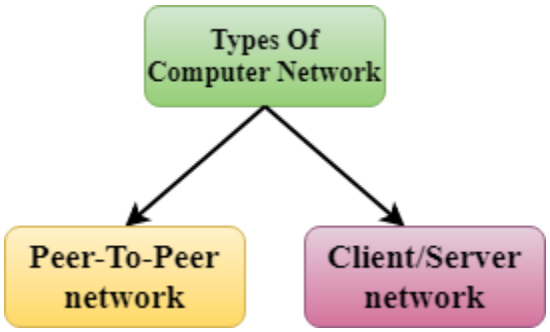
Chapter-1

Network Fundamentals

Computer Network Architecture

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

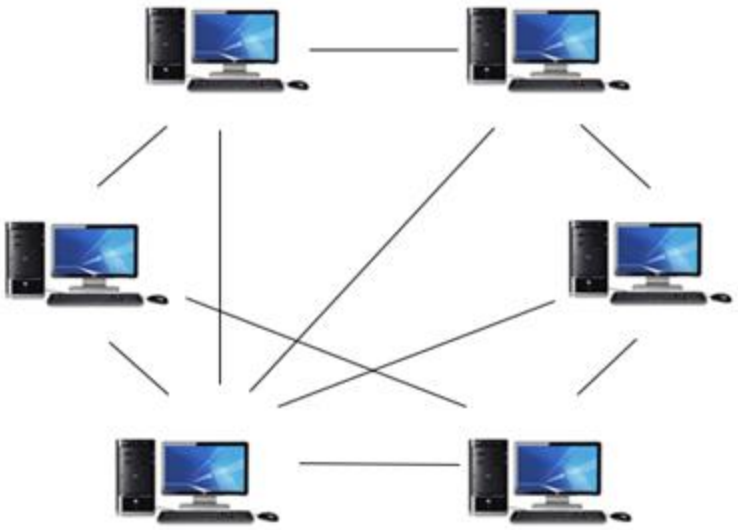
The two types of network architectures are used:



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

Peer-To-Peer network

- Peer-To-Peer network is a network in which all the computers are linked together with equal privilege and responsibilities for processing the data.
- Peer-To-Peer network is useful for small environments, usually up to 10 computers.
- Peer-To-Peer network has no dedicated server.
- Special permissions are assigned to each computer for sharing the resources, but this can lead to a problem if the computer with the resource is down.



Advantages Of Peer-To-Peer Network:

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

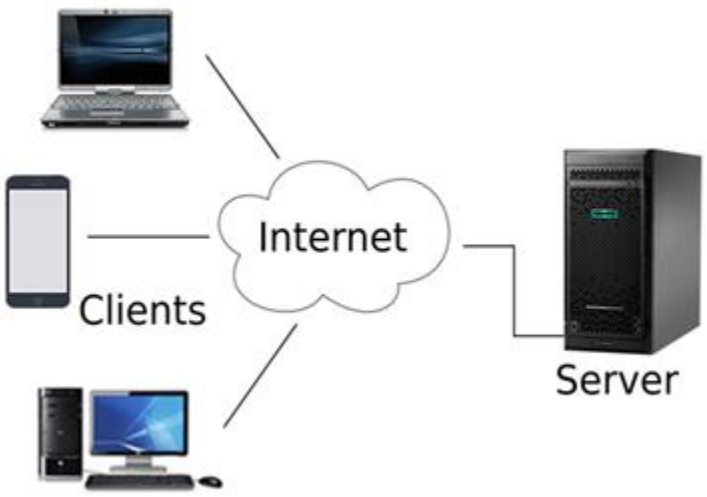
Disadvantages Of Peer-To-Peer Network:

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

Client/Server Network

- Client/Server network is a network model designed for the end users called clients, to access the resources such as songs, video, etc. from a central computer known as Server.
- The central controller is known as a **server** while all other computers in the network are called **clients**.
- A server performs all the major operations such as security and network management.
- A server is responsible for managing all the resources such as files, directories, printer, etc.

- All the clients communicate with each other through a server. For example, if client1 wants to send some data to client 2, then it first sends the request to the server for the permission. The server sends the response to the client 1 to initiate its communication with the client 2.



Advantages Of Client/Server network:

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

Disadvantages Of Client/Server network:

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

Computer Network Components

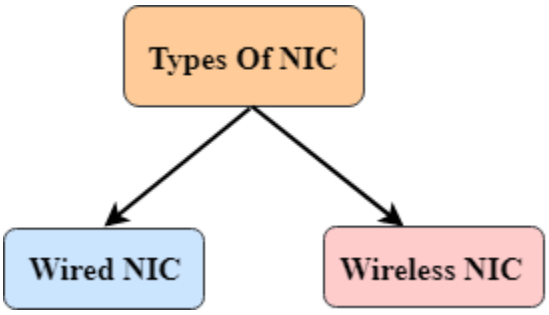
Computer network components are the *major parts* which are needed to *install the software*. Some important network components are **NIC**, **switch**, **cable**, **hub**, **router**, and **modem**. Depending on the type of network that we need to install, some network components can also be removed.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

Following are the major components required to install a network:

NIC

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

There are two types of NIC:



1. Wired NIC
2. Wireless NIC

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

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

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Hub

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

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

Switch

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

Router

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

Advantages Of Router:

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

the network. Router splits the single network into two networks of 12 workstations each, reduces the traffic load by half.

Modem

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

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

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

Cables and Connectors

Cable is a transmission media used for transmitting a signal.

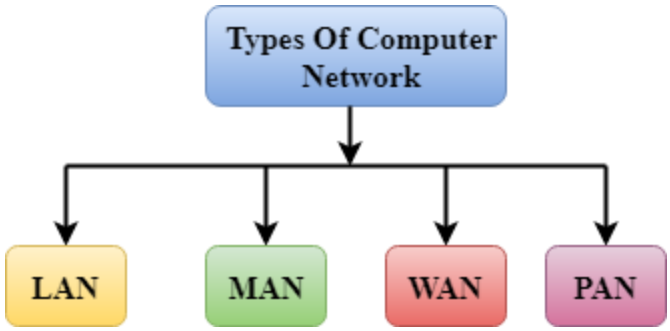
There are three types of cables used in transmission:

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

Computer Network Types

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

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



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

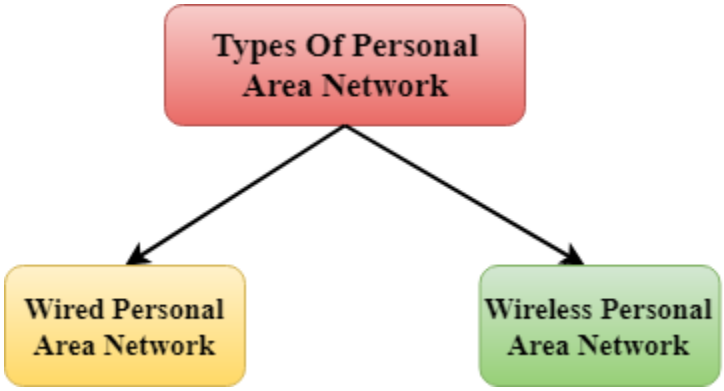
LAN(Local Area Network)

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

PAN(Personal Area Network)

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

There are two types of Personal Area Network:



- Wired Personal Area Network
- Wireless Personal Area Network

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

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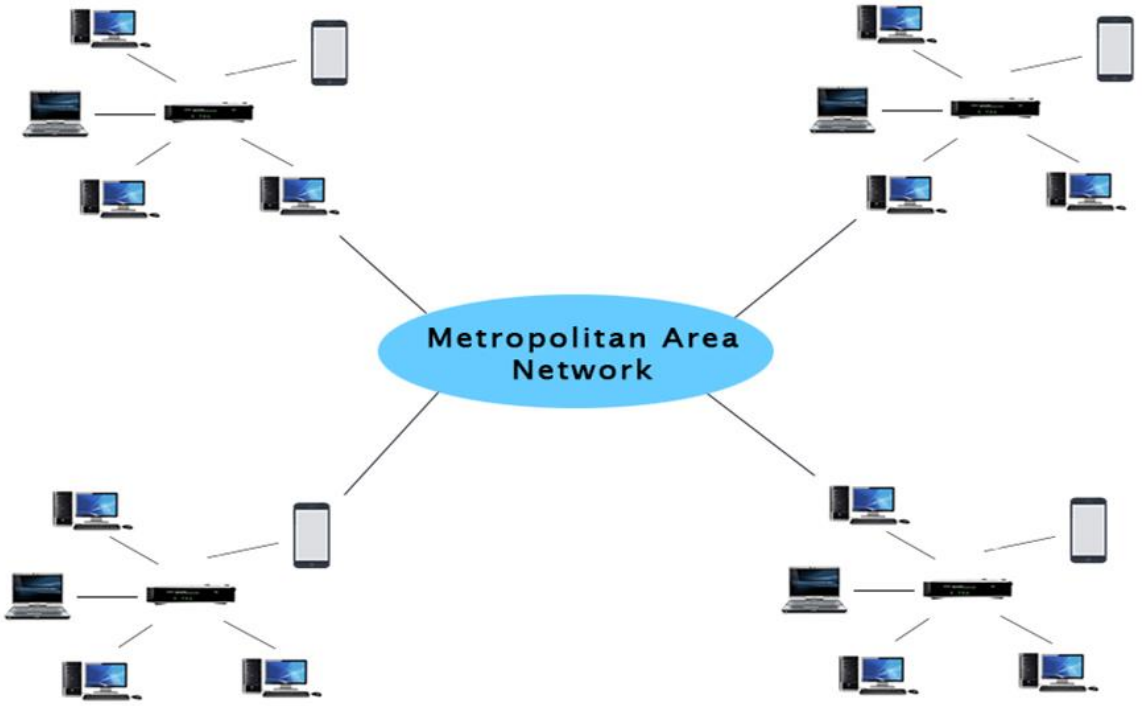
Wired Personal Area Network: Wired Personal Area Network is created by using the USB.

Examples Of Personal Area Network:

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

MAN(Metropolitan Area Network)

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

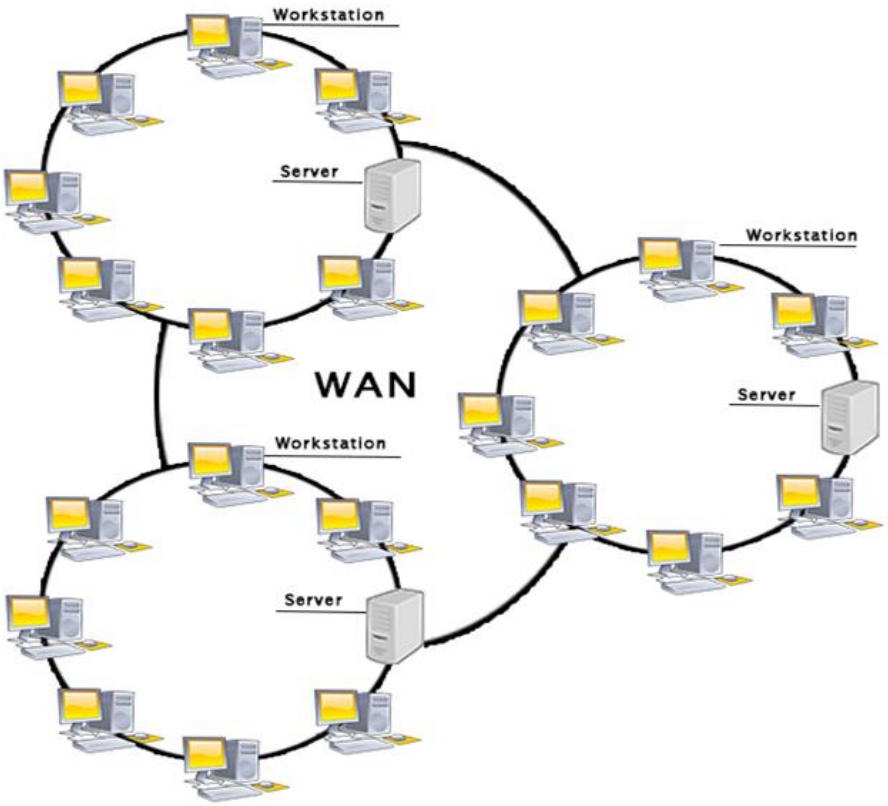


Uses Of Metropolitan Area Network:

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

WAN(Wide Area Network)

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



Examples Of Wide Area Network:

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

Advantages Of Wide Area Network:

Following are the advantages of the Wide Area Network:

- **Geographical area:** A Wide Area Network provides a large geographical area. Suppose if the branch of our office is in a different city then we can connect with them through WAN. The internet provides a leased line through which we can connect with another branch.
- **Centralized data:** In case of WAN network, data is centralized. Therefore, we do not need to buy the emails, files or back up servers.

- **Get updated files:** Software companies work on the live server. Therefore, the programmers get the updated files within seconds.
- **Exchange messages:** In a WAN network, messages are transmitted fast. The web application like Facebook, Whatsapp, Skype allows you to communicate with friends.
- **Sharing of software and resources:** In WAN network, we can share the software and other resources like a hard drive, RAM.
- **Global business:** We can do the business over the internet globally.
- **High bandwidth:** If we use the leased lines for our company then this gives the high bandwidth. The high bandwidth increases the data transfer rate which in turn increases the productivity of our company.

Disadvantages of Wide Area Network:

The following are the disadvantages of the Wide Area Network:

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

Internetwork

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

Types Of Internetwork:

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

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

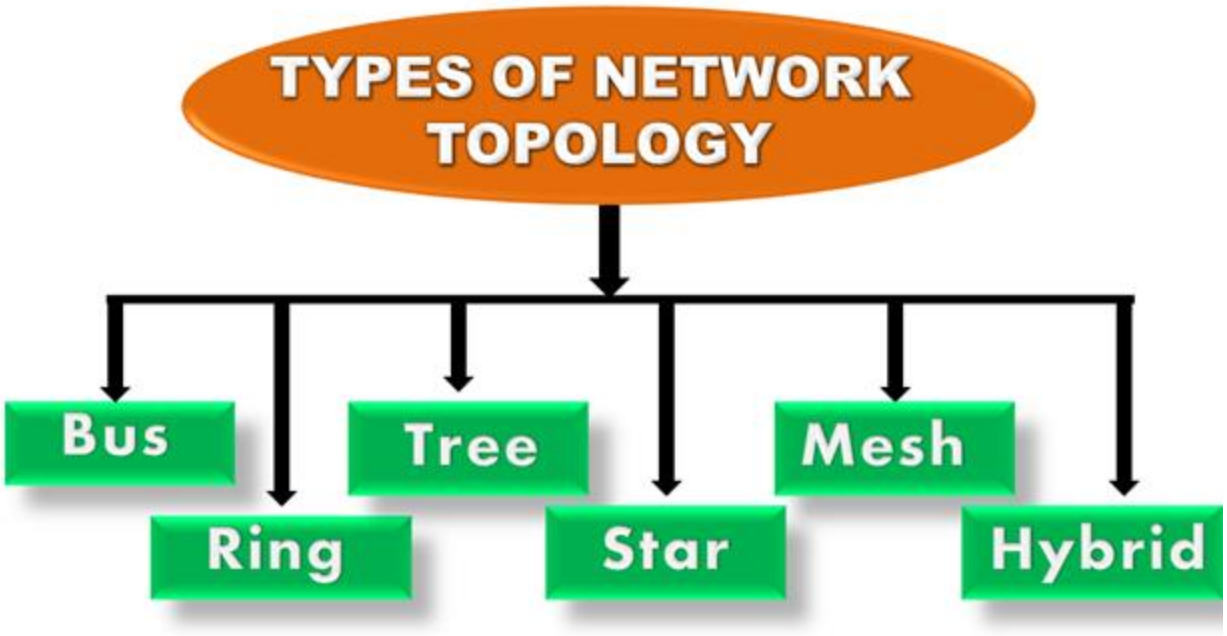
Intranet advantages:

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

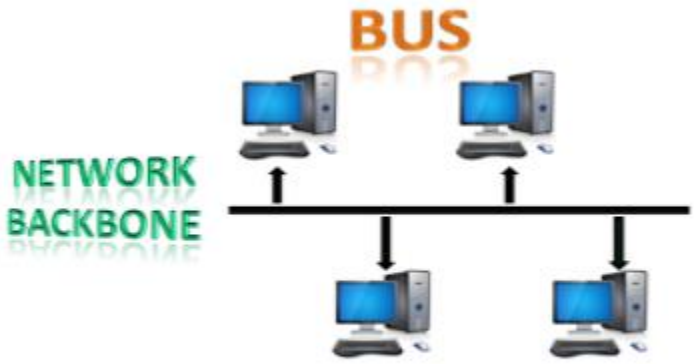
What is Topology?

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

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



Bus Topology



- The bus topology is designed in such a way that all the stations are connected through a single cable known as a backbone cable.
- Each node is either connected to the backbone cable by drop cable or directly connected to the backbone cable.
- When a node wants to send a message over the network, it puts a message over the network. All the stations available in the network will receive the message whether it has been addressed or not.

- The bus topology is mainly used in 802.3 (ethernet) and 802.4 standard networks.
- The configuration of a bus topology is quite simpler as compared to other topologies.
- The backbone cable is considered as a **"single lane"** through which the message is broadcast to all the stations.
- The most common access method of the bus topologies is **CSMA** (Carrier Sense Multiple Access).

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

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

Advantages of Bus topology:

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

Disadvantages of Bus topology:

- **Extensive cabling:** A bus topology is quite simpler, but still it requires a lot of cabling.
- **Difficult troubleshooting:** It requires specialized test equipment to determine the cable faults. If any fault occurs in the cable, then it would disrupt the communication for all the nodes.

- **Signal interference:** If two nodes send the messages simultaneously, then the signals of both the nodes collide with each other.
- **Reconfiguration difficult:** Adding new devices to the network would slow down the network.
- **Attenuation:** Attenuation is a loss of signal leads to communication issues. Repeaters are used to regenerate the signal.

Ring Topology

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

Working of Token passing

- A token moves around the network, and it is passed from computer to computer until it reaches the destination.
- The sender modifies the token by putting the address along with the data.
- The data is passed from one device to another device until the destination address matches. Once the token received by the destination device, then it sends the acknowledgment to the sender.
- In a ring topology, a token is used as a carrier.

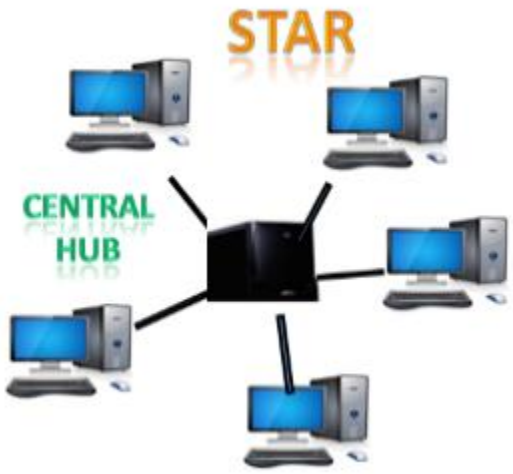
Advantages of Ring topology:

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

Disadvantages of Ring topology:

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

Star Topology



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

Advantages of Star topology

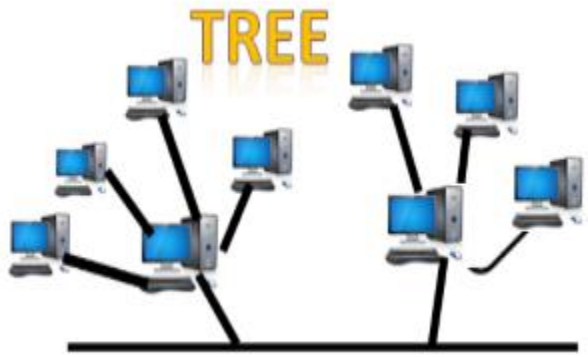
- **Efficient troubleshooting:** Troubleshooting is quite efficient in a star topology as compared to bus topology. In a bus topology, the manager has to inspect the kilometers of cable. In a star topology, all the stations are connected to the centralized network. Therefore, the network administrator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

- **Cost effective:** Star topology networks are cost-effective as it uses inexpensive coaxial cable.
- **High data speeds:** It supports a bandwidth of approx 100Mbps. Ethernet 100BaseT is one of the most popular Star topology networks.

Disadvantages of Star topology

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

Tree topology



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

Advantages of Tree topology

- **Support for broadband transmission:** Tree topology is mainly used to provide broadband transmission, i.e., signals are sent over long distances without being attenuated.

- **Easily expandable:** We can add the new device to the existing network. Therefore, we can say that tree topology is easily expandable.
- **Easily manageable:** In tree topology, the whole network is divided into segments known as star networks which can be easily managed and maintained.
- **Error detection:** Error detection and error correction are very easy in a tree topology.
- **Limited failure:** The breakdown in one station does not affect the entire network.
- **Point-to-point wiring:** It has point-to-point wiring for individual segments.

Disadvantages of Tree topology

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

Mesh topology

- Mesh technology is an arrangement of the network in which computers are interconnected with each other through various redundant connections.
- There are multiple paths from one computer to another computer.

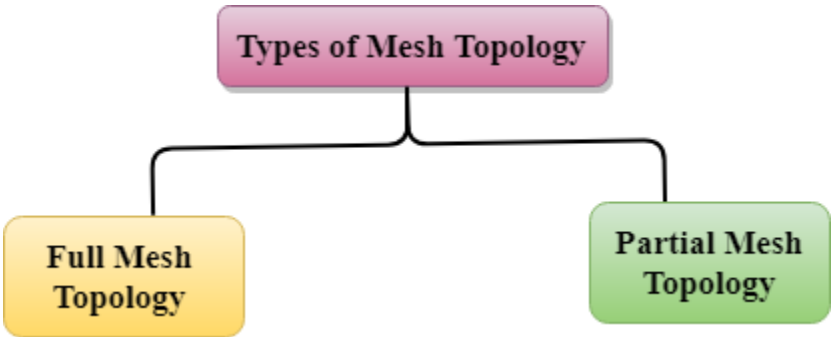
- It does not contain the switch, hub or any central computer which acts as a central point of communication.
- The Internet is an example of the mesh topology.
- Mesh topology is mainly used for WAN implementations where communication failures are a critical concern.
- Mesh topology is mainly used for wireless networks.
- Mesh topology can be formed by using the formula: **Number of cables = (n*(n-1))/2;**

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

How to find Nth Highest Salary in SQL

Mesh topology is divided into two categories:

- Fully connected mesh topology
- Partially connected mesh topology



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

Advantages of Mesh topology:

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

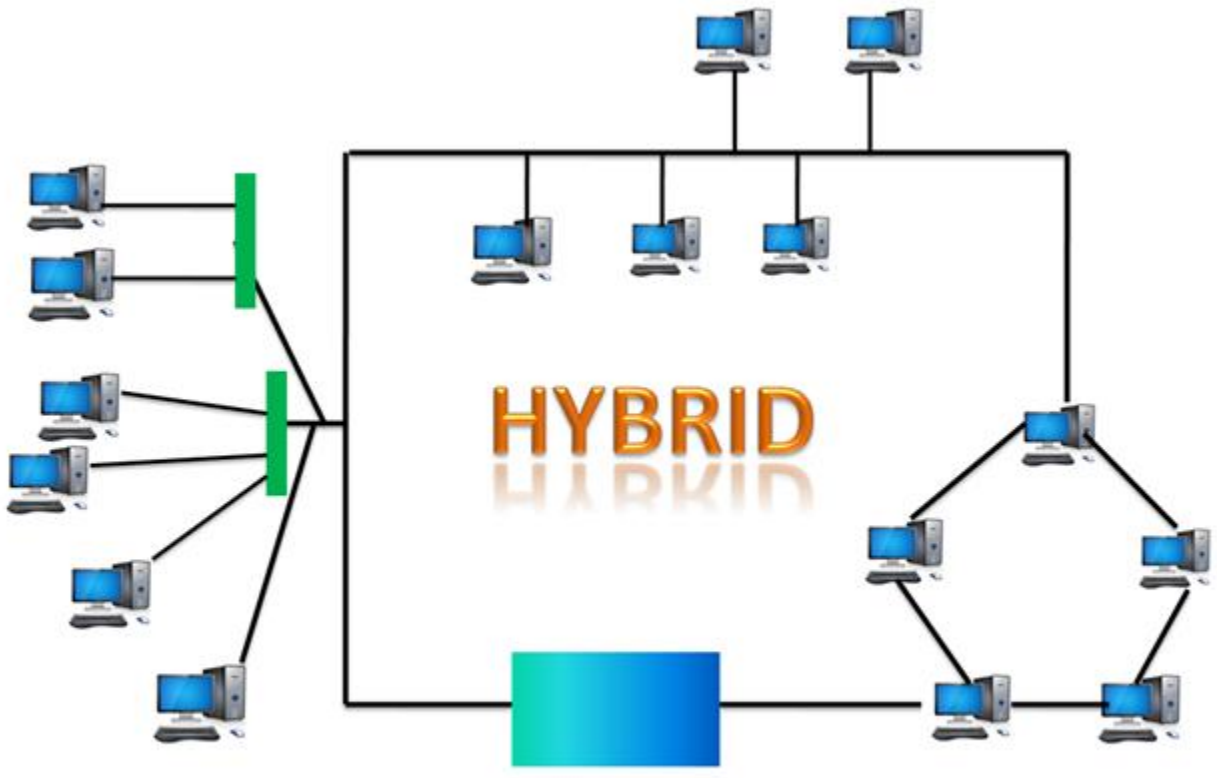
Fast Communication: Communication is very fast between the nodes.

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

Disadvantages of Mesh topology

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

Hybrid Topology



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

Advantages of Hybrid Topology

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

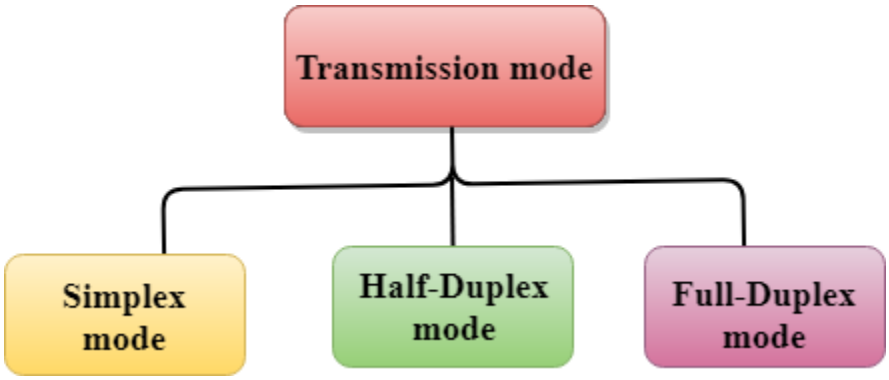
Disadvantages of Hybrid topology

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

Transmission modes

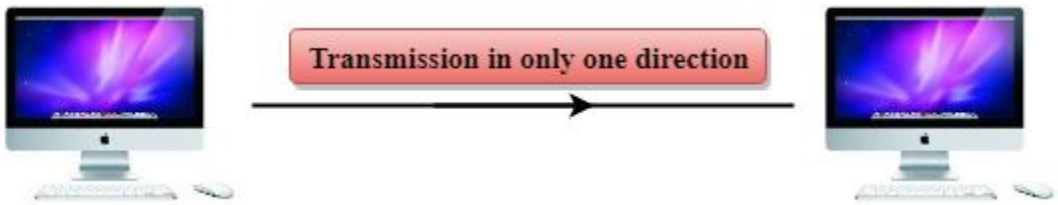
- The way in which data is transmitted from one device to another device is known as **transmission mode**.
- The transmission mode is also known as the communication mode.
- Each communication channel has a direction associated with it, and transmission media provide the direction. Therefore, the transmission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

The Transmission mode is divided into three categories:



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

Simplex mode



- In Simplex mode, the communication is unidirectional, i.e., the data flow in one direction.
- A device can only send the data but cannot receive it or it can receive the data but cannot send the data.

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

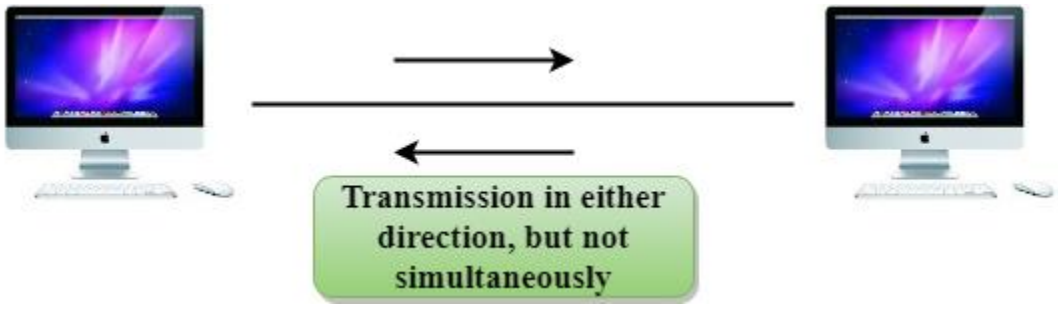
Advantage of Simplex mode:

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

Disadvantage of Simplex mode:

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

Half-Duplex mode



- In a Half-duplex channel, direction can be reversed, i.e., the station can transmit and receive the data as well.
- Messages flow in both the directions, but not at the same time.
- The entire bandwidth of the communication channel is utilized in one direction at a time.
- In half-duplex mode, it is possible to perform the error detection, and if any error occurs, then the receiver requests the sender to retransmit the data.

- A **Walkie-talkie** is an example of the Half-duplex mode. In Walkie-talkie, one party speaks, and another party listens. After a pause, the other speaks and first party listens. Speaking simultaneously will create the distorted sound which cannot be understood.

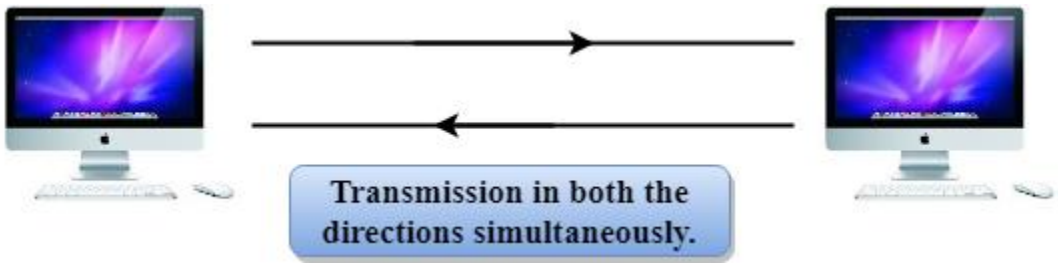
Advantage of Half-duplex mode:

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

Disadvantage of Half-Duplex mode:

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

Full-duplex mode



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

Advantage of Full-duplex mode:

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

Disadvantage of Full-duplex mode:

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

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

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

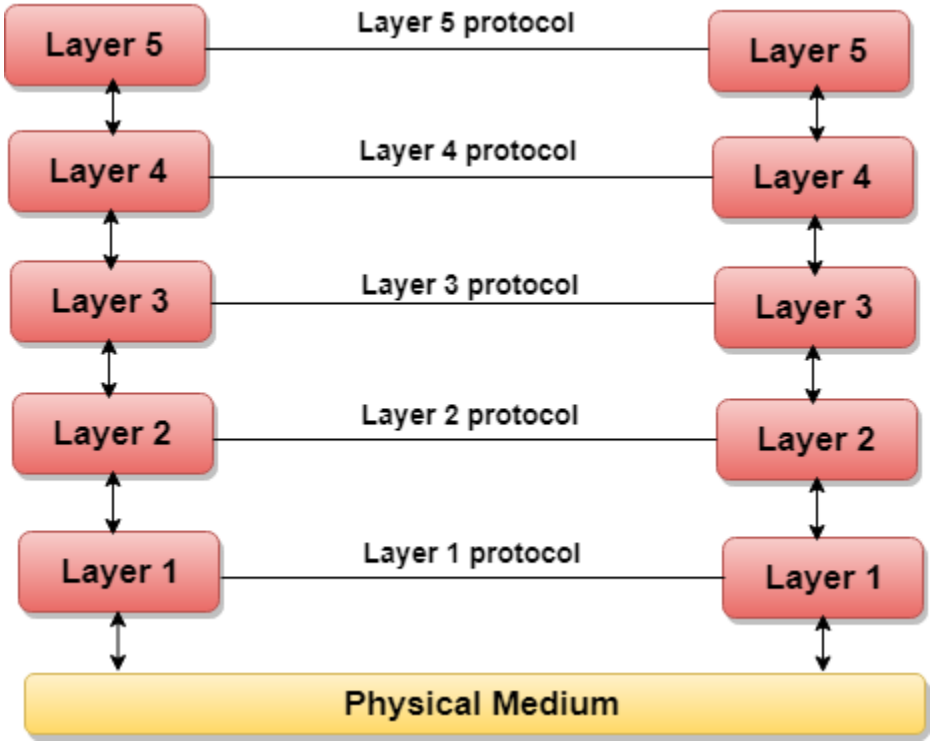
Computer Network Models

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

Layered Architecture

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

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



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

Why do we require Layered architecture?

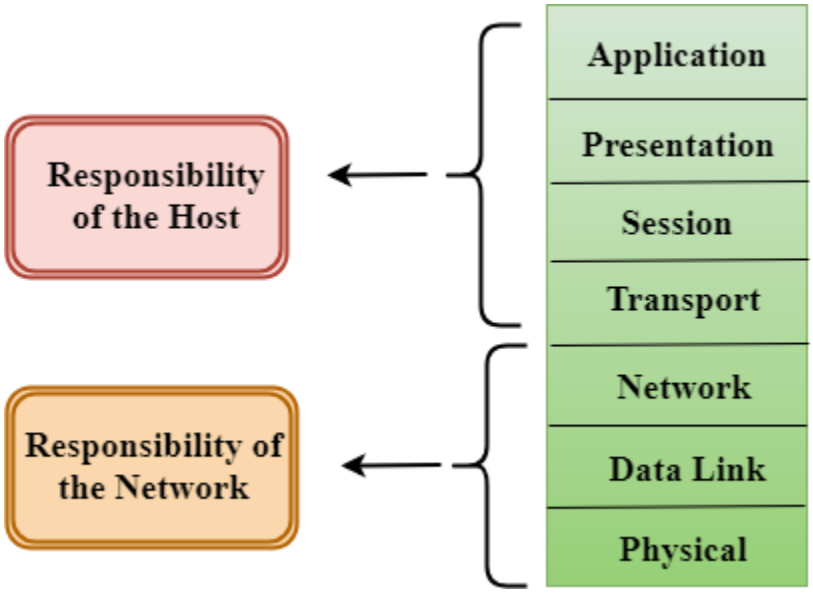
- **Divide-and-conquer approach:** Divide-and-conquer approach makes a design process in such a way that the unmanageable tasks are divided into small and manageable tasks. In short, we can say that this approach reduces the complexity of the design.

- **Modularity:** Layered architecture is more modular. Modularity provides the independence of layers, which is easier to understand and implement.
- **Easy to modify:** It ensures the independence of layers so that implementation in one layer can be changed without affecting other layers.
- **Easy to test:** Each layer of the layered architecture can be analyzed and tested individually.

OSI Model

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

Characteristics of OSI Model:



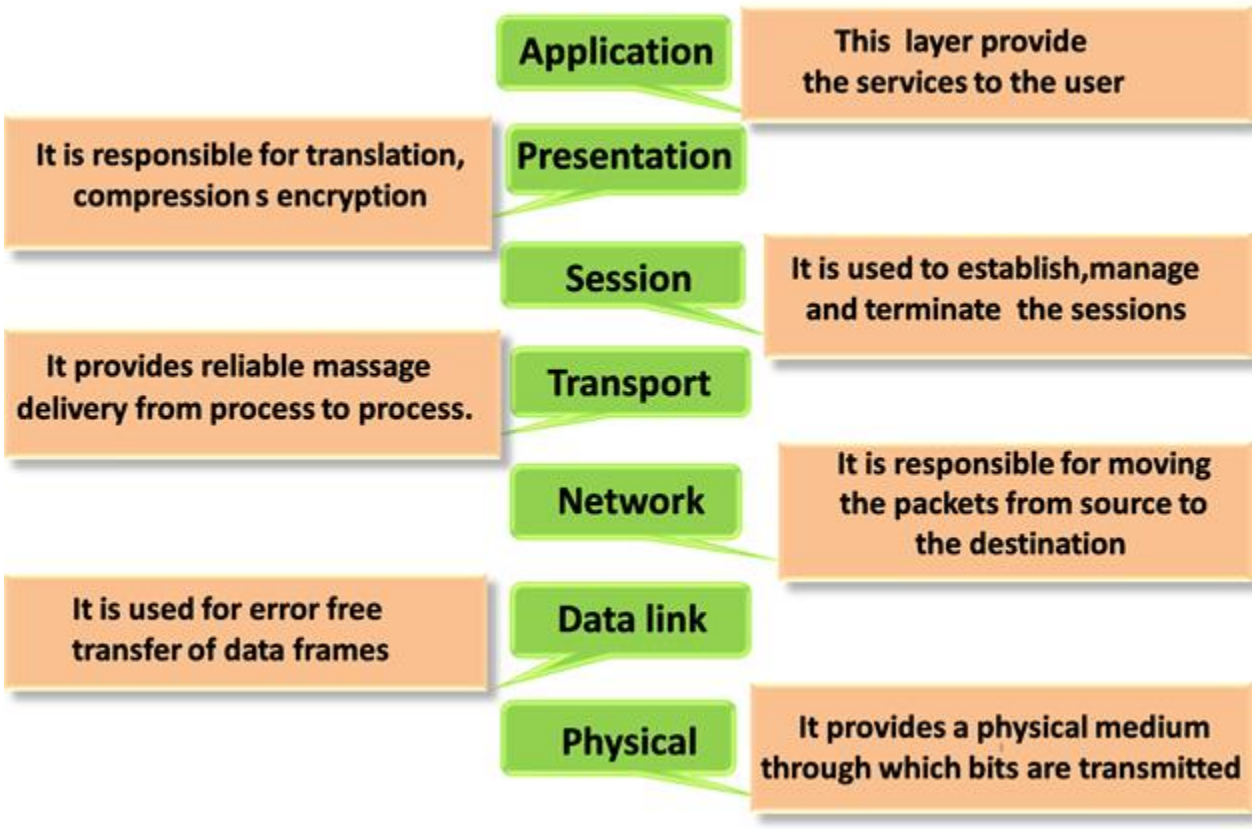
- The OSI model is divided into two layers: upper layers and lower layers.
- The upper layer of the OSI model mainly deals with the application related issues, and they are implemented only in the software. The application layer is closest to the end user. Both the end user and the application layer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

of the OSI model and is closest to the physical medium. The physical layer is mainly responsible for placing the information on the physical medium.

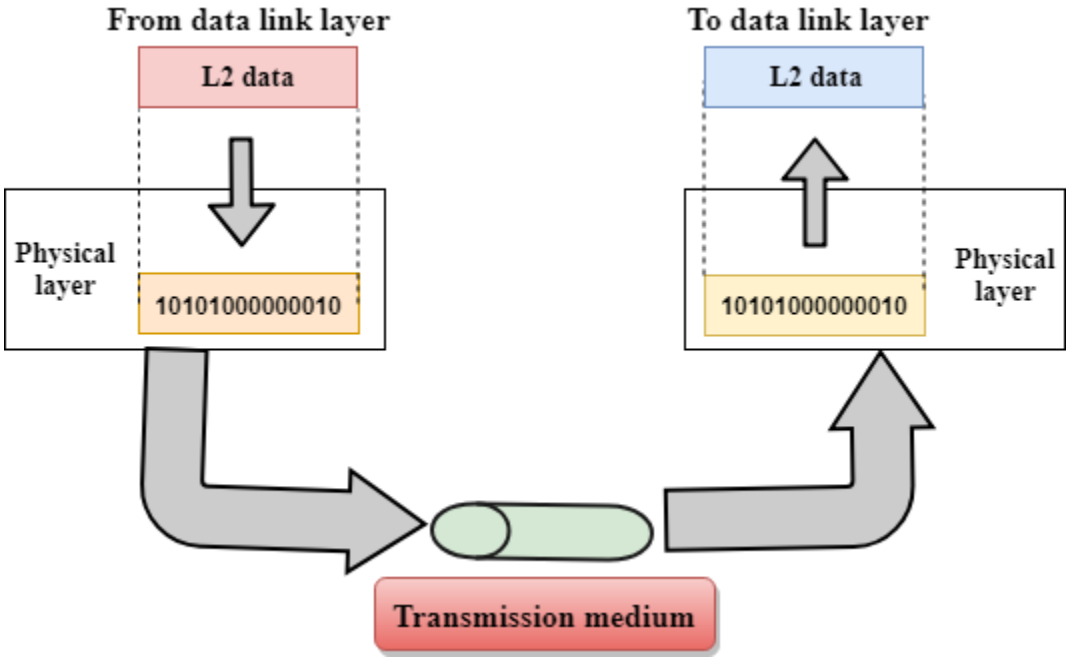
Functions of the OSI Layers

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

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



Physical layer

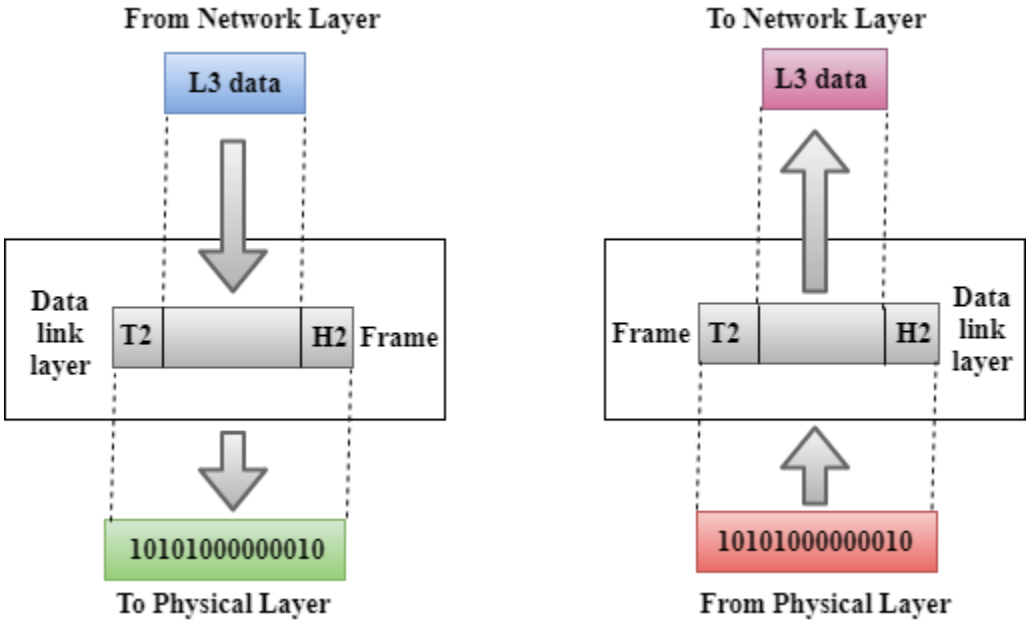


- The main functionality of the physical layer is to transmit the individual bits from one node to another node.
- It is the lowest layer of the OSI model.
- It establishes, maintains and deactivates the physical connection.
- It specifies the mechanical, electrical and procedural network interface specifications.

Functions of a Physical layer:

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

Data-Link Layer



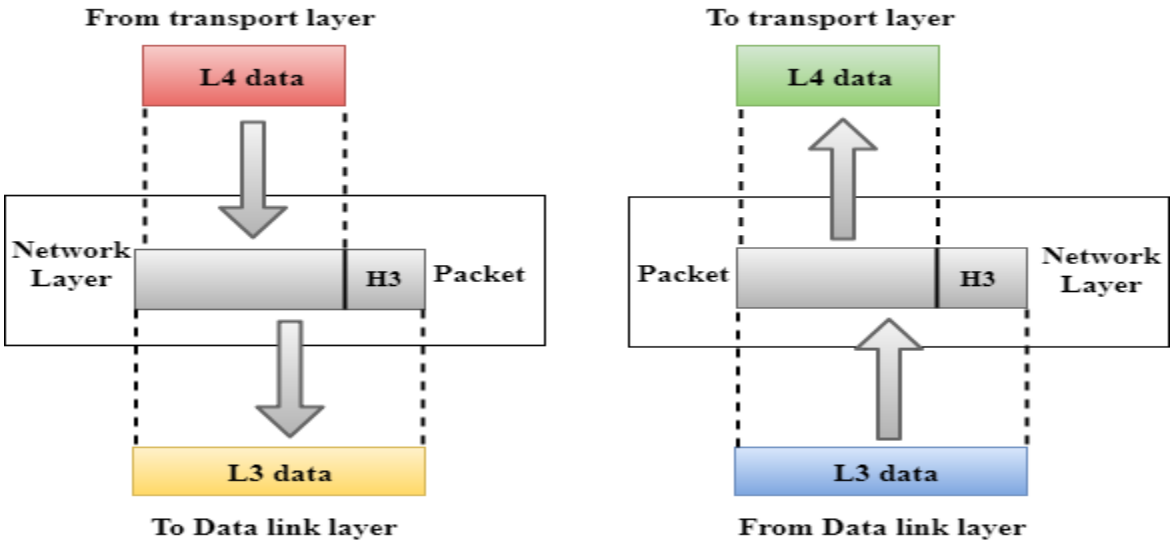
- This layer is responsible for the error-free transfer of data frames.
- It defines the format of the data on the network.
- It provides a reliable and efficient communication between two or more devices.
- It is mainly responsible for the unique identification of each device that resides on a local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

Functions of the Data-link layer

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

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

Network Layer

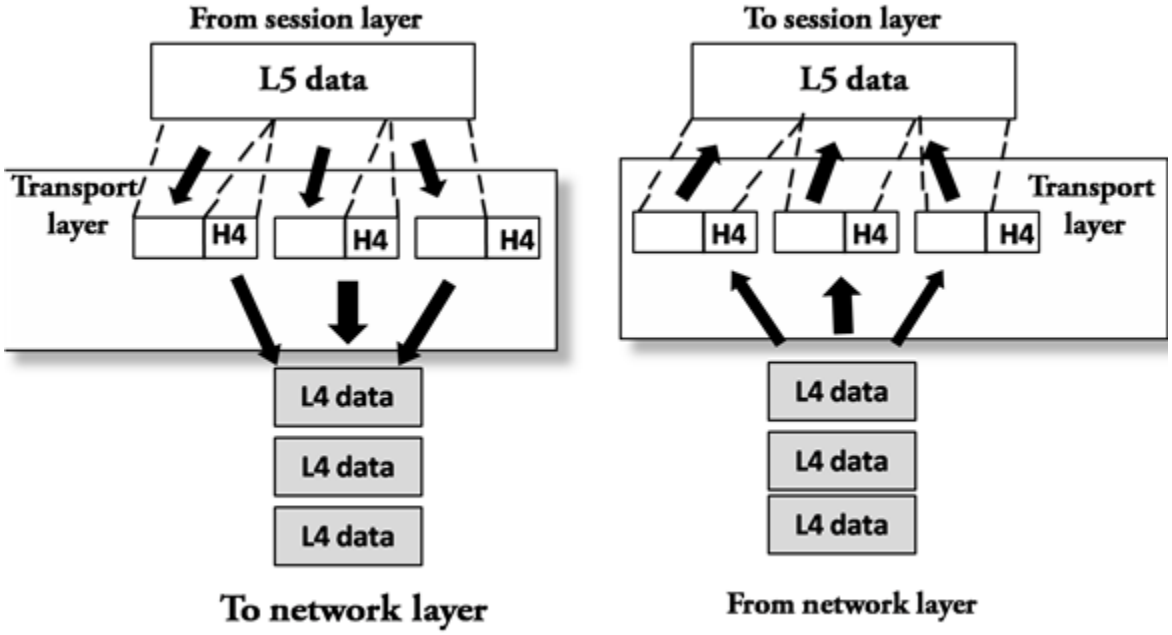


- It is a layer 3 that manages device addressing, tracks the location of devices on the network.
- It determines the best path to move data from source to the destination based on the network conditions, the priority of service, and other factors.
- The Data link layer is responsible for routing and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

Functions of Network Layer:

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

Transport Layer



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

The two protocols used in this layer are:

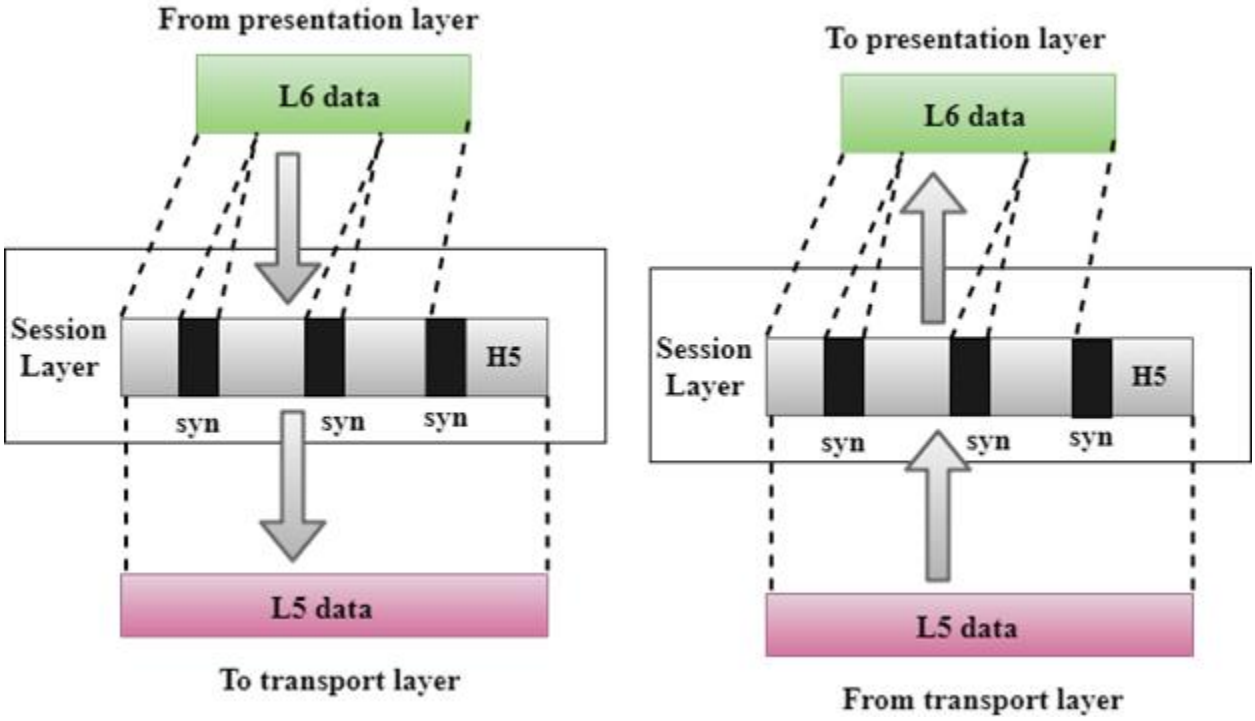
- **Transmission Control Protocol**
  - It is a standard protocol that allows the systems to communicate over the internet.
  - It establishes and maintains a connection between hosts.
  - When data is sent over the TCP connection, then the TCP protocol divides the data into smaller units known as segments.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

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

Functions of Transport Layer:

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

Session Layer

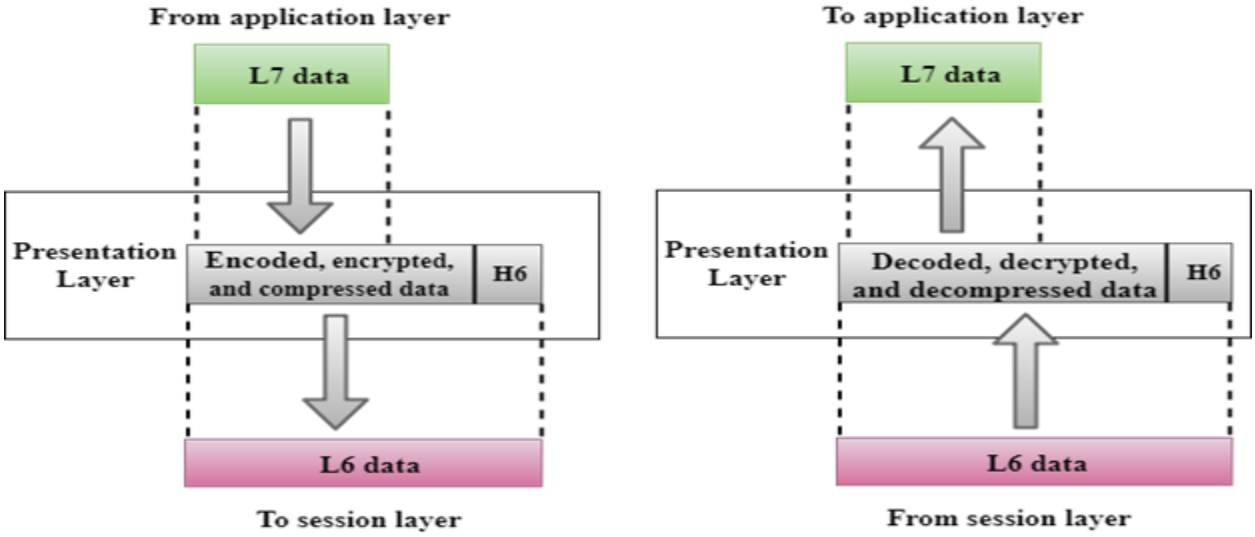


- It is a layer 3 in the OSI model.
- The Session layer is used to establish, maintain and synchronizes the interaction between communicating devices.

Functions of Session layer:

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

Presentation Layer

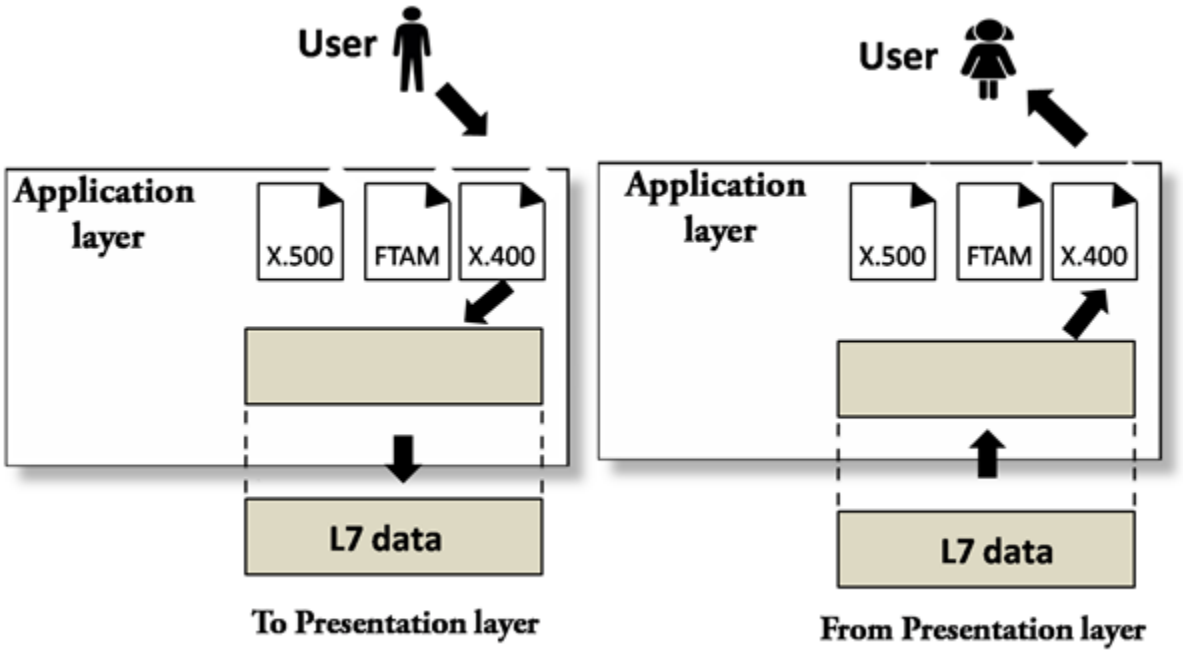


- A Presentation layer is mainly concerned with the syntax and semantics of the information exchanged between the two systems.
- It acts as a data translator for a network.
- This layer is a part of the operating system that converts the data from one presentation format to another format.
- The Presentation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

Functions of Presentation layer:

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

Application Layer



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

Functions of Application layer:

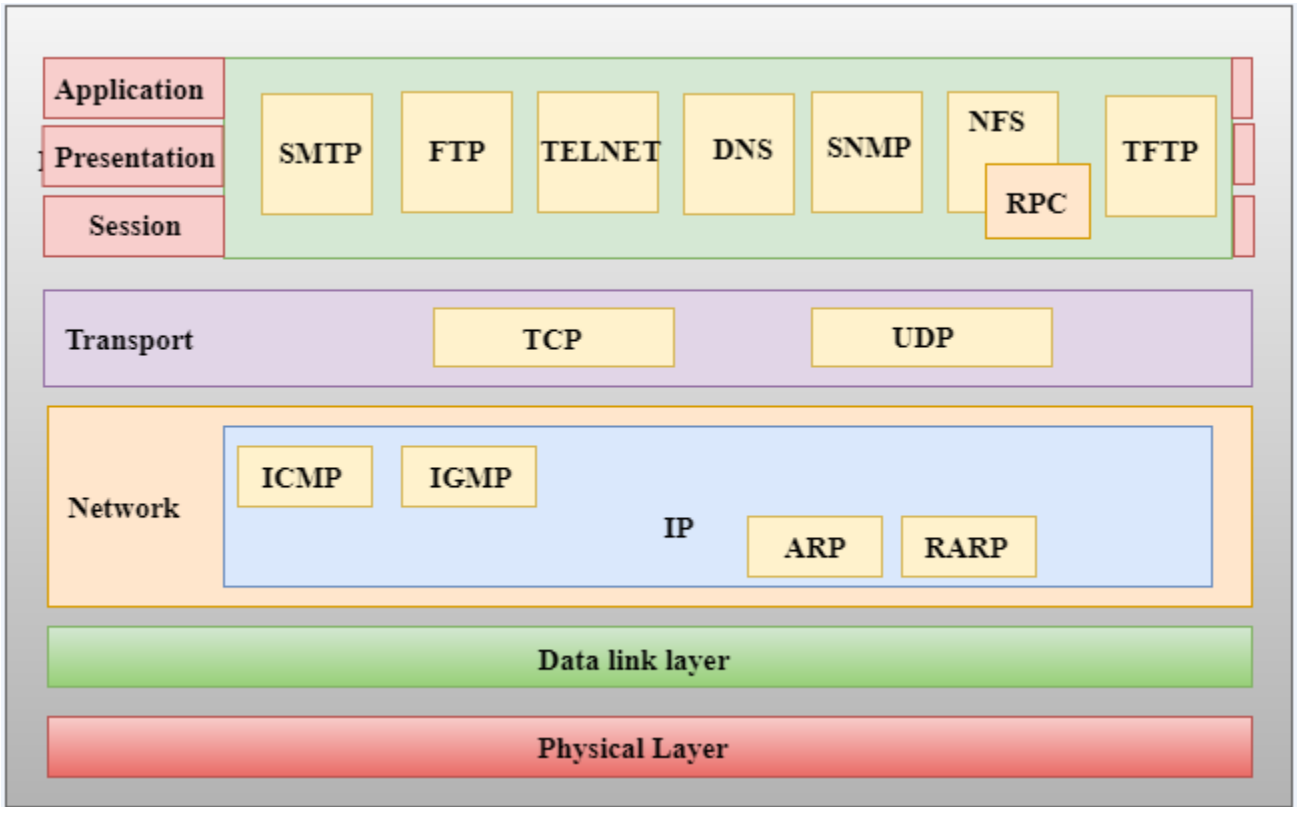
- **File transfer, access, and management (FTAM):** An application layer allows a user to access the files in a remote computer, to retrieve the files from a computer and to manage the files in a remote computer.
- **Mail services:** An application layer provides the facility for email forwarding and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

TCP/IP model

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

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

Functions of TCP/IP layers:



Network Access Layer

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

Internet Layer

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

Following are the protocols used in this layer are:

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

Following are the responsibilities of this protocol:

- **IP Addressing:** This protocol implements logical host addresses known as IP addresses. The IP addresses are used by the internet and higher layers to identify the device and to provide internetwork routing.
- **Host-to-host communication:** It determines the path through which the data is to be transmitted.
- **Data Encapsulation and Formatting:** An IP protocol accepts the data from the transport layer protocol. An IP protocol ensures that the data is sent and received securely, it encapsulates the data into message known as IP datagram.
- **Fragmentation and Reassembly:** The limit imposed on the size of the IP datagram by data link layer protocol is known as Maximum Transmission unit (MTU). If the size of IP datagram is greater than the MTU unit, then the IP protocol splits the datagram into smaller units so that they

can travel over the local network. Fragmentation can be done by the sender or intermediate router. At the receiver side, all the fragments are reassembled to form an original message.

- **Routing:** When IP datagram is sent over the same local network such as LAN, MAN, WAN, it is known as direct delivery. When source and destination are on the distant network, then the IP datagram is sent indirectly. This can be accomplished by routing the IP datagram through various devices such as routers.

ARP Protocol

Prime Ministers of India | List of Prime Minister of India (1947-2020)

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

ICMP Protocol

- **ICMP** stands for Internet Control Message Protocol.
- It is a mechanism used by the hosts or routers to send notifications regarding datagram problems back to the sender.
- A datagram travels from router-to-router until it reaches its destination. If a router is unable to route the data because of some unusual conditions such as disabled links, a device is on fire or network congestion, then the ICMP protocol is used to inform the sender that the datagram is undeliverable.
- An ICMP protocol mainly uses two terms:
  - **ICMP Test:** ICMP Test is used to test whether the destination is reachable or not.
  - **ICMP Reply:** ICMP Reply is used to check whether the destination device is responding or not.

- The core responsibility of the ICMP protocol is to report the problems, not correct them. The responsibility of the correction lies with the sender.
- ICMP can send the messages only to the source, but not to the intermediate routers because the IP datagram carries the addresses of the source and destination but not of the router that it is passed to.

Transport Layer

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

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

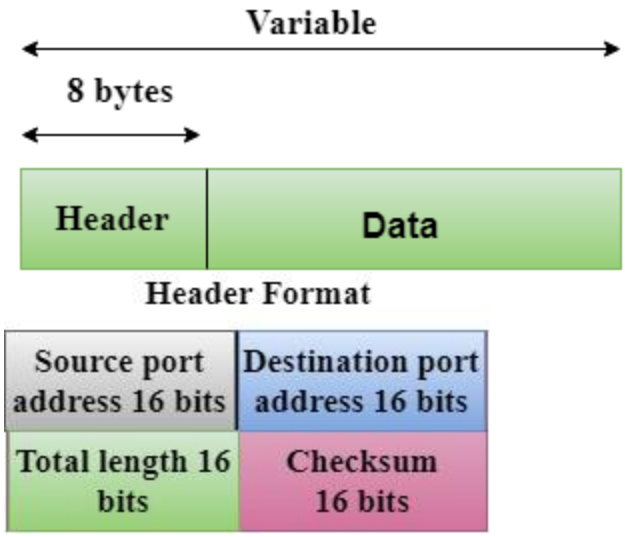
- **User Datagram Protocol (UDP)**
  - It provides connectionless service and end-to-end delivery of transmission.
  - It is an unreliable protocol as it discovers the errors but not specify the error.
  - User Datagram Protocol discovers the error, and ICMP protocol reports the error to the sender that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

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

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

    Total length: It defines the total number of bytes of the user datagram in bytes. **Checksum:** The checksum is a 16-bit field used in error detection.

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



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

Application Layer

- An application layer is the topmost layer in the TCP/IP model.
- It is responsible for handling high-level protocols, issues of representation.
- This layer allows the user to interact with the application.
- When one application layer protocol wants to communicate with another application layer, it forwards its data to the transport layer.

- There is an ambiguity occurs in the application layer. Every application cannot be placed inside the application layer except those who interact with the communication system. For example: text editor cannot be considered in application layer while web browser using **HTTP** protocol to interact with the network where **HTTP** protocol is an application layer protocol.

Following are the main protocols used in the application layer:

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

Enterprise network constructs

Think of the Enterprise Network as the internet, except that it's local to your organization.

An enterprise network helps employees and machines communicate, share files, access systems, and analyze the performance of an IT environment that drives business operations. Enterprise networks are configured to:

- Connect a limited number of authorized systems, apps, and individuals.
- Enable a secure and efficient communication channel to perform specific business operations.

In this article, we will discuss the enterprise network, how it helps the business, and industry-proven best practices to run secure, high performance, and highly dependable enterprise networking systems.

What is the enterprise network?

The term 'enterprise network' refers to the physical, virtual, or logical connectivity infrastructure that enables systems and apps to:

- Communicate
- Share information
- Run services and programs
- Analyze system performance

The enterprise network effectively comprises the infrastructure, hardware and software systems, and the communication protocols used to deliver end-to-end services. The network (or its subset) may be architected, designed, deployed, optimized, and configured to perform a unique set of business and technical objectives.
To establish an enterprise network at geographically disparate locations, use Virtual Private Networks (VPNs) to connect these regions.

Types of enterprise networks

Some of the common types of enterprise networks include:

- Local Area Networks
- Wide Area Networks
- Cloud networks

Local Area Network (LAN)

A LAN is a computer network that interconnects systems within a small building or room. Typically used for personal, non-commercial use cases, LANs can also be used as small-scale prototyping or testbed networks.

You can also establish LANs logically and virtually within a larger network. For example, each department within the enterprise network can operate a small LAN where multiple computers are connected to the same switch but decoupled from other departmental LANs.

Wide Area Network (WAN)

Think of a LAN that spans across buildings and disparate geographic locations—even globally.

WAN connectivity differs from LANs in terms of the protocols and components across the layers of the OSI model used to transmit data. While LAN technologies are used to transmit data at higher rates within close proximity, WANs are set up for communication that is:

- Long-distance
- Energy efficient
- Secure
- Dependable

WANs can be deployed as a private or public network and are usually set up by the internet service providers (ISPs).

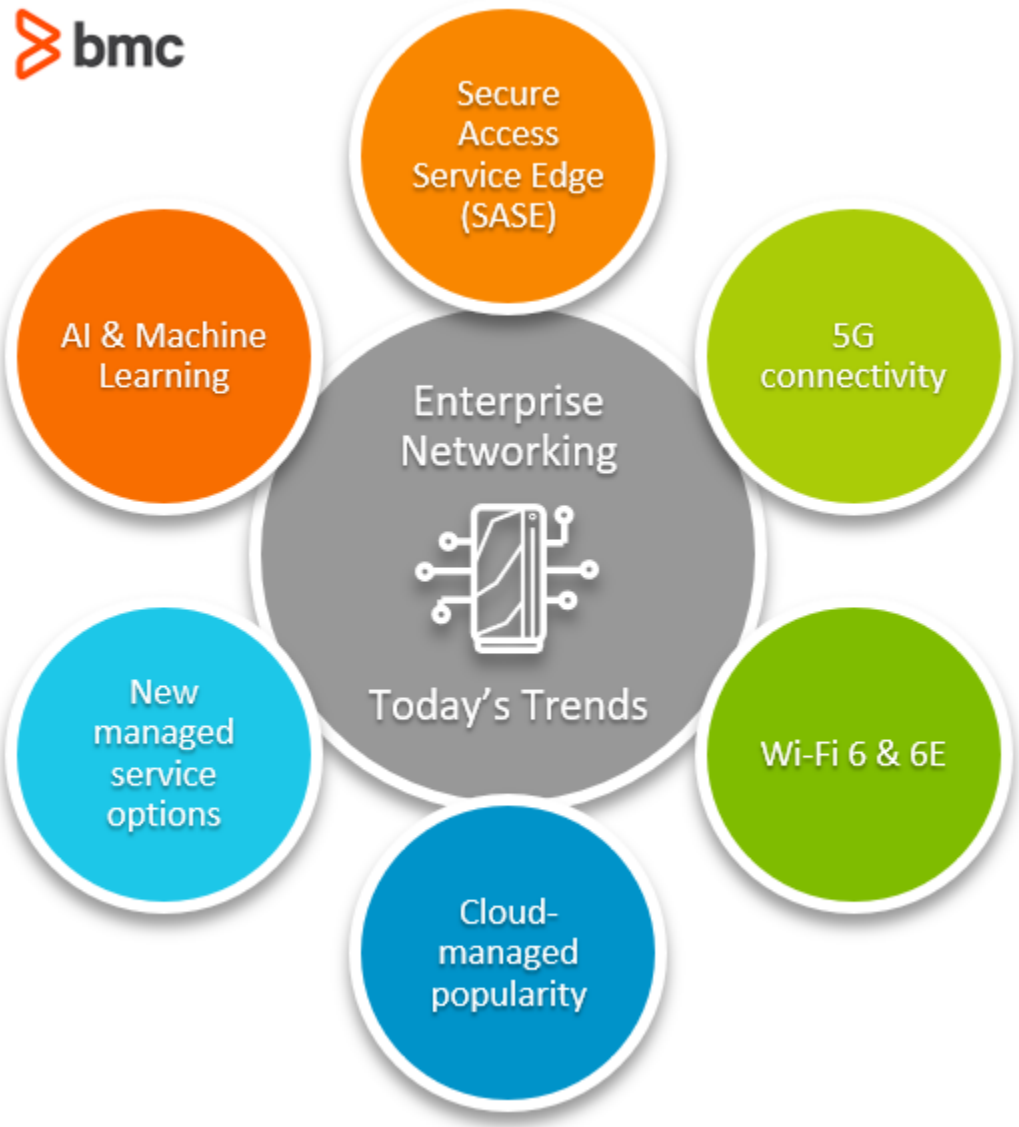
You can also have a software-defined WAN, or SD-WAN. This is a virtual WAN architecture controlled by software technologies that create an abstraction of the virtualized WAN from the underlying infrastructure components. This technology enables secure WAN operations while decoupling the performance from the underlying components.
An SD-WAN offers more flexible and dependable connectivity services that can be controlled at the application level, without sacrificing security and quality of service (QoS).

Cloud networks

Most enterprise IT services are delivered from data centers and cloud networks. The IT environment may be a hybrid mix of on-premise servers and off-site cloud networks. The cloud stack may consist of multiple cloud computing models—private, public, and hybrid cloud.
Additionally, you likely employ multi-cloud services to deliver various application components and services as an optimal tradeoff between cost, performance, and security offered by different cloud models.
The infrastructure components and software technologies enable the connectivity between data center hardware, applications. and services running across these various IT environments. The cloud resources

and the services running on the hardware are accessed and controlled over the internet, usually through private and secure network channels (unless used for public-facing applications).

Conceptually, cloud networks can be seen as a WAN (often an SD-WAN) that may comprise multiple subset of networks shared or distributed privately among customers of cloud computing services.



Enterprise networking trends & concepts

Already embarked on your enterprise networking strategy? It can be interesting to follow some of the latest trends in the enterprise networking domain.

Today's technology advancements and improvements are generally centered around service dependability, security, and readiness to integrate new technology standards and systems.

Some new innovations and trends include:

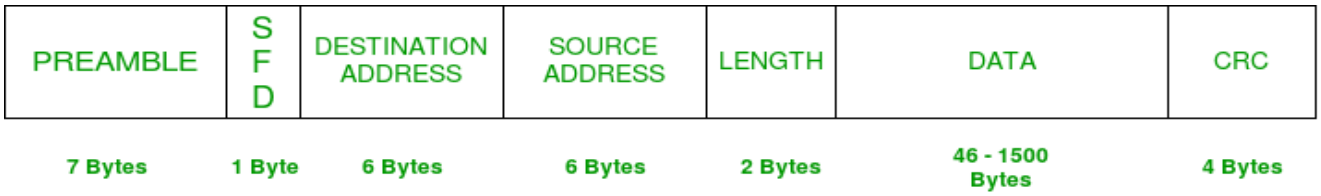
- **Secure Access Service Edge (SASE).** This network architecture introduces an additional security layer for edge network technologies.
- **5G connectivity.** With significant investments and adoption recently, the new 5G networking standard is set to reach maturity in coming years. Organizations taking advantage of the technology are early adopters and disruptors, especially since 5G connectivity offers significantly better user experience with high data transmission rates.
- **Wi-Fi 6 and 6E.** These new connectivity standards are around 30% faster than Wi-Fi 5. They're especially useful for simple in-house LAN implementations.
- **Cloud-managed popularity.** According to a recent IDC publication, cloud-managed WAN, SD-WAN, and Unified Communications adoption continues to rise.
- **Managed service options.** New service delivery models, like Networking as a Service (NaaS), enable organizations to leverage advanced enterprise networking capabilities on a subscription cost basis.
- **AI and machine learning.** AI- and ML-enabled enterprise networking will greatly enhance visibility and control into enterprise networks and the IT infrastructure that generates a vast deluge of information at every node and network endpoint.

Ethernet Frame Format

Ethernet is most widely used LAN Technology, which is defined under IEEE standards 802.3. The reason behind its wide usability is Ethernet is easy to understand, implement, maintain and allows low-cost network implementation. Also, Ethernet offers flexibility in terms of topologies which are allowed.Ethernet generally uses Bus Topology. Ethernet operates in two layers of the OSI model, Physical Layer, and Data Link Layer. For Ethernet, the protocol data unit is Frame since we mainly deal with DLL. In order to handle collision, the Access control mechanism used in Ethernet is CSMA/CD.

Basic frame format which is required for all MAC implementation is defined in **IEEE 802.3 standard**. Though several optional formats are being used to extend the protocol's basic capability. Ethernet frame starts with Preamble and SFD, both works at the physical layer. Ethernet header contains both Source and Destination MAC address, after which the payload of the frame is present. The last field is CRC which is used to detect the error. Now, let's study each field of basic frame format.

Ethernet (IEEE 802.3) Frame Format –



IEEE 802.3 ETHERNET Frame Format

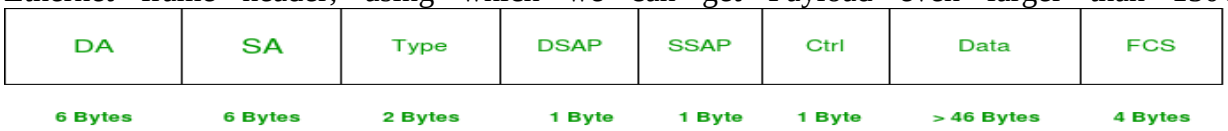
- **PREAMBLE –** Ethernet frame starts with 7-Bytes Preamble. This is a pattern of alternative 0's and 1's which indicates starting of the frame and allow sender and receiver to establish bit synchronization. Initially, PRE (Preamble) was introduced to allow for the loss of a few bits due to signal delays. But today's high-speed Ethernet don't need Preamble to protect the frame bits. PRE (Preamble) indicates the receiver that frame is coming and allow the receiver to lock onto the data stream before the actual frame begins.
- **Start of frame delimiter (SFD) –** This is a 1-Byte field which is always set to 10101011. SFD indicates that upcoming bits are starting of the frame, which is the destination address. Sometimes SFD is considered the part of PRE, this is the reason Preamble is described as 8 Bytes in many places. The SFD warns station or stations that this is the last chance for synchronization.
- **Destination Address –** This is 6-Byte field which contains the MAC address of machine for which data is destined.
- **Source Address –** This is a 6-Byte field which contains the MAC address of source machine. As Source Address is always an individual address (Unicast), the least significant bit of first byte is always 0.

- **Length –** Length is a 2-Byte field, which indicates the length of entire Ethernet frame. This 16-bit field can hold the length value between 0 to 65534, but length cannot be larger than 1500 because of some own limitations of Ethernet.
- **Data –** This is the place where actual data is inserted, also known as **Payload**. Both IP header and data will be inserted here if Internet Protocol is used over Ethernet. The maximum data present may be as long as 1500 Bytes. In case data length is less than minimum length i.e. 46 bytes, then padding 0's is added to meet the minimum possible length.
- **Cyclic Redundancy Check (CRC) –** CRC is 4 Byte field. This field contains a 32-bits hash code of data, which is generated over the Destination Address, Source Address, Length, and Data field. If the checksum computed by destination is not the same as sent checksum value, data received is corrupted.

Note – Size of frame of Ethernet IEEE 802.3 varies 64 bytes to 1518 bytes including data length (46 to 1500 bytes).

Brief overview on Extended Ethernet Frame (Ethernet II Frame) :

Standard IEEE 802.3 basic frame format is discussed above in detail. Now let's see the extended Ethernet frame header, using which we can get Payload even larger than 1500 Bytes.



Proposed ETHERNET Frame Extension

DA [Destination MAC Address] : ***6 bytes***

SA [Source MAC Address] : ***6 bytes***

Type [0x8870 (Ethertype)] : ***2 bytes***

DSAP [802.2 Destination Service Access Point] : ***1 byte***

SSAP [802.2 Source Service Access Point] : ***1 byte***

Ctrl [802.2 Control Field] : ***1 byte***

Data [Protocol Data] : ***> 46 bytes***

FCS [Frame Checksum] : ***4 bytes***

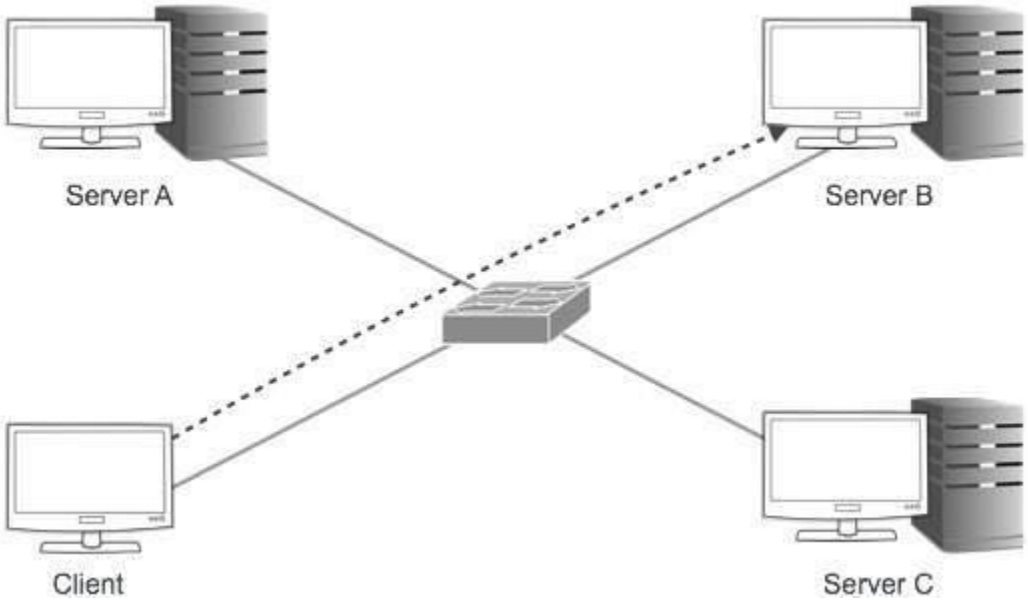
Building block	Size	Function
PreambleStart frame delimiter (SFD)	8 bytes	Synchronization of the receiversBit sequence that initiates the frame
Destination address (MAC)	6 bytes	Hardware address of the destination network adapter
Source address (MAC)	6 bytes	Hardware address of the source network adapter
Tag	4 bytes	Optional VLAN tag for integration in VLAN networks (IEEE 802.1q)
Type	2 bytes	Ethernet II: labeling of layer 3 protocols
Length	2 bytes	Length information about the record
Destination service access point (DSAP)	1 byte	Individual address of the addressed service access point
Source service access point (SSAP)	1 byte	Source address of the sending device
Control	1 byte	Defines the LLC frame (logical link)
SNAP	5 bytes	Field for the definition of the organizationally unique identifier (OUI) of the manufacturer and the protocol number (like "Type")
Data	44-1,500 bytes (limit depending on frame structure)	The data to be transmitted
Frame check sequence (FCS)	4 bytes	Checksum that computes the entire frame
Inter frame gap (IFS)	-	Transmission break of 9.6 µs

IPv4 - Addressing

IPv4 supports three different types of addressing modes. −

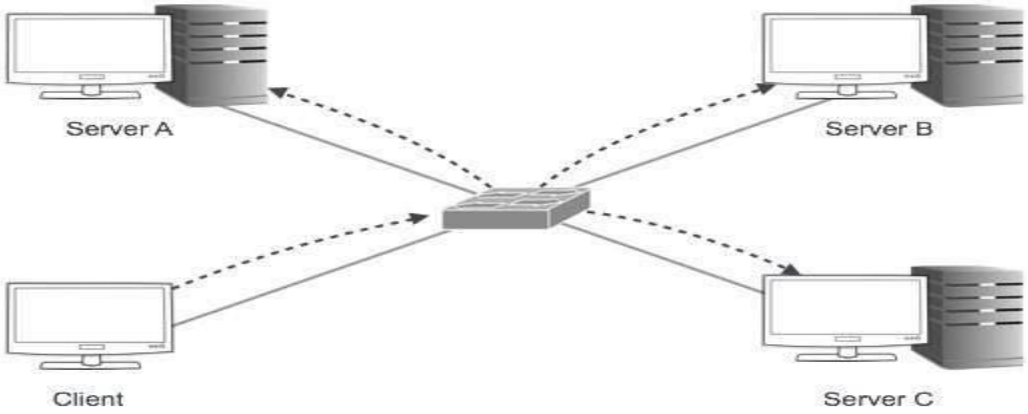
Unicast Addressing Mode

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



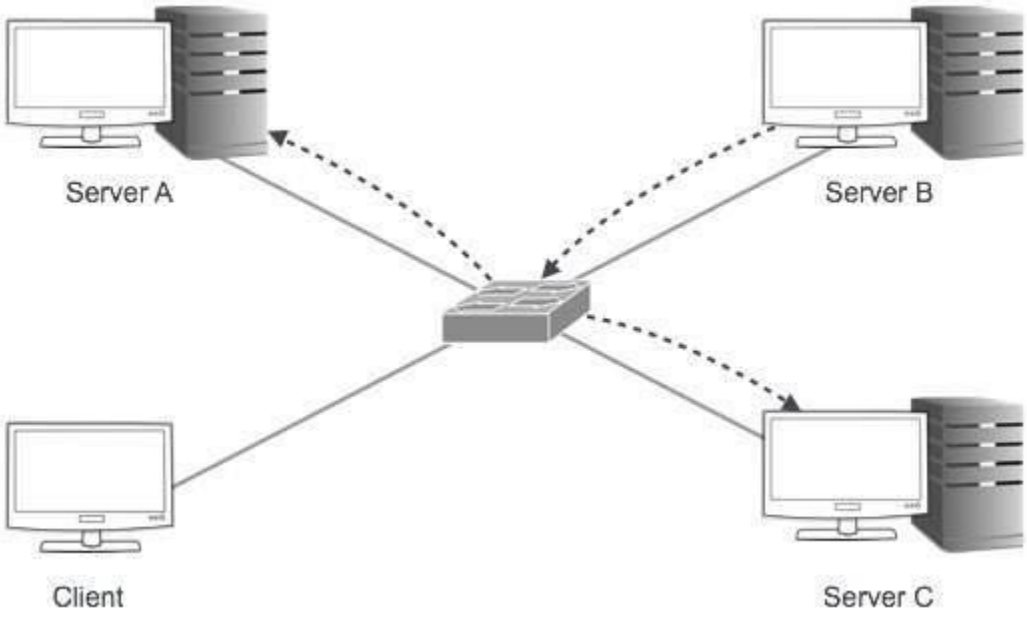
Broadcast Addressing Mode

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



Multicast Addressing Mode

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



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

Hierarchical Addressing Scheme

IPv4 uses hierarchical addressing scheme. An IP address, which is 32-bits in length, is divided into two or three parts as depicted −

A single IP address can contain information about the network and its sub-network and ultimately the host. This scheme enables the IP Address to be hierarchical where a network can have many sub-networks which in turn can have many hosts.

Subnet Mask

The 32-bit IP address contains information about the host and its network. It is very necessary to distinguish both. For this, routers use Subnet Mask, which is as long as the size of the network address in the IP address. Subnet Mask is also 32 bits long. If the IP address in binary is ANDed with its Subnet Mask, the result yields the Network address. For example, say the IP Address is 192.168.1.152 and the Subnet Mask is 255.255.255.0 then −

IP	192.168.1.152	11000000	10101000	00000001	10011000	ANDed
Mask	255.255.255.0	11111111	11111111	11111111	00000000	
Network	192.168.1.0	11000000	10101000	00000001	00000000	Result

This way the Subnet Mask helps extract the Network ID and the Host from an IP Address. It can be identified now that 192.168.1.0 is the Network number and 192.168.1.152 is the host on that network.

Binary Representation

The positional value method is the simplest form of converting binary from decimal value. IP address is 32 bit value which is divided into 4 octets. A binary octet contains 8 bits and the value of each bit can be determined by the position of bit value '1' in the octet.

MSB	8th	7th	6th	5th	4th	3rd	2nd	1st	LSB
	1	1	1	1	1	1	1	1	
Positional Value	128	64	32	16	8	4	2	1	

Positional value of bits is determined by 2 raised to power (position – 1), that is the value of a bit 1 at position 6 is 2^(6-1) that is 2^5 that is 32. The total value of the octet is determined by adding up the positional value of bits. The value of 11000000 is 128+64 = 192. Some examples are shown in the table below −

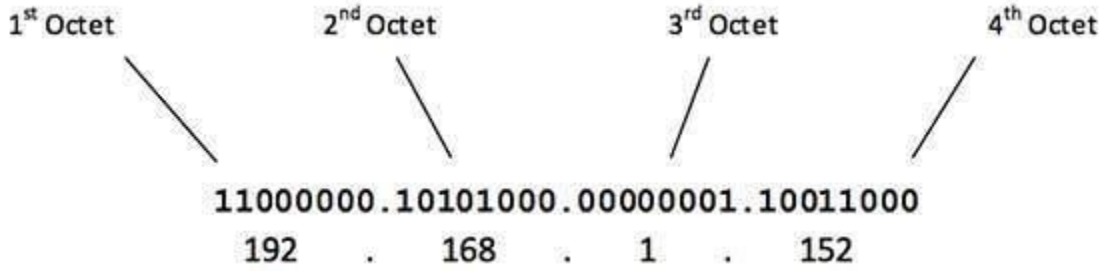
128	64	32	16	8	4	2	1	Value
0	0	0	0	0	0	0	1	1
0	0	0	0	0	0	1	0	2
0	0	0	0	0	0	1	1	3
0	0	0	0	0	1	0	0	4
0	0	0	0	0	1	0	1	5
0	0	0	0	0	1	1	0	6
0	0	0	0	0	1	1	1	7
0	0	0	0	1	0	0	0	8
0	0	0	0	1	0	0	1	9
0	0	0	0	1	0	1	0	10
0	0	0	1	0	0	0	0	16
0	0	1	0	0	0	0	0	32
0	1	0	0	0	0	0	0	64
0	1	1	0	0	1	0	0	100
0	1	1	1	1	1	1	1	127
1	0	0	0	0	0	0	0	128
1	0	1	0	1	0	0	0	168
1	1	0	0	0	0	0	0	192
1	1	1	1	1	1	1	1	255

IPv4 - Address Classes

Internet Protocol hierarchy contains several classes of IP Addresses to be used efficiently in various situations as per the requirement of hosts per network. Broadly, the IPv4 Addressing system is divided into five classes of IP Addresses. All the five classes are identified by the first octet of IP Address.

Internet Corporation for Assigned Names and Numbers is responsible for assigning IP addresses.

The first octet referred here is the left most of all. The octets numbered as follows depicting dotted decimal notation of IP Address −



The number of networks and the number of hosts per class can be derived by this formula −

Number of networks = 2^network_bits

Number of Hosts/Network = 2^host_bits – 2

When calculating hosts' IP addresses, 2 IP addresses are decreased because they cannot be assigned to hosts, i.e. the first IP of a network is network number and the last IP is reserved for Broadcast IP.

Class A Address

The first bit of the first octet is always set to 0 (zero). Thus the first octet ranges from 1 – 127, i.e.

00000001 – 01111111
1 – 127

Class A addresses only include IP starting from 1.x.x.x to 126.x.x.x only. The IP range 127.x.x.x is reserved for loopback IP addresses.

The default subnet mask for Class A IP address is 255.0.0.0 which implies that Class A addressing can have 126 networks (2^7-2) and 16777214 hosts (2^{24}-2).

Class A IP address format is thus: **0NNNNNNN**.HHHHHHHH.HHHHHHHH.HHHHHHHH

Class B Address

An IP address which belongs to class B has the first two bits in the first octet set to 10, i.e.

10000000 – 10111111
128 – 191

Class B IP Addresses range from 128.0.x.x to 191.255.x.x. The default subnet mask for Class B is 255.255.x.x.

Class B has 16384 (2^{14}) Network addresses and 65534 (2^{16}-2) Host addresses.

Class B IP address format is: **10NNNNNN.NNNNNNNN**.HHHHHHHH.HHHHHHHH

Class C Address

The first octet of Class C IP address has its first 3 bits set to 110, that is −

11000000 – 11011111
192 – 223

Class C IP addresses range from 192.0.0.x to 223.255.255.x. The default subnet mask for Class C is 255.255.255.x.

Class C gives 2097152 (2^{21}) Network addresses and 254 (2^{8}-2) Host addresses.

Class C IP address format is: **110NNNNN.NNNNNNNN.NNNNNNNN**.HHHHHHHH

Class D Address

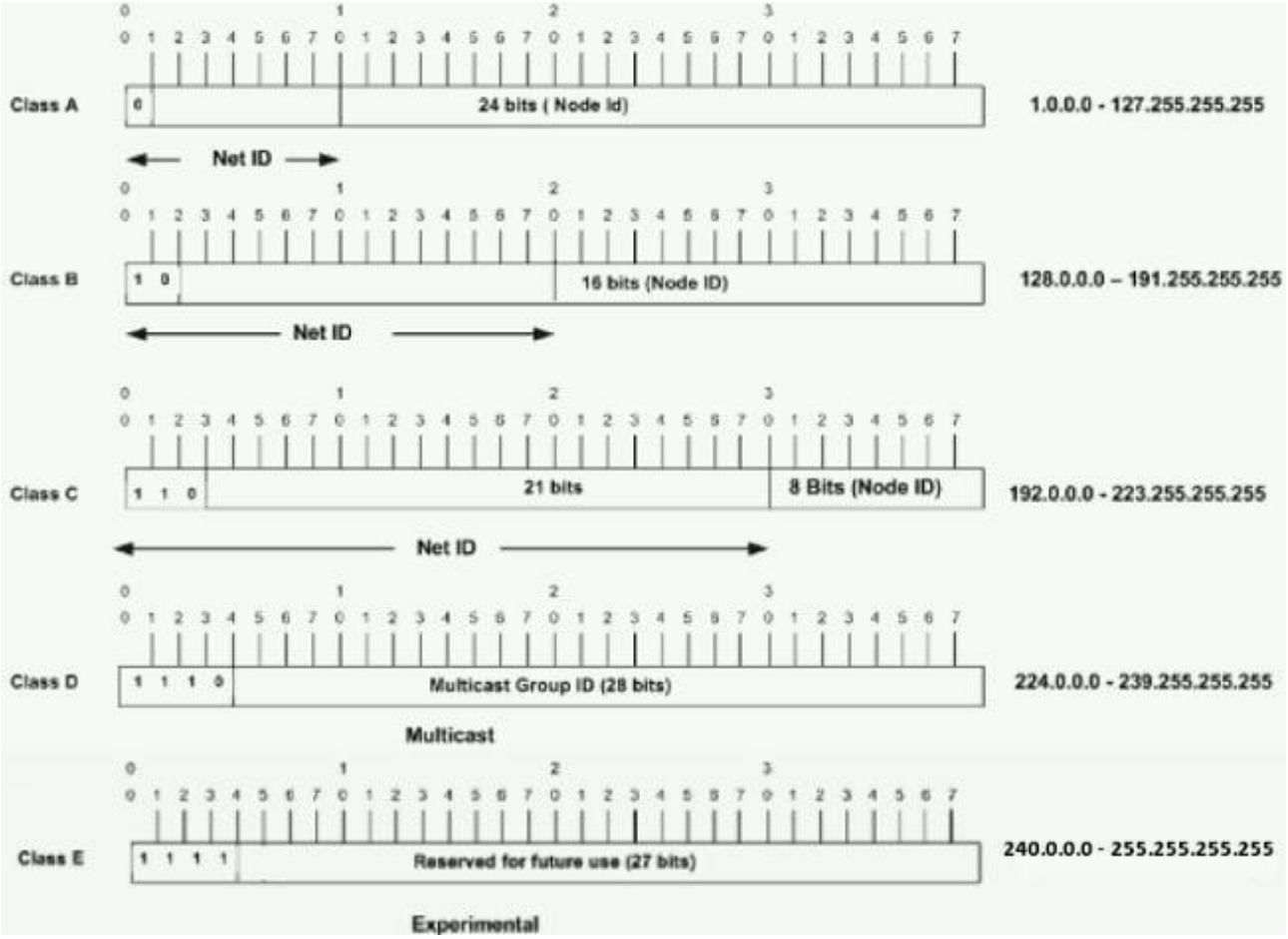
Very first four bits of the first octet in Class D IP addresses are set to 1110, giving a range of −

11100000 – 11101111
224 – 239

Class D has IP address range from 224.0.0.0 to 239.255.255.255. Class D is reserved for Multicasting. In multicasting data is not destined for a particular host, that is why there is no need to extract host address from the IP address, and Class D does not have any subnet mask.

Class E Address

This IP Class is reserved for experimental purposes only for R&D or Study. IP addresses in this class ranges from 240.0.0.0 to 255.255.255.254. Like Class D, this class too is not equipped with any subnet mask.


Class A
0
24 bits (Node Id)
1.0.0.0 - 127.255.255.255
Net ID
Class B
1 0
16 bits (Node ID)
128.0.0.0 – 191.255.255.255
Net ID
Class C
1 1 0
21 bits
8 Bits (Node ID)
192.0.0.0 - 223.255.255.255
Net ID
Class D
1 1 1 0
Multicast Group ID (28 bits)
224.0.0.0 - 239.255.255.255
Multicast
Class E
1 1 1 1
Reserved for future use (27 bits)
240.0.0.0 - 255.255.255.255
Experimental

IP

IP stands for **Internet Protocol** and **v4** stands for **Version Four** (IPv4). IPv4 was the primary version brought into action for production within the ARPANET in 1983. IP version four addresses are 32-bit integers which will be expressed in decimal notation. Example- 192.0.2.126 could be an IPv4 address.

Parts of IPv4

- **Network part:** The network part indicates the distinctive variety that's appointed to the network. The network part conjointly identifies the category of the network that's assigned.
- **Host Part:** The host part uniquely identifies the machine on your network. This part of the IPv4 address is assigned to every host. For each host on the network, the network part is the same, however, the host half must vary.
- **Subnet number:** This is the nonobligatory part of IPv4. Local networks that have massive numbers of hosts are divided into subnets and subnet numbers are appointed to that.

Characteristics of IPv4

- IPv4 could be a 32-Bit IP Address.
- IPv4 could be a numeric address, and its bits are separated by a dot.
- The number of header fields is twelve and the length of the header field is twenty.
- It has Unicast, broadcast, and multicast style of addresses.
- IPv4 supports VLSM (Virtual Length Subnet Mask).
- IPv4 uses the Post Address Resolution Protocol to map to the MAC address.
- RIP may be a routing protocol supported by the routed daemon.
- Networks ought to be designed either manually or with DHCP.
- Packet fragmentation permits from routers and causing host.

Advantages of IPv4

- IPv4 security permits encryption to keep up privacy and security.
- IPV4 network allocation is significant and presently has quite 85000 practical routers.
- It becomes easy to attach multiple devices across an outsized network while not NAT.
- This is a model of communication so provides quality service also as economical knowledge transfer.
- IPV4 addresses are redefined and permit flawless encoding.
- Routing is a lot of scalable and economical as a result of addressing is collective more effectively.
- Data communication across the network becomes a lot of specific in multicast organizations.
    - Limits net growth for existing users and hinders the use of the net for brand new users.
    - Internet Routing is inefficient in IPv4.
    - IPv4 has high System Management prices and it's labor-intensive, complex, slow & frequent to errors.
    - Security features are nonobligatory.
    - Difficult to feature support for future desires as a result of adding it on is extremely high overhead since it hinders the flexibility to attach everything over IP

Internet Control Message Protocol (ICMP)

The ICMP stands for Internet Control Message Protocol. It is a network layer protocol. It is used for error handling in the network layer, and it is primarily used on network devices such as routers. As different types of errors can exist in the network layer, so ICMP can be used to report these errors and to debug those errors.

For example, some sender wants to send the message to some destination, but the router couldn't send the message to the destination. In this case, the router sends the message to the sender that I could not send the message to that destination.

The IP protocol does not have any error-reporting or error-correcting mechanism, so it uses a message to convey the information. For example, if someone sends the message to the destination, the message is somehow stolen between the sender and the destination. If no one reports the error, then the sender might think that the message has reached the destination. If someone in-between reports the error, then the sender will resend the message very quickly.

Position of ICMP in the network layer

The ICMP resides in the IP layer, as shown in the below diagram.

The ICMP messages are usually divided into two categories:

ICMP messages

Category	Type	Message
Error-reporting messages	3	Destination unreachable
	4	Source quench
	11	Time exceeded
	12	Parameter problem
	5	Redirection
Query messages	8 or 0	Echo request or reply
	13 or 14	Timestamp request or reply

- **Error-reporting messages**

The error-reporting message means that the router encounters a problem when it processes an IP packet then it reports a message.

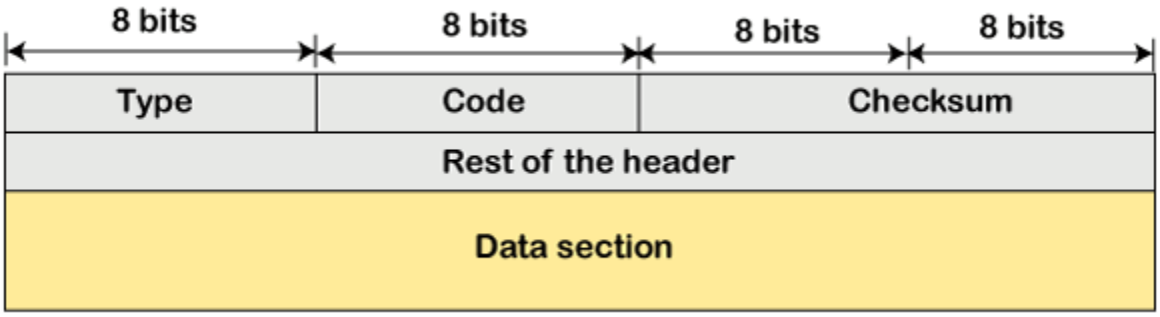
- **Query messages**

The query messages are those messages that help the host to get the specific information of another host. For example, suppose there are a client and a server, and the client wants to know whether the server is live or not, then it sends the ICMP message to the server.

ICMP Message Format

The message format has two things; one is a category that tells us which type of message it is. If the message is of error type, the error message contains the type and the code. The type defines the type of message while the code defines the subtype of the message.

The ICMP message contains the following fields:

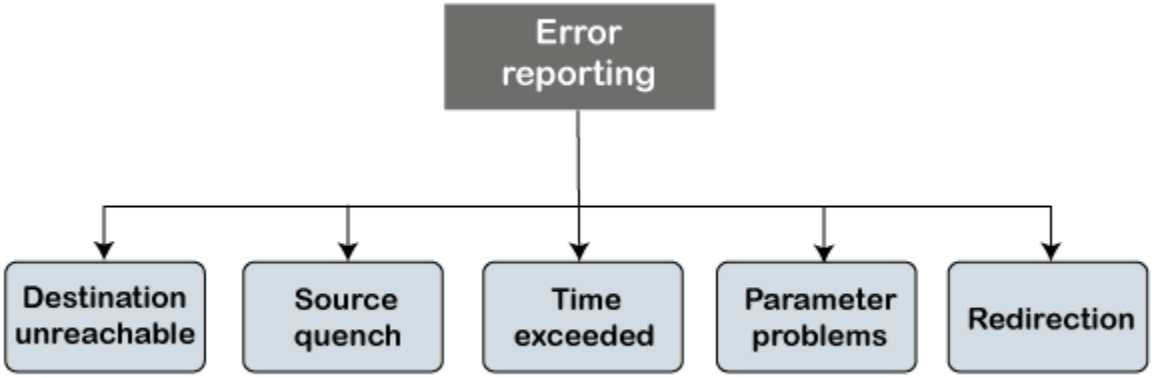


- **Type:** It is an 8-bit field. It defines the ICMP message type. The values range from 0 to 127 are defined for ICMPv6, and the values from 128 to 255 are the informational messages.
- **Code:** It is an 8-bit field that defines the subtype of the ICMP message
- **Checksum:** It is a 16-bit field to detect whether the error exists in the message or not.

Note: The ICMP protocol always reports the error messages to the original source. For example, when the sender sends the message, if any error occurs in the message then the router reports to the sender rather than the receiver as the sender is sending the message.

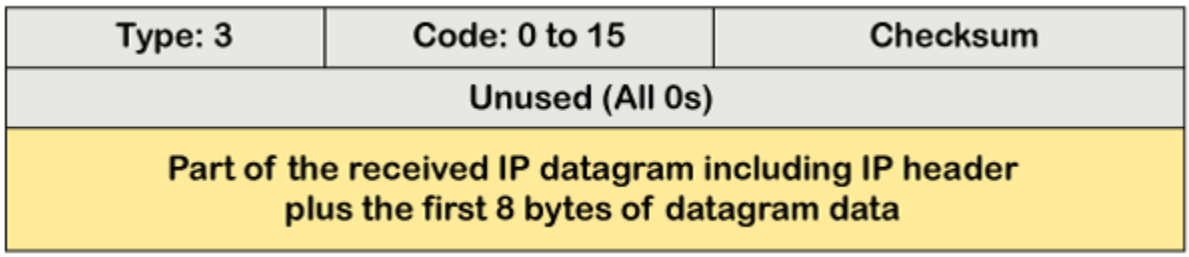
Types of Error Reporting messages

The error reporting messages are broadly classified into the following categories:



- **Destination unreachable**

The destination unreachable error occurs when the packet does not reach the destination. Suppose the sender sends the message, but the message does not reach the destination, then the intermediate router reports to the sender that the destination is unreachable.



The above diagram shows the message format of the destination unreachable message. In the message format:

Type: It defines the type of message. The number 3 specifies that the destination is unreachable.

Code (0 to 15): It is a 4-bit number which identifies whether the message comes from some intermediate router or the destination itself.

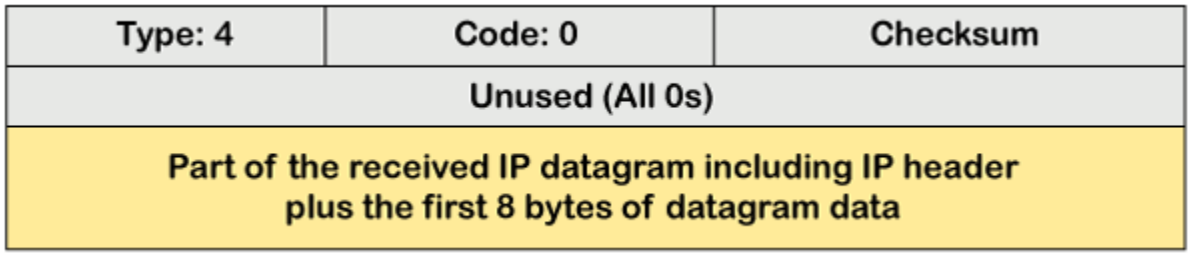
Note: If the destination creates the destination unreachable message then the code could be either 2 or 3.

Sometimes the destination does not want to process the request, so it sends the destination unreachable message to the source. A router does not detect all the problems that prevent the delivery of a packet.

- **Source quench**

There is no flow control or congestion control mechanism in the network layer or the IP protocol. The sender is concerned with only sending the packets, and the sender does not think whether the receiver is

ready to receive those packets or is there any congestion occurs in the network layer so that the sender can send a lesser number of packets, so there is no flow control or congestion control mechanism. In this case, ICMP provides feedback, i.e., source quench. Suppose the sender resends the packet at a higher rate, and the router is not able to handle the high data rate. To overcome such a situation, the router sends a source quench message to tell the sender to send the packet at a lower rate.



The above diagram shows the message format of the source quench message. It is a type 4 message, and code is zero.

Note: A source quench message informs the sender that the datagram has been discarded due to the congestion occurs in the network layer.

So, the sender must either stop or slow down the sending of datagrams until the congestion is reduced. The router sends one source-quench message for each datagram that is discarded due to the congestion in the network layer.

- **Time exceeded**

Sometimes the situation arises when there are many routers that exist between the sender and the receiver. When the sender sends the packet, then it moves in a routing loop. The time exceeded is based on the time-to-live value. When the packet traverses through the router, then each router decreases the value of TTL by one. Whenever a router decreases a datagram with a time-to-live value to zero, then the router discards a datagram and sends the time exceeded message to the original source.

Each of the MAC layers has different data units. For example, some layers can handle upto 1500 data units, and some can handle upto 300 units. When the packet is sent from a layer having 1500 units to the layer having 300 units, then the packet is divided into fragments; this process is known as fragmentation. These 1500 units are divided into 5 fragments, i.e., f1, f2, f3, f4, f5, and these fragments reach the destination in a sequence. If all the fragments are not reached to the destination in a set time, they discard all the received fragments and send a time-exceeded message to the original source.

In the case of fragmentation, the code will be different as compared to TTL. Let's observe the message format of time exceeded.

Type: 11	Code: 0 or 1	Checksum
Unused (All 0s)		
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

The above message format shows that the type of time-exceeded is 11, and the code can be either 0 or 1. The code 0 represents TTL, while code 1 represents fragmentation. In a time-exceeded message, the code 0 is used by the routers to show that the time-to-live value is reached to zero.

The code 1 is used by the destination to show that all the fragments do not reach within a set time.

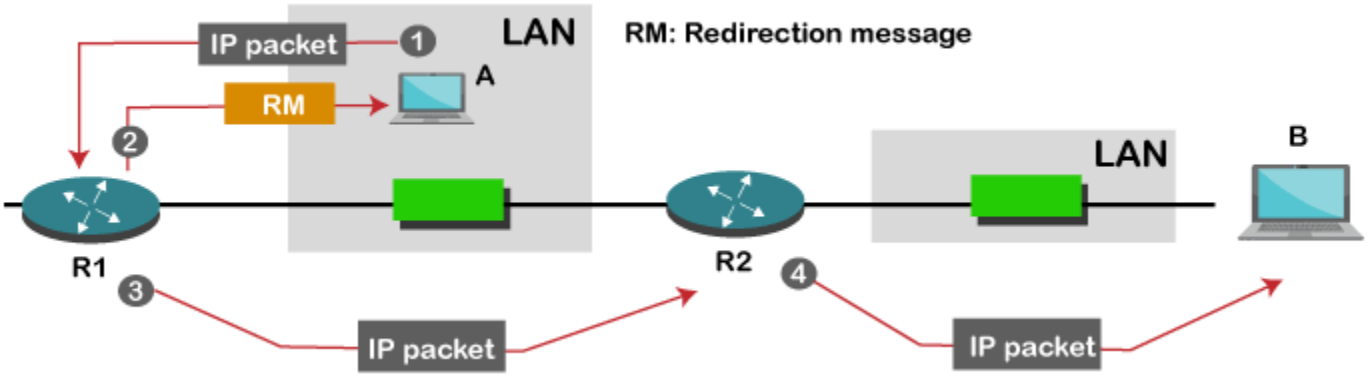
Parameter problems

The router and the destination host can send a parameter problem message. This message conveys that some parameters are not properly set.

Type: 12	Code: 0 or 1	Checksum
Pointer	Unused (All 0s)	
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

The above diagram shows the message format of the parameter problem. The type of message is 12, and the code can be 0 or 1.

Redirection



When the packet is sent, then the routing table is gradually augmented and updated. The tool used to achieve this is the redirection message. For example, A wants to send the packet to B, and there are two routers exist between A and B. First, A sends the data to the router 1. The router 1 sends the IP packet to router 2 and redirection message to A so that A can update its routing table.

Note: A redirection message is sent from the router to the host on the same network.

ICMP Query Messages

The ICMP Query message is used for error handling or debugging the internet. This message is commonly used to ping a message.

Echo-request and echo-reply message

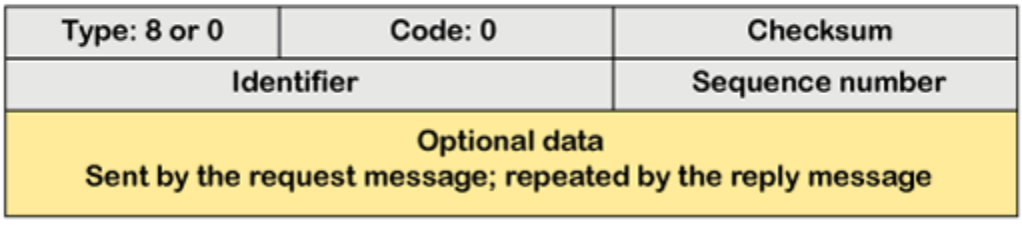
A router or a host can send an echo-request message. It is used to ping a message to another host that "Are you alive". If the other host is alive, then it sends the echo-reply message. An echo-reply message is sent by the router or the host that receives an echo-request message.

Key points of Query messages

1. The echo-request message and echo-reply message can be used by the network managers to check the operation of the IP protocol. Suppose two hosts, i.e., A and B, exist, and A wants to communicate with host B. The A host can communicate to host B if the link is not broken between A and B, and B is still alive.
2. The echo-request message and echo-reply message check the host's reachability, and it can be done by invoking the ping command.

The message format of echo-request and echo-reply message

Type 8: Echo request
Type 0: Echo reply



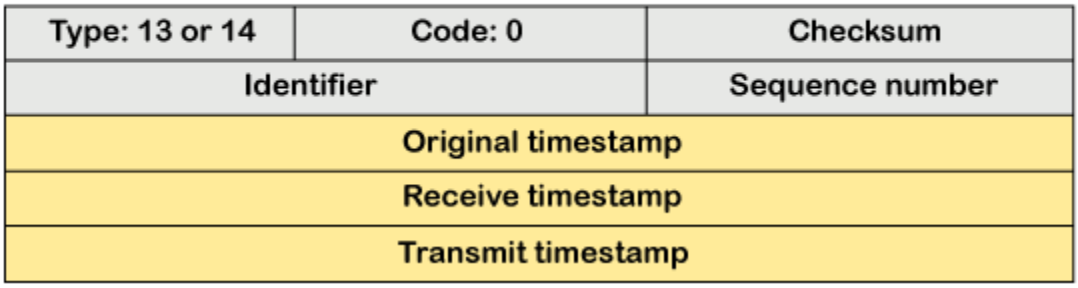
The above diagram shows the message format of the echo-request and echo-reply message. The type of echo-request is 8, and the request of echo-reply is 0. The code of this message is 0.

Timestamp-request and timestamp-reply message

The timestamp-request and timestamp-reply messages are also a type of query messages. Suppose the computer A wants to know the time on computer B, so it sends the timestamp-request message to computer B. The computer B responds with a timestamp-reply message.

Message format of timestamp-request and timestamp-reply

Type 13: request
Type 14: reply



The type of timestamp-request is 13, and the type of timestamp-reply is 14. The code of this type of message is 0.

Key points related to timestamp-request and timestamp-reply message

- It can be used to calculate the round-trip time between the source and the destination, even if the clocks are not synchronized.
- It can also be used to synchronize the clocks in two different machines if the exact transit time is known.

If the sender knows the exact transit time, then it can synchronize the clock. The sender asks the time on the receiver's clock, and then it adds the time and propagation delay. Suppose the time is 1:00 clock and propagation delay is 100 ms, then time would be 1:00 clock plus 100 ms.

Debugging tools

There are several tools used for debugging. In this topic, we will learn two tools that use ICMP for debugging. The two tools are **ping** and **traceroute**. We have learned about ping in echo-request and echo-reply messages that check whether the host or a router is alive or running.

Now we will take a look at the traceroute.

Traceroute is a tool that tracks the route taken by a packet on an IP network from source to destination. It records the time taken by the packet on each hop during its route from source to destination. Traceroute uses ICMP messages and TTL values. The TTL value is calculated; if the TTL value reaches zero, the packet gets discarded. Traceroute uses small TTL values as they get quickly expired. If the

TTL value is 1 then the message is produced by router 1; if the TTL value is 2 then the message is produced by router 2, and so on.

Let's understand the traceroute through an example.

Suppose A and B are two different hosts, and A wants to send the packet to the host B. Between A and B, 3 routers exist. To determine the location of the routers, we use the traceroute tool.

TTL value =1: First, host A sends the packet to router 1 with TTL value 1, and when the packet reaches to router 1 then router reduces the value of TTL by one and TTL values becomes 0. In this case, router 1 generates the time-exceeded message and host A gets to know that router 1 is the first router in a path.

TTL value=2: When host A sends the packet to router 1 with TTL value 2, and when the packet reaches to router 1 then the TTL value gets decremented by 1 and the TTL value becomes 1. Then router 1 sends the packet to router 2, and the TTL value becomes 0, so the router generates a time-exceeded message. The host A gets to know that router 2 is the second router on the path.

TTL value=3: When host A sends the packet to router 1 with TTL value 3, then the router decrements its value by one, and the TTL value becomes 2. Then, router 1 sends the packet to router 2, and the TTL value becomes 1. Then, router 2 sends the packet to router 3, and the TTL value becomes 0. As TTL value becomes 0, router 3 generates a time-exceeded message. In this way, host A is the third router on a path.

Address Resolution Protocol (ARP)

Address Resolution Protocol (ARP) is a communication protocol used to find the MAC (Media Access Control) address of a device from its IP address. This protocol is used when a device wants to communicate with another device on a Local Area 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 that is in a different network from the sender. The Proxy ARP configured router responds to

the ARP and map the MAC address of 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 necessary information.

Example - If Host A wants to transmit data to Host B, which is on the different network, then Host A sends an ARP request message to receive a MAC address for Host B. The router responds to Host A with its own MAC address pretend itself as a destination. When the data is transmitted to the destination by Host A, it will send to the gateway so that it sends to Host 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 is a broadcast 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 no ARP responses are received, so all other nodes cannot use the IP address allocated to that switch or router. Yet if a router or switch sends an ARP request for its IP address and receives an ARP response, another node uses the IP address allocated to the switch 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 request its IPv4 address from the ARP gateway router table. A table is created by the network administrator in the gateway-router that 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 the IP address of the device. The device sends a RARP broadcast packet, including its own MAC address in the address field of both the sender and the receiver hardware. A host installed inside of the local network called the RARP-server is prepared to respond to such type of broadcast packet. The RARP server is then trying to locate a mapping table entry in the IP to MAC address. If any entry matches the item in the table, then the RARP server sends the response packet along with the IP address to 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 from the data link layer addresses. These are mainly used for the frame relays, and ATM networks, where Layer 2 virtual circuit addressing are often acquired from Layer 2 signaling. When using these virtual circuits, the relevant Layer 3 addresses are available.

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

What Does ARP Do and How Does It Work?

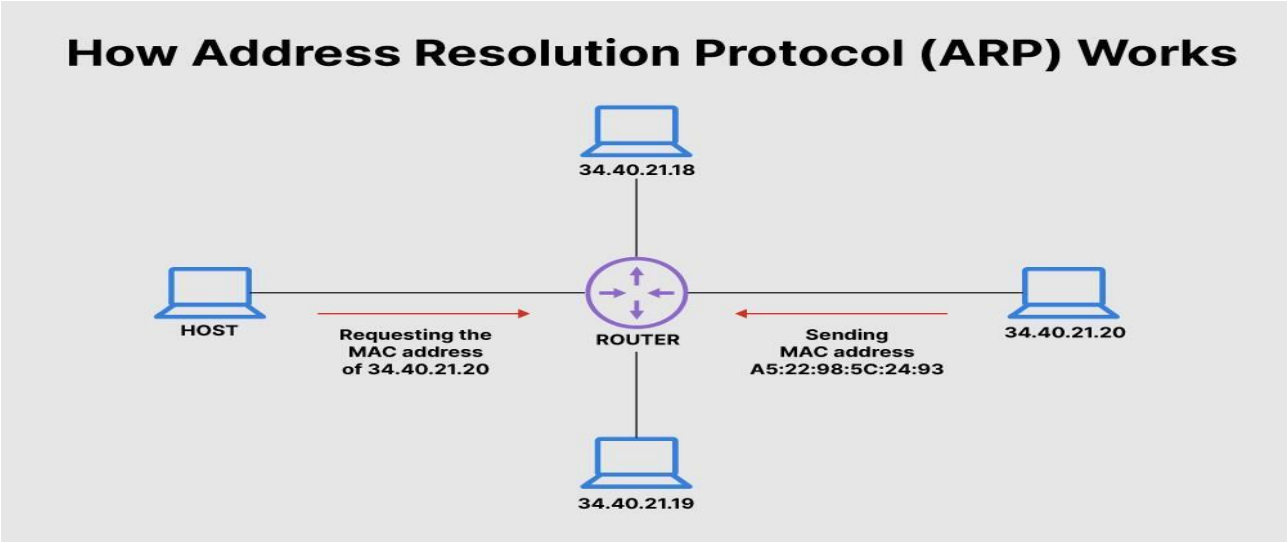
When a new computer joins a local area network (LAN), it will receive a unique IP address to use for identification and communication.

Packets of data arrive at a gateway, destined for a particular host machine. The gateway, or the piece of hardware on a network that allows data to flow from one network to another, asks the ARP program to find a MAC address that matches the IP address. The ARP cache keeps a list of each IP address and its matching MAC address. The ARP cache is dynamic, but users on a network can also configure a static ARP table containing IP addresses and MAC addresses.

ARP caches are kept on all operating systems in an IPv4 Ethernet network. Every time a device requests a MAC address to send data to another device connected to the LAN, the device verifies its ARP cache to see if the IP-to-MAC-address connection has already been completed. If it exists, then a new request is unnecessary. However, if the translation has not yet been carried out, then the request for network addresses is sent, and ARP is performed.

An ARP cache size is limited by design, and addresses tend to stay in the cache for only a few minutes. It is purged regularly to free up space. This design is also intended for privacy and security to prevent IP addresses from being stolen or spoofed by cyberattackers. While MAC addresses are fixed, IP addresses are constantly updated.

In the purging process, unutilized addresses are deleted; so is any data related to unsuccessful attempts to communicate with computers not connected to the network or that are not even powered on.



Transport Layer Protocols:

Transport Layer responsibilities

Transport Layer is the second layer of the TCP/IP model. It is an **end-to-end** layer used to deliver messages to a host. It is termed as an end-to-end layer because it provides a point-to-point connection **rather than** hop-to- hop, between the source host and destination host to deliver the services reliably. The unit of data encapsulation in Transport Layer is a segment.

The standard protocols used by Transport Layer to enhance its functionalities are TCP(Transmission Control Protocol), UDP(User Datagram Protocol), DCCP(Datagram Congestion Control Protocol) etc.

Various responsibilities of a Transport Layer –

- **Process to process delivery –**
  While Data Link Layer requires the MAC address (48 bits address contained inside the Network Interface Card of every host machine) of source-destination hosts to correctly deliver a frame and Network layer requires the IP address for appropriate routing of packets , in a similar way Transport Layer requires a Port number to correctly deliver the segments of data to the correct process amongst the multiple processes running on a particular host. A **port number** is a 16 bit address used to identify any client-server program uniquely.
- **End-to-end Connection between hosts –**
  The transport layer is also responsible for creating the end-to-end Connection between hosts for which it mainly uses TCP and UDP. TCP is a secure, connection- orientated protocol which uses a handshake protocol to establish a robust connection between two end- hosts. TCP ensures reliable delivery of messages and is used in various applications. UDP, on the other hand, is a stateless and unreliable protocol which ensures best-effort delivery. It is suitable for the applications which have little concern with flow or error control and requires to send the bulk of data like video conferencing. It is often used in multicasting protocols.
- **Multiplexing and Demultiplexing –**
  Multiplexing allows simultaneous use of different applications over a network which is running on a host. The transport layer provides this mechanism which enables us to send packet streams from various applications simultaneously over a network. The transport layer accepts these packets from different processes differentiated by their port numbers and passes them to the network layer after adding proper headers. Similarly, Demultiplexing is required at the receiver side to obtain the data coming from various processes. Transport receives the segments of data from the network layer and delivers it to the appropriate process running on the receiver's machine.
- **Congestion Control –**
  Congestion is a situation in which too many sources over a network attempt to send data and the router buffers start overflowing due to which loss of packets occur. As a result retransmission of packets from the sources increases the congestion further. In this situation, the Transport layer provides Congestion Control in different ways. It uses **open loop** congestion control to prevent the congestion and **closed loop** congestion control to remove the congestion in a network once it occurred. TCP provides AIMD- additive increase multiplicative decrease, leaky bucket technique for congestion control.
- **Data integrity and Error correction –**
  Transport layer checks for errors in the messages coming from application layer by using error detection codes, computing checksums, it checks whether the received data is not corrupted and

uses the ACK and NACK services to inform the sender if the data has arrived or not and checks for the integrity of data.

- **Flow control –**
  The transport layer provides a flow control mechanism between the adjacent layers of the TCP/IP model. TCP also prevents data loss due to a fast sender and slow receiver by imposing some flow control techniques. It uses the method of sliding window protocol which is accomplished by the receiver by sending a window back to the sender informing the size of data it can receive.

The transport layer is part of the TCP/IP networking model, sometimes called the **networking architecture**. It contains a comprehensive set of documents that describes everything required for a computer network to function.

The transport layer is responsible for the logical communication between applications running on different hosts, thus providing services to application layer protocols on a higher layer of the TCP/IP network model.

Even though many transport layer protocols exist, the two most commonly used protocols are the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP).

These protocols provide different functionalities for different application requirements.

A few of the most important functionalities are:

- Tracking of individual conversation.
- Ordered data transfer and data segmentation.
- Multiplexing conversation using port numbers.

Tracking of individual conversation

Data flowing from one application to another is known as a conversation.

A host can have multiple applications communicating with each other, either within a local network or a remote network. The transport layer has a mechanism that makes it possible for each application on a host, to communicate with another application on a different host, either within a local network or a remote network.

According to Cisco, this mechanism assigns an identifier called a **port number** to each application, such that each software process that needs to access a particular network has an unique identifier.

Ordered data transfer

A continuous stream of bytes is broken down into segments for transmission and delivery by the transport layer services.

According to this article, most networks have a limitation on the amount of data that a single packet can contain. Because of this, the sending device transport layer prepares the data into **segments**.

Similarly, the receiving device transport layer receives these segments and uses the header to reconstruct them into complete data.

Multiplexing conversation using port numbers

When using an application, the data or services provided usually appear as a stream of continuous data.

But sending data (e.g., video) across a network as a complete stream can consume all of the available network bandwidth. This prevent other services such as an email from using the medium and makes error recovery and retransmission of damage data more difficult.

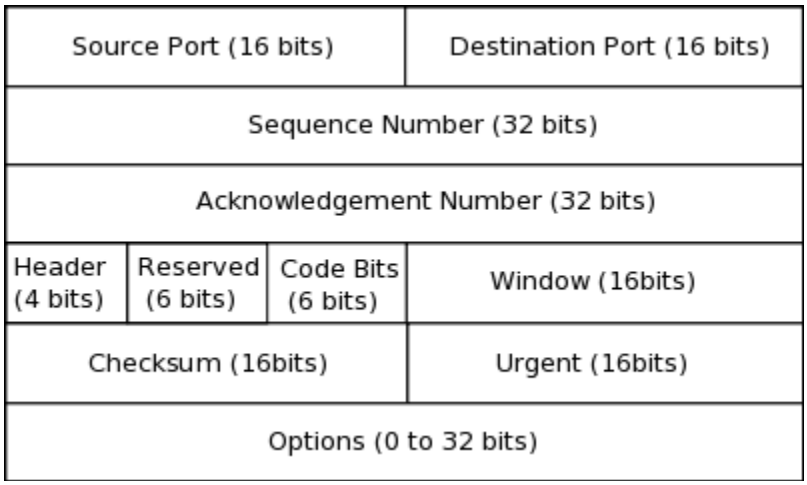
The multiplexing mechanism segments TCP and UDP data into small chunks to enable communication from different users to interleave on the same network. This mechanism relies on a concept known as a **socket.**

Transmission Control Protocol

According to this article, Transmission Control Protocol (TCP) can be defined as a standard that defines how to establish and maintain a network conversation through which application programs can exchange data.

The type of transport layer protocol an application chooses to use depends on the application requirement.

TCP is analogous to sending a package with a tracker that tracks the package from its source to its destination.



As defined in Request For Comment (RFC) 7913, TCP has the following features:

- Connection establishment and termination.
- Multiplexing using ports.
- Flow control using windowing.
- Error recovery.
- Ordered data transfer and data segmentation.

Connection establishment and termination

Before any TCP feature can occur, TCP connection establishment must take place first, because TCP is a connection-oriented protocol.

A connection-oriented protocol is a protocol that establishes a permanent connection between client and server before the transfer of data can begin.

During this connection establishment, a device negotiates the amount of traffic to be forwarded during the three-way handshake, which must be completed before data transfer can begin.

A three-way handshake is established using two flags:

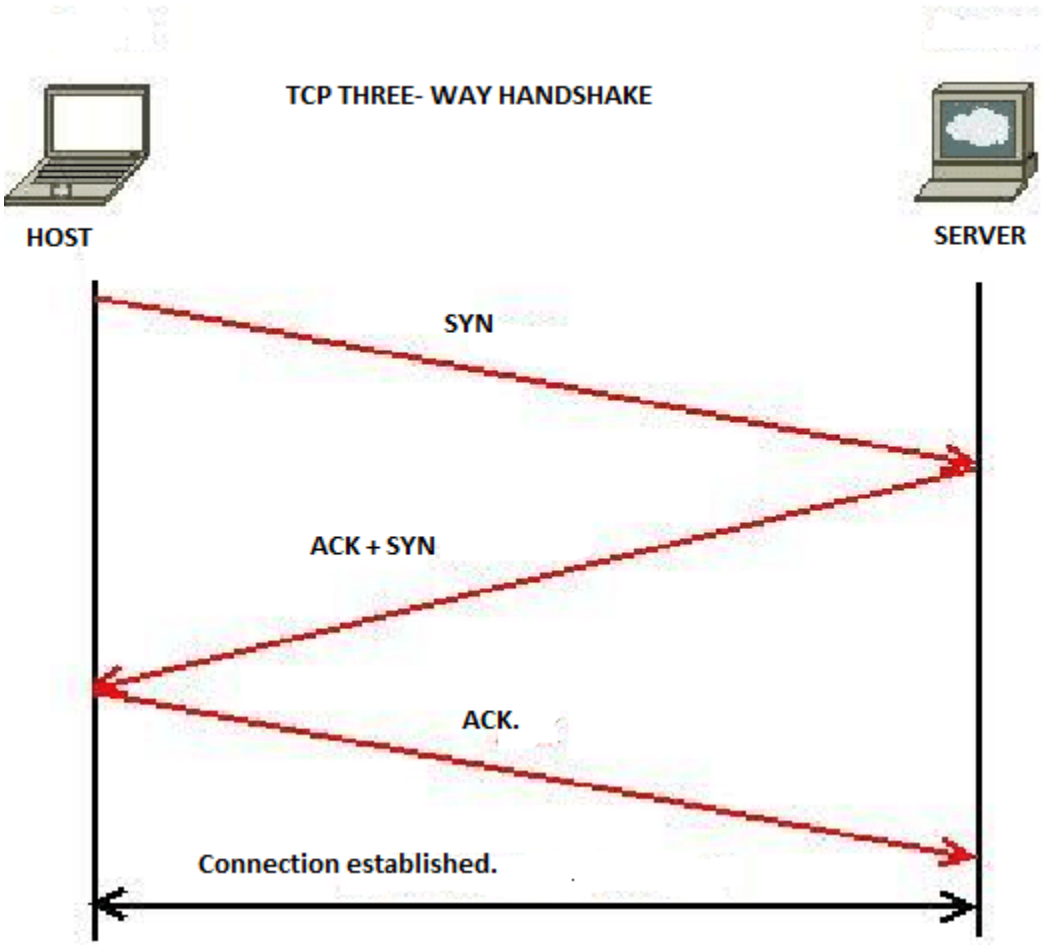
1. **Synchronization (SYN) flag**
2. **Acknowledge (ACK) flag**

The SYN flag is used in the first step of the connection establishment between the two hosts. This flag is found only in the first packet from the server and the host containing a synchronizing sequence number.

The ACK flag is used to acknowledge packets that are successfully received by a device.

For example, to create a three-way handshake between a server and a host, the host sends a SYN flag to a server providing all the necessary information such as its port number (source port) and destination port number (signifying which services it wants access to).

When the server receives the SYN flag from the host, it sends back another SYN and an ACK flag. This contains a source port number (the port number used as the destination port number on the SYN flag sent by the host) and a destination port number (the port number that the host used as source port number). The host acknowledges those flags' reception with an ACK flag, and a connection is established, thus forming a **three-way handshake.**

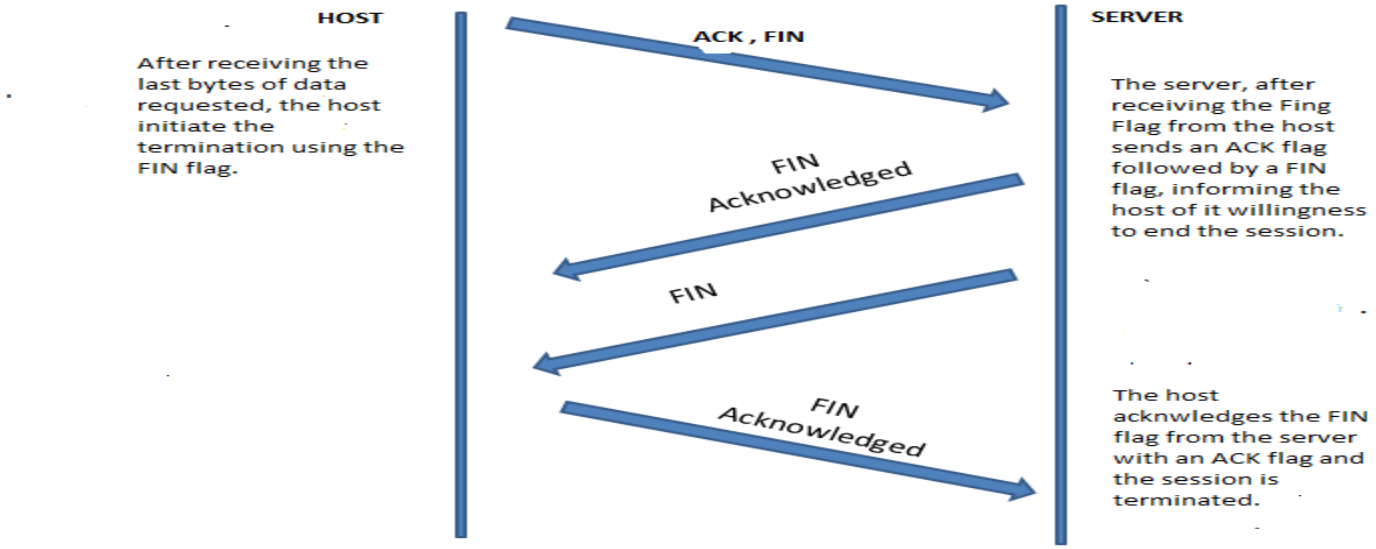


Three-way handshake

Now, let's have a look at how three-way handshake is performed in detail:

- Each flag must contain a source port number and destination port number.
- When a device is sending an SYN flag, its port number becomes the source port number, while the sending device becomes the destination port number.

- After the device receives the SYN flag and wants to send an acknowledgment, it uses the ACK flag and then reverses these port numbers.
- The session uses a four-way termination sequence. An additional flag, the FINISHED (FIN) flag, is used together along with the SYN and ACK flag.
- The finish (FIN) flag is used to request connection termination when there is no more data from the sender. This is the last packet sent by the sender.
- Using the session created above, after the host device receives the last packet from the server, it sends an acknowledgment (ACK) flag informing the server that it has received the packet.
- If the host intends to terminate the session, it sends a FIN flag and the ACK flag, informing the server that it has received all the information it requires from the server and intends to terminate it.
- The server replies with an ACK flag notifying the client that it has received the FIN flag and is aware of the hosts' readiness to terminate the session.
- The server replies with ACK and FIN flags, informing the host of its willingness to end the session. The session ends immediately after the host sends an ACK flag to the server completing the four way-handshake.



Session Termination

In the example above, the host initiates the session termination. But in practice, any device can terminate a session.

Error recovery

TCP provides reliable data transfer, that means that all packet sent from a source reaches its destination without any failure. In a situation where an error occurs along the route, TCP uses a mechanism to resend the faulty segment.

It uses acknowledgment (ACK) and sequence fields in the TCP header, to number the data bytes and track them. By so doing, it achieves reliability.

For example, if a web server has 400 bytes of data to send to a requesting web client. On establishing the session, the server breaks down the data into smaller segments, let's say, 100 bytes each.

Then, the server sends the first 100 bytes (0-99) of the data, with a sequence number of 1. The host after receiving this first segment, sends an ACK informing the server that the packet has reached its destination and starts waiting for the arrival of the next segment i.e., 100-199 with a sequence number of 2.

This mechanism continues up to 400 bytes, with the host sending an acknowledgment for each segment received. The explanation above does not recover any error, it will be the same TCP mechanism used for error recovery.

For error recovery, TCP uses the sequence and ACK flag so that the receiving host can notice missing data and request the sending device to resend the segment and uses the ACK flag to acknowledge the received of the missing piece.

Assuming in the example above, the host received the first hundred bytes (0-99) with a sequence number of 1, and instead of receiving the next bytes (100-199) with a sequence number of 2, the host receives a segment with (200-299) bytes with a sequence number of 3, it will send a packet requesting for that missing segment, i.e., 100-99 with a sequence number of 2.

The sending device can also resend a segment if the receiving device does not acknowledge all the data sent. The sending device, in this case, is a server.

The server waits for a few moments using a timer called the retransmission timer, to make sure that no other acknowledgment arrives, after that it then decides to resend that particular segment that it did not receive its acknowledgment flag.

Flow control using windowing

Because network host has limited resources such as limited space and processing power, TCP implements a mechanism called flow control using a window concept. This is applied to the amount of data that can be awaiting acknowledgment at any one point of time.

The receiving device uses the windowing concept to inform the sender how much data it can receive at any given time. This allows the sender to either speed up or slow down the sending of segments through a **window sliding process**.

User Datagram Protocol

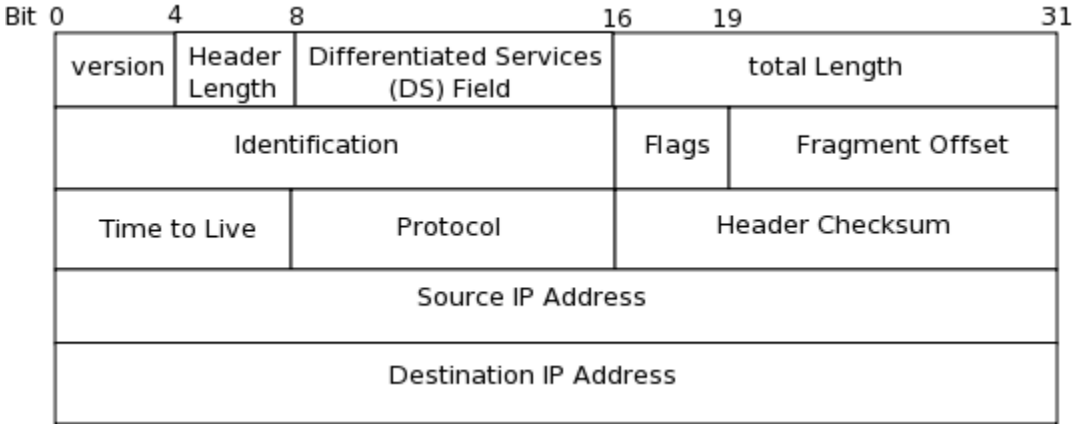
User datagram protocol (UDP) is considered as a best-effort transport protocol because it is a light-weighted transport protocol. UDP is a connectionless protocol, meaning it provides no reliability or reordering of the data segment and flow control like TCP. Because of this, UDP is faster than TCP in transporting data.

However, according to Cisco, UDP provides some similar benefits to TCP, such as data segmentation and multiplexing using port numbers. UDP is used by applications that are tolerant to loss of data but not delay.

For example, TCP's requirement will make it difficult to stream live video, as all packets must be sent and acknowledged, which will consume many resources and can cause severe delay.

But with UDP, if a packet is missing, the streaming will continue unnoticed. It only becomes apparent when many segments are missing, which is seen in low video quality and lack of synchronization between video and audio.

The significant difference between TCP and UDP is that TCP offers a wide range of services to applications, while UDP does not, this does not make UDP inferior to TCP, but by providing fewer services, UDP has fewer bytes in its header, and this makes UDP is faster when transporting data.



Differences between UDP and TCP

UDP	TCP
Because UDP has a low-overhead, it has faster transmission of data.	Because of TCP high-overhead, it has slow transmission of data
UDP does not acknowledge receiving of the data and does not resend lost data. Thus, it is not reliable.	TCP is reliable, because it acknowledges the received data and resends any lost data.
UDP delivers data as it arrives without an ordered arrangement of the segment.	It delivers data in a sequenced order.

Application

Protocols supported by UDP are:

- Dynamic Host Transfer Protocol (DHCP)
- Domain Name System (DNS)
- Trivial File Transfer Protocol (TFTP)
- Voice over Internet Protocol (VoIP)

Protocols supported by TCP are:

- File Transfer Protocol (FTP)
- Hyper Text Transfer Protocol (HTTP)
- Secure Shell (SSH)

Multiple separate conversations

The whole purpose of building an enterprise network or connecting a small office home office (SOHO) network to the internet, is for applications such as text messaging, email, video streaming, video, and audio conversations to occur.

To manage these multiple simultaneous conversations, TCP and UDP uses a header field that can uniquely identifies these applications running simultaneously. This unique identifier is called port number.

Port number

Each service running on a device uses a specific well-known port number. These port numbers identify each application or service running on a client uniquely.

For every connection from clients, the segment header contains two types of port numbers:

1. Source port number
2. Destination port number

Source port number

Source port numbers are port numbers dynamically generated by the sending device transport layer, that identifies each conversation between the two end devices.

Destination port number

In a segment sent by a client, a destination port number is placed within it to tell the destination server, the services that the client is requesting.

This mechanism is possible because, unlike on a client machine in which request can originate from any locally unused port, services provided by a server have a well-known dedicated port assigned to them. As such, the destination port number is inserted by the client, informs the server.

For example, Telnet uses TCP transport protocol and has a destination port number of 23. When a server receives a segment with a destination port number of 23, it knows that the client is requesting a Telnet service.

Socket pair

The source and the destination port numbers placed within a segment, only identifies which application in a client, requests for that service from a server. But the segment does not have any mechanism to specify which device is requesting the service.

To identify which device is requesting a particular service, the internet protocol (IP) encapsulates the segment containing the source and the destination port number.

This IP packet includes the source IP address to identify which device the request originates from and the destination IP address to determine the destination device. Thus, creating a socket.

A socket is a combination of the destination IP address and destination port or source IP address and source port.

A socket is handy to the transport layer because it keeps track of services and devices requesting such services, to properly forward the data to the requesting application as stated by Cisco.

Port number groups

Internet Assigned Numbers Authority (IANA), an organization responsible for assigning various addressing standards, has grouped port numbers into three major groups, these groups are:

- Well-known ports
- Registered ports
- Dynamic or private ports

Well-known ports (0-1023)

Well-known ports are port numbers assigned to services such as web browsers, email clients, HTTPS, and Telnet.

The RFC6335 outlines the registration procedures for these services and port numbers.

The table below shows us some well-known port numbers, the transport layer protocol that they support, and their applications. These port numbers are assigned as listed in RFC6335.

Port number	Protocol	Application
20	TCP	FTP data
21	TCP	FTP control
22	TCP	SSH
23	TCP	SMTP
53	UDP	DNS
67	UDP	DHCP Server

Registered port numbers (102-49151)

Organizations such as Cisco have port numbers assigned to some of their well-known services by IANA.

IANA assigned these port numbers to request entities to use with specific processes or applications.

Dynamic (49152-65535)

Dynamic port numbers are usually assigned by a client operating system (OS) dynamically when establishing a server connection.

VRP(Versatile routing platform)

This is just a quick intro guide into Huawei Versatile routing platform. We tend to do a lot of work on these day to day but documentation is very sparse.

We are going to aim to put a few guides on here for common scenarios and command line options.

What is Huawei VRP?

Huawei essentially have their own* OS which allows configuration via the commandline. These routers are pretty popular as they do provide a lot of functionality. They do have downsides though (I'll get to that) but overall they are very capable. One of the main advantages is price, they are cheaper in comparison to a Cisco equivalent mainly due to the fact that their chipsets for DSL / VDSL is their own. Whereas Cisco or Juniper use a 3rd Party and have to therefore buy the chips, Huawei have their own.

Downsides

So there are some downsides with this. When buying one or looking at the specs they are good on paper, all singing all dancing , can do this can do that etc. however when you have a large estate of them and you work on them day to day you notice some of the issues they have. Here are a few I have personally come accorss:

- Cisco but not – So as you probably know Cisco filed against Huawei stating intellectual property, basically Huawei had copied portions of Cisco IOS. It is so similar it is uncanny but its not similar enough. It becomes more of an annoyance so for example to configure a Huawei you enter system-view, to run a show command its display. Its only a small thing but its annoying when jumping between OS's. Its not different enough to think agh yes its an Huawei, because they server the same purpose as a Cisco I find myself jumping on and forgetting its a Huawei.
- Save sometimes doesn't apply – Usually you except a command to be saved when you enter it, although it mine show in the current-config it sometimes doesn't apply / become active / work until you reboot. They are unreliable and if something doesn't work which you apply its usually "Its a Huawei, give it a reboot"
- Reboot takes forever – This is the biggest annoyance in my opinion, coupled with the above a reboot takes literally 10 minutes or so, its the biggest downside of Huawei's.
- Routing metrics are odd – The admin distance of the routing protocols is so odd, for example by default an OSPF route is preferred over a static route, its not a major issue but its more of a why?

Good Things?

So its not all bad, this isn't a post about why Huawei's are rubbish and you shouldn't buy one, its more of an intro into what we have seen whilst working with them on a larger scale. Here are some of what I consider to be upsides

- Vendor support is ace – They actually are really good, one example I have is we had WiFi issues accross a whole estate for a customer where speeds were actually terrible for any device for a certain model and firmware, unfortunately we rolled this out to 500+ sites. The support worked with us labbing this up and the long and short is they wrote a new patch which could be applied to the router and solved it. I have never gotten support like that from a vendor before.

- They are capable – The devices themselves are actually very capable its just configuring them which is a pain. They support everything you would excpect from an enterprise grade router.
- Cost – They are obviously a little bit cheaper than a Cisco and pretty much do the same thing.

VRP network operating system

The **VRP (versatile Routing Platform)** , the universal routing platform, is the research experience of Huawei in the field of communication, and is the operating system platform of all data communication products based on IP/ATM architecture. **Huawei products that run the VRP operating system include routers, LAN switches, ATM switches, dial-up access servers, IP telephony gateways, carrier-grade integrated service access platforms, intelligent service selection gateways, and dedicated hardware firewalls** . The core switching platform is based on IP or ATM.

The operating system adopts layered design, which is divided into physical layer hardware related drive interface, real-time operating system and task dispatch interface, IP/ATM forwarding center and Routing policy Management, system management and Configuration service, routing application layer and service layer.

The Huawei VRP provides a modular architecture with rich functional features and application-based tailoring and scalability capabilities. The VRP is Huawei's fully autonomous intellectual property network operating system that provides a consistent network interface, user interface, and management interface for a variety of hardware platforms, and offers a flexible and versatile application solution with more than 300 features. The VRP, with IP/ATM switching platform as its core, integrates the communication elements such as routing technology, QoS service technology, VPN tunneling technology, security technology and digital video/voice technology. Taking the IP Turboenginetm technology independently developed by Huawei as an example, the traditional processing method of network beginning is to pass through the physical layer, link layer, and network layer and routing strategy layer, etc., through the layer task scheduling and analysis processing, the message processing speed is limited by each layer protocol stack processing flow and the bottleneck of operating system scheduling. Only by the increase of CPU and bus to improve the forwarding rate, and IP Turboenginetm technology completely changed message forwarding scheduling process and routing lookup algorithm, composed of two core technologies: Hardware forwarding/interrupt forwarding and fast routing lookup algorithm. Hardware forwarding/interrupt forwarding bypasses the traditional layer beginning processing scheduling process, and in the Interrupt service program of receiving the packet in the physical layer buffer, it realizes the type recognition and routing lookup of the message, and points the head pointer of the message to the sending queue of the corresponding interface within the interrupt, thus realizing the interrupt forwarding. Due to the interruption of the program instructions can not be too long and affect the system operation, high-speed routing lookup algorithm is also the core technology, it can be guaranteed at the IPV4 address, up to four times to find the destination route and related interface index, and the size of the routing table is independent, this technology to ensure that the very short program instructions to complete the message routing To provide the basis for the implementation of interrupt forwarding, so that the device's message forwarding speed increased by 5-10 times! Coupled with the Distributed processing technology and QoS technology, Huawei Network products in the performance indicators with international first-class standards.

VRP (Versatile Routing Platform) is the Operating System that many **Huawei** devices operate. This is like Cisco's **IOS**, Nokia's (Alcatel-Lucent's) **TiMOS** and Juniper's **JUNOS**. There is only one difference. There is no "**OS**" at the end of the abbreviation :)

There are different versions of **Huawei VRP (Versatile Routing Platform)**. Beginning with VRP 1.X, now, VRP 8.X isbeing used. All the **VRP versions** from the beginning up to nowis given below. VRP Versions:

- **VRP 1.X**
- **VRP 3.X**
- **VRP 2.X**
- **VRP 5.X**
- **VRP 8.X**

Here, we will not talk about all the versions but it can be useful to talk about the last **VRP version**. The last VRP version VRP 8.X provide many new feature that is not supported in the previous **VRP versions**.

First of all, VRP 8.X support virtualization and cloud routing. Beside, it provides Multi concept of many features. VRP 8.X support Multi-CPU, Multi-Chassis, Multi-Process, Multi-Product and Multi-Solution.

Huawei VRP Upgrade

All Operating systems need upgrades for new features and developments. As other Operations Systems, **Huawei VRP** is also need upgrades.