UNIVERSITAS KOMPUTER INDONESIA

Wireless and Mobile Communication

Chap 6
Signal Encoding Technique

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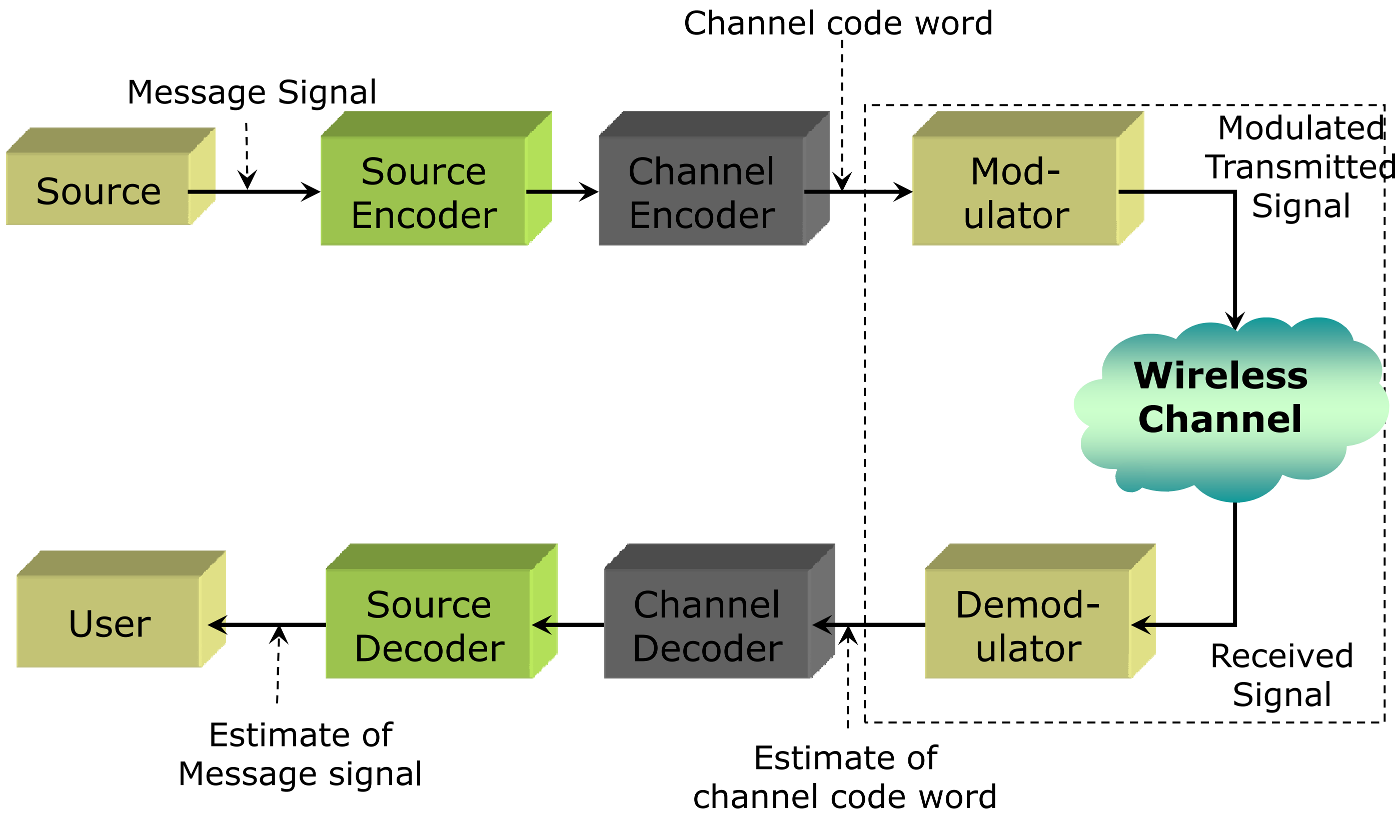
Wireless Communication System



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., 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

Comparing 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

Comparing 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

Digital Data to Analog Signals

- 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

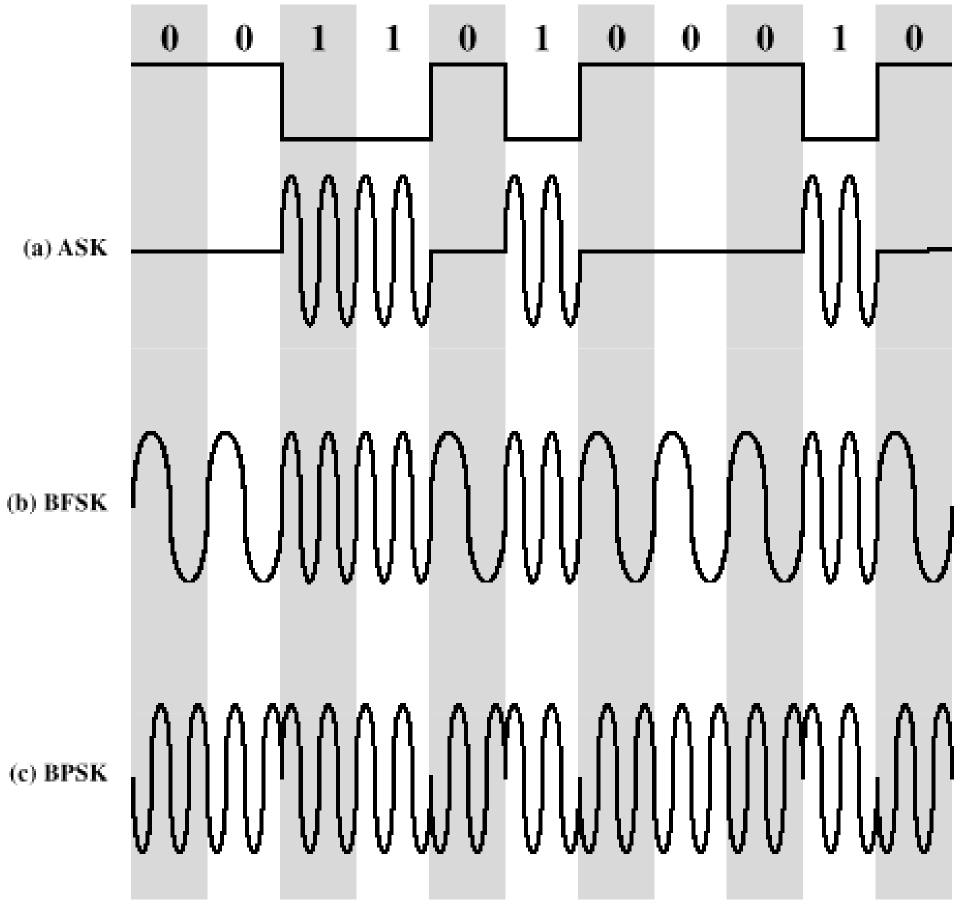


Figure 6.2 Modulation of Analog Signals for Digital Data

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

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

- $f_i = f_c + (2i - 1 - M)f_d$
- 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)

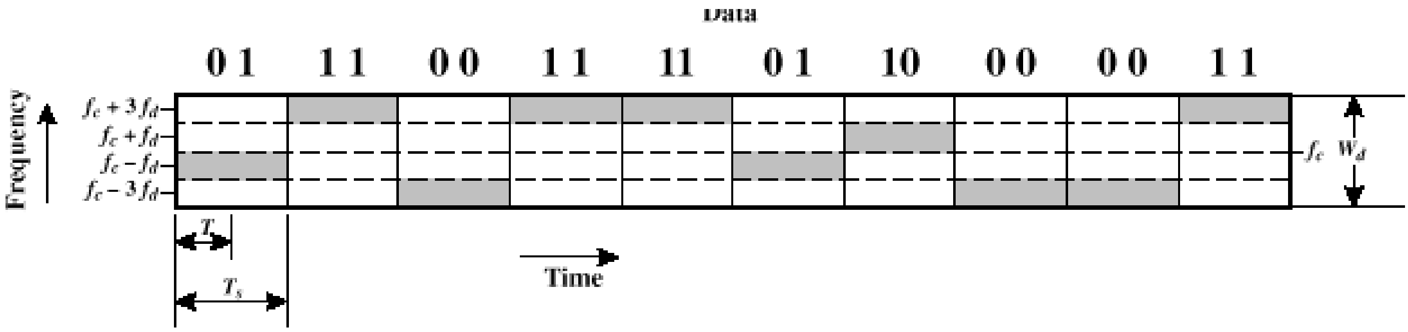


Figure 6.4 MFSK Frequency Use ($M = 4$)

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 $B_T=2\text{D}F+(1+r)R$
    - R = bit rate
    - 0 < r < 1; related to how signal is filtered
    - $\text{D}F = f_2-f_c=f_c-f_1$

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

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

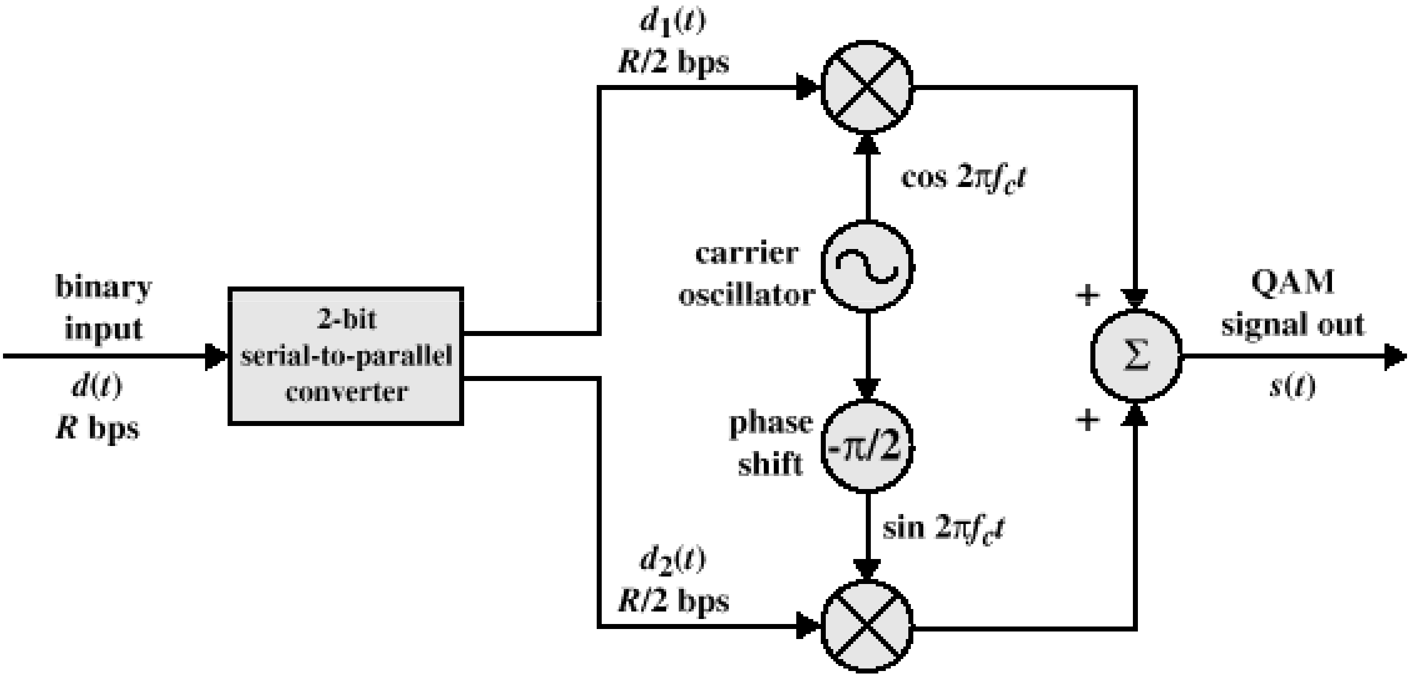


Figure 6.10 QAM Modulator

Analog Data to Analog Signal

- 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

Modulation Techniques

- 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 (< 1)
    - Ratio of amplitude of input signal to carrier
- a.k.a double sideband transmitted carrier (DSBTC)

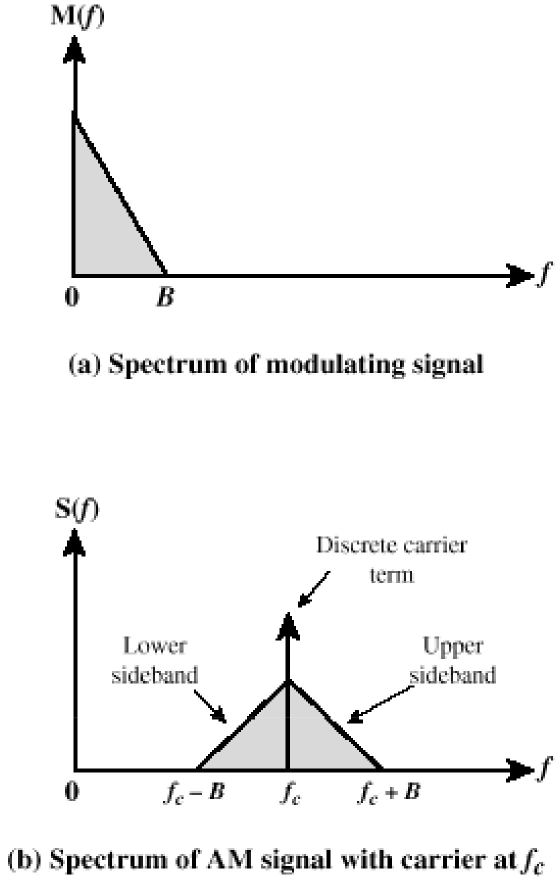


Figure 6.12 Spectrum of an AM Signal

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

- 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

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

Analog Data to Digital Signal

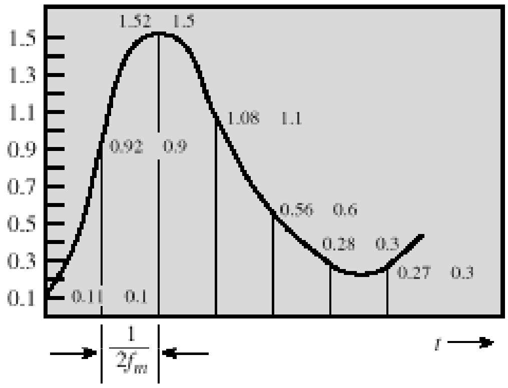
- Digitization: Often analog data are converted to digital form
- Once analog data have been converted to digital signals, the digital data:
  – can be transmitted using NRZ-L
  – can be encoded as a digital signal using a code other than NRZ-L
  – can be converted to an analog signal, using previously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

Analog data to digital signal

- Pulse code modulation (PCM)
- Delta modulation (DM)

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 (δ) at each sampling interval
- The bit stream approximates derivative of analog signal (rather than amplitude)
  - 1 is generated if function goes up
  - 0 otherwise

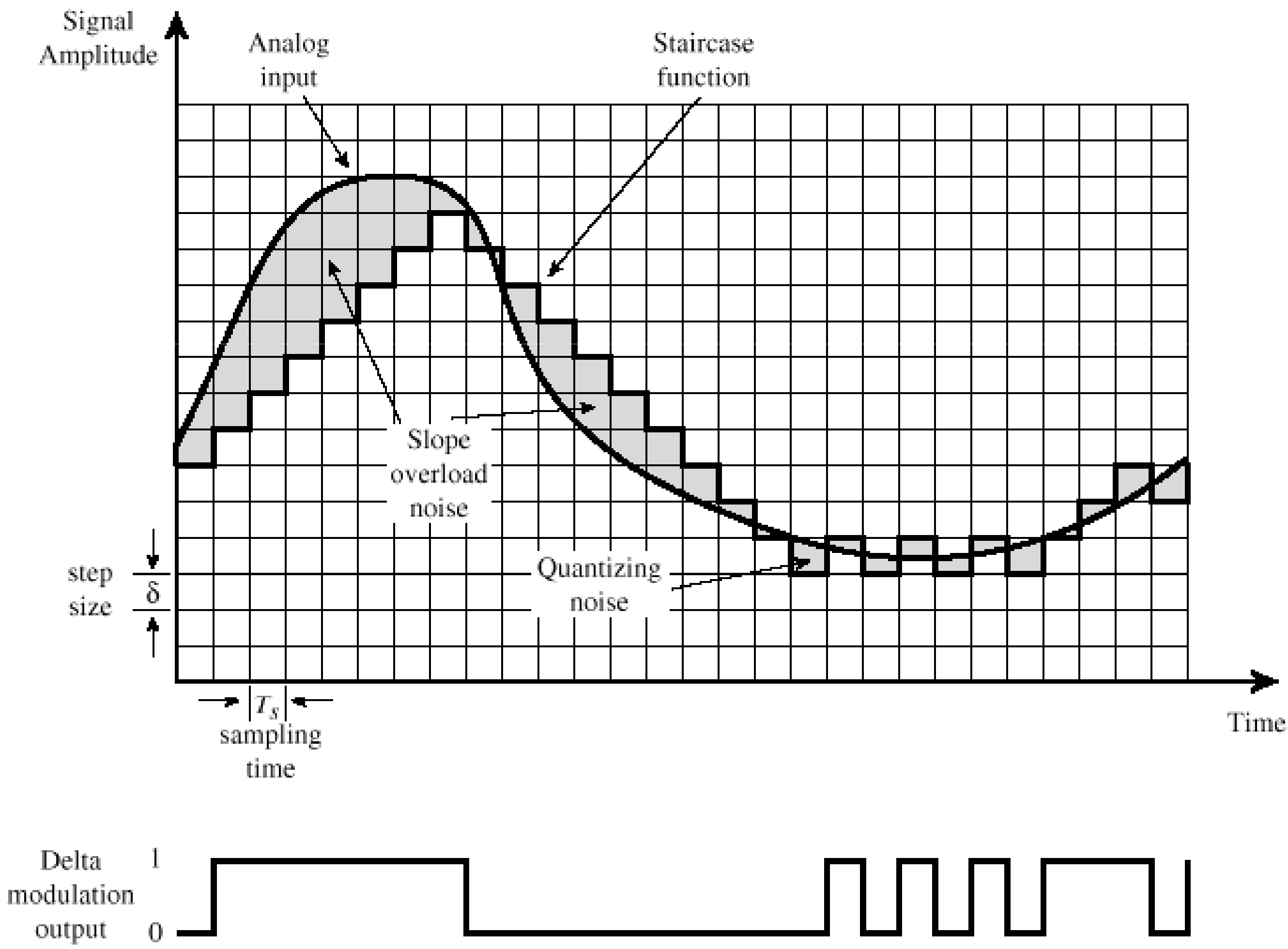


Figure 6.18 Example of Delta Modulation

Delta Modulation

- Two important parameters
  - Size of step assigned to each binary digit (δ)
  - 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

1-Mbps Data Rate

To minimize the effect of a low SNR and data loss in cases of narrowband interference, each bit of data is encoded as a sequence of 11 bits called a *Barker 11 code*. The goal is to add enough additional information to each bit of data that its integrity will be preserved when it is sent in a noisy environment.

There are only two possible values for the Barker chips—one corresponding to a 0 data bit (10110111000) and one for a 1 data bit (01001000111). The receiver must also expect the Barker chips and convert them back into single bits of data. The number and sequence of the Barker chip bits have been defined to allow data bits to be recovered if some of the chip bits are lost. In fact, up to 9 of the 11 bits in a single chip can be lost before the original data bit cannot be restored.

Each bit in a Barker chip can be transmitted by using the *differential binary phase shift keying* (DBPSK) modulation scheme. The phase of the carrier signal is shifted or rotated according to the data bit being transmitted, as follows:

0: The phase is not changed.

1: The phase is "rotated" or shifted 180 degrees, such that the signal is suddenly inverted.

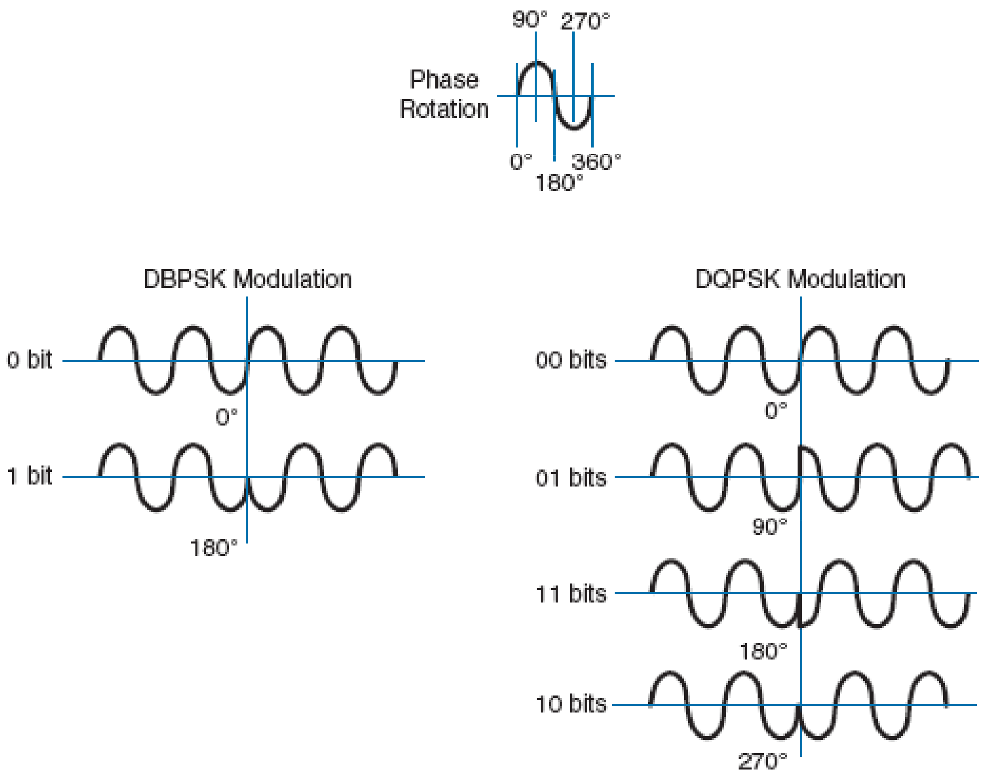
2-Mbps Data Rate

It is possible to couple the 1-Mbps strategy with a different modulation scheme to double the data rate. As before, each data bit is coded into an 11-bit Barker code with an 11-MHz chipping rate. This time, chips are taken two at a time and modulated onto the carrier signal by using *differential quadrature phase shift keying* (DQPSK). The two chips are used to affect the carrier signal's phase in four possible ways, each one 90 degrees apart (hence, the name quadrature). The bit patterns produce the following phase shifts:

- **00:** The phase is not changed.
- **01:** Rotate the phase 90 degrees.
- **11:** Rotate the phase 180 degrees.
- **10:** Rotate the phase 270 degrees.

Because DQPSK can modulate data bits in pairs, it is able to transmit twice the data rate of DBPSK, or 2 Mbps.

Example Phase Changes During DBPSK and DQPSK Modulation.



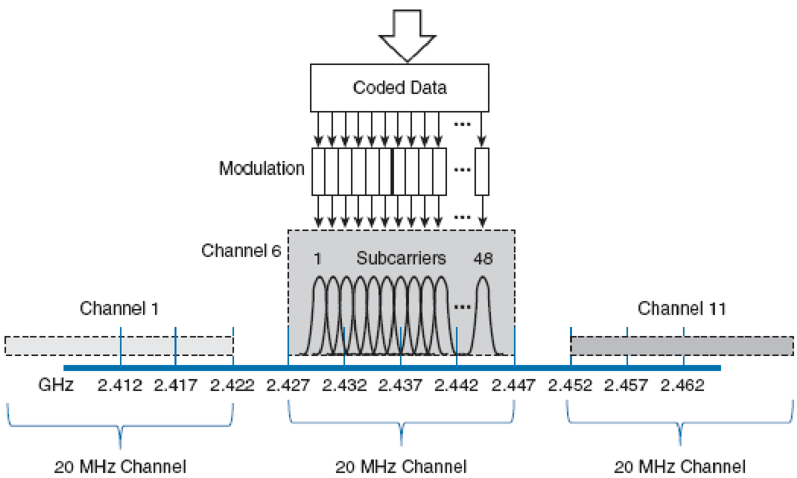
OFDM

DSSS spreads the chips of a single data stream into one wide 22 MHz channel. It is inherently limited to an 11 Mbps data rate because of the consistent 11-MHz chipping rate that feeds into the RF modulation. To scale beyond that limit, a vastly different approach is needed.

In contrast, orthogonal frequency division multiplexing (OFDM) sends data bits in *parallel* over multiple frequencies, all contained in a single 20 MHz channel. Each channel is divided into 64 subcarriers (also called subchannels or tones) that are spaced 312.5 kHz apart. The subcarriers are broken down into the following types:

- **Guard**—12 subcarriers are used to help set one channel apart from another.
- **Pilot**—4 subcarriers are equally spaced to help receivers lock onto the channel.
- **Data**—48 subcarriers are devoted to carrying data.

OFDM Operation with 48 Parallel Subcarriers



Examples of Phase and Amplitude Changes with 16-QAM.