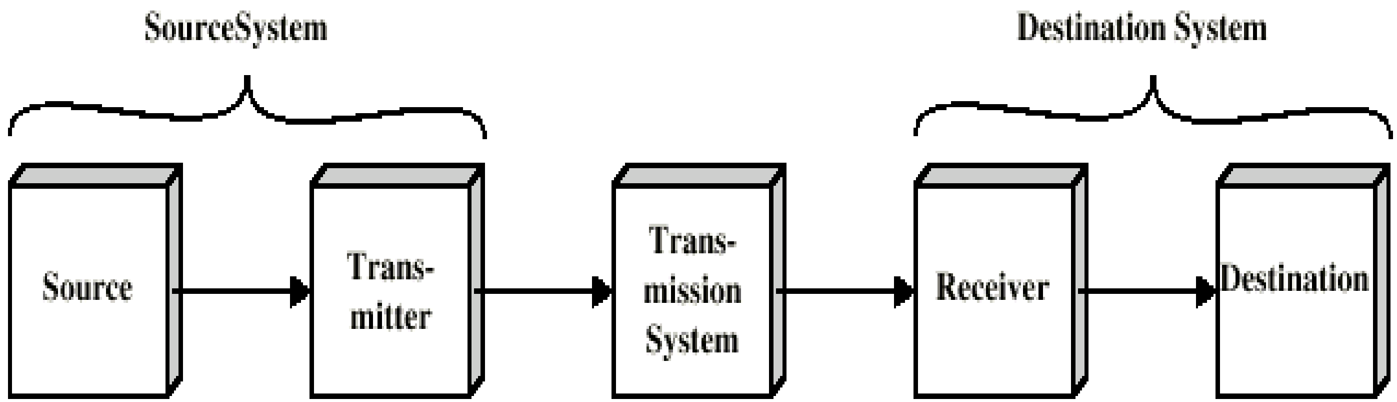
Data Communication

- The word data refers to information presented in whatever form is agreed upon by the parties creating and using the data.
- E.g. text, audio, video, image etc.
- The term telecommunication means communication at a distance.
- E.g. telephony, telegraphy, television etc.
- Data communication: The exchange of data between two parties Via some form of transmission medium.

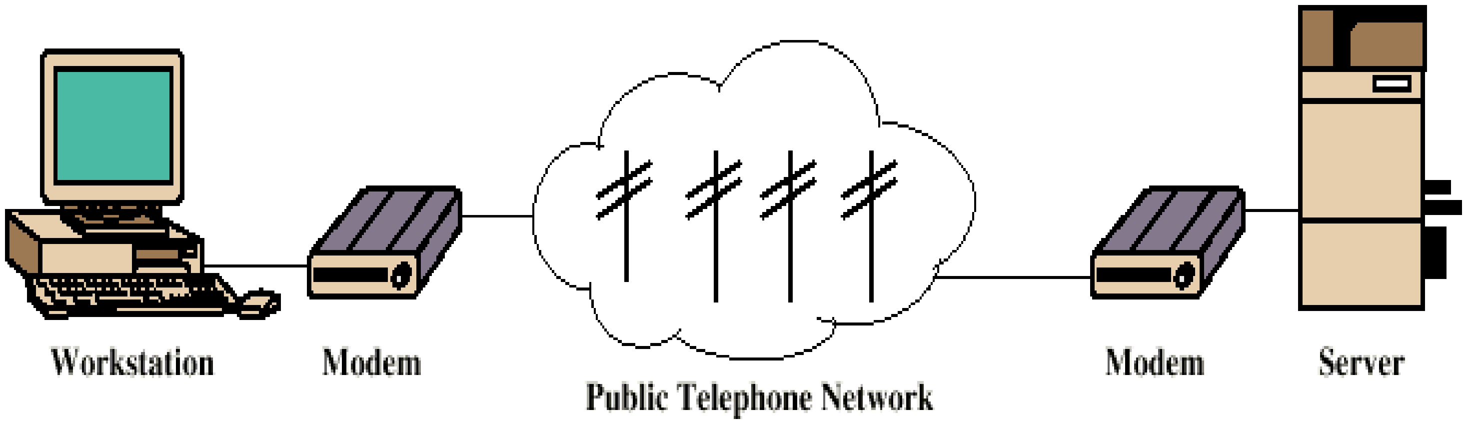
Data Communication Model

- The key elements of this model are
  - **Source** - generates data to be transmitted
  - **Transmitter** - converts data into transmittable signals
  - **Transmission System** - carries data from source to destination
  - **Receiver** - converts received signal into data
  - **Destination** - takes incoming data

Data Communication Model

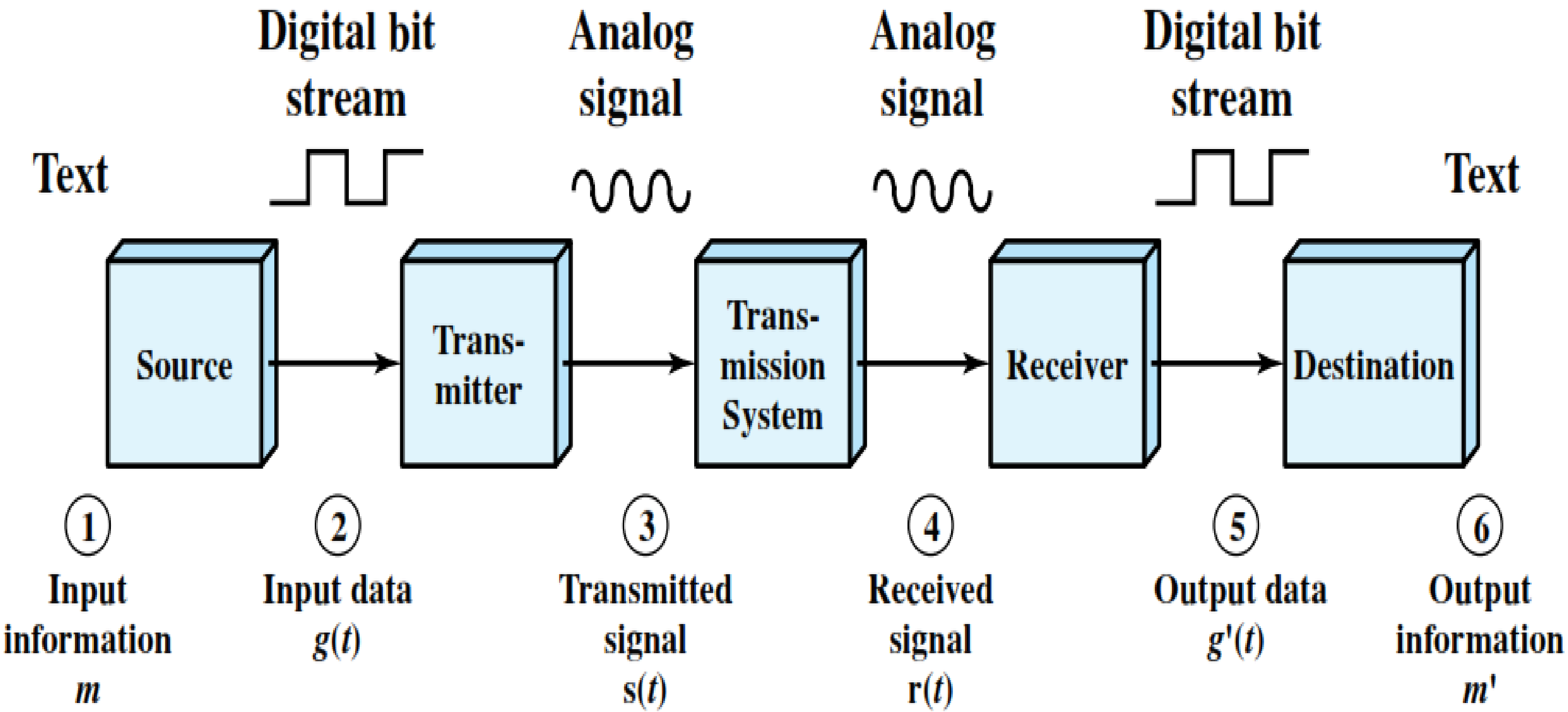


(a) General block diagram



(b) Example

Data Communication Model

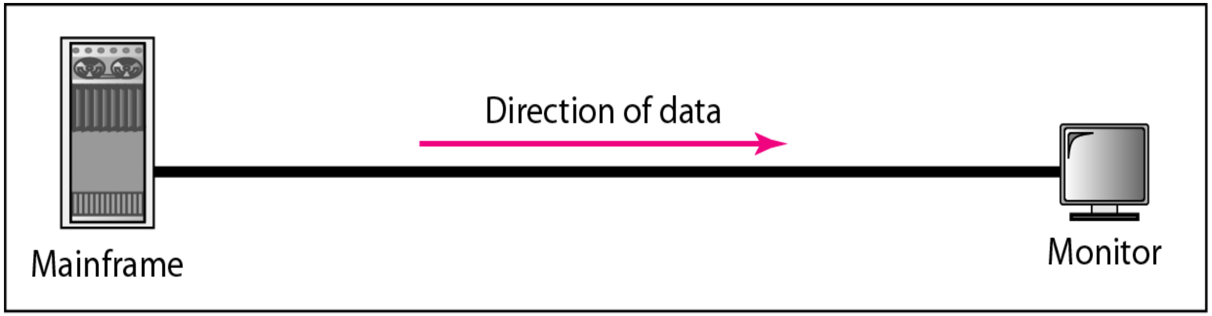


Tasks of a communication system

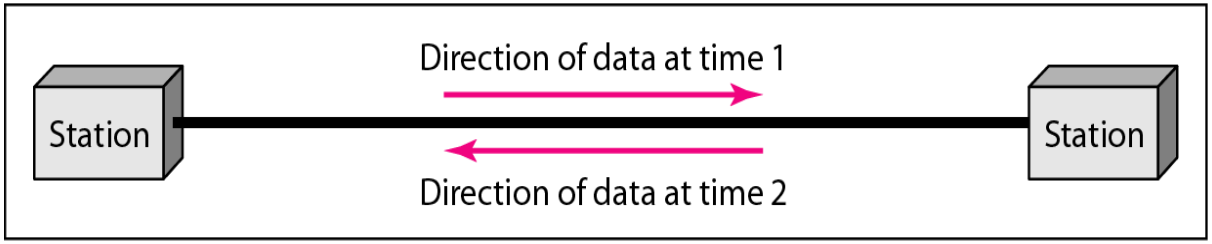
- Transmission System Utilization
- Interfacing
- Signal Generation
- Synchronization
- Exchange Management
- Error detection and correction
- Flow Control
- Addressing
- Routing
- Recovery
- Message formatting
- Security
- Network Management

Transmission\Communication Mode

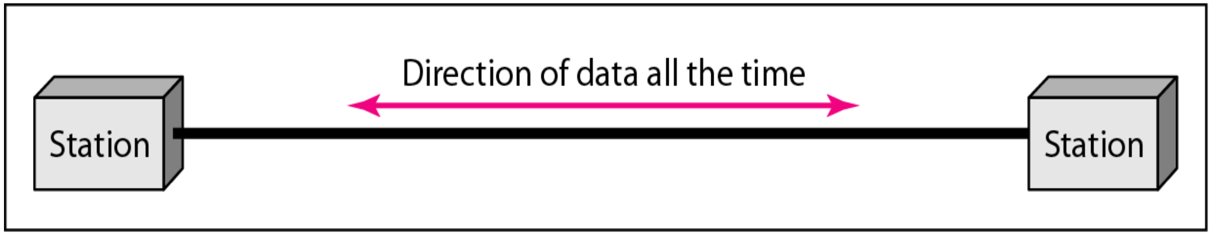
- Simplex
- Half-duplex
- Full-duplex



a. Simplex



b. Half-duplex



c. Full-duplex

Networks

- A network is a set of devices (often referred to as nodes) connected by communication links.
- A node can be a computer, printer, or any other device capable of sending and/or receiving data.
- A link can be a cable, air, optical fiber, or any medium which can transport a signal carrying information.

Network Criteria

- Performance
  - Depends on Network Elements
    - Number of users
    - Transmission Medium (capacity)
    - Hardware capabilities
    - Software efficiency
  - Measured in terms of Delay and Throughput

Network Criteria

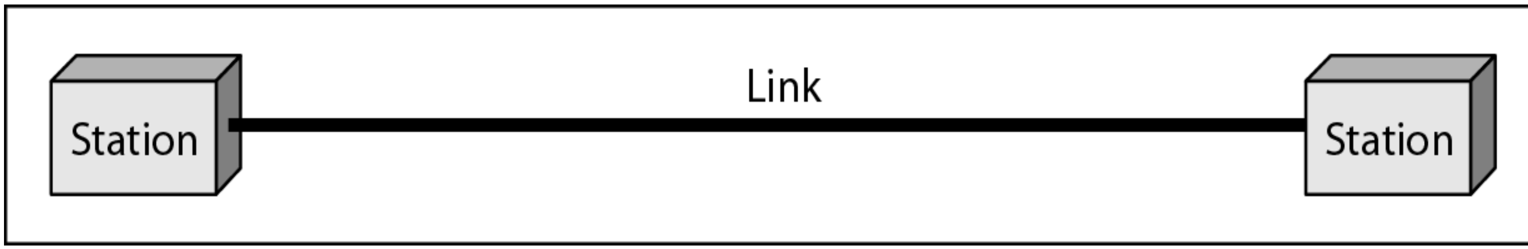
- Reliability
  - Failure rate of network components
  - Measured in terms of availability/robustness
- Security
  - Data protection against corruption/loss of data due to:
    - Errors
    - Malicious users

Physical Structure of Network

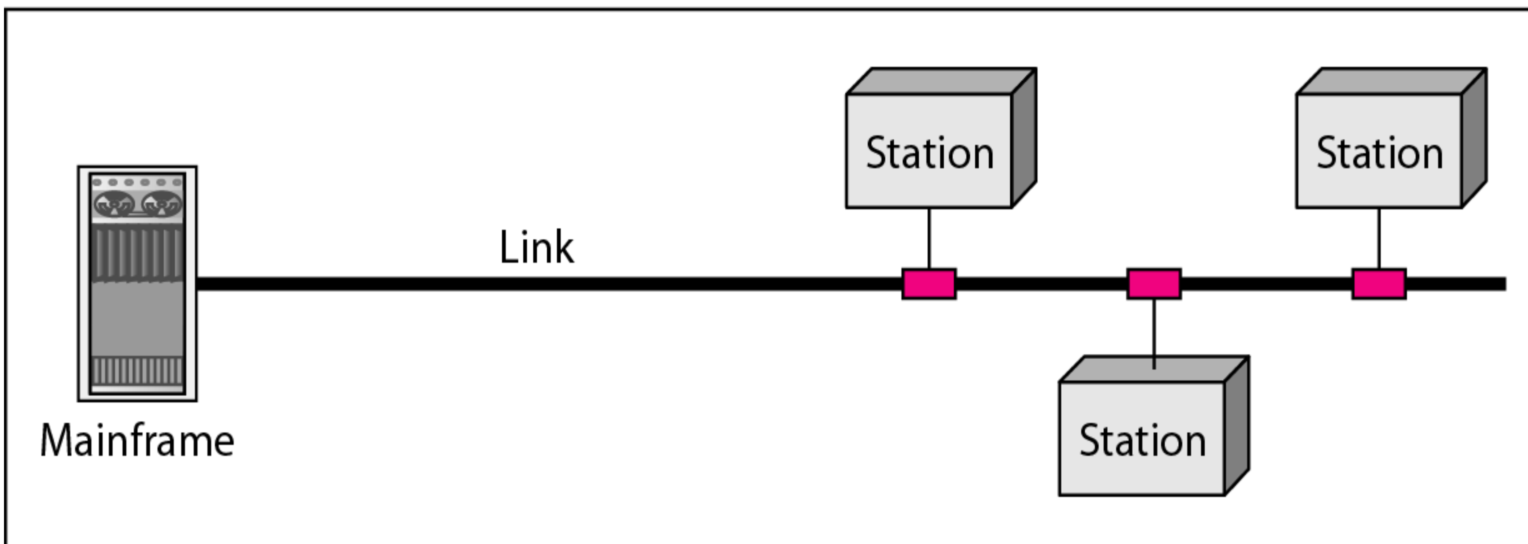
- Type of connections
  - Point to Point - single transmitter and receiver
  - Multipoint - multiple recipients of single transmission
    - Spatially shared connections
    - Timeshared connections
- Physical Topology
  - A geometric representation of all the links and nodes

Physical Structure of Network

- Type of connections

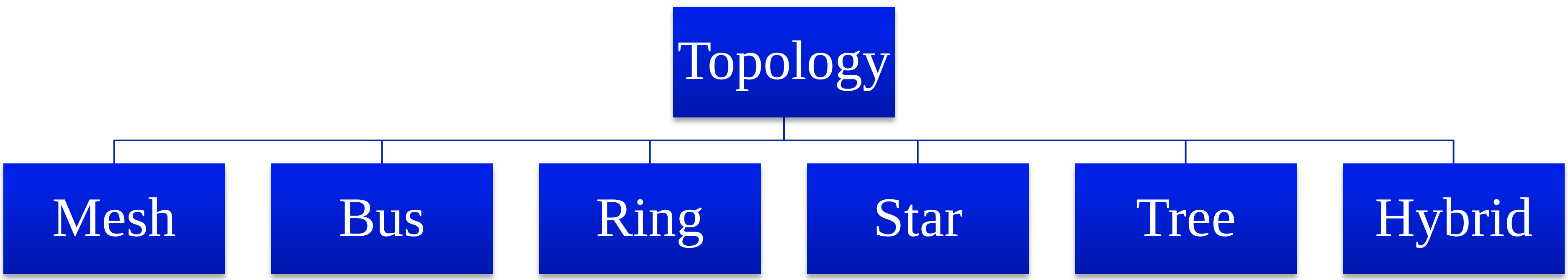


a. Point-to-point



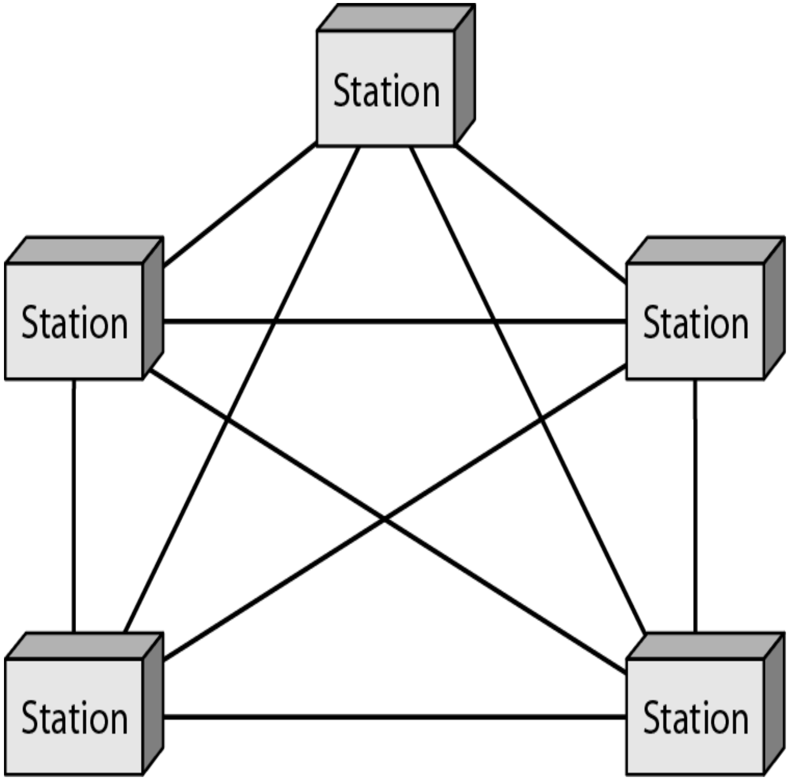
b. Multipoint

Physical Topology (Network Topology)



Mesh Topology

- Every device has a dedicated point-to-point link to every other device.
- It does not contain the switch, hub or any central computer which acts as a central point of communication.

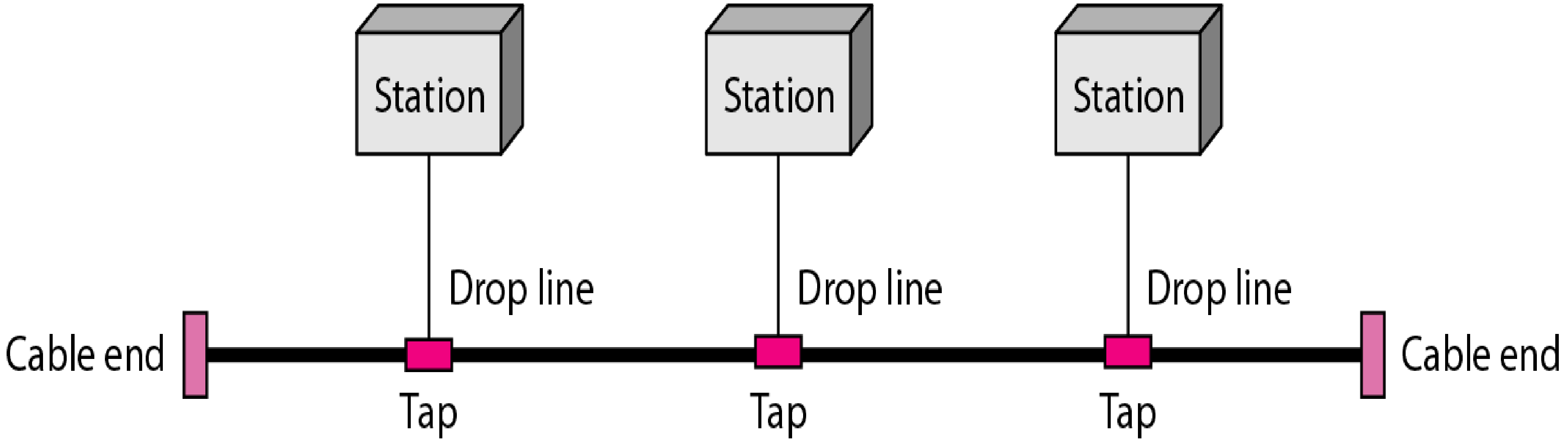


Mesh Topology

- Advantages
  - Reliable and Robust
  - Easier Reconfiguration
  - Secure
  - Fault detection is easy
- Disadvantages
  - Cost is high
  - very difficult to maintain and manage.
  - Wires requires more space

Bus Topology

- One long cable acts as a backbone to link all other devices.
- Nodes connect with backbone by drop lines and taps.
- It has multipoint connections.
- A terminator is required at each end to absorb the signal so it does not reflect back across the bus.

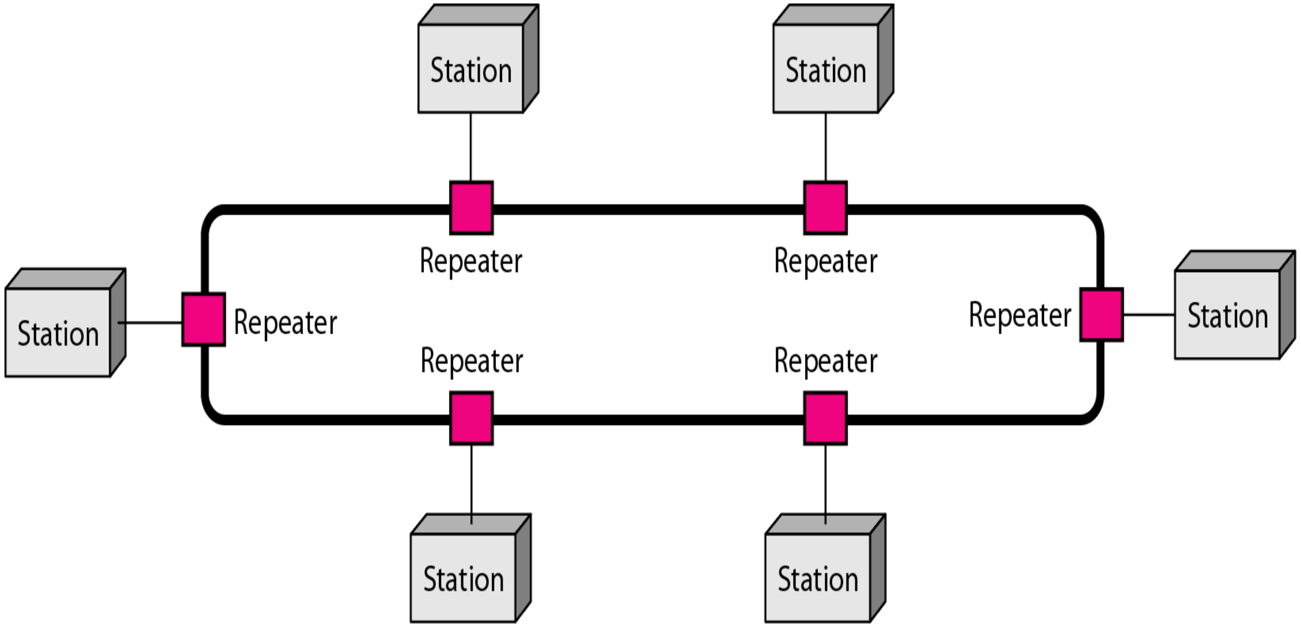


Bus Topology

- Advantages
  - Easy to implement and extend.
  - Cheaper than other topologies.
- Disadvantages
  - Limited cable length and number of stations.
  - If there is a problem with the cable, the entire network goes down.
  - Degraded Performance
  - It is slower than the other topologies.

Ring Topology

- Network forms a circle.
- Each device has dedicated point-to-point link with two other devices.
- Signal is transmitted unidirectional
- Each device acts as a repeater

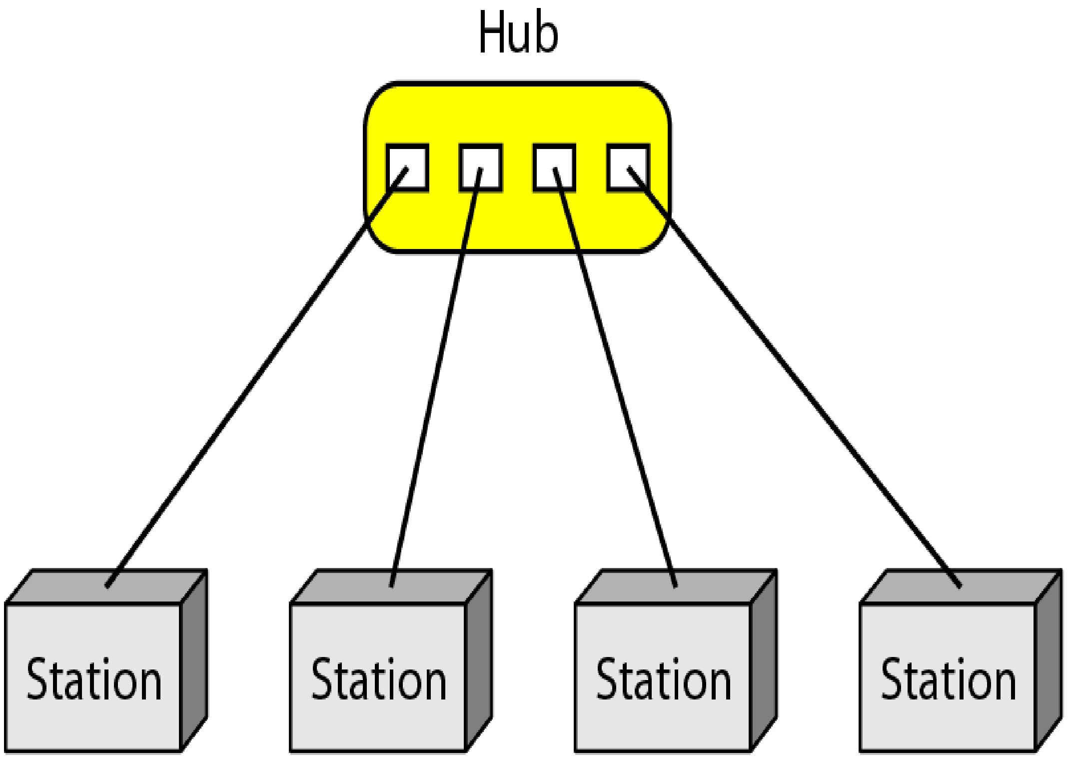


Ring Topology

- Advantages
  - Easy to install and reconfigure.
  - Less chance of packet collisions.
- Disadvantages
  - Communication delay is directly proportional to the number of nodes.
  - If one workstation shuts down, it affects whole network.

Star Topology

- Each device has a dedicated point-to-point link only to a central controller (hub).

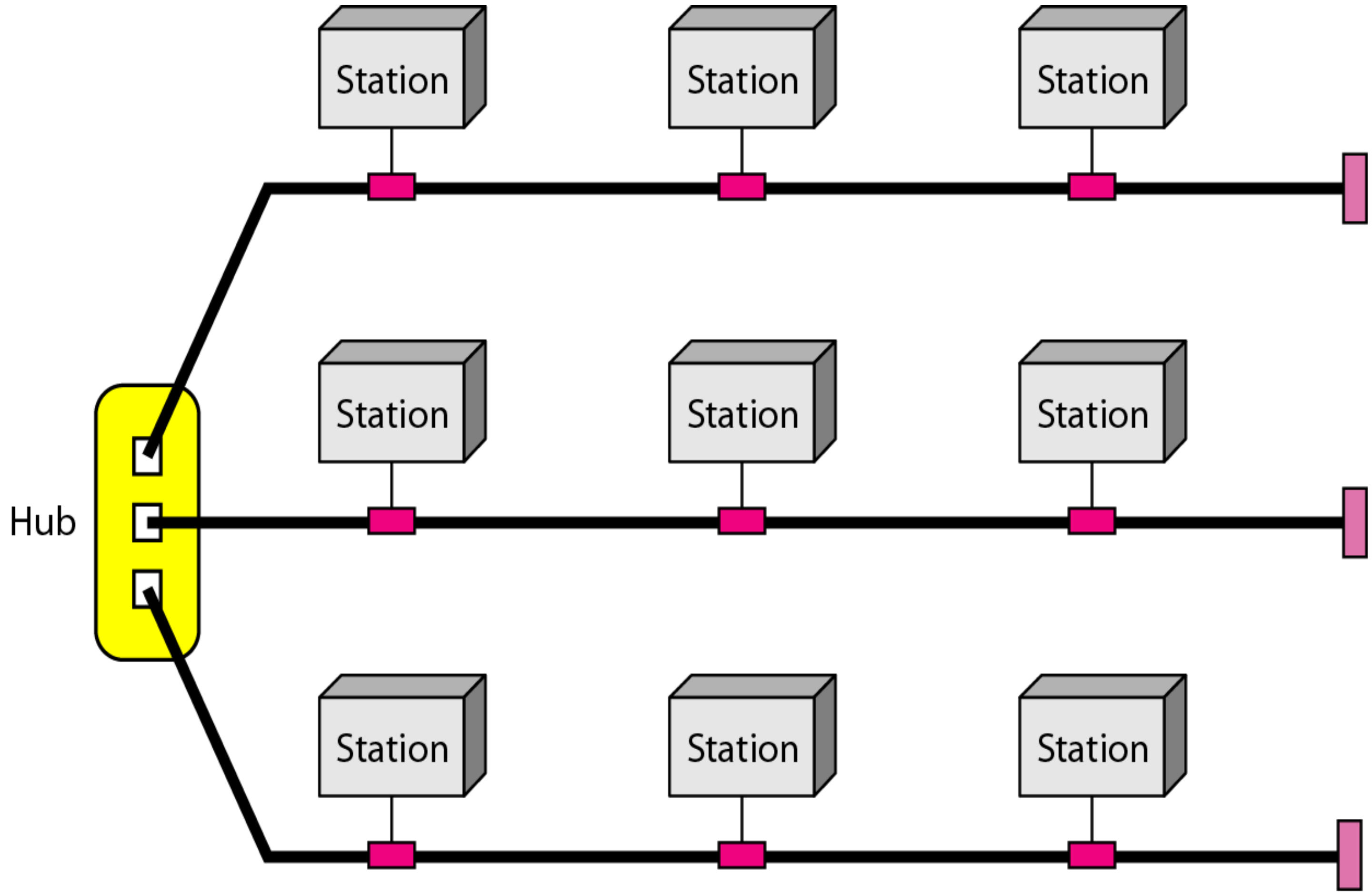


Star Topology

- Advantages
  - Easy to install and reconfigure.
  - Low network traffic.
- Disadvantages
  - Cost is high.
  - Performance depends on the hub's capacity.
  - Single point of failure.

Hybrid Topology

- Combination of two or more topologies.



Types of Computer Network

- Local Area Network (LAN)
  - LAN is a small size computer networks that covers a school, building, office, home etc.
- Metropolitan Area Network (MAN)
  - MAN is larger than LAN and connects multiple LANs with each other through a city or town.
- Wide area network (WAN)
  - WAN connects multiple LAN or MAN
  - A WAN can cover country, continent or even a whole world.

Review Questions

- For n devices in a network, what is the number of cable links required for a mesh, ring, bus, and star topology?
- What are the two types of line configuration?
- Categorize the four basic topologies in terms of line configuration.
- What are some of the factors that determine whether a communication system is a LAN or WAN?

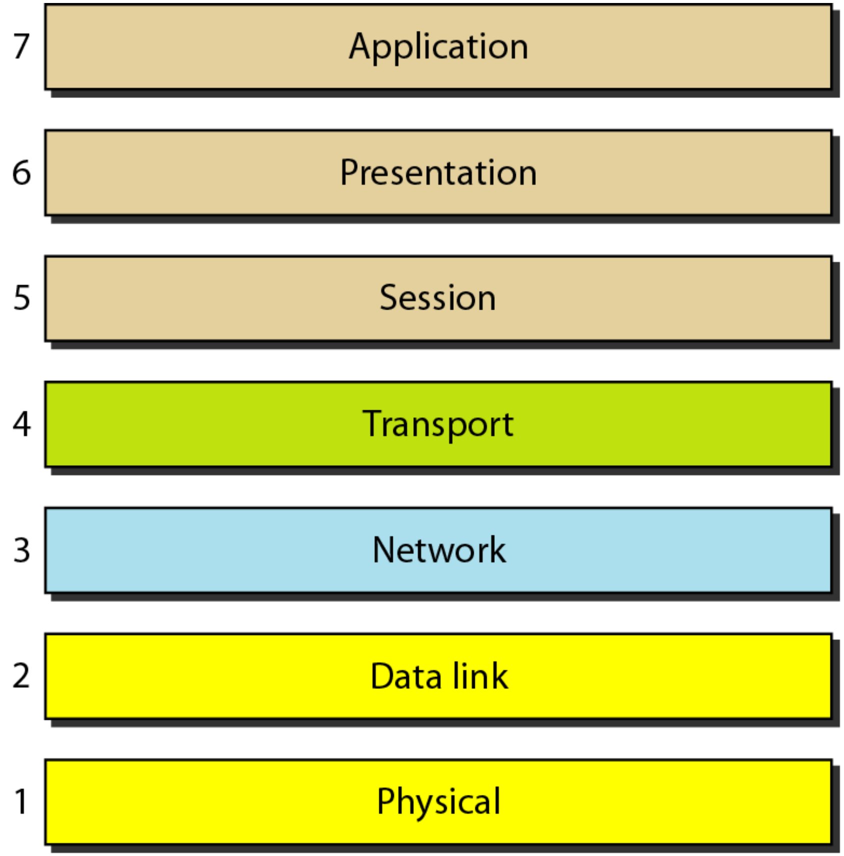
Network Models

Network Models

- **Network architectures:** Represented as a set of layers & protocols
- **Purpose of layered approach:**
  - To divide a complex task into smaller and simpler tasks
  - To ensure independence of layers, so that each layer can be changed or modified without affecting other layers
  - Each layer can be analyzed and tested independently
- **Principles of protocol layering:**
  - Each layer should be able to perform two opposite tasks. For example, encryption & decryption at presentation layer (in OSI model)
  - The two objects under each layer at both the ends should be identical. For example, at the top most layer objects should be message (eg. plain text, or image, or audio, or video etc)

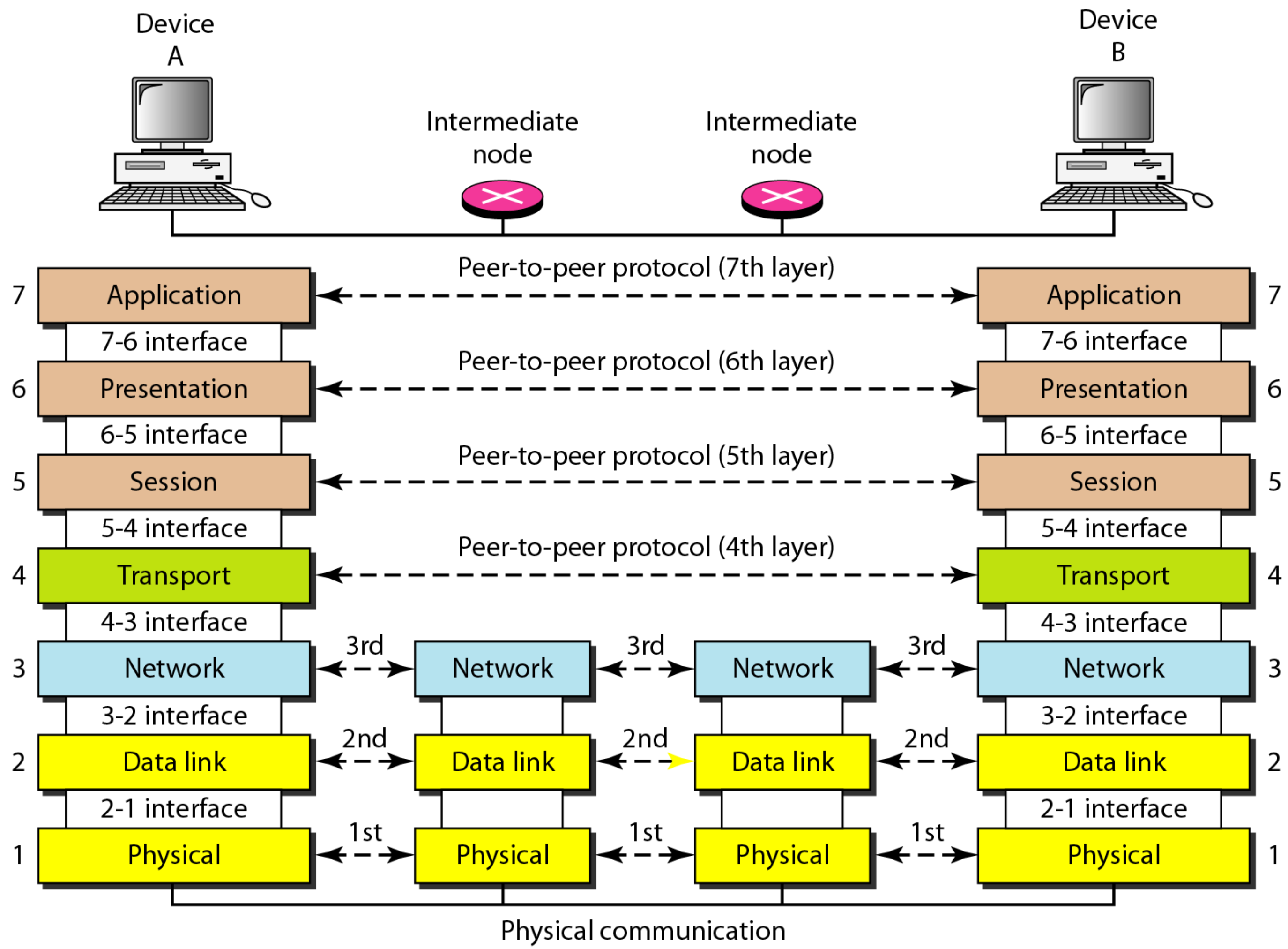
Open Systems Interconnection (OSI) Model

- OSI model is proposed by the ISO (International Standards Organization)
- It has seven layers



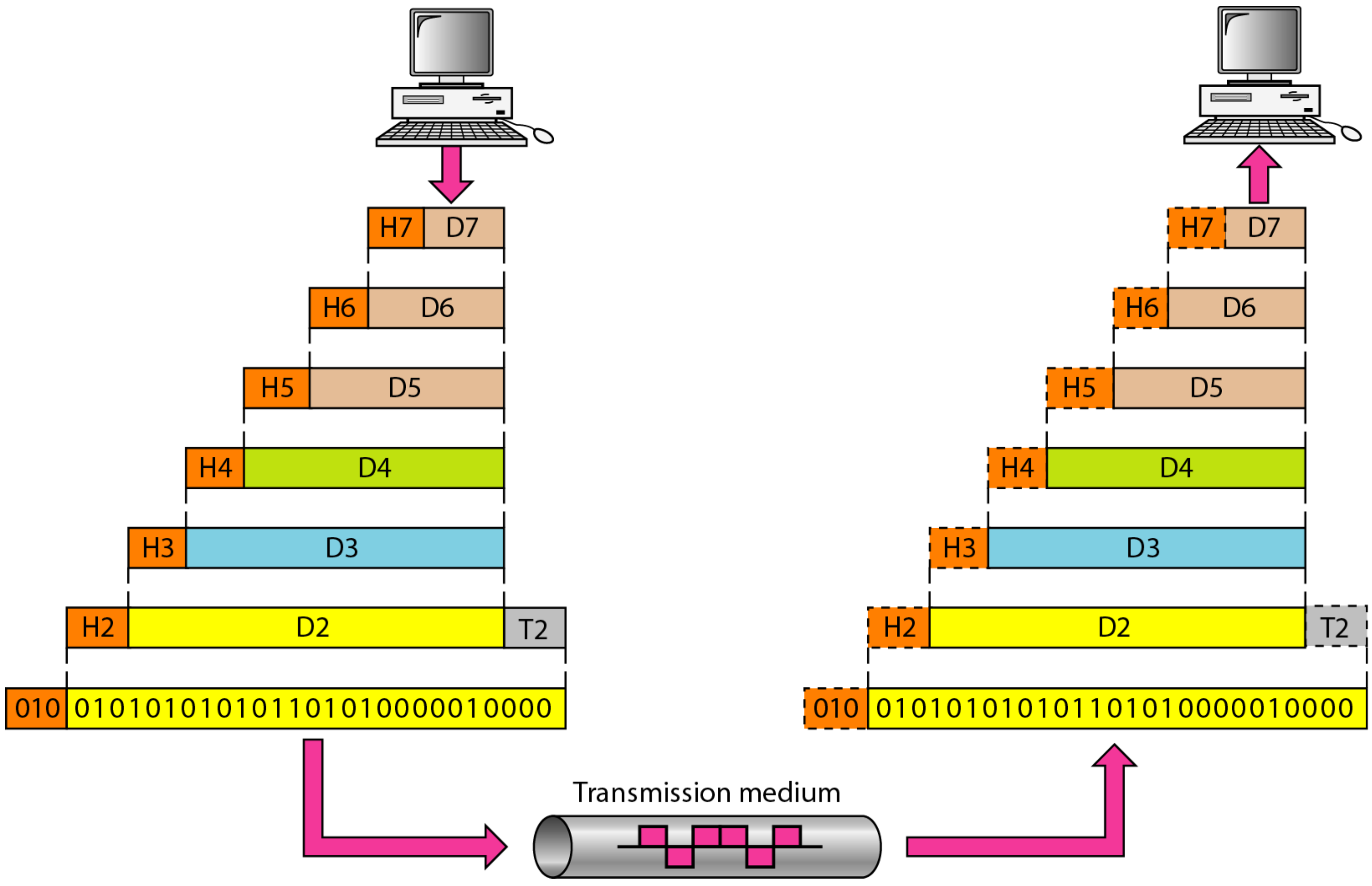
Open Systems Interconnection (OSI) Model

- The interaction between layers in the OSI model



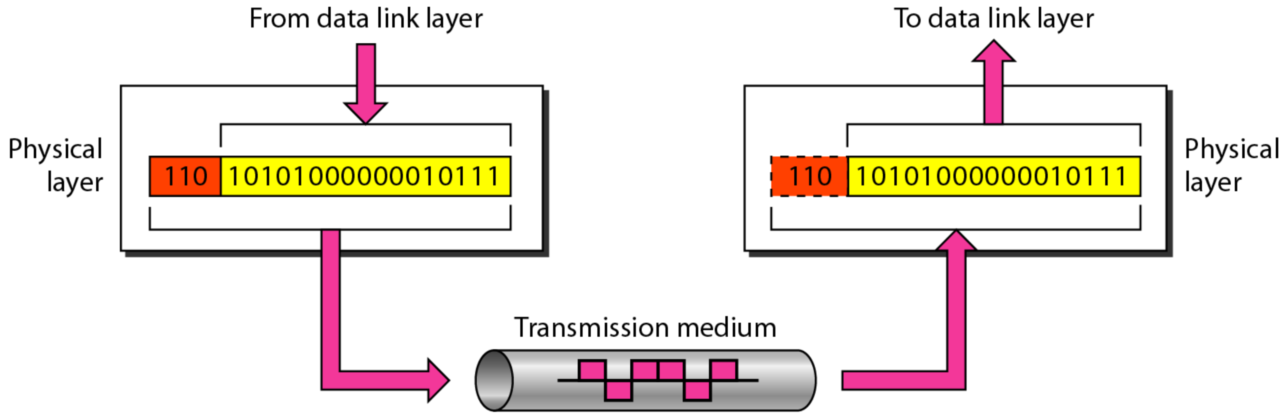
Open Systems Interconnection (OSI) Model

- An exchange using the OSI model



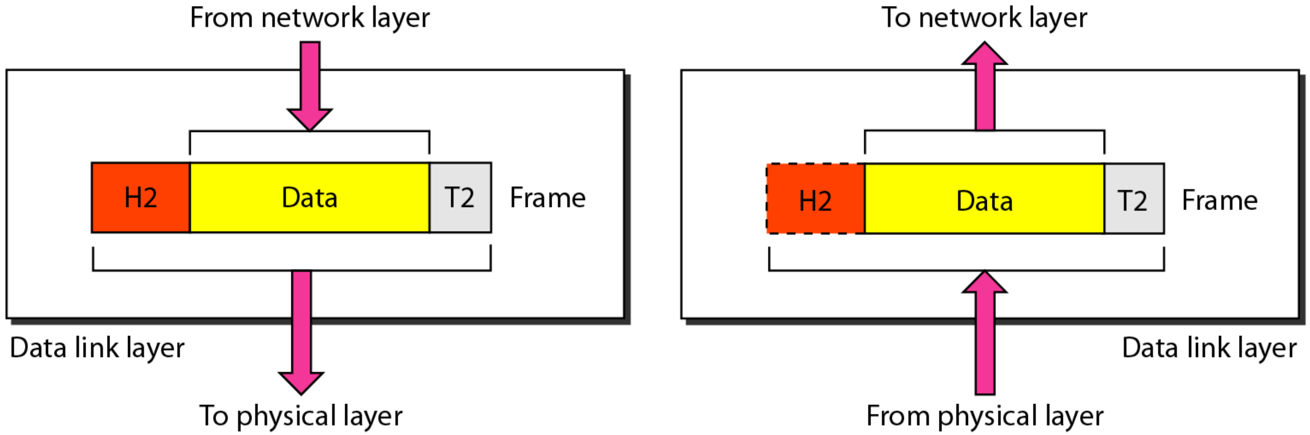
OSI Model

- **Physical layer:**
  - Controls the transmission of actual data across the link
  - At sender end, converts bits into signals (**encoding**)
  - At receiver end, converts signals into bit stream
  - Transmission rate
  - Synchronization of bits
  - Transmission mode
  - Line configuration



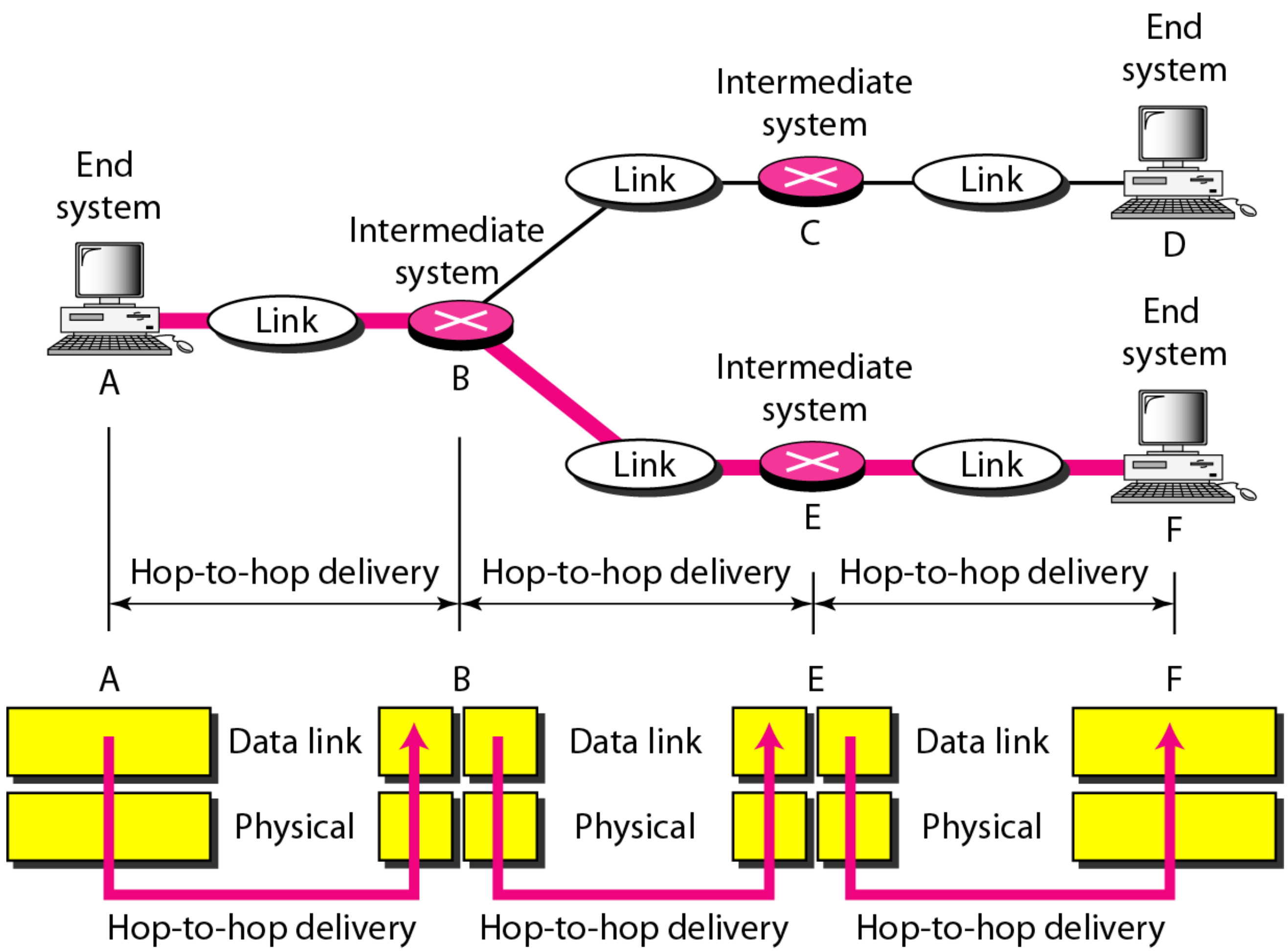
OSI Model

- **Data link layer:**
  - It is responsible for **physical addressing** via MAC addresses
  - MAC addresses are 48 bit addresses printed on NIC (network interface card). For example, 07:01:02:2C:4B:B7 (12 hexadecimal digits)
  - **Flow Control:** It controls how data are placed & received from the media (hop-to-hop)
  - Performs **media access control** using CSMA/CD
  - Performs **error detection & correction**: For that, this layer adds control data as a trailer at the end of the data packet (hop-to-hop)



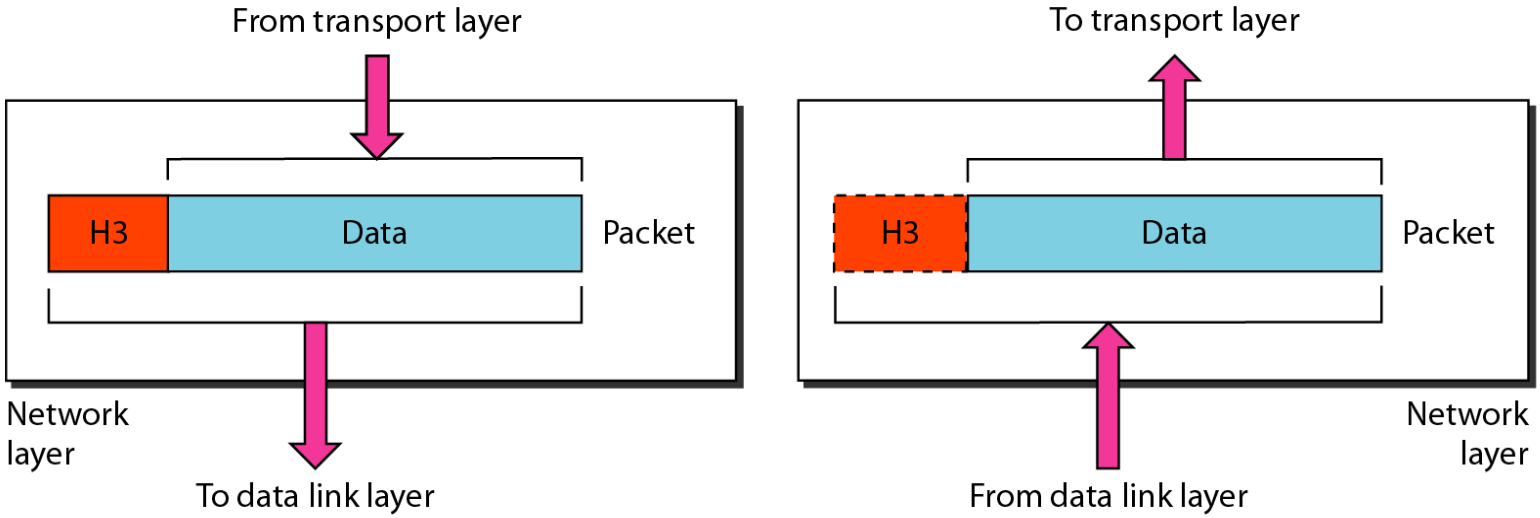
OSI Model

- **Data link layer**



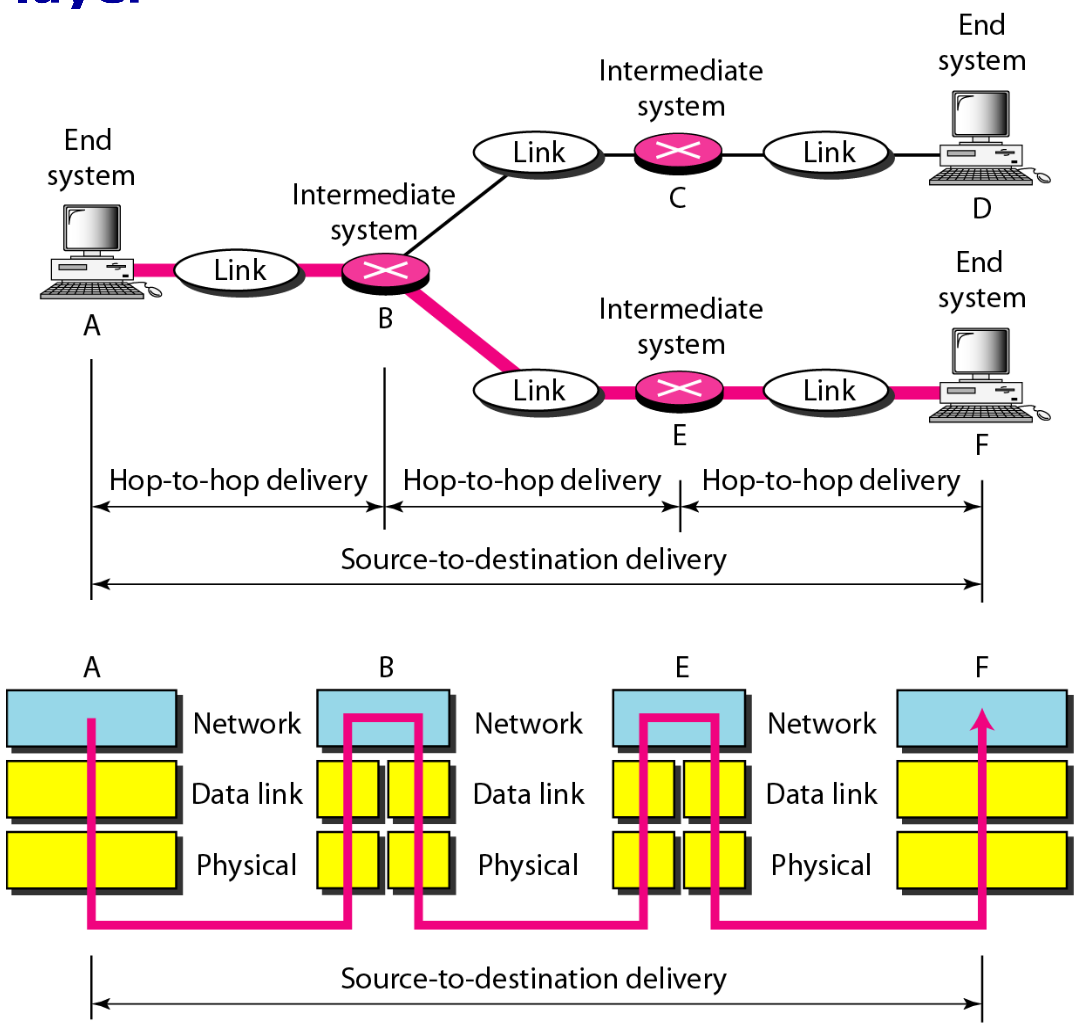
OSI Model

- **Network layer:** It is responsible for:
  - **Logical addressing** via IPv4 (32 bit address like 172.16.32.133) or IPv6 (128 bit address)
  - **Routing:** Routing is a process which is performed by network layer devices in order to deliver the datagrams by choosing an optimal path between sender and receiver



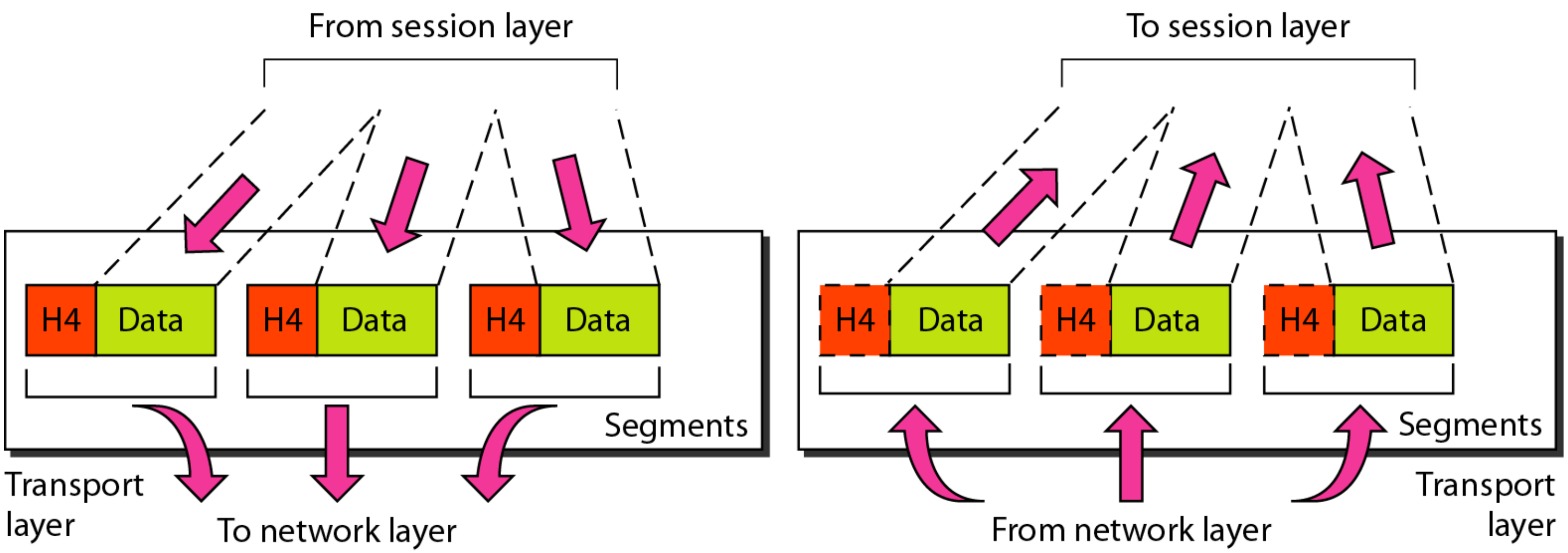
OSI Model

- **Network layer**



OSI Model

- **Transport layer:** It is responsible for:
  - Giving services to the session layer
  - Process to process delivery
  - Providing end-to-end connection between hosts

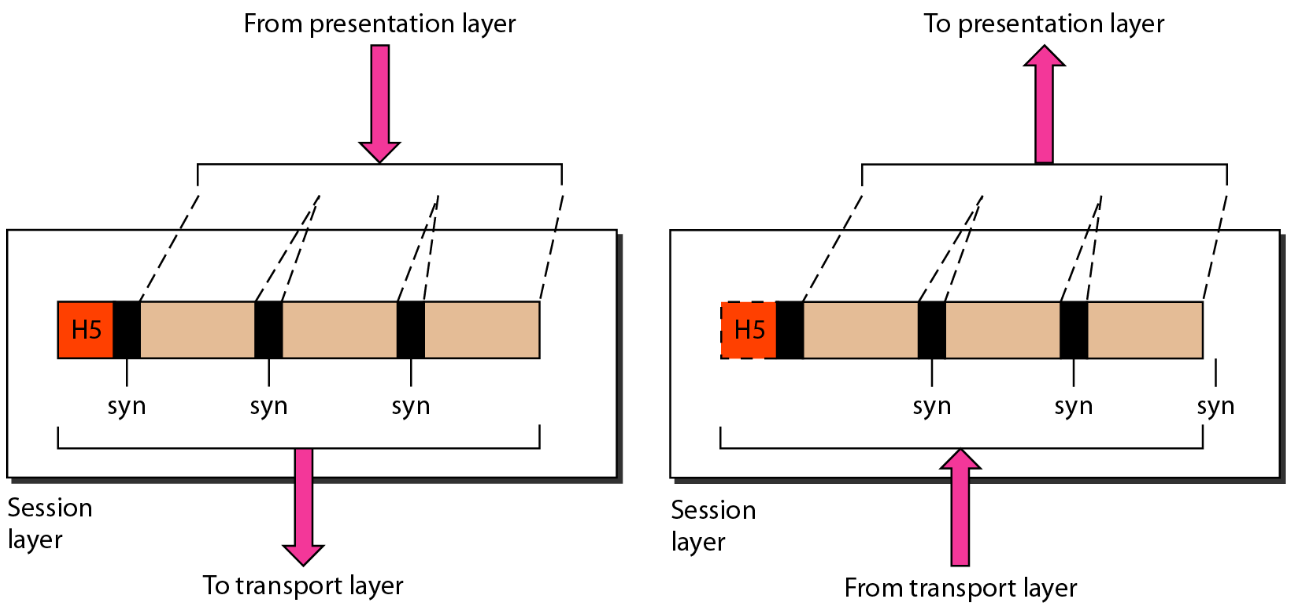


OSI Model

- **Transport layer:** (Contd.)
  - Service point addressing (e.g. port number)
  - Error control (through retransmission) (end-to-end)
  - Segmentation and reassembly
  - Flow Control (end-to-end)
  - Connection Control
    - Provides connectionless or connection-oriented services
    - TCP: Connection-oriented. For example, e-mail, www, ftp etc.
    - UDP: Connectionless; faster than TCP. For example, video streaming, games etc.

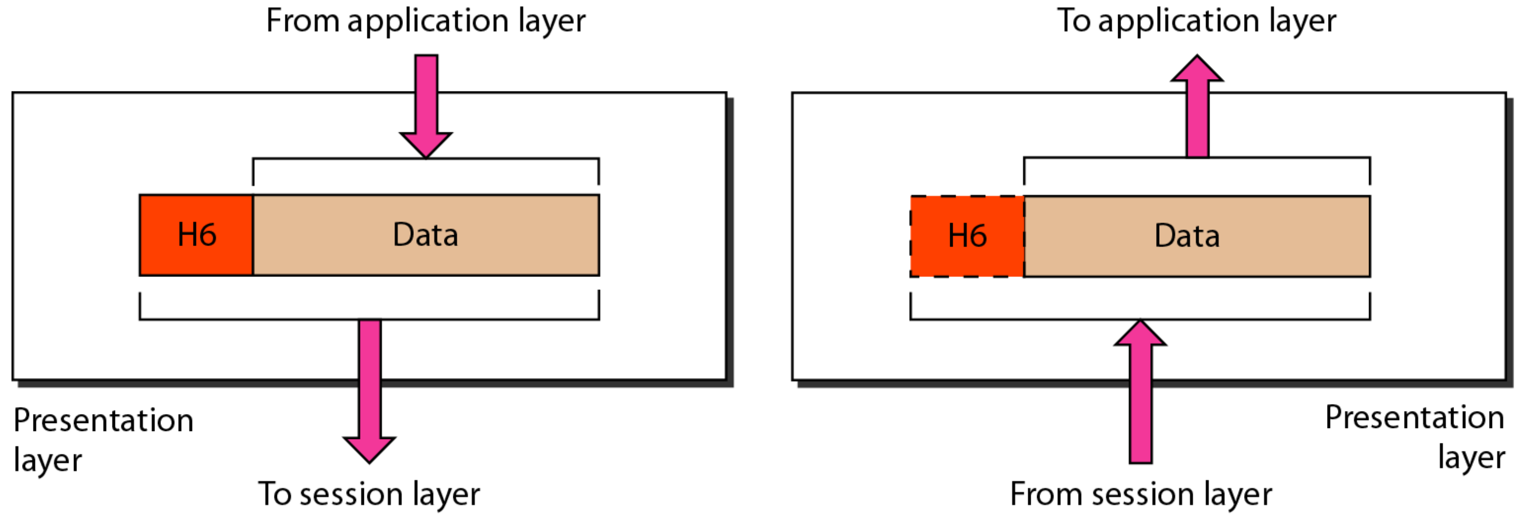
OSI Model

- **Session layer:**
  - Manages connections between two applications
    - Dialog control
    - Add checkpoints
  - Authentication is used to establish session/connection
  - Provides authorization



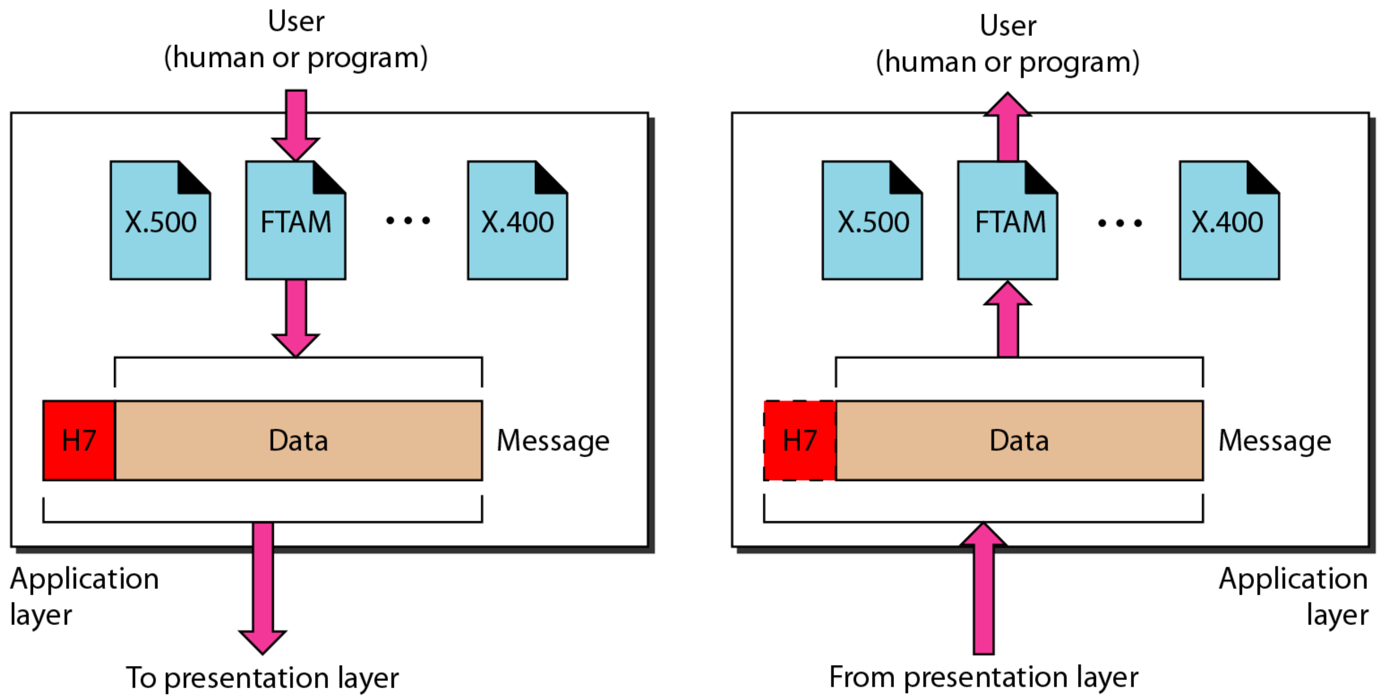
OSI Model

- **Presentation layer:**
  - Determines the format used to exchange data
  - This layer is responsible for:
    - **Translation:** For example, data → binary format
    - **Data compression:** Reduces amount of space used to store original files. It is helpful in real time video streaming
    - **Encryption/decryption**



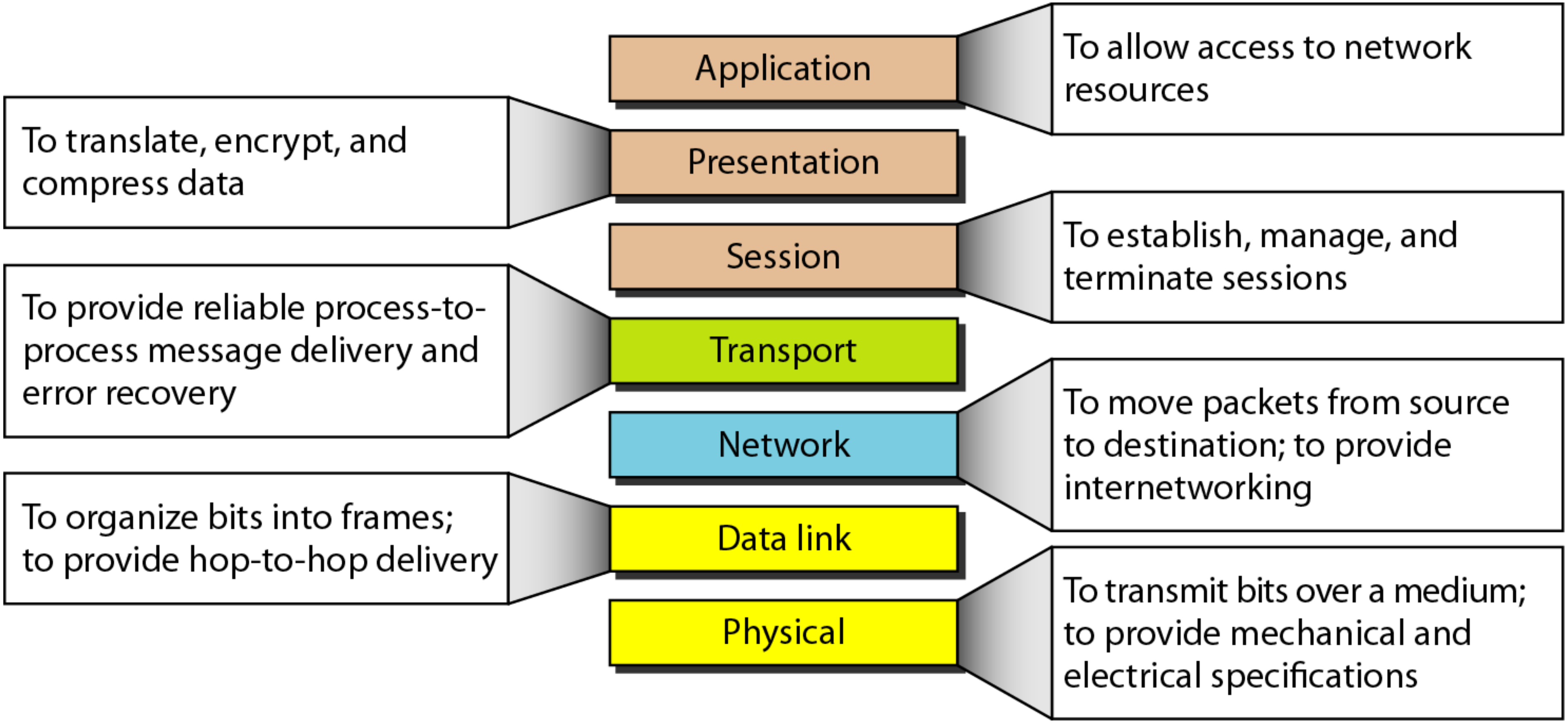
OSI Model

- **Application layer:**
  - Used by network applications. For example, Google chrome, Skype etc.
  - It is responsible for providing services to the users.
  - Protocols: HTTP (for web services), FTP (for transferring files), SMTP (for e-mail service) etc.



OSI Model

- **Summary of layers**

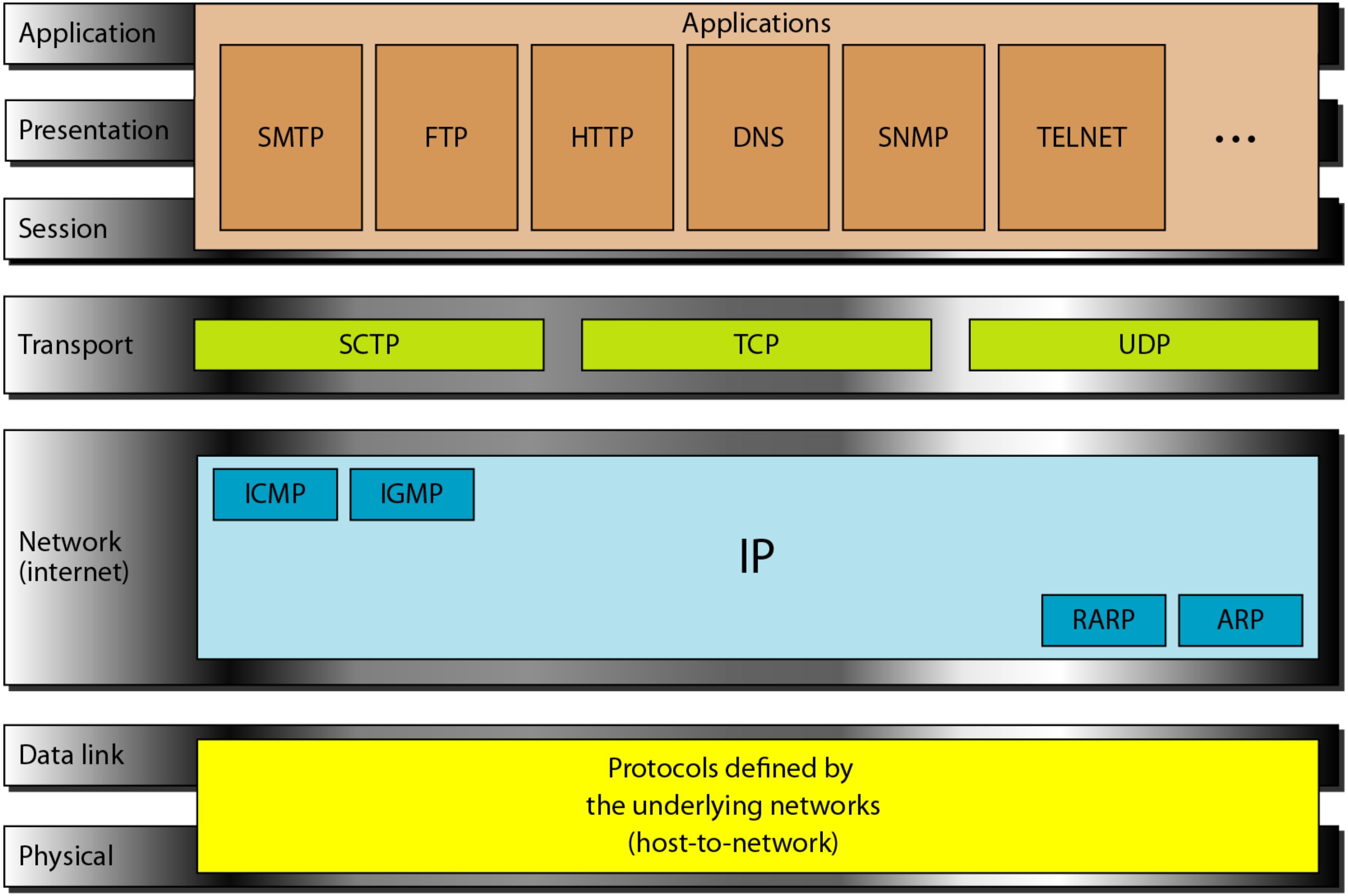


TCP/IP Protocol Suite

TCP/IP Protocol Suite

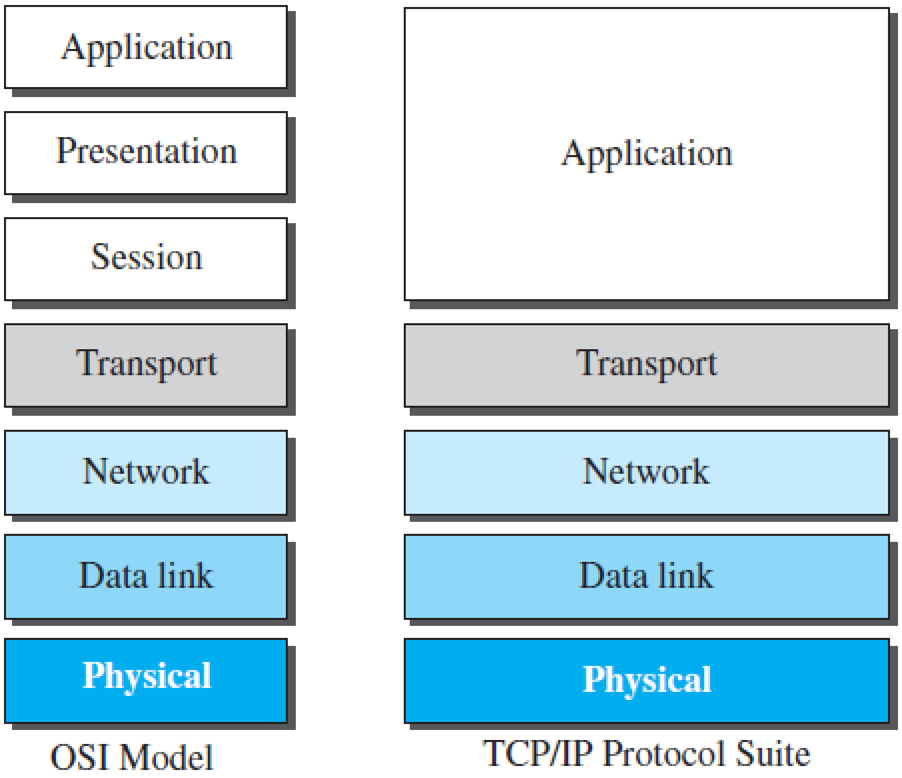
- TCP/IP (Transmission Control Protocol/Internet Protocol) is a protocol suite used in the Internet today
- Original TCP/IP had four layers
  - host-to-network
  - internet
  - transport
  - application
- It is the network model used in current Internet architecture.

OSI versus TCP/IP



OSI versus TCP/IP

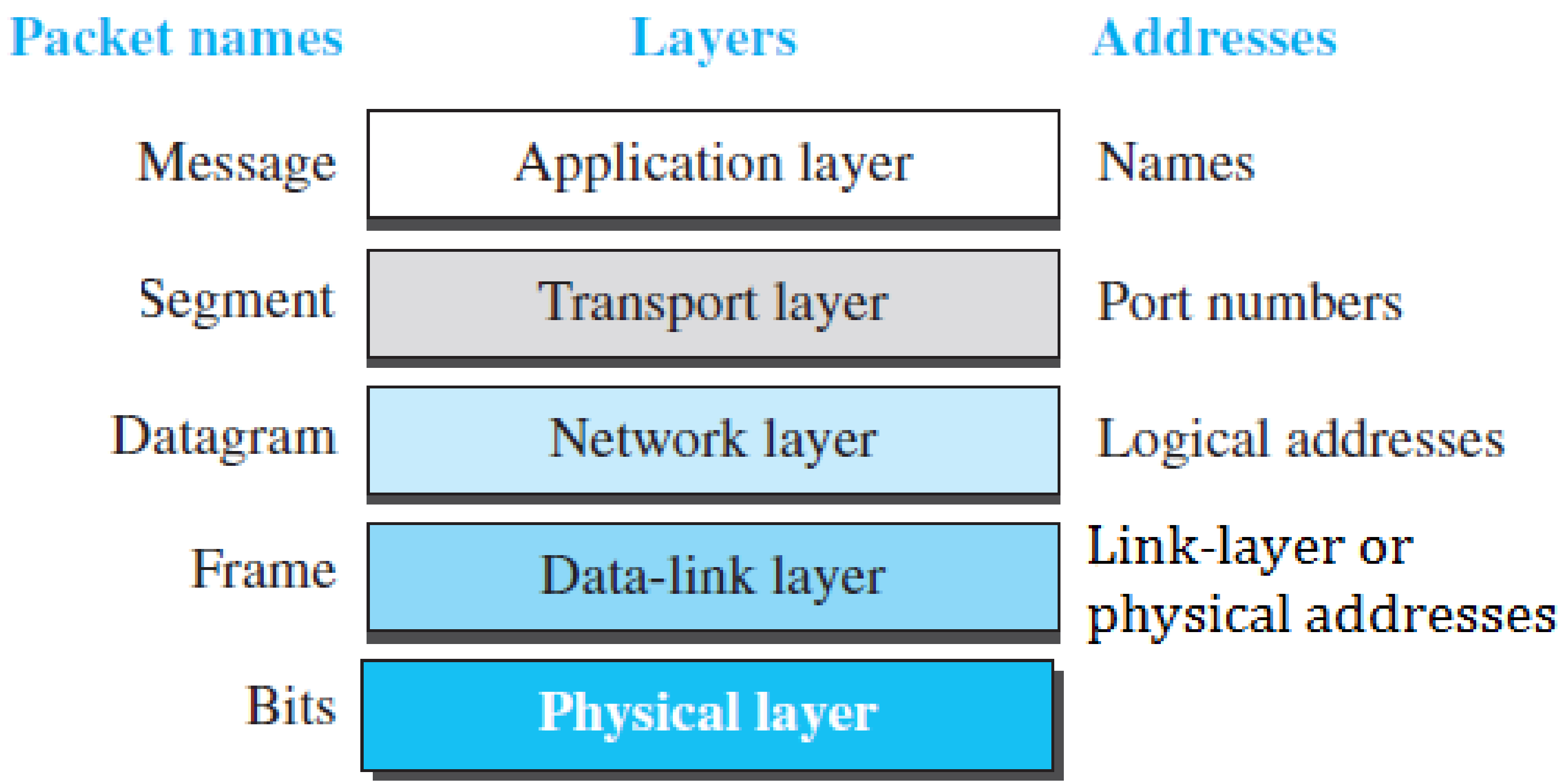
- Session and presentation layers are missing from TCP/IP
- Some of the functionalities of the session layer are available in some of the transport-layer protocols
- If some functionalities of the session and presentation layers are needed in an application, they can be included in the application software



TCP/IP and OSI model

TCP/IP Protocol Suite

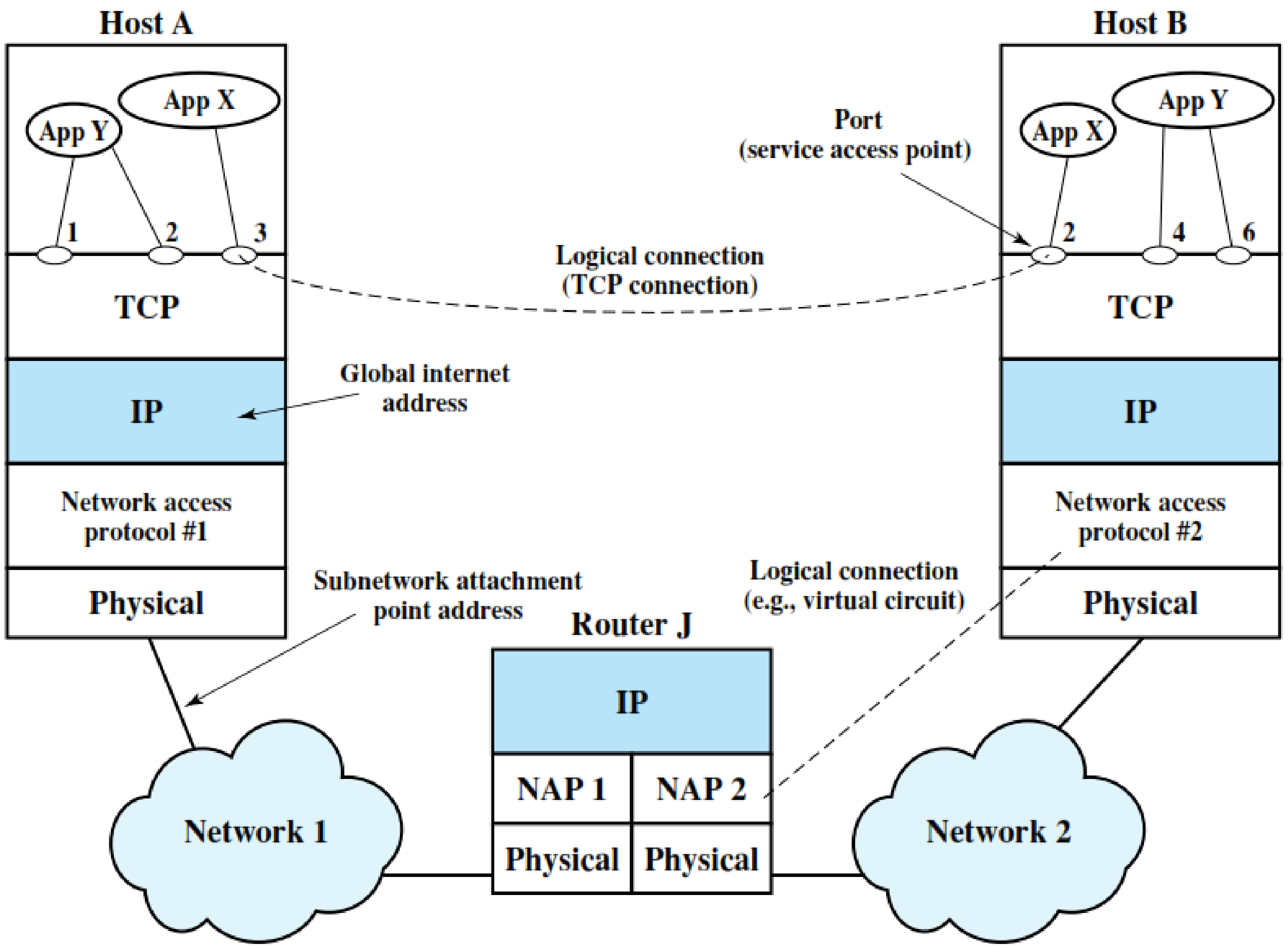
- **Addressing in the TCP/IP protocol suite:**



Addressing in the TCP/IP protocol suite

TCP/IP Protocol Suite

- **Operation of TCP and IP**



Review Questions

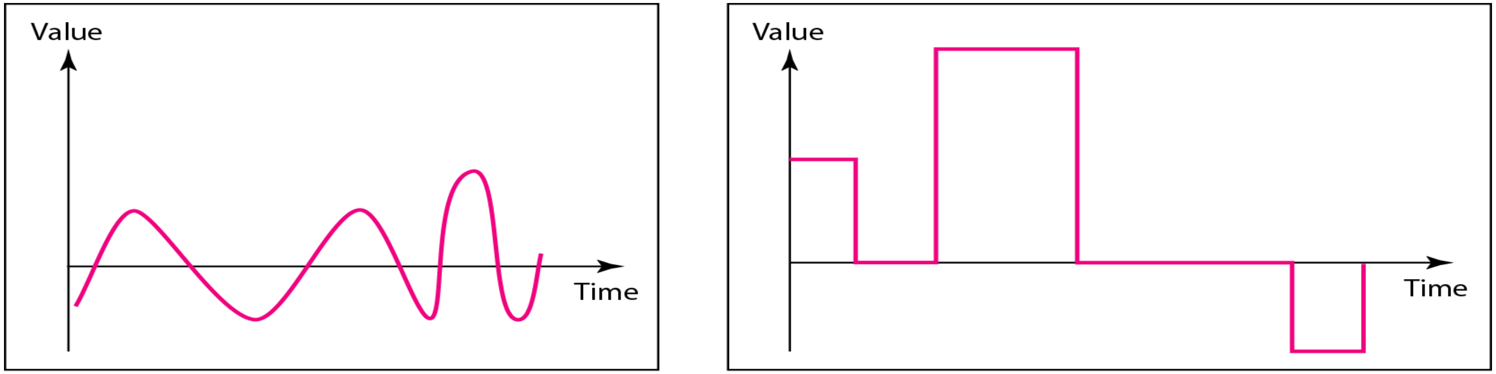
- Assume a system uses five protocol layers. If the application program creates a message of 100 bytes and each layer (including the fifth and the first) adds a header of 10 bytes to the data unit, what is the efficiency (the ratio of application layer bytes to the number of bytes transmitted) of the system?
- In an internet, we change the LAN technology to a new one. Which layers in the TCP/IP protocol suite need to be changed?
- Suppose the algorithms used to implement the operations at layer k is changed. How does this impact operations at layers k − 1 and k + 1?
- Suppose there is a change in the service (set of operations) provided by layer k. How does this impact services at layers k-1 and k+1?

Introduction to Physical Layer

Data and Signals: Analog and Digital

- Data can be analog or digital
  - Analog data are continuous and take continuous values
  - Digital data have discrete states and take discrete values
- To be transmitted, data must be transformed to electromagnetic signals
- 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
- In data communications, we commonly use

periodic analog signals and non-periodic digital signals

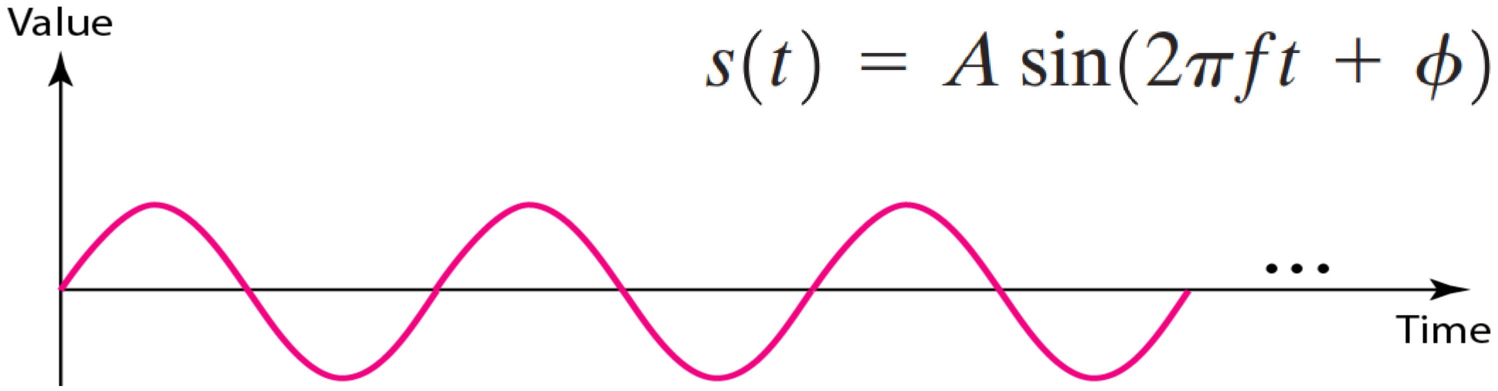


a. Analog signal b. Digital signal

Periodic Analog Signals

- A periodic signal completes a pattern within a measurable time frame, called **period**, and repeats that pattern over subsequent identical periods
- Periodic analog signals can be classified as **simple** or **composite**
- A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals
- A composite periodic analog signal is composed of multiple sine waves (will be discussed later)

$$s(t) = A\sin(2\pi f t + \phi)$$



A sine wave

Periodic Analog Signals

Period:

- The amount of time, in seconds, a signal needs to complete 1 cycle

Frequency:

- The number of cycles in one second. It is formally expressed in **Hertz (Hz)**

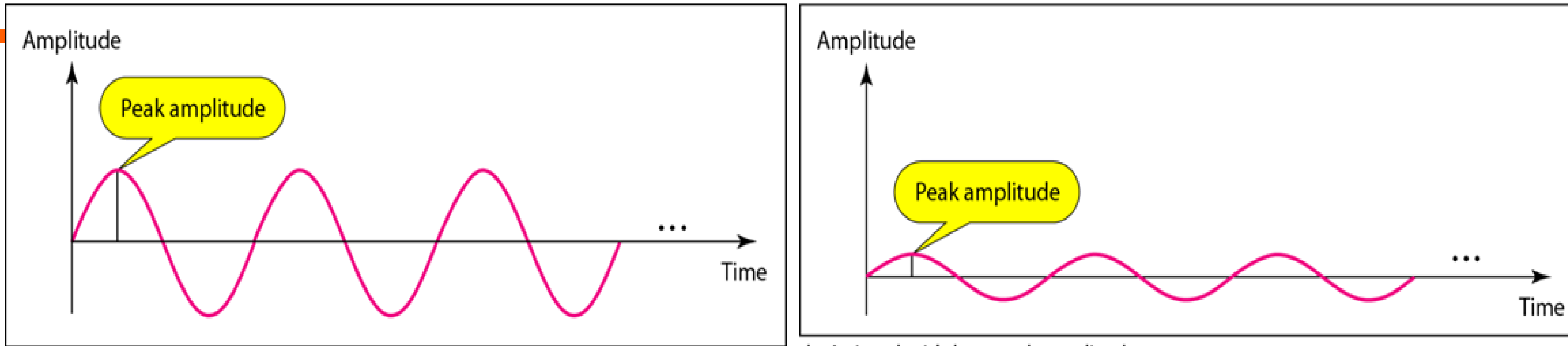
Note: Frequency and period are the inverse of each other

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

Peak amplitude:

- The peak amplitude of a signal is the absolute value of its highest intensity, proportional to the energy it carries
- It is normally measured in volts

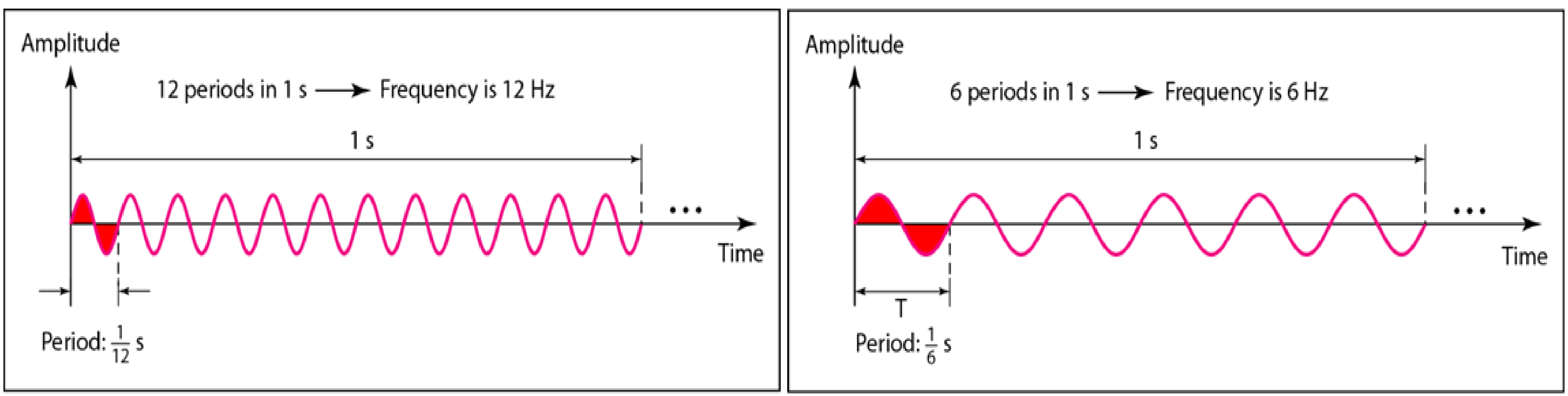
Two signals with the same phase and frequency, but different amplitudes



a. A signal with high peak amplitude

b. A signal with low peak amplitude

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 1:

The power we use at home has a frequency of 50 Hz. Find out the period of this sine wave.

$$T = \frac{1}{f} = \frac{1}{50} = .02 \text{ s} = 20 \text{ ms}$$

Example 2:

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

Solution: First we change 100 ms to seconds, and then we calculate the frequency from the period (1 Hz = 10^{-3} kHz)

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 10^{-1} \text{ s}$$

$$f = \frac{1}{T} = \frac{1}{10^{-1}} \text{ Hz} = 10 \text{ Hz} = 10 \times 10^{-3} \text{ kHz} = 10^{-2} \text{ kHz}$$

Periodic Analog Signals

Note:

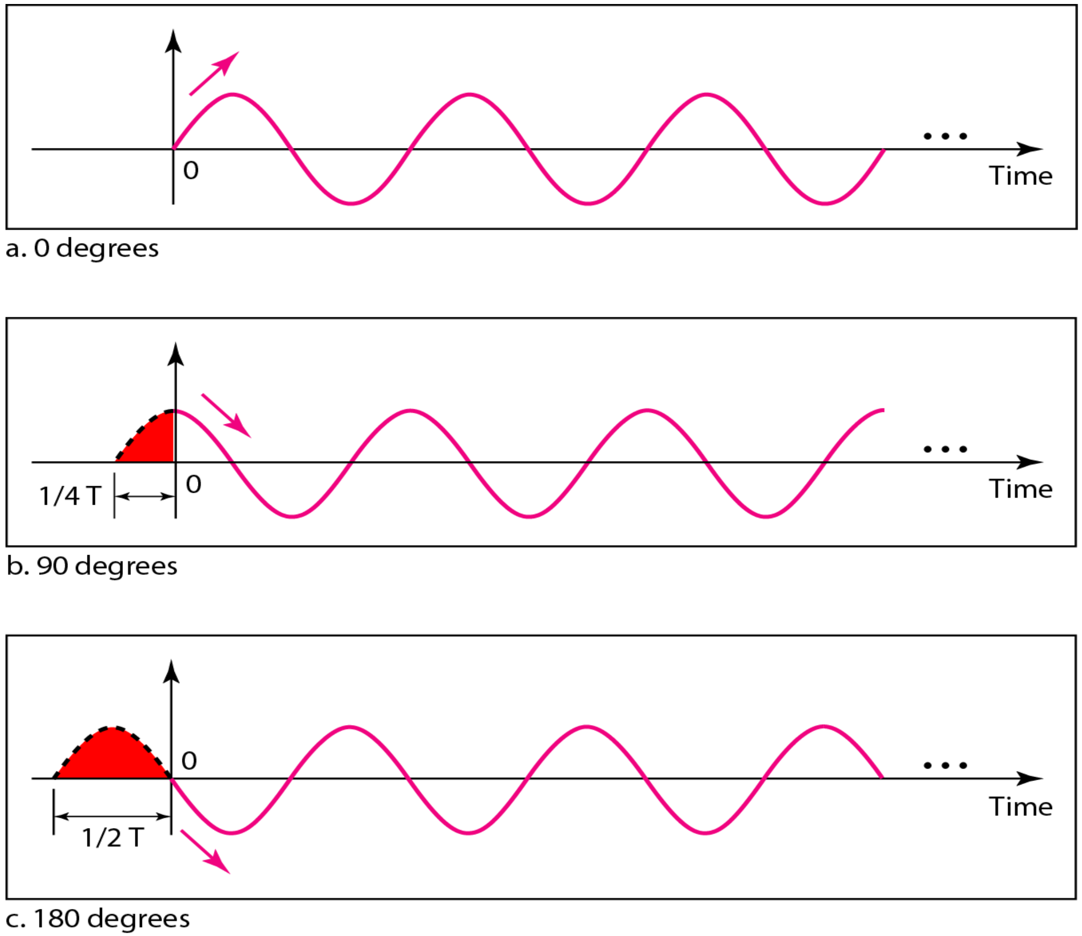
- Frequency is the rate of change with respect to time
- Change in a short span of time means high frequency
- Change over a long span of time means low frequency
- If a signal does not change at all, its frequency is zero
- If a signal changes instantaneously, its frequency is infinite

Phase:

- Phase describes the position of the waveform relative to time 0
- It is measured in degrees or radians [$360^{o} = 2\ \Pi$ rad]

Periodic Analog Signals

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



Example 3

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

Solution:

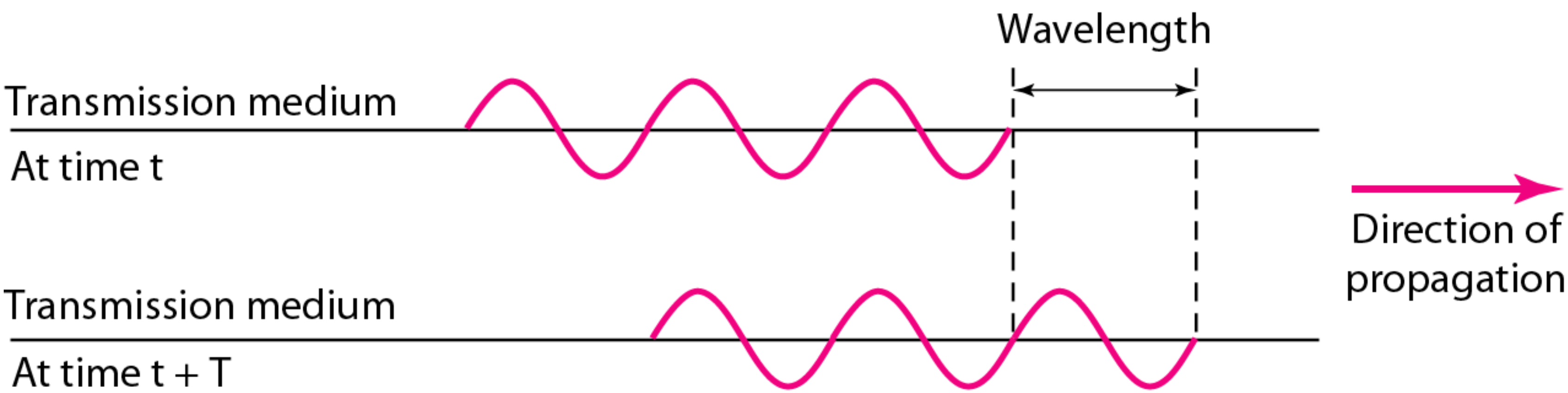
We know that 1 complete cycle is 360°. Therefore, 1/6 cycle is

$$\boxed{\frac{1}{6} \times 360 = 60^\circ = 60 \times \frac{2\pi}{360} \text{ rad} = \frac{\pi}{3} \text{ rad} = 1.046 \text{ rad}}$$

Wavelength and period

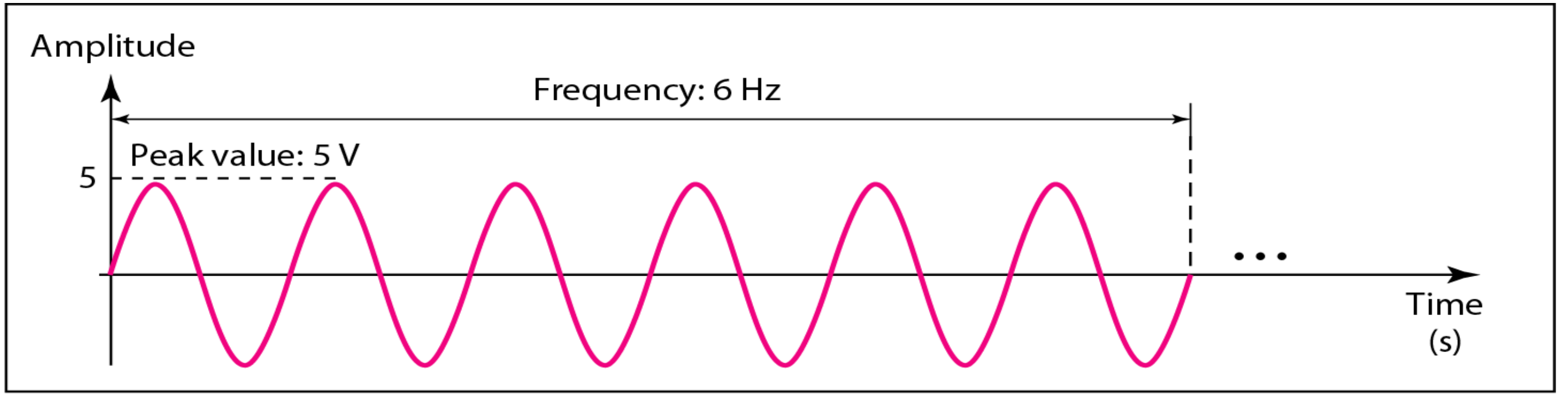
Wavelength (λ): It is the distance a simple signal can travel in one period.

Wavelength = Propagation speed x Period
= Propagation speed/Frequency

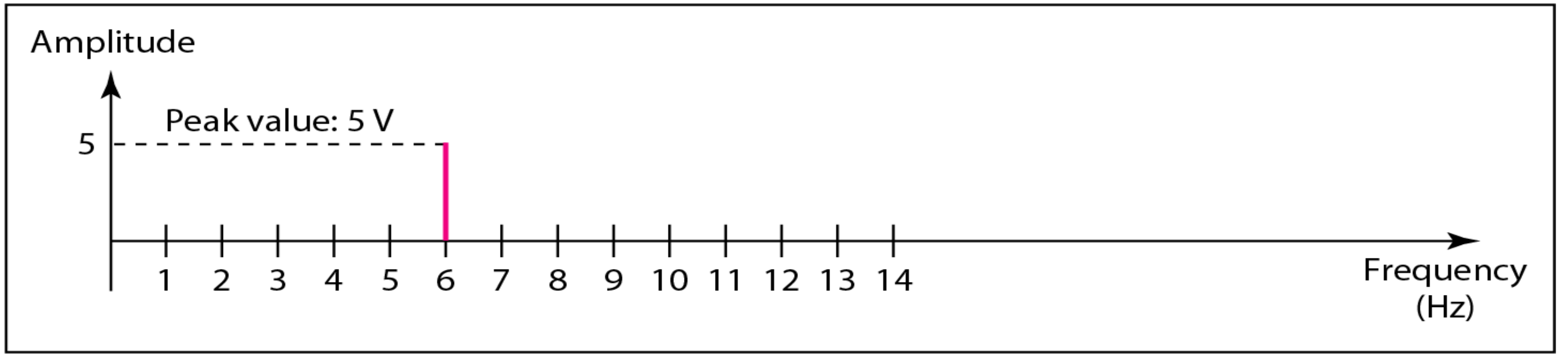


The time-domain and frequency-domain plots of a sine wave

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



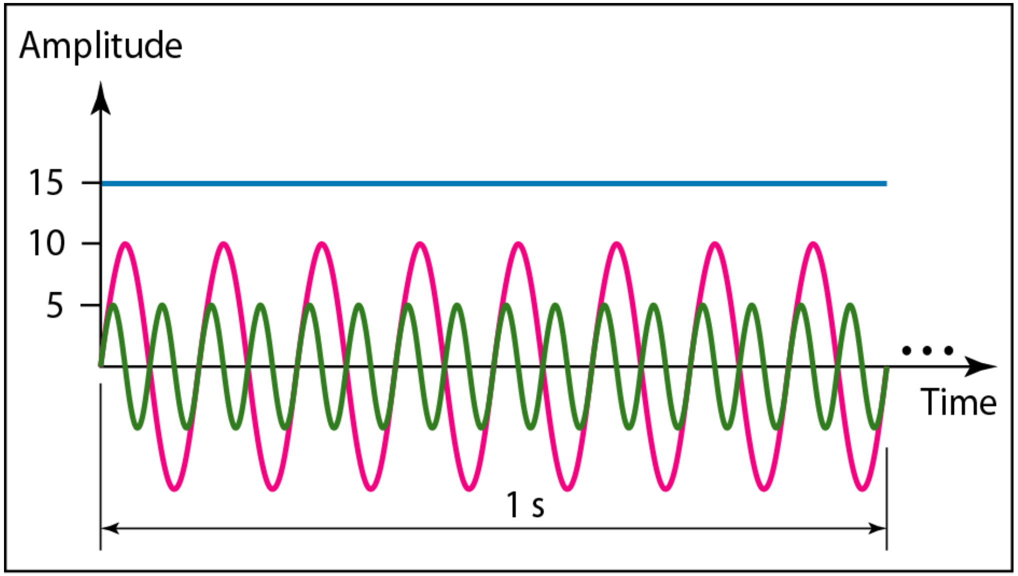
a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)



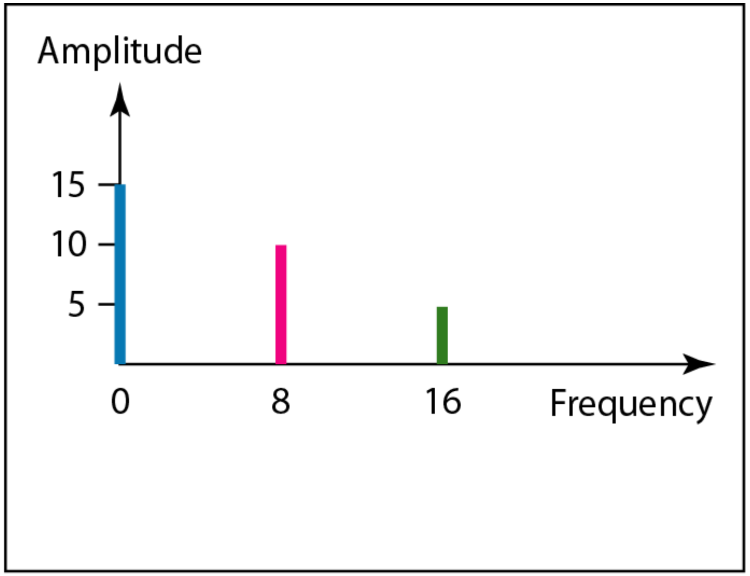
b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

The time domain and frequency domain of three sine waves

- The frequency domain is more compact and useful when we are dealing with more than one sine wave



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



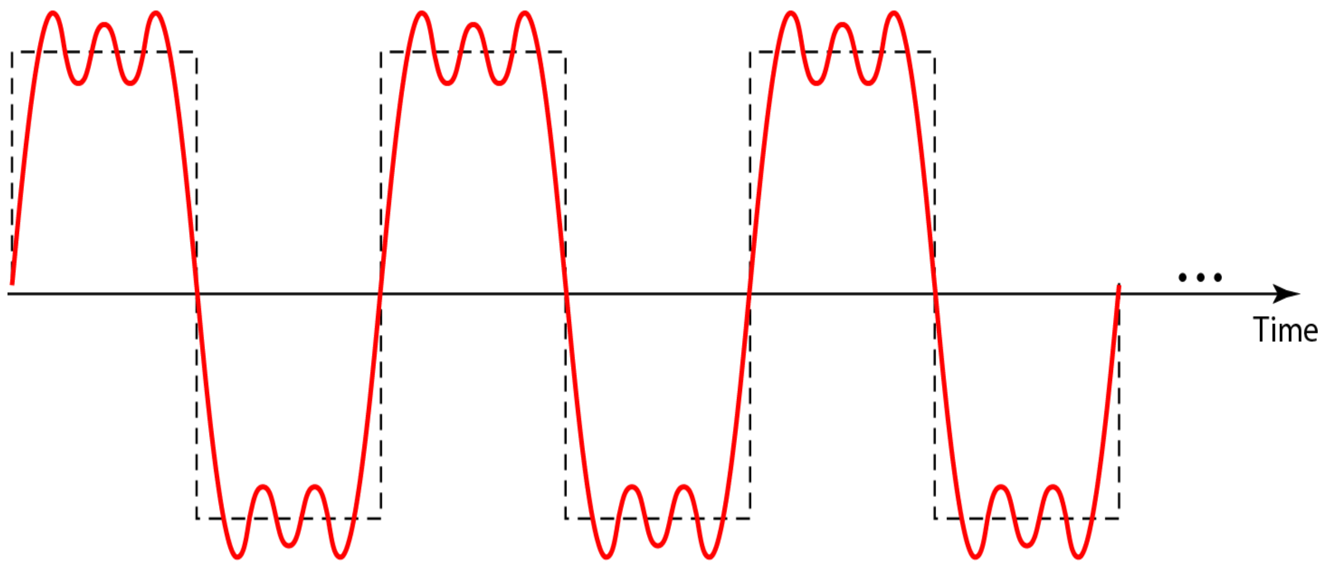
b. Frequency-domain representation of the same three signals

Basics of Analog & Digital Signals

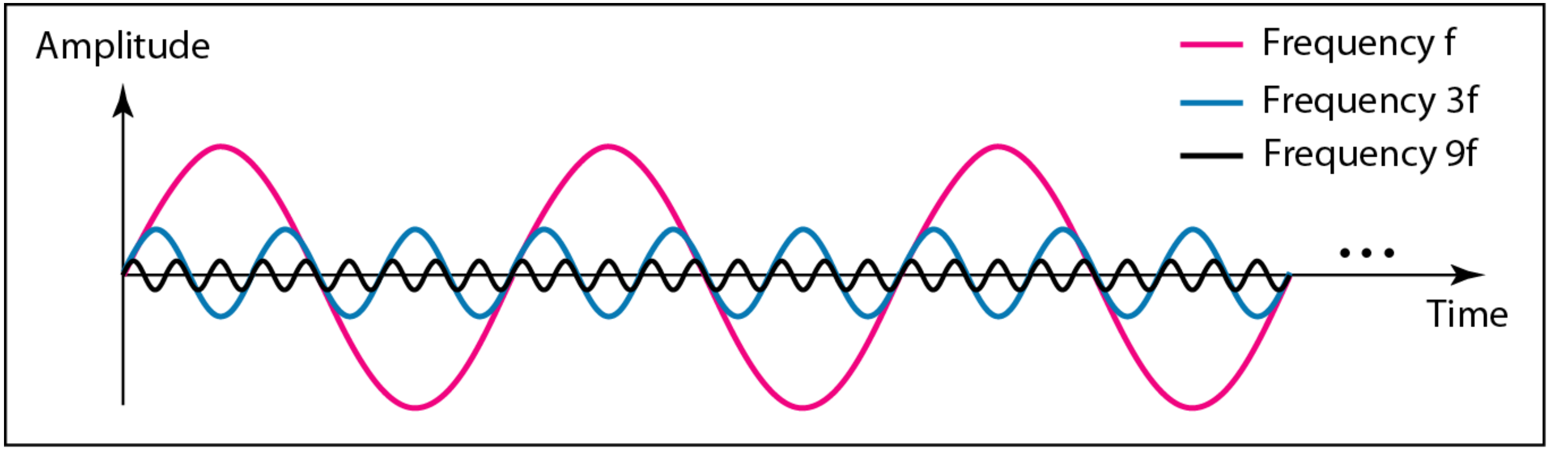
Periodic Analog Signals

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

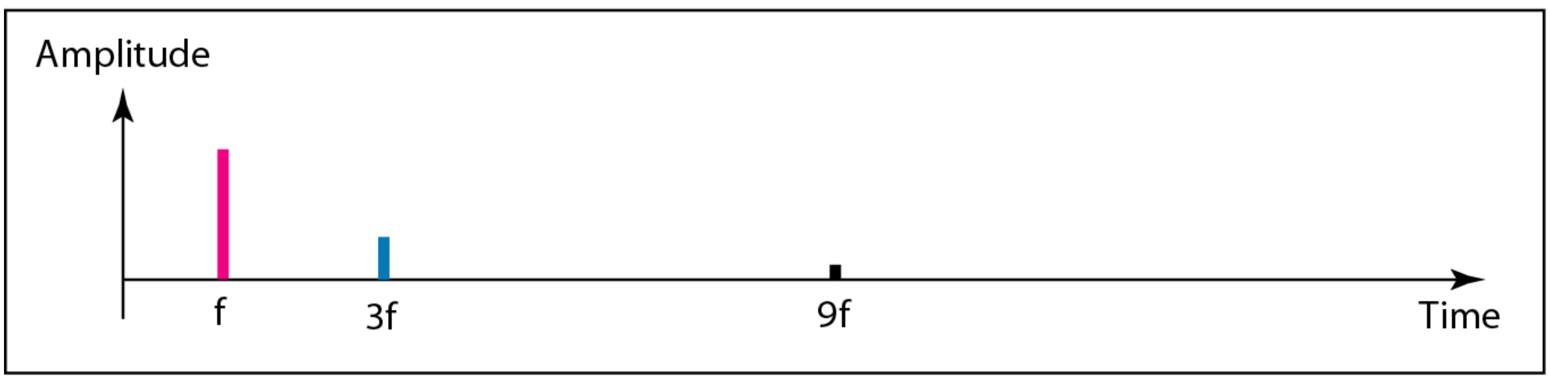
A composite periodic signal



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



a. Time-domain decomposition of a composite signal



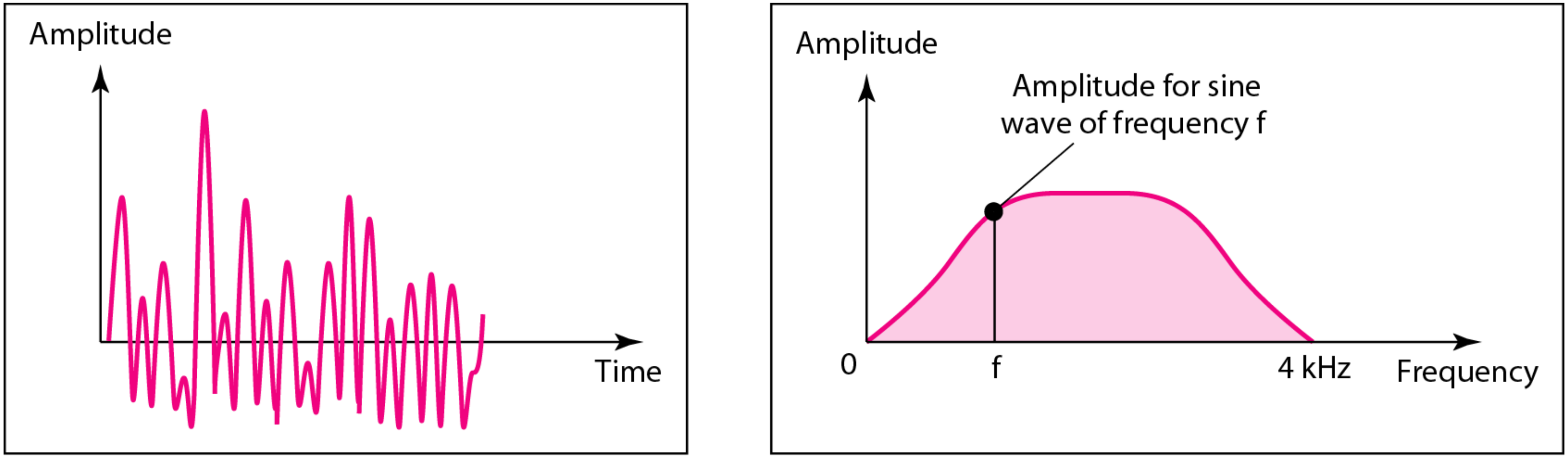
b. Frequency-domain decomposition of the composite signal

fundamental frequency = f, also known as first harmonic

Analog Signals

- If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies
- If the composite signal is non-periodic or aperiodic (signal created by a microphone or a telephone set), the decomposition gives a combination of sine waves with continuous frequencies

The time and frequency domains of a nonperiodic signal



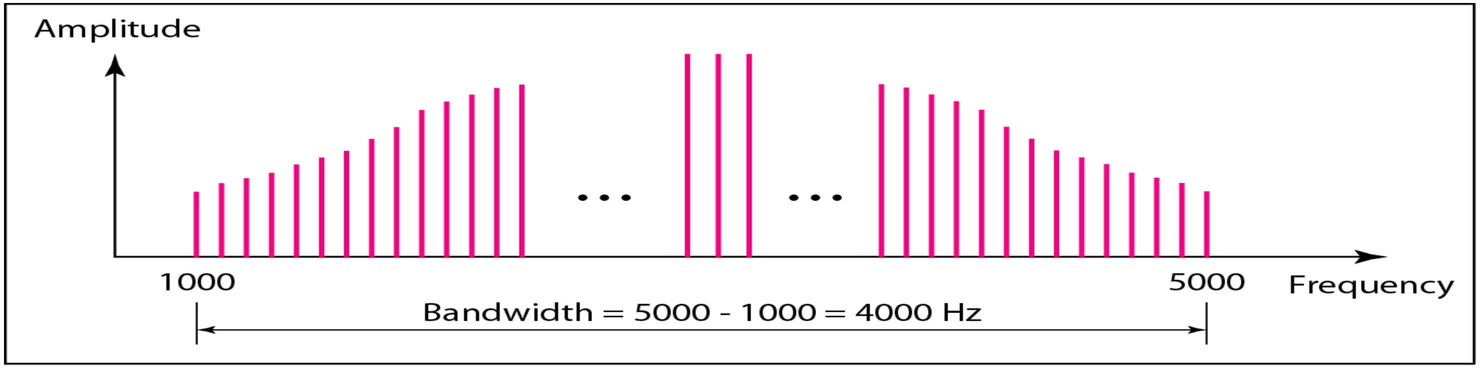
a. Time domain

b. Frequency domain

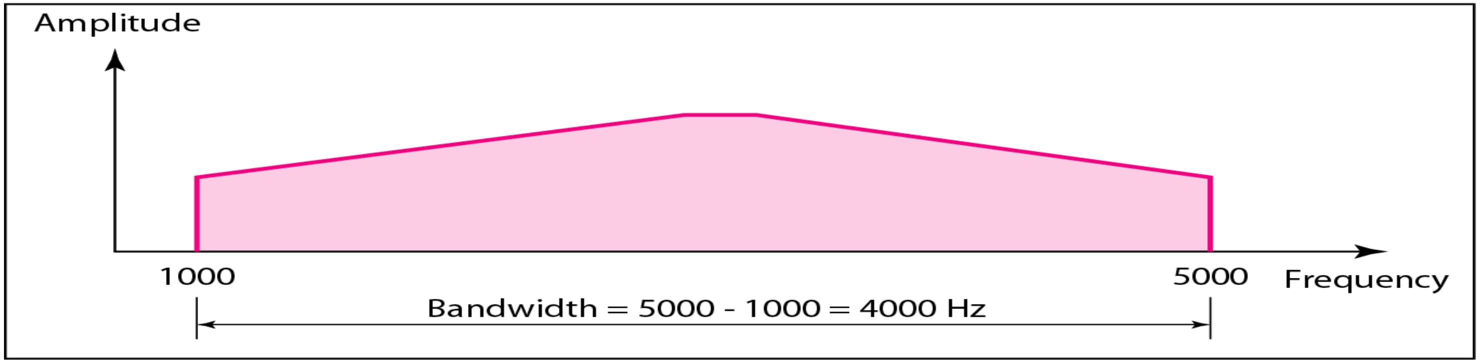
Bandwidth

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

The bandwidth of periodic and nonperiodic composite signals



a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

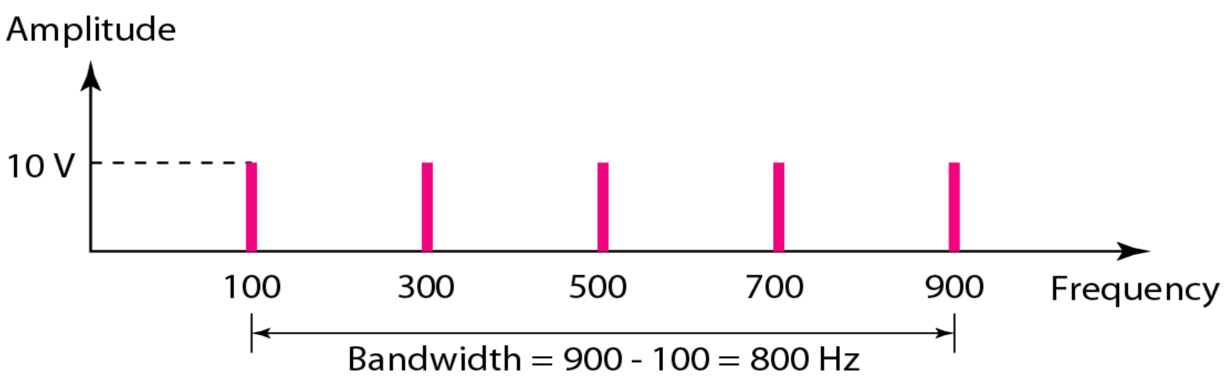
Example 1

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

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



Example 2

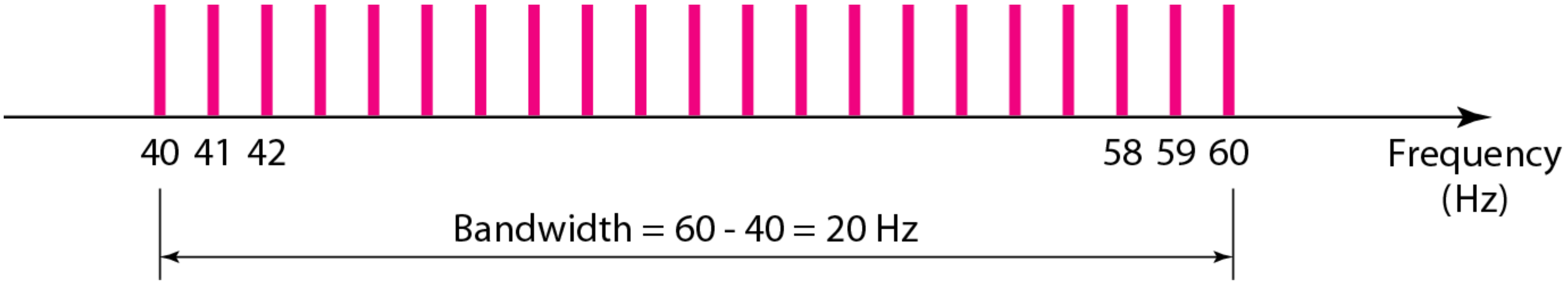
A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

Solution:

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

$$\boxed{B = f_h - f_l \;\Rightarrow\; 20 = 60 - f_l \;\Rightarrow\; f_l = 60 - 20 = 40 \text{ Hz}}$$

The spectrum contains all integer frequencies

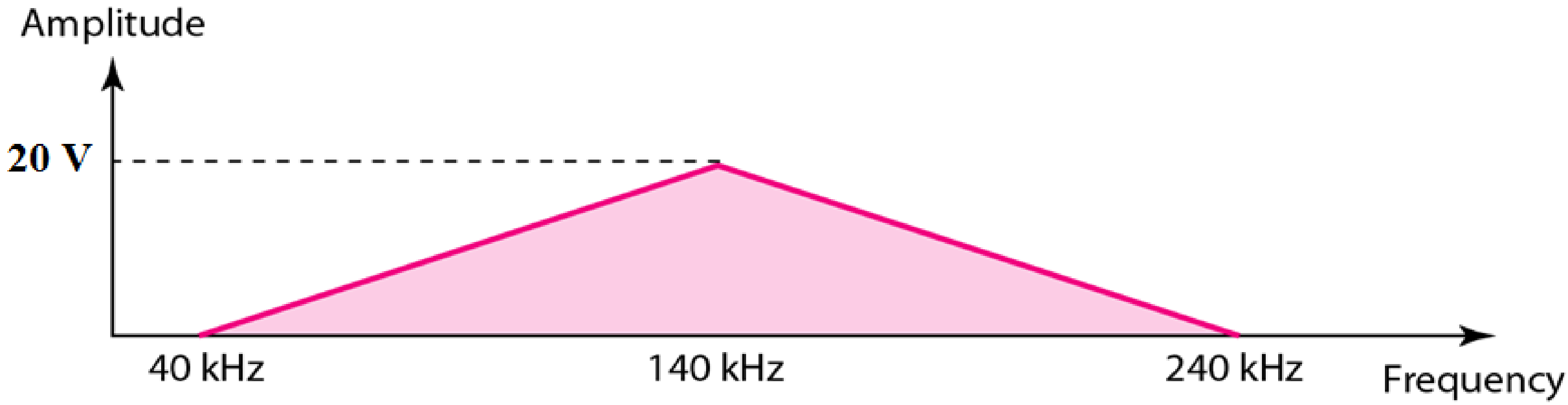


Example 3

A non-periodic 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 plot of the signal.

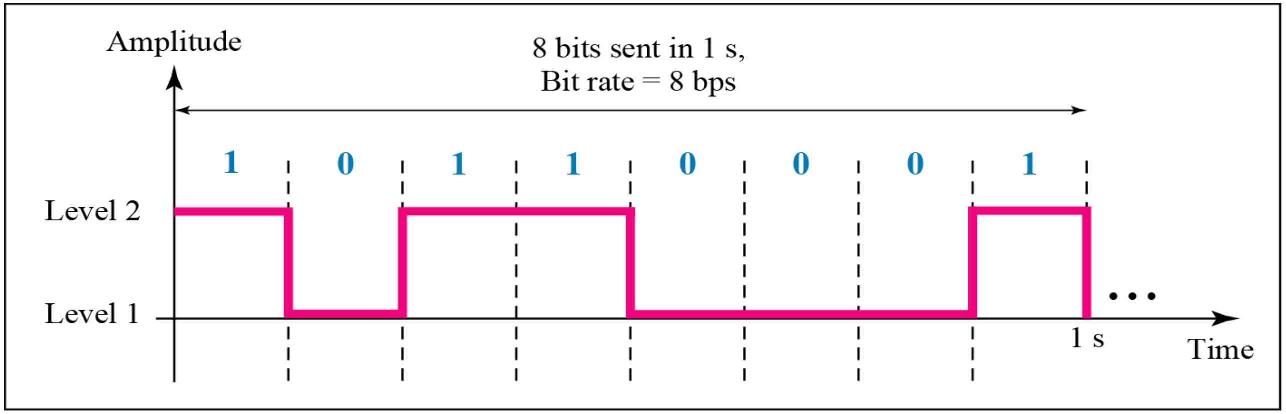
Solution:

The lowest frequency must be at 40 kHz and the highest at 240 kHz. Following figure shows the frequency domain plot and the bandwidth

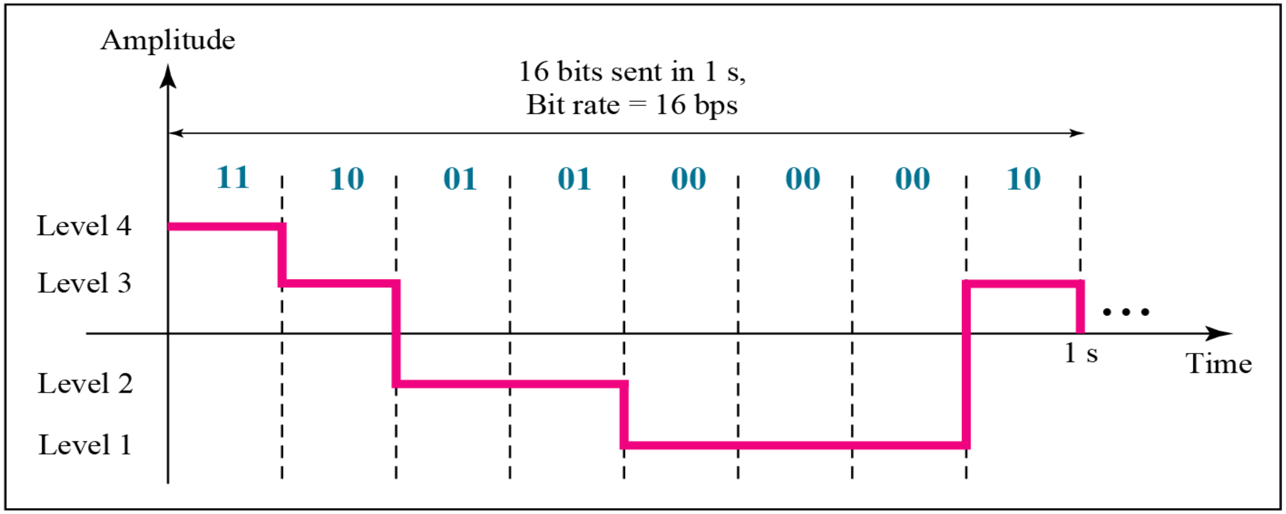


Digital Signals

- 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



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

Digital Signals

Bit Rate:

- The **bit rate** is the number of bits sent in 1 second
- It is expressed in bits per second (bps)

Bit Length

- It is the distance one bit occupies on the transmission medium

Bit length = Propagation speed x bit duration

Example 1

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

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

Solution: 100 x 24 x 80 x 8 = 1,536,000 bps = 1.536 Mbps

Example 2

A digitized voice channel, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?

Solution:

The bit rate can be calculated as

$$\boxed{2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}}$$

Example 3: What is the bit rate for high-definition TV (HDTV)?

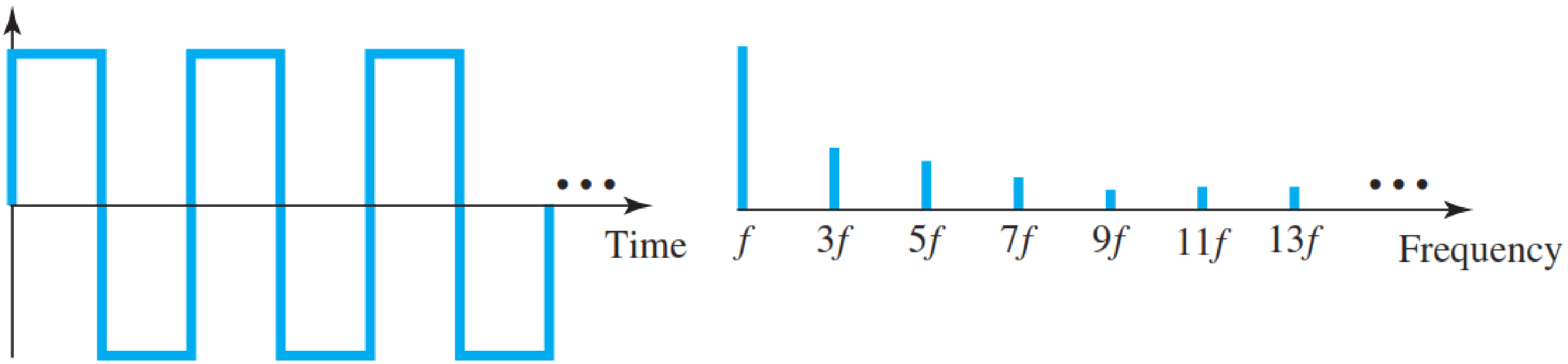
There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represents one color pixel.

Solution:

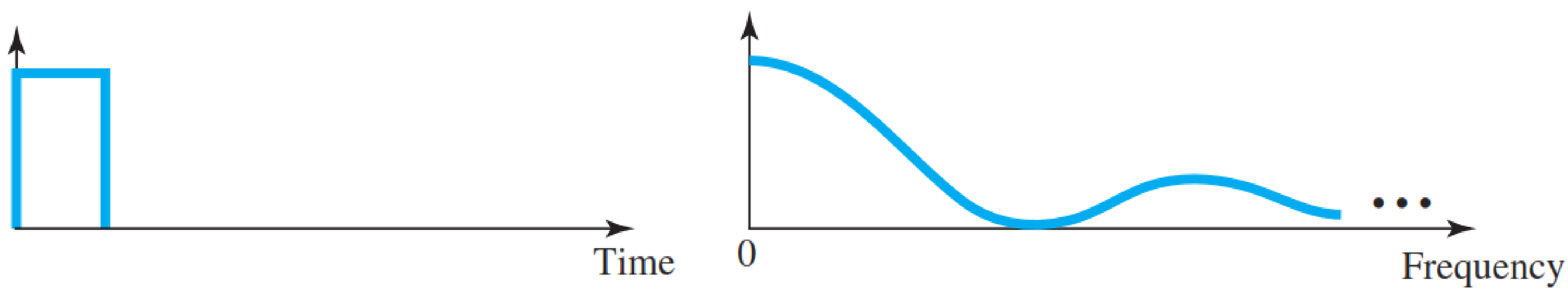
$$\boxed{1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \text{ or } 1.5 \text{ Gbps}}$$

Digital Signal as a Composite Analog Signal

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



a. Time and frequency domains of periodic digital signal



b. Time and frequency domains of nonperiodic digital signal

Transmission of Digital Signals

Baseband Transmission [Local area network]

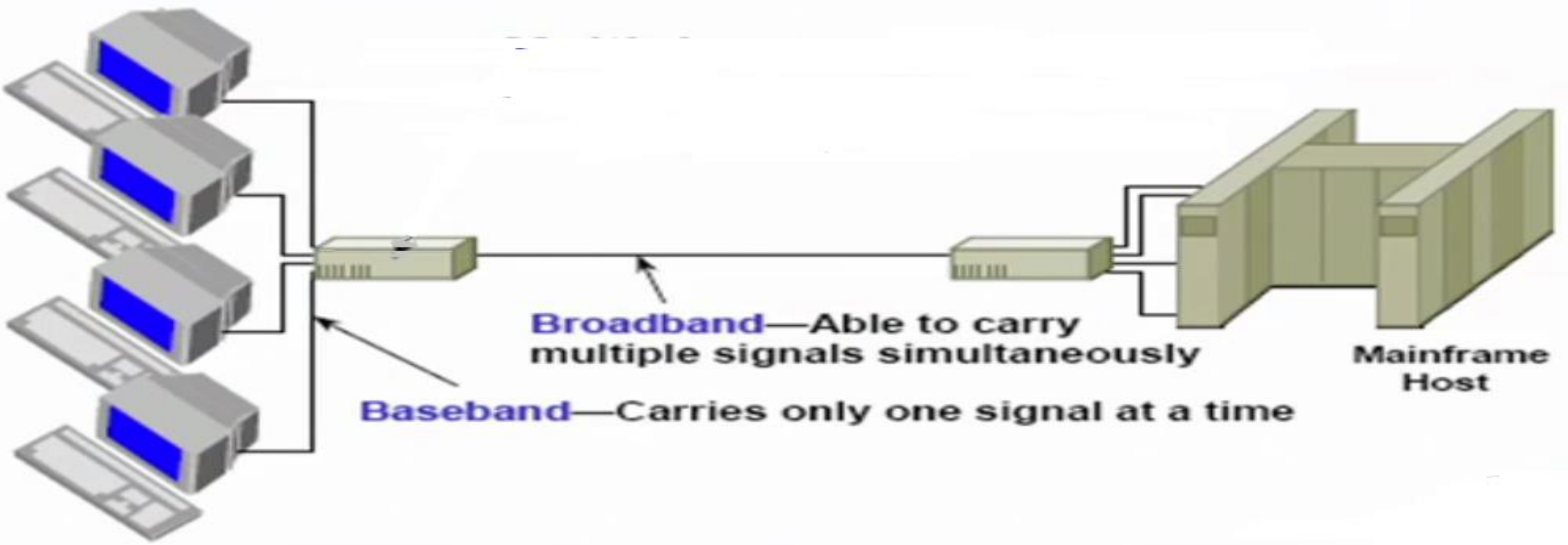
- Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal
- It typically use digital signaling over a single wire; the transmissions themselves take the form of either electrical pulses or light
- The digital signal used in baseband transmission occupies the entire bandwidth of the network media to transmit a single data signal
- Baseband communication is bidirectional, allowing computer to both send and receive data using single cable. However, the sending and receiving cannot occur on the same wire at the same time.
- Using baseband transmissions, it is possible to transmit multiple signals on a single cable by using multiplexing. Baseband typically uses Time-Division Multiplexing (TDM)

Transmission of Digital Signals

Broadband Transmission [Wide area network]

- Broadband transmission or modulation means changing the digital signal to an analog signal for transmission
- Broadband uses analog signals in the form of optical or electromagnetic waves over multiple transmission frequencies
- For signals to be both sent and received, the transmission media must be split into two channels. Alternatively, two cables can be used: one to send and one to receive transmissions
- Multiple channels are created in a broadband system by using a multiplexing technique known as Frequency Division Multiplexing (FDM)

Transmission of Digital Signals

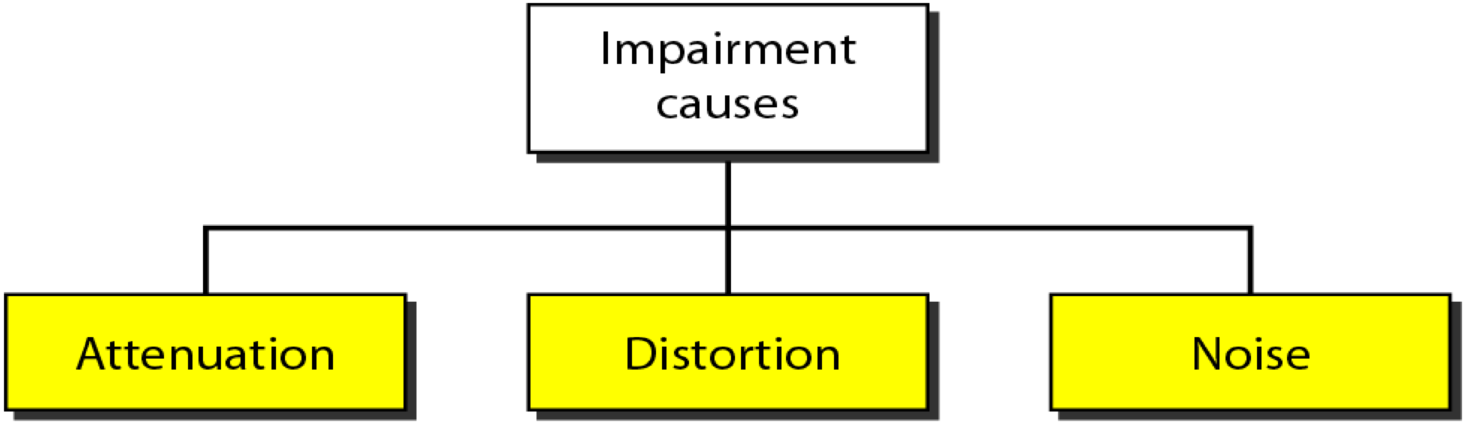


Transmission Impairment & Data Rate Limits

Transmission Impairment

- Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment
- This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium. What is sent is not what is received

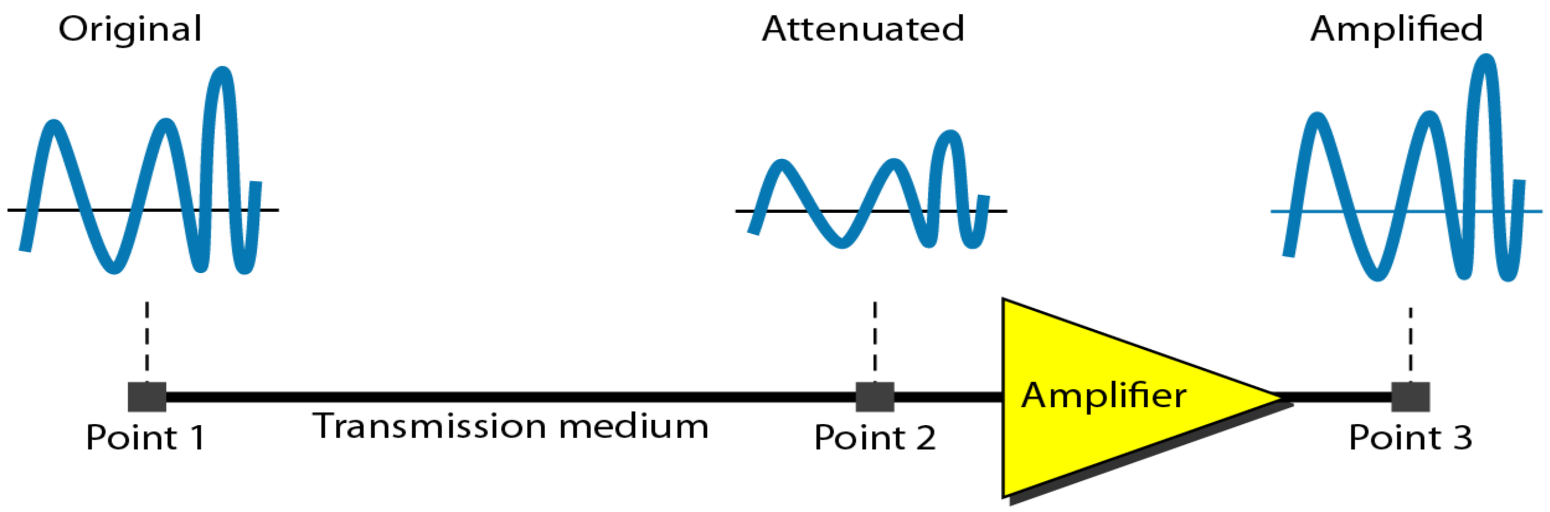
Causes of impairment



Attenuation

- Loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium. To compensate for this loss, amplifiers are used to amplify the signal
- The decibel (dB) measures the relative strengths of two signals or one signal at two different points

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



Example 1

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

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

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

Example 2

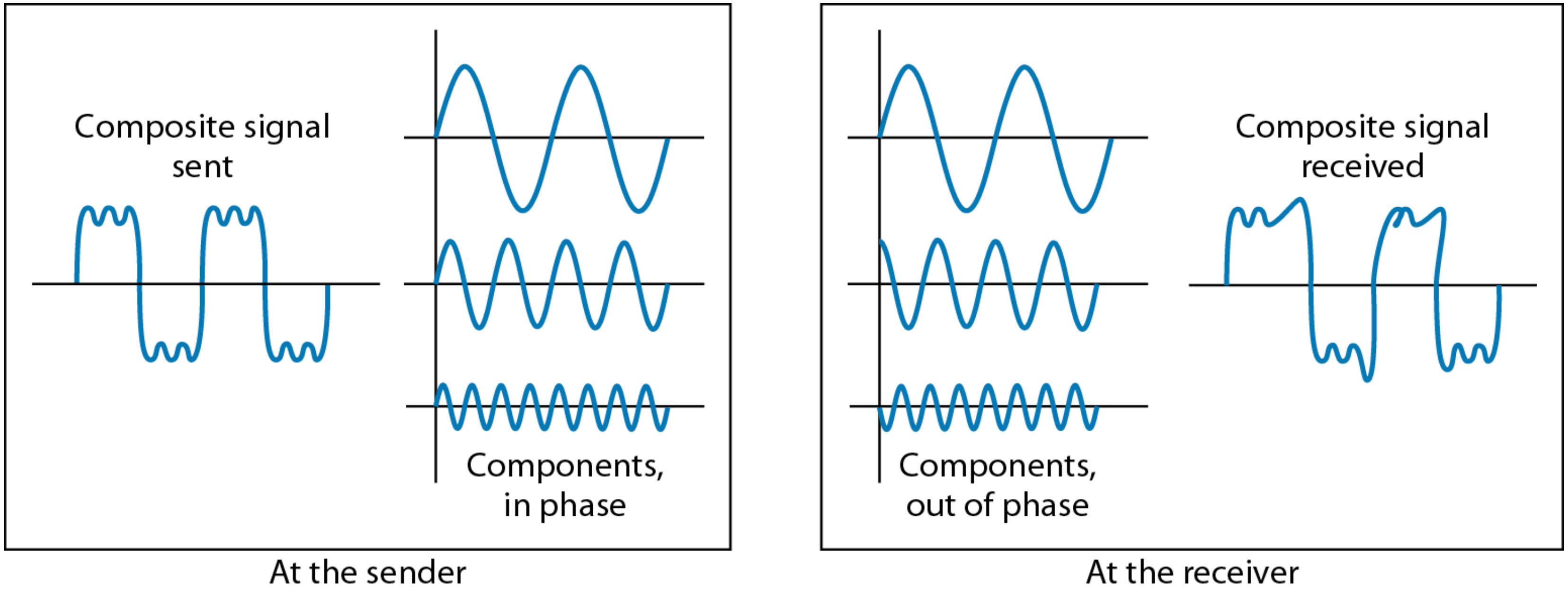
A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

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

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

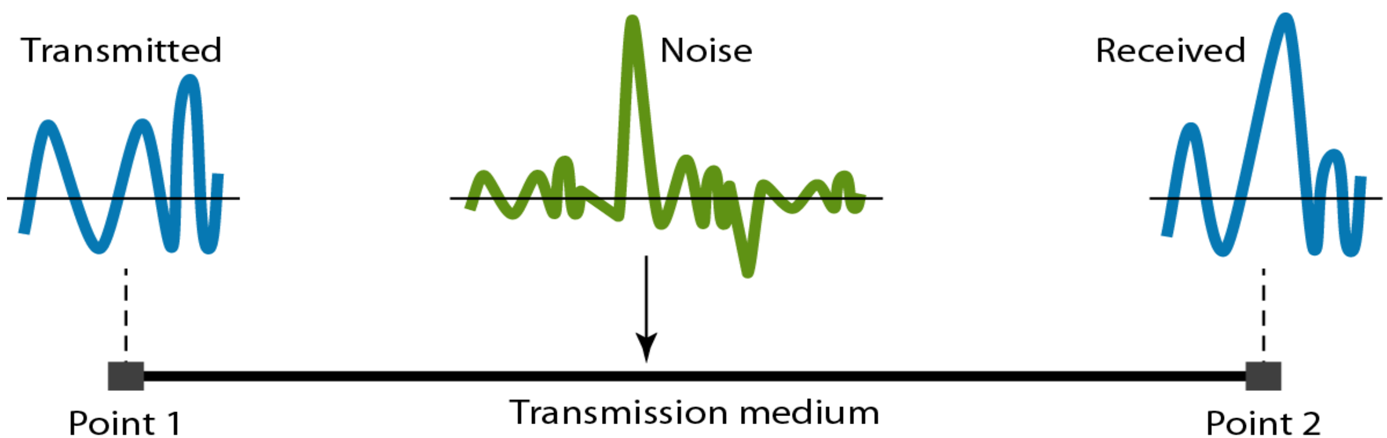
Distortion

- Distortion means that the signal changes its form or shape
- It can occur in a composite signal made of different frequencies. Each signal component has its own propagation speed through a medium and, therefore, its own delay in arriving at the final destination. Differences in delay may create a difference in phase



Noise

- Several types of noise, such as thermal noise, induced noise, crosstalk, and impulse noise, may corrupt the signal
  - Thermal noise is the random motion of electrons in a wire, which creates an extra signal not originally sent by the transmitter
  - Induced noise comes from sources such as motors and appliances
  - Crosstalk is the effect of one wire on the other
  - Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning etc.



Signal-to-Noise Ratio

- To find the theoretical bit rate limit, we need to know the ratio of the signal power to the noise power. The signal-to-noise ratio is defined as:

SNR = average signal power/average noise power

- A high SNR means the signal is less corrupted by noise; a low SNR means the signal is more corrupted by noise
- Since SNR is the ratio of two powers, it is often described in decibel units:

$$\mathbf{SNR_{dB} = 10\log_{10}SNR}$$

Example

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}?

Solution:

The values of SNR and SNR_{dB} can be calculated as follows:

$$SNR = (10{,}000\ \mu W)/(1\ \mu W) = 10{,}000$$

$$SNR_{dB} = 10\log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40\ dB$$

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:
    - The bandwidth available
    - The level of the signals we use
    - The quality of the channel (the level of noise)
- Two theoretical formulas were developed to calculate the data rate: one by **Nyquist** for a noiseless channel, another by **Shannon** for a noisy channel
- For a **noiseless** channel, the **Nyquist bit rate** formula defines the theoretical maximum bit rate

$$\textbf{BitRate} = 2 * \textbf{bandwidth} * \log_2 L$$

where L is the number of signal levels used to represent data

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

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

Noisy Channel: Shannon Capacity

- In reality, we cannot have a noiseless channel. In 1944, Claude Shannon introduced a formula, called the Shannon capacity, to determine the theoretical highest data rate for a noisy channel:

Capacity = bandwidth x $\log_2$ (1 + SNR)

Example 1

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 2

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

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

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

$$\text{BitRate} = 2 * \text{bandwidth} * \log_2 L$$

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

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

Performance

- One important issue in networking is the performance of the network. One characteristic that affect network performance is bandwidth
- 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
- Mainly, performance of a network can be measured in terms of:
  - Throughput
  - Latency (Delay)

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

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

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

Latency (Delay)

- A packet starts in a host (the source), passes through a series of routers, and ends its journey in another host (the destination)
- The latency or delay defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source.

Latency = propagation + transmission + queuing + processing

Latency (Delay)

- Propagation time measures the time required for a bit to travel from the source to the destination.

Propagation time = Distance / (Propagation Speed)

- **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 $\times 10^{8}$ m/s in cable.

$$\text{Propagation time} = (12{,}000 \times 10{,}000) / (2.4 \times 2^{8}) = 50 \text{ ms}$$

Latency (Delay)

- The transmission time of a message depends on the size of the message and the bandwidth of the channel.

Transmission time = (Message size) / Bandwidth

Example

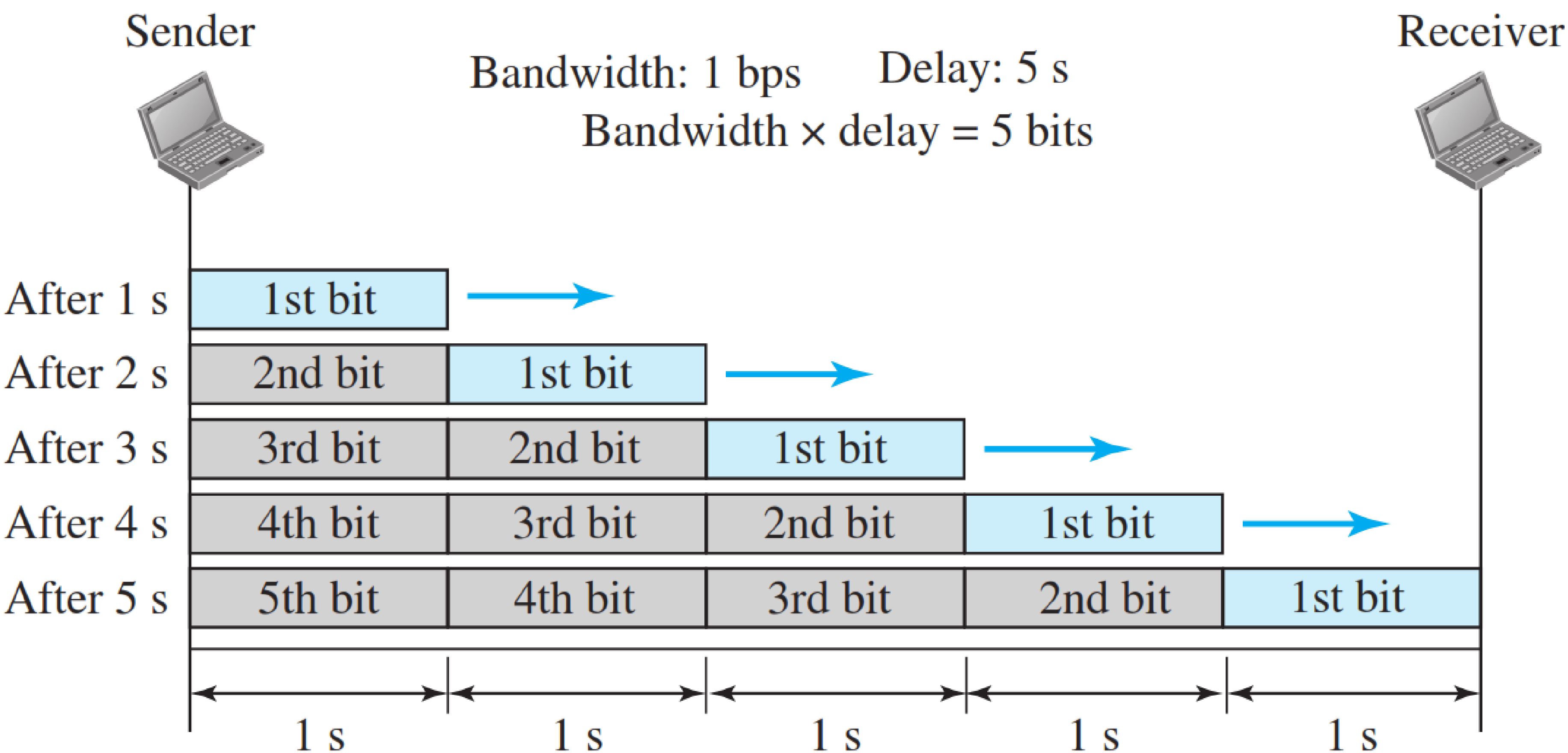
- What are the propagation time and the transmission time for a 2.5-KB (kilobyte) message (an email) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s .

$$\textbf{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \text{ ms}$$

$$\textbf{Transmission time} = (2500 \times 8) / 10^9 = 0.020 \text{ ms}$$

Bandwidth-Delay Product

The bandwidth-delay product defines the number of bits that can fill the link.

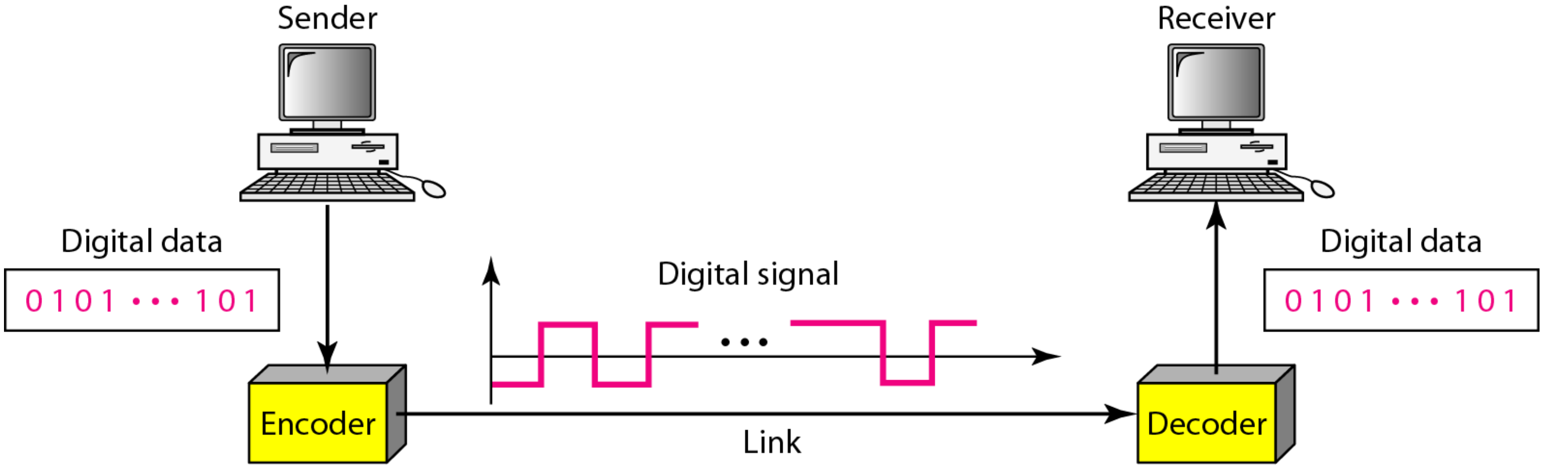


Digital Transmission

Digital-to-digital Conversion

- We can represent digital data by using digital signals
- The conversion involves three techniques:
  - Line coding
  - Block coding
  - Scrambling
- Line coding is always needed; block coding and scrambling may or may not be needed

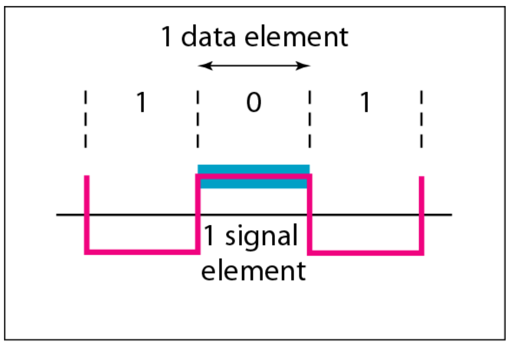
Line coding and decoding



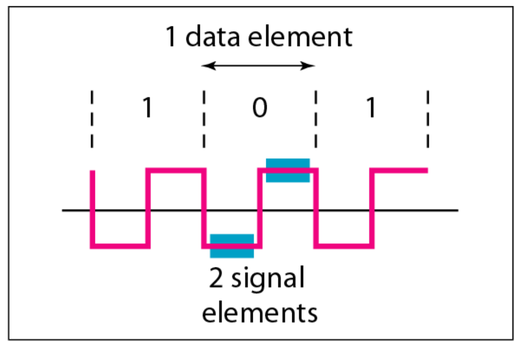
Line Coding

Signal element versus data element

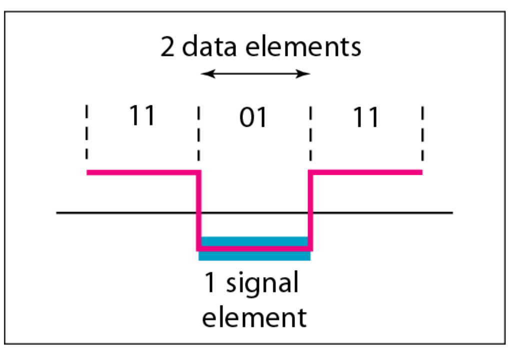
Data Element : smallest entity that can represent a piece of information
Signal Element : shortest unit (timewise) of a digital signal



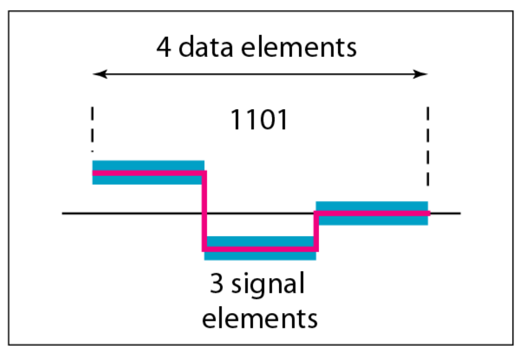
a. One data element per one signal element (r = 1)



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 $\left(r = \frac{4}{3}\right)$

ratio r is the number of data elements carried by each signal element.

Line Coding

- Line coding is the process of converting digital data to digital signals

Data Rate versus Signal Rate:

- **Data rate (or bit rate):** Number of data elements (bits) sent in one second. Unit is bits per second (bps)
- **Signal rate (or pulse rate or modulation rate or baud rate):** Number of signal elements sent in one second. The unit is baud

Signal rate (S) = c x Data rate (N)/r

Where r = number of data elements carried by each signal element

c = case factor , best worst average

Line Coding

Example

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?

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

Line Coding

Bandwidth

> **Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.**

- bandwidth (range of frequencies) is proportional to the signal rate (baud rate).

$$B_{\min} = c \times N \times (1/r)$$

$$N_{\max} = (1/c) \times B \times r$$

Line coding characteristics

Baseline Wandering

- Baseline : a running average of the received signal power
- Baseline Wandering :

  A long string of **0**s or **1**s can cause a drift in the baseline
- It makes difficult for the receiver to decode correctly.
- A good line coding scheme needs to prevent baseline wandering.

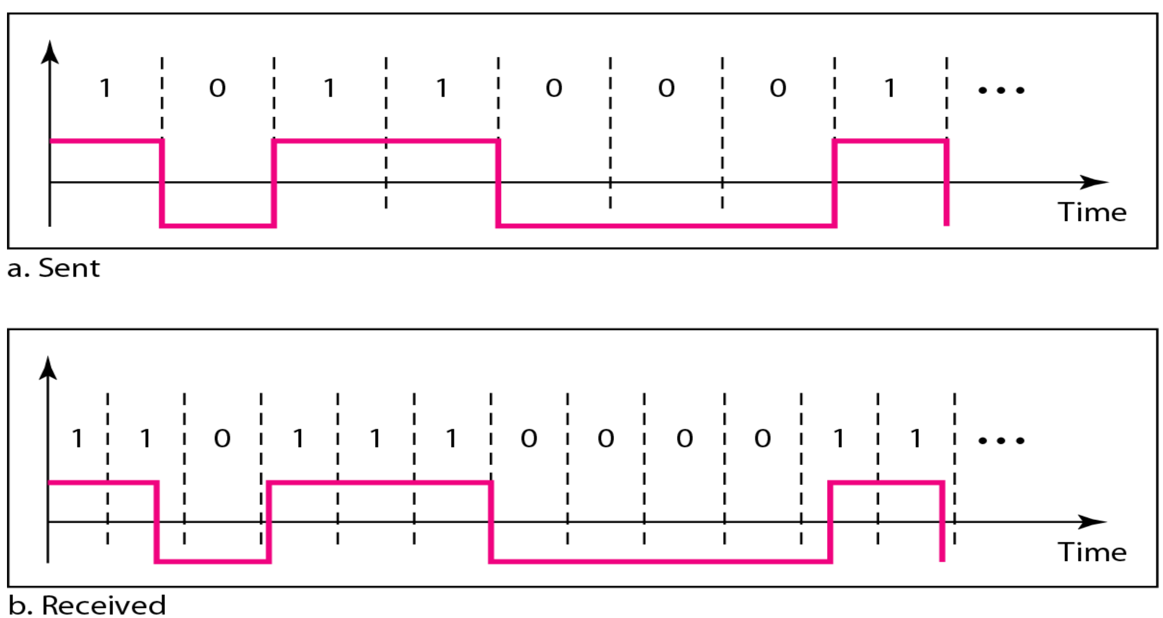
DC Components

- When the voltage level in a digital signal is constant for a while
- It creates very low frequencies around zero
- Creates problems for a system that cannot pass low frequencies
- For example, a telephone line cannot pass frequencies below 200 Hz.

Line coding characteristics

Self-synchronization

- A self-synchronizing digital signal includes timing information in the data being transmitted.
- If the receiver clock is faster or slower, the receiver might misinterpret the signals
- In the following figure, receiver has a shorter bit duration



Line coding characteristics

Self-synchronization : Example

- 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 coding characteristics

Built-in Error Detection

- It is desirable to have a built-in error-detecting capability in the generated code

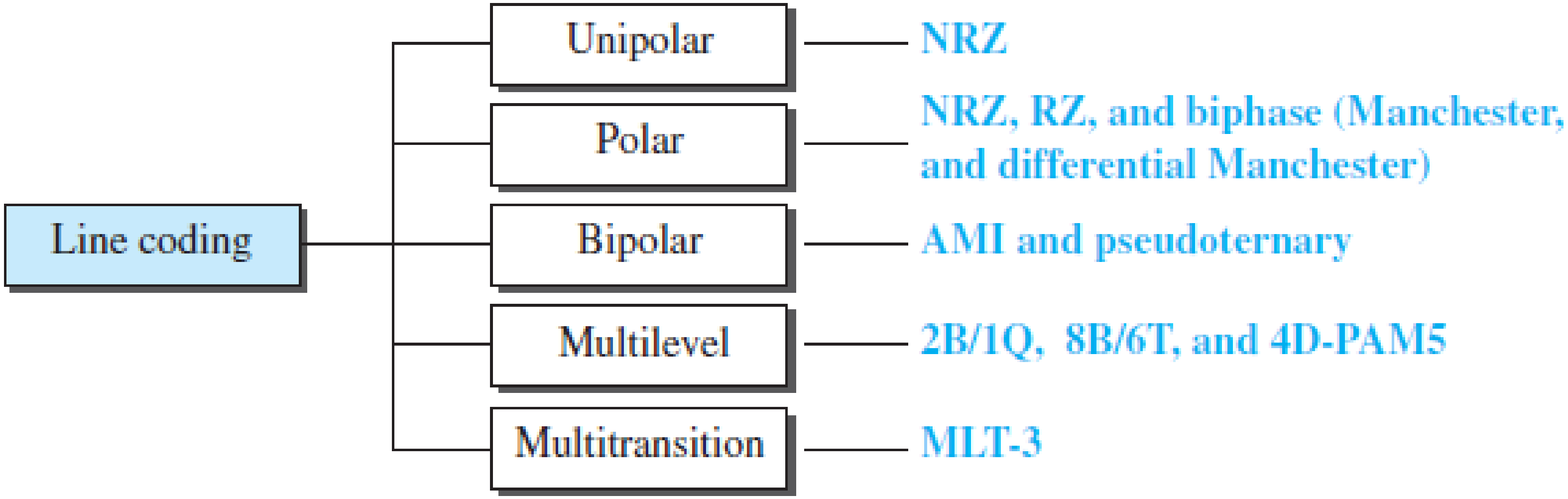
Immunity to Noise and Interference

- It a code that is immune to noise and other interferences.

Complexity

- A complex scheme is more costly to implement than a simple one.

Line coding schemes



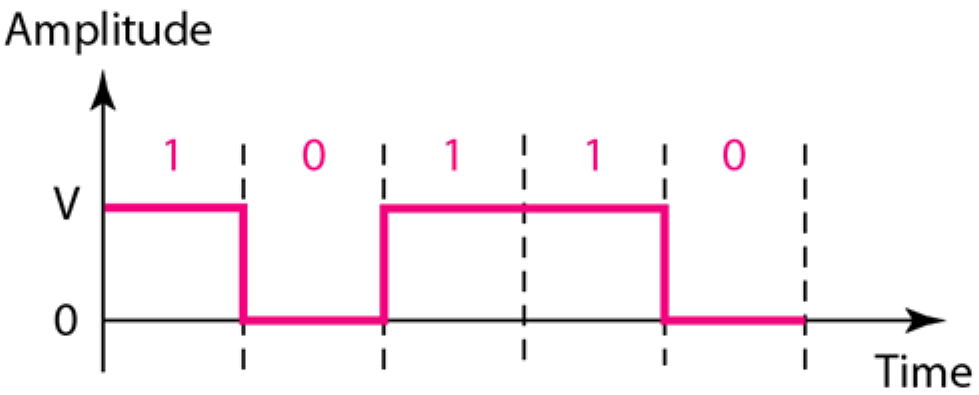
Unipolar NRZ (Nonreturn to Zero) scheme

Unipolar: All the signal levels are on one side of the time axis, either above or below

- Signal does not return to zero at the middle of the bit

Unipolar NRZ:

- 0 = low level
- 1 = high level



Polar schemes: NRZ-L (Nonreturn to Zero-Level) and NRZ-I (Nonreturn to Zero Inverted)

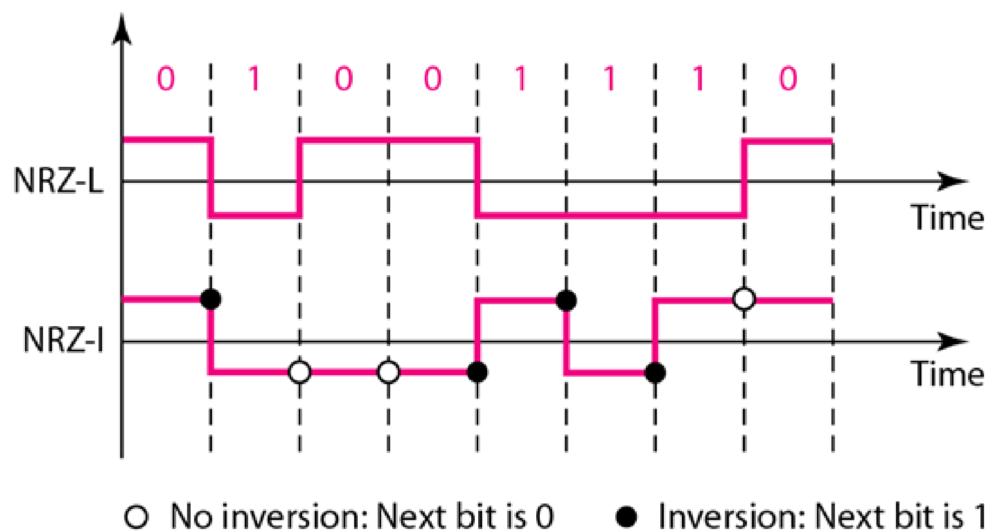
- **Polar schemes:** The voltages are on both sides of the time axis
- In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit

NRZ-L:

- 0 = high level
- 1 = low level

NRZ-I:

- 0 = no transition at beginning of interval
- 1 = transition at beginning of interval



NRZ-S and NRZ-M (Two versions of NRZ-I)

- Mark and Space
  - **Mark:** Binary 1
  - **Space:** Binary 0
- **NRZ-S:** A binary encoding scheme in which a signal parameter, such as voltage, undergoes **a change in a level** every time when a **"0" occurs**, but when a "one" occurs, it remains the same, *i.e.* , no transition occurs
- **NRZ-M:** A binary encoding scheme in which a signal parameter, such as voltage, undergoes **a change in a level every** time when a **"1" occurs**, but when a "zero" occurs, it remains the same, *i.e.* , no transition occurs

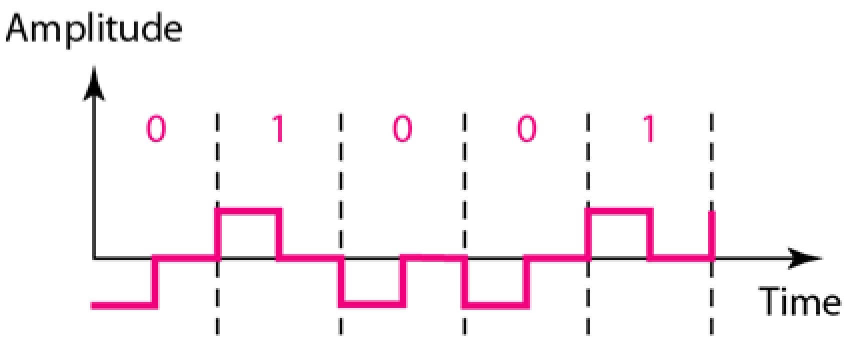
Note: NRZ-S and NRZ-M schemes are inverse to each other

Polar RZ (Return-to-zero) scheme

- The main problem with NRZ encoding occurs when the sender and receiver clocks are not synchronized
- In RZ, the signal changes not between bits but during the bit
- In RZ, signal goes to 0 in the middle of each bit and it remains there until the beginning of the next bit

RZ:

- 0 = low level to zero
- 1 = high level to zero



Polar biphase: Manchester and differential Manchester schemes

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

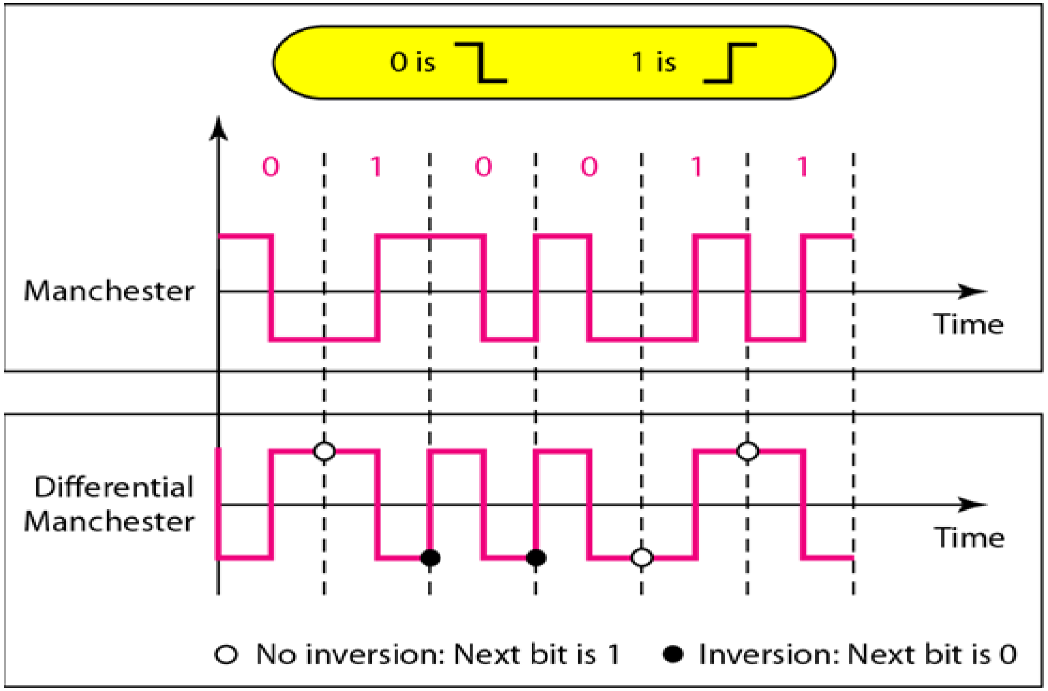
Manchester:

- 0 = transition from high to low
- 1 = transition from low to high

Differential Manchester:

Always a transition in middle of interval

- 0 = transition at beginning of interval
- 1 = no transition at beginning of interval



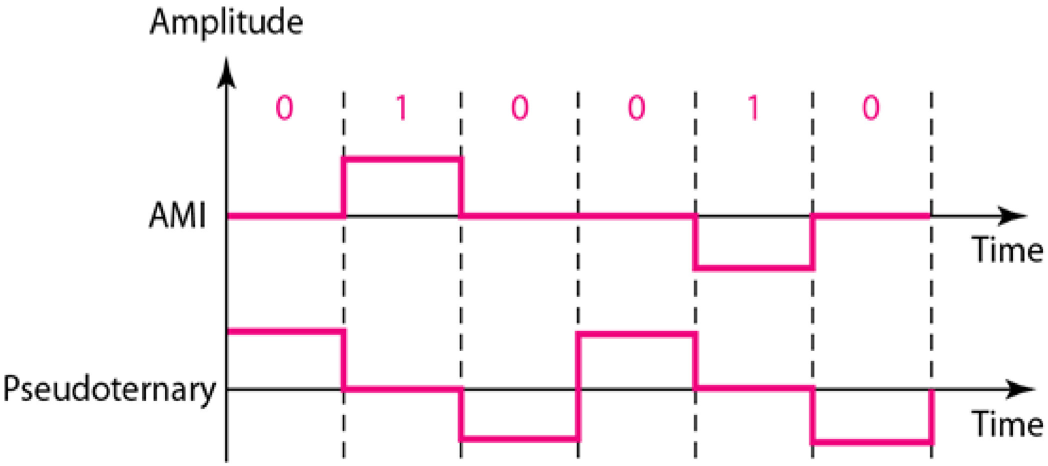
Bipolar schemes: AMI (alternate mark inversion) and pseudoternary

AMI:

- 0 = neutral zero voltage (or no line signal)
- 1 = positive or negative level, alternating for successive ones

Pseudoternary:

- 0 = positive or negative level, alternating for successive zeros
- 1 = no line signal



Multilevel: mBnL

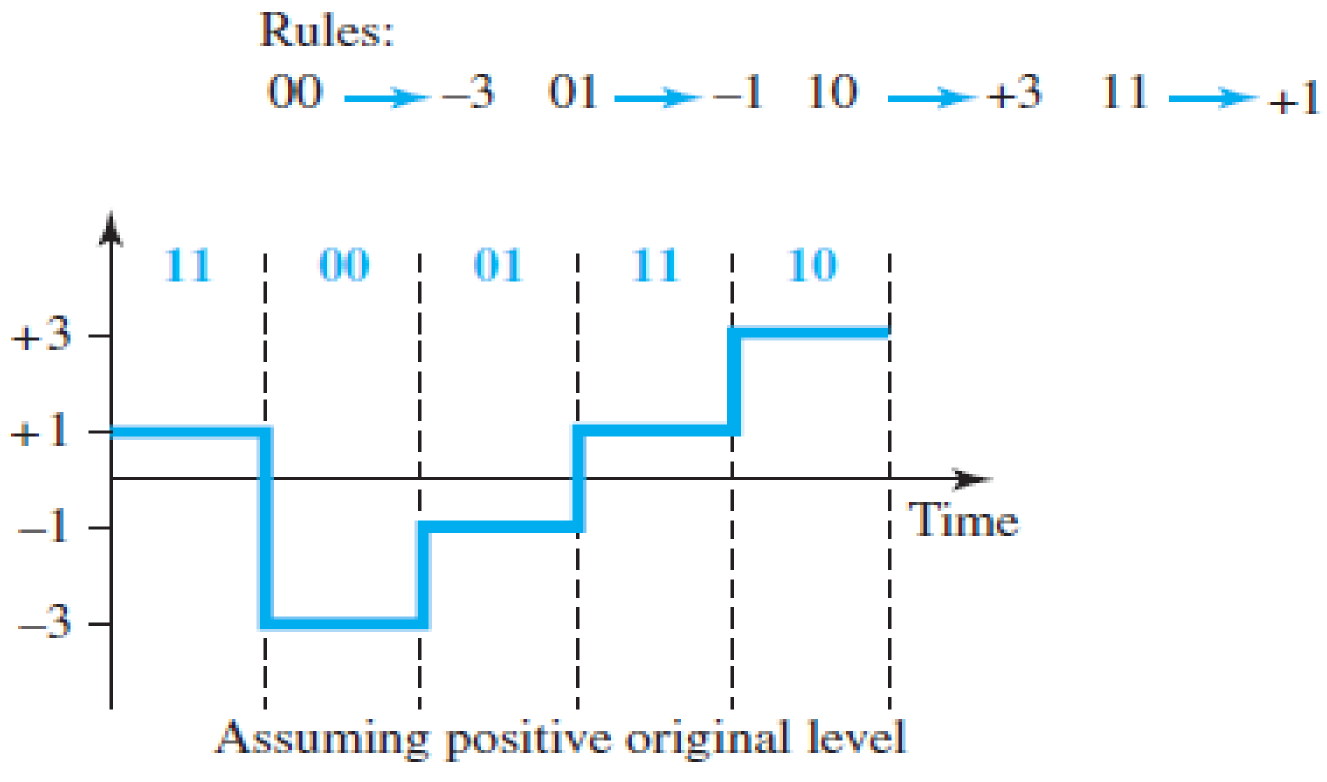
In *mBnL* schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which $2^m \leq L^n$.

- **m:** Length of the binary pattern
- **B:** Means binary data
- **n:** Length of signal pattern
- **L:** Number of levels

Multilevel: 2B1Q (two binary one quaternary) scheme

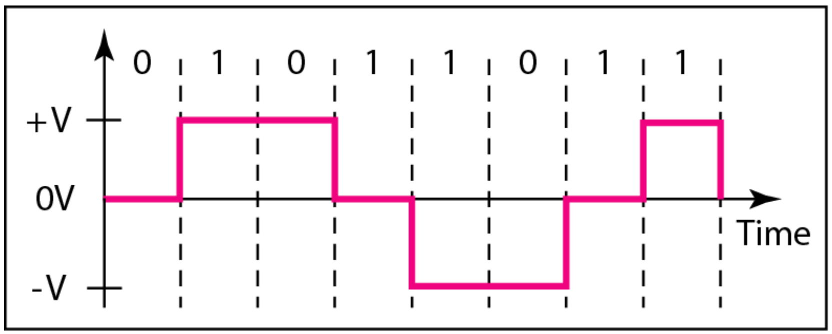
- ***2B1Q*** scheme uses data pattern of size 2 and encodes the 2-bit patters as 1 signal element belonging to a four-level signal

- Here, m = 2, n = 1, and L = 4 (quaternary)

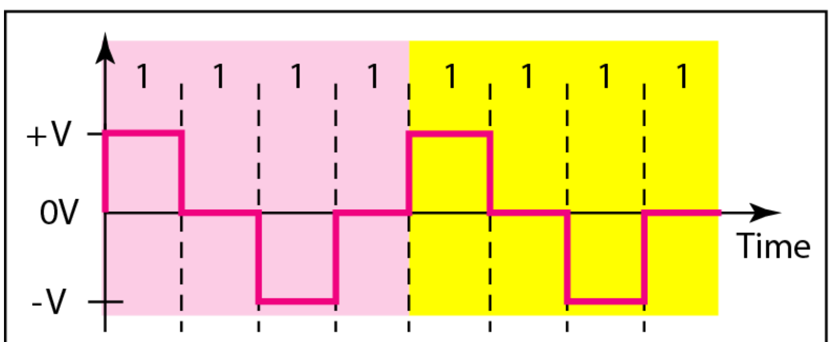


Multitransition : MLT-3

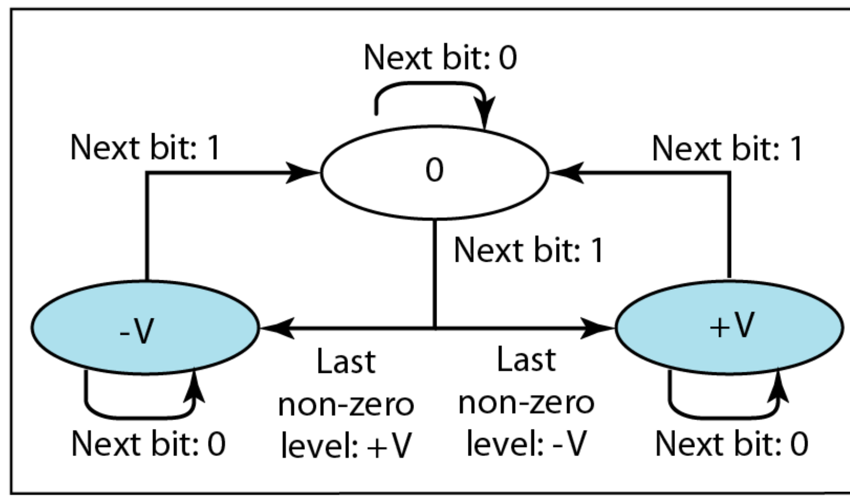
- Because of synchronization requirements we force transitions.
- Codes can be created that are differential at the bit level forcing transitions at bit boundaries.



a. Typical case



b. Worse case



c. Transition states

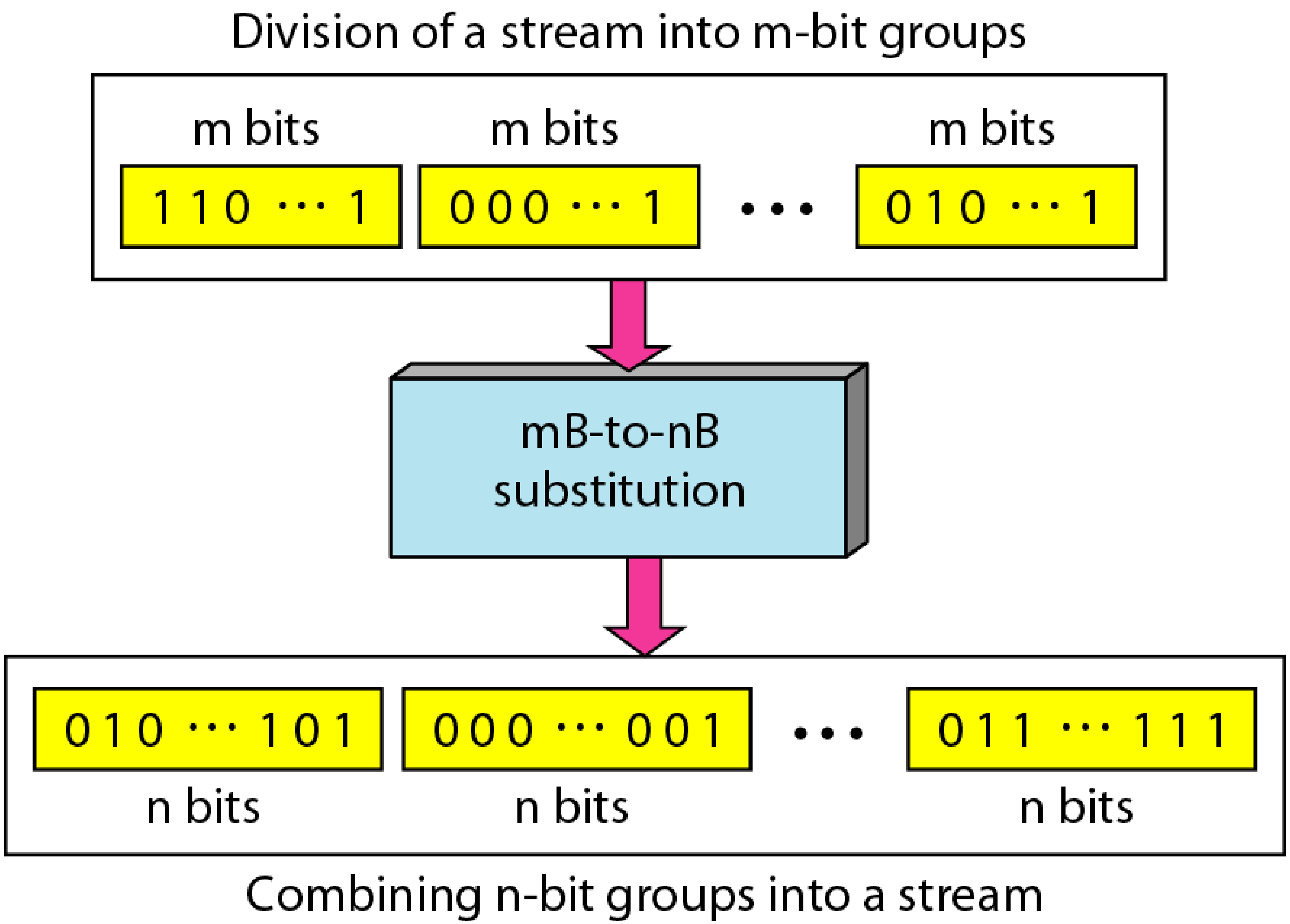
Summary of line coding schemes

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Unipolar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multiline	MLT-3	$B = N/3$	No self-synchronization for long 0s

Block Coding

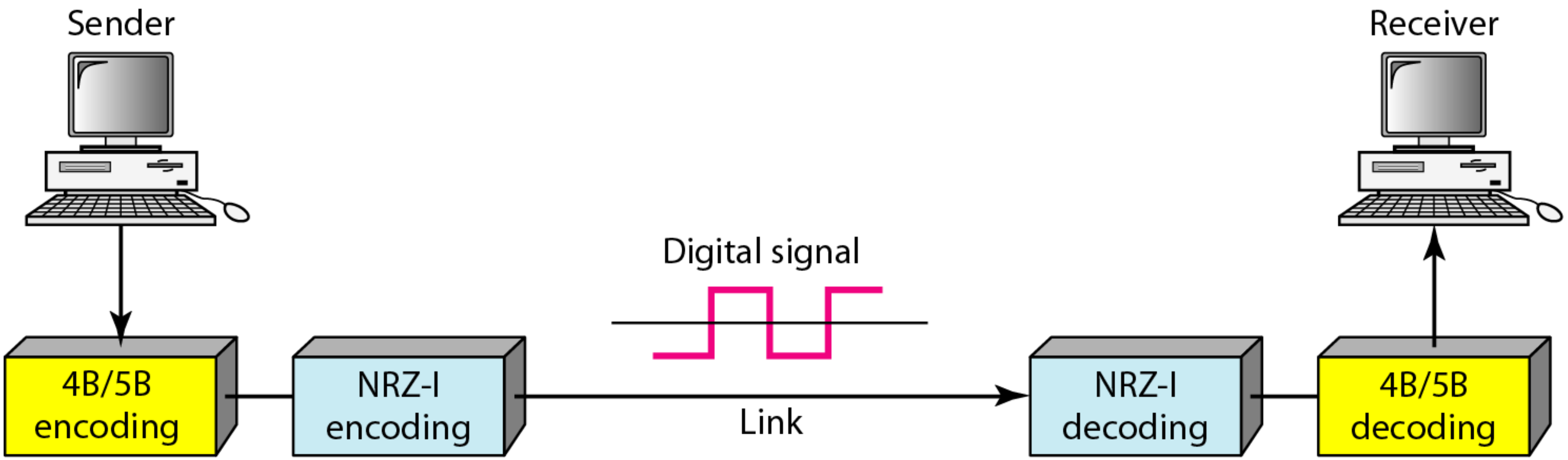
- For a code to be capable of error detection, we need to add redundancy, i.e., extra bits to the data bits.
- Synchronization also requires redundancy - transitions are important in the signal flow and must occur frequently.
- Block coding is normally referred to as mB/nB coding.
- It replaces each m-bit group with an n-bit group.
- Block coding is done in three steps:
  - division,
  - substitution and
  - combination

Block Coding



Block Coding

Using block coding 4B/5B with NRZ-I line coding scheme

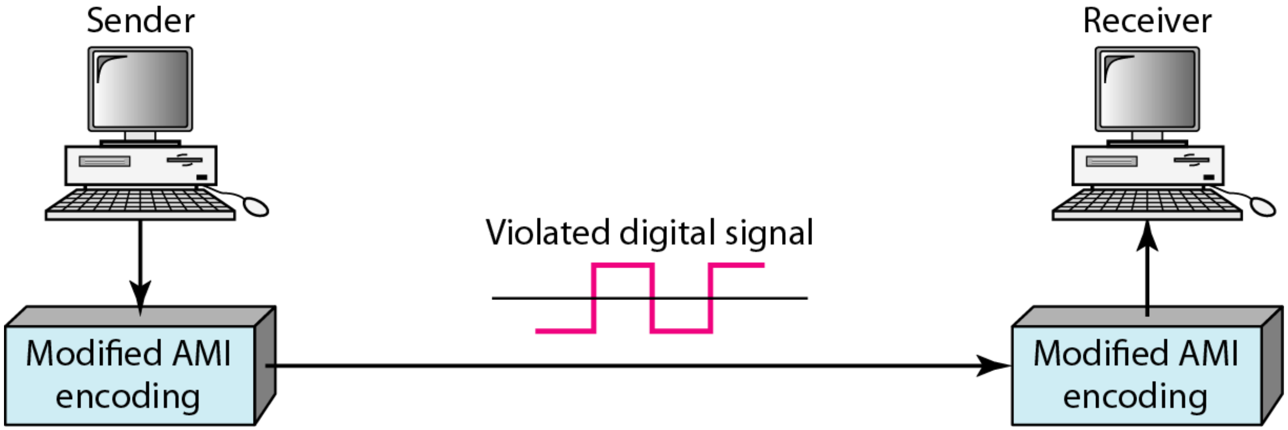


4B/5B

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

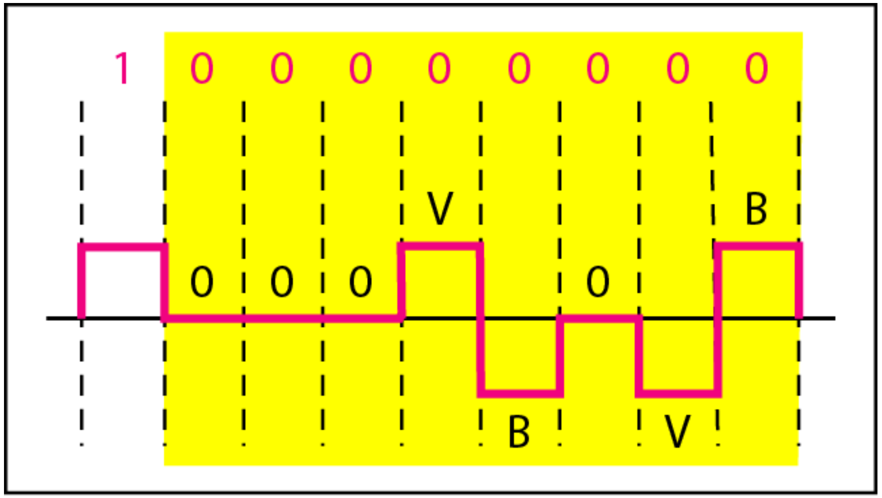
Scrambling

- The best code is one that does not increase the bandwidth for synchronization and has no DC components.
- Scrambling is a technique used to create a sequence of bits that has the required c/c's for transmission - self clocking, no low frequencies, no wide bandwidth.
- It is implemented at the same time as encoding.
- It replaces 'unfriendly' runs of bits with a violation code that is easy to recognize.

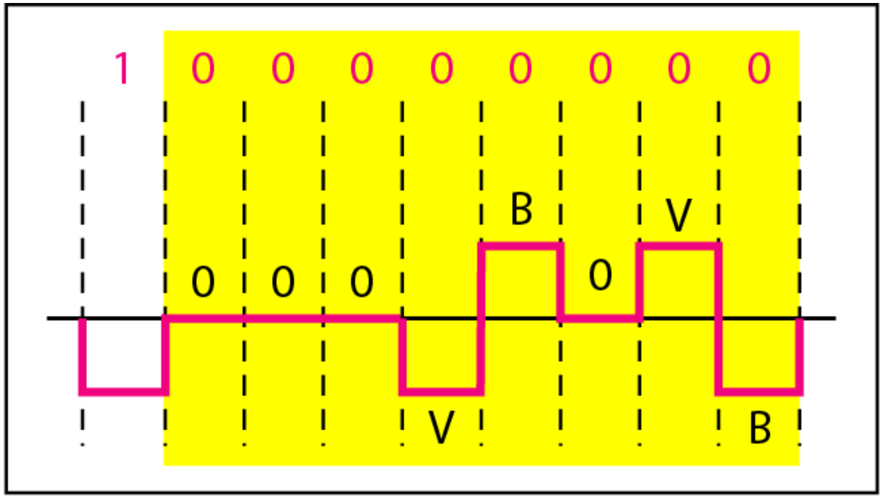


Scrambling : B8ZS

- B8ZS substitutes eight consecutive zeros with 000VB0VB.
- V stands for violation, It violates the line encoding rule.
- B stands for bipolar, it implements the bipolar line encoding rule.



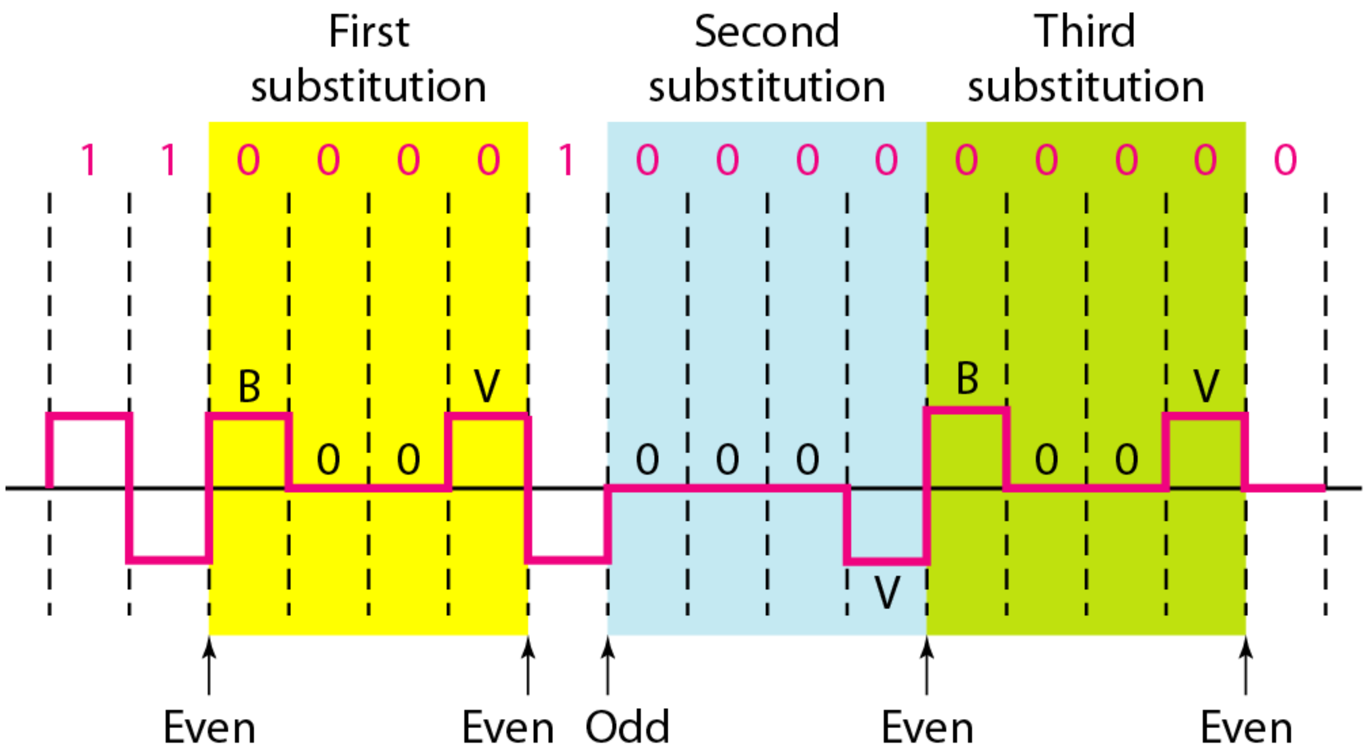
a. Previous level is positive.



b. Previous level is negative.

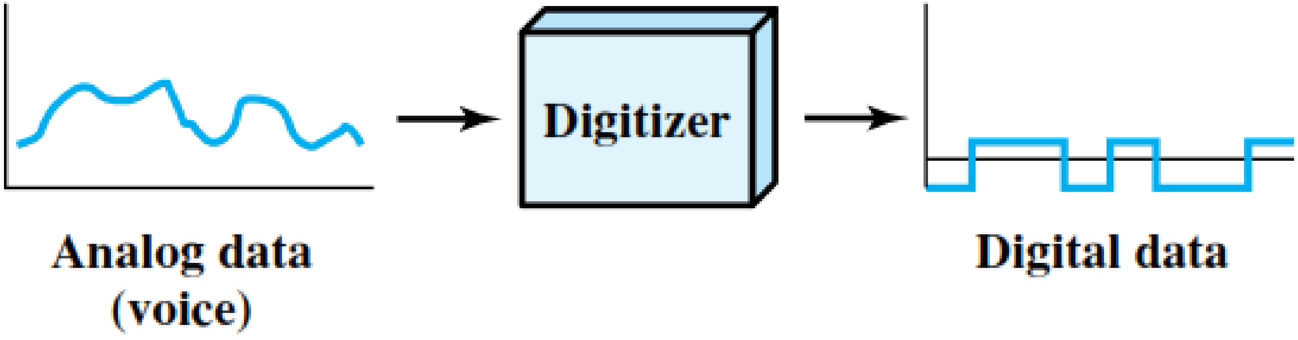
Scrambling : HDB3

- HDB3 substitutes four consecutive zeros with 000V or B00V.
- If # of non zero pulses is even the substitution is B00V.
- If # of non zero pulses is odd the substitution is 000V.



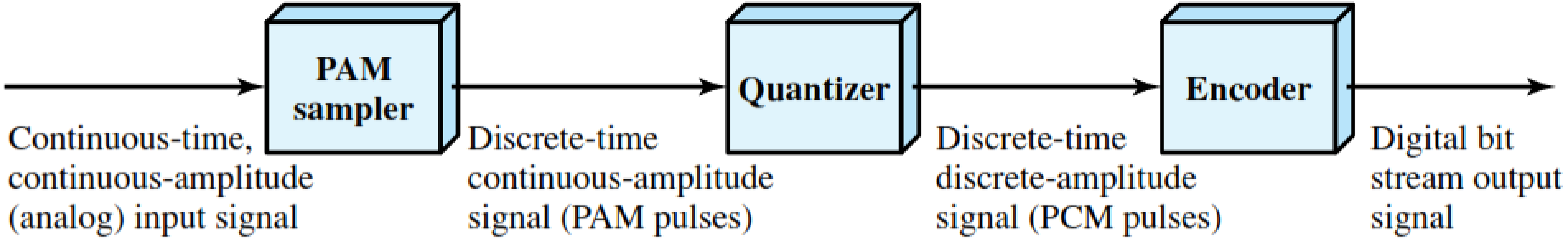
Analog-to-digital Conversion

- A digital signal is superior to an analog signal
- Because it is more robust to noise and can easily be recovered, corrected and amplified.
- The tendency today is to change an analog signal to digital data.
- The device used for converting analog data into digital form for transmission, and subsequently recovering the original analog data from the digital, is known as a codec (coder-decoder).
- Techniques for digitization
  - Pulse Code Modulation (PCM)
  - Delta Modulation (DM)



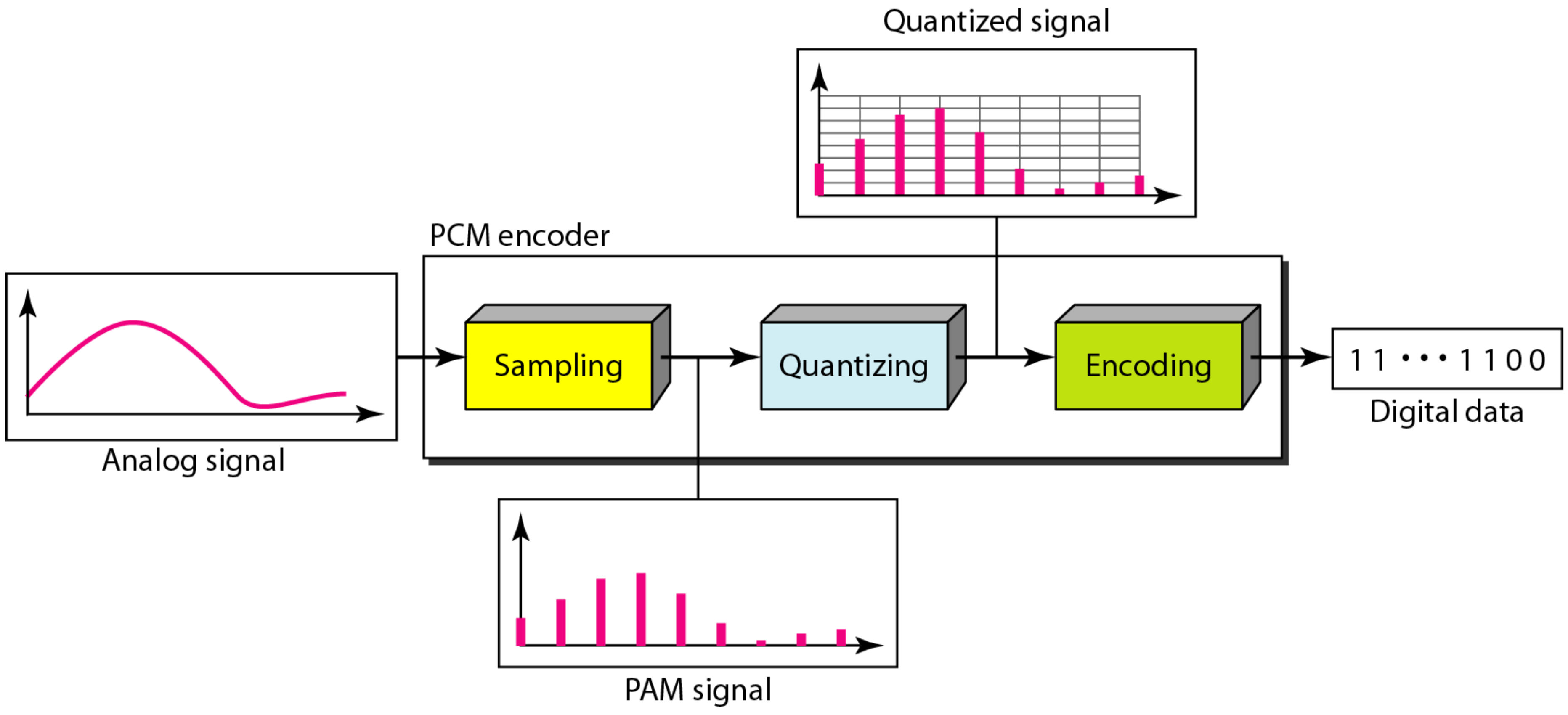
Pulse Code Modulation (PCM)

- PCM consists of three steps to digitize an analog signal:
  - Sampling
  - Quantization
  - Binary encoding



Pulse Code Modulation (PCM)

- Components of PCM encoder

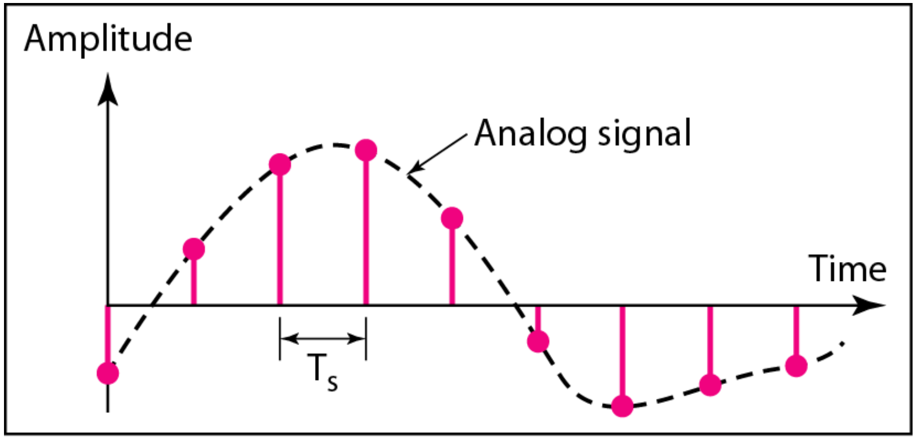


Pulse Code Modulation (PCM)

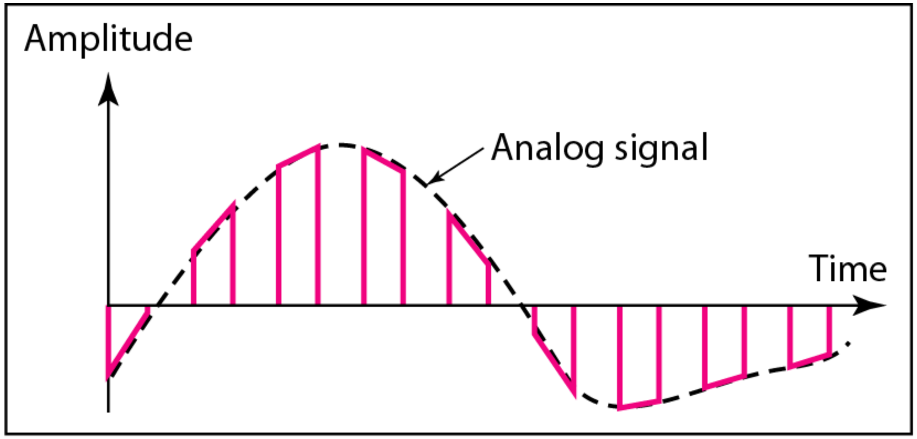
- Sampling
  - Analog signal is sampled every T_S secs.
  - T_S is referred to as the sampling interval.
  - $f_s = 1/ T_S$ is called the sampling rate or sampling frequency.
  - There are 3 sampling methods:
    - Ideal - an impulse at each sampling instant
    - Natural - a pulse of short width with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

Pulse Code Modulation (PCM)

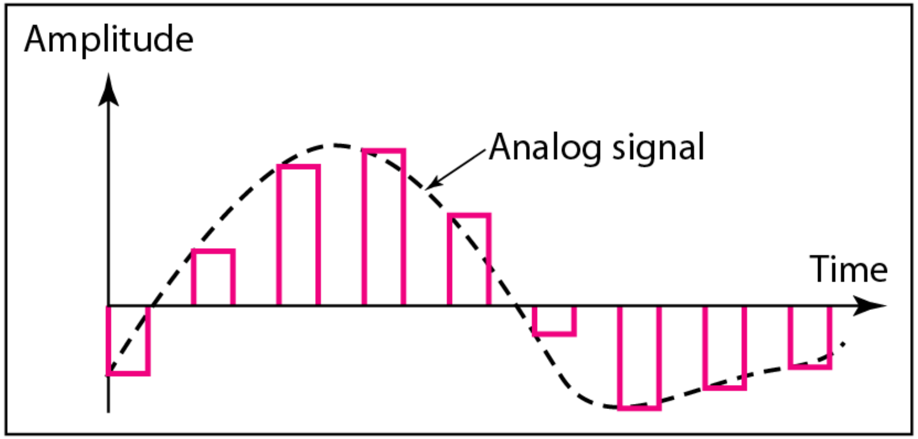
- Sampling



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

Pulse Code Modulation (PCM)

- Sampling
  - Pulse code modulation (PCM) is based on the sampling theorem:

SAMPLING THEOREM: If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal.

The function f(t) may be reconstructed from these samples by the use of a low pass filter.

Pulse Code Modulation (PCM)

- Example

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Pulse Code Modulation (PCM)

- 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

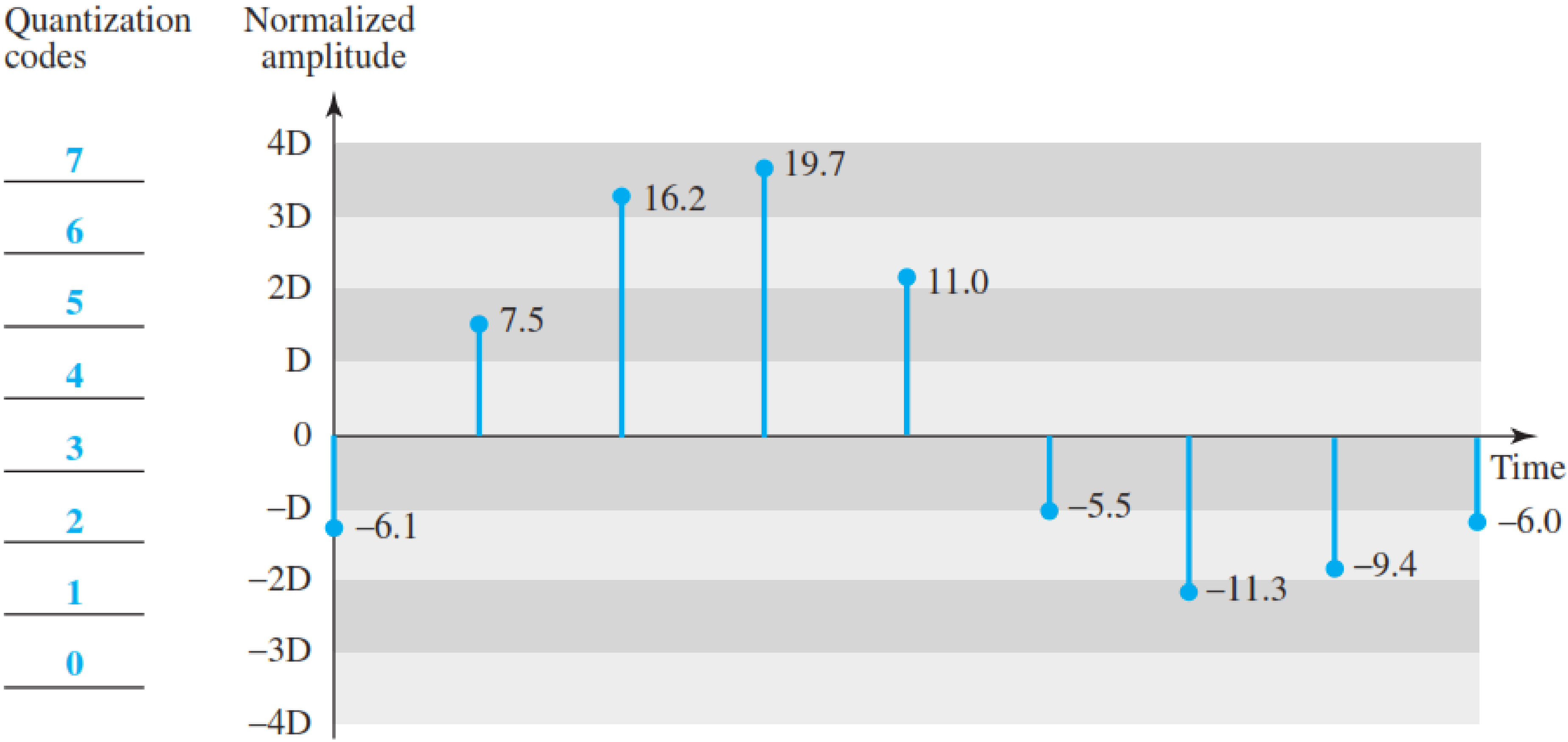
$$\Delta = \frac{\max - min}{L}$$

Pulse Code Modulation (PCM)

- 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.

- Encoding
  - Each quantized sample can be changed to an n-bit code word.
  - The bit rate can be found from the formula

$$\textbf{Bit rate} = \textbf{sampling rate} \times \textbf{number of bits per sample} = f_s \times n_b$$



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

Pulse Code Modulation (PCM)

- 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

The signal-to-noise ratio for quantizing noise can be expressed as

$$\mathbf{SN_QR_{dB} = 6.02n + 1.76\ dB}$$

Pulse Code Modulation (PCM)

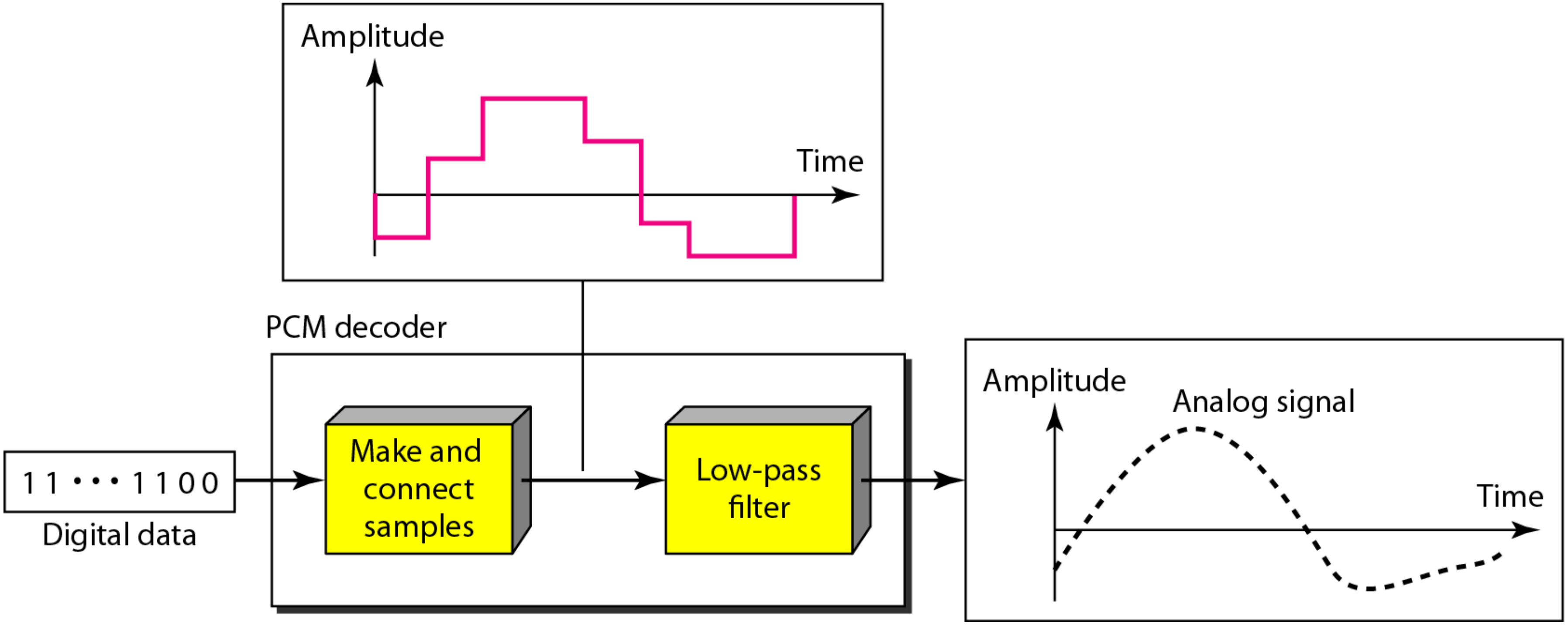
- Quantization Error
  - Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.
  - Non linear quantization is used to alleviate this problem. Goal is to keep SN_QR fixed for all sample values.
  - Two approaches:
    - 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.

Pulse Code Modulation (PCM)

- PCM Decoder
  - To recover an analog signal from a digitized signal we follow the following steps:
    - We use a hold circuit that holds the amplitude value of a pulse till the next pulse arrives.
    - We pass this signal through a low pass filter with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.
  - The higher the value of L, the less distorted a signal is recovered.

Pulse Code Modulation (PCM)

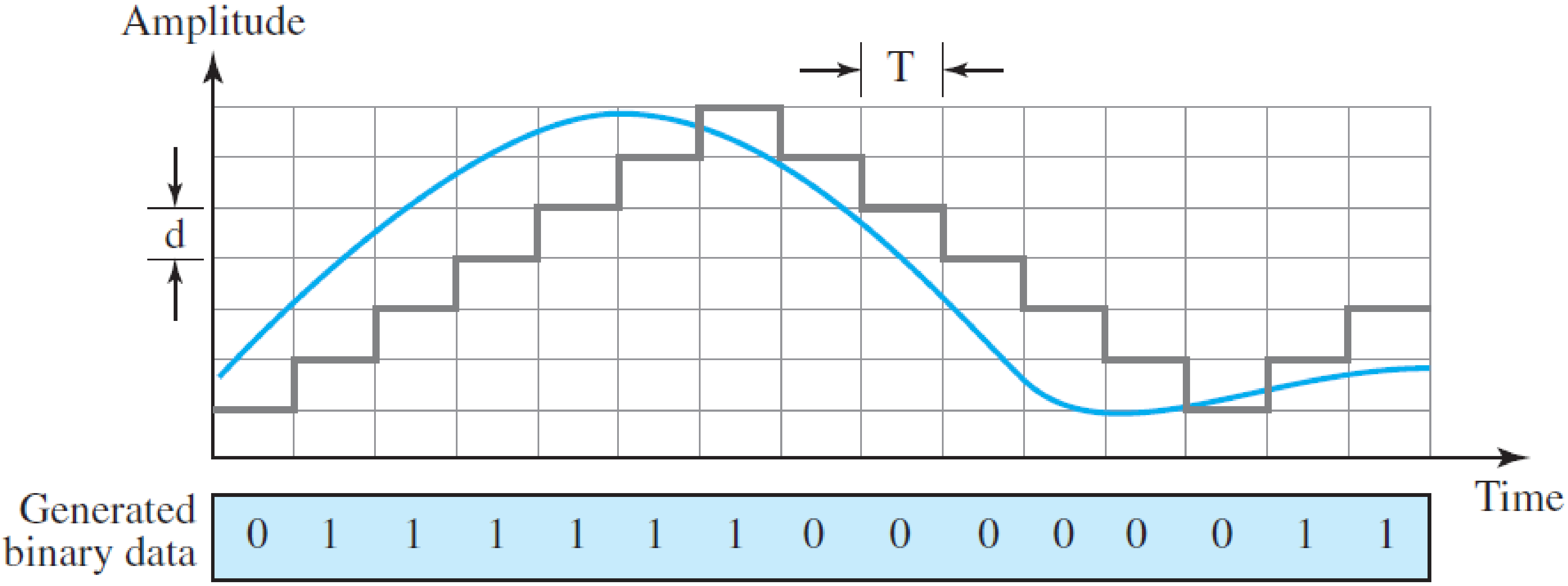
- PCM Decoder



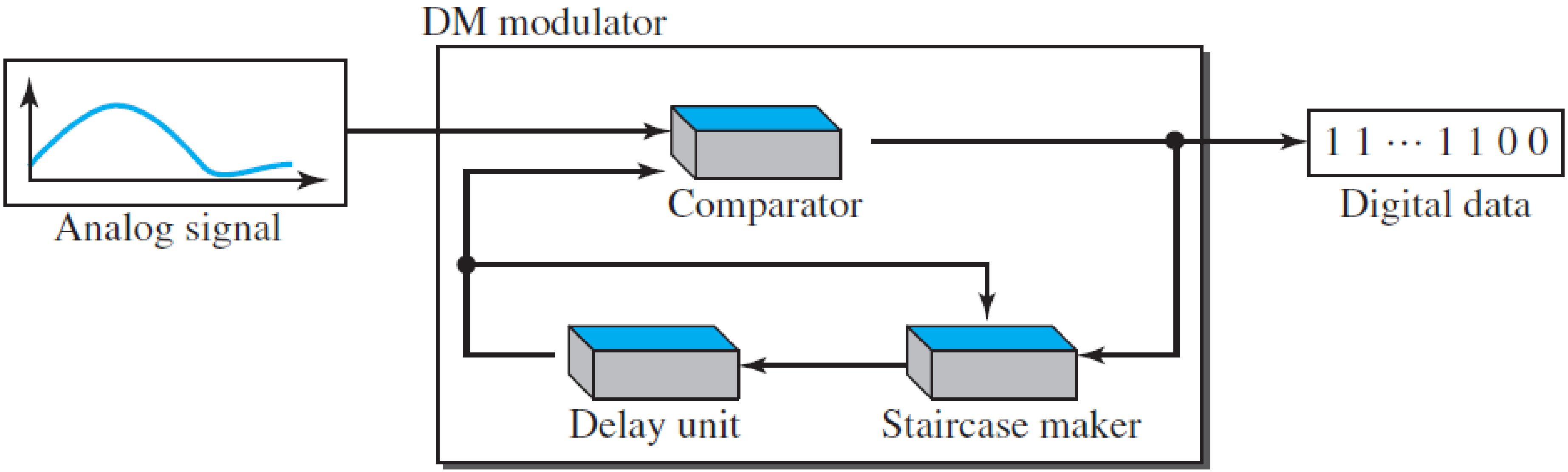
Delta Modulation (DM)

- Delta modulation is simplest technique.
- DM finds the change in amplitude from the previous sample.
- Modulator
  - The process records the small positive or negative changes, called delta δ.
  - If the δ is positive, the process records a 1
  - If the δ is negative, the process records a 0

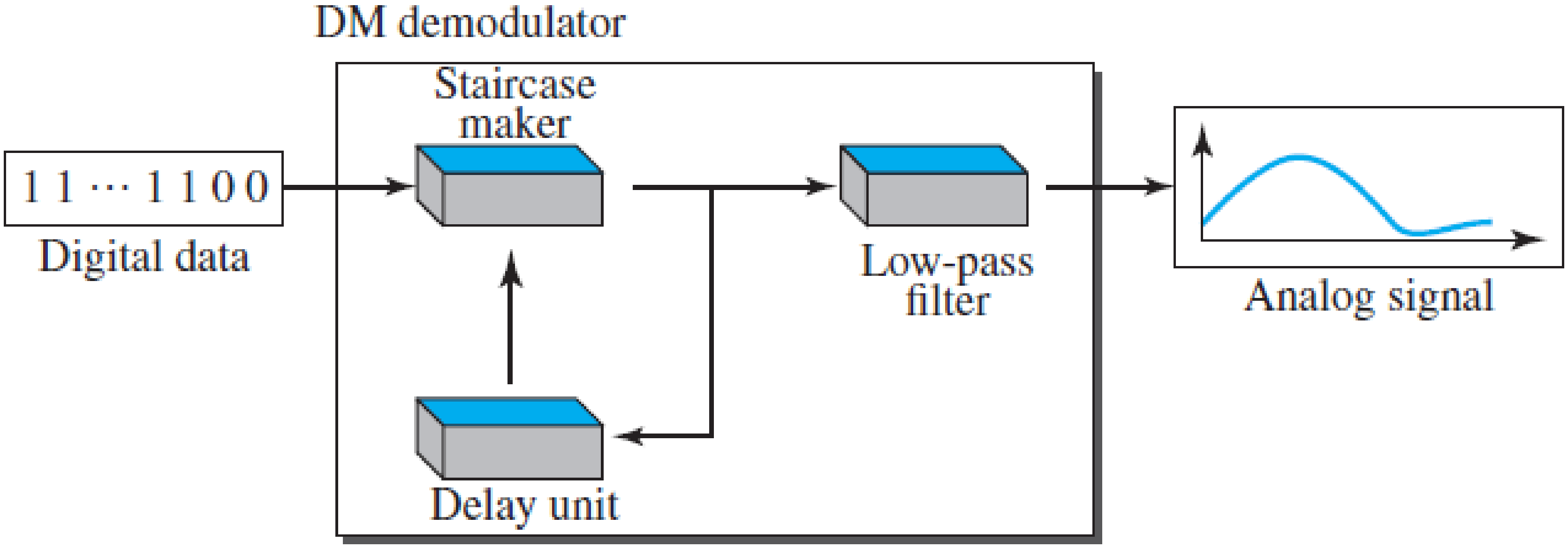
Delta Modulation (DM)



Delta Modulation (DM)



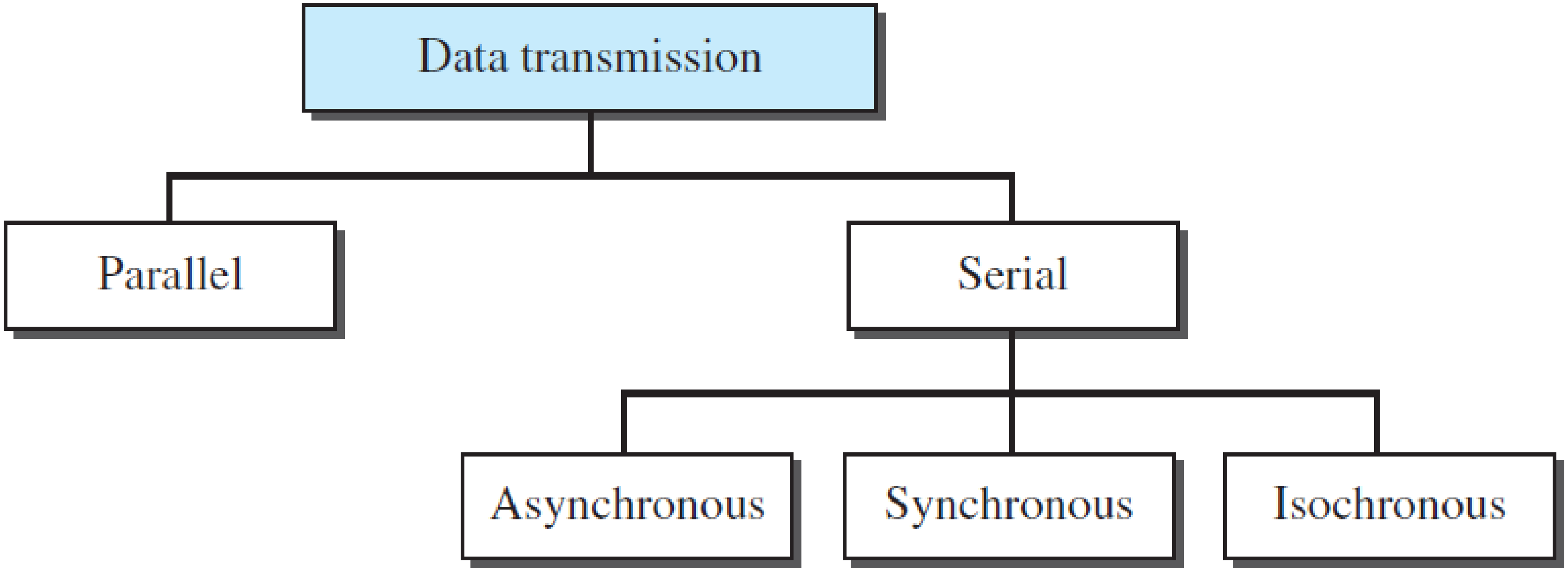
Delta Modulation (DM)



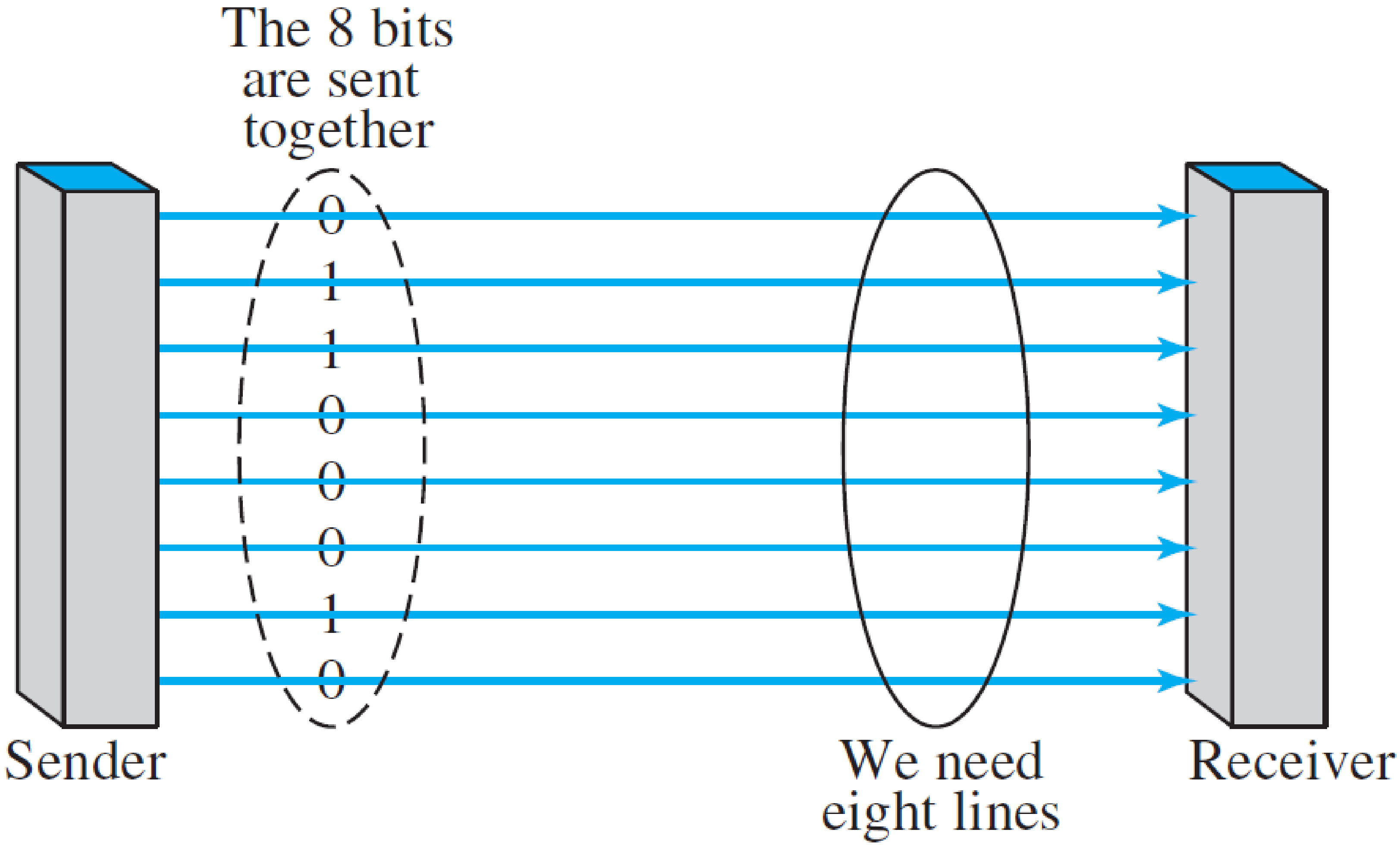
Delta Modulation (DM)

- Adaptive DM
  - The value of delta is not fixed.
  - The value of delta changes according to the amplitude.
- Quantization error of DM is less than PCM.

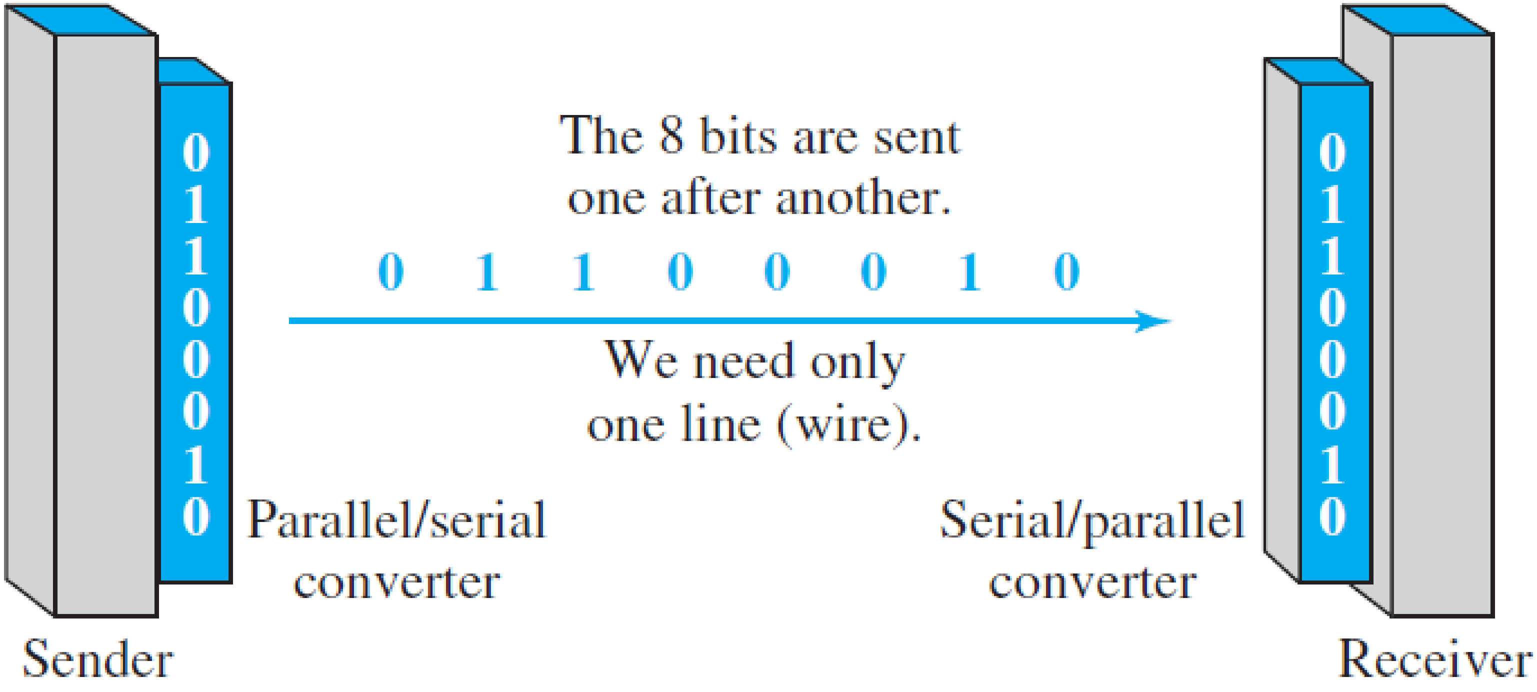
Transmission Modes



Parallel Transmission

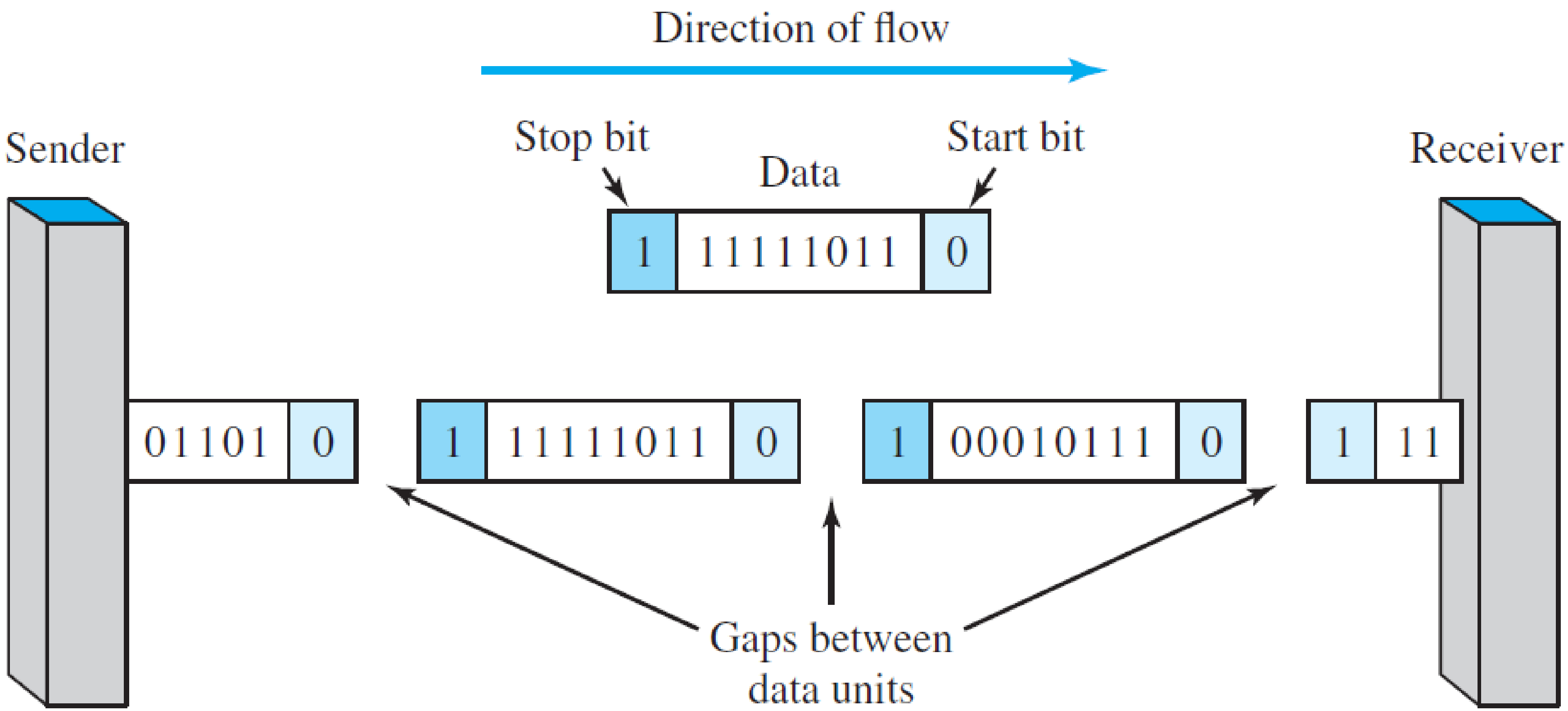


Serial Transmission



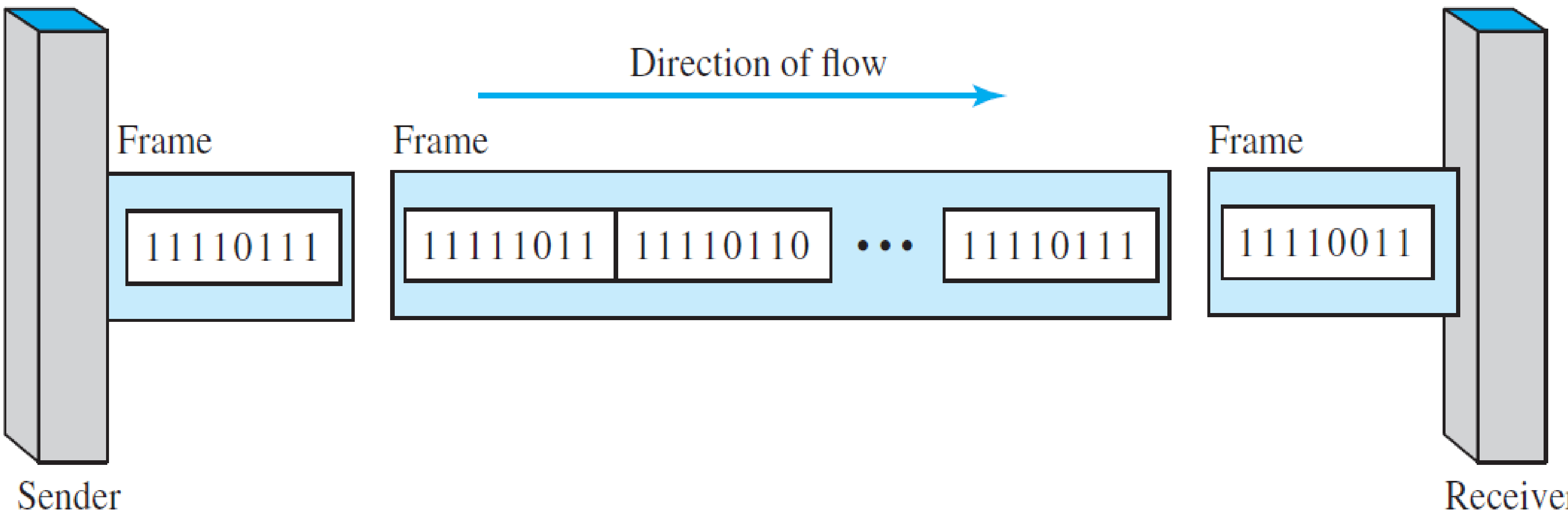
Asynchronous Transmission

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between bytes.



Synchronous Transmission

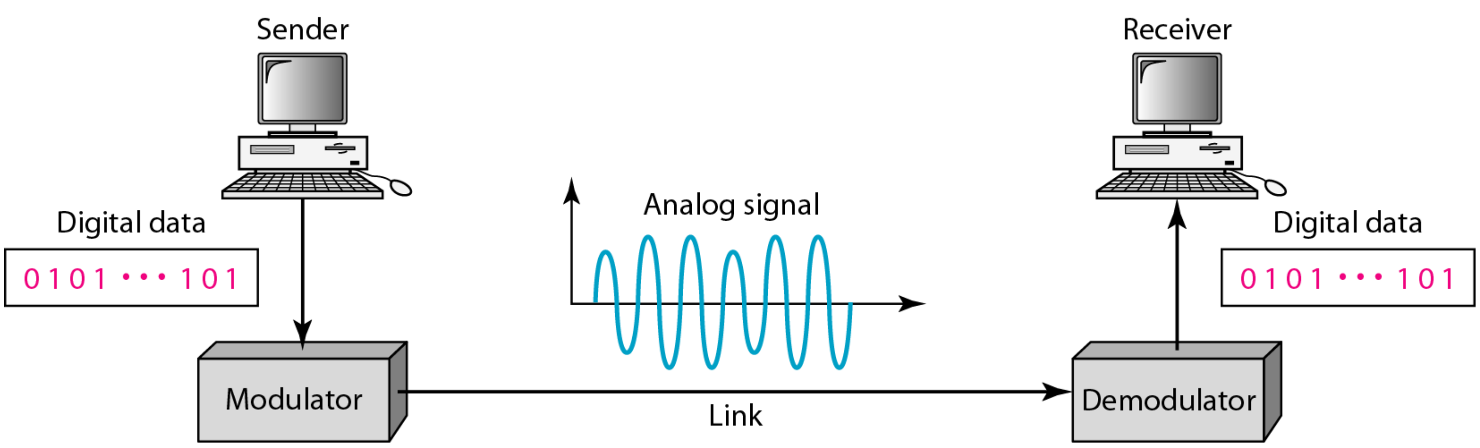
In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.



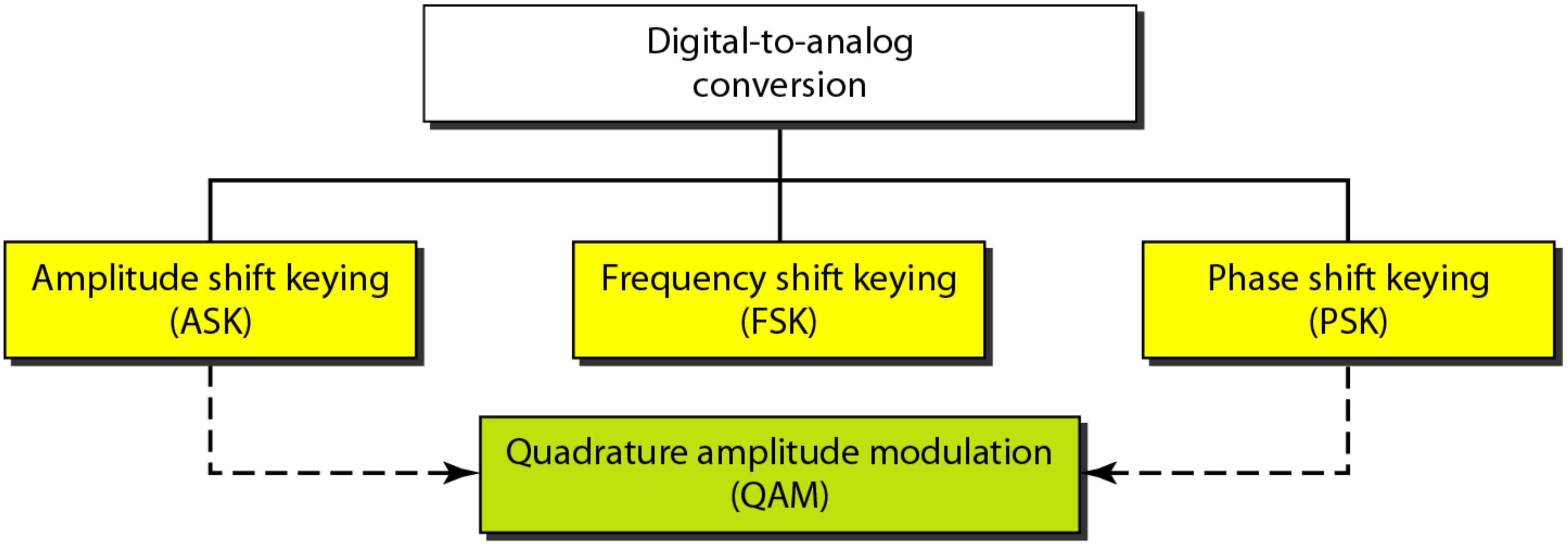
Analog Transmission

Digital-to-Analog Conversion

- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency fc) performs the function of transporting the digital data in an analog waveform.
- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.
- Characteristics are:
  - Amplitude
  - Frequency
  - Phase



Types of digital-to-analog conversion



Data Rate versus Signal Rate

- In the analog transmission of digital data, the signal or baud rate is less than or equal to the bit rate.

  Signal rate (S) = Data rate (N)/r bauds

  Where r = number of data bits carried by each signal element

- Bandwidth
  - The required bandwidth for analog transmission of digital data is proportional to the signal rate except for FSK
  - The difference between the carrier signals needs to be added in FSK.
- Carrier Signal
  - The sending device produces a high-frequency signal that acts as a base for the information signal.

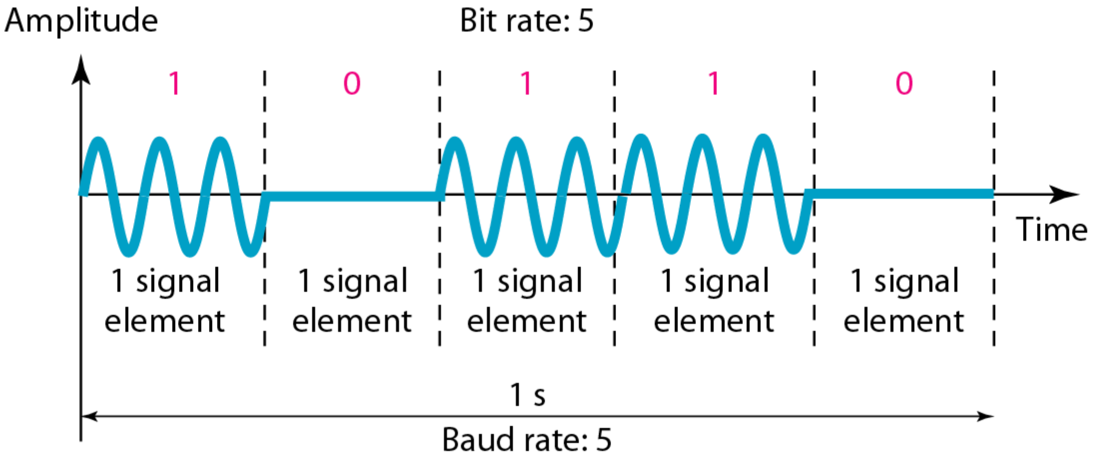
Amplitude Shift Keying (ASK)

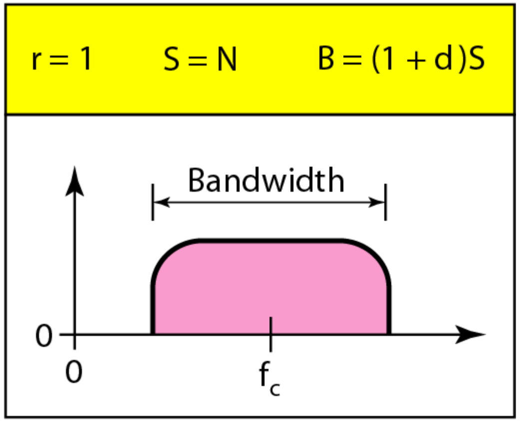
- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- The bandwidth B of ASK is proportional to the signal rate S.

$$\mathbf{B = (1+d)S}$$

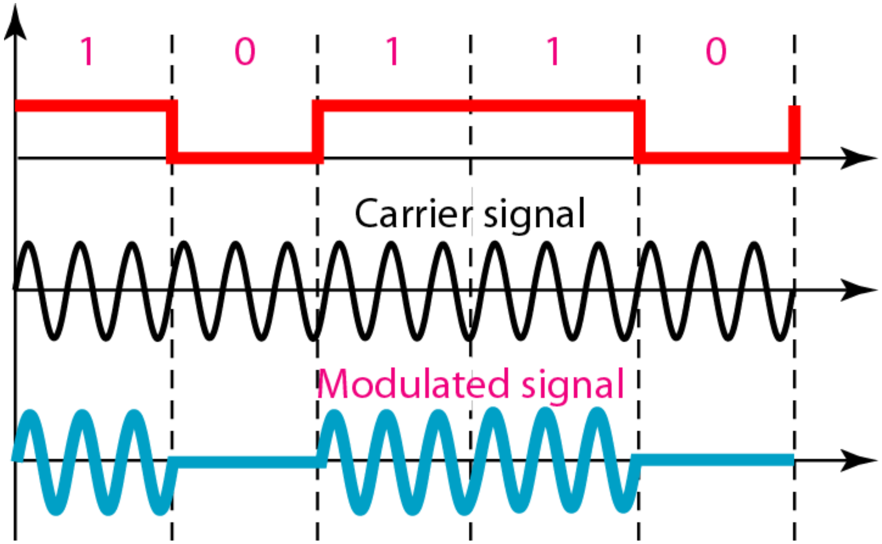
- “d” is due to modulation and filtering, lies between 0 and 1.

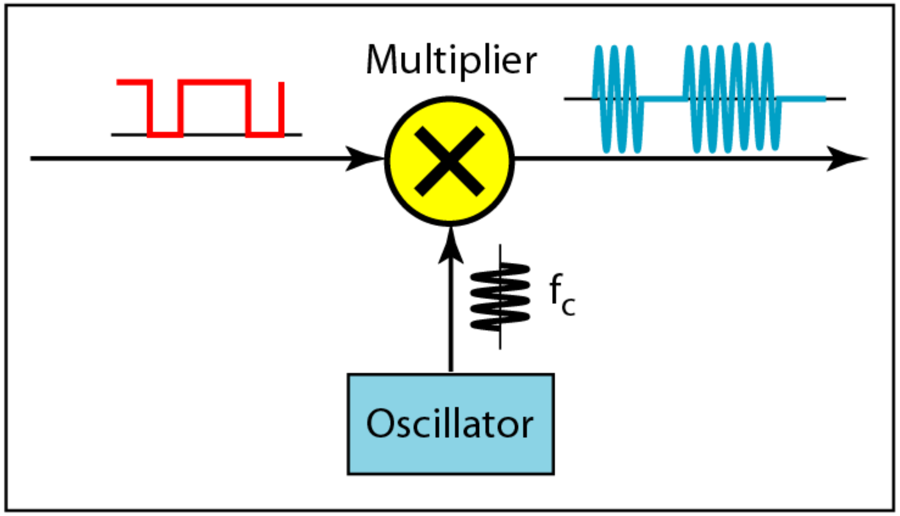
Binary ASK (BASK) or on-off keying (OOK).





Implementation of binary ASK



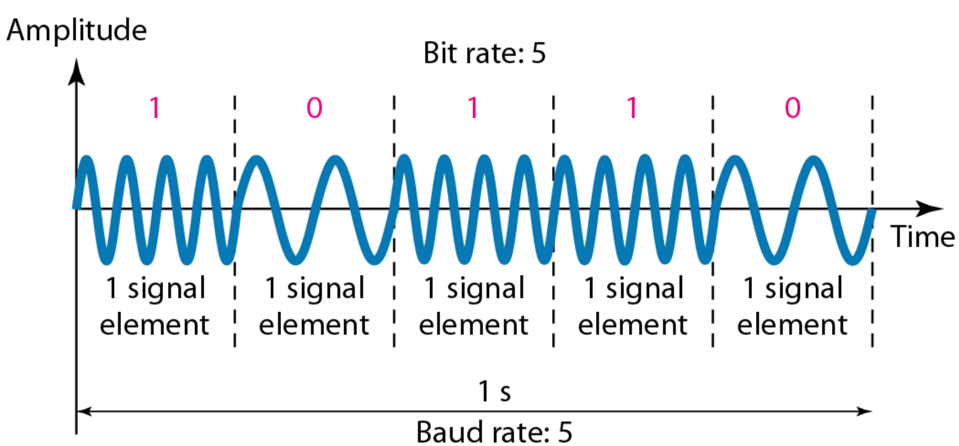


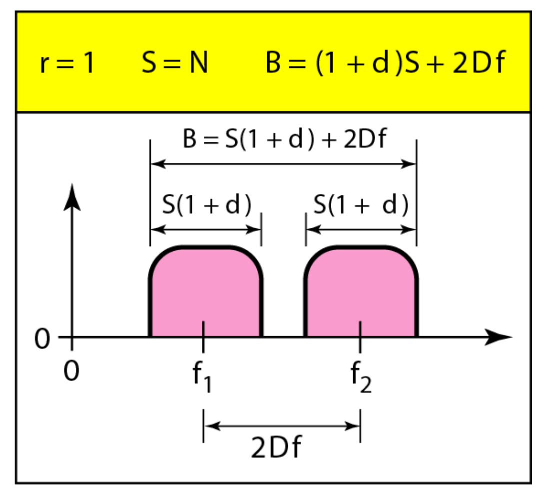
Frequency Shift Keying (FSK)

- The digital data stream changes the frequency of the carrier signal, fc.
- a “0” could be represented by $f_1=f_c - \Delta f$
- a “1” could be represented by $f_2=f_c+\Delta f$
- The bandwidth B of FSK is:

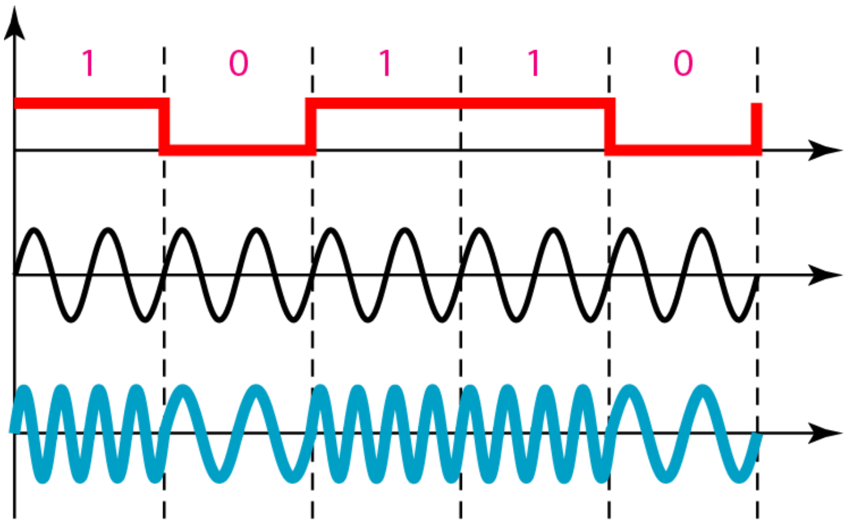
$$\mathbf{B = (1+d)S +2\,\Delta f}$$

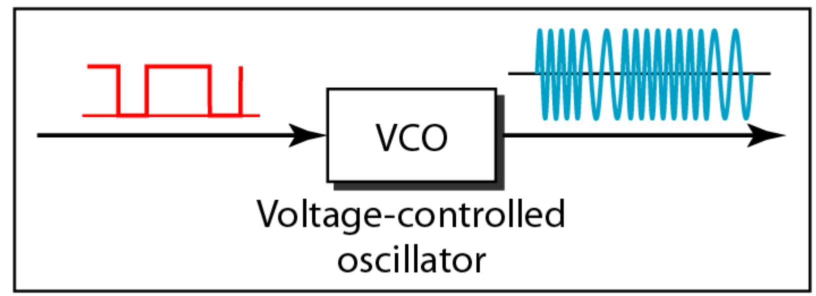
Binary FSK (BFSK)





Implementation of binary FSK



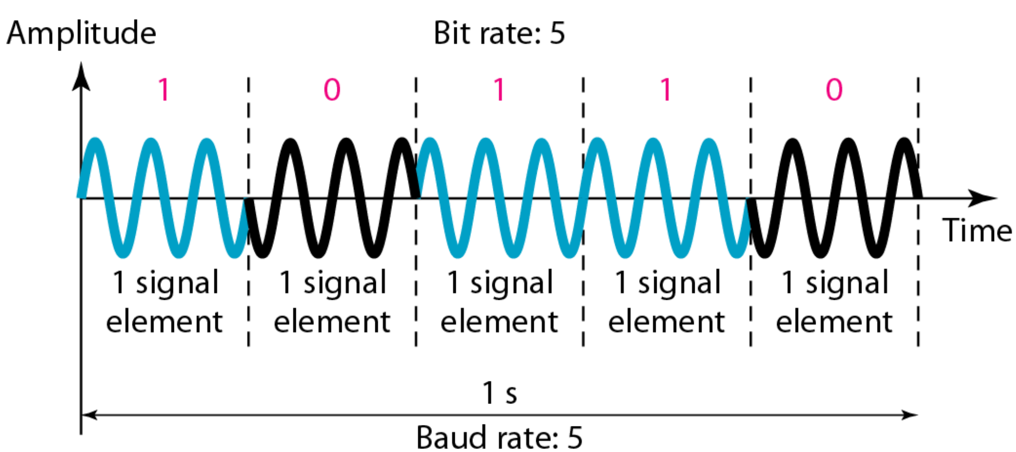


Phase Shift Keying (PSK)

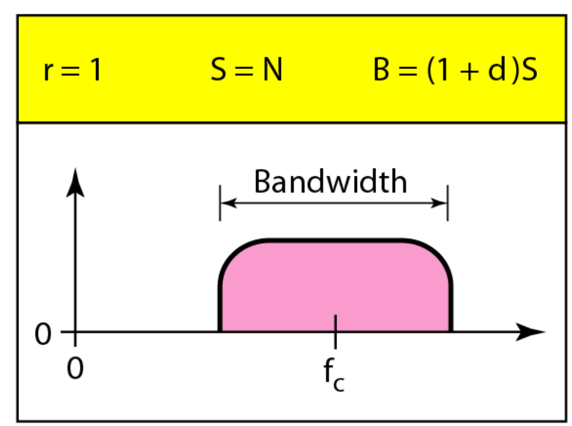
- The digital data stream changes the phase of the carrier signal, fc.
- a “0” could be represented 180 degree
- a “1” could be represented 0 degree
- The bandwidth B of PSK is:

$$\mathbf{B = (1+d)S}$$

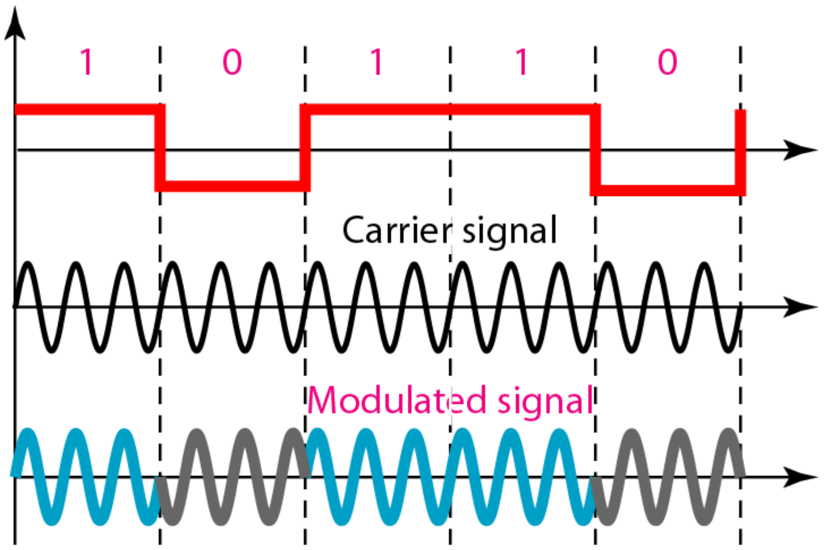
- PSK is much more robust than ASK as it is not that vulnerable to noise

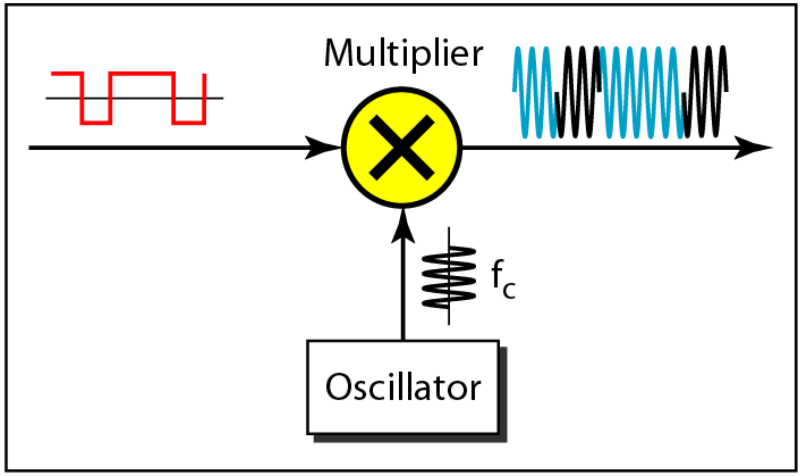


Binary PSK (BPSK)



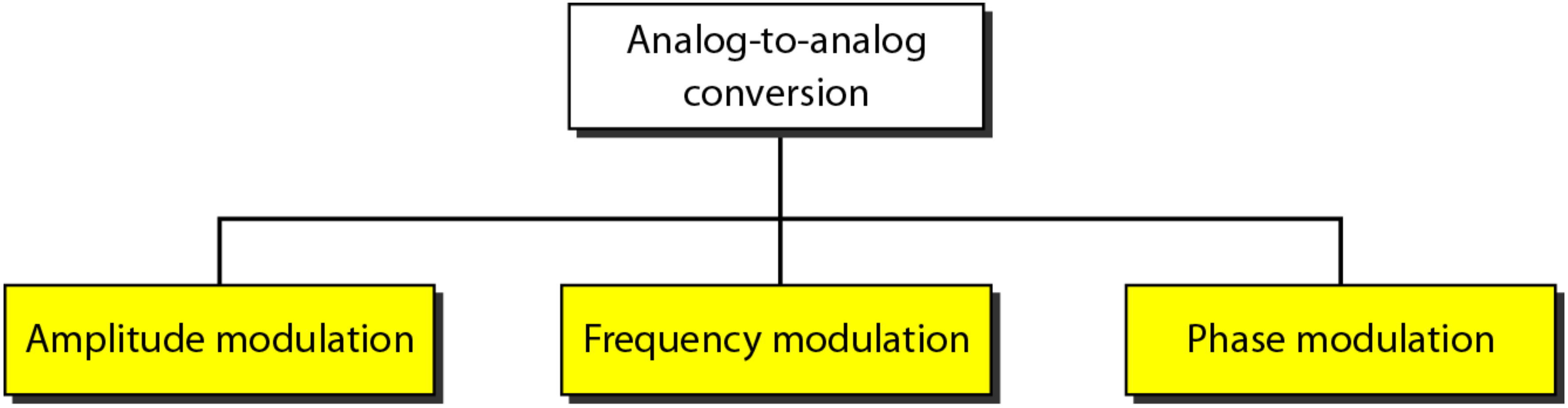
Implementation of binary PSK





Analog-to-Analog Conversion

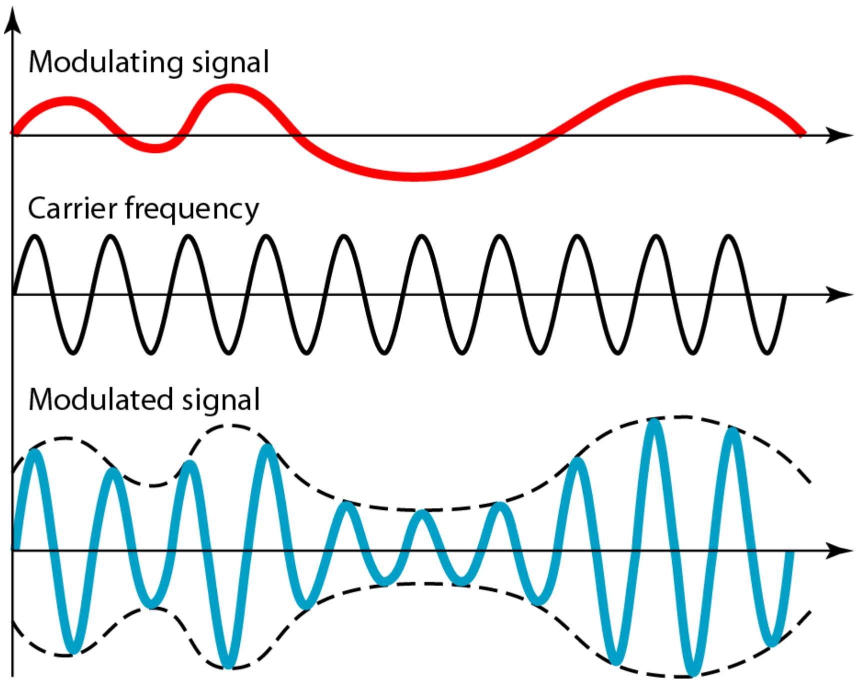
- Analog-to-analog conversion is the representation of analog information by an analog signal.
- Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us.

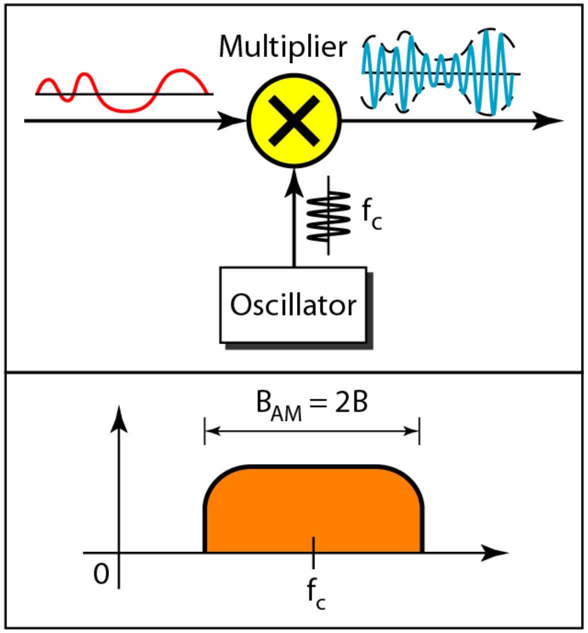


Amplitude Modulation(AM)

- A carrier signal is modulated only in amplitude value.
- The modulating signal is the envelope of the carrier.
- The required bandwidth

$$\mathbf{B_{AM} = 2B}$$

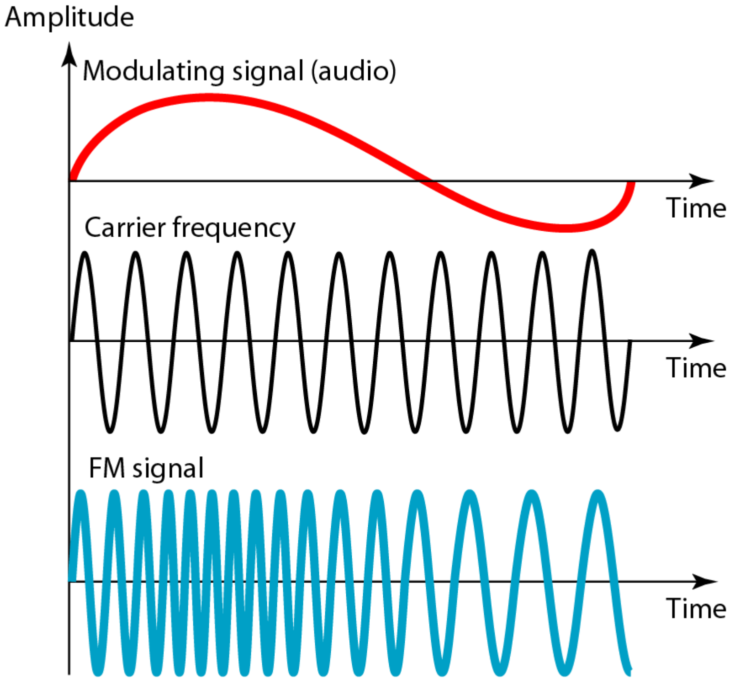


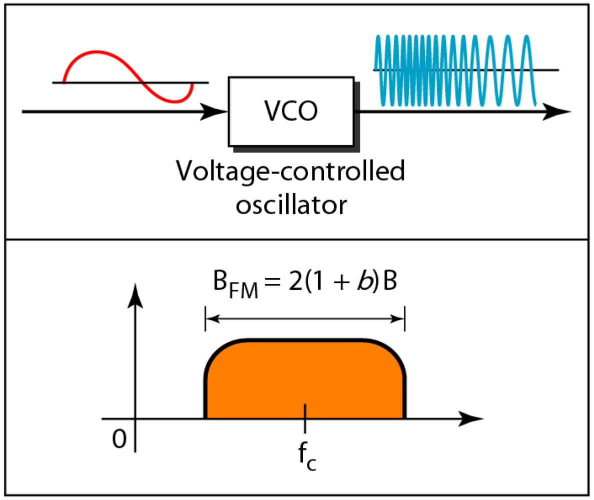


Frequency Modulation(FM)

- The modulating signal changes the freq. fc of the carrier signal.
- The bandwidth for FM is high.
- The required bandwidth

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



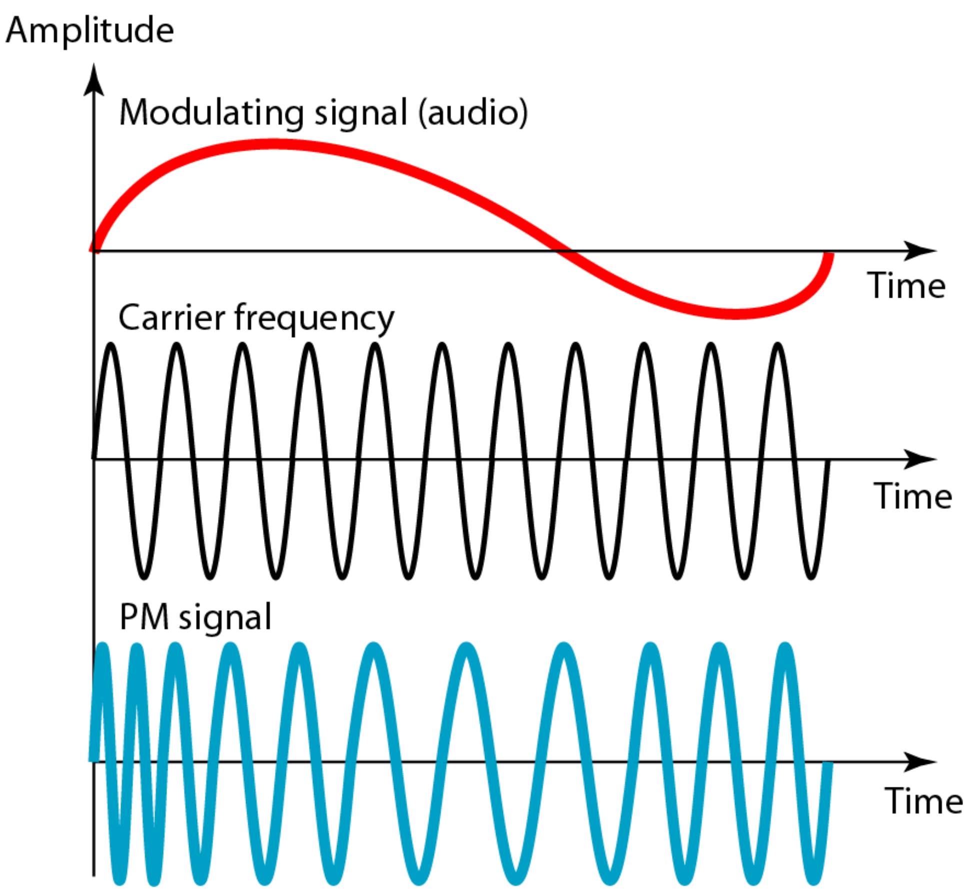


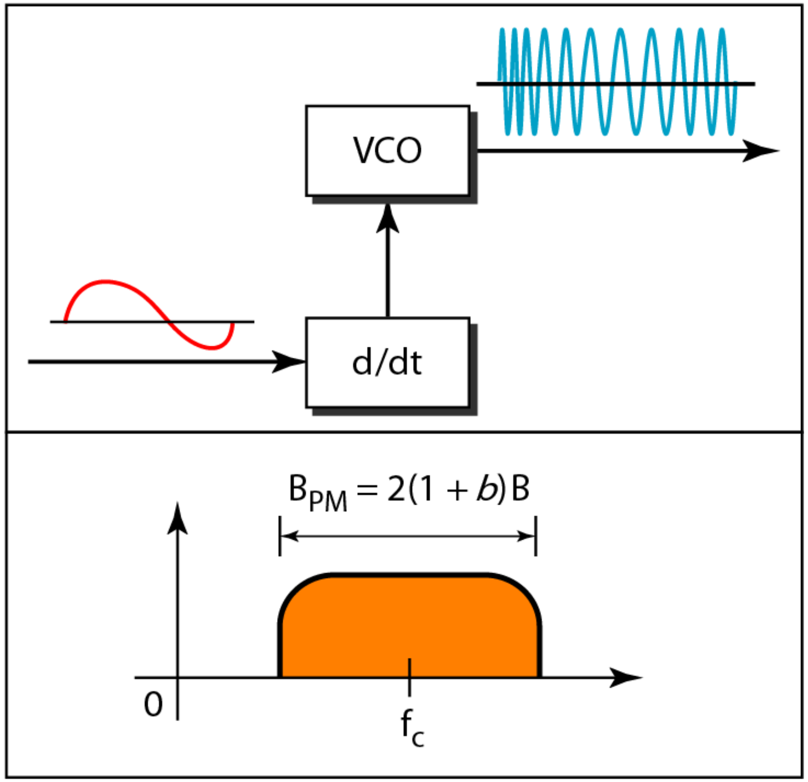
Phase Modulation (PM)

- 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 AM
- The required bandwidth

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

Phase Modulation (PM)





Multiplexing Techniques

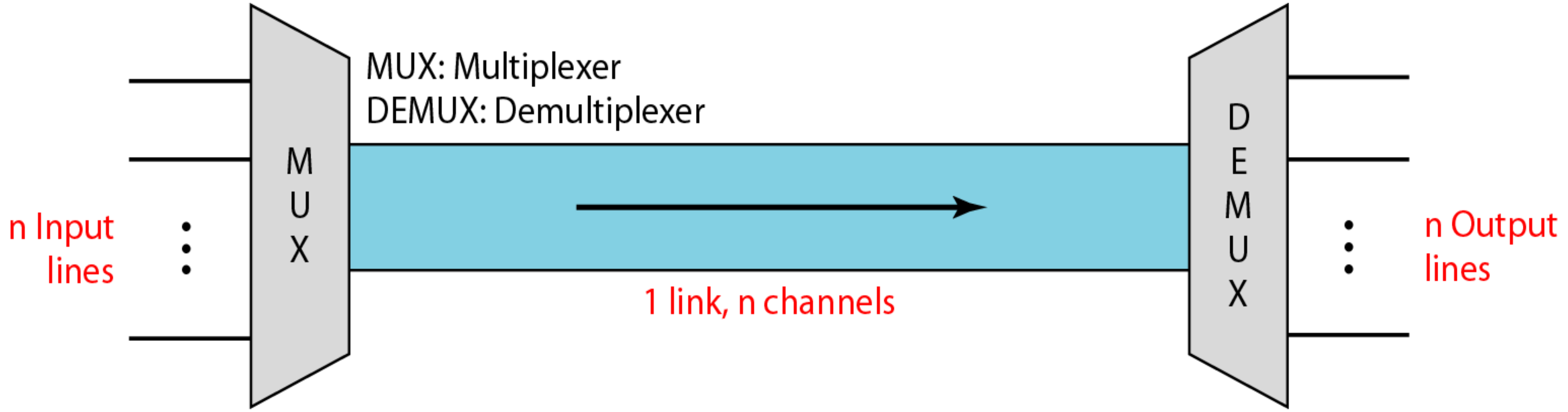
Bandwidth Utilization: Multiplexing

- In real life, we have links with limited bandwidths
- The wise use of these bandwidths has been, and will be, one of the main challenges in data communications
- **Bandwidth utilization** is the wise use of available bandwidth to achieve specific goals
- Efficiency can be achieved by multiplexing
- Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared
- **Multiplexing** is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link

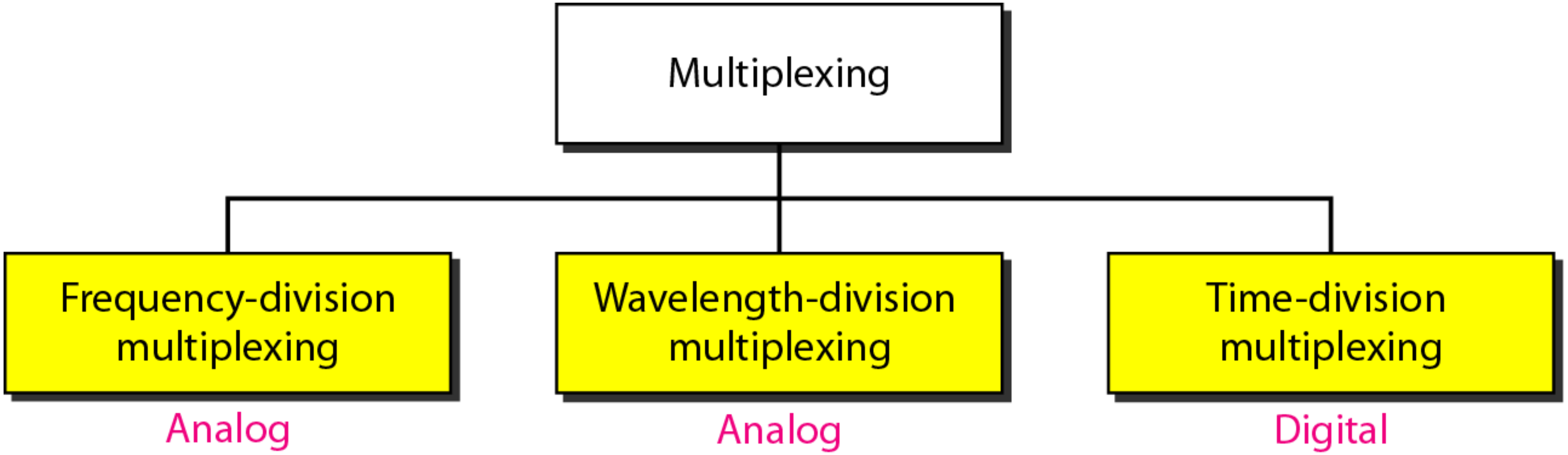
Multiplexing

- As data and telecommunications use increases, so does traffic
- We can accommodate this increase by continuing to add individual links each time a new channel is needed; or we can install higher-bandwidth links and use each to carry multiple signals
- In a multiplexed system, *n* lines share the bandwidth of one link

Dividing a link into channels



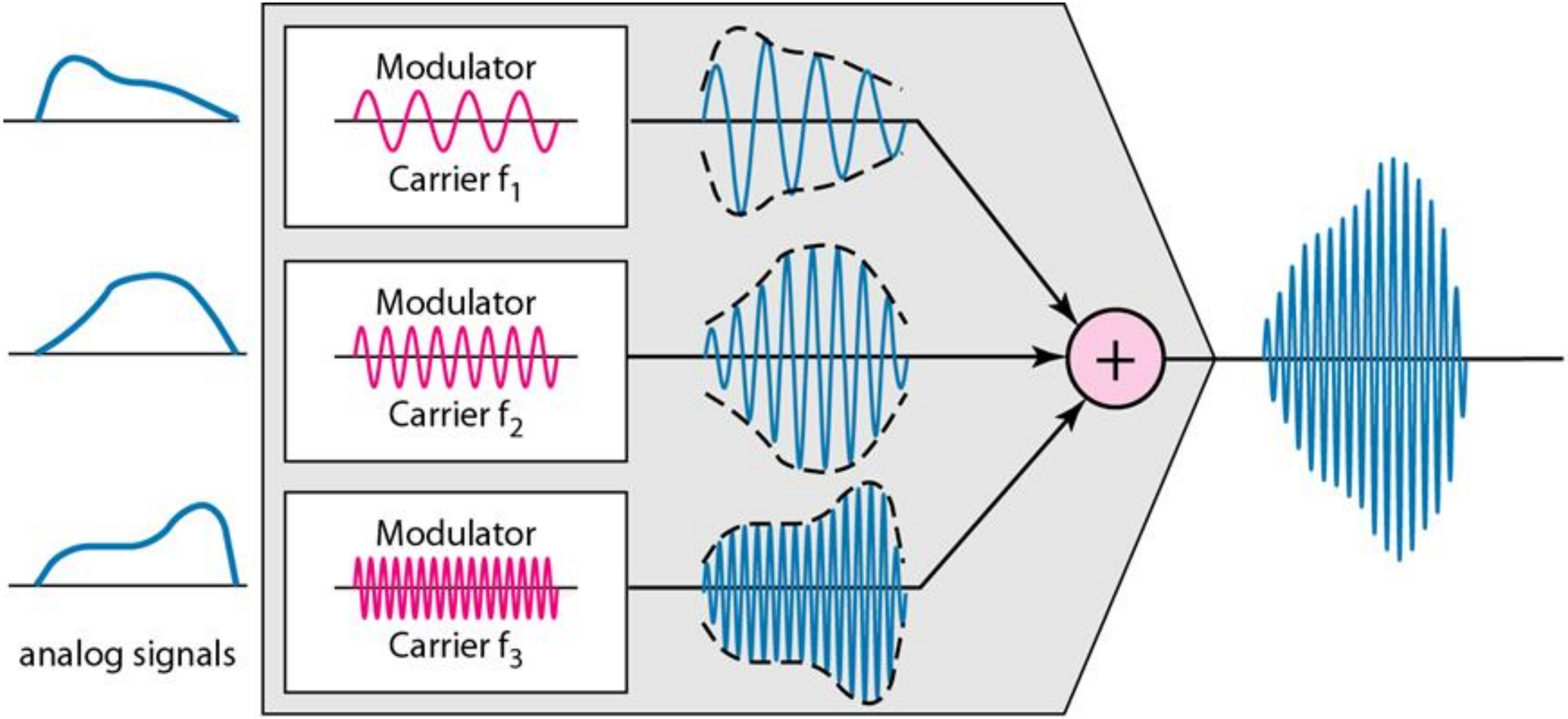
Categories of multiplexing



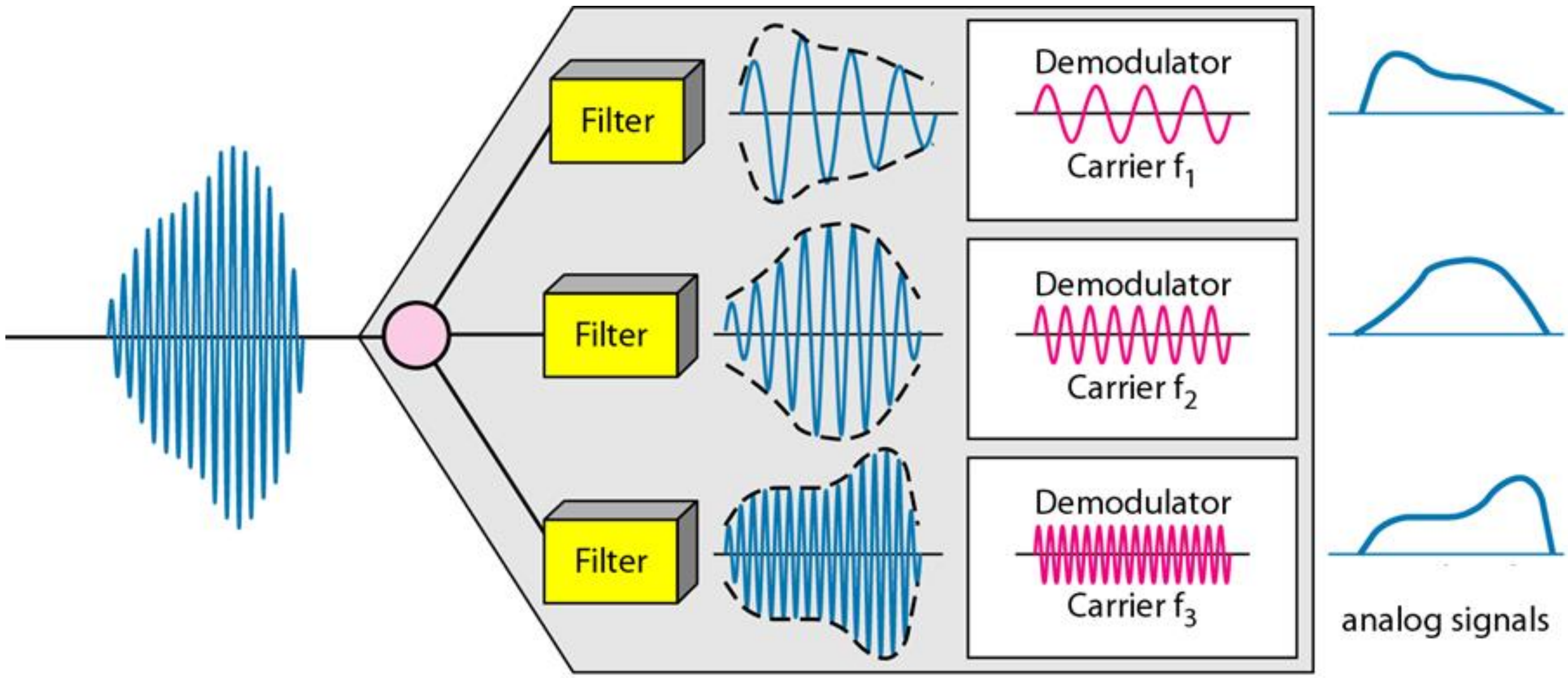
Frequency-division multiplexing (FDM)

- FDM is an analog technique that can be applied when the bandwidth of a link is greater than the combined bandwidths of the signals to be transmitted
- In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link
- Channels can be separated by strips of unused bandwidth - **guard bands** - to prevent signals from overlapping
- **Applications:** AM and FM radio broadcasting; television broadcasting; telephone networks etc.

FDM multiplexing process



FDM demultiplexing process



Example

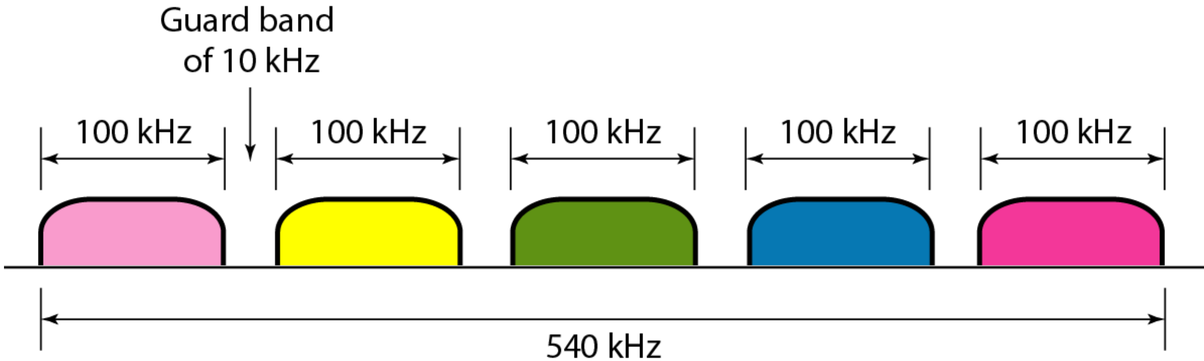
Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least

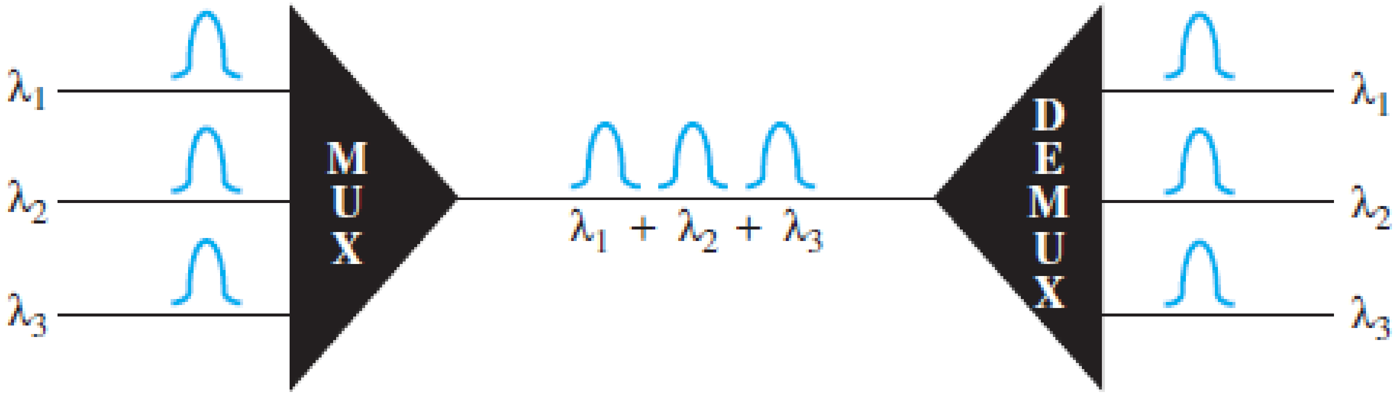
$$5 \times 100 + 4 \times 10 = 540 \text{ kHz},$$

as shown in Figure

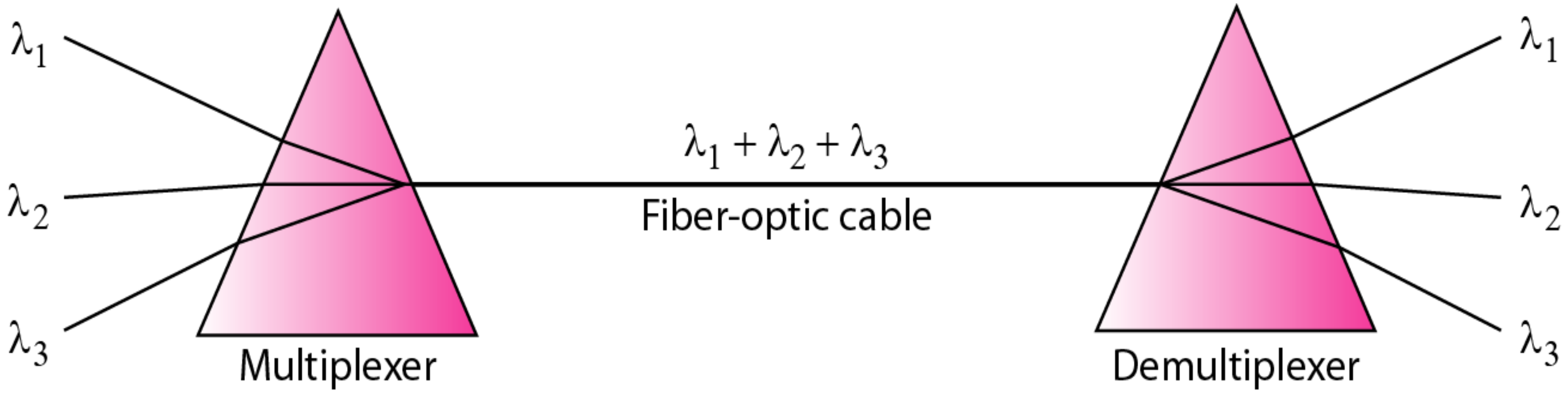


Wavelength-division multiplexing (WDM)

- WDM is an analog multiplexing technique to combine optical signals
- It is designed to use the high-data-rate capability of fiber-optic cable
- It is conceptually the same as FDM, except that the multiplexing and demultiplexing involve optical signals. In WDM, we combine different signals of different frequencies. The difference is that the frequencies are very high
- **Application:** SONET (synchronous optical network) network

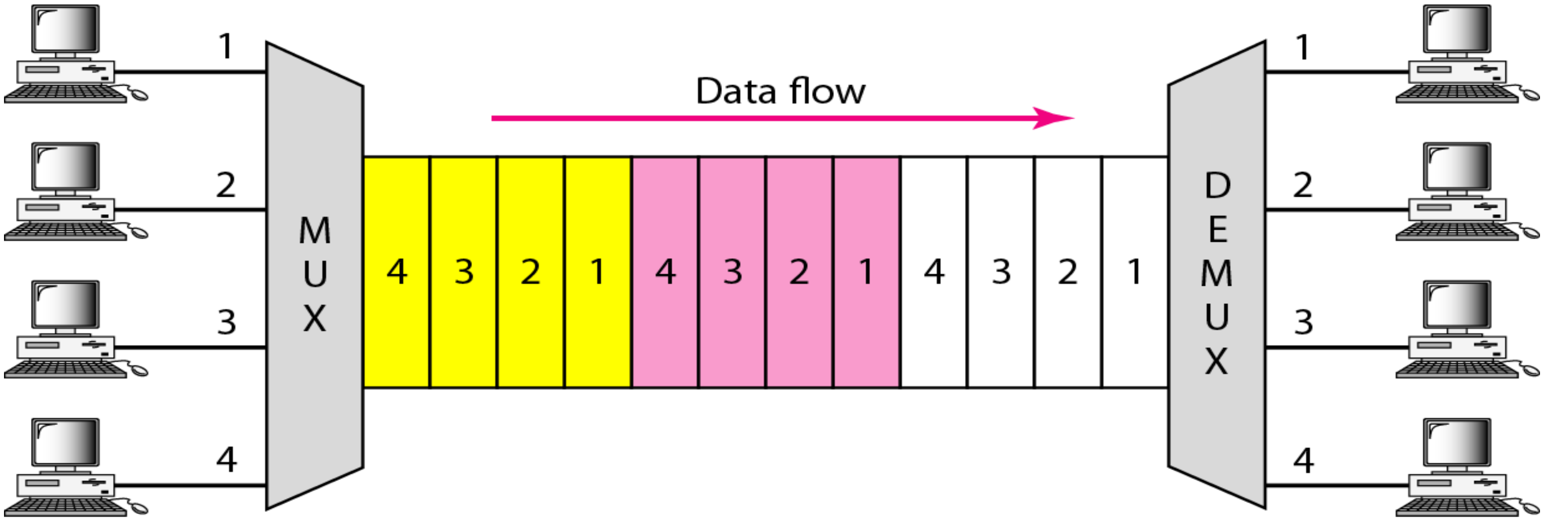


Prisms in wavelength-division multiplexing and demultiplexing



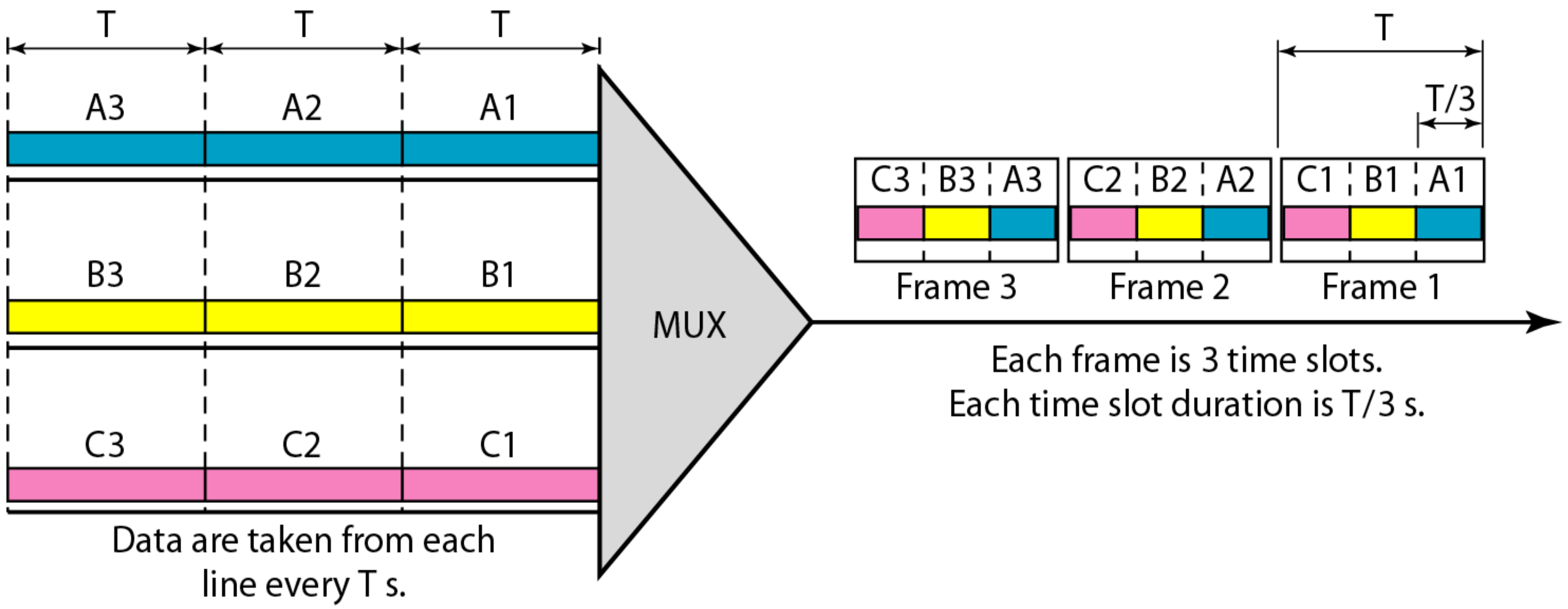
Time-Division Multiplexing (TDM)

- TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate link
- TDM allows several connections to share the high bandwidth of a link
- Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link
- **Application:** Telephone networks



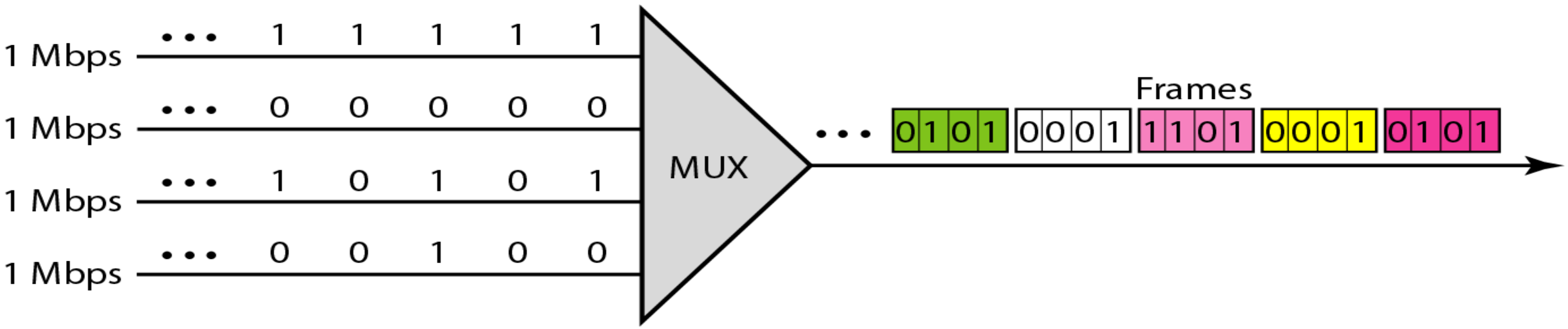
Synchronous time-division multiplexing

- In synchronous TDM, each input connection has an allotment in the output even if it is not sending data
- In synchronous TDM, the data rate of the link is n times faster, and the unit duration is n times shorter



Example 1

Following figure shows synchronous TDM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, and (c) the output bit rate.



Solution:

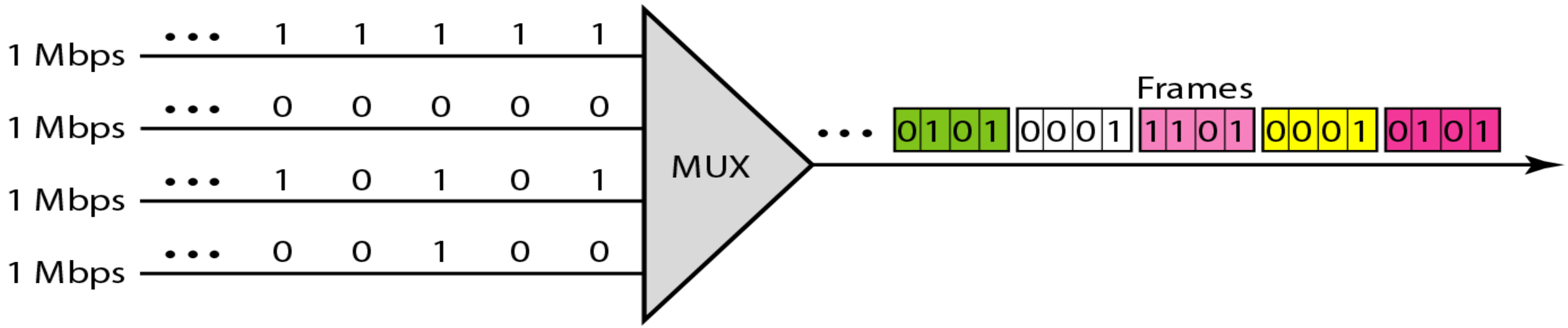
We can answer the questions as follows:

a. The input bit duration is the inverse of the bit rate:

  1/1 Mbps = 1 μs

b. The output bit duration is one-fourth of the input bit duration, or ¼ μs

Example 1 (Contd…)



c. The output bit rate is the inverse of the output bit duration or 1/(4μs) or 4 Mbps.

This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps

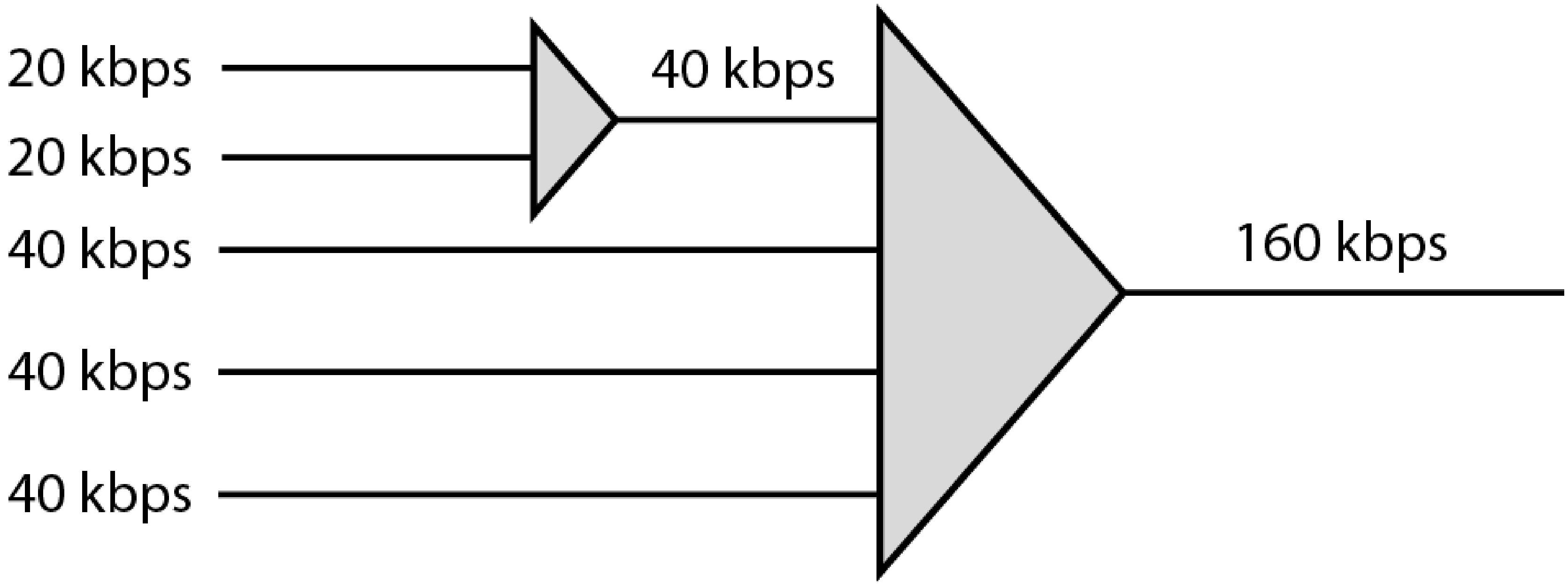
Data Rate Management

- Not all input links maybe have the same data rate.
- Some links maybe slower. There maybe several different input link speeds
- There are three strategies that can be used to overcome the data rate mismatch: multilevel, multislot and pulse stuffing

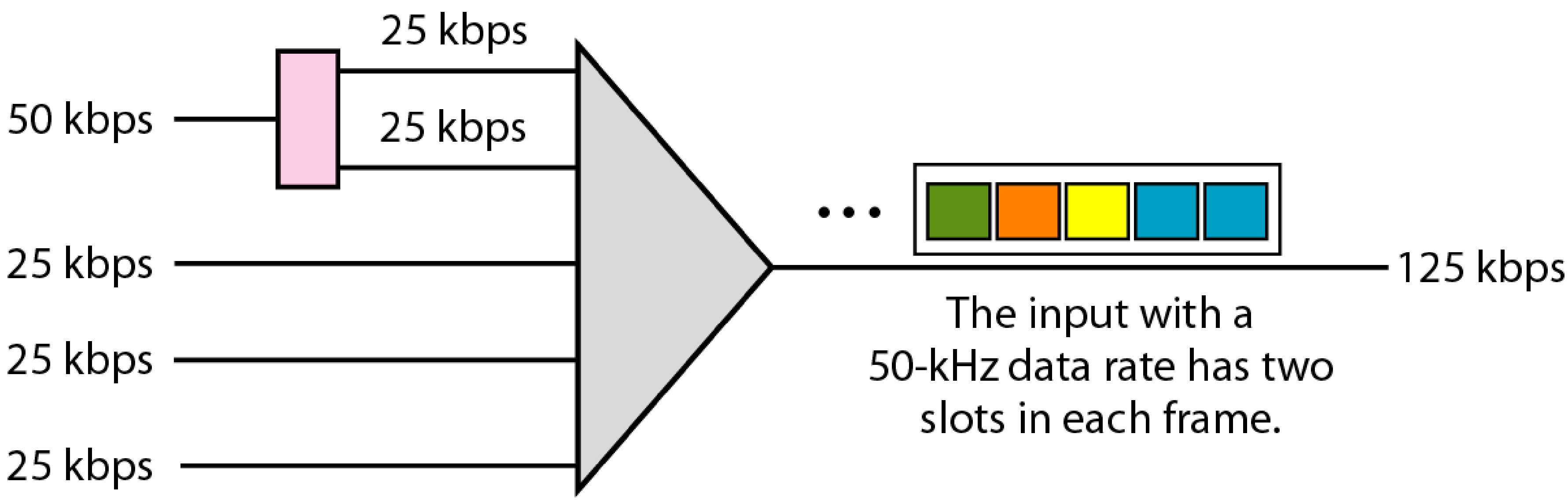
Data rate matching

- Multilevel: used when the data rate of the input links are multiples of each other.
- Multislot: used when there is a GCD between the data rates. The higher bit rate channels are allocated more slots per frame, and the output frame rate is a multiple of each input link.
- Pulse Stuffing: used when there is no GCD between the links. The slowest speed link will be brought up to the speed of the other links by bit insertion, this is called pulse stuffing.

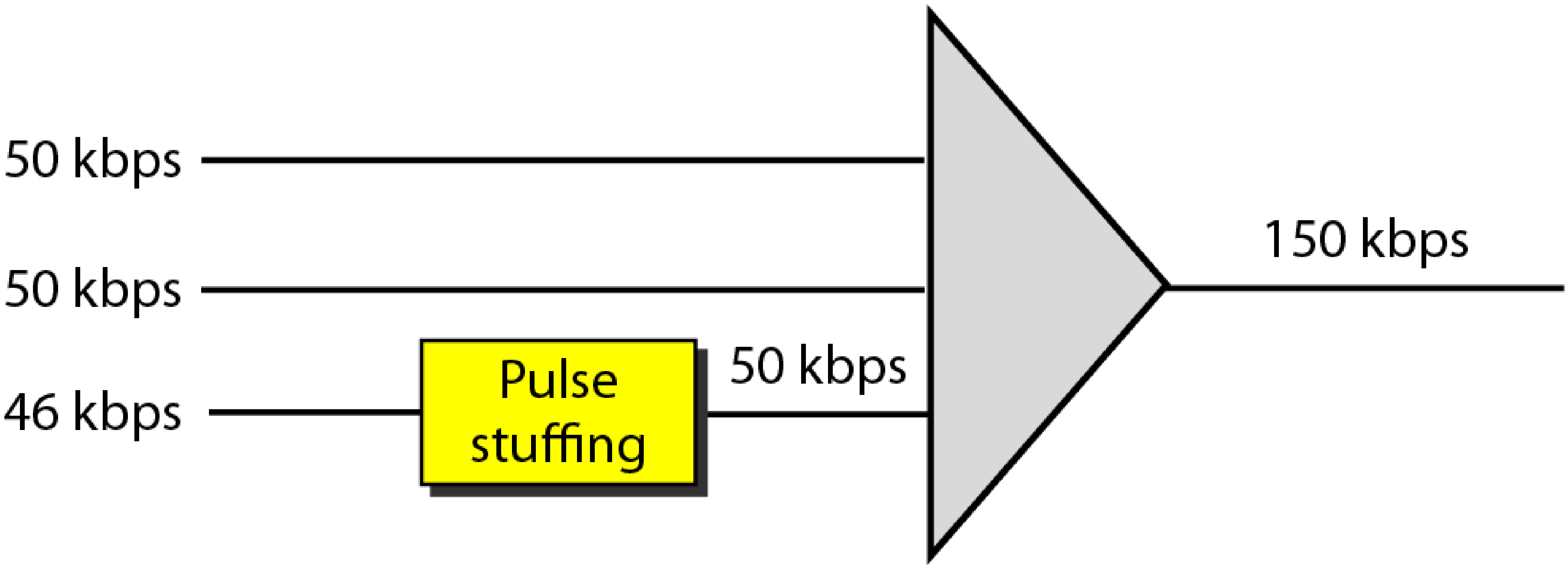
Multilevel multiplexing



Multiple-slot multiplexing



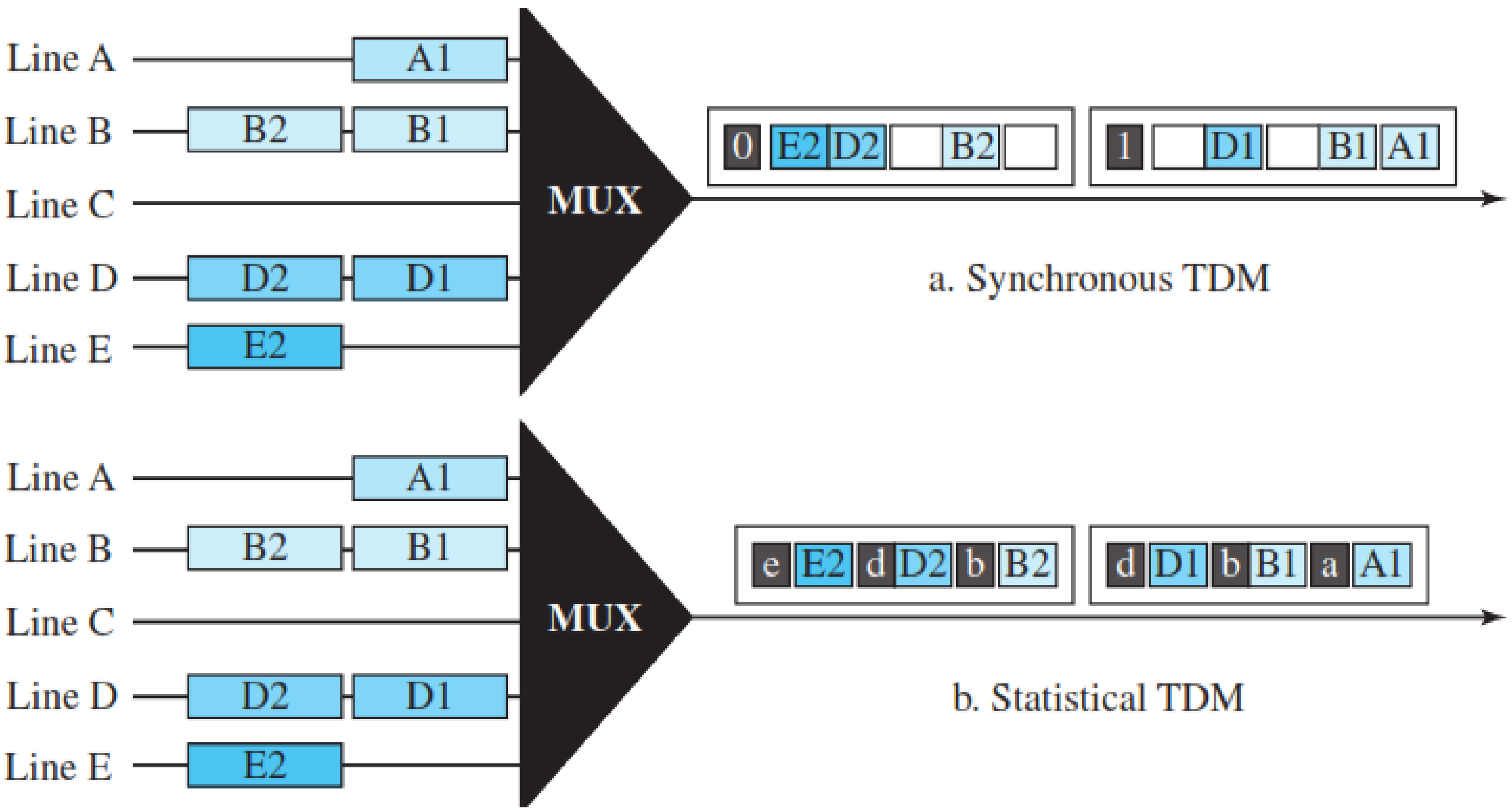
Pulse stuffing



Statistical Time-Division Multiplexing

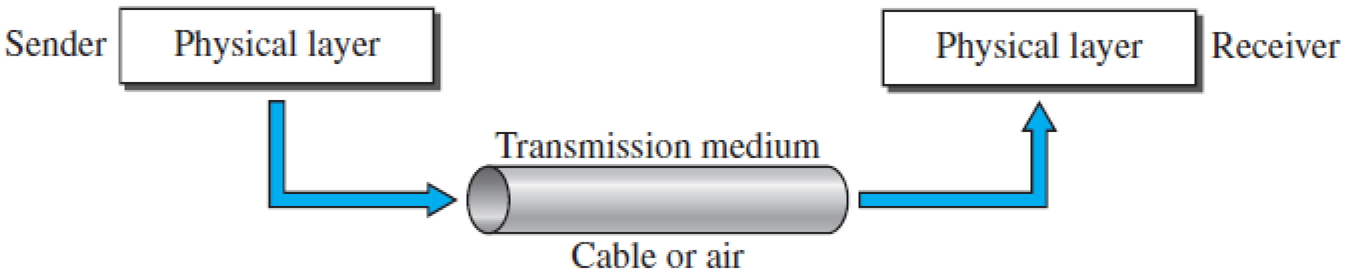
- In synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send
- In statistical time-division multiplexing, slots are dynamically allocated to improve bandwidth efficiency
- The multiplexer checks each input line in round robin fashion; it allocates a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line
- An output slot in synchronous TDM is totally occupied by data; in statistical TDM, a slot needs to carry data as well as the address of the destination because there is no fixed relationship between the inputs and outputs
- In statistical multiplexing, the number of slots in each frame is less than the number of input lines

Synchronous vs Statistical Time-Division Multiplexing



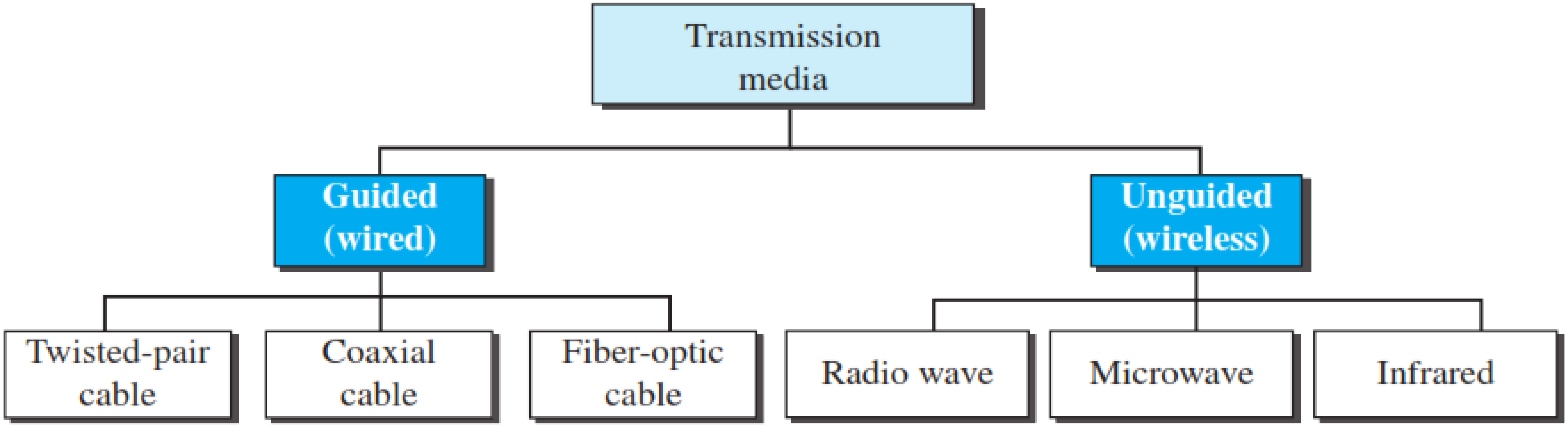
Transmission media

Transmission Media



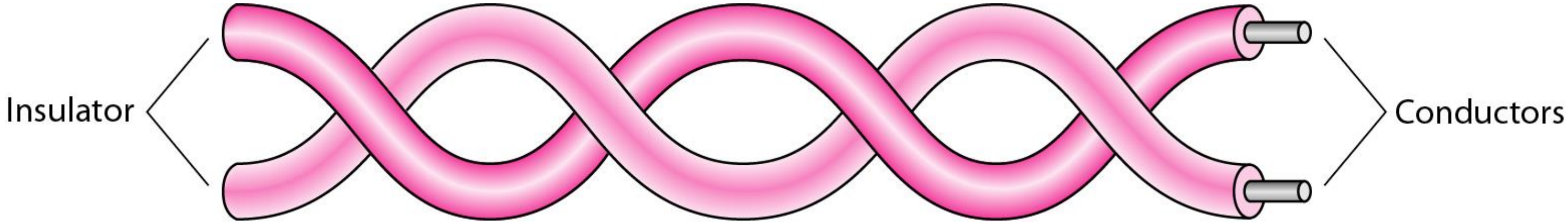
- Located below the physical layer and are directly controlled by the physical layer
- A transmission medium can be broadly defined as anything that can carry information from a source to a destination

Classes of transmission media



Twisted-Pair Cable

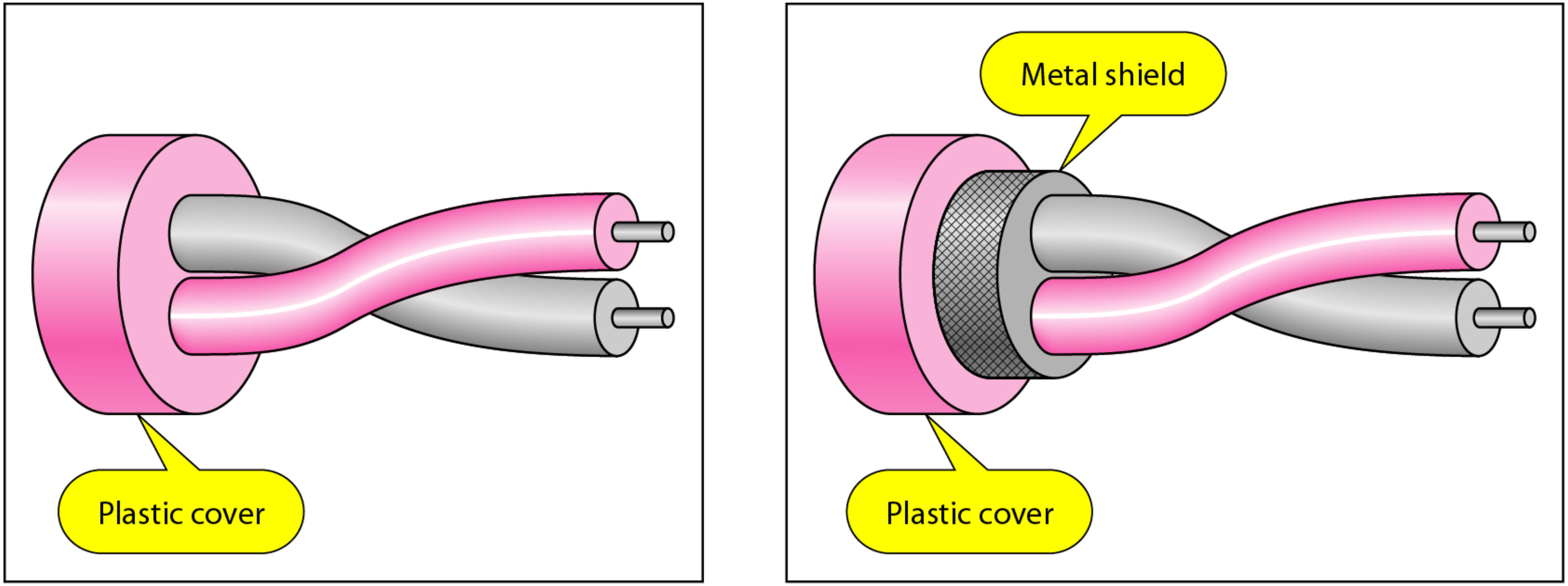
- A twisted pair consists of two insulated copper wires
- The wires are twisted together in a helical form
- The pairs are twisted to provide protection against crosstalk.
- Twisted pairs can run several kilometers without amplification.
- But for longer distances the signal becomes too attenuated and repeaters are needed.



Types of twisted pair cable

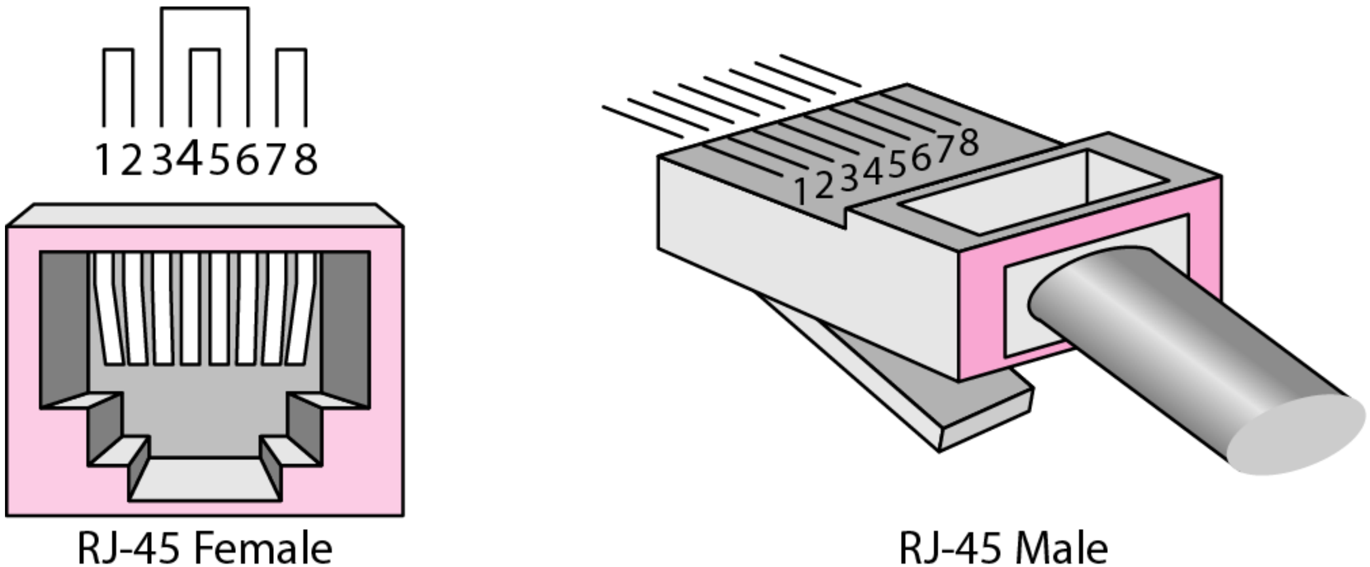
Two types of twisted pair cables:

- Unshielded Twisted Pair (UTP)
- Shielded Twisted Pair (STP)

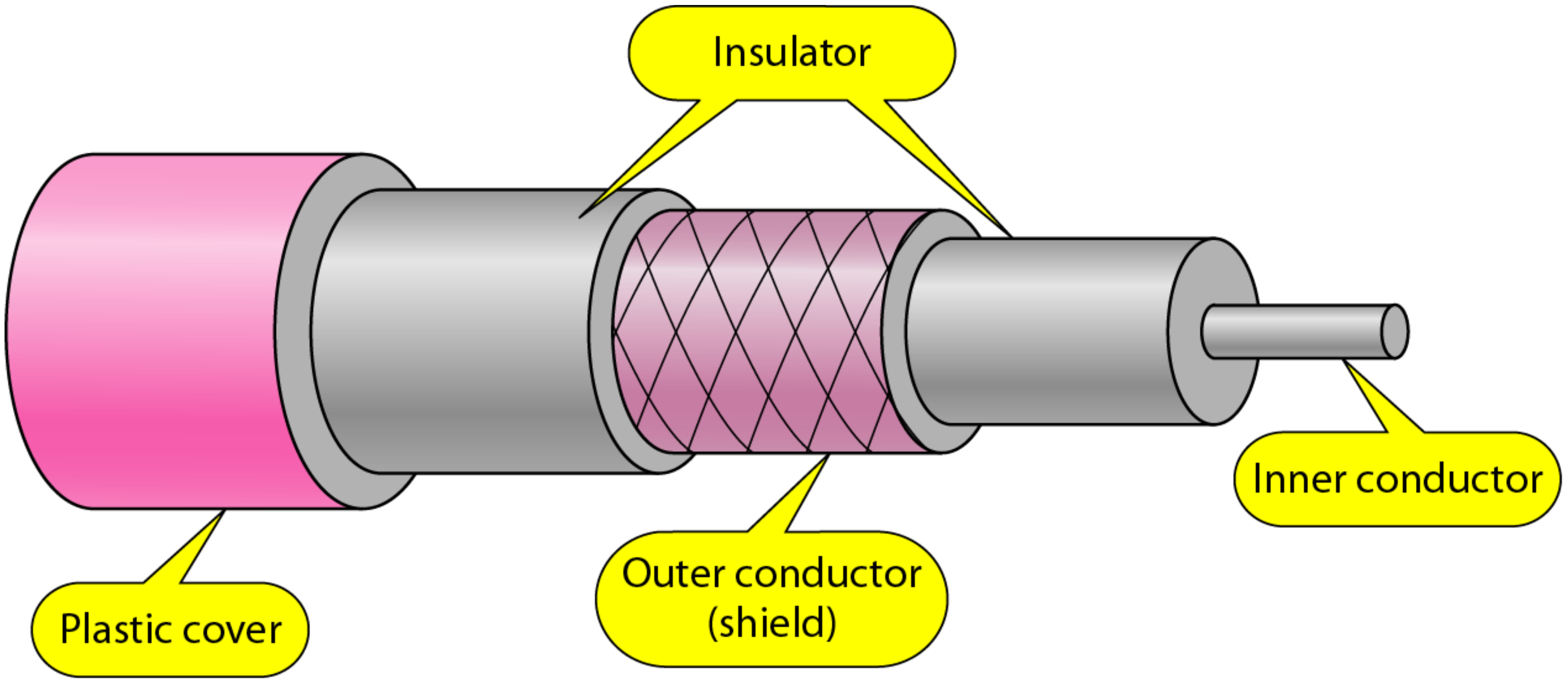


a. UTP b. STP

UTP connector



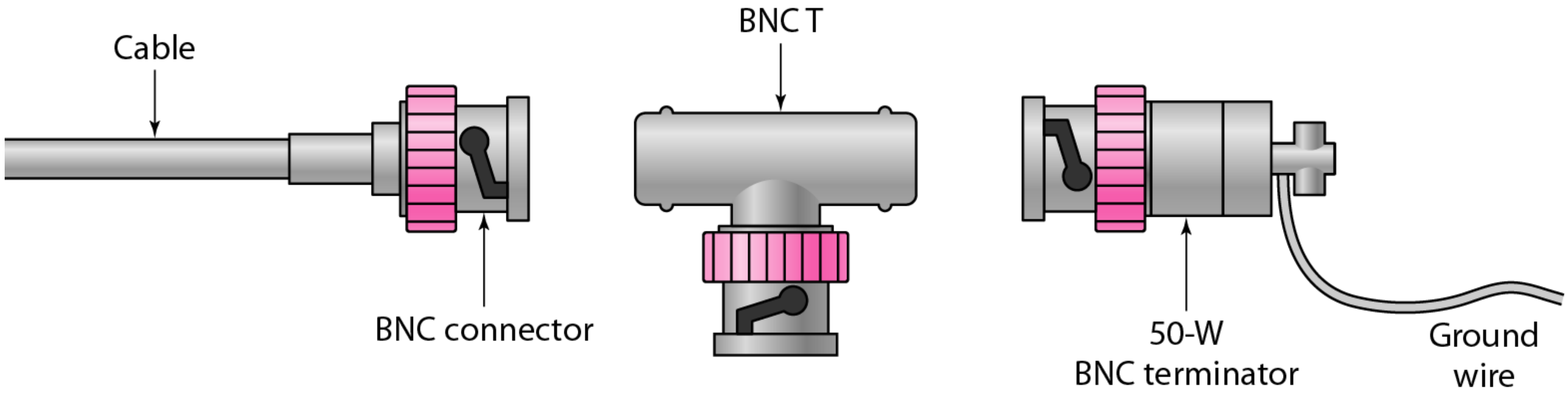
Coaxial Cable



Categories of coaxial cables

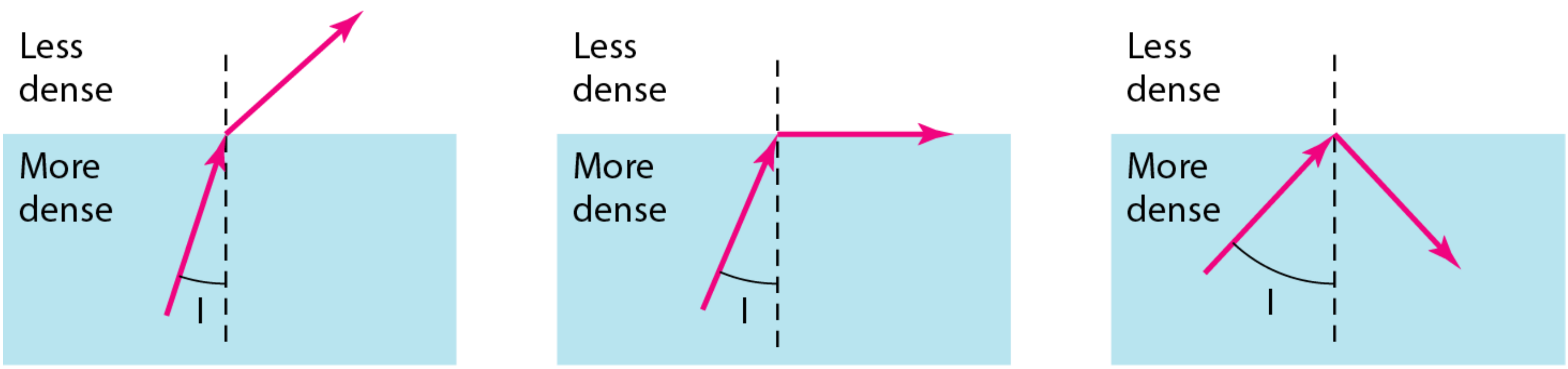
Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

BNC (Bayonet Neill-Concelman) connectors



Fiber-Optic Cable

- A fiber-optic cable is made of glass or plastic.
- Transmits signals in the form of light.
- Optical fibers use reflection to guide light through a channel.
- Bending of light ray



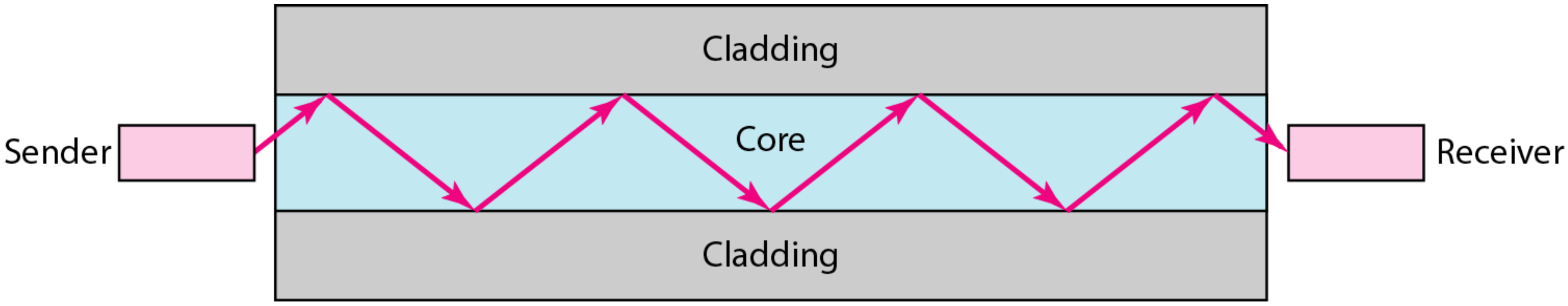
I < critical angle, refraction

I = critical angle, refraction

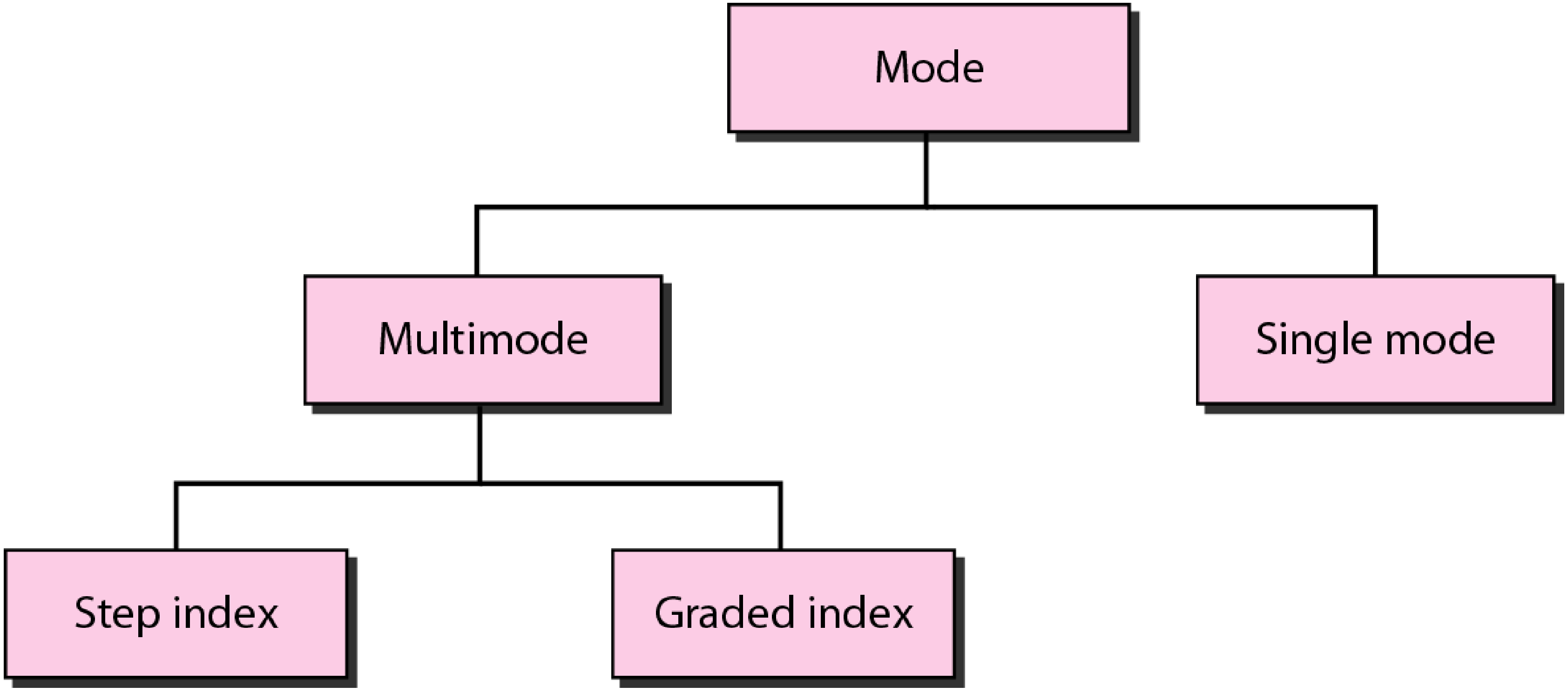
I > critical angle, reflection

Fiber-Optic Cable

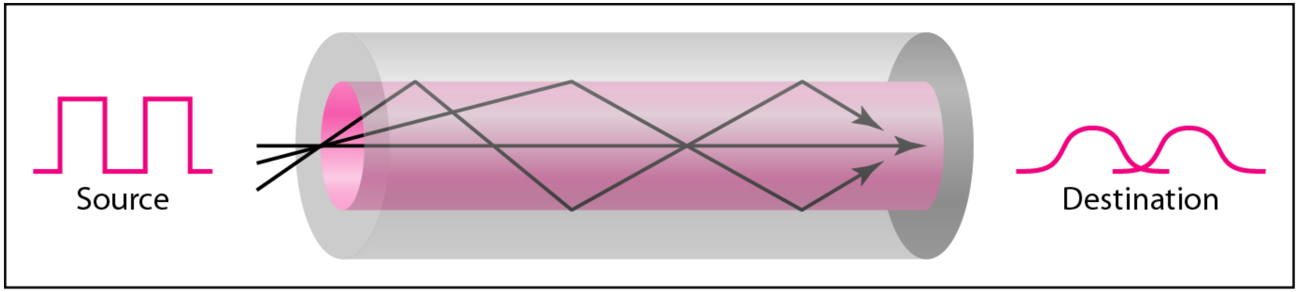
- A glass or plastic core is surrounded by a cladding of less dense glass or plastic.
- Transmits signals in the form of light.
- Bending of light ray



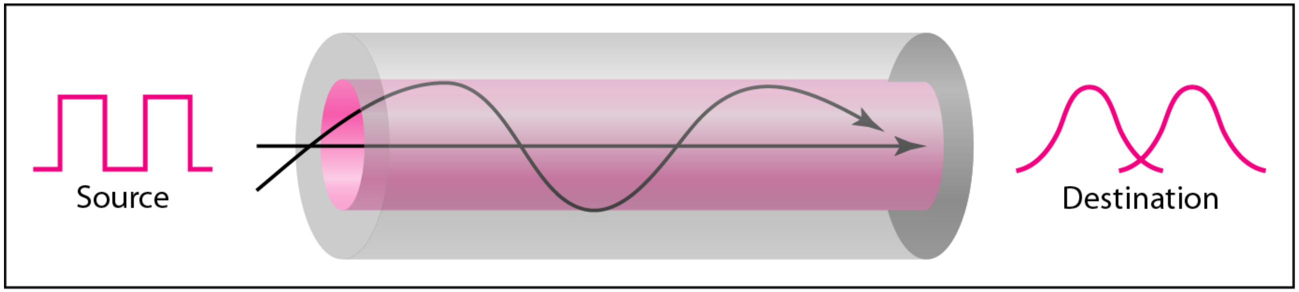
Propagation modes



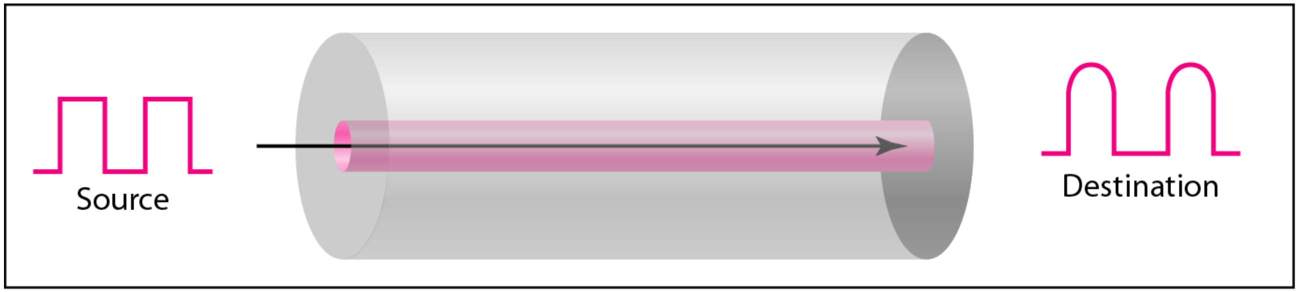
Propagation modes



a. Multimode, step index

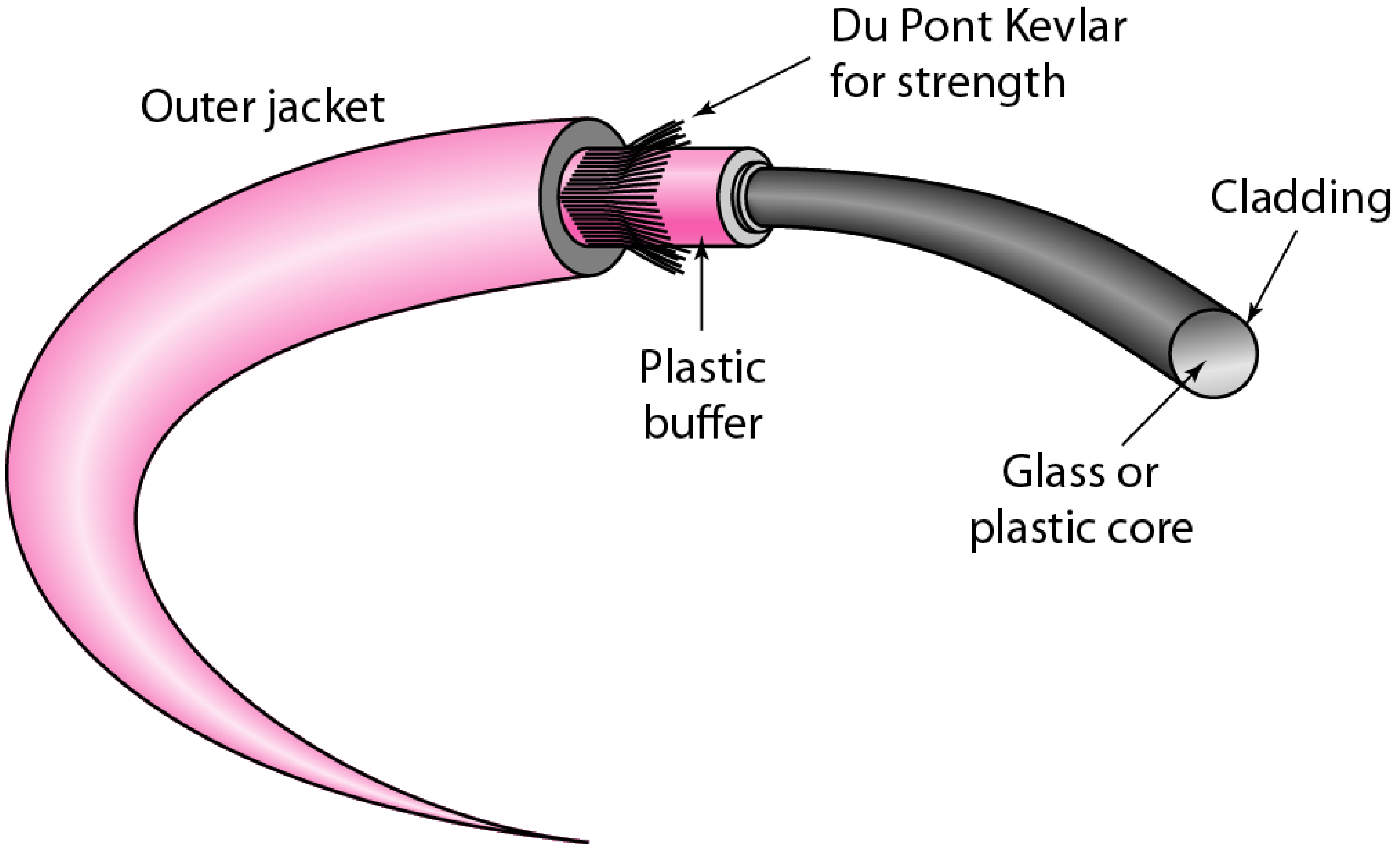


b. Multimode, graded index

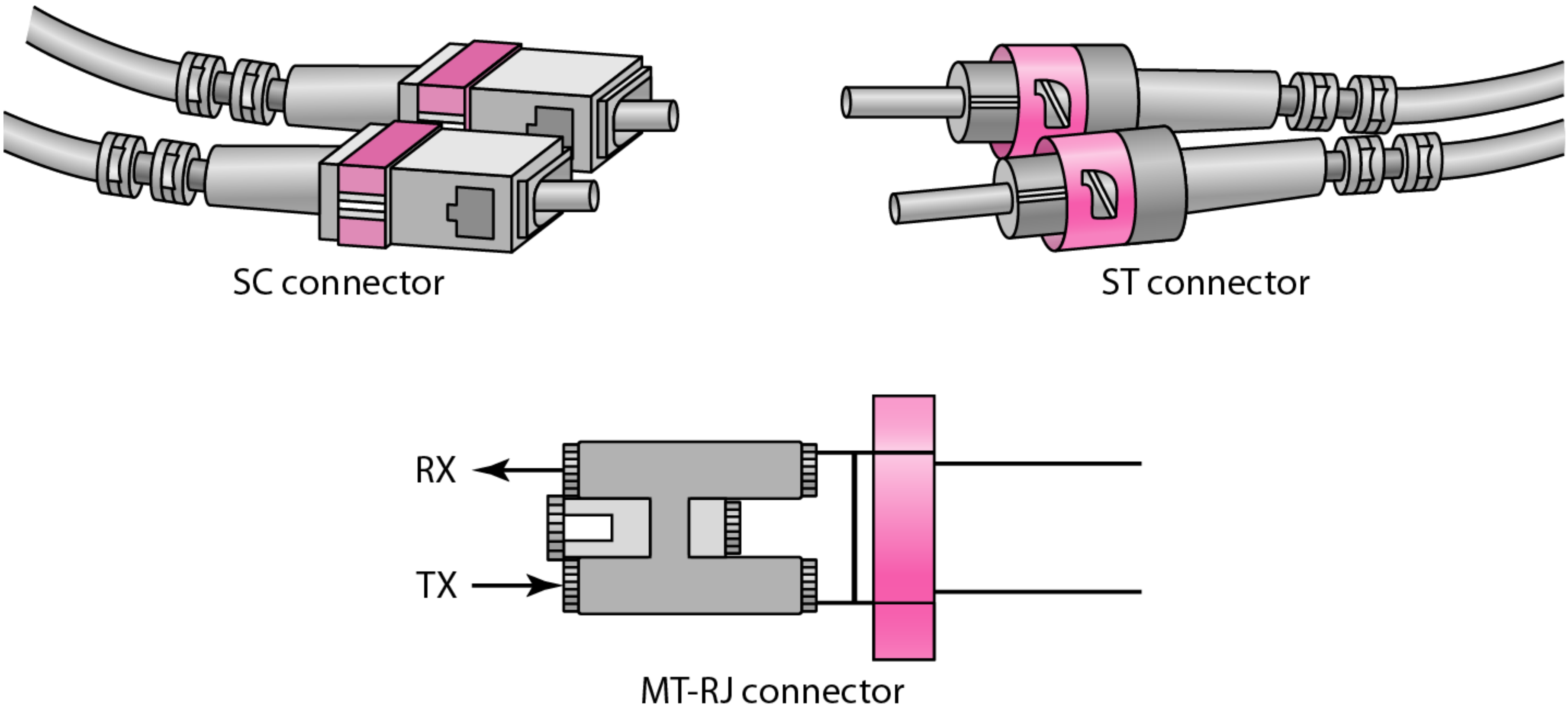


c. Single mode

Fiber construction

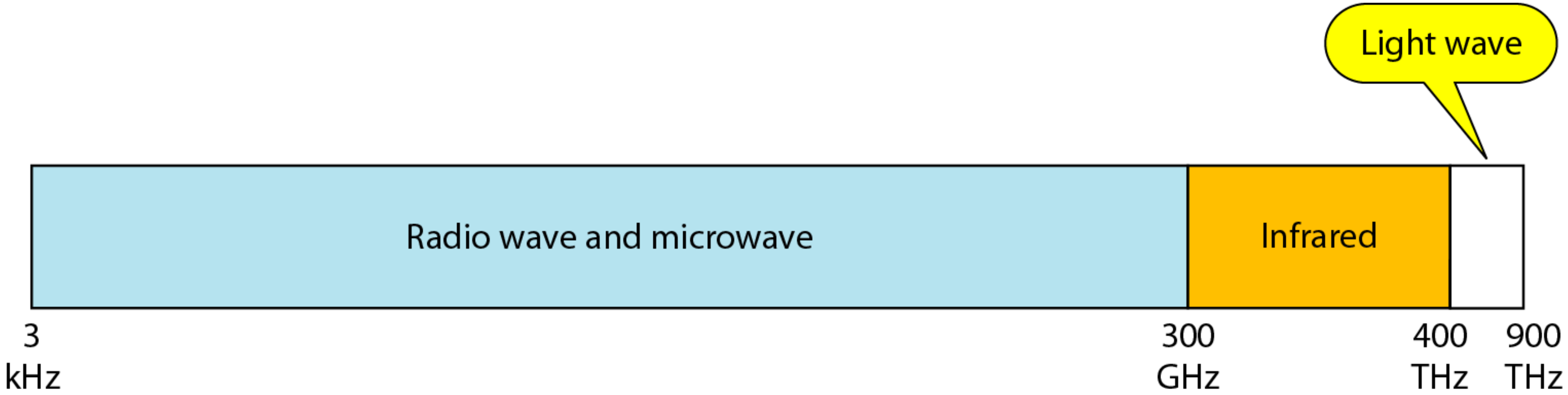


Fiber-optic cable connectors



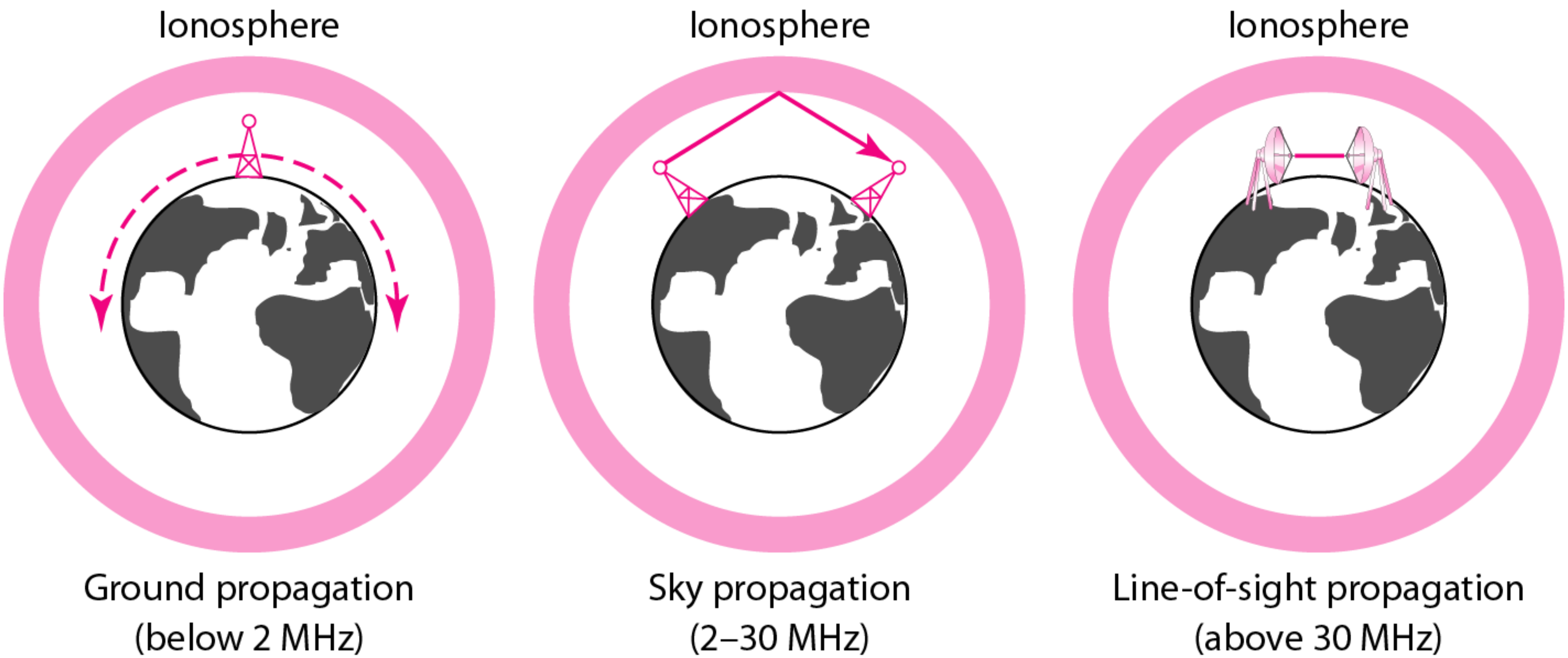
UNGUIDED MEDIA: WIRELESS

- Unguided media transport electromagnetic waves without using a physical conductor.



Electromagnetic spectrum for wireless communication

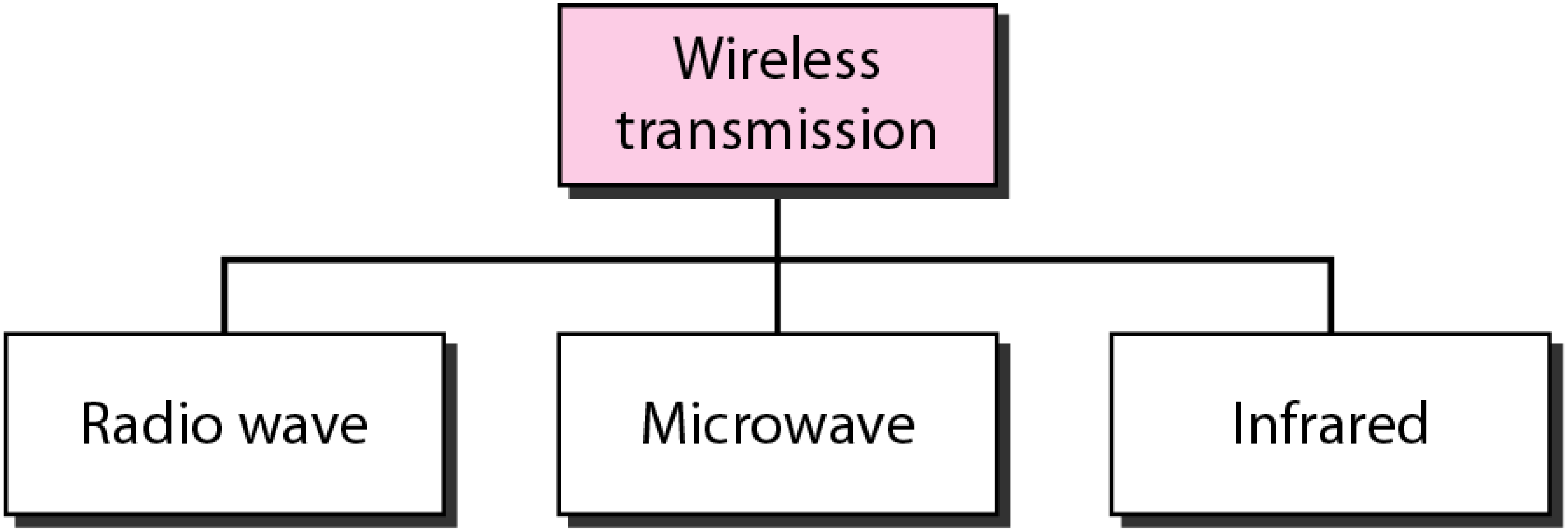
Propagation methods



Wireless Bands

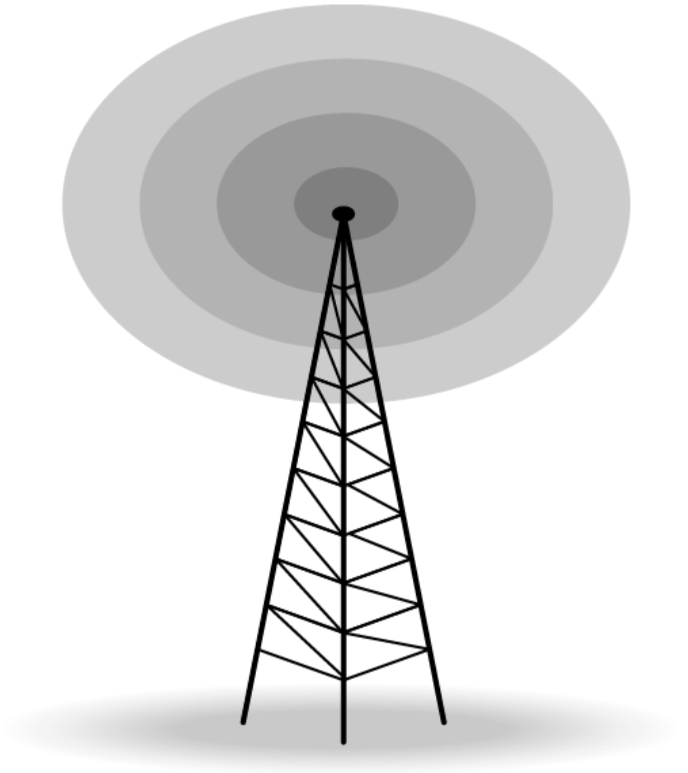
Band	*Range*	*Propagation*	*Application*
VLF (very low frequency)	3–30 kHz	Ground	Long-range radio navigation
LF (low frequency)	30–300 kHz	Ground	Radio beacons and navigational locators
MF (middle frequency)	300 kHz–3 MHz	Sky	AM radio
HF (high frequency)	3–30 MHz	Sky	Citizens band (CB), ship/aircraft 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



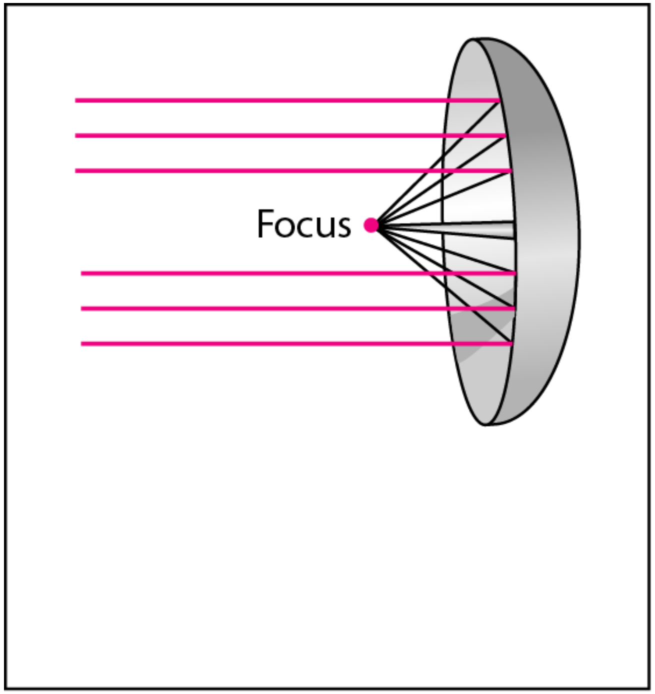
Radio Waves

- Radio waves are used for multicast communications.
- They can penetrate through walls.
- Highly regulated.
- Use omni directional antennas.

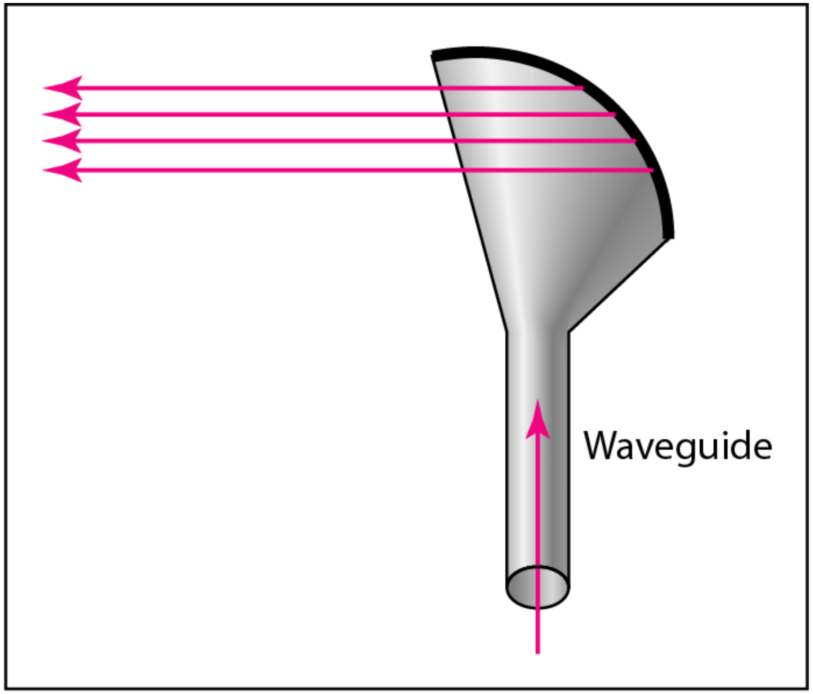


Microwaves

- Microwaves are used for unicast communication.
- Higher frequency ranges cannot penetrate walls.
- Use directional antennas.
- point to point line of sight communications.



a. Dish antenna



b. Horn antenna

Infrared

- Infrared signals can be used for short-range communication.
- line-of-sight propagation.