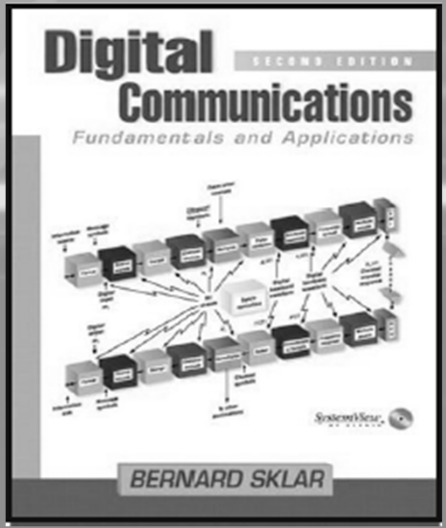



# ENE 467
# Digital Communications

**TEACHING BY**

**ASST. PROF. SUWAT PATTARAMALAI, PH.D.**

# 2. Formatting and Baseband Modulation

- Outcome
  - Can explain Baseband systems and formatting texture data
  - Can design analog formatting procedure (Sampling, Aliasing, oversample)
  - Can explain source corruption (effects of quantizing and channel)
  - Can design Pulse Code Modulation
  - Can explain the differences of uniform and non-uniform Quantization
  - Can explain Baseband modulation (waveform types, bit per symbol, M-ary)
  - Can explain Correlative Coding

# Formatting and Baseband Modulation

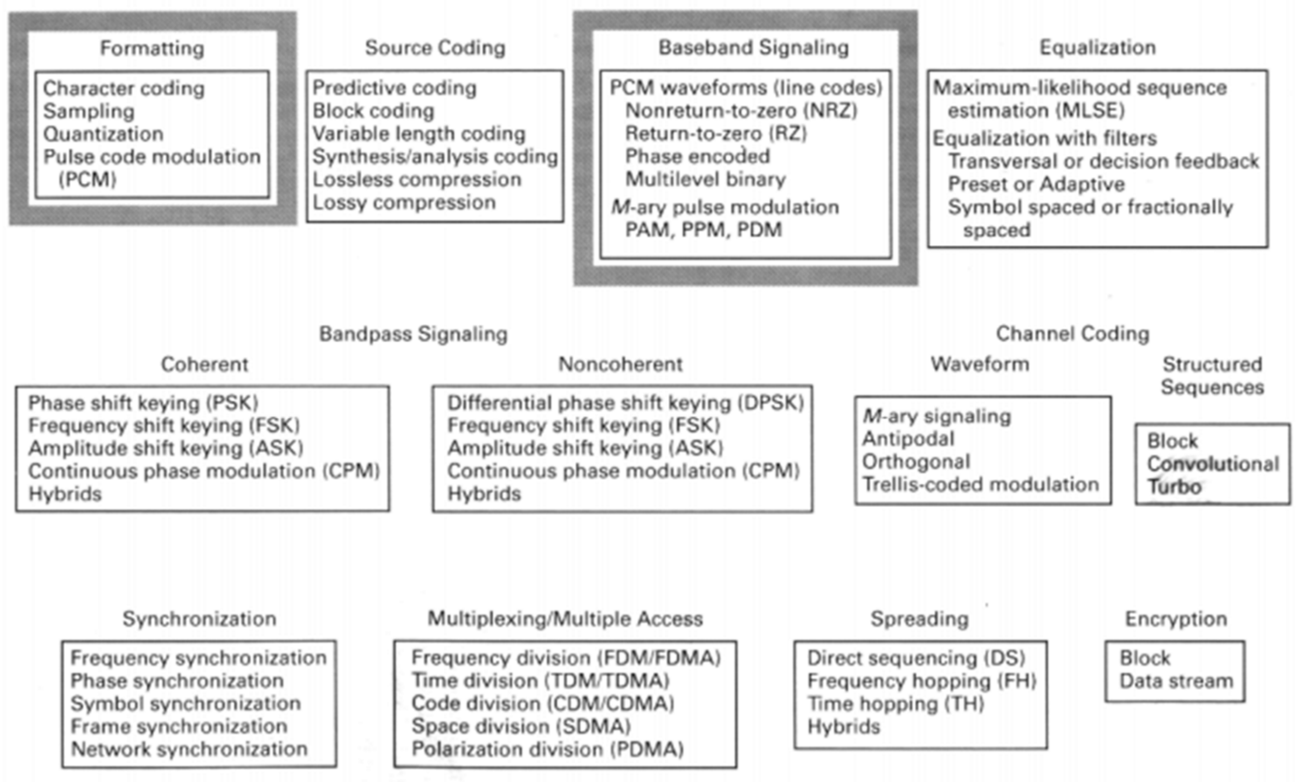



**Figure 2.1** Basic digital communication transformations

# Formatting and Baseband Modulation

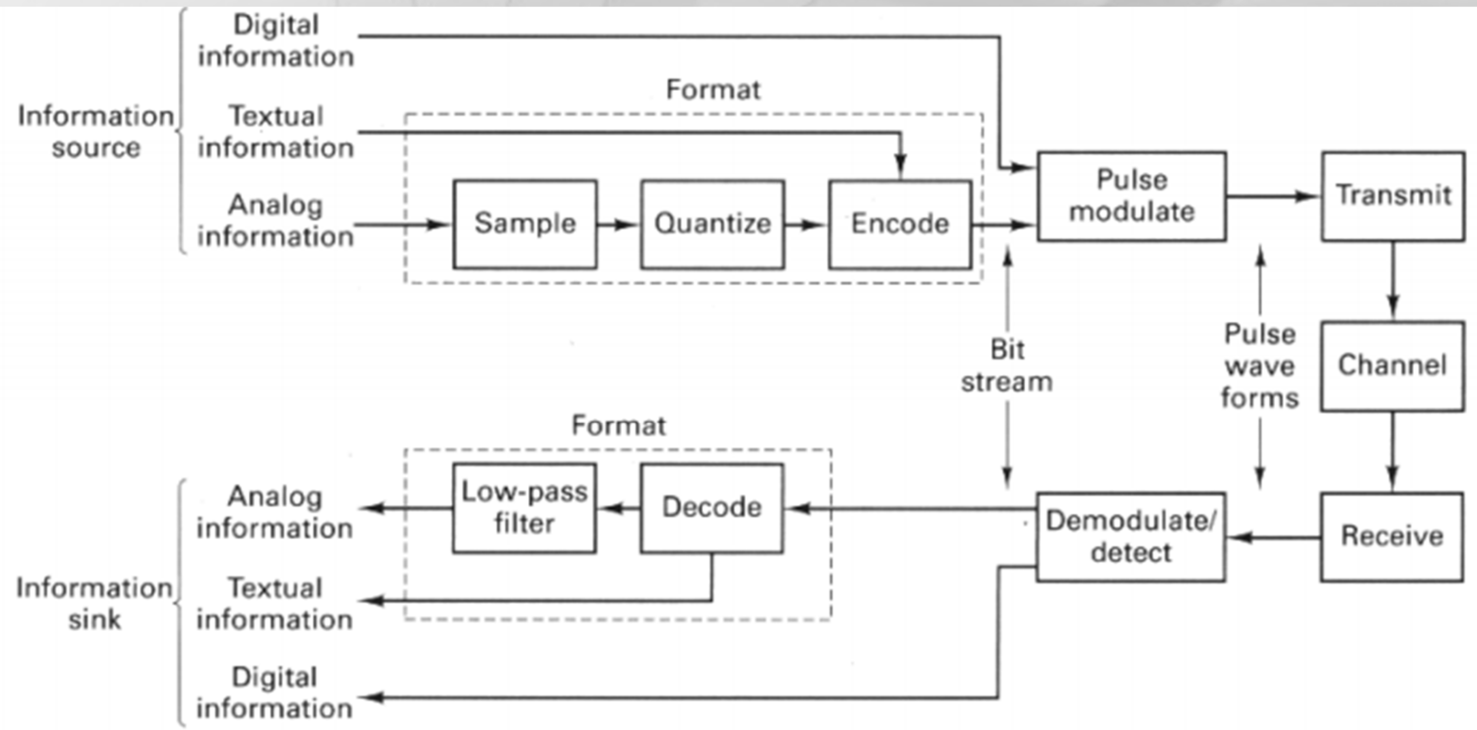



**Figure 2.2** Formatting and transmission of baseband signals.

# Formatting Textual Data (Character Coding)

| Bits | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | | | | 5 | 0 | 1 | 0 | 1 | 0 | 1 | 0 | 1 |
| | | | | 6 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 |
| | | | | 7 | 0 | 0 | 0 | 0 | 1 | 1 | 1 | 1 |
| 1 | 2 | 3 | 4 | | | | | | | | | |
| 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
SOH Start of heading
STX Start of text
ETX End of text
EOT End of transmission
ENQ Enquiry
ACK Acknowledge
BEL Bell, or alarm
BS Backspace
HT Horizontal tabulation
LF Line feed
VT Vertical tabulation
FF Form feed
CR Carriage return
SO Shift out
SI Shift in
DLE Data link escape

DC1 Device control 1
DC2 Device control 2
DC3 Device control 3
DC4 Device control 4
NAK Negative acknowledge
SYN Synchronous idle
ETB End of transmission
CAN Cancel
EM End of medium
SUB Substitute
ESC Escape
FS File separator
GS Group separator
RS Record separator
US Unit separator
SP Space
DEL Delete

**Figure 2.3** Seven-bit American standard code for information interchange (ASCII).

# Formatting Textual Data (Character Coding)

| Bits | 5 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | 6 | 0 | 0 | 0 | 0 | 1 | 1 | 1 | 1 | 0 | 0 | 0 | 0 | 1 | 1 | 1 | 1 |
| | 7 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 |
| | 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 transmissio |
| BYP | Bypass |
| LF | Line feed |
| EOB | End of block |
| PRE | Prefix (ESC) |
| RS | Reader stop |
| SM | Start message |
| DS | Digit select |
| SOS | Start of significanc |
| IFS | Interchange file separator |
| IGS | Interchange group separator |
| IRS | Interchange record separator |
| IUS | Interchange unit separator |
| Others | Same as ASCII |

**Figure 2.4** EBCDIC character code set.

# Message, Characters, and Symbols

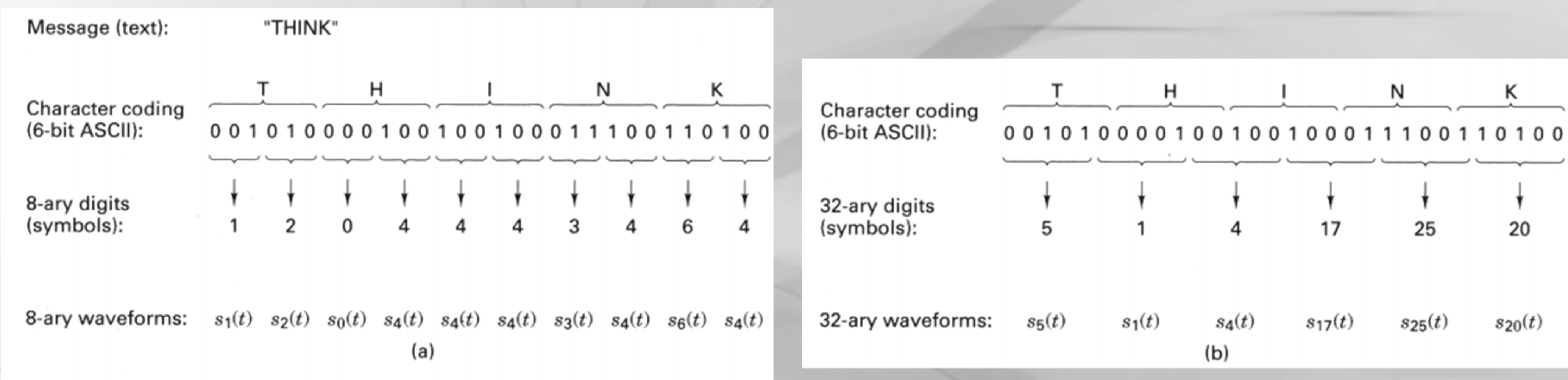



**Figure 2.5** Messages, characters, and symbols. (a) 8-ary example. (b) 32-ary example.

# Formatting Analog Information

- Sampling

$$f_s \geq 2f_m$$

$$T_s \leq \frac{1}{2f_m} \text{ sec}$$

- Impulse Sampling

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

$$X_\delta(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

$$\begin{aligned} x_s(t) = x(t)x_\delta(t) &= \sum_{n=-\infty}^{\infty} x(t)\delta(t - nT_s) \\ &= \sum_{n=-\infty}^{\infty} x(nT_s)\delta(t - nT_s) \end{aligned}$$

$$\begin{aligned} X_s(f) = X(f) * X_\delta(f) &= X(f) * \left[\frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)\right] \\ &= \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - nf_s) \end{aligned}$$

# Impulse Sampling

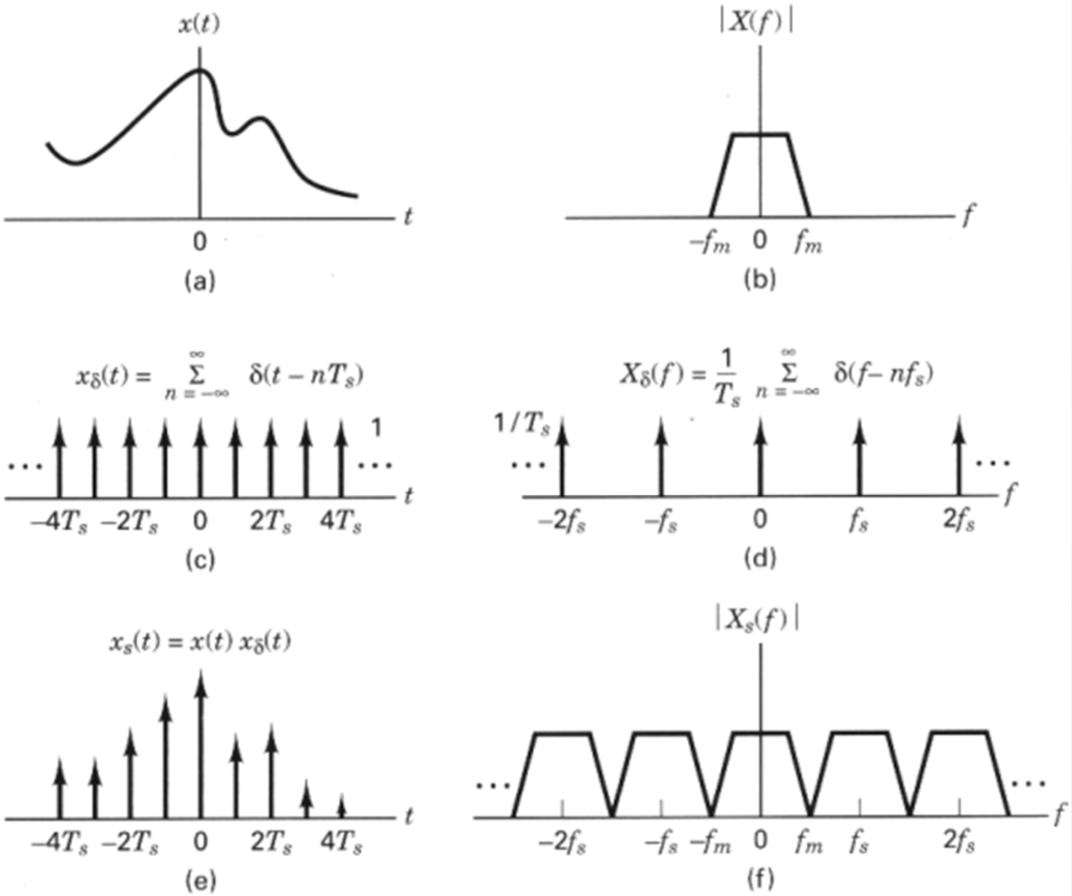



**Figure 2.6** Sampling theorem using the frequency convolution property of the Fourier transform.

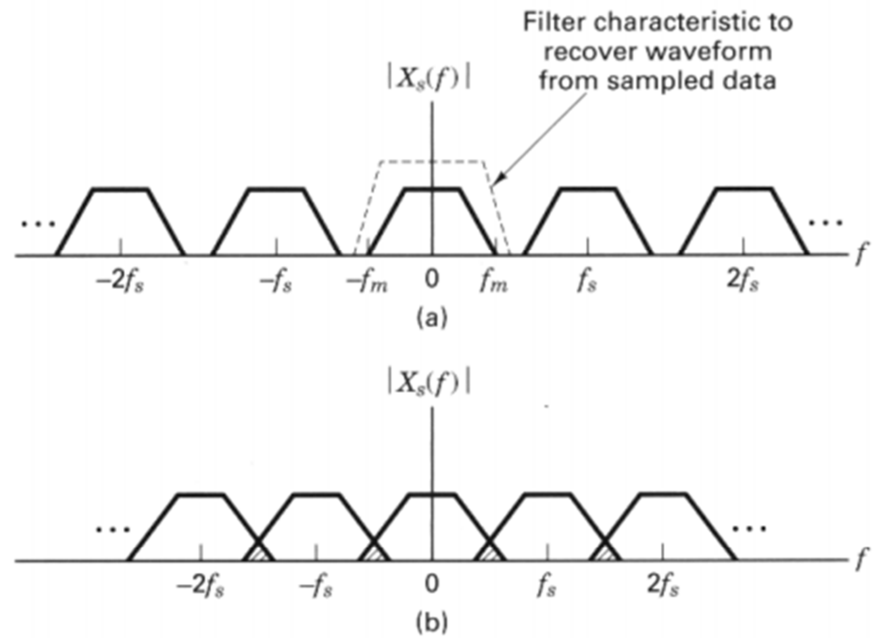



**Figure 2.7** Spectra for various sampling rates. (a) Sampled spectrum ($f_s > 2f_m$). (b) Sampled spectrum ($f_s < 2f_m$).

# Natural Sampling

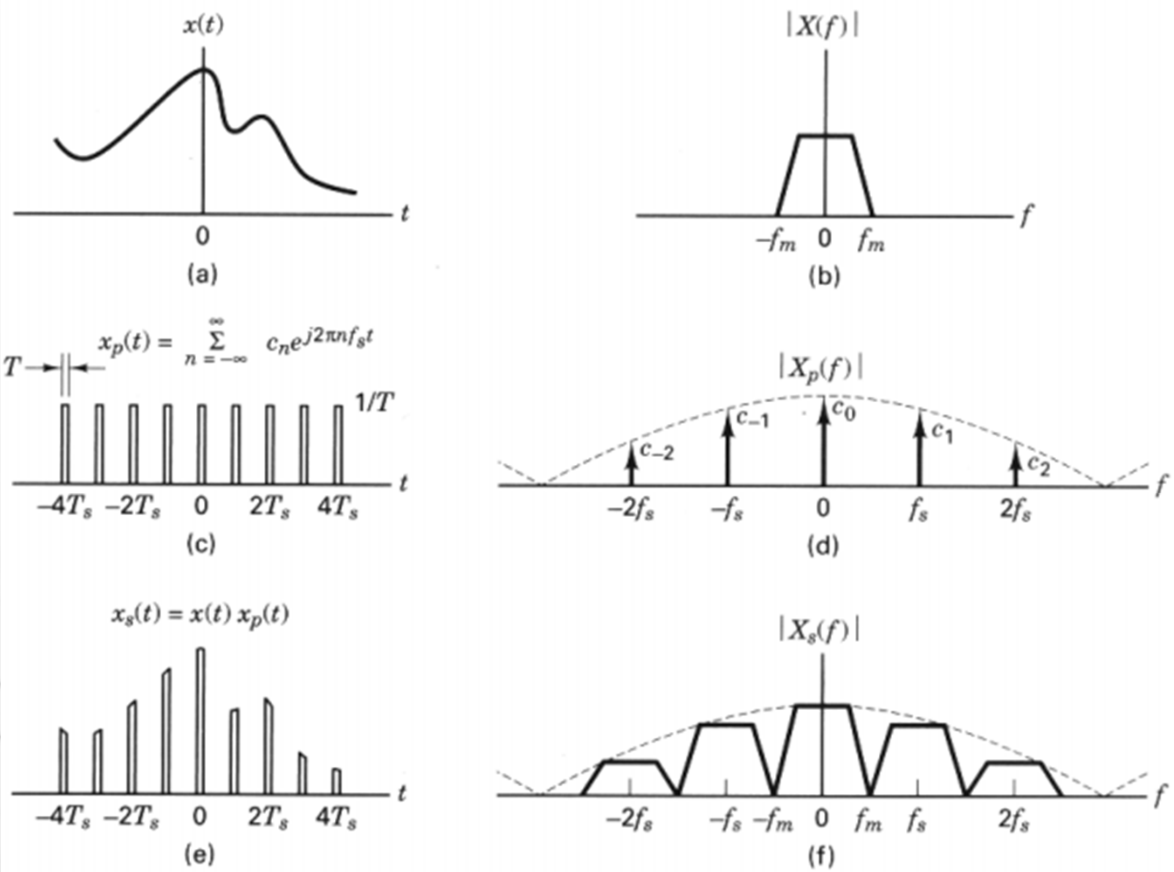



**Figure 2.8** Sampling theorem using the frequency shifting property of the Fourier transform.

$$x_s(t) = x(t)x_p(t)$$

$$x_p(t) = \sum_{n=-\infty}^{\infty} c_n e^{j 2\pi n f_s t}$$

$$x_s(t) = x(t) \sum_{n=-\infty}^{\infty} c_n e^{j 2\pi n f_s t}$$

$$X_s(f) = \mathscr{F}\left\{x(t) \sum_{n=-\infty}^{\infty} c_n e^{j 2\pi n f_s t}\right\}$$

$$X_s(f) = \sum_{n=-\infty}^{\infty} c_n \mathscr{F}\{x(t)e^{j 2\pi n f_s t}\}$$

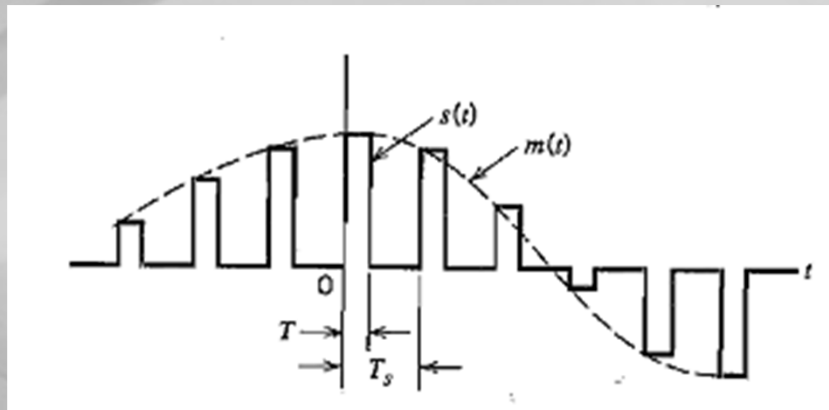
$$X_s(f) = \sum_{n=-\infty}^{\infty} c_n X(f - nf_s)$$

# Sample and Hold Operation

unity amplitude rectangular pulse $p(t)$ of pulse width $T_s$.

$$x_s(t) = p(t) * [x(t)x_\delta(t)]$$
$$= p(t) * \left[x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)\right]$$

$$X_s(f) = P(f)\mathcal{F}\left\{x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)\right\}$$
$$= P(f)\left\{X(f) * \left[\frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)\right]\right\}$$
$$= P(f)\frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - nf_s)$$




$$\operatorname{sinc} y = \frac{\sin \pi y}{\pi y}$$

$P(f)$ is of the form $T_s \operatorname{sinc} fT_s$.

# Aliasing

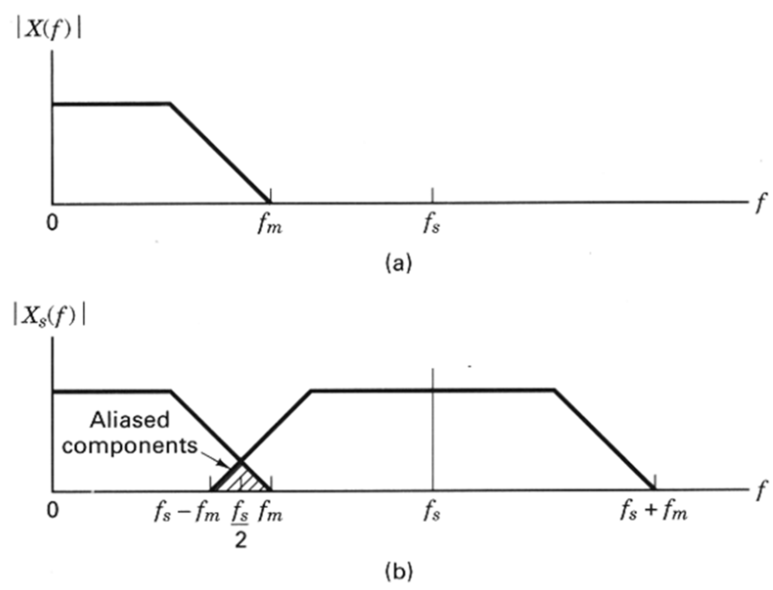



**Figure 2.9** Aliasing in the frequency domain. (a) Continuous signal spectrum. (b) Sampled signal spectrum.

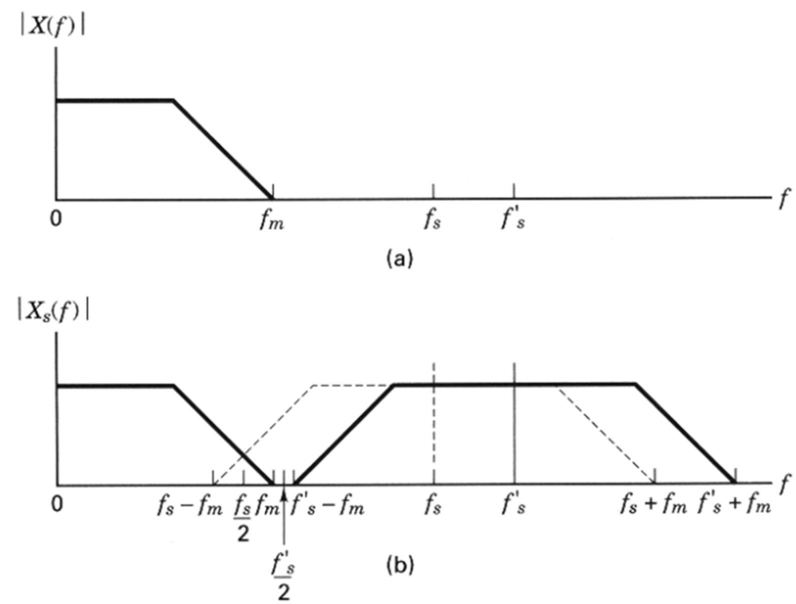



**Figure 2.10** Higher sampling rate eliminates aliasing. (a) Continuous signal spectrum. (b) Sampled signal spectrum.

# Why Oversample?

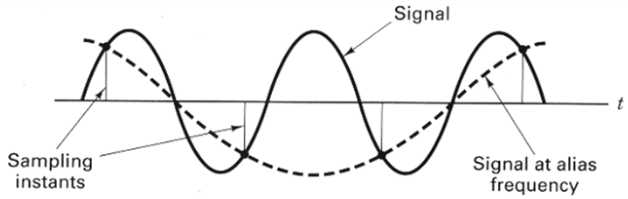


**Figure 2.13** Alias frequency generated by sub-Nyquist sampling rate.

***Without Oversampling***

1. The signal passes through a high performance analog lowpass filter to limit its bandwidth.
2. The filtered signal is sampled at the Nyquist rate for the (approximated) bandlimited signal. As described in Section 1.7.2, a strictly bandlimited signal is not realizable.
3. The samples are processed by an analog-to-digital converter that maps the continuous-valued samples to a finite list of discrete output levels.

***With Oversampling***

1. The signal is passed through a low performance (less costly) analog low-pass filter (prefilter) to limit its bandwidth.
2. The pre-filtered signal is sampled at the (now higher) Nyquist rate for the (approximated) bandlimited signal.
3. The samples are processed by an analog-to-digital converter that maps the continuous-valued samples to a finite list of discrete output levels.
4. The digital samples are then processed by a high performance digital filter to reduce the bandwidth of the digital samples.
5. The sample rate at the output of the digital filter is reduced in proportion to the bandwidth reduction obtained by this digital filter.

# Signal Interface for a Digital System

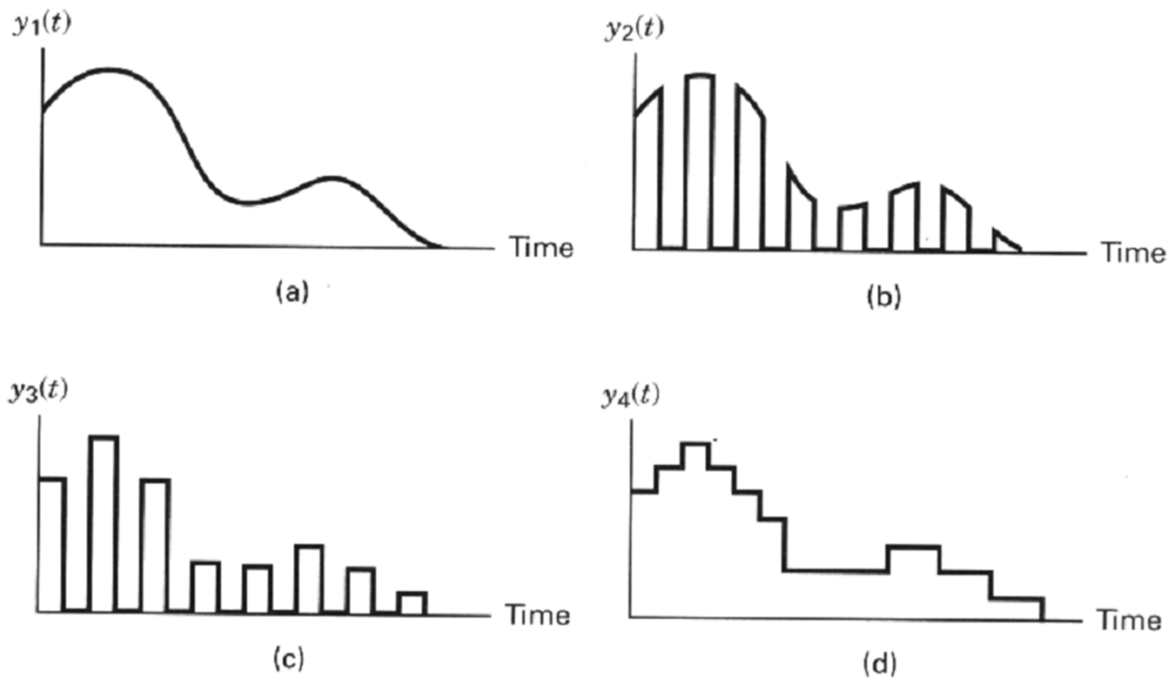



**Figure 2.14** Amplitude and time coordinates of source data. (a) Original analog waveform. (b) Natural-sampled data. (c) Quantized samples. (d) Sample and hold.

# Source of Corruption

- Source of Corruption
  - Quantization Noise – from round-off or truncation error
  - Quantize Saturation – input exceeds the range of quantize
  - Timing Jitter – slight jitter in the position of the sample – no longer uniform sample
- Channel Effects
  - Channel Noise – Thermal noise, interference from circuit switching
  - Inter-symbol Interference – causes by bandlimited channel (BW close signal BW)
  - Signal-to-Noise Ratio for Quantized Pulses

# Source of Corruption

- Signal-to-Noise Ratio for Quantized Pulses

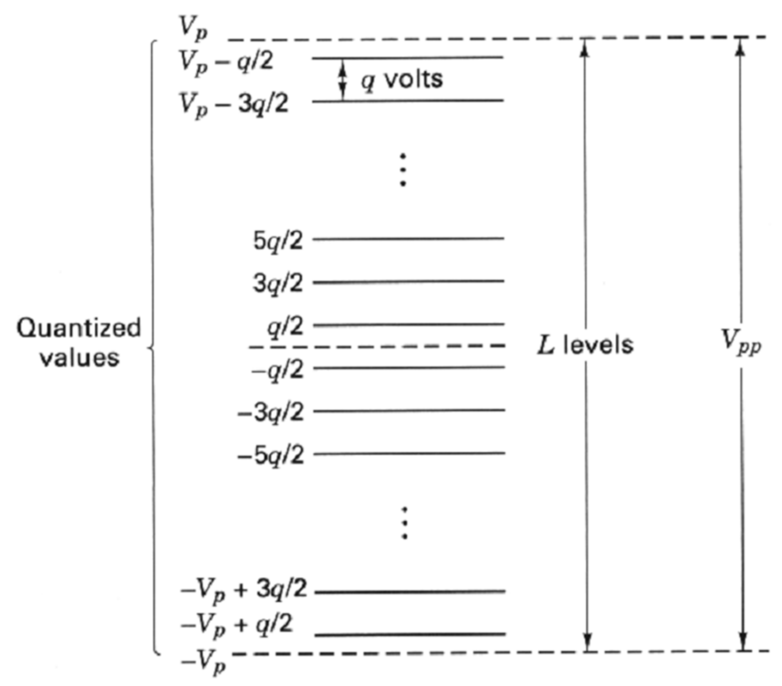



**Figure 2.15** Quantization levels.

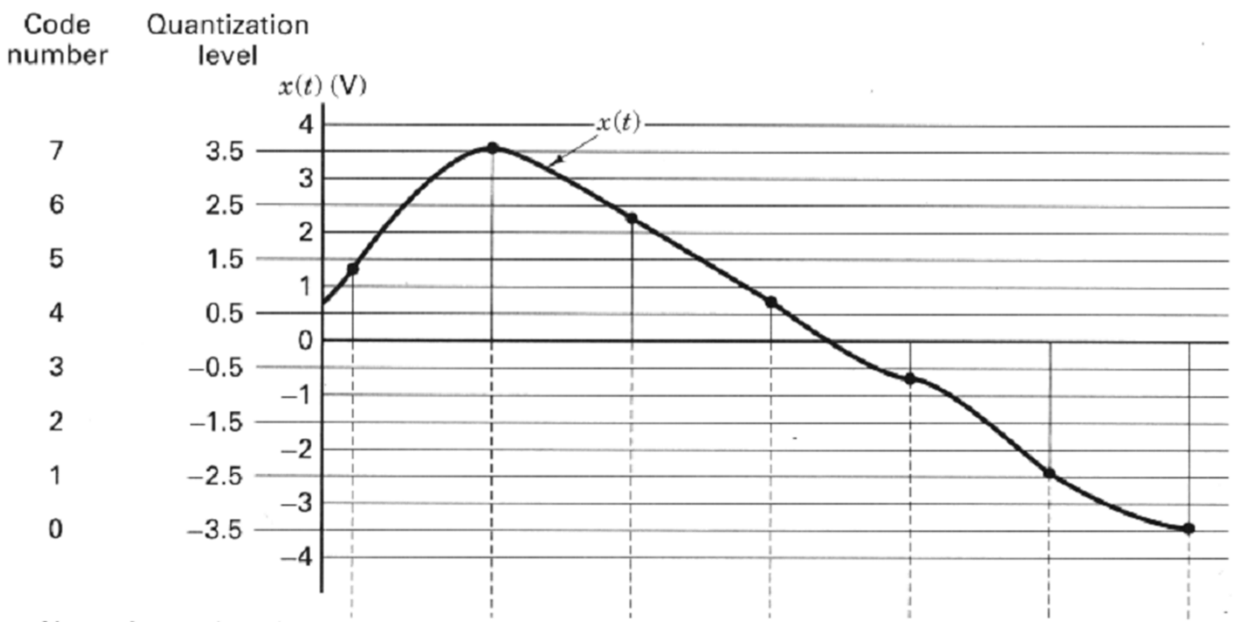
$$\sigma^2 = \int_{-q/2}^{+q/2} e^2 p(e)\, de$$

$$= \int_{-q/2}^{+q/2} e^2 \frac{1}{q}\, de = \frac{q^2}{12}$$

$$V_p^2 = \left(\frac{V_{\mathrm{pp}}}{2}\right)^2 = \left(\frac{Lq}{2}\right)^2 = \frac{L^2 q^2}{4}$$

$$\left(\frac{S}{N}\right)_q = \frac{L^2 q^2/4}{q^2/12} = 3L^2$$

# Pulse Code Modulation




| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 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 |

**Figure 2.16** Natural samples, quantized samples, and pulse code modulation. (Reprinted with permission from Taub and Schilling, *Principles of Communications Systems,* McGraw-Hill Book Company, New York, 1971, Fig. 6.5-1, p. 205.)

# Uniform and Non-uniform Quantization

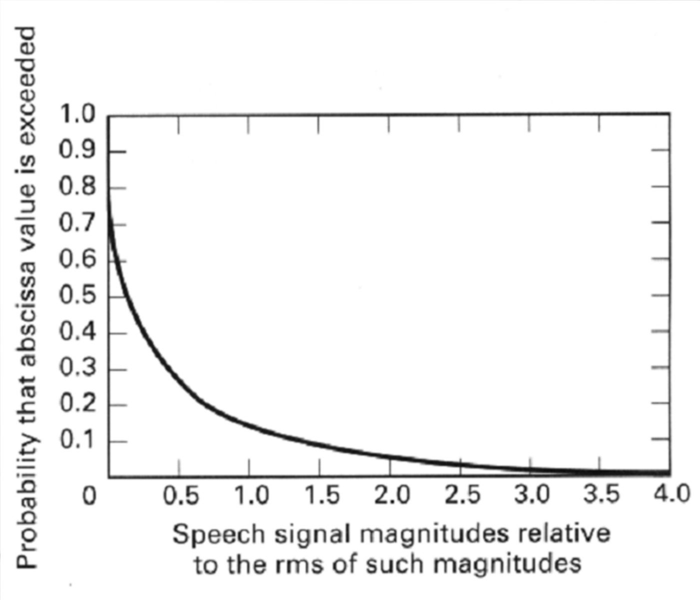



**Figure 2.17** Statistical distribution of single-talker speech signal magnitudes.

$$\text{number of dB} = 10 \log_{10} \frac{P_2}{P_1}$$

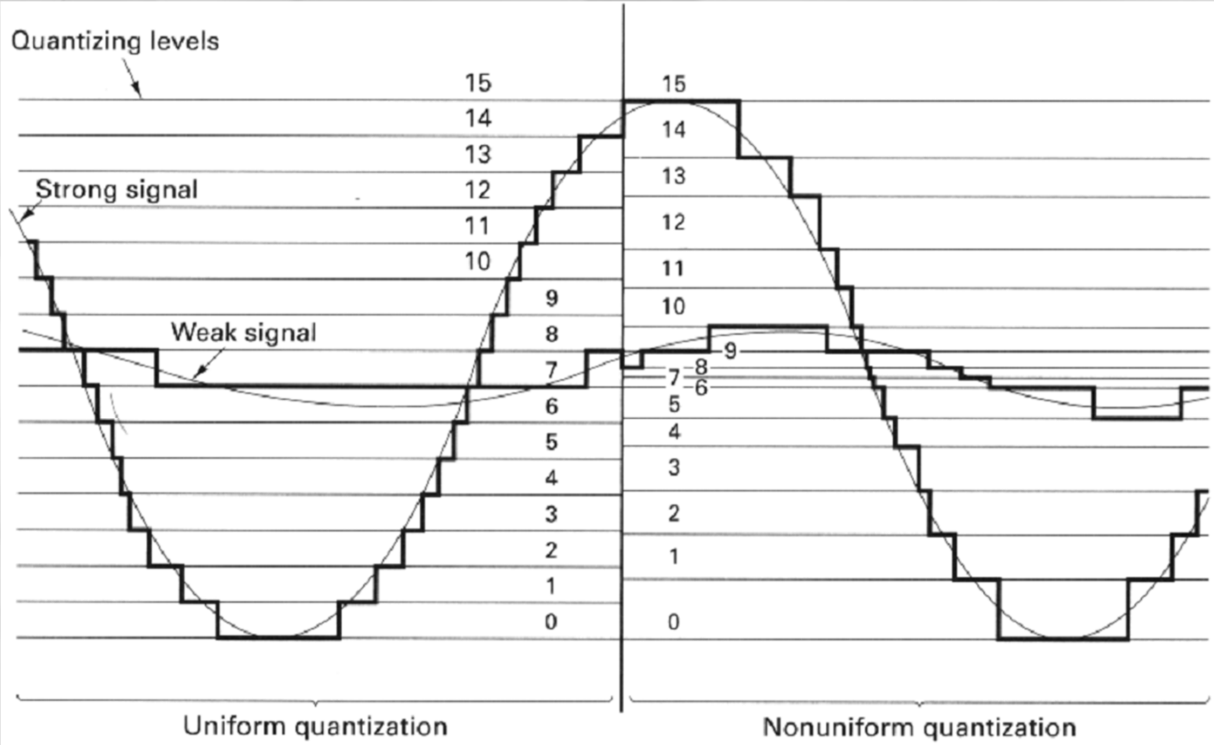



**Figure 2.18** Uniform and nonuniform quantization of signals.

# Uniform and Non-uniform Quantization

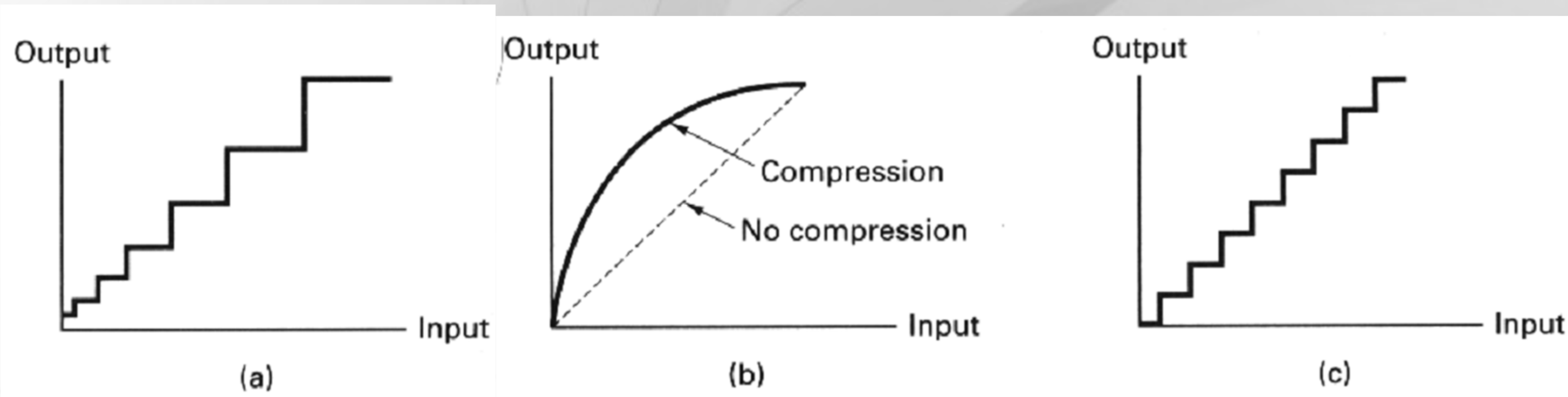



**Figure 2.19** (a) Nonuniform quantizer characteristic. (b) Compression characteristic. (c) Uniform quantizer characteristic.

# Uniform and Non-uniform Quantization

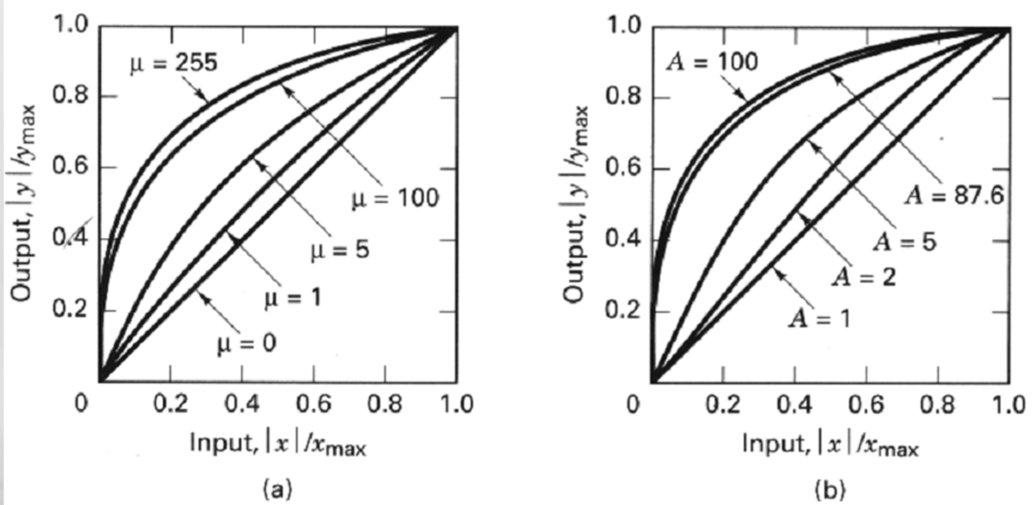



**Figure 2.20** Compression characteristics. (a) μ-law characteristic. (b) *A*-law characteristic.

$$y = y_{\max} \frac{\log_e[1 + \mu(|x|/x_{\max})]}{\log_e(1 + \mu)} \operatorname{sgn} x$$

$$\operatorname{sgn} x = \begin{cases} +1 & \text{for } x \geq 0 \\ -1 & \text{for } x < 0 \end{cases}$$

$$y = \begin{cases} y_{\max} \dfrac{A(|x|/x_{\max})}{1 + \log_e A} \operatorname{sgn} x & 0 < \dfrac{|x|}{x_{\max}} \leq \dfrac{1}{A} \\ y_{\max} \dfrac{1 + \log_e[A(|x|/x_{\max})]}{1 + \log_e A} \operatorname{sgn} x & \dfrac{1}{A} < \dfrac{|x|}{x_{\max}} < 1 \end{cases}$$

# Baseband Transmission

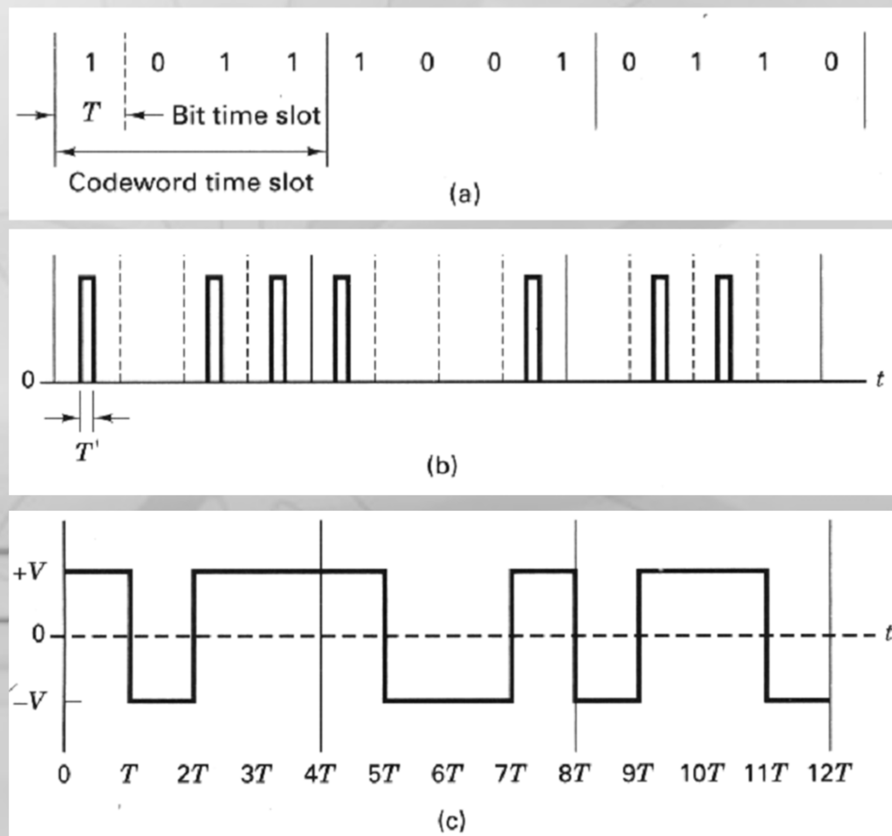



**Figure 2.21** Example of waveform representation of binary digits. (a) PCM sequence. (b) Pulse representation of PCM. (c) Pulse waveform (transition between two levels).

**Figure 2.22** Various PCM waveforms.

# Baseband Transmission

1. *Dc component.* Eliminating the dc energy from the signal's power spectrum enables the system to be ac coupled. Magnetic recording systems, or systems using transformer coupling, have little sensitivity to very low frequency signal components. Thus low-frequency information could be lost.
2. *Self-Clocking.* Symbol or bit synchronization is required for any digital communication system. Some PCM coding schemes have inherent synchronizing or clocking features that aid in the recovery of the clock signal. For example, the Manchester code has a transition in the middle of every bit interval whether a one or a zero is being sent. This guaranteed transition provides a clocking signal.
3. *Error detection.* Some schemes, such as duobinary, provide the means of detecting data errors without introducing additional error-detection bits into the data sequence.
4. *Bandwidth compression.* Some schemes, such as multilevel codes, increase the efficiency of bandwidth utilization by allowing a reduction in required bandwidth for a given data rate; thus there is more information transmitted per unit bandwidth.
5. *Differential encoding.* This technique is useful because it allows the polarity of differentially encoded waveforms to be inverted without affecting the data detection. In communication systems where waveforms sometimes experience inversion, this is a great advantage. (Differential encoding is treated in greater detail in Chapter 4, Section 4.5.2.)
6. *Noise immunity.* The various PCM waveform types can be further characterized by probability of bit error versus signal-to-noise ratio. Some of the schemes are more immune than others to noise. For example, the NRZ waveforms have better error performance than does the unipolar RZ waveform.

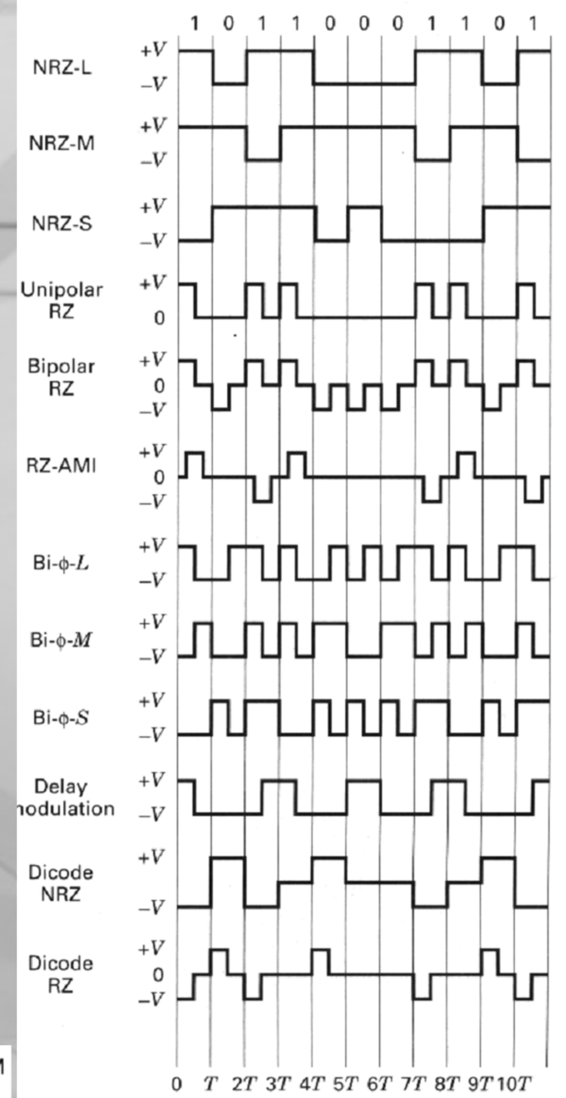



**Figure 2.22** Various PCM waveforms.

# Baseband Transmission

- Spectral Attributes of PCM Waveforms

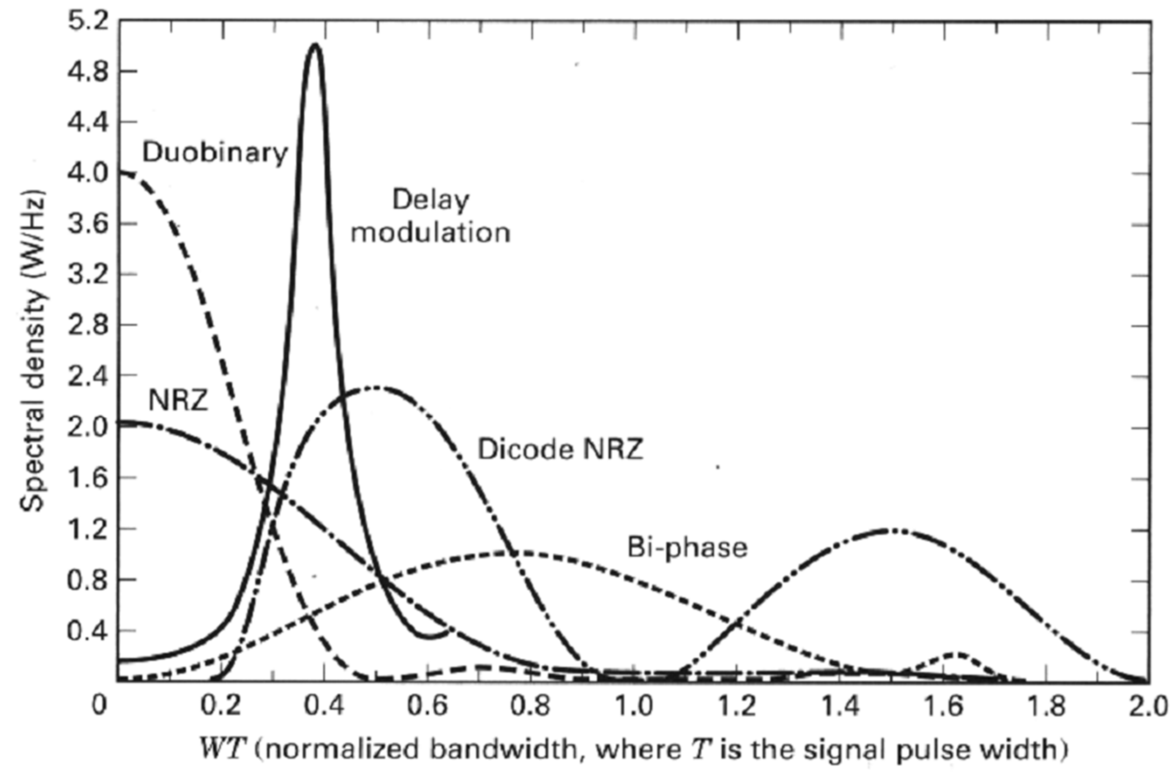



**Figure 2.23** Spectral densities of various PCM waveforms.

# Baseband Transmission

- Bits per PCM Word and Bits per Symbol $M = 2^k$

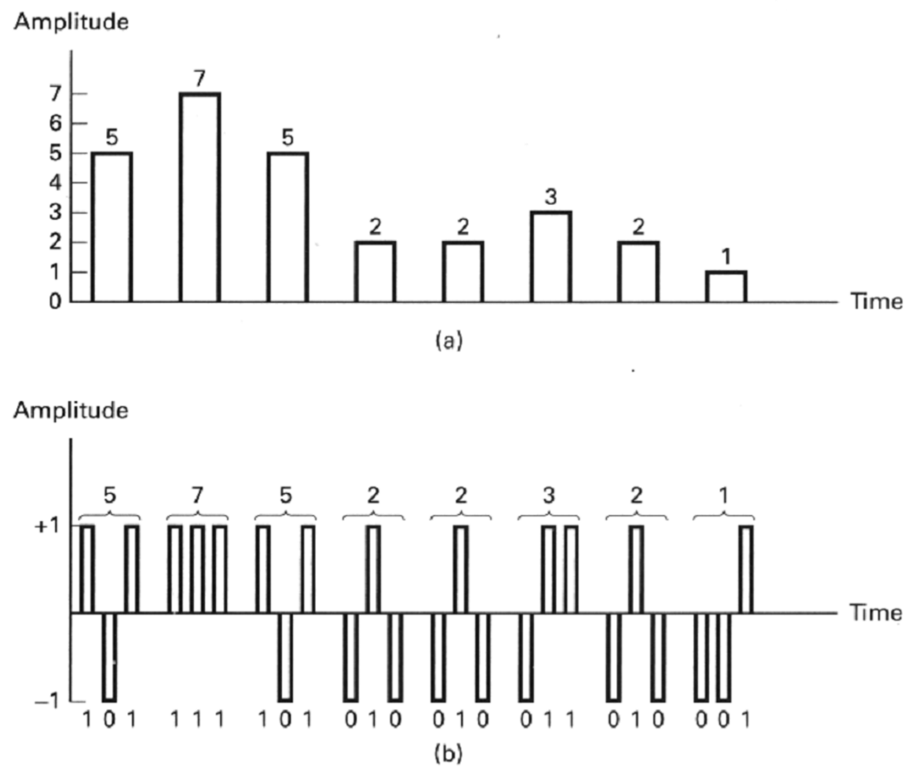



**Figure 2.24** Pulse code modulation signaling. (a) Eight-level signaling. (b) Two-level signaling.

# Correlative Coding

- Duo-binary Signaling $y_k = x_k + x_{k-1}$

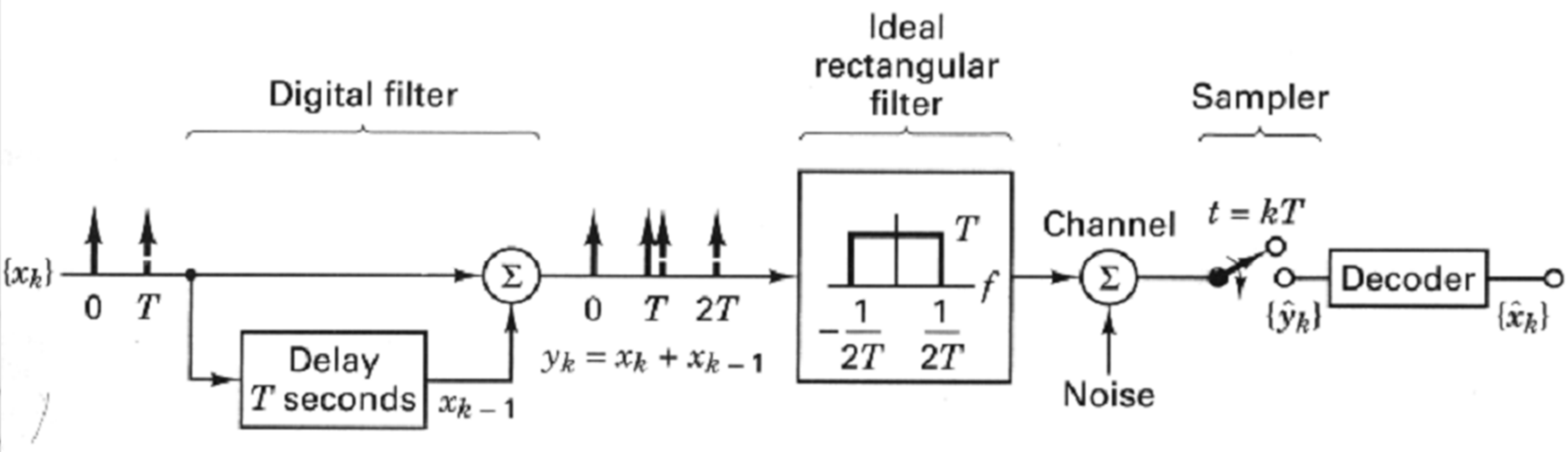



**Figure 2.25** Duobinary signaling.

# Correlative Coding

- Duo-binary Decoding $y_k = x_k + x_{k-1}$

**Example 2.4 Duobinary Coding and Decoding**

Use Equation (2.29) to demonstrate duobinary coding and decoding for the following sequence: $\{x_k\} = 0\ 0\ 1\ 0\ 1\ 1\ 0$. Consider the first bit of the sequence to be a startup digit, not part of the data.

*Solution*

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Binary digit sequence $\{x_k\}$: | 0 | 0 | 1 | 0 | 1 | 1 | 0 |
| Bipolar amplitudes $\{x_k\}$: | –1 | –1 | +1 | –1 | +1 | +1 | –1 |
| Coding rule: $y_k = x_k + x_{k-1}$: | | –2 | 0 | 0 | 0 | 2 | 0 |

Decoding decision rule:

If $\hat{y}_k = 2$, decide that $\hat{x}_k = +1$ (or binary one).

If $\hat{y}_k = -2$, decide that $\hat{x}_k = -1$ (or binary zero).

If $\hat{y}_k = 0$, decide opposite of the previous decision.

| | | | | | | |
|---|---|---|---|---|---|---|
| Decoded bipolar sequence $\{\hat{x}_k\}$: | –1 | +1 | –1 | +1 | +1 | –1 |
| Decoded binary sequence $\{\hat{x}_k\}$: | 0 | 1 | 0 | 1 | 1 | 0 |

# Correlative Coding

- Precoding

$$w_k = x_k \oplus w_{k-1}$$

$$0 \oplus 0 = 0$$
$$0 \oplus 1 = 1$$
$$1 \oplus 0 = 1$$
$$1 \oplus 1 = 0$$

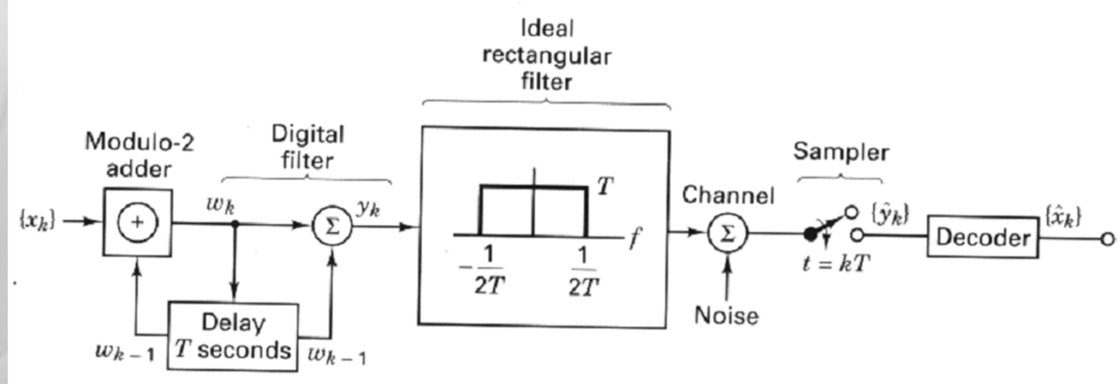



**Figure 2.26** Precoded duobinary signaling.

**Example 2.5 Duobinary Precoding**

Illustrate the duobinary coding and decoding rules when using the differential precoding of Equation (2.30). Assume the same $\{x_k\}$ sequence as that given in Example 2.4.

*Solution*

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Binary digit sequence $\{x_k\}$: | 0 | 0 | 1 | 0 | 1 | 1 | 0 |
| Precoded sequence $w_k = x_k \oplus w_{k-1}$: | 0 | 0 | 1 | 1 | 0 | 1 | 1 |
| Bipolar sequence $\{w_k\}$: | –1 | –1 | +1 | +1 | –1 | +1 | +1 |
| Coding rule: $y_k = w_k + w_{k-1}$: | | –2 | 0 | +2 | 0 | 0 | +2 |

Decoding decision rule: If $\hat{y}_k = \pm 2$, decide that $\hat{x}_k$ = binary zero. If $\hat{y}_k = 0$, decide that $\hat{x}_k$ = binary one.

| | | | | | | |
|---|---|---|---|---|---|---|
| Decoded binary sequence $\{x_k\}$: | 0 | 1 | 0 | 1 | 1 | 0 |

# Problems and Questions

- Conclusion

In this chapter we have considered the first important step in any digital communication system, transforming the source information (both textual and analog) to a form that is compatible with a digital system. We treated various aspects of sampling, quantization (both uniform and nonuniform), and pulse code modulation (PCM). We considered the selection of pulse waveforms for the transmission of baseband signals through the channel. We also introduced the duobinary concept of adding a controlled amount of ISI to achieve an improvement in bandwidth efficiency at the expense of an increase in power.

# Problems and Questions

- Problems

**2.2.** We want to transmit 800 characters/s, where each character is represented by its 7-bit ASCII codeword, followed by an eighth bit for error detection, per character, as in Problem 2.1. A multilevel PAM waveform with $M = 16$ levels is used.

**(a)** What is the effective transmitted bit rate?

**(b)** What is the symbol rate?

**2.9.** A waveform, $x(t) = 10 \cos (1000t + \pi/3) + 20 \cos (2000t + \pi/6)$ is to be uniformly sampled for digital transmission.

**(a)** What is the maximum allowable time interval between sample values that will ensure perfect signal reproduction?

**(b)** If we want to reproduce 1 hour of this waveform, how many sample values need to be stored?

**2.15.** A signal in the frequency range 300 to 3300 Hz is limited to a peak-to-peak swing of 10 V. It is sampled at 8000 samples/s and the samples are quantized to 64 evenly spaced levels. Calculate and compare the bandwidths and ratio of peak signal power to rms quantization noise if the quantized samples are transmitted either as binary pulses or as four-level pulses. Assume that the system bandwidth is defined by the main spectral lobe of the signal.

# Problems and Questions

- Questions

**2.1** What are the similarities and differences between the terms "formatting" and "source coding"? (See Chapter 2, introduction.)

**2.2** In the process of *formatting* information, why is it often desirable to perform *over-sampling?* (See Section 2.4.3.)

**2.3** In using pulse code modulation (PCM) for digitizing analog information, explain how the parameters *fidelity, bandwidth,* and *time delay* can be traded off. (See Section 2.6.)

**2.4** Why is it often preferred to use units of normalized bandwidth, $WT$ (or time-bandwidth product), compared with bandwidth alone? (See Section 2.8.3.)