CHAPTER-ONE: INTRODUCTION

What is telecommunication?

- Telecommunications are **the means of electronic transmission of information over distances**.
- The information may be in the form of voice telephone calls, data, text, images, or video. Today, telecommunications are used to organize more or less remote computer systems into telecommunications networks.

PSTN (Public Switch Telephone Network)

- 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.

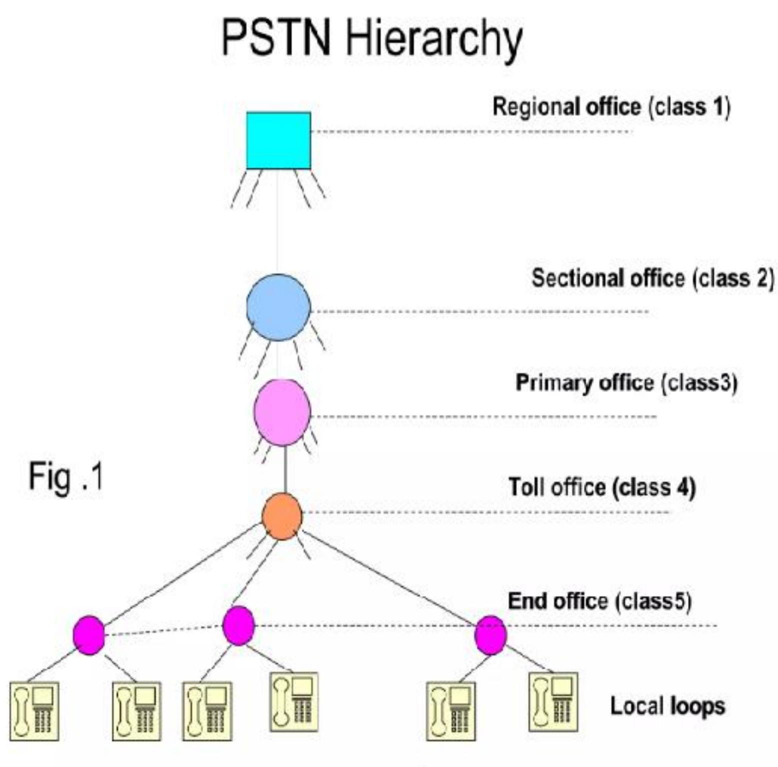
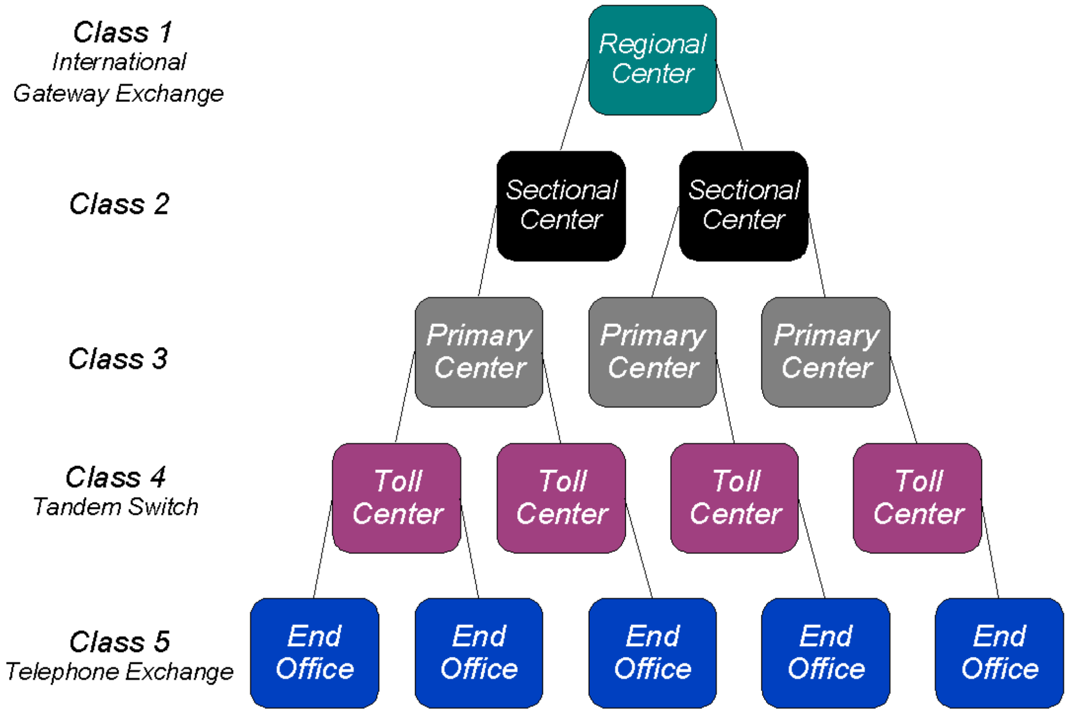


Fig .1



Class 1 (regional center)

The class 1 office was the Regional Center (RC). Regional centers served three purposes in the North American toll network

(a) their connections were the "last resort" for final setup of calls when routes between centers lower in the hierarchy were not available.

(b) they were initially staffed by engineers who had the authority to block portions of the network within the region in case of emergencies or network congestion - although these functions were transferred after 1962 to the Network Control/Operations Center and the distributed Network Management Centers.

(c) they provided collection points (until the development of more advanced computer hardware and software for toll operators) for circuits that would be passed along to one of the international overseas gateways (which operated as special centers outside the formal North American hierarchy).

The regional centers updated each other on the status of every circuit in the network. These centers would then reroute traffic around the trouble spots and keep each informed at all times.

Class 2 (sectional center)

The class 2 office was the Sectional Center (SC). The sectional center typically connected major toll centers within one or two states or provinces, or a significant portion of a large state or province, to provide interstate or interprovincial connections for long-distance calls. At various times, there were between 50 and 75 active class two offices in the network.

Class 3 (primary center)

The class 3 office was the Primary Center (PC). Calls being made beyond the limits of a small geographical area where circuits are not connected directly between class 4 toll offices would be passed from the toll center to the primary center. These locations use high usage trunks to complete connection between toll centers. The primary center never served dial tone to the user. The number of primary centers in the network fluctuated from time to time, ranging between 150 and 230.

Class 4 (toll center (tandem exchange))

The class 4 office is the Toll Center (TC), Toll Point (TP), or Intermediate Point (IP). A call going between two end offices not directly connected, or whose direct trunks are busy, is routed through the toll center. The toll center is also used to connect to the long-distance network for calls where added costs are incurred, such as operator handled services. This toll center may also be called the tandem office because calls have to pass through this location to get to another part of the network. Toll centers might have been operated either as interstate facilities, under the operation of AT&T Long Lines (GTE in a few cases), or by local telephone companies, handling long-distance traffic to points within a particular operating company territory. Class 4 offices continue to exist, although with considerable changes, as they handle local exchange company interconnections, locally charged or long-distance rated, or provide facilities for connection to long-distance company points of presence.

- Also known as a junction network, a tandem office serves a large geographical area comprising several local exchanges while managing switches between local exchanges.

- Let’s say you dialed the number of a client who lives in the same city but in another suburb. In this case your call will be routed to a tandem office from your local exchange, and the tandem office will route the signal on to the local exchange near your client’s location.

Class 5 (local exchange)

The class 5 office is the local exchange or end office. It delivers dial tone to the customer. The end office, also called a branch exchange, is the closest connection to the end customer. Over 19,000 end offices in the United States alone provide basic dial tone services.

- A local exchange – which may consist of one or more exchanges – hooks up subscribers to a PSTN line. Also known as a central office or a switching exchange, a telephone exchange may have as many as 10,000 lines.

- All telephones are connected to the local exchange in a specific area. Interestingly, if you were to dial the number of your supplier located in the building next to yours, the call won't leave your local exchange and will be routed to the supplier as soon as it reaches the exchange.
- The exchange then identifies the number dialed so it can route the call towards the correct end destination. This process works as follows.

- The first three digits of a phone number represent the exchange (the local switch), while the last four digits identify the individual subscriber within that exchange.

- This means that when you dial a number and it reaches your local exchange, your call is immediately linked to the subscriber without the need for any further routing.

Network Topology

- A network topology is the physical and logical arrangement of nodes and connections in a network.
- Nodes usually include devices such as switches, routers and software with switch and router features.
- Network topologies are often represented as a graph.

Types of Network Topology are:

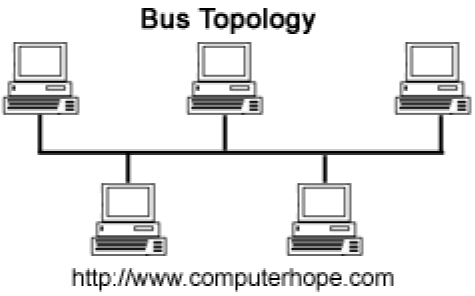
Comparison Chart

BASIS FOR COMPARISON	PHYSICAL TOPOLOGY	LOGICAL TOPOLOGY
Basic	Refer to how a network look and functions.	Fashion in which data travels logistically.
Types	Bus, star, ring and mesh topologies.	Logical bus and the logical ring.
Founded on	Physical connections of cables and devices.	Path traveled by data in a network.
Can affect	Cost, scalability, flexibility, bandwidth capacity, etcetera.	Data delivery causing lost packets or congestion.

- **A logical topology is how devices appear connected to the user. A physical topology is how they are actually interconnected with wires and cables**.

Physical topology

Bus Topology:



- **Bus topology** is a network setup where each computer and network device is connected to a single cable or backbone.
- Bus topology, also known as line topology, is **a type of network topology in which all devices in the network are connected by one central RJ-45 network cable or coaxial cable**.

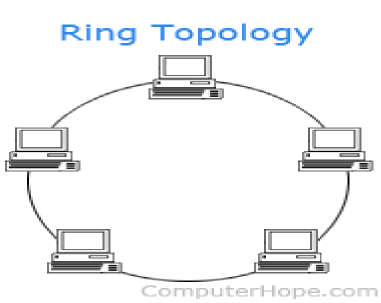
Advantages:

- It works well when you have a small network.
- It's the easiest network topology for connecting computers or peripherals in a linear fashion.
- It requires less cable length than a star topology.

Disadvantages:

- It can be difficult to identify the problems if the whole network goes down.
- It can be hard to troubleshoot individual device issues.
- Bus topology is not great for large networks.
- Terminators are required for both ends of the main cable.
- Additional devices slow the network down.
- If a main cable is damaged, the network fails or splits into two.

Ring Topology:



- A **ring topology** is a network configuration where device connections create a circular data path.
- Each networked device is connected to two others, like points on a circle. Together, devices in a ring topology are referred to as a **ring network**.

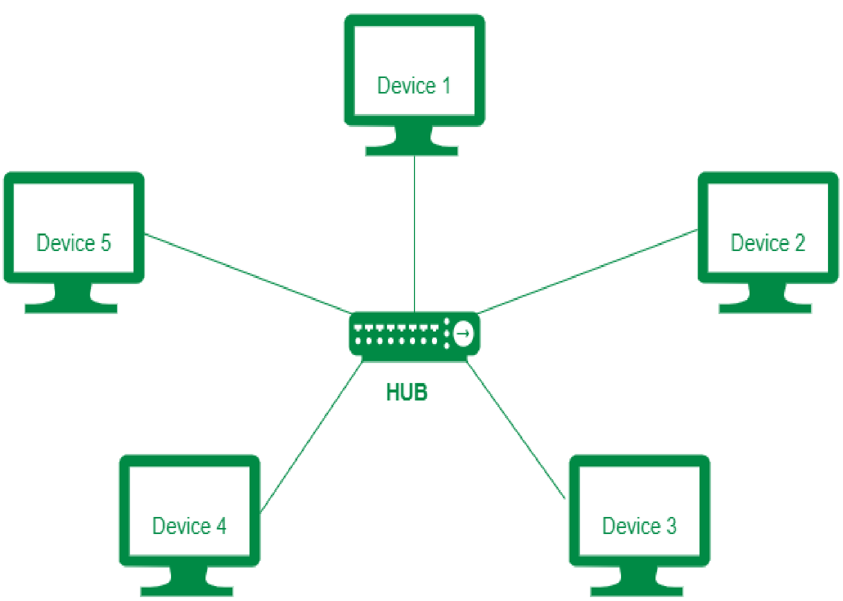
Advantages of a ring topology

- All data flows in one direction, reducing the chance of packet collisions.
- A network server is not needed to control network connectivity between each workstation.
- Data can transfer between workstations at high speeds.
- Additional workstations can be added without impacting performance of the network.

Disadvantages of a ring topology

- All data being transferred over the network must pass through each workstation on the network, which can make it slower than a star topology.
- The entire network will be impacted if one workstation shuts down.
- The hardware needed to connect each workstation to the network is more expensive than Ethernet cards and hubs/switches.

Star topology:



Star Topology

- Star topology is a network topology in which each network component is physically connected to a central node such as a router, hub or switch.
- The central network device acts as a server, and the peripheral devices act as clients.

Advantages:

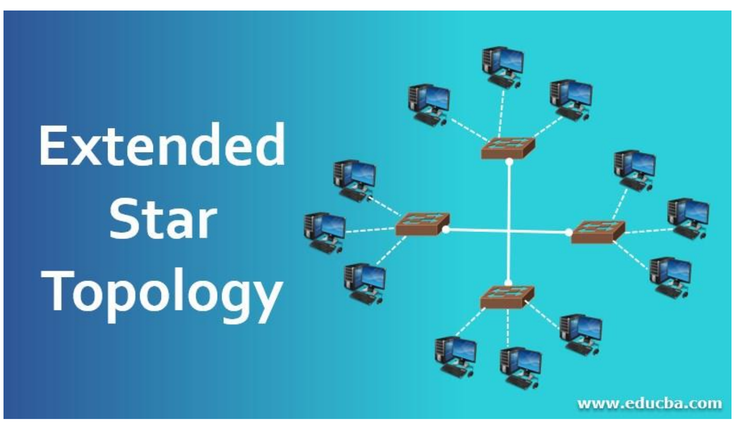
- It is very reliable – if one cable or device fails then all the others will still work
- It is high-performing as no data collisions can occur
- Easier to put in
- Robust in nature

Disadvantage:

- Requires more cable than a linear bus.
- If the connecting network device (network switch) fails, nodes attached are disabled and can't participate in network communication.
- If hub goes down everything goes down, none of the devices can work without hub.
- Hub requires more resources and regular maintenance because it's the central system of star.
- Extra hardware is required (hubs or switches) which adds to cost
- Performance is predicated on the one concentrator i.e., hub.

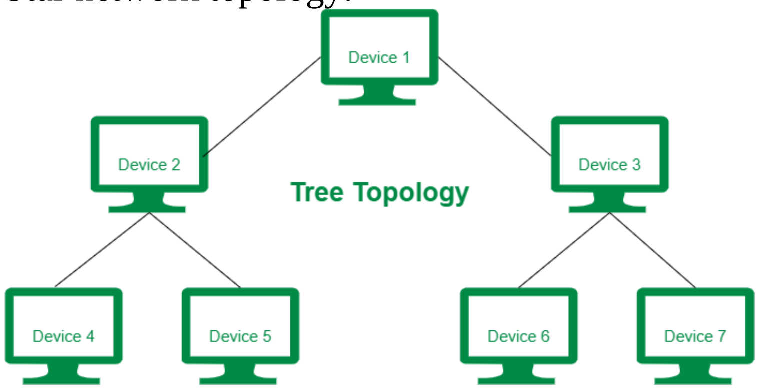
Extended star topology:

- It is the extended version of the star topology. This topology helps to control multiple nodes at the same time.
- The data can be easily transmitted across the network and there is less chance of network failure compared to other topologies.
- The central hub controls the whole network in the extended star topology.



Tree Topology:

- Tree Topology is a topology which is having a tree structure in which all the computers are connected like the branches which are connected with the tree.
- In Computer Network, tree topology is called a combination of a Bus and Star network topology.



Advantages:

- This topology is the combination of bus and star topology.
- This topology provides a hierarchical as well as central data arrangement of the nodes.
- As the leaf nodes can add one or more nodes in the hierarchical chain, this topology provides high scalability.
- Tree Topology is highly secure.
- It is used in WAN.
- Tree Topology is reliable.

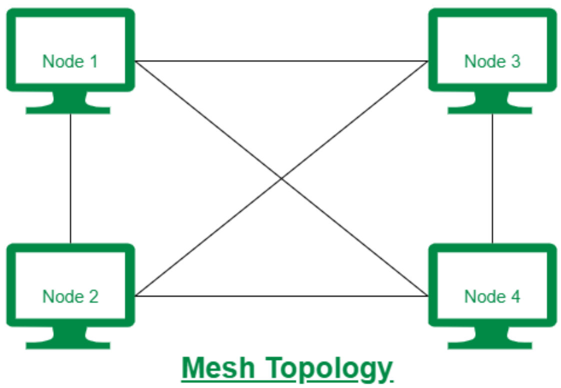
Disadvantages:

- Requires a large number of cables compared to star and ring topology.
- As the data needs to travel from the central cable this creates dense network traffic.
- The Backbone appears as the failure point of the entire segment of the network.
- Treatment of the topology is pretty complex.
- The establishment cost increases as well.
- If the bulk of nodes is added to this network, then the maintenance will become complicated.

- This network is very difficult to configure as compared to the other network topologies.

Mesh Topology:

- A mesh topology is a network setup where each computer and network device is interconnected with one another.
- This topology setup allows for most transmissions to be distributed even if one of the connections goes down.
- It is a topology commonly used for wireless networks.



Advantages of a mesh topology

- Manages high amounts of traffic, because multiple devices can transmit data simultaneously.
- A failure of one device does not cause a break in the network or transmission of data.
- Adding additional devices does not disrupt data transmission between other devices.

Disadvantages of a mesh topology

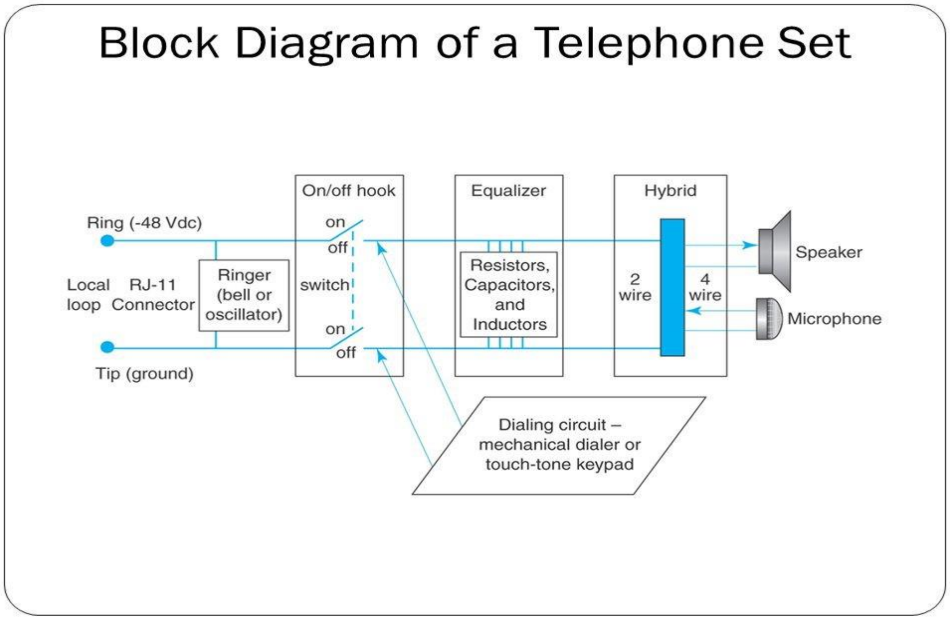
- The cost to implement is higher than other network topologies, making it a less desirable option.
- Building and maintaining the topology is difficult and time consuming.
- The chance of redundant connections is high, which adds to the high costs and potential for reduced efficiency.

Central office switch

- A telephone exchange, telephone switch, or central office is a telecommunications system used in the public switched telephone network (PSTN) or in large enterprises.
- It interconnects telephone subscriber lines or virtual circuits of digital systems to establish telephone calls between subscribers.
- The term telephone exchange is often used synonymously with central office, a Bell System term.
- Often, a central office is defined as a building used to house the inside plant equipment of potentially several telephone exchanges, each serving a certain geographical area.
- Such an area has also been referred to as the exchange or exchange area. In North America, a central office location may also be identified as a wire center, designating a facility to which a telephone is connected and obtains dial tone.

Subscriber Telephone:

- Telephone subscriber or 'subscriber' means a person or entity to whom exchange telephone service, either residential or commercial, is provided and in return for which the person or entity is billed on a monthly basis.
- When the same person, business, or organization has several telephone accesses lines, each exchange access facility constitutes a separate subscription
- Device which converts human speech in the form of sound waves produced by the vocal cord to electrical signals basically a transducer i.e., a device that converts one form of energy into a different form
- Signals are then transmitted over telephone wires and then converted back to sound waves for human ears
  - Transmitter (Microphone)
  - Receiver (Earphone)
  - Signaling functions



Transmitter (Microphone)

- consists of a movable speaker diaphragm that is sensitive to both amplitude and frequency
- The diaphragm contains carbon particles that can conduct electricity
- As the human voice spoken into the transmitter varies, the amount of carbon granules that strike the electrical contacts in the mouthpiece also varies—thereby sending varying analog electrical signals out into the voice network.

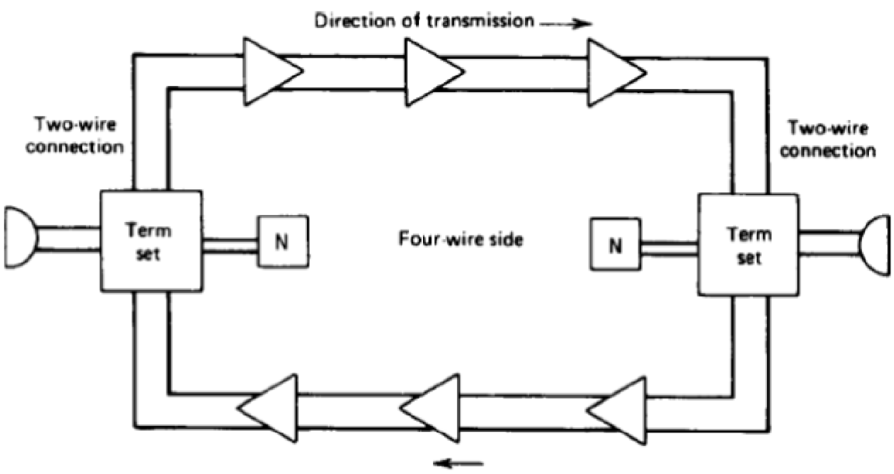
Receiver (Earphone)

- Acts in an opposite direction to the mouthpiece
- The electrical signal/waves produced by the transmitter are received at an electromagnet in the receiver
- Varying levels of electricity produce varying levels of magnetism—that, in turn, causes the diaphragm to move in direct proportion to the magnetic variance
- The moving diaphragm produces a varying sound that corresponds to the sound waves that were input at the transmitter

Four wire Circuits

- Necessary to use amplifiers to compensate for the attenuation of the transmission path most amplifiers are unidirectional

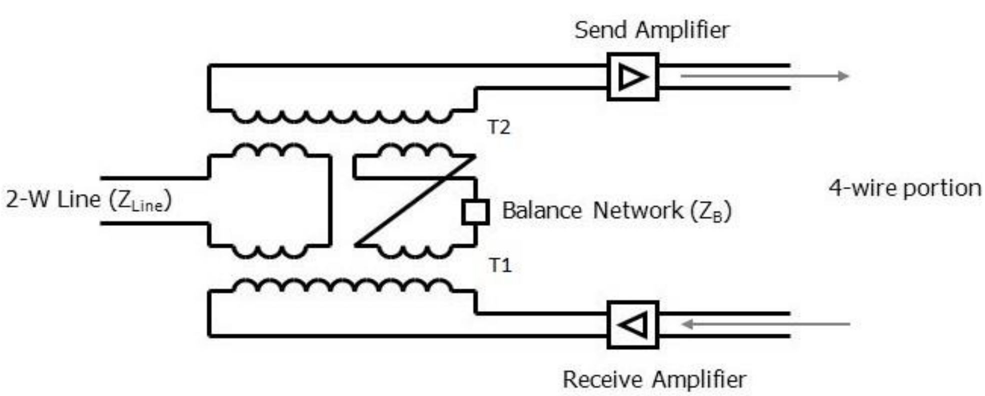
- Necessary to provide separate channels for transmit (go) and receive (return) signal directions of transmission called four-wire circuit
- At each end, the four-wire circuit must be connected to a two-wire line leading to a telephone
- If both paths of the four-wire circuit were connected directly to the two-wire circuit at each end a signal could circulate round the complete loop thus created
- This leads result in continuous oscillation known as singing results if the loop gain at some frequency is greater than unity.

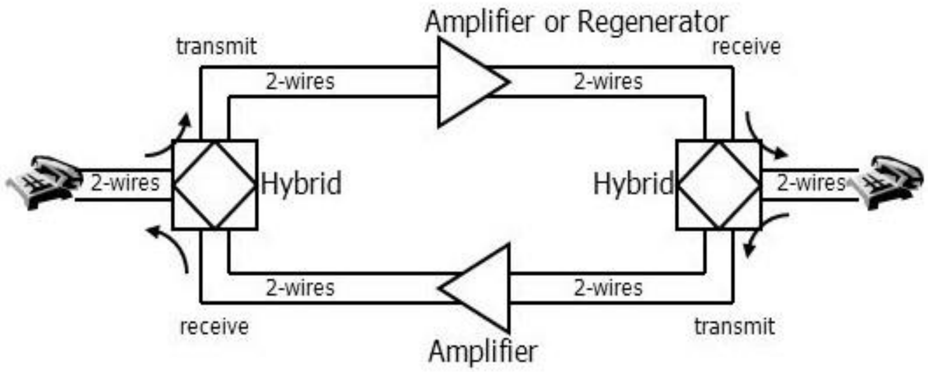


2W/4W hybrid in long distance connection

Hybrid Transformer

- To avoid the signing, the two-wire line at each end is connected to the four-wire circuit by hybrid transformer
- It is used to convert a 2-wire circuit at the phone/terminal end to a 4-wire system in the switching network consists of two cross connected transformer and a line balance network whose impedance is similar to that of the two-wire circuit over the required frequency band.



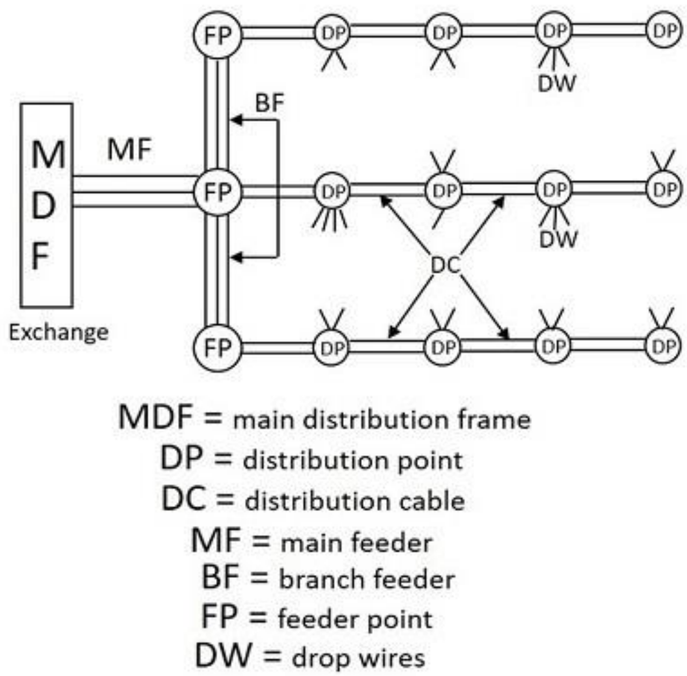


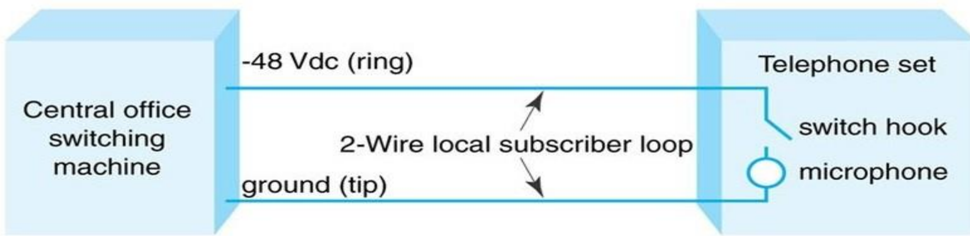
- The output signal from the receive amplifier causes equal voltages to be induced in the secondary windings of transformer T1
- If the impedances of the two-wire line and the line balance are equal, i.e., $Z_{line} = Z_B$ equal currents flow in the primary windings of transformer T2
- windings are connected in antiphase thus no EMF induced in the secondary windings of transformer T2
- No signal is applied to the input of the send amplifier

Subscriber Loop:

Subscriber Loop Systems

- A **subscriber loop carrier** or **subscriber line carrier** (**SLC**) provides telephone exchange-like telephone interface functionality.
- In a general telephone network, every subscriber has two dedicated lines connecting to the nearest switching exchange, which are called the **Loop lines** of that subscriber.
- The laying of lines to the subscriber premises from the exchange office is called **Cabling**. As it is difficult to run cables from each subscriber's premises to the exchange, large cables are used through which the drop wires (subscriber lines) are taken to a distribution point.
- The drop wires are connected to wire pairs at the distribution point, in the cables. Such distribution cables from nearby geographical area are connected at a same feeder point where they connected to branch feeder cables which in turn, are connected to the main feeder cable.
- This whole process can be understood with the help of the following figure





- The subscriber cable pairs from the exchange will also terminate at MDF through main feeder cables that carry large number of wire pairs.
- These subscriber pairs and exchange pairs are interconnected at the MDF using jumpers, which makes MDF to provide flexible mechanism for reallocating cable pairs and subscriber numbers.
- This means a subscriber who shifts to a different location though in the same exchange area, can be allowed to use the same number using appropriate jumper, while his old drop wires can be used by another subscriber with a new number.

Inlets and Outlets:

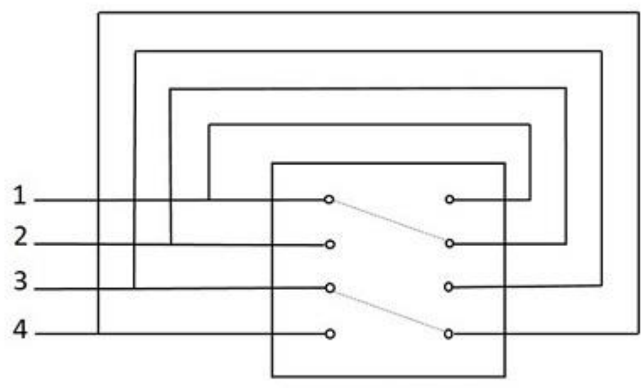
The set of input circuits of an exchange are called **Inlets** and the set of output circuits are called the **Outlets.** The primary function of a switching system is to establish an electrical path between a given inlet-outlet pair.

Usually, **N** indicates the inlets and the outlets are indicated by **M**. So, a switching network has **N** inlets and **M** outlets.

Folded Network:

- When the number of inlets is equal to the number of outlets for a switching network, such a network is called the **Symmetric Network**, which means N=M. A network where the outlets are connected to the inlets, is called the **Folded Network.**

- In a Folded Network, the N number of inlets which come as outlets are again folded back to the inlets. Nevertheless, the switching network provides connections to the inlets and outlets as per the requirement. The following figure will help you understand how the Switching Network works.



- As one connection can be given to one line per time, only N/2 connections are established for N inlets of a folded network. Such a network can be called as **Nonblocking network.** In a non-blocking network, as long as the called subscriber is free, a calling subscriber will be able to establish a connection to the called subscriber.

- In the above figure, only 4 subscribers were considered - where line 1 is busy with line 2 and line 3 is busy with line 4. While the call is in progress, there used to be no chance for making another call and hence, only a single connection was made. Hence for N inlets, only N/2 lines are connected.

- At times, it might happen that the inlet and outlet connections are continuously used to make Transit calls through trunk lines only, but not among the local subscribers.

- The inlet and outlet connections if used in an **Inter-exchange transmission** such that the exchange does not support connection between local subscribers, then it is called the Transit Exchange.

- A switching network of such kind is called the **Non-folded network.** This is shown in the following figure −

Difference between analog and digital signals

S. No.	Analog signal	Digital Signal
1	Analog signals are continuous signals	Digital signals are discrete signals.
2	Analog signal uses continuous values for representing the information.	A digital signal uses discrete values for representing the information.
3	Analog signals can be affected by the noise during the transmission.	Digital signals cannot be affected by the noise during transmission.
4	Accuracy of Analog signal is affected by the noise.	Digital signals are noise-immune hence there accuracy is less affected
5	Devices which are using analog signals are less flexible	Device using digital signals are very flexible
6	Analog signals consumes less bandwidth	Digital signals consume more bandwidth.
7	Analog signal are stored in the form of continuous wave form.	Digital signals are stored in the form of binary bits "0", "1".
8	Analog signals have low cost.	Digital signals have high cost.
9	Analog signals are portable.	Digital signals are not Portable.
10	Analog signals give observation error	Digital Signals doesn't give observation error.

Advantages of Digital system over the analog system:

- Several digital signals can be multiplexed together.
- Regenerative repeaters can be used in case of digital signals.
- Encryption and decryption in digital system.

TRANSMISSION Impairments:

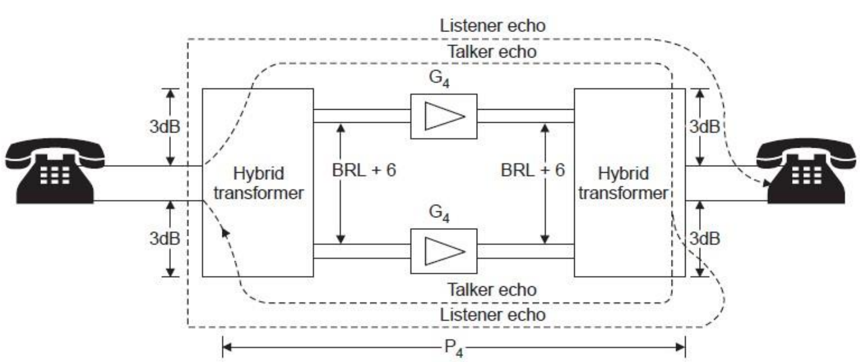
- Transmission impairment occurs when the received signal is different from the transmitted signal. As we know, a signal can be transmitted as Analog signal or it can be transmitted as a digital signal.
- In Analog signals due to transmission impairment the resulting received signal gets different amplitude or the shape. In the case of digitally transmitted signals at the receiver side we get changes in bits (0's or 1's).

Causes of Transmission impairments

1**. Jitter**: It is the delay in time while transmitting data packets over the network. Change in route and network congestion may cause Jitter. More the jitter poorer with be the quality of communication. i.e., audio video streaming.

2. **Echo and Singing**: When the transmitted signal is fed back to the source again then this is considered as echo as well as singing. They are the repetition of waveform when transmitted signal are coupled into the returned path.

- In a four-wire circuit when Z line ≠ ZB, an imperfect line balance causes part of the signal energy transmitted in one direction to return in the other
- A signal reflected to the speaker's end of the circuit is called talker echo
- A signal reflected at the listener's end is called listener's echo When the returning signal is repeatedly coupled back into the forward path to produce oscillations, singing occurs
- Echoes and singing both occur as a result of transmitted signals being coupled into a return path and fed back to the respective sources.



$$T2 = T24 + T42 - G4\ dB \qquad (1)$$

where,

T_{24}: attenuation between 2 to 4 wire line

T_{42}: attenuation on between 4 to 2 wire line

G_4: total gain of one side of four wire circuit (in dB)

- To avoid signing, the T_{24}, T_{42} are normally $3dB$

$$T_2 = 6 - G_4\, dB \qquad (2)$$

$$T2 = T24 + T42 - G4\ dB \qquad (1)$$

where,

T_{24}: attenuation between 2 to 4 wire line

T_{42}: attenuation on between 4 to 2 wire line

G_4: total gain of one side of four wire circuit (in dB)

- To avoid signing, the T_{24}, T_{42} are normally $3dB$

$$T_2 = 6 - G_4\, dB \qquad (2)$$

$$L_t = 3 - G_4 + (6 + BRL) - G_4 + 3 = 2T_2 + BRL\ dB \quad (4)$$

- Echo is delayed by a time

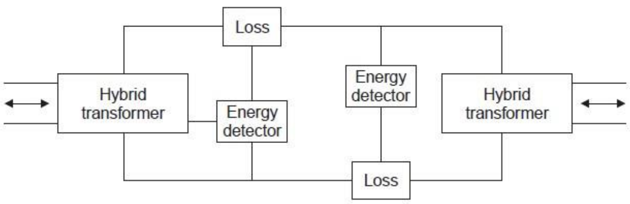
$$D_t = 2T_4 \qquad (5)$$

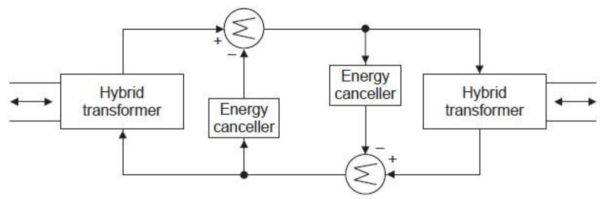
- T_4: delay of the four-wire circuit (between 2-wire terminations)
- The attenuation L_t of the echo that reaches the listener's 2-wire line is

$$L_s = (6 + BRL) - G_4 + (6 + BRL) - G_4 = 2T_2 + 2BRL\ dB \ (6)$$

- An echo suppressor operates in four wire circuits by measuring the speech power in each leg and inserts a large amount of loss (35 dB typically) in the opposite leg when the power level exceeds a threshold

  When full duplex transmission is used, the echo suppressors must be disabled



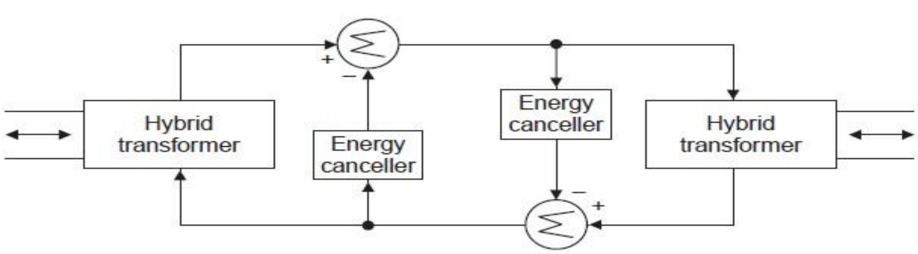


  An echo suppressor converts a full duplex circuit into a half-duplex circuit with energy sensing being the means of turning the line around

- An echo canceller operates by simulating the echo path to subtract a properly delayed and attenuated copy of a transmitted signal
- The transmitted speech is stored for a period of time equal to the round-trip delay of the circuit
- The stored signal is attenuated and then subtracted from the incoming signal

3. **Cross talk:**

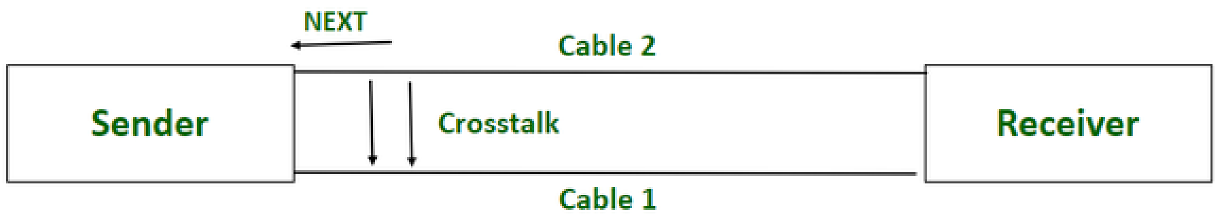
- This is the effect caused when signal transmitted on one channel creates an unwanted effect in another channel during communication. Simply when one signal is shared in two or more transmission system then there arises cross talk. It reduces the quality of communication.
- Crosstalk occurs when a signal transmitted on one copper twisted pair in a bundle radiates and potentially interferes with and degrades the transmission on another pair.
- Crosstalk can and will typically happen between pairs/services within the same copper cable or binder. Cross talk is strongest within the same binder,



and depends on the placement and proximity of pairs. It can also be strong between binders – but tends to be negligible between cables.

Near-End Crosstalk (NEXT):

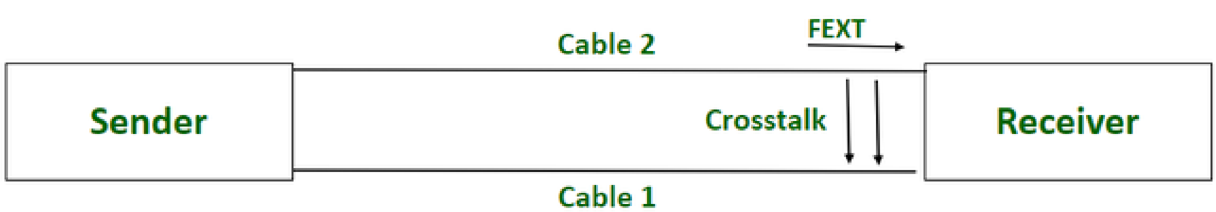
- Near-End Crosstalk refers to the disturbance in analog signal in one of the twisted pair cables due to the signal in another pair of twisted pair cables at the near end of the transmission medium i.e., near the source of data transmission.
- It occurs when outgoing data signal leaks and corrupts incoming data signal. As a result, the incoming signal gets mixed with the outgoing signal at the near end of the transmitting station.



- *NEXT*

Far-End Crosstalk (FEXT):

- Far End Crosstalk refers to the disturbance in analog signal in one of twisted pair cable due to the signal in other twisted pair cable at the far end of the transmission medium i.e., near the destination of data transmission.
- It occurs when incoming data signal leaks and corrupts outgoing data signal at the receiver end. As a result, the outgoing signal gets mixed with the incoming signal at the far end of the transmitting station.

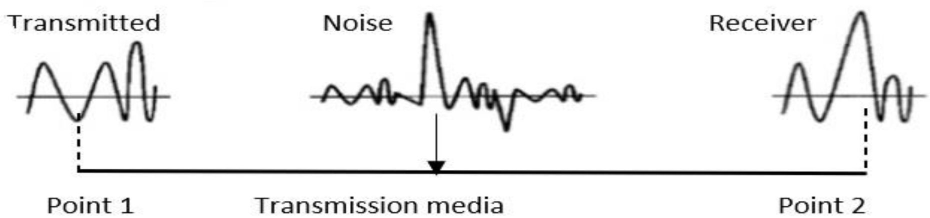


- *FEXT)*

4. Noise

- Noise is the major factor for the transmission distortion as any unwanted signal gets added to the transmitted signal by which the resulting transmitted signal gets modified and at the receiver side it is difficult to remove the unwanted noise signal.
- These noises are various kinds like shot noise, impulse noise, thermal noise etc.

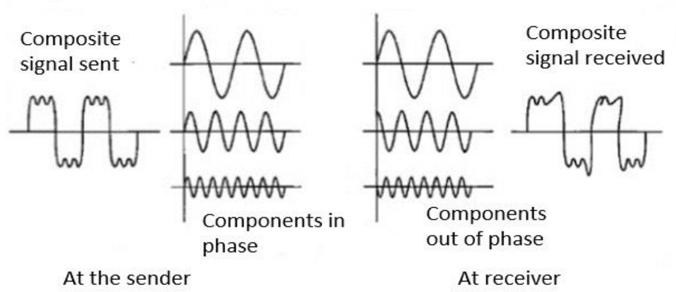
Noise is diagrammatically represented as follows −



5. Distortion

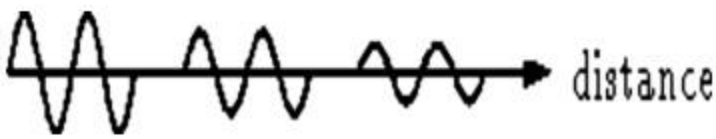
- This kind of distortion is mainly appearing in case of composite signals in which a composite signal has various frequency components in it and each frequency component has some time constraint which makes a complete signal.
- But while transmitting this composite signal, if a certain delay happens between the frequency's components, then there may be the chance that the frequency component will reach the receiver end with a different delay constraint from its original which leads to the change in shape of the signal.
- The delay happens due to environmental parameters or from the distance between transmitter and receiver etc.

Distortion is diagrammatically represented as follows −



6. Attenuation

- Attenuation is generally decreased in signal strength, by which the received signal will be difficult to receive at the receiver end.
- This attenuation happens due to the majority factor by environment as environment imposes a lot of resistance and the signal strength decreases as it tries to overcome the resistance imposed.



The above picture shows that the signal loses power at its travels time.

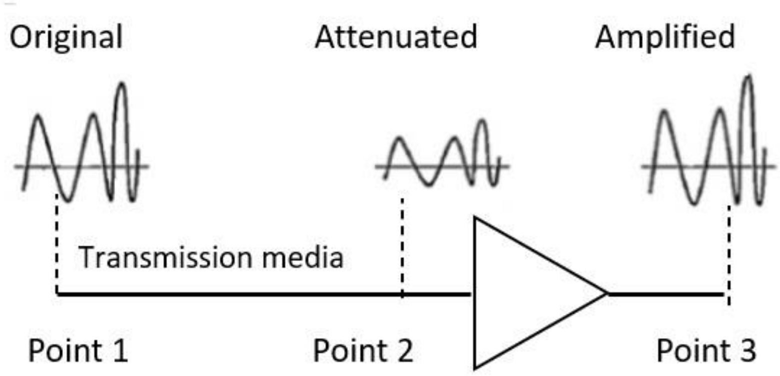
Attenuation is diagrammatically represented as follows –

- Attenuation is measured in **decibels(dB)**. It measures the relative strengths of two signals or one signal at two different point.

Attenuation(dB) = 10log10(P2/P1)
P1 is the power at sending end and P2 is the power at receiving end.

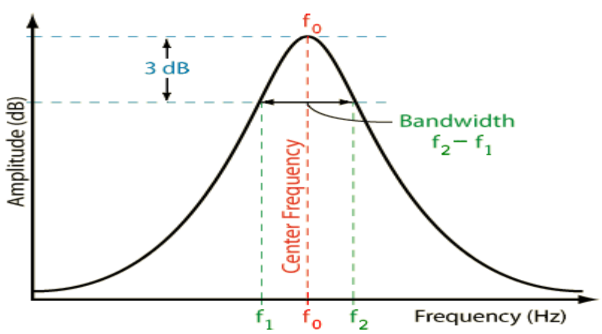
Some where the decibel is also defined in terms of voltage instead of power. In this case because power is proportional to the square of the voltage the formula is

Attenuation(dB) = 20log10(V2/V1)
V1 is the voltage at sending end and V2 is the voltage at receiving end.



7. **Bandwidth:**

- The rate of data transmission in a given medium per unit time is known as bandwidth.
- It is generally measured in terms of bps, Kbps, Mbps etc. Higher the bandwidth, higher will be the data transmission and faster will be the communication.
- But if bandwidth is low then it leads to network congestion causing network impairments.



8. **Interference:**

- In telecommunications, an **interference** is that which modifies a signal in a disruptive manner, as it travels along a communication channel between its source and receiver.

- The term is often used to refer to the addition of unwanted signals to a useful signal. Common examples include:
    - Electromagnetic interference (EMI)
    - Co-channel interference (CCI), also known as crosstalk
    - Adjacent-channel interference (ACI)
    - Intersymbol interference (ISI)
    - Inter-carrier interference (ICI), caused by doppler shift in OFDM modulation (multitone modulation).
    - Common-mode interference (CMI)
    - Conducted interference

Transmission medium:

BASIS FOR COMPARISON	GUIDED MEDIA	UNGUIDED MEDIA
Basic	**The signal requires a physical path for transmission.**	**The signal is broadcasted through air or sometimes water.**
Alternative name	**It is called wired communication or bounded transmission media.**	**It is called wireless communication or unbounded transmission media.**

BASIS FOR COMPARISON	GUIDED MEDIA	UNGUIDED MEDIA
Direction	**It provides direction to signal for travelling.**	**It does not provide any direction.**
Types	**Twisted pair cable, coaxial cable and fiber optic cable.**	**Radio wave, microwave and infrared.**

Note:

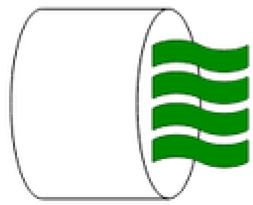
- **The guided media formed the different network topologies whereas the unguided media formed the continuous network topologies.**
- **In Guided Media the signals are in the state of current and voltage whereas in Unguided media the signals are in the state of electromagnetic waves.**

Guided media:

a) **Twisted Pair Cable –**
   - It consists of 2 separately insulated conductor wires wound about each other.
   - Generally, several such pairs are bundled together in a protective sheath.
   - They are the most widely used Transmission Media. Twisted Pair is of two types:

i) **Unshielded Twisted Pair (UTP):**
UTP consists of two insulated copper wires twisted around one another.

This type of cable has the ability to block interference and does not depend on a physical shield for this purpose. It is used for telephonic applications.



Unshielded Twisted Pair

Advantages:

⇢ Least expensive

⇢ Easy to install

⇢ High-speed capacity

Disadvantages:

⇢ Susceptible to external interference

⇢ Lower capacity and performance in comparison to STP

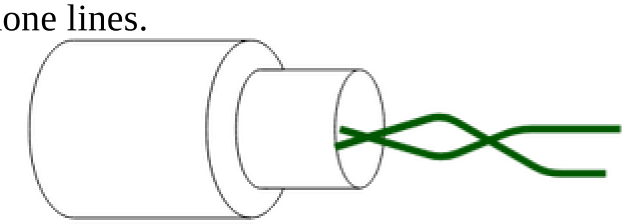
⇢ Short distance transmission due to attenuation

Applications:

Used in telephone connections and LAN networks

ii) Shielded Twisted Pair (STP):

- This type of cable consists of a special jacket (a copper braid covering or a foil shield) to block external interference.
- It is used in fast-data-rate Ethernet and in voice and data channels of telephone lines.



Shielded Twisted Pair

Advantages:

⇢ Better performance at a higher data rate in comparison to UTP

⇢ Eliminates crosstalk

⇢ Comparatively faster

Disadvantages:

⇢ Comparatively difficult to install and manufacture

⇢ More expensive

⇢ Bulky

Applications:

- The shielded twisted pair type of cable is most frequently used in extremely cold climates, where the additional layer of outer covering makes it perfect for withstanding such temperatures or for shielding the interior components.

b) Coaxial Cable –

- It has an outer plastic covering containing an insulation layer made of PVC or Teflon and 2 parallel conductors each having a separate insulated protection cover.
- The coaxial cable transmits information in two modes: Baseband mode (dedicated cable bandwidth) and Broadband model (cable bandwidth is split into separate ranges).
- Cable TVs and analog television networks widely use Coaxial cables.

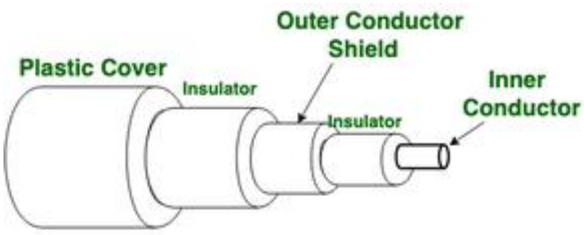


Figure of Coaxial Cable

Advantages:

- High Bandwidth
- Better noise Immunity
- Easy to install and expand
- Inexpensive

Disadvantages:

- Single cable failure can disrupt the entire network

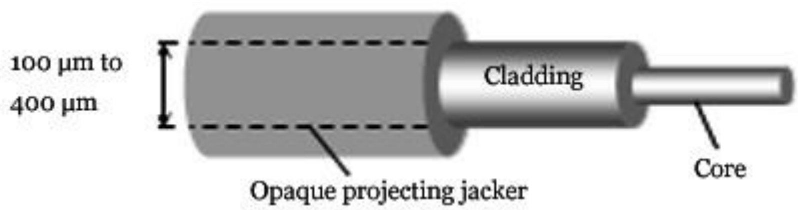
Applications:

- Radio frequency signals are sent over coaxial wire.
- It can be used for cable television signal distribution, digital audio (S/PDIF), computer network connections (like Ethernet), and feedlines that connect radio transmitters and receivers to their antennas.

c) Optical Fiber Cable –

- Optical fiber is the technology associated with data transmission using light pulses travelling along with a long fiber which is usually made of plastic or glass.

Design Of an Optical Fiber



- Optical fiber is made of a thin glass core (diameter 10 to 100µm) surrounded by a glass coating called cladding, protected by a jacket of plastic.

A Fiber Optic Relay System consists of the following components:

1. The Transmitter – It produces the light signals and encodes them to fit to transmit.
2. The Optical Fiber – The medium for transmitting the light pulse (signal).
3. The Optical Receiver – It receives the transmitted light pulse (signal) and decodes them to be fit to use.
4. The Optical Regenerator – Necessary for long-distance data transmission.

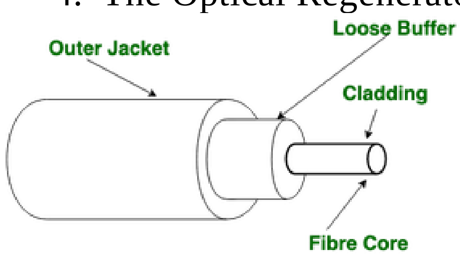


Figure of Optical Fibre Cable

Advantages:

- Increased capacity and bandwidth
- Lightweight, Less signal attenuation
- Immunity to electromagnetic interference
- Resistance to corrosive materials
- Economical and cost-effective, Less power consumption
- Thin and non-flammable, Less signal degradation
- Flexible and lightweight, Excellent data security

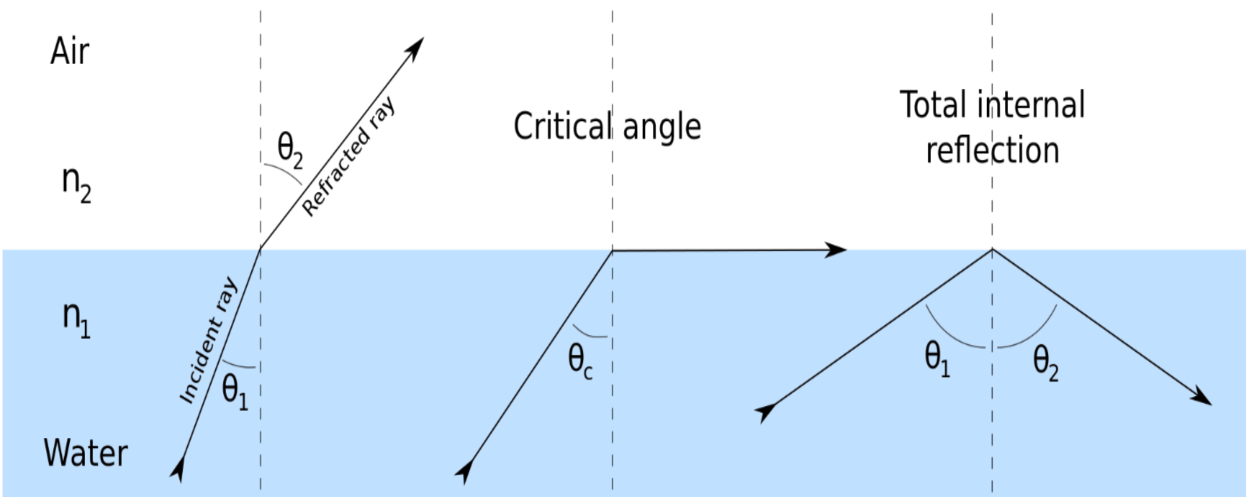
- Cost-effective, Unaffected by interference

Disadvantages:

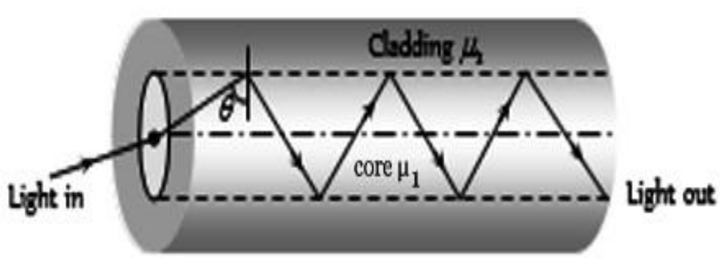
- Difficult to install and maintain
- High cost
- Fragile

Working of Optical Fiber

- Optical fiber works on the principle of total internal reflection.
- When light travelling in an optically dense medium hits a boundary at a steep angle (larger than the critical angle for the boundary), the light is completely reflected.
- This is called total internal reflection.



- This effect is used in optical fibers to confine light in the core.
- Light travels through the fiber core, bouncing back and forth off the boundary between the core and cladding.
- Because the light must strike the boundary with an angle greater than the critical angle, the only light that enters the fiber within a certain range of angles can travel down the fiber without leaking out.



- This range of angles is called the acceptance cone of the fiber.
- The size of this acceptance cone is a function of the refractive index difference between the fiber's core and cladding.

1. Critical Angle (θ_C)

At core-cladding interface, if $\theta_c = \theta$, then

$$\cos\theta_c = \sqrt{\frac{\mu_1{}^2 - \mu_2{}^2}{\mu_1}} \Rightarrow \theta_c = \cos^{-1}\left(\frac{\sqrt{\mu_1{}^2 - \mu_2{}^2}}{\mu_1}\right)$$

2. Acceptance Angle (θ_a)

The value of maximum angle of incidence with the axis of fibre in the air for which all the incident light is totally reflected is known as acceptance angle.

If θ_a = Acceptance angle, ?$_1$= refractive index of core and ?$_2$=refractive index of cladding, then

$$\sin\theta_a = \frac{\sqrt{\mu_1{}^2 - \mu_2{}^2}}{\mu_0} \Rightarrow \theta_a = \sin^{-1}\sqrt{\mu_1{}^2 - \mu_2{}^2}$$

3. Numerical Aperture

Light gathering capability of an optical fibre is related to its numerical aperture. This is defined as the sine of its acceptance angle. That is,

$$NA = \sin i = \sqrt{\mu_1{}^2 - \mu_2{}^2}$$

The numerical aperture can also be given in terms of relative core, cladding index difference (Δ), where

$$\Delta = \frac{\mu_1{}^2 - \mu_2{}^2}{2\mu_1{}^2}$$

Thus,

$$NA = \sqrt{\mu_1{}^2 - \mu_2{}^2} = \mu_1\sqrt{2\Delta}$$

4. Fibre Attenuation

In practice, a very small part of light energy is lost from an optical fibre. This reduction in the light energy is called attenuation and is described by

$$I = I_0 e^{-\frac{a}{x}}$$

where,

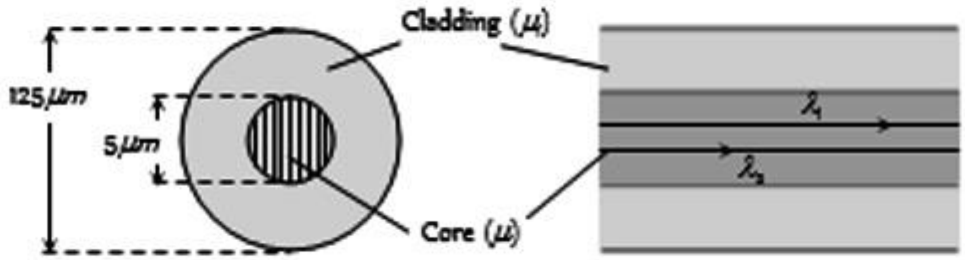
I = Intensity of light when it enters the fibre
I_0= Intensity of light as a distance x along the fibre
a= absorption coefficient or attenuation coefficient

Types Of Optical Fiber

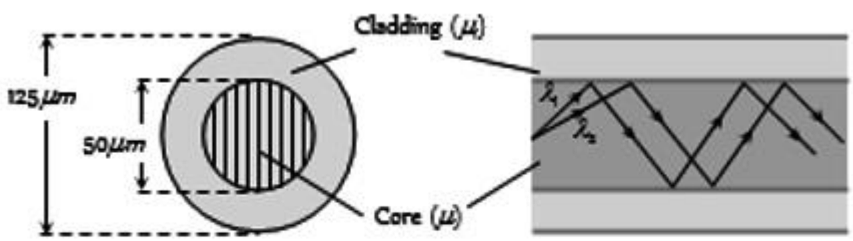
- Optical fibers are classified on different parameters such as refractive index, materials used and mode of propagation of light. Let's see each classification in detail.

<u>Based on Refractive Index</u>

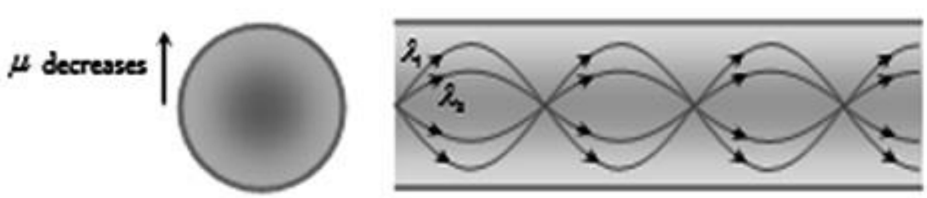
1. <u>Mono Mode Optical Fiber</u>: It has a very narrow core of diameter about 5µm or less, cladding is relatively big.



2. <u>Multi-mode Optical Fiber</u>: It is again of two types:
i) Step Index Multi Mode Fiber: The diameter of the core is about 50 µm. The core has a constant R. The refractive index then changes to a lower value of ? which remains constant through the cladding.



(ii) ***Graded Index Multi Mode Fiber*****:** Refractive index decreases smoothly from its center to the outer surface of the fiber (cladding). There is no noticeable boundary between core and cladding.



Based on Materials Used

The classification based on the materials used is as follows:

1. **Plastic Optical Fibers:** Polymethylmethacrylate is used as a core material for the transmission of light.
2. **Glass Fibers:** It consists of extremely fine glass fibers.

Based on Mode of Propagation of Light

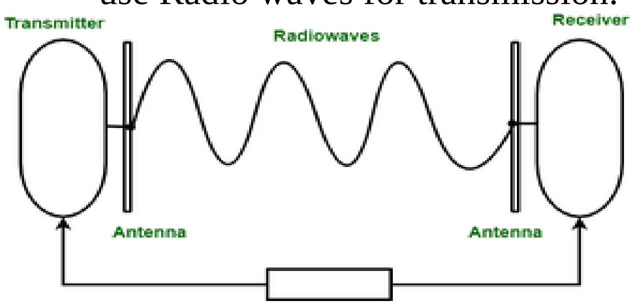
The classification based on the mode of propagation of light is as follows:

- **Single-Mode Fibers:** These fibers are used for the long-distance transmission of signals.
- **Multimode Fibers:** These fibers are used for the short-distance transmission of signals.

Unguided Media

(i)Radio waves –

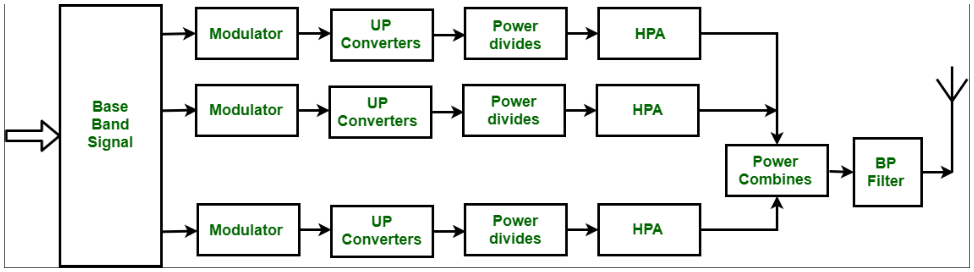
- These are easy to generate and can penetrate through buildings.
- The sending and receiving antennas need not be aligned.
- Frequency Range:3KHz – 1GHz. AM and FM radios and cordless phones use Radio waves for transmission.



Further Categorized as (i) Terrestrial and (ii) Satellite.

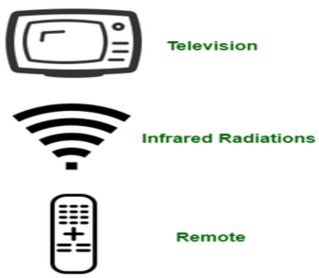
(ii) Microwaves –

- It is a line-of-sight transmission i.e., the sending and receiving antennas need to be properly aligned with each other.
- The distance covered by the signal is directly proportional to the height of the antenna. Frequency Range:1GHz – 300GHz.
- These are majorly used for mobile phone communication and television distribution.



(iii) Infrared – Infrared waves are used for very short distance communication.

- They cannot penetrate through obstacles.
- This prevents interference between systems.
- Frequency Range:300GHz – 400THz. It is used in TV remotes, wireless mouse, keyboard, printer, etc.



Teleservices vs Bearer services

Teleservices:

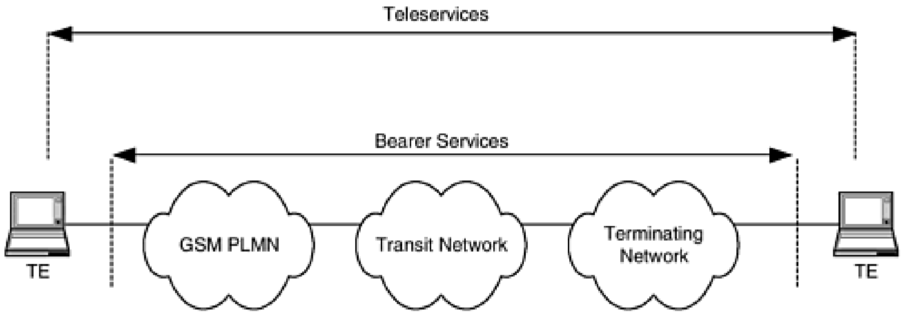
- Teleservices provide service capability up to the point the service is used, i.e., to the user.
- Therefore, they also include the terminal equipment functions.
- They can provide end-to-end (user-to-user) communication.
- A teleservice requires the use of an appropriate bearer service for information transfer.
- Examples are telephony, facsimile, telex, teletex, or videotex. Teleservice involves both lower-layer and higher-layer functions (i.e., layers 1 to 7).

Bearer services:

- Bearer services define the transmission capacity and functions required from the network.
- They provide the capability and means to transfer the information between the ISDN user-network interfaces.
- They do not include the user terminal functions.
- Bearer services allow the user to send information from one device to another and involve only lower-layer functions (i.e., layers 1 to 3 of the OSI model).

Supplementary services:

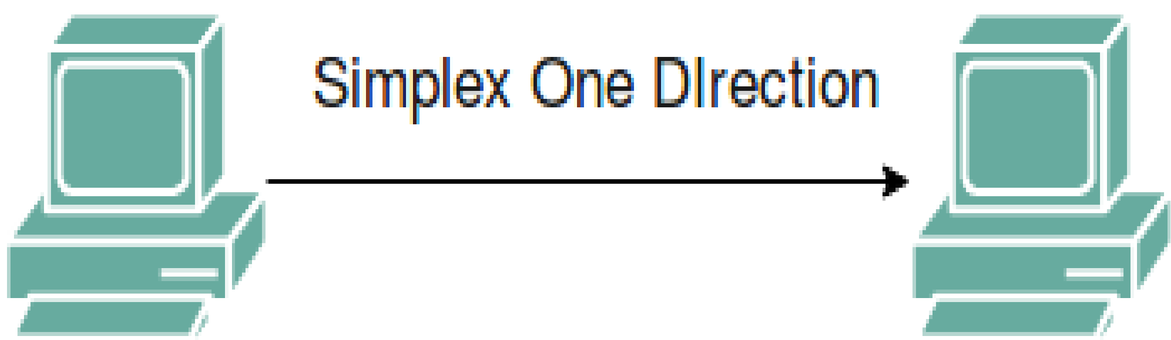
- Supplementary services enhance the functionality of basic services by providing additional features and capabilities.
- They are always offered with association with a basic service.
- Several supplementary services may be associated with a basic service.
- The number of associated supplementary services depends on the user's needs.



Transmission Modes:

1.Simplex Mode –

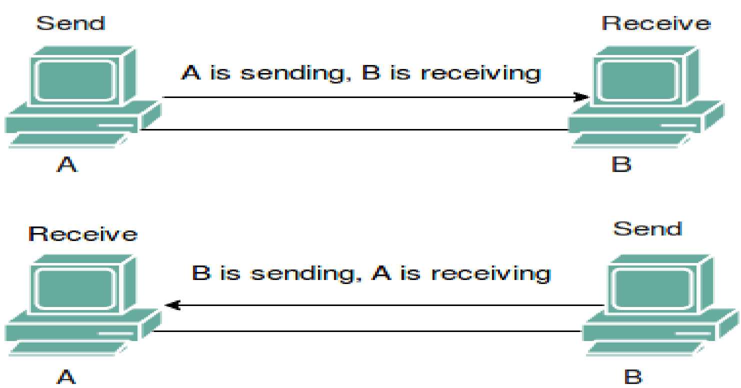
- In Simplex mode, the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit, the other can only receive.
- The simplex mode can use the entire capacity of the channel to send data in one direction.
  Example: Keyboard and traditional monitors. The keyboard can only introduce input, the monitor can only give the output.



2.Half-Duplex Mode –

- In half-duplex mode, each station can both transmit and receive, but not at the same time.

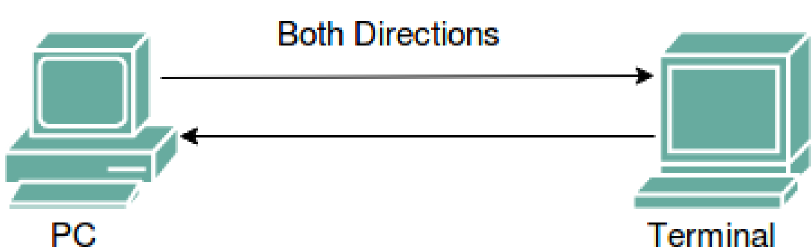
- When one device is sending, the other can only receive, and vice versa.
- The half-duplex mode is used in cases where there is no need for communication in both directions at the same time.
- The entire capacity of the channel can be utilized for each direction.
- Example: Walkie-talkie in which message is sent one at a time and messages are sent in both directions.
- Channel capacity=Bandwidth * Propagation Delay



3.Full-Duplex Mode –

- In full-duplex mode, both stations can transmit and receive simultaneously. In fullduplex mode, signals going in one direction share the capacity of the link with signals going in another direction, this sharing can occur in two ways:
- Full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.
- Example: Telephone Network in which there is communication between two persons by a telephone line, through which both can talk and listen at the same time.

- Channel Capacity=2* Bandwidth*propagation Delay



What are the different optical sources and the detectors in the optical communication?

- There are two main categories of optical signal sources—light emitting diodes and infrared laser diodes.

  - Light emitting diodes (LEDs) are the lower-cost, lower-performance source. They're used in applications where lower data rates and/or shorter distances are acceptable.

  - Infrared laser diodes operate at much higher speeds, dissipate higher power levels, and require temperature compensation or control to maintain specified performance levels. They are also more costly.

- Signal detectors also fall into two main categories—PIN photodiodes and avalanche photodiodes.

  - Similar to sources, the two types provide much different cost/performance ratios.

  - PIN photodiodes are more commonly used, especially in less stringent applications.

  - Avalanche photodiodes, on the other hand, are very sensitive and can be used where longer distances and higher data rates are involved.

Why are the trunk transit exchange, AAR, core network and the international gateway exchange are needed in PSTN?

- The trunk transit exchange, also known as a trunk exchange, is a key component of the public switched telephone network (PSTN).
- It is responsible for connecting telephone exchanges and switching calls between them.
- The AAR, or Automatic Alternate Routing, is a feature of the PSTN that enables calls to be automatically rerouted to another destination if the original route is unavailable due to a fault or congestion.
- The core network refers to the central part of the PSTN that connects all the exchanges and switches together.
- It is responsible for routing calls and providing connectivity between different telephone exchanges.
- The international gateway exchange is a special type of exchange that connects the PSTN to other countries' telephone networks.
- It is responsible for routing calls between different countries and for providing international connectivity.
- All of these components are necessary for the PSTN to function properly and to provide reliable telephone service to users.

- The local exchange is the telephone exchange serving a particular local area. It connects individual telephone lines to the PSTN and routes calls to other local exchanges or to the trunk exchange.
- The trunk exchange, also known as a trunk transit exchange, is responsible for connecting local exchanges to each other and to the core network. It switches calls between different local exchanges and routes them to their destination.
- The AAR, or Automatic Alternate Routing, is a feature of the PSTN that enables calls to be automatically rerouted to another destination if the original route is unavailable due to a fault or congestion.
- The core network is the central part of the PSTN that connects all the exchanges and switches together. It is responsible for routing calls and providing connectivity between different telephone exchanges.
- The international gateway exchange is a special type of exchange that connects the PSTN to other countries' telephone networks. It is responsible for routing calls between different countries and for providing international connectivity.

What is a Microwave?

- Microwaves are electromagnetic radiations, also known as microwave radiation. Microwaves have a frequency ranging between 300 MHz and 300 GHz.
- The wavelength of microwaves ranges from 1 mm to around 30 cm. Microwave radiations lie in between the radio waves and infrared radiations.
- Microwaves are short-wavelength radio waves having frequencies in the gigahertz (GHz) range.
- Microwaves can pass through glass.
- Microwaves are produced by special vacuum tubes referred to as klystrons, magnetrons, and Gunn diodes.

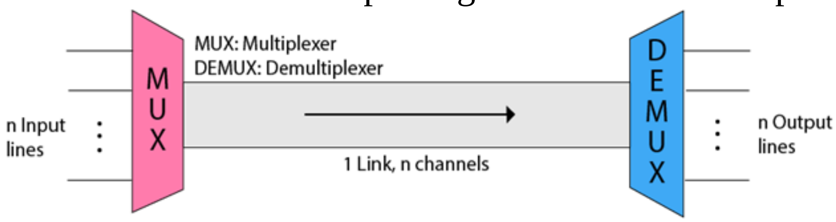
Properties of microwaves or microwave radiation are listed below.

- They are the radiations capable of radiating electromagnetic energy with shorter wavelengths.
- Microwaves are reflected by metal surfaces.
- The transmission of microwaves is affected by phenomena like refraction, diffraction, reflection, and interference.
- They can easily pass-through glass and plastics, and hence are used in heating and cooking in an oven.
- They are easily attenuated within shorter distances.
- These radiations are not reflected by the Ionosphere.
- Microwave radiation can pass through the atmosphere. Hence, microwaves are used in the satellite communication sector to transmit information back and forth to the satellite. We can know why satellite dishes are made of metal since they reflect microwave radiation.
- Microwaves travel in a straight line and are reflected by the conducting surfaces.
- Microwave currents have the capacity to flow through a thin layer of a cable.

Chapter 2: Multiplexing and Multiple Access Technique

Multiplexing:

- Multiplexing is a technique used to combine and send the multiple data streams over a single medium.
- The process of combining the data streams is known as multiplexing and hardware used for multiplexing is known as a multiplexer.



Advantages of Multiplexing:

- More than one signal can be sent over a single medium.
- The bandwidth of a medium can be utilized effectively.

Concentrator:

- A concentrator differs from a multiplexer in that it allows data to be transmitted from only one terminal at a time over a communication channel.
- Concentration is used in circuit switched systems. It is typically used in access network part.
- A concentrator usually provides communication capability between many low-speed, usually asynchronous channels and one or more high-speed, usually synchronous channels.
- A concentrator aggregates and forwards data packets within a system.
- A concentrator may also administrate various dial-up internet calls and function as a network router. EXAMPLE: A remote access Hub Another example of **concentrator** is VPN 3000 series from Cisco.

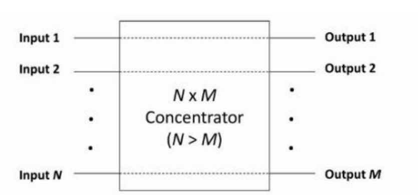
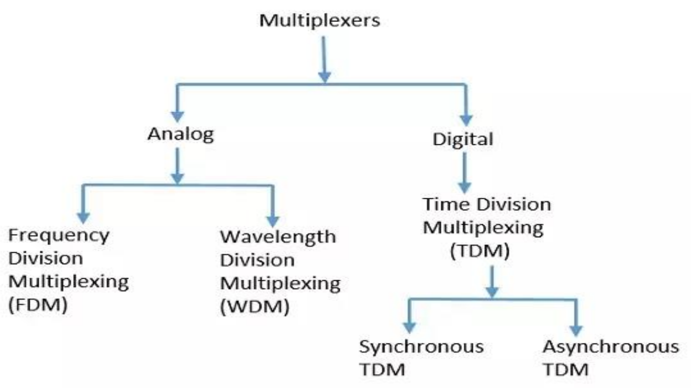


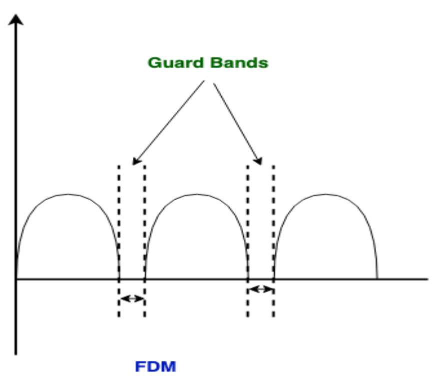
Fig: NxM concentrator

Types of Multiplexing:



FDM:

- FDM stands for Frequency division multiplexing.
- FDM works with only analog signals.
- It has high conflict.
- It’s wiring or chip is complex rather than simple.
- It is inefficient.
- In this, frequency sharing takes place.
- In it Guard band is necessary.

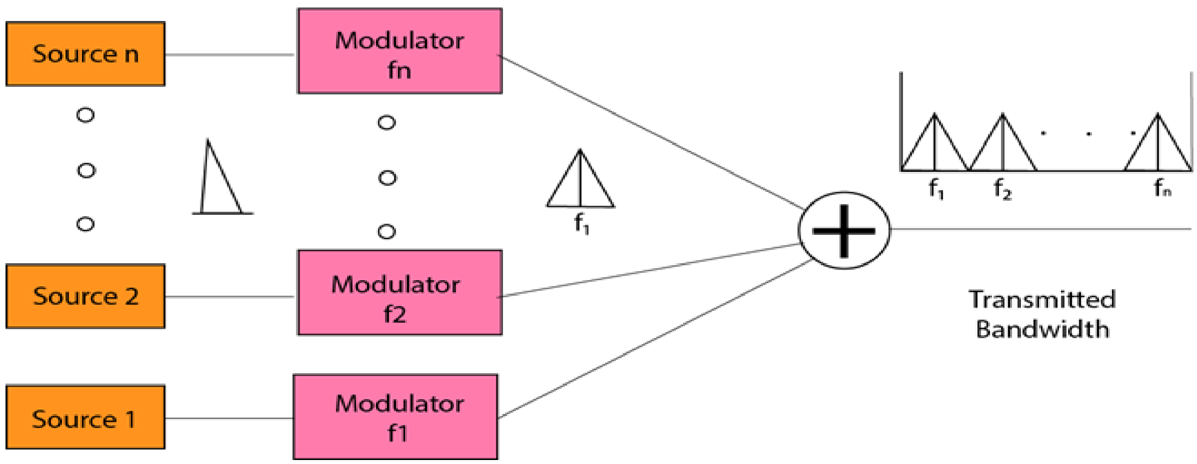


Advantages Of FDM:

- FDM process is very simple and easy modulation.
- A Large number of signals can be sent through an FDM simultaneously.
- It does not require any synchronization between sender and receiver.

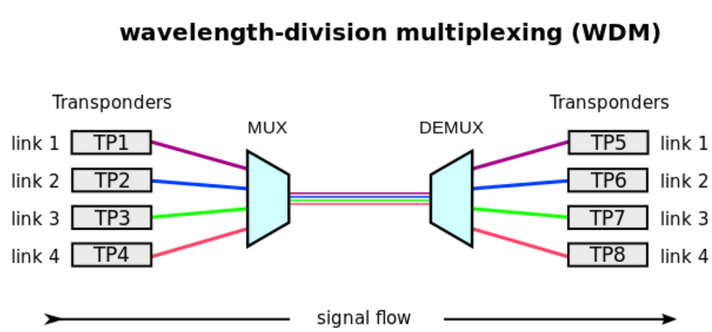
Disadvantages Of FDM:

- FDM technique is used only when low-speed channels are required.
- It suffers the problem of crosstalk.
- A Large number of modulators are required.
- It requires a high bandwidth channel.



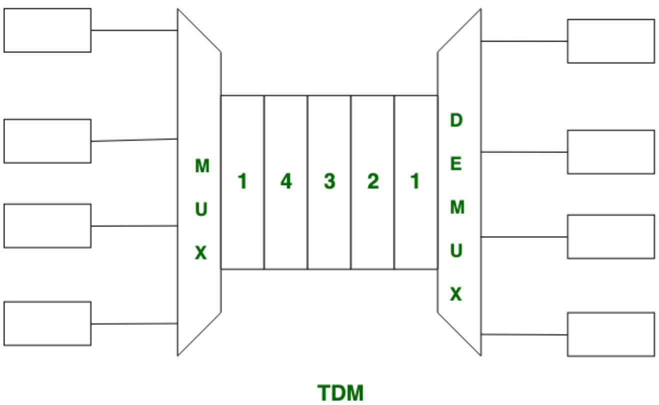
WDM:

- Wavelength Division Multiplexing is same as FDM except that the optical signals are transmitted through the fiber optic cable.
- WDM is used on fiber optics to increase the capacity of a single fiber.
- It is used to utilize the high data rate capability of fiber optic cable.
- The disadvantage of WDM is that it needs many fiber components and increases the failure probability.



TDM:

- TDM stands for Time division multiplexing.
- TDM works with digital signals as well as analog signals.
- TDM has low conflict.
- Wiring or chip of TDM is simple.
- TDM is efficient.
- In TDM, time sharing takes place.
- In TDM, synchronization pulse is necessary.



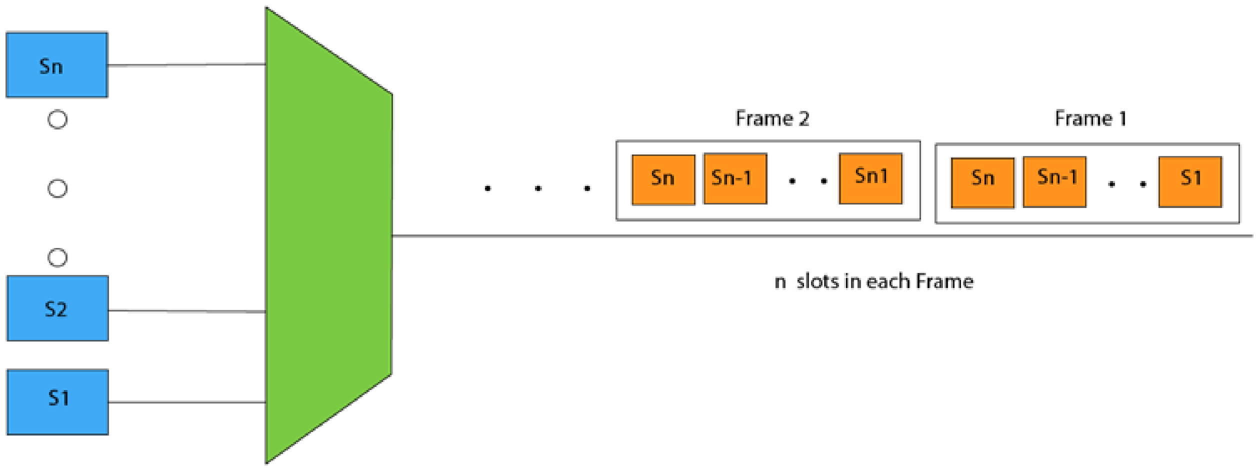
There are two types of TDM:

- Synchronous TDM
- Asynchronous TDM

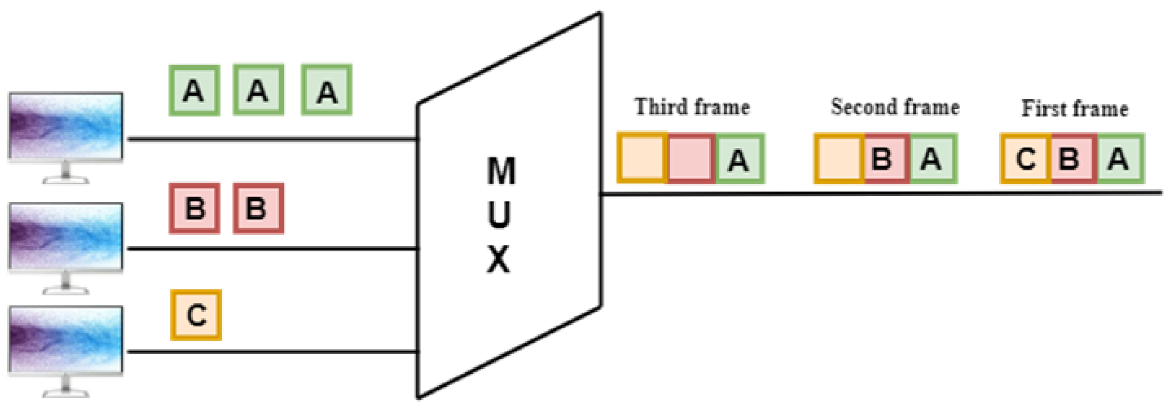
Synchronous TDM

- A Synchronous TDM is a technique in which time slot is preassigned to every device.
- In Synchronous TDM, each device is given some time slot irrespective of the fact that the device contains the data or not.
- If the device does not have any data, then the slot will remain empty.
- In Synchronous TDM, signals are sent in the form of frames. Time slots are organized in the form of frames. If a device does not have data for a particular time slot, then the empty slot will be transmitted.

- The most popular Synchronous TDM are T-1 multiplexing, ISDN multiplexing, and SONET multiplexing.
- If there are n devices, then there are n slots.



Concept Of Synchronous TDM



In the above figure, the Synchronous TDM technique is implemented. Each device is allocated with some time slot. The time slots are transmitted irrespective of whether the sender has data to send or not.

Asynchronous TDM

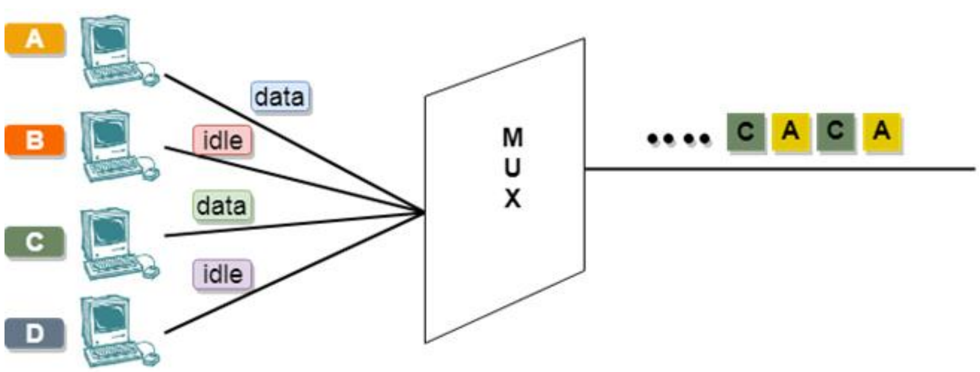
- An asynchronous TDM is also known as Statistical TDM.
- An asynchronous TDM is a technique in which time slots are not fixed as in the case of Synchronous TDM. Time slots are allocated to only those devices which have the data to send. Therefore, we can say that Asynchronous Time Division multiplexor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

- An asynchronous TDM technique dynamically allocates the time slots to the devices.
- In Asynchronous TDM, total speed of the input lines can be greater than the capacity of the channel.
- Asynchronous Time Division multiplexor accepts the incoming data streams and creates a frame that contains only data with no empty slots.
- In Asynchronous TDM, each slot contains an address part that identifies the source of the data.

ADDRESS	DATA

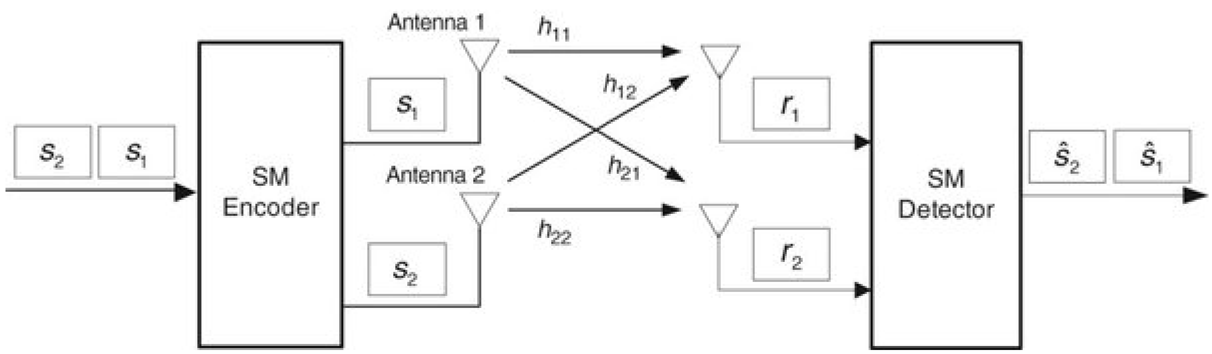
- The difference between Asynchronous TDM and Synchronous TDM is that many slots in Synchronous TDM are unutilized, but in Asynchronous TDM, slots are fully utilized. This leads to the smaller transmission time and efficient utilization of the capacity of the channel.
- In Synchronous TDM, if there are n sending devices, then there are n time slots. In Asynchronous TDM, if there are n sending devices, then there are m time slots where m is less than n (**m<n**).
- The number of slots in a frame depends on the statistical analysis of the number of input lines.

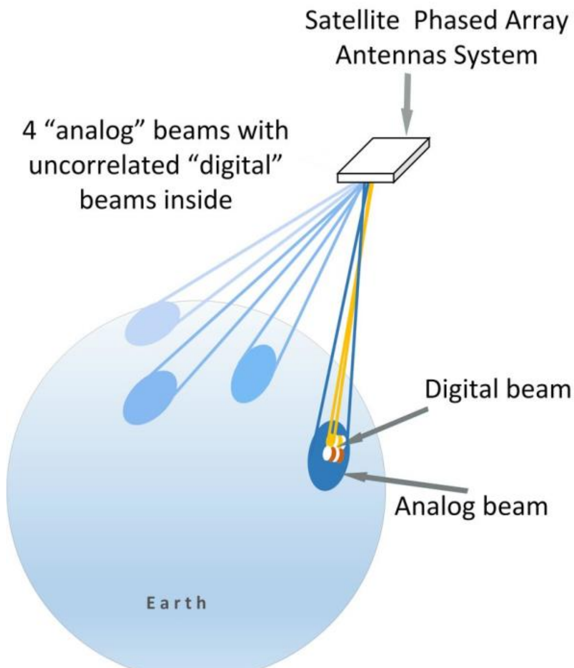
Concept Of Asynchronous TDM



SDM:

- **Space-division multiplexing (SDM).** Signal paths are spatially separated through the use of multiple conductors, such as optical fibers or electrical wires.
- The conductors are bundled into a single transport medium but are physically separated, with each conductor handling a transmitted channel.
- Individual conductors can be further multiplexed through the use of FDM, TDM or other techniques.
- SDM is often used in submarine cable systems to help increase capacity, but it can also be used for wireless communications.





Compare FDM, WDM, and TDM

FDM	WDM	TDM
1. The communication channel is divided by frequency.	1. The communication channel is divided by wavelength.	1. The communication channel is divided by time.
2. Analog technique.	2. Analog technique.	2. Digital technique.
3. Synchronization is not required.	3. Synchronization is not required.	3. Synchronization is required.
4. It requires complex circuitry at the transmitter and receiver.	4. It requires complex circuitry at the transmitter and receiver.	4. It does not require complex circuitry.
5. In FDM, the problem of crosstalk is severe.	5. In WDM, the problem of crosstalk is severe.	5. In TDM, the problem of crosstalk is not severe.
6. The channel bandwidth is effectively used.	6. The channel bandwidth is effectively used.	6. The channel bandwidth is wasted.
7. FDM stands for Frequency Division Multiplexing.	7. WDM stands for Wavelength Division Multiplexing.	7. TDM stands for Time Division Multiplexing.

Multiple Access:

- Access methods are multiplexing techniques that provide communications services to multiple users in a single-bandwidth wired or wireless medium. C

1. Frequency Division Multiple Access (FDMA):

- FDMA is a type of channelization protocol.
- In this bandwidth is divided into various frequency bands.
- Each station is allocated with band to send data and that band is reserved for particular station for all the time which is as follows:

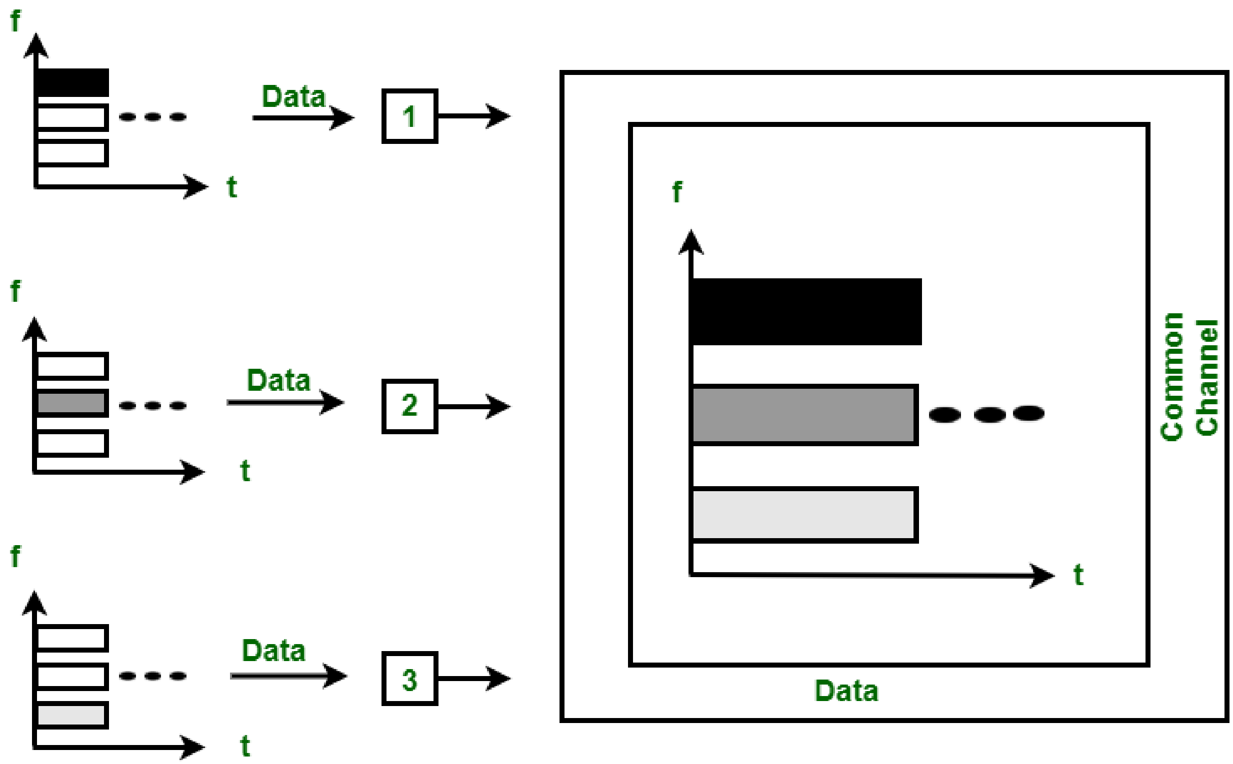


Figure – FDMA

- The frequency bands of different stations are separated by small band of unused frequency and that unused frequency bands are called as guard bands that prevents the interference of stations.
- It is similar to the data link layer access approach, in which the data link layer at each station instructs the physical layer to generate a bandpass signal from the data provided to it.

Features of FDMA:

- In FDMA, every user shares the frequency channel or satellite transponder simultaneously; however, every user transmits at single frequency.
- FDMA is compatible with both digital and analog signals.
- FDMA demands highly efficient filters in the radio hardware, contrary to CDMA and TDMA.
- FDMA is devoid of timing issues that exist in TDMA.
- As a result of the frequency filtering, FDMA is not prone to the near-far problem that exists in CDMA.
- All users transmit and receive at different frequencies because every user receives an individual frequency slot.

Advantages

1. In terms of hardware resources, it is very simple and easy to use.
2. Because FDMA is efficient, it can manage a smaller user population.
3. The system's complexity is modest, it reduces inter-symbol interference.

Disadvantages

1. It only works with analogue signals.
2. The transponders require extensive bandwidth.
3. The traffic's carrying capacity is not very high.
4. The highest bit rate per channel is small and fixed.

2. Time Division Multiple Access (TDMA):

- TDMA is the channelization protocol in which bandwidth of channel is divided into various stations on the time basis.
- There is a time slot given to each station, the station can transmit data during that time slot only which is as follows:

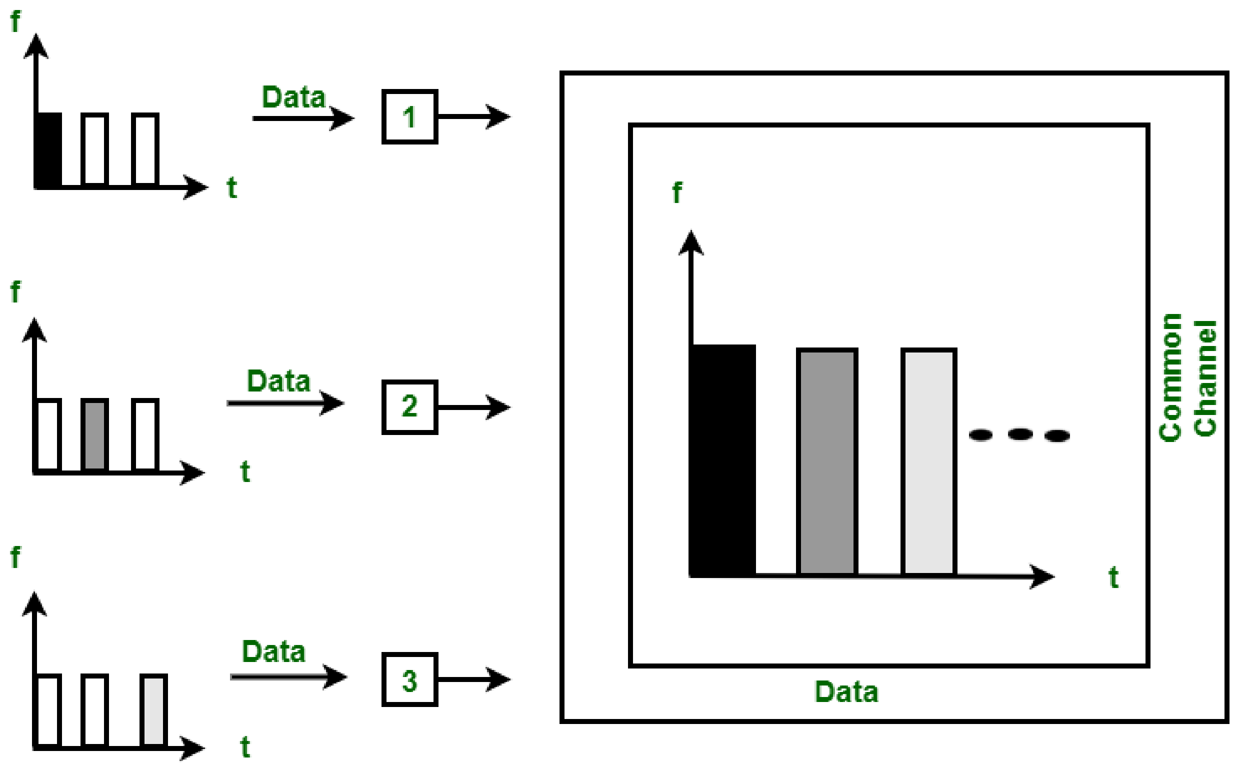


Figure – TDMA

- Here, Each station must aware of its beginning of time slot and the location of the time slot.
- TDMA requires synchronization between different stations.
- It is type of access method in the data link layer.
- At each station data link layer tells the station to use the allocated time slot.
- TDMA is a form of ***time-division multiplexing (TDM)*** in which numerous transmitters are connected to a single receiver rather than a single transmitter.

Features of TDMA

- Shares single carrier frequency with multiple users.
- Non-continuous transmission makes handoff simpler.
- Slots can be assigned on demand in dynamic TDMA.
- Less stringent power control than CDMA due to reduced intra cell interference.
- Higher synchronization overhead than CDMA.

Advantages

1. It may transmit data at speeds ranging from 64 kbps to 120 Mbps.
2. TDMA enables administrators to do administrations such as fax, voiceband information, SMS, and applications such as mixed media and video conferencing.
3. It extends the client's battery life by communicating alone for a part of the time during discussions.
4. TDMA may surely adapt to information transmission and voice correspondence.

Disadvantages

1. In TDMA, frequency/slot allocation will be complicated.
2. In TDMA, high data rates needed equalization.
3. Network and spectrum planning is a complex and time-consuming process requiring great expertise and resources.
4. The focus is on organization and range arranging.

3. Code Division Multiple Access (CDMA):

- In CDMA, all the stations can transmit data simultaneously.
- It allows each station to transmit data over the entire frequency all the time.
- Multiple simultaneous transmissions are separated by unique code sequence.
- Each user is assigned with a unique code sequence.

Features of CDMA:

- It allows more users to connect at a given time and thus provides improved data and voice communication capacity.
- A full spectrum is used by all the channels in CDMA.
- CDMA systems make the use of power control to eliminate the interference and noise and to thus improve the network quality.
- In CDMA systems all the cells can thus use the same frequency.

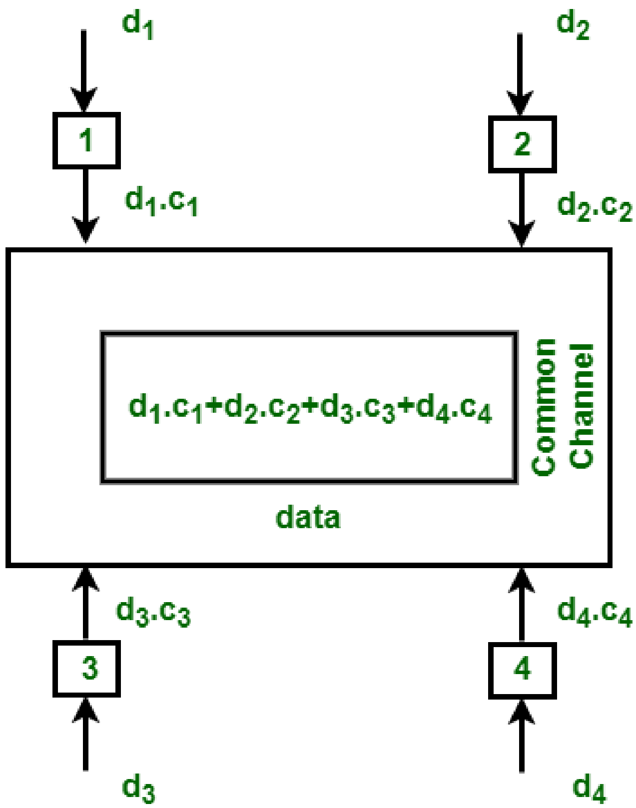


Figure – CDMA

- In the above figure, there are 4 stations marked as 1, 2, 3 and 4.
- Data assigned with respective stations as d_1, d_2, d_3 and d_4 and the code assigned with respective stations as c_1, c_2, c_3 and c_4.

Advantages

1. There is no need for synchronization.
2. CDMA channels are difficult to decode, so they enhance cellular communication security.
3. It offers better secure transmission.

Disadvantages

1. CDMA needed time synchronization.
2. As the number of users rises, the CDMA system's performance decreases.
3. It has a high price because of the greater equipment.

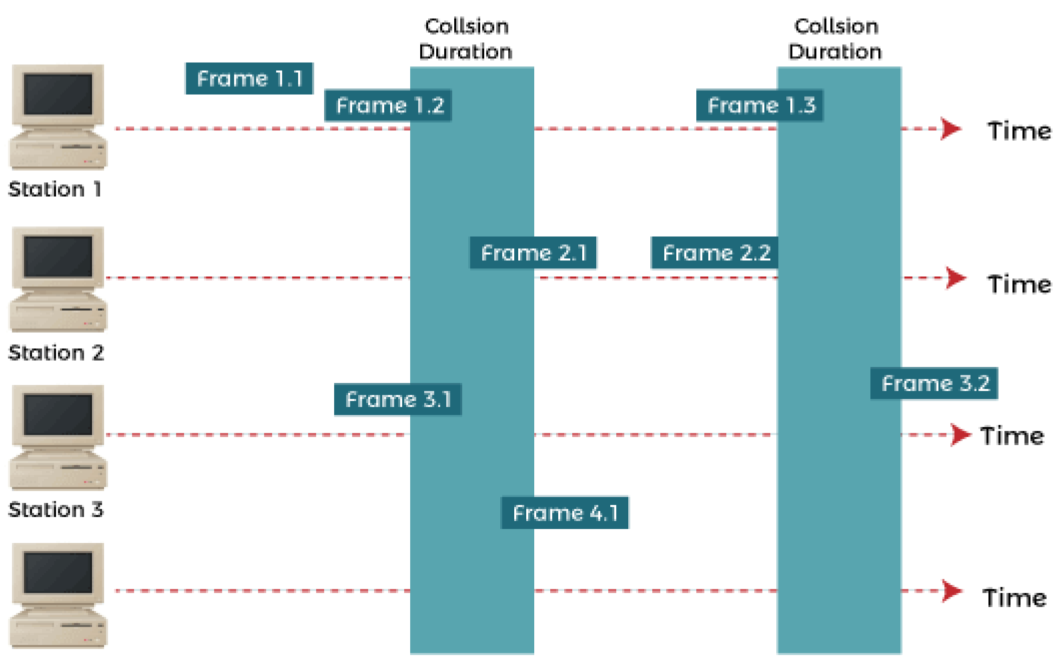
Features	FDMA	TDMA	CDMA
Full Forms	FDMA is an abbreviation for Frequency Division Multiple Access.	TDMA is an abbreviation for Time Division Multiple Access.	CDMA is an abbreviation for Code Division Multiple Access.
Flexibility	It has a little flexible.	It has moderate flexibility.	It has high flexibility.
Codeword	It doesn't require a codeword.	It also doesn't require a codeword.	It needs a codeword.
Rate of Data	It has a low data rate.	It has a medium data rate.	It has a high data rate.
Mode of Data transfer	It uses continuous signals for data transmission.	It uses signals in bursts for data transmission.	It uses digital signals for data transmission.
Synchronization	It doesn't need any synchronization.	It requires synchronization.	It also doesn't require any synchronization.
Cells Capacity	It has a limited cell capacity.	It also has a limited cell capacity.	It has no capacity restriction for a channel, although it is interference-limited.
Cost	It has a high cost.	It has a low cost.	Its installation cost is high, but the operational cost is low.
Guard times and Bands	It needed guard bands.	It needed guard times.	It needed both guard times and guard bands.

Aloha:

- Aloha is designed for wireless LAN (Local Area Network) but can also be used in a shared medium to transmit data.
- In aloha, any station can transmit data to a channel at any time.
- It does not require any carrier sensing.

Pure Aloha

- Pure aloha is used when data is available for sending over a channel at stations.
- In pure Aloha, when each station transmits data to a channel without checking whether the channel is idle or not, the chances of collision may occur, and the data frame can be lost.



Frames in Pure ALOHA

- When a station transmits the data frame to a channel without checking whether the channel is free or not, there will be a possibility of the collision of data frames.
- Station expects the acknowledgement from the receiver, and if the acknowledgement of the frame is received at the specified time, then it will be OK; otherwise, the station assumes that the frame is destroyed.

- Then station waits for a random amount of time, and after that, it retransmits the frame until all the data are successfully transmitted to the receiver.

Efficiency-

Efficiency of Pure Aloha (η) = $G \times e^{-2G}$

where G = Number of stations willing to transmit data

Maximum Efficiency-

For maximum efficiency,

- We put $d\eta / dG = 0$
- Maximum value of η occurs at $G = 1/2$
- Substituting $G = 1/2$ in the above expression, we get-

Maximum efficiency of Pure Aloha

$= 1/2 \times e^{-2 \times 1/2}$

$= 1 / 2e$

$= 0.184$

= 18.4%

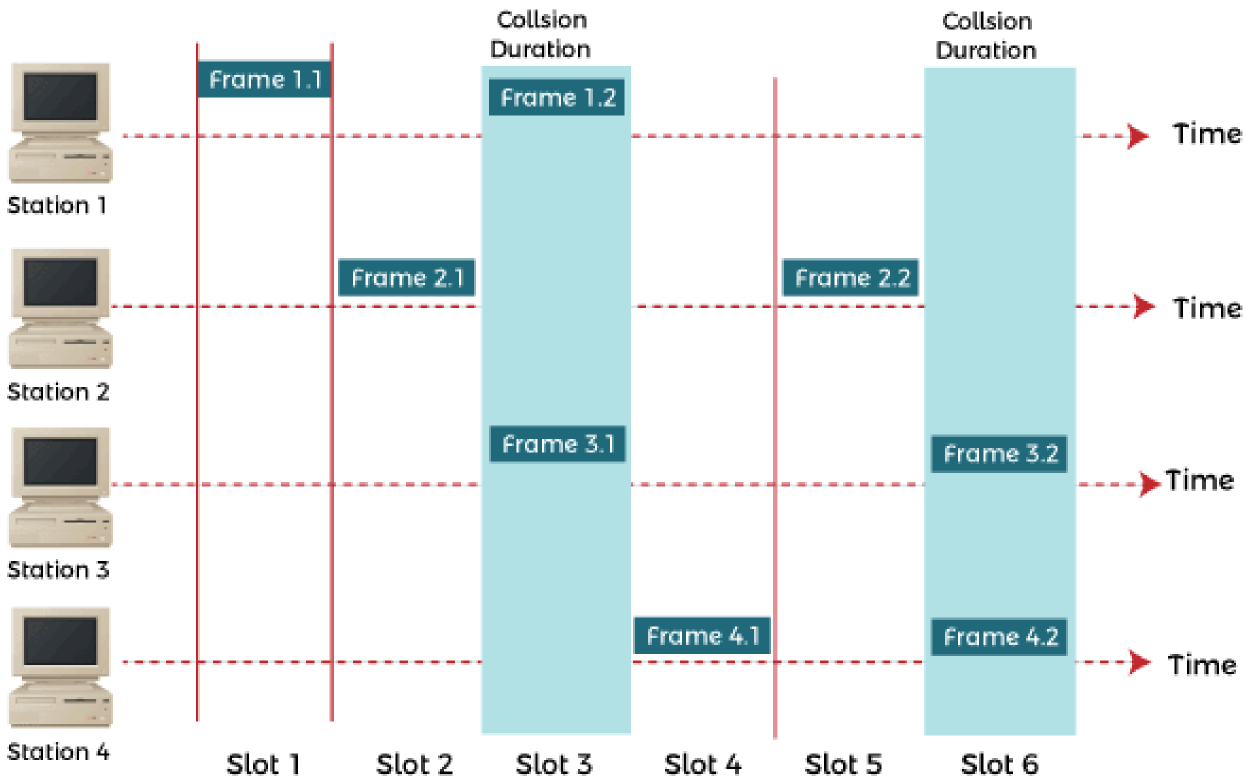
Thus,

Maximum Efficiency of Pure Aloha (η) = 18.4%

The maximum efficiency of Pure Aloha is very less due to large number of collisions.

Slotted Aloha

- There is a high possibility of frame hitting in pure aloha, so slotted aloha is designed to overcome it.
- Unlike pure aloha, slotted aloha does not allow the transmission of data whenever the station wants to send it.



Frames in Slotted ALOHA

- In slotted Aloha, the shared channel is divided into a fixed time interval called slots.
- So that, if a station wants to send a frame to a shared channel, the frame can only be sent at the beginning of the slot, and only one frame is allowed to be sent to each slot.
- If the station is failed to send the data, it has to wait until the next slot.
- However, there is still a possibility of a collision because suppose if two stations try to send a frame at the beginning of the time slot.

Efficiency-

Efficiency of Slotted Aloha (η) = G x e^{-G}

where G = Number of stations willing to transmit data at the beginning of the same time slot

Maximum Efficiency-

For maximum efficiency,

- We put $d\eta$ / dG = 0
- Maximum value of η occurs at G = 1
- Substituting G = 1 in the above expression, we get-

Maximum efficiency of Slotted Aloha

= 1 x e^{-1}

= 1 / e

= 0.368

= 36.8%

Thus,

Maximum Efficiency of Slotted Aloha (η) = 36.8%

The maximum efficiency of Slotted Aloha is high due to less number of collisions.

Difference Between Pure Aloha And Slotted Aloha-

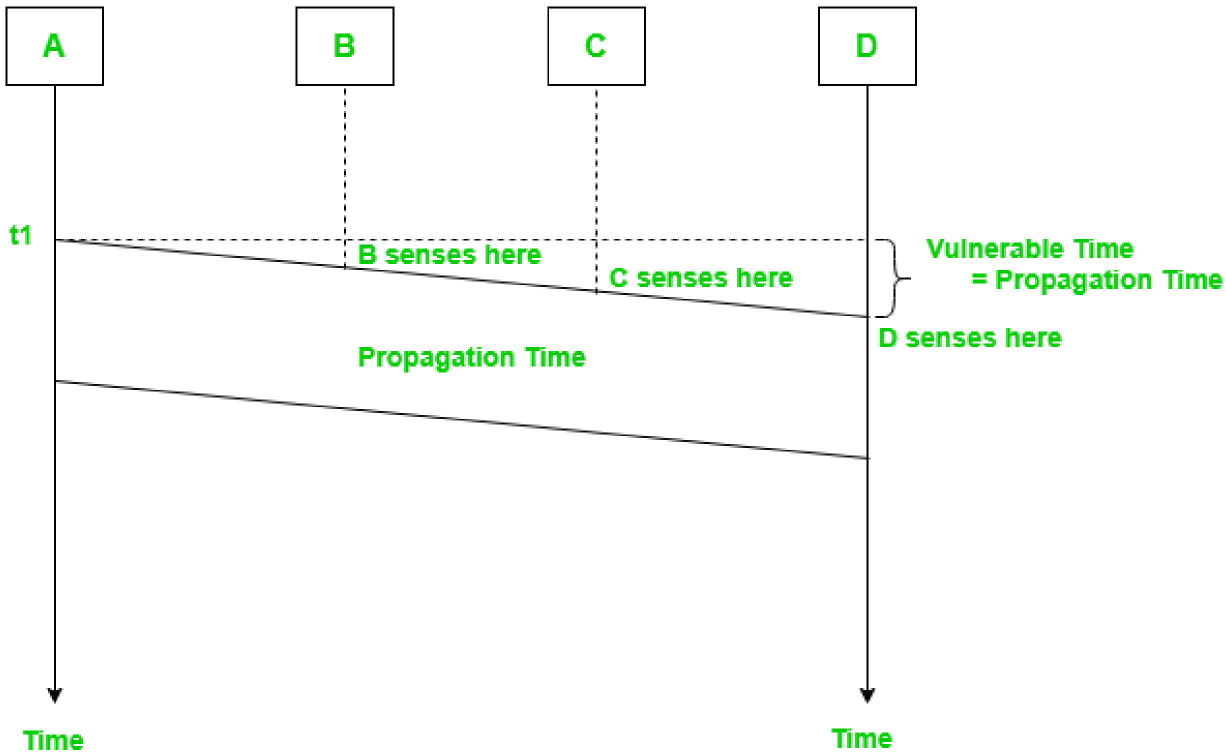
Pure Aloha	**Slotted Aloha**
Any station can transmit the data at any time.	Any station can transmit the data at the beginning of any time slot.
The time is continuous and not globally synchronized.	The time is discrete and globally synchronized.
Vulnerable time in which collision may occur $= 2 \times T_t$	Vulnerable time in which collision may occur $= T_t$
Probability of successful transmission of data packet $= G \times e^{-2G}$	Probability of successful transmission of data packet $= G \times e^{-G}$
Maximum efficiency = 18.4% (Occurs at G = 1/2)	Maximum efficiency = 36.8% (Occurs at G = 1)
The main advantage of pure aloha is its simplicity in implementation.	The main advantage of slotted aloha is that it reduces the number of collisions to half and doubles the efficiency of pure aloha.

Carrier Sense Multiple Access (CSMA)

- This method was developed to decrease the chances of collisions when two or more stations start sending their signals over the data link layer.
- Carrier Sense multiple access requires that each station **first check the state of the medium** before sending.

Vulnerable Time:

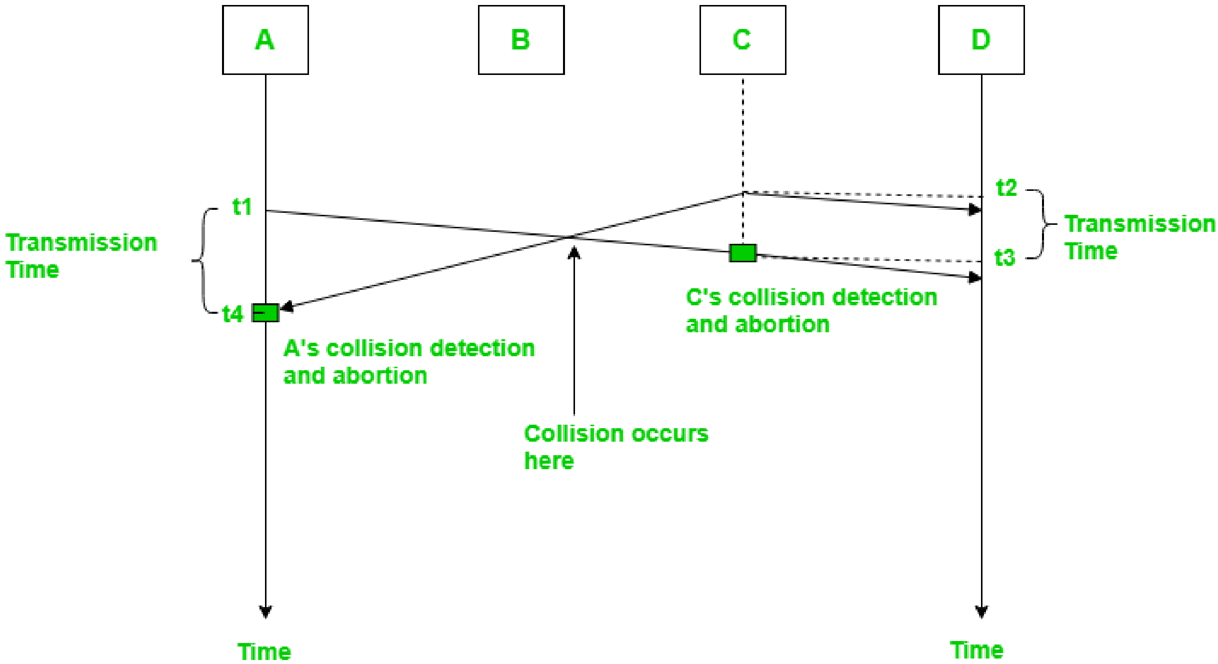
Vulnerable time = Propagation time (Tp)



- The persistence methods can be applied to help the station take action when the channel is busy/idle.

1. Carrier Sense Multiple Access with Collision Detection (CSMA/CD):

- In this method, a station monitors the medium after it sends a frame to see if the transmission was successful.
- If successful, the transmission is finished, if not, the frame is sent again.

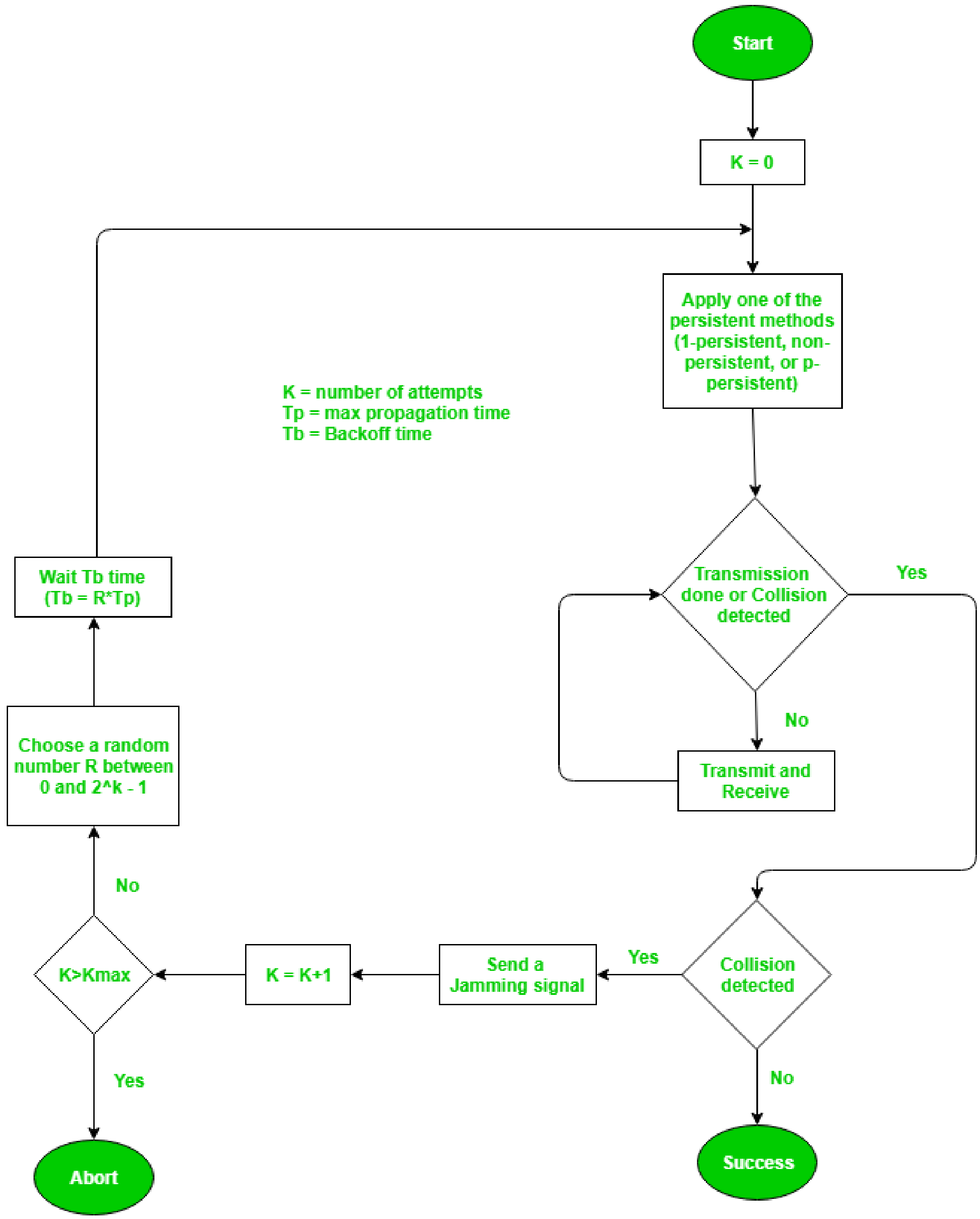


- In the diagram, *starts* sending the first bit of its frame at t1 and since C sees the channel idle at t2, starts sending its frame at t2.
- C detects A's frame at t3 and aborts transmission. A detects C's frame at t4 and aborts its transmission.
- Transmission time for C's frame is, therefore, t3-t2 and for A's frame is t4-t1
- So, the **frame transmission time (Tfr) should be at least twice the maximum propagation time (Tp)**. This can be deduced when the two stations involved in a collision are a maximum distance apart.

The algorithm of CSMA/CD is:

- When a frame is ready, the transmitting station checks whether the channel is idle or busy.
- If the channel is busy, the station waits until the channel becomes idle.
- If the channel is idle, the station starts transmitting and continually monitors the channel to detect collision.
- If a collision is detected, the station starts the collision resolution algorithm.
- The station resets the retransmission counters and completes frame transmission.

Process: The entire process of collision detection can be explained as follows:



Throughput and Efficiency: The throughput of CSMA/CD is much greater than pure or slotted ALOHA.

- For the 1-persistent method, throughput is 50% when G=1.
- For the non-persistent method, throughput can go up to 90%.

Characteristics of CSMA/CD

1)The device with the electronic token is the only one that can transmit after a collision.
2) A device listens and waits until the media is not busy before transmitting.
3) After detecting a collision, hosts can attempt to resume transmission after a random time delay has expired.
4) All of the devices on a segment see data that passes on the network medium.
5) A jam signal indicates that the collision has cleared and the media is not busy.
6) Devices can be configured with a higher transmission priority.

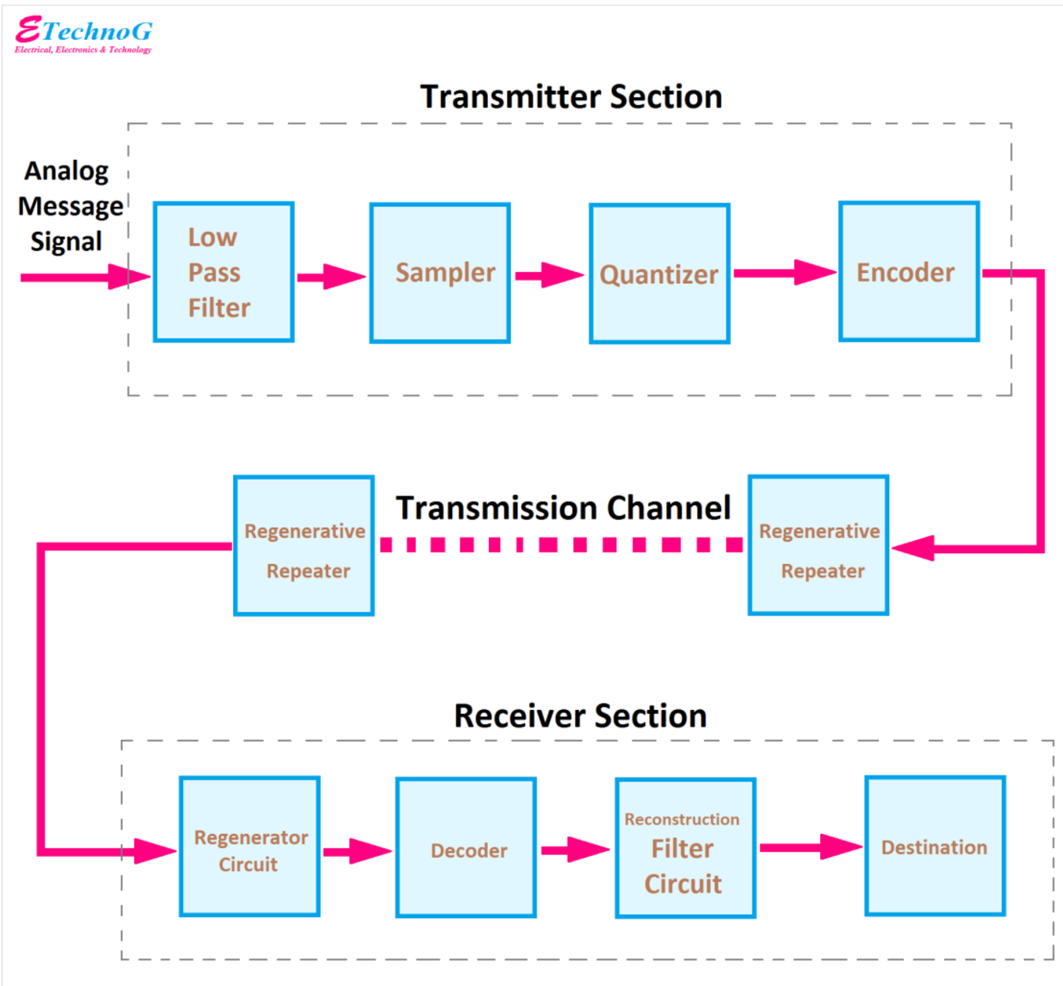
What do we mean by contention system?

- Contention is **a method of line control in which the terminals request to transmit**.
- **A communication system in which multiple users share a common channel in a way that can lead to conflicts**.

Chapter 3: PCM

What is PCM?

- PCM (**Pulse code modulation**) is a method used to converter analog signals, such as a telephone call, to digital signals, In PCM, an analog signal is sampled and converted to a series of binary bits.
- In pulse code modulation, the analog message signal is first sampled, and then the amplitude of the sample is approximated to the nearest set of quantization level. This allows the representation of time and amplitude in a discrete manner. Thereby, generating a discrete signal.



Block Diagram of PCM System

PCM Working Principle
If we know the function of each part of the system then we can easily understand the working principle of PCM. Now let's see the important parts or blocks of the PCM system are,

1. **Low Pass Filter**
As its name suggests, the low pass filter will pass only low-frequency signals. So, the analog signal is applied to the input of the low pass filter and the low pass filter will block the high-frequency signal and pass the low-frequency signal to the next stage. A frequency is set as per the frequency of the message signal in the filter circuit with respect to which the signal will be cut off.

2**. Sampler**
The sampler circuit helps to take sample data from the message signal at the instantaneous value. It can be called a process of cutting the continuous signal into discrete from. The low-frequency analog signal comes from the filter circuit to the sampler circuit and it is sampled at regular intervals. Generally, the sampling rate or sample frequency is twice than the highest frequency component of the message signal. So, **$fs \geq 2fm$**

Here, Fs= sampled frequency, and Fm = message signal frequency

3. Quantizer
The quantizer is a circuit that does quantizing or compress or reduces the no of samples. It removes the excessive bits from the sampled signal provided by the sampler circuit. Basically, it confines the data and makes rounds off each sample. Actually, it gives the shape of digital bis or digital pulse from the sinusoidal shape. For example, if the voltage is 5V then it is considered as high but if the voltage is 0V, then it is considered as low. So the quantizer can make the 4.8V to 5V or 0.5V to 0V or remove the 3V signals. There are two types of quantizing processes - uniform quantizing and non-uniform quantizing. In the uniform quantizing process there is a uniform spacing between the levels whereas, in the non-uniform quantizing, the spacing between the levels is non-uniformed.

4. Encoder
The encoder circuit converts the quantizing signal into binary codes. So this unit generates the actual digital encoded signals. This digital signal is consists of binary pulses and they act as modulated output from the transmitter section of the whole system.

5. Regenerative Repeater

The regenerative repeater is placed in both ends of the transmission channel sending and receiving end. at the sending end, it makes ready the signal for long-distance transmission. And at the receiving end, it removes noise from the signal and gives reshape to the signal to get the actual signal. There may be multiple numbers of regenerative repeaters for long-distance transmission.

6. Transmission Channel

A transmission channel is a medium or path for data transmission. Through this channel or medium data or signal is transmitted.

7. Regenerator Circuit

At the receiving end of the transmission channel, the regenerator circuit is used. It also works the same as the regenerator repeater. It removes the noise and distortion from the signal received from the transmission channel and generates the actual signal sent.

8. Decoder

The decoder circuit converts the received digitally coded binary signal into the original signal. This circuit also demodulates the signal. The generative circuit and decoder both are responsible to convert the received digital signal into an analog signal.

9. Reconstruction Filter

The filter at the receiving end of the system is known as the reconstruction filter. The reconstruction filter provides the actual original signal sent by the sender.

What is T1 lines?

- **T1 lines in used for north American standard.**
- The T1 line uses two wire pairs (one for sending, one for receiving) and time division multiplexing (TDM) to interleave 24channels 64-Kbps voice or data.
- The standard T1 frame is 193 bits, which holds 24 8-bit voice samples and one synchronization bit with 8,000 frames transmitted per second.

- T1 lines are widely used to connect an organization's PBX to the telephone company or to connect a local network (LAN) to an Internet provider (ISP).

- They are also used to access the internet in buildings without DSL, cable or fixed wireless coverage.

The basics E1 Carrier features are shown as below:

● Transmission of signals at the rate of 1.544 Mbps, here is how it is calculated:

◆ T1 carries 24 channels
◆ 24 channel for voice/data + It has 1framing bit
◆ 1 Channel = 8bits
◆ Total bit= (8 *24) +1=192+1=193
◆ Sampling frequency = 8KHz
◆ Data rate per channel = 8*8000 =64Kbps
◆ (64*24=1536) +(1*8=8) =1536+8=1544=1.544Mbps

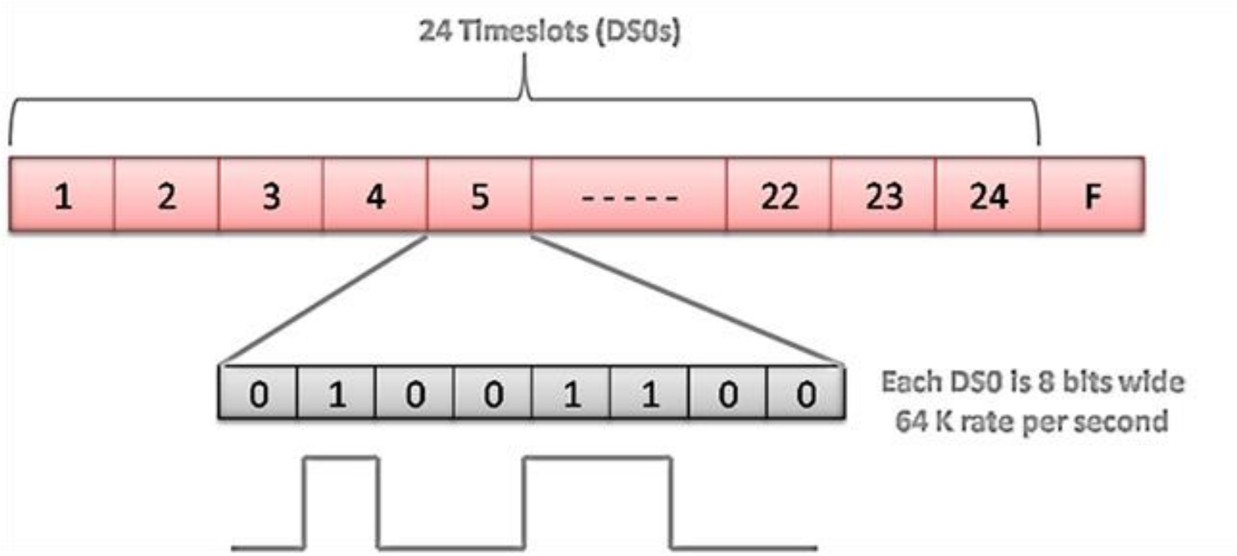
● T1 media types including twisted pair cable, coaxial cable, microwave radio, fiber optic cable, and satellite.
● A T1 line is point-to-point. T1 lines may be used fractionally or at their full bandwidth.
● T1 frame consists of 24 timeslots.

T1 Frame Types

The T1 interface supports 4 different frame structures, dictated by the mode of operation, they are Frame, Super Frame (SF), Extended Super Frame (ESF) and Unframed.

Frame

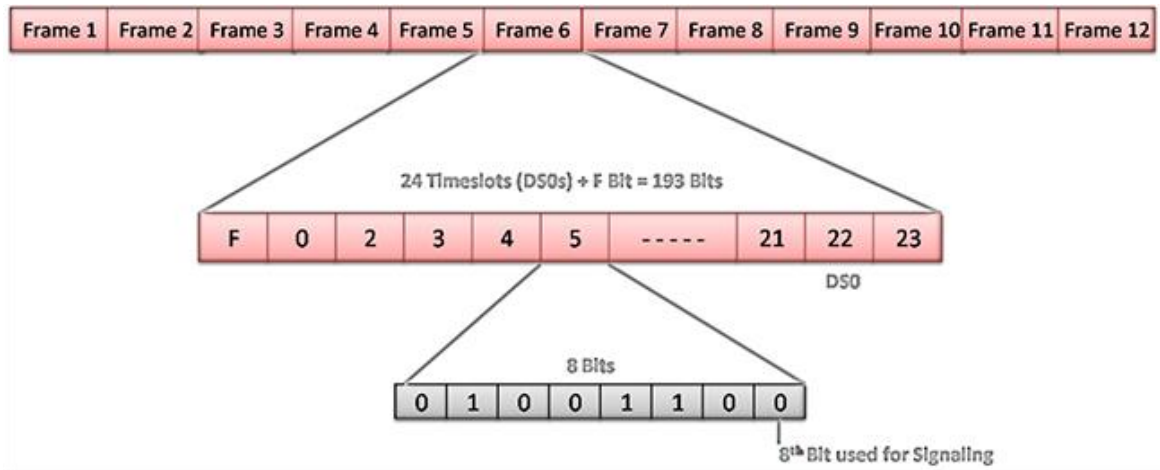
- A basic T1 frame DS1 (digital signal-level one) contains 24 DS0 (64kbps) time slots, numbered from 1 to 24, each time slot has 8 bits, a total of 192 bits.
- The T1 basic frame also includes an F bit (framing bit), which is used as a frame synchronization bit to indicate the end of the current frame and the beginning of the next frame.
- The transmission rate of DS1=193*8k= 1.544 Mbps.



● T1 frame is constructed of 24 timeslots (each timeslot is of 8 bits) plus one framing bit added.
● Total frame length is 193 bits.
● Each TS is regarded as a channel of 64 kbit/s bandwidth.
● Framing bit creates a channel of 8kbit/s and is used for messages, synchronization, and alarms.
● A frame is the basic building block for the SF and the ESF.

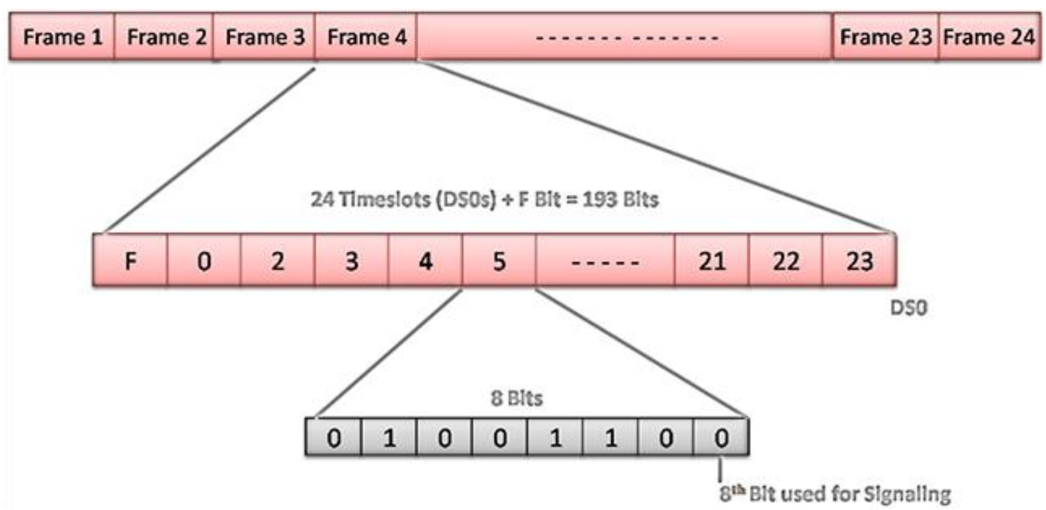
Super Frame (SF)

The SF frame format (also known as the D4 frame format) is the most commonly used format in the current public switched telephone network. Each SF consists of 12 basic DS1 frames. The 193th bit of each frame is used as the control bit, and the 12th 193bit of the SF Combine them to form a 12-bit control word (e.g. 100011011100) to provide frame synchronization and signaling management information. The odd bits of the 12-bit control word of the SF frame (called the Ft bit, and the corresponding frame is called the terminal frame) are used to mark the frame and super frame boundary so that the receiving device can correctly process user data; the even bits of the control word (called Fs) Bit, the corresponding frame is called signaling frame) used to carry signaling flags.



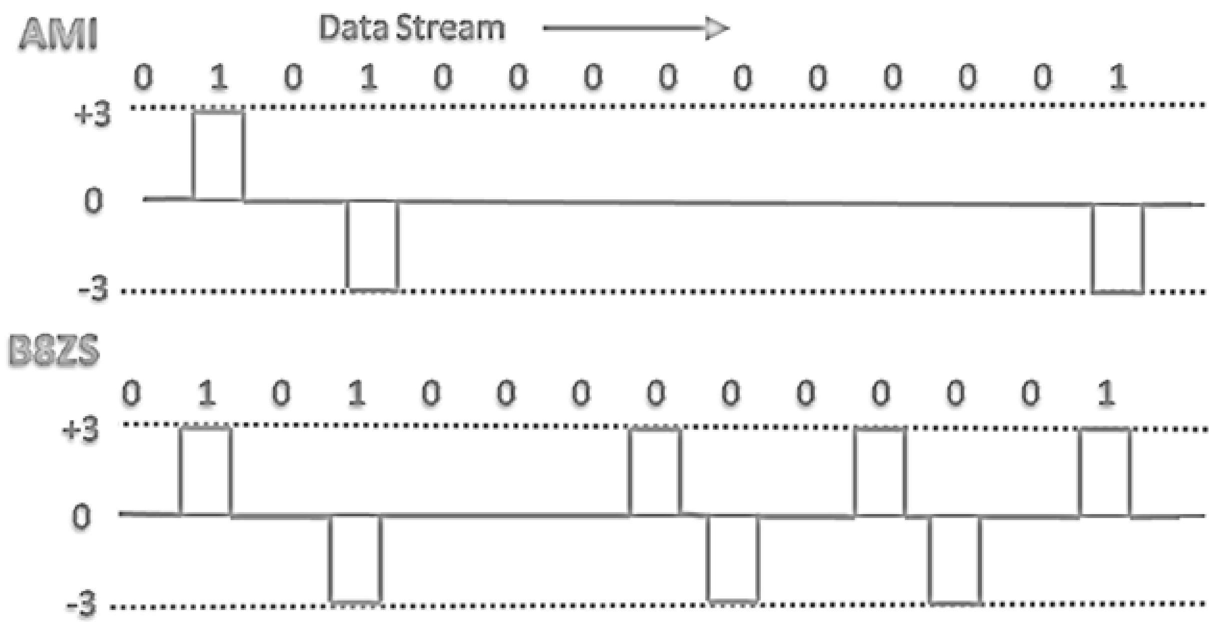
Extended Super Frame (ESF)

The ESF frame format extends the SF frame mode from 12 frames to 24 frames, a total of 193* 24 = 4632 bits. ESF and SF frame format are basically the same.



T1 Line Encoding

T1 commonly used line codes include B8ZS code and AMI code. B8ZS is the abbreviation of Bipolar with 8-Zero Substitution, mainly to solve the defect that the timing signal cannot be extracted when the long 0 appears in the AMI coding format.



What is E1 Line

- **E1 lines is used for European standard.**
- The E1 line uses two wire pairs (one for sending, one for receiving) and time division multiplexing (TDM) to transmit 32channels 64-Kbps voice or data.
- it is a 2.048 Mbps point-to-point dedicated, digital circuit provided by the telephone companies in Europe.
- E1 and T1 lines can be interconnected for international use.

The basics E1 Carrier features are shown as below:

● E1 is a digital communication link that enables the transmission of voice, data, and video signals at the rate of 2.048 million bits per second (Mbps). here is how it is calculated:

◆ E1 carries 30+2 channel.
◆ 30 channel for voice/data, 2 channel for controlling,

- ◆ 1channel=8 bits
- ◆ Total bit=8*32=256
- ◆ Sampling frequency=8kHz
- ◆ Date rate per channel= 8*8000=64Kbps
- ◆ 32*64=2048=2.048Mbps

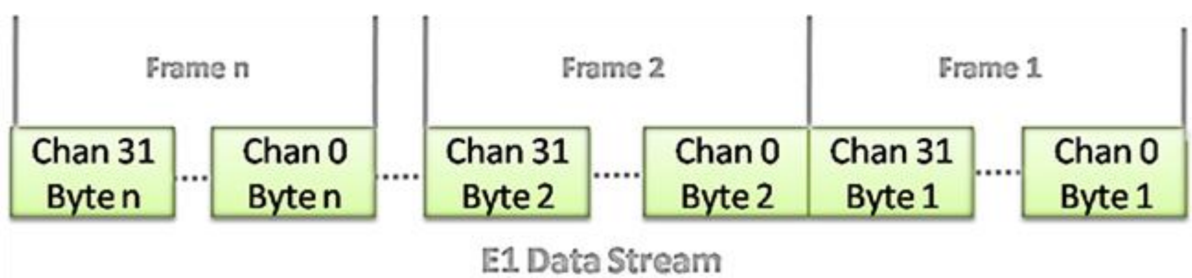
● Deployed primarily in Europe and Asia.
● E1 frame consists of 32 timeslots.
● E1 specifications defined in CCITT Recommendation G.704, although Recommendation G.732 supplements G.704.
● Data is sent over one signal pair and simultaneously received on another signal pair. (full duplex transmission).
● Before the data is output to the E1 line, it must be conditioned by the line driver to meet the electrical characteristics of the E1 span (pulse width, pulse height, and pulse voltages).
● Line driver converts the unipolar signal output from the multiplexer into a bipolar signal (each successive digital 1 has the opposite polarity of the previous one).
● E1 transmission uses Bi-polar Return to Zero (BRZ) framing format.

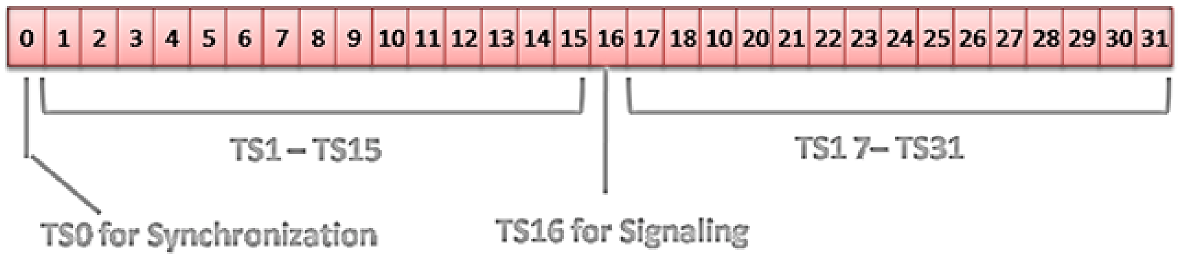
E1 Frame Types

E1 support various modes, and all use 2048 Kb/s, including Unframed (UNF), Framed (FR) and Multi-Framed (MF).

Framed (FR)

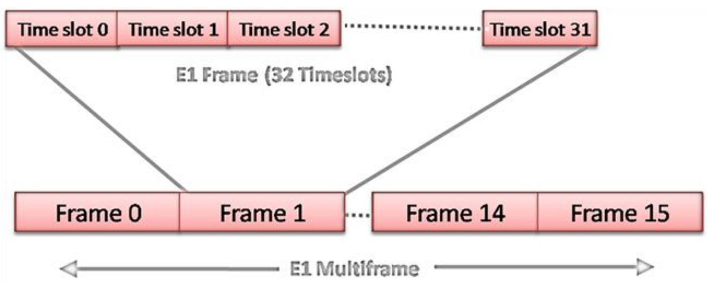
In the Framed mode, all 32 slots are used for data, detection of boundaries is gained with TS0.





Multi-Framed (MF)

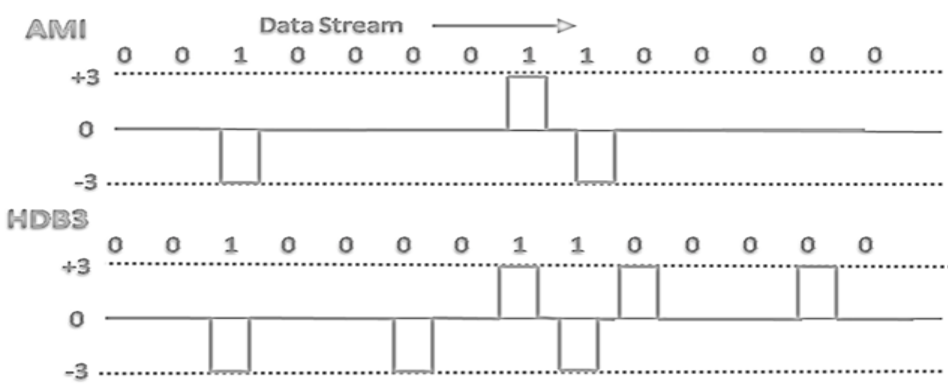
In the Multi-Framed mode, TS0 is used for synchronization, all other channels are unaffected.



The E1 multiframe consists of 16 consecutive E1 frames, the extra features to multi-frame is the addition of a Cyclic Redundancy Check (CRC), and Channel Associated Signaling (CAS).

E1 Line Encoding

Commonly used line codes for E1 include HDB3 code and AMI code.



Common characteristics	E1 and T1
Sampling frequency	8kHz
Number of samples per telephone signal	8000 per second
Length of PCM frame	1/b = 1/8000/s = 125µs
Number of bits in each code word	8
Telephone channel bit rate	b x d = 8000/s x 8 bit = 64kbit/s

Differing characteristics	E1	T1
Encoding/decoding Number of segments in characteristic	A-law 13	µ-law 15
Number of timeslots per PCM frame	32	24
Number of bits per PCM frame (* signifies an additional bit)	d x g = 8 x 32 = 256 bits	d x g + 1* = 8 x 24 + 1* = 193 bits
Length of an 8-bit timeslot	(c x d)/h = (125µs x 8)/256 = approx. 3.9µs	(c x d)/h = (125µs x 8)/193 = approx. 5.2µs
Bit rate of time-division multiplexed signal	b x h 8000/s x 256 bits 2048kbit/s	b x h 8000/s x 193 bits 1544kbit/s

if more information needed of t1 and e1 lines go through:
https://www.rfwireless-world.com/Terminology/T1-frame-vs-E1-frame.html

T1 lines VS E1 lines details:

- T1 frame vs E1 frame structure and mentions difference between T1 and E1 frame structure used in PCM telecommunication systems.
- It mentions number of bits, channels and bit rates used in E1 and T1 frame structure.
- Both T1 and E1 frames are related to PCM 24 channels and 30 channels respectively.
- T1 supports 1.544 Mbps data rate and E1 supports 2.048 Mbps data rate.
- T1 is widely used to transmit telephone calls between local central offices over distances of about 5 to 50 miles.

For long distance transmission, T1 signals are multiplexed into other faster TDM systems.

T1 frame structure used in PCM telephone system

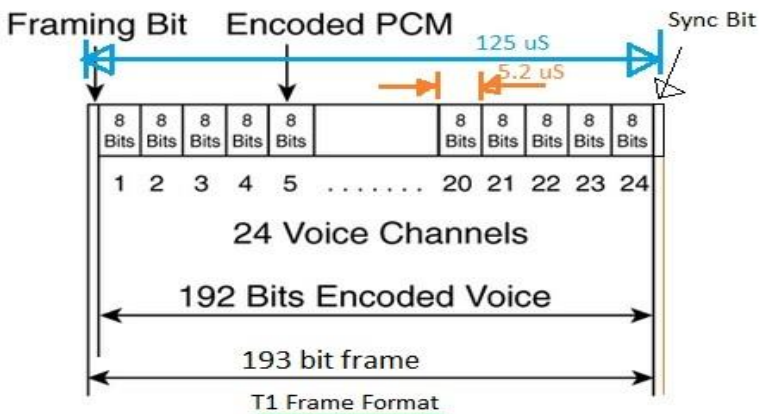
- 

Figure-1: T1 Frame Structure

The figure-1 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.
- The samples are converted to serial digital words by ADC.
- These serial digital words produced from total 24 channels are transmitted one after the other.
- Each sample is an 8-bit word, 7 bits of magnitude and 1 bit representing polarity.
- Channel Sampling rate is 125µS/24 = 5.2 µS per cycle or 192 KHz.
- This represents a total of 24 (8) = 192 bits.
- 1 more bit is added as sync pulse.
- This 193rd bit will help in synchronization between transmission and reception.

E1 frame structure used in PCM telephone system

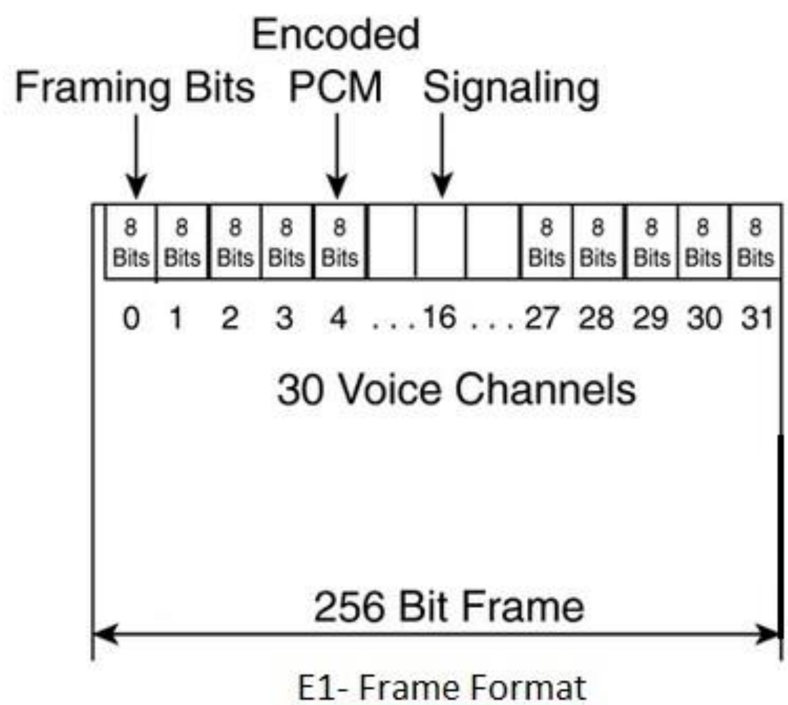


Figure-2: E1 Frame Structure

- The figure-2 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 accommodates 30 voice channels including two additional 64 kbps channels.
- These two additional channels are used to transmit synchronization, service and signaling data. T
- his results into E1 transmission rate of 2048 kbps = 32 * 64 kbps.
- Each **E1 frame** consists of 256 bits. These bits are arranged in 32 timeslots of 8 bits each. The frame repetition rate is 8,000 per second. Hence data rate supported by each timeslot is 64 kbps. The number of timeslots available for user data is maximum 31 as timeslot 0 is always reserved as explained below.
- Each 8-bit sample or data word occupies a Timeslot in the E1 stream. A cycle of 32 timeslots repeats every 125 us. A complete cycle of the 32 timeslots is called a E1-Frame. The timeslots in a frame are numbered TS0 to TS31. The first timeslot sent (i.e., TS0) contains synchronization and service data. The timeslots that carry user data are called payload.

  A fixed 7-bit pattern "0011011" known as FAS (i.e., Frame Alignment Signal) is transmitted in timeslot 0 in each even frame. The second bit of timeslot 0 alternates between 0 and 1. When the receiver detects this pattern it achieves frame alignment. If three or more incorrect FAS patterns are received in a row, the frame alignment is lost.

  When frame alignment can't be achieved or no signal is received by the E1 equipment it transmits RAI (Remote Alarm Indication) by setting the A bits of the TS0 to 1 (see table 6) or by sending all ones pattern called AIS (Alarm Indication Signal). The AIS is usually sent by network equipment. Also refer difference between CAS and CCS➤➤ which describes Channel Associated Signaling and Common Channel Signaling.

As explained above, T1 and E1 are PCM frame structures. Let us compare T1 vs E1 and summarize difference between T1 and E1 frame structures.

➤T1 supports 1.544 Mbps data rate where as E1 supports 2.048 Mbps.

➤T1 is used over distances of about 5 to 50 miles. E1 cover longer than T1.

➤T1 offers 24 voice channels where as E1 offers 30 voice channels.

➤T1 frame structure contains 193 bits, E1 frame structure contains 256 bits.

Explain the Companding process in PCM

- Companding is the process of compression and then expansion.
- With Companding system, the higher amplitude analog signals are compressed (amplified less than lower amplitude signals) prior to transmission and then expanded (amplified more than the lower amplitude signals) in the receiver.

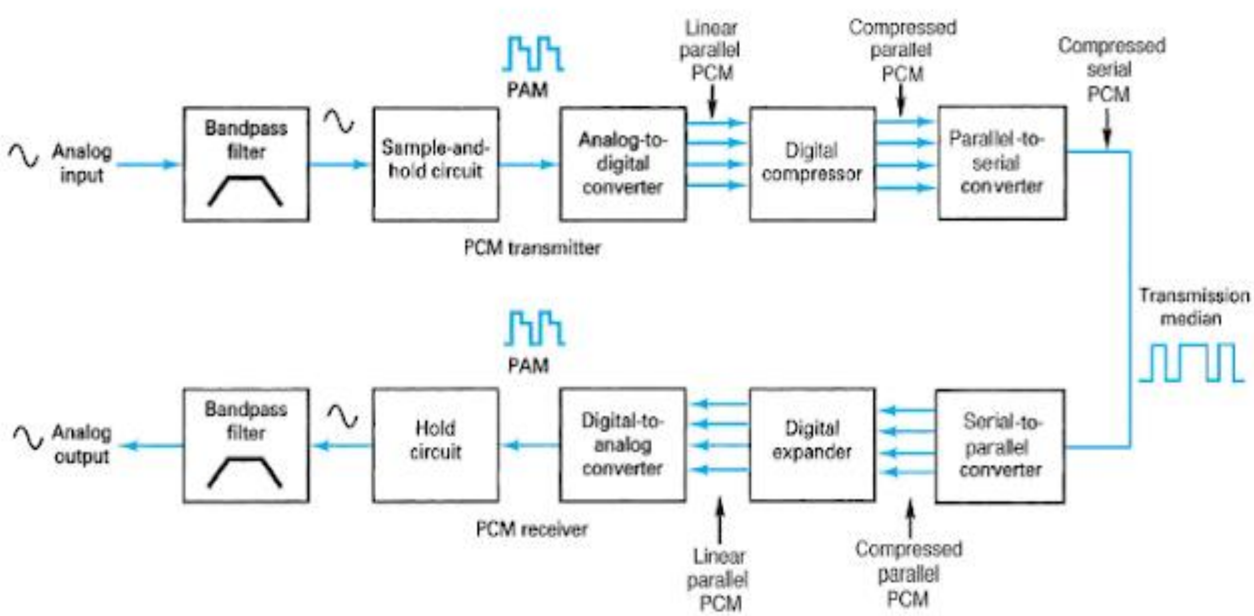


Fig. 1 Block diagram of Companded PCM system

- Companding is used in pulse code modulation (PCM). The process involves decreasing the number of bits used to record the strongest (loudest) signals.
- In the digital file format, Companding improves the signal-to-noise ratio at reduced bit rates. For example, a 16-bit PCM signal may be converted to an eight-bit ".wav" or ".au" file.
- As we know in non-uniform quantization, the step size varies according to the signal level.
- If the signal level is low then step size will be small. So, the step size will be low for weak signal.
- Thus, the quantization noise will also be low.
- So, in order to maintain proper signal to quantization noise ratio, the step size must be variable according to the signal level.
- Thus, in order to achieve non-uniform quantization, the process of Companding is used.

There are two types of Companding techniques. They are −

A-law Companding Technique

- Uniform quantization is achieved at A = 1, where the characteristic curve is linear and no compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law Companding is used for PCM telephone systems.

µ-law Companding Technique

- Uniform quantization is achieved at µ = 0, where the characteristic curve is linear and no compression is done.
- µ-law has mid-tread at the origin. Hence, it contains a zero value.
- µ-law Companding is used for speech and music signals.
- µ-law is used in North America and Japan.

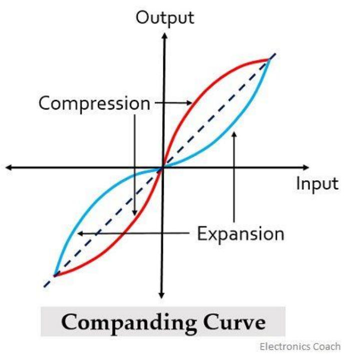
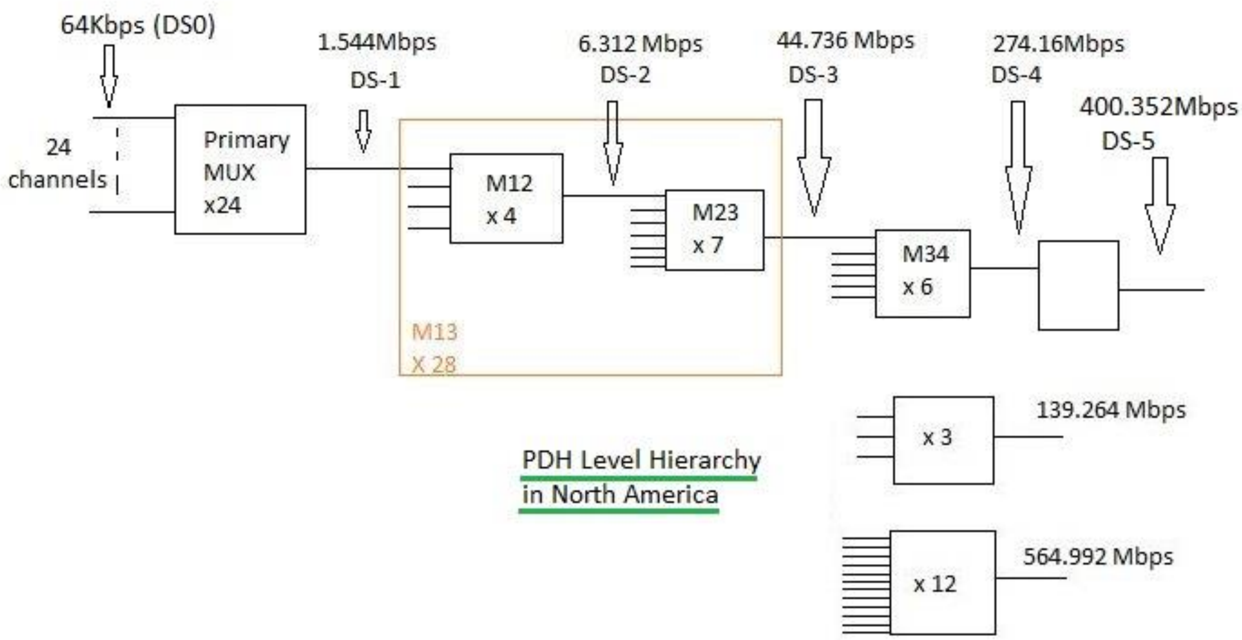


Fig: Companding process in PCM

- The compressor and expander perform inverse operations thus in the above figure the dotted line represents the linear characteristic of the compander indicating that the originally transmitted signal is recovered at the receiver.

Plesiochronous Digital Hierarchy (PDH):

- PDH stands for Plesiochronous Digital Hierarchy.
- It is a telecommunications network transmission technology designed for the transport of large data volumes across large-scale digital networks.
- In PDH, multiplexing of 2 Mbit/s signals into higher order multiplexed signals.
- The laying cable between switch sites is very expensive.
- It increases the traffic capacity of a cable by increasing the bit rate.
- In **PDH**, digital multiplexer's inputs (bit streams) are of same bit rate and are derived from different clocks from different oscillators.
- Each will differ within tolerance of few clock periods. Hence it is called Plesiochronous.
- Bit Interleaving is used in PDH to combine digital signals.



- In PDH there are two main standards i.e., 30 channel one used in Europe and 24 channel one used in North America/Japan.
- Basic rate is 64 Kbps in North America (designated as DS0) and in Europe (designated as E0).
- Rates derived from 2.048 Mbps basic rate including bit stuffing in 30 channel case are mentioned below.

  2.048 x 4 gives 8.448 Mbps (120 channels)

  8.448 x 4 gives 34.368 Mbps (480 channels)

  34.368 x 4 gives 139.264 Mbps (1920 channels)

  139.264 x 4 gives 564.992 Mbps (7680 channels)

Advantages of PDH:

- It was designed to support transportation of huge amounts of data over digital equipment's using various transmission mediums such as microwave radio or fiber optic systems.
- Network management has been enhanced to greater extent.

- It performs well in North America, Europe and Japan independently as per standard specifications.

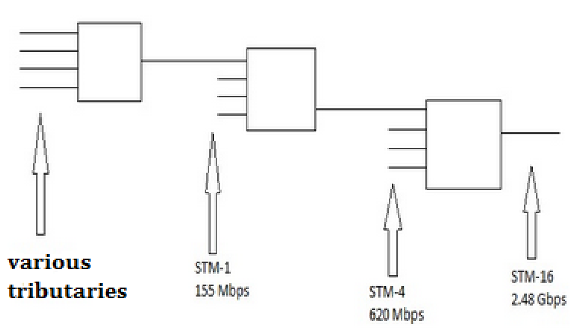
Disadvantages of PDH:

- In PDH, different frame is used for transmission and in data layer. Hence multiplexing and de-multiplexing is very complex.
- The maximum capacity for PDH is 566 Mbps, which is limited in bandwidth.
- Tolerance is allowed in bit rates.
- PDH allows only Point-to-Point configuration.
- PDH does not support Hub.

Synchronous Digital Hierarchy (SDH) :

- SDH stands for Synchronous Digital Hierarchy and it refers to as a multiplex technology used in telecommunication.
- Synchronous Optical Network is internationally used.
- It is said to be a variation of SONET and is taken equal to SDH.
- Its characteristics are founded on high-order multiplexing.

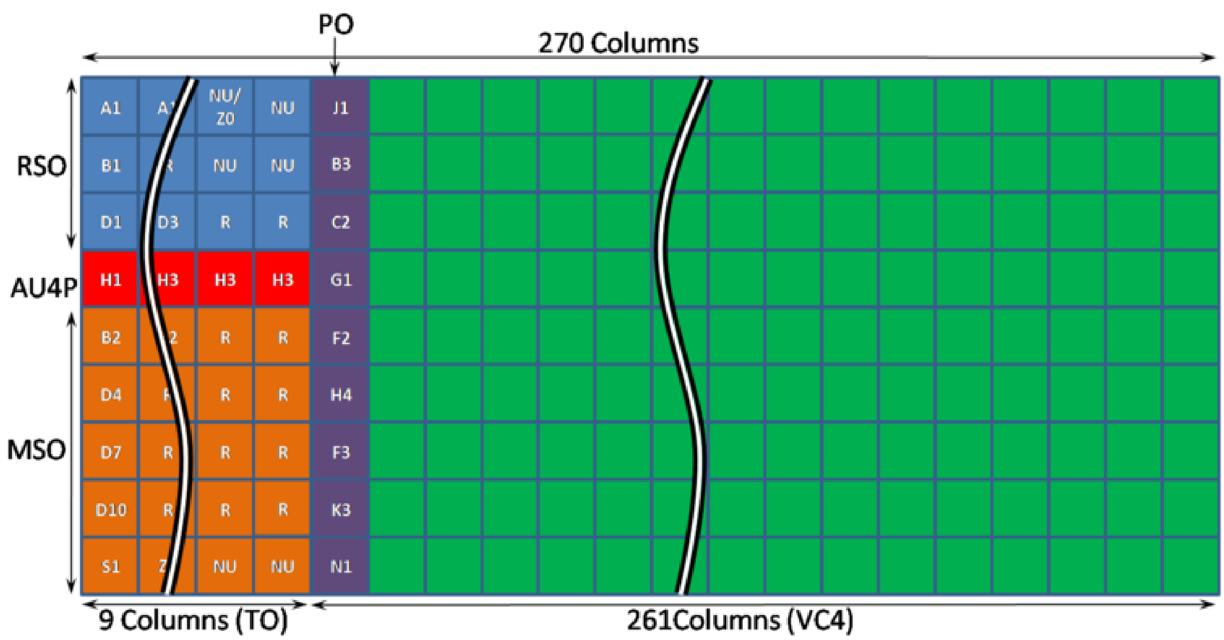
Working of SDH:



- In SDH, digital multiplexer's inputs are of same bit rate and are derived from common clock. Hence it is called synchronous.

- At low transmission rates data can also be transferred via an electrical interface.
- SDH system is designed to replace PDH system for transporting large quantities of telephone calls and data traffic over the same fiber without synchronization problems, providing a simple and flexible network infrastructure.

The STM-1 frame is on the basic transmission format for SDH (Synchronous Digital Hierarchy). An STM-1 frame has a byte-oriented structure with 9 rows and 270 columns of bytes, for a total of 2,430 bytes (9 rows * 270 columns = 2430 bytes). Each byte corresponds to a 64kbit/s channel.



The STM-1 base frame is structured with the following characteristics:

- **Length**: 270 column × 9 row = 2430 bytes
- **Byte**: 1-byte = 8 bit
- **Duration** (Frame repetition time): 125 μs i.e. 8000 frame/s
- **Rate** (Frame capacity): 2430 × 8 × 8000 = 155.52 Mbit/s
- **Payload** = 2349 bytes × 8 bits × 8000 frames/sec = 150.336 Mbit/s

Advantages of SDH:

- It consistently uses more simplified multiplexing and demultiplexing techniques.

- The optical fiber bandwidth can increase without limit.
- It has improved maintenance protocols with easy growth to higher bit rates.
- Rings provide switching protection to data traffic.
- It quickly interconnects with various networks.
- It has a comprehensive network management system.
- It has a flexible self-healing network.
- It can transport existing PDH, broadband and broadcast signals.
- It continues to remain popular within telecommunications networks and operators.
- It enables rapid recovery from failure.
- It offers network transmission services on local area networks for interactive multimedia, like video conferencing.
- It supports multiple operators or vendors.
- It supports multipoint networking.

Disadvantages of SDH:

- Complexity is increased by directly adding and dropping lower-rate signals that were archived using pointers.
- It demands complicated SDH equipment to manage different traffic types and options.
- It provides a lower bandwidth utilization ratio.
- It does not carry E2 because the container is unavailable.
- It is largely software-based and is vulnerable to cyber attacks.

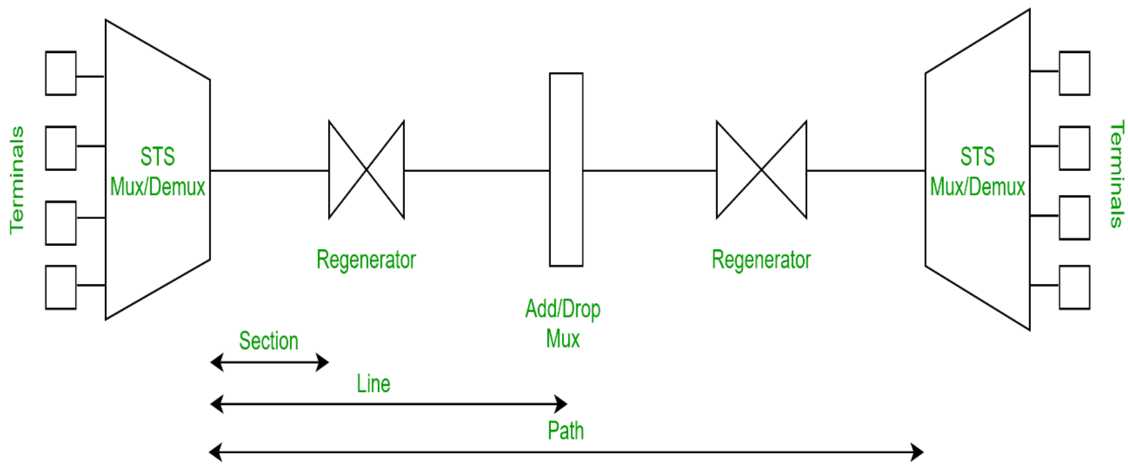
SONET:

- SONET stands for Synchronous Optical Network.
- It is used to transmit a large amount of data over relatively large distances using optical fiber.
- With SONET, multiple digital data streams are transferred at the same time over the optical fiber.

Why SONET is called a Synchronous Network?

- A single clock (Primary Reference Clock, PRC) handles the timing of transmission of signals & equipment's across the entire network.

- **SONET Network Elements:**



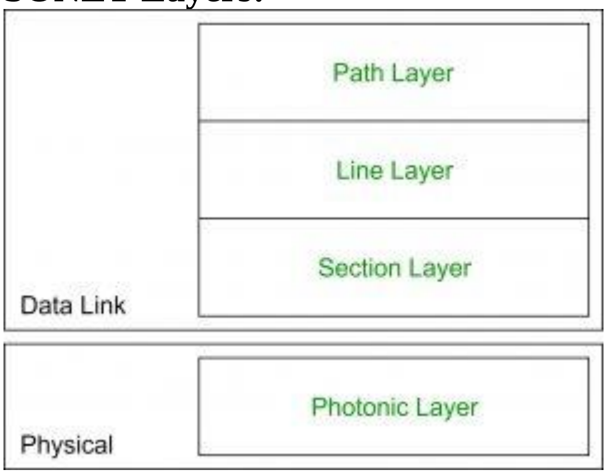
1. **STS Multiplexer:**
   - Performs multiplexing of signals
   - Converts electrical signal to optical signal
2. **STS Demultiplexer:**
   - Performs demultiplexing of signals
   - Converts optical signal to electrical signal
3. **Regenerator:**
   It is a repeater, that takes an optical signal and regenerates (increases the strength) it.

4. **Add/Drop Multiplexer:**
   It allows to add signals coming from different sources into a given path or remove a signal.

Why SONET is used?
SONET is used to convert an electrical signal into an optical signal so that it can travel longer distances.

SONET Connections:
- **Section:** Portion of network connecting two neighboring devices.
- **Line:** Portion of network connecting two neighboring multiplexers.
- **Path:** End-to-end portion of the network.

SONET Layers:



SONET includes four functional layers:

1. **Path Layer:**
   - It is responsible for the movement of signals from its optical source to its optical destination.
   - STS Mux/Demux provides path layer functions.
2. **Line Layer:**
   - It is responsible for the movement of signal across a physical line.
   - STS Mux/Demux and Add/Drop Mux provides Line layer functions.
3. **Section Layer:**
   - It is responsible for the movement of signal across a physical section.
   - Each device of network provides section layer functions.
4. **Photonic Layer:**
   - It corresponds to the physical layer of the OSI model.
   - It includes physical specifications for the optical fiber channel (presence of light = 1 and absence of light = 0).

Advantages of SONET:

- Transmits data to large distances
- Low electromagnetic interference
- High data rates
- Large Bandwidth
- It is compatible with legacy and future network
- Very high efficiency
- Allows transportation of all forms of traffic
- Standard optical interference
- De-multiplexing is easy

Disadvantages of SONET:

- No interoperable standard
- Tributary services require SONET mux services
- Low cost effective for low channel numbers.
- Bandwidth efficiency is a problem at higher capacity
- More overhead is required

SDH/SONET Digital rate hierarchy table

SONET Rate Name	SDH name	Line Rate (Mbps)	Synchronous payload envelope rate(Mbps)	Transport Overhead rate(Mbps)
STS-1	None	51.84	50.112	1.728
STS-3	STM-1	155.52	150.336	5.184
STS-12	STM-4	622.08	601.344	20.736

STS-48	STM-16	2488.32	2405.376	84.672
STS-192	STM-64	9953.28	9621.504	331.776
STS-768	STM-256	39813.12	38486.016	1327.104

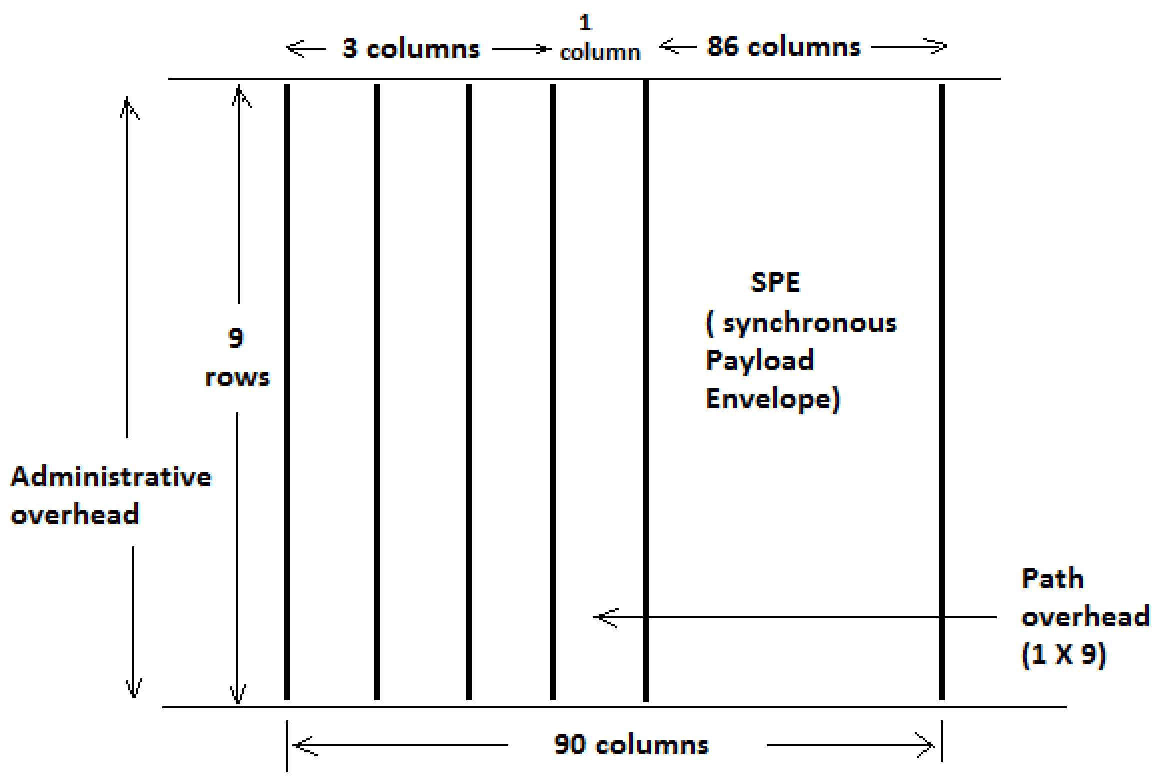
PDH VS SDH:

PDH	SDH
Plesiochronous Digital Hierarchy	Synchronous Digital Hierarchy
In PDH, reference clock is not synchronized throughout the network.	In SDH, Reference clock is synchronized throughout the network.
There is no synchronization between payload and frame.	There is synchronization between payload and frame.
PDH system has different frame structures at different hierarchy levels.	SDH system has consistent frame structures throughout the hierarchy.
Physical cross-connections are provided on the same level in PDH.	Digital cross-connections are provided at different signal levels in SDH.

In PDH, rates are derived from basic rate of 1.544 Mbps. The maximum capacity is about 566 Mbps.	In SDH, rates are derived from basic rate of 155.52 Mbps. The maximum up to 40 Gbps rates can be derived from basic rate mentioned.
There is no universal standard for PDH.	Universal standard exists for SDH.
PDH is incompatible with other signals such as ATM, FDDI, DQDB etc.	SDH is compatible with other signals such as ATM, FDDI, DQDB etc.
Multiplying method used in PDH is complex.	Multiplying method used in SDH is simple.
Implementation cost of PDH is lower.	Implementation cost of SDH is higher.

Draw the frame format of STS-1?

- **STS-1 frame format:** The STS-1 is synchronous transport signal is SONET.
- The frame format is two dimensional as shown below.



- STS-1 frame format is mainly divided into two main areas (a) Administrative overhead (b) Synchronous payload envelope

(a) Administrative overhead

- It is also known as transport overhead which is composed of section and line overhead.
- Section overhead is of 3X3 bytes and line overhead of 6X3 bytes.

(b) Synchronous payload envelope

- SPE also has two parts

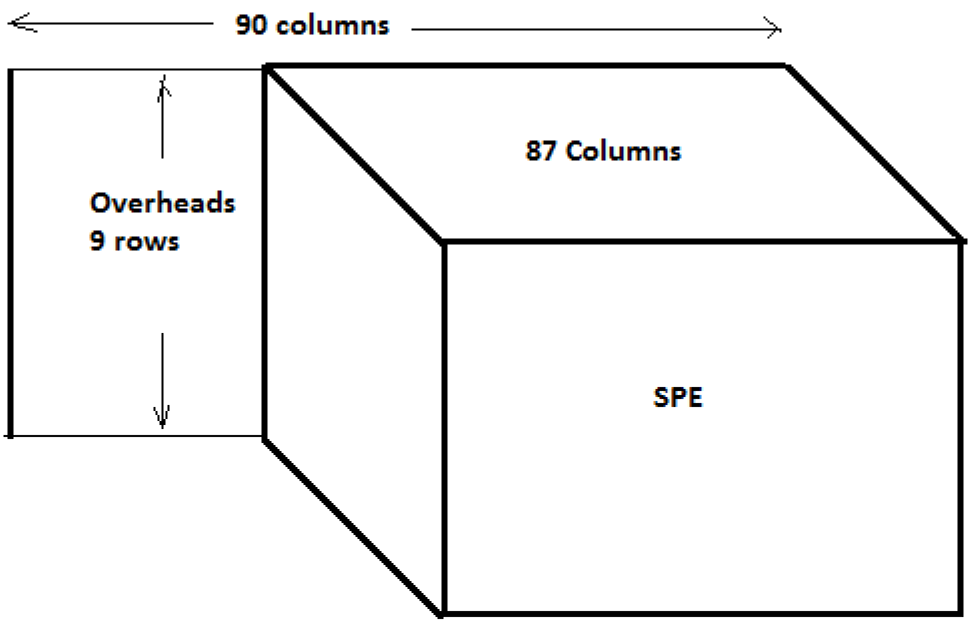
(i) Path overhead

- The STS-1 path overhead is **part of the synchronous payload envelope**. or combinations of above.

- STS-1 is a specific sequence of 810 bytes (6480 bits), which includes various overhead bytes and an envelope capacity for transporting payloads.
- It can be depicted as a 90 column by 9 row structure.

(ii) Payload

- The Payload is the revenue producing traffic being transported and routed over the SONET network.
- Once the payload is multiplexed into the SPE, It can be transported and switched through SONET without having to examined and possibly de-multiplexed at intermediate nodes.
- STS-1 is specific sequence of 810 bytes (6480 bits) which includes various overhead bytes and an row by row from top to bottom and from left to right.



- The first 3-coloumns contain 9 bytes, out of which (3X3) 9 bytes are of section overhead and remaining (6X3) 18 bytes are for line overhead.
- The remaining 87 columns constitute the STS-1 envelope capacity or SPE.
- SPEs can have any alignment within the frame and this alignment is indicated bye the H1 and H2 pointer bytes in the line overhead.
- At a rate 8000 frames per second that works out to be at a rate of 51.84 Mbps at STS-1 data rate as shown below,
- **9 X (90 bytes/frame) X (8 bits/byte) X 8000 frame= 51.84 Mbps date rate of STS-1**

What is Frame and Multiple Frame?

- Framing is a point-to-point connection between two devices that consists of a wire in which data is transmitted as a stream of bits.

How T1 and E1 lines are accommodated in STM-1 Frame structure?

- T1 and E1 are two different types of digital communication systems that are commonly used for transmitting voice and data over long distances.
- They both use pulse code modulation (PCM) to encode and transmit the data, and they both operate at different rates.
- T1 is a North American standard that uses a transmission rate of 1.544 megabits per second (Mbps) and is capable of carrying up to 24 voice or data channels.
- E1 is a European standard that uses a transmission rate of 2.048 Mbps and is capable of carrying up to 30 voice or data channels.
- Both T1 and E1 lines can be accommodated within the STM-1 frame structure, which is a standard used in synchronous optical networking (SONET) and synchronous digital hierarchy (SDH) systems. STM-1 stands for Synchronous Transport Module level 1 and refers to a transmission rate of 155.52 Mbps.
- The STM-1 frame structure is used to multiplex multiple T1 or E1 lines into a single, higher-capacity transmission stream.
- It consists of a series of fixed-length frames, each containing a number of time slots that can be used to carry T1 or E1 channels.
- The STM-1 frame structure also includes a number of overhead bytes that are used for various control and management functions.
- Overall, the STM-1 frame structure provides a flexible and efficient way to transmit T1 and E1 lines over long distances using SONET or SDH systems.

What is virtual container (virtual tributary) in SDH?

Virtual Container

- SDH supports a concept called virtual containers (VC).
- Through the use of pointers and offset values, VCs can be carried in the SDH payload as independent data packages.
- VCs are used to transport lower-speed tributary signals.
- Figure 2 illustrates the location of a VC-4 within the STM1 frame.
- Note that it can start (indicated by the J1 path overhead byte) at any point within the STM-1 frame.
- The start location of the J1 byte is indicated by the pointer byte values.

Virtual containers can also be concatenated to provide more capacity in a flexible fashion.

Line Coding

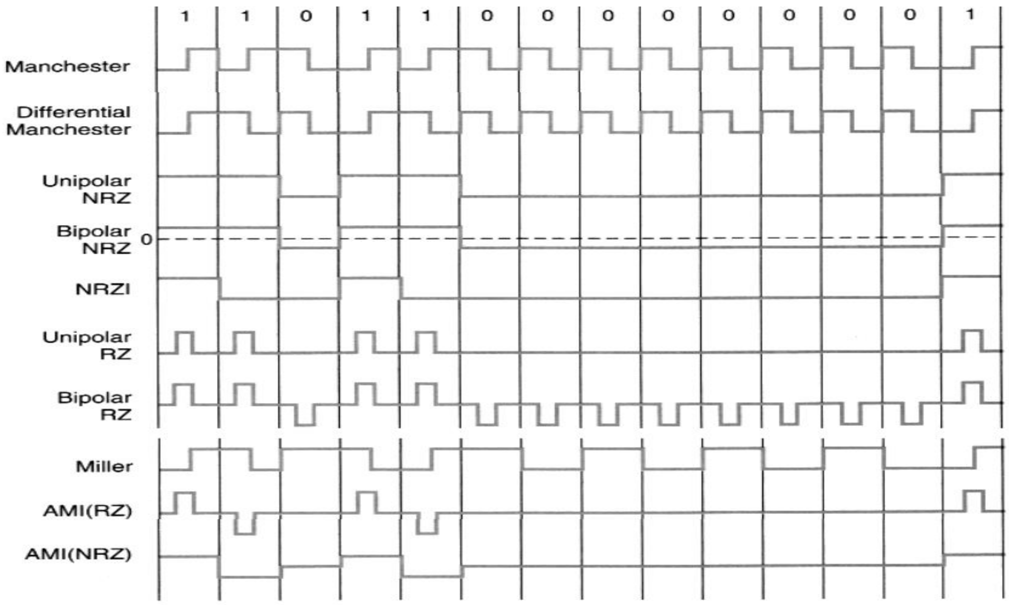
- Line coding is the process of converting binary data, a sequence of bits to a digital signal.
- Binary data can be transmitted using a number of different types of pulses.
- The choice of a particular pair of pulses to represent the symbols 1 and 0 is called Line Coding.

Properties of Line Coding

Following are the properties of line coding −

- As the coding is done to make more bits transmit on a single signal, the bandwidth used is much reduced.
- For a given bandwidth, the power is efficiently used.
- The probability of error is much reduced.
- Error detection is done and the bipolar too has a correction capability.
- Power density is much favorable.
- The timing content is adequate.

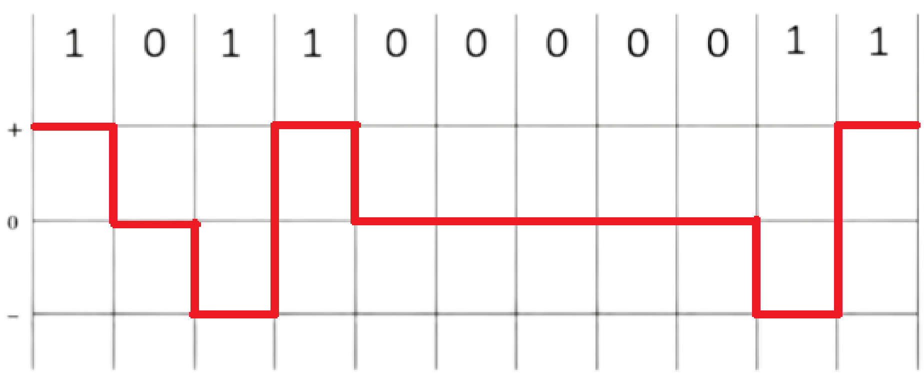
- Long strings of **1s** and **0s** is avoided to maintain transparency.



Bipolar-AMI

- It stands for ***Bipolar Alternate Mark Inversion (AMI).***
- In this type of encoding, Bit – 0 is represented by 0 voltage line, and Bit – 1 is represented by alternate positive and negative voltages.

Waveform for the Binary Sequence 10110000011 using Bipolar-AMI:



Bipolar-AMI encoding for 10110000011

B8ZS

- B8ZS stands for ***Bipolar 8-Zero Substitution.***
- It is one type of **Scrambling** method.
- It is used to solve the problem of the long sequence of zeros in the Bipolar-AMI encoding method.
- It is used when the data stream consists of ***8 consecutive zeros.***
- This B8ZS work is based on the below scenarios:

Scenario 1 - If the data stream does not contain 8 consecutive zeros:

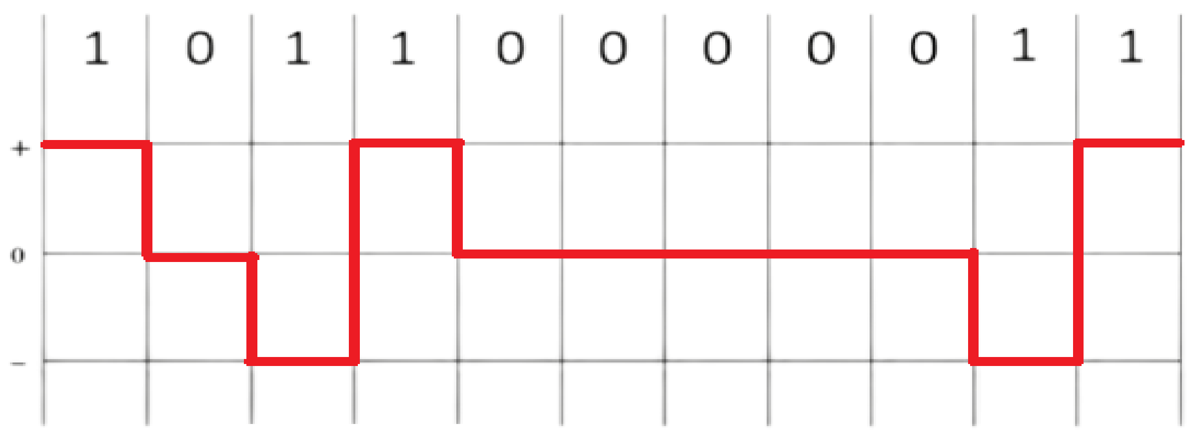
- Then B8ZS generates the same waveform as Bipolar-AMI.

Scenario 2 - If the data stream contains 8 consecutive zeros:

- Then it checks the ***previous voltage level.***
    - ***Case 1 -*** If previous voltage level is **POSITIVE** then it substitute or generates the waveform in the sequence of **0 0 0 + – 0 – + voltage format.**
    - ***Case 2 -*** If previous voltage level is **NEGATIVE** then it substitute or generates the waveform in the sequence of **0 0 0 – + 0 + – voltage format.**
- This encoding method creates ***two violations*** in AMI type of waveforms.

The given Binary Sequence 10110000011 does not contain 8 consecutive zeros. Therefore B8ZS encoding technique generates the same waveform as Bipolar-AMI.

Waveform for the Binary Sequence 10110000011 using B8ZS:



B8ZS encoding for 10110000011

Because of the absence of 8 consecutive 0's, there is no change in the waveform and generates the same waveform as Bipolar-AMI.

HDB3

- HDB3 stands for ***High-Density Bipolar 3.***
- It is one type of **Scrambling** method.
- It is also used to solve the problem of the long sequence of zeros in the Bipolar-AMI encoding method.
- It is used when the data stream consists of ***4 consecutive zeros.***
- This HDB3 work is based on the below scenarios:

Scenario 1 - If the data stream does not contain 4 consecutive zeros:

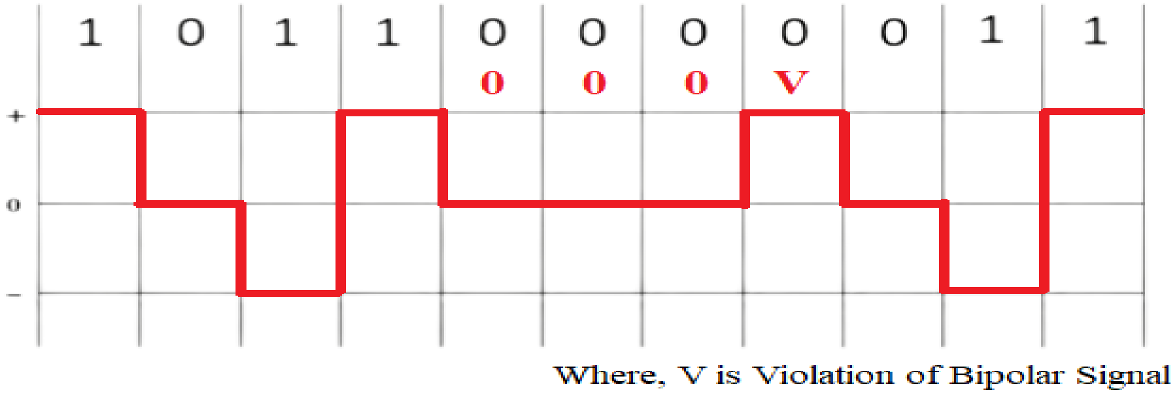
- Then HDB3 generates the same waveform as Bipolar-AMI.

Scenario 2 - If the data stream consists of 4 consecutive zeros:

- Then it computes the ***previous non-zero voltages*** and then ***checks the last voltage level.***
    - ***Case 1 -*** If the count of non-zero voltage level is **EVEN** and the last voltage level is **POSITIVE** then it substitutes or generates the waveform in the sequence of **– 0 0 – voltage format.**
    - ***Case 2 -*** If the count of non-zero voltage level is **EVEN** and the last voltage level is **NEGATIVE** then it substitutes or generates the waveform in the sequence of **+ 0 0 + voltage format.**
    - ***Case 3 -*** If the count of non-zero voltage level is **ODD** and the last voltage level is **POSITIVE** then it substitutes or generates the waveform in the sequence of **0 0 0 + voltage format.**
    - ***Case 4 -*** If the count of non-zero voltage level is **ODD** and the last voltage level is **NEGATIVE** then it substitutes or generates the waveform in the sequence of **0 0 0 – voltage format.**
- This encoding method creates ***one violation*** in AMI type of waveforms.

- The given Binary Sequence 10110000011 contains 4 consecutive zeros.
- But, up to 1011 binary sequence HDB3 generate the same waveform as Bipolar-AMI because no consecutive 4 zeros were found.
- After finding 4 consecutive 0's checks for the **non-zero voltages that are 3 that means ODD** number and **last voltage level is POSITIVE**.
- Therefore, it is categorized under **Case 3.**
- Hence generate waveform according to ***Case 3*** in the sequence ***0 0 0 + voltage format.***
- The remaining bits after the 4 consecutive 0's again represented according to the Bipolar-AMI.

Waveform for the Binary Sequence 10110000011 using HDB3I:



HDB3 encoding of 10110000011

CHAPTER 4- Switching system

Switching

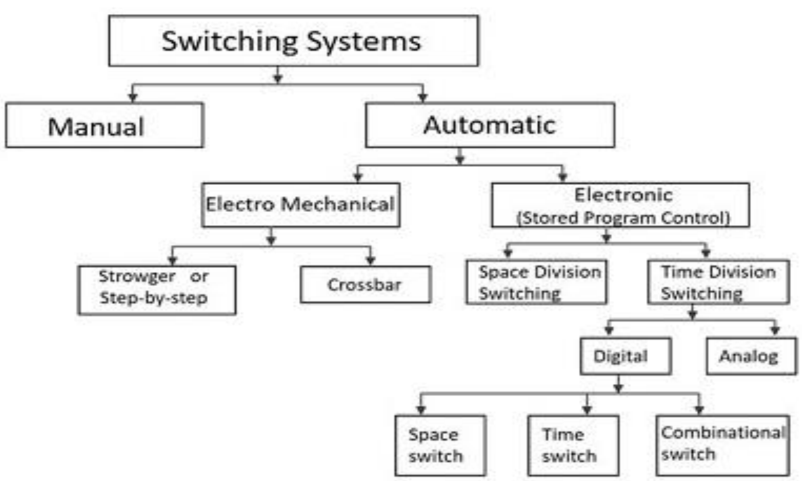
- Switching is the method that is used to establish connections between nodes within a network.
- Switching in computer network is what help in deciding the best route for data transmission in a larger network if there are multiple path.
- Once a connection has been made, information can be sent. Telephone switching usually refers to the switching of voice channels.
- There are a number of different types of switches including local switches, tandem switches and transit switches and any number of them can play a part in creating a connection.

Local switch: This provides switching for a specific area. Subscriber loops connect to the local switch in that area.

Tandem switch: This is used to interconnect switches at various sites within the network.

Transit switch: This is very similar to a tandem switch, except it is used for long-distance connections.

Classification of switching system:

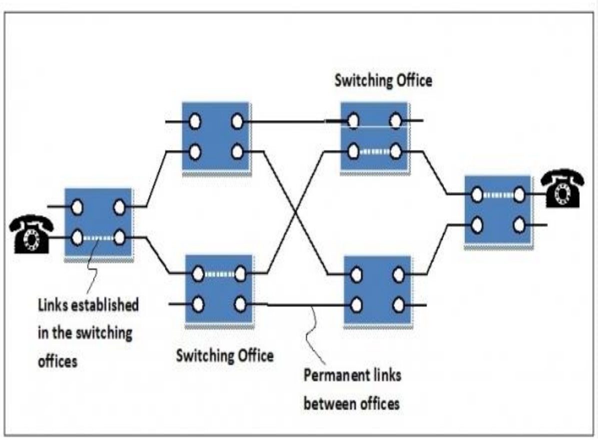


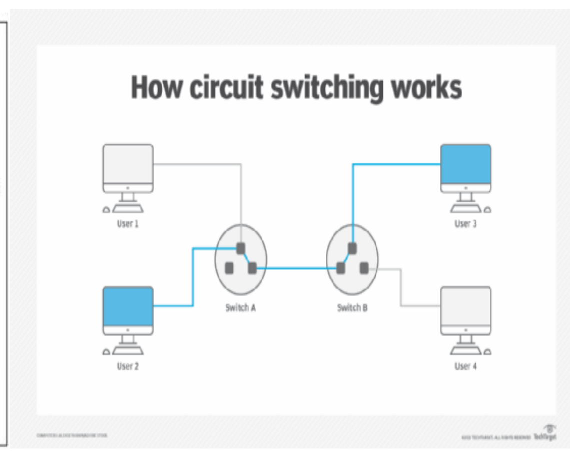
What Is Circuit Switching?

- In circuit switching, a dedicated path or route is established between the sender and the receiver.
- In other words, before data can be transferred connection must be established.
- This type of technique is used in telephone network where by before you can communicate with your friend via the telephone, a connection is established between you and your friend as soon as you dial up to call your friend.

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 transmit 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 relinquished. The disconnection is initiated by any one of the users. Disconnection involves removal of all intermediate links from the sender to the receiver.





The following diagram represents circuit established between two telephones connected by circuit switched connection. The blue boxes represent the switching offices and their connection with other switching offices. The black lines connecting the switching offices represents the permanent link between the offices. When a connection is requested, links are established within the switching offices as denoted by white dotted lines, in a manner so that a dedicated circuit is established between the communicating parties. The links remains as long as communication continues.

Advantages of circuit switching

1. Once the communication channel is established, it is dedicated and no other device can use that channel.
2. It is suitable for real-time services or communications.

Disadvantages of circuit switching

1. There is time delay due to the establishment of connection.
2. There is inefficient use of channel.
3. It is more expensive than other switching techniques.

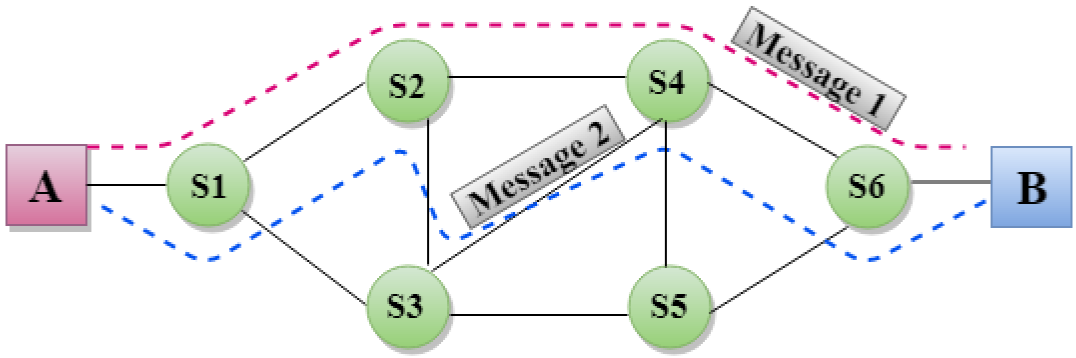
What Is Message Switching?

- Unlike in circuit switching where a dedicated path is established, a message switching does not need a dedicated path.
- It uses a store and forward mechanism where by a message is broken into pieces or chunks and are transferred to an intermediary node which helps to store the messages and when the messages are completely stored, they are forwarded to their destination.

Message Switching

- Message Switching is a switching technique in which a message is transferred as a complete unit and routed through intermediate nodes at which it is stored and forwarded.
- In Message Switching technique, there is no establishment of a dedicated path between the sender and receiver.
- The destination address is appended to the message. Message Switching provides a dynamic routing as the message is routed through the intermediate nodes based on the information available in the message.
- Message switches are programmed in such a way so that they can provide the most efficient routes.
- Each and every node stores the entire message and then forward it to the next node. This type of network is known as **store and forward network.**

- Message switching treats each message as an independent entity.



Advantage of message switching

1. It is more efficient than circuit switching as there is not establishment of dedicated path and more device can share the channel.
2. Congestion of traffic can be reduced as the message is store temporarily along the transmission path.
3. Due to store and forward mechanism, message priorities can be established.

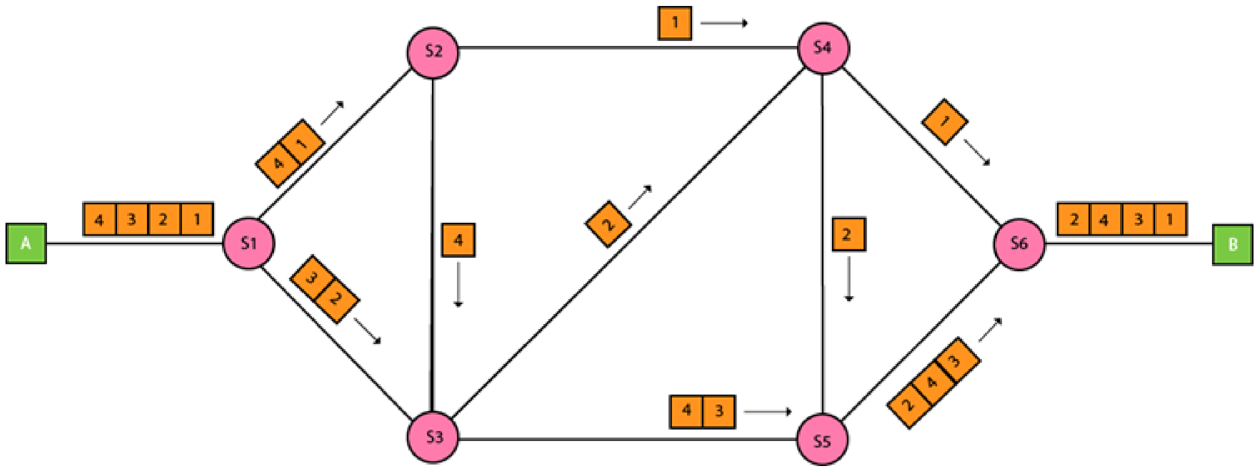
Disadvantages of message switching

1. It is not suitable for real-time communications or applications.
2. The store and forward devices can be quite expensive because they must have large storage capacity to hold large amount of data.

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.



Advantage of packet switching

1. In packet switching, there is an efficient use of channels.
2. Packet switching is cost effective as switching devices must not have a huge storage capacity.
3. There is an improvement in delay characteristics as messages are broken into chunks.
4. If there is problem with a path, data can be rerouted.

Disadvantages of packet switching

1. It is not suitable for real-time application also.
2. Initial implementation cost is high.
3. If a packet is lost, sender needs to resend the missing packet.
4. The protocol for packet switching are quite complex.

Packet switching is further divided into two;

- Datagram packet switching and
- Virtual circuit packet switching

What Is A Datagram Packet Switching?

- Datagram packet switching is also known as connectionless switching and does not have a fixed path.
- The datagram (or message) contains destination information which allow the intermediary node to take routing decisions and to forward the message.

What Is a Virtual Packet Switching?

- Virtual circuit packet switching is also known as connection-oriented switching. A preplanned route (or logical path) is established before messages can be sent.
- The connection between the sender and the receiver is established with the use of call request and call accept packet.
- Due to call request and call accept which do result in an acknowledgement, virtual circuit packet switching is more reliable.
- Message arrives in the order that it is been sent.

Differences between circuit vs message vs packet switching.

Basics	Circuit Switching	Message Switching	Packet Switching
Connection Creation	Connection is created between the source and destination by establishing a dedicated path between source and destination.	Links are created independently one by one between the nodes on the way.	Links are created independently one by one between the nodes on the way.
Queuing	No queue is formed.	Queue is formed.	Queue is formed.
Message and Packets	There is one big entire data stream called a message.	There is one big entire data stream called a message.	The big message is divided into a small number of packets.

Basics	Circuit Switching	Message Switching	Packet Switching
Routing	One single dedicated path exists between the source and destination.	Messages follow the independent route to reach a destination.	Packets follow the independent path to hold the destination.
Addressing and sequencing	Messages need not be addressed as there is one dedicated path.	Messages are addressed as independent routes are established.	Packets are addressed, and sequencing is done as all the packets follow the independent route.
Propagation Delay	No	Yes	Yes
Transmission Capacity	Low	Maximum	Maximum
Sequence Order	Message arrives in Sequence.	Message arrives in Sequence.	Packets do not appear in sequence at the destination.
Use Bandwidth	Wastage	Bandwidth is used to its maximum extent.	Bandwidth is used to its maximum extent.

Circuit Switching	Datagram Packet Switching	Virtual Circuit Packet Switching
Dedicated transmission path	No dedicated path	No dedicated path
Continuous transmission of data	Transmission of packets	Transmission of packets
Fast enough for interactive	Fast enough for interactive	Fast enough for interactive
Messages are not stored	Packets may be stored until delivered	Packets stored until delivered
The path is established for entire conversation	Route established for each packet	Route established for entire conversation
Call setup delay; negligible transmission delay	Packet transmission delay	Call setup delay; packet transmission delay
Busy signal if called party busy	Sender may be notified if packet not delivered	Sender notified of connection denial
Overload may block call setup; no delay for established calls	Overload increases packet delay	Overload may block call setup; increases packet delay
Electromechanical or computerized switching nodes	Small switching nodes	Small switching nodes
User responsible for message loss protection	Network may be responsible for individual packets	Network may be responsible for packet sequences
Usually no speed or code conversion	Speed and code conversion	Speed and code conversion
Fixed bandwidth	Dynamic use of bandwidth	Dynamic use of bandwidth
No overhead bits after call setup	Overhead bits in each packet	Overhead bits in each packet

Manual Switching system:

- In a manual switching system, **the operator has full control of a connection. He/She enables the signaling systems, performs switching and releases a connection after a conversation**.

call cannot be stored, as in message switching, ... Circuit switching is thus an example of a *lost-call* system. The theory of traffic in lost-call systems is also explained in Chapter 4.

3.4 Manual systems

The earliest form of switchboard[5] had incoming circuits connected to vertical metal bars and outgoing links connected to horizontal metal bars, as shown in Figure 3.2. The operator makes a connection by inserting a brass peg where the appropriate vertical and horizontal bars crossed, i.e. at a *crosspoint*. This was the forerunner of the crosspoint matrices used in modern switching systems.

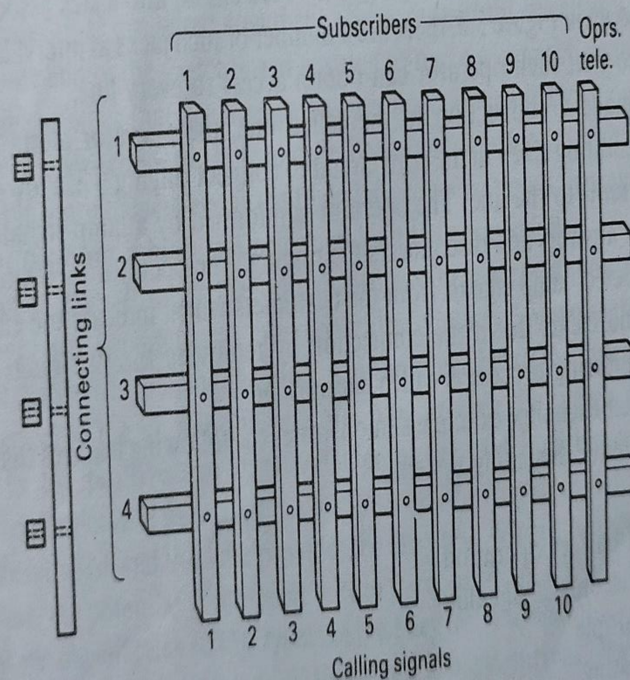


Figure 3.2 Early crossbar switchboard.

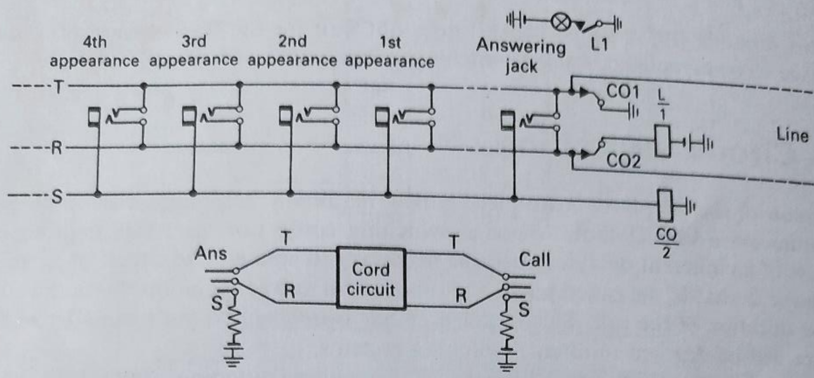


Figure 3.3 Multiple switchboard.

If all customers' lines are connected to vertical bars, the operator makes a connection from a calling line j to a called line k by choosing a horizontal link z and inserting pegs at the crosspoints with coordinates (j,z) and (k,z). Thus, the connection is made through two stages of crosspoint switching and an intermediate link. Such *link systems*, with two or more switching stages, are used in modern telephone exchanges.

The need for larger exchanges, with many operators to handle the traffic, led to the cord type of switchboard.[6–8] As shown in Figure 3.3, each operator answers calls from one group of customers. When one of these calls, the operator responds to a lamp signal by inserting a plug into the corresponding answering jack and operating a key to connect the headset to the cord circuit attached to that plug. The operator obtains the number of the called line by speaking to the caller and then completes the connection, if the line is free, by inserting the other plug of the cord circuit into a jack associated with that line. As shown in Figure 3.3, there are a number of such jacks at intervals along the switchboard, so that each operator can obtain access to every line.

Having made a connection to the called line, the operator alerts the called customer by operating a key in the cord circuit to connect ringing (i.e. a low-frequency alternating current) to the line. The operator is informed by a lamp signal when the called customer answers and disconnects the ringing. The operator then monitors the connection to detect lamp signals from the customers which indicate the end of their conversation and clears down the connection by removing the two plugs from the jacks. This monitoring process is called *supervision*.

The manual-exchange example demonstrates the following features that are also present in automatic switching systems:

- Central-battery operation
- Loop/disconnect signalling
- The multiple
- Busy testing

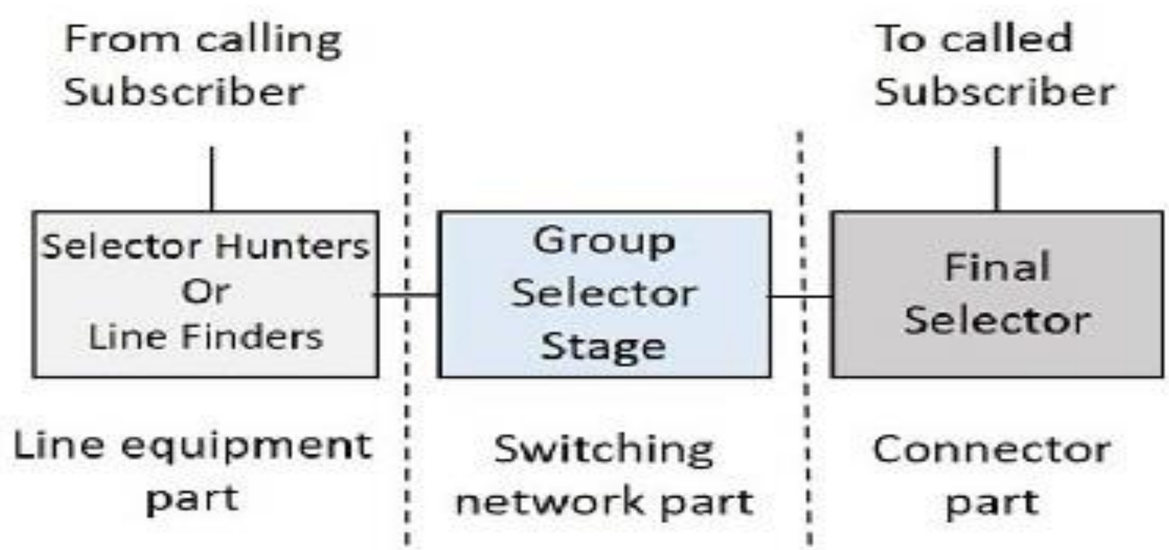
The **Automatic** switching systems are classified as the following –

- **Electromechanical Switching Systems –** Here, mechanical switches are electrically operated.
- **Electronic Switching Systems –** Here, the usage of electronic components such as diodes, transistors and ICs are used for the switching purposes.

ELECTROMECHANICAL SWITCHING SYSTEM:

i)Step-by-Step Switching

- The Step-by-step switching system is a very popular and widely-used switching system, which may be constructed using Uni-selectors or two-motion selectors or the combination of both.
- The wiper present in this switching, steps forward by one contact and then moves forward according to the number of dialed pulses or according to the signaling conditions and hence the name, **step-by-step** switching is given.
- A step-by-step switching is also called the **Direct control** system as the relevant signaling tones are sent out to the subscriber by the switching elements or selectors at the appropriate stages of switching.
- This system has three main stages of configuration. The following figure shows the different stages.



Let us now see how these blocks function.

Selector Hunters

As soon as the calling subscriber gets ready to dial the number, by lifting the handset from the telephone, a dial tone is heard. We have already learnt that a number is not accepted unless the dial tone is heard. But to get that dial tone, the line has to be established when the handset is lifted up. The **Selector Hunter** circuit, establishes the line to make a call as immediately as the calling subscriber lifts up the handset to make a call.

The Selector Hunters hunt for selecting a switching matrix part. Usually, 24-outlet Uni-selectors are used as selector hunters. and so this can be called as **Subscriber Uni-selector** scheme as there is a dedicated Uni-selector for each subscriber in the system. These can also be build using two-motion selectors.

The selector hunter mechanism can also be replaced with the line finder mechanism, where there is small difference between the two in construction. Here, we shall discuss the selector hunter mechanism. The figure below gives an idea about its construction.

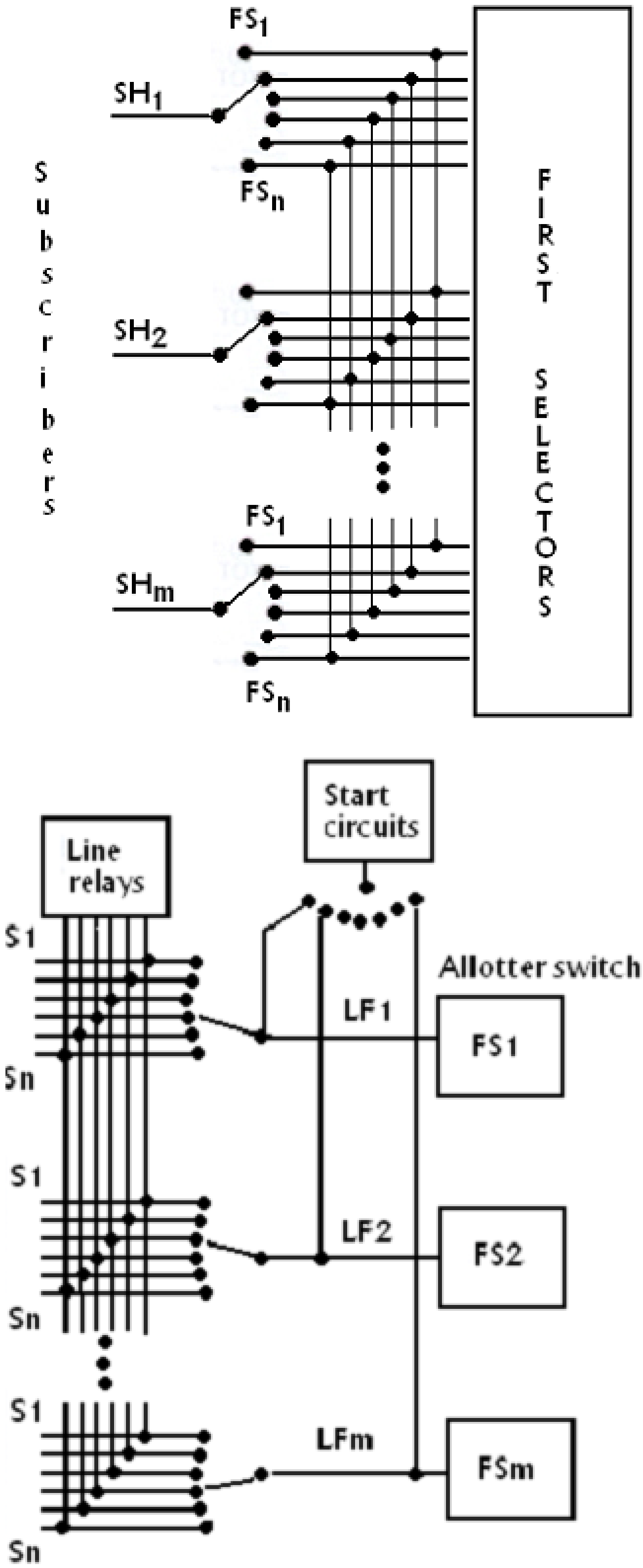


Fig: subscriber accessing the stowger switching system

When a calling subscriber lifts the handset to make a call, the selector hunter activates the interrupter mechanism, which steps up the wiper until a free first group selector is found at the outlet. One of the bank contacts of the selector hunter, at this point, senses whether the first group selector is free or busy. Once a free first selector is sensed, the interrupter is disabled and the connection is established, where the first selector sends out a dialer tone to the calling subscriber.

The line finder approach is used where the traffic is low and the exchange is small, whereas the selector hunter mechanism described above is used for large exchanges with heavy traffic and this approach is cost-effective.

Group Selector Stage

The Group Selector stage has the main switching network. The calling subscriber dials the number after hearing the dial tone. The first number when dialed activates the first selector. To be more precise, the group selector consists of certain selector stages. We used to have 5 numbers as an identification number, for the land connection. Hence, there were three selector stages present.

To dial the first number, the number plate is rotated by placing the finger in the finger gap given according to the subscriber number. After taking out the finger, the number plate is rotated back to its previous position, which sends the dialing pulses to the first selector. The first selector then moves accordingly, to place a contact.

When the subscriber starts dialing, the dial tone produced till then, cuts off and the pulse train is received according to the number dialed. The wiper assembly of the first selector then moves vertically upward, according to the number dialed. The wipers then move in the horizontal plane across the contacts until they come across a contact to which a free second group selector is connected. This horizontal stepping is completed within the inter-digit gap of about 240ms. From there, the first group selector connects the electrical path to the available second group selector.

Likewise, every group selector connects path according to the number dialed and then extends the connection to the next selector until the final selector. The action of the final selector is a bit different. As discussed above, three selectors are present and the fourth and the fifth numbers are connected to the matrix by the final selector.

Final Selector

The last two digits are processed by the final selector. This selector moves vertically according to the fourth digit dialed and then it moves horizontally according to the last digit, as there are no further digits to connect it to some other connector. The last digit dialed, establishes electrical connection to the called subscriber.

Since the final selector responds to both the digits in vertical and horizontal directions unlike the group selectors, this final selector is also called a **Numerical Selector.** If the called subscriber is free, as sensed from a signal at the corresponding bank contact, the final selector sends out a ringing current to the called subscriber and a ringing tone to the calling subscriber.

When the called subscriber lifts his handset, the ringing current and the ringing tone provided till then, are cut off and the call metering circuits are enabled by the control circuits associated with the final selectors. Otherwise, if the called subscriber is found to be busy on some other line, then the final selector sends out a busy tone to the calling subscriber. At any stage of switching, if there is no free selector available at the next stage, a busy tone is returned to the calling subscriber.

The magnets and mechanical linkages used in rotating the shafts vertically and horizontally while connecting a call, will release the magnet (generally called the release magnet) and armature release the shaft when the call is completed.

Advantages of step by step switching although it has the several disadvantages it is used for following purpose:

- **Faster establishment and release of calls is done**.
- Number of calls made in a given period can be increased.
- Calls can be made irrespective of the load on the system or the time of the day.
- Inexpensive for small system and highly reliable due to the distributed nature of equipment.

DISADVANTAGES:

1. As this switching involves heavy mechanical displacements, regular maintenance by the skilled technicians is necessary.
2. It is not feasible to select an alternate route for interoffice calls, if all the trunks are busy as the switching is by step through various selectors.
3. Step by step switching is limited to dial pulses. For touchtone telephones, special devices have to be introduced between line finder and first selector to convert the tones into dial pulses.
4. If calling rate is high, heavy operation is performed by the system and the life time of the system is less.
5. The last two digits of the called line numbers are specifically determined by their location on the connector. Congestion could arise when the switching system is heavily loaded.
6. The strowger system can accept only 7 to 9 pulses in 1 second. Hence if we dial fast, the system cannot give correct performance.

2.**CROSSBAR SWITCHING SYSTEM**

The Features of Crossbar Switches

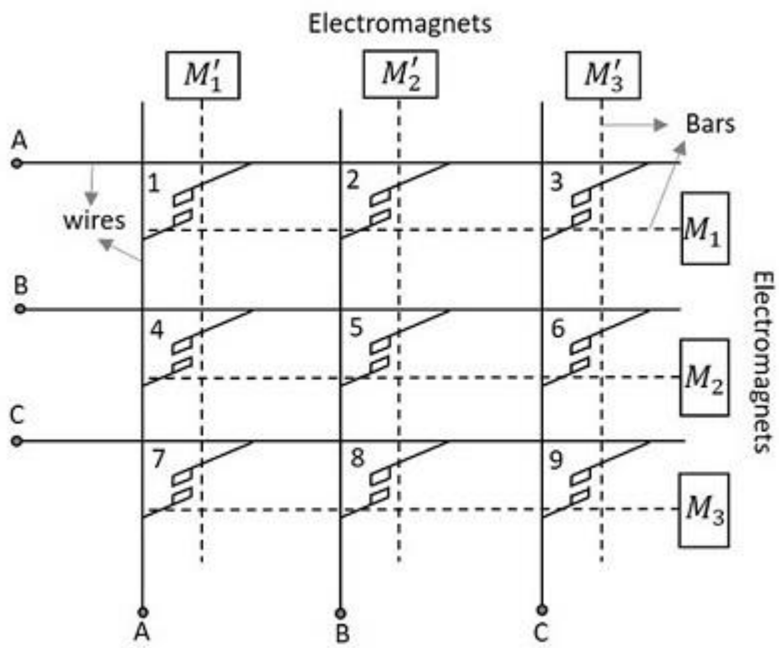
In this section, we will discuss the different features of the Crossbar Switches. The features are described in brief below –

- While processing a call, the common control system helps in the sharing of resources.
- The specific route functions of call processing are hardwired because of the Wire logic computers.
- The flexible system design helps in the appropriate ratio selection is allowed for a specific switch.
- Fewer moving parts ease the maintenance of Crossbar switching systems.

The Crossbar switching system uses the common control networks which enable the switching network to perform event monitoring, call processing, charging, operation and maintenance as discussed previously. The common control also provides uniform numbering of subscribers in a multi-exchange area like big cities and routing of calls from one exchange to another using the same intermediate exchanges. This method helps to avoid the disadvantages associated with the step-by-step switching method through its unique process of receiving and storing the complete number to establish a call connection.

Crossbar Switching Matrix

The Crossbar arrangement is a matrix which is formed by the M X N sets of contacts arranged as vertical and horizontal bars with contact points where they meet. They need nearly M + N number of activators to select one of the contacts. The Crossbar matrix arrangement is shown in the following figure.



3 X 3 Crossbar Switching

The Crossbar matrix contains an array of horizontal and vertical wires shown by solid lines in the following figure, which are both connected to initially separated contact points of switches. The horizontal and vertical bars shown in dotted lines in the above figure are mechanically connected to these contact points and attached to the electromagnets.

Advantage crossbar:

1. It is a Non Blocking Network that allows multiple i/o connections to be achieved simultaneously.
2. It provides full connectivity.

   *Example: any permutation can be implemented using a crossbar.
3. It is highly useful in a multiprocessor system, as all processors can send memory requests independently and asynchronously.
4. It gives maximum utilization of bandwidth as compared to other networks like bus system, multistage networks.

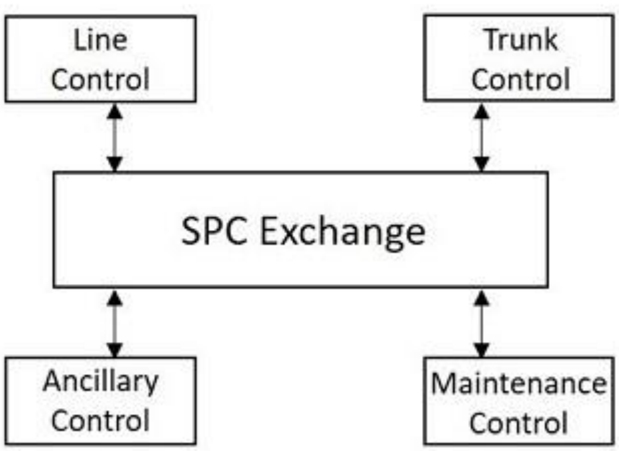
Disadvantage crossbar:

1. The crossbar switch is singled-layered switch.
2. At each point, there is a switch when closed, connects one of the inputs to one of the outputs.

ELECTRIC SWITCHING SYSTEM (Stored Program Control):

- The **Stored Program Control**, in short **SPC** is the concept of electronics that ringed in a change in telecommunication.
- It permits the features like abbreviated dialing, call forwarding, call waiting, etc.
- The Stored Program Control concept is where a program or a set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor.

As the exchange control functions are carried out through programs stored in the memory of a computer, it is called the **Stored Program Control (SPC)**. The following figure shows the basic control structure of an SPC telephony exchange.



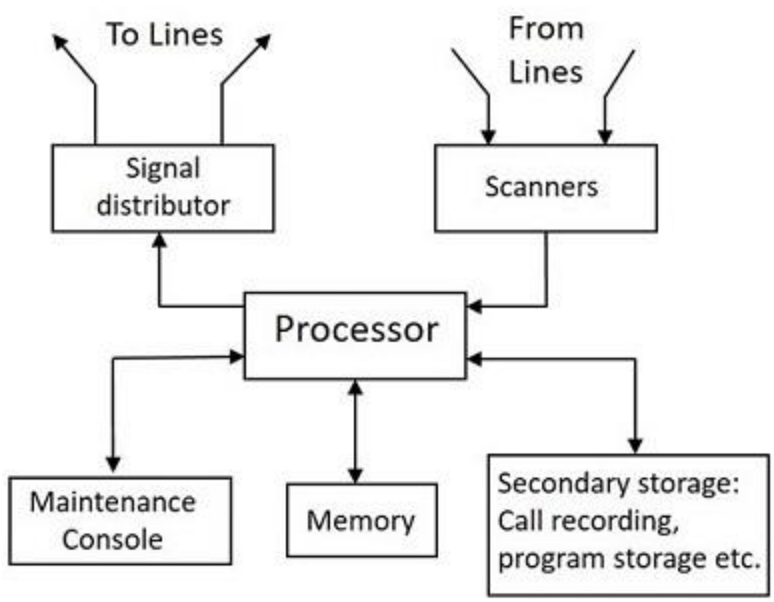
- The processors used by SPC are designed based on the requirements of the exchange. The processors are duplicated; and, using more than one processor makes the process reliable. A separate processor is used for the maintenance of the switching system.

There are two types of SPCs –

- Centralized SPC
- Distributed SPC

Centralized SPC

The previous version of Centralized SPC used a single main processor to perform the exchange functions. The dual processor replaced the single main processor at a later stage of advancement. This made the process more reliable. The following figure shows the organization of a typical Centralized SPC.



A dual processor architecture may be configured to operate in three modes like –

- Standby Mode
- Synchronous Duplex Mode
- Load Sharing Mode

Standby Mode

- As the name implies, in the two processors present, one processor is active and the other is in the standby mode.
- The processor in the standby mode is used as a backup, in case the active one fails.
- This mode of exchange uses a secondary storage common to both the processors. The active processor copies the status of the system periodically and stores in the axis secondary storage, but the processors are not directly connected.
- The programs and instructions related to the control functions, routine programs and other required information are stored in the Secondary storage.

Synchronous Duplex Mode

- In the Synchronous Duplex mode, two processors are connected and operated in synchronism. Two processors P1 and P2 are connected and separate memories like M1 and M2 are used. These processors are coupled to exchange the stored data. A Comparator is used in between these two processors. The Comparator helps in comparing the results.
- During the normal operation, both of the processors function individually receiving all the information from the exchange and also related data from their memories. However, only one processor controls the exchange; the other one remains in synchronism with the previous one. The comparator, which compares the results of both the processors, identifies if any fault occurs and then the faulty processor among them is identified by operating them individually. The faulty processor is brought into service only after the rectification of fault and the other processor serves meanwhile.

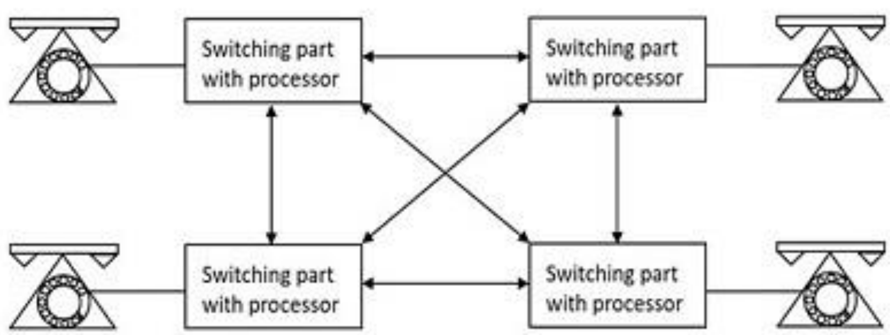
Load Sharing Mode

- Load sharing mode is where a task is shared between two processors. The Exclusion Device (ED) is used instead of the comparator in this mode. The processors call for ED to share the resources, so that both the processors do not seek the same resource at the same time.
- In this mode, both the processors are simultaneously active. These processors share the resources of the exchange and load. In case one of the processor fails, the other one takes over the entire load of the exchange with the help of ED. Under normal operation, each processor handles one-half of the calls on a

statistical basis. The exchange operator can however vary the processor load for maintenance purpose.

Distributed SPC

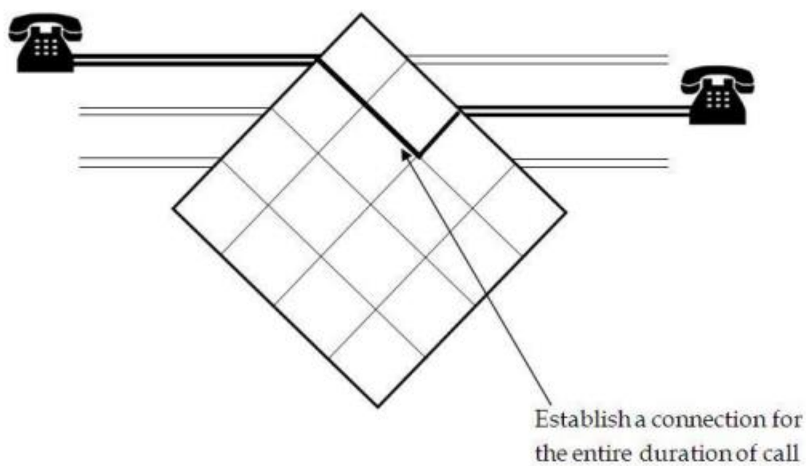
- Unlike Electromechanical switches and Centralized SPC, the introduction of Distributed SPC has enabled to provide a wide range of services.
- This SPC has separate small processors called the **Regional Processors** that deal with different works, rather than just one or two processors working on the whole thing like in the centralized system.
- However, when these regional processors are required to perform complex tasks, the centralized SPC helps by directing them.
- The Distributed SPC has more availability and reliability than Centralized SPC, because entire exchange control functions may be decomposed either horizontally or vertically for distributed processing.
- Such distributed control where switching equipment is divided into parts, each of which have its own processor, is indicated in the figure below.



- The exchange environment in vertical decomposition is divided into several blocks and each block is assigned to a processor that performs all the control functions that are related to specific block of equipment, whereas each processor in horizontal decomposition performs one or some of the exchange control functions.

Space Switching:

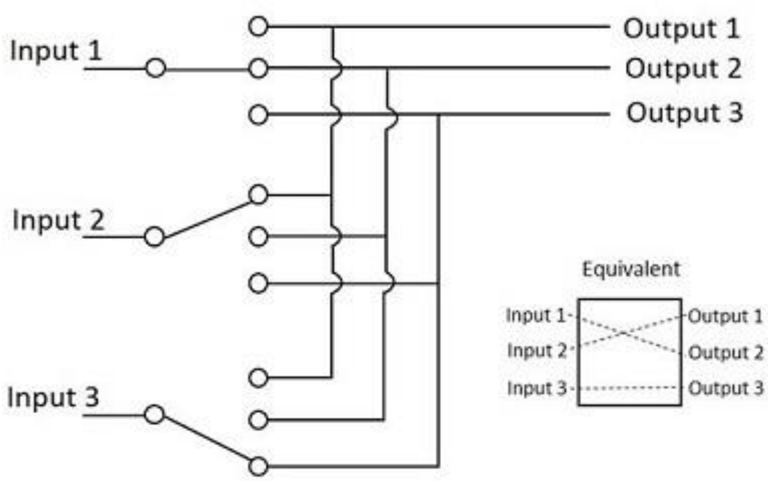
- When we consider Space switching there is a dedicated path (two parallel wires) established between the caller and called subscribers for the entire duration of call in the exchange by the switch.
- It was originally designed for analog networks, but is used currently in both digital and analog switching.
- This means then the conversation is going on the switch create the link between two sides. At that time only that call is going in the path.



Space Division Switching

- In space division switching, a dedicated path is established between the calling and the called subscribers for the entire duration of the call.
- The paths in a circuit are separated from each other, spatially in space division switching.
- Though initially designed for analog networks, it is being used for both analog and digital switching.

- A Crosspoint switch is mostly referred to as a space division switch because it moves a bit stream from one circuit or bus to another.
- The switching system where any channel of one of its incoming PCM highway is connected to any channel of an outgoing PCM highway, where both of them are spatially separated is called the Space Division Switching.
- The Crosspoint matrix connects the incoming and outgoing PCM highways, where different channels of an incoming PCM frame may need to be switched by different Crosspoints in order to reach different destinations.



- Though the space division switching was developed for the analog environment, it has been carried over to digital communication as well.
- This requires separate physical path for each signal connection, and uses metallic or semiconductor gates.

Advantages of Space Division Switching

Following is the advantage of Space Division Switching –

- It is instantaneous.

Disadvantages of Space Division Switching

- Number of Crosspoints required to make space-division switching are acceptable in terms of blocking.

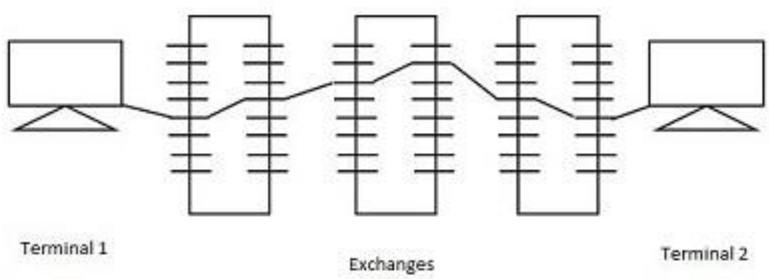
Time Switching

- Time division switching involves the sharing of cross points for shorter periods of time. This paves way for the reassign of cross points and its associated circuits for other needed connections.
- Therefore, in time division switching, greater savings in cross points can be achieved. Hence, by using dynamic control mechanisms, a switching element can be assigned to many inlet-outlet pairs for few microseconds.
- This is the principle of time division switching.
- Time division switching uses time division multiplexing to achieve switching.
- Two popular methods that are used in time division multiplexing are
- (a) the time slot interchange (TSI) and
- (b) the TDM bus.
- In ordinary time division multiplexing, the data reaches the output in the same order as they sent. But TSI changes the ordering of slots based on the desired connections.

Time Division Switching

- In time division switching, sampled values of speech signals are transferred at fixed intervals.
- Time division switching comes under digital switching techniques, where the Pulse Code Modulated signals are mostly present at the input and the output ports.
- A digital Switching system is one, where the inputs of any PCM highway can be connected to the outputs of any PCM highway, to establish a call.
- The incoming and outgoing signals when received and re-transmitted in a different time slot, is called **Time Division Switching.**

- The digitized speech information is sliced into a sequence of time intervals or slots. Additional voice circuit slots, corresponding to other users are inserted into this bit stream of data.
- Hence, the data is sent in time frames.
- The main difference between space division multiplexing and time division multiplexing is sharing of Crosspoints.
- Crosspoints are not shared in space division switching, whereas they can be shared in time division multiplexing, for shorter periods.
- This helps in reassigning the Crosspoints and its associated circuitry for other connections as well.



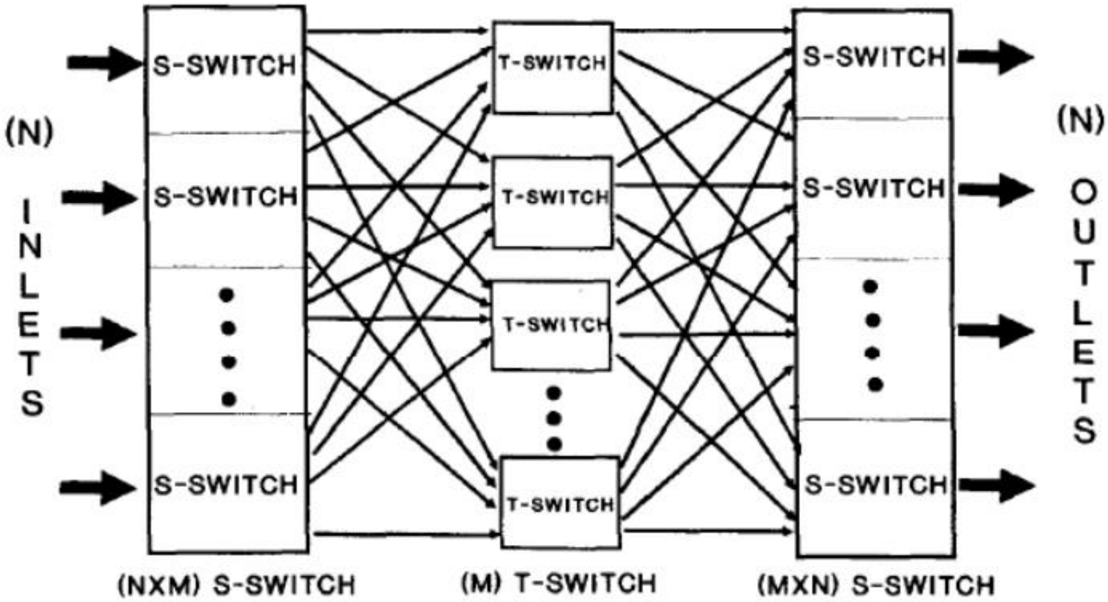
- Time division switches use time division multiplexing, in switching.
- The two popular methods of TDM are TSI (Time and Slot Interchange) and TDM bus.
- The data sent at the transmitter reaches the receiver in the same order, in an ordinary time division multiplexing whereas, in TSI mechanism, the data sent is changed according to the ordering of slots based on the desired connections.
- It consists of RAM with several memory locations such as input, output locations and control unit.
- Both of the techniques are used in digital transmission.
- The TDM bus utilizes multiplexing to place all the signals on a common transmission path.
- The bus must have higher data rate than individual I/O lines.
- The main advantage of time division multiplexing is that, there is no need of Crosspoints. However, processing each connection creates delay as each time slot must be stored by RAM, then retrieved and then passed on.

ADVANTAGES & DISADVANTAGES OF TIME DIVISION SWITCHING

- The advantage of Time-division switching is that it needs no crosspoints.
- Its disadvantage in the case of TSI,is that processing each connection creates delays.
- Each time slot must be stored in the RAM,then retreived and passed on.

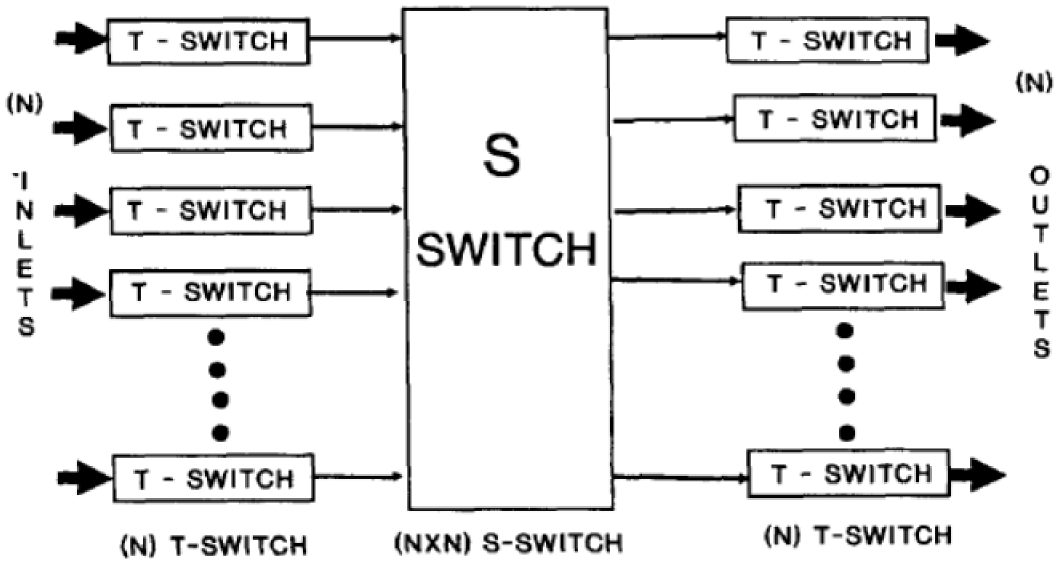
Space-Time-Space (STS) Switching:

- One objective in the design of a modern digital switching system is to re-duce costs and improve the switching efficiency of the fabric.
- Obviously, there is a practical limit to the size of a single switching stage that can be effectively utilized.
- At present, various combinations of S switches and T switches are used to accomplish the above objective. One combination uses an S switch followed by a T switch and a final S switch.
- This arrangement, referred to as STS fabric, is shown in Fig. 2.6.
- This particular arrangement depicts N x M (meaning N inputs and M outputs) size, with NS switches separated by MT switches. In an STS switching fabric, a path through the network is established via smart network controllers that link an incoming time slot with an outgoing time slot.
- This type of time slot linkage is then dynamically updated throughout the duration of a call.



Time - Space-Time (TST) Switching:

- One of the most popular switching fabric arrangements currently deployed by digital switching systems is based on time-space-time (TST) architecture, as shown in Fig. 2.7
- An incoming time slot enters a T switch; a path is hunted through the S switch for an appropriate outgoing time slot; and once identified, the path through the switching fabric is established and dynamically updated throughout the duration of the call.
- One of the basic advantages of the TST architecture over the STS architecture is that it can be implemented at a lower cost, since T switches are less expensive than S switches and under heavy traffic offer more efficient utilization of time slots with lower blocking probabilities.



Compare STS VS TST:

SR. No.	STS	TST
1.	Input space block is interfaced to output space block using a time block in between.	Input time block interfaced to output time block by using a space block in between.
2.	Not preferred now since size of 's' block being a matrix increases as square of inputs.	Preferred now as size of time switch increases linearity with number of input and output buses.
3.	Now costly due to cost of switching hardware because two space switch blocks are required.	Now economical due to availability of low cost high speed memories required for T-blocks.
4.	For large size, size can be limited by splitting S blocks S-S-T-S-S.	Small size can be further reduced by T-S-S-S-T.
5.	Peripheral functions cannot be incorporated into S block.	Peripheral functions such as super MUX alignment of PCM with exchange frame can be incorporated into T-S block.
6.	Design is very simple.	Design is complicated.
7.	Memory size is large.	Memory size is small.
8.	Only few applications.	Applications are More.

Strowger Rotary dial mechanism:

Rotary Dial Telephone

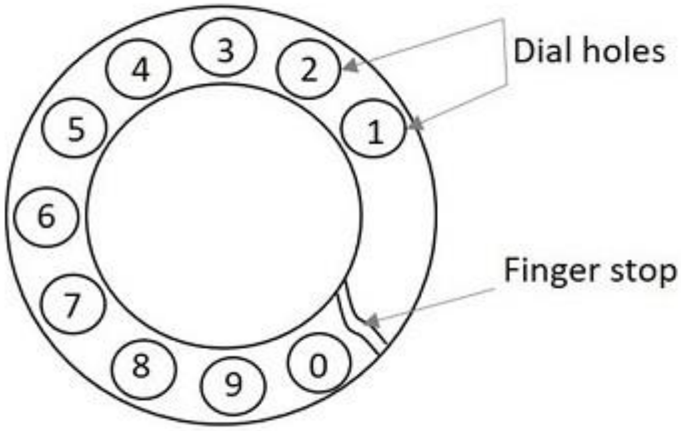
In this section, we will learn about what the Rotary Dial Telephone is and how it works. To start with, we will discuss the drawbacks that were prevalent before the invention of the Rotary Dial Telephone.

The pulse dialing technique is where there is making and breaking of the subscriber loops. This might disturb and affect the performance of speaker, microphone and bell contained in the telephone. In addition, the dialing timings should not affect the timing of the pulse train as this will lead to the dialing of a wrong number.

The Rotary Dial Telephone came into existence to solve the problems prevailing then. The microphone and the loudspeaker are combined and placed in the receiver set. The set has a finger plate the arrangement of which makes the dialing time appropriate. The below figure shows how a rotary dial looks like.

The dial is operated by placing the finger in the hole appropriate to the digit to be dialed. Now, drawing the fingerplate round in the clockwise direction to the finger stop position and letting the dial free by withdrawing the finger, makes a number dialed. The fingerplate and the associated mechanism now return to the rest position under the influence of a spring. The dial is ready for the next number.

The dial pulses are produced during the return travel of the fingerplate, thus eliminating the human element in pulse timings. The following figure shows the dial holes and finger stop.

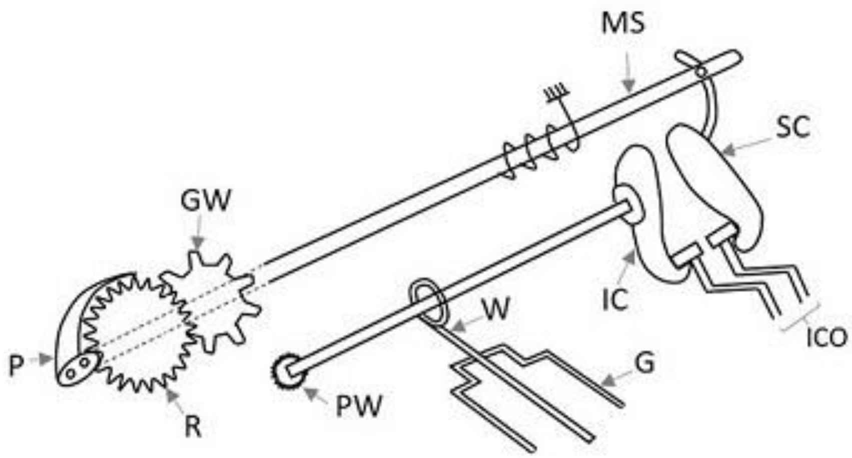


A rotary dial phone uses the following for implementing pulse dialing –

- Finger plate and spring
- Shaft, gear and Pinion wheel
- Pawl and ratchet mechanism
- Impulsing cam and suppressor cam or a trigger mechanism
- Impulsing contact
- Centrifugal governor and worm gear
- Transmitter, Receiver and bell by-pass circuits

Internal Mechanism

The cam mechanism or trigger mechanism helps in dialing. This mechanism is used in operating the Impulsing contact. Let us consider the operation of the rotary dial telephone using the cam mechanism. The following figure will help you understand the internal mechanism.



G = Governo
GW = Gear Wheel
IC = Impulsing Cam
ICO = Impulsing Contacts
MS = Main Shaft

P = Pawl
PW = Pinion Wheel
R = Rachet
SC = Suppressor Cam
W =Worm gear

- The suppressor cam helps in keeping the Impulsing cam away from the Impulsing contacts. When the rotary dial is in rest position, then the Impulsing contacts are away from the Impulsing cam. When a number is dialed, by placing the finger in the dial hole, which means the dial is displaced from its position, then the Impulsing contacts come near the Impulsing cam. This rotation of the finger plate, causes the rotation of the Main shaft.

Multi stage switching:

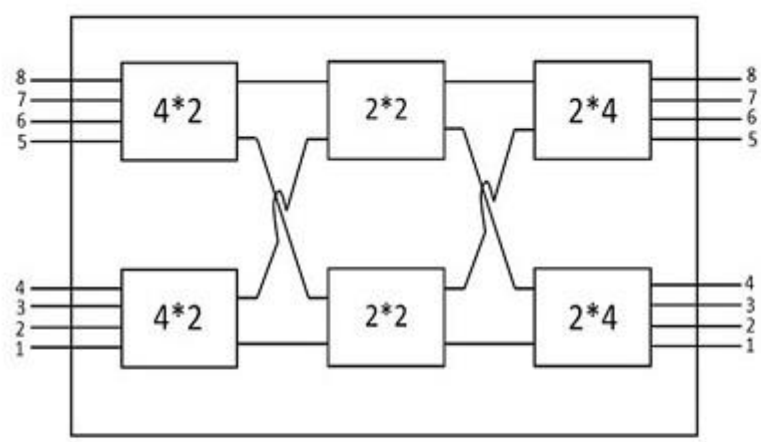
Multi-Stage Networks

The multi-stage networks are the networks built to provide connections between more subscribers more efficiently than the Crossbar switching systems.

The Crossbar switching networks discussed previously have some limitations as described below –

- The number of Crosspoint will be the square of the number of attached stations and hence this is costly for a large switch.
- The failure of Crosspoint prevents connection with those two subscribers between which the Crosspoint is connected.
- Even if all the attached devices are active, only few of the Crosspoints are utilized

In order to find a solution to subsidize these disadvantages, the multistage space division switches were built. By splitting the Crossbar switch into smaller units and interconnecting them, it is possible to build multistage switches with fewer Crosspoints. The following figure shows an example of a multistage switch.



The multistage switch like the above one needs less number of Crosspoints than the ones needed in Crossbar switching. According to the example shown above, for the 8 (input) and 8 (output) various subscribers (both called and calling subscribers), the Crosspoints needed in a normal Crossbar network will be square of them, which is 64. However, in the multistage Crossbar network, just 40 Crosspoints are enough. This is as shown in the diagram above. In a large multistage Crossbar switch, the reduction is more significant.

Advantages of a Multistage Network

The advantages of a multistage network are as follows –

- The number of Crossbars are reduced.
- The number of paths of connection can be more.

- It reduces the number of cross points.
- If one path fails, then there will be an availability of another path.

Disadvantages of a Multistage Network

The disadvantage of a multistage network are as follows –

- Multistage switches may cause **Blocking**.
- The number or size of the intermediate switches if increased can solve this problem, but the cost increases with this.

One stage two stage and three stage networks in the book [IN BOOK PAGE NO 129]

The basic function of a switching system in page no 56-57 in book.

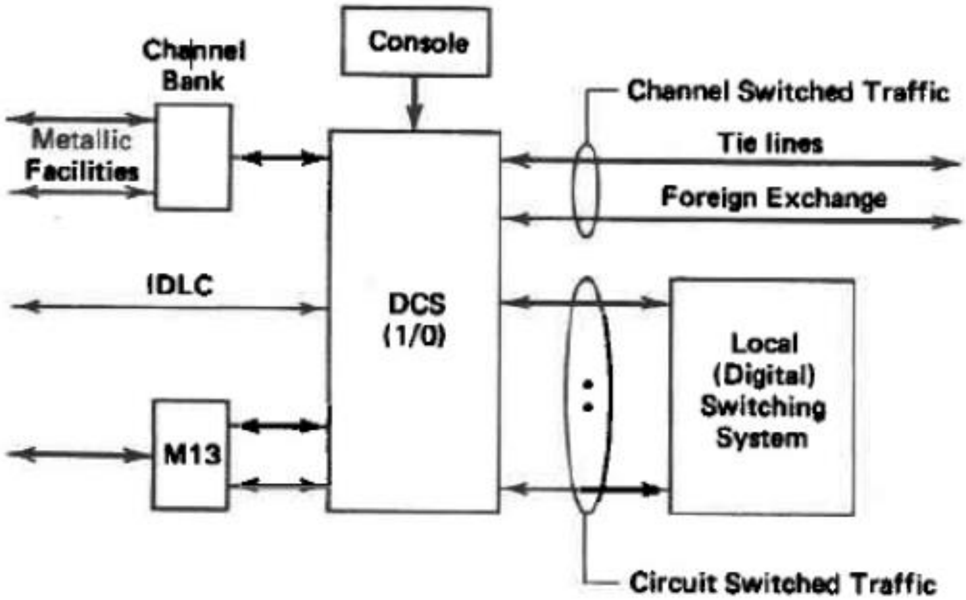
Differences between single and multistage network:

Sr. No.	Single Stage	Multi Stage
1.	Inlet to outlet connection is by a single cross point.	Inlet to Outlet connection is by multiple cross points
2.	Utilization of single cross point per connection results in better quality link.	Utilization of multiple cross points may degrade the quality of a connection.
3.	All individual cross point can be utilized for only single inlet/outlet pair connection.	Similar cross point can be used establish connection in between a number of inlet/outlet pairs.
4.	An exact cross point is required for each exact connection.	An exact connection may be established through using sets of cross points.
5.	If a cross points fails, connected connection cannot be establish. Means there is no redundancy.	Alternative paths and cross-points are available.
	Cross points are incompetently used. Only one cross point in every row or column of a triangular or square switch matrix is even in utilize, even if every line is active.	Cross points are used

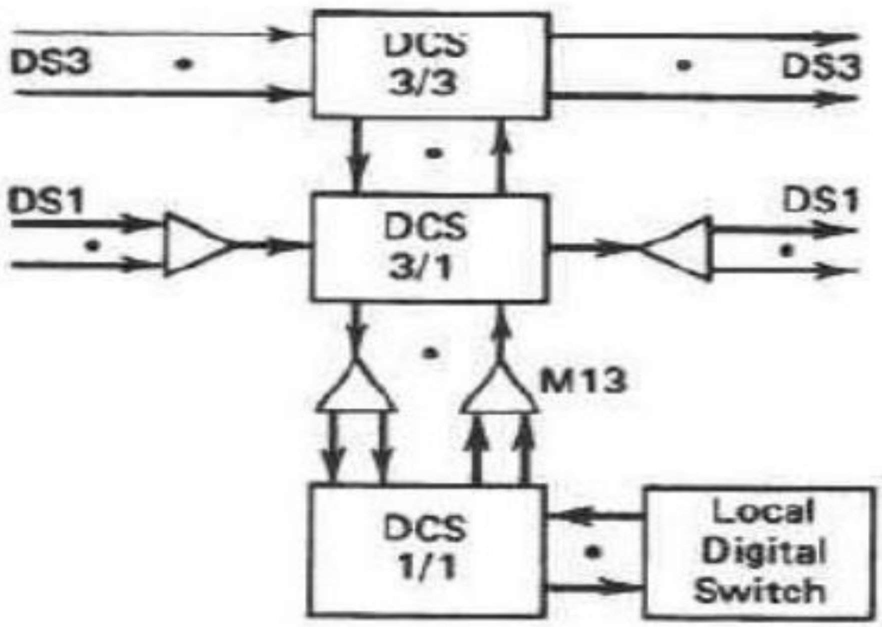
6.		Efficiently
7.	Number of cross points is Prohibitive	Number of cross points is reduced considerably
8.	A huge number of cross points in every inlet/outlet leads to capacitive loading.	There is no problem of capacitive loading
9.	The network is non-blocking in nature	The network is blocking in nature
10.	Less Time for establishing a call.	More time for establishing a call.

Digital Cross Connect

- A digital cross-connect system (DCS or DXC) is a piece of circuit-switched network equipment, used in telecommunications networks, that allows lower-level TDM bit streams, such as DS0 bit streams, to be rearranged and interconnected among higher-level TDM signals, such as DS1 bit streams.
- DCS units are available that operate on both older T-carrier/E-carrier bit streams, as well as newer SONET/SDH bit streams.
- DCS devices can be used for "grooming" telecommunications traffic, switching traffic from one circuit to another in the event of a network failure, supporting automated provisioning, and other applications.
- Having a DCS in a circuit-switched network provides important flexibility that can otherwise only be obtained at higher cost using manual "DSX" cross-connect patch panels.
- It is important to realize that while DCS devices "switch" traffic, they are *not* packet switches—
- they switch *circuits*, not packets, and the circuit arrangements they are used to manage tend to persist over very long-time spans, typically months or longer, as compared to packet switches, which can route every packet differently, and operate on micro- or millisecond time spans.
- DCS units are also sometimes colloquially called "DACS" units, after a proprietary brand name of DCS units created and sold by AT&T's Western Electric division, now Alcatel-Lucent.
- Modern digital access and cross-connect systems are not limited to the T-carrier system, and may accommodate high data rates such as those of SONET.



Electronic digital cross-connect system.



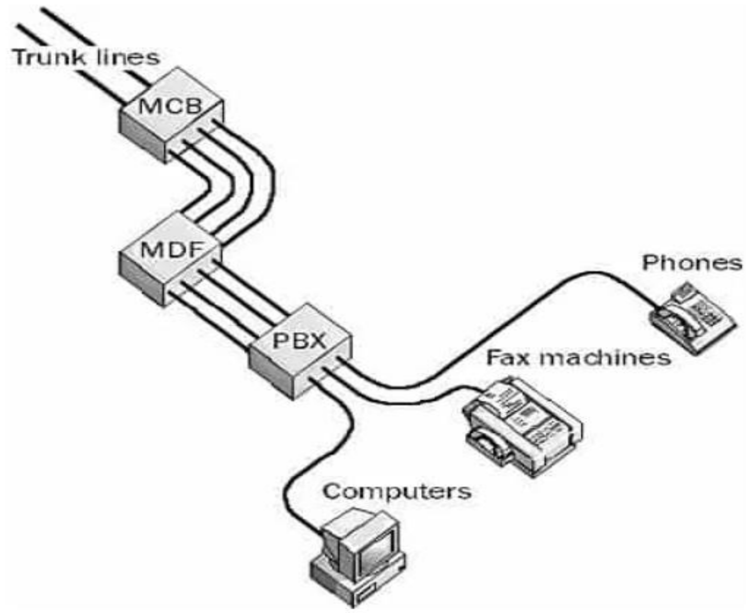
DCS hierarchy.

Private Branch Exchange (PBX)

- A private branch exchange is a telephone system within an enterprise that switches calls between users on local lines, while enabling all users to share a certain number of external phone lines.
- In contrast to a public switched telephone network (PSTN), the main purpose of a PBX is to save the cost of requiring a line for each user to the telephone company's central office.
- PBX solutions used to be tools specifically designed to allow for internal networks to operate within a business environment.
- PBX redesigned how companies handle calls, minimizing the demands on corporate budget and opening the door to digital transformation.
- Today's PBX system can split a single phone line into various private lines, all identified by extensions.
- This means anyone can reach someone else in an office through a single phone number.

Features of PBX systems include:

- **Managing and completing calls:** PBX solutions allow for the management of calls and the control of various PBX branches. Operators can restrict or permit international and local dialing options based on phone service.
- **Call transferring:** You can transfer calls between users and departments with ease. This makes it easy to establish and maintain connections in a company with minimal dropped calls.
- **Advanced calling features:** PBX solutions can now be equipped with tools for Direct Inward Dialing (DID), IVR (Interactive Voice Response), call recording, and other features.
- **Local connections:** Offering users access to local numbers in cities where they aren't physically present for a more local brand experience.
- **Call control:** Call blocking, forwarding, logging, transferring, call waiting, and other calling features.
- **Office connections:** Connecting multiple office locations within the same phone system keeps employees in different destinations connected.



PBX – Private Branch Exchange

PBXs support a number of features, including the following:

- **Direct Inward Dialing (DID):** A form of call routing that allows outside users to dial directly to any of the extensions
- **Direct Outward Dialing (DOD):** A form of call routing that allows extensions to dial directly to any outside phone number
- **Station-to-Station Dialing (SSD):** Allows any extension to call any other extension without using a business line

Advantages

- Powerful calling features included
- Flexibility to work remotely
- Superior reliability and uptime
- Encrypted communications

Disadvantages

✗ Needs a broadband connection

✗ Adjusting network settings

✗ Users might need guidance

Explain random write sequential and read operations in time switch.

- In a time-switched system, random write and sequential read operations refer to the way in which data is accessed and processed in a computer system.
- Random write operations involve writing data to random locations in the system's memory or storage.
- This type of operation is often used when data is being added or updated in a database, for example. It requires the system to locate the appropriate location in memory and write the data there, which can take some time.
- Sequential read operations, on the other hand, involve reading data in a sequential or linear fashion.
- This means that the system reads data from one location and then moves on to the next location in sequence.
- Sequential read operations are often used when data is being read from a file or database and processed in a specific order.
- These operations can be faster than random write operations because the system does not have to spend time locating the appropriate location in memory.

- In a time-switched system, these operations may be performed at different times or in different orders, depending on the needs of the system and the tasks it is performing.

- For example, a system may perform a batch of random write operations and then switch to a series of sequential read operations, or it may alternate between the two types of operations as needed.

Design the Trucking Diagram of 10,000-line step-by-step exchange.

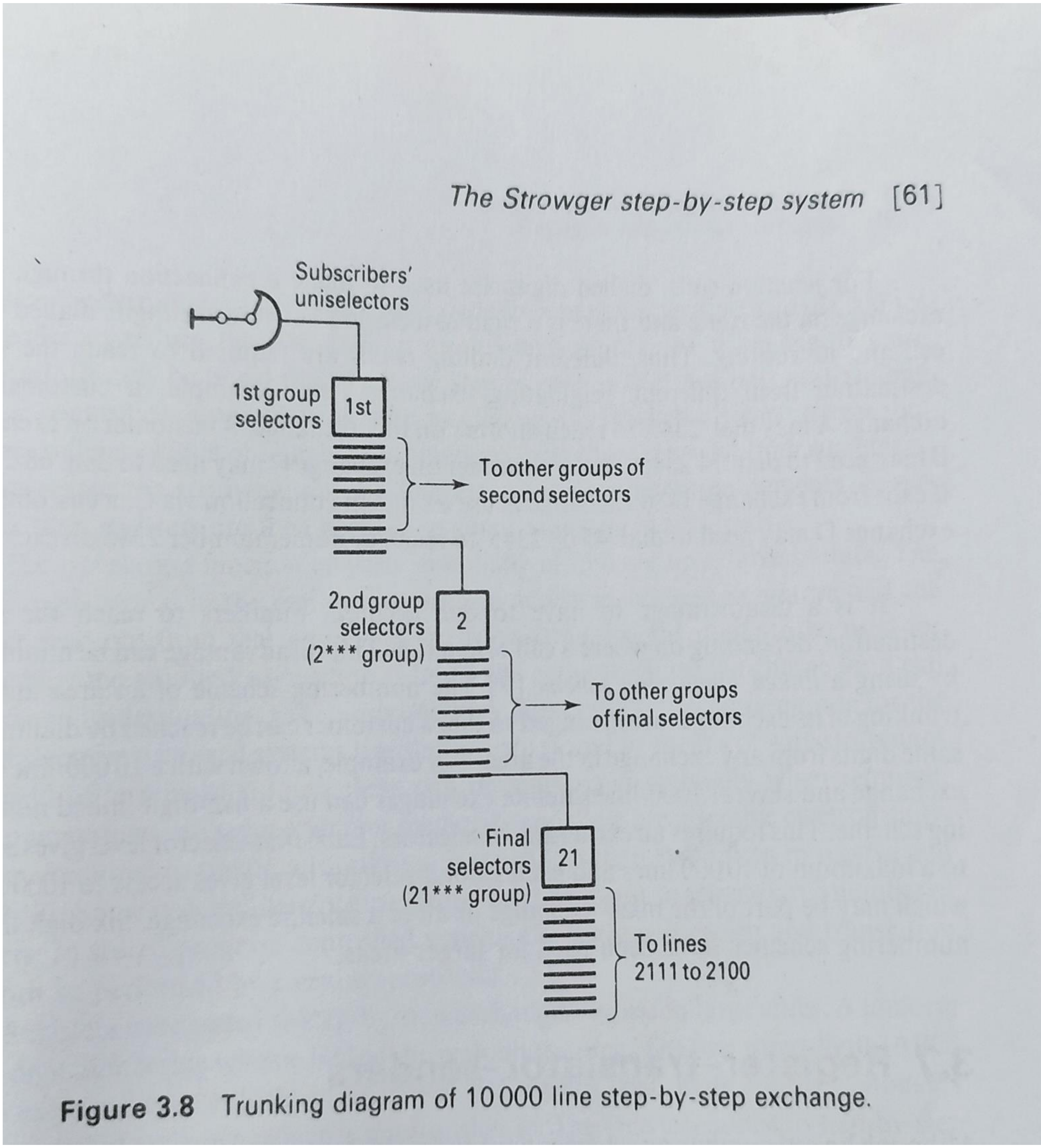

The Strowger step-by-step system [61]

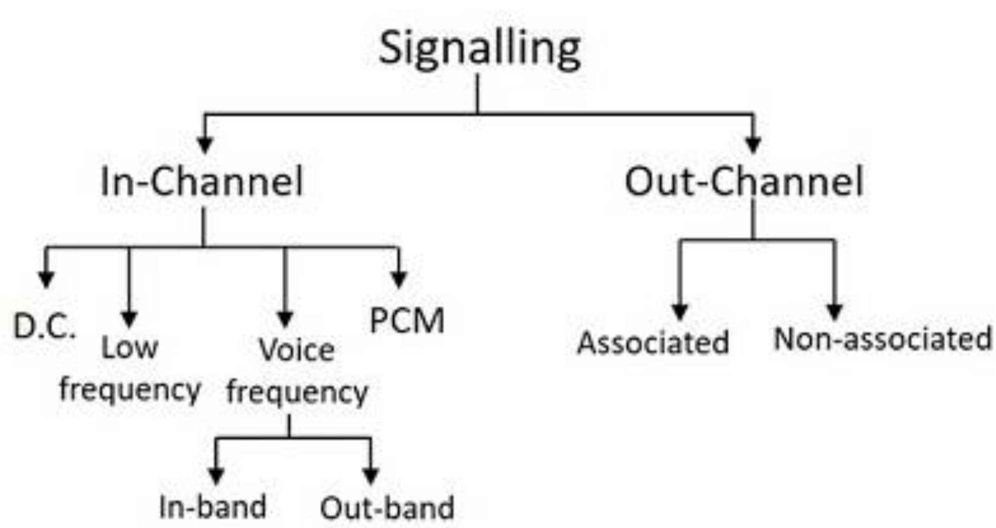
Figure 3.8 Trunking diagram of 10 000 line step-by-step exchange.

CHAPTER 6: SIGNALLING IN TELEPHONE

Signaling

- In telecommunication, signaling is the use of signals for controlling communications.
- Signaling refers to the information exchanges between terminal devices, exchanges, and routers for setting up circuits, termination, billing, advanced network services, etc.
- Signaling techniques enable the circuit to function as a whole by inter connecting all varieties of switching systems.
- There are three forms of signaling involved in a telecommunication network.
    - **Subscriber loop signaling**: The subscriber loop signaling depends upon the type of telephone instrument used.
    - **Intra exchange or register signaling**: The intra exchange signaling refers to the internal portion of a switching system that is heavily dependent upon the type and design of a switching system, which varies depending upon the model.
    - **Inter exchange or inter-register signaling:** The inter-exchange signaling takes place between exchanges. This helps in the exchange of address digits, which pass from exchange to exchange on a link-by-link basis.

Types of Signalling:



i) In-Channel Signaling

- **I**n-Channel Signaling is also known as **Per Trunk Signaling**.
- This uses the same channel, which carries user voice or data to pass control signals related to that call or connection.
- No additional transmission facilities are needed, for In-channel signaling.
- This type of signaling is used to carry voice or data and pass control signals related to a call or connection.
- There are different types of In-channels Signaling, as seen in the above figure. The D.C. signaling is simple, cheap and reliable even for unamplified audio circuits. However, for amplified audio circuits, low frequency A.C. signaling may be adopted.
- The Voice Frequency signaling is used when FDM (Frequency Division Multiplexing) transmission systems are used, because low frequency signaling and D.C. signaling cannot be provided. This Voice Frequency signaling may be **In-band** or **Out-band**.

a) In-band Signaling

- In-band voice frequency uses the same frequency band as the voice, which is 300-3400 Hz, which has to be protected against false operation by speech.
- One such instant took place when a lady's voice which has generated a tone at around 2600Hz lasting for a duration of 100ms was detected as the line disconnect signal due to which her calls were frequently being disconnected in the middle of her conversation. Such problems precluded the in-band signaling during speech phase.

The advantages of In-band signaling are −

- The control signals can be sent to every part where a speech signal can reach.
- The control signals will be independent of the transmission systems as they are carried along with the speech signals.
- The Analog to digital and Digital to analog conversion processes will not affect them.

b) Out-band Signaling

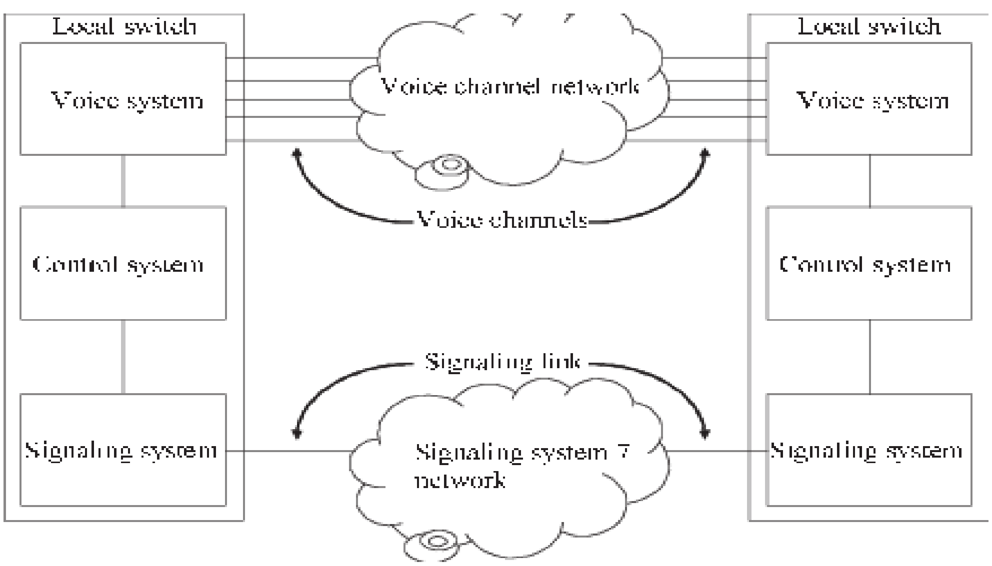
- The out-band signaling uses frequencies which are above the voice band but below the upper limit of 4000 Hz of the nominal voice channel spacing.

- The signaling is done throughout the speech period and thus continuous supervision of the call is allowed.
- Extra circuits are needed to handle the extremely narrow band width of this signaling, due to which it is seldom used.
- Both of these in-band and out-band voice frequency signaling techniques have limited information transmission capacity. In order to provide enhanced facilities, common channel signaling is used.

ii) Common Channel Signaling

- Common Channel Signaling uses a separate common channel for passing control signals for a group of trunks or information paths as it does not use the speech or the data path for signaling.
- The common channel signaling consists of two types of nodes such as **Signaling Transfer Points** (STP) and **Signaling Points** (SP).
- A Signaling point is capable of handling control messages directly addressed to it but is incapable of routing messages.
- Signaling transfer point is capable of routing messages and can perform the functions of SP.

Ccs between the central processor:



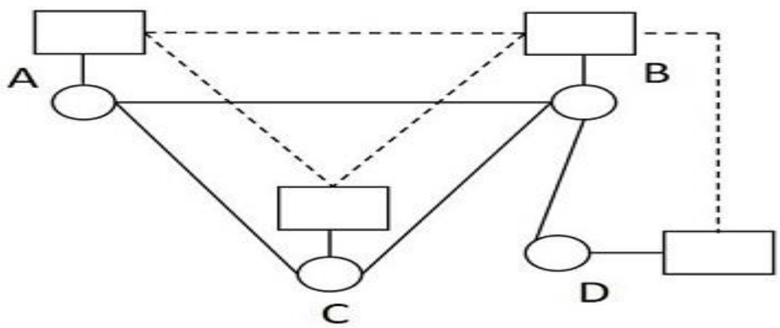
This common channel signaling is implemented in two modes −

- Channel associated mode
- Channel non-associated mode

a) Channel-associated Mode

- In the channel-associated mode, the channel closely tracks the trunk groups along the entire length of the connection.
- Here, the signaling is done on a separate channel; the signaling path passes through the same set of switches, as does the speech path.

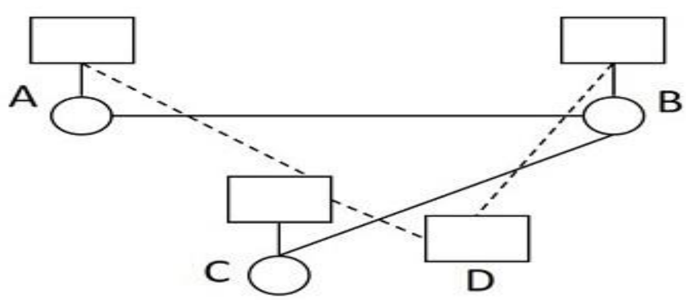
The following figure shows the associated mode of operation in common channel Signalling



- The signaling paths for the speech paths A-B, A-C-B and B-D are A-B, A-C-B and B-D respectively. The advantages of this signaling are −
  - The implementation is economic
  - The assignment of trunk groups is simple

b) Channel Non-associated Mode

- In the channel non-associated mode, there is no close or simple assignment of the control channels to trunk groups.
- It follows a different path from that of the speech signal as shown in the following figure.



- The signaling paths for the speech paths A-B and B-C are A-C-D-B and B-D-C respectively.
- The network topologies are different for signaling and speech networks.
- Though this scheme offers flexibility as there is no switching center, it is a bit complex, as the signal messages may be transferred between the two end switching systems via any available path in the common channel signaling network according to its own routing principles.

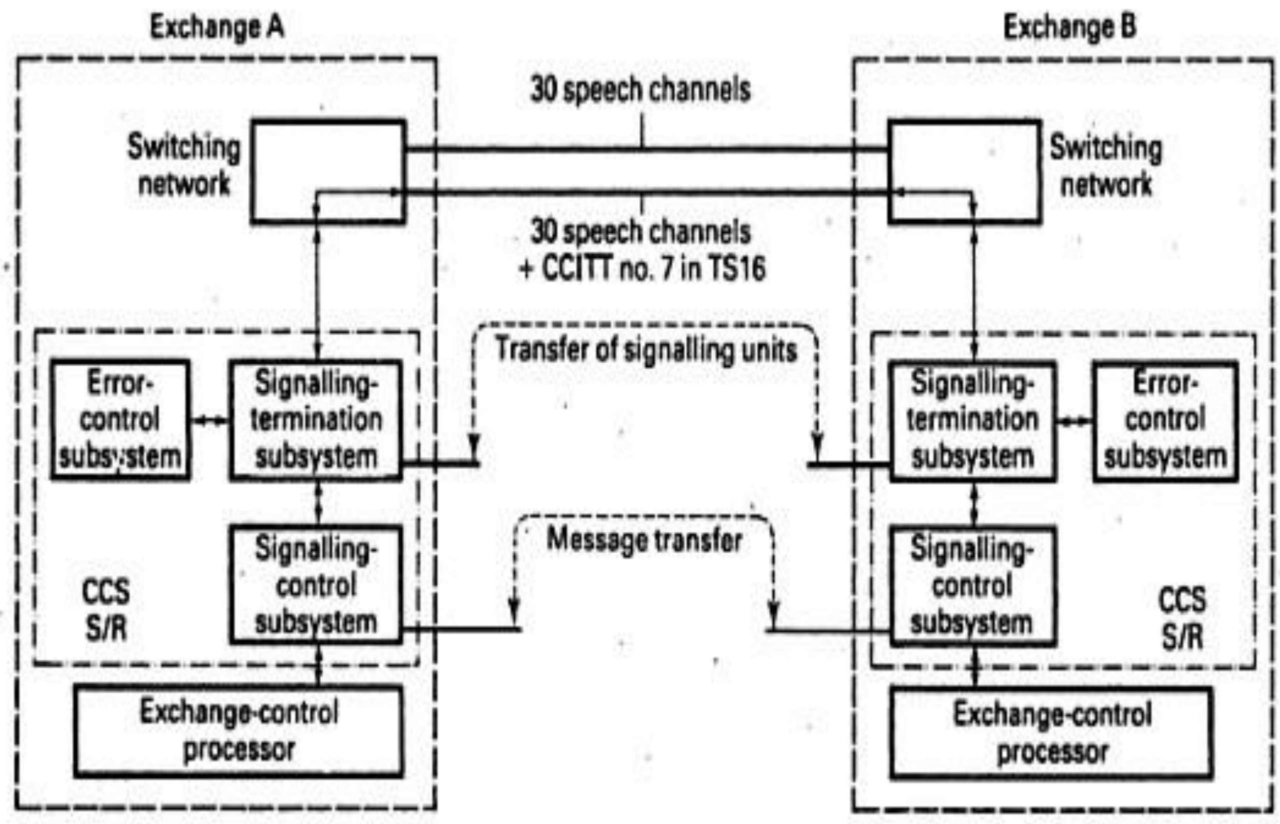
Differences between In Channel Signalling and Common Channel Signalling

In Channel	**Common Channel**
• Trunks are held up during Signalling.	• Trunks aren't required for Signalling.
• Signal repertoire is limited.	• Extensive signal repertoire is possible.
• Interference between voice and Control signals may occur.	• No interference as the two Channels is physically separate.
• Separate Signalling	• Only one set of Signalling
• Equipment is needed for each trunk and thus expensive.	• equipment's is essential for a whole group of trunk Circuits and hence CCS is economical
• Voice channel being the control channel, there is a possibility of potential misuse by the customers.	• Control Channel is in general in accessible to users.
• Signalling is comparatively slow.	• Signalling is significantly fast.
• It is difficult to change or add signals.	• There is flexibility to change or add signals.
• It is difficult to handle Signalling during speech period.	• Signals during speech. There is freedom to handle
• Reliability of Signalling path is not Critical.	• Reliability of the signaling Path is critical.

- Speech circuit reliability is assured.
- There is no automatic test of speech circuit.

Common channel signaling is better than In-channel signaling.

Explain the architecture of SS7.



Block schematic diagram of CCITT no.7 signalling system.

- A block schematic diagram of the CCITT no. 7 signaling system is demonstrated in figure.

- Signal messages are passed by the central processor of the sending exchange to the CCS system. It consists of the microprocessor depends subsystem.
- The signaling termination subsystem, the error control subsystem and the signaling control subsystems.
- The signaling control subsystem formations the messages in the suitable format and queues them for transmission.
- While there are no messages to send, this generates filler messages to remain the link active.
- Messages after that passed to the signaling termination sub-system, here whole signal units (SU) are assembled BY using sequence numbers and check bits generated through the error control subsystem.
- On the receiving terminal, the reverse sequence is continued. The levels are given below as:

Level 1: The Physical Layer

Level 2: The Data Link Level

Level 3: The signaling network level

Level 4: The User Part

Signalling Units Used in SS7

- Signaling information is passed over the signaling links in messages, which are called signal units.
- Signal units are continuously transmitted in both directions on any link that is in service.
- SS7 uses three different types of signal units:
  - Message Signal Units (MSUs)
  - Link Status Signal Units (LSSUs)
  - Fill-In Signal Units (FISUs)

  A signaling point sends FISUs over the link when it does not have any MSUs or LSSUs to transmit.

Signal Unit Structure

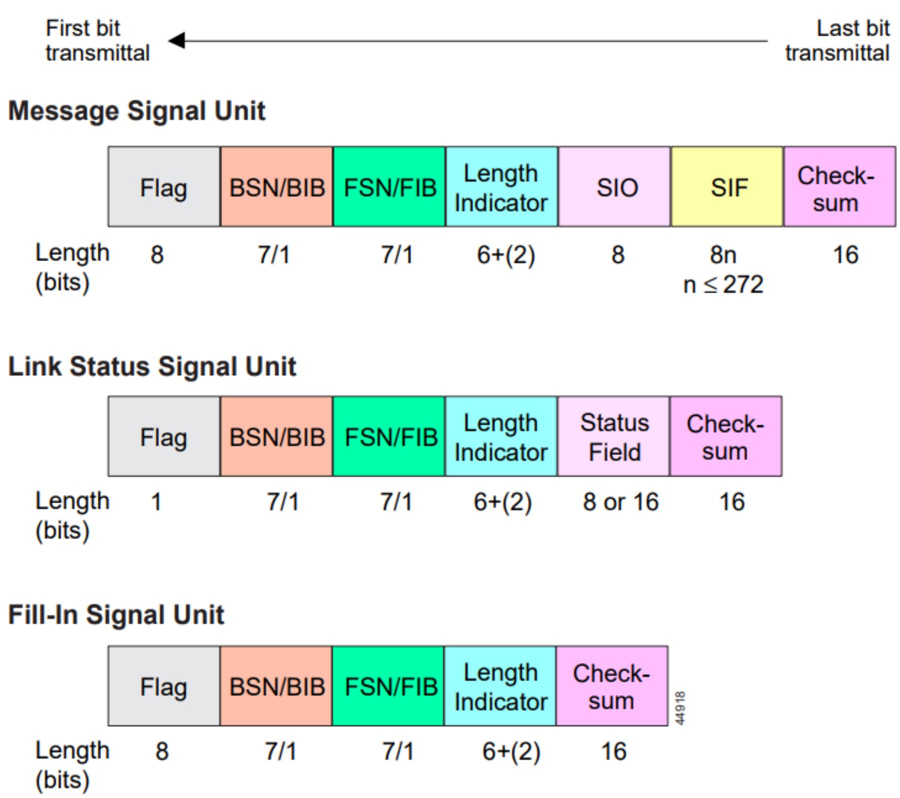
All types of signal units (MSU, LSSU, FISU) have a set of common fields which are used by MTP Level 2. Field types include the following:

- **Flag**—Delimiter in a signal unit which marks the end of one signal unit and the beginning of another. All signal units begin with a distinct 8-bit pattern (0111 1110).

Note Although the protocol allows an opening and closing flag, only one flag is used in North America.

- **Checksum**—An 8-bit sum calculated from the transmitted message by the transmitting signaling point and inserted in the message. It is recalculated by the receiving signaling point, and if corrupted, a retransmission is requested.
- **Length Indicator**—The number of octets between itself and the checksum. Checks the integrity of the signal unit and discriminates between different types of signal units. The default values are: FISU=0, LSSU=1 or 2, MSU>2
- **BSN/BIB FSN/FIB**—Octets that hold the backward sequence number (BSN) and backward indicator bit (BIB); the forward sequence number (FSN) and the forward indicator bit (FIB).

Figure 4-1 SS7 Signal Unit Types



Signal Unit Flow Control

- The BSN/BIB and FSN/FIB fields in a signal unit (SU) confirm receipt of SUs and ensure that they are received in the order in which they were transmitted.
- These fields also provide flow control.
- MSUs and LSSUs are assigned a sequence number when transmitted.
- That sequence number is placed in the FSN field of the outgoing signal unit, which is stored by the transmitting signaling point until it is acknowledged by the receiving signaling point.
- Signaling points acknowledge receipt of SUs by putting the sequence number of the last correctly received (and in sequence) SU in the backward sequence number (BSN) of every SU they transmit.

SU Error Detection

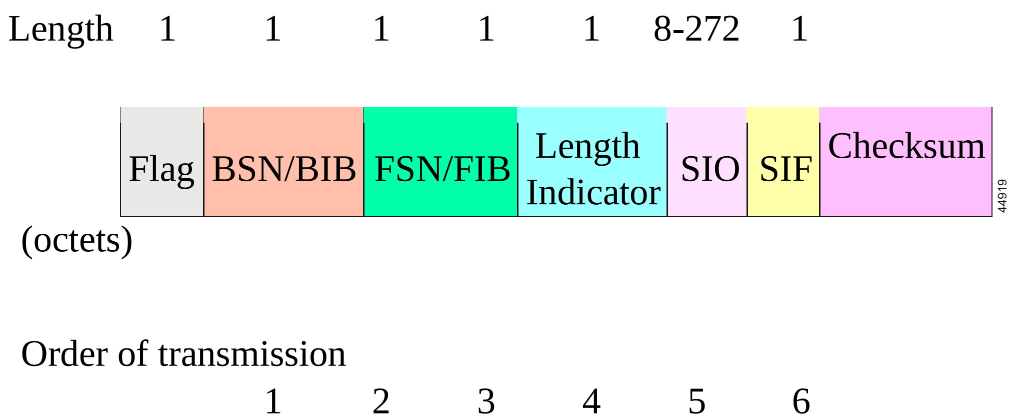
- The check bit field and the sequence number of the signal unit are used to detect errors. Seven-bit sequence numbering is used.
- The forward sequence number (FSN) is incremented by one after every transmission.
- The backward sequence number (BSN) is used to acknowledge received signal units.

Types of Signal Units

a) Message Signal Units

- MSUs are the workhorses of the SS7 network.
- All signaling associated with call setup and teardown, database query and response, and SS7 management requires the use of MSUs.
- MSUs provide MTP protocol fields, service indicator octet (SIO) and service information field (SIF).
- The SIO identifies the type of protocol (ISUP, TCAP) and standard (ITU-TS, ANSI).
- The SIF transfers control information and routing label.

Figure4-2 MSU Format



SIO Structure

- The functionality of the MSU lies in the contents of the service indicator octet (SIO) and the service information fields (SIF).
- The SIO is an 8-bit field that contains three types of information:
- Four bits to indicate the type of information contained in the service information field (referred to as the service indicator). (Refer to Table 4-1.)
- Two bits to indicate whether the message is for use in a national or international network.
- Two bits to identify the message priority. Not used to control the order of transmittal, but used when network is congested to determine if a message can be discarded. Value is from 0–3, with 3 the highest priority.

Table4-1SIO Service Indicator Bits

Value	Function
0	Signaling Network Management
1	Signaling Network Testing and Maintenance
2	Signaling Connection Control Part (SCCP)
3	ISDN User Part (ISUP)

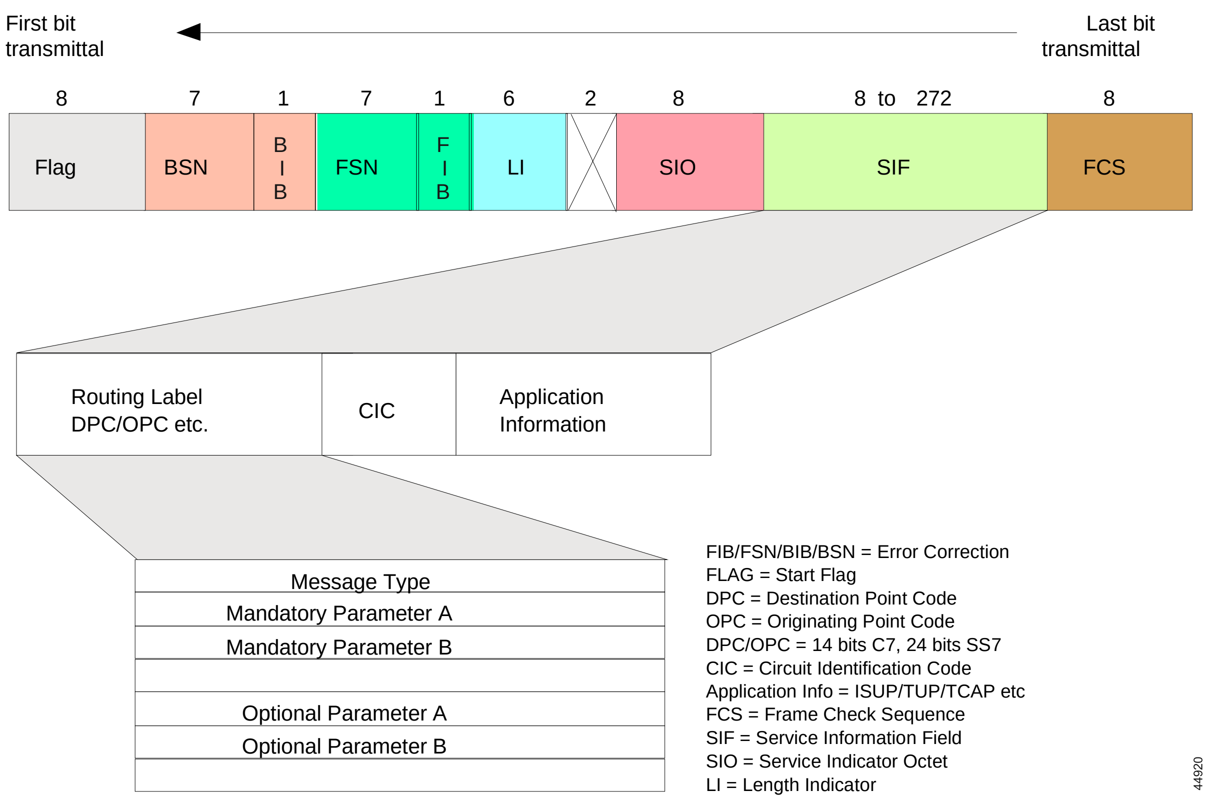
SIF Structure

- The service information field (SIF) provides the first piece of information necessary for routing and decoding the message.
- The SIF transfers control information and the routing label used by Level 3.
- The routing label consists of the destination point code (DPC), originating point code (OPC) and signaling link selection (SLS) fields.

Note An ANSI point code consists of network, cluster and member octets (245-16-0). ANSI routing label uses 7 octets; ITU-T routing label uses 4 octets.

- The SIF can contain up to 272 octets and is used by network management, ISUP, TCAP and MAP. (See Figure 4-3.)

Figure4-3 MSU SIF Structure

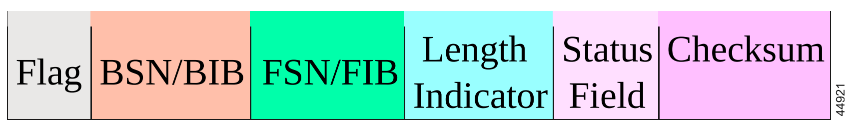


b) Link Status Signal Unit

- LSSUs communicate information about the signaling link between the nodes on either end of the link.
- This information is contained in the status field of the signal unit. (See Figure 4-4.)
- They signal the initiation of link alignment, quality of received traffic, and status of processors at either end of the link.
- LSSUs do not require any addressing information because they are only sent between signaling points.

Figure4-4 LSSU Format

Length 1 1 1 1 1 or 2 1



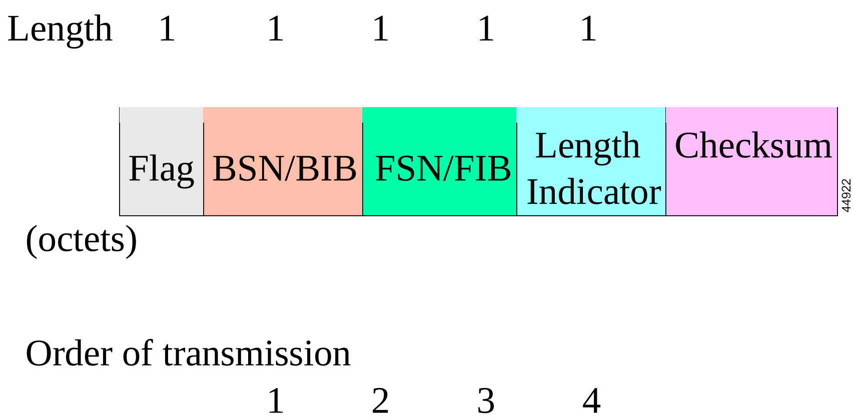
(octets)

Order of transmission
1 2 3 4 5

c) Fill-in Signal Unit

- FISUs do not carry any information; they simply occupy the link when there are no LSSUs or MSUs.
- FISUs support the monitoring of link traffic because they undergo error checking.
- They can also be used to acknowledge the receipt of messages using backward sequence number (BSN) and backward indicator bit (BIB). (See Figure 4-5.)

Figure4-5 FISU Format



Compare the architecture of SS7 with seven-layer OSI architecture

The relationship among these levels and the layers of the OSI model is demonstrated in figure. The user part includes layers 4 to 7 of the OSI model.

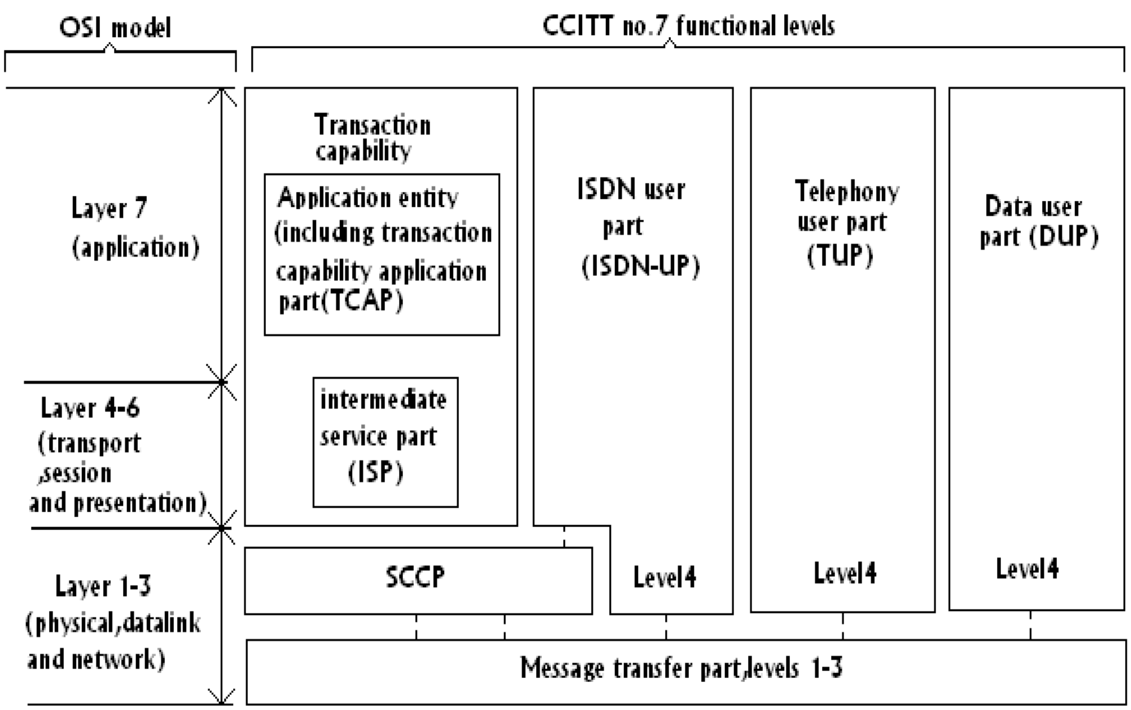


FIG - Relationship between CCITT No.7 Functional levels and OSI layers

- Level 1 is implying that of sending bit streams over a physical path. This uses times slot 24 of a 1.5 M bit/s system or times lot 16 of a 2 M bit/s PCM system.
- Level 2 performs the functions of link initialization, error control, error rate monitoring, delineation of messages and flow control.
- Level 3 gives the functions needed for a signaling network. All nodes in the network have a single point odd that is a 14-bit address. All messages contain point codes of the originating and terminating nodes for such messages.
- Levels 1 to 3 by the message transfer part (MTP) of CCITT number 7.
- Level 4 is the user part. It consists of the processes for handling the service being supported through the signaling system. The message transfer part is able of supporting various different user parts. Therefore, as far, three have been explained: the telephone user part (TUE), the (ISDN) user part (ISDN-UP) and the data user part (DUP).

DTMF

- The abbreviation DTMF stands for Dual Tone Multiple-Frequency, also called Touch Tone.

- DTMF signaling was developed to signal the destination telephone number of calls without requiring a telephone operator.

- It was standardized by the International Telecommunication Union (ITU) Telecommunication

- DTMF tones are also used by cable television broadcasters to indicate the start and stop times of commercial insertion points during station breaks for cable company benefit.

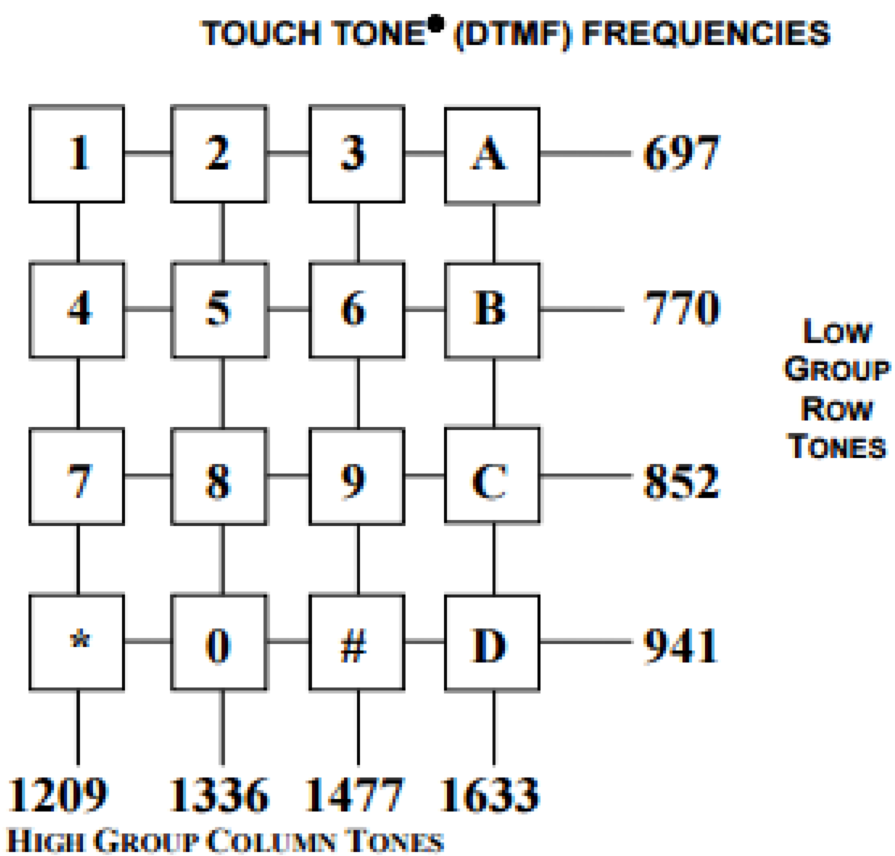
- The frequencies used prevent harmonics from being incorrectly detected by receivers as other DTMF frequencies.

- DTMF keypads are laid out on a 4x4 matrix, in which each row represents low frequency and each column represents high frequency.

- With DTMF, each key pressed on a phone generates two tones of specific frequencies.

- One tone is generated from a high-frequency group of tones, while the other is from a low-frequency group.

- DTMF systems use eight different frequency signals transmitted in pairs to represent 16 different numbers, letters and symbols.



- Dual-tone multi-frequency (DTMF) signaling is a method used to transmit information over telephone lines.
- It is used to dial telephone numbers and to send commands to voicemail systems and other automated systems.

- DTMF signals are sent by pressing the keys on a telephone keypad.
- Each key on the keypad is assigned a specific frequency, and when a key is pressed, two frequencies are sent over the line.
- The receiving system is able to detect the frequencies and determine which key was pressed based on the combination of frequencies received.

There are several advantages to using DTMF signaling:

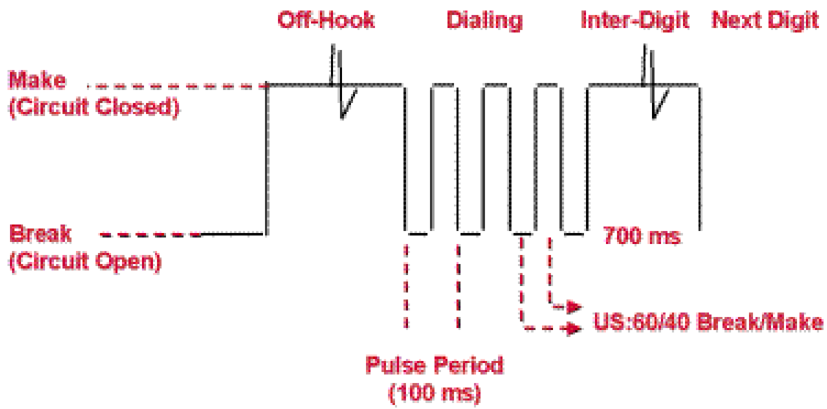
1. It is easy to use: Most people are familiar with using a telephone keypad, so it is easy for them to use DTMF signaling.
2. It is reliable: DTMF signals are relatively easy for the receiving system to detect, so there is a low error rate.
3. It can transmit information over long distances: DTMF signals are able to travel long distances over telephone lines without degrading in quality.
4. It can be used with a variety of systems: DTMF signaling can be used with a wide range of systems, including voicemail systems, automated attendant systems, and interactive voice response (IVR) systems.
5. It is secure: Because DTMF signals are transmitted over telephone lines, they are less vulnerable to interception or tampering than other forms of signaling.

There are also some disadvantages to using DTMF signaling:

1. Limited information capacity: DTMF signaling can only transmit a limited amount of information at a time, as it is limited to the 16 keys on a telephone keypad.
2. Limited accuracy: DTMF signaling is not as accurate as some other methods of transmitting information, as it is susceptible to noise and interference on the telephone line.
3. Not suitable for high-speed data transmission: DTMF signaling is not well-suited for transmitting large amounts of data at high speeds.
4. Not widely used for newer technologies: DTMF signaling is not as commonly used with newer technologies, such as internet-based communication systems.
5. May not work with some older phone systems: Some older phone systems may not be able to detect DTMF signals, which can cause problems if you are trying to use DTMF signaling with these systems.

Pulse Dialing:

- Pulse dialing is a method of signaling to a telephone exchange to indicate the desired telephone number to be connected.
- It was the most widely used form of dialing before the advent of tone dialing.
- In pulse dialing, the phone sends a series of electrical pulses, or "clicks," to the exchange to indicate each digit of the telephone number.
- The exchange then uses these pulses to route the call to the desired number.
- To make a call using pulse dialing, the caller would pick up the handset of the phone and listen for a dial tone.
- Once the dial tone was heard, the caller would then press the keys on the phone's Dialpad to enter the desired telephone number.
- Each time a key was pressed, the phone would generate a pulse, or "click," which would be transmitted to the exchange.
- Pulse dialing is no longer commonly used, as it has been largely replaced by tone dialing, which uses a series of tones to signal the desired telephone number to the exchange.



What are the distinctive signaling tones provided in all automatic systems?

There are many different kinds of signaling tones that are used in automatic systems, depending on the specific application and the needs of the system. Some common types of signaling tones include:

1. **Dial tones:** These are the tones that you hear when you pick up a phone and are used to indicate that the phone is ready to make a call.
2. **Busy tones:** These are the tones that you hear when you try to call a phone that is already in use.
3. **Ringback tones:** These are the tones that you hear when you make a call and the phone is ringing on the other end.
4. **Reorder tones:** These are the tones that you hear when there is a problem with the phone line or the call cannot be completed.
5. **Call waiting tones:** These are the tones that you hear when you are on a call and another call comes in.
6. **Confirmation tones:** These are tones that are used to confirm that a particular action or request has been completed or accepted.
7. **Warning tones:** These are tones that are used to alert the user to a potential problem or danger.
8. **Error tones:** These are tones that are used to indicate that an error has occurred or that a request cannot be completed.
9. **Attention tones:** These are tones that are used to get the attention of the user or to indicate that a message is coming.

10. In paging systems, signaling tones are used to alert the user to an incoming message or page.
11. In alarm systems, signaling tones are used to alert the user to a potential danger or emergency.
12. In transportation systems, signaling tones are used to indicate the status of a train or bus, such as when it is arriving or departing.
13. In aviation, signaling tones are used to indicate the status of a flight or to provide information to pilots and air traffic controllers.
14. In military communications, signaling tones are used to indicate the status of a transmission and to ensure the secure transmission of sensitive information.
15. In emergency response systems, signaling tones are used to alert first responders to an emergency and to provide them with important information about the situation.

Numbering plan

- A numbering plan is a system for assigning telephone numbers to subscribers and routing telephone calls in a telephone network.

- It is a set of rules that dictate how telephone numbers are organized and used.

- There are many different numbering plans in use around the world, and each country generally has its own numbering plan.

- The numbering plan for a particular country or region is typically administered by the national or regional telecommunications regulatory agency.

- Numbering plans usually specify the format and length of telephone numbers, as well as the structure of the numbering system.

- They may also specify how telephone numbers are assigned to different types of service, such as landline, mobile, and VoIP, and how numbers are used for different types of calls, such as local, long distance, and international.

Types of numbering plan

1. **Closed numbering plan:** In a closed numbering plan, the range of available telephone numbers is limited and pre-allocated to a specific geographic area or service provider. This means that once all of the available numbers have been assigned, no new numbers can be added to the system without significant changes to the numbering plan.
2. **Open numbering plan:** In an open numbering plan, there is no limit to the number of telephone numbers that can be assigned. New numbers can be added to the system as needed, either by adding additional digits to the existing numbering format or by creating a new range of numbers.
3. **Geographical numbering plan:** In a geographical numbering plan, telephone numbers are assigned based on the location of the subscriber. The first few digits of the telephone number typically correspond to the area code, which identifies the geographic region where the number is located.

4. **Non-geographical numbering plan**: In a non-geographical numbering plan, telephone numbers are not tied to a specific location. These numbers may be used for various types of services, such as freephone, premium rate, or virtual numbers, and may be used in multiple locations around the world.
5. **E.164 numbering plan**: The E.164 numbering plan is a standardized numbering plan developed by the International Telecommunication Union (ITU). It is used in many countries around the world and specifies a format for telephone numbers that includes a country code, area code, and local number.

6. **A semi-open numbering plan** is a hybrid between a closed and an open numbering plan. In a semi-open numbering plan, a limited range of telephone numbers is pre-allocated to a specific geographic area or service provider. However, unlike a closed numbering plan, there is a mechanism in place to allow for the addition of new numbers to the system as needed.

7. **A semi-close numbering plan** is similar to a semi-open numbering plan, but it has additional restrictions on the assignment of new telephone numbers. In a semi-close numbering plan, the addition of new numbers may be limited to certain circumstances, such as when there is a significant increase in demand for telephone service in a particular area.

- It is worth noting that the terms "semi-open" and "semi-close" are not formally defined in the telecommunications industry, and their usage may vary.

- In general, a numbering plan is considered either open or closed, with no intermediate categories.

- However, some countries or regions may use the terms "semi-open" and "semi-close" informally to describe their numbering plans.

HOW numbering plan is achieved in modern Telephony System?

- In modern telephony systems, numbering plans are typically administered by the national or regional telecommunications regulatory agency.

- The regulatory agency is responsible for managing the allocation and assignment of telephone numbers, as well as enforcing the rules of the numbering plan.

To achieve a numbering plan in a modern telephony system, the following steps are typically followed:

1. **Determine the structure and format of the telephone numbers:** The numbering plan specifies the structure and format of the telephone numbers that will be used in the system. This includes the number of digits in the telephone number, the arrangement of the digits, and the meaning of each digit or group of digits.
2. **Allocate blocks of telephone numbers:** The regulatory agency allocates blocks of telephone numbers to service providers or geographic areas as needed. The size of the block and the specific numbers within it are determined by the numbering plan.
3. **Assign telephone numbers to subscribers**: Service providers or other organizations assign telephone numbers to their subscribers according to the rules of the numbering plan. This may involve assigning numbers in sequence from the allocated block, or using a more complex algorithm to determine the specific number to be assigned.
4. **Set up routing rules:** The regulatory agency or the service providers set up routing rules to determine how telephone calls are routed through the network based on the destination telephone number. These rules may be based on the structure and format of the telephone numbers, as well as on the location or type of service associated with the numbers.
5. **Update and maintain the numbering plan:** The numbering plan is typically reviewed and updated on a regular basis to ensure that it is still effective and efficient. This may involve making changes to the structure and format of the telephone numbers, adjusting the allocation and assignment of numbers, or modifying the routing rules.

What is charging plan?

- A charging plan is a set of rules and rates that determine how a customer is charged for the goods or services that they use.
- In the context of telecommunications, a charging plan might specify the rates for making phone calls, sending text messages, or using data services.
- Charging plans can be based on a variety of factors, such as the duration of the call, the time of day, the location of the call, and the type of service being used.

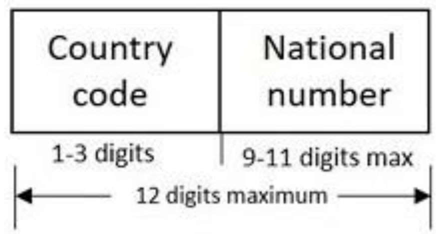
- Some charging plans offer a fixed rate for a certain number of minutes or a certain amount of data, while others charge based on usage beyond a certain threshold.
- Charging plans can be offered by a variety of different types of businesses, including phone companies, internet service providers, and utility companies.

Types of charging plan:

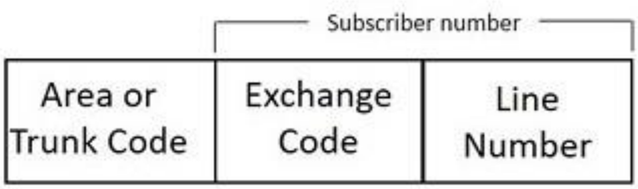
1. **Flat rate plans**: These charging plans offer a fixed rate for a certain number of minutes or a certain amount of data. For example, a phone company might offer a flat rate plan that includes unlimited calls and texts for a fixed monthly fee.
2. **Pay-as-you-go plans:** These charging plans allow customers to pay for the goods or services they use on an as-needed basis. For example, a customer might purchase a certain amount of credit that they can use to make phone calls or send texts, and their balance will be reduced each time they use the service.
3. **Tiered plans:** These charging plans offer different rates or allowances depending on the usage level of the customer. For example, a phone company might offer a tiered plan with different rates for different levels of data usage.
4. **Usage-based plans:** These charging plans charge customers based on their actual usage of the goods or services. For example, a phone company might charge customers based on the number of minutes they use or the amount of data they consume.
5. **Bundle plans:** These charging plans offer a package of goods or services at a discounted rate. For example, a phone company might offer a bundle plan

that includes a certain amount of minutes, texts, and data for a single monthly fee.

6. **Subscription plans:** These charging plans offer a set of goods or services on a recurring basis, usually for a fixed fee. For example, a customer might subscribe to a streaming service and pay a monthly fee to access a certain number of movies or TV shows.



International telephone number



National Telephone Number

How the charging plan has been achieved in the modern telephony system?

- In modern telephony systems, charging plans are typically implemented using software that tracks the usage of the phone or other communication device and calculates the charges based on the applicable rate plan.

- This usage data is often stored in a database, which can be accessed by the billing system to generate invoices or statements for the customer.

- When a customer makes a call, sends a text message, or uses data on their phone, the telephony system records the duration of the call, the number or address of the recipient, and the type of service being used.

- This usage data is then used to calculate the charges based on the rate plan that the customer is subscribed to.
- Some modern telephony systems also allow customers to track their usage and charges in real-time using online portals or mobile apps.
- This can help customers to better understand their usage patterns and make informed decisions about their charging plan.

Derivation of 2 stage and 3 stage networks in the book page number 129-143

Chapter 7: Synchronization and Network Management

Synchronization and its principle:

- In telecommunications, synchronization is the process of coordinating the timing of the transmission of signals between two or more devices in order to ensure that the information being transmitted is received correctly.

There are several principles that are important in synchronization:

- **Time base:** A common time base is necessary to ensure that all devices are using the same timing reference.
- **Frequency accuracy:** The transmission frequency must be accurately maintained to ensure that the signal is received correctly.
- **Phase accuracy:** The phase of the transmitted signal must be accurately maintained to ensure that the signal is received correctly.
- **Jitter:** Jitter is the variation in the timing of the signal, and it can cause errors if it is too large.
- **Drift:** Drift is the slow change in the timing of the signal over time, and it can also cause errors if it is not compensated for.

Modes of Operations:

- **Free-running mode:**
  - In this mode, each device operates independently and is not synchronized to any external reference.
  - This is the simplest mode of operation, but it can result in significant errors if the devices are not accurately time-aligned.
- **Self-synchronizing mode:**
  - In this mode, each device is able to synchronize itself to an external reference, such as a clock signal transmitted over a network.
  - This can be a more accurate method of synchronization, but it requires additional hardware and complexity.
- **Synchronous mode:**
  - In this mode, all devices are continuously synchronized to a common reference, such as a central clock.

    - This is the most accurate method of synchronization, but it requires a dedicated reference source and a means of distributing the reference signal to all devices.
- **Asynchronous mode:**
    - In this mode, devices are not continuously synchronized to a common reference, but they can exchange messages and synchronize their timing as needed.
    - This is a more flexible method of synchronization, but it can be less accurate if the devices are not able to exchange messages frequently enough.

Synchronizer Circuit:

- A synchronizer circuit is a device that is used to synchronize the timing of two or more signals.
- The primary function of a synchronizer is to ensure that the timing of the signals is aligned correctly, so that the information being transmitted is received correctly.

There are several types of synchronizer circuits, including:

- **Phase-locked loops (PLLs):** PLLs are used to synchronize the phase of a signal to a reference signal. They are commonly used to lock the frequency of a signal to a reference frequency.
- **Digital sync circuits:** Digital sync circuits are used to synchronize the timing of digital signals. They are typically used to ensure that the timing of data bits is aligned correctly, so that the data can be correctly interpreted.
- **Clock and data recovery (CDR) circuits:** CDR circuits are used to recover the clock signal from a data signal. They are commonly used in high-speed communications systems to recover the clock signal from a data stream.
- **Timing recovery circuits:** Timing recovery circuits are used to recover the timing of a signal that has been distorted or degraded. They are commonly used in systems that transmit signals over long distances or through noisy channels.

PLL:

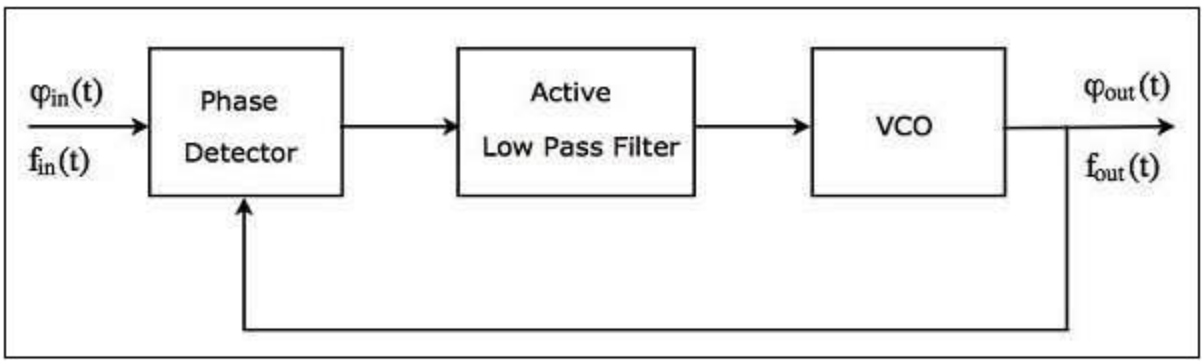
- Phase Locked Loop **(PLL)** is one of the vital blocks in linear systems.
- It is useful in communication systems such as radars, satellites, FMs, etc.

Block Diagram of PLL

A Phase Locked Loop (PLL) mainly consists of the following **three blocks** −

- Phase Detector
- Active Low Pass Filter
- Voltage Controlled Oscillator (VCO)

The **block diagram** of PLL is shown in the following figure −



The output of a phase detector is applied as an input of active low pass filter. Similarly, the output of active low pass filter is applied as an input of VCO.

The **working** of a PLL is as follows −

i) Phase Detector:

- **Phase detector** produces a DC voltage, which is proportional to the phase difference between the input signal having frequency of *fin* and feedback (output) signal having frequency of *fout.*
- A **Phase detector** is a multiplier and it produces two frequency components at its output − sum of the frequencies fin and fout and difference of frequencies fin & fout.

ii) Active low pass filter:

- An **active low pass filter** produces a DC voltage at its output, after eliminating high frequency component present in the output of the phase detector.
- It also amplifies the signal.

iii) VCO:

- A **VCO** produces a signal having a certain frequency, when there is no input applied to it.
- This frequency can be shifted to either side by applying a DC voltage to it.
- Therefore, the frequency deviation is directly proportional to the DC voltage present at the output of a low pass filter.
- The above operations take place until the VCO frequency equals to the input signal frequency.
- Based on the type of application, we can use either the output of active low pass filter or output of a VCO.

PLLs are used in many **applications** such as FM demodulator, clock generator etc.

PLL operates in one of the **following three modes** −

- Free running mode
- Capture mode
- Lock mode

Initially, PLL operates in **free running mode** when no input is applied to it. When an input signal having some frequency is applied to PLL, then the output signal frequency of VCO will start change. At this stage, the PLL is said to be operating in the **capture mode**. The output signal frequency of VCO will change continuously until it is equal to the input signal frequency. Now, it is said to be PLL is operating in the **lock mode**.

Sampling time recovery and the frame time recovery:

- Sampling time recovery and frame time recovery are both techniques that are used to recover the timing of a signal that has been distorted or degraded.

- Sampling time recovery is a technique that is used to recover the timing of a signal that is being sampled.

- It is commonly used in systems that use analog-to-digital converters (ADCs) to convert analog signals into digital form.

- Frame time recovery is a technique that is used to recover the timing of a signal that is transmitted in frames.
- It is commonly used in systems that transmit data in frames, such as Ethernet or ATM.
- In both cases, the goal is to recover the original timing of the signal, so that the information being transmitted can be correctly interpreted.
- This may involve techniques such as phase-locked loops (PLLs) or clock and data recovery (CDR) circuits.

Sampling Time Recovery vs Frame Time Recovery:

	Sampling Time Recovery	Frame Time Recovery
Purpose	To recover the timing of a signal that is being sampled	To recover the timing of a signal that is transmitted in frames
Example systems	Analog-to-digital converters (ADCs)	Ethernet, ATM
Techniques used	Phase-locked loops (PLLs), clock and data recovery (CDR) circuits	PLLs, CDR circuits

Signal type	Continuous analog signal	Discrete digital frames
Timing reference	Internal clock	External reference signal
Timing accuracy	Depends on the accuracy of the internal clock and the stability of the analog signal	Depends on the accuracy of the external reference signal and the stability of the frame structure
Complexity	Typically simpler than frame time recovery	Typically more complex than sampling time recovery
Applications	Analog-to-digital conversion, signal processing	Data communication, networking

Timing Inaccuracies:

- Timing inaccuracies can occur in any system that relies on synchronization, and they can have a variety of causes.
- Some common sources of timing inaccuracies include:

1. **Drift:** Drift is the slow change in the timing of a signal over time, and it can be caused by temperature changes, aging of components, or other factors.
2. **Jitter**: Jitter is the variation in the timing of a signal, and it can be caused by noise, interference, or other external factors.
3. **Frequency errors:** If the transmission frequency is not accurately maintained, it can cause timing errors in the received signal.
4. **Phase errors**: If the phase of the transmitted signal is not accurately maintained, it can cause timing errors in the received signal.
5. **Skew:** Skew is the difference in the timing of two or more signals, and it can be caused by differences in the transmission paths or differences in the timing of the transmitting devices.

- Timing inaccuracies can cause errors in the transmitted information and can degrade the performance of a system.

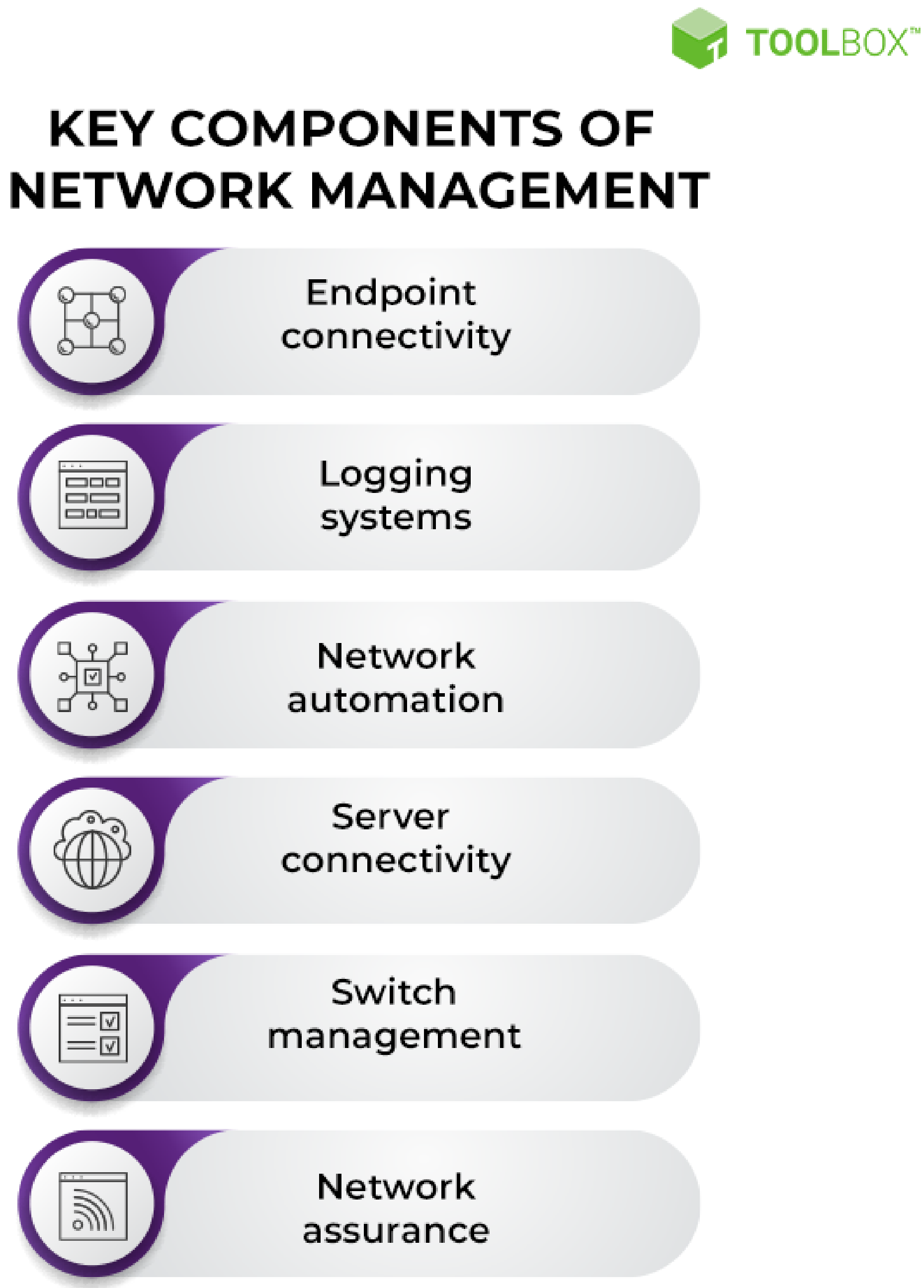
- They can be minimized through the use of synchronization techniques and devices, such as phase-locked loops (PLLs) and clock and data recovery (CDR) circuits.

Ways to Reduce the timing accuracies:

1. **Use a stable reference signal:** A stable reference signal, such as a high-quality clock signal or a GPS signal, can help to minimize drift and other long-term timing errors.
2. **Use synchronization devices**: Devices such as phase-locked loops (PLLs) and clock and data recovery (CDR) circuits can be used to synchronize the timing of signals and reduce jitter and other short-term timing errors.
3. **Use error correction techniques:** Error correction techniques, such as forward error correction (FEC) and interleaving, can help to mitigate the effects of timing errors on the transmitted information.
4. **Use high-quality components:** Using high-quality components, such as low-jitter clocks and stable amplifiers, can help to reduce timing inaccuracies caused by component-level errors.
5. **Monitor and maintain the system:** Regular monitoring and maintenance of the system can help to identify and correct any issues that may be causing timing inaccuracies.

Define network management.

- Network management is the procedure of administering, managing and working a data network using a network management system.



AIM of Network management:

Objectives of Network Management

- Main objective of network management is:
  - User Satisfaction.
    - Performance of the network.
      - Hint: Response Time, Imagine a bank teller waiting, Emergency situations.
    - Availability & Reliability of the network.
      - Hint: Minimize impact during maintenance, Backup components / AMC.
    - Information availability.
      - Hint: Open communication with users, Schedule of preventive maintenance.
    - Network uptime.
      - Hint: Patch around the Problem, Replacement, Repair
    - Backup facilities.
      - Hint: Software Backup (RAID), Hardware Backup (PGVCL).
    - Cost effectiveness.
      - Hint: Prior Planning for Expansion in future.

Routing Control:

- Routing control refers to the process of determining the path that data should take through a network to reach its destination.
- This is typically done using routing protocols, which are algorithms that determine the best path based on various factors such as cost, distance, and network congestion.

Routing Metrics:

- Hops
- Bandwidth
- Load
- Cost
- Reliability

Routing in Network Management:

- Routing is an important aspect of network management, as it determines the path that data takes through the network to reach its destination.
- Routing is used in network management to:

1. **Ensure the efficient and reliable transmission of data:** By choosing the best path through the network based on various factors such as cost, distance, and network congestion, routing can help to ensure that data is transmitted efficiently and reliably.
2. **Make the best use of network resources:** By selecting the most efficient path through the network, routing can help to minimize the use of network resources such as bandwidth and processing power.
3. **Adapt to changes in the network:** Dynamic routing protocols allow the network to adapt to changes automatically, such as the failure of a link or the addition of a new device. This helps to maintain the performance of the network and reduce downtime.
4. **Provide security:** Routing can be used to implement security measures such as access control lists (ACLs), which can help to prevent unauthorized access to the network.

- Overall, routing is a key component of network management and is essential for ensuring the smooth operation of the network.

Types of routing:

a) Static Routing

- Static Routing is also known as Nonadaptive Routing.
- It is a technique in which the administrator manually adds the routes in a routing table.
- A Router can send the packets for the destination along the route defined by the administrator.
- In this technique, routing decisions are not made based on the condition or topology of the networks

Advantages Of Static Routing

Following are the advantages of Static Routing:

- **No Overhead:** It has ho overhead on the CPU usage of the router. Therefore, the cheaper router can be used to obtain static routing.
- **Bandwidth:** It has not bandwidth usage between the routers.
- **Security:** It provides security as the system administrator is allowed only to have control over the routing to a particular network.

Disadvantages of Static Routing:

Following are the disadvantages of Static Routing:

- For a large network, it becomes a very difficult task to add each route manually to the routing table.
- The system administrator should have a good knowledge of a topology as he has to add each route manually.

b) Default Routing

- Default Routing is a technique in which a router is configured to send all the packets to the same hop device, and it doesn't matter whether it belongs to a particular network or not. A Packet is transmitted to the device for which it is configured in default routing.
- Default Routing is used when networks deal with the single exit point.
- It is also useful when the bulk of transmission networks have to transmit the data to the same hp device.
- When a specific route is mentioned in the routing table, the router will choose the specific route rather than the default route. The default route is chosen only when a specific route is not mentioned in the routing table.

c) Dynamic Routing

- It is also known as Adaptive Routing.
- It is a technique in which a router adds a new route in the routing table for each packet in response to the changes in the condition or topology of the network.
- Dynamic protocols are used to discover the new routes to reach the destination.
- In Dynamic Routing, RIP and OSPF are the protocols used to discover the new routes.
- If any route goes down, then the automatic adjustment will be made to reach the destination.

The Dynamic protocol should have the following features:

- All the routers must have the same dynamic routing protocol in order to exchange the routes.
- If the router discovers any change in the condition or topology, then router broadcast this information to all other routers.

Advantages of Dynamic Routing:

- It is easier to configure.
- It is more effective in selecting the best route in response to the changes in the condition or topology.

Disadvantages of Dynamic Routing:

- It is more expensive in terms of CPU and bandwidth usage.
- It is less secure as compared to default and static routing.

Flow control:

- Flow control is a mechanism that is used to regulate the flow of data in telecommunications systems, to prevent overloading of the communication resources.
- It is an important aspect of telecommunications, as it helps to ensure the reliable transmission of data and to make the best use of the available communication resources.

There are several different types of flow control that are used in telecommunications, including:

Window-based flow control:

- In window-based flow control, the sender is allowed to transmit only a certain amount of data before waiting for an acknowledgement from the receiver.
- The size of the window determines the amount of data that can be transmitted before an acknowledgement is required.

Credit-based flow control:

- In credit-based flow control, the sender is granted a certain amount of credit, which allows it to transmit data without waiting for an acknowledgement.
- The receiver sends credits to the sender as it is able to process the data, and the sender can transmit data as long as it has credits available.

Rate-based flow control:

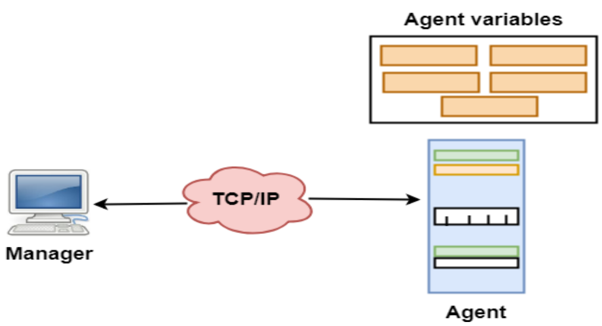
- In rate-based flow control, the sender is required to transmit data at a certain rate, and it is not allowed to exceed this rate.
- This can be used to prevent overloading of the communication resources or to prioritize certain types of traffic.

Flow control is typically implemented at the transport layer of the communication protocol stack, and it can be used in combination with other techniques such as error correction and congestion control to manage the flow of data in the network.

SNMP

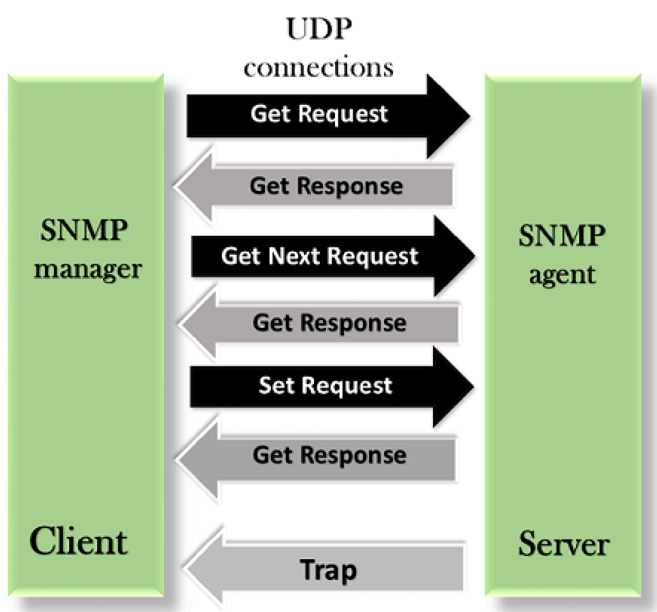
- SNMP stands for **Simple Network Management Protocol**.
- SNMP is a framework used for managing devices on the internet.
- It provides a set of operations for monitoring and managing the internet.

SNMP Concept



- SNMP has two components Manager and agent.
- The manager is a host that controls and monitors a set of agents such as routers.

SNMP defines five types of messages: GetRequest, GetNextRequest, SetRequest, GetResponse, and Trap.



GetRequest:

- The GetRequest message is sent from a manager (client) to the agent (server) to retrieve the value of a variable.

GetNextRequest:

- The GetNextRequest message is sent from the manager to agent to retrieve the value of a variable.
- This type of message is used to retrieve the values of the entries in a table.
- If the manager does not know the indexes of the entries, then it will not be able to retrieve the values.
- In such situations, GetNextRequest message is used to define an object.

GetResponse:

- The GetResponse message is sent from an agent to the manager in response to the GetRequest and GetNextRequest message.
- This message contains the value of a variable requested by the manager.

SetRequest:

- The SetRequest message is sent from a manager to the agent to set a value in a variable.

Trap:

- The Trap message is sent from an agent to the manager to report an event.
- For example, if the agent is rebooted, then it informs the manager as well as sends the time of rebooting.

What are the main sources of clock instability in a Network?

- There are several sources of clock instability in a network, which can affect the accuracy of the timing of the signals transmitted over the network.
- Some common sources of clock instability include:

Temperature changes: Temperature changes can cause the frequency of a clock signal to drift, as the components of the clock oscillator expand and contract with temperature changes.

Aging of components: As components age, their performance can degrade, which can cause the frequency of the clock signal to drift.

Power supply fluctuations: Fluctuations in the power supply can cause the frequency of the clock signal to vary, as the oscillator may not be able to maintain a stable frequency under changing power conditions.

Noise: Noise from external sources, such as electromagnetic interference (EMI) or radio frequency interference (RFI), can cause the clock signal to become unstable.

Jitter: Jitter is the variation in the timing of a signal, and it can be caused by a variety of factors such as noise, interference, or data traffic patterns.

- **Overall, clock instability can cause errors in the transmitted information and can degrade the performance of a network.**
- **It can be minimized through the use of high-quality clock sources and synchronization techniques such as phase-locked loops (PLLs).**

Elastic Store:

- The elastic store inside of Maxim T1, E1, and J1 devices serves as a dual port buffer between the line side and the system side of the device.
- It allows the two sides to operate in different clock domains or even at different frequencies.

Elastic Store Operation

The elastic store is a dual port buffer that has a depth of 512 bits. Since the bit length of a frame varies between T1 and E1, the amount of buffer bits that are used depends on the mode of operation. There are four basic modes of operation:

- T1 Mode: 193-bit frame
- E1 Mode: 256-bit frame
- T1 to E1 Rate Conversion Mode: 193-bit frames on the line (network) side and 256-bit frames on the system (backplane) side
- Interleave Bus Operation Mode: 193-bit or 256-bit frames on the line (network) side and 256-bit frames on the system (backplane) side with a high-speed gapped system clock.

Here, each example was based on the receive elastic store in E1 mode with 256-bit frames. write pointer indicates the white dots and read pointers indicated the black dots.

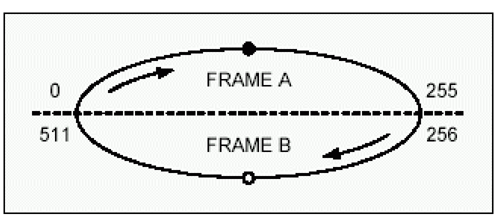


Figure 1. Elastic store read and write pointers are exactly one frame apart.

Figure 1 illustrates the condition where the write pointer and read pointer are "ideally" centered at exactly one frame apart.

In Figure 2, the read pointer (black dot) is about to enter frame B. When either pointer crosses a frame boundary, the distance between the two pointers is compared in the forward direction. Any distance below a set threshold will cause a frame slip and the pointer that just crossed the frame boundary is moved to the beginning of the next frame. Depending on which pointer slipped, there will either be a repeated or deleted frame. The threshold depends on the mode of operation: 16 bits in E1 mode and 9 bits for all other modes. In the example below, the write pointer (white dot) is almost a frame away thus no slip occurs and the read pointer will continue into frame B.

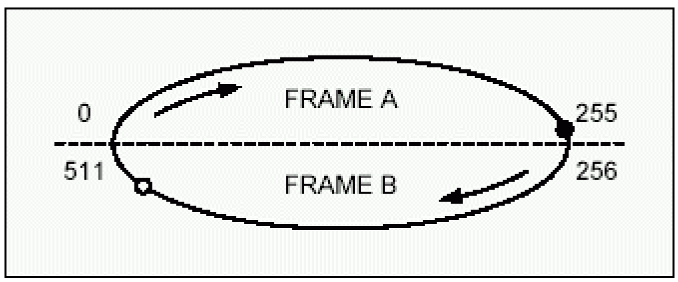


Figure 2. Read and write pointers are far enough apart that no slip occurs after a compare.

In Figure 3, the read pointer (black dot) is running faster than the write pointer (white dot) and will eventually catch up. As the read pointer crosses the boundary into frame A, it detects that the write pointer is within 16 bits of the start of frame A. Rather than enter frame A, the read pointer slips and returns to the start of frame B at bit position 256. Because the read pointer slipped, the last frame read from the buffer is repeated and a receive elastic store empty event is reported.

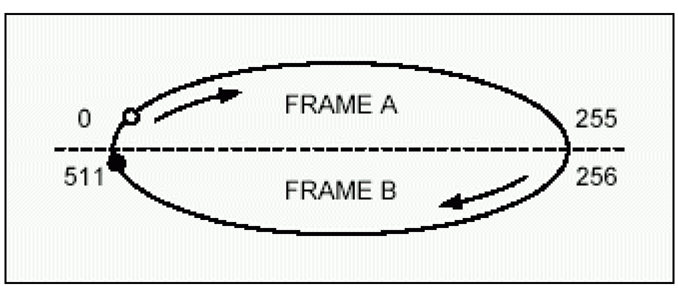


Figure 3. The read pointer is too close to the write pointer during a compare and cause a slip.

In figure 4, The same is true when the write pointer is running faster than the read pointer. If the write pointer detects that the read pointer is within 16 bits of the start of frame B, the write pointer slips and returns to the start of frame A at bit position 0. Since the write pointer slipped, the last frame written to the buffer is deleted and a receive elastic store full event is reported. An example of the write pointer causing the slip is shown in Figure 4.

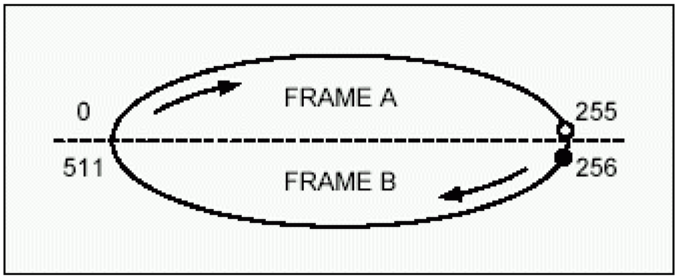


Figure 4. The write pointer is too close to the read pointer during a compare and causes a slip.

Elastic store in time synchronization:

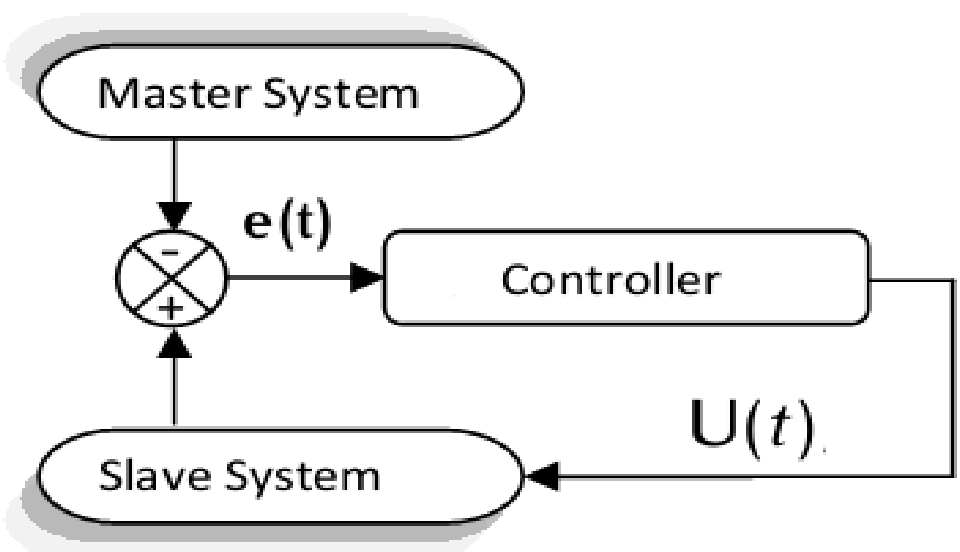
- Elastic Store is a plugin for Elasticsearch that allows you to store and retrieve data from Elasticsearch using a simple key-value interface.
- It does not have any specific functionality related to time synchronization.
- If you are looking to synchronize the time on your Elasticsearch nodes, you can use the NTP (Network Time Protocol) to ensure that the clocks on all of your nodes are accurate.
- You can also use the Elasticsearch API to retrieve the current time on the nodes and use this information to synchronize the clocks.
- Alternatively, you can use a tool like Elasticsearch's own `clock skew` monitoring feature to detect and alert you if there are any significant differences in the clock times between your nodes.

Master-Slave Synchronization:

- In a master-slave synchronization system, the master device sends commands or signals to one or more slave devices.
- The slave devices receive the commands and respond by carrying out the required actions.
- The master device may also receive feedback from the slave devices, which it can use to adjust its own actions or to send further commands to the slaves.

Here is a simple example of a master-slave synchronization system:

1. The master device sends a signal indicating the current time to the slave devices.
2. The slave devices adjust their internal clocks to match the time indicated by the master.
3. The slave devices carry out their assigned tasks according to the synchronized time.
4. The slave devices may send feedback to the master device indicating the status of their tasks.
5. The master device may use this feedback to adjust its own actions or to send further commands to the slave devices.



Why synchronization is important? [4 marks]

- Synchronization is important in telecommunication systems because it ensures that all devices are operating in a coordinated and consistent manner. It helps to improve the reliability and performance of the system as a whole.

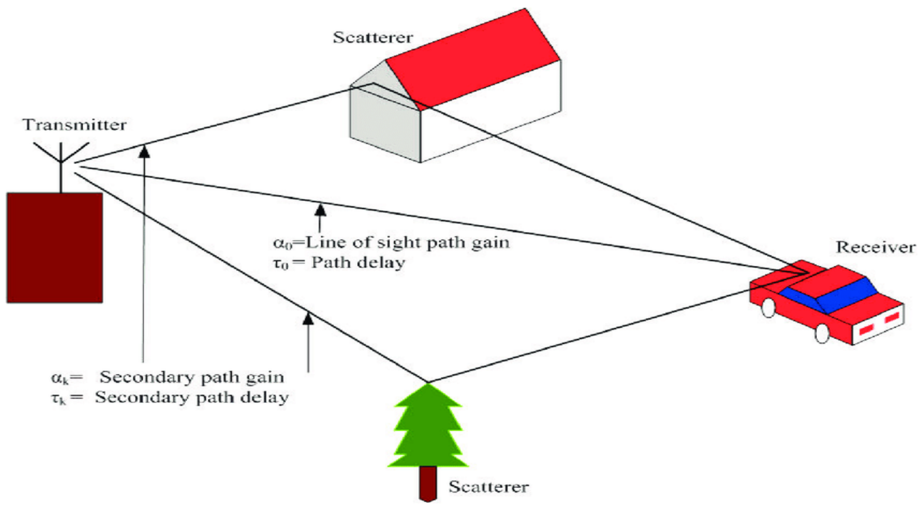
There are several different types of synchronization that can be used in telecommunication systems, including time synchronization, frequency synchronization, and phase synchronization.

- **Time synchronization** is important in telecommunication systems because it ensures that all devices are using the same time reference. This can be critical for applications such as scheduling, routing, and network management, where accurate timing is essential.
- **Frequency synchronization** is important in telecommunication systems because it ensures that all devices are operating at the same frequency. This can be critical for applications such as radio and television broadcasting, where multiple devices need to operate at the same frequency in order to avoid interference.
- **Phase synchronization** is important in telecommunication systems because it ensures that all devices are operating in phase with each other. This can be critical for applications such as radar and other systems that rely on precise phase relationships between signals.

Overall, synchronization is a key aspect of telecommunication systems, and it plays a critical role in ensuring the reliability and performance of these systems.

Chapter 8: Diversity Techniques

Multipath propagation



- Multipath is the propagation phenomenon that results in radio signals reaching the receiving antenna by two or more paths.
- In telecommunications, multipath propagation is the presence of multiple paths that a signal can travel from a transmitter to a receiver.
- This can occur when the signal is reflected off of obstacles or surfaces before reaching the receiver.
- As a result, the receiver may receive multiple copies of the same signal, each with a different delay and possibly different phase or amplitude.
- This can cause fading and interference, which can degrade the performance of a communication system.
- There are various techniques that can be used to mitigate the effects of multipath propagation, such as diversity techniques and equalization.

Advantages of Multipath Propagation

1. **Increased coverage**: Multipath propagation can allow a signal to reach areas that would be shadowed or blocked by obstacles if the signal were only able to propagate in a straight line.
2. **Increased capacity:** In some cases, multipath propagation can allow multiple signals to be transmitted simultaneously on the same frequency

without interference, which can increase the capacity of a communication system.

3. **Improved resistance to fading**: Fading, which is the loss of signal strength that can occur when a signal travels through a medium with varying properties, can be reduced by multipath propagation. This is because the multiple paths that the signal takes can cancel out the effects of fading on some of the paths.

Disadvantages of Multipath Propagation

1. **Interference:** The multiple copies of a signal that are received by the receiver can interfere with each other, which can cause errors and degrade the performance of the communication system.

2. **Increased complexity:** Dealing with multipath propagation can be more complex than dealing with a single path, as it requires the use of techniques such as diversity and equalization to mitigate the effects of multipath propagation.

3. **Increased power consumption:** Some techniques used to mitigate the effects of multipath propagation, such as diversity, may require additional hardware or transmitters, which can increase the power consumption of the communication system.

LOS AND NON-LOS

- In telecommunications, the line-of-sight (LOS) model is a model that assumes that a direct, unobstructed path exists between the transmitter and the receiver.
- This means that there are no obstacles or surfaces that can reflect or absorb the signal.
- The LOS model is often used to predict the performance of communication systems, such as those used in satellite or microwave communications.
- On the other hand, the non-line-of-sight (NLOS) model is a model that takes into account the presence of obstacles or surfaces that can reflect or absorb the signal.

- This can result in multipath propagation, where the signal takes multiple paths from the transmitter to the receiver.

- The NLOS model is often used to predict the performance of communication systems in urban or indoor environments, where there are many obstacles that can affect the signal.

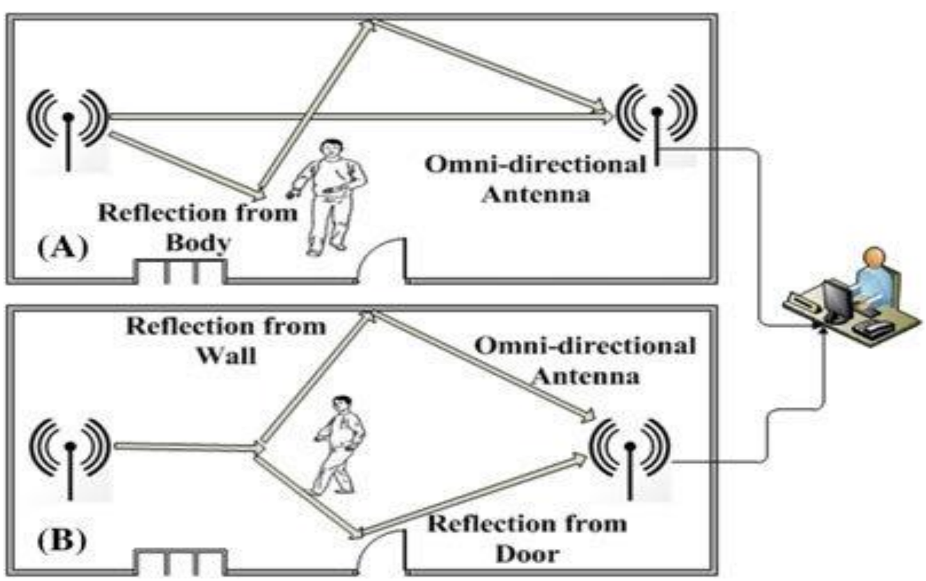
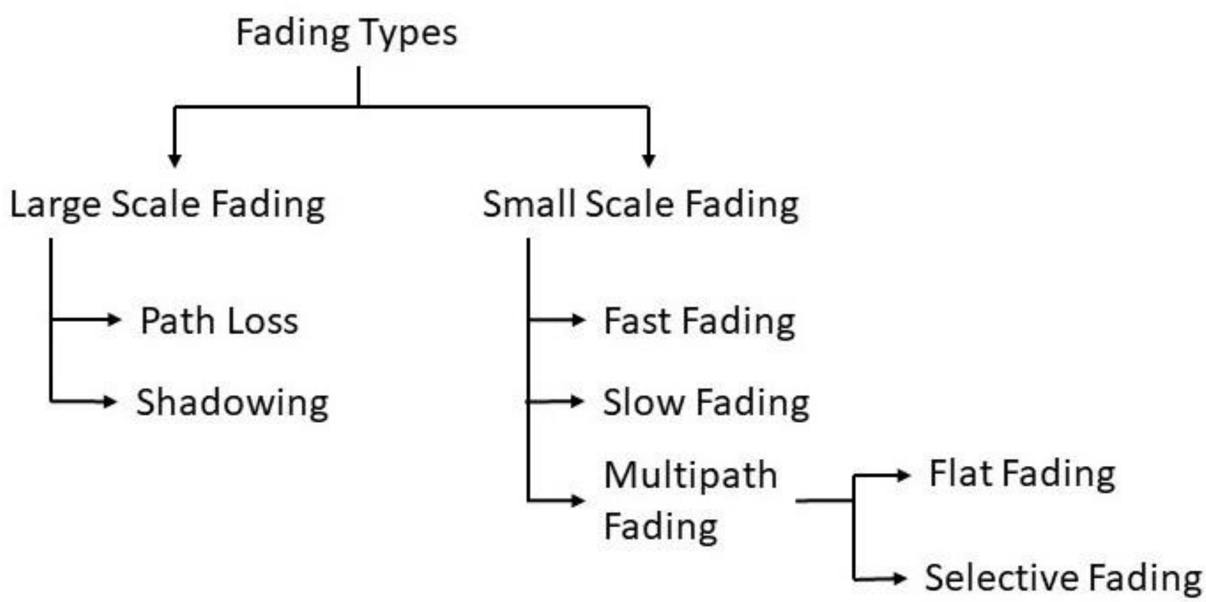


Fig: A) LOS   B) NON-LOS

Fading

- Superposition of multiple signals at the receiver cause random amplitude variation which is known as Fading.
- Fading is a phenomenon that occurs due to varying parameters and conditions of the channel during wireless propagation.



1. **Large Scale Fading:**

- This refers to the attenuation of signal power due to obstacles between the transmitter and receiver.
- It also covers the attenuation and fluctuations of signal when the signal is transmitted over a long distance (usually in kilometers).

i) Path Loss:

- It refers to the attenuation when a signal is transmitted over large distances.
- Wireless signals spread as they propagate through the medium and as the distance increases, the energy per unit area starts decreasing (Click here to try the Path Loss Calculator).

- This is a fundamental loss that is independent of the type of transmitter and medium.

ii) Shadowing:

- This refers to the loss in signal power due to the obstructions in the path of propagation.
- There are a few ways in which shadowing effects can minimize signal loss. One that is most effective, is to have a Line-Of-Sight propagation.

2. Small Scale Fading:

- This refers to the fluctuations in signal strength and phase over short distance and small duration of time.
- It is also called Rayleigh Fading.
- Small Scale Fading affects almost all forms of wireless communication and overcoming them is a necessity to increase efficiency and decrease error.

a) Fast Fading:

- It occurs mainly due to reflections for surfaces and movement of transmitter or receiver.

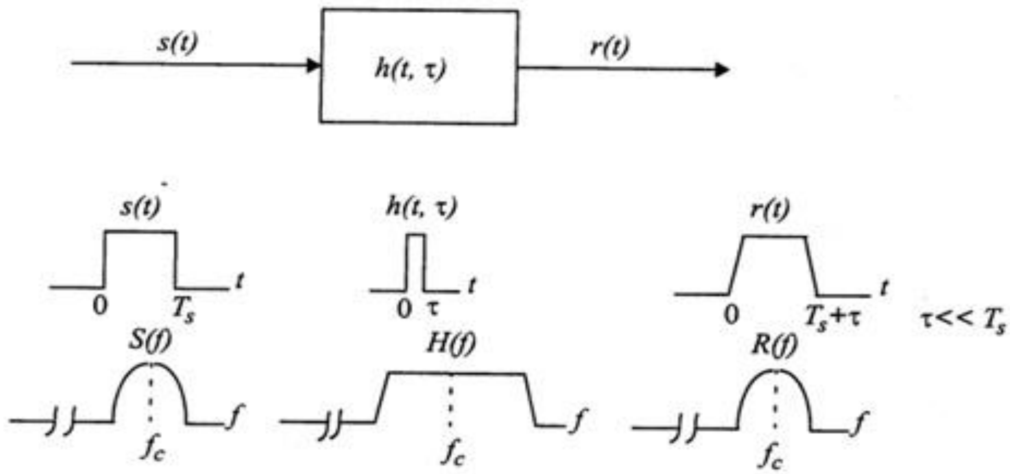
b) Slow Fading:

- It occurs mainly due to shadowing where large buildings or geographical structures obstruct the LOS.

c) Multipath Fading:

- It occurs when a signal reaches the receiver from various path i.e., when multipath propagation takes place.
- Multipath fading can affect all ranges of frequencies starting from low frequency to microwave and beyond.
- It affects both the amplitude and the phase of the signal causing phase distortions and ISI.
- Multipath fading can affect signal transmission in two ways:

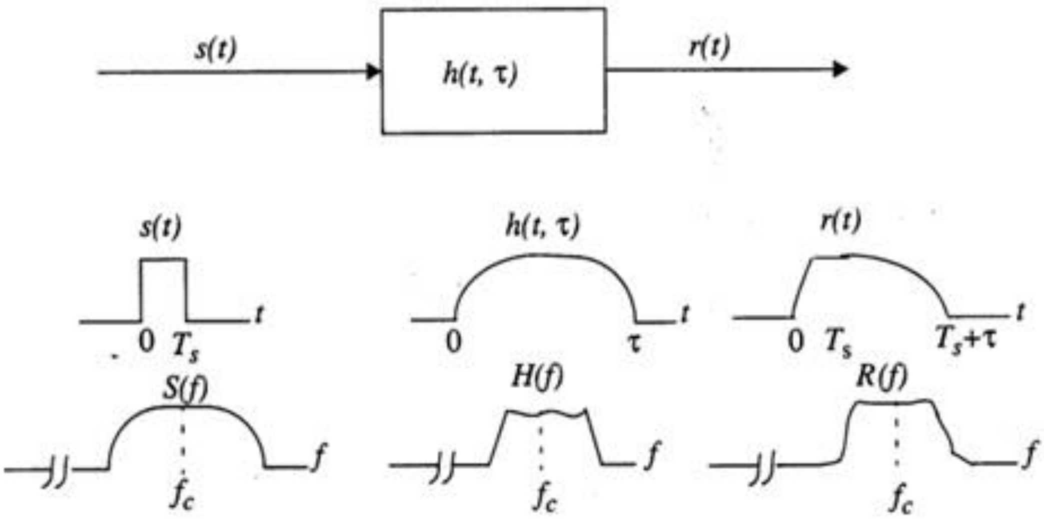
i) Flat Fading



Flat Fading

- Channel response in flat fading impaired signal has flat gain/linear phase over bandwidth (BW) which is greater than signal BW.
- Spectral characteristics of flat fading impaired signal are preserved over time.
- The figure-1 depicts time domain and frequency domain flat fading channel characteristics.
- As gain of the signal varies over time, flat fading channels are known as amplitude varying channels. They are also called as narrowband channels as signal BW is narrow compare to channel BW.
- Signal undergoes flat fading if following conditions are met
  ➡ Bs <<Bc
  ➡ Ts >>σ_τ

d) Frequency Selective Fading



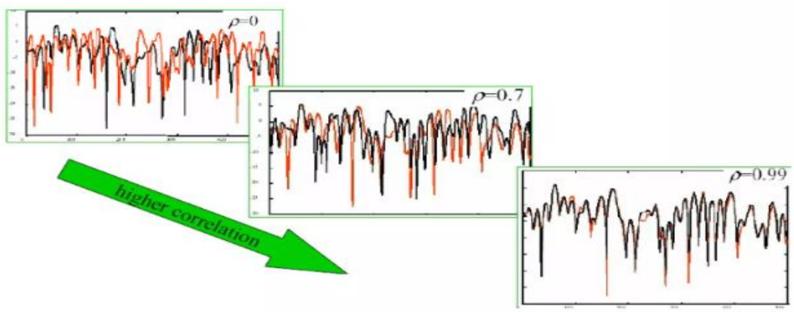
Frequency Selective Fading

- In this faded signal, channel response of signal has constant gain/linear phase over bandwidth (BW) which is less than that of signal BW.
- It is caused by ISI (Inter Symbol Interference) wherein received signal consists of multiple delayed and attenuated versions of the transmitted signal.
- The figure-2 depicts time domain and frequency domain frequency selective fading channel characteristics.
- Signal undergoes frequency selective fading if following conditions are met
  ➡ Bs >Bc
  ➡ Ts <σ_τ
- Rule of thumb for channel to be frequency selective is Ts <=10*σ_τ

Diversity System

- Diversity is a powerful communication receiver technique that provides wireless link improvement at a relatively low cost.
- **Diversity techniques** are used in wireless communications systems to primarily to improve performance over a fading radio channel.

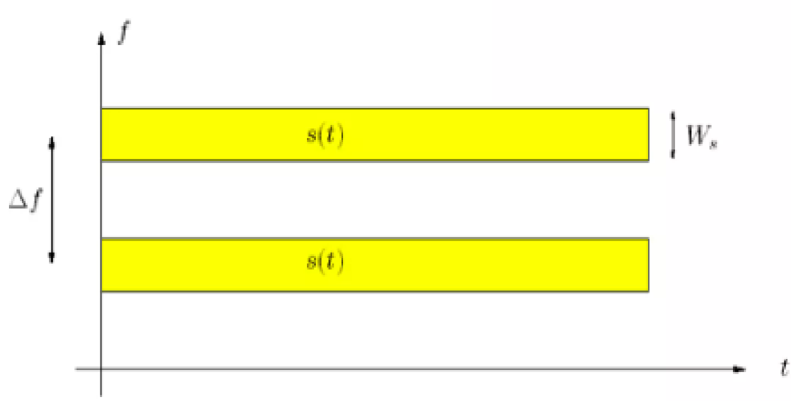
- Requirements for Diversity:
  1- Multiple branches
  2- Low correlation between branches



Frequency Diversity

- The same information signal is transmitted on different carriers, the frequency separation between them being at least the coherence bandwidth.

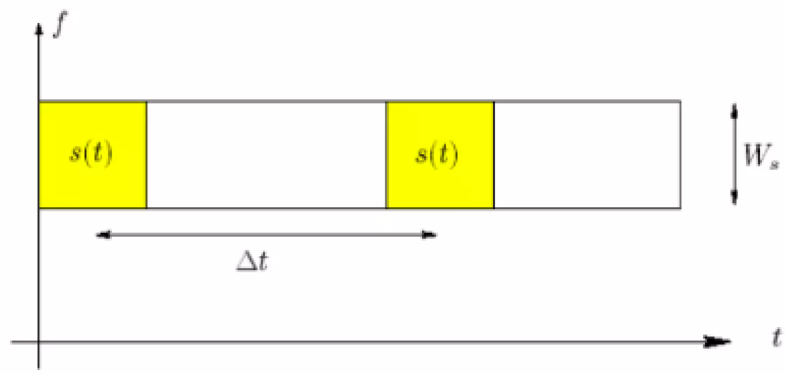
- **Frequency Diversity**
  - Is implemented by transmitting same information on more than one carrier frequency.
  - The separation between the carriers should be at least the coherent bandwidth $(\Delta f)c$.
  - Different copies undergo independent fading.
  - Only one antenna is needed.



Time Diversity

- The information signal is transmitted repeatedly in time at regularly intervals.
- The separation between the **transmit times should be greater than the coherence time, T_c**.
- The time interval depends on the fading rate, and increases with the decrease in the rate of fading.

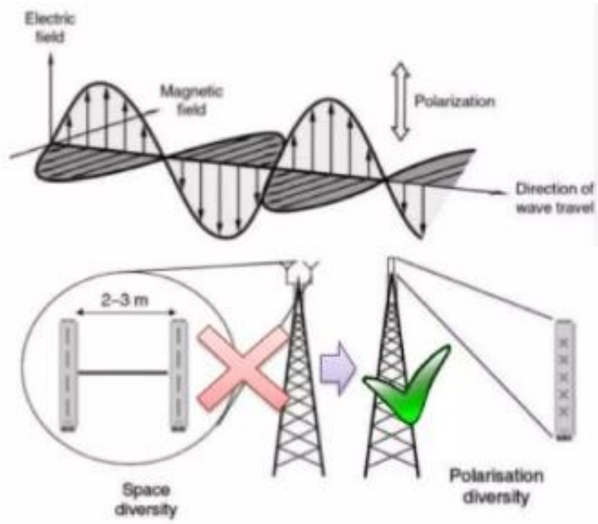
- **Time Diversity**
  - Repeatedly transmits information at the time spacing that exceeds the coherence time of the channel.
  - The interval between transmission of same symbol should be at least the coherence time (Δt)c.
  - Different copies undergo independent fading.
  - Reduction in efficiency (effective data rate < real data rate).



Polarization diversity

- Here, the electric and magnetic fields of the signal carrying the information are modified and many such signals are used to send the same information.
- Thus, **orthogonal type of polarization is obtained**.

- **Polarization Diversity**
  - Uses antennas of different polarizations i.e. horizontal and vertical.
  - The antennas take advantage of the multipath propagation characteristics to receive separate uncorrelated signals.



Angle Diversity

- Here, directional antennas are used to create independent copies of the transmitted signal over multiple paths.
- Angle diversity refers to the use of multiple antennas at different angles to improve the performance of a wireless communication system.
- By using multiple antennas that are spaced apart and oriented at different angles, a system can take advantage of the diversity of the multipath fading environment to improve the reliability and performance of the communication link.
- There are a number of different techniques that can be used to achieve angle diversity in telecommunications, including antenna arrays, beamforming, and space-time coding.
- These techniques can be used in a variety of applications, including mobile communication systems, wireless local area networks (WLANs), and satellite communication systems.

- Angle diversity can be particularly important in environments where the signal strength may be variable or subject to interference, as it can help to improve the reliability and robustness of the communication link.

Space Diversity

- Space diversity also known as antenna diversity, is one of the most popular forms of diversity used in wireless communication.
- In this there are multiple antennas placed at different spatial locations which results into different possibly independent received signals.
- Antenna space diversity is also used in base station design. At each cell site, multiple base station receiving antennas are used to provide diversity reception. Since the important scatterers are generally on the ground in the vicinity of the mobile, the base station antennas must be spaced far apart to achieve decorrelation.
- There are four types of space diversity methods:
    - Selection diversity
    - Feedback diversity
    - Maximal ratio combining
    - Equal gain diversity

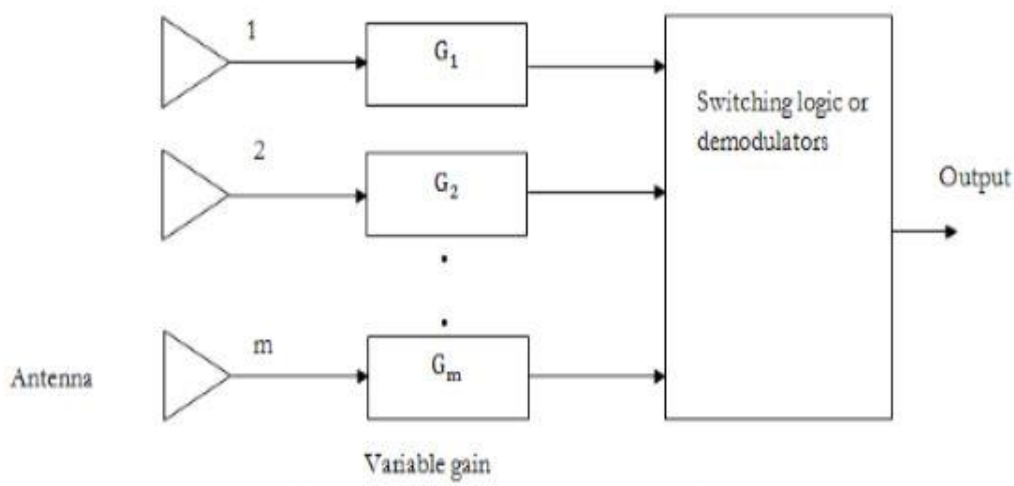


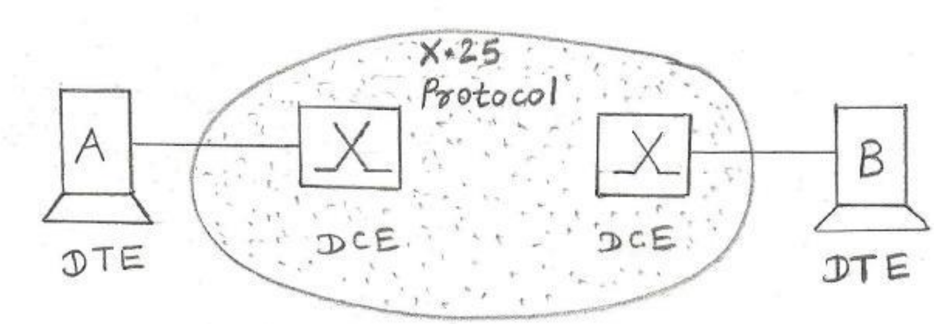
Fig: Block diagram of space diversity

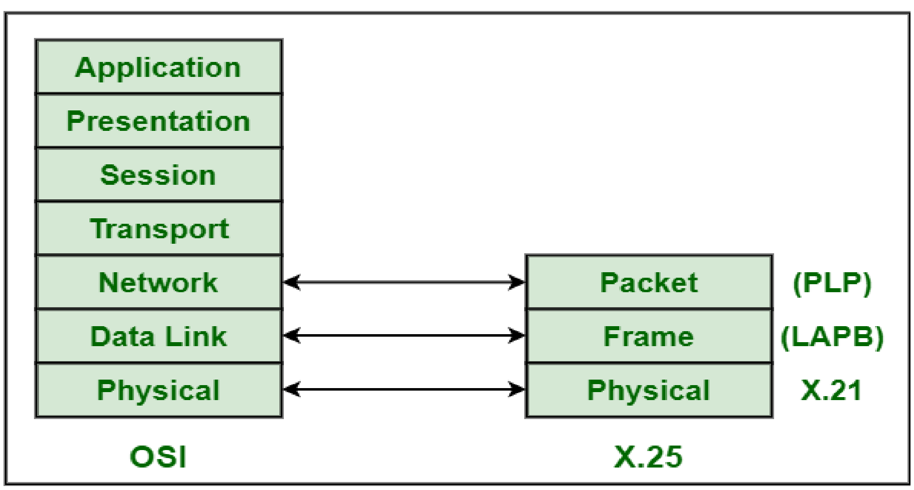
Sr. No.	Diversity scheme	Advantages	Disadvantages
1.	Polarization Diversity	• No space and extra bandwidth is required.	• 3 dB extra power is a must • Two branch diversity schemes is only possible.
2.	Space Diversity	• Several diversity branches are allowed. • It is also applicable to macroscopic diversity. • No extra bandwidth or power is required.	• Large hardware size is required. • Larger antenna spacing is a must for the microscopic diversity at the base station.
3.	Frequency Diversity	• Several diversity branches are allowed.	• Relevant power level of frequency spectrum are important.
4.	Time Diversity	• Hardware is simple. • Several diversity branches are allowed.	• Larger buffer memory is a must when diversity frequency is small. • More frequency spectrum is necessary according to the number of diversity branches.
5.	Angle Diversity	• Doppler spread can be reduced.	• Diversity gain will depend on the number of obstacles available around the terminal.
6.	Path Diversity	• No space is required. • No extra bandwidth and power are required.	• The diversity gain will depend on the delay status.

CHAPTER 9: Protocols in Telecommunications

X.25

- X.25 is a protocol suite defined by ITU-T for packet switched communications over WAN (Wide Area Network).
- Presently, it is used for networks for ATMs and credit card verification.
- It allows multiple logical channels to use the same physical line.
- It also permits data exchange between terminals with different communication speeds.
- An *X.25 network* is an interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) that operates in the packet mode.
- An X.25 network connects to public data networks by dedicated circuits. X.25 networks use the connection-mode network service.





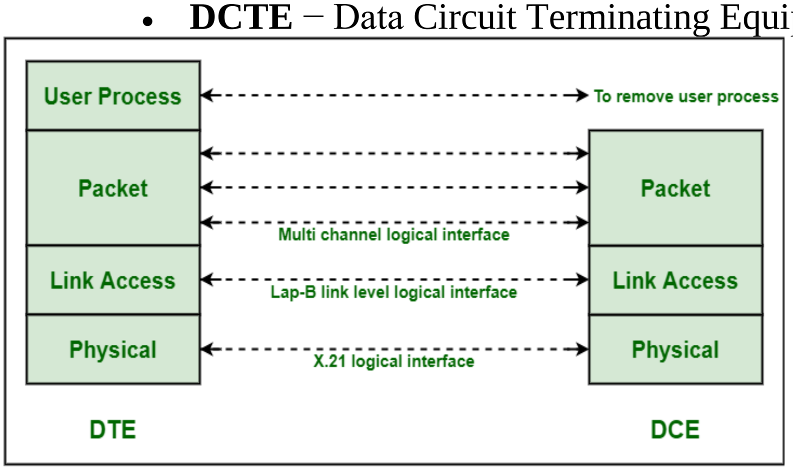
X.25 Layer Mapping with OSI Model

X.25 has three protocol layers

- **Physical Layer:** It lays out the physical, electrical and functional characteristics that interface between the computer terminal and the link to the packet switched node. X.21 physical implementer is commonly used for the linking.
- **Data Link Layer:** It comprises the link access procedures for exchanging data over the link. Here, control information for transmission over the link is attached to the packets from the packet layer to form the LAPB frame (Link Access Procedure Balanced). This service ensures a bit-oriented, error-free, and ordered delivery of frames.
- **Packet Layer:** This layer defines the format of data packets and the procedures for control and transmission of the data packets. It provides external virtual circuit service. Virtual circuits may be of two types: virtual call and permanent virtual circuit. The virtual call is established dynamically when needed through call set up procedure, and the circuit is relinquished through call clearing procedure. Permanent virtual circuit, on the other hand, is fixed and network assigned.

Equipment used

- **X.21** implementer
- **DTE** − Data Terminal Equipment
- **DCTE** − Data Circuit Terminating Equipment



Different Layers of X.25 and Interface between DTE and DCE

Benefits or advantages of X.25

➡It is reliable protocol as it uses error control and retransmission of bad packets.

➡It has faster response times.

➡It handles both high speed and low speed data requirements.

➡The network is highly available due to use of distributed routing.

➡It uses addressing capabilities.

➡It can be statistically multiplexed.

Drawbacks or disadvantages of X.25

➡It offers low data rate which is about 64 Kbps.

➡It utilizes flow control and error control at data link and network layer. This results into larger overhead and consecutively slows down the transmissions.

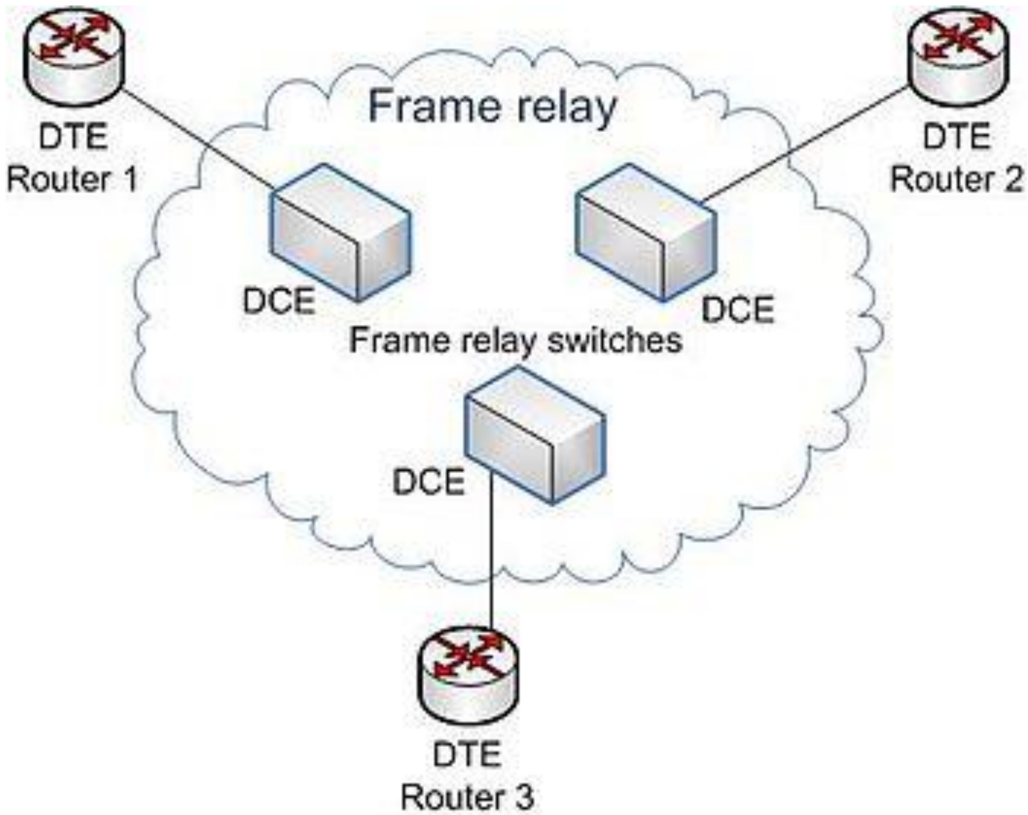
➡Queuing delays, small packet size, No QoS guarantee and used for data only.

Frame Relay

- Frame Relay is a packet switched communication service from LANs (Local Area Network) to backbone networks and WANs.
- It operates at two layers: physical layer and data link layer. It supports all standard physical layer protocols. It is mostly implemented at the data link layer.
- Frame Relay uses virtual circuits to connect a single router to multiple remote sites.
- In most cases, permanent virtual circuits are used, i.e., a fixed network-assigned circuit is used through which the user sees a continuous uninterrupted line. However, switched virtual circuits may also be used.
- Frame relay is a fast packet technology based on X.25. Data is transmitted by encapsulating them in multiple sized frames.
- The protocol does not attempt to correct errors and so it is faster.
- Error correction is handled by the endpoints, which are responsible for retransmission of dropped frames

Frame Relay Devices are

- **DTE** − Data Terminal Equipment
- **DCTE:** − Data Circuit Terminating Equipment



Advantages:

1. High speed
2. Scalable
3. Reduced network congestion
4. Cost-efficient
5. Secured connection

Disadvantages:

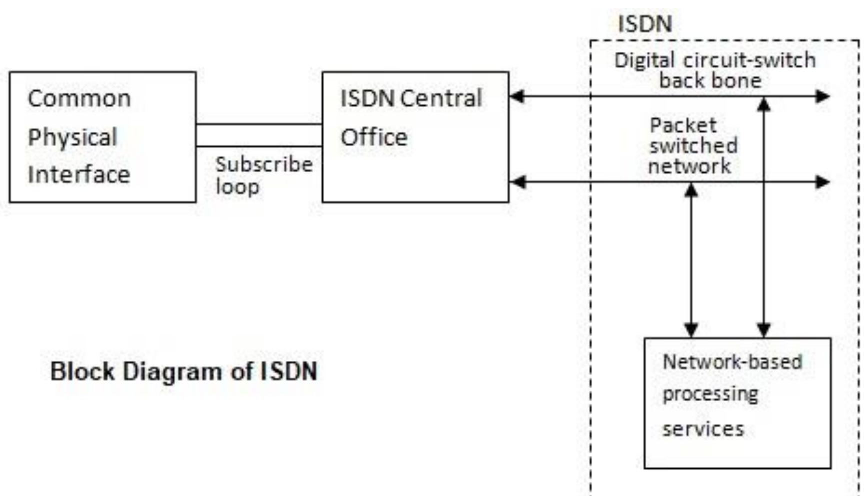
1. Lacks error control mechanism
2. Delay in packet transfer
3. Less reliable

Differences between Frame Relay and X.25

Frame Relay	X.25
• It has informal data rate	• It has fixed data rate
• It performs Multiplexing and switching at data link layer	• It performs Multiplexing and switching at network layer
• Frame Relay does not support Hop-to-Hop error and flow control.	• It performs Hop-to-Hop error and flow control at data link layer.
• It does not support End-to-End flow and error control.	• It performs End-to-End flow and error control at network layer
• Congestion control is required in frame relay.	• Congestion control is not required in X.25.
• Call control signaling requires separate logical connection from user data.	• X.25 uses same data for call control signaling.

ISDN

- ISDN is a set of protocols that is based on high-speed fully digitized telephone service.
- The main aim of ISDN is to provide a fully integrated digital service to the users.
- ISDN is a network concept providing a integration of data, voice and video.
- It's based on 64Kbps digital Communication channel.
- ISDN is a generic term for any network which connects homes and business together with a service company such as bank, air-lines, stock market etc. using a digital network.



Block Diagram of ISDN

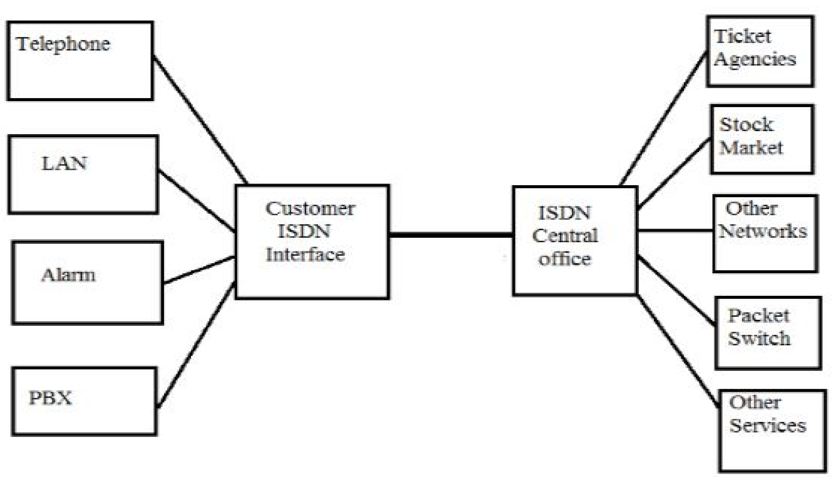


Fig no.2: Architecture of ISDN

Features of ISDN

- Offers point-to-point delivery.
- Network access and network interconnection for multimedia.
- Different data rates from 64 Kbps up to 2 Mbps are commercially available which can meet many needs for transporting multimedia and is four to many times more than today's analog modems
- Call set-up times are under one second. ISD N can dramatically speed up transfer of information over the Internet or over a remote LAN connection, especially rich media like graphics, audio or video or applications that normally run at LAN speeds.
- ISDN will be the feeder network for broadband ISDN based on ATM standards.

ISDN Services:

ISDN provides a fully integrated digital service to users. These services fall into 3 categories- bearer services, teleservices, and supplementary services.

1. **Bearer Services –**
    - Transfer of information (voice, data, and video) between users without the network manipulating the content of that information is provided by the bearer network.
    - There is no need for the network to process the information and therefore does not change the content.
    - Bearer services belong to the first three layers of the OSI model.
    - They are well defined in the ISDN standard.
    - They can be provided using circuit-switched, packet-switched, frame-switched, or cell-switched networks.

2. **Teleservices –**

    - In this, the network may change or process the contents of the data.
    - These services correspond to layers 4-7 of the OSI model.
    - Teleservices rely on the facilities of the bearer services and are designed to accommodate complex user needs.

- The user need not be aware of the details of the process. Teleservices include telephony, teletex, telefax, videotex, telex, and teleconferencing.
- Though the ISDN defines these services by name yet they have not yet become standards.

3. **Supplementary Service –**

- Additional functionality to the bearer services and teleservices are provided by supplementary services.
- Reverse charging, call waiting, and message handling are examples of supplementary services which are all familiar with today's telephone company services.
- Some of the examples of supplementary services are reverse charging, call waiting, and message handling.

Principles of ISDN:

Following are the principles of ISDN are:

- It supports both circuit switching & packet switching with the connections at 64 kbps.
- In ISDN layered protocol architecture is used for specification.
- ISDN services provides maintenance.
- ISDN services includes some network management functions.
- In ISDN network several configurations are possible for implementing.

Applications of ISDN are following –

* It has a high speed image applications that is used to transfer data between two or more users.

* It also has a high speed data transfer as the bit transfer rate through ISDN is very high.

* It also has very good voice service.

* It is also used in the video conferencing in which we have used the various devices like camera, microphone, speakers, TV etc. for carrying out communications with various users for formal purposes.

* It also provides Additional telephone lines in the homes etc.

Two Types of services offered by ISDN are following-

* **BRI services**
* **PRI services**

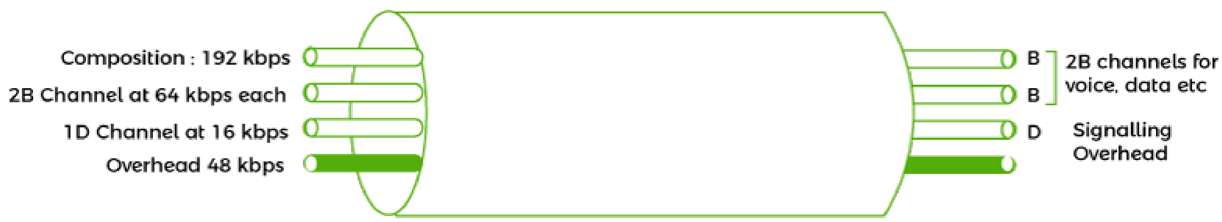
BRI services

- BRI stands for the Basic Rate Interface.
- The ISDN BRI services offer two B channels and one D channel.

- The B stands for bearer and D for data services.
- The B channel services operate at 64 KBPS. They are meant to carry user data.

- The D channel services are meant to carry control and signaling information.
- D-Channel service operates at 164 Kbps.
- The total bit rate of BRI services is 192 Kbps and it provides framing control and other overheads too.

- **The following figure shows the basic structure of the frame in the Basic Rate Interface is:**



Basic Rate Interface (BRI)

The 48 bit frame consists of

- 16 bits of B1 Channel
- 16 bits of B2 Channel
- 4 bits of D channel
- 12 overhead bits

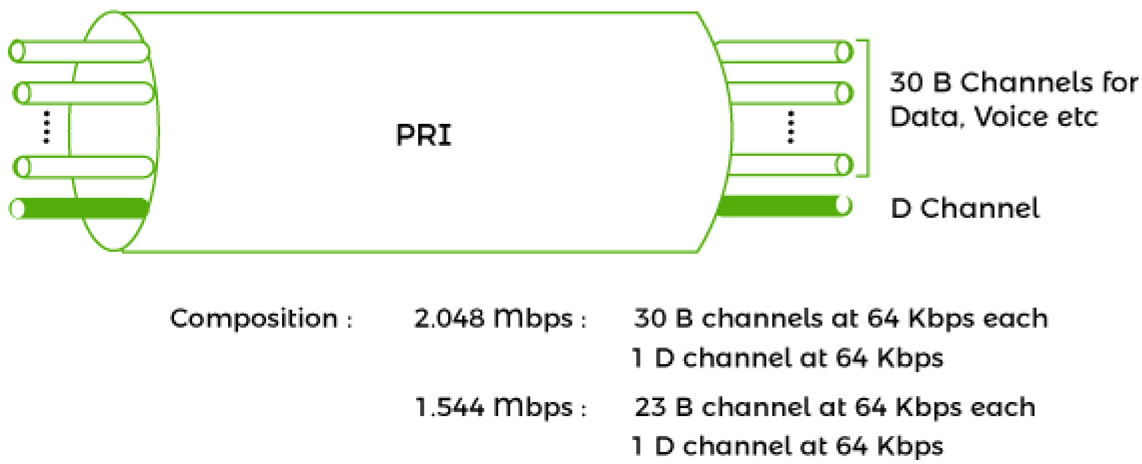
The frame is transmitted in 250 µsec, which results in the following bit rates:

- In frame each B channel = 16 / 250 µsec = 64 kbps
- In frame D channel = 4 / 250 µsec = 16 kbps
- In frame Overhead Bits = 12 / 250 µsec = 48 kbps
- In frame Overall Bit rate = 48 / 250 µsec = 192 kbps

PRI Services

- PRI stands for Primary Rate Interface Service.
- The ISDN PRI service offers 23 B Channels and one D channel.
- It provides total bit rate of 1.544 Mbps.

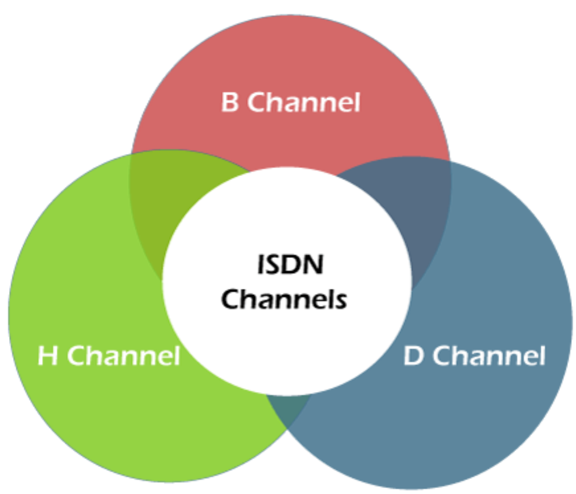
The following figure shows the basic structure of the frame in the Primary Rate Interface is:



ISDN CHANNELS:

- ISDN structure have a central ISDN office in which all the users are linked to this through a digital pipe.
- This digital pipe has different capacities and have a different data transfer rates and these are organized into multiple channels of different sizes.

ISDN standard have the following three types of channels:



B Channel:

- It stands for Bearer channel.
- It has a 64 kbps standard data rate.
- It is a basic user channel and can carry any digital information in full-duplex mode. In this transmission rate does not exceed 64 kbps.
- It can carry digital voice, digital data, and any other low data rate information.

D Channel:

- It stands for Data Channel.
- This channel carry control signal for bearer services.
- This channel is required for signaling or packet-switched data and all-controlling signals such as establishing calls, ringing, call interrupt, etc.
- Data (D) channel - 16 or 64 kbps

H Channel:

- It stands for Hybrid Channel.
- It provides user information at higher bit rates.

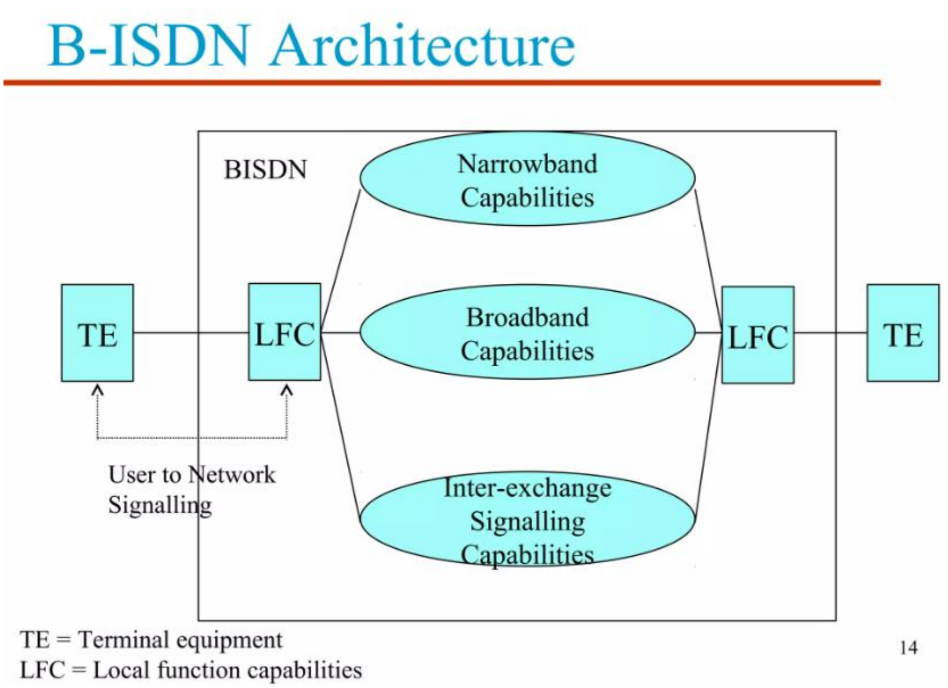
There are 3 types of Hybrid Channel depending on the data rates. Following are the hybrid channels types:

- Hybrid Channel 0 with 384 kbps data rate.
- Hybrid Channel 11 with 1536 kbps data rate.
- Hybrid Channel 12 with 1920 kbps data rate.

Broad-Band ISDN

- BISDN is an extension of ISDN, that is, it has narrowband capability of ISDN but also the broadband capability.
- The purpose of BISDN is to achieve complete integration of services, ranging from low-bit- rate burst signals to high-bit-rate continuous real-time signals.
- It provides user with additional data rates
  - 155.52 Mbps full-duplex
  - 155.52Mbps/ 622.08 Mbps
  - 622.08 Mbps full-duplex
- It has very high-performance switches.
- It exploits optical fiber transmission Technology.
- Here, ATM is specified for Information transfer across the user-network interface.
- Fixed size 53 octet packet with a 5-octet header.
- Implies that internal switching will be packet-based.

Architecture of B-ISDN



- The architecture of the B-ISDN includes low Layer capabilities and high Layer capabilities.
- These capabilities support the services within the B-ISDN and other networks by means of interworking B-ISDN with those networks.

Low Layer capabilities

The low layer capabilities of B-ISDN architecture are explained below −

- From the functional capabilities of the B-ISDN, as shown in Figure, the information transfer capabilities require further description.
- Broadband information transfer is provided by an ATM at the B-ISDN user-network interface (UNI) and at switching entities inside the network.

High Layer capabilities

The high layer capabilities of B-ISDN architecture are explained below −

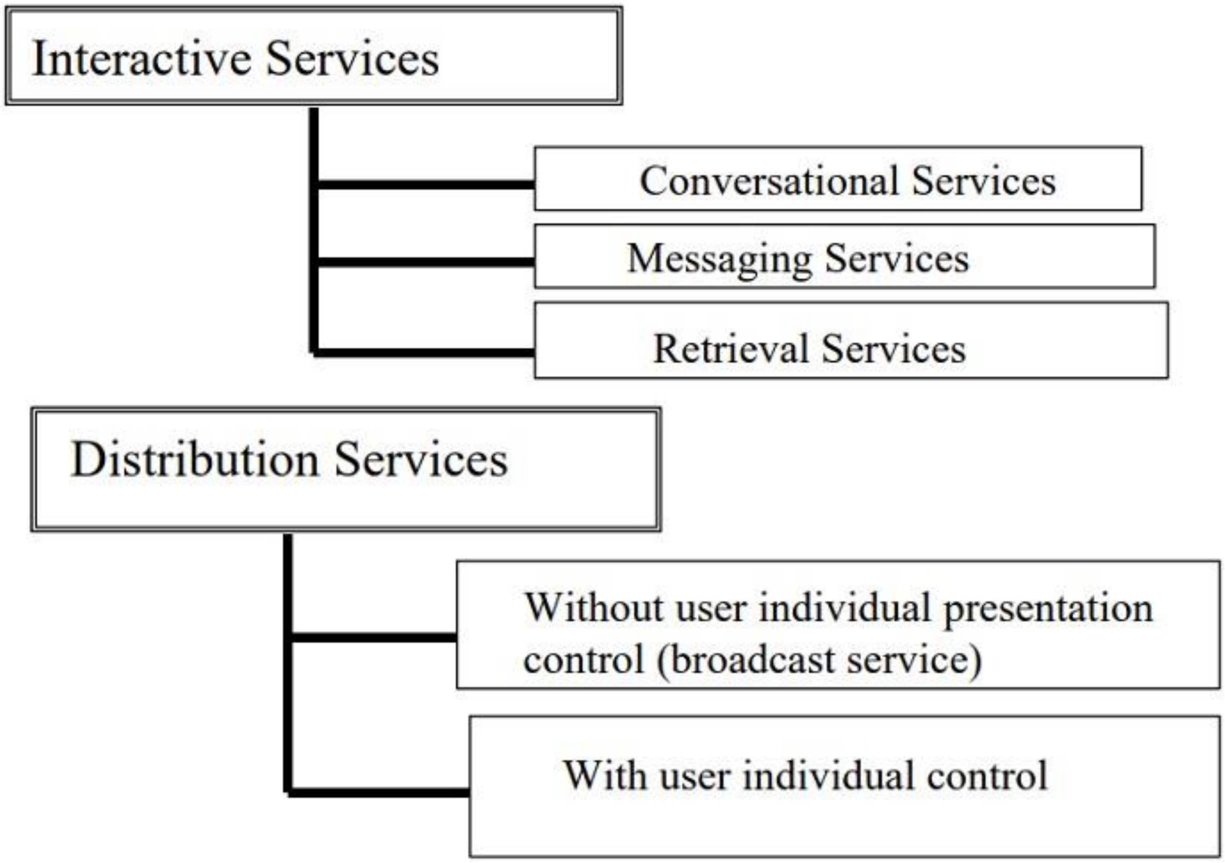
- Normally, the high Layer functional capabilities are involved only in the terminal equipment.
- The support of some services, provision of high layer functions could be made through special nodes in the B-ISDN belonging to the public network or to centers operated by other organizations and accessed via B-ISDN user-network or network node interfaces (NNIs).

B-ISDN Services

There are two types of B-ISDN services which are as follows −

- **Interactive Services** − Two-way exchange of information (other than control Signalling information) between two subscribers or between a subscriber and a service provider.
- **Distribution Services** − Primarily one way transfer of information, from service provider to B-ISDN subscriber.

These services are shown in the diagram format below −



ISDN has three layers:

1. The Physical Layer:

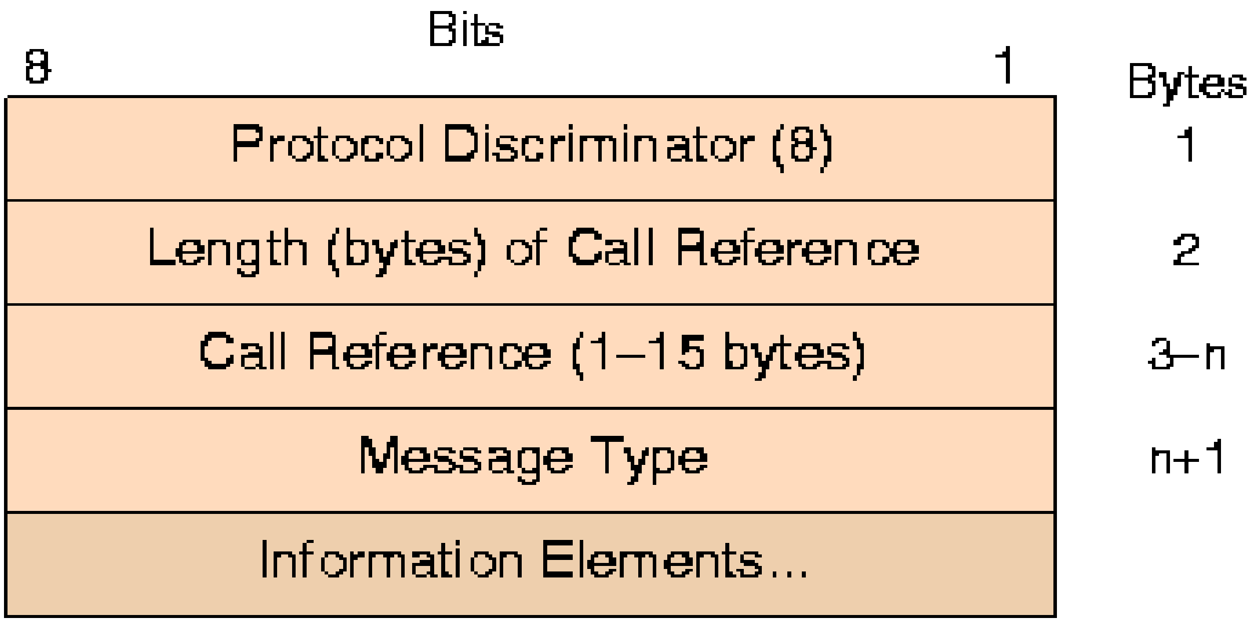
- This layer is responsible for the transmission of bits over a physical medium such as a copper wire or fiber optic cable.
- It includes the hardware components, such as cables and modems, that are needed to transmit and receive data.

2. The Data Link Layer:

- This layer is responsible for the reliable transmission of data over the physical medium.
- It includes protocols such as HDLC (High-Level Data Link Control) that ensure that data is correctly transmitted and received.

3.The Network Layer (ISDN layer three Protocol Explanations)

- This layer is responsible for routing data between devices on the network.
- It includes protocols such as ISDN Q.931, which is used to establish, maintain, and terminate connections between devices on the network.



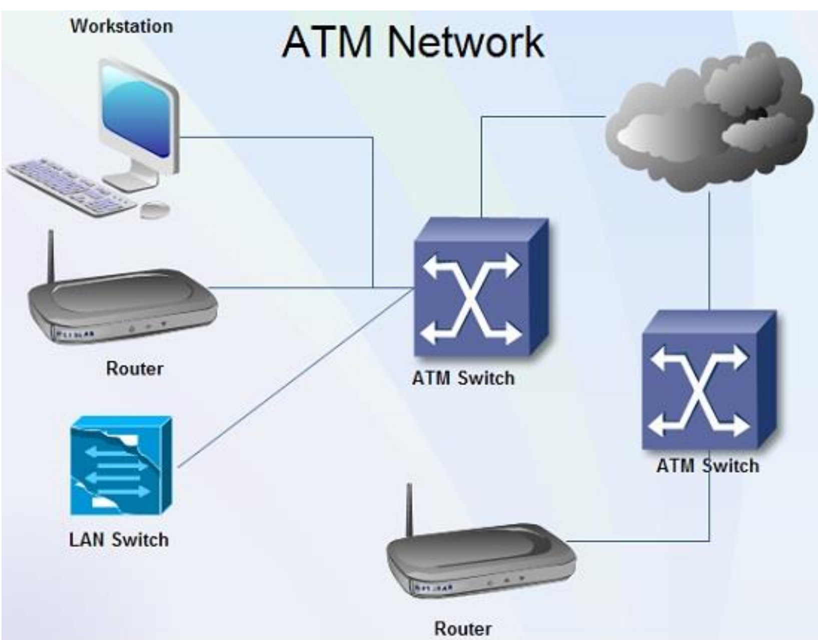
- Q.931 and TCP/IP work at different layers and share few features.
- Q.931 is an Integrated Services Digital Network (ISDN) signal connection control protocol that is an International Telecommunication Union (ITU)-T Recommendation.
- Q.931 was designed for ISDN call establishment, maintenance, and release of network connections between two DTEs on the ISDN D channel.
- Q.931 has reliable layers but does not provide retransmission or flow control. Q.931 frame elements are as follows:
  - Protocol discriminator (PD): Specifies connection signaling protocol. (e.g., PD=8 for DSS1)
  - Call reference value (CR): Addresses concurrent connections. he value is valid only during the actual time period of the connection
  - Message type (MT): Specifies three-message layer type (i.e., call setup, release and feature control) from Q.931 call control message set. There are messages defined for the call setup, the call release and the control of call features.
  - Information elements (IE): Specifies additional message details, that is, name, length and variable content field. Specify further information

which is associated to the actual message. An IE contains the IE name (e.g., bearer capability), their length and a variable field of contents.

ISDN is no longer in widespread use, as it has largely been replaced by newer technologies such as broadband internet and VoIP (Voice over Internet Protocol).

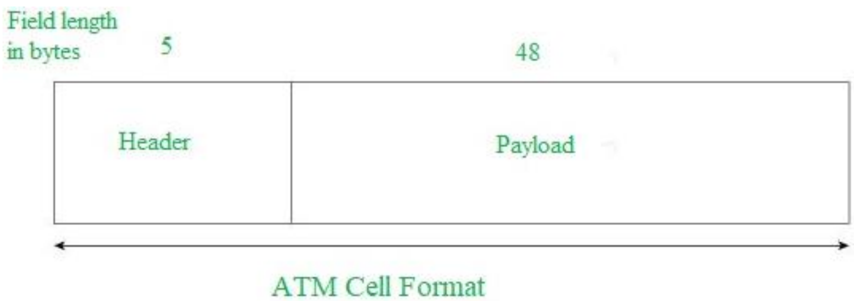
ATM

- Asynchronous Transfer Mode (ATM) is a network technology that is used to transmit data, voice, and video over a network.
- It is a connection-oriented technology that uses fixed-sized packets, called cells, to transmit data.
- ATM is designed to support high-speed communication over long distances, and it is often used in wide area networks (WANs) and metropolitan area networks (MANs).
- ATM can support multiple types of traffic, including real-time traffic such as voice and video, as well as non-real-time traffic such as data and file transfers.
- It is often used in conjunction with other technologies, such as SONET/SDH, to provide a high-capacity, reliable communications infrastructure.

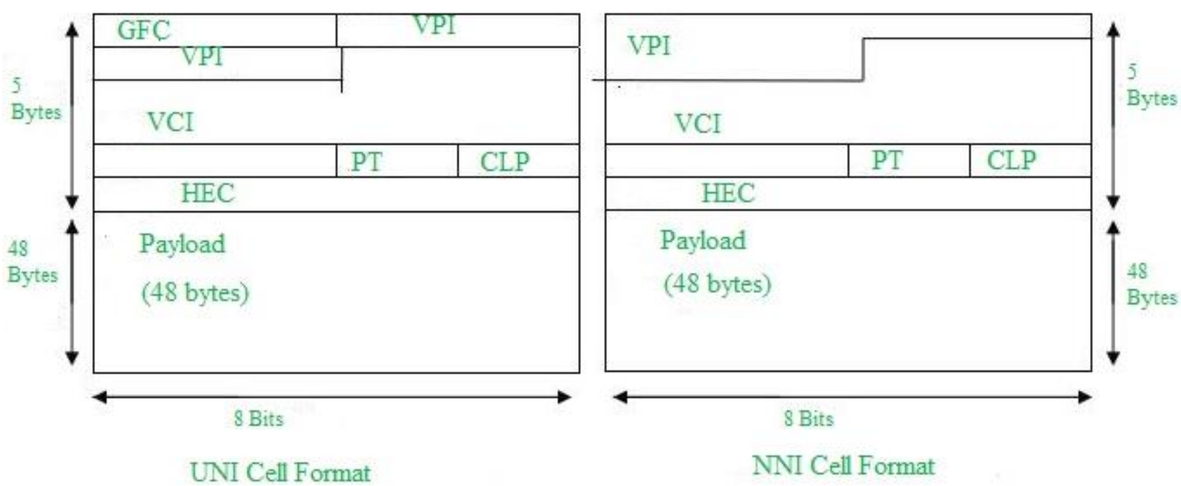


ATM Cell Format –

- As information is transmitted in ATM in the form of fixed-size units called **cells**.
- As known already each cell is 53 bytes long which consists of a 5 bytes header and 48 bytes payload.



Asynchronous Transfer Mode can be of two format types which are as follows:



1. **UNI Header:**
    - This is used within private networks of ATMs for communication between ATM endpoints and ATM switches.
    - It includes the Generic Flow Control (GFC) field.

2. **NNI Header:**

- It is used for communication between ATM switches, and it does not include the Generic Flow Control (GFC) instead it includes a Virtual Path Identifier (VPI) which occupies the first 12 bits.

Explanation of Header Format

- An ATM header is a specific type of header that is used to identify and transmit data within an ATM cell.
- The header is located at the beginning of the cell and contains information about the cell, including its type, destination, and source.

- The ATM header format typically includes the following fields:

GFC (General Flow Control) field: This field is used to identify the type of data being transmitted in the cell.

VPI (Virtual Path Identifier) field: This field is used to identify the path that the cell will take through the network.

VCI (Virtual Circuit Identifier) field: This field is used to identify the specific circuit or connection that the cell is associated with.

PT (Payload Type) field: This field is used to identify the type of data contained in the payload of the cell.

CLP (Cell Loss Priority) field: This field is used to indicate the priority of the cell. Cells with a higher priority are more likely to be transmitted before cells with a lower priority.

HEC (Header Error Check) field: This field is used to detect errors in the header of the cell. If an error is detected, the cell is not transmitted and is discarded.

Services provided by Asynchronous Transfer Mode (ATM)

Asynchronous Transfer Mode (ATM) is a networking technology that is used to transmit data, voice, and video over a network. Some of the services that can be provided using ATM include:

- **Data transmission:** ATM can be used to transmit data between computers and other devices over a network.

- **Voice transmission:** ATM can be used to transmit voice over a network, either as a standalone service or as part of a Voice over IP (VoIP) system.

- **Video transmission:** ATM can be used to transmit video over a network, either as a standalone service or as part of a video conferencing system.

- **High-speed internet access:** ATM can be used to provide high-speed internet access to individuals and businesses.

- **Virtual private networks (VPNs):** ATM can be used to create a secure, private network over a public network, such as the internet.

- **Quality of Service (QoS):** ATM can be used to prioritize different types of traffic and ensure that real-time traffic, such as voice and video, is transmitted with minimal delay and loss.

- **Network management:** ATM can be used to manage and monitor network performance and ensure that the network is running efficiently.

Application of ATM

- **Wide area networks (WANs):** ATM is often used in WANs to provide high-speed connectivity between different locations.

- **Metropolitan area networks (MANs):** ATM is often used in MANs to provide high-speed connectivity within a city or metropolitan area.

- **Internet service providers (ISPs):** ATM is often used by ISPs to provide high-speed internet access to individuals and businesses.

- **Private networks**: ATM is often used to create private networks, such as virtual private networks (VPNs), to allow secure communication over a public network.

- **Voice over IP (VoIP):** ATM can be used to transmit voice over a network using VoIP technology.

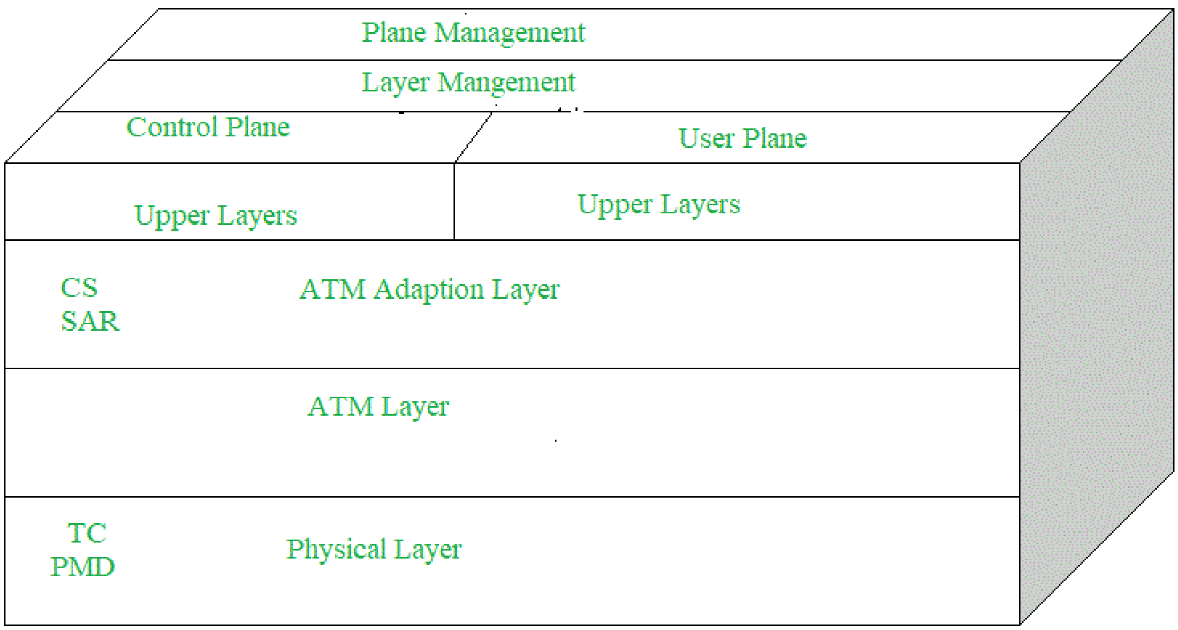
- **Video conferencing**: ATM can be used to transmit video for applications such as video conferencing.

- **Data centers**: ATM can be used to connect servers and other devices within a data center.

- **Wireless networks:** ATM can be used to transmit data over wireless networks, such as cellular networks.

ATM LAYERS (NETWORK ACCESS)

- Asynchronous Transfer Mode (ATM) is a networking technology that uses fixed-sized packets, called cells, to transmit data.
- In order to access an ATM network, a device must be equipped with an ATM network interface, which is responsible for transmitting and receiving ATM cells.
- There are several types of ATM network interfaces, including:



1. **ATM Adaption Layer (AAL) –**
    - It is meant for isolating higher-layer protocols from details of ATM processes and prepares for conversion of user data into cells and segments it into 48-byte cell payloads.
    - AAL protocol excepts transmission from upper-layer services and helps them in mapping applications, e.g., voice, data to ATM cells.

2. **Physical Layer –**
    - It manages the medium-dependent transmission and is divided into two parts physical medium-dependent sublayer and transmission convergence sublayer.

- The main functions are as follows:
    - It converts cells into a bitstream.
    - It controls the transmission and receipt of bits in the physical medium.
    - It can track the ATM cell boundaries.
    - Look for the packaging of cells into the appropriate type of frames.

3. **ATM Network Interface Card (NIC):**
    - An ATM NIC is a hardware device that is installed in a computer or other device to enable it to communicate over an ATM network.
    - The NIC includes both an AAL and a PHY.

In order to access an ATM network, a device must have an ATM NIC installed and be connected to the network via a physical medium, such as a copper or fiber optic cable. The device must also be configured to use the appropriate AAL for the type of traffic it will be transmitting.

ATM Signalling:

- Asynchronous Transfer Mode (ATM) is a networking technology that uses fixed-sized packets, called cells, to transmit data.
- ATM uses a signaling protocol to establish, maintain, and terminate connections between devices on the network.
- The signaling protocol is responsible for establishing and maintaining the connection between two devices, as well as allocating resources and negotiating parameters for the connection.

There are several types of ATM signaling protocols, including:

i) ATM Forum UNI Signaling (AFS):

- AFS is a signaling protocol that is used to establish connections between devices on an ATM network.

- It is used in the User-Network Interface (UNI) between a user device and an ATM network.

ii) **ATM Forum Network-Network Interface (NNI) Signaling:**

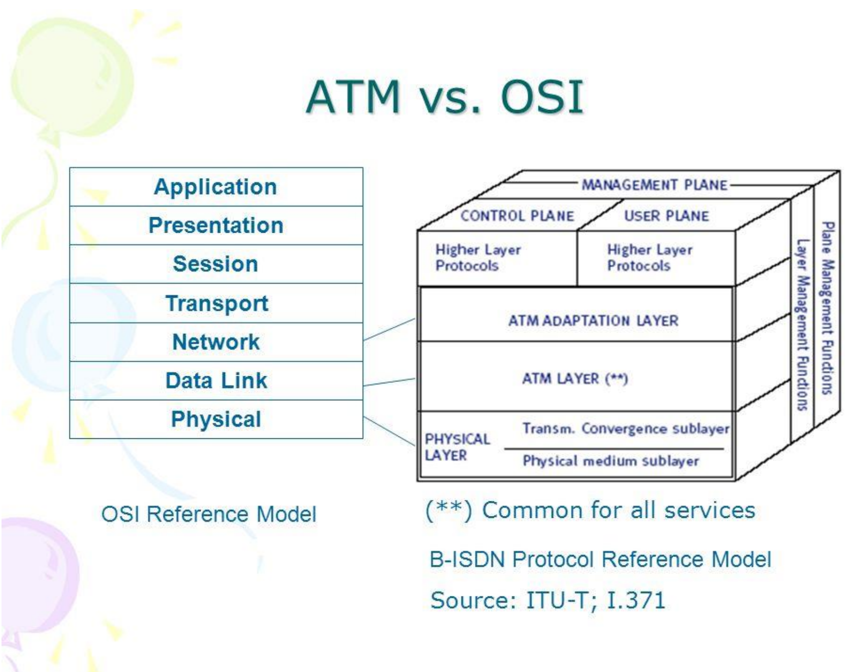
- NNI signaling is a signaling protocol that is used between two ATM switches to establish and maintain connections between devices on the network.

iii) **Signaling System 7 (SS7):**

- SS7 is a signaling protocol that is used in telecommunications networks to establish and maintain connections between devices.
- It is often used in conjunction with ATM to provide signaling for voice and data transmission over an ATM network.

ATM signaling plays a critical role in the operation of an ATM network, as it enables devices on the network to communicate with each other and establish connections for the transmission of data, voice, and video.

ATM with OSI layer



Advancement of ATM over Frame Relay:

Criteria	Frame Relay	ATM
Other Name	Frame Relay	Cell Relay
Frame Size	Variable & depends on the information sends.	Fixed & each cell is 53 bytes (header and data).
Platform	Software Controlled	Hardware implementation convenience.
Cost	Low	High
Overhead	Medium	Less
Throughpu t	Medium	High
Reliability	High	Low
Speed	Low	High
Data Rate	64 Kbps up to 45 Mbps (T3).	155.520 Mbps or 622.080 Mbps.
Coverage Area	Suitable for WAN and MAN only.	Suitable for WAN, MAN, LAN and CAN
Error Control	It does not support error and flow control, it let these function for upper layers.	It provides minimal error and flows control via 8 bits in ATM cell header called HEC (Header Error Control) that used at UNI interface format only.

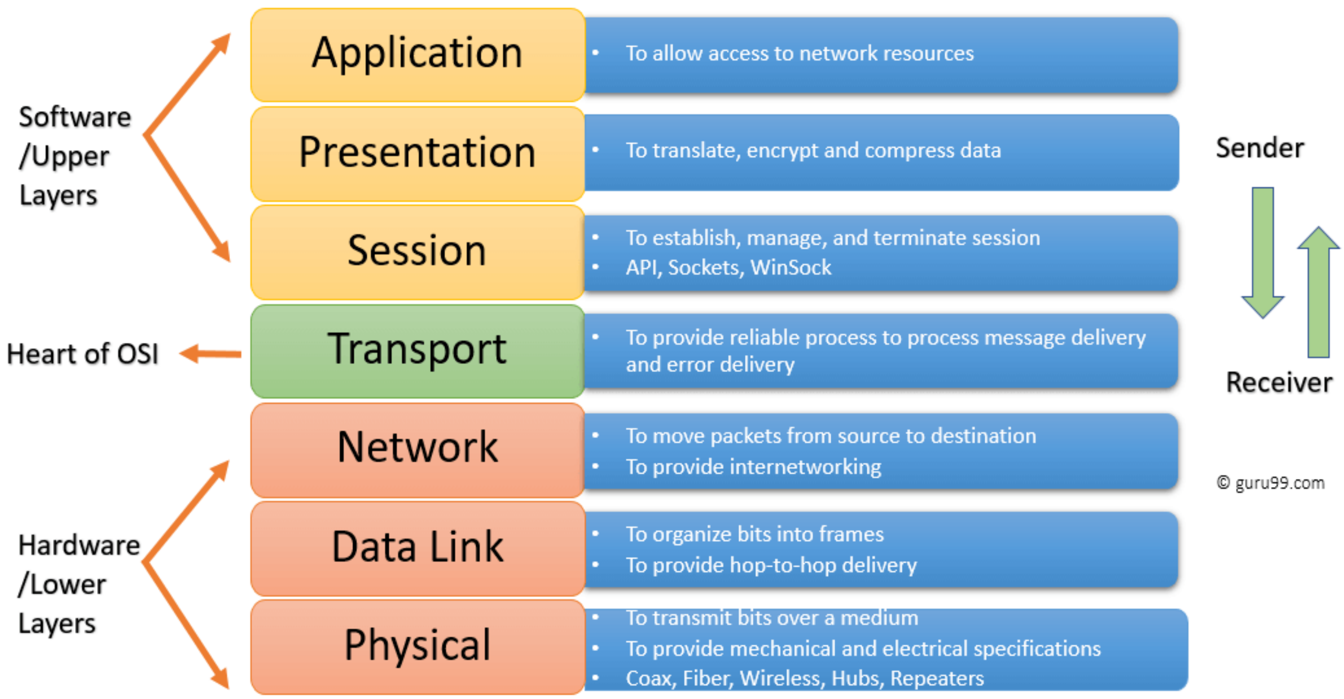
Display OSI model with DCTA units' functions and examples of protocols used in each layer, to establish common caption in the network.

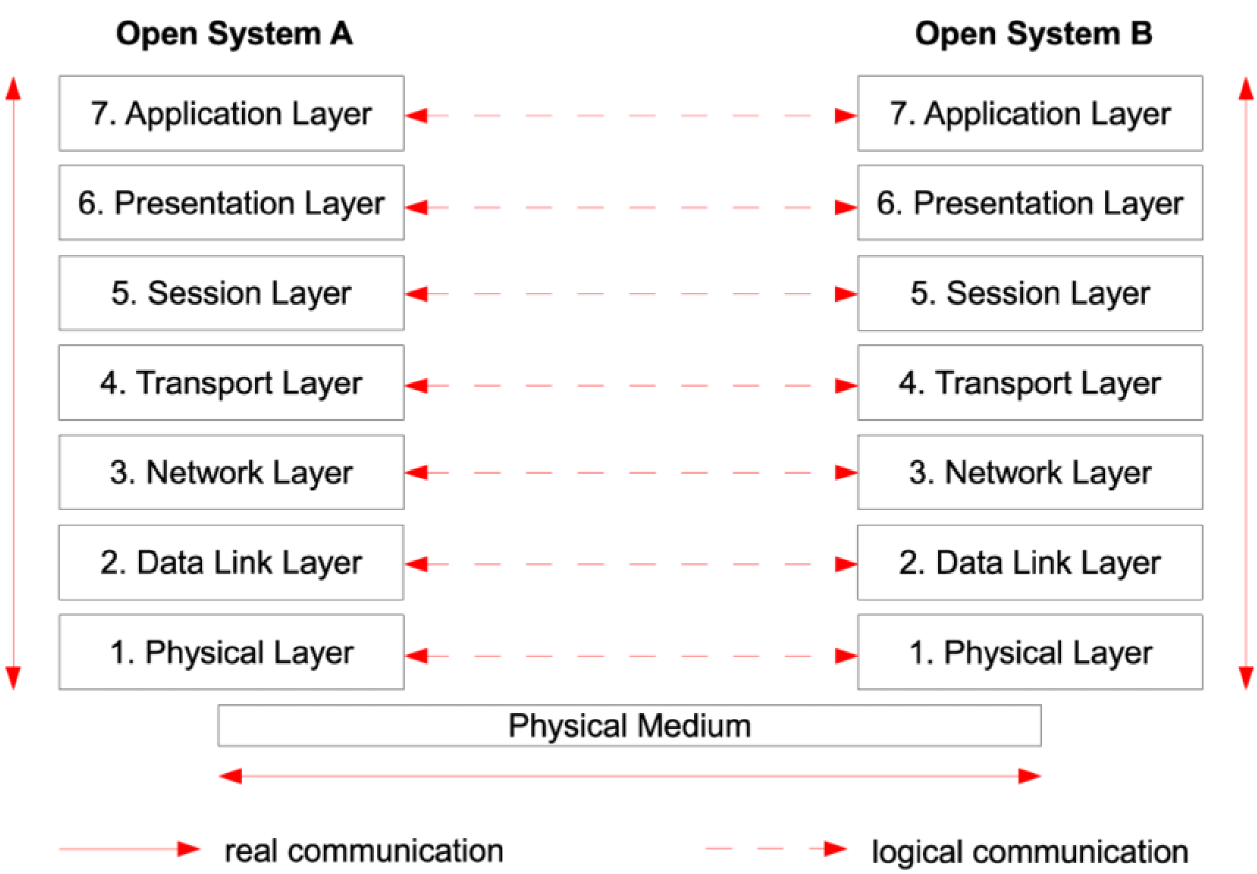
OSI model:

- The OSI Model is a logical and conceptual model that defines network communication used by systems open to interconnection and communication with other systems.
- The Open System Interconnection (OSI Model) also defines a logical network and effectively describes computer packet transfer by using various layers of protocols.

Why of OSI Model?

- Helps you to understand communication over a network.
- Troubleshooting is easier by separating functions into different network layers.
- Helps you to understand new technologies as they are developed.
- Allows you to compare primary functional relationships on various network layers.


Software
/Upper
Layers
Heart of OSI
Hardware
/Lower
Layers
Application
To allow access to network resources
Presentation
To translate, encrypt and compress data
Session
To establish, manage, and terminate session
API, Sockets, WinSock
Transport
To provide reliable process to process message delivery and error delivery
Network
To move packets from source to destination
To provide internetworking
Data Link
To organize bits into frames
To provide hop-to-hop delivery
Physical
To transmit bits over a medium
To provide mechanical and electrical specifications
Coax, Fiber, Wireless, Hubs, Repeaters
Sender
Receiver
© guru99.com


Open System A
Open System B
7. Application Layer
6. Presentation Layer
5. Session Layer
4. Transport Layer
3. Network Layer
2. Data Link Layer
1. Physical Layer
Physical Medium
real communication
logical communication

1. **Physical layer:**
    - The physical layer is responsible for transmitting raw data over a physical medium, such as a cable.
    - Protocols used in this layer include Ethernet and Wi-Fi.

2. **Data link layer:**
    - The data link layer is responsible for transmitting data over a link between two devices.
    - It is responsible for error detection and correction, as well as flow control.
    - Protocols used in this layer include ARP, PPP, and Frame Relay.

3. **Network layer:**
    - The network layer is responsible for routing data through the network.
    - It is responsible for deciding the best path for data to take, based on factors such as network congestion and available resources.
    - Protocols used in this layer include IP, ICMP, and IGMP.

4. **Transport layer:**
    - The transport layer is responsible for ensuring that data is delivered reliably from one device to another.
    - It is responsible for error checking, flow control, and retransmission of lost or damaged data.
    - Protocols used in this layer include TCP and UDP.

5. **Session layer:**
    - The session layer is responsible for establishing, maintaining, and terminating connections between devices.
    - Protocols used in this layer include RPC and SQL.

6. **Presentation layer:**
    - The presentation layer is responsible for formatting and encoding data for transmission.
    - It is also responsible for converting data between different formats, such as text and binary.
    - Protocols used in this layer include TLS and SSL.

7. **Application layer:**
    - The application layer is responsible for providing services to the user, such as email and file transfer.
    - Protocols used in this layer include HTTP, FTP, and SMTP.

DCTA

- DCTA stands for Data Communications Technology Association.
- It is an organization that promotes the development and use of data communications technologies.
- It is not directly related to the OSI model.
- The DCTA is an independent organization that promotes the development and use of data communications technologies.
- It is not directly related to the OSI model or any specific layer of the model.
- It may, however, work with different technologies and protocols that are used in the various layers of the OSI model.

Advantages of the OSI Model

Here, are major benefits/pros of using the OSI model:

- It helps you to standardize router, switch, motherboard, and other hardware
- Reduces complexity and standardizes interfaces
- Facilitates modular engineering
- Helps you to accelerate the evolution.
- Provide support for connection-oriented services as well as connectionless service.

- It is a standard model in computer networking.
- Supports connectionless and connection-oriented services.
- Offers flexibility to adapt to various types of protocols

Disadvantages of the OSI Model

Here are some cons/ drawbacks of using OSI Model:

- Fitting of protocols is a tedious task.
- You can only use it as a reference model.
- Doesn't define any specific protocol.
- In the OSI network layer model, some services are duplicated in many layers such as the transport and data link layers
- Layers can't work in parallel as each layer need to wait to obtain data from the previous layer.

Layer	Name	Function	Protocols
Layer 7	Application	To allow access to network resources.	SMTP, HTTP, FTP, POP3, SNMP
Layer 6	Presentation	To translate, encrypt and compress data.	MPEG, ASCH, SSL, TLS
Layer 5	Session	To establish, manage, and terminate the session	NetBIOS, SAP
Layer 4	Transport	The transport layer builds on the network layer to provide data transport from a process on a source machine to a process on a destination machine.	TCP, UDP
Layer 3	Network	To provide internetworking. To move packets from source to destination	IPV5, IPV6, ICMP, IPSEC, ARP, MPLS.
Layer 2	Data Link	To organize bits into frames. To provide hop-to-hop delivery	RAPA, PPP, Frame Relay, ATM, Fiber Cable, etc.

Layer	Name	Function	Protocols
Layer 1	Physical	To transmit bits over a medium. To provide mechanical and electrical specifications	RS232, 100BaseTX, ISDN, 11.

Chapter 11- NGN

Convergence of the Media

- Convergence of media refers to the integration of various forms of communication and media, such as television, radio, print, and the Internet, into a single platform.
- This can be achieved through the use of digital technologies, which allow different types of media to be accessed and shared on a single device, such as a smartphone or a computer.
- Convergence of media has led to significant changes in the way that people consume media and access information, as it has made it easier for them to access a wide range of content from a single source.
- It has also had an impact on the media industry, as traditional media companies have had to adapt to the changing landscape and compete with newer digital media companies.

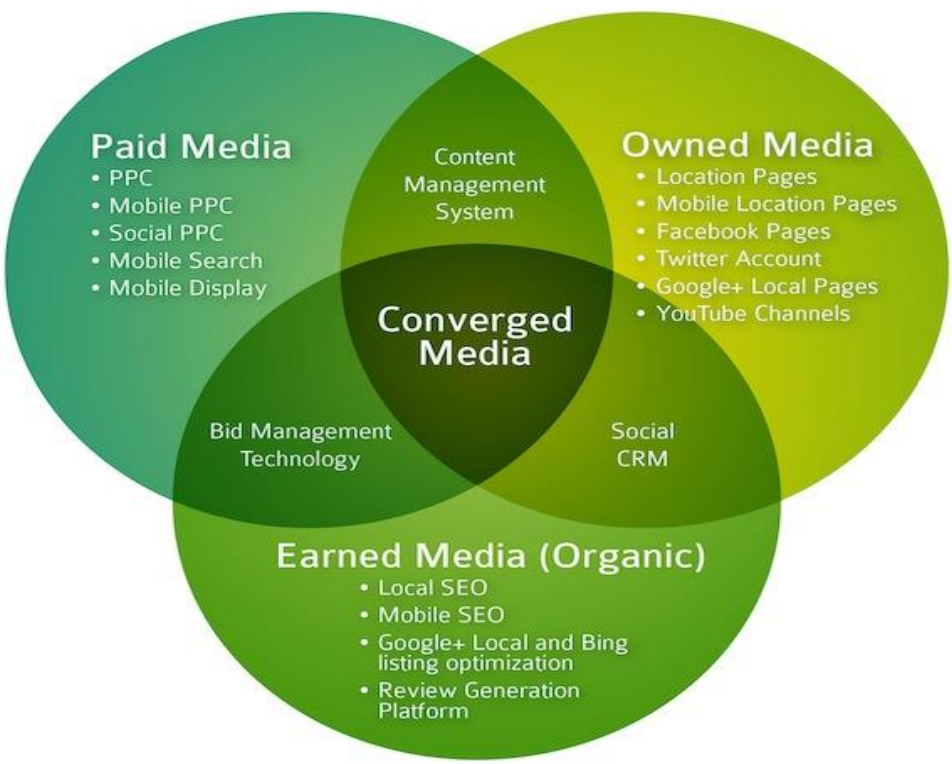
There are several types of convergence of media, including:

1. **Technological convergence:** This refers to the integration of different types of media and communication technologies, such as the convergence of television and the Internet.

2. **Industrial convergence**: This refers to the consolidation of media companies and the emergence of media conglomerates, which own multiple media outlets in different sectors, such as television, radio, and print.
3. **Cultural convergence:** This refers to the mixing and blending of different cultures and media content, as a result of increased global connectivity and the sharing of media across borders.
4. **Structural convergence:** This refers to the convergence of media content and distribution platforms, such as the emergence of streaming services that offer a variety of TV shows, movies, and other content.
5. **Convergence of media ownership:** This refers to the concentration of media ownership in the hands of a few large companies, which can have significant implications for the diversity of media content and the freedom of the press.

Need of Media Convergence

- ✓ Technologically rich societies have entered the digital age
- ✓ Media industries are grappling with new opportunities - and threats - afforded by what is called "convergence".
- ✓ Media people tend to get very excited about convergence, because it holds so much promise.
- ✓ The melding together of different media, incorporating new personalized services is both impressive and overwhelming.



There are several potential disadvantages of media convergence:

Loss of diversity: The consolidation of media ownership can lead to a reduction in the diversity of viewpoints and perspectives represented in the media.

Decreased competition: The convergence of media can lead to fewer media companies, which can reduce competition and lead to higher prices for media content.

Loss of jobs: The integration of different types of media can lead to the automation of certain tasks, which can result in job losses for media workers.

Privacy concerns: The convergence of media can lead to the collection and sharing of large amounts of personal data, which can raise concerns about privacy and the misuse of personal information.

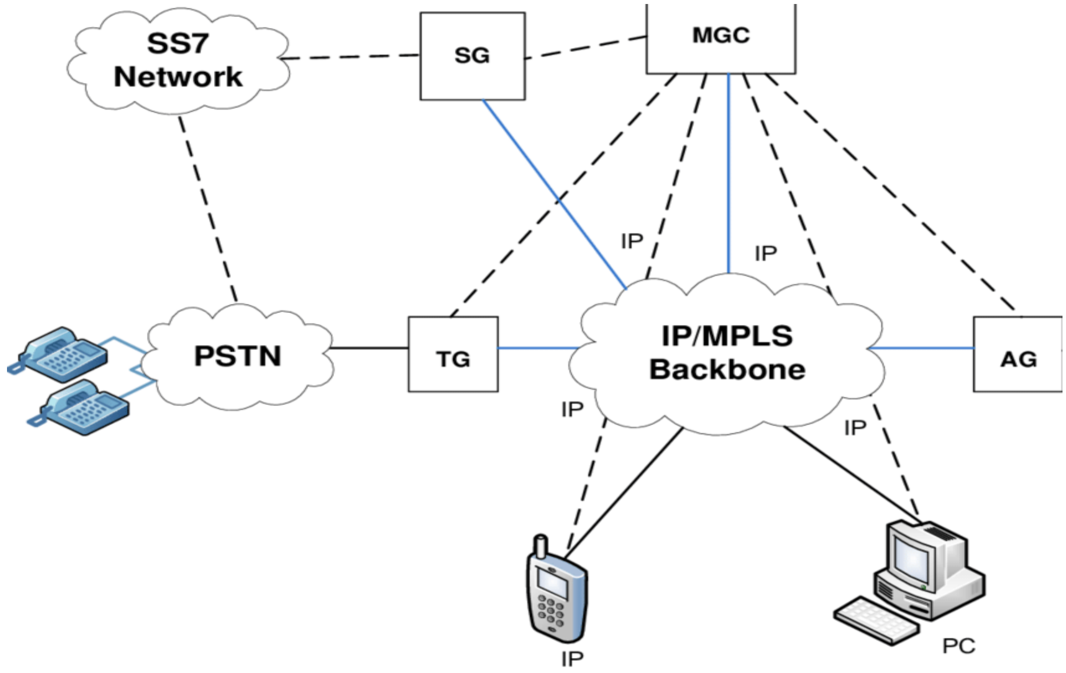
Dependence on technology: The reliance on technology for the convergence of media can make it vulnerable to disruptions or outages, which can impact access to media content.

Digital divide: Not everyone has equal access to technology and the Internet, which can create a digital divide and limit access to media for some individuals.

NGN

- **Next Generation Network (NGN)** refers to a packet-based network and it can be used for both telecommunication services as well as data and it supports mobility.
- NGNs are based on a packet-switched architecture, which means that data is transmitted in small packets that are routed through the network to their destination.
- This allows NGNs to support a wide range of services, including voice, data, and video, and to be more flexible and efficient than traditional circuit-switched networks.
- NGNs are often used to support the convergence of media, as they can transmit multiple types of communication over a single network.

Architecture of NGN



- Next-Generation Network (NGN) architecture refers to the overall design and structure of an NGN, which is a type of telecommunications network that is designed to support a variety of communication services, including traditional circuit-switched services, as well as newer packet-based services such as Internet Protocol (IP) telephony and IPTV.
- NGN architecture is typically based on a packet-switched architecture, which means that data is transmitted in small packets that are routed through the network to their destination.
- This allows NGNs to be more flexible and efficient than traditional circuit-switched networks, as packets can be routed through the network based on their destination rather than being transmitted along a dedicated circuit.
- NGN architecture is also designed to be scalable and modular, allowing new services and technologies to be easily integrated into the network.
- It typically consists of three main components: a core network, which is responsible for routing and switching packets; an access network, which connects users to the core network; and service platforms, which provide various communication services to users.

- Overall, the goal of NGN architecture is to create a seamless communication experience for users, allowing them to access a wide range of services from a single device, such as a smartphone or a computer.

Services supported in NGN:

- Data communication services.
- Multimedia Services.
- Public interest services.
- Public Switched Telephone Network (PSTN)/Integrated Services Digital Network (ISDN) simulation services.
- Public Switched Telephone Network (PSTN)/ Integrated Services Digital Network (ISDN) emulation services

There are several advantages to using a Next-Generation Network (NGN):

1. **Flexibility:** NGNs are based on a packet-switched architecture, which allows them to be more flexible and efficient than traditional circuit-switched networks. They can support a wide range of services, including voice, data, and video, and can easily be adapted to support new technologies and services.
2. **Scalability:** NGNs are designed to be scalable, allowing them to easily expand to meet the growing demand for communication services.
3. **Cost-effectiveness:** NGNs can be more cost-effective than traditional circuit-switched networks, as they can support multiple services over a single network and do not require dedicated circuits for each service.
4. **Quality of service:** NGNs can offer improved quality of service compared to traditional circuit-switched networks, as they can prioritize different types of traffic and adjust the routing of packets based on the needs of the service being provided.

However, there are also some potential disadvantages to using an NGN:

1. **Complexity:** NGNs can be complex to implement and maintain, as they involve multiple components and technologies.
2. **Interoperability issues:** NGNs may not be compatible with older technologies and systems, which can create interoperability issues.
3. **Dependence on technology:** NGNs rely on technology, which can make them vulnerable to disruptions or outages.

4. **Upfront costs:** The implementation of an NGN can be expensive, as it requires the deployment of new infrastructure and the integration of multiple technologies.

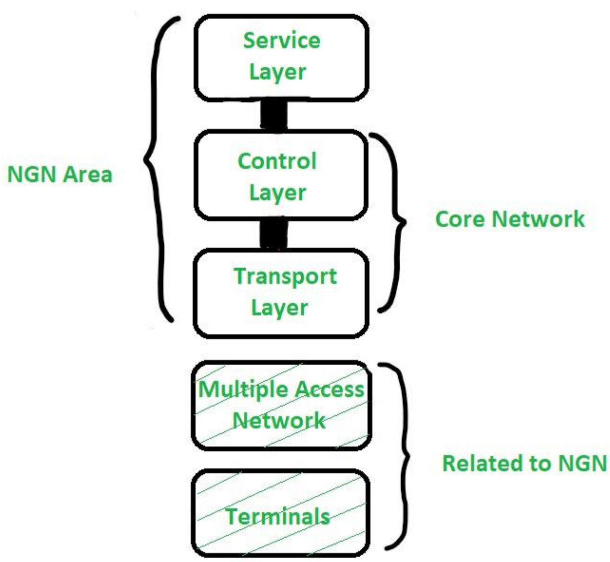
Next Generation Network Layer:

- In Access Layer, different types of media gateways that support connection to and from access network with the core network are included.

- Core network layer is network handling converged services based on Internet Protocol (IP). Control layer works as call server. It provides call control functions also provides control of a media gateway.

- Service layer is an IT platform that creates a service creation environment extending its functionality in order to cover new network scenarios as an intelligent network.

The below figure illustrates **Next Generation Network (NGN) Layers** as follows.



Next Generation Network (NGN) Layer

Scope of NGN:

1. Support for multiple services: NGNs can support a variety of communication services, including voice, data, and video, over a single network.

2. Packet-switched architecture: NGNs are based on a packet-switched architecture, which allows them to transmit data in small packets that are routed through the network to their destination. This makes them more efficient and flexible than traditional circuit-switched networks.

3. Quality of service: NGNs can offer improved quality of service compared to traditional networks, as they can prioritize different types of traffic and adjust the routing of packets based on the needs of the service being provided.

4. Scalability: NGNs are designed to be scalable, allowing them to easily expand to meet the growing demand for communication services.

5. Support for new technologies: NGNs are designed to be modular and flexible, allowing new technologies and services to be easily integrated into the network.

Challenges of NGN

1. **Interoperability:** Ensuring that different types of equipment and devices can work together seamlessly can be a challenge.

2. **Security:** NGN networks have a large attack surface, which makes them vulnerable to cyber threats. Protecting against these threats is critical.

3. **Quality of Service (QoS):** NGN networks must be able to provide a high level of QoS to ensure that different types of traffic, such as voice and video, are delivered with minimal delays and disruptions.

4. **Scalability:** NGN networks must be able to scale up or down as needed to accommodate changes in demand.

5. **Regulation:** NGN networks are subject to a range of regulatory requirements, which can be challenging to navigate.

6. **Cost:** Building and maintaining NGN networks can be expensive, which can be a challenge for service providers.

Real Time traffic using packet

- The real-time traffic, such as audio or video, is captured by a device, such as a microphone or camera.
- The traffic is divided into small chunks of data, which are then packaged into packets.
- The packets are transmitted over the network using a protocol, such as UDP, which is optimized for real-time traffic.
- The packets are received by the destination device, which reassembles the packets into their original form.
- The real-time traffic is played back by the destination device.

Packet Length

- In telecommunications, the packet length refers to the size of the data packets that are transmitted over a network.
- Packet length is typically measured in bytes, and it can vary depending on the type of data being transmitted and the protocol being used.
- For example, in some protocols, the packet length is fixed and cannot be changed.
- In other protocols, the packet length may be variable, meaning that it can be adjusted depending on the amount of data being transmitted.
- The packet length can have an impact on the performance of a network, as smaller packets may be transmitted more quickly, but they may also require more overhead (e.g., more packets may be needed to transmit the same amount of data).
- On the other hand, larger packets may be transmitted more slowly, but they may require less overhead.
- It is important to choose an appropriate packet length for a given network, as the packet length can affect the efficiency and performance of the network.

Packet Length:

- In telecommunications, packet length refers to the size of a packet of data that is transmitted over a network.
- Packet length is typically measured in bits or bytes and can vary depending on the type of network and the protocols being used.
- In general, the packet length is determined by the amount of data that needs to be transmitted and the size of the packets that can be handled by the network.
- The packet length can also be influenced by factors such as the type of network, the bandwidth available, and the latency of the network.

Buffering:

- Buffering is a process that helps to smooth out variations in the rate at which data is received.
- When data is received more quickly than it can be processed, it is temporarily stored in a buffer until it can be used.

- This helps to ensure that the data can be used at a consistent rate, even if the rate at which it is received is not consistent.

Types of Buffering:

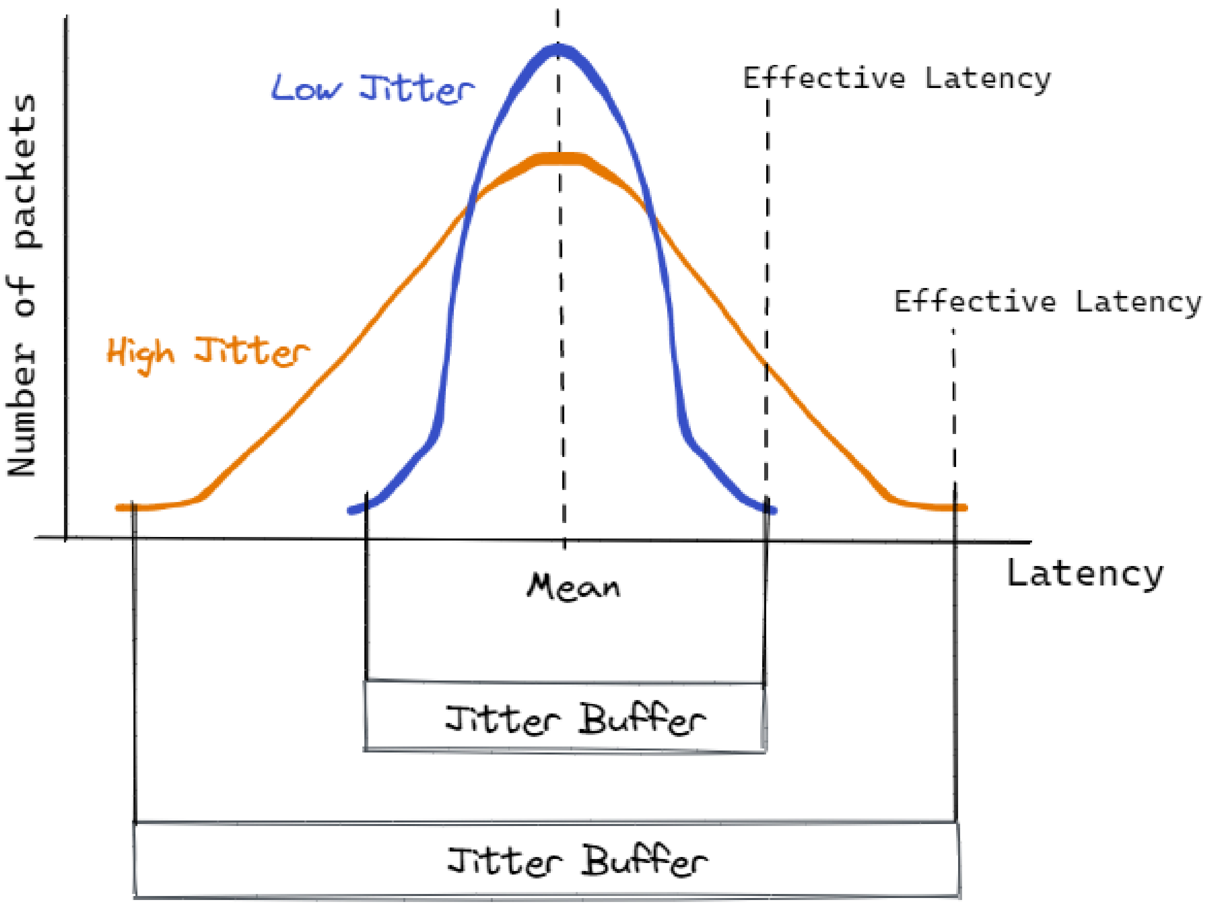
1. **Zero Capacity –**
   This queue cannot keep any message waiting in it. Thus it has maximum length 0. For this, a sending process must be blocked until the receiving process receives the message. It is also known as no buffering.
2. **Bounded Capacity –**
   This queue has finite length n. Thus, it can have n messages waiting in it. If the queue is not full, new message can be placed in the queue, and a sending process is not blocked. It is also known as automatic buffering.
3. **Unbounded Capacity –**
   This queue has infinite length. Thus, any number of messages can wait in it. In such a system, a sending process is never blocked.

Needs of Buffering:

- It helps in matching speed between two devices, between which the data is transmitted. For example, a hard disk has to store the file received from the modem.
- It helps the devices with different data transfer size to get adapted to each other.
- It helps devices to manipulate data before sending or receiving.
- It also supports copy semantics.

Jitter:

- Jitter refers to the variation in the delay of packet transmissions.
- When there is a lot of jitters, the packets may arrive at irregular intervals rather than at a consistent rate.
- This can cause problems for real-time applications such as video conferencing, where packets arriving at irregular intervals can lead to choppy or distorted audio and video.



Jitter causes:

1. Congestion: When the network is congested with a lot of traffic, packets may be delayed or lost, leading to jitter.
2. Distance: The distance between the devices communicating over the network can also contribute to jitter. Longer distances can result in more delay, which can cause jitter.
3. Interference: Interference from other devices or sources can disrupt the transmission of packets and cause jitter.
4. Hardware: The quality of the hardware being used on the network can also affect jitter. If the hardware is of poor quality or not functioning properly, it can lead to jitter.
5. Network protocols: The protocols being used on the network can also affect jitter. If the protocols are not efficient or are misconfigured, it can lead to jitter.
6. Power supply: The power supply to the devices on the network can also be a factor in jitter. If the power supply is unstable, it can cause problems with the transmission of packets and lead to jitter.

Ways to Reduce the Jitter:

1. Use Quality of Service (QoS) protocols: QoS protocols allow you to prioritize certain types of traffic on the network, which can help to reduce jitter for real-time applications such as voice and video.
2. Use a network with low latency: Low latency networks, such as fiber optic networks, can help to reduce jitter because the packets are transmitted more quickly.
3. Upgrade hardware: Upgrading to higher quality hardware can help to reduce jitter because the hardware is more reliable and can handle the transmission of packets more efficiently.
4. Use a network with a high bandwidth: A high bandwidth network allows for more data to be transmitted at once, which can help to reduce jitter.
5. Use error correction: Error correction techniques can help to reduce jitter by identifying and correcting errors in the transmission of packets.
6. Monitor the network: Monitoring the network can help you to identify problems that may be causing jitter and take steps to fix them.

Delay

- The delays, here, means the time for which the processing of a particular packet takes place. We have the following types of delays in computer networks:

1. **Transmission Delay:**

- The time taken to transmit a packet from the host to the transmission medium is called Transmission delay.

- For example, if bandwidth is 1 bps (every second 1 bit can be transmitted onto the transmission medium) and data size is 20 bits then what is the transmission delay? If in one second, 1 bit can be transmitted. To transmit 20 bits, 20 seconds would be required.
- Let B bps is the bandwidth and L bit is the size of the data then transmission delay is,
- Tt = L/B

This delay depends upon the following factors:

- If there are multiple active sessions, the delay will become significant.
- Increasing bandwidth decreases transmission delay.
- MAC protocol largely influences the delay if the link is shared among multiple devices.
- Sending and receiving a packet involves a context switch in the operating system, which takes a finite time.

2. Propagation delay:

- After the packet is transmitted to the transmission medium, it has to go through the medium to reach the destination.
- Hence the time taken by the last bit of the packet to reach the destination is called propagation delay.

Factors affecting propagation delay:

1. **Distance –** It takes more time to reach the destination if the distance of the medium is longer.
2. **Velocity –** If the velocity(speed) of the signal is higher, the packet will be received faster.

Tp = Distance / Velocity

Note:
Velocity =3 X 10^8 m/s (for air)

Velocity= 2.1 X 10^8 m/s (for optical fiber)

3. **Queueing delay:**

- Let the packet is received by the destination, the packet will not be processed by the destination immediately.
- It has to wait in a queue in something called a buffer.
- So the amount of time it waits in queue before being processed is called queueing delay.

In general, we can’t calculate queueing delay because we don’t have any formula for that.

This delay depends upon the following factors:

- If the size of the queue is large, the queuing delay will be huge. If the queue is empty there will be less or no delay.
- If more packets are arriving in a short or no time interval, queuing delay will be large.
- The less the number of servers/links, the greater is the queuing delay.

4. **Processing delay:**

- Now the packet will be taken for the processing which is called processing delay.
- Time is taken to process the data packet by the processor that is the time required by intermediate routers to decide where to forward the packet, update TTL, perform header checksum calculations.
- It also doesn’t have any formula since it depends upon the speed of the processor and the speed of the processor varies from computer to computer.

Quality of Service (QOS)

- Quality of service (QoS) refers to any technology that manages data traffic to reduce packet loss, latency and jitter on a network.
- QoS controls and manages network resources by setting priorities for specific types of data on the network.

QoS parameters

Organizations can measure QoS quantitatively by using several parameters, including the following:

- **Packet loss.** This happens when network links become congested, and routers and switches start dropping packets. When packets are dropped during real-time communication, such as in voice or video calls, these sessions can experience jitter and gaps in speech. Packets can be dropped when a queue, or line of packets waiting to be sent, overflows.
- **Jitter.** This is the result of network congestion, timing drift and route changes. Too much jitter can degrade the quality of voice and video communication.
- **Latency.** This the time it takes a packet to travel from its source to its destination. Latency should be as close to zero as possible. If a voice over IP call has a high amount of latency, users can experience echo and overlapping audio.
- **Bandwidth.** This is the capacity of a network communications link to transmit the maximum amount of data from one point to another in a given amount of time. QoS optimizes the network performance by managing bandwidth and giving high priority applications with stricter performance requirements more resources than others.
- **Mean opinion score (MOS).** This is a metric to rate voice quality that uses a five-point scale, with a five indicating the highest quality.

Importance Of QOS:

Improved performance: QoS can help to improve the performance of critical applications by prioritizing their traffic and allocating the necessary resources.

Better user experience: By ensuring that real-time applications such as voice and video receive a higher level of performance, QoS can help to improve the user experience.

Increased efficiency: By optimizing the use of network resources, QoS can help to increase the overall efficiency of the network.

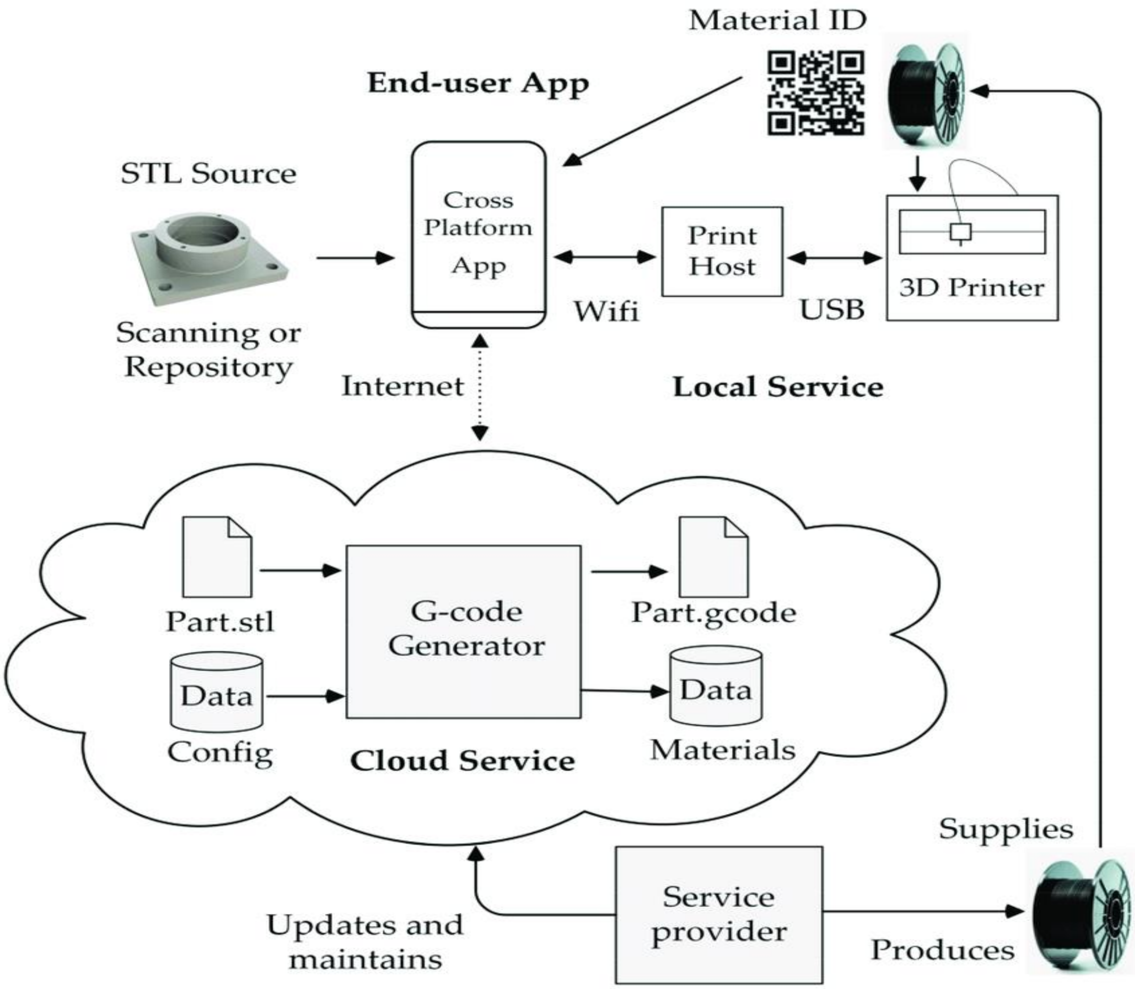
Better utilization of resources: QoS can help to ensure that resources such as bandwidth are used more efficiently, which can help to reduce costs.

Improved reliability: By prioritizing certain types of traffic, QoS can help to improve the reliability of the network and reduce the risk of disruptions.

Enhanced security: QoS can be used to implement security measures such as firewall rules, which can help to improve the overall security of the network.

Integrated Service Architecture:

- Integrated service architecture refers to the design of a system that integrates multiple services or components in order to support a particular function or set of functions.
- This can include both hardware and software components, and may involve the integration of services from multiple vendors or providers.
- Integrated service architectures are often used in large, complex systems that require the coordination of multiple functions in order to deliver a desired service or capability.
- They can provide a number of benefits, such as increased efficiency, reduced costs, and improved reliability, but can also be challenging to design and implement due to the need to coordinate and integrate multiple components and services.



Advantages of ISA:

1. Improved efficiency: An integrated service architecture can help to streamline processes and eliminate redundancies, which can lead to increased efficiency and productivity.
2. Reduced costs: By integrating multiple services or components into a single system, it is often possible to reduce costs through economies of scale and the elimination of duplicate infrastructure.
3. Improved reliability: An integrated service architecture can help to improve the reliability of a system by providing redundant components and failover capabilities.
4. Increased flexibility: An integrated service architecture can make it easier to add or modify components and services, allowing the system to adapt and evolve over time.

5. Enhanced security: An integrated service architecture can help to improve security by centralizing control and monitoring of access to resources and data.

Disadvantages of ISA:

1. Complexity: An integrated service architecture can be complex to design and implement, particularly in systems that involve the integration of multiple vendors or services.
2. Dependency issues: If one component or service in an integrated system fails, it can affect the entire system. This can create dependency issues and make the system more vulnerable to disruptions.
3. Upgrade challenges: Upgrading or modifying components in an integrated service architecture can be challenging, as it may require changes to multiple parts of the system.
4. Lack of standardization: If an integrated service architecture involves the integration of components from multiple vendors or providers, it may be difficult to ensure that all components are fully compatible and work together seamlessly.
5. Cost: Implementing an integrated service architecture can be expensive, particularly if it requires the development of custom components or integration with proprietary systems.

Differentiated Service Architecture:

- Differentiated services refer to a multiple service model that can satisfy many requirements.

- In other words, it supports multiple mission-critical applications. Moreover, these services help to minimize the burden of the network devices and also support the scaling of the network. Some major differentiated services are as follows.

Traffic conditioning – Ensures that the traffic entering the DiffServ domain.

Packet classification – Categorizes the packet within a specific group using the traffic descriptor.

Packet marking – Classify a packet based on a specific traffic descriptor.

Congestion Management – Achieve queuing and traffic scheduling.

Congestion avoidance – Monitor traffic loads to minimize congestion. In involves packet dropping.

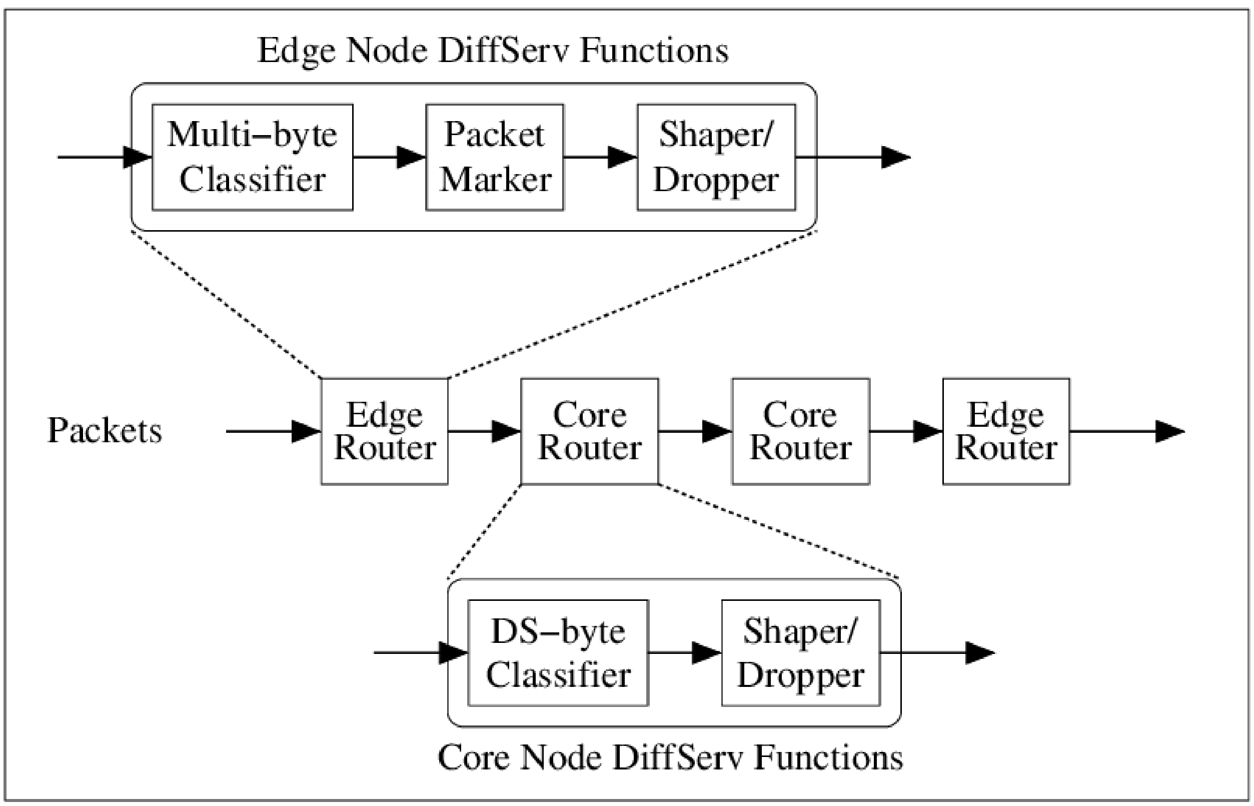
Advantages of DiffServ Architecture

1. Improved quality of service: DiffServ allows network administrators to prioritize different types of traffic, ensuring that critical traffic such as voice and video is given priority over non-critical traffic. This can improve the quality of service for users of real-time applications.
2. Better resource utilization: DiffServ allows administrators to allocate network resources more efficiently by giving priority to the traffic that needs it most. This can help to prevent congestion and improve overall network performance.
3. Flexibility: DiffServ allows administrators to create custom policies for different types of traffic, enabling them to tailor the service levels to the specific needs of their network.
4. Scalability: DiffServ can be implemented at different levels of the network, from the edge to the core, allowing it to scale with the needs of the network.
5. Interoperability: DiffServ is a standard-based approach that is supported by a wide range of networking equipment, making it easy to integrate into existing networks.
6. Cost-effective: Because DiffServ allows administrators to allocate resources more efficiently, it can help to reduce the overall cost of operating a network.

Disadvantages of DiffServ Architecture:

1. Complexity: Implementing differentiated service architecture can be complex, especially in large networks with many different types of traffic. It requires careful planning and configuration to ensure that traffic is properly classified and prioritized.

2. Limited scalability: Differentiated service architecture relies on a limited set of service levels, which can make it difficult to accommodate new types of traffic or services that do not fit into the existing service levels.
3. Potential for unfairness: If some traffic is given higher priority than others, it could lead to unfairness for traffic that is assigned to lower priority levels. This could be especially problematic if the traffic being given lower priority is critical for certain applications or services.
4. Dependence on network configuration: The effectiveness of differentiated service architecture depends on the network being configured properly. If the network is not configured correctly, it could lead to poor performance or even failures.



INTEGRATED SERVICES
VERSUS
DIFFERENTIATED SERVICES

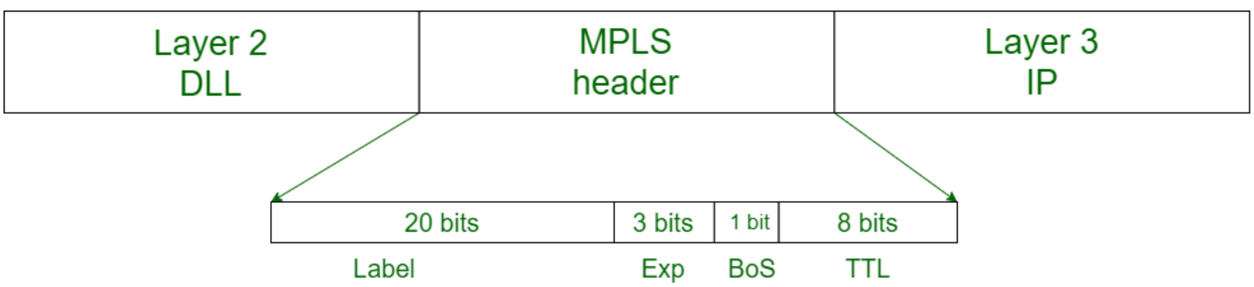
INTEGRATED SERVICES	DIFFERENTIATED SERVICES
Architecture that specifies the elements to guarantee Quality of Service (QoS) on network	Architecture that specifies a simple and scalable mechanism for classifying and managing network traffic and providing QoS on modern IP networks
Involve prior reservation of resources before sending to achieve the required Quality of Service	Mark the packets with priority and send it to the network and do not require prior reservation
Also called IntServ	Also called DiffSer
Not scalable	Scalable
Involve per flow setup	Involve long term setup
Involve end to end service scope	Involve domain service scope

Multi-Protocol Label Switching (MPLS)

- **Multi-Protocol Label Switching (MPLS)** is an IP packet routing technique that routes IP packet through paths via labels instead of looking at complex routing tables of routers.
- This feature helps in increasing the delivery rate of IP packets.

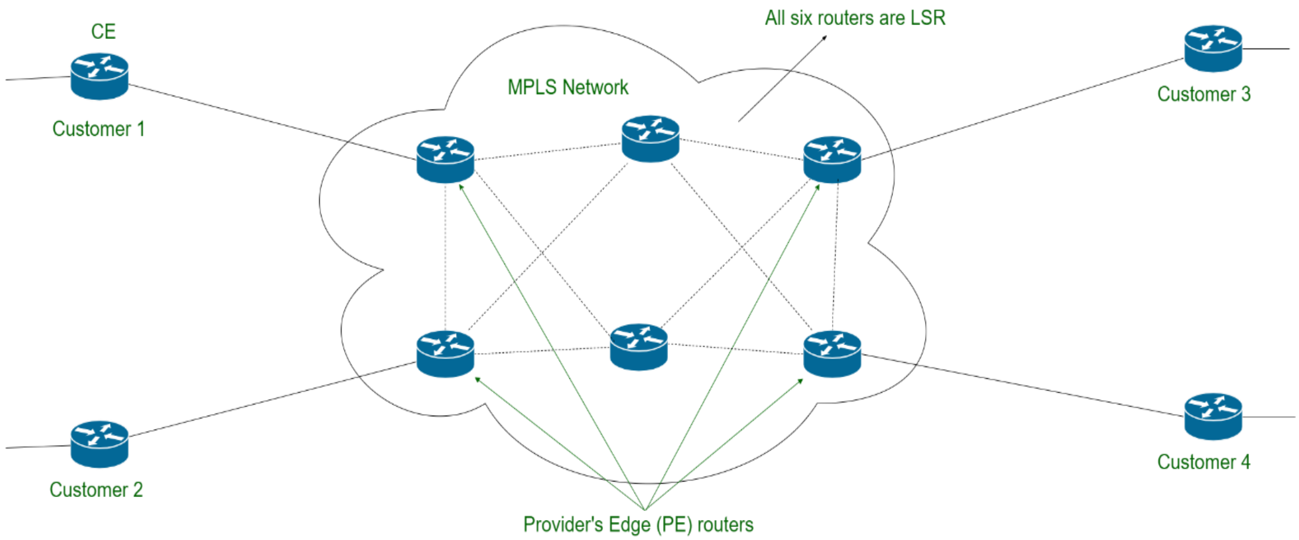
MPLS Header –
The MPLS Header is 32 bit long and is divided into four parts –

1. **Label –** This field is 20 bit long and can take value b/w 0 & $2^{20} - 1$.
2. **Exp –** They are 3 bits long and used for *Quality of Service (QoS)*.
3. **Bottom of stack (S) –** It is of size 1 bit. MPLS labels are stacked one over other. If there is only one label remained in MPLS header, then its value is 1 otherwise 0.
4. **Time to Live (TTL) –** It is 8 bit long and its value is decreased by one at each hop to prevent packet to get stuck in network.



Working of MPLS

- In a MPLS network, data packets are assigned labels when they enter the network.
- These labels are used to forward the packets to their destination, rather than using the traditional method of looking up the destination IP address in a routing table.
- This allows MPLS networks to operate more efficiently and at a higher speed, as the labels can be used to route the packets along predetermined paths.
- MPLS networks can also be used to prioritize certain types of traffic, such as real-time voice or video, by assigning them higher priority labels.



Advantages of MPLS

- Improved Network Utilization
- Consistent Network Performance.
- Obscures Network Complexity.
- Easier Global Changes.
- Reduced Network Congestion.
- Increased Uptime.
- Scalable IP VPNs.

Disadvantages of MPLS

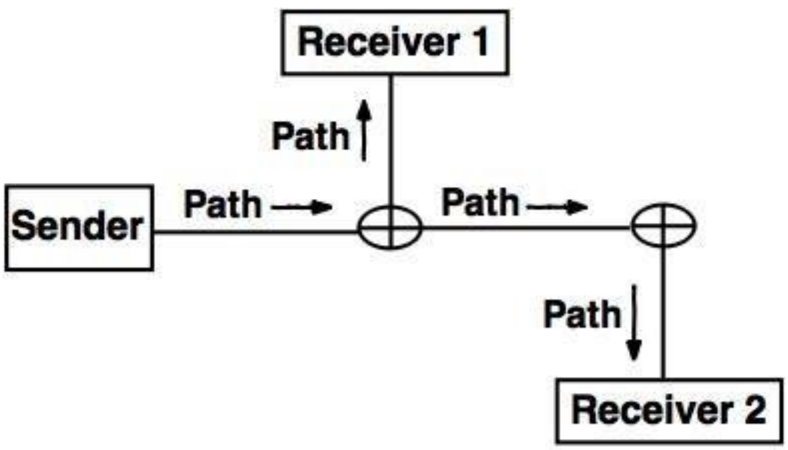
- High cost
- Limited compatibility
- Limited flexibility
- Complexity
- Dependence on a single service provider

Resources Reservation Protocols:

- **Resource Reservation Protocol (RSVP)** is used in real-time systems for an efficient quality band transmission to a particular receiver.
- It is generally used by the receiver side for the fast delivery of the transmission packets from the sender to the receiver.
- Resource Reservation Protocol (RSVP) is a transport layer protocol used to reserve network resources and enable running Internet applications to gain quality of service (QoS).
- The two important types of RSVP messages are:

1. Path messages:

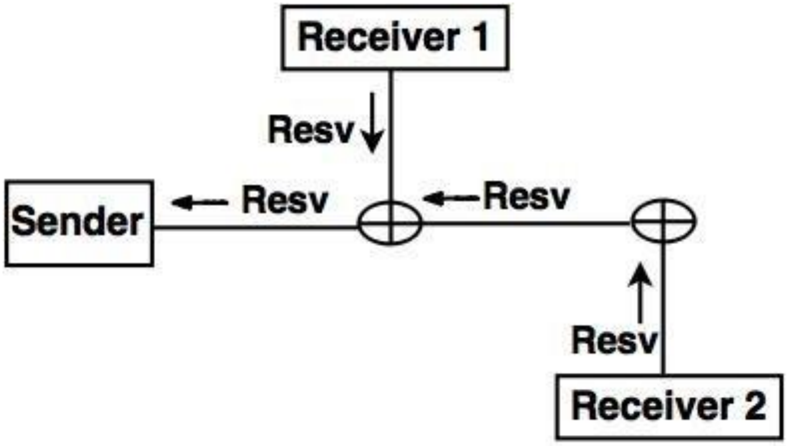
- The receivers in a flow make the reservation in RSVP, but the receivers do not know the path traveled by the packets before the reservation.
- The path is required for reservation To solve this problem the RSVP uses the path messages.
- A path message travels from the sender and reaches to all receivers by multi-casting and path message stores the necessary information for the receivers.



Path messages

2. Resv messages:

After receiving path message, the receiver sends a Resv message. The Resv message travels to the sender and makes a resource reservation on the routers which supports for RSVP.



Resv Messages

Working of RSVP

- Let's assume that a particular video program is to be multicast at a certain time on Monday evening. Expecting to receive it, you send an RSVP request before the broadcast (you'll need a special client program or perhaps your browser includes one) to allocate sufficient bandwidth and priority of packet scheduling for the program. This request will go to your nearest Internet gateway with an RSVP server.
- It will determine whether you are eligible to have such a reservation set up and, if so, whether sufficient bandwidth remains to be reserved to you without affecting earlier reservations. Assuming you can make the reservation and it is entered, the gateway then forwards your reservation to the next gateway toward the destination (or source of multicast). In this manner, your reservation is ensured all the way to the destination. (If the reservation can't be made all the way to the destination, all reservations are removed.)

Advantages of RSVP:

1. Quality of service: RSVP allows users to specify the QoS they require for their data streams, which can be useful for applications that require a certain level of performance (e.g., real-time audio or video).
2. Efficient use of resources: RSVP helps ensure that the network resources are used efficiently, by allowing users to reserve the resources they need in advance and releasing them when they are no longer needed.
3. Scalability: RSVP is designed to scale to large networks, making it suitable for use in enterprise and Internet environments.

4. Integration with other protocols: RSVP can be used in conjunction with other protocols, such as Multiprotocol Label Switching (MPLS), to provide a more comprehensive approach to QoS management.

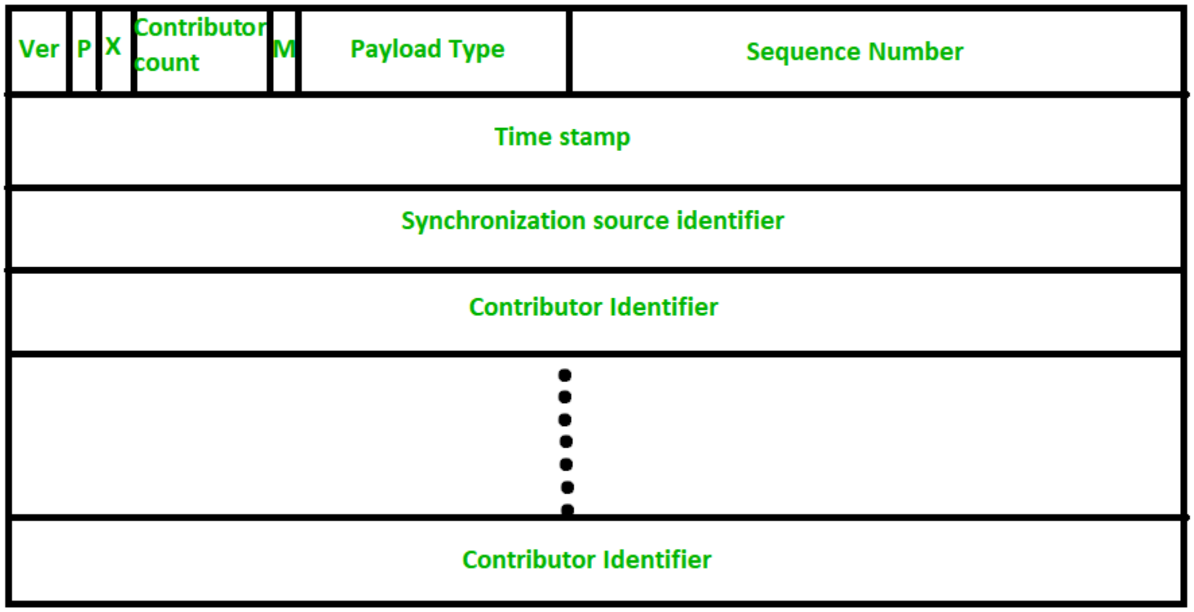
Disadvantages

1. Complexity: RSVP can be complex to implement and manage, particularly in large or highly dynamic networks. This can make it challenging for network administrators to set up and maintain.
2. Overhead: RSVP can add overhead to the network, as it requires additional signaling between nodes to set up and maintain reservations. This can impact the performance of the network, especially in environments where there are many reservations being made.
3. Limited support: RSVP may not be supported by all network devices or applications, which can limit its usefulness in some environments.
4. Security concerns: RSVP relies on trust between the sender and receiver, which can be a potential security vulnerability if the sender is not authentic or is requesting more resources than they are entitled to.

Real Time Transport Protocol:

- A protocol is designed to handle real-time traffic (like audio and video) of the Internet, is known as **Real Time Transport Protocol (RTP)**.
- RTP must be used with UDP.
- It does not have any delivery mechanism like multicasting or port numbers.
- RTP supports different formats of files like MPEG and MJPEG.
- It is very sensitive to packet delays and less sensitive to packet loss.

RTP Header Format: The diagram of header format of RTP packet is shown below:



The header format of RTP is very simple and it covers all real-time applications. The explanation of each field of header format is given below:

Version: This 2-bit field defines version number. The current version is 2.

1. **P –** The length of this field is 1-bit. If value is 1, then it denotes presence of padding at end of packet and if value is 0, then there is no padding.
2. **X –** The length of this field is also 1-bit. If value of this field is set to 1, then it indicates an extra extension header between data and basic header and if value is 0 then, there is no extra extension.
3. **Contributor count –** This 4-bit field indicates number of contributors. Here maximum possible number of contributors is 15 as a 4-bit field can allows number from 0 to 15.
4. **M –** The length of this field is 1-bit and it is used as end marker by application to indicate end of its data.
5. **Payload types –** This field is of length 7-bit to indicate type of payload. We list applications of some common types of payloads.
6. **Sequence Number –** The length of this field is 16 bits. It is used to give serial numbers to RTP packets. It helps in sequencing. The sequence number for first packet is given a random number and then every next packet's sequence number is incremented by 1. This field mainly helps in checking lost packets and order mismatch.
7. **Time Stamp –** The length of this field is 32-bit. It is used to find relationship between times of different RTP packets. The timestamp for

first packet is given randomly and then time stamp for next packets given by sum of previous timestamp and time taken to produce first byte of current packet. The value of 1 clock tick is varying from application to application.

8. **Synchronization Source Identifier –** This is a 32-bit field used to identify and define the source. The value for this source identifier is a random number that is chosen by source itself. This mainly helps in solving conflict arises when two sources started with the same sequencing number.
9. **Contributor Identifier –** This is also a 32-bit field used for source identification where there is more than one source present in session. The mixer source use Synchronization source identifier and other remaining sources (maximum 15) use Contributor identifier.

Advantages of RTP

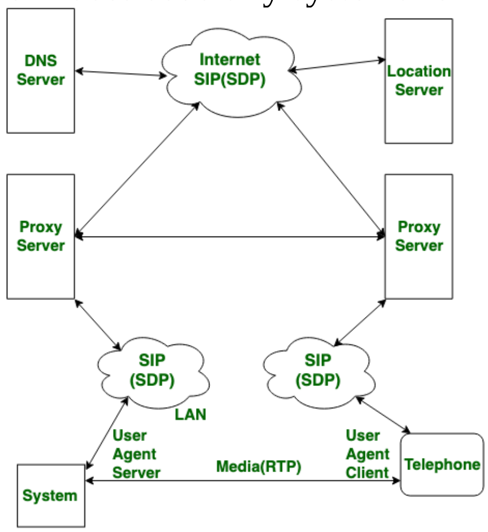
- Generally low latency.
- Packets are sequence-numbered and timestamped for reassembly if they arrive out of order. This lets data sent using RTP be delivered on transports that don't guarantee ordering or even guarantee delivery at all.
- This means RTP can be — but is not required to be — used atop UDP for its performance as well as its multiplexing and checksum features.
- RTP supports multicast;
- RTP isn't limited to use in audiovisual communication.

Disadvantages of RTP:

1. Limited security
2. No congestion controls
3. Limited error recovery
4. Limited support for interoperation

Session Initiation Protocol:

- Session Initiation Protocol (SIP) is a signaling protocol used for initiating, maintaining, modifying and terminating real-time sessions that involve video, voice, messaging and other communications between two or more endpoints on the Internet.
- SIP is used to establish, maintain and terminate voice and video calls, as well as other types of communications sessions such as video conferencing and instant messaging.
- SIP is designed to be independent of the underlying transport layer and can be used with a variety of transport protocols, including UDP, TCP and TLS.
- SIP is also designed to be independent of the underlying media, which means that it can be used to transmit a variety of media types, such as audio, video, and text.
- SIP is widely used in Voice over IP (VoIP) systems and is an important part of many modern communication systems, including Internet telephony, instant messaging, and web conferencing.
- SIP can handle all types of media.   .
- SIP is more flexible.
- SIP is not depending any other device.
- It is protocol of application layer.
- It is related with signaling protocol to manipulate the technologies.
- It is compatible with internet.
- SIP uses severally systems for handling completely different operation

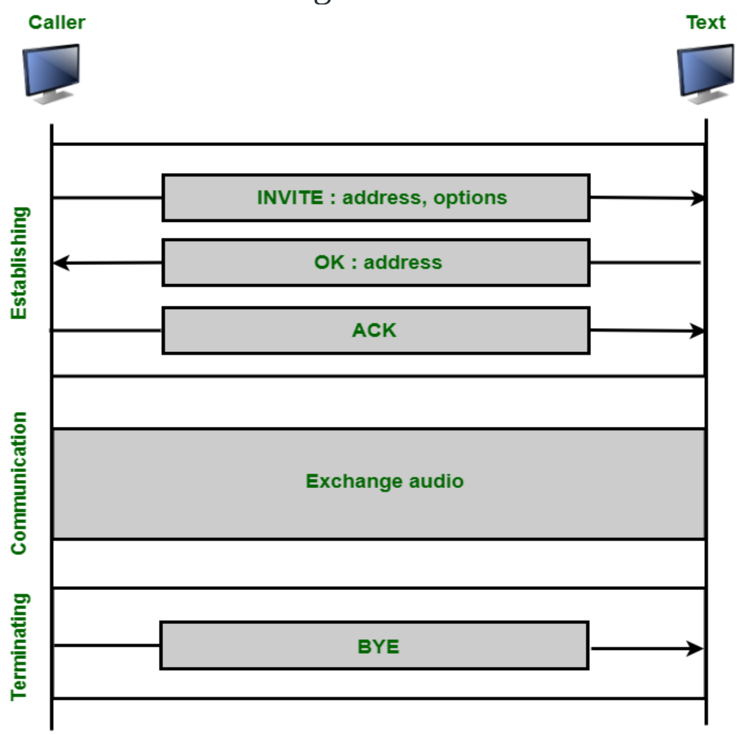


SIP Architecture Model

SIP Session:

A simple session using SIP consists of the following.

1. Establishing a session.
2. Communication.
3. Terminating the session



SIP Session

Establishing a session:

It requires a three-way handshake.

- The caller will send INVITE message.
- If the caller is willing to start out, he/she sends a reply message to verify that a reply code is received, the caller send an ACK message.

Communication:

- After establishment of session, the caller and callee communicate using two temporary ports.

Terminating the session:

- The session can often be terminated by using BYE message send by either caller or callee.

Differences between NGN vs traditional circuit Network:

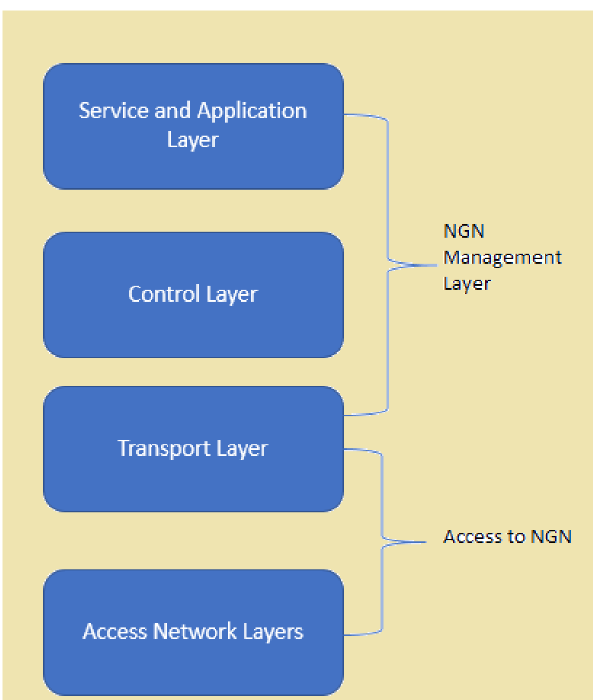
Difference between Next Generation Network and Traditional Network :

NGN	Traditional
IP Based	TDM Based
Use of different signaling and protocols to support multiple services	Use of standard SS7 signaling
Cost Effective	High initial deployment and expansion costs
Packet/label switched - Bandwidth acquires & releases it as it is needed	Circuit switched - It reserves the required bandwidth in advance
Resources not consumed when idle	Path reserved for the duration of the call
Dynamic policy based traffic routing.	Pre-determined routing of calls based on dialed numbers
Multiple Services, One Platform (IP) Approach	Different Services, Different Platform (voice, data or video)
Flat network - It simplifies management, operation and maintenance of the network.	Hierarchical design
Designed for bursty data transmission	Poorly matched for bursty data transmission
Supports variable information transfer rates (voice, data, and video)	Non-variable information transfer rates
Distributed Switch Functions with standard open interfaces	Switch function in a single box.
Allows choices of network elements from multiple vendors	switching vendors based upon feature availability and overall performance
Services and applications can be implemented and customized by vendors or third party developers	Services and features depend on vendor implementation

Driving Factors of NGN:

1. Convergence: NGN allows different types of communication services to be provided over a single network, which can reduce the complexity and cost of deploying and maintaining multiple networks.
2. Interoperability: NGN uses standardized protocols, such as IP, that are designed to be interoperable with other systems and devices. This makes it easier for different vendors' equipment to work together and enables new services to be developed more quickly.
3. Scalability: NGN networks are designed to be scalable, which means that they can be easily expanded to support more users and a wider range of services.
4. Flexibility: NGN networks are designed to be flexible and can be customized to meet the specific needs of different users and applications.
5. Efficiency: NGN networks are designed to be more efficient in terms of bandwidth utilization, which can reduce the cost of providing services and improve the overall quality of service.
6. Mobility: NGN networks support mobility, which means that users can access the network and its services from anywhere, using a variety of devices.

NGN layer Architecture



Access layer: This layer consists of the physical infrastructure that connects users to the NGN, such as cables, switches, and routers.

Network layer: This layer consists of the core network infrastructure, including routers and switches, that is responsible for routing data packets between different parts of the network.

Service layer: This layer consists of the applications and services that are provided to users, such as voice, video, and data services.

Management layer: This layer consists of the systems and tools that are used to manage and monitor the operation of the NGN, including network monitoring, performance management, and security management.

Applications layer: This layer consists of the applications that run on top of the NGN, such as video conferencing, VoIP, and instant messaging.

End-user devices: This layer consists of the devices that are used by users to access the NGN, such as phones, computers, and tablets.

Why MPLS is called 2.5-layer protocol?

- MPLS is a layer 2.5 protocol because it combines some of the functions of both the data link layer and the network layer, allowing it to provide both layer 2 switching and layer 3 routing capabilities.

- Because MPLS is able to work at both the data link and network layers, it is able to provide a wide range of networking services, including traffic engineering, virtual private networks (VPNs), and quality of service (QoS).

MPLS Features:

Basic Multiprotocol Label Switching (MPLS) Features

- MPLS reduces routing lookups.
- MPLS forwards packets based on labels.
- Labels usually correspond to IP destination networks (equal to traditional IP forwarding).
- Labels can also correspond to other parameters:
  - Layer 3 VPN destination
  - Layer 2 circuit
  - Outgoing interface on the egress router
  - QoS
  - Source address
- MPLS supports forwarding of all Layer 3 protocols, not just IP.

SHORT NOTES AND EXTRA SOLUTIONS

NGN BY ITU

- A Next Generation Networks (NGN) is a packet-based network able to provide Telecommunication Services to users and able to make use of multiple broadbands, QoS-enabled transport technologies and in which service-related functions are independent of the underlying transport-related technologies.
- It enables unfettered access for users to networks and to competing service providers and services of their choice.
- It supports generalized mobility which will allow consistent and ubiquitous provision of services to users.

Advantages of layered Communication Protocol:

1. Modularity: Each layer of a protocol can be developed and modified independently of the other layers, which makes it easier to update and improve the system.
2. Interoperability: Because each layer serves a specific purpose and follows a specific set of rules, different devices and systems can communicate with each other even if they use different hardware and software.
3. Ease of troubleshooting: If there is a problem with the communication, it is easier to identify which layer is causing the issue and fix it.
4. Extensibility: New functions can be added to the system by adding new layers or modifying existing ones, rather than having to redesign the entire system.
5. Reusability: The layers of a protocol can be used in a variety of different communication systems, which helps to reduce development time and costs.

Overall, the use of a layered communication protocol helps to make communication systems more reliable, flexible, and efficient.

How can we reduce the trunk group outlets and inlets of a switch to some reasonable amount by hierarchical network?

- One way to reduce the number of trunk group outlets and inlets in a switch is to use a hierarchical network design.

In a hierarchical network, the network is divided into smaller subnetworks, or subnets, each of which is connected to a central network device, such as a switch or router.

- This central device is often referred to as a "core" or "distribution" layer device.
- By dividing the network into smaller subnets and connecting them to a central device, you can reduce the number of outlets and inlets needed on the switch, as the central device can then handle the communication between the subnets.
- This can help to improve network performance and reduce the complexity of the network.
- To implement a hierarchical network, you can use a combination of switches, routers, and other networking devices to create the various layers of the network.
- You may also need to configure VLANs (Virtual LANs) to segment the network into the different subnets and ensure that communication is properly routed between them.

EPBAX:

- EPABX is an acronym of **"Electronic Private Automatic Branch Exchange"**.

- It is an electronic device that is used for voice communication in Industries, Hotels, offices, and many more places.

- This electronic device is independent, and we can use it without any service providers (or trunk lines).

- EPABX is a switching system that enables both internal and external stitching functions for any organization.

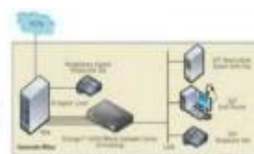
- In order to select an appropriate EPABX, one should have proper information about the traffic pattern in the office.

The EPABX system offers many features. Some are listed as follows -

- Transferring of calls
- Call picks up
- Conference
- Call back up, and many more.

FUNCTION OF THE EPABX SYSTEM

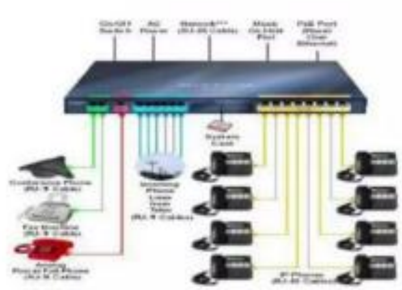
TYPES OF EPABX



1. **Analog PBX**

- "Phone-Box" system is used to connect different phone lines.
- Requires a person to operate the switchboard controlling the calls.
- Companies providing service – Nortel, Toshiba,

  -- Panasonic.



2. **Digital PBX:**

- Sound converted to digital signal and sent on a channel through circuit switching.
- It is the way contemporary phone companies adopt to connect to callers.
- Companies providing service – Uni-phone Telecommunications,

  -- Rexon Technology

  -- Panasonic.

ADVANTAGES

- Optimum utilization of resources.
- Easier in programming
- System can be expanded
- Occupies Small Space

DISADVANTAGES

- Required electricity
- Need careful handling
- Installed and fixed at one place
- Maintenance cost is higher

Explain the meaning in telecommunication of echo, singing, stored-program control conditional selection, charge advice and tandem exchange.

In telecommunications, the meanings of the terms you requested are as follows:

1. **Echo:**
    - In telecommunications, an echo is an unwanted reflection of a transmitted signal that returns to the sender, causing the sender to hear their own voice or transmission repeated back to them.
    - Echoes can be caused by a variety of factors, such as reflections off of walls or other objects, or by delays in the transmission path.

2. **Singing:**
    - In telecommunications, "singing" typically refers to the phenomenon of electrical or electronic devices generating a high-pitched, whine-like sound.
    - This can be caused by various factors, such as electrical interference, improper grounding, or faulty components.

3. **Stored-program control:**
    - In telecommunications, stored-program control refers to the use of a computer program or set of instructions stored in a device's memory to control the device's operations.
    - This can be used to control a variety of telecommunications equipment, such as telephone exchanges, switches, and routers.

4. **Conditional selection:**
    - In telecommunications, conditional selection refers to the use of control logic to make decisions about how to handle a given signal or transmission based on certain conditions.

- For example, a telecom system might use conditional selection to determine whether to route a call over a certain network or to a particular destination based on the phone number being called or the availability of certain network resources.

5. **Charge advice:**
    - In telecommunications, a charge advice is a message sent by a phone company or other service provider to inform a customer that they have been charged for a call or other service.
    - The charge advice typically includes details about the charge, such as the amount, the date and time of the call, and the phone number or service used.

6. **Tandem exchange:**
    - In telecommunications, a tandem exchange is a telephone exchange that is connected to two or more other telephone exchanges and is used to route calls between them.
    - Tandem exchanges are used to connect different phone networks and provide a means for calls to be transferred between them.

Needs For Telecom Traffic Dimensioning: