Computer Network

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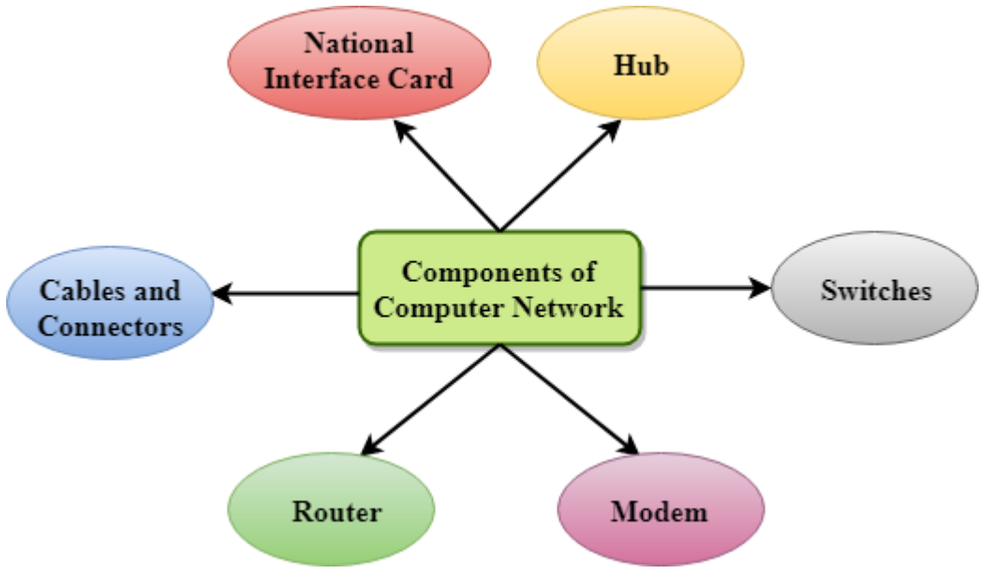
What is a Computer Network?

- **Computer Network** is a group of computers connected with each other through wires, optical fibres or optical links so that various devices can interact with each other through a network.
- The aim of the computer network is the sharing of resources among various devices.
- In the case of computer network technology, there are several types of networks that vary from simple to complex level.

Components Of Computer Network:



Major components of a computer network are:

NIC(National interface card)

NIC is a device that helps the computer to communicate with another device. The network interface card contains the hardware addresses, the data-link layer protocol use this address to identify the system on the network so that it transfers the data to the correct destination.

There are two types of NIC: wireless NIC and wired NIC.

- **Wireless NIC:** All the modern laptops use the wireless NIC. In Wireless NIC, a connection is made using the antenna that employs the **radio wave technology**.
- **Wired NIC:** Cables use the **wired NIC** to transfer the data over the medium.

Hub

Hub is a central device that splits the network connection into multiple devices. When computer requests for information from a computer, it sends the request to the Hub. Hub distributes this request to all the interconnected computers.

Switches

Switch is a networking device that groups all the devices over the network to transfer the data to another device. A switch is better than Hub as it does not broadcast the message over the network, i.e., it sends the message to the device for which it belongs to. Therefore, we can say that switch sends the message directly from source to the destination.

Cables and connectors

Cable is a transmission media that transmits the communication signals. **There are three types of cables:**

- **Twisted pair cable:** It is a high-speed cable that transmits the data over **1Gbps** or more.
- **Coaxial cable:** Coaxial cable resembles like a TV installation cable. Coaxial cable is more expensive than twisted pair cable, but it provides the high data transmission speed.
- **Fibre optic cable:** Fibre optic cable is a high-speed cable that transmits the data using light beams. It provides high data transmission speed as compared to other cables. It is more expensive as compared to other cables, so it is installed at the government level.

Router

Router is a device that connects the LAN to the internet. The router is mainly used to connect the distinct networks or connect the internet to multiple computers.

Modem

Modem connects the computer to the internet over the existing telephone line. A modem is not integrated with the computer motherboard. A modem is a separate part on the PC slot found on the motherboard.

Uses Of Computer Network

- **Resource sharing:** Resource sharing is the sharing of resources such as programs, printers, and data among the users on the network without the requirement of the physical location of the resource and user.
- **Server-Client model:** Computer networking is used in the **server-client model**. A server is a central computer used to store the information and maintained by the system administrator. Clients are the machines used to access the information stored in the server remotely.
- **Communication medium:** Computer network behaves as a communication medium among the users. For example, a company contains more than one computer has an email system which the employees use for daily communication.
- **E-commerce:** Computer network is also important in businesses. We can do the business over the internet. For example, amazon.com is doing their business over the internet, i.e., they are doing their business over the internet.

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Computer Network Tutorial

Computer Network tutorial provides basic and advanced concepts of Data Communication & Networks (DCN). Our Computer Networking Tutorial is designed for beginners and professionals.

Our Computer Network tutorial includes all topics of Computer Network such as introduction, features, types of computer network, architecture, hardware, software, internet, intranet, website, LAN, WAN, etc.

What is Computer Network?

A computer network is a set of devices connected through links. A node can be computer, printer, or any other device capable of sending or receiving the data. The links connecting the nodes are known as communication channels.

Computer Network uses distributed processing in which task is divided among several computers. Instead, a single computer handles an entire task, each separate computer handles a subset.

Following are the advantages of Distributed processing:

- **Security:** It provides limited interaction that a user can have with the entire system. For example, a bank allows the users to access their own accounts through an ATM without allowing them to access the bank's entire database.
- **Faster problem solving:** Multiple computers can solve the problem faster than a single machine working alone.
- **Security through redundancy:** Multiple computers running the same program at the same time can provide the security through redundancy. For example, if four computers run the same program and any computer has a hardware error, then other computers can override it.

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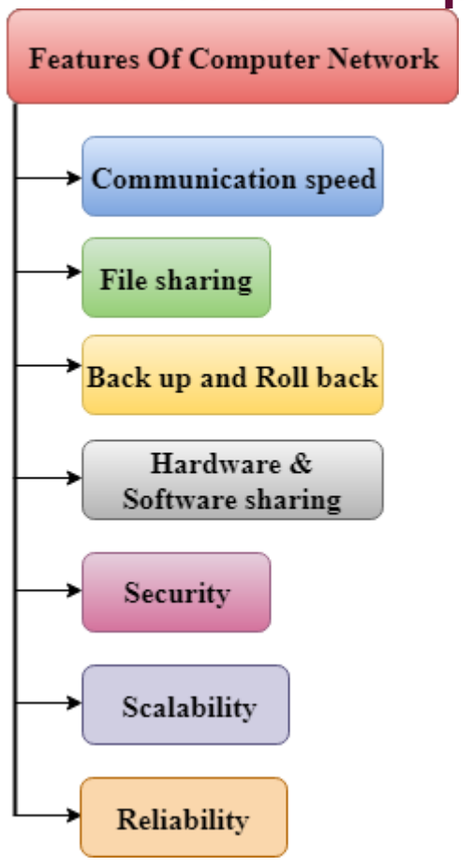
Features Of Computer network



A list Of Computer network features is given below.

- Communication speed
- File sharing
- Back up and Roll back is easy
- Software and Hardware sharing
- Security
- Scalability
- Reliability

Communication speed

Network provides us to communicate over the network in a fast and efficient manner. For example, we can do video conferencing, email messaging, etc. over the internet. Therefore, the computer network is a great way to share our knowledge and ideas.

File sharing

File sharing is one of the major advantage of the computer network. Computer network provides us to share the files with each other.

Back up and Roll back is easy

Since the files are stored in the main server which is centrally located. Therefore, it is easy to take the back up from the main server.

Software and Hardware sharing

We can install the applications on the main server, therefore, the user can access the applications centrally. So, we do not need to install the software on every machine. Similarly, hardware can also be shared.

Security

Network allows the security by ensuring that the user has the right to access the certain files and applications.

Scalability

Scalability means that we can add the new components on the network. Network must be scalable so that we can extend the network by adding new devices. But, it decreases the speed of the connection and data of the transmission speed also decreases, this increases the chances of error occurring. This problem can be overcome by using the routing or switching devices.

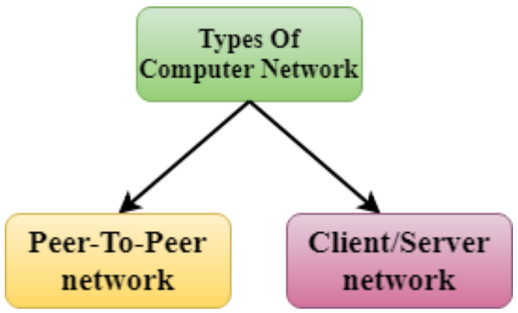
Reliability

Computer network can use the alternative source for the data communication in case of any hardware failure.

Computer Network Architecture

Computer Network Architecture is defined as the physical and logical design of the software, hardware, protocols, and media of the transmission of data. Simply we can say that how computers are organized and how tasks are allocated to the computer.

The two types of network architectures are used:

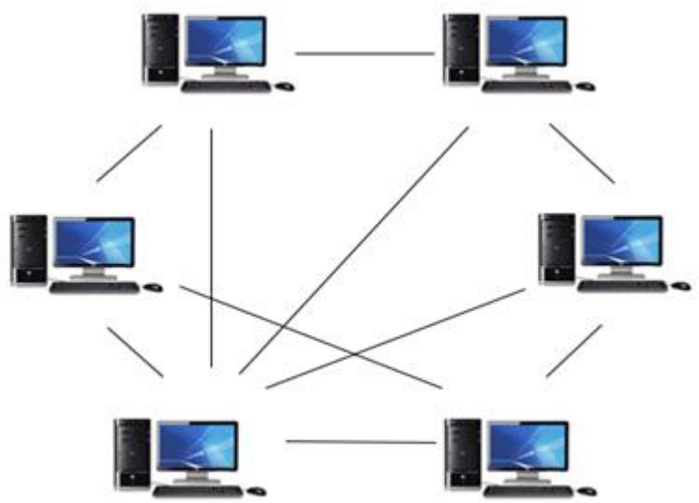


- Peer-To-Peer network
- Client/Server network

Peer-To-Peer network

- Peer-To-Peer network is a network in which all the computers are linked together with equal privilege and responsibilities for processing the data.
- Peer-To-Peer network is useful for small environments, usually up to 10 computers.
- Peer-To-Peer network has no dedicated server.

- Special permissions are assigned to each computer for sharing the resources, but this can lead to a problem if the computer with the resource is down.



Advantages Of Peer-To-Peer Network:

- It is less costly as it does not contain any dedicated server.
- If one computer stops working but, other computers will not stop working.
- It is easy to set up and maintain as each computer manages itself.

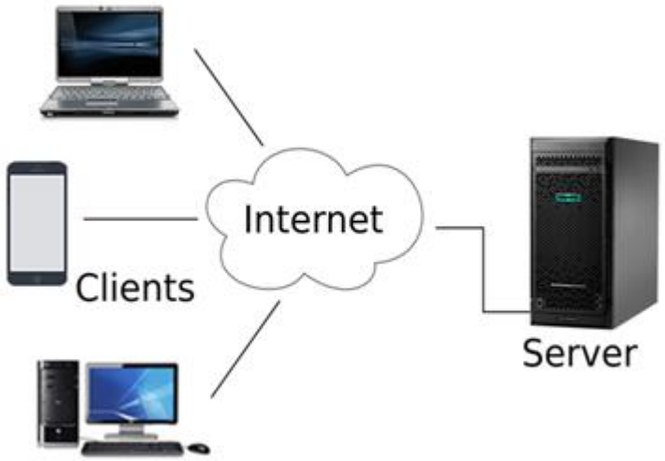
Disadvantages Of Peer-To-Peer Network:

- In the case of Peer-To-Peer network, it does not contain the centralized system . Therefore, it cannot back up the data as the data is different in different locations.
- It has a security issue as the device is managed itself.

Client/Server Network

- Client/Server network is a network model designed for the end users called clients, to access the resources such as songs, video, etc. from a central computer known as Server.
- The central controller is known as a **server** while all other computers in the network are called **clients**.

- A server performs all the major operations such as security and network management.
- A server is responsible for managing all the resources such as files, directories, printer, etc.
- All the clients communicate with each other through a server. For example, if client1 wants to send some data to client 2, then it first sends the request to the server for the permission. The server sends the response to the client 1 to initiate its communication with the client 2.



Advantages Of Client/Server network:

- A Client/Server network contains the centralized system. Therefore we can back up the data easily.
- A Client/Server network has a dedicated server that improves the overall performance of the whole system.
- Security is better in Client/Server network as a single server administers the shared resources.
- It also increases the speed of the sharing resources.

Disadvantages Of Client/Server network:

- Client/Server network is expensive as it requires the server with large memory.
- A server has a Network Operating System(NOS) to provide the resources to the clients, but the cost of NOS is very high.
- It requires a dedicated network administrator to manage all the resources.

Computer Network Components

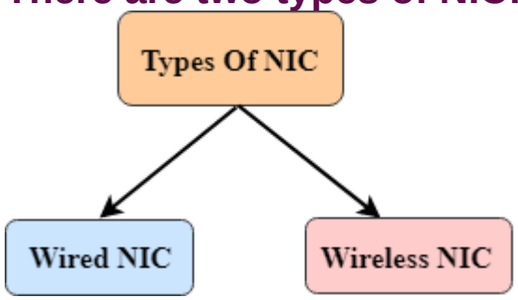
Computer network components are the *major parts* which are needed to *install the software*. Some important network components are **NIC**, **switch**, **cable**, **hub**, **router**, and **modem**. Depending on the type of network that we need to install, some network components can also be removed. For example, the wireless network does not require a cable.

Following are the major components required to install a network:

NIC

- NIC stands for network interface card.
- NIC is a hardware component used to connect a computer with another computer onto a network
- It can support a transfer rate of 10,100 to 1000 Mb/s.
- The MAC address or physical address is encoded on the network card chip which is assigned by the IEEE to identify a network card uniquely. The MAC address is stored in the PROM (Programmable read-only memory).

There are two types of NIC:



1. Wired NIC
2. Wireless NIC

Wired NIC: The Wired NIC is present inside the motherboard. Cables and connectors are used with wired NIC to transfer data.

Wireless NIC: The wireless NIC contains the antenna to obtain the connection over the wireless network. For example, laptop computer contains the wireless NIC.

Hub

A Hub is a hardware device that divides the network connection among multiple devices. When computer requests for some information from a network, it first sends the request to the Hub through cable. Hub will broadcast this request to the entire network. All the devices will check whether the request belongs to them or not. If not, the request will be dropped.

The process used by the Hub consumes more bandwidth and limits the amount of communication. Nowadays, the use of hub is obsolete, and it is replaced by more advanced computer network components such as Switches, Routers.

Switch

A switch is a hardware device that connects multiple devices on a computer network. A Switch contains more advanced features than Hub. The Switch contains the updated table that decides where the data is transmitted or not. Switch delivers the message to the correct destination based on the physical address present in the incoming message. A Switch does not broadcast the message to the entire network like the Hub. It determines the device to whom the message is to be transmitted. Therefore, we can say that switch provides a direct connection between the source and destination. It increases the speed of the network.

Router

- A router is a hardware device which is used to connect a LAN with an internet connection. It is used to receive, analyze and forward the incoming packets to another network.
- A router works in a **Layer 3 (Network layer)** of the OSI Reference model.
- A router forwards the packet based on the information available in the routing table.
- It determines the best path from the available paths for the transmission of the packet.

Advantages Of Router:

- **Security:** The information which is transmitted to the network will traverse the entire cable, but the only specified device which has been addressed can read the data.
- **Reliability:** If the server has stopped functioning, the network goes down, but no other networks are affected that are served by the router.
- **Performance:** Router enhances the overall performance of the network. Suppose there are 24 workstations in a network generates a same amount of traffic. This increases the traffic load on the network. Router splits the single network into two networks of 12 workstations each, reduces the traffic load by half.
- **Network range**

Modem

- A modem is a hardware device that allows the computer to connect to the internet over the existing telephone line.
- A modem is not integrated with the motherboard rather than it is installed on the PCI slot found on the motherboard.
- It stands for Modulator/Demodulator. It converts the digital data into an analog signal over the telephone lines.

Based on the differences in speed and transmission rate, a modem can be classified in the following categories:

- Standard PC modem or Dial-up modem
- Cellular Modem
- Cable modem

Cables and Connectors

Cable is a transmission media used for transmitting a signal.

There are three types of cables used in transmission:

- Twisted pair cable
- Coaxial cable
- Fibre-optic cable

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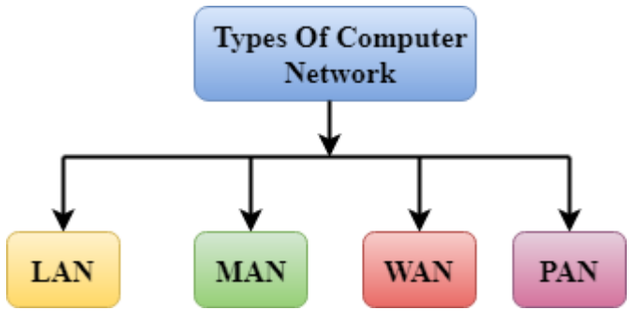
Computer Network Types

A computer network is a group of computers linked to each other that enables the computer to communicate with another computer and share their resources, data, and applications.

A computer network can be categorized by their size. A **computer network** is mainly of **four types**:



- LAN(Local Area Network)
- PAN(Personal Area Network)
- MAN(Metropolitan Area Network)
- WAN(Wide Area Network)

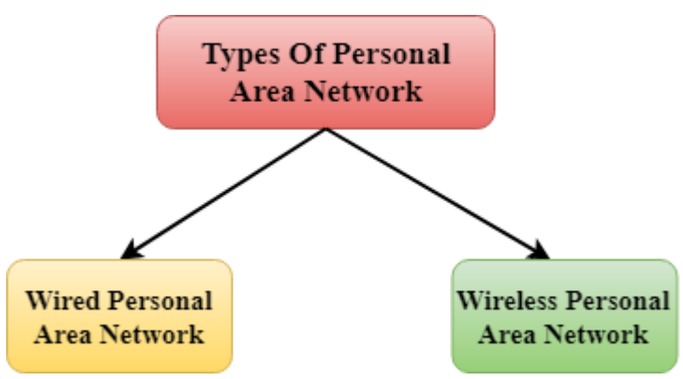
LAN(Local Area Network)

- Local Area Network is a group of computers connected to each other in a small area such as building, office.
- LAN is used for connecting two or more personal computers through a communication medium such as twisted pair, coaxial cable, etc.
- It is less costly as it is built with inexpensive hardware such as hubs, network adapters, and ethernet cables.
- The data is transferred at an extremely faster rate in Local Area Network.
- Local Area Network provides higher security.

PAN(Personal Area Network)

- Personal Area Network is a network arranged within an individual person, typically within a range of 10 meters.
- Personal Area Network is used for connecting the computer devices of personal use is known as Personal Area Network.
- **Thomas Zimmerman** was the first research scientist to bring the idea of the Personal Area Network.
- Personal Area Network covers an area of **30 feet**.
- Personal computer devices that are used to develop the personal area network are the laptop, mobile phones, media player and play stations.

There are two types of Personal Area Network:



- Wired Personal Area Network
- Wireless Personal Area Network

Wireless Personal Area Network: Wireless Personal Area Network is developed by simply using wireless technologies such as WiFi, Bluetooth. It is a low range network.

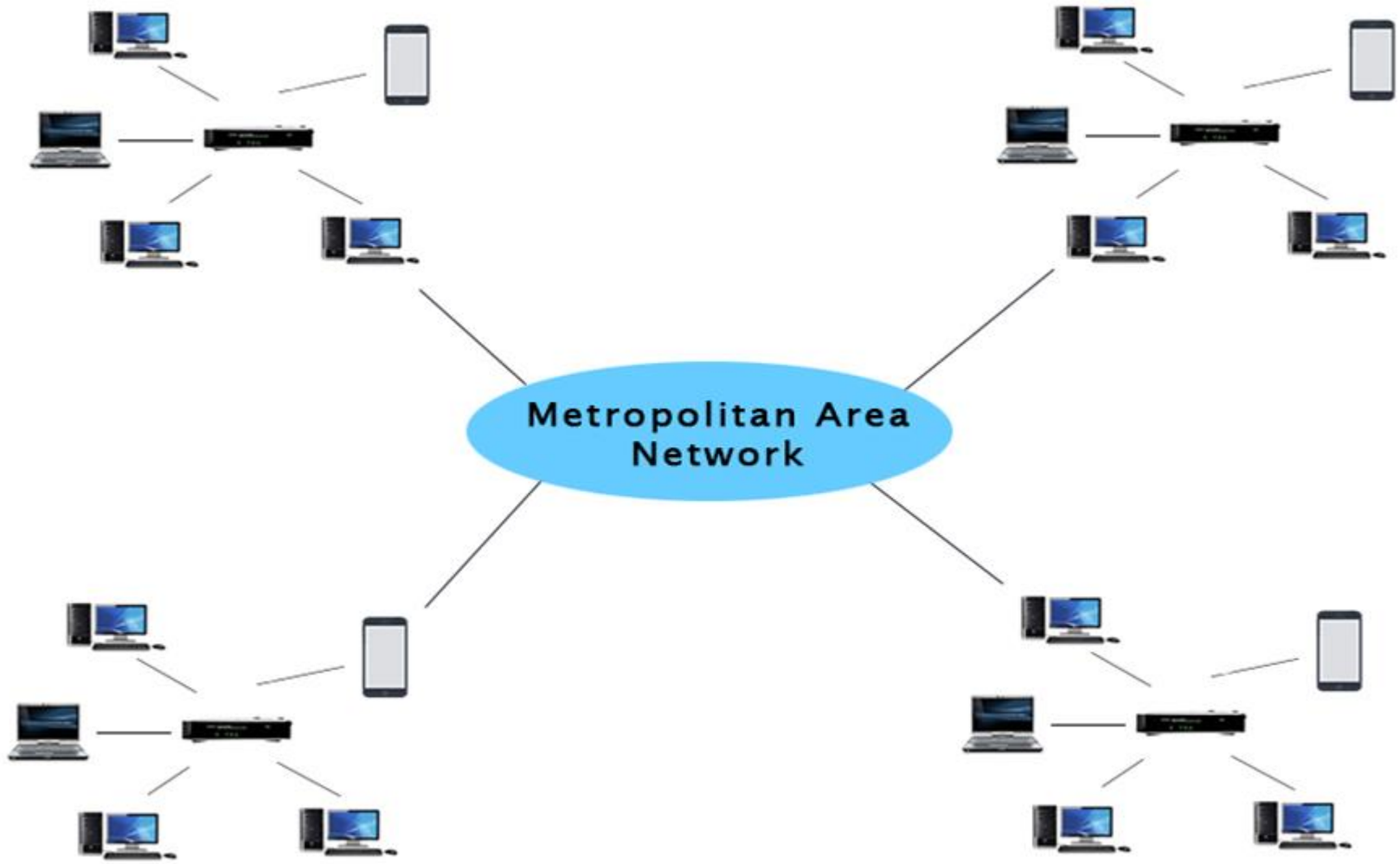
Wired Personal Area Network: Wired Personal Area Network is created by using the USB.

Examples Of Personal Area Network:

- **Body Area Network:** Body Area Network is a network that moves with a person. **For example**, a mobile network moves with a person. Suppose a person establishes a network connection and then creates a connection with another device to share the information.
- **Offline Network:** An offline network can be created inside the home, so it is also known as a **home network**. A home network is designed to integrate the devices such as printers, computer, television but they are not connected to the internet.
- **Small Home Office:** It is used to connect a variety of devices to the internet and to a corporate network using a VPN

MAN(Metropolitan Area Network)

- A metropolitan area network is a network that covers a larger geographic area by interconnecting a different LAN to form a larger network.
- Government agencies use MAN to connect to the citizens and private industries.
- In MAN, various LANs are connected to each other through a telephone exchange line.
- The most widely used protocols in MAN are RS-232, Frame Relay, ATM, ISDN, OC-3, ADSL, etc.
- It has a higher range than Local Area Network(LAN).

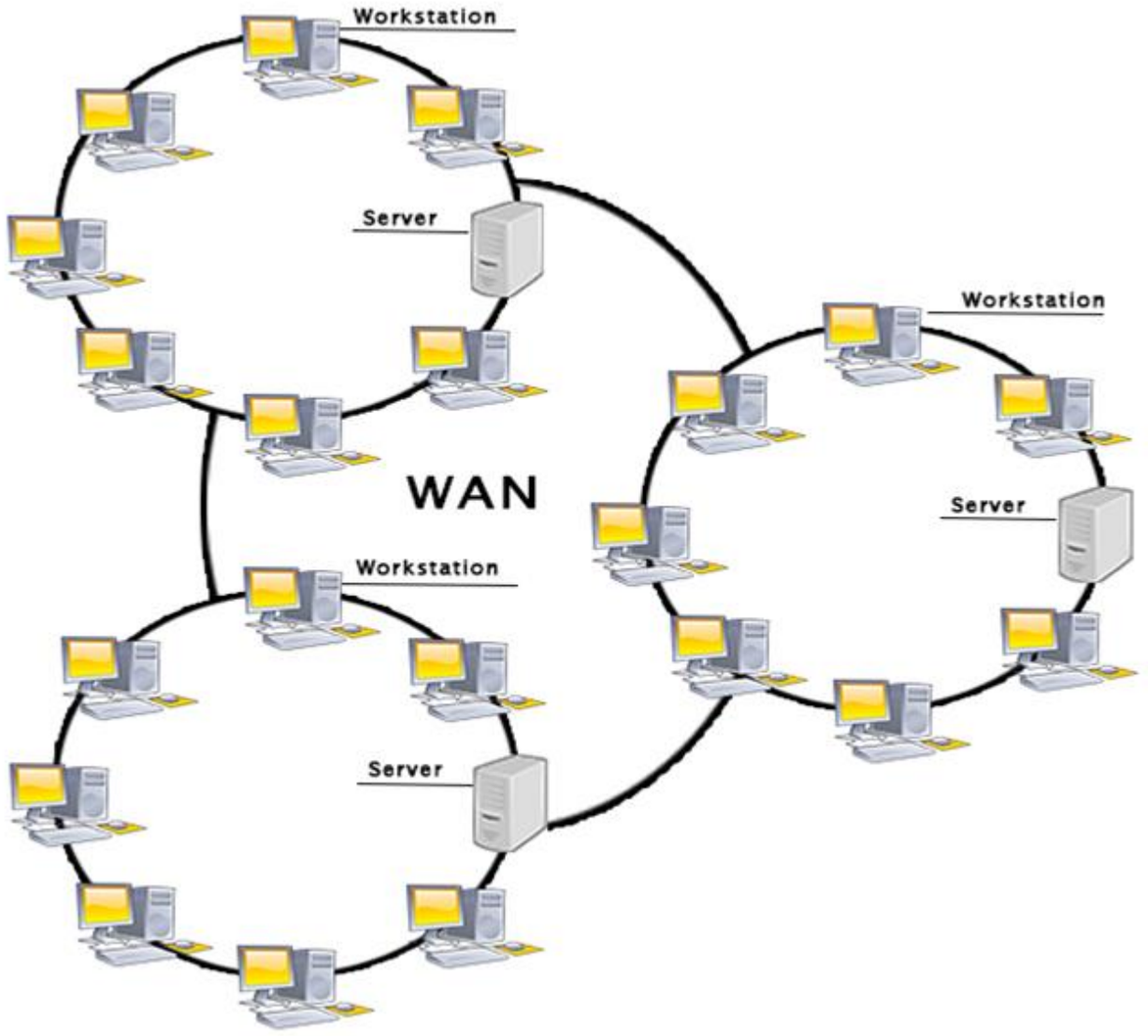


Uses Of Metropolitan Area Network:

- MAN is used in communication between the banks in a city.
- It can be used in an Airline Reservation.
- It can be used in a college within a city.
- It can also be used for communication in the military.

WAN(Wide Area Network)

- A Wide Area Network is a network that extends over a large geographical area such as states or countries.
- A Wide Area Network is quite bigger network than the LAN.
- A Wide Area Network is not limited to a single location, but it spans over a large geographical area through a telephone line, fibre optic cable or satellite links.
- The internet is one of the biggest WAN in the world.
- A Wide Area Network is widely used in the field of Business, government, and education.



Examples Of Wide Area Network:

- **Mobile Broadband:** A 4G network is widely used across a region or country.
- **Last mile:** A telecom company is used to provide the internet services to the customers in hundreds of cities by connecting their home with fiber.
- **Private network:** A bank provides a private network that connects the 44 offices. This network is made by using the telephone leased line provided by the telecom company.

Advantages Of Wide Area Network:

Following are the advantages of the Wide Area Network:

- **Geographical area:** A Wide Area Network provides a large geographical area. Suppose if the branch of our office is in a different city then we can connect with them through WAN. The internet provides a leased line through which we can connect with another branch.
- **Centralized data:** In case of WAN network, data is centralized. Therefore, we do not need to buy the emails, files or back up servers.
- **Get updated files:** Software companies work on the live server. Therefore, the programmers get the updated files within seconds.
- **Exchange messages:** In a WAN network, messages are transmitted fast. The web application like Facebook, Whatsapp, Skype allows you to communicate with friends.
- **Sharing of software and resources:** In WAN network, we can share the software and other resources like a hard drive, RAM.
- **Global business:** We can do the business over the internet globally.
- **High bandwidth:** If we use the leased lines for our company then this gives the high bandwidth. The high bandwidth increases the data transfer rate which in turn increases the productivity of our company.

Disadvantages of Wide Area Network:

The following are the disadvantages of the Wide Area Network:

- **Security issue:** A WAN network has more security issues as compared to LAN and MAN network as all the technologies are combined together that creates the security problem.
- **Needs Firewall & antivirus software:** The data is transferred on the internet which can be changed or hacked by the hackers, so the firewall needs to be used. Some people can inject the virus in our system so antivirus is needed to protect from such a virus.
- **High Setup cost:** An installation cost of the WAN network is high as it involves the purchasing of routers, switches.
- **Troubleshooting problems:** It covers a large area so fixing the problem is difficult.

Internetwork

- An internetwork is defined as two or more computer network LANs or WAN or computer network segments are connected using devices, and they are configured by a local addressing scheme. This process is known as **internetworking**.
- An interconnection between public, private, commercial, industrial, or government computer networks can also be defined as **internetworking**.
- An internetworking uses the **internet protocol**.
- The reference model used for internetworking is **Open System Interconnection(OSI)**.

Types Of Internetwork:

1. **Extranet:** An extranet is a communication network based on the internet protocol such as **Transmission Control protocol** and **internet protocol**. It is used for information sharing. The access to the extranet is restricted to only those users who have login credentials. An extranet is the lowest level of internetworking. It can be categorized as **MAN**, **WAN** or other computer networks. An extranet cannot have a single **LAN**, atleast it must have one connection to the external network.

2. **Intranet:** An intranet is a private network based on the internet protocol such as **Transmission Control protocol** and **internet protocol**. An intranet belongs to an organization which is only accessible by the **organization's employee** or members. The main aim of the intranet is to share the information and resources among the organization employees. An intranet provides the facility to work in groups and for teleconferences.

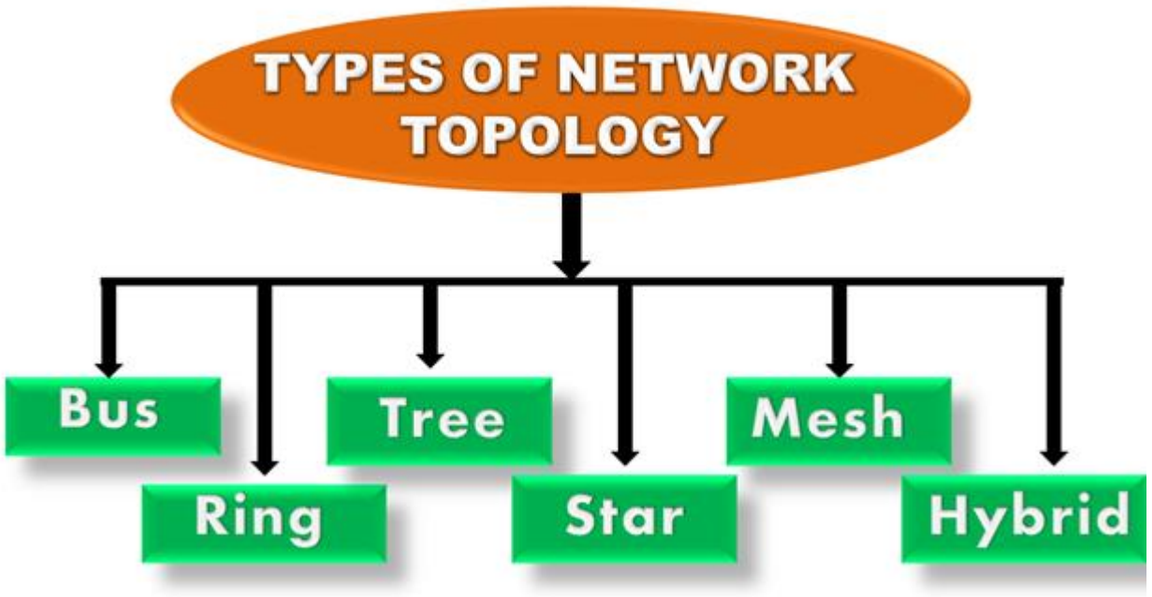
Intranet advantages:

- **Communication:** It provides a cheap and easy communication. An employee of the organization can communicate with another employee through email, chat.
- **Time-saving:** Information on the intranet is shared in real time, so it is time-saving.
- **Collaboration:** Collaboration is one of the most important advantage of the intranet. The information is distributed among the employees of the organization and can only be accessed by the authorized user.
- **Platform independency:** It is a neutral architecture as the computer can be connected to another device with different architecture.
- **Cost effective:** People can see the data and documents by using the browser and distributes the duplicate copies over the intranet. This leads to a reduction in the cost.

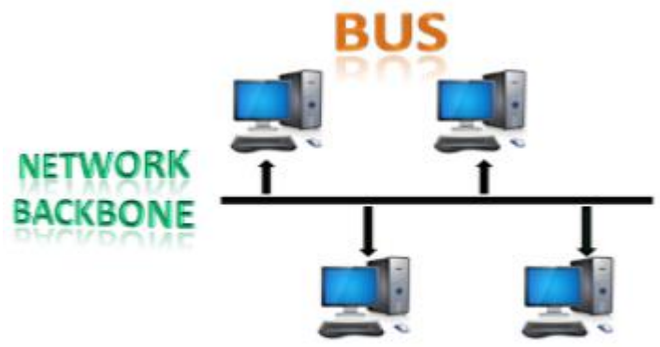
What is Topology?

Topology defines the structure of the network of how all the components are interconnected to each other. There are two types of topology: physical and logical topology.

Physical topology is the geometric representation of all the nodes in a network.



Bus Topology



- The bus topology is designed in such a way that all the stations are connected through a single cable known as a backbone cable.

- Each node is either connected to the backbone cable by drop cable or directly connected to the backbone cable.
- When a node wants to send a message over the network, it puts a message over the network. All the stations available in the network will receive the message whether it has been addressed or not.
- The bus topology is mainly used in 802.3 (ethernet) and 802.4 standard networks.
- The configuration of a bus topology is quite simpler as compared to other topologies.
- The backbone cable is considered as a **"single lane"** through which the message is broadcast to all the stations.
- The most common access method of the bus topologies is **CSMA** (Carrier Sense Multiple Access).

CSMA: It is a media access control used to control the data flow so that data integrity is maintained, i.e., the packets do not get lost. There are two alternative ways of handling the problems that occur when two nodes send the messages simultaneously.

- **CSMA CD:** CSMA CD (**Collision detection**) is an access method used to detect the collision. Once the collision is detected, the sender will stop transmitting the data. Therefore, it works on "**recovery after the collision**".
- **CSMA CA: CSMA CA (Collision Avoidance)** is an access method used to avoid the collision by checking whether the transmission media is busy or not. If busy, then the sender waits until the media becomes idle. This technique effectively reduces the possibility of the collision. It does not work on "recovery after the collision".

Advantages of Bus topology:

- **Low-cost cable:** In bus topology, nodes are directly connected to the cable without passing through a hub. Therefore, the initial cost of installation is low.
- **Moderate data speeds:** Coaxial or twisted pair cables are mainly used in bus-based networks that support upto 10 Mbps.
- **Familiar technology:** Bus topology is a familiar technology as the installation and troubleshooting techniques are well known, and hardware components are easily available.
- **Limited failure:** A failure in one node will not have any effect on other nodes.

Disadvantages of Bus topology:

- **Extensive cabling:** A bus topology is quite simpler, but still it requires a lot of cabling.
- **Difficult troubleshooting:** It requires specialized test equipment to determine the cable faults. If any fault occurs in the cable, then it would disrupt the communication for all the nodes.
- **Signal interference:** If two nodes send the messages simultaneously, then the signals of both the nodes collide with each other.
- **Reconfiguration difficult:** Adding new devices to the network would slow down the network.

- **Attenuation:** Attenuation is a loss of signal leads to communication issues. Repeaters are used to regenerate the signal.

Ring Topology

- Ring topology is like a bus topology, but with connected ends.
- The node that receives the message from the previous computer will retransmit to the next node.
- The data flows in one direction, i.e., it is unidirectional.
- The data flows in a single loop continuously known as an endless loop.
- It has no terminated ends, i.e., each node is connected to other node and having no termination point.
- The data in a ring topology flow in a clockwise direction.
- The most common access method of the ring topology is **token passing**.
    - **Token passing:** It is a network access method in which token is passed from one node to another node.
    - **Token:** It is a frame that circulates around the network.

Working of Token passing

- A token moves around the network, and it is passed from computer to computer until it reaches the destination.
- The sender modifies the token by putting the address along with the data.

- The data is passed from one device to another device until the destination address matches. Once the token received by the destination device, then it sends the acknowledgment to the sender.
- In a ring topology, a token is used as a carrier.

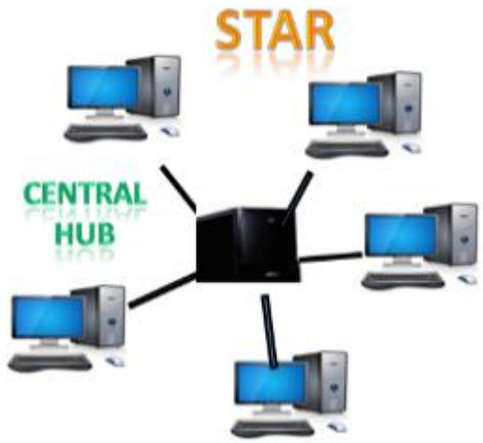
Advantages of Ring topology:

- **Network Management:** Faulty devices can be removed from the network without bringing the network down.
- **Product availability:** Many hardware and software tools for network operation and monitoring are available.
- **Cost:** Twisted pair cabling is inexpensive and easily available. Therefore, the installation cost is very low.
- **Reliable:** It is a more reliable network because the communication system is not dependent on the single host computer.

Disadvantages of Ring topology:

- **Difficult troubleshooting:** It requires specialized test equipment to determine the cable faults. If any fault occurs in the cable, then it would disrupt the communication for all the nodes.
- **Failure:** The breakdown in one station leads to the failure of the overall network.
- **Reconfiguration difficult:** Adding new devices to the network would slow down the network.
- **Delay:** Communication delay is directly proportional to the number of nodes. Adding new devices increases the communication delay.

Star Topology



- Star topology is an arrangement of the network in which every node is connected to the central hub, switch or a central computer.
- The central computer is known as a **server**, and the peripheral devices attached to the server are known as **clients**.
- Coaxial cable or RJ-45 cables are used to connect the computers.
- Hubs or Switches are mainly used as connection devices in a **physical star topology**.
- Star topology is the most popular topology in network implementation.

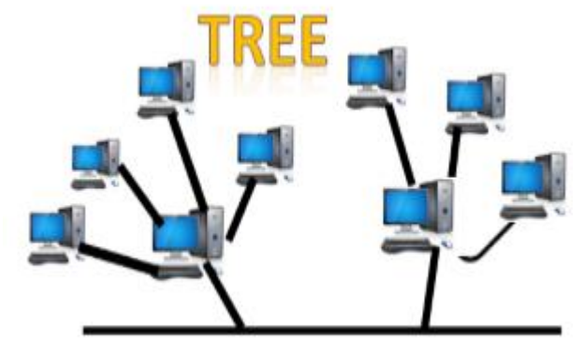
Advantages of Star topology

- **Efficient troubleshooting:** Troubleshooting is quite efficient in a star topology as compared to bus topology. In a bus topology, the manager has to inspect the kilometers of cable. In a star topology, all the stations are connected to the centralized network. Therefore, the network administrator has to go to the single station to troubleshoot the problem.
- **Network control:** Complex network control features can be easily implemented in the star topology. Any changes made in the star topology are automatically accommodated.
- **Limited failure:** As each station is connected to the central hub with its own cable, therefore failure in one cable will not affect the entire network.
- **Familiar technology:** Star topology is a familiar technology as its tools are cost-effective.
- **Easily expandable:** It is easily expandable as new stations can be added to the open ports on the hub.
- **Cost effective:** Star topology networks are cost-effective as it uses inexpensive coaxial cable.
- **High data speeds:** It supports a bandwidth of approx 100Mbps. Ethernet 100BaseT is one of the most popular Star topology networks.

Disadvantages of Star topology

- **A Central point of failure:** If the central hub or switch goes down, then all the connected nodes will not be able to communicate with each other.
- **Cable:** Sometimes cable routing becomes difficult when a significant amount of routing is required.

Tree topology



- Tree topology combines the characteristics of bus topology and star topology.
- A tree topology is a type of structure in which all the computers are connected with each other in hierarchical fashion.
- The top-most node in tree topology is known as a root node, and all other nodes are the descendants of the root node.
- There is only one path exists between two nodes for the data transmission. Thus, it forms a parent-child hierarchy.

Advantages of Tree topology

- **Support for broadband transmission:** Tree topology is mainly used to provide broadband transmission, i.e., signals are sent over long distances without being attenuated.
- **Easily expandable:** We can add the new device to the existing network. Therefore, we can say that tree topology is easily expandable.
- **Easily manageable:** In tree topology, the whole network is divided into segments known as star networks which can be easily managed and maintained.
- **Error detection:** Error detection and error correction are very easy in a tree topology.
- **Limited failure:** The breakdown in one station does not affect the entire network.
- **Point-to-point wiring:** It has point-to-point wiring for individual segments.

Disadvantages of Tree topology

- **Difficult troubleshooting:** If any fault occurs in the node, then it becomes difficult to troubleshoot the problem.
- **High cost:** Devices required for broadband transmission are very costly.
- **Failure:** A tree topology mainly relies on main bus cable and failure in main bus cable will damage the overall network.
- **Reconfiguration difficult:** If new devices are added, then it becomes difficult to reconfigure.

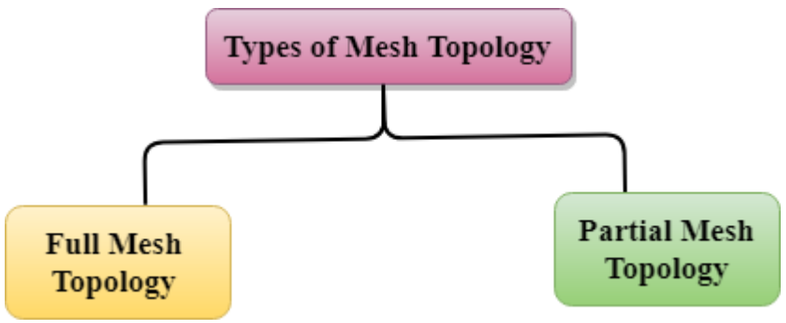
Mesh topology

- Mesh technology is an arrangement of the network in which computers are interconnected with each other through various redundant connections.
- There are multiple paths from one computer to another computer.
- It does not contain the switch, hub or any central computer which acts as a central point of communication.
- The Internet is an example of the mesh topology.
- Mesh topology is mainly used for WAN implementations where communication failures are a critical concern.
- Mesh topology is mainly used for wireless networks.
- Mesh topology can be formed by using the formula:
  Number of cables = (n*(n-1))/2;

Where n is the number of nodes that represents the network.

Mesh topology is divided into two categories:

- Fully connected mesh topology
- Partially connected mesh topology



- **Full Mesh Topology:** In a full mesh topology, each computer is connected to all the computers available in the network.
- **Partial Mesh Topology:** In a partial mesh topology, not all but certain computers are connected to those computers with which they communicate frequently.

Advantages of Mesh topology:

Reliable: The mesh topology networks are very reliable as if any link breakdown will not affect the communication between connected computers.

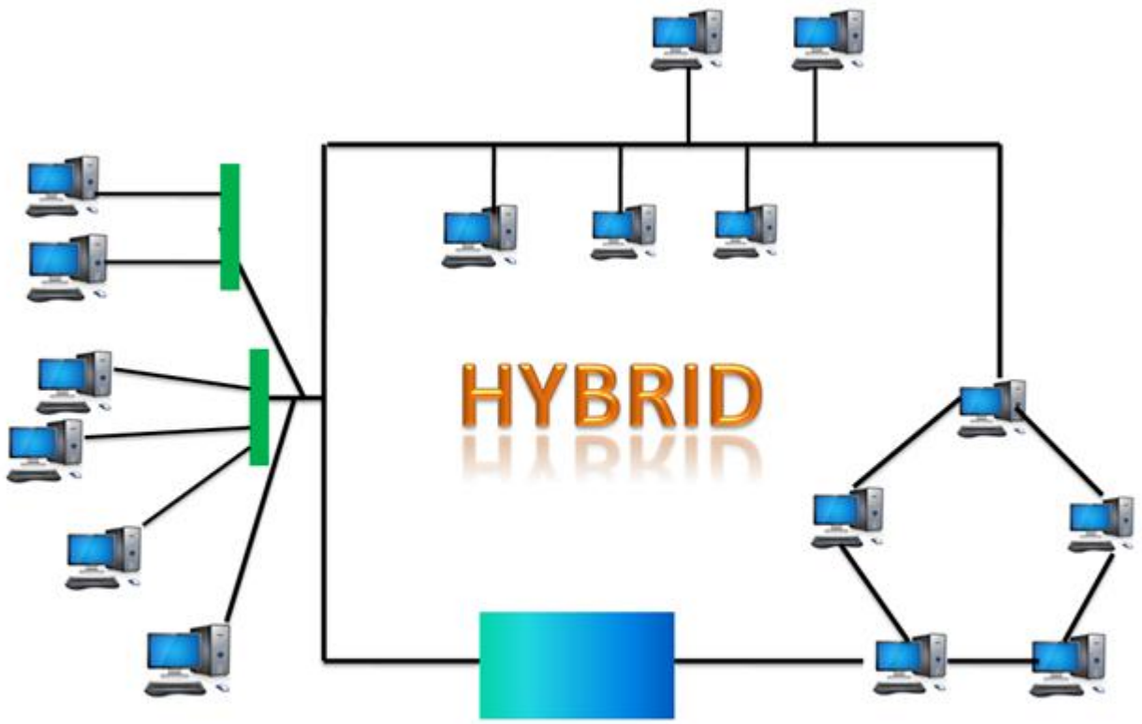
Fast Communication: Communication is very fast between the nodes.

Easier Reconfiguration: Adding new devices would not disrupt the communication between other devices.

Disadvantages of Mesh topology

- **Cost:** A mesh topology contains a large number of connected devices such as a router and more transmission media than other topologies.
- **Management:** Mesh topology networks are very large and very difficult to maintain and manage. If the network is not monitored carefully, then the communication link failure goes undetected.
- **Efficiency:** In this topology, redundant connections are high that reduces the efficiency of the network.

Hybrid Topology



- The combination of various different topologies is known as **Hybrid topology**.
- A Hybrid topology is a connection between different links and nodes to transfer the data.
- When two or more different topologies are combined together is termed as Hybrid topology and if similar topologies are connected with each other will not result in Hybrid topology. For example, if there exist a ring topology in one branch of ICICI bank and bus topology in another branch of ICICI bank, connecting these two topologies will result in Hybrid topology.

Advantages of Hybrid Topology

- **Reliable:** If a fault occurs in any part of the network will not affect the functioning of the rest of the network.
- **Scalable:** Size of the network can be easily expanded by adding new devices without affecting the functionality of the existing network.
- **Flexible:** This topology is very flexible as it can be designed according to the requirements of the organization.

- **Effective:** Hybrid topology is very effective as it can be designed in such a way that the strength of the network is maximized and weakness of the network is minimized.

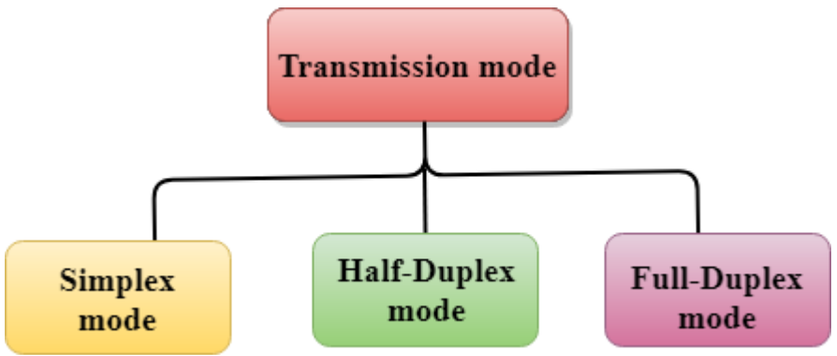
Disadvantages of Hybrid topology

- **Complex design:** The major drawback of the Hybrid topology is the design of the Hybrid network. It is very difficult to design the architecture of the Hybrid network.
- **Costly Hub:** The Hubs used in the Hybrid topology are very expensive as these hubs are different from usual Hubs used in other topologies.
- **Costly infrastructure:** The infrastructure cost is very high as a hybrid network requires a lot of cabling, network devices, etc.

Transmission modes

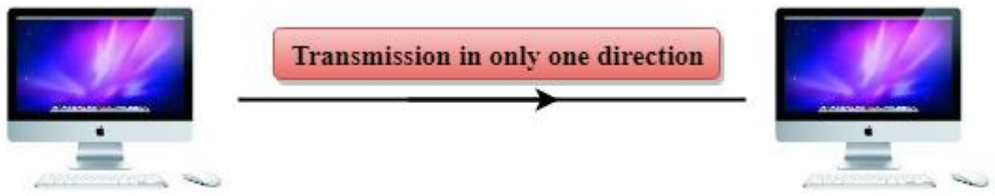
- The way in which data is transmitted from one device to another device is known as **transmission mode**.
- The transmission mode is also known as the communication mode.
- Each communication channel has a direction associated with it, and transmission media provide the direction. Therefore, the transmission mode is also known as a directional mode.
- The transmission mode is defined in the physical layer.

The Transmission mode is divided into three categories:



- Simplex mode
- Half-duplex mode
- Full-duplex mode

Simplex mode



- In Simplex mode, the communication is unidirectional, i.e., the data flow in one direction.
- A device can only send the data but cannot receive it or it can receive the data but cannot send the data.
- This transmission mode is not very popular as mainly communications require the two-way exchange of data. The simplex mode is used in the business field as in sales that do not require any corresponding reply.
- The radio station is a simplex channel as it transmits the signal to the listeners but never allows them to transmit back.
- Keyboard and Monitor are the examples of the simplex mode as a keyboard can only accept the data from the user and monitor can only be used to display the data on the screen.
- The main advantage of the simplex mode is that the full capacity of the communication channel can be utilized during transmission.

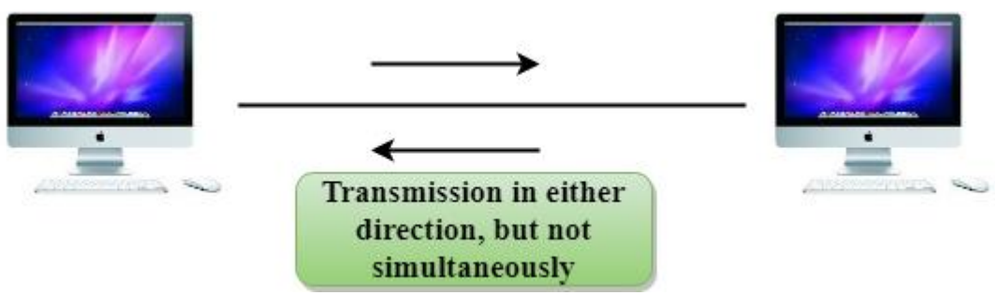
Advantage of Simplex mode:

- In simplex mode, the station can utilize the entire bandwidth of the communication channel, so that more data can be transmitted at a time.

Disadvantage of Simplex mode:

- Communication is unidirectional, so it has no inter-communication between devices.

Half-Duplex mode



- In a Half-duplex channel, direction can be reversed, i.e., the station can transmit and receive the data as well.
- Messages flow in both the directions, but not at the same time.
- The entire bandwidth of the communication channel is utilized in one direction at a time.
- In half-duplex mode, it is possible to perform the error detection, and if any error occurs, then the receiver requests the sender to retransmit the data.
- A **Walkie-talkie** is an example of the Half-duplex mode. In Walkie-talkie, one party speaks, and another party listens. After a pause, the other speaks and first party listens. Speaking simultaneously will create the distorted sound which cannot be understood.

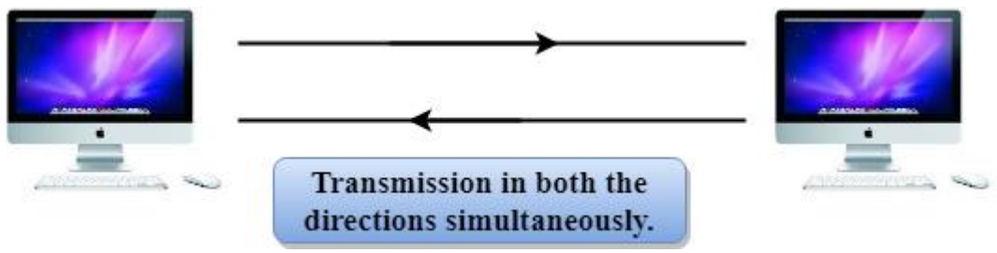
Advantage of Half-duplex mode:

- In half-duplex mode, both the devices can send and receive the data and also can utilize the entire bandwidth of the communication channel during the transmission of data.

Disadvantage of Half-Duplex mode:

- In half-duplex mode, when one device is sending the data, then another has to wait, this causes the delay in sending the data at the right time.

Full-duplex mode



- In Full duplex mode, the communication is bi-directional, i.e., the data flow in both the directions.
- Both the stations can send and receive the message simultaneously.
- Full-duplex mode has two simplex channels. One channel has traffic moving in one direction, and another channel has traffic flowing in the opposite direction.
- The Full-duplex mode is the fastest mode of communication between devices.
- The most common example of the full-duplex mode is a telephone network. When two people are communicating with each other by a telephone line, both can talk and listen at the same time.

Advantage of Full-duplex mode:

- Both the stations can send and receive the data at the same time.

Disadvantage of Full-duplex mode:

- If there is no dedicated path exists between the devices, then the capacity of the communication channel is divided into two parts.

Differences b/w Simplex, Half-duplex and Full-duplex mode

Basis for comparison	Simplex mode	Half-duplex mode	Full-duplex mode
Direction of communication	In simplex mode, the communication is unidirectional.	In half-duplex mode, the communication is bidirectional, but one at a time.	In full-duplex mode, the communication is bidirectional.
Send/Receive	A device can only send the data but cannot receive it or it can only receive the data but cannot send it.	Both the devices can send and receive the data, but one at a time.	Both the devices can send and receive the data simultaneously.
Performance	The performance of half-duplex mode is better than the simplex mode.	The performance of full-duplex mode is better than the half-duplex mode.	The Full-duplex mode has better performance among simplex and half-duplex mode as it doubles the utilization of the capacity of the communication channel.

Example	Examples of Simplex mode are radio, keyboard, and monitor.	Example of half-duplex is Walkie-Talkies.	Example of the Full-duplex mode is a telephone network.

Models

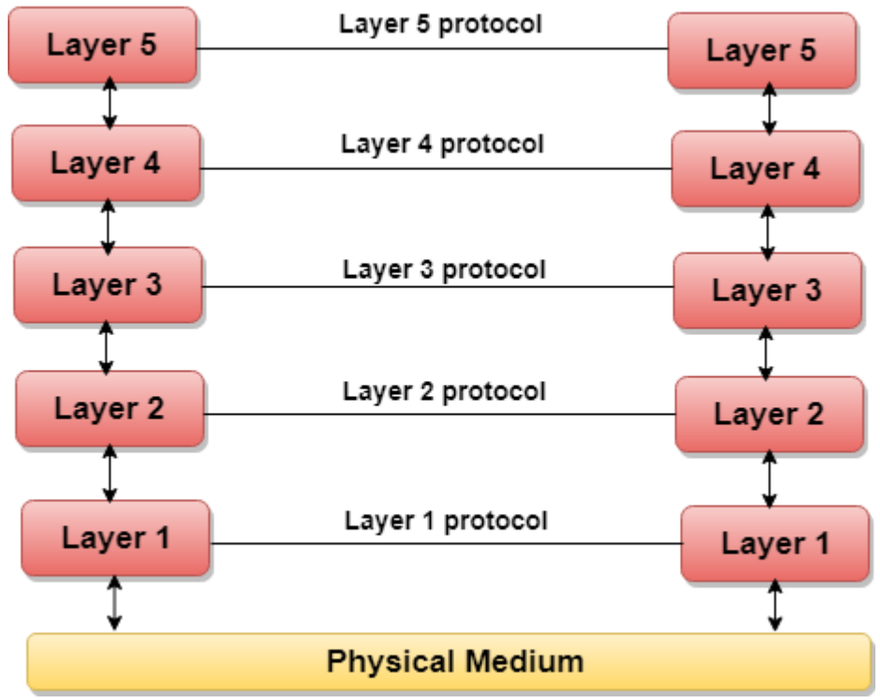
Computer Network Models

A communication subsystem is a complex piece of Hardware and software. Early attempts for implementing the software for such subsystems were based on a single, complex, unstructured program with many interacting components. The resultant software was very difficult to test and modify. To overcome such problem, the ISO has developed a layered approach. In a layered approach, networking concept is divided into several layers, and each layer is assigned a particular task. Therefore, we can say that networking tasks depend upon the layers.

Layered Architecture

- The main aim of the layered architecture is to divide the design into small pieces.
- Each lower layer adds its services to the higher layer to provide a full set of services to manage communications and run the applications.
- It provides modularity and clear interfaces, i.e., provides interaction between subsystems.
- It ensures the independence between layers by providing the services from lower to higher layer without defining how the services are implemented. Therefore, any modification in a layer will not affect the other layers.
- The number of layers, functions, contents of each layer will vary from network to network. However, the purpose of each layer is to provide the service from lower to a higher layer and hiding the details from the layers of how the services are implemented.
- The basic elements of layered architecture are services, protocols, and interfaces.
    - **Service:** It is a set of actions that a layer provides to the higher layer.
    - **Protocol:** It defines a set of rules that a layer uses to exchange the information with peer entity. These rules mainly concern about both the contents and order of the messages used.
    - **Interface:** It is a way through which the message is transferred from one layer to another layer.
- In a layer n architecture, layer n on one machine will have a communication with the layer n on another machine and the rules used in a conversation are known as a layer-n protocol.

Let's take an example of the five-layered architecture.



- In case of layered architecture, no data is transferred from layer n of one machine to layer n of another machine. Instead, each layer passes the data to the layer immediately just below it, until the lowest layer is reached.
- Below layer 1 is the physical medium through which the actual communication takes place.
- In a layered architecture, unmanageable tasks are divided into several small and manageable tasks.
- The data is passed from the upper layer to lower layer through an interface. A Layered architecture provides a clean-cut interface so that minimum information is shared among different layers. It also ensures that the implementation of one layer can be easily replaced by another implementation.
- A set of layers and protocols is known as network architecture.

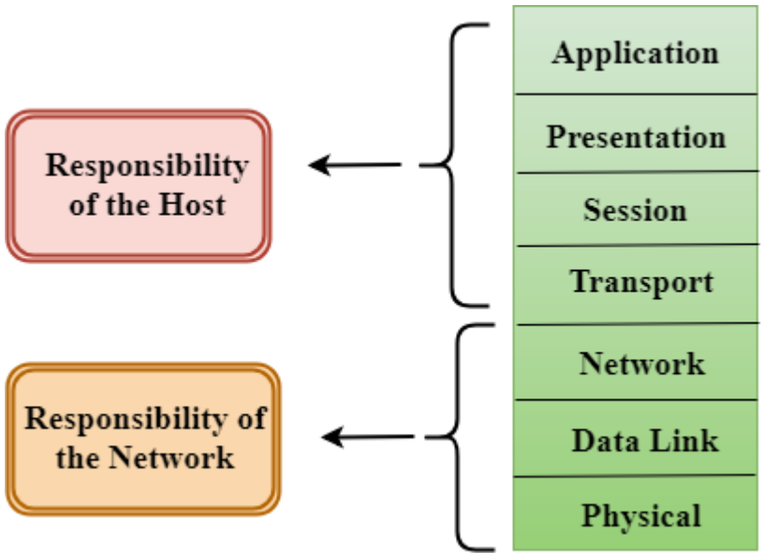
Why do we require Layered architecture?

- **Divide-and-conquer approach:** Divide-and-conquer approach makes a design process in such a way that the unmanageable tasks are divided into small and manageable tasks. In short, we can say that this approach reduces the complexity of the design.
- **Modularity:** Layered architecture is more modular. Modularity provides the independence of layers, which is easier to understand and implement.
- **Easy to modify:** It ensures the independence of layers so that implementation in one layer can be changed without affecting other layers.
- **Easy to test:** Each layer of the layered architecture can be analyzed and tested individually.

OSI Model

- OSI stands for **Open System Interconnection** is a reference model that describes how information from a software application in one computer moves through a physical medium to the software application in another computer.
- OSI consists of seven layers, and each layer performs a particular network function.
- OSI model was developed by the International Organization for Standardization (ISO) in 1984, and it is now considered as an architectural model for the inter-computer communications.
- OSI model divides the whole task into seven smaller and manageable tasks. Each layer is assigned a particular task.
- Each layer is self-contained, so that task assigned to each layer can be performed independently.

Characteristics of OSI Model:



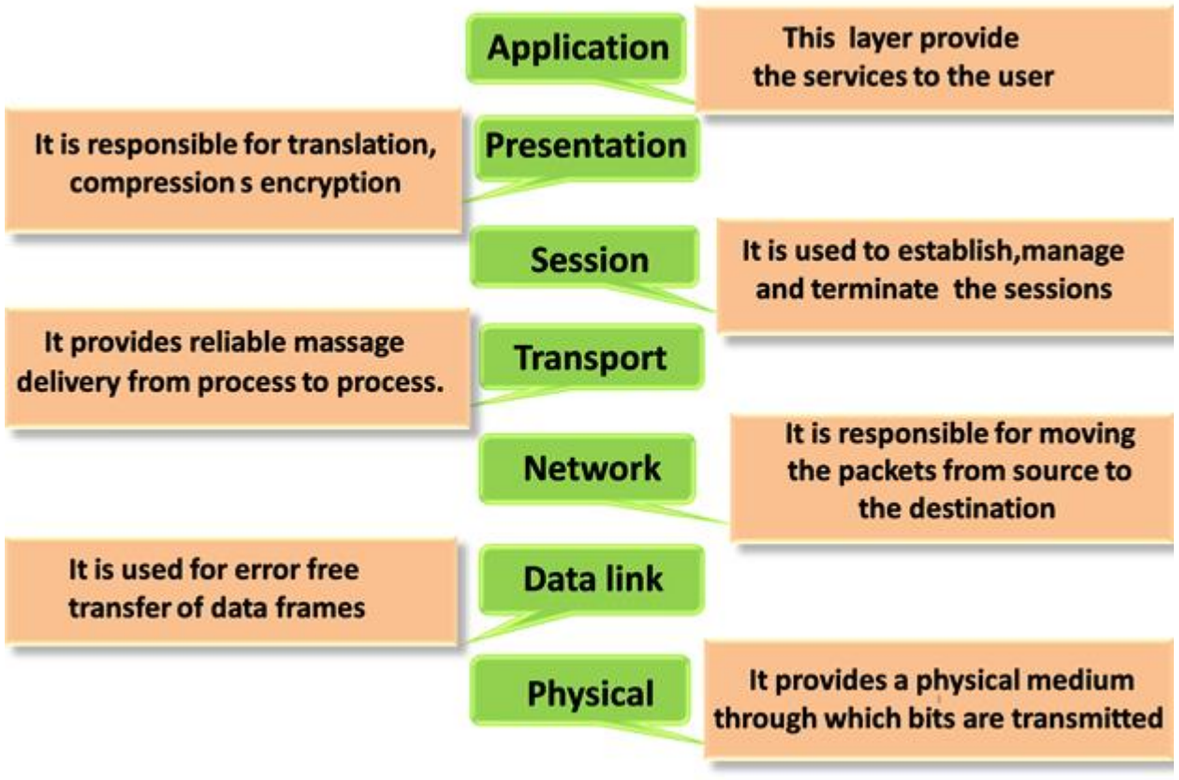
- The OSI model is divided into two layers: upper layers and lower layers.
- The upper layer of the OSI model mainly deals with the application related issues, and they are implemented only in the software. The application layer is closest to the end user. Both the end user and the application layer interact with the software applications. An upper layer refers to the layer just above another layer.

- The lower layer of the OSI model deals with the data transport issues. The data link layer and the physical layer are implemented in hardware and software. The physical layer is the lowest layer of the OSI model and is closest to the physical medium. The physical layer is mainly responsible for placing the information on the physical medium.

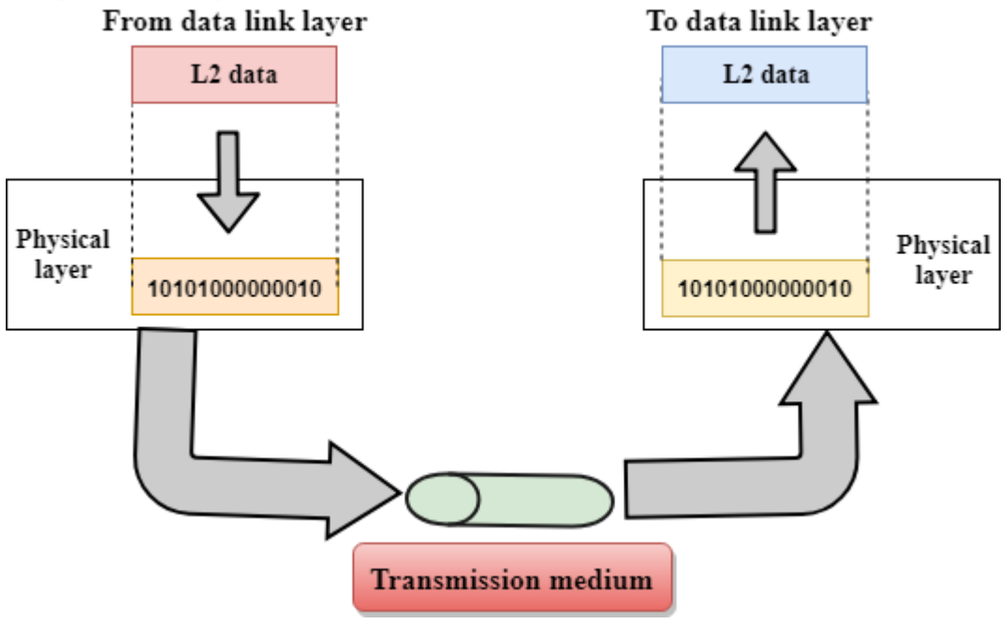
Functions of the OSI Layers

There are the seven OSI layers. Each layer has different functions. A list of seven layers are given below:

1. Physical Layer
2. Data-Link Layer
3. Network Layer
4. Transport Layer
5. Session Layer
6. Presentation Layer
7. Application Layer


Application
This layer provide the services to the user
Presentation
It is responsible for translation, compression s encryption
Session
It is used to establish,manage and terminate the sessions
Transport
It provides reliable massage delivery from process to process.
Network
It is responsible for moving the packets from source to the destination
Data link
It is used for error free transfer of data frames
Physical
It provides a physical medium through which bits are transmitted

Physical layer

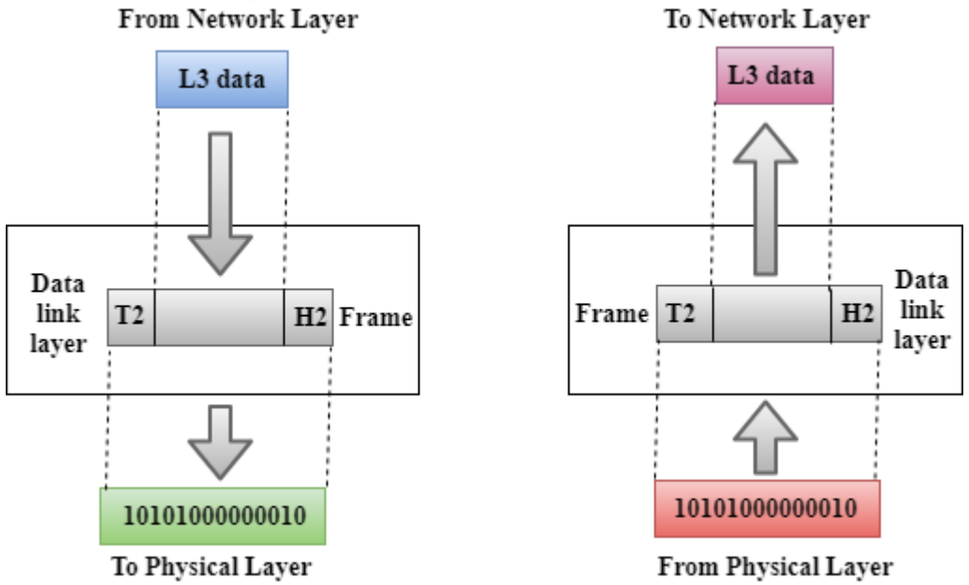


- The main functionality of the physical layer is to transmit the individual bits from one node to another node.
- It is the lowest layer of the OSI model.
- It establishes, maintains and deactivates the physical connection.
- It specifies the mechanical, electrical and procedural network interface specifications.

Functions of a Physical layer:

- **Line Configuration:** It defines the way how two or more devices can be connected physically.
- **Data Transmission:** It defines the transmission mode whether it is simplex, half-duplex or full-duplex mode between the two devices on the network.
- **Topology:** It defines the way how network devices are arranged.
- **Signals:** It determines the type of the signal used for transmitting the information.

Data-Link Layer



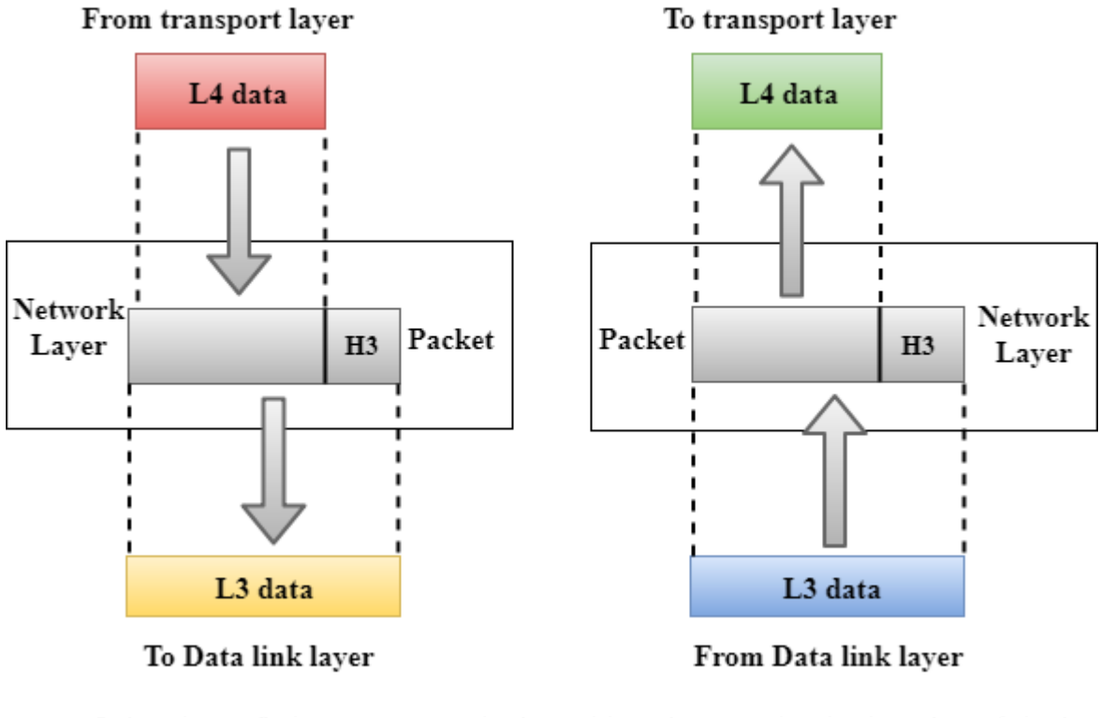
- This layer is responsible for the error-free transfer of data frames.
- It defines the format of the data on the network.
- It provides a reliable and efficient communication between two or more devices.
- It is mainly responsible for the unique identification of each device that resides on a local network.
- It contains two sub-layers:
  - **Logical Link Control Layer**
    - It is responsible for transferring the packets to the Network layer of the receiver that is receiving.
    - It identifies the address of the network layer protocol from the header.
    - It also provides flow control.
  - **Media Access Control Layer**
    - A Media access control layer is a link between the Logical Link Control layer and the network's physical layer.
    - It is used for transferring the packets over the network.

Functions of the Data-link layer

- **Framing:** The data link layer translates the physical's raw bit stream into packets known as Frames. The Data link layer adds the header and trailer to the frame. The header which is added to the frame contains the hardware destination and source address.

- **Physical Addressing:** The Data link layer adds a header to the frame that contains a destination address. The frame is transmitted to the destination address mentioned in the header.
- **Flow Control:** Flow control is the main functionality of the Data-link layer. It is the technique through which the constant data rate is maintained on both the sides so that no data get corrupted. It ensures that the transmitting station such as a server with higher processing speed does not exceed the receiving station, with lower processing speed.
- **Error Control:** Error control is achieved by adding a calculated value CRC (Cyclic Redundancy Check) that is placed to the Data link layer's trailer which is added to the message frame before it is sent to the physical layer. If any error seems to occurr, then the receiver sends the acknowledgment for the retransmission of the corrupted frames.
- **Access Control:** When two or more devices are connected to the same communication channel, then the data link layer protocols are used to determine which device has control over the link at a given time.

Network Layer

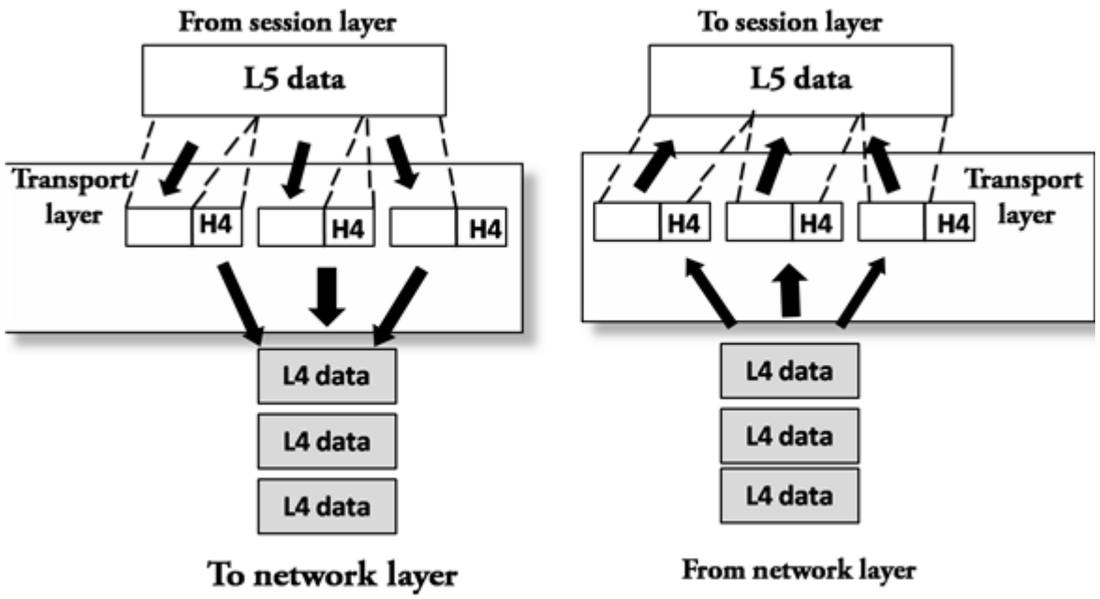


- It is a layer 3 that manages device addressing, tracks the location of devices on the network.
- It determines the best path to move data from source to the destination based on the network conditions, the priority of service, and other factors.
- The Data link layer is responsible for routing and forwarding the packets.
- Routers are the layer 3 devices, they are specified in this layer and used to provide the routing services within an internetwork.
- The protocols used to route the network traffic are known as Network layer protocols. Examples of protocols are IP and Ipv6.

Functions of Network Layer:

- **Internetworking:** An internetworking is the main responsibility of the network layer. It provides a logical connection between different devices.
- **Addressing:** A Network layer adds the source and destination address to the header of the frame. Addressing is used to identify the device on the internet.
- **Routing:** Routing is the major component of the network layer, and it determines the best optimal path out of the multiple paths from source to the destination.
- **Packetizing:** A Network Layer receives the packets from the upper layer and converts them into packets. This process is known as Packetizing. It is achieved by internet protocol (IP).

Transport Layer



- The Transport layer is a Layer 4 ensures that messages are transmitted in the order in which they are sent and there is no duplication of data.
- The main responsibility of the transport layer is to transfer the data completely.
- It receives the data from the upper layer and converts them into smaller units known as segments.
- This layer can be termed as an end-to-end layer as it provides a point-to-point connection between source and destination to deliver the data reliably.

The two protocols used in this layer are:

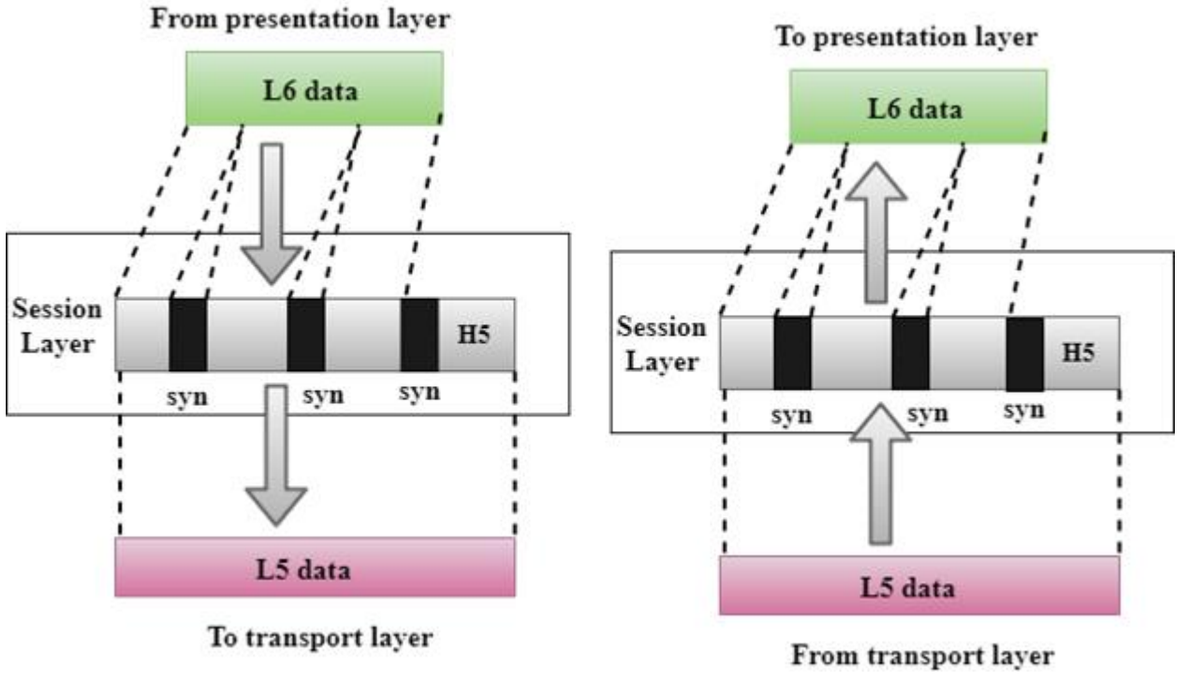
- **Transmission Control Protocol**
  - It is a standard protocol that allows the systems to communicate over the internet.
  - It establishes and maintains a connection between hosts.
  - When data is sent over the TCP connection, then the TCP protocol divides the data into smaller units known as segments. Each segment travels over the internet using multiple routes, and they arrive in different orders at the destination. The transmission control protocol reorders the packets in the correct order at the receiving end.
- **User Datagram Protocol**
  - User Datagram Protocol is a transport layer protocol.

- It is an unreliable transport protocol as in this case receiver does not send any acknowledgment when the packet is received, the sender does not wait for any acknowledgment. Therefore, this makes a protocol unreliable.

Functions of Transport Layer:

- **Service-point addressing:** Computers run several programs simultaneously due to this reason, the transmission of data from source to the destination not only from one computer to another computer but also from one process to another process. The transport layer adds the header that contains the address known as a service-point address or port address. The responsibility of the network layer is to transmit the data from one computer to another computer and the responsibility of the transport layer is to transmit the message to the correct process.
- **Segmentation and reassembly:** When the transport layer receives the message from the upper layer, it divides the message into multiple segments, and each segment is assigned with a sequence number that uniquely identifies each segment. When the message has arrived at the destination, then the transport layer reassembles the message based on their sequence numbers.
- **Connection control:** Transport layer provides two services Connection-oriented service and connectionless service. A connectionless service treats each segment as an individual packet, and they all travel in different routes to reach the destination. A connection-oriented service makes a connection with the transport layer at the destination machine before delivering the packets. In connection-oriented service, all the packets travel in the single route.
- **Flow control:** The transport layer also responsible for flow control but it is performed end-to-end rather than across a single link.
- **Error control:** The transport layer is also responsible for Error control. Error control is performed end-to-end rather than across the single link. The sender transport layer ensures that message reach at the destination without any error.

Session Layer

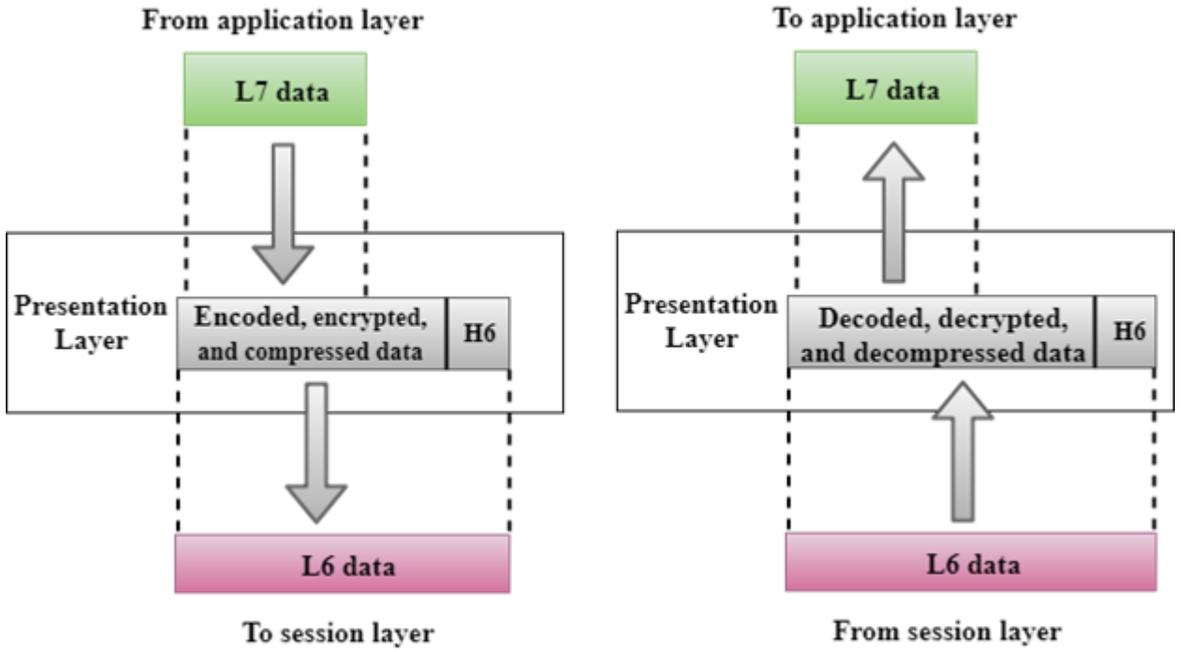


- It is a layer 3 in the OSI model.
- The Session layer is used to establish, maintain and synchronizes the interaction between communicating devices.

Functions of Session layer:

- **Dialog control:** Session layer acts as a dialog controller that creates a dialog between two processes or we can say that it allows the communication between two processes which can be either half-duplex or full-duplex.
- **Synchronization:** Session layer adds some checkpoints when transmitting the data in a sequence. If some error occurs in the middle of the transmission of data, then the transmission will take place again from the checkpoint. This process is known as Synchronization and recovery.

Presentation Layer

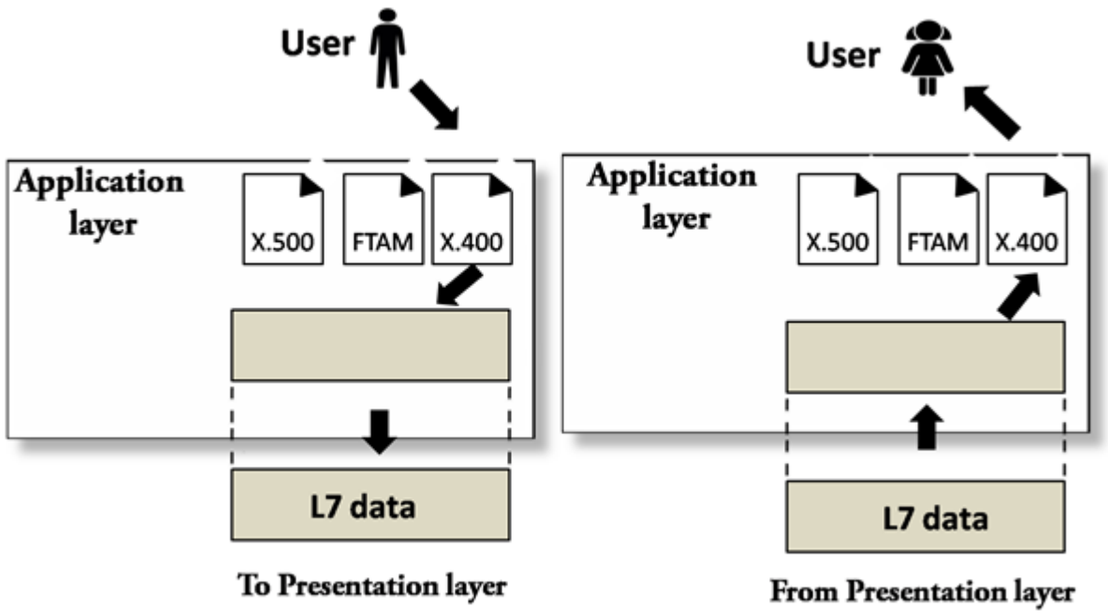


- A Presentation layer is mainly concerned with the syntax and semantics of the information exchanged between the two systems.
- It acts as a data translator for a network.
- This layer is a part of the operating system that converts the data from one presentation format to another format.
- The Presentation layer is also known as the syntax layer.

Functions of Presentation layer:

- **Translation:** The processes in two systems exchange the information in the form of character strings, numbers and so on. Different computers use different encoding methods, the presentation layer handles the interoperability between the different encoding methods. It converts the data from sender-dependent format into a common format and changes the common format into receiver-dependent format at the receiving end.
- **Encryption:** Encryption is needed to maintain privacy. Encryption is a process of converting the sender-transmitted information into another form and sends the resulting message over the network.
- **Compression:** Data compression is a process of compressing the data, i.e., it reduces the number of bits to be transmitted. Data compression is very important in multimedia such as text, audio, video.

Application Layer



- An application layer serves as a window for users and application processes to access network service.
- It handles issues such as network transparency, resource allocation, etc.
- An application layer is not an application, but it performs the application layer functions.
- This layer provides the network services to the end-users.

Functions of Application layer:

- **File transfer, access, and management (FTAM):** An application layer allows a user to access the files in a remote computer, to retrieve the files from a computer and to manage the files in a remote computer.
- **Mail services:** An application layer provides the facility for email forwarding and storage.
- Directory services: An application provides the distributed database sources and is used to provide that global information about various objects.

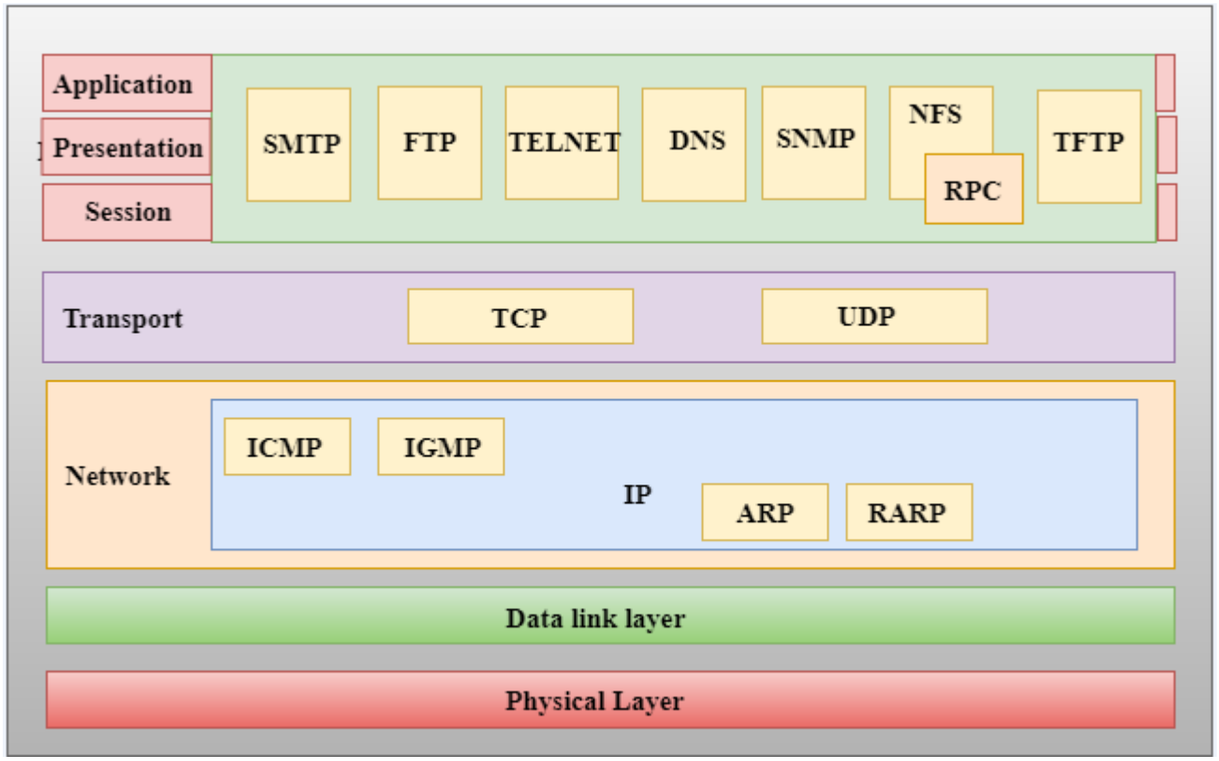
TCP/IP model

- The TCP/IP model was developed prior to the OSI model.

- The TCP/IP model is not exactly similar to the OSI model.
- The TCP/IP model consists of five layers: the application layer, transport layer, network layer, data link layer and physical layer.
- The first four layers provide physical standards, network interface, internetworking, and transport functions that correspond to the first four layers of the OSI model and these four layers are represented in TCP/IP model by a single layer called the application layer.
- TCP/IP is a hierarchical protocol made up of interactive modules, and each of them provides specific functionality.

Here, hierarchical means that each upper-layer protocol is supported by two or more lower-level protocols.

Functions of TCP/IP layers:



Network Access Layer

- A network layer is the lowest layer of the TCP/IP model.
- A network layer is the combination of the Physical layer and Data Link layer defined in the OSI reference model.

- It defines how the data should be sent physically through the network.
- This layer is mainly responsible for the transmission of the data between two devices on the same network.
- The functions carried out by this layer are encapsulating the IP datagram into frames transmitted by the network and mapping of IP addresses into physical addresses.
- The protocols used by this layer are ethernet, token ring, FDDI, X.25, frame relay.

Internet Layer

- An internet layer is the second layer of the TCP/IP model.
- An internet layer is also known as the network layer.
- The main responsibility of the internet layer is to send the packets from any network, and they arrive at the destination irrespective of the route they take.

Following are the protocols used in this layer are:

IP Protocol: IP protocol is used in this layer, and it is the most significant part of the entire TCP/IP suite.

Following are the responsibilities of this protocol:

- **IP Addressing:** This protocol implements logical host addresses known as IP addresses. The IP addresses are used by the internet and higher layers to identify the device and to provide internetwork routing.
- **Host-to-host communication:** It determines the path through which the data is to be transmitted.
- **Data Encapsulation and Formatting:** An IP protocol accepts the data from the transport layer protocol. An IP protocol ensures that the data is sent and received securely, it encapsulates the data into message known as IP datagram.
- **Fragmentation and Reassembly:** The limit imposed on the size of the IP datagram by data link layer protocol is known as Maximum Transmission unit (MTU). If the size of IP datagram is greater than the MTU unit, then the IP protocol splits the datagram into smaller units so that they can travel over the local network. Fragmentation can be done by the sender or intermediate router. At the receiver side, all the fragments are reassembled to form an original message.
- **Routing:** When IP datagram is sent over the same local network such as LAN, MAN, WAN, it is known as direct delivery. When source and destination are on the distant network, then the IP datagram is sent indirectly. This can be accomplished by routing the IP datagram through various devices such as routers.

ARP Protocol

- ARP stands for **Address Resolution Protocol**.

- ARP is a network layer protocol which is used to find the physical address from the IP address.
- **The two terms are mainly associated with the ARP Protocol:**
  - **ARP request:** When a sender wants to know the physical address of the device, it broadcasts the ARP request to the network.
  - **ARP reply:** Every device attached to the network will accept the ARP request and process the request, but only recipient recognize the IP address and sends back its physical address in the form of ARP reply. The recipient adds the physical address both to its cache memory and to the datagram header

ICMP Protocol

- **ICMP** stands for Internet Control Message Protocol.
- It is a mechanism used by the hosts or routers to send notifications regarding datagram problems back to the sender.
- A datagram travels from router-to-router until it reaches its destination. If a router is unable to route the data because of some unusual conditions such as disabled links, a device is on fire or network congestion, then the ICMP protocol is used to inform the sender that the datagram is undeliverable.
- An ICMP protocol mainly uses two terms:
  - **ICMP Test:** ICMP Test is used to test whether the destination is reachable or not.
  - **ICMP Reply:** ICMP Reply is used to check whether the destination device is responding or not.
- The core responsibility of the ICMP protocol is to report the problems, not correct them. The responsibility of the correction lies with the sender.
- ICMP can send the messages only to the source, but not to the intermediate routers because the IP datagram carries the addresses of the source and destination but not of the router that it is passed to.

Transport Layer

The transport layer is responsible for the reliability, flow control, and correction of data which is being sent over the network.

The two protocols used in the transport layer are **User Datagram protocol and Transmission control protocol**.

- **User Datagram Protocol (UDP)**
  - It provides connectionless service and end-to-end delivery of transmission.
  - It is an unreliable protocol as it discovers the errors but not specify the error.
  - User Datagram Protocol discovers the error, and ICMP protocol reports the error to the sender that user datagram has been damaged.

- **UDP consists of the following fields:**

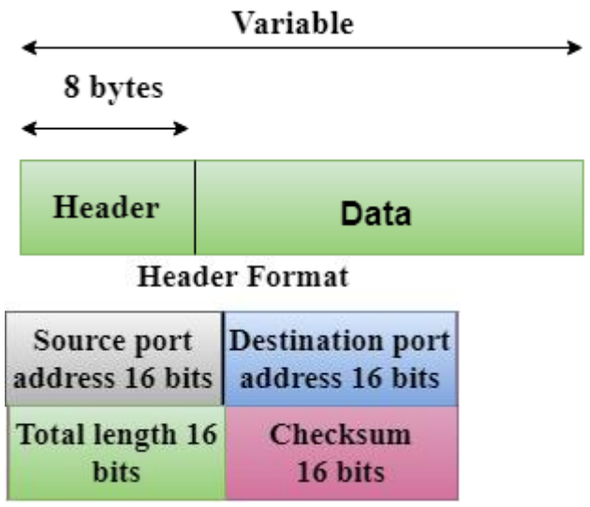
  Source port address: The source port address is the address of the application program that has created the message.

  Destination port address: The destination port address is the address of the application program that receives the message.

  Total length: It defines the total number of bytes of the user datagram in bytes.

  Checksum: The checksum is a 16-bit field used in error detection.

- UDP does not specify which packet is lost. UDP contains only checksum; it does not contain any ID of a data segment.



- **Transmission Control Protocol (TCP)**
  - It provides a full transport layer services to applications.
  - It creates a virtual circuit between the sender and receiver, and it is active for the duration of the transmission.
  - TCP is a reliable protocol as it detects the error and retransmits the damaged frames. Therefore, it ensures all the segments must be received and acknowledged before the transmission is considered to be completed and a virtual circuit is discarded.
  - At the sending end, TCP divides the whole message into smaller units known as segment, and each segment contains a sequence number which is required for reordering the frames to form an original message.
  - At the receiving end, TCP collects all the segments and reorders them based on sequence numbers.

Application Layer

- An application layer is the topmost layer in the TCP/IP model.
- It is responsible for handling high-level protocols, issues of representation.
- This layer allows the user to interact with the application.
- When one application layer protocol wants to communicate with another application layer, it forwards its data to the transport layer.
- There is an ambiguity occurs in the application layer. Every application cannot be placed inside the application layer except those who interact with the communication system. For example: text editor cannot be considered in application layer while web browser using **HTTP** protocol to interact with the network where **HTTP** protocol is an application layer protocol.

Following are the main protocols used in the application layer:

- **HTTP:** HTTP stands for Hypertext transfer protocol. This protocol allows us to access the data over the world wide web. It transfers the data in the form of plain text, audio, video. It is known as a Hypertext transfer protocol as it has the efficiency to use in a hypertext environment where there are rapid jumps from one document to another.
- **SNMP:** SNMP stands for Simple Network Management Protocol. It is a framework used for managing the devices on the internet by using the TCP/IP protocol suite.
- **SMTP:** SMTP stands for Simple mail transfer protocol. The TCP/IP protocol that supports the e-mail is known as a Simple mail transfer protocol. This protocol is used to send the data to another e-mail address.
- **DNS:** DNS stands for Domain Name System. An IP address is used to identify the connection of a host to the internet uniquely. But, people prefer to use the names instead of addresses. Therefore, the system that maps the name to the address is known as Domain Name System.
- **TELNET:** It is an abbreviation for Terminal Network. It establishes the connection between the local computer and remote computer in such a way that the local terminal appears to be a terminal at the remote system.
- **FTP:** FTP stands for File Transfer Protocol. FTP is a standard internet protocol used for transmitting the files from one computer to another computer.

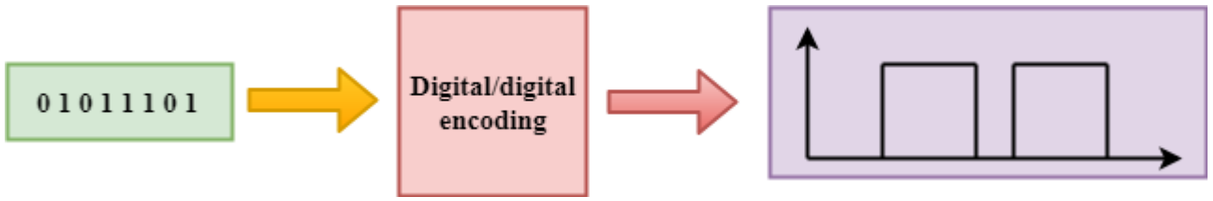
Physical Layer

Digital Transmission

Data can be represented either in analog or digital form. The computers used the digital form to store the information. Therefore, the data needs to be converted in digital form so that it can be used by a computer.

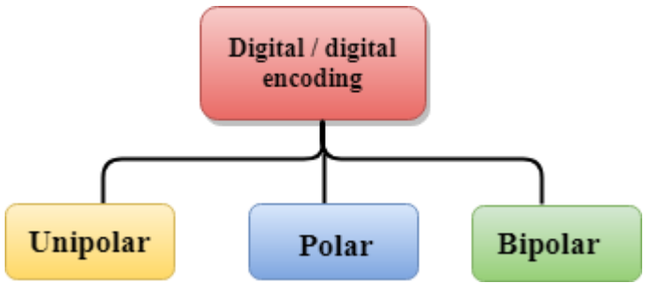
DIGITAL-TO-DIGITAL CONVERSION

Digital-to-digital encoding is the representation of digital information by a digital signal. When binary 1s and 0s generated by the computer are translated into a sequence of voltage pulses that can be propagated over a wire, this process is known as digital-to-digital encoding.



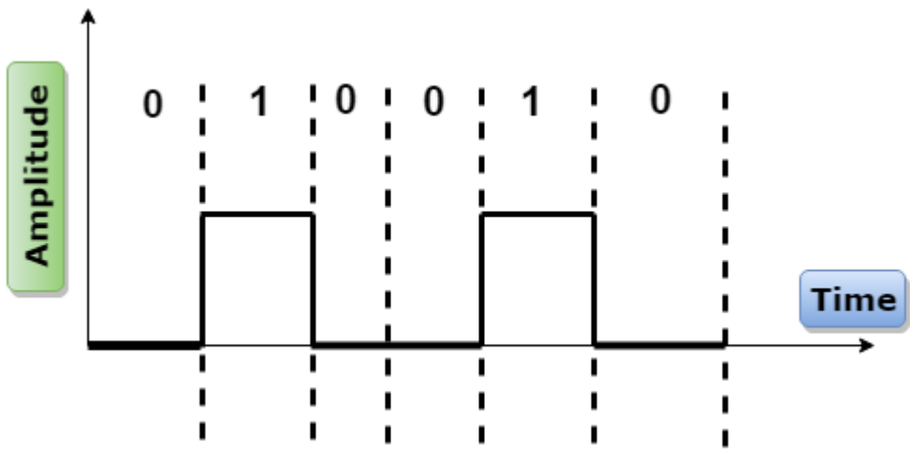
Digital-to-digital encoding is divided into three categories:

- Unipolar Encoding
- Polar Encoding
- Bipolar Encoding



Unipolar

- Digital transmission system sends the voltage pulses over the medium link such as wire or cable.
- In most types of encoding, one voltage level represents 0, and another voltage level represents 1.
- The polarity of each pulse determines whether it is positive or negative.
- This type of encoding is known as Unipolar encoding as it uses only one polarity.
- In Unipolar encoding, the polarity is assigned to the 1 binary state.
- In this, 1s are represented as a positive value and 0s are represented as a zero value.
- In Unipolar Encoding, '1' is considered as a high voltage and '0' is considered as a zero voltage.
- Unipolar encoding is simpler and inexpensive to implement.

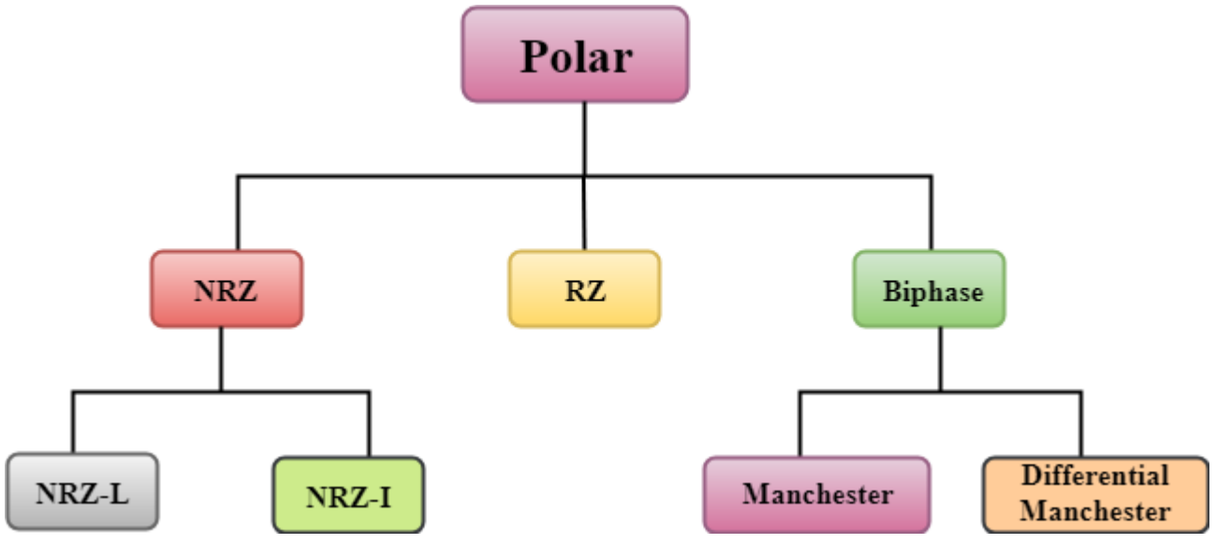


Unipolar encoding has two problems that make this scheme less desirable:

- DC Component
- Synchronization

Polar

- Polar encoding is an encoding scheme that uses two voltage levels: one is positive, and another is negative.
- By using two voltage levels, an average voltage level is reduced, and the DC component problem of unipolar encoding scheme is alleviated.



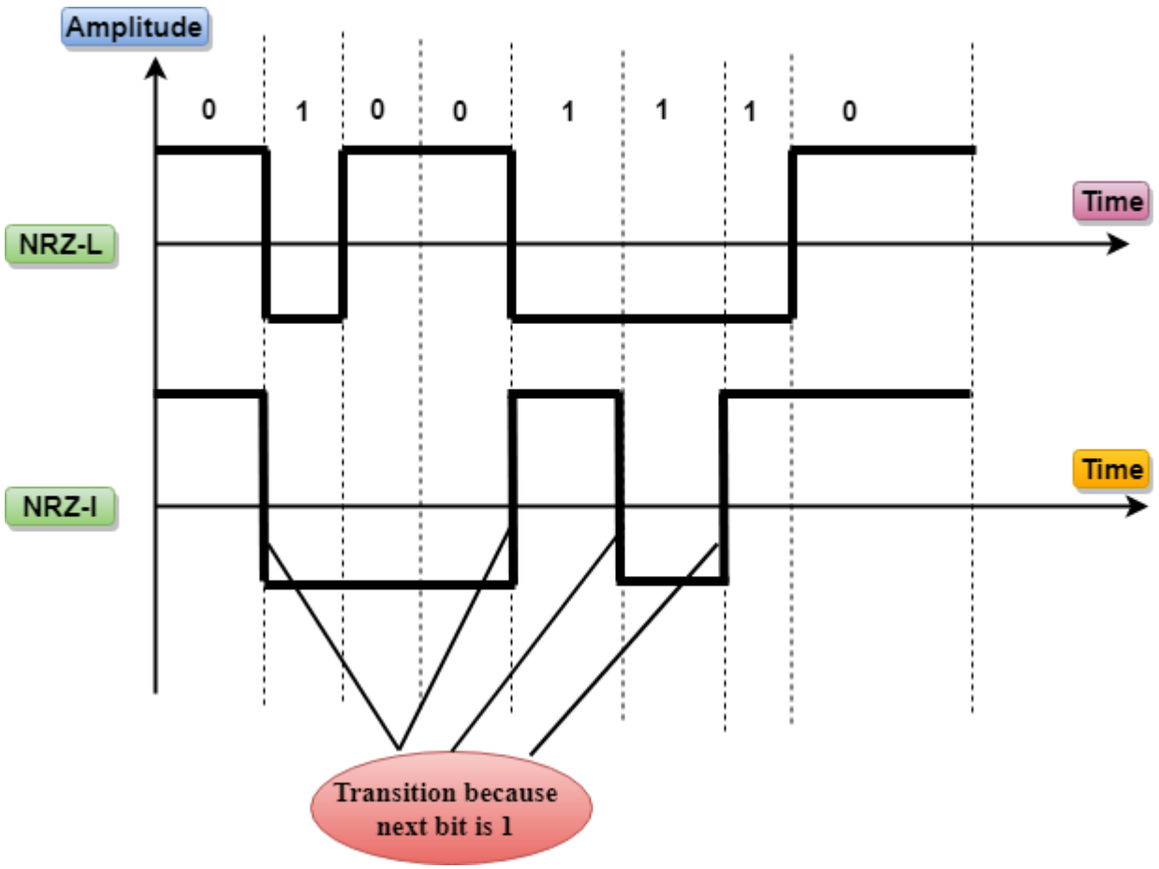
NRZ

- NRZ stands for Non-return zero.
- In NRZ encoding, the level of the signal can be represented either positive or negative.

The two most common methods used in NRZ are:

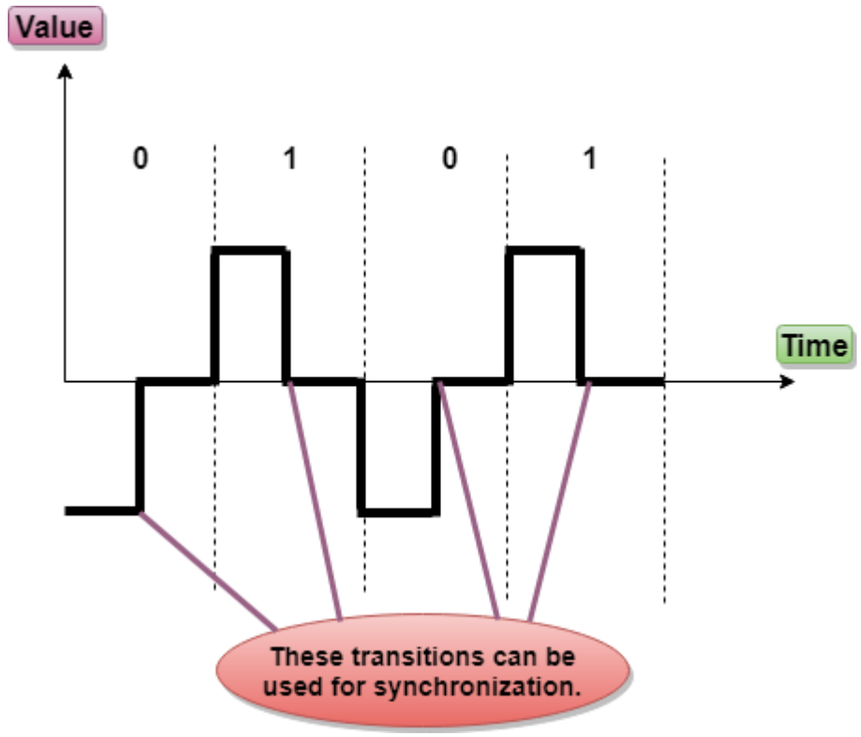
NRZ-L: In NRZ-L encoding, the level of the signal depends on the type of the bit that it represents. If a bit is 0 or 1, then their voltages will be positive and negative respectively. Therefore, we can say that the level of the signal is dependent on the state of the bit.

NRZ-I: NRZ-I is an inversion of the voltage level that represents 1 bit. In the NRZ-I encoding scheme, a transition occurs between the positive and negative voltage that represents 1 bit. In this scheme, 0 bit represents no change and 1 bit represents a change in voltage level.



RZ

- RZ stands for Return to zero.
- There must be a signal change for each bit to achieve synchronization. However, to change with every bit, we need to have three values: positive, negative and zero.
- RZ is an encoding scheme that provides three values, positive voltage represents 1, the negative voltage represents 0, and zero voltage represents none.
- In the RZ scheme, halfway through each interval, the signal returns to zero.
- In RZ scheme, 1 bit is represented by positive-to-zero and 0 bit is represented by negative-to-zero.



Disadvantage of RZ:

It performs two signal changes to encode one bit that acquires more bandwidth.

Biphase

- Biphase is an encoding scheme in which signal changes at the middle of the bit interval but does not return to zero.

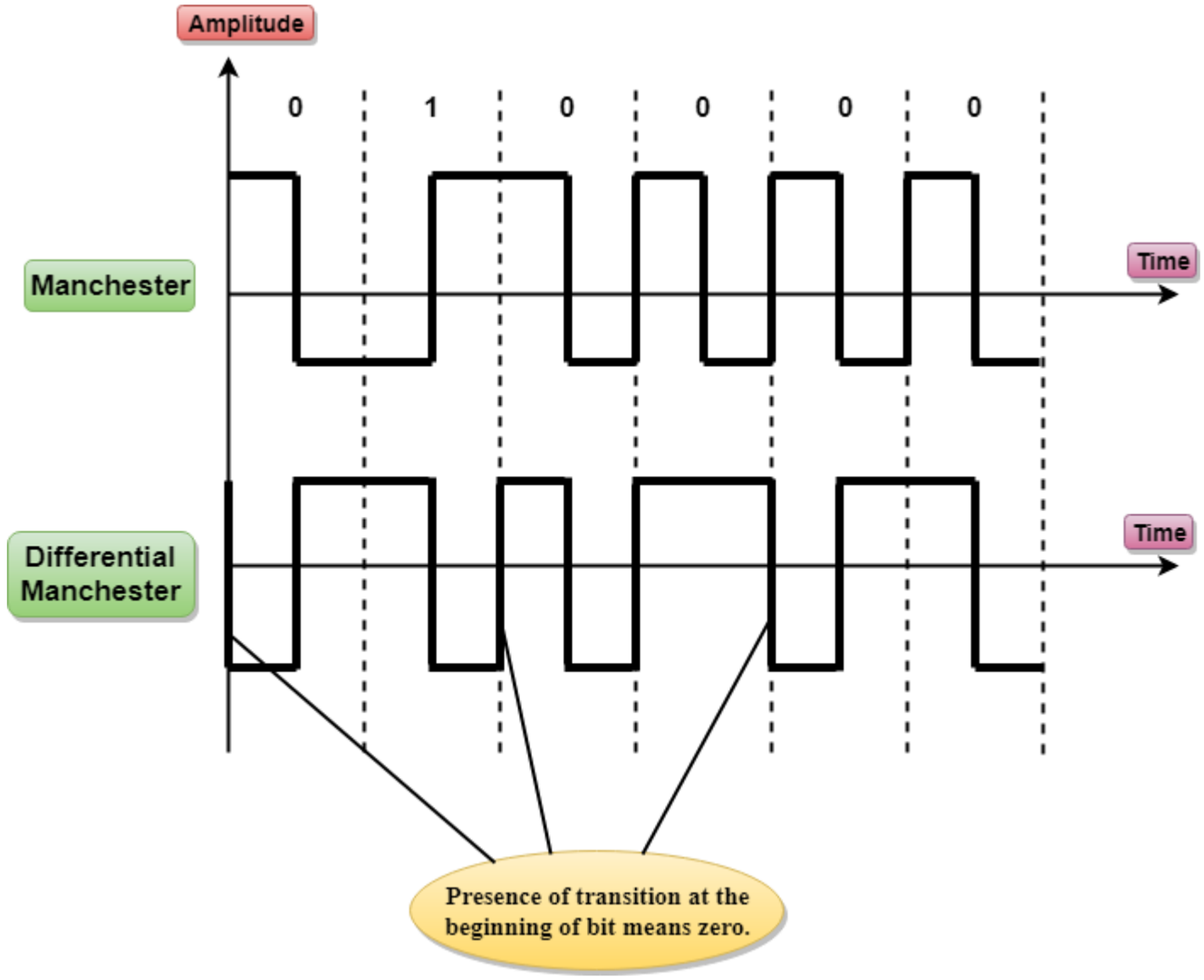
Biphase encoding is implemented in two different ways:

Manchester

- It changes the signal at the middle of the bit interval but does not return to zero for synchronization.
- In Manchester encoding, a negative-to-positive transition represents binary 1, and positive-to-negative transition represents 0.
- Manchester has the same level of synchronization as RZ scheme except that it has two levels of amplitude.

Differential Manchester

- It changes the signal at the middle of the bit interval for synchronization, but the presence or absence of the transition at the beginning of the interval determines the bit. A transition means binary 0 and no transition means binary 1.
- In Manchester Encoding scheme, two signal changes represent 0 and one signal change represent 1.

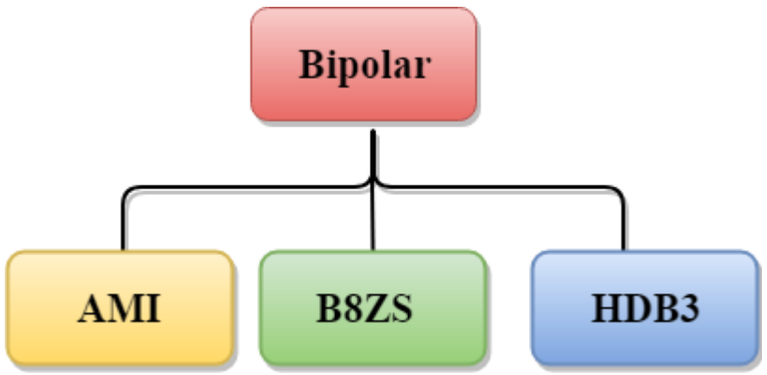


Bipolar

- Bipolar encoding scheme represents three voltage levels: positive, negative, and zero.

- In Bipolar encoding scheme, zero level represents binary 0, and binary 1 is represented by alternating positive and negative voltages.
- If the first 1 bit is represented by positive amplitude, then the second 1 bit is represented by negative voltage, third 1 bit is represented by the positive amplitude and so on. This alternation can also occur even when the 1bits are not consecutive.

Bipolar can be classified as:



AMI

- AMI stands for ***alternate mark inversion*** where mark work comes from telegraphy which means 1. So, it can be redefined as **alternate 1 inversion**.
- In Bipolar AMI encoding scheme, 0 bit is represented by zero level and 1 bit is represented by alternating positive and negative voltages.

Advantage:

- DC component is zero.
- Sequence of 1s bits are synchronized.

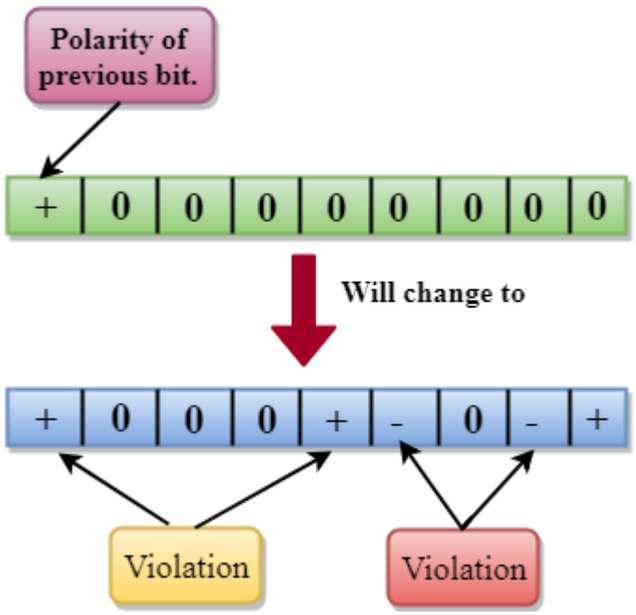
Disadvantage:

- This encoding scheme does not ensure the synchronization of a long string of 0s bits.

B8ZS

- B8ZS stands for **Bipolar 8-Zero Substitution**.
- This technique is adopted in North America to provide synchronization of a long sequence of 0s bits.
- In most of the cases, the functionality of B8ZS is similar to the bipolar AMI, but the only difference is that it provides the synchronization when a long sequence of 0s bits occur.

- B8ZS ensures synchronization of a long string of 0s by providing force artificial signal changes called violations, within 0 string pattern.
- When eight 0 occurs, then B8ZS implements some changes in 0s string pattern based on the polarity of the previous 1 bit.
- If the polarity of the previous 1 bit is positive, the eight 0s will be encoded as zero, zero, zero, positive, negative, zero, negative, positive.

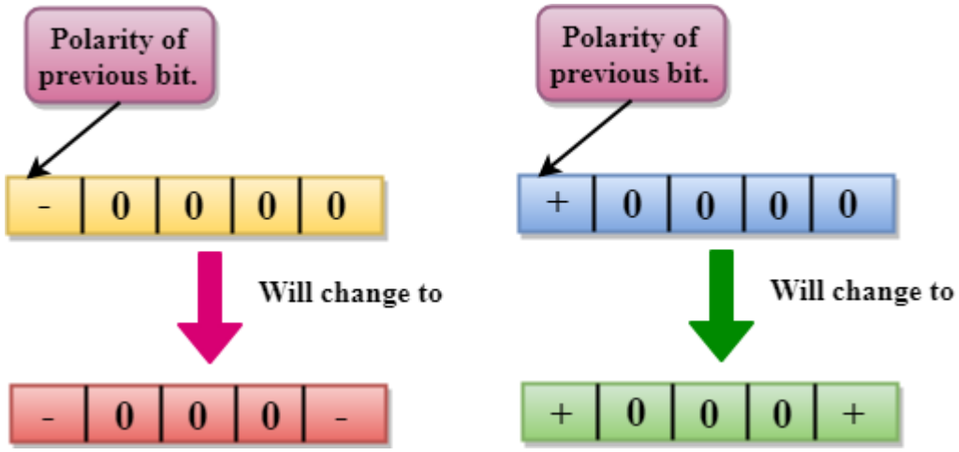


- If the polarity of previous 1 bit is negative, then the eight 0s will be encoded as zero, zero, zero, negative, positive, zero, positive, negative.

HDB3

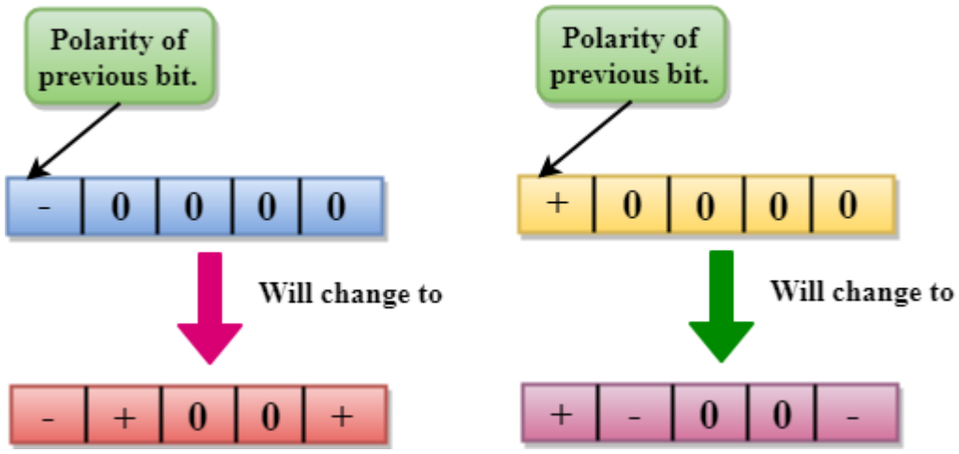
- HDB3 stands for **High-Density Bipolar 3**.
- HDB3 technique was first adopted in Europe and Japan.
- HDB3 technique is designed to provide the synchronization of a long sequence of 0s bits.
- In the HDB3 technique, the pattern of violation is based on the polarity of the previous bit.
- When four 0s occur, HDB3 looks at the number of 1s bits occurred since the last substitution.
- If the number of 1s bits is odd, then the violation is made on the fourth consecutive of 0. If the polarity of the previous bit is positive, then the violation is positive. If the polarity of the previous bit is negative, then the violation is negative.

If the number of 1s bits since the last substitution is odd.



If the number of 1s bits is even, then the violation is made on the place of the first and fourth consecutive 0s. If the polarity of the previous bit is positive, then violations are negative, and if the polarity of the previous bit is negative, then violations are positive.

If the number of 1s bits since the last substitution is even.



ANALOG-TO-DIGITAL CONVERSION

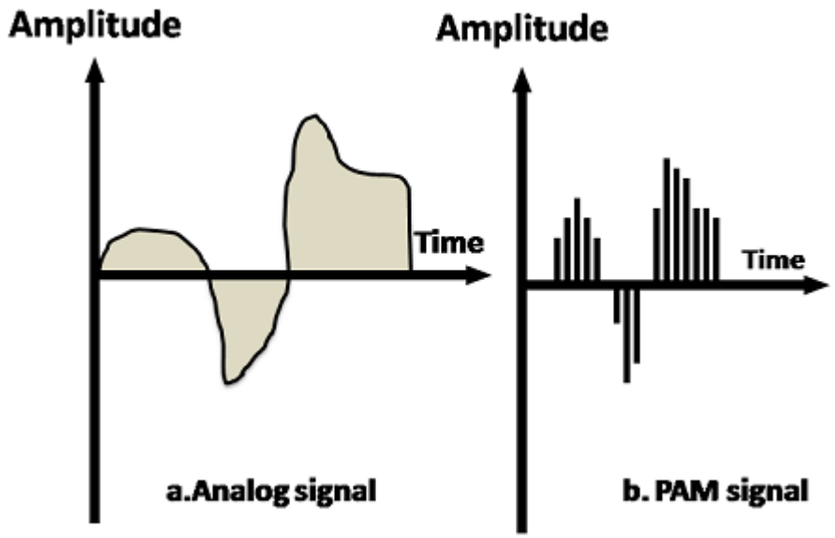
- When an analog signal is digitalized, this is called an analog-to-digital conversion.
- Suppose human sends a voice in the form of an analog signal, we need to digitalize the analog signal which is less prone to noise. It requires a reduction in the number of values in an analog message so that they can be represented in the digital stream.

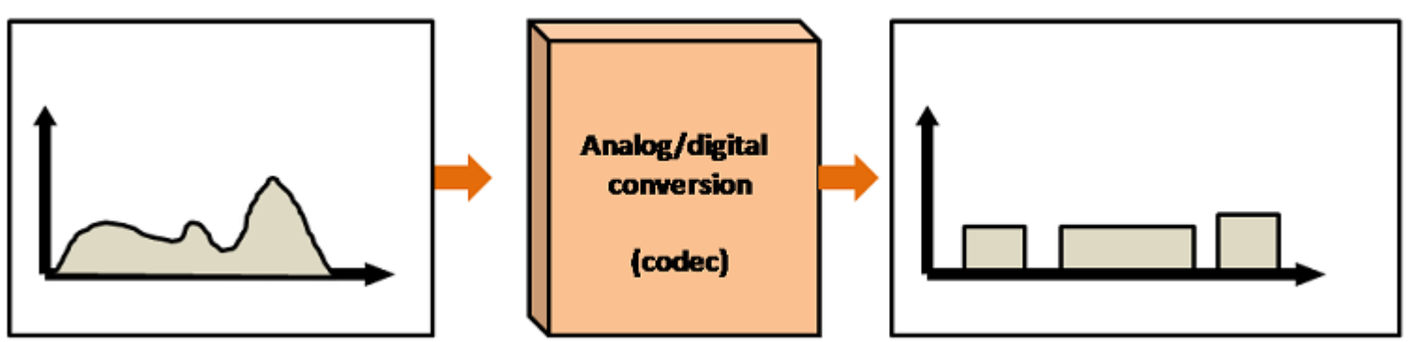
- In analog-to-digital conversion, the information contained in a continuous wave form is converted in digital pulses.

Techniques for Analog-To-Digital Conversion

PAM

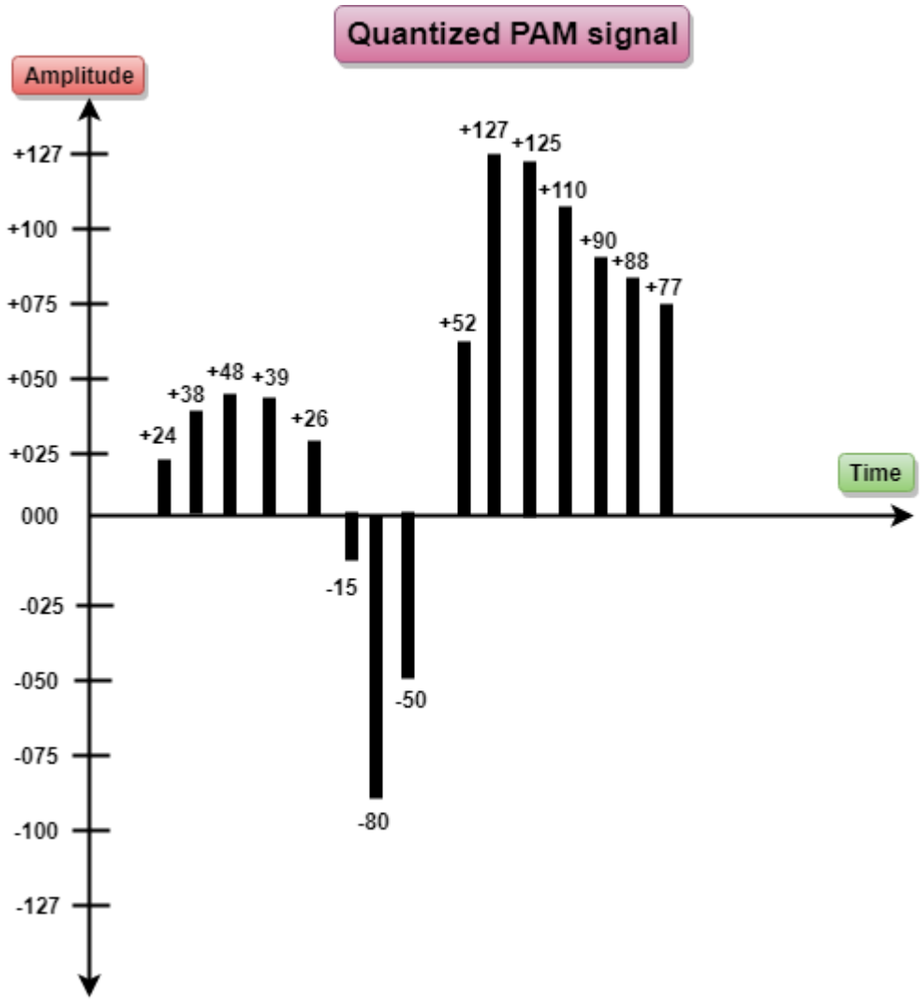
- PAM stands for **pulse amplitude modulation**.
- PAM is a technique used in analog-to-digital conversion.
- PAM technique takes an analog signal, samples it, and generates a series of digital pulses based on the result of sampling where sampling means measuring the amplitude of a signal at equal intervals.
- PAM technique is not useful in data communication as it translates the original wave form into pulses, but these pulses are not digital. To make them digital, PAM technique is modified to PCM technique.



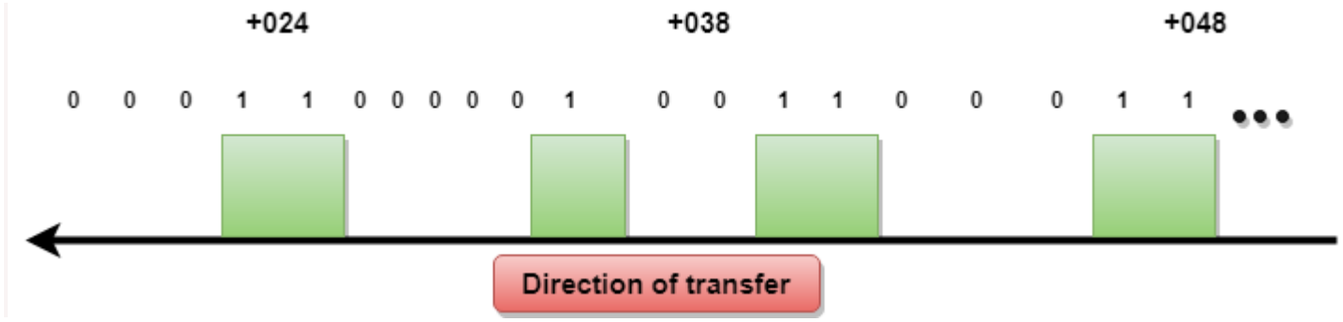


PCM

- PCM stands for **Pulse Code Modulation**.
- PCM technique is used to modify the pulses created by PAM to form a digital signal. To achieve this, PCM quantizes PAM pulses. Quantization is a process of assigning integral values in a specific range to sampled instances.
- PCM is made of four separate processes: PAM, quantization, binary encoding, and digital-to-digital encoding.



PCM



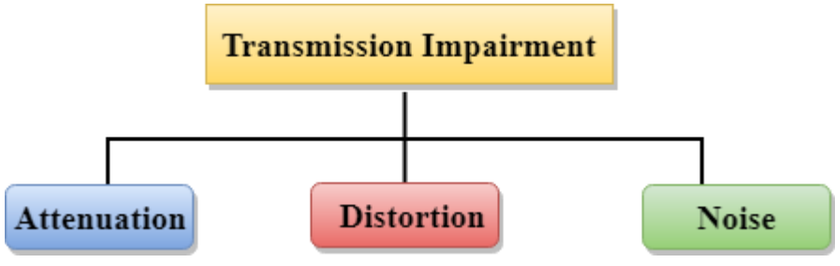
What is Transmission media?

- Transmission media is a communication channel that carries the information from the sender to the receiver. Data is transmitted through the electromagnetic signals.
- The main functionality of the transmission media is to carry the information in the form of bits through **LAN**(Local Area Network).
- It is a physical path between transmitter and receiver in data communication.
- In a copper-based network, the bits in the form of electrical signals.
- In a fibre based network, the bits in the form of light pulses.
- In **OSI**(Open System Interconnection) phase, transmission media supports the Layer 1. Therefore, it is considered to be as a Layer 1 component.
- The electrical signals can be sent through the copper wire, fibre optics, atmosphere, water, and vacuum.
- The characteristics and quality of data transmission are determined by the characteristics of medium and signal.
- Transmission media is of two types are wired media and wireless media. In wired media, medium characteristics are more important whereas, in wireless media, signal characteristics are more important.
- Different transmission media have different properties such as bandwidth, delay, cost and ease of installation and maintenance.
- The transmission media is available in the lowest layer of the OSI reference model, i.e., **Physical layer**.

Some factors need to be considered for designing the transmission media:

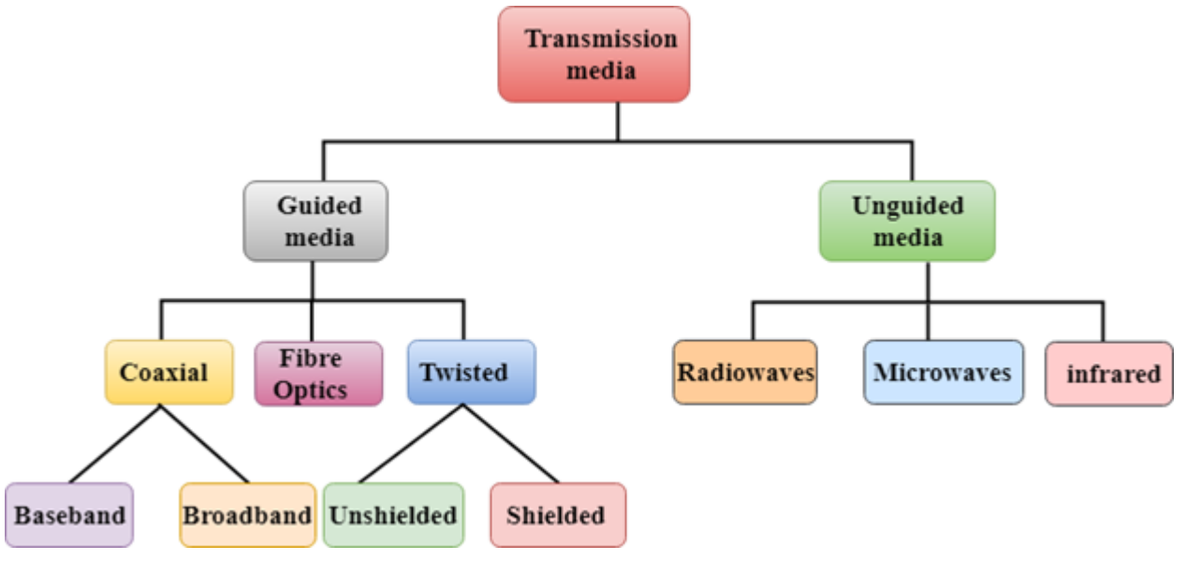
- **Bandwidth:** All the factors are remaining constant, the greater the bandwidth of a medium, the higher the data transmission rate of a signal.
- **Transmission impairment:** When the received signal is not identical to the transmitted one due to the transmission impairment. The quality of the signals will get destroyed due to transmission impairment.
- **Interference:** An interference is defined as the process of disrupting a signal when it travels over a communication medium on the addition of some unwanted signal.

Causes Of Transmission Impairment:



- **Attenuation:** Attenuation means the loss of energy, i.e., the strength of the signal decreases with increasing the distance which causes the loss of energy.
- **Distortion:** Distortion occurs when there is a change in the shape of the signal. This type of distortion is examined from different signals having different frequencies. Each frequency component has its own propagation speed, so they reach at a different time which leads to the delay distortion.
- **Noise:** When data is travelled over a transmission medium, some unwanted signal is added to it which creates the noise.

Classification Of Transmission Media:



- Guided Transmission Media
- UnGuided Transmission Media

Guided Media

It is defined as the physical medium through which the signals are transmitted. It is also known as Bounded media.

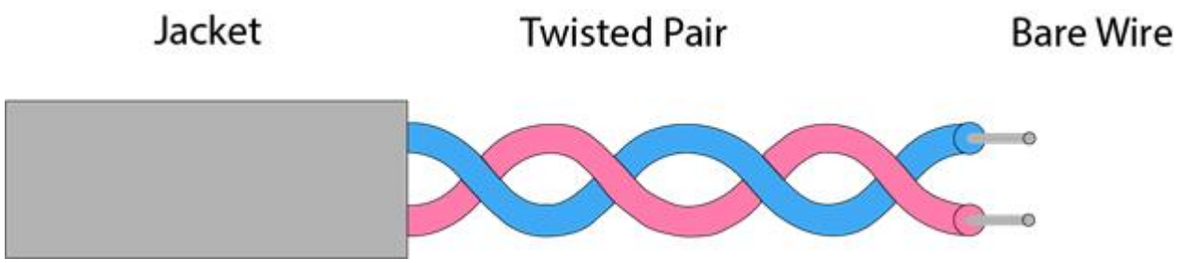
Types Of Guided media:

Twisted pair:

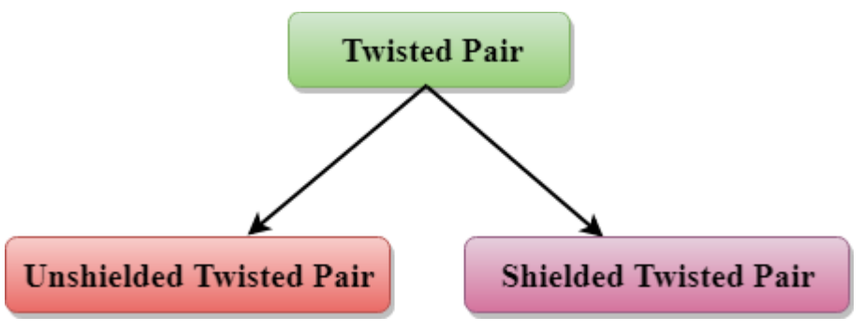
Twisted pair is a physical media made up of a pair of cables twisted with each other. A twisted pair cable is cheap as compared to other transmission media. Installation of the twisted pair cable is easy, and it is a lightweight cable. The frequency range for twisted pair cable is from 0 to 3.5KHz.

A twisted pair consists of two insulated copper wires arranged in a regular spiral pattern.

The degree of reduction in noise interference is determined by the number of turns per foot. Increasing the number of turns per foot decreases noise interference.



Types of Twisted pair:



Unshielded Twisted Pair:

An unshielded twisted pair is widely used in telecommunication. Following are the categories of the unshielded twisted pair cable:

- **Category 1:** Category 1 is used for telephone lines that have low-speed data.

- **Category 2:** It can support upto 4Mbps.
- **Category 3:** It can support upto 16Mbps.
- **Category 4:** It can support upto 20Mbps. Therefore, it can be used for long-distance communication.
- **Category 5:** It can support upto 200Mbps.

Advantages Of Unshielded Twisted Pair:

- It is cheap.
- Installation of the unshielded twisted pair is easy.
- It can be used for high-speed LAN.

Disadvantage:

- This cable can only be used for shorter distances because of attenuation.

Shielded Twisted Pair

A shielded twisted pair is a cable that contains the mesh surrounding the wire that allows the higher transmission rate.

Characteristics Of Shielded Twisted Pair:

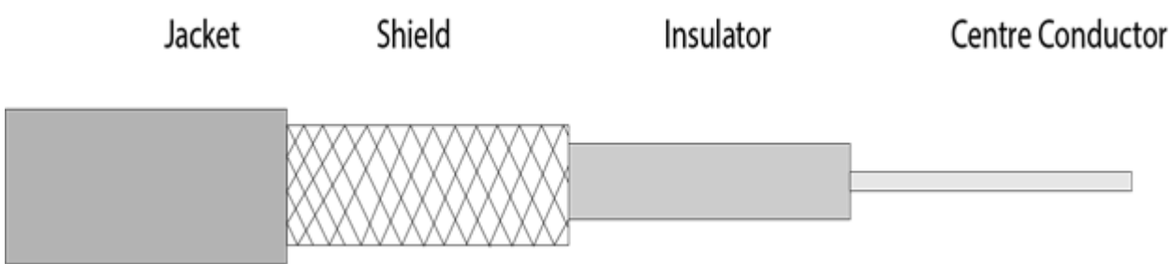
- The cost of the shielded twisted pair cable is not very high and not very low.
- An installation of STP is easy.
- It has higher capacity as compared to unshielded twisted pair cable.
- It has a higher attenuation.
- It is shielded that provides the higher data transmission rate.

Disadvantages

- It is more expensive as compared to UTP and coaxial cable.
- It has a higher attenuation rate.

Coaxial Cable

- Coaxial cable is very commonly used transmission media, for example, TV wire is usually a coaxial cable.
- The name of the cable is coaxial as it contains two conductors parallel to each other.
- It has a higher frequency as compared to Twisted pair cable.
- The inner conductor of the coaxial cable is made up of copper, and the outer conductor is made up of copper mesh. The middle core is made up of non-conductive cover that separates the inner conductor from the outer conductor.
- The middle core is responsible for the data transferring whereas the copper mesh prevents from the **EMI**(Electromagnetic interference).



Coaxial cable is of two types:

1. **Baseband transmission:** It is defined as the process of transmitting a single signal at high speed.
2. **Broadband transmission:** It is defined as the process of transmitting multiple signals simultaneously.

Advantages Of Coaxial cable:

- The data can be transmitted at high speed.
- It has better shielding as compared to twisted pair cable.
- It provides higher bandwidth.

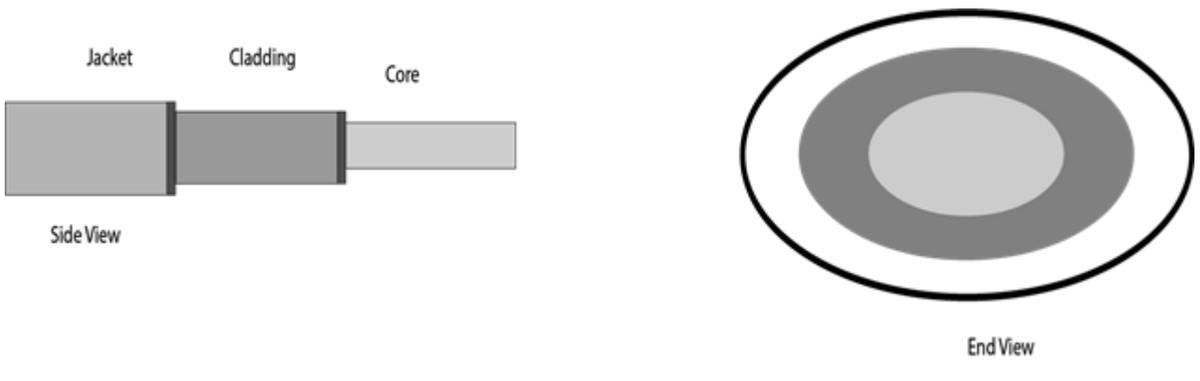
Disadvantages Of Coaxial cable:

- It is more expensive as compared to twisted pair cable.
- If any fault occurs in the cable causes the failure in the entire network.

Fibre Optic

- Fibre optic cable is a cable that uses electrical signals for communication.
- Fibre optic is a cable that holds the optical fibres coated in plastic that are used to send the data by pulses of light.
- The plastic coating protects the optical fibres from heat, cold, electromagnetic interference from other types of wiring.
- Fibre optics provide faster data transmission than copper wires.

Diagrammatic representation of fibre optic cable:



Basic elements of Fibre optic cable:

- **Core:** The optical fibre consists of a narrow strand of glass or plastic known as a core. A core is a light transmission area of the fibre. The more the area of the core, the more light will be transmitted into the fibre.
- **Cladding:** The concentric layer of glass is known as cladding. The main functionality of the cladding is to provide the lower refractive index at the core interface as to cause the reflection within the core so that the light waves are transmitted through the fibre.
- **Jacket:** The protective coating consisting of plastic is known as a jacket. The main purpose of a jacket is to preserve the fibre strength, absorb shock and extra fibre protection.

Following are the advantages of fibre optic cable over copper:

- **Greater Bandwidth:** The fibre optic cable provides more bandwidth as compared copper. Therefore, the fibre optic carries more data as compared to copper cable.
- **Faster speed:** Fibre optic cable carries the data in the form of light. This allows the fibre optic cable to carry the signals at a higher speed.

- **Longer distances:** The fibre optic cable carries the data at a longer distance as compared to copper cable.
- **Better reliability:** The fibre optic cable is more reliable than the copper cable as it is immune to any temperature changes while it can cause obstruct in the connectivity of copper cable.
- **Thinner and Sturdier:** Fibre optic cable is thinner and lighter in weight so it can withstand more pull pressure than copper cable.

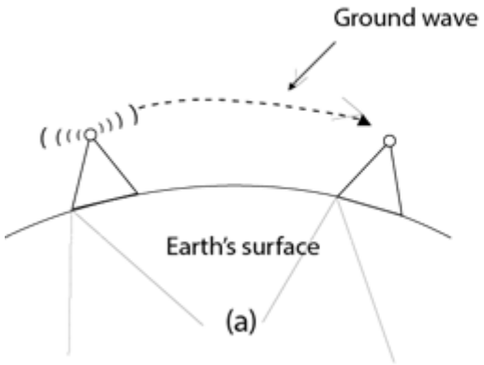
UnGuided Transmission

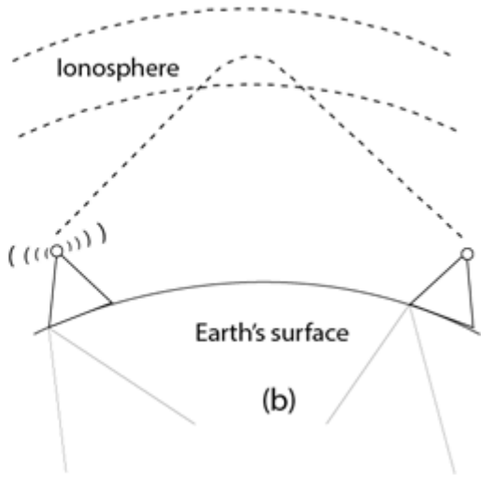
- An unguided transmission transmits the electromagnetic waves without using any physical medium. Therefore it is also known as **wireless transmission**.
- In unguided media, air is the media through which the electromagnetic energy can flow easily.

Unguided transmission is broadly classified into three categories:

Radio waves

- Radio waves are the electromagnetic waves that are transmitted in all the directions of free space.
- Radio waves are omnidirectional, i.e., the signals are propagated in all the directions.
- The range in frequencies of radio waves is from 3Khz to 1 khz.
- In the case of radio waves, the sending and receiving antenna are not aligned, i.e., the wave sent by the sending antenna can be received by any receiving antenna.
- An example of the radio wave is **FM radio**.





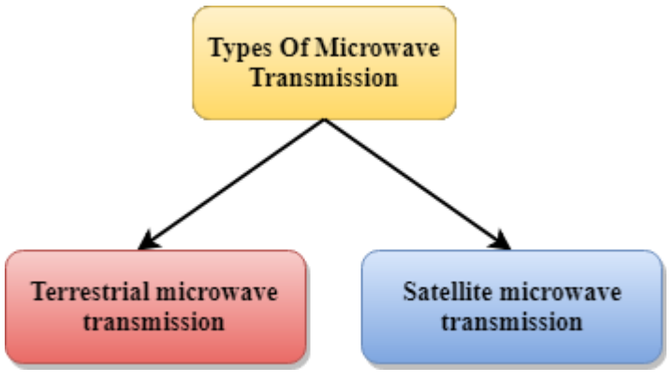
Applications Of Radio waves:

- A Radio wave is useful for multicasting when there is one sender and many receivers.
- An FM radio, television, cordless phones are examples of a radio wave.

Advantages Of Radio transmission:

- Radio transmission is mainly used for wide area networks and mobile cellular phones.
- Radio waves cover a large area, and they can penetrate the walls.
- Radio transmission provides a higher transmission rate.

Microwaves



Microwaves are of two types:

- Terrestrial microwave
- Satellite microwave communication.

Terrestrial Microwave Transmission

- Terrestrial Microwave transmission is a technology that transmits the focused beam of a radio signal from one ground-based microwave transmission antenna to another.
- Microwaves are the electromagnetic waves having the frequency in the range from 1GHz to 1000 GHz.

- Microwaves are unidirectional as the sending and receiving antenna is to be aligned, i.e., the waves sent by the sending antenna are narrowly focussed.
- In this case, antennas are mounted on the towers to send a beam to another antenna which is km away.
- It works on the line of sight transmission, i.e., the antennas mounted on the towers are the direct sight of each other.

Characteristics of Microwave:

- **Frequency range:** The frequency range of terrestrial microwave is from 4-6 GHz to 21-23 GHz.
- **Bandwidth:** It supports the bandwidth from 1 to 10 Mbps.
- **Short distance:** It is inexpensive for short distance.
- **Long distance:** It is expensive as it requires a higher tower for a longer distance.
- **Attenuation:** Attenuation means loss of signal. It is affected by environmental conditions and antenna size.

Advantages Of Microwave:

- Microwave transmission is cheaper than using cables.
- It is free from land acquisition as it does not require any land for the installation of cables.
- Microwave transmission provides an easy communication in terrains as the installation of cable in terrain is quite a difficult task.
- Communication over oceans can be achieved by using microwave transmission.

Disadvantages of Microwave transmission:

- **Eavesdropping:** An eavesdropping creates insecure communication. Any malicious user can catch the signal in the air by using its own antenna.
- **Out of phase signal:** A signal can be moved out of phase by using microwave transmission.
- **Susceptible to weather condition:** A microwave transmission is susceptible to weather condition. This means that any environmental change such as rain, wind can distort the signal.
- **Bandwidth limited:** Allocation of bandwidth is limited in the case of microwave transmission.

Satellite Microwave Communication

- A satellite is a physical object that revolves around the earth at a known height.
- Satellite communication is more reliable nowadays as it offers more flexibility than cable and fibre optic systems.
- We can communicate with any point on the globe by using satellite communication.

How Does Satellite work?

The satellite accepts the signal that is transmitted from the earth station, and it amplifies the signal. The amplified signal is retransmitted to another earth station.

Advantages Of Satellite Microwave Communication:

- The coverage area of a satellite microwave is more than the terrestrial microwave.
- The transmission cost of the satellite is independent of the distance from the centre of the coverage area.
- Satellite communication is used in mobile and wireless communication applications.
- It is easy to install.
- It is used in a wide variety of applications such as weather forecasting, radio/TV signal broadcasting, mobile communication, etc.

Disadvantages Of Satellite Microwave Communication:

- Satellite designing and development requires more time and higher cost.
- The Satellite needs to be monitored and controlled on regular periods so that it remains in orbit.
- The life of the satellite is about 12-15 years. Due to this reason, another launch of the satellite has to be planned before it becomes non-functional.

Infrared

- An infrared transmission is a wireless technology used for communication over short ranges.
- The frequency of the infrared in the range from 300 GHz to 400 THz.
- It is used for short-range communication such as data transfer between two cell phones, TV remote operation, data transfer between a computer and cell phone resides in the same closed area.

Characteristics Of Infrared:

- It supports high bandwidth, and hence the data rate will be very high.
- Infrared waves cannot penetrate the walls. Therefore, the infrared communication in one room cannot be interrupted by the nearby rooms.
- An infrared communication provides better security with minimum interference.
- Infrared communication is unreliable outside the building because the sun rays will interfere with the infrared waves.

What is 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.

Multiplexing is achieved by using a device called Multiplexer (**MUX**) that combines n input lines to generate a single output line. Multiplexing follows many-to-one, i.e., n input lines and one output line.

Demultiplexing is achieved by using a device called Demultiplexer (**DEMUX**) available at the receiving end. DEMUX separates a signal into its component signals (one input and n outputs). Therefore, we can say that demultiplexing follows the one-to-many approach.

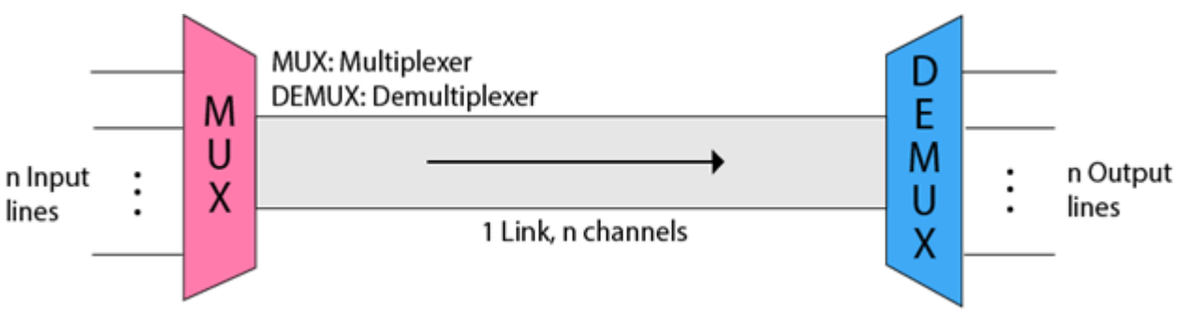
Why Multiplexing?

- The transmission medium is used to send the signal from sender to receiver. The medium can only have one signal at a time.
- If there are multiple signals to share one medium, then the medium must be divided in such a way that each signal is given some portion of the available bandwidth. For example: If there are 10 signals and bandwidth of medium is100 units, then the 10 unit is shared by each signal.
- When multiple signals share the common medium, there is a possibility of collision. Multiplexing concept is used to avoid such collision.
- Transmission services are very expensive.

History of Multiplexing

- Multiplexing technique is widely used in telecommunications in which several telephone calls are carried through a single wire.
- Multiplexing originated in telegraphy in the early 1870s and is now widely used in communication.
- George Owen Squier developed the **telephone carrier multiplexing** in 1910.

Concept of Multiplexing



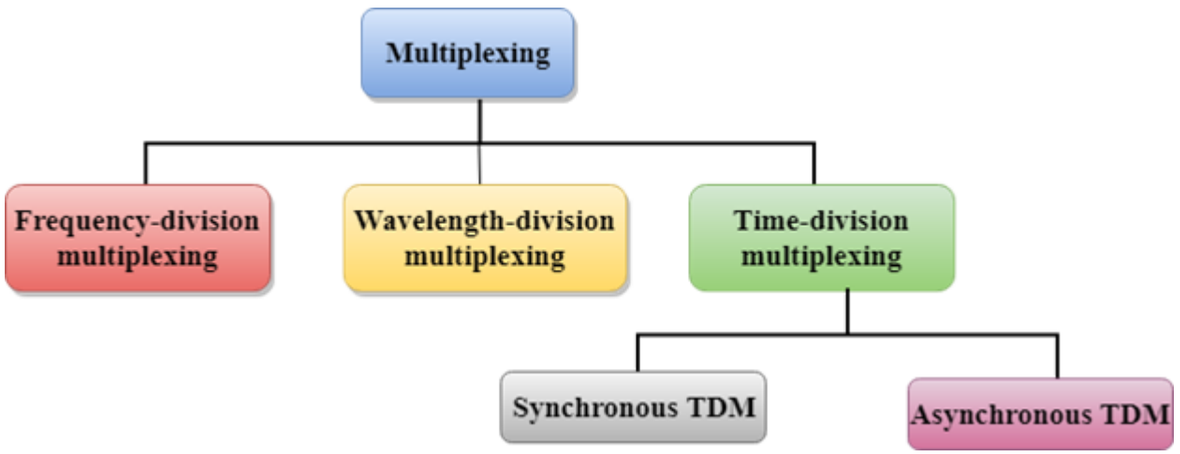
- The 'n' input lines are transmitted through a multiplexer and multiplexer combines the signals to form a composite signal.
- The composite signal is passed through a Demultiplexer and demultiplexer separates a signal to component signals and transfers them to their respective destinations.

Advantages of Multiplexing:

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

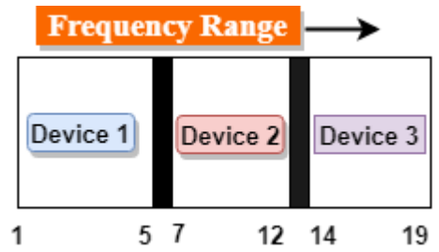
Multiplexing Techniques

Multiplexing techniques can be classified as:

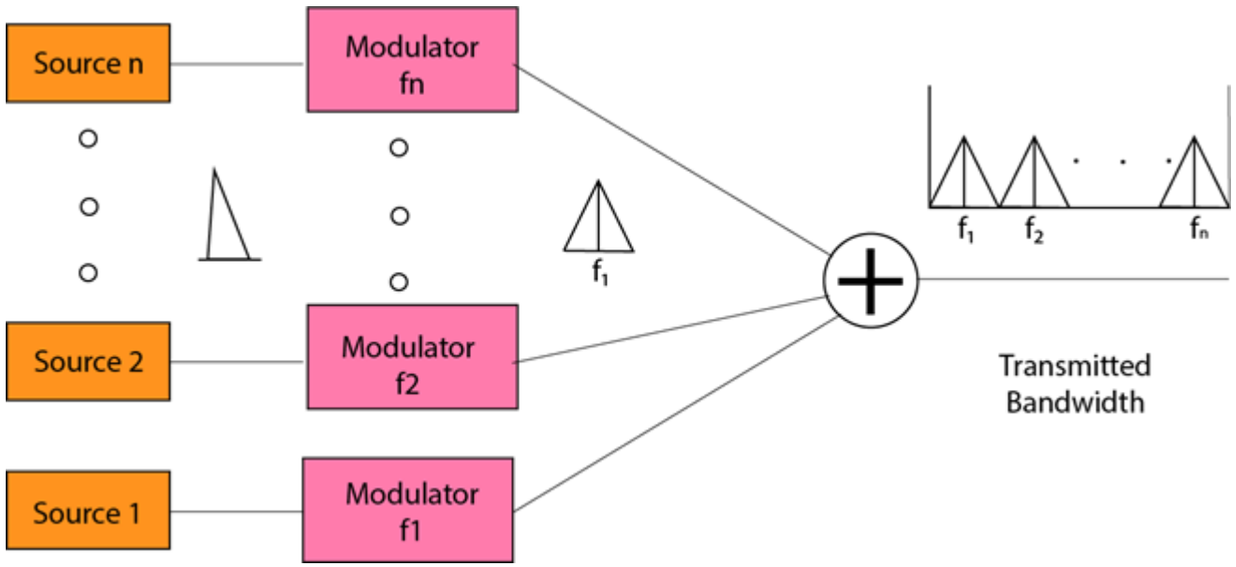


Frequency-division Multiplexing (FDM)

- It is an analog technique.
- **Frequency Division Multiplexing** is a technique in which the available bandwidth of a single transmission medium is subdivided into several channels.



- In the above diagram, a single transmission medium is subdivided into several frequency channels, and each frequency channel is given to different devices. Device 1 has a frequency channel of range from 1 to 5.
- The input signals are translated into frequency bands by using modulation techniques, and they are combined by a multiplexer to form a composite signal.
- The main aim of the FDM is to subdivide the available bandwidth into different frequency channels and allocate them to different devices.
- Using the modulation technique, the input signals are transmitted into frequency bands and then combined to form a composite signal.
- The carriers which are used for modulating the signals are known as **sub-carriers**. They are represented as f1,f2..fn.
- **FDM** is mainly used in radio broadcasts and TV networks.



Advantages Of FDM:

- FDM is used for analog signals.
- 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.

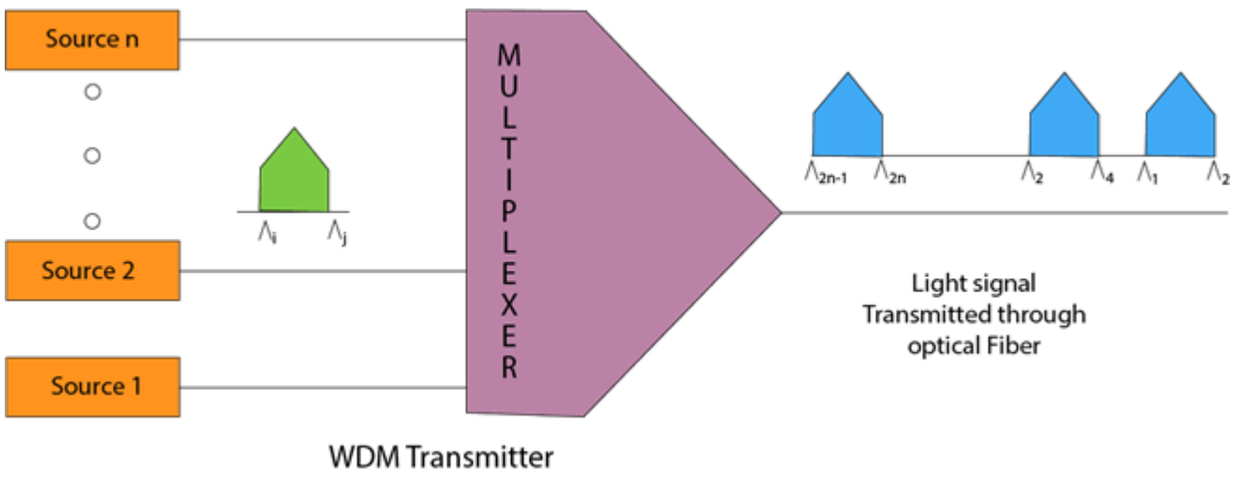
Disadvantages Of FDM:

- 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.

Applications Of FDM:

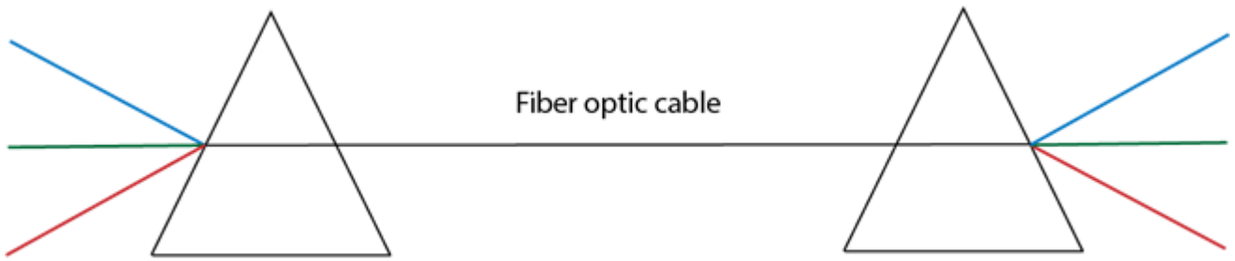
- FDM is commonly used in TV networks.
- It is used in FM and AM broadcasting. Each FM radio station has different frequencies, and they are multiplexed to form a composite signal. The multiplexed signal is transmitted in the air.

Wavelength Division Multiplexing (WDM)



- Wavelength Division Multiplexing is same as FDM except that the optical signals are transmitted through the fibre optic cable.
- WDM is used on fibre optics to increase the capacity of a single fibre.

- It is used to utilize the high data rate capability of fibre optic cable.
- It is an analog multiplexing technique.
- Optical signals from different source are combined to form a wider band of light with the help of multiplexer.
- At the receiving end, demultiplexer separates the signals to transmit them to their respective destinations.
- Multiplexing and Demultiplexing can be achieved by using a prism.
- Prism can perform a role of multiplexer by combining the various optical signals to form a composite signal, and the composite signal is transmitted through a fibre optical cable.
- Prism also performs a reverse operation, i.e., demultiplexing the signal.



Time Division Multiplexing

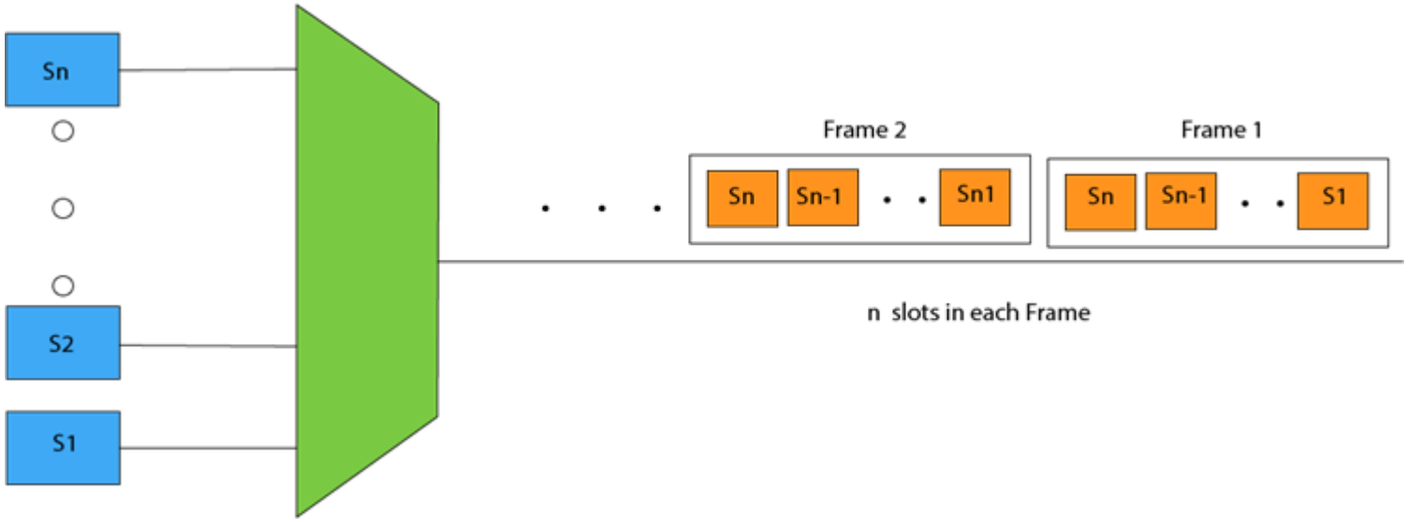
- It is a digital technique.
- In Frequency Division Multiplexing Technique, all signals operate at the same time with different frequency, but in case of Time Division Multiplexing technique, all signals operate at the same frequency with different time.
- In **Time Division Multiplexing technique**, the total time available in the channel is distributed among different users. Therefore, each user is allocated with different time interval known as a Time slot at which data is to be transmitted by the sender.
- A user takes control of the channel for a fixed amount of time.
- In Time Division Multiplexing technique, data is not transmitted simultaneously rather the data is transmitted one-by-one.
- In TDM, the signal is transmitted in the form of frames. Frames contain a cycle of time slots in which each frame contains one or more slots dedicated to each user.
- It can be used to multiplex both digital and analog signals but mainly used to multiplex digital signals.

There are two types of TDM:

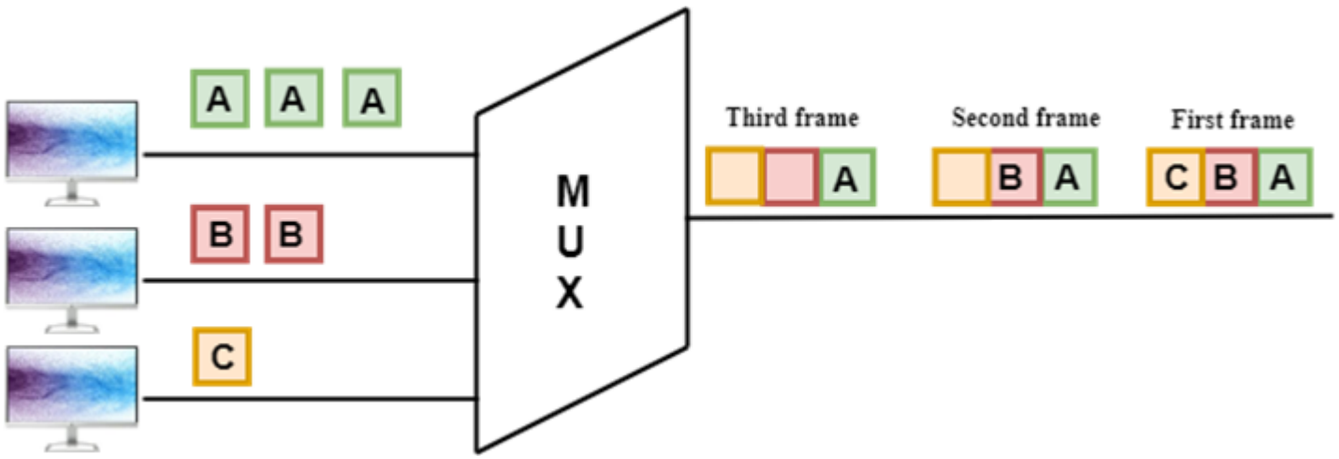
- Synchronous TDM
- Asynchronous TDM

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



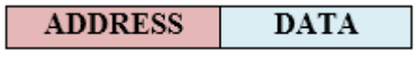
In the above figure, the Synchronous TDM technique is implemented. Each device is allocated with some time slot. The time slots are transmitted irrespective of whether the sender has data to send or not.

Disadvantages Of Synchronous TDM:

- The capacity of the channel is not fully utilized as the empty slots are also transmitted which is having no data. In the above figure, the first frame is completely filled, but in the last two frames, some slots are empty. Therefore, we can say that the capacity of the channel is not utilized efficiently.
- The speed of the transmission medium should be greater than the total speed of the input lines. An alternative approach to the Synchronous TDM is Asynchronous Time Division Multiplexing.

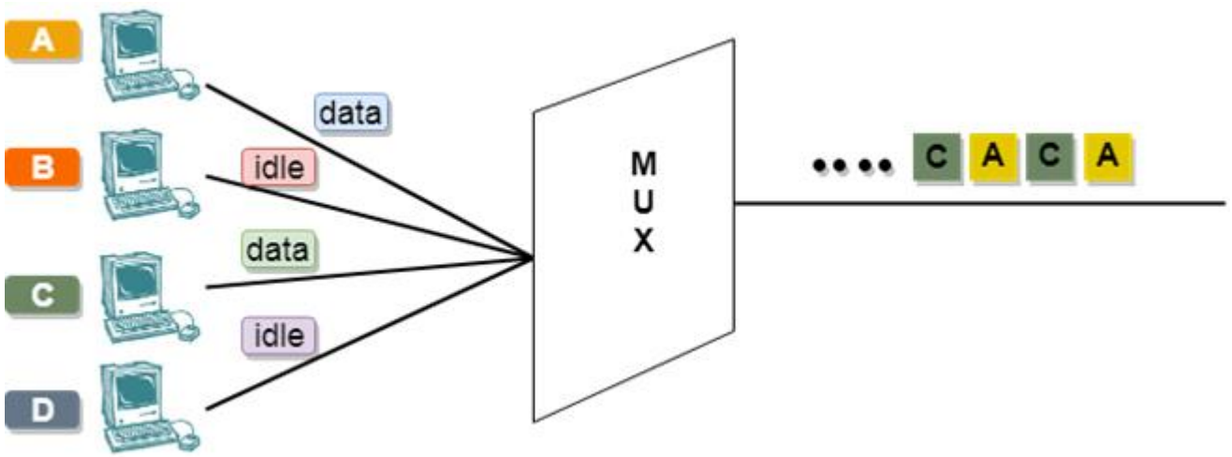
Asynchronous TDM

- 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.



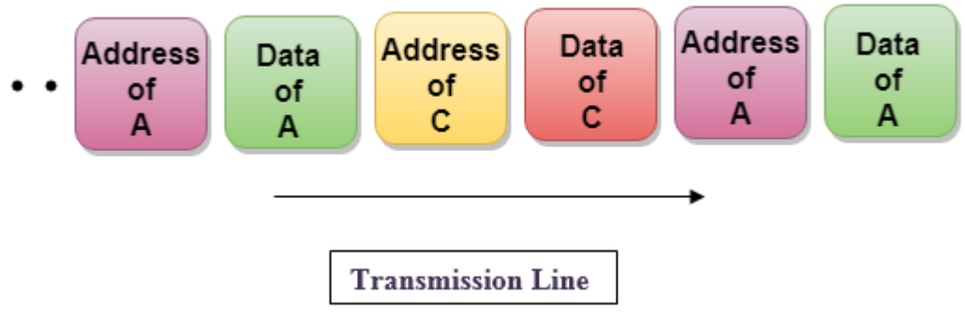
- The difference between Asynchronous TDM and Synchronous TDM is that many slots in Synchronous TDM are unutilized, but in Asynchronous TDM, slots are fully utilized. This leads to the smaller transmission time and efficient utilization of the capacity of the channel.
- In Synchronous TDM, if there are 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



In the above diagram, there are 4 devices, but only two devices are sending the data, i.e., A and C. Therefore, the data of A and C are only transmitted through the transmission line.

Frame of above diagram can be represented as:



The above figure shows that the data part contains the address to determine the source of the data.

Switching

- When a user accesses the internet or another computer network outside their immediate location, messages are sent through the network of transmission media. This technique of transferring the information from one computer network to another network is known as **switching**.
- Switching in a computer network is achieved by using switches. A switch is a small hardware device which is used to join multiple computers together with one local area network (LAN).
- Network switches operate at layer 2 (Data link layer) in the OSI model.
- Switching is transparent to the user and does not require any configuration in the home network.
- Switches are used to forward the packets based on MAC addresses.
- A Switch is used to transfer the data only to the device that has been addressed. It verifies the destination address to route the packet appropriately.
- It is operated in full duplex mode.
- Packet collision is minimum as it directly communicates between source and destination.
- It does not broadcast the message as it works with limited bandwidth.

Why is Switching Concept required?

Switching concept is developed because of the following reasons:

- **Bandwidth:** It is defined as the maximum transfer rate of a cable. It is a very critical and expensive resource. Therefore, switching techniques are used for the effective utilization of the bandwidth of a network.
- **Collision:** Collision is the effect that occurs when more than one device transmits the message over the same physical media, and they collide with each other. To overcome this problem, switching technology is implemented so that packets do not collide with each other.

Advantages of Switching:

- Switch increases the bandwidth of the network.
- It reduces the workload on individual PCs as it sends the information to only that device which has been addressed.
- It increases the overall performance of the network by reducing the traffic on the network.
- There will be less frame collision as switch creates the collision domain for each connection.

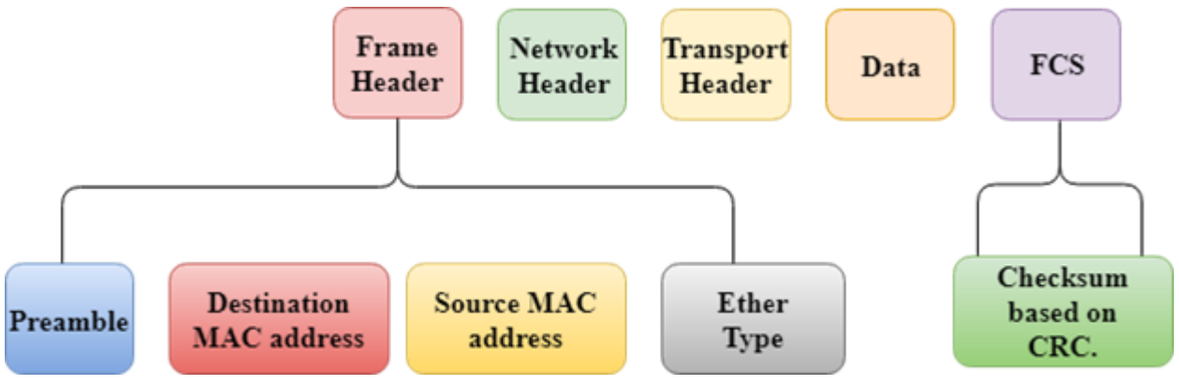
Disadvantages of Switching:

- A Switch is more expensive than network bridges.

- A Switch cannot determine the network connectivity issues easily.
- Proper designing and configuration of the switch are required to handle multicast packets.

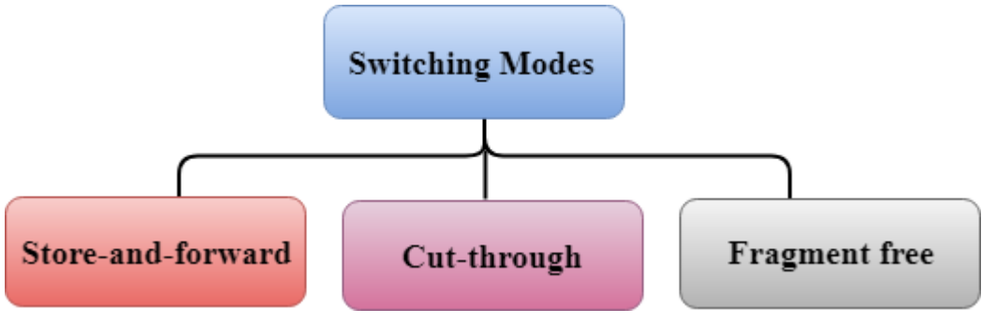
Switching Modes

- The layer 2 switches are used for transmitting the data on the data link layer, and it also performs error checking on transmitted and received frames.
- The layer 2 switches forward the packets with the help of MAC address.
- Different modes are used for forwarding the packets known as **Switching modes**.
- In **switching mode**, Different parts of a frame are recognized. The frame consists of several parts such as preamble, destination MAC address, source MAC address, user's data, FCS.

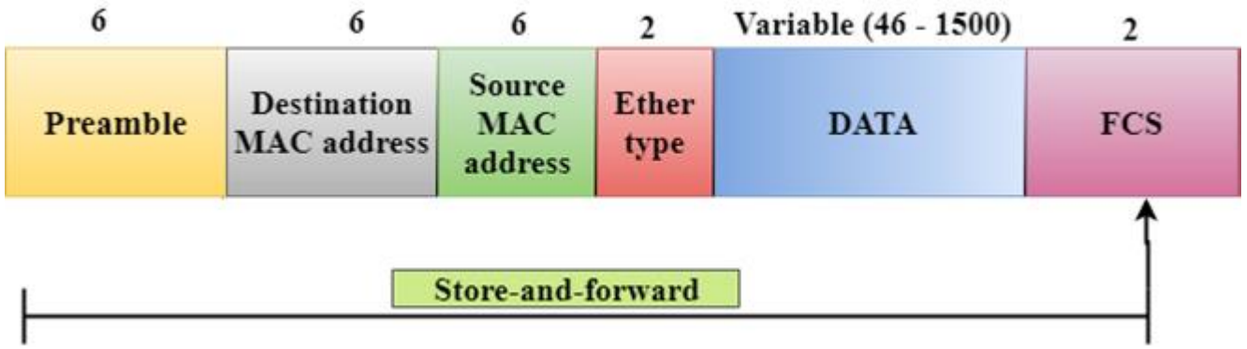


There are three types of switching modes:

- Store-and-forward
- Cut-through
- Fragment-free

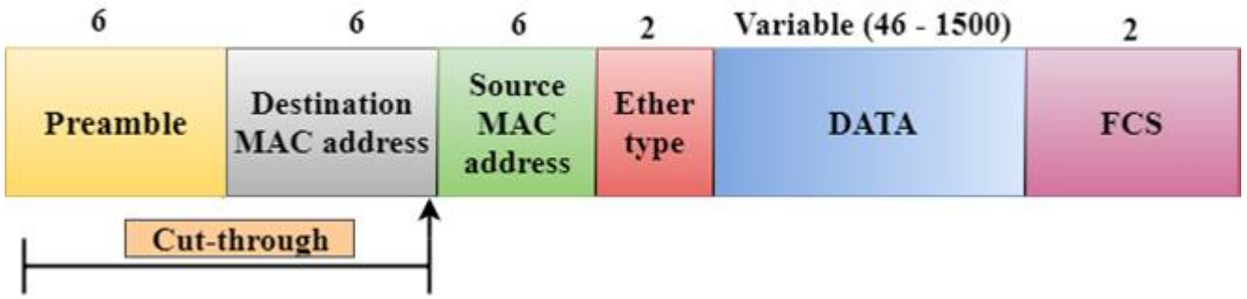


Store-and-forward



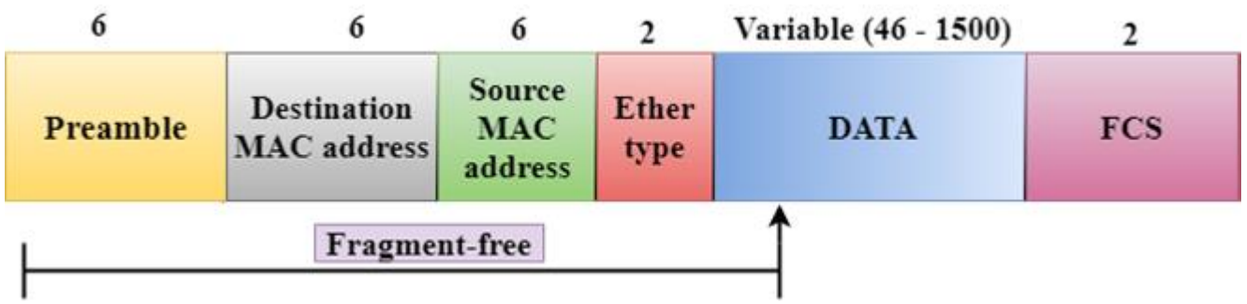
- Store-and-forward is a technique in which the intermediate nodes store the received frame and then check for errors before forwarding the packets to the next node.
- The layer 2 switch waits until the entire frame has received. On receiving the entire frame, switch store the frame into the switch buffer memory. This process is known as **storing the frame**.
- When the frame is stored, then the frame is checked for the errors. If any error found, the message is discarded otherwise the message is forwarded to the next node. This process is known as **forwarding the frame**.
- CRC (Cyclic Redundancy Check) technique is implemented that uses a number of bits to check for the errors on the received frame.
- The store-and-forward technique ensures a high level of security as the destination network will not be affected by the corrupted frames.
- Store-and-forward switches are highly reliable as it does not forward the collided frames.

Cut-through Switching



- Cut-through switching is a technique in which the switch forwards the packets after the destination address has been identified without waiting for the entire frame to be received.
- Once the frame is received, it checks the first six bytes of the frame following the preamble, the switch checks the destination in the switching table to determine the outgoing interface port, and forwards the frame to the destination.
- It has **low latency** rate as the switch does not wait for the entire frame to be received before sending the packets to the destination.
- It has no **error checking technique**. Therefore, the errors can be sent with or without errors to the receiver.
- A Cut-through switching technique has **low wait time** as it forwards the packets as soon as it identifies the destination MAC address.
- In this technique, collision is not detected, if frames have collided will also be forwarded.

Fragment-free Switching



- A Fragment-free switching is an advanced technique of the Cut-through Switching.
- A Fragment-free switching is a technique that reads atleast 64 bytes of a frame before forwarding to the next node to provide the error-free transmission.
- It combines the speed of Cut-through Switching with the error checking functionality.
- This technique checks the 64 bytes of the ethernet frame where addressing information is available.

- A collision is detected within 64 bytes of the frame, the frames which are collided will not be forwarded further.

Differences b/w Store-and-forward and Cut-through Switching.

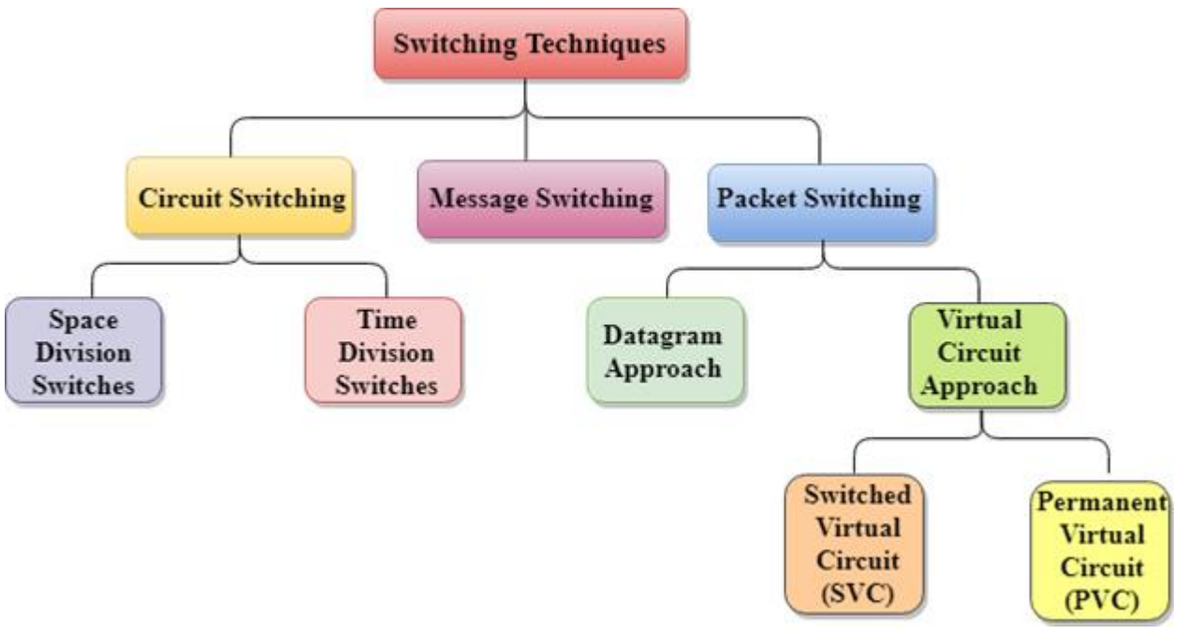
Store-and-forward Switching	Cut-through Switching
Store-and-forward Switching is a technique that waits until the entire frame is received.	Cut-through Switching is a technique that checks the first 6 bytes following the preamble to identify the destination address.
It performs error checking functionality. If any error is found in the frame, the frame will be discarded otherwise forwarded to the next node.	It does not perform any error checking. The frame with or without errors will be forwarded.
It has high latency rate as it waits for the entire frame to be received before forwarding to the next node.	It has low latency rate as it checks only six bytes of the frame to determine the destination address.
It is highly reliable as it forwards only error-free packets.	It is less reliable as compared to Store-and-forward technique as it forwards error prone packets as well.
It has a high wait time as it waits for the entire frame to be received before taking any forwarding decisions.	It has low wait time as cut-through switches do not store the whole frame or packets.

Switching techniques

In large networks, there can be multiple paths from sender to receiver. The switching technique will decide the best route for data transmission.

Switching technique is used to connect the systems for making one-to-one communication.

Classification Of Switching Techniques



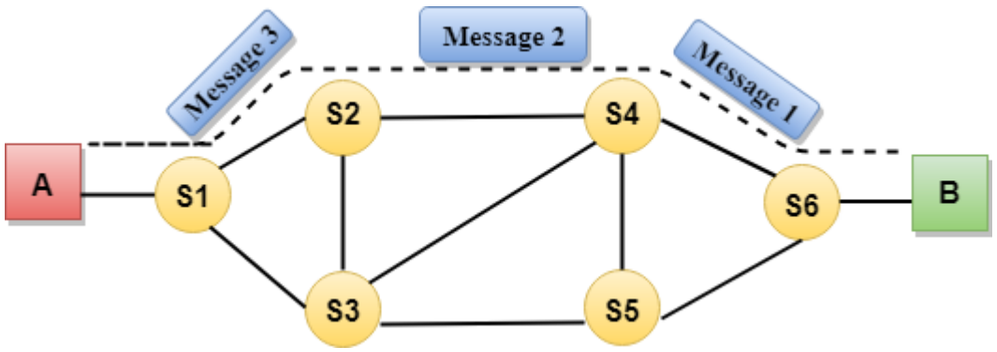
Circuit Switching

- Circuit switching is a switching technique that establishes a dedicated path between sender and receiver.
- In the Circuit Switching Technique, once the connection is established then the dedicated path will remain to exist until the connection is terminated.
- Circuit switching in a network operates in a similar way as the telephone works.
- A complete end-to-end path must exist before the communication takes place.
- In case of circuit switching technique, when any user wants to send the data, voice, video, a request signal is sent to the receiver then the receiver sends back the acknowledgment to ensure the availability of the dedicated path. After receiving the acknowledgment, dedicated path transfers the data.
- Circuit switching is used in public telephone network. It is used for voice transmission.
- Fixed data can be transferred at a time in circuit switching technology.

Communication through circuit switching has 3 phases:

- Circuit establishment
- Data transfer

- Circuit Disconnect



Circuit Switching can use either of the two technologies:

Space Division Switches:

- Space Division Switching is a circuit switching technology in which a single transmission path is accomplished in a switch by using a physically separate set of crosspoints.
- Space Division Switching can be achieved by using crossbar switch. A crossbar switch is a metallic crosspoint or semiconductor gate that can be enabled or disabled by a control unit.
- The Crossbar switch is made by using the semiconductor. For example, Xilinx crossbar switch using FPGAs.
- Space Division Switching has high speed, high capacity, and nonblocking switches.

Space Division Switches can be categorized in two ways:

- **Crossbar Switch**
- **Multistage Switch**

Crossbar Switch

The Crossbar switch is a switch that has n input lines and n output lines. The crossbar switch has n^2 intersection points known as **crosspoints.**

Disadvantage of Crossbar switch:

The number of crosspoints increases as the number of stations is increased. Therefore, it becomes very expensive for a large switch. The solution to this is to use a multistage switch.

Multistage Switch

- Multistage Switch is made by splitting the crossbar switch into the smaller units and then interconnecting them.
- It reduces the number of crosspoints.
- If one path fails, then there will be an availability of another path.

Advantages Of Circuit Switching:

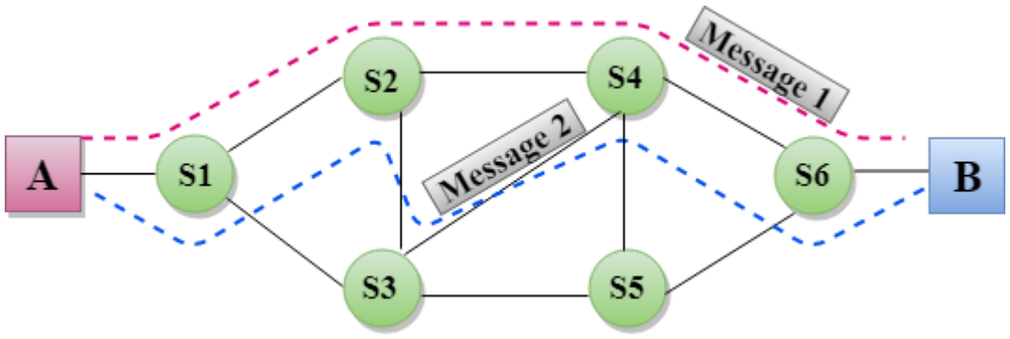
- In the case of Circuit Switching technique, the communication channel is dedicated.
- It has fixed bandwidth.

Disadvantages Of Circuit Switching:

- Once the dedicated path is established, the only delay occurs in the speed of data transmission.
- It takes a long time to establish a connection approx 10 seconds during which no data can be transmitted.
- It is more expensive than other switching techniques as a dedicated path is required for each connection.
- It is inefficient to use because once the path is established and no data is transferred, then the capacity of the path is wasted.
- In this case, the connection is dedicated therefore no other data can be transferred even if the channel is free.

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.



Advantages Of Message Switching

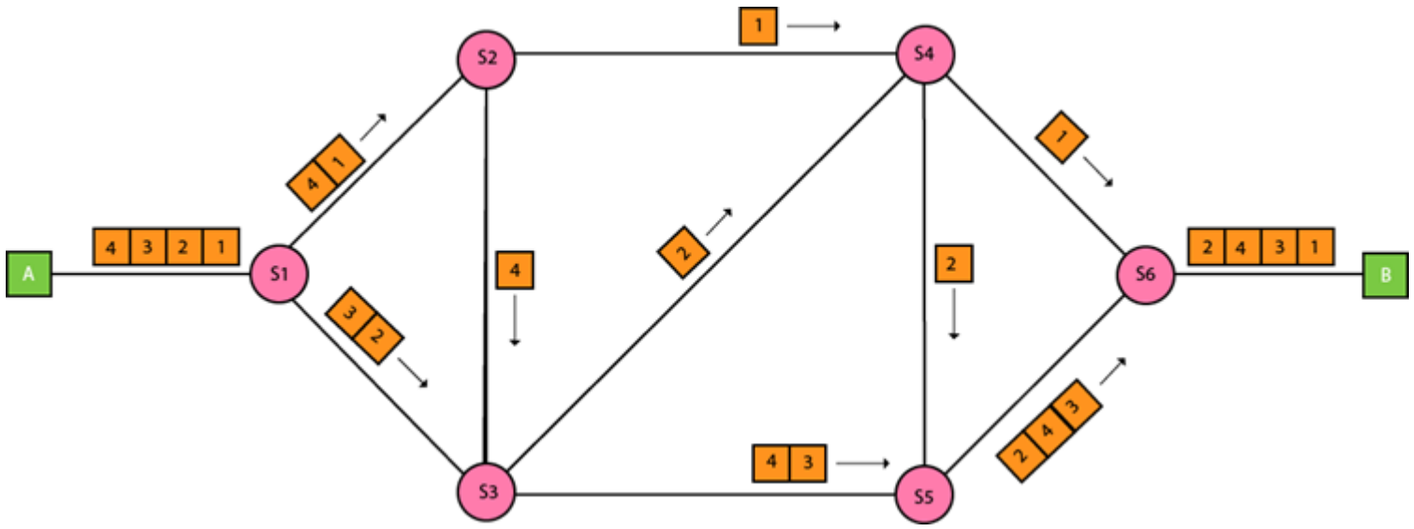
- Data channels are shared among the communicating devices that improve the efficiency of using available bandwidth.
- Traffic congestion can be reduced because the message is temporarily stored in the nodes.
- Message priority can be used to manage the network.
- The size of the message which is sent over the network can be varied. Therefore, it supports the data of unlimited size.

Disadvantages Of Message Switching

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

Packet Switching

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



Approaches Of Packet Switching:

There are two approaches to Packet Switching:

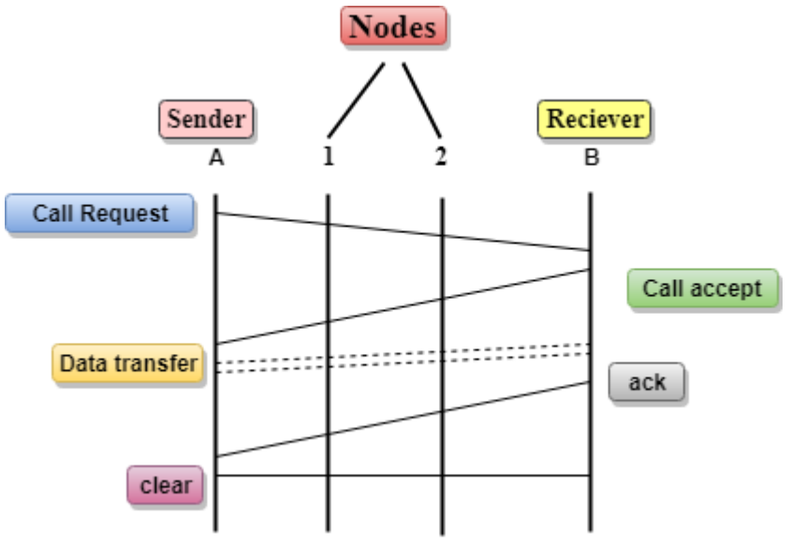
Datagram Packet switching:

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

Virtual Circuit Switching

- Virtual Circuit Switching is also known as connection-oriented switching.
- In the case of Virtual circuit switching, a preplanned route is established before the messages are sent.
- Call request and call accept packets are used to establish the connection between sender and receiver.
- In this case, the path is fixed for the duration of a logical connection.

Let's understand the concept of virtual circuit switching through a diagram:



- In the above diagram, A and B are the sender and receiver respectively. 1 and 2 are the nodes.
- Call request and call accept packets are used to establish a connection between the sender and receiver.
- When a route is established, data will be transferred.
- After transmission of data, an acknowledgment signal is sent by the receiver that the message has been received.
- If the user wants to terminate the connection, a clear signal is sent for the termination.

Differences b/w Datagram approach and Virtual Circuit approach

Datagram approach	Virtual Circuit approach
Node takes routing decisions to forward the packets.	Node does not take any routing decision.
Congestion cannot occur as all the packets travel in different directions.	Congestion can occur when the node is busy, and it does not allow other packets to pass through.

It is more flexible as all the packets are treated as an independent entity.	It is not very flexible.

Advantages Of Packet Switching:

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

Disadvantages Of Packet Switching:

- Packet Switching technique cannot be implemented in those applications that require low delay and high-quality services.
- The protocols used in a packet switching technique are very complex and requires high implementation cost.
- If the network is overloaded or corrupted, then it requires retransmission of lost packets. It can also lead to the loss of critical information if errors are nor recovered

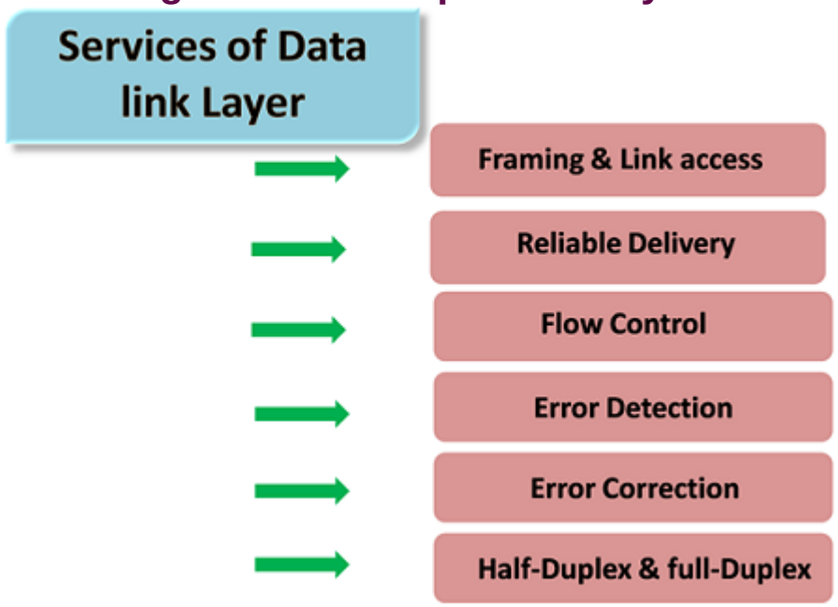
Data Link layer

Data Link Layer

- In the OSI model, the data link layer is a 4^{th} layer from the top and 2^{nd} layer from the bottom.
- The communication channel that connects the adjacent nodes is known as links, and in order to move the datagram from source to the destination, the datagram must be moved across an individual link.
- The main responsibility of the Data Link Layer is to transfer the datagram across an individual link.
- The Data link layer protocol defines the format of the packet exchanged across the nodes as well as the actions such as Error detection, retransmission, flow control, and random access.
- The Data Link Layer protocols are Ethernet, token ring, FDDI and PPP.

- An important characteristic of a Data Link Layer is that datagram can be handled by different link layer protocols on different links in a path. For example, the datagram is handled by Ethernet on the first link, PPP on the second link.

Following services are provided by the Data Link Layer:



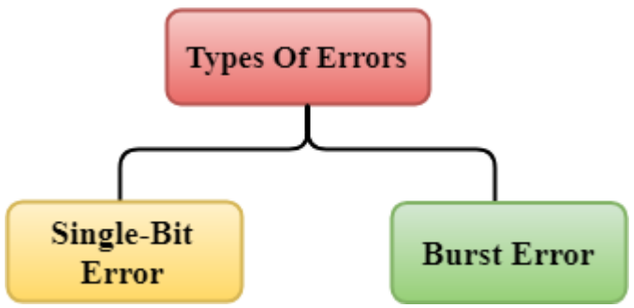
- **Framing & Link access:** Data Link Layer protocols encapsulate each network frame within a Link layer frame before the transmission across the link. A frame consists of a data field in which network layer datagram is inserted and a number of data fields. It specifies the structure of the frame as well as a channel access protocol by which frame is to be transmitted over the link.
- **Reliable delivery:** Data Link Layer provides a reliable delivery service, i.e., transmits the network layer datagram without any error. A reliable delivery service is accomplished with transmissions and acknowledgements. A data link layer mainly provides the reliable delivery service over the links as they have higher error rates and they can be corrected locally, link at which an error occurs rather than forcing to retransmit the data.
- **Flow control:** A receiving node can receive the frames at a faster rate than it can process the frame. Without flow control, the receiver's buffer can overflow, and frames can get lost. To overcome this problem, the data link layer uses the flow control to prevent the sending node on one side of the link from overwhelming the receiving node on another side of the link.
- **Error detection:** Errors can be introduced by signal attenuation and noise. Data Link Layer protocol provides a mechanism to detect one or more errors. This is achieved by adding error detection bits in the frame and then receiving node can perform an error check.
- **Error correction:** Error correction is similar to the Error detection, except that receiving node not only detect the errors but also determine where the errors have occurred in the frame.

- **Half-Duplex & Full-Duplex:** In a Full-Duplex mode, both the nodes can transmit the data at the same time. In a Half-Duplex mode, only one node can transmit the data at the same time.

Error Detection

When data is transmitted from one device to another device, the system does not guarantee whether the data received by the device is identical to the data transmitted by another device. An Error is a situation when the message received at the receiver end is not identical to the message transmitted.

Types Of Errors

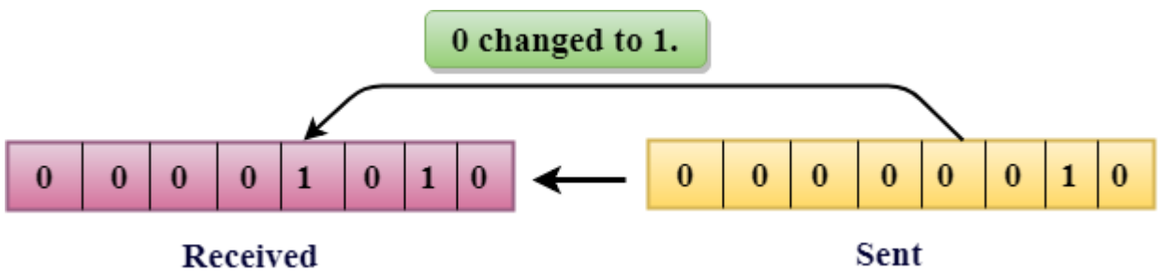


Errors can be classified into two categories:

- Single-Bit Error
- Burst Error

Single-Bit Error:

The only one bit of a given data unit is changed from 1 to 0 or from 0 to 1.



In the above figure, the message which is sent is corrupted as single-bit, i.e., 0 bit is changed to 1.

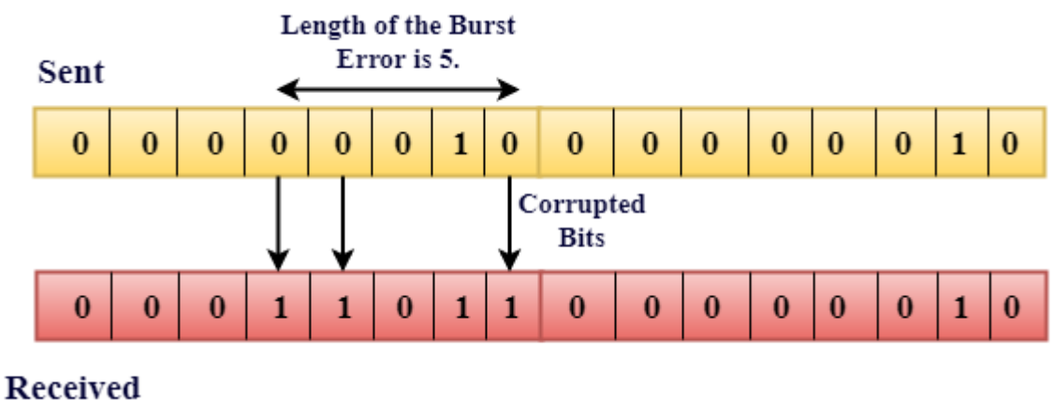
Single-Bit Error does not appear more likely in Serial Data Transmission. For example, Sender sends the data at 10 Mbps, this means that the bit lasts only for 1 ?s and for a single-bit error to occurred, a noise must be more than 1 ?s.

Single-Bit Error mainly occurs in Parallel Data Transmission. For example, if eight wires are used to send the eight bits of a byte, if one of the wire is noisy, then single-bit is corrupted per byte.

Burst Error:

The two or more bits are changed from 0 to 1 or from 1 to 0 is known as Burst Error.

The Burst Error is determined from the first corrupted bit to the last corrupted bit.



The duration of noise in Burst Error is more than the duration of noise in Single-Bit.

Burst Errors are most likely to occurr in Serial Data Transmission.

The number of affected bits depends on the duration of the noise and data rate.

Error Detecting Techniques:

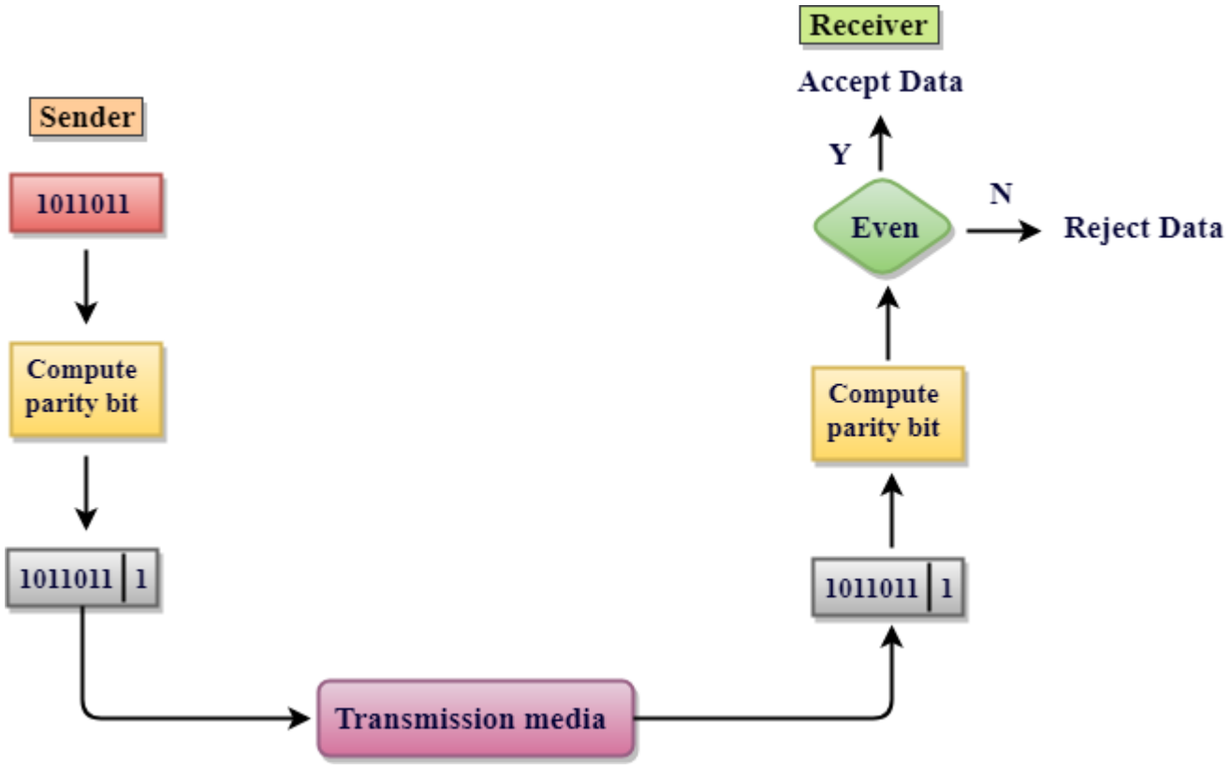
The most popular Error Detecting Techniques are:

- Single parity check
- Two-dimensional parity check
- Checksum

- Cyclic redundancy check

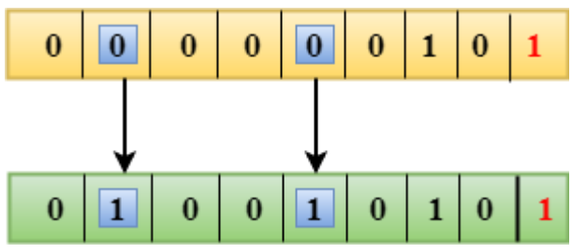
Single Parity Check

- Single Parity checking is the simple mechanism and inexpensive to detect the errors.
- In this technique, a redundant bit is also known as a parity bit which is appended at the end of the data unit so that the number of 1s becomes even. Therefore, the total number of transmitted bits would be 9 bits.
- If the number of 1s bits is odd, then parity bit 1 is appended and if the number of 1s bits is even, then parity bit 0 is appended at the end of the data unit.
- At the receiving end, the parity bit is calculated from the received data bits and compared with the received parity bit.
- This technique generates the total number of 1s even, so it is known as even-parity checking.



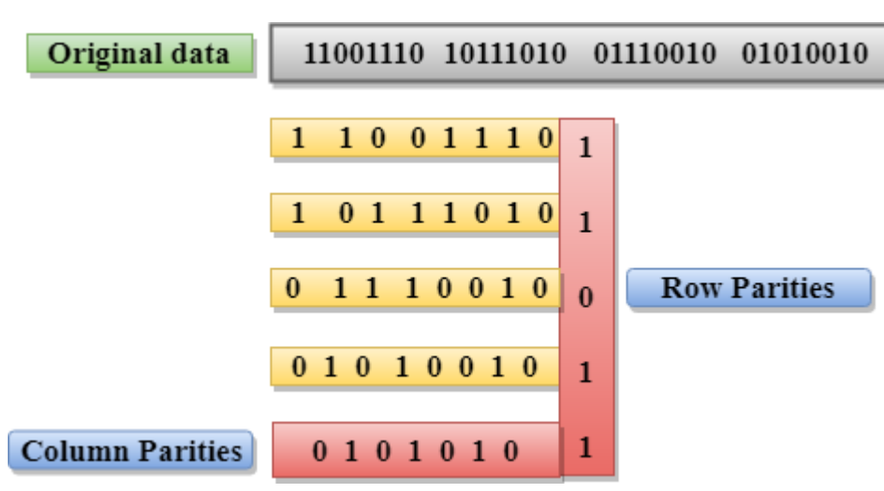
Drawbacks Of Single Parity Checking

- It can only detect single-bit errors which are very rare.
- If two bits are interchanged, then it cannot detect the errors.



Two-Dimensional Parity Check

- Performance can be improved by using **Two-Dimensional Parity Check** which organizes the data in the form of a table.
- Parity check bits are computed for each row, which is equivalent to the single-parity check.
- In Two-Dimensional Parity check, a block of bits is divided into rows, and the redundant row of bits is added to the whole block.
- At the receiving end, the parity bits are compared with the parity bits computed from the received data.



Drawbacks Of 2D Parity Check

- If two bits in one data unit are corrupted and two bits exactly the same position in another data unit are also corrupted, then 2D Parity checker will not be able to detect the error.
- This technique cannot be used to detect the 4-bit errors or more in some cases.

Checksum

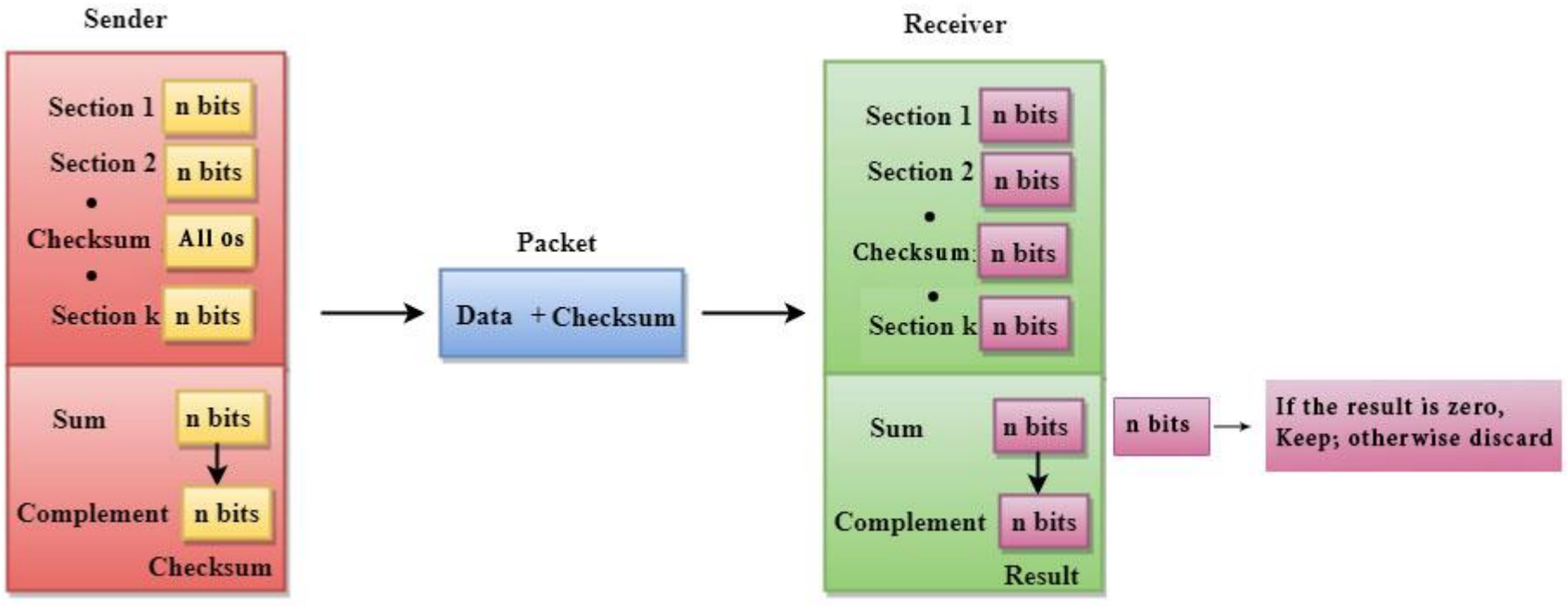
A Checksum is an error detection technique based on the concept of redundancy.

It is divided into two parts:

Checksum Generator

A Checksum is generated at the sending side. Checksum generator subdivides the data into equal segments of n bits each, and all these segments are added together by using one's complement arithmetic. The sum is complemented and appended to the original data, known as checksum field. The extended data is transmitted across the network.

Suppose L is the total sum of the data segments, then the checksum would be ?L



1. The Sender follows the given steps:
2. The block unit is divided into k sections, and each of n bits.
3. All the k sections are added together by using one's complement to get the sum.
4. The sum is complemented and it becomes the checksum field.
5. The original data and checksum field are sent across the network.

Checksum Checker

A Checksum is verified at the receiving side. The receiver subdivides the incoming data into equal segments of n bits each, and all these segments are added together, and then this sum is complemented. If the complement of the sum is zero, then the data is accepted otherwise data is rejected.

1. The Receiver follows the given steps:
2. The block unit is divided into k sections and each of n bits.
3. All the k sections are added together by using one's complement algorithm to get the sum.
4. The sum is complemented.
5. If the result of the sum is zero, then the data is accepted otherwise the data is discarded.

Cyclic Redundancy Check (CRC)

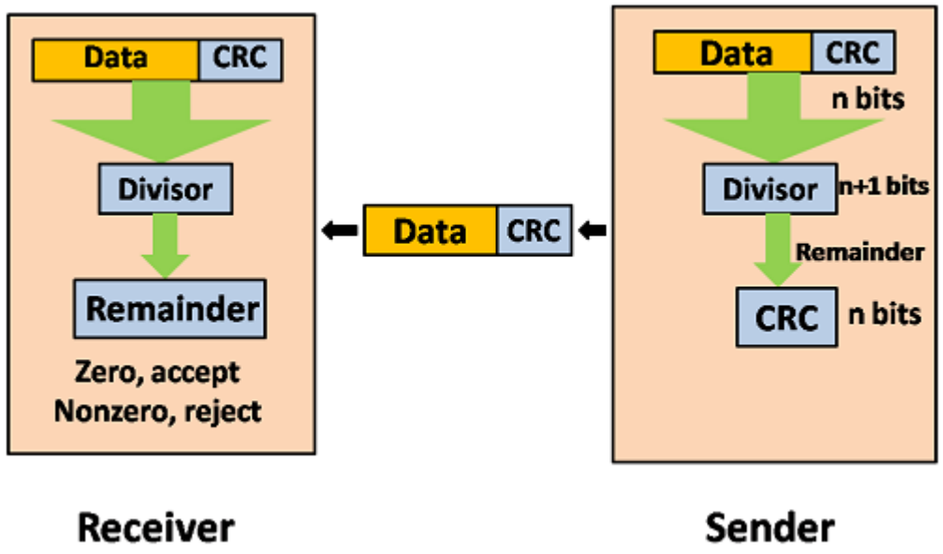
CRC is a redundancy error technique used to determine the error.

Following are the steps used in CRC for error detection:

- In CRC technique, a string of n 0s is appended to the data unit, and this n number is less than the number of bits in a predetermined number, known as division which is n+1 bits.
- Secondly, the newly extended data is divided by a divisor using a process is known as binary division. The remainder generated from this division is known as CRC remainder.
- Thirdly, the CRC remainder replaces the appended 0s at the end of the original data. This newly generated unit is sent to the receiver.
- The receiver receives the data followed by the CRC remainder. The receiver will treat this whole unit as a single unit, and it is divided by the same divisor that was used to find the CRC remainder.

If the resultant of this division is zero which means that it has no error, and the data is accepted.

If the resultant of this division is not zero which means that the data consists of an error. Therefore, the data is discarded.

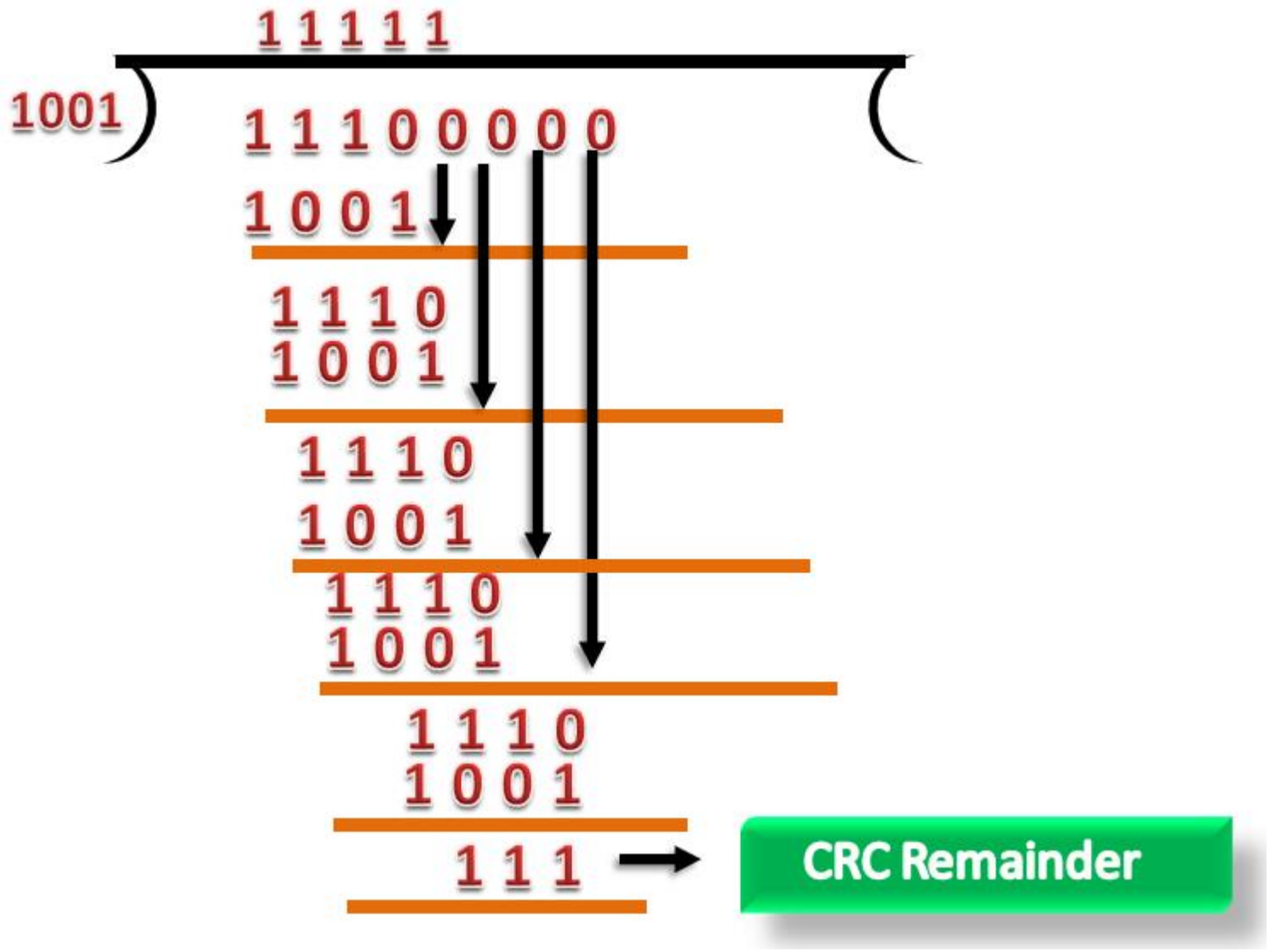


Let's understand this concept through an example:

Suppose the original data is 11100 and divisor is 1001.

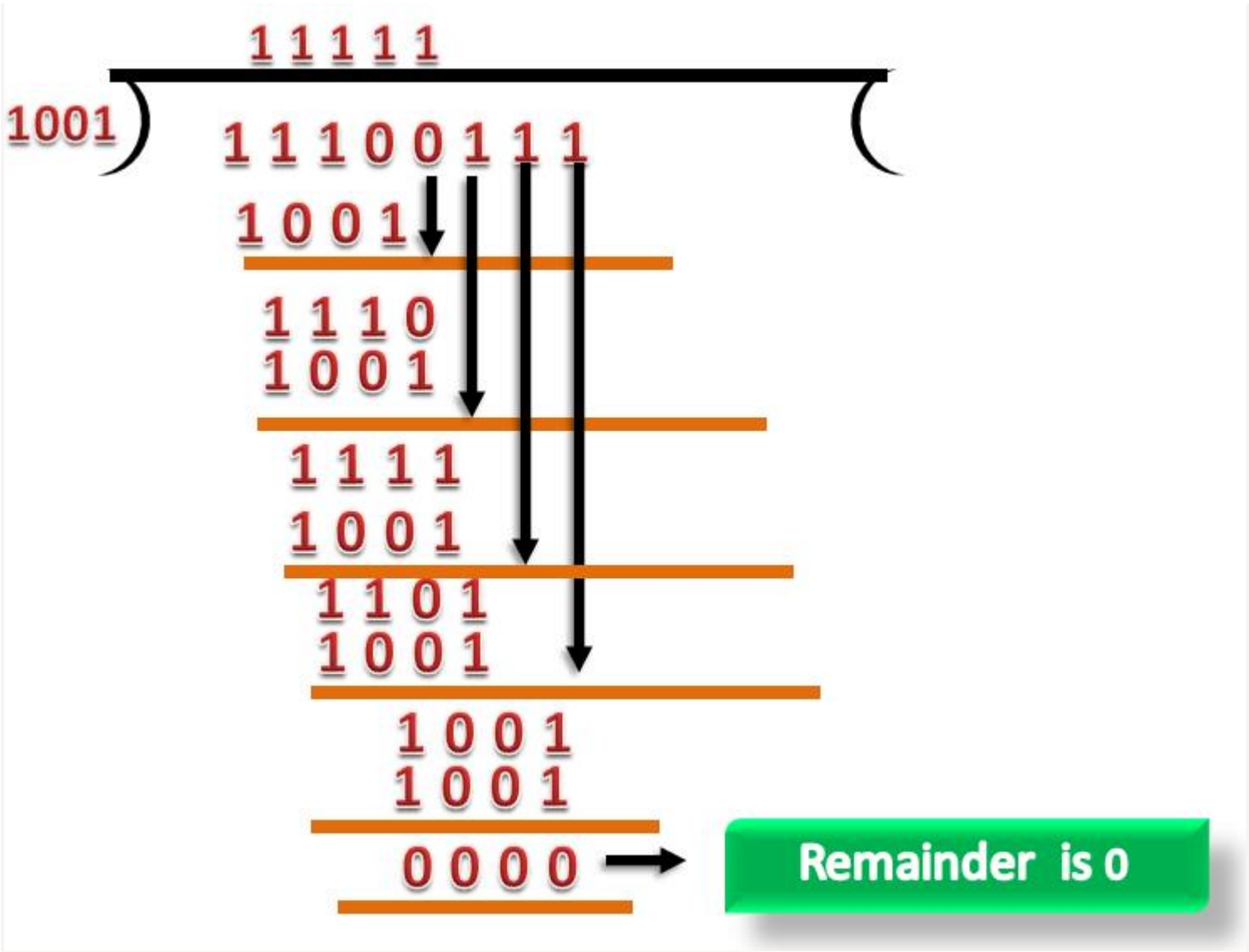
CRC Generator

- A CRC generator uses a modulo-2 division. Firstly, three zeroes are appended at the end of the data as the length of the divisor is 4 and we know that the length of the string 0s to be appended is always one less than the length of the divisor.
- Now, the string becomes 11100000, and the resultant string is divided by the divisor 1001.
- The remainder generated from the binary division is known as CRC remainder. The generated value of the CRC remainder is 111.
- CRC remainder replaces the appended string of 0s at the end of the data unit, and the final string would be 11100111 which is sent across the network.



CRC Checker

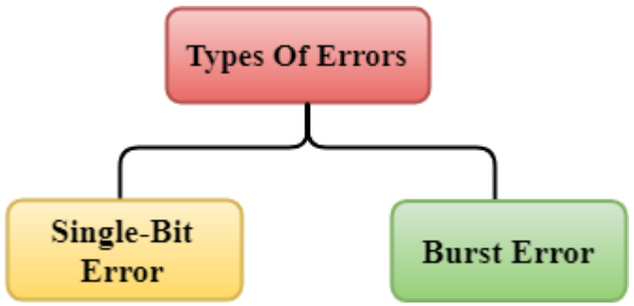
- The functionality of the CRC checker is similar to the CRC generator.
- When the string 11100111 is received at the receiving end, then CRC checker performs the modulo-2 division.
- A string is divided by the same divisor, i.e., 1001.
- In this case, CRC checker generates the remainder of zero. Therefore, the data is accepted.



Error Detection

When data is transmitted from one device to another device, the system does not guarantee whether the data received by the device is identical to the data transmitted by another device. An Error is a situation when the message received at the receiver end is not identical to the message transmitted.

Types Of Errors

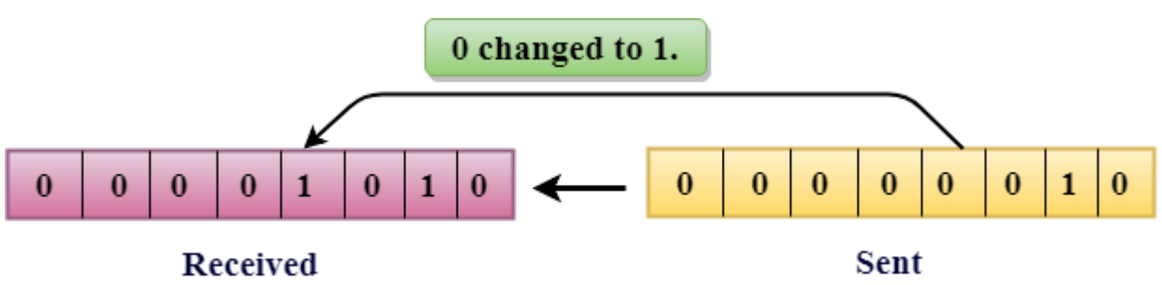


Errors can be classified into two categories:

- Single-Bit Error
- Burst Error

Single-Bit Error:

The only one bit of a given data unit is changed from 1 to 0 or from 0 to 1.



In the above figure, the message which is sent is corrupted as single-bit, i.e., 0 bit is changed to 1.

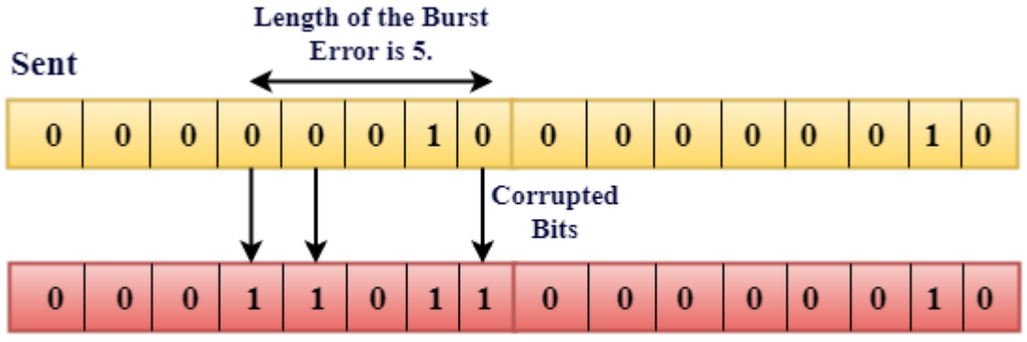
Single-Bit Error does not appear more likely in Serial Data Transmission. For example, Sender sends the data at 10 Mbps, this means that the bit lasts only for 1 ?s and for a single-bit error to occurred, a noise must be more than 1 ?s.

Single-Bit Error mainly occurs in Parallel Data Transmission. For example, if eight wires are used to send the eight bits of a byte, if one of the wire is noisy, then single-bit is corrupted per byte.

Burst Error:

The two or more bits are changed from 0 to 1 or from 1 to 0 is known as Burst Error.

The Burst Error is determined from the first corrupted bit to the last corrupted bit.



The duration of noise in Burst Error is more than the duration of noise in Single-Bit.

Burst Errors are most likely to occurr in Serial Data Transmission.

The number of affected bits depends on the duration of the noise and data rate.

Error Detecting Techniques:

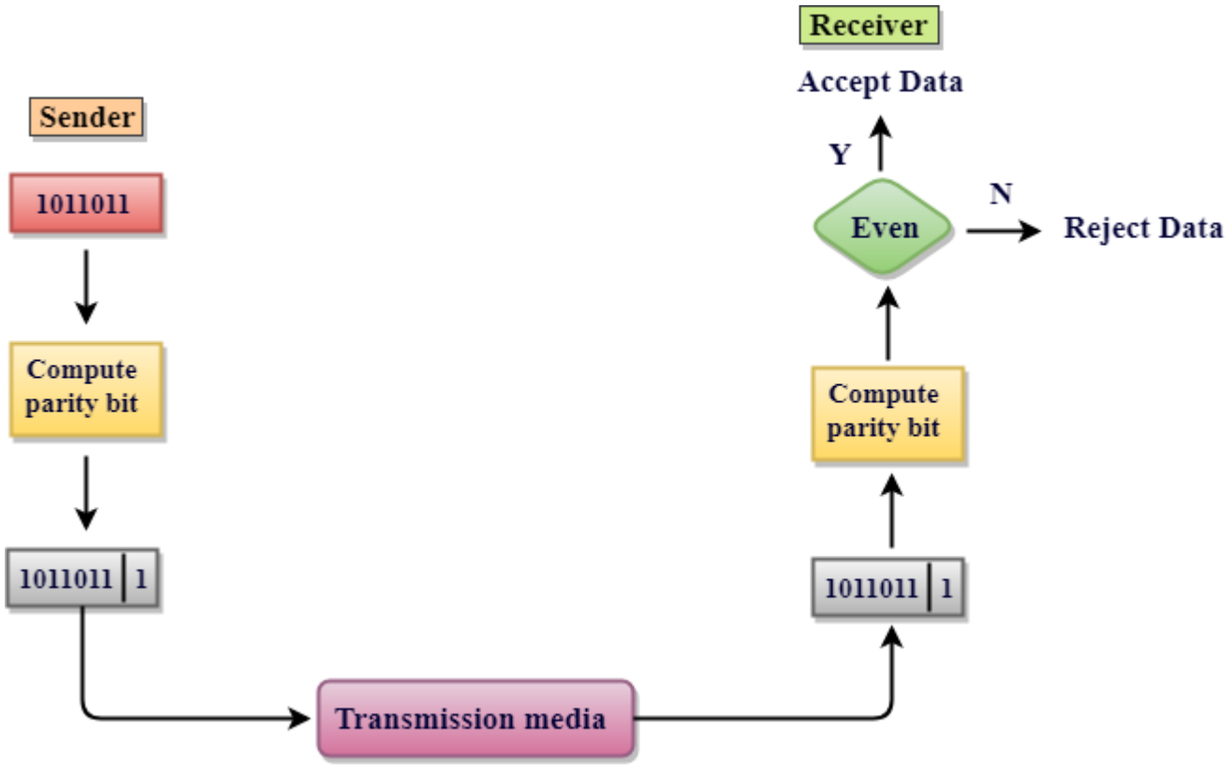
The most popular Error Detecting Techniques are:

- Single parity check
- Two-dimensional parity check
- Checksum
- Cyclic redundancy check

Single Parity Check

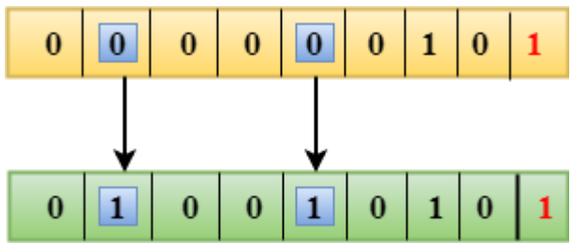
- Single Parity checking is the simple mechanism and inexpensive to detect the errors.
- In this technique, a redundant bit is also known as a parity bit which is appended at the end of the data unit so that the number of 1s becomes even. Therefore, the total number of transmitted bits would be 9 bits.
- If the number of 1s bits is odd, then parity bit 1 is appended and if the number of 1s bits is even, then parity bit 0 is appended at the end of the data unit.
- At the receiving end, the parity bit is calculated from the received data bits and compared with the received parity bit.

- This technique generates the total number of 1s even, so it is known as even-parity checking.



Drawbacks Of Single Parity Checking

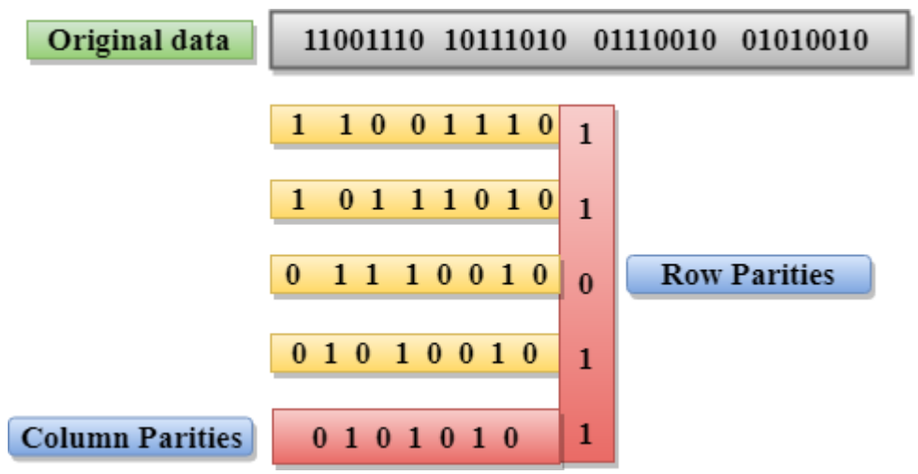
- It can only detect single-bit errors which are very rare.
- If two bits are interchanged, then it cannot detect the errors.



Two-Dimensional Parity Check

- Performance can be improved by using **Two-Dimensional Parity Check** which organizes the data in the form of a table.

- Parity check bits are computed for each row, which is equivalent to the single-parity check.
- In Two-Dimensional Parity check, a block of bits is divided into rows, and the redundant row of bits is added to the whole block.
- At the receiving end, the parity bits are compared with the parity bits computed from the received data.



Drawbacks Of 2D Parity Check

- If two bits in one data unit are corrupted and two bits exactly the same position in another data unit are also corrupted, then 2D Parity checker will not be able to detect the error.
- This technique cannot be used to detect the 4-bit errors or more in some cases.

Checksum

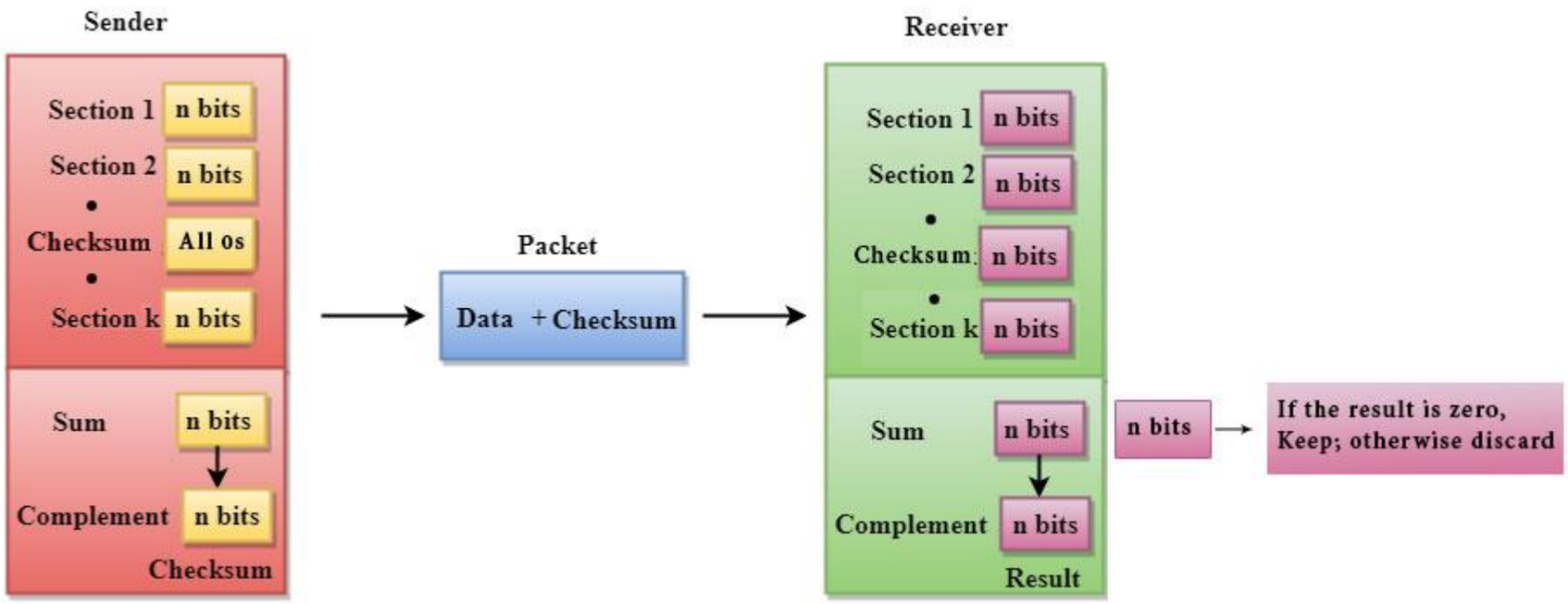
A Checksum is an error detection technique based on the concept of redundancy.

It is divided into two parts:

Checksum Generator

A Checksum is generated at the sending side. Checksum generator subdivides the data into equal segments of n bits each, and all these segments are added together by using one's complement arithmetic. The sum is complemented and appended to the original data, known as checksum field. The extended data is transmitted across the network.

Suppose L is the total sum of the data segments, then the checksum would be ?L



1. The Sender follows the given steps:
2. The block unit is divided into k sections, and each of n bits.
3. All the k sections are added together by using one's complement to get the sum.
4. The sum is complemented and it becomes the checksum field.
5. The original data and checksum field are sent across the network.

Checksum Checker

A Checksum is verified at the receiving side. The receiver subdivides the incoming data into equal segments of n bits each, and all these segments are added together, and then this sum is complemented. If the complement of the sum is zero, then the data is accepted otherwise data is rejected.

1. The Receiver follows the given steps:
2. The block unit is divided into k sections and each of n bits.
3. All the k sections are added together by using one's complement algorithm to get the sum.
4. The sum is complemented.
5. If the result of the sum is zero, then the data is accepted otherwise the data is discarded.

Cyclic Redundancy Check (CRC)

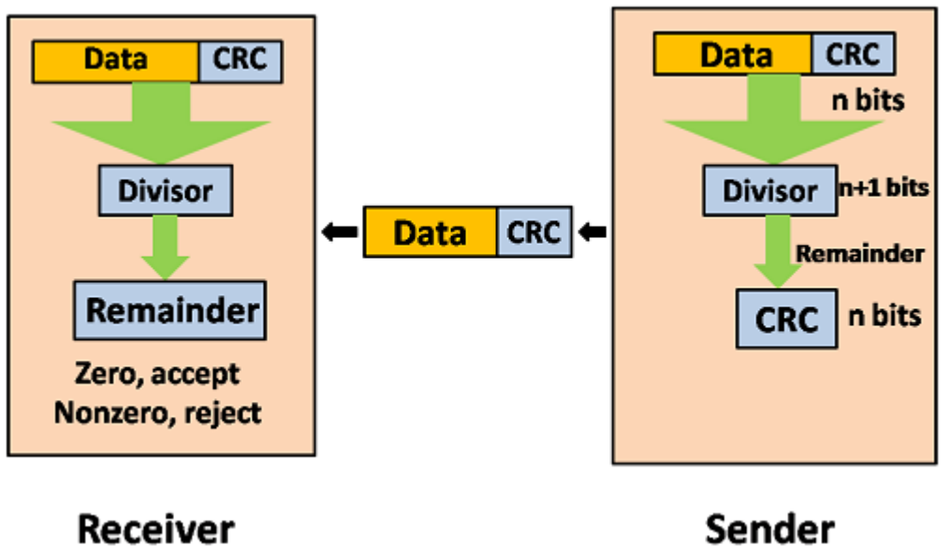
CRC is a redundancy error technique used to determine the error.

Following are the steps used in CRC for error detection:

- In CRC technique, a string of n 0s is appended to the data unit, and this n number is less than the number of bits in a predetermined number, known as division which is n+1 bits.
- Secondly, the newly extended data is divided by a divisor using a process is known as binary division. The remainder generated from this division is known as CRC remainder.
- Thirdly, the CRC remainder replaces the appended 0s at the end of the original data. This newly generated unit is sent to the receiver.
- The receiver receives the data followed by the CRC remainder. The receiver will treat this whole unit as a single unit, and it is divided by the same divisor that was used to find the CRC remainder.

If the resultant of this division is zero which means that it has no error, and the data is accepted.

If the resultant of this division is not zero which means that the data consists of an error. Therefore, the data is discarded.

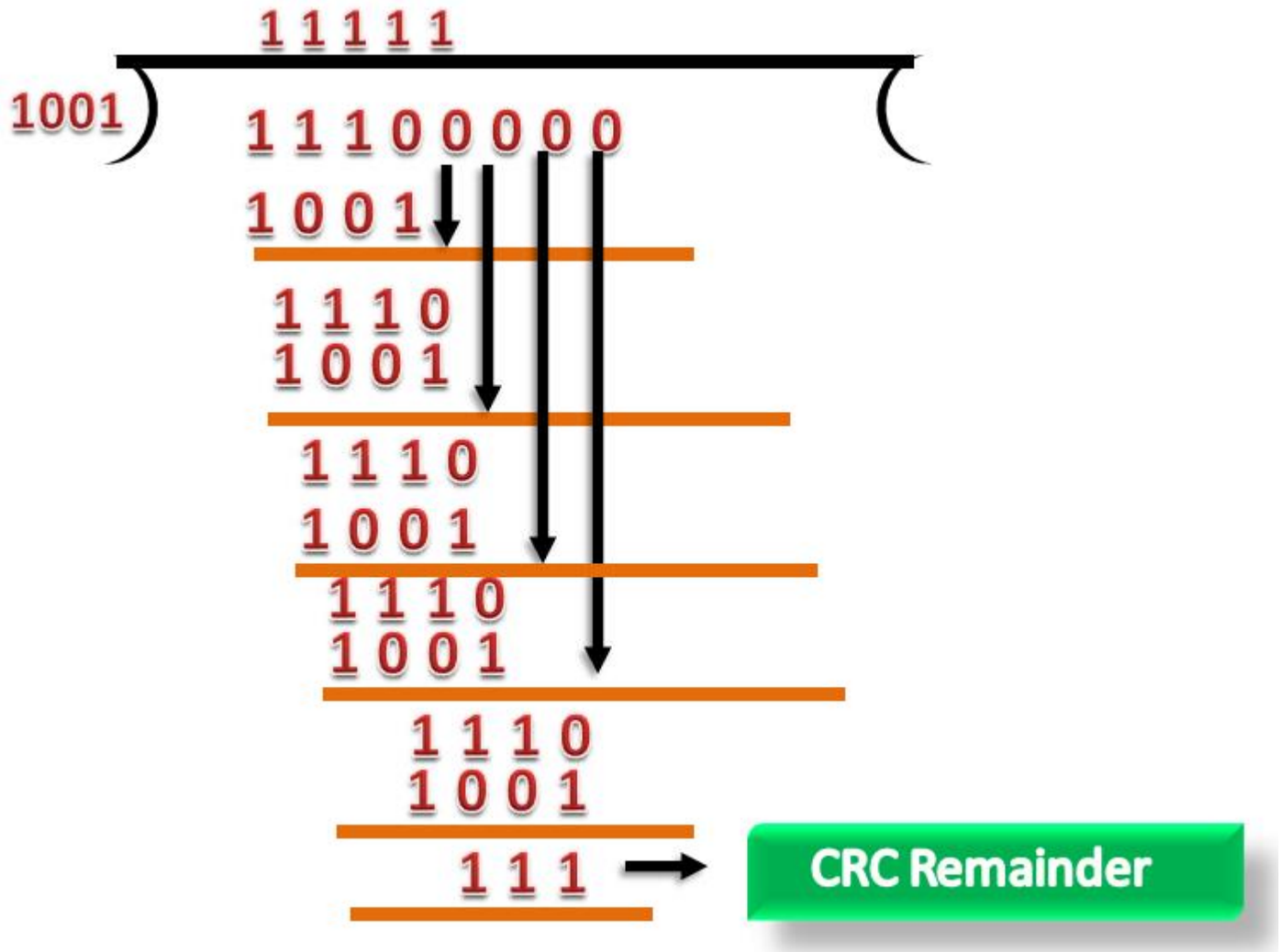


Let's understand this concept through an example:

Suppose the original data is 11100 and divisor is 1001.

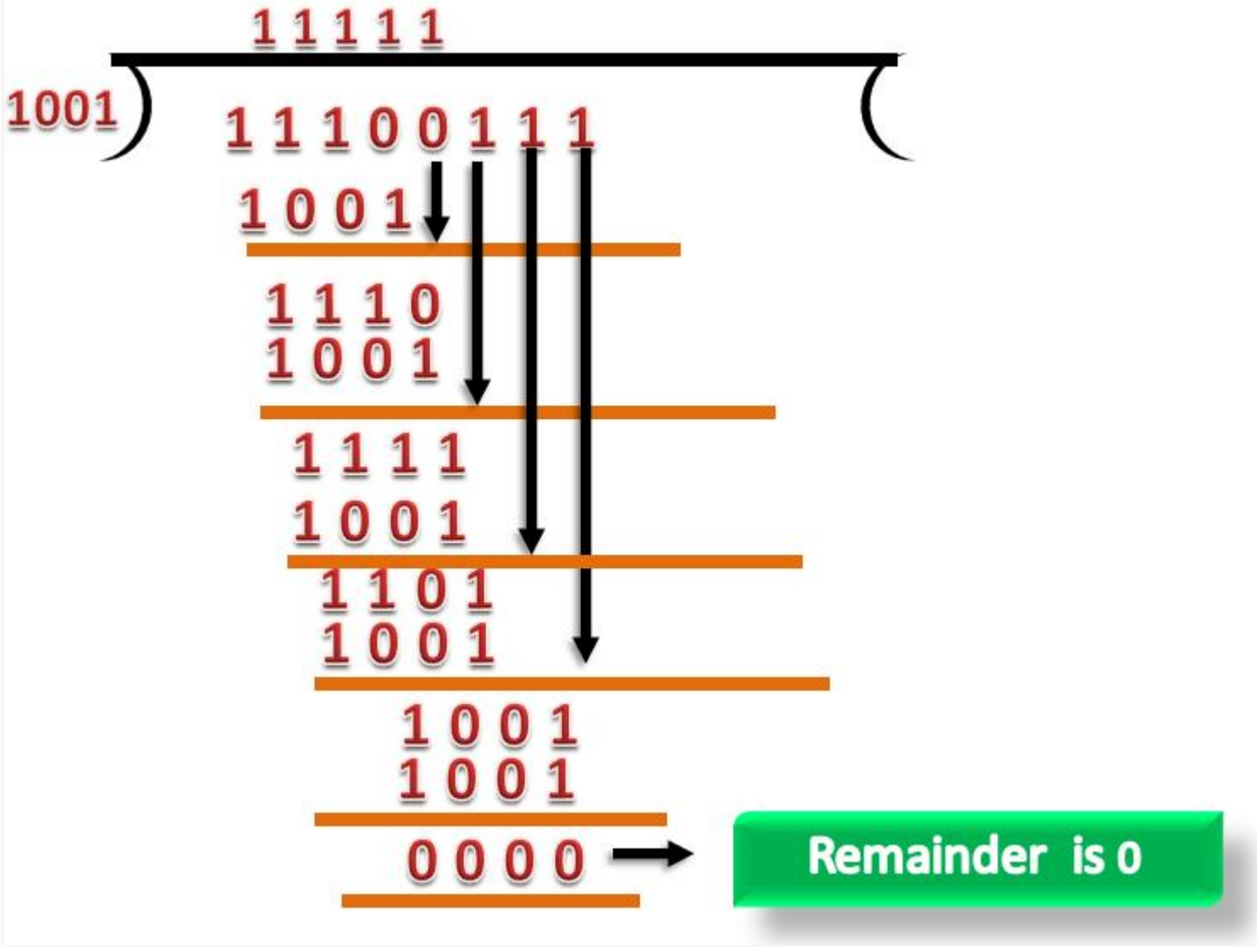
CRC Generator

- A CRC generator uses a modulo-2 division. Firstly, three zeroes are appended at the end of the data as the length of the divisor is 4 and we know that the length of the string 0s to be appended is always one less than the length of the divisor.
- Now, the string becomes 11100000, and the resultant string is divided by the divisor 1001.
- The remainder generated from the binary division is known as CRC remainder. The generated value of the CRC remainder is 111.
- CRC remainder replaces the appended string of 0s at the end of the data unit, and the final string would be 11100111 which is sent across the network.



CRC Checker

- The functionality of the CRC checker is similar to the CRC generator.
- When the string 11100111 is received at the receiving end, then CRC checker performs the modulo-2 division.
- A string is divided by the same divisor, i.e., 1001.
- In this case, CRC checker generates the remainder of zero. Therefore, the data is accepted.



Error Correction

Error Correction codes are used to detect and correct the errors when data is transmitted from the sender to the receiver.

Error Correction can be handled in two ways:

- **Backward error correction:** Once the error is discovered, the receiver requests the sender to retransmit the entire data unit.
- **Forward error correction:** In this case, the receiver uses the error-correcting code which automatically corrects the errors.

A single additional bit can detect the error, but cannot correct it.

For correcting the errors, one has to know the exact position of the error. For example, If we want to calculate a single-bit error, the error correction code will determine which one of seven bits is in error. To achieve this, we have to add some additional redundant bits.

Suppose r is the number of redundant bits and d is the total number of the data bits. The number of redundant bits r can be calculated by using the formula:

$$2^r >= d+r+1$$

The value of r is calculated by using the above formula. For example, if the value of d is 4, then the possible smallest value that satisfies the above relation would be 3.

To determine the position of the bit which is in error, a technique developed by R.W Hamming is Hamming code which can be applied to any length of the data unit and uses the relationship between data units and redundant units.

Hamming Code

Parity bits: The bit which is appended to the original data of binary bits so that the total number of 1s is even or odd.

Even parity: To check for even parity, if the total number of 1s is even, then the value of the parity bit is 0. If the total number of 1s occurrences is odd, then the value of the parity bit is 1.

Odd Parity: To check for odd parity, if the total number of 1s is even, then the value of parity bit is 1. If the total number of 1s is odd, then the value of parity bit is 0.

Algorithm of Hamming code:

- An information of 'd' bits are added to the redundant bits 'r' to form d+r.
- The location of each of the (d+r) digits is assigned a decimal value.

- The 'r' bits are placed in the positions 1,2,.....2^{k-1}.
- At the receiving end, the parity bits are recalculated. The decimal value of the parity bits determines the position of an error.

Relationship b/w Error position & binary number.

Error Position	Binary Number
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Let's understand the concept of Hamming code through an example:

Suppose the original data is 1010 which is to be sent.

Total number of data bits 'd' = 4

Number of redundant bits r : $2^r >= d+r+1$

$2^r >= 4+r+1$

Therefore, the value of r is 3 that satisfies the above relation.

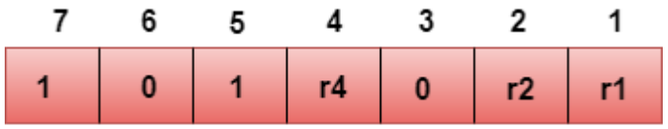
Total number of bits = d+r = 4+3 = 7;

Determining the position of the redundant bits

The number of redundant bits is 3. The three bits are represented by r1, r2, r4. The position of the redundant bits is calculated with corresponds to the raised power of 2. Therefore, their corresponding positions are **1, 2^1, 2^2**.

1. The position of r1 = 1
2. The position of r2 = 2
3. The position of r4 = 4

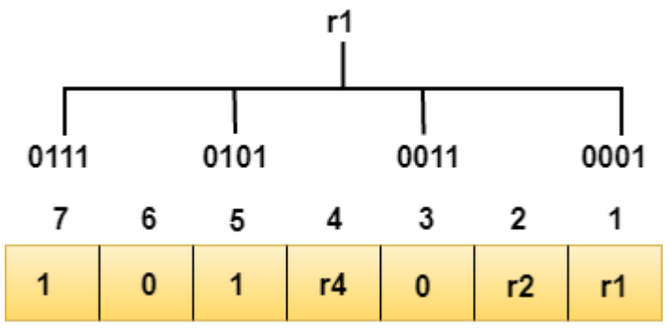
Representation of Data on the addition of parity bits:



Determining the Parity bits

Determining the r1 bit

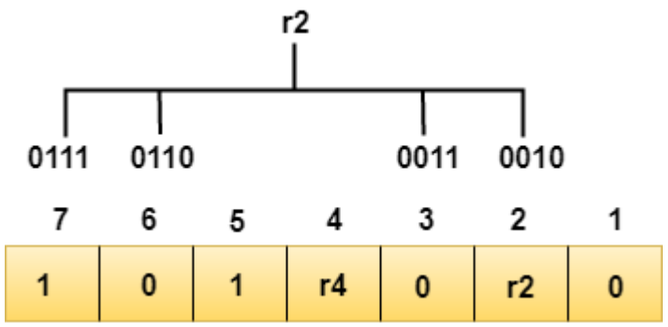
The r1 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the first position.



We observe from the above figure that the bit positions that includes 1 in the first position are 1, 3, 5, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r1 is **even, therefore, the value of the r1 bit is 0**.

Determining r2 bit

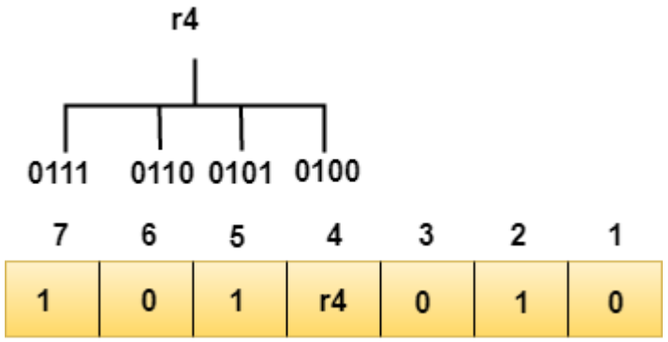
The r2 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the second position.



We observe from the above figure that the bit positions that includes 1 in the second position are **2, 3, 6, 7**. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r2 is **odd, therefore, the value of the r2 bit is 1**.

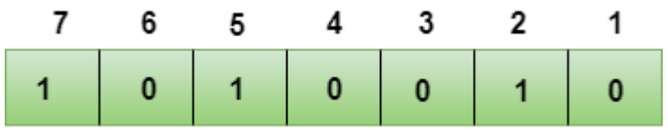
Determining r4 bit

The r4 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the third position.



We observe from the above figure that the bit positions that includes 1 in the third position are **4, 5, 6, 7**. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r4 is **even, therefore, the value of the r4 bit is 0**.

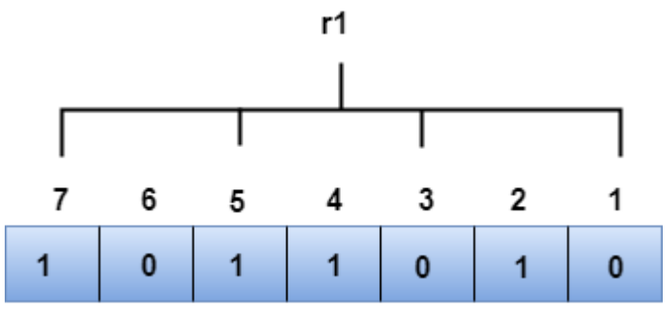
Data transferred is given below:



Suppose the 4th bit is changed from 0 to 1 at the receiving end, then parity bits are recalculated.

R1 bit

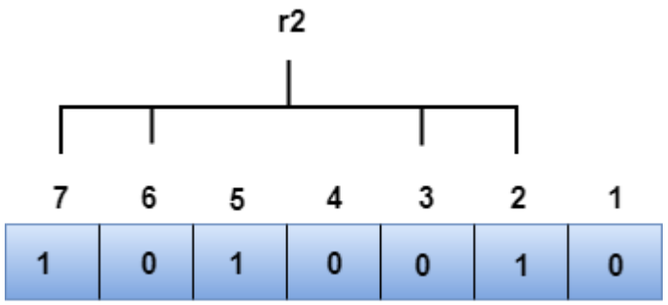
The bit positions of the r1 bit are 1,3,5,7



We observe from the above figure that the binary representation of r1 is 1100. Now, we perform the even-parity check, the total number of 1s appearing in the r1 bit is an even number. Therefore, the value of r1 is 0.

R2 bit

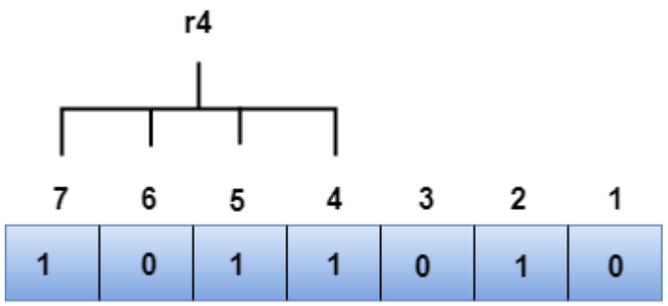
The bit positions of r2 bit are 2,3,6,7.



We observe from the above figure that the binary representation of r2 is 1001. Now, we perform the even-parity check, the total number of 1s appearing in the r2 bit is an even number. Therefore, the value of r2 is 0.

R4 bit

The bit positions of r4 bit are 4,5,6,7.



We observe from the above figure that the binary representation of r4 is 1011. Now, we perform the even-parity check, the total number of 1s appearing in the r4 bit is an odd number. Therefore, the value of r4 is 1.

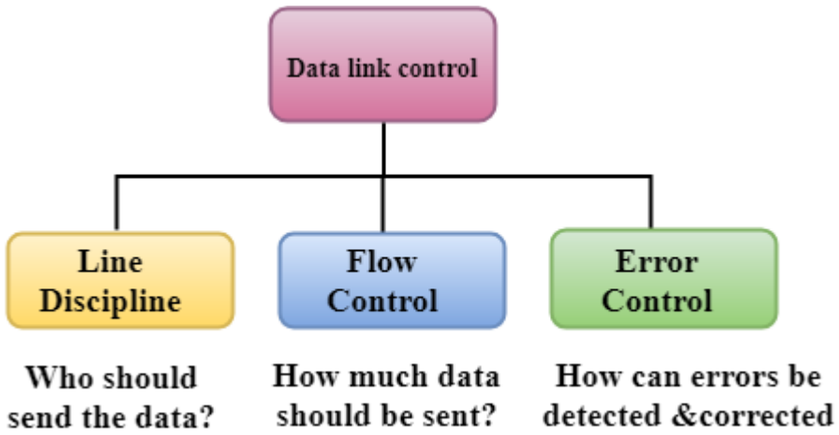
- *The binary representation of redundant bits, i.e., r4r2r1 is 100, and its corresponding decimal value is 4. Therefore, the error occurs in a 4^{th} bit position. The bit value must be changed from 1 to 0 to correct the error.*

Data Link Controls

Data Link Control is the service provided by the Data Link Layer to provide reliable data transfer over the physical medium. For example, In the half-duplex transmission mode, one device can only transmit the data at a time. If both the devices at the end of the links transmit the data simultaneously, they will collide and leads to the loss of the information. The Data link layer provides the coordination among the devices so that no collision occurs.

The Data link layer provides three functions:

- Line discipline
- Flow Control
- Error Control



Line Discipline

- Line Discipline is a functionality of the Data link layer that provides the coordination among the link systems. It determines which device can send, and when it can send the data.

Line Discipline can be achieved in two ways:

- ENQ/ACK
- Poll/select

END/ACK

END/ACK stands for Enquiry/Acknowledgement is used when there is no wrong receiver available on the link and having a dedicated path between the two devices so that the device capable of receiving the transmission is the intended one.

END/ACK coordinates which device will start the transmission and whether the recipient is ready or not.

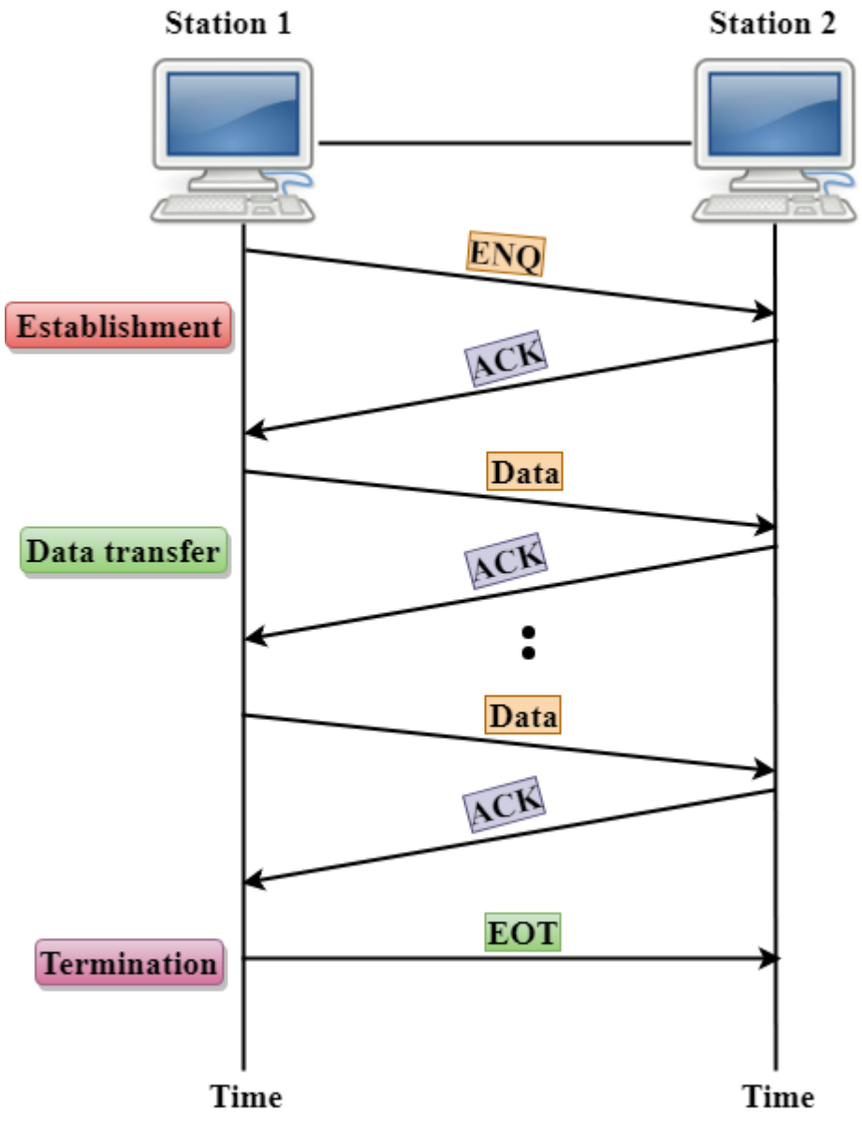
Working of END/ACK

The transmitter transmits the frame called an Enquiry (ENQ) asking whether the receiver is available to receive the data or not.

The receiver responses either with the positive acknowledgement(ACK) or with the negative acknowledgement(NACK) where positive acknowledgement means that the receiver is ready to receive the transmission and negative acknowledgement means that the receiver is unable to accept the transmission.

Following are the responses of the receiver:

- If the response to the ENQ is positive, the sender will transmit its data, and once all of its data has been transmitted, the device finishes its transmission with an EOT (END-of-Transmission) frame.
- If the response to the ENQ is negative, then the sender disconnects and restarts the transmission at another time.
- If the response is neither negative nor positive, the sender assumes that the ENQ frame was lost during the transmission and makes three attempts to establish a link before giving up.



Poll/Select

The Poll/Select method of line discipline works with those topologies where one device is designated as a primary station, and other devices are secondary stations.

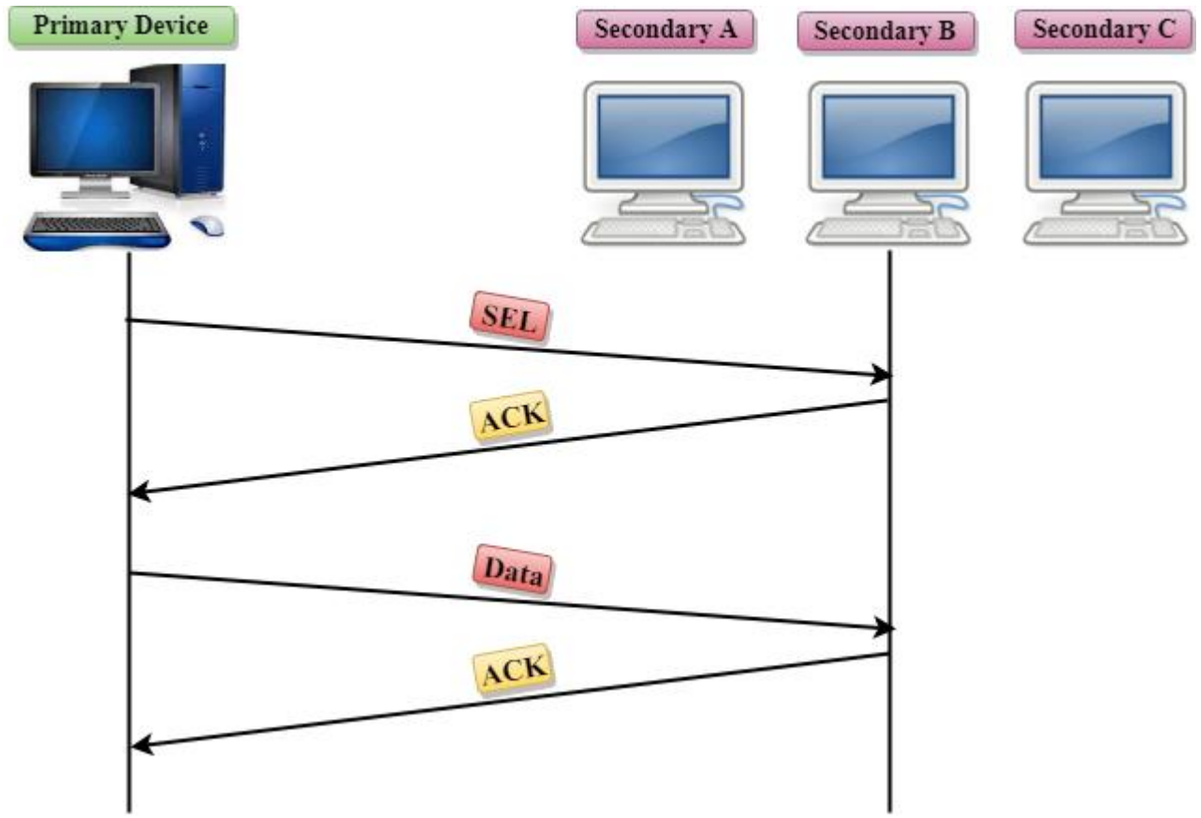
Working of Poll/Select

- In this, the primary device and multiple secondary devices consist of a single transmission line, and all the exchanges are made through the primary device even though the destination is a secondary device.

- The primary device has control over the communication link, and the secondary device follows the instructions of the primary device.
- The primary device determines which device is allowed to use the communication channel. Therefore, we can say that it is an initiator of the session.
- If the primary device wants to receive the data from the secondary device, it asks the secondary device that they anything to send, this process is known as polling.
- If the primary device wants to send some data to the secondary device, then it tells the target secondary to get ready to receive the data, this process is known as selecting.

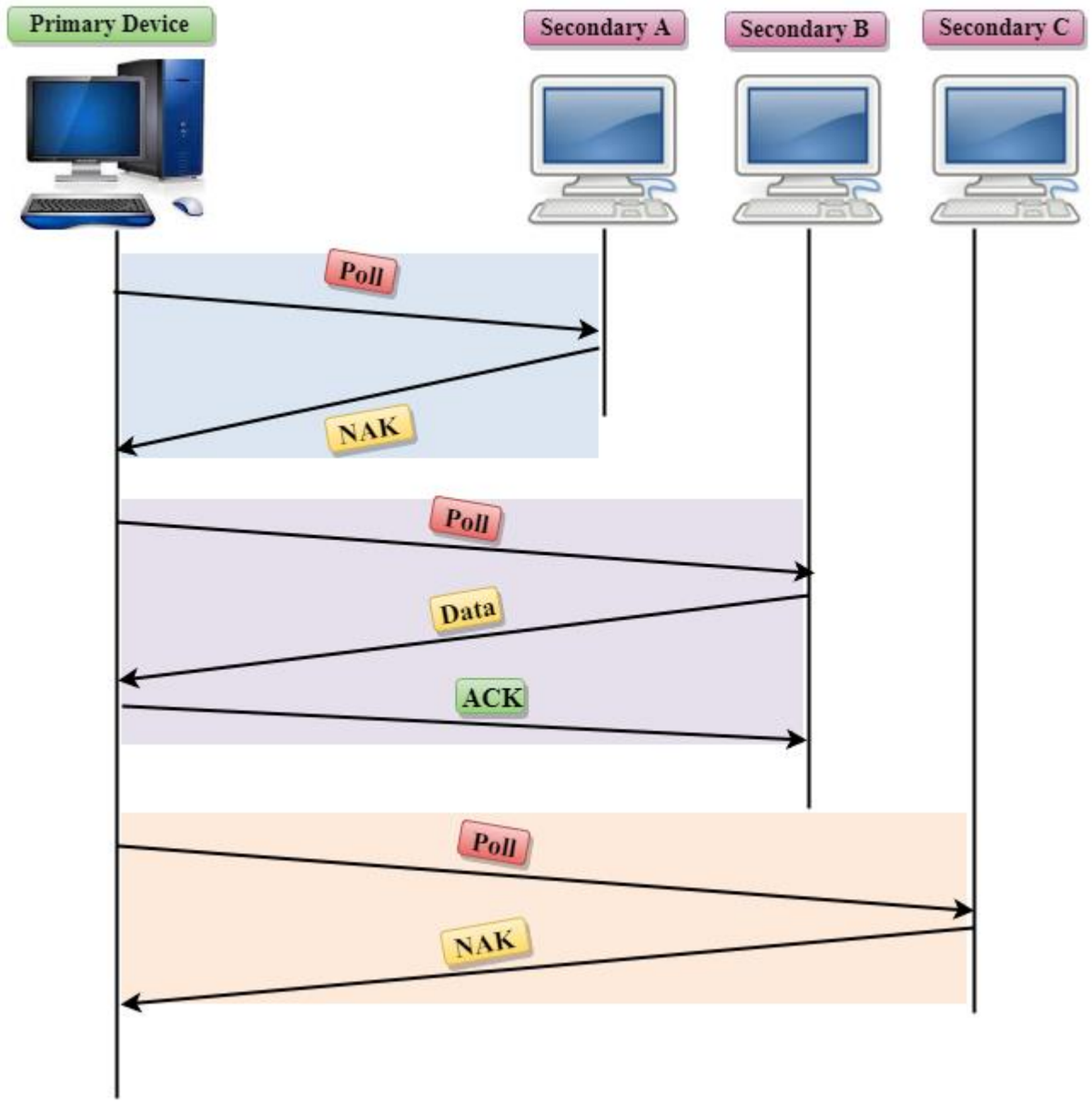
Select

- The select mode is used when the primary device has something to send.
- When the primary device wants to send some data, then it alerts the secondary device for the upcoming transmission by transmitting a Select (SEL) frame, one field of the frame includes the address of the intended secondary device.
- When the secondary device receives the SEL frame, it sends an acknowledgement that indicates the secondary ready status.
- If the secondary device is ready to accept the data, then the primary device sends two or more data frames to the intended secondary device. Once the data has been transmitted, the secondary sends an acknowledgement specifies that the data has been received.



Poll

- The Poll mode is used when the primary device wants to receive some data from the secondary device.
- When a primary device wants to receive the data, then it asks each device whether it has anything to send.
- Firstly, the primary asks (poll) the first secondary device, if it responds with the NACK (Negative Acknowledgement) means that it has nothing to send. Now, it approaches the second secondary device, it responds with the ACK means that it has the data to send. The secondary device can send more than one frame one after another or sometimes it may be required to send ACK before sending each one, depending on the type of the protocol being used.



Flow Control

- It is a set of procedures that tells the sender how much data it can transmit before the data overwhelms the receiver.
- The receiving device has limited speed and limited memory to store the data. Therefore, the receiving device must be able to inform the sending device to stop the transmission temporarily before the limits are reached.
- It requires a buffer, a block of memory for storing the information until they are processed.

Two methods have been developed to control the flow of data:

- Stop-and-wait
- Sliding window

Stop-and-wait

- In the Stop-and-wait method, the sender waits for an acknowledgement after every frame it sends.
- When acknowledgement is received, then only next frame is sent. The process of alternately sending and waiting of a frame continues until the sender transmits the EOT (End of transmission) frame.

Advantage of Stop-and-wait

The Stop-and-wait method is simple as each frame is checked and acknowledged before the next frame is sent.

Disadvantage of Stop-and-wait

Stop-and-wait technique is inefficient to use as each frame must travel across all the way to the receiver, and an acknowledgement travels all the way before the next frame is sent. Each frame sent and received uses the entire time needed to traverse the link.

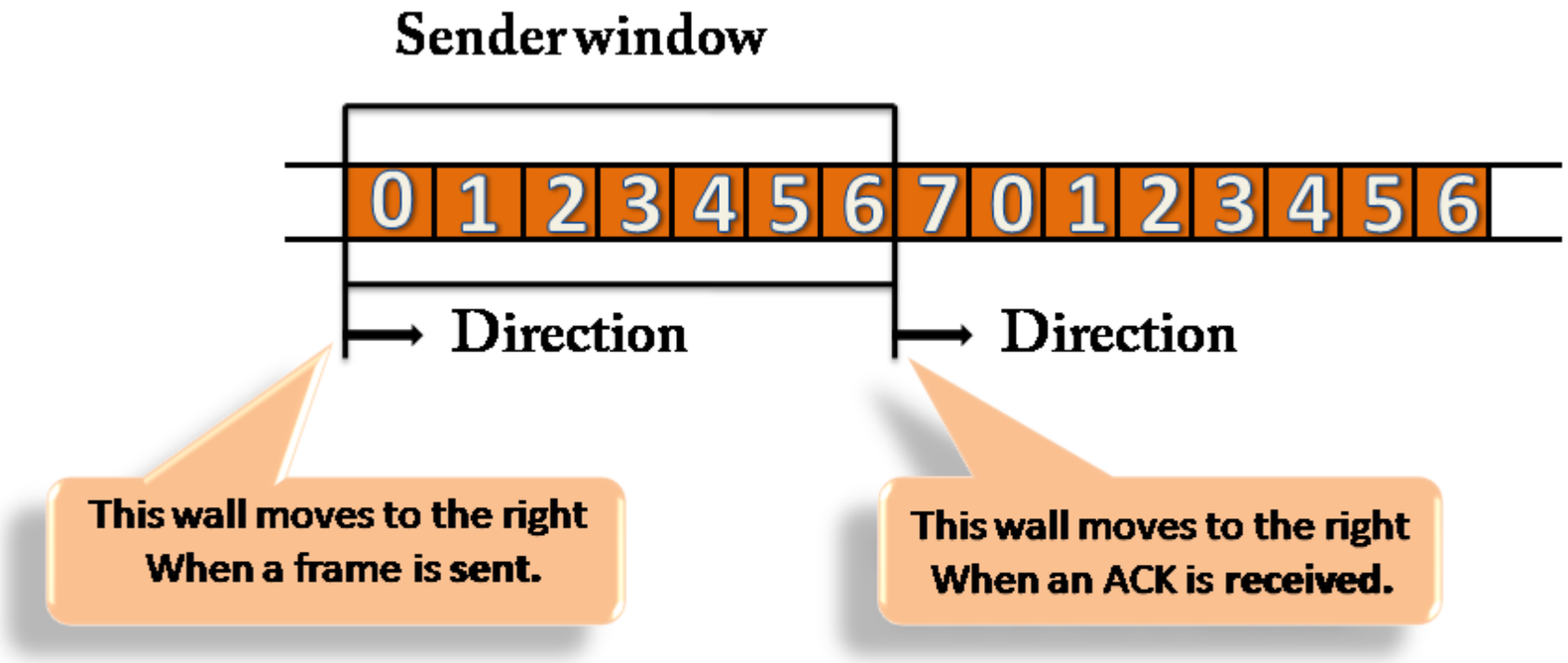
Sliding Window

- The Sliding Window is a method of flow control in which a sender can transmit the several frames before getting an acknowledgement.
- In Sliding Window Control, multiple frames can be sent one after the another due to which capacity of the communication channel can be utilized efficiently.
- A single ACK acknowledge multiple frames.
- Sliding Window refers to imaginary boxes at both the sender and receiver end.
- The window can hold the frames at either end, and it provides the upper limit on the number of frames that can be transmitted before the acknowledgement.
- Frames can be acknowledged even when the window is not completely filled.

- The window has a specific size in which they are numbered as modulo-n means that they are numbered from 0 to n-1. For example, if n = 8, the frames are numbered from 0,1,2,3,4,5,6,7,0,1,2,3,4,5,6,7,0,1.....
- The size of the window is represented as n-1. Therefore, maximum n-1 frames can be sent before acknowledgement.
- When the receiver sends the ACK, it includes the number of the next frame that it wants to receive. For example, to acknowledge the string of frames ending with frame number 4, the receiver will send the ACK containing the number 5. When the sender sees the ACK with the number 5, it got to know that the frames from 0 through 4 have been received.

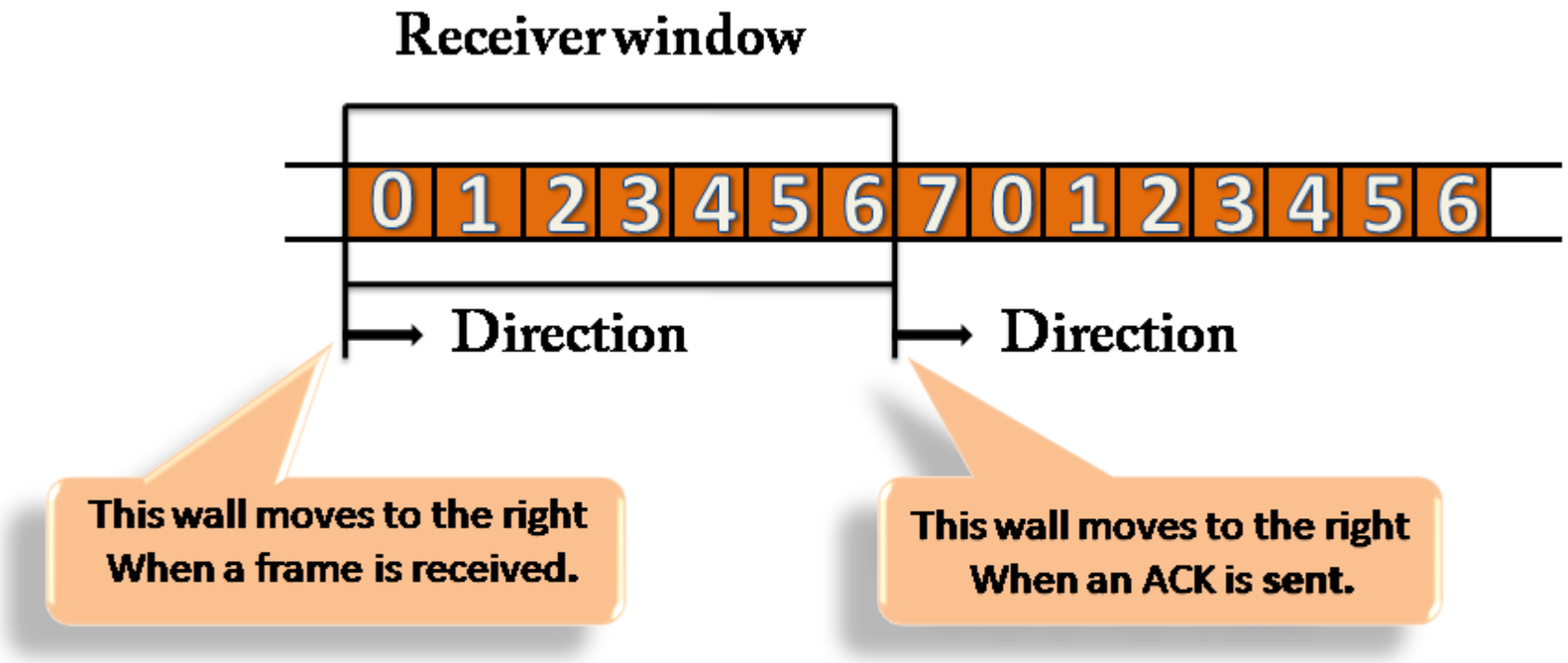
Sender Window

- At the beginning of a transmission, the sender window contains n-1 frames, and when they are sent out, the left boundary moves inward shrinking the size of the window. For example, if the size of the window is w if three frames are sent out, then the number of frames left out in the sender window is w-3.
- Once the ACK has arrived, then the sender window expands to the number which will be equal to the number of frames acknowledged by ACK.
- For example, the size of the window is 7, and if frames 0 through 4 have been sent out and no acknowledgement has arrived, then the sender window contains only two frames, i.e., 5 and 6. Now, if ACK has arrived with a number 4 which means that 0 through 3 frames have arrived undamaged and the sender window is expanded to include the next four frames. Therefore, the sender window contains six frames (5,6,7,0,1,2).



Receiver Window

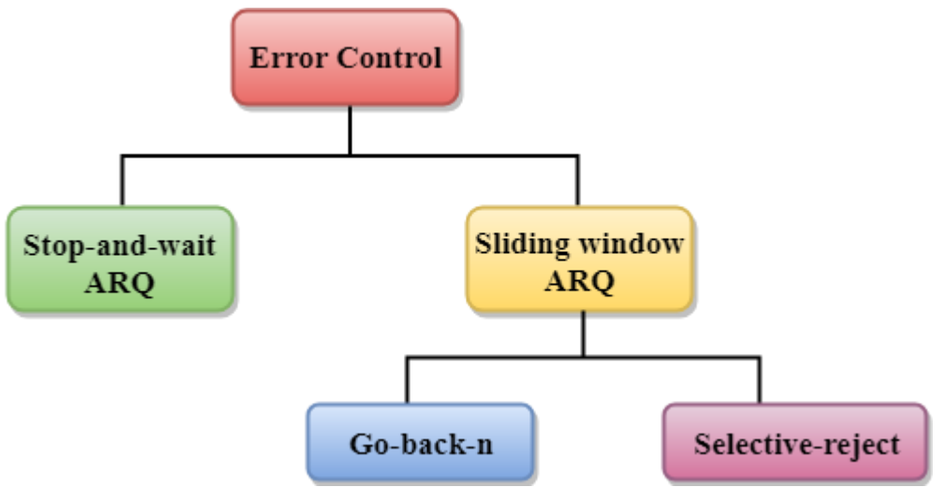
- At the beginning of transmission, the receiver window does not contain n frames, but it contains n-1 spaces for frames.
- When the new frame arrives, the size of the window shrinks.
- The receiver window does not represent the number of frames received, but it represents the number of frames that can be received before an ACK is sent. For example, the size of the window is w, if three frames are received then the number of spaces available in the window is (w-3).
- Once the acknowledgement is sent, the receiver window expands by the number equal to the number of frames acknowledged.
- Suppose the size of the window is 7 means that the receiver window contains seven spaces for seven frames. If the one frame is received, then the receiver window shrinks and moving the boundary from 0 to 1. In this way, window shrinks one by one, so window now contains the six spaces. If frames from 0 through 4 have sent, then the window contains two spaces before an acknowledgement is sent.



Error Control

Error Control is a technique of error detection and retransmission.

Categories of Error Control:



Stop-and-wait ARQ

Stop-and-wait ARQ is a technique used to retransmit the data in case of damaged or lost frames.

This technique works on the principle that the sender will not transmit the next frame until it receives the acknowledgement of the last transmitted frame.

Four features are required for the retransmission:

- The sending device keeps a copy of the last transmitted frame until the acknowledgement is received. Keeping the copy allows the sender to retransmit the data if the frame is not received correctly.
- Both the data frames and the ACK frames are numbered alternately 0 and 1 so that they can be identified individually. Suppose data 1 frame acknowledges the data 0 frame means that the data 0 frame has been arrived correctly and expects to receive data 1 frame.
- If an error occurs in the last transmitted frame, then the receiver sends the NAK frame which is not numbered. On receiving the NAK frame, sender retransmits the data.
- It works with the timer. If the acknowledgement is not received within the allotted time, then the sender assumes that the frame is lost during the transmission, so it will retransmit the frame.

Two possibilities of the retransmission:

- **Damaged Frame:** When the receiver receives a damaged frame, i.e., the frame contains an error, then it returns the NAK frame. For example, when the data 0 frame is sent, and then the receiver sends the ACK 1 frame means that the data 0 has arrived correctly, and transmits the data 1 frame. The sender transmits the next frame: data 1. It reaches undamaged, and the receiver returns ACK 0. The sender transmits the next frame: data 0. The receiver reports an error and returns the NAK frame. The sender retransmits the data 0 frame.
- **Lost Frame:** Sender is equipped with the timer and starts when the frame is transmitted. Sometimes the frame has not arrived at the receiving end so that it can be acknowledged neither positively nor negatively. The sender waits for acknowledgement until the timer goes off. If the timer goes off, it retransmits the last transmitted frame.

Sliding Window ARQ

SlidingWindow ARQ is a technique used for continuous transmission error control.

Three Features used for retransmission:

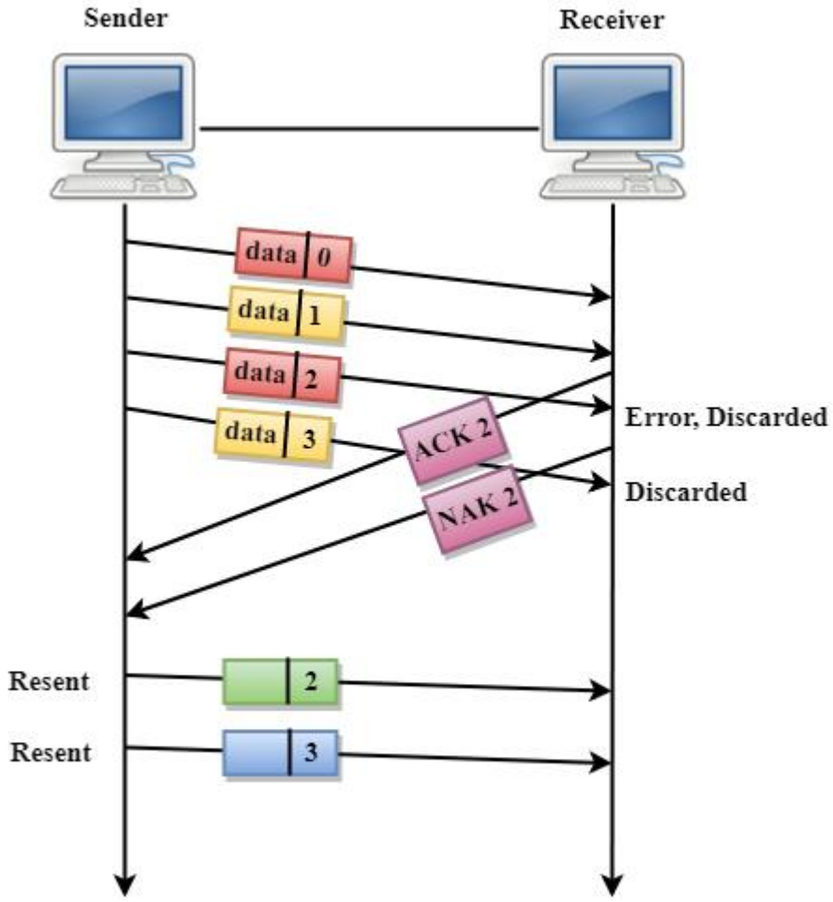
- In this case, the sender keeps the copies of all the transmitted frames until they have been acknowledged. Suppose the frames from 0 through 4 have been transmitted, and the last acknowledgement was for frame 2, the sender has to keep the copies of frames 3 and 4 until they receive correctly.
- The receiver can send either NAK or ACK depending on the conditions. The NAK frame tells the sender that the data have been received damaged. Since the sliding window is a continuous transmission mechanism, both ACK and NAK must be numbered for the identification of a frame. The ACK frame consists of a number that represents the next frame which the receiver expects to receive. The NAK frame consists of a number that represents the damaged frame.
- The sliding window ARQ is equipped with the timer to handle the lost acknowledgements. Suppose then n-1 frames have been sent before receiving any acknowledgement. The sender waits for the acknowledgement, so it starts the timer and waits before sending any more. If the allotted time runs out, the sender retransmits one or all the frames depending upon the protocol used.

Two protocols used in sliding window ARQ:

- **Go-Back-n ARQ:** In Go-Back-N ARQ protocol, if one frame is lost or damaged, then it retransmits all the frames after which it does not receive the positive ACK.

Three possibilities can occur for retransmission:

- **Damaged Frame:** When the frame is damaged, then the receiver sends a NAK frame.

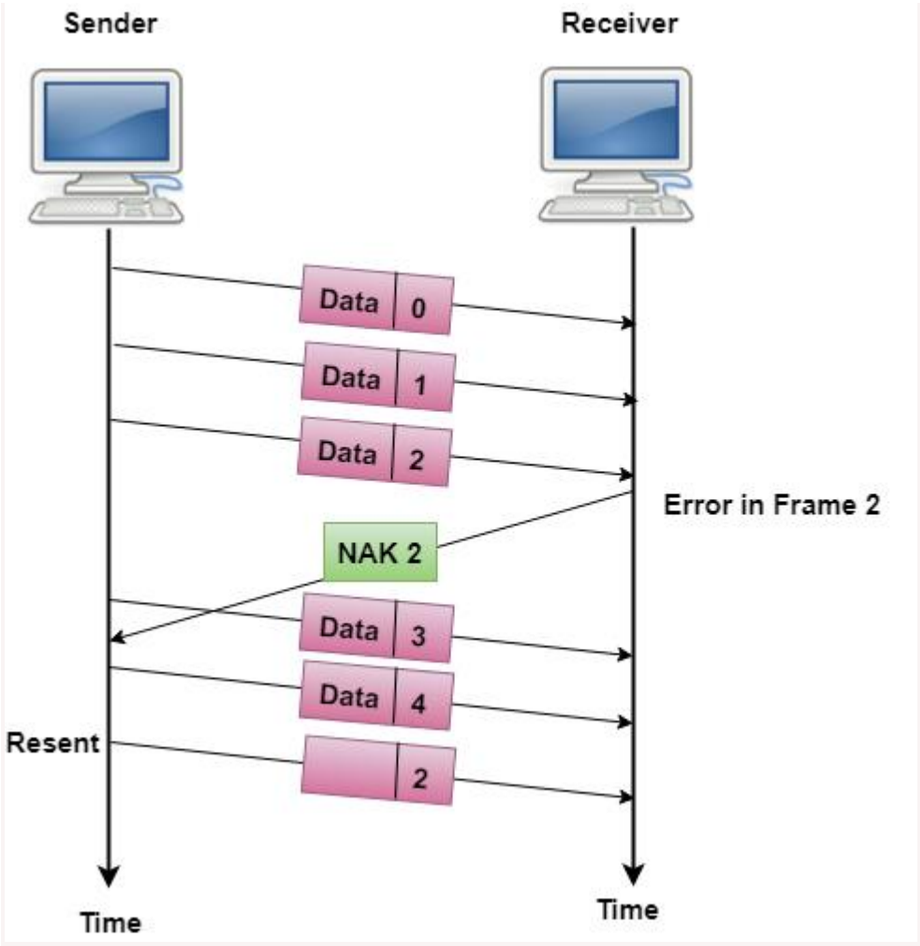


In the above figure, three frames have been transmitted before an error discovered in the third frame. In this case, ACK 2 has been returned telling that the frames 0,1 have been received successfully without any error. The receiver discovers the error in data 2 frame, so it returns the NAK 2 frame. The frame 3 is also discarded as it is transmitted after the damaged frame. Therefore, the sender retransmits the frames 2,3.

- **Lost Data Frame:** In Sliding window protocols, data frames are sent sequentially. If any of the frames is lost, then the next frame arrive at the receiver is out of sequence. The receiver checks the sequence number of each of the frame, discovers the frame that has been skipped, and returns the NAK for the missing frame. The sending device retransmits the frame indicated by NAK as well as the frames transmitted after the lost frame.
- **Lost Acknowledgement:** The sender can send as many frames as the windows allow before waiting for any acknowledgement. Once the limit of the window is reached, the sender has no more frames to send; it must wait for the acknowledgement. If the acknowledgement is lost, then the sender could wait forever. To avoid such situation, the sender is equipped with the timer that starts counting whenever the window capacity is reached. If the acknowledgement has not been received within the time limit, then the sender retransmits the frame since the last ACK.

Selective-Reject ARQ

- Selective-Reject ARQ technique is more efficient than Go-Back-n ARQ.
- In this technique, only those frames are retransmitted for which negative acknowledgement (NAK) has been received.
- The receiver storage buffer keeps all the damaged frames on hold until the frame in error is correctly received.
- The receiver must have an appropriate logic for reinserting the frames in a correct order.
- The sender must consist of a searching mechanism that selects only the requested frame for retransmission.



Network Layer

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Network Layer

- The Network Layer is the third layer of the OSI model.
- It handles the service requests from the transport layer and further forwards the service request to the data link layer.
- The network layer translates the logical addresses into physical addresses
- It determines the route from the source to the destination and also manages the traffic problems such as switching, routing and controls the congestion of data packets.
- The main role of the network layer is to move the packets from sending host to the receiving host.

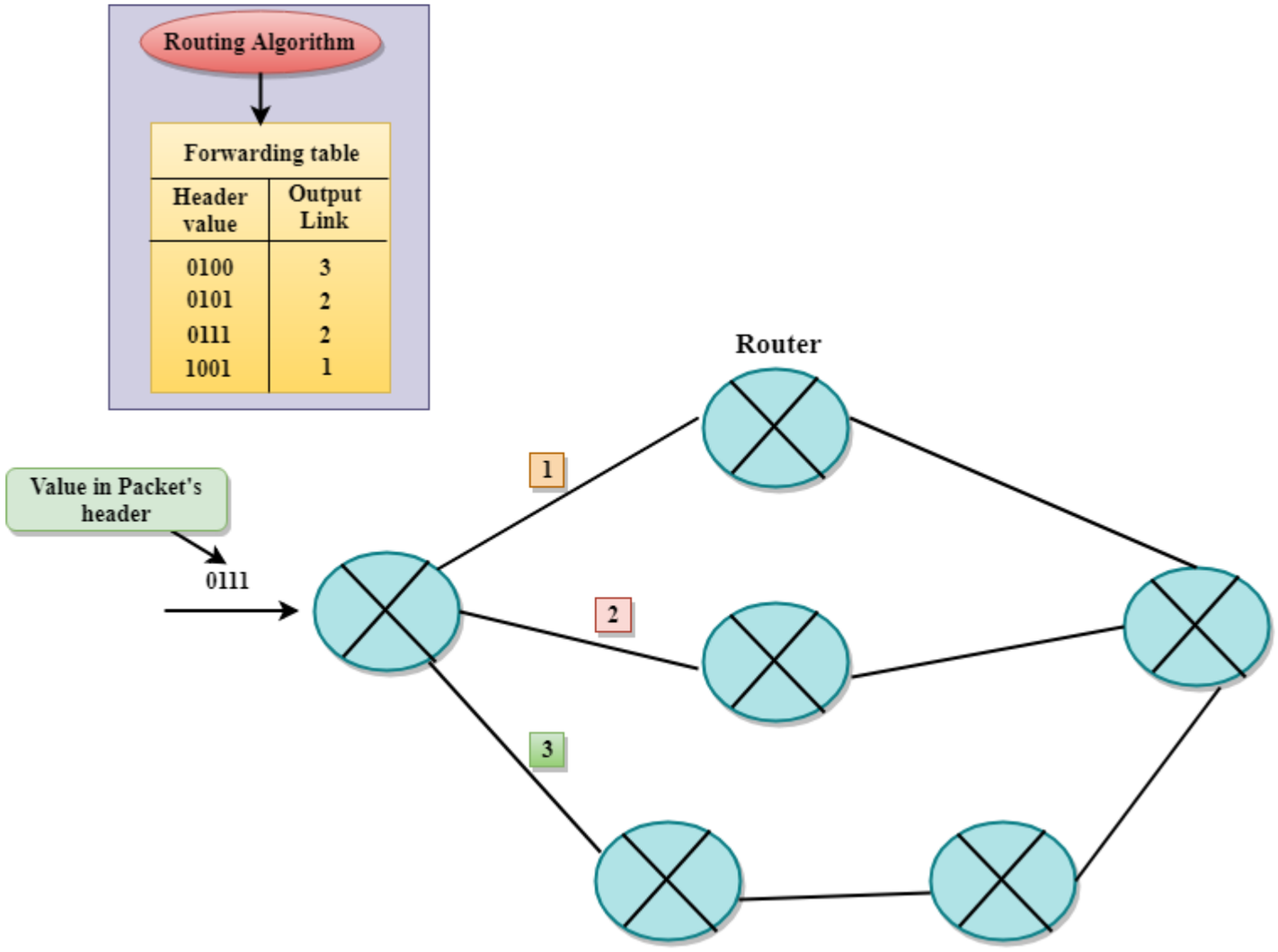
The main functions performed by the network layer are:

- **Routing:** When a packet reaches the router's input link, the router will move the packets to the router's output link. For example, a packet from S1 to R1 must be forwarded to the next router on the path to S2.
- **Logical Addressing:** The data link layer implements the physical addressing and network layer implements the logical addressing. Logical addressing is also used to distinguish between source and destination system. The network layer adds a header to the packet which includes the logical addresses of both the sender and the receiver.
- **Internetworking:** This is the main role of the network layer that it provides the logical connection between different types of networks.
- **Fragmentation:** The fragmentation is a process of breaking the packets into the smallest individual data units that travel through different networks.

Forwarding & Routing

In Network layer, a router is used to forward the packets. Every router has a forwarding table. A router forwards a packet by examining a packet's header field and then using the header field value to index into the forwarding table. The value stored in the forwarding table corresponding to the header field value indicates the router's outgoing interface link to which the packet is to be forwarded.

For example, the router with a header field value of 0111 arrives at a router, and then router indexes this header value into the forwarding table that determines the output link interface is 2. The router forwards the packet to the interface 2. The routing algorithm determines the values that are inserted in the forwarding table. The routing algorithm can be centralized or decentralized.



Routing algorithms determine the values in forwarding table

Services Provided by the Network Layer

- **Guaranteed delivery:** This layer provides the service which guarantees that the packet will arrive at its destination.
- **Guaranteed delivery with bounded delay:** This service guarantees that the packet will be delivered within a specified host-to-host delay bound.
- **In-Order packets:** This service ensures that the packet arrives at the destination in the order in which they are sent.

- **Guaranteed max jitter:** This service ensures that the amount of time taken between two successive transmissions at the sender is equal to the time between their receipt at the destination.
- **Security services:** The network layer provides security by using a session key between the source and destination host. The network layer in the source host encrypts the payloads of datagrams being sent to the destination host. The network layer in the destination host would then decrypt the payload. In such a way, the network layer maintains the data integrity and source authentication services.

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Network Addressing

- Network Addressing is one of the major responsibilities of the network layer.
- Network addresses are always logical, i.e., software-based addresses.
- A host is also known as end system that has one link to the network. The boundary between the host and link is known as an interface. Therefore, the host can have only one interface.
- A router is different from the host in that it has two or more links that connect to it. When a router forwards the datagram, then it forwards the packet to one of the links. The boundary between the router and link is known as an interface, and the router can have multiple interfaces, one for each of its links. Each interface is capable of sending and receiving the IP packets, so IP requires each interface to have an address.
- Each IP address is 32 bits long, and they are represented in the form of "dot-decimal notation" where each byte is written in the decimal form, and they are separated by the period. An IP address would look like 193.32.216.9 where 193 represents the decimal notation of first 8 bits of an address, 32 represents the decimal notation of second 8 bits of an address.

- **Let's understand through a simple example.**

  - In the above figure, a router has three interfaces labeled as 1, 2 & 3 and each router interface contains its own IP address.
  - Each host contains its own interface and IP address.
  - All the interfaces attached to the LAN 1 is having an IP address in the form of 223.1.1.xxx, and the interfaces attached to the LAN 2 and LAN 3 have an IP address in the form of 223.1.2.xxx and 223.1.3.xxx respectively.
  - Each IP address consists of two parts. The first part (first three bytes in IP address) specifies the network and second part (last byte of an IP address) specifies the host in the network.

Classful Addressing

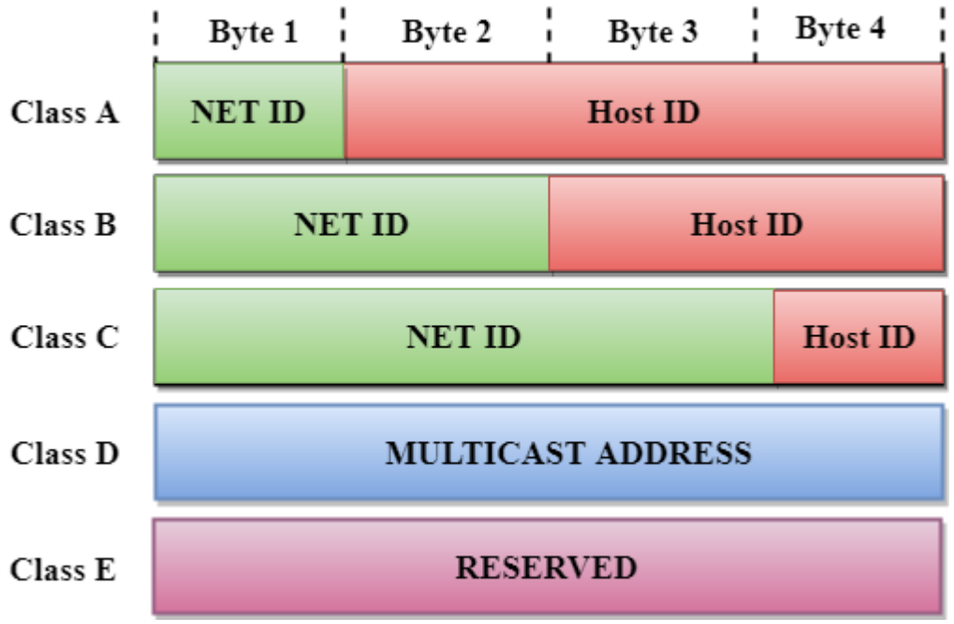
An IP address is 32-bit long. An IP address is divided into sub-classes:

  - Class A
  - Class B

- Class C
- Class D
- Class E

An ip address is divided into two parts:

- **Network ID:** It represents the number of networks.
- **Host ID:** It represents the number of hosts.



In the above diagram, we observe that each class have a specific range of IP addresses. The class of IP address is used to determine the number of bits used in a class and number of networks and hosts available in the class.

Class A

In Class A, an IP address is assigned to those networks that contain a large number of hosts.

- The network ID is 8 bits long.

- The host ID is 24 bits long.

In Class A, the first bit in higher order bits of the first octet is always set to 0 and the remaining 7 bits determine the network ID. The 24 bits determine the host ID in any network.

The total number of networks in Class A = 2^7 = 128 network address

The total number of hosts in Class A = 2^{24} - 2 = 16,777,214 host address

Class B

In Class B, an IP address is assigned to those networks that range from small-sized to large-sized networks.

- The Network ID is 16 bits long.
- The Host ID is 16 bits long.

In Class B, the higher order bits of the first octet is always set to 10, and the remaining14 bits determine the network ID. The other 16 bits determine the Host ID.

The total number of networks in Class B = 2^{14} = 16384 network address

The total number of hosts in Class B = 2^{16} - 2 = 65534 host address

Class C

In Class C, an IP address is assigned to only small-sized networks.

- The Network ID is 24 bits long.
- The host ID is 8 bits long.

In Class C, the higher order bits of the first octet is always set to 110, and the remaining 21 bits determine the network ID. The 8 bits of the host ID determine the host in a network.

The total number of networks = 2^{21} = 2097152 network address

The total number of hosts = 2^8 - 2 = 254 host address

Class D

In Class D, an IP address is reserved for multicast addresses. It does not possess subnetting. The higher order bits of the first octet is always set to 1110, and the remaining bits determines the host ID in any network.

Class E

In Class E, an IP address is used for the future use or for the research and development purposes. It does not possess any subnetting. The higher order bits of the first octet is always set to 1111, and the remaining bits determines the host ID in any network.

Rules for assigning Host ID:

The Host ID is used to determine the host within any network. The Host ID is assigned based on the following rules:

- The Host ID must be unique within any network.
- The Host ID in which all the bits are set to 0 cannot be assigned as it is used to represent the network ID of the IP address.
- The Host ID in which all the bits are set to 1 cannot be assigned as it is reserved for the multicast address.

Rules for assigning Network ID:

If the hosts are located within the same local network, then they are assigned with the same network ID. The following are the rules for assigning Network ID:

- The network ID cannot start with 127 as 127 is used by Class A.
- The Network ID in which all the bits are set to 0 cannot be assigned as it is used to specify a particular host on the local network.
- The Network ID in which all the bits are set to 1 cannot be assigned as it is reserved for the multicast address.

Classful Network Architecture

Class	Higher bits	NET ID bits	HOST ID bits	No.of networks	No.of hosts per network	Range
A	0	8	24	2^7	2^{24}	0.0.0.0 to 127.255.255.255
B	10	16	16	2^{14}	2^{16}	128.0.0.0 to 191.255.255.255
C	110	24	8	2^{21}	2^8	192.0.0.0 to 223.255.255.255
D	1110	Not Defined	Not Defined	Not Defined	Not Defined	224.0.0.0 to 239.255.255.255
E	1111	Not Defined	Not Defined	Not Defined	Not Defined	240.0.0.0 to 255.255.255.255

Routing

- A Router is a process of selecting path along which the data can be transferred from source to the destination. Routing is performed by a special device known as a router.
- A Router works at the network layer in the OSI model and internet layer in TCP/IP model
- A router is a networking device that forwards the packet based on the information available in the packet header and forwarding table.
- The routing algorithms are used for routing the packets. The routing algorithm is nothing but a software responsible for deciding the optimal path through which packet can be transmitted.
- The routing protocols use the metric to determine the best path for the packet delivery. The metric is the standard of measurement such as hop count, bandwidth, delay, current load on the path, etc. used by the routing algorithm to determine the optimal path to the destination.
- The routing algorithm initializes and maintains the routing table for the process of path determination.

Routing Metrics and Costs

Routing metrics and costs are used for determining the best route to the destination. The factors used by the protocols to determine the shortest path, these factors are known as a metric.

Metrics are the network variables used to determine the best route to the destination. For some protocols use the static metrics means that their value cannot be changed and for some other routing protocols use the dynamic metrics means that their value can be assigned by the system administrator.

The most common metric values are given below:

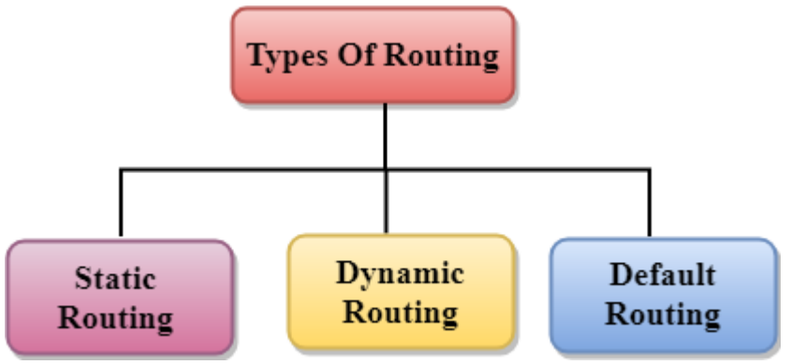
- **Hop count:** Hop count is defined as a metric that specifies the number of passes through internetworking devices such as a router, a packet must travel in a route to move from source to the destination. If the routing protocol considers the hop as a primary metric value, then the path with the least hop count will be considered as the best path to move from source to the destination.
- **Delay:** It is a time taken by the router to process, queue and transmit a datagram to an interface. The protocols use this metric to determine the delay values for all the links along the path end-to-end. The path having the lowest delay value will be considered as the best path.
- **Bandwidth:** The capacity of the link is known as a bandwidth of the link. The bandwidth is measured in terms of bits per second. The link that has a higher transfer rate like gigabit is preferred over the link that has the lower capacity like 56 kb. The protocol will determine the bandwidth capacity for all the links along the path, and the overall higher bandwidth will be considered as the best route.
- **Load:** Load refers to the degree to which the network resource such as a router or network link is busy. A Load can be calculated in a variety of ways such as CPU utilization, packets processed per second. If the traffic increases, then the load value will also be increased. The load value changes with respect to the change in the traffic.
- **Reliability:** Reliability is a metric factor may be composed of a fixed value. It depends on the network links, and its value is measured dynamically. Some networks go down more often than others. After network failure, some network links repaired more easily than other network links. Any reliability factor can be considered for the assignment of reliability ratings, which are generally numeric values assigned by the system administrator.

Types of Routing

Routing can be classified into three categories:

- Static Routing

- Default Routing
- Dynamic Routing



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.

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.

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.

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Network Layer Protocols

TCP/IP supports the following protocols:

ARP

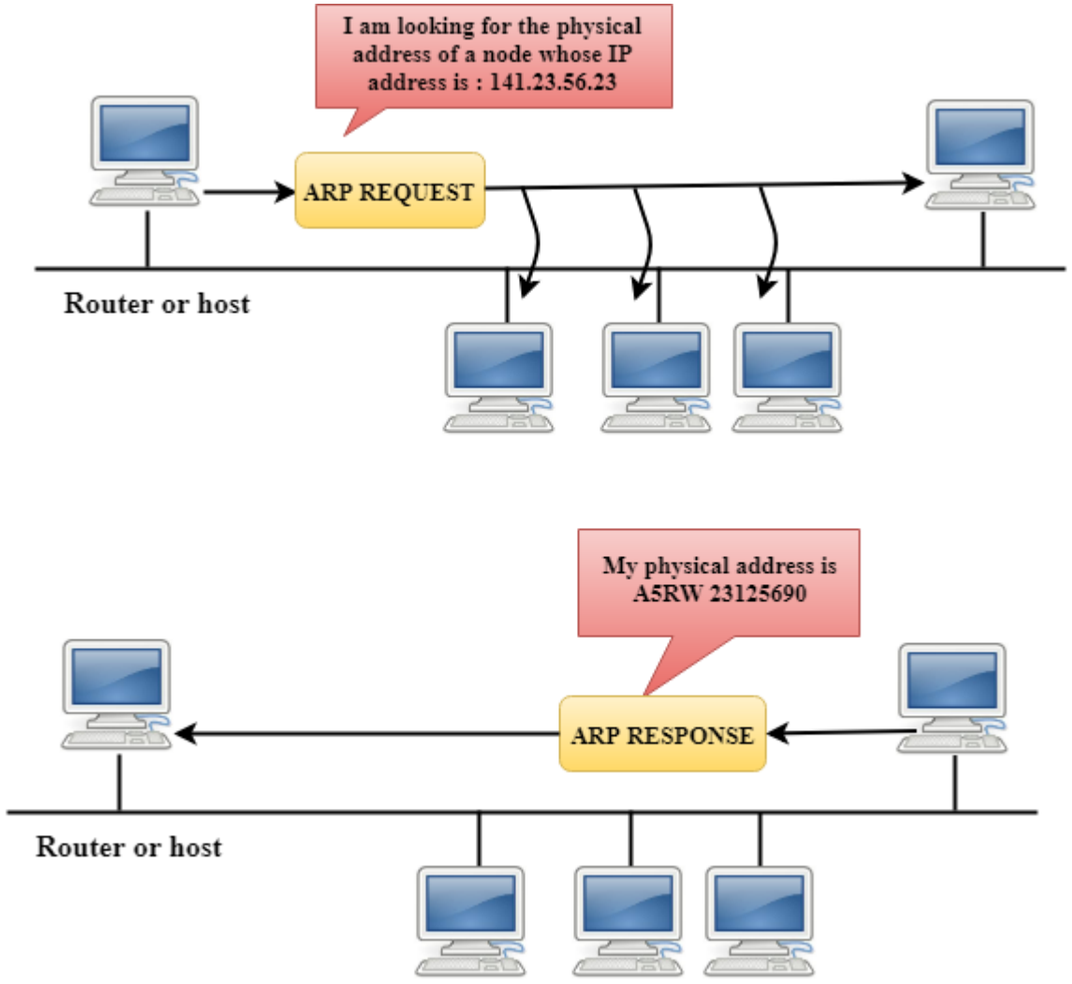
- ARP stands for Address Resolution Protocol.
- It is used to associate an IP address with the MAC address.
- Each device on the network is recognized by the MAC address imprinted on the NIC. Therefore, we can say that devices need the MAC address for communication on a local area network. MAC address can be changed easily. For example, if the NIC on a particular machine fails, the MAC address changes but IP address does not change. ARP is used to find the MAC address of the node when an internet address is known.

Note: MAC address: The MAC address is used to identify the actual device.

IP address: It is an address used to locate a device on the network.

How ARP works

If the host wants to know the physical address of another host on its network, then it sends an ARP query packet that includes the IP address and broadcast it over the network. Every host on the network receives and processes the ARP packet, but only the intended recipient recognizes the IP address and sends back the physical address. The host holding the datagram adds the physical address to the cache memory and to the datagram header, then sends back to the sender.

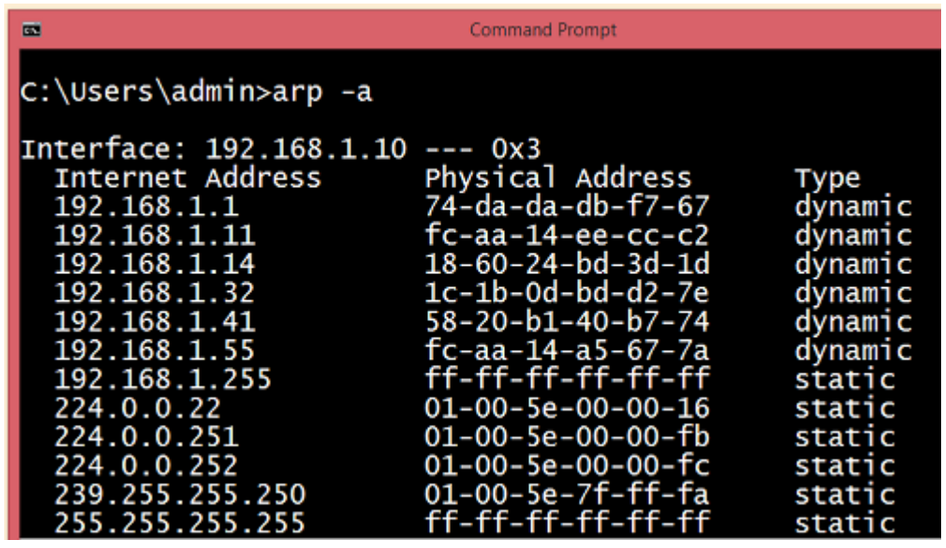


Steps taken by ARP protocol

If a device wants to communicate with another device, the following steps are taken by the device:

- The device will first look at its internet list, called the ARP cache to check whether an IP address contains a matching MAC address or not. It will check the ARP cache in command prompt by using a command **arp-a**.

- If ARP cache is empty, then device broadcast the message to the entire network asking each device for a matching MAC address.
- The device that has the matching IP address will then respond back to the sender with its MAC address
- Once the MAC address is received by the device, then the communication can take place between two devices.
- If the device receives the MAC address, then the MAC address gets stored in the ARP cache. We can check the ARP cache in command prompt by using a command arp -a.



Note: ARP cache is used to make a network more efficient.

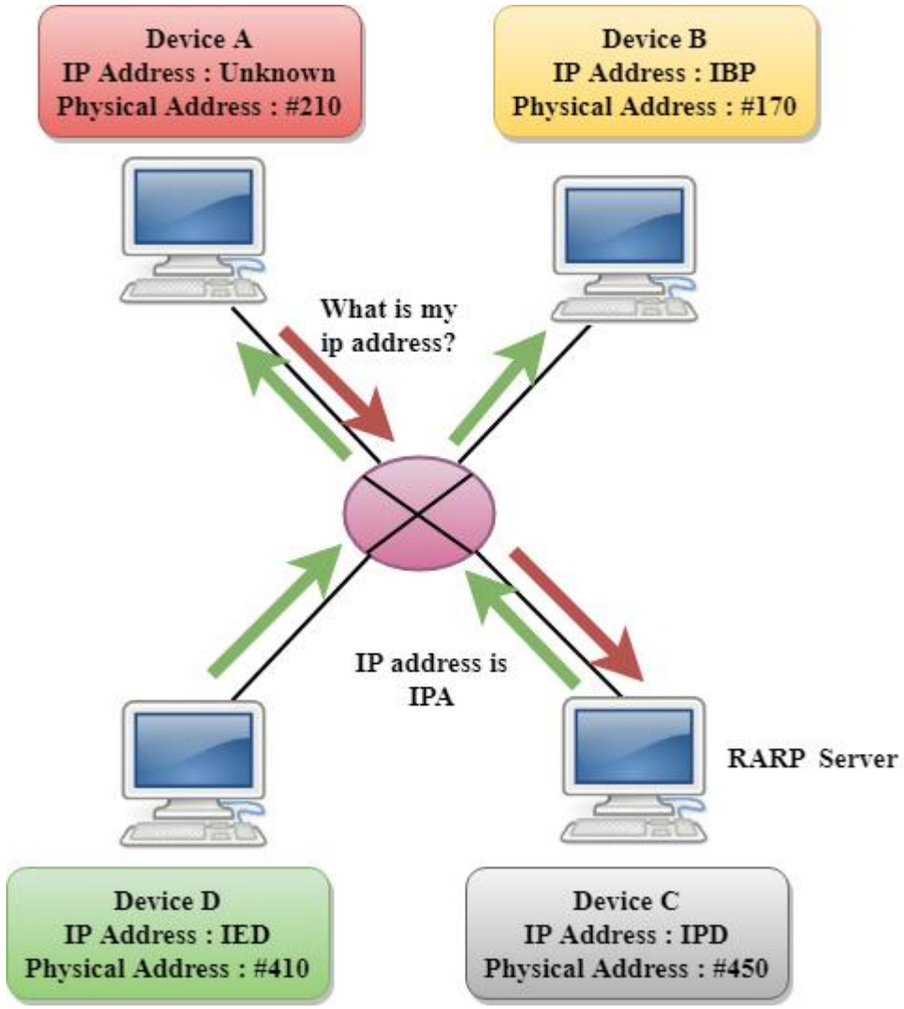
In the above screenshot, we observe the association of IP address to the MAC address.

There are two types of ARP entries:

- **Dynamic entry:** It is an entry which is created automatically when the sender broadcast its message to the entire network. Dynamic entries are not permanent, and they are removed periodically.
- **Static entry:** It is an entry where someone manually enters the IP to MAC address association by using the ARP command utility.

RARP

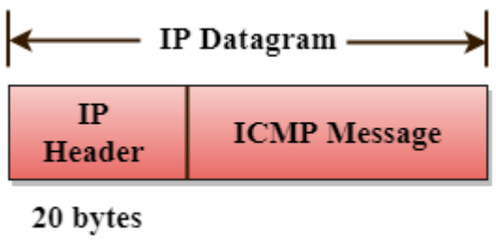
- RARP stands for **Reverse Address Resolution Protocol**.
- If the host wants to know its IP address, then it broadcast the RARP query packet that contains its physical address to the entire network. A RARP server on the network recognizes the RARP packet and responds back with the host IP address.
- The protocol which is used to obtain the IP address from a server is known as **Reverse Address Resolution Protocol**.
- The message format of the RARP protocol is similar to the ARP protocol.
- Like ARP frame, RARP frame is sent from one machine to another encapsulated in the data portion of a frame.



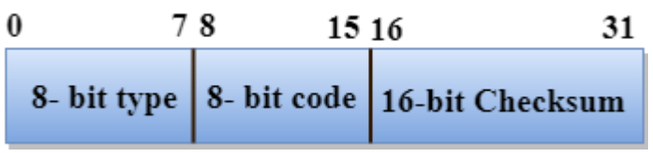
ICMP

- ICMP stands for Internet Control Message Protocol.
- The ICMP is a network layer protocol used by hosts and routers to send the notifications of IP datagram problems back to the sender.
- ICMP uses echo test/reply to check whether the destination is reachable and responding.
- ICMP handles both control and error messages, but its main function is to report the error but not to correct them.

- An IP datagram contains the addresses of both source and destination, but it does not know the address of the previous router through which it has been passed. Due to this reason, ICMP can only send the messages to the source, but not to the immediate routers.
- ICMP protocol communicates the error messages to the sender. ICMP messages cause the errors to be returned back to the user processes.
- ICMP messages are transmitted within IP datagram.



The Format of an ICMP message



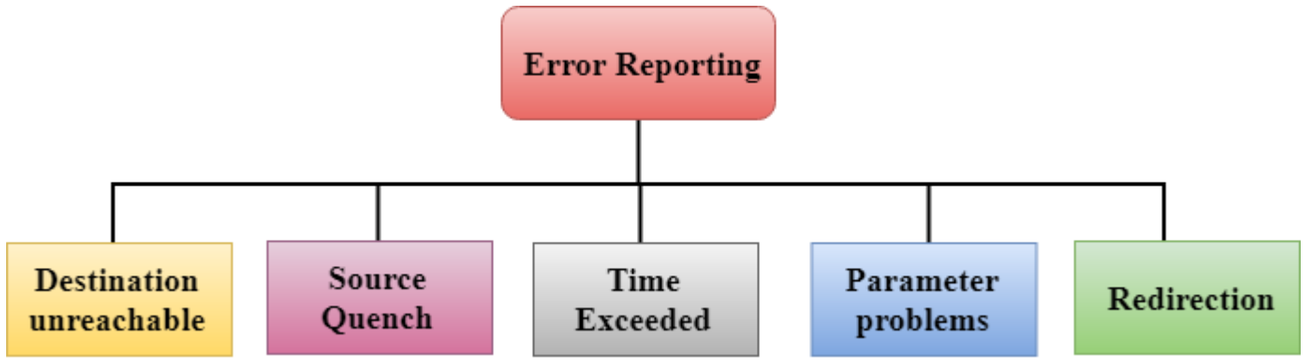
- The first field specifies the type of the message.
- The second field specifies the reason for a particular message type.
- The checksum field covers the entire ICMP message.

Error Reporting

ICMP protocol reports the error messages to the sender.

Five types of errors are handled by the ICMP protocol:

- Destination unreachable
- Source Quench
- Time Exceeded
- Parameter problems
- Redirection



- **Destination unreachable:** The message of "Destination Unreachable" is sent from receiver to the sender when destination cannot be reached, or packet is discarded when the destination is not reachable.
- **Source Quench:** The purpose of the source quench message is congestion control. The message sent from the congested router to the source host to reduce the transmission rate. ICMP will take the IP of the discarded packet and then add the source quench message to the IP datagram to inform the source host to reduce its transmission rate. The source host will reduce the transmission rate so that the router will be free from congestion.
- **Time Exceeded:** Time Exceeded is also known as "Time-To-Live". It is a parameter that defines how long a packet should live before it would be discarded.

There are two ways when Time Exceeded message can be generated:

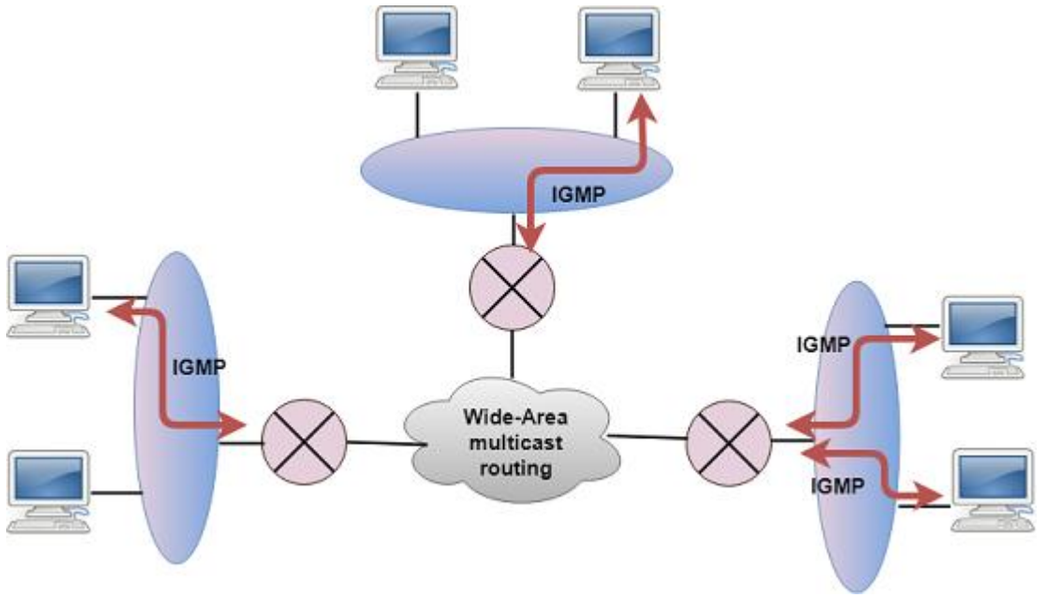
Sometimes packet discarded due to some bad routing implementation, and this causes the looping issue and network congestion. Due to the looping issue, the value of TTL keeps on decrementing, and when it reaches zero, the router discards the datagram. However, when the datagram is discarded by the router, the time exceeded message will be sent by the router to the source host.

When destination host does not receive all the fragments in a certain time limit, then the received fragments are also discarded, and the destination host sends time Exceeded message to the source host.

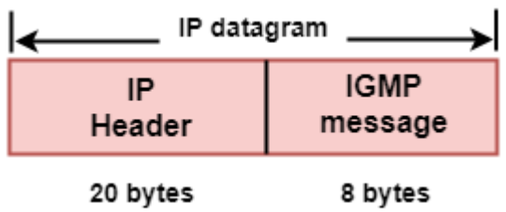
- **Parameter problems:** When a router or host discovers any missing value in the IP datagram, the router discards the datagram, and the "parameter problem" message is sent back to the source host.
- **Redirection:** Redirection message is generated when host consists of a small routing table. When the host consists of a limited number of entries due to which it sends the datagram to a wrong router. The router that receives a datagram will forward a datagram to a correct router and also sends the "Redirection message" to the host to update its routing table.

IGMP

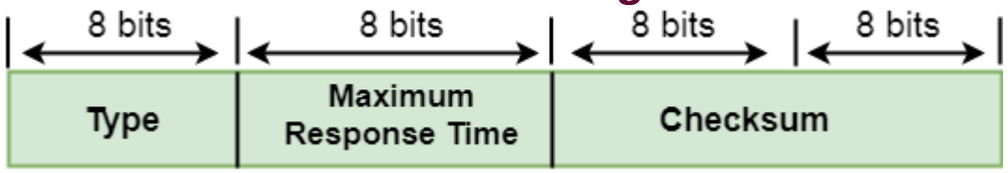
- IGMP stands for **Internet Group Message Protocol**.
- The IP protocol supports two types of communication:
  - **Unicasting:** It is a communication between one sender and one receiver. Therefore, we can say that it is one-to-one communication.
  - **Multicasting:** Sometimes the sender wants to send the same message to a large number of receivers simultaneously. This process is known as multicasting which has one-to-many communication.
- The IGMP protocol is used by the hosts and router to support multicasting.
- The IGMP protocol is used by the hosts and router to identify the hosts in a LAN that are the members of a group.



- IGMP is a part of the IP layer, and IGMP has a fixed-size message.
- The IGMP message is encapsulated within an IP datagram.



The Format of IGMP message



Where,

Type: It determines the type of IGMP message. There are three types of IGMP message: Membership Query, Membership Report and Leave Report.

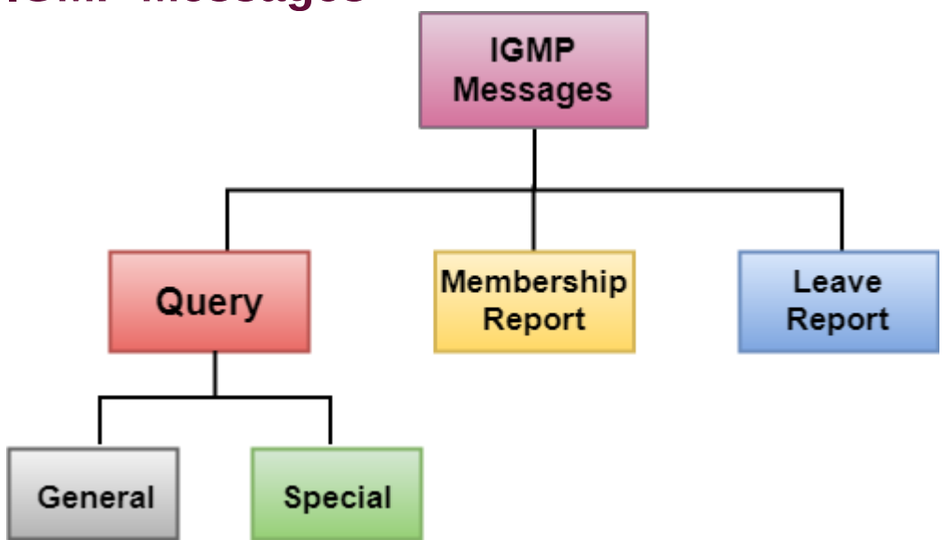
Maximum Response Time: This field is used only by the Membership Query message. It determines the maximum time the host can send the Membership Report message in response to the Membership Query message.

Checksum: It determines the entire payload of the IP datagram in which IGMP message is encapsulated.

Group Address: The behavior of this field depends on the type of the message sent.

- **For Membership Query**, the group address is set to zero for General Query and set to multicast group address for a specific query.
- **For Membership Report**, the group address is set to the multicast group address.
- **For Leave Group**, it is set to the multicast group address.

IGMP Messages



- **Membership Query message**
  - This message is sent by a router to all hosts on a local area network to determine the set of all the multicast groups that have been joined by the host.
  - It also determines whether a specific multicast group has been joined by the hosts on a attached interface.
  - The group address in the query is zero since the router expects one response from a host for every group that contains one or more members on that host.
- **Membership Report message**
  - The host responds to the membership query message with a membership report message.
  - Membership report messages can also be generated by the host when a host wants to join the multicast group without waiting for a membership query message from the router.
  - Membership report messages are received by a router as well as all the hosts on an attached interface.
  - Each membership report message includes the multicast address of a single group that the host wants to join.
  - IGMP protocol does not care which host has joined the group or how many hosts are present in a single group. It only cares whether one or more attached hosts belong to a single multicast group.
  - The membership Query message sent by a router also includes a "**Maximum Response time**". After receiving a membership query message and before sending the membership report message, the host waits for the random amount of time from 0 to the maximum response time. If a host

observes that some other attached host has sent the "**Maximum Report message**", then it discards its "**Maximum Report message**" as it knows that the attached router already knows that one or more hosts have joined a single multicast group. This process is known as feedback suppression. It provides the performance optimization, thus avoiding the unnecessary transmission of a "**Membership Report message**".

- **Leave Report**
  When the host does not send the "Membership Report message", it means that the host has left the group. The host knows that there are no members in the group, so even when it receives the next query, it would not report the group.

Routing Algorithm

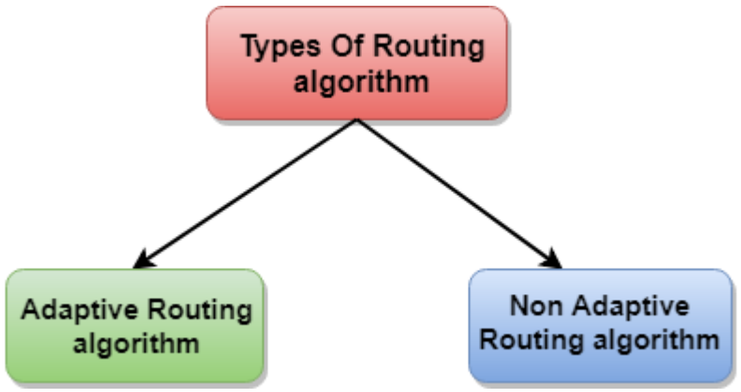
Routing algorithm

- In order to transfer the packets from source to the destination, the network layer must determine the best route through which packets can be transmitted.
- Whether the network layer provides datagram service or virtual circuit service, the main job of the network layer is to provide the best route. The routing protocol provides this job.
- The routing protocol is a routing algorithm that provides the best path from the source to the destination. The best path is the path that has the "least-cost path" from source to the destination.
- Routing is the process of forwarding the packets from source to the destination but the best route to send the packets is determined by the routing algorithm.

Classification of a Routing algorithm

The Routing algorithm is divided into two categories:

- Adaptive Routing algorithm
- Non-adaptive Routing algorithm



Adaptive Routing algorithm

- An adaptive routing algorithm is also known as dynamic routing algorithm.
- This algorithm makes the routing decisions based on the topology and network traffic.
- The main parameters related to this algorithm are hop count, distance and estimated transit time.

An adaptive routing algorithm can be classified into three parts:

- **Centralized algorithm:** It is also known as global routing algorithm as it computes the least-cost path between source and destination by using complete and global knowledge about the network. This algorithm takes the connectivity between the nodes and link cost as input, and this information is obtained before actually performing any calculation. **Link state algorithm** is referred to as a centralized algorithm since it is aware of the cost of each link in the network.
- **Isolation algorithm:** It is an algorithm that obtains the routing information by using local information rather than gathering information from other nodes.
- **Distributed algorithm:** It is also known as decentralized algorithm as it computes the least-cost path between source and destination in an iterative and distributed manner. In the decentralized algorithm, no node has the knowledge about the cost of all the network links. In the beginning, a node contains the information only about its own directly attached links and through an iterative process of calculation computes the least-cost path to the destination. A Distance vector algorithm is a decentralized algorithm as it never knows the complete path from source to the destination, instead it knows the direction through which the packet is to be forwarded along with the least cost path.

Non-Adaptive Routing algorithm

- Non Adaptive routing algorithm is also known as a static routing algorithm.

- When booting up the network, the routing information stores to the routers.
- Non Adaptive routing algorithms do not take the routing decision based on the network topology or network traffic.

The Non-Adaptive Routing algorithm is of two types:

Flooding: In case of flooding, every incoming packet is sent to all the outgoing links except the one from it has been reached. The disadvantage of flooding is that node may contain several copies of a particular packet.

Random walks: In case of random walks, a packet sent by the node to one of its neighbors randomly. An advantage of using random walks is that it uses the alternative routes very efficiently.

Differences b/w Adaptive and Non-Adaptive Routing Algorithm

Basis Of Comparison	Adaptive Routing algorithm	Non-Adaptive Routing algorithm
Define	Adaptive Routing algorithm is an algorithm that constructs the routing table based on the network conditions.	The Non-Adaptive Routing algorithm is an algorithm that constructs the static table to determine which node to send the packet.
Usage	Adaptive routing algorithm is used by dynamic routing.	The Non-Adaptive Routing algorithm is used by static routing.
Routing decision	Routing decisions are made based on topology and network traffic.	Routing decisions are the static tables.
Categorization	The types of adaptive routing algorithm, are Centralized, isolation and distributed algorithm.	The types of Non Adaptive routing algorithm are flooding and random walks.
Complexity	Adaptive Routing algorithms are more complex.	Non-Adaptive Routing algorithms are simple.

Distance Vector Routing Algorithm

- **The Distance vector algorithm is iterative, asynchronous and distributed.**
  - **Distributed:** It is distributed in that each node receives information from one or more of its directly attached neighbors, performs calculation and then distributes the result back to its neighbors.
  - **Iterative:** It is iterative in that its process continues until no more information is available to be exchanged between neighbors.
  - **Asynchronous:** It does not require that all of its nodes operate in the lock step with each other.
- The Distance vector algorithm is a dynamic algorithm.
- It is mainly used in ARPANET, and RIP.
- Each router maintains a distance table known as **Vector**.

Three Keys to understand the working of Distance Vector Routing Algorithm:

- **Knowledge about the whole network:** Each router shares its knowledge through the entire network. The Router sends its collected knowledge about the network to its neighbors.
- **Routing only to neighbors:** The router sends its knowledge about the network to only those routers which have direct links. The router sends whatever it has about the network through the ports. The information is received by the router and uses the information to update its own routing table.
- **Information sharing at regular intervals:** Within 30 seconds, the router sends the information to the neighboring routers.

Distance Vector Routing Algorithm

Let $d_x(y)$ be the cost of the least-cost path from node x to node y. The least costs are related by Bellman-Ford equation,

$$d_x(y) = \min_v\{c(x,v) + d_v(y)\}$$

Where the minv is the equation taken for all x neighbors. After traveling from x to v, if we consider the least-cost path from v to y, the path cost will be $c(x,v)+d_v(y)$. The least cost from x to y is the minimum of $c(x,v)+d_v(y)$ taken over all neighbors.

With the Distance Vector Routing algorithm, the node x contains the following routing information:

- For each neighbor v, the cost c(x,v) is the path cost from x to directly attached neighbor, v.
- The distance vector x, i.e., $D_x = [\ D_x(y) : y \text{ in } N\ ]$, containing its cost to all destinations, y, in N.
- The distance vector of each of its neighbors, i.e., $D_v = [\ D_v(y) : y \text{ in } N\ ]$ for each neighbor v of x.

Distance vector routing is an asynchronous algorithm in which node x sends the copy of its distance vector to all its neighbors. When node x receives the new distance vector from one of its neighboring vector, v, it saves the distance vector of v and uses the Bellman-Ford equation to update its own distance vector. The equation is given below:

$$d_x(y) = \min_v\{ c(x,v) + d_v(y)\} \quad \text{for each node y in N}$$

The node x has updated its own distance vector table by using the above equation and sends its updated table to all its neighbors so that they can update their own distance vectors.

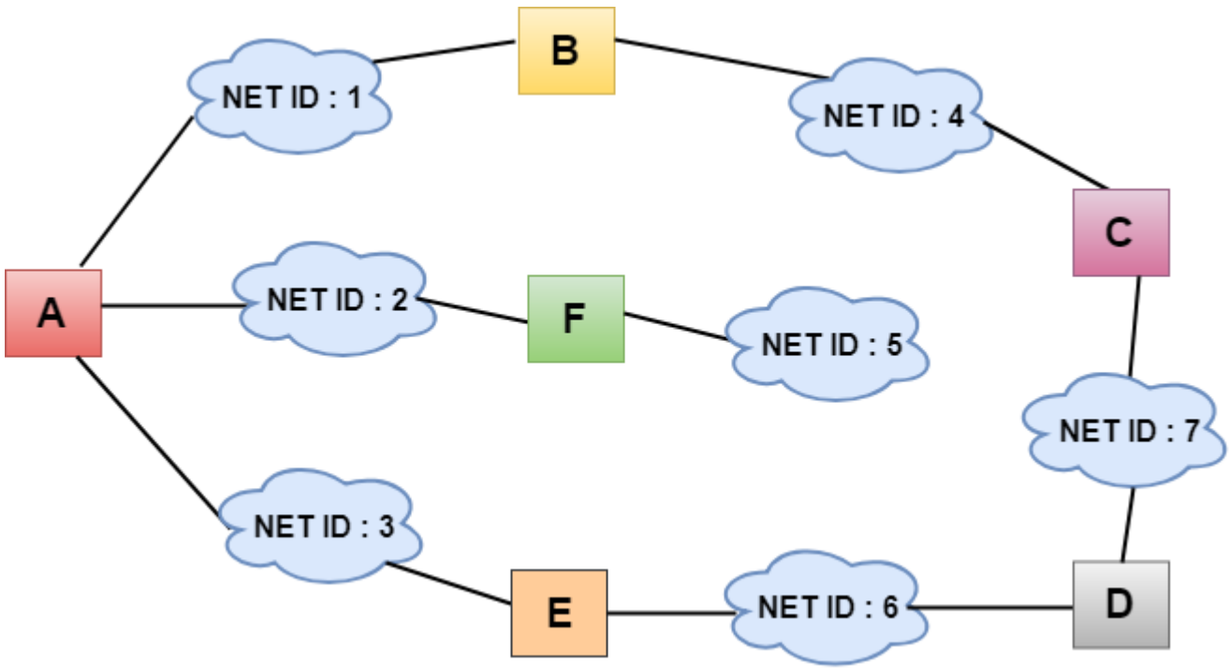
Algorithm

```
At each node x,

Initialization


for all destinations y in N:
Dx(y) = c(x,y)     // If y is not a neighbor then c(x,y) = ∞
for each neighbor w
Dw(y) = ?     for all destination y in N.
for each neighbor w
send distance vector Dx = [ Dx(y)  : y in N ] to w
loop
  wait(until I receive any distance vector from some neighbor w)
  for each y in N:
  Dx(y) = minv{c(x,v)+Dv(y)}
If Dx(y) is changed for any destination y
Send distance vector Dx = [ Dx(y) : y in N ] to all neighbors
forever
```

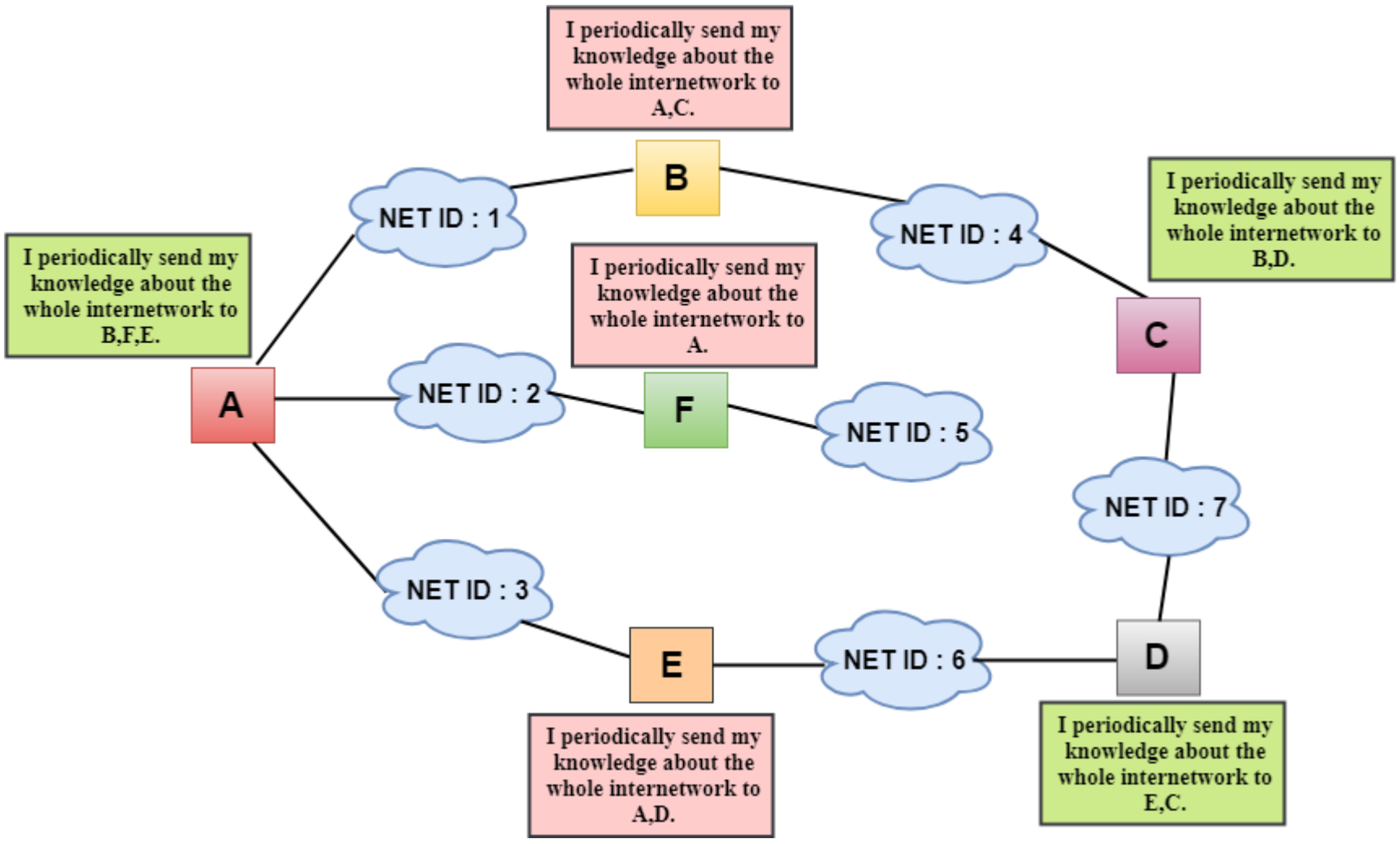
Note: In Distance vector algorithm, node x update its table when it either see any cost change in one directly linked nodes or receives any vector update from some neighbor.

Let's understand through an example:

Sharing Information



- In the above figure, each cloud represents the network, and the number inside the cloud represents the network ID.
- All the LANs are connected by routers, and they are represented in boxes labeled as A, B, C, D, E, F.
- Distance vector routing algorithm simplifies the routing process by assuming the cost of every link is one unit. Therefore, the efficiency of transmission can be measured by the number of links to reach the destination.
- In Distance vector routing, the cost is based on hop count.



In the above figure, we observe that the router sends the knowledge to the immediate neighbors. The neighbors add this knowledge to their own knowledge and sends the updated table to their own neighbors. In this way, routers get its own information plus the new information about the neighbors.

Routing Table

Two process occurs:

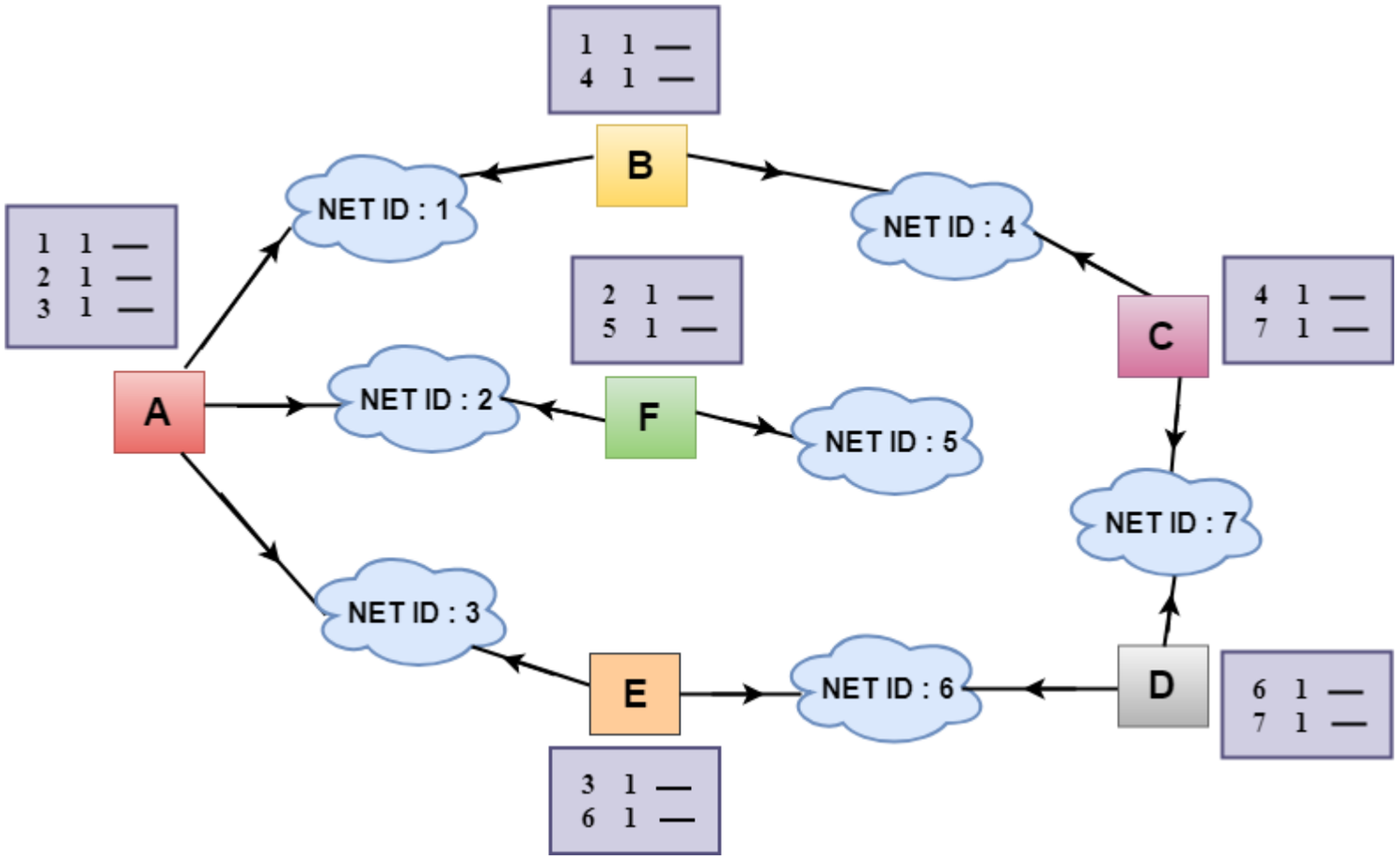
- Creating the Table
- Updating the Table

Creating the Table

Initially, the routing table is created for each router that contains atleast three types of information such as Network ID, the cost and the next hop.

NET ID	Cost	Next Hop
- - - - -	- - - -	- - - - -
- - - - -	- - - -	- - - - -
- - - - -	- - - -	- - - - -
- - - - -	- - - -	- - - - -
- - - - -	- - - -	- - - - -

- **NET ID:** The Network ID defines the final destination of the packet.
- **Cost:** The cost is the number of hops that packet must take to get there.
- **Next hop:** It is the router to which the packet must be delivered.



- In the above figure, the original routing tables are shown of all the routers. In a routing table, the first column represents the network ID, the second column represents the cost of the link, and the third column is empty.
- These routing tables are sent to all the neighbors.

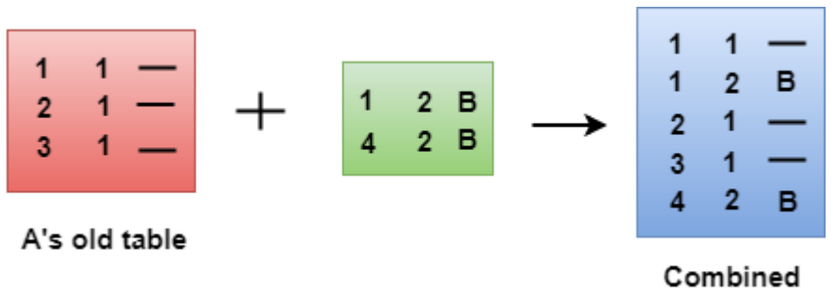
For Example:

1. A sends its routing table to B, F & E.
2. B sends its routing table to A & C.
3. C sends its routing table to B & D.
4. D sends its routing table to E & C.
5. E sends its routing table to A & D.
6. F sends its routing table to A.

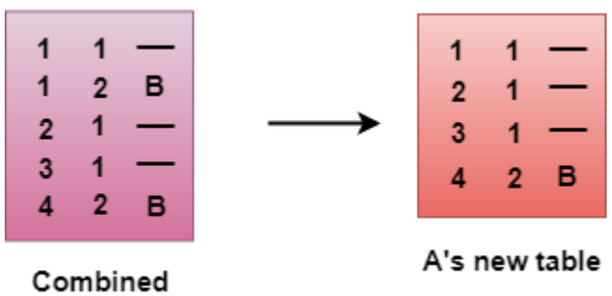
Updating the Table

- When A receives a routing table from B, then it uses its information to update the table.
- The routing table of B shows how the packets can move to the networks 1 and 4.
- The B is a neighbor to the A router, the packets from A to B can reach in one hop. So, 1 is added to all the costs given in the B's table and the sum will be the cost to reach a particular network.

- After adjustment, A then combines this table with its own table to create a combined table.



- The combined table may contain some duplicate data. In the above figure, the combined table of router A contains the duplicate data, so it keeps only those data which has the lowest cost. For example, A can send the data to network 1 in two ways. The first, which uses no next router, so it costs one hop. The second requires two hops (A to B, then B to Network 1). The first option has the lowest cost, therefore it is kept and the second one is dropped.



- The process of creating the routing table continues for all routers. Every router receives the information from the neighbors, and update the routing table.

Final routing tables of all the routers are given below:

Router A

6	2	E
1	1	–
3	1	–
4	2	B
7	3	E
2	1	–
5	2	F

Router B

6	3	E
1	1	–
3	2	A
4	1	–
7	2	C
2	2	A
5	3	A

Router C

6	2	D
1	2	B
3	3	D
4	1	–
7	1	–
2	3	B
5	4	B

Router D

6	1	–
1	3	E
3	2	E
4	2	C
7	1	–
2	3	E
5	4	E

Router E

6	1	–
1	2	A
3	1	–
4	3	A
7	2	D
2	2	A
5	3	A

Router F

6	3	A
1	2	A
3	2	A
4	3	A
7	4	A
2	1	–
5	1	–

Next →← Prev

Link State Routing

Link state routing is a technique in which each router shares the knowledge of its neighborhood with every other router in the internetwork.

The three keys to understand the Link State Routing algorithm:

- **Knowledge about the neighborhood:** Instead of sending its routing table, a router sends the information about its neighborhood only. A router broadcast its identities and cost of the directly attached links to other routers.

- **Flooding:** Each router sends the information to every other router on the internetwork except its neighbors. This process is known as Flooding. Every router that receives the packet sends the copies to all its neighbors. Finally, each and every router receives a copy of the same information.
- **Information sharing:** A router sends the information to every other router only when the change occurs in the information.

Link State Routing has two phases:

Reliable Flooding

- **Initial state:** Each node knows the cost of its neighbors.
- **Final state:** Each node knows the entire graph.

Route Calculation

Each node uses Dijkstra's algorithm on the graph to calculate the optimal routes to all nodes.

- The Link state routing algorithm is also known as Dijkstra's algorithm which is used to find the shortest path from one node to every other node in the network.
- The Dijkstra's algorithm is an iterative, and it has the property that after k^{th} iteration of the algorithm, the least cost paths are well known for k destination nodes.

Let's describe some notations:

- **c(i , j):** Link cost from node i to node j. If i and j nodes are not directly linked, then c(i , j) = ∞.
- **D(v):** It defines the cost of the path from source code to destination v that has the least cost currently.
- **P(v):** It defines the previous node (neighbor of v) along with current least cost path from source to v.
- **N:** It is the total number of nodes available in the network.

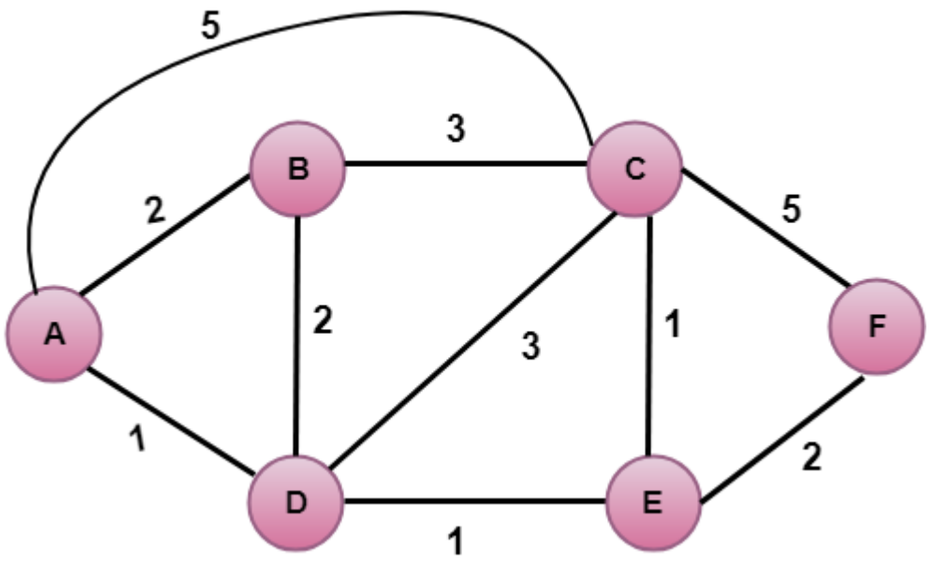
Algorithm

```
Initialization
```

```
N = {A}     // A is a root node.
for all nodes v
if v adjacent to A
then D(v) = c(A,v)
else D(v) = infinity
loop
find w not in N such that D(w) is a minimum.
Add w to N
Update D(v) for all v adjacent to w and not in N:
D(v) = min(D(v) , D(w) + c(w,v))
Until all nodes in N
```

In the above algorithm, an initialization step is followed by the loop. The number of times the loop is executed is equal to the total number of nodes available in the network.

Let's understand through an example:



In the above figure, source vertex is A.

Step 1:

The first step is an initialization step. The currently known least cost path from A to its directly attached neighbors, B, C, D are 2,5,1 respectively. The cost from A to B is set to 2, from A to D is set to 1 and from A to C is set to 5. The cost from A to E and F are set to infinity as they are not directly linked to A.

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)
1	A	2,A	5,A	1,A	∞	∞

Step 2:

In the above table, we observe that vertex D contains the least cost path in step 1. Therefore, it is added in N. Now, we need to determine a least-cost path through D vertex.

a) Calculating shortest path from A to B

1. v = B, w = D
2. D(B) = min(D(B) , D(D) + c(D,B))
3. = min(2, 1+2)>
4. = min(2, 3)
5. The minimum value is 2. Therefore, the currently shortest path from A to B is 2.

b) Calculating shortest path from A to C

1. v = C, w = D
2. D(B) = min(D(C) , D(D) + c(D,C))
3. = min(5, 1+3)
4. = min(5, 4)
5. The minimum value is 4. Therefore, the currently shortest path from A to C is 4.</p>

c) Calculating shortest path from A to E

1. v = E, w = D
2. D(B) = min(D(E) , D(D) + c(D,E))
3. = min(∞, 1+1)
4. = min(∞, 2)
5. The minimum value is 2. Therefore, the currently shortest path from A to E is 2.

Note: The vertex D has no direct link to vertex E. Therefore, the value of D(F) is infinity.

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)

1	A	2,A	5,A	1,A	∞	∞
2	AD	2,A	4,D		2,D	∞

Step 3:

In the above table, we observe that both E and B have the least cost path in step 2. Let's consider the E vertex. Now, we determine the least cost path of remaining vertices through E.

a) Calculating the shortest path from A to B.

1. v = B, w = E
2. D(B) = min(D(B) , D(E) + c(E,B))
3. = min(2 , 2+ ∞)
4. = min(2, ∞)
5. The minimum value is 2. Therefore, the currently shortest path from A to B is 2.

b) Calculating the shortest path from A to C.

1. v = C, w = E
2. D(B) = min(D(C) , D(E) + c(E,C))
3. = min(4 , 2+1)
4. = min(4,3)
5. The minimum value is 3. Therefore, the currently shortest path from A to C is 3.

c) Calculating the shortest path from A to F.

1. v = F, w = E
2. D(B) = min(D(F) , D(E) + c(E,F))
3. = min(∞ , 2+2)
4. = min(∞ ,4)
5. The minimum value is 4. Therefore, the currently shortest path from A to F is 4.

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)
1	A	2,A	5,A	1,A	∞	∞
2	AD	2,A	4,D		2,D	∞
3	ADE	2,A	3,E			4,E

Step 4:

In the above table, we observe that B vertex has the least cost path in step 3. Therefore, it is added in N. Now, we determine the least cost path of remaining vertices through B.

a) Calculating the shortest path from A to C.

1. v = C, w = B
2. D(B) = min(D(C) , D(B) + c(B,C))
3. = min(3 , 2+3)
4. = min(3,5)
5. The minimum value is 3. Therefore, the currently shortest path from A to C is 3.

b) Calculating the shortest path from A to F.

1. v = F, w = B
2. D(B) = min(D(F) , D(B) + c(B,F))
3. = min(4, ∞)
4. = min(4, ∞)
5. The minimum value is 4. Therefore, the currently shortest path from A to F is 4.

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)

1	A	2,A	5,A	1,A	∞	∞
2	AD	2,A	4,D		2,D	∞
3	ADE	2,A	3,E			4,E
4	ADEB		3,E			4,E

Step 5:

In the above table, we observe that C vertex has the least cost path in step 4. Therefore, it is added in N. Now, we determine the least cost path of remaining vertices through C.

a) Calculating the shortest path from A to F.

1. v = F, w = C
2. D(B) = min(D(F) , D(C) + c(C,F))
3. = min(4, 3+5)
4. = min(4,8)
5. The minimum value is 4. Therefore, the currently shortest path from A to F is 4.

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)
1	A	2,A	5,A	1,A	∞	∞
2	AD	2,A	4,D		2,D	∞
3	ADE	2,A	3,E			4,E
4	ADEB		3,E			4,E

5	ADEBC					4,E

Final table:

Step	N	D(B),P(B)	D(C),P(C)	D(D),P(D)	D(E),P(E)	D(F),P(F)
1	A	2,A	5,A	1,A	∞	∞
2	AD	2,A	4,D		2,D	∞
3	ADE	2,A	3,E			4,E
4	ADEB		3,E			4,E
5	ADEBC					4,E
6	ADEBCF					

Disadvantage:

Heavy traffic is created in Line state routing due to Flooding. Flooding can cause an infinite looping, this problem can be solved by using Time-to-leave field

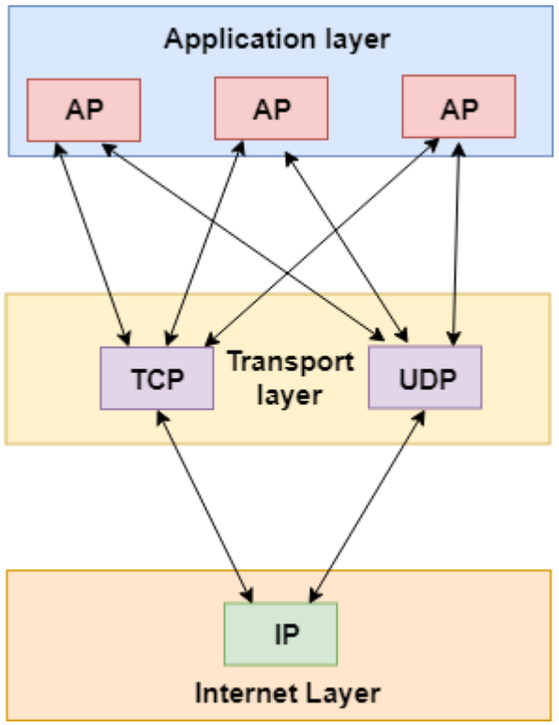
Transport Layer

Next →← Prev

Transport Layer

- The transport layer is a 4th layer from the top.
- The main role of the transport layer is to provide the communication services directly to the application processes running on different hosts.

- The transport layer provides a logical communication between application processes running on different hosts. Although the application processes on different hosts are not physically connected, application processes use the logical communication provided by the transport layer to send the messages to each other.
- The transport layer protocols are implemented in the end systems but not in the network routers.
- A computer network provides more than one protocol to the network applications. For example, TCP and UDP are two transport layer protocols that provide a different set of services to the network layer.
- All transport layer protocols provide multiplexing/demultiplexing service. It also provides other services such as reliable data transfer, bandwidth guarantees, and delay guarantees.
- Each of the applications in the application layer has the ability to send a message by using TCP or UDP. The application communicates by using either of these two protocols. Both TCP and UDP will then communicate with the internet protocol in the internet layer. The applications can read and write to the transport layer. Therefore, we can say that communication is a two-way process.

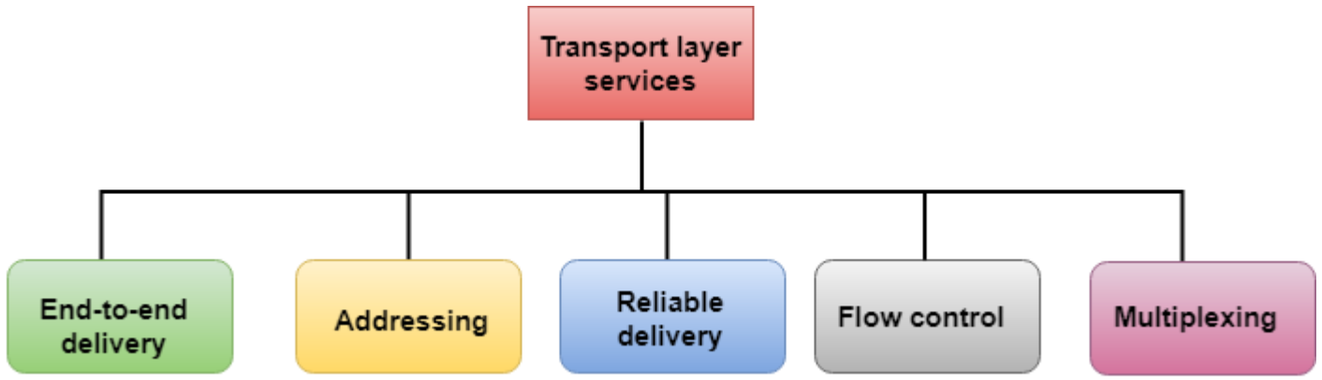


Services provided by the Transport Layer

The services provided by the transport layer are similar to those of the data link layer. The data link layer provides the services within a single network while the transport layer provides the services across an internetwork made up of many networks. The data link layer controls the physical layer while the transport layer controls all the lower layers.

The services provided by the transport layer protocols can be divided into five categories:

- End-to-end delivery
- Addressing
- Reliable delivery
- Flow control
- Multiplexing



End-to-end delivery:

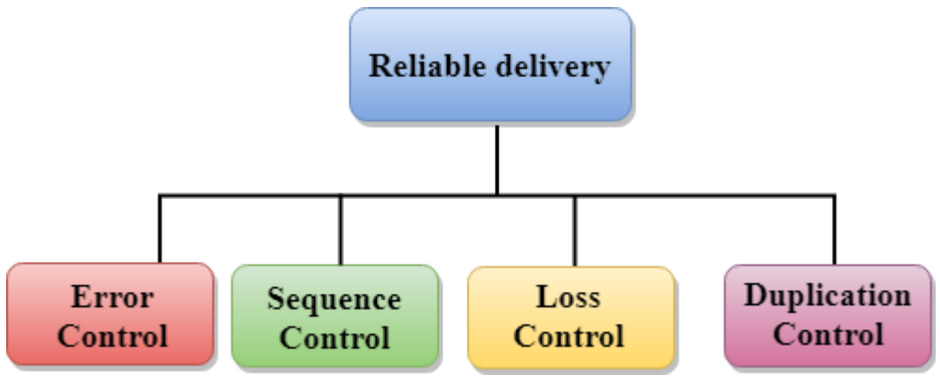
The transport layer transmits the entire message to the destination. Therefore, it ensures the end-to-end delivery of an entire message from a source to the destination.

Reliable delivery:

The transport layer provides reliability services by retransmitting the lost and damaged packets.

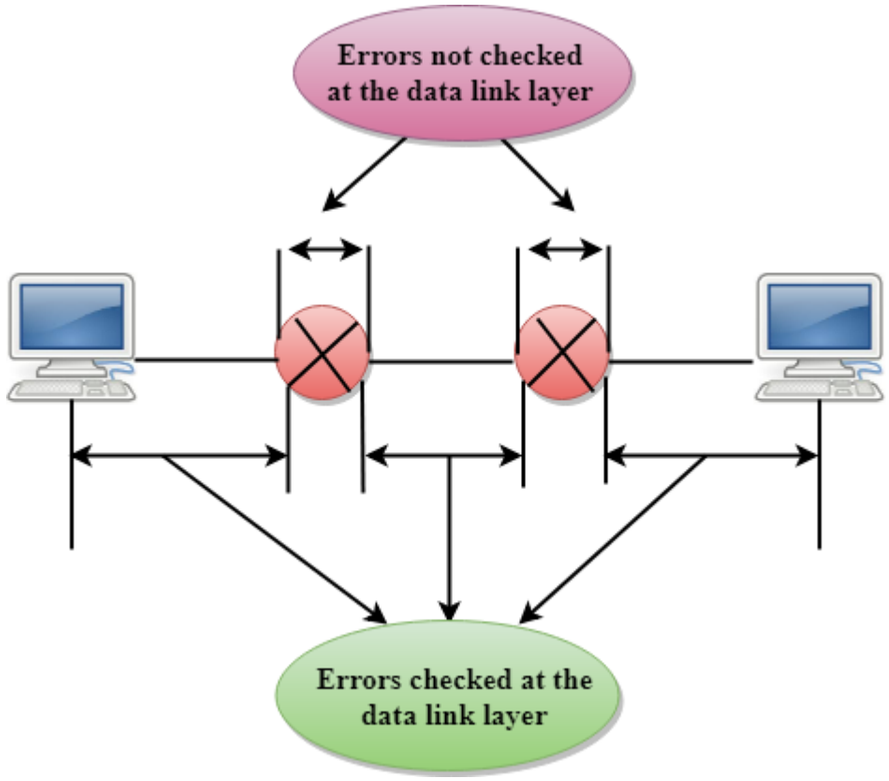
The reliable delivery has four aspects:

- Error control
- Sequence control
- Loss control
- Duplication control



Error Control

- The primary role of reliability is **Error Control**. In reality, no transmission will be 100 percent error-free delivery. Therefore, transport layer protocols are designed to provide error-free transmission.
- The data link layer also provides the error handling mechanism, but it ensures only node-to-node error-free delivery. However, node-to-node reliability does not ensure the end-to-end reliability.
- The data link layer checks for the error between each network. If an error is introduced inside one of the routers, then this error will not be caught by the data link layer. It only detects those errors that have been introduced between the beginning and end of the link. Therefore, the transport layer performs the checking for the errors end-to-end to ensure that the packet has arrived correctly.



Sequence Control

- The second aspect of the reliability is sequence control which is implemented at the transport layer.
- On the sending end, the transport layer is responsible for ensuring that the packets received from the upper layers can be used by the lower layers. On the receiving end, it ensures that the various segments of a transmission can be correctly reassembled.

Loss Control

Loss Control is a third aspect of reliability. The transport layer ensures that all the fragments of a transmission arrive at the destination, not some of them. On the sending end, all the fragments of transmission are given sequence numbers by a transport layer. These sequence numbers allow the receiver?s transport layer to identify the missing segment.

Duplication Control

Duplication Control is the fourth aspect of reliability. The transport layer guarantees that no duplicate data arrive at the destination. Sequence numbers are used to identify the lost packets; similarly, it allows the receiver to identify and discard duplicate segments.

Flow Control

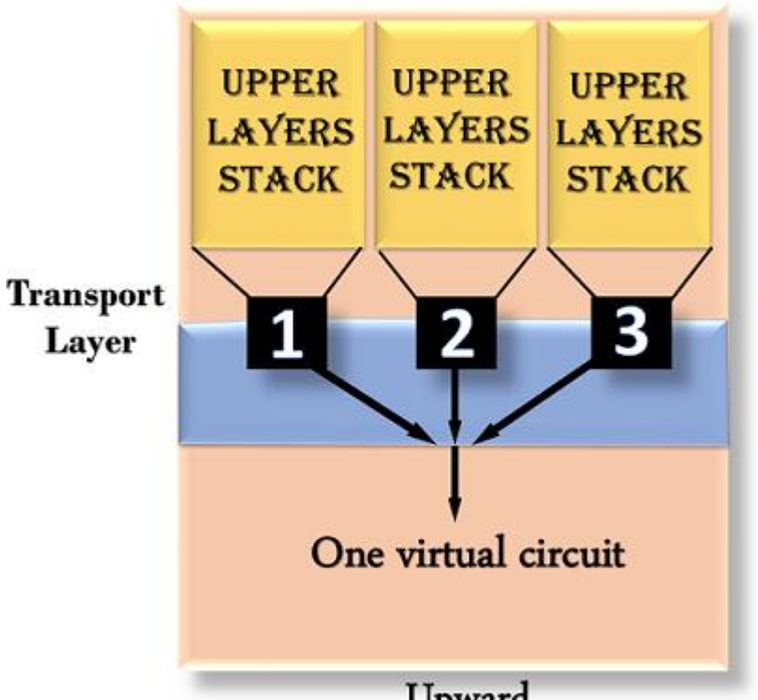
Flow control is used to prevent the sender from overwhelming the receiver. If the receiver is overloaded with too much data, then the receiver discards the packets and asking for the retransmission of packets. This increases network congestion and thus, reducing the system performance. The transport layer is responsible for flow control. It uses the sliding window protocol that makes the data transmission more efficient as well as it controls the flow of data so that the receiver does not become overwhelmed. Sliding window protocol is byte oriented rather than frame oriented.

Multiplexing

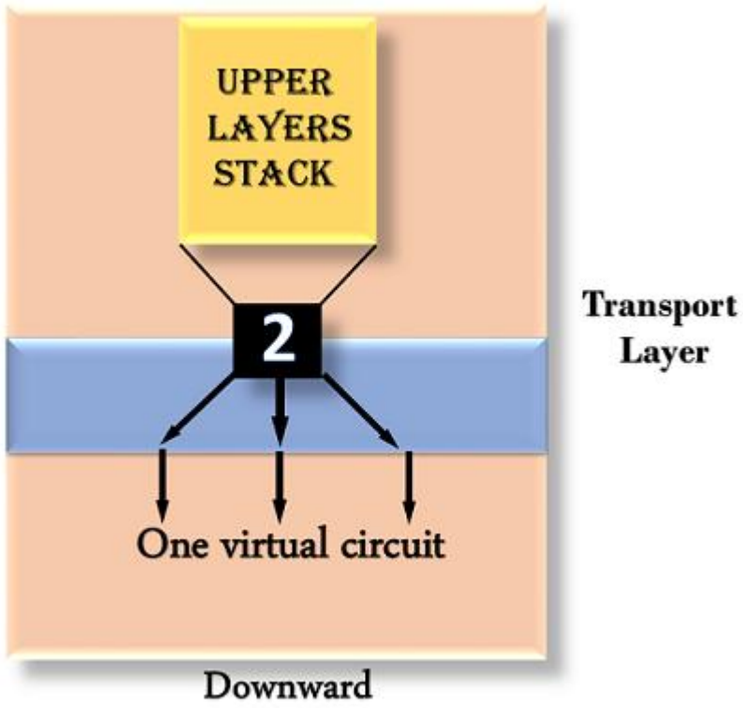
The transport layer uses the multiplexing to improve transmission efficiency.

Multiplexing can occur in two ways:

- **Upward multiplexing:** Upward multiplexing means multiple transport layer connections use the same network connection. To make more cost-effective, the transport layer sends several transmissions bound for the same destination along the same path; this is achieved through upward multiplexing.



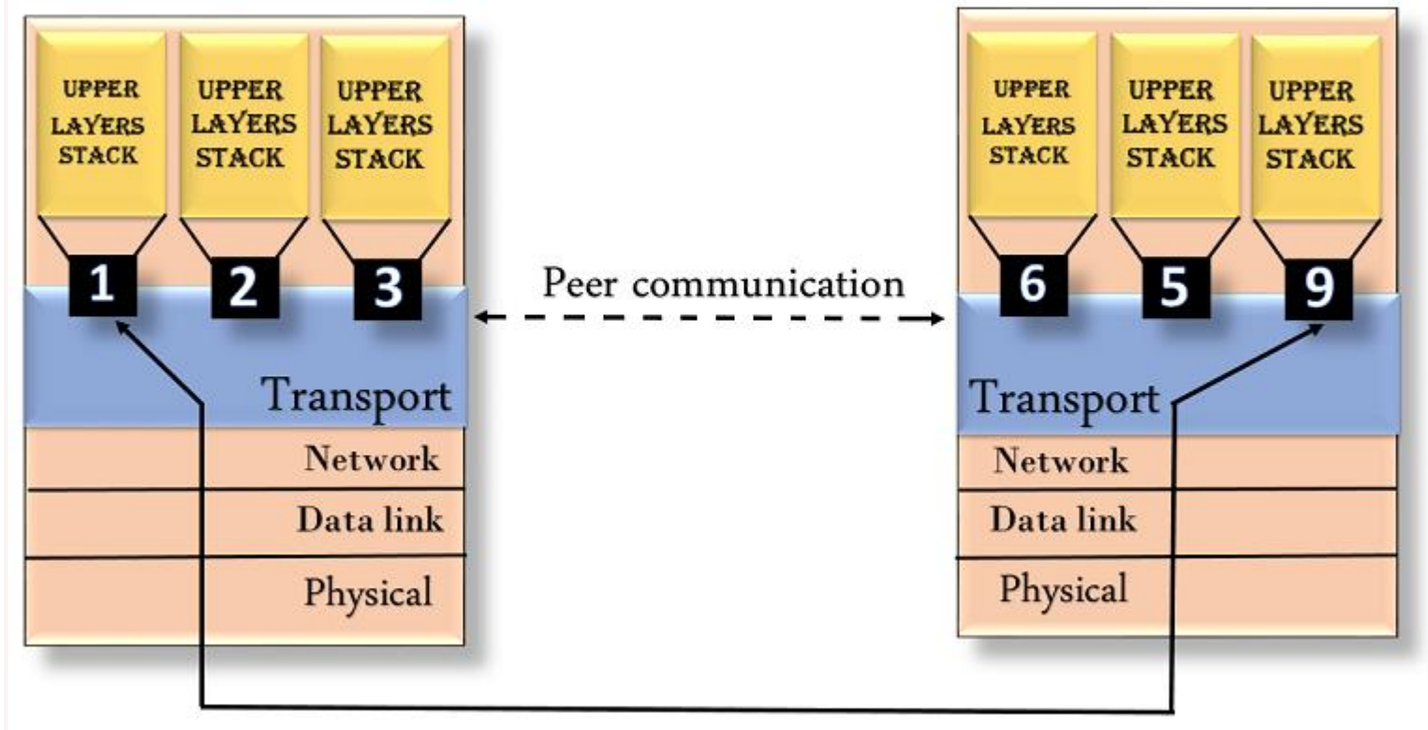
- **Downward multiplexing:** Downward multiplexing means one transport layer connection uses the multiple network connections. Downward multiplexing allows the transport layer to split a connection among several paths to improve the throughput. This type of multiplexing is used when networks have a low or slow capacity.



Transport
Layer

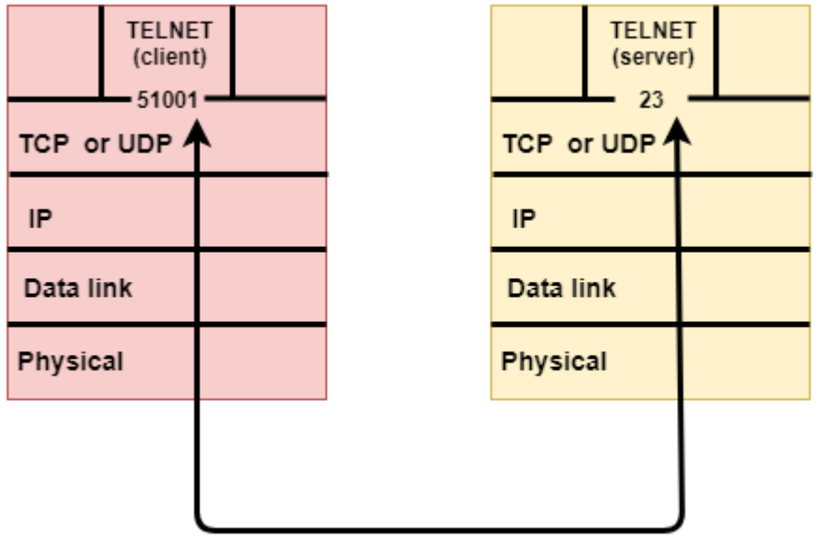
Addressing

- According to the layered model, the transport layer interacts with the functions of the session layer. Many protocols combine session, presentation, and application layer protocols into a single layer known as the application layer. In these cases, delivery to the session layer means the delivery to the application layer. Data generated by an application on one machine must be transmitted to the correct application on another machine. In this case, addressing is provided by the transport layer.
- The transport layer provides the user address which is specified as a station or port. The port variable represents a particular TS user of a specified station known as a Transport Service access point (TSAP). Each station has only one transport entity.
- The transport layer protocols need to know which upper-layer protocols are communicating.



Transport Layer protocols

- The transport layer is represented by two protocols: TCP and UDP.
- The IP protocol in the network layer delivers a datagram from a source host to the destination host.
- Nowadays, the operating system supports multiuser and multiprocessing environments, an executing program is called a process. When a host sends a message to other host means that source process is sending a process to a destination process. The transport layer protocols define some connections to individual ports known as protocol ports.
- An IP protocol is a host-to-host protocol used to deliver a packet from source host to the destination host while transport layer protocols are port-to-port protocols that work on the top of the IP protocols to deliver the packet from the originating port to the IP services, and from IP services to the destination port.
- Each port is defined by a positive integer address, and it is of 16 bits.



UDP

- UDP stands for **User Datagram Protocol**.
- UDP is a simple protocol and it provides nonsequenced transport functionality.
- UDP is a connectionless protocol.
- This type of protocol is used when reliability and security are less important than speed and size.
- UDP is an end-to-end transport level protocol that adds transport-level addresses, checksum error control, and length information to the data from the upper layer.
- The packet produced by the UDP protocol is known as a user datagram.

User Datagram Format

The user datagram has a 16-byte header which is shown below:

Source port address 16 bits	Destination port address 16 bits
Total Length 16 bits	Checksum 16 bits
Data	

Where,

- **Source port address:** It defines the address of the application process that has delivered a message. The source port address is of 16 bits address.
- **Destination port address:** It defines the address of the application process that will receive the message. The destination port address is of a 16-bit address.
- **Total length:** It defines the total length of the user datagram in bytes. It is a 16-bit field.
- **Checksum:** The checksum is a 16-bit field which is used in error detection.

Disadvantages of UDP protocol

- UDP provides basic functions needed for the end-to-end delivery of a transmission.
- It does not provide any sequencing or reordering functions and does not specify the damaged packet when reporting an error.
- UDP can discover that an error has occurred, but it does not specify which packet has been lost as it does not contain an ID or sequencing number of a particular data segment.

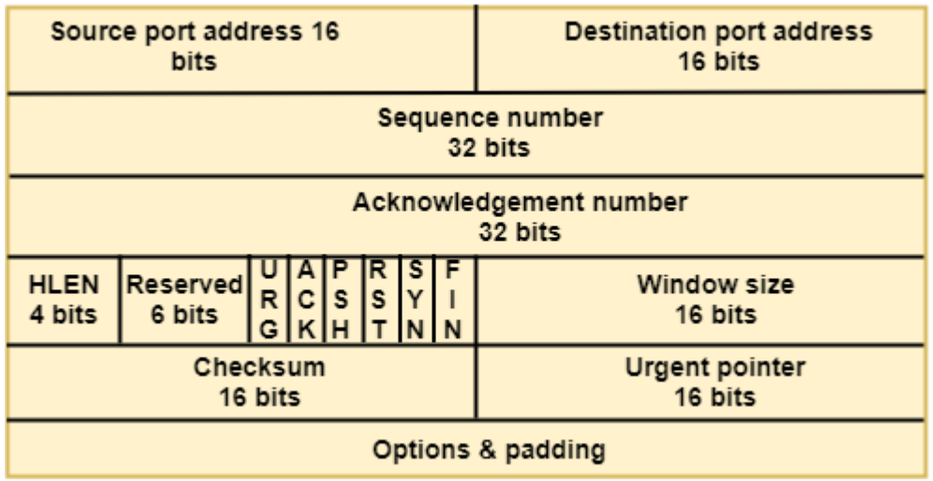
TCP

- TCP stands for Transmission Control Protocol.
- It provides full transport layer services to applications.
- It is a connection-oriented protocol means the connection established between both the ends of the transmission. For creating the connection, TCP generates a virtual circuit between sender and receiver for the duration of a transmission.

Features Of TCP protocol

- **Stream data transfer:** TCP protocol transfers the data in the form of contiguous stream of bytes. TCP group the bytes in the form of TCP segments and then passed it to the IP layer for transmission to the destination. TCP itself segments the data and forward to the IP.
- **Reliability:** TCP assigns a sequence number to each byte transmitted and expects a positive acknowledgement from the receiving TCP. If ACK is not received within a timeout interval, then the data is retransmitted to the destination.
  The receiving TCP uses the sequence number to reassemble the segments if they arrive out of order or to eliminate the duplicate segments.
- **Flow Control:** When receiving TCP sends an acknowledgement back to the sender indicating the number the bytes it can receive without overflowing its internal buffer. The number of bytes is sent in ACK in the form of the highest sequence number that it can receive without any problem. This mechanism is also referred to as a window mechanism.
- **Multiplexing:** Multiplexing is a process of accepting the data from different applications and forwarding to the different applications on different computers. At the receiving end, the data is forwarded to the correct application. This process is known as demultiplexing. TCP transmits the packet to the correct application by using the logical channels known as ports.
- **Logical Connections:** The combination of sockets, sequence numbers, and window sizes, is called a logical connection. Each connection is identified by the pair of sockets used by sending and receiving processes.
- **Full Duplex:** TCP provides Full Duplex service, i.e., the data flow in both the directions at the same time. To achieve Full Duplex service, each TCP should have sending and receiving buffers so that the segments can flow in both the directions. TCP is a connection-oriented protocol. Suppose the process A wants to send and receive the data from process B. The following steps occur:
  - Establish a connection between two TCPs.
  - Data is exchanged in both the directions.
  - The Connection is terminated.

TCP Segment Format



Where,

- **Source port address:** It is used to define the address of the application program in a source computer. It is a 16-bit field.
- **Destination port address:** It is used to define the address of the application program in a destination computer. It is a 16-bit field.
- **Sequence number:** A stream of data is divided into two or more TCP segments. The 32-bit sequence number field represents the position of the data in an original data stream.
- **Acknowledgement number:** A 32-field acknowledgement number acknowledge the data from other communicating devices. If ACK field is set to 1, then it specifies the sequence number that the receiver is expecting to receive.
- **Header Length (HLEN):** It specifies the size of the TCP header in 32-bit words. The minimum size of the header is 5 words, and the maximum size of the header is 15 words. Therefore, the maximum size of the TCP header is 60 bytes, and the minimum size of the TCP header is 20 bytes.
- **Reserved:** It is a six-bit field which is reserved for future use.
- **Control bits:** Each bit of a control field functions individually and independently. A control bit defines the use of a segment or serves as a validity check for other fields.

There are total six types of flags in control field:

- **URG:** The URG field indicates that the data in a segment is urgent.
- **ACK:** When ACK field is set, then it validates the acknowledgement number.
- **PSH:** The PSH field is used to inform the sender that higher throughput is needed so if possible, data must be pushed with higher throughput.
- **RST:** The reset bit is used to reset the TCP connection when there is any confusion occurs in the sequence numbers.

- **SYN:** The SYN field is used to synchronize the sequence numbers in three types of segments: connection request, connection confirmation (with the ACK bit set), and confirmation acknowledgement.
- **FIN:** The FIN field is used to inform the receiving TCP module that the sender has finished sending data. It is used in connection termination in three types of segments: termination request, termination confirmation, and acknowledgement of termination confirmation.
    - **Window Size:** The window is a 16-bit field that defines the size of the window.
    - **Checksum:** The checksum is a 16-bit field used in error detection.
    - **Urgent pointer:** If URG flag is set to 1, then this 16-bit field is an offset from the sequence number indicating that it is a last urgent data byte.
    - **Options and padding:** It defines the optional fields that convey the additional information to the receiver.

Differences b/w TCP & UDP

Basis for Comparison	TCP	UDP
Definition	TCP establishes a virtual circuit before transmitting the data.	UDP transmits the data directly to the destination computer without verifying whether the receiver is ready to receive or not.
Connection Type	It is a Connection-Oriented protocol	It is a Connectionless protocol
Speed	slow	high
Reliability	It is a reliable protocol.	It is an unreliable protocol.
Header size	20 bytes	8 bytes

acknowledgement	It waits for the acknowledgement of data and has the ability to resend the lost packets.	It neither takes the acknowledgement, nor it retransmits the damaged frame.

Application Layer

Application Layer

The application layer in the OSI model is the closest layer to the end user which means that the application layer and end user can interact directly with the software application. The application layer programs are based on client and servers.

The Application layer includes the following functions:

- **Identifying communication partners:** The application layer identifies the availability of communication partners for an application with data to transmit.
- **Determining resource availability:** The application layer determines whether sufficient network resources are available for the requested communication.
- **Synchronizing communication:** All the communications occur between the applications requires cooperation which is managed by an application layer.

Services of Application Layers

- **Network Virtual terminal:** An application layer allows a user to log on to a remote host. To do so, the application creates a software emulation of a terminal at the remote host. The user's computer talks to the software terminal, which in turn, talks to the host. The remote host thinks that it is communicating with one of its own terminals, so it allows the user to log on.
- **File Transfer, Access, and Management (FTAM):** An application allows a user to access files in a remote computer, to retrieve files from a computer and to manage files in a remote computer. FTAM defines a hierarchical virtual file in terms of file structure, file attributes and the kind of operations performed on the files and their attributes.
- **Addressing:** To obtain communication between client and server, there is a need for addressing. When a client made a request to the server, the request contains the server address and its own address. The server response to the client request, the request contains the destination address, i.e., client address. To achieve this kind of addressing, DNS is used.
- **Mail Services:** An application layer provides Email forwarding and storage.
- **Directory Services:** An application contains a distributed database that provides access for global information about various objects and services.

Authentication: It authenticates the sender or receiver's message or both.

Network Application Architecture

Application architecture is different from the network architecture. The network architecture is fixed and provides a set of services to applications. The application architecture, on the other hand, is designed by the application developer and defines how the application should be structured over the various end systems.

Application architecture is of two types:

- **Client-server architecture:** An application program running on the local machine sends a request to another application program is known as a client, and a program that serves a request is known as a server. For example, when a web server receives a request from the client host, it responds to the request to the client host.

Characteristics Of Client-server architecture:

- In Client-server architecture, clients do not directly communicate with each other. For example, in a web application, two browsers do not directly communicate with each other.
- A server is fixed, well-known address known as IP address because the server is always on while the client can always contact the server by sending a packet to the sender's IP address.

Disadvantage Of Client-server architecture:

It is a single-server based architecture which is incapable of holding all the requests from the clients. For example, a social networking site can become overwhelmed when there is only one server exists.

- **P2P (peer-to-peer) architecture:** It has no dedicated server in a data center. The peers are the computers which are not owned by the service provider. Most of the peers reside in the homes, offices, schools, and universities. The peers communicate with each other without passing the information through a dedicated server, this architecture is known as peer-to-peer architecture. The applications based on P2P architecture includes file sharing and internet telephony.

Features of P2P architecture

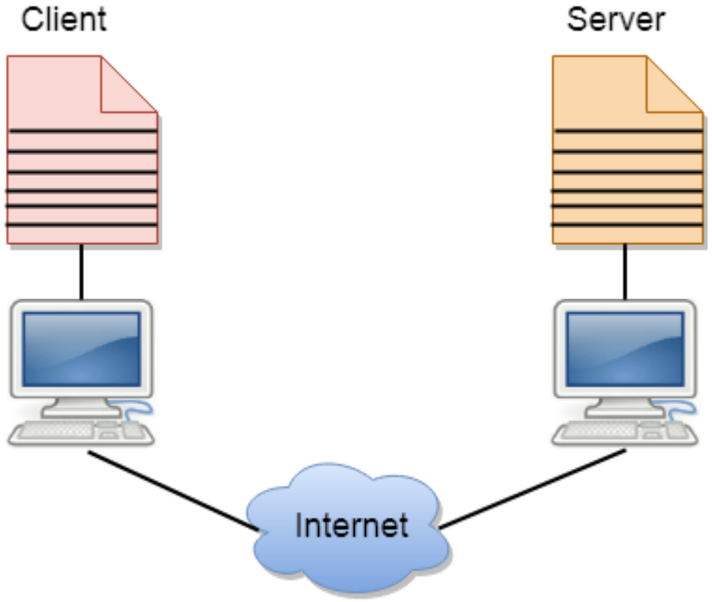
- **Self scalability:** In a file sharing system, although each peer generates a workload by requesting the files, each peer also adds a service capacity by distributing the files to the peer.
- **Cost-effective:** It is cost-effective as it does not require significant server infrastructure and server bandwidth.

Client and Server processes

- A network application consists of a pair of processes that send the messages to each other over a network.
- In P2P file-sharing system, a file is transferred from a process in one peer to a process in another peer. We label one of the two processes as the client and another process as the server.
- With P2P file sharing, the peer which is downloading the file is known as a client, and the peer which is uploading the file is known as a server. However, we have observed in some applications such as P2P file sharing; a process can be both as a client and server. Therefore, we can say that a process can both download and upload the files.

Client and Server model

- A client and server networking model is a model in which computers such as servers provide the network services to the other computers such as clients to perform a user based tasks. This model is known as client-server networking model.
- The application programs using the client-server model should follow the given below strategies:



- An application program is known as a client program, running on the local machine that requests for a service from an application program known as a server program, running on the remote machine.

- A client program runs only when it requests for a service from the server while the server program runs all time as it does not know when its service is required.
- A server provides a service for many clients not just for a single client. Therefore, we can say that client-server follows the many-to-one relationship. Many clients can use the service of one server.
- Services are required frequently, and many users have a specific client-server application program. For example, the client-server application program allows the user to access the files, send e-mail, and so on. If the services are more customized, then we should have one generic application program that allows the user to access the services available on the remote computer.

Client

A client is a program that runs on the local machine requesting service from the server. A client program is a finite program means that the service started by the user and terminates when the service is completed.

Server

A server is a program that runs on the remote machine providing services to the clients. When the client requests for a service, then the server opens the door for the incoming requests, but it never initiates the service.

A server program is an infinite program means that when it starts, it runs infinitely unless the problem arises. The server waits for the incoming requests from the clients. When the request arrives at the server, then it responds to the request.

Advantages of Client-server networks:

- **Centralized:** Centralized back-up is possible in client-server networks, i.e., all the data is stored in a server.
- **Security:** These networks are more secure as all the shared resources are centrally administered.
- **Performance:** The use of the dedicated server increases the speed of sharing resources. This increases the performance of the overall system.
- **Scalability:** We can increase the number of clients and servers separately, i.e., the new element can be added, or we can add a new node in a network at any time.

Disadvantages of Client-Server network:

- **Traffic Congestion** is a big problem in Client/Server networks. When a large number of clients send requests to the same server may cause the problem of Traffic congestion.
- It does not have a robustness of a network, i.e., when the server is down, then the client requests cannot be met.

- A client/server network is very decisive. Sometimes, regular computer hardware does not serve a certain number of clients. In such situations, specific hardware is required at the server side to complete the work.
- Sometimes the resources exist in the server but may not exist in the client. For example, If the application is web, then we cannot take the print out directly on printers without taking out the print view window on the web.

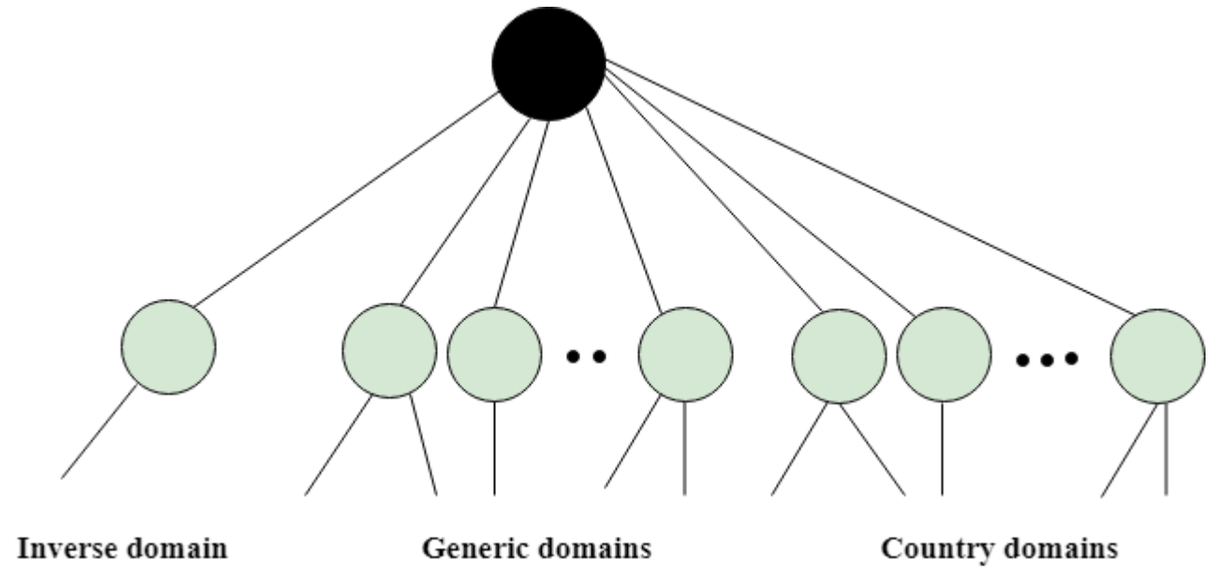
Application Protocols

DNS

An application layer protocol defines how the application processes running on different systems, pass the messages to each other.

- DNS stands for Domain Name System.
- DNS is a directory service that provides a mapping between the name of a host on the network and its numerical address.
- DNS is required for the functioning of the internet.
- Each node in a tree has a domain name, and a full domain name is a sequence of symbols specified by dots.
- DNS is a service that translates the domain name into IP addresses. This allows the users of networks to utilize user-friendly names when looking for other hosts instead of remembering the IP addresses.
- For example, suppose the FTP site at EduSoft had an IP address of 132.147.165.50, most people would reach this site by specifying ftp.EduSoft.com. Therefore, the domain name is more reliable than IP address.

DNS is a TCP/IP protocol used on different platforms. The domain name space is divided into three different sections: generic domains, country domains, and inverse domain.

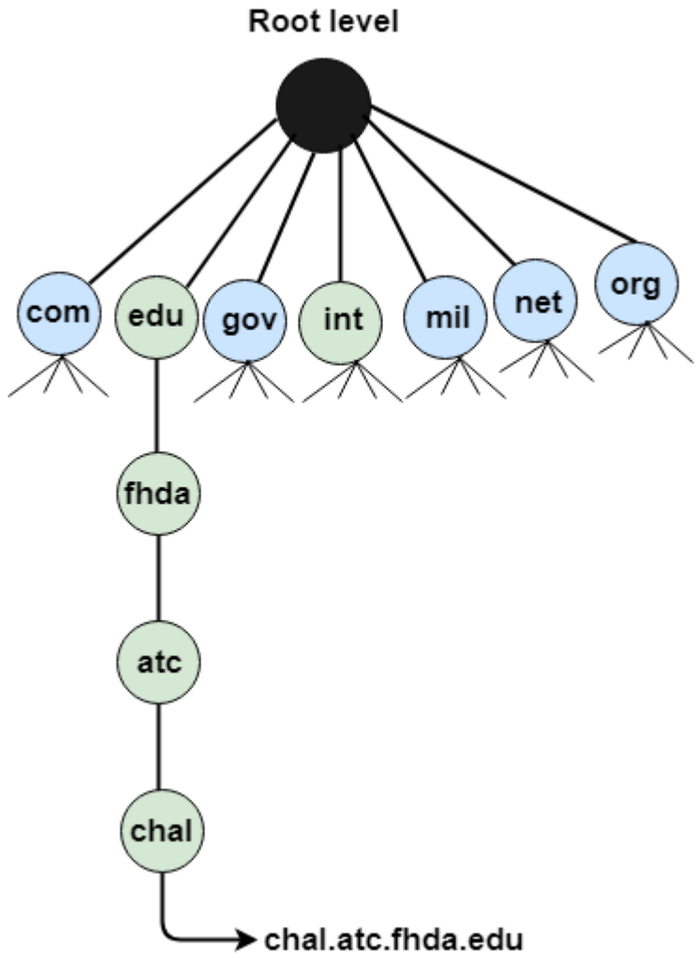


Generic Domains

- It defines the registered hosts according to their generic behavior.
- Each node in a tree defines the domain name, which is an index to the DNS database.
- It uses three-character labels, and these labels describe the organization type.

Label	Description
aero	Airlines and aerospace companies
biz	Businesses or firms
com	Commercial Organizations
coop	Cooperative business Organizations

edu	Educational institutions
gov	Government institutions
info	Information service providers
int	International Organizations
mil	Military groups
museum	Museum & other nonprofit organizations
name	Personal names
net	Network Support centers
org	Nonprofit Organizations
pro	Professional individual Organizations



Country Domain

The format of country domain is same as a generic domain, but it uses two-character country abbreviations (e.g., us for the United States) in place of three character organizational abbreviations.

Inverse Domain

The inverse domain is used for mapping an address to a name. When the server has received a request from the client, and the server contains the files of only authorized clients. To determine whether the client is on the authorized list or not, it sends a query to the DNS server and ask for mapping an address to the name.

Working of DNS

- DNS is a client/server network communication protocol. DNS clients send requests to the. server while DNS servers send responses to the client.
- Client requests contain a name which is converted into an IP address known as a forward DNS lookups while requests containing an IP address which is converted into a name known as reverse DNS lookups.
- DNS implements a distributed database to store the name of all the hosts available on the internet.
- If a client like a web browser sends a request containing a hostname, then a piece of software such as **DNS resolver** sends a request to the DNS server to obtain the IP address of a hostname. If DNS server does not contain the IP address associated with a hostname, then it forwards the request to another DNS server. If IP address has arrived at the resolver, which in turn completes the request over the internet protocol.

FTP

- FTP stands for File transfer protocol.
- FTP is a standard internet protocol provided by TCP/IP used for transmitting the files from one host to another.
- It is mainly used for transferring the web page files from their creator to the computer that acts as a server for other computers on the internet.
- It is also used for downloading the files to computer from other servers.

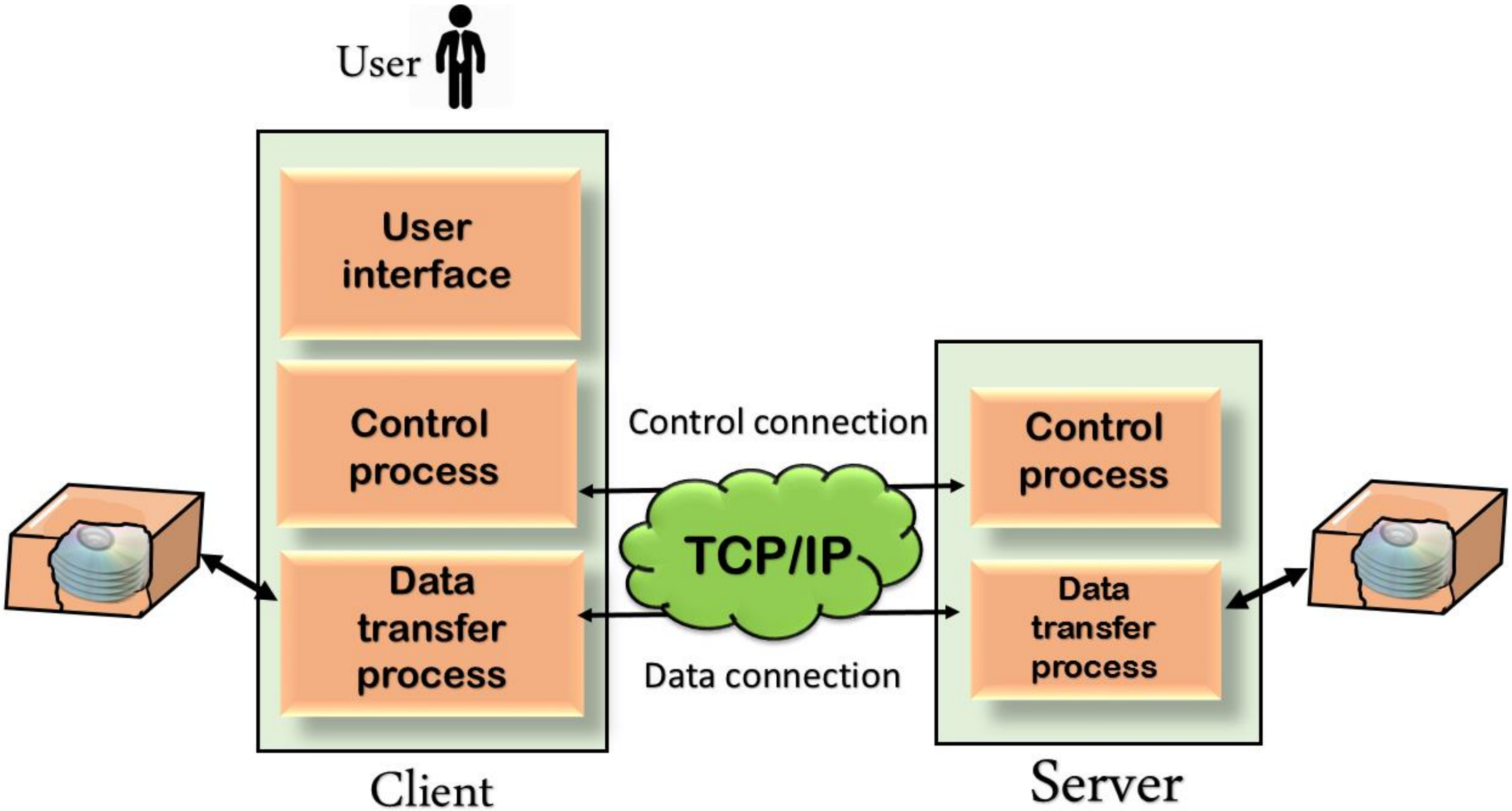
Objectives of FTP

- It provides the sharing of files.
- It is used to encourage the use of remote computers.
- It transfers the data more reliably and efficiently.

Why FTP?

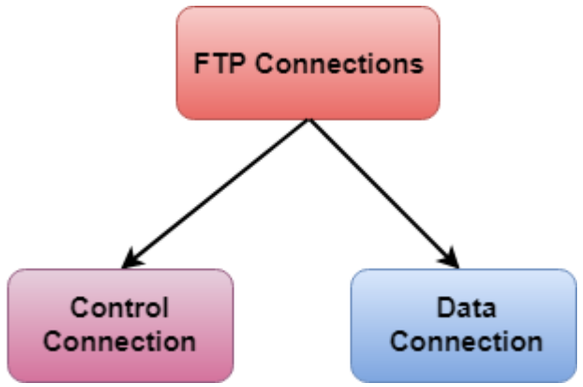
Although transferring files from one system to another is very simple and straightforward, but sometimes it can cause problems. For example, two systems may have different file conventions. Two systems may have different ways to represent text and data. Two systems may have different directory structures. FTP protocol overcomes these problems by establishing two connections between hosts. One connection is used for data transfer, and another connection is used for the control connection.

Mechanism of FTP



The above figure shows the basic model of the FTP. The FTP client has three components: the user interface, control process, and data transfer process. The server has two components: the server control process and the server data transfer process.

There are two types of connections in FTP:



- **Control Connection:** The control connection uses very simple rules for communication. Through control connection, we can transfer a line of command or line of response at a time. The control connection is made between the control processes. The control connection remains connected during the entire interactive FTP session.
- **Data Connection:** The Data Connection uses very complex rules as data types may vary. The data connection is made between data transfer processes. The data connection opens when a command comes for transferring the files and closes when the file is transferred.

FTP Clients

- FTP client is a program that implements a file transfer protocol which allows you to transfer files between two hosts on the internet.
- It allows a user to connect to a remote host and upload or download the files.
- It has a set of commands that we can use to connect to a host, transfer the files between you and your host and close the connection.
- The FTP program is also available as a built-in component in a Web browser. This GUI based FTP client makes the file transfer very easy and also does not require to remember the FTP commands.

Advantages of FTP:

- **Speed:** One of the biggest advantages of FTP is speed. The FTP is one of the fastest way to transfer the files from one computer to another computer.
- **Efficient:** It is more efficient as we do not need to complete all the operations to get the entire file.
- **Security:** To access the FTP server, we need to login with the username and password. Therefore, we can say that FTP is more secure.
- **Back & forth movement:** FTP allows us to transfer the files back and forth. Suppose you are a manager of the company, you send some information to all the employees, and they all send information back on the same server.

Disadvantages of FTP:

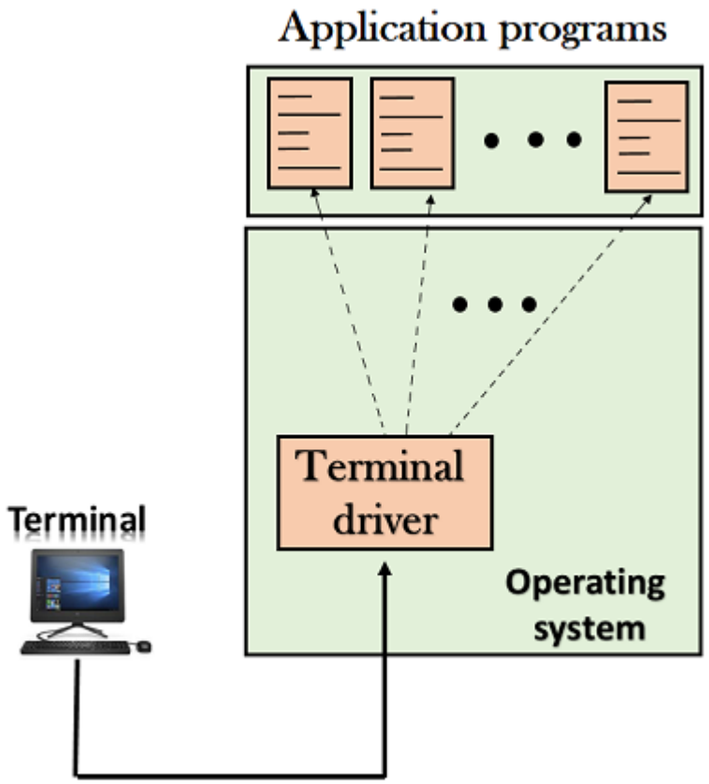
- The standard requirement of the industry is that all the FTP transmissions should be encrypted. However, not all the FTP providers are equal and not all the providers offer encryption. So, we will have to look out for the FTP providers that provides encryption.
- FTP serves two operations, i.e., to send and receive large files on a network. However, the size limit of the file is 2GB that can be sent. It also doesn't allow you to run simultaneous transfers to multiple receivers.
- Passwords and file contents are sent in clear text that allows unwanted eavesdropping. So, it is quite possible that attackers can carry out the brute force attack by trying to guess the FTP password.
- It is not compatible with every system.

Telnet

- The main task of the internet is to provide services to users. For example, users want to run different application programs at the remote site and transfers a result to the local site. This requires a client-server program such as FTP, SMTP. But this would not allow us to create a specific program for each demand.
- The better solution is to provide a general client-server program that lets the user access any application program on a remote computer. Therefore, a program that allows a user to log on to a remote computer. A popular client-server program Telnet is used to meet such demands. Telnet is an abbreviation for **Terminal Network**.
- Telnet provides a connection to the remote computer in such a way that a local terminal appears to be at the remote side.

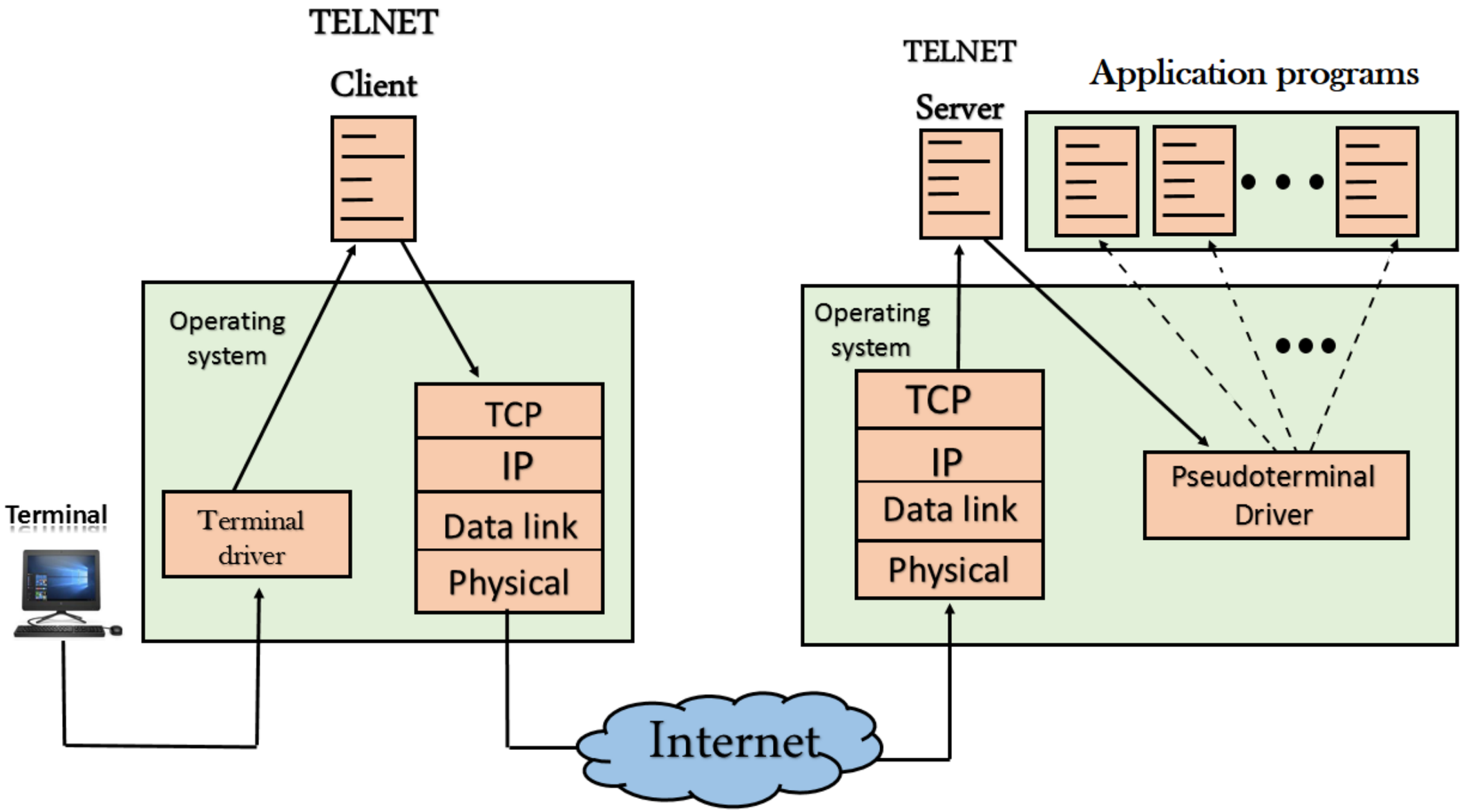
There are two types of login:

Local Login



- When a user logs into a local computer, then it is known as local login.
- When the workstation running terminal emulator, the keystrokes entered by the user are accepted by the terminal driver. The terminal driver then passes these characters to the operating system which in turn, invokes the desired application program.
- However, the operating system has special meaning to special characters. For example, in UNIX some combination of characters have special meanings such as control character with "z" means suspend. Such situations do not create any problem as the terminal driver knows the meaning of such characters. But, it can cause the problems in remote login.

Remote login



- When the user wants to access an application program on a remote computer, then the user must perform remote login.

How remote login occurs

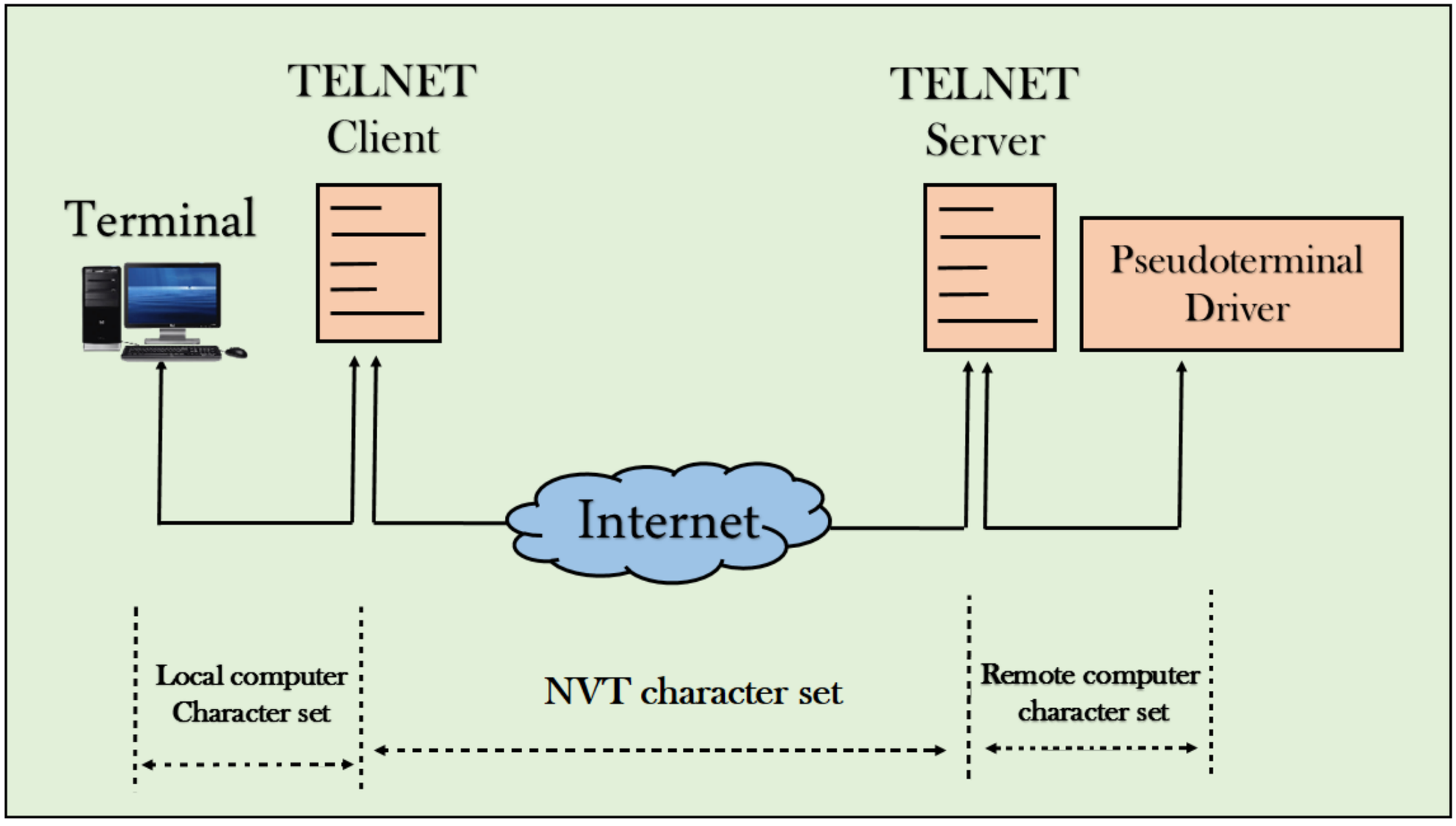
At the local site

The user sends the keystrokes to the terminal driver, the characters are then sent to the TELNET client. The TELNET client which in turn, transforms the characters to a universal character set known as network virtual terminal characters and delivers them to the local TCP/IP stack

At the remote site

The commands in NVT forms are transmitted to the TCP/IP at the remote machine. Here, the characters are delivered to the operating system and then pass to the TELNET server. The TELNET server transforms the characters which can be understandable by a remote computer. However, the characters cannot be directly passed to the operating system as a remote operating system does not receive the characters from the TELNET server. Therefore it requires some piece of software that can accept the characters from the TELNET server. The operating system then passes these characters to the appropriate application program.

Network Virtual Terminal (NVT)



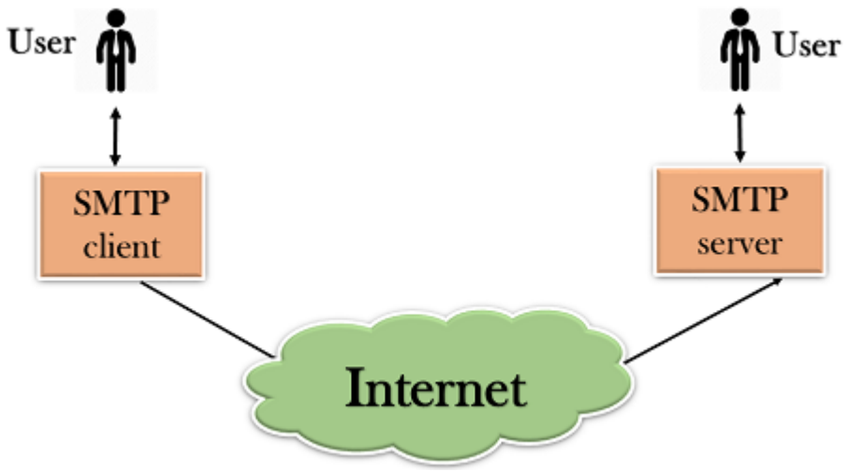
- The network virtual terminal is an interface that defines how data and commands are sent across the network.
- In today's world, systems are heterogeneous. For example, the operating system accepts a special combination of characters such as end-of-file token running a DOS operating system *ctrl+z* while the token running a UNIX operating system is *ctrl+d*.

- TELNET solves this issue by defining a universal interface known as network virtual interface.
- The TELNET client translates the characters that come from the local terminal into NVT form and then delivers them to the network. The Telnet server then translates the data from NVT form into a form which can be understandable by a remote computer.

SMTP

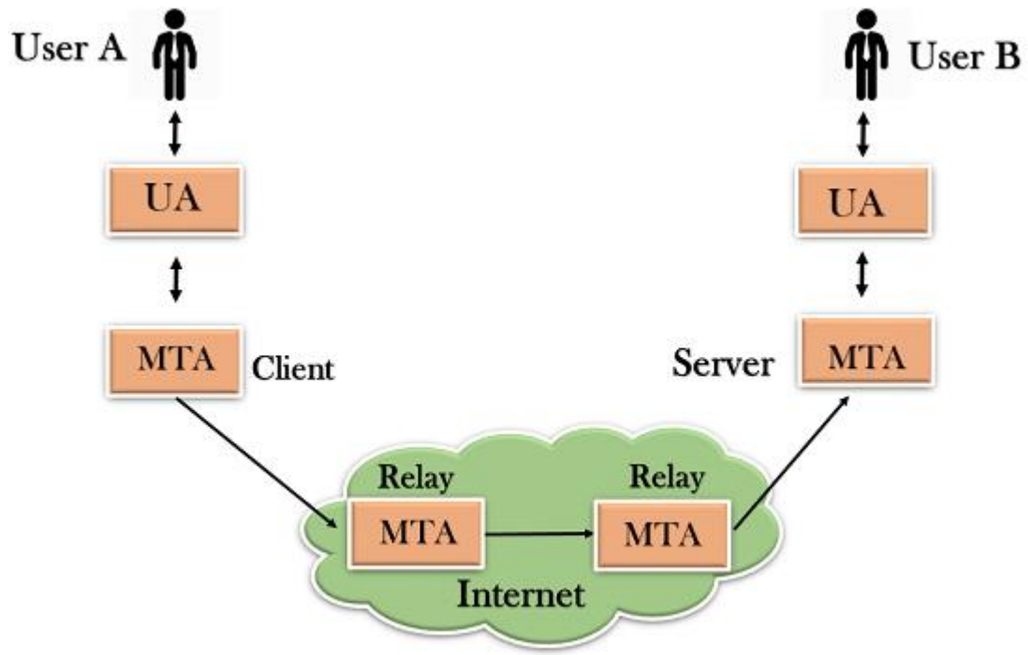
- SMTP stands for Simple Mail Transfer Protocol.
- SMTP is a set of communication guidelines that allow software to transmit an electronic mail over the internet is called **Simple Mail Transfer Protocol**.
- It is a program used for sending messages to other computer users based on e-mail addresses.
- It provides a mail exchange between users on the same or different computers, and it also supports:
  - It can send a single message to one or more recipients.
  - Sending message can include text, voice, video or graphics.
  - It can also send the messages on networks outside the internet.
- The main purpose of SMTP is used to set up communication rules between servers. The servers have a way of identifying themselves and announcing what kind of communication they are trying to perform. They also have a way of handling the errors such as incorrect email address. For example, if the recipient address is wrong, then receiving server reply with an error message of some kind.

Components of SMTP

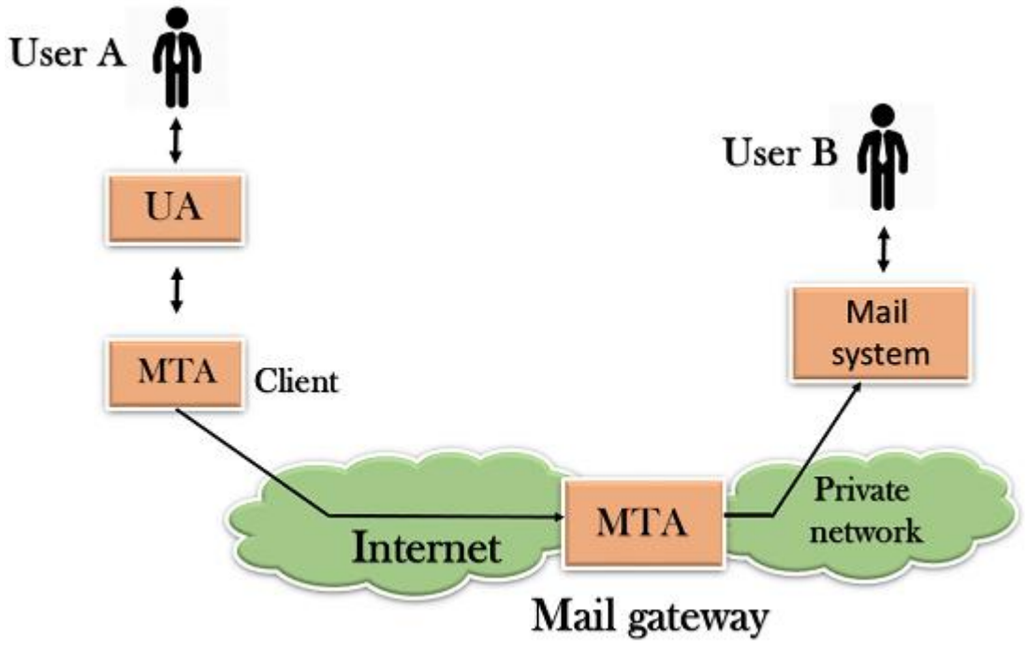


- First, we will break the SMTP client and SMTP server into two components such as user agent (UA) and mail transfer agent (MTA). The user agent (UA) prepares the message, creates the envelope and then puts the message in the envelope. The mail transfer agent (MTA) transfers this mail across the internet.

- SMTP allows a more complex system by adding a relaying system. Instead of just having one MTA at sending side and one at receiving side, more MTAs can be added, acting either as a client or server to relay the email.



- The relaying system without TCP/IP protocol can also be used to send the emails to users, and this is achieved by the use of the mail gateway. The mail gateway is a relay MTA that can be used to receive an email.



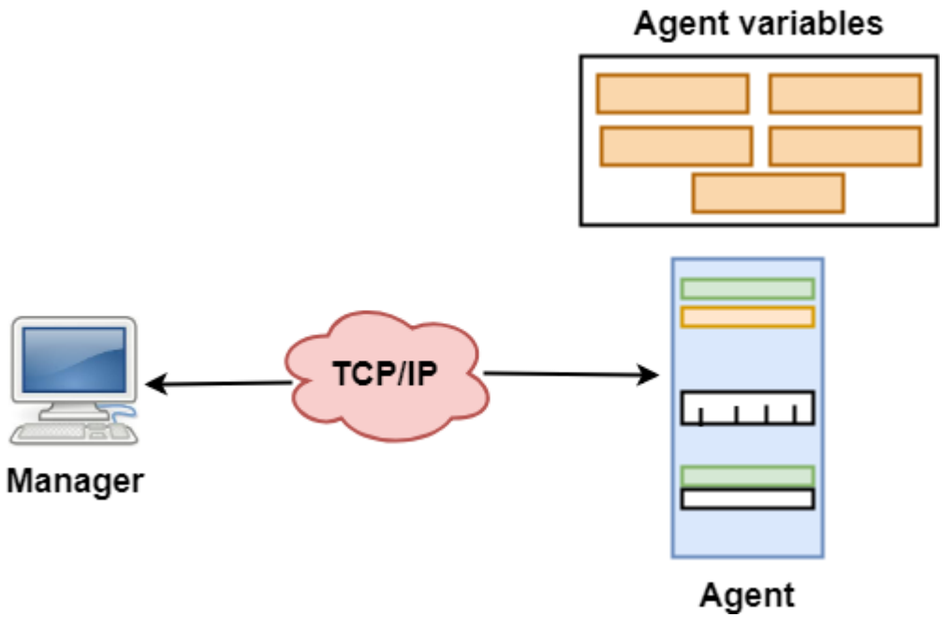
Working of SMTP

1. **Composition of Mail:** A user sends an e-mail by composing an electronic mail message using a Mail User Agent (MUA). Mail User Agent is a program which is used to send and receive mail. The message contains two parts: body and header. The body is the main part of the message while the header includes information such as the sender and recipient address. The header also includes descriptive information such as the subject of the message. In this case, the message body is like a letter and header is like an envelope that contains the recipient's address.
2. **Submission of Mail:** After composing an email, the mail client then submits the completed e-mail to the SMTP server by using SMTP on TCP port 25.
3. **Delivery of Mail:** E-mail addresses contain two parts: username of the recipient and domain name. For example, vivek@gmail.com, where "vivek" is the username of the recipient and "gmail.com" is the domain name.
   If the domain name of the recipient's email address is different from the sender's domain name, then MSA will send the mail to the Mail Transfer Agent (MTA). To relay the email, the MTA will find the target domain. It checks the MX record from Domain Name System to obtain the target domain. The MX record contains the domain name and IP address of the recipient's domain. Once the record is located, MTA connects to the exchange server to relay the message.
4. **Receipt and Processing of Mail:** Once the incoming message is received, the exchange server delivers it to the incoming server (Mail Delivery Agent) which stores the e-mail where it waits for the user to retrieve it.
5. **Access and Retrieval of Mail:** The stored email in MDA can be retrieved by using MUA (Mail User Agent). MUA can be accessed by using login and password.

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.
- It is an application layer protocol in which a few manager stations can handle a set of agents.
- The protocol designed at the application level can monitor the devices made by different manufacturers and installed on different physical networks.
- It is used in a heterogeneous network made of different LANs and WANs connected by routers or gateways.

Managers & Agents

- A manager is a host that runs the SNMP client program while the agent is a router that runs the SNMP server program.

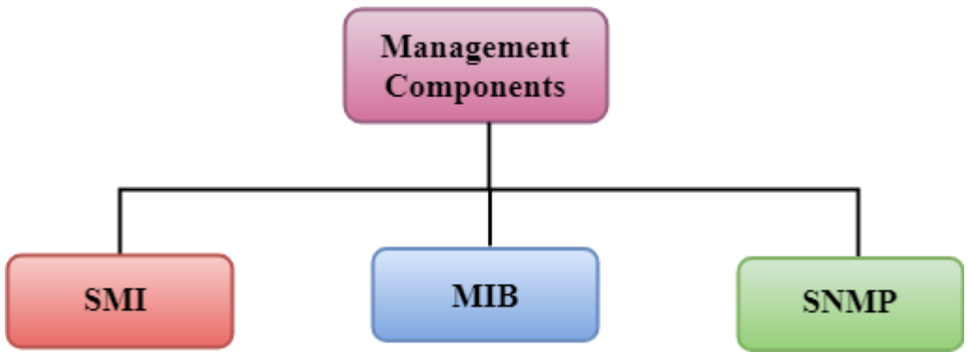
- Management of the internet is achieved through simple interaction between a manager and agent.
- The agent is used to keep the information in a database while the manager is used to access the values in the database. For example, a router can store the appropriate variables such as a number of packets received and forwarded while the manager can compare these variables to determine whether the router is congested or not.
- Agents can also contribute to the management process. A server program on the agent checks the environment, if something goes wrong, the agent sends a warning message to the manager.

Management with SNMP has three basic ideas:

- A manager checks the agent by requesting the information that reflects the behavior of the agent.
- A manager also forces the agent to perform a certain function by resetting values in the agent database.
- An agent also contributes to the management process by warning the manager regarding an unusual condition.

Management Components

- Management is not achieved only through the SNMP protocol but also the use of other protocols that can cooperate with the SNMP protocol. Management is achieved through the use of the other two protocols: SMI (Structure of management information) and MIB(management information base).
- Management is a combination of SMI, MIB, and SNMP. All these three protocols such as abstract syntax notation 1 (ASN.1) and basic encoding rules (BER).

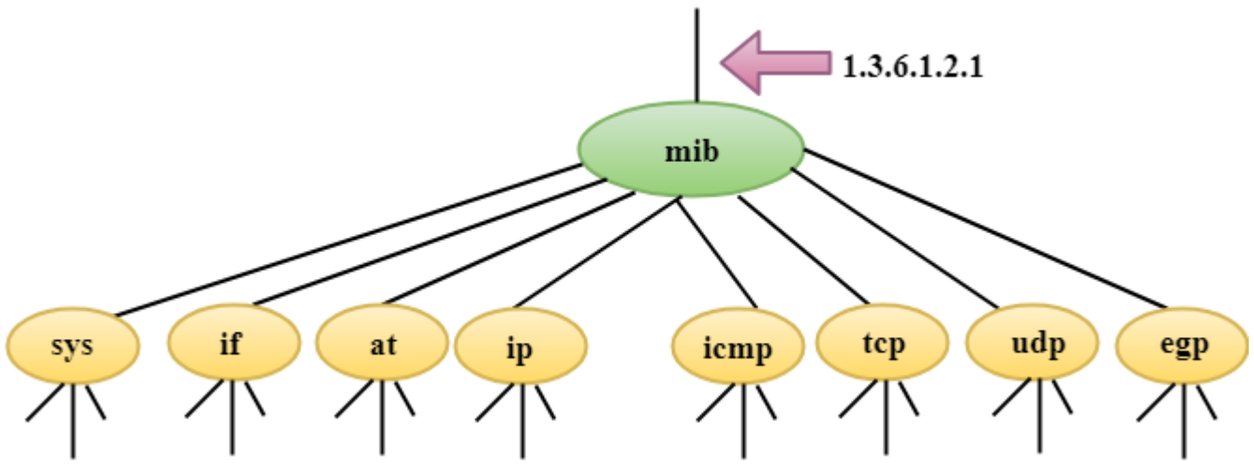


SMI

The SMI (Structure of management information) is a component used in network management. Its main function is to define the type of data that can be stored in an object and to show how to encode the data for the transmission over a network.

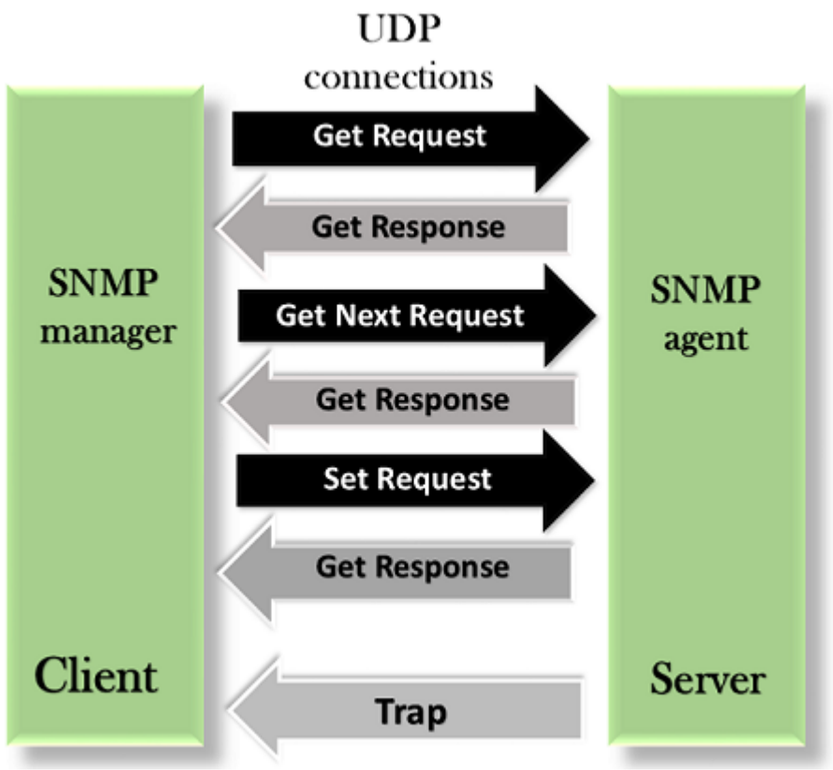
MIB

- The MIB (Management information base) is a second component for the network management.
- Each agent has its own MIB, which is a collection of all the objects that the manager can manage. MIB is categorized into eight groups: system, interface, address translation, ip, icmp, tcp, udp, and egp. These groups are under the mib object.



SNMP

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.

HTTP

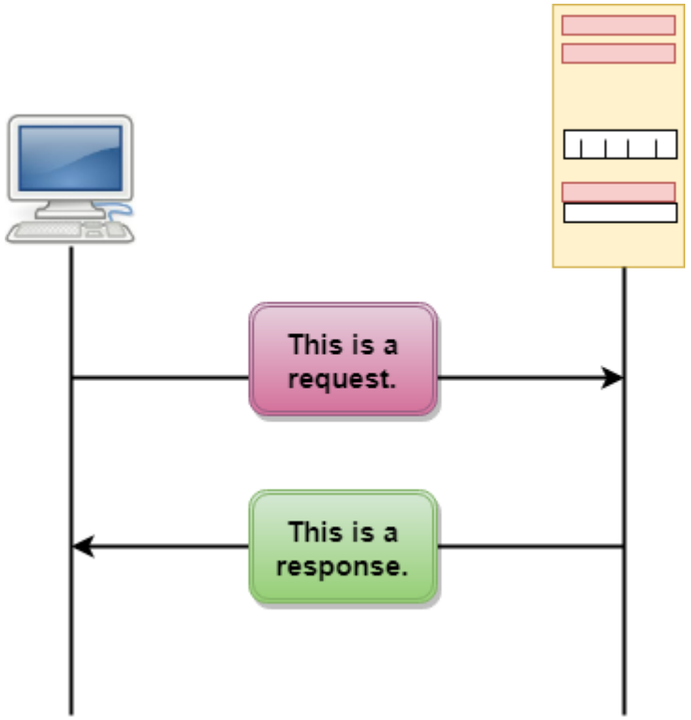
- HTTP stands for **HyperText Transfer Protocol**.

- It is a protocol used to access the data on the World Wide Web (www).
- The HTTP protocol can be used to transfer the data in the form of plain text, hypertext, audio, video, and so on.
- This protocol is known as HyperText Transfer Protocol because of its efficiency that allows us to use in a hypertext environment where there are rapid jumps from one document to another document.
- HTTP is similar to the FTP as it also transfers the files from one host to another host. But, HTTP is simpler than FTP as HTTP uses only one connection, i.e., no control connection to transfer the files.
- HTTP is used to carry the data in the form of MIME-like format.
- HTTP is similar to SMTP as the data is transferred between client and server. The HTTP differs from the SMTP in the way the messages are sent from the client to the server and from server to the client. SMTP messages are stored and forwarded while HTTP messages are delivered immediately.

Features of HTTP:

- **Connectionless protocol:** HTTP is a connectionless protocol. HTTP client initiates a request and waits for a response from the server. When the server receives the request, the server processes the request and sends back the response to the HTTP client after which the client disconnects the connection. The connection between client and server exist only during the current request and response time only.
- **Media independent:** HTTP protocol is a media independent as data can be sent as long as both the client and server know how to handle the data content. It is required for both the client and server to specify the content type in MIME-type header.
- **Stateless:** HTTP is a stateless protocol as both the client and server know each other only during the current request. Due to this nature of the protocol, both the client and server do not retain the information between various requests of the web pages.

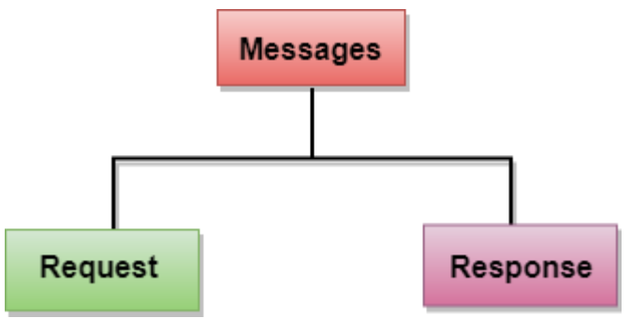
HTTP Transactions



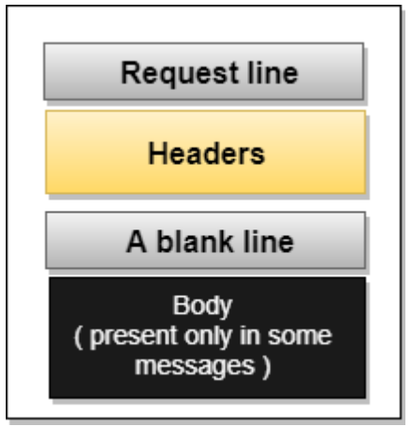
The above figure shows the HTTP transaction between client and server. The client initiates a transaction by sending a request message to the server. The server replies to the request message by sending a response message.

Messages

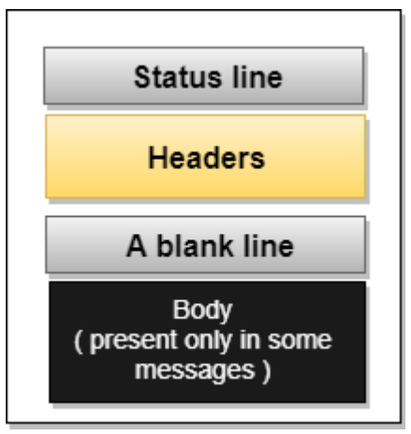
HTTP messages are of two types: request and response. Both the message types follow the same message format.



Request Message: The request message is sent by the client that consists of a request line, headers, and sometimes a body.



Response Message: The response message is sent by the server to the client that consists of a status line, headers, and sometimes a body.



Uniform Resource Locator (URL)

- A client that wants to access the document in an internet needs an address and to facilitate the access of documents, the HTTP uses the concept of Uniform Resource Locator (URL).
- The Uniform Resource Locator (URL) is a standard way of specifying any kind of information on the internet.
- The URL defines four parts: method, host computer, port, and path.

- **Method:** The method is the protocol used to retrieve the document from a server. For example, HTTP.
- **Host:** The host is the computer where the information is stored, and the computer is given an alias name. Web pages are mainly stored in the computers and the computers are given an alias name that begins with the characters "www". This field is not mandatory.
- **Port:** The URL can also contain the port number of the server, but it's an optional field. If the port number is included, then it must come between the host and path and it should be separated from the host by a colon.
- **Path:** Path is the pathname of the file where the information is stored. The path itself contain slashes that separate the directories from the subdirectories and files.

Network Security

Computer Network Security

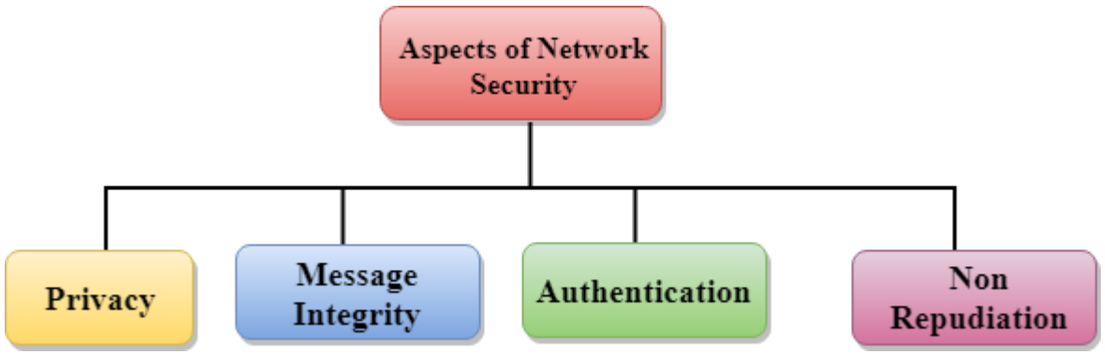
Computer network security consists of measures taken by business or some organizations to monitor and prevent unauthorized access from the outside attackers.

Different approaches to computer network security management have different requirements depending on the size of the computer network. For example, a home office requires basic network security while large businesses require high maintenance to prevent the network from malicious attacks.

Network Administrator controls access to the data and software on the network. A network administrator assigns the user ID and password to the authorized person.

Aspects of Network Security:

Following are the desirable properties to achieve secure communication:



- **Privacy:** Privacy means both the sender and the receiver expects confidentiality. The transmitted message should be sent only to the intended receiver while the message should be opaque for other users. Only the sender and receiver should be able to understand the transmitted message as eavesdroppers can intercept the message. Therefore, there is a requirement to encrypt the message so that the message cannot be intercepted. This aspect of confidentiality is commonly used to achieve secure communication.
- **Message Integrity:** Data integrity means that the data must arrive at the receiver exactly as it was sent. There must be no changes in the data content during transmission, either maliciously or accident, in a transit. As there are more and more monetary exchanges over the internet, data integrity is more crucial. The data integrity must be preserved for secure communication.
- **End-point authentication:** Authentication means that the receiver is sure of the sender?s identity, i.e., no imposter has sent the message.
- **Non-Repudiation:** Non-Repudiation means that the receiver must be able to prove that the received message has come from a specific sender. The sender must not deny sending a message that he or she send. The burden of proving the identity comes on the receiver. For example, if a customer sends a request to transfer the money from one account to another account, then the bank must have a proof that the customer has requested for the transaction.

Privacy

The concept of how to achieve privacy has not been changed for thousands of years: the message cannot be encrypted. The message must be rendered as opaque to all the unauthorized parties. A good encryption/decryption technique is used to achieve privacy to some extent. This technique ensures that the eavesdropper cannot understand the contents of the message.

Encryption/Decryption

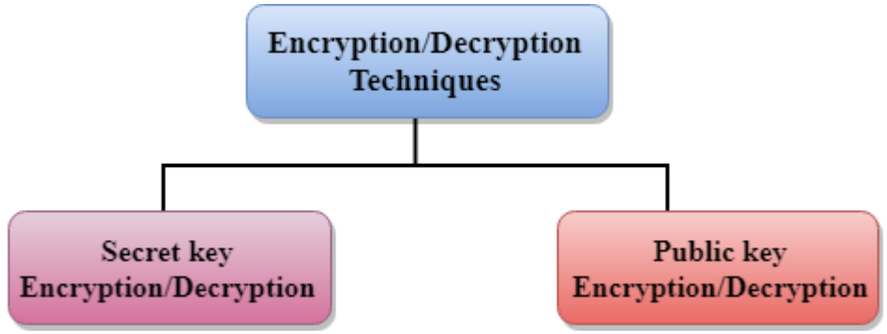
Encryption: Encryption means that the sender converts the original information into another form and sends the unintelligible message over the network.

Decryption: Decryption reverses the Encryption process in order to transform the message back to the original form.

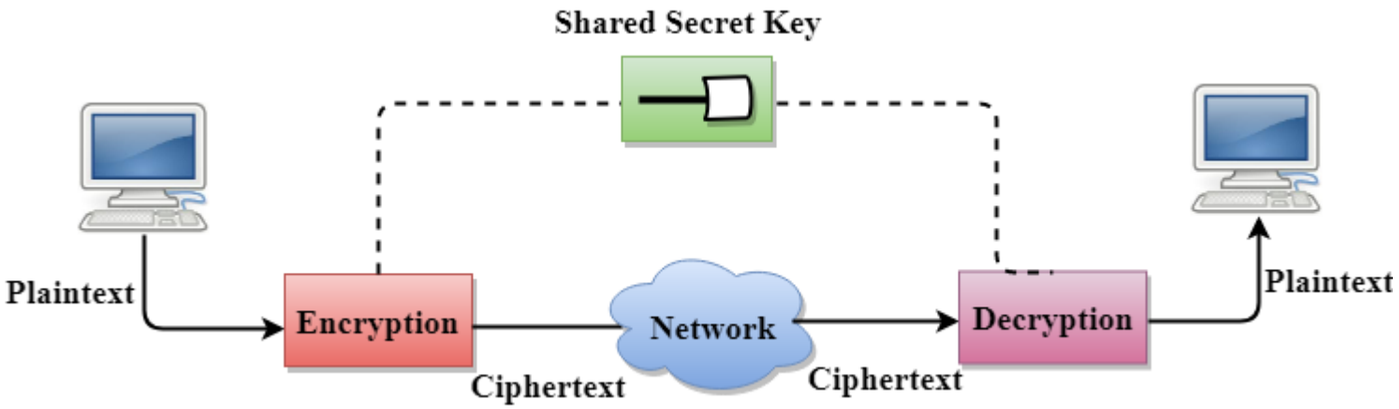
The data which is to be encrypted at the sender site is known as plaintext, and the encrypted data is known as ciphertext. The data is decrypted at the receiver site.

There are two types of Encryption/Decryption techniques:

- Privacy with secret key Encryption/Decryption
- Privacy with public key Encryption/Decryption



Secret Key Encryption/Decryption technique



- In Secret Key Encryption/Decryption technique, the same key is used by both the parties, i.e., the sender and receiver.
- The sender uses the secret key and encryption algorithm to encrypt the data; the receiver uses this key and decryption algorithm to decrypt the data.
- In Secret Key Encryption/Decryption technique, the algorithm used for encryption is the inverse of the algorithm used for decryption. It means that if the encryption algorithm uses a combination of addition and multiplication, then the decryption algorithm uses a combination of subtraction and division.
- The secret key encryption algorithm is also known as symmetric encryption algorithm because the same secret key is used in bidirectional communication.
- In secret key encryption/decryption algorithm, the secret code is used by the computer to encrypt the information before it is sent over the network to another computer.

- The secret key requires that we should know which computers are talking to each other so that we can install the key on each computer.

Data Encryption Standard (DES)

- The Data Encryption Standard (DES) was designed by IBM and adopted by the U.S. government as the standard encryption method for nonmilitary and nonclassified use.
- The Data Encryption Standard is a standard used for encryption, and it is a form of Secret **Key Cryptography**.

Advantage

Efficient: The secret key algorithms are more efficient as it takes less time to encrypt the message than to encrypt the message by using a public key encryption algorithm. The reason for this is that the size of the key is small. Due to this reason, Secret Key Algorithms are mainly used for encryption and decryption.

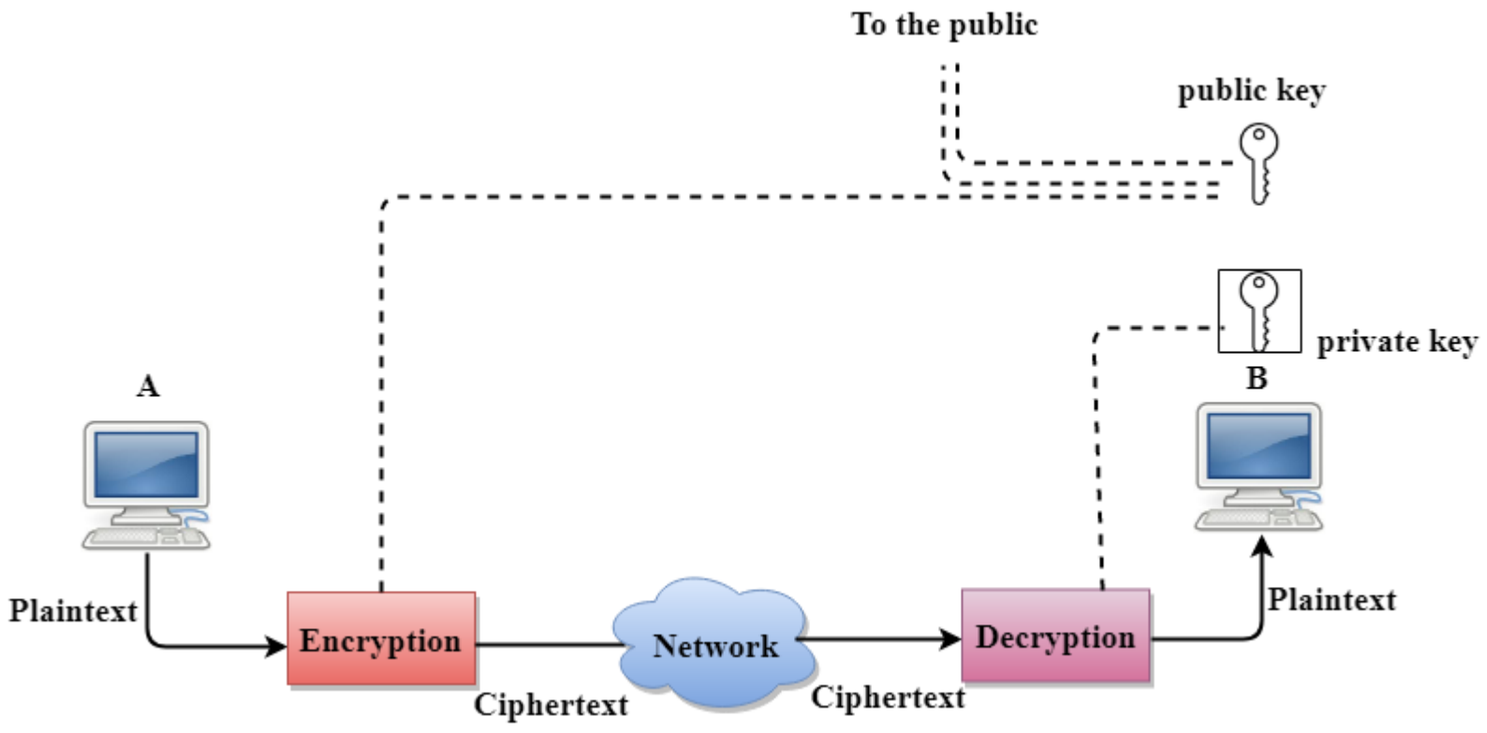
Disadvantages of Secret Key Encryption

The Secret Key Encryption/Decryption has the following disadvantages:

- Each pair of users must have a secret key. If the number of people wants to use this method in the world is N, then there are N(N-1)/2 secret keys. For example, for one million people, then there are half billion secret keys.
- The distribution of keys among different parties can be very difficult. This problem can be resolved by combining the Secret Key Encryption/Decryption with the Public Key Encryption/Decryption algorithm.

Public Key Encryption/Decryption technique

- There are two keys in public key encryption: a private key and a public key.
- The private key is given to the receiver while the public key is provided to the public.



In the above figure, we see that A is sending the message to user B. 'A' uses the public key to encrypt the data while 'B' uses the private key to decrypt the data.

- In public key Encryption/Decryption, the public key used by the sender is different from the private key used by the receiver.
- The public key is available to the public while the private key is kept by each individual.
- The most commonly used public key algorithm is known as RSA.

Advantages of Public Key Encryption

- The main restriction of private key encryption is the sharing of a secret key. A third party cannot use this key. In public key encryption, each entity creates a pair of keys, and they keep the private one and distribute the public key.
- The number of keys in public key encryption is reduced tremendously. For example, for one million users to communicate, only two million keys are required, not a half-billion keys as in the case of secret key encryption.

Disadvantages of Public Key Encryption

- **Speed:** One of the major disadvantage of the public-key encryption is that it is slower than secret-key encryption. In secret key encryption, a single shared key is used to encrypt and decrypt the message which speeds up the process while in public key encryption, different two keys are used, both related to each other by a complex mathematical process. Therefore, we can say that encryption and decryption take more time in public key encryption.
- **Authentication:** A public key encryption does not have a built-in authentication. Without authentication, the message can be interpreted or intercepted without the user's knowledge.
- **Inefficient:** The main disadvantage of the public key is its complexity. If we want the method to be effective, large numbers are needed. But in public key encryption, converting the plaintext into ciphertext using long keys takes a lot of time. Therefore, the public key encryption algorithms are efficient for short messages not for long messages.

Differences b/w Secret Key Encryption & Public Key Encryption

Basis for Comparison	Secret Key Encryption	Public Key Encryption
Define	Secret Key Encryption is defined as the technique that uses a single shared key to encrypt and decrypt the message.	Public Key Encryption is defined as the technique that uses two different keys for encryption and decryption.
Efficieny	It is efficient as this technique is recommended for large amounts of text.	It is inefficient as this technique is used only for short messages.
Other name	It is also known as Symmetric Key encryption.	It is also known as Asymmetric Key Encryption.
Speed	Its speed is high as it uses a single key for encryption and decryption.	Its speed is slow as it uses two different keys, both keys are related to each other through the complicated mathematical process.

Algorithms	The Secret key algorithms are DES, 3DES, AES & RCA.	The Public key algorithms are Diffie-Hellman, RSA.
Purpose	The main purpose of the secret key algorithm is to transmit the bulk data.	The main purpose of the public key algorithm is to share the keys securely.

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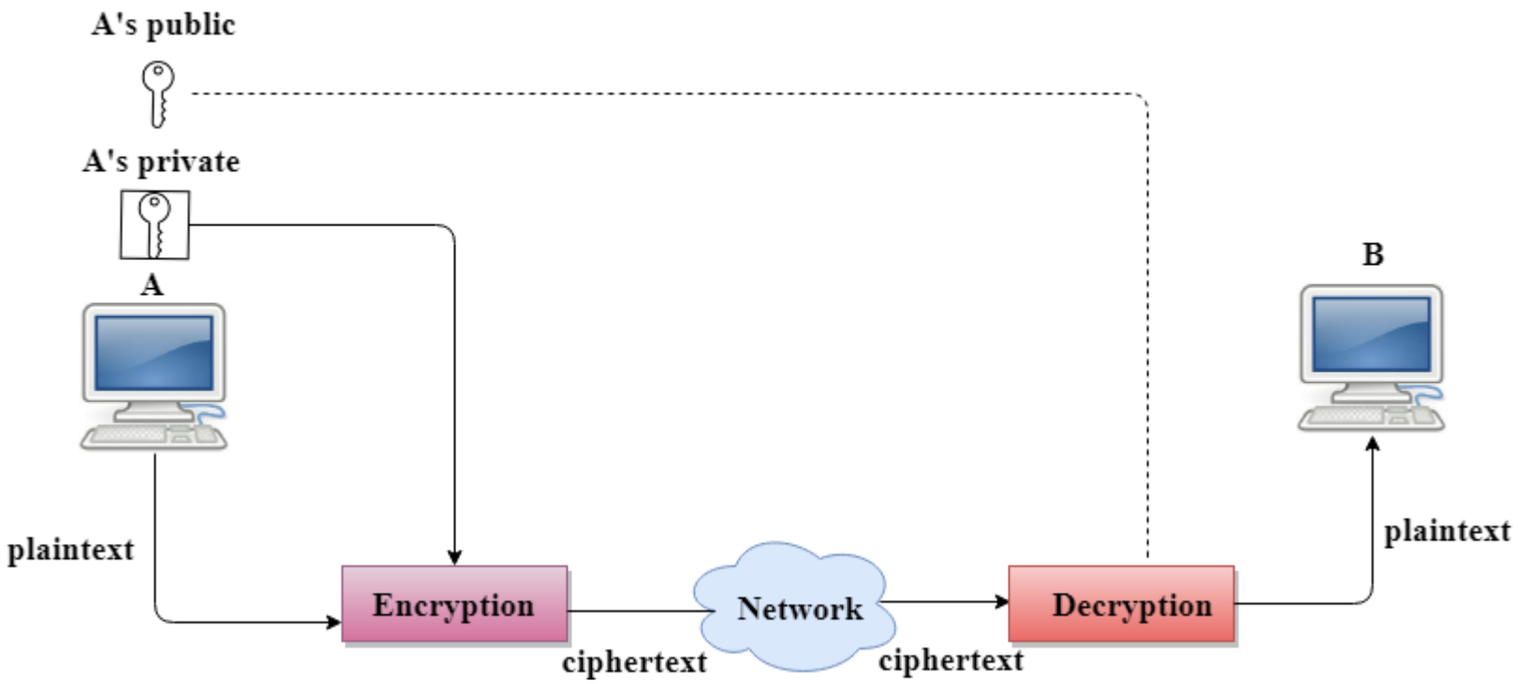
Digital Signature

The Digital Signature is a technique which is used to validate the authenticity and integrity of the message. We know that there are four aspects of security: privacy, authentication, integrity, and non-repudiation. We have already discussed the first aspect of security and other three aspects can be achieved by using a digital signature.

The basic idea behind the Digital Signature is to sign a document. When we send a document electronically, we can also sign it. We can sign a document in two ways: to sign a whole document and to sign a digest.

Signing the Whole Document

- In Digital Signature, a public key encryption technique is used to sign a document. However, the roles of a public key and private key are different here. The sender uses a private key to encrypt the message while the receiver uses the public key of the sender to decrypt the message.
- In Digital Signature, the private key is used for encryption while the public key is used for decryption.
- Digital Signature cannot be achieved by using secret key encryption.



Digital Signature is used to achieve the following three aspects:

- **Integrity:** The Digital Signature preserves the integrity of a message because, if any malicious attack intercepts a message and partially or totally changes it, then the decrypted message would be impossible.
- **Authentication:** We can use the following reasoning to show how the message is authenticated. If an intruder (user X) sends a message pretending that it is coming from someone else (user A), user X uses her own private key to encrypt the message. The message is decrypted by using the public key of user A. Therefore this makes the message unreadable. Encryption with X's private key and decryption with A's public key results in garbage value.
- **Non-Repudiation:** Digital Signature also provides non-repudiation. If the sender denies sending the message, then her private key corresponding to her public key is tested on the plaintext. If the decrypted message is the same as the original message, then we know that the sender has sent the message.

Note: Digital Signature does not provide privacy. If there is a need for privacy, then another layer of encryption/decryption is applied.

Signing the Digest

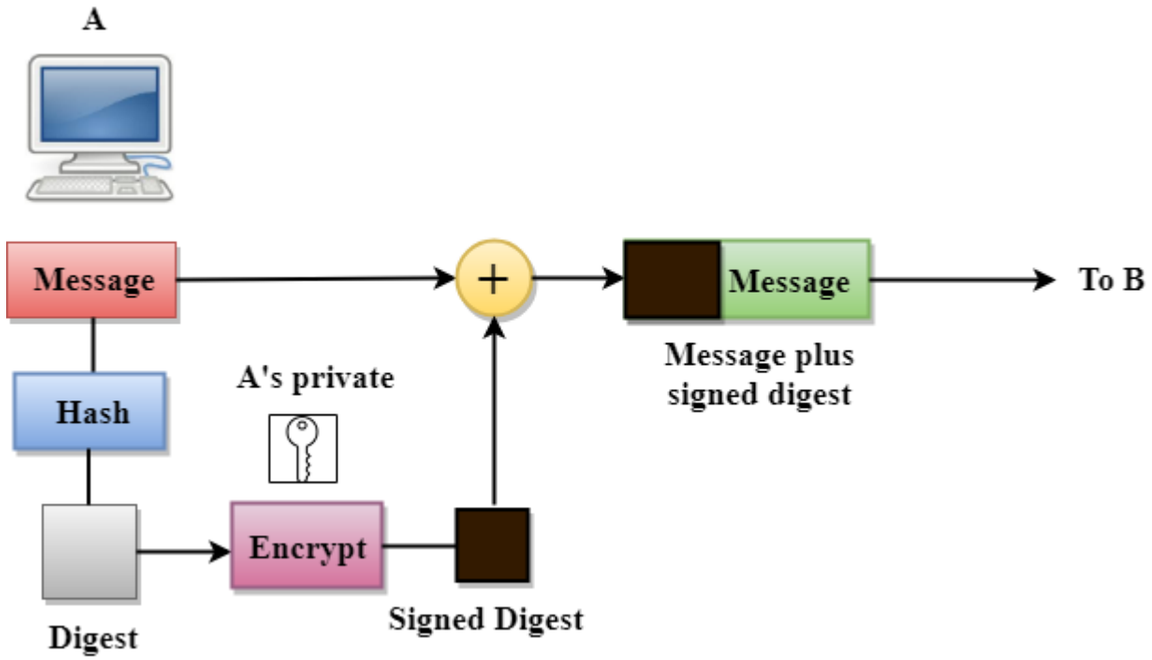
- Public key encryption is efficient if the message is short. If the message is long, a public key encryption is inefficient to use. The solution to this problem is to let the sender sign a digest of the document instead of the whole document.
- The sender creates a miniature version (digest) of the document and then signs it, the receiver checks the signature of the miniature version.

- The hash function is used to create a digest of the message. The hash function creates a fixed-size digest from the variable-length message.
- The two most common hash functions used: MD5 (Message Digest 5) and SHA-1 (Secure Hash Algorithm 1). The first one produces 120-bit digest while the second one produces a 160-bit digest.
- A hash function must have two properties to ensure the success:
  - First, the digest must be one way, i.e., the digest can only be created from the message but not vice versa.
  - Second, hashing is a one-to-one function, i.e., two messages should not create the same digest.

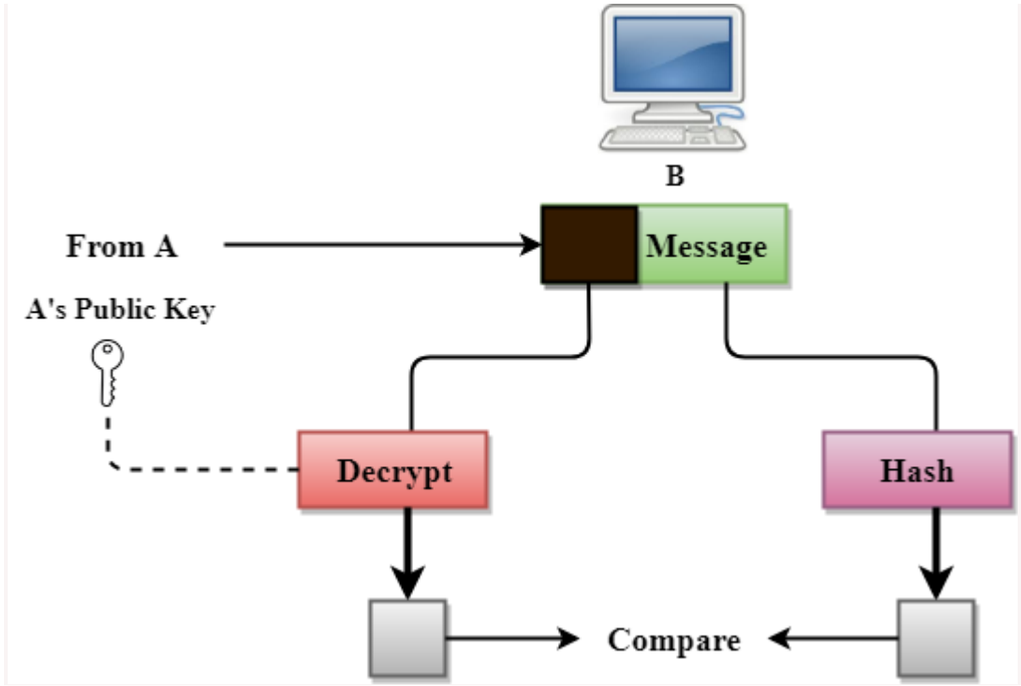
Following are the steps taken to ensure security:

- The miniature version (digest) of the message is created by using a hash function.
- The digest is encrypted by using the sender's private key.
- After the digest is encrypted, then the encrypted digest is attached to the original message and sent to the receiver.
- The receiver receives the original message and encrypted digest and separates the two. The receiver implements the hash function on the original message to create the second digest, and it also decrypts the received digest by using the public key of the sender. If both the digests are same, then all the aspects of security are preserved.

At the Sender site



At the Receiver site



PGP

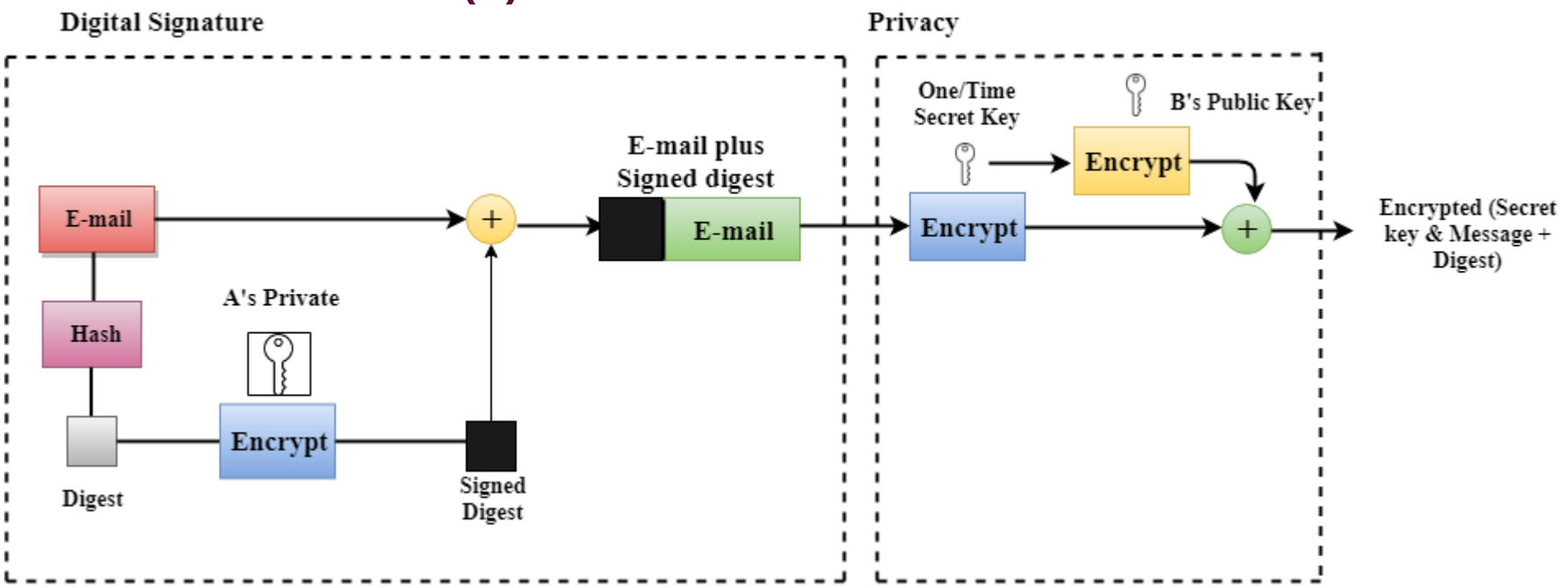
- PGP stands for Pretty Good Privacy (PGP) which is invented by Phil Zimmermann.
- PGP was designed to provide all four aspects of security, i.e., privacy, integrity, authentication, and non-repudiation in the sending of email.
- PGP uses a digital signature (a combination of hashing and public key encryption) to provide integrity, authentication, and non-repudiation. PGP uses a combination of secret key encryption and public key encryption to provide privacy. Therefore, we can say that the digital signature uses one hash function, one secret key, and two private-public key pairs.
- PGP is an open source and freely available software package for email security.
- PGP provides authentication through the use of Digital Signature.
- It provides confidentiality through the use of symmetric block encryption.
- It provides compression by using the ZIP algorithm, and EMAIL compatibility using the radix-64 encoding scheme.

Following are the steps taken by PGP to create secure e-mail at the sender site:

- The e-mail message is hashed by using a hashing function to create a digest.
- The digest is then encrypted to form a signed digest by using the sender's private key, and then signed digest is added to the original email message.

- The original message and signed digest are encrypted by using a one-time secret key created by the sender.
- The secret key is encrypted by using a receiver's public key.
- Both the encrypted secret key and the encrypted combination of message and digest are sent together.

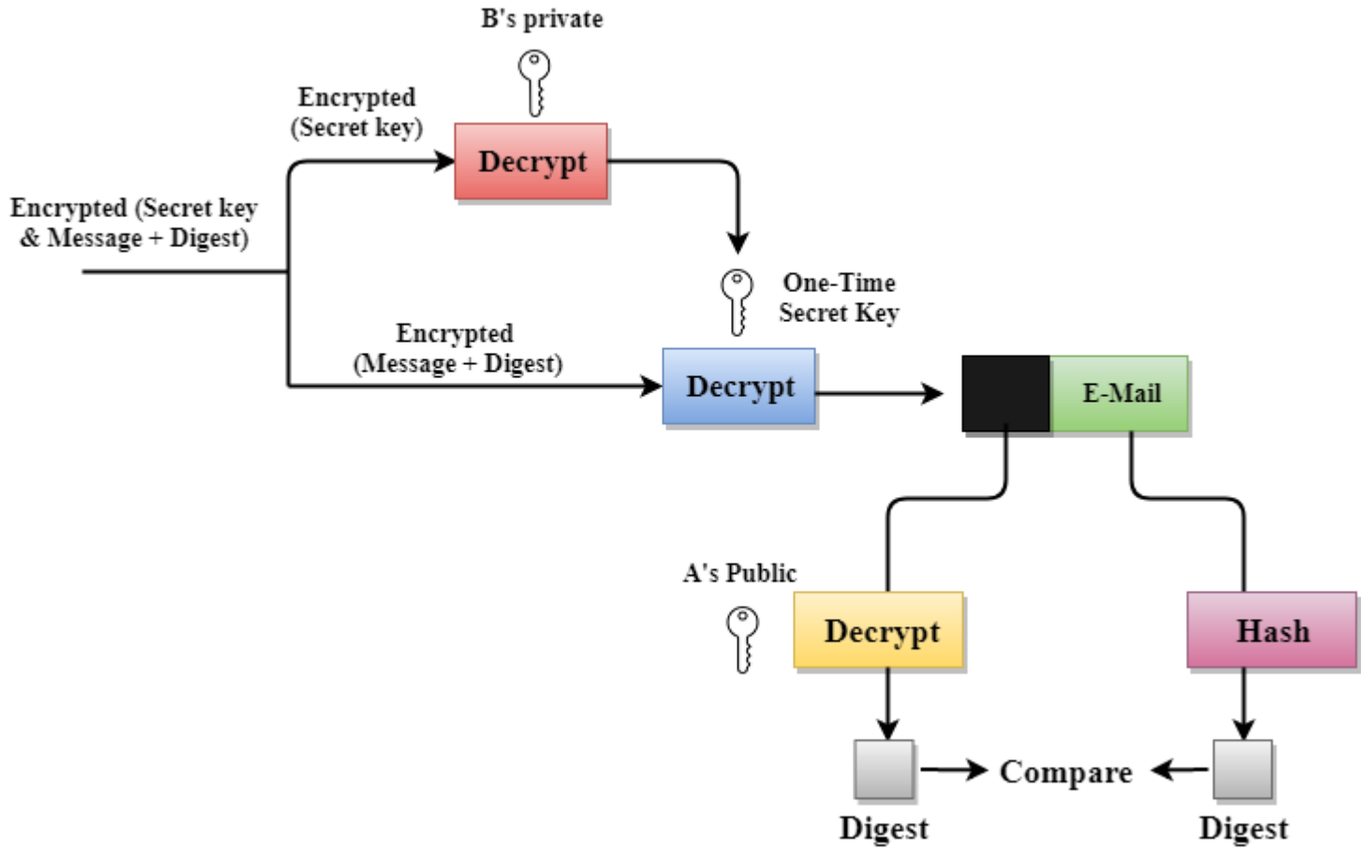
PGP at the Sender site (A)



Following are the steps taken to show how PGP uses hashing and a combination of three keys to generate the original message:

- The receiver receives the combination of encrypted secret key and message digest is received.
- The encrypted secret key is decrypted by using the sender's private key to get the one-time secret key.
- The secret key is then used to decrypt the combination of message and digest.
- The digest is decrypted by using the sender's public key, and the original message is hashed by using a hash function to create a digest.
- Both the digests are compared if both of them are equal means that all the aspects of security are preserved.

PGP at the Receiver site (B)



Disadvantages of PGP Encryption

- **The Administration is difficult:** The different versions of PGP complicate the administration.
- **Compatibility issues:** Both the sender and the receiver must have compatible versions of PGP. For example, if you encrypt an email by using PGP with one of the encryption technique, the receiver has a different version of PGP which cannot read the data.
- **Complexity:** PGP is a complex technique. Other security schemes use symmetric encryption that uses one key or asymmetric encryption that uses two different keys. PGP uses a hybrid approach that implements symmetric encryption with two keys. PGP is more complex, and it is less familiar than the traditional symmetric or asymmetric methods.
- **No Recovery:** Computer administrators face the problems of losing their passwords. In such situations, an administrator should use a special program to retrieve passwords. For example, a technician has physical access to a PC which can be used to retrieve a password. However, PGP does not offer such a special program for recovery; encryption methods are very strong so, it does not retrieve the forgotten passwords results in lost messages or lost files.