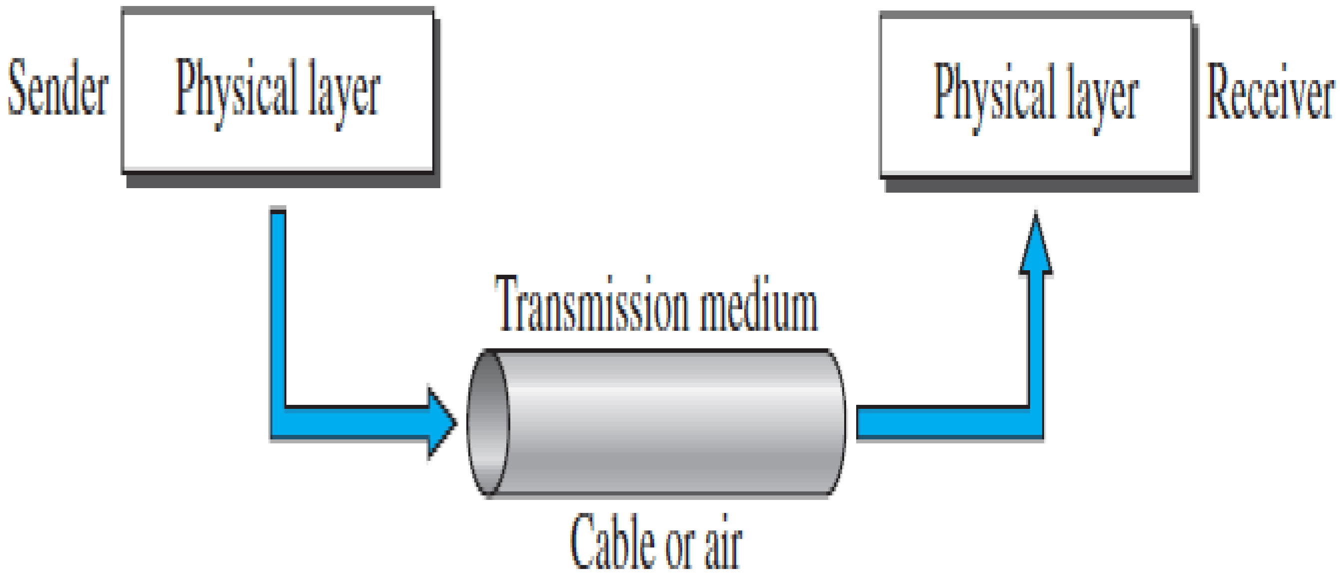
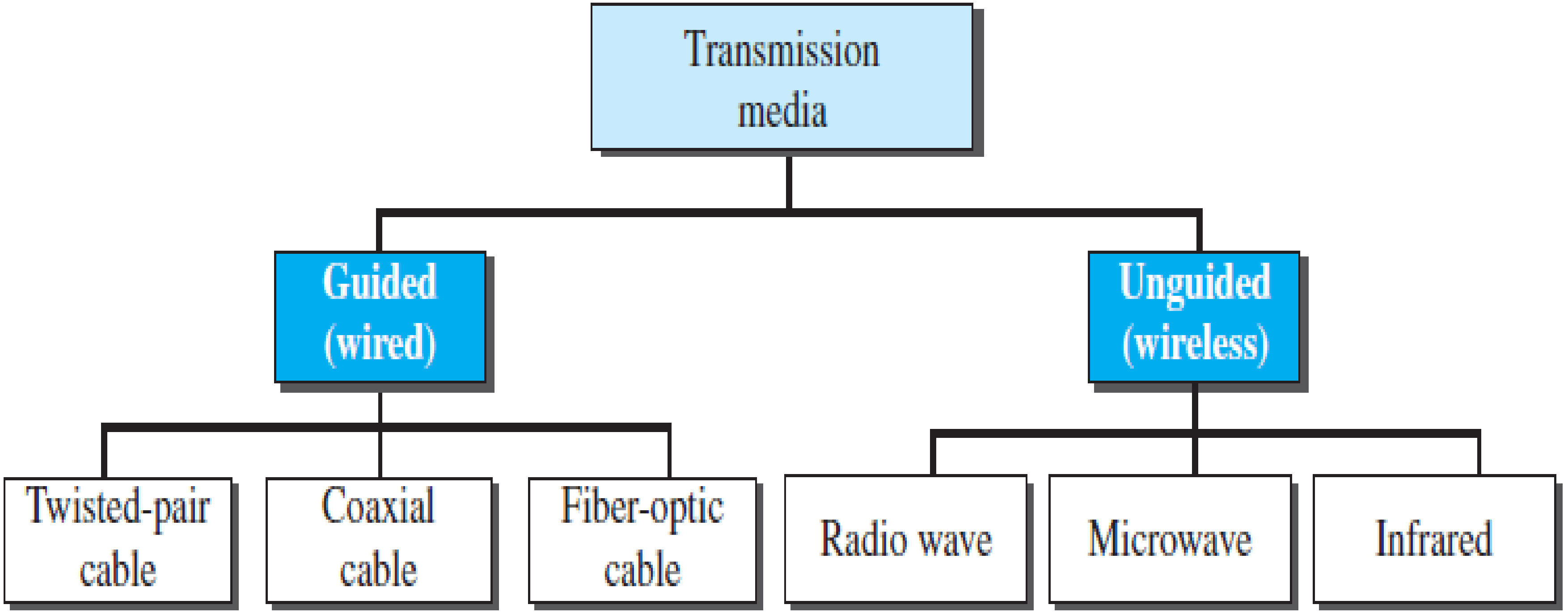
Transmission Media

- A transmission **medium** can be broadly defined as anything that can carry information from a source to a destination.

- For example, the transmission medium for two people having a dinner conversation is the air.


Sender
Physical layer
Physical layer
Receiver
Transmission medium
Cable or air

Classes of transmission media

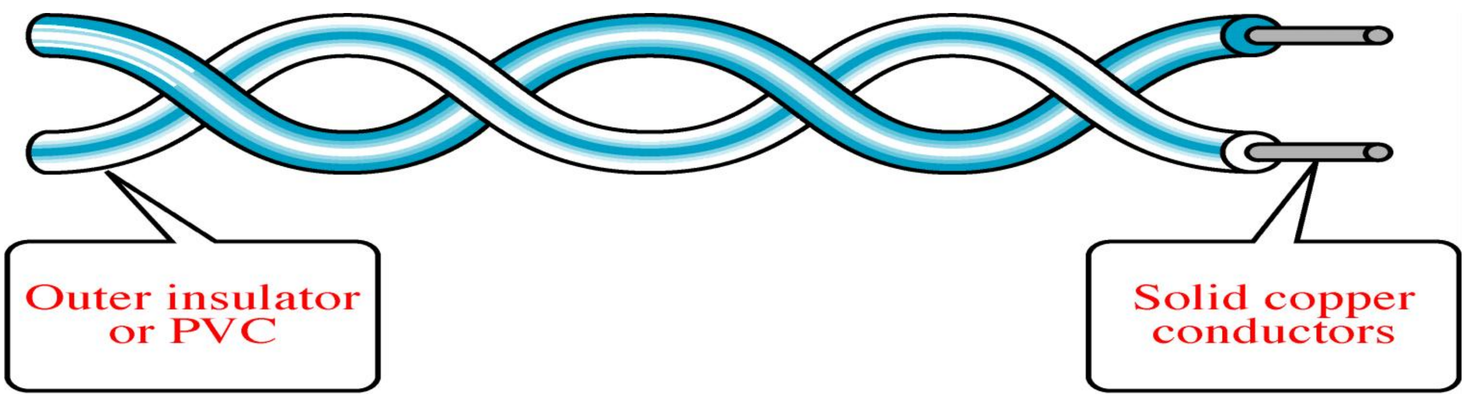


Guided media, which are those that provide a conduit from one device to another.

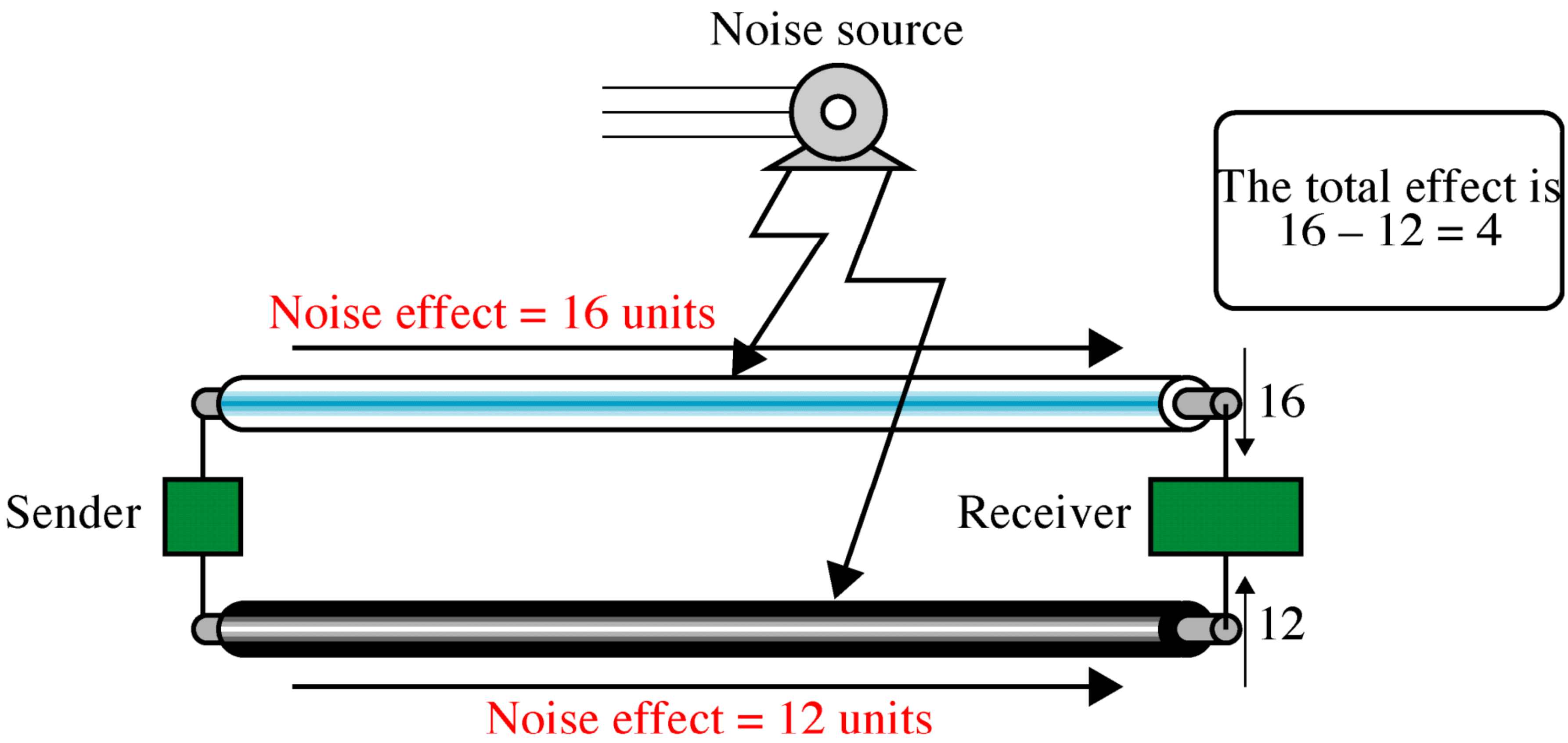
Unguided media transport electromagnetic waves without using a physical conductor.

Twisted-Pair Cable

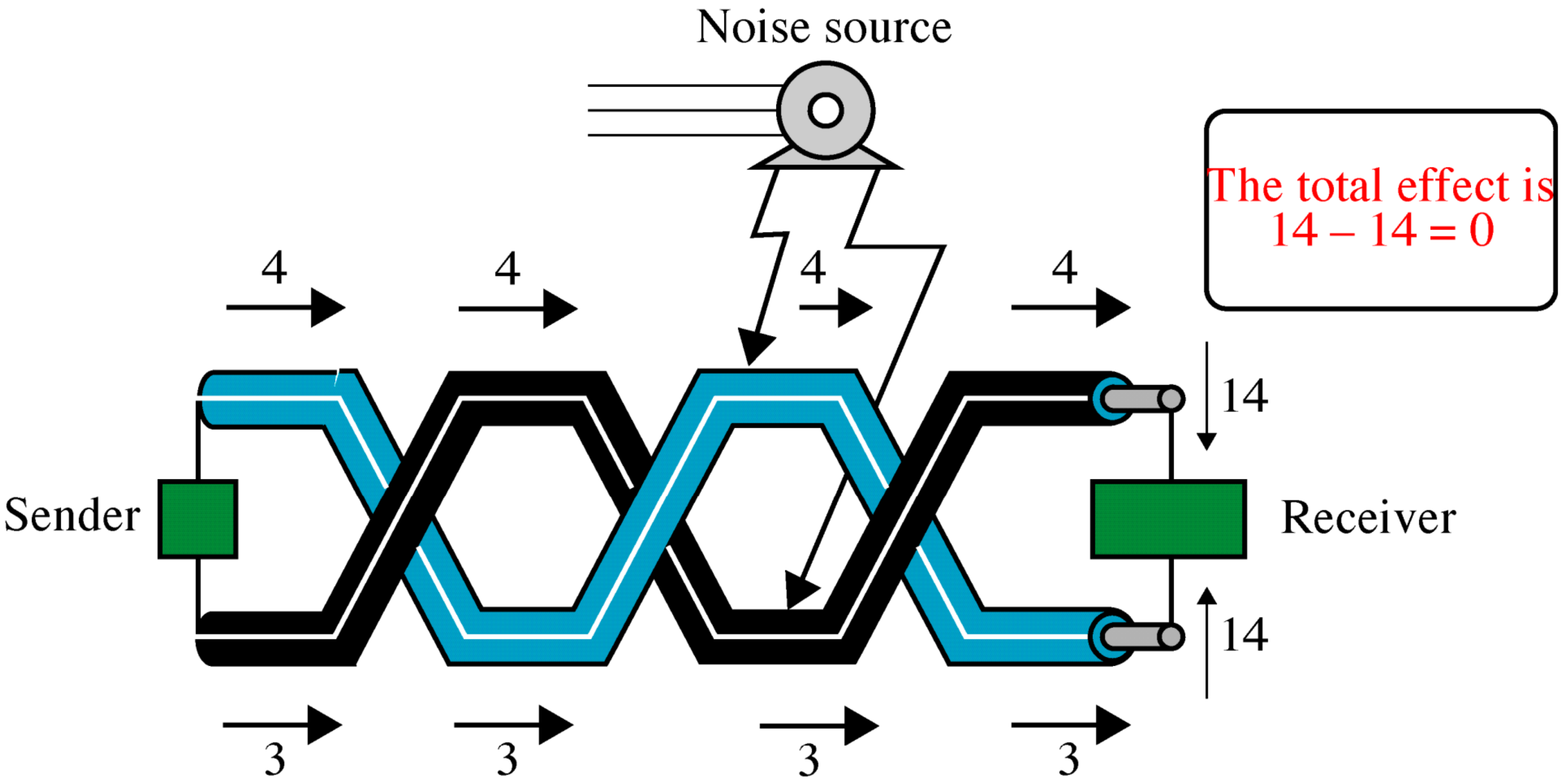
Twisted Pair and Coax use metallic(Copper) conductors that accept and transport the signals in the form of Electrical Current.



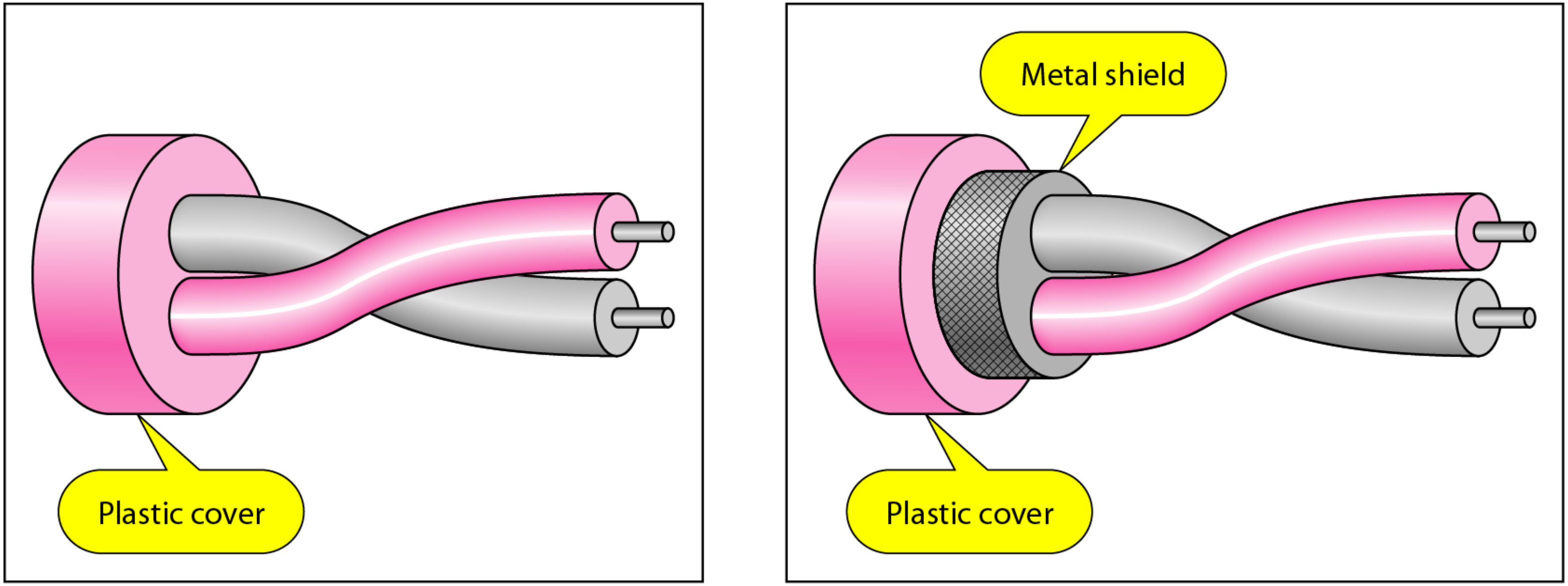
Effect of Noise on Parallel Lines



Noise on Twisted-Pair Lines



UTP and STP cables

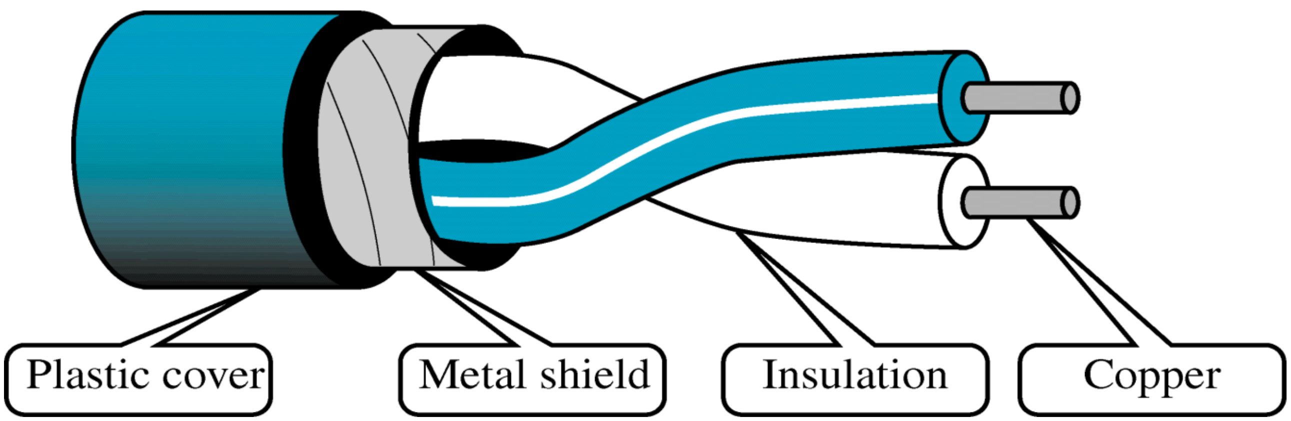


a. UTP

b. STP

Shielded Twisted-Pair Cable

- Metal casing prevents the penetration of electromagnetic noise.
- Eliminate the phenomenon , called CROSSTALK



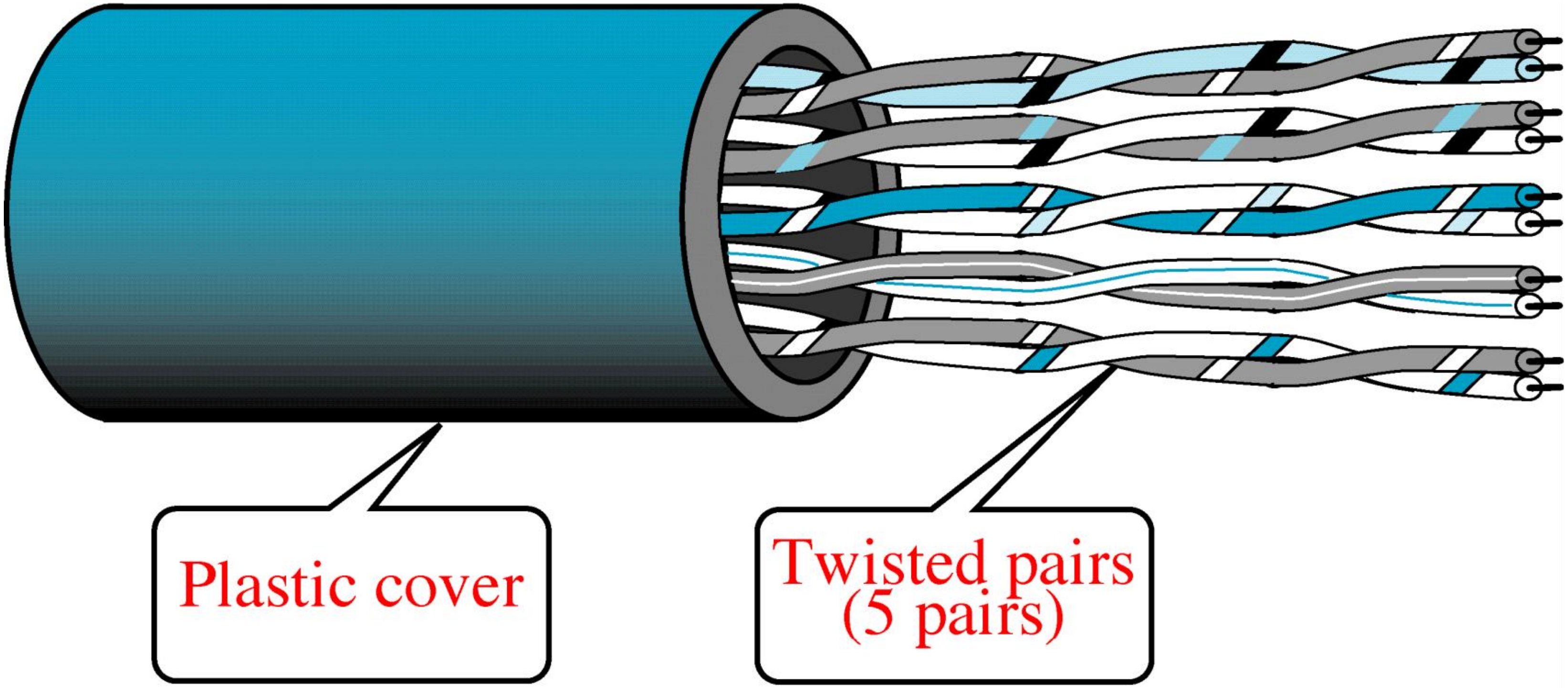
Advantages :

1. Cheaper
2. Less susceptible to electrical interference caused by nearby equipment or wires.
3. In turn are less likely to cause interference themselves.
4. Because it is electrically "cleaner", STP wire can carry data at a faster speed.

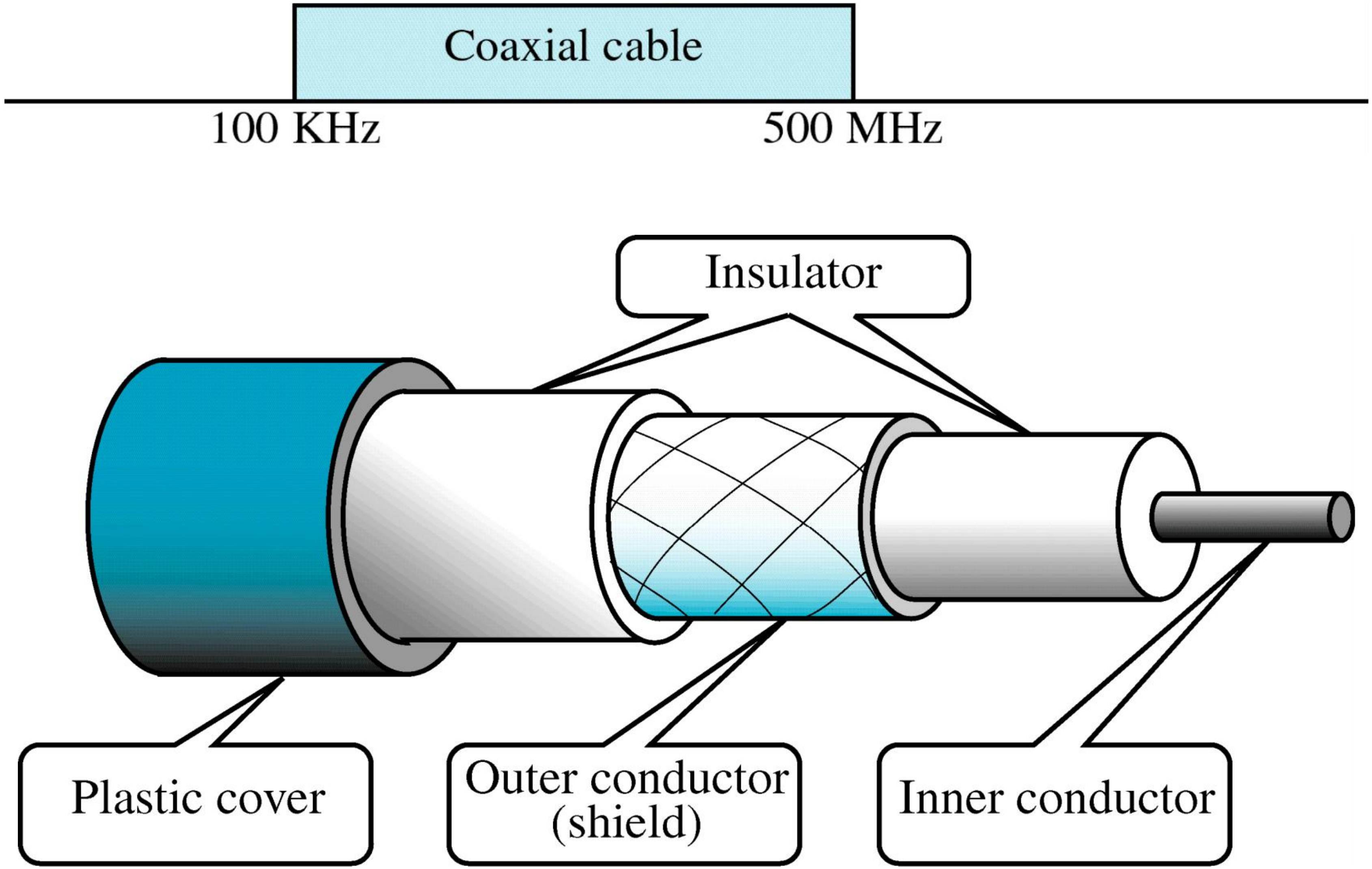
Disadvantages :

1. STP wire is that it is physically larger and more expensive than twisted pair wire.

Unshielded Twisted-Pair Cable



Coaxial Cable



Two kinds of coaxial cable

- ✓ One kind, 50-ohm cable, is commonly used when it is intended for digital transmission from the start.
- ✓ The other kind, 75-ohm cable, is commonly used for analog transmission and cable television.
  - ✓ Cable TV operators began to provide Internet access over cable, which has made 75-ohm cable more important for data communication.

- High bandwidth
- Excellent noise immunity.
- The bandwidth possible depends on the cable quality and length.
- Used within the telephone system, cable television and MAN
- For long-distance lines, but have now replaced by fiber optics on long distance routes.

Categories of coaxial cables

Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Optical Fiber Cable

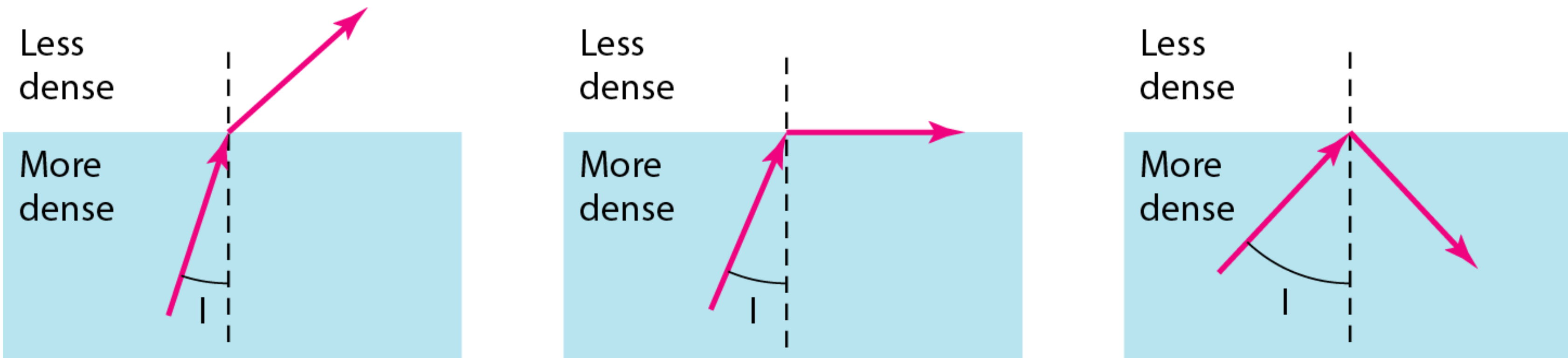
Optical Fiber is a glass or plastic cable that accept and transport the signals in the form of Light.

Advantages:

- Noise Resistance
- Less Signal Attenuation
- Higher BW

Disadvantages:

- Cost
- Installation/Maintenance
- Fragility(Broken Wire)

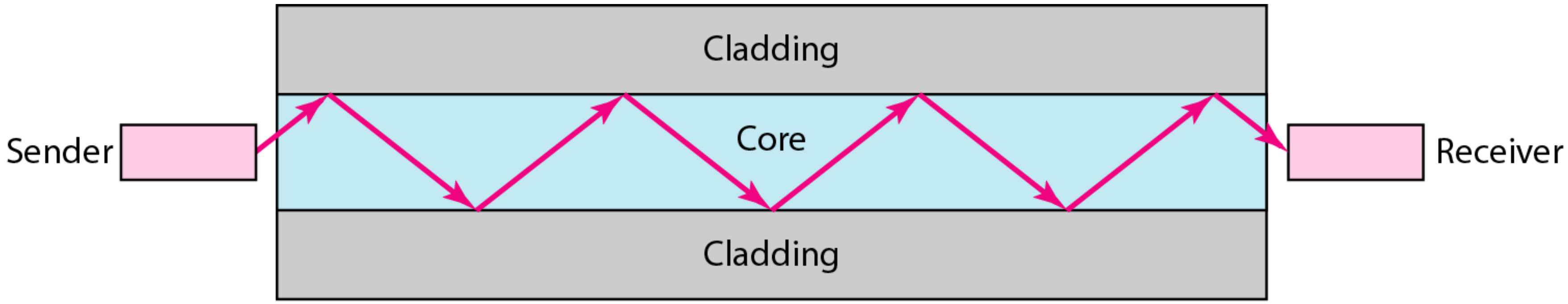


I < critical angle, refraction

I = critical angle, refraction

I > critical angle, reflection

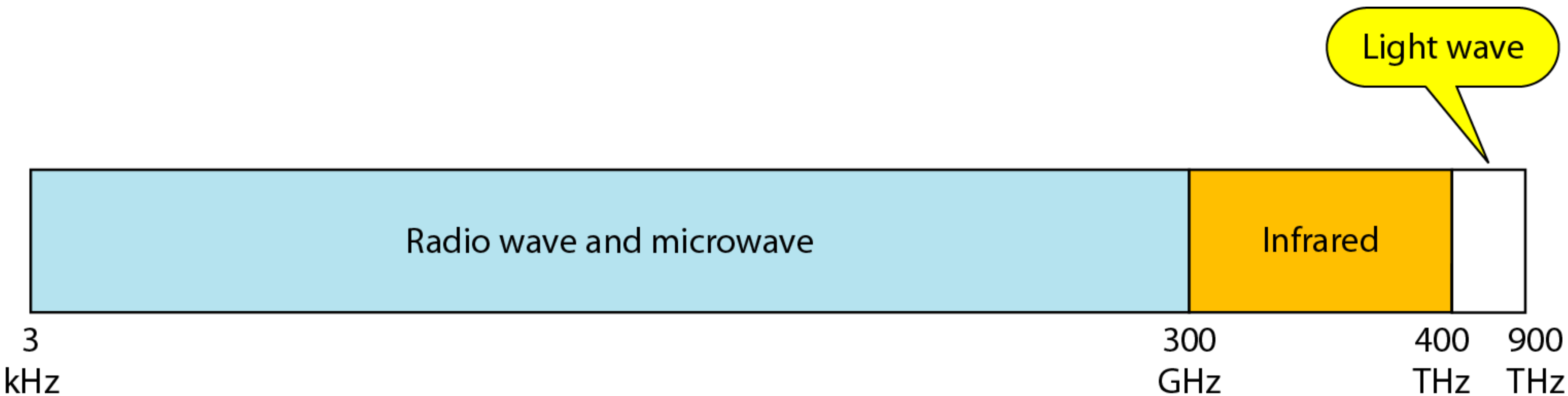
Optical fiber



UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication.

Electromagnetic spectrum for wireless communication



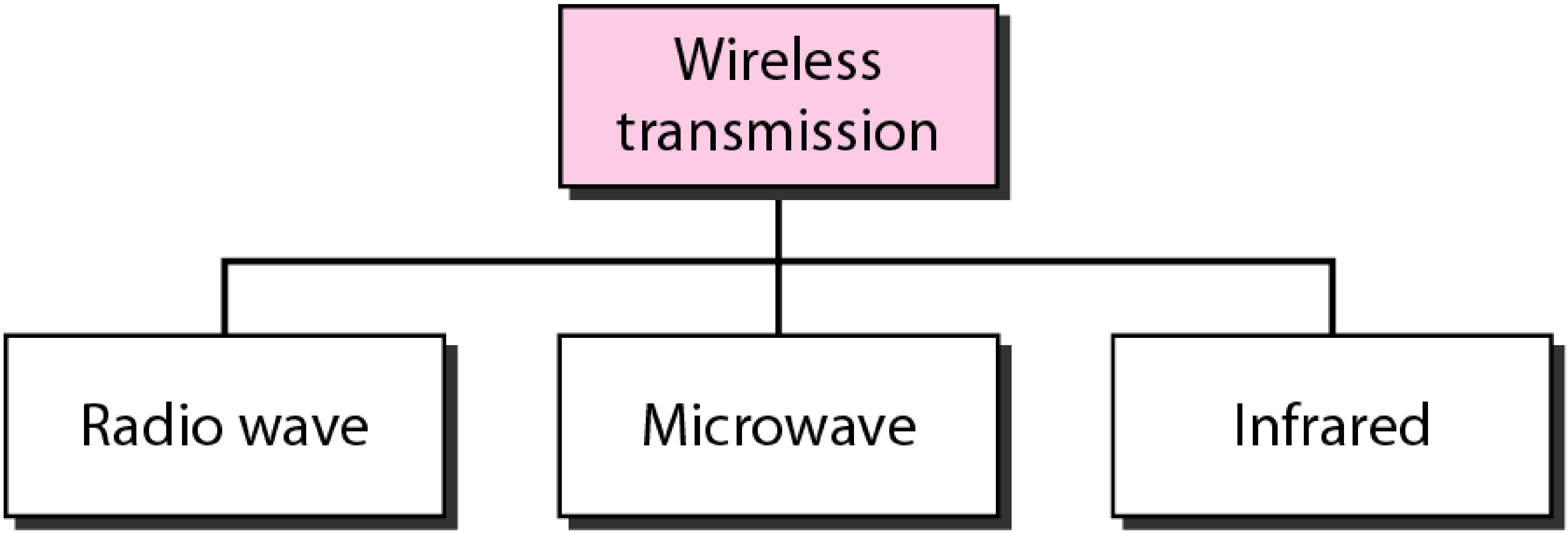
Propagation Methods

- Ground Propagation
- Sky Propagation
- Line-of-Sight Propagation

Bands

Band	*Range*	*Propagation*	*Application*
VLF (very low frequency)	3–30 kHz	Ground	Long-range radio navigation
LF (low frequency)	30–300 kHz	Ground	Radio beacons and navigational locators
MF (middle frequency)	300 kHz–3 MHz	Sky	AM radio
HF (high frequency)	3–30 MHz	Sky	Citizens band (CB), ship/aircraft communication
VHF (very high frequency)	30–300 MHz	Sky and line-of-sight	VHF TV, FM radio
UHF (ultrahigh frequency)	300 MHz–3 GHz	Line-of-sight	UHF TV, cellular phones, paging, satellite
SHF (superhigh frequency)	3–30 GHz	Line-of-sight	Satellite communication
EHF (extremely high frequency)	30–300 GHz	Line-of-sight	Radar, satellite

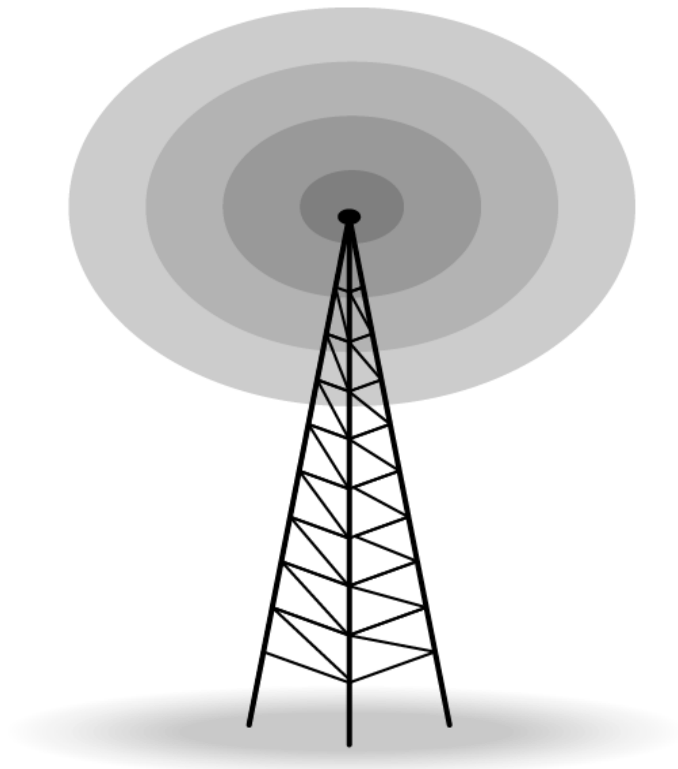
Wireless transmission waves



Note

Radio waves are used for multicast communications, such as radio and television.
They can penetrate through walls.
Use Omni directional antennas

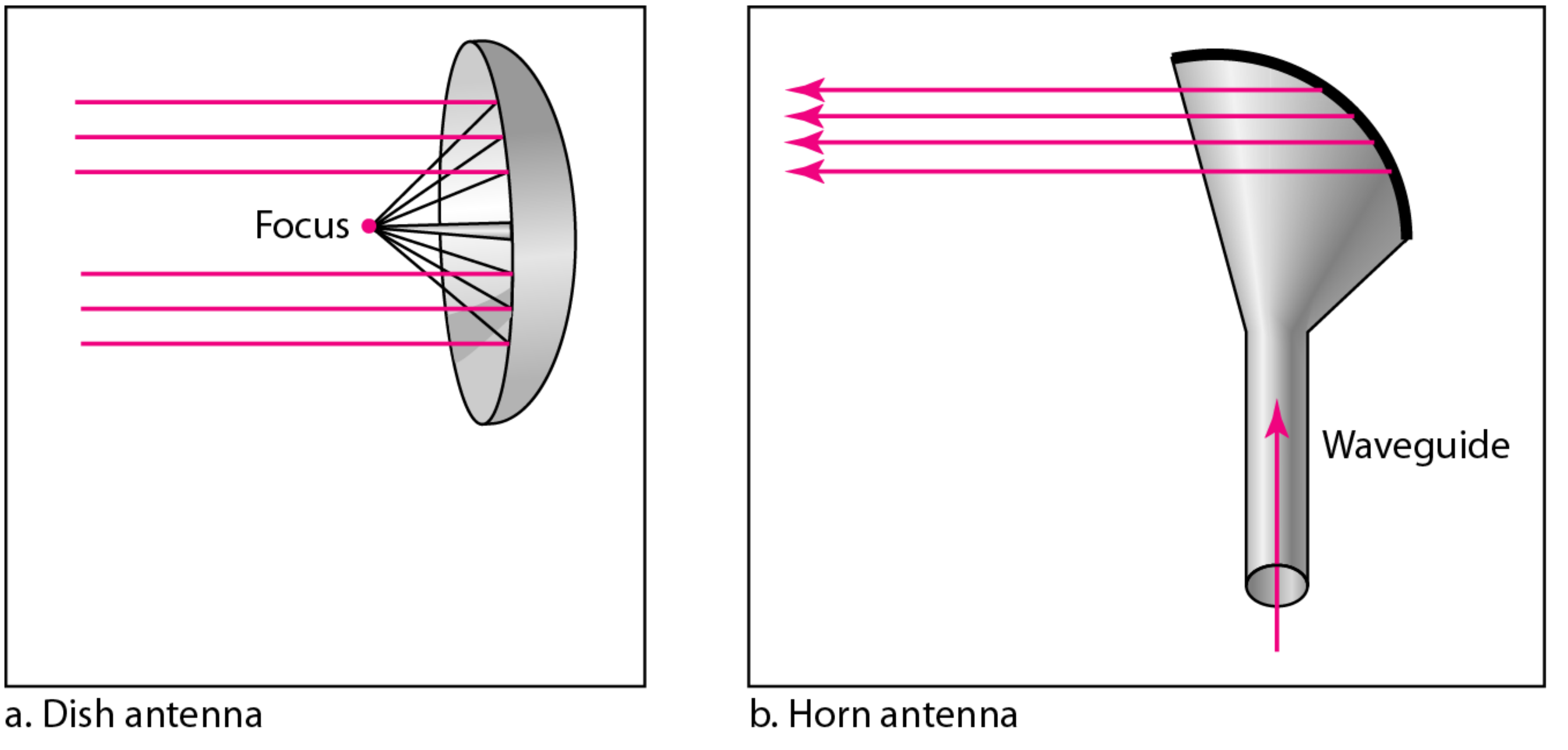
Omnidirectional antenna



Note

Microwaves are used for unicast communication such as cellular telephones and wireless LANs.
Higher frequency ranges cannot penetrate walls.
Use directional antennas - point to point line of sight communications.

Unidirectional antennas



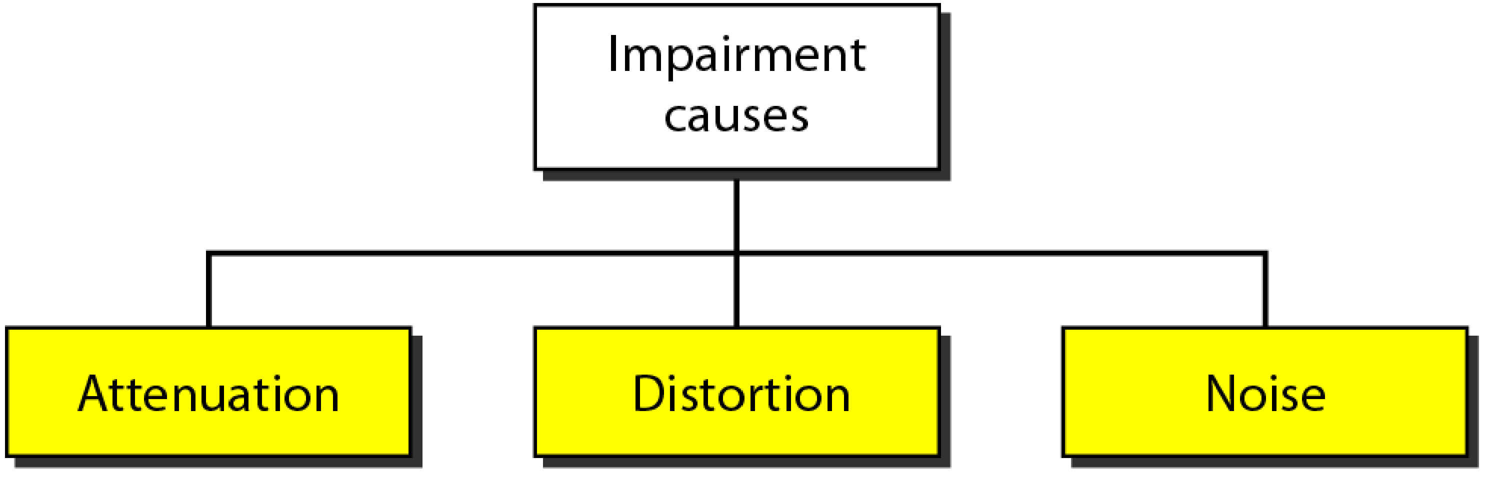
Note

Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

Transmission Impairment

- Signal transmit through medium that are not perfect.
- This imperfection cause signal impairment.
- What is sent is not received.

Causes of impairment



Attenuation

- Means loss of energy -> weaker signal
- When a signal travels through a medium it loses energy overcoming the resistance of the medium
- Amplifiers are used to compensate for this loss of energy by amplifying the signal.

Measurement of Attenuation

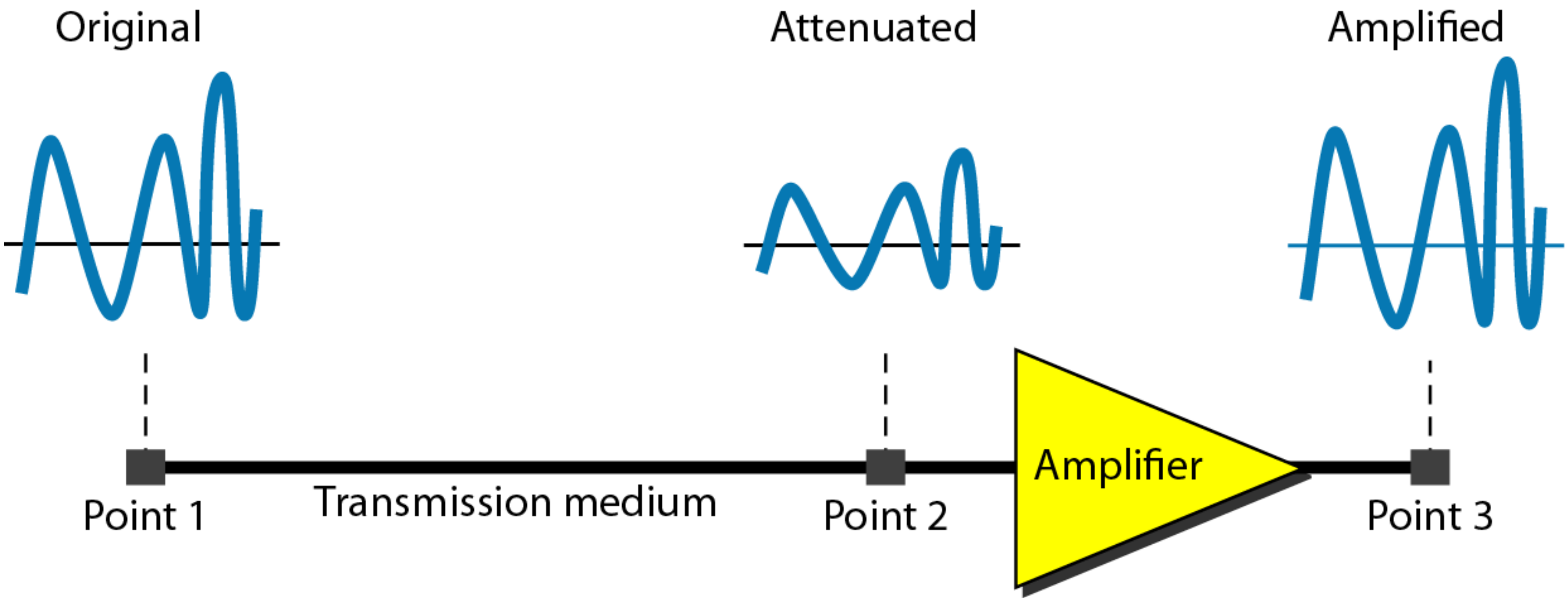
- To show the loss or gain of energy the unit "decibel" is used.

$$dB = 10\log_{10} P_2/P_1$$

P_1 - input signal

P_2 - output signal

Attenuation



- *Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that P_2 is $(1/2)P_1$. In this case, the attenuation (loss of power) can be calculated as*

Example 3.26

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5 P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

A loss of 3 dB (–3 dB) is equivalent to losing one-half the power.

Example 3.27

A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1}$$

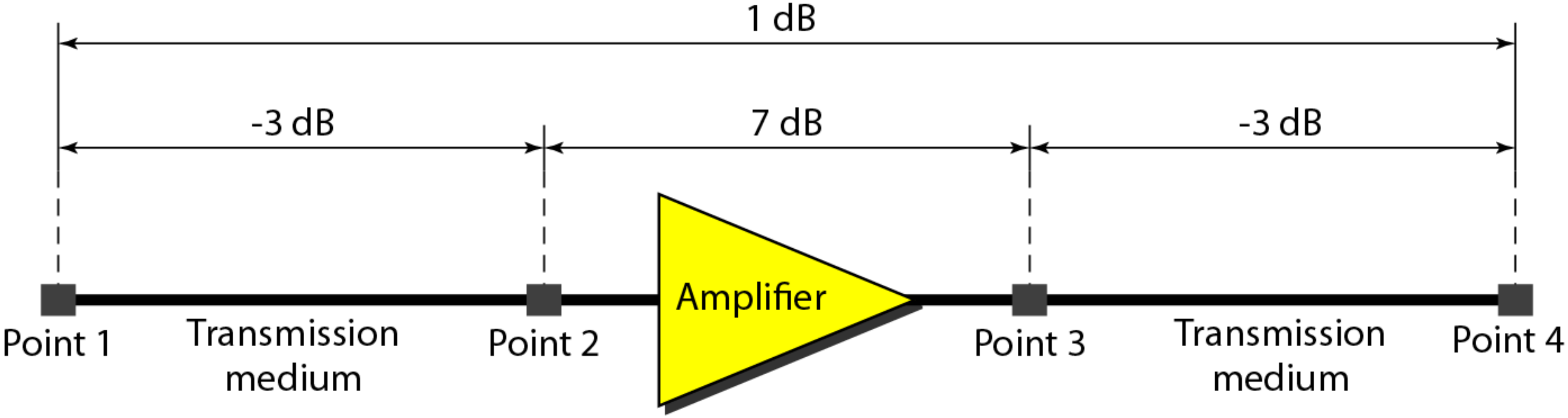
$$= 10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

Example 3.28

One reason that engineers use the decibel to measure the changes in the strength of a signal is that decibel numbers can be added (or subtracted) when we are measuring several points (cascading) instead of just two. In Figure 3.27 a signal travels from point 1 to point 4. In this case, the decibel value can be calculated as

$$\boxed{\mathrm{dB} = -3 + 7 - 3 = +1}$$

Decibels for Example 3.28



Example 3.29

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as dB_m and is calculated as $dB_m = 10 \log10 P_m$, where P_m is the power in milliwatts. Calculate the power of a signal with $dB_m = -30$.

Solution

We can calculate the power in the signal as

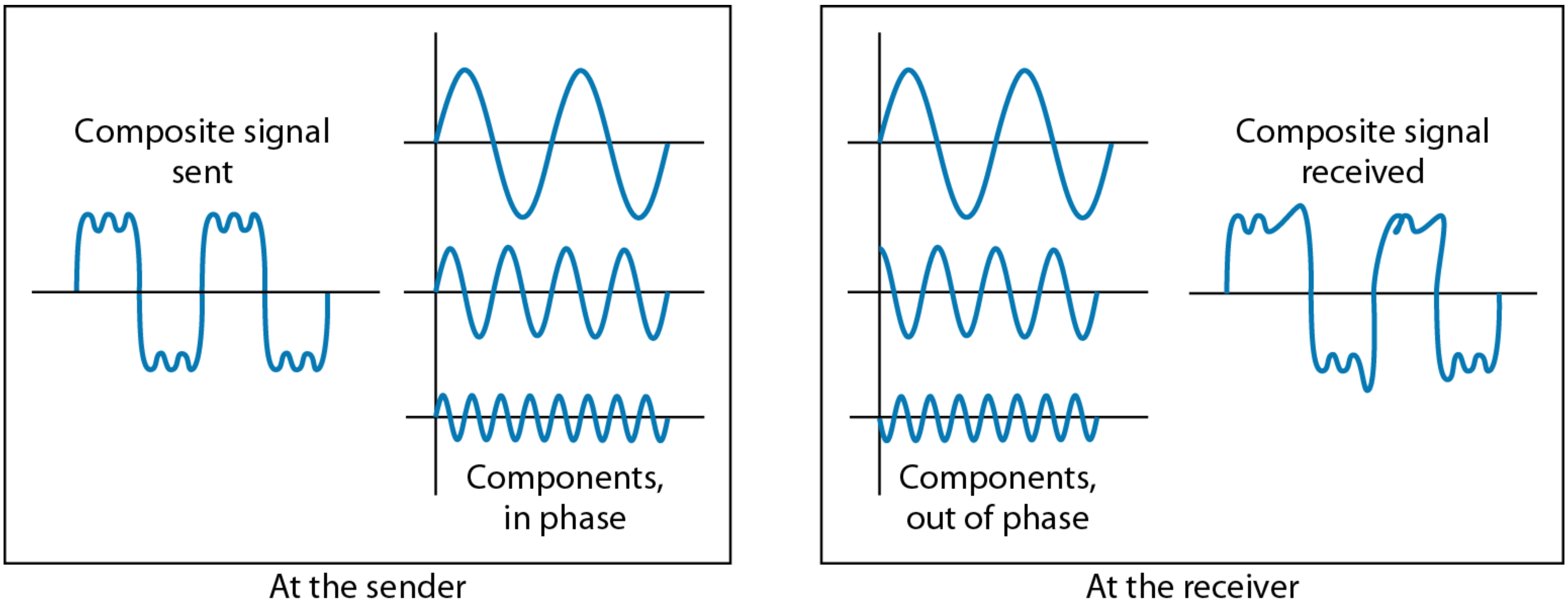
$$\mathrm{dB_m} = 10 \log_{10} P_m = -30$$

$$\log_{10} P_m = -3 \qquad P_m = 10^{-3}\ \mathrm{mW}$$

Distortion

- Means that the signal changes its form or shape
- Distortion occurs in composite signals
- Each frequency component has its own propagation speed traveling through a medium.
- The different components therefore arrive with different delays at the receiver.
- That means that the signals have different phases at the receiver than they did at the source.

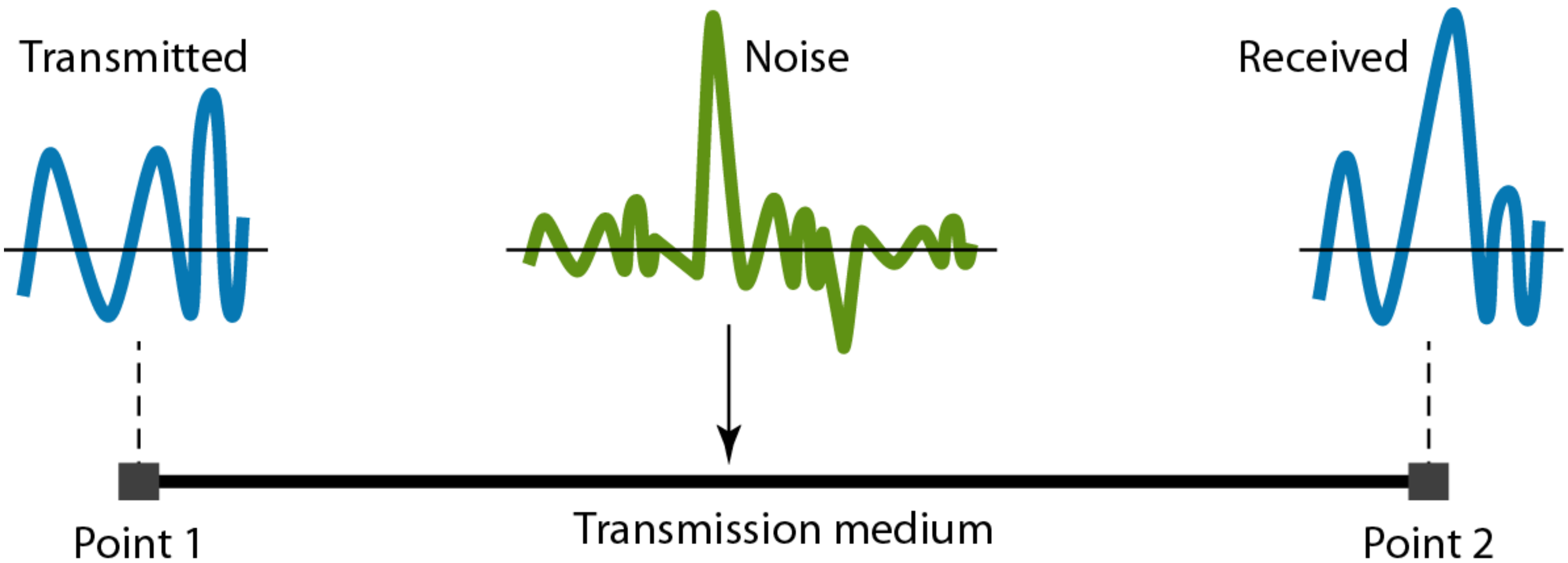
Distortion


Composite signal
sent
Components,
in phase
At the sender
Components,
out of phase
Composite signal
received
At the receiver

Noise

- There are different types of noise
  - Thermal - random noise of electrons in the wire creates an extra signal
  - Crosstalk - same as above but between two wires.
  - Impulse - Spikes that result from power lines, lightening, etc.
  - Induced

Noise


Transmitted
Noise
Received
Point 1
Transmission medium
Point 2

Signal to Noise Ratio (SNR)

- To measure the quality of a system the SNR is often used. It indicates the strength of the signal wrt the noise power in the system.
- It is the ratio between two powers.
- It is usually given in dB and referred to as $SNR_{dB.}$

Example 3.31

The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}?

Example 3.32

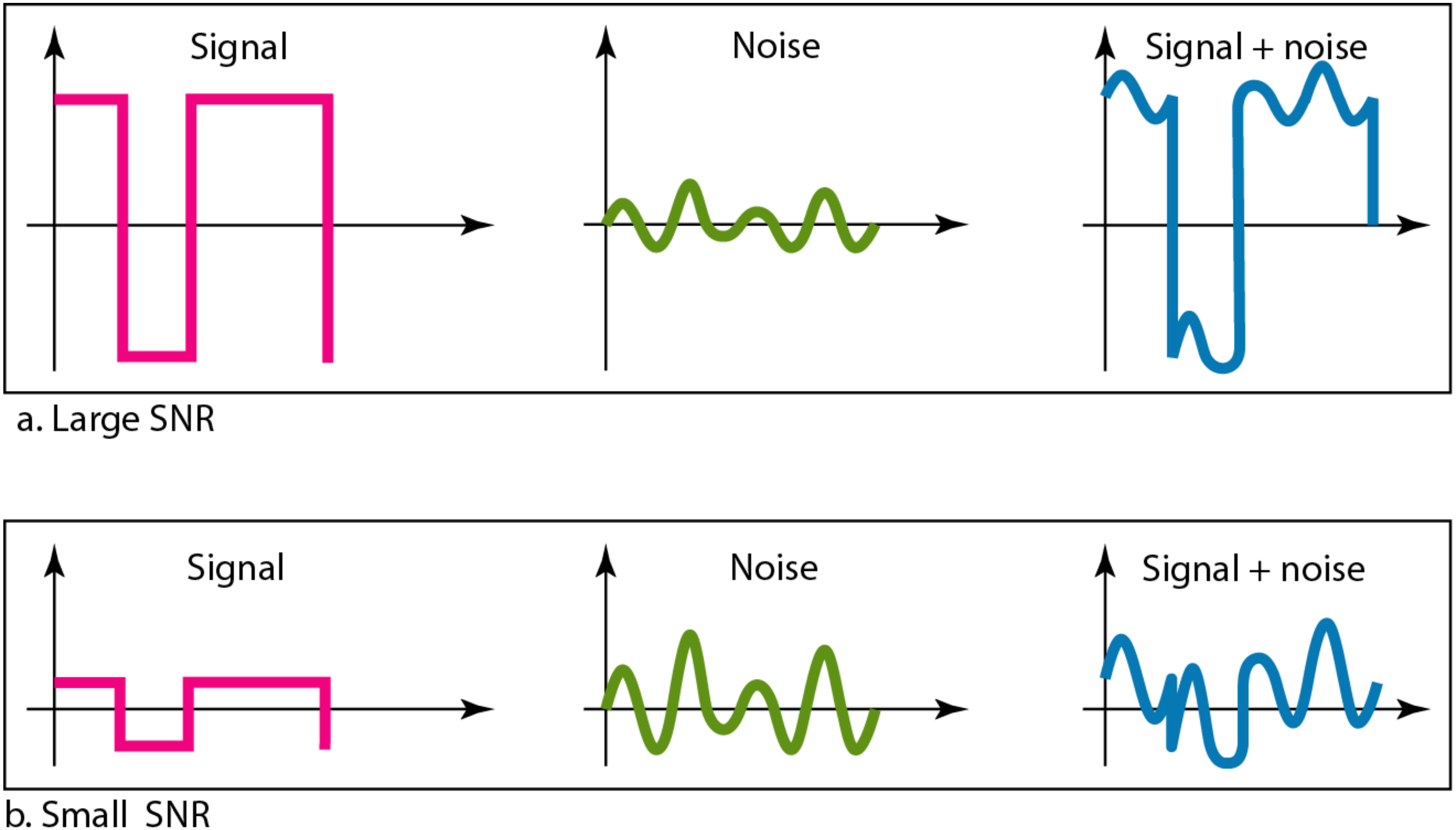
The values of SNR and SNR_{dB} for a noiseless channel are

$$\mathrm{SNR} = \frac{\text{signal power}}{0} = \infty$$

$$\mathrm{SNR_{dB}} = 10 \log_{10} \infty = \infty$$

We can never achieve this ratio in real life; it is an ideal.

Figure 3.30 ***Two cases of SNR: a high SNR and a low SNR***



DATA AND SIGNALS

To be transmitted, data must be transformed to electromagnetic signals.

ANALOG AND DIGITAL

Data can be analog or digital. The term analog data refers to information that is continuous; digital data refers to information that has discrete states. Analog data take on continuous values. Digital data take on discrete values.

Topics discussed in this section:

Analog and Digital Data
Analog and Digital Signals
Periodic and Nonperiodic Signals

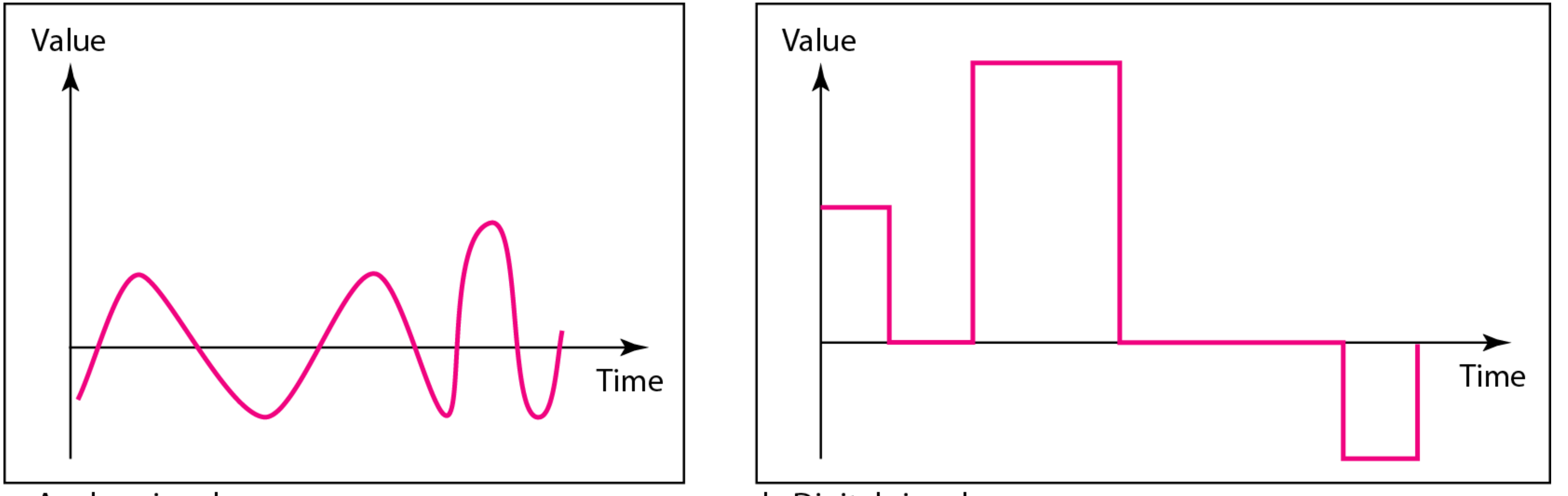
Note

Data can be analog or digital.
Analog data are continuous and take continuous values.
Digital data have discrete states and take discrete values.

Note

Signals can be analog or digital. Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values.

Comparison of analog and digital signals



a. Analog signal

b. Digital signal

Periodic and Non Periodic

- **A periodic signal** completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods.

- The completion of one full pattern is called as a **cycle.**

- A **non-periodic signal** changes without exhibiting a pattern or a cycle.

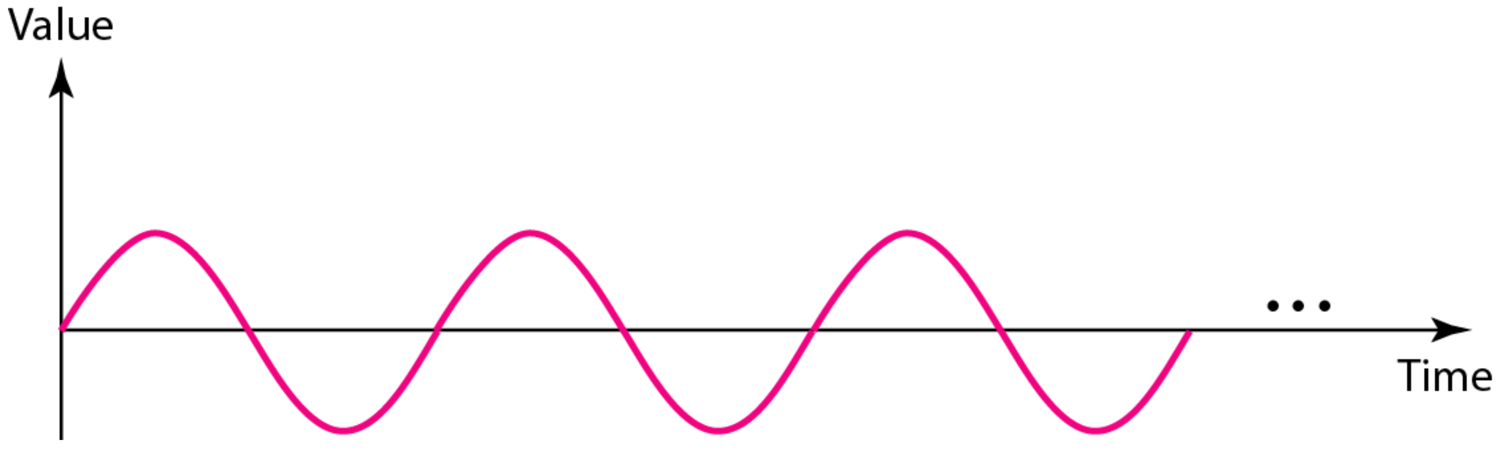
Note

In data communications, we commonly use periodic analog signals and nonperiodic digital signals.

PERIODIC ANALOG SIGNALS

Periodic analog signals can be classified as simple or composite.

Figure ***A sine wave***



A sine wave is represented by:

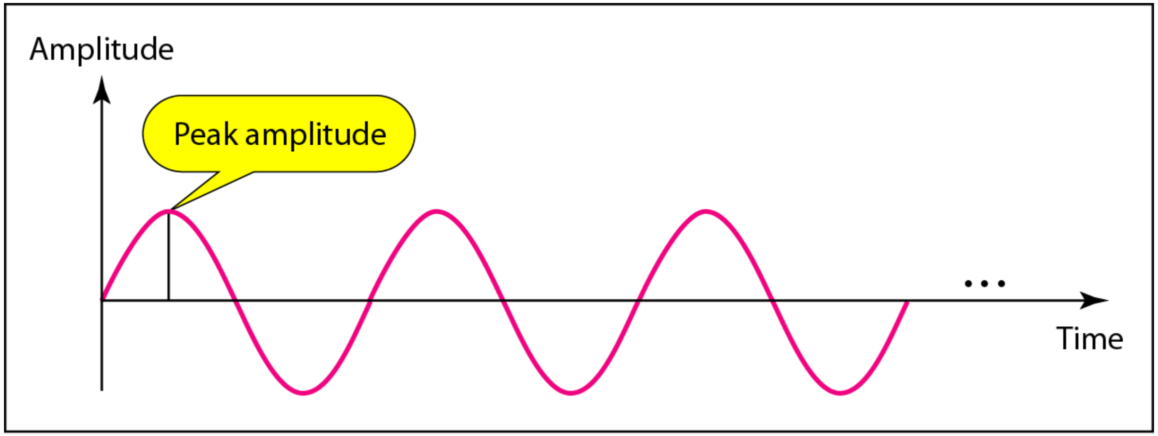
- Peak Amplitude
- Frequency
- Phase
- Wavelength

Peak Amplitude

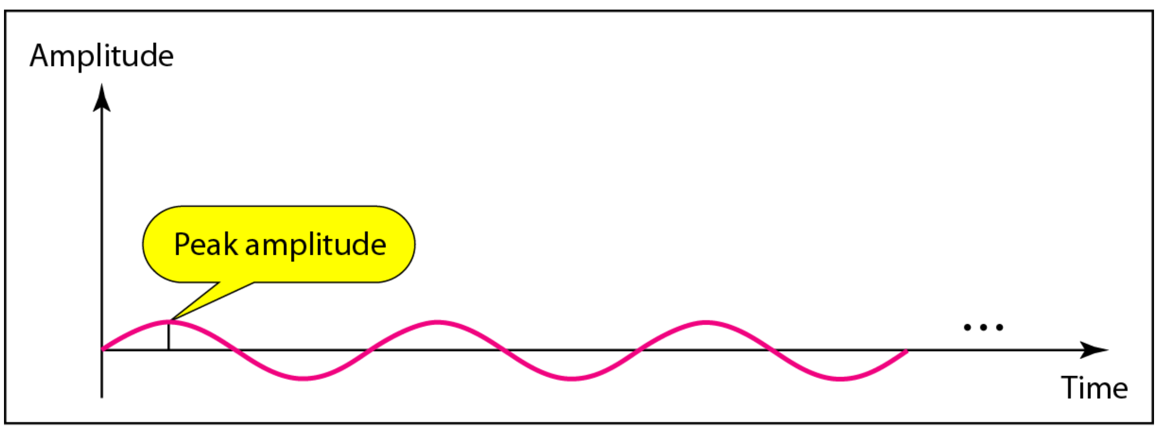
- The peak amplitude of a signal is the absolute value of its highest intensity, proportional to energy it carries.

- Measured in volts.

Two signals with the same phase and frequency, but different amplitudes



a. A signal with high peak amplitude



b. A signal with low peak amplitude

Example

The voltage of a battery is a constant; this constant value can be considered a sine wave, as we will see later. For example, the peak value of an AA battery is normally 1.5 V.

Period and Frequency

- Period refers to amount of time, in seconds, a signal takes to complete one cycle.
- Frequency refers to the number of periods in 1 second.
- Period is expressed in seconds and frequency is expressed in hertz (Hz)

Note

Frequency and period are the inverse of each other.

$$f = \frac{1}{T} \quad \text{and} \quad T = \frac{1}{f}$$

Note

Frequency is the rate of change with respect to time.

Change in a short span of time means high frequency.

Change over a long span of time means low frequency.

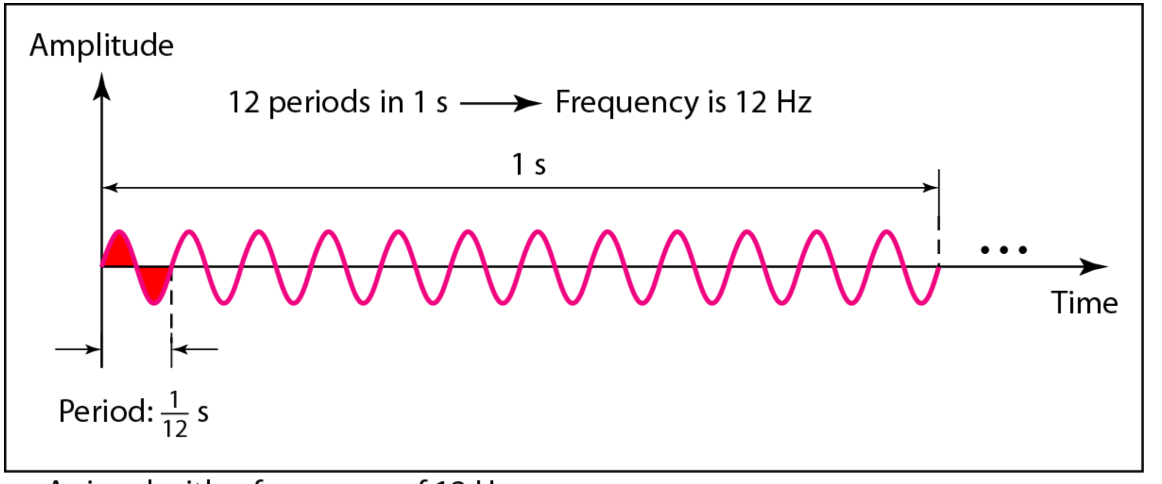
Note

If a signal does not change at all, its frequency is zero.
If a signal changes instantaneously, its frequency is infinite.

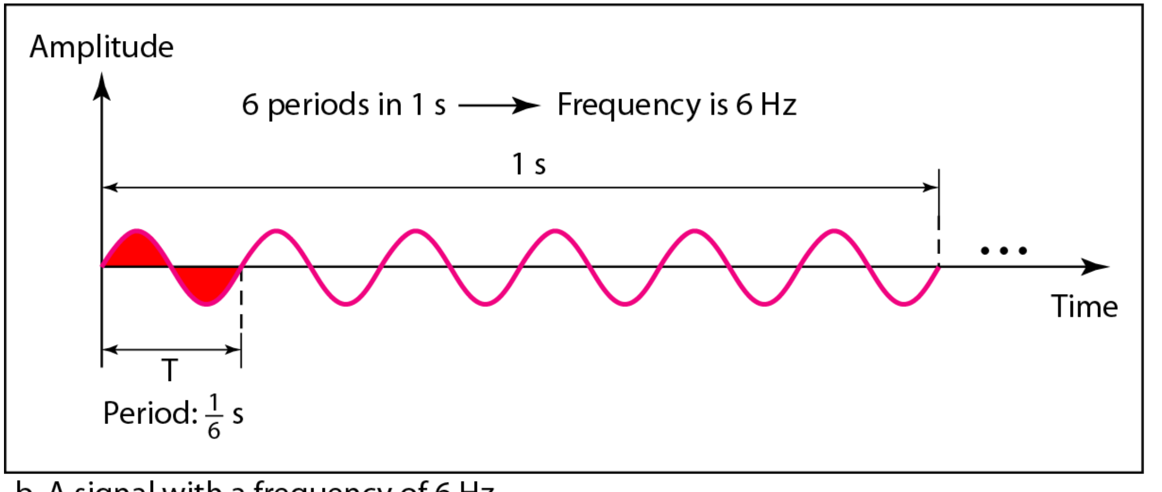
Table ***Units of period and frequency***

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

Figure ***Two signals with the same amplitude and phase, but different frequencies***



a. A signal with a frequency of 12 Hz



b. A signal with a frequency of 6 Hz

Example

The power we use at home has a frequency of 60 Hz. Determined the period of this sine wave?

Example

Express a period of 100 ms in microseconds.

Example

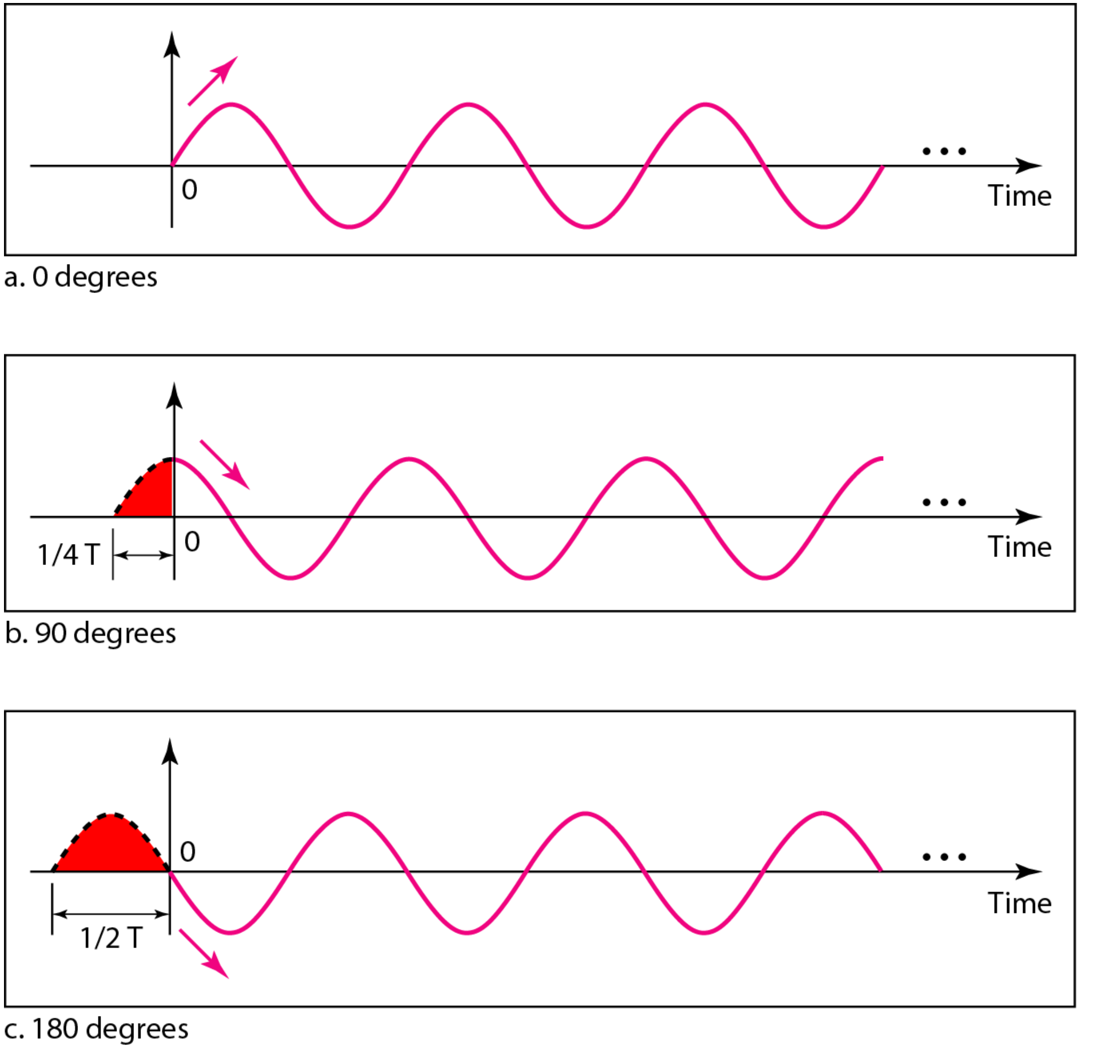
The period of a signal is 100 ms. What is its frequency in kilohertz?

Solution

Note

Phase describes the position of the waveform relative to time 0.

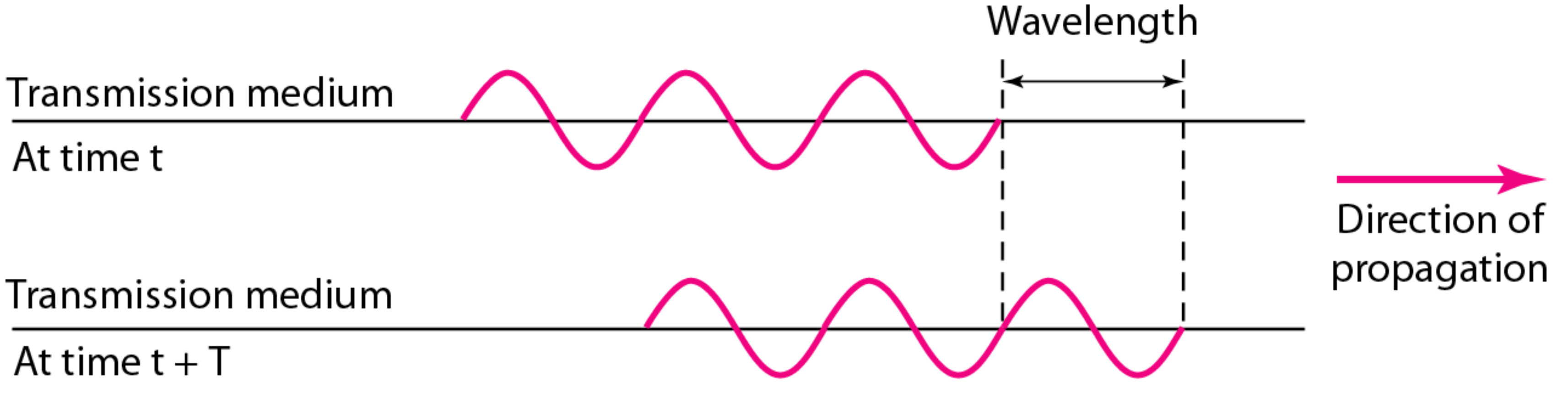
Figure *Three sine waves with the same amplitude and frequency, but different phases*



Example

A sine wave with value 1/6 cycle with respect to time 0. What is its phase in degrees and radians?

Figure *Wavelength and period*



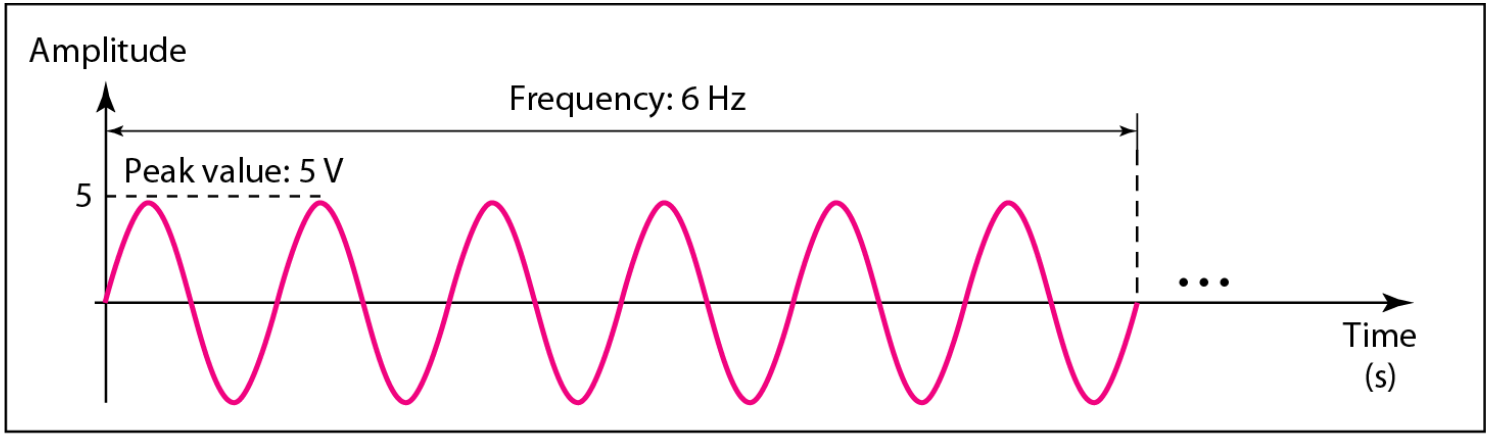
Wavelength

- Wavelength is the distance a signal can travel in one period.
- Wavelength= propagation speed* period

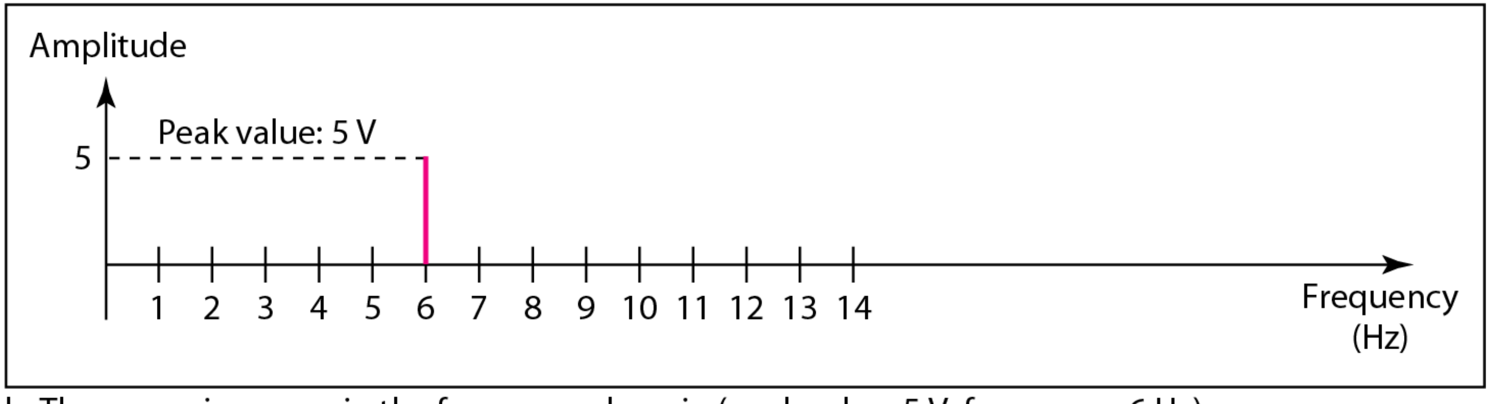
Or

- Wavelength= propagation speed/ frequency

Figure *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)

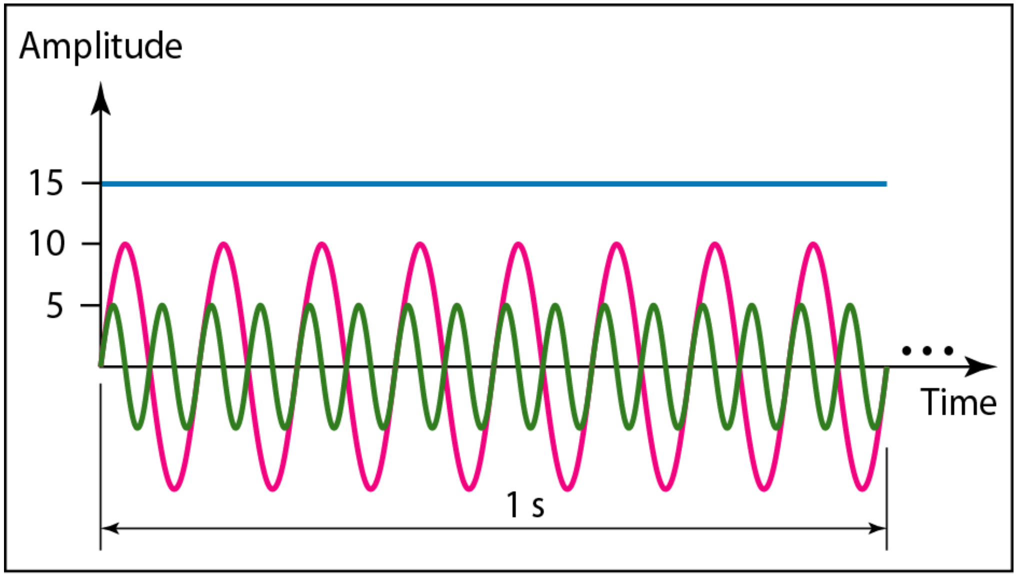
Note

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

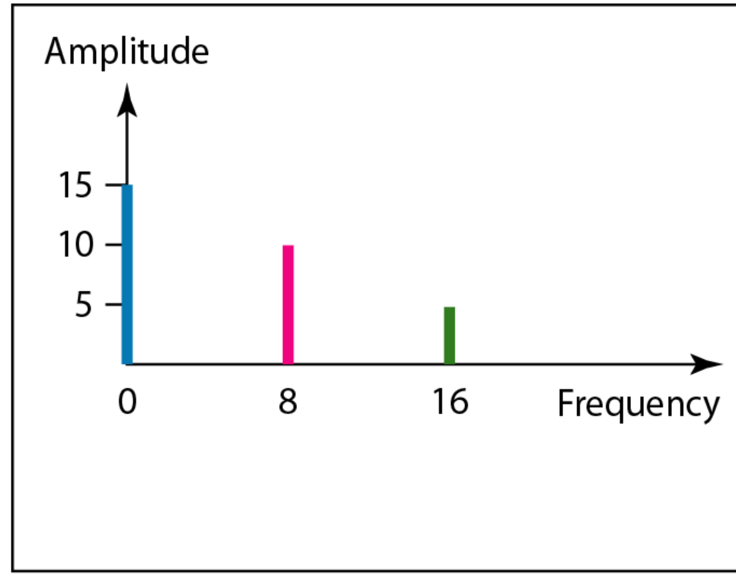
Example

The frequency domain is more compact and useful when we are dealing with more than one sine wave. For example, Figure 3.8 shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.

Figure *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

Note

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.

Note

According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases.

Note

**If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies;
if the composite signal is nonperiodic, the decomposition gives a combination of sine waves with continuous frequencies.**

Example

Figure 3.9 shows a periodic composite signal with frequency f. This type of signal is not typical of those found in data communications. We can consider it to be three alarm systems, each with a different frequency. The analysis of this signal can give us a good understanding of how to decompose signals.

Figure *A composite periodic signal*

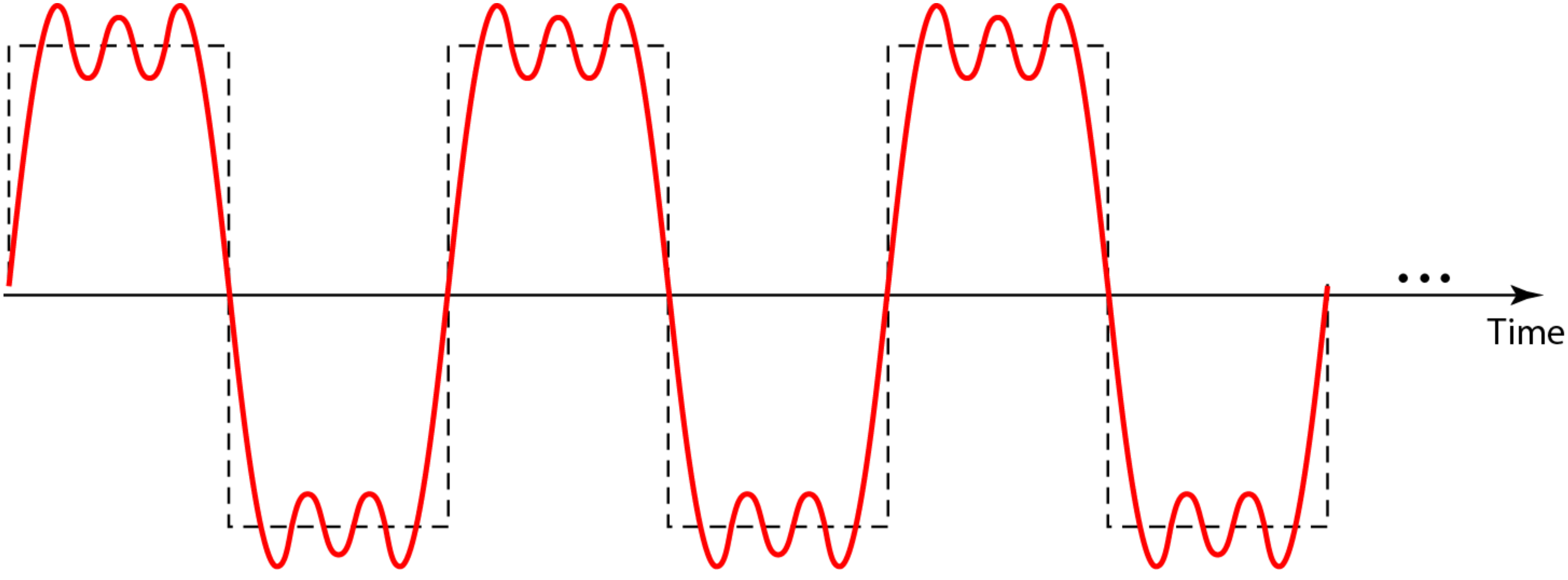
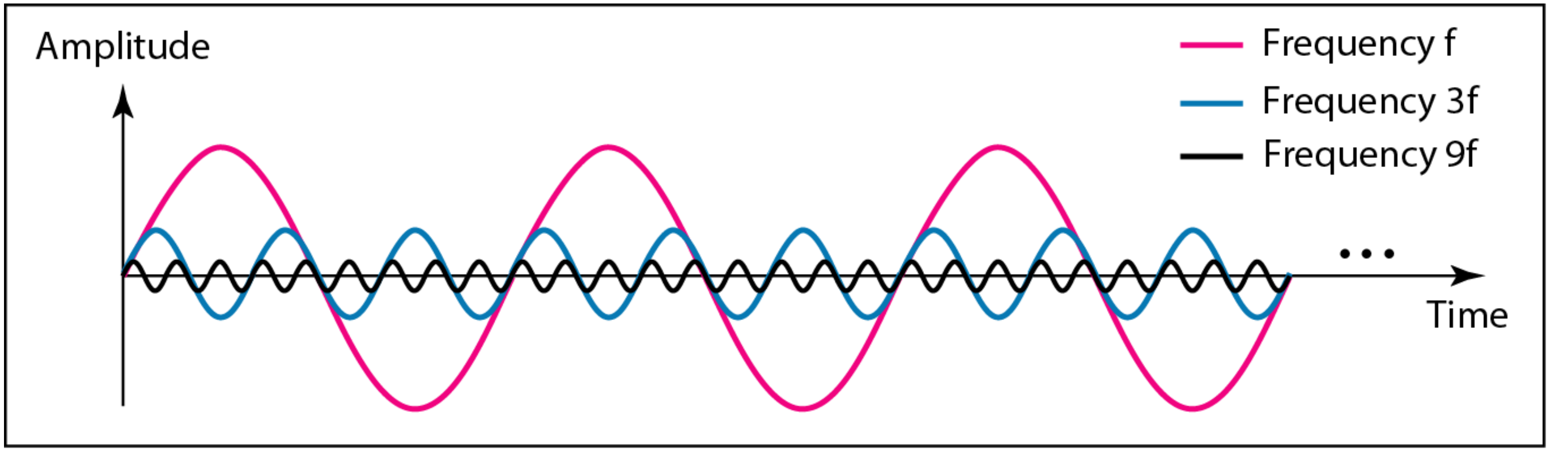
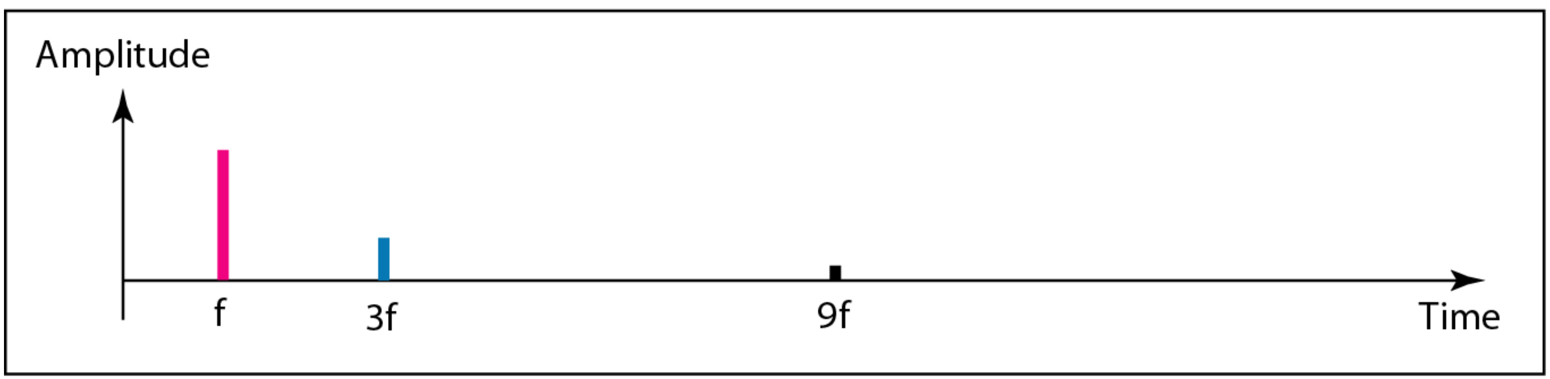


Figure ***Decomposition of a composite periodic signal in the time and frequency domains***



a. Time-domain decomposition of a composite signal

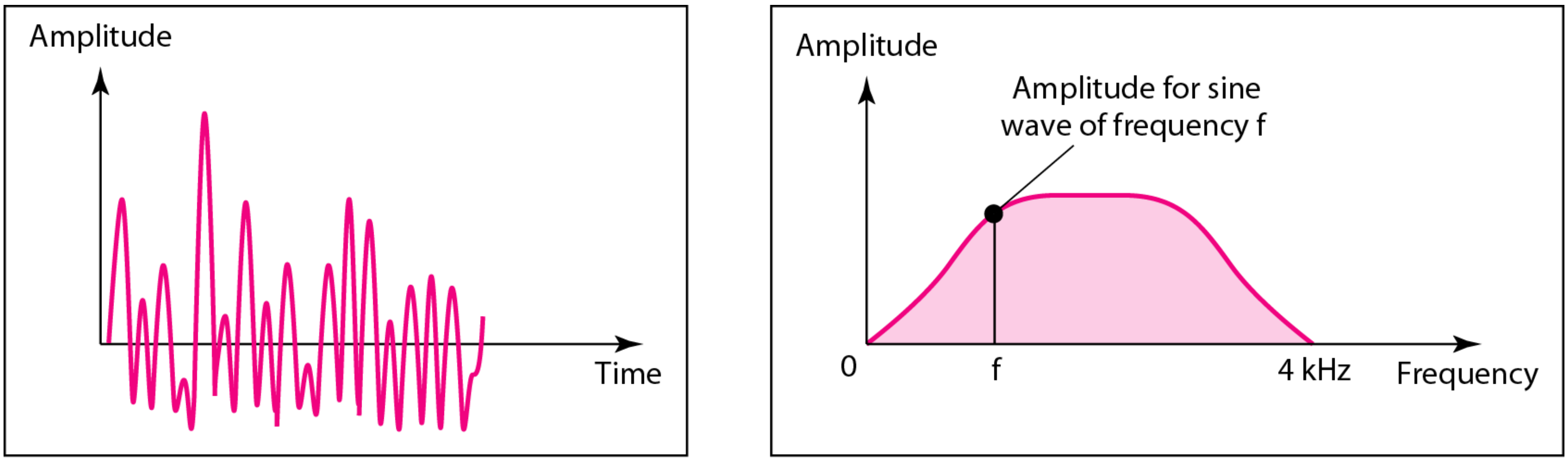


b. Frequency-domain decomposition of the composite signal

Example

Figure 3.11 shows a nonperiodic composite signal. It can be the signal created by a microphone or a telephone set when a word or two is pronounced. In this case, the composite signal cannot be periodic, because that implies that we are repeating the same word or words with exactly the same tone.

Figure *The time and frequency domains of a nonperiodic signal*



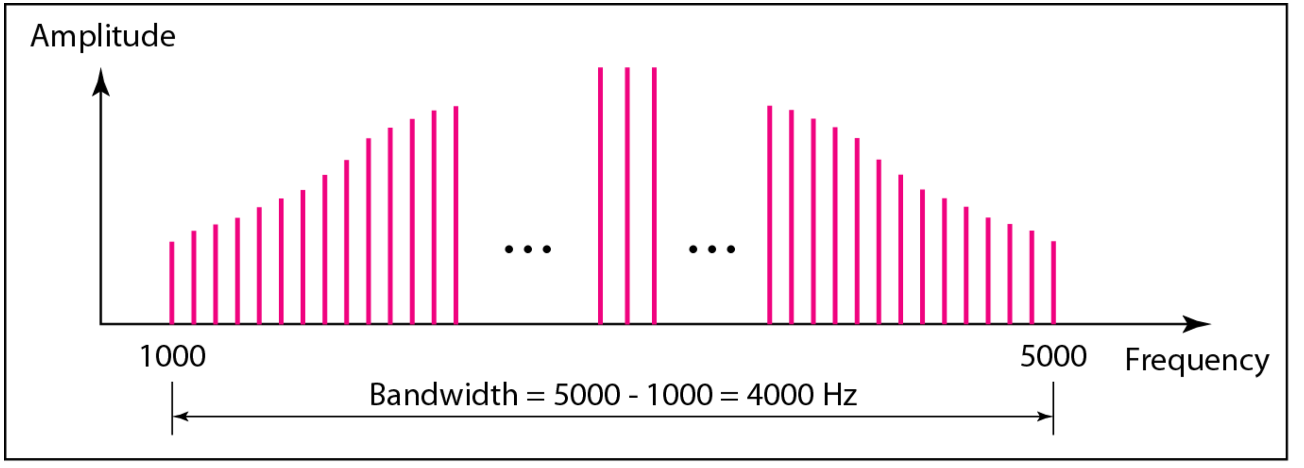
a. Time domain

b. Frequency domain

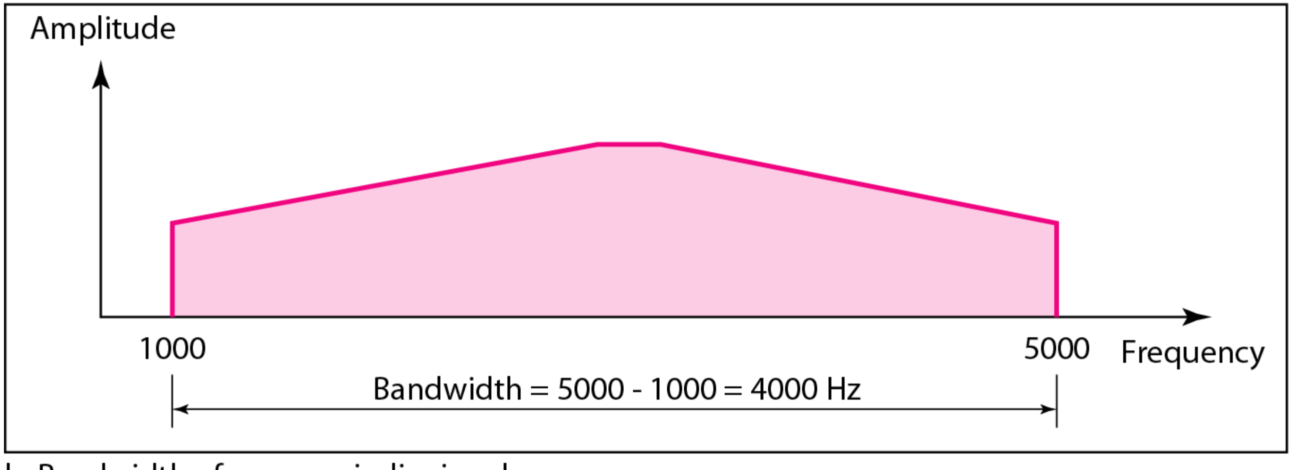
Note

The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

Figure *The bandwidth of periodic and nonperiodic composite signals*



a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

Example

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth? Draw the spectrum, assuming all components have a maximum amplitude of 10 V.

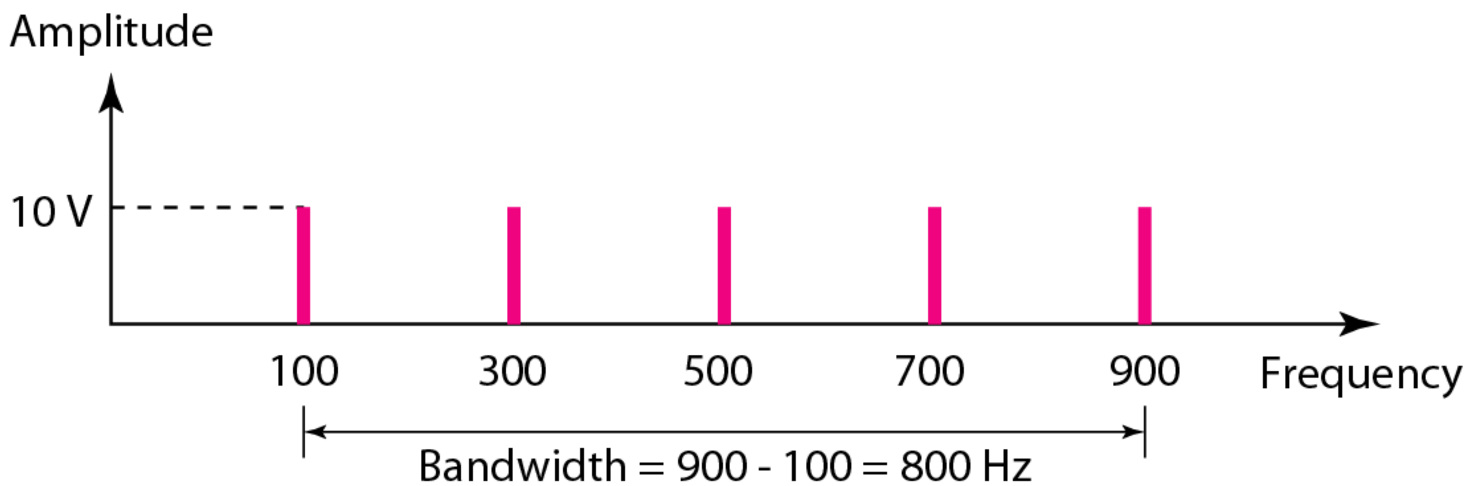
Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$\boxed{B = f_h - f_l = 900 - 100 = 800 \text{ Hz}}$$

The spectrum has only five spikes, at 100, 300, 500, 700, and 900 Hz (see Figure 3.13).

Figure *The bandwidth for Example 3.10*



Example

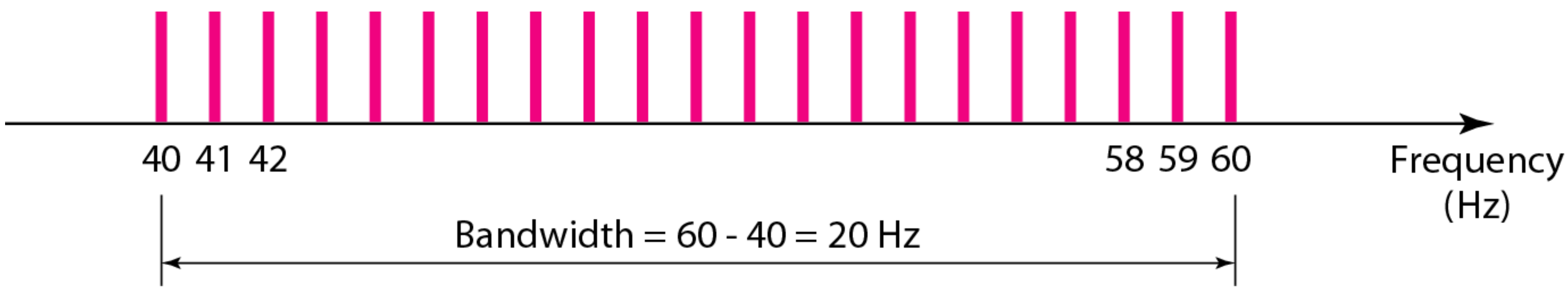
A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

The spectrum contains all integer frequencies. We show this by a series of spikes (see Figure 3.14).

Figure ***The bandwidth for Example 3.11***



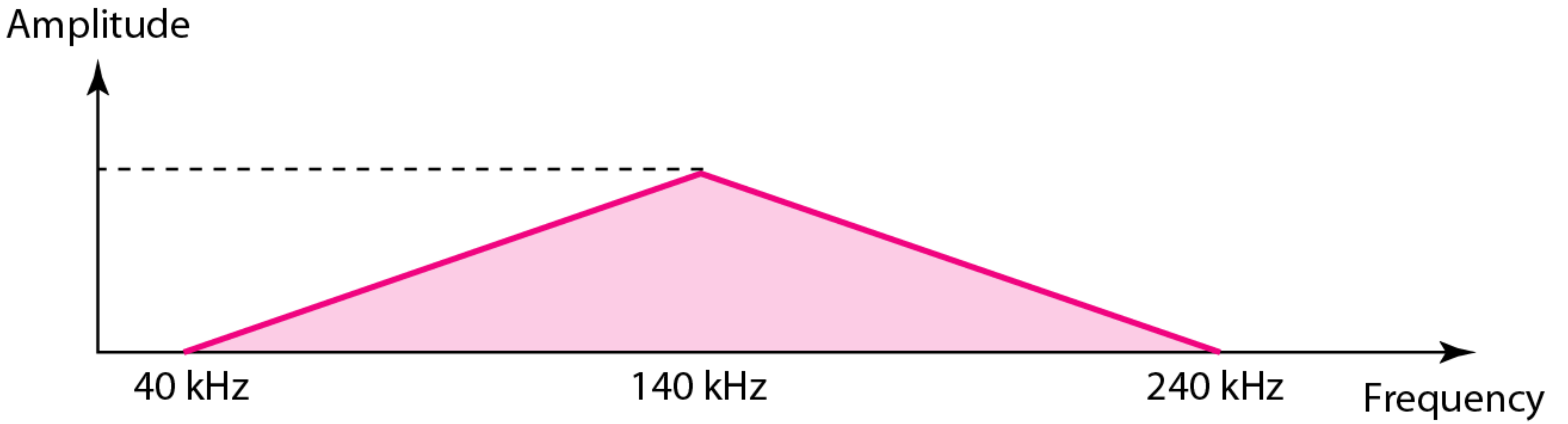
Example

A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal.

Solution

The lowest frequency must be at 40 kHz and the highest at 240 kHz. Figure 3.15 shows the frequency domain and the bandwidth.

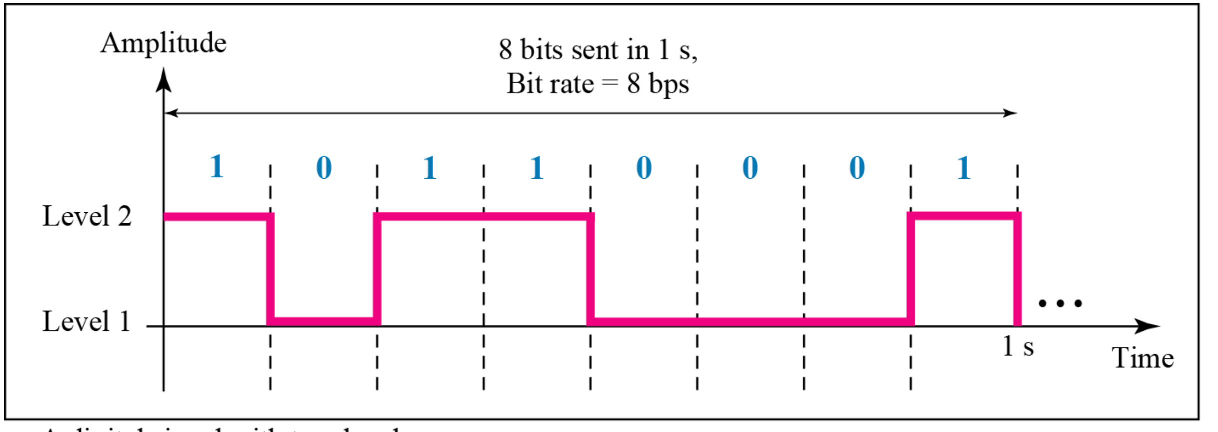
Figure *The bandwidth for Example 3.12*



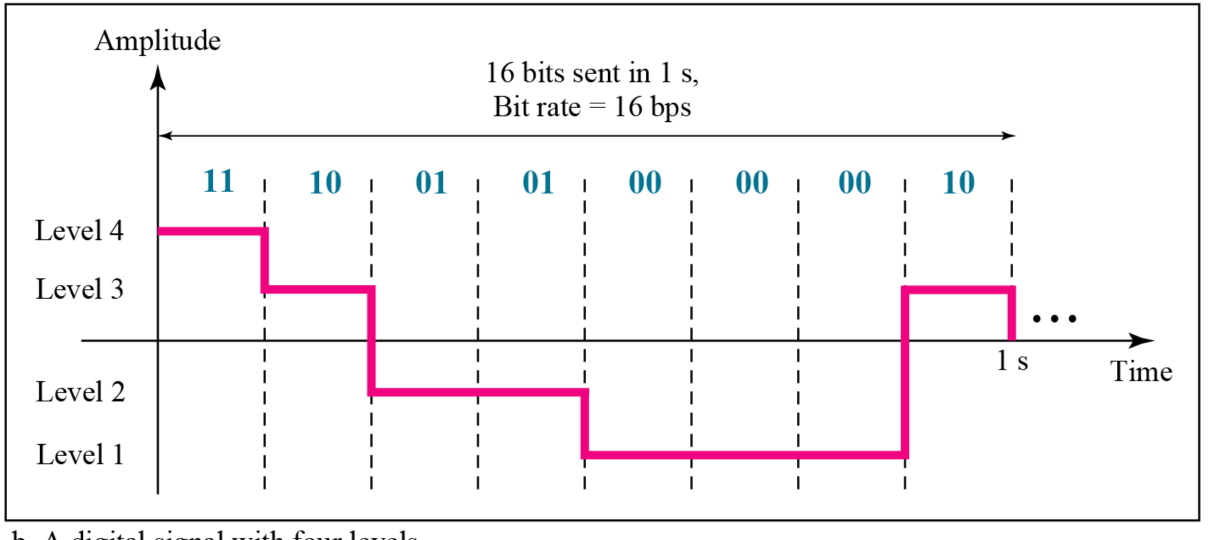
3-3 DIGITAL SIGNALS

In addition to being represented by an analog signal, information can also be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.

Figure ***Two digital signals: one with two signal levels and the other with four signal levels***



a. A digital signal with two levels



b. A digital signal with four levels

Example

A digital signal has eight levels. How many bits are needed per level? We calculate the number of bits from the formula

$$\boxed{\text{Number of bits per level} = \log_2 8 = 3}$$

Each signal level is represented by 3 bits.

Example

A digital signal has nine levels. How many bits are needed per level? We calculate the number of bits by using the formula. Each signal level is represented by 3.17 bits. However, this answer is not realistic. The number of bits sent per level needs to be an integer as well as a power of 2. For this example, 4 bits can represent one level.

Bit Rate

- Most Digital Signals are non-periodic. Therefore, period and frequency are not appropriate characteristics.
- Bit rate is used.
- Bit rate is number of bits sent in 1s. Expressed in **bps.**

Example

Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?

Solution

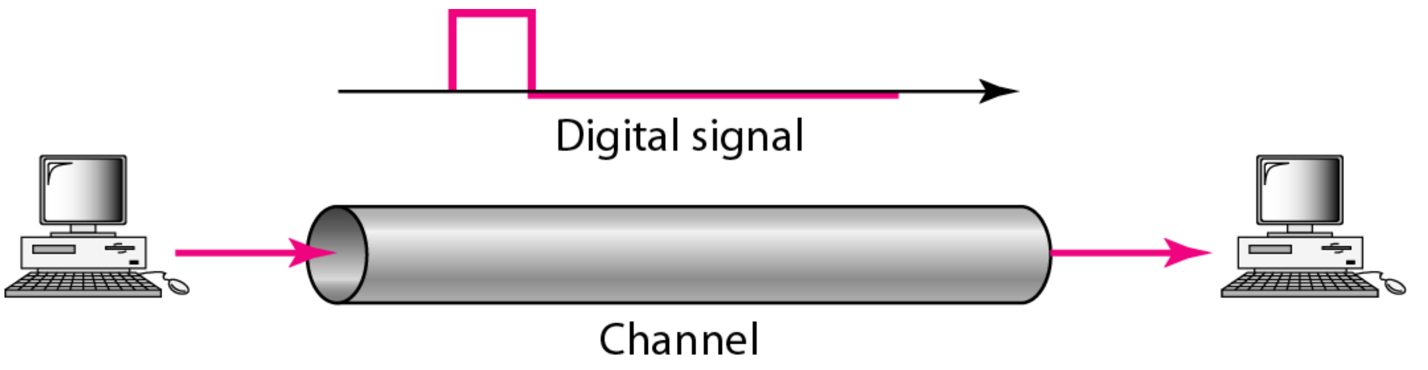
A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is

$$\boxed{100 \times 24 \times 80 \times 8 = 1{,}636{,}000 \text{ bps} = 1.636 \text{ Mbps}}$$

Bit Length

- Bit length is distance one bit occupies on transmission medium.
- Bit length= propagation speed* bit duration

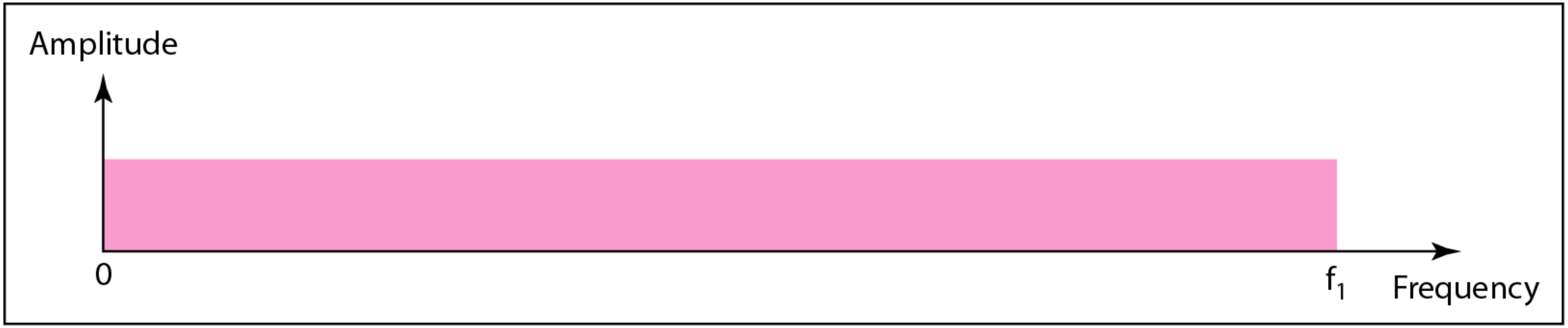
Figure *Baseband transmission*



Note

A digital signal is a composite analog signal with an infinite bandwidth.

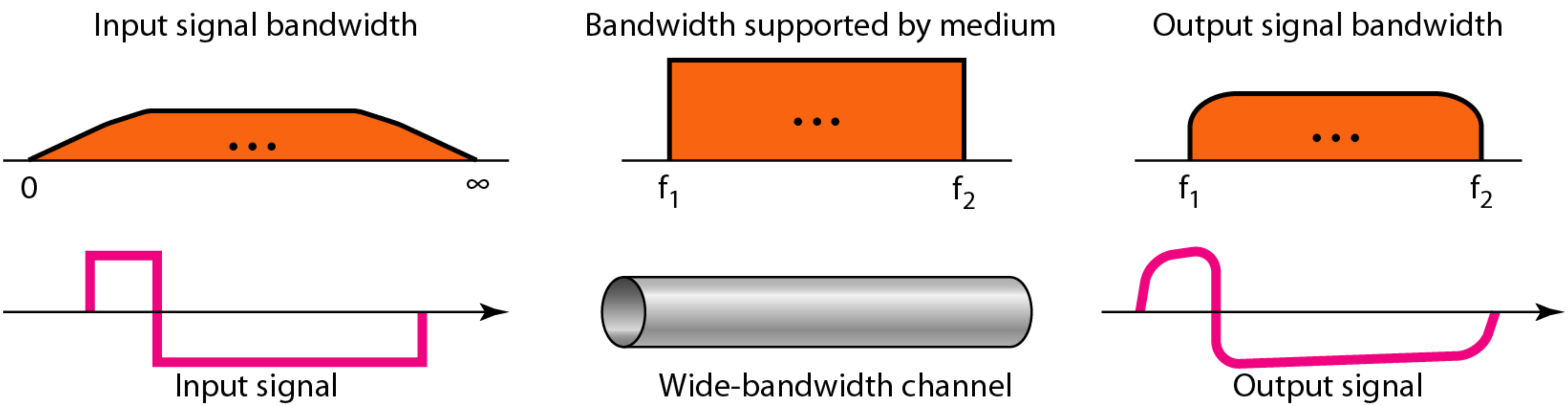
Figure *Bandwidths of two low-pass channels*



a. Low-pass channel, wide bandwidth

b. Low-pass channel, narrow bandwidth

Figure *Baseband transmission using a dedicated medium*



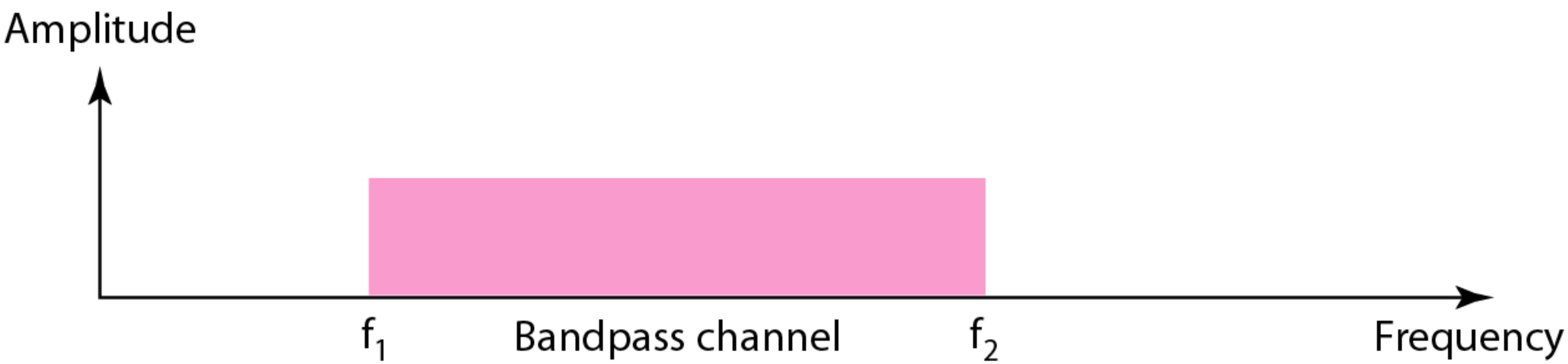
Note

Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.

Note

In baseband transmission, the required bandwidth is proportional to the bit rate; if we need to send bits faster, we need more bandwidth. Bit rate is twice the bandwidth required.

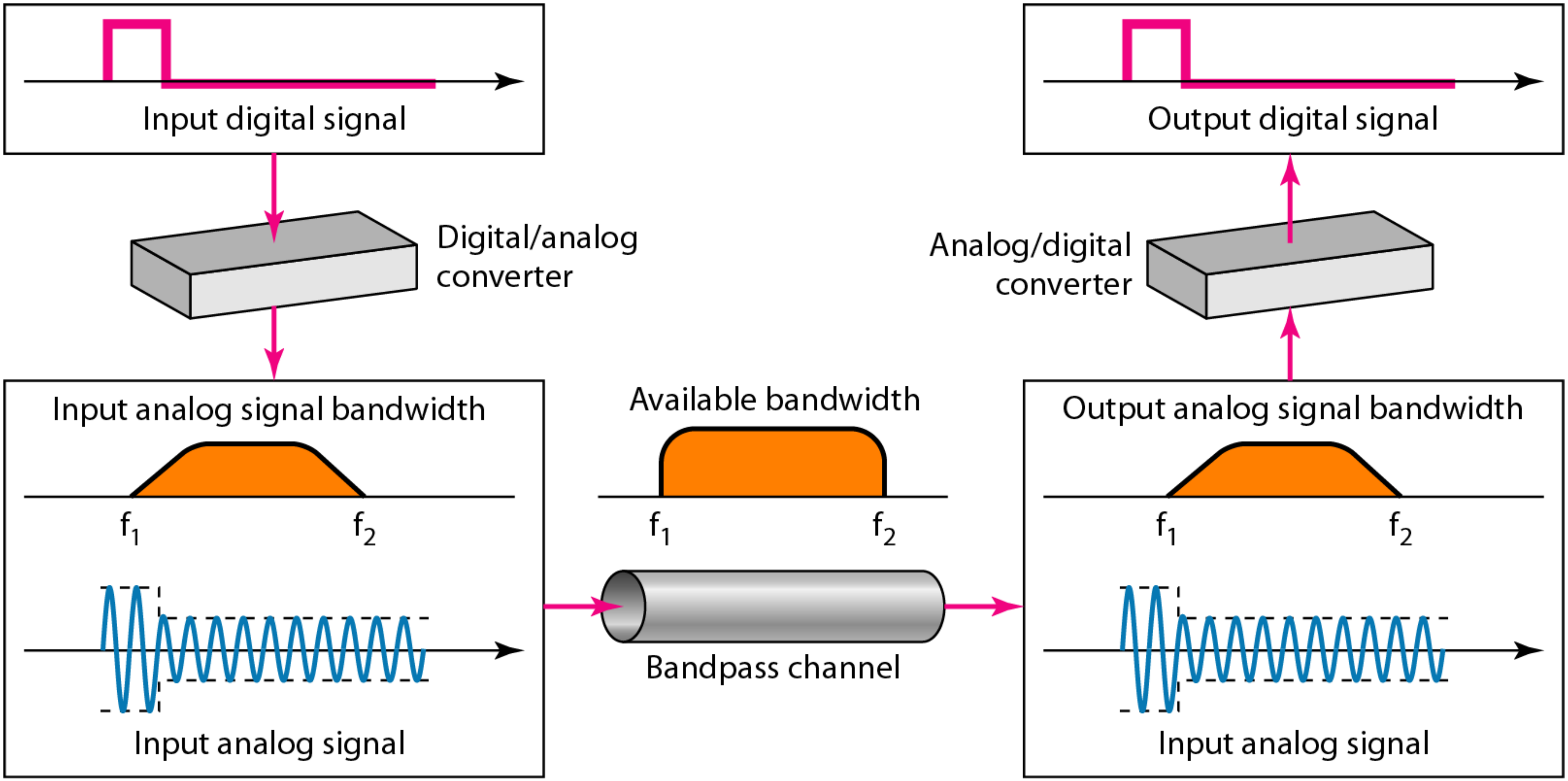
Broadband Transmission: *Bandwidth of a bandpass channel*



Note

**If the available channel is a bandpass channel, we cannot send the digital signal directly to the channel;
we need to convert the digital signal to an analog signal before transmission.**

Figure ***Modulation of a digital signal for transmission on a bandpass channel***



Example

An example of broadband transmission using modulation is the sending of computer data through a telephone subscriber line, the line connecting a resident to the central telephone office. These lines are designed to carry voice with a limited bandwidth. The channel is considered a bandpass channel. We convert the digital signal from the computer to an analog signal, and send the analog signal. We can install two converters to change the digital signal to analog and vice versa at the receiving end. The converter, in this case, is called a modem

DATA RATE LIMITS

A very important consideration in data communications is how fast we can send data, in bits per second, over a channel. Data rate depends on three factors:

1. ***The bandwidth available***
2. ***The level of the signals we use***
3. ***The quality of the channel (the level of noise)***

Topics discussed in this section:

Noiseless Channel: Nyquist Bit Rate
Noisy Channel: Shannon Capacity
Using Both Limits

Note

Increasing the levels of a signal may reduce the reliability of the system.

Noiseless Channel: Nyquist Bit Rate

$$\text{Bit Rate} = 2 * \text{bandwidth} * \log_2 L$$

Example

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}}$$

Example

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}}$$

Example

We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?

Solution

We can use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L$$

$$\log_2 L = 6.625 \qquad L = 2^{6.625} = 98.7 \text{ levels}$$

Since this result is not a power of 2, we need to either increase the number of levels or reduce the bit rate. If we have 128 levels, the bit rate is 280 kbps. If we have 64 levels, the bit rate is 240 kbps.

Noisy Channel : Shannon Capacity

- Capacity= Bandwidth * $\log_2$(1+SNR)

Example

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as

$$\boxed{C = B \log_2(1 + \text{SNR}) = B \log_2(1 + 0) = B \log_2 1 = B \times 0 = 0}$$

This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

Example

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1 + \text{SNR}) = 3000 \log_2 (1 + 3162) = 3000 \log_2 3163$$
$$= 3000 \times 11.62 = 34{,}860 \text{ bps}$$

This means that the highest bit rate for a telephone line is 34.860 kbps. If we want to send data faster than this, we can either increase the bandwidth of the line or improve the signal-to-noise ratio.

Example

The signal-to-noise ratio is often given in decibels. Assume that SNR_{dB} = 36 and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$SNR_{dB} = 10 \log_{10} SNR \rightarrow SNR = 10^{SNR_{dB}/10} \rightarrow SNR = 10^{3.6} = 3981$$

$$C = B \log_2 (1 + SNR) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Example

For practical purposes, when the SNR is very high, we can assume that SNR + 1 is almost the same as SNR. In these cases, the theoretical channel capacity can be simplified to

$$C = B \times \frac{\text{SNR}_{\text{dB}}}{3}$$

For example, we can calculate the theoretical capacity of the previous example as

$$C = 2\ \text{MHz} \times \frac{36}{3} = 24\ \text{Mbps}$$

Example

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

Solution

First, we use the Shannon formula to find the upper limit.

$$C = B \log_2(1 + \text{SNR}) = 10^6 \log_2(1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}$$

Example

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example. Then we use the Nyquist formula to find the number of signal levels.

$$4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \quad \Rightarrow \quad L = 4$$

Note

The Shannon capacity gives us the upper limit; the Nyquist formula tells us how many signal levels we need.

3–6 PERFORMANCE

One important issue in networking is the performance of the network—how good is it?.

Topics discussed in this section:

Bandwidth
Throughput
Latency (Delay)
Bandwidth-Delay Product

Note

In networking, we use the term bandwidth in two contexts.

- **The first, bandwidth in hertz, refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.**

- **The second, bandwidth in bits per second, refers to the speed of bit transmission in a channel or link.**

Example

If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps by using the same technology

Example

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

Solution

We can calculate the throughput as

$$\boxed{\text{Throughput} = \frac{12{,}000 \times 10{,}000}{60} = 2 \text{ Mbps}}$$

The throughput is almost one-fifth of the bandwidth in this case.

Latency (Delay)

Latency= Propagation Time+ transmission time +
queuing time+ processing delay

Example

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be 2.4 × 108 m/s in cable.

Solution

We can calculate the propagation time as

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^{8}} = 50 \text{ ms}$$

The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

Example

What are the propagation time and the transmission time for a 2.5-kbyte message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4 × 108 m/s.

Solution

We can calculate the propagation and transmission time as shown on the next slide:

Example

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{2500 \times 8}{10^9} = 0.020 \text{ ms}$$

Note that in this case, because the message is short and the bandwidth is high, the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

Example

What are the propagation time and the transmission time for a 5-Mbyte message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission times as shown on the next slide.

Example

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5{,}000{,}000 \times 8}{10^6} = 40 \text{ s}$$

Note that in this case, because the message is very long and the bandwidth is not very high, the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

Analog Transmission

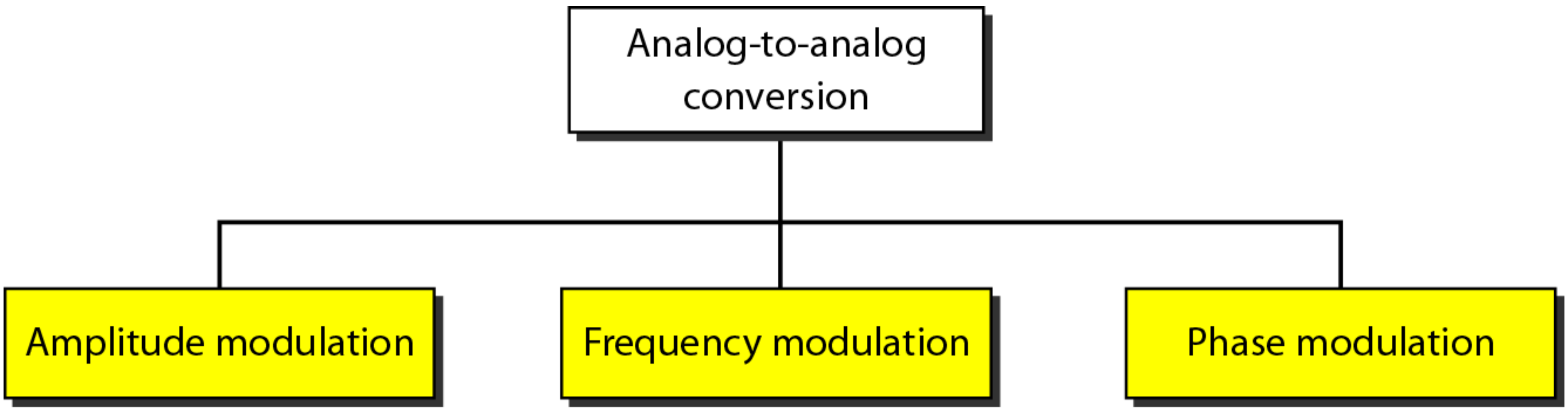
5-2 ANALOG AND DIGITAL

Analog-to-analog conversion is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us.

Topics discussed in this section:

- **Amplitude Modulation**
- **Frequency Modulation**
- **Phase Modulation**

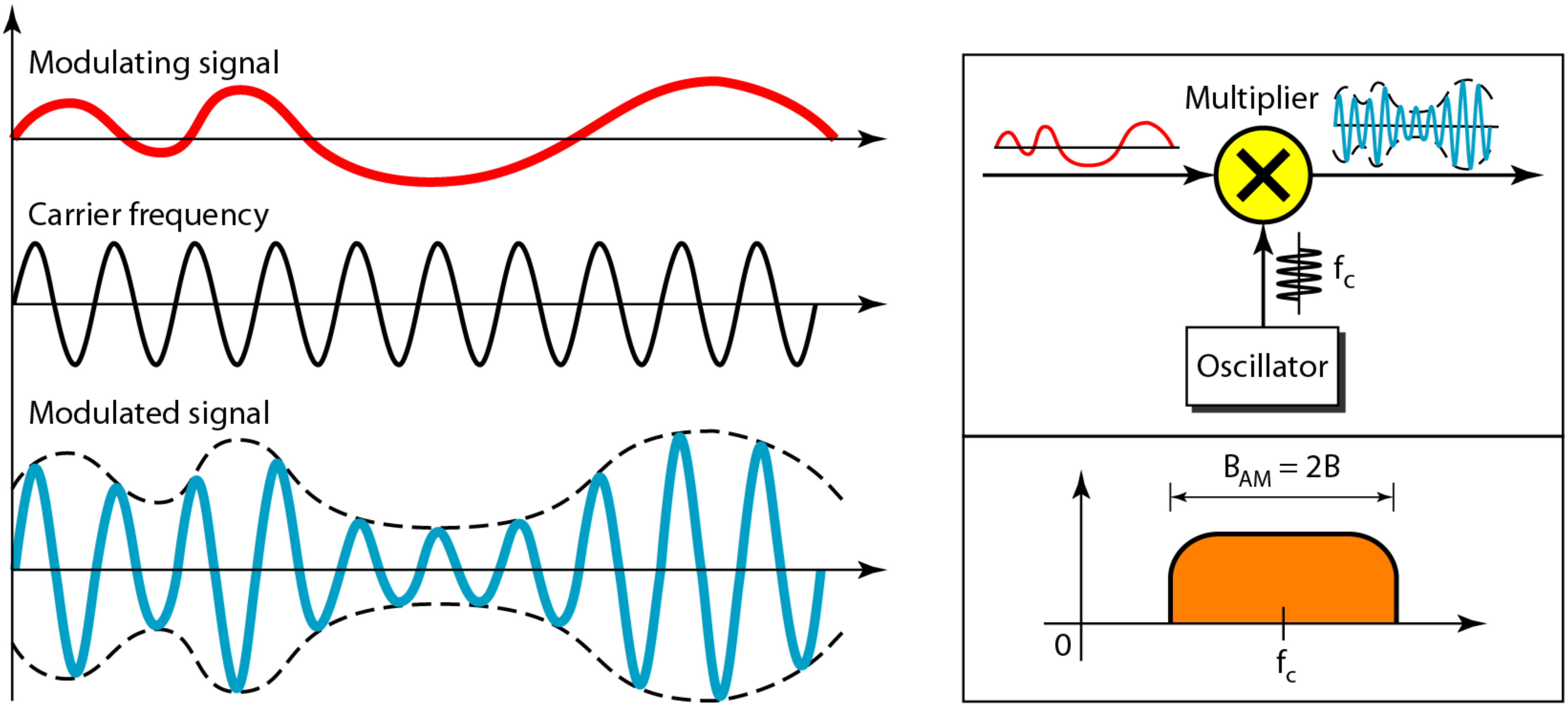
Figure 5.15 *Types of analog-to-analog modulation*



Amplitude Modulation

- A carrier signal is modulated only in amplitude value
- The modulating signal is the envelope of the carrier
- The required bandwidth is 2B, where B is the bandwidth of the modulating signal
- Since on both sides of the carrier freq. f_c, the spectrum is identical, we can discard one half, thus requiring a smaller bandwidth for transmission.

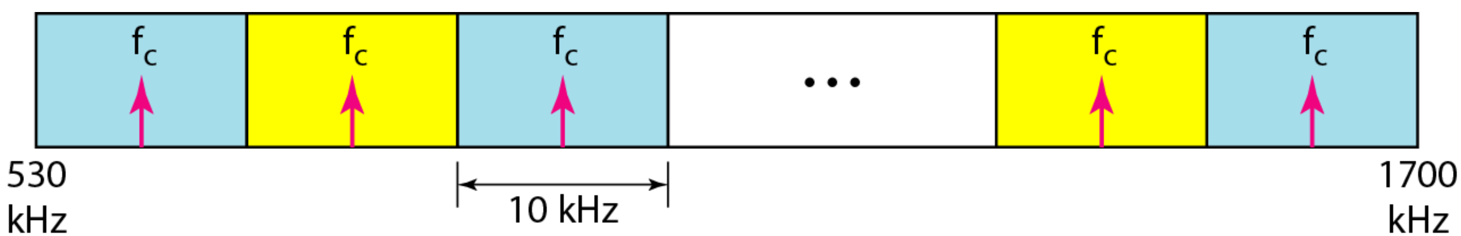
Figure 5.16 *Amplitude modulation*



Note

The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

Figure 5.17 *AM band allocation*



Frequency Modulation

- The modulating signal changes the freq. f_c of the carrier signal
- The bandwidth for FM is high
- It is approx. 10x the signal frequency

Note

The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$. Where β is usually 4.

Figure 5.18 *Frequency modulation*

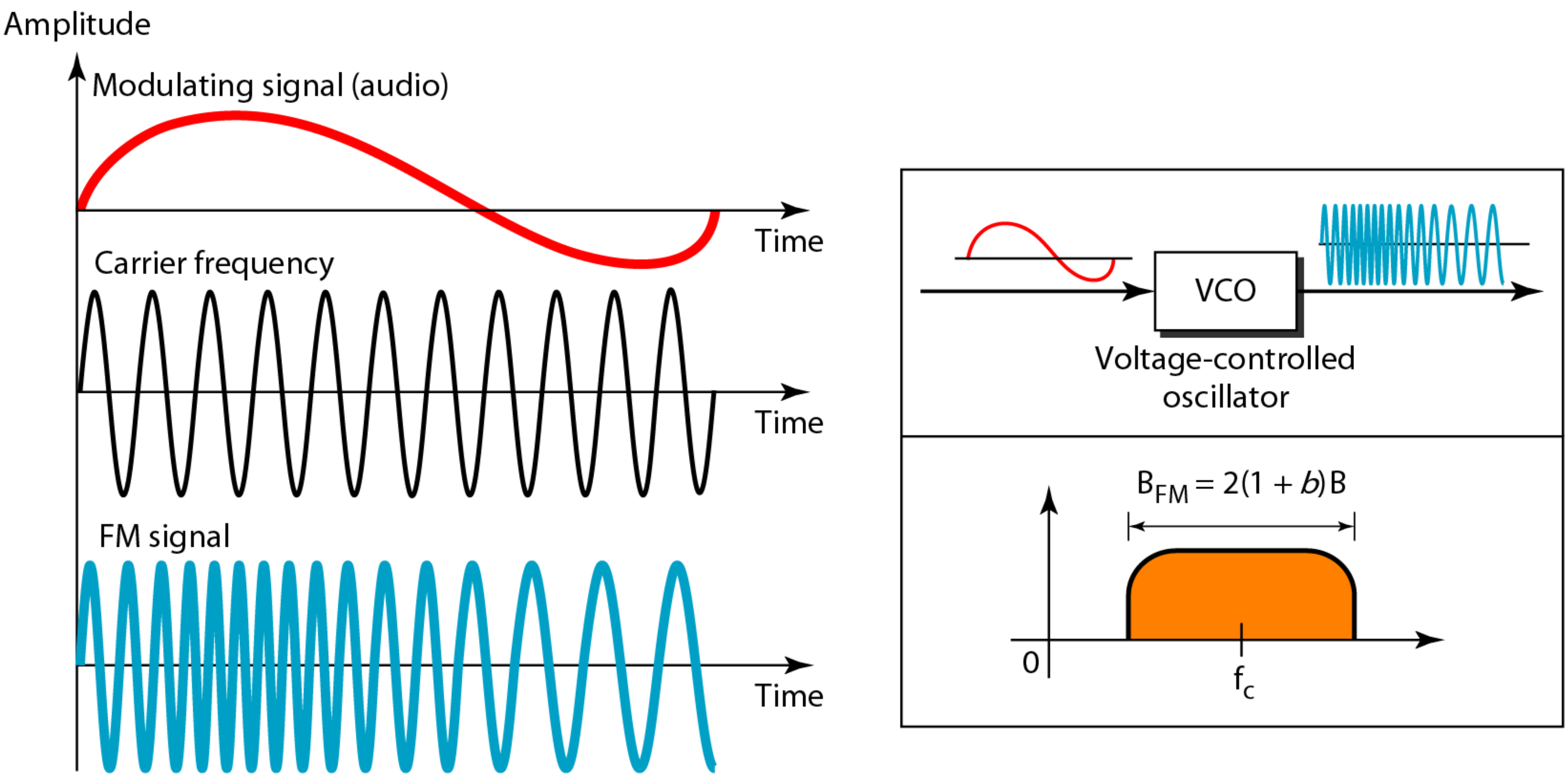
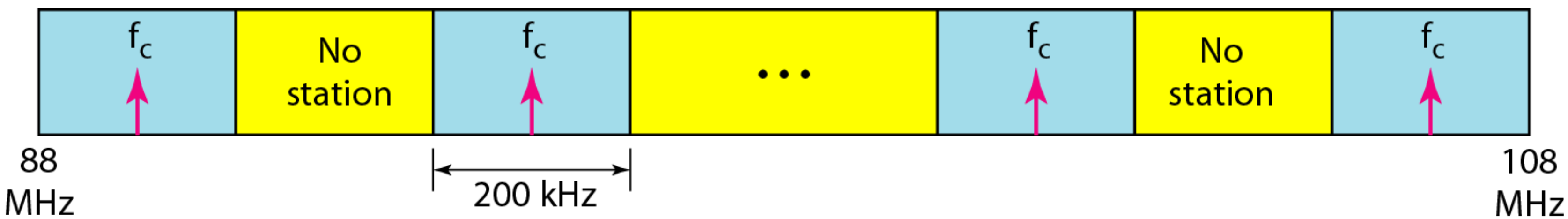


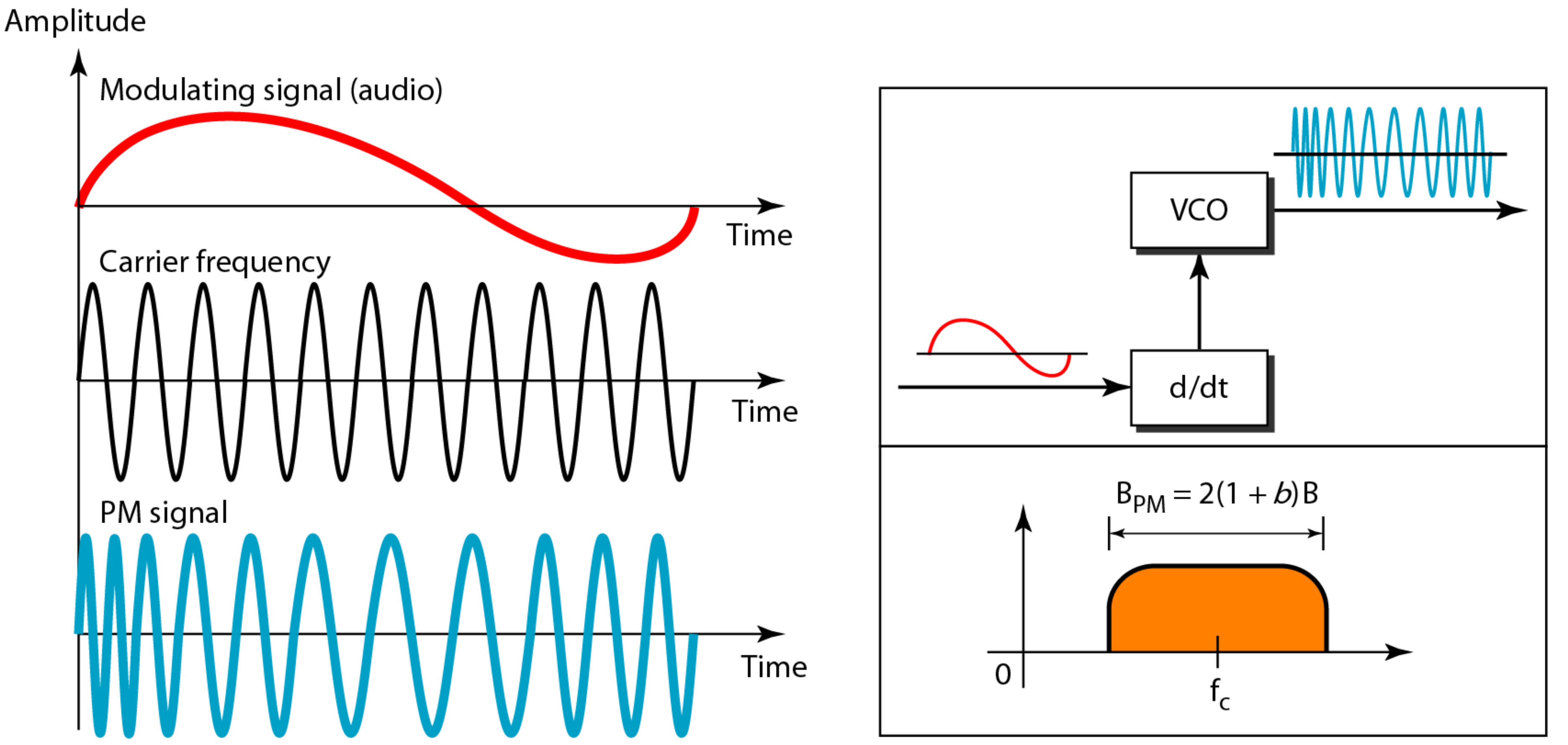
Figure 5.19 *FM band allocation*



Phase Modulation (PM)

- The modulating signal only changes the phase of the carrier signal.
- The phase change manifests itself as a frequency change but the instantaneous frequency change is proportional to the derivative of the amplitude.
- The bandwidth is higher than for AM.

Figure 5.20 *Phase modulation*



Note

The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal:

$$B_{PM} = 2(1 + \beta)B.$$

Where $\beta = 2$ most often.

Digital Transmission

4-2 ANALOG-TO-DIGITAL CONVERSION

A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, the tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation.

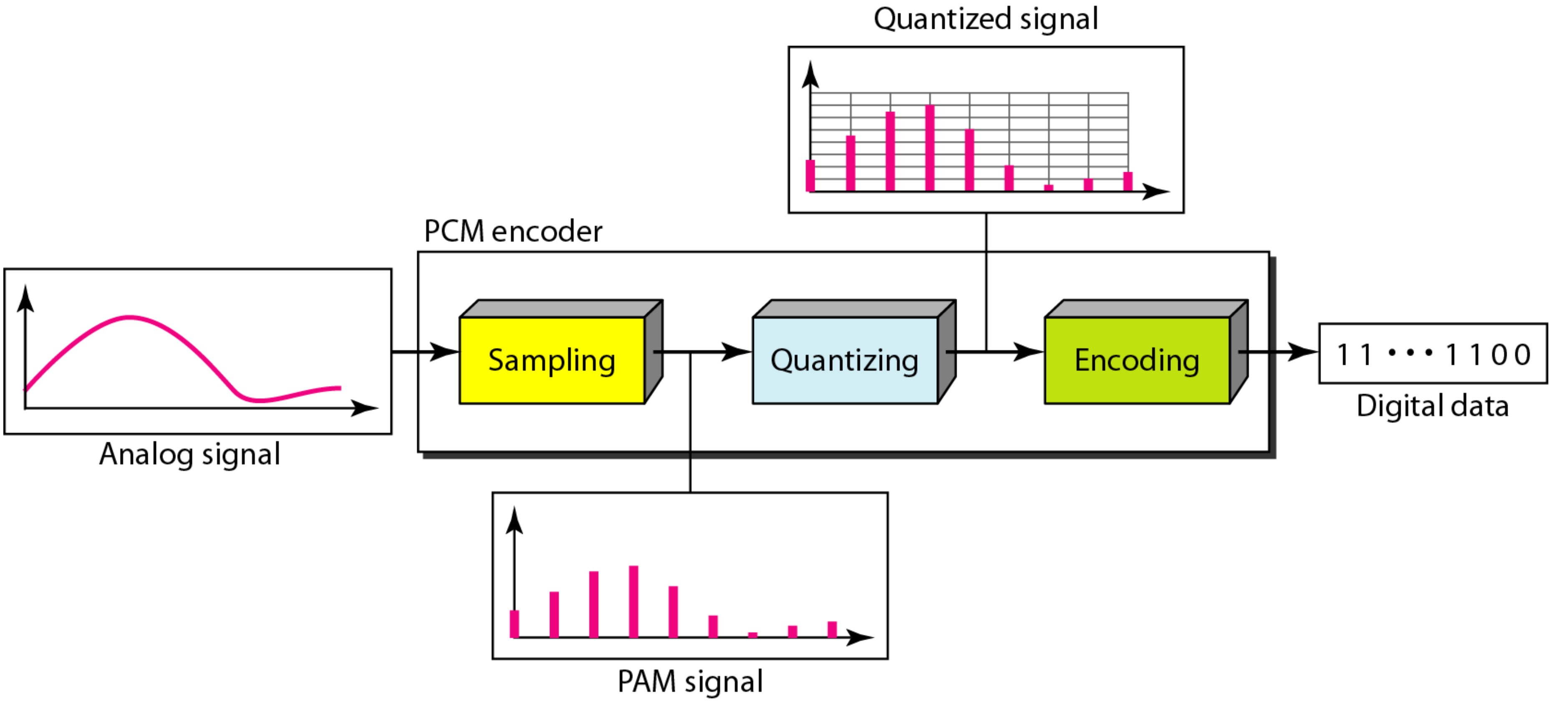
Topics discussed in this section:

- **Pulse Code Modulation (PCM)**

PCM

- PCM consists of three steps to digitize an analog signal:
  1. Sampling
  2. Quantization
  3. Binary encoding
- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, ie remove high frequency components that affect the signal shape.

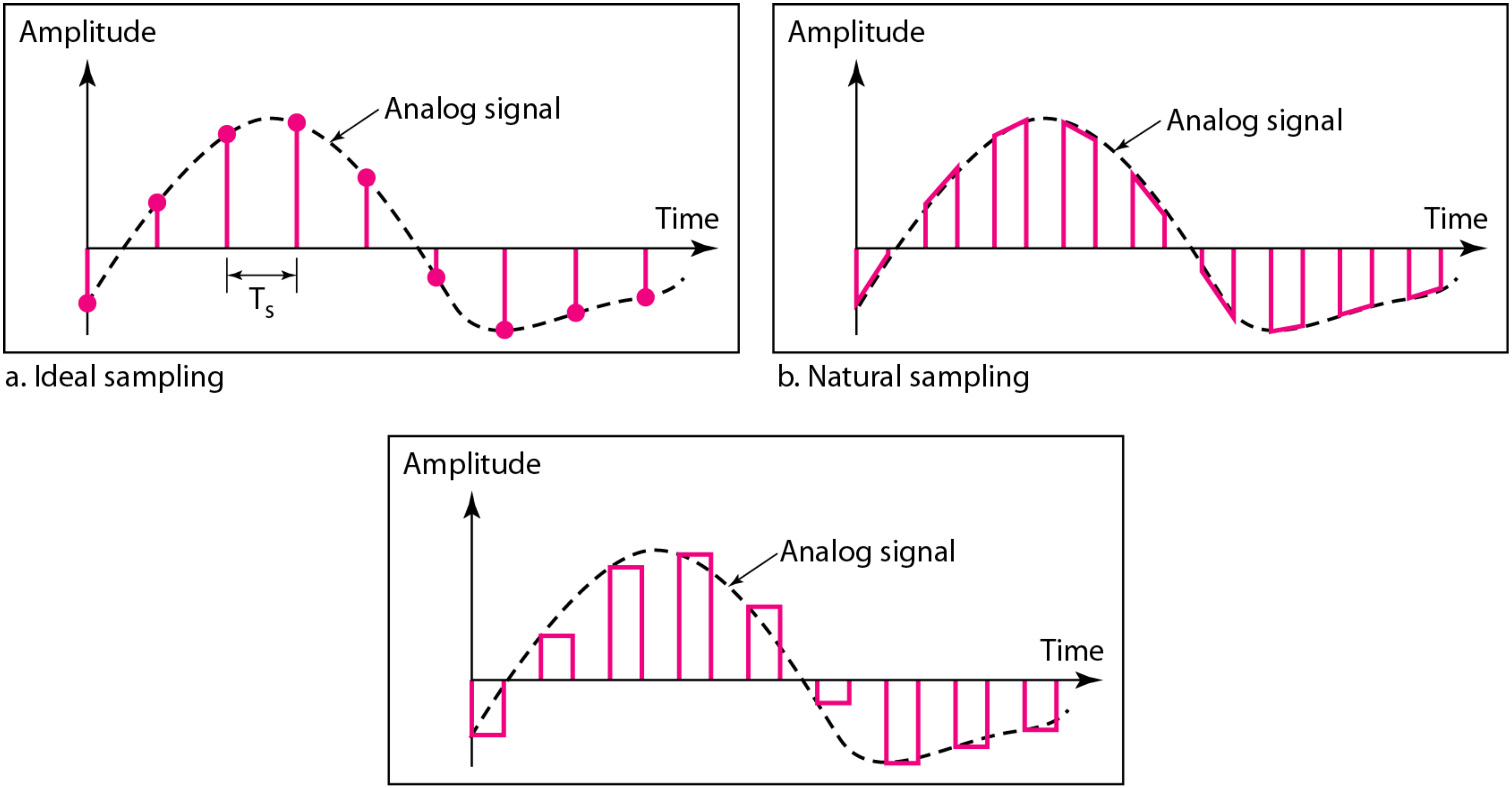
Figure 4.21 *Components of PCM encoder*



Sampling

- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - sample and hold, like natural but with single amplitude value
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

Figure 4.22 *Three different sampling methods for PCM*



a. Ideal sampling

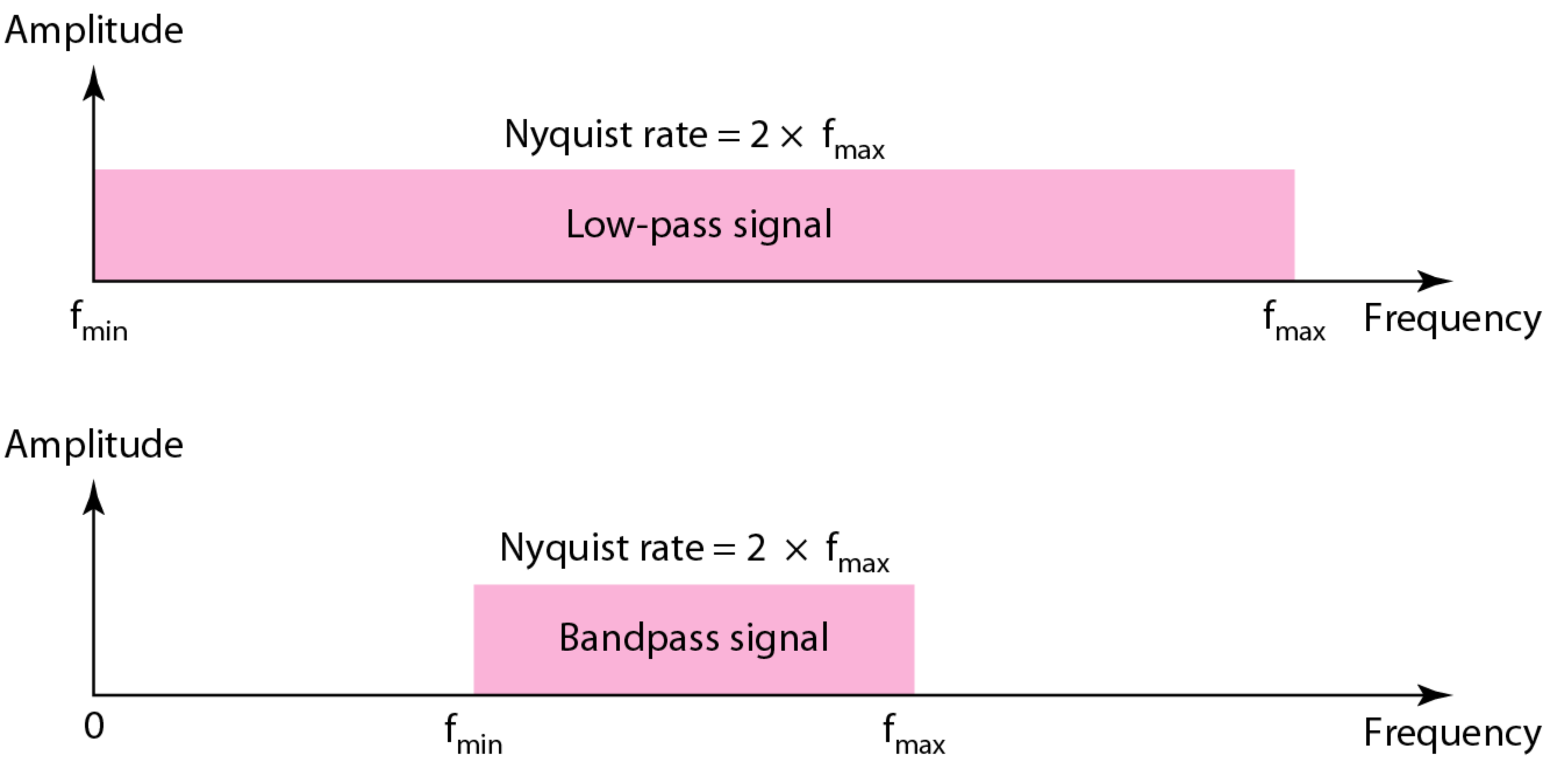
b. Natural sampling

c. Flat-top sampling

Note

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*



Quantization

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into L zones, each of height Δ.

$$\Delta = (\max - \min)/L$$

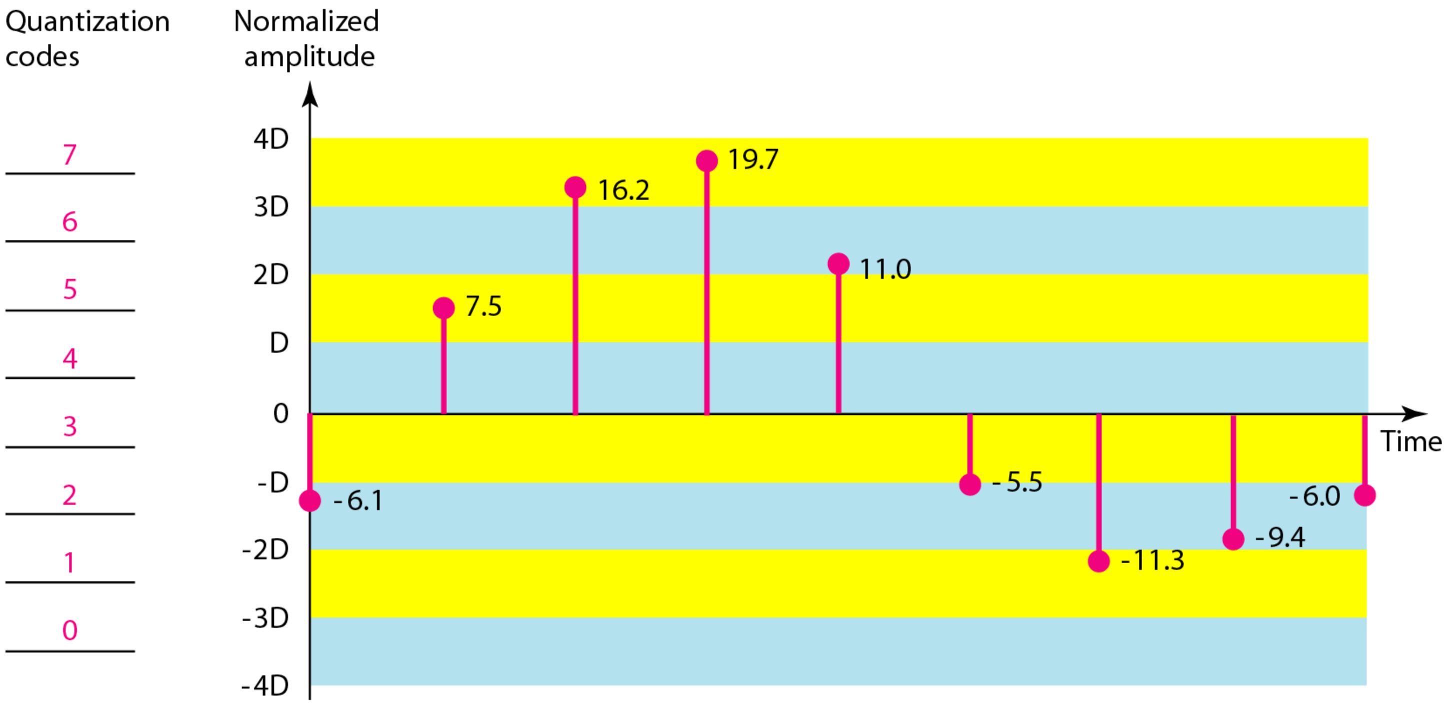
Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitutes V_{min}=-20V and V_{max}=+20V.
- We want to use L=8 quantization levels.
- Zone width Δ = (20 - -20)/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

Figure 4.26 *Quantization and encoding of a sampled signal*



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Quantization Error

- When a signal is quantized, we introduce an error - the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- The more zones, the smaller Δ which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples -> higher bit rate

Quantization Error and SN_QR

- Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.
- Non linear quantization is used to alleviate this problem. Goal is to keep SN_QR fixed for all sample values.
- Two approaches:
  - The quantization levels follow a logarithmic curve. Smaller Δ's at lower amplitudes and larger Δ's at higher amplitudes.
  - Companding: The sample values are compressed at the sender into logarithmic zones, and then expanded at the receiver. The zones are fixed in height.

Bit rate and bandwidth requirements of PCM

- The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate

$$\text{Bit rate} = n_b \times f_s$$

- The bandwidth required to transmit this signal depends on the type of line encoding used. Refer to previous section for discussion and formulas.
- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Analog Transmission

5-1 DIGITAL-TO-ANALOG CONVERSION

***Digital-to-analog* conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.**

Topics discussed in this section:

- **Aspects of Digital-to-Analog Conversion**
- **Amplitude Shift Keying**
- **Frequency Shift Keying**
- **Phase Shift Keying**
- **Quadrature Amplitude Modulation**

Digital to Analog Conversion

- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency f_c) performs the function of transporting the digital data in an analog waveform.
- The analog carrier signal is manipulated to uniquely identify the digital data being carried.

Figure 5.1 *Digital-to-analog conversion*

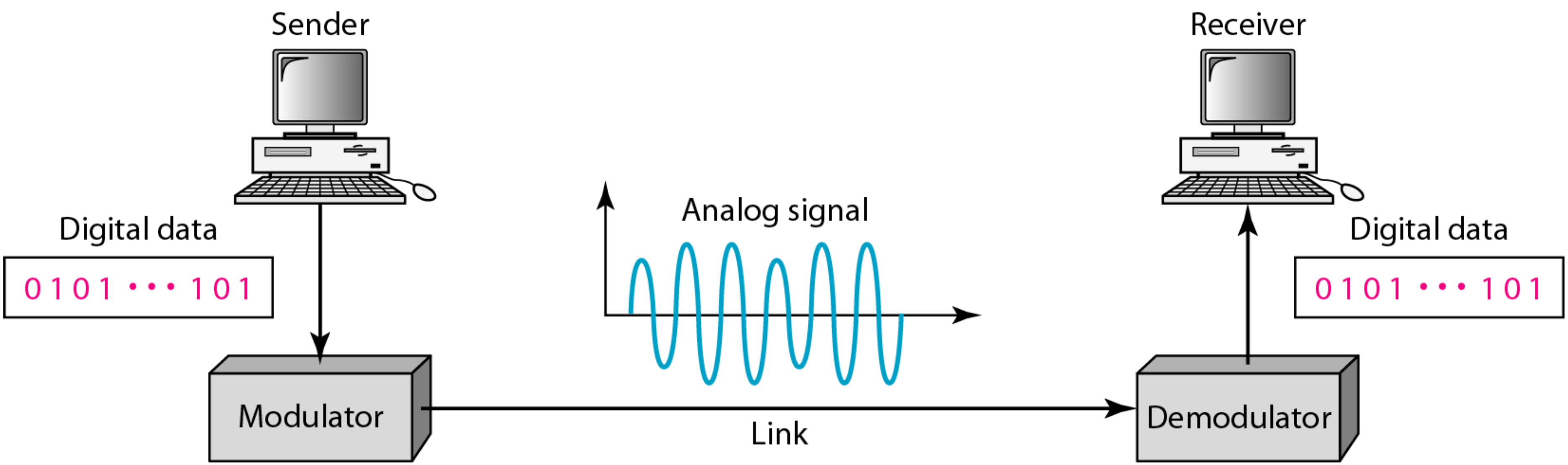
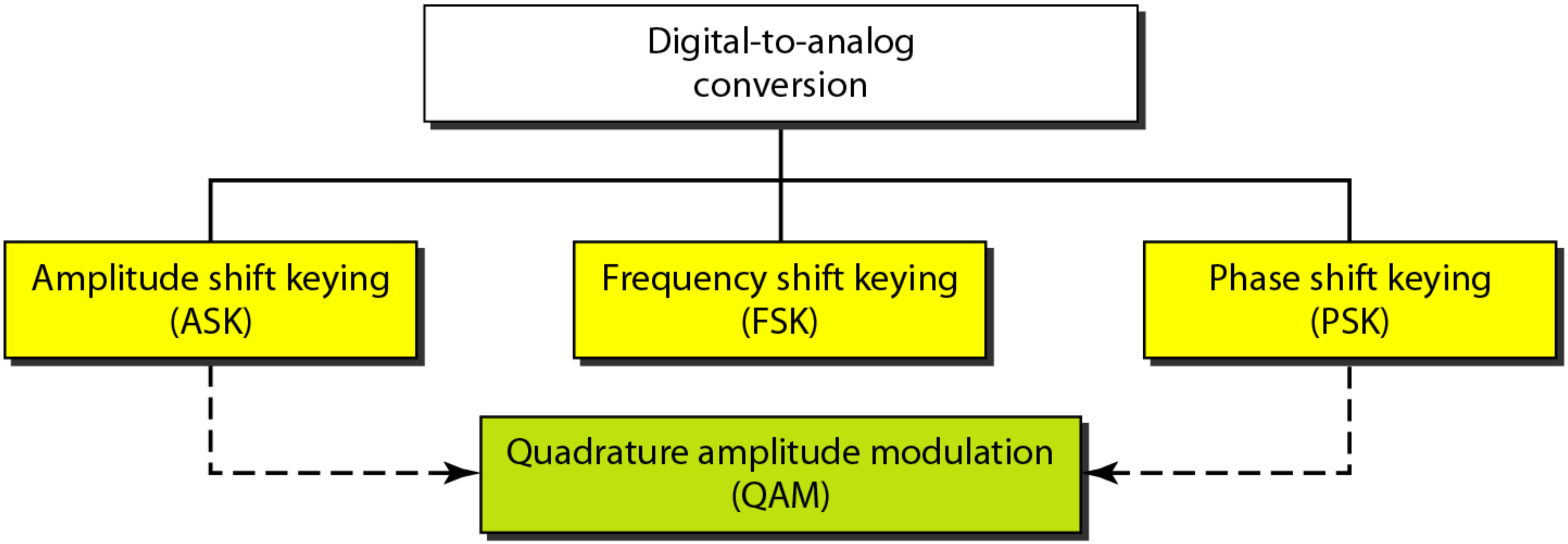


Figure 5.2 *Types of digital-to-analog conversion*



Note

Bit rate, N, is the number of bits per second (bps). Baud rate is the number of signal elements per second (bauds). In the analog transmission of digital data, the signal or baud rate is less than or equal to the bit rate.

S=Nx1/r bauds

Where r is the number of data bits per signal element.

Example 5.1

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, r = 4, S = 1000, and N is unknown. We can find the value of N from

$$S = N \times \frac{1}{r} \qquad \text{or} \qquad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Example 5.2

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, S = 1000, N = 8000, and r and L are unknown. We find first the value of r and then the value of L.

$$S = N \times \frac{1}{r} \quad \longrightarrow \quad r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \quad \longrightarrow \quad L = 2^r = 2^8 = 256$$

Amplitude Shift Keying (ASK)

- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- For example: a digital “1” could not affect the signal, whereas a digital “0” would, by making it zero.
- The line encoding will determine the values of the analog waveform to reflect the digital data being carried.

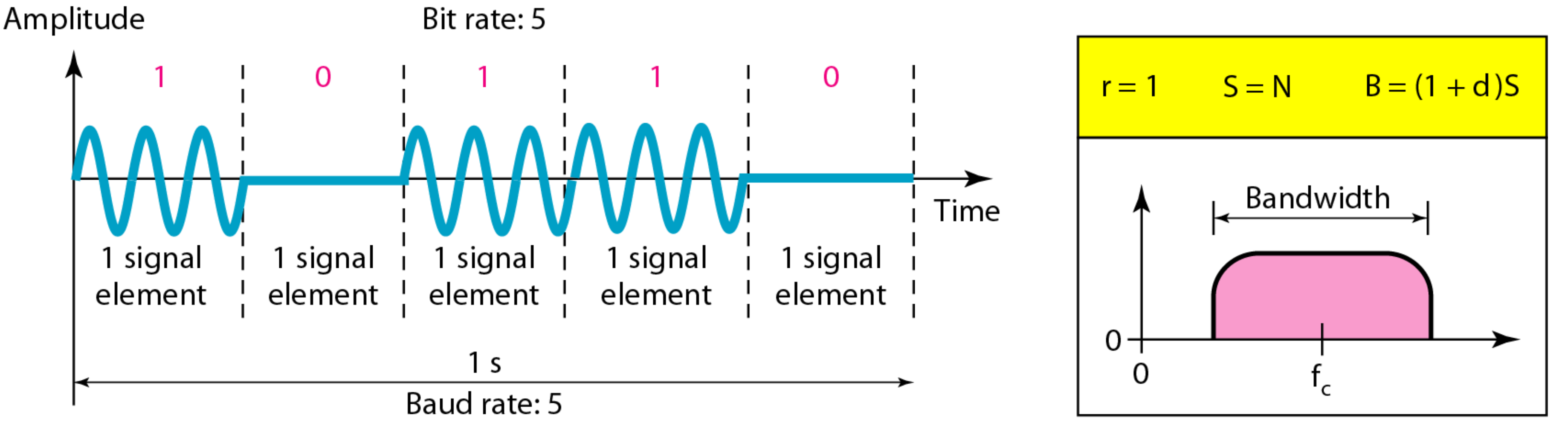
Bandwidth of ASK

- The bandwidth B of ASK is proportional to the signal rate S.

$$B = (1+d)S$$

- "d" is due to modulation and filtering, lies between 0 and 1.

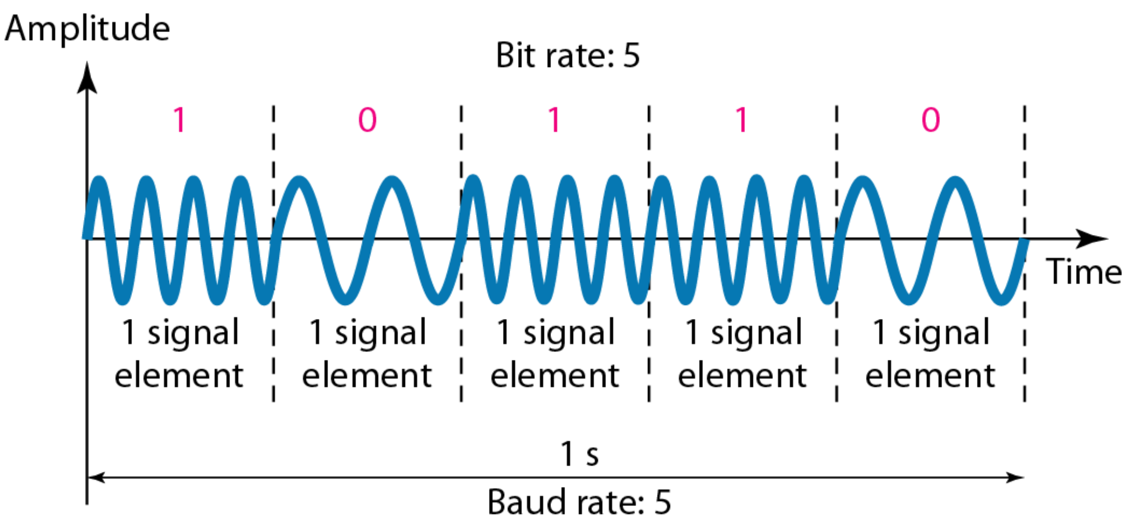
Figure 5.3 *Binary amplitude shift keying*

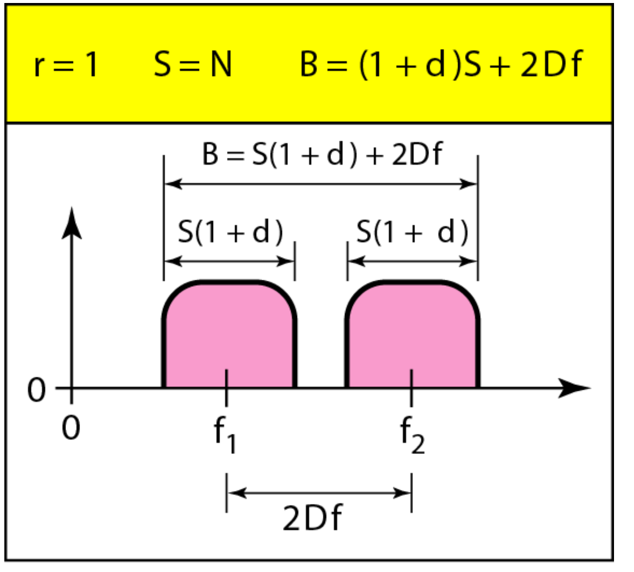


Frequency Shift Keying

- The digital data stream changes the frequency of the carrier signal, f_c.
- For example, a “1” could be represented by $f_1=f_c+\Delta f$, and a “0” could be represented by $f_2=f_c-\Delta f$.

Figure 5.6 *Binary frequency shift keying*





Bandwidth of FSK

- If the difference between the two frequencies (f_1 and f_2) is $2\Delta f$, then the required BW B will be:

$$B = (1+d)xS + 2\Delta f$$

Example 5.5

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with d = 1?

Solution

This problem is similar to Example 5.3, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose 2Δf to be 50 kHz; this means

$$B = (1 + d) \times S + 2\Delta f = 100 \rightarrow 2S = 50 \text{ kHz} \quad S = 25 \text{ kbaud} \quad N = 25 \text{ kbps}$$

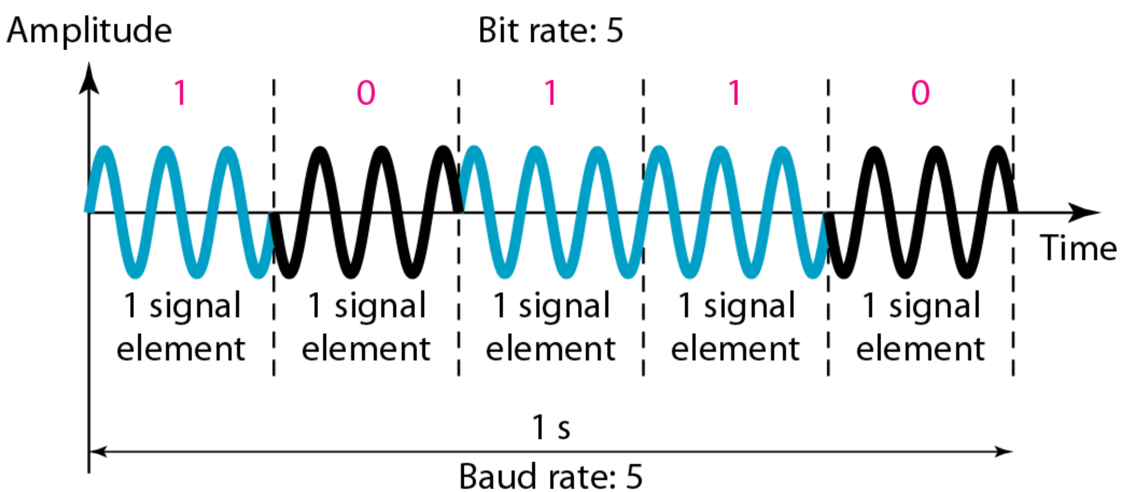
Phase Shift Keyeing

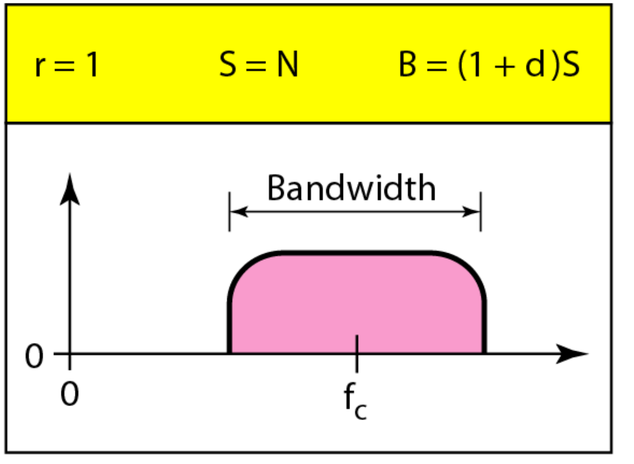
- We vary the phase shift of the carrier signal to represent digital data.
- The bandwidth requirement, B is:

$$B = (1+d)xS$$

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

Figure 5.9 *Binary phase shift keying*

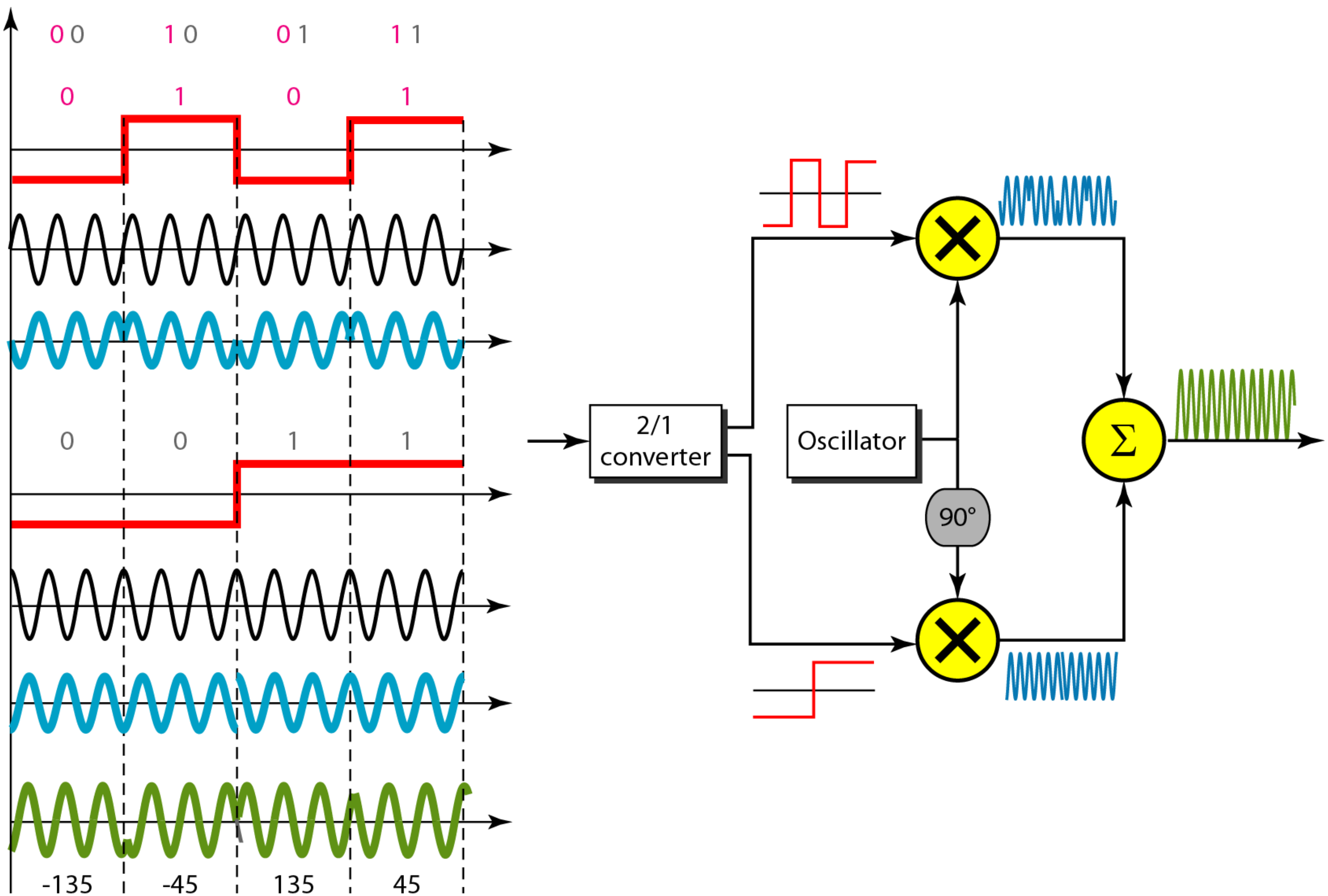




Quadrature PSK

- 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.

Figure 5.11 *QPSK and its implementation*



Note

Quadrature amplitude modulation is a combination of ASK and PSK.

Digital Transmission

4-1 DIGITAL-TO-DIGITAL CONVERSION

In this section, we see how we can represent digital data by using digital signals. The conversion involves three techniques: line coding, block coding, and scrambling. Line coding is always needed; block coding and scrambling may or may not be needed.

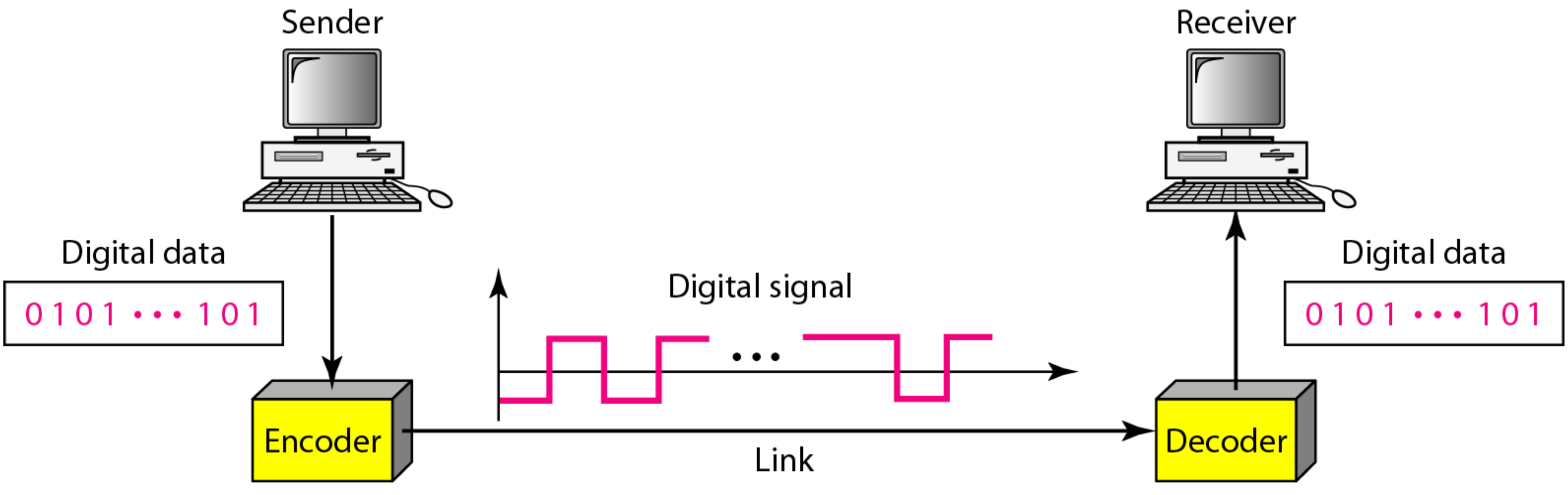
Topics discussed in this section:

- **Line Coding**
- **Line Coding Schemes**
- **Block Coding**
- **Scrambling**

Line Coding

- Converting a string of 1’s and 0’s (digital data) into a sequence of signals that denote the 1’s and 0’s.
- For example a high voltage level (+V) could represent a “1” and a low voltage level (0 or -V) could represent a “0”.

Figure 4.1 *Line coding and decoding*



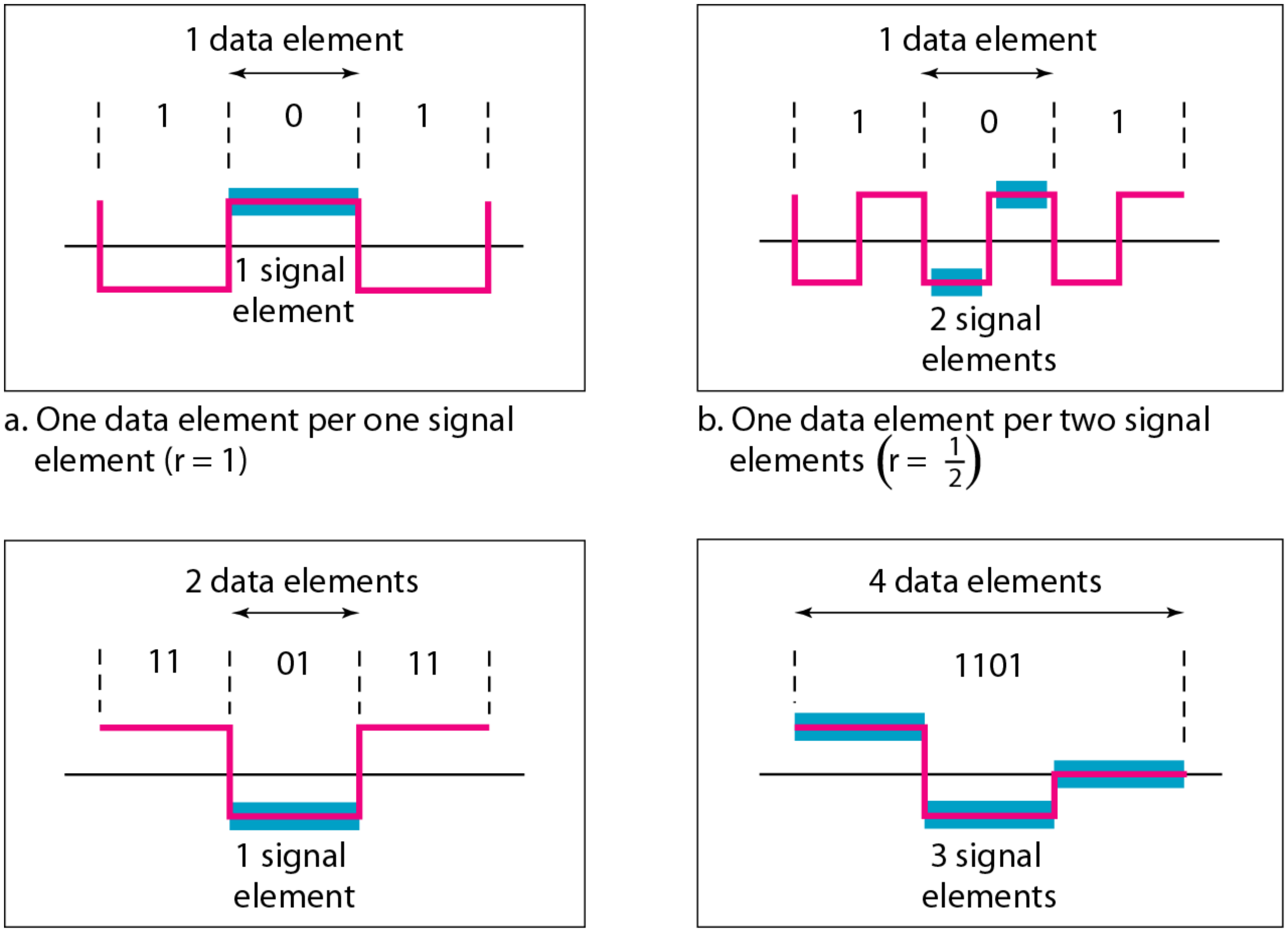
Mapping Data symbols onto Signal levels

- A data symbol (or element) can consist of a number of data bits:
  - 1 , 0 or
  - 11, 10, 01, ……
- A data symbol can be coded into a single signal element or multiple signal elements
  - 1 -> +V, 0 -> -V
  - 1 -> +V and -V, 0 -> -V and +V
- The ratio ‘r’ is the number of data elements carried by a signal element.

Relationship between data rate and signal rate

- The data rate defines the number of bits sent per sec - bps. It is often referred to the bit rate.
- The signal rate is the number of signal elements sent in a second and is measured in bauds. It is also referred to as the modulation rate.
- Goal is to increase the data rate whilst reducing the baud rate.

Figure 4.2 ***Signal element versus data element***



a. One data element per one signal element (r = 1)

b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$

c. Two data elements per one signal element (r = 2)

d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

Data rate and Baud rate

- The baud or signal rate can be expressed as:

$$S = c \times N \times 1/r \text{ bauds}$$

where N is data rate

c is the case factor (worst, best & avg.)

r is the ratio between data element & signal element

Example 4.1

A signal is carrying data in which one data element is encoded as one signal element (r = 1). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

Solution

We assume that the average value of c is 1/2 . The baud rate is then

$$S = c \times N \times \frac{1}{r} = \frac{1}{2} \times 100{,}000 \times \frac{1}{1} = 50{,}000 = 50 \text{ kbaud}$$

Considerations for choosing a good signal element referred to as line encoding

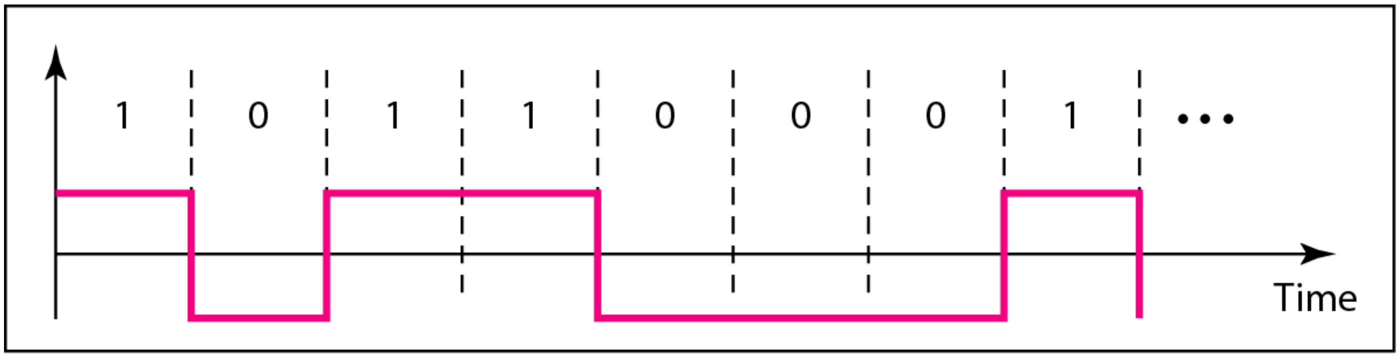
Line encoding C/Cs

- DC components - when the voltage level remains constant for long periods of time, there is an increase in the low frequencies of the signal. Most channels are bandpass and may not support the low frequencies.
- This will require the removal of the dc component of a transmitted signal.

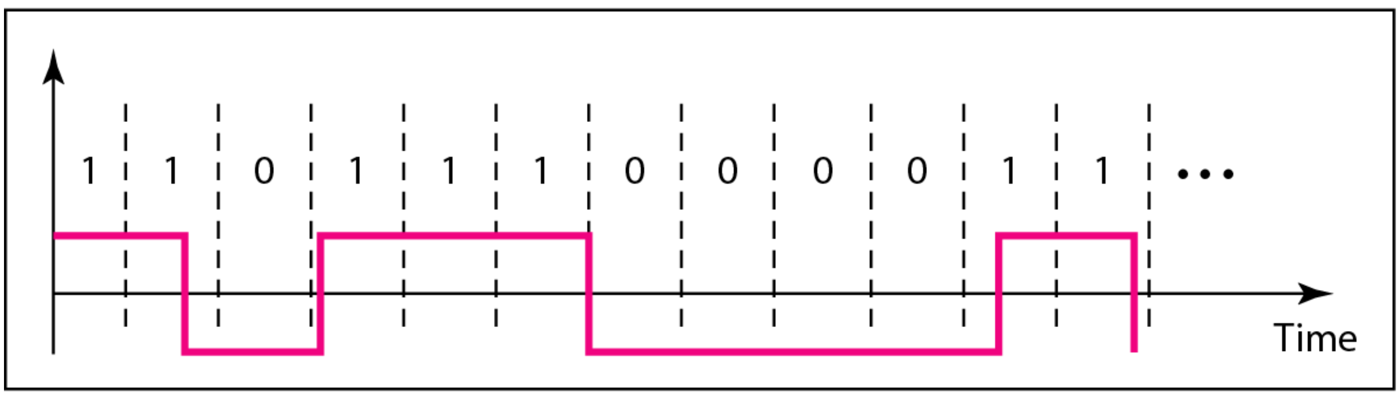
Line encoding C/Cs

- Self synchronization - the clocks at the sender and the receiver must have the same bit interval.
- If the receiver clock is faster or slower it will misinterpret the incoming bit stream.

Figure 4.3 *Effect of lack of synchronization*



a. Sent



b. Received

Example 4.3

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent	1001 bits received	1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

1,000,000 bits sent	1,001,000 bits received	1000 extra bps

Line encoding C/Cs

- Error detection - errors occur during transmission due to line impairments.
- Some codes are constructed such that when an error occurs it can be detected. For example: a particular signal transition is not part of the code. When it occurs, the receiver will know that a symbol error has occurred.

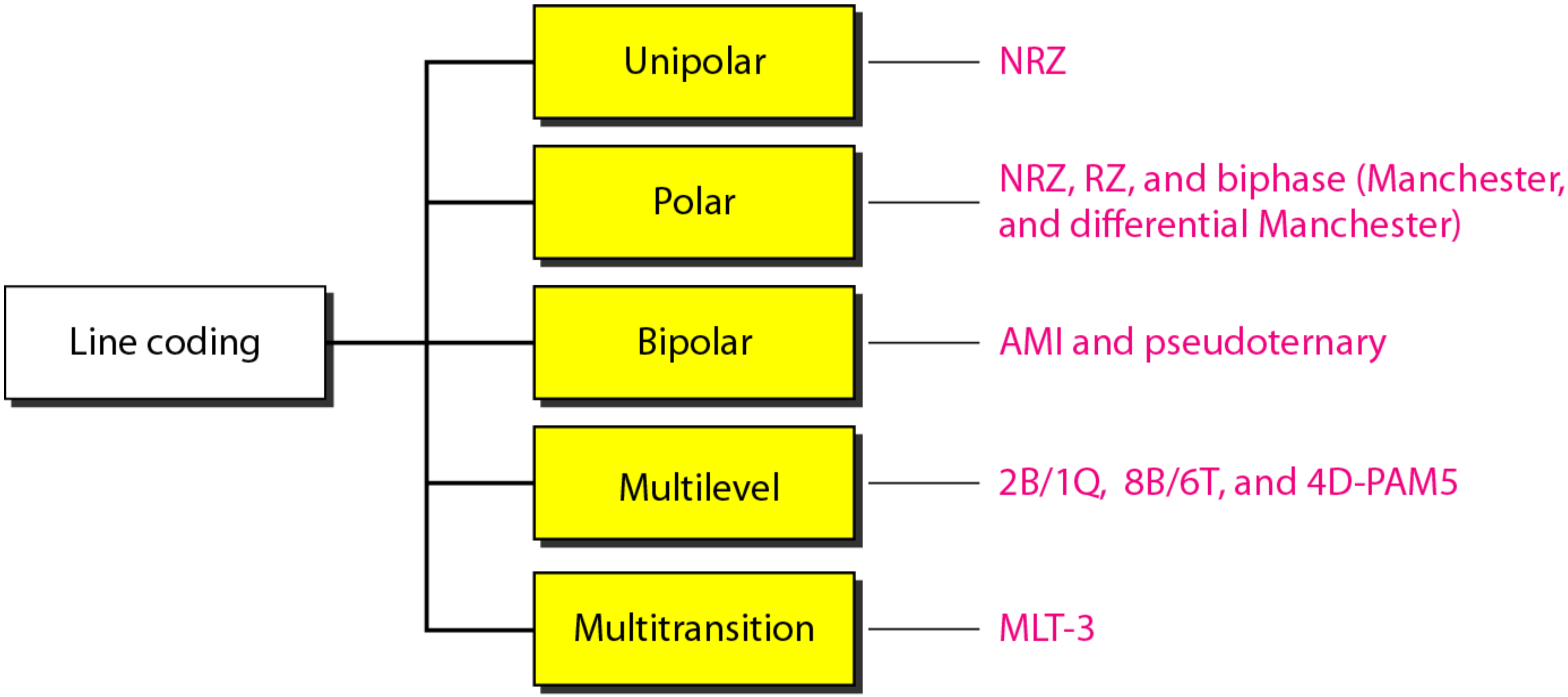
Line encoding C/Cs

- Noise and interference - there are line encoding techniques that make the transmitted signal "immune" to noise and interference.
- This means that the signal cannot be corrupted, it is stronger than error detection.

Line encoding C/Cs

- Complexity - the more robust and resilient the code, the more complex it is to implement and the price is often more than simple one.

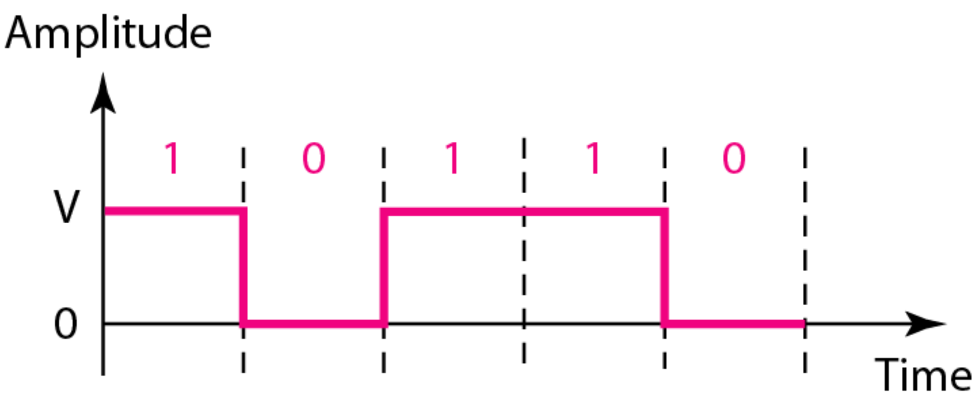
Figure 4.4 *Line coding schemes*



Unipolar

- All signal levels are on one side of the time axis - either above or below
- NRZ - Non Return to Zero scheme is an example of this code. The signal level does not return to zero during a symbol transmission.
- Scheme is prone to DC components. It has no synchronization or any error detection. It is simple but costly in power consumption.

Figure 4.5 *Unipolar NRZ scheme*

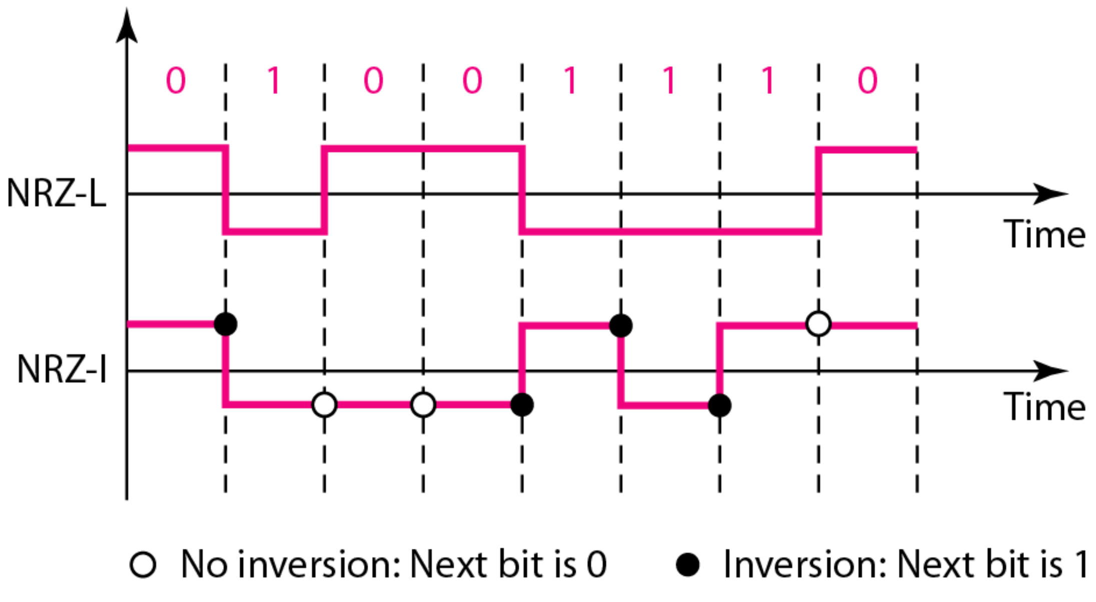


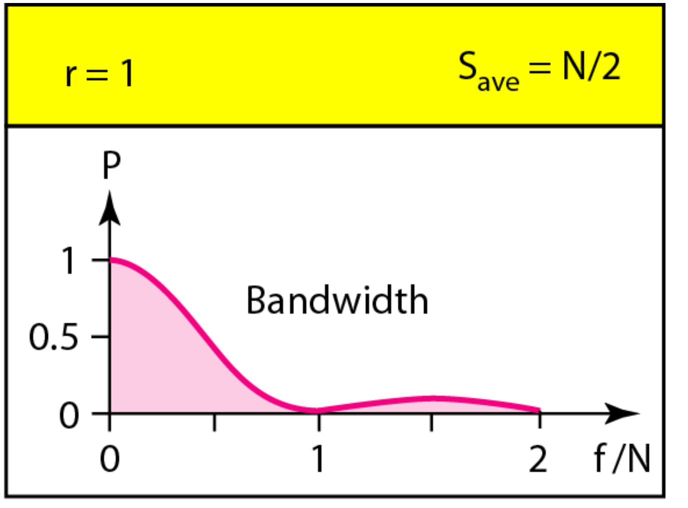
$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Polar - NRZ

- The voltages are on both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 1 and -V for 0.
- There are two versions:
  - NZR - Level (NRZ-L) - positive voltage for one symbol and negative for the other
  - NRZ - Inversion (NRZ-I) - the change or lack of change in polarity determines the value of a symbol. E.g. a “1” symbol inverts the polarity a “0” does not.

Figure 4.6 *Polar NRZ-L and NRZ-I schemes*





Note

In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit.

Note

NRZ-L and NRZ-I both have an average signal rate of N/2 Bd.

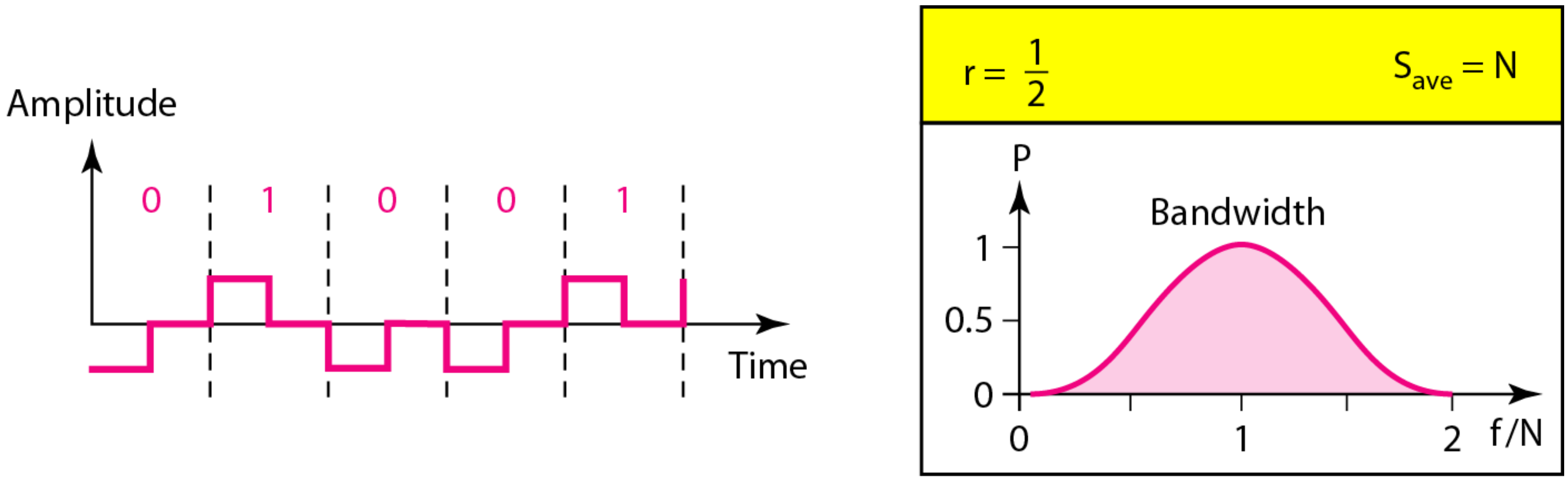
Note

NRZ-L and NRZ-I both have a DC component problem, it is worse for NRZ-L. Both have no self synchronization &no error detection. Both are relatively simple to implement.

Polar - RZ

- The Return to Zero (RZ) scheme uses three voltage values. +, 0, -.
- Each symbol has a transition in the middle. Either from high to zero or from low to zero.
- This scheme has more signal transitions (two per symbol) and therefore requires a wider bandwidth.
- No DC components or baseline wandering.
- Self synchronization - transition indicates symbol value.
- More complex as it uses three voltage level. It has no error detection capability.

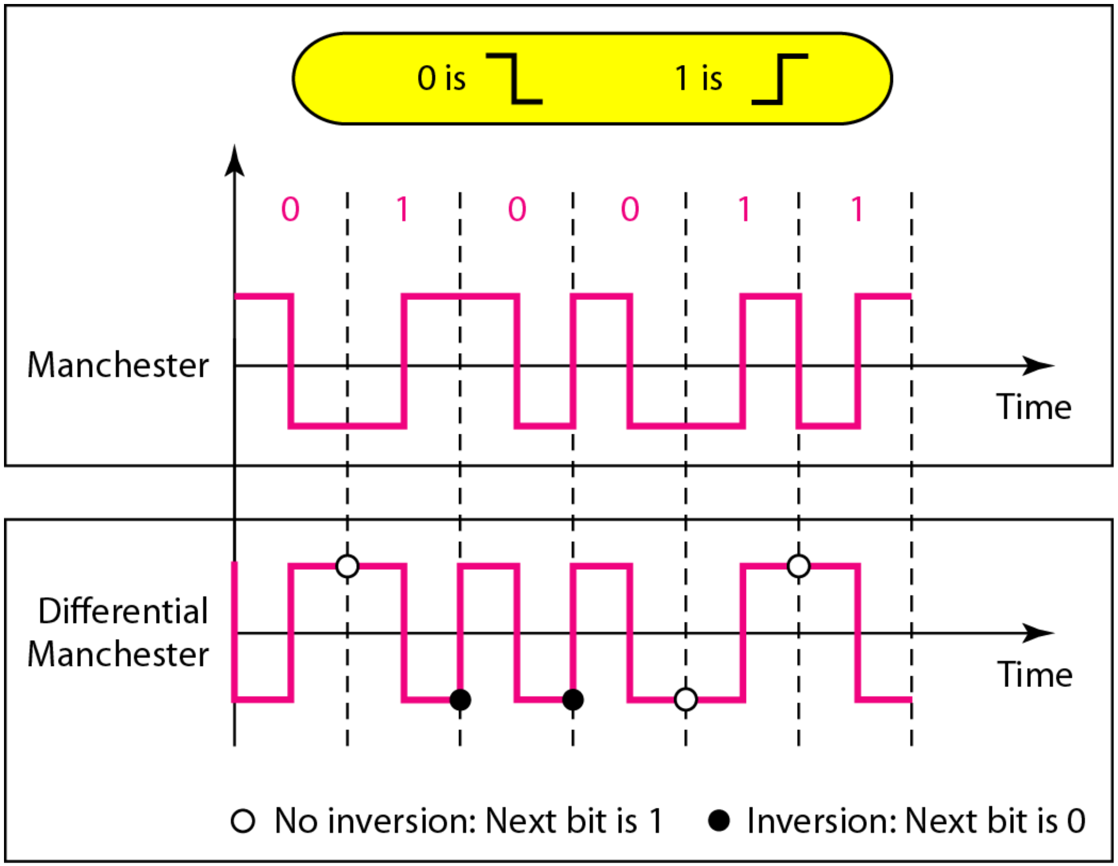
Figure 4.7 *Polar RZ scheme*

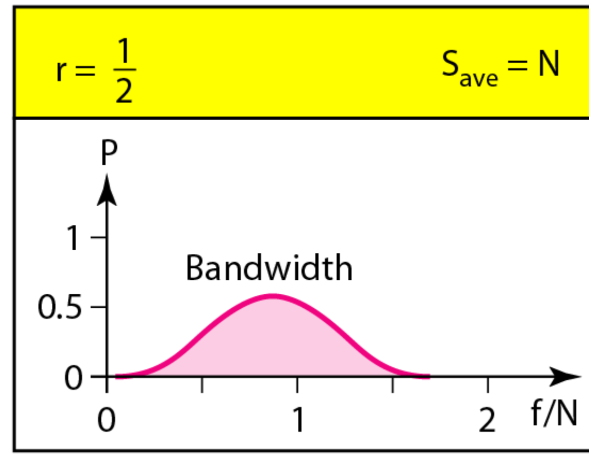


Polar - Biphase: Manchester and Differential Manchester

- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*





Note

In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

Note

The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ. The is no DC component. None of these codes has error detection.