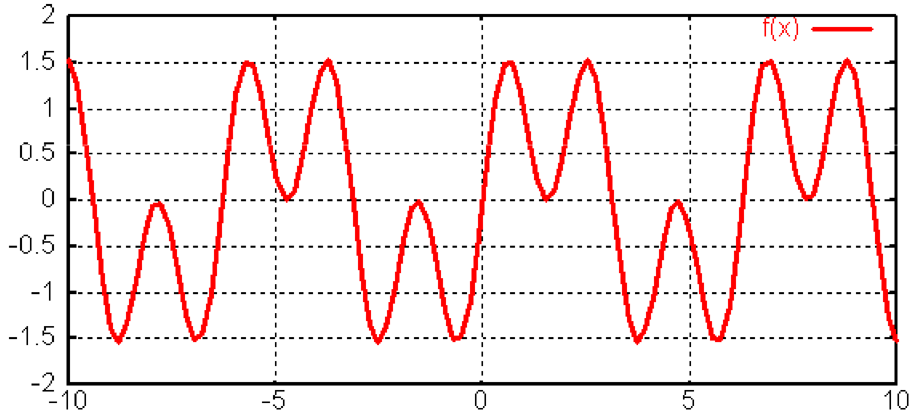
Signal Encoding Techniques

Chapter 6

Signals



Signals are electric or electromagnetic encoding of data

Computers Use Signals for Communcation

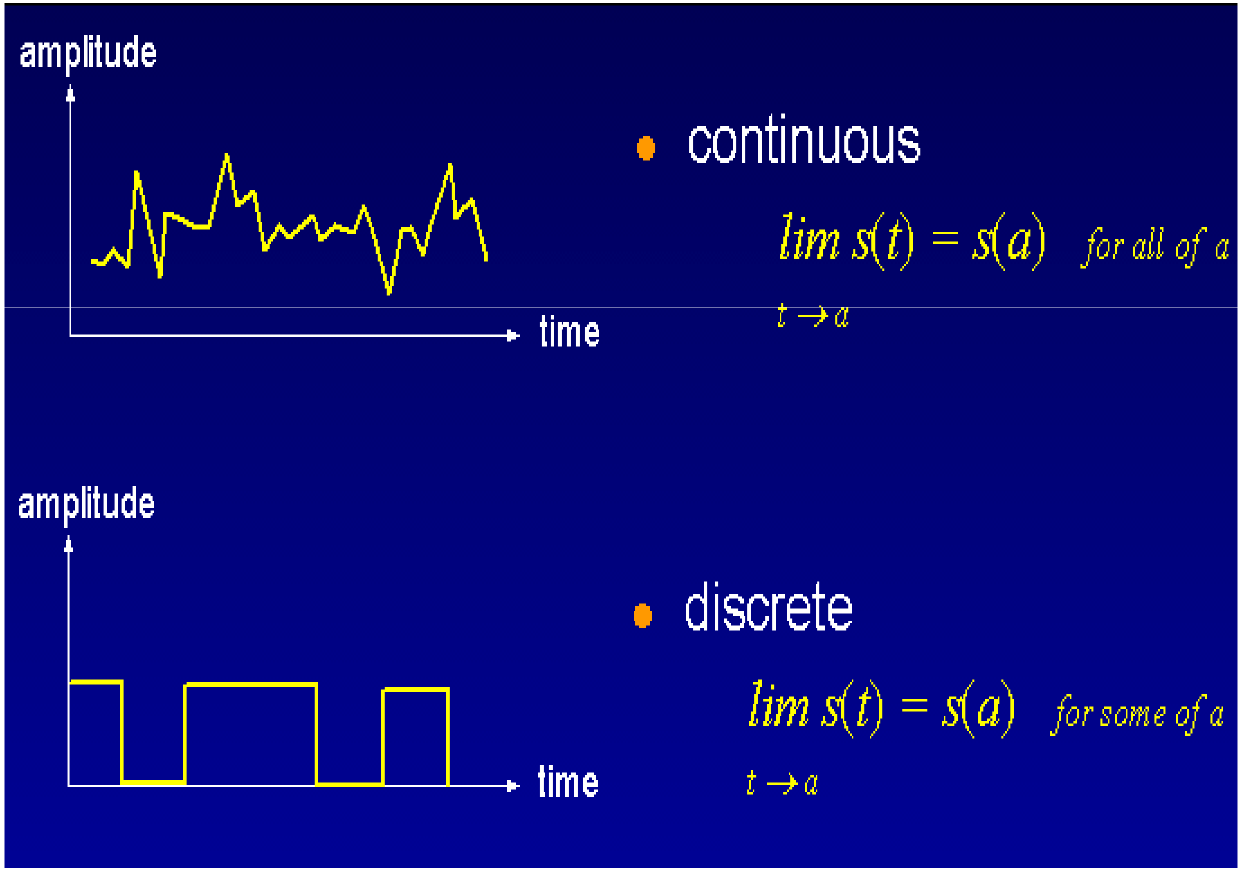
- Computers transmit data using digital signals, sequences of specified voltage levels. Graphically they are often represented as a square wave.
- Computers sometimes communicate over telephone line using analog signals, which are formed by continuously varying voltage levels.

Signal = Function of Time

- The signal is a function of time. Horizontal axis represents time and the vertical axis represents the voltage level.
- Signal represents data OR Data is encoded by means of a signal
- Signal is what travels on a communication medium
- An understanding of signals is required so that suitable signal may be chosen to represent data

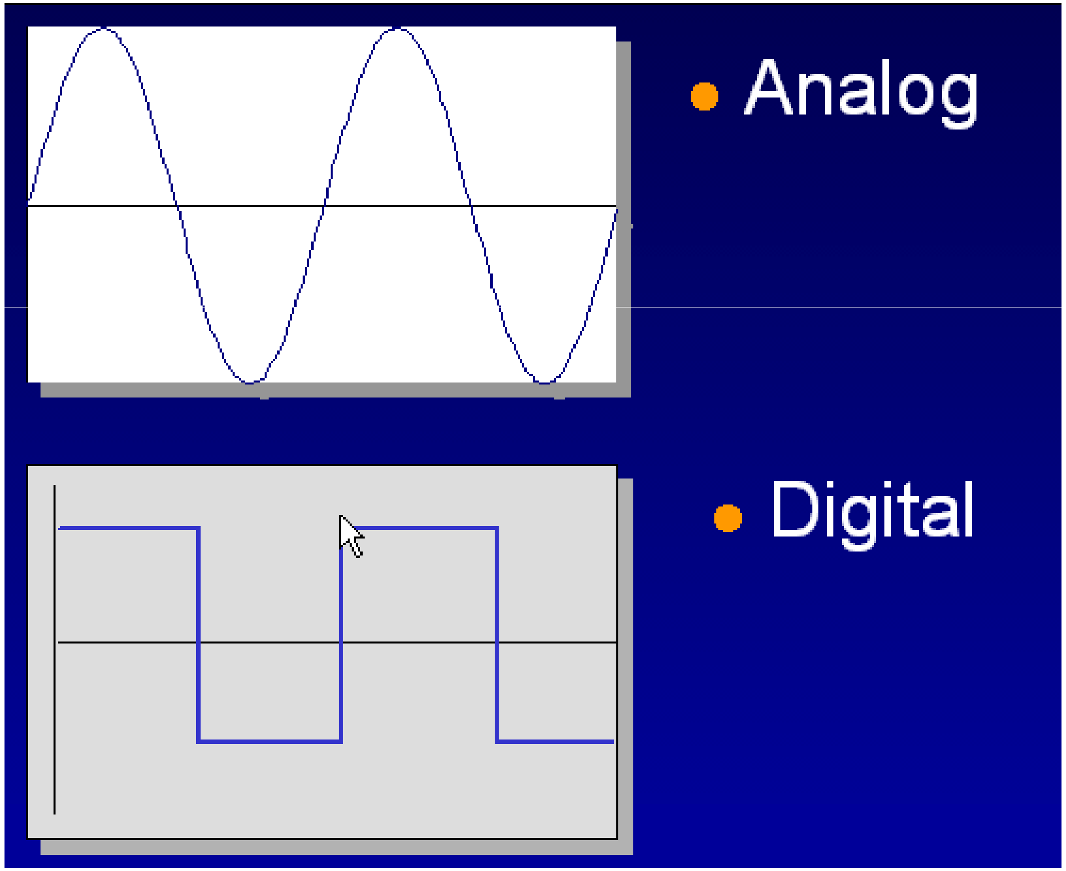
Continuous and Discrete Signal

- Continuous or Analog signals take on all possible values of amplitude
- Digital or Discrete Signals take on finite set of voltage levels

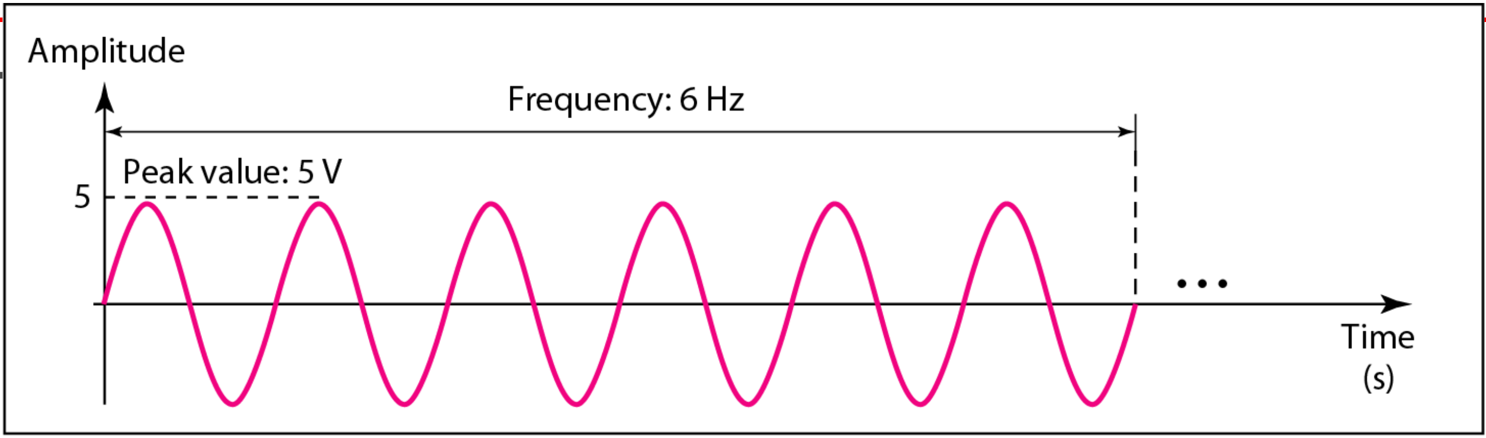


Analog and Digital Signal

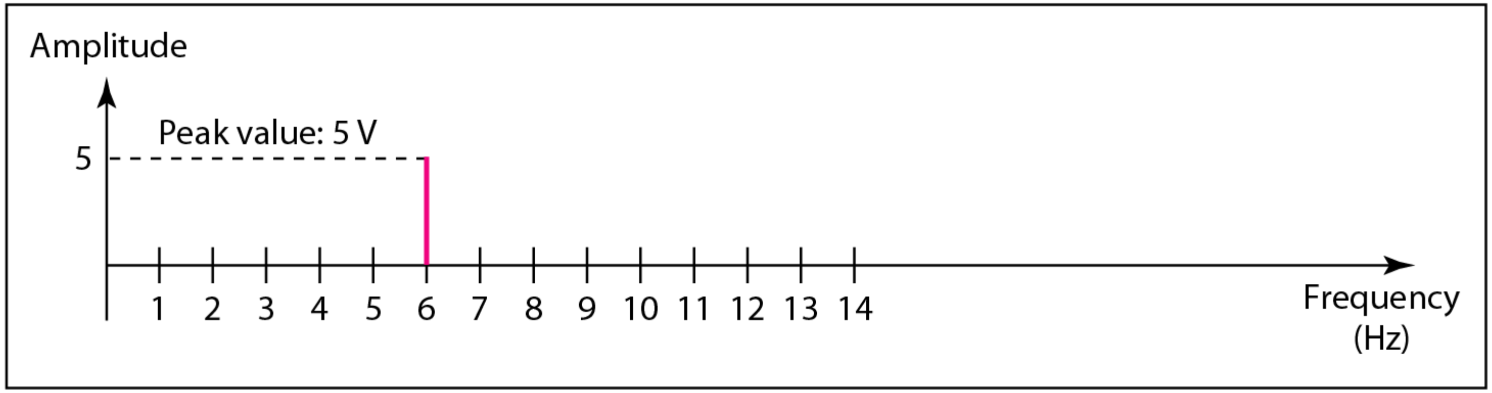
- Continuous/Analog signals take on all possible values of amplitude
- Digital or Discrete Signals take on finite set of voltage levels



The time-domain and frequency-domain plots of a sine wave



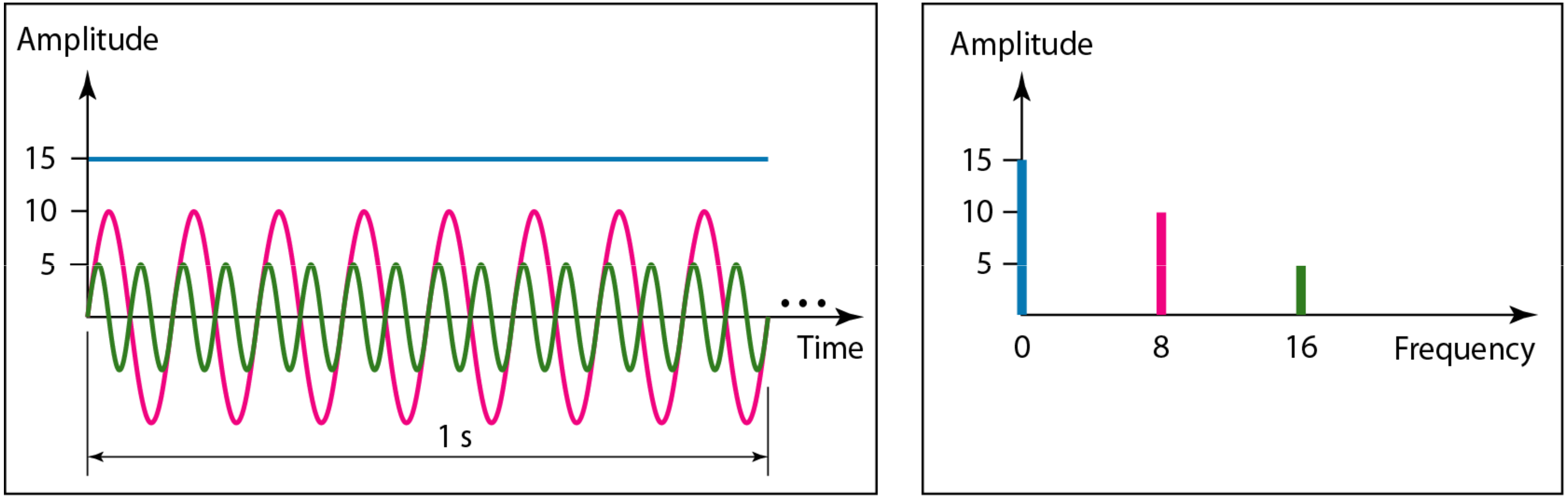
a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)



b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

A complete sine wave in the time domain can be represented by one single spike in the frequency domain.

Figure 3.8 *The time domain and frequency domain of three sine waves*

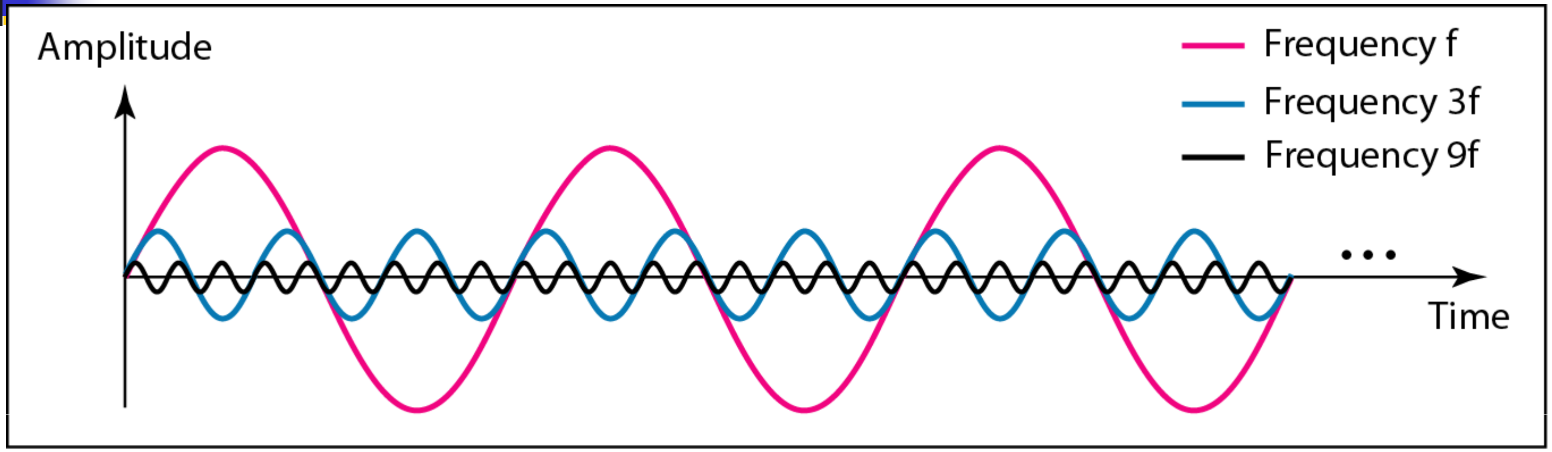


a. Time-domain representation of three sine waves with frequencies 0, 8, and 16

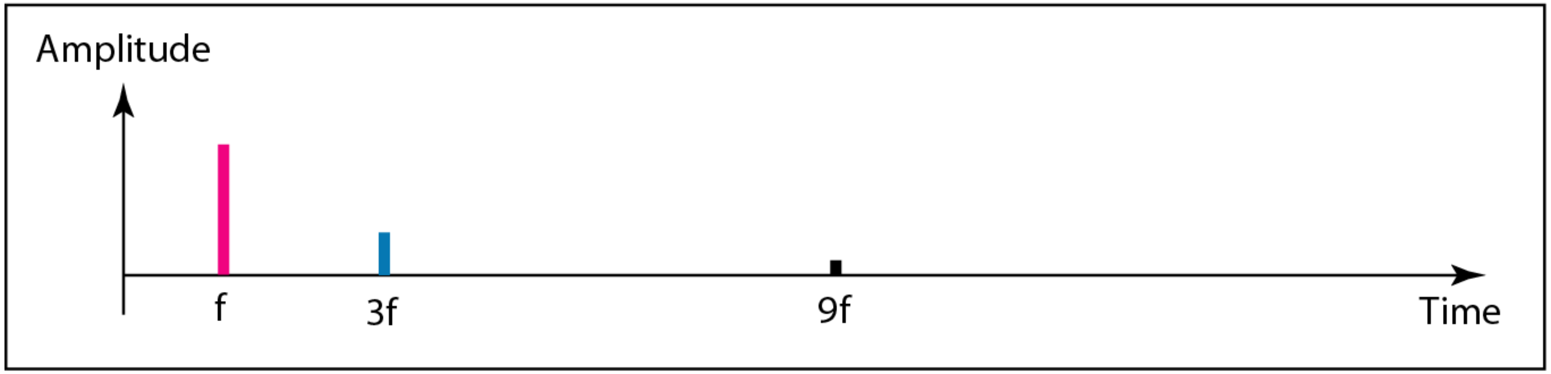
b. Frequency-domain representation of the same three signals

A single-frequency sine wave is not useful in data communications We need to send a composite signal, a signal made of many simple sine waves.

Figure 3.10 *Decomposition of a composite periodic signal in the time and frequency domains*

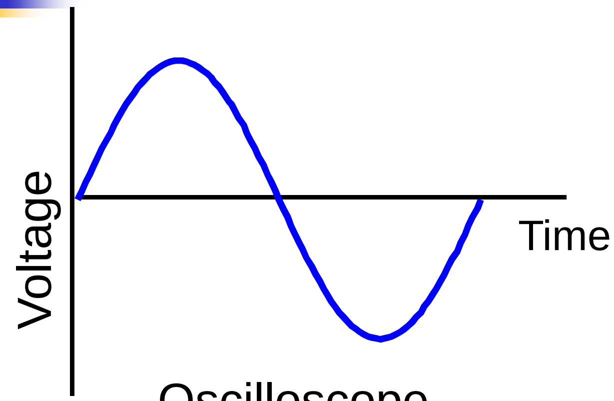


a. Time-domain decomposition of a composite signal

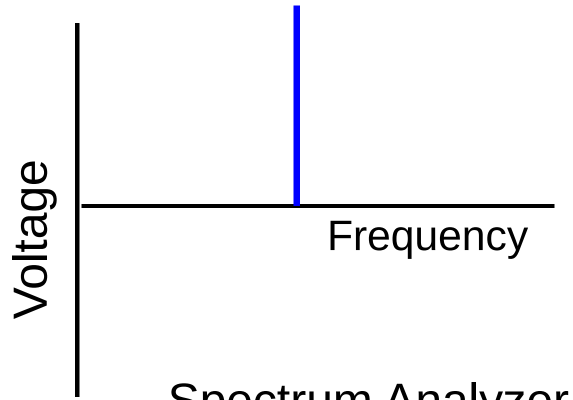


b. Frequency-domain decomposition of the composite signal

Sources Generate Sine Waves

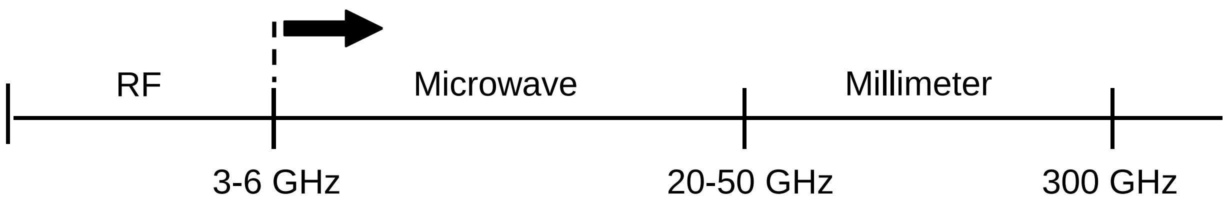


Oscilloscope



Spectrum Analyzer

This is the ideal output: most specs deal with deviations from the ideal and adding modulation to a sine wave



Analog and Digital Data

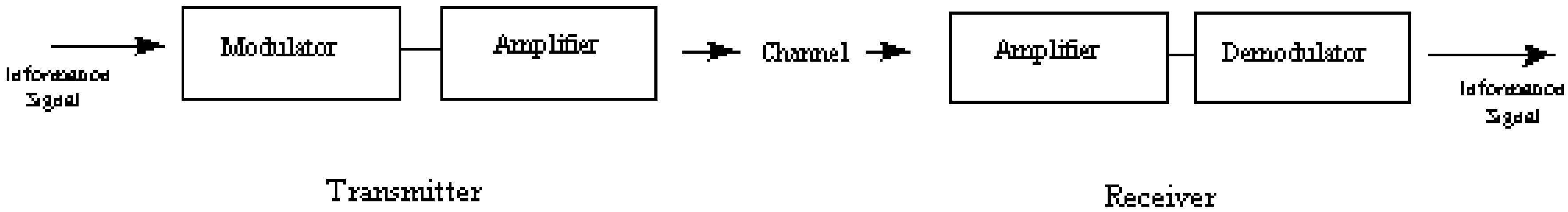
- Analog data take on all possible values. Voice and video are continuously varying patterns of intensity
- Digital data take on finite (countable) number of values. Example, ASCII characters, integers

- The result of modulating the carrier signal is called the modulated signal.

Introduction

- In the early days of mobiles, the wavelength of carrier waves was about 30 cm, so early mobile phones relied on extractable antennas
- A carrier wave with higher frequency allows for faster rates of data encoding and more information capacity, or "bandwidth." In addition, (e.g., 2 GHz)
- Today's carrier frequencies are about three times faster than the old 900 megahertz waves, resulting in minimum antennas sizes
- INCREASE IN FREQUENCY CAUSES SIZE OF ANTENNA TO DECREASE.
- "For efficient radiation of electromagnetic energy the radiating antenna should **be at least** of the order of one-tenth or more the wavelength of signal radiated" - the optimum antenna size is ½ or ¼ of a wavelength).

The communication system



Transmitter: *The transmitter modulates the information onto a carrier signal, amplifies the signal and broadcasts it over the channel*
Channel: *The medium which transports the modulated signal to the receiver.*
Receiver: *The sub-system that takes in the transmitted signal from the channel and processes it to retrieve the information signal. The receiver must be able to discriminate the signal from other signals which may using the same channel (called tuning), amplify the signal for processing and demodulate (remove the carrier) to retrieve the information*

Why is Modulation Required?

- **To achieve easy radiation:** If the communication channel consists of free space, antennas are required to radiate and receive the signal. Dimension of the antennas is limited by the corresponding wavelength.

Example: Voice signal bandwidth f=3kHz

$$\lambda = \frac{c}{f} = \frac{3 \cdot 10^8}{3 \cdot 10^3} = 10^5 \text{ m}$$

$\rightarrow$ λ/4=25000m!!

If we modulate a carrier wave @ $f_c = 100\text{MHz}$ with the voice signal

$$\lambda = \frac{c}{f} = \frac{3 \cdot 10^8}{100 \cdot 10^6} = 3 \text{ m}$$

$\rightarrow$ λ/4=75cm

Why Carrier frequency is used?

- To reduce the wavelength for efficient transmission and reception (the optimum antenna size is ½ or ¼ of a wavelength). A typical audio frequency of 3000 Hz will have a wavelength of 100 km and would need an effective antenna length of 25 km! By comparison, a typical carrier for FM is 100 MHz, with a wavelength of 3 m, and could use an antenna only 80 cm long.

- To allow simultaneous use of the same channel, called *multiplexing*. Each unique signal can be assigned a different carrier frequency (like radio stations) and still share the same channel.

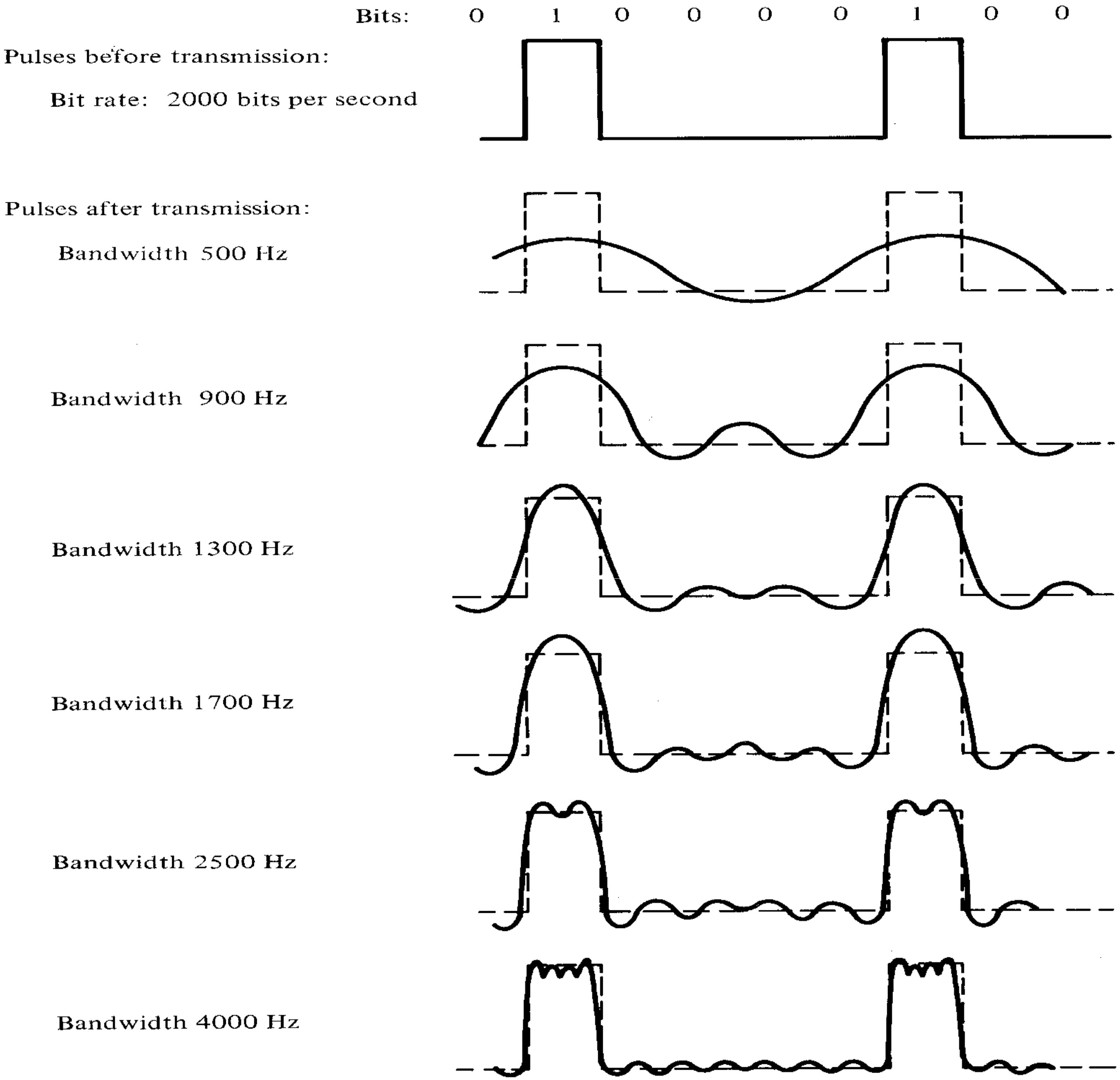
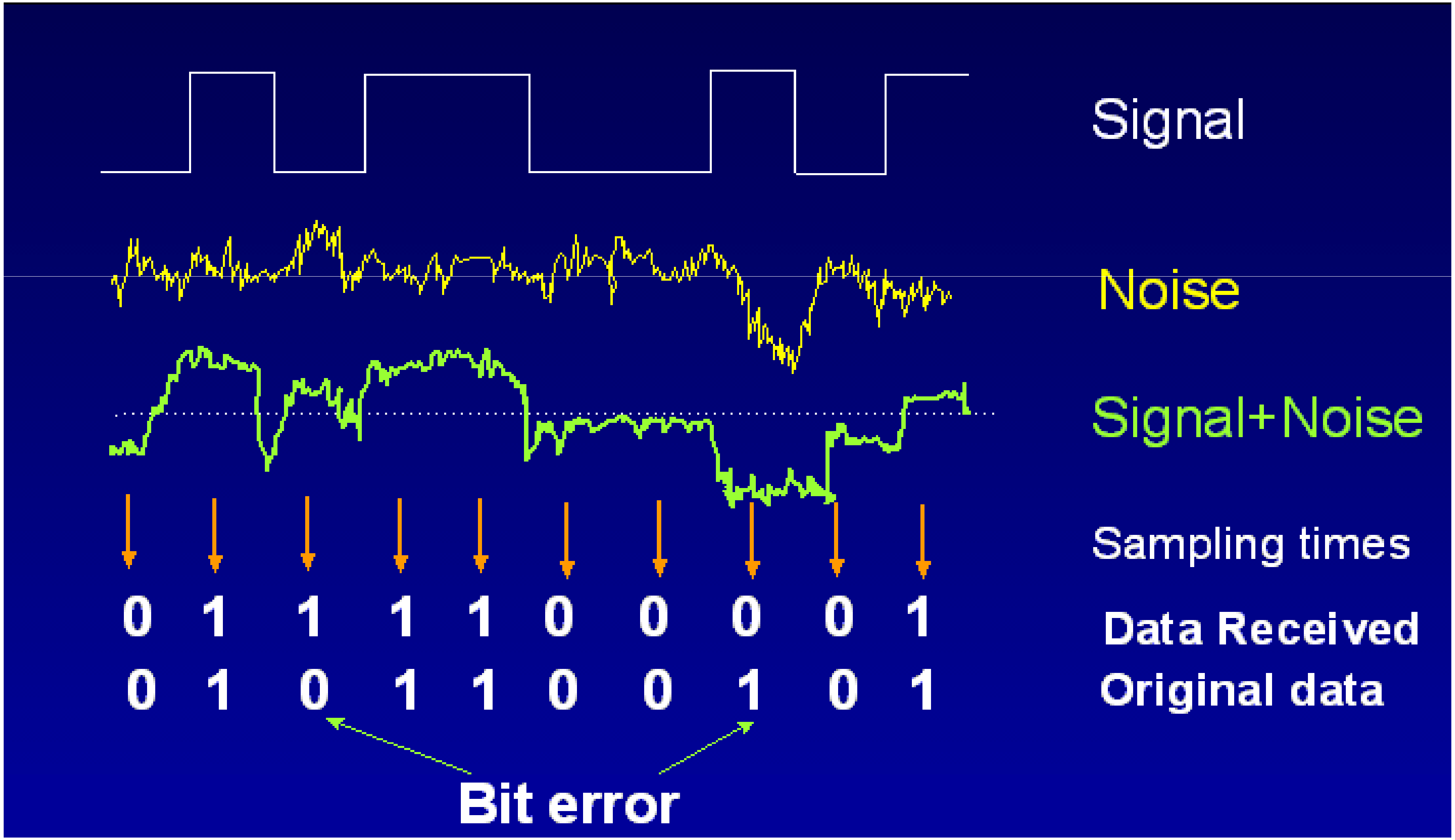


FIGURE 2.9. Effect of bandwidth on a digital signal.

Effect of Noise

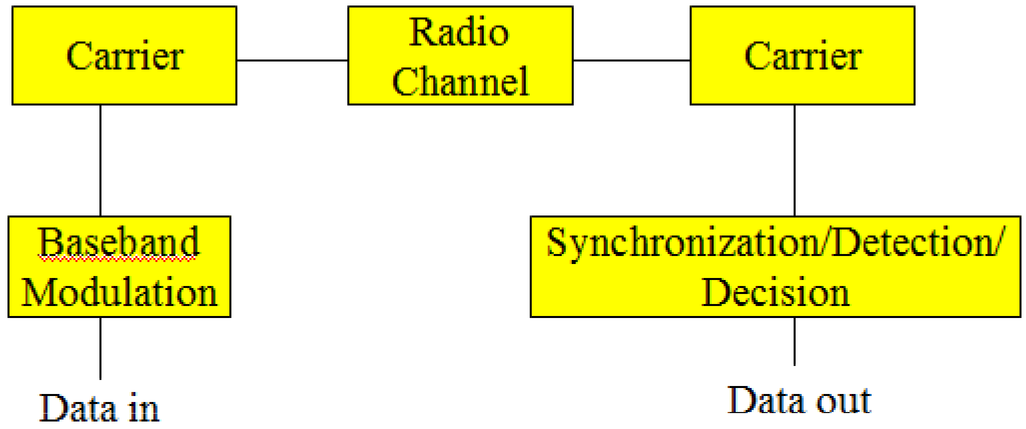


Signal Modulation

- The basic sine wave goes like *$V(t) = V_o \sin (2 p f t + \varphi)$* where the parameters are defined below:
- V(t) *the voltage of the signal as a function of time.* V_o *the amplitude of the signal* , f *the frequency of oscillation,* φ *the phase of the signal, representing the starting point of the cycle.*
- To modulate the signal just means to systematically vary one of the three parameters of the signal: amplitude, frequency or phase. Therefore, the type of modulation may be categorized as either
- AM: amplitude modulation
- FM: frequency modulation or
- PM: phase modulation

The form of encoding is chosen to optimize transmission medium (e.g., conserve bandwidth, minimum error

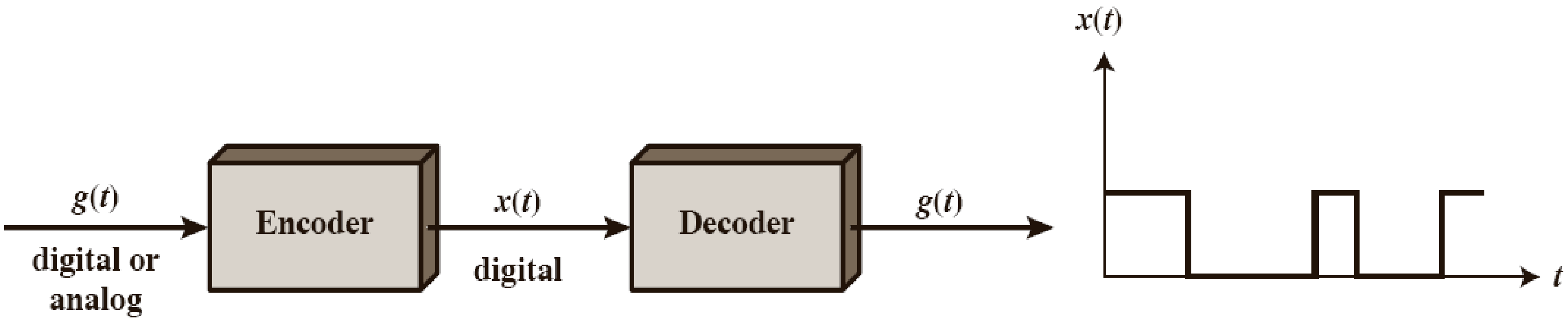
Modulation & Demodulation



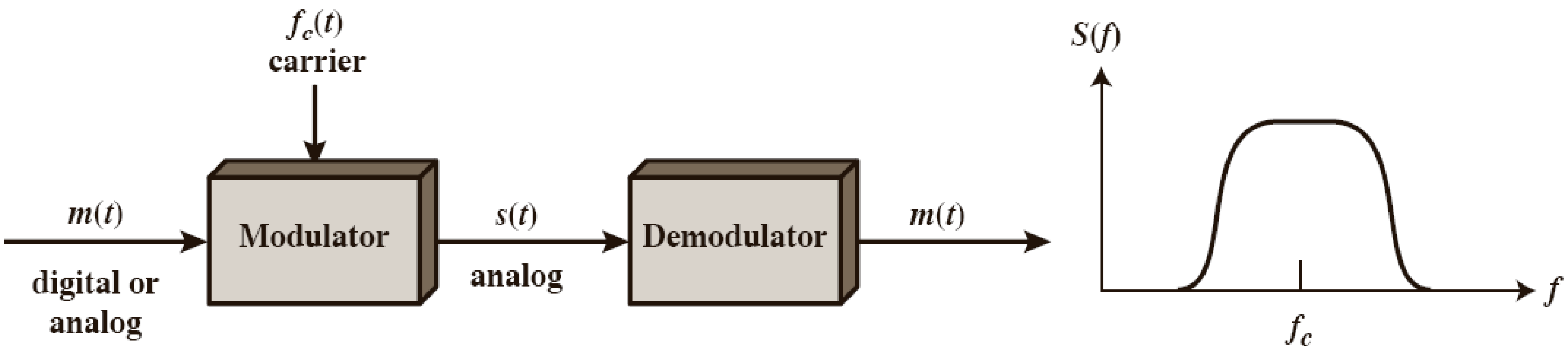
Data may be transmitted using a carrier signal by modulation. Modulation is the process of encoding source data onto a carrier signal with frequency fc. Amplitude, phase, frequency. Bandlimited

- **Sinusoidal waves, pulse train, square wave, etc. can be used as carriers**

Digital Vs Analog



(a) Encoding onto a digital signal



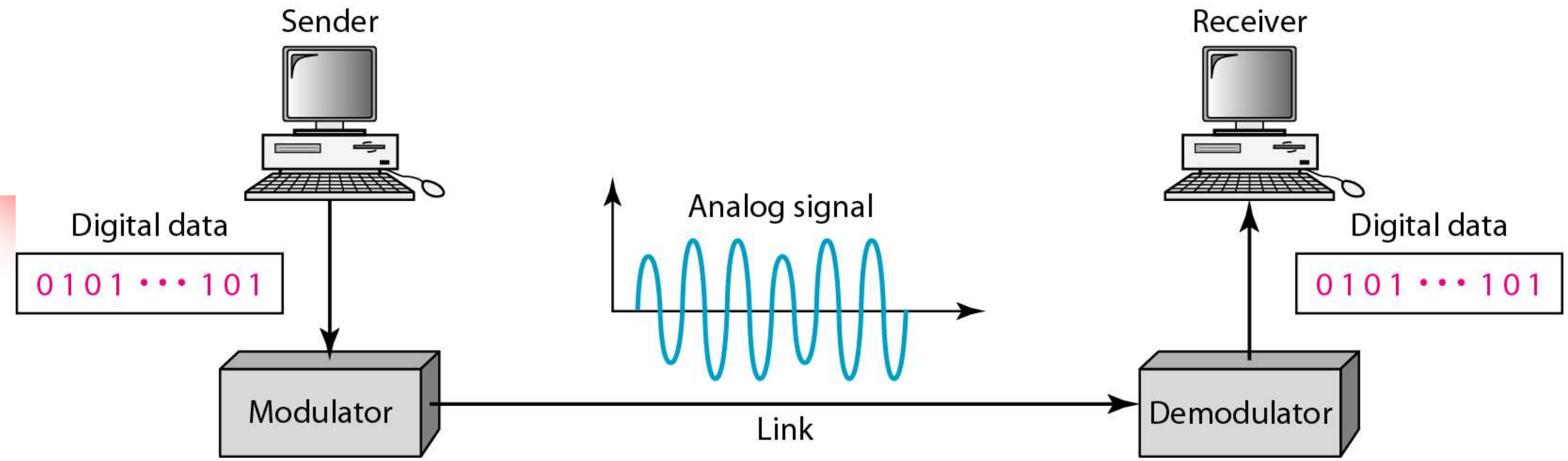
(b) Modulation onto an analog signal

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 (wireless)
  - A carrier signal (frequency f_c) performs the function of transporting the digital data in an analog waveform.
- Analog data, analog signal
  - Analog data in electrical form can be transmitted easily and cheaply
  - Done with voice transmission over voice-grade lines



Terms

- Data element, bits, a signal binary 0 or 1
- Data rate, bits per second, the rate at which data elements are transmitted (bps).
- **Signal rate or modulation rate**, the rate at which signal elements are transmitted per second (baud) (at which the signal level is changed).

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

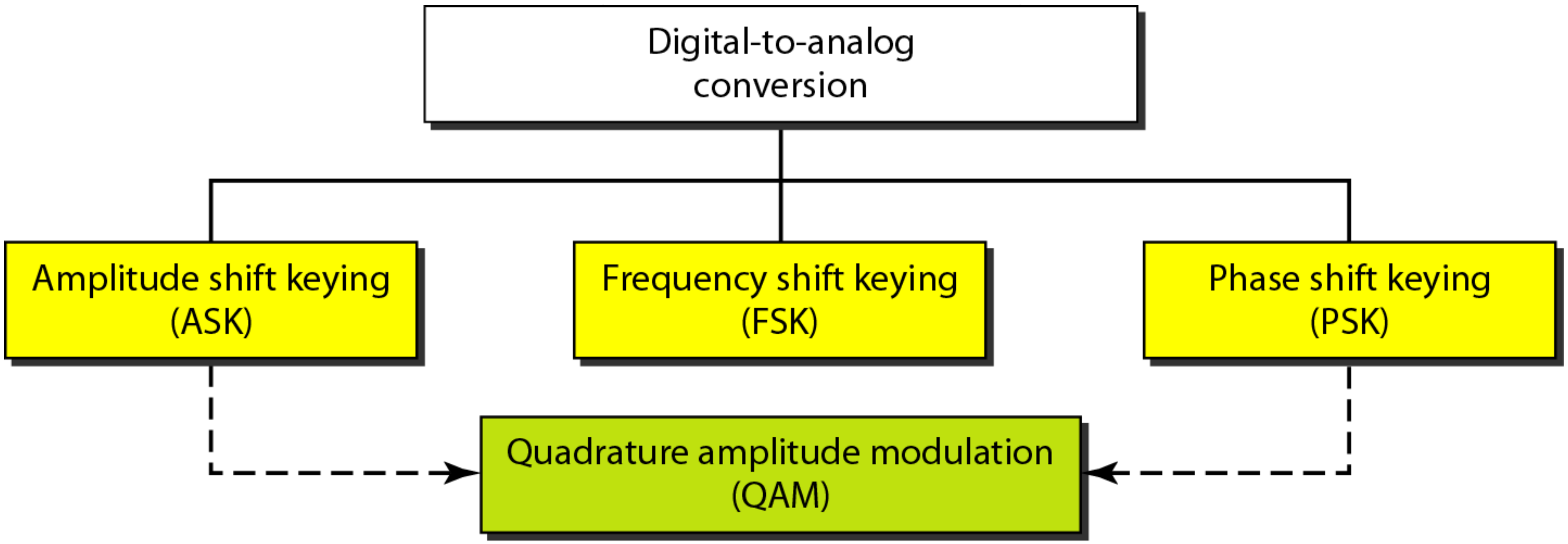
- Signal spectrum
  - With lack of high-frequency components, less bandwidth required
  - With no dc component, ac coupling via transformer possible (excellent electrical isolation and may solve the interference problem)
  - Transfer function of a channel is worse near band edges. A good signal design should concentrate the transmitted power near the middle of the transmission bandwidth.
- Clocking
  - The receiver must determine the begining and end of each bit position. One expensive approach is to provide a separate clock channel to synchronize the transmitter and receiver.
  - The alternative is to provide some synchronization methods that is based on the transmitted signal.

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)



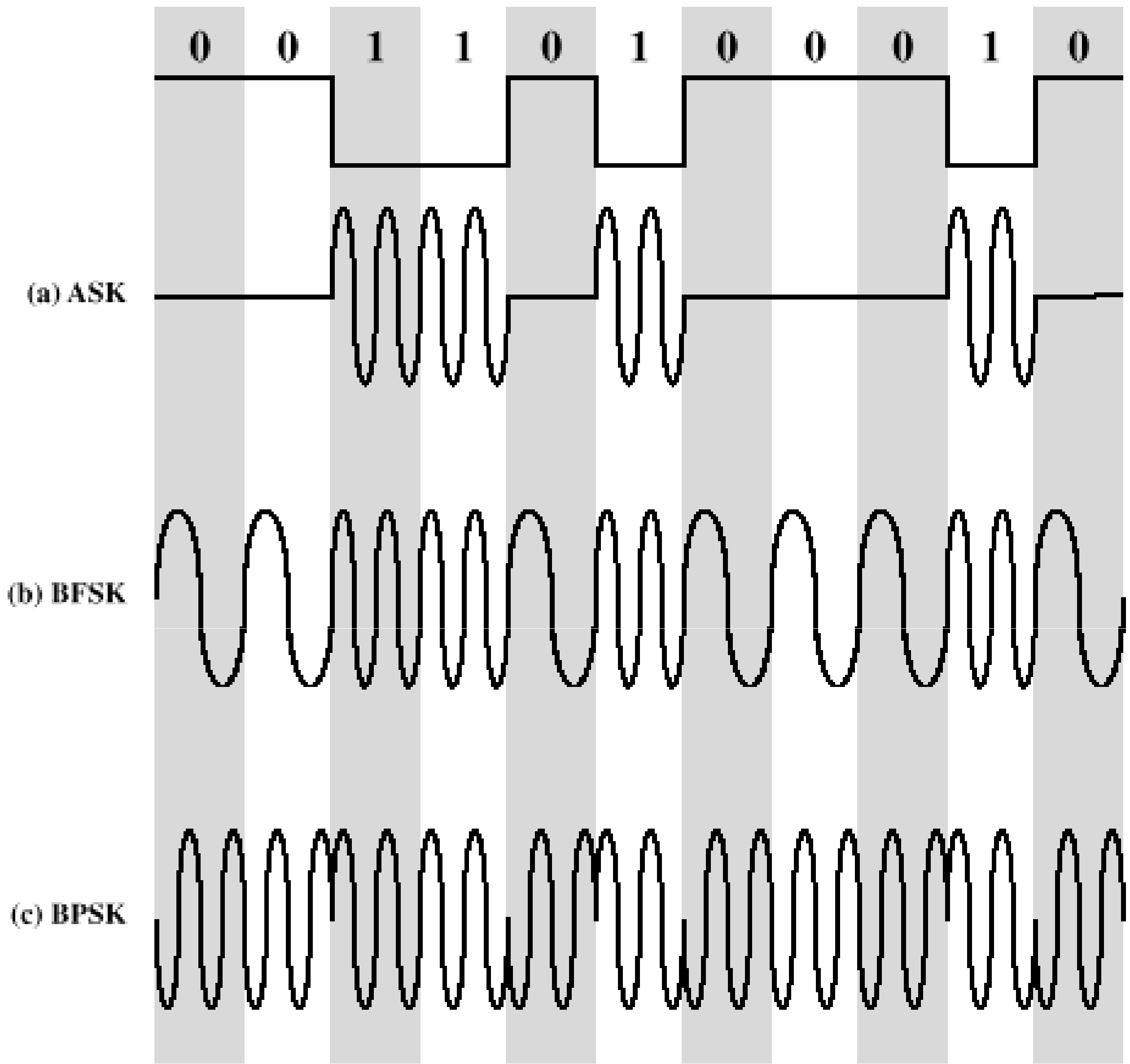


Figure 6.2 Modulation of Analog Signals for Digital Data

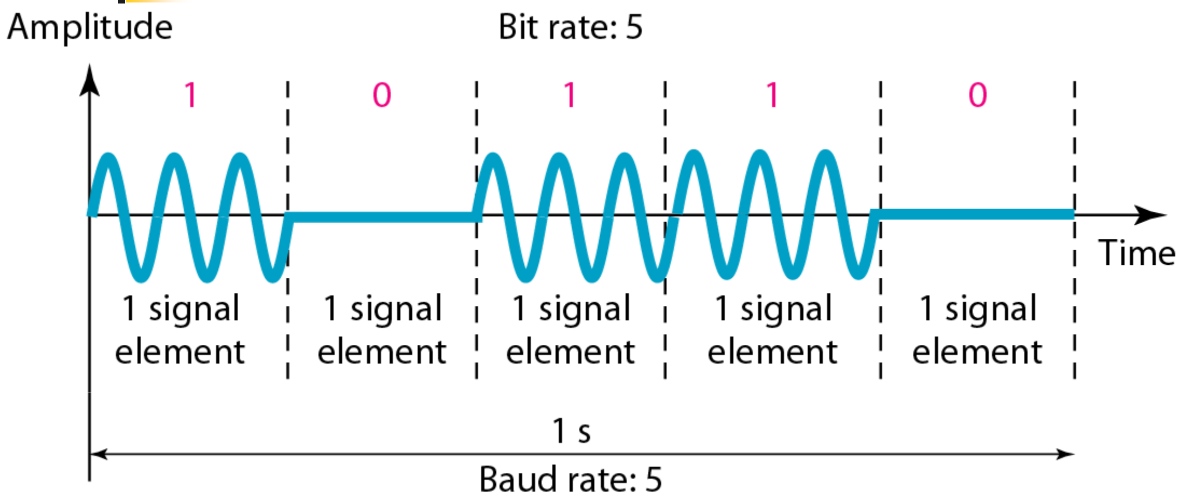
Amplitude-Shift Keying

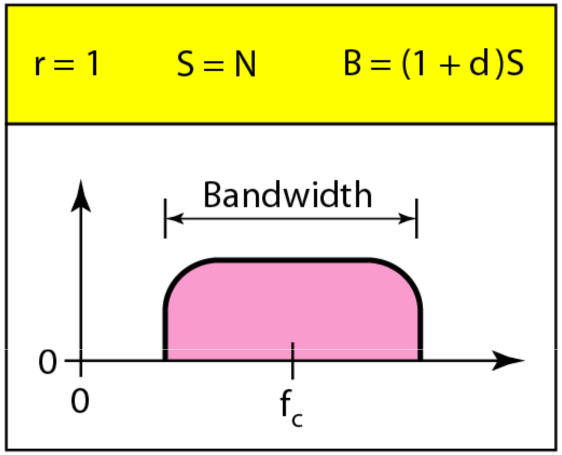
- One binary digit represented by presence of carrier, at constant amplitude
- Other binary digit represented by absence of carrier

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

- 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)

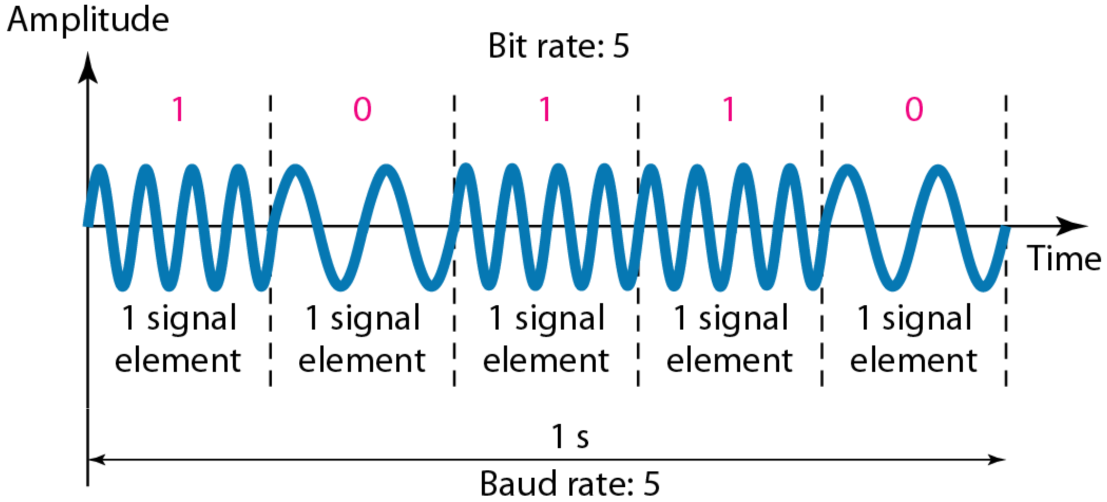
- Two binary digits represented by two different frequencies near the carrier frequency

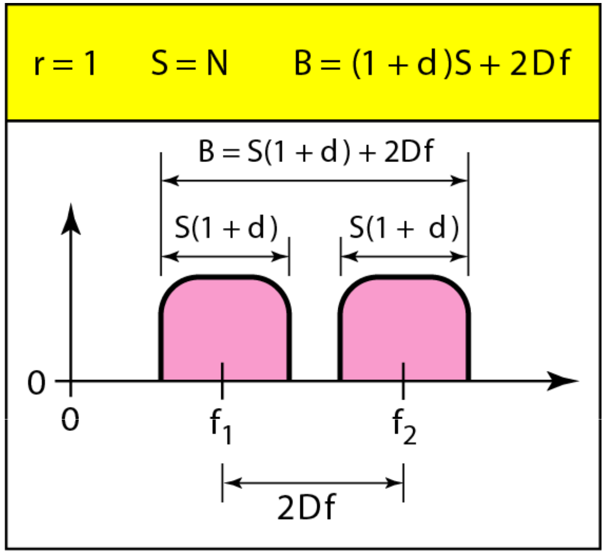
$$s(t) = \begin{cases} A\cos(2\pi f_1 t) & \text{binary } 1 \\ A\cos(2\pi f_2 t) & \text{binary } 0 \end{cases}$$

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

- For example, a "1" could be represented by *f1=fc +Δf*, and a "0" could be represented by *f2=fc-Δf*.

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 even 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

Binary Frequency-Shift Keying (BFSK)

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- On voice-grade lines, used up to 1200bps
- Used for high-frequency (3 to 30 MHz) radio transmission
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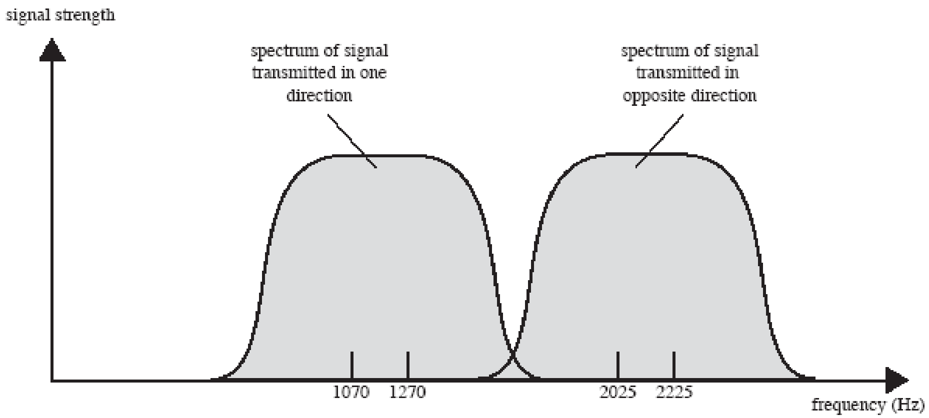


Figure 6.3 Full-Duplex FSK Transmission on a Voice-Grade Line

Multiple Frequency-Shift Keying (MFSK)

- To match the data rate of the input bit stream, each output signal element is held for a period of Ts=LT second,where T is the bit period (data rate=1/T) - So, one signal element encodes L bits
- 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)

Example of MFSK with M=4. an input bit stream of 20 bits is encoded 2 bit at a time, with each of the four possible 2-bit combinations transmitted as a different frequency. Each column represents a time unit Ts in which a single 2-bit signal elements is transmitted. The shaded rectangle in the column indicates the frequency transmitted during that time unit.

With fc=250khz, fd=25khz, and M=8, we have the following frequency assignments for each of the 8 possible 3 bit combination This scheme can support a data rate =1/T = 2Lfd = 150 kbps

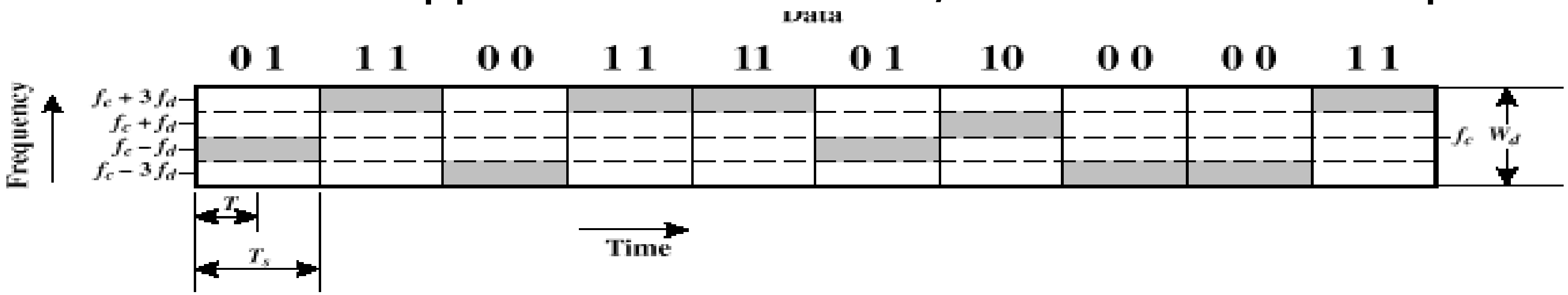


Figure 6.4 MFSK Frequency Use ($M = 4$)

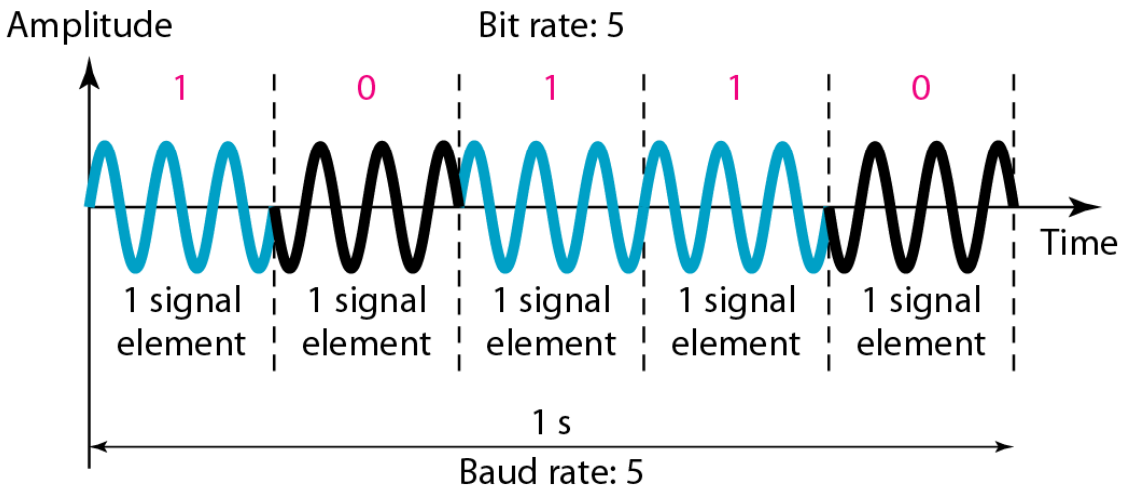
Phase-Shift Keying (PSK)

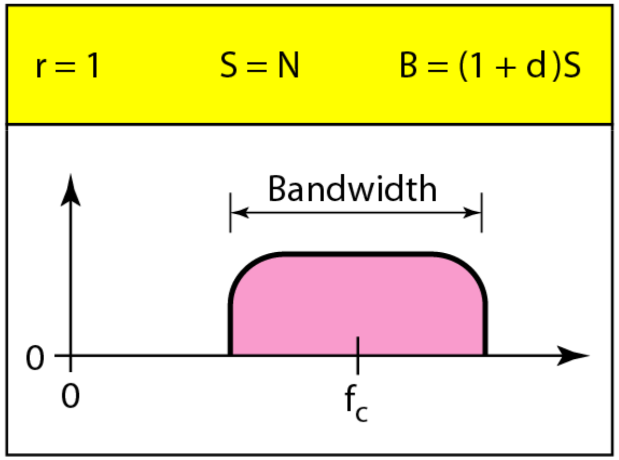
- Two-level PSK (BPSK)
- Uses two phases to represent binary digits

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

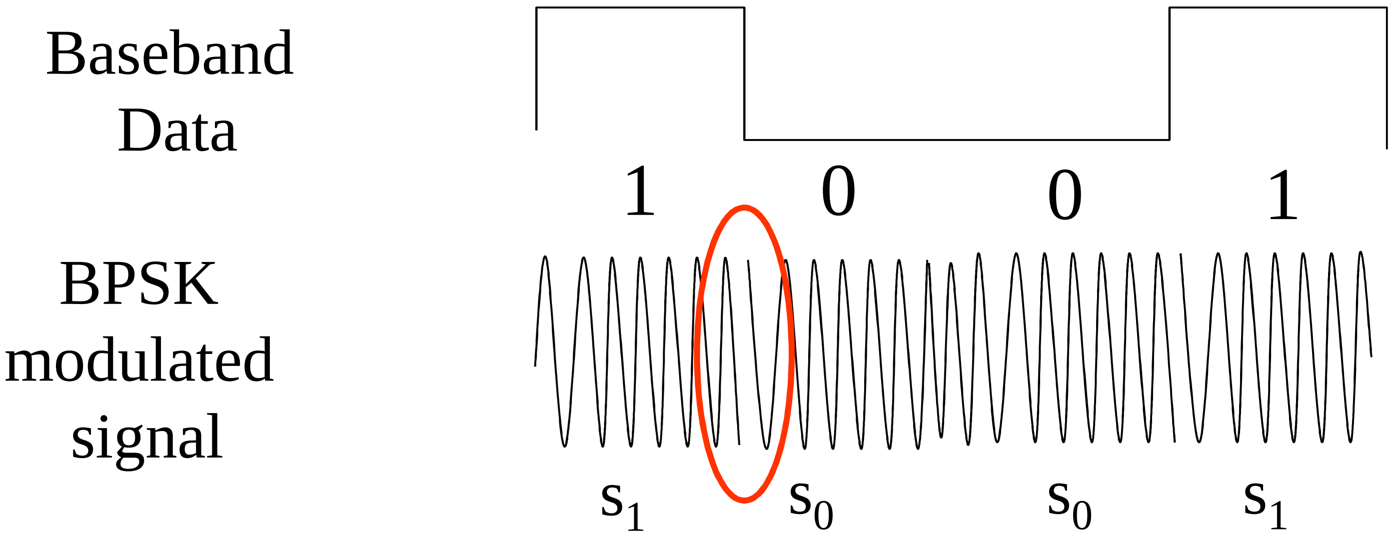
$$= \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ -A\cos(2\pi f_c t) & \text{binary } 0 \end{cases}$$

Phase-Shift Keying (PSK)





Phase Shift Keying (PSK)



where s_0 =-Acos(ω_ct) and s_1 =Acos(ω_ct)

- Major drawback – rapid amplitude change between symbols due to phase discontinuity, which requires infinite bandwidth. Binary Phase Shift Keying (BPSK) demonstrates better performance than ASK and BFSK
- BPSK can be expanded to a M-ary scheme, employing multiple phases and amplitudes as different states

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
- PSK is much more robust than ASK as it is not that vulnerable to noise, which changes amplitude of the signal.

Phase-Shift Keying (PSK)

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

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

Quadrature PSK

- To increase the bit rate, we can code 2 or more bits onto one signal element.
- In QPSK, we parallelize the bit stream so that every two incoming bits are split up and PSK a carrier frequency. One carrier frequency is phase shifted 90^o from the other - in quadrature.
- The two PSKed signals are then added to produce one of 4 signal elements. L = 4 here.

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

Phase-Shift Keying (PSK)

- Differential PSK (DPSK)
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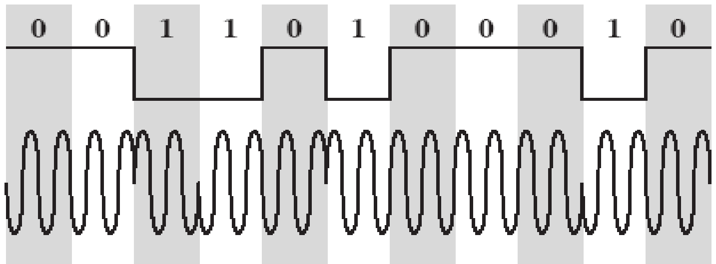


Figure 6.5 Differential Phase-Shift Keying (DPSK)

Summary: Digital Modulation

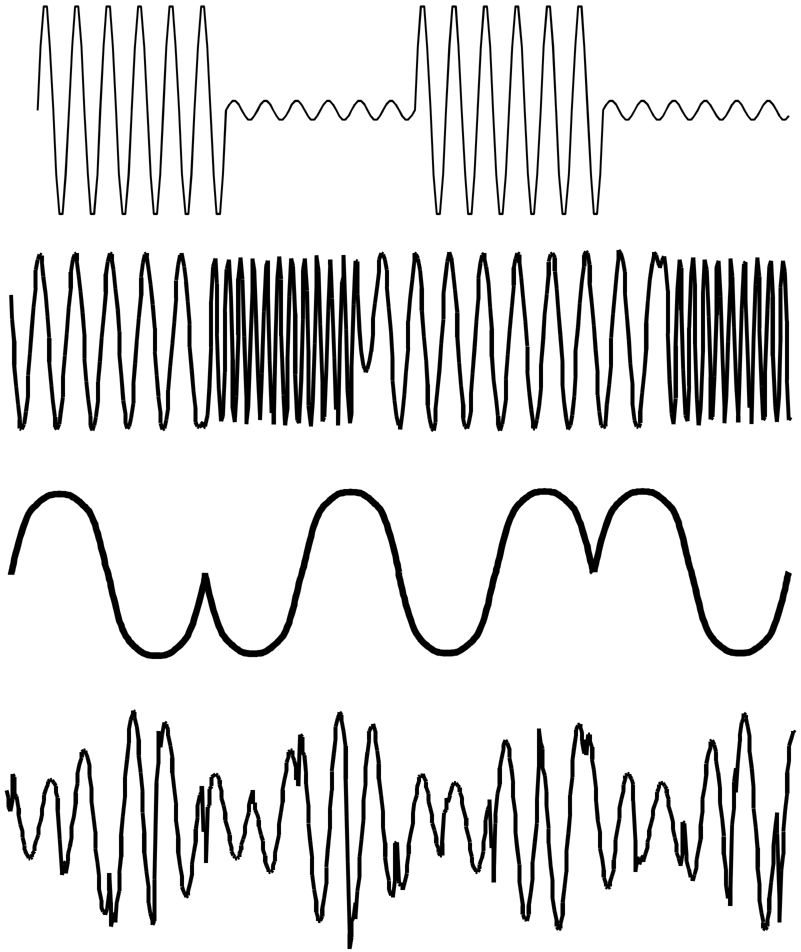
...signal characteristics to modify

Amplitude

Frequency

Phase

Both Amplitude
and Phase



Performance

- Bandwidth of modulated signal (B_T)
  - ASK, PSK $B_T=(1+r)R$
  - FSK $B_T=2\mathrm{D}F+(1+r)R$

    - R = bit rate
    - 0 < r < 1; related to how signal is filtered
    - $\mathrm{D}F = f_2 - f_c = f_c - f_1$

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

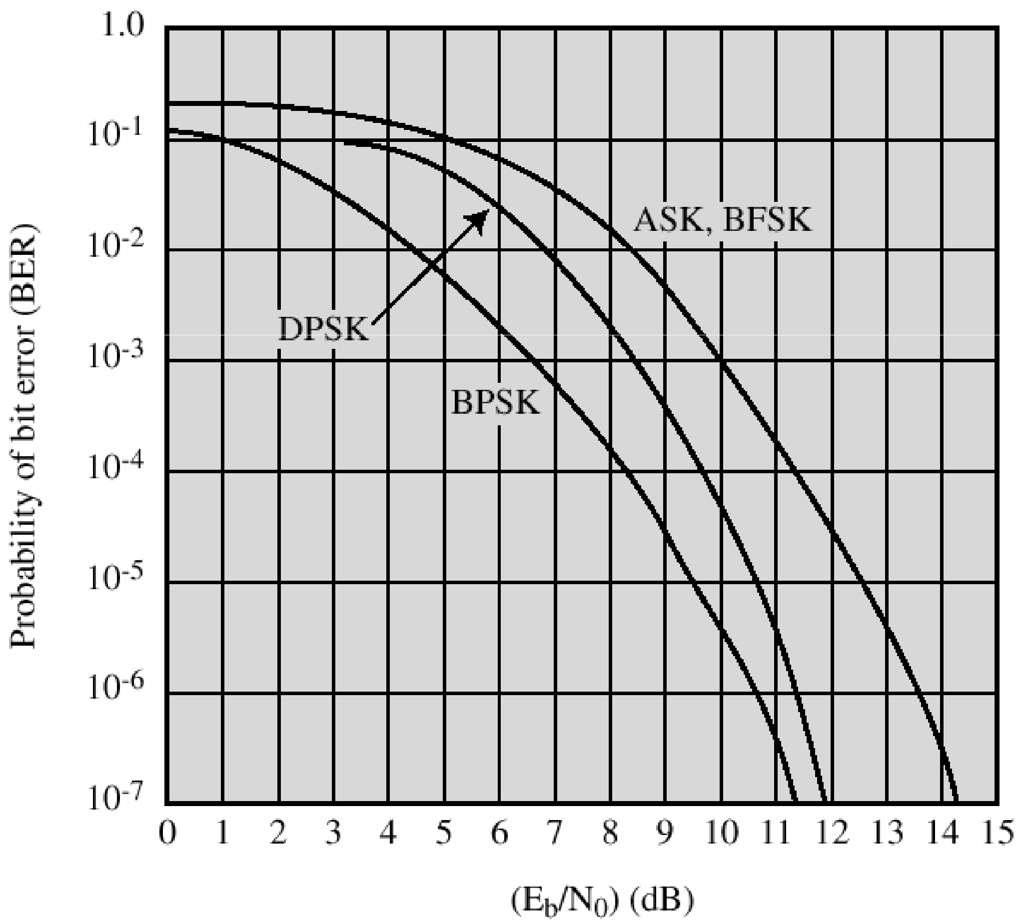
Performance

- Bandwidth efficiency — The ratio of data rate to transmission bandwidth (R/B_T)

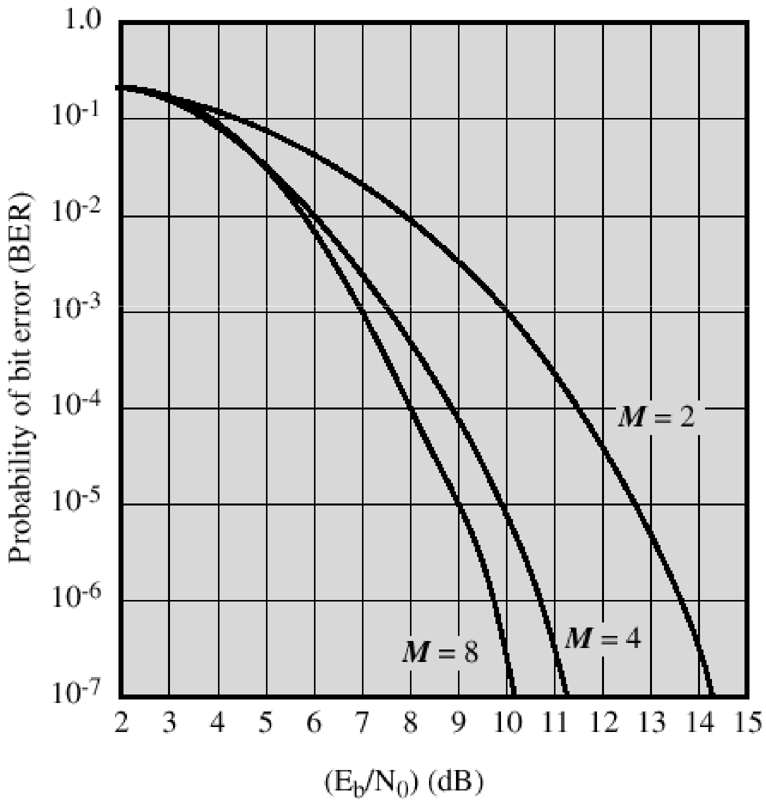
- For MFSK, with the increase of M, the bandwidth efficiency is decreased.

- For MPSK, with the increase of M, the bandwidth efficiency is increased.

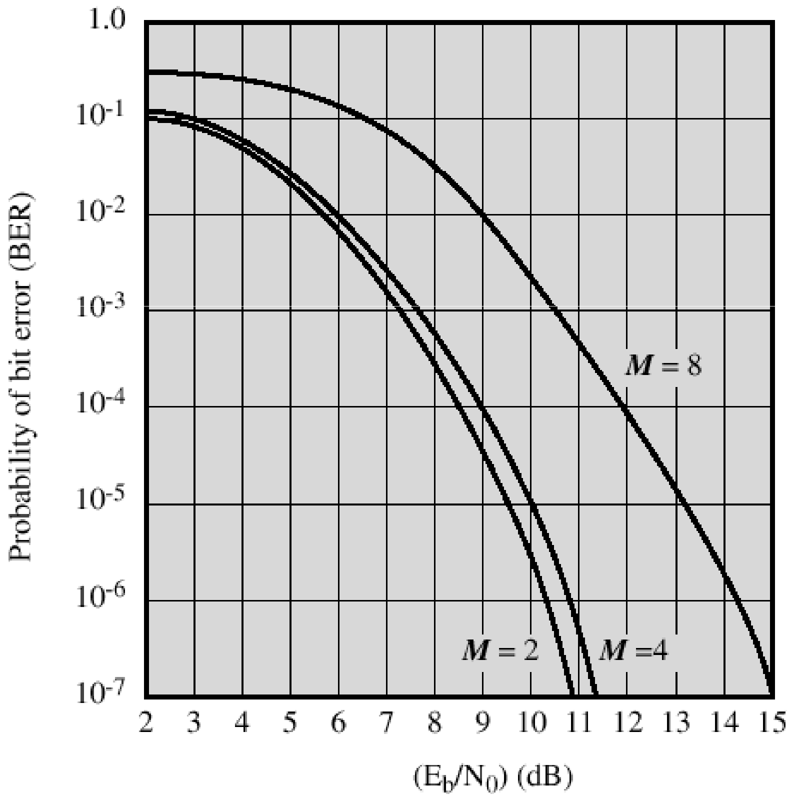
Performance



Performance



(a) Multilevel FSK (MFSK)



(b) Multilevel PSK (MPSK)

Performance

- Tradeoff between bandwidth efficiency and error performances: an increase in bandwidth efficiency results in an increase in error probability.

Minimum shift keying

$$MSK \quad s(t) = \begin{cases} \sqrt{\dfrac{2E_b}{T_b}} \cos(2\pi f_1 t + \theta(0)) & binary\ 1 \\ \sqrt{\dfrac{2E_b}{T_b}} \cos(2\pi f_2 t + \theta(0)) & binary\ 0 \end{cases}$$

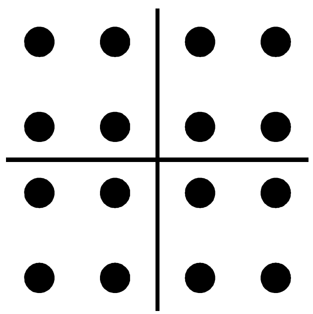
Where E_b is the transmitted signal energy per bit, and T_b is the bit duration, the phase $\theta(0)$ denotes the value of the phase at time t=0.

Quadrature Amplitude Modulation

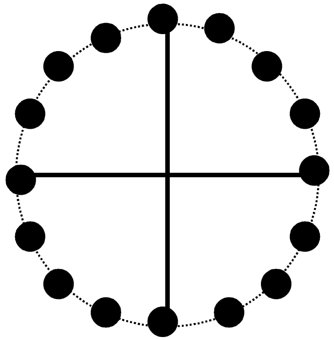
- QAM is a combination of ASK and PSK
  - Two different signals sent simultaneously on the same carrier frequency

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

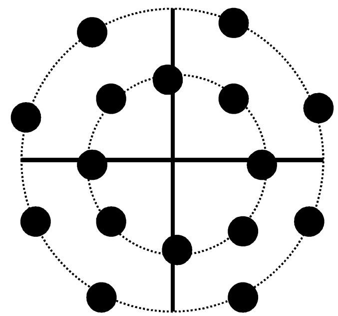
Multi-level (M-ary) Phase and Amplitude Modulation



16 QAM



16 PSK



16 APSK

- Amplitude and phase shift keying can be combined to transmit several bits per symbol.
  - Often referred to as *linear* as they require linear amplification.
  - More bandwidth-efficient, but more susceptible to noise.
- For M=4, 16QAM has the largest distance between points, but requires very linear amplification. 16PSK has less stringent linearity requirements, but has less spacing between constellation points, and is therefore more affected by noise.

Comparison of Modulation Types

Modulation Format	Bandwidth efficiency C/B	Log2(C/B)	Error-free Eb/No
16 PSK	4	2	18dB
16 QAM	4	2	15dB
8 PSK	3	1.6	14.5dB
4 PSK	2	1	10dB
4 QAM	2	1	10dB
BFSK	1	0	13dB
BPSK	1	0	10.5dB

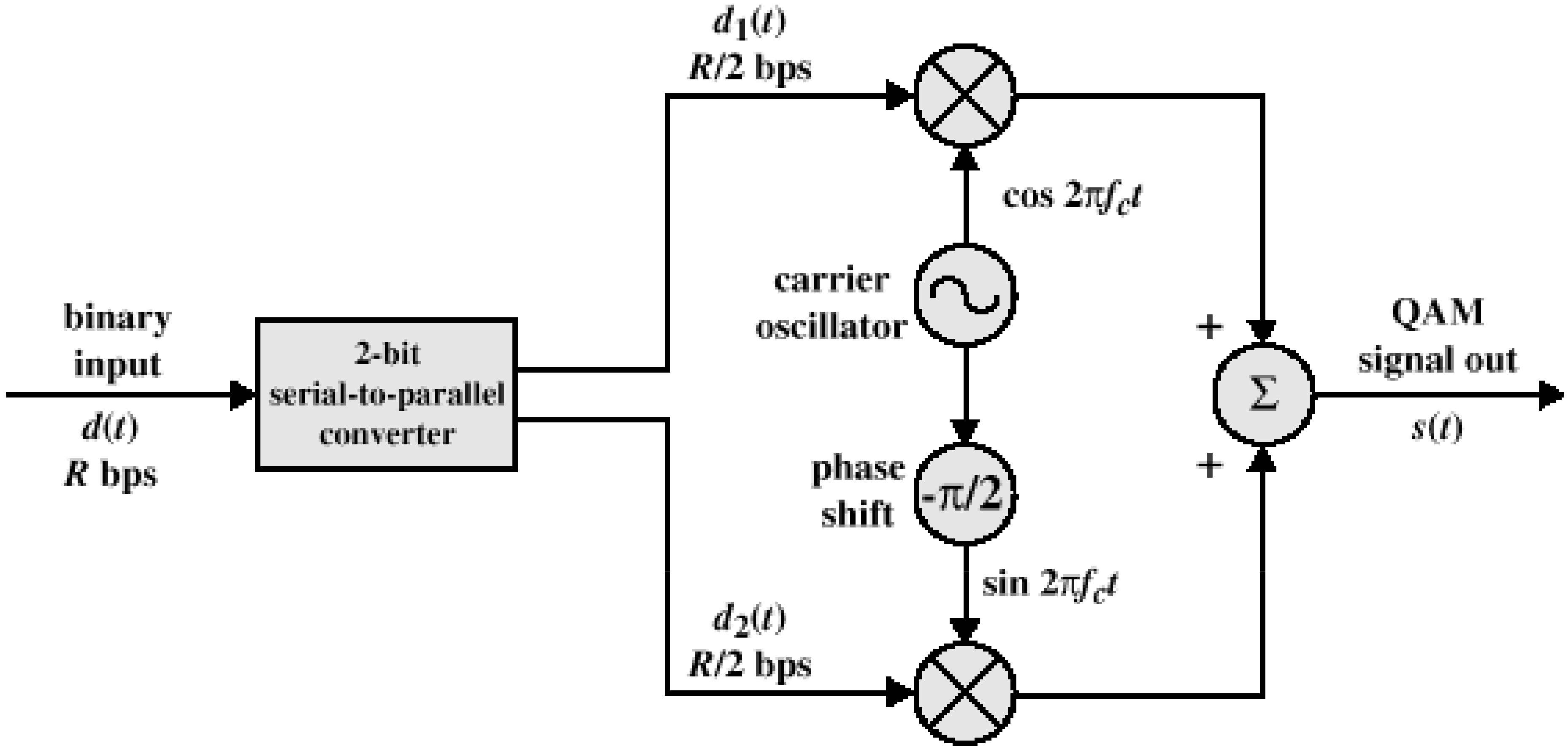


Figure 6.10 QAM Modulator

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

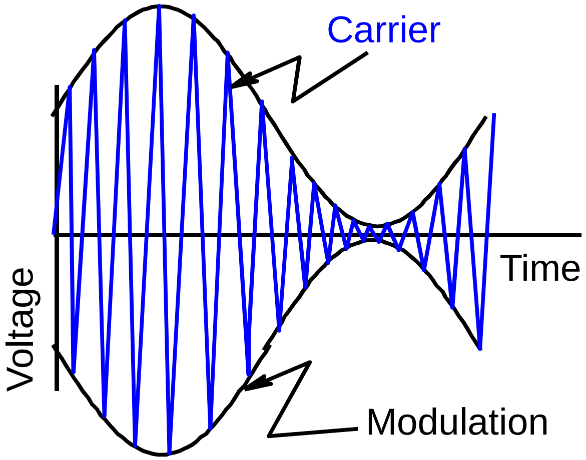
- Amplitude Modulation

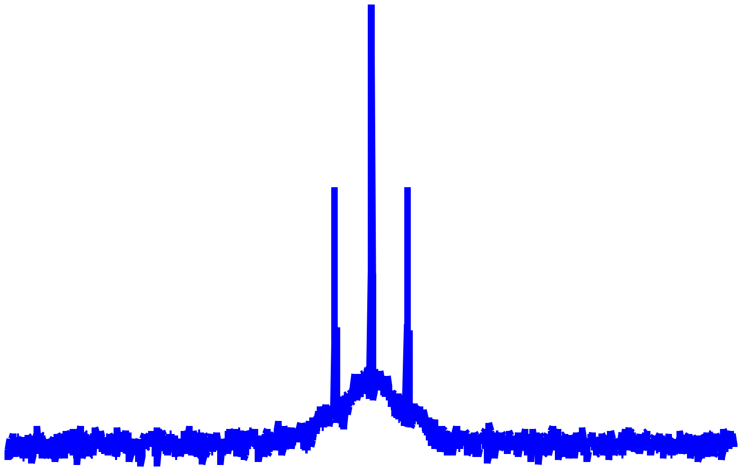
$$s(t) = [1 + n_a x(t)]\cos 2\pi f_c t$$

  - $\cos 2\pi f_c t$ = 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)

Modulation: Analog

Amplitude Modulation

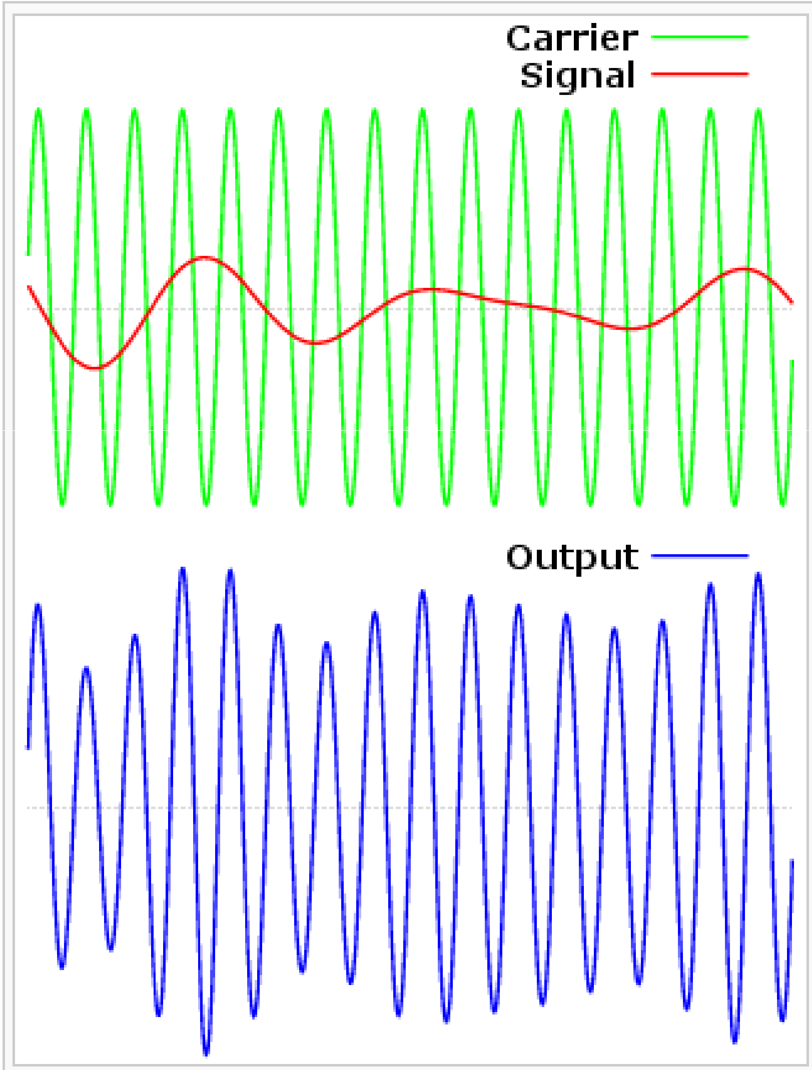


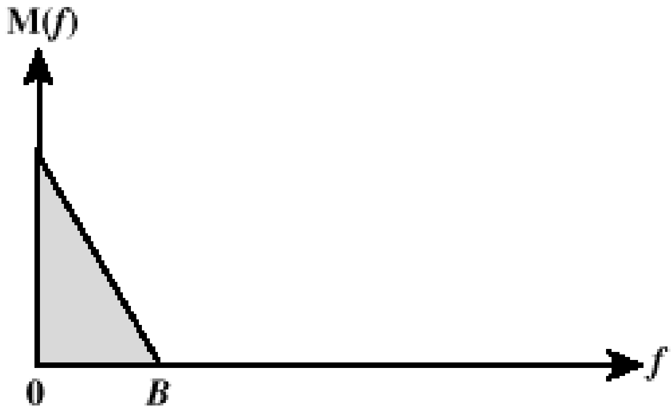


Important Signal Generator Specs for Amplitude Modulation

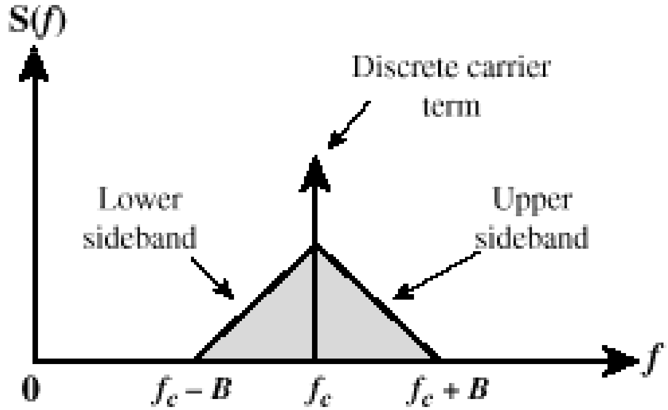
- Modulation frequency
- Linear AM
- Log AM
- Depth of modulation (Mod Index)

Amplitude modulation





(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

Other variants

- Double sideband suppressed carrier (DSBSC): filters out the carrier frequency and sends both sidebands.
- Vestigial sideband (VSB), uses one sideband and reduced-power carrier.

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

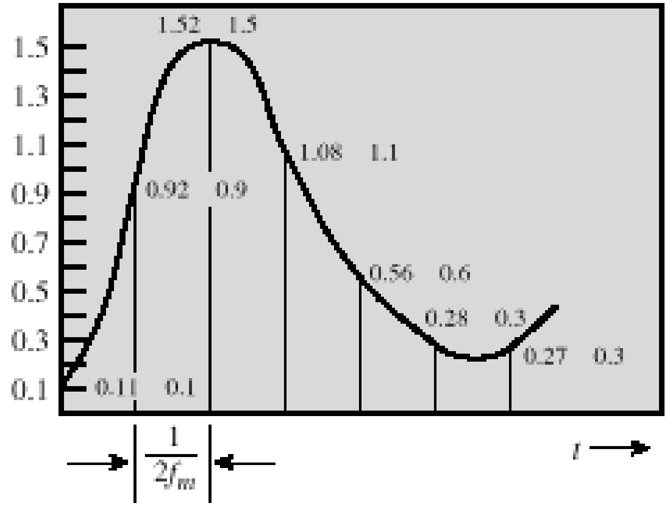
- Analog data to digital signal
  - Pulse code modulation (PCM)
  - Delta modulation (DM)

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
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	

Digit	Binary Equivalent	PCM waveform
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
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

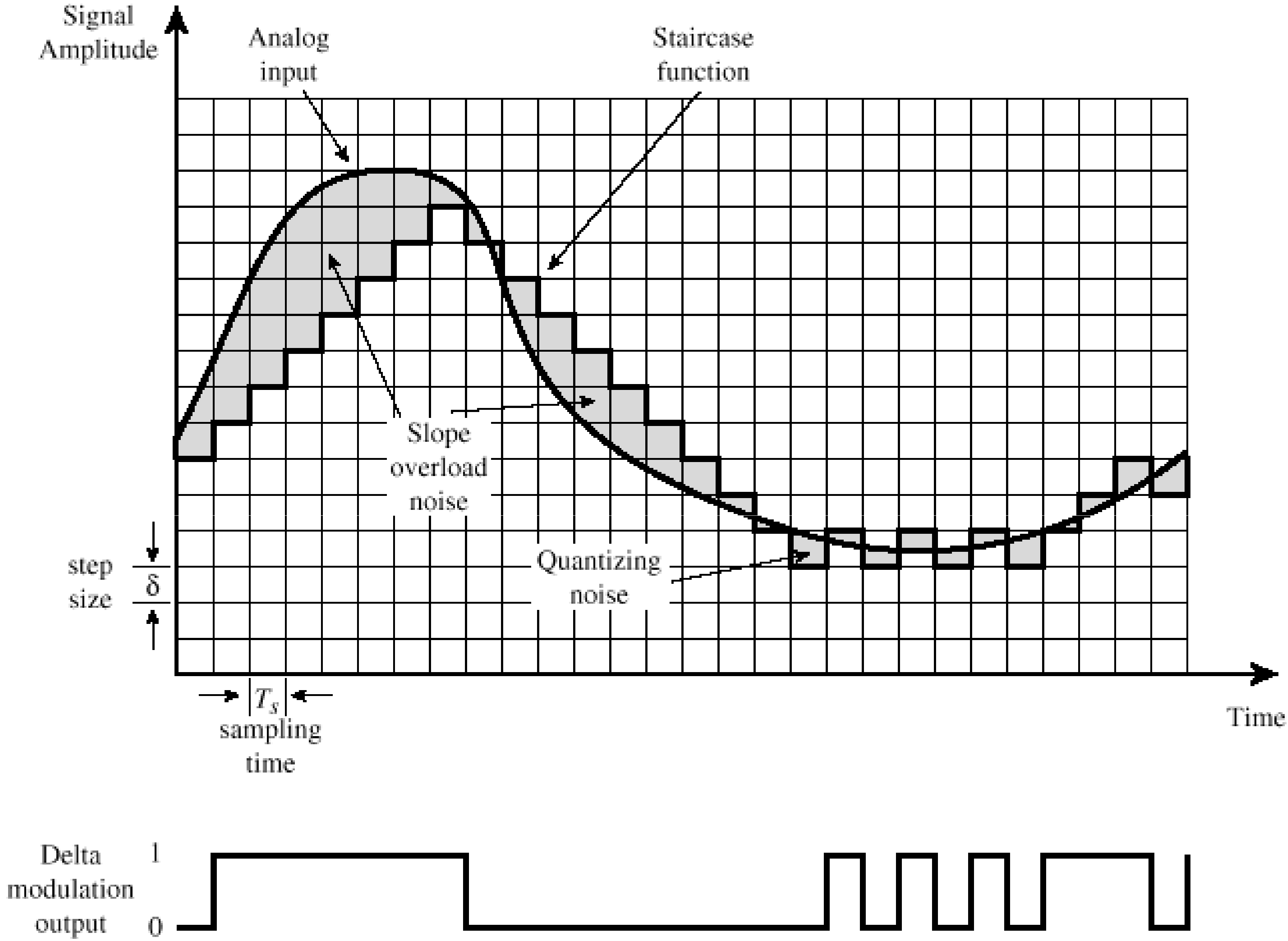


Figure 6.18 Example of Delta Modulation

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

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