baseband demodulator does not need to scale by two to extract the baseband signal because the carrier is suppressed.)

Thus, noise at passband demodulator model output equals `2*noise(t)*cos(wc*t)`. The filtered noise power density is then `4*(`*`input noise density`*`)/2`. The factor of 1/2 comes from the cosine. The filtered output noise density is twice the input noise density.

In the baseband model, doubling the noise injected at the passband nodes was not simply a matter of convenience.

So, to answer questions 1 and 2,

- Yes, noise injected at a passband node splits evenly between the two equivalent baseband nodes

  but

- Because each split is doubled, the ratio of *peak* signal to total noise equals the true signal-to-noise ratio.

Question 3 is rendered moot by using the noise summary and integrating over the proper band. If the pnoise analysis only integrated from 1GHz to 1.1GHz, (instead of from 900MHz to 1.1GHz), there would be a mysterious factor of two error.

In the baseband model, phase noise entering on the phase error pin propagates to both the I and Q outputs. In the baseband model, the same noise power appears on just the one output. Again, the total noise in the baseband model is twice that of the passband model to maintain the correct signal-to-noise ratio.

Noise injected at the modulator input (resistors and modulator noise)

Total noise in the baseband model due to modulator and input resistor noise is twice what it is from just one phase. Thus, the total noise due to sources on the input sides of the modulators differ by a factor of two. This occurs because the passband model is a real multiplier, which modulates the noise. If the peak signal voltage agrees with the baseband model, the passband modulator model attenuates input noise most of the time. The important thing is that the signal-to-noise ratios in the passband and baseband models agree anywhere in the system.

Now, copy the circuit, remove the capacitors at the modulator inputs, and repeat the last set of noise analyses. You see that in the passband model, the input resistors, R20 and R21, together contribute twice as much as the baseband counterpart, R22. With the capacitors, R20 and R21 together contribute just as much as R22. Without the capacitors, the input noise is truly white over the frequencies of interest. The same thing happens to the modulator noise itself. Figure B-25 on page 2578 shows the results.

In particular, the modulator now also has noise at twice the carrier frequency and that noise mixes down to the carrier frequency. The baseband model is just that, a baseband model. The answer to question 4 is *no*. The baseband models do not account for noise, or signals, at carrier harmonics. The baseband equivalent noise analysis is valid only if noise injected into the modulators has no power beyond the local oscillator frequency. Phase noise injected at the phase noise pins should also be band limited.

Figure B-25 Noise Summaries with the input capacitors removed.

Device	Param	Noise Contribution	% Of Total
/I2	IQ_modulator_q	4.22571e-09	29.21
/I2	IQ_modulator_i	4.22571e-09	29.21
/I77	PA_PB	2.3769e-09	16.43
/R21	rn	1.80619e-09	12.49
/R20	rn	1.80619e-09	12.49
/I84	IQ_demodulator	2.42376e-11	0.17

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 1.44651e-08
No input referred noise available

Device	Param	Noise Contribution	% Of Total
/I22	IQ_mod_BB_i	4.22561e-09	50.11
/I76	PA_BB_i	2.3769e-09	28.19
/R22	rn	1.80615e-09	21.42
/I82	IQ_demod_BB_i	2.42376e-11	0.29
/PORT9	rn	1.19682e-13	0.00
/I22	IQ_mod_BB_q	1.01619e-13	0.00

Integrated Noise Summary (in V^2) Sorted By Noise Contributors
Total Output Noise = 8.43322e-09
Total Input Referred Noise = 9.76997e-11

C

Documents That Ship in the Software Hierarchy

In addition to the SpectreRF documents visible through CDNSHelp, there are many documents that ship in the software hierarchy along with workshops. To find these documents, go to

`$MMSIMHOME/tools/spectre/examples/SpectreRF_workshop`

The structure of the `SpectreRF_workshop` directory is given in the following table. To make the best use of this information, look at the summaries of contents to determine which document contains the information you need. That document will tell you which tar ball contains the corresponding database. The tar balls are also included in the `SpectreRF_workshop` directory, but are not listed here.

Document or directory	For a brief summary of contents, see
`SpectreRF_simulink_example.pdf`	SpectreRF_simulink_example.pdf on page 2580
`EnvelopeAN.pdf`	EnvelopeAN.pdf on page 2580
`HB_AN.pdf`	HB_AN.pdf on page 2583
`JitterAN.pdf`	JitterAN.pdf on page 2585
`LSSP_AN.pdf`	LSSP_AN.pdf on page 2581
`LTJM_AN.pdf`	LTJM_AN.pdf on page 2583
`MatlabAN.pdf`	MatlabAN.pdf on page 2586
`MatlabWorkshop.pdf`	MatlabWorkshop.pdf on page 2581
`NS_AN.pdf`	NS_AN.pdf on page 2582
`PerturbationAN.pdf`	PerturbationAN.pdf on page 2584
`PLL_Jitter_AN.pdf`	PLL_Jitter_AN.pdf on page 2582

Document or directory	For a brief summary of contents, see
PSRR_Drv_AN.pdf	PSRR_Drv_AN.pdf on page 2583
PSRR_Osc_AN.pdf	PSRR_Osc_AN.pdf on page 2584
PstbAN.pdf	PstbAN.pdf on page 2586
readme.txt	readme.txt on page 2583
RF_Blocks_AN.pdf	RF_Blocks_AN.pdf on page 2585
hbsp.pdf	
OFDM_tutorial.pdf	
GFSK_tutorial.pdf	

SpectreRF_simulink_example.pdf

Title: *Co-Simulation with SpectreRF MATLAB/Simulink Tutorial*

This document is a tutorial that describes how to run a co-simulation between SpectreRF and MATLAB/Simulink.

Tutorial 1 demonstrates Simulink, SpectreRF co-simulation while the SpectreRF simulator runs a tran analysis, for the system-level module of a wireless LAN. The three steps required for the co-simulation include:

1. Inserting and configuring the SpectreRF Engine in the Simulink schematic.
2. Setting up the netlist to adopt the SpectreRF and MATLAB co-simulation.
3. Running the co-simulation.

Tutorial 2 demonstrates Simulink, SpectreRF co-simulation while the SpectreRF simulator runs an ENVLP analysis. This tutorial illustrates:

1. The use mode of manually running Spectre with ADE and calling MATLAB from ADE.
2. The use mode of manually running Simulink and activating Spectre through Simulink.

EnvelopeAN.pdf

Title: *A User’s Guide to Envelope Following Analysis Using SpectreRF*

This document is an application note that describes how to use envelope following analysis with SpectreRF. The topics covered include:

1. Envelope following algorithm
2. Using the envelope following analysis
3. Parameters used in the analysis
4. Plotting results
5. An automatic gain control (AGC) example
6. Comparing simulation and measurement results for a GSM power amplifier
7. Using autonomous envelope following, with examples
8. Simulating circuits with FM sources

LSSP_AN.pdf

Title: *Measuring Large-Signal S-Parameters Using SpectreRF*

This document illustrates how to measure large signal S-parameters using Spectre and SpectreRF from within ADE. The contents include:

1. Calculating LSSP in terms of input power
2. Calculating LSSP in terms of beat frequency
3. Comparing large and small-signal S-parameters

MatlabWorkshop.pdf

Title: *SpectreRF Workshop: Using the SpectreRF MATLAB Toolbox*

This document is a workshop that describes how to use SpectreRF and the SpectreRF MATLAB toolbox to measure parameters that are important in the design verification of low noise amplifiers (LNA). The contents include:

1. Measuring the small signal gain (SP) of an LNA
2. Conducting a large signal noise analysis (PSS and Pnoise)
3. Measuring gain compression and total harmonic distortion (Swept PSS)
4. Measuring IP3 (PSS plus PAC)

5. Measuring IP3 (QPSS analysis with Shooting or the Harmonic Balance engine)
6. Measuring IP3 (Rapid IP3 using AC analysis)
7. Measuring IP3 (Rapid IP3 using PSS Plus PAC)

PLL_Jitter_AN.pdf

Title: *Application Note. Pll jitter measurements*

This document illustrates how to use the Spectre and SpectreRF simulators with ADE to measure jitter characteristics of phase-locked loop (PLL) circuits. The contents include:

1. Using SpectreRF to simulate the individual blocks and measure the jitter and operating parameters of the models
2. Creating or modifying the behavioral models of the blocks to incorporate the jitter
3. Time domain simulation of the PLL using the behavioral models of the blocks.
4. Post processing of the simulation results to obtain the jitter and noise characteristics of the entire PLL.

NS_AN.pdf

Title: *The Application Note of Noise Separation in Pnoise/Qpnoise Analysis*

This document is an application note that describes how to use noise separation features in Pnoise and Qpnoise analysis to extract noise sources. The contents include:

1. The principles of noise separation in RF circuits
2. The graphical user interface used for noise separation
3. The flow to follow to determine which noise sources add the most noise to the output
4. Understanding the noise source summary report
5. An example is provided that illustrates how to locate the noise sources that contribute the most noise to the output in an NE600 mixer.

PSRR_Drv_AN.pdf

Title: *Power Supply Rejection Ratio Characterization Using SpectreRF Driven Circuits*

This document illustrates how to characterize the power supply rejection ratio (PSRR) of driven circuits. The contents include:

1. How to set up the PXF analysis
2. Using Direct Plot for the PXF analysis to plot the periodic transfer function

readme.txt

This text document lists the main topic of each of the `.pdf` files in the `SpectreRF_workshop`.

HB_AN.pdf

Title: *Harmonic Balance Release Note*

This document illustrates how to use the Harmonic Balance (HB) method, which is very efficient in simulating circuits with only low order harmonics, such as low-noise amplifiers. The contents include:

1. The parameters used to specify HB for driven circuit large signal simulations
2. How to use HB from within ADE
3. Discussion of when it is most appropriate to use time domain Shooting and when it is better to use HB

LTJM_AN.pdf

Title: *Application Note. Long term jitter measurements*

This document describes the procedures used to measure long term jitter characteristics from within the Analog Design Environment (ADE). Long term jitter characterizes the variation in the accumulated width of a large number of clock periods. The contents include:

1. Plotting the phase noise power spectral density, which will be used to compute the long term jitter

2. Selecting the number of cycles to use for the measurement
3. Selecting the minimum frequency of the integration. Long term jitter is described as very sensitive to the lower frequency limit that is used in the integration of the phase noise.

PerturbationAN.pdf

Title: *Perturbation Based Measurements Using SpectreRF*

This document describes how to use perturbation based rapid IP3, rapid IP2, the IM2 distortion summary, and the compression distortion summary to measure the compression distortion and intercept points of RF circuits. Contents include:

1. A discussion of intermodulation distortion
2. A discussion of compression distortion
3. A discussion of the perturbation method used in the analyses
4. An example using IP3 and compression distortion summary measurements of a mixer
5. An example using IP2 and IM2 distortion summary measurements of a Gilbert direct conversion mixer
6. An example using IPa3 and compression distortion summary measurements of a power amplifier

PSRR_Osc_AN.pdf

Title: *Power Supply Rejection Ratio Characterization Using SpectreRF Autonomous Circuits*

This document describes how to measure the power supply rejection ratio (PSRR) of oscillators and other autonomous circuits. Knowing PSRR values can help you protect against noise sources such as power supply ripples. The same methodology can be used to measure the effect of substrate noise. Contents include:

1. Setting up the PXF analysis to measure the PSRR of a VCO.
2. A discussion of the different definitions of PSRR
3. A discussion of confirming the results of the PXF analysis by taking quasi-static measurements of the tuning sensitivity and frequency pushing of the VCO

RF_Blocks_AN.pdf

Title: *Using Verilog-A Baseband Models of RF Blocks Application Note*

This document describes how to create and use Verilog-A signal flow and conservative baseband models of RF blocks. Contents include:

1. Several examples exploring the correspondence between passband (PB) and baseband (BB) representations of RF signals
2. Examples illustrating baseband modeling of linear time invariant blocks
3. Examples illustrating baseband modeling of nonlinear blocks using an amplifier
4. Code examples for Verilog-A modules, including
   - An ideal baseband to passband converter
   - A behavioral model of a passband to baseband converter
   - A behavioral model of a deviation frequency meter
   - A baseband model of a passband low pass filter with transfer function `H(s) = a/(a+s)`
   - A signal flow baseband model of a nonlinear RF block based on the complex BB gain approach

JitterAN.pdf

Title: *Jitter Measurements Using SpectreRF Application Note*

This document illustrates how to measure the jitter characteristics of typical blocks, working within ADE. Examples are presented for both driven and autonomous circuits. Contents include:

1. A discussion of how time (or phase) jitter is defined. RMS jitter and peak-to-peak jitter measurements are introduced
2. An example illustrating how to measure jitter for an autonomous circuit (differential VCO)
3. An example illustrating how to measure phase noise and jitter at the output of a divider, in a circuit that uses a VCO with divide-by-2 prescaler

MatlabAN.pdf

Title: *Spectre/RF Matlab Toolbox Application Note*

This document describes how to use SpectreRF MATLAB toolbox to read in a simulation result and perform standard RF measurements. Contents include:

1. Instructions for obtaining and setting up the toolbox
2. A list and description of the basic functions included in the toolbox:
    - `cds_srr`
    - `cds_evalsig`
    - `cds_plotsig`
    - `cds_harmonic`
    - `cds_interpsig`
    - `cds_fft`
    - `cds_compression`
    - `cds_ipn`
3. An example illustrating how to obtain the 1dB compression point, the first order harmonic signal, and the total harmonic distortion
4. A discussion of compatibility with the Aptivia MATLAB functions

PstbAN.pdf

Title: *Stability analysis of Linear Periodical Time-Varying Circuit using SpectreRF PSTB Analysis Application Note*

This document discusses how to use STB analysis to evaluate the stability of a linear periodical time-varying circuit. Contents include:

1. An example illustrating how to run an STB and PSTB analysis on a VCO with an inherent nonlinearity
2. An example illustrating how to examine the local stability of an injection-locked oscillator

An example illustrating how to examine the global stability of an injection-locked oscillator

Index

Numerics

1dB compression point
  low noise amplifier, plotting 877

A

analysis
  descriptions 127, 179, 229, 253

B

batch mode, setting in .cdsinit file 672
before Simulation value of Start MATLAB field 794
Bench Type 674

C

Cadence libraries, setting up 805, 806
conventions, typographic and syntax 29
conversion gain measurement, mixer 1047
  conversion gain, plotting 1052
  power supply rejection, plotting 1054
cosimulation
  generating netlist for lower-level block 791
  inputs, selecting 792
  options, enabling 793
  outputs, selecting 792
  running 800
  setting up and running 786
  simulation response timeout, setting 789
  socket mode for, setting 789
  socket port 793
  socket port for, setting 789
  software required for 786
  Spectre command used in 789
  start method for 791, 794
  starting ADE manually and MATLAB automatically 795
  starting MATLAB and Spectre RF separately 795, 800
  starting MATLAB manually and Spectre RF automatically 797, 801
  starting Spectre RF manually and MATLAB automatically 801
  stop time 795
  with MATLAB, introduction to 786
coupler block, connecting to system-level Simulink schematic 787
current mismatch, effect on PFD-CP 703

D

delay mismatch, effect on PFD-CP 704
Designer’s Guide to SPICE and Spectre 28
divider jitter, generating Verilog-A model of 672
documents, shipped in the software hierarchy 1413

E

Enable PLL Macro Model option 674
equivalent noise resistance, low noise amplifier, plotting 855
examples
  low noise amplifier 819
    1dB compression point, plotting 877
    noise calculations 864
    output voltage distribution 832
    PXF analysis 894
    S-parameter analysis 838
      voltage standing wave ratio, plotting 845
    S-parameter Noise analysis 846
      equivalent noise resistance, plotting 855
      load stability circles, plotting 857
      minimum noise figure 852
      noise circles, plotting 861
      noise figure 852
      source stability circles 859
    third-order intercept point, plotting 887

voltage gain
calculation 824
plotting 829
mixer
conversion gain measurement 1047
conversion gain, plotting 1052
power supply rejection, plotting 1054
harmonic distortion measurement 993
intermodulation distortion measurement
with QPSS 1072
noise figure measurement 1020
1dB compression point 1056
noise figure measurement with PSP 1029
periodic S-parameter plots 1029
third-order intercept measurement with PSS sweep and PAC 1063
oscillator simulation, of differential oscillator 924
output noise, plotting 937
oscillator simulation, of tline3oscRF
phase noise, plotting 921
steady state solution, plotting 918

F

Fast Frac-N VCO template 678
Fast VCO template 678
flicker noise computation 264
Format — Port/Signal Displays — Signal Dimensions 790
Fourier analysis 127
fractional-N VCO template 678
frame size, specifying 788
freq_meter instance, for measuring PLL noise 717
frequency synthesizer, block diagram of 672

H

harmonic distortion measurement, mixer 993

I

input pins, number of in SpectreRF Engine block 788
intermodulation distortion
measurement 182
with QPSS, mixer 1072
isnoisy parameter 849

J

jitter variation with phase difference, in VCO 705
jitter, divider 672

L

libpll_sh.so 673
libpllMMpsd_sh.so 673
libpllPPVoscModel_sh.so 673
libpllTTpfd_cpModel_sh.so 673
load stability circle
low noise amplifier, plotting 857
low noise amplifier
1dB compression point, plotting 877
noise calculations 864
output voltage distribution 832
PXF analysis 894
S-parameter analysis 838
voltage standing wave ratio, plotting 845
S-parameter Noise analysis 846
equivalent noise resistance, plotting 855
load stability circles, plotting 857
minimum noise figure 852
noise circles, plotting 861
noise figure 852
source stability circles 859
third-order intercept point, plotting 887
voltage gain
calculation 824
plotting 829
low noise amplifier, simulation example 819
lteratio parameter 140

M

macro-models
PLL simulation with 717
Math Operations library 790
MATLAB
design name 794, 796
start command 794
startup directory 794
maxacfreq parameter 141
maxsideband parameter 257
minimum noise figure, low noise amplifier 852
mixer
conversion gain measurement 1047
conversion gain, plotting 1052
power supply rejection, plotting 1054
harmonic distortion measurement 993
intermodulation distortion measurement with QPSS 1072
noise figure measurement 1020
1dB compression point 1056
noise figure with PSP measurement 1029
periodic S-parameter plots measurement 1029
third-order intercept measurement with PSS sweep and PAC 1063

N

netlist
generating for cosimulation 791
generating for lower-level block in cosimulation 791
preparing for cosimulation in ADE 791
preparing for cosimulation without using a GUI 798
newlink OutputParamsPXF 234
no command value, of Start MATLAB field 794
noise calculations, low noise amplifier 864
noise circles
low noise amplifier, plotting 861
noise figure measurement 263
low noise amplifier 852
mixer 1020
1dB compression point calculating 1056
noise figure measurement with PSP, mixer 1029
Noise Summary form 874
noise temperature 755
noisefile parameter 263
noisetemp parameter 263, 755
noisevec parameter 263
Normal VCO template 678
now command value
in MATLAB cosimulation 794
of Start MATLAB field 794
using to test whether ADE starts MATLAB 794

O

oprobe parameter 263
oscillators
differential oscillator 924
output noise, plotting 937
starting in simulations 903
tline3oscRF
phase noise, plotting 921
steady state solution, plotting 918
troubleshooting simulations for 982
oscmm model 681
oscmm Symbol 682
oscmm_fast symbol 683
oscmm_frac symbol 685
oscmm_frac_fast Symbol 688, 689, 691, 692, 694, 695, 697, 698, 700
output noise, plotting with oscillators 937
output pins, number of in SpectreRF Engine block 788
output voltage distribution, low noise amplifier 832
overview, Spectre RF analyses 2

P

PAC 179
PAC analysis 179
intermodulation distortion measurement with 182
synopsis 181
parameters
leratio 140
maxacfreq 141

maxsideband 257
noisefile 263
noisetemp 263
noisevec 263
oprobe 263
period 137
refsideband 255
reltol 140
saveinit 138
sidebands 257
steadyratio 140
tinit 137
tonset 136
tstart 136
tstop 137
values 264
parameters tonset 136
period parameter 137
periodic AC analysis, *See* PAC analysis
periodic noise analysis, *See* Pnoise analysis
periodic S-parameter plots, mixer 1029
periodic steady-state analysis. *See* PSS analysis
periodic transfer function analysis, *See* PXF analysis
pfd_cp_bench 706
PFD-CP
input and output 702
nonideal effects on performance of 702
PFD-CP block, generating Verilog-A model of 672
PFD-CP macro-model, extracting 701, 706
PFD-CP macro-model, using 710
PFDCP Sweep field, use of 708
phase noise, discussion of 631
and SpectreRF simulation 645
frequently asked questions 652
further reading 657
models 634
troubleshooting 647
phase noise, with oscillators 903
plotting 921
pins, input, in SpectreRF Engine block 788
pins, output, in SpectreRF Engine block 788
PLL Bench 674
PLL Macro Model Wizard 674
PLL macro-model test bench (fractional-N), schematic of 719
PLL macro-model test bench (integer-N), schematic of 718
PLL simulation, challenges presented by 670
PLL test benches, found in pllMMLib 672
pllmmnoise.vcsv, produced if PSD plugin is used 718
PLLs
simulating with macro-models 717
pllTTpfd_cp symbol 710
plugin, command for using with Spectre 673
Pnoise 253
Pnoise analysis 253
flicker noise computation 264
noise figure computation 263
synopsis 257
PSS 127
PSS analysis 127
PSS Noise Summary form 874
PXF 229
PXF analysis 229
low noise amplifier 894
synopsis 234

R

refsideband parameter 255
related documents 1413
reltol parameter 140
resistor noise generation 849

S

sample time, setting for block used in cosimulation 789
saveinit parameter 138
setting up the software 803
setting up the Cadence libraries 805
using the UNIX shell window 806
Setup — Matlab/Simulink — Log file 794
Setup — Matlab/Simulink — Setting 792
sidebands parameter 257
Simulation — Configuration Parameters 790
Simulation — Start 800, 801
simulation response timeout for cosimulation 789
socket mode, for cosimulation 789
socket port, for cosimulation 789
source stability circle

low noise amplifier 859
S-parameter analysis, low noise amplifier 838
noise 846
voltage standing wave ratio, plotting 845
S-parameter Noise analysis, low noise amplifier
equivalent noise resistance plotting 855
load stability circles plotting 857
minimum noise figure 852
noise circles plotting 861
noise figure 852
source stability circles plotting 859
S-parameter simulation data, plotting
equations for S-parameter calculator 769
Spectre command, for cosimulation 789
Spectre RF
analyses, overview 2
SpectreRF Engine
block, number of input pins in 788
block, number of output pins in 788
showing port labels for 789
SpectreRF-workshop 1413
Start MATLAB field
before Simulation value 794
no value 794
now value 794
values for 794
starting the oscillator 903
steady state solution, plotting with oscillators 918
steadyratio parameter 140
stop time, for cosimulation 795
synopses
PAC 181
Pnoise 257
PXF 234

T

third-order intercept point
low noise amplifier, plotting 887
measurement with PSS sweep and PAC 1063
tinit parameter 137
tonset parameter 136
Tools — Analog Environment 792
troubleshooting, with oscillator simulation 982
tstab parameter 136
tstart parameter 136
tstop parameter 137
tuning curve, of VCO 675
typographical conventions 29

U

using in SpectreRF simulation
port parameter types 725

V

values parameter 264
VCO
effect of buffers and prescalars on simulation 680
jitter variation with phase difference in 705
noiseless macro-model of 680
white noise supported by 680
VCO block, generating CMI macromodel of 672
VCO extraction 675
VCO internal noise 676
VCO macro-model 676
VCO macro-model, extracting 677
VCO sweep, setting 679
VCO templates 678
VCO test bench for macro-model extraction 678
vco_vdd 680
vco_vout 680
vco_vss 680
vco_vtune 680
VCO, turning curve of 675
Vctrl sweep range 680
VDD sweep 680
vddstart VCO sweep parameter, vddstop VCO sweep parameter, vddsteps VCO sweep parameter 679
View — Simulink library 790
voltage gain
calculation, low noise amplifier 824
plotting, low noise amplifier 829
VSS sweep 680

W

white noise, in VCO 680