INTRODUCTION

Introduction To Communication System:

The fundamental purpose of a communications system is the exchange of data between two parties. Below figure presents one particular example, which is communication between a workstation and a server over a public telephone network. Another example is the exchange of voice signals between two telephones over the same network.

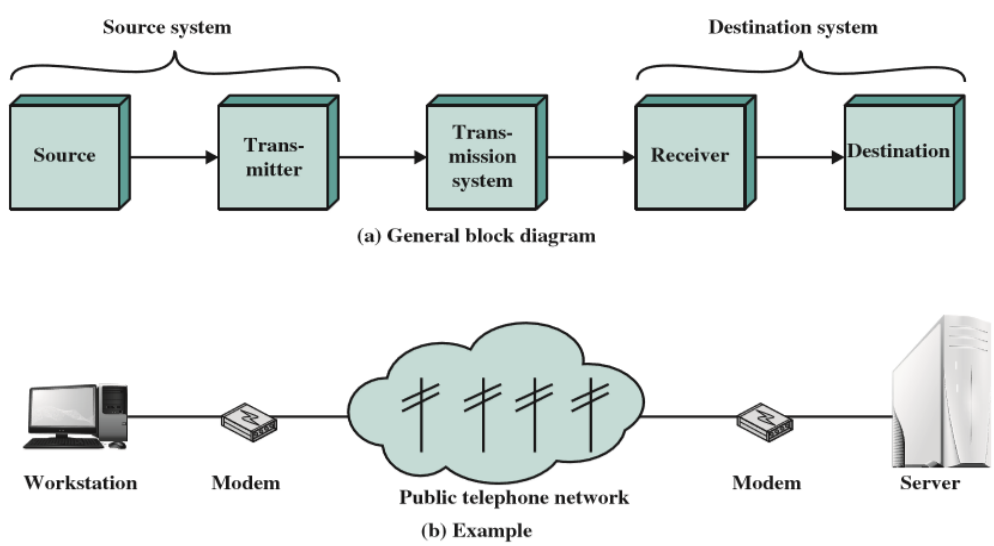


Fig: Communication Model

Following are the key elements of the system:

•Source: This device generates the data to be transmitted; examples are telephones and personal computers.

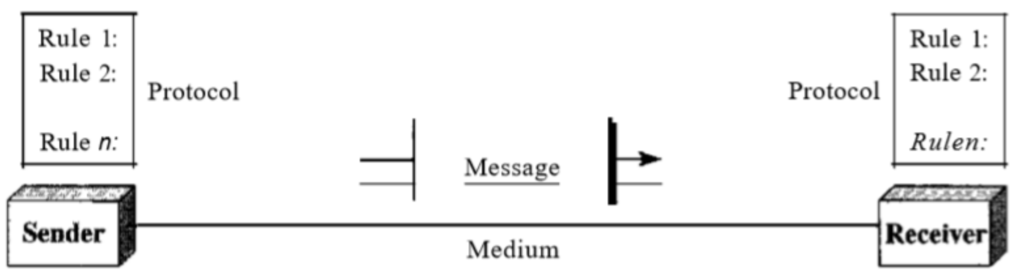
•Transmitter: Usually, the data generated by a source system are not transmitted directly in the form in which they were generated. Rather, a transmitter transforms and encodes the information in such a way as to produce electromagnetic signals that can be transmitted across some sort of transmission system. For example, a modem takes a digital bit stream from an attached device such as a personal computer and transforms that bit stream into an analog signal that can be handled by the telephone network.

•Transmission system: This can be a single transmission line or a complex network connecting source and destination.

•Receiver: The receiver accepts the signal from the transmission system and converts it into a form that can be handled by the destination device. For example, a modem will accept an analog signal coming from a network or transmission line and convert it into a digital bit stream.

•Destination: Takes the incoming data from the receiver.

Generic Communication Model



A generic communication model has five components:

Message: The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.

Sender: The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.

Receiver: The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.

Transmission medium: The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.

Protocol: A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.

Cellular wireless communication:

Cellular network is an underlying technology for mobile phones, personal communication systems, wireless networking etc. The technology is developed for mobile, radio, telephone to replace high power transmitter/receiver systems. Cellular networks use lower power, shorter range and more transmitters for data transmission. Cellular communications systems are wireless mobile communications systems that divide a large geographic area into smaller sections or cells, each with a low-power wireless transmitter, for the purpose of optimizing the use of a limited number of frequencies.

What is a Network?

♣A network is a set of devices (often referred to as nodes) connected by communication links. A node can be a computer, printer, or any other device capable of sending and/or receiving data generated by other nodes on the network.

♣Network is the collection of computer, software and hardware that are all connected to each other to help their work together.

♣A network connects computers by means of cabling system (or wireless media), specialized software and devices that manage data traffic.

♣A network enables users to share files and resources such as printer as well as send message electrically to each other.

Network Topologies:

Network topology is the arrangement of the various elements of a communications networks.

Two types:

♣**Physical Topology**: It describes the geometric arrangement of components that make up the LAN. It refers to the way the computers are cabled together.

♣**Logical Topology**: It describes the possible connections between pairs of networked end –points that can communicate.

Physical Topologies are Bus topology, Ring topology, Star topology, Mesh topology, Tree topology, Hybrid topology, and Daisy Chain topology.

Logical topologies are Token Bus, Token Ring, CSMA/CD etc.

A) Bus Topology:

♣All nodes are connected to a single common cable known as the backbone.

♣Both end of the backbone must be terminated with a terminating resister to prevent signal bounce and complete the circuit.

♣If the backbone cable fails, the entire network effectively becomes unusable.

♣Bus (backbone) carries all network data.

♣Bus networks work best with a limited number of devices.

♣When one computer send a signal up the wire all the computers receive the information but only one with the address that matches accepts the information, the rest disregard the message.

Advantages

♣Easy to implement.

♣Requires least amount of cable to connect the computers together.

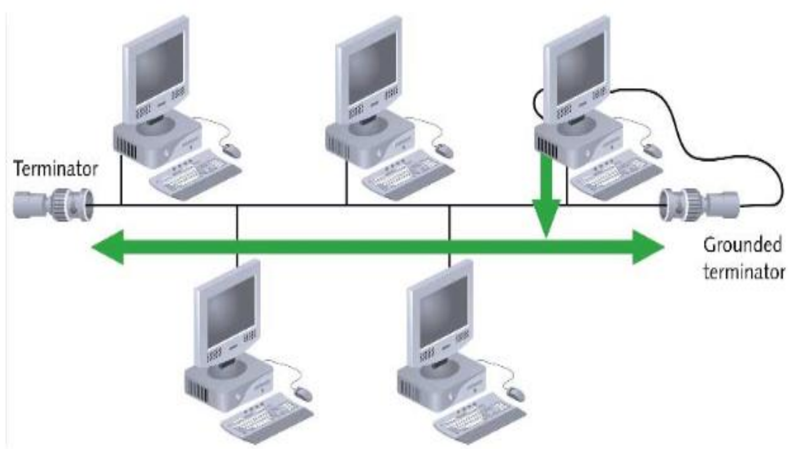
♣Failures of one station does not affect the others.

Disadvantages

•A central cable break can disable the entire network.

• Difficult to troubleshoot.

•Collisions occurs when two nodes send message simultaneously.



B) Star Topology:

♣Most dominant topology type in contemporary LANs.

♣Every node on the network is connected through a central device.

♣Each computer on a star network communicates with a central device that resends the message either to each computer or only to the destination computer.

♣A central device (hub) connects hubs and nodes to the network.

-Each node connects to its own dedicated port on the hub.

-Hubs broadcast transmitted signals to all connected devices.

-we can connect multiple hubs to form a hierarchical star topology.

Advantages:

♣Single computer failure does not necessarily bring down the whole star network

♣Easy to connect new nodes or devices.

♣Centralized management.

♣Most popular topology in use; wide variety of equipment available

♣ The center of the star network is a good place to diagnose the faults.

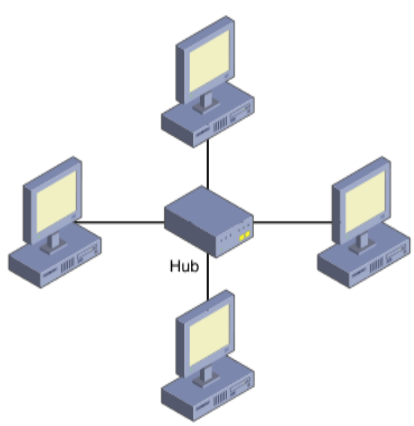
♣Compared to Bus topology it gives far much better performance.

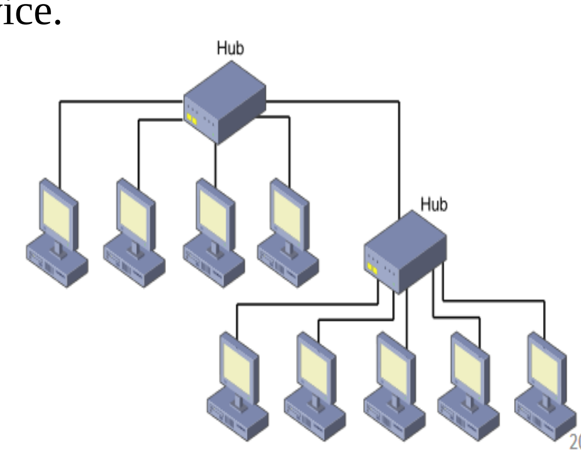
Disadvantages:

♣If central device fails, the entire network goes down.

♣Requires more cable than the bus topology.

♣Performance is depends on central device.





C) Ring Topology:

♣Each node is connected to the two nearest nodes so the entire network forms a circle.

♣Ring network consists of nodes that are joined by point- to-point connections to form a ring.

♣Data are transmitted around the ring using token passing either clockwise or counterclockwise.

♣No central hub

♣Each node will repeat any signal that is on the network regardless its destination. The destination station recognizes its address and copies the frame into a local buffer as it goes by. The frame continues to circulate until it returns to the source station, where it is removed.

♣A failure in any cable or device breaks the loop and can take down the entire network.

Advantages:

•Each computer has equal access to resource.

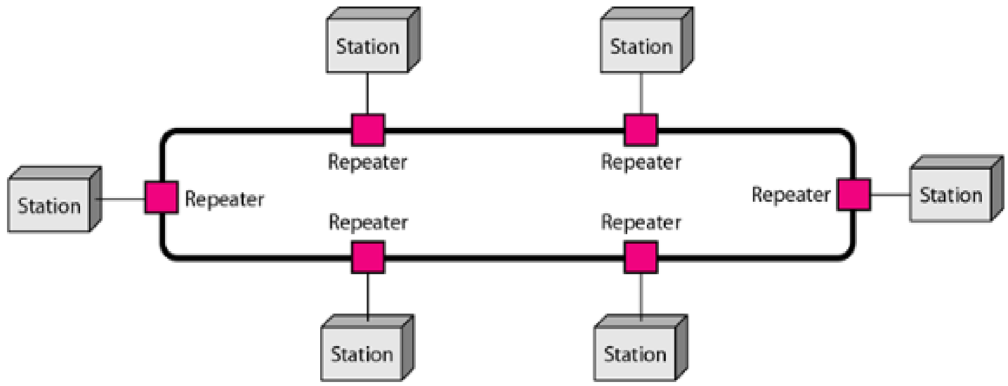
•Performance is better than that of Bus topology

•Network is point- to- point connections. Hence, easier to locate defective node.

Disadvantages:

•Failure of one computer on the ring can affect the whole network.

•Each packet of data must pass through all the computers between source and destination, slower than star topology.



D) Mesh Topology:

The Internet is a mesh topology.

Two Types:

♣Fully Connected

♣Partially Connected

♣Fully Connected Mesh Topology:

•All nodes are interconnected.

•Each and every node has a unique point to point link with all the other nodes. This features leads to the reliability and fault tolerance.

•In mesh topology it will connect or share traffic between two nodes only.

•Messages sent on a mesh network can take any of several possible paths from source to destination.

♣In Partial mesh topology, nodes are connected to only some, not all, of the other nodes.

Advantages:

•When a link of other nodes fail to connect it will not affect the entire network.

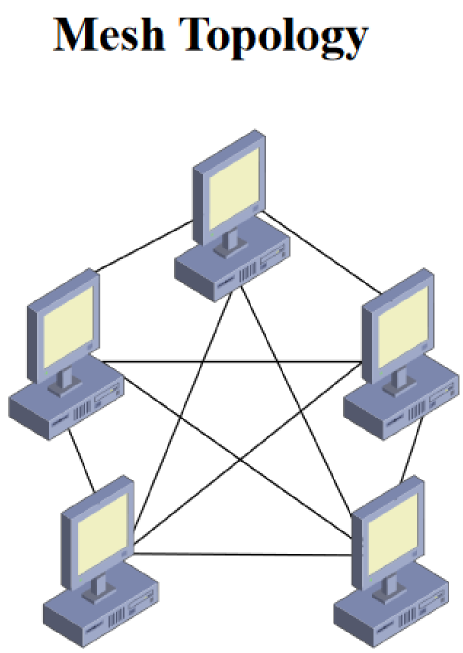
•There is a facility of a unique link between nodes to ensure higher finest data rate and remove traffic issues.

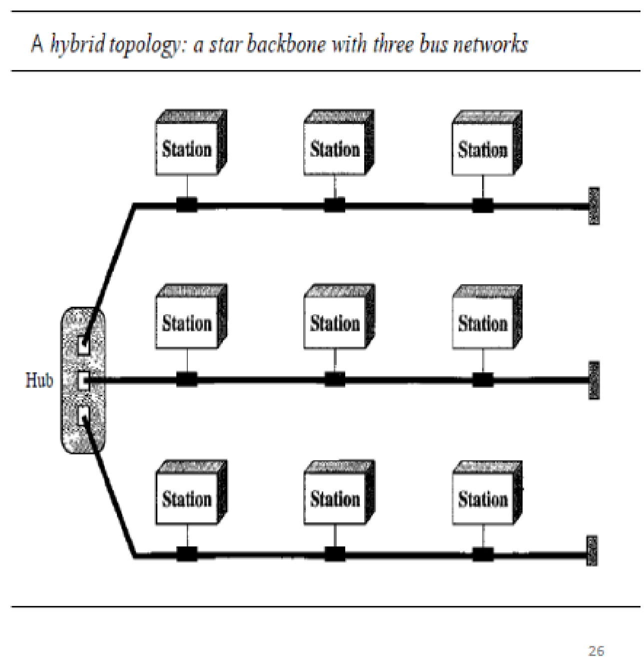
•Error identification and error isolation can be found easy.

•It is robust.

Disadvantages:

- It is most expensive network from the point of view of link cost i.e. cost of cable.
- Bulk wiring is required.
- Installation and configuration are difficult if the connectivity gets more.





PSTN:

PSTN (public switched telephone network) is the world's collection of interconnected voice-oriented public telephone networks. PSTN stands for public switched telephone network, or the traditional circuit-switched telephone network. PTSN comprises all the switched telephone networks around the world that are operated by local, national or international carriers. These networks provide the infrastructure and the services for public telecommunication.

How PSTN works

A public switched telephone network is a combination of telephone networks used worldwide, including telephone lines, fiber optic cables, switching centers, cellular networks, satellites and cable systems. A PSTN lets users make landline telephone calls to one another.

A PSTN is made up of switches at centralized points on a network that function as nodes to enable communication between two points on the network. A call is placed after being routed through multiple switches. Voice signals can then travel over the connected phone lines.

The PSTN phone line is used with traditional dial-up network modems to connect a computer to the Internet. Dial-up Internet connections support up to 56 Kbps. In the early days of the Internet, this was the main method for home Internet access but it became obsolete with the introduction of broadband Internet services.

Assignment:

1. Explain Generic Communication model with suitable diagram?
2. What is Network Topology? Explain different types of network topologies in detail.
3. Explain cellular wireless communication in detail?
4. What is PSTN? Explain how PSTN works?

Chapter 2: Data Transmission

Transmission of Signal

Data or signal transmission occurs between transmitter and receiver over a transmission medium. The successful transmission of data depends on two factors i.e., the quality of signal being transmitted and characteristic of transmission medium. Data is transformed in the form of electromagnetic signals across a transmission medium.

Analog and Digital Data Transmission

Analog and Digital Data:

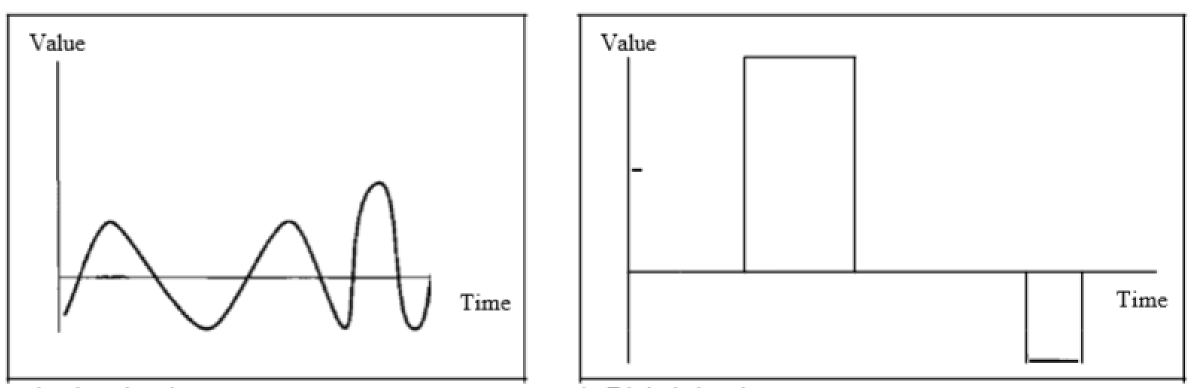
Data can be analog or digital. The term analog data refers to information that is continuous; digital data refers to information that has discrete states. For example, an analog clock that has hour, minute, and second hands gives information in a continuous form; the movements of the hands are continuous. On the other hand, a digital clock that reports the hours and the minutes will change suddenly from 8:05 to 8:06.

Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal. Digital data take on discrete values. For example, data are stored in computer memory in the form of 0s and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

Analog Signal and Digital Signal:

An analog signal has infinitely many levels of intensity over a period of time. As the wave moves from value A to value B, it passes through and includes an infinite number of values along its path. A digital signal, on the other hand, can have only a limited number of defined values. Although each value can be any number, it is often as simple as 1 and O.

Figure 3.1 *Comparison ofanalog and digital signals*



a. Analog signal
b. Digital signal

Analog Transmission:

Analog transmission is a transmission of analog data such as human voice using a continuous signal which varies continuously in amplitude, phase, or some other property of the variable.

During transmission of analog signal, the signal will become weaker after a certain distance. To achieve larger distance, the analog transmission system include amplifier that boosts the energy (strength) of the signal.

→Amplifier also amplifies noise.

Digital Transmission:

Digital Transmission is the transmission of signals that vary discretely with time. Digital signals use discrete values for the transmission of binary information over a communication medium such as a network cable or a telecommunications link.

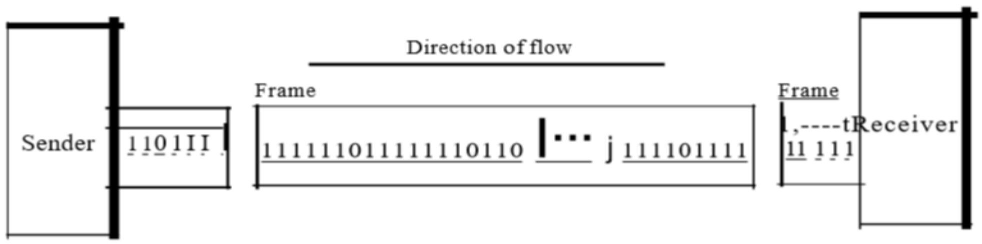
Unlike analog transmission, digital transmission uses repeaters to enhance the signal strength. A repeater receives the digital signal, recovers the pattern of 0's and 1's and retransmits the signal.

→Repeater regenerates the signal as a result noise gets eliminated.

Synchronous Transmission:

Synchronous data transmission is a data transfer method in which a continuous stream of data signals is accompanied by timing signals (generated by an electronic clock) to ensure that the transmitter and the receiver are in step (synchronized) with one another. The data is sent in blocks (called frames or packets) spaced by fixed time intervals.

In other words, data are transmitted as an unbroken string of 1s and Os, and the receiver separates that string into the bytes, or characters. In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.

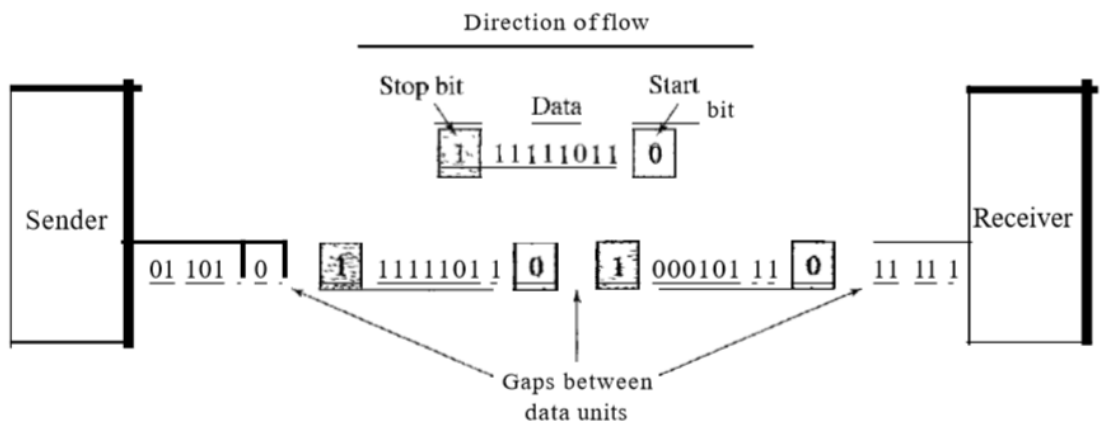


Asynchronous Transmission:

Asynchronous transmission is so named because the timing of a signal is not important. Instead, information is received and translated by agreed upon patterns. Patterns are based on grouping the bit stream into bytes. Each group, usually 8 bits, is sent along the link as a unit.

Without synchronization, the receiver cannot use timing to predict when the next group will arrive. To alert the receiver to the arrival of a new group, therefore, an extra bit is added to the beginning of each byte. This bit, usually a 0, is called the start bit. To let the receiver, know that the byte is finished, 1 or more additional bits are appended to the end of the byte. These bits, usually 1s, are called stop bits.

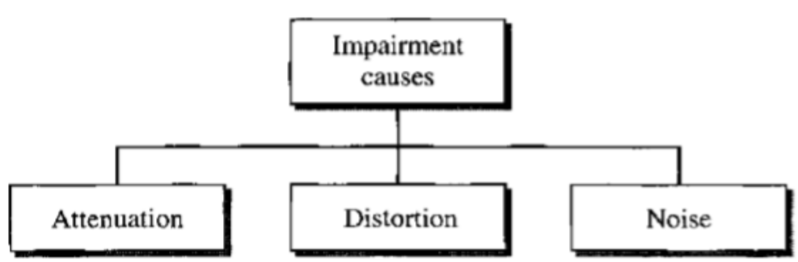
Note: In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.



Transmission Impairments:

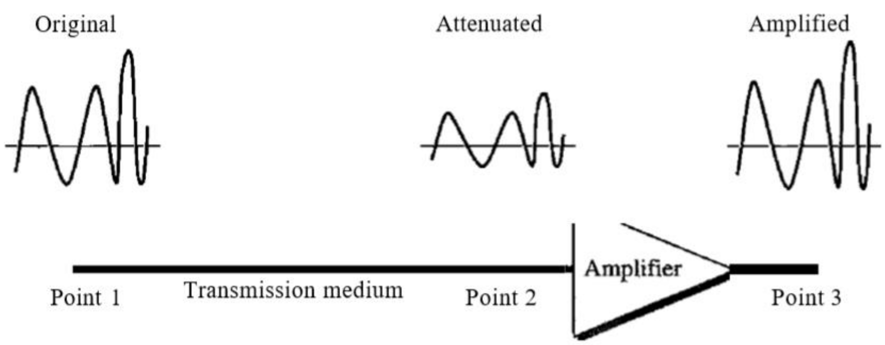
Transmission impairment is defined as a condition in which the signal's strength and quality gets deteriorated (degraded) as it gets transmission. In any communication system, the received signal is never identical to the transmitted one due to some transmission impairments. Three causes of impairment are attenuation, distortion, and noise.

Causes of impairment:



Attenuation

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. That is why a wire carrying electric signals gets warm, if not hot, after a while. Some of the electrical energy in the signal is converted to heat. To compensate for this loss, amplifiers are used to amplify the signal. Figure 3.26 shows the effect of attenuation and amplification.



Attenuation may be expressed in decibels(dB) as under:

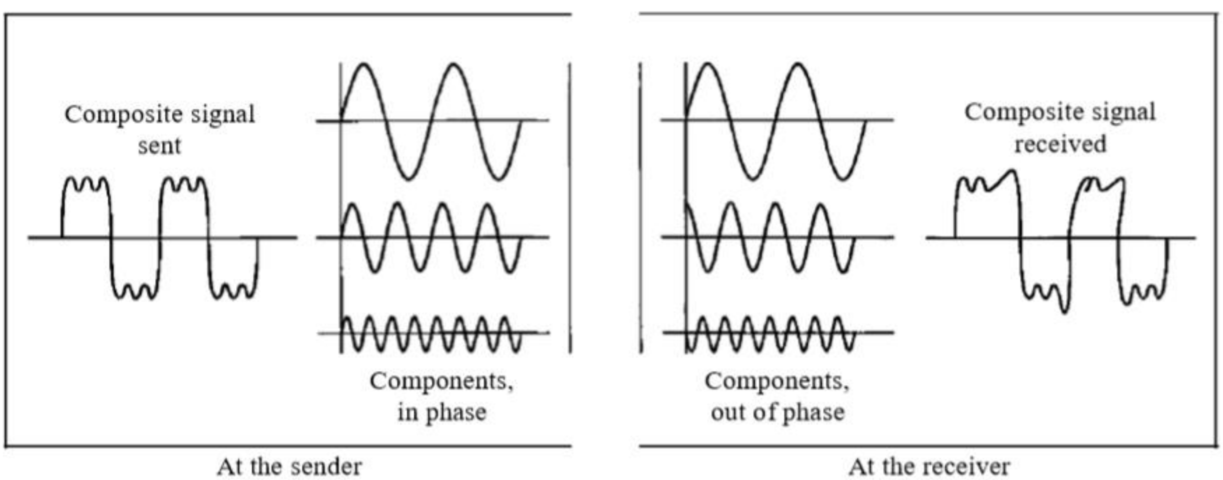
$$dB = 10 \log_{10} \frac{P_2}{P_1}$$

Where, P2 = power at receiving end

P1= power at sending end

Distortion:

Distortion means that the signal changes its form or shape. Distortion can occur in a composite signal made of different frequencies. Each signal component has its own propagation speed through a medium and, therefore, its own delay in arriving at the final destination. Differences in delay may create a **difference in phase** if the delay is not exactly the same as the period duration. In other words, 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.



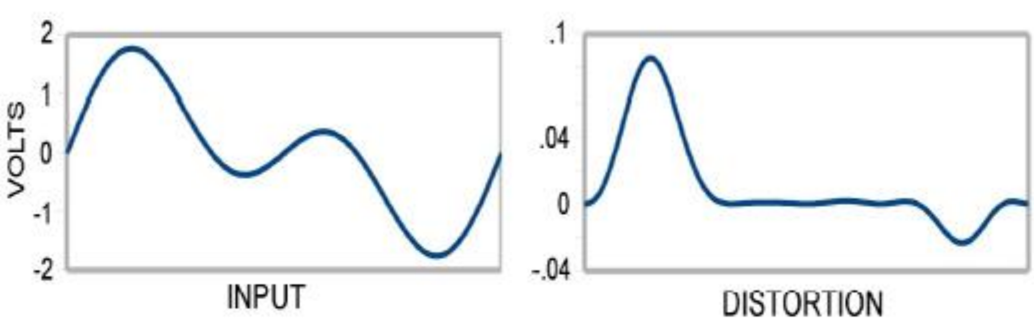
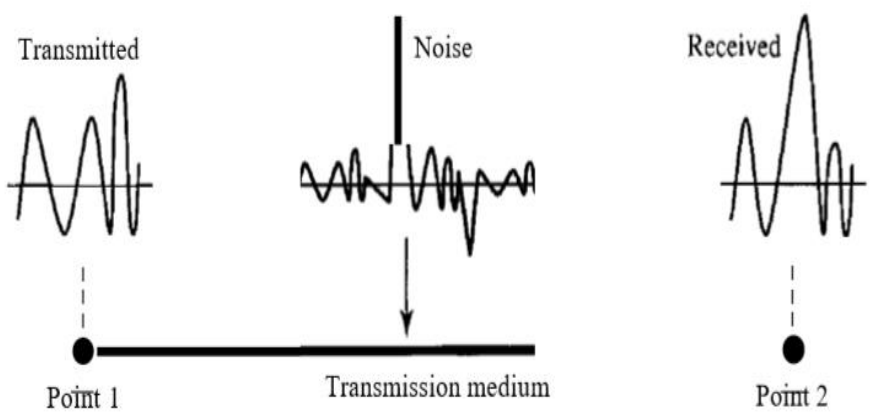


FIG 7 SIMPLE INTERMODULATION DISTORTION

Noise:

Noise is another cause of impairment. 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. These devices act as a sending antenna, and the transmission medium acts as the receiving antenna. Crosstalk is the effect of one wire on the other. One wire act as a sending antenna and the other as the receiving antenna. Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on.



Channel Capacity (Maximum data rate):

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

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

Two theoretical formulas were developed to calculate the data rate: one by Nyquist for a noiseless channel, another by Shannon for a noisy channel.

Noiseless Channel: Nyquist Bit Rate

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

Bit-Rate= 2 x bandwidth x Log2 L

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data, and BitRate is the bit rate in bits per second.

Example:

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

$$\text{Bit-Rate} = 2 \times 3000 \times \log_2(2) = 6000 \text{ bps}$$

Noisy Channel: Shannon Capacity

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

Capacity =bandwidth X log2 (1 +SNR)

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ratio, and capacity is the capacity of the channel in bits per second.

Signal-to-noise ratio is defined as the ratio of the power of a signal to the power of background noise: i.e SNR = Signal Power/Noise Power.

Example

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 to 3300 Hz) assigned for data communications. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

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

Signal Encoding and Decoding:

Encoding is the process of converting the data or a given sequence of characters, symbols, alphabets etc., into a specified format, for the secured transmission of data.

Decoding is the reverse process of encoding which is to extract the information from the converted format.

In the sender side there is a device called encoder encodes the data to the required signal and the signal is transmitted through the channel. In the receiver side there is another device called decoder which decodes the signal to data using the encoding information of the signal.

Encoding Techniques

The data encoding technique is divided into the following types, depending upon the type of data conversion.

- **Analog data to Analog signals** − The modulation techniques such as Amplitude Modulation, Frequency Modulation and Phase Modulation of analog signals, fall under this category.
- **Analog data to Digital signals** − This process can be termed as digitization, which is done by Pulse Code Modulation *PCM*. Hence, it is nothing but digital modulation. As we have already discussed, sampling and quantization are the important factors in this. Delta Modulation gives a better output than PCM.

- Digital **data to Analog signals** − The modulation techniques such as Amplitude Shift Keying *ASK*, Frequency Shift Keying *FSK*, Phase Shift Keying *PSK*, etc., fall under this category. These will be discussed in subsequent chapters.

- **Digital data to Digital signals −** These are in this section. There are several ways to map digital data to digital signals. Some of them are: RZ, NRZ, Differential Manchester etc.

Transmission Media:

A communication channel is called a medium. There are two types of media:

1**. Guided Media**: twisted pair cable, coaxial cable and fiber optic cable

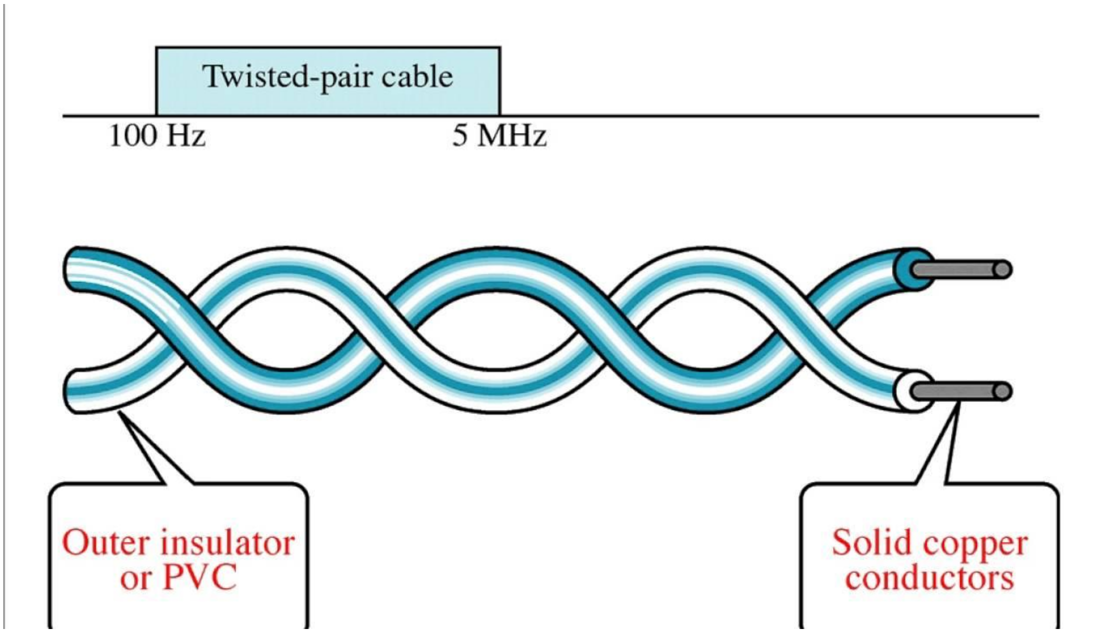
2. **Unguided Media**: radio wave, microwave, infrared and satellite.

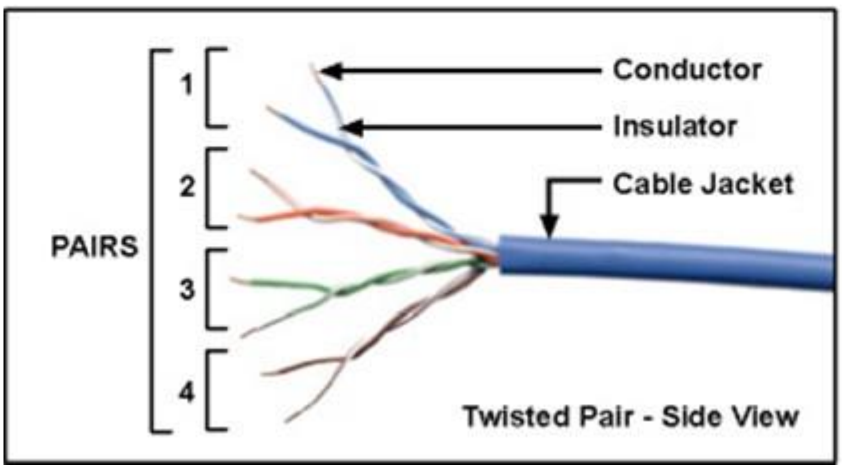
Guided (Wired) Media: Guided Media is a communication medium which allows the data to get guided along it i.e. physical connection is need.

Unguided (Wireless) Media: The wireless media is called unguided media. The signal propagates in form of wireless electromagnetic waves.

GUIDED MEDIA

1. Twisted Pair Cable
   - A twisted pair consists of two conductors (copper), each with its own plastic insulation, twisted together.
   - One of the wires is used to carry signals to receiver, and the other is used only as a ground reference.
   - A signal is usually carried as the difference in voltage between the two wires in the pair. This provides better immunity to external noise because the noise tends to affect both wire the same, leaving the differential voltage unchanged.
   - Frequency range for twisted pair cable is 100Hz to 5MHz.





- Two types :
  - Unshielded twisted pair(UTP)

    UTP contains no shielding and is more susceptible to external noise but is the most frequently used because it is inexpensive and easier to install. E.g. Cat5e, Cat6, Cat6a and Cat7
  - Shielded twisted pair(STP)

    STP cable contains an outer conductive shield that is electrically grounded to insulate the signals from external electrical noise. STP also uses inner foil shields to protect each wire pair from noise generated by the other pairs. Expensive than UTP.eg. IBM STP-A

Characteristics:

a. Easy to install
b. Low cost (cheaper than coaxial and optical fiber cable)
c. High speed capacity i.e. data transfer rate of CAT5e is up to 1 Gbps.

Applications:

i. In telephone lines to carry voice and data channels.
j. To create a local area network.
k. Ethernet

2. **Coaxial Cable:**
   - Coaxial cable carries signals of higher frequency range than those in twisted pair cable.
   - It has a central core conductor of solid or standard wire (usually copper) enclosed in an insulating sheath, which is in turn encased in an outer conductor of metal foil.
   - The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath and the whole cable is protected by a plastic cover.

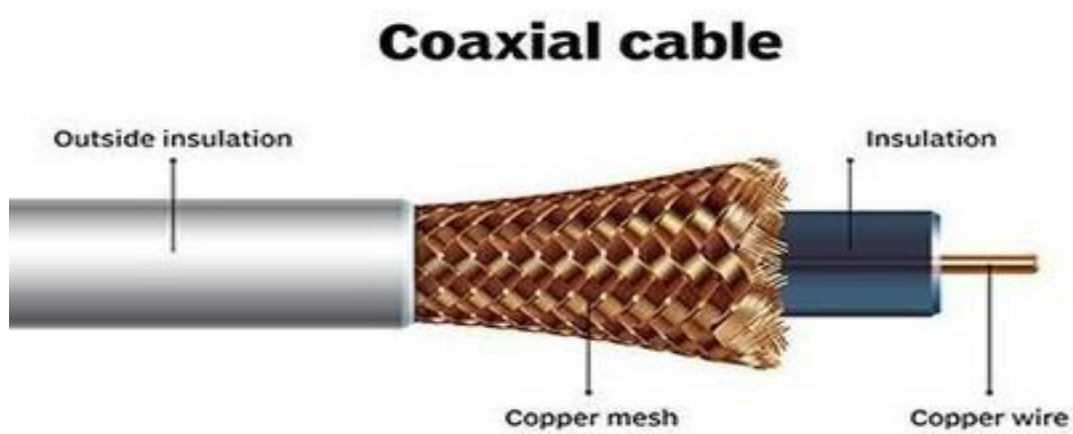
   Characteristics:

   a. Due to shield provided, this cable has excellent noise immunity.
   b. It has large bandwidth and low losses.

c. Expensive than twisted pair cables but cheaper than fiber optic cable.
d. Data rate of 10 Mbps and can be increased with increase in diameter of the inner conductor.

Category of coaxial cable:

Category	Impedance	Use
RG-59	75-ohm	cable TV (analog transmission)
RG-58	50-ohm	Thin Ethernet (Digital trans.)
RG-11	50-ohm	Thick Ethernet (Digital trans.)

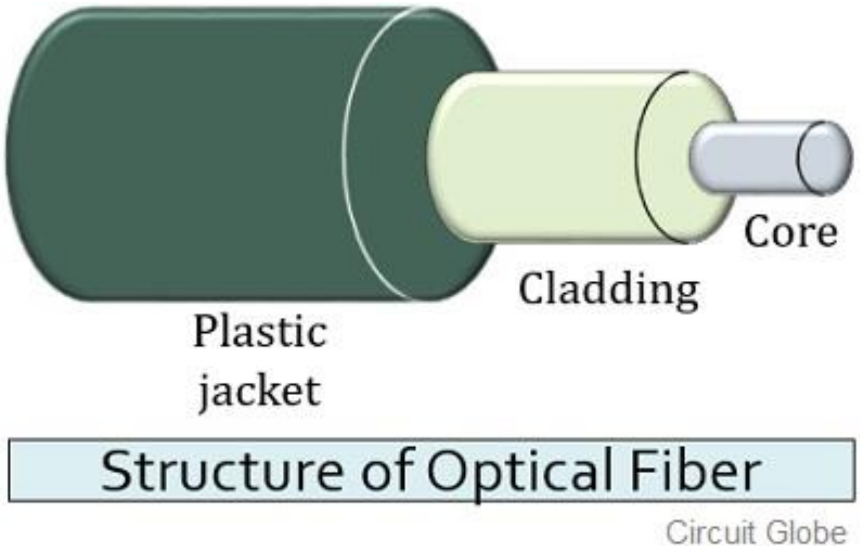


Applications:

a. Analog and digital telephone network.
b. Cable TV.
c. Thick and thin Ethernet.

Optical Fiber:

Optical fiber is a thin, flexible medium capable of guiding an optical ray. Optical fiber cable has a cylindrical shape and consists of three sections: core, cladding and jacket.



The core is a inner layer and is made up of fiber or glass. It is surrounded by a glass cladding which has a lower refractive index than the core. The outer most layer is of plastic which is called as jacket.

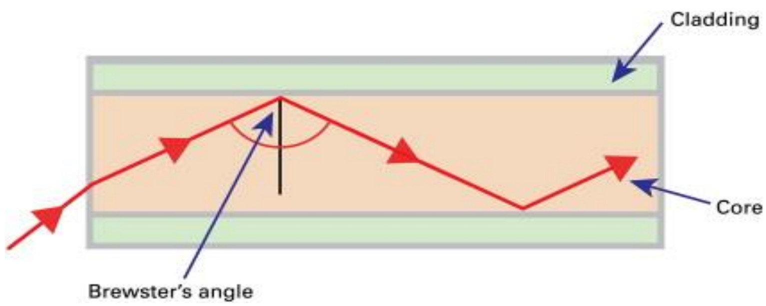


Fig: transfer of light ray inside the core

Characteristics:

a. The light ray inside the fiber does not escape through the walls because of the total internal reflection.
b. Very high data transfer rate up to 10 Gbps.
c. It is smaller and light weight but expensive.
d. It is not affected by noise signals
e. These cables have much lower attenuation and can carry signal to longer distances without using amplifiers and repeaters in between.
f. The installation of fiber optic cables is difficult and tedious.
g. Joining the optical fibers is a difficult job.

Unguided Transmission:

Unguided media transports electromagnetic wave without using a physical conductor. This type of communication is called wireless communication. Signals are normally broadcasted through air and thus are available to anyone who has a device (antenna) capable to receive them.

Unguided signals can travel from source to destination in several ways.

1. Ground propagation
2. Sky propagation
3. Line-of-sight propagation

1. Radio Transmission:

Basic idea is to transmit signal in the form of radio waves. Radiowaves have frequencies between 3 kHz and 1 GHz. Radiowaves can broadcast omnidirectionally or directionally. Various kinds of antennas are used to broadcast these signals.

→ Omnidirectional i.e. all directions
→ Radiowaves can penetrate through walls.
→ Used TV broadcasting and F.M radio broadcasting

Applications:

a. Cellular communication
b. Wireless LAN
c. Satellite communication

2. Microwave transmission

The microwave transmission uses higher frequency range than radio frequency as a result, they produce better throughput and performance. Microwave are basically electromagnetic waves having frequency range between 1 GHz and 300 GHz.

→ Microwaves are unidirectional.
→ Microwaves propagation is a line of sight propagation.
→ Cannot penetrate through walls.

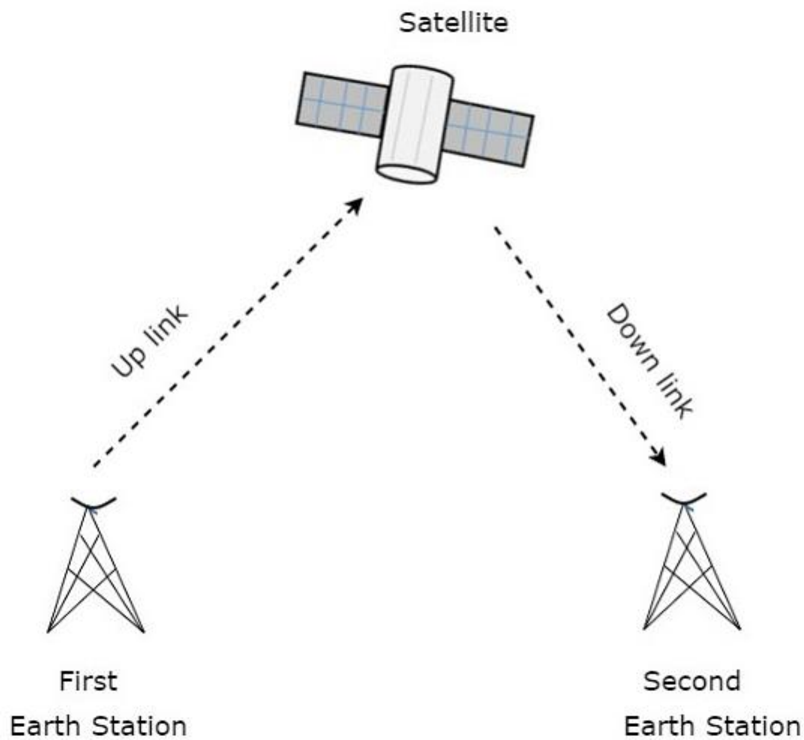
Applications:

a. In cellular phones
b. In satellite networks
c. Wireless LAN

3. Satellite Transmission

A satellite is simply a repeater. It consists of several transponders each of which listens to some portion of the spectrum, amplifies the incoming signal and the rebroadcasts it at another frequency to avoid interference with the incoming signal.

Satellite communication system consists of ground stations for transmitting and receiving signals and a communication satellite in the space. The range of frequencies used for transmission of signals from ground station to the satellite is **uplink** frequency and those used for transmission of signals from satellites to ground station is **downlink** frequency. Uplink and downlink frequencies are different **to avoid interference.**



Types of satellite:

1. **LEO (Low Earth Orbit)**

   They are very close to earth ranging from 500 to 1500 km above the surface. They are only visible for 15 to 20 minutes. As they are very close to earth surface their coverage area is small but less propagation delay.

2. **MEO (Medium Earth Orbit)**

   They are somewhere between 8000 km to 18000 km above earth surface. Visible for around 2 to 8 hours and has larger coverage area than LEO satellite.

3. **GEO(Geostationary Earth Orbit)**

   These satellites are in orbit 35,863 km above the earth’s surface along the equator. They rotate at the speed of earth rotation so they are visible all the time.

Larger coverage area than LEO and MEO but more

S.N	Guided Media	Unguided Media
1.	The signal energy propagates through wires in guided media.	The signal energy propagates through air in unguided media
2.	Guided media is used for point to point communication	Unguided media is used for point to point and multipoint communication.
3.	Signals are in the form of voltage, current or photons in guided media.	Signals are in the form of electromagnetic waves in unguided media.
4.	There is specific direction for sending a signal.	There is no particular direction to send signals.
5.	e.g. twisted pair, coaxial cable and optical fiber	E.g. radiowaves, microwaves and infrared.

propagation delay.

Advantages:

a. Larger coverage area than a terrestrial system.
b. Higher bandwidth.

Disadvantages:

a. Larger propagation delay than terrestrial system.
b. Launching satellite into orbit is costly.

Comparison between guided and unguided media:

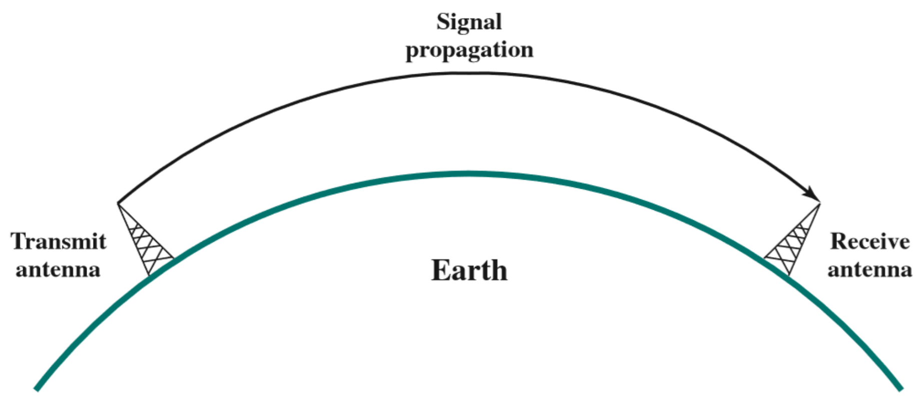
Wireless propagation:

Wireless propagation occurs through one of the following three routes:

1. Ground wave Propagation

   Ground Wave propagation is a method of radio wave propagation that uses the area between the surface of the earth and the ionosphere for transmission. Ground wave propagation can be found for the frequency range of few kHz to 2MHz. There are several factors that help an electromagnetic wave to be transmitted through ground wave propagation. One factor is that the electromagnetic wave induces a current in the Earth’s surface, the result of which is to slow the wavefront near the Earth, causing the wavefront to tilt downward and hence follow the Earth’s curvature. Another factor is **diffraction**, which is a phenomenon having to do with the behavior of electromagnetic waves in the presence of obstacles. Electromagnetic waves in this frequency range are scattered by the atmosphere in such a way that they do not penetrate the upper atmosphere.

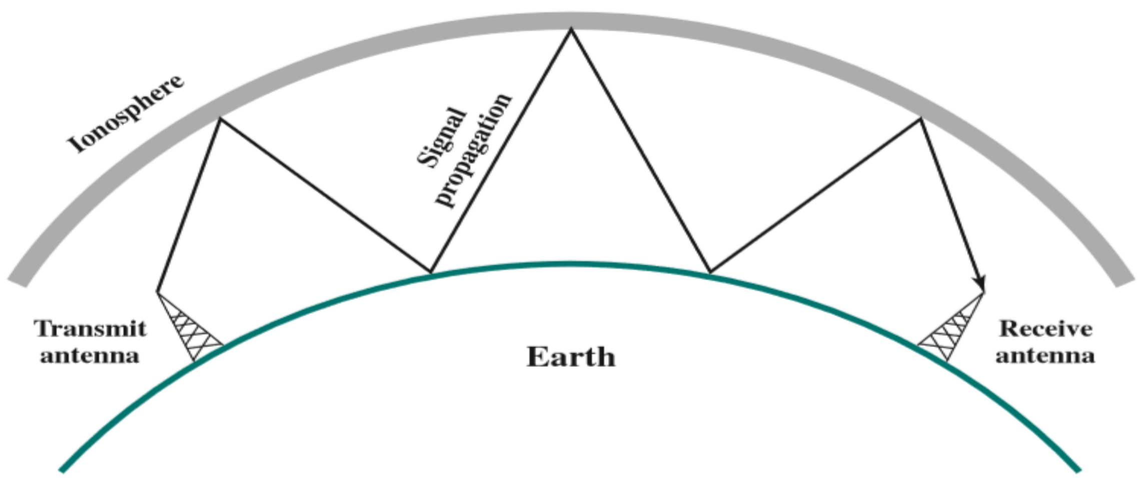
   e.g. AM radio uses ground wave propagation.



(a) Ground wave propagation (below 2 MHz)

2. Sky wave Propagation

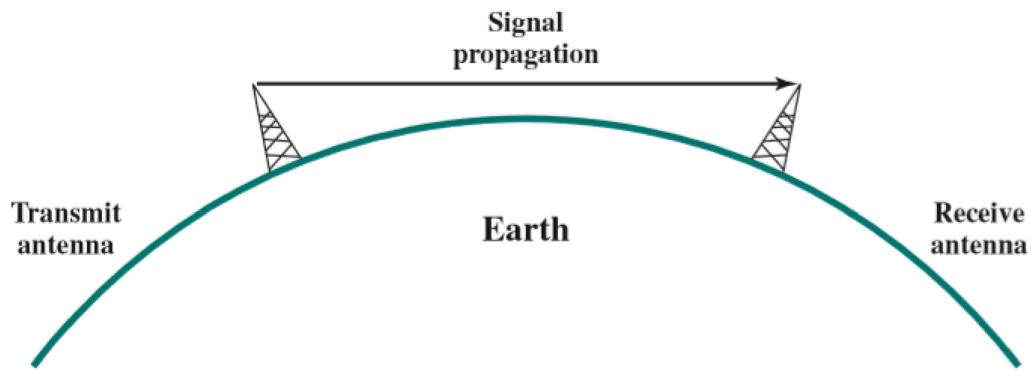
With sky wave propagation, a signal from an earth-based antenna is reflected from the ionized layer of the upper atmosphere (ionosphere) back down to Earth. Waves having frequency higher than 2MHz and less than 30 MHz can be propagated through sky wave propagation. With this propagation mode, a signal can be picked up thousands of kilometers from the transmitter. Eg. Are BBC news



(b) Sky wave propagation (2 to 30 MHz)

3. Line-of-sight propagation

Above 30 MHz, neither ground wave nor sky wave propagation modes operate, and communication must be by line of sight. For satellite communication, a signal above 30 MHz is not reflected by the ionosphere and therefore a signal can be transmitted between an earth station and a satellite. For ground-based communication, the transmitting and receiving antennas must be within an effective line of sight of each other. Eg. satellite communication, FM radio and microwave.

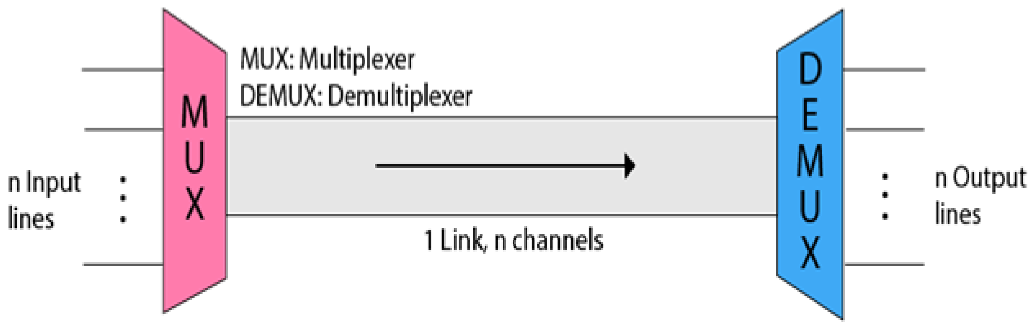


(c) Line-of-sight (LOS) propagation (above 30 MHz)

Multiplexing and Multiple Access Technique

Multiplexing concept

- Multiplexing is a technique that allows the simultaneous transmission of multiple signal across a single data link.
- Multiplexing is done using a device called multiplexer (MUX) that combines 'n' input lines to generate one output line (i.e. many to one).
- At the receiving end a device called demultiplexer (DEMUX) is used that separates signal into its component signal i.e. one input and many output.



Advantages:

1. More than one signal can be send over a single link.
2. Effective use of bandwidth.

Modes of signal Transmission:

Simplex

In simplex transmission mode, the communication between sender and receiver occurs in only one direction. The sender can only send the data, and the receiver can only receive the data. The receiver cannot reply to the sender.

To take a keyboard / monitor relationship as an example, the keyboard can only send the input to the monitor, and the monitor can only receive the input and display it on the screen. The monitor cannot reply, or send any feedback, to the keyboard.

Half Duplex

The communication between sender and receiver occurs in both directions in half duplex transmission, but only one at a time. The sender and receiver can both send and receive the information, but only one is allowed to send at any given time.

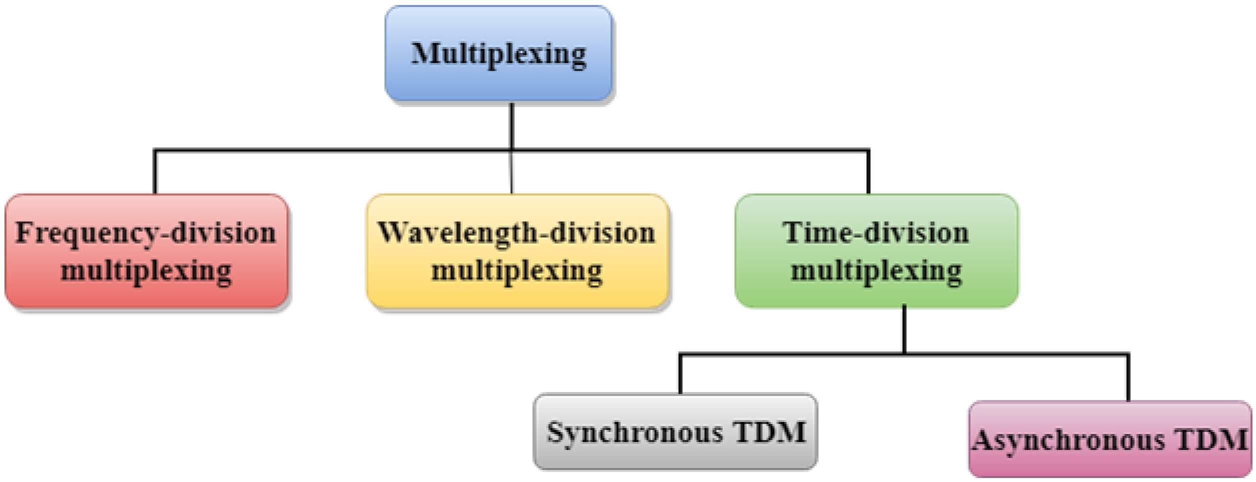
For example, in walkie-talkies, the speakers at both ends can speak, but they have to speak one by one. They cannot speak simultaneously.

Full Duplex

In full duplex transmission mode, the communication between sender and receiver can occur simultaneously. The sender and receiver can both transmit and receive at the same time.

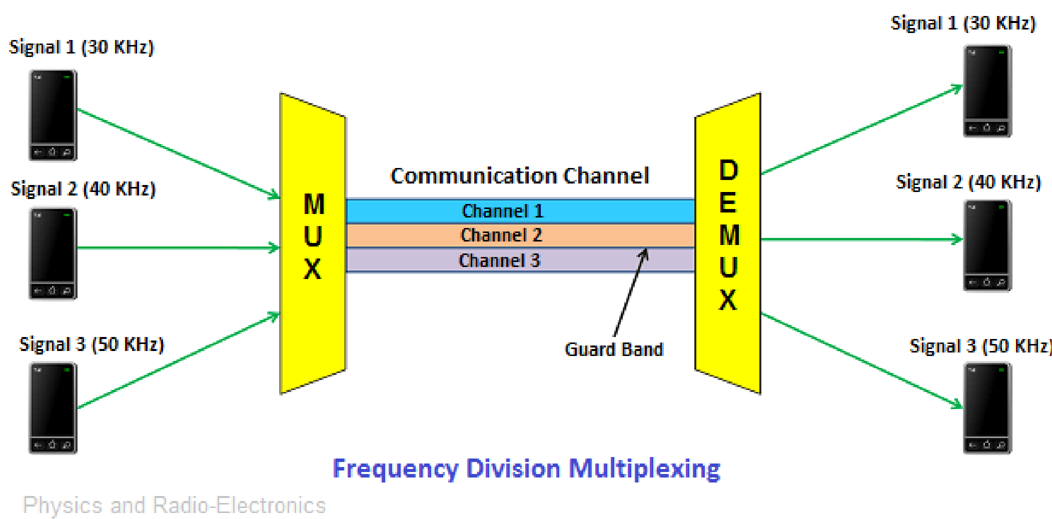
For example, in a telephone conversation, two people communicate, and both are free to speak and listen at the same time.

<u>Types of Multiplexing:</u>



Frequency Division Multiplexing (FDM):

- FDM is an analog technique that combines **analog signals**
- FDM can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidth of the signals to be transmitted.



→The frequency division multiplexing divides the bandwidth of a channel into several logical sub-channels. Each logical sub-channel is allocated for a different signal frequency. The individual signals are filtered and then modulated (frequency is shifted), in order to fit exactly into logical sub-channels.

→Each logical sub-channel is separated by an unused bandwidth called Guard Band to prevent overlapping of signals. In other words, there exists a frequency gap between two adjacent signals to prevent signal overlapping.

Above Figure Explanation:

The above figure shows the schematic diagram of an FDM system. The transmitter end contains multiple transmitters and the receiver end contains multiple receivers. The communication channel is present between the transmitter and receiver.

At transmitter end, each transmitter sends a signal of different frequency. In the above figure, the transmitter 1 sends a signal of 30 kHz, transmitter 2 sends a signal of 40 kHz, and transmitter 3 sends a signal of 50 kHz. These signals of different frequencies are then multiplexed or combined by using a device called multiplexer. It then transmits the multiplexed signals over a communication channel.

At the receiver end, the multiplexed signals are separated by using a device called demultiplexer. It then sends the separated signals to the respective receivers. In the above figure, the receiver 1 receives signal of 30 kHz, receiver 2 receives signal of 40 kHz, and receiver 3 receives signal of 50 kHz.

Advantages:

1. A large number of signals can be transmitted simultaneously.
2. FDM doesn't need synchronization between sender and receiver.

Disadvantages:

1. The communication channel must have a very large bandwidth.

2. Large number of modulator and filters are required.

Application:

1. Frequency division multiplexing is used for FM and AM radio broadcasting.
2. It is used in television broadcasting.

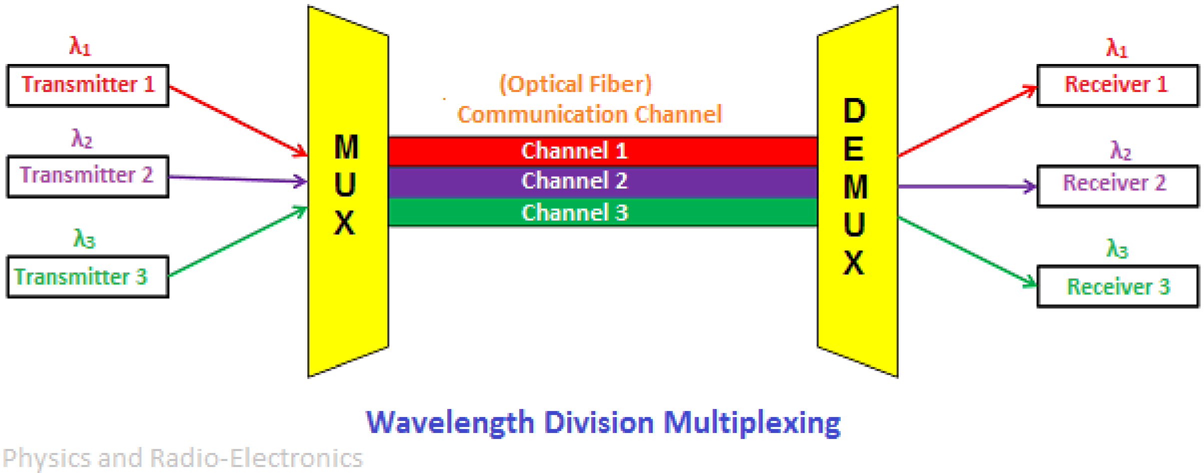
Wavelength Division Multiplexing (WDM):

Wavelength division multiplexing is an **analog technique**. It is the most important and most popular method to increase the capacity of an optical fiber. The working principle of wavelength division multiplexing is similar to frequency division multiplexing. The only difference is in wavelength division multiplexing optical signals are used instead of electrical signals.

Wavelength division multiplexing (WDM) is a technique of multiplexing multiple optical carrier signals through a single optical fiber channel by varying the wavelengths or colors of laser lights.

Wavelength division multiplexing is a technology that increases the bandwidth of a communication channel (optical fiber) by simultaneously allowing multiple optical signals through it.

The combined signal is transmitted via a single optical fiber cable. At the receiving end, a demultiplexer splits the incoming beam into its components and each of the beams is send to the corresponding receivers.



→The wavelength division multiplexing divides the bandwidth of a channel into several logical sub-channels according to its wavelength. It allots each logical sub-channel for a different light color or optical signal wavelength. The individual signals are filtered and then modulated (wavelength is shifted), to fit exactly into logical sub-channels.

Advantages of Wavelength Division Multiplexing (WDM)

1. WDM allows transmission of data in two directions simultaneously (Full duplex).

2. Greater transmission capacity

3. High security

4. Long distance communication with low signal loss

Disadvantages:

1. The cost of the system increases with the addition of more optical components.
2. Wavelength tuning is a difficult job.

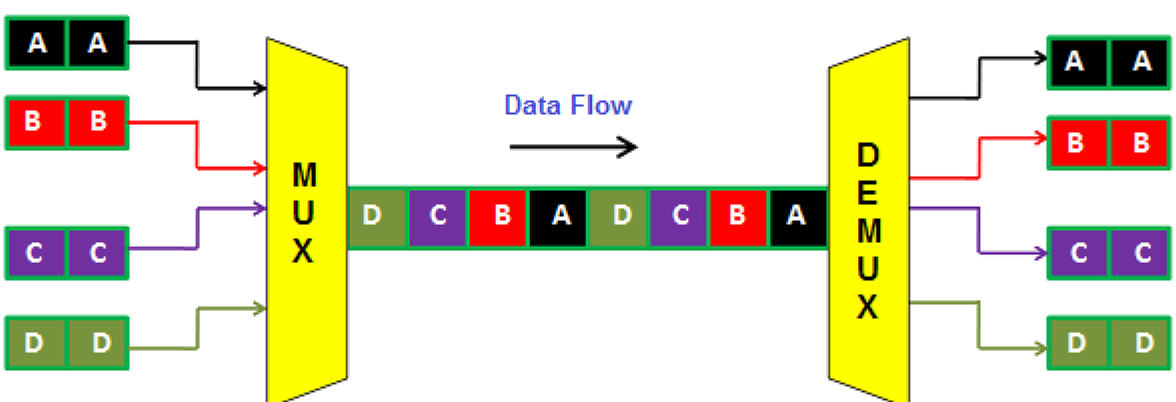
Time Division Multiplexing (TDM):

Time Division Multiplexing is digital technique, i.e. it is the process of combining multiple digital signals into one signal.

Time Division Multiplexing is a technique in which multiple signals are combined and transmitted one after another on the same communication channel.

At the receiver side, the signals are separated and received. Each signal is received by a user at a different time.

In frequency division multiplexing, all signals of different frequencies are transmitted simultaneously. But in time division multiplexing, all signals operate with the same frequency are transmitted at different times.



Time Division Multiplexing

In time division multiplexing, each user is allotted a particular time interval called time slot during which data is transmitted. The time interval (time slot) allotted to each receiver (user) is so small that the receiver will not detect that some time was used to serve another receiver (user).

In time division multiplexing, all signals are not transmitted simultaneously; instead, they are transmitted one after another. For example, as shown in the above figure, at first, we send signal A. Then after second signal B and then after third signal C and finally, we send last signal D. Thus, each user occupies an entire bandwidth for a short period of time.

Types of TDM (Time Division Multiplexing)

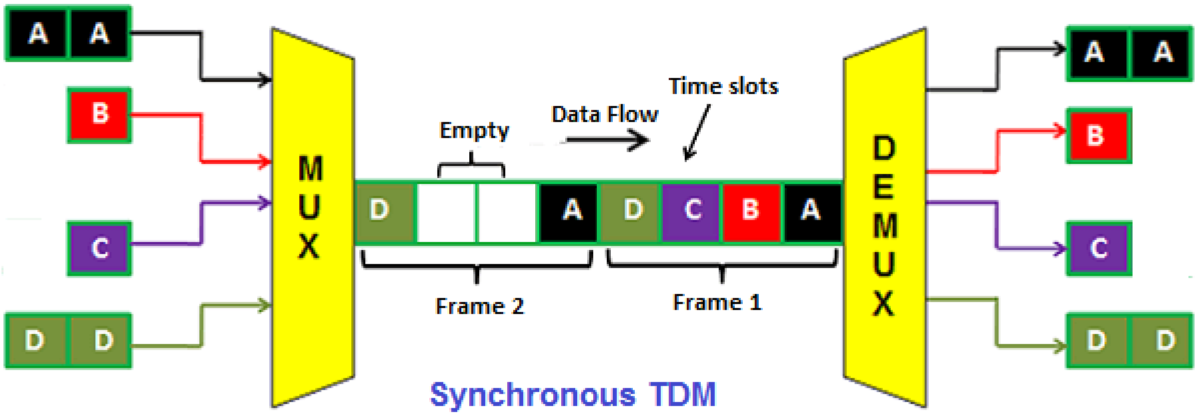
Time Division Multiplexing is mainly classified into two types:

- Synchronous TDM (Time Division Multiplexing)
- Asynchronous TDM (Time Division Multiplexing)

Synchronous TDM (Time Division Multiplexing)

In synchronous time division multiplexing, each device (transmitter) is allotted with a fixed time slot, regardless of the fact that the device (transmitter) has any data to transmit or not. The device has to transmit data within this time slot. If the device (transmitter) does not have any data to send then its time slot remains empty.

The various time slots are arranged into frames and each frame consists of one or more time slots dedicated to each device (transmitter).



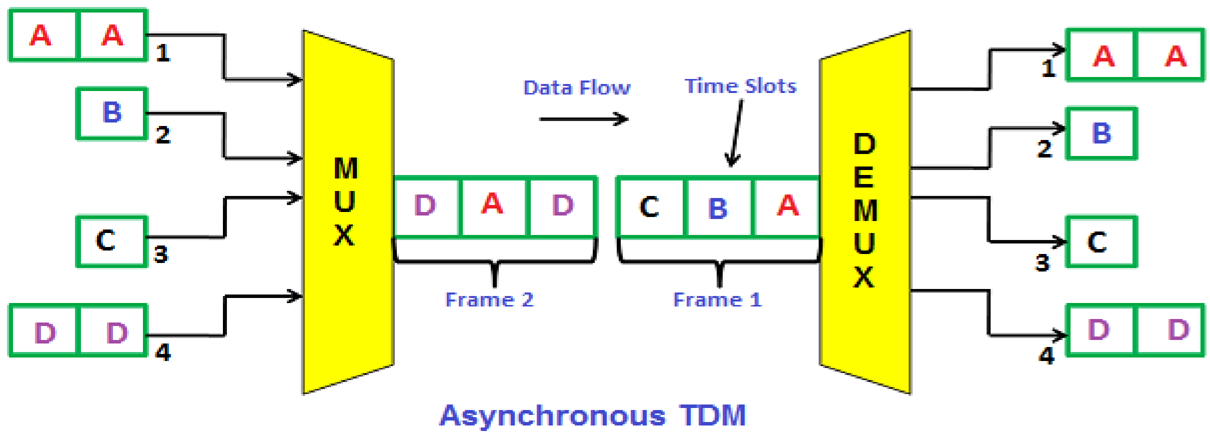
From above figure we can see that in time frame 2, the transmitter B and C does not have any data to send so the time slot B and C remains empty.

The main drawback of synchronous time division multiplexing is that the channel capacity is not fully utilized. Hence, the bandwidth goes wasted.

Asynchronous/statistical TDM

An asynchronous TDM is a technique in which time slots are not fixed as in the case of Synchronous TDM. Time slots are allocated to only those devices which have the data to send. Therefore, we can say that Asynchronous Time Division multiplexor transmits only the data from active workstations.

The time slots in asynchronous TDM are always less than the number of devices (transmitter). For example, if we have X devices and Y time slots. Y should always be less than X (I.e. Y < X).



In the above figure, it is shown that the number of devices are 4 and time slots are 3. The timeframe 1 (all slots) is completely filled with data from devices A, B, and C. The timeframe 1 has only 3 time-slots. So the data from device D is filled in the next timeframe (I.e. timeframe 2) in timeslot 1. The data from devices A and D will be filled in timeslots 2 and 3 in timeframe 2.

In asynchronous time division multiplexing, the multiplexer scans all the devices (transmitters) and accepts input only from the devices that have actual data to send and fills all the frames, and then sends it to the receiver.

Advantages of Time Division Multiplexing (TDM)

1. Full bandwidth is utilized by a user at a particular time.

2. The time division multiplexing technique is more flexible than frequency division multiplexing.

3. In time division multiplexing, the problem of crosstalk is very less.

Disadvantages of Time Division Multiplexing (TDM)

1. Synchronization is required in time division multiplexing.
2. Complex to implement.

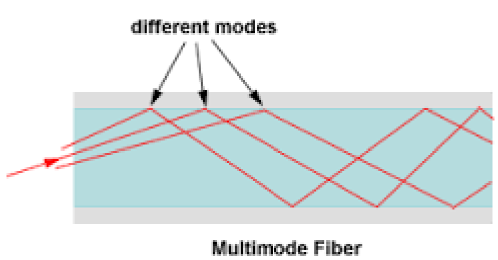
Space Division Multiplexing:

Space Division Multiplexing is a new technology for optical data transmission. Space Division Multiplexing is used in optical fiber cables to achieve higher data rate. In SDM, multiple spatial channels are created for transmission of different signals in a single fiber. This is achieved either by using a multimode fiber or by using a fiber having multiple cores.

→ Multicore fiber has several cores embedded in the fiber cladding. This causes crosstalk between them, which ultimately limits transmission performance. So the cores needs to be sufficiently far so that no crosstalk is possible.

Fig: multicore fiber

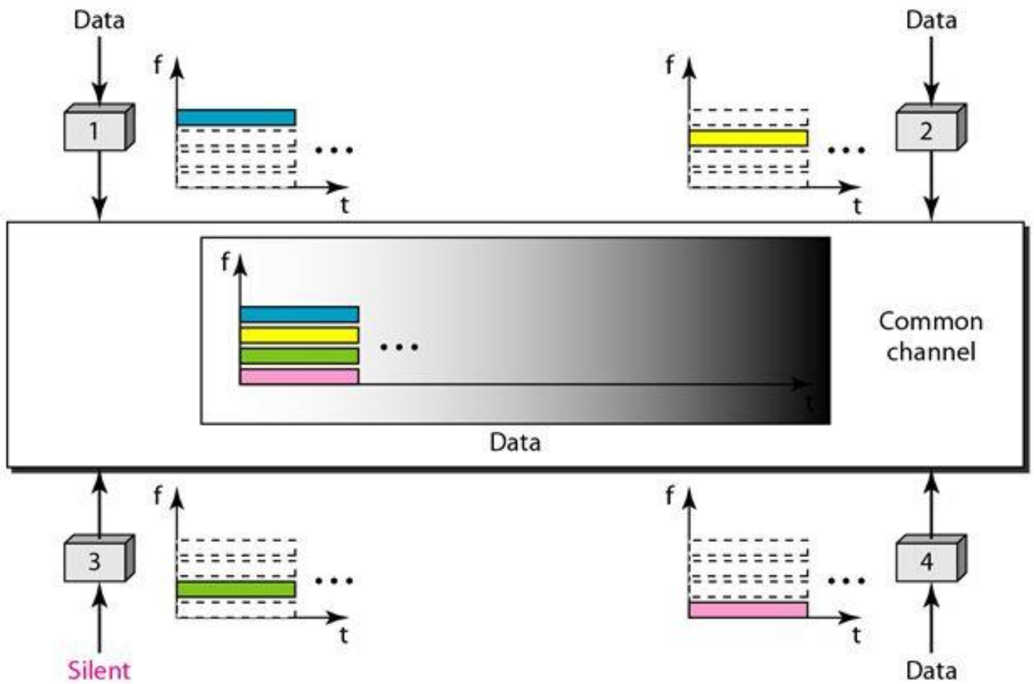
→In multi-mode fiber the propagation of several independent modes is done within a single core. The number of modes is determined by the core size and the refractive index of the fiber. Increasing the size of the allows for more modes to be supported within the fiber.



- → Only suitable for short distance transmission.

Frequency Division Multiple Access:

In frequency-division multiple access (FDMA), the available bandwidth is divided into frequency bands. Each station is allocated a band to send its data. In other words, each band is reserved for a specific station, and it belongs to the station all the time. To prevent station interferences, the allocated bands are separated from one another by small **guard bands.**



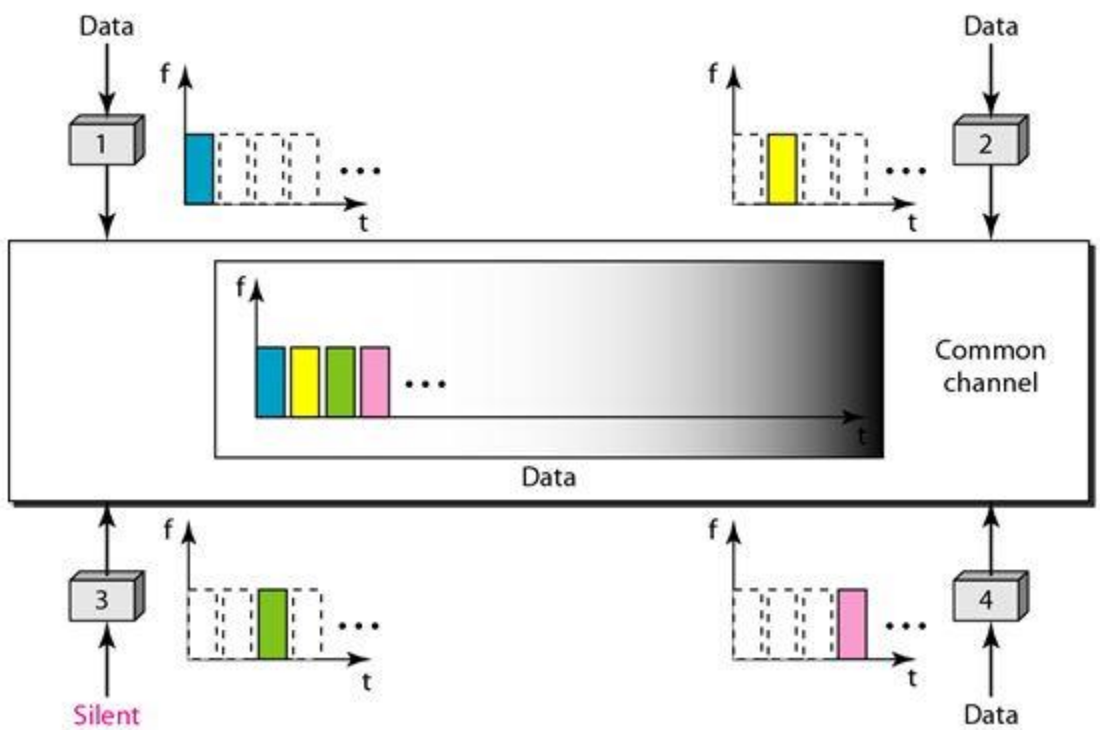
The differences between FDM and FDMA are as follows:

FDM, is a physical layer technique that combines the loads from low-bandwidth channels and transmits them by using a high-bandwidth channel. The channels that are combined are low-pass. The multiplexer modulates the signals, combines them, and creates a bandpass signal. The bandwidth of each channel is shifted by the multiplexer.

FDMA, on the other hand, is an access method in the data link layer. The data link layer in each station tells its physical layer to make a bandpass signal from the data passed to it. The signal must be created in the allocated band. There is no physical multiplexer at the physical layer. The signals created at each station are automatically bandpass-filtered. They are mixed when they are sent to the common channel.

Time-Division Multiple Access (TDMA):

In time-division multiple access (TDMA), the stations share the bandwidth of the channel in time. Each station is allocated a time slot during which it can send data. Each station transmits its data in is assigned time slot. The following figure shows the idea behind TDMA.



The main problem with TDMA lies in achieving synchronization between the different stations. Each station needs to know the beginning of its slot and the location of its slot. This may be difficult because of propagation delays introduced in the system if the stations are spread over a large area. To compensate for the delays, we can insert guard times. Synchronization is normally accomplished by having some synchronization bits at the beginning of each slot.

The differences between TDMA and TDM are :

• TDM is a physical layer technique that combines the data from slower channels and transmits them by using a faster channel. The process uses a physical multiplexer that interleaves data units from each channel.

• TDMA, on the other hand, is an access method in the data link layer. The data link layer in each station tells its physical layer to use the allocated time slot. There is no physical multiplexer at the physical layer.

Code-Division Multiple Access (CDMA):

CDMA simply means communication with different codes. CDMA differs from FDMA because only one channel occupies the entire bandwidth of the link. It differs from TDMA because all stations can send data simultaneously; there is no timesharing.

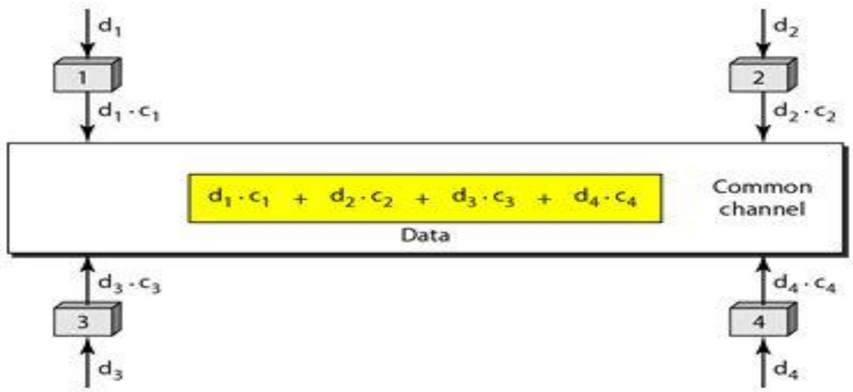
Implementation:

Let us assume we have four stations 1, 2, 3, and 4 connected to the same channel. The data from station 1 are d1 , from station 2 are d2, and so on. The code assigned to the first station is c1, to the second is c2, and so on. We assume that the assigned codes have two properties.

1. If we multiply each code by another, we get 0.

2. If we multiply each code by itself, we get 4 (the number of stations).

With these two properties in mind, how the above four stations can send data using the same common channel, as shown in the following figure.



Station 1 multiplies (a special kind of multiplication, as we will see) its data by its code to get d1.c1. Station 2 multiplies its data by its code to get d2.c2. And so on. The data that go on the channel are the sum of all these terms, as shown in the box.

Any station that wants to receive data from one of the other three multiplies the data on the channel by the code of the sender. For example, suppose stations 1 and 2 are talking to each other. Station 2 wants to hear what station 1 is saying. It multiplies the data on the channel by c1 the code of station1.
Because (c1.c1) is 4, but (c2 . c1), (c3. c1), and (c4 .c1) are all 0s, station 2 divides the result by 4 to get the data from station1.

data =(d1.c1+d2.c2+d3.c3+d4.c4).c1

= c1. d1. c1+ c1. d2. c2+ c1. d3. c3+ c1. d4. c4= 4d1

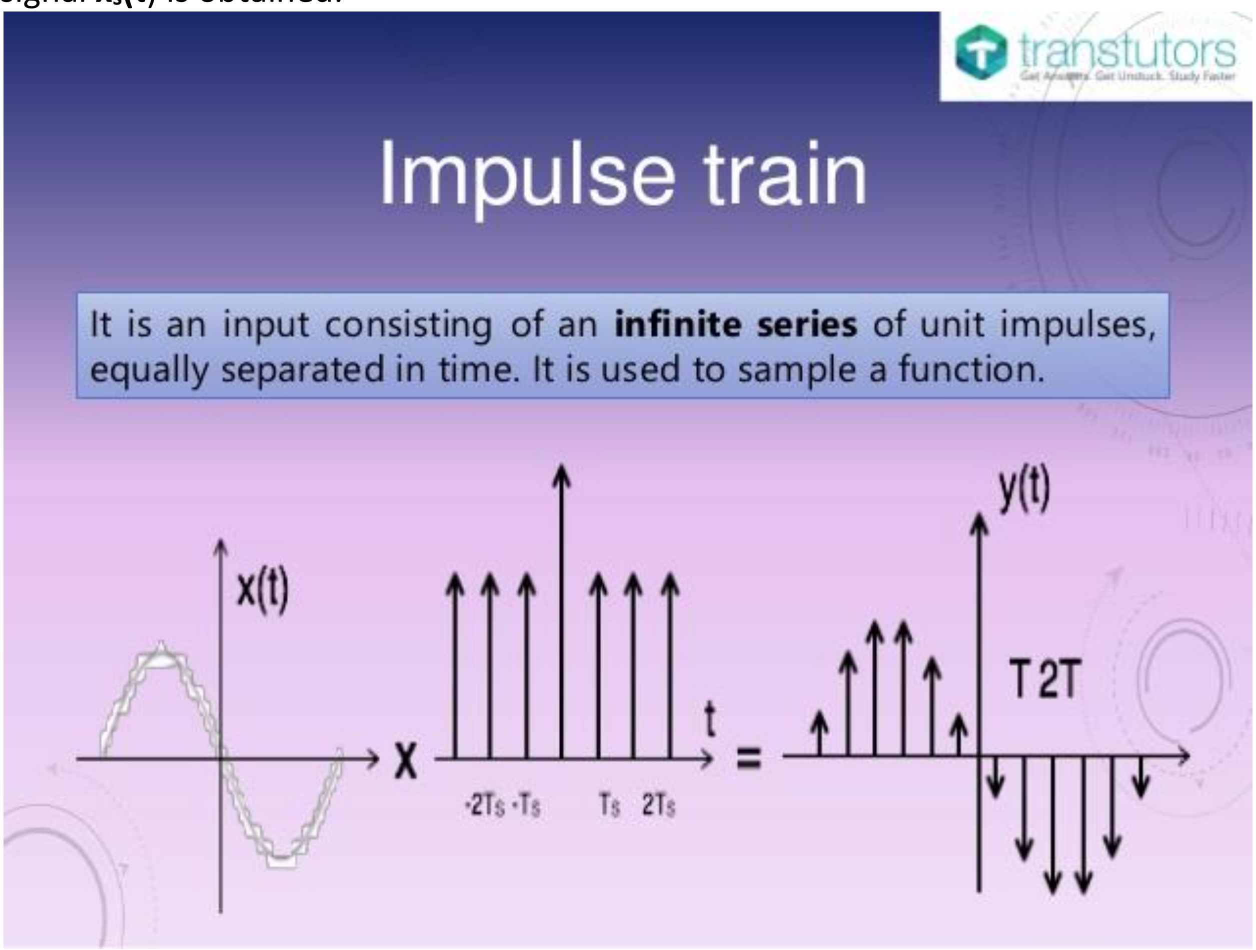
PULSE CODE MODULATION

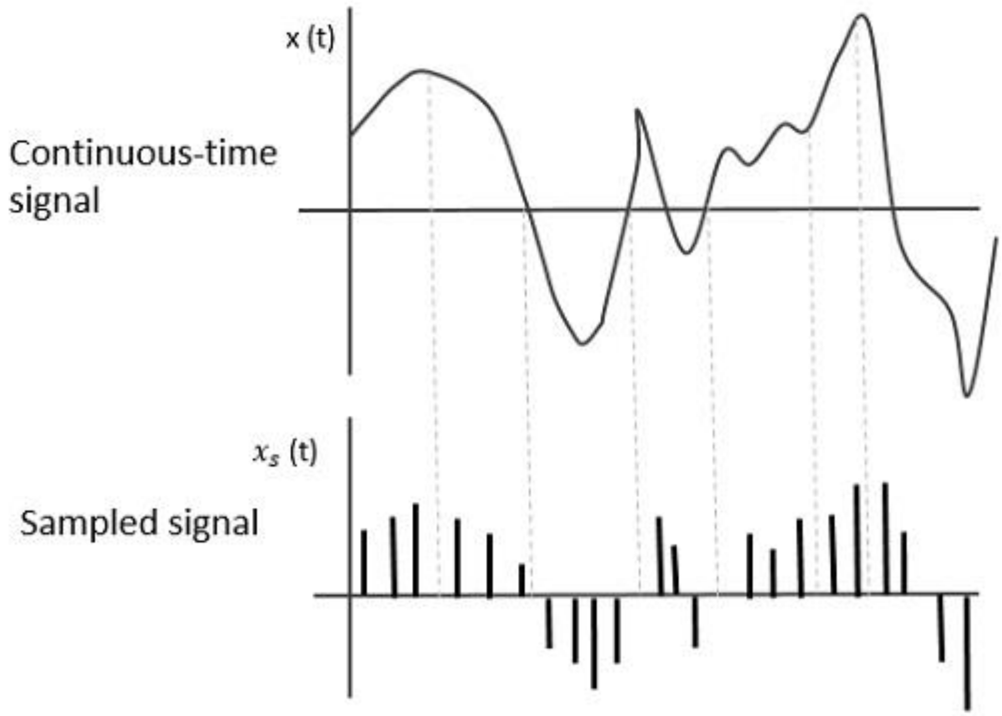
Sampling:

The process of converting continuous time signals into equivalent discrete time signals can be termed as **Sampling**.

When a source generates an analog signal and if that has to be digitized, having **1s** and **0s** i.e., High or Low, the signal has to be discretized in time. This discretization of analog signal is called as Sampling.

The following figure indicates a continuous-time signal **x** (t) and a sampled signal $\mathbf{x_s}$(t). When $\mathbf{x_s}$(t) is multiplied by a periodic impulse train, the sampled signal $\mathbf{x_s}$(t) is obtained.





A **sample** is a value or set of values at a point in time and/or space. So the **sampling time (period**) is the time difference between two consecutive samples in a sampled signal.

Hence, a **sampling rate** is defined as the number of samples produced per second. i.e. the reciprocal of time period is called as sampling rate(frequency).

$$Sampling\ Frequency = \frac{1}{T_s} = f_s$$

Where, T_S is the sampling period and f_S is the sampling frequency.

- ➔ For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as **Nyquist rate.**
- ➔ For this Nyquist gave a theorem which is also called as sampling theorem. This theorem states that "**a signal can be exactly reproduced if it is**

sampled at the rate f_s which is greater than twice the maximum frequency f_m".

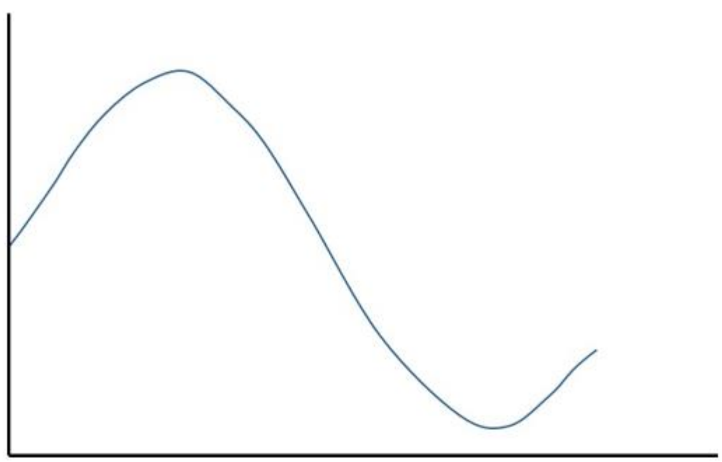
i.e.

$$f_s \geq 2f_m$$

Quantization:

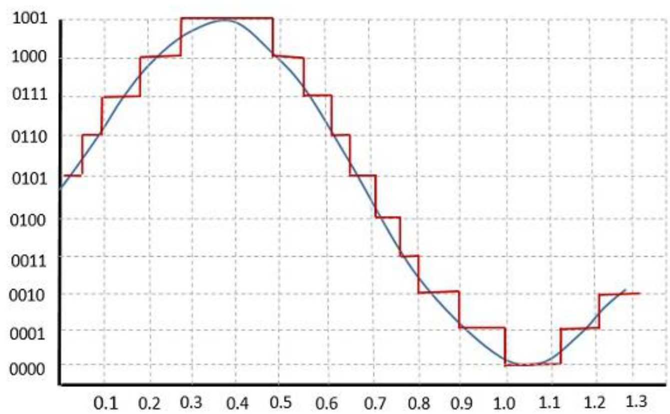
Quantization is the process in which the continuous range of values of analog signal is sampled into discrete values and these discrete values are joined to round off the value to a near stabilized value. The final values are the approximately equal to the analog values.

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.



The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. **Quantization** is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

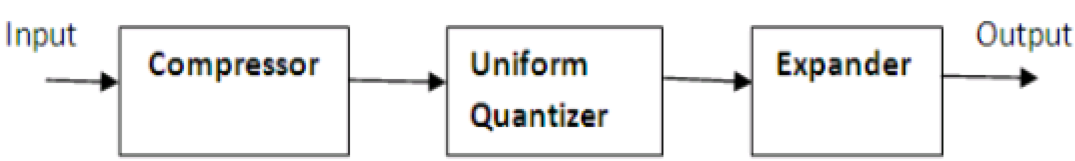
The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.

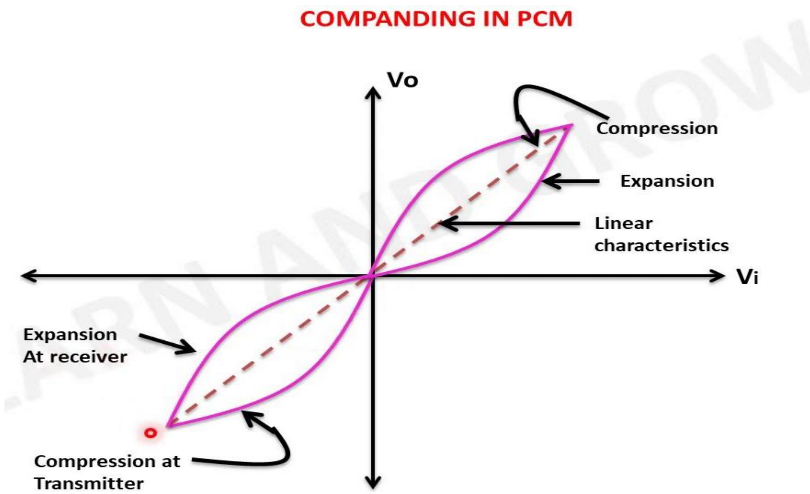


Companding:

Companding refers to a technique for compressing and then expanding (or decompressing) an analog or digital signal. Data is compressed before being transmitted. Then, it is expanded at the receiving end using the same non-linear scale to restore it to its original form.

- ➔ Companding is used as a complement to the process of modulation and demodulation. In this process a signal is compressed, then changed from analog to digital, transmitted and converted back from digital to analog before it is expanded again.
- ➔ The received signal will have reduced noise and crosstalk levels.





The compressor will compress the dynamic range of the signal so that high dynamic range signal can be passed through components of low dynamic range capability, the uniform quantizer will undergo the quantization process of the compressed signal and the lastly the expander will undergo expansion and invert the compression function to reconstruct the original signal.

Two types:

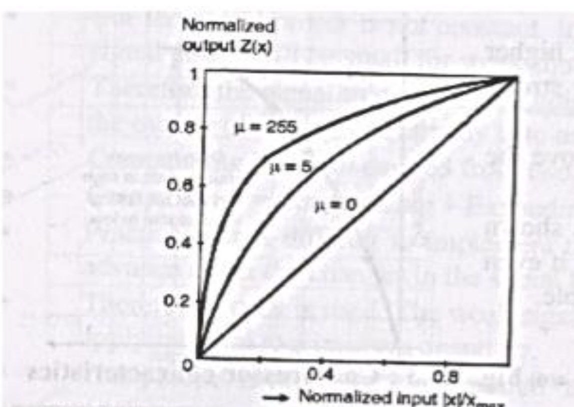
➔ Mu law:

→here, compressor characteristic is continuous.

→approx. linear for smaller values of input and logarithmic for higher value of input levels.

→mu law companding is used for speech and music signals.

→also used for PCM telephone systems in U.S , Canada and Japan (where T1 circuits are used).



(a) Compressor characteristic of a μ-law compressor

➔ A-law:

→An **A-law algorithm** is a standard companding algorithm, used in European 8-bit PCM digital communications systems to optimize, i.e. modify, the dynamic range of an analog signal for digitizing.
→here, compressor characteristic is piece-wise made up of linear segment for low input and levels and logarithmic segment for high level inputs.
→used for telephone systems elsewhere.(European standard where E-1 circuits are used.)

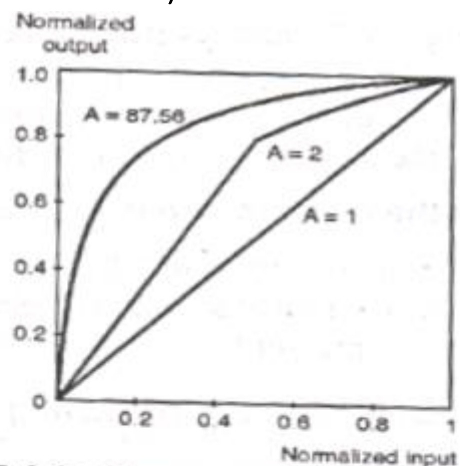
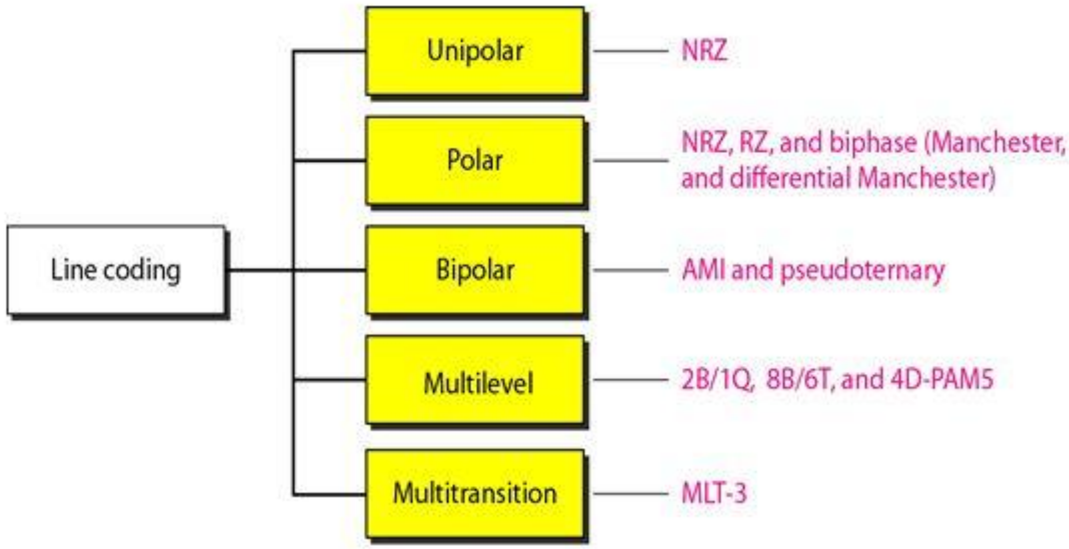


Fig. 7.6.6 : Compressor characteristics of A-law compressor

Line Coding:

Line coding is the process of converting digital data to digital signals. By this technique we convert a sequence of bits to a digital signal. At the sender side, digital data are encoded into a digital signal and at the receiver side the digital data are recreated by decoding the digital signal.

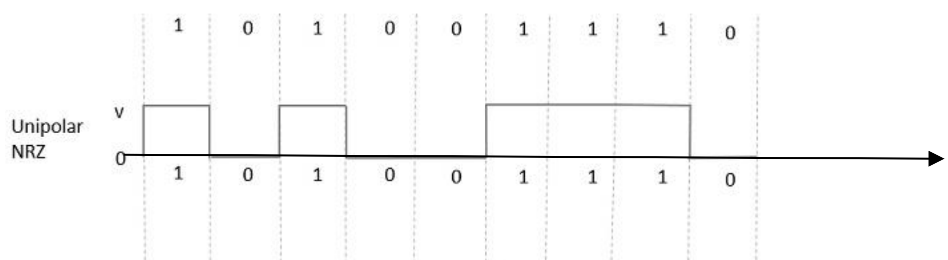
We assume that data, in the form of text, numbers, graphical images, audio, or video, are stored in computer memory as sequences of bits. Line coding converts a sequence of bits to a digital signal. Line coding scheme can be categorized into following five categories:



Unipolar codes:

Unipolar codes use only one voltage level other than zero. Hence, the encoded signal will have either +V voltage value or 0.

<u>Non return to zero (NRZ)</u> – It is unipolar line coding scheme in which positive voltage defines bit 1 and the zero voltage defines bit 0. Signal does not return to zero at the middle of the bit thus it is called NRZ. For example:

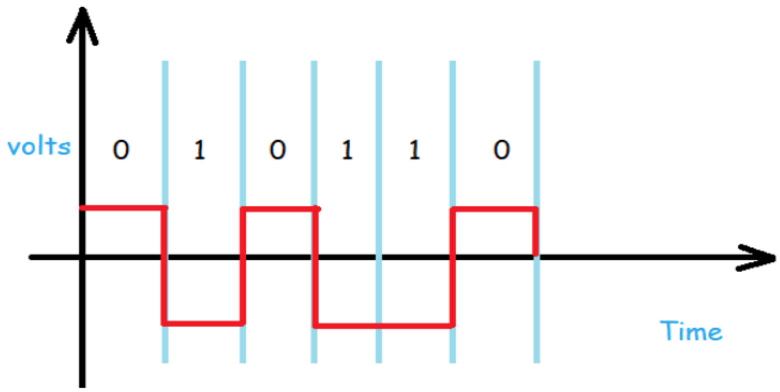


Polar schemes –

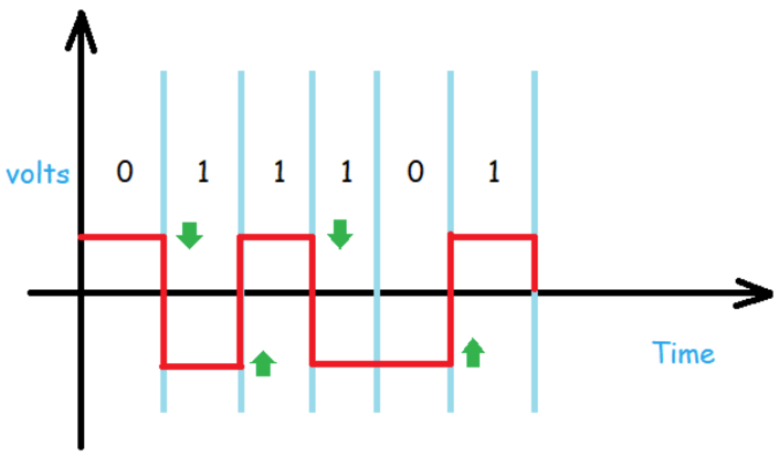
In polar schemes, the voltages are on the both sides of the time axis. For example, the voltage level for 0 can be positive and the voltage level for 1 can be negative.

<u>Non-Return-to-Zero (NRZ):</u> In polar NRZ encoding, we use two levels of voltage amplitude. We can have two versions of polar NRZ: NRZ-L (level) and NRZ-I (Invert).

Let's see how these are represented:



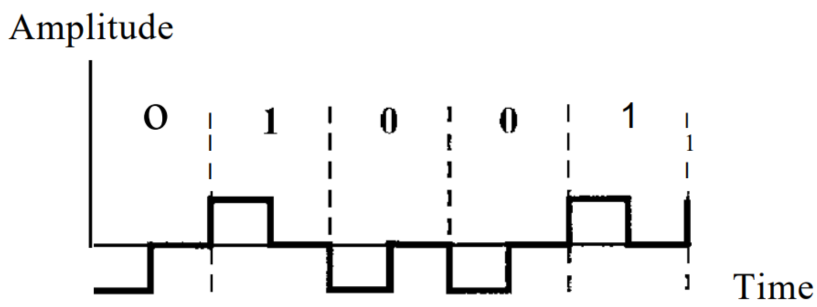
In the above diagram, we can simply notice that high volt is for logical 0 and low volt is for logical 1. This is the representation of **NRZ-Level**.



Now in this one, the idea is that whenever we encounter logical 1 then the signal will be inverted, but when it encounters logical 0 then it remains on the same side. This is the **NRZ-Invert**.

<u>Return to Zero (RZ):</u>

Return to zero proved out to be a nice alternative or say a solution to NRZ drawbacks. Unlike NRZ, RZ uses three values of voltage i.e. positive, negative, zero. And as the name suggests it returns to zero in the middle of each bit.

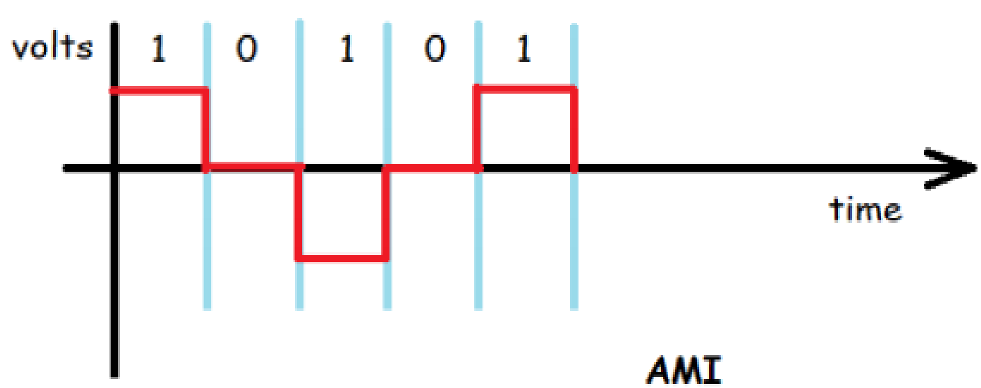


The idea behind the above representation is that logical 1 is represented as half-positive and half-zero volts and the logical 0 is represented as half negative and half-zero volts.

Bipolar Line Coding

Bipolar consists of three voltage levels which are positive negative and zero. While representing, the voltage level for one bit of data is at zero, and the other bit inverts of transits or alternates between positive and negative voltage.

ALTERNATE MARK INVERSION (AMI): the representation here follows a simple logic that is, for representing logical 0 we use zero voltage, and while representing logical 1 we use alternating positive and negative voltages, which can be seen in the image below.



HDB3:

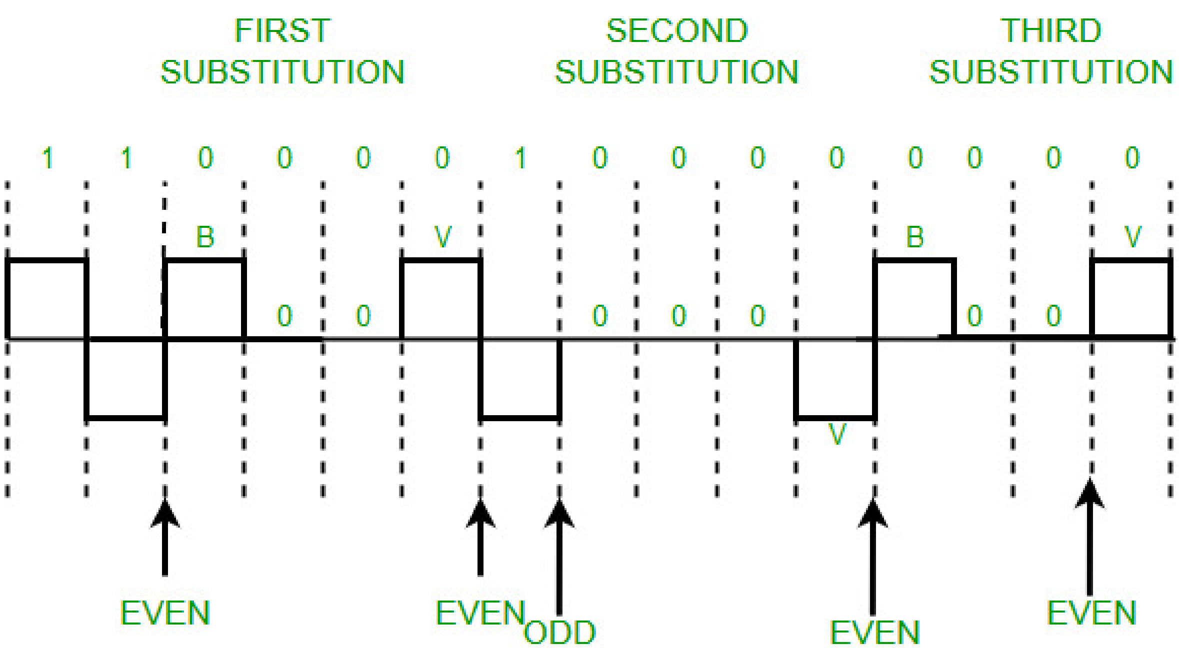
High-density bipolar 3-zero (HDB3) is based on Alternate Mark Inversion (AMI), but extends this by inserting violation codes whenever there is a run of 4 or more 0's.

The two rules can be stated as follows:

1. If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be OOOV, which makes the total number of nonzero pulses even.

2. If the number of nonzero pulses after the last substitution is even, the substitution pattern will be BOOV, which makes the total number of nonzero pulses even.

For example:

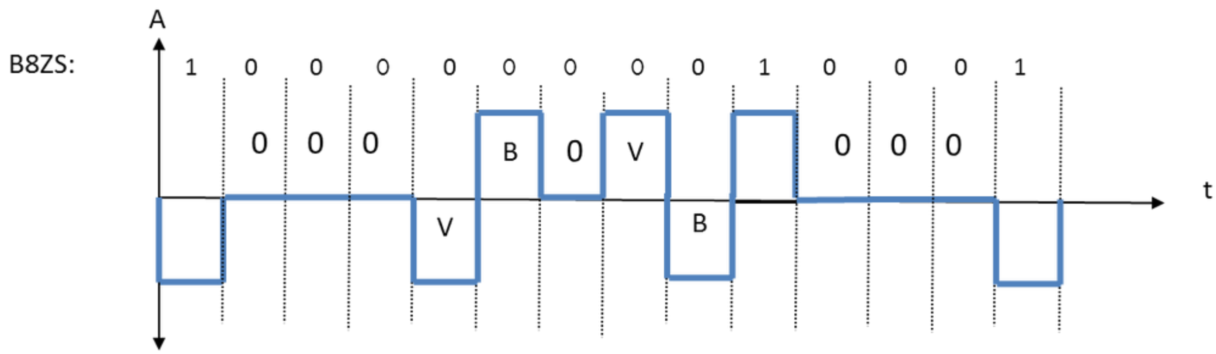


Note: HDB3 substitutes four consecutive zeros with OOOV or BOOV depending on the number of nonzero pulses after the last substitution.

B8ZS:

Binary 8 Zero Substitution (B8ZS) works in a similar way to AMI by changing poles for each binary 1. Moreover when it found 8 consecutive 0's it replace them by '000VB0VB'. Here, V means violation (same to previous non-zero bit) and B means Balancing (opposite of previous non-zero bit). I.e. The V means the same polarity as the polarity of the previous nonzero pulse; B means the polarity opposite to the polarity of the previous nonzero pulse.

For example:



Modulation:

Modulation is the process of varying one or more parameters of a carrier signal before being transmitted through the transmission media.

Pulse Code Modulation (PCM)

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). i.e. a signal is pulse code modulated to convert its analog information into a binary sequence, i.e., **1s** and **0s**. The output of a PCM will resemble a binary sequence.

A PCM encoder has three processes, as shown in Figure below:

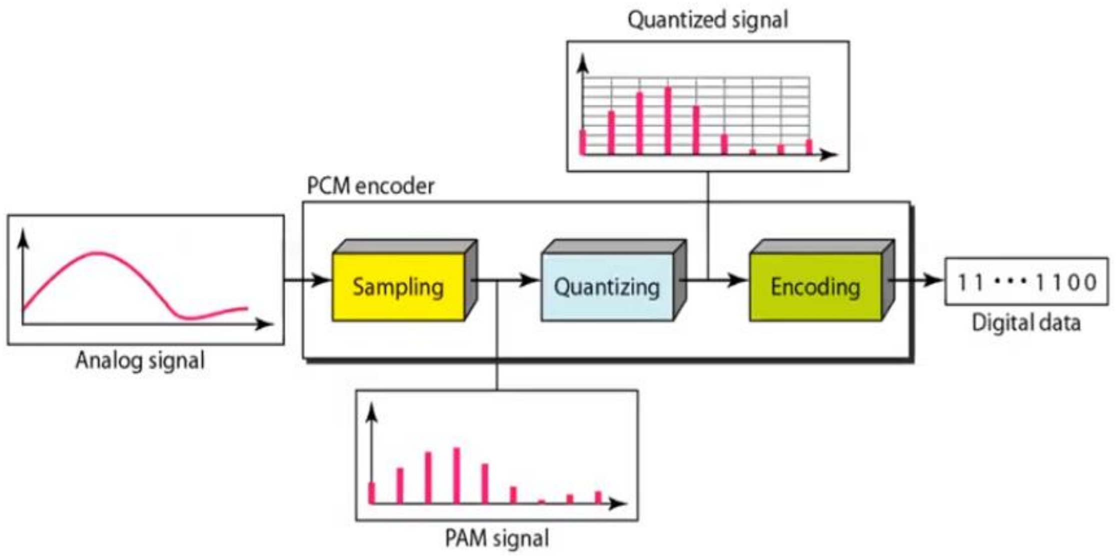


Fig: PCM Generation

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

Sampling:

The first step in PCM is sampling. The analog signal is sampled every *Ts* time, where *Ts* is the sample interval or period. The inverse of the sampling interval is called the sampling rate or sampling frequency and denoted by *fs,* where *fs = 1/Ts.*

According to the Nyquist theorem, to reproduce the original analog signal, one necessary condition is that the sampling rate be at least twice the highest frequency in the original signal.

Quantization:

The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal. The set of amplitudes can be

infinite with nonintegral values between the two limits. These values cannot be used in the encoding process. The following are the steps in quantization:
1. We assume that the original analog signal has instantaneous amplitudes between Vmin and Vmax'

2. We divide the range into L zones, each of height Δ (delta)

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

3. We assign quantized values of 0 to L - I to the midpoint of each zone.

4. We approximate the value of the sample amplitude to the quantized values.

Quantization Error

Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error.

Encoder

The last step in PCM is encoding. The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code and each samples can be changed to binary code.

PCM transmission format (E1 and T1):

- ➔ E1 and T1 are digital telecommunication carrier standards, initially developed in different continents to carry voice conversations simultaneously using time division multiplexing. Both standards use two pairs of wires for transmit and receive paths to achieve full duplex communication.

- ➔ Both E1 and T1 carrier methods are initially developed to transmit and receive pulse code modulated (PCM) voice over time multiplexed copper wires.
- ➔ E1 is an European standard which comprises of 32 simultaneous channels, while T1 is North American standard which consists of 24 simultaneous channels. Each channel is capable of transmitting 64kbps of data.
- ➔ Data rate of E1 is 2 Mbps, while the data rate of T1 is 1.544 Mbps.

T1 frame format used in PCM telephone system

The following figure depicts **T1-frame structure** used in PCM based telephone systems. As shown, T1 system multiplexes 24 voice channels into a single line using TDM techniques. Each analog voice channel is sampled at 8KHz rate. In other words, analog waveform is sampled at every 125 µS.

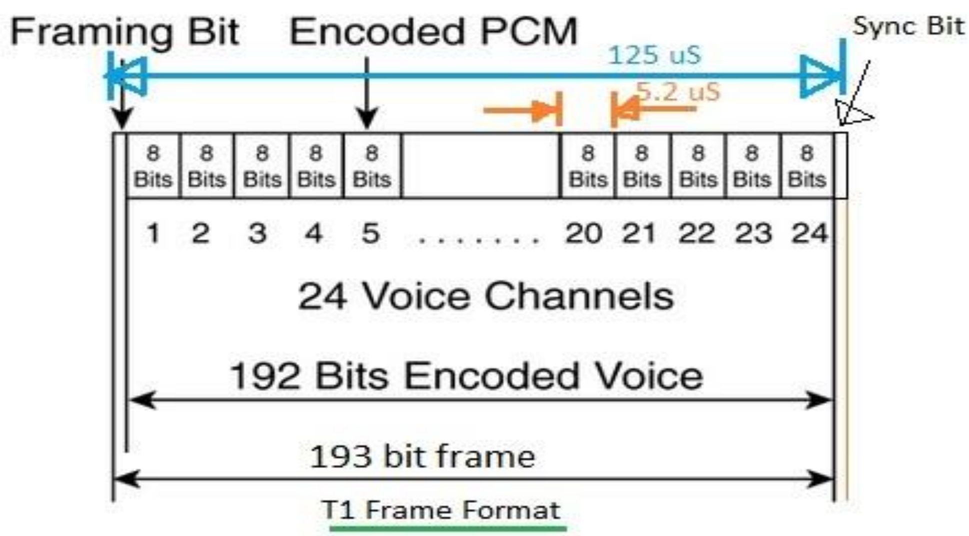


Fig: T1 Frame Format

E1 frame format used in PCM telephone system

The following figure depicts E1-frame structure. E1 frame takes 8 bits from each of the 64 kbps channels one by one. It retransmits these channels at 2048 kbps. E1 frame accomodates 30 voice channels including two additional 64 kbps channels. These two additional channels are used to transmit synchronization, service and signaling data. This results into E1 transmission rate of 2048 kbps = 32 * 64 kbps.

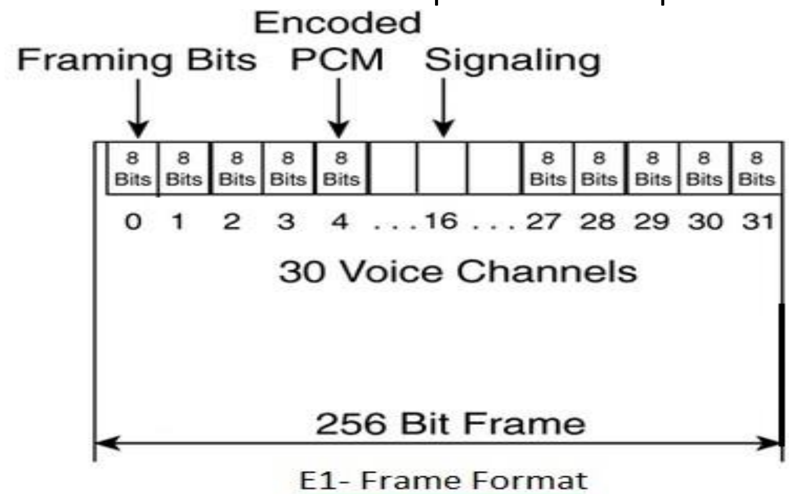


Fig: E1-Frame Format

Switching Techniques

Switching:

Switching is the process of forwarding message (data packets) from one node to another so that it can reach to the destination. A switched network consists of a series of interlinked nodes, called switches. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (e.g. computers or telephones). Others are used only for routing.

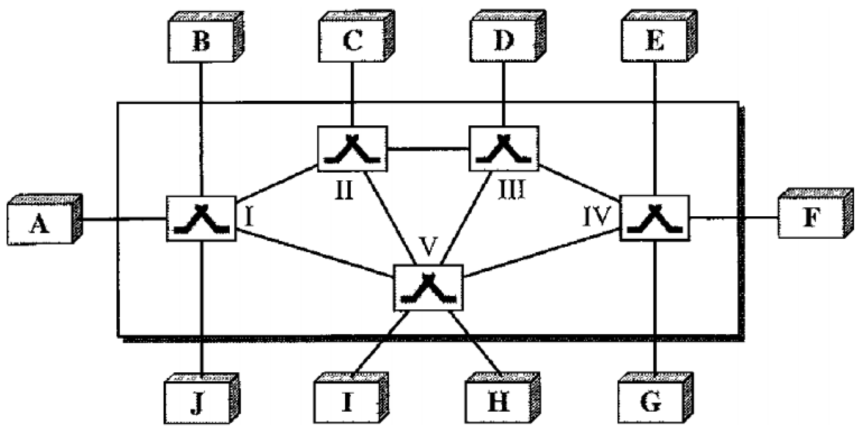


Fig: Switched Network

There are multiple switching techniques:

1. **<u>Circuit switching:</u>**
   - ➔ In circuit switching network, a communications path between end devices (nodes) needs to be set up before they can communicate.
   - ➔ There is a need of pre-specified route from which data will travel and no other data is permitted.
   - ➔ It was designed for voice applications. Telephone is the best suitable example of circuit switching.

Phases of Circuit Switch Connection

- **Circuit Establishment**: In this phase, a dedicated circuit is established from the source to the destination through a number of intermediate switching centers. The sender and receiver transmits communication signals to request and acknowledge establishment of circuits.
- **Data Transfer**: Once the circuit has been established, data and voice are transferred from the source to the destination. The dedicated connection remains as long as the end parties communicate.
- **Circuit Disconnection**: When data transfer is complete, the connection is terminated. The disconnection is initiated by any one of the user. Disconnection involves removal of all intermediate links from the sender to the receiver.

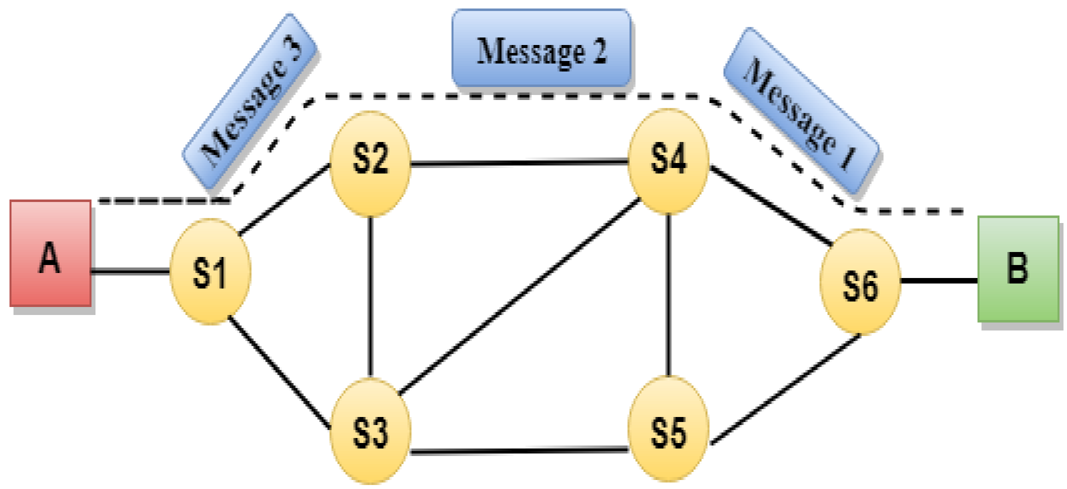


Fig: circuit switched Network.

Advantages:

- It is suitable for long continuous transmission, since a continuous transmission route is established, that remains throughout the conversation.
- The dedicated path ensures a steady data rate of communication.
- No intermediate delays are found once the circuit is established. So, they are suitable for real time communication of both voice and data transmission.

Disadvantages:

- Bandwidth requirement is high even in cases of low data volume.
- There is underutilization of system resources. Once resources are allocated to a particular connection, they cannot be used for other connections.
- Time required in establishing a connection may be high.

2. **Message switching:**

Message switching was a technique developed as an alternate to circuit switching, before packet switching was introduced. **In message switching, end users communicate by sending and receiving *messages* that included the entire data to be shared.** Messages are the smallest individual unit.

Also, the sender and receiver are not directly connected. There are a number of intermediate nodes which transfer data and ensure that the message reaches its destination.

→In Message Switching technique, there is no establishment of a dedicated path between the sender and receiver.

Message switching is performed in 2 steps:

Store and forward – The intermediate nodes have the responsibility of transferring the entire message to the next node. Hence, each node must have storage capacity. A message will only be delivered if the next hop and the link connecting it are both available. A store-and-forward switch forwards a message only if sufficient resources are available and the next hop is accepting data.

Message delivery – This implies the entire information is transferred in a single message from source to the destination node. Each message must have a ***header*** that contains the message routing information, including the source and destination.

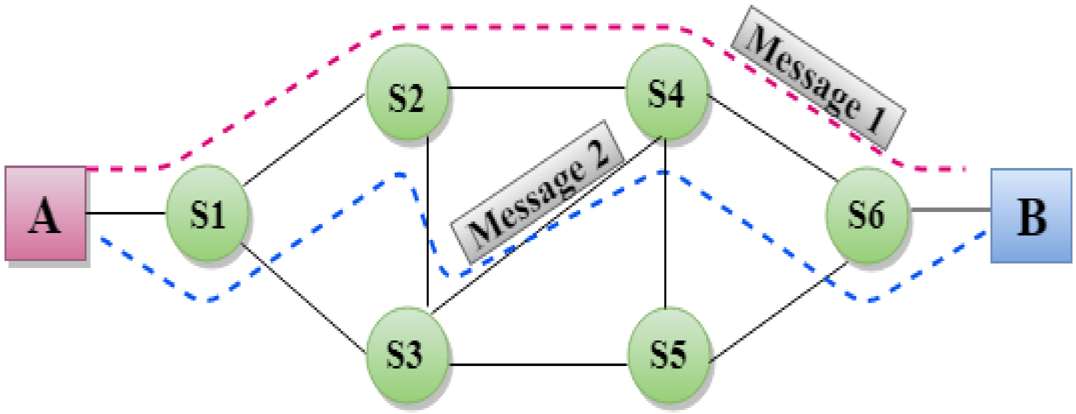


Fig: Message switching network

Advantages:

- Data channels are shared among the communicating devices that improve the efficiency of using available bandwidth.
- Traffic congestion can be reduced because the message is temporarily stored in the nodes.
- Message priority can be used to manage the network.

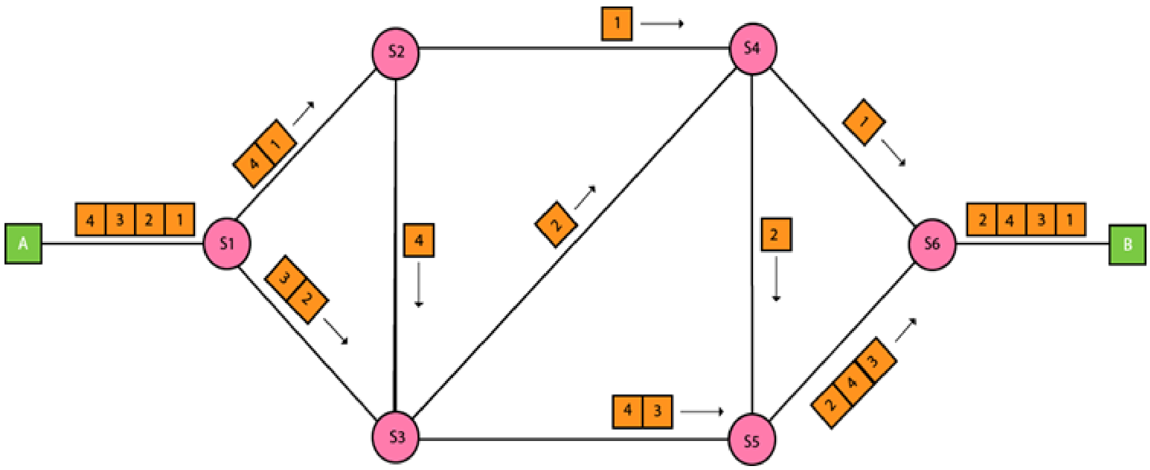
Disadvantages:

- The message switches must be equipped with sufficient storage to enable them to store the messages until the message is forwarded.
- The Long delay can occur due to the storing and forwarding facility provided by the message switching technique.

Packet Switching

- The packet switching is a switching technique in which the message is sent in one go, but it is divided into smaller pieces, and they are sent individually.
- The message splits into smaller pieces known as packets and packets are given a unique number to identify their order at the receiving end.
- Every packet contains some information in its headers such as source address, destination address and sequence number.
- Packets will travel across the network, taking the shortest path as possible.

- All the packets are reassembled at the receiving end in correct order.
- If any packet is missing or corrupted, then the message will be sent to resend the message.
- If the correct order of the packets is reached, then the acknowledgment message will be sent.



Advantages Of Packet Switching:

- **Cost-effective:** In packet switching technique, switching devices do not require massive secondary storage to store the packets, so cost is minimized to some extent. Therefore, we can say that the packet switching technique is a cost-effective technique.
- **Reliable:** If any node is busy, then the packets can be rerouted. This ensures that the Packet Switching technique provides reliable communication.
- **Efficient:** Packet Switching is an efficient technique. It does not require any established path prior to the transmission, and many users can use the same communication channel simultaneously, hence makes use of available bandwidth very efficiently

Disadvantages Of Packet Switching:

- Packet Switching technique cannot be implemented in those applications that require low delay and high-quality services.
- The protocols used in a packet switching technique are very complex and requires high implementation cost.

Two approaches for Packet switching:

1) **Datagram Packet Switching (Connectionless packet switching)**

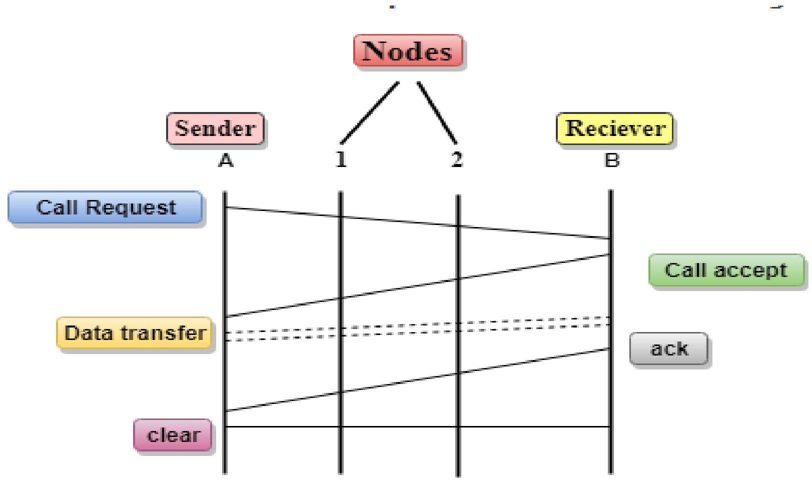
    - It is a packet switching technology in which packet is known as a datagram, is considered as an independent entity. Each packet contains the information about the destination and switch(router) uses this information to forward the packet to the correct destination.
    - The packets are reassembled at the receiving end in correct order.
    - In Datagram Packet Switching technique, the path is not fixed.
    - Intermediate nodes take the routing decisions to forward the packets.

2) **Virtual Circuit Switching (Connection oriented packet switching)**

    - In the case of Virtual circuit switching, a preplanned route is established before the messages are sent.
    - the data stream is transferred over a packet switched network, in such a way that it seems to the user that there is a dedicated path from the sender to the receiver.
    - **Call request** and **call accept** packets are used to establish the connection between sender and receiver.
    - In this case, the path is fixed for the duration of a logical connection.

Lets understand through a diagram below:

- In the above diagram, A and B are the sender and receiver respectively. 1 and 2 are the nodes.
- **Call request** and call **accept packets** are used to establish a connection between the sender and receiver.



- When a route is established, data will be transferred.
- After transmission of data, an acknowledgment signal is sent by the receiver that the message has been received.
- If the user wants to terminate the connection, a clear signal is sent for the termination.

Switching systems Classification

- **Manual**
- **Automatic**
    - **Electromechanical**
    - **Electronic**

<u>Manual switching:</u>

From the earliest days of the telephone, it was observed that it was more practical to connect different telephone instruments by running wires from each instrument to a central switching point, or **telephone exchange**, than it was to run wires between all the instruments.

An exchange consists of electronic component (switchboard) and human operator that interconnect telephone subscriber line of digital system to establish telephone call between subscribers. Earlier 21 customers are permitted to reach one another by means of a manually operated central switchboard. The manual switchboard was quickly extended from 21 lines to hundreds of lines. Each line was terminated on the switchboard in a socket (called a jack), and a number of short, flexible circuits (called cords) with a plug on both ends of each cord were also provided. Thus, two lines can be interconnected by inserting the two ends of a cord in the appropriate jacks.

Limitations:

→ In a manual exchange, the subscriber needs to communicate with the operator and the operator will communicate with another subscriber through the switchboard which is time consuming.

→ Sufficient privacy is not there as an operator is involved in setting up and monitoring the calls.

Electromechanical Switching:

The Electromechanical switching systems are a combination of mechanical and electrical switching types. Here mechanical switches are electrically operated. The Manual Switching system requires an operator who after receiving a request, places a call. Here, the operator is the sole in-charge for establishing or releasing the connections.

The idea of automatic switching appeared as early as 1879, and the first fully automatic switch to achieve commercial success was invented in 1889 by Almon B. Strowger.

Electronic switching:

The Electronic Switching systems are operated with the help of a processor or a computer which control the switching timings. The instructions are programmed and stored on a processor or computer that control the operations. This method of storing the programs on a processor or computer is called the **Stored Program Control (SPC)** technology. New facilities can be added to a **SPC** system by changing the control program.