UNIT - 5

LOCAL AREA NETWORK

PHYSICAL LAYER

SYLLABUS

Unit V

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.

ETHERNET

- Ethernet is a family of wired computer networking technologies.
- A Local Area Network(LAN) is a computer network that is designed for a limited geographic area such as a building or campus.
- LAN can be used as an isolated network to connect computers in an organization for the sole purpose of sharing resources, most LANs today are also linked to a Wide Area Network(WAN) or the internet.
- In the 1980s and 1990s, several different types of wired LANs were used.
- 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).

- The IEEE has also created several physical-layer standards for different LAN protocols.
- All these wired LANs use a media access method to solve the problem of sharing the media.
- The relationship of the IEEE 802 standard to the TCP/IP protocol suite is shown in figure.

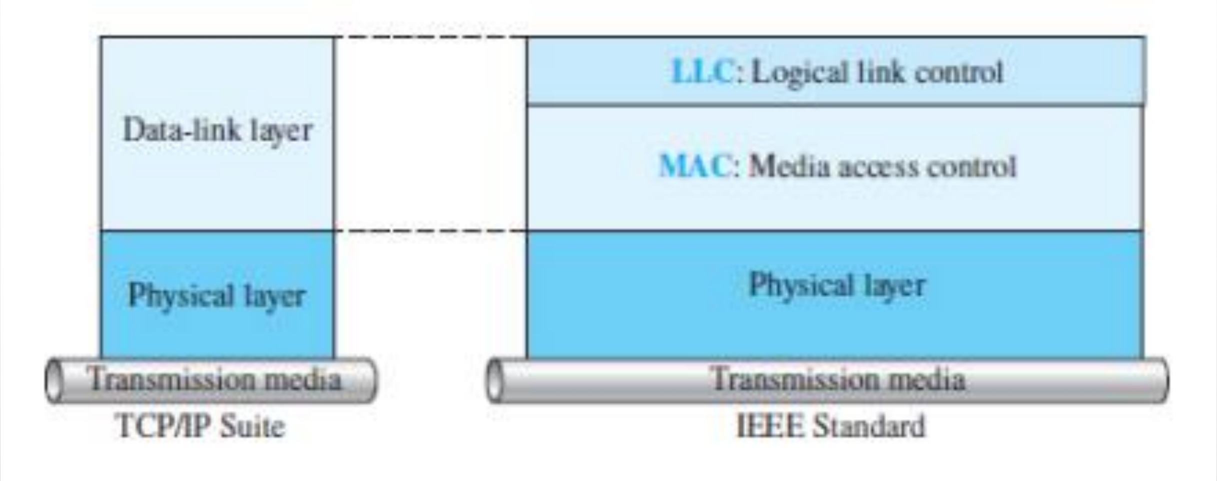


Figure : IEEE standard for wired LANs

- The Ethernet LAN was developed in the 1970s by Robert Metcalfe and David Boggs.
- It has gone through four generations:
    - Standard Ethernet (10 mbps),
    - Fast Ethernet (100 mbps),
    - Gigabit Ethernet (1 gbps), and
    - 10 gigabit Ethernet (10 gbps).

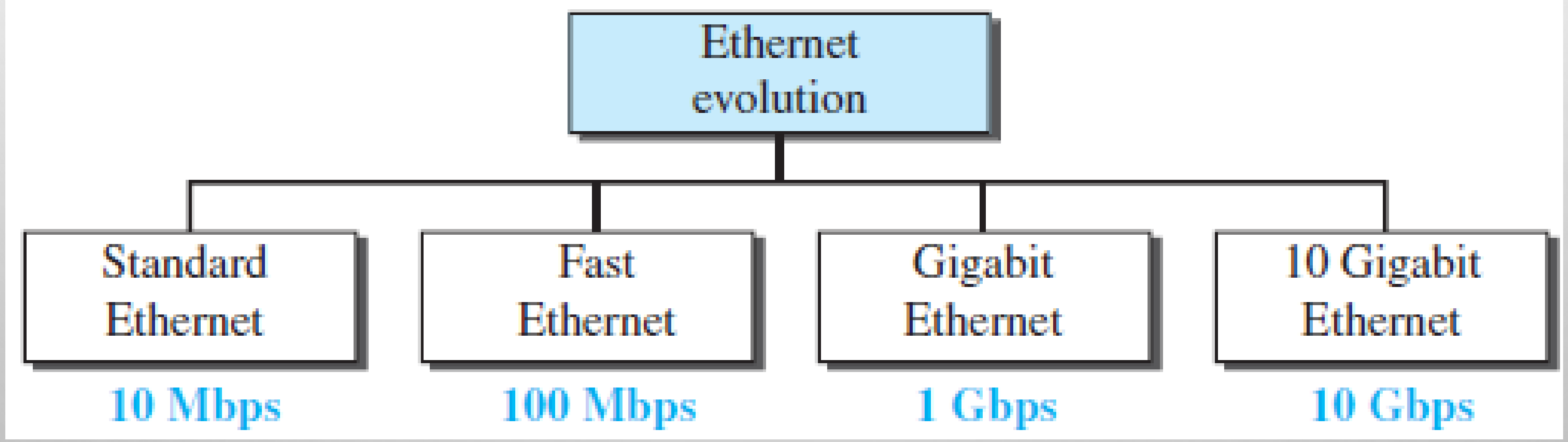


Figure : Ethernet evolution through four generations

STANDARD ETHERNET (10 MBPS)

- The original Ethernet technology with the data rate of 10Mbps is referred as the standard Ethernet.
- Although most implementations have moved to other technologies in the Ethernet evolution, there are some features of the standard Ethernet that have not changed during the evolution.
- Standard version pave the way for understanding the other three technologies.
  - Connectionless and unreliable service
  - Frame format
  - Frame length

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:

- The Ethernet frame contains seven fields, as shown in figure.

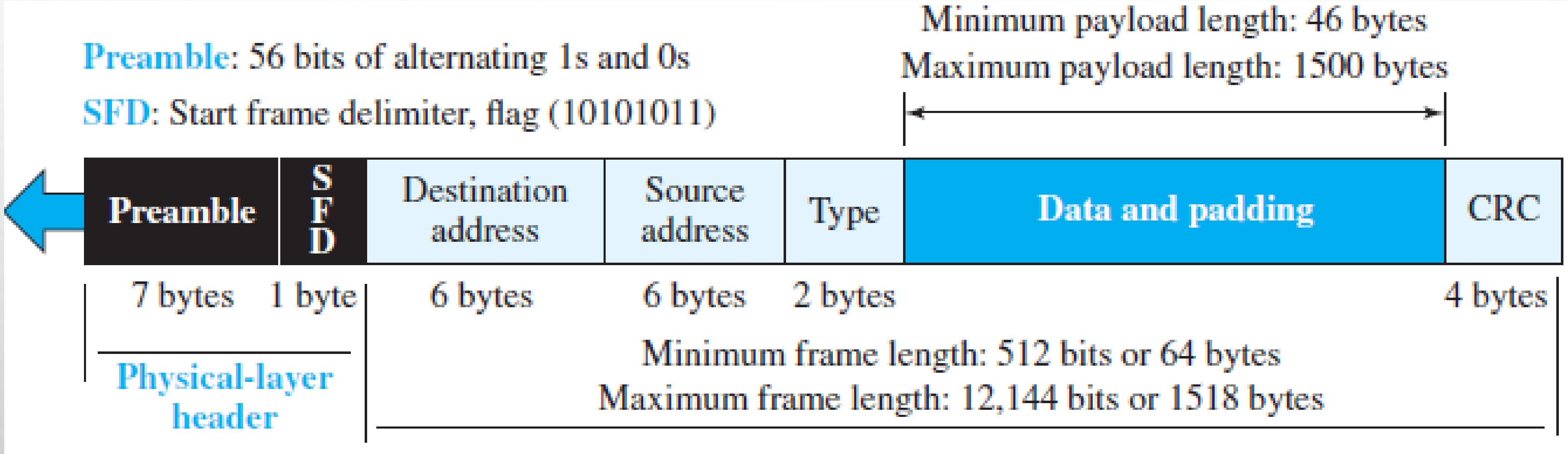


Figure : Ethernet frame

Preamble :

- Contains 7 bytes (56 bits) of alternating 0s and 1s that alert the receiving system to the coming frame and 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.
- The preamble is actually added at the physical layer and is not (formally) part of the frame.

Start frame delimiter (SFD):

- This field (1 byte: 10101011) signals the beginning of the frame.
- 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. This field is actually a flag that defines the beginning of the frame. We need to remember that an Ethernet frame is a variable-length frame. It needs a flag to define the beginning of the frame. The SFD field is also added at the physical layer.

Destination address (da) :

- This field is 6 bytes (48 bits) and contains the link-layer address of the destination station or stations to receive the packet.

Source address (SA) :

- This field is also 6 bytes and contains the link-layer address of the sender of the packet.

Type :

- This field defines the upper-layer protocol whose packet is encapsulated in the frame. This protocol can be IP, ARP, open shortest path first (OSPF), and so on.

Data :

- This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes.

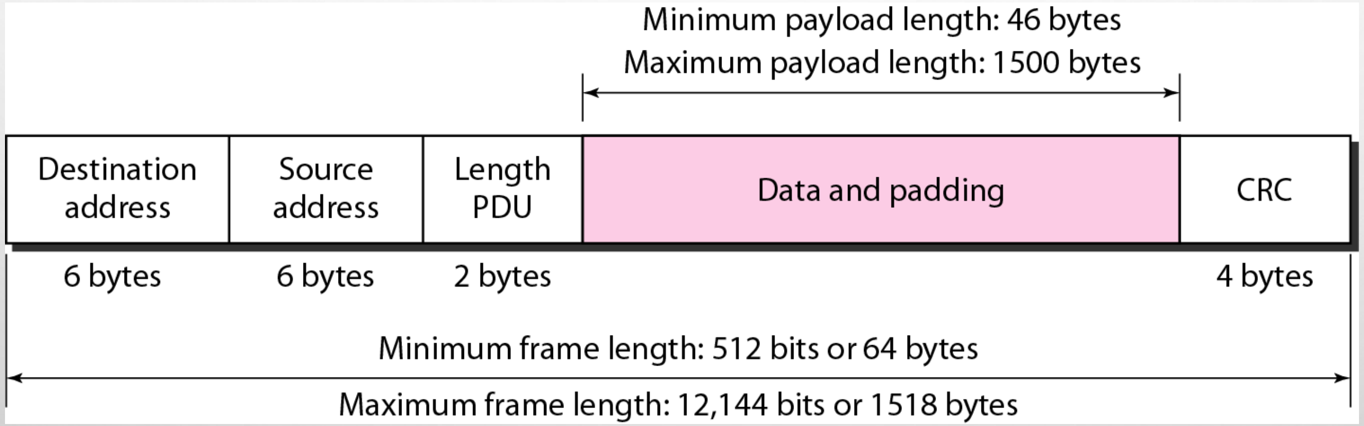
CRC :

- The last field contains error-detection information, in this case a crc32. The CRC is calculated over the addresses, types, and data field. If the receiver calculates the CRC and finds that it is not zero, then it discards the frame.

Frame length :

- Ethernet imposes 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 needs to have a minimum length of 512 bits or 64 bytes. Part of this length is the header and the trailer.
    - If we count 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.

- There are two historical reasons for the maximum length restriction.
  - 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.



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.
- For example, the following shows an Ethernet mac address:

47:20:1B:2E:08:EE

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.

Example:
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

Unicast, Multicast, and Broadcast addresses

- A source address is always a *unicast address—the frame comes from only one station.*
- The destination address can be *unicast, multicast, or broadcast.*
  - If the least significant bit of the first byte in a destination address is 0, the address is unicast; otherwise, it is multicast.
  - 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.

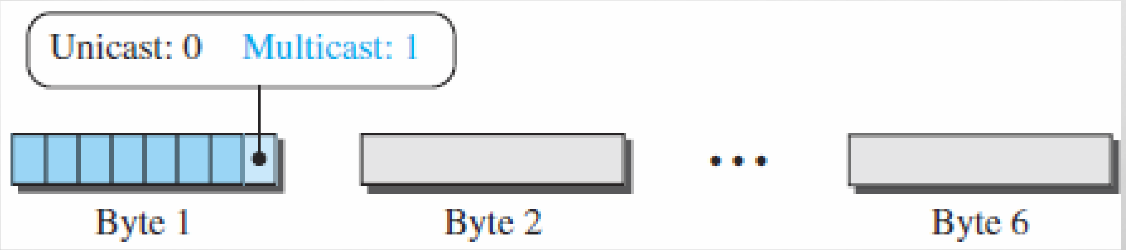


Figure : Unicast and multicast addresses

Example :

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

Solution:

Look at the second hexadecimal digit from the left to find the type of the address.

a. if it is even, the address is unicast → This is a unicast address because A in binary is 1010 (even).

b. If it is odd, the address is multicast → This is a multicast address because 7 in binary is 0111 (odd).

c. If all digits are Fs, the address is broadcast →This is a broadcast address because all digits are fs in hexadecimal

Implementation :

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

Figure : Summary of Standard Ethernet implementations

- The standard Ethernet defined several implementations, but only four of them became popular during the 1980s.
- In the nomenclature 10basex, the number defines the data rate (10 mbps).
- The term base means baseband (digital) signal, and
- X approximately defines either the maximum size of the cable in 100 m

    (for example, 5 for 500 or 2 for 185 m) or

- The type of the cable [t for unshielded twisted-pair (UTP) cable and f for fiber-optic cable].
- The standard Ethernet uses a baseband signal, which means that the bits are changed to a digital signal and directly sent on the line.

Encoding and Decoding

- All standard implementations use digital signaling(baseband) at 10 mbps.
- At the sender, data are converted to a digital signal using the line coding scheme.
- At the receiver, the received signal is interpreted as coded and decoded into data.

WIFI, IEEE 802.11 PROJECT

- IEEE has defined the specifications for a wireless LAN, called IEEE 802.11, which covers the physical and data-link layers.
- It is sometimes called **Wireless Ethernet**.
- In some countries, the public uses the term Wi-Fi (short for wireless fidelity) as a synonym for wireless LAN.

Architecture :

- The IEEE standard defines two kinds of services:
  - Basic Service Set(BSS) and
  - 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).

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

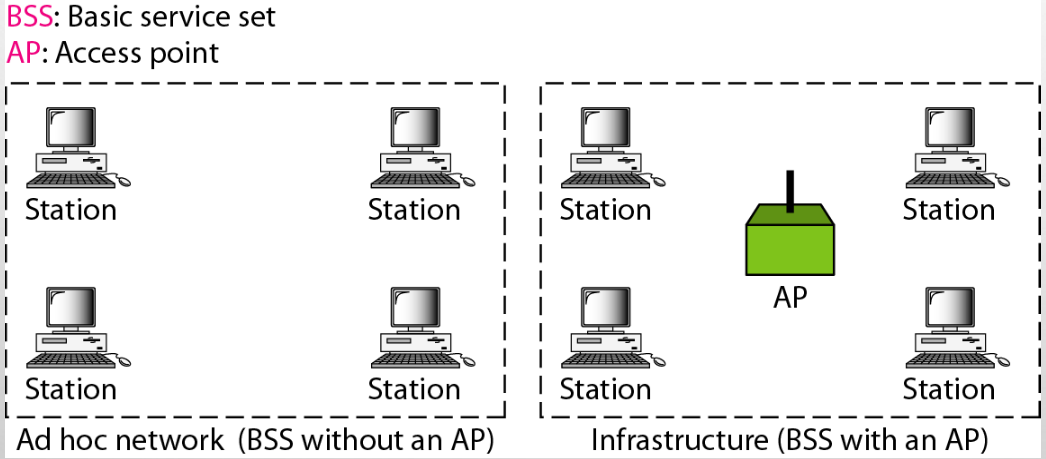
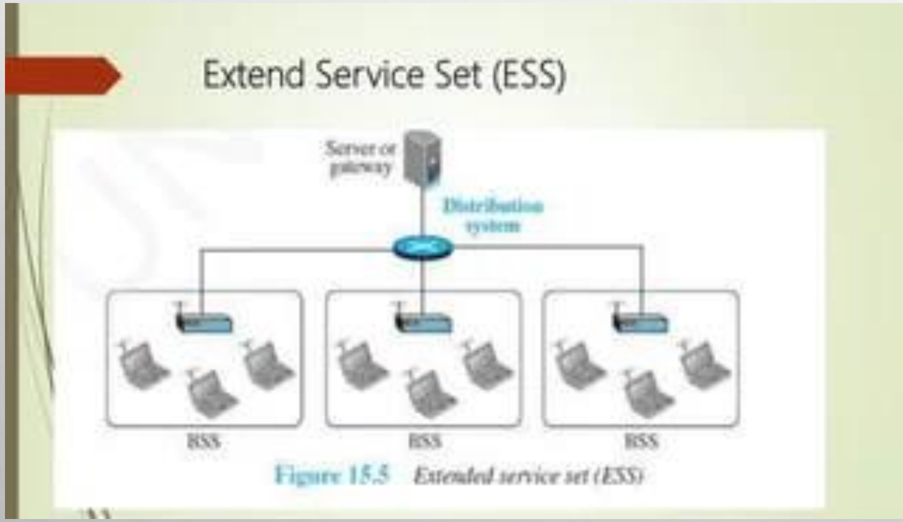


Figure : Basic service sets (BSSs)

Extended service set :

- An extended service set(ESS) is made up of two or more BSSs with APs.
- Here, 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.



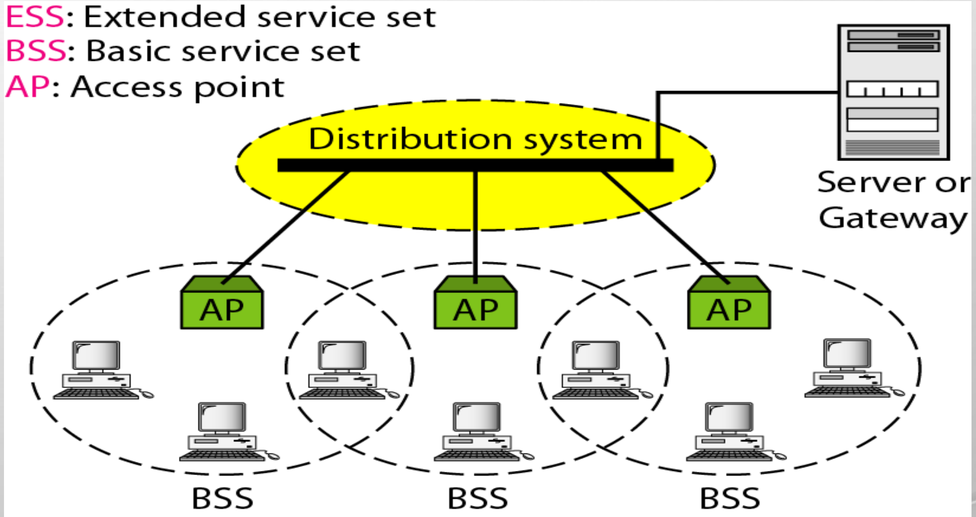


Figure : Extended service set (ESS)

- ESS uses two types of stations: mobile and stationary.
- The mobile stations are normal stations inside a BSS. The stationary stations are AP stations that are part of a wired LAN.
- When BSSs are connected, the stations within reach of one another can communicate without the use of an AP.
- However, communication between a station in a BSS and the outside BSS occurs via the AP.

Station types

IEEE 802.11 defines three types of stations based on their mobility in a wireless LAN:

- No-Transition
  - A station with no-transition mobility is either stationary (not moving) or moving only inside a BSS.
- BSS-Transition
  - A station with BSS-transition mobility can move from one BSS to another, but the movement is confined inside one ESS.
- ESS-Transition mobility
  - A station with ESS-transition mobility can move from one ESS to another.

MAC SUBLAYER

- IEEE 802.11 defines two MAC sublayers:
  1. The Distributed Coordination Function (DCF) and
  2. The Point Coordination Function (PCF).
- Figure shows the relationship among the two mac sublayers, the LLC sublayer, and the physical layer.

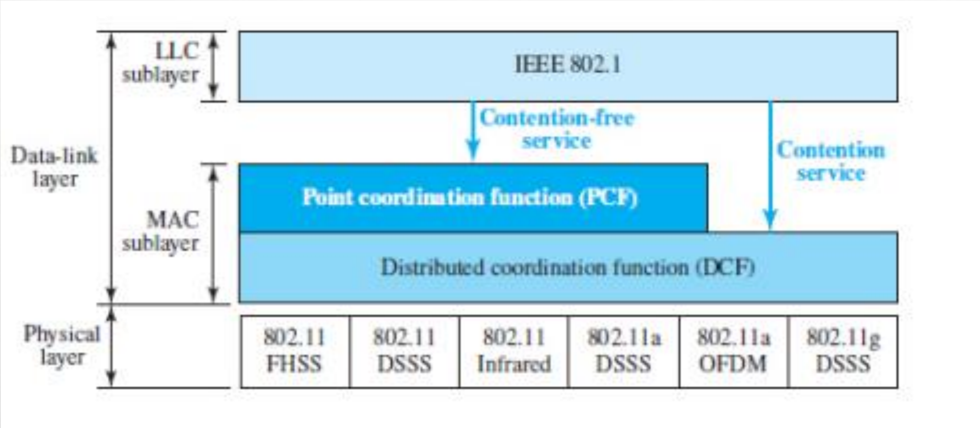


Figure : MAC layers in the IEEE 802.11 standard

Distributed coordination function

- One of the two protocols defined by the IEEE at the MAC sublayer is called the **Distributed Coordination Function (DCF).**
- DCF uses CSMA/CA as the access method.
- Frame exchange time line.

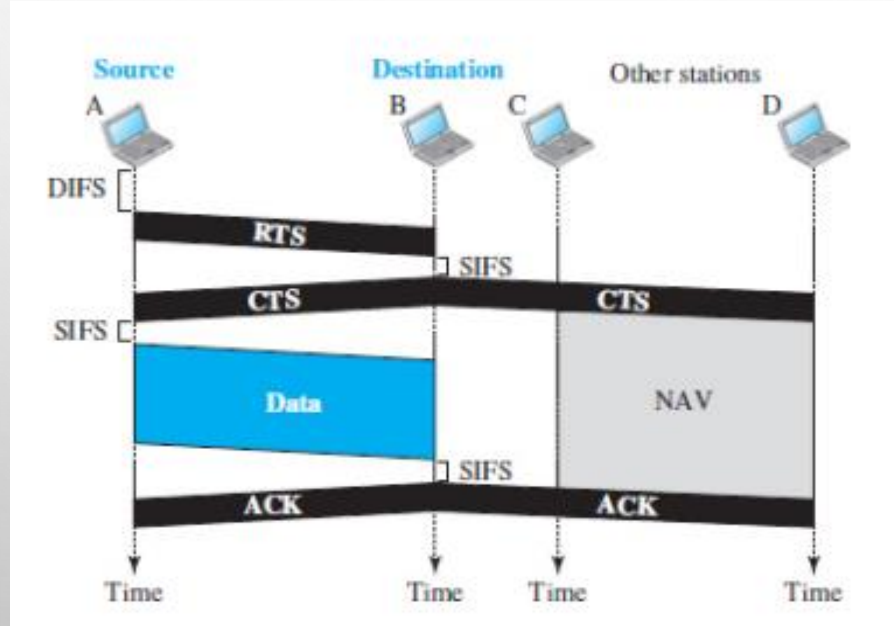


Figure : CSMA/CA and NAV

- Before sending a frame, the source station senses the medium by checking the energy level at the carrier frequency.
- The channel uses a persistence strategy with back off until the channel is idle.
- After the station is found to be idle, the station waits for a period of time called the DCF interframe space (DIFS); then the station sends a control frame called the request to send (RTS).
- After receiving the RTS and waiting a period of time called the short interframe space (SIFS), the destination station sends a control frame, called the clear to send (CTS), to the source station. This control frame indicates that the destination station is ready to receive data.
- The source station sends data after waiting an amount of time equal to the SIFS.
- The destination station, after waiting an amount of time equal to the SIFS, sends an acknowledgment to show that the frame has been received.

Network allocation vector :

- How do other stations defer sending their data if one station acquires access? In other words, how is the collision avoidance aspect of this protocol accomplished?

  The key is a feature called NAV.

- 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.
- Each time a station accesses the system and sends an rts frame, other stations start their NAV.
- In other words, each station, before sensing the physical medium to see if it is idle, first checks its NAV to see if it has expired

Collision during handshaking :

- Two or more stations may try to send RTS frames at the same time.
- These control frames may collide. However, because there is no mechanism for collision detection, the sender assumes there has been a collision if it has not received a CTS frame from the receiver.
- The back off strategy is employed, and the sender tries again.

Hidden-station problem:

- The solution to the hidden-station problem is the use of the handshake frames (RTS and CTS).
- However, because both b and c are within the range of a, the CTS message, which contains the duration of data transmission from b to a, reaches c.
- Station c knows that some hidden station is using the channel and refrains from transmitting until that duration is over.

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.

- To give priority to PCF over DCF, another **interframe space,** has been defined → **Point Coordination Function Interframe Space (PCF IFS(PIFS)).**
- PIFS (PCF IFS) is shorter than the DIFS.
- If, at the same time, a station wants to use only DCF and an AP wants to use PCF, the AP has priority.
- **Repetition interval** has been designed to cover both contention-free (PCF) and contention-based (DCF) traffic to allow DCF accessing the media.

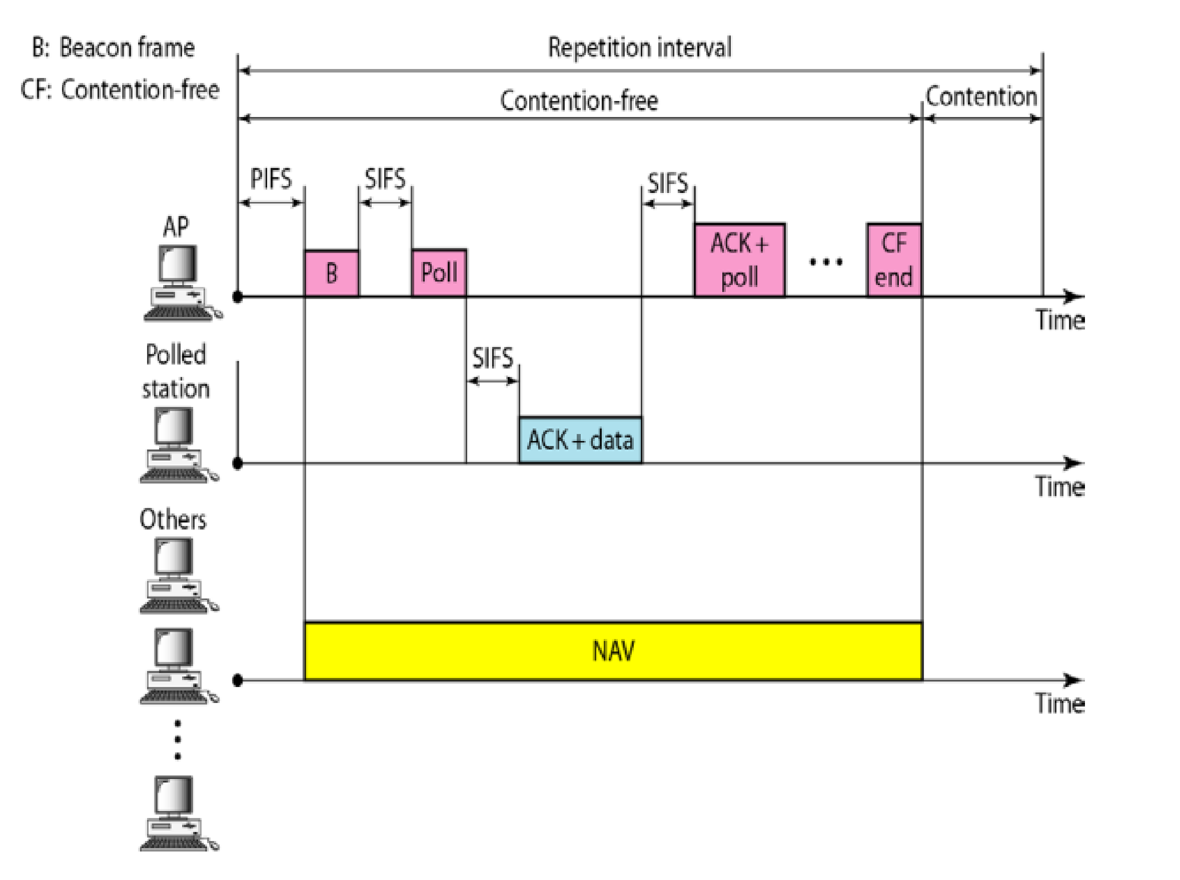


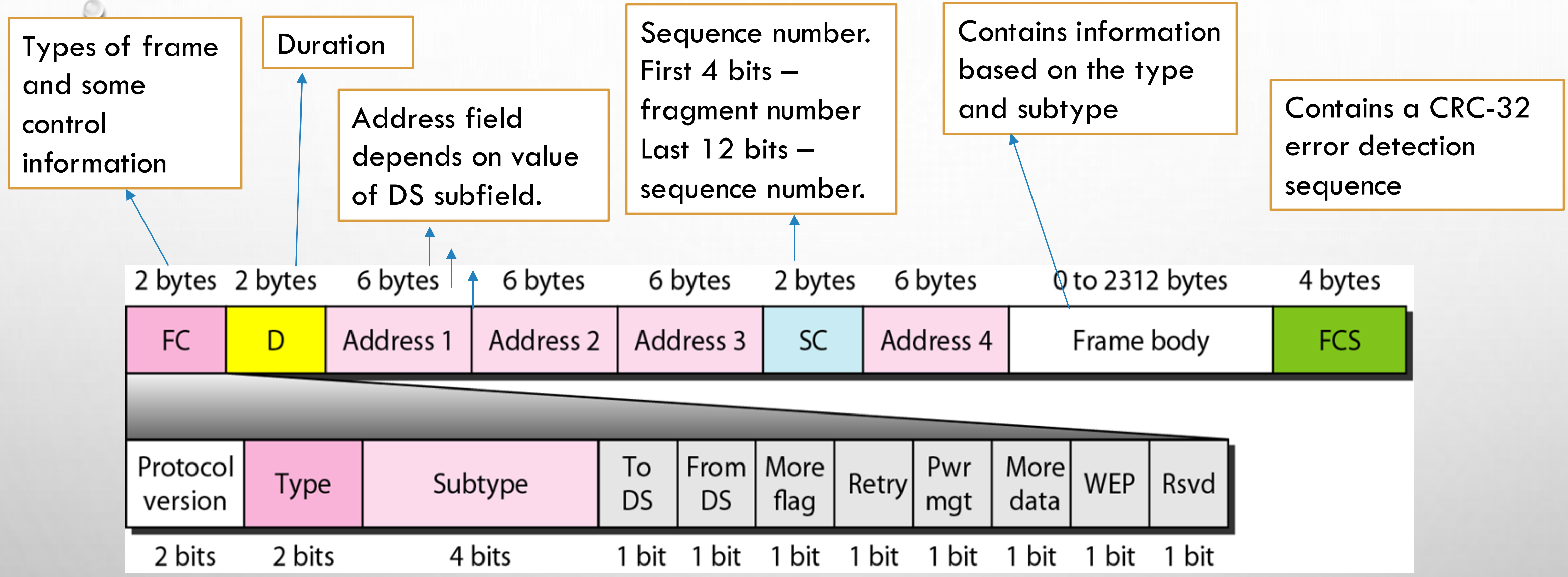
Figure: Example of repetition interval

- The RP starts with a **beacon control frame,** that broadcasts system parameters.
  - e.g., Hopping sequence, dwell-time (for FHS), clock synch, polling invitations to join the system. It also declares the length of the CF (contention-free) period to be used as shown below.
- Each station hearing the Beacon frame starts its NAV counter for the contention free duration.

FRAGMENTATION

- The wireless environment is very noisy.
- Corrupt frame has to be retransmitted.
- Fragmentation is recommended.
    - The division of a large frame into smaller ones.
- It is more efficient to resend a small frame than a large one.

Frame format: The MAC layer frame consists of nine fields.



- Subfield in FC field:

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 14.2)
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

Frame types:

- A wireless LAN defined by IEEE 802.11 has three categories of frames:
  - Management frames
    - Used for the initial communication between stations and access points.
  - Control frames
    - Used for accessing the channel and acknowledging.
  - Data frames
    - Used for carrying data and control information.

```
Frame Control Field: 0x8000
    .... ..00 = Version: 0
    .... 00.. = Type: Management frame (0)
    1000 .... = Subtype: 8
```

```
Frame Control Field: 0xb400
    .... ..00 = Version: 0
    .... 01.. = Type: Control frame (1)
    1011 .... = Subtype: 11
```

```
Frame Control Field: 0x8841
    .... ..00 = Version: 0
    .... 10.. = Type: Data frame (2)
    1000 .... = Subtype: 8
```

Figure: Type and Subtype

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

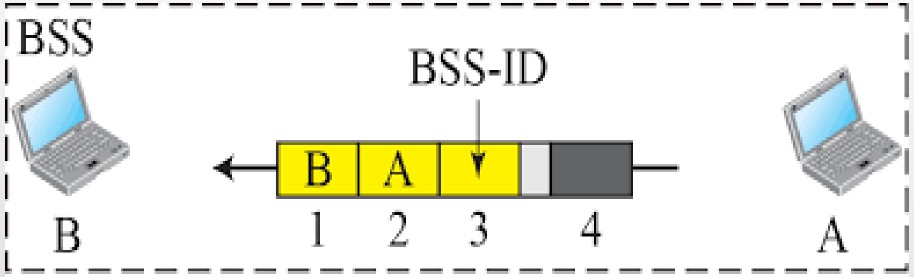
Figure: values of subfields in control frames.

Addressing mechanism:

- IEEE 802.11 specifies four addressing mechanism cases, defined by the value of the two flags in the FC field → to DS and from DS.
- Each flag can be either 0 or 1, resulting in four different situations.
- The interpretation of four addresses in the MAC frame depends on the value of these flag.

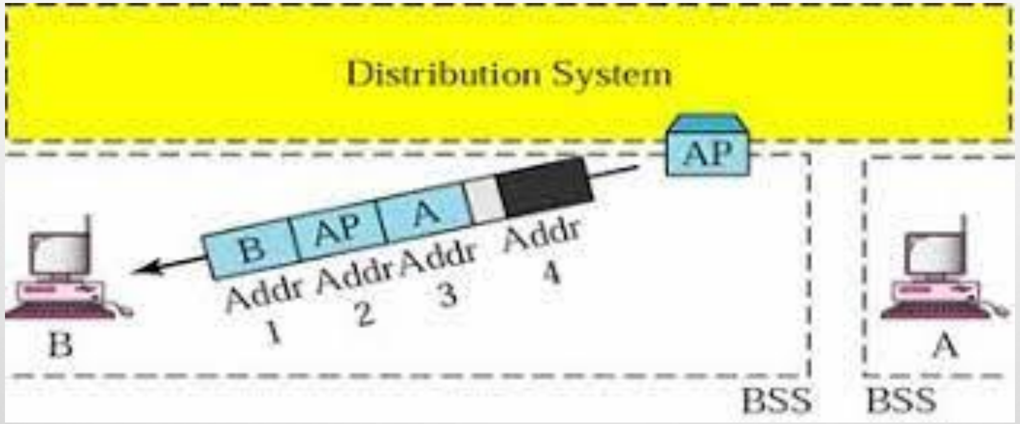
To DS	*From DS*	*Address 1*	*Address 2*	*Address 3*	*Address 4*
0	0	Destination	Source	BSS ID	N/A
0	1	Destination	Sending AP	Source	N/A
1	0	Receiving AP	Source	Destination	N/A
1	1	Receiving AP	Sending AP	Destination	Source

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

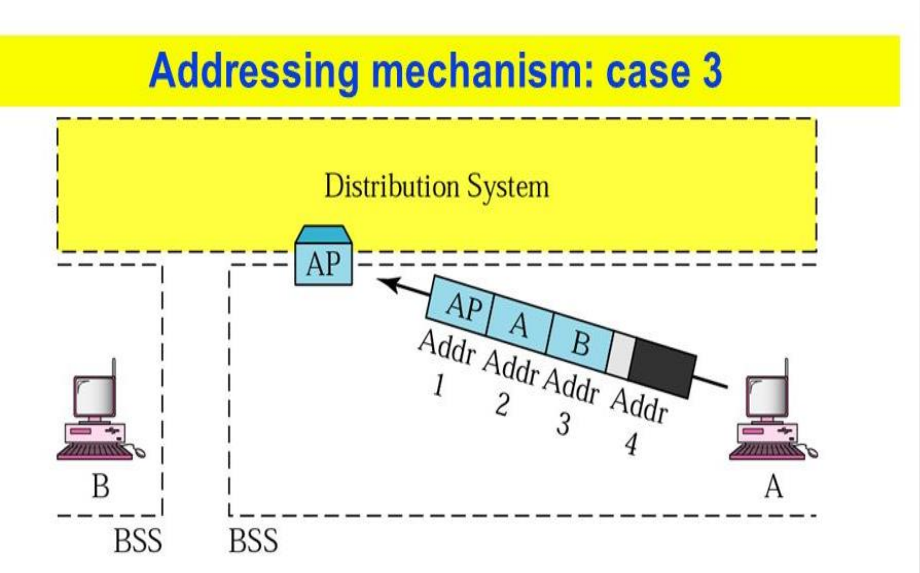


a. Case 1

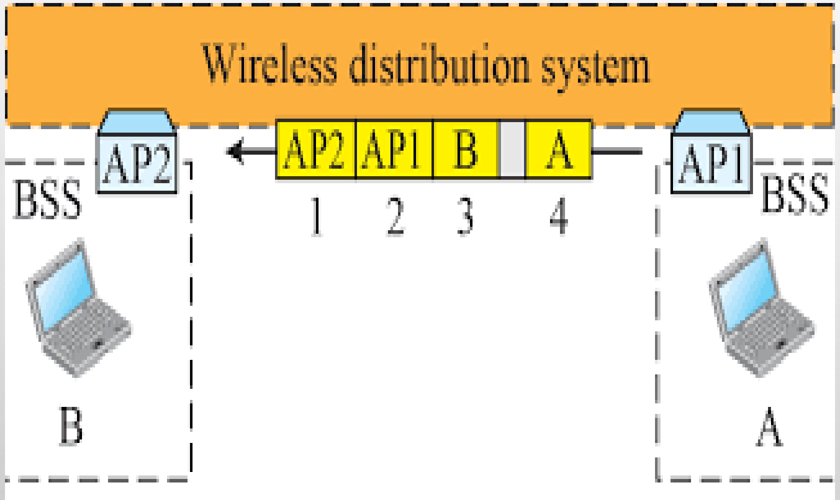
- Case 2 - 01.
  - In this case, to DS = 0 and from DS = 1. This means that the frame is coming from a distributed system.
    - The frame is coming from an AP and going to a station.
  - Address 3 contains the original sender of the frame.



- Case 3 - 10.
  - In this case, to DS = 1 and from DS = 0. This means that the frame is going to a distributed system.
    - The frame is going from a station to an AP.
    - ACK is sent to the original station.
  - Address 3 contains the final destination of the frame in the distributed system.



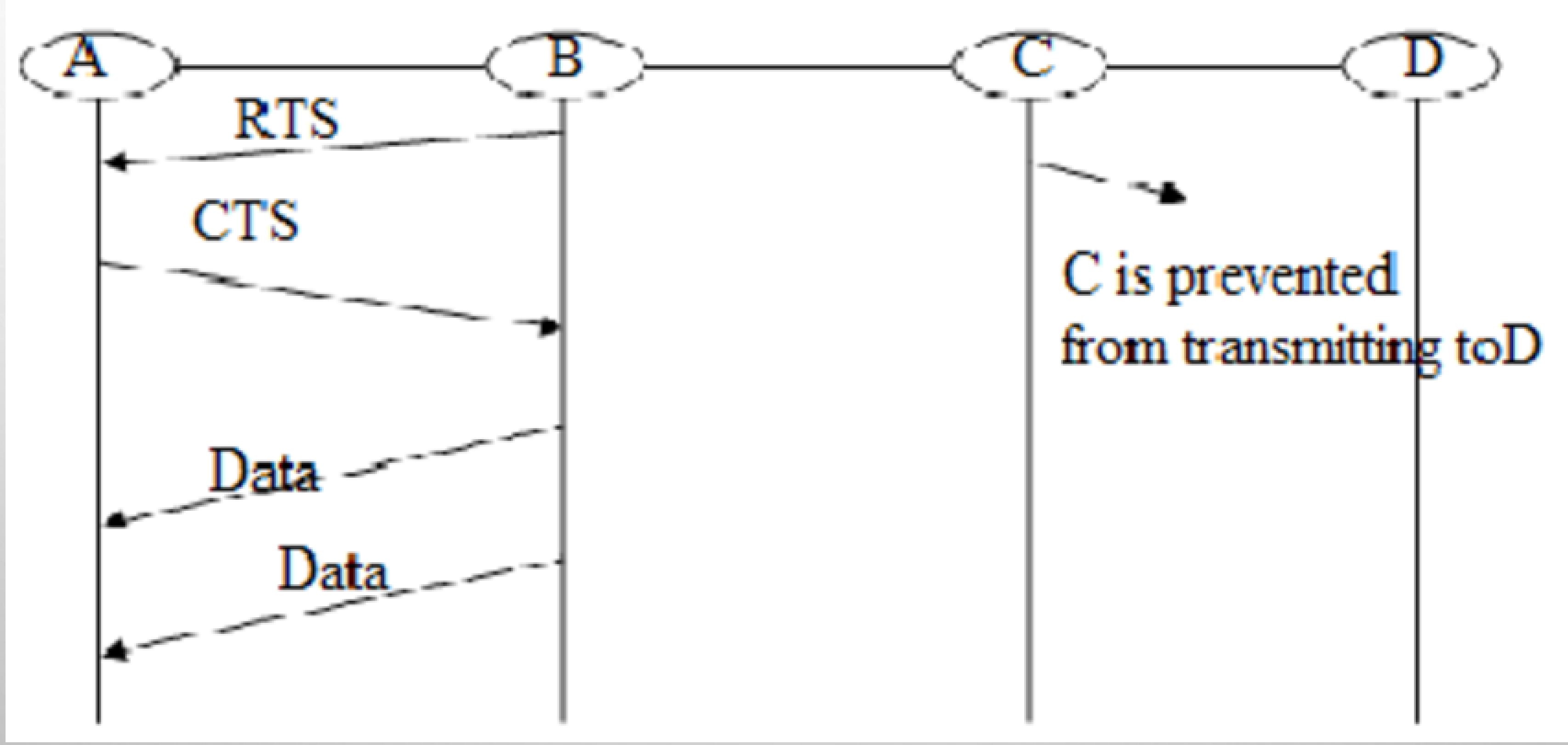
- Case 4 - 11.
  - In this case, to DS = 1 and from DS = 1. This is the case in which the distributed system is also wireless.
    - The frame is going from one AP to another AP in a wireless distributed system.
    - ACK is sent to the original station.
  - Here, we need four addresses to define the original sender, the final destination, and two intermediate APs.



d. Case 4

Exposed-station problem:

- Similar to hidden station problem.

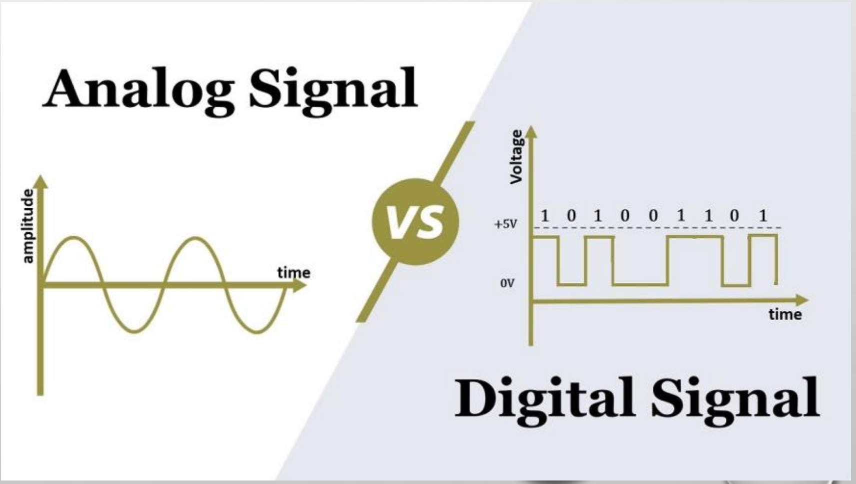


- In this problem, a station refrains from using a channel when it is available.
- Example: Station A is transmitting to station B. station C has some data to send to station D, which can be sent without interfering with the transmission from A to B. as it hears RTS from station A, it refrain from sending, even though the communication between C and D cannot cause a collision in the zone between A and C.
- C cannot know that station A’s transmission does not affect the zone between C and D.
- C is too conservative and wastes the capacity of the channel.

PHYSICAL LAYER

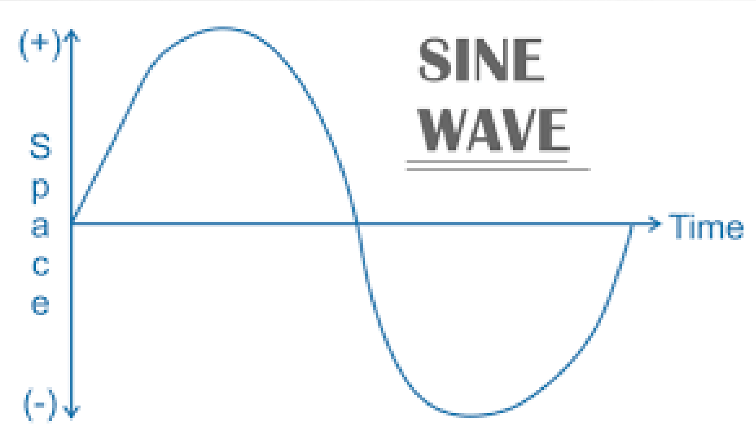
SIGNALS

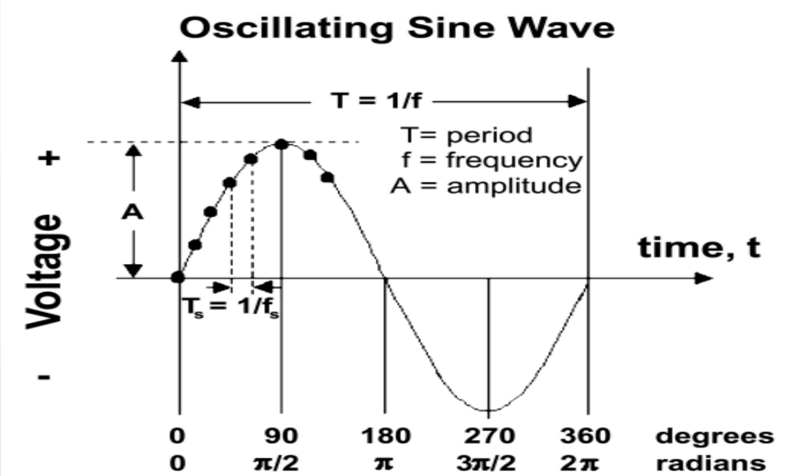
- What exchanged between two entities? → Data (information).
- What goes through the network connecting two entities at the physical layer? → Signals.
- Example:
  - When Alice sends a message to bob, the message is changed to electrical signals; when bob receives the message, the signals are changed back to the message.
- The signal can be analog or digital.



Analog signals

- Analog Signal can take one of two forms: Periodic and aperiodic (nonperiodic).
- Periodic analog signal can be classified as simple or composite.
- The sine wave is the most fundamental form of a periodic analog signal.



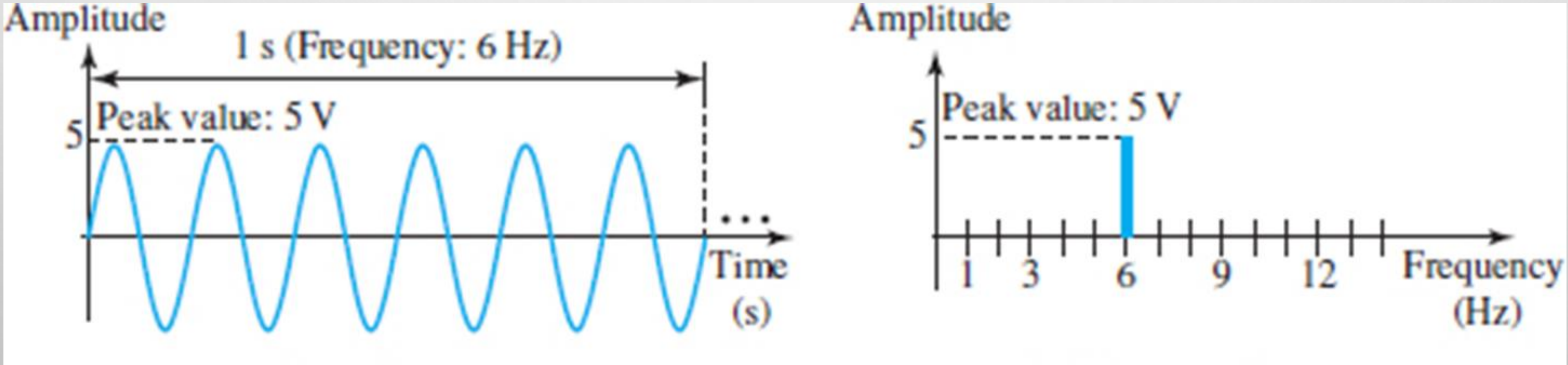


- A sine wave is a simple oscillating curve, its change over a course of a cycle is smooth and consistent.
- A sine wave can be represented by three parameters: **period, peak amplitude and phase.**
- Peak amplitude → absolute value of its highest intensity. Measured in volts.
- Period → amount of time, in seconds, a signal needs to complete in one cycle.
- Frequency → number of period in 1s.
- Frequency is inverse of period. → $f = 1/T$.
- Phase → position of the waveform relative to time 0. measured in degrees or radians.

- Wavelength → the distance a simple signal can travel in one period.

$$\text{Freq} = \frac{c}{\lambda} \qquad \lambda = \frac{c}{\text{Freq}}$$

- Time and frequency Domains → Sine wave shown by using what is called a **time domain plot, which shows changes in signal amplitude with respect to time.**
- To show the relationship between amplitude and frequency, we can use a **frequency-domain plot**



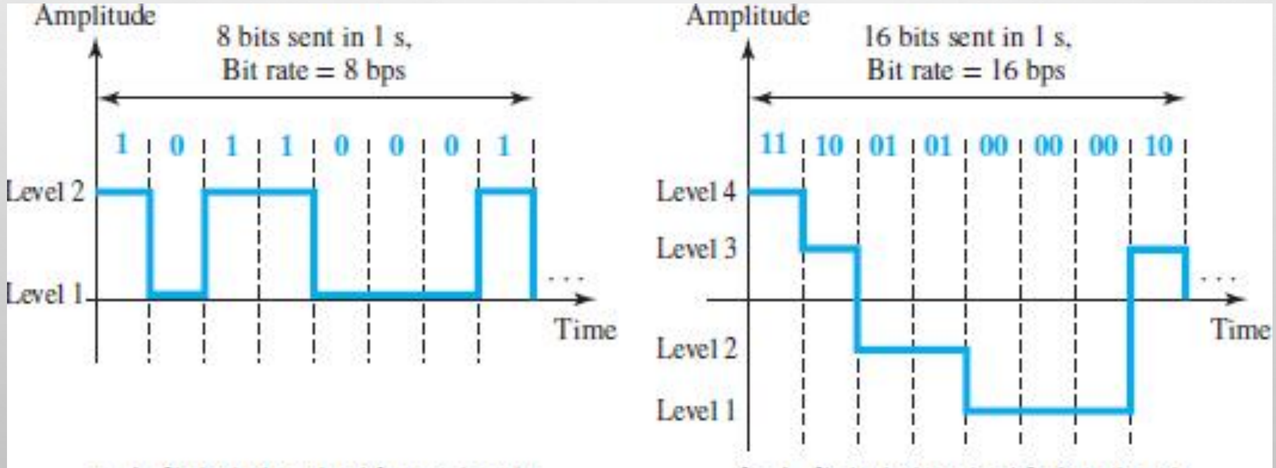
a. Time domain

b. Frequency domain

- Composite signals → A composite signal is made up of many simple sine waves. We need to send a composite signal to communicate data.
- Bandwidth → The range of frequencies contained in a composite signal is its bandwidth.
  - The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

Digital signals

- Information can also be represented by a digital signal.
- Example: a value 1 can be encoded as a positive voltage and a value 0 as a zero voltage.
- A digital signal can have more than two levels.
- In general, if the signal has L levels, each level needs $\log_2 L$ bits.



a. A digital signal with two levels

b. A digital signal with four levels

Bit Rate:

- Bit rate is used to describe the digital signal.
- Bit rate is the number of bits sent in 1s, expressed in bits per second(bps).
- Bit rate can be represented in kbps or Mbps.

Example:

- Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?
  - 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 X 24 X 80 X 8 = 1,536,000bps = 1.536Mbps.

Bit length:

- The bit length is the distance 1 bit occupies on the transmission medium.

  Bit length = 1/(bit rate)

Example:

- The length of the bit for previous example is

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

Transmission of Digital Signals:

- We can transmit a digital signal by using one of two different approaches : **Baseband transmission or Broadband transmission.**
- 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.

Signal Impairment

- Signal travel through transmission media, which are not perfect.
- The imperfection causes signal impairment.
- Three causes of impairment are **attenuation, distortion and noise.**

Attenuation and Amplification:

- Attenuation means a loss of energy.
- When a simple or composite signal 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.

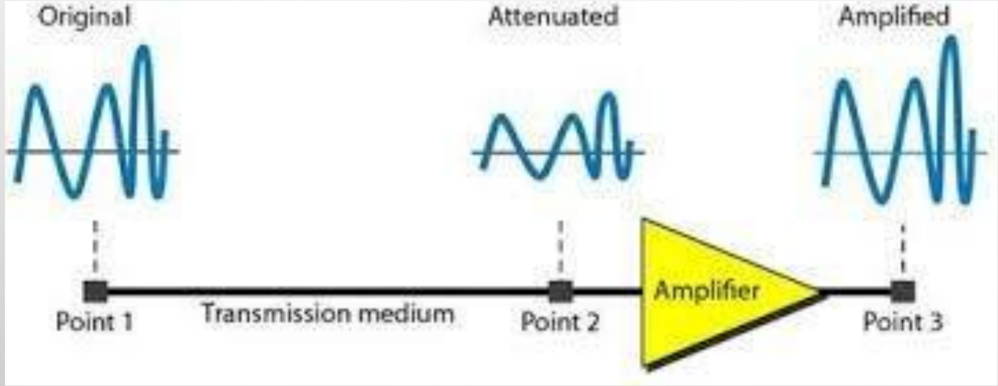


Figure: Attenuation and amplification

- The lost or gained strength of a signal is measured by a unit called **decibel(dB).**
- The decibel is negative if a signal is attenuated and positive if a signal is amplified.
- Variable P1 and P2 are the powers of a signal at points 1 and 2, respectively.

$$dB = 10 \log_{10}(P2/P1)$$

- Example:
  - 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:

$$\mathbf{10 \log_{10}(P2/P1) = 10 \log_{10}((0.5P1)/P1) = 10 \log_{10} 0.5 = 10 \times (-0.3) = -3dB.}$$

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

Distortion:

- Distortion means that the signal changes its form or shape.
- Distortion can occur in a composite signal made up of different frequencies.

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 that comes from power lines, lightning and so on.

Signal to Noise ratio(SNR):

- To find the theoretical bit-rate limit, we need to know the ratio of the signal power to the noise power.

$$SNR = (\text{average signal power}) / (\text{average noise power})$$

- Because, SNR is the ratio of two powers, it is often described in decibel units

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

Data Rate Limits:

- It 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)

Noiseless channel: Nyquist Bit Rate:

- For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate.

$$\text{BitRate} = 2 \times B \times \log_2 L$$

B → Bandwidth of the channel

L → Number of signal levels used to represent data

Bitrate → number of bits per second.

- Practically, there is a limit. When we increase the number of signal levels, we impose a burden on the receiver.
- 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.

Example:

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

we can use the Nyquist formula as shown:

$$265{,}000 = 2 \text{ X } 20{,}000 \text{ X } \log_2 L \rightarrow \log_2 L = 6.625 \rightarrow L = 2^{6.625} = 98.7 \text{ levels.}$$

Noisy Channel: Shannon Capacity:

- In reality, we cannot have a noiseless channel; the channel is always noisy.
- Shannon introduced a formula, called the Shannon Capacity, to determine the theoretical highest data rate for a noisy channel:

$$C = B \times \log_2(1+SNR)$$

- In the Shannon formula there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel.

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

$$C = B \log_2(1+SNR) = B \log_2(1+0) = B \log_2 1 = B \times 0 = 0$$

This means that the capacity of this channel is zero, the data are so corrupted in this channel that they are useless when received.

Example 2:

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

$$C = B \log_2(1+SNR) = 3000 \log_2(1+3162) = 34{,}881 \text{bps}$$

This means that the highest bit rate for a telephone line is 34.881kbps. 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.

Using both limits:

We need to use both methods to find the limits and signal levels.

Example:

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

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

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

Then, we use Nyquist formula to find the number of signal levels.

$$4Mbps = 2 \times 1Mbps \times \log_2 L \rightarrow \log_2 L = 2 \rightarrow L = 4 \rightarrow \text{signal levels}$$

Performance:

- One characteristics that measures network performance is **Bandwidth.**
- The term can be used in two different contexts with two different measuring values: Bandwidth in hertz and bandwidth in bits per second.
- Bandwidth in hertz is the range of frequencies involved.
- Bandwidth in bits per second can be defined as the number of bits a channel can pass.

Throughput:

- Measure of how fast we can actually send data through a network.

Latency(Delay):

- It 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.
- There are four types of delay:
  - Propogation delay
  - Transmission delay
  - Queuing delay
  - Processing delay

The latency or total delay is

Latency = propogation delay + transmission delay + queuing delay + processing delay

Bandwidth-Delay product

Let us elaborate this issue, using two hypothetical cases as examples:

1. Case 1: let us assume that we have a link with a bandwidth of 1bps. We also assume that the delay of the link is 5s. We can see the bandwidth-delay product (1 X 5) is the maximum number of bits that can fill the link. there can be no more than 5 bits at any one time on the link.
2. Case 2: Now assume we have a bandwidth of 5bps with a delay of 5s. We can see that there can be a maximum of 5 X 5 = 25 bits on the line. The reason is that, at each second, there are 5 bits on the line.

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

Jitter:

- Another performance issue that is related to delay is jitter.
- Jitter is a problem if different packets of data encounters different delays.
- The application using the data at the receiver site is time-sensitive.

Digital transmission

- A computer network is designed to send information from one point to another.
- Information needs to be converted to either a digital signal or an analog signal for transmission.
- In digital transmission, if data are digital, we need to use digital-to-digital conversion techniques.
- If data are analog, we need to use analog-to-digital conversion.

Digital-to-digital Conversion:

- If we use digital transmission and our data are already digital, we need digital-to-digital conversion.
- The conversion involves three techniques:
  - Line coding
  - Block coding
  - Scrambling

Line Coding:

- It is the process of converting digital to digital signals.
- Data → text, numbers, graphical images, audio or video. → stored in sequence of bits.

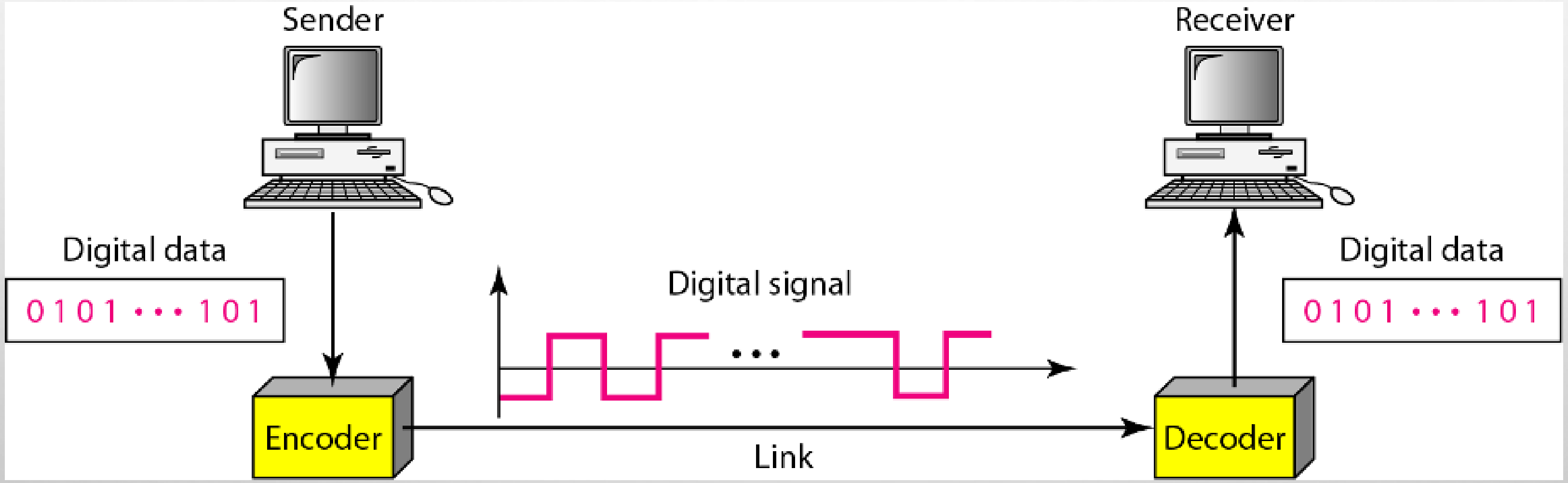


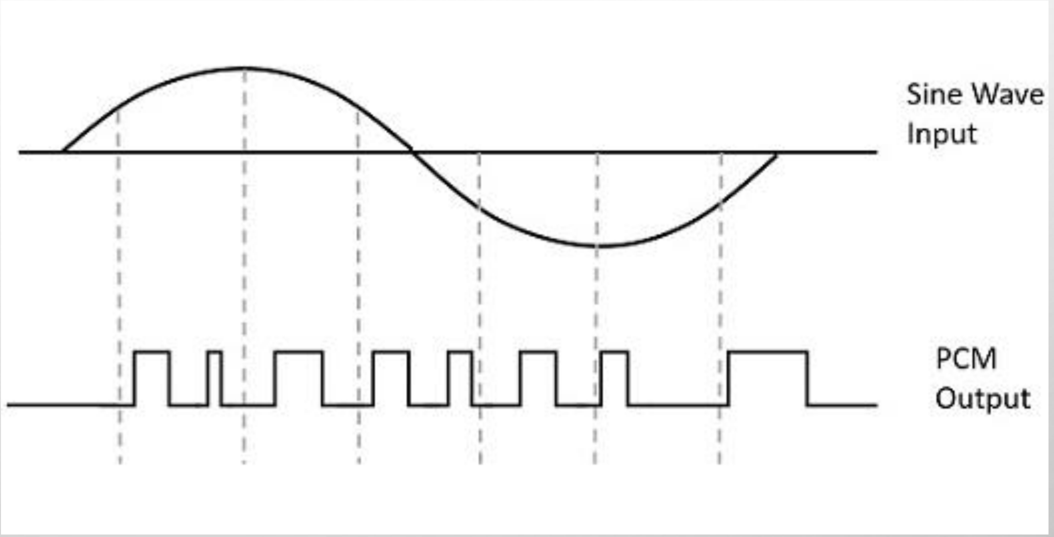
Figure: Line coding and decoding.

Block coding:

- Block coding give us redundancy and improve performance of line coding.
- Block coding changes a block of m bits into a block of n bits.
- Block coding normally involves three steps: division, substitution and combination.

Analog-to-digital Conversion:

- The tendency today is to change an analog signal to digital data because the digital signal is less susceptible to noise.
- Here, we describe two techniques:
  - Pulse code modulation
  - Delta modulation



Pulse Code Modulation(PCM)

- The most common technique used to change an analog signal to digital data is called **pulse code modulation(PCM).**

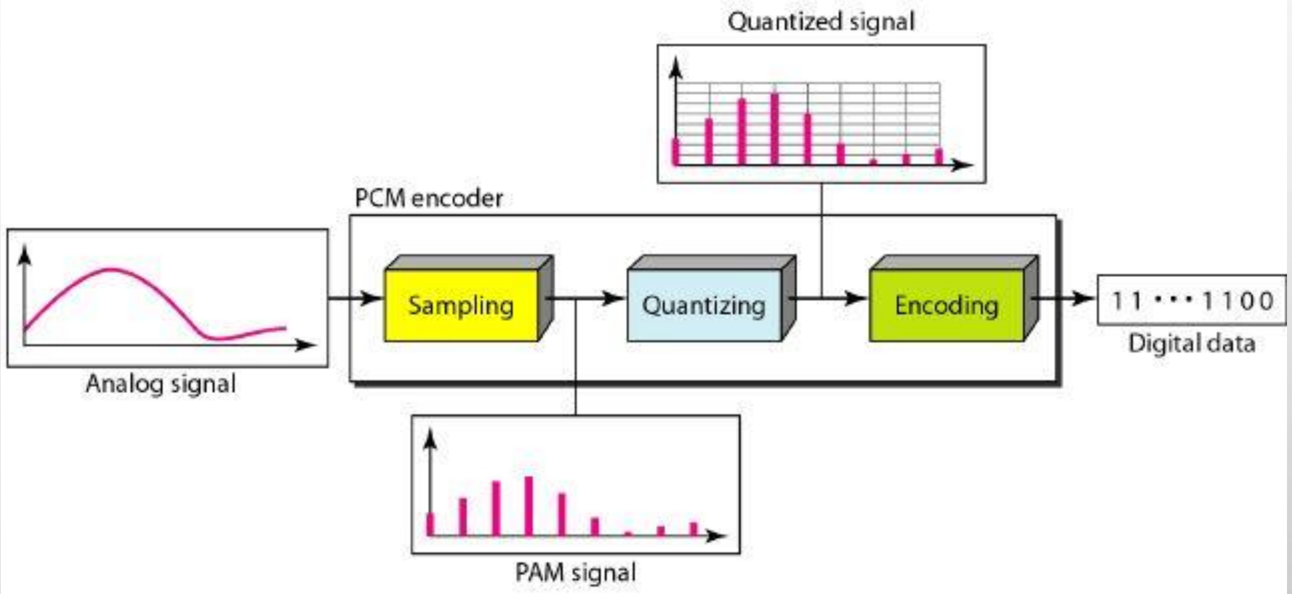


Figure: Components of a PCM encoder

- A PCM encoder has three processes

1. The analog signal is sampled every T s.
2. The sampled signal is quantized, which means every sample is considered as a pulse.
3. The quantized values(pulses) are encoded as streams of bits.

Example:

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

The human voice normally contains frequencies from 0 to 4000Hz. So the sampling rate and bit rate are calculated as

Sampling rate = 4000 X 2 = 8000 samples

Bit rate = 8000 X 8 = 64,000bps = 64 kbps

PCM Bandwidth:

- It can be proven that the minimum bandwidth of the digital signal is

$$B_{min} = n_b \times B_{analog}$$

Delta Modulation(DM)

- PCM is a very complex technique. To reduce its complexity we use Delta Modulation Technique.
- Delta Modulation is the simplest technique.
- PCM finds the value of the signal amplitude for each sample, DM finds the change from the previous sample.

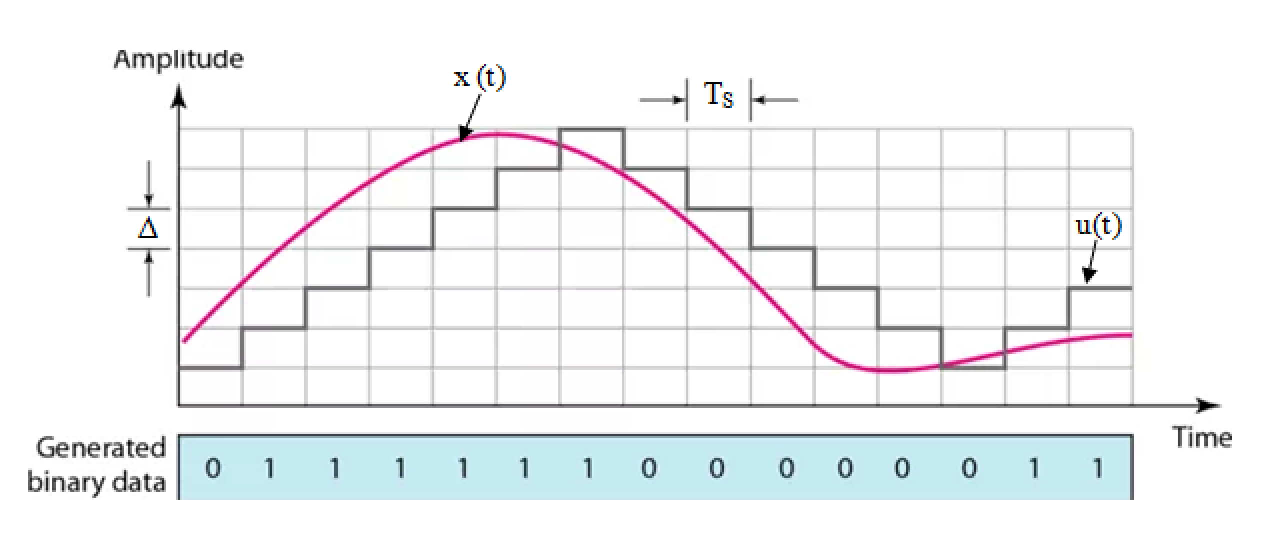
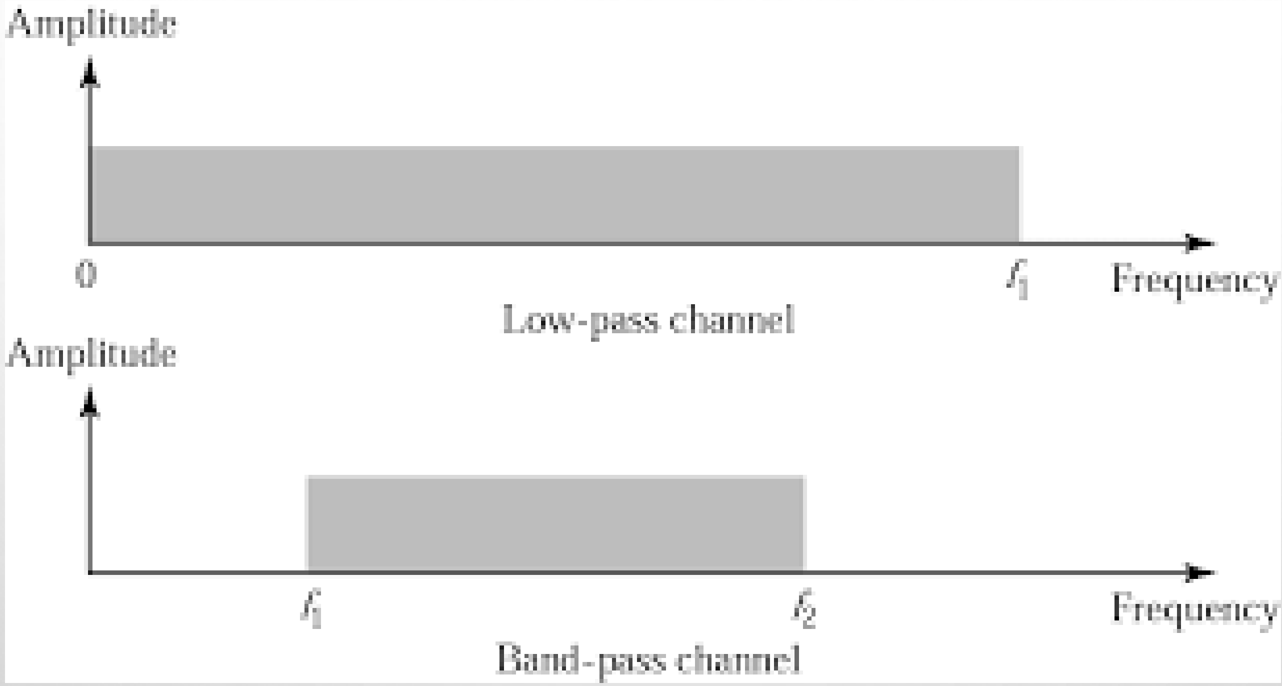


Figure: The process of delta modulation

Analog transmission

- Digital transmission needs low pass channel(a channel that starts from 0).
- Analog transmission is the only choice if we have a bandpass channel(a channel that does not start from zero).
- Converting digital data to a bandpass analog signal is called digital to analog conversion.
- Converting a low pass analog signal to a bandpass analog signal is called analog to analog conversion.


Amplitude
0
f_1
Frequency
Low-pass channel
Amplitude
f_1
f_2
Frequency
Band-pass channel

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.
- Any of three characteristics of sine wave can be altered, gives us three mechanisms for modulating digital data into an analog signal.
  - Amplitude Shift keying(ASK).
  - Frequency Shift Keying(FSK).
  - Phase Shift Keying(PSK).
  - Quadrature amplitude modulation(QAM).

Amplitude Shift Keying

- In amplitude shift keying, the amplitude of the carrier signal is varied to create signal element.
- Both frequency and phase remain constant.

Binary ASK(BASK):

- ASK is normally implemented using only two levels, referred to as binary amplitude shift keying or on-off keying(OOK).

Multilevel ASK:

- In multilevel ASK there are more than two levels.

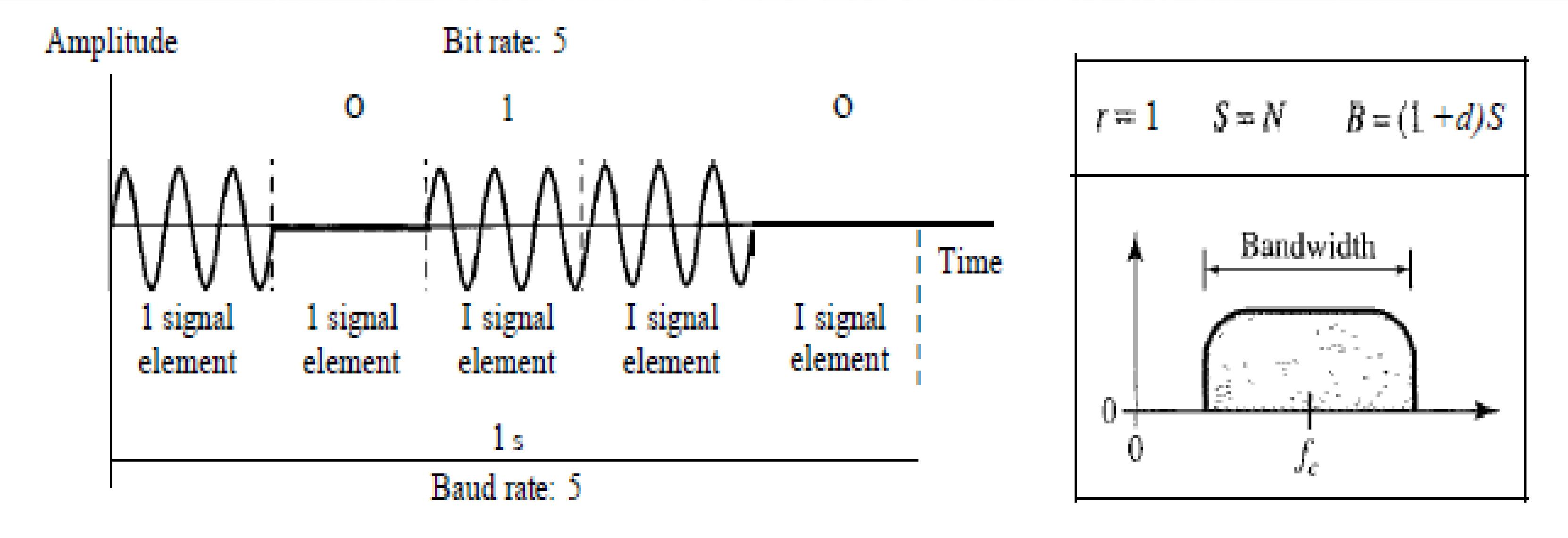


Figure: Binary amplitude shift keying

Frequency Shift Keying(FSK)

- In frequency shift keying, the frequency of the carrier signal is varied to represented 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.
- Peak amplitude and phase remain constant for all signal element.
- Binary FSK(BFSK) → has two carrier frequency.

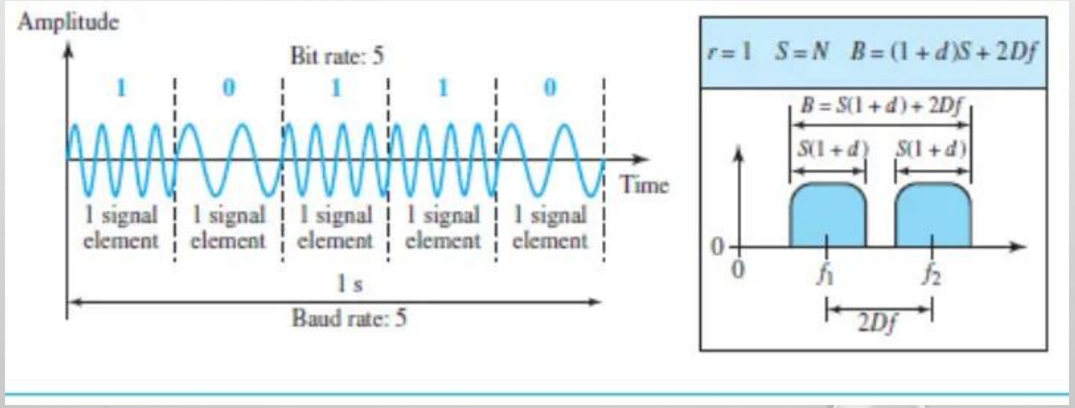


Figure: Binary frequency shift keying

Phase Shift Keying(PSK)

- In phase shift keying, the phase of the carrier signal is varied to represented two or more signal elements.
- Peak amplitude and frequency remain constant for all signal element.
- PSK is more common than ASK or FSK.
- Binary PSK → the simplest PSK is the Binary PSK in which we have only two signal elements, one with a phase of 0^0 , and the other with a phase of 180^0 .
- Binary PSK is as simple as binary ASK with one big advantage – it is less susceptible to noise.

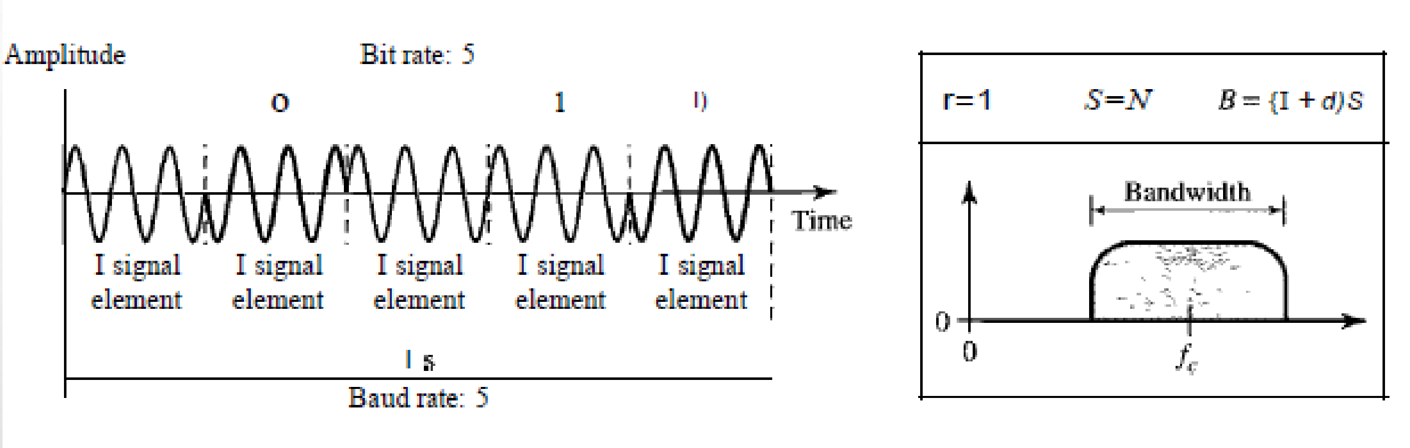


Figure: Binary phase shift keying

Analog to analog conversion:

- Analog to analog conversion or analog modulation, 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.
- Example: Radio.
- Analog to analog conversion can be accomplished in three ways: amplitude modulation(AM), frequency modulation(FM) and Phase modulation(PM).

Amplitude Modulation:

- In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal.

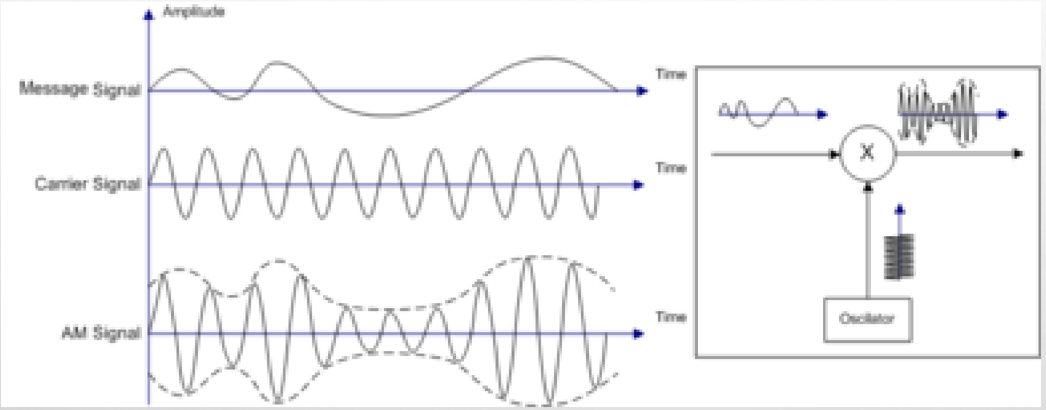


Figure: Amplitude Modulation

Frequency Modulation:

- In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level(amplitude) of the modulating signal.
- The Peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.

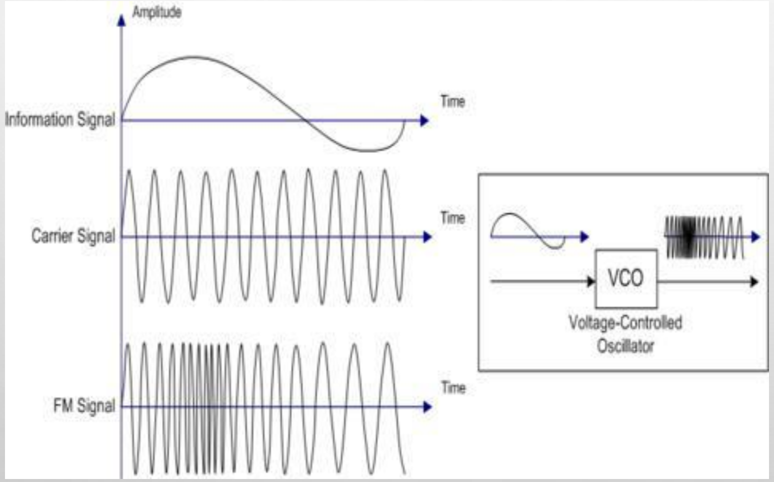


Figure: Frequency Modulation

Phase Modulation:

- In PM transmission, the frequency phase of the carrier signal is modulated to follow the changing voltage level(amplitude) of the modulating signal.
- The Peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly.
- It can be proven mathematically that PM is the same as FM with one difference.
    - In FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal.
    - In PM, the instantaneous change in the carrier frequency is proportional to the derivation of the amplitude of the modulating signal.

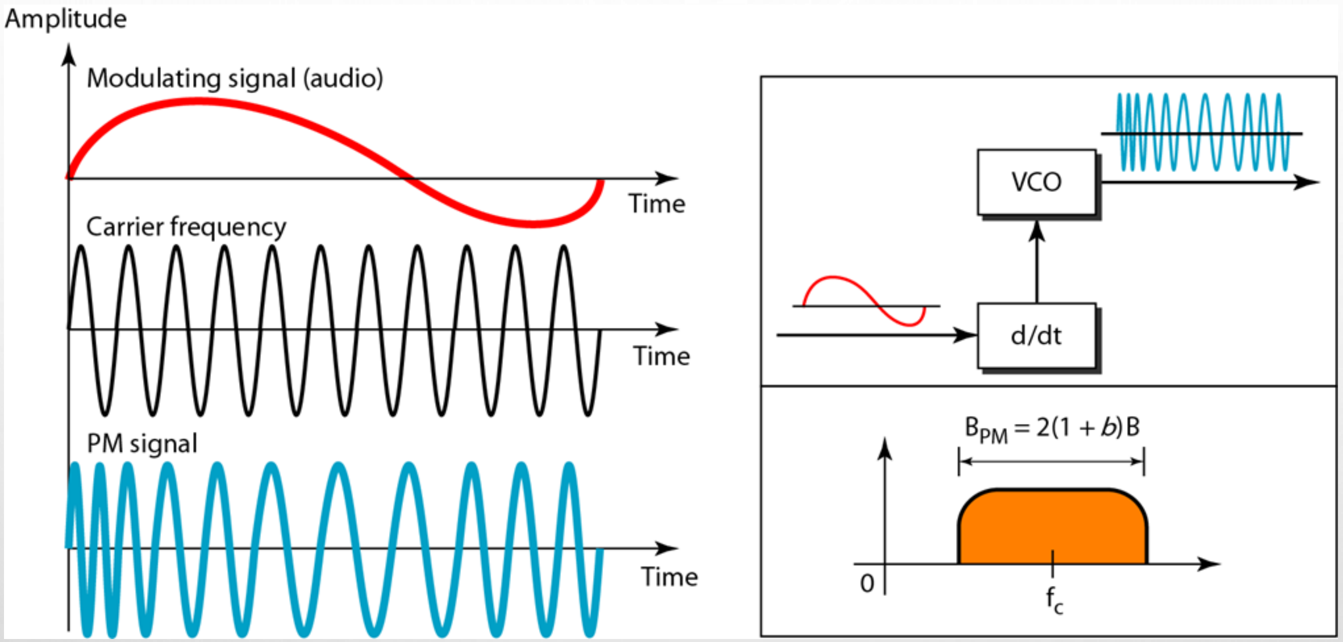


Figure: Phase Modulation

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.

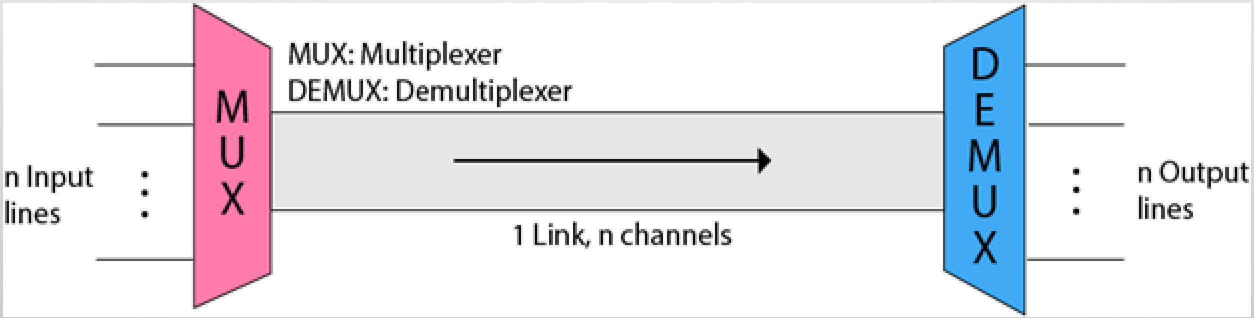


Figure: Dividing a link into channels.

- Multiplexer combines into a single stream(Many to one).
- Demultiplexer separates the stream back into its component transmissions(one to many) and directs them to their corresponding lines.
- Link refers to physical path.
- Channel refers to the portion of a link that carries a transmission between a given pair of lines
- One link can have many channels.
- There are three basic multiplexing techniques:
  - Frequency-division multiplexing(FDM)
  - Wavelength-division multiplexing(WDM) and
  - Time-division multiplexing(TDM).

Frequency – Division Multiplexing:

- Can be applied when the bandwidth of a link in (hertz) is greater than the combined bandwidths of the signal to be transmitted
- Each signal is modulated to a different carrier frequency.
- Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal.
- Channels can be separated by strips of unused bandwidth - **guard bands** -to prevent signals from overlapping

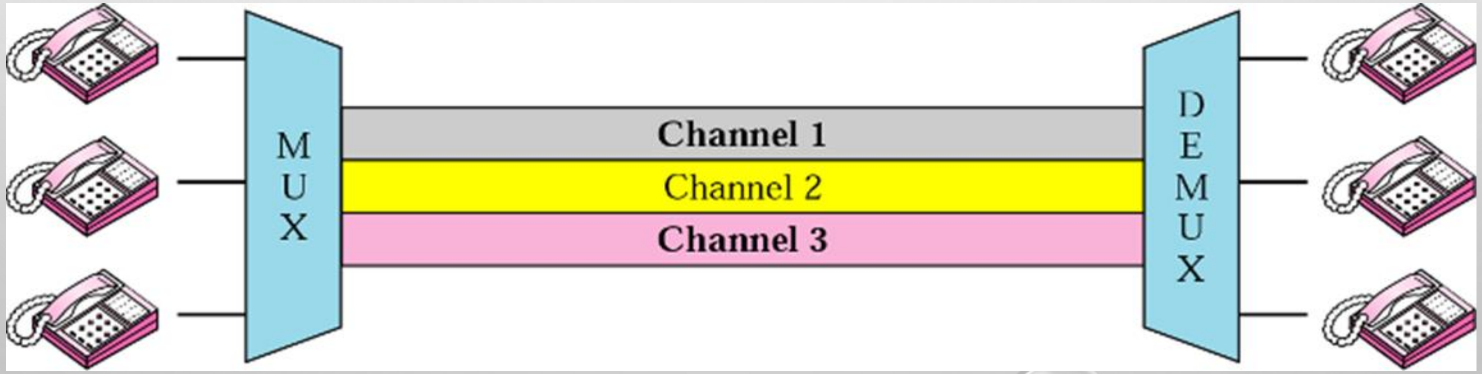


Figure: Frequency-Division Multiplexing

Time – Division Multiplexing:

- 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.
- 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

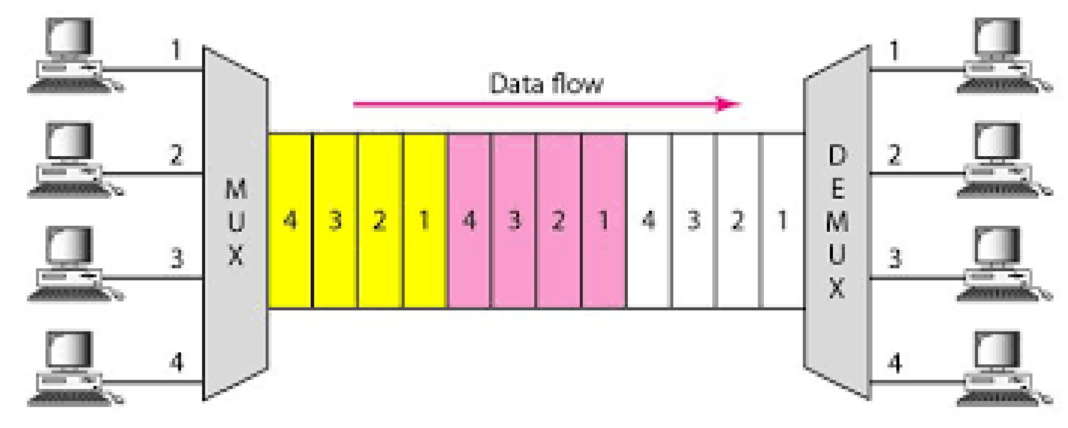


Figure: Time-Division Multiplexing

THANK YOU

END