ANNAI WOMEN'S COLLEGE

(Arts & Science)

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Course Material

Paper Name : Computer Networks

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No.Of Units : 5 units

COMPUTER NETWORKS

UNIT-I

PHYSICAL LAYER

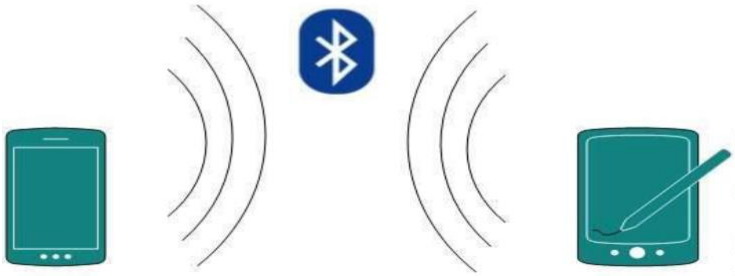
Data Communication

Data communications refers to the transmission of this digital data between two or more computers and a computer network or data network is a telecommunications network that allows computers to exchange data. The physical connection between networked computing devices is established using either cable media or wireless media. The best-known computer network is the Internet.

NETWORK TYPES

Personal Area Network

A Personal Area Network (PAN) is smallest network which is very personal to a user. This may include Bluetooth enabled devices or infra-red enabled devices. PAN has connectivity range up to 10 meters. PAN may include wireless computer keyboard and mouse, Bluetooth enabled headphones, wireless printers and TV remotes.

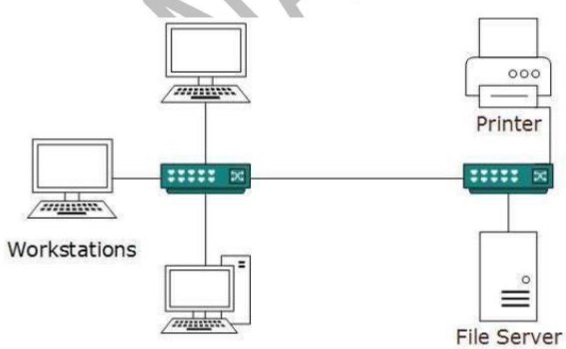


For example, Piconet is Bluetooth-enabled Personal Area Network which may contain up to 8 devices connected together in a master-slave fashion.

Local Area Network

A computer network spanned inside a building and operated under single administrative system is generally termed as Local Area Network (LAN). Usually,LAN covers an organization' offices, schools, colleges or universities. Number of systems connected in LAN may vary from as least as two to as much as 16 million.

LAN provides a useful way of sharing the resources between end users.The resources such as printers, file servers, scanners, and internet are easily sharable among computers.



LANs are composed of inexpensive networking and routing equipment. It may contains local servers serving file storage and other locally shared applications. It mostly operates on private IP addresses and does not involve heavy routing. LAN works under its own local domain and controlled centrally.

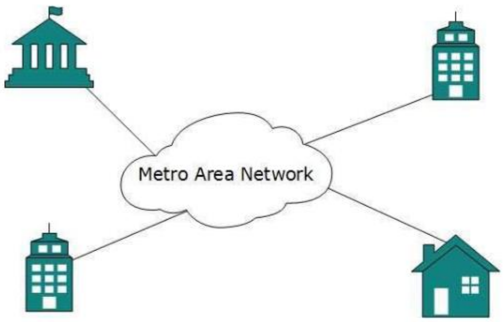
LAN uses either Ethernet or Token-ring technology. Ethernet is most widely employed LAN technology and uses Star topology, while Token-ring is rarely seen.

LAN can be wired,wireless, or in both forms at once.

Metropolitan Area Network

The Metropolitan Area Network (MAN) generally expands throughout a city such as cable TV network. It can be in the form of Ethernet,Token-ring, ATM, or Fiber Distributed Data Interface (FDDI).

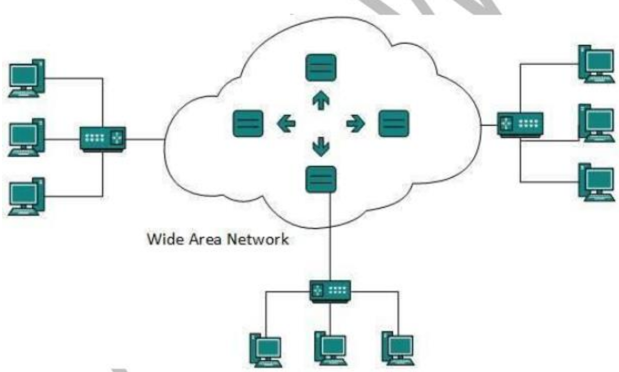
Metro Ethernet is a service which is provided by ISPs. This service enables its users to expand their Local Area Networks. For example, MAN can help an organization to connect all of its offices in a city.



Backbone of MAN is high-capacity and high-speed fiber optics. MAN works in between Local Area Network and Wide Area Network. MAN provides uplink for LANs to WANs or internet.

Wide Area Network

As the name suggests,the Wide Area Network (WAN) covers a wide area which may span across provinces and even a whole country. Generally, telecommunication networks are Wide Area Network. These networks provide connectivity to MANs and LANs. Since they are equipped with very high speed backbone, WANs use very expensive network equipment.



WAN may use advanced technologies such as Asynchronous Transfer Mode (ATM), Frame Relay, and Synchronous Optical Network (SONET). WAN may be managed by multiple administration.

Internetwork

A network of networks is called an internetwork, or simply the internet. It is the largest network in existence on this planet.The internet hugely connects all WANs and it can have connection to LANs and Home networks. Internet uses TCP/IP protocol suite and uses IP as its addressing protocol. Present day, Internet is widely implemented using IPv4. Because of shortage of address spaces, it is gradually migrating from IPv4 to IPv6.

Internet enables its users to share and access enormous amount of information worldwide. It uses WWW, FTP, email services, audio and video streaming etc. At huge level, internet works on Client-Server model.

Internet uses very high speed backbone of fiber optics. To inter-connect various continents, fibers are laid under sea known to us as submarine communication cable.

Internet is widely deployed on World Wide Web services using HTML linked pages and is accessible by client software known as Web Browsers. When a user requests a page using some web browser located on some Web Server anywhere in the world, the Web Server responds with the proper HTML page. The communication delay is very low.

Internet is serving many proposes and is involved in many aspects of life. Some of them are:

- Web sites
- E-mail
- Instant Messaging
- Blogging
- Social Media
- Marketing
- Networking
- Resource Sharing
- Audio and Video Streaming

NETWORK MODELS

The OSI Model - Features, Principles and Layers

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

Feature of OSI Model

1. Big picture of communication over network is understandable through this OSI model.
2. We see how hardware and software work together.
3. We can understand new technologies as they are developed.
4. Troubleshooting is easier by separate networks.
5. Can be used to compare basic functional relationships on different networks.

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 unwieldy.

Functions of Different Layers

Following are the functions performed by each layer of the OSI model. This is just an introduction, we will cover each layer in details in the coming tutorials.

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.

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.

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.

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.

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.

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.

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.

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.

MULTIPLEXING

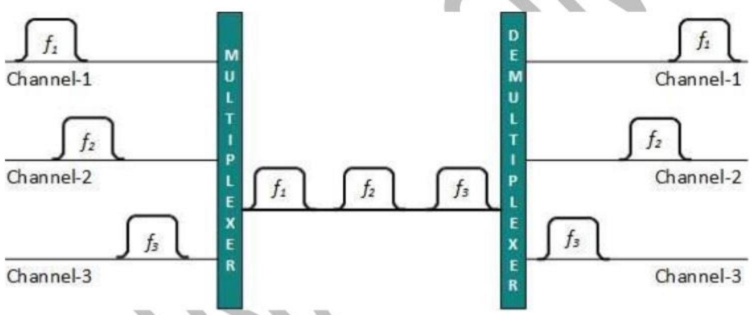
Multiplexing is a technique by which different analog and digital streams of transmission can be simultaneously processed over a shared link. Multiplexing divides the high capacity medium into low capacity logical medium which is then shared by different streams.

Communication is possible over the air (radio frequency), using a physical media (cable), and light (optical fiber). All mediums are capable of multiplexing.

When multiple senders try to send over a single medium, a device called Multiplexer divides the physical channel and allocates one to each. On the other end of communication, a De-multiplexer receives data from a single medium, identifies each, and sends to different receivers.

Frequency Division Multiplexing

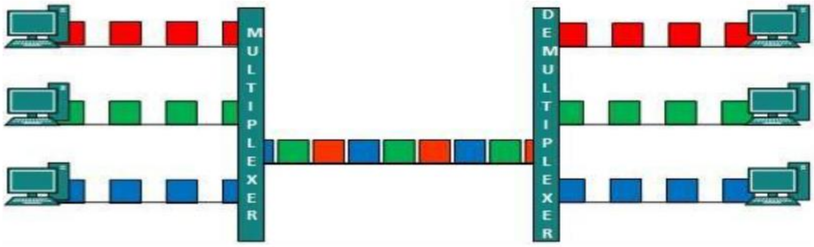
When the carrier is frequency, FDM is used. FDM is an analog technology. FDM divides the spectrum or carrier bandwidth in logical channels and allocates one user to each channel. Each user can use the channel frequency independently and has exclusive access of it. All channels are divided in such a way that they do not overlap with each other. Channels are separated by guard bands. Guard band is a frequency which is not used by either channel.



Time Division Multiplexing

TDM is applied primarily on digital signals but can be applied on analog signals as well. In TDM the shared channel is divided among its user by means of time slot. Each user can transmit data within the provided time slot only. Digital signals are divided in frames, equivalent to time slot i.e. frame of an optimal size which can be transmitted in given time slot.

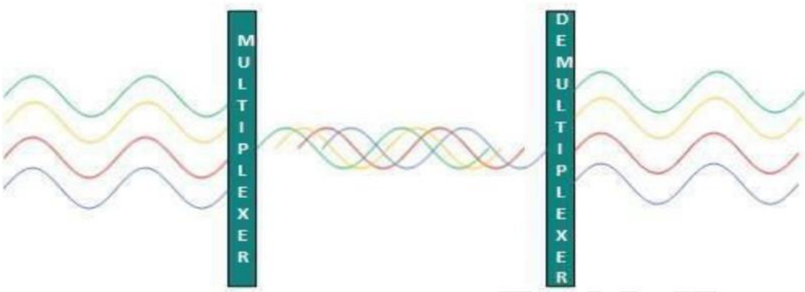
TDM works in synchronized mode. Both ends, i.e. Multiplexer and De-multiplexer are timely synchronized and both switch to next channel simultaneously.



When channel A transmits its frame at one end,the De-multiplexer provides media to channel A on the other end.As soon as the channel A's time slot expires, this side switches to channel B. On the other end, the De-multiplexer works in a synchronized manner and provides media to channel B. Signals from different channels travel the path in interleaved manner.

Wavelength Division Multiplexing

Light has different wavelength (colors). In fiber optic mode, multiple optical carrier signals are multiplexed into an optical fiber by using different wavelengths. This is an analog multiplexing technique and is done conceptually in the same manner as FDM but uses light as signals.



Further, on each wavelength time division multiplexing can be incorporated to accommodate more data signals.

Code Division Multiplexing

Multiple data signals can be transmitted over a single frequency by using Code Division Multiplexing. FDM divides the frequency in smaller channels but CDM allows its users to full bandwidth and transmit signals all the time using a unique code. CDM uses orthogonal codes to spread signals.

Each station is assigned with a unique code, called chip. Signals travel with these codes independently, inside the whole bandwidth.The receiver knows in advance the chip code signal it has to receive.

SPREAD SPECTRUM

A collective class of signaling techniques are employed before transmitting a signal to provide a secure communication, known as the **Spread Spectrum Modulation**. The main advantage of spread spectrum communication technique is to prevent "interference" whether it is intentional or unintentional.

The signals modulated with these techniques are hard to interfere and cannot be jammed. An intruder with no official access is never allowed to crack them. Hence, these techniques are used for military purposes. These spread spectrum signals transmit at low power density and has a wide spread of signals.

These are of two types.

- Frequency Hopped Spread Spectrum (FHSS)
- Direct Sequence Spread Spectrum (DSSS)

Frequency Hopped Spread Spectrum (FHSS)

This is frequency hopping technique, where the users are made to change the frequencies of usage, from one to another in a specified time interval, hence called as **frequency hopping**. For example, a frequency was allotted to sender 1 for a particular period of time. Now, after a while, sender 1 hops to the other frequency and sender 2 uses the first frequency, which was previously used by sender 1. This is called as **frequency reuse**.

The frequencies of the data are hopped from one to another in order to provide a secure transmission. The amount of time spent on each frequency hop is called as **Dwell time**.

Direct Sequence Spread Spectrum (DSSS)

Whenever a user wants to send data using this DSSS technique, each and every bit of the user data is multiplied by a secret code, called as **chipping code**. This chipping code is nothing but the spreading code which is multiplied with the original message and transmitted. The receiver uses the same code to retrieve the original message.

Comparison between FHSS and DSSS/CDMA

Both the spread spectrum techniques are popular for their characteristics. To have a clear understanding, let us take a look at their comparisons.

FHSS	DSSS / CDMA
Multiple frequencies are used	Single frequency is used
Hard to find the user's frequency at any instant of time	User frequency, once allotted is always the same
Frequency reuse is allowed	Frequency reuse is not allowed
Sender need not wait	Sender has to wait if the spectrum is busy
Power strength of the signal is high	Power strength of the signal is low
Stronger and penetrates through the obstacles	It is weaker compared to FHSS

It is never affected by interference	It can be affected by interference
It is cheaper	It is expensive
This is the commonly used technique	This technique is not frequently used

Advantages of Spread Spectrum

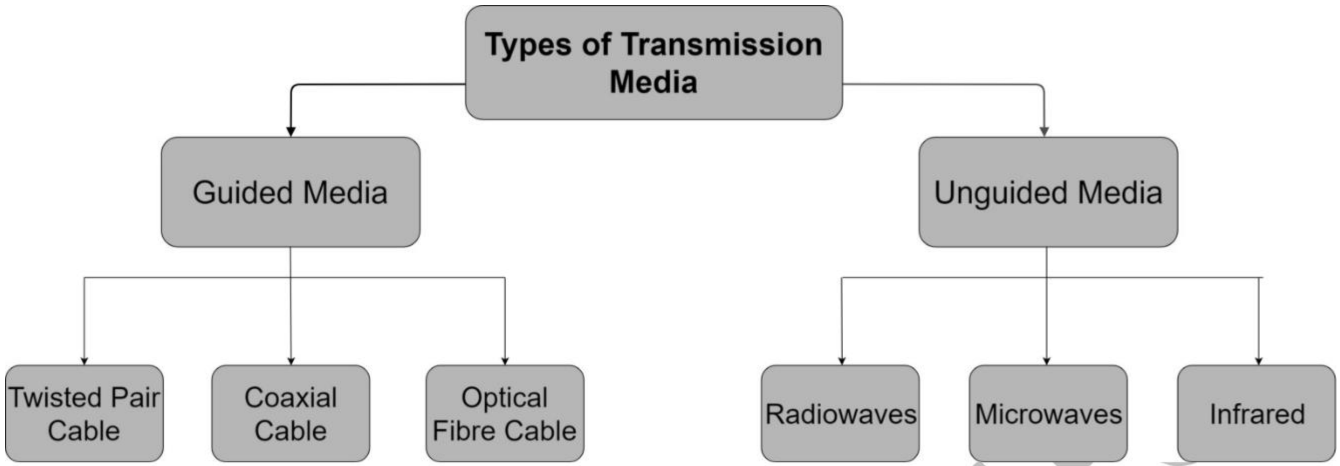
Following are the advantages of spread spectrum −

- Cross-talk elimination
- Better output with data integrity
- Reduced effect of multipath fading
- Better security
- Reduction in noise
- Co-existence with other systems
- Longer operative distances
- Hard to detect
- Not easy to demodulate/decode
- Difficult to jam the signals

Although spread spectrum techniques were originally designed for military uses, they are now being used widely for commercial purpose.

Types of Transmission Media

In data communication terminology, a transmission medium is a physical path between the transmitter and the receiver i.e it is the channel through which data is sent from one place to another. Transmission Media is broadly classified into the following types:



1. Guided Media:

It is also referred to as Wired or Bounded transmission media. Signals being transmitted are directed and confined in a narrow pathway by using physical links. Features:

- High Speed
- Secure
- Used for comparatively shorter distances

There are 3 major types of Guided Media:

(i) Twisted Pair Cable –

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

1. **Unshielded Twisted Pair (UTP):**
   This type of cable has the ability to block interference and does not depend on a physical shield for this purpose. It is used for telephonic applications.
   Advantages:
   - Least expensive
   - Easy to install
   - High speed capacity

   Disadvantages:
   - Susceptible to external interference
   - Lower capacity and performance in comparison to STP
   - Short distance transmission due to attenuation
2. **Shielded Twisted Pair (STP):**
   This type of cable consists of a special jacket to block external interference. It is used in fast-data-rate Ethernet and in voice and data channels of telephone lines.
   Advantages:
   - Better performance at a higher data rate in comparison to UTP
   - Eliminates crosstalk
   - Comparitively faster

   Disadvantages:

- Comparitively difficult to install and manufacture
- More expensive
- Bulky

(ii) Coaxial Cable –

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

Advantages:

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

Disadvantages:

- Single cable failure can disrupt the entire network

(iii) Optical Fibre Cable –

It uses the concept of reflection of light through a core made up of glass or plastic. The core is surrounded by a less dense glass or plastic covering called the cladding. It is used for transmission of large volumes of data.

Advantages:

- Increased capacity and bandwidth
- Light weight
- Less signal attenuation
- Immunity to electromagnetic interference
- Resistance to corrosive materials

Disadvantages:

- Difficult to install and maintain
- High cost
- Fragile
- unidirectional, ie, will need another fibre, if we need bidirectional communication

2. Unguided Media:

It is also referred to as Wireless or Unbounded transmission media.No physical medium is required for the transmission of electromagnetic signals.

Features:

- Signal is broadcasted through air
- Less Secure
- Used for larger distances

There are 3 major types of Unguided Media:

(i) Radiowaves –

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

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

(ii) Microwaves –
It is a line of sight transmission i.e. the sending and receiving antennas need to be properly aligned with each other. The distance covered by the signal is directly proportional to the height of the antenna. Frequency Range:1GHz – 300GHz. These are majorly used for mobile phone communication and television distribution.
(iii) Infrared –
Infrared waves are used for very short distance communication. They cannot penetrate through obstacles. This prevents interference between systems. Frequency Range:300GHz – 400THz. It is used in TV remotes, wireless mouse, keyboard, printer, etc.

SWITCHING

Switching is process to forward packets coming in from one port to a port leading towards the destination. When data comes on a port it is called ingress, and when data leaves a port or goes out it is called egress. A communication system may include number of switches and nodes. At broad level, switching can be divided into two major categories:

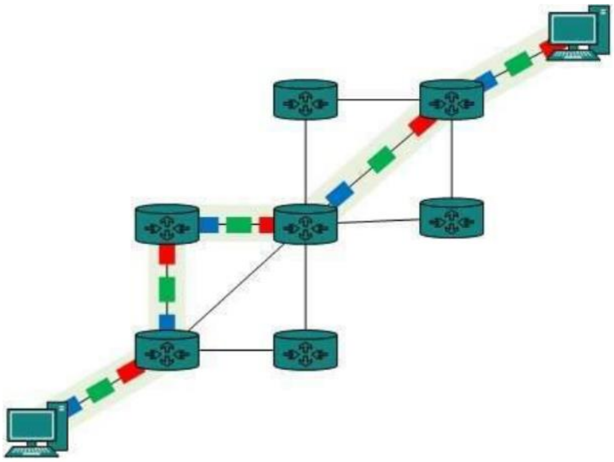
- **Connectionless:** The data is forwarded on behalf of forwarding tables. No previous handshaking is required and acknowledgements are optional.
- **Connection Oriented:** Before switching data to be forwarded to destination, there is a need to pre-establish circuit along the path between both endpoints. Data is then forwarded on that circuit. After the transfer is completed, circuits can be kept for future use or can be turned down immediately.

Circuit Switching

When two nodes communicate with each other over a dedicated communication path, it is called circuit switching.There 'is a need of pre-specified route from which data will travels and no other data is permitted.In circuit switching, to transfer the data, circuit must be established so that the data transfer can take place.

Circuits can be permanent or temporary. Applications which use circuit switching may have to go through three phases:

- Establish a circuit
- Transfer the data
- Disconnect the circuit

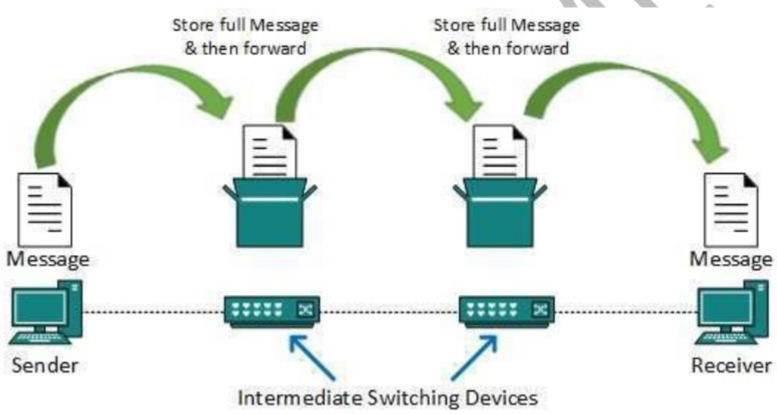


Circuit switching was designed for voice applications. Telephone is the best suitable example of circuit switching. Before a user can make a call, a virtual path between caller and callee is established over the network.

Message Switching

This technique was somewhere in middle of circuit switching and packet switching. In message switching, the whole message is treated as a data unit and is switching / transferred in its entirety.

A switch working on message switching, first receives the whole message and buffers it until there are resources available to transfer it to the next hop. If the next hop is not having enough resource to accommodate large size message, the message is stored and switch waits.



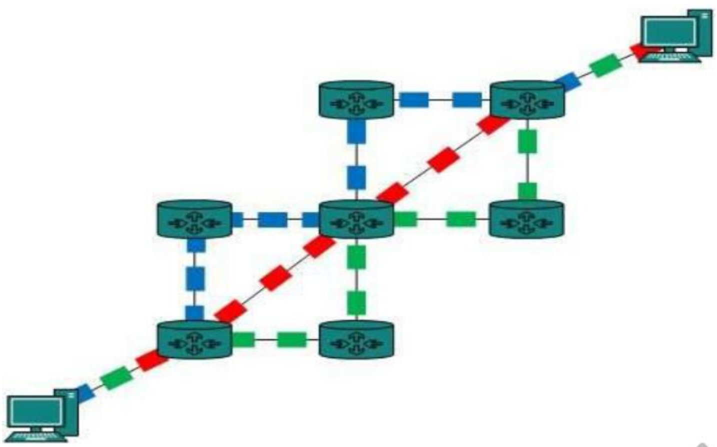
This technique was considered substitute to circuit switching. As in circuit switching the whole path is blocked for two entities only. Message switching is replaced by packet switching. Message switching has the following drawbacks:

- Every switch in transit path needs enough storage to accommodate entire message.
- Because of store-and-forward technique and waits included until resources are available, message switching is very slow.
- Message switching was not a solution for streaming media and real-time applications.

Packet Switching

Shortcomings of message switching gave birth to an idea of packet switching. The entire message is broken down into smaller chunks called packets. The switching information is added in the header of each packet and transmitted independently.

It is easier for intermediate networking devices to store small size packets and they do not take much resources either on carrier path or in the internal memory of switches.



Packet switching enhances line efficiency as packets from multiple applications can be multiplexed over the carrier. The internet uses packet switching technique. Packet switching enables the user to differentiate data streams based on priorities. Packets are stored and forwarded according to their priority to provide quality of service.

UNIT-II

DATALINK LAYER

DATA LINK LAYER

Data Link Layer is second layer of OSI Layered Model. This layer is one of the most complicated layers and has complex functionalities and liabilities. Data link layer hides the details of underlying hardware and represents itself to upper layer as the medium to communicate.

Data link layer works between two hosts which are directly connected in some sense. This direct connection could be point to point or broadcast. Systems on broadcast network are said to be on same link. The work of data link layer tends to get more complex when it is dealing with multiple hosts on single collision domain.

Data link layer is responsible for converting data stream to signals bit by bit and to send that over the underlying hardware. At the receiving end, Data link layer picks up data from hardware which are in the form of electrical signals, assembles them in a recognizable frame format, and hands over to upper layer.

Data link layer has two sub-layers:

- **Logical Link Control:** It deals with protocols, flow-control, and error control
- **Media Access Control:** It deals with actual control of media

Functionality of Data-link Layer

Data link layer does many tasks on behalf of upper layer. These are:

- **Framing**

  Data-link layer takes packets from Network Layer and encapsulates them into Frames.Then, it sends each frame bit-by-bit on the hardware. At receiver' end, data link layer picks up signals from hardware and assembles them into frames.

- **Addressing**

  Data-link layer provides layer-2 hardware addressing mechanism. Hardware address is assumed to be unique on the link. It is encoded into hardware at the time of manufacturing.

- **Synchronization**

  When data frames are sent on the link, both machines must be synchronized in order to transfer to take place.

- **Error Control**

  Sometimes signals may have encountered problem in transition and the bits are flipped.These errors are detected and attempted to recover actual data bits. It also provides error reporting mechanism to the sender.

- **Flow Control**

  Stations on same link may have different speed or capacity. Data-link layer ensures flow control that enables both machine to exchange data on same speed.

- **Multi-Access** When host on the shared link tries to transfer the data, it has a high probability of collision. Data-link layer provides mechanism such as CSMA/CD to equip capability of accessing a shared media among multiple Systems.

Types of Errors

There may be three types of errors:

- **Single bit error**

  In a frame, there is only one bit, anywhere though, which is corrupt.

- **Multiple bits error**

Frame is received with more than one bits in corrupted state.

- **Burst error**

Frame contains more than1 consecutive bits corrupted.

Error control mechanism may involve two possible ways:

- **Error detection**
- **Error correction**

Error Detection

Errors in the received frames are detected by means of Parity Check and Cyclic Redundancy Check (CRC). In both cases, few extra bits are sent along with actual data to confirm that bits received at other end are same as they were sent. If the counter-check at receiver' end fails, the bits are considered corrupted.

Parity Check

One extra bit is sent along with the original bits to make number of 1s either even in case of even parity, or odd in case of odd parity.

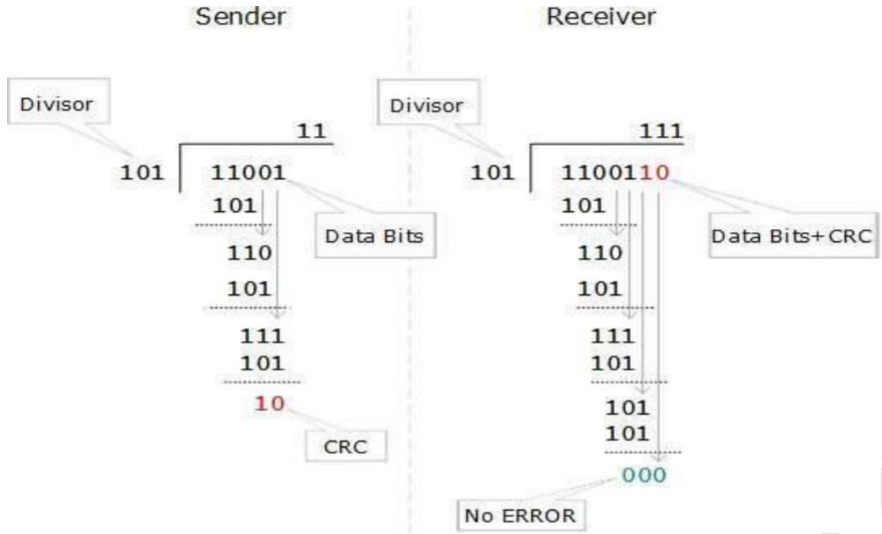
The sender while creating a frame counts the number of 1s in it. For example, if even parity is used and number of 1s is even then one bit with value 0 is added. This way number of 1s remains even.If the number of 1s is odd, to make it even a bit with value 1 is added.

The receiver simply counts the number of 1s in a frame. If the count of 1s is even and even parity is used, the frame is considered to be not-corrupted and is accepted. If the count of 1s is odd and odd parity is used, the frame is still not corrupted.

If a single bit flips in transit, the receiver can detect it by counting the number of 1s. But when more than one bits are erro neous, then it is very hard for the receiver to detect the error.

Cyclic Redundancy Check (CRC)

CRC is a different approach to detect if the received frame contains valid data. This technique involves binary division of the data bits being sent. The divisor is generated using polynomials. The sender performs a division operation on the bits being sent and calculates the remainder. Before sending the actual bits, the sender adds the remainder at the end of the actual bits. Actual data bits plus the remainder is called a codeword. The sender transmits data bits as codewords.



At the other end, the receiver performs division operation on codewords using the same CRC divisor. If the remainder contains all zeros the data bits are accepted, otherwise it is considered as there some data corruption occurred in transit.

Error Correction

In the digital world, error correction can be done in two ways:

- **Backward Error Correction** When the receiver detects an error in the data received, it requests back the sender to retransmit the data unit.
- **Forward Error Correction** When the receiver detects some error in the data received, it executes error-correcting code, which helps it to auto-recover and to correct some kinds of errors.

The first one, Backward Error Correction, is simple and can only be efficiently used where retransmitting is not expensive. For example, fiber optics. But in case of wireless transmission retransmitting may cost too much. In the latter case, Forward Error Correction is used.

To correct the error in data frame, the receiver must know exactly which bit in the frame is corrupted. To locate the bit in error, redundant bits are used as parity bits for error detection.For example, we take ASCII words (7 bits data), then there could be 8 kind of information we need: first seven bits to tell us which bit is error and one more bit to tell that there is no error.

For m data bits, r redundant bits are used. r bits can provide 2r combinations of information. In m+r bit codeword, there is possibility that the r bits themselves may get corrupted. So the number of r bits used must inform about m+r bit locations plus no-error information, i.e. m+r+1.

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

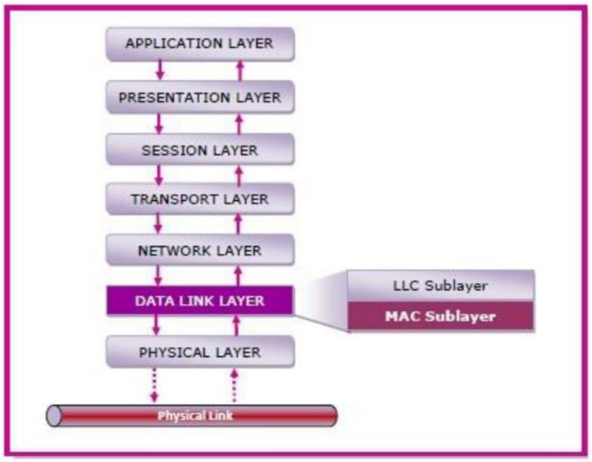
The medium access control (MAC) is a sublayer of the data link layer of the open system interconnections (OSI) reference model for data transmission. It is responsible for flow control and multiplexing for transmission medium. It controls the transmission of data packets via remotely shared channels. It sends data over the network interface card.

MAC Layer

The Open System Interconnections (OSI) model is a layered networking framework that conceptualizes how communications should be done between heterogeneous systems. The data link layer is the second lowest layer. It is divided into two sublayers −

- The logical link control (LLC) sublayer
- The medium access control (MAC) sublayer

The following diagram depicts the position of the MAC layer −



Functions of MAC Layer

- It provides an abstraction of the physical layer to the LLC and upper layers of the OSI network.
- It is responsible for encapsulating frames so that they are suitable for transmission via the physical medium.
- It resolves the addressing of source station as well as the destination station, or groups of destination stations.
- It performs multiple access resolutions when more than one data frame is to be transmitted. It determines the channel access methods for transmission.
- It also performs collision resolution and initiating retransmission in case of collisions.
- It generates the frame check sequences and thus contributes to protection against transmission errors.

MAC Addresses

MAC address or media access control address is a unique identifier allotted to a network interface controller (NIC) of a device. It is used as a network address for data transmission within a network segment like Ethernet, Wi-Fi, and Bluetooth.

MAC address is assigned to a network adapter at the time of manufacturing. It is hardwired or hard-coded in the network interface card (NIC). A MAC address comprises of six groups of two hexadecimal digits, separated by hyphens, colons, or no separators. An example of a MAC address is 00:0A:89:5B:F0:11.

Random Access

In this, all stations have same superiority that is no station has more priority than another station. Any station can send data depending on medium's state(idle or busy). It has two features:

1. There is no fixed time for sending data
2. There is no fixed sequence of stations sending data

The Random access protocols are further subdivided as:

(a) ALOHA – It was designed for wireless LAN but is also applicable for shared medium. In this, multiple stations can transmit data at the same time and can hence lead to collision and data being garbled.

- **Pure Aloha:**
  When a station sends data it waits for an acknowledgement. If the acknowledgement doesn't come within the allotted time then the station waits for a random amount of time called back-off time (Tb) and re-sends the data. Since different stations wait for different amount of time, the probability of further collision decreases.
- **Slotted Aloha:**
  It is similar to pure aloha, except that we divide time into slots and sending of data is allowed only at the beginning of these slots. If a station misses out the allowed time, it must wait for the next slot. This reduces the probability of collision.

Controlled Access

In controlled access, the stations seek information from one another to find which station has the right to send. It allows only one node to send at a time, to avoid collision of messages on shared medium. The three controlled-access methods are:

1. Reservation
2. Polling
3. Token Passing

Reservation

- In the reservation method, a station needs to make a reservation before sending data.
- The time line has two kinds of periods:
  1. Reservation interval of fixed time length
  2. Data transmission period of variable frames.
- If there are M stations, the reservation interval is divided into M slots, and each station has one slot.
- Suppose if station 1 has a frame to send, it transmits 1 bit during the slot 1. No other station is allowed to transmit during this slot.
- In general, i^{th} station may announce that it has a frame to send by inserting a 1 bit into i^{th} slot. After all N slots have been checked, each station knows which stations wish to transmit.
- The stations which have reserved their slots transfer their frames in that order.
- After data transmission period, next reservation interval begins.
- Since everyone agrees on who goes next, there will never be any collisions.

The following figure shows a situation with five stations and a five slot reservation frame. In the first interval, only stations 1, 3, and 4 have made reservations. In the second interval, only station 1 has made a reservation.

Polling

- Polling process is similar to the roll-call performed in class. Just like the teacher, a controller sends a message to each node in turn.
- In this, one acts as a primary station(controller) and the others are secondary stations. All data exchanges must be made through the controller.
- The message sent by the controller contains the address of the node being selected for granting access.
- Although all nodes receive the message but the addressed one responds to it and sends data, if any. If there is no data, usually a “poll reject”(NAK) message is sent back.
- Problems include high overhead of the polling messages and high dependence on the reliability of the controller.

Token Passing

- In token passing scheme, the stations are connected logically to each other in form of ring and access of stations is governed by tokens.
- A token is a special bit pattern or a small message, which circulate from one station to the next in the some predefined order.
- In Token ring, token is passed from one station to another adjacent station in the ring whereas incase of Token bus, each station
  uses the bus to send the token to the next station in some predefined order.
- In both cases, token represents permission to send. If a station has a frame queued for transmission when it receives the token, it can send that frame before it passes the token to the next station. If it has no queued frame, it passes the token simply.
- After sending a frame, each station must wait for all N stations (including itself) to send the token to their neighbors and the other N – 1 stations to send a frame, if they have one.
- There exists problems like duplication of token or token is lost or insertion of new station, removal of a station, which need be tackled for correct and reliable operation of this scheme.

Performance

Performance of token ring can be concluded by 2 parameters:-

1. **Delay**, which is a measure of time between when a packet is ready and when it is delivered.So, the average time (delay) required to send a token to the next station = a/N.
2. **Throughput**, which is a measure of the successful traffic.

WIRELESS NETWORKS

Computer networks that are not connected by cables are called wireless networks. They generally use radio waves for communication between the network nodes. They allow devices to be connected to the network while roaming around within the network coverage.

Types of Wireless Networks

1. Wireless LANs: Connects two or more network devices using wireless distribution techniques.
2. Wireless MANs: Connects two or more wireless LANs spreading over a metropolitan area.

3. Wireless WANs: Connects large areas comprising LANs, MANs and personal networks.

Advantages of Wireless Networks

1. It provides clutter-free desks due to the absence of wires and cables.
2. It increases the mobility of network devices connected to the system since the devices need not be connected to each other.
3. Accessing network devices from any location within the network coverage or Wi-Fi hotspot becomes convenient since laying out cables is not needed.
4. Installation and setup of wireless networks are easier.
5. New devices can be easily connected to the existing setup since they needn't be wired to the present equipment. Also, the number of equipment that can be added or removed to the system can vary considerably since they are not limited by the cable capacity. This makes wireless networks very scalable.
6. Wireless networks require very limited or no wires. Thus, it reduces the equipment and setup costs.

Examples of wireless networks

1. Mobile phone networks
2. Wireless sensor networks
3. Satellite communication networks
4. Terrestrial microwave networks

BLUETOOTH

Bluetooth wireless technology is a short range communications technology intended to replace the cables connecting portable unit and maintaining high levels of security. Bluetooth technology is based on **Ad-hoc technology** also known as **Ad-hoc Pico nets**, which is a local area network with a very limited coverage.

History of Bluetooth

WLAN technology enables device connectivity to infrastructure based services through a wireless carrier provider. The need for personal devices to communicate wirelessly with one another without an established infrastructure has led to the emergence of **Personal Area Networks (PANs)**.

- Ericsson's Bluetooth project in 1994 defines the standard for PANs to enable communication between mobile phones using low power and low cost radio interfaces.
- In May 1988, Companies such as IBM, Intel, Nokia and Toshiba joined Ericsson to form the Bluetooth Special Interest Group (SIG) whose aim was to develop a defacto standard for PANs.
- IEEE has approved a Bluetooth based standard named IEEE 802.15.1 for Wireless Personal Area Networks (WPANs). IEEE standard covers MAC and Physical layer applications.

Bluetooth specification details the entire protocol stack. Bluetooth employs Radio Frequency (RF) for communication. It makes use of **frequency modulation** to generate radio waves in the **ISM** band.

Symbol of Bluetooth

An example of a Bluetooth device

The usage of Bluetooth has widely increased for its special features.

- Bluetooth offers a uniform structure for a wide range of devices to connect and communicate with each other.
- Bluetooth technology has achieved global acceptance such that any Bluetooth enabled device, almost everywhere in the world, can be connected with Bluetooth enabled devices.
- Low power consumption of Bluetooth technology and an offered range of up to ten meters has paved the way for several usage models.
- Bluetooth offers interactive conference by establishing an adhoc network of laptops.
- Bluetooth usage model includes cordless computer, intercom, cordless phone and mobile phones.

Piconets and Scatternets

Bluetooth enabled electronic devices connect and communicate wirelessly through shortrange devices known as **Piconets**. Bluetooth devices exist in small ad-hoc configurations with the ability to act either as master or slave the specification allows a mechanism for **master** and **slave** to switch their roles. Point to point configuration with one master and one slave is the simplest configuration.

When more than two Bluetooth devices communicate with one another, this is called a **PICONET**. A Piconet can contain up to seven slaves clustered around a single master. The device that initializes establishment of the Piconet becomes the **master**.

The master is responsible for transmission control by dividing the network into a series of time slots amongst the network members, as a part of **time division multiplexing**scheme which is shown below.

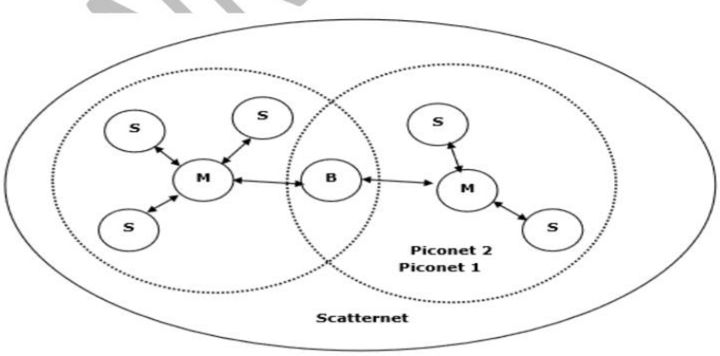


Figure: Piconets and Scatternets

The features of Piconets are as follows −

- Within a Piconet, the timing of various devices and the frequency hopping sequence of individual devices is determined by the clock and unique **48-bit address** of master.
- Each device can communicate simultaneously with up to seven other devices within a single Piconet.
- Each device can communicate with several piconets simultaneously.
- Piconets are established dynamically and automatically as Bluetooth enabled devices enter and leave piconets.
- There is no direct connection between the slaves and all the connections are essentially master-to-slave or slave-to-master.
- Slaves are allowed to transmit once these have been polled by the master.
- Transmission starts in the slave-to-master time slot immediately following a polling packet from the master.
- A device can be a member of two or more piconets, jumping from one piconet to another by adjusting the transmission regime-timing and frequency hopping sequence dictated by the master device of the second piconet.
- It can be a slave in one piconet and master in another. It however cannot be a master in more than once piconet.
- Devices resident in adjacent piconets provide a bridge to support inner-piconet connections, allowing assemblies of linked piconets to form a physically extensible communication infrastructure known as **Scatternet**.

CELLULAR NETWORK

Cellular network is an underlying technology for mobile phones, personal communication systems, wireless networking etc. The technology is developed for mobile radio telephone to replace high power transmitter/receiver systems. Cellular networks use lower power, shorter range and more transmitters for data transmission.

Features of Cellular Systems

Wireless Cellular Systems solves the problem of spectral congestion and increases user capacity. The features of cellular systems are as follows −

- Offer very high capacity in a limited spectrum.
- Reuse of radio channel in different cells.
- Enable a fixed number of channels to serve an arbitrarily large number of users by reusing the channel throughout the coverage region.
- Communication is always between mobile and base station (not directly between mobiles).
- Each cellular base station is allocated a group of radio channels within a small geographic area called a cell.
- Neighboring cells are assigned different channel groups.

- By limiting the coverage area to within the boundary of the cell, the channel groups may be reused to cover different cells.
- Keep interference levels within tolerable limits.
- Frequency reuse or frequency planning.
- Organization of Wireless Cellular Network.

Cellular network is organized into multiple low power transmitters each 100w or less.

Shape of Cells

The coverage area of cellular networks are divided into **cells**, each cell having its own antenna for transmitting the signals. Each cell has its own frequencies. Data communication in cellular networks is served by its base station transmitter, receiver and its control unit.

The shape of cells can be either square or hexagon −

Square

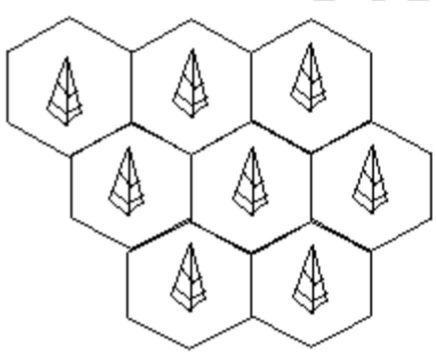
A square cell has four neighbors at distance **d** and four at distance Root **2 d**

- Better if all adjacent antennas equidistant
- Simplifies choosing and switching to new antenna

Hexagon

A hexagon cell shape is highly recommended for its easy coverage and calculations. It offers the following advantages −

- Provides equidistant antennas
- Distance from center to vertex equals length of side



Frequency Reuse

Frequency reusing is the concept of using the same radio frequencies within a given area, that are separated by considerable distance, with minimal interference, to establish communication.

Frequency reuse offers the following benefits −

- Allows communications within cell on a given frequency
- Limits escaping power to adjacent cells
- Allows re-use of frequencies in nearby cells
- Uses same frequency for multiple conversations
- 10 to 50 frequencies per cell

A satellite is an object that revolves around another object. For example, earth is a satellite of The Sun, and moon is a satellite of earth.

A **communication satellite** is a **microwave repeater station** in a space that is used for telecommunication, radio and television signals. A communication satellite processes the data coming from one earth station and it converts the data into another form and send it to the second earth station.

SATELLITE NETWORKS

How a Satellite Works

Two stations on earth want to communicate through radio broadcast but are too far away to use conventional means. The two stations can use a relay station for their communication. One earth station transmits the signal to the satellite.

Uplink frequency is the frequency at which ground station is communicating with satellite. The satellite transponder converts the signal and sends it down to the second earth station, and this is called **Downlink frequency**. The second earth station also communicates with the first one in the same way.

Advantages of Satellite

The advantages of Satellite Communications are as follows −

- The Coverage area is very high than that of terrestrial systems.
- The transmission cost is independent of the coverage area.
- Higher bandwidths are possible.

Disadvantages of Satellite

The disadvantages of Satellite Communications are as follows −

- Launching satellites into orbits is a costly process.
- The bandwidths are gradually used up.
- High propagation delay for satellite systems than the conventional terrestrial systems.

Satellite Communication Basics

The process of satellite communication begins at an **earth station**. Here an installation is designed to transmit and receive signals from a satellite in orbit around the earth. Earth stations send information to satellites in the form of high powered, high frequency (GHz range) signals.

The satellites **receive** and **retransmit** the signals back to earth where they are received by other earth stations in the coverage area of the satellite. **Satellite's footprint** is the area which receives a signal of useful strength from the satellite.

The transmission system from the earth station to the satellite through a channel is called the **uplink**. The system from the satellite to the earth station through the channel is called the **downlink**.

Satellite Frequency Bands

The satellite frequency bands which are commonly used for communication are the **Cband, Ku-band,** and **Ka-band**. C-band and Ku-band are the commonly used frequency spectrums by today's satellites.

It is important to note that there is an inverse relationship between frequency and wavelength i.e. when frequency increases, wavelength decreases this helps to understand the relationship between **antenna diameter** and **transmission frequency**. Larger antennas (satellite dishes) are necessary to gather the signal with increasing wavelength.

Earth Orbits

A satellite when launched into space, needs to be placed in certain orbit to provide a particular way for its revolution, so as to maintain accessibility and serve its purpose whether scientific, military or commercial. Such orbits which are assigned to satellites, with respect to earth are called as **Earth Orbits**. The satellites in these orbits are Earth Orbit Satellites.

The important kinds of Earth Orbits are −

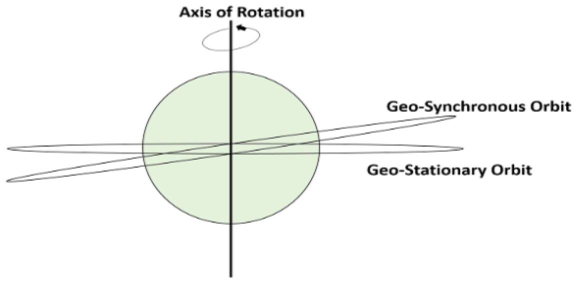
- Geo-synchronous Earth Orbit
- Geo-stationary Earth Orbit
- Medium Earth Orbit
- Low Earth Orbit

Geo-synchronous Earth Orbit (GEO) Satellites

A Geo-synchronous Earth orbit Satellite is one which is placed at an altitude of 22,300 miles above the Earth. This orbit is synchronized with a **side real day** (i.e., 23hours 56minutes). This orbit can **have inclination and eccentricity**. It may not be circular. This orbit can be tilted at the poles of the earth. But it appears stationary when observed from the Earth.

The same geo-synchronous orbit, if it is **circular** and in the plane of equator, it is called as geo-stationary orbit. These Satellites are placed at 35,900kms (same as geosynchronous) above the Earth's Equator and they keep on rotating with respect to earth's direction (west to east). These satellites are considered **stationary** with respect to earth and hence the name implies.

Geo-Stationary Earth Orbit Satellites are used for weather forecasting, satellite TV, satellite radio and other types of global communications.



The above figure shows the difference between Geo-synchronous and Geo- Stationary orbits. The Axis of rotation indicates the movement of Earth.

The main point to note here is that every Geo-Stationary orbit is a Geo-Synchronous orbit. But every Geo-Synchronous orbit is NOT a Geo-stationary orbit.

Medium Earth Orbit (MEO) Satellites

Medium earth orbit (MEO) satellite networks will orbit at distances of about 8000 miles from earth's surface. Signals transmitted from a MEO satellite travel a shorter distance. This translates to improved signal strength at the receiving end. This shows that smaller, more lightweight receiving terminals can be used at the receiving end.

Since the signal is travelling a shorter distance to and from the satellite, there is less transmission delay. **Transmission delay** can be defined as the time it takes for a signal to travel up to a satellite and back down to a receiving station.

For real-time communications, the shorter the transmission delay, the better will be the communication system. As an example, if a GEO satellite requires 0.25 seconds for a round trip, then MEO satellite requires less than 0.1 seconds to complete the same trip. MEOs operates in the frequency range of 2 GHz and above.

Low Earth Orbit (LEO) Satellites

The LEO satellites are mainly classified into three categories namely, little LEOs, big LEOs, and Mega-LEOs. LEOs will orbit at a distance of 500 to 1000 miles above the earth's surface.

This relatively short distance reduces transmission delay to only 0.05 seconds. This further reduces the need for sensitive and bulky receiving equipment. Little LEOs will operate in the 800 MHz (0.8 GHz) range. Big LEOs will operate in the 2 GHz or above range, and Mega-LEOs operates in the 20-30 GHz range.

The higher frequencies associated with **Mega-LEOs** translates into more information carrying capacity and yields to the capability of real-time, low delay video transmission scheme.

Network Devices (Hub, Repeater, Bridge, Switch, Router, Gateways and Brouter)

1. Repeater – A repeater operates at the physical layer. Its job is to regenerate the signal over the same network before the signal becomes too weak or corrupted so as to extend the length to which the signal can be transmitted over the same network. An important point to be noted about repeaters is that they do not amplify the signal. When the signal becomes weak, they copy the signal bit by bit and regenerate it at the original strength. It is a 2 port device.

2. **Hub** – A hub is basically a multiport repeater. A hub connects multiple wires coming from different branches, for example, the connector in star topology which connects different stations. Hubs cannot filter data, so data packets are sent to all connected devices. In other words, collision domain of all hosts connected through Hub remains one. Also, they do not have intelligence to find out best path for data packets which leads to inefficiencies and wastage.

Types of Hub

- **Active Hub :-** These are the hubs which have their own power supply and can clean , boost and relay the signal along the network. It serves both as a repeater as well as wiring center. These are used to extend maximum distance between nodes.
- **Passive Hub :-** These are the hubs which collect wiring from nodes and power supply from active hub. These hubs relay signals onto the network without cleaning and boosting them and can't be used to extend distance between nodes.

3. Bridge – A bridge operates at data link layer. A bridge is a repeater, with add on functionality of filtering content by reading the MAC addresses of source and destination. It is also used for interconnecting two LANs working on the same protocol. It has a single input and single output port, thus making it a 2 port device.

Types of Bridges

- **Transparent Bridges :-** These are the bridge in which the stations are completely unaware of the bridge's existence i.e. whether or not a bridge is added or deleted from the network , reconfiguration of the stations is unnecessary. These bridges makes use of two processes i.e. bridge forwarding and bridge learning.
- **Source Routing Bridges :-** In these bridges, routing operation is performed by source station and the frame specifies which route to follow. The hot can discover frame by sending a specical frame called discovery frame, which spreads through the entire network using all possible paths to destination.

4. **Switch** – A switch is a multi port bridge with a buffer and a design that can boost its efficiency(large number of ports imply less traffic) and performance. Switch is data link layer device. Switch can perform error checking before forwarding data, that makes it very efficient as it does not forward packets that have errors and forward good packets selectively to correct port only. In other words, switch divides collision domain of hosts, but broadcast domain remains same.

5. Routers – A router is a device like a switch that routes data packets based on their IP addresses. Router is mainly a Network Layer device. Routers normally connect LANs and WANs together and have a dynamically updating routing table based on which they make decisions on routing the data packets. Router divide broadcast domains of hosts connected through it. **6. Gateway** – A gateway, as the name suggests, is a passage to connect two networks together that may work upon different networking models. They basically works as the messenger agents that take data from one system, interpret it, and transfer it to another system. Gateways are also called protocol converters and can operate at any network layer. Gateways are generally more complex than switch or router.

7. Brouter – It is also known as bridging router is a device which combines features of both bridge and router. It can work either at data link layer or at network layer. Working as router, it is capable of routing packets across networks and working as bridge, it is capable of filtering local area network traffic.

UNIT-III

NETWORK LAYER

Network Layer

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

The main functions performed by the network layer are:

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

SWITCHING

Switching is process to forward packets coming in from one port to a port leading towards the destination. When data comes on a port it is called ingress, and when data leaves a port or goes out it is called egress. A communication system may include number of switches and nodes. At broad level, switching can be divided into two major categories:

- **Connectionless:** The data is forwarded on behalf of forwarding tables. No previous handshaking is required and acknowledgements are optional.

- **Connection Oriented:** Before switching data to be forwarded to destination, there is a need to pre-establish circuit along the path between both endpoints. Data is then forwarded on that circuit. After the transfer is completed, circuits can be kept for future use or can be turned down immediately.

Packet Switching

- In packet switching, messages are divided into packets of fixed or variable size.
- The size of packet is decided by the network and the governing protocol.
- Resource allocation for a packet is not done in packet switching.
- Resources are allocated on demand.
- The resource allocation is done on first-come, first-served basis.
- Each switching node has a small amount of buffer space to hold packets temporarily.
- If the outgoing line is busy, the packet stays in queue until the line becomes available.

Packet switching method uses two routing methods:

1. Datagram Packet Switching

- Datagram packet switching is normally implemented in the network layer.
- In datagram network, each packet is routed independently through the network.
- Each packet carries a header that contains the full information about the destination.
- When the switch receives the packet, the destination address in the header of the packet is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded.

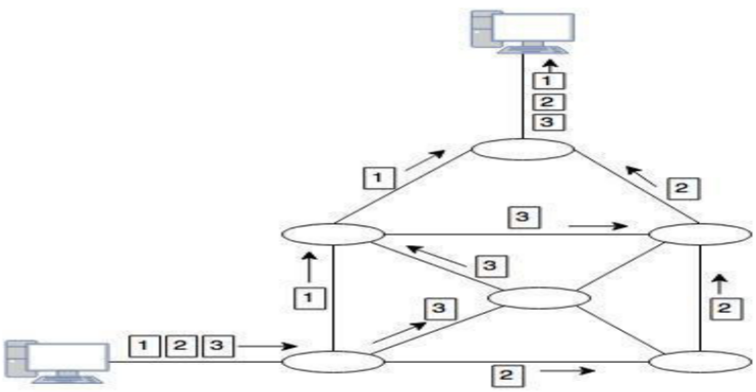


Fig: Datagram Packet Switching

2. Virtual Circuit Packet Switching

- Virtual circuit packet switching is normally done at the data link layer.
- Virtual circuit packet switching establishes a fixed path between a source and a destination to transfer the packets.
- It is also called as **connection oriented network.**

A source and destination have to go through three phases in a virtual circuit packet switching:

I. Setup phase
ii. Data transfer phase
iii. Connection release phase

- A logical connection is established when a sender sends a setup request to the receiver and the receiver sends back an acknowledgement to the sender if the receiver agree.
- All packets belonging to the same source and destination travel the same path.
- The information is delivered to the receiver in the same order as transmitted by the sender.

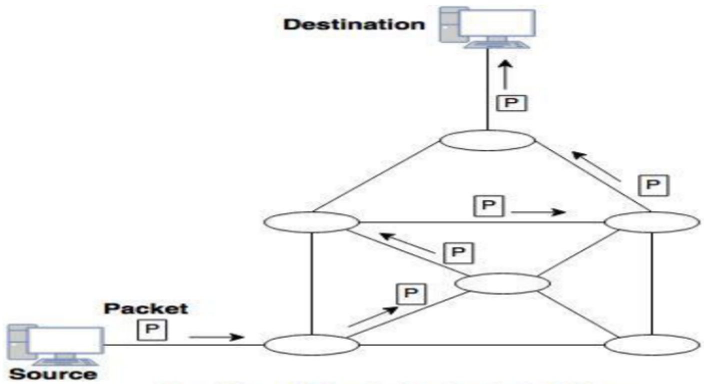


Fig: Virtual Circuit Packet Switching

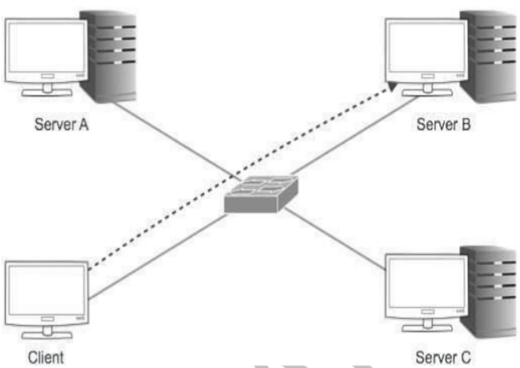
IPV4 ADDRESSING

The **IPv4** address is a 32-bit number that uniquely identifies a network interface on a system, as explained in How IP **Addresses** Apply to Network Interfaces. An **IPv4** address is written in decimal digits, divided into four 8-bit fields that are separated by periods. Each 8-bit field represents a byte of the **IPv4** address.

IPv4 supports three different types of addressing modes.

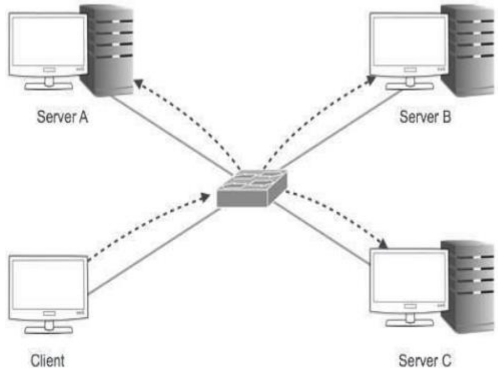
− **Unicast Addressing Mode**

In this mode, data is sent only to one destined host. The Destination Address field contains 32- bit IP address of the destination host. Here the client sends data to the targeted server −



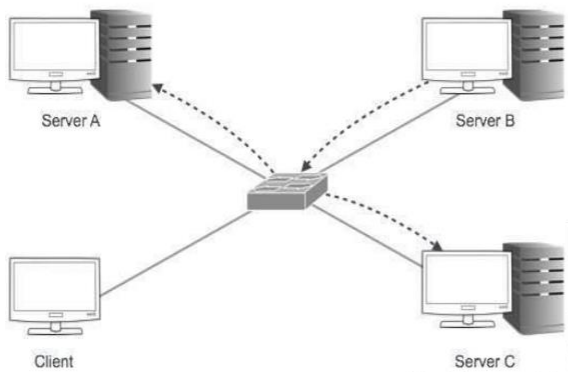
Broadcast Addressing Mode

In this mode, the packet is addressed to all the hosts in a network segment. The Destination Address field contains a special broadcast address, i.e. **255.255.255.255**. When a host sees this packet on the network, it is bound to process it. Here the client sends a packet, which is entertained by all the Servers −



Multicast Addressing Mode

This mode is a mix of the previous two modes, i.e. the packet sent is neither destined to a single host nor all the hosts on the segment. In this packet, the Destination Address contains a special address which starts with 224.x.x.x and can be entertained by more than one host.



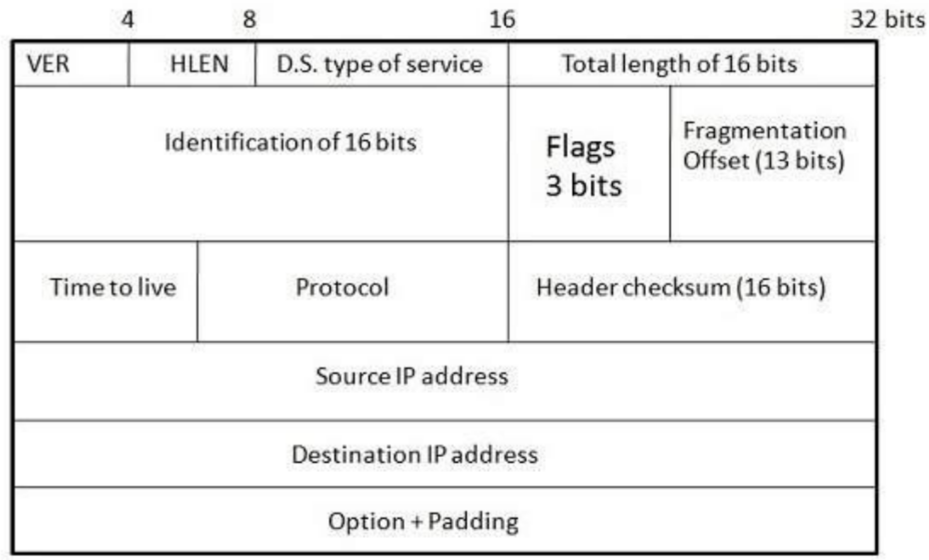
Here a server sends packets which are entertained by more than one servers. Every network has one IP address reserved for the Network Number which represents the network and one IP address reserved for the Broadcast Address, which represents all the hosts in that network.

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:



Points to remember:

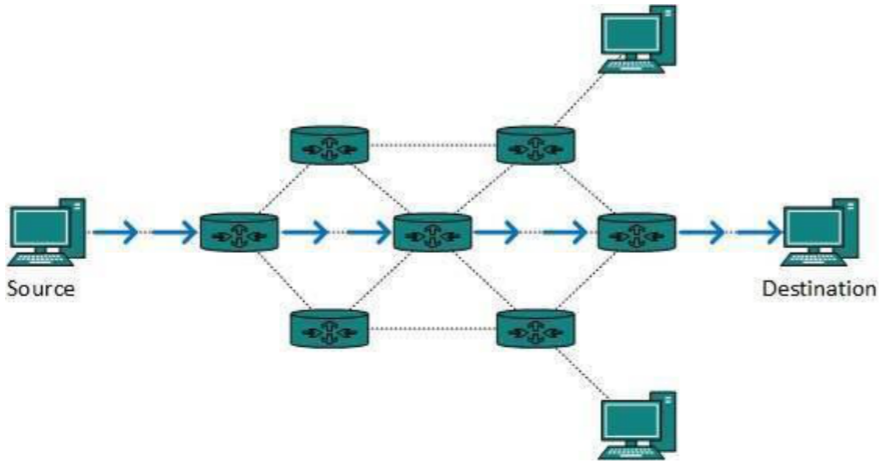
- The length of datagram is variable.
- The Datagram is divided into two parts: **header** and **data.**
- The length of header is 20 to 60 bytes.
- The header contains information for routing and delivery of the packet.

Routing

When a device has multiple paths to reach a destination, it always selects one path by preferring it over others. This selection process is termed as Routing. Routing is done by special network devices called routers or it can be done by means of software processes.

Unicast routing

Most of the traffic on the internet and intranets known as unicast data or unicast traffic is sent with specified destination. Routing unicast data over the internet is called unicast routing. It is the simplest form of routing because the destination is already known. Hence the router just has to look up the routing table and forward the packet to next hop.



Broadcast routing

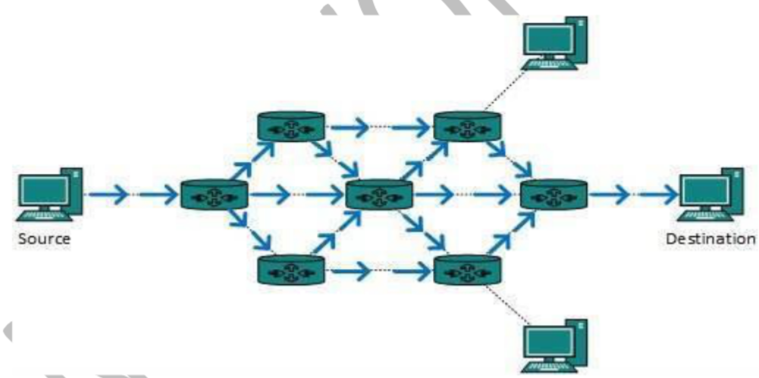
By default, the broadcast packets are not routed and forwarded by the routers on any network. Routers create broadcast domains. But it can be configured to forward broadcasts in some special cases. A broadcast message is destined to all network devices.

Broadcast routing can be done in two ways (algorithm):

- A router creates a data packet and then sends it to each host one by one. In this case, the router creates multiple copies of single data packet with different destination addresses. All packets are sent as unicast but because they are sent to all, it simulates as if router is broadcasting.

  This method consumes lots of bandwidth and router must destination address of each node.

- Secondly, when router receives a packet that is to be broadcasted, it simply floods those packets out of all interfaces. All routers are configured in the same way.

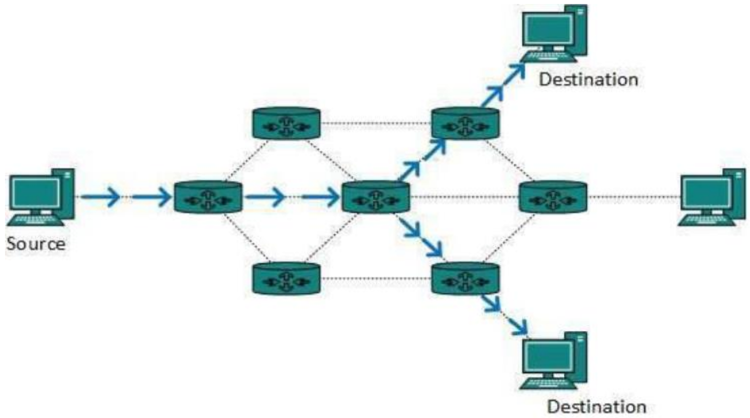


This method is easy on router's CPU but may cause the problem of duplicate packets received from peer routers.

Reverse path forwarding is a technique, in which router knows in advance about its predecessor from where it should receive broadcast. This technique is used to detect and discard duplicates.

Multicast Routing

Multicast routing is special case of broadcast routing with significance difference and challenges. In broadcast routing, packets are sent to all nodes even if they do not want it. But in Multicast routing, the data is sent to only nodes which wants to receive the packets.

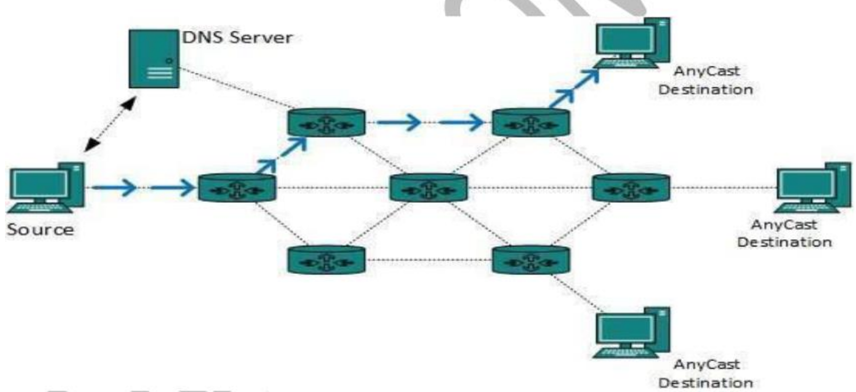


The router must know that there are nodes, which wish to receive multicast packets (or stream) then only it should forward. Multicast routing works spanning tree protocol to avoid looping.

Multicast routing also uses reverse path Forwarding technique, to detect and discard duplicates and loops.

Anycast Routing

Anycast packet forwarding is a mechanism where multiple hosts can have same logical address. When a packet destined to this logical address is received, it is sent to the host which is nearest in routing topology.



Anycast routing is done with help of DNS server. Whenever an Anycast packet is received it is enquired with DNS to where to send it. DNS provides the IP address which is the nearest IP configured on it.

Unicast Routing Protocols

There are two kinds of routing protocols available to route unicast packets:

- **Distance Vector Routing Protocol**

Distance Vector is simple routing protocol which takes routing decision on the number of hops between source and destination. A route with less number of hops is considered as the best route. Every router advertises its set best routes to other routers. Ultimately, all routers build up their network topology based on the advertisements of their peer routers,

For example Routing Information Protocol (RIP).

- **Link State Routing Protocol**

  Link State protocol is slightly complicated protocol than Distance Vector. It takes into account the states of links of all the routers in a network. This technique helps routes build a common graph of the entire network. All routers then calculate their best path for routing purposes.for example, Open Shortest Path First (OSPF) and Intermediate System to Intermediate System (ISIS).

Multicast Routing Protocols

Unicast routing protocols use graphs while Multicast routing protocols use trees, i.e. spanning tree to avoid loops. The optimal tree is called shortest path spanning tree.

- **DVMRP** - Distance Vector Multicast Routing Protocol
- **MOSPF** - Multicast Open Shortest Path First
- **CBT** - Core Based Tree
- **PIM** - Protocol independent Multicast

Protocol Independent Multicast is commonly used now. It has two flavors:

- **PIM Dense Mode**

  This mode uses source-based trees. It is used in dense environment such as LAN.

- **PIM Sparse Mode**

  This mode uses shared trees. It is used in sparse environment such as WAN.

Routing Algorithms

The routing algorithms are as follows:

Flooding

Flooding is simplest method packet forwarding. When a packet is received, the routers send it to all the interfaces except the one on which it was received. This creates too much burden on the network and lots of duplicate packets wandering in the network.

Time to Live (TTL) can be used to avoid infinite looping of packets. There exists another approach for flooding, which is called Selective Flooding to reduce the overhead on the network. In this method, the router does not flood out on all the interfaces, but selective ones.

Shortest Path

Routing decision in networks, are mostly taken on the basis of cost between source and destination. Hop count plays major role here. Shortest path is a technique which uses various algorithms to decide a path with minimum number of hops.

Common shortest path algorithms are:

- Dijkstra's algorithm
- Bellman Ford algorithm
- Floyd Warshall algorithm

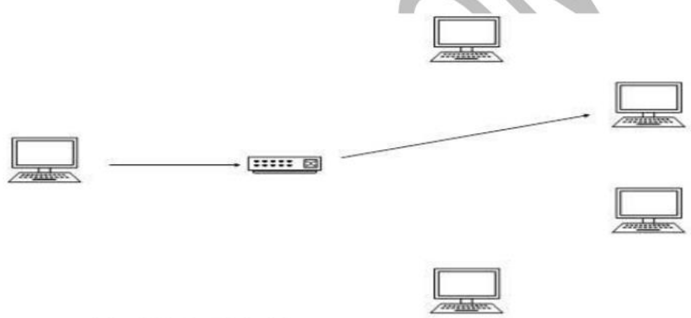
IPV6 ADDRESSING

An Internet Protocol Version 6 **address** (**IPv6 address**) is a numerical label that is used to identify a network interface of a computer or a network node participating in an **IPv6** computer network and for locating it in the network.

In computer networking, addressing mode refers to the mechanism of hosting an address on the network. IPv6 offers several types of modes by which a single host can be addressed. More than one host can be addressed at once or the host at the closest distance can be addressed.

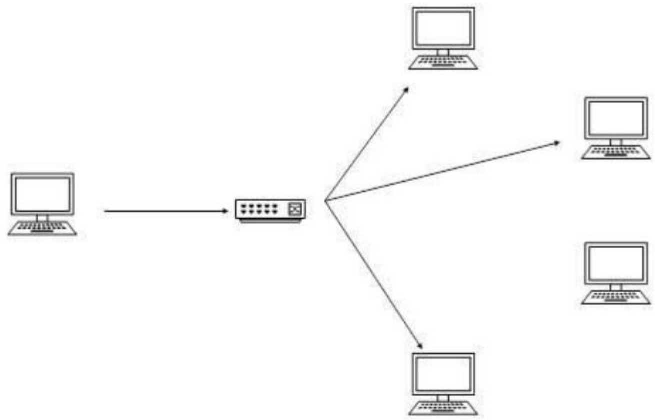
Unicast

In unicast mode of addressing, an IPv6 interface (host) is uniquely identified in a network segment. The IPv6 packet contains both source and destination IP addresses. A host interface is equipped with an IP address which is unique in that network segment.When a network switch or a router receives a unicast IP packet, destined to a single host, it sends out one of its outgoing interface which connects to that particular host.



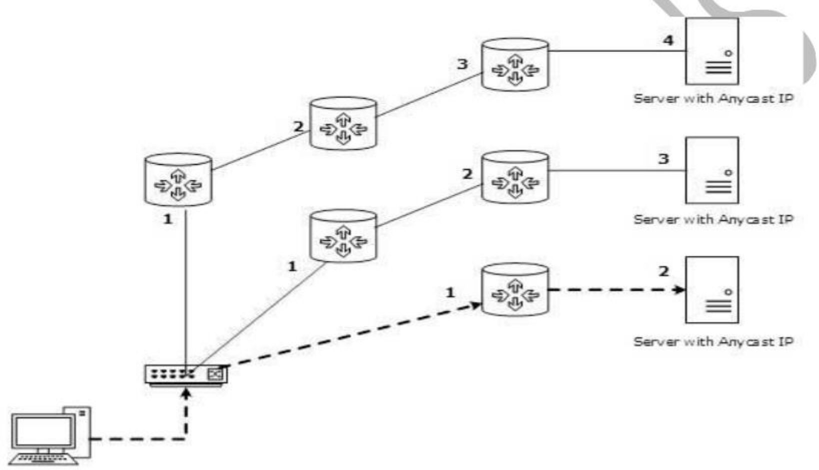
Multicast

The IPv6 multicast mode is same as that of IPv4. The packet destined to multiple hosts is sent on a special multicast address. All the hosts interested in that multicast information, need to join that multicast group first. All the interfaces that joined the group receive the multicast packet and process it, while other hosts not interested in multicast packets ignore the multicast information.



Anycast

IPv6 has introduced a new type of addressing, which is called Anycast addressing. In this addressing mode, multiple interfaces (hosts) are assigned same Anycast IP address. When a host wishes to communicate with a host equipped with an Anycast IP address, it sends a Unicast message. With the help of complex routing mechanism, that Unicast message is delivered to the host closest to the Sender in terms of Routing cost.



In the above picture, when a client computer tries to reach a server, the request is forwarded to the server with the lowest Routing Cost.

UNIT IV

TRANSPORT LAYER

Transport Layer protocols

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

UDP

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

User Datagram Format

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

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.

Differences b/w TCP & UDP

Basis for Comparison	**TCP**	**UDP**
Definition	TCP establishes a virtual circuit before transmitting the data.	UDP transmits the data directly to the destination computer without verifying whether the receiver is ready to receive or not.
Connection Type	It is a Connection-Oriented protocol	It is a Connectionless protocol
Speed	slow	high
Reliability	It is a reliable protocol.	It is an unreliable protocol.
Header size	20 bytes	8 bytes
acknowledgement	It waits for the acknowledgement of data and has the ability to resend the lost packets.	It neither takes the acknowledgement, nor it retransmits the damaged frame.

FLOW AND ERROR CONTROL

Flow Control: Flow control coordinates that amount of data that can be sent before receiving an acknowledgement.

- It is one of the most important duties of the data link layer.
- Flow control tells the sender how much data to send.
- It makes the sender wait for some sort of an acknowledgement (ACK) before continuing to send more data.
- **Flow Control Techniques: Stop-and-wait, and Sliding Window**

Error Control: Error control in the data link layer is based on ARQ (automatic repeat request), which is the retransmission of data.

- The term error control refers to methods of error detection and retransmission.
- Everytime an error is detected in an exchange, specified frames are retransmitted. This process is called ARQ.

To ensure reliable communication, there needs to exist flow control (managing the amount of data the sender sends), and error control (that data arrives at the destination error free).

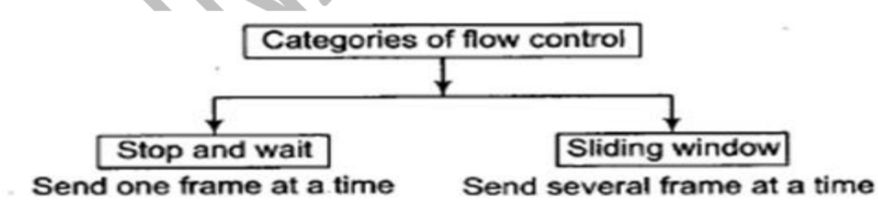
- Flow and error control needs to be done at several layers.
- For node-to-node links, flow and error control is carried out in the data-link layer.
- For end-point to end-point, flow and error control is carried out in the transport layer.

Flow & Error control:

- Error Detection and ARQ (error detection with retransmissions) must be combined with methods that intelligently limit the number of 'outstanding' (unACKed) frames.
- *Flow & Error control techniques: Stop-and-Wait ARQ, Go-Back-N ARQ, and Selective Repeat ARQ*

Flow Control Techniques:

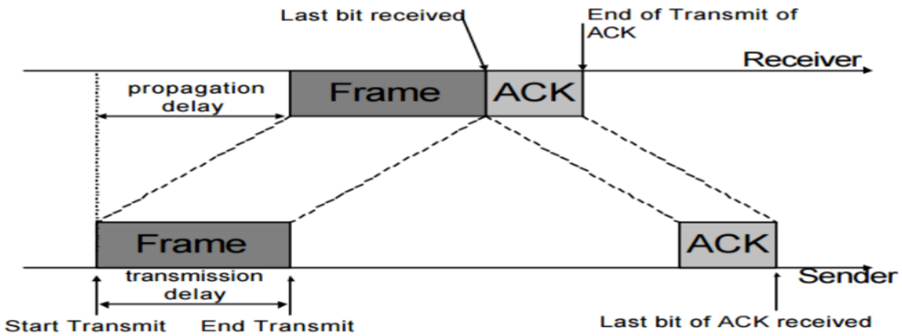
- One important aspect of the data link layer is flow control.
- Flow control refers to a set of procedures used to restrict the amount of data the sender can send before waiting for acknowledgement.



Stop and Wait Flow control:

- The sender has to wait for an acknowledgment of every frame that it sends.
- Only when a acknowledgment has been received is the next frame sent. This process continues until the sender transmits an End of Transmission (EOT) frame.

- In Stop-and-Wait flow control, the receiver indicates its readiness to receive data for each frame.



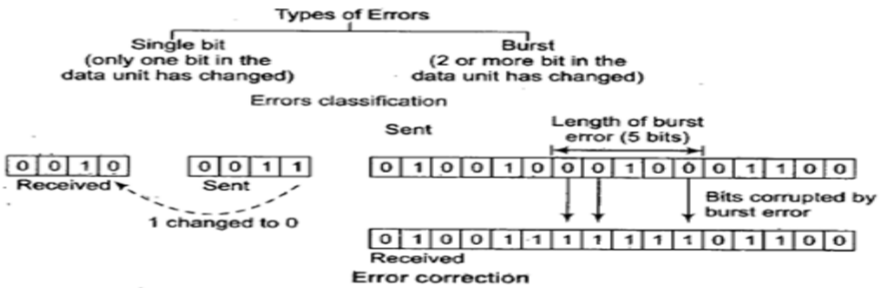
- For every frame that is sent, there needs to be an acknowledgment, which takes a similar amount of propagation time to get back to the sender.
- Only one frame can be in transmission at a time. This leads to inefficiency if propagation delay is much longer than the transmission delay
- **Advantages of Stop and Wait:**
    - It's simple and each frame is checked and acknowledged well.
- **Disadvantages of Stop and Wait:**
    - Only one frame can be in transmission at a time.
    - It is inefficient, if the distance between devices is long. Reason is propagation delay is much longer than the transmission delay.
    - The time spent for waiting acknowledgements between each frame can add significant amount to the total transmission time.

Sliding Window Flow Control:

- It works by having the sender and receiver have a “window” of frames.
- Each frame has to be numbered in relation to the sliding window. For a window of size n, frames get a number from 0 to n - 1. Subsequent frames get a number mod n.
- The sender can send as many frames as would fit into a window.
- The receiver, upon receiving enough frames, will respond with an acknowledgment of all frames up to a certain point in the window. It is called slide.
- This window can hold frames at either end and provides the upper limit on the number of frames that can be transmitted before requiring an acknowledgement.

Error Control Techniques:

- Many factors including line noise can alter or wipe out one or more bits of a given data unit.



- Reliable systems must have mechanism for detecting and correcting such errors.
- Error detection and correction are implemented either at the data link layer or the transport layer of the OSI model.

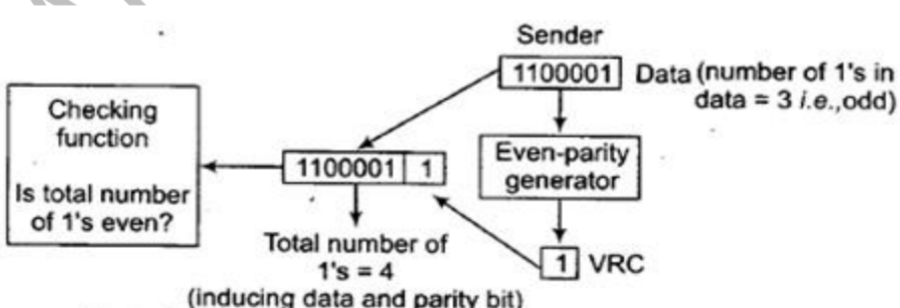
Error Detection

- Error detection uses the concept of redundancy, which means adding extra bits for detecting errors at the destination.

- Checking function performs the action that the received bit stream passes the checking criteria, the data portion of the data unit is accepted else rejected.

Vertical Redundancy Check (VRC)

- In this technique, a redundant bit, called parity bit, is appended to every data unit, so that the total number of 1's in the unit (including the parity bit) becomes even.
- If number of 1's are already even in data, then parity bit will be 0.



Detection of error using vertical redundancy check method

- Some systems may use odd parity checking, where the number of 1's should be odd. The principle is the same, the calculation is different.

Checksum

- There are two algorithms involved in this process, checksum generator at sender end and checksum checker at receiver end.
- The sender follows these steps
    - The data unit is divided into k sections each of n bits.
    - All sections are added together using 1's complement to get the sum.
    - The sum is complemented and becomes the checksum.
    - The checksum is sent with the data.
- The receiver follows these steps
    - The received unit is divided into k sections each of n bits.
    - All sections are added together using 1's complement to get the sum.
    - The sum is complemented.
    - If the result is zero, the data are accepted, otherwise they are

rejected. Limitation of checksum:

- It is not possible to detect the vertical error from the data which is received at receivers end.
- If noise modify the data in such a way that vertically placed bits can cancel the change made to them then calculated checksum will always be same as received checksum. Such errors cannot be detected and they are known as vertical errors.

Cyclic Redundancy Check (CRC):

- CRC is based on binary division.
- A sequence of redundant bits called CRC or the CRC remainder is appended to the end of a data unit, so that the resulting data unit becomes exactly divisible by a second, predetermined binary number.
- At its destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit is assumed to be intact and therefore is accepted.

Selection Criteria for CRC generator:

- Generator should be of more than 1 bit.
- when x is part of our generator, than it will detect all the errors. So for a generator to detect all type of errors, it should not contain x.
- if generator contains x+1, then all the odd bit errors are detected.

- A good generator always contain x other it will bw multiple of x.
- CRC 32 will always detect all type of errors in the network. it is considered as ideal network detector.

Error Correction:

- Error correction in data link layer is implemented simply anytime.
- An error is detected in an exchange, a negative acknowledgement NAK is returned and the specified frames are retransmitted. This process is called Automatic Repeat Request (ARQ).
- Retransmission of data happens in three Cases: Damaged frame, Lost frame and Lost acknowledgement.

Tcp Congestion Control

When large amount of data is fed to system which is not capable of handling it, congestion occurs. TCP controls congestion by means of Window mechanism. TCP sets a window size telling the other end how much data segment to send. TCP may use three algorithms for congestion control:

- Additive increase, Multiplicative Decrease
- Slow Start
- Timeout React

Timer Management

TCP uses different types of timer to control and management various tasks:

Keep-alive timer:

- This timer is used to check the integrity and validity of a connection.
- When keep-alive time expires, the host sends a probe to check if the connection still exists.

Retransmission timer:

- This timer maintains stateful session of data sent.
- If the acknowledgement of sent data does not receive within the Retransmission time, the data segment is sent again.

Persist timer:

- TCP session can be paused by either host by sending Window Size 0.
- To resume the session a host needs to send Window Size with some larger value.
- If this segment never reaches the other end, both ends may wait for each other for infinite time.
- When the Persist timer expires, the host re-sends its window size to let the other end know.
- Persist Timer helps avoid deadlocks in communication.

Timed-Wait:

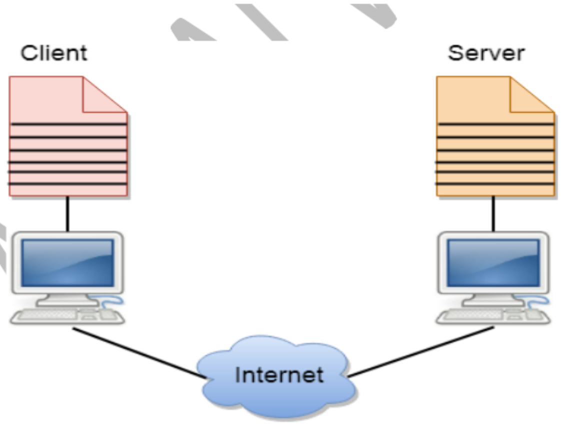
- After releasing a connection, either of the hosts waits for a Timed-Wait time to terminate the connection completely.
- This is in order to make sure that the other end has received the acknowledgement of its connection termination request.
- Timed-out can be a maximum of 240 seconds (4 minutes).

UNIT V

APPLICATION LAYER

Client and Server model

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



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

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

Client

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

Server

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

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

Advantages of Client-server networks:

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

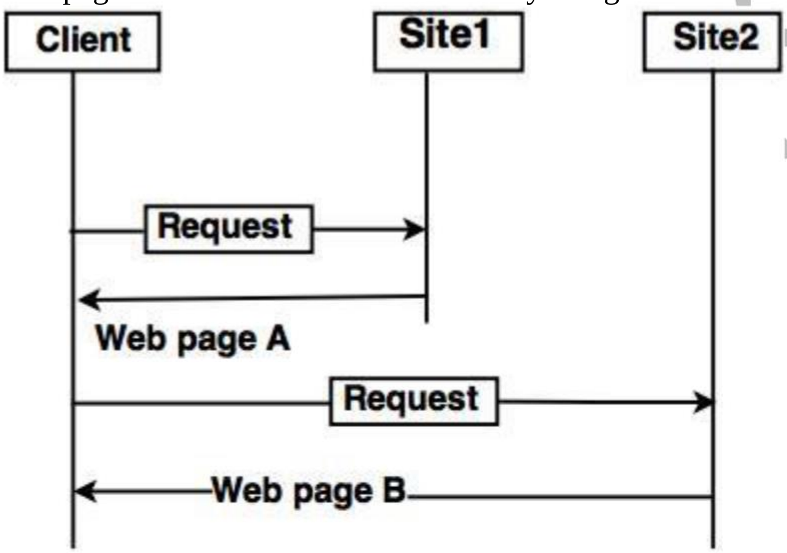
Disadvantages of Client-Server network:

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

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

World Wide Web

- The World Wide Web (WWW) is a collection of documents and other web resources which are identified by URLs, interlinked by hypertext links, and can be accessed and searched by browsers via the Internet.
- World Wide Web is also called the Web and it was invented by Tim Berners-Lee in 1989.
- Website is a collection of web pages belonging to a particular organization.
- The pages can be retrieved and viewed by using browser.



Architecture of WWW

- The client wants to see some information that belongs to site 1.
- It sends a request through its browser to the server at site 2.
- The server at site 1 finds the document and sends it to the client.

Client (Browser):

- Web browser is a program, which is used to communicate with web server on the Internet.

- Each browser consists of three parts: a controller, client protocol and interpreter.
- The controller receives input from input device and use the programs to access the documents.
- After accessing the document, the controller uses one of the interpreters to display the document on the screen.

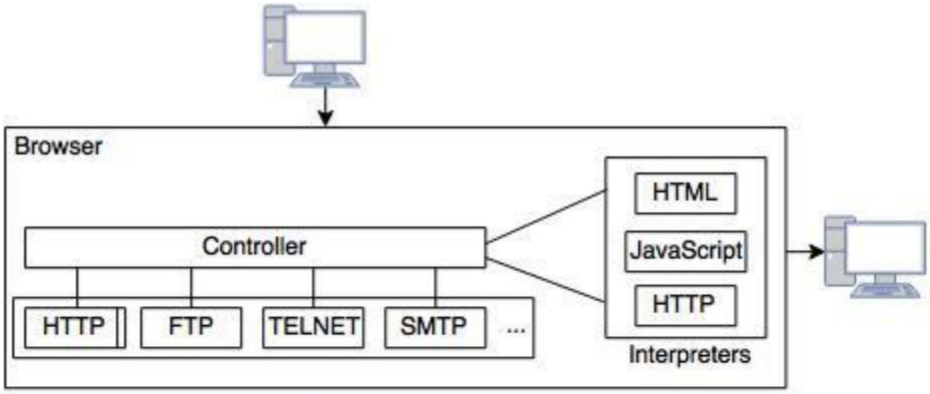


Fig: Cllient (Browser)

Server:

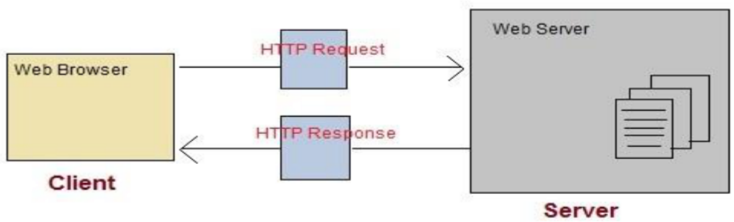
- A computer which is available for the network resources and provides service to the other computer on request is known as server.
- The web pages are stored at the server.
- Server accepts a TCP connection from a client browser.
- It gets the name of the file required.
- Server gets the stored file. Returns the file to the client and releases the top connection.

Uniform Resource Locater (URL)

- The URL is a standard for specifying any kind of information on the Internet.
- The URL consists of four parts: protocol, host computer, port and path.
- The protocol is the client or server program which is used to retrieve the document or file. The protocol can be ftp or http.
- The host is the name of computer on which the information is located.
- The URL can optionally contain the port number and it is separated from the host name by a colon.
- Path is the pathname of the file where the file is stored.

HTTP (Hypertext Transfer Protocol)

- HTTP is a protocol that clients and servers use on the web to communicate.
- It is similar to other internet protocols such as SMTP(Simple Mail Transfer Protocol) and FTP(File Transfer Protocol) but there is one fundamental difference.
- HTTP is a **stateless protocol** i.e HTTP supports only one request per connection. This means that with HTTP the clients connect to the server to send one request and then disconnects. This mechanism allows more users to connect to a given server over a period of time.
- The client sends an HTTP request and the server answers with an HTML page to the client, using HTTP.



HTTP METHODS

HTTP request can be made using a variety of methods, but the ones you will use most often are **Get** and **Post**. The method name tells the server the kind of request that is being made, and how the rest of the message will be formated.

HTTP Methods and Descriptions :

Method Name	**Description**
OPTIONS	Request for communication options that are available on the request/response chain.
GET	Request to retrieve information from server using a given URI.

Method Name	Description
HEAD	Identical to GET except that it does not return a message-body, only the headers and status line.
POST	Request for server to accept the entity enclosed in the body of HTTP method.
DELETE	Request for the Server to delete the resource.
CONNECT	Reserved for use with a proxy that can switch to being a tunnel.
PUT	This is same as POST, but POST is used to create, PUT can be used to create as well as update. It replaces all current representations of the target resource with the uploaded content.

DIFFERENCE BETWEEN GET AND POST REQUESTS

GET Request	POST Request
Data is sent in header to the server	Data is sent in the request body
Get request can send only limited amount of data	Large amount of data can be sent.
Get request is not secured because data is exposed in URL	Post request is secured because data is not exposed in URL.

GET Request	POST Request
Get request can be bookmarked and is more efficient.	Post request cannot be bookmarked.

FTP

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

Objectives of FTP

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

Why FTP?

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

Mechanism of FTP

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

There are two types of connections in FTP:

- **Control Connection:** The control connection uses very simple rules for communication. Through control connection, we can transfer a line of command or line of response at a time. The control connection is made between the control processes. The control connection remains connected during the entire interactive FTP session.

- **Data Connection:** The Data Connection uses very complex rules as data types may vary. The data connection is made between data transfer processes. The data connection opens when a command comes for transferring the files and closes when the file is transferred.

FTP Clients

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

Advantages of FTP:

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

Disadvantages of FTP:

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

ELECTRONIC MAIL

ARCHITECTURE AND SERVICES:

E-mail systems consist of two subsystems. They are:-

(1). User Agents, which allow people to read and send e-mail

(2). Message Transfer Agents, which move messages from source to destination

E-mail systems support 5 basic functions:-

a. Composition b. Transfer c. Reporting d. Displaying e. Disposition

(a). Composition: It refers to the process of creating messages and answers. Any text editor is used for body of the message. While the system itself can provide assistance with addressing and numerous header fields attached to each message.

(b). Reporting: It has to do with telling the originator what happened to the message that is, whether it was delivered, rejected (or) lost.

(c). Transfer: It refers to moving messages from originator to the recipient.

(d). Displaying: Incoming messages are to be displayed so that people can read their email.

(e). Disposition: It concerns what the recipient dose with the message after receiving it. Possibilities include throwing it away before reading (or) after reading, saving it and so on.

Most systems allow users to create mailboxes to store incoming e-mail. Commands are needed to create and destroy mailboxes, inspect the contents of mailboxes, insert and delete messages from mailboxes, and so on.

THE USER AGENT

A user agent is normally a program (sometimes called a mail reader) that accepts a variety of commands for composing, receiving, and replying to messages, as well as for manipulating mailboxes.

SENDING E-MAIL

To send an e-mail message, a user must provide the message, the destination address, and possibly some other parameters. The message can be produced with a free-standing text editor, a word processing program, or possibly with a specialized text editor built into the user agent. The destination address must be in a format that the user agent can deal with. Many user agents expect addresses of the form user@dns-address.

READING E-MAIL

When a user agent is started up, it looks at the user's mailbox for incoming e-mail before displaying anything on the screen. Then it may announce the number of messages in the mailbox or display a one-line summary of each one and wait for a command.

(2) MESSAGE
FORMATS RFC 822

Messages consist of a primitive envelope (described in RFC 821), some number of header fields, a blank line, and then the message body. Each header field (logically) consists of a single line of ASCII text containing the field name, a colon, and, for most fields, a value.

MIME — The Multipurpose Internet Mail Extensions

RFC 822 specified the headers but left the content entirely up to the users. Nowadays, on the worldwide Internet, this approach is no longer adequate. The problems include sending and receiving

1. Messages in languages with accents (e.g., French and German).
2. Messages in non-Latin alphabets (e.g., Hebrew and Russian).
3. Messages in languages without alphabets (e.g., Chinese and Japanese).
4. Messages not containing text at all (e.g., audio or images).

A solution was proposed in RFC 1341 called **MIME** (**Multipurpose Internet Mail Extensions**)

The basic idea of MIME is to continue to use the RFC 822 format, but to add structure to the message body and define encoding rules for non-ASCII messages. By not deviating from RFC 822, MIME messages can be sent using the existing mail programs and protocols. All that has to be changed are the sending and receiving programs, which users can do for themselves.

DNS

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

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

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

Generic Domains

- It defines the registered hosts according to their generic behavior.
- Each node in a tree defines the domain name, which is an index to the DNS database.

- It uses three-character labels, and these labels describe the organization type.

Label	Description
aero	Airlines and aerospace companies
biz	Businesses or firms
com	Commercial Organizations
coop	Cooperative business Organizations
edu	Educational institutions
gov	Government institutions
info	Information service providers
int	International Organizations
mil	Military groups
museum	Museum & other nonprofit organizations
name	Personal names
net	Network Support centers
org	Nonprofit Organizations
pro	Professional individual Organizations

Country Domain

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

Inverse Domain

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

Working of DNS

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