P.S.R. ENGINEERING COLLEGE, SIVAKASI – 626140
(An Autonomous Institution – Affiliated to Anna University)

DEPARTMENT OF COMPUTER SCIENCE & ENGINEERING
(ACADEMIC YEAR 2022-2023 ODD SEMESTER)

191CS54 COMPUTER NETWORKS COURSE MATERIAL

191CS54	COMPUTER NETWORKS			L	T	P	C
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Programme:	**B.E.** Computer Science and Engineering	**Sem:**	5	**Category:**			PC
Prerequisite:	NIL						
Aim:	To understand the concepts of computer networks, protocols and data communication.						

Course Outcomes: The Students will be able to

CO1:	Recall the fundamentals of a computer networks.
CO2:	Identify and understand various IEEE Standards for LAN.
CO3:	Illustrate multi-channel access Routing Protocols.
CO4:	Outline the elements and protocols of transport layer
CO5:	Describe congestion in network layer with routing algorithms.
CO6:	Discuss various Application Layer Protocols.

S.No	UNIT-I COMPUTER NETWORKS FUNDAMENTALS
1.	Building a network
2.	Wired and Wireless Networks- Requirements
3.	Layering and protocols
4.	Internet Architecture
5.	Network software Performance
6.	Link Layer Services – Framing
7.	Error Detection – Flow control.

Computer network

A **computer network** is a system in which multiple computers are connected to each other to share information and resources.

Characteristics of a Computer Network

- Share resources from one computer to another.
- Create files and store them in one computer, access those files from the other computer(s) connected over the network.
- Connect a printer, scanner, or a fax machine to one computer within the network and let other computers of the network use the machines available over the network.

Following is the list of hardware's required to set up a computer network.

- Network Cables
- Distributors
- Routers
- Internal Network Cards
- External Network Cards

Network Cables

Network cables are used to connect computers. The most commonly used cable is Category 5 cable RJ-45.

Distributors

A computer can be connected to another one via a serial port but if we need to connect many computers to produce a network, this serial connection will not work.

The solution is to use a central body to which other computers, printers, scanners, etc. can be connected and then this body will manage or distribute network traffic.

Router

A router is a type of device which acts as the central point among computers and other devices that are a part of the network. It is equipped with holes called ports. Computers and other devices are connected to a router using network cables. Now-a-days router comes in wireless modes using which computers can be connected without any physical cable.

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Network Card

Network card is a necessary component of a computer without which a computer cannot be connected over a network. It is also known as the network adapter or Network Interface Card (NIC). Most branded computers have network card pre-installed. Network cards are of two types: Internal and External Network Cards.

Internal Network Cards

Motherboard has a slot for internal network card where it is to be inserted. Internal network cards are of two types in which the first type uses Peripheral Component Interconnect (PCI) connection, while the second type uses Industry Standard Architecture (ISA). Network cables are required to provide network access.

External Network Cards

External network cards are of two types: Wireless and USB based. Wireless network card needs to be inserted into the motherboard, however no network cable is required to connect to the network.

Universal Serial Bus (USB)

USB card is easy to use and connects via USB port. Computers automatically detect USB card and can install the drivers required to support the USB network card automatically.

Types of Network Topology

The arrangement of a network that comprises nodes and connecting lines via sender and receiver is referred to as network topology. The various network topologies are:

Mesh Topology:

In a mesh topology, every device is connected to another device via a particular channel. In Mesh Topology, the protocols used are AHCP (Ad Hoc Configuration Protocols), DHCP (Dynamic Host Configuration Protocol), etc.

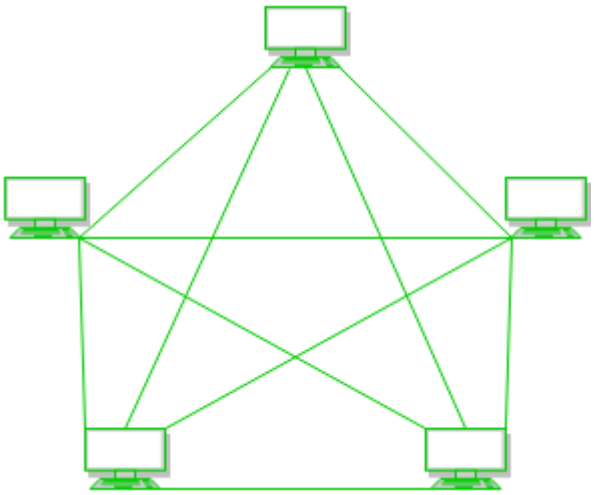


Figure 1: Every device is connected to another via dedicated channels. These channels are known as links.

- Suppose, the N number of devices are connected with each other in a mesh topology, the total number of ports that are required by each device is N-1. In Figure 1, there are 5 devices connected to each other, hence the total number of ports required by each device is 4. The total number of ports required=N*(N-1).
- Suppose, N number of devices are connected with each other in a mesh topology, then the total number of dedicated links required to connect them is ${}^{N}C_2$ i.e. N(N-1)/2. In Figure 1, there are 5 devices connected to each other, hence the total number of links required is 5*4/2 = 10.

Advantages of this topology:

- Communication is very fast between the nodes.
- It is robust.
- The fault is diagnosed easily. Data is reliable because data is transferred among the devices through dedicated channels or links.
- Provides security and privacy.

Problems with this topology:

- Installation and configuration are difficult.
- The cost of cables is high as bulk wiring is required, hence suitable for less number of devices.
- The cost of maintenance is high.

Star Topology:

In star topology, all the devices are connected to a single hub through a cable. This hub is the central node and all other nodes are connected to the central node. The hub can be passive in nature i.e., not an intelligent hub such as broadcasting devices, at the same time the hub can be intelligent known as an active hub. Active hubs have repeaters in them. Coaxial cables or RJ-45 cables are used to connect the computers. In Star Topology, many popular Ethernet LAN protocols are used as CD(Collision Detection), CSMA (Carrier Sense Multiple Access), etc.

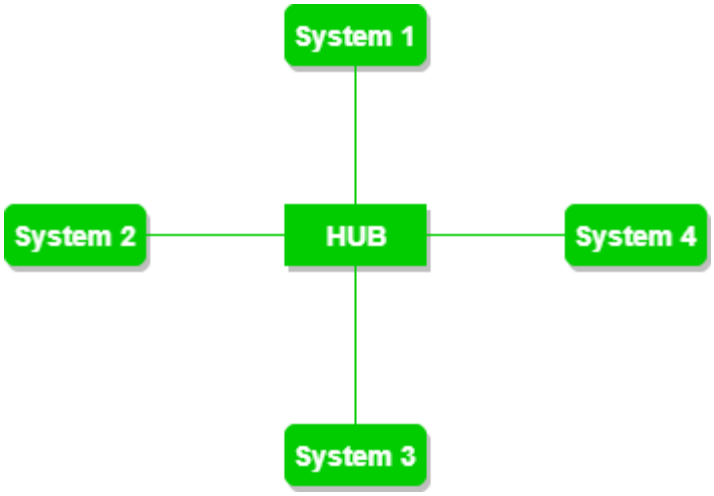


Figure 2: A star topology having four systems connected to a single point of connection i.e. hub.

Advantages of this topology:

- If N devices are connected to each other in a star topology, then the number of cables required to connect them is N. So, it is easy to set up.
- Each device requires only 1 port i.e. to connect to the hub, therefore the total number of ports required is N.
- It is Robust. If one link fails only that link will affect and not other than that.
- Easy to fault identification and fault isolation.
- Star topology is cost-effective as it uses inexpensive coaxial cable.

Problems with this topology:

- If the concentrator (hub) on which the whole topology relies fails, the whole system will crash down.
- The cost of installation is high.
- Performance is based on the single concentrator i.e. hub.

Bus Topology:

Bus topology is a network type in which every computer and network device is connected to a single cable. It is bi-directional. It is a multi-point connection and a non-robust topology because if the backbone fails the topology crashes. In Bus Topology, various MAC (Media Access Control) protocols are followed by LAN ethernet connections like TDMA, Pure Aloha, CDMA, Slotted Aloha, etc.

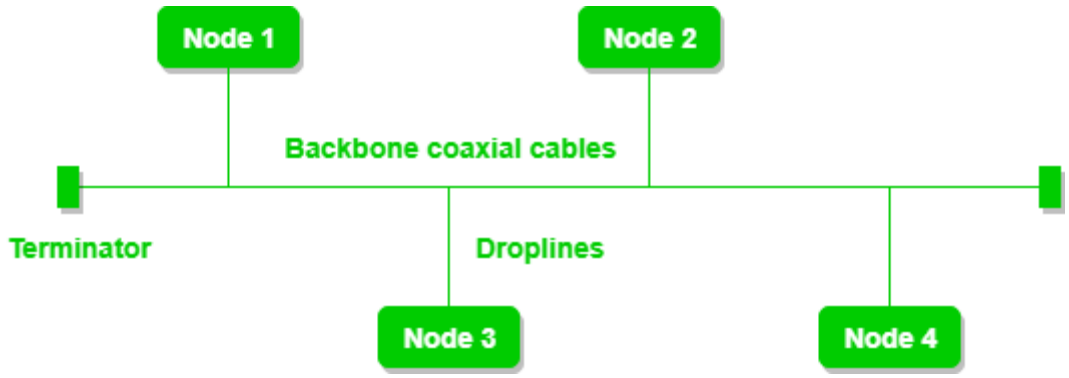


Figure 3: A bus topology with shared backbone cable. The nodes are connected to the channel via drop lines.

Advantages of this topology:

- If N devices are connected to each other in a bus topology, then the number of cables required to connect them is 1, known as backbone cable, and N drop lines are required.
- Coaxial or twisted pair cables are mainly used in bus-based networks that support up to 10 Mbps.
- The cost of the cable is less compared to other topologies, but it is used to build small networks.
- Bus topology is familiar technology as installation and troubleshooting techniques are well known.

Problems with this topology:

- A bus topology is quite simpler, but still, it requires a lot of cabling.
- If the common cable fails, then the whole system will crash down.
- If the network traffic is heavy, it increases collisions in the network. To avoid this, various protocols are used in the MAC layer known as Pure Aloha, Slotted Aloha, CSMA/CD, etc.
- Adding new devices to the network would slow down networks.
- Security is very low.

Ring Topology:

In this topology, it forms a ring connecting devices with exactly two neighboring devices.

A number of repeaters are used for Ring topology with a large number of nodes, because if someone wants to send some data to the last node in the ring topology with 100 nodes, then the data will have to pass through 99 nodes to reach the 100th node. Hence to prevent data loss repeaters are used in the network.

The data flows in one direction, i.e.., it is unidirectional, but it can be made bidirectional by having 2 connections between each Network Node, it is called Dual Ring Topology. In-Ring Topology, the Token Ring Passing protocol is used by the workstations to transmit the data.

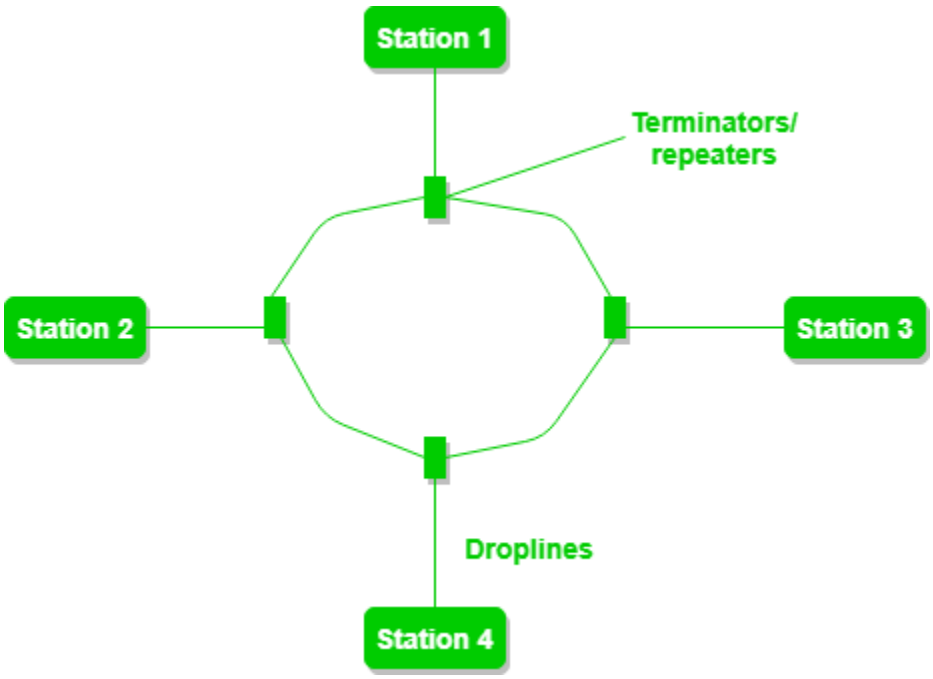


Figure 4: A ring topology comprises 4 stations connected with each forming a ring.

The most common access method of ring topology is token passing.

- **Token passing:** It is a network access method in which a token is passed from one node to another node.

- **Token:** It is a frame that circulates around the network.

The following operations take place in ring topology are :

1. One station is known as a **monitor** station which takes all the responsibility for performing the operations.
2. To transmit the data, the station has to hold the token. After the transmission is done, the token is to be released for other stations to use.
3. When no station is transmitting the data, then the token will circulate in the ring.
4. There are two types of token release techniques: **Early token release** releases the token just after transmitting the data and **Delayed token release** releases the token after the acknowledgment is received from the receiver.

Advantages of this topology:

- The data transmission is high-speed.
- The possibility of collision is minimum in this type of topology.
- Cheap to install and expand.
- It is less costly than a star topology.

Problems with this topology:

- The failure of a single node in the network can cause the entire network to fail.
- Troubleshooting is difficult in this topology.
- The addition of stations in between or the removal of stations can disturb the whole topology.
- Less secure.

Tree Topology :

This topology is the variation of the Star topology. This topology has a hierarchical flow of data. In Tree Topology, protocols like DHCP and SAC (Standard Automatic Configuration) are used.

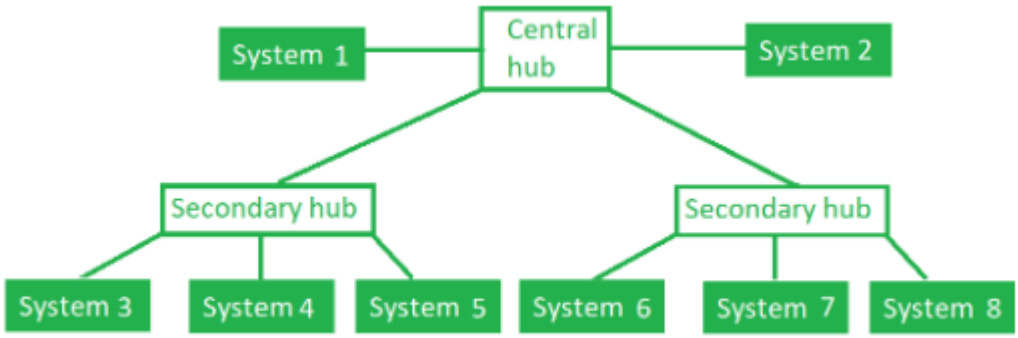


Figure 5: In this, the various secondary hubs are connected to the central hub which contains the repeater. This data flow from top to bottom i.e. from the central hub to the secondary and then to the devices or from bottom to top i.e. devices to the secondary hub and then to the central hub. It is a multi-point connection and a non-robust topology because if the backbone fails the topology crashes.

Advantages of this topology :

- It allows more devices to be attached to a single central hub thus it decreases the distance that is traveled by the signal to come to the devices.
- It allows the network to get isolated and also prioritize from different computers.
- We can add **new devices to the existing network.**
- **Error detection** and **error correction** are very easy in a tree topology.

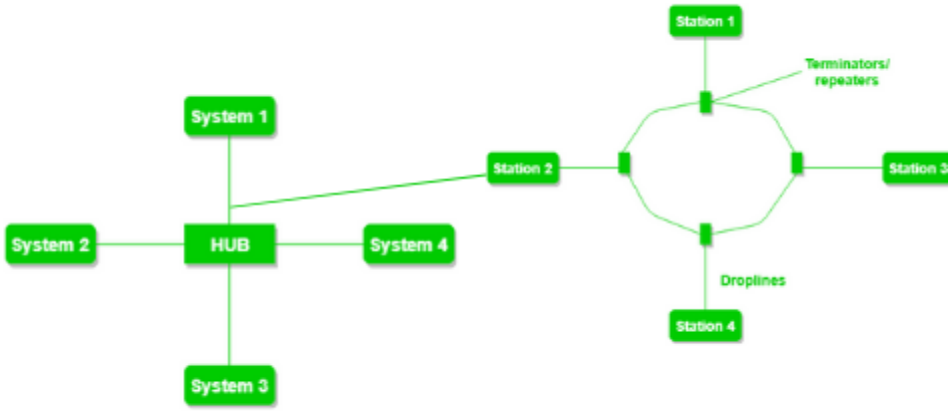
Problems with this topology:

- If the central hub gets fails the entire system fails.

- The cost is high because of the cabling.
- If new devices are added, it becomes difficult to reconfigure.

Hybrid Topology :

This topological technology is the combination of all the various types of topologies we have studied above. It is used when the nodes are free to take any form. It means these can be individuals such as Ring or Star topology or can be a combination of various types of topologies seen above. Each individual topology uses the protocol that has been discussed earlier.



Hybrid Topology

Figure 6: The above figure shows the structure of the Hybrid topology. As seen it contains a combination of all different types of networks.

Advantages of this topology :

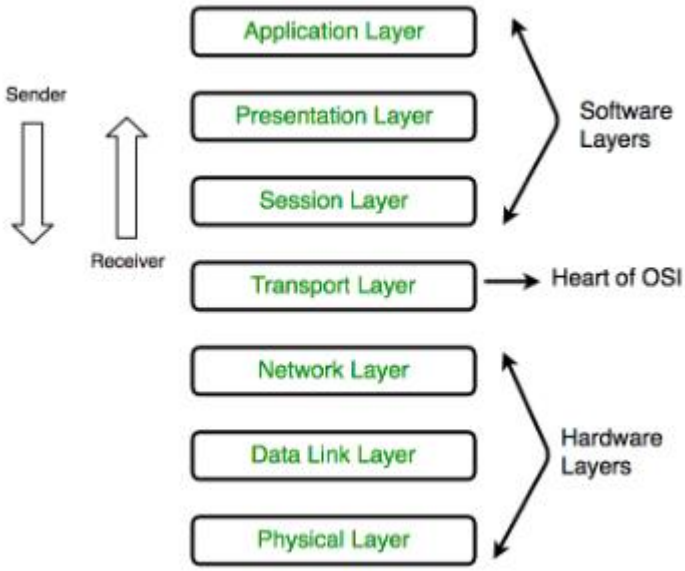
- This topology is **very flexible**.
- The size of the network can be easily expanded by **adding new devices.**

Problems with this topology :

- It is challenging **to design the architecture** of the Hybrid Network.
- **Hubs** used in this topology are **very expensive.**
- The infrastructure cost is very high as a hybrid network **requires a lot of cabling and network devices**.

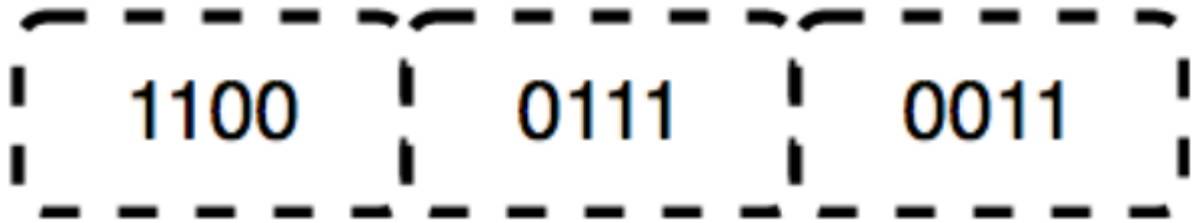
Layers of OSI Model

OSI stands for **Open Systems Interconnection**. It has been developed by ISO – '**International Organization for Standardization**', in the year 1984. It is a 7 layer architecture with each layer having specific functionality to perform. All these 7 layers work collaboratively to transmit the data from one person to another across the globe.



1. Physical Layer (Layer 1) :

The lowest layer of the OSI reference model is the physical layer. It is responsible for the actual physical connection between the devices. The physical layer contains information in the form of **bits.** It is responsible for transmitting individual bits from one node to the next. When receiving data, this layer will get the signal received and convert it into 0s and 1s and send them to the Data Link layer, which will put the frame back together.



The functions of the physical layer are as follows:

1. **Bit synchronization:** The physical layer provides the synchronization of the bits by providing a clock. This clock controls both sender and receiver thus providing synchronization at bit level.
2. **Bit rate control:** The Physical layer also defines the transmission rate i.e. the number of bits sent per second.
3. **Physical topologies:** Physical layer specifies the way in which the different, devices/nodes are arranged in a network i.e. bus, star, or mesh topology.
4. **Transmission mode:** Physical layer also defines the way in which the data flows between the two connected devices. The various transmission modes possible are Simplex, half-duplex and full-duplex.

* Hub, Repeater, Modem, Cables are Physical Layer devices.

** Network Layer, Data Link Layer, and Physical Layer are also known as **Lower Layers** or **Hardware Layers**.

2. Data Link Layer (DLL) (Layer 2) :

The data link layer is responsible for the node-to-node delivery of the message. The main function of this layer is to make sure data transfer is error-free from one node to another,

over the physical layer. When a packet arrives in a network, it is the responsibility of DLL to transmit it to the Host using its MAC address.
Data Link Layer is divided into two sublayers:

1. Logical Link Control (LLC)
2. Media Access Control (MAC)

The packet received from the Network layer is further divided into frames depending on the frame size of NIC(Network Interface Card). DLL also encapsulates Sender and Receiver's MAC address in the header.

The Receiver's MAC address is obtained by placing an ARP(Address Resolution Protocol) request onto the wire asking "Who has that IP address?" and the destination host will reply with its MAC address.

The functions of the Data Link layer are :

1. **Framing:** Framing is a function of the data link layer. It provides a way for a sender to transmit a set of bits that are meaningful to the receiver. This can be accomplished by attaching special bit patterns to the beginning and end of the frame.
2. **Physical addressing:** After creating frames, the Data link layer adds physical addresses (MAC address) of the sender and/or receiver in the header of each frame.
3. **Error control:** Data link layer provides the mechanism of error control in which it detects and retransmits damaged or lost frames.
4. **Flow Control:** The data rate must be constant on both sides else the data may get corrupted thus, flow control coordinates the amount of data that can be sent before receiving acknowledgement.
5. **Access control:** When a single communication channel is shared by multiple devices, the MAC sub-layer of the data link layer helps to determine which device has control over the channel at a given time.

** Packet in Data Link layer is referred to as **Frame.***

** Data Link layer is handled by the NIC (Network Interface Card) and device drivers of host machines.

*** Switch & Bridge are Data Link Layer devices.

3. Network Layer (Layer 3) :

The network layer works for the transmission of data from one host to the other located in different networks. It also takes care of packet routing i.e. selection of the shortest path to transmit the packet, from the number of routes available. The sender & receiver's IP addresses are placed in the header by the network layer.

The functions of the Network layer are :

1. **Routing:** The network layer protocols determine which route is suitable from source to destination. This function of the network layer is known as routing.
2. **Logical Addressing:** In order to identify each device on internetwork uniquely, the network layer defines an addressing scheme. The sender & receiver's IP addresses are placed in the header by the network layer. Such an address distinguishes each device uniquely and universally.

* *Segment* in Network layer is referred to as **Packet**.

** Network layer is implemented by networking devices such as routers.

4. Transport Layer (Layer 4) :

The transport layer provides services to the application layer and takes services from the network layer. The data in the transport layer is referred to as *Segments*. It is responsible for the End to End Delivery of the complete message. The transport layer also provides the acknowledgement of the successful data transmission and re-transmits the data if an error is found.

At sender's side: Transport layer receives the formatted data from the upper layers, performs **Segmentation**, and also implements **Flow & Error control** to ensure proper data transmission. It also adds Source and Destination port numbers in its header and forwards the segmented data to the Network Layer.

Note: *The sender needs to know the port number associated with the receiver's application.*

Generally, this destination port number is configured, either by default or manually. For example, when a web application makes a request to a web server, it typically uses port number 80, because this is the default port assigned to web applications. Many applications have default ports assigned.

At receiver's side: Transport Layer reads the port number from its header and forwards the Data which it has received to the respective application. It also performs sequencing and reassembling of the segmented data.

The functions of the transport layer are as follows:

1. **Segmentation and Reassembly:** This layer accepts the message from the (session) layer, and breaks the message into smaller units. Each of the segments produced has a header associated with it. The transport layer at the destination station reassembles the message.
2. **Service Point Addressing:** In order to deliver the message to the correct process, the transport layer header includes a type of address called service point address or port address. Thus by specifying this address, the transport layer makes sure that the message is delivered to the correct process.

The services provided by the transport layer :

A. Connection-Oriented Service: It is a three-phase process that includes
–ConnectionEstablishment
–DataTransfer
– Termination / disconnection

In this type of transmission, the receiving device sends an acknowledgement, back to the source after a packet or group of packets is received. This type of transmission is reliable and secure.

B. Connectionless service: It is a one-phase process and includes Data Transfer. In this type of transmission, the receiver does not acknowledge receipt of a packet. This approach

allows for much faster communication between devices. Connection-oriented service is more reliable than connectionless Service.
* Data in the Transport Layer is called as **Segments**.
** Transport layer is operated by the Operating System. It is a part of the OS and communicates with the Application Layer by making system calls.
*Transport Layer is called as **Heart of OSI** model.*

5. Session Layer (Layer 5) :

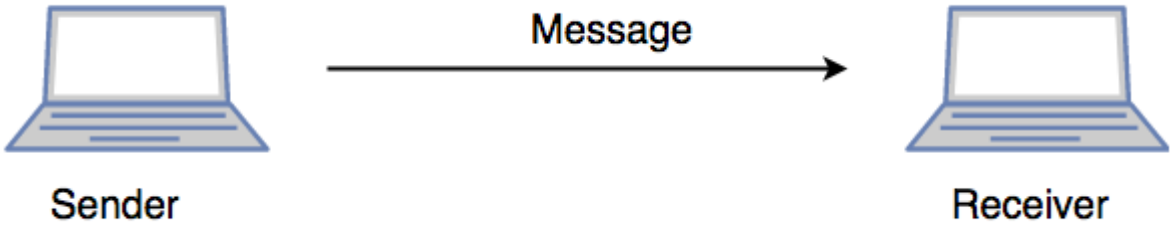
This layer is responsible for the establishment of connection, maintenance of sessions, authentication, and also ensures security.
The functions of the session layer are :

1. **Session establishment, maintenance, and termination:** The layer allows the two processes to establish, use and terminate a connection.
2. **Synchronization:** This layer allows a process to add checkpoints which are considered synchronization points into the data. These synchronization points help to identify the error so that the data is re-synchronized properly, and ends of the messages are not cut prematurely and data loss is avoided.
3. **Dialog Controller:** The session layer allows two systems to start communication with each other in half-duplex or full-duplex.

**All the below 3 layers(including Session Layer) are integrated as a single layer in the TCP/IP model as "Application Layer".
Implementation of these 3 layers is done by the network application itself. These are also known as **Upper Layers or **Software Layers**.

Scenario:

Let us consider a scenario where a user wants to send a message through some Messenger application running in his browser. The "Messenger" here acts as the application layer which provides the user with an interface to create the data. This message or so-called Data is compressed, encrypted (if any secure data), and converted into bits (0's and 1's) so that it can be transmitted.



6. Presentation Layer (Layer 6):

The presentation layer is also called the **Translation layer**. The data from the application layer is extracted here and manipulated as per the required format to transmit over the network.
The functions of the presentation layer are :

- **Translation:** For example, ASCII to EBCDIC.
- **Encryption/ Decryption:** Data encryption translates the data into another form or code. The encrypted data is known as the ciphertext and the decrypted data is known as plain text. A key value is used for encrypting as well as decrypting data.

- **Compression:** Reduces the number of bits that need to be transmitted on the network.

7. Application Layer (Layer 7) :

At the very top of the OSI Reference Model stack of layers, we find the Application layer which is implemented by the network applications. These applications produce the data, which has to be transferred over the network. This layer also serves as a window for the application services to access the network and for displaying the received information to the user.

Example: Application – Browsers, Skype Messenger, etc.

***Application Layer is also called Desktop Layer.*

Internet Architecture

The Internet architecture evolved out of experiences with an earlier packet-switched network called the ARPANET. Both the Internet and the ARPANET were funded by the Advanced Research Projects Agency (ARPA), one of the research and development funding agencies of the U.S. Department of Defense. The Internet and ARPANET were around before the OSI architecture, and the experience gained from building them was a major influence on the OSI reference model.

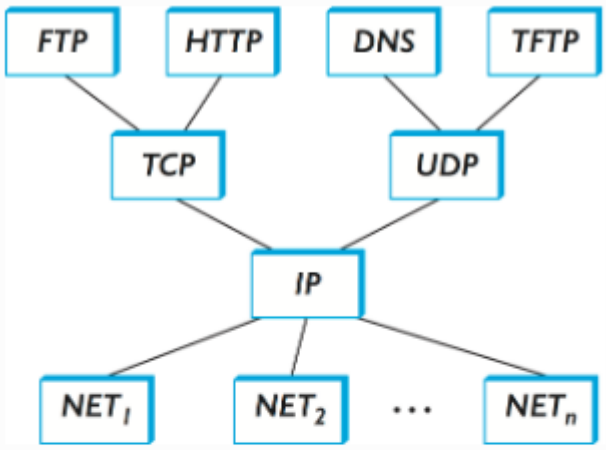


Figure 14. *Internet protocol graph.*

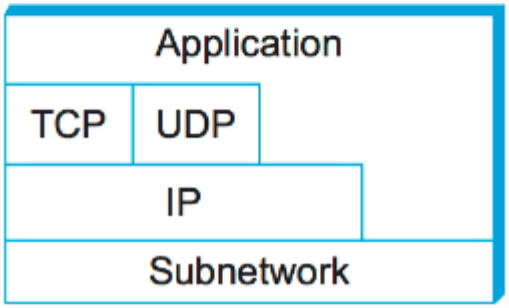


Figure: Internet *protocol graph.*ℑ

While the 7-layer OSI model can, with some imagination, be applied to the Internet, a simpler stack is often used instead. At the lowest level is a wide variety of network protocols, denoted NET_1, NET_2, and so on. In practice, these protocols are implemented by a combination of hardware (e.g., a network adaptor) and software (e.g., a network device driver). For example, you might find Ethernet or wireless protocols (such as the 802.11 Wi-Fi standards) at this

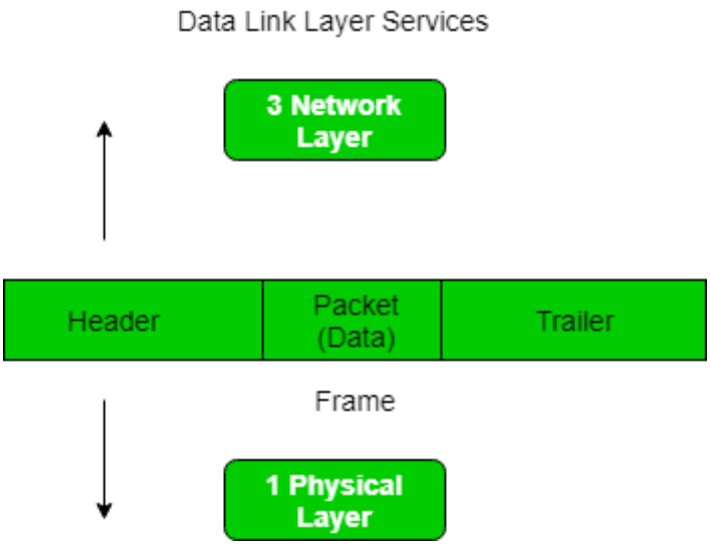
layer. (These protocols in turn may actually involve several sublayers, but the Internet architecture does not presume anything about them.) The next layer consists of a single protocol—the *Internet Protocol* (IP). This is the protocol that supports the interconnection of multiple networking technologies into a single, logical internetwork. The layer on top of IP contains two main protocols—the *Transmission Control Protocol* (TCP) and the *User Datagram Protocol* (UDP). TCP and UDP provide alternative logical channels to application programs: TCP provides a reliable byte-stream channel, and UDP provides an unreliable datagram delivery channel (*datagram* may be thought of as a synonym for message). In the language of the Internet, TCP and UDP are sometimes called *end-to-end* protocols, although it is equally correct to refer to them as *transport* protocols.

Running above the transport layer is a range of application protocols, such as HTTP, FTP, Telnet (remote login), and the Simple Mail Transfer Protocol (SMTP), that enable the interoperation of popular applications. To understand the difference between an application layer protocol and an application, think of all the different World Wide Web browsers that are or have been available (e.g., Firefox, Chrome, Safari, Netscape, Mosaic, Internet Explorer). There is a similarly large number of different implementations of web servers. The reason that you can use any one of these application programs to access a particular site on the Web is that they all conform to the same application layer protocol: HTTP. Confusingly, the same term sometimes applies to both an application and the application layer protocol that it uses (e.g., FTP is often used as the name of an application that implements the FTP protocol).

Framing in Data Link Layer

Frames are the units of digital transmission, particularly in computer networks and telecommunications. Frames are comparable to the packets of energy called photons in the case of light energy. Frame is continuously used in Time Division Multiplexing process.

Framing is a point-to-point connection between two computers or devices consists of a wire in which data is transmitted as a stream of bits. However, these bits must be framed into discernible blocks of information. Framing is a function of the data link layer. It provides a way for a sender to transmit a set of bits that are meaningful to the receiver. Ethernet, token ring, frame relay, and other data link layer technologies have their own frame structures. Frames have headers that contain information such as error-checking codes.



At the data link layer, it extracts the message from the sender and provides it to the receiver by providing the sender's and receiver's addresses. The advantage of using frames is that data is broken up into recoverable chunks that can easily be checked for corruption.

The process of dividing the data into frames and reassembling it is transparent to the user and is handled by the data link layer.

Framing is an important aspect of data link layer protocol design because it allows the transmission of data to be organized and controlled, ensuring that the data is delivered accurately and efficiently.

Problems in Framing –

- **Detecting start of the frame:** When a frame is transmitted, every station must be able to detect it. Station detects frames by looking out for a special sequence of bits that marks the beginning of the frame i.e. SFD (Starting Frame Delimiter).
- **How does the station detect a frame:** Every station listens to link for SFD pattern through a sequential circuit. If SFD is detected, sequential circuit alerts station. Station checks destination address to accept or reject frame.
- **Detecting end of frame:** When to stop reading the frame.

Types of framing – There are two types of framing:

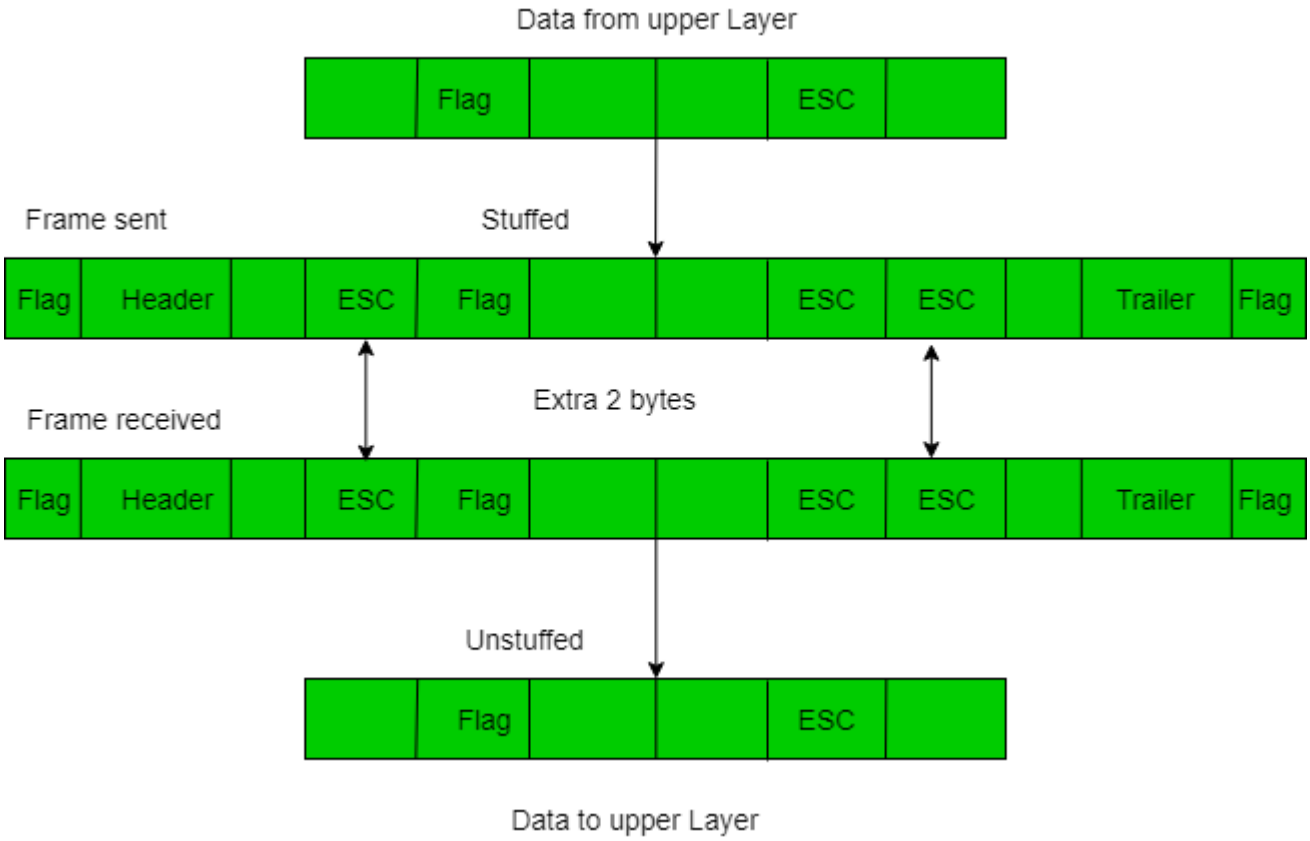
1. Fixed size – The frame is of fixed size and there is no need to provide boundaries to the frame, the length of the frame itself acts as a delimiter.

- **Drawback:** It suffers from internal fragmentation if the data size is less than the frame size
- **Solution:** Padding

2. Variable size – In this, there is a need to define the end of the frame as well as the beginning of the next frame to distinguish. This can be done in two ways:

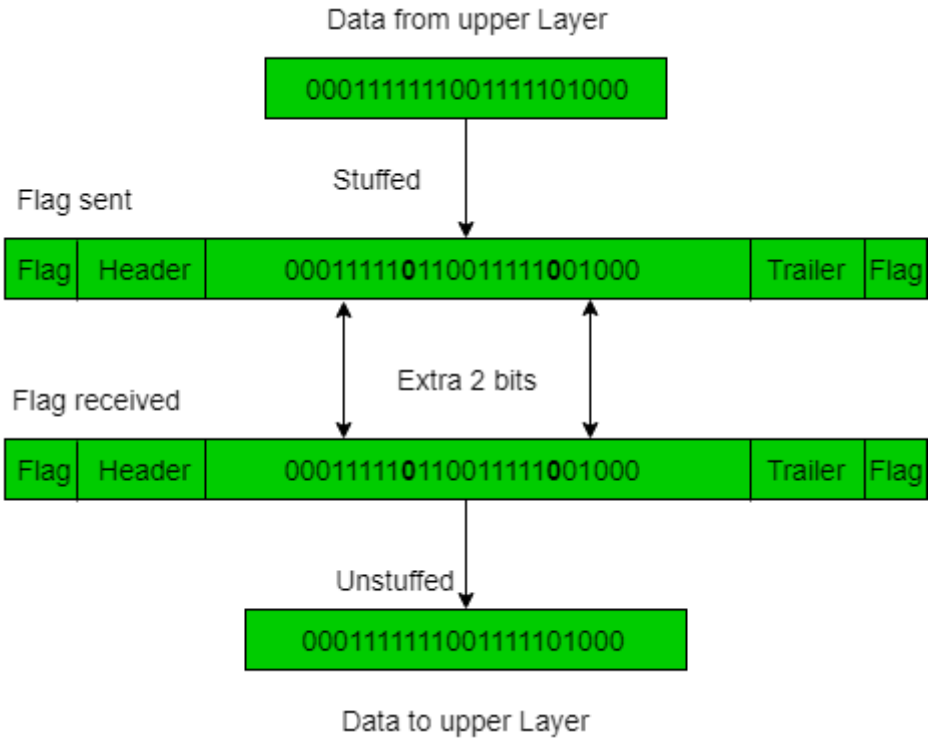
1. **Length field –** We can introduce a length field in the frame to indicate the length of the frame. Used in **Ethernet(802.3)**. The problem with this is that sometimes the length field might get corrupted.
2. **End Delimiter (ED) –** We can introduce an ED(pattern) to indicate the end of the frame. Used in **Token Ring**. The problem with this is that ED can occur in the data. This can be solved by:
   1. Character/Byte Stuffing: Used when frames consist of characters. If data contains ED then, a byte is stuffed into data to differentiate it from ED.

Let ED = "$" –> if data contains '$' anywhere, it can be escaped using '\O' character.
–> if data contains '\O$' then, use '\O\O\O$'($ is escaped using \O and \O is escaped using \O).



Disadvantage – It is very costly and obsolete method.

2. Bit Stuffing: Let ED = 01111 and if data = 01111
–> Sender stuffs a bit to break the pattern i.e. here appends a 0 in data = 0111**0**1.
–> Receiver receives the frame.
–> If data contains 01110**1**, receiver removes the 0 and reads the data.



Examples –

- If Data –> 011100011110 and ED –> 0111 then, find data after bit stuffing?
  –>011**0**10001**1**0**1**1**0**0

- If Data –> 110001001 and ED –> 1000 then, find data after bit stuffing?
  –>1100**1**0100**1**1

Flow and Error Control

Flow control and Error control are the two main responsibilities of the Data link layer. Let us understand what these two terms specify. For the node-to-node delivery of the data, the flow and error control are done at the data link layer.

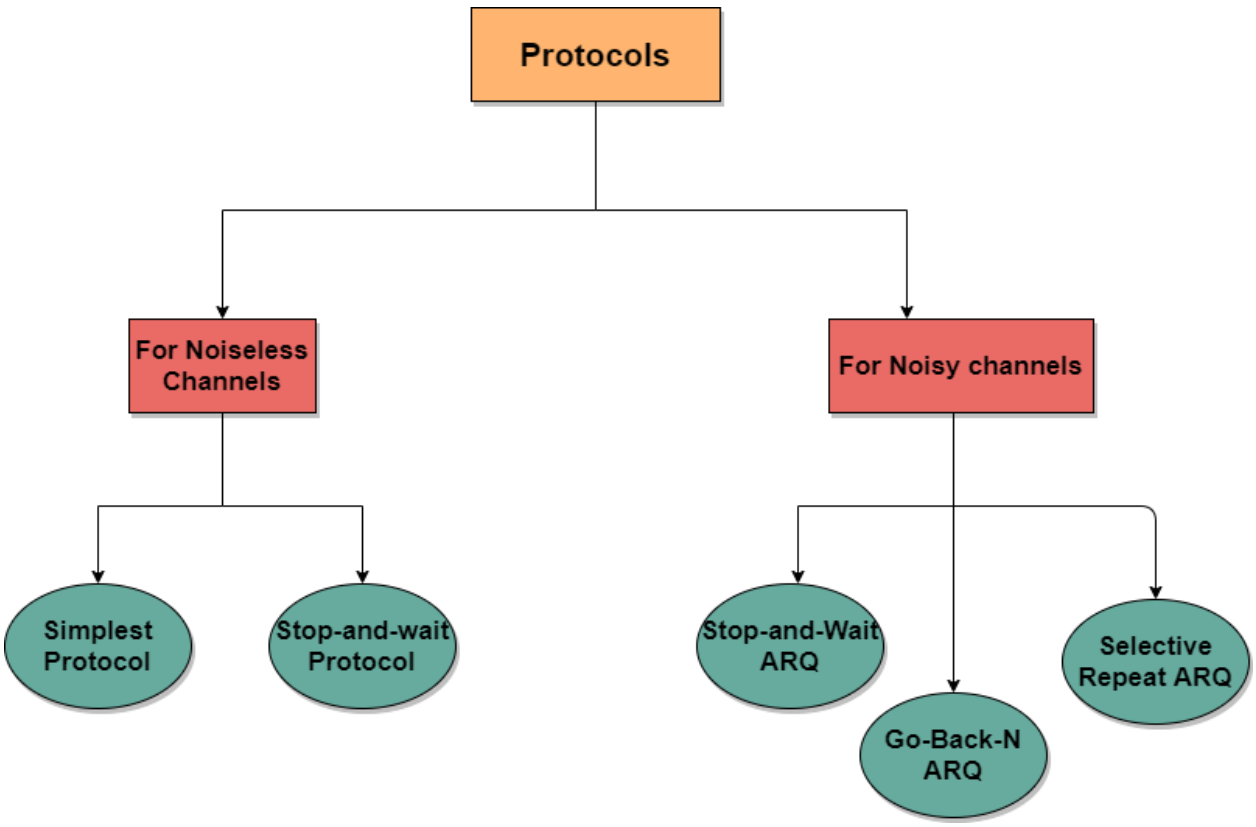
Flow Control mainly coordinates with the amount of data that can be sent before receiving an acknowledgment from the receiver and it is one of the major duties of the data link layer.

- For most of the protocols, **flow control** is a set of procedures that mainly tells the sender how much data the sender can send before it must **wait for an acknowledgment** from **the receiver**.
- The data flow must not be allowed to overwhelm the receiver; because any receiving device has a very limited speed at which the device can process the incoming data and the limited amount of memory to store the incoming data.
  - The processing rate is slower than the transmission rate; due to this reason each receiving device has a block of memory that is commonly known as **buffer,** that is used to store the incoming data until this data will be processed. In case the buffer begins to fillup then the receiver must be able to tell the sender to halt the transmission until once again the receiver become able to receive.
- Thus the flow control makes the sender; wait for the acknowledgment from the receiver before the continuation to send more data to the receiver.

- Some of the common flow control techniques are: Stop-and-Wait and sliding window technique.
- **Error Control** contains both error detection and error correction. It mainly allows the receiver to inform the sender about any damaged or lost frames during the transmission and then it coordinates with the retransmission of those frames by the sender.
- The term Error control in the data link layer mainly refers to the methods of error detection and retransmission. Error control is mainly implemented in a simple way and that is whenever there is an error detected during the exchange, then specified frames are retransmitted and this process is also referred to as **Automatic Repeat request(ARQ).**

Protocols

- The implementation of protocols is mainly implemented in the software by using one of the common programming languages. The classification of the protocols can be mainly done on the basis of where they are being used.
- Protocols can be used for **noiseless channels**(that is **error-free**) and also used for noisy channels(that is **error-creating**). The protocols used for noiseless channels mainly cannot be used in real-life and are mainly used to serve as the basis for the protocols used for **noisy channels.**



- All the above-given protocols are unidirectional in the sense that the data frames travel from one node i.e Sender to the other node i.e receiver.

- The special frames called acknowledgment (ACK) and negative acknowledgment (NAK) both can flow in opposite direction for flow and error control purposes and the data can flow in only one direction.
- But in the real-life network, the protocols of the data link layer are implemented as bidirectional which means the flow of the data is in both directions. And in these protocols, the flow control and error control information such as ACKs and NAKs are included in the data frames in a technique that is commonly known as **piggybacking.**
- Also, bidirectional protocols are more complex than the unidirectional protocol.

Simplest Protocol

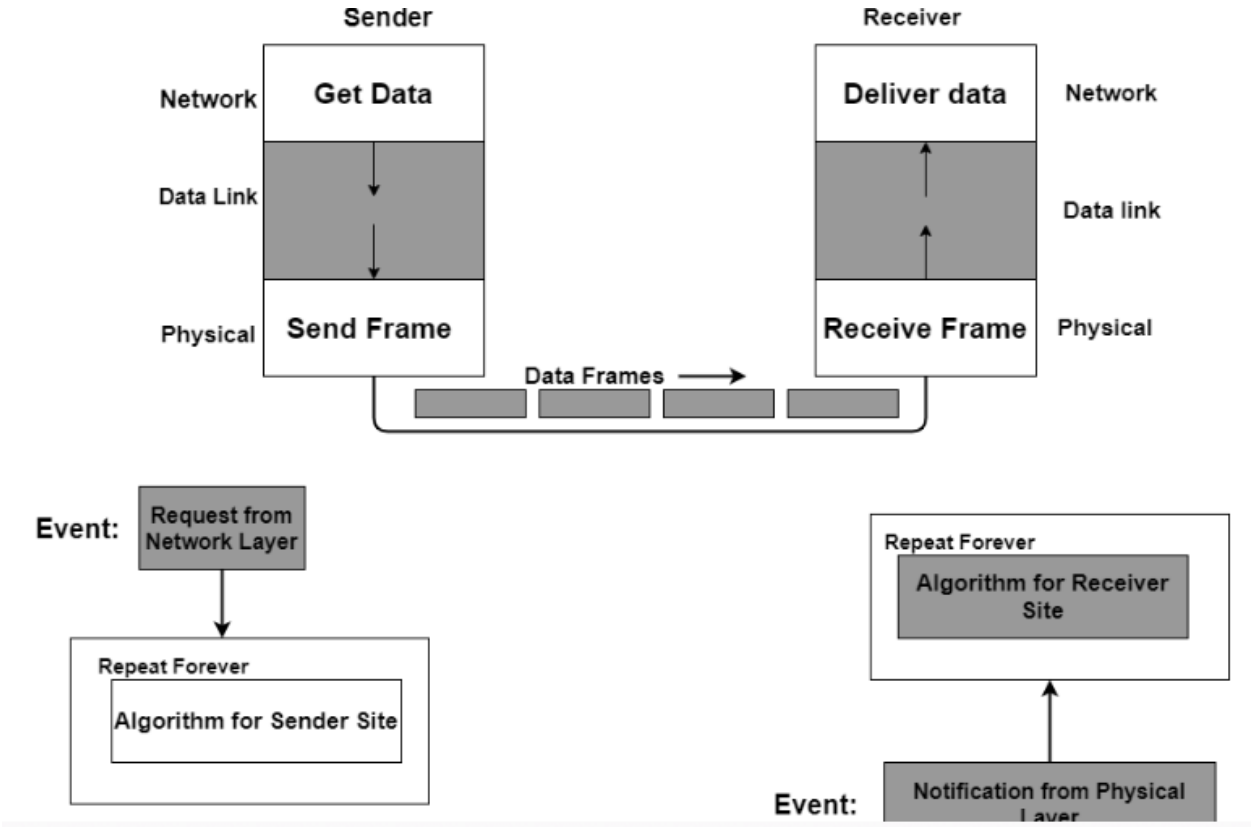
In this tutorial, we will be covering the Simplest Protocol that lies under the category Noiseless Channels in the Data link layer.

Simplest Protocol is a protocol that neither has **flow control** nor has **error control**(as we have already told you that it lies under the category of Noiseless channels).

- The simplest protocol is basically a **unidirectional protocol** in which data frames only travel in one direction; from the sender to the receiver.
- In this, the receiver can immediately handle the frame it receives whose processing time is small enough to be considered as negligible.
- Basically, the data link layer of the receiver immediately removes the header from the frame and then hand over the data packet to the network layer that also accepts the data packet immediately.
- We can also say that in the case of this protocol the receiver never gets overwhelmed with the incoming frames from the sender.

Design of the Simplest Protocol

The flow control is not needed by the Simplest Protocol. The data link layer at the sender side mainly gets the data from the network layer and then makes the frame out of data and sends it. On the Receiver site, the data link layer receives the frame from the physical layer and then extracts the data from the frame, and then delivers the data to its network layer.



The Datalink layers of both sender and receiver mainly provide transmission services for their Network layers. The data link layer also uses the services provided by the physical layer such as signaling, multiplexing, etc for the physical transmission of the bits.

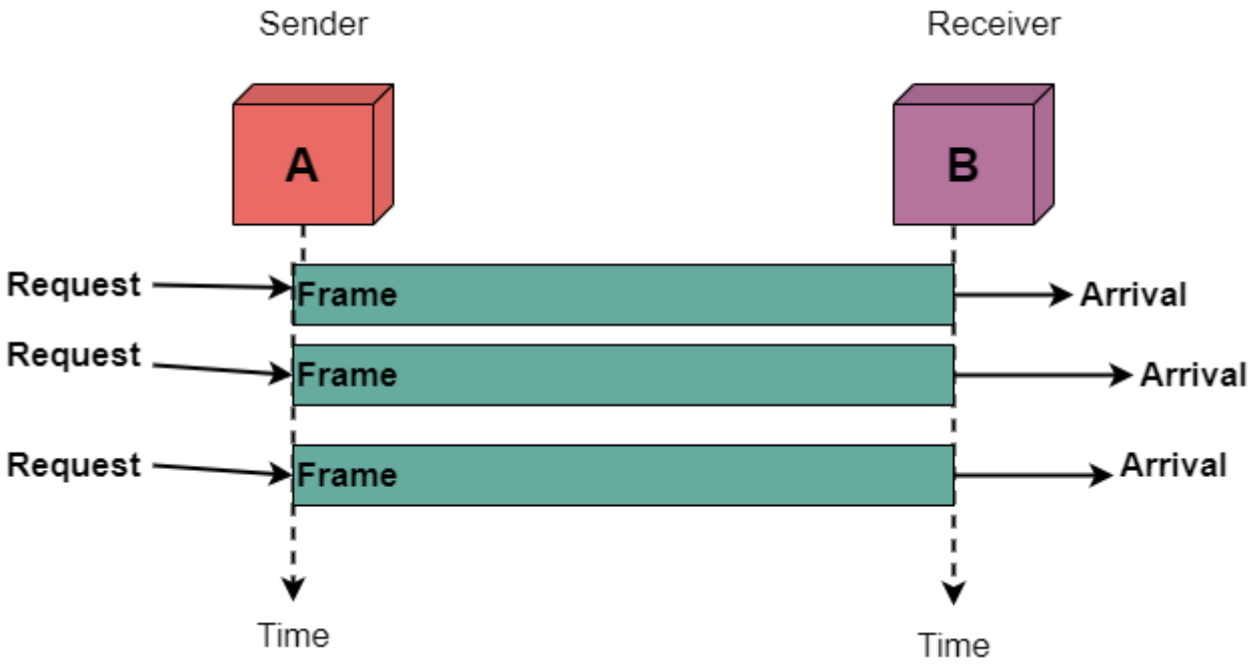
The procedure used by the data link layer

Let us now take a look at the procedure used by the data link layer at both sides(sender as well as the receiver).

- There is no frame send by the data link layer of the sender site until its network layer has a data packet to send.
- Similarly, the receiver site cannot deliver a data packet to its network layer until a frame arrives.
- In case if the implementation of the protocol is done as a procedure then there is a need to introduce the idea of events in the protocol.
- The procedure at the sender site runs constantly; there is no action until there is a request from the network layer.
- Also, the procedure at the receiver site runs constantly; there is no action until there is a notification from the physical layer.

Flow Diagram for Simplest Protocol

Using the simplest protocol the sender A sends a sequence of frames without even thinking about receiver B.



In order to send the three frames, there will be an occurrence of three events at sender A and three events at the receiver B.

It is important to note that in the above figure the data frames are shown with the help of boxes.

The height of the box mainly indicates the transmission time difference between the first bit and the last bit of the frame.

Stop-and-Wait Protocol

In this tutorial, we will be covering another protocol used in the **Noiseless** channels in the Data link layer.

Stop-and-wait Protocol is used in the data link layer for the transmission in the noiseless channels. Let us first understand why there is a need to use this protocol then we will cover this protocol in detail.

We have studied the simplest protocol in the previous tutorial, suppose there is a scenario in which the data frames arrive at the receiver's site faster than they can be processed means the rate of transmission is more than the processing rate of the frames. Also, it is normal that the receiver does not have enough space, and the data is also coming from multiple sources. Then due to all these, there may occur discarding of frames or denial of service.

In order to prevent the receiver from overwhelming, there is a need to tell the sender to slow down the transmission of frames. We can make use of feedback from the receiver to the sender.

Now from the next section, we will cover the **concept of the Stop-and-wait protocol.**

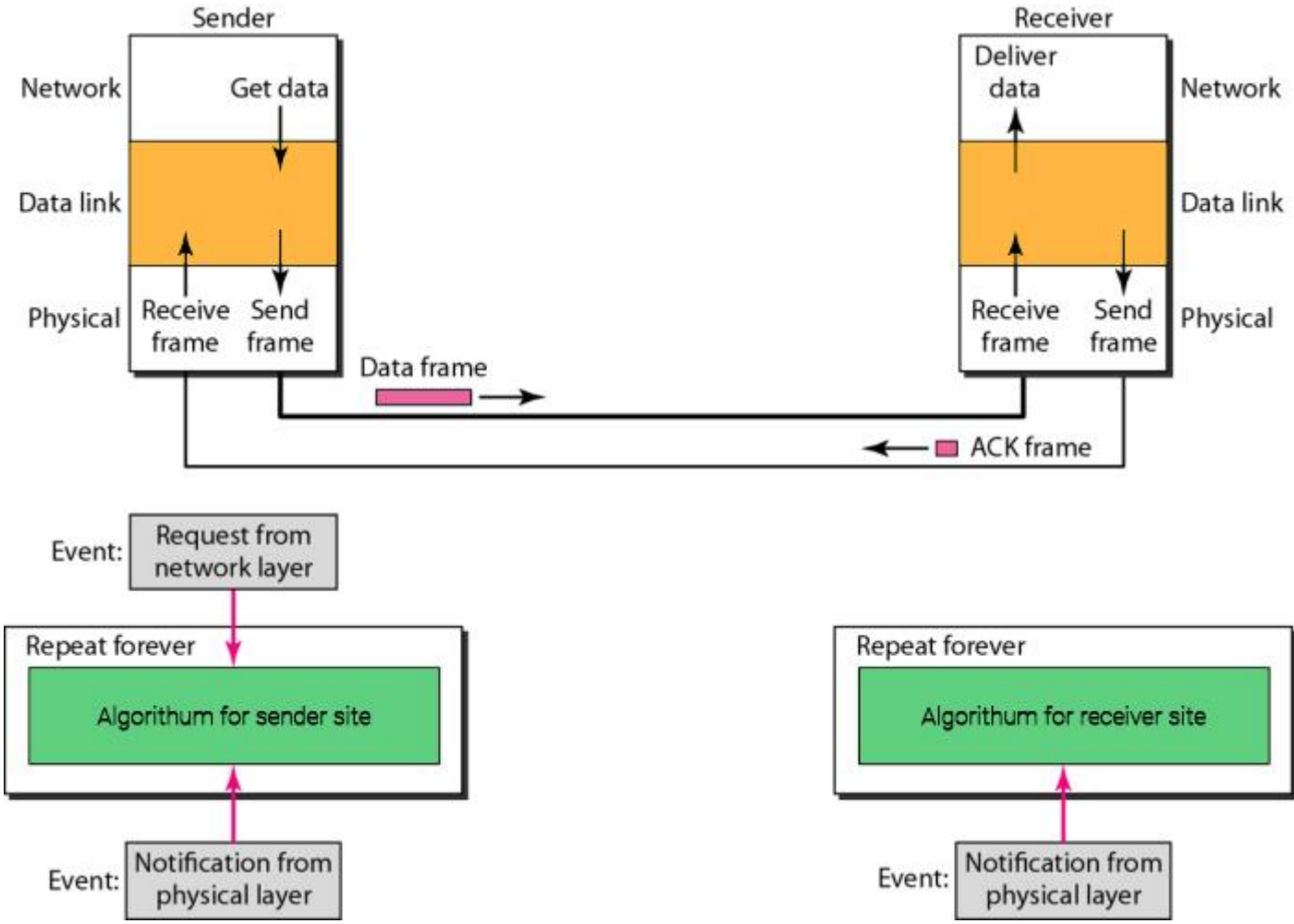
As the name suggests, when we use this protocol during transmission, then the sender sends one frame, then stops until it receives the confirmation from the receiver, after receiving the confirmation sender sends the next frame.

- There is **unidirectional communication** for the data frames, but the acknowledgment or ACK frames travel from the other direction. Thus the flow control is added here.
- Thus the stop-and-wait is one of the flow control protocol which makes the use of flow control service provided by the data link layer.
    - For every sent frame, the acknowledgment is needed and it takes the same amount of time for propagation in order to get back to the sender.

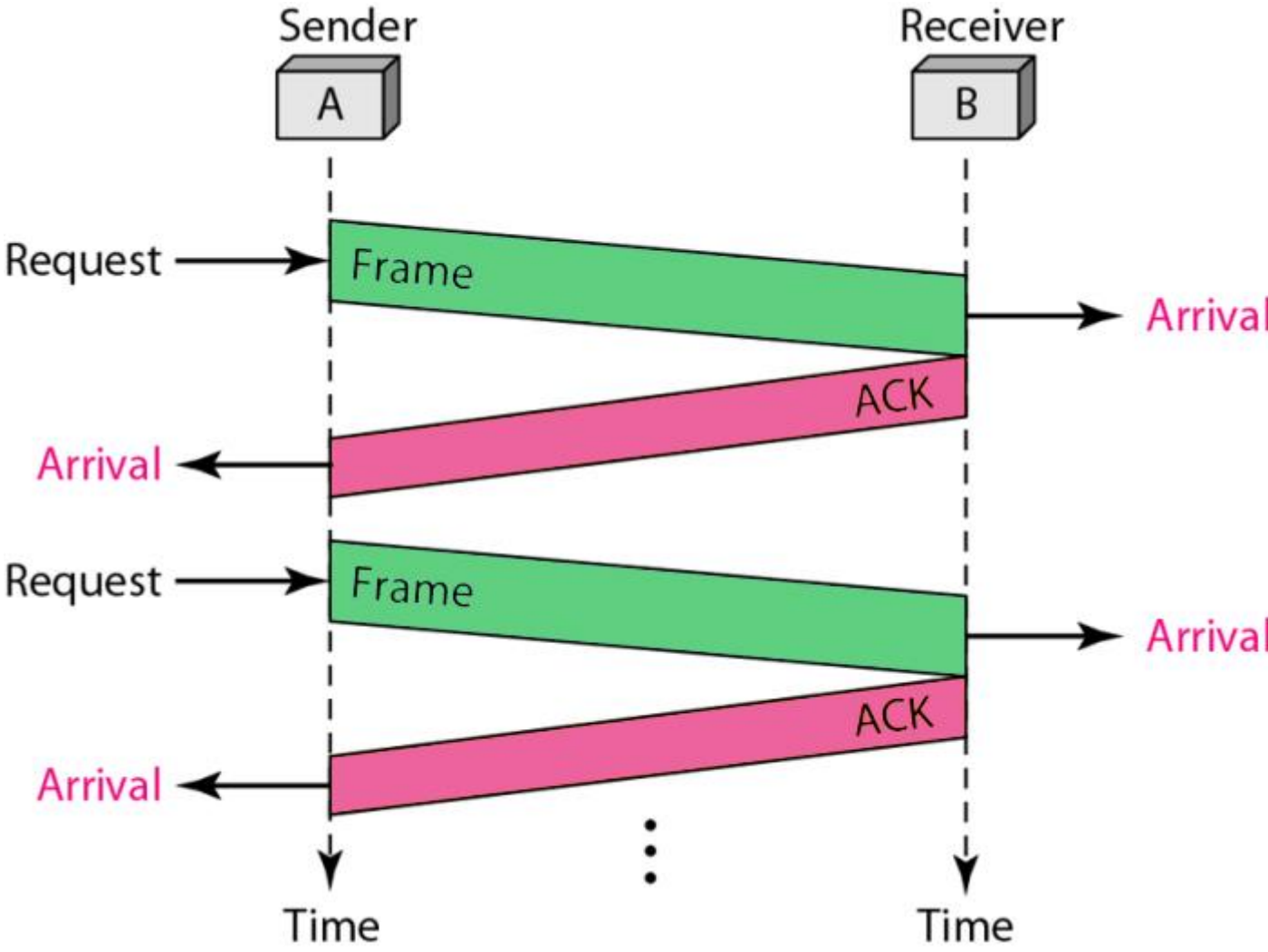
Design of the Stop-and-Wait protocol

Datalink layer at the sender side waits for its network layer in order to send the data packet. After that data link checks that it can send the frame or not. In case of receiving a positive notification from the physical layer; the data link layer makes the frame out of the data provided by the network layer and then sends it to the physical layer. After sending the data it will then wait for the acknowledgment before sending the next frame.

The data link layer on the receiver side waits for the frame to arrive. When the frame arrives then the receiver processes the frame and then delivers it to the network layer. After that, it will send the acknowledgment or we can say that ACK frame back to the sender.


Sender
Receiver
Network
Get data
Deliver data
Network
Data link
Data link
Physical
Receive frame
Send frame
Receive frame
Send frame
Physical
Data frame
ACK frame
Event:
Request from network layer
Repeat forever
Algorithum for sender site
Repeat forever
Algorithum for receiver site
Event:
Notification from physical layer
Event:
Notification from physical layer

Flow diagram of the stop-and-wait protocol



Advantages

One of the main advantages of the stop-and-wait protocol is the accuracy provided. As the transmission of the next frame is only done after receiving the acknowledgment of the previous frame. Thus there is no chance for data loss.

Disadvantages

Given below are some of the drawbacks of using the stop-and-wait Protocol:

- Using this protocol only one frame can be transmitted at a time.
- Suppose in a case, the frame is sent by the sender but it gets lost during the transmission and then the receiver can neither get it nor can send an acknowledgment back to the sender. Upon not receiving the acknowledgment the sender will not send the next frame. Thus there will occur two situations and these are: The receiver has to wait for an infinite amount of time for the data and the sender has to wait for an infinite amount of time in order to send the next frame.

- In the case of the transmission over a long distance, this is not suitable because the propagation delay becomes much longer than the transmission delay.
- In case the sender sends the data and this data has also been received by the receiver. After receiving the data the receiver then sends the acknowledgment but due to some reasons, this acknowledgment is received by the sender after the timeout period. Now as this acknowledgment is received too late; thus it can be wrongly considered as the acknowledgment of another data packet.
- The time spent waiting for the acknowledgment for each frame also adds up in the total transmission time.

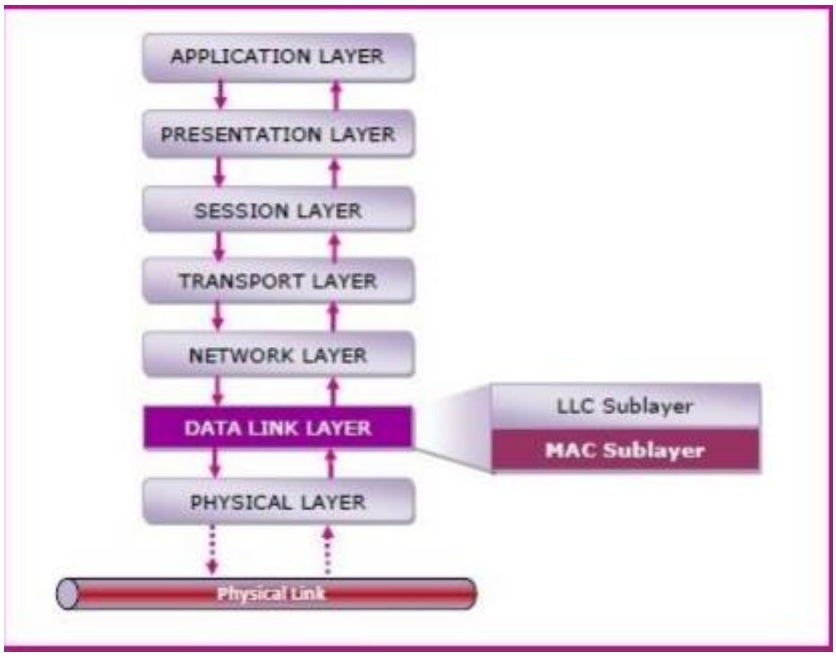
UNIT- II INTERNETWORKING AND MEDIA ACCESS CONTROL	
8.	Media Access Control
9.	Protocol Formats-Ethernet (802.3)
10.	Wireless LANs – 802.11 – Bluetooth
11.	Switching and Bridging
12.	Basic Internetworking-Protocols
13.	IP, CIDR, ARP, RARP, DHCP

Medium access control (MAC)

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 in the OSI Model 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



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.

Ethernet

Ethernet is a set of technologies and protocols that are used primarily in LANs. It was first standardized in 1980s by IEEE 802.3 standard. IEEE 802.3 defines the physical layer and the medium access control (MAC) sub-layer of the data link layer for wired Ethernet networks. Ethernet is classified into two categories: classic Ethernet and switched Ethernet.

Classic Ethernet is the original form of Ethernet that provides data rates between 3 to 10 Mbps. The varieties are commonly referred as 10BASE-X. Here, 10 is the maximum throughput, i.e. 10 Mbps, BASE denoted use of baseband transmission, and X is the type of medium used. Most varieties of classic Ethernet have become obsolete in present communication scenario.

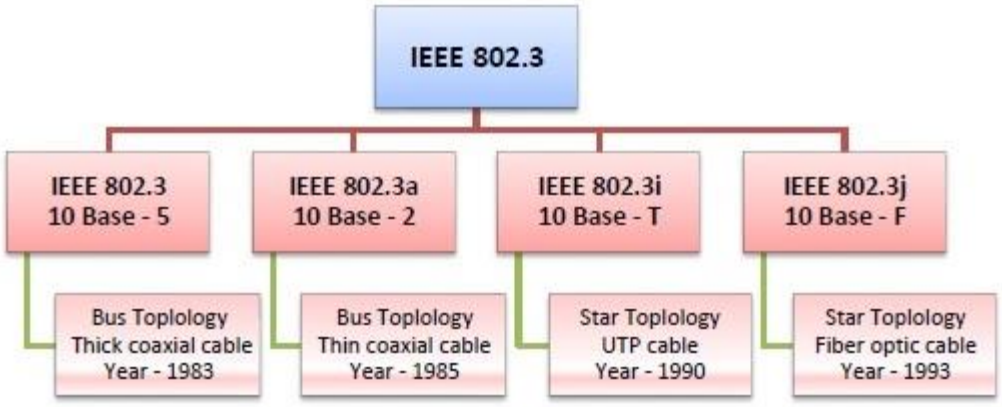
A switched Ethernet uses switches to connect to the stations in the LAN. It replaces the repeaters used in classic Ethernet and allows full bandwidth utilization.

IEEE 802.3 Popular Versions

There are a number of versions of IEEE 802.3 protocol. The most popular ones are -

- **IEEE 802.3**: This was the original standard given for 10BASE-5. It used a thick single coaxial cable into which a connection can be tapped by drilling into the cable to the core. Here, 10 is the maximum throughput, i.e. 10 Mbps, BASE denoted use of baseband transmission, and 5 refers to the maximum segment length of 500m.
- **IEEE 802.3a**: This gave the standard for thin coax (10BASE-2), which is a thinner variety where the segments of coaxial cables are connected by BNC connectors. The 2 refers to the maximum segment length of about 200m (185m to be precise).
- **IEEE 802.3i**: This gave the standard for twisted pair (10BASE-T) that uses unshielded twisted pair (UTP) copper wires as physical layer medium. The further variations were given by IEEE 802.3u for 100BASE-TX, 100BASE-T4 and 100BASE-FX.

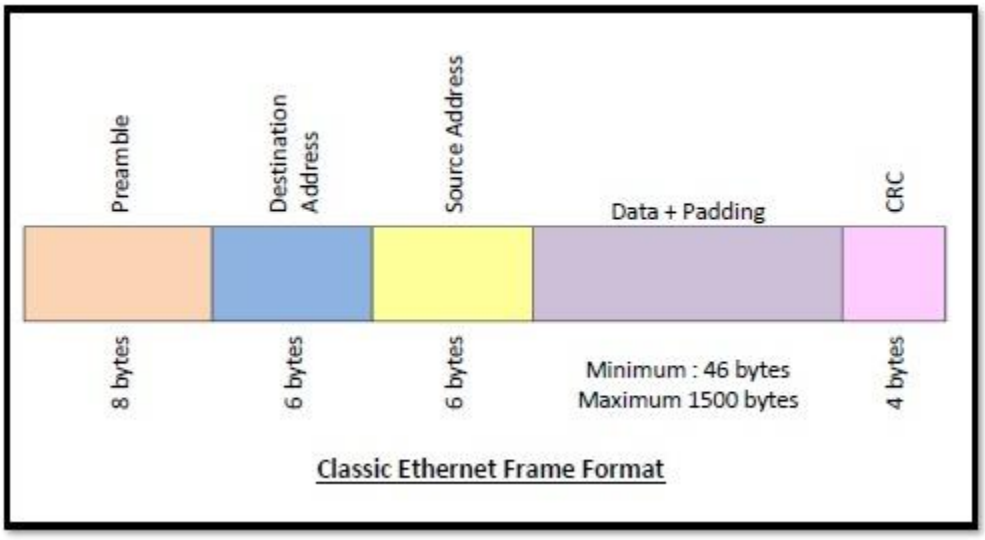
- **IEEE 802.3i**: This gave the standard for Ethernet over Fiber (10BASE-F) that uses fiber optic cables as medium of transmission.
- er optic cables as medium of transmission.

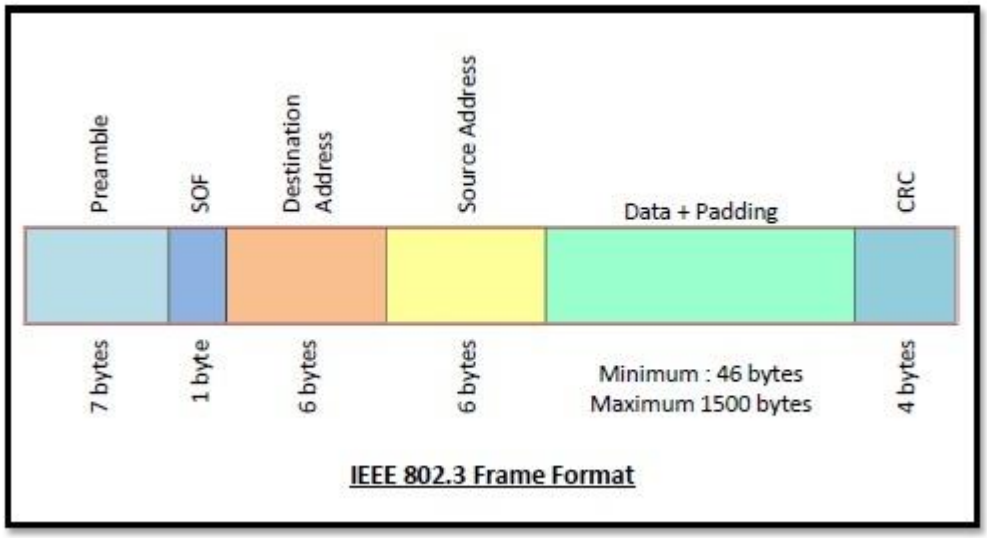


Frame Format of Classic Ethernet and IEEE 802.3

The main fields of a frame of classic Ethernet are -

- **Preamble**: It is the starting field that provides alert and timing pulse for transmission. In case of classic Ethernet it is an 8 byte field and in case of IEEE 802.3 it is of 7 bytes.
- **Start of Frame Delimiter**: It is a 1 byte field in a IEEE 802.3 frame that contains an alternating pattern of ones and zeros ending with two ones.
- **Destination Address**: It is a 6 byte field containing physical address of destination stations.
- **Source Address**: It is a 6 byte field containing the physical address of the sending station.
- **Length**: It a 7 bytes field that stores the number of bytes in the data field.
- **Data**: This is a variable sized field carries the data from the upper layers. The maximum size of data field is 1500 bytes.
- **Padding**: This is added to the data to bring its length to the minimum requirement of 46 bytes.
- **CRC**: CRC stands for cyclic redundancy check. It contains the error detection information.





Wireless LANs

Wireless LANs are those Local Area Networks that use high frequency radio waves instead of cables for connecting the devices in LAN. Users connected by WLANs can move around within the area of network coverage. Most WLANs are based upon the standard IEEE 802.11 or WiFi.

IEEE 802.11 Architecture

The components of an IEEE 802.11 architecture are as follows

1) Stations (STA) − Stations comprise all devices and equipments that are connected to the wireless LAN. A station can be of two types:

- **Wireless Access Pointz (WAP)** − WAPs or simply access points (AP) are generally wireless routers that form the base stations or access.
- **Client.** − Clients are workstations, computers, laptops, printers, smartphones, etc.

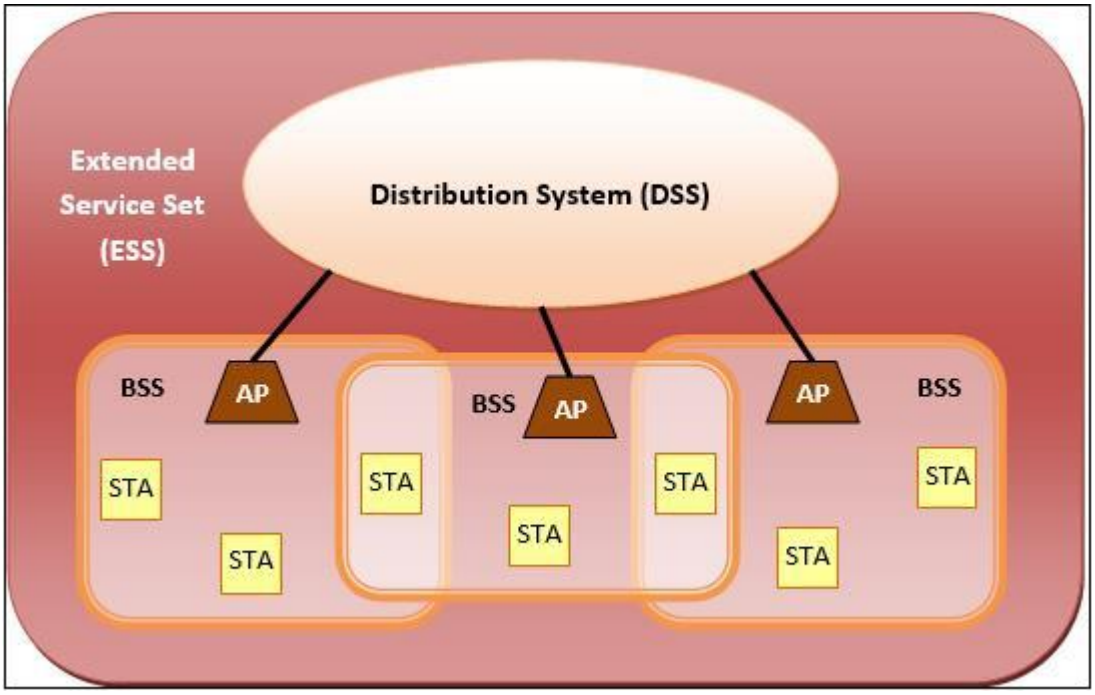
Each station has a wireless network interface controller.

2) Basic Service Set (BSS) −A basic service set is a group of stations communicating at physical layer level. BSS can be of two categories depending upon mode of operation:

- **Infrastructure BSS** − Here, the devices communicate with other devices through access points.
- **Independent BSS** − Here, the devices communicate in peer-to-peer basis in an ad hoc manner.

3) Extended Service Set (ESS) − It is a set of all connected BSS.

4) Distribution System (DS) − It connects access points in ESS.



Advantages of WLANs

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

Disadvantages of WLANs

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

Bluetooth

Blootooth is an universal for short range wireless voice and data communication.It is a Wireless Personal Area Network (WPAN) technology and is used for exchanging data over smaller distances. This technology was invented by Ericson in 1994. It operates in the unlicensed, industrial, scientific and medical (ISM) band from 2.4 GHz to 2.485 GHz. Maximum devices that can be connected at the same time are 7. Bluetooth ranges up to 10 meters. It provides data rates up to 1 Mbps or 3 Mbps depending upon the version. The spreading technique that it uses is FHSS (Frequency-hopping spread spectrum). A Bluetooth network is called a **piconet** and a collection of interconnected piconets is called **scatternet**.

What is bluetooth.

Bluetooth Transmission capacity 720 kbps.

Bluetooth is Wireless.

Bluetooth is Low cost short distance radio communications standard .

Bluetooth is robust and flexible .

Bluetooth is cable replacement technology that can be used to connect almost any device to any other device.

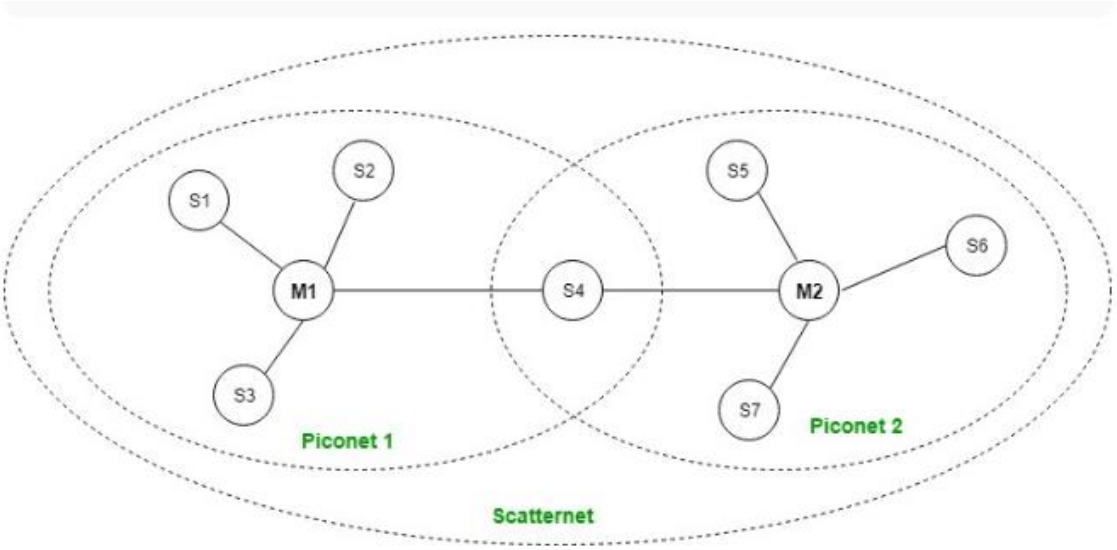
The basic architecture unit of a bluetooth is a piconet.

Bluetooth Architecture:

The architecture of Bluetooth defines two types of networks:

1. Piconet

2. Scatternet



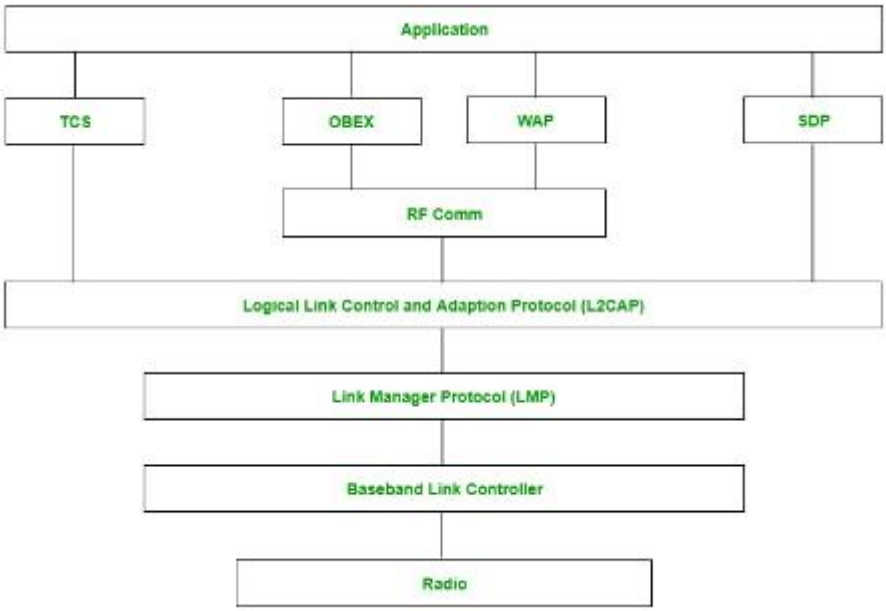
Piconet:

Piconet is a type of Bluetooth network that contains **one primary node** called the master node and **seven active secondary nodes** called slave nodes. Thus, we can say that there is a total of 8 active nodes which are present at a distance of 10 meters. The communication between the primary and secondary nodes can be one-to-one or one-to-many. Possible communication is only between the master and slave; Slave-slave communication is not possible. It also has **255 parked nodes**, these are secondary nodes and cannot take participation in communication unless it gets converted to the active state.

Scatternet:

It is formed by using **various piconets**. A slave that is present in one piconet can act as master or we can say primary in another piconet. This kind of node can receive a message from a master in one piconet and deliver the message to its slave in the other piconet where it is acting as a slave. This type of node is referred to as a bridge node. A station cannot be mastered in two piconets.

Bluetooth protocol stack:



1. **Radio (RF) layer:** It specifies the details of the air interface, including frequency, the use of frequency hopping and transmit power. It performs modulation/demodulation of the data into RF signals. It defines the physical characteristics of Bluetooth transceivers. It defines two types of physical links: connection-less and connection-oriented.

2. **Baseband Link layer:** The baseband is the digital engine of a Bluetooth system and is equivalent to the MAC sublayer in LANs. It performs the connection establishment within a piconet, addressing, packet format, timing and power control.

3. **Link Manager protocol layer:** It performs the management of the already established links which includes authentication and encryption processes. It is responsible for creating the links, monitoring their health, and terminating them gracefully upon command or failure.

4. **Logical Link Control and Adaption (L2CAP) Protocol layer:** It is also known as the heart of the Bluetooth protocol stack. It allows the communication between upper and lower layers of the Bluetooth protocol stack. It packages the data packets received from upper layers into the form expected by lower layers. It also performs segmentation and multiplexing.

5. **Service Discovery Protocol (SDP) layer:** It is short for Service Discovery Protocol. It allows discovering the services available on another Bluetooth-enabled device.

6. **RF comm layer:** It is a cabal replacement protocol. It is short for Radio Frontend Component. It provides a serial interface with WAP and OBEX. It also provides emulation of serial ports over the logical link control and adaption protocol(L2CAP). The protocol is based on the ETSI standard TS 07.10.

7. **OBEX:** It is short for Object Exchange. It is a communication protocol to exchange objects between 2 devices.

8. **WAP:** It is short for Wireless Access Protocol. It is used for internet access.

9. **TCS:** It is short for Telephony Control Protocol. It provides telephony service. The basic function of this layer is call control (setup & release) and group management for gateway serving multiple devices.

10. **Application layer:** It enables the user to interact with the application.

Advantage:

- Low cost.
- Easy to use.
- It can also penetrate through walls.
- It creates an Ad-hoc connection immediately without any wires.
- It is used for voice and data transfer.

Disadvantages:

- It can be hacked and hence, less secure.
- It has a slow data transfer rate: of 3 Mbps.
- It has a small range: 10 meters.
- Bluetooth communication does not support routing.
- The issues of handoffs have not been addressed.

Applications:

- Used in laptops, and in wireless PCs.
- In printers.
- In wireless headsets.
- Connecting digital camera wirelessly to a mobile phone.
- Data transfer from one cell phone to other cell phone or computer.
- Medical health care
- sports and fitness
- military
- security
- Consumer ,
- games,

- professional.
- Services.
- Industry

Switching

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

Why is Switching Concept required?

Switching concept is developed because of the following reasons:

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

Advantages of Switching:

- Switch increases the bandwidth of the network.

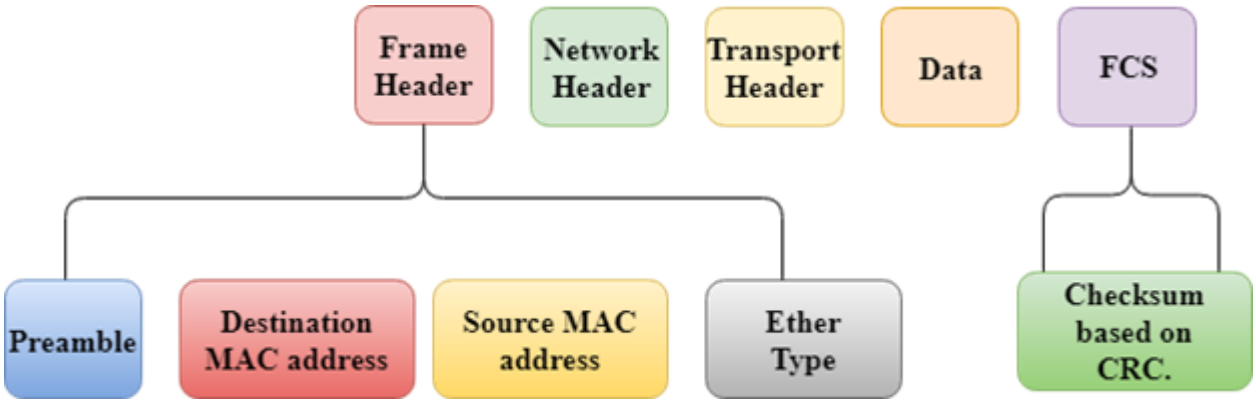
- It reduces the workload on individual PCs as it sends the information to only that device which has been addressed.
- It increases the overall performance of the network by reducing the traffic on the network.
- There will be less frame collision as switch creates the collision domain for each connection.

Disadvantages of Switching:

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

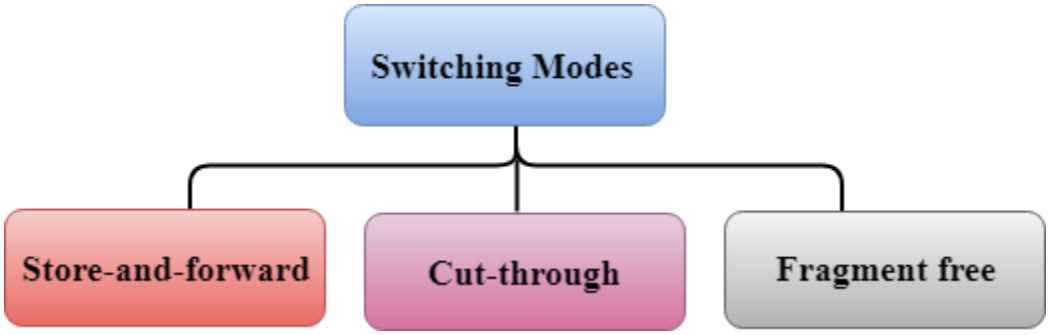
Switching Modes

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

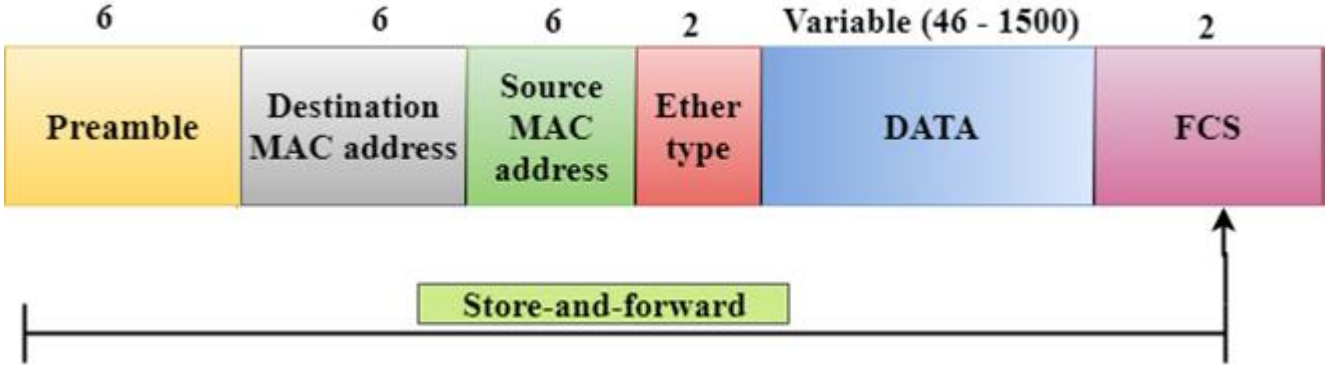


There are three types of switching modes:

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

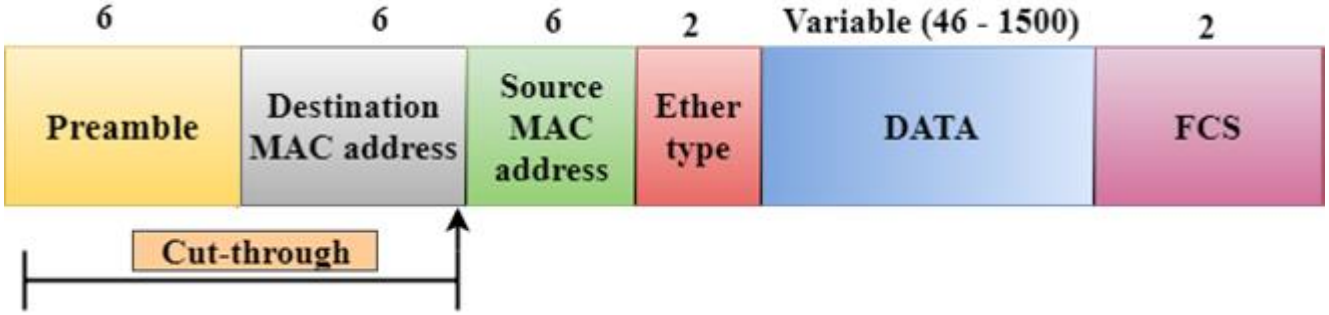


Store-and-forward



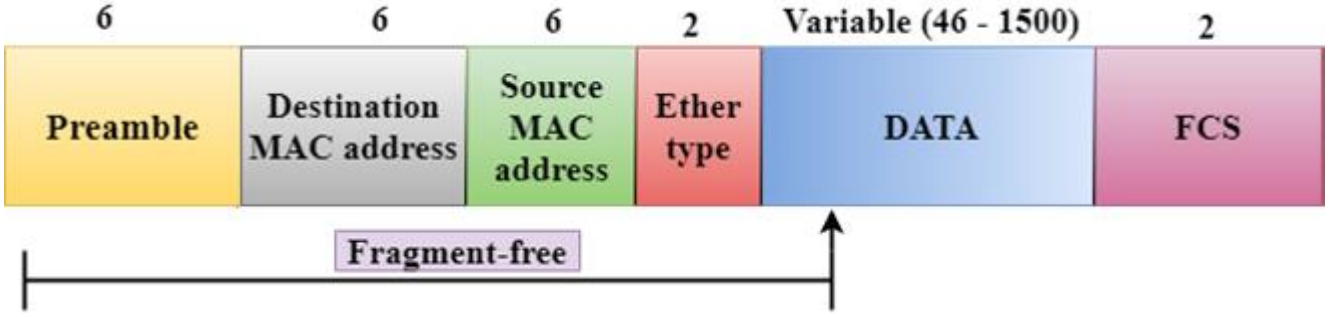
- Store-and-forward is a technique in which the intermediate nodes store the received frame and then check for errors before forwarding the packets to the next node.
- The layer 2 switch waits until the entire frame has received. On receiving the entire frame, switch store the frame into the switch buffer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

Cut-through Switching



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

Fragment-free Switching



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

- This technique checks the 64 bytes of the ethernet frame where addressing information is available.
- A collision is detected within 64 bytes of the frame, the frames which are collided will not be forwarded further.

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

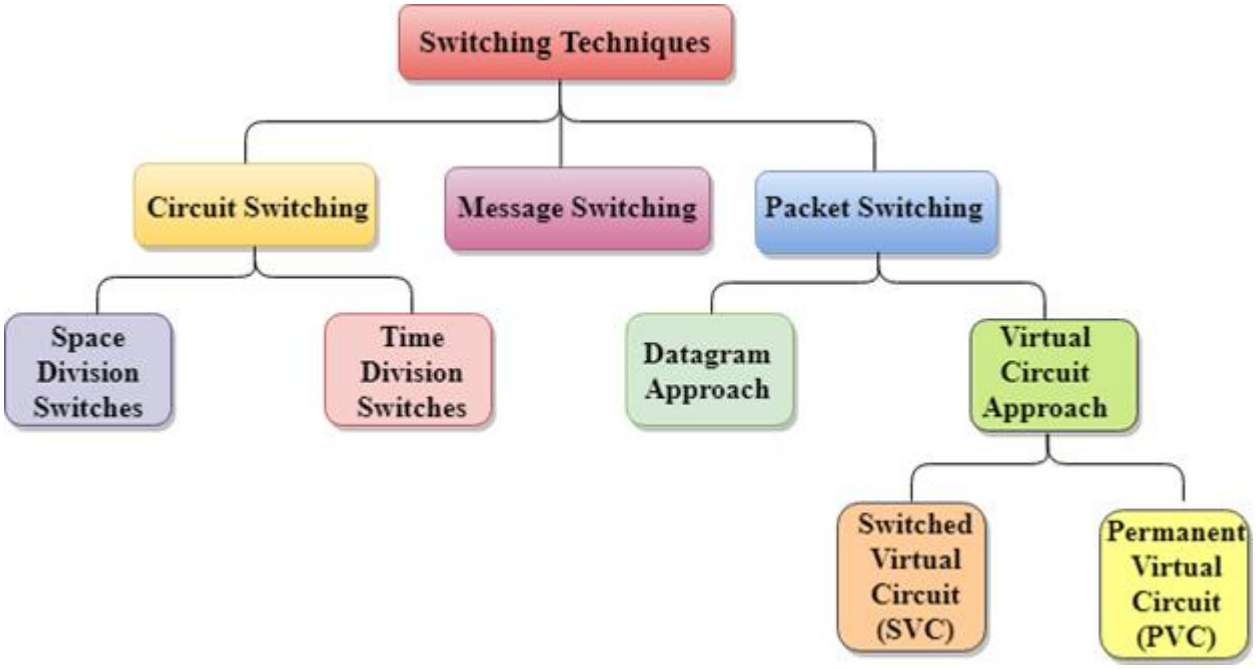
Store-and-forward Switching	Cut-through Switching
Store-and-forward Switching is a technique that waits until the entire frame is received.	Cut-through Switching is a technique that checks the first 6 bytes following the preamble to identify the destination address.
It performs error checking functionality.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

Switching techniques

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

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

Classification Of Switching Techniques

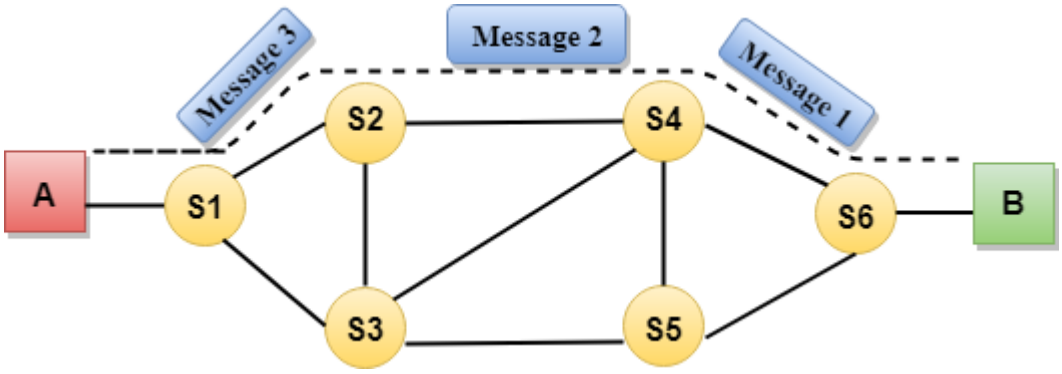


Circuit Switching

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

Communication through circuit switching has 3 phases:

- Circuit establishment
- Data transfer
- Circuit Disconnect



Circuit Switching can use either of the two technologies:

Space Division Switches:

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

Space Division Switches can be categorized in two ways:

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

Crossbar Switch

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

Disadvantage of Crossbar switch:

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

Multistage Switch

- Multistage Switch is made by splitting the crossbar switch into the smaller units and then interconnecting them.
- It reduces the number of crosspoints.

- If one path fails, then there will be an availability of another path.

Advantages Of Circuit Switching:

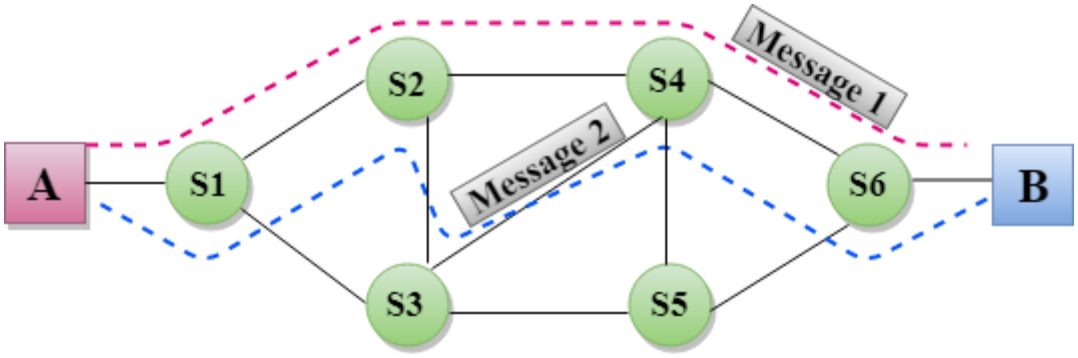
- In the case of Circuit Switching technique, the communication channel is dedicated.
- It has fixed bandwidth.

Disadvantages Of Circuit Switching:

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

Message Switching

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



Advantages Of Message Switching

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

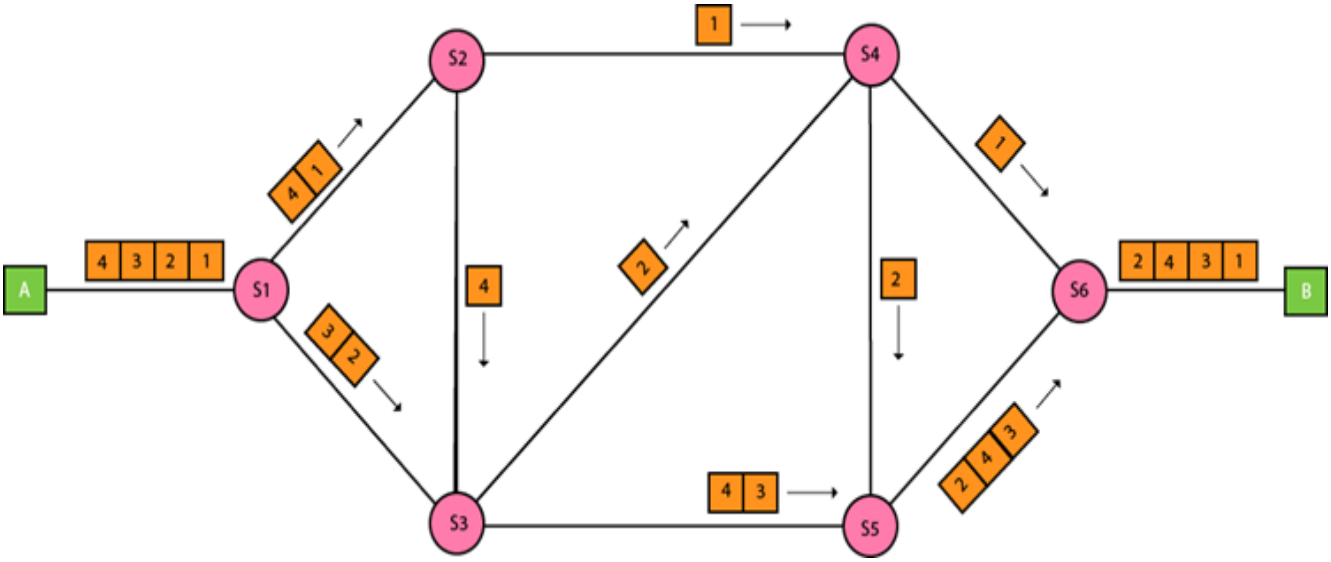
Disadvantages of Message Switching

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

Packet Switching

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

- If the correct order of the packets is reached, then the acknowledgment message will be sent.



Approaches Of Packet Switching:

There are two approaches to Packet Switching:

Datagram Packet switching:

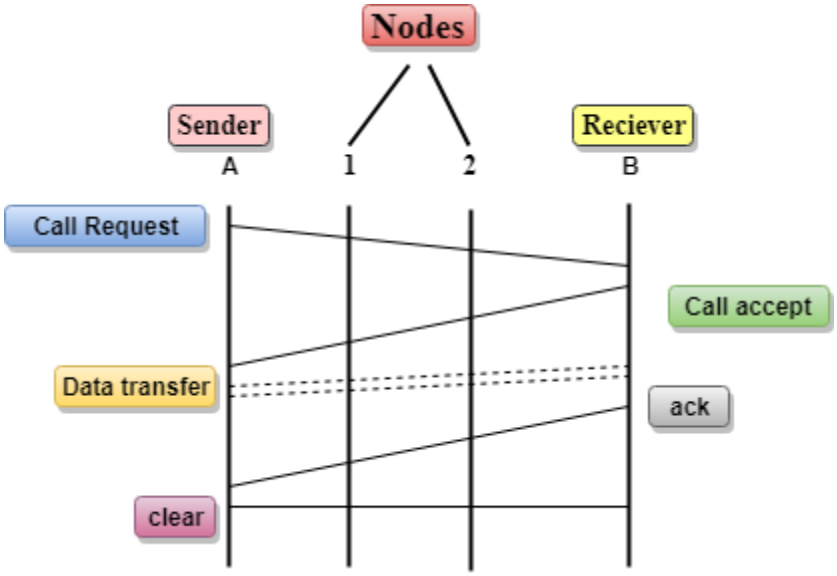
- It is a packet switching technology in which packet is known as a datagram, is considered as an independent entity. Each packet contains the information about the destination and switch uses this information to forward the packet to the correct destination.
- The packets are reassembled at the receiving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

Virtual Circuit Switching

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

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

Datagram approach	Virtual Circuit approach



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

Differences b/w Datagram approach and Virtual Circuit approach

Node takes routing decisions to forward the packets.	Node does not take any routing decision.
Congestion cannot occur as all the packets travel in different directions.	Congestion can occur when the node is busy, and it does not allow other packets to pass through.
It is more flexible as all the packets are treated as an independent entity.	It is not very flexible.

Advantages Of Packet Switching:

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

Disadvantages Of Packet Switching:

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

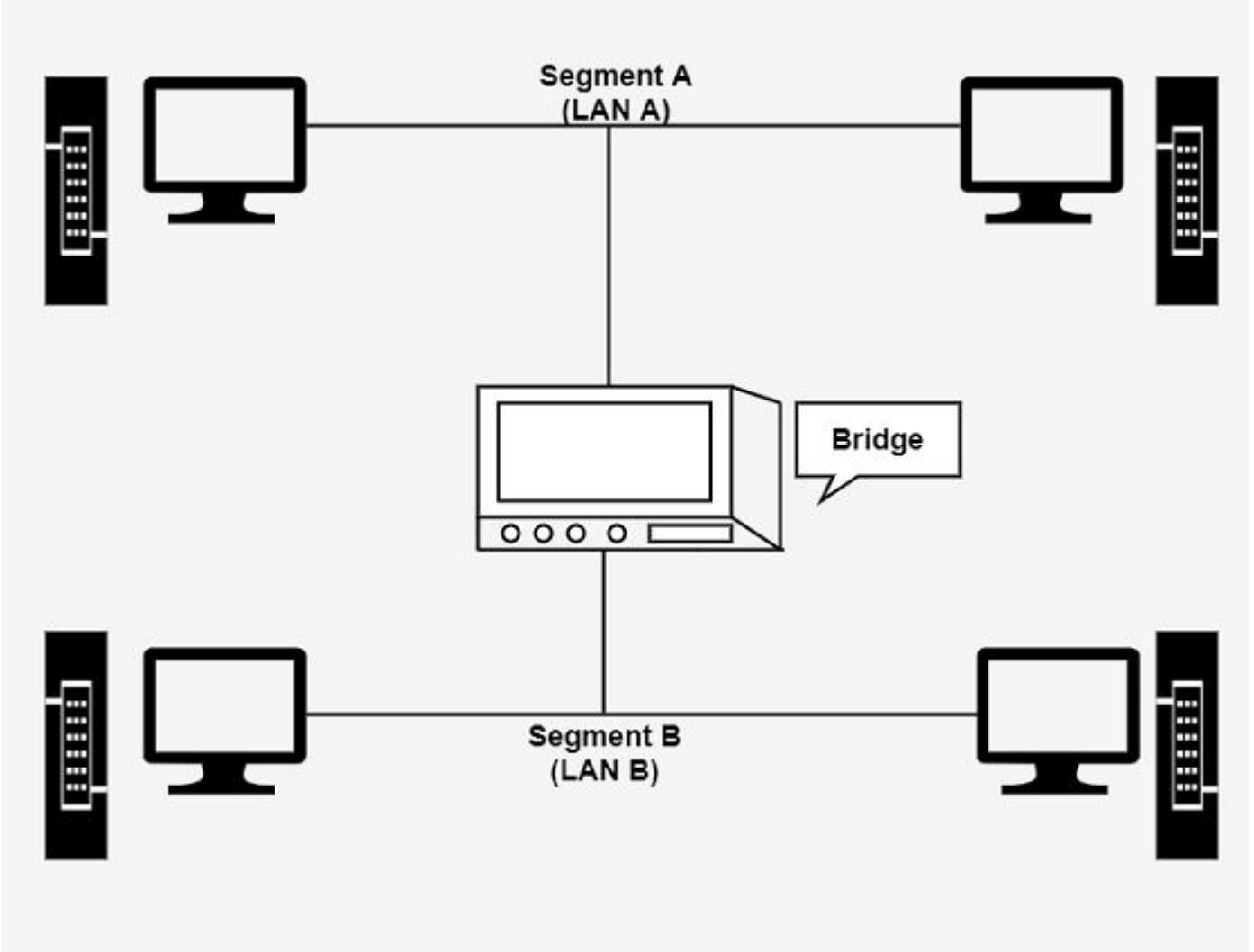
Bridges

Bridges are used to connect two subnetworks that use interchangeable protocols. It combines two LANs to form an extended LAN. The main difference between the bridge and repeater is that the bridge has a penetrating efficiency.

Working of Bridges

A bridge accepts all the packets and amplifies all of them to the other side. The bridges are intelligent devices that allow the passing of only selective packets from them. A bridge only

passes those packets addressed from a node in one network to another node in the other network.

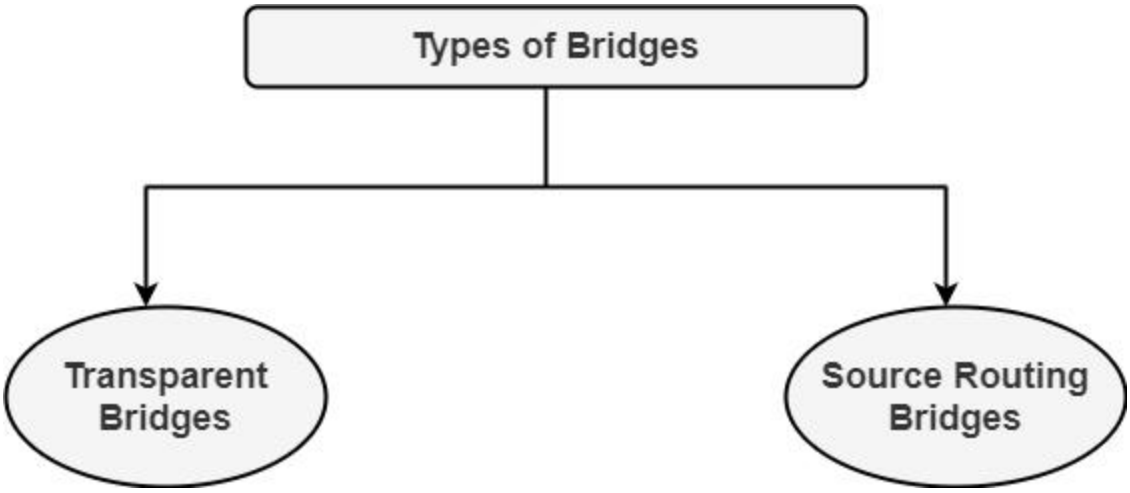


A bridge performs in the following aspect −

- A bridge receives all the packets or frame from both LAN (segment) A and B.
- A bridge builds a table of addresses from which it can identify that the packets are sent from which LAN (or segment) to which LAN.
- The bridge reads the send and discards all packets from LAN A sent to a computer on LAN A and that packets from LAN A send to a computer on LAN B are retransmitted to LAN B.
- The packets from LAN B are considered in the same method.

Types of Bridges

There are generally two types of bridges which are as follows −



Transparent Bridges

It is also called learning bridges. Bridge construct its table of terminal addresses on its own as it implements connecting two LANs. It facilitates the source location to create its table. It is self-updating. It is a plug and plays bridge.

Source Routing Bridge

This sending terminal means the bridges that the frames should stay. This type of bridge is used to prevent looping problem.

Uses of Bridges

The main uses of bridges are −

- Bridges are used to divide large busy networks into multiple smaller and interconnected networks to improve performance.
- Bridges also can increase the physical size of a network.
- Bridges are also used to connect a LAN segment through a synchronous modem relation to another LAN segment at a remote area.

Internet Protocols

- Internet Protocols are a set of rules that governs the communication and exchange of data over the internet. Both the sender and receiver should follow the same protocols in order to communicate the data. In order to understand it better, let's take an example of a language. Any language has its own set of vocabulary and grammar which we need to know if we want to communicate in that language. Similarly, over the internet whenever we access a website or exchange some data with another device then these processes are governed by a set of rules called the internet protocols.
- **Working of internet protocol:** The internet and many other data networks work by organizing data into small pieces called packets. Each large data sent between two network devices is divided into smaller packets by the underlying hardware and software. Each network protocol defines the rules for how its data packets must be organized in specific ways according to the protocols the network supports.
- **Why do we need protocols?**
- It may be that the sender and receiver of data are parts of different networks, located in different parts of the world having different data transfer rates. So, we need

protocols to manage the flow control of data, access control of the link being shared in the communication channel. Suppose there is a sender X who has a data transmission rate of 10 Mbps. And, there is a receiver Y who has a data receiving rate of 5Mbps. Since the rate of receiving the data is slow so some data will be lost during transmission. In order to avoid this, the receiver Y needs to inform sender X about the speed mismatch so that the sender X can adjust its transmission rate. Similarly, the access control decides the node which will access the link shared in the communication channel at a particular instant of time. If not the transmitted data will collide if many computers send data simultaneously through the same link resulting in the corruption or loss of data.

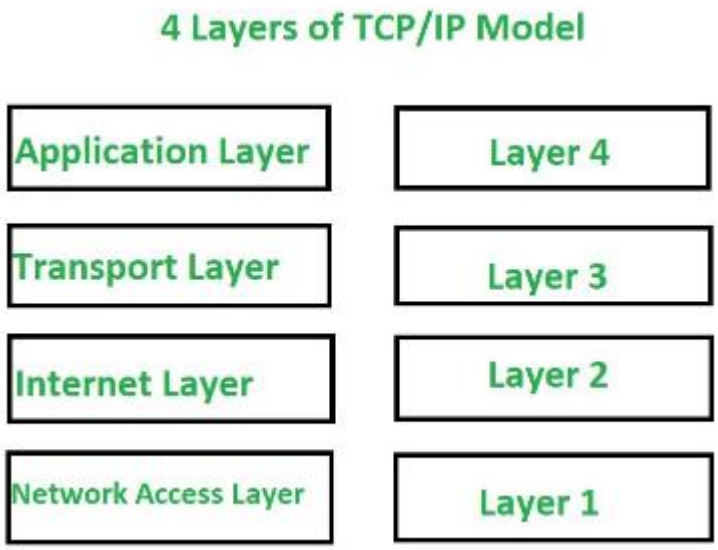
Types of internet protocol

The Internet Protocols are of different types having different uses:-

1. TCP/IP(Transmission Control Protocol/ Internet Protocol): These are a set of standard rules that allows different types of computers to communicate with each other. The IP protocol ensures that each computer that is connected to the Internet is having a specific serial number called the IP address. TCP specifies how data is exchanged over the internet and how it should be broken into IP packets. It also makes sure that the packets have information about the source of the message data, the destination of the message data, the sequence in which the message data should be re-assembled, and checks if the message has been sent correctly to the specific destination. The TCP is also known as a connection-oriented protocol.

The functionality of TCP/IP is divided into 4 layers with each one having specific protocols:

1. **Application Layer:** The application layer makes sure that the data from the sending end is received in a format that is acceptable and supported at the receiving end.
2. **Transport Layer:** The transport layer is responsible for the smooth transmission of data from one end to the other. It is also responsible for reliable connectivity, error recovery, and flow control of the data.
3. **Internet Layer:** This Internet Layer moves packets from source to destination by connecting independent networks.
4. **Network Access Layer:** The Network Access Layer sees how a computer connects to a network.



2. SMTP(Simple Mail Transfer Protocol): These protocols are important for sending and distributing outgoing emails. This protocol uses the header of the mail to get the email id of the receiver and enters the mail into the queue of outgoing mails. And as soon as, it delivers the mail to the receiving email id, it removes the email from the outgoing list. The message or the electronic mail may consider of text, video, image etc. It helps in setting up of some communication server rules.

3. PPP(Point to Point Protocol): It is a communication protocol that is used to create a direct connection between two communicating devices. This protocol defines the rules using which two devices will authenticate with each other and exchange information with each other. For example, A user connects his PC to the server of an Internet Service Provider also uses PPP. Similarly, for connecting two routers for direct communication it uses PPP.

4. FTP (File Transfer Protocol): This protocol is used for transferring files from one system to the other. This works on a client-server model. When a machine requests for file transfer from another machine, the FTO sets up a connection between the two and authenticates each other using their ID and Password. And, the desired file transfer takes place between the machines.

5. SFTP(Secure File Transfer Protocol): SFTP which is also known as SSH FTP refers to File Transfer Protocol (FTP) over Secure Shell (SSH) as it encrypts both commands and data while in transmission. SFTP acts as an extension to SSH and encrypts files and data then sends them over a secure shell data stream. This protocol is used to remotely connect to other systems while executing commands from the command line.

6. HTTP(Hyper Text Transfer Protocol): This protocol is used to transfer hypertexts over the internet and it is defined by the www(world wide web) for information transfer. This protocol defines how the information needs to be formatted and transmitted. And, it also defines the various actions the web browsers should take in response to the calls made to access a particular web page. Whenever a user opens their web browser, the user will indirectly use HTTP as this is the protocol that is being used to share text, images, and other multimedia files on the World Wide Web.

Note: *Hypertext refers to the special format of the text that can contain links to other texts.*

7. HTTPS(HyperText Transfer Protocol Secure): HTTPS is an extension of the Hypertext Transfer Protocol (HTTP). It is used for secure communication over a computer network with the SSL/TLS protocol for encryption and authentication. So, generally, a website has an HTTP protocol but if the website is such that it receives some sensitive information such as credit card details, debit card details, OTP, etc then it requires an SSL certificate installed to make the website more secure. So, before entering any sensitive information on a website, we should check if the link is HTTPS or not. If it is not HTTPS then it may not be secure enough to enter sensitive information.

8. TELNET(Terminal Network): TELNET is a standard TCP/IP protocol used for virtual terminal service given by ISO. This enables one local machine to connect with another. The computer which is being connected is called a remote computer and which is connecting is called the local computer. TELNET operation lets us display anything being performed on the remote computer in the local computer. This operates on the client/server principle. The local computer uses the telnet client program whereas the remote computer uses the telnet server program.

9. POP3(Post Office Protocol 3): POP3 stands for Post Office Protocol version 3. It has two Message Access Agents (MAAs) where one is client MAA (Message Access Agent) and another is server MAA(Message Access Agent) for accessing the messages from the mailbox. This protocol helps us to retrieve and manage emails from the mailbox on the

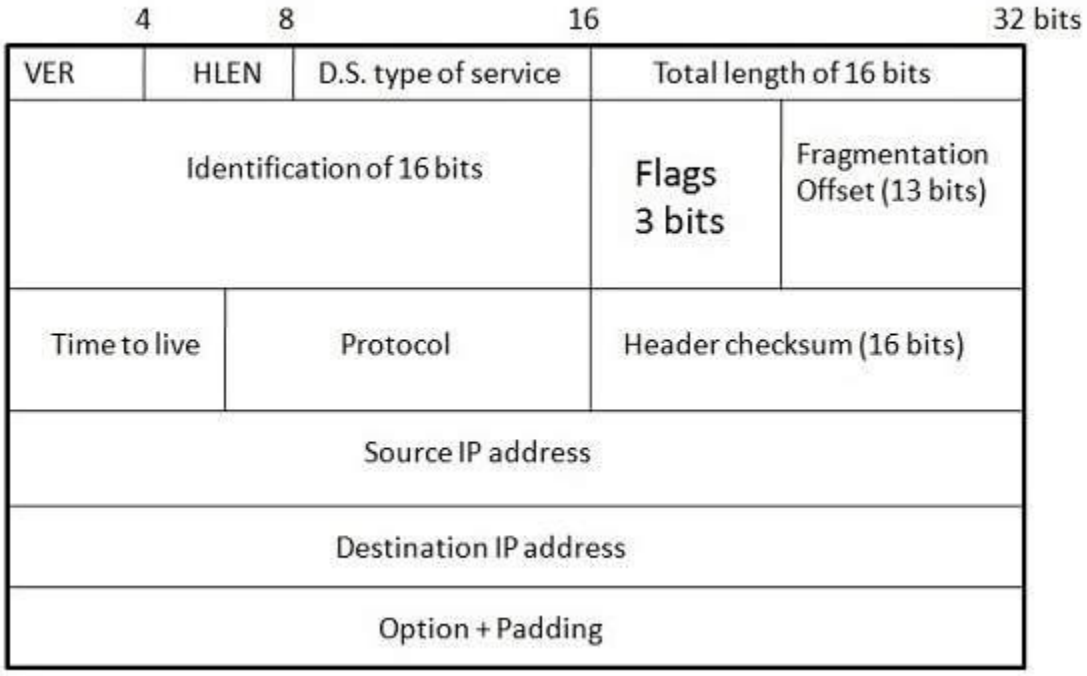
receiver mail server to the receiver's computer. This is implied between the receiver and receiver mail server. It can also be called as one way client server protocol. The POP3 WORKS ON THE 2 PORTS I.E. PORT 110 AND PORT 995.

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.

What is CIDR (Classless Inter-Domain Routing or supernetting)?

CIDR (Classless Inter-Domain Routing or supernetting) is a method of assigning IP addresses that improves the efficiency of address distribution and replaces the previous system based on Class A, Class B and Class C networks.

The initial goal of CIDR was to slow the increase of routing tables on routers across the internet and decrease the rapid exhaustion of IPv4 addresses. As a result, the number of available internet addresses has greatly increased.

The original classful network design of the internet included inefficiencies that drained the pool of unassigned IPv4 addresses faster than necessary. The classful design included the following:

- Class A, with over 16 million identifiers
- Class B, with 65,535 identifiers
- Class C, with 254 host identifiers

If an organization needed more than 254 host machines, it would be switched into Class B. However, this could potentially waste over 60,000 hosts if the business didn't need to use them, thus unnecessarily decreasing the availability of IPv4 addresses. The Internet Engineering Task Force introduced CIDR in 1993 to fix this problem.

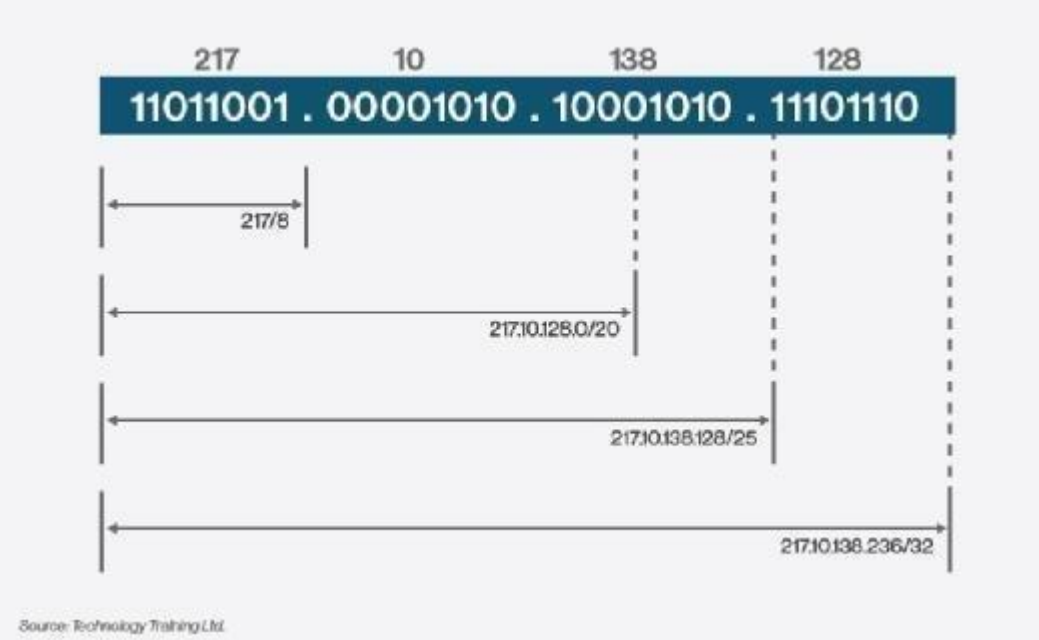
CIDR is based on variable-length subnet masking (VLSM), which enables network engineers to divide an IP address space into a hierarchy of subnets of different sizes. This makes it possible to create subnetworks with different host counts without wasting large numbers of addresses.

CIDR addresses are made up of two sets of numbers:

1. **Prefix.** The prefix is the binary representation of the network address -- similar to what would be seen in a normal IP address.
2. **Suffix.** The suffix declares the total number of bits in the entire address.

For example, CIDR notation might look like: 192.168.129.23/17 -- with 17 being the number of bits in the address. IPv4 addresses support a maximum of 32 bits.

The same CIDR notation can be applied to IPv6 addresses. The only difference is IPv6 addresses can contain up to 128 bits.


217
10
138
128
11011001 . 00001010 . 10001010 . 11101110
217/8
217.10.128.0/20
217.10.138.128/25
217.10.138.236/32
Source: Technology Training Ltd.

Address Resolution Protocol (ARP) and its types

Address Resolution Protocol (ARP) is a communication protocol used to find the MAC (Media Access Control) a device from its IP address. This protocol is used when a device wants to communicate with another device on a a Network or Ethernet.

Types of ARP

There are four types of Address Resolution Protocol, which is given below:

- Proxy ARP
- Gratuitous ARP
- Reverse ARP (RARP)
- Inverse ARP

Proxy ARP - Proxy ARP is a method through which a Layer 3 devices may respond to ARP requests for a target a different network from the sender. The Proxy ARP configured router responds to the ARP and map the MAC the router with the target IP address and fool the sender that it is reached at its destination.

At the backend, the proxy router sends its packets to the appropriate destination because the packets contain the

information.

Example - If Host A wants to transmit data to Host B, which is on the different network, then Host A sends an est message to receive a MAC address for Host B. The router responds to Host A with its own MAC address self as a destination. When the data is transmitted to the destination by Host A, it will send to the gateway so that it lost B. This is known as proxy ARP.

Gratuitous ARP - Gratuitous ARP is an ARP request of the host that helps to identify the duplicate IP address. It cast request for the IP address of the router. If an ARP request is sent by a switch or router to get its IP address and esponses are received, so all other nodes cannot use the IP address allocated to that switch or router. Yet if a router sends an ARP request for its IP address and receives an ARP response, another node uses the IP address allocated ch or router.

There are some primary use cases of gratuitous ARP that are given below:

- The gratuitous ARP is used to update the ARP table of other devices.
- It also checks whether the host is using the original IP address or a duplicate one.

Reverse ARP (RARP) - It is a networking protocol used by the client system in a local area network (LAN) to IPv4 address from the ARP gateway router table. A table is created by the network administrator in the gateway- is used to find out the MAC address to the corresponding IP address.

When a new system is set up or any machine that has no memory to store the IP address, then the user has to find ress of the device. The device sends a RARP broadcast packet, including its own MAC address in the address field e sender and the receiver hardware. A host installed inside of the local network called the RARP-server is prepared l to such type of broadcast packet. The RARP server is then trying to locate a mapping table entry in the IP to ress. If any entry matches the item in the table, then the RARP server sends the response packet along with the IP the requesting computer.

Inverse ARP (InARP) - Inverse ARP is inverse of the ARP, and it is used to find the IP addresses of the nodes lata link layer addresses. These are mainly used for the frame relays, and ATM networks, where Layer 2 virtual dressing are often acquired from Layer 2 signaling. When using these virtual circuits, the relevant Layer 3 are available.

ARP conversions Layer 3 addresses to Layer 2 addresses. However, its opposite address can be defined by InARP. P has a similar packet format as ARP, but operational codes are different.

Example - If Host A wants to transmit data to Host B, which is on the different network, then Host A sends an est message to receive a MAC address for Host B. The router responds to Host A with its own MAC address self as a destination. When the data is transmitted to the destination by Host A, it will send to the gateway so that it lost B. This is known as proxy ARP.

Gratuitous ARP - Gratuitous ARP is an ARP request of the host that helps to identify the duplicate IP address. It cast request for the IP address of the router. If an ARP request is sent by a switch or router to get its IP address and esponses are received, so all other nodes cannot use the IP address allocated to that switch or router. Yet if a router sends an ARP request for its IP address and receives an ARP response, another node uses the IP address allocated ch or router.

There are some primary use cases of gratuitous ARP that are given below:

- The gratuitous ARP is used to update the ARP table of other devices.
- It also checks whether the host is using the original IP address or a duplicate one.

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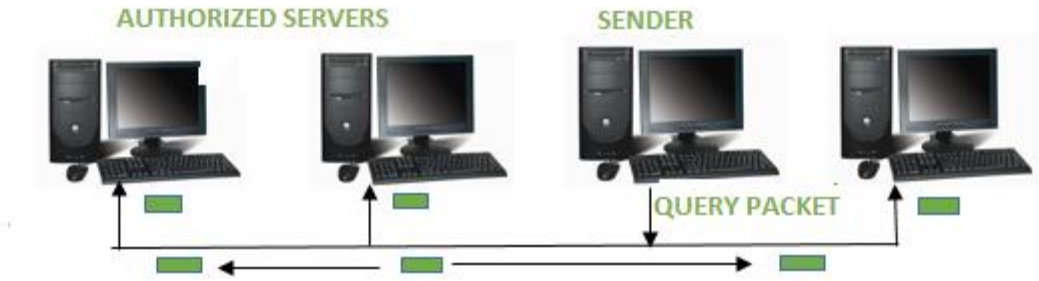
RARP

RARP is abbreviation of **Reverse Address Resolution Protocol** which is a protocol based on computer networking which is employed by a client computer to request its IP address from a gateway server's Address Resolution Protocol table or cache. The network administrator creates a table in gateway-router, which is used to map the MAC address to corresponding IP address.

This protocol is used to communicate data between two points in a server. The client doesn't necessarily need prior knowledge the server identities capable of serving its request. Media Access Control (MAC) addresses requires individual configuration on the servers done by an administrator. RARP limits to the serving of IP addresses only.

When a replacement machine is set up, the machine may or might not have an attached disk that may permanently store the IP Address so the RARP client program requests IP Address from the RARP server on the router. The RARP server will return the IP address to the machine under the belief that an entry has been setup within the router table.



REVERSE ADDRESS RESOLUTION PROTOCOL

History of RARP :

RARP was proposed in 1984 by the university Network group. This protocol provided the IP Address to the workstation. These diskless workstations were also the platform for the primary workstations from Sun Microsystems.

Working of RARP :

The RARP is on the Network Access Layer and is employed to send data between two points in a very network. Each network participant has two unique addresses:- IP address (a logical address) and MAC address (the physical address).

The IP address gets assigned by software and after that the MAC address is constructed into the hardware. The RARP server that responds to RARP requests, can even be any normal computer within the network. However, it must hold the data of all the MAC addresses with their assigned IP addresses. If a RARP request is received by the network, only these RARP servers can reply to it. The info packet needs to be sent on very cheap layers of the network. This implies that the packet is transferred to all the participants at the identical time.

The client broadcasts a RARP request with an Ethernet broadcast address and with its own physical address. The server responds by informing the client its IP address.

How is RARP different from ARP ?

RARP	ARP
RARP stands for Reverse Address Resolution Protocol	ARP stands for Address Resolution Protocol

In RARP, we find our own IP address	In ARP, we find the IP address of a remote machine
The MAC address is known and the IP address is requested	The IP address is known, and the MAC address is being requested
It uses the value 3 for requests and 4 for responses	It uses the value 1 for requests and 2 for responses

Uses of RARP :

RARP is used to convert the Ethernet address to an IP address. It is available for the LAN technologies like FDDI, token ring LANs, etc.

Disadvantages of RARP :

The Reverse Address Resolution Protocol had few disadvantages which eventually led to its replacement by BOOTP and DHCP. Some of the disadvantages are listed below:

- The RARP server must be located within the same physical network.
- The computer sends the RARP request on very cheap layer of the network. Thus, it's unattainable for a router to forward the packet because the computer sends the RARP request on very cheap layer of the network.
- The RARP cannot handle the subnetting process because no subnet masks are sent. If the network is split into multiple subnets, a RARP server must be available with each of them.
- It isn't possible to configure the PC in a very modern network.
- It doesn't fully utilize the potential of a network like Ethernet.

RARP has now become an obsolete protocol since it operates at low level. Due to this, it requires direct address to the network which makes it difficult to build a server.

UNIT- III ROUTING AND ROUTING PROTOCOLS			
Routing	**T1**	**1**	**19**
RIP-OSPF-IGRP- EIGRP-Metrics	**T1**	**2**	**21**
Switch Basics-Global Internet-Domains- BGP-IPv6	**T1**	**1**	**22**
Multicast Communication	**T1**	**1**	**23**
IP addresses – Address Classes	**T1**	**1**	**24**
Sub netting- Super netting- Examples	**T1**	**1**	**25**
Multicast Routing	**T1**	**1**	**26**
DVMRP-PIM	**T1**	**1**	**27**

Routing

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

Routing Metrics and Costs

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

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

The most common metric values are given below:

- **Hop count:** Hop count is defined as a metric that specifies the number of passes through internetworking devices such as a router, a packet must travel in a route to move from source to the destination. If the routing protocol considers the hop as a primary metric value, then the path with the least hop count will be considered as the best path to move from source to the destination.
- **Delay:** It is a time taken by the router to process, queue and transmit a datagram to an interface. The

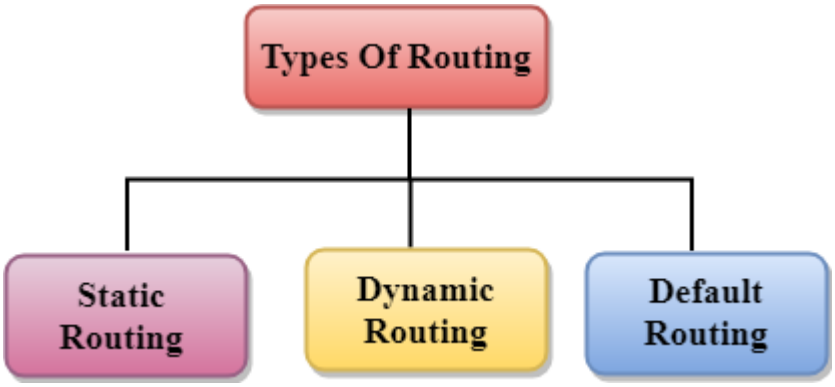
protocols use this metric to determine the delay values for all the links along the path end-to-end. The path having the lowest delay value will be considered as the best path.

- **Bandwidth:** The capacity of the link is known as a bandwidth of the link. The bandwidth is measured in terms of bits per second. The link that has a higher transfer rate like gigabit is preferred over the link that has the lower capacity like 56 kb. The protocol will determine the bandwidth capacity for all the links along the path, and the overall higher bandwidth will be considered as the best route.
- **Load:** Load refers to the degree to which the network resource such as a router or network link is busy. A Load can be calculated in a variety of ways such as CPU utilization, packets processed per second. If the traffic increases, then the load value will also be increased. The load value changes with respect to the change in the traffic.
- **Reliability:** Reliability is a metric factor may be composed of a fixed value. It depends on the network links, and its value is measured dynamically. Some networks go down more often than others. After network failure, some network links repaired more easily than other network links. Any reliability factor can be considered for the assignment of reliability ratings, which are generally numeric values assigned by the system administrator.

Types of Routing

Routing can be classified into three categories:

- Static Routing
- Default Routing
- Dynamic Routing



Static Routing

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

Advantages Of Static Routing

Following are the advantages of Static Routing:

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

Disadvantages of Static Routing:

Following are the disadvantages of Static Routing:

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

Default Routing

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

Dynamic Routing

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

The Dynamic protocol should have the following features:

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

Advantages of Dynamic Routing:

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

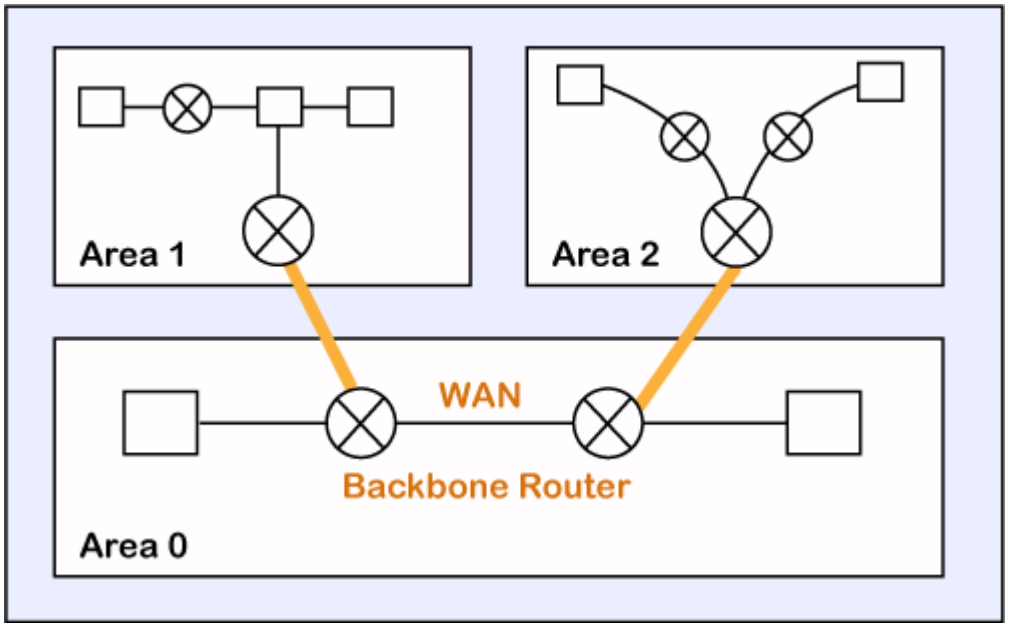
Disadvantages of Dynamic Routing:

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

OSPF PROTOCOL

The OSPF stands for **Open Shortest Path First**. It is a widely used and supported routing protocol. It is an intradomain protocol, which means that it is used within an area or a network. It is an interior gateway protocol that has been designed within a single autonomous system. It is based on a link-state routing algorithm in which each router contains the information of every domain, and based on this information, it determines the shortest path. The goal of routing is to learn routes. The OSPF achieves by learning about every router and subnet within the entire network. Every router contains the same information about the network. The way the router learns this information by sending LSA (Link State Advertisements). These LSAs contain information about every router, subnet, and other networking information. Once the LSAs have been flooded, the OSPF stores the information in a link-state database known as LSDB. The main goal is to have the same information about every router in an LSDBs.

OSPF Areas



OSPF divides the autonomous systems into areas where the area is a collection of networks, hosts, and routers.

Like internet service providers divide the internet into a different autonomous system for easy management and OSPF further divides the autonomous systems into Areas.

Routers that exist inside the area flood the area with routing information

In Area, the special router also exists. The special routers are those that are present at the border of an area, and these special routers are known as Area Border Routers. This router summarizes the information about an area and shares the information with other areas.

How does OSPF work?

There are three steps that can explain the working of OSPF:

Step 1: The first step is to become OSPF neighbors. The two connecting routers running OSPF on the same link creates a neighbor relationship.

Step 2: The second step is to exchange database information. After becoming the neighbors, the two routers exchange the LSDB information with each other.

Step 3: The third step is to choose the best route. Once the LSDB information has been exchanged with each other, the router chooses the best route to be added to a routing table based on the calculation of SPF.

How a router forms a neighbor relationship?

The first thing is happened before the relationship is formed is that each router chooses the router ID.

Router ID (RID): The router ID is a number that uniquely identifies each router on a network. The router ID is in the format of the IPv4 address. There are few ways to set the router ID, the first way is to set the router ID manually and the other way is to let the router decides itself.

- Manually assigned: The router checks whether the router ID is manually set or not. If it manually set, then it is a router ID. If it is not manually set, then it will choose the highest 'up' status loopback interface IP address. If there are no loopback interfaces, then it will choose the highest 'up' status non-loopback interface IP address.

Two routers connected to each other through point to point or multiple routers are connected can communicate with each other through an OSPF protocol. The two routers are adjacent only when both the routers send the HELLO packet to each other. When both the routers receive the acknowledgment of the HELLO packet, then they come in a two-way state. As OSPF is a link state routing protocol, so it allows to create the neighbor relationship between the routers. The two routers can be neighbors only when they belong to the same subnet, share the same area id, subnet mask, timers, and authentication. The OSPF relationship is a relationship formed between the routers so that they can know each other. The two routers can be neighbors if atleast one of them is designated router or backup designated router in a network, or connected through a point-to-point link.

Types of links in OSPF

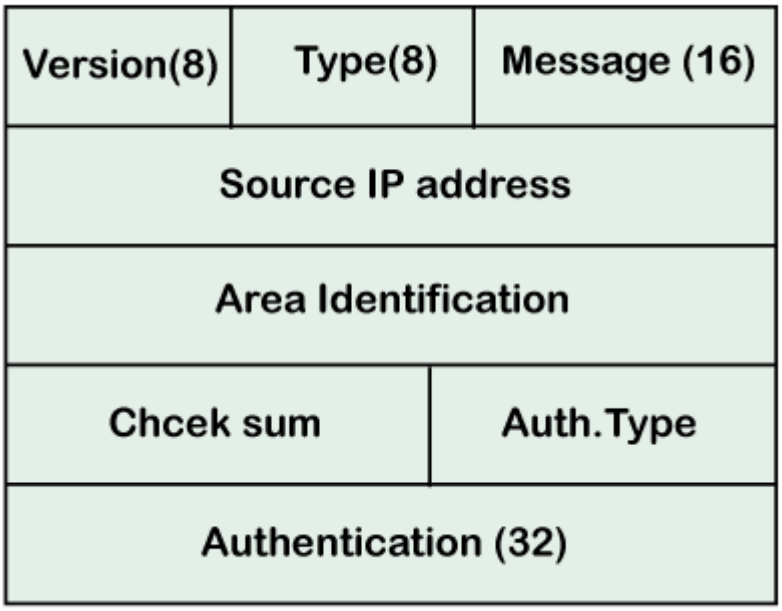
A link is basically a connection, so the connection between two routers is known as a link.

There are four types of links in OSPF:

1. **Point-to-point link:** The point-to-point link directly connects the two routers without any host or router in between.
2. **Transient link:** When several routers are attached in a network, they are known as a transient link. The transient link has two different implementations:
   Unrealistic topology: When all the routers are connected to each other, it is known as an unrealistic topology.
   Realistic topology: When some designated router exists in a network then it is known as a realistic topology. Here designated router is a router to which all the routers are connected. All the packets sent by the routers will be passed through the designated router.
3. **Stub link:** It is a network that is connected to the single router. Data enters to the network through the single router and leaves the network through the same router.
4. **Virtual link:** If the link between the two routers is broken, the administration creates the virtual path between the routers, and that path could be a long one also.

OSPF Message Format

The following are the fields in an OSPF message format:



- **Version:** It is an 8-bit field that specifies the OSPF protocol version.
- **Type:** It is an 8-bit field. It specifies the type of the OSPF packet.
- **Message:** It is a 16-bit field that defines the total length of the message, including the header. Therefore, the total length is equal to the sum of the length of the message and header.

- **Source IP address:** It defines the address from which the packets are sent. It is a sending routing IP address.
- **Area identification:** It defines the area within which the routing takes place.
- **Checksum:** It is used for error correction and error detection.
- **Authentication type:** There are two types of authentication, i.e., 0 and 1. Here, 0 means for none that specifies no authentication is available and 1 means for pwd that specifies the password-based authentication.
- **Authentication:** It is a 32-bit field that contains the actual value of the authentication data.

OSPF Packets

There are five different types of packets in OSPF:

- Hello
- Database Description
- Link state request
- Link state update
- Link state Acknowledgment

Let's discuss each packet in detail.

1. Hello packet

The Hello packet is used to create a neighborhood relationship and check the neighbor's reachability. Therefore, the Hello packet is used.

2. Database Description

After establishing a connection, if the neighbor router is communicating with the system first time, it sends the database information about the network topology to the system so that the system can update or modify accordingly.

3. Link state request

The link-state request is sent by the router to obtain the information of a specified route. Suppose there are two routers, i.e., router 1 and router 2, and router 1 wants to know the information about the router 2, so router 1 sends the link state request to the router 2. When router 2 receives the link state request, then it sends the link-state information to router 1.

4. Link state update

The link-state update is used by the router to advertise the state of its links. If any router wants to broadcast the state of its links, it uses the link-state update.

5. Link state acknowledgment

The link-state acknowledgment makes the routing more reliable by forcing each router to send the acknowledgment on each link state update. For example, router A sends the link state update to the router B and router C, then in return, the router B and C sends the link- state acknowledgment to the router A, so that the router A gets to know that both the routers have received the link-state update.

OSPF States

The device running the OSPF protocol undergoes the following states:

- **Down:** If the device is in a down state, it has not received the HELLO packet. Here, down does not mean that the device is physically down; it means that the OSPF process has not been started yet.
- **Init:** If the device comes in an init state, it means that the device has received the HELLO packet from the other router.
- **2WAY:** If the device is in a 2WAY state, which means that both the routers have received the HELLO packet from the other router, and the connection gets established between the routers.
- **Exstart:** Once the exchange between the routers get started, both the routers move to the Exstart state. In this state, master and slave are selected based on the router's id. The master controls the sequence of numbers, and starts the exchange process.
- **Exchange:** In the exchange state, both the routers send a list of LSAs to each other that contain a database description.
- **Loading:** On the loading state, the LSR, LSU, and LSA are exchanged.
- **Full:** Once the exchange of the LSAs is completed, the routers move to the full state.

IGRP Routing Protocol

In a host network, the Interior Gateway Routing Protocol (IGRP) is a proprietary distance vector routing protocol that is used to exchange routing information. Cisco was the one who came up with the idea.

The Internet Geolocation Routing Protocol (IGRP) regulates the transfer of routing information among linked routers in the host network or autonomous system. The protocol guarantees that every router’s routing table is kept up to date with the most direct route available. IGRP also helps to minimize routing loops by updating itself in response to changes that occur on the network and by implementing error management.

Characteristics

The following are the characteristics of the IGRP (Interior Gateway Routing Protocol):

1. The Internet Group Routing Technology (IGRP) is a distance-vector routing protocol created by Cisco.
2. In addition to bandwidth, delay (by default), reliability, load, and MTU are all measured in the IGRP

protocol.

3. It transmits updates every 90 seconds, with a hold-down time of 280 seconds between each broadcasting session.
4. When network changes occur, triggered updates are utilized to expedite the convergence process.
5. The IGRP router command needs the inclusion of an AS number.
6. For routers to communicate routing information, they must be in the same Associated System Number (AS).
7. The maximum number of hops allowed by IGRP is 255. It has a default value of 100 and is often changed to 50 or less.
8. The IGRP AD value is 100.

Goals of IGRP

The International Geophysical Research Program (IGRP) has two primary objectives:

1. Its primary function is to provide routing information to all linked routers within its border or inside its autonomous system
2. It will automatically update whenever the network topology changes.

Every 90 seconds, it sends out a notice to its neighbors to inform them of any new modifications.

IGRP Timers

IGRP Timers are a kind of timer that is used in the Internet of Things (IoT).

Every 90 seconds, the Internet Geolocation Routing Protocol (IGRP) delivers its routing table to its neighbors. When compared to RIP, which might use excessive bandwidth when sending updates every 30 seconds, IGRP's default update duration of 90 seconds is a plus. RIP's default update period of 90 seconds is a disadvantage. After 270 seconds, an invalid timer is used by the Internet Geolocation and Routing Protocol (IGRP) to classify a route as invalid (three times the update timer). When a route is removed from the routing database, IGRP utilizes a flush timer, similar to the way RIP does. The default flush duration is set to 630 seconds (seven times the update period and more than 10 minutes).

In the event that a network goes down or the metric for the network rises, the route is put in hold down mode. Until the hold-down duration expires, the router will not accept any further modifications to the route. This configuration prevents routing loops from forming in the network. The hold-down timer is set at 280 seconds by default (three times the update timer plus 10 seconds).

Functions

The IGRP performs a variety of functions:

Interior Gateway Routing Protocol (IGRP) was developed by Cisco in response to the restrictions of the Routing Information Protocol (RIP), which manages a maximum hop count of 15 per connection. The Internet Geolocation Routing Protocol (IGRP) allows for a maximum hop count of 255. The fundamental two objectives of the IGRP are as follows:

1. Route information should be sent between all linked routers inside its border or autonomous system.
2. Continue to update anytime there is a topological, network, or route change that takes place.

Every 90 seconds, the IGRP broadcasts to its neighbors a notice of any new modifications as well as information about its current condition.

IGRP is responsible for maintaining a routing table with the most optimum route to the corresponding nodes and networks inside the parent network, as determined by the parent network. Given that it is a distance-vector protocol, the IGRP calculates the metric for the shortest route to a certain destination based on a number of different criteria.

Advantages:

1. The procedure is simple and uncomplicated.
2. It takes into account the latency, bandwidth, reliability, and load of a network connection while calculating the score. Consequently, it is quite accurate when it comes to selecting the most suited approach.
3. The use of composite metrics
4. Configuration is straightforward.
5. When compared to RIP, it has more scalability (255 hops, 100 by default)

Disadvantages:

1. The hop count is limited to 15; if a packet has traveled through 15 routers and still has another router to travel to, it will be discarded.
2. Does not support a variable-length subnet mask (VLSM), which means that it sends routing updates based only on a fixed-length subnet mask (FLSM) or routes that fall on classful boundaries. As a result, RIP V1 will not function on a network that has been subnetted beyond the standard /8, /16, and /24 (255.0.0.0, 255.255.255.0) or Class A, B, and C network borders (255.0.0.0, 255.255.255.0).
3. Convergence occurs slowly, particularly on large networks.
4. Doesn't know how much bandwidth is available on a given connection.
5. Doesn't allow numerous pathways for the same route
6. Routing updates may use a substantial amount of bandwidth since the whole routing database is delivered whenever the state of a connection changes. Routers are prone to routing loops.

Border Gateway Protocol (BGP) is used to Exchange routing information for the internet and is the protocol used between ISP which are different ASes.
The protocol can connect together any internetwork of autonomous system using an arbitrary topology. The only requirement is that each AS have at least one router that is able to run BGP and that is router connect to at least one other AS's BGP router. BGP's main function is to exchange network reach-ability information with other BGP systems. Border Gateway Protocol constructs an autonomous systems' graph based on the information exchanged between BGP routers.

Characteristics of Border Gateway Protocol (BGP):

- **Inter-Autonomous System Configuration:** The main role of BGP is to provide communication between two autonomous systems.
- BGP supports Next-Hop Paradigm.
- Coordination among multiple BGP speakers within the AS (Autonomous System).
- **Path Information:** BGP advertisement also include path information, along with the reachable destination and next destination pair.
- **Policy Support:** BGP can implement policies that can be configured by the administrator. For ex:- a router running BGP can be configured to distinguish between the routes that are known within the AS and that which are known from outside the AS.
- Runs Over TCP.
- BGP conserve network Bandwidth.
- BGP supports CIDR.
- BGP also supports Security.

Functionality of Border Gateway Protocol (BGP):
BGP peers performs 3 functions, which are given below.

1. The first function consist of initial peer acquisition and authentication. both the peers established a TCP connection and perform message exchange that guarantees both sides have agreed to communicate.
2. The second function mainly focus on sending negative or positive reach-ability information.
3. The third function verifies that the peers and the network connection between them are functioning correctly.

BGP Route Information Management Functions:

- **Route Storage:** Each BGP stores information about how to reach other networks.
- **Route Update:** In this task, Special techniques are used to determine when and how to use the information received from peers to properly update the routes.
- **Route Selection:** Each BGP uses the information in its route databases to select good routes to each network on the internet network.
- **Route advertisement:** Each BGP speaker regularly tells its peer what is knows about various networks and methods to reach them.

IPV6

Internet Protocol Version 6 is a network layer protocol that allows communication to take place over the network. IPv6 was designed by Internet Engineering Task Force (IETF) in December 1998 with the purpose of superseding the IPv4 due to the global exponentially growing internet users.

IPv4 vs IPv6

The common type of IP address (is known as IPv4, for "version 4"). Here's an example of what an IP address might look like:

25.59.209.224

An IPv4 address consists of four numbers, each of which contains one to three digits, with a single dot (.) separating each number or set of digits. Each of the four numbers can range from 0 to 255. This group of separated numbers creates the addresses that let you and everyone around the globe to send and retrieve data over our Internet connections. The IPv4 uses a 32-bit address scheme allowing to store 2^32 addresses which is more than 4 billion addresses. To date, it is considered the primary Internet Protocol and carries 94% of Internet traffic. Initially, it was assumed it would never run out of addresses but the present situation paves a new way to IPv6, let's see why? An IPv6 address consists of eight groups of four hexadecimal digits. Here's an example IPv6 address:

3001:0da8:75a3:0000:0000:8a2e:0370:7334

This new IP address version is being deployed to fulfil the need for more Internet addresses. It was aimed to resolve issues which are associated with IPv4. With 128-bit address space, it allows 340 undecillion unique address space. IPv6 also called IPng (Internet Protocol next generation).

IPv6 support a theoretical maximum of 340, 282, 366, 920, 938, 463, 463, 374, 607, 431, 768, 211, 456. To keep it straightforward, we will never run out of IP addresses again.

Types of IPv6 Address

Now that we know about what is IPv6 address let's take a look at its different types.

- **Unicast addresses** It identifies a unique node on a network and usually refers to a single sender or a single receiver.
- **Multicast addresses** It represents a group of IP devices and can only be used as the destination of a datagram.
- **Anycast addresses** It is assigned to a set of interfaces that typically belong to different nodes.

Advantages of IPv6

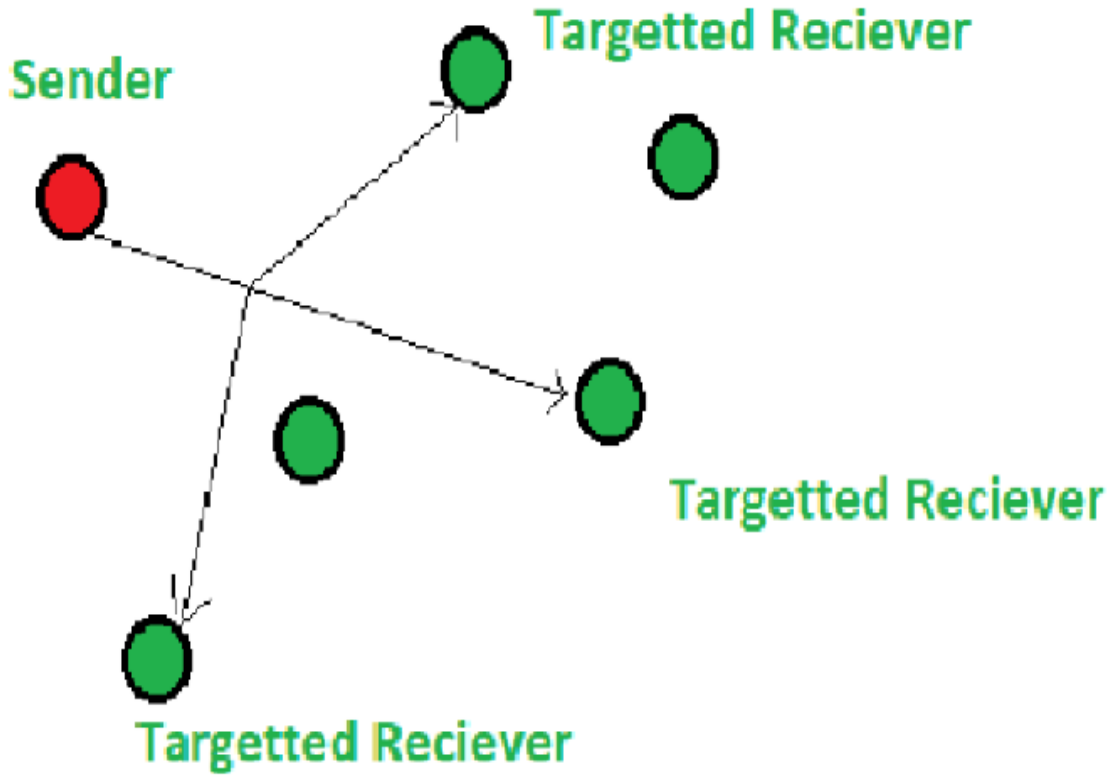
- Reliability
- **Faster Speeds:** IPv6 supports multicast rather than broadcast in IPv4.This feature allows bandwidth-intensive packet flows (like multimedia streams) to be sent to multiple destinations all at once.
- **Stronger Security:** IPSecurity, which provides confidentiality, and data integrity, is embedded into IPv6.
- Routing efficiency
- Most importantly it's the final solution for growing nodes in Global-network.

Disadvantages of IPv6

- **Conversion:** Due to widespread present usage of IPv4 it will take a long period to completely shift to IPv6.
- **Communication:** IPv4 and IPv6 machines cannot communicate directly with each other. They need an intermediate technology to make that possible.

Multicast is a method of group communication where the sender sends data to multiple receivers or nodes present in the network simultaneously. Multicasting is a type of one-to-many and many-to-many communication as it allows sender or senders to send data packets to multiple receivers at once across LANs or WANs. This process helps in minimizing the data frame of the network.

Multicasting works in similar to Broadcasting, but in Multicasting, the information is sent to the targeted or specific members of the network. This task can be accomplished by transmitting individual copies to each user or node present in the network, but sending individual copies to each user is inefficient and might increase the network latency. To overcome these shortcomings, multicasting allows a single transmission that can be split up among the multiple users, consequently, this reduces the bandwidth of the signal.



Applications :

Multicasting is used in many areas like:

-

1. Internet protocol (IP)
2. Streaming Media

It also supports video conferencing applications and webcasts.

IP Multicast :
Multicasting that takes place over the Internet is known as IP Multicasting. These multicast follow the internet protocol(IP) to transmit data. IP multicasting uses a mechanism known as 'Multicast trees' to transmit to information among the users of the network. Multicast trees; allows a single transmission to branch out to the desired receivers. The branches are created at the Internet routers, the branches are created such that the length of the transmission will be minimum.
IP multicasts also use two other essential protocols to function; Internet Group Management Protocol (IGMP), Protocol Independent Multicast (PIM). IGMP allows the recipients to access the data or information. The network routers use PIM to create multicast trees.

To sum up, Multicasting is an efficient way of communication; it reduces the bandwidth usage.

Multicast Routing: When a router receives a multicast packet, the situation is different fromwhen it receives a unicast packet. A multicast packet may have destinations in more than one network. Forwarding of a single packet to members of a group requires a shortest path tree. If we have n groups, we may need *n* shortest path trees. We can imagine the complexity of multicast routing. Two approaches have been used to solve the problem: source-based trees and group-shared trees.

a. Source-Based Tree: In the source-based tree approach, each router needs to have one shortestpath tree for each group. The shortest path tree for a group defines the next hop for each network that has loyal member(s) for that group. Five groups in the domain: GI, G2, G3, G4, and G5.

At the moment G1 has loyal members in four networks, G2 in three, G3 in two, G4 in two, and G5 in two. We have shown the names of the groups with loyal members on each network. There is one shortest path tree for each group; therefore there are five shortest path trees for five groups

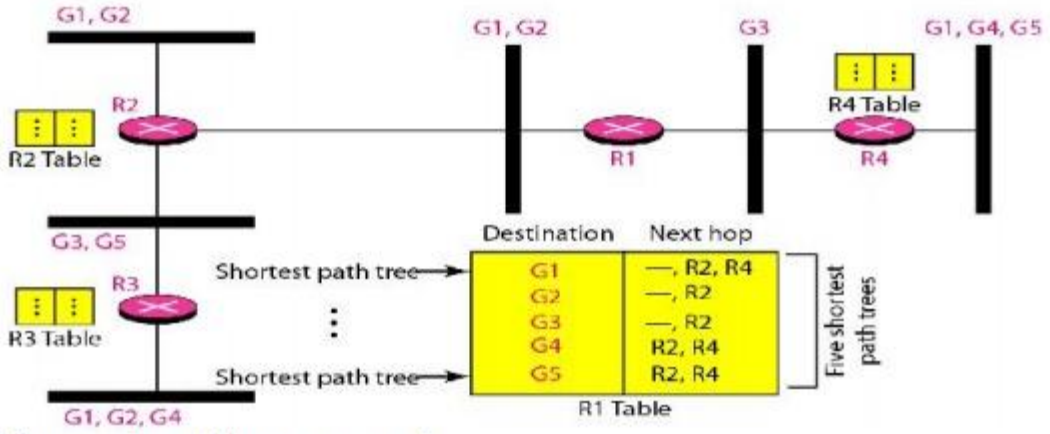


Figure 3.58 Source-based tree approach

b. Group-Shared Tree:. In the group-shared tree approach, instead of each router having*m*shortest path trees, only one designated router, called the center core, or rendezvous router, takes the responsibility of distributing multicast traffic. The core has *m* shortest path trees in its routing table. The rest of the routers in the domain have none. If a router receives a multicast packet, it encapsulates the packet in a unicast packet and sends it to the core router. The core router removes the multicast packet from its capsule, and consults its routing table to route the packet.

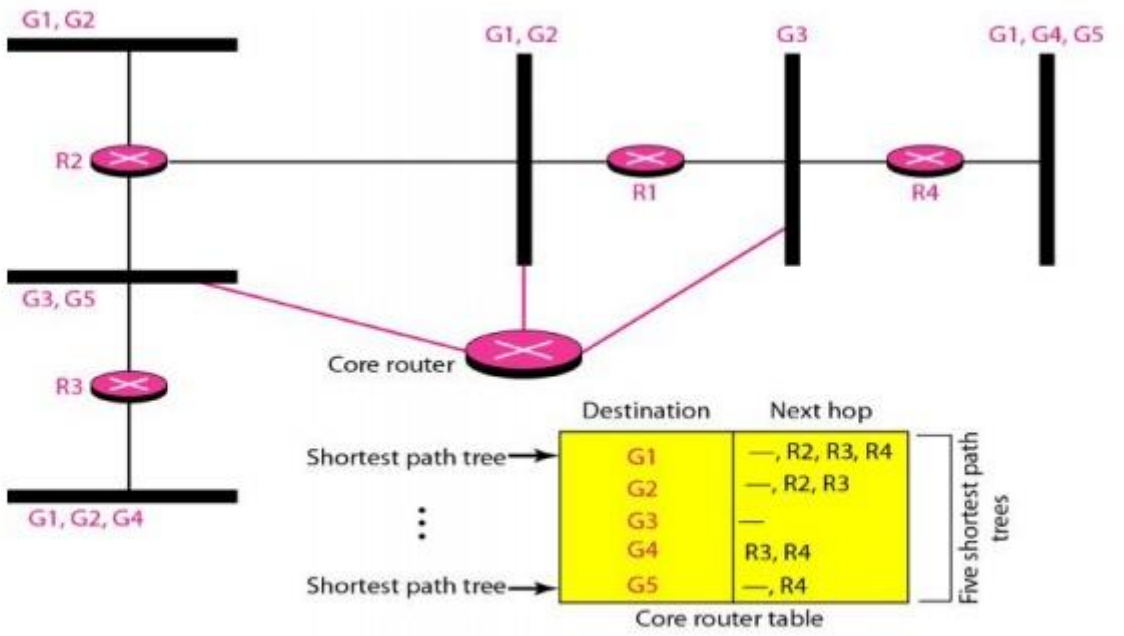


Figure 3.59 Group-shared tree approach

8. Routing Protocols

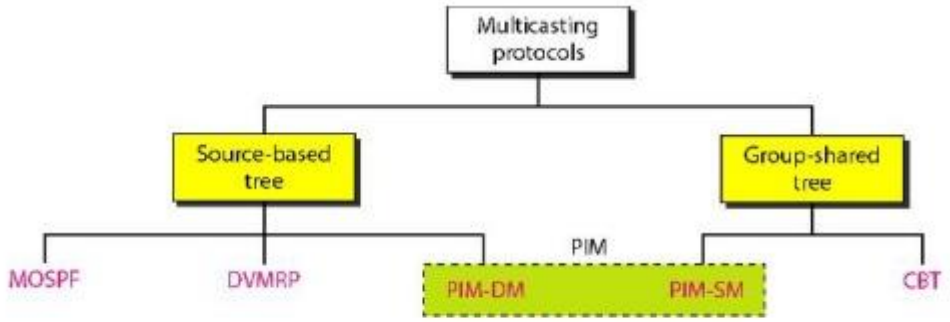


Figure 3.60 Taxonomy of common multicast protocols

a. Multicast Link State Routing: MOSPF

Multicast link state routing uses the source-based tree approach. links. For multicast routing, a node needs to revise the interpretation of *state.* A node advertises every group which has any loyal member on the link. Here the meaning of state is "what groups are active on this link." The information about the group comes from IGMP. Each router running IGMP solicits the hosts on the link to find out the membership status.

MOSPF Multicast Open Shortest Path First (MOSPF) protocol is an extension of theOSPF protocol that uses multicast link state routing to create source-based trees. The protocol requires a new link state update packet to associate the unicast address of a host with the group address or addresses the host is sponsoring. This packet is called the group-membership LSA.

b. Multicast Distance Vector: DVMRP

Multicast Distance Vector Routing Unicast distance vector routing is very simple; extending itto support multicast routing is complicated. Multicast routing does not allow a router to send its routing table to its neighbors. The idea is to create a table from scratch by using the information from the unicast distance vector tables.

Multicast distance vector routing uses source-based trees, but the router never actually makes a routing table. When a router receives a multicast packet, it forwards the packet as though it is consulting a routing table.

a) **Flooding**

b) Reverse Path Forwarding (RPF)

c) Reverse Path Broadcasting (RPB)

d) Reverse Path Multicasting (RPM)

DVMRP The Distance Vector Multicast Routing Protocol (DVMRP) is animplementation of multicast distance vector routing. It is a source-based routing protocol, based on RIP.

c. CBT

The Core-Based Tree (CBT) protocol is a group-shared protocol that uses a core as the root of the tree. The autonomous system is divided into regions, and a core (center router or rendezvous router) is chosen for each region.

The Core-Based Tree (CBT) is a group-shared tree, center-based protocol using one tree per group. One of the routers in the tree is called the core. A packet is sent from the source to members of the group following this procedure:

The source, which may or may not be part of the tree, encapsulates the multicast packet inside a unicast packet with the unicast destination address of the core and sends it to the core. This part of delivery is done using a unicast address; the only recipient is the core router.

b) The core decapsulates the unicast packet and forwards it to all interested intetfaces.

c) Each router that receives the multicast packet, in tum, forwards it to all interested interfaces.

PIM

Protocol Independent Multicast (PIM) is the name given to two independent multicastrouting protocols: Protocol Independent Multicast, Dense Mode (PIM-DM) and Protocol Independent Multicast, Sparse Mode (PIM-SM). Both protocols are unicast protocol-dependent, but the similarity ends here.

PIM-DM

PIM-DM is used when there is a possibility that each router is involved in multicasting (dense mode). In this environment, the use of a protocol that broadcasts the packet is justified because almost all routers are involved in the process. PIM-DM is a source-based tree routing protocol that uses RPF and pruning and grafting strategies for multicasting. Its operation is like that of DVMRP.

PIM-SM

PIM-SM is used when there is a slight possibility that each router is involved in multicasting (sparse mode). In this environment, the use of a protocol that broadcasts the packet is not justified; a protocol such as CBT that uses a group-shared tree is more appropriate. PIM-SM is used in a sparse multicast environment such as a WAN. PIM-SM is a group-shared tree routing protocol that has a rendezvous point (RP) as the source of the tree.

UNIT – IV TRANSPORT LAYER

Overview of Transport layer – TCP and UDP

Reliable byte Stream (TCP)

Connection Management- Flow control – Retransmission

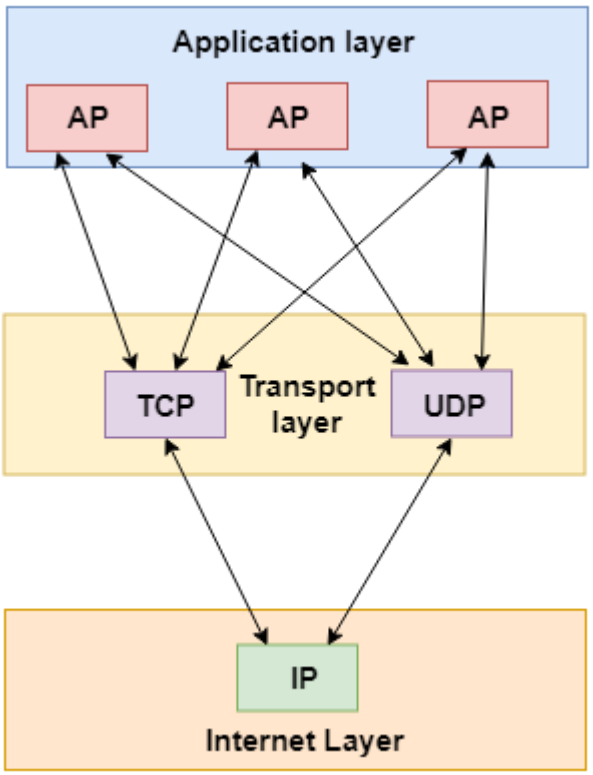
TCP Congestion control

Congestion avoidance (DEC bit, RED)

QoS – Application requirements

Transport Layer

- The transport layer is a 4th layer from the top.
- The main role of the transport layer is to provide the communication services directly to the application processes running on different hosts.
- The transport layer provides a logical communication between application processes running on different hosts. Although the application processes on different hosts are not physically connected, application processes use the logical communication provided by the transport layer to send the messages to each other.
- The transport layer protocols are implemented in the end systems but not in the network routers.
- A computer network provides more than one protocol to the network applications. For example, TCP and UDP are two transport layer protocols that provide a different set of services to the network layer.
- All transport layer protocols provide multiplexing/demultiplexing service. It also provides other services such as reliable data transfer, bandwidth guarantees, and delay guarantees.
- Each of the applications in the application layer has the ability to send a message by using TCP or UDP. The application communicates by using either of these two protocols. Both TCP and UDP will then communicate with the internet protocol in the internet layer. The applications can read and write to the transport layer. Therefore, we can say that communication is a two-way process.

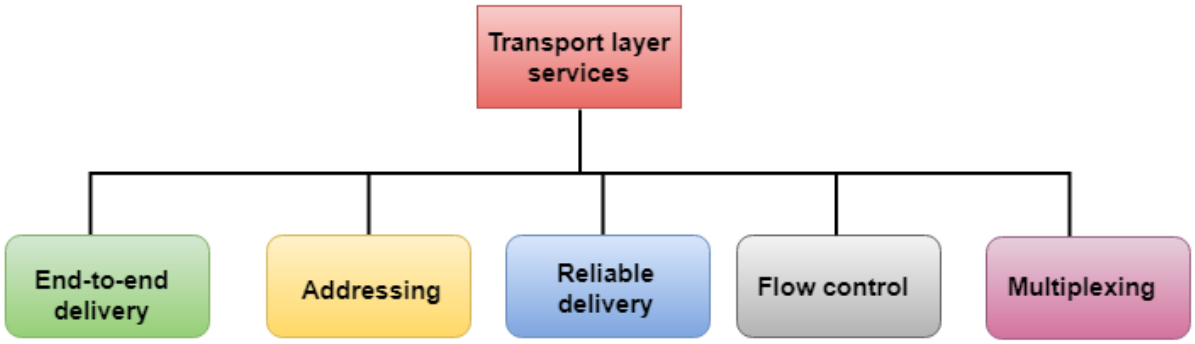


Services provided by the Transport Layer

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

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

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



End-to-end delivery:

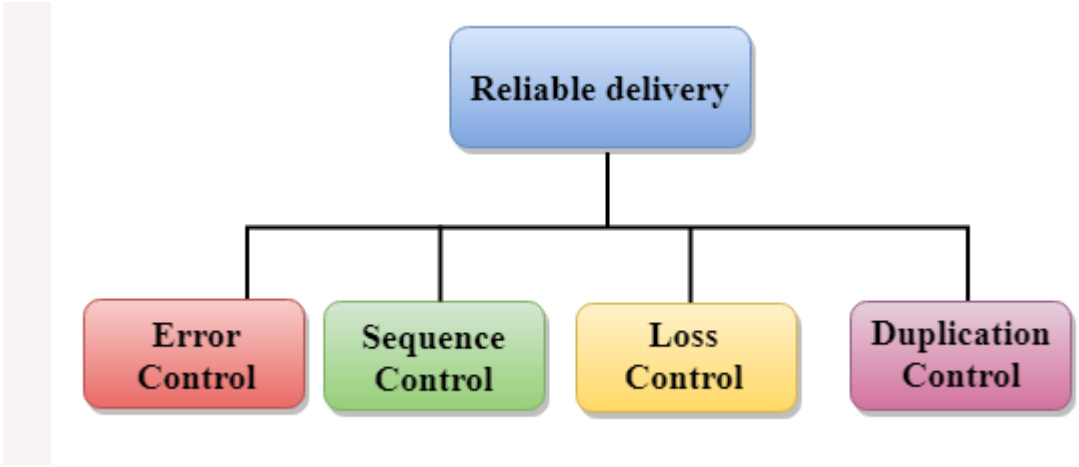
The transport layer transmits the entire message to the destination. Therefore, it ensures the end-to-end delivery of an entire message from a source to the destination.

Reliable delivery:

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

The reliable delivery has four aspects:

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



Error Control

- The primary role of reliability is **Error Control**. In reality, no transmission will be 100 percent error-free

delivery. Therefore, transport layer protocols are designed to provide error-free transmission.

- The data link layer also provides the error handling mechanism, but it ensures only node-to-node error-free delivery. However, node-to-node reliability does not ensure the end-to-end reliability.
- The data link layer checks for the error between each network. If an error is introduced inside one of the routers, then this error will not be caught by the data link layer. It only detects those errors that have been introduced between the beginning and end of the link. Therefore, the transport layer performs the checking for the errors end-to-end to ensure that the packet has arrived correctly.

Sequence Control

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

Loss Control

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

Duplication Control

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

Flow Control

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

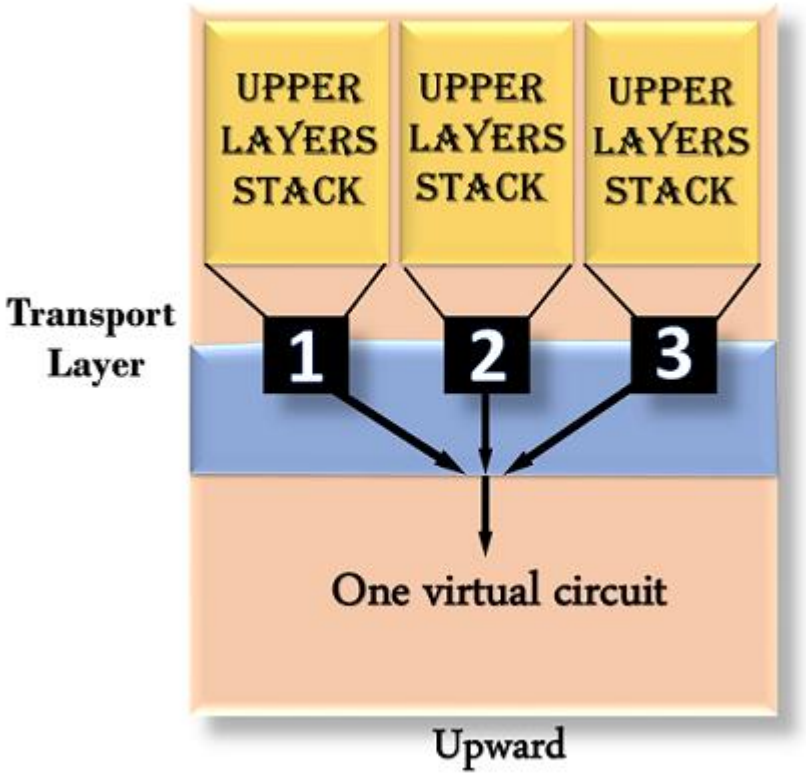
Multiplexing

The transport layer uses the multiplexing to improve transmission efficiency.

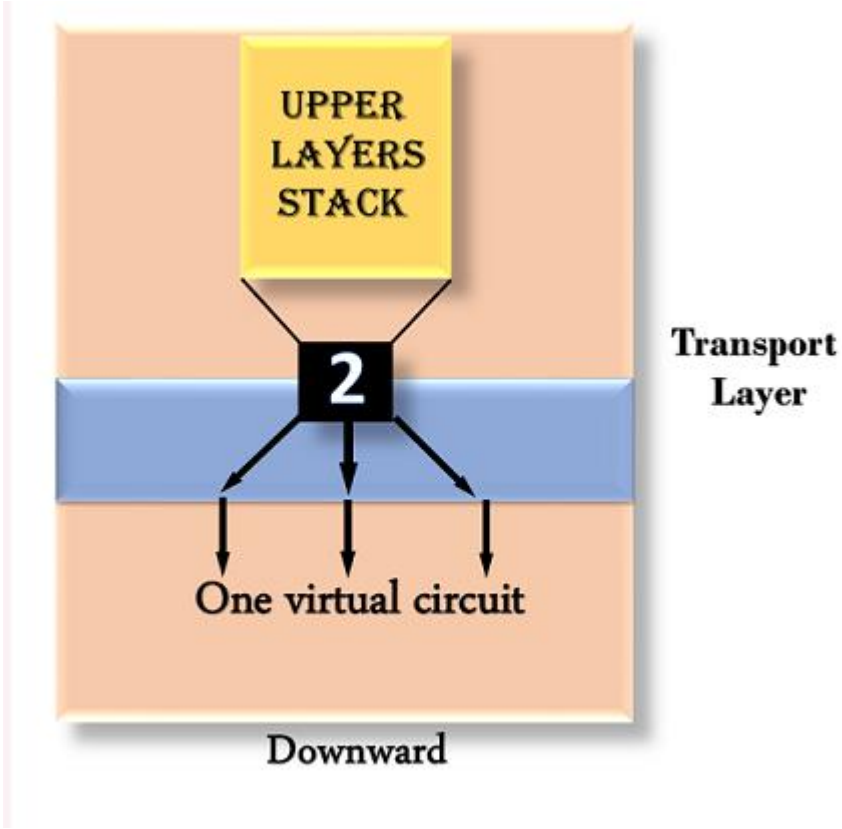
Multiplexing can occur in two ways:

- **Upward multiplexing:** Upward multiplexing means multiple transport layer connections use the same

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

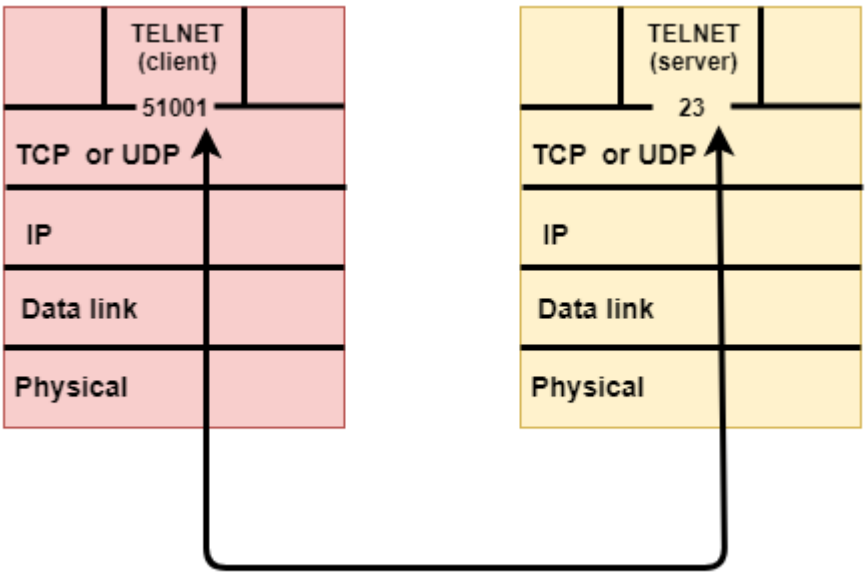


- **Downward multiplexing:** Downward multiplexing means one transport layer connection uses the multiple network connections. Downward multiplexing allows the transport layer to split a connection among several paths to improve the throughput. This type of multiplexing is used when networks have a low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



Transport Layer 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.

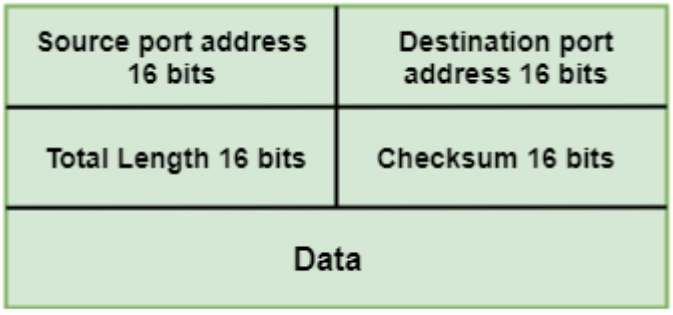


UDP

- 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.

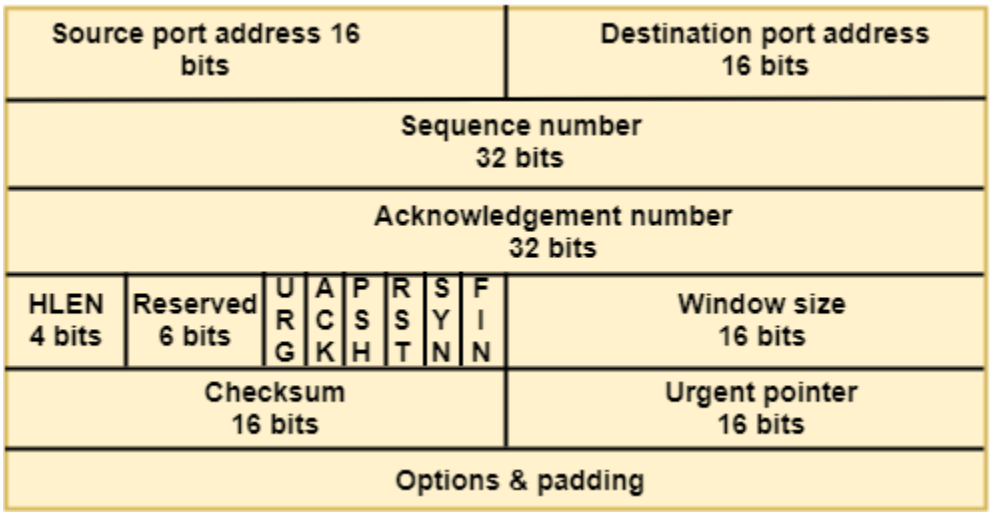
Features Of TCP protocol

- **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.

TCP Segment Format



Where,

- **Source port address:** It is used to define the address of the application program in a source computer. It is a 16-bit field.
- **Destination port address:** It is used to define the address of the application program in a destination computer. It is a 16-bit field.
- **Sequence number:** A stream of data is divided into two or more TCP segments. The 32-bit sequence number field represents the position of the data in an original data stream.

- **Acknowledgement number:** A 32-field acknowledgement number acknowledge the data from other communicating devices. If ACK field is set to 1, then it specifies the sequence number that the receiver is expecting to receive.
- **Header Length (HLEN):** It specifies the size of the TCP header in 32-bit words. The minimum size of the header is 5 words, and the maximum size of the header is 15 words. Therefore, the maximum size of the TCP header is 60 bytes, and the minimum size of the TCP header is 20 bytes.
- **Reserved:** It is a six-bit field which is reserved for future use.
- **Control bits:** Each bit of a control field functions individually and independently. A control bit defines the use of a segment or serves as a validity check for other fields.

There are total six types of flags in control field:

- **URG:** The URG field indicates that the data in a segment is urgent.
- **ACK:** When ACK field is set, then it validates the acknowledgement number.
- **PSH:** The PSH field is used to inform the sender that higher throughput is needed so if possible, data must be pushed with higher throughput.
- **RST:** The reset bit is used to reset the TCP connection when there is any confusion occurs in the sequence numbers.
- **SYN:** The SYN field is used to synchronize the sequence numbers in three types of segments: connection request, connection confirmation (with the ACK bit set), and confirmation acknowledgement.
- **FIN:** The FIN field is used to inform the receiving TCP module that the sender has finished sending data. It is used in connection termination in three types of segments: termination request, termination confirmation, and acknowledgement of termination confirmation.
    - **Window Size:** The window is a 16-bit field that defines the size of the window.
    - **Checksum:** The checksum is a 16-bit field used in error detection.
    - **Urgent pointer:** If URG flag is set to 1, then this 16-bit field is an offset from the sequence number indicating that it is a last urgent data byte.
    - **Options and padding:** It defines the optional fields that convey the additional information to the receiver.

Differences b/w TCP & UDP

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

TCP CONGESTION CONTROL

TCP uses a congestion window and a congestion policy that avoid congestion. Previously, we assumed that only the receiver can dictate the sender's window size. We ignored another entity here, the network. If the network cannot deliver the data as fast as it is created by the sender, it must tell the sender to slow down. In other words, in addition to the receiver, the network is a second entity that determines the size of the sender's window.

Congestion policy in TCP –

1. Slow Start Phase: starts slowly increment is exponential to threshold
2. Congestion Avoidance Phase: After reaching the threshold increment is by 1
3. Congestion Detection Phase: Sender goes back to Slow start phase or Congestion avoidance phase.

Slow Start Phase : exponential increment – In this phase after every RTT the congestion window size increments exponentially.
Initially cwnd = 1

After 1 RTT, cwnd = 2^(1) = 2

2 RTT, cwnd = 2^(2) = 4

3 RTT, cwnd = 2^(3) = 8

Congestion Avoidance Phase : additive increment – This phase starts after the threshold value also denoted as *ssthresh*. The size of *cwnd*(congestion window) increases additive. After each RTT cwnd = cwnd + 1.
Initially cwnd = i

After 1 RTT, cwnd = i+1

2 RTT, cwnd = i+2

3 RTT, cwnd = i+3

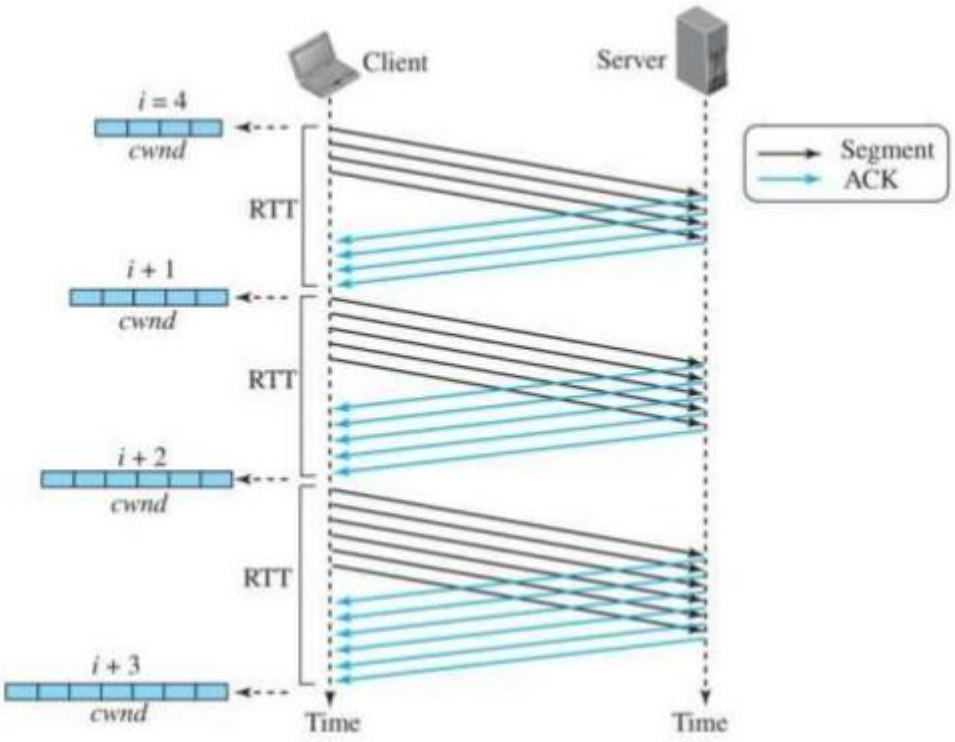
Congestion Detection Phase : multiplicative decrement – If congestion occurs, the congestion window size is decreased. The only way a sender can guess that congestion has occurred is the need to retransmit a segment. Retransmission is needed to recover a missing packet that is assumed to have been dropped by a router due to congestion. Retransmission can occur in one of two cases: when the RTO timer times out or when three duplicate ACKs are received.

- **Case 1 : Retransmission due to Timeout –** In this case congestion possibility is high.
  (a) ssthresh is reduced to half of the current window size.
  (b) set cwnd = 1
  (c) start with slow start phase again.
- **Case 2 : Retransmission due to 3 Acknowledgement Duplicates –** In this case congestion possibility is less.
  (a) ssthresh value reduces to half of the current window size.
  (b) set cwnd= ssthresh
  (c) start with congestion avoidance phase

Example – Assume a TCP protocol experiencing the behavior of slow start. At 5th transmission round with a threshold (ssthresh) value of 32 goes into congestion avoidance phase and continues till 10th transmission. At 10th transmission round, 3 duplicate ACKs are received by the receiver and enter into additive increase mode. Timeout occurs at 16th transmission round. Plot the transmission round (time) vs congestion window size of TCP segments.

Congestion Avoidance

Congestion Avoidance: Additive Increase : To avoid congestion before it happens, we must slow down this exponential growth. TCP defines another algorithm called congestion avoidance, which increases the cwnd additively instead of exponentially. When the size of the congestion window reaches the slow-start threshold in the case where cwnd = i, the slow-start phase stops and the additive phase begins. In this algorithm, each time the whole "window" of segments is acknowledged, the size of the congestion window is increased by one. A window is the number of segments transmitted during RTT



The sender starts with cwnd = 4. This means that the sender can send only four segments. After four ACKs arrive, the acknowledged segments are purged from the window, which means there is now one extra empty segment slot in the window. The size of the congestion window is also increased by 1. The size of window is now 5. After sending five segments and receiving five acknowledgments for them, the size of the congestion window now becomes 6, and so on. In other words, the size of the congestion window in this algorithm is also a function of the number of ACKs that have arrived and can be determined as follows: If an ACK arrives, cwnd 5 cwnd 1 (1/cwnd). The size of the window increases only 1/cwnd portion of MSS (in bytes). In other words, all segments in the previous window should be acknowledged to increase the window 1 MSS bytes. If we look at the size of the cwnd in terms of round-trip times (RTTs), we find that the growth rate is linear in terms of each round-trip time, which is much more conservative than the slow-start approach.

Start	→	$cwnd = i$
After 1 RTT	→	$cwnd = i + 1$
After 2 RTT	→	$cwnd = i + 2$
After 3 RTT	→	$cwnd = i + 3$

In the congestion-avoidance algorithm, the size of the congestion window increases additively until congestion is detected.

Congestion Detection: Multiplicative Decrease: If congestion occurs, the congestion window size must be decreased. The only way the sender can guess that congestion has

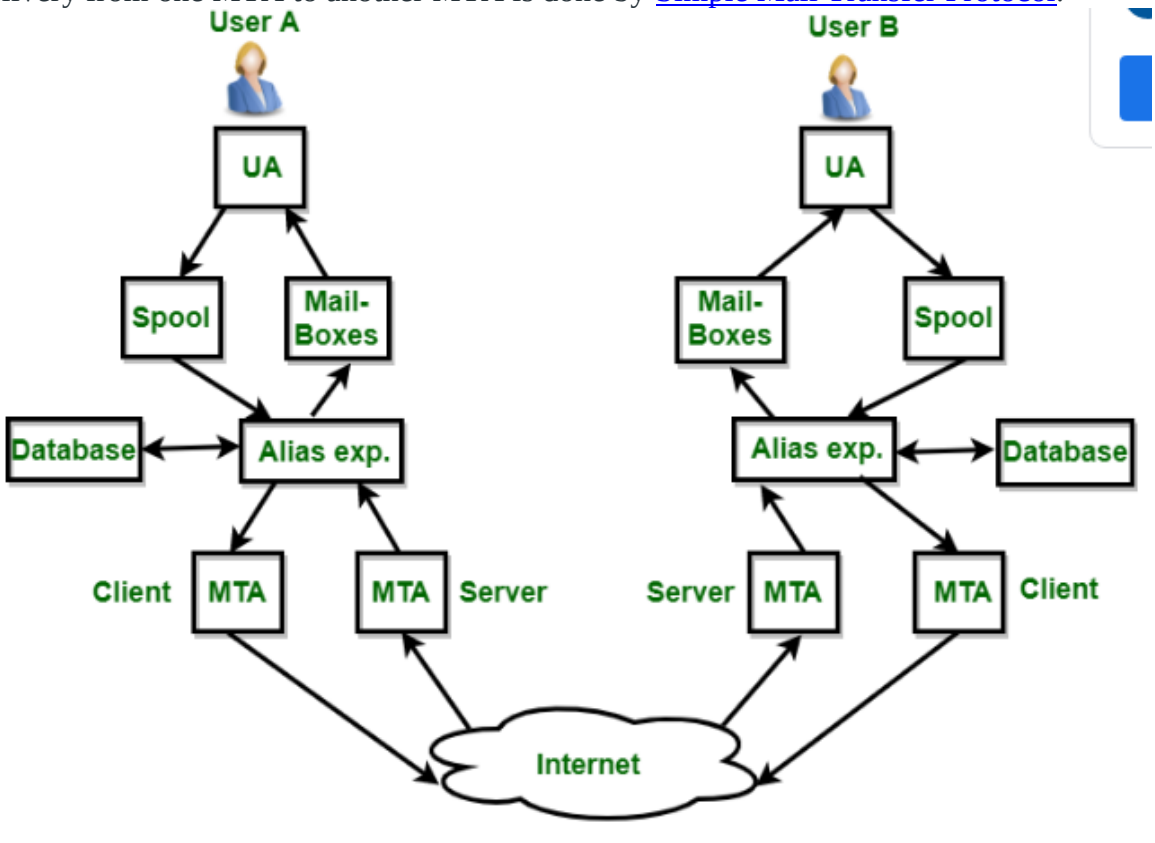
occurred is by the need to retransmit a segment. However, retransmission can occur in one of two cases: when a timer times out or when three Duplicate ACKs are received. In both cases, the size of the threshold is dropped to one-half, a multiplicative decrease. TCP implementations have two reactions : 1. If a time-out occurs, there is a stronger possibility of congestion; a segment has probably been dropped in the network, and there is no news about the sent segments. In this case TCP reacts strongly: a. It sets the value of the threshold to one-half of the current window size.

b. It sets cwnd to the size of one segment. c. It starts the slow-start phase again. 2. If three ACKs are received, there is a weaker possibility of congestion; a segment may have been dropped, but some segments after that may have arrived safely since three ACKs are received. This is called fast transmission and fast recovery. In this case, TCP has a weaker reaction: a. It sets the value of the threshold to one-half of the current window size. b. It sets cwnd to the value of the threshold. c. It starts the congestion avoidance phase. An implementations reacts to congestion detection in one of the following ways : If detection is by time-out, a new slow-start phase starts. If detection is by three ACKs, a new congestion avoidance phase starts.

UNIT – V APPLICATION LAYER AND CASE STUDIES	
LEARNING AND TEACHING AID Chalk & Board, Presentation Slides, Videos	
28.	Electronic Mail- SMTP, POP3, IMAP, MIME
29.	HTTP and HTTPS – Web Services
30.	DNS – SNMP
31.	Wireshark Packet Capturing Tool -Cisco Packet Tracer Tool
32.	Open Source Network Tools-Manage Engine Tool.

Electronic Mail (e-mail) is one of most widely used services of Internet. This service allows an Internet user to send a **message in formatted manner (mail)** to the other Internet user in any part of world. Message in mail not only contain text, but it also contains images, audio and videos data. The person who is sending mail is called **sender** and person who receives mail is called **recipient**. It is just like postal mail service. **Components of E-Mail System :** The basic components of an email system are : User Agent (UA), Message Transfer Agent (MTA), Mail Box, and Spool file. These are explained as following below.

1. **User Agent (UA) :** The UA is normally a program which is used to send and receive mail. Sometimes, it is called as mail reader. It accepts variety of commands for composing, receiving and replying to messages as well as for manipulation of the mailboxes.
2. **Message Transfer Agent (MTA) :** MTA is actually responsible for transfer of mail from one system to another. To send a mail, a system must have client MTA and system MTA. It transfer mail to mailboxes of recipients if they are connected in the same machine. It delivers mail to peer MTA if destination mailbox is in another machine. The delivery from one MTA to another MTA is done by Simple Mail Transfer Protocol.



1. **Mailbox :** It is a file on local hard drive to collect mails. Delivered mails are present in this file. The user can read it delete it according to his/her requirement. To use e-mail system each user must have a mailbox . Access to mailbox is only to owner of mailbox.
2. **Spool file :** This file contains mails that are to be sent. User agent appends outgoing mails in this file using SMTP. MTA extracts pending mail from spool file for their delivery. E-mail allows one name, an **alias**, to represent several different e-mail addresses. It is known as **mailing list**, Whenever user have to sent a message, system checks recipient's name against alias database. If mailing list is present for defined alias, separate messages, one for each entry in the list, must be prepared and handed to MTA. If for defined alias, there is no such mailing list is present, name itself becomes naming address and a single message is delivered to mail transfer entity.

Services provided by E-mail system :

- **Composition –** The composition refer to process that creates messages and answers. For composition any kind of text editor can be used.
- **Transfer –** Transfer means sending procedure of mail i.e. from the sender to recipient.
- **Reporting –** Reporting refers to confirmation for delivery of mail. It help user to check whether their mail is delivered, lost or rejected.
- **Displaying –** It refers to present mail in form that is understand by the user.
- **Disposition –** This step concern with recipient that what will recipient do after receiving mail i.e save mail, delete before reading or delete after reading.

Electronic Mail (e-mail) is one of the most widely used services of the Internet. This service allows an Internet user to send a **message in a formatted manner (mail)** to other Internet users in any part of the world. Message in the mail not only contain text, but it also contains images, audio and videos data. The person who is sending mail is called **sender** and person who receives mail is called the **recipient**. It is just like postal mail service.

Format of E-mail :

An e-mail consists of three parts that are as follows :

1. Envelope
2. Header
3. Body

These are explained as following below.

1. Envelope :

The envelope part encapsulates the message. It contains all information that is required for sending any e-mail such as destination address, priority and security level. The envelope is used by MTAs for routing message.

2. Header :

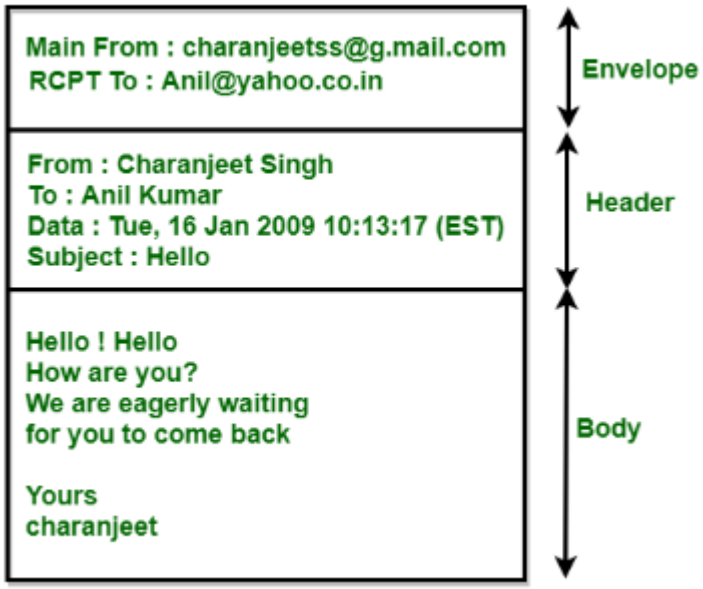
The header consists of a series of lines. Each header field consists of a single line of ASCII text specifying field name, colon and value. The main header fields related to message transport are :

1. **To:** It specifies the DNS address of the primary recipient(s).
2. **Cc :** It refers to carbon copy. It specifies address of secondary recipient(s).
3. **BCC:** It refers to blind carbon copy. It is very similar to Cc. The only difference between Cc and Bcc is that it allow user to send copy to the third party without primary and secondary recipient knowing about this.
4. **From :** It specifies name of person who wrote message.
5. **Sender :** It specifies e-mail address of person who has sent message.
6. **Received :** It refers to identity of sender's, data and also time message was received. It also contains the information which is used to find bugs in routing system.

7. **Return-Path:** It is added by the message transfer agent. This part is used to specify how to get back to the sender.

3. Body:- The body of a message contains text that is the actual content/message that needs to be sent, such as "Employees who are eligible for the new health care program should contact their supervisors by next Friday if they want to switch." The message body also may include signatures or automatically generated text that is inserted by the sender's email system.

The above-discussed field is represented in tabular form as follows :



Advantages of E-mail :

1. E-mails provides faster and easy mean of communication. One can send message to any person at any place of world by just clicking mouse.
2. Various folders and sub-folders can be created within inbox of mail, so it provide management of messages.
3. It is effective and cheap means of communication because single message can be send to multiple people at same time.
4. E-mails are very easy to filter. User according to his/her priority can prioritize e-mail by specifying subject of e-mail.
5. E-mail is not just only for textual message. One can send any kind of multimedia within mail.
6. E-mail can be send at any hour of day, thus ensures timeliness of message.
7. It is secure and reliable method to deliver our message.
8. It also provide facility for edition and formatting of textual messages.
9. There is also facility of auto-responders in e-mail i.e. to send automated e-mails with certain text.
10. To write an e-mail there is no need of any kind of paper, thus it is environment friendly.

Disadvantages of E-mail :

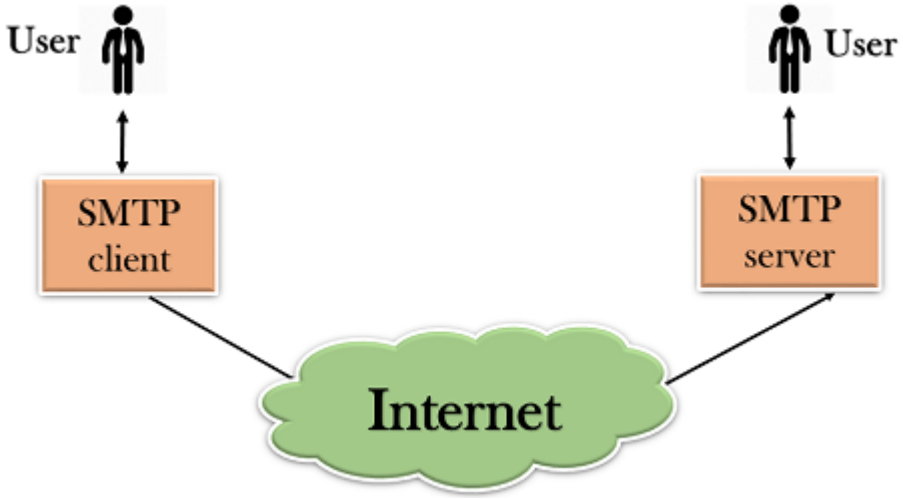
1. It is source of viruses. It is capable to harm one's computer and read out user's e-mail address book and send themselves to number of people around the world.
2. It can be source of various spams. These spam mails can fill up inbox and to deletion of these mail consumes lot of time.

3. It is informal method of communication. The documents those require signatures are not managed by e-mail.
4. To use facility of e-mail, user must have an access to internet and there are many parts of world where people does not have access to Internet.
5. Sometimes, e-mails becomes misunderstood as it is not capable of expressing emotions.
6. To be updated, user have to check inbox from time-to-time.

SMTP

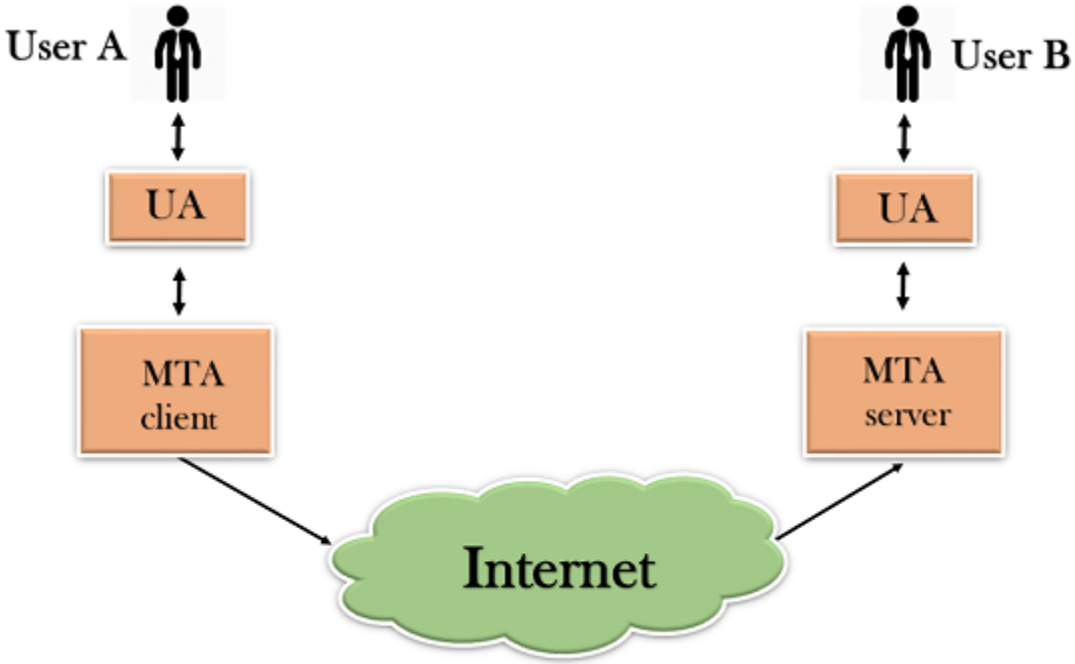
- SMTP stands for Simple Mail Transfer Protocol.
- SMTP is a set of communication guidelines that allow software to transmit an electronic mail over the internet is called **Simple Mail Transfer Protocol**.
- It is a program used for sending messages to other computer users based on e-mail addresses.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

Components of SMTP

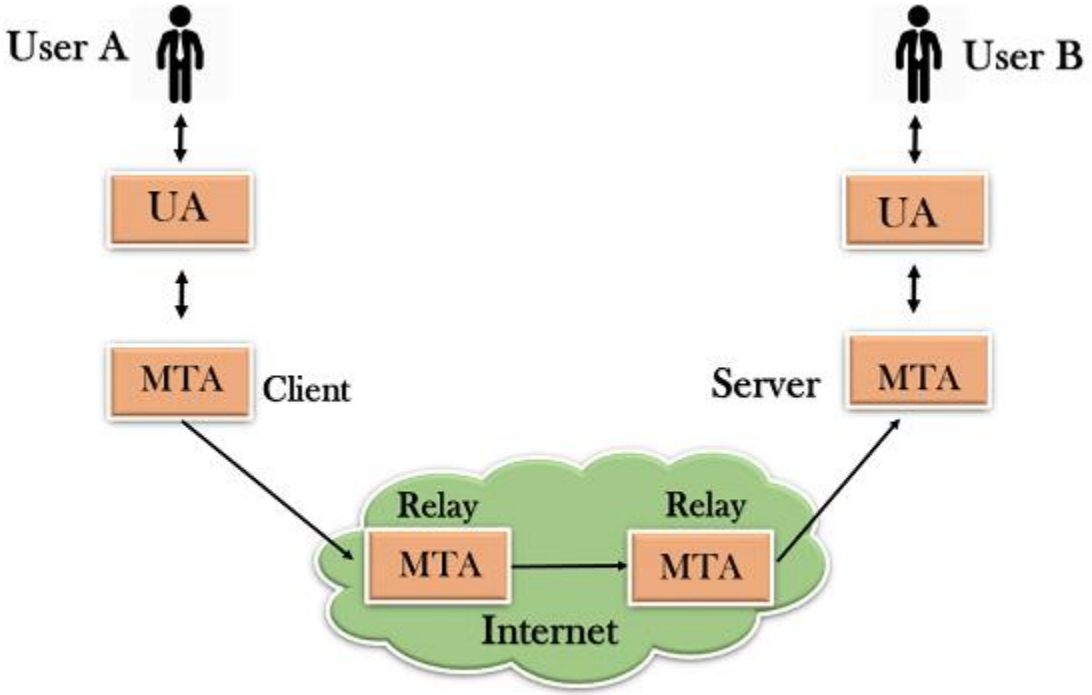


- First, we will break the SMTP client and SMTP server into two components such as user agent (UA) and mail transfer agent (MTA). The user agent (UA) prepares the

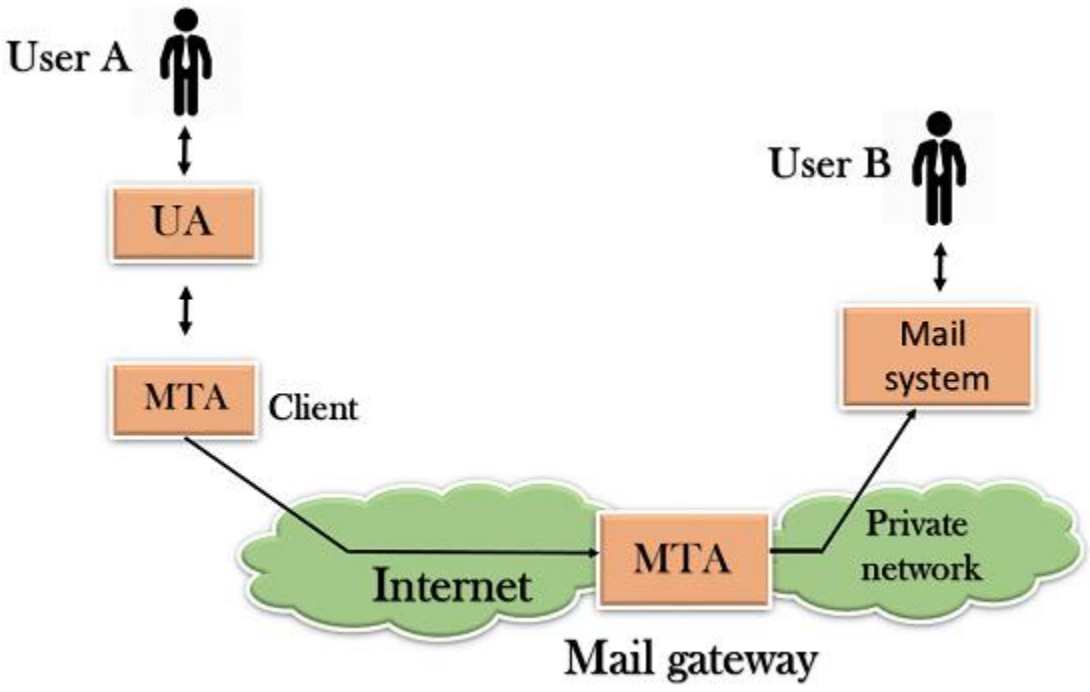
message, creates the envelope and then puts the message in the envelope. The mail transfer agent (MTA) transfers this mail across the internet.



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



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



Working of SMTP

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

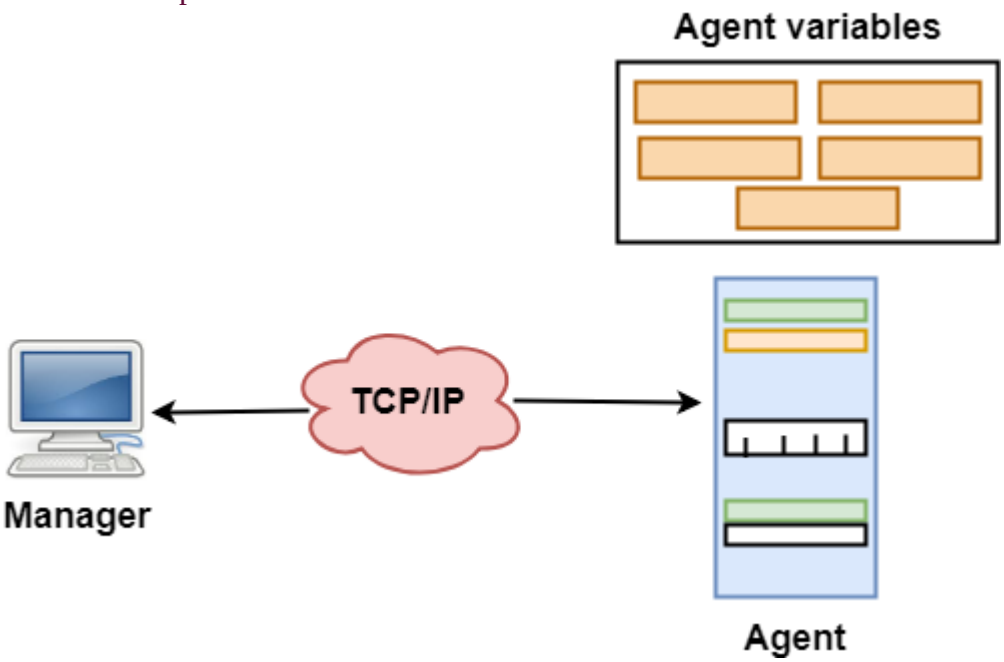
domain name and IP address of the recipient's domain. Once the record is located, MTA connects to the exchange server to relay the message.

4. **Receipt and Processing of Mail:** Once the incoming message is received, the exchange server delivers it to the incoming server (Mail Delivery Agent) which stores the e-mail where it waits for the user to retrieve it.
5. **Access and Retrieval of Mail:** The stored email in MDA can be retrieved by using MUA (Mail User Agent). MUA can be accessed by using login and password.

SNMP

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

SNMP Concept



- SNMP has two components Manager and agent.
- The manager is a host that controls and monitors a set of agents such as routers.
- It is an application layer protocol in which a few manager stations can handle a set of agents.
- The protocol designed at the application level can monitor the devices made by different manufacturers and installed on different physical networks.
- It is used in a heterogeneous network made of different LANs and WANs connected by routers or gateways.

Managers & Agents

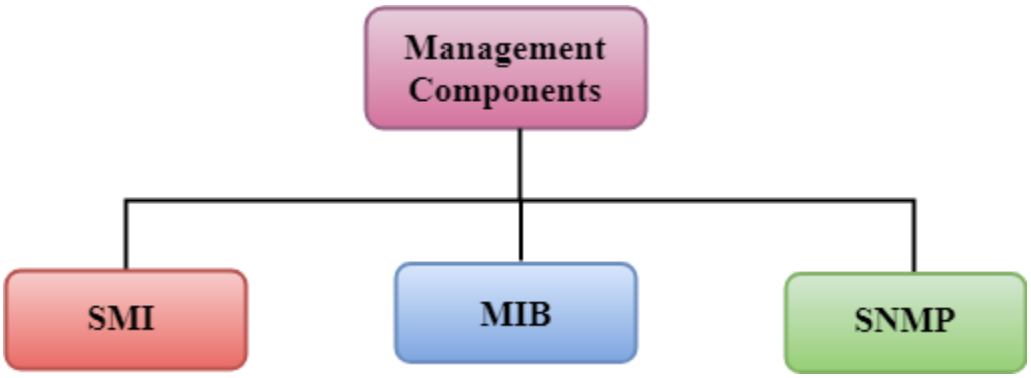
- A manager is a host that runs the SNMP client program while the agent is a router that runs the SNMP server program.
- Management of the internet is achieved through simple interaction between a manager and agent.
- The agent is used to keep the information in a database while the manager is used to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

Management with SNMP has three basic ideas:

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

Management Components

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

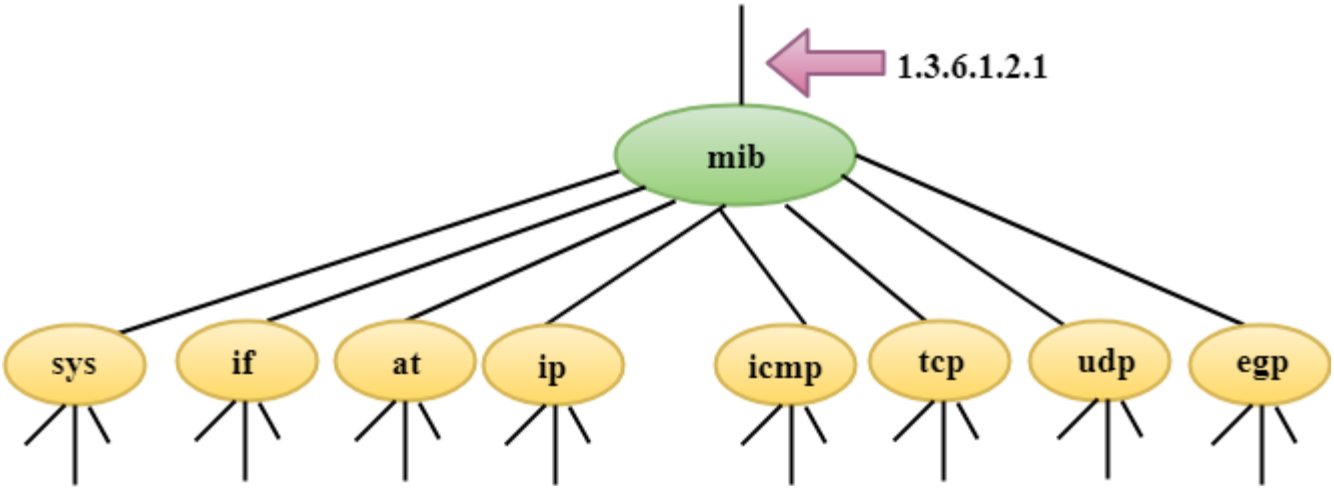


SMI

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

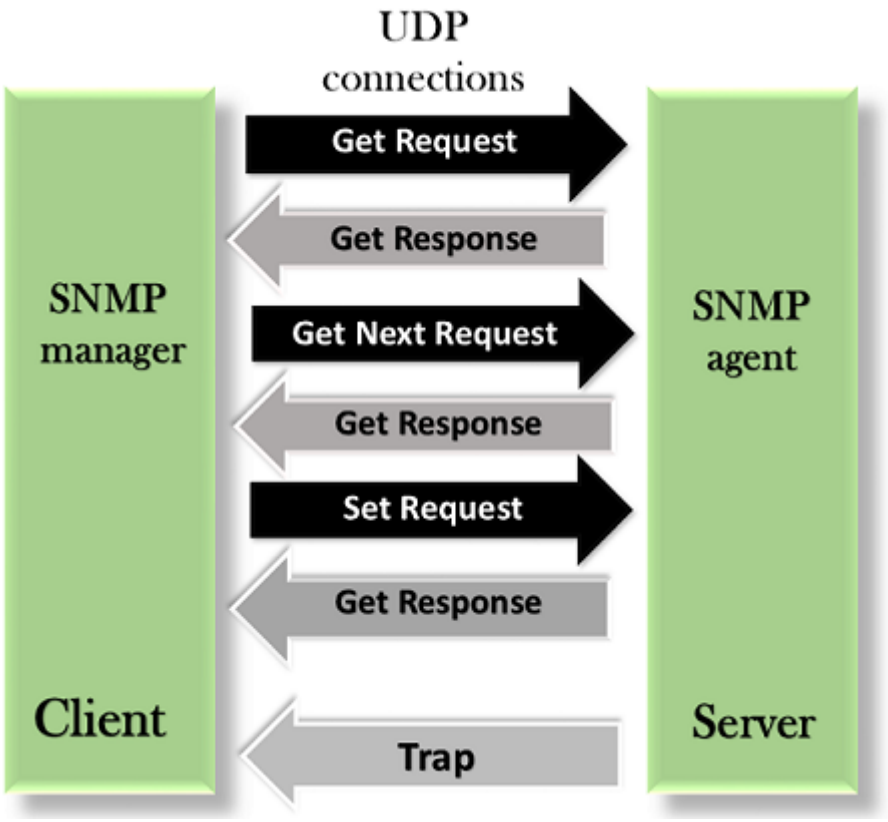
MIB

- The MIB (Management information base) is a second component for the network management.
- Each agent has its own MIB, which is a collection of all the objects that the manager can manage. MIB is categorized into eight groups: system, interface, address



translation, ip, icmp, tcp, udp, and egp. These groups are under the mib object.

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



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

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

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

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

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

HTTP

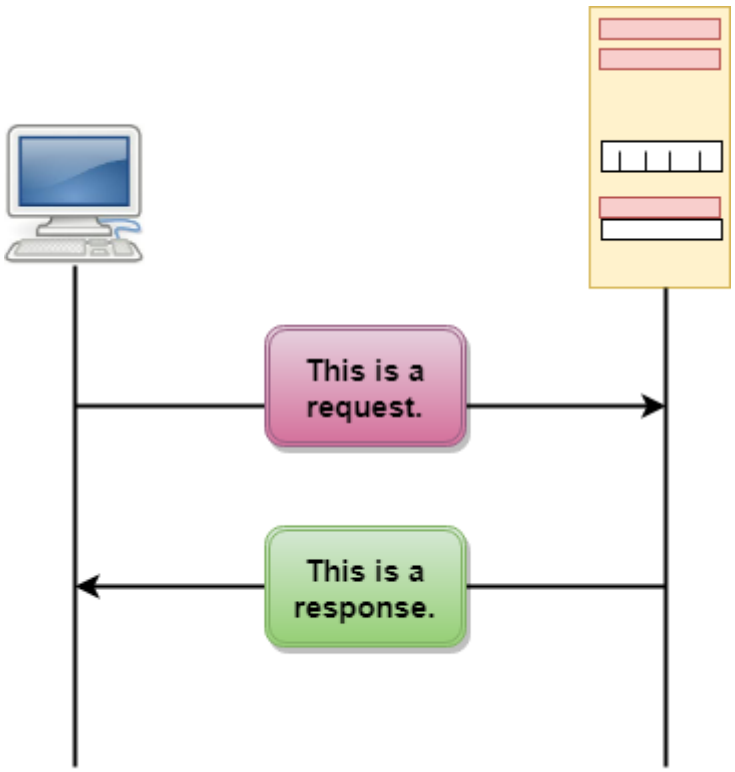
- HTTP stands for **HyperText Transfer Protocol**.
- It is a protocol used to access the data on the World Wide Web (www).
- The HTTP protocol can be used to transfer the data in the form of plain text, hypertext, audio, video, and so on.

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

Features of HTTP:

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

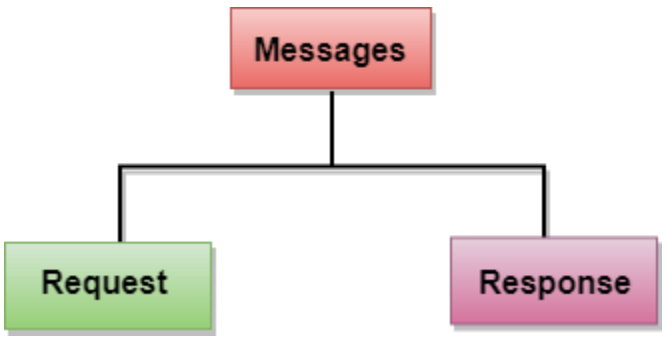
HTTP Transactions

- 

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

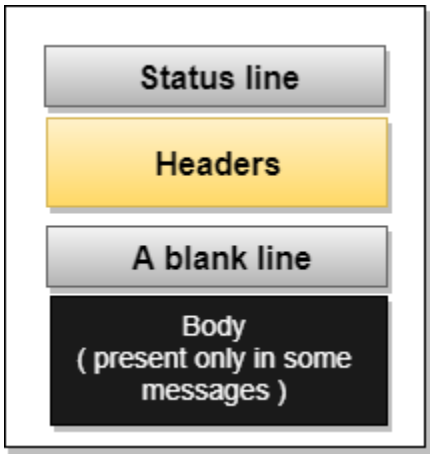
Messages

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

- 

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

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



Uniform Resource Locator (URL)

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

- **Method:** The method is the protocol used to retrieve the document from a server. For example, HTTP.

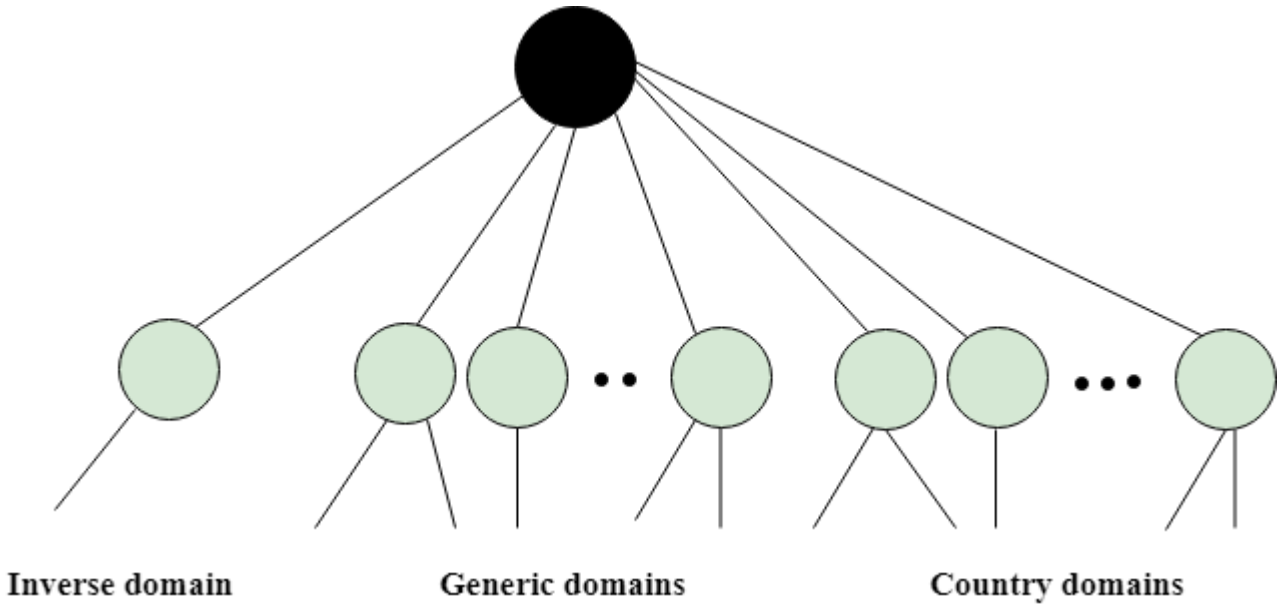
- **Host:** The host is the computer where the information is stored, and the computer is given an alias name. Web pages are mainly stored in the computers and the computers are given an alias name that begins with the characters "www". This field is not mandatory.
- **Port:** The URL can also contain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

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.

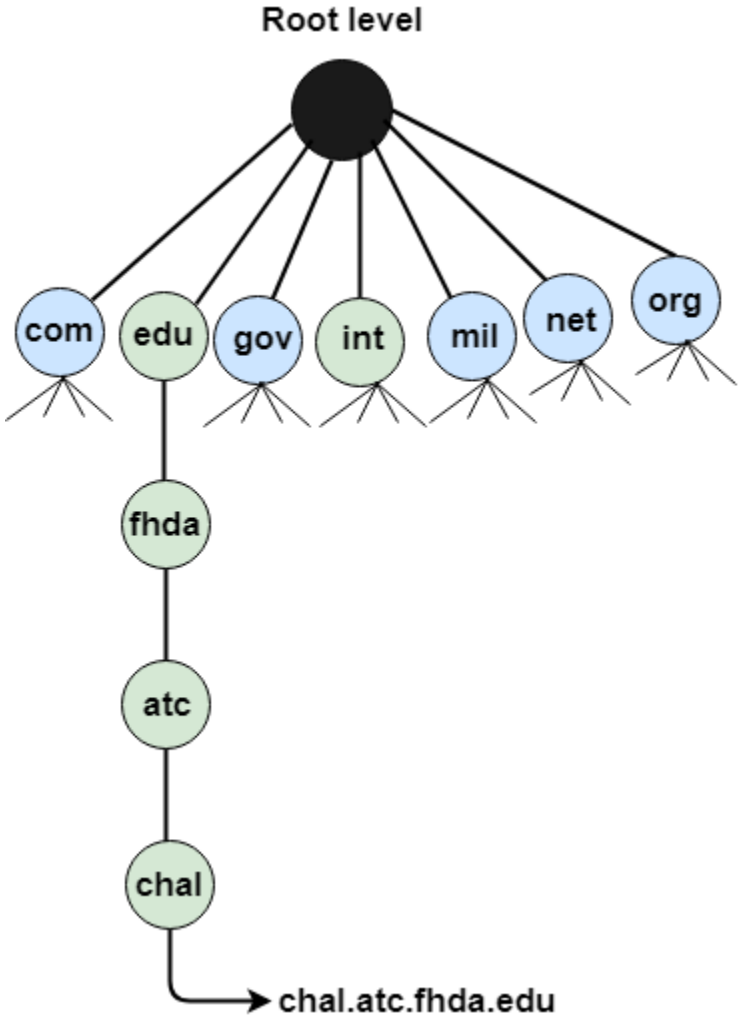


Generic Domains

- 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.
- **CISCO PACKET TRACER**

The main purpose of Cisco Packet Tracer is to help students learn the principles of networking with hands-on experience as well as develop Cisco technology specific skills. Since the protocols are implemented in software only method, this tool cannot replace the hardware Routers or Switches. Interestingly, this tool does not only include Cisco products but also many more networking devices.

- Using this tool is widely encouraged as it is part of the curriculum like CCNA, CCENT where Faculties use Packet Trace to demonstrate technical concepts and networking systems. Students complete assignments using this tool, working on their own or in teams.

Workspace :

1. **Logical –**
   Logical workspace shows the logical network topology of the network the user has built. It represents the placing, connecting and clustering virtual network devices.
2. **Physical –**
   Physical workspace shows the graphical physical dimension of the logical network. It

depicts the scale and placement in how network devices such as routers, switches and hosts would look in a real environment. It also provides geographical representation of networks, including multiple buildings, cities and wiring closets.

Key Features:

- Unlimited devices
- E-learning
- Customize single/multi user activities
- Interactive Environment
- Visualizing Networks
- Real-time mode and Simulation mode
- Self-paced
- Supports majority of networking protocols
- International language support
- Cross platform compatibility