# Wireless Communication

## Lecture 5

## Signal Encoding Techniques

# Reasons for Choosing Encoding Techniques

- Digital data, digital signal
  - Equipment less complex and expensive than digital-to-analog modulation equipment
- Analog data, digital signal
  - Permits use of modern digital transmission and switching equipment

# Reasons for Choosing Encoding Techniques

- Digital data, analog signal
  - Some transmission media will only propagate analog signals
  - E.g., optical fiber and unguided media
- Analog data, analog signal
  - Analog data in electrical form can be transmitted easily and cheaply
  - Done with voice transmission over voice-grade lines

# Signal Encoding Criteria

- What determines how successful a receiver will be in interpreting an incoming signal?
  - Signal-to-noise ratio
  - Data rate
  - Bandwidth
- An increase in data rate increases bit error rate
- An increase in SNR decreases bit error rate
- An increase in bandwidth allows an increase in data rate

# Factors Used to Compare Encoding Schemes

- Signal spectrum
  - With lack of high-frequency components, less bandwidth required
  - With no dc component, ac coupling via transformer possible
  - Transfer function of a channel is worse near band edges
- Clocking
  - Ease of determining beginning and end of each bit position

# Factors Used to Compare Encoding Schemes

- Signal interference and noise immunity
  - Performance in the presence of noise
- Cost and complexity
  - The higher the signal rate to achieve a given data rate, the greater the cost

# Basic Encoding Techniques

- Digital data to analog signal
  - Amplitude-shift keying (ASK)
    - Amplitude difference of carrier frequency
  - Frequency-shift keying (FSK)
    - Frequency difference near carrier frequency
  - Phase-shift keying (PSK)
    - Phase of carrier signal shifted

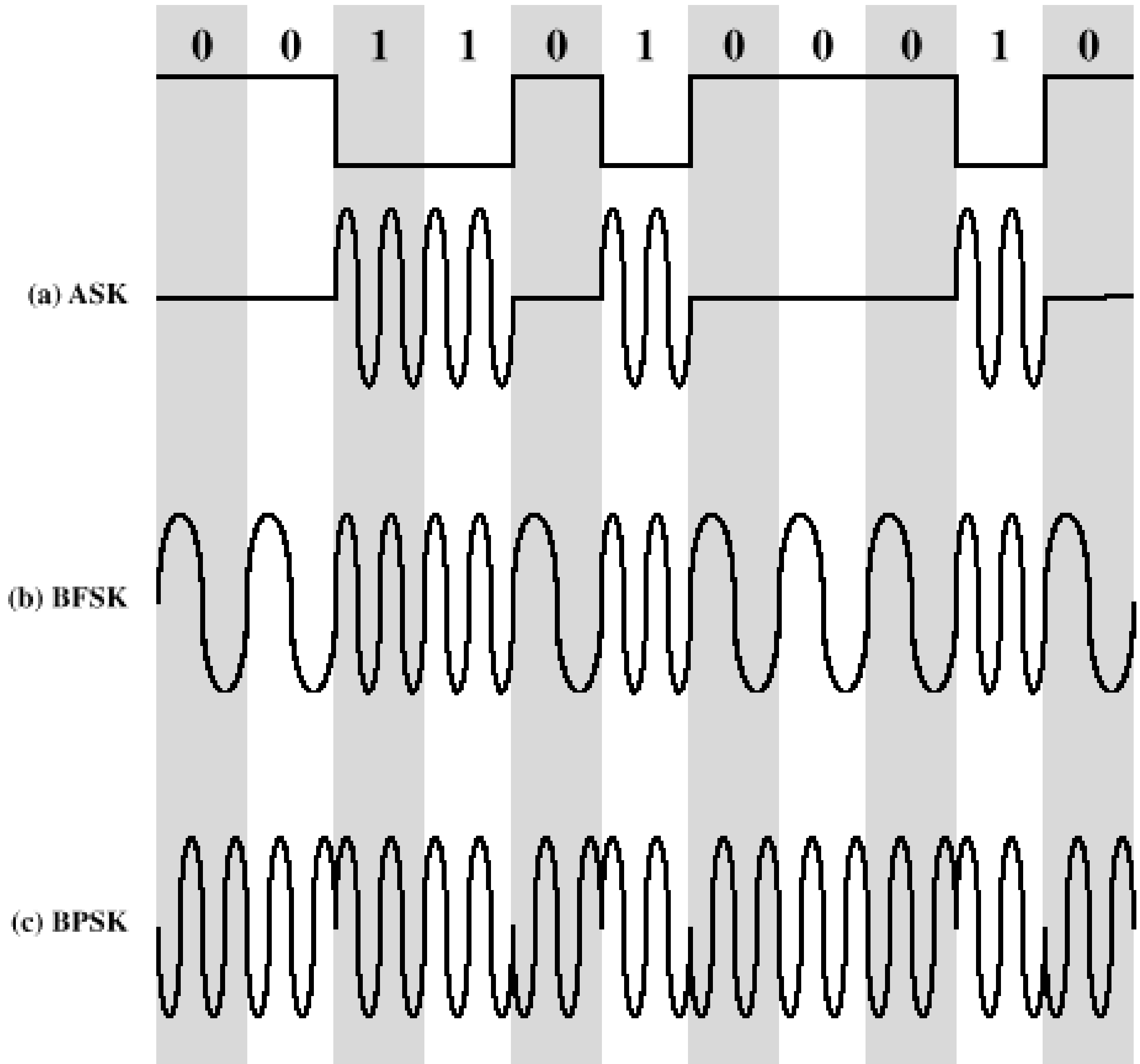



**Figure 6.2 Modulation of Analog Signals for Digital Data**

# Amplitude-Shift Keying

- One binary digit represented by presence of carrier, at constant amplitude
- Other binary digit represented by absence of carrier

$$s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ 0 & \text{binary } 0 \end{cases}$$

  - where the carrier signal is $A\cos(2\pi f_c t)$

# Amplitude-Shift Keying

- Susceptible to sudden gain changes
- Inefficient modulation technique
- On voice-grade lines, used up to 1200 bps
- Used to transmit digital data over optical fiber

# Binary Frequency-Shift Keying (BFSK)

- Two binary digits represented by two different frequencies near the carrier frequency

$$s(t)=\begin{cases} A\cos(2\pi f_1 t) & \text{binary } 1 \\ A\cos(2\pi f_2 t) & \text{binary } 0 \end{cases}$$

  - where $f_1$ and $f_2$ are offset from carrier frequency $f_c$ by equal but opposite amounts

# Binary Frequency-Shift Keying (BFSK)

- Less susceptible to error than ASK
- On voice-grade lines, used up to 1200bps
- Used for high-frequency (3 to 30 MHz) radio transmission
- Can be used at higher frequencies on LANs that use coaxial cable

# Multiple Frequency-Shift Keying (MFSK)

- More than two frequencies are used
- More bandwidth efficient but more susceptible to error

$$s_i(t) = A\cos 2\pi f_i t \qquad 1 \leq i \leq M$$

  - $f_i = f_c + (2i - 1 - M) f_d$
  - $f_c$ = the carrier frequency
  - $f_d$ = the difference frequency
  - $M$ = number of different signal elements = $2^L$
  - $L$ = number of bits per signal element

# Multiple Frequency-Shift Keying (MFSK)

- To match data rate of input bit stream, each output signal element is held for:

$$T_s = LT \text{ seconds}$$

  - where $T$ is the bit period (data rate = $1/T$)
- So, one signal element encodes $L$ bits

# Multiple Frequency-Shift Keying (MFSK)

- Total bandwidth required

$$2Mf_d$$

- Minimum frequency separation required

$$2f_d = 1/T_s$$

- Therefore, modulator requires a bandwidth of

$$W_d = 2^L/LT = M/T_s$$

# Multiple Frequency-Shift Keying (MFSK)

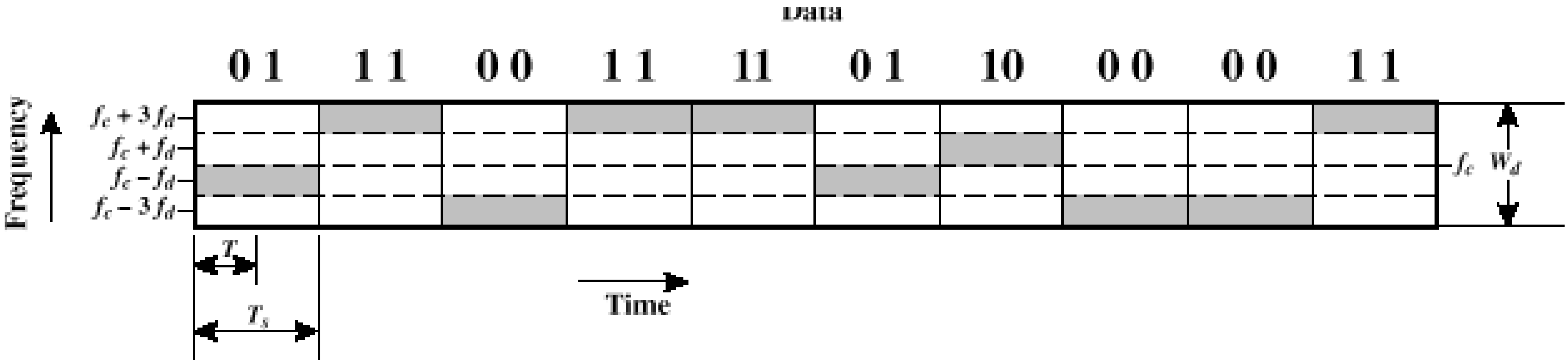



**Figure 6.4 MFSK Frequency Use ($M = 4$)**

# Phase-Shift Keying (PSK)

- Two-level PSK (BPSK)
  - Uses two phases to represent binary digits

$$s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ A\cos(2\pi f_c t + \pi) & \text{binary } 0 \end{cases}$$

$$= \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ -A\cos(2\pi f_c t) & \text{binary } 0 \end{cases}$$

# Phase-Shift Keying (PSK)

- Differential PSK (DPSK)
  - Phase shift with reference to previous bit
    - Binary 0 – signal burst of same phase as previous signal burst
    - Binary 1 – signal burst of opposite phase to previous signal burst

# Phase-Shift Keying (PSK)

- Four-level PSK (QPSK)
  - Each element represents more than one bit

$$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \dfrac{\pi}{4}\right) & 11 \\ A\cos\left(2\pi f_c t + \dfrac{3\pi}{4}\right) & 01 \\ A\cos\left(2\pi f_c t - \dfrac{3\pi}{4}\right) & 00 \\ A\cos\left(2\pi f_c t - \dfrac{\pi}{4}\right) & 10 \end{cases}$$

# Phase-Shift Keying (PSK)

- Multilevel PSK
  - Using multiple phase angles with each angle having more than one amplitude, multiple signals elements can be achieved

$$D = \frac{R}{L} = \frac{R}{\log_2 M}$$

- $D$ = modulation rate, baud
- $R$ = data rate, bps
- $M$ = number of different signal elements = $2^L$
- $L$ = number of bits per signal element

# Performance

- Bandwidth of modulated signal ($B_T$)
  - ASK, PSK $B_T=(1+r)R$
  - FSK $B_T=2\Delta F+(1+r)R$
    - $R$ = bit rate
    - 0 < r < 1; related to how signal is filtered
    - $\Delta F = f_2 - f_c = f_c - f_1$

# Performance

- Bandwidth of modulated signal ($B_T$)
  - MPSK $$B_T = \left(\frac{1+r}{L}\right)R = \left(\frac{1+r}{\log_2 M}\right)R$$
  - MFSK $$B_T = \left(\frac{(1+r)M}{\log_2 M}\right)R$$
    - $L$ = number of bits encoded per signal element
    - $M$ = number of different signal elements

# Quadrature Amplitude Modulation

- QAM is a combination of ASK and PSK
  - Two different signals sent simultaneously on the same carrier frequency

$$s(t) = d_1(t)\cos 2\pi f_c t + d_2(t)\sin 2\pi f_c t$$

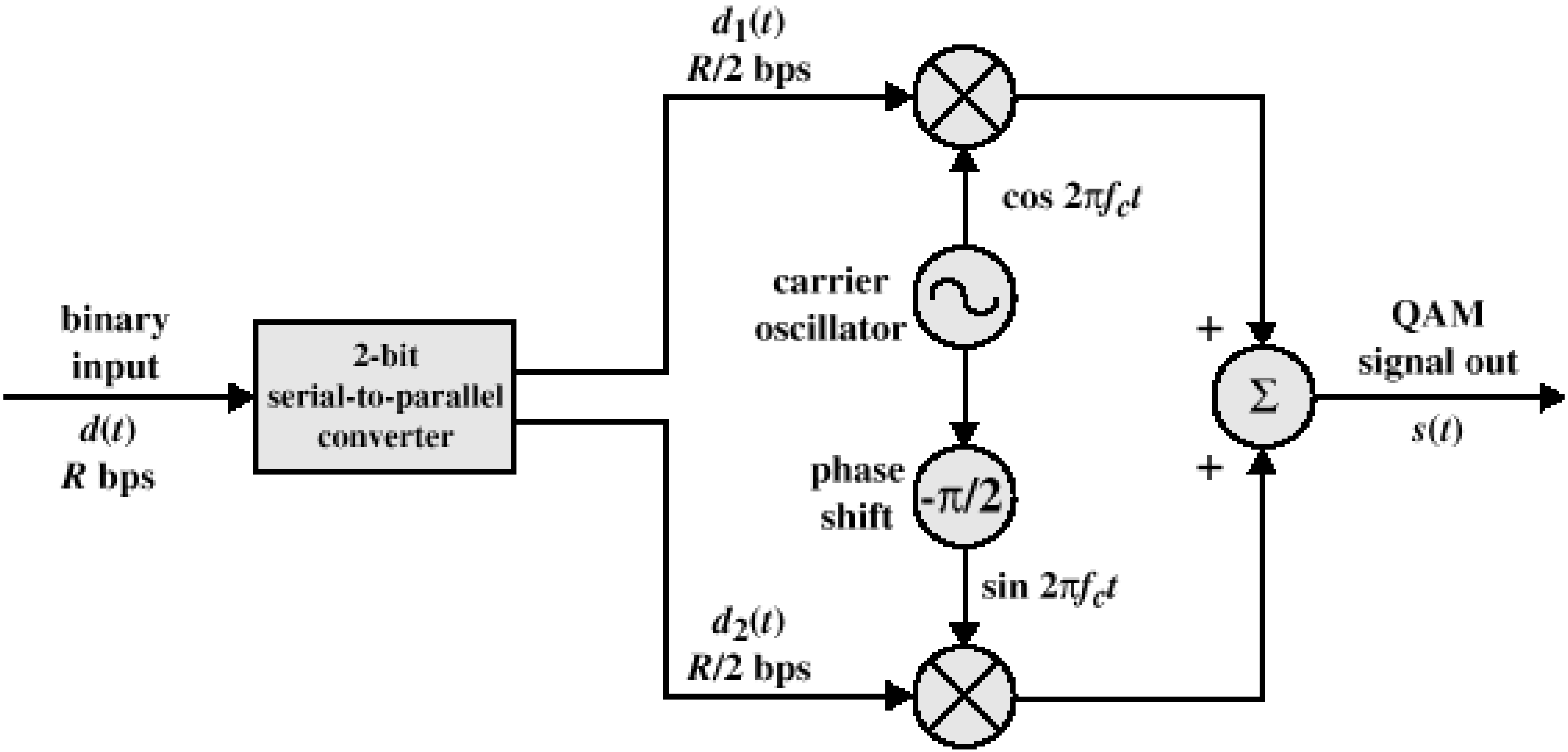



Figure 6.10 QAM Modulator

# Reasons for Analog Modulation

- Modulation of digital signals
  - When only analog transmission facilities are available, digital to analog conversion required
- Modulation of analog signals
  - A higher frequency may be needed for effective transmission
  - Modulation permits frequency division multiplexing

# Basic Encoding Techniques

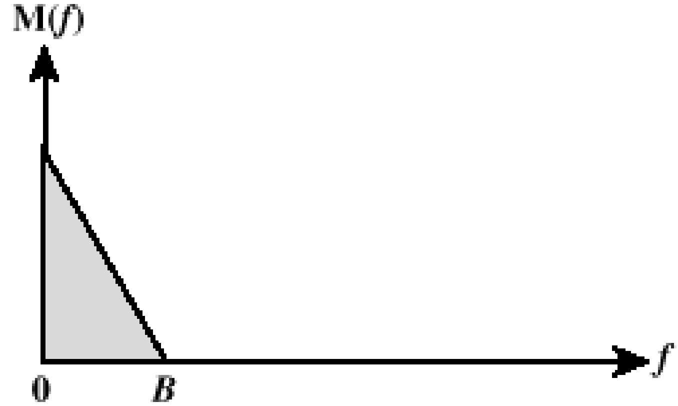
- Analog data to analog signal
  - Amplitude modulation (AM)
  - Angle modulation
    - Frequency modulation (FM)
    - Phase modulation (PM)

# Amplitude Modulation

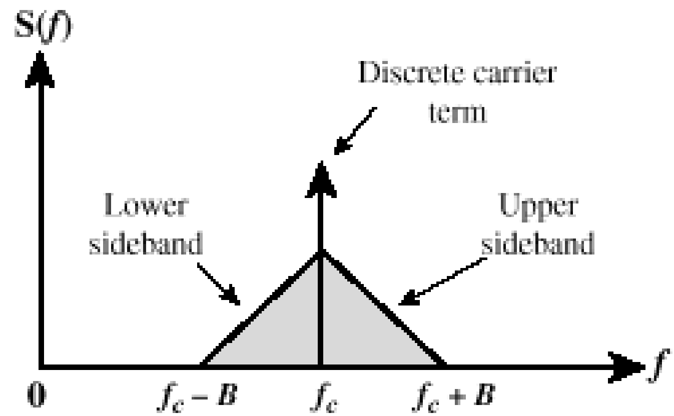
- Amplitude Modulation

$$s(t) = [1 + n_a x(t)]\cos 2\pi f_c t$$

  - $\cos 2\pi f_c t$ = carrier
  - $x(t)$ = input signal
  - $n_a$ = modulation index
    - Ratio of amplitude of input signal to carrier
- a.k.a double sideband transmitted carrier (DSBTC)




(a) Spectrum of modulating signal




(b) Spectrum of AM signal with carrier at $f_c$

**Figure 6.12 Spectrum of an AM Signal**

# Amplitude Modulation

- Transmitted power

$$P_t = P_c\left(1 + \frac{n_a^{\;2}}{2}\right)$$

  - $P_t$ = total transmitted power in $s(t)$
  - $P_c$ = transmitted power in carrier

# Single Sideband (SSB)

- Variant of AM is single sideband (SSB)
  - Sends only one sideband
  - Eliminates other sideband and carrier
- Advantages
  - Only half the bandwidth is required
  - Less power is required
- Disadvantages
  - Suppressed carrier can't be used for synchronization purposes

# Angle Modulation

- Angle modulation

$$s(t) = A_c \cos[2\pi f_c t + \phi(t)]$$

- Phase modulation
  - Phase is proportional to modulating signal

$$\phi(t) = n_p m(t)$$

    - $n_p$ = phase modulation index

# Angle Modulation

- Frequency modulation
  - Derivative of the phase is proportional to modulating signal

$$\phi'(t) = n_f m(t)$$

    - $n_f$ = frequency modulation index

# Angle Modulation

- Compared to AM, FM and PM result in a signal whose bandwidth:
  - is also centered at $f_c$
  - but has a magnitude that is much different
    - Angle modulation includes $\cos(\varnothing\,(t))$ which produces a wide range of frequencies
- Thus, FM and PM require greater bandwidth than AM

# Angle Modulation

- Carson's rule

$$B_T = 2(\beta + 1)B$$

where

$$\beta = \begin{cases} n_p A_m & \text{for PM} \\ \dfrac{\Delta F}{B} = \dfrac{n_f A_m}{2\pi B} & \text{for FM} \end{cases}$$

- The formula for FM becomes

$$B_T = 2\Delta F + 2B$$

# Basic Encoding Techniques

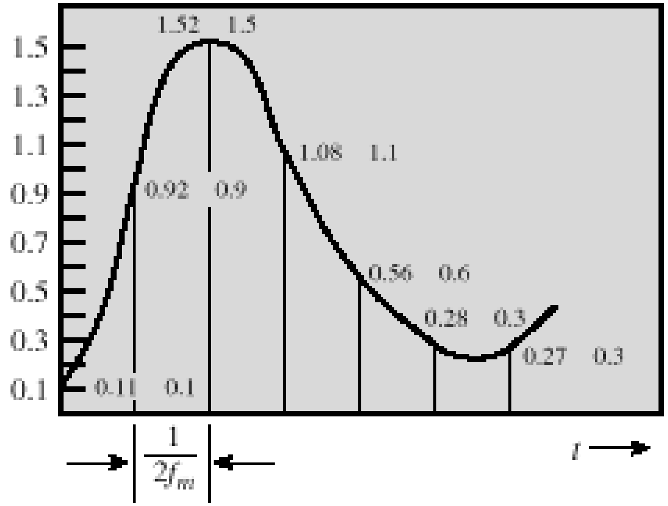
- Analog data to digital signal
  - Pulse code modulation (PCM)
  - Delta modulation (DM)

# Analog Data to Digital Signal

- Once analog data have been converted to digital signals, the digital data:
  - can be transmitted using NRZ-L
  - can be encoded as a digital signal using a code other than NRZ-L
  - can be converted to an analog signal, using previously discussed techniques

# Pulse Code Modulation

- Based on the sampling theorem
- Each analog sample is assigned a binary code
  - Analog samples are referred to as pulse amplitude modulation (PAM) samples
- The digital signal consists of block of *n* bits, where each *n*-bit number is the amplitude of a PCM pulse




(a)

| Digit | Binary Equivalent | PCM waveform | Digit | Binary Equivalent | PCM waveform |
|---|---|---|---|---|---|
| 0 | 0000 | | 8 | 1000 | |
| 1 | 0001 | | 9 | 1001 | |
| 2 | 0010 | | 10 | 1010 | |
| 3 | 0011 | | 11 | 1011 | |
| 4 | 0100 | | 12 | 1100 | |
| 5 | 0101 | | 13 | 1101 | |
| 6 | 0110 | | 14 | 1110 | |
| 7 | 0111 | | 15 | 1111 | |

(b)

**Figure 6.15 Pulse-Code Modulation**

# Pulse Code Modulation

- By quantizing the PAM pulse, original signal is only approximated
- Leads to quantizing noise
- Signal-to-noise ratio for quantizing noise

$$\mathrm{SNR}_{\mathrm{dB}} = 20\log 2^{n} + 1.76\,\mathrm{dB} = 6.02n + 1.76\,\mathrm{dB}$$

- Thus, each additional bit increases SNR by 6 dB, or a factor of 4

# Delta Modulation

- Analog input is approximated by staircase function
  - Moves up or down by one quantization level ($\delta$) at each sampling interval
- The bit stream approximates derivative of analog signal (rather than amplitude)
  - 1 is generated if function goes up
  - 0 otherwise

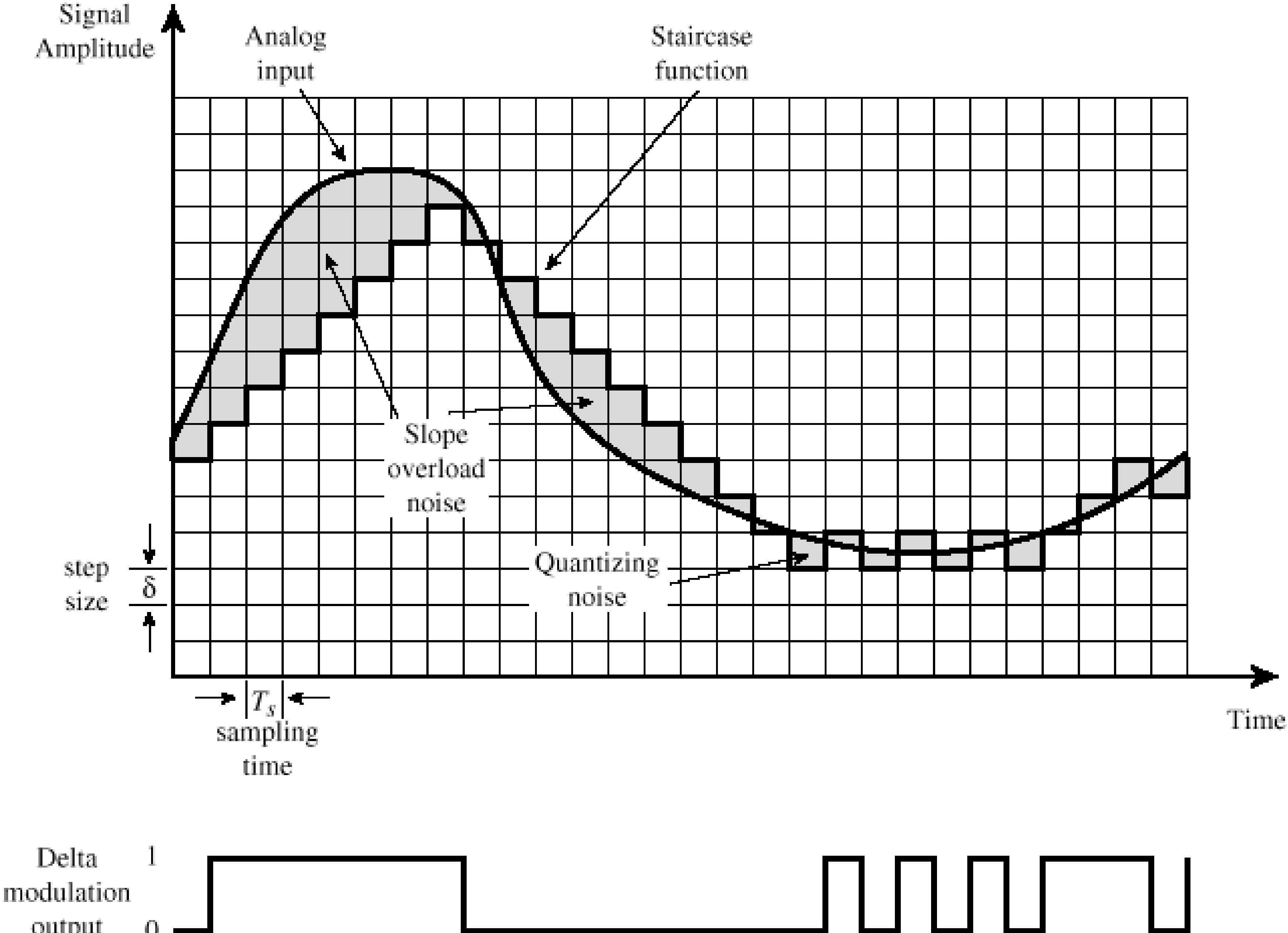



**Figure 6.18 Example of Delta Modulation**

# Delta Modulation

- Two important parameters
  - Size of step assigned to each binary digit ($\delta$)
  - Sampling rate
- Accuracy improved by increasing sampling rate
  - However, this increases the data rate
- Advantage of DM over PCM is the simplicity of its implementation

# Reasons for Growth of Digital Techniques

- Growth in popularity of digital techniques for sending analog data
  - Repeaters are used instead of amplifiers
    - No additive noise
  - TDM is used instead of FDM
    - No intermodulation noise
  - Conversion to digital signaling allows use of more efficient digital switching techniques