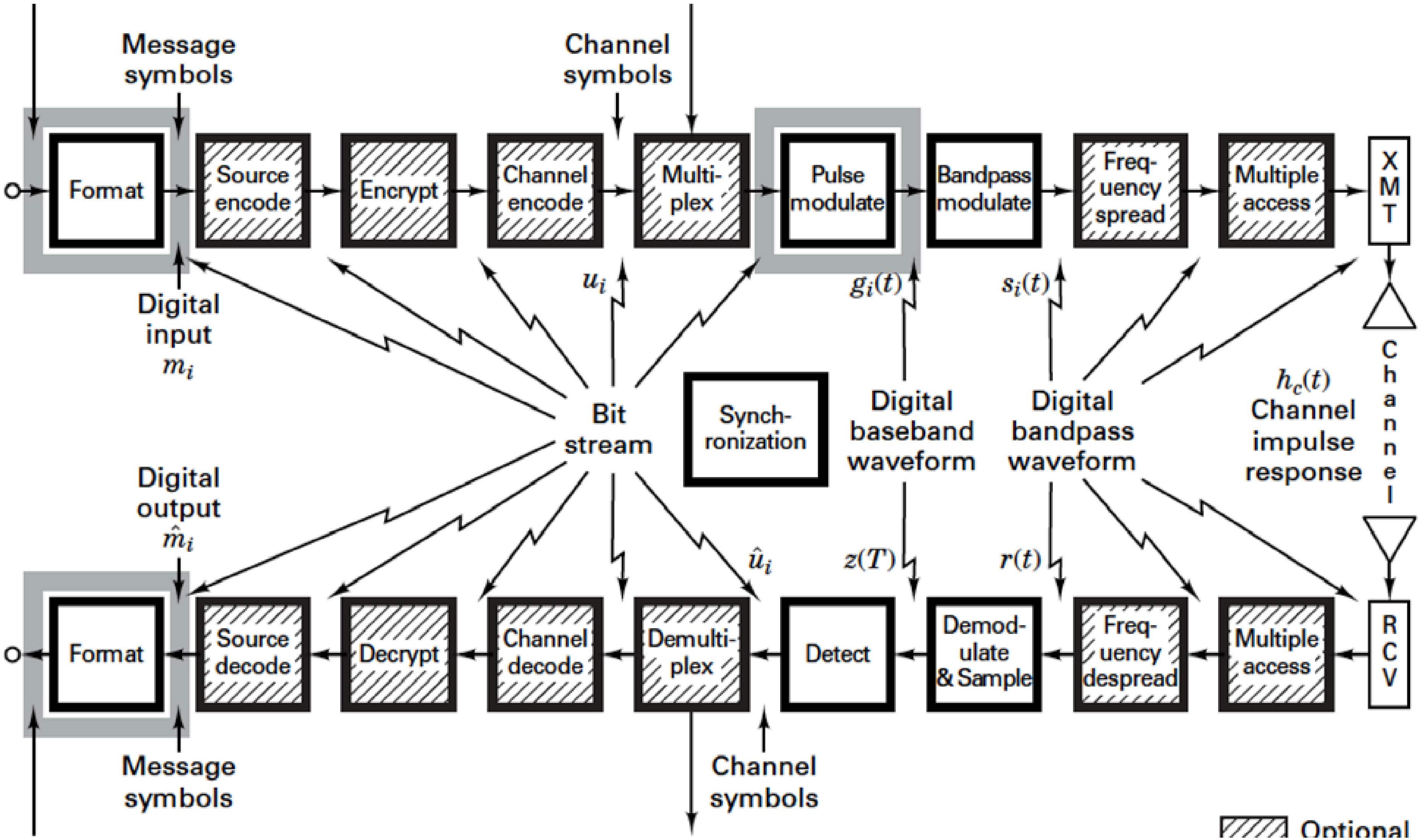
Chapter 2

Formatting and Baseband Modulation

Formatting

Formatting & Baseband



Formatting and Baseband

Formatting

- Character coding
- Sampling
- Quantization
- Pulse code modulation (PCM)

Source Coding

- Predictive coding
- Block coding
- Variable length coding
- Synthesis/analysis coding
- Lossless compression
- Lossy compression

Baseband Signaling

- PCM waveforms (line codes)
  - Nonreturn-to-zero (NRZ)
  - Return-to-zero (RZ)
  - Phase encoded
  - Multilevel binary
- *M*-ary pulse modulation
  - PAM, PPM, PDM

Equalization

- Maximum-likelihood sequence estimation (MLSE)
- Equalization with filters
  - Transversal or decision feedback
  - Preset or Adaptive
  - Symbol spaced or fractionally spaced

Bandpass Signaling

Coherent

- Phase shift keying (PSK)
- Frequency shift keying (FSK)
- Amplitude shift keying (ASK)
- Continuous phase modulation (CPM)
- Hybrids

Noncoherent

- Differential phase shift keying (DPSK)
- Frequency shift keying (FSK)
- Amplitude shift keying (ASK)
- Continuous phase modulation (CPM)
- Hybrids

Channel Coding

Waveform

- *M*-ary signaling
- Antipodal
- Orthogonal
- Trellis-coded modulation

Structured Sequences

- Block
- Convolutional
- Turbo

Synchronization

- Frequency synchronization
- Phase synchronization
- Symbol synchronization
- Frame synchronization
- Network synchronization

Multiplexing/Multiple Access

- Frequency division (FDM/FDMA)
- Time division (TDM/TDMA)
- Code division (CDM/CDMA)
- Space division (SDMA)
- Polarization division (PDMA)

Spreading

- Direct sequencing (DS)
- Frequency hopping (FH)
- Time hopping (TH)
- Hybrids

Encryption

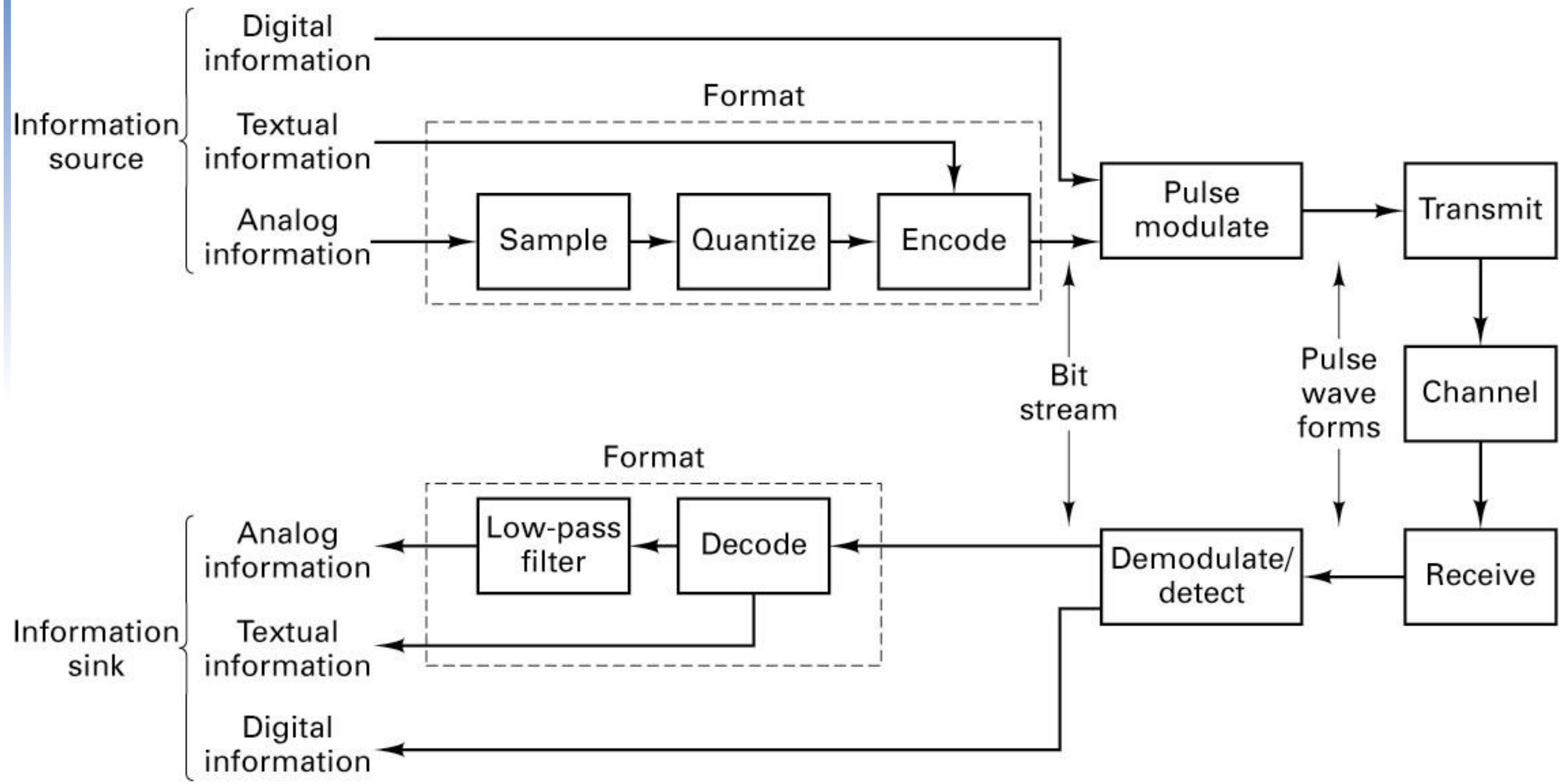
- Block
- Data stream

What is Formatting?

- Information can take either of the three forms:
  1. Textual information
  2. Analog signals
  3. Digital data
- Before the signals are transmitted over a digital communication channel, an information bearing signal must be converted to digital symbols (Formatting).
- The resulting digital symbols are then represented by baseband waveforms (Pulse Modulation or Line Coding).

Block Diagram

Block diagram representing formatting and transmission of baseband signals.



Textual Data (1)

American Standard Code for Information Interchange (ASCII) for encoding alphanumerics

Bits											
				5	0	1	0	1	0	1	0	1
				6	0	0	1	1	0	0	1	1
1	2	3	4	7	0	0	0	0	1	1	1	1
0	0	0	0		NUL	DLE	SP	0	@	P	'	p
1	0	0	0		SOH	DC1	!	1	A	Q	a	q
0	1	0	0		STX	DC2	"	2	B	R	b	r
1	1	0	0		ETX	DC3	#	3	C	S	c	s
0	0	1	0		EOT	DC4	$	4	D	T	d	t
1	0	1	0		ENQ	NAK	%	5	E	U	e	u
0	1	1	0		ACK	SYN	&	6	F	V	f	v
1	1	1	0		BEL	ETB	'	7	G	W	g	w
0	0	0	1		BS	CAN	(	8	H	X	h	x
1	0	0	1		HT	EM	)	9	I	Y	i	y
0	1	0	1		LF	SUB	*	:	J	Z	j	z
1	1	0	1		VT	ESC	+	;	K	[	k	{
0	0	1	1		FF	FS	,	<	L	\	l	|
1	0	1	1		CR	GS	–	=	M	]	m	}
0	1	1	1		SO	RS	.	>	N	∧	n	~
1	1	1	1		SI	US	/	?	O	–	o	DEL

NUL	Null, or all zeros	DC1	Device control 1
SOH	Start of heading	DC2	Device control 2
STX	Start of text	DC3	Device control 3
ETX	End of text	DC4	Device control 4
EOT	End of transmission	NAK	Negative acknowledge
ENQ	Enquiry	SYN	Synchronous idle
ACK	Acknowledge	ETB	End of transmission
BEL	Bell, or alarm	CAN	Cancel
BS	Backspace	EM	End of medium
HT	Horizontal tabulation	SUB	Substitute
LF	Line feed	ESC	Escape
VT	Vertical tabulation	FS	File separator
FF	Form feed	GS	Group separator
CR	Carriage return	RS	Record separator
SO	Shift out	US	Unit separator
SI	Shift in	SP	Space
DLE	Data link escape	DEL	Delete

Textual Data (2)

Extended Binary Coded Decimal Interchange Information (EBCDIC) for encoding alphanumerics

Bits 5	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1
Bits 6	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1
Bits 7	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1
Bits 8	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
1 2 3 4																
0 0 0 0	NUL	SOH	STX	ETX	PF	HT	LC	DEL			SMM	VT	FF	CR	SO	SI
0 0 0 1	DLE	DC1	DC2	DC3	RES	NL	BS	IL	CAN	EM	CC		IFS	IGS	IRS	IUS
0 0 1 0	DS	SOS	FS		BYP	LF	EOB	PRE			SM			ENQ	ACK	BEL
0 0 1 1			SYN		PN	RS	US	EOT					DC4	NAK		SUB
0 1 0 0	SP										¢		<	(	+	!
0 1 0 1	&										!	$	*	)	;	¬
0 1 1 0	–	/										,	%	_	>	?
0 1 1 1											:	#	@	'	=	"
1 0 0 0		a	b	c	d	e	f	g	h	i						
1 0 0 1		j	k	l	m	n	o	p	q	r						
1 0 1 0			s	t	u	v	w	x	y	z						
1 0 1 1																
1 1 0 0		A	B	C	D	E	F	G	H	I						
1 1 0 1		J	K	L	M	N	O	P	Q	R						
1 1 1 0			S	T	U	V	W	X	Y	Z						
1 1 1 1	0	1	2	3	4	5	6	7	8	9						

PF	Punch off
HT	Horizontal tab
LC	Lower case
DEL	Delete
SP	Space
UC	Upper case
RES	Restore
NL	New line
BS	Backspace
IL	Idle
PN	Punch on
EOT	End of transmission
BYP	Bypass
LF	Line feed
EOB	End of block
PRE	Prefix (ESC)
RS	Reader stop
SM	Start message
DS	Digit select
SOS	Start of significance
IFS	Interchange file separator
IGS	Interchange group separator
IRS	Interchange record separator
IUS	Interchange unit separator
Others	Same as ASCII

Message and Symbol

- Textual message comprises a sequence of alphanumeric characters.
  - Example: Hello, how are you.
- Textual message is converted into a sequence of bits, i.e. bit stream or baseband signal.
- Symbols are formed by a group of k bits from a finite symbol set of $M=2^k$ such symbols.
- A system using a symbol set size of M is referred to as an *M-ary* system.

Message and Symbol: Example

Message (text): "THINK"

Character coding (6-bit ASCII):

	T	H	I	N	K
Character coding (6-bit ASCII):	0 0 1 0 1 0	0 0 0 1 0 0	1 0 0 1 0 0	0 1 1 1 0 0	1 1 0 1 0 0

3-bit groups	0 0 1	0 1 0	0 0 0	1 0 0	1 0 0	1 0 0	0 1 1	1 0 0	1 1 0	1 0 0
8-ary digits (symbols):	1	2	0	4	4	4	3	4	6	4
8-ary waveforms:	$s_1(t)$	$s_2(t)$	$s_0(t)$	$s_4(t)$	$s_4(t)$	$s_4(t)$	$s_3(t)$	$s_4(t)$	$s_6(t)$	$s_4(t)$

(a)

	T	H	I	N	K
Character coding (6-bit ASCII):	0 0 1 0 1 0	0 0 0 1 0 0	1 0 0 1 0 0	0 1 1 1 0 0	1 1 0 1 0 0

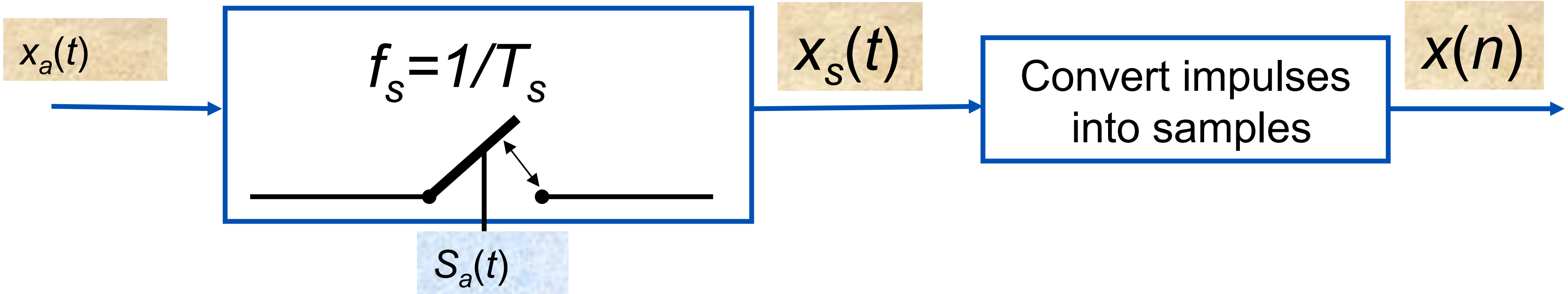
5-bit groups	0 0 1 0 1	0 0 0 0 1	0 0 1 0 0	1 0 0 0 1	1 1 0 0 1	1 0 1 0 0
32-ary digits (symbols):	5	1	4	17	25	20

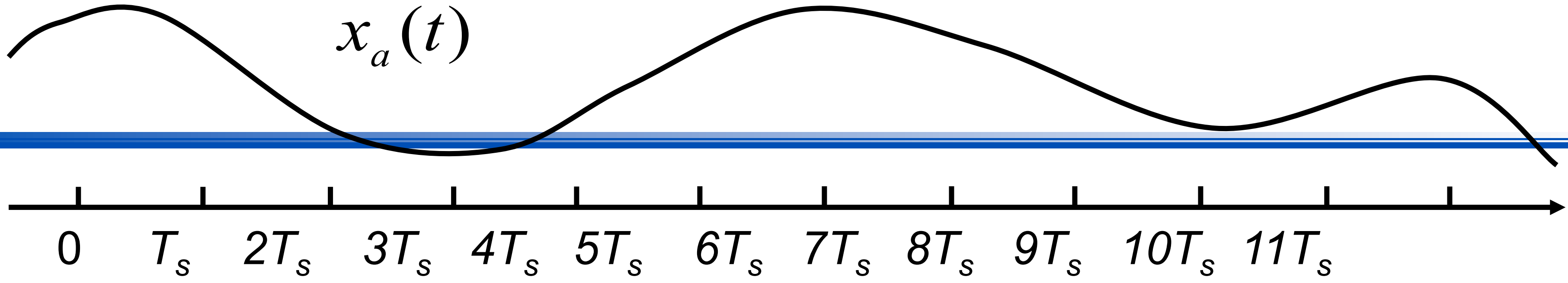
Formatting Analog Information

Periodic Sampling

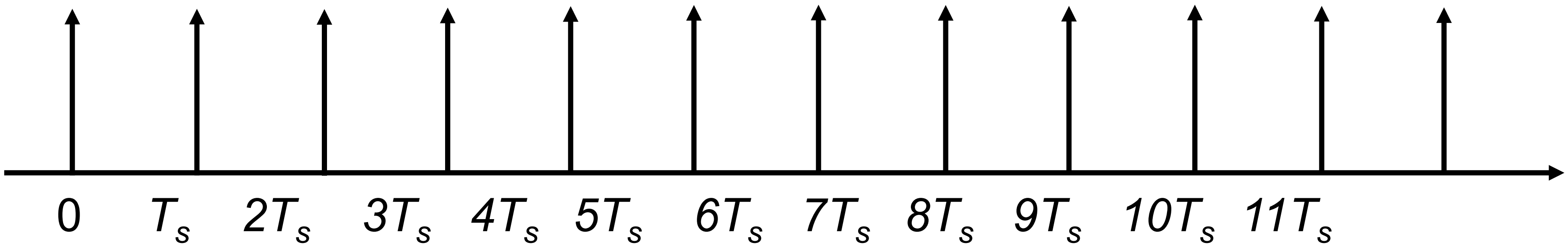
- Typically, discrete-time signals are formed by periodically sampling a continuous-time signal : $x(n)=x_a(nT_s)$

  The sampling interval T_s is the sampling period, and $f_s=1/T_s$ is the sampling frequency in samples per second.
- The sampling process:





$$s_a(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$



$$x_s(t) = x_a(t)s_a(t) = \sum_{n=-\infty}^{\infty} x_a(t)\delta(t - nT_s) = \sum_{n=-\infty}^{\infty} x_a(nT_s)\delta(t - nT_s)$$

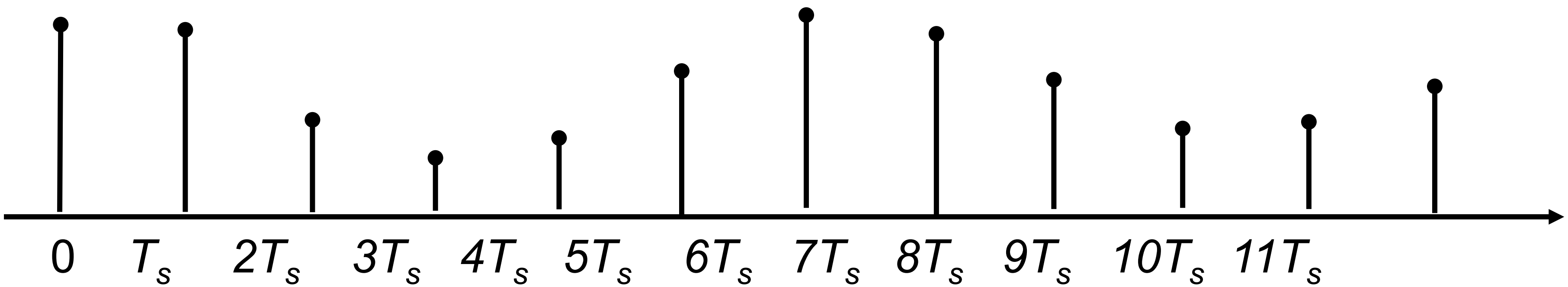
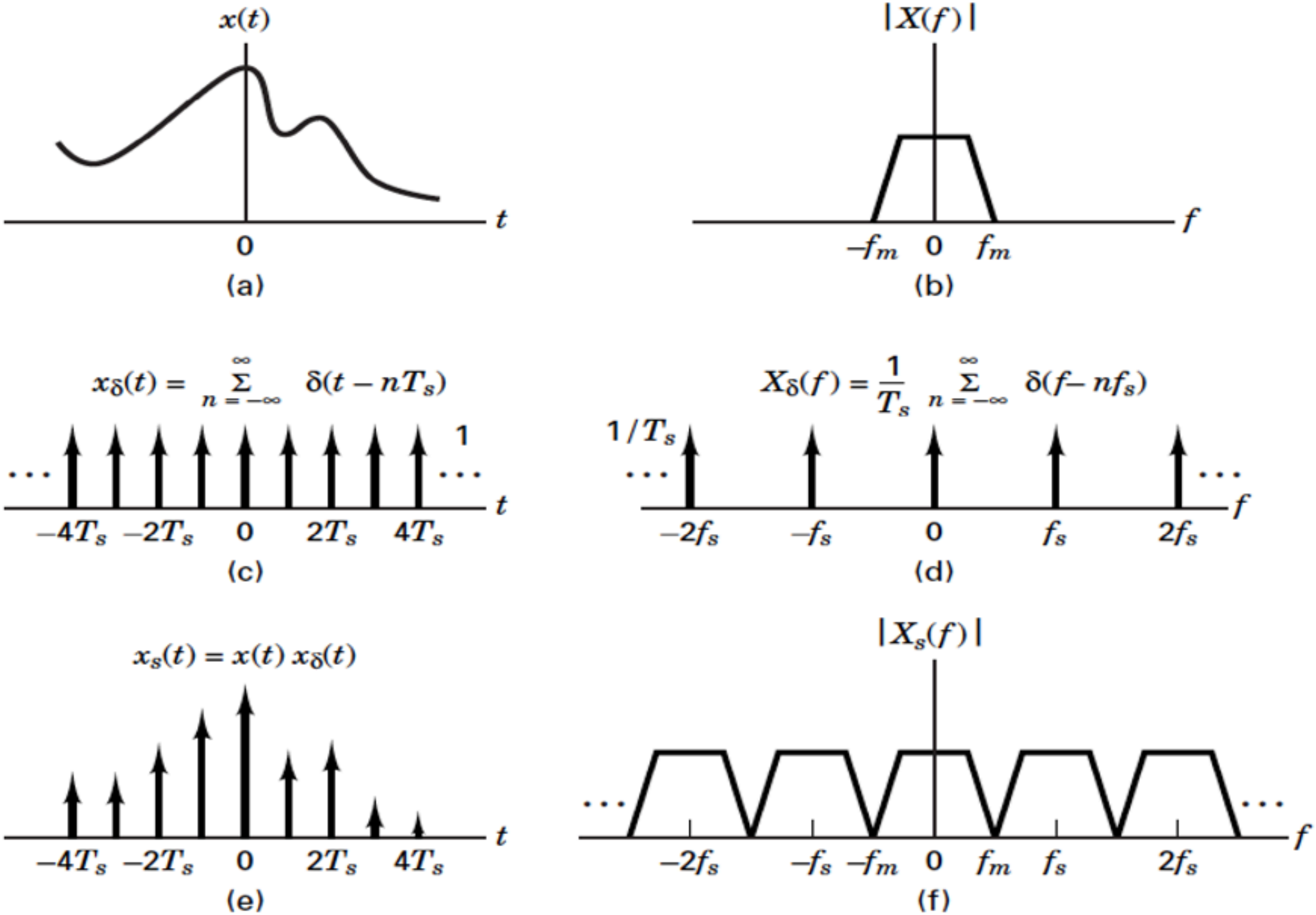


Illustration of Ideal Sampling



Fourier Transform of a CT Sampled Signal

- Fourier transform pair:

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

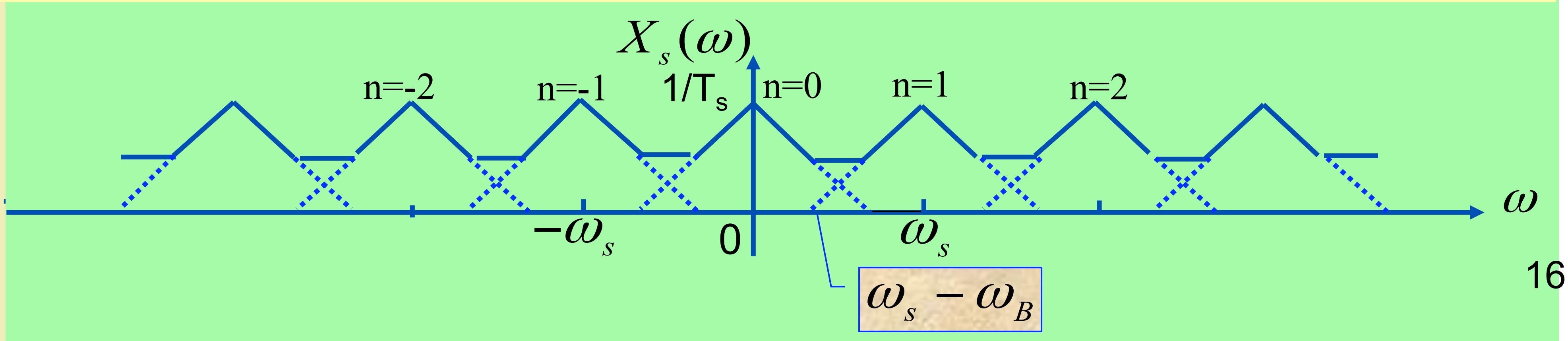
- Fourier transform of sampled signal :

$$X_s(\omega) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X_a(\omega - n\omega_s), \quad \omega_s = \frac{2\pi}{T_s}$$

$$X_s(\omega) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X_a(\omega - n\omega_s), \quad \omega_s = \frac{2\pi}{T_s}$$

- The Fourier transform of the continuous-time sampled signal $X_s(\omega)$ is a periodic function of ω consisting of a superposition of shifted replicas of $X_a(\omega)$, scaled by $1/T_s$.

The overlap of the Fourier transform of each of the terms of the sampled signal is called aliasing



Sampling Theorem :

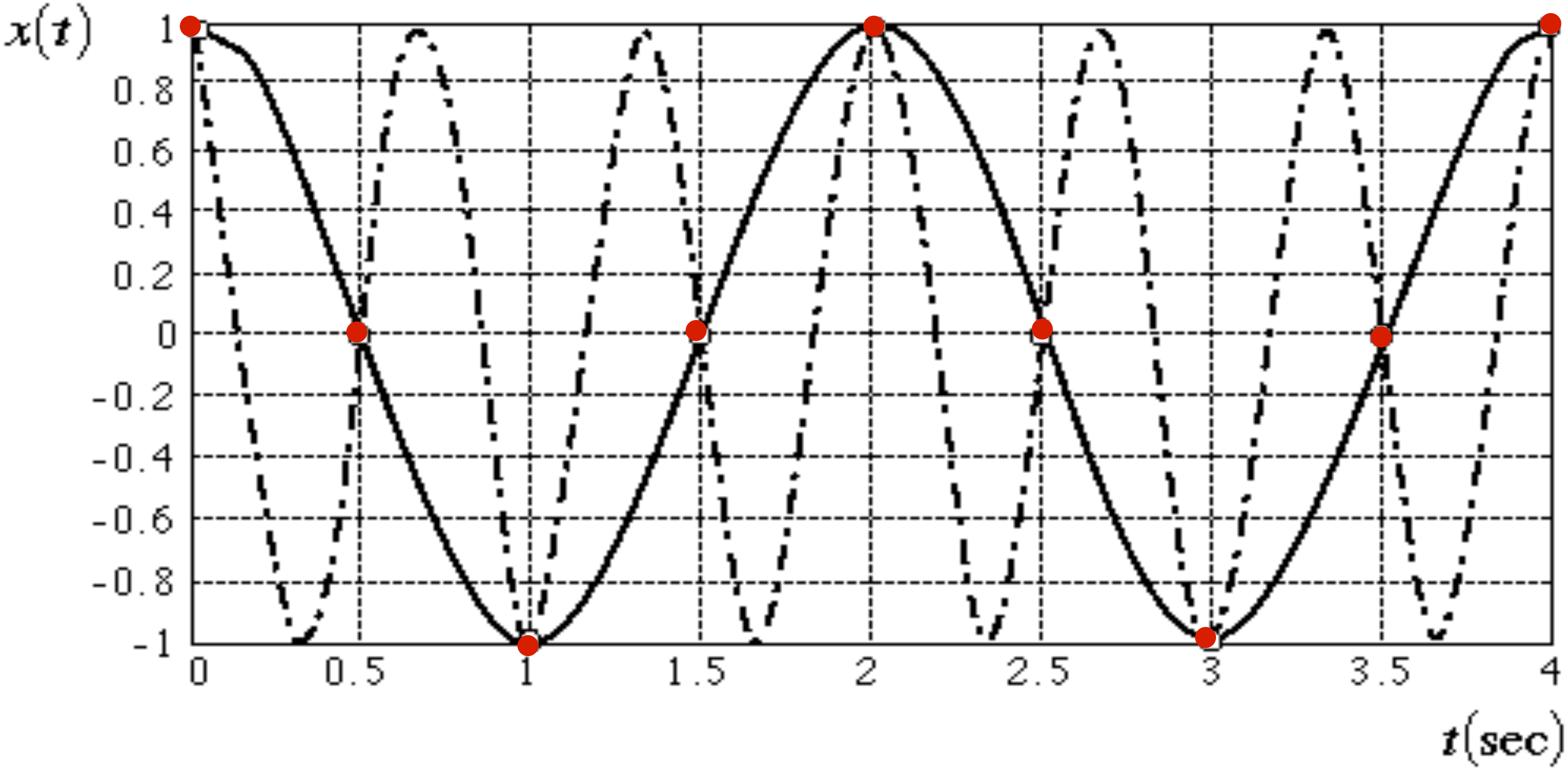
- A bandlimited continuous-time signal, with highest frequency(bandwidth) B Hz, can be uniquely recovered from its samples provided that the sampling rate $F_s \geq 2B$ samples per second.
- The frequency $F_s = 2B$ is called the *Nyquist sampling frequency*.
- If the signal is sampled at less than the Nyquist rate, then the *aliasing* occurs.

Example

$x_1(t) = \cos 2\pi F_1 t, \quad F_1 = 0.5Hz$

$x_2(t) = \cos 2\pi F_2 t, \quad F_2 = 1.5Hz$

$F_S = 2 \ samples/sec \quad \Rightarrow \quad x_1(nTs) = x_2(nTs)$



Natural Sampling

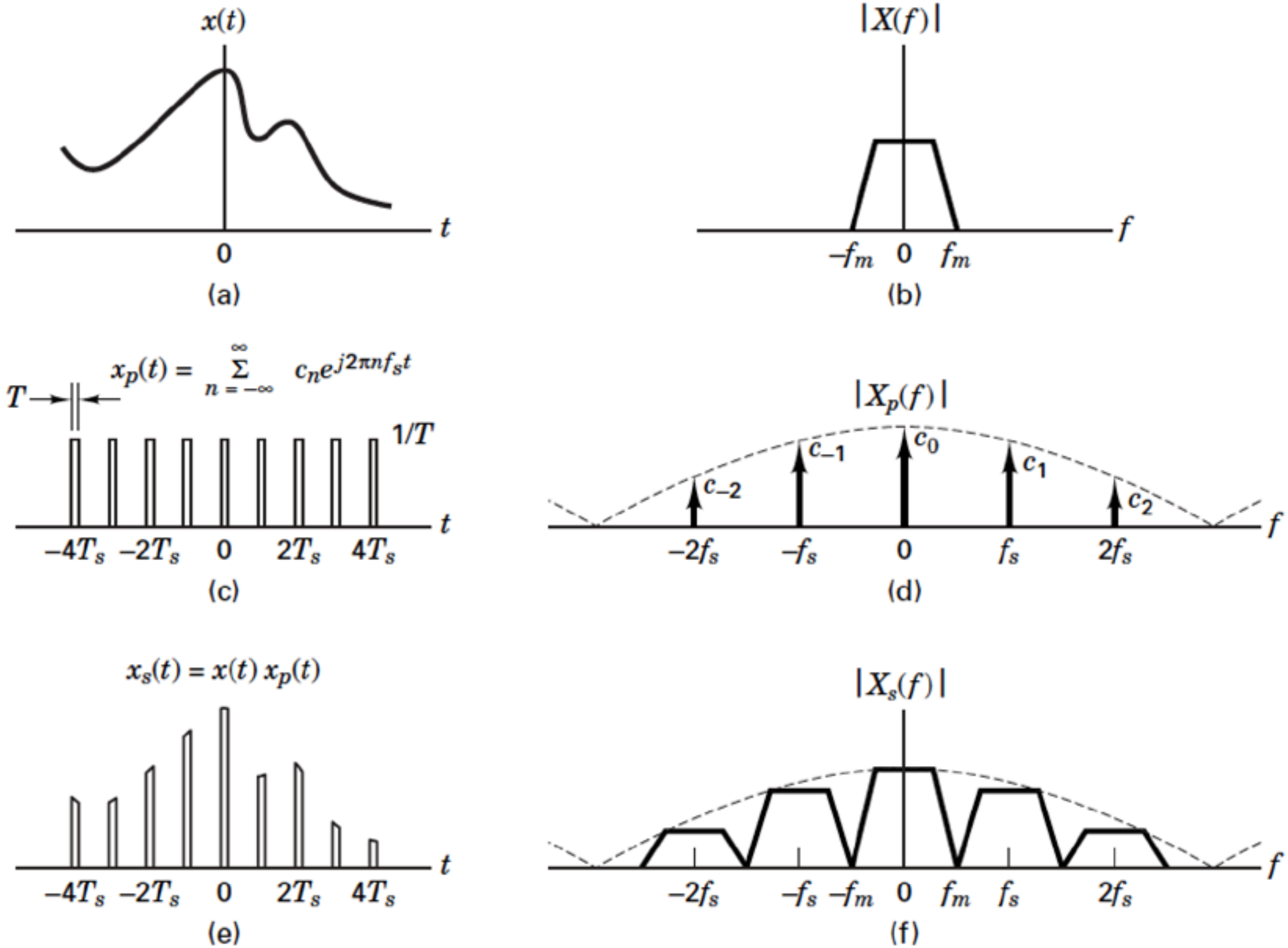
- Replace impulse train in ideal sampling with a pulse train *p*(*t*) (also know as the gating waveform).
- The pulse train

$$p(t) = \sum_{n=-\infty}^{\infty} h(t - nT_s)$$

where $h(t) = 1$ for $0 \leq t \leq \tau$ and $h(t) = 0$ otherwise.

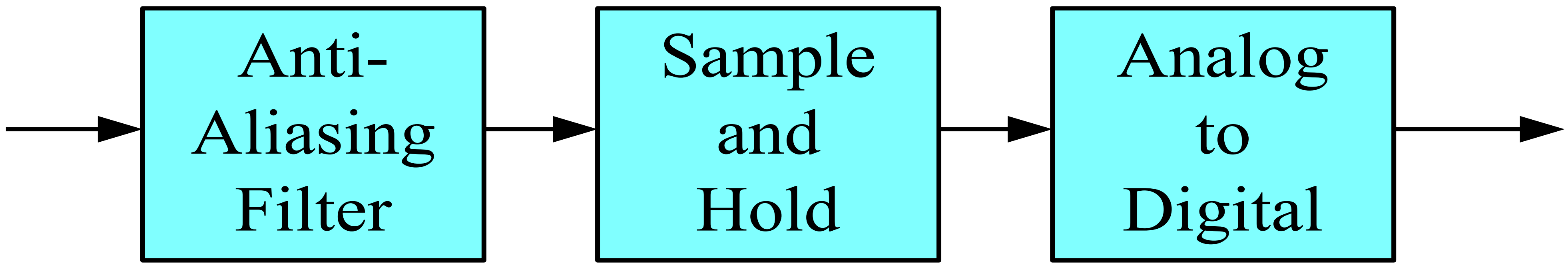
- The pulse train can be implemented by an on/off switch.

Illustration of Natural Sampling



Analog-to-Digital Conversion

- Components : anti-aliasing filter, sample and hold, analog-to-digital converter (quantization).



Block Diagram of an ADC

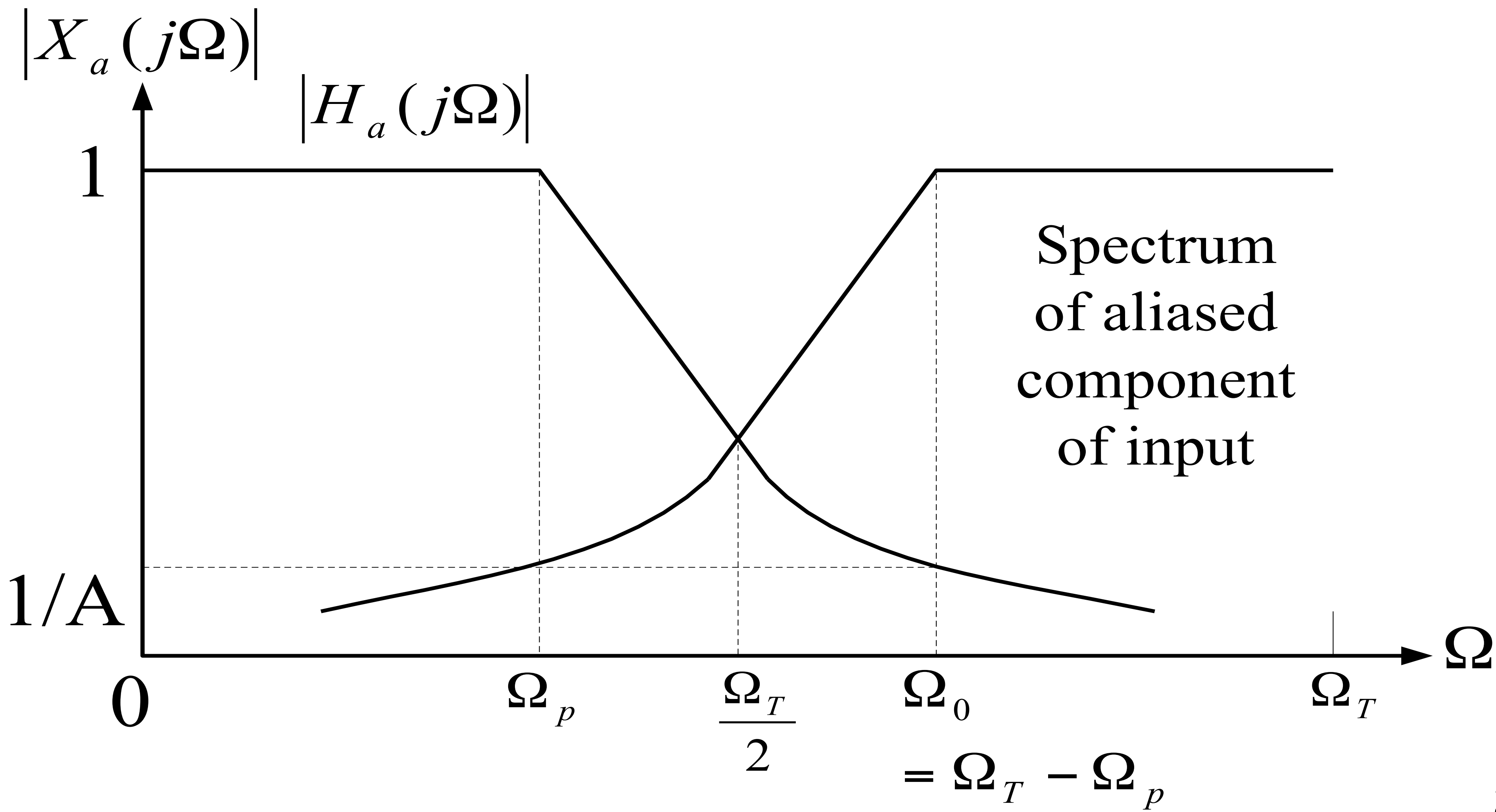
Anti-aliasing Filter

- The role of anti-aliasing filter is to cut off the frequency components that is higher than the half of sampling frequency.
- Ideally, the anti-aliasing filter should have a lowpass frequency response,

$$H_a(j\Omega) = \begin{cases} 1, & |\Omega| < \Omega_T/2 \\ 0, & |\Omega| \geq \Omega_T/2 \end{cases}$$

Such a “brickwall” filter can’t be realized using practical analog circuit, hence, must be approximated.

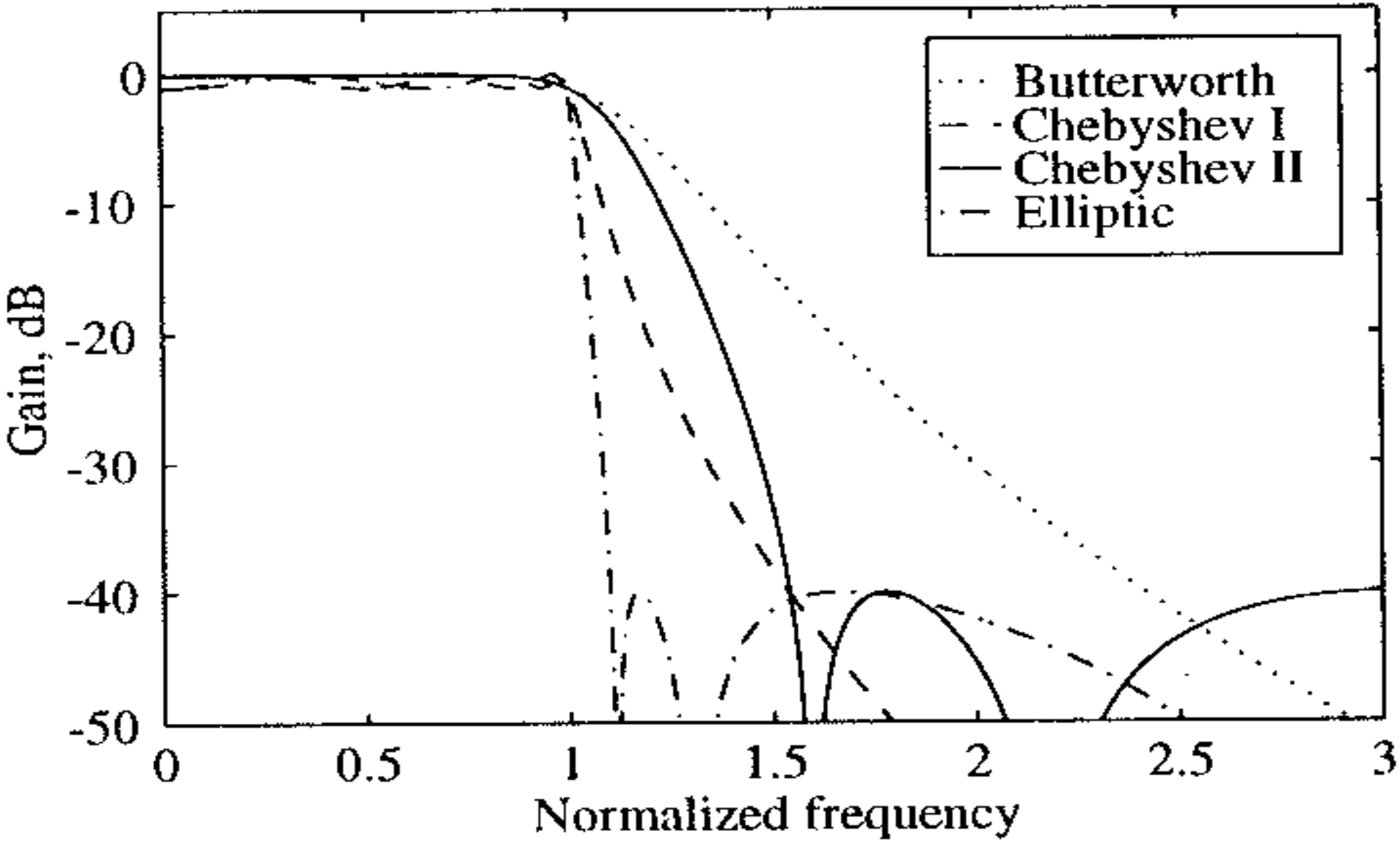
Anti-aliasing Filter’s Effect on Signal Band



Anti-Aliasing Filter Design

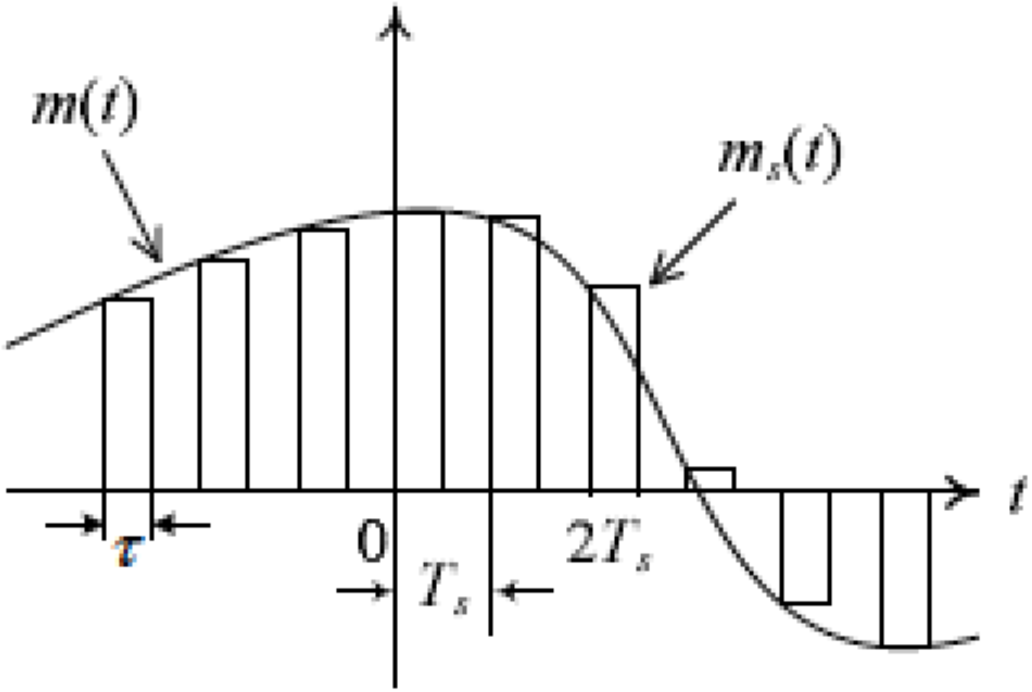
- Requirement :
  1. Approximate linear phase in passband
  2. Passband edge > highest frequency in signal
  3. Stopband edge < half of sampling frequency
- Four types of analog filter
  1. Butterworth filter : good passband, slow roll-off
  2. Chebyshev filter : good roll-off and linear phase
  3. Elliptic filter : fast roll-off, non-linear phase
  4. Bessel filter : close to linear phase, wide transitionband
- Design can be done in Matlab

Frequency Response of 4 Types of Filter



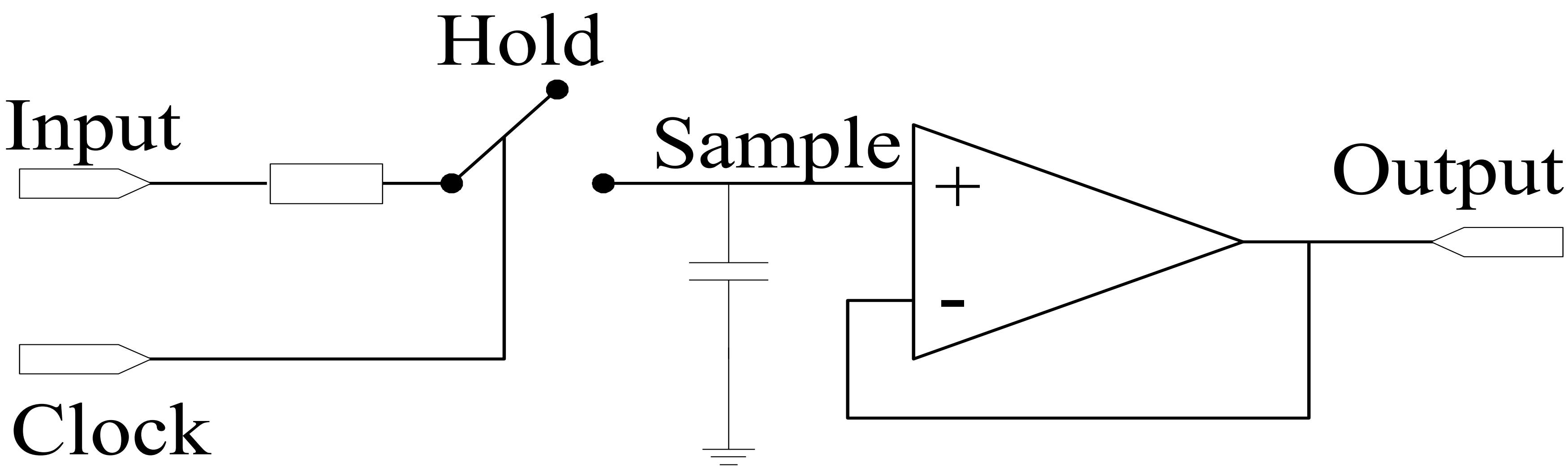
Sample and Hold

- Sample and hold is the most popular sampling method.
- Involves two operations:
  - Sample and hold



Sample and Hold Circuit

- Samples the analog signal at uniform intervals and holds the sampled value after each sampling operation for sufficient time for accurate conversion by the A/D converter.



Analog-to-Digital Converter

- Converts an analog signal into a binary coded digital signal.
- Types of A/D converter

1. Integrating converter
2. Successive approximation converter
3. Flash converter
4. Folding A/D converter
5. Pipelined A/D converter

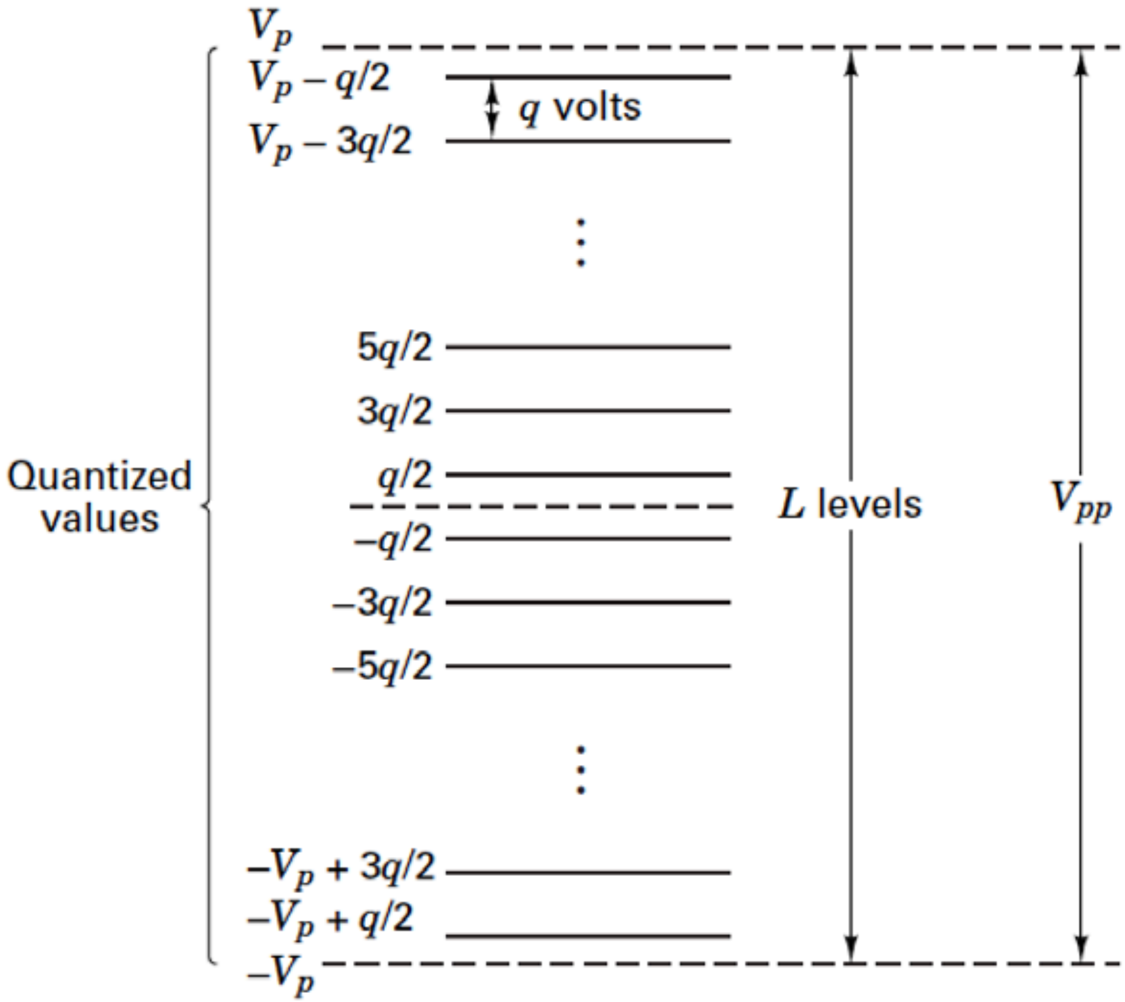
Quantization

A/D Conversion

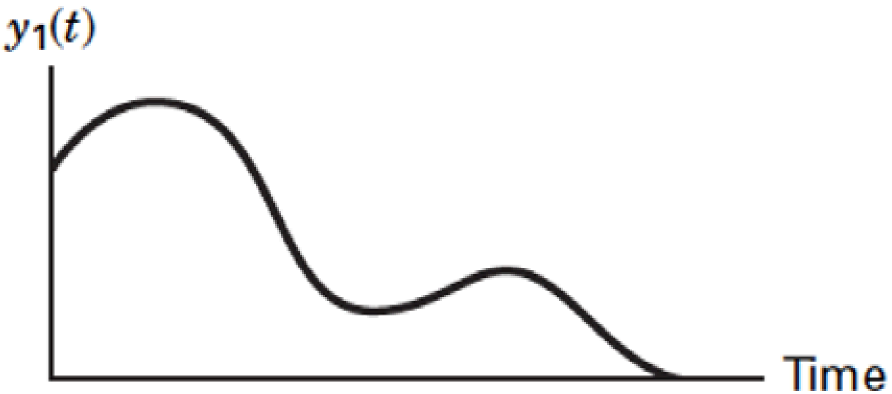
- Uniform quantizer
  - Peak signal power to average quantization noise power is:

$$\left(\frac{S}{N}\right)_q \leq 3L^2$$

  - SNR increases as a function of the number of quantization level squared.



Examples of Sampling

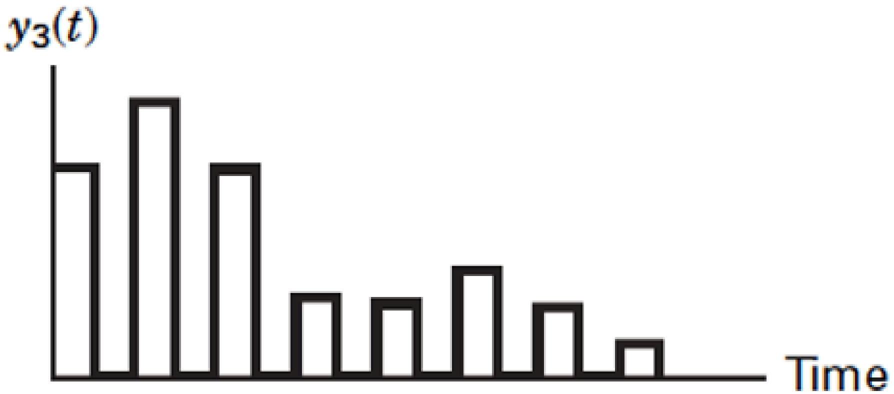


(a)

Original analog waveform

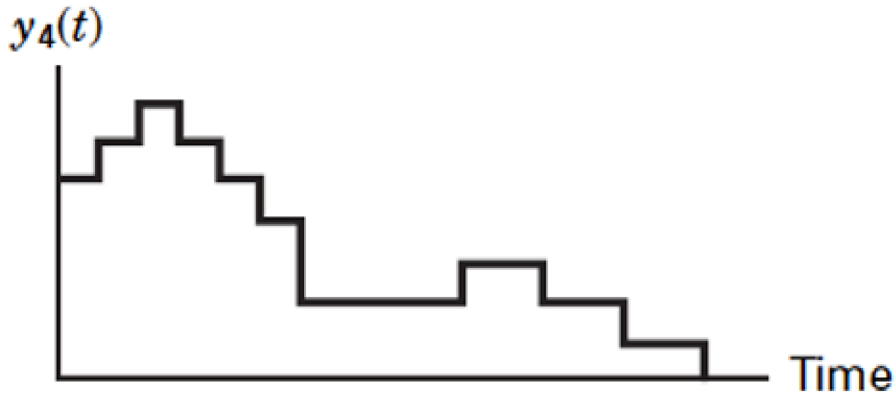
(b)

Natural-sampled data



(c)

Quantized samples



(d)

Sample and Hold

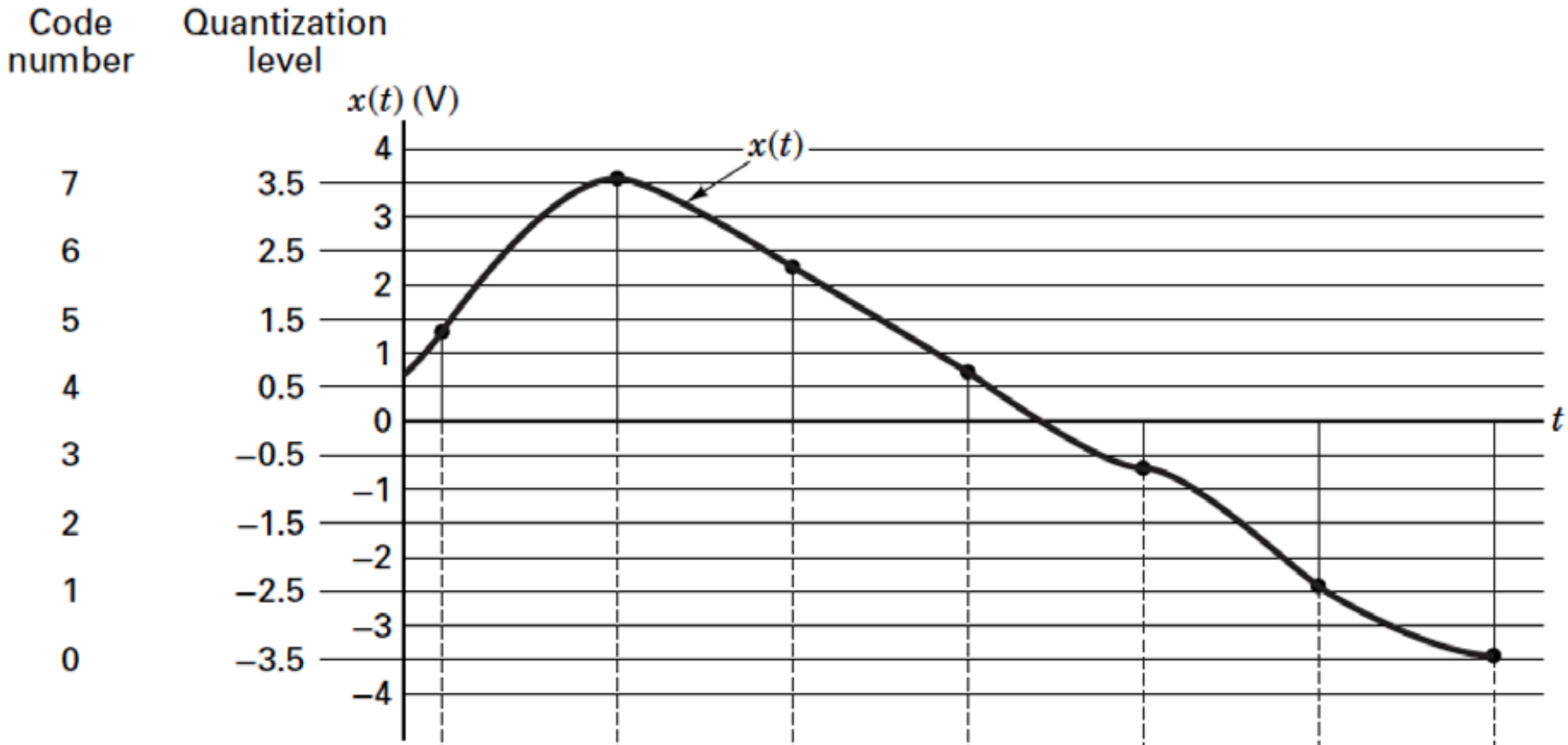
Pulse Code Modulation (PCM)

- In pulse modulation, some parameter of a pulse train is varied in accordance with the sample values of a message signal.
- Pulse-amplitude modulation (PAM)
  - Amplitudes of regularly spaced pulses are varied.
- Pulse-width modulation (PWM)
  - Widths of the individual pulses are varied.
- Pulse-position modulation (PPM)
  - Position of a pulse relative to its original of occurrence is varied.
- Pulse modulation techniques are still analog modulation. For digital communications of an analog source, quantization of sampled values is needed.

PCM

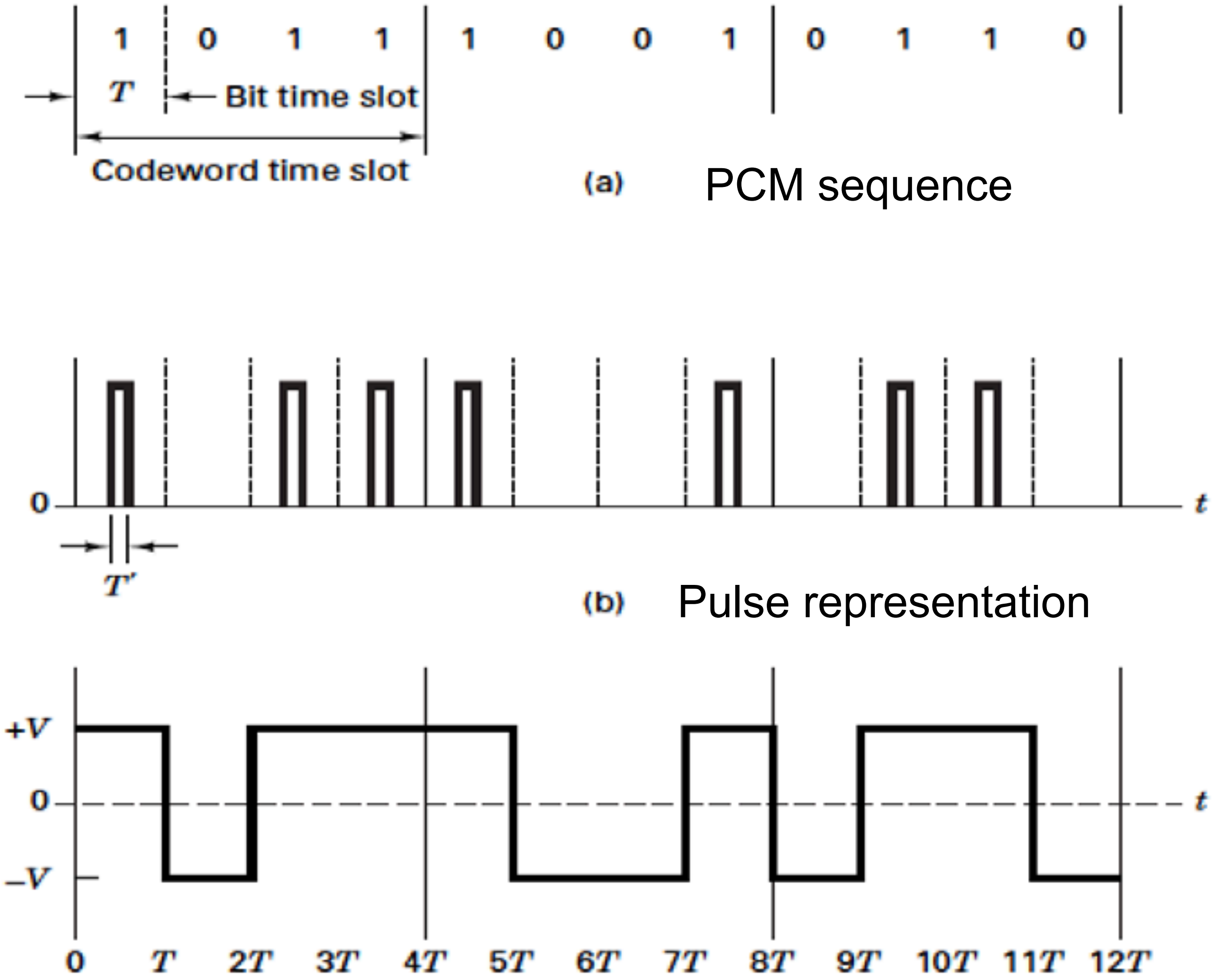
- A PCM signal is obtained from the quantized PAM signal by encoding each quantized sample to a digital codeword
- In binary PCM each quantized sample is digitally encoded into an R-bit binary codeword.
- Binary digits of a PCM signal can be transmitted using many efficient modulation schemes.

PCM - Example



Natural sample value	1.3	3.6	2.3	0.7	–0.7	–2.4	–3.4
Quantized sample value	1.5	3.5	2.5	0.5	–0.5	–2.5	–3.5
Code number	5	7	6	4	3	1	0
PCM sequence	101	111	110	100	011	001	000

PCM Waveform Example



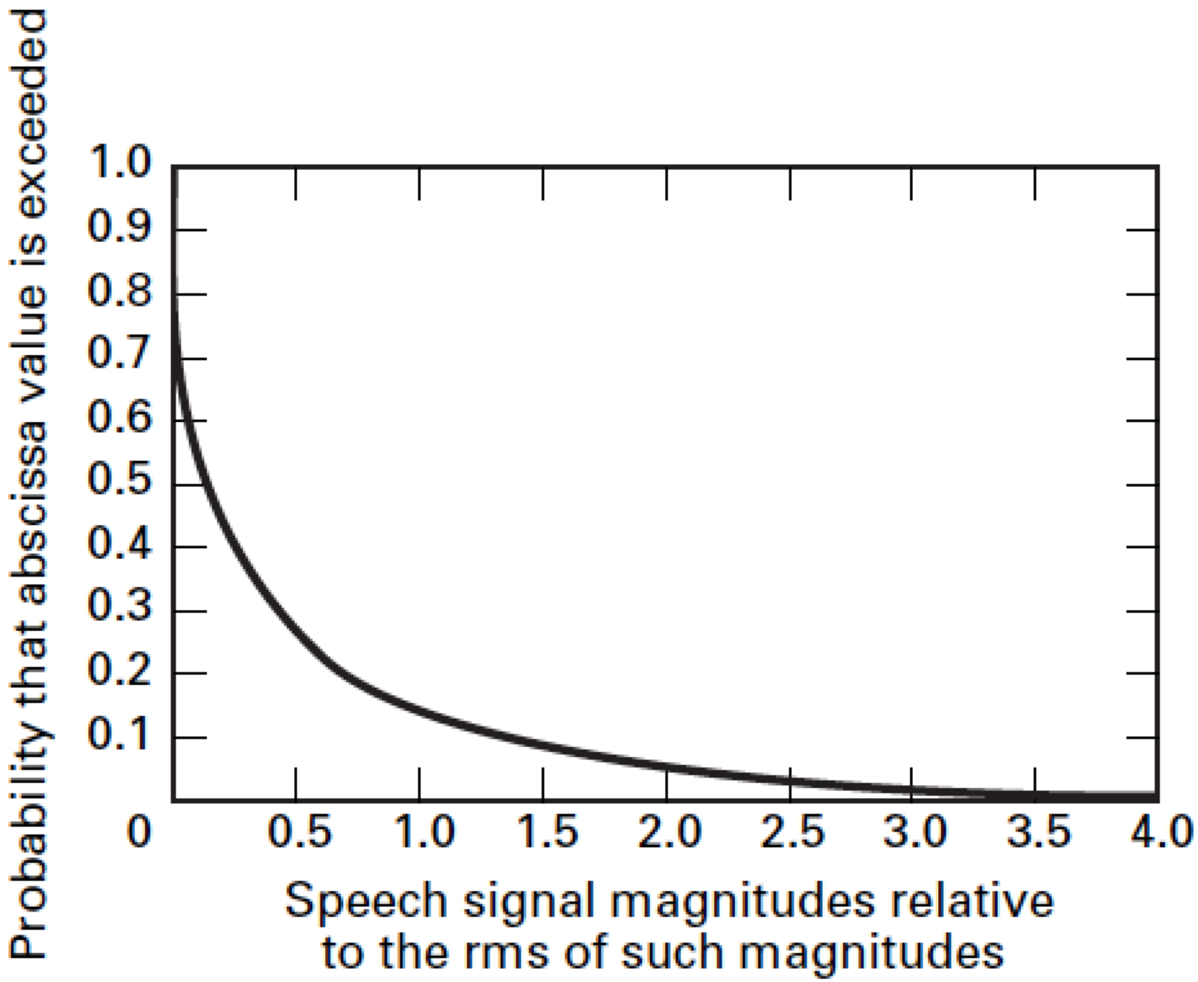
(a) PCM sequence

(b) Pulse representation

(c) Pules waveform

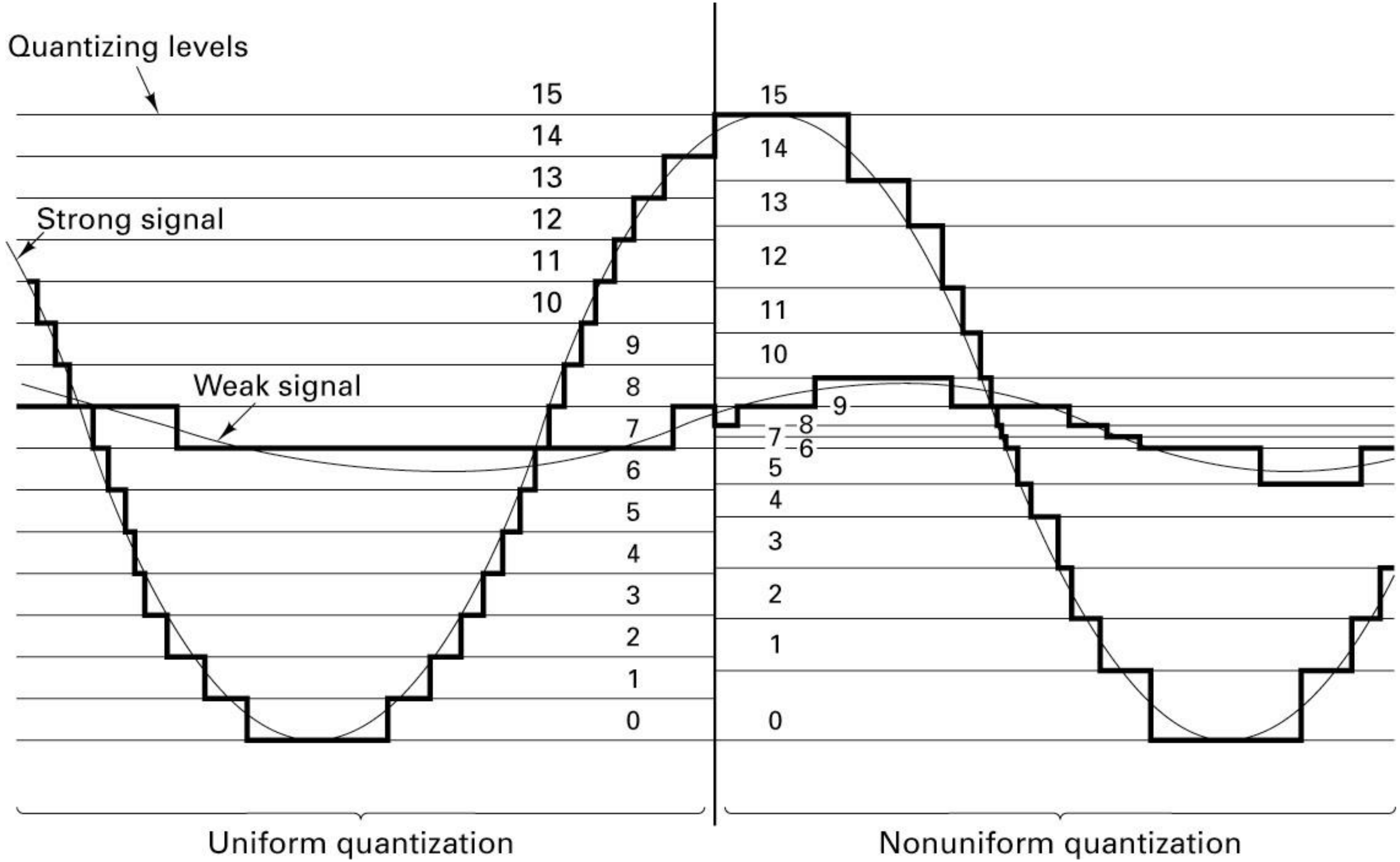
Uniform Quantization (1)

- For most voice communications, very low speech volumes predominate.
- Large amplitudes are very rare while low amplitudes are more often



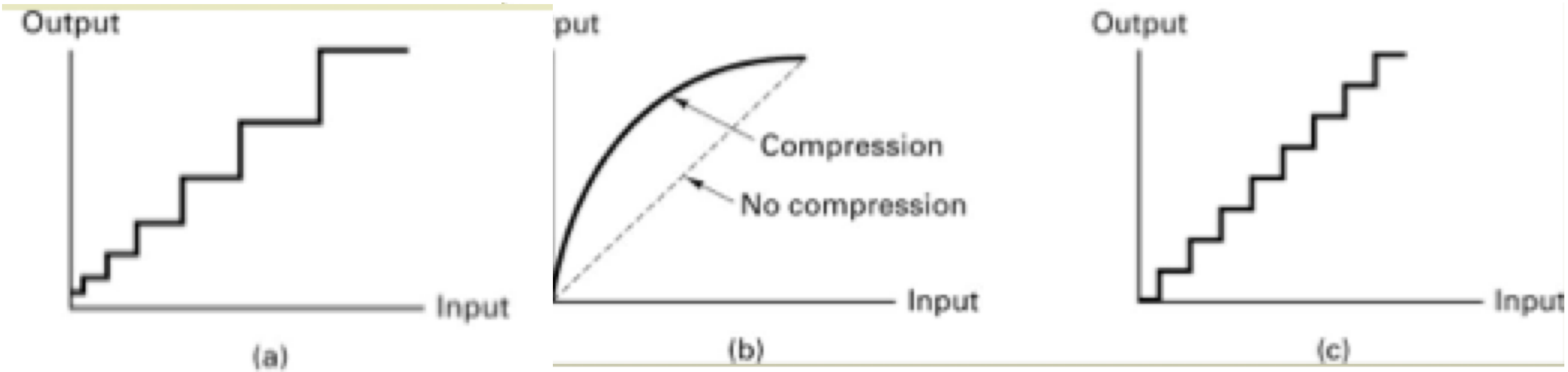
Uniform Quantization (2)

- Using a uniform quantizer for speech signals provides coarse quantization at low amplitudes



Nonuniform Quantization (1)

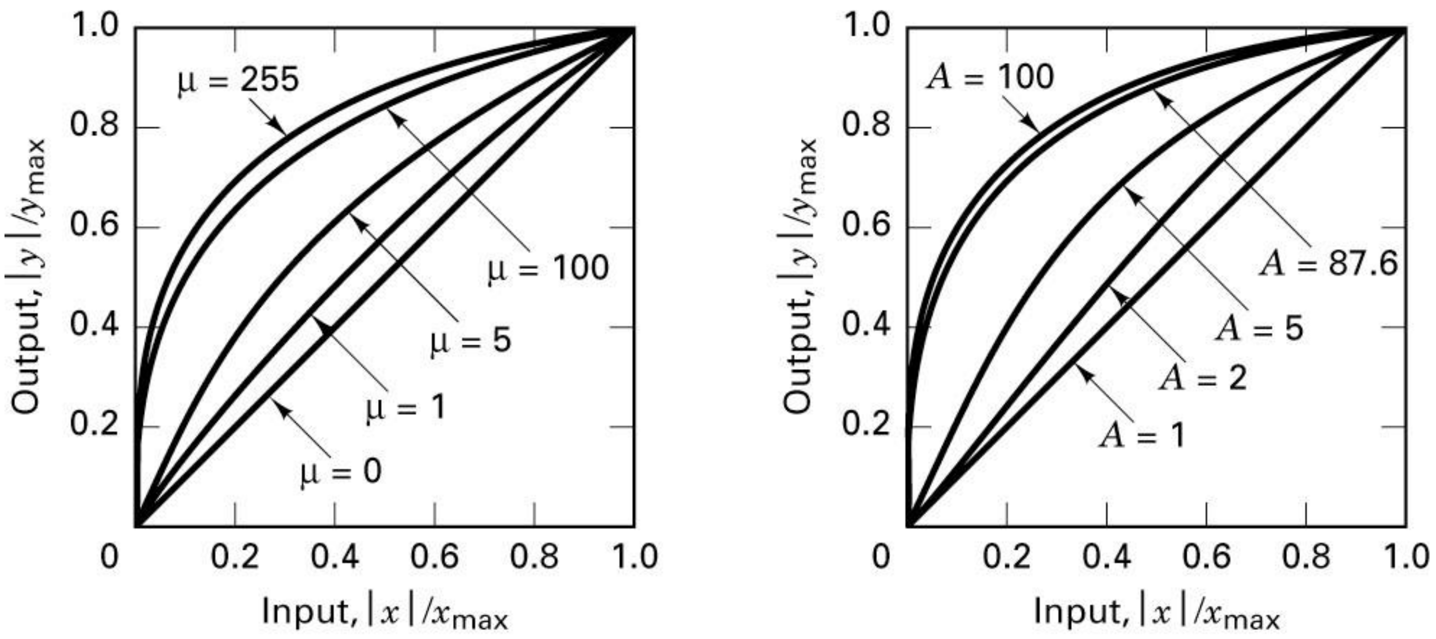
- Nonuniform quantizers are used for speech signals, which provide coarse quantization at high amplitudes and fine quantization at low amplitudes.
- Nonuniform quantization is achieved by the process of companding followed by uniform quantization.



Nonuniform Quantization (2)

- Two commonly used companders are:

μ – law compander	A – law compander
$y = y_{max} \dfrac{\log_e\left[1+\mu(\lvert x\rvert / x_{max})\right]}{\log_e\left[1+\mu\right]} \operatorname{sgn}(x)$	$y = \begin{cases} y_{max} \dfrac{A(\lvert x\rvert / x_{max})}{1+\log_e A} \operatorname{sgn}(x) & 0 < \dfrac{\lvert x\rvert}{x_{max}} \le \dfrac{1}{A} \\ y_{max} \dfrac{1+\log_e\left[A(\lvert x\rvert / x_{max})\right]}{1+\log_e A} \operatorname{sgn}(x) & \dfrac{1}{A} < \dfrac{\lvert x\rvert}{x_{max}} \le 1 \end{cases}$



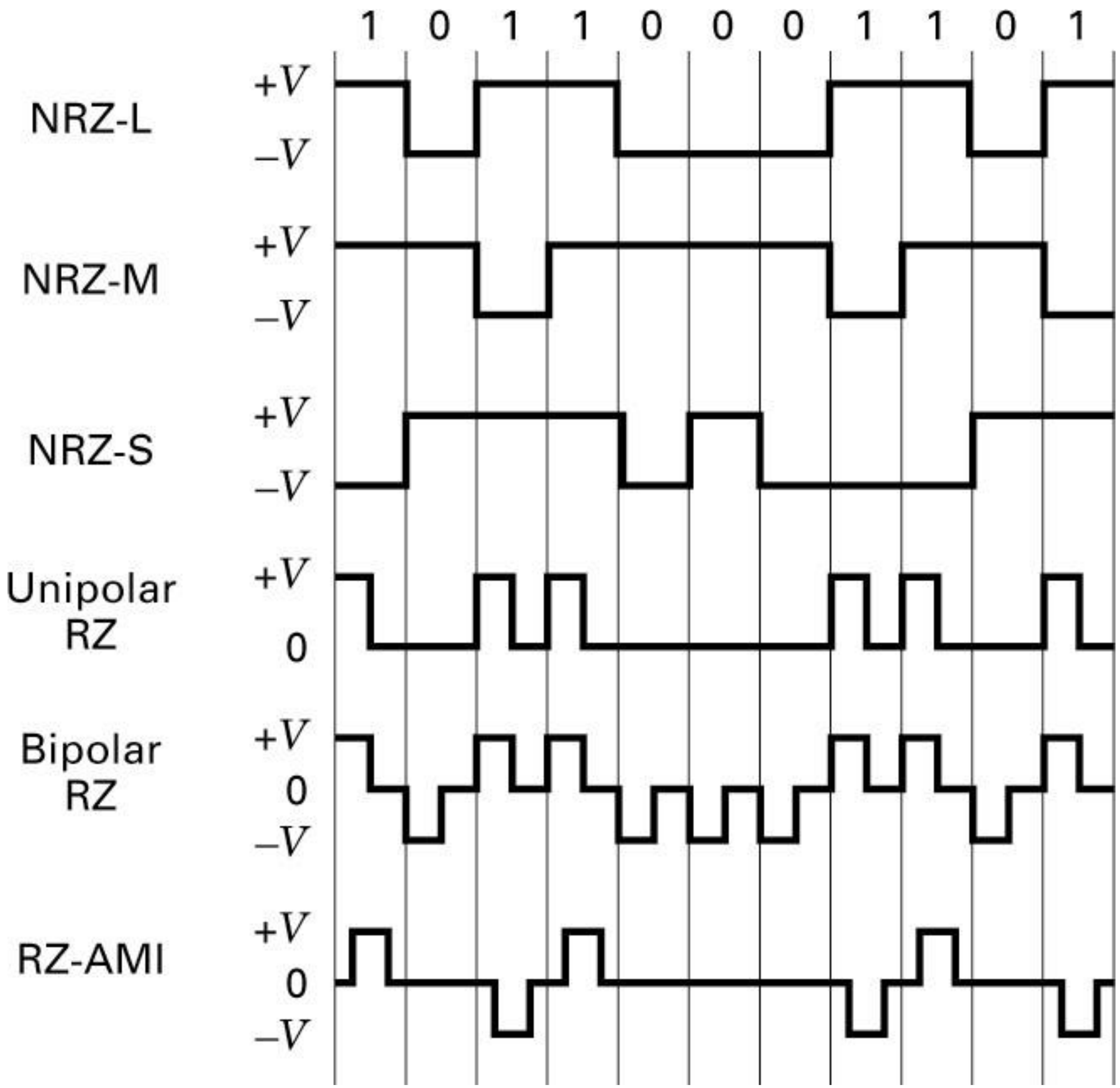
(a) (b)

Baseband Transmission

PCM Waveform Types

- Nonreturn-to-zero (NRZ)
  - NRZ is most commonly used PCM waveform
  - NRZ-L (L for level)
  - NRZ-M (M for mark)
  - NRZ-S (S for space)
- Return-to-zero (RZ)
  - Unipolar-RZ, bipolar-RZ, RZ-AMI(alternate mark inversion)
- Phase encoded
- Multilevel binary

PCM Coding (1)



1 = 1 voltage level,
0=another voltage level

1 (mark)=change in level,
0 (space) = no change in level

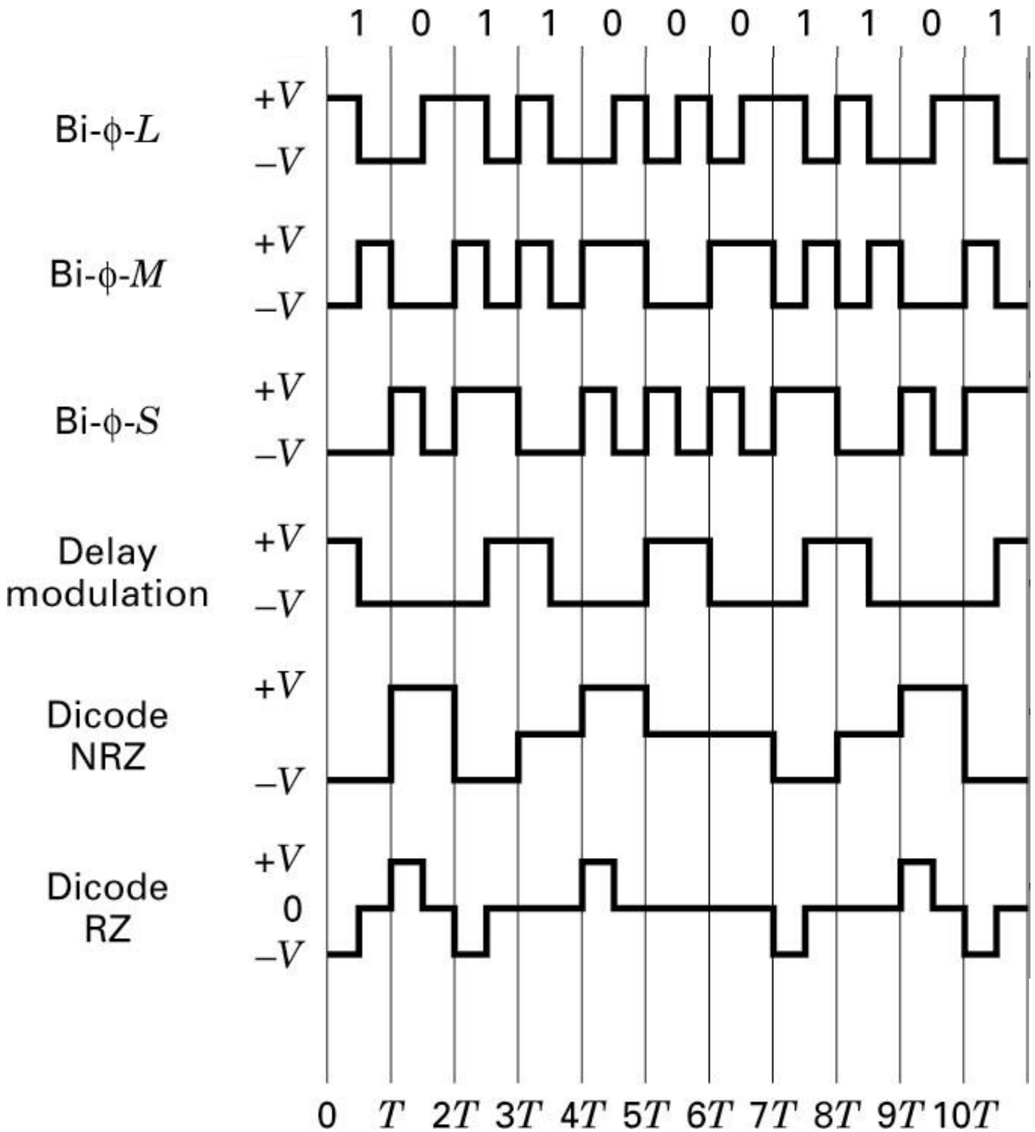
A complement of NRZ-M

1=half-bit-wide pulse,
0=no pulse

1 and 0 by opposite-level pulses

1=equal amplitude alternating pulse
0=no pulse

PCM Coding (2)



1 0 1 1 0 0 0 1 1 0 1
Bi-ϕ-L
Bi-ϕ-M
Bi-ϕ-S
Delay modulation
Dicode NRZ
Dicode RZ
+V
0
–V
0 T 2T 3T 4T 5T 6T 7T 8T 9T 10T

Bits per PCM Word and Bits per Symbol

- PCM word size
  - How many bits shall we assign to each analog sample?

$$|e| \le pV_{pp}$$

$$|e_{\max}| = \frac{q}{2} = \frac{V_{pp}}{2L}$$

$$\frac{V_{pp}}{2L} \le pV_{pp} \rightarrow 2^l = L \le \frac{1}{2p}$$

$$l \ge \log_2\left(\frac{1}{2p}\right)$$

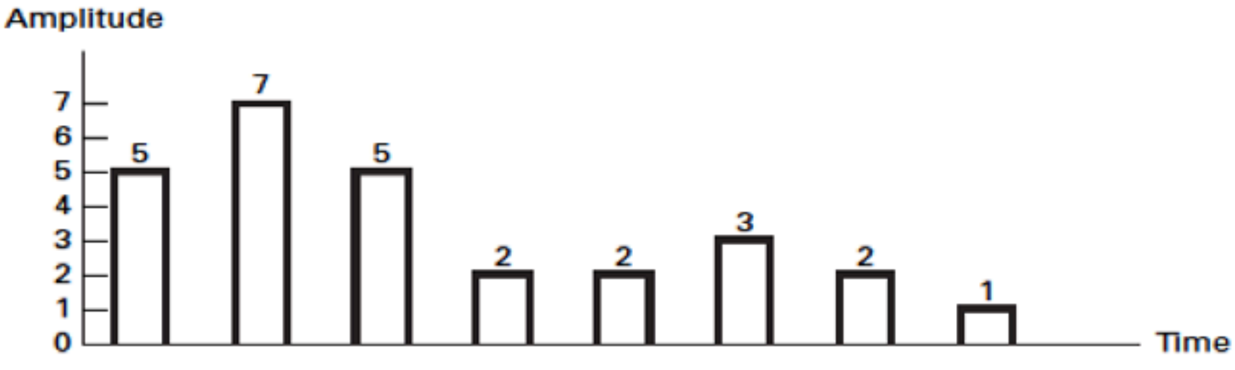
e: quantization error,

V_{pp} peak-to-peak voltage

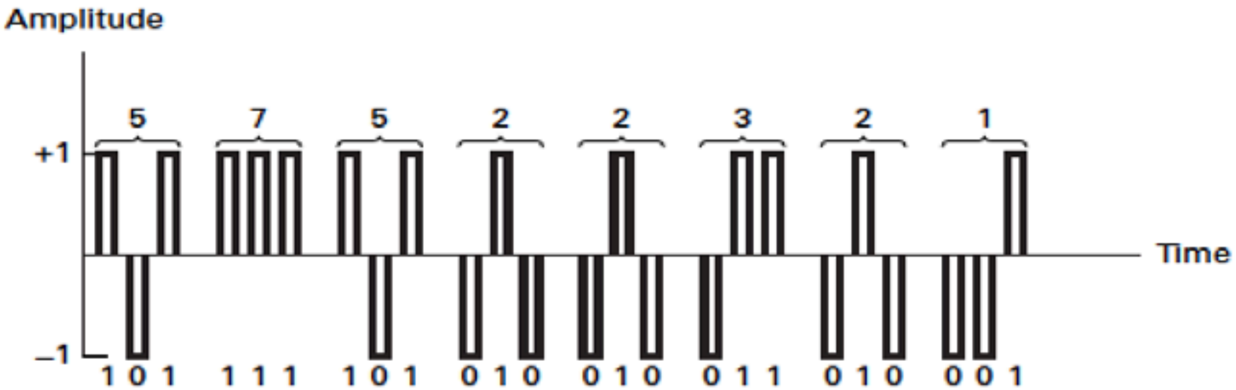
q: quantization level

M-ary Pulse-Modulation

- Multilevel signaling - a group of k-bit is transmitted by $M=2^k$ level pulse.



(a)



(b)

Activity 1

The information in an analog waveform, with maximum frequency f_m=3kHz, is to be transmitted over an M-ary PAM system, where the number of pulse levels is M=16. The quantization error is specified not to exceed (+/-)1% of the peak-to-peak analog signal.

(a) What is the minimum number of bits/samples, or PCM word size that should be used in digitizing the analog waveform?

(b) What is the minimum required sampling rate, and what is the resulting bit transmission rate?

(c) What is the PAM pulse or symbol transmit rate?

(d) If the transmission bandwidth equals 12 kHz, determine the bandwidth efficiency for this system.