Physical Layer

CS44 Data Communications and Networking

Manjula L

Department of Computer Science and Engineering
RIT, Bangalore

Syllabus

Local Area Network: Ethernet: Standard Ethernet, WIFI, IEEE 802.11 project- architecture, MAC sub layer, addressing mechanism.

Physical layer: Signals - Analog signals, Digital signals. Signal Impairment -Attenuation and Amplification, Distortion, Data Rate Limits, Performance.

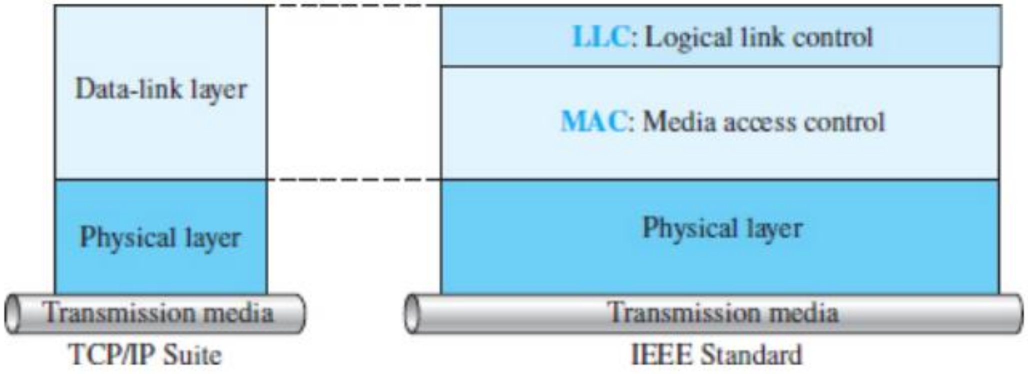
Digital Transmission - Digital-to-Digital Conversion, Analog-to-Digital Conversion. Analog Transmission- Digital-to-Analog Conversion,

Analog-to-Analog Conversion. Multiplexing: Frequency-Division Multiplexing, Time-Division Multiplexing.

Local Area Network: Ethernet: Standard Ethernet

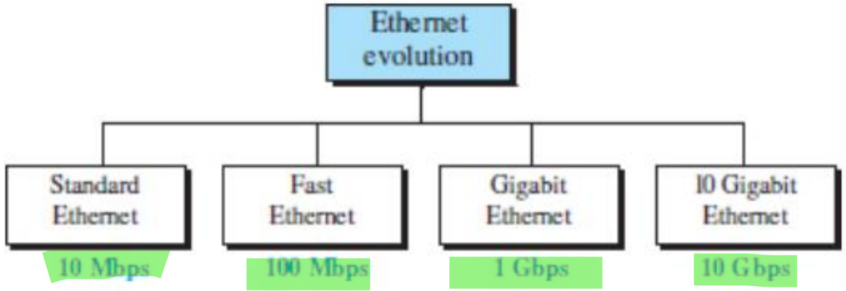
- The Institute of Electrical and Electronics Engineers (IEEE) has subdivided the data-link layer into two sublayers: logical link control (LLC) and media access control (MAC)

Figure 4.1 *IEEE standard for wired LANs*



Ethernet Evolution

Figure 4.2 *Ethernet evolution through four generations*



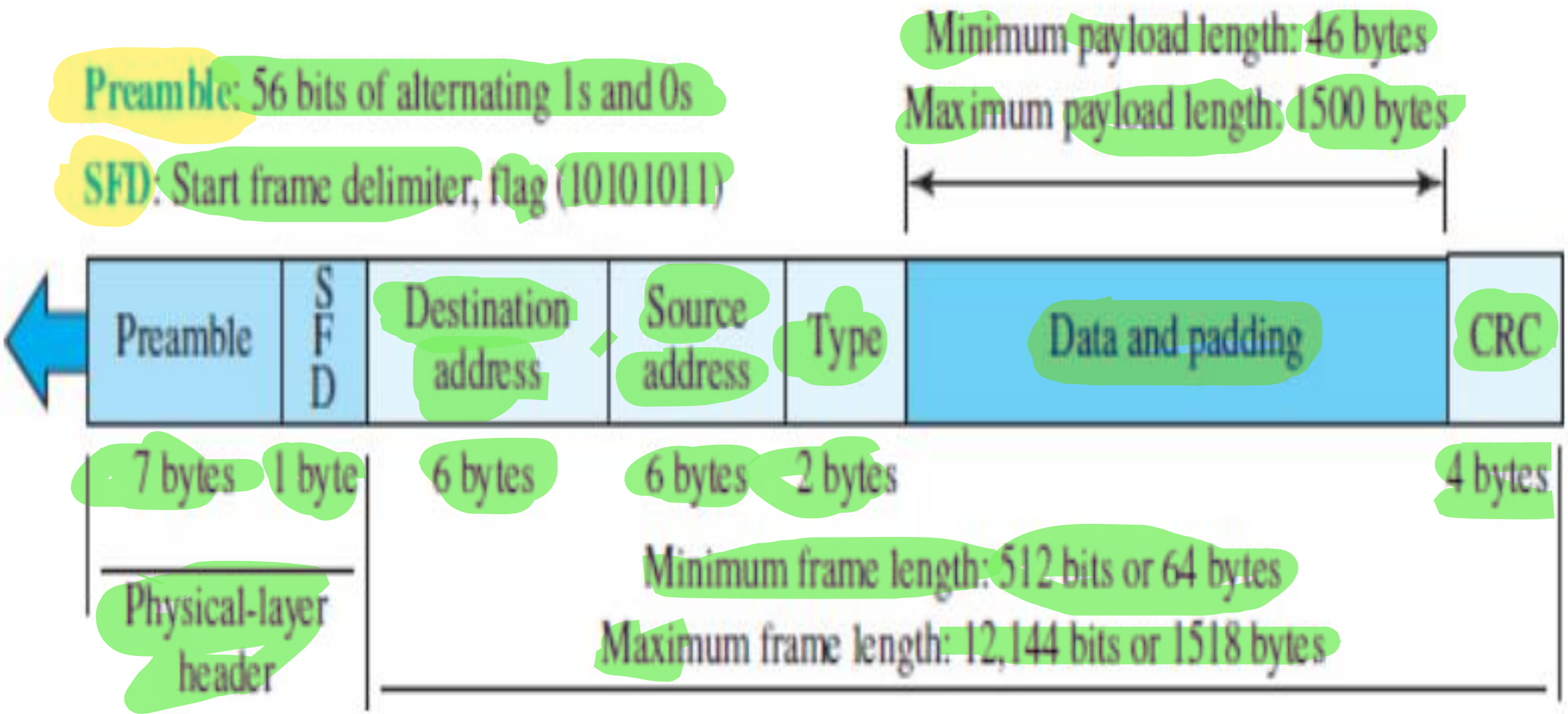
Standard Ethernet (10 Mbps)

- Ethernet technology with the data rate of 10 Mbps is referred as Standard Ethernet.
- Connectionless and Unreliable Service
- Frame Format
- Frame Length
- Addressing
- Transmission of Address Bits
- Implementation

Connectionless and Unreliable Service

- Ethernet provides a connectionless service, which means each frame sent is independent of the previous or next frame.
- Ethernet has no connection establishment or connection termination phases. The sender sends a frame whenever it has it; the receiver may or may not be ready for it.
- The sender may overwhelm the receiver with frames, which may result in dropped frames. If a frame drops, the sender data-link layer will not know about it unless an upper layer protocol takes care of it.
- Ethernet is also unreliable. If a frame is corrupted during transmission and the receiver finds out about the corruption, the receiver drops the frame silently. It is the duty of high-level protocols to find out about it

Frame Format



- **Preamble:** This field contains 7 bytes (56 bits) of alternating 0s and 1s
  - alert the receiving system to the coming frame
  - enable it to synchronize its clock if it's out of synchronization
  - The pattern provides only an alert and a timing pulse
  - The 56-bit pattern allows the stations to miss some bits at the beginning of the frame
  - actually added at the physical layer and is not (formally) part of the frame

Ethernet frame

- Start frame delimiter (SFD)
  - 1 byte: 10101011) signals the beginning of the frame as the size of Ethernet frame is variable size
  - The SFD warns the station or stations that this is the last chance for synchronization
  - The last 2 bits are $(11)_2$ and alert the receiver that the next field is the destination address
  - The SFD field is also added at the physical layer

Ethernet frame

- Destination address (DA)
  - Six bytes (48 bits)
  - contains the linklayer address of the destination station or stations to receive the packet
  - When the receiver sees its own link-layer address, or a multicast address for a group that the receiver is a member of, or a broadcast address, it decapsulates the data from the frame and passes the data to the upperlayer protocol defined by the value of the type field
- Source address (SA)
  - Six bytes and contains the link-layer address of the sender of the packet

Ethernet frame

- Type
  - Defines the upper-layer protocol whose packet is encapsulated in the frame
    - This protocol can be IP, ARP, OSPF, and so on
  - It is used for multiplexing and demultiplexing.

Ethernet frame

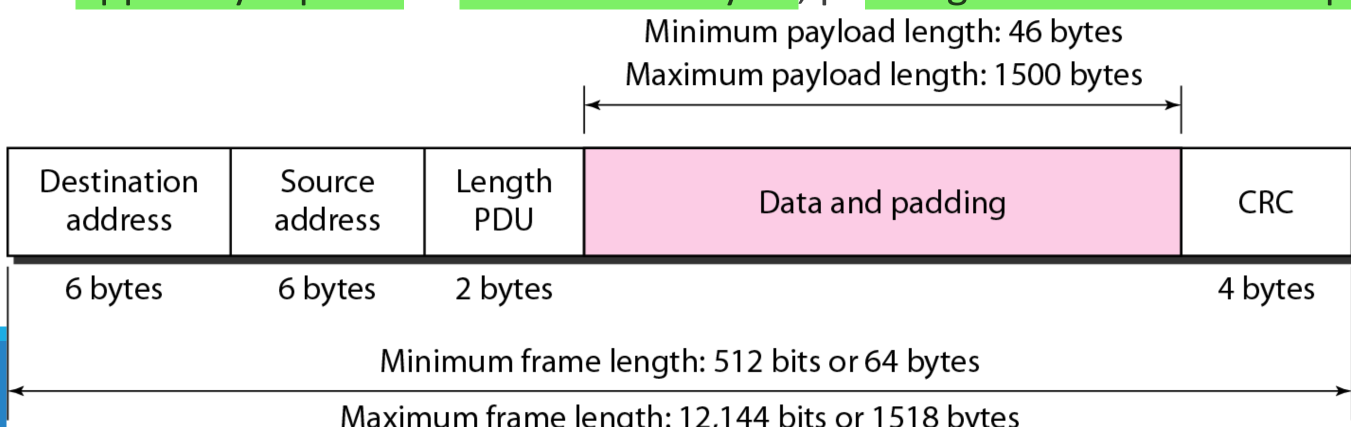
- Data
  - encapsulated from the upper-layer protocols
  - minimum of 46 and a maximum of 1500 bytes
    - If the data coming from the upper layer is more than 1500 bytes, it should be fragmented and encapsulated in more than one frame
    - If it is less than 46 bytes, it needs to be padded with extra 0s
      - A padded data frame is delivered to the upper-layer protocol as it is (without removing the padding)
        - means that it is the responsibility of the upper layer to remove or, in the case of the sender, to add the padding
        - The upper-layer protocol needs to know the length of its data. For example, a datagram has a field that defines the length of the data

Ethernet frame

- CRC
  - contains error detection information, in this case a CRC-32
  - The CRC is calculated over the addresses, types, and data field
  - If the receiver calculates the CRC and finds that it is not zero (corruption in transmission), it discards the frame.

Frame Length

- Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame
- The minimum length restriction is required for the correct operation of CSMA/CD
  - An Ethernet frame - a minimum length of 512 bits or 64 bytes
- Part of the length is the header and the trailer
  - If 18 bytes of header and trailer (6 bytes of source address, 6 bytes of destination address, 2 bytes of length or type, and 4 bytes of CRC), then the minimum length of data from the upper layer is 64 − 18 = 46 bytes
    - If the upper-layer packet is less than 46 bytes, padding is added to make up the difference



Frame Length

- The standard defines the maximum length of a frame (without preamble and SFD field) as 1518 bytes
- If we subtract the 18 bytes of header and trailer, the maximum length of the payload is 1500 bytes
- The maximum length restriction has two historical reasons
  - First, memory was very expensive when Ethernet was designed; a maximum length restriction helped to reduce the size of the buffer
  - Second, the maximum length restriction prevents one station from monopolizing the shared medium, blocking other stations that have data to send

frame length in bytes		data length in bytes	
Minimum	Maximum	Minimum	Maximum
64	1518	46	1500

Standard Ethernet - Addressing

- Each station on an Ethernet network (such as a PC, workstation, or printer) has its own network interface card **(NIC)**
- The NIC fits inside the station and provides the station with a **link-layer address**
- The Ethernet address is **6 bytes** (48 bits), normally written in hexadecimal notation, with a **colon** between the bytes

06 : 01 : 02 : 01 : 2C : 4B

6 bytes = 12 hex digits = 48 bits

Transmission of Address Bits

The way the addresses are sent out online is different from the way they are written in hexadecimal notation.

The transmission is left to right, byte by byte; however, for each byte, the least significant bit is sent first and the most significant bit is sent last.

This means that the bit that defines an address as unicast or multicast arrives first at the receiver. This helps the receiver to immediately know if the packet is unicast or multicast.

This example shows how the address 47:20:1B:2E:08:EE is sent out online.

Hexadecimal	47	20	1B	2E	08	EE
Binary	**01000111**	**00100000**	**00011011**	**00101110**	**00001000**	**11101110**

This example shows how the address 47:20:1B:2E:08:EE is sent out online.

Hexadecimal	47	20	1B	2E	08	EE
Binary	**01000111**	**00100000**	**00011011**	**00101110**	**00001000**	**11101110**
Transmitted	**11100010**	**00000100**	**11011000**	**01110100**	**00010000**	**01110111**

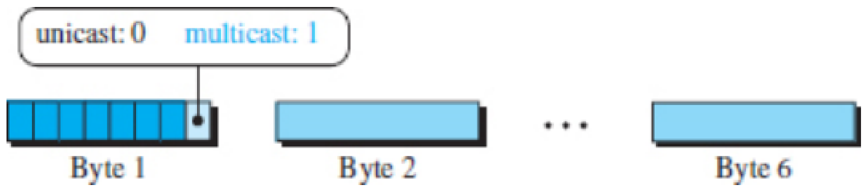
←

Multicast
cuz first is 1

Unicast, Multicast, and Broadcast Addresses

- A source address is always a unicast address—the frame comes from only one station.
- The destination address, however, can be unicast, multicast, or broadcast.
- Figure 4.4 shows how to distinguish a unicast address from a multicast address. If the least significant bit of the first byte in a destination address is 0, the address is unicast; otherwise, it is multicast.
- Note that with the way the bits are transmitted, the unicast/multicast bit is the first bit that is transmitted or received. The broadcast address is a special case of the multicast address: The recipients are all the stations on the LAN. A broadcast destination address is forty-eight 1s.

Figure 4.4 *Unicast and multicast addresses*



Example 4.2

Define the type of the following destination addresses:

a. **4A:30:10:21:10:1A**

b. **47:20:1B:2E:08:EE**

c. **FF:FF:FF:FF:FF:FF**

convert the hex of 1st byte into binary
reverse it
UNICAST - if the first bit is 0
MULTICAST - if the first bit is 1
BROADCAST - all 48 bits / 6bytes are 1/FF

a. This is a unicast address because A in binary is 1010 (even).

b. This is a multicast address because 7 in binary is 0111 (odd).

c. This is a broadcast address because all digits are Fs in hexadecimal.

Implementation	*Medium*	*Medium Length(m)*	*Encoding*
10Base5	Thick coax	500	Line coding
10Base2	Thin coax	185	Line coding
10Base-T	2 UTP	100	Line coding
10Base-F	2 Fiber	2000	Line coding

WIFI, IEEE 802.11 PROJECT

- Architecture
- MAC
- Addressing Set

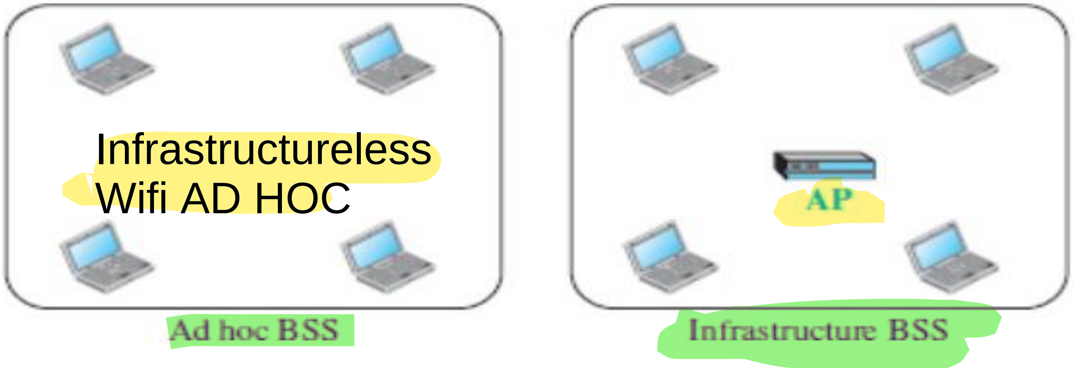
Architecture Wi-Fi

The IEEE standard defines two kinds of services: the basic service set (BSS) and the extended service set (ESS).

Basic Service Set: IEEE 802.11 defines the basic service set (BSS) as the building blocks of a wireless LAN.

A basic service set is made up of stationary or mobile wireless stations and an optional central base station, known as the access point (AP).

Figure 4.7 *Basic service sets (BSSs)*

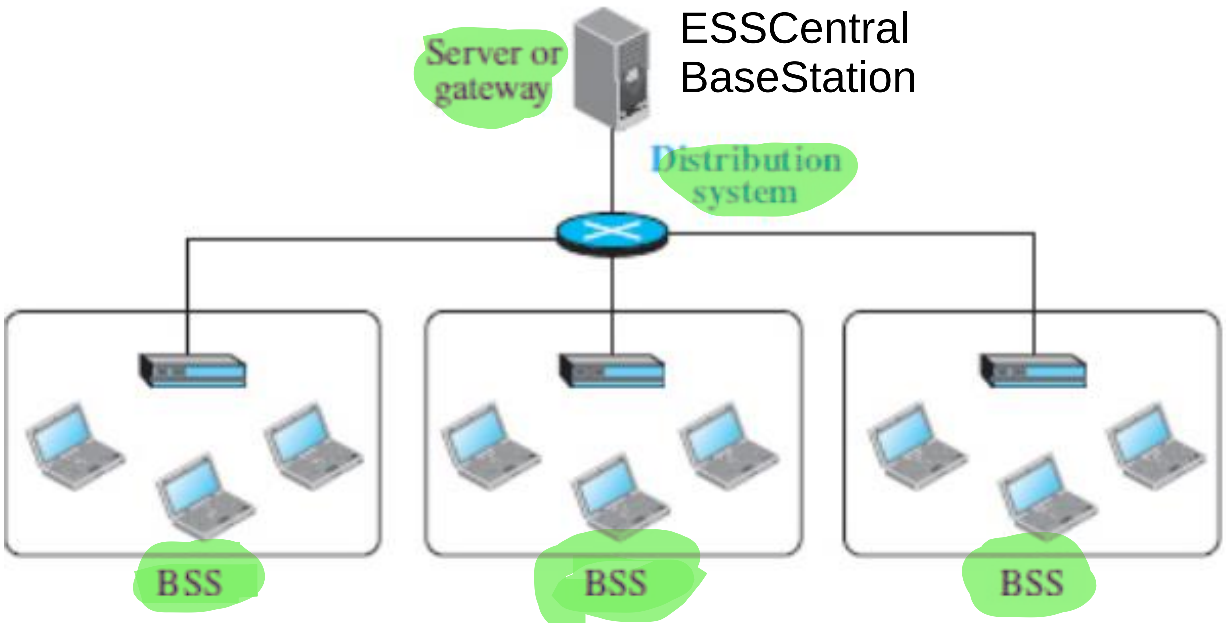


The BSS without an AP is a stand-alone network and cannot send data to other BSSs. It is called an ad hoc architecture.

In this architecture, stations can form a network without the need of an AP; they can locate one another and agree to be part of a BSS.

A BSS with an AP is sometimes referred to as an infrastructure BSS

Extended Service Set



An extended service set (ESS) is made up of two or more BSSs with APs. In this case, the BSSs are connected through a distribution system, which is a wired or a wireless network.

The distribution system connects the APs in the BSSs.

IEEE 802.11 does not restrict the distribution system; it can be any IEEE LAN such as an Ethernet

MAC Sublayer

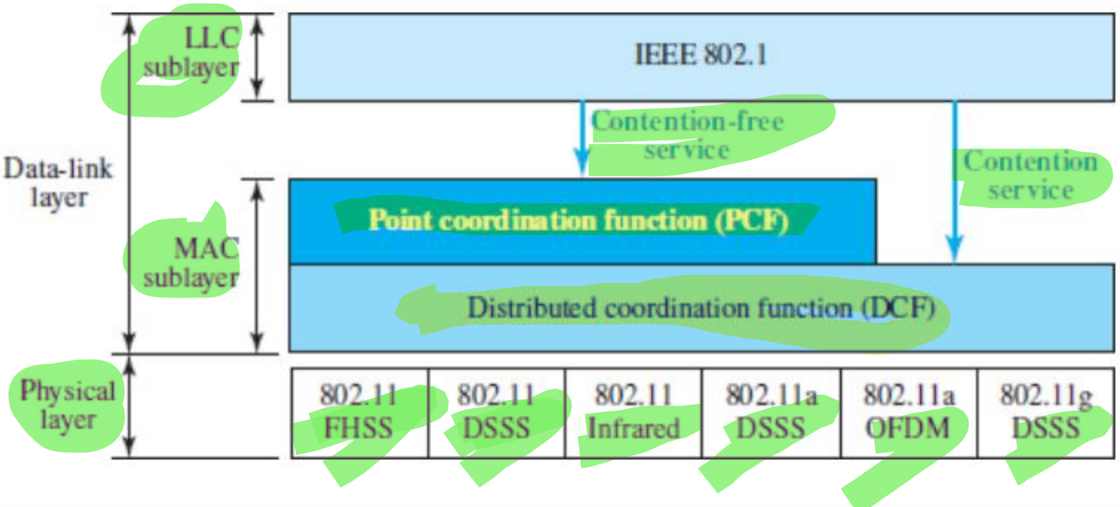
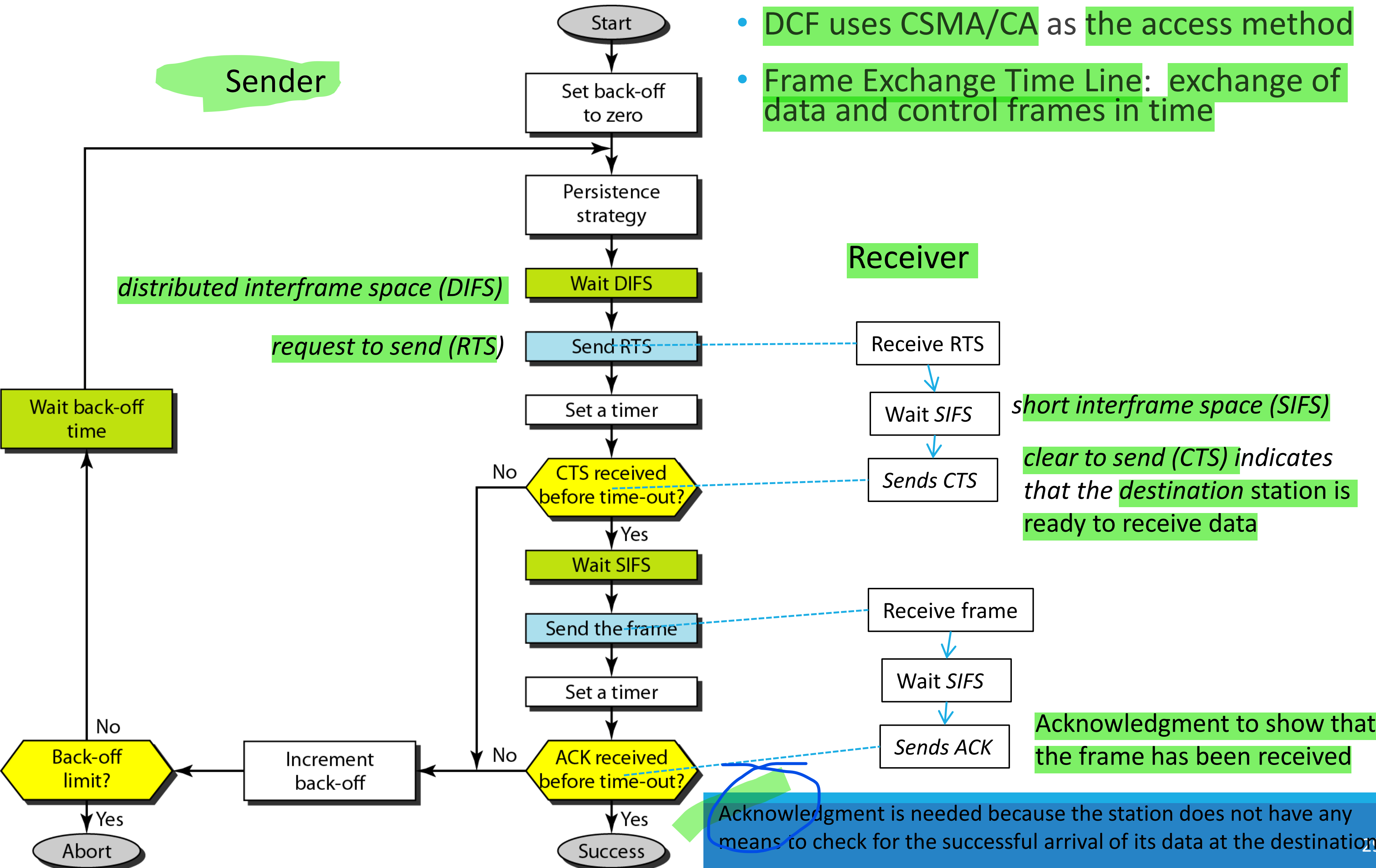


Figure 4.9 *MAC layers in the IEEE 802.11 standard*

Distributed Coordination Function (DCF)

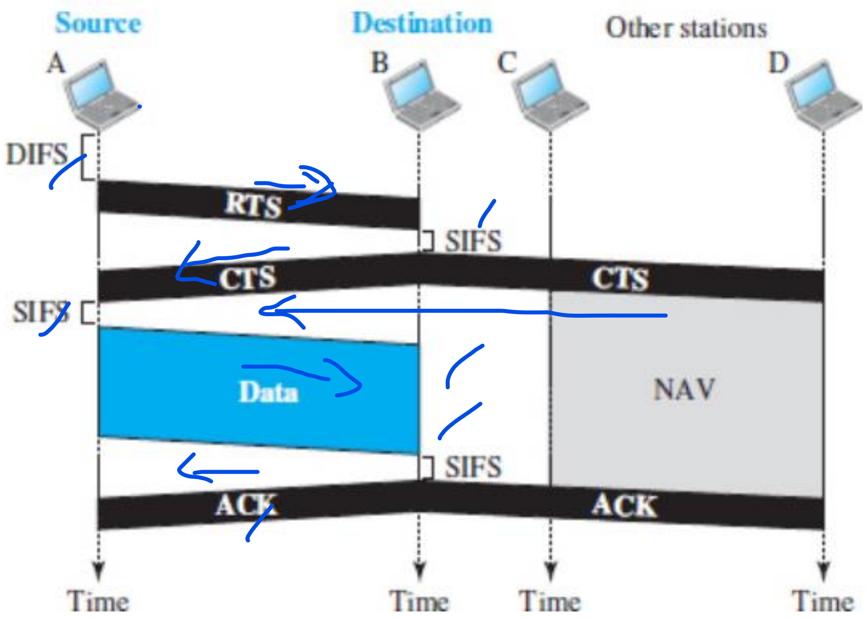
- DCF uses CSMA/CA as the access method
- Frame Exchange Time Line: exchange of data and control frames in time



Acknowledgment is needed because the station does not have any means to check for the successful arrival of its data at the destination

Distributed Coordination Function

Network Allocation Vector:



When a station sends an RTS frame, it includes the duration of time that it needs to occupy the channel. The stations that are affected by this transmission create a timer called a network allocation vector (NAV) that shows how much time must pass before these stations are allowed to check the channel for idleness

Point Coordination Function (PCF)

- The point coordination function (PCF) is an optional access method that can be implemented in an infrastructure network (not in an ad hoc network).

- It is implemented on top of the DCF and is used mostly for time-sensitive transmission

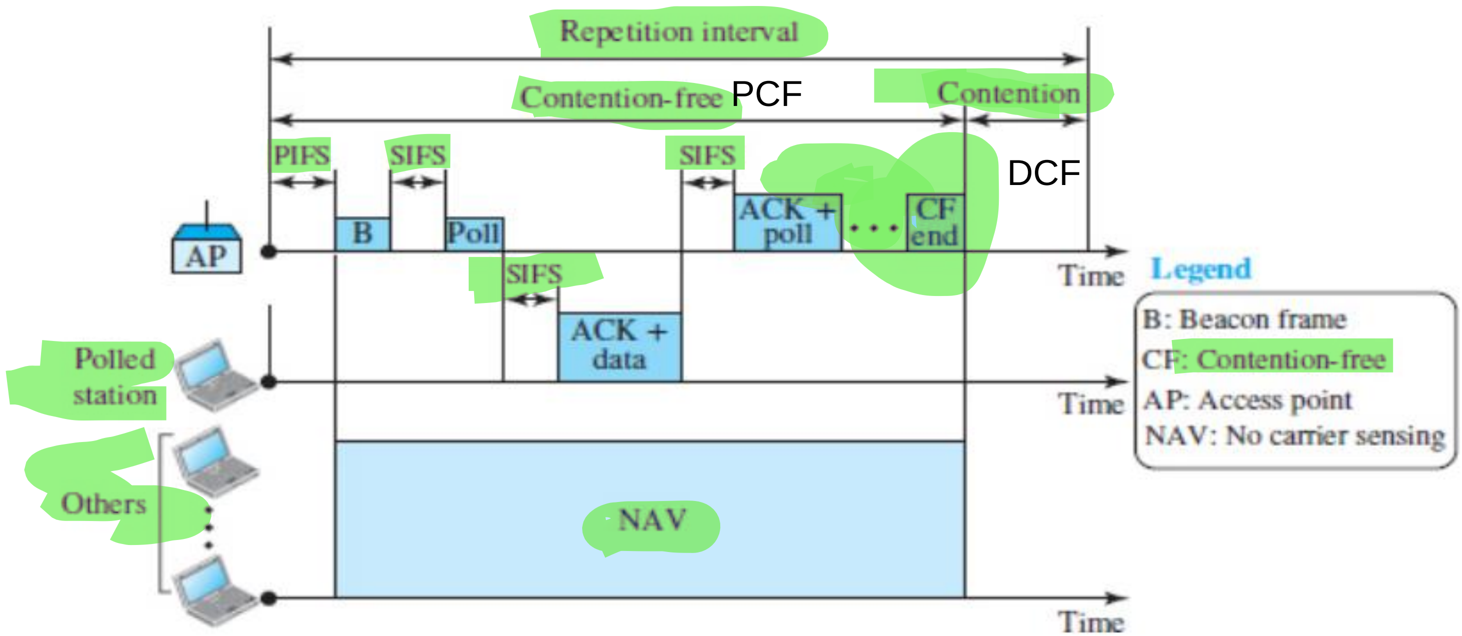
- PCF has a centralized, contention-free polling access method. The access point performs polling for stations that are capable of being polled. The stations are polled one after another, sending any data they have to the access point.

Point Coordination Function (PCF)

- To give priority to PCF over DCF, another interframe space, has been defined: point coordination function interframe space [PCF IFS (PIFS)].
- At the same time, a station wants to use only DCF and an access point wants to use PCF, the access point has priority.
- Because of the priority of PCF over DCF, stations that only use DCF may not gain access to the medium. To prevent this, a repetition interval has been designed to cover both contention-free PCF and contention-based DCF traffic.
- The repetition interval, which is repeated continuously, starts with a special control frame, called a beacon frame. When the stations hear the beacon frame, they start their NAV for the duration of the contention-free period of the repetition interval. Figure 4.11 shows an example of a repetition interval.

Point Coordination Function (PCF)

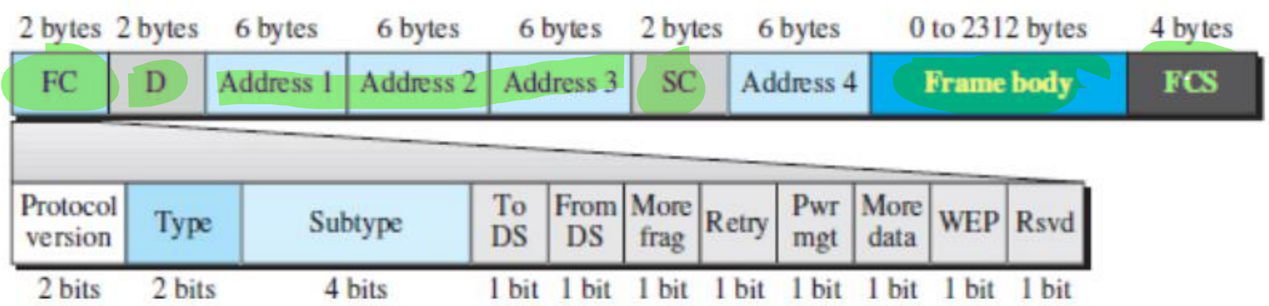
Figure 4.11 *Example of repetition interval*



During the repetition interval, the point controller (PC) can send a poll frame, receive data, send an ACK, receive an ACK, or do any combination of these (802.11 uses piggybacking). At the end of the contention-free (CF) period, the PC sends a CF end frame to allow the contention-based stations to use the medium

Frame Format

Figure 4.12 *Frame format*



- Frame control (FC). The FC field is 2 bytes long and defines the type of frame and some control information. Table 4.5 describes the subfields. We will discuss each frame type next.

Field	*Explanation*
Version	Current version is 0
Type	Type of information: management (00), control (01), or data (10)
Subtype	Subtype of each type (see Table 4.6)
To DS	Defined later
From DS	Defined later
More flag	When set to 1, means more fragments
Retry	When set to 1, means retransmitted frame
Pwr mgt	When set to 1, means station is in power management mode
More data	When set to 1, means station has more data to send
WEP	Wired equivalent privacy (encryption implemented)
Rsvd	Reserved

- ~~D~~. This field defines the duration of the transmission that is used to set the value of NAV. In one control frame, it defines the ID of the frame.

- ❑ Addresses. There are four address fields, each 6 bytes long. The meaning of each address field depends on the value of the To DS and From DS subfields. DS - Distrbn System
- ❑ Sequence control. This field, often called the SC field, defines a 16-bit value. The first 4 bits define the fragment number; the last 12 bits define the sequence number, which is the same in all fragments.
- ❑ Frame body. This field, which can be between 0 and 2312 bytes, contains information based on the type and the subtype defined in the FC field
- . ❑ FCS. The FCS field is 4 bytes long and contains a CRC-32 error-detection sequence.

Frame Types

- A wireless LAN defined by IEEE 802.11 has three categories of frames: management frames, control frames, and data frames.
- Management Frames : Management frames are used for the initial communication between stations and access points.
- Control Frames: Control frames are used for accessing the channel and acknowledging frames. Figure 4.13 shows the format.

Figure 4.13 *Control frames*

Table 4.6 *Values of subfields in control frames*

Subtype	*Meaning*
1011	Request to send (RTS)
1100	Clear to send (CTS)
1101	Acknowledgment (ACK)

Data Frames

- Data frames are used for carrying data and control information.

Addressing Mechanism

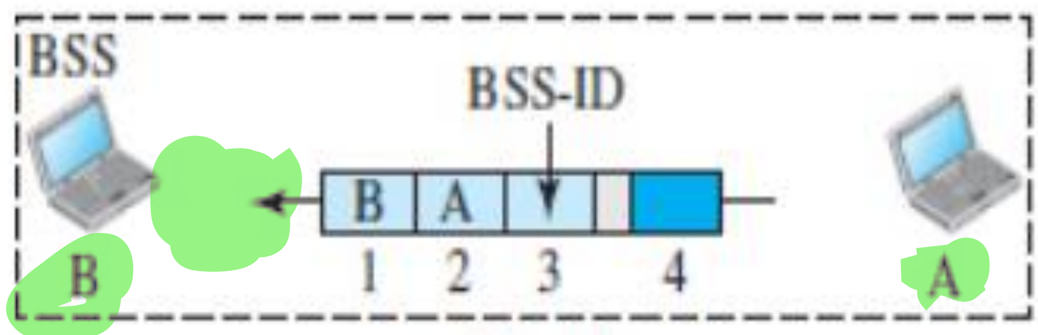
			Next device	Previous Device	Final Dest	Original Src	
	To DS	*From DS*	*Address 1*	*Address 2*	*Address 3*	*Address 4*	
DSB	0	0	Destination	Source	BSS ID	N/A	Station to Station
DSS	0	1	Destination	Sending AP	Source	N/A	AP to Station
RSD	1	0	Receiving AP	Source	Destination	N/A	Station to AP
RSDS	1	1	Receiving AP	Sending AP	Destination	Source	AP to AP WIRELESS

Addressing Mechanism

- Note that address 1 is always the address of the next device that the frame will visit.
- Address 2 is always the address of the previous device that the frame has left.
- Address 3 is the address of the final destination station if it is not defined by address 1 or the original source station if is not defined by address 2.
- Address 4 is the original source when the distribution system is also wireless.

Addressing mechanisms

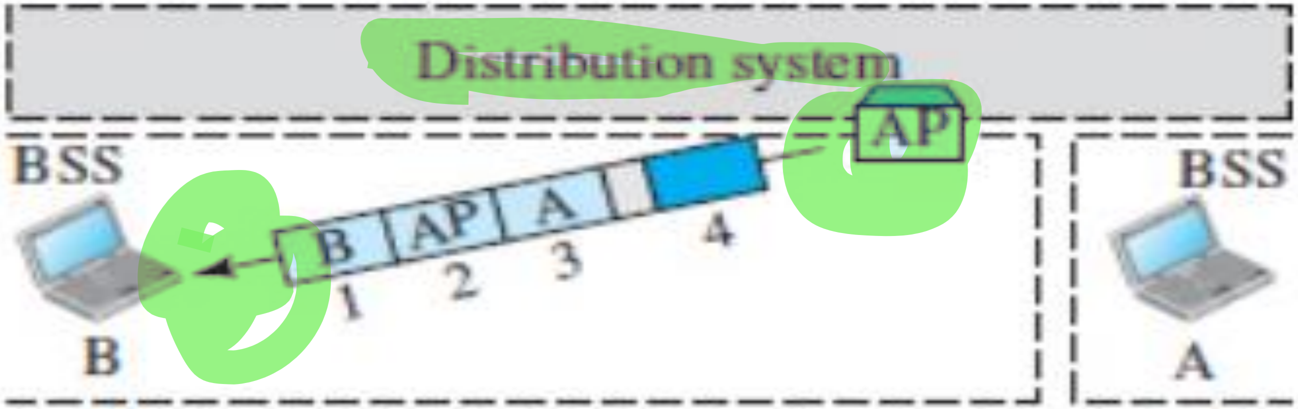
- Case 1:00. In this case, To DS = 0 and From DS = 0. This means that the frame is not going to a distribution system (To DS = 0) and is not coming from a distribution system (From DS = 0). The frame is going from one station in a BSS to another without passing through the distribution system. Th



a. Case 1

Addressing mechanisms

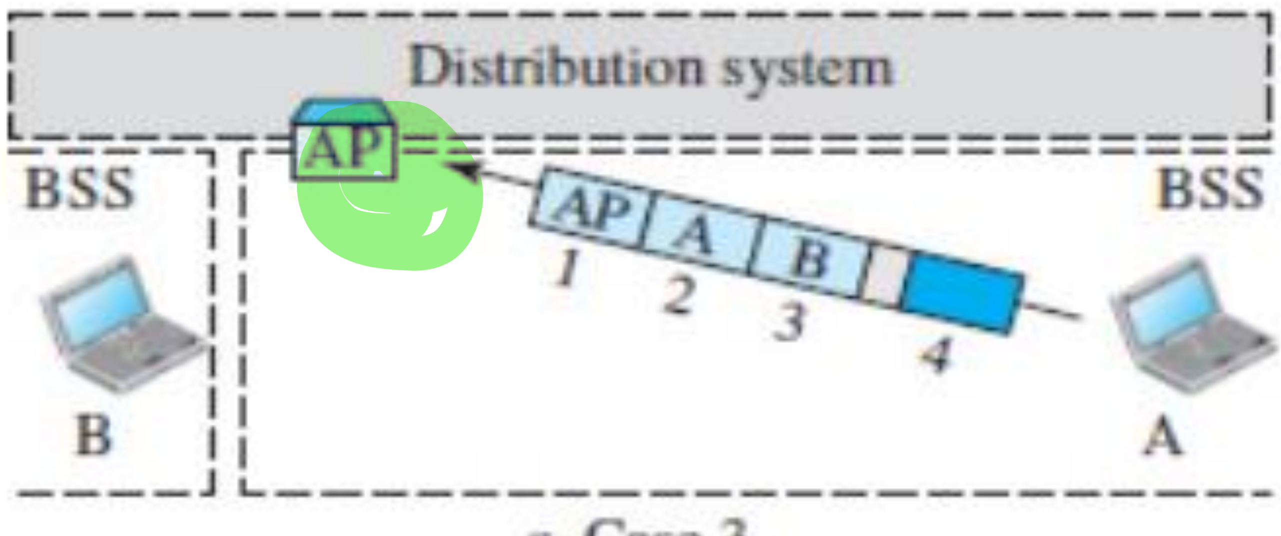
- Case 2:01. In this case, To DS = 0 and From DS = 1. This means that the frame is coming from a distribution system (From DS = 1). The frame is coming from an AP and going to a station. The addresses are as shown in Figure 4.14. Note that address 3 contains the original sender of the frame (in another BSS).



b. Case 2

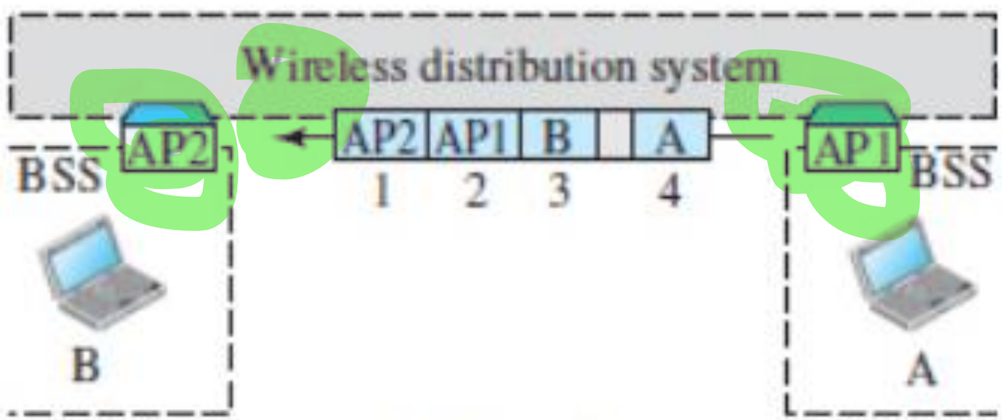
Addressing mechanisms

- Case 3:10. In this case, To DS = 1 and From DS = 0. This means that the frame is going to a distribution system (To DS = 1). The frame is going from a station to an AP. The ACK is sent to the original station. The addresses are as shown in Figure 4.14. Note that address 3 contains the final destination of the frame in the distribution system



Addressing mechanisms

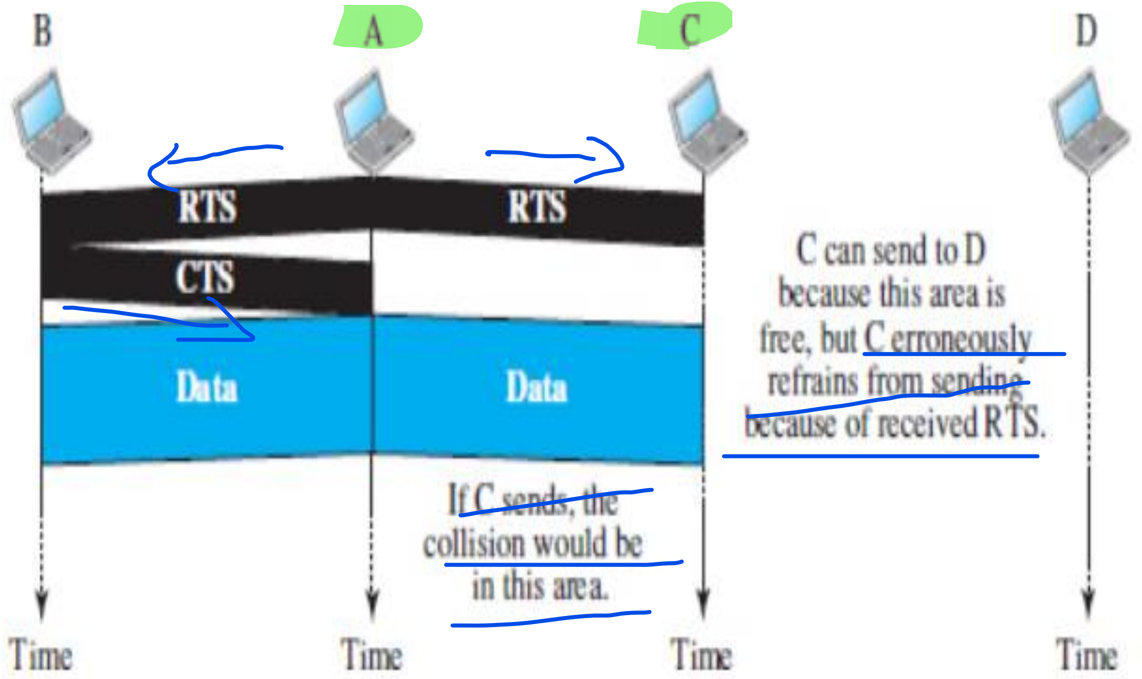
- Case 4:11. In this case, To DS = 1 and From DS = 1. This is the case in which the distribution system is also wireless. The frame is going from one AP to another AP in a wireless distribution system. Here, we need four addresses to define the original sender, the final destination, and two intermediate APs. Figure 4.14 shows the situation.



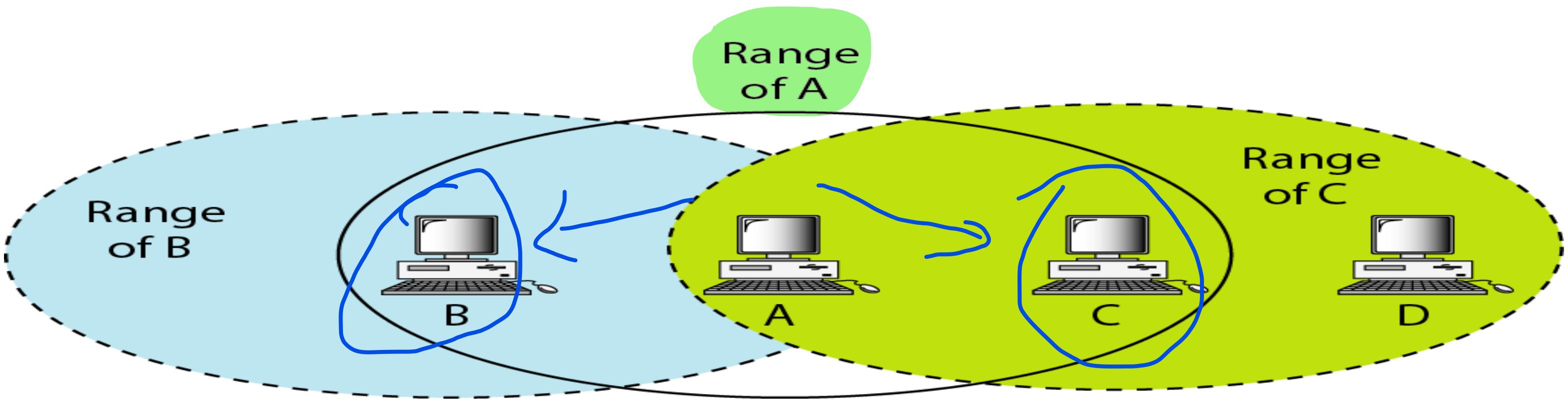
d. Case 4

Exposed Station problem

Figure 4.15 *Exposed-station problem*



Exposed Station problem



C is exposed to transmission from A to B.

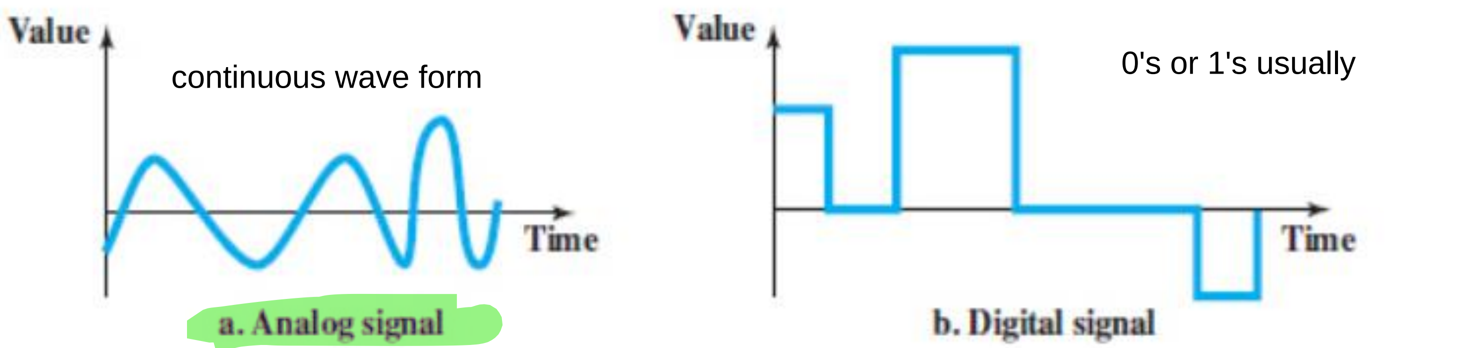
Exposed Station problem

- In this problem a station refrains from using a channel when it is, in fact, available. In Figure 4.15, station A is transmitting to station B. Station C has some data to send to station D, which can be sent without interfering with the transmission from A to B.
- However, station C is exposed to transmission from A; it hears what A is sending and thus refrains from sending. In other words, C is too conservative and wastes the capacity of the channel.
- The handshaking messages RTS and CTS cannot help in this case. Station C hears the RTS from A and refrains from sending, even though the communication between C and D cannot cause a collision in the zone between A and C; station C cannot know that station A's transmission does not affect the zone between C and D.

Physical Layer -Signals

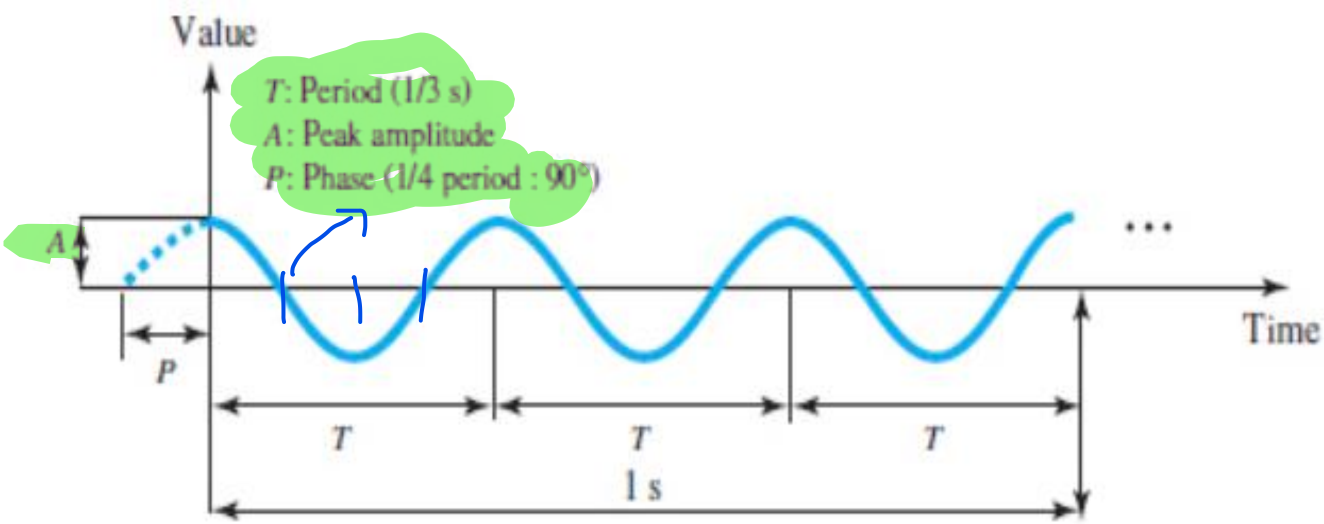
- The signals can be analog or digital. An analog signal takes many values; a digital signal takes a limited number of values as shown in Figure 2.2.

Figure 2.2 *Comparison of analog and digital signals*



Analog Signals

- An analog signal can take one of two forms: periodic or aperiodic (nonperiodic). In data communications, we commonly use periodic analog signals. They can be classified as simple or composite. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals.



Analog Signals

- **Peak Amplitude**: The peak amplitude of a signal is the absolute value of its highest intensity. For electrical signals, peak amplitude is normally measured in volts. The period (T) refers to the amount of time, in seconds, a signal needs to complete one cycle.

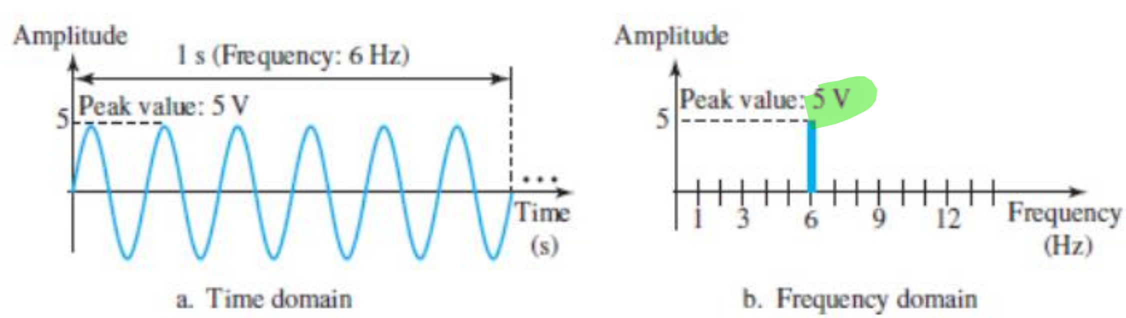
- The **frequency (f)**, measured in hertz (Hz), refers to the number of periods in 1 s. Note that period and frequency are just one characteristic defined in two ways. Period and frequency are inverses of each other, in other words (f = 1/T).

- **Phase**: The term phase describes the position of the waveform relative to time 0. If we think of the wave as something that can be shifted backward or forward along the time axis, phase describes the amount of that shift. It indicates the status of the first cycle. Phase is measured in degrees or radians (360° is 2π rad).
- **Wavelength:** The wavelength is the distance a simple signal can travel in one period. The wavelength binds the period or the frequency of a simple sine wave to the propagation speed in the medium.
- While the frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium. The wavelength can be calculated if one is given the propagation speed of the medium and the frequency of the signal. If we represent wavelength by λ, propagation speed by c, and frequency by f, we get
- $\boxed{\lambda = c / f = c \times T}$

Bandwidth :

The range of frequencies contained in a composite signal is its bandwidth. The bandwidth is the difference between the lowest and highest frequencies in the signal. For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 − 1000, or 4000.

Figure 2.4 *The time-domain and frequency-domain plots of a sine wave*



Digital Signals

- Bit Rate: The bit rate is the number of bits sent in 1 s, expressed in bits per second (bps). The bit rate can be represented as kbps (kilo bits per second, where kilo means one thousand) or Mbps (mega bits per second, where mega means one million)

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

Baud Rate: The baud rate is the number of signal changes (symbols) transmitted per second in a communication channel. It measures how many times the signal can change per second, not necessarily the data rate (bits per second). Each signal change can represent one or more bits depending on the modulation.

Bit rate

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

$$100 \times 24 \times 80 \times 8 = 1{,}536{,}000 \text{ bps} = 1.536 \text{ Mbps}$$

- The **bit length** is the distance 1 bit occupies on the transmission medium. Bit length = 1 / (bit rate)

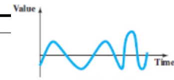
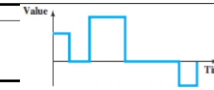
- Find the length of bit for previous example:

- 1/1,536,000 = 0.000000651 s = 0.651 µs

Transmission of Digital Signals

- A digital signal is a composite analog signal with frequencies between zero and infinity
- how can we send a digital signal from point A to point B?
- We can transmit a digital signal by using one of two different approaches:
  - ***Baseband transmission means*** sending a digital signal over a channel without changing it to an analog signal
  - Broadband transmission or modulation means changing the digital signal to an analog signal for transmission

Analog Vs Digital

Analog Signal 	Digital Signal 
a continuous wave that changes over a time period.	a discrete wave that carries information in binary form.
represented by a sine wave.	represented by square waves.
described by the amplitude, period or frequency, and phase.	described by bit rate and bit intervals.
has no fixed range.	has a finite range i.e. between 0 and 1.
more prone to distortion.	less prone to distortion.
transmit data in the form of a wave.	carries data in the binary form i.e. 0 and 1.
The human voice is the best example of an analog signal.	Signals used for transmission in a computer are the digital signal.
analog transmission is the only choice if we have a bandpass channel.	while digital transmission is very desirable, a low-pass channel is needed.
Converting a low-pass analog signal to a bandpass analog signal is traditionally called analog-to-analog conversion.	Converting digital data to a bandpass analog signal is traditionally called digital-to-analog conversion.

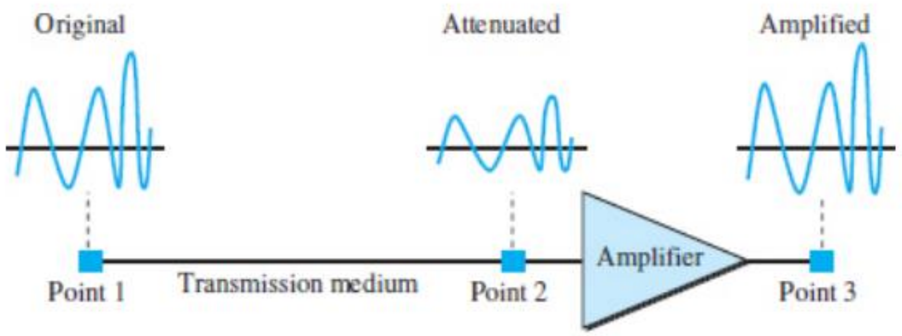
SIGNAL IMPAIRMENT

- 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.
- Three causes of impairment are
  - Attenuation
  - Distortion and
  - noise.

Attenuation and Amplification

- Attenuation means a 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, we need amplification.

Figure 2.6 *Attenuation and amplification*



Decibel

- To show that a signal has lost or gained strength, engineers use the unit of the decibel. The decibel (dB) measures the relative strengths of two signals or one signal at two different points. Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified. Variables P1 and P2 are the powers of a signal at points 1 and 2, respectively.

- $\boxed{\mathbf{dB = 10 \log_{10} (P2 / P1)}}$ $P_2 \rightarrow$ Dest

- Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that P2 = 0.5P1 . In this case, the attenuation (loss of power) can be calculated as

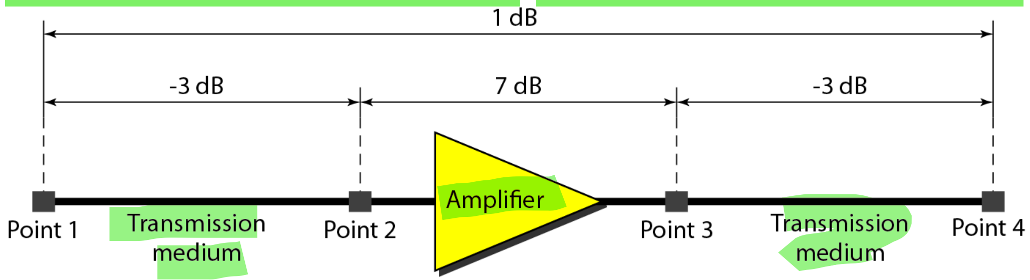
Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that $P_2 = 0.5P_1$. In this case, the attenuation (loss of power) can be calculated as

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

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

Attenuation Example 3

- One reason that engineers use the decibel to measure the changes in the strength of a signal → decibel numbers can be added (or subtracted) when we are measuring several points (cascading) instead of just two
- A signal travels from point 1 to point 4



  - The signal is attenuated by the time it reaches point 2
  - Between points 2 and 3, the signal is amplified
  - Again, between points 3 and 4, the signal is attenuated
- In this case, the decibel value can be calculated as

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

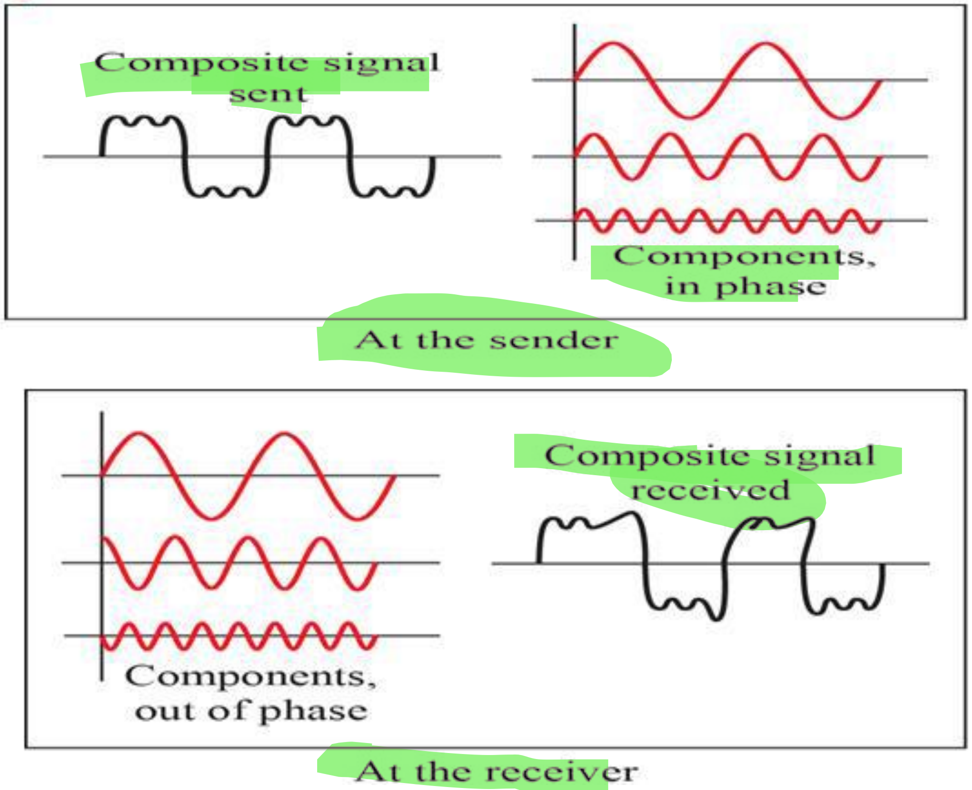
The signal has gained in power.

Distortion

- Distortion means that the signal changes its form or shape. Distortion can occur in a composite signal made up 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 if the delay is not exactly the same as the period duration.

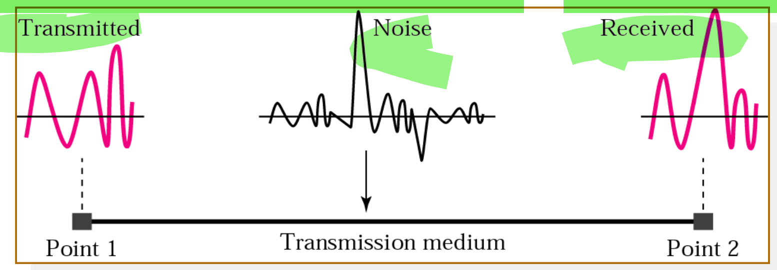
Distortion …

- If the delay is not exactly the same as the period duration, it may create a difference in phase
    - Signal components at the receiver have phases different from what they had at the sender
    - The shape of the composite signal is therefore not the same



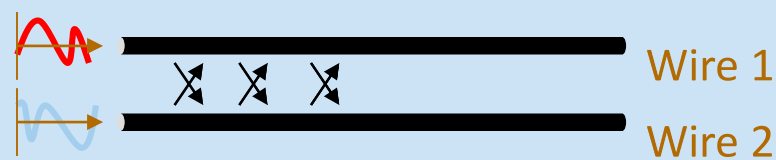
Noise

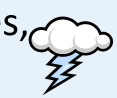
- Noise ⇨ Undesirable signals added between the transmitter and the receiver
  - may corrupt the signal



Types of noise	Description
thermal noise	• random motion of electrons in a wire ◦ creates an extra signal not originally sent by the transmitter

induced noise	• comes from sources such as motors and appliances
	  ◦ Devices act as a sending antenna
	  ◦ Transmission medium acts as the receiving antenna

crosstalk	• effect of one wire on the other
	  ◦ One wire acts as a sending antenna
	  ◦ Other wire as the receiving antenna
	

impulse noise	a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on 

Noise : Signal-to-Noise Ratio

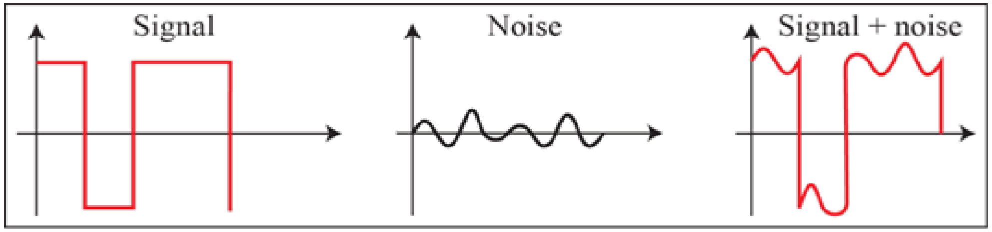
- SNR is the ratio of what is wanted (signal) to what is not wanted (noise).
  - A measurement of signal reception's quality
    - often described in decibel units, SNR_{dB}

$$\boxed{SNR = \frac{\text{average signal power}}{\text{average noise power}}}$$

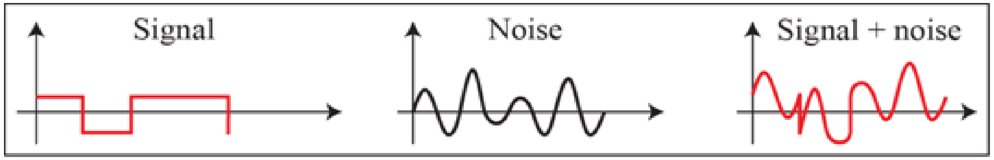
$$SNR_{dB} = 10 \log_{10} SNR$$

- high SNR means the signal is less corrupted by noise.
- Low SNR means the signal is more corrupted by noise

Noise : Signal-to-Noise Ratio



a. High SNR



b. Low SNR

- 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} ?

$$\boxed{SNR = \frac{10000\,\mu w}{1\,\mu w} = 10000\,\mu w}$$

$$\boxed{SNR_{dB} = 10\log_{10} 10000 = 10\log_{10} 10^{4} = 40}$$

Data Rate Limits

- Data rate: 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)**
- Two theoretical formulas were developed to calculate the data rate:
  - Nyquist for a noiseless channel
  - Shannon for a noisy channel.

Noiseless Channel: Nyquist Bit Rate

- Nyquist bit rate formula defines the theoretical maximum bit rate

$$\boxed{\text{Bit Rate} = 2 \times \text{Bandwidth} \times \log_2 L}$$

  - Bandwidth is the bandwidth of the channel
  - *L is the number of signal* levels used to represent data
  - BitRate is the bit rate in bits per second.
- theoretically , given a specific bandwidth, any bit rate can be calculated by increasing the number of signal levels
- Practically there is a limit
  - Increase in the number of signal levels, a burden on the receiver is imposed
    - If the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1
    - If the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels.
      - increasing the levels of a signal reduces the reliability of the system.

Harry Nyquist (1889-1976)

Noiseless Channel: Nyquist Bit Rate

- Does the Nyquist theorem bit rate agree with the intuitive bit rate described in baseband transmission?
  - They match when we have only two levels.
  - In baseband transmission, the bit rate is 2 times the bandwidth if we use only the first harmonic in the worst case
  - However, the Nyquist formula is more general and can be applied to baseband transmission and modulation
  - It can be applied when we have two or more levels of signals.

Noiseless Channel: Nyquist Bit Rate

- 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

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

- 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

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

Noiseless Channel: Nyquist Bit Rate

- We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?
  - 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 (L) 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's Capacity*

- In reality, the channel is always noisy.
- Claude Shannon introduced a formula
  - determine the theoretical highest data rate for a noisy channel: *Shannon's Capacity*

$$\boxed{\text{Capacity} = \text{Bandwidth} \times \log_2(1+\text{SNR})}$$

  - bandwidth is the bandwidth of the channel (Hz)
  - SNR is the signal-to-noise ratio
  - capacity is the capacity of the channel in bits per second
- No indication of the signal level - means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel
- It is a characteristic of the channel, not the method of transmission

Claude Elwood Shannon (1916-2001)

Example 1 – *Shannon's Capacity*

- Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero.
  - 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
  - We cannot receive any data through this channel

Example 2 – *Shannon's Capacity*

- 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 3 – *Shannon's Capacity*

- 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

Example 1: 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?

Using Both Limits

- In practice, both methods used to find the limits and signal levels
- Example 1: 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?
  - 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.
  - Use the Nyquist formula to find the number of signal levels

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

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

Network Performance

- important issue in networking → performance of the network
- Performance parameters:
  - Bandwidth
    - Analog – Hertz
    - Digital – Bits per second (bps)
  - Throughput
    - Actual data rate
  - Latency (delay)
    - Time it takes for an entire message to completely arrive at the destination

Bandwidth

- used in two different contexts with two different measuring values:
- bandwidth in hertz
  - range of frequencies contained in a composite signal or the range of frequencies a channel can pass
    - Ex. bandwidth of a subscriber telephone line is 4 kHz
- bandwidth in bits per second
  - number of bits per second that a channel, a link, or even a network can transmit
    - Ex. bandwidth of a Fast Ethernet network (or the links in this network) is a maximum of 100 Mbps - means that this network can send 100 Mbps.
- relationship between the bandwidth in hertz and bandwidth in bits per second
  - increase in bandwidth in hertz means an increase in bandwidth in bits per second
    - depends on whether we have baseband transmission or transmission with modulation

Bandwidth – examples

- The bandwidth of a subscriber line is 4 kHz for voice or data.
- The bandwidth of this line for data transmission can be up to 56,000 bps using a sophisticated modem to change the digital signal to analog.
- If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps.

Throughput

- Measure of how fast we can actually send data through a network
- Bandwidth in bits per second and throughput are different
  - A link may have a bandwidth of *B bps, but we can only send T bps* through this link with *T always less than B.*
  - *Bandwidth is a potential* measurement of a link; the throughput is an actual measurement of how fast we can send data.
  - For example, we may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps
    - cannot send more than 200 kbps through this link.
  - Imagine a highway designed to transmit 1000 cars per minute from one point to another.
  - Due to congestion on the road, 100 cars per minute are transmitted
    - The bandwidth is 1000 cars per minute; the throughput is 100 cars per minute.

Throughput - 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?

Throughput - 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?

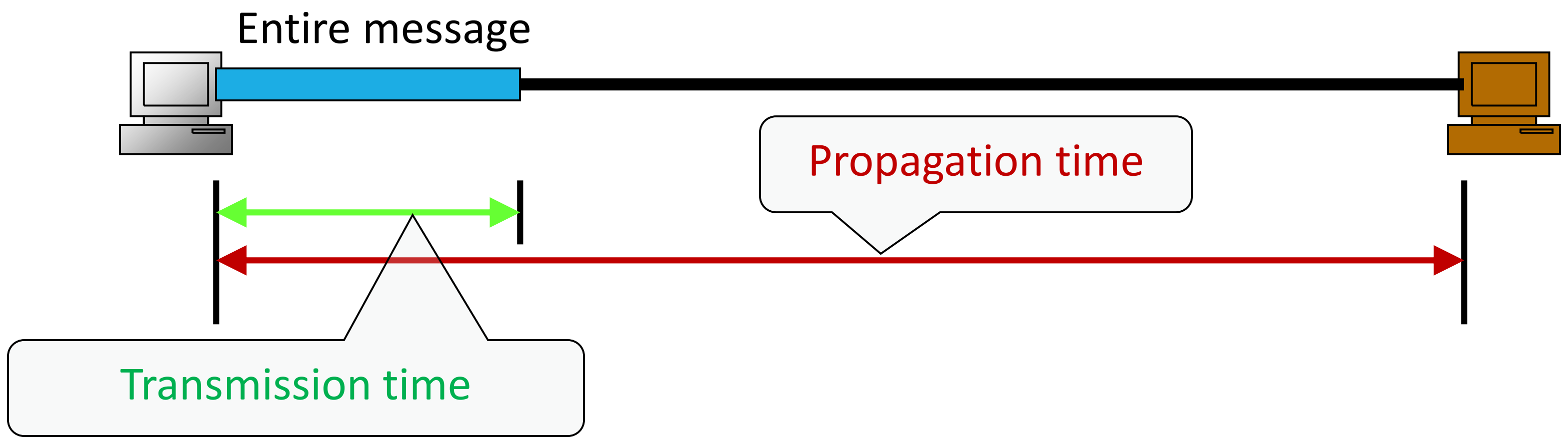
$$\boxed{\text{Throughput} = \frac{12{,}000 \times 10{,}000}{60} = 2\ \text{Mbps}}$$

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

Latency

- **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
- Composed of
  - Propagation time
  - Transmission time
  - Queuing time
  - Processing time

Latency = propagation time
+ transmission time
+ queuing time
+ processing delay



Latency - *Propagation* time

- **time required for a bit to travel from the source to the** destination.
- **Propagation time = Distance / (Propagation Speed)**
- The propagation speed of electromagnetic signals depends on the medium and on the frequency of the signal
  - For example, in a vacuum, light is propagated with a speed of 3 × 108 m/s. It is lower in air; it is much lower in cable.

Example Latency - *Propagation* time

- 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.

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

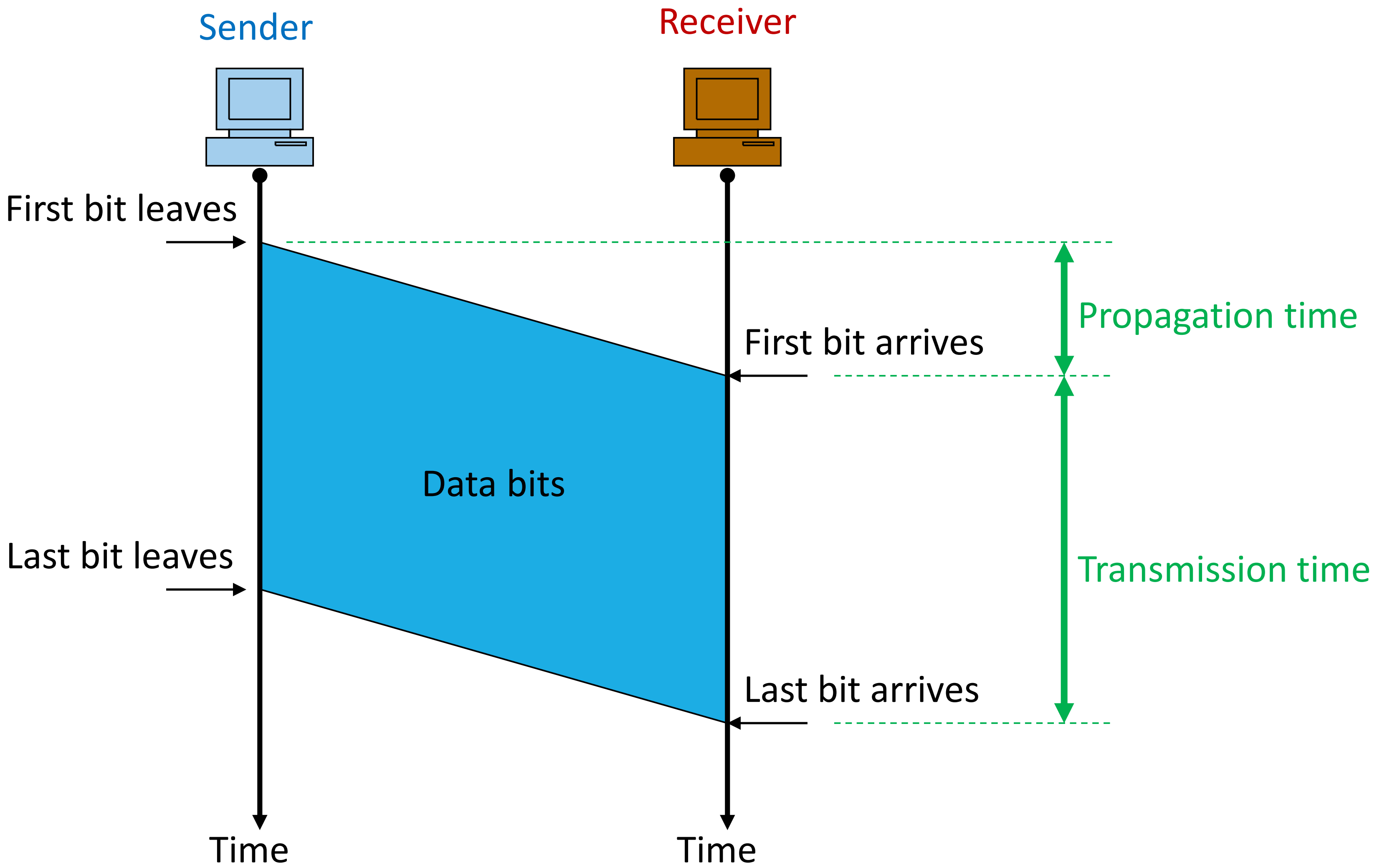
- The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

Latency - Transmission time

- In data communications , message is sent not just 1 bit
  - The first and last bit may take a time equal to the propagation time to reach its destination
    - The first bit leaves earlier and arrives earlier; the last bit leaves later and arrives later.
- **Transmission time** → time between the first bit leaving the sender and the last bit arriving at the receiver
- The **transmission time of a message** depends on the size of the message and the bandwidth of the channel

Transmission time = (Message size) / Bandwidth

Latency - Transmission time



Example 1 - Transmission time

- 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 2.4×10^8 m/s.

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

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

Note : Message is short and the bandwidth is high so the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

Example 2 - Transmission time

- 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.

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

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

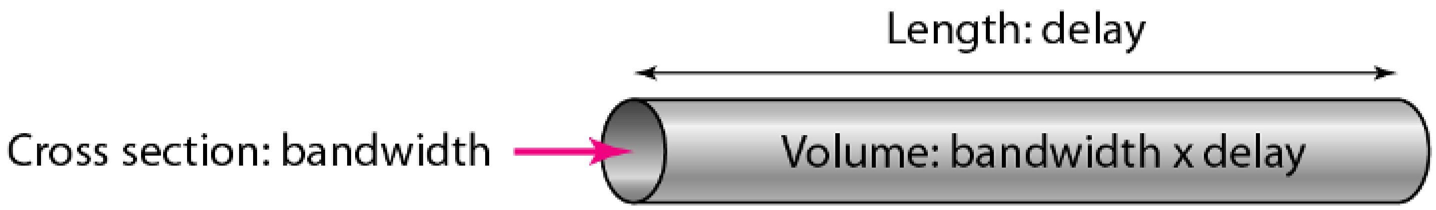
Note: Message is very long and the bandwidth is not very high so the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

Latency – Queuing time

- time needed for each intermediate or end device to hold the message before it can be processed.
  - not a fixed factor - changes with the load imposed on the network.
    - When there is heavy traffic on the network, the queuing time increases
  - An intermediate device, such as a router, queues the arrived messages and processes them one by one
  - If there are many messages, each message will have to wait

Bandwidth-Delay Product

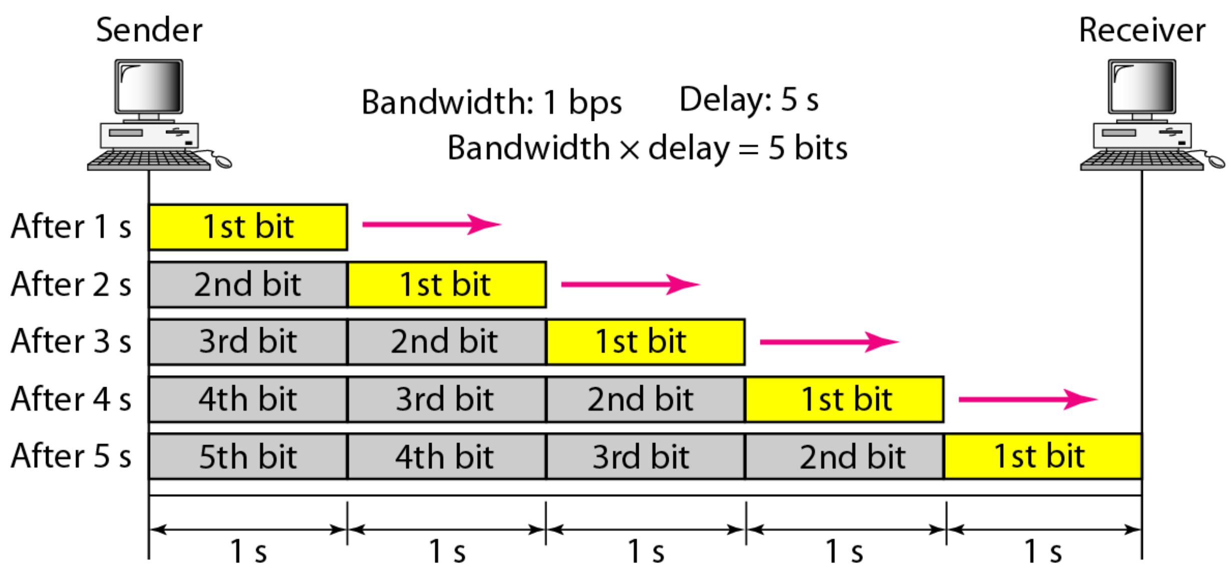
- The product of bandwidth and delay is the number of bits that can fill the link



- two hypothetical cases as examples
  - Filling the links with bits
  - Filling the pipe with bits
- Two cases show that product of bandwidth and delay is the number of bits that can fill the link

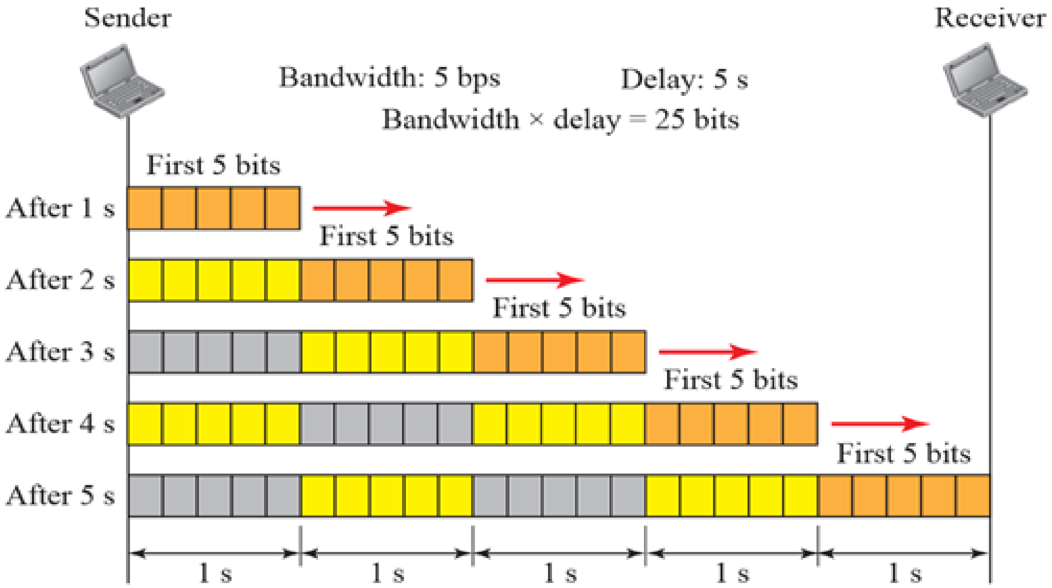
Bandwidth-Delay Product

- Filling the links with bits
  - assuming that we have a link with a bandwidth of 1 bps (unrealistic).
  - We also assume that the delay of the link is 5 s (also unrealistic).
  - We want to see what the bandwidth-delay product means in this case?
  - This product 1 × 5 is the maximum number of bits that can fill the link.
  - There can be no more than 5 bits at any time on the link.



Bandwidth-Delay Product

- Filling the pipe with bits
  - Assuming we have a bandwidth = 5 bps, and delay of the link = 5 s
  - There can be maximum 5 × 5 = 25 bits on the line.
  - The reason is that, at each second, there are 5 bits on the line
  - The duration of each bit is: 1/5 = 0.20 s.



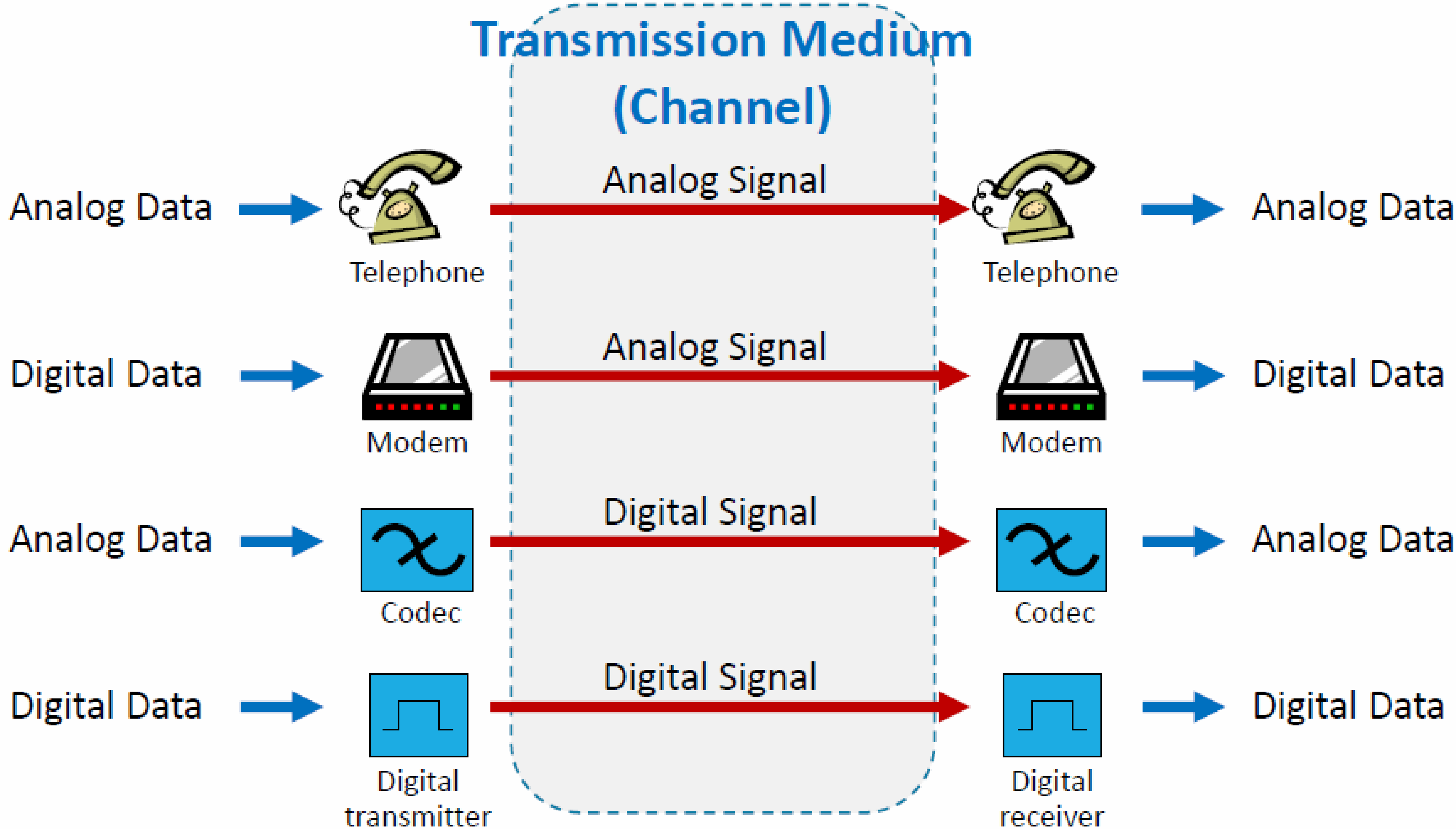
Bandwidth-Delay Product

- The number of bits that can fill the link is important if we need to send data in bursts and wait for the acknowledgment of each burst before sending the next one
- To use the maximum capability of the link, we need to make the size of our burst 2 times the product of bandwidth and delay
  - sender should send a burst of data of (2 × bandwidth × delay) bits
  - sender then waits for receiver acknowledgment for part of the burst before sending another burst
  - amount 2 × bandwidth × delay is the number of bits that can be in transition at any time

Jitter

- Related to delay is **jitter**
- **Jitter** is a problem if different packets of data encounter different delays and the application using the data at the receiver site is time-sensitive (audio and video data, for example)
- If the delay for the first packet is 20 ms, for the second is 45 ms, and for the third is 40 ms, then the real-time application that uses the packets endures jitter.

Analog and Digital



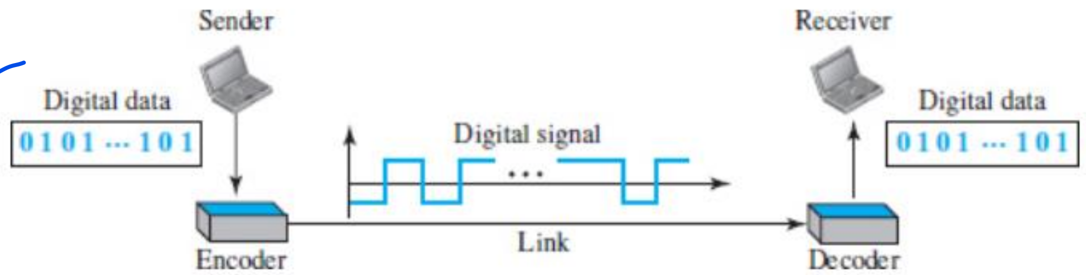
Conversion Techniques

- Digital-to-digital Conversion
  - Line Coding
    - Line Coding Schemes
  - Block Coding
  - Scrambling
- Analog-to-digital Conversion
  - Pulse Code Modulation (PCM)
  - Delta Modulation (DM)

- Digital-to-analog Conversion
  - Aspects of Digital-to-Analog Conversion
  - Amplitude Shift Keying
  - Frequency Shift Keying
  - Phase Shift Keying
  - Quadrature Amplitude Modulation
- Analog-to-analog Conversion
  - Amplitude Modulation (AM)
  - Frequency Modulation (FM)
  - Phase Modulation (PM)

Digital-to-Digital Conversion

- The conversion involves three techniques: line coding, block coding, and scrambling.
- **Line coding** is the process of converting digital data to digital signals. We assume that data, in the form of text, numbers, graphical images, audio, or video, are stored in computer memorys as sequences of bits

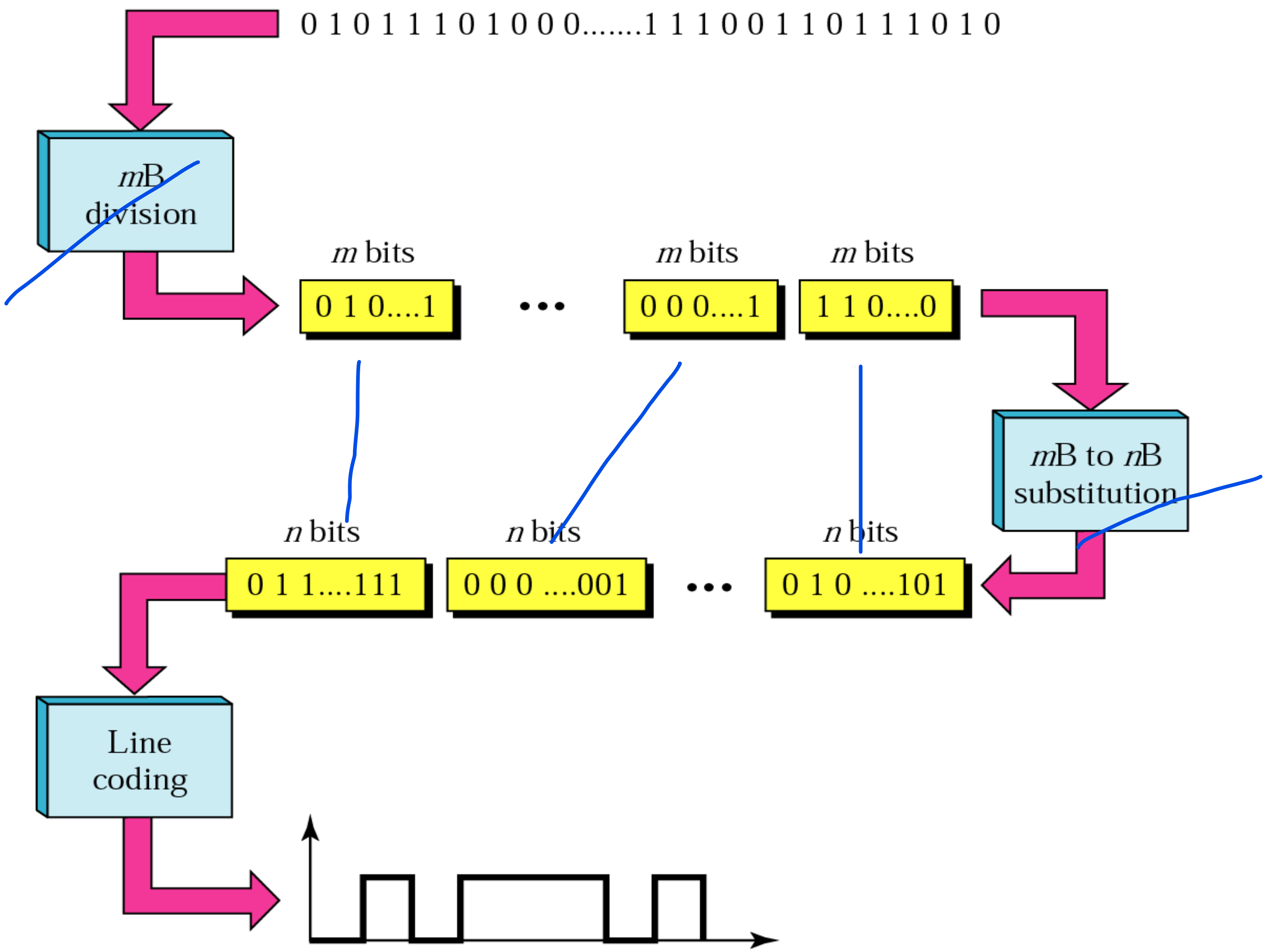


Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are re-created by decoding the digital signal

Block Coding

- How to ensure synchronization and provide some kind of inherent error detecting?
  - Solution – need redundancy
    - Block coding → give this redundancy and improve the performance of line coding
- Block coding changes a block of *m bits into a block of n bits, where n > m*
  - *referred to as an mB/nB encoding*) → distinguishes block encoding from multilevel encoding
    - multilevel encoding → written without a slash eg. 8B6T
    - it replaces each *m*-bit group with an *n*-bit group
    - involves three steps:
      - **Division:** sequence of bits is divided into groups of *m bits*
      - **Substitution**: substitute an *m-bit group with an n-bit* group
      - **Combination**: *n-bit groups are combined to form a stream* (n > m)

Block Coding

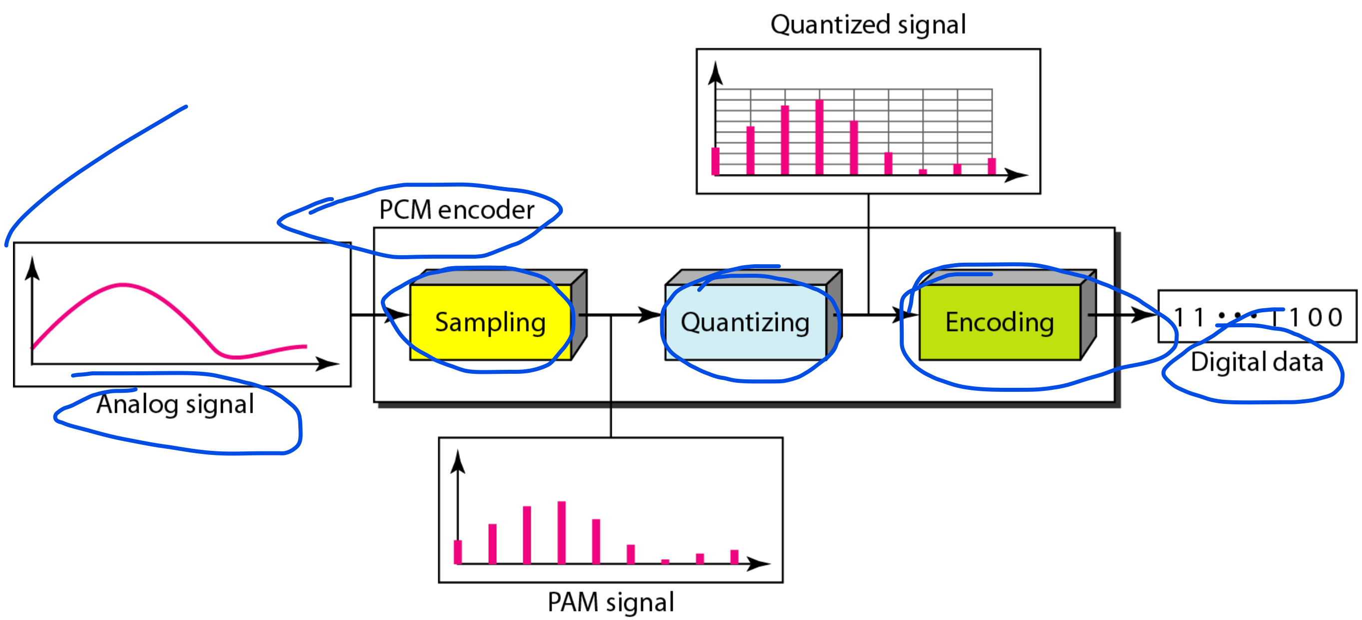


Analog to Digital Conversion

- an analog signal → created by a microphone or camera
- The amplitude of analog signal can take any value over a continuous range i.e. it can take on an infinite values.
- Digital signal amplitude can take on finite values.
- Digital signal is superior to an analog signal
- Analog signal can be converted into digital by sampling and quantizing-- digitization
  - two techniques
    - pulse code modulation
    - delta modulation
- Digital data are converted to digital signal using line coding, block coding or scrambling techniques

Pulse Code Modulation (PCM)

- Change an analog signal to digital data (digitization)
- A PCM encoder has three processes
  - Sampling - analog signal is sampled
  - Quantization- sampled signal is quantized
  - encoding - quantized values are encoded as streams of bits



PCM - Sampling

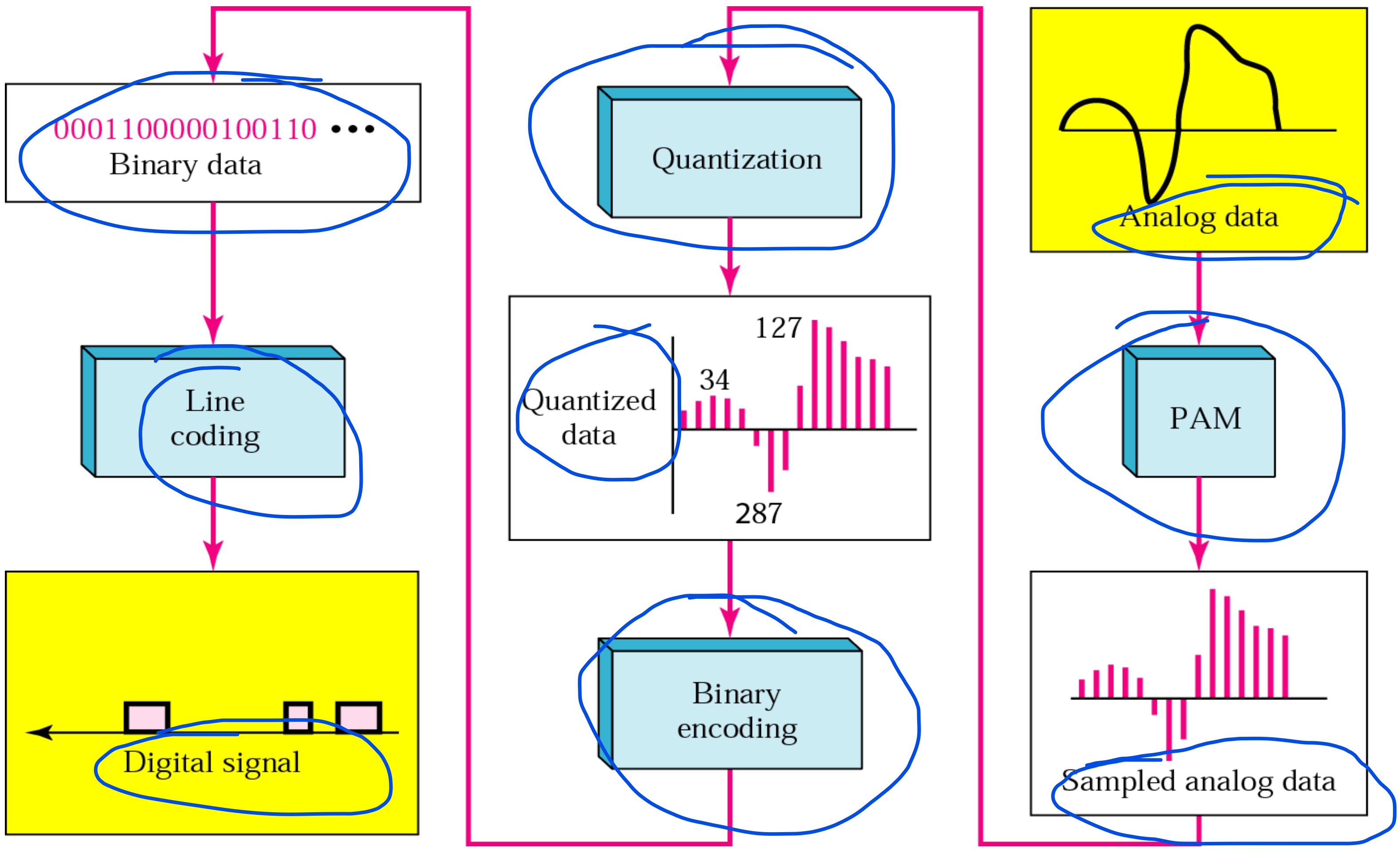
- The analog signal is sampled every T_s *s, where* T_s *is* the sample interval or period
  - **Called as pulse amplitude modulation (PAM)**
  - Result →an analog signal with nonintegral values
- The inverse of the sampling interval is called the *sampling rate or sampling frequency and denoted by* f_s
  - $f_s = 1/T_s$

PCM: Encoding Example

- We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?
- **Solution**
- The human voice normally contains frequencies from 0 to 4000 Hz
- So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

PCM: The Whole Process



Minimum Required Bandwidth

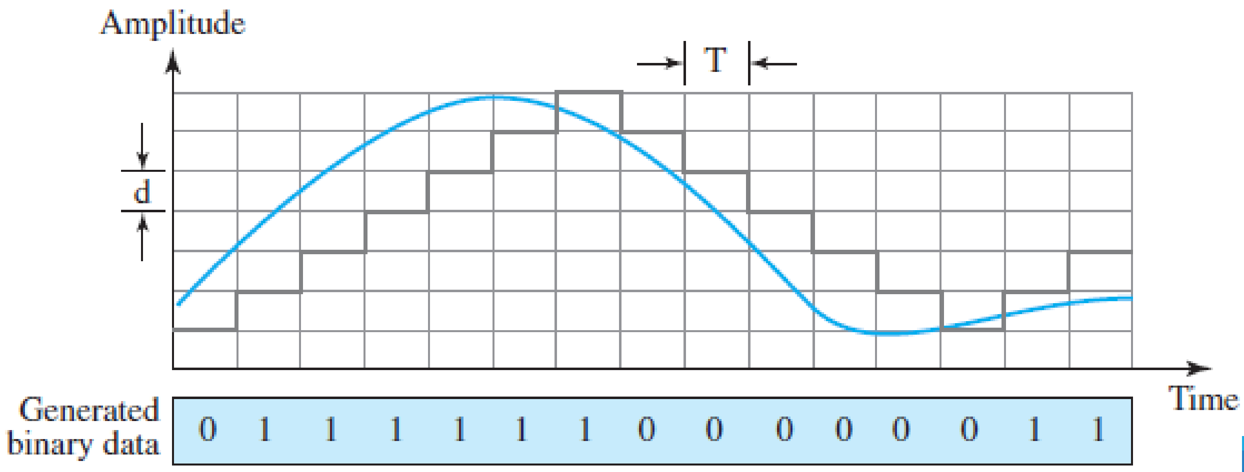
- It can be proved that the minimum bandwidth of the digital signal is $\boldsymbol{B_{min} = n_b \times B_{analog}}$

Delta Modulation (DM)

- PCM is a very complex technique
- Delta modulation reduce the complexity of PCM
- PCM finds the value of the signal amplitude for each sample whereas DM finds the change from the previous sample → bit
- DM → no code words; bits are sent one after another
  - Modulator is used at the sender site to create a stream of bits from an analog signal
  - Demodulator is used at receiver site to creates the analog signal from the received digital data
- DM is not perfect → Quantization error introduced in the process
  - The quantization error of DM is much less than that for PCM

DM...

- The process records the small positive or negative changes, called delta δ
  - If the delta is positive, the process records a 1
  - If it is negative, the process records a 0
- A better performance achieved if the value of δ is not fixed
  - Adaptive delta modulation - value of δ changes according to the amplitude of the analog signal.

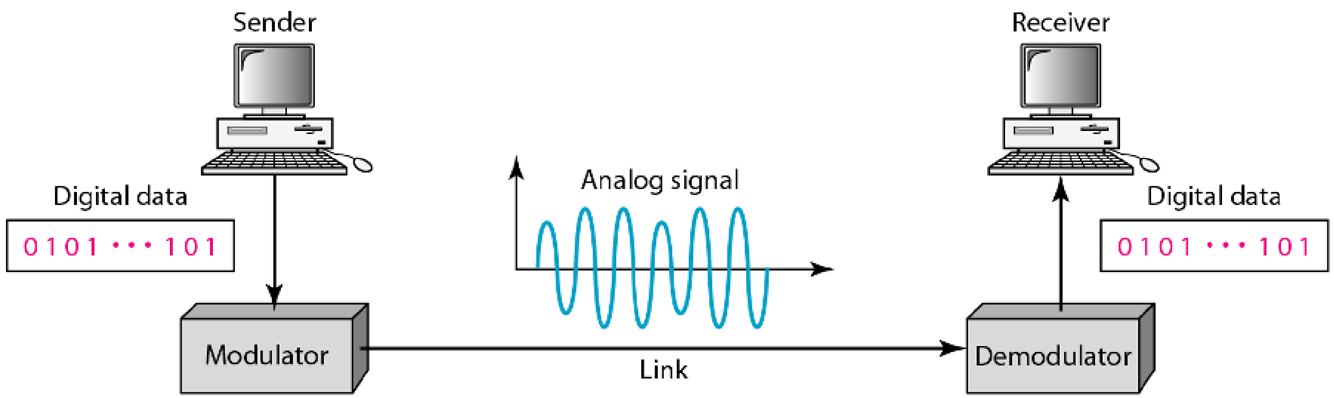


Analog-to-analog conversion

- Change an analog signal to a new analog signal with a smaller bandwidth
  - used when only a band-pass channel is available
- Three methods:
  - Amplitude modulation (AM) : amplitude of a carrier is changed based on the changes in the original analog signal
  - Frequency modulation (FM): the phase of a carrier is changed based on the changes in the original analog signal
  - Phase modulation (PM): the phase of a carrier signal is changed to show the changes in the original signal

Digital-to-Analog Conversion

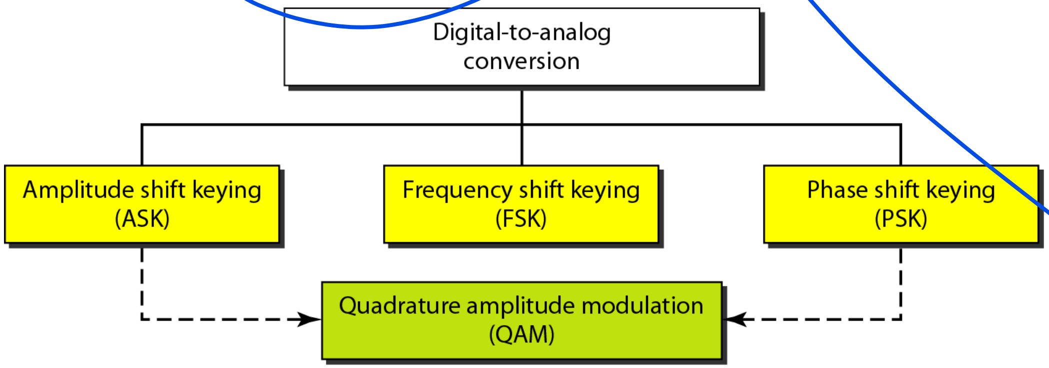
- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data when a band-pass channel is available
- Required to send digital data over a band-pass channel
  - Also known as ***modulation***



(band-pass channel)

Digital-to-analog conversion

- Four methods:
  - Amplitude shift keying (ASK): amplitude of a carrier is changed using the digital data
  - Frequency shift keying (FSK): frequency of a carrier is changed using the digital data.
  - Phase shift keying (PSK): phase of a carrier signal is changed to represent digital data.
  - Quadrature amplitude modulation (QAM): both amplitude and phase of a carrier signal are changed to represent digital data.

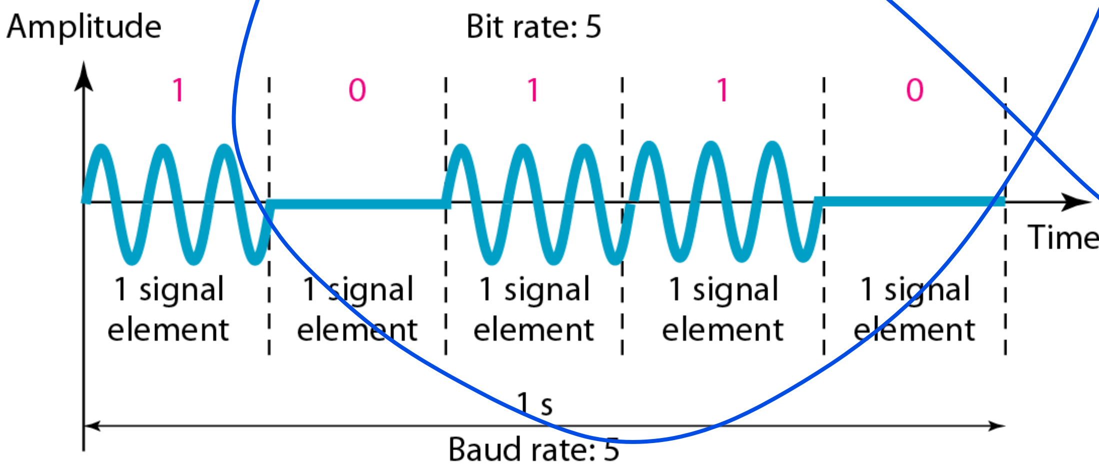


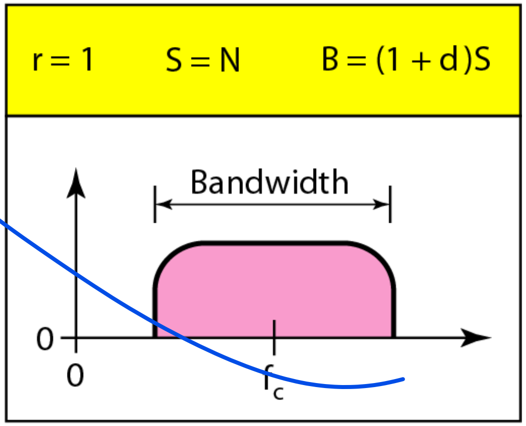
Amplitude shift keying

- Amplitude of the carrier signal is varied to create signal elements
- Both frequency and phase remain constant while the amplitude changes
- Binary ASK (BASK)
  - uses only two amplitude levels
- Multilevel ASK (implemented with QAM)
  - more than two levels

Binary Amplitude Shift Keying

- Several levels (kinds) of signal elements with a different amplitude can exists
- ASK → normally implemented using only two levels
  - *Binary amplitude shift keying or on-off keying (OOK)*
  - Peak amplitude of
    - One signal level is 0
    - Other signal level is the same as the amplitude of the carrier frequency





middle of the bandwidth - *fc* : Carrier frequency

$0 < d < 1$

Bandwidth for ASK

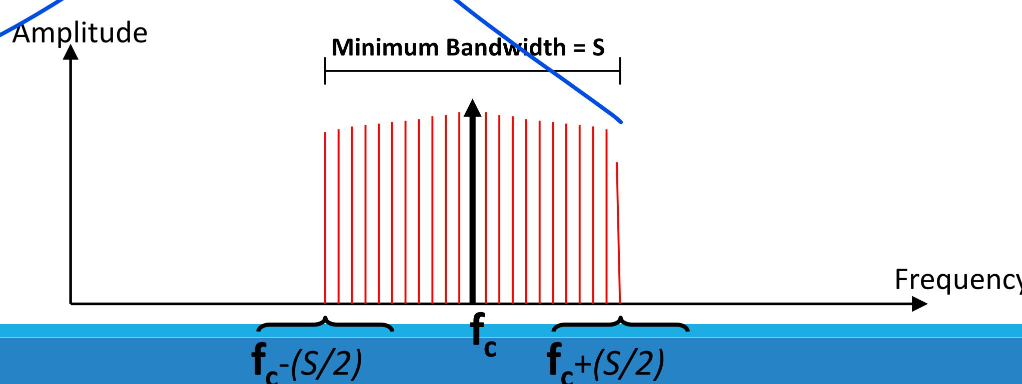
- carrier signal is only one simple sine wave -> has a continuous set of frequencies
- modulation produces a nonperiodic composite signal
- bandwidth is proportional to the signal rate (baud rate)

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

  - S is the signal rate
  - *B is the bandwidth*
  - *d depends on the modulation and filtering* process
  - The value of *d is between 0 and 1*

    $B_{min} = S$

    $B_{max} = 2S$

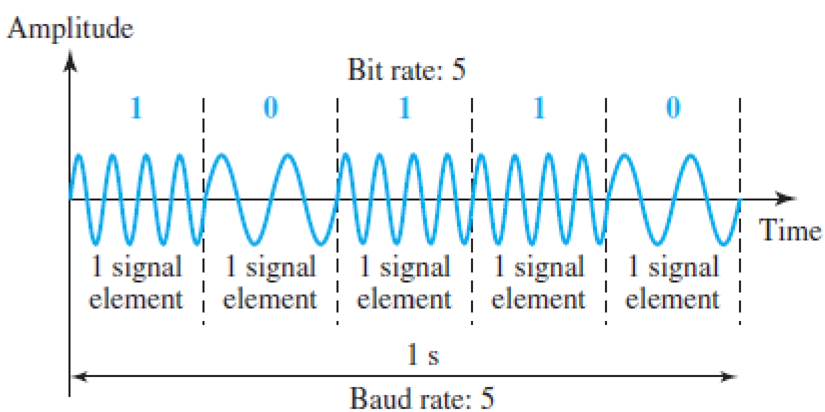


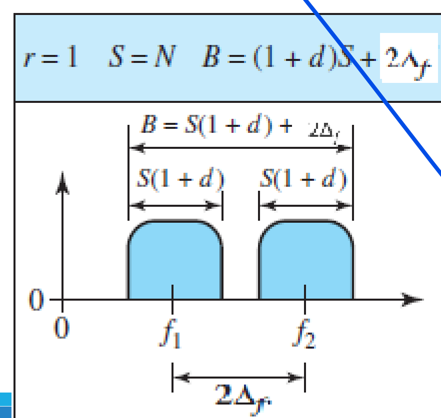
Frequency Shift Keying

- frequency of the carrier signal is varied to represent data
  - The frequency of the modulated signal is constant for the duration of one signal element
  - but changes for the next signal element if the data element changes
- Both peak amplitude and phase remain constant for all signal elements
- Binary FSK (or BFSK)
  - consider two carrier frequencies *f1 and f2*
- Multilevel FSK (MFSK)
  - more than two frequencies

Binary Frequency Shift Keying

- Consider two carrier frequencies *f1 and f2*
  - *first carrier frequency for* the data element 0
  - Second *carrier frequency for* the data element 1
  - carrier frequencies are very high and the difference between them is very small
- *Both f1 and f2 are Δ_f apart from the midpoint between the two bands*
- *The difference* between the two frequencies is $2\Delta_f$





Bandwidth for *BFSK*

- carrier signal is only one simple sine wave -> has a continuous set of frequencies
- modulation produces a nonperiodic composite signal
- Consider FSK is two ASK signals, each with its own carrier frequency (*f1 or f2)*
- *difference between the two frequencies is* $2\Delta_f$
  - minimum value of $2\Delta_f$ should be at least *S for* the proper operation of modulation and demodulation.

$$B = (1 + d) \times S + \text{frequency shift} = (1 + d) \times S + 2\Delta_f$$

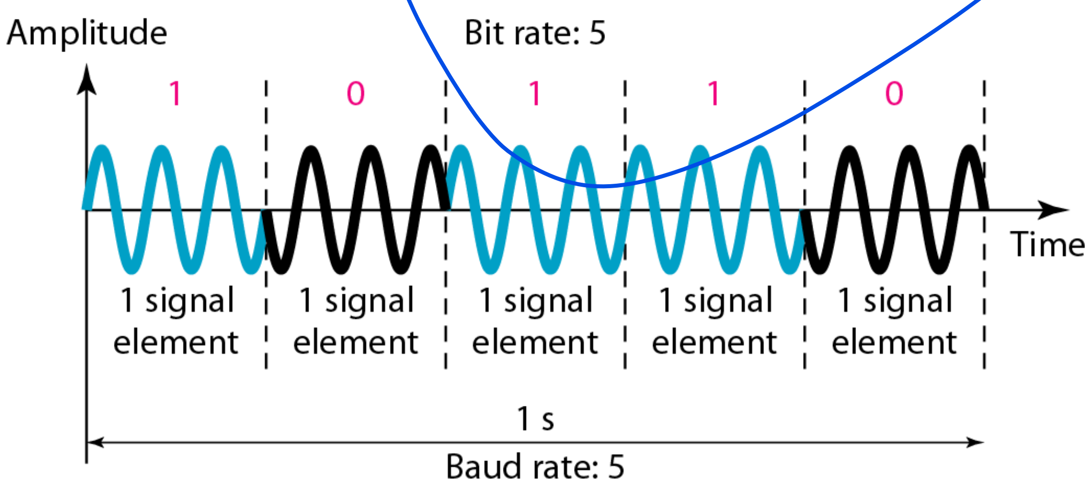
  - S is the signal rate
  - *B is the bandwidth*
  - *d depends on the modulation and filtering* process
  - The value of *d is between 0 and 1*

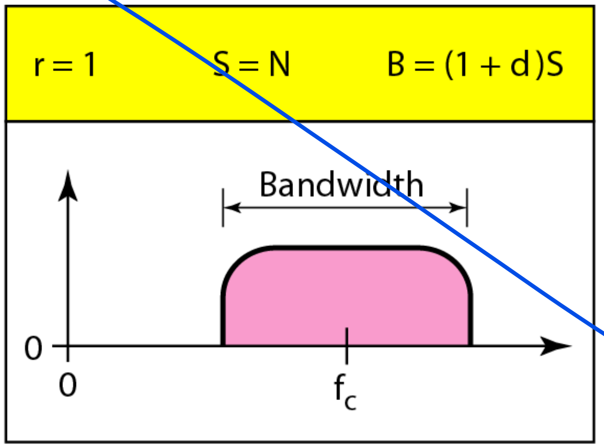
Phase Shift Keying

- Phase of the carrier is varied to represent two or more different signal elements
- Both peak amplitude and frequency remain constant as the phase changes
- Today, PSK is more common than ASK or FSK
- Binary Phase Shift Keying
  - only two signal elements
- Quadrature PSK (QPSK)
  - uses two separate BPSK modulations

Binary Phase Shift Keying

- Or Binary PSK
- only two signal elements
  - one with a phase of 0◦
  - other with a phase of 180◦





Binary PSK

- The bandwidth is the same as that for binary ASK, but less than that for BFSK
- No bandwidth is wasted for separating two carrier signals

PSK	ASK
PSK is less susceptible to noise than ASK as noise can change the amplitude easier than it can change the phase	ASK is more susceptible to noise than PSK
the criterion for bit detection is the phase of the signal	the criterion for bit detection is the amplitude of the signal
superior to FSK because we do not need two carrier signals	need two carrier signals
needs more sophisticated hardware to be able to distinguish between phases	No sophisticated hardware

Comparison ASK, FSK, PSK

Parameters	ASK	FSK	PSK
Variable characteristics	Amplitude	Frequency	Phase
Bandwidth	B =(1+d)S d is due to modulation & filtering ,lies between 0 & 1.	B=(1+d)×S+2Δf	B=(1+d)×S
Noise immunity	low	High	High
Complexity	Simple	Moderately complex	Very complex
Error probability	High	Low	Low
Performance in presence of noise	Poor	Better than ASK	Better than FSK
Bit rate	Suitable upto 100 bits/sec	Suitable upto about 1200 bits/sec	Suitable for high bit rates

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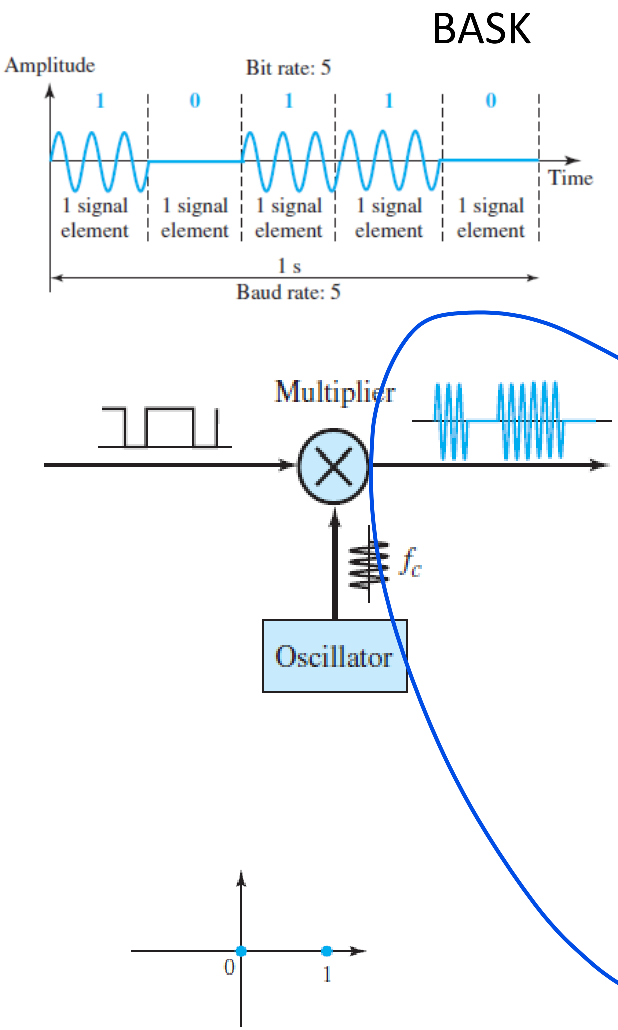
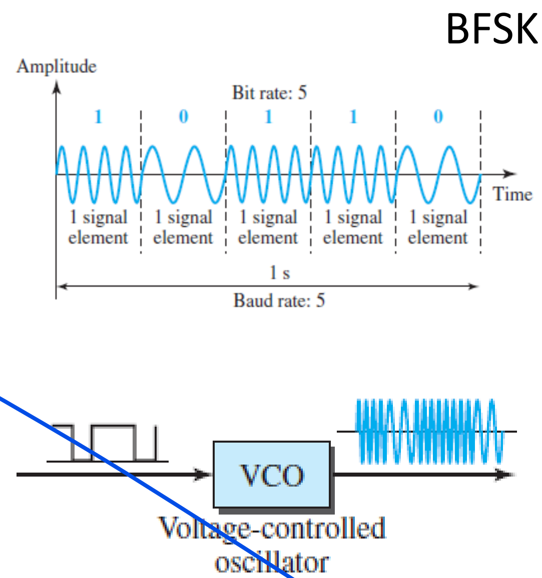
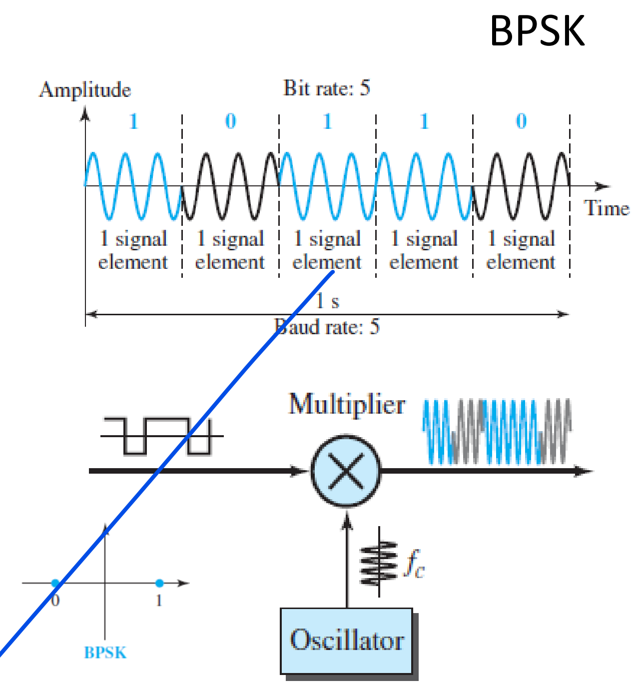
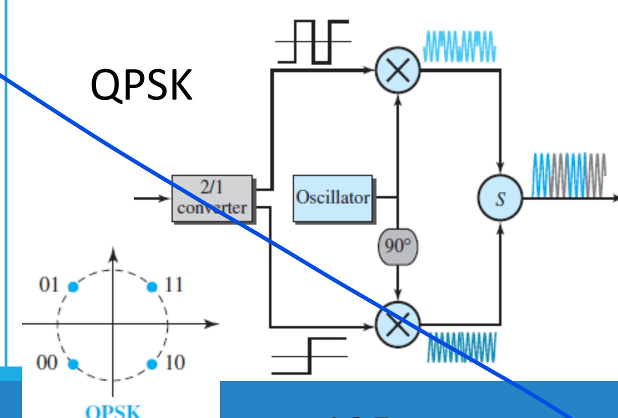
Comparison ASK, FSK, PSK









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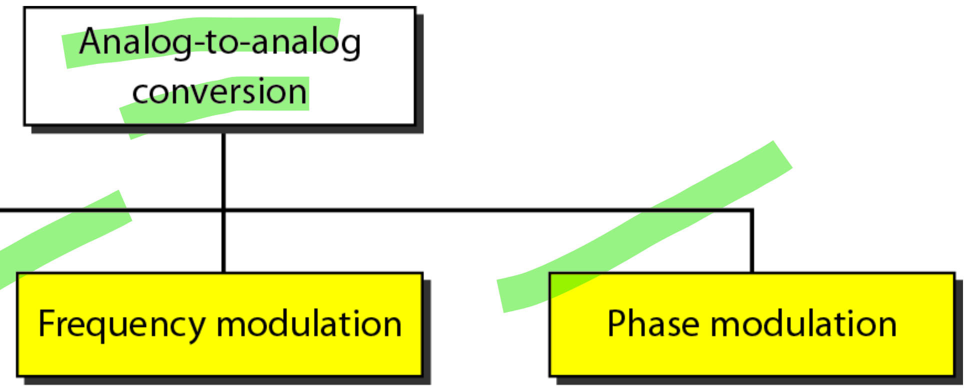
Analog-to-Analog Conversion

Modulation

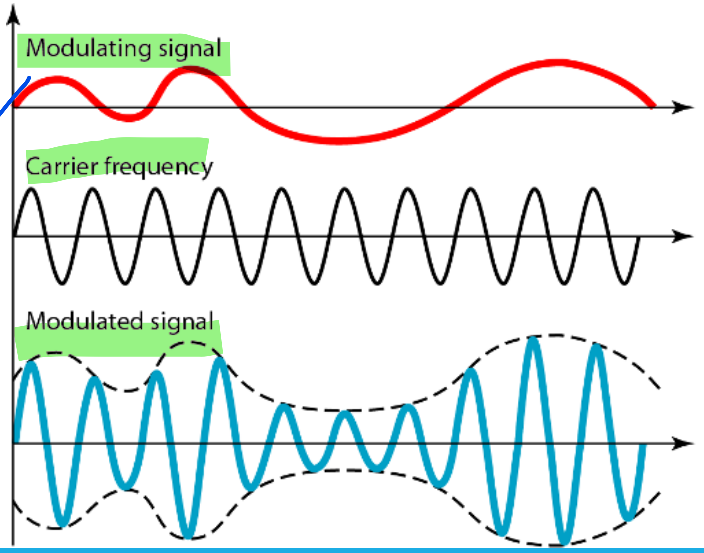
- Process of transmitting analog information by an analog signal
- Although the signal is already analog, modulation is needed if a band-pass channel is available to us eg. Radio
  - The government assigns a narrow bandwidth to each radio station
  - each radio station – produces low-pass signal
    - all stations low analog pass signal in the same range
  - Listening to different stations, the low-pass signals need to be shifted, each to a different range → bans pass signal

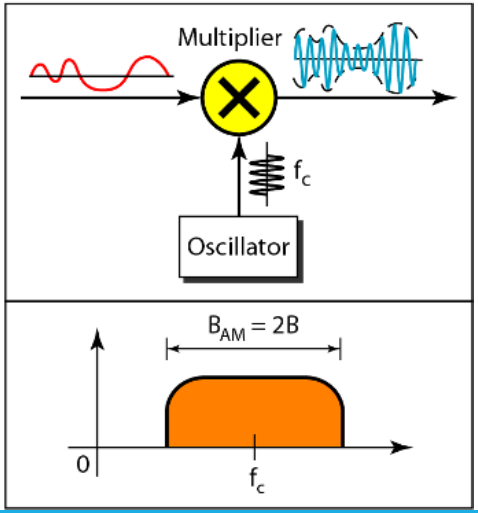


amplitude is modulated

Amplitude Modulation (AM)

- Amplitude of the carrier signal is modulated to follow the changing amplitudes of the modulating signal
  - Only the amplitude changes to follow variations in the information
  - The frequency and phase of the carrier remain the same
- The modulating signal is the envelope of the carrier
- The amplitude of the carrier signal changed according to the amplitude of the modulating signal using a simple multiplier





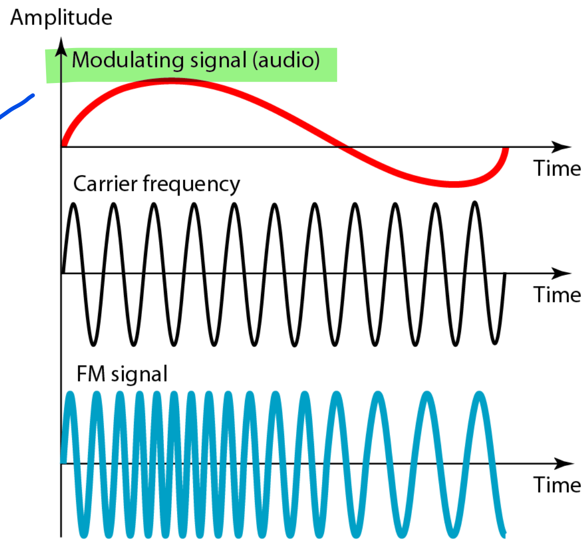
AM Bandwidth

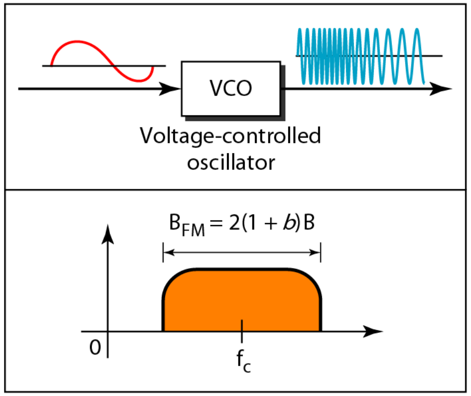
- The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency
  - $B_{AM}=2B$
- As the signal components above and below the carrier frequency carry exactly the same information, some implementations discard one-half of the signals and cut the bandwidth in half.

Frequency Modulation (FM)

Frequency is modulated

- Frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal
  - Peak amplitude and phase of the carrier signal remain constant,
  - As the amplitude of the information signal changes, the frequency of the carrier changes correspondingly
- Implemented by using a voltage-controlled oscillator as with FSK
  - The frequency of the oscillator changes according to the input voltage which is the amplitude of the modulating signal





FM Bandwidth

- The actual bandwidth is difficult to determine exactly
- Can be shown empirically that it is
  - several times that of the analog signal or
  - $2(1 + \beta)B$ *where* β *is a factor that depends on modulation technique* with a common value of 4.

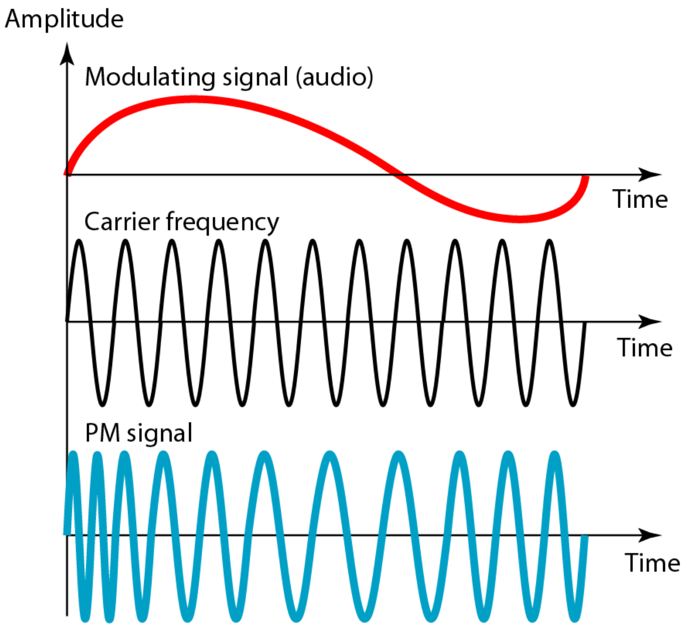
Phase is modulated

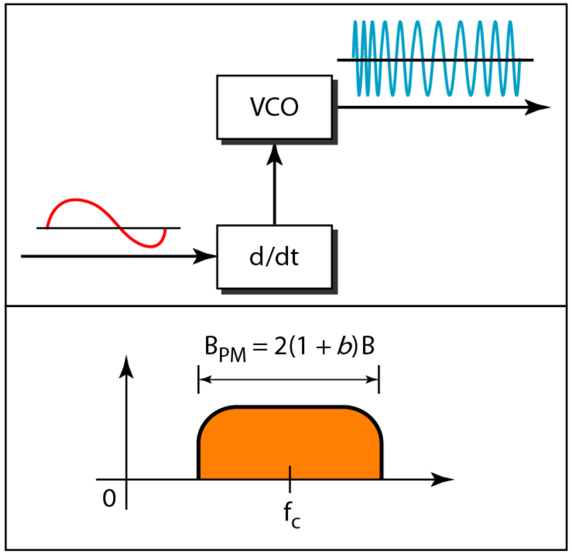
Phase Modulation (PM)

- phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal
  - peak amplitude and frequency of the carrier signal remain constant
  - As the amplitude of the information signal changes, the phase of the carrier changes correspondingly
- It can be proved mathematically that PM is the same as FM with one difference
  - FM: the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal
  - PM: the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal

Phase Modulation (PM)

- PM is normally implemented by using a voltage-controlled oscillator along with a derivative
- Frequency of the oscillator changes according to the derivative of the input voltage, which is the amplitude of the modulating signal.





PM Bandwidth

- The actual bandwidth is difficult to determine exactly, but it can be shown empirically that it is several times that of the analog signal

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

- Although the formula shows the same bandwidth for FM and PM, the value of β is lower in the case of PM (around 1 for narrowband and 3 for wideband)

Bandwidth: Revisited

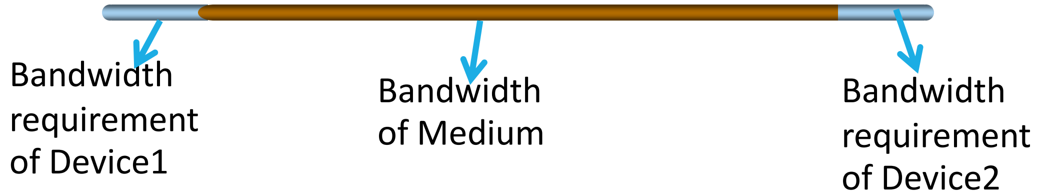
- Bandwidth = highest frequency – lowest frequency
  - Used to measures maximum data transfer rate of a network or Internet connection in a given amount of time
  - Describe network speeds
    - does not measure **how fast bits of data move** from one location to another (data rate)
    - measures **how much data can flow** through a specific connection at one time
    - The wider the bandwidth, the faster data can be sent
- Analog devices bandwidth → measured in hertz (cycles per second)
- Digital devices→ bandwidth is measured in bits per second

Bandwidth Utilization

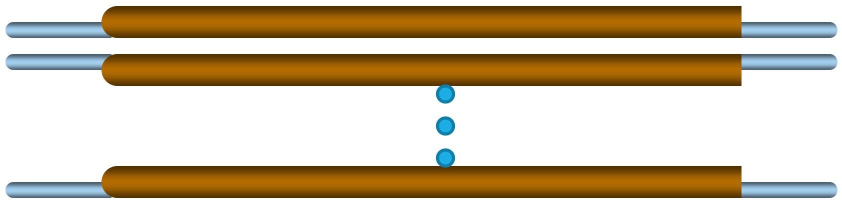
- Reality→ limited bandwidth
  - Requires wise use → main challenge of electronic communication
  - Bandwidth Utilization → depends on application
- Two goal of bandwidth utilization
  - Efficiency
    - Achieved by combining several low-bandwidth channels into one with larger bandwidth
      - Technique is called **multiplexing**
  - Privacy and anti-jamming
    - Achieved by expanding the bandwidth of a channel to insert redundancy
      - Technique is called **spectrum spreading**

Transmission Medium

- Devices connected to transmission medium as



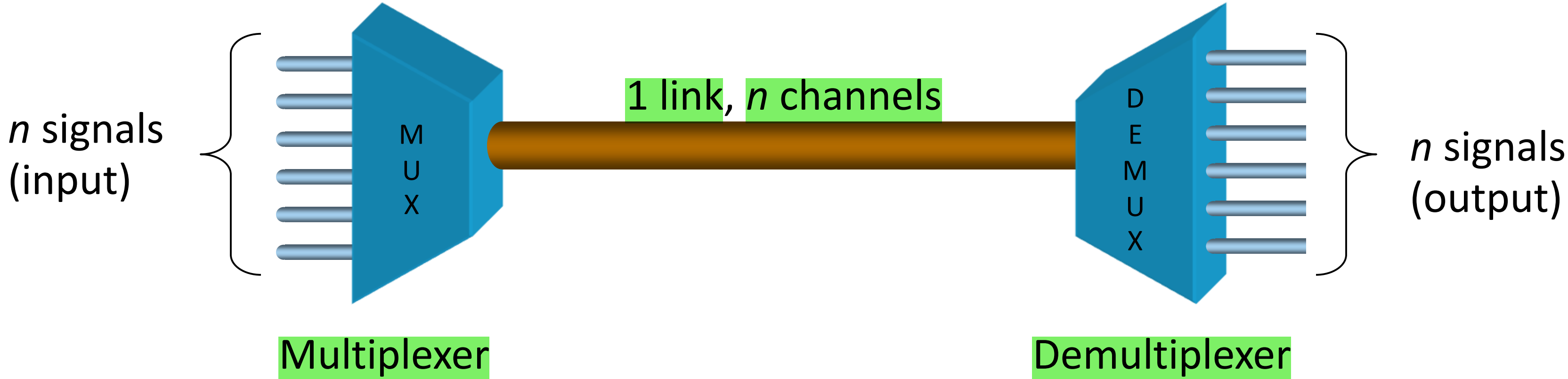
- Increased usage of data and telecommunications results into increased traffic
- How to accommodate this increase?
  - Add individual links each time a new channel is needed
- Today's technology includes high bandwidth media such as optical fiber and terrestrial and satellite microwaves



  - Each medium has a bandwidth far in excess of that needed for the average transmission signal- costly and also become difficult to manage

Shared Link

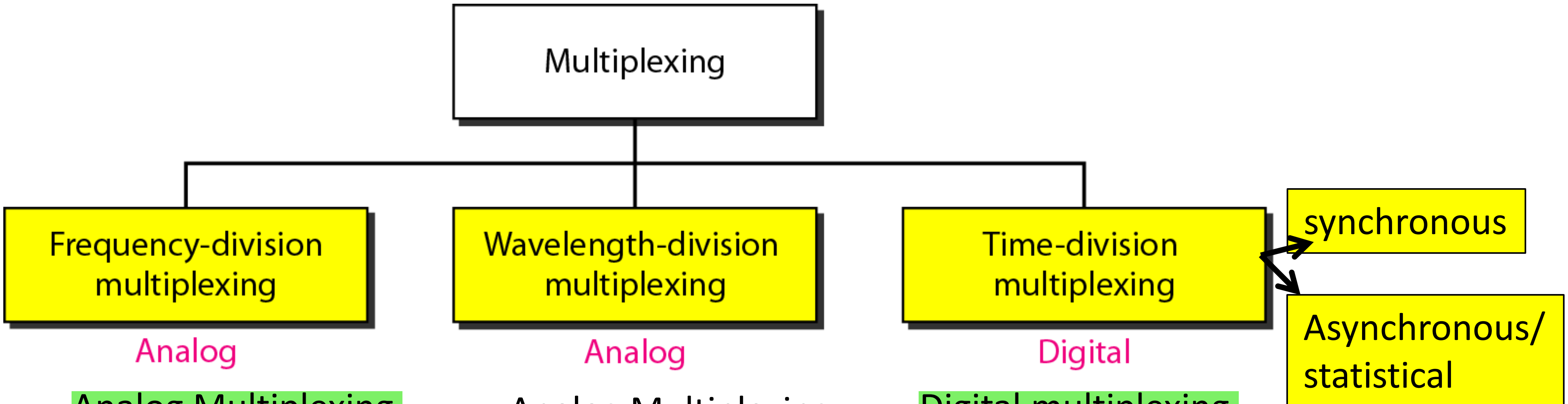
- Bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted
  - Solution: the higher bandwidth link can be shared to carry multiple signals from different devices connected to it
    - Maximizes bandwidth utilization → gives efficient system
    - reduce the line cost and also it would be easier to keep track of one line than several lines
- Shared Link: uses multiplexing technique



Multiplexing

- Process of combining multiple signals (analog or digital), commonly from slow devices, onto one very fast communications link
  - achieved by a device called a Multiplexer (MUX) and Demultiplexer (DEMUX)
    - MUX: combines the lines at sender on the left to direct their transmission streams into a single stream (many-to-one)
    - DEMUX: separates the stream at receiver back into its component transmissions (one-to-many) and directs them to their corresponding lines
- allow the simultaneous transmission of multiple signals across a single data link
  - Link → physical path
  - Channel →portion of a link that carries a transmission between a given pair of lines
    - One link can have many *(n) channels*

Multiplexing …



Analog Multiplexing technique to combine analog signals

Analog Multiplexing technique to combine optical signals

Digital multiplexing technique to combine several low rate channels into high rate one

- Carrier division multiple access (CDMA) is considered as a fourth multiplexing category

Frequency Division Multiplexing (FDM)

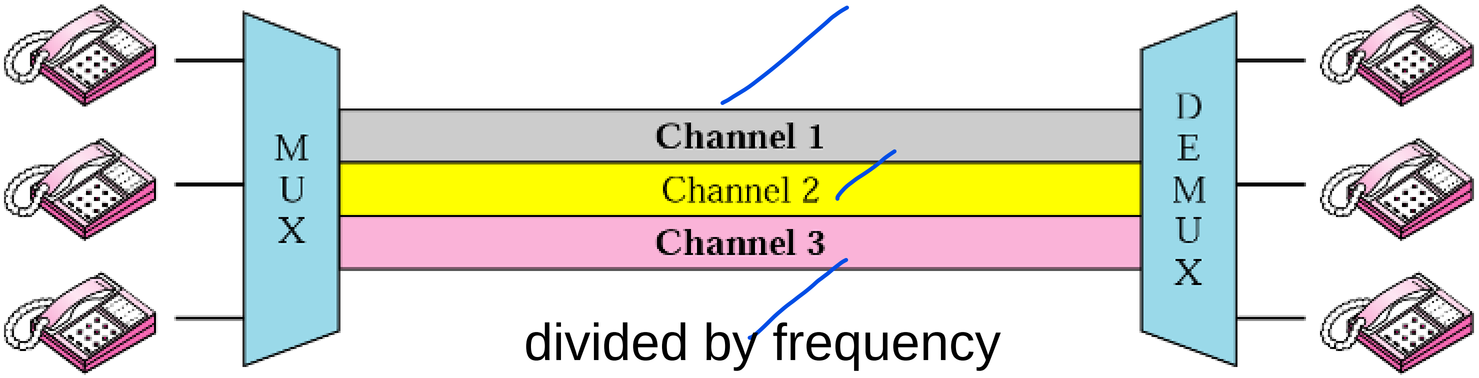
An analog multiplexing technique to combine signals

- Can be applied when the bandwidth of a link in (hertz) is greater than the combined bandwidths of the signal to be transmitted
  - Medium BW > Channel BW
- Transmitting all of the signals along the same high speed link simultaneously

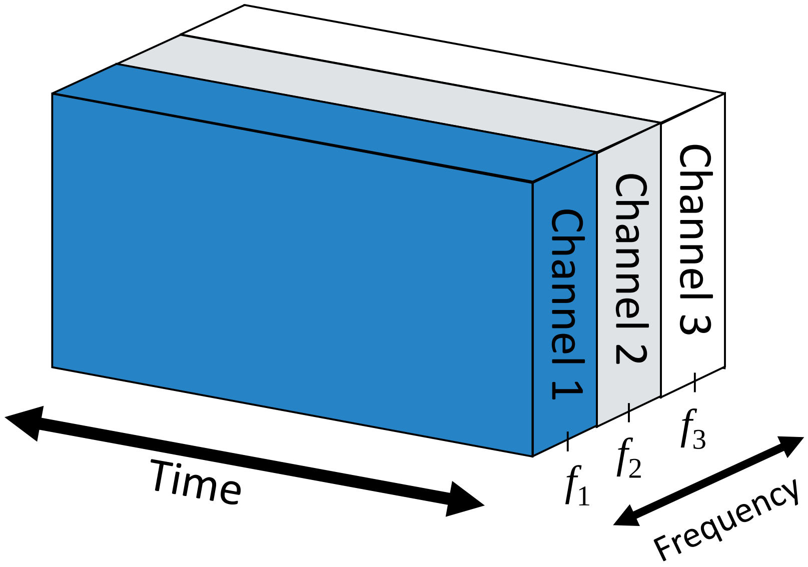
FDM …

- Each signal is modulated to a different carrier frequency
  - Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal
    - These bandwidth ranges are the channels through which the various signals travel
      - Channels can be separated by strips of unused bandwidth -guard bands -to prevent signals from overlapping
    - In addition, carrier frequencies must not interfere with the original data frequencies
- These modulated signals are then combined into a single composite signal that can be transported by the link
- E.g., broadcast radio
- Channel allocated even if no data

Conceptual View of FDM



divided by frequency
same time

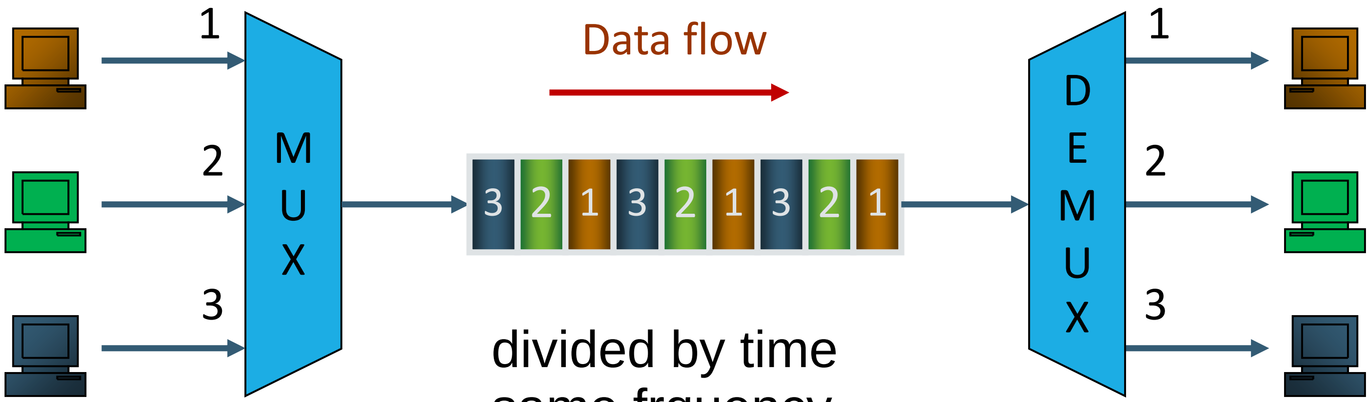


Time Division Multiplexing (TDM)

A digital multiplexing technique to combine data

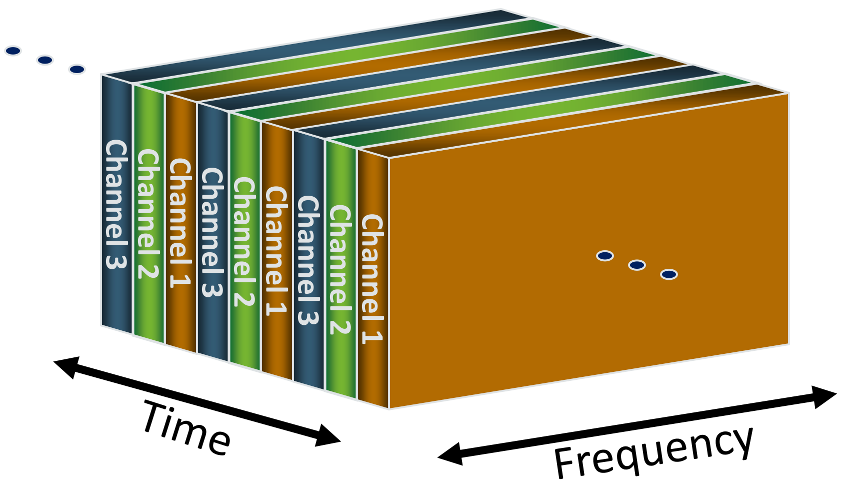
- **digital process that allows several connections** to share the high bandwidth of a link
- Time is shared
  - Each connection occupies a portion of time in the link
  - Same link of FDM is shown sectioned by time rather than by frequency
- Digital data from different sources are combined into one timeshared link
- Analog data can be sampled, changed to digital data, and then multiplexed by using TDM

Conceptual View of TDM



divided by time
same frquency

- Portions of signals 1, 2, and 3 occupy the link sequentially
- Data in a message from source 1 always go to one specific destination 1, 2, 3, or 4
- The delivery is fixed and unvarying, unlike switching



Summary

- Signals get impaired by attenuation, distortion, and noise
- For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate.
- For a noisy channel, we need to use the Shannon capacity to find the maximum bit rate.
- Attenuation, distortion, and noise can impair a signal.
- Attenuation is the loss of a signal's energy due to the resistance of the medium.
- Distortion is the alteration of a signal due to the differing propagation speeds of each of the frequencies that make up a signal.
- Noise is the external energy that corrupts a signal.
- The bandwidth-delay product defines the number of bits that can fill the link.

Summary

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in the digital data
- Digital-to-analog conversion can be accomplished in several ways: amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK)
- Quadrature amplitude modulation (QAM) combines ASK and PSK
- A constellation diagram shows us the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature)
- Analog-to-analog conversion is the representation of analog information by an analog signal
- Conversion is needed if the medium is bandpass in nature or if only a bandpass bandwidth is available to us
- Analog-to-analog conversion can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM).