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PG DEPARTMENT OF
COMPUTER SCIENCE

SUBJECT NAME: COMPUTER NETWORKS

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UNIT-1

1.1 Introduction to Computer Networks

Modern world scenario is ever changing. Data Communication and network have changed the way business and other daily affair works. Now, they highly rely on computer networks and internetwork.

A set of devices often mentioned as nodes connected by media link is called a Network.

A node can be a device which is capable of sending or receiving data generated by other nodes on the network like a computer, printer etc. These links connecting the devices are called **Communication channels**.

Computer network is a telecommunication channel using which we can share data with other computers or devices, connected to the same network. It is also called Data Network. The best example of computer network is Internet.

A network must be able to meet certain criteria, these are mentioned below:

1. Performance
2. Reliability
3. Scalability

Performance

It can be measured in the following ways:

- **Transit time :** It is the time taken to travel a message from one device to another.
- **Response time :** It is defined as the time elapsed between enquiry and response.

Other ways to measure performance are :

1. Efficiency of software
2. Number of users
3. Capability of connected hardware

Reliability

It decides the frequency at which network failure take place. More the failures are, less is the network's reliability.

Security

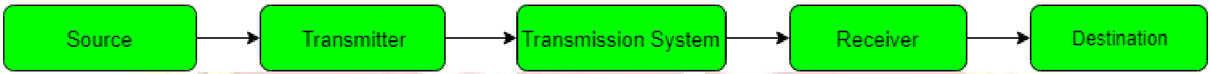
It refers to the protection of data from any unauthorised user or access. While travelling through network, data passes many layers of network, and data can be traced if attempted. Hence security is also a very important characteristic for Networks.

Properties of a Good Network

1. **Interpersonal Communication:** We can communicate with each other efficiently and easily. Example: emails, chat rooms, video conferencing etc, all of these are possible because of computer networks.
2. **Resources can be shared:** We can share physical resources by making them available on a network such as printers, scanners etc.
3. **Sharing files, data:** Authorised users are allowed to share the files on the network.

1.2 Basic Communication Model

A Communication model is used to exchange data between two parties. For example: communication between a computer, server and telephone (through modem).



Source

Data to be transmitted is generated by this device, example: telephones, personal computers etc.

Transmitter

The data generated by the source system is not directly transmitted in the form its generated. The transmitter transforms and encodes the data in such a form to produce electromagnetic waves or signals.

Transmission System

A transmission system can be a single transmission line or a complex network connecting source and destination.

Receiver

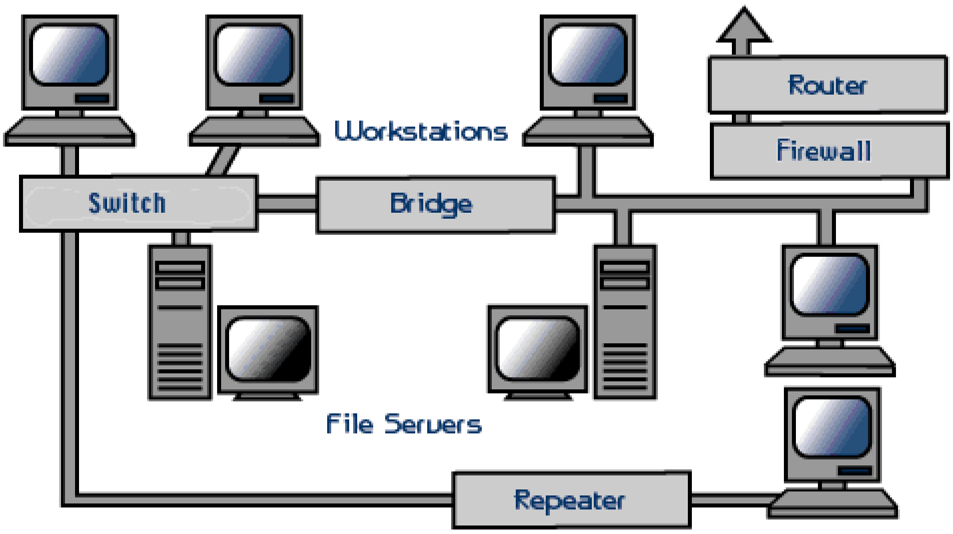
Receiver accepts the signal from the transmission system and converts it into a form which is easily managed by the destination device.

Destination

Destination receives the incoming data from the receiver.

1.3 Network Hardware

Networking hardware includes all computers, peripherals, interface cards and other equipment needed to perform data-processing and communications within the network.



The basic computer hardware components that are needed to set up a network are as follows –

Network Cables

Network cables are the transmission media to transfer data from one device to another. A commonly used network cable is category 5 cable with RJ – 45 connector, as shown in the image below:

Routers

A router is a connecting device that transfers data packets between different computer networks. Typically, they are used to connect a PC or an organization's LAN to a broadband internet connection. They contain RJ-45 ports so that computers and other devices can connect with them using network cables.

Repeaters, Hubs, and Switches

Repeaters, hubs and switches connect network devices together so that they can function as a single segment.

A repeater receives a signal and regenerates it before re-transmitting so that it can travel longer distances.

A hub is a multiport repeater having several input/output ports, so that input at any port is available at every other port.

A switch receives data from a port, uses packet switching to resolve the destination device and then forwards the data to the particular destination, rather than broadcasting it as a hub.

REPEATER

HUB

SWITCH

Bridges

A bridge connects two separate Ethernet network segments. It forwards packets from the source network to the destined network.

Gateways

A gateway connects entirely different networks that work upon different protocols. It is the entry and the exit point of a network and controls access to other networks.

Network Interface Cards

NIC is a component of the computer to connect it to a network. Network cards are of two types: Internal network cards and external network cards.

1.4 Network software

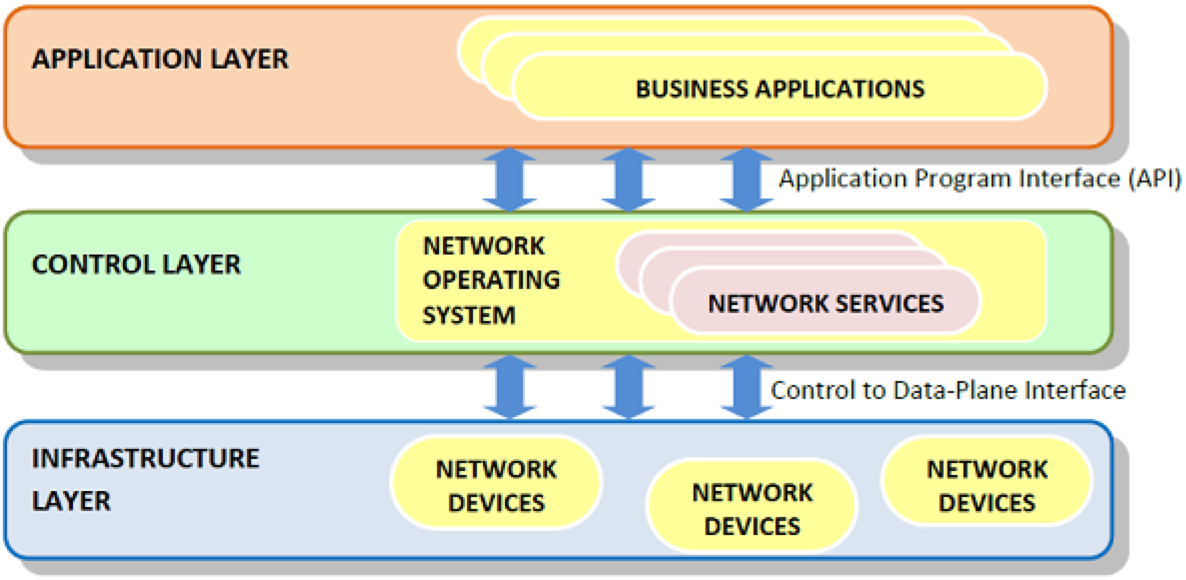
Network software encompasses a broad range of software used for design, implementation, and operation and monitoring of computer networks. Traditional networks were hardware based with software embedded. With the advent of Software – Defined Networking (SDN), software is separated from the hardware thus making it more adaptable to the ever-changing nature of the computer network.

Functions of Network Software

- Helps to set up and install computer networks
- Enables users to have access to network resources in a seamless manner
- Allows administrations to add or remove users from the network
- Helps to define locations of data storage and allows users to access that data
- Helps administrators and security system to protect the network from data breaches, unauthorized access and attacks on a network
- Enables network virtualizations

SDN Framework

The Software Defined Networking framework has three layers as depicted in the following diagram –



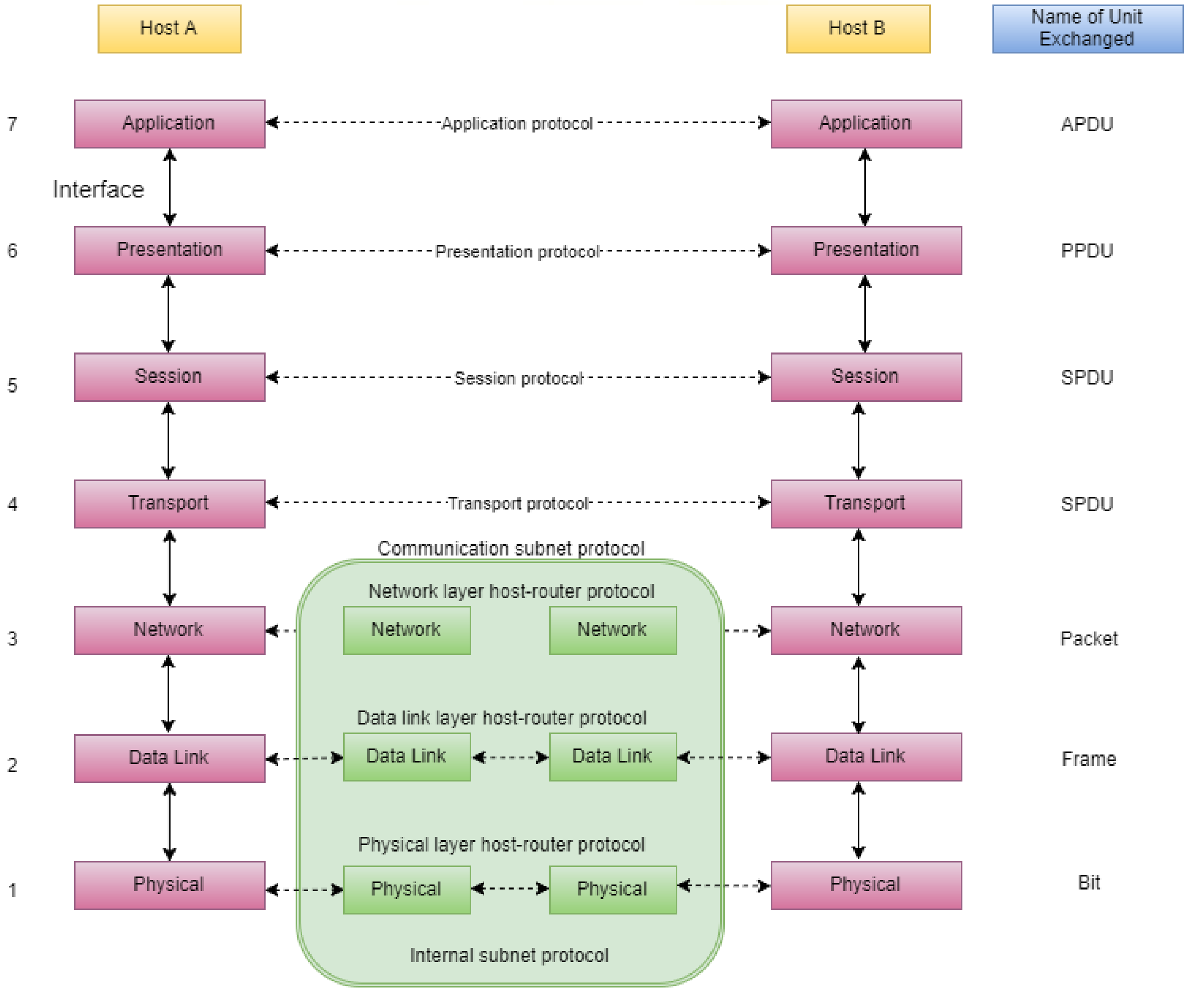
- **APPLICATION LAYER** – SDN applications reside in the Application Layer. The applications convey their needs for resources and services to the control layer through APIs.
- **CONTROL LAYER** – The Network Control Software, bundled into the Network Operating System, lies in this layer. It provides an abstract view of the underlying network infrastructure. It receives the requirements of the SDN applications and relays them to the network components.
- **INFRASTRUCTURE LAYER** – Also called the Data Plane Layer, this layer contains the actual network components. The network devices reside in this layer that shows their network capabilities through the Control to data-Plane Interface.

1.7 The OSI Model

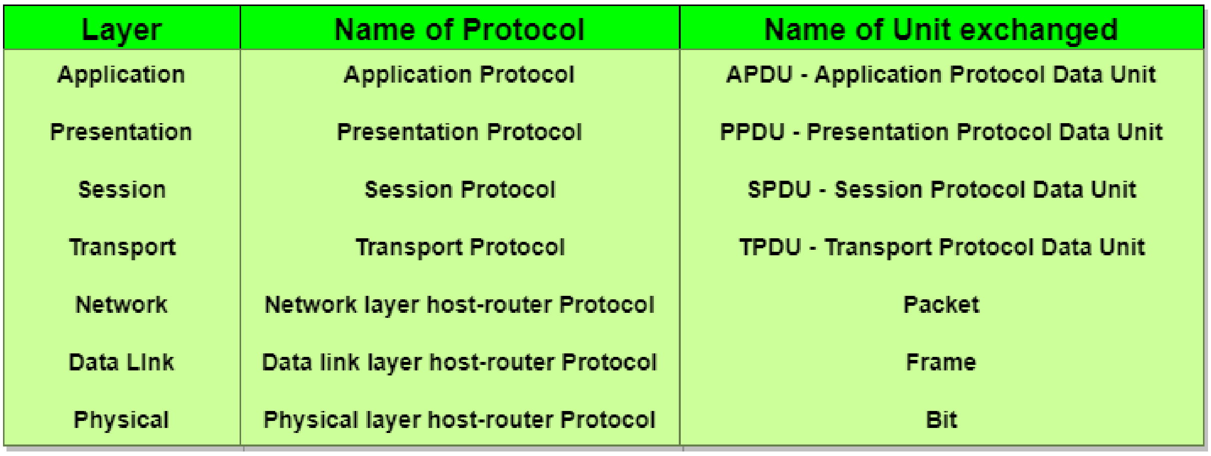
There are n numbers of users who use computer network and are located over the world. So to ensure, national and worldwide data communication, systems must be developed which are compatible to communicate with each other ISO has developed a standard. ISO stands for **International organization of Standardization**. This is called a model for **Open System Interconnection** (OSI) and is commonly known as OSI model.

The ISO-OSI model is a seven layer architecture. It defines seven layers or levels in a complete communication system. They are:

1. Application Layer
2. Presentation Layer
3. Session Layer
4. Transport Layer
5. Network Layer
6. Datalink Layer
7. Physical Layer



In the table below, we have specified the **protocols** used and the **data unit** exchanged by each layer of the OSI Model.



Layer	Name of Protocol	Name of Unit exchanged
Application	Application Protocol	APDU - Application Protocol Data Unit
Presentation	Presentation Protocol	PPDU - Presentation Protocol Data Unit
Session	Session Protocol	SPDU - Session Protocol Data Unit
Transport	Transport Protocol	TPDU - Transport Protocol Data Unit
Network	Network layer host-router Protocol	Packet
Data Link	Data link layer host-router Protocol	Frame
Physical	Physical layer host-router Protocol	Bit

Principles of OSI Reference Model

The OSI reference model has 7 layers. The principles that were applied to arrive at the seven layers can be briefly summarized as follows:

1. A layer should be created where a different abstraction is needed.
2. Each layer should perform a well-defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that architecture does not become unwieldly.

Functions of Different Layers

OSI Model Layer 1: The Physical Layer

1. Physical Layer is the lowest layer of the OSI Model.
2. It activates, maintains and deactivates the physical connection.
3. It is responsible for transmission and reception of the unstructured raw data over network.
4. Voltages and data rates needed for transmission is defined in the physical layer.
5. It converts the digital/analog bits into electrical signal or optical signals.
6. Data encoding is also done in this layer.

Functions of Physical Layer

Following are the various functions performed by the Physical layer of the OSI model.

1. **Representation of Bits:** Data in this layer consists of stream of bits. The bits must be encoded into signals for transmission. It defines the type of encoding i.e. how 0's and 1's are changed to signal.
2. **Data Rate:** This layer defines the rate of transmission which is the number of bits per second.
3. **Synchronization:** It deals with the synchronization of the transmitter and receiver. The sender and receiver are synchronized at bit level.
4. **Interface:** The physical layer defines the transmission interface between devices and transmission medium.
5. **Line Configuration:** This layer connects devices with the medium: Point to Point configuration and Multipoint configuration.
6. **Topologies:** Devices must be connected using the following topologies: Mesh, Star, Ring and Bus.
7. **Transmission Modes:** Physical Layer defines the direction of transmission between two devices: Simplex, Half Duplex, Full Duplex.
8. Deals with baseband and broadband transmission.

OSI Model Layer 2: Data Link Layer

1. Data link layer synchronizes the information which is to be transmitted over the physical layer.
2. The main function of this layer is to make sure data transfer is error free from one node to another, over the physical layer.
3. Transmitting and receiving data frames sequentially is managed by this layer.
4. This layer sends and expects acknowledgements for frames received and sent respectively. Resending of non-acknowledgement received frames is also handled by this layer.
5. This layer establishes a logical layer between two nodes and also manages the Frame traffic control over the network. It signals the transmitting node to stop, when the frame buffers are full.

Functions of Data Link Layer

1. **Framing:** Frames are the streams of bits received from the network layer into manageable data units. This division of stream of bits is done by Data Link Layer.

2. **Physical Addressing:** The Data Link layer adds a header to the frame in order to define physical address of the sender or receiver of the frame, if the frames are to be distributed to different systems on the network.
3. **Flow Control:** A flow control mechanism to avoid a fast transmitter from running a slow receiver by buffering the extra bit is provided by flow control. This prevents traffic jam at the receiver side.
4. **Error Control:** Error control is achieved by adding a trailer at the end of the frame. Duplication of frames are also prevented by using this mechanism. Data Link Layers adds mechanism to prevent duplication of frames.
5. **Access Control:** Protocols of this layer determine which of the devices has control over the link at any given time, when two or more devices are connected to the same link.

OSI Model Layer 3: The Network Layer

1. Network Layer routes the signal through different channels from one node to other.
2. It acts as a network controller. It manages the Subnet traffic.
3. It decides by which route data should take.
4. It divides the outgoing messages into packets and assembles the incoming packets into messages for higher levels.

Functions of Network Layer

1. It translates logical network address into physical address. Concerned with circuit, message or packet switching.
2. Routers and gateways operate in the network layer. Mechanism is provided by Network Layer for routing the packets to final destination.
3. Connection services are provided including network layer flow control, network layer error control and packet sequence control.
4. Breaks larger packets into small packets.

OSI Model Layer 4: Transport Layer

1. Transport Layer decides if data transmission should be on parallel path or single path.
2. Functions such as Multiplexing, Segmenting or Splitting on the data are done by this layer
3. It receives messages from the Session layer above it, convert the message into smaller units and passes it on to the Network layer.
4. Transport layer can be very complex, depending upon the network requirements.

Transport layer breaks the message (data) into small units so that they are handled more efficiently by the network layer.

Functions of Transport Layer

1. **Service Point Addressing:** Transport Layer header includes service point address which is port address. This layer gets the message to the correct process on the computer unlike Network Layer, which gets each packet to the correct computer.
2. **Segmentation and Reassembling:** A message is divided into segments; each segment contains sequence number, which enables this layer in reassembling the message. Message is reassembled correctly upon arrival at the destination and replaces packets which were lost in transmission.
3. **Connection Control:** It includes 2 types:
    - Connectionless Transport Layer : Each segment is considered as an independent packet and delivered to the transport layer at the destination machine.
    - Connection Oriented Transport Layer : Before delivering packets, connection is made with transport layer at the destination machine.
4. **Flow Control:** In this layer, flow control is performed end to end.
5. **Error Control:** Error Control is performed end to end in this layer to ensure that the complete message arrives at the receiving transport layer without any error. Error Correction is done through retransmission.

OSI Model Layer 5: The Session Layer

1. Session Layer manages and synchronize the conversation between two different applications.
2. Transfer of data from source to destination session layer streams of data are marked and are resynchronized properly, so that the ends of the messages are not cut prematurely and data loss is avoided.

Functions of Session Layer

1. **Dialog Control :** This layer allows two systems to start communication with each other in half-duplex or full-duplex.
2. **Token Management:** This layer prevents two parties from attempting the same critical operation at the same time.
3. **Synchronization :** This layer allows a process to add checkpoints which are considered as synchronization points into stream of data. Example: If a system is sending a file of

800 pages, adding checkpoints after every 50 pages is recommended. This ensures that 50 page unit is successfully received and acknowledged. This is beneficial at the time of crash as if a crash happens at page number 110; there is no need to retransmit 1 to100 pages.

OSI Model Layer 6: The Presentation Layer

1. Presentation Layer takes care that the data is sent in such a way that the receiver will understand the information (data) and will be able to use the data.
2. While receiving the data, presentation layer transforms the data to be ready for the application layer.
3. Languages(syntax) can be different of the two communicating systems. Under this condition presentation layer plays a role of translator.
4. It perfroms Data compression, Data encryption, Data conversion etc.

Functions of Presentation Layer

1. **Translation:** Before being transmitted, information in the form of characters and numbers should be changed to bit streams. The presentation layer is responsible for interoperability between encoding methods as different computers use different encoding methods. It translates data between the formats the network requires and the format the computer.
2. **Encryption:** It carries out encryption at the transmitter and decryption at the receiver.
3. **Compression:** It carries out data compression to reduce the bandwidth of the data to be transmitted. The primary role of Data compression is to reduce the number of bits to be transmitted. It is important in transmitting multimedia such as audio, video, text etc.

OSI Model Layer 7: Application Layer

1. Application Layer is the topmost layer.
2. Transferring of files disturbing the results to the user is also done in this layer. Mail services, directory services, network resource etc are services provided by application layer.
3. This layer mainly holds application programs to act upon the received and to be sent data.

Functions of Application Layer

1. **Mail Services:** This layer provides the basis for E-mail forwarding and storage.
2. **Network Virtual Terminal:** It allows a user to log on to a remote host. The application creates software emulation of a terminal at the remote host. User's computer talks to the software terminal which in turn talks to the host and vice versa. Then the remote host believes it is communicating with one of its own terminals and allows user to log on.
3. **Directory Services:** This layer provides access for global information about various services.
4. **File Transfer, Access and Management (FTAM):** It is a standard mechanism to access files and manages it. Users can access files in a remote computer and manage it. They can also retrieve files from a remote computer.

Merits of OSI reference model

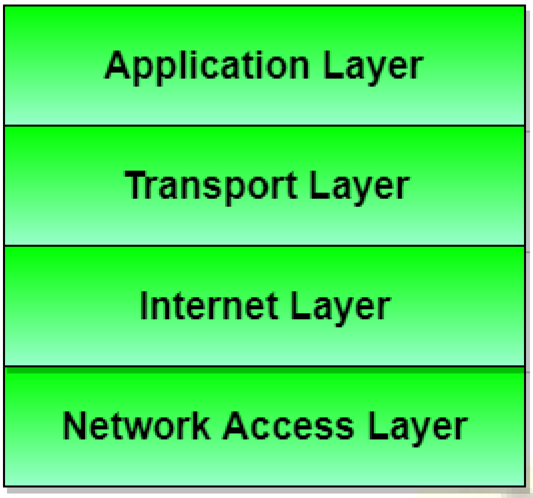
1. OSI model distinguishes well between the services, interfaces and protocols.
2. Protocols of OSI model are very well hidden.
3. Protocols can be replaced by new protocols as technology changes.
4. Supports connection oriented services as well as connectionless service.

Demerits of OSI reference model

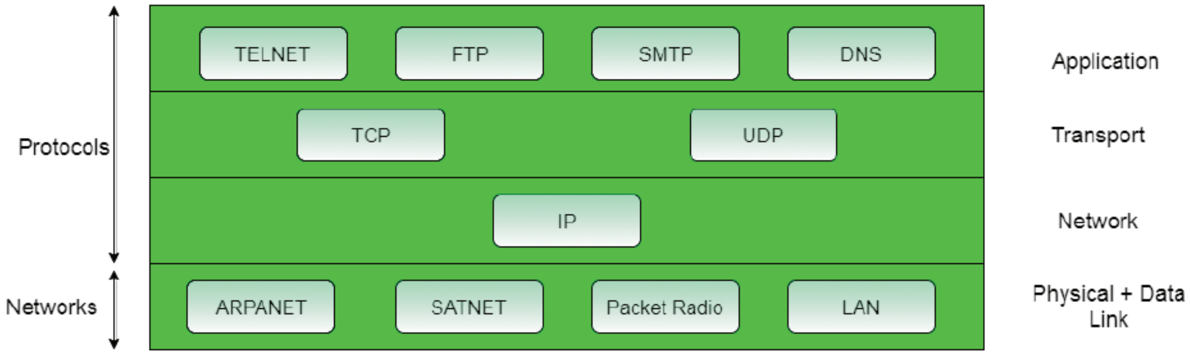
1. Model was devised before the invention of protocols.
2. Fitting of protocols is tedious task.
3. It is just used as a reference model.

1.8 The TCP/IP Reference Model

TCP/IP means Transmission Control Protocol and Internet Protocol. It is the network model used in the current Internet architecture as well. **Protocols** are set of rules which govern every possible communication over a network. These protocols describe the movement of data between the source and destination or the internet. They also offer simple naming and addressing schemes.



Protocols and networks in the TCP/IP model:



Overview of TCP/IP reference model

TCP/IP that is Transmission Control Protocol and Internet Protocol was developed by Department of **Defence's Project Research Agency** (ARPA, later DARPA) as a part of a research project of network interconnection to connect remote machines.

The features that stood out during the research, which led to making the TCP/IP reference model were:

- Support for a flexible architecture. Adding more machines to a network was easy.
- The network was robust, and connections remained intact untill the source and destination machines were functioning.

Different Layers of TCP/IP Reference Model

Below we have discussed the 4 layers that form the TCP/IP reference model:

Layer 1: Host-to-network Layer

1. Lowest layer of the all.
2. Protocol is used to connect to the host, so that the packets can be sent over it.
3. Varies from host to host and network to network.

Layer 2: Internet layer

1. Selection of a packet switching network which is based on a connectionless internetwork layer is called a internet layer.
2. It is the layer which holds the whole architecture together.
3. It helps the packet to travel independently to the destination.
4. Order in which packets are received is different from the way they are sent.
5. IP (Internet Protocol) is used in this layer.
6. The various functions performed by the Internet Layer are:
    - Delivering IP packets
    - Performing routing
    - Avoiding congestion

Layer 3: Transport Layer

1. It decides if data transmission should be on parallel path or single path.
2. Functions such as multiplexing, segmenting or splitting on the data is done by transport layer.
3. The applications can read and write to the transport layer.
4. Transport layer adds header information to the data.
5. Transport layer breaks the message (data) into small units so that they are handled more efficiently by the network layer.
6. Transport layer also arrange the packets to be sent, in sequence.

Layer 4: Application Layer

The TCP/IP specifications described a lot of applications that were at the top of the protocol stack. Some of them were TELNET, FTP, SMTP, DNS etc.

1. **TELNET** is a two-way communication protocol which allows connecting to a remote machine and run applications on it.
2. **FTP**(File Transfer Protocol) is a protocol, that allows File transfer amongst computer users connected over a network. It is reliable, simple and efficient.

3. **SMTP**(Simple Mail Transport Protocol) is a protocol, which is used to transport electronic mail between a source and destination, directed via a route.
4. **DNS**(Domain Name Server) resolves an IP address into a textual address for Hosts connected over a network.
5. It allows peer entities to carry conversation.
6. It defines two end-to-end protocols: TCP and UDP
   - **TCP(Transmission Control Protocol):** It is a reliable connection-oriented protocol which handles byte-stream from source to destination without error and flow control.
   - **UDP(User-Datagram Protocol):** It is an unreliable connection-less protocol that do not want TCPs, sequencing and flow control. Eg: One-shot request-reply kind of service.

Merits of TCP/IP model

1. It operated independently.
2. It is scalable.
3. Client/server architecture.
4. Supports a number of routing protocols.
5. Can be used to establish a connection between two computers.

Demerits of TCP/IP

1. In this, the transport layer does not guarantee delivery of packets.
2. The model cannot be used in any other application.
3. Replacing protocol is not easy.
4. It has not clearly separated its services, interfaces and protocols.

1.9 Example Networks

1.9.1 Internet

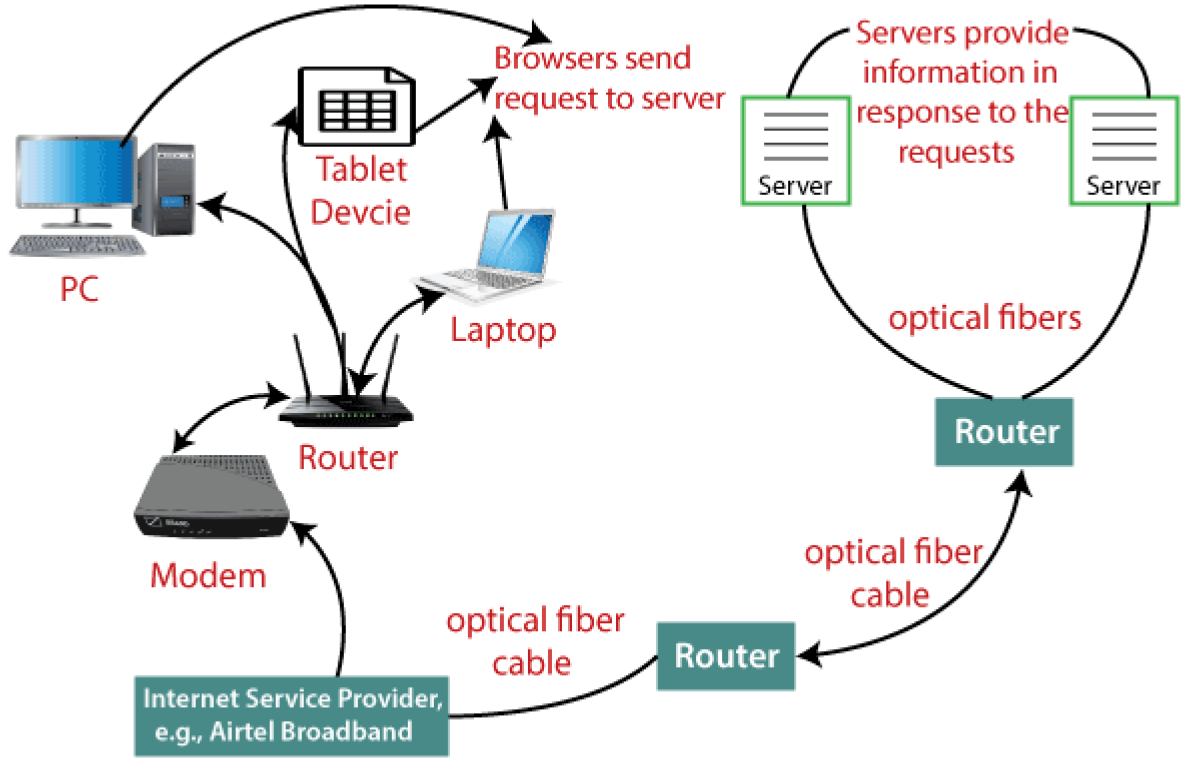
Internet is defined as an Information super Highway, to access information over the web. However, It can be defined in many ways as follows:

- Internet is a world-wide global system of interconnected computer networks.
- Internet uses the standard Internet Protocol (TCP/IP).
- Every computer in internet is identified by a unique IP address.

- IP Address is a unique set of numbers (such as 110.22.33.114) which identifies a computer location.
- A special computer DNS (Domain Name Server) is used to give name to the IP Address so that user can locate a computer by a name.
- Internet is accessible to every user all over the world.

The internet works with the help of clients and servers. A device such as a laptop, which is connected to the internet is called a client, not a server as it is not directly connected to the internet. However, it is indirectly connected to the internet through an Internet Service Provider (ISP) and is identified by an IP address, which is a string of numbers.

A server is a large computer that stores websites. It also has an IP address. A place where a large number of servers are stored is called a data center. The server accepts requests send by the client through a browser over a network (internet) and responds accordingly.



To access the internet we need a domain name, which represents an IP address number, i.e., each IP address has been assigned a domain name.

1.9.2 3G Mobile Phone Network

Third generation mobile phones, or "3G Internet" mobile phones, is a set of standards for wireless mobile communication systems, that promises to deliver quality multimedia services along with high quality voice transmission.

Features

- 3G systems comply with the International Mobile Telecommunications-2000 (IMT-2000)
- specifications by the International Telecommunication Union (ITU).
- The first 3G services were available in 1998.
- It provides high speed transmission having data transfer rate more than 0.2Mbps.
- Global roaming services are available for both voice and data.
- It offers advanced multimedia access like playing music, viewing videos, television services etc.
- It provides access to all advanced Internet services, for example surfing webpages with audio and video.
- It paved the way for the increased usage of smartphones with wide screens as they provided better viewing of mobile webpages, videos and mobile televisions.

Specifications for 3G

3G specifications are laid down by two groups, 3GPP and 3GPP2.

- 3GPP (Third Generation Partnership Project) − These specifications are based upon Global System for Mobile (GSM) communications, and are known as Universal Mobile Telecommunications Systems (UMTS). The technologies includes in it are −
    - Universal Terrestrial Radio Access (UTRA)
    - General Packet Radio Service (GPRS)
    - Enhanced Data rates for GSM Evolution (EDGE)
- 3GPP2 − These specifications are based upon Code Division Multiple Access (CDMA). Two main specifications under this are −
    - Wideband CDMA (WCDMA)
    - CDMA2000

Areas of Application

- Wireless voice telephony
- Fixed wireless Internet access

- Mobile Internet access
- Video calls
- Mobile TV technologies
- Video-on-demand
- Video conferencing
- Tele-medicine
- Global Positioning System (GPS)
- Location-based services

1.9.4 Wireless LANs (WLANs)

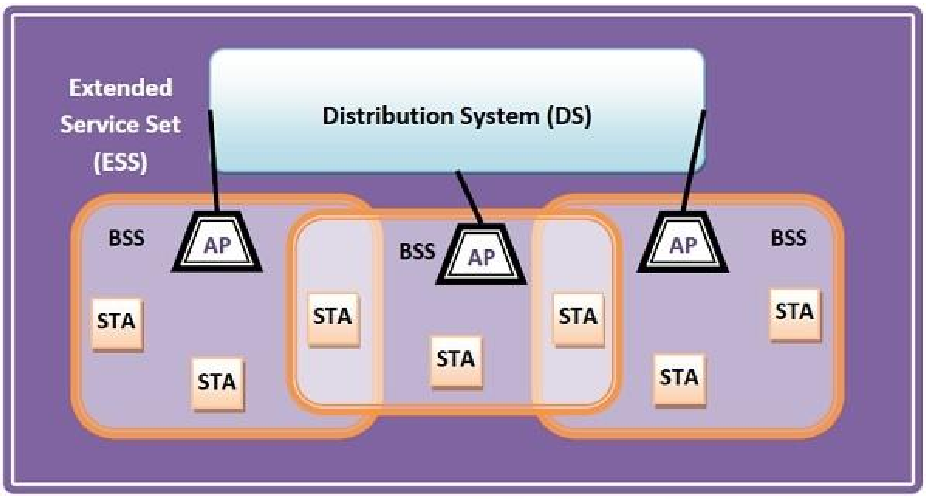
Wireless LANs are wireless computer networks that use high-frequency radio waves instead of cables for connecting the devices within a limited area forming LAN (Local Area Network). Users connected by wireless LANs can move around within this limited area such as home, school, campus, office building, railway platform, etc.

Most WLANs are based upon the standard IEEE 802.11 standard or WiFi.

Components of WLANs

The components of WLAN architecture as laid down in IEEE 802.11 are −

- **Stations (STA)** − Stations comprises of all devices and equipment that are connected to the wireless LAN. Each station has a wireless network interface controller. A station can be of two types −
    - Wireless Access Point (WAP or AP)
    - Client
- **Basic Service Set (BSS)** − A basic service set is a group of stations communicating at the physical layer level. BSS can be of two categories −
    - Infrastructure BSS
    - Independent BSS
- **Extended Service Set (ESS)** − It is a set of all connected BSS.
- **Distribution System (DS)** − It connects access points in ESS.



Types of WLANS

WLANs, as standardized by IEEE 802.11, operates in two basic modes, infrastructure, and ad hoc mode.

- **Infrastructure Mode** − Mobile devices or clients connect to an access point (AP) that in turn connects via a bridge to the LAN or Internet. The client transmits frames to other clients via the AP.
- **Ad Hoc Mode** − Clients transmit frames directly to each other in a peer-to-peer fashion.

Advantages of WLANs

- They provide clutter-free homes, offices and other networked places.
- The LANs are scalable in nature, i.e. devices may be added or removed from the network at greater ease than wired LANs.
- The system is portable within the network coverage. Access to the network is not bounded by the length of the cables.
- Installation and setup are much easier than wired counterparts.
- The equipment and setup costs are reduced.

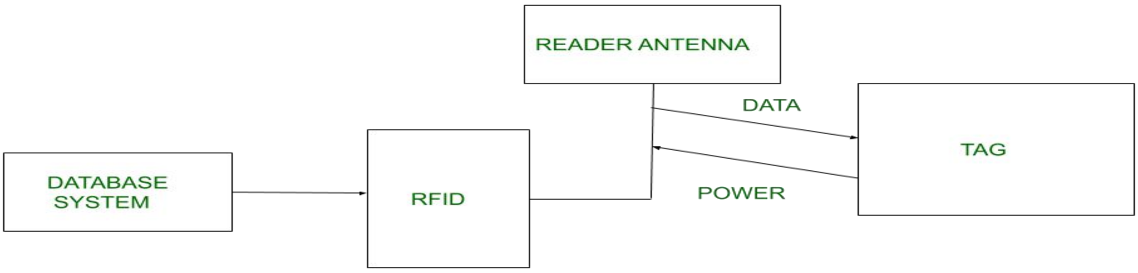
Disadvantages of WLANs

- Since radio waves are used for communications, the signals are noisier with more interference from nearby systems.
- Greater care is needed for encrypting information. Also, they are more prone to errors. So, they require greater bandwidth than the wired LANs.

- WLANs are slower than wired LANs.

1.9.5 Radio Frequency Identification (RFID)

It is a method which is used to track or identify a object by radio transmission uses over the web. Data digitally encoded in RFID tag which might be read by the reader. This is device work as a tag or label during which data read from tags which is stored in database through reader as compared traditional barcodes and QR codes. It is often read outside the road of sight either passive or active RFID.



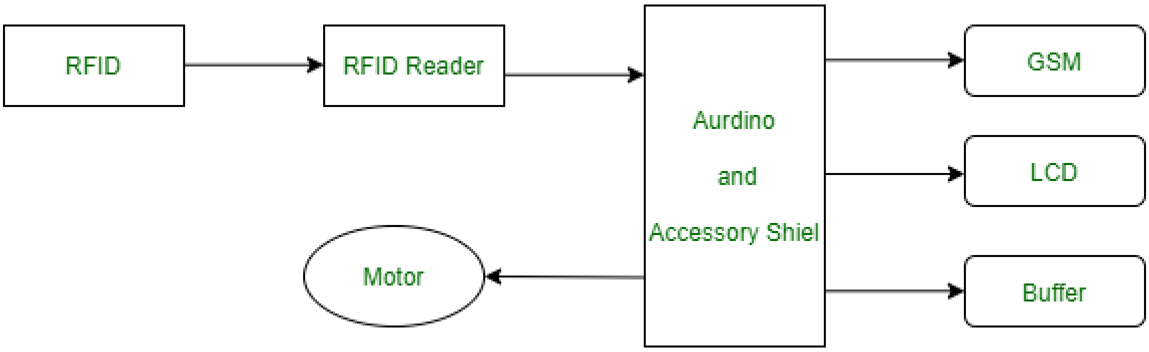
There are two types of RFID :

1. **Passive RFID** –
   In this device, RF tags are not attached by a power supply and passive RF tag stored their power. When it is emitted from active antennas and the RF tag are used specific frequency like 125-134MHZ as low frequency, 13.56MHZ as a high frequency and 856MHZ to 960MHZ as ultra-high frequency.
2. **Active RFID** –
   In this device, RF tags are attached by a power supply that emits a signal and there is an antenna which receives the data.

Working Principle of RFID :

Generally, RFID uses radio waves to perform AIDC function. AIDC stands for Automatic Identification and Data Capture technology which performs object identification and collection and mapping of the data.

An antenna is an device which converts power into radio waves which are used for communication between reader and tag. RFID readers retrieve the information from RFID tag which detects the tag and reads or writes the data into the tag. It may include one processor, package, storage and transmitter and receiver unit.



Features of RFID :

- An RFID tag consists of two-part which is an microcircuit and an antenna.
- This tag is covered by protective material which acts as a shield against the outer environment effect.
- This tag may active or passive in which we mainly and widely used passive RFID.

Application of RFID :

- It utilized in tracking shipping containers, trucks and railroad, cars.
- It uses in Asset tracking.
- It utilized in credit-card shaped for access application.
- It uses in Personnel tracking.
- Controlling access to restricted areas.
- It uses ID badging.
- Supply chain management.
- Counterfeit prevention (e.g., in the pharmaceutical industry).

Advantages of RFID :

- It provides data access and real-time information without taking to much time.
- RFID tags follow the instruction and store a large amount of information.
- The RFID system is non-line of sight nature of the technology.
- It improves the Efficiency, traceability of production.
- In RFID hundred of tags read in a short time.

Disadvantages of RFID :

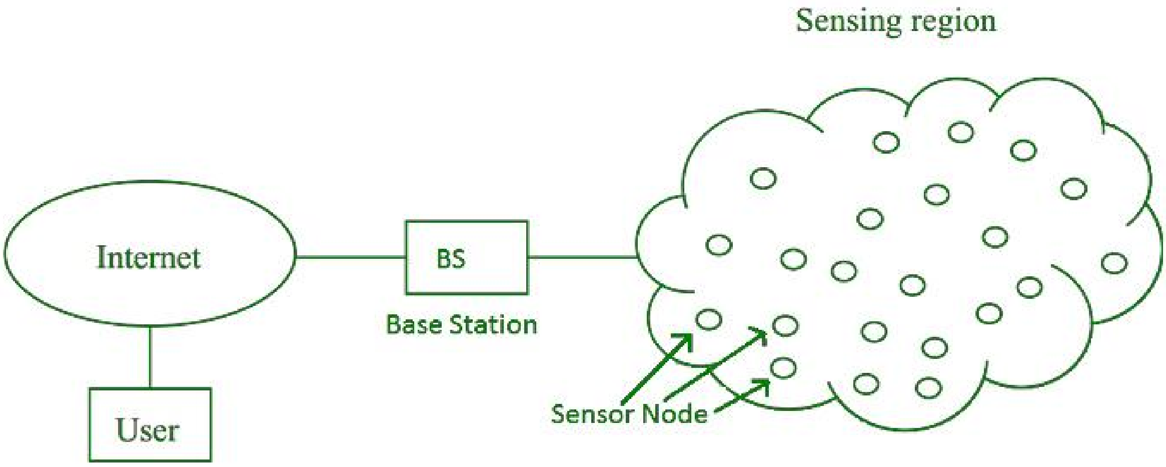
- It takes longer to program RFID Devices.
- RFID intercepted easily even it is Encrypted.
- In an RFID system, there are two or three layers of ordinary household foil to dam the radio wave.

- There is privacy concern about RFID devices anybody can access information about anything.
- Active RFID can costlier due to battery.

1.9.6 Wireless Sensor Network (WSN)

It is an infrastructure-less wireless network that is deployed in a large number of wireless sensors in an ad-hoc manner that is used to monitor the system, physical or environmental conditions.

Sensor nodes are used in WSN with the onboard processor that manages and monitors the environment in a particular area. They are connected to the Base Station which acts as a processing unit in the WSN System. Base Station in a WSN System is connected through the Internet to share data.



WSN can be used for processing, analysis, storage, and mining of the data.

Applications of WSN:

1. Internet of Things (IOT)
2. Surveillance and Monitoring for security, threat detection
3. Environmental temperature, humidity, and air pressure
4. Noise Level of the surrounding
5. Medical applications like patient monitoring
6. Agriculture
7. Landslide Detection

Challenges of WSN:

1. Quality of Service
2. Security Issue
3. Energy Efficiency
4. Network Throughput
5. Performance

6. Ability to cope with node failure
7. Cross layer optimisation
8. Scalability to large scale of deployment

Components of WSN:

1. **Sensors:**

   Sensors in WSN are used to capture the environmental variables and which is used for data acquisition. Sensor signals are converted into electrical signals.

2. **Radio Nodes:**

   It is used to receive the data produced by the Sensors and sends it to the WLAN access point. It consists of a microcontroller, transceiver, external memory, and power source.

3. **WLAN Access Point:**

   It receives the data which is sent by the Radio nodes wirelessly, generally through the internet.

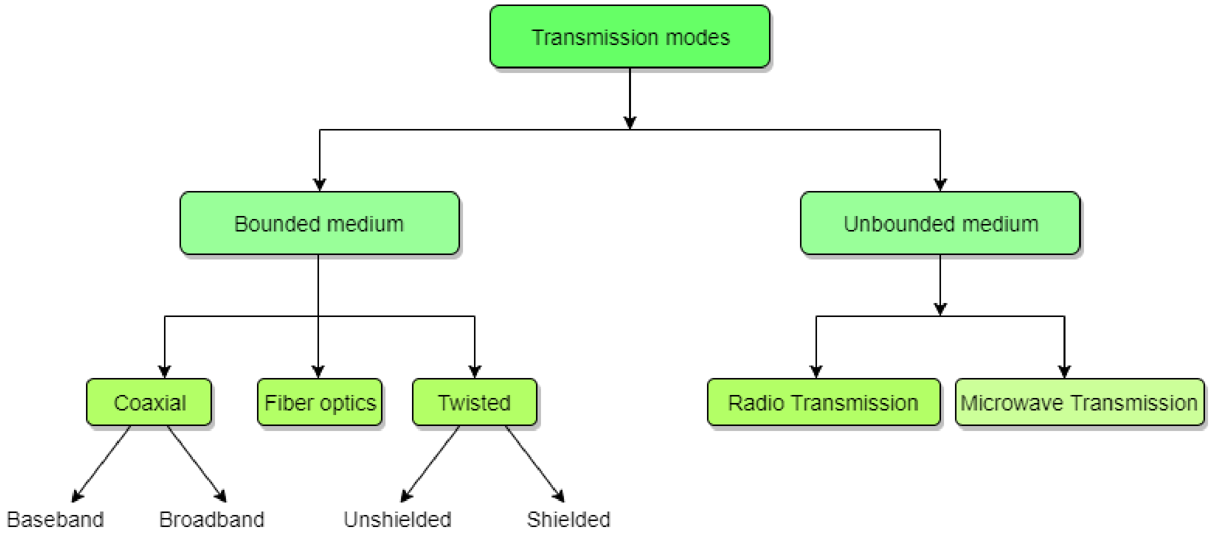
4. **Evaluation Software:**

   The data received by the WLAN Acess Poing is processed by a software called as Evaluation Software for presenting the report to the users for further processing of the data which can be used for processing, analysis, storage, and mining of the data.

1.10 Transmission Mediums in Computer Networks

Data is represented by computers and other telecommunication devices using signals. Signals are transmitted in the form of electromagnetic energy from one device to another. Electromagnetic signals travel through vacuum, air or other transmission mediums to move from one point to another(from sender to receiver).

Electromagnetic energy (includes electrical and magnetic fields) consists of power, voice, visible light, radio waves, ultraviolet light, gamma rays etc.



Factors to be considered while selecting a Transmission Medium

1. Transmission Rate
2. Cost and Ease of Installation
3. Resistance to Environmental Conditions
4. Distances

1.10.1 Bounded or Guided Transmission Media

Guided media, which are those that provide a conduit from one device to another, include **Twisted-Pair Cable**, **Coaxial Cable**, and **Fibre-Optic Cable**.

A signal travelling along any of these media is directed and contained by the physical limits of the medium. Twisted-pair and coaxial cable use metallic (copper) conductors that accept and transport signals in the form of electric current. **Optical fibre** is a cable that accepts and transports signals in the form of light.

1.10.2 Twisted Pair Cable

This cable is the most commonly used and is cheaper than others. It is lightweight, cheap, can be installed easily, and they support many different types of network. Some important points :

- Its frequency range is 0 to 3.5 kHz.
- Typical attenuation is 0.2 dB/Km @ 1kHz.
- Typical delay is 50 µs/km.
- Repeater spacing is 2km.

A twisted pair consists of two conductors(normally copper), each with its own plastic insulation, twisted together. One of these wires is used to carry signals to the receiver, and the other is used only as ground reference. The receiver uses the difference between the two. In addition to the signal sent by the sender on one of the wires, interference(noise) and crosstalk may affect both wires and create unwanted signals. If the two wires are parallel, the effect of these unwanted signals is not the same in both wires because they are at different locations relative to the noise or crosstalk sources. This results in a difference at the receiver.

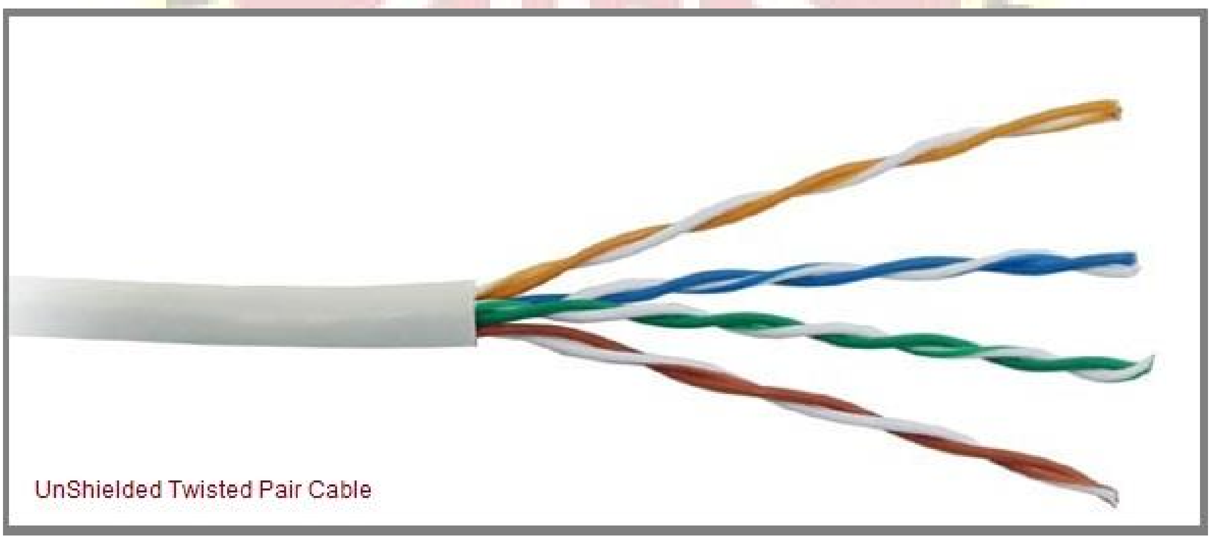
Twisted Pair is of two types:

- **Unshielded Twisted Pair (UTP)**
- **Shielded Twisted Pair (STP)**

1.10.3 Unshielded Twisted Pair Cable

It is the most common type of telecommunication when compared with Shielded Twisted Pair Cable which consists of two conductors usually copper, each with its own colour plastic insulator. Identification is the reason behind coloured plastic insulation.

UTP cables consist of 2 or 4 pairs of twisted cable. Cable with 2 pair use **RJ-11** connector and 4 pair cable use **RJ-45** connector.



Advantages of Unshielded Twisted Pair Cable

- Installation is easy
- Flexible
- Cheap
- It has high speed capacity,
- 100 meter limit

- Higher grades of UTP are used in LAN technologies like Ethernet.

It consists of two insulating copper wires (1mm thick). The wires are twisted together in a helical form to reduce electrical interference from similar pair.

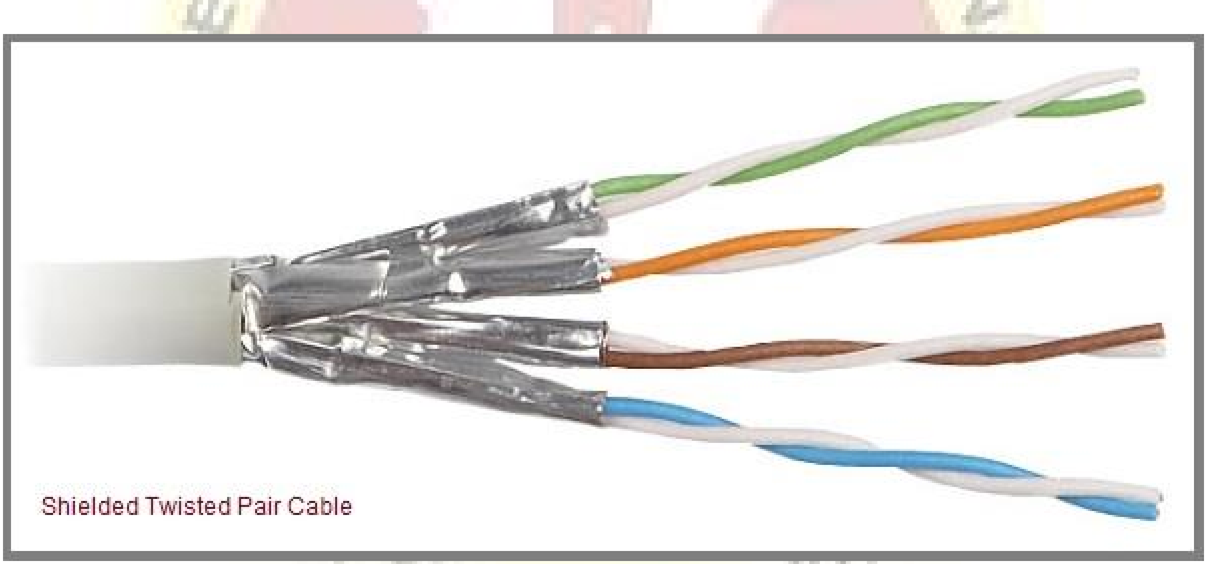
Disadvantages of Unshielded Twisted Pair Cable

- Bandwidth is low when compared with Coaxial Cable
- Provides less protection from interference.

1.10.4 Shielded Twisted Pair Cable

This cable has a metal foil or braided-mesh covering which encases each pair of insulated conductors. Electromagnetic noise penetration is prevented by metal casing. Shielding also eliminates crosstalk .

It has same attenuation as unshielded twisted pair. It is faster the unshielded and coaxial cable. It is more expensive than coaxial and unshielded twisted pair.



Shielded Twisted Pair Cable

Advantages of Shielded Twisted Pair Cable

- Easy to install
- Performance is adequate
- Can be used for Analog or Digital transmission
- Increases the signalling rate
- Higher capacity than unshielded twisted pair
- Eliminates crosstalk

Disadvantages of Shielded Twisted Pair Cable

- Difficult to manufacture
- Heavy

Applications of Shielded Twisted Pair Cable

- In telephone lines to provide voice and data channels. The DSL lines that are used by the telephone companies to provide high-data-rate connections also use the high-bandwidth capability of unshielded twisted-pair cables.
- Local Area Network, such as 10Base-T and 100Base-T, also use twisted-pair cables.

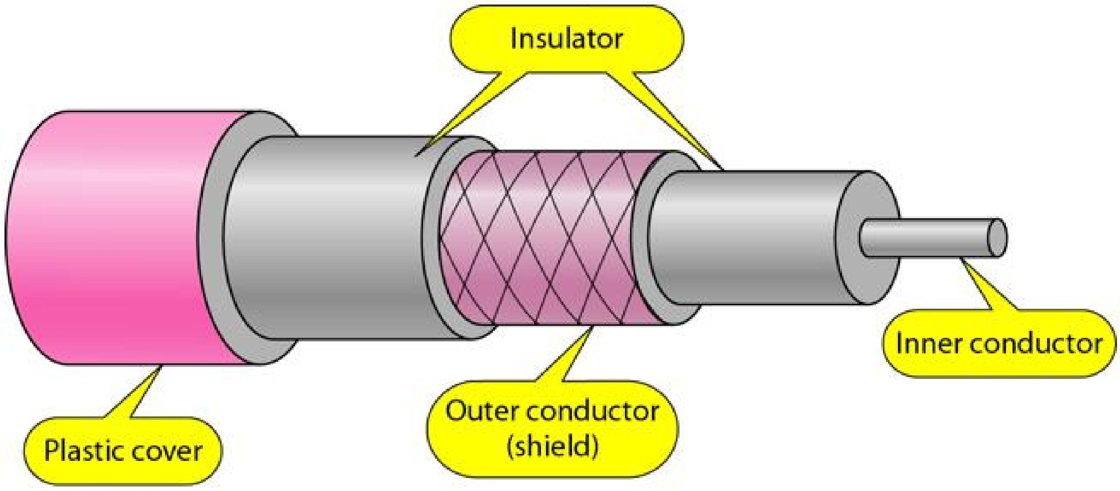
1.10.5 Coaxial Cable

Coaxial is called by this name because it contains two conductors that are parallel to each other. Copper is used in this as centre conductor which can be a solid wire or a standard one. It is surrounded by PVC installation, a sheath which is encased in an outer conductor of metal foil, barid or both.

Outer metallic wrapping is used as a shield against noise and as the second conductor which completes the circuit. The outer conductor is also encased in an insulating sheath. The outermost part is the plastic cover which protects the whole cable.

Here the most common coaxial standards.

- 50-Ohm RG-7 or RG-11 : used with thick Ethernet.
- 50-Ohm RG-58 : used with thin Ethernet
- 75-Ohm RG-59 : used with cable television
- 93-Ohm RG-62 : used with ARCNET.



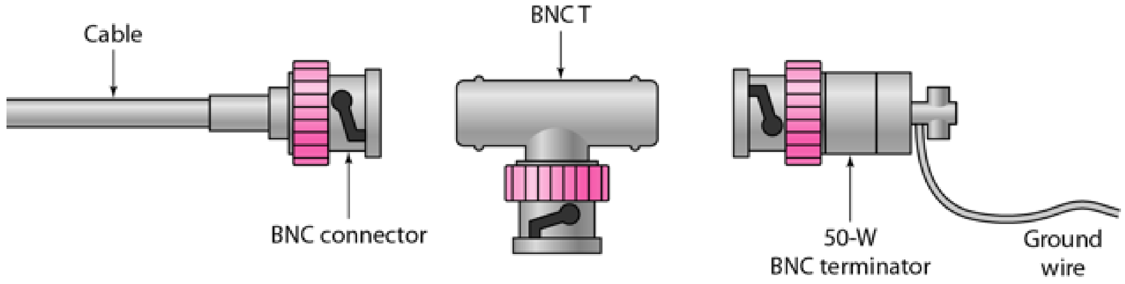
Coaxial Cable Standards

Coaxial cables are categorized by their Radio Government(RG) ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the inner conductor, the thickness and the type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function, as shown in the table below:

Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the Bayonet Neill-Concelman (BNC) connector. The below figure shows 3 popular types of these connectors: the BNC Connector, the BNC T connector and the BNC terminator.



The BNC connector is used to connect the end of the cable to the device, such as a TV set. The BNC T connector is used in Ethernet networks to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

There are two types of Coaxial cables:

1. BaseBand

This is a 50 ohm (Ω) coaxial cable which is used for digital transmission. It is mostly used for LAN's. Baseband transmits a single signal at a time with very high speed. The major drawback is that it needs amplification after every 1000 feet.

2. BroadBand

This uses analog transmission on standard cable television cabling. It transmits several simultaneous signal using different frequencies. It covers large area when compared with Baseband Coaxial Cable.

Advantages of Coaxial Cable

- Bandwidth is high
- Used in long distance telephone lines.
- Transmits digital signals at a very high rate of 10Mbps.
- Much higher noise immunity
- Data transmission without distortion.
- The can span to longer distance at higher speeds as they have better shielding when compared to twisted pair cable

Disadvantages of Coaxial Cable

- Single cable failure can fail the entire network.
- Difficult to install and expensive when compared with twisted pair.
- If the shield is imperfect, it can lead to grounded loop.

Applications of Coaxial Cable

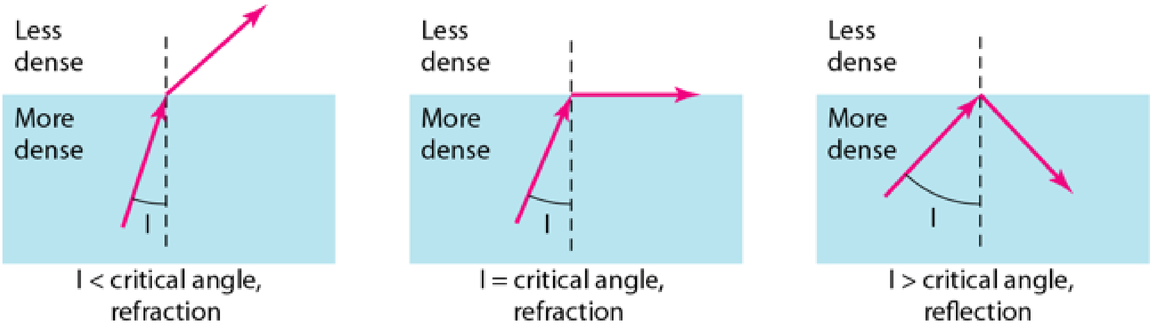
- Coaxial cable was widely used in analog telephone networks, where a single coaxial network could carry 10,000 voice signals.
- Cable TV networks also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Cable TV uses RG-59 coaxial cable.
- In traditional Ethernet LANs. Because of it high bandwidth, and consequence high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The 10Base-2, or Thin Ethernet, uses RG-58 coaxial cable with BNC connectors to transmit data at 10Mbps with a range of 185 m.

1.10.6 Fiber Optic Cable

A fibre-optic cable is made of glass or plastic and transmits signals in the form of light.

Light travels in a straight line as long as it is mobbing through a single uniform substance. If ray of light travelling through one substance suddenly enters another substance (of a different density), the ray changes direction.

The below figure shows how a ray of light changes direction when going from a more dense to a less dense substance.

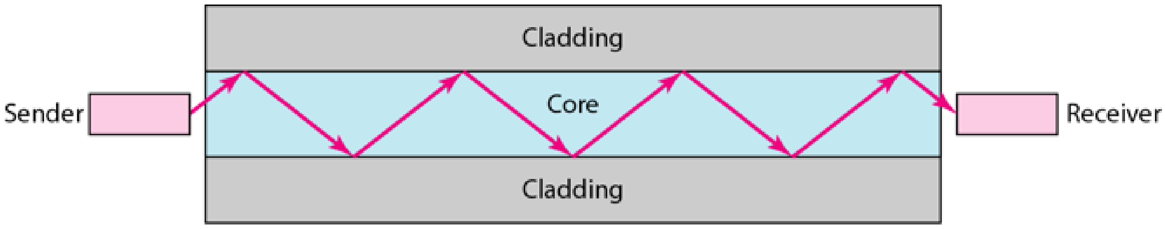

Less dense
More dense
I
I < critical angle, refraction
Less dense
More dense
I
I = critical angle, refraction
Less dense
More dense
I
I > critical angle, reflection

Bending of a light ray

As the figure shows:

- If the **angle of incidence I**(the angle the ray makes with the line perpendicular to the interface between the two substances) is **less** than the **critical angle**, the ray **refracts** and moves closer to the surface.
- If the angle of incidence is **greater** than the critical angle, the ray **reflects**(makes a turn) and travels again in the denser substance.
- If the angle of incidence is **equal** to the critical angle, the ray refracts and **moves parallel** to the surface as shown.

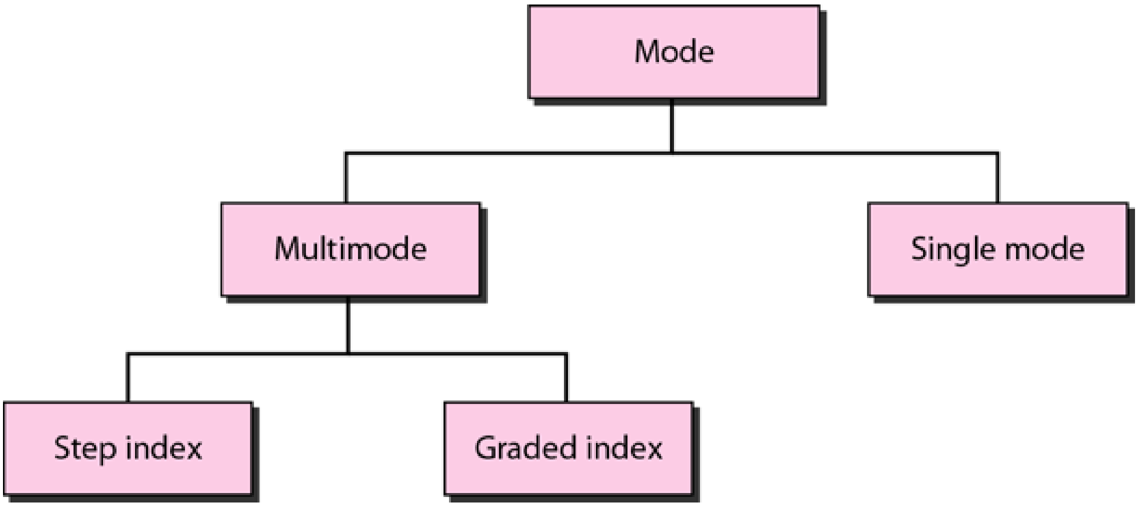
Optical fibres use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it.



Internal view of an Optical fibre

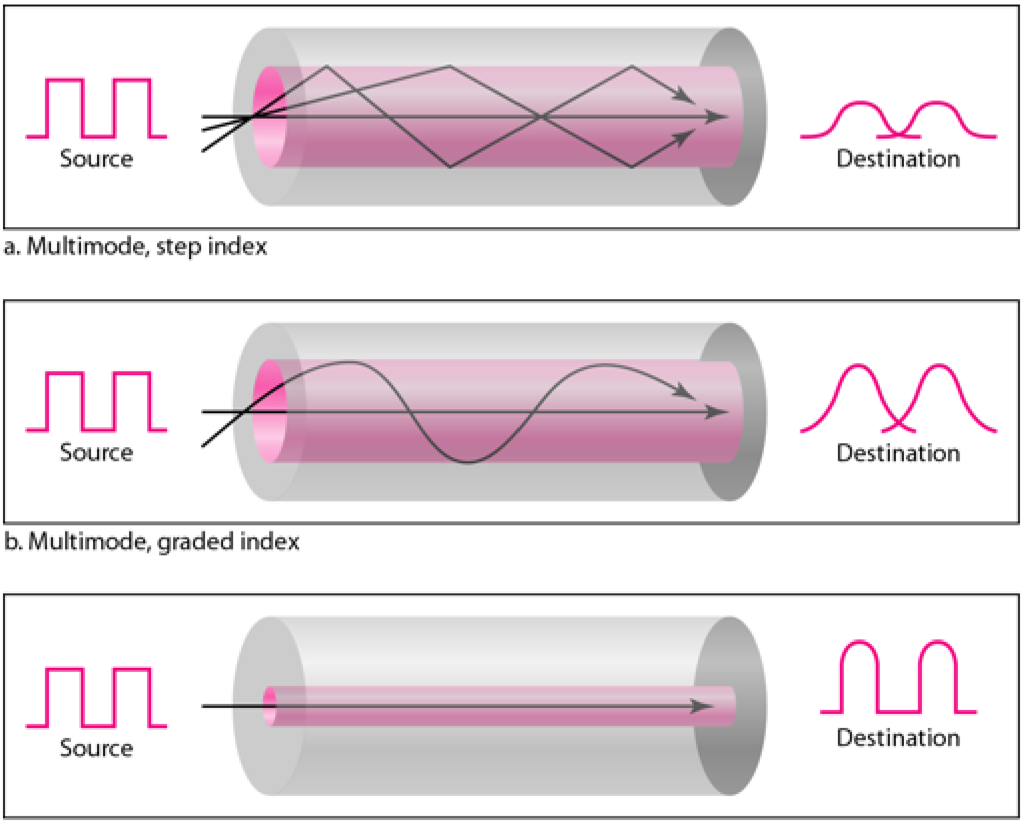
Propagation Modes of Fiber Optic Cable

Current technology supports two modes(**Multimode** and **Single mode**) for propagating light along optical channels, each requiring fibre with different physical characteristics. Multimode can be implemented in two forms: **Step-index** and **Graded-index**.



Multimode Propagation Mode

Multimode is so named because multiple beams from a light source move through the core in different paths. How these beams move within the cable depends on the structure of the core as shown in the below figure.



a. Multimode, step index

b. Multimode, graded index

c. Single mode

- In **multimode step-index fibre**, the density of the core remains constant from the centre to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. The term step-index refers to the suddenness of this change, which contributes to the distortion of the signal as it passes through the fibre.

- In **multimode graded-index fibre**, this distortion gets decreases through the cable. The word index here refers to the index of refraction. This index of refraction is related to the density. A graded-index fibre, therefore, is one with varying densities. Density is highest at the centre of the core and decreases gradually to its lowest at the edge.

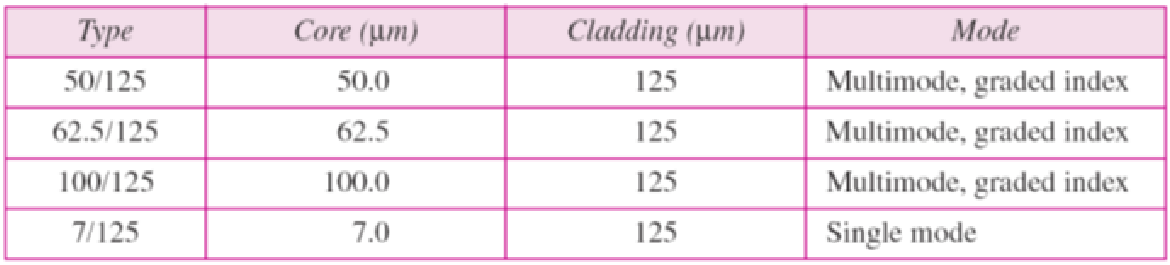
Single Mode

Single mode uses step-index fibre and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal. The single-mode fibre itself is manufactured with a much smaller diameter than that of multimode fibre, and with substantially lower density.

The decrease in density results in a critical angle that is close enough to 90 degree to make the propagation of beams almost horizontal.

Fibre Sizes for Fiber Optic Cable

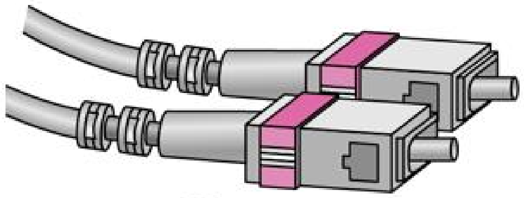
Optical fibres are defined by the ratio of the diameter or their core to the diameter of their cladding, both expressed in micrometers. The common sizes are shown in the figure below:



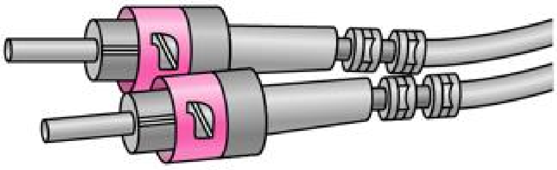
Type	*Core (μm)*	*Cladding (μm)*	*Mode*
50/125	50.0	125	Multimode, graded index
62.5/125	62.5	125	Multimode, graded index
100/125	100.0	125	Multimode, graded index
7/125	7.0	125	Single mode

Fibre Optic Cable Connectors

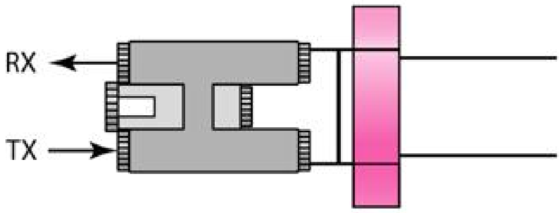
There are three types of connectors for fibre-optic cables, as shown in the figure below.



SC connector



ST connector



MT-RJ connector

The **Subscriber Channel(SC)** connector is used for cable TV. It uses push/pull locking system. The **Straight-Tip(ST)** connector is used for connecting cable to the networking devices. MT-RJ is a connector that is the same size as RJ45.

Advantages of Fibre Optic Cable

Fibre optic has several advantages over metallic cable:

- Higher bandwidth
- Less signal attenuation
- Immunity to electromagnetic interference
- Resistance to corrosive materials
- Light weight
- Greater immunity to tapping

Disadvantages of Fibre Optic Cable

There are some disadvantages in the use of optical fibre:

- Installation and maintenance
- Unidirectional light propagation
- High Cost

Applications of Fibre Optic Cable

- Often found in backbone networks because its wide bandwidth is cost-effective.
- Some cable TV companies use a combination of optical fibre and coaxial cable thus creating a hybrid network.

- Local-area Networks such as 100Base-FX network and 1000Base-X also use fibre-optic cable.

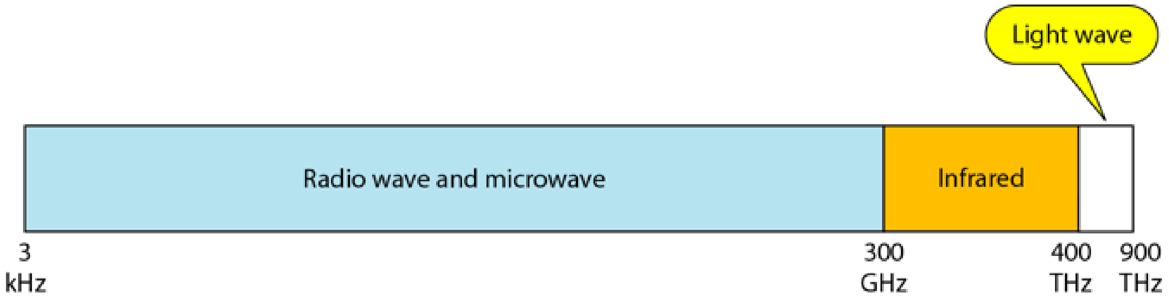
UNIT-II

2.1 Wireless Transmission

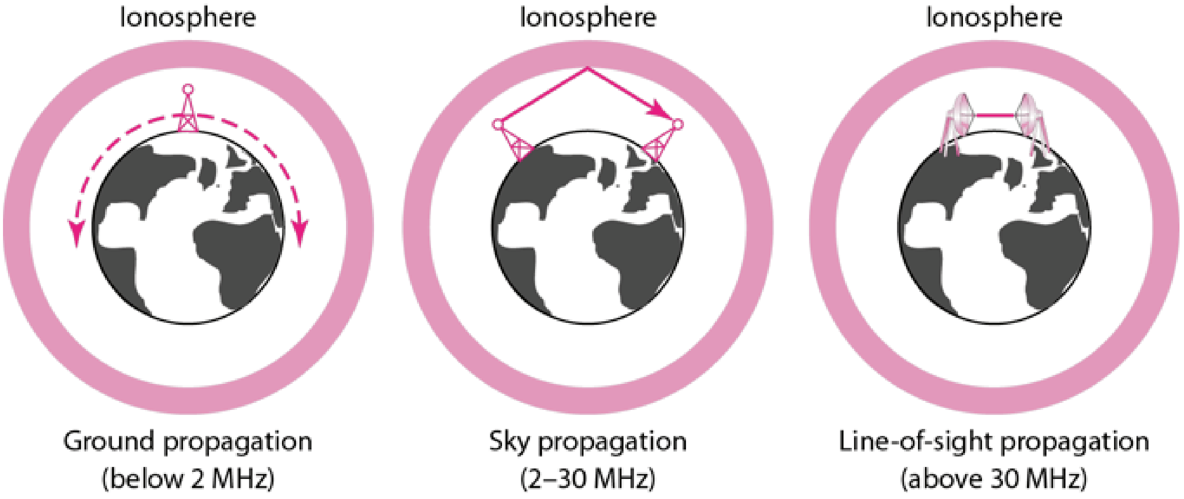
UnBounded or UnGuided Transmission Media

Unguided medium transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication. Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them.

The below figure shows the part of the electromagnetic spectrum, ranging from 3 kHz to 900 THz, used for wireless communication.



Unguided signals can travel from the source to the destination in several ways: **Gound propagation**, **Sky propagation** and **Line-of-sight propagation** as shown in below figure.



Propagation Modes

- **Ground Propagation:** In this, radio waves travel through the lowest portion of the atmosphere, hugging the Earth. These low-frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet.
- **Sky Propagation:** In this, higher-frequency radio waves radiate upward into the ionosphere where they are reflected back to Earth. This type of transmission allows for greater distances with lower output power.
- **Line-of-sight Propagation:** in this type, very high-frequency signals are transmitted in straight lines directly from antenna to antenna.

We can divide wireless transmission into three broad groups:

1. Radio waves
2. Micro waves
3. Infrared waves

2.2 Radio Waves

Electromagnetic waves ranging in frequencies between 3 KHz and 1 GHz are normally called radio waves.

Radio waves are omnidirectional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna send waves that can be received by any receiving antenna. The omnidirectional property has disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signal suing the same frequency or band.

Radio waves, particularly with those of low and medium frequencies, can penetrate walls. This characteristic can be both an advantage and a disadvantage. It is an advantage because, an AM radio can receive signals inside a building. It is a disadvantage because we cannot isolate a communication to just inside or outside a building.

2.2.1 Omnidirectional Antenna for Radio Waves

Radio waves use omnidirectional antennas that send out signals in all directions.

Applications of Radio Waves

- The omnidirectional characteristics of radio waves make them useful for multicasting in which there is one sender but many receivers.
- AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.

2.3 Micro Waves

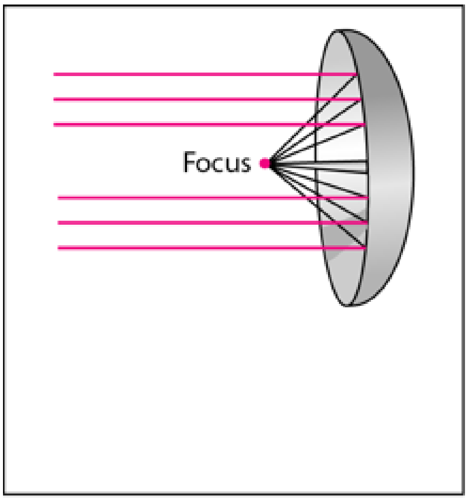
Electromagnetic waves having frequencies between 1 and 300 GHz are called micro waves. Micro waves are unidirectional. When an antenna transmits microwaves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas.

The following describes some characteristics of microwaves propagation:

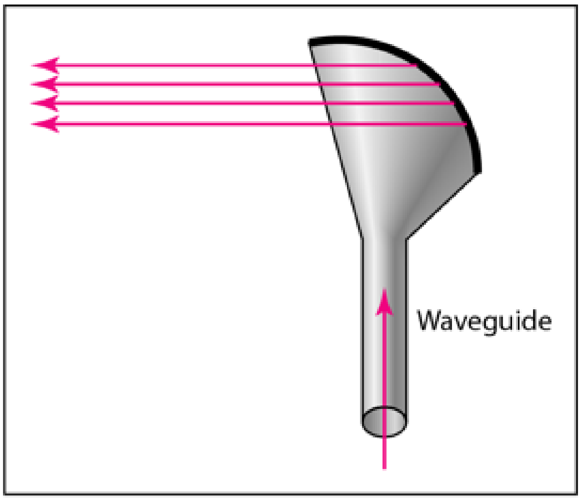
- Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall.
- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside the buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore, wider sub-bands can be assigned and a high date rate is possible.
- Use of certain portions of the band requires permission from authorities.

2.3.1 Unidirectional Antenna for Micro Waves

Microwaves need unidirectional antennas that send out signals in one direction. Two types of antennas are used for microwave communications: **Parabolic Dish** and **Horn**.



a. Dish antenna



b. Horn antenna

A parabolic antenna works as a funnel, catching a wide range of waves and directing them to a common point. In this way, more of the signal is recovered than would be possible with a single-point receiver.

A horn antenna looks like a gigantic scoop. Outgoing transmissions are broadcast up a stem and deflected outward in a series of narrow parallel beams by the curved head. Received transmissions are collected by the scooped shape of the horn, in a manner similar to the parabolic dish, and are deflected down into the stem.

Applications of Micro Waves

Microwaves, due to their unidirectional properties, are very useful when unicast(one-to-one) communication is needed between the sender and the receiver. They are used in cellular phones, satellite networks and wireless LANs.

There are 2 types of Microwave Transmission :

1. Terrestrial Microwave
2. Satellite Microwave

Advantages of Microwave Transmission

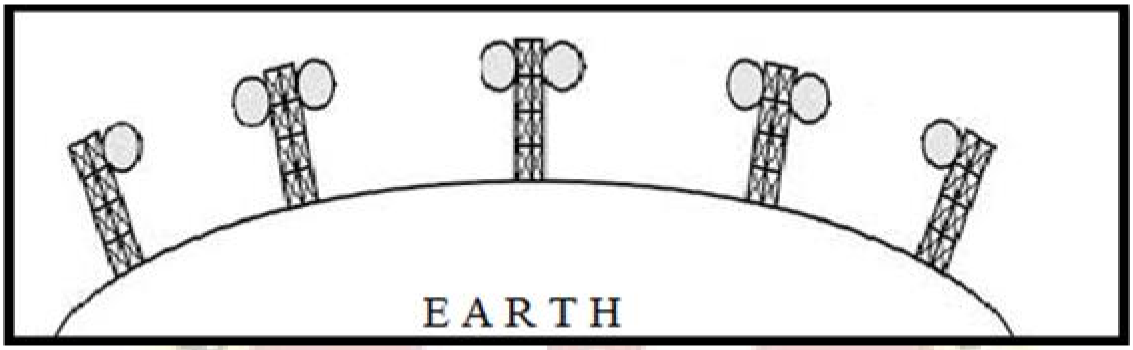
- Used for long distance telephone communication
- Carries 1000's of voice channels at the same time

Disadvantages of Microwave Transmission

- It is very costly

2.4 Terrestrial Microwave

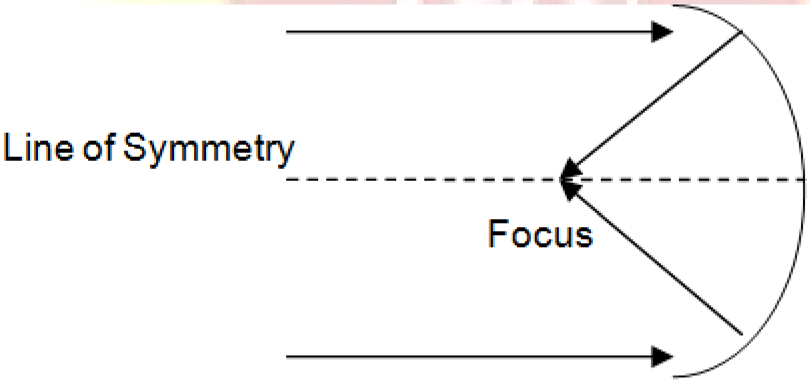
For increasing the distance served by terrestrial microwave, repeaters can be installed with each antenna .The signal received by an antenna can be converted into transmittable form and relayed to next antenna as shown in below figure. It is an example of telephone systems all over the world



There are **two types of antennas** used for terrestrial microwave communication :

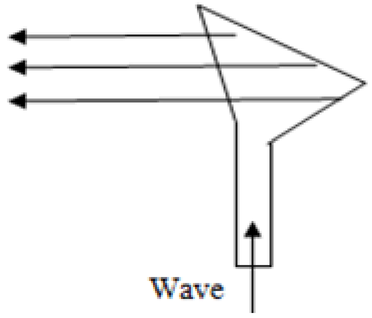
2.4.1. Parabolic Dish Antenna

In this every line parallel to the line of symmetry reflects off the curve at angles in a way that they intersect at a common point called focus. This antenna is based on geometry of parabola.



2.4.2. Horn Antenna

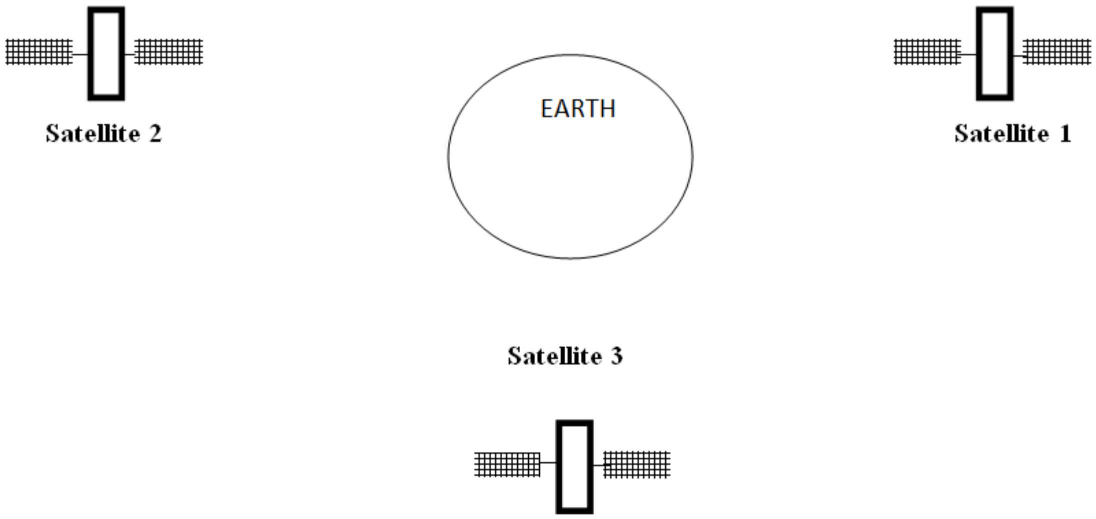
It is a like gigantic scoop. The outgoing transmissions are broadcast up a stem and deflected outward in a series of narrow parallel beams by curved head.



2.5 Satellite Microwave

This is a microwave relay station which is placed in outer space. The satellites are launched either by rockets or space shuttles carry them.

These are positioned 36000 Km above the equator with an orbit speed that exactly matches the rotation speed of the earth. As the satellite is positioned in a geo-synchronous orbit, it is stationery relative to earth and always stays over the same point on the ground. This is usually done to allow ground stations to aim antenna at a fixed point in the sky.



Features of Satellite Microwave

- Bandwidth capacity depends on the frequency used.
- Satellite microwave deployment for orbiting satellite is difficult.

Advantages of Satellite Microwave

- Transmitting station can receive back its own transmission and check whether the satellite has transmitted information correctly.
- A single microwave relay station which is visible from any point.

Disadvantages of Satellite Microwave

- Satellite manufacturing cost is very high
- Cost of launching satellite is very expensive
- Transmission highly depends on whether conditions, it can go down in bad weather

2.6 Infrared Waves

Infrared waves, with frequencies from 300 GHz to 400 THz, can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another, a short-range communication system in on room cannot be affected by another system in the next room.

When we use infrared remote control, we do not interfere with the use of the remote by our neighbours. However, this same characteristic makes infrared signals useless for long-range communication. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications of Infrared Waves

- The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate.
- The Infrared Data Association(IrDA), an association for sponsoring the use of infrared waves, has established standards for using these signals for communication between devices such as keyboards, mouse, PCs and printers.
- Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

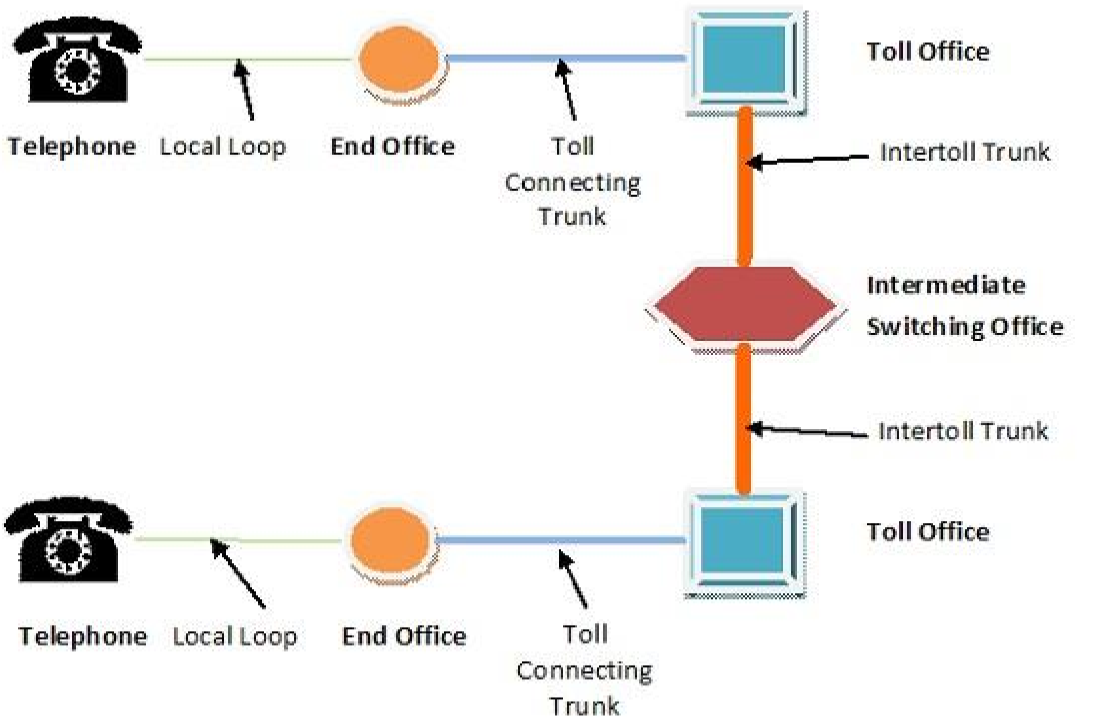
2.7 Telephone Network Structure

The telephone system model is organized as a highly redundant, multilevel hierarchy. It comprises of the following components −

- Telephone of the subscriber or end user
- End office − Local central office directly connected to end user at a distance of 1 – 10km.
- Local loop − A two-way connection between the telephone and the end office.
- Toll office − switching centres which are called tandem offices when located within the same local area.
- Toll connecting trunk − Lines that connect end offices with toll offices.

- Intermediate switching offices – Interconnected non-hierarchical switching offices for connecting toll offices.
- Inter toll trunk – Very high bandwidth channels that connect either two toll offices via intermediate switching offices.

The model can be diagrammatically represented as follows –



Communication for different Transmissions

- Both caller and callee are attached to same end office –

  In this case, a direct electrical connection is set up between the local loops of the subscribers by the switching mechanism of the end office.

- Caller and callee are attached to different end offices –

  In this case, the end office of the caller sets up a connection with one or more connected toll offices, which then performs the switching job. This again has two cases –

  - If the end offices of the caller and the callee have a common toll office, then the toll office establishes a connection within itself.

- If there are no common toll office between the caller and callee, then a path is established between the different toll offices, through intermediate switching office via intertoll trunks.

Transmission Media Used

- Local loop − analog twisted pair cables.
- Toll connecting trunk − fibre optic links
- Between switching offices − fibre optic cables and microwaves

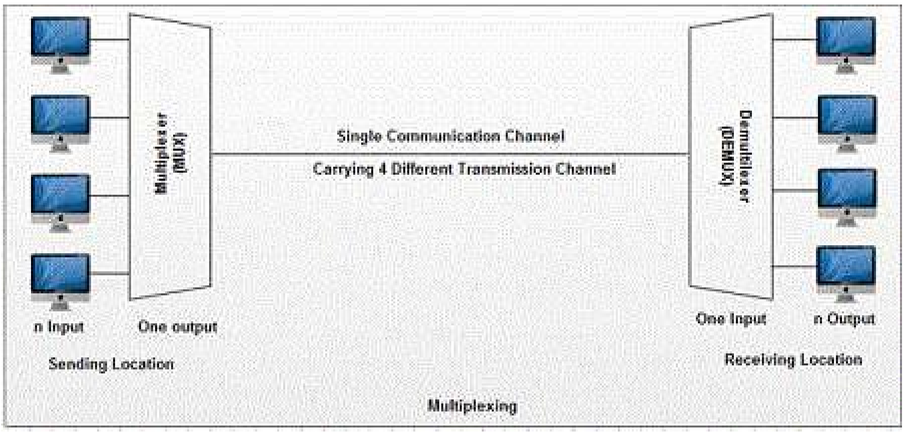
2.8 Multiplexing

Multiplexing is done by using a device called multiplexer (**MUX**) that combines n input lines to generate one output line i.e. (many to one). Therefore multiplexer (**MUX**) has several inputs and one output.

At the receiving end, a device called demultiplexer (**DEMUX**) or (*demuxing*) is used that separates signal into its component signals. So DEMUX has one input and several outputs.

Concept of Multiplexing

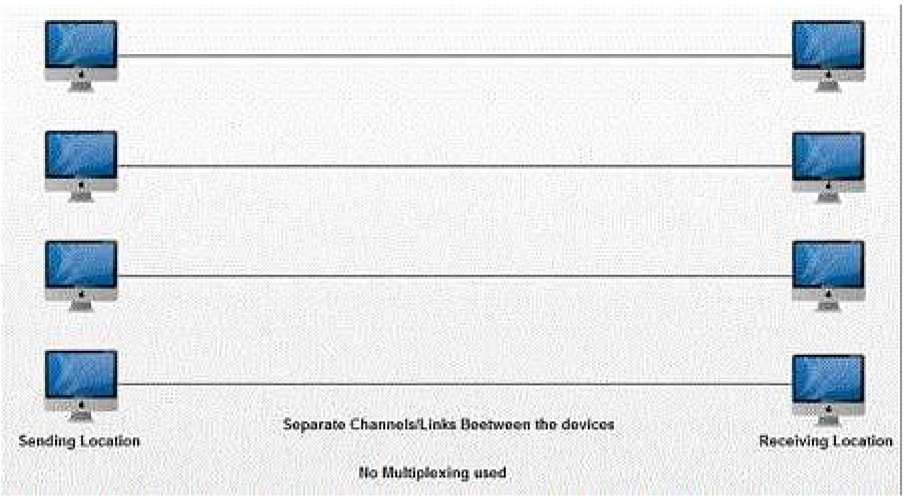
As shown in fig multiplexer takes 4 input lines and diverts them to single output line. The signal from 4 different devices is combined and carried by this single line. At the receiving side, a demultiplexer takes this signal from a single line & breaks it into the original signals and passes them to the 4 different receivers.



Advantages of Multiplexing

If no multiplexing is used between the users at two different sites that are distance apart, then separate communication lines would be required as shown in fig.

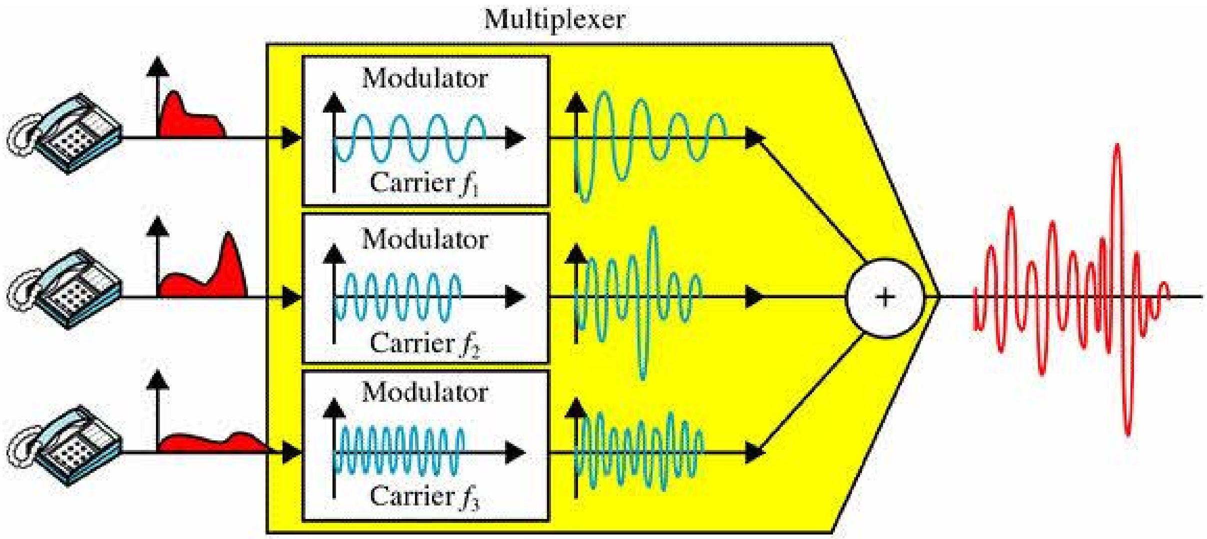
This is not only costly but also become difficult to manage. If multiplexing is used then, only one line is required. This leads to the reduction in the line cost and also it would be easier to keep track of one line than several lines. Multiplexing efficient for utilization of bandwidth.



The following are several examples of different multiplexing methods:

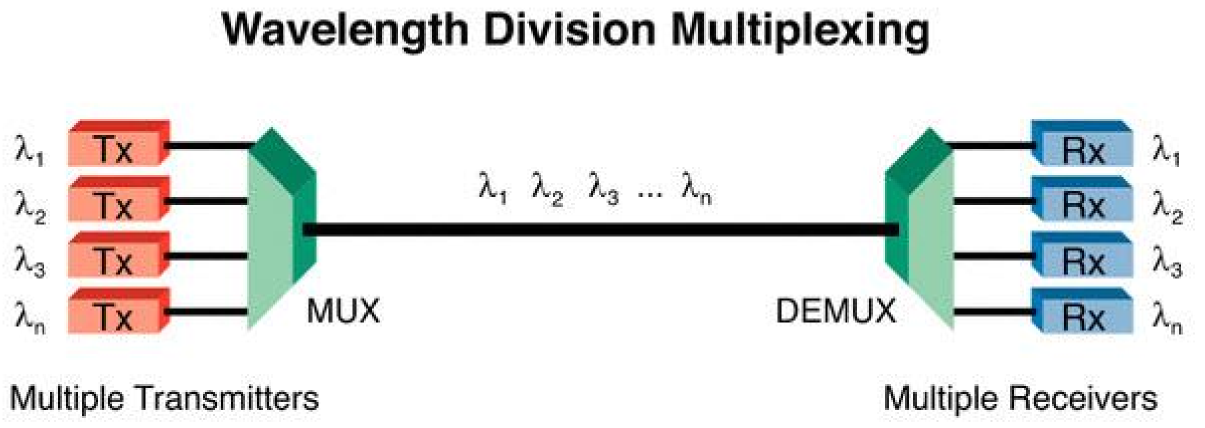
2.8.1 Frequency Division Multiplexing (FDM) –

It is used in analog signal, a multiplexing technique that uses different frequencies to combine multiple streams of data for transmission over a communications medium. FDM assigns a different carrier frequency to each data stream and then combines many modulated carrier frequencies for transmission. For example, television transmitters use **FDM to broadcast several channels at once**. **Band pass filter** is used for separating channels and allows to pass a specific range of frequencies, During transmission of streams it blocks lower and higher frequencies. In Frequency Division Multiplexing, channels are separated by unused strips of guard bands to prevent Overlapping. Guard bands increase the bandwidth for FDM.



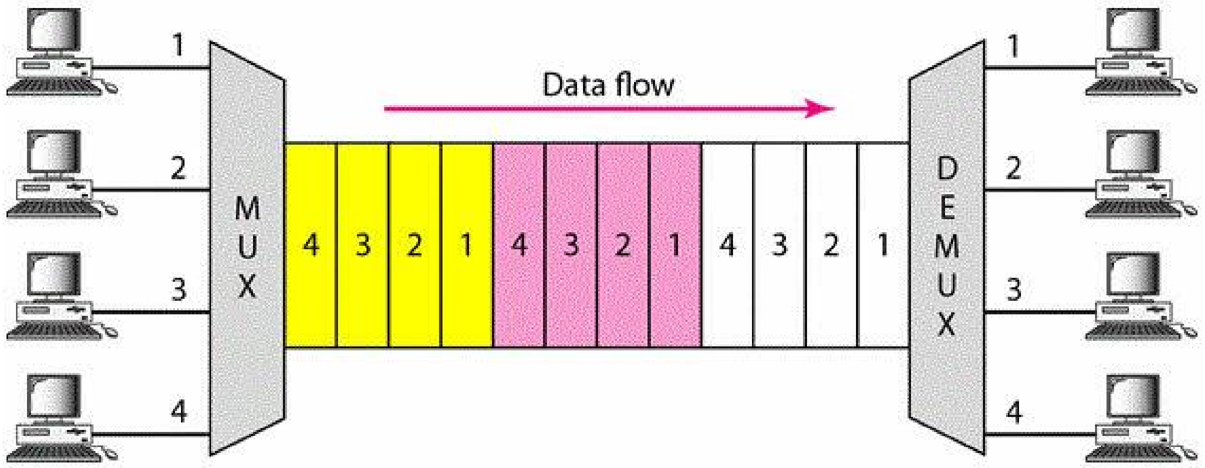
2.8.2 Wavelength Division Multiplexing (WDM)

It is used in analog signal, a type of multiplexing developed for use on optical fiber. WDM modulates each of several data streams onto a different part of the **light spectrum**. **WDM is the optical equivalent of FDM**.



2.8.3Time Division Multiplexing (TDM)

A type of multiplexing that combines data streams by assigning each stream a different time slot in a set. TDM is designed for **digital signals,** which combining several low-rate channels into high-rate one. TDM repeatedly transmits a fixed sequence of time slots over a single transmission channel. Within T-Carrier systems, such as T-1 and T-3, TDM combines **Pulse Code Modulated (PCM)**streams created for each conversation or data stream. TDM, slots are further divided into Frames. In order to separate channels **AND gates** are used in a TDM receiver.



Types of TDM

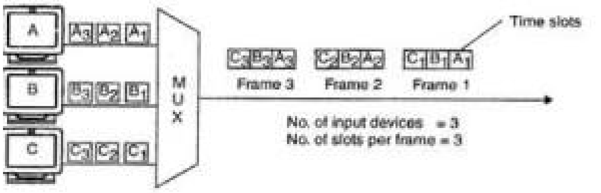
Time division multiplexing is classifieds into four types:

- Synchronous time-division multiplexing
- Asynchronous time-division multiplexing

- Interleaving time-division multiplexing
- Statistical time-division multiplexing

Synchronous Time Division Multiplexing

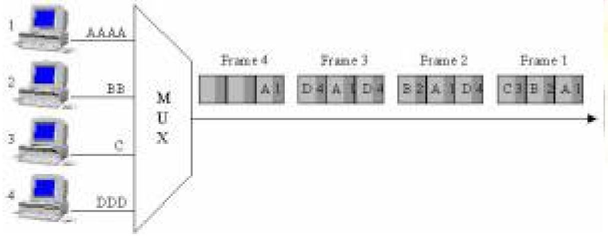
Synchronous time division multiplexing can be used for both analog and digital signals. In synchronous TDM, the connection of input is connected to a frame. If there are ‘n’ connections, then a frame is divided into ‘n’ time slots – and, for each unit, one slot is allocated – one for each input line. In this synchronous TDM sampling, the rate is same for all the signals, and this sampling requires a common clock signal at both the sender and receiver end. In synchronous TDM, the multiplexer allocates the same slot to each device at all times.



Synchronous Time Division Multiplexing

Asynchronous Time-Division Multiplexing

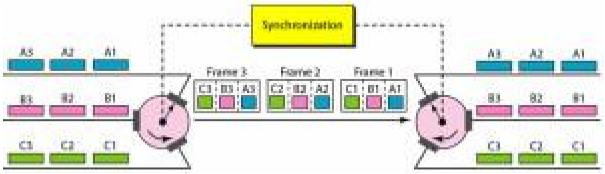
In asynchronous time-division multiplexing, the sampling rate is different for different signals, and it doesn’t require a common clock. If the devices have nothing to transmit, then their time slot is allocated to another device. Designing of a commutator or de-commutator is difficult and the bandwidth is less for time-division multiplexing. This type of time-division multiplexing is used in asynchronous transfer mode networks.



Asynchronous Time-Division Multiplexing

Interleaving

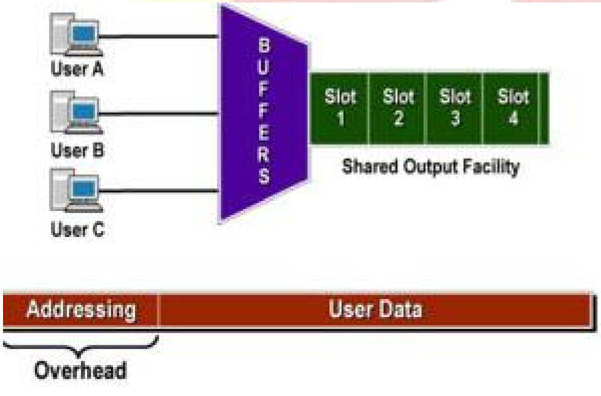
Time-division multiplexing can be visualized as two fast rotating switches on the multiplexing and demultiplexing side. At the same speed these switches rotate and synchronize, but in opposite directions. When the switch opens at the multiplexer side in front of a connection, it has the opportunity to send a unit into the path. In the same way, when the switch opens on the demultiplexer side in front of a connection that has the opportunity to receive a unit from the path. This process is called interleaving.



Interleaving

Statistical Time-Division Multiplexing

Statistical time-division multiplexing is used to transmit several types of data concurrently across a single transmission cable. This is often used for managing data being transmitted via LAN or WAN. The data is simultaneously transmitted from the input devices that are connected to the network including printers, fax machines, and computers. This type of multiplexing is also used in telephone switch board settings to manage the calls. Statistical TDM is similar to dynamic bandwidth allocation, an in this type of time-division multiplexing, a communication channel is divided into an arbitrary number of data streams.



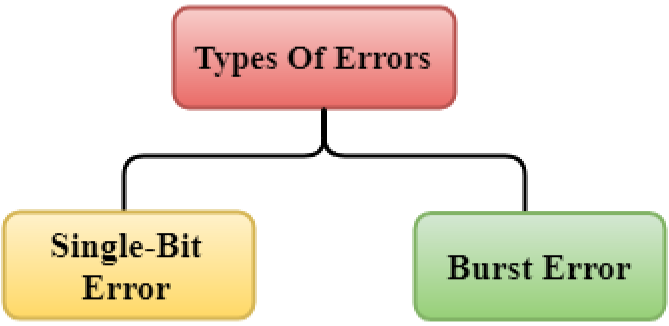
Statistical Time-Division Multiplexing

These are the different types of multiplexing techniques used in communication system for efficient transferring and receiving of the data.

2.9 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

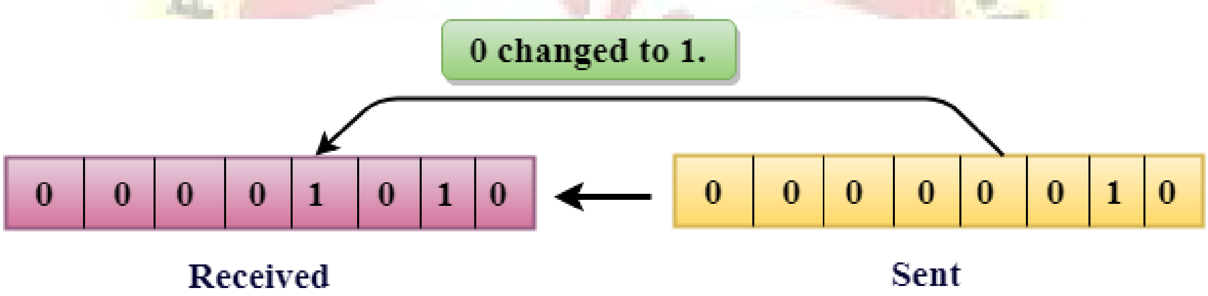


Errors can be classified into two categories:

- 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.

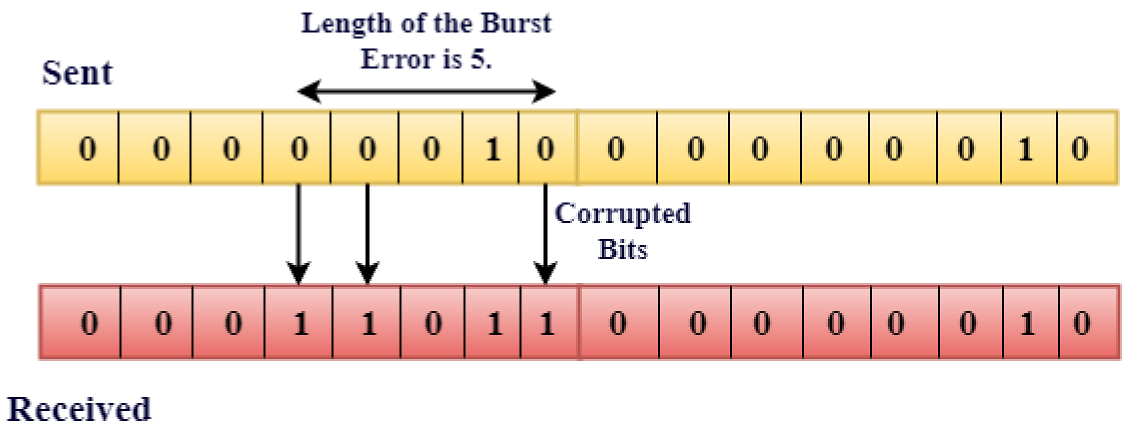


In the above figure, the message which is sent is corrupted as single-bit, i.e., 0 bit is changed to 1.

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.

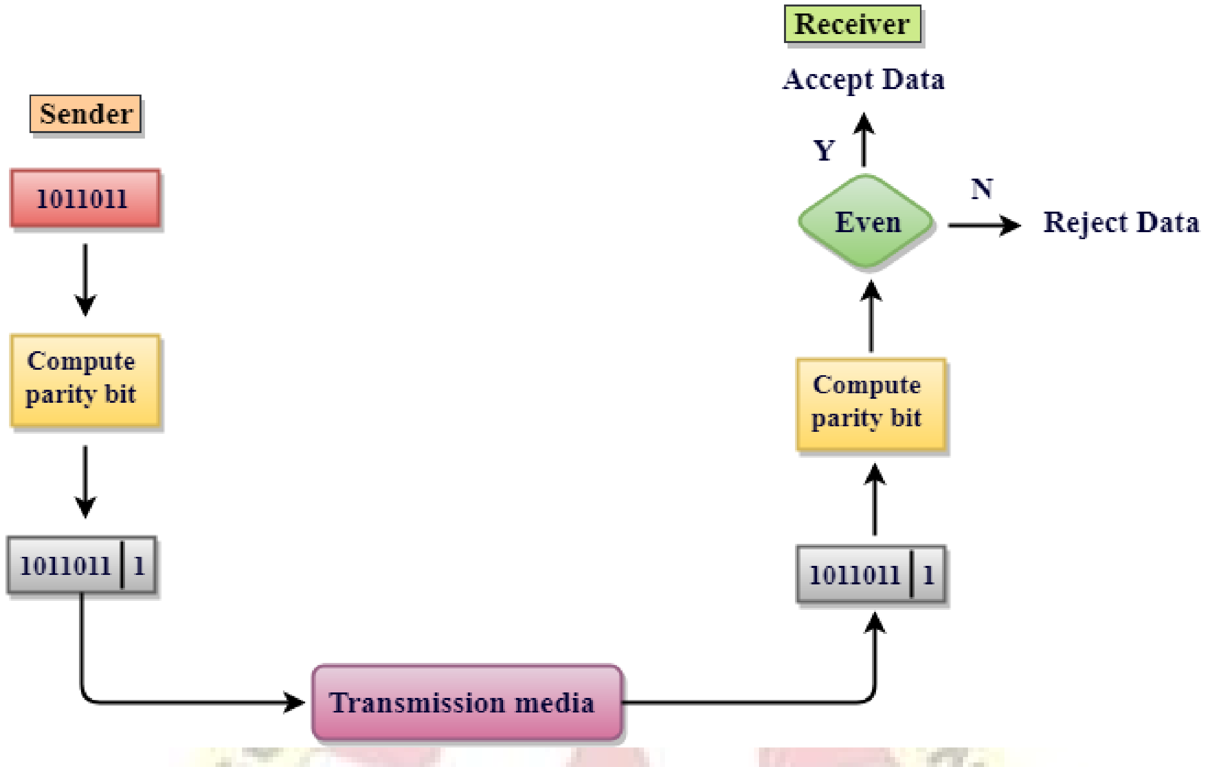
2.10 Error Detecting Techniques:

The most popular Error Detecting Techniques are:

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

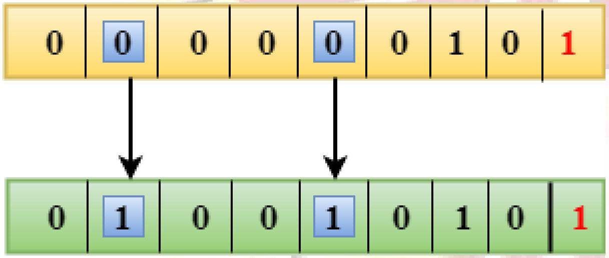
2.10.1 Single Parity Check

- 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.



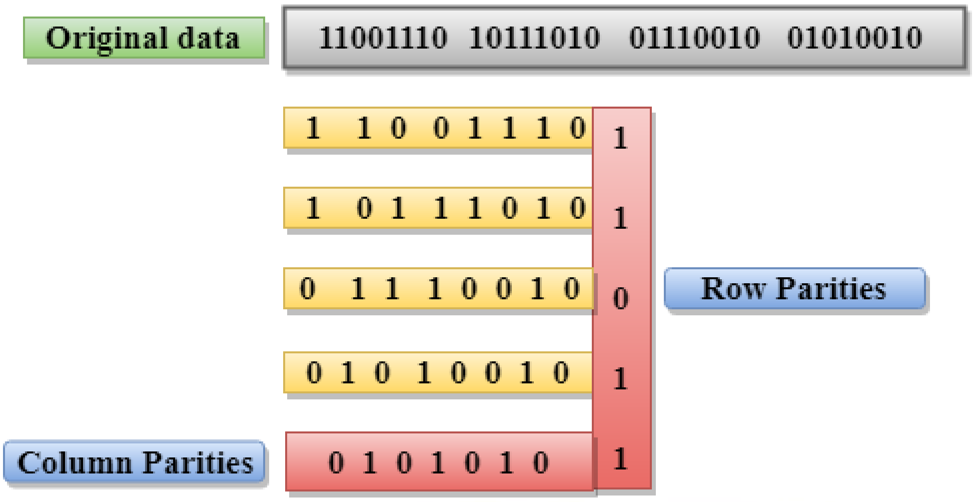
Drawbacks of Single Parity Checking

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



2.10.2 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.

2.10.3 Checksum

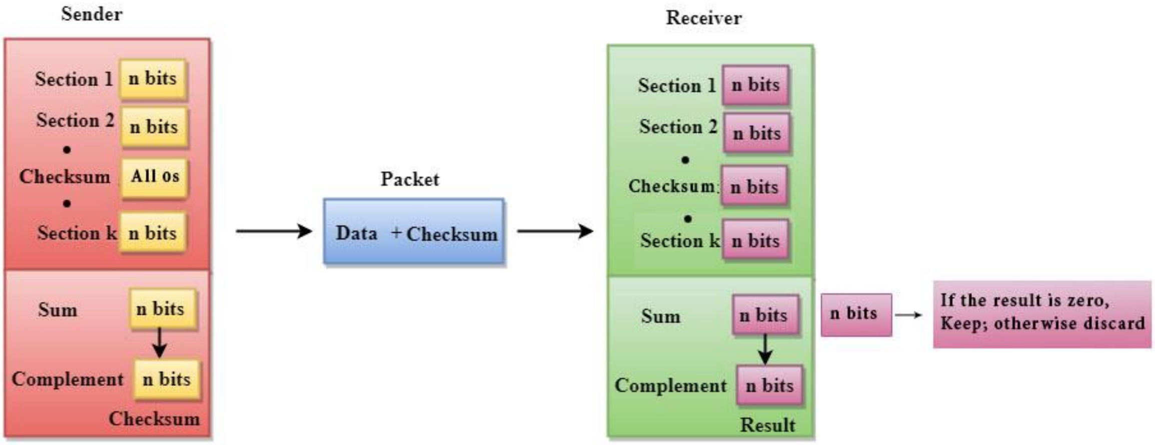
A Checksum is an error detection technique based on the concept of redundancy.

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 discard ed.

2.10.4 Cyclic Redundancy Check (CRC)

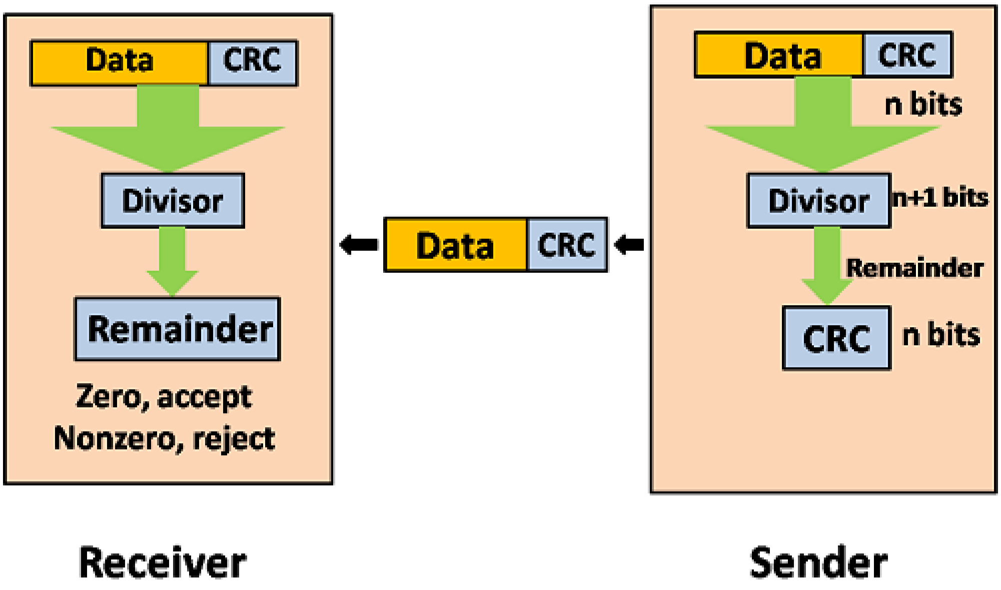
CRC is a redundancy error technique used to determine the error.

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.

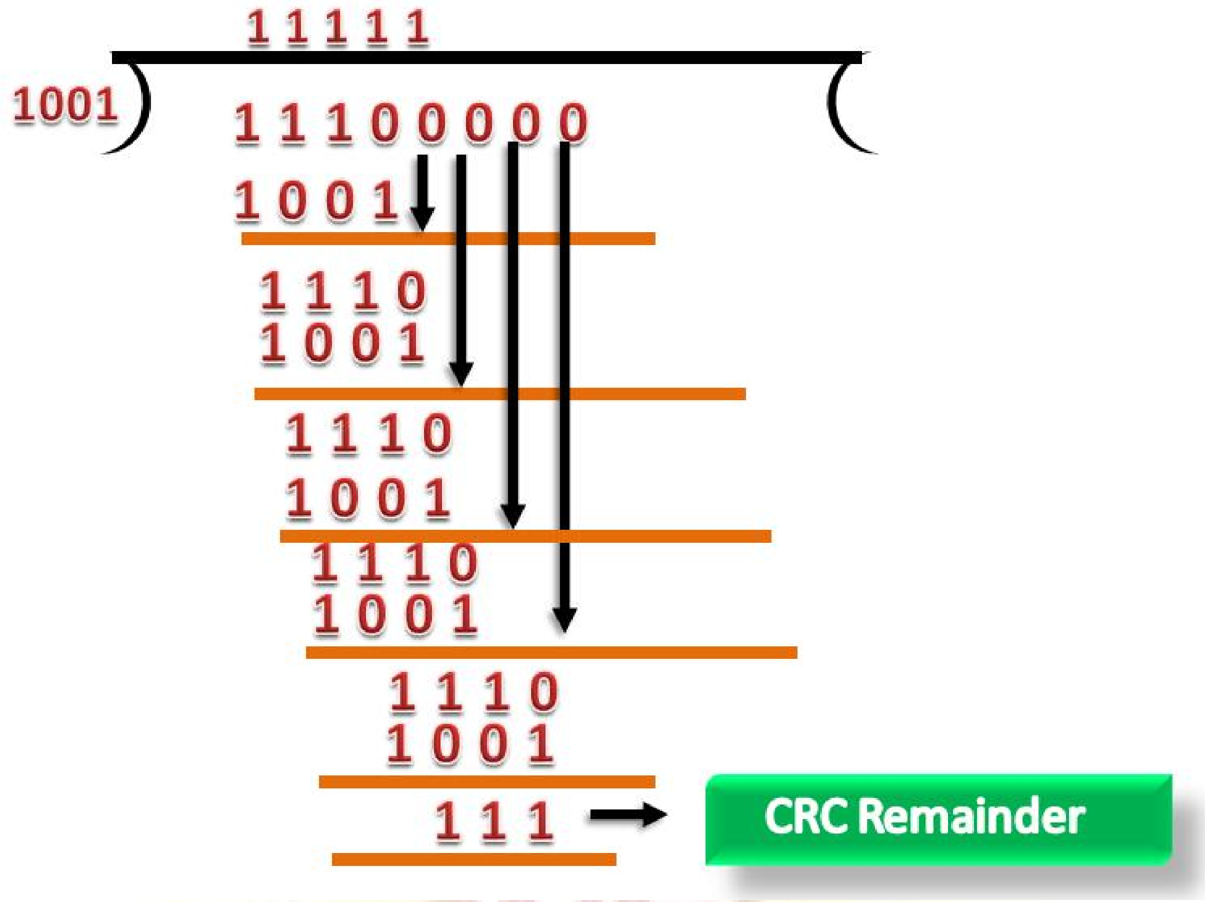


Suppose the original data is 11100 and divisor is 1001.

CRC Generator

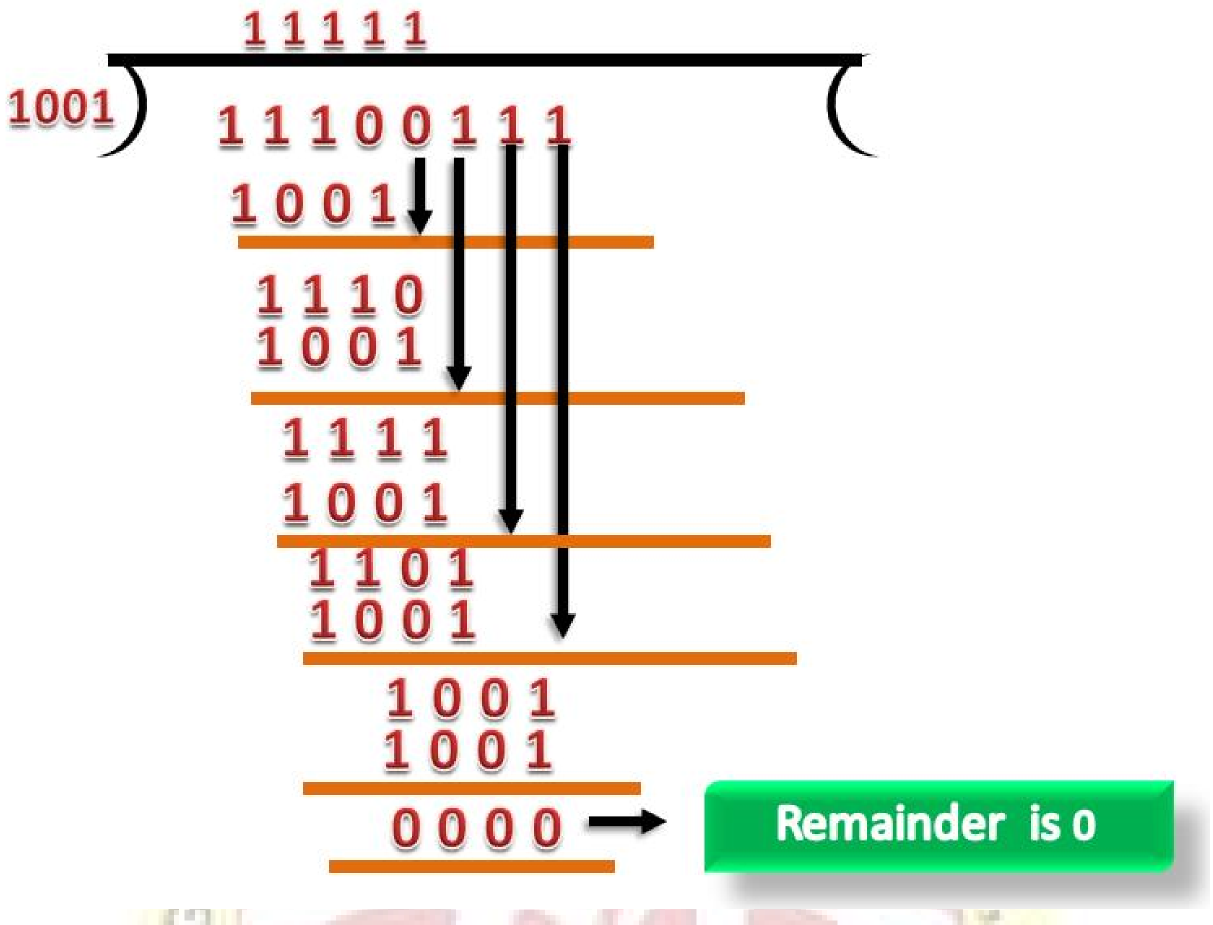
- A CRC generator uses a modulo-2 division. Firstly, three zeroes are appended at the end of the data as the length of the divisor is 4 and we know that the length of the string 0s to be appended is always one less than the length of the divisor.

- 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.



2.11 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.

2.11.1 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.

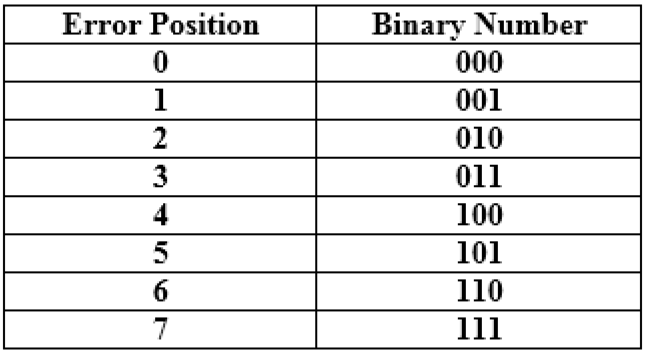
Even parity: To check for even parity, if the total number of 1s is even, then the value of the parity bit is 0. If the total number of 1s occurrences is odd, then the value of the parity bit is 1.

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

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.

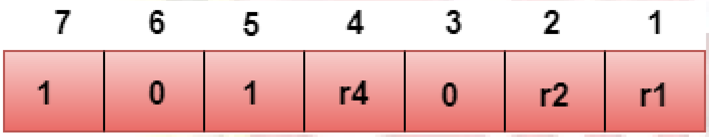
Total number of bits = d+r = 4+3 = 7;

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:

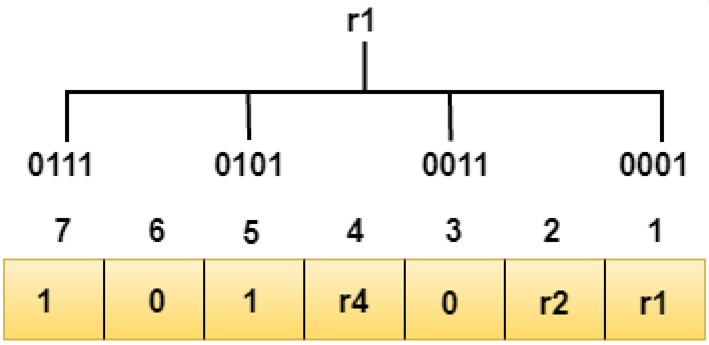


7	6	5	4	3	2	1
1	0	1	r4	0	r2	r1

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.

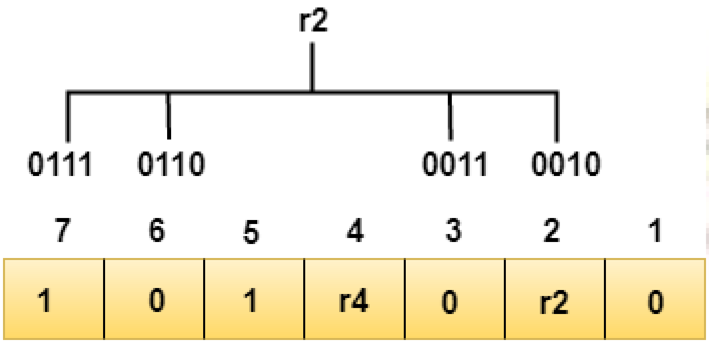


7	6	5	4	3	2	1
1	0	1	r4	0	r2	r1

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

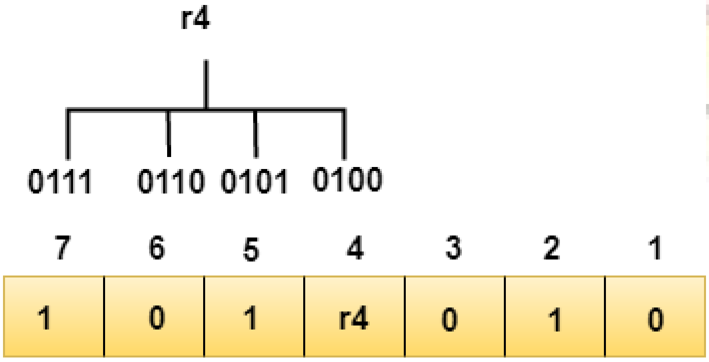
The r2 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the second position.



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**.

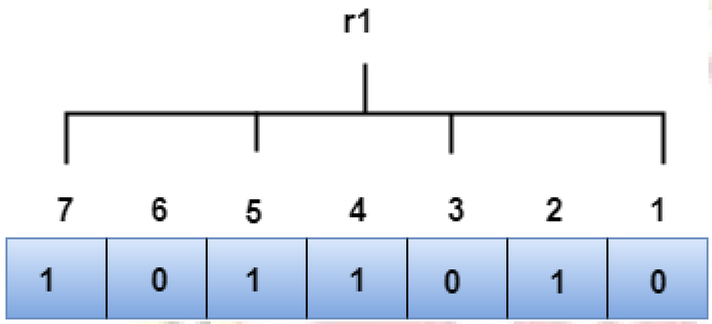
Data transferred is given below:

7	6	5	4	3	2	1
1	0	1	0	0	1	0

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

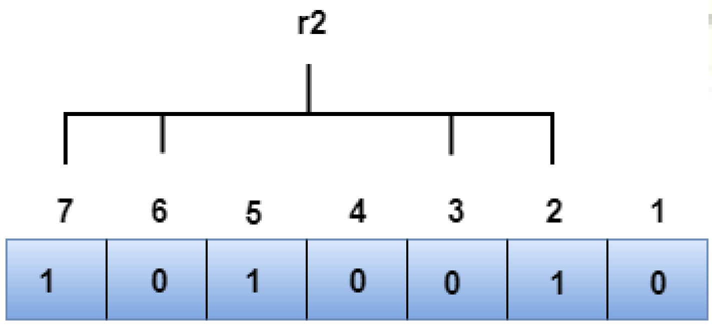


7	6	5	4	3	2	1
1	0	1	1	0	1	0

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.

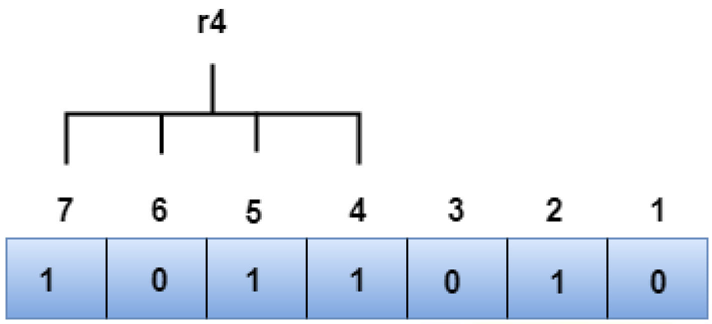


7	6	5	4	3	2	1
1	0	1	0	0	1	0

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.

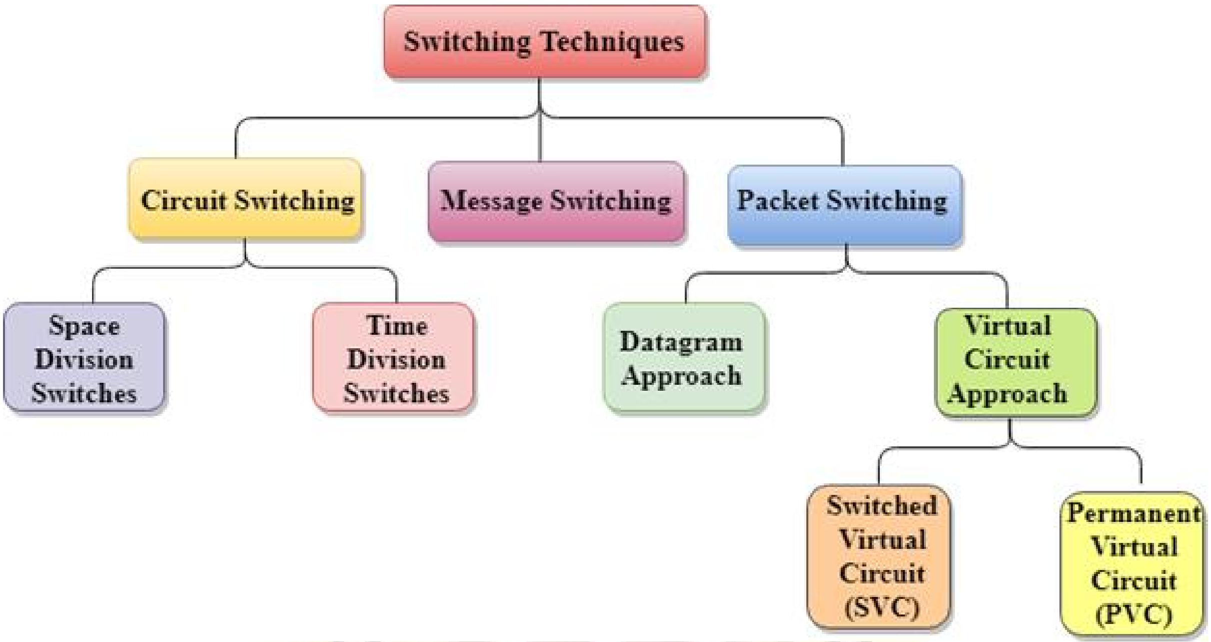
- The binary representation of redundant bits, i.e., r4r2r1 is 100, and its corresponding decimal value is 4. Therefore, the error occurs in a 4th bit position. The bit value must be changed from 1 to 0 to correct the error.

2.12 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

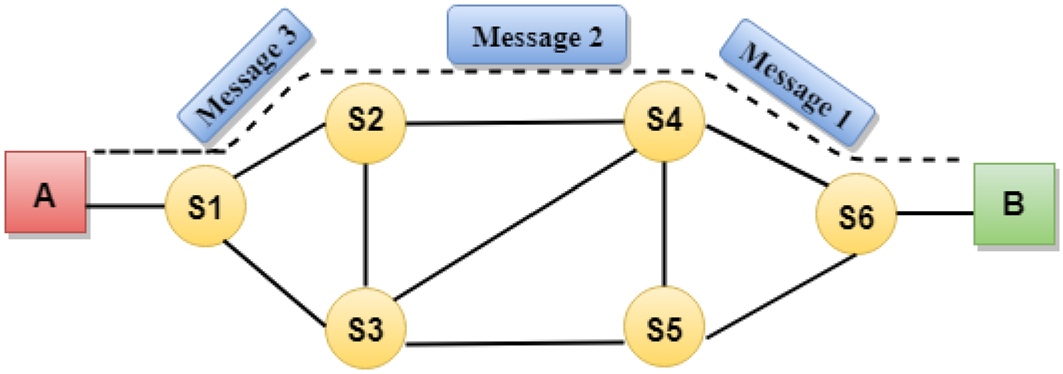


2.12.1 Circuit Switching

- 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.

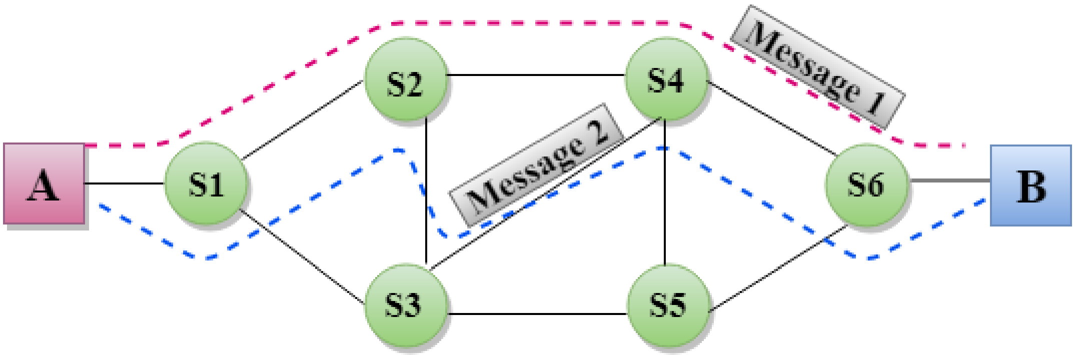
Disadvantages of Circuit Switching:

- 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.

2.12.2 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.

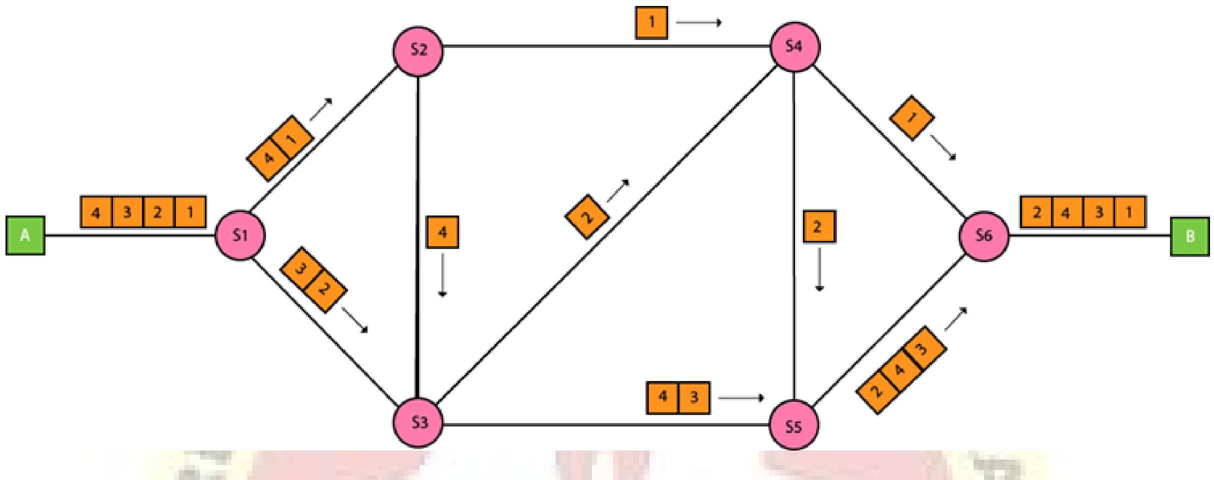
Disadvantages of Message Switching

- 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.

2.12.3 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:

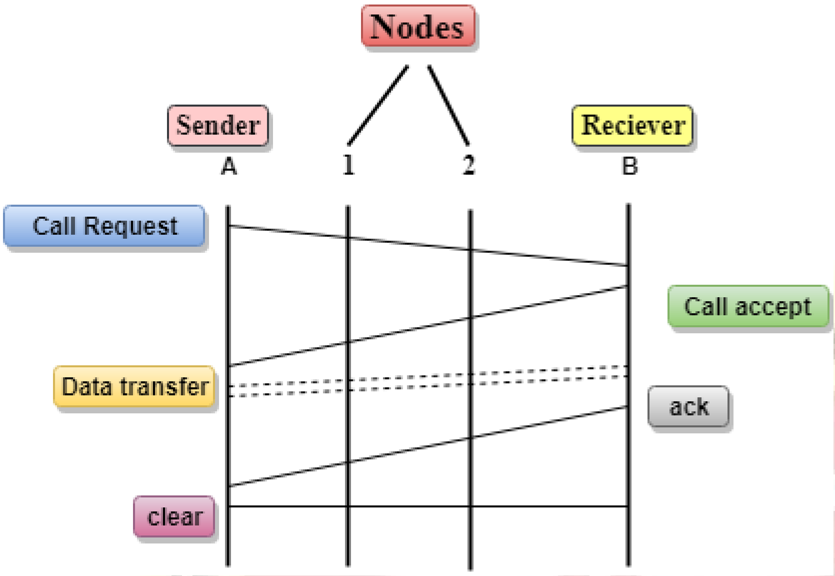
- It is a packet switching technology in which packet is known as a datagram, is considered as an independent entity. Each packet contains the information about the destination and switch uses this information to forward the packet to the correct destination.
- The packets are reassembled at the receiving 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.

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.

2.13 Data Link Layer Design Issues:

Reliable Delivery:

Frames are delivered to the receiver reliably and in the same order as generated by the sender. *Connection state* keeps track of sending order and which frames require retransmission. For example, receiver state includes which frames have been received, which ones have not, etc.

Best Effort:

The receiver does not return acknowledgments to the sender, so the sender has no way of knowing if a frame has been successfully delivered.

Acknowledged Delivery:

The receiver returns an acknowledgment frame to the sender indicating that a data frame was properly received. This sits somewhere between the other two in that the sender keeps connection state, but may not necessarily retransmit unacknowledged frames. Likewise, the receiver may hand received packets to higher layers in the order in which the arrive, regardless of the original sending order.

Framing

The DLL translates the physical layer's raw bit stream into discrete units (messages) called *frames*. How can the receiver detect frame boundaries? That is, how can the receiver recognize the start and end of a frame?

Length Count:

Make the first field in the frame's header be the length of the frame. That way the receiver knows how big the current frame is and can determine where the next frame ends.

Bit Stuffing:

Use reserved bit patterns to indicate the start and end of a frame. For instance, use the 4-bit sequence of 0111 to delimit consecutive frames. A frame consists of everything between two delimiters.

Character stuffing:

Use reserved characters to indicate the start and end of a frame. For instance, use the two-character sequence DLE STX (Data-Link Escape, Start of TeXt) to signal the beginning of a frame, and the sequence DLE ETX (End of TeXt) to flag the frame's end.

Encoding Violations:

Send an signal that doesn't conform to any legal bit representation. In Manchester encoding, for instance, 1-bits are represented by a high-low sequence, and 0-bits by low-high sequences. The start/end of a frame could be represented by the signal low-low or high-high.

The advantage of encoding violations is that no extra bandwidth is required as in bit-stuffing. The IEEE 802.4 standard uses this approach.

Error Control

Error control is concerned with insuring that all frames are eventually delivered to a destination.

Acknowledgements:

Typically, reliable delivery is achieved using the ``acknowledgments with retransmission'' paradigm, whereby the receiver returns a special *acknowledgment* (ACK) frame to the sender indicating the correct receipt of a frame.

In some systems, the receiver also returns a *negative acknowledgment* (NACK) for incorrectly-received frames. This is nothing more than a hint to the sender so that it can retransmit a frame right away without waiting for a timer to expire.

Timers:

One problem that simple ACK/NACK schemes fail to address is recovering from a frame that is lost, and as a result, fails to solicit an ACK or NACK. What happens if an ACK or NACK becomes lost?

Retransmission timers are used to resend frames that don't produce an ACK. When sending a frame, schedule a timer to expire at some time after the ACK should have been returned. If the timer goes off, retransmit the frame.

Sequence Numbers:

Retransmissions introduce the possibility of duplicate frames. To suppress duplicates, add sequence numbers to each frame, so that a receiver can distinguish between new frames and old copies.

Flow Control

Flow control deals with throttling the speed of the sender to match that of the receiver. Usually, this is a dynamic process, as the receiving speed depends on such changing factors as the load, and availability of buffer space.

One solution is to have the receiver extend *credits* to the sender. For each credit, the sender may send one frame. Thus, the receiver controls the transmission rate by handing out credits.

Link Management

In some cases, the data link layer service must be ``opened'' before use:

- The data link layer uses open operations for allocating buffer space, control blocks, agreeing on the maximum message size, etc.
- Synchronize and initialize send and receive sequence numbers with its peer at the other end of the communications channel

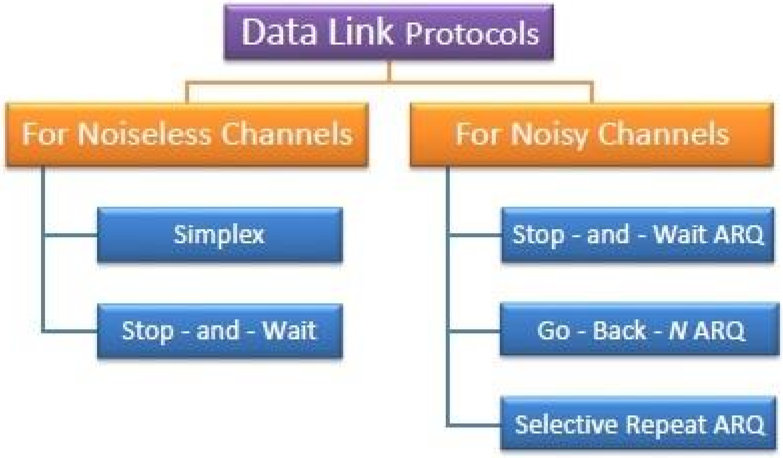
UNIT-III

3.1 Elementary Data Link Protocol

Protocols in the data link layer are designed so that this layer can perform its basic functions: framing, error control and flow control. Framing is the process of dividing bit -streams from physical layer into data frames whose size ranges from a few hundred to a few thousand bytes. Error control mechanisms deals with transmission errors and retransmission of corrupted and lost frames. Flow control regulates speed of delivery and so that a fast sender does not drown a slow receiver.

Types of Data Link Protocols

Data link protocols can be broadly divided into two categories, depending on whether the transmission channel is noiseless or noisy.



Simplex Protocol

The Simplex protocol is hypothetical protocol designed for unidirectional data transmission over an ideal channel, i.e. a channel through which transmission can never go wrong. It has distinct procedures for sender and receiver. The sender simply sends all its data available onto the channel as soon as they are available its buffer. The receiver is assumed to process all incoming data instantly. It is hypothetical since it does not handle flow control or error control.

Stop – and – Wait Protocol

Stop – and – Wait protocol is for noiseless channel too. It provides unidirectional data transmission without any error control facilities. However, it provides for flow control so that a fast sender does not drown a slow receiver. The receiver has a finite buffer size with finite processing speed. The sender can send a frame only when it has received indication from the receiver that it is available for further data processing.

Stop – and – Wait ARQ

Stop – and – wait Automatic Repeat Request (Stop – and – Wait ARQ) is a variation of the above protocol with added error control mechanisms, appropriate for noisy channels. The sender keeps a copy of the sent frame. It then waits for a finite time to receive a positive acknowledgement from receiver. If the timer expires or a negative acknowledgement is received, the frame is retransmitted. If a positive acknowledgement is received then the next frame is sent.

Go – Back – N ARQ

Go – Back – N ARQ provides for sending multiple frames before receiving the acknowledgement for the first frame. It uses the concept of sliding window, and so is also called sliding window protocol. The frames are sequentially numbered and a finite number of frames are sent. If the acknowledgement of a frame is not received within the time period, all frames starting from that frame are retransmitted.

Selective Repeat ARQ

This protocol also provides for sending multiple frames before receiving the acknowledgement for the first frame. However, here only the erroneous or lost frames are retransmitted, while the good frames are received and buffered.

3.2 Sliding Window Protocol

The sliding window is a technique for sending multiple frames at a time. It controls the data packets between the two devices where reliable and gradual delivery of data frames is needed.

In this technique, each frame has sent from the sequence number. The sequence numbers are used to find the missing data in the receiver end. The purpose of the sliding window technique is to avoid duplicate data, so it uses the sequence number.

Types of Sliding Window Protocol

Sliding window protocol has two types:

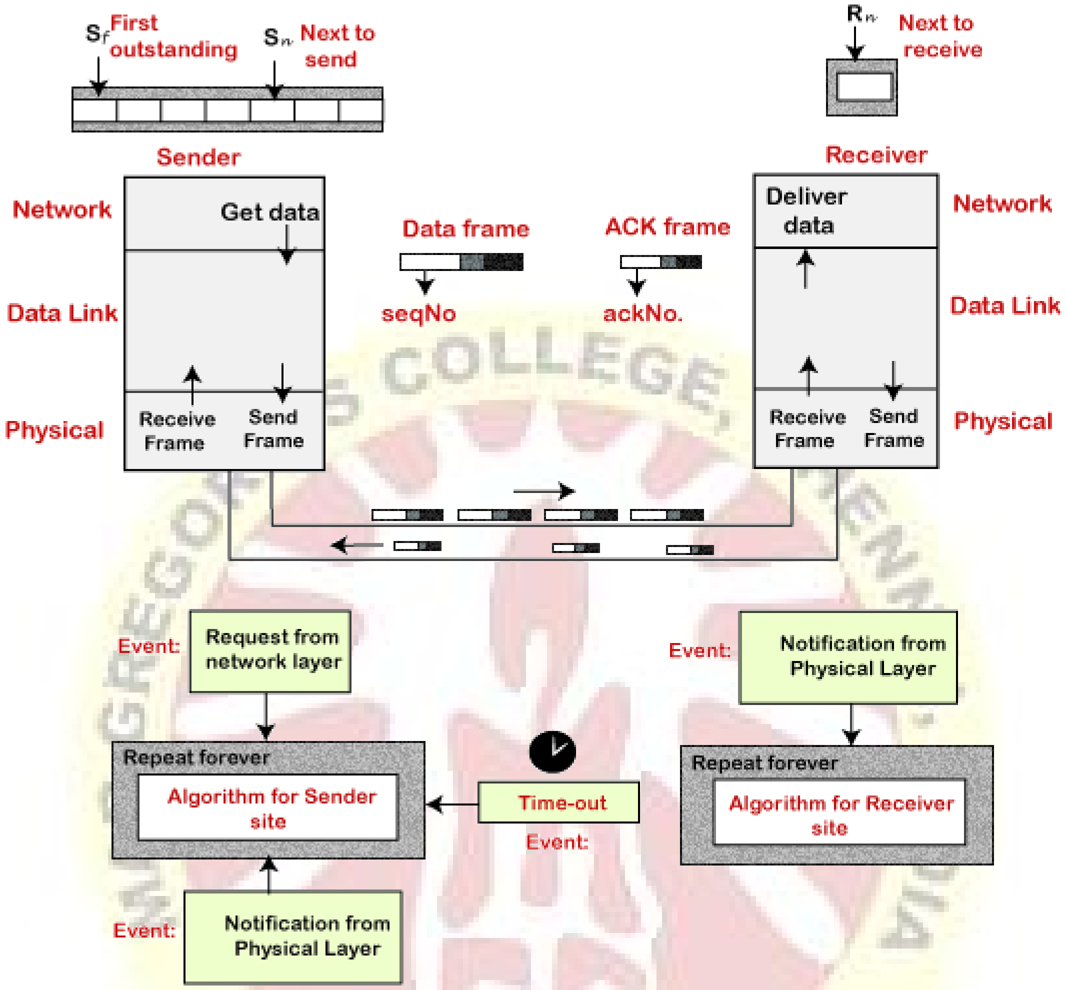
1. Go-Back-N ARQ
2. Selective Repeat ARQ

Go-Back-N ARQ

Go-Back-N ARQ protocol is also known as Go-Back-N Automatic Repeat Request. It is a data link layer protocol that uses a sliding window method. In this, if any frame is corrupted or lost, all subsequent frames have to be sent again.

The size of the sender window is N in this protocol. For example, Go-Back-8, the size of the sender window, will be 8. The receiver window size is always 1.

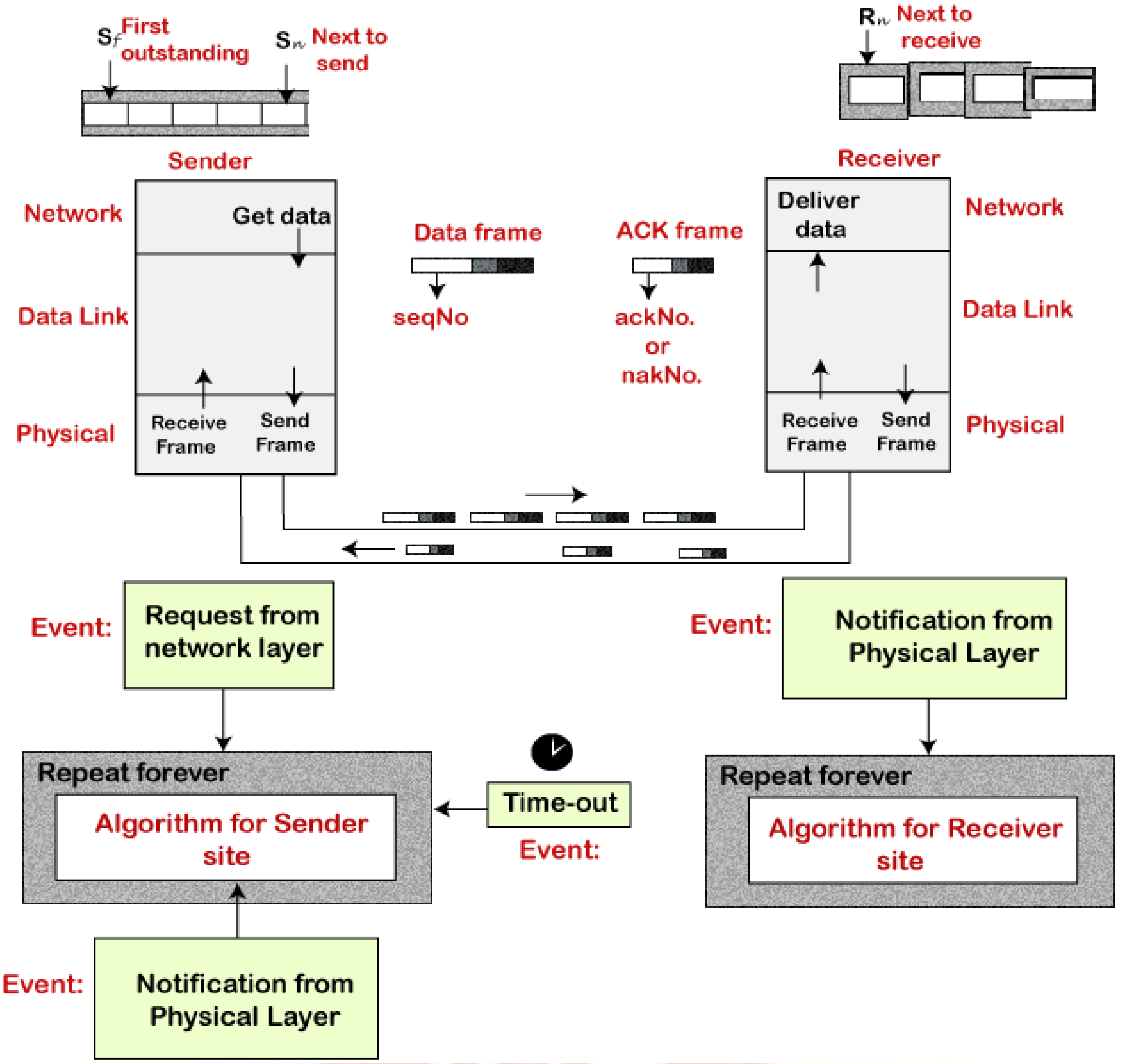
If the receiver receives a corrupted frame, it cancels it. The receiver does not accept a corrupted frame. When the timer expires, the sender sends the correct frame again. The design of the Go-Back-N ARQ protocol is shown below.



Selective Repeat ARQ

Selective Repeat ARQ is also known as the Selective Repeat Automatic Repeat Request. It is a data link layer protocol that uses a sliding window method. The Go-back-N ARQ protocol works well if it has fewer errors. But if there is a lot of error in the frame, lots of bandwidth loss in sending the frames again. So, we use the Selective Repeat ARQ protocol. In this protocol, the size of the sender window is always equal to the size of the receiver window. The size of the sliding window is always greater than 1.

If the receiver receives a corrupt frame, it does not directly discard it. It sends a negative acknowledgment to the sender. The sender sends that frame again as soon as on the receiving negative acknowledgment. There is no waiting for any time-out to send that frame. The design of the Selective Repeat ARQ protocol is shown below.

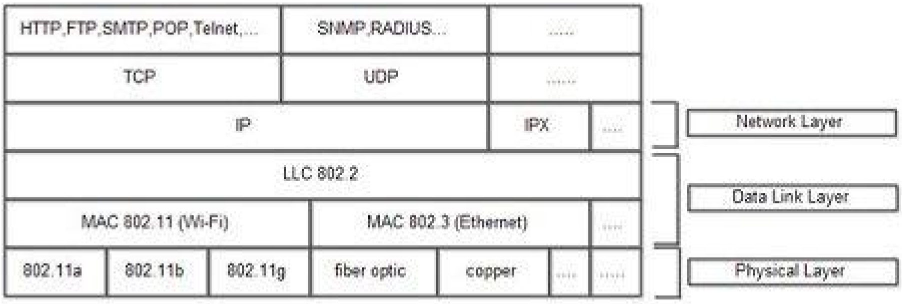


3.3 Medium Access Layer

The Media Access Control (MAC) data communication Networks protocol sub-layer, also known as the Medium Access Control, is a sub-layer of the data link layer specified in the seven-layer OSI model. The medium access layer was made necessary by systems that share a common communications medium. Typically these are local area networks. The MAC layer is the "low" part of the second OSI layer, the layer of the "data link". In fact, the IEEE divided this layer into two layers "above" is the control layer the logical connection (Logical Link Control, LLC) and "down" the control layer The medium access (MAC).

The MAC sub-layer has two primary responsibilities:

Data encapsulation, including frame assembly before transmission, and frame parsing/error detection during and after reception. Media access control, including initiation of frame transmission and recovery from transmission failure.

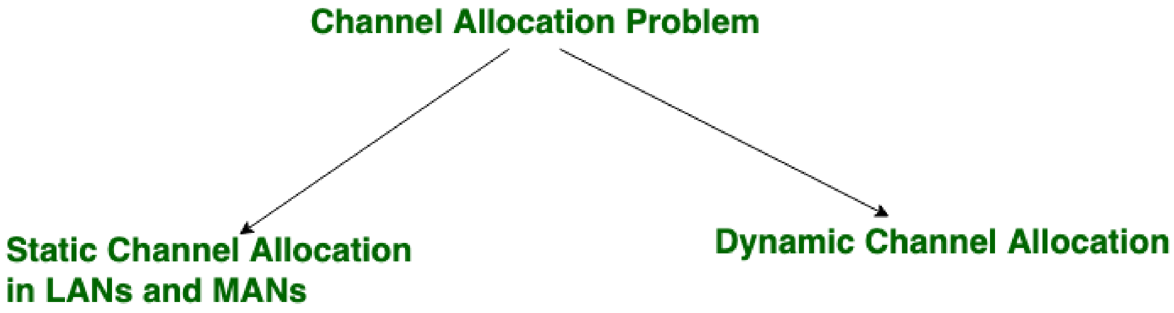


Network layers.

3.4 Channel Allocation Problem

Channel allocation is a process in which a single channel is divided and allotted to multiple users in order to carry user specific tasks. There are user's quantity may vary every time the process takes place. If there are N number of users and channel is divided into N equal-sized sub channels, Each user is assigned one portion. If the number of users are small and don't vary at times, than Frequency Division Multiplexing can be used as it is a simple and efficient channel bandwidth allocating technique.

Channel allocation problem can be solved by two schemes: Static Channel Allocation in LANs and MANs, and Dynamic Channel Allocation.


Channel Allocation Problem
Static Channel Allocation
in LANs and MANs
Dynamic Channel Allocation

These are explained as following below.

1. Static Channel Allocation in LANs and MANs:

It is the classical or traditional approach of allocating a single channel among multiple competing users Frequency Division Multiplexing (FDM). if there are N users, the bandwidth is divided into N equal sized portions each user being assigned one portion. since each user has a private frequency band, there is no interface between users.

It is not efficient to divide into fixed number of chunks.

T = $1/(U*C-L)$

T(FDM) = $N*T(1/U(C/N)-L/N)$

Where,

T = mean time delay,

C = capacity of channel,

L = arrival rate of frames,

1/U = bits/frame,

N = number of sub channels,

T(FDM) = Frequency Division Multiplexing Time

2. Dynamic Channel Allocation:

Possible assumptions include:

1. **Station Model:**
   Assumes that each of N stations independently produce frames. The probability of producing a packet in the interval IDt where I is the constant arrival rate of new frames.
2. **Single Channel Assumption:**
   In this allocation all stations are equivalent and can send and receive on that channel.
3. **Collision Assumption:**
   If two frames overlap in time-wise, then that's collision. Any collision is an error, and both frames must re transmitted. Collisions are only possible error.
4. **Time** can be divided into Slotted or Continuous.
5. **Stations** can sense a channel is busy before they try it.

Protocol Assumption:

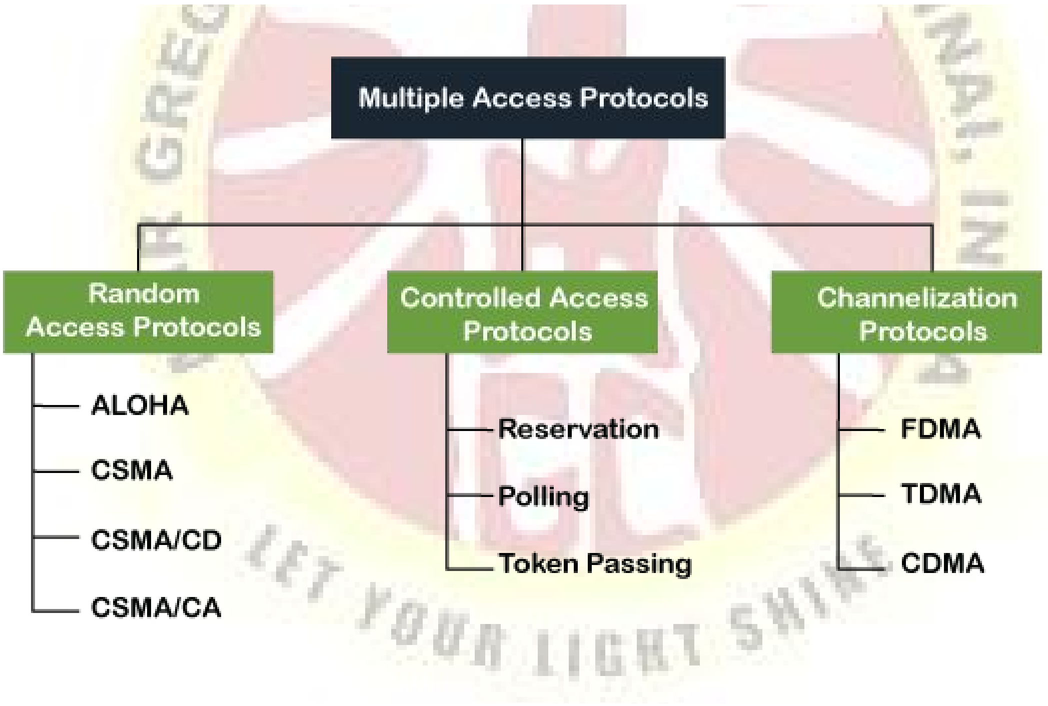
- N independent stations.
- A station is blocked untill its generated frame is transmitted.
- probability of a frame being generated in a period of length Dt is IDt where I is the arrival rate of frames.
- Only a single Channel available.
- Time can be either: Continuous or slotted.

- **Carrier Sense:** A station can sense if a channel is already busy before transmission.
- **No Carrier Sense:** Time out used to sense loss data.

3.5 **Multiple Access Protocol**

When a sender and receiver have a dedicated link to transmit data packets, the data link control is enough to handle the channel. Suppose there is no dedicated path to communicate or transfer the data between two devices. In that case, multiple stations access the channel and simultaneously transmits the data over the channel. It may create collision and cross talk. Hence, the multiple access protocol is required to reduce the collision and avoid crosstalk between the channels.

Following are the types of multiple access protocol that is subdivided into the different process as:



A. Random Access Protocol

In this protocol, all the station has the equal priority to send the data over a channel. In random access protocol, one or more stations cannot depend on another station nor any station control another station. Depending on the channel's state (idle or busy), each station transmits the data frame. However, if more than one station sends the data over a channel, there may be a collision or data conflict. Due to the collision, the data frame packets may be lost or changed. And hence, it does not receive by the receiver end.

Following are the different methods of random-access protocols for broadcasting frames on the channel.

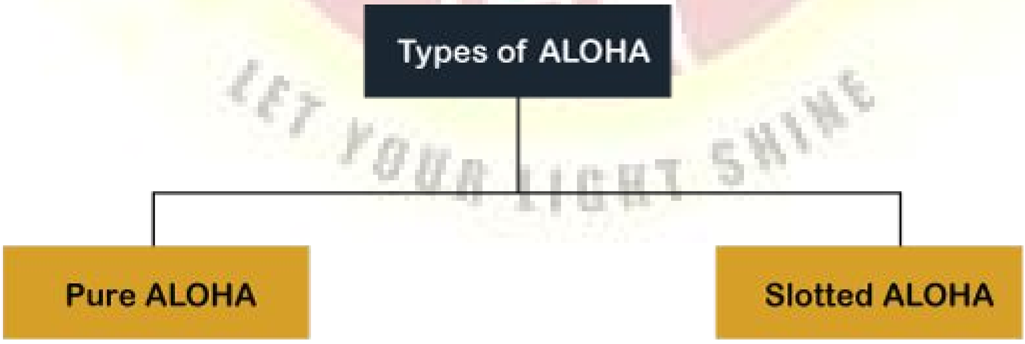
- Aloha
- CSMA
- CSMA/CD
- CSMA/CA

ALOHA Random Access Protocol

It is designed for wireless LAN (Local Area Network) but can also be used in a shared medium to transmit data. Using this method, any station can transmit data across a network simultaneously when a data frameset is available for transmission.

Aloha Rules

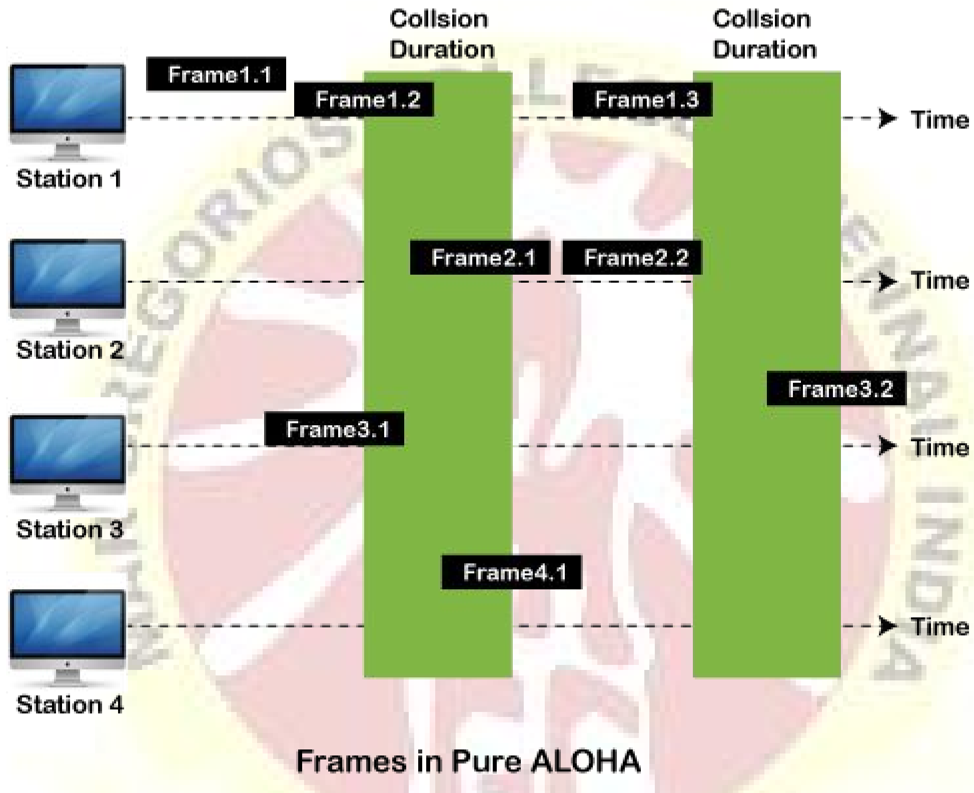
1. Any station can transmit data to a channel at any time.
2. It does not require any carrier sensing.
3. Collision and data frames may be lost during the transmission of data through multiple stations.
4. Acknowledgment of the frames exists in Aloha. Hence, there is no collision detection.
5. It requires retransmission of data after some random amount of time.



Pure Aloha

Whenever data is available for sending over a channel at stations, we use Pure Aloha. In pure Aloha, when each station transmits data to a channel without checking whether the channel is idle or not, the chances of collision may occur, and the data frame can be lost. When any station transmits the data frame to a channel, the pure Aloha waits for the receiver's acknowledgment. If it does not acknowledge the receiver end within the specified time, the station waits for a random amount of time, called the backoff time (Tb). And the station may assume the frame has been lost or destroyed. Therefore, it retransmits the frame until all the data are successfully transmitted to the receiver.

1. The total vulnerable time of pure Aloha is 2 * Tfr.
2. Maximum throughput occurs when G = 1/ 2 that is 18.4%.
3. Successful transmission of data frame is S = G * e ^ - 2 G.



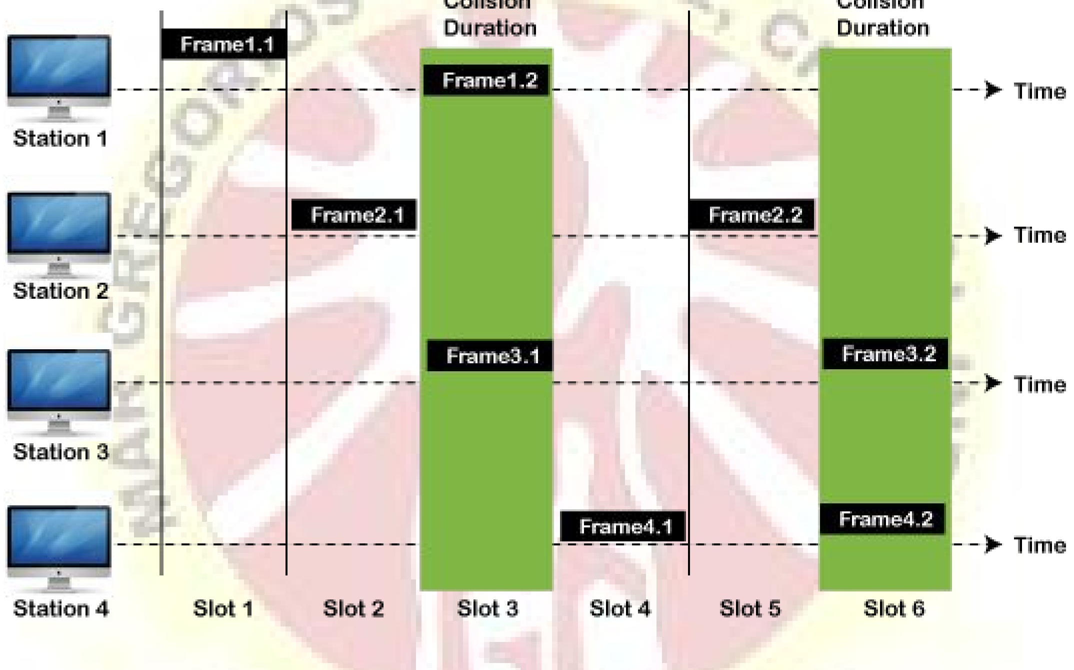
Frames in Pure ALOHA

As we can see in the figure above, there are four stations for accessing a shared channel and transmitting data frames. Some frames collide because most stations send their frames at the same time. Only two frames, frame 1.1 and frame 2.2, are successfully transmitted to the receiver end. At the same time, other frames are lost or destroyed. Whenever two frames fall on a shared channel simultaneously, collisions can occur, and both will suffer damage. If the new frame's first bit enters the channel before finishing the last bit of the second frame. Both frames are completely finished, and both stations must retransmit the data frame.

Slotted Aloha

The slotted Aloha is designed to overcome the pure Aloha's efficiency because pure Aloha has a very high possibility of frame hitting. In slotted Aloha, the shared channel is divided into a fixed time interval called **slots**. So that, if a station wants to send a frame to a shared channel, the frame can only be sent at the beginning of the slot, and only one frame is allowed to be sent to each slot. And if the stations are unable to send data to the beginning of the slot, the station will have to wait until the beginning of the slot for the next time. However, the possibility of a collision remains when trying to send a frame at the beginning of two or more station time slot.

1. Maximum throughput occurs in the slotted Aloha when G = 1 that is 37%.
2. The probability of successfully transmitting the data frame in the slotted Aloha is S = G * e ^ - 2 G.
3. The total vulnerable time required in slotted Aloha is Tfr.



Frames in Slotted ALOHA

CSMA (Carrier Sense Multiple Access)

It is a **carrier sense multiple access** based on media access protocol to sense the traffic on a channel (idle or busy) before transmitting the data. It means that if the channel is idle, the station can send data to the channel. Otherwise, it must wait until the channel becomes idle. Hence, it reduces the chances of a collision on a transmission medium.

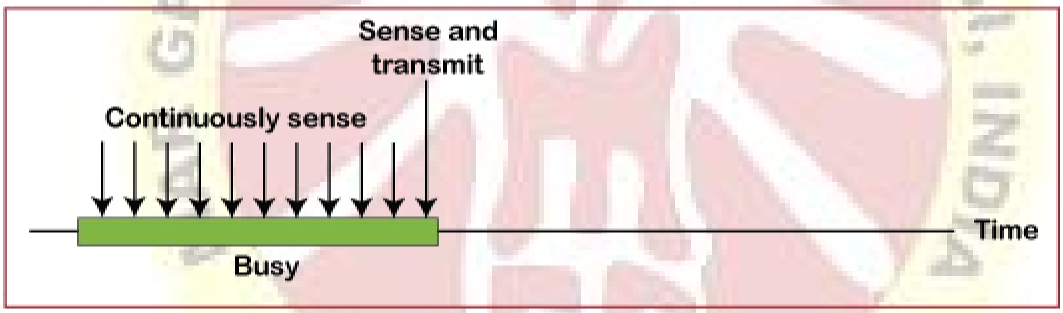
CSMA Access Modes

1-Persistent: In the 1-Persistent mode of CSMA that defines each node, first sense the shared channel and if the channel is idle, it immediately sends the data. Else it must wait and keep track of the status of the channel to be idle and broadcast the frame unconditionally as soon as the channel is idle.

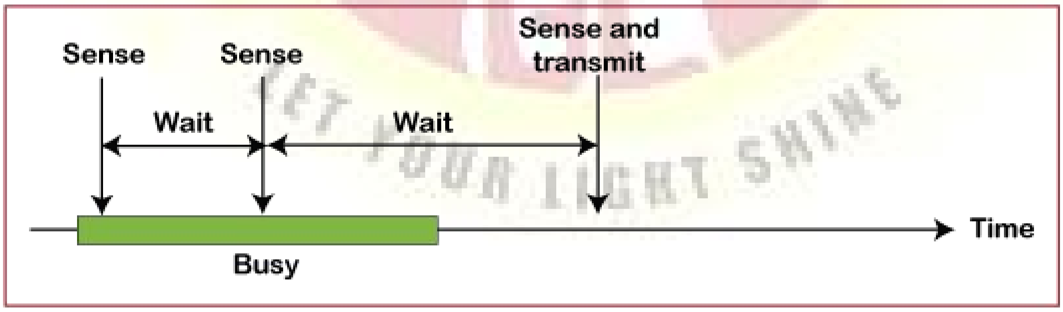
Non-Persistent: It is the access mode of CSMA that defines before transmitting the data, each node must sense the channel, and if the channel is inactive, it immediately sends the data. Otherwise, the station must wait for a random time (not continuously), and when the channel is found to be idle, it transmits the frames.

P-Persistent: It is the combination of 1-Persistent and Non-persistent modes. The P-Persistent mode defines that each node senses the channel, and if the channel is inactive, it sends a frame with a **P** probability. If the data is not transmitted, it waits for a (**q = 1-p probability**) random time and resumes the frame with the next time slot.

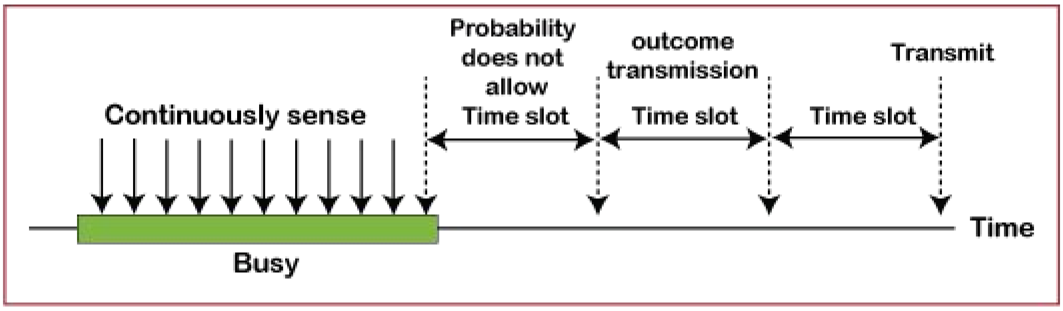
O- Persistent: It is an O-persistent method that defines the superiority of the station before the transmission of the frame on the shared channel. If it is found that the channel is inactive, each station waits for its turn to retransmit the data.



a. 1-persistent



b. Nonpersistent



c. p-persistent

CSMA/ CD

It is a **carrier sense multiple access/ collision detection** network protocol to transmit data frames. The CSMA/CD protocol works with a medium access control layer. Therefore, it first senses the shared channel before broadcasting the frames, and if the channel is idle, it transmits a frame to check whether the transmission was successful. If the frame is successfully received, the station sends another frame. If any collision is detected in the CSMA/CD, the station sends a jam/ stop signal to the shared channel to terminate data transmission. After that, it waits for a random time before sending a frame to a channel.

CSMA/ CA

It is a **carrier sense multiple access/collision avoidance** network protocol for carrier transmission of data frames. It is a protocol that works with a medium access control layer. When a data frame is sent to a channel, it receives an acknowledgment to check whether the channel is clear. If the station receives only a single (own) acknowledgments, that means the data frame has been successfully transmitted to the receiver. But if it gets two signals (its own and one more in which the collision of frames), a collision of the frame occurs in the shared channel. Detects the collision of the frame when a sender receives an acknowledgment signal.

Following are the methods used in the CSMA/ CA to avoid the collision:

Interframe space: In this method, the station waits for the channel to become idle, and if it gets the channel is idle, it does not immediately send the data. Instead of this, it waits for some time, and this time period is called the **Interframe** space or IFS. However, the IFS time is often used to define the priority of the station.

Contention window: In the Contention window, the total time is divided into different slots. When the station/ sender is ready to transmit the data frame, it chooses a random slot number of slots as **wait time**. If the channel is still busy, it does not restart the entire process, except that it restarts the timer only to send data packets when the channel is inactive.

Acknowledgment: In the acknowledgment method, the sender station sends the data frame to the shared channel if the acknowledgment is not received ahead of time.

B. Controlled Access Protocol

It is a method of reducing data frame collision on a shared channel. In the controlled access method, each station interacts and decides to send a data frame by a particular station approved by all other stations. It means that a single station cannot send the data frames unless all other stations are not approved. It has three types of controlled access: **Reservation, Polling**, and **Token Passing**.

C. Channelization Protocols

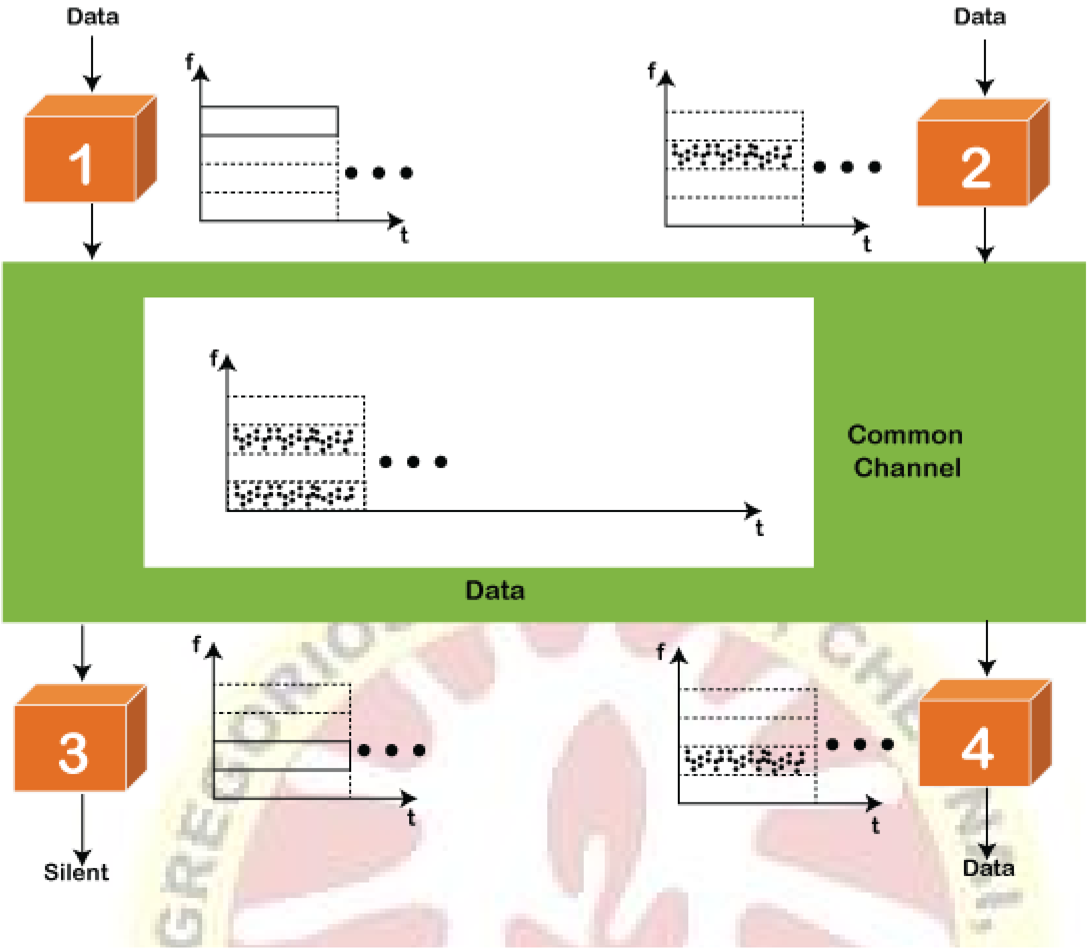
It is a channelization protocol that allows the total usable bandwidth in a shared channel to be shared across multiple stations based on their time, distance and codes. It can access all the stations at the same time to send the data frames to the channel.

Following are the various methods to access the channel based on their time, distance and codes:

1. FDMA (Frequency Division Multiple Access)
2. TDMA (Time Division Multiple Access)
3. CDMA (Code Division Multiple Access)

FDMA

It is a frequency division multiple access (**FDMA**) method used to divide the available bandwidth into equal bands so that multiple users can send data through a different frequency to the subchannel. Each station is reserved with a particular band to prevent the crosstalk between the channels and interferences of stations.



TDMA

Time Division Multiple Access (**TDMA**) is a channel access method. It allows the same frequency bandwidth to be shared across multiple stations. And to avoid collisions in the shared channel, it divides the channel into different frequency slots that allocate stations to transmit the data frames. The same **frequency** bandwidth into the shared channel by dividing the signal into various time slots to transmit it. However, TDMA has an overhead of synchronization that specifies each station's time slot by adding synchronization bits to each slot.

CDMA

The code division multiple access (CDMA) is a channel access method. In CDMA, all stations can simultaneously send the data over the same channel. It means that it allows each station to transmit the data frames with full frequency on the shared channel at all times. It does not require the division of bandwidth on a shared channel based on time slots. If multiple stations send data to a channel simultaneously, their data frames are separated by a unique code sequence. Each station has a different unique code for transmitting the data over a shared channel. For example, there are multiple users in a room that are continuously speaking. Data is received by the users if only two-person interact with each other using the same language. Similarly, in the network, if different stations communicate with each other simultaneously with different code language.

UNIT -IV

4.1 NETWORK LAYER DESIGN ISSUES:

1) Store and formed packet switching.

2) Service provided to the transport layer.

3) Implementation of connectionless service.

4) Implementation of connection-oriented source.

5) Comparison of virtual circuit and datagram submits.

1) Store and forward operation : -

i) Host transmits packet to router across LAN or oval point to point link.

ii) Packet is stored on router until fully arrived and processed.

iii) Packet is forward to next router.

2) Service provide to transport layer :

The network layer services have been designed with the goals : -

i) The advice should independent of router telenet

ii) The transport layer should be shilded from the number type and topology of the router present.

iii) The network addresses maid arailable to transport

3)Implementation of Connectionless service

Connectionless service is offered packets are injected into the subnet individually and routed independently of each other. Each packet is transmitted indenpendently. Connectionless service used in network layer ID and transport layer. Packet are frequently called datagram connectionless service is largely for data communication the internet.

3) Implementation of Connection-oriented service
Connection-oriented service is used a path from the source router to the destination router must be established before any data packet can be sent. Connection oriented service also called virtual circuit service. This service used network layer for ATM. It also used in transport layer for TCP. A connection must be established before any can be sent packets order preserved logical connection is also established here.

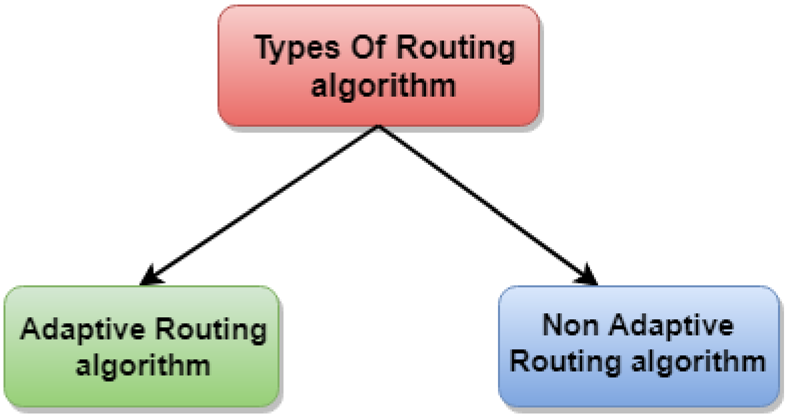
4.2 Routing algorithm

- 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


Types Of Routing algorithm
Adaptive Routing algorithm
Non Adaptive Routing algorithm

4.2.1 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.

4.2.2 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.

4.2.3 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:

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

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:
$D_x(y) = c(x,y)$     // If y is not a neighbor then $c(x,y) = \infty$
for each neighbor w
$D_w(y) = ?$     for all destination y in N.
for each neighbor w
send distance vector $D_x = [ D_x(y) : y \in N ]$ to w
loop
  wait(until I receive any distance vector from some neighbor w)
  for each y in N:
  $D_x(y) = \min_v\{c(x,v)+D_v(y)\}$
If $D_x(y)$ is changed for any destination y
Send distance vector $D_x = [ D_x(y) : y \in N ]$ to all neighbors
forever
```

4.2.4 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
```

4.3 Congestion

A state occurring in network layer when the message traffic is so heavy that it slows down network response time.

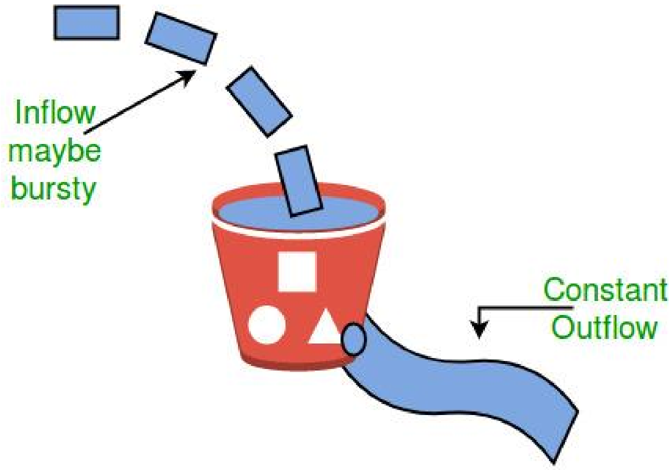
Effects of Congestion

- As delay increases, performance decreases.
- If delay increases, retransmission occurs, making situation worse.

4.3.1 Congestion control algorithms

4.3.1.1 Leaky Bucket Algorithm

Imagine a bucket with a small hole in the bottom. No matter at what rate water enters the bucket, the outflow is at constant rate. When the bucket is full with water additional water entering spills over the sides and is lost.



Similarly, each network interface contains a leaky bucket and the following **steps** are involved in leaky bucket algorithm:

1. When host wants to send packet, packet is thrown into the bucket.
2. The bucket leaks at a constant rate, meaning the network interface transmits packets at a constant rate.
3. Bursty traffic is converted to a uniform traffic by the leaky bucket.
4. In practice the bucket is a finite queue that outputs at a finite rate.

4.3.1.2 Token bucket Algorithm

Need of token bucket Algorithm:-

The leaky bucket algorithm enforces output pattern at the average rate, no matter how bursty the traffic is. So in order to deal with the bursty traffic we need a flexible algorithm so that the data is not lost. One such algorithm is token bucket algorithm.

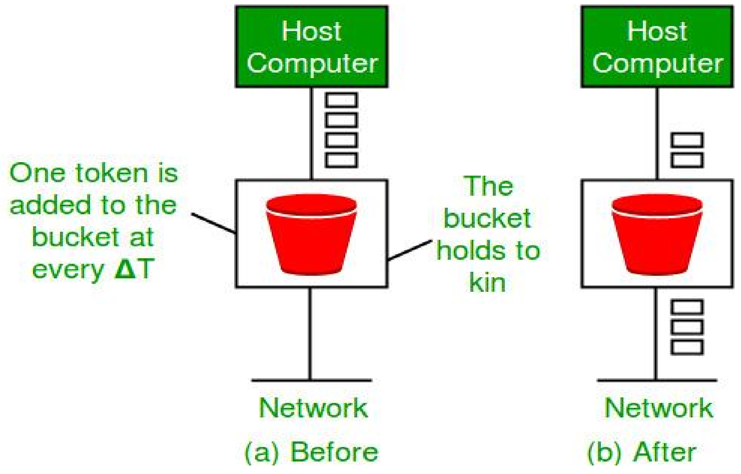
Steps of this algorithm can be described as follows:

1. In regular intervals tokens are thrown into the bucket. ƒ
2. The bucket has a maximum capacity. ƒ
3. If there is a ready packet, a token is removed from the bucket, and the packet is send.
4. If there is no token in the bucket, the packet cannot be send.

Let's understand with an example,

In figure (A) we see a bucket holding three tokens, with five packets waiting to be transmitted.For a packet to be transmitted, it must capture and destroy one token. In figure (B) We see that three of the five packets have gotten through, but the other two are stuck waiting for more tokens to be generated.

Let's understand with an example,



(a) Before (b) After

4.4 Internet Protocols

4.4.1Transmission Control Protocol (TCP)

TCP is a connection oriented protocol and offers end-to-end packet delivery. It acts as back bone for connection.It exhibits the following key features:

- Transmission Control Protocol (TCP) corresponds to the Transport Layer of OSI Model.

- TCP is a reliable and connection oriented protocol.
- TCP offers:
    - Stream Data Transfer.
    - Reliability.
    - Efficient Flow Control
    - Full-duplex operation.
    - Multiplexing.
- TCP offers connection oriented end-to-end packet delivery.
- TCP ensures reliability by sequencing bytes with a forwarding acknowledgement number that indicates to the destination the next byte the source expect to receive.
- It retransmits the bytes not acknowledged with in specified time period.

TCP Services

TCP offers following services to the processes at the application layer:

- Stream Delivery Service
- Sending and Receiving Buffers
- Bytes and Segments
- Full Duplex Service
- Connection Oriented Service
- Reliable Service

STREAM DELIVER SERVICE

TCP protocol is stream oriented because it allows the sending process to send data as stream of bytes and the receiving process to obtain data as stream of bytes.

SENDING AND RECEIVING BUFFERS

It may not be possible for sending and receiving process to produce and obtain data at same speed, therefore, TCP needs buffers for storage at sending and receiving ends.

BYTES AND SEGMENTS

The Transmission Control Protocol (TCP), at transport layer groups the bytes into a packet. This packet is called segment. Before transmission of these packets, these segments are encapsulated into an IP datagram.

FULL DUPLEX SERVICE

Transmitting the data in duplex mode means flow of data in both the directions at the same time.

CONNECTION ORIENTED SERVICE

TCP offers connection oriented service in the following manner:

1. TCP of process-1 informs TCP of process – 2 and gets its approval.
2. TCP of process – 1 and TCP of process – 2 and exchange data in both the two directions.
3. After completing the data exchange, when buffers on both sides are empty, the two TCP's destroy their buffers.

RELIABLE SERVICE

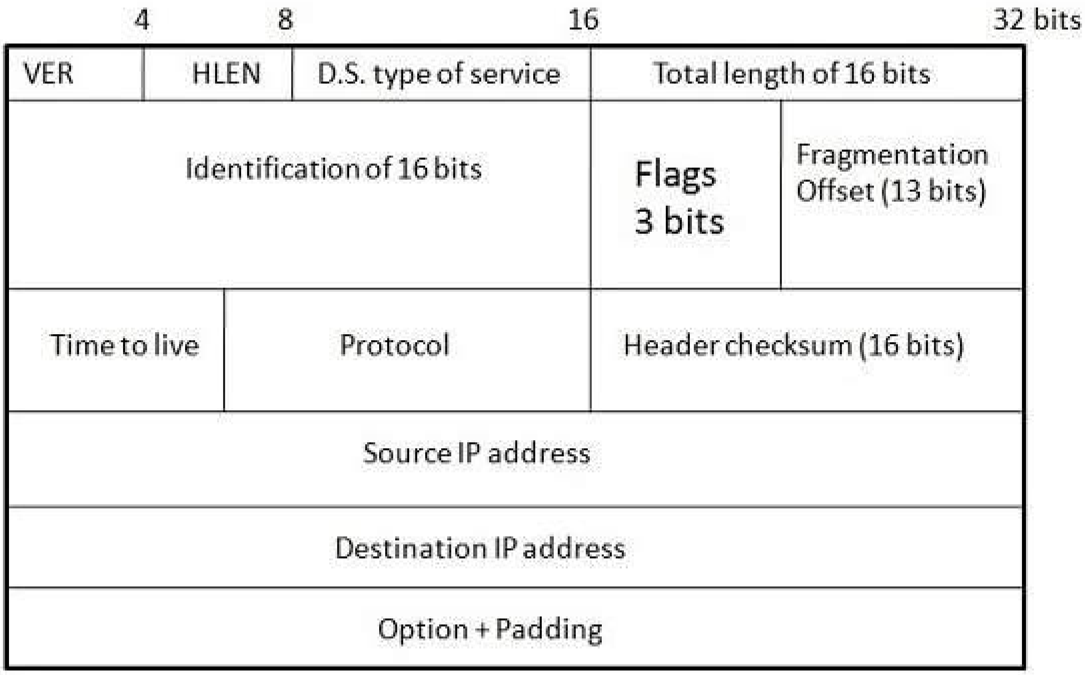
For sake of reliability, TCP uses acknowledgement mechanism.

4.4.2 Internet Protocol (IP)

Internet Protocol is **connectionless** and **unreliable** protocol. It ensures no guarantee of successfully transmission of data.

In order to make it reliable, it must be paired with reliable protocol such as TCP at the transport layer.

Internet protocol transmits the data in form of a datagram as shown in the following diagram:



4 8 16 32 bits

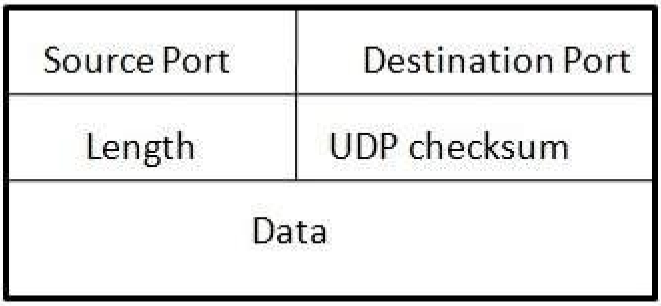
<table>
<tr><td>VER</td><td>HLEN</td><td>D.S. type of service</td><td colspan="2">Total length of 16 bits</td></tr>
<tr><td colspan="3">Identification of 16 bits</td><td>Flags
3 bits</td><td>Fragmentation Offset (13 bits)</td></tr>
<tr><td colspan="2">Time to live</td><td>Protocol</td><td colspan="2">Header checksum (16 bits)</td></tr>
<tr><td colspan="5">Source IP address</td></tr>
<tr><td colspan="5">Destination IP address</td></tr>
<tr><td colspan="5">Option + Padding</td></tr>
</table>

- The length of datagram is variable.
- The Datagram is divided into two parts: **header** and **data.**
- The length of header is 20 to 60 bytes.
- The header contains information for routing and delivery of the packet.

4.4.3 User Datagram Protocol (UDP)

Like IP, UDP is connectionless and unreliable protocol. It doesn't require making a connection with the host to exchange data. Since UDP is unreliable protocol, there is no mechanism for ensuring that data sent is received.

UDP transmits the data in form of a datagram. The UDP datagram consists of five parts as shown in the following diagram:



<table>
<tr><td>Source Port</td><td>Destination Port</td></tr>
<tr><td>Length</td><td>UDP checksum</td></tr>
<tr><td colspan="2">Data</td></tr>
</table>

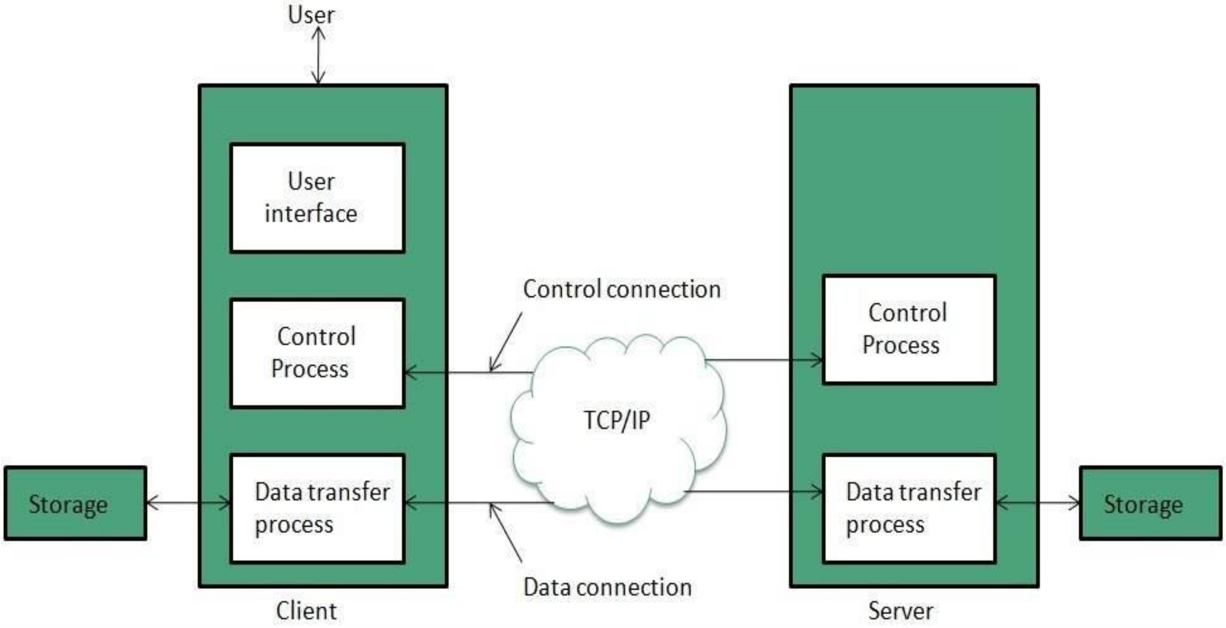
- UDP is used by the application that typically transmit small amount of data at one time.

- UDP provides protocol port used i.e. UDP message contains both source and destination port number, that makes it possible for UDP software at the destination to deliver the message to correct application program.

4.4.4 File Transfer Protocol (FTP)

FTP is used to copy files from one host to another. FTP offers the mechanism for the same in following manner:

- FTP creates two processes such as Control Process and Data Transfer Process at both ends i.e. at client as well as at server.
- FTP establishes two different connections: one is for data transfer and other is for control information.
- **Control connection** is made between **control processes** while **Data Connection** is made between<="" b="" style="box-sizing: border-box;">
- FTP uses **port 21** for the control connection and **Port 20** for the data connection.



4.4.5 Trivial File Transfer Protocol (TFTP)

Trivial File Transfer Protocol is also used to transfer the files but it transfers the files without authentication. Unlike FTP, TFTP does not separate control and data information. Since there is no authentication exists, TFTP lacks in security features therefore it is not recommended to use TFTP.

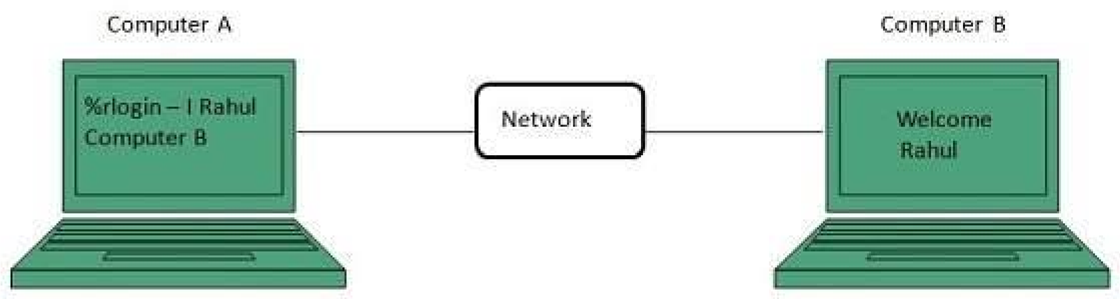
Key points

- TFTP makes use of UDP for data transport. Each TFTP message is carried in separate UDP datagram.

- The first two bytes of a TFTP message specify the type of message.
- The TFTP session is initiated when a TFTP client sends a request to upload or download a file.
- The request is sent from an ephemeral UDP port to the **UDP port 69** of an TFTP server.

4.4.6 Telnet

Telnet is a protocol used to log in to remote computer on the internet. There are a number of Telnet clients having user friendly user interface. The following diagram shows a person is logged in to computer A, and from there, he remote logged into computer B.



4.4.7 Hyper Text Transfer Protocol (HTTP)

HTTP is a communication protocol. It defines mechanism for communication between browser and the web server. It is also called request and response protocol because the communication between browser and server takes place in request and response pairs.

HTTP Request

HTTP request comprises of lines which contains:

- Request line
- Header Fields
- Message body
- The first line i.e. the **Request line** specifies the request method i.e. **Get** or **Post.**
- The second line specifies the header which indicates the domain name of the server from where index.htm is retrieved.

HTTP Response

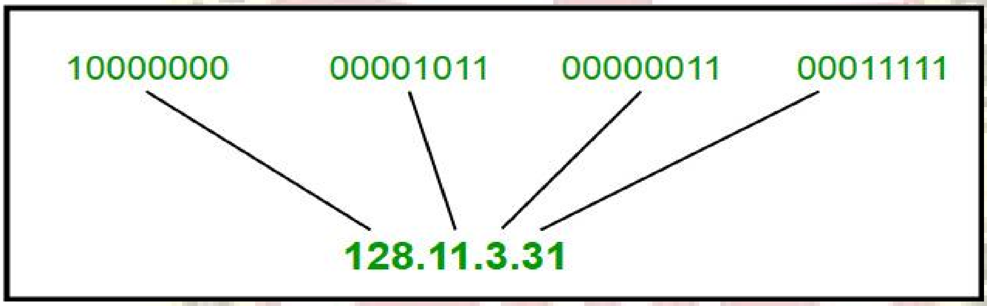
Like HTTP request, HTTP response also has certain structure. HTTP response contains:

- Status line
- Headers
- Message body

4.5 IP Addressing

IP address is an address having information about how to reach a specific host, especially outside the LAN. An IP address is a 32 bit unique address having an address space of 2^{32}. Generally, there are two notations in which IP address is written, dotted decimal notation and hexadecimal notation.

Dotted Decimal Notation



Classful Addressing

The 32 bit IP address is divided into five sub-classes. These are:

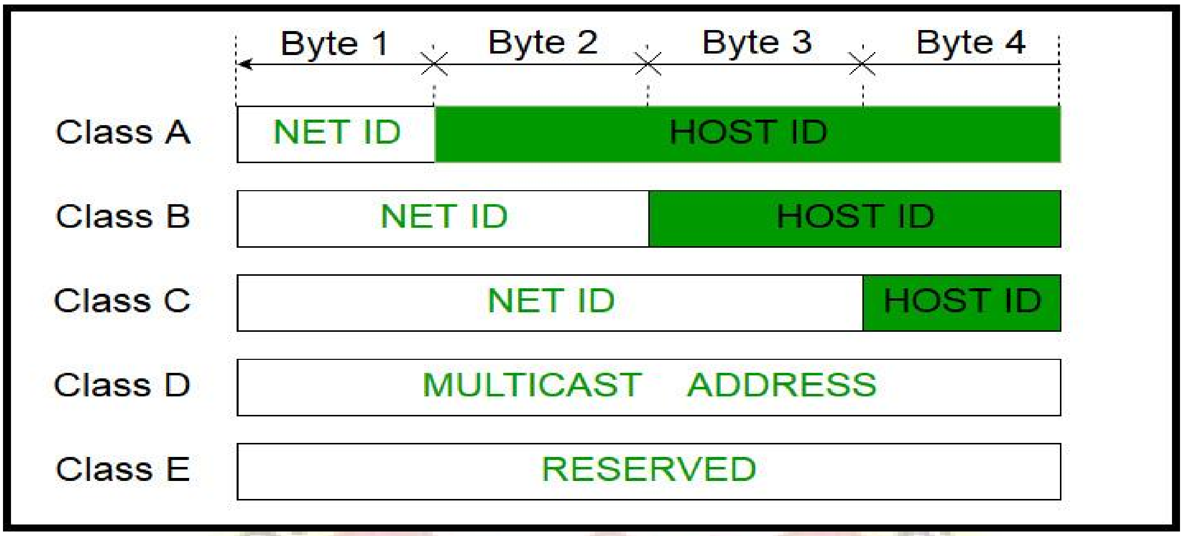
- Class A
- Class B
- Class C
- Class D
- Class E

Each of these classes has a valid range of IP addresses. Classes D and E are reserved for multicast and experimental purposes respectively. The order of bits in the first octet determine the classes of IP address.

IPv4 address is divided into two parts:

- **Network ID**
- **Host ID**

The class of IP address is used to determine the bits used for network ID and host ID and the number of total networks and hosts possible in that particular class. Each ISP or network administrator assigns IP address to each device that is connected to its network.



IP addresses are globally managed by Internet Assigned Numbers Authority(IANA) and regional Internet registries(RIR).

Class A:

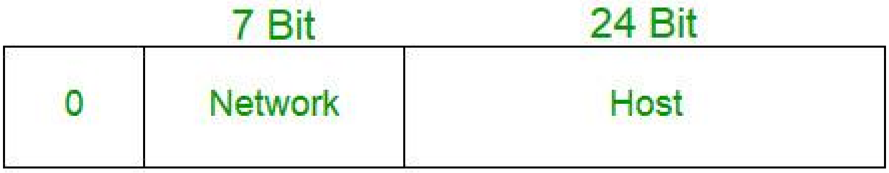
IP address belonging to class A are assigned to the networks that contain a large number of hosts.

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

The higher order bit of the first octet in class A is always set to 0. The remaining 7 bits in first octet are used to determine network ID. The 24 bits of host ID are used to determine the host in any network. The default sub-net mask for class A is 255.x.x.x. Therefore, class A has a total of:

- 2^7= 128 network ID
- 2^24 – 2 = 16,777,214 host ID

IP addresses belonging to class A ranges from 1.x.x.x – 126.x.x.x



Class A

Class B:

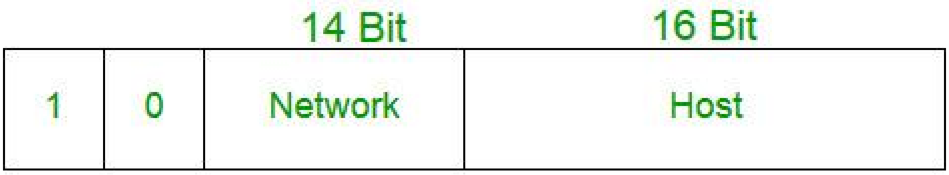
IP address belonging to class B are assigned to the networks that ranges from medium-sized to large-sized networks.

- The network ID is 16 bits long.
- The host ID is 16 bits long.

The higher order bits of the first octet of IP addresses of class B are always set to 10. The remaining 14 bits are used to determine network ID. The 16 bits of host ID is used to determine the host in any network. The default sub-net mask for class B is 255.255.x.x. Class B has a total of:

- 2^14 = 16384 network address
- 2^16 – 2 = 65534 host address

IP addresses belonging to class B ranges from 128.0.x.x – 191.255.x.x.



Class B

Class C:

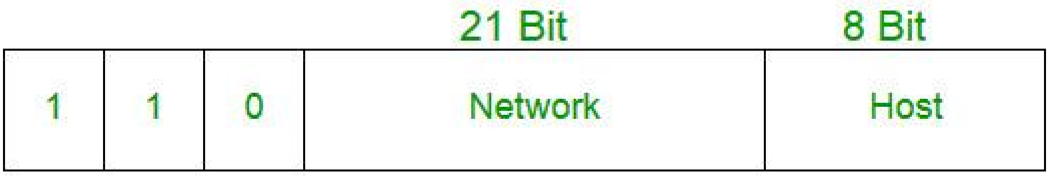
IP address belonging to class C are assigned to small-sized networks.

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

The higher order bits of the first octet of IP addresses of class C are always set to 110. The remaining 21 bits are used to determine network ID. The 8 bits of host ID is used to determine the host in any network. The default sub-net mask for class C is 255.255.255.x. Class C has a total of:

- 2^21 = 2097152 network address
- 2^8 – 2 = 254 host address

IP addresses belonging to class C ranges from 192.0.0.x – 223.255.255.x.

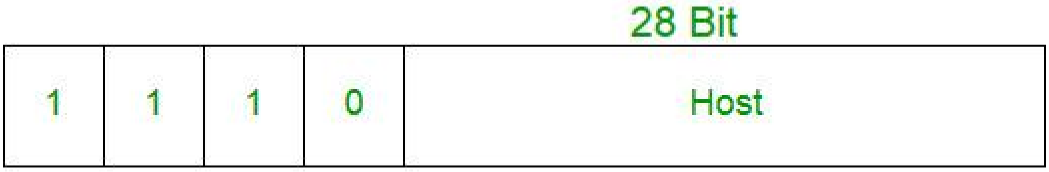


Class C

Class D:

IP address belonging to class D are reserved for multi-casting. The higher order bits of the first octet of IP addresses belonging to class D are always set to 1110. The remaining bits are for the address that interested hosts recognize.

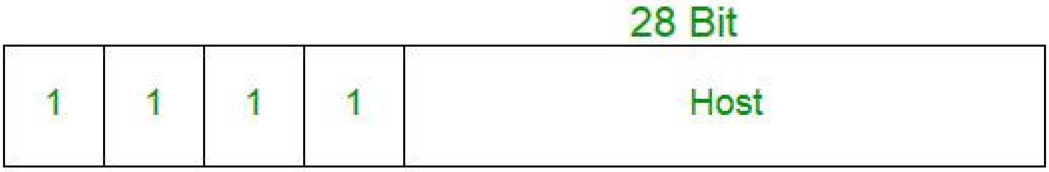
Class D does not posses any sub-net mask. IP addresses belonging to class D ranges from 224.0.0.0 – 239.255.255.255.



Class D

Class E:

IP addresses belonging to class E are reserved for experimental and research purposes. IP addresses of class E ranges from 240.0.0.0 – 255.255.255.254. This class doesn’t have any sub-net mask. The higher order bits of first octet of class E are always set to 1111.



Class E

Range of special IP addresses:

169.254.0.0 – 169.254.0.16 : Link local addresses
127.0.0.0 – 127.0.0.8 : Loop-back addresses
0.0.0.0 – 0.0.0.8 : used to communicate within the current network.

4.6 Internet Control Protocol

- IP packets use logical (host to host) addresses and need to be encapsulated in a frame with the help of physical (node-to-node) addresses.
- Some protocols are needed to create mapping between physical and logical addresses.

Static Mapping

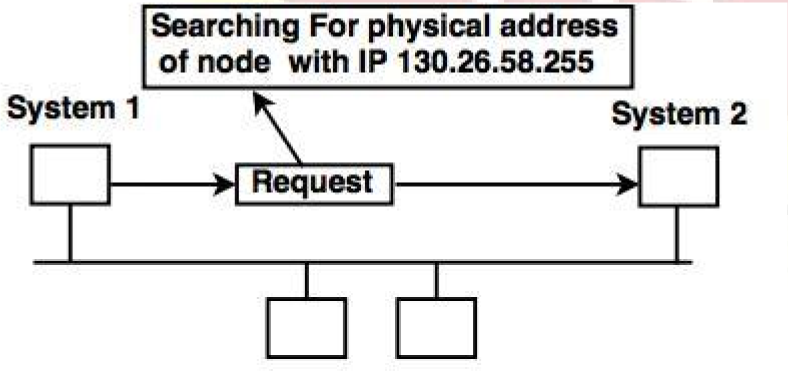
- It creates a table that associates a logical address with a physical address.
- This address is stored on each machine in the network.
- Each machine has an IP address of another machine but not its physical address. Hence, physical addresses are usually seen in the table.

Dynamic Mapping

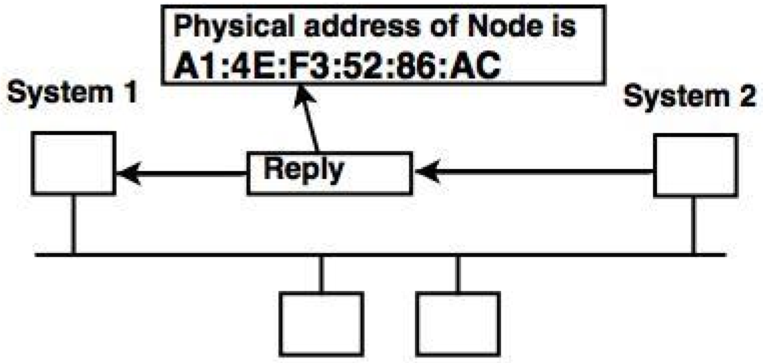
In this mapping, each machine knows one of the two addresses (logical or physical address) and tries to find the other one.

Address Resolution Protocol (ARP)

- Host or router has an IP address and needs to send another host or router (it has the logical (IP) address of the receiver).
- The logical address is obtained from the routing table, if the sender is a router.
- But, the IP datagram is encapsulated in a frame, which is able to pass through the physical network. This means that the sender needs the physical address of the receiver.
- The host or the router sends an ARP query packet (packet contains the physical and IP addresses of the sender and the IP address of the receiver).
- Considering that, the sender does not know the physical address of the receiver, the query is broadcast over the network.
- Every host or router on the network receives and processes the **ARP query packet**, but only the desired recipient recognizes its IP address and **sends back ARP response** (response packet contains the recipients IP and physical addresses).
- The packet is **unicasted** directly to the inquirer by using the physical address which is received in the query packet.

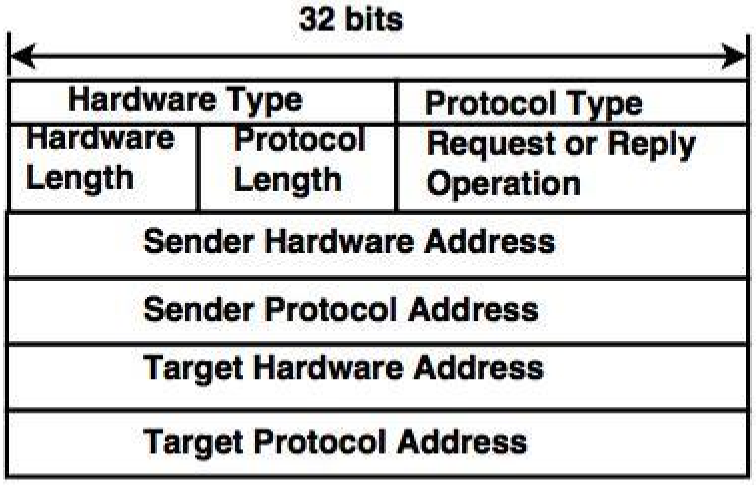

Searching For physical address of node with IP 130.26.58.255
System 1
System 2
Request

ARP Request Broadcast.



ARP Reply Unicast.

ARP Packet Format



32 bits

Hardware Type		Protocol Type
Hardware Length	Protocol Length	Request or Reply Operation
Sender Hardware Address		
Sender Protocol Address		
Target Hardware Address		
Target Protocol Address		

ARP Packet

1. Hardware type

This is 16 bit field used to define the type of the network on which ARP is running.

2. Protocol Length

This is 16 bit length used to define the protocol. For example, the value of this field in IPv4 is 0800H.

3. Hardware length

This is 8 bit field used to define the length of physical address in bytes. This value is 6 for ethernet.

4. Protocol Length

This is 8 bit field used to define the length of logical address in bytes. This value is 4 for IPv4.

5. Operation

This is 16 bit field used to define a type of packet; ARP reply or request.

6. Sender Hardware Length

This is a variable length field used to define the physical address of the sender.

7. Sender Protocol Address

This is a variable length field used to define the logical address of the sender. This field is 4 bytes long for IP protocol.

8. Target Hardware Address

This is a variable length field used to define the physical address of the target. This field is 6 bytes long for ethernet. For ARP request message, this field is '0' because the sender does not know the physical address of the target.

9. Target Protocol Address

This is a variable length used to define the logical address of the target. This is 4 byte long for the IPv4 protocol.

UNIT -V

5.1 Transport Layer Design Issues

The transport layer delivers the message from one process to another process running on two different hosts. Thus, it has to perform number of functions to ensure the accurate delivery of message. The various functions of transport layer are:

- Establishing, Maintaining & Releasing Connection
- Addressing
- Data Transfer
- Flow Control
- Error Control
- Congestion Control

Establishing, Maintaining & Releasing Connection:

The transport layer establishes, maintains & releases end-to-end transport connection on the request of upper layers. Establishing a connection involves allocation of buffers for storing user data, synchronizing the sequence numbers of packets etc. A connection is released at the request of upper layer.

Addressing:

In order to deliver the message from one process to another, an addressing scheme is required. Several process may be running on a system at a time. In order to identify the correct process out of the various running processes, transport layer uses an addressing scheme called port number. Each process has a specific port number.

Data Transfer:

Transport layer breaks user data into smaller units and attaches a transport layer header to each unit forming a TPDU (Transport Layer Data Unit). The TPDU is handed over to the network layer for its delivery to destination. The TPDU header contains port number, sequence number, acknowledgement number, checksum and other fields.

Flow Control:

Like data link layer, transport layer also performs flow control. However, flow control at transport layer is performed end-to-end rather than node-to-node. Transport Layer uses a sliding window protocol to perform flow control.

Error Control:

Transport layer also provides end-to-end error control facility. Transport layer deals with several different types of errors:

Error due to damaged bits.

Error due to non delivery of TPDUs.

Error due to duplicate delivery of TPDUs.

Error due to delivery of TPDU to a wrong destination.

5.2 Connection Management

3 Way Handshake-

Three Way Handshake is a process used for establishing a TCP connection.

Consider-

- Client wants to establish a connection with the server.

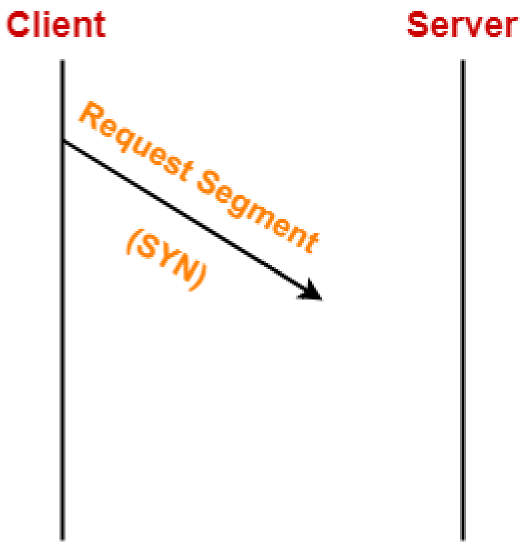
- Before Three Way Handshake, both client and server are in closed state.

TCP Handshake involves the following steps in establishing the connection-

Step-01: SYN-

For establishing a connection,

- Client sends a request segment to the server.
- Request segment consists only of TCP Header with an empty payload.
- Then, it waits for a reply segment from the server.



Request segment contains the following information in TCP header-

1. Initial sequence number
2. SYN bit set to 1
3. Maximum segment size
4. Receiving window size

1. Initial Sequence Number-

Client sends the initial sequence number to the server.

- It is contained in the sequence number field.

- It is a randomly chosen 32 bit value.

2. SYN Bit Set To 1-

Client sets SYN bit to 1 which indicates the server-

- This segment contains the initial sequence number used by the client.
- It has been sent for synchronizing the sequence numbers.

3. Maximum Segment Size (MSS)-

Client sends its MSS to the server.

- It dictates the size of the largest data chunk that client can send and receive from the server.
- It is contained in the Options field.

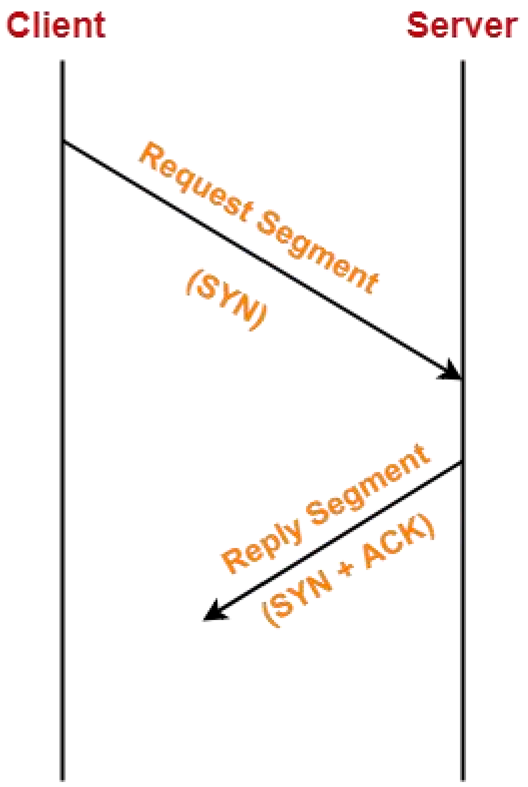
4. Receiving Window Size-

- Client sends its receiving window size to the server.
- It dictates the limit of unacknowledged data that can be sent to the client.
- It is contained in the window size field.

Step-02: SYN + ACK-

After receiving the request segment,

- Server responds to the client by sending the reply segment.
- It informs the client of the parameters at the server side.



Reply segment contains the following information in TCP header-

1. Initial sequence number
2. SYN bit set to 1
3. Maximum segment size
4. Receiving window size
5. Acknowledgment number
6. ACK bit set to 1

1. Initial Sequence Number-

- Server sends the initial sequence number to the client.
- It is contained in the sequence number field.
- It is a randomly chosen 32 bit value.

2. SYN Bit Set To 1-

Server sets SYN bit to 1 which indicates the client-

- This segment contains the initial sequence number used by the server.
- It has been sent for synchronizing the sequence numbers.

3. Maximum Segment Size (MSS)-

- Server sends its MSS to the client.
- It dictates the size of the largest data chunk that server can send and receive from the client.
- It is contained in the Options field.

4. Receiving Window Size-

- Server sends its receiving window size to the client.
- It dictates the limit of unacknowledged data that can be sent to the server.
- It is contained in the window size field.

5. Acknowledgement Number-

- Server sends the initial sequence number incremented by 1 as an acknowledgement number.
- It dictates the sequence number of the next data byte that server expects to receive from the client.

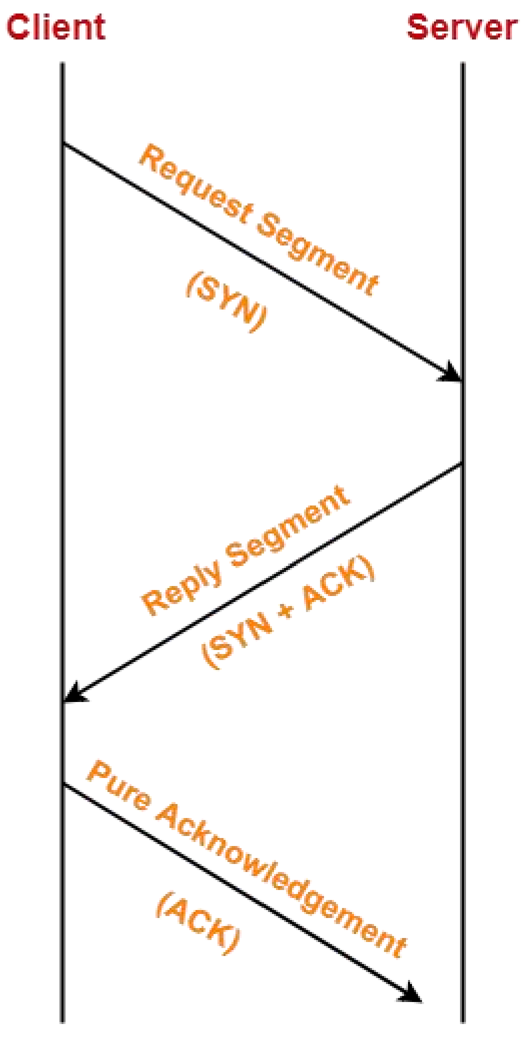
6. ACK Bit Set To 1-

- Server sets ACK bit to 1.
- It indicates the client that the acknowledgement number field in the current segment is valid.

Step-03: ACK-

After receiving the reply segment,

- Client acknowledges the response of server.
- It acknowledges the server by sending a pure acknowledgement.

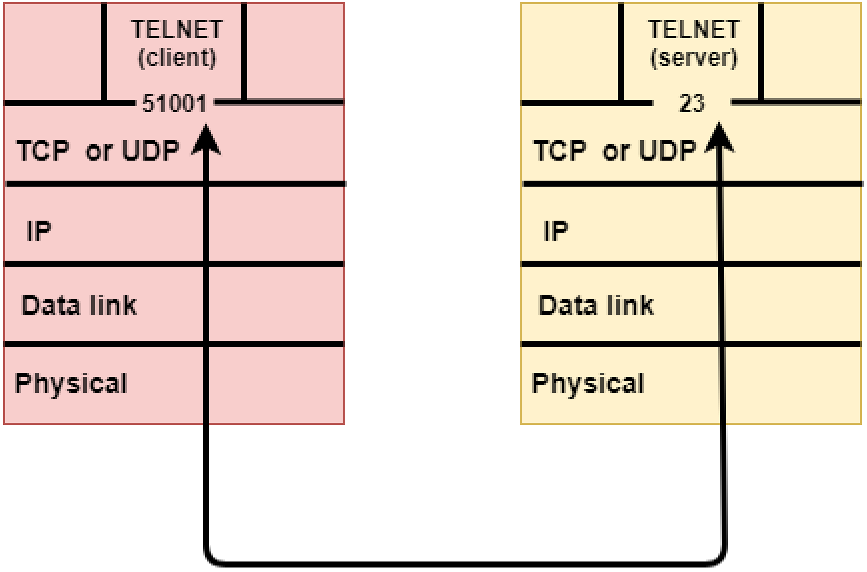


3 Way Handshake

5.3Transport 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.

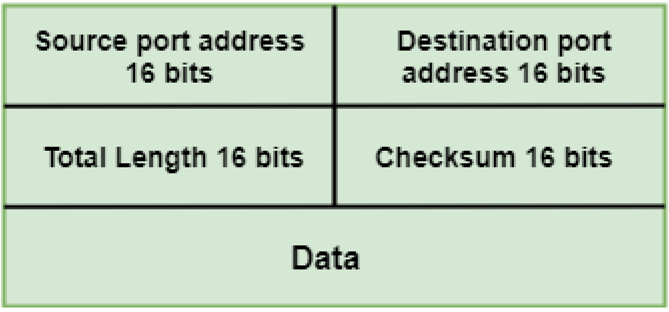


UDP

- 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.

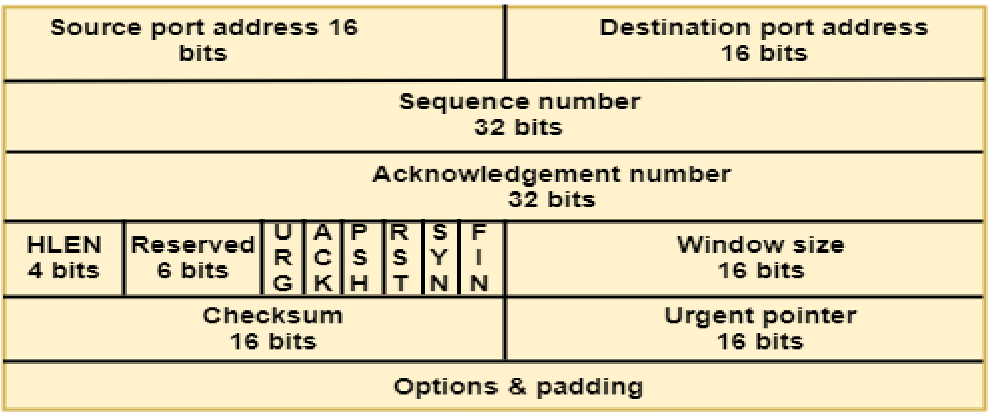
TCP

- 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



<table>
<tr><td colspan="8">Source port address 16
bits</td><td>Destination port address
16 bits</td></tr>
<tr><td colspan="9">Sequence number
32 bits</td></tr>
<tr><td colspan="9">Acknowledgement number
32 bits</td></tr>
<tr><td>HLEN
4 bits</td><td>Reserved
6 bits</td><td>U
R
G</td><td>A
C
K</td><td>P
S
H</td><td>R
S
T</td><td>S
Y
N</td><td>F
I
N</td><td>Window size
16 bits</td></tr>
<tr><td colspan="8">Checksum
16 bits</td><td>Urgent pointer
16 bits</td></tr>
<tr><td colspan="9">Options &amp; padding</td></tr>
</table>

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.

5.4 Network Security

It refers to the measures taken by any enterprise or organisation to secure its computer network and data using both hardware and software systems. This aims at securing the confidentiality and accessibility of the data and network. Every company or organisation that handles large amount of data, has a degree of solutions against many **cyber threats**.

The most basic example of Network Security is password protection where the user of the network oneself chooses. In the recent times, Network Security has become the central topic of cyber security with many organisations inviting applications of people who have skills in this area. The network security solutions protect various **vulnerabilities of the computer systems** such as:

1. Users

2. Locations

3. Data

4. Devices

5. Applications

Network Security:

The basic principle of network security is protecting huge stored data and network in layers that ensures a bedding of rules and regulations that have to be acknowledged before

performing any activity on the data.
These levels are:

1. Physical

2. Technical

3. Administrative

1. **Physical Network Security:**

   This is the most basic level that includes protecting the data and network though unauthorized personnel from acquiring the control over the confidentiality of the network. These includes external peripherals and routers might be used for cable connections. The same can be achieved by using devices like bio-metric systems.

2. **Technical Network Security:**

   It primarily focusses on protecting the data stored in the network or data involved in transitions through the network. This type serves two purposes. One, protection from the unauthorized users and the other being protection from malicious activities.

3. **Administrative Network Security:**

   This level of network security protects user behavior like how the permission has been granted and how the authorization process takes place. This also ensures the level of sophistication the network might need for protecting it through all the attacks. This level also suggests necessary amendments that have to be done over the infrastructure.

Types of Network Security:

The few types of network securities are discussed as below:

1. Access Control**:**

   Not every person should have complete allowance to the accessibility to the network or its data. The one way to examine this is by going through each personnel's details. This is done through Network Access Control which ensures that only a handful of authorized personnel must be able to work with allowed number of resources.

2. Antivirus **and Anti-malware Software:**

   This type of network security ensures that any malicious software does not enter the network and jeopardize the security of the data. The malicious software like Viruses, Trojans, Worms are handled by the same. This ensure that not only the entry of the malware is protected but also that the system is well equipped to fight once it has entered.

3. **Cloud Security:**

   Now a day, a lot many organisations are joining hands with the cloud technology where a large amount of important data is stored over the internet. This is very vulnerable to the malpractices that few unauthorized dealers might pertain Many businesses embrace SaaS applications for providing some of its employees the allowance of accessing the data stored over the cloud. This type of security ensures in creating gaps in visibility of the data.

5.7 Cryptography

It is technique of securing information and communications through use of codes so that only those person for whom the information is intended can understand it and process it. Thus preventing unauthorized access to information. The prefix "crypt" means "hidden" and suffix graphy means "writing". The term **cryptography** is a Greek word which means "secret writing".

In Cryptography the techniques which are use to protect information are obtained from mathematical concepts and a set of rule based calculations known as algorithms to convert messages in ways that make it hard to decode it. These algorithms are used for cryptographic key generation, digital signing, verification to protect data privacy, web browsing on internet and to protect confidential transactions such as credit card and debit card transactions.

Features of Cryptography are as follows:

1. **Confidentiality:**

   Information can only be accessed by the person for whom it is intended and no other person except him can access it.

2. **Integrity:**

   Information cannot be modified in storage or transition between sender and intended receiver without any addition to information being detected.

3. **Non-repudiation:**

   The creator/sender of information cannot deny his or her intention to send information at later stage.

4. **Authentication:**

   The identities of sender and receiver are confirmed. As well as destination/origin of information is confirmed.

The terminology used in cryptography is given below:

1. **Plaintext.** The original message or data that is fed into the algorithm as input is called plaintext.

2**. Encryption algorithm**. The encryption algorithm is the algorithm that performs various substitutions and transformations on the plaintext. Encryption is the process of changing plaintext into cipher text.

3. **Ciphertext.** Ciphertext is the encrypted form the message. It is the scrambled message produced as output. It depends upon the plaintext and the key.

4. **Decryption algorithm.** The process of changing Ciphertext into plain text is known as decryption. Decryption algorithm is essentially the encryption algorithm run in reverse. It takes the Ciphertext and the key and produces the original plaintext.

5. **Key**. It also acts as input to the encryption algorithm. The exact substitutions and transformations performed by the algorithm depend on the key. Thus a key is a number or a set of number that the algorithm uses to perform encryption and decryption.

Types of Cryptography:

In general there are three types of cryptography:

Symmetric Key Cryptography:

It is an encryption system where the sender and receiver of message use a single common key to encrypt and decrypt messages. Symmetric Key Systems are faster and simpler but the problem is that sender and receiver have to somehow exchange key in a secure manner. The most popular symmetric key cryptography system is Data Encryption System(DES).

1. **Hash Functions:**

   There is no usage of any key in this algorithm. A hash value with fixed length is calculated as per the plain text which makes it impossible for contents of plain text to be recovered. Many operating systems use hash functions to encrypt passwords.

2. **Asymmetric Key Cryptography:**

   Under this system a pair of keys is used to encrypt and decrypt information. A public key is used for encryption and a private key is used for decryption. Public key and Private Key are different. Even if the public key is known by everyone the intended receiver can only decode it because he alone knows the private key.

****