Conceptualization of Computer Networks

Dr.G. Ignisha Rajathi, Ph.D.,

Assistant Professor,

Department of Computer Science and Engineering,

Sri Krishna College of Engineering and Technology (Autonomous),

Coimbatore.

S. Nagajothi, M.E.,

Assistant Professor,

Department of Computer Science and Engineering,

Sri Krishna College of Engineering and Technology (Autonomous),

Coimbatore.

Dr.P. Balamurugan, Ph.D.,

Associate Professor, Department of Information Technology,

SRM Institute of Science and Technology, Kattankulathur, Chengalpattu,

Tamilnadu, India.

Dr.R. Johny Elton, Ph.D.,

Indsoft Technologies,

Tirunelveli.

Published by

Centivens Institute of Innovative Research Private Limited

Conceptualization of Computer Networks

ISBN 978-81-942938-5-9

Month: August 2020

Authors

Dr.G. Ignisha Rajathi

S. Nagajothi

Dr.P. Balamurugan

Dr.R. Johny Elton

Centivens Institute of Innovative Research

307, 5th Street Extension, Gandhipuram,

Coimbatore - 641012, Tamilnadu, India.

E-mail: publications@centivens.com

Website: www.centivens.com

Preface

***Always be prepared for the worst; but hope for the best* – Lee**

A rich knowledge of a structured approach is disseminated in a detailed version in this book. Even if you have not known about networking before and desire to know the different concepts of networking, then *Conceptualization of Computer Networks* will get you started. It is a quick book to quick cook the state of art in Computer Networks. You will know how easy it is to quintessence on the conceptual resolution to the prevailing inter-juncture which is given as all under one roof.

It explains how networks work from inside out. It is a thorough text for an introductory level learner which is clearly researched with consistent notation and style, full set of diagrammatic and theoretic explanation. It can also be taken as a supplemental text or reference for a huge variety of networking related courses. The in- depth coverage of the five different modules include the fundamentals and detailed version of link layer; media access and internetworking; routing; transport layer; application layer; Having grabbed the fundamental framework, the detailed conceptualization of MAC, Ethernet, Wireless LANs, Wi-Fi 802.11, Bluetooth 802.15.1, Switching and Bridging, IP versions and many more topics have been discussed clearly in this book. The further chapters focus on different routing protocols, global internet followed by transport layer with protocols, connection management, congestion control, quality of service. Most of all, the application layer concept instigates the practical implications of E-mail, http, web services, SNMP based MIB and security. This book supports meticulously to understand the ideas behind the implementation of the massive network connectivity all over the galaxy.

Be assured of learning a plethora of computer networking concepts in a single capsule – *Conceptualization of Computer networks.* Get ready for the treasure hunt in Networks.

Acknowledgement

"Each one has his own gift from God"

Gratitude is the fairest blossom which springs from the soul.
Burden-bearing, laughter-sharing, forever-caring ***FAMILY***!!!...a very happy, bliss-filled, hearty thanks to each one of you! Our gratitude knew no bounds!

We are very much grateful to ***our friends and well-wishers, near and dear, superiors and colleagues*** who have been a great source of encouragement and support to us all through our endeavors.

Happiness cannot be expressed by words and help taken cannot be left without thanking. We would like to thank all who are a part of our lives and our work.
A simple but strong word of real sense – ***"THANK YOU!".***

Author Biographies

Dr.G. Ignisha Rajathi received her degrees - Bachelor of Engineering and Master of Engineering as a rank holder, in the discipline of Computer Science and Engineering under Anna University, Chennai. She completed her Doctorate in the Faculty of Information and Communication Engineering under Anna University, Chennai. Having 13 years of teaching experience, she is presently working as Assistant Professor in the Department of Computer Science and Engineering at Sri Krishna College of Engineering and Technology, Coimbatore, India. She has marked her areas of interest as Medical imaging, Image processing, Soft Computing. She has published more than 15 research articles in Journals, including high-impact versions and in various Conferences. She has published books and patents. She has trained the police force in fundamentals of computers and has delivered many invited talks and guest lectures, also engaged in consultancy projects.

S. Nagajothi, Assistant Professor, Department of Computer Science and Engineering, Sri Krishna College of Engineering and Technology, Coimbatore. She has completed her Bachelor of Technology - Information Technology in the year 2014 and Master of Engineering - Computer Science and Engineering in the year 2016 as a Rank holder. I have 3 years of Teaching Experience. Her area of research is IoT, Machine Learning, Cloud computing and Wireless Networks. She has published around 06 papers in Scopus indexed journals, 02 book chapters and life member of ICSCS, IAENG and IEEE**.** She has also received Research Grant from ICMR and CSIR for Conducting Seminars and Workshops.

Dr.P. Balamurugan is Associate Professor, Department of Information Technology, SRM Institute of Science and Technology, Kattankulathur, Chengalpattu, Tamilnadu, India. He passed BE in Computer Science and Engineering from Madurai Kamaraj University in 2003 and ME in Computer Science and Engineering from Manonmaniam Sundaranar University in 2006. He completed his doctorate from Anna University in 2013. He obtained GATE score in 2003. He is recognized as a Supervisor for Ph.D Programme and M.Tech (By Research) in Information and Communication Engineering for Anna University and Vel Tech Rangarajan Dr. Sagunthala R & D Institute of Science and Technology. He is the member of IEEE, ACM, ISTE etc... He is trained certificate holder of 'High Impact Teaching Skills' through WIPRO MISSION 10X and Trained Evaluator of NBA and ABET. He is in the reviewer member of 6 International/National Journals. His research interest is mainly in Networks, Ad-hoc and Sensor Networks, and Data Structures. He has filed and published more than five patents. He has published books "Theory of Computation" and "Problem Solving and python Programming" and also a number of research papers in international journals and presented papers in international conferences. He has done a number of consultancies work at various organizations. He has delivered a many informative guest lectures at various organizations. He has also organized and conducted many workshops, conferences and seminars at his zone.

Dr.R. Johny Elton is a Research Fanatic with ardent passion in scientific exploration of detailed delineation on current technologies. He did his Bachelor of Engineering Degree in Noorul Islam College of Engineering, Thuckalay, Master of Engineering in Manonmaniam Sundaranar University and Doctoral degree from Anna University, Chennai. His research interests include Natural Language Processing, Computer Vision and has published research papers in peer-reviewed Journals. Currently, he is working for Indsoft Technologies, Tirunelveli, on various innovative research works.

TABLE OF CONTENTS

CHAPTER 1

1. Fundamentals & Link Layer

Objectives

- To understand about Network requirements and building a network.
- To explain basics of networking.
- To know about internet architecture.
- To explain about layers and protocols.
- To explain about flow and error control.

1. Introduction

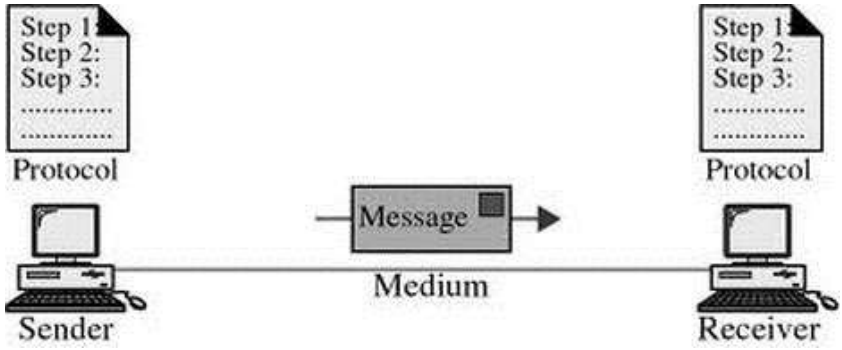
1.1. Building a Network

Data Communication

Data communication is defined as the exchange of data or information between different devices through any destined transmission medium, example, wire cable. Data communication occurs with the communicating devices or systems which are composed of a combination of hardware and software. Hardware is stated as any physical equipment and software is stated as programs using any programming language. The efficiency of this system relies on 4 important features such as

- Accuracy
- Timelines
- Delivery
- Jitter

Components of a Network



- Sender
- Receiver
- Message
- Protocol
- Transmission medium

Accuracy - The system is deemed to deliver the data exactly.

Timelines - Data should be delivered in a2 timely manner.

Delivery - The system has to deliver data to the destined location.

Jitter - The difference in the time of arrival of packets.

Sender - Device that sends the data to receiver.

Receiver - Device that receives the data from sender.

Message - The information to be communicated.

Protocol - Set of guidelines to administer data communication.

Transmission medium - Physical pattern where a message migrates from sender to receiver.

1.2. Requirements

Perspectives

Network design depends upon the following perspectives.

- **Application programme -** Specifies the list of services needed by application.
- **Network designer -** Lists attributes of low cost but efficient design.
- **Network provider -** Lists the features of a system which is easy to manage and provide security.

Scalable Connectivity

In network, only few nodes were selected for privacy and security. A system which is capable of supporting the growth of the system to an arbitrary large size is meant to be scalable.

Data Representation

- Text
- Numbers
- Audio
- Video
- Images

Data Flow

Two devices can be communicated through 3 different forms such as simplex, half duplex and full duplex.

- **Simplex**

The connectivity is unidirectional ie., one way in simplex mode. Any one of any two systems in a connection can send and only the next system can receive.

E.g: television used in our day to day life.

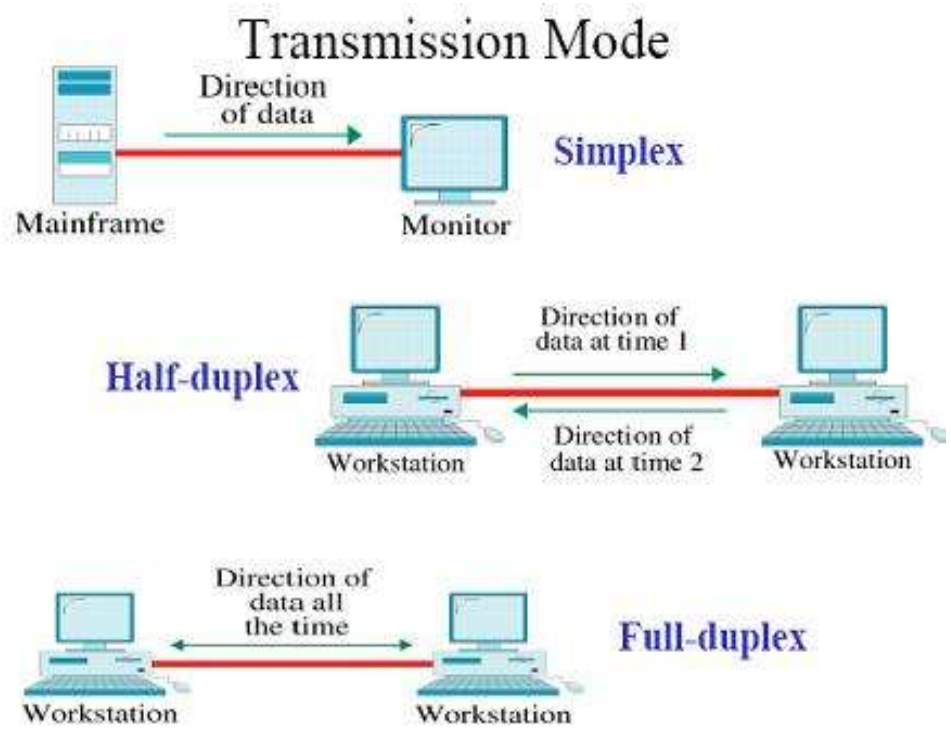
- **Half duplex**

All stations send and receive the messages in different timelines, which is not simultaneously passed at the same time in half duplex. This is given as when one device transmits data, the other receives data.

E.g: walky-talky, CB radios.

- **Full duplex**

Both stations send and receive simultaneously, where the data can be passed at the same time under full duplex mode. Communication can be performed in both directions at the same time. E.g: Telephone.



Network

A set of devices or nodes linked through communication links is called as network. Nodes->computer, printer, and scanner. All devices send and receive data, formulated by other nodes on network. Network uses distributed systems where there is sharing of any process by different systems.

Network Criteria

The metrics of network criteria are performance, reliability and security.

Physical Structure

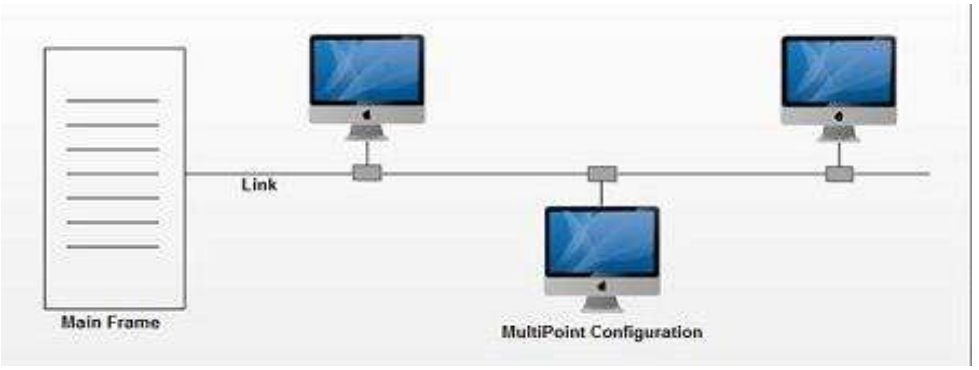
Connection and its types – Connection establishment is the connectivity between devices in a network. A link is a pathway to communicate and send and receive data between devices. The types are given as follows.

- Point to point
- Multipoint

Point to point – The establishment of connection between 2 individual devices.

E.g: Television satellite link.

Multipoint – when a single link is shared by more than two devices, it is called as multipoint (multidrop).



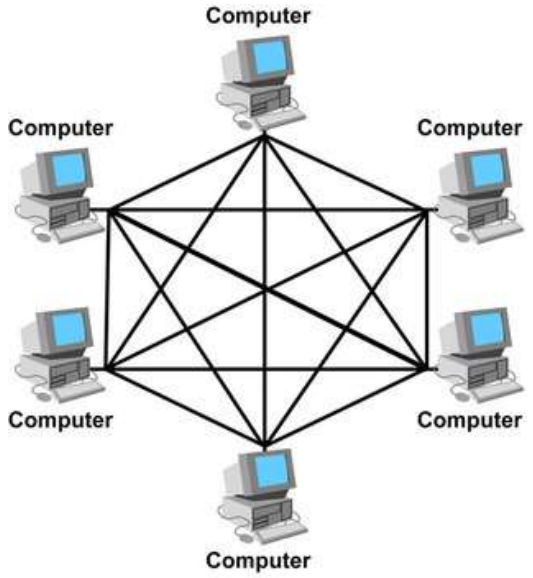
Categories of Topology

The physical establishment of network connectivity is called as topology. They are categorised as mesh, star, bus, ring and hybrid.

- **Mesh**

All devices establish a point to point connectivity to other devices in its scope of contact, in mesh topology. The connected 2 devices carry messages in this topology.

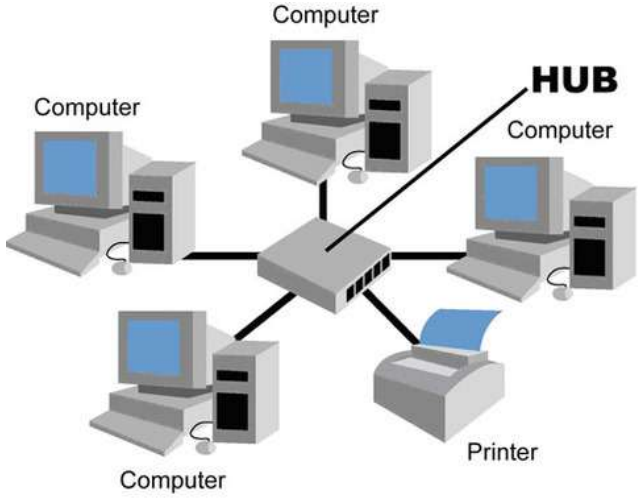
No of links = n (n-1)/2, where n stands for nodes



- **Star**

In star topology, a central controller holds the connectivity with the devices using devoted link and the central controller is hub.

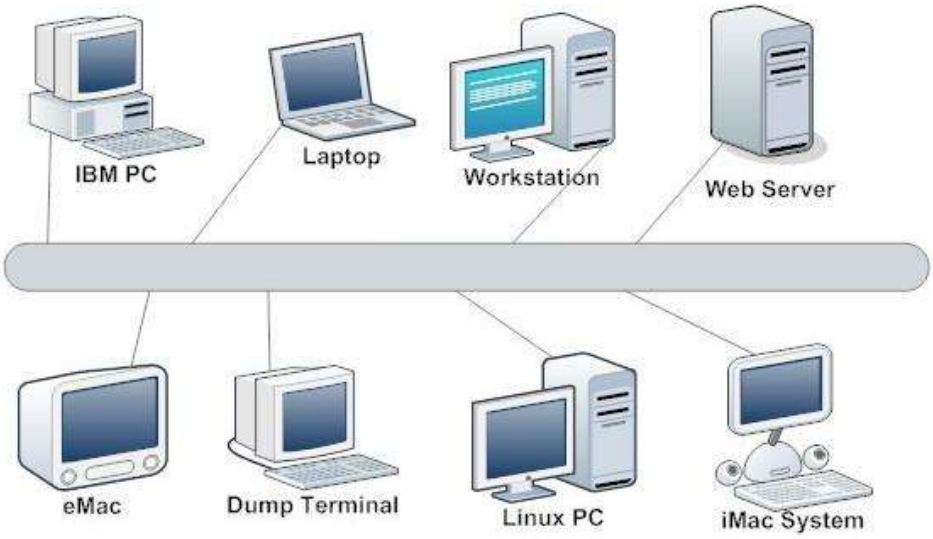
No of links = n, n stands for nodes.



- **Bus**

A bus is multipoint link. The connection between the device and the main cable is done by drop line. To establish connectivity with the metallic core a connector splices into the main cable or punctures the sheathing of a cable through tap.

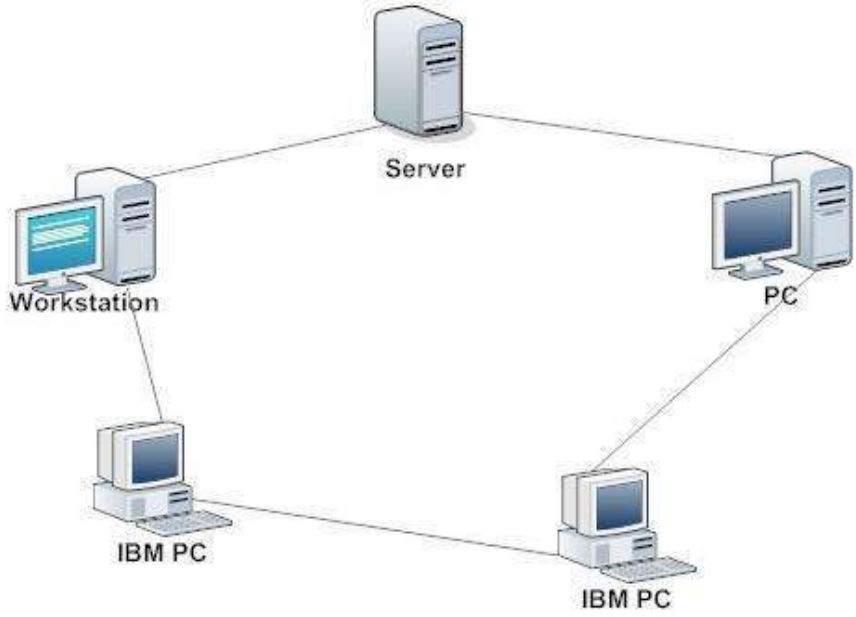
No. of links = 1 backbone, n droplines



- **Ring**

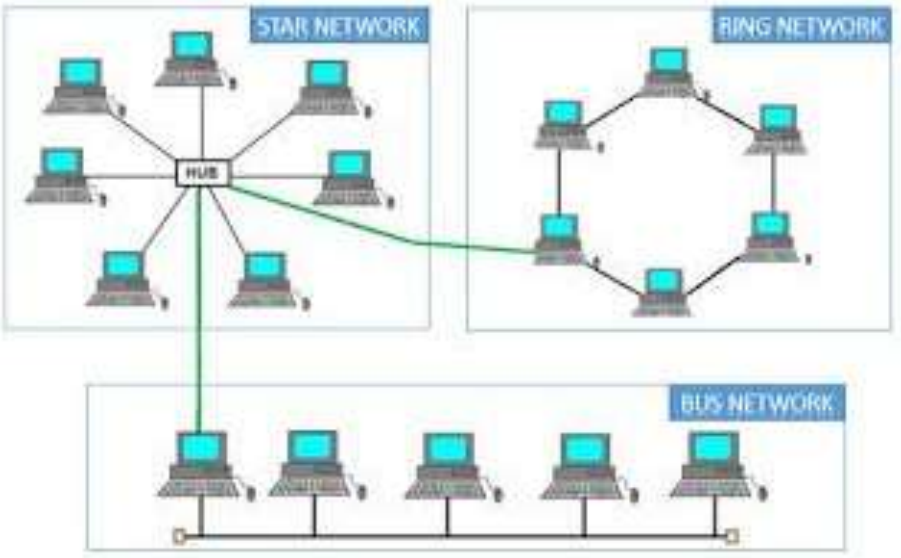
A connectivity is established between 2 devices in ring topology which are in close proximity on both sides. The transmission proceeds through the ring in a direction. Each system in ring connects with repeater. Repeater generates bits and passes them.

No of links= n-1, n is the number of nodes



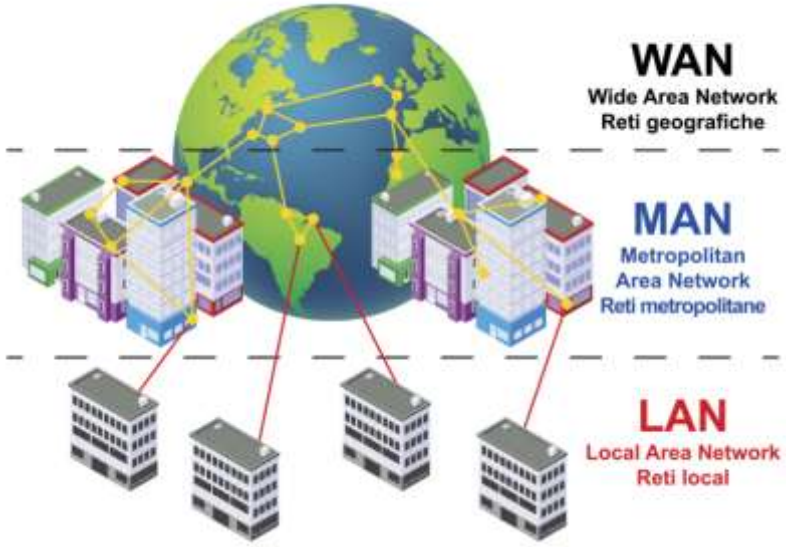
- **Hybrid**

The mixture of different topologies is Hybrid topology.



Network Models or Categories

- **LAN -** Local Area Network (less than 2m).
- **WAN -** Wide Area Network (world-wide connectivity).
- **MAN -** Metropolitan Area Network (span ten of miles).



1.3. Layering and Protocols

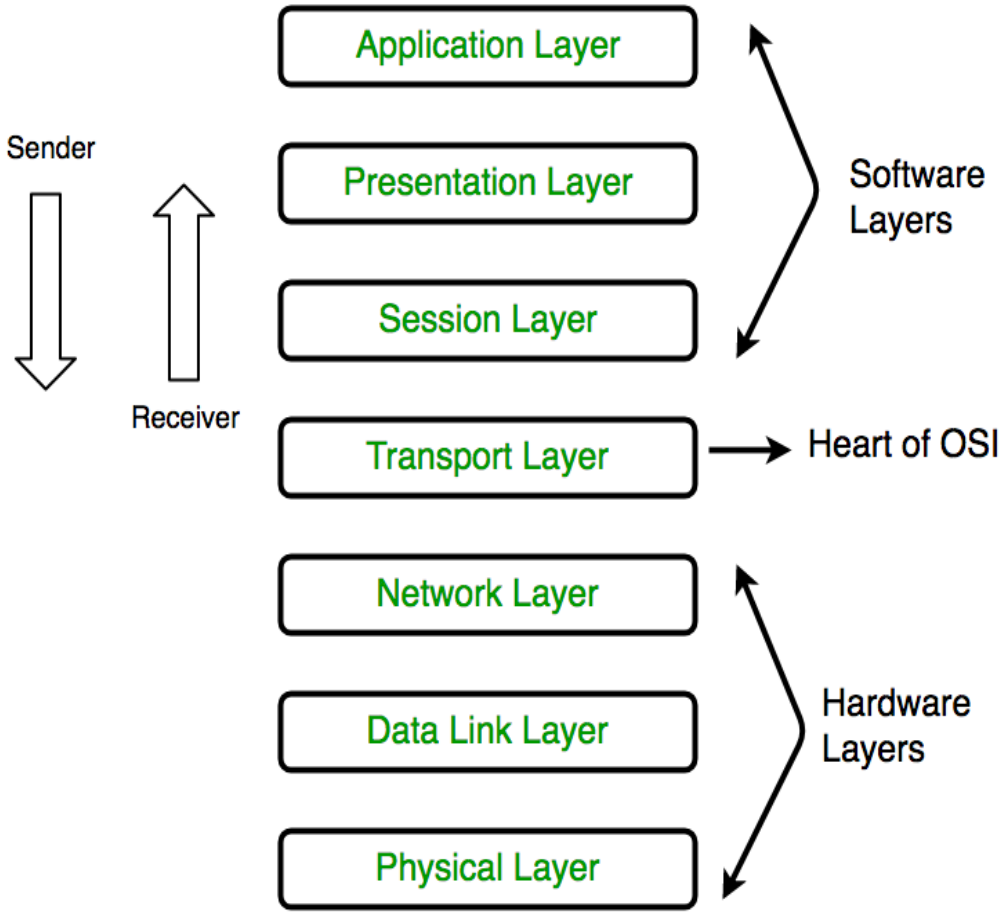
Protocols → syntax, semantics, timing

Standards → de facto, de jure

1.3.1. Networks Models

The International Standards Organization (ISO) devoted worldwide International Standards in 1947. In 1970, an ISO standard covers all aspects of network communications in Open System Interconnection (OSI) models. In order to formulate connectivity with various systems irrespective of logic of software in hardware, is the motive of OSI model. It is used to design network architecture, which is flexible, robust and interoperable. ISO holds OSI as model.

It consists of 7 layers which used to explore data communication**.** The communication is governed by instructions and resolutions as protocol. The event in a system that enables a communication is peer to peer process.



Over the adjacent layers of the sender and the receiver the data along with network information is traversed or passed by interfacing.

Network Support Layers

Physical

Datalink

Network

Transport → link these 2 layers

User Support Layers

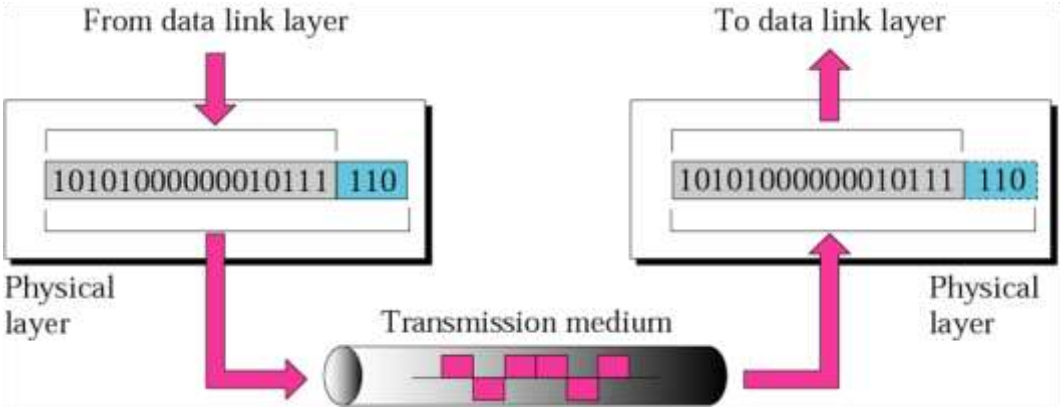
Session

Presentation

Application

Physical Layer

It performs the process to carry a bit stream through a physical medium. It instructs the physical systems and interfaces perform the data transfer, being properly transferring individual bits from one system to another. Physical layer positions process to data link layer, through the transmission medium.

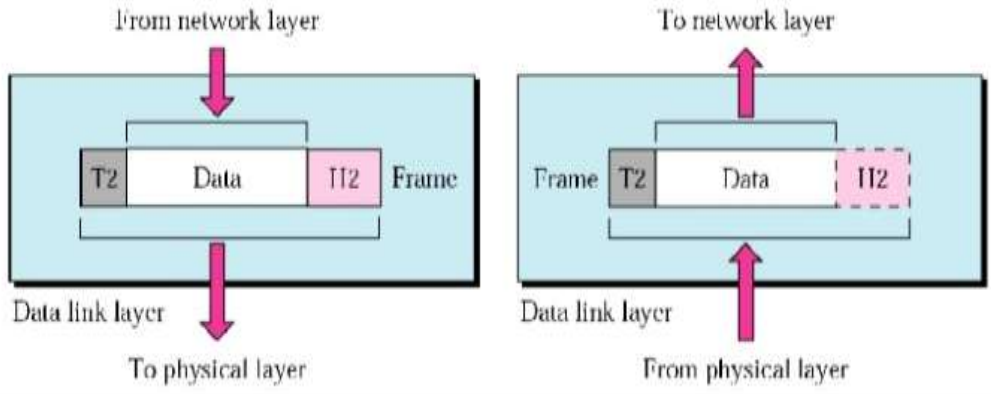


Characteristics

1. Physical characteristics of interfaces and medium
2. Representation of bits
3. Data rate (transmission rate)
4. Synchronization of bits
5. Line configuration
6. Physical topology
7. Transmission mode

1. *Data Link Layer*

It is the next layer to the physical layer makes it look error-free to network layer. The frame movement between devices is done by the data link layer. The data units being transferred to network layer as bit streams is frames. This illustrates node-to-node delivery.



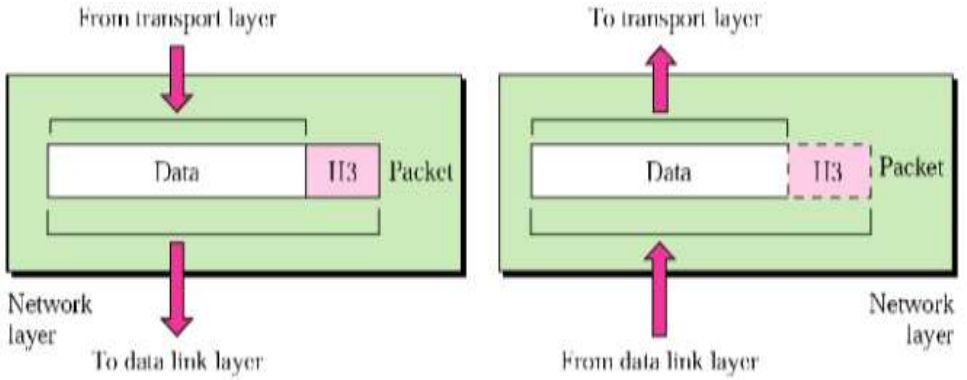
Characteristics

Different characteristics are listed as follows.

1. Framing
2. Physical addressing
3. Flow control
4. Error control
5. Access control

2. *Network Layer*

It facilitates the delivery of packets across networks, through links from source to destination. It delivers packets between two devices in same network. Both networks and links are connected to create networks of networks or mass networks called as routers or switches.



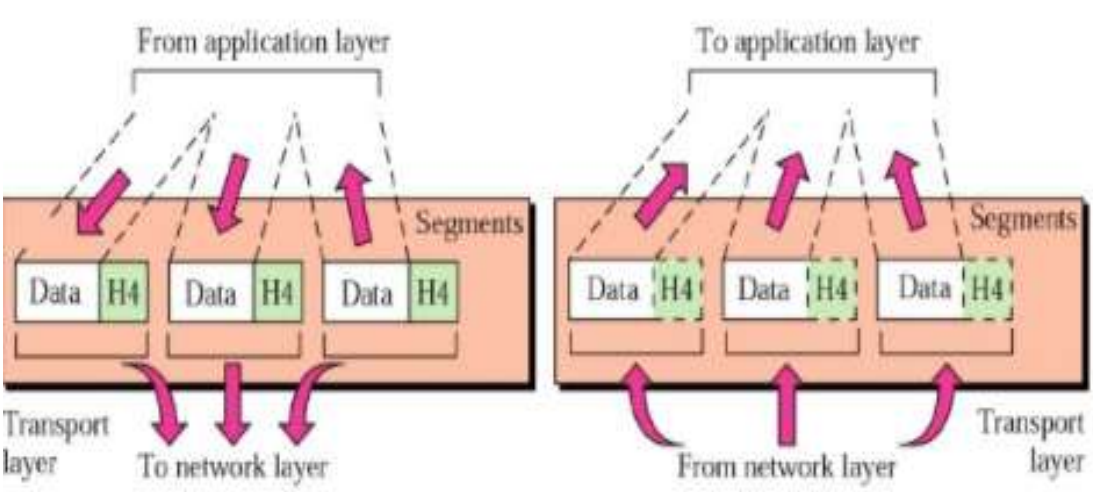
Characteristics

The various characteristics are listed as follows.

- Logical addressing
- Routing

3. Transport Layer

The delivery of the complete messages without damage is the purpose of this layer. An application program executed on a host machine performs the said process.



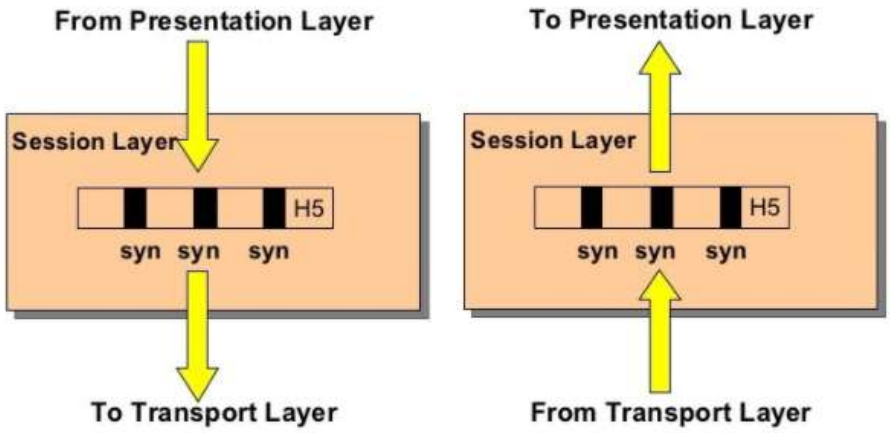
Characteristics

The various characteristics are listed as follows.

1. Service-point addressing
2. Segmentation and reassembly
3. Connection control
4. Flow control
5. Error control

4. Session Layer

The facilitation, maintenance and synchronization are the activities of session layer. It is the network dialog controller.



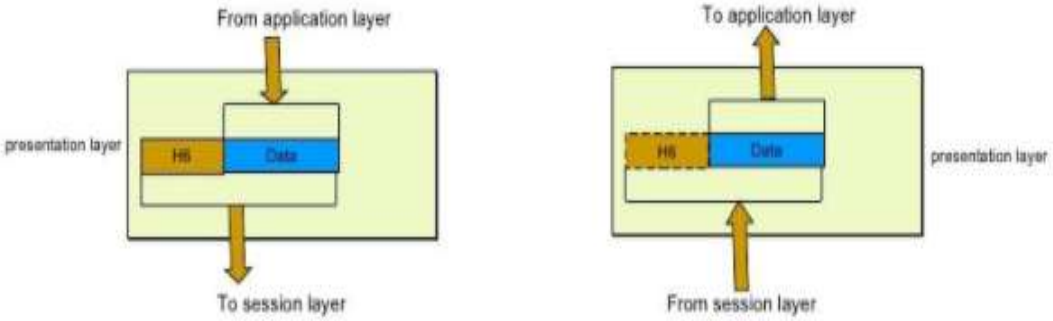
Characteristics

The various characteristics are listed as follows.

- Dialog control
- Synchronization

5. Presentation Layer

The syntax and semantics of messages are preserved in this layer. It holds the responsibilities of performing translation, compression and encryption.



Characteristics

The various characteristics are listed as follows.

- Translation
- Encryption
- Compression

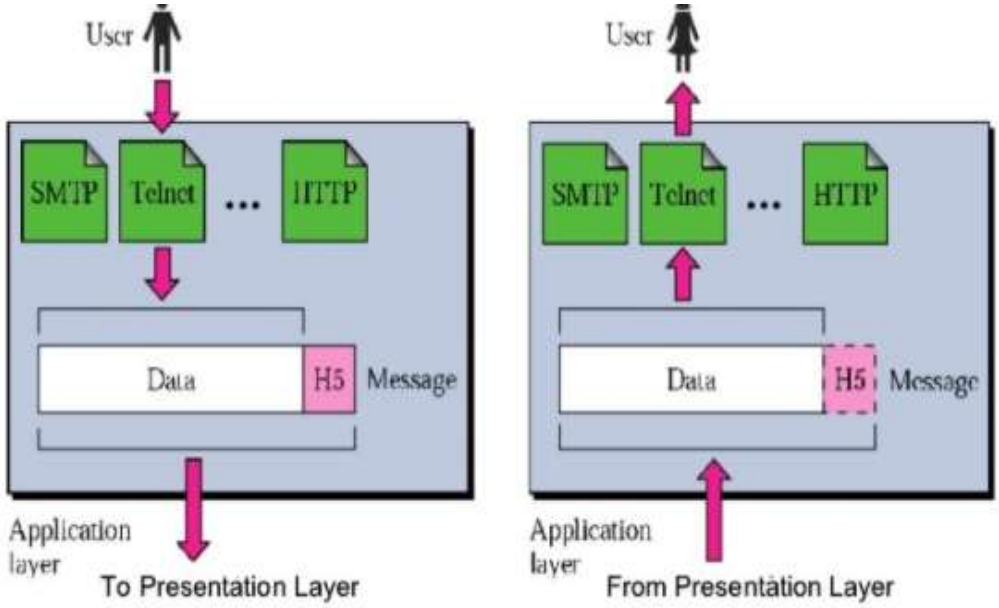
6. Application Layer

User interfacing is enunciated in this layer which supports various services like e-mail, remote file control, shared database management and distributed information services.

Characteristics

The various characteristics are listed as follows.

- Network virtual terminal
- File management
- Mailing services
- Directory services



Types of Connections

Link means two or more devices connected to each other through physical connection in a network**.** Computers connected in link is also referred as nodes, hosts, work stations.

1. Point to point connection
2. Multipoint connection

1. Point – to – point

A dedicated direct connectivity is established between systems.

Example - Remote control, TV control system.

2. Multipoint

Multipoint connection supports sharing of channel capacity among stations in the network.

- Spatially shared communication.
- Time shared connection.

Spatially shared - more than one device sharing the link simultaneously.

Time shared - devices share the link on turn by turn basis.

Switched Network

Switching is a methodology which interconnects multiple connectivity to establish a large network to have an effective communication. A switched network contains a sequence of nodes that are interlinked with each other known as switches. The circuit switching, packet switching and message switching are the categories of switching.

Circuit switching - It is established by physical connectivity to form networks of 'n' number of channels.

Packet switching - The message is divided into packets of fixed or variable size and transmitted.

Message switching - Messages are received, stored and transmitted.

Internetwork

When two or more devices are connected by an established communication link for sharing data or resources or exchanging messages is called as network or networking. When two or more networks need to be connected for the same purpose is called an internetworking or network of computer network. The connecting devices, routers, gateways are used to connect independent networks to form internetwork.

Addressing

The address of a node given by LAN or WAN or MAN is the physical address. Logical address is essential for universal communications to identify each host uniquely which are not basically dependent on underlying physical networks**.** Physical address changes hop-to-hop. Logical address remains same. Process of forwarding the messages to the destined node as per its addressing is called as routing.

Types of Address

1. **Unicast -** Once source & one specific destination.
2. **Broadcast -** One source & all nodes on the network.
3. **Multicast -** One source & some subsets of nodes on the network.

Cost Effective Resource Sharing

1. Modem (Modulator+ Demodulator)

It is used to perform both modulation and demodulation according to the requirement.

2. Multiplexer & Demultiplexer

The process of transmitting more signals simultaneously on one path is termed as multiplexer. The process used to perform demultiplexing, which separates the signal and send it to the appropriate destination device.

Reliability

1. Error Control

The data must be delivered to their destination accurately as it was sent from the source. Reliability is achieved by check summing each packet in source and verifying the checksum at the destination. Internet protocol (IP) is the mechanism which is used by TCP/IP protocols for transmission of data.

Types of Error

1. single bit
2. multiple bit(or) burst error

Single bit – If only one bit in a given data string is permissible to change during the transmission.

Burst bit - if two or more consecutive bits in a data string are permissible to change.

- Single bits affect only one character.
- Burst bits affect one or more characters.

1. Congestion

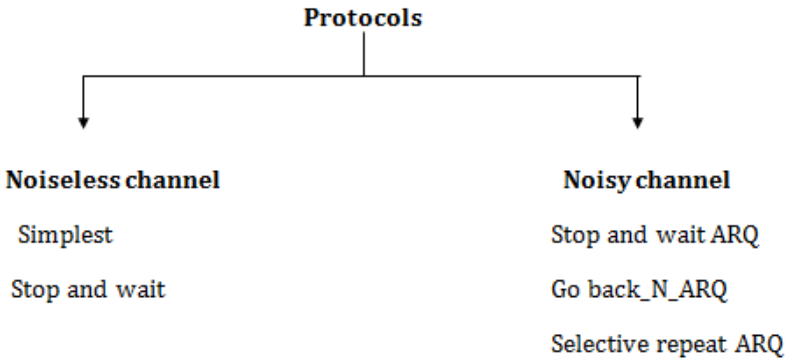
Packets are lost due to congestion in the link to overcome in the network uses congestion control mechanisms. Congestion occurs if the users of the network send data at a rate that is greater than the network can handle (number of packets). When enormous packets are present in the subnet, the performance of the network will be degraded. To handle congestion as prevention or control, this is used.

2. Retransmission

If a packet is damaged, lost, delayed during transit (or) if the acknowledgment has not yet been received, then it will be retransmitted.

1.3.2. Protocols

The process of framing and achieving appropriate control over flow and error handling in delivery of data is implemented in datalink layer using protocols.



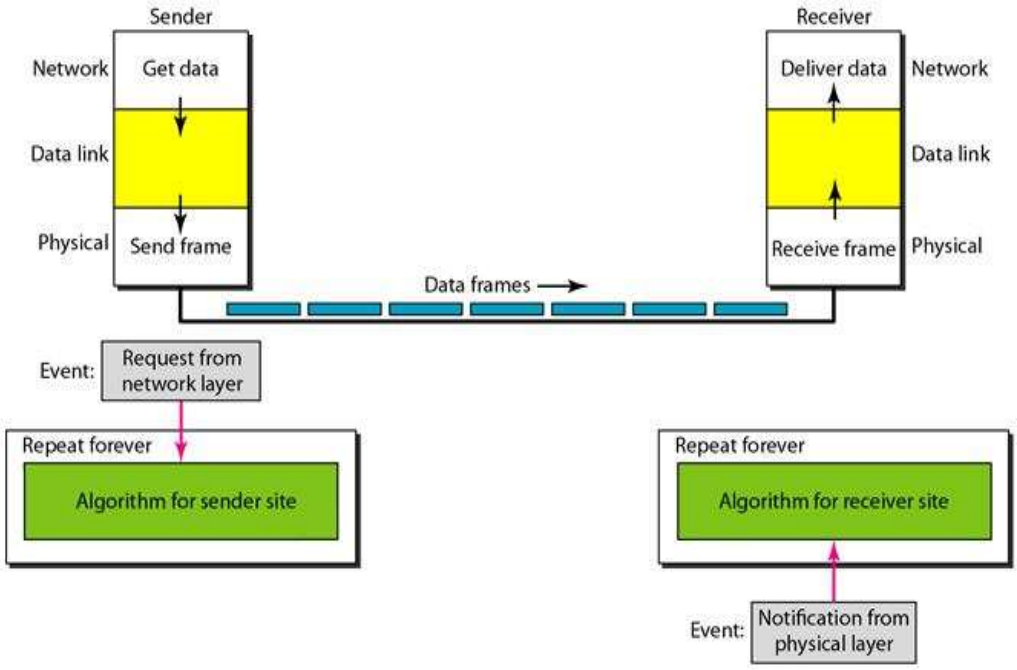
In general, the data frames travel from sender to receiver because of unidirectional property. The acknowledgement (ACK) and negative acknowledgement (NAK) which are identified as special frames flow in direction which contradict one another with the data flow direction. Piggybacking is the process of including ACK and NAK in the data frames, to hold the flow and error control information.

Noiseless Channel

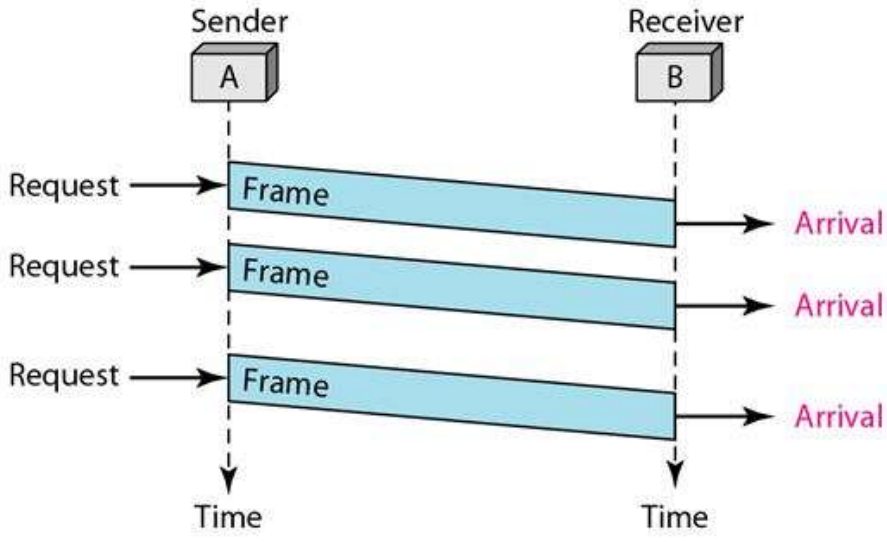
The protocol of noiseless channel devoid usage of flow control.

1. Simplest Protocol

This protocol lacks flow control. At the transmitter's site, data in the data link layer is obtained from network layer by preparing a frame and transmitting it. On the other end, at the receiver end, the frame is received via the physical layer and the data is extracted from the received frame and delivered to the network layer. It is noted that the datalink layers offer transmission to the sender and receiver and vice versa, to its network layers. Also, in order to transmit bits physically, these data link layers use the services provided by their physical layers. A frame is transmitted from the sender's site only when a data packet is held by the network layer and this data packet is delivered only when the frame arrives the receiver site. When the event at the sender site or receiver site is running constantly, and no action will be encountered until the network layer requests at the sender site or any notification is received at the receiver's site.



Flow Diagram

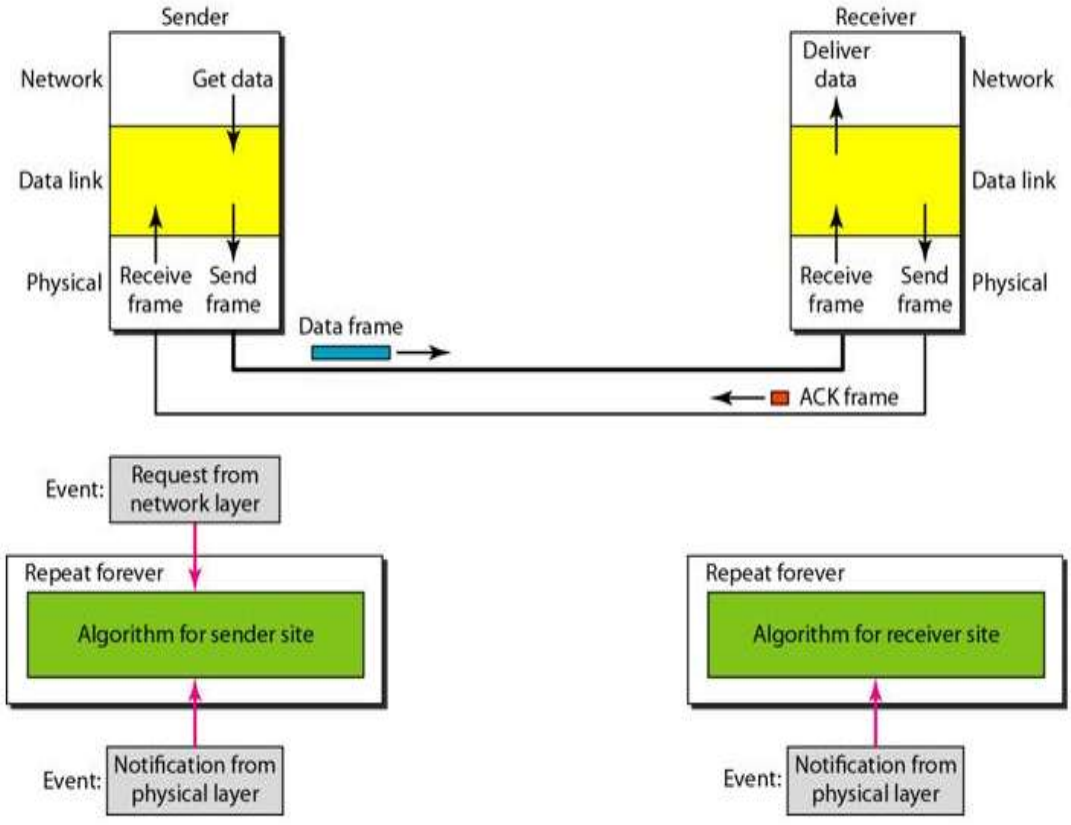


2. Stop and Wait Protocol

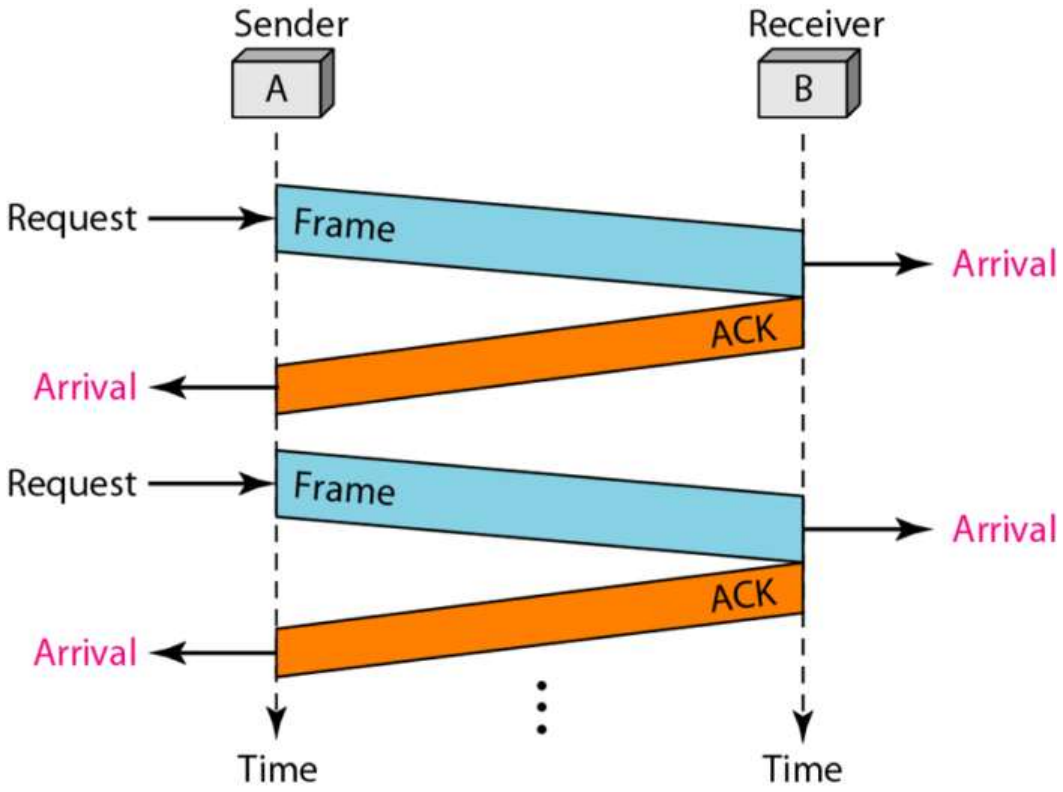
The arrival of data being earlier at receiver location, the data frames have to be retained before its deployment. Particularly when receiving data from several sources, the receiver just doesn't have enough storage capacity.

This either results in the abandoning of frames or denial - of - service. To stop the receiver being overloaded with frames, instruct the sender to slow down. Feedback from the recipient to the sender must be present. The sender sends a frame in stop-and-wait protocol, stops until the recipient receives the approval and then continues to transfer. For data frames it is unidirectional communication but ACK auxiliary frames move from the other direction. The traffic is visible on the channels enunciating the forward for data and backward trail for ACK. Here they employ half-duplex connection. Either a request from network layer or an arrival note from physical layer or both are done.

Until the recognition of the frame, the reply has to be deferred or ignored. The channel does not repeat the frames and is error free. However, the network layer can post a request continuously ignoring in-between arrival. It stops the data frame from being sent straight away. When a frame is sent and when the receiver receives the data frame, the variable is set to send ACK. When an ACK is received it sets it as true to allow the next frame to be sent. If it's a false it can't allow the sender to send the next frame. Note the protocol sending two frames includes the sender in four events, and the recipient performs two events.



Flow Diagram



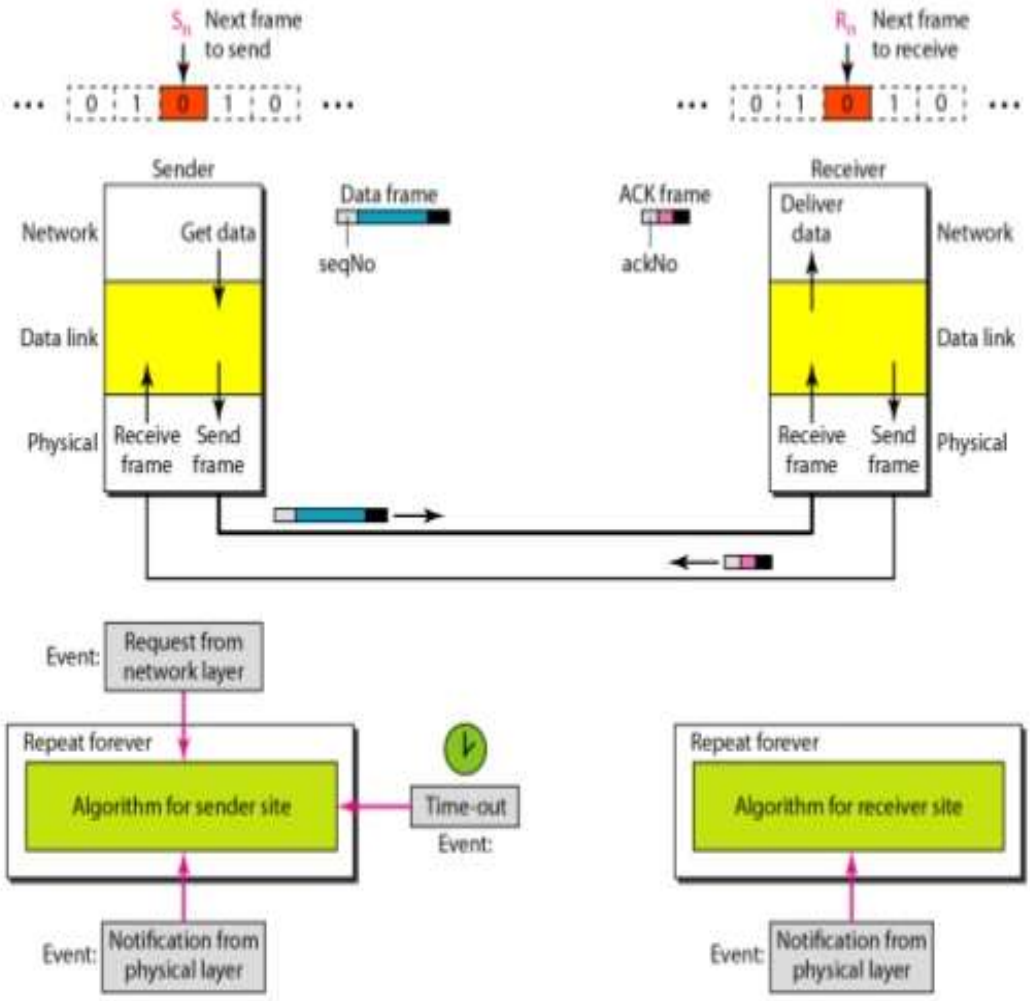
Noisy Channels (It Adds Idea to Flow Control)

1. Stop and Wait Automatic Repeat Channels

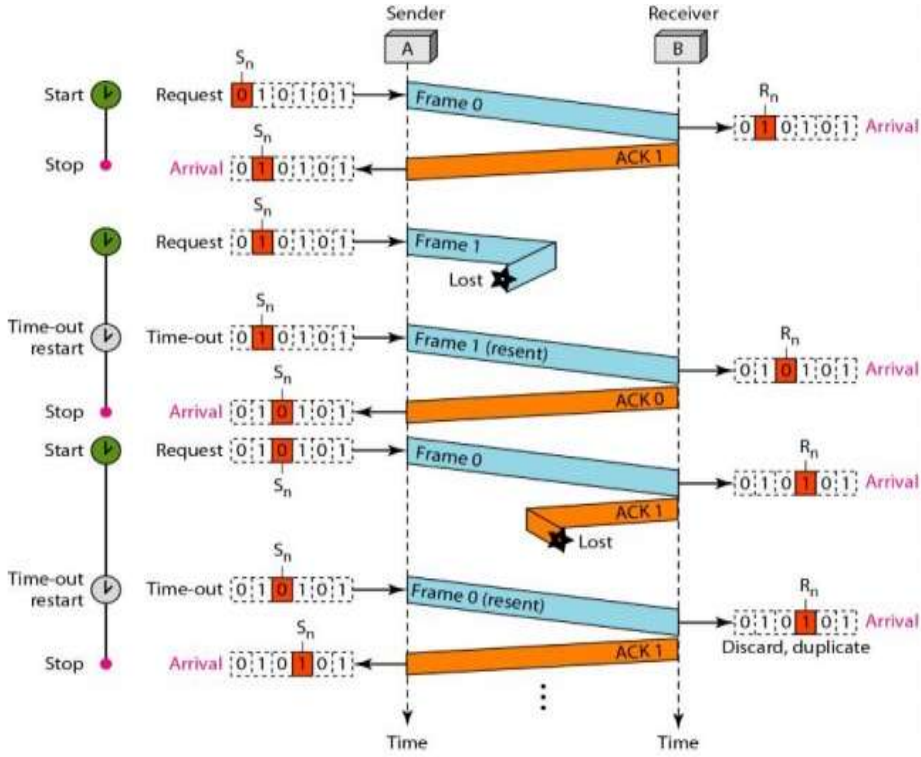
Stop and Wait-ARQ is a basic protocol which aids in error control mechanism. It is important to add redundancy bits to the data frame to identify and correct corrupted frames. The broken message is tested and discarded at the receiver. In this protocol, the detection of errors is manifested by the receiver's silence. Lost frames are toughest to manage than corrupted ones. There is no way for a frame to be marked. The right one or a duplicate frame or a frame out of order might be the receiver frame. When the receiver receives an out-of-order data frame, that means frames have either been missed or recreated.

Timer is used in this protocol, as and when the timer expires and if the sent frame is not acknowledged, the frame is resent, the copy is preserved and the timer is restarted. The protocols use the stop and wait system, as redundant data prevails in the network, there is only one unique frame that requires an ACK. By holding a copy of the sent frame and retransmitting the frame when the timer expires, error correction is done in stop and wait ARQ.

Whereas an ACK frame can sometimes be distorted or destroyed, redundancy bits and a sequence number are required as well. The ACK frame for that portal has a number field sequence. The sender actually discards a damaged ACK frame or avoids an out-of-order in this protocol.

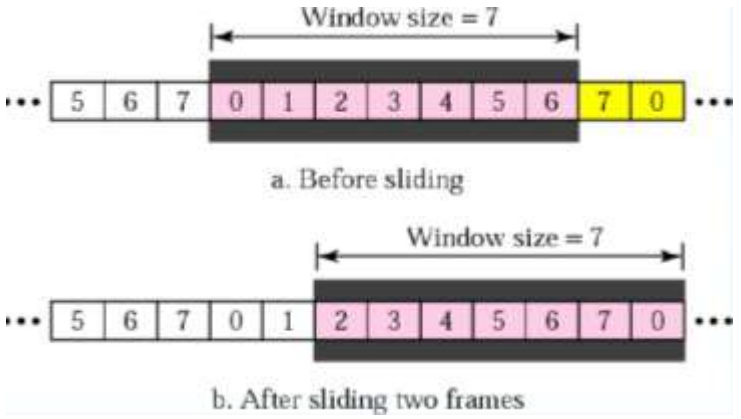


It states that sequence numbers must be designated to each frame. A field is added to the data frame by storing the sequence number of that frame. We provide sequence number frames in Stop and Wait ARQ. The sequence numbers are based on the arithmetic of mod-2. The data frames and the ACK frames have to possess same sequence number. The number of the acknowledgment often alerts the recipient with next frame by sharing its sequence number.

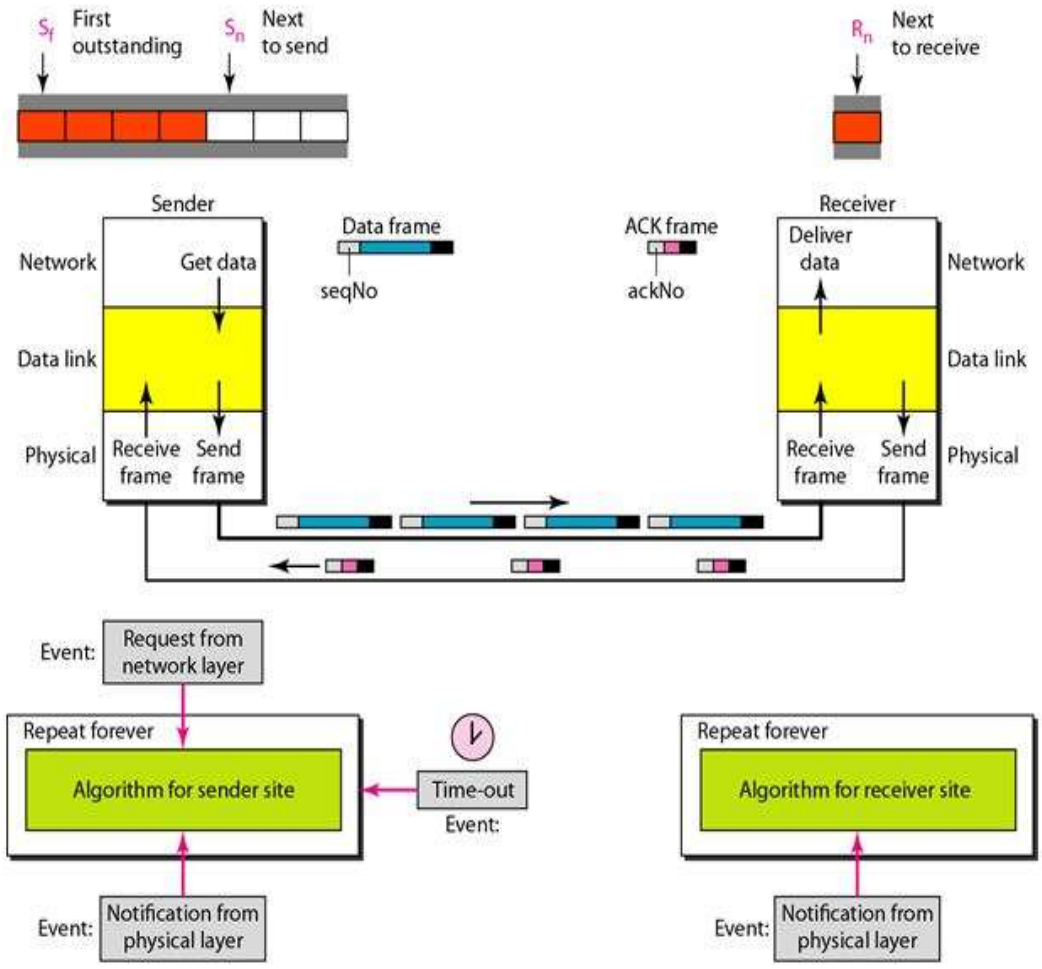


2. Go Back 'N' ARQ

In the stop-and-wait procedure, the transmission time needed for a frame to enter the receiver plus the transmission time for the recognition to return is negligible. In system like satellite system, the round-trip time can be as long as 500 ms (propagation delay) this protocol is also known as back N ARQ**.** This technique is used to resolve the stop and wait ARQ inefficiency by enabling the transmitter to continue to send adequate frames so that the channel is kept occupied while the transmitter is waiting for recognition. If one frame is damaged or lost, all frames are sent after retransmission of last frame.

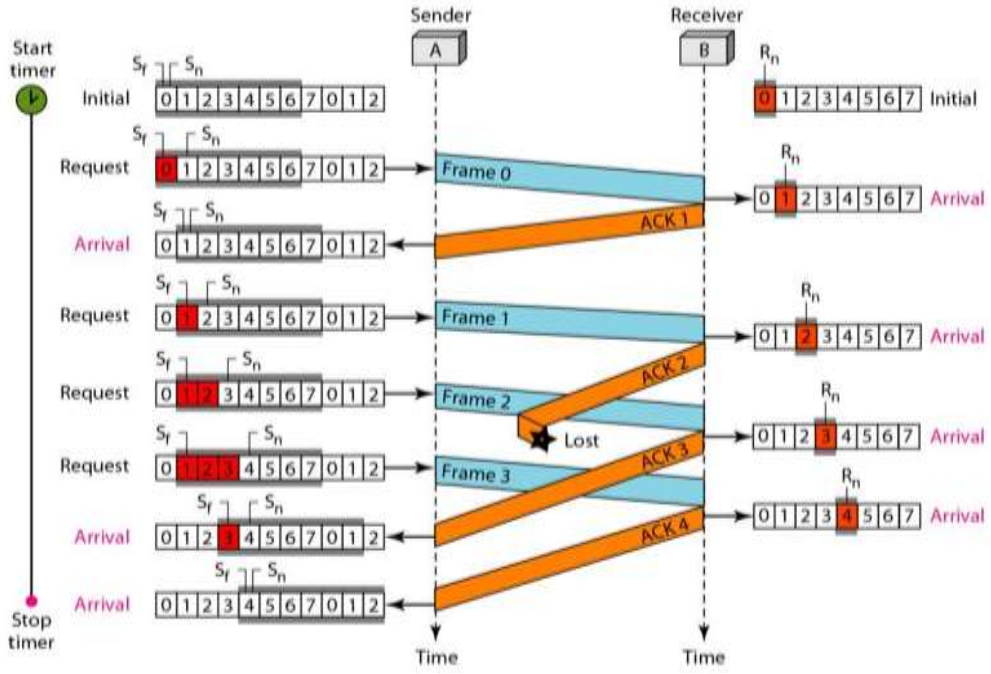


The sender does not wait for an ACK signal for the next frame to be transmitted. It continuously transmits the frames as long as the NAK signal is not received by it. NAK is sent to sender by the recipient. If the transmitted frames are damaged or destroyed, or if the acknowledgment is destroyed, the error can be implemented. If the second frame is impaired, error is detected and NAK – 2 signal is sent to the receiver back. The transmitter begins retransmission from frame 2 upon receiving this signal. The receiver discards all of the frames obtained after frame 2.



If the receiver does not receive a specific data frame, it sends a NAK to the transmitter and the transmitter retransmits all the frames received from the last recognised frame. After each data frame the transmitter does not anticipate an acknowledgment in return N. The transmitter can send as many frames as the window allows until an acknowledgment is awaited. It must wait until the timer goes off and retransmit all frames again until limit has been reached or the transmitter has no more frames to transmit. The selective repeat ARQ is

most powerful but complex of all ARQ protocols. In this process, the sender retransmits only the frame that is damaged or lost. Like go-back N process, the lost ACK or NAK frames are handled in the same way. This method employs pipelining. The previous task is completed as pipelining in the networking of a task is always initiated. It increases transmission quality.



1.4. Internet Architecture

Internet Architecture or Internetworking Architecture → Transmission Control Protocol / Interconnecting Architecture → TCP/IP

TCP / IP is a compilation of rules and procedures setting out how all transmissions are transmitted over the Internet In 1969, TCP/IP was originally developed as protocol for networks that was connected to advanced research project agency network (ARPANET) ponded by defence advanced research projects agency (DARPA) in us. This protocol is composed of a broad set protocol, released as standards.

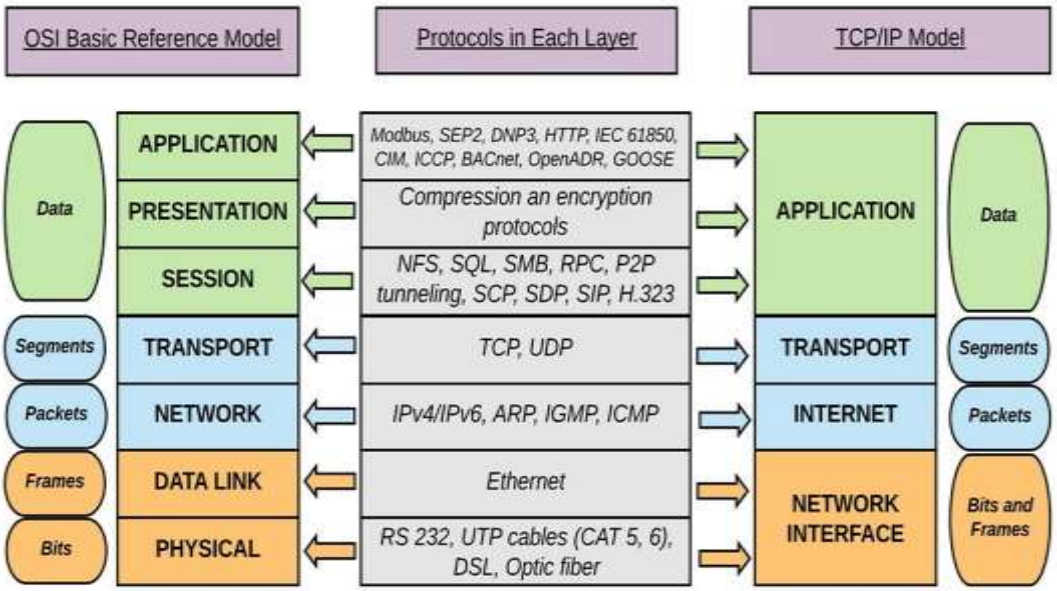
Internet Standards by the Internet Activities Board (IAB)

There are 4 layers in TCP/IP:

1. Application Layer
2. Transport Layer
3. Internet Layer
4. Host to Network

TCP/IP layers was developed prior to OSI model. Host to network is a mix of physical as well as datalink layer. The Internet layer is comparable to the network layer of the network. The application layer consists of a mixture of session, presentation, and application layer with transport layer support. The basic functionalities of these 4 layers are given so.

At transport layer, TCP/IP supports 3 protocols namely, TCP (Transmission Control Protocol), UDP (User Datagram Protocol) and SCTP (Stream Control Transmission Protocol).



TCP/IP is hierarchical consisting of networking devices, each with a particular feature.

1. Host to Network (Physical and Datalink Layer)

- At physical and datalink layers, no specific protocol is given in TCP/IP.
- It adheres to work with standards and propriety procedures.
- TCP/IP can be a LAN or WAN.

2. Network Layer (Internet Layer)

- The IP in the network layer is supported by TCP/IP.

Internetworking Protocol (IP)

- This protocol is unstable and connectivity-free as the strongest service delivery effort.
- Best effort means IP doesn't check or monitor errors.
- IP transports data in packets called datagrams.

Address Resolution Protocol (ARP)

- ARP is used to connect a physical or station address with a logical address.
- A station address is defined by each device on a connexion.
- ARP is used specifically to locate a node's physical address when its internet address is identified.

Reverse Address Resolution Protocol (RARP)

- RARP helps a host to explore when its physical address is identified at international addresses.
- This is invoked when a computer initially establishes connectivity with network or during the booting of a diskless computer.

Internet Control Message Protocol (ICMP)

- ICMP is enabled during data issues and intimates to the host through gateway to notify sender.
- ICMP sends messages about query and costs from accidental loss.

Internet Group Message Protocol (ICMP)

- IGMP facilitates the simultaneous transmission of messages to a set of receivers.

3. Transport Layer

- Two protocols addressed the transport layer in TCP / IP: TCP, UDP.
- IP is a device communication -to-host which can send a packet from one physical device to the next.
- UDP and TCP are transport-level protocols which carry a process-to - process communication.

User Datagram Protocol (UDP)

- The simplest of TCP / IP protocol is the UDP.
- It is a process - to - process protocol that adds data from the data to the upper layer with only port address, checksum, error control and length information.

Transmission Control Protocol (TCP)

- TCP facilitates applications with complete transport facilities.
- A stable stream transport protocol is TCP.

- A link between both ends of the transmission must be formed before the data can be transmitted either.
- TCP breaks a data stream into smaller units at the transmitting end, called segments.
- TCP collects each datagram at the receiving end, as it reorders the transmission based on sequence numbers at the receiving end.

Stream Control Transmission Protocol (SCTP)

The SCTP supports new applications such as voice over internet.

The best features of UDP and TCP are combined in the transport layer protocol.

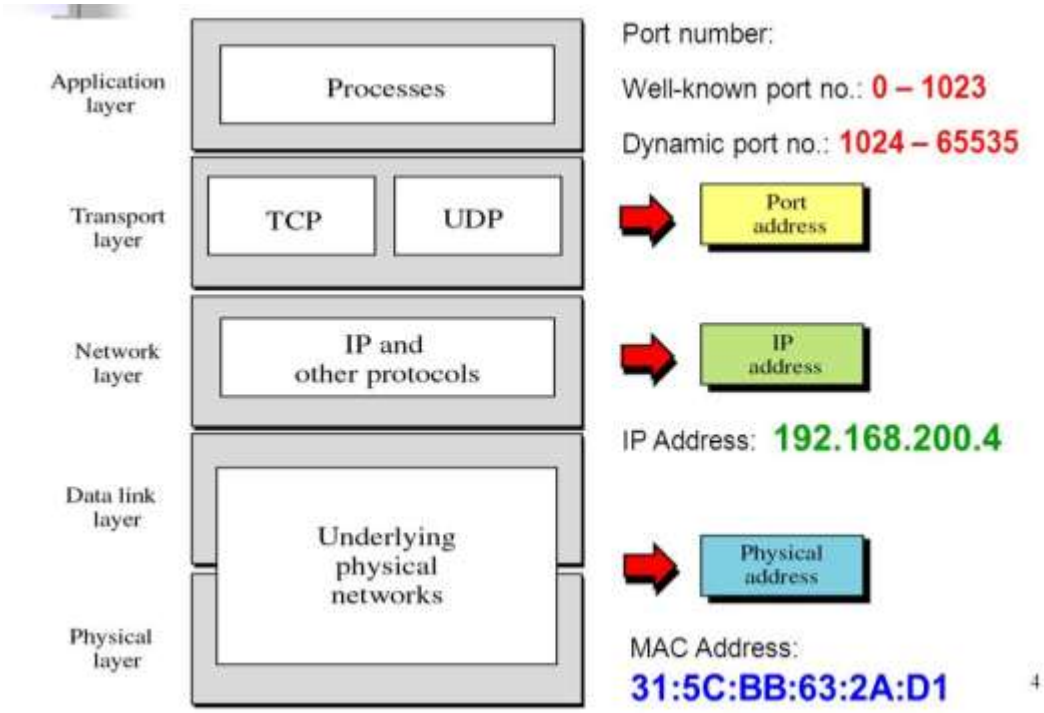
4. Application Layer

The TCP / IP application layer equals OSI session, presentation and application layers.

Addressing in TCP/IP

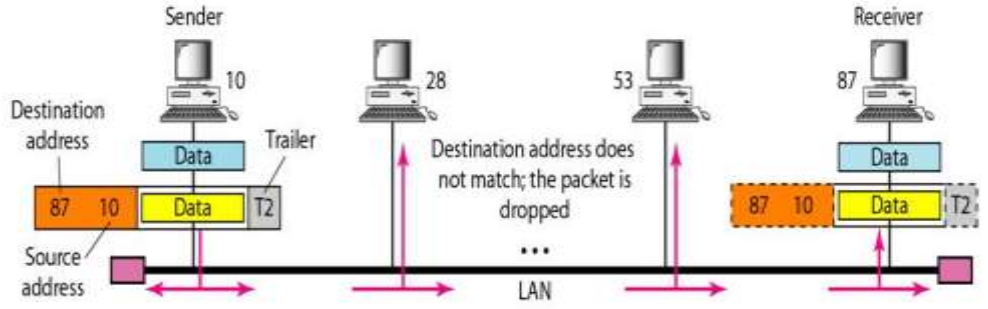
Various levels of addresses used in TCP/IP protocols are:

1. Physical (link) Address
2. Logical (IP) Address
3. Port Address
4. Specific Address



1. Physical Address

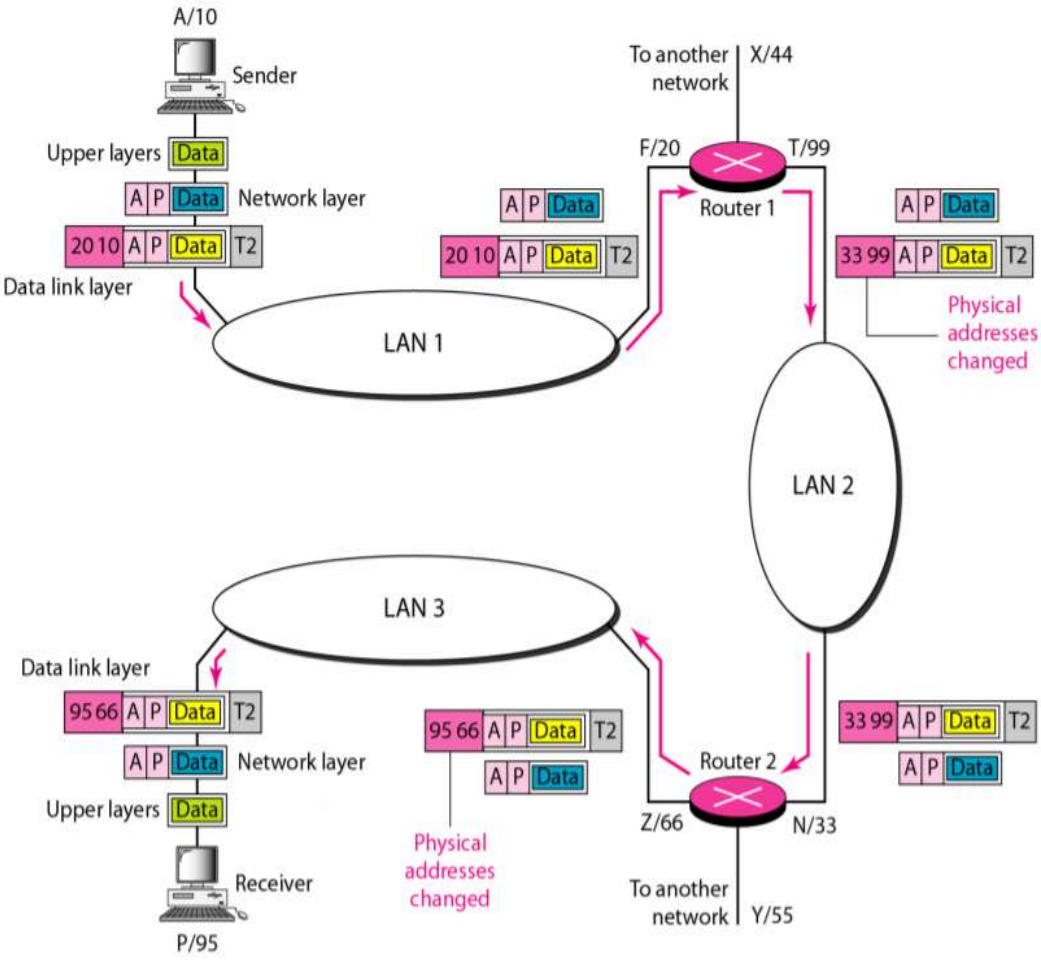
- The physical address is known as the link address.
- It is a node's address as specified by its LAN or WAN.
- It is included in the frame that the data link layer uses.
- Physical Address holds the authority over Networks (LAN or WAN).
- These address size and format differ depending on the network.



Physical Address is given by 6 Bytes (12 Hexadecimal Digits)

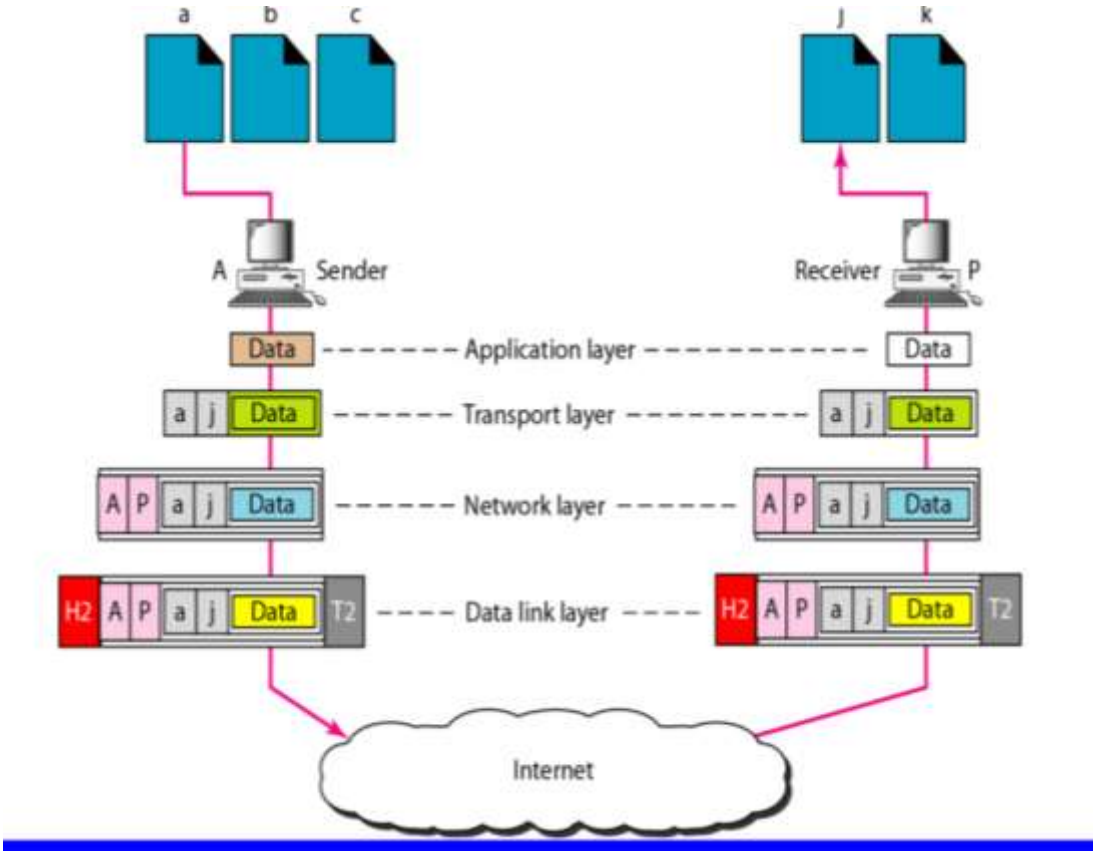
2. Logical Address

- It establishes global connectivity irrespective of the physical grids underlying them.
- A logical internet address is usually a 32-bit address that can be uniquely connected to the interconnect host
- The physical addresses will vary between hop to hop, but typically the logical addresses stay the same.
- Publicly addressed and visible hosts on the internet must have different IP addresses.



3. Port Address

- The IP and Physical address are required for information to be transported from source to destination host.
- Computer devices can concurrently run several processes.
- Internet networking is an interacting mechanism with other processes.
- In ICP / IP, a port address is called the label assigned to a method.
- The TCP / IP port address is 16 bits long.
- The Port address typically remains unchanged.
- A 16-bit port address is represented by a single number.



4. Specific Address

- User friendly addresses also exist for certain applications.
  E.g: E-Mail id, URL, WWW
- Sender can change the addresses with the corresponding port and logical address by the sender.

1.5. Network Software

Network design and protocol requirements are significant; however, the design is inadequate in justifying the internet's remarkable performance. The number of internet-connected computers has exponentially increased because of the functionality provided by the internet. Computer networks are capable in principle of transporting any kind of information such as digitized images, digital voice and so on. It was very slow to send and receive data to do something useful with the information. Today's networks are increasingly being used to carry multimedia, and their support for it can only increase as hardware becomes faster. Software

applications are designed to communicate with users and a communication protocol set to communicate across the globe.

Application Programming Interface (API)

(sockets)

- The network-export interface is the place to start when implementing a network application.
- Most of the network protocols are software and network protocols along with operating system are utilized by neighbouring computers. nearby all computer systems implement their network protocols as part of the operating system (exported by the network).
- When the "exported via the network" interface normally refers to that given to its networking subsystem by the OS.
- This interface is also called the programming interface for network applications (APIs)
- Each operating system is allowed to define its own network API using socket interface.
- The benefits of any single API supported by industry are, the applications can be easily transferrable from one OS to another and the applications are simple to be developed.
- To describe socket interface, a set of services along with the syntax that can be provided on a particular computer system.
- Generalization is positively a goal of the socket interface.
- A good way to think of the socket is, at the point where a local application process gets connected to the network.
- The interface defines operations for
    1. Creating the socket
    2. Attaching the socket to the network
    3. Sending/receiving messages though the socket
    4. Closing the socket

1. Creating a Socket

int socket_fd (int domain, int type, int protocol)

The protocol family in the domain specifies 3 arguments namely, PF_INET - Internet family, PF_UNIX - Unix pipe facility and PF-PACKET - Direct access to the network interface.

The type argument indicates the semantic of the communication,

SOCK_STREAM_byte stream

SOCK_DGRAM_message-oriented service

The protocol argument identifies the specific protocol has been used,

UNSPEC combination of PP_INET&SOCK_SYSTEM

2. Attaching Socket to the Network

The attachment depends on the client and the server. A passive open is performed by the application process, on the server machine.

The server invokes 3 operations namely,

int bind (int socket_fd, struct sockaddr*address, int addr_ len)

int listen (int socket_fd, int backlog)

int accept (int socket_fd, struct sockaddr*address, int*addr_len)

To link the newly formed socket to the designated location, the link operation is used. Here, the local participant server addresses the network. The IP address of the server and the number of ports of the TCP is available at the address. The listening operation determines how many communications on the socket listed are pending. The accept procedure performs accessible passive. The blocking process will not return until the combination has been established by a remote participant. A new socket that is already in relation is returned once it is full and the address statement includes the address of the remote participants. An active open on the customer computer is performed by the application process, and a single operation given below is invoked by stating who seeks to make a connection,

int connect (int socket_fd, struct sockaddr *address, int addr_len)

The address includes the address of the remote participant. Typically, the client specifies only the address of the remote participant and the device fills in the local information. On a well-known port, the server listens to message.

3. Sending\Receiving Messages through Socket

In general, two operations are invoked by the application process once a connection is established, in order to send and receive the data messages,

int trx (int socket_fd, char*message, int mes_len, int flags)

int rcx (int socket_fd, char*buffer, intbuf-len, int flags)

Certain details of operations are controlled by both the operations taken as the set of flags.

1.6. Performance

Performance

- In network design, performance is an important factor for any computers.
- Two approaches measures network performance.
  - Bandwidth
  - latency
- Both are put in together, to define the performance of the given link.

Bandwidth

- "The number of bits that can be transmitted over the network in a certain period of time" is defined as the bandwidth of the network.

Bits Per Second (**BPS**)

Latency

- The time occupied by the transmit of messages from one end of the network to another end.

Round Trip Time (RTT)

- The consumption of time for message to travel between different ends.
- It has 3 components.

Latency=propagation + transmit + queue

Propagation = distance / speed of light

Transmit = size / bandwidth

Where distance = length of wire in which data travels

Speed of light = speed of light over that particular wire

Size = size of packet

Bandwidth = bandwidth at which is packet to be transmitted

Jitter

- Jitter is a parameter related to delay.
- Jitter time is the interval between the maximum effect (or minimal effect) of a signal in two periods.
- It is produced by electromagnetic interference and cross talking with other signal carriers.
- Different data packets encounter various delays.
- The data packets that hit the recipient at various times, triggering jitter.

Throughput = packet transfer size / packet transfer time

Transfer time = RTT + 1 / bandwidth + packet transfer size

Problems

1. If bandwidth is 10mbps, what is the bit duration time?

If bandwidth is 10mbps

The bit duration is

Bit duration=1/bandwidth

1/10*1000000

=10 microseconds

2. For I mb over a 1Gbps network with RTT 100 milliseconds, bind out the transfer time and throughput of the link

Transfer time =RTT+1/bandwidth*transfer size

=100ms+1/1Gbps*1MB

100+1/1*1000000000*1*1000000*8

=100ms+8ms =108ms

Throughput=transfer size/transfer time

=I MB/108ms

=1*8*1000000/108*1000

=74.1Mbps

3. Consider a p-p link 50 km in length. At what bandwidth would propagation delay (speed 2*100000000m/s) equal transmit delay for 100 bytes packet? What will be the 5/2 bytes packets?

Propagation delay = distance/speed of light

=50*1000/2*100000000 m/s =250 micro seconds

Propagation delay is equal to transmit delay

Transmit delay=size/bandwidth

=packet size/transmit delay

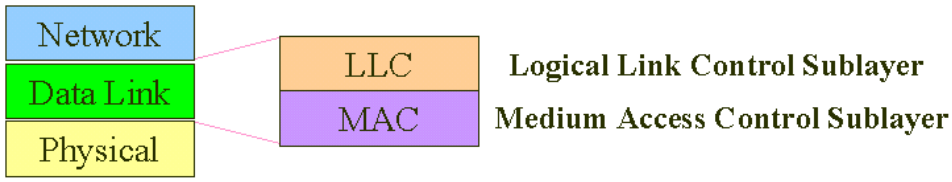
=800 bits/250 micro seconds

=3.2 micro bps

1000 bytes = 100*8 = 800 bits

1.7. Datalink Layer

The data link layer is responsible for converting a new transmission capacity into a link that is accountable for node-to-node (hop-to-hop) communication. The most important tasks of the data link layer are given below.



1. Logical Link Layer Access

Framing

Addressing

Flow control

Error control

2. Media Access Control

The layer of data links divides the stream of bits obtained from the network layer onto 1 manageable data unit called frames. A header is added to each frame at the data link layer in order to specify the frame addresses of the sender and receiver. By incorporating mechanisms for detecting and retransmitting defective, redundant or final frames, the datalink layer also adds stability to the physical layer.

1.8. Framing

Data transfer in the physical layer involves transferring bits from the source to the destination in the form of a single.

- The physical layer provides bit synchronisation to ensure the same bit and durations and timing are used by the sender and recipient.
- It packets bits into frames in the data link layer, which can be easily separated from each other.
- Framing the data link layer distinguishes the message from one source to the destination or from other messages to other destinations by adding the address of the sender and destination.
- The destination address specifies where the packet should go and the sender address determines which allows the receiver to recognise the receipt.

- If the frame is very wide, the flow and error management is very hard to perform efficiently.
- A 1 – bit error can cause retransmission of messages even though it is a big datagram.
- When a message is broken into smaller frames, only the small frame is influenced by a single bit.

There are two types of framing.

1. Fixed size framing
2. Variable size framing

1. Fixed Size Framing

Here the boundaries of the frames need not be defined**.** The size acts as the delimiter.

E.g: ATM wide area network, Frames of fixed size called cells.

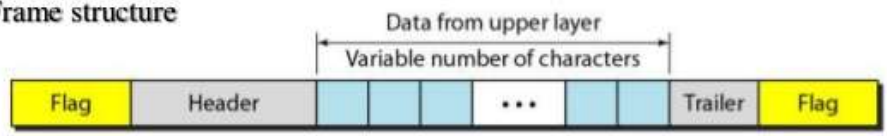
2. Variable Size Framing

Here the end of the frame and also the beginning of the next frame are defined. There are two types - Character oriented approach and Bit oriented approach.

a. Character Oriented Approach

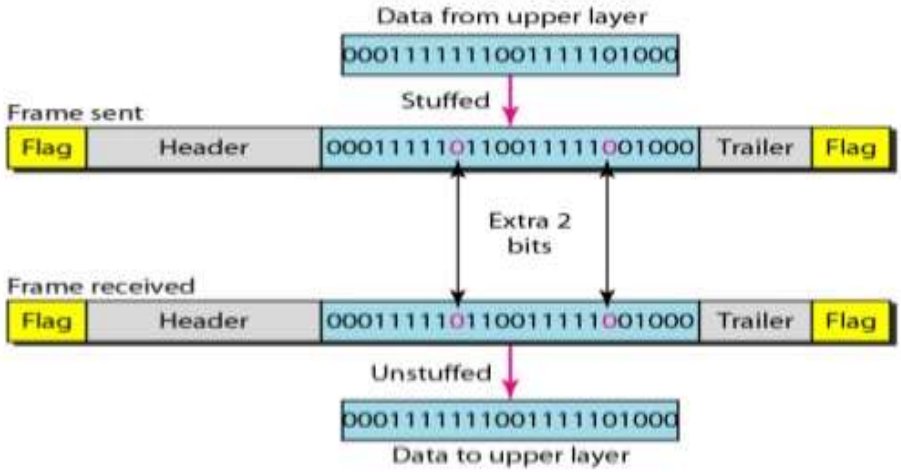
- Data carried from a coding scheme, such as ASCII, with 8-bit characters.
- The header carries the addresses of the source and destination and other control information, and the trailer carrying error detection or error correction redundant bits also has a number of 8 bits.
- To distinguish one frame and the end of the frame from the next frame.
- A flag composed of special characters based on the protocol, signals the start or end of the frame.
- Only text was exchanged in character-oriented framing via the data link layers.
- Byte stuffing (stuffing of characters) is a special byte applied to the frame's data segment as there is a character of the same pattern as the flag.
- There is an extra byte in the data segment called escape character (ESC). Insertion of 1 additional byte to the frame, as a flag or escape character is byte stuffing.

- Frame structure



b. Bit Oriented Approach

- A series of bits to be interpreted by the upper layer is the data portion of a frame.
- Delimiter is used separate frames from one another.
- Most protocols use a special flag 01111110 of 8-bit pattern which defines the beginning and the end of frame, as the delimiter.
- Here flag creates with byte_ oriented protocols.
- Bit stuffing is the process of inserting an extra 0 for every five consecutive times following a 0 in the data, so that the receiver does not mistake the 01111110 pattern for a flag.



1.9. Error Detection

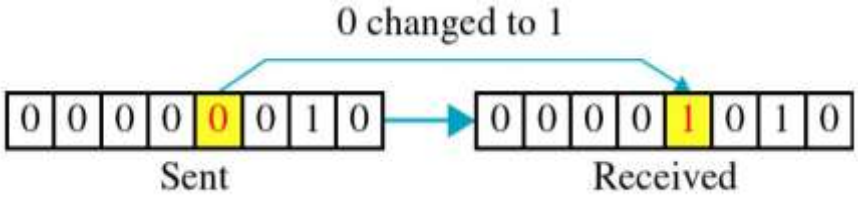
Errors

- As bits are transferred, certain unforeseeable changes occur due to interference.
- This can change the signal shape that some applications need to find and resolve errors.
- There are 2 types of errors.

1. single bit error
2. burst bit error

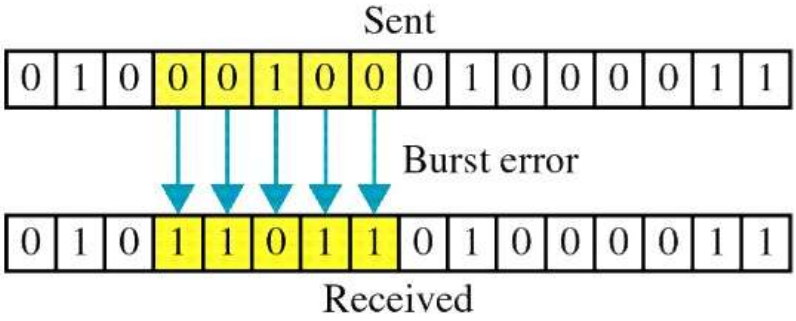
1. Single Bit Error

- Single-bit error means that only 1 bit of the data unit (byte, character, packet) has been inverted as 1 to 0 or 0 to 1.
- Only one Bit Data Unit has been modified in single bit error.



2. Burst Error

- A Term error means that the data unit has inverted 2 or more bits.
- When more bits are changed in the data device, it is called as Burst error.



Redundancy

- The principle is redundancy in identifying or fixing errors.
- Extra bits of original data (redundant) are needed to detect or correct errors.
- The sender attaches some unnecessary bits and the recipient eliminates them.
- This helps the receiver to spot corrupted bits or their degradation.

Error Detection

In frames, bit errors are inserted. Detecting mistakes is just one aspect of the issue, and error correction is another issue. There are two simple approaches taken. Where a message receiver discovers an error. When the sender sends the damaged message, a copy of the message can be retransmitted. If bits are uncommon, then the retransmitted copy would most likely be error-free. In certain forms of error detection, the algorithm requires the receiver to rely on Error Correcting codes to recreate the correct algorithm. In almost all connect level protocols, 2-dimensional parity and check sums are used. Cyclic redundancy check (CRC) is employed. The basic concept of error detection scheme is to appends details about redundancies, to determine if errors have been introduced.

Example

- If the receiver finds two copies, then both are accurate.
- If they vary, an error has been put into one or both of them and discarded.
- Two factors for weak identification of errors.
    1. It sends n redundant bits for an 1-bit message.
    2. Any mistake that appears to corrupt the same bit positions in the first and second copies of the message is undetected by several errors.
- The main aim of error detection codes holds high likelihood detection of errors.

Error Detecting Codes

- No new information will be added if the bits are redundant.
- Extracted by some well-defined algorithm directly from the original post.
- The algorithm is well-known to both the sender and recipient.
- The redundant bits produced may use the message algorithm.
- Both the message and a few additional bits are evoked.
- The same outcome of the sender is expected at the recipient when the same algorithm is applied.
- It compares the outcome with the one that the sender sends to it.
- If they match, it can be inferred that during transmission, no errors were inserted in the message.
- They are referred to as codes that detect errors.

Checksum

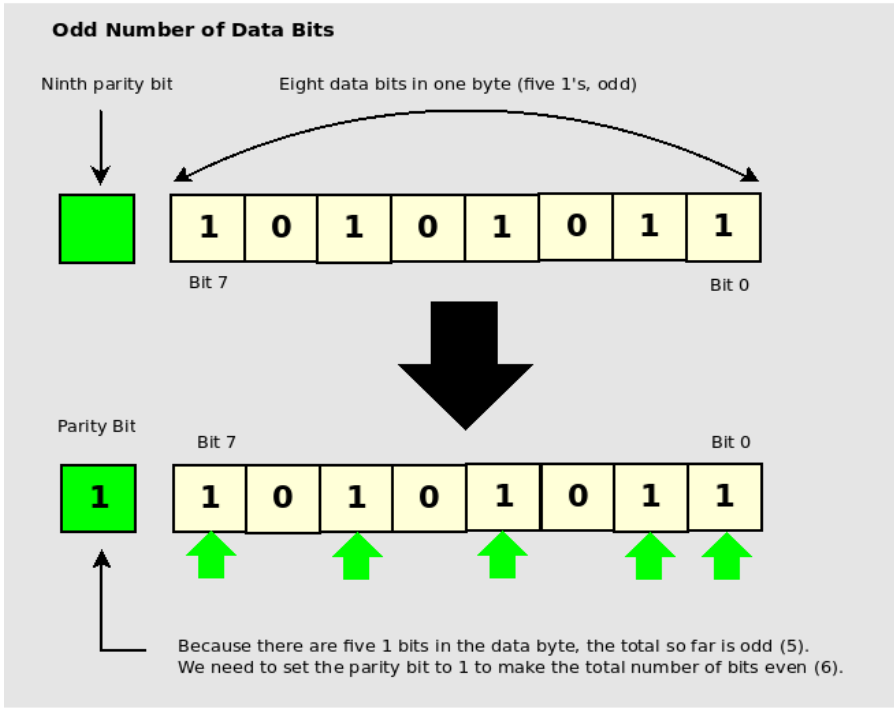
- A checksum can be named after the algorithm used for generating code gets applied.
- It is an error check which utilizes an algorithm for summing.
- The term checksum is sometimes used imprecisely to denote any type of code that detects errors like CRCs.

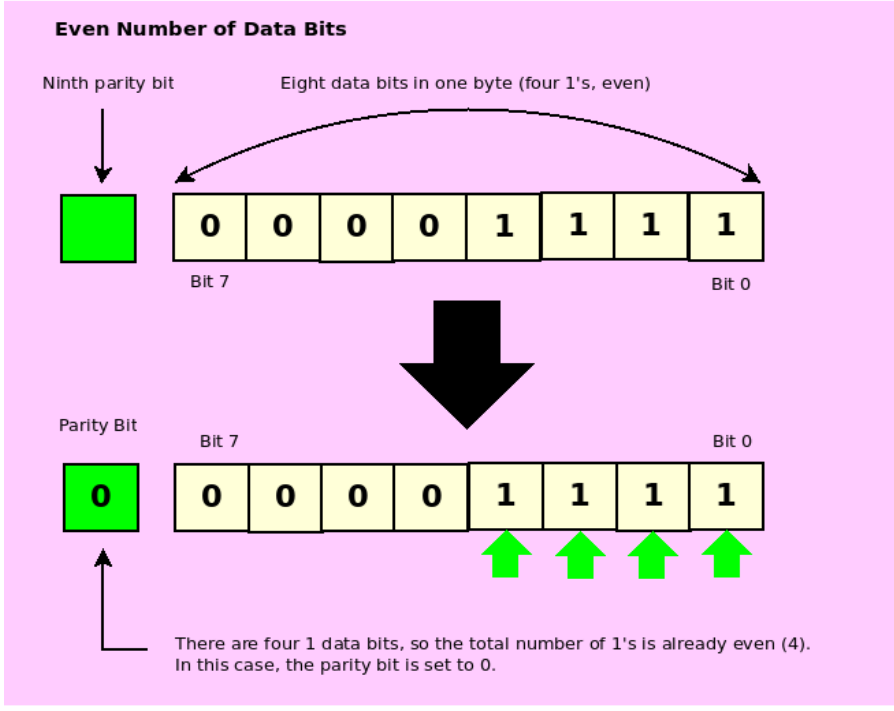
Two-Dimensional Parity

It is based on a "simple" (one-dimensional) parity typically involving adding an extra bit to a 7-bit code to match the number of 1s in the byte. In each location of bits, the two-dimensional parity manipulates in a similar way and holds a bit every byte as parity.

Even Parity

Even parity ensures that the number of 1 bits (8 data bits + 1 parity bit) is even.





Internet Checksum Algorithm

- It offers the same kind of accessibility and parity as the CRCs.
- It adds up all the words that are transmitted in the internet checksum, and then transmits the results of that number. This is called checksum.
- On the received data, the receiver performs the same calculation and the results are compared with the checksum received.
- The results end up in a mismatch when any of the data transmitted is corrupted that includes the checksum too, which intimates discrepancy to the receiver.

Cyclic Redundancy Check (CRC)

- The key objective in developing algorithms for error detection is to increase the possibility of detecting errors using only a limited number of redundant bits.
- The conceptual underpinning of the CRC is embedded in a mathematical branch termed as the finite fields.

Error **correction** appears to be most beneficial when:

1. Errors are relatively reliable.
   E.g: wireless environment.
2. The rate value of retransmission is too high.
   E.g: satellite link.
   - Although error detection involves sending more bits when mistakes occur, error correction requires sending more bits all the time.
   - The use of networking error correction codes is referred to as forward error correction (FEC) as error correction is done in advance by transmitting additional details, rather than watching for problems to occur and grappling with it later through retransmission.

Eg:- 802.11 (wireless network)

- It involves vital methodologies like.
  1. Acknowledgements
  2. Timeouts
- An acknowledgment (ACK) is a tiny control frame sent back to its peer by a protocol indicating that it has received an earlier frame.
- Retransmission of the originating frame happens when the sender fails to acquire a receipt for a rational amount of time. This is often referred as a timeout.

- Identification of a data frame sending back in the opposite direction and receipt of acknowledgement is known as a piggy bank.

Error Control

- Error management is the detection of errors as well as error correction.
- It enables the recipient to notify the transmitter of any frames missing or disrupted in the propagation and schedules the sender's retransmission of those frames.
- Error management prefers error detection and retransmission approaches.
- Automatic Repeat Request (ARQ) is termed as the occurrence of errors obtained during a specific frame is transferred.
- Data link layer error management is based on the Automatic Repeat Request (ARQ), which is data retransmission.

1.10. Flow Control

- Flow control can be explained as the amount of information that can be controlled at the submission even before an acknowledgment is received.
- It is a collection of processes that involves in instructing the sender how much data can be transmitted before awaiting a receiver's acknowledgement.
- Data flow is restricted to overpower the receive.
- The recipient system is offered a limited speed to process the incoming data.
- It is also responsible for warning the transmitting system and request sending a lower number of frames or temporarily stop them, when the limit is observed.
- Reviewing and analysing the arriving information is to be taken place before it can be used.
- The rate of such processing is habitually leisurelier than the rate of transmission.
- Each receiving system has a memory block called a reversed buffer to store the incoming data before it is being processed.
- If the buffer starts filling up, the receiver must be able to tell the transmitter to interrupt transmission before it can be received again.
- Set of trials to limit the quantity of message sent by the sender unless awaiting the receipts.

CHAPTER 2

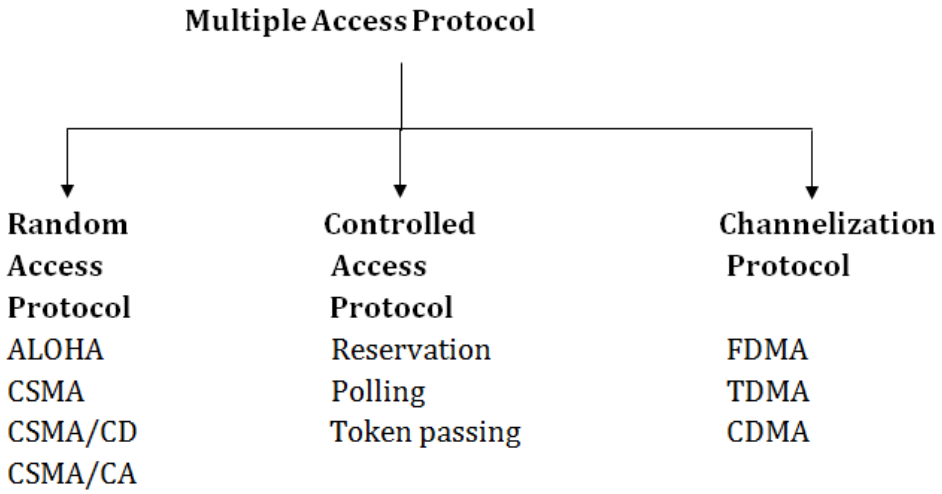
2. Media Access and Internet Working

Objectives

- To understand about Media Access Control and Internetworking.
- To explain wired and wireless networks.
- To know about switching and bridging.
- To explain about basic internetworking – IP, ARP, RARP, ICMP, BOOTP, DHCP.

2.1. Medium Access Control (MAC)

A multi-access medium based computer network needs a protocol for efficient media sharing. A Multipoint is formed when connecting more nodes or stations using a direct bond. It is otherwise called as broad cast connection. A protocol which can manage synchronized multiple access to the link is needed.



2.1.1. Random Access Control

All the stations are given equal priority or importance in any random-access system. There cannot be any station given authority over the next station in contention system. No node allows or hinders another node from sending data. A station with a data decides whether to send it using a protocol-defined procedure. This shows dependency on the environment, whether it stays idle otherwise it is busy. The protocol checks for the availability and each station will transmit when it wishes.

Features

- A station does not have a scheduled time to transmit. Between the stations, transmission is random, so it is called random access.
- No ruling stated to compete such as which station to send and which one to access the medium and it is called Contention method.
- In random method of access each station is entitled to the medium for better management.
- If more than one station attempts to submit a conflict of access and the frames are either lost or changed.

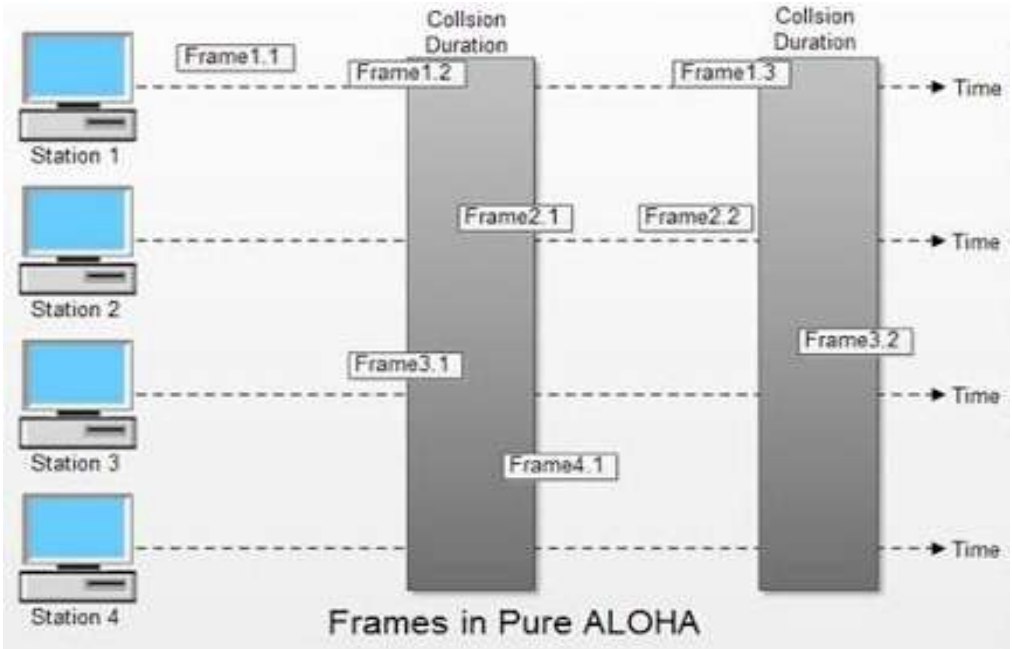
- Multiple Access is the methodology observed by ALOHA.
- The method approved with the addition of the method facing the station is called Carrier Sense Multiple Access (CSMA) that senses the medium prior to transmission.
- There are two parallel CSMA approaches that are Carrier Sense Multi Access with Collision Detection (CSMA / CD) and Carrier Sense Multi-Access with Collision Avoidance (CSMA / CA).
- Appropriate station is informed with the remedial action when collision has occurred by CSMA / CD and CSMA/CA avoids collision.

a) ALOHA

It is the initial of method for accessing the data randomly, established in 1970 at the University of Hawaii. Wireless radio LAN was its major target and also it worked out well with a medium in sharing with others too. When a station transmits information, a different station can simultaneously attempt to do just that. The two stations' data clash with each other.

Pure ALOHA

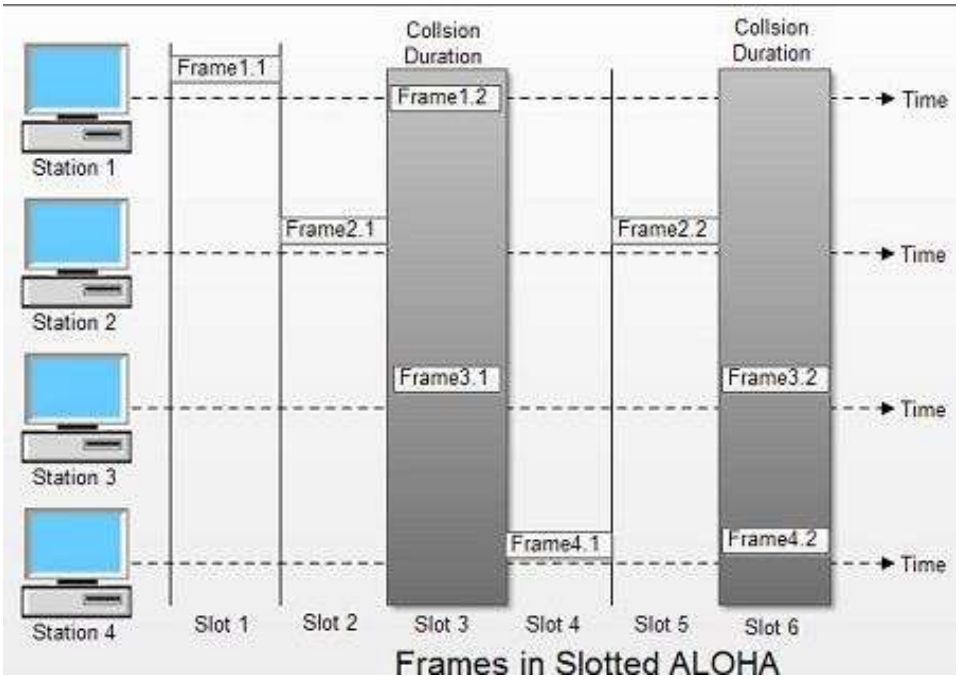
- It is the protocol which stands modest and effective.
- If it has data to send, each station transmits it as a frame through the single channel available that ends up in collision. It may involve more stations too.



- Collision happens when one bit of a frame collocates with another frame and the frames are lost.
- This protocol is based upon receiver acknowledgments.
- An acknowledgement is necessary, for a station that transmits a frame.
- On delay of receiving the acknowledgement, the sender retransmits the same frame, with the prediction that the previous one is missed on ten go.
- The frames would clash again while trafficking with resending of the same frames from these stations.
- Pure ALOHA dictates wait for the station for some duration and instructs to retransmit the data frames after the pause.
- The randomness helps prevent further collisions.

Slotted ALOHA

- The time in slotted ALOHA is divided as time force and force slots and instruct the node to submit by start of time slot.
- On this, channel is permitted to deliver at the start of the coordinated time slot, if a station fails that moment, it must wait until the start of the next time slot.
- The station that began to transmit by the start of time slot has completed frame transmission.



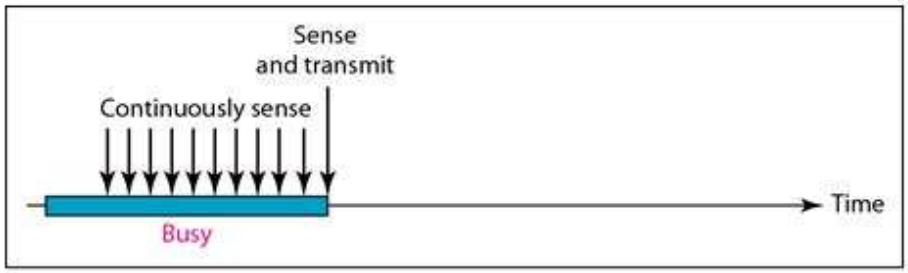
b) Carrier Sense Multiple Access (CSMA)

- This protocol operates on carrier sensing principles. A station listens to the nature of the transmission (carrier) on the cable in this protocol and intends to behave accordingly.
  - Non - Persistent CSMA.
  - I - Persistent CSMA.
  - P - Persistent CSMA.

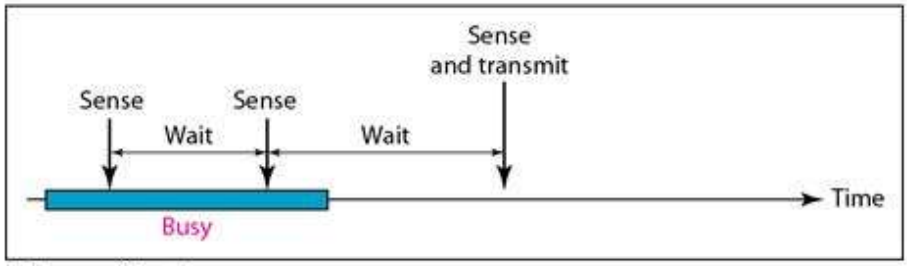
- **I-persistent CSMA**
  - A modest and straight forward method.
  - The frames are sent immediately after the station finds the line idle.
  - This approach leads to maximum collision, as two or more stations can idle the line and start sending their frames.

- **Non-persistent**
  - The transmission of data is done in this mode automatically, when the line stays passive.
  - For a predicted time duration it holds on and when the connection becomes active, further checks for medium's availability.
  - Decreases collision, since the nodes defer and then trace the path for submission concurrently.
  - Decreases network reliability, since the medium remains idle while there are several frame stations to transmit.

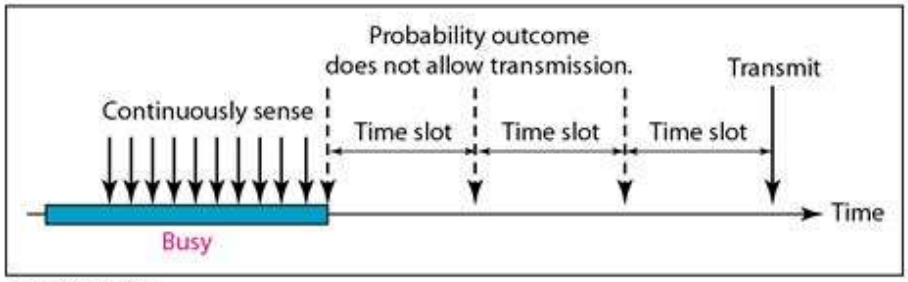
- **P-persistent**
  - This method works when the medium holds time slots of length same as or bigger one to the longest time of propagation.
  - P-persistent methods incorporate the benefits of those two other methods.
  - It decreases the risk of collision and proves efficiency.



a. 1-persistent



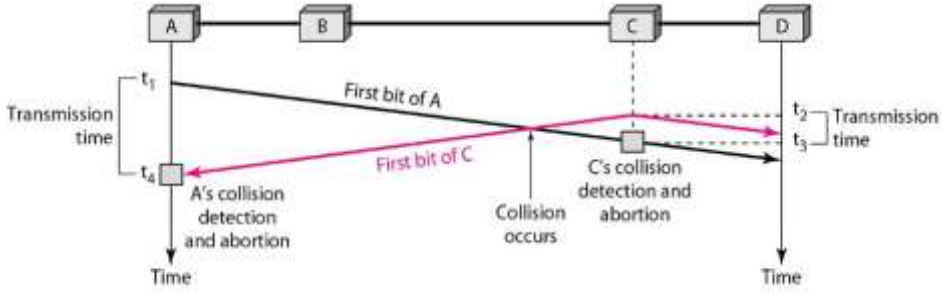
b. Nonpersistent



c. p-persistent

c) Carrier Sense Multiple Access with Collision Detection (CSMA\CD)

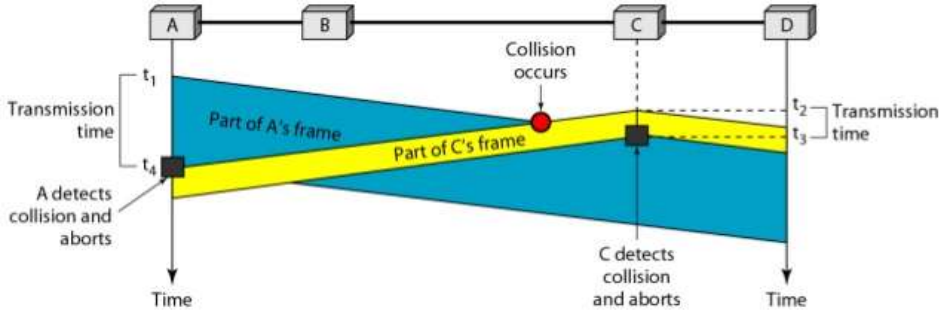
The clarity of an algorithm is not specified with this CSMA method, as how to resolve collision by employing its CSMA/CD arguments. A station tracks the medium in its system post the transmission of a frame to see if the transmission has been carried out properly. Then the work gets done or it resends the frame if collision occurs.



Time

- Using persistence procedure Node, A executes its procedure at time t1, and begins to submit frames.
- Node C is yet to receive the initial bit which was transmitted by A at t2.
- C performs the process of perseverance so that begins transmitting bits on both directions.
- After t2, collision occurs at t3 and it is identified by C as initial bit of frame A is delivered. So, C shall abort transmission.
- By t4, A understands that collision has occurred as the initial bit of C is received, and so it is instantly aborted.
- Finally the transmission is resolved as A during t4-t1 and C during t3-t2.

Restriction on frame size should be included in CSMA / CD. Collison should be detected and the transmitter must abort it before transmitting end bit of the data frame. On transmitting complete frame, no more back up is available, the monitoring goes to halt state. The transmission time of Tfr should be minimum twice the longest time of propagation Tp = 2Tp.

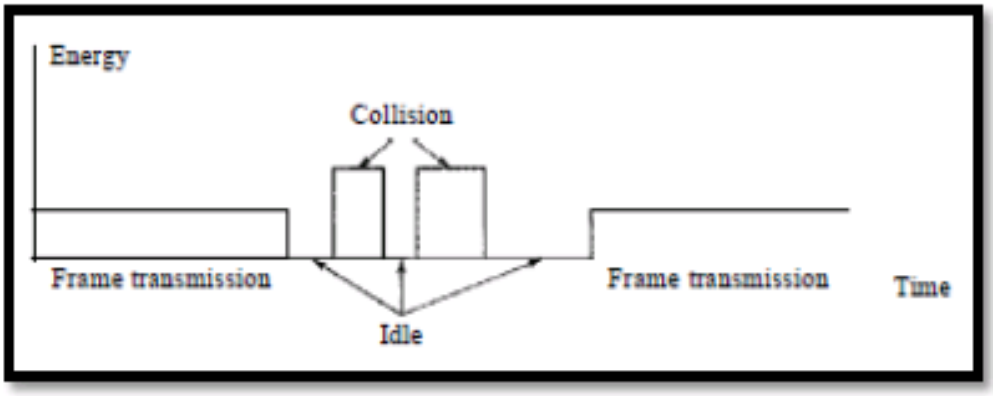


CSMA / CD is similar to the protocol for ALOHA. Until we begin sending the frame employing any of the persistence procedure, the persistence mechanism is used to sense the channel. Transmission is used to send the whole frame, and hold on to receive an acknowledgment and detect collision as well. In the event that other stations have not yet detected the collision, the jamming signal is being used to trigger the collision.

Energy level's of a channel has 3 values as: Zero, Normal, Abnormal.

- **Zero level** - the channel is idle.
- **Normal level –** the channel is acquired and has started transmitting the frame.
- **Abnormal level -** the normal energy level is doubled with collision.

A station with a frame to submit needs to track the energy level to decide if the channel is active, idle or in collision mode.



Energy level during transmission, idle and collision

CSMA/CD's throughput is max of ALOHA control or otherwise it is considered to be a slotted one. The max throughput occurs at a different value of G, which holds dependency caused by process of persistence while the value of p refers to p-persistence.

d) Carrier Sense Multiple Access/Collision Avoidance

As a collision is detected, the CSMA / CA station must be capable of receiving when transmitting. The transmission of its data occurs with a station when it receives a signal of non-collision and during collision, 2 signals are received, either its signal or a signal received from other station.

The obtained signal must be substantially different in these two situations. The signal of the second station has to append a considerable amount of energy to that provided by previous station.

The energy of the signal stays unchangeable to that of the energy preserved during transmission in **wired network**, since either cable duration seems low or repeaters to amplify the energy exists with sender and receiver. Most of the transmitted energy in **wireless network** is lost in transmission. There is very little energy in the received signal.

CSMA/CA is mainly invented for wireless network.

1. **Frame space**
2. **Contention window**
3. **Acknowledgements**

Inter Frame Space

- Even if the channel is found idle, collision is prevented by deferring transmission.
- The identification of a channel staying idle will not instantly mount the process.
- It awaits interframe space (IFS) for a period of time.
- The IFS time enables this station to be read from the pre-positioning of the signal from a station.
- The IFS variable may also be used to set the preferences for stations or frame styles.
- Continuing to be idle after the IFS time the station will give, but if it still has to wait the same time as the time of the contention.
- The IFS can also be used in CSMA / CA to describe a station or frame's priority.

Contention Window

- The slots are segregated to form the window of contention.
- A ready-to-send station selects any slot for its idle time.
- As per the binary exponential back-off technique, the slot's count in the window varies.
- For the first time it takes up one slot and doubles after the ifs duration every time the station does not identify an inactive stream.
- The station monitors the channel in contention window at the end of every time slot.
- If the station finds the channel busy, rather than restarting the process it just ends the timer and restarts it while the channel is noticed to be idle. This provides preference to longest waiting station.
- In CSMA / CA, if the station notices the channel busy, the contention window timer doesn't restart, the timer restarts when the channel becomes passive.

Acknowledgement

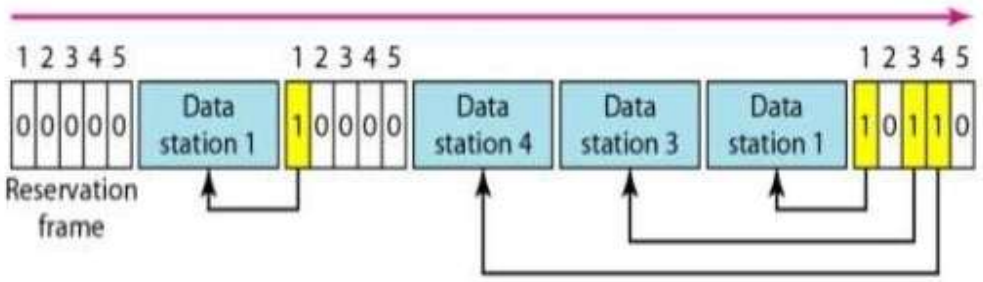
- Collision results in data loss.
- Data can be compromised during forwarding.
- As the recipient receives the data frame, a positive acknowledgment and time-out timer is initiated.

2.1.2. Controlled Access Control

In controlled access the station consults with each other in identifying the appropriate station to transmit. Any identified station doesn't be granted priority to send without authorization from connected stations.

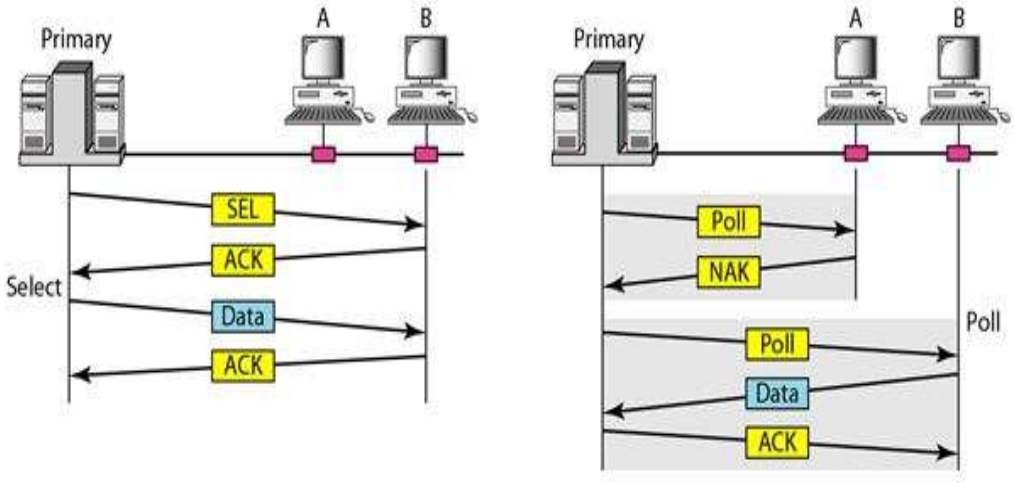
a) Reservation

During this process, reservation had been made by the station before sending data. Intervals are obtained by splitting the time and usually the data frames the reservation frame within that duration. Having the availability of N stations in the system, the reservation frame is exactly N reservation mini slots. Each mini slot is station owned. Therefore, whenever a data frame has to be sent by a station, its own mini slots are reserved. The stations with these reservations hold the authority to send the required data frame that is usually preceded by the reservation frame.



b) Polling

This methodology operates based on topology where each unit is chosen as either a primary or a secondary station. And when the ultimate destination is a secondary device, all data exchanges must be performed via the primary device. The first device manages the connection and it is followed by the secondary device. The primary computer is always session initiator.



- If primary wants to receive data, it tells the secondary to get ready to receive.

Select

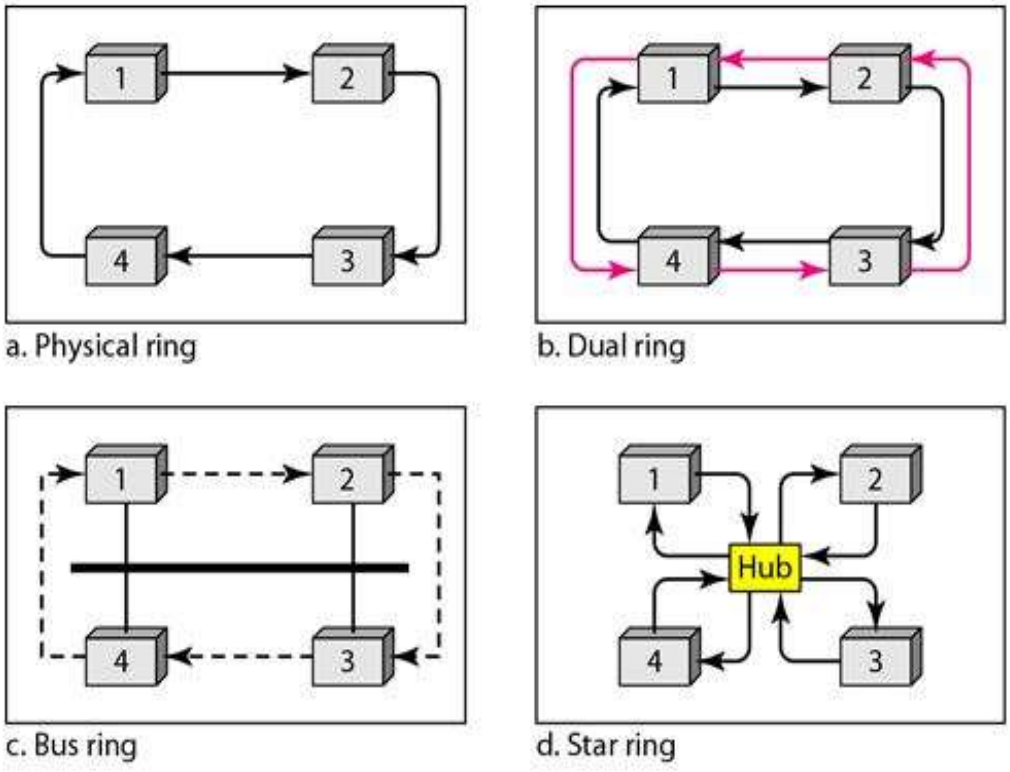
- If the main computer has anything to send, the select function is used.
- If it has anything to attach, attach it to the primary computer.
- The primary must alert the secondary to the upcoming transmission and wait for the secondary's ready state to be recognised.
- A select (SEL) frame is created and transmitted by the primary before sending the data, and address of the secondary is held by a field.

Poll

- Whenever a primary device attempts to invoke a transmission from any other secondary devices, this poll mechanism is utilized by the primary device.
- Each system must seek to POLL, if it needs to send, whenever the primary data is ready for receiving.
- When the first secondary is contacted, the response might be a NAK or any data frame.
- Assuming reflex is negative with NAK, the first one elects the next one similar to it, till it identifies the station with transmittable data.
- In case of a positive answer, the first one reads data frame and sends an acknowledgment (ACK) confirming its reception.

c) Token Passing

On a network connectivity, stations are arranged in such a way that it appears to be a logical ring. There exists a predecessor and successor stations for every station. The channel can be accessed at the current station. A special packet called a token, flows through the ring owning to token grants the station the might to view and transfer the data to the station. Therefore, a station had to wait before sending a data until its predecessor receives the token. The token is preserved till the results are sent and is released once it is left out with no more data. Unless the token is received in the next round, the station is unable to send data. When a station receives the token in this process and has no data to send, it only transfers the data to the next station. The token needs to be controlled to make sure it is not lost or ruined. Usually, the tokens are controlled by the token management and its primary role is to allocate the station priorities and the types of data for transmission. This token management is highly required for identifying high priority stations, where the token held by it is higher than the token held by other stations.



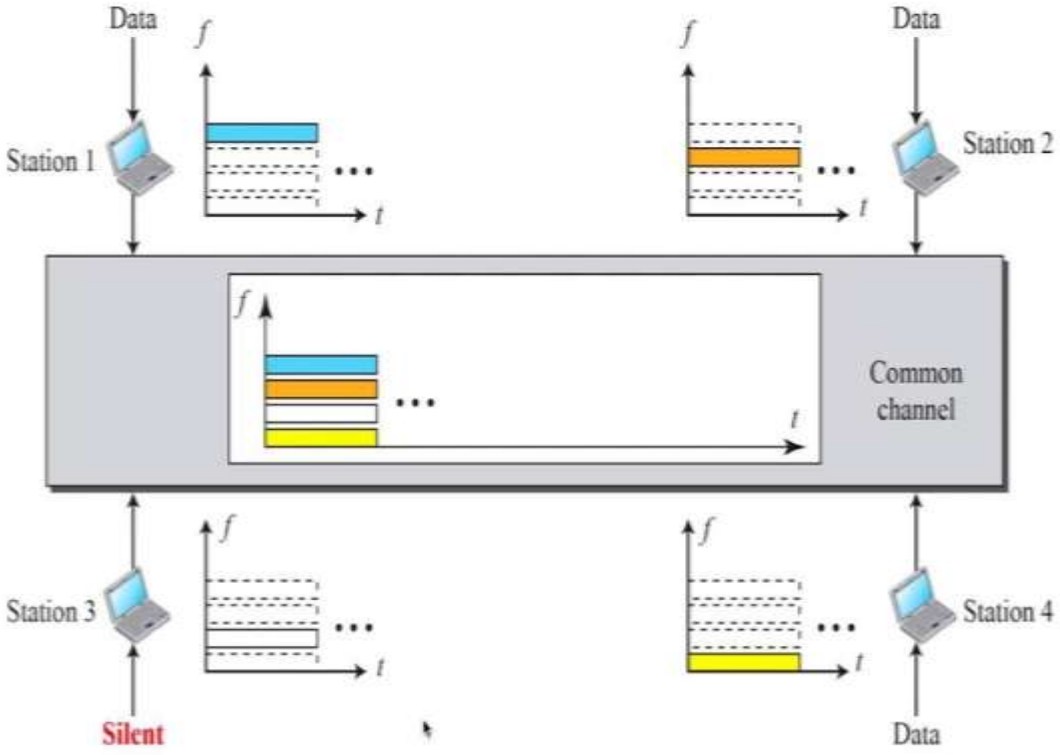
2.1.3. Channelization

It is a multiple access method where the bandwidth available for establishing a connectivity can be shared either by frequency or time or code.

a. FDMA → Frequency Division Multiple Access.
b. TDMA → Time Division Multiple Access.
c. CDMA → Code Division Multiple Access.

a) FDMA – Frequency Division Multiple Access

In FDMA, the accessible bandwidth is split as different bands of frequency and a band is reserved to each station for data transmission. The transmitter frequencies are confined by means of a band pass filter. To avoid station interfaces, tiny guard bands separate the assigned bands from each other. In FDMA, guard bands segregate the bandwidth of shared channel.



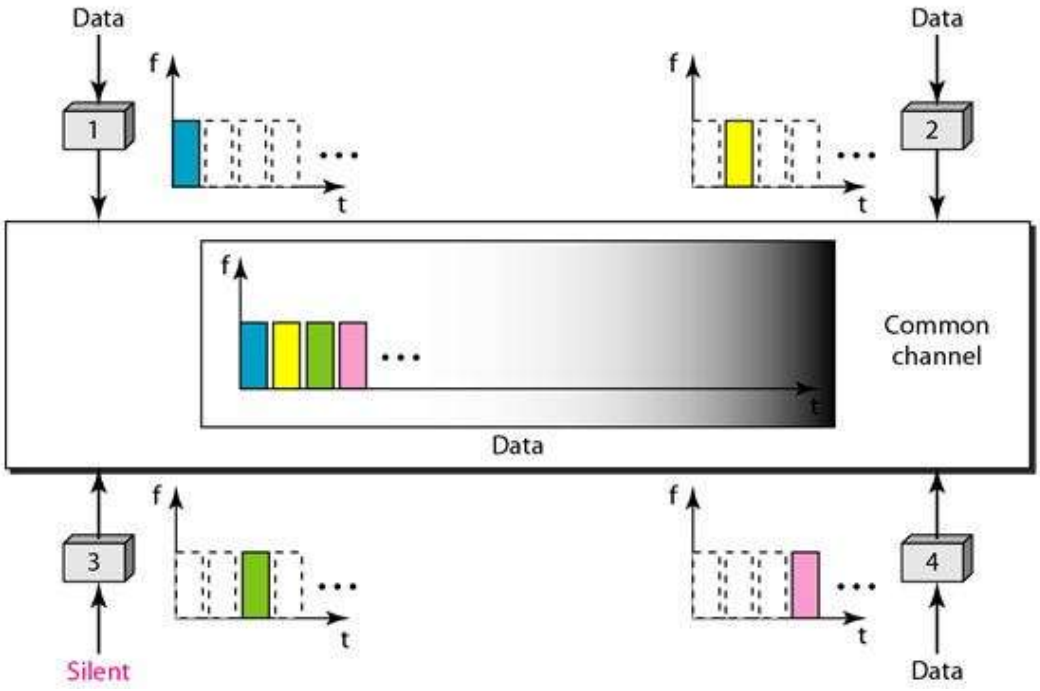
- FDMA specifies a predefined frequency bands or entire duration of and established connectivity. Eg: Cellular telephone systems.

Here, the physical layer techniques integrate low-bandwidth channel loads. Low-pass is the channels that are integrated. The multiplexer modulates, integrates and generates a band pass signal for the signals. Every channel's bandwidth is transferred by the multiplexer. The access method tells each station in its physical layer in the datalink layer to propagate a band pass signal out if data passed to it. In the allocated band, the signal must be generated. The physical layer does not contain a physical multiplexer. Automatically band pass-filter filters the signals produced at each station. When they are sent to a popular channel, they are mixed.

b) TDMA-time Division Multiple Access

The stations share the channel's bandwidth over time in this methodology. A time slot is assigned to each station during which it will be submitting data.

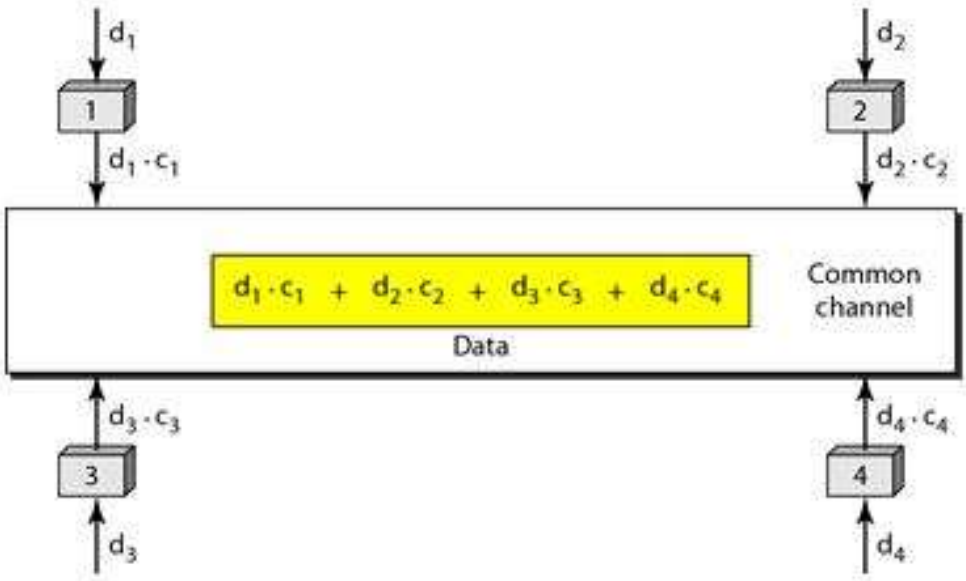
Every data is sent during the allotted slot of time duration.



The primary issue with TDMA is to achieve synchronisation between the various stations. The station must be alert about the start and end of the time durations along with its location. If the stations are scattered over a wide area of propagation delays, the delay guard times are added to compensate. Synchronization is typically done by providing some synchronisation bits by the start position. TDMA holds the bandwidth as a channel which shares among various stations over time. In the physical layer, the slower data is combined and transmitted using a faster source. The interleaving of data units are accomplished by using multiplexer. In the data link layer, TDMA is an access process and conveys the physical layer to occupy the designated slot of duration.

c) CDMA-code Division Multiple Access

In code division multiple access, it occupies only an allotted channel of bandwidth in the link. It sends all the data simultaneously in no time-sharing mode. In CDMA, a channel transmits all data concurrently.



For Example

Four stations 1,2,3,4 as d1, d2, d3, d4

Codes c1, c2, c3, c4

There are 2 properties

1. The multiplication of each code by another, results in 0
2. The multiplication of each code by itself, results in 4

   Message = d1.c1 + d2.c2 + d3.c3 + d4.c4

2.2. Ethernet (802.3) or Wired LAN

LAN - Local Area Network

The LAN has numerous connecting methodologies stated as Ethernet, token bus or ring, FDDI, ATM LAN etc. In 1985, IEEE developed 802 project and ANSI adopted in 1987. The physical standards are established by subdividing data link layer as two other layers by IEEE.

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

Physical Layer

It depends upon how it is implemented and the types of physical media used. The specifications for all LAN are detailly defined by IEEE.

Datalink Layer

As per the IEEE standard, data link layers can be divided as LCC and MAC.

1. LCC - Logical Link Control

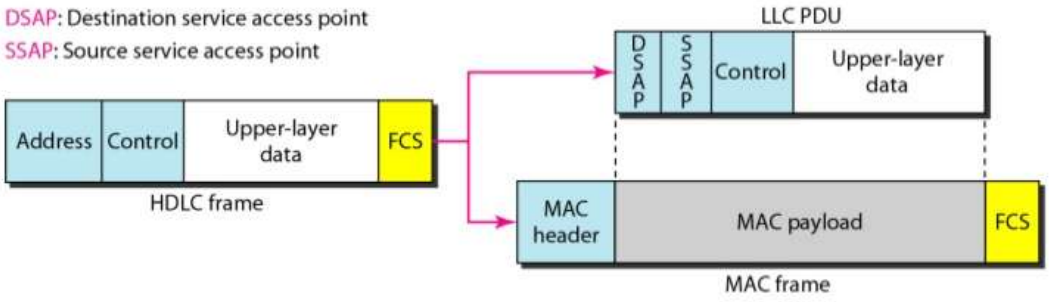
The framing, control over the data flow and errors are managed by data link control. In IEEE project 802, flow control, error control and certain framing duties are dealt in logical link control. In both LCC & MAC, framing is done. For all IEEE LANs, the LCC provides a single data link control protocol. Different LANs are provided with various protocols. One LCC Protocol offers interconnection between various LANs, as MAC sublayer performs the transport operation.

Framing

- LCC defines a Protocol Data Unit (PDU).
- The leader contains a control field of both error and flow control.
- The protocol of higher layer and LCC are defined at the source and destination respectively by the other two header files. These fields are called as Destination Service Access Point (DSAP) and Source Service Access Point (SSAP).

Need for LCC

The purpose of LCC is to provide flow and error control for the upper layer protocol that actually demand their service. LCC facilitates error control, flow control over protocols of application layer. IP is not used as the service provided by LCC in upper layers.

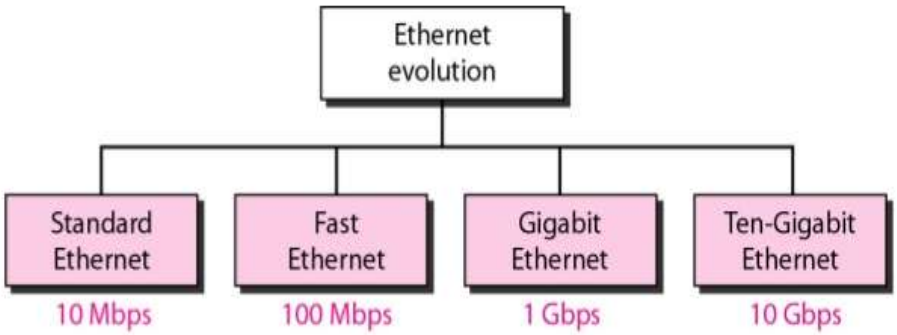


2. MAC - Media Access Control

- It includes random access, controlled access and channelization.
- A sub layer called MAC has been developed by IEEE project 802 with the unique access method which is specified for each LAN.

- It identifies CSMA / CD as the Ethernet LAN media access system and the LAN token ring and token bus system of token passing.
- A variety of distinct modules are included in the MAC layer. They are specifying the method of access and the unique format of framing in LAN.

2.2.1. Evolution of Ethernet



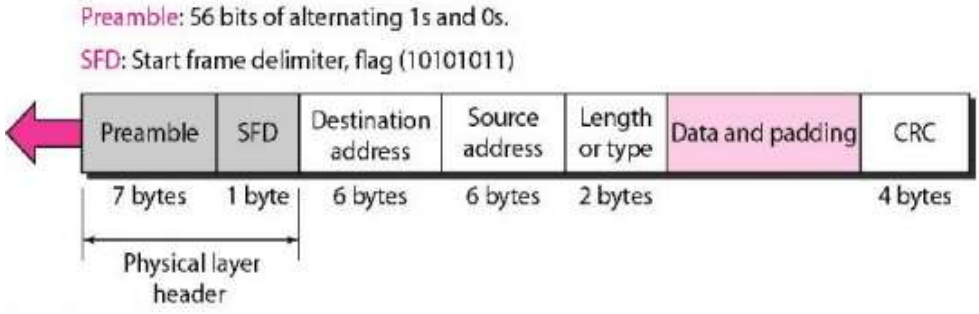
1. Standard Ethernet

MAC Sublayer

- All the operations of the methods are administrated by MAC sublayer.
- The data of the upper layer is framed and transferred to the physical layer.

Frame Format

- The frame of ethernet has 7 fields.



Preamble: 56 bits of 0's and 1's

SFD: start frame delimiter flag.

Preamble

The first frame field has 7 bytes of the 802.3 and are alerted by the receiving device and allows synchronise the timing of the input.

Start Frame Delimiter (SFD)

The next field that is obviously the second is of 1 byte and signals the frame's beginning.

The SFD frame alerts the station of synchronisation or lack of opportunity.

It also alerts the receiver that the next field is the destination address.

Destination Address (DA)

The DA field is (6 bytes) and includes the physical address of the packet receiving station or station.

Source Address (SA)

The SA field is (6 bytes) and includes the physical address of the packet sender.

Length Type

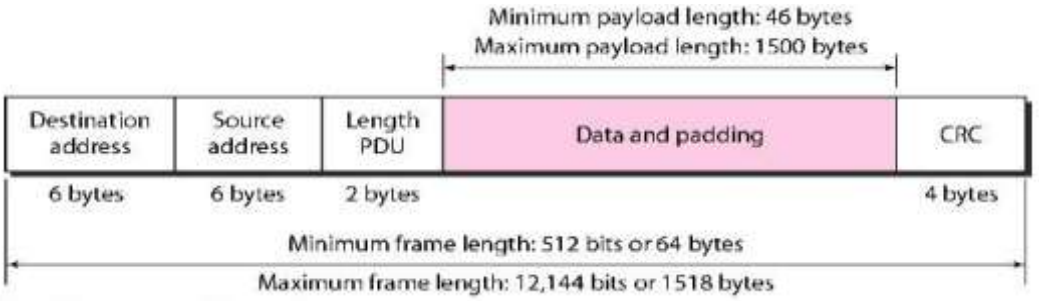
It defines the type field or length field.

Data

This area carries data encapsulated from protocols of the higher layers. 46 bytes minimum and -1500 bytes maximum.

CRC

This field includes knowledge about error detection.

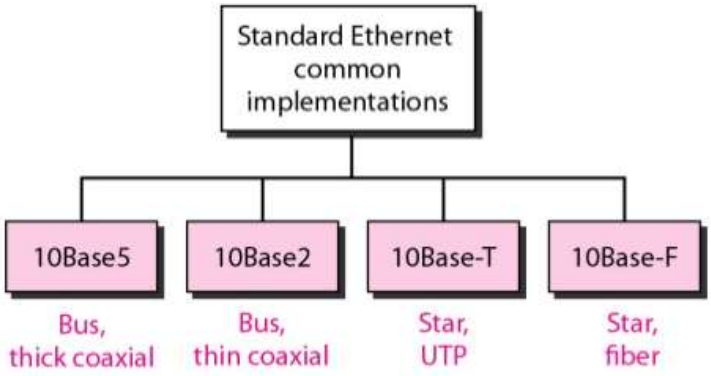


Addressing

Every station holds unique Network Interface Card (NIC) on Ethernet. Inside the station, NIC offers a physical address of about 6-bytes, to station. This is specified by the least significant bit of the first byte. For unicast the bit is 0 and multicast, otherwise. The link between the sender and the receiver, determines the destination address of the unicast that is available with one recipient. A multicast destination address identifies a collection of

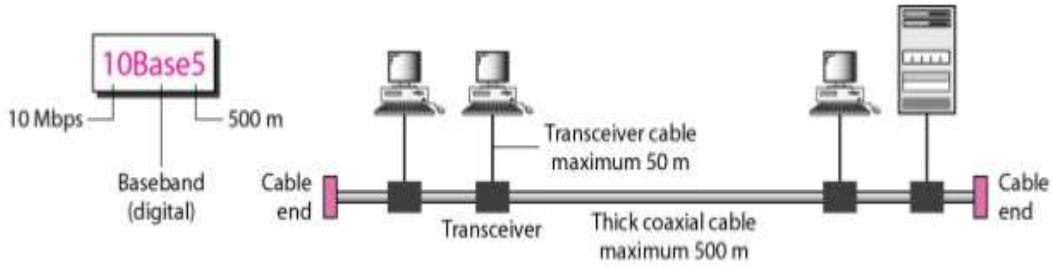
addresses, one to many are the links between the sender and receive. The destination address of the broadcast holds the bits as IS in a special sect of a multicast address.

2.2.2. Physical Layer



10 Base 5: Thick Ethernet

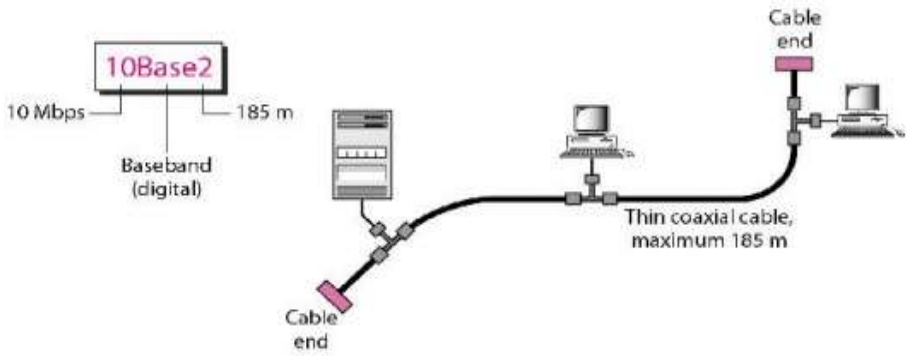
This is a thicknet → This name is formulated by referring to the size of the cable. Ethernet specification uses bus topology with transceiver connected to a cable with thick coaxial, through a tap.



The responsibilities of transceiver are transmitting, receiving and also collision detection. It is associated to the cable that provides path to send and receive data through coaxial cables.

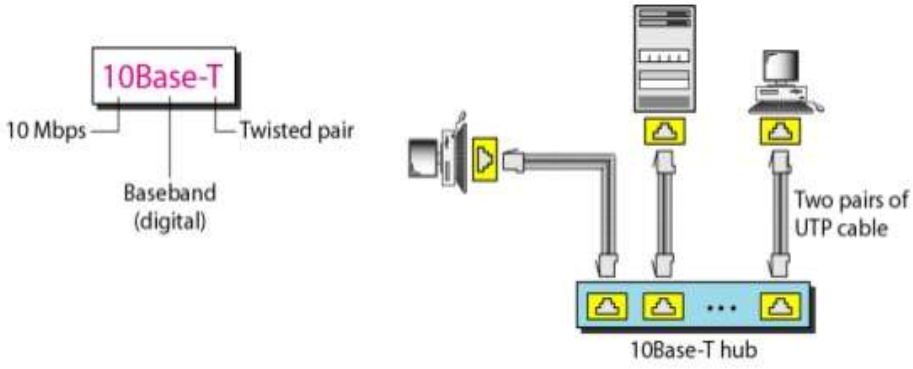
10 Base 2: Thin Ethernet

10 base 2, thin ethernet, cheaper net → It is connected through bus with thinner and supple cable. The cable will have the ability to bend to pass though the stations very close by. The transceiver is considered as a part of NIC installed in station. It is highly cost effective when compared to other 10 base.



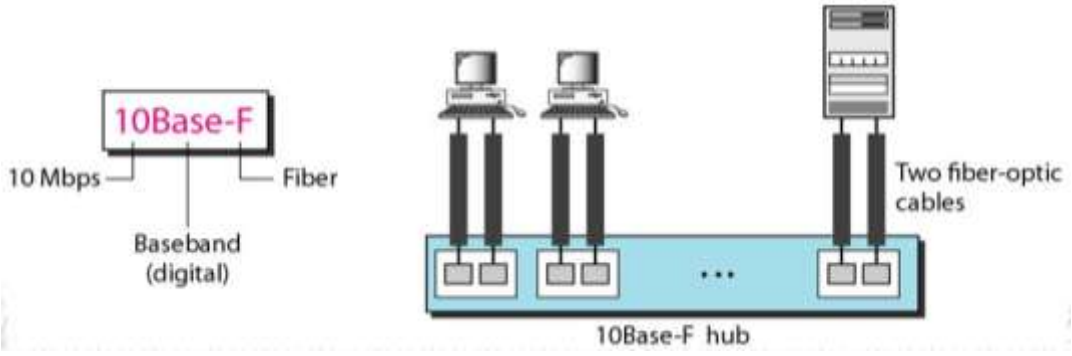
10 Base T: Twisted Pair Ethernet

Mostly through two pairs of twisted cables, these stations are linked to a hub. One path for sending and one path between the station and the centre for receiving. So if a collision is involved, the hub replaces the coaxial cable. The actual length of the pair being twisted is 100 M.



10 base F: Fiber Ethernet

For connecting stations to a hub, 10 base F uses a star topology. Using two fibre optic cables, the stations are connected to the hub.



2. Fast Ethernet

Fast ethernet was intended for LAN protocol through fiber channel. IEEE fast ethernet is named as 802.30. It works at 100mbps rate which is 10 times faster.

Goals of Fast Ethernet

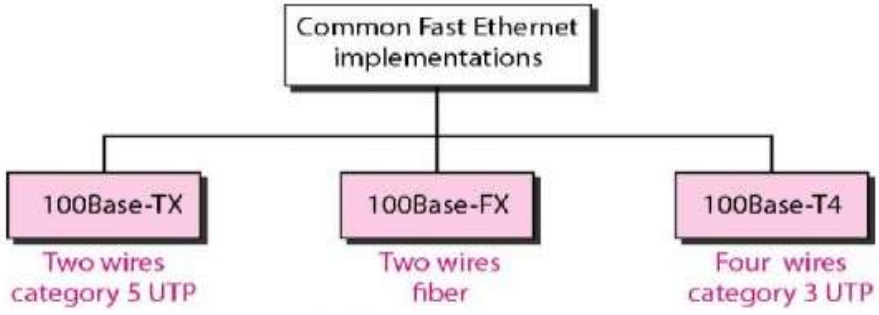
1. Elevation of information rate shoot up to 100 MBPS.
2. Making consistency with the ethernet standard.
3. Maintain a constant 48 bits of address.
4. Maintain a constant format for frame.
5. Maintain the constant length of the frame.

2.2.3. MAC Sublayer

Evolution of ethernet from 10 Mbps**.** Star topology is used to establish the connectivity by full duplex and half duplex. In full duplex, stations are connected via hub and in half duplex, the connections is made via a switch with buffer at each port.

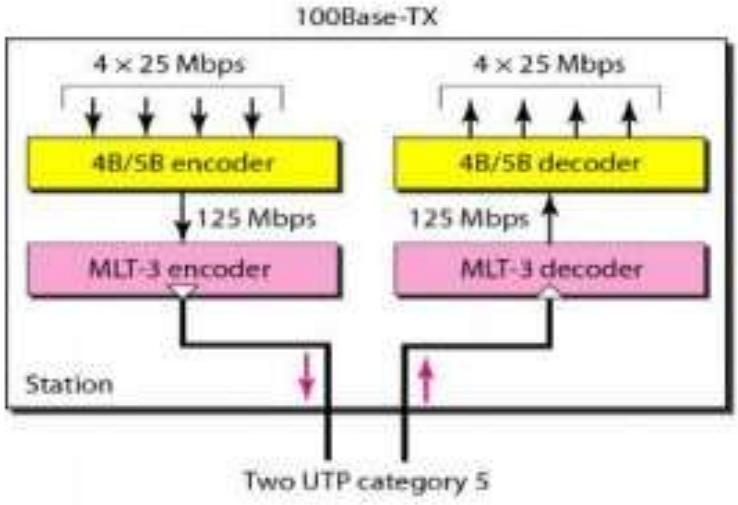
Uses of Fast Ethernet

- To allow incompatible devices to get connected with each other.
- To activate the multiple capabilities of one system.
- Allowing a station to verify the capabilities of a centre.



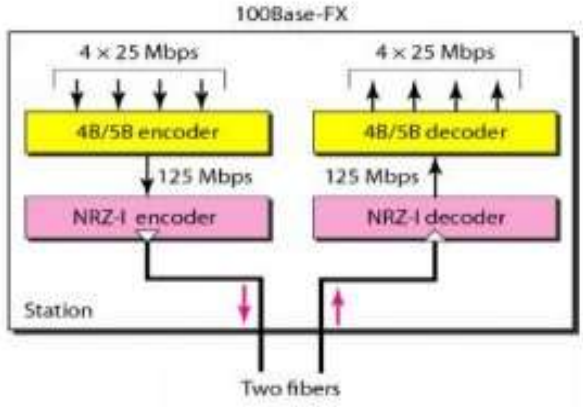
100 base -TX

- Two pairs of twisted-pair cables are used.
- Data rate 125Mbps.
- MLT3 scheme is used for good bandwidth.
- Block coding 4B15B is for bit synchronisation by preventing a long series of OS and LS from occurring.



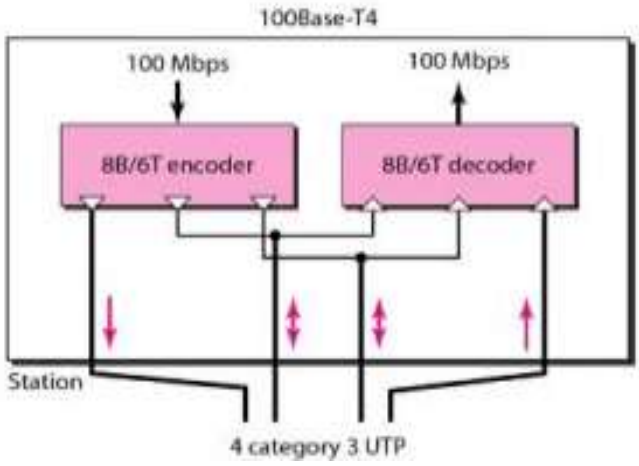
100 Base-FX

- It uses two fibre optic cables in pairs.
- By using simple encoding schemes, it can easily handle high bandwidth efficiently.
- NRZ-1 encoding scheme is used for bit synchronisation issue for long sequences of OS 100 to 125Mbps.



100 Base-T4

- It utilises UTP category 3 or higher.
- It uses 4 UTP pairs to relay 100Mbps.
- Each twisted pair cannot manage more than 25 M band easily.
- One pair switch between sending and receiving. It handles 75Mband (25Mband each).
- In 8B/6T, six-signal elements are obtained as a result of eight data elements.



3. Gigabit Ethernet

Gigabit Ethernet protocol has higher data rate of 1000Mbps. It is also known as standard 802.3z.

Goals of Gigabit Ethernet

- Upgrade the speed of data to 1Gbps.
- It must be made as a standard one or compatible for Fast Ethernet.
- The same address of 48-bits and similar format is used for frames.
- The frame lengths are maintained the same for all sizes.
- Promoting auto-negation to be described in Fast Ethernet.

MAC Sublayer

Gigabit Ethernet has two approaches as given below:

a. Half-duplex
b. Full-duplex

a) Full-Duplex

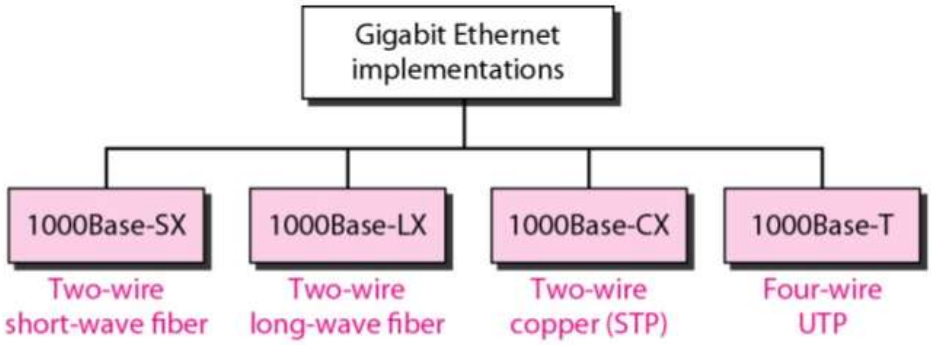
- There is no collision in the full-duplex Gigabit Ethernet mode; the attenuation of signal identifies the overall cable length.
- Each switch connected with buffers to store data until they are transmitted.

b) Half Duplex

- A switch may be alternated with a hub which serves as a common cable if there is a collision.
- It uses CSMA/CD, which depends on minimum frame size.

- Traditional - 512 bits
- Carrier extension - 4096 bits
- Frame bursting-multiple frames sent with frame padding

Gigabit Ethernet Implementation



4. Ten - Gigabit Ethernet

- 10 gigabit ethernet is also mentioned as standard 802.3 AE.
- The motive behind 10 Gigabit are given as follows.
  - Elevation of data speed to 10Gbps.
  - Assuring compatibility over standard, fast and gigabit Ethernet.
  - Using determined and relevant address with 48 bits.
  - Maintain the size of frame lengths.
  - Enable existing LANs to be interconnected to Metropolitan Area or Wide Area Network.
  - Frame relay & ATM.

2.3. Wireless LAN

2.3.1. IEEE 802.11 - WIFI

IEEE 802.11, a wireless LAN was designed by IEEE that conceals physical layer and data link layer.

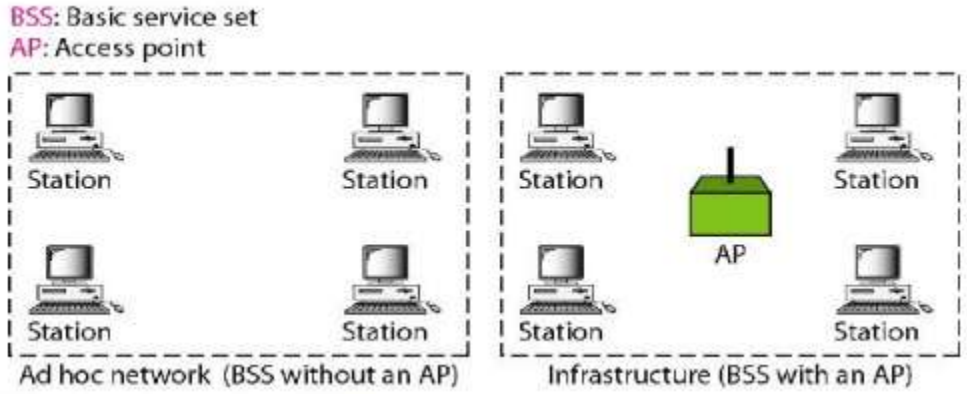
2.3.2. Architecture

The services are given below,

1. The Basic Service Set (BSS)
2. The Extended Service Set (ESS)

1. Basic Service Set (BSS)

A wireless LAN is the basic service set (BSS), made of stationary or mobile wireless stations and an optional central base station known as Access point (AP). It is a standalone network without an AP and does not transmit data to another BSS. It is referred to as ADHOC architecture. On the other hand, BSS with AP is deemed as infrastructure network.

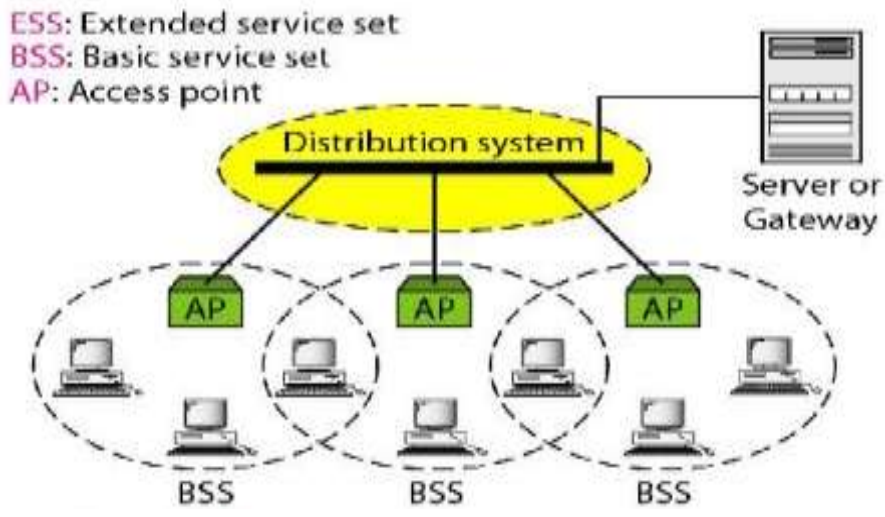


2. Extended Service Set (ESS)

Two or more BSS with AP forms an extended service set (ESS). BSS is connected over a distributed structure by wired LAN. Distributed system connects AP in the BSS.

There are two types of stations.

- Mobile-normal stations inside BSS.
- Stationary-AP stations using a LAN with wired connectivity.



Station Types

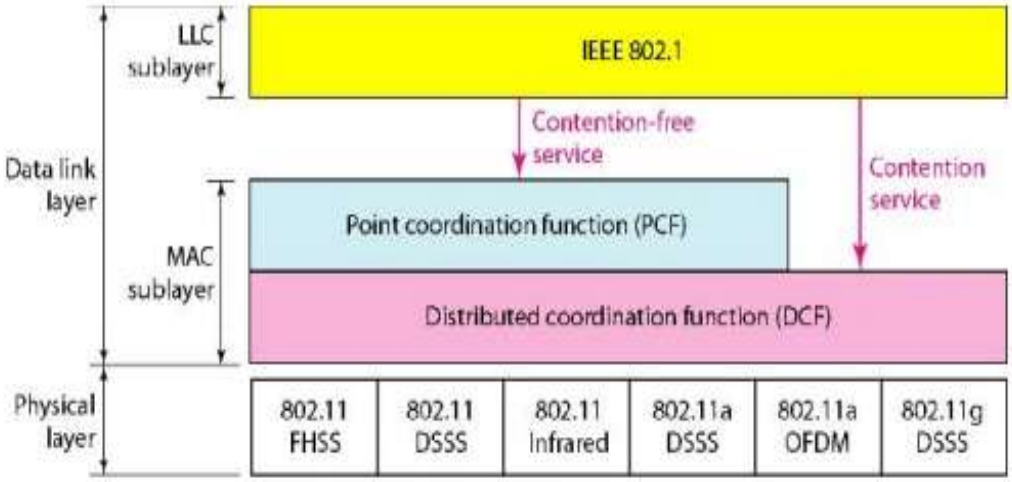
Three different stations depending upon their mobility in a wireless LAN are with no-transition, BSS-transition and ESS-transition in IEEE 802.11.

Location which has no – transmission movement is a stationary one or BSS that moves. BSS transfer mobility may make a station to switch between different BSS, while the communication is designed within a BSS. From one ESS to another, a station with ESS transfer mobility may switch from one another.

2.3.3. MAC Sublayer

IEEE 802.11 Expresses 2 MAC Sublayers

- **DCF**- Distributed Coordination Function
- **PCF**- Point Coordination Function



Distribution Coordination Function (DCF)

Access method used by **DCF** is CSMA/CA. **Wireless LAN** will not be possible to device CSMA/CD because of few points.

1. While detecting a collision, a station should send data and receive alert related to collision concurrently. Here it means expensive stations possess higher specifications of bandwidth.
2. The identification of collision will not be acknowledged due to the hidden station issue.
3. Great signal fading may be the distance between stations, which may prevent a station from hearing a collision at one end and at the other end.

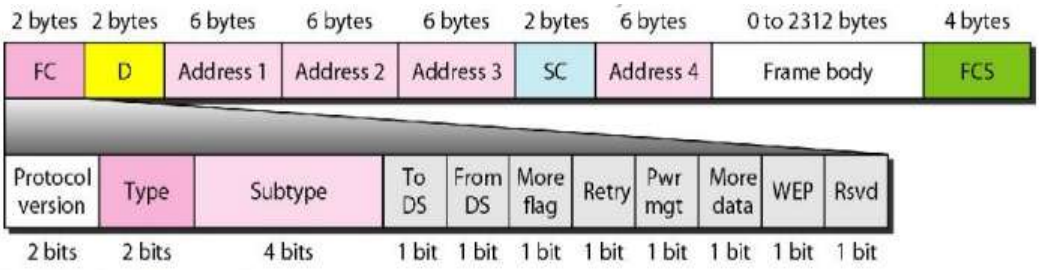
When a station sends an RTS frame it includes the duration of time that it needs to occupy the channel. The Network Allocation Vector (NAV) resembles a timer, that is produced on the nodes which impacts the transmission indicating how much time it takes. Two or more stations can attempt to simultaneously submit RTS frames. These control frames can collide. The non-reception of CTS frame from the receiver, states the sender that there has been a collision. The back off technique is used and the sender resends.

Point Coordination Function (PCF)

PCF is an optional form of connectivity that can be introduced in a network of infrastructures. It is a centralised method of contention-free polling access. The AP performs polling for stations that are capable of being polled. The stations are polled one after another, **sending** whatever data they have to the AP. The Point Controller (PC) will be able to transmit data to poll frame, send acknowledgement, receive acknowledgement, further use one of the combinations called as piggybacking.

2.3.4. Frame Format

The MAC layer comprises fields of nine, namely:



Frame control (FC) - The FC field has a length of 2 bytes and specifies the frame's type and some other additional information related to control.

D - This field defines the duration of transmission that is used to set the value of NAV

Address - They are four in number with each containing 6 bytes of length, Address fields depend on the value to DS and from DS subfield.

Sequence control (SC) - Number of the Sequence of frames are defined that is necessary for flow control.

Frame body - It ranges from 0 to 2312 bytes which holds data depending on the type and subtype specified.

FCS - It measures a length of 4 bytes and comprises of CRC-32 error detection sequence.

Sub Fields in FC Field

Version	current version 0
Type	management (00), control (00), and data (10)
Subtype	1011- request to Send (RTS)
	1100- Clear to Send (CTS)
	1101- Acknowledgement (ACK)
To DS	to Distributed System
From DS	from Distributed System
More flag	1 with more number of fragments
Retry	1 which retransmits data frame
Pwr mgt	1 when mode is on for power management
More data	1 when power to send data
WEP	wired equivalent privacy
RSRD	reserved

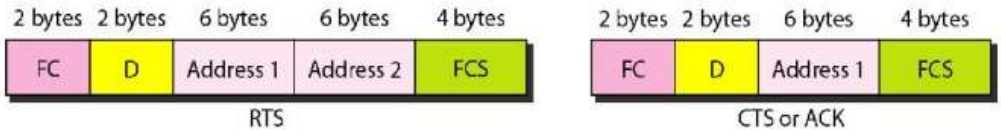
Frame Types

The frames are segregated as management frames, control frames and data frames

Management frames - Send early message with station and that of the access point.

Control frames - Responsibly acts towards retrieving the channel and recognizing it.

Data frames - Carries data and control information.



2.3.5. Physical Layer

IEEE 802.11 FHSS - IEEE 802.11 Frequency Hopping Spread Spectrum uses 2.4 GHZ ISM band. The band is divided in 79 sub bands of I MHZ.

IEEE 802.11 Infrared - It is also referred as Pulse Position Modulation (PPM) and uses Infrared light that ranges between 800 and 900 nm.

IEEE 802.11 DSSS - IEEE 80.11 Direct Sequence Spread Spectrum uses 2.4 GHZ ISM band.

IEEE 802.11 OFDM - IEEE 802.11a Orthogonal Frequency Division Multiplexing method for signal generation in 5 GHZ ISM band. This band is divided into 52 sub bands, where 48 sub bands for sending 48 groups of bits at a time and 4 sub bands for control information.

IEEE 802.11 DSSS - This describes the Direct Sequence method that is of higher rata and the signal is generated in 2.4GHZ ISM band.

IEEE 802.11g - It gives forward error correction and OFDM using the 2.4 GHZ ISM band. It is backward compatible with 802.11b.

2.4. Bluetooth

Bluetooth is a kind of wireless LAN technology that is designed for connecting various devices with different functionalities such as telephone, computers, cameras, printers, and so on. Bluetooth LAN (IEEE 802.15) is an ad hoc network, where the devices are connected impulsively referred as gadgets. Bluetooth was initiated by Ericsson Company. This is more suited for small area and defined as a wireless Personal Area Network (PAN).

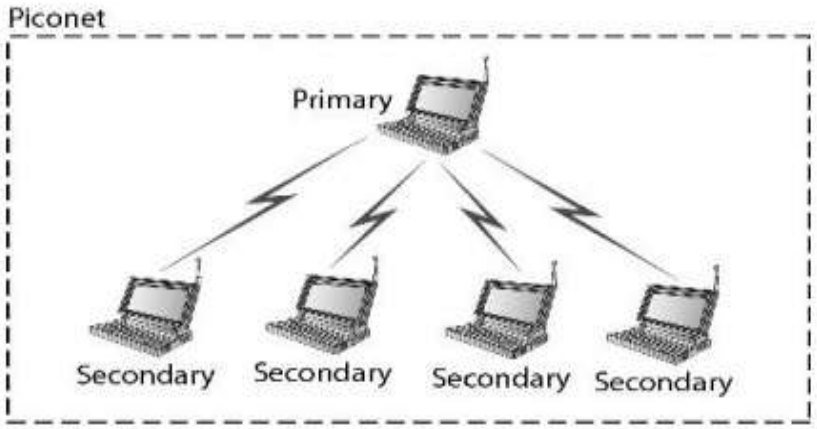
2.4.1. Architecture

Bluetooth defines two types of network.

- piconet
- scatternet

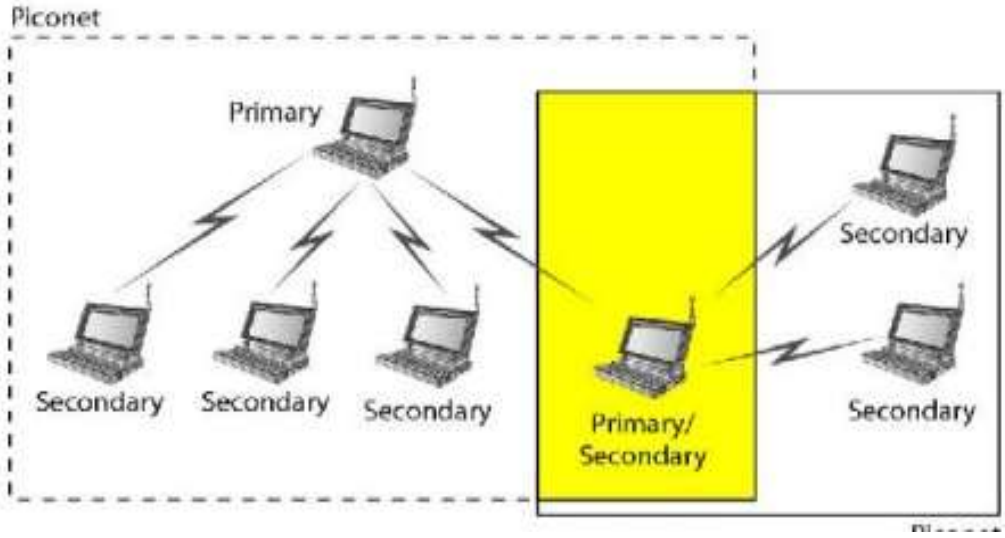
Piconet

A Bluetooth is network that can be called as a piconet or otherwise called as a small net. A piconet can have up to 8 primary stations and rest as secondary stations. All the secondary stations synchronise with the primary with their clocks and hopping sequence. A piconet may have only one primary station. One to one or one to many may be the contact between the main and secondary. A piconet has a maximum of 7 secondaries and 8th can be a parked state. A secondary is synchronised with the primary in a parked state.



Scatternet

Piconets are fused to create what is referred to as a scatternet. A secondary station in one piconet may be the main in another piconet. This station can receive messages from the main in the first piconet (secondary) and functions as a main, delivers them to secondary in the second piconet. Any station might take part in 2 piconets.



A built-in short-range radio transmitter is found in a Bluetooth system with 2.4 GHz bandwidth. It is the interface between IEEE 802.11b wireless LAN and Bluetooth LAN.

2.4.2. Bluetooth Layers

2.4.2.1. Radio Layer

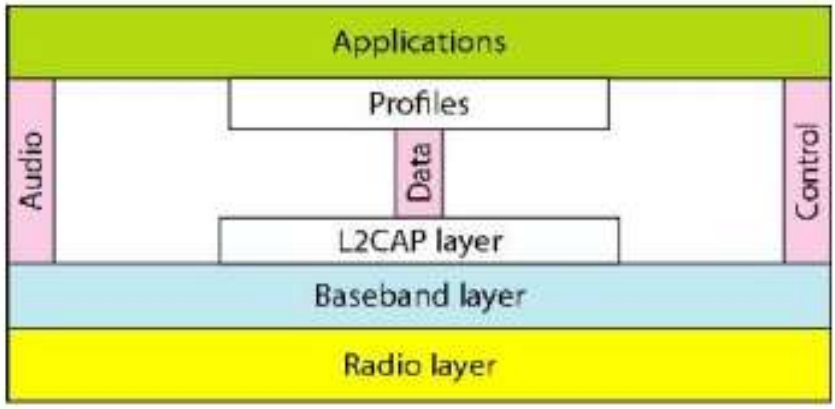
It is equivalent to the physical layer in internet model**.** A low power devices are Bluetooth with a range of 10m.

Band

The 2.4 GHz ISM band used by Bluetooth is segregated into 1 MHz each that contributes 79 channels.

Modulation

Sophisticated version of FSK called GFSK (Gaussian bandwidth filtering)**,** GFSK is a carrier frequency where bit 1 represents frequency over the carrier and bit 0 represents frequency below the carrier is used in Bluetooth.



FHSS - Frequency Hopping Spread Spectrum.

This methodology is used by physical layer that avoids interferences caused by different systems or networks. It is observed that bluetooth can hop upto 1600 times in a second i.e., frequency modulation of the devices can be varied to 1600 times in a second. The frequency of 625 micro seconds (1/1600 s) is utilised.

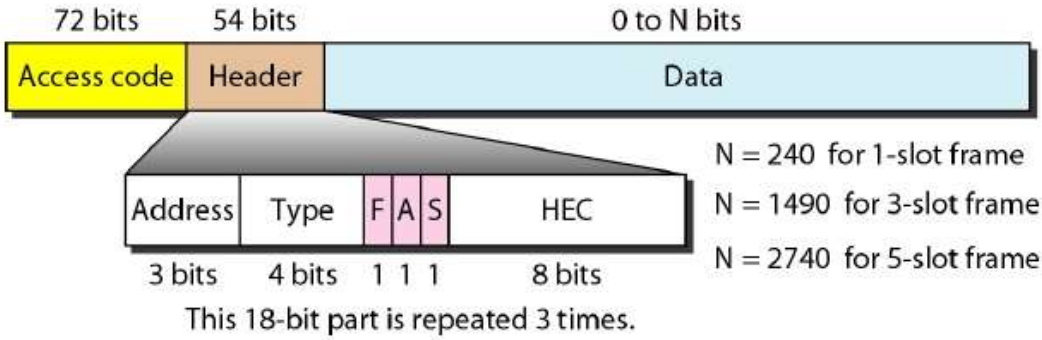
2.4.2.2. Base Band Layer

MAC sublayer in LAN resembles the baseband layer. The different nodes contact each other in the timing duration. The allotted slot is 625 micro seconds. TDMA is used as Time Division Duplex TDMA (TDD-TDMA) by the Bluetooth with half-duplex type of communication i.e., data is sent or received non-synchronously.

Physical Links

There are 2 types of links given as Synchronous Connection-Oriented link (SCO) which evades delay and integrity & Asynchronous Connection-Oriented link (ACL) which overrides latency avoidance by data integrity.

Frame Format



Access code - It consists of 72 bits for synchronization as well as for identifying the primary frame from one piconet to another.

Header - 18-bit pattern is used to repeat the 54 bits field.

Address - 3 bits address define up to 7 secondaries if address is 0 and it is used for broadcast.

Type - Type of data traversing through higher layers where,

F - Flow control

A - Acknowledgement

S - Sequence number

HEC - Header Error Correction

2.4.2.3. L2CAP

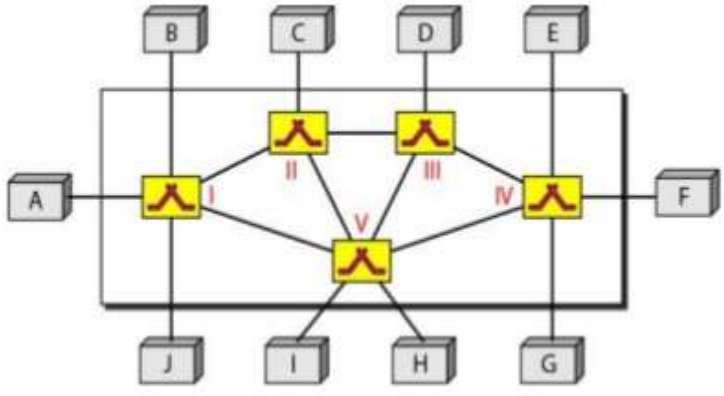
The Logical Link Control and Adaptation Protocol (L2CAP) resembles LLC sublayer. It is used for exchange of data, on an ACL link, while an SCO channel does not use L2CAP. The 16-**bit** length field defines the size of the data from upper layers. The L2CAP performs multiplexing, segmentation, reassembly, Quality Of Service (QOS) and group management.

2.4.2.4. Data Field

This field contains the data or control bits. It can be of length 0 to 2740 bits, which holds either data otherwise control bits traversing from upper layers.

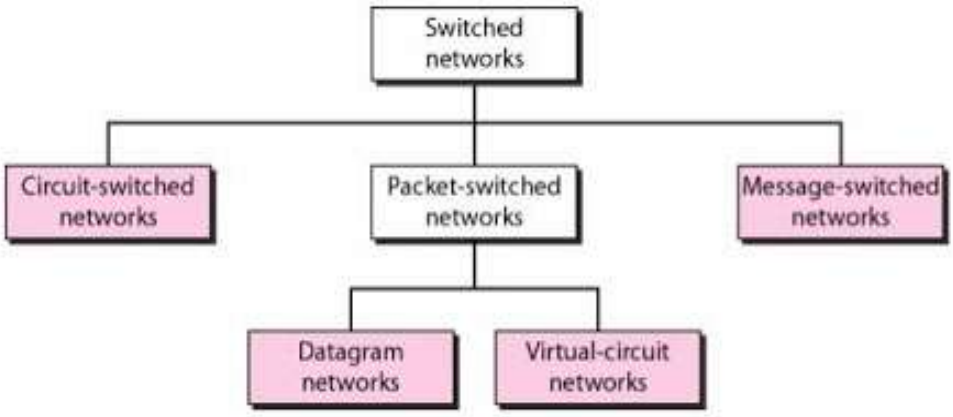
2.5. Switching

This network is made up of nodes named switches. These switches provide an adhoc link with many switch-linked devices. Few of them are linked with the end systems in the switched network.



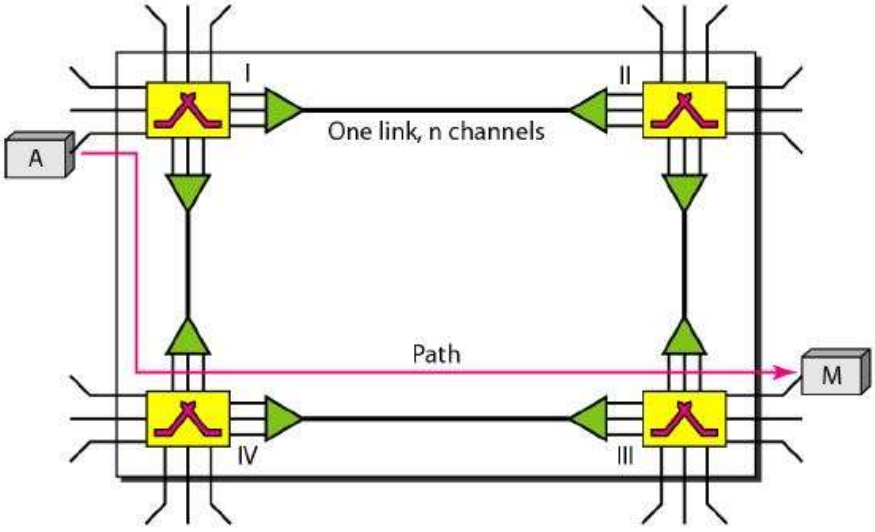
The end systems are A, B, C, D, E, F, G, H, I, J, and switches are 1,2,3,4,5.

2.5.1. Taxonomy of Switched Networks



2.5.1.1. Circuit Switched Networks

Series of continuous switches are linked through physical links in network switched circuit. A link connecting various nodes holds a separate path composed of many connectors. This network holds a constant flow of physical-connected switches where every connection is segregated as h-channels.

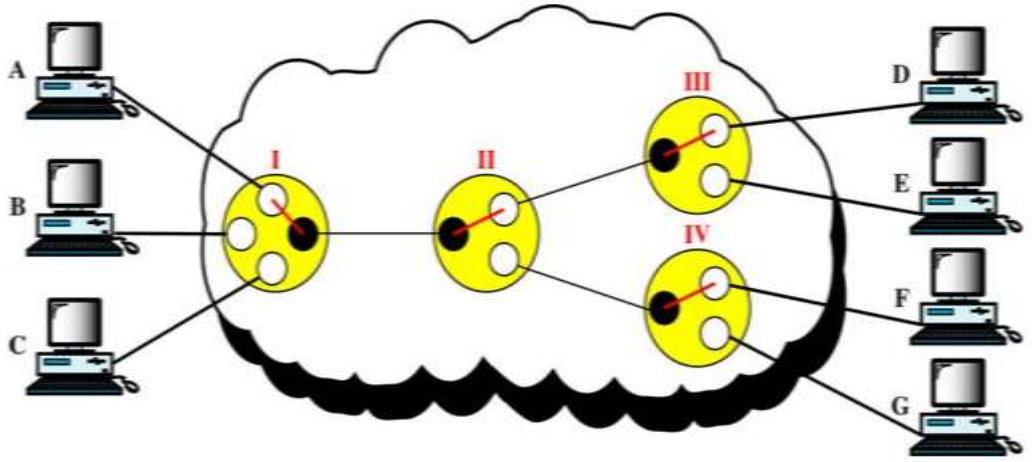


A switch is attached to the final device. In order to get connected with the system x, it is expected to appeal a link with x such as all switches and x agrees. The process is done in the setup phase. On each connection, a circuit (channel) is reserved and the dedicated path is specified by the combination of circuits or channels. At the physical stage, circuit switching

takes place. For the services to be used during the interaction, the station must make a reservation. Resources such as channels, switch buffers, switch processing time, input / output switch ports have to be preserved with confinement until the tear-down step for the overall cycle of message transfer. During data transfer, there is no source and addressing involved.

Three Phases

1. **Setup phase -** A connection between the two end systems are created with the address required for communication.
2. **Data transfer phase -** Two nodes (channels) can transfer data.
3. **Teardown phase -** When one of the parties needs to disconnect a signal is sent to each switch to release the resources.



The telephone network is connection oriented. It involves in the transfer through a network that has a dedicated connectivity and this is termed circuit switching.

Advantages

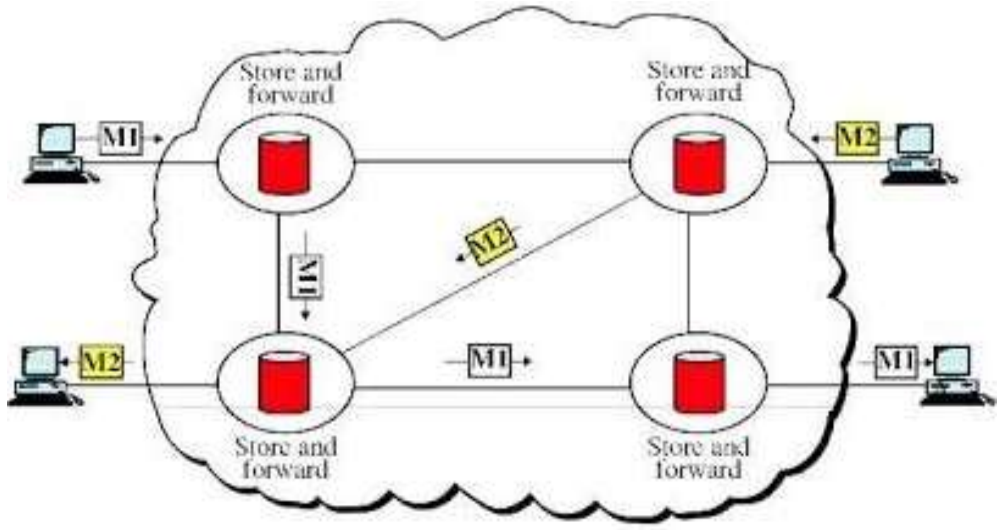
- Dedicated transmission channel establishment provides a guaranteed data rate of transmission.
- Time delay is not found in data flow.

Disadvantages

- Although the channel is idle, it cannot be used to transmit any other data.
- More Bandwidth is demanded by dedicated channels.
- It takes longer time to establish connection.

2.5.1.2. Message Switching (OR) Telegraph Network

Text message is encoded using different code to frame a sequence of data with dashes. The text message is sent from the telegraph office which is the source to the telegraph switching station. An operator takes the decision of routing the message based on the destination address information the operator will either forward the message is a connectivity to the destination is free or store the message till the communication line becomes free. Every message is considered as an independent unit and includes both the source and destination addresses. All complete messages are transmitted from a device to another device through the connectivity.



Each intermediate device receives the message, ensures the readiness of the connecting device and then forwards it to the next one. This is called as a store and forward network. The information is more efficient and other switches that can be used to forward messages to their ultimate destination.

Advantages

- Effective traffic management is offered by prioritizing messages that needs to be switched.
- Network traffic congestion is minimized.
- The network devices share the data channels.
- Asynchronous communication is provided across different time zones.

Disadvantages

- Storing and forwarding introduces delay.
- A large storing capacity is needed for intermediate devices in storing the messages.

2.5.1.3. Packet Switching

The message to be transmitted is fragmented into smaller packets and each packet contains information about the source and destination in a header along with the address details of the intermediate nodes. Each and every packet can take various routes in order to reach the destination. There are 2 advantages.

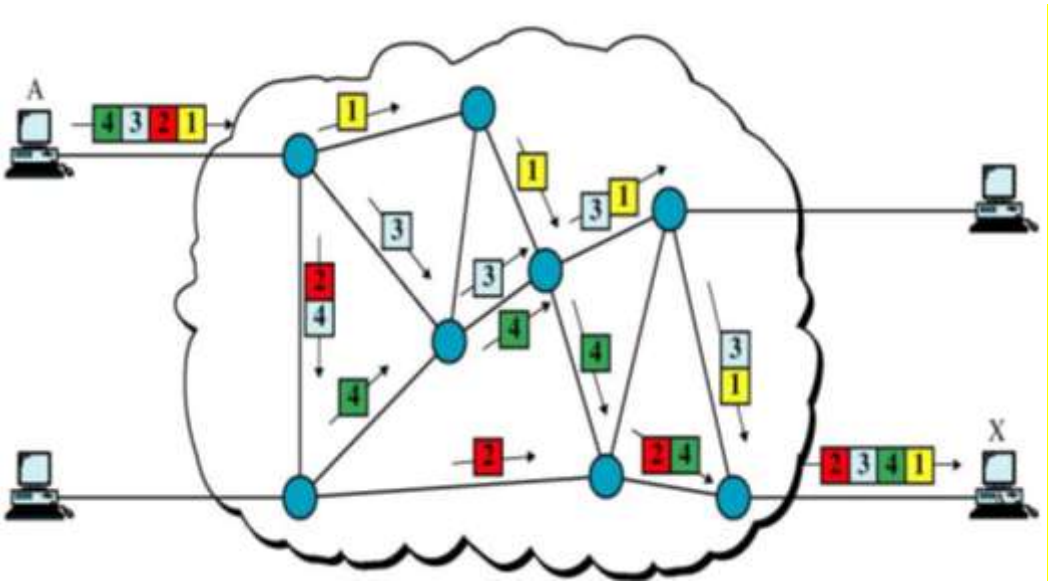
1. Bandwidth is reduced by splitting data into different routes in a busy circuit.
2. If a certain link in the network goes down during the transmission, the remaining packets can send through another route.
    - The packet length is restricted to a maximum length.
    - There are two methods of packet switching.
        - Datagram packet switching.
        - Virtual circuit packet switching.

a) Datagram Packet Switching

A stream of packets is obtained by dividing the message into packets and each packet has its own control instruction and acts an independent unit. Also, each packet is routed independently using the switching devices with individual intermediate node which helps in identifying the next route. Once it is ready for transmission, control information is exchanged in order to establish the packet sequence and destination.

b) Virtual Circuit Packet Switching

Virtual circuit is nothing but establishing a logical connection between the sender and receiver. Based on the agreement of the commination parameter with the receiving device, a communication is established with a conversation initiated by the sending device. The devices use it for the rest of the conversion. All the packet travel through the logical connection established between the sender and receiver.



Advantages

- The bandwidth of the network can be increased in order to communicate with numerous devices via the same network channel.
- A packet can be routed using a switching node based on its requirements.
- Time and transmission delay are reduced.

Disadvantages

- In Packet switching, the size of the RAM is proportional to the quantity of the packets.
- It requires more processing power.

2.5.2. Bridging

Bridge is used to connect two or more detached networks that are connected for exchanging data or resources. It utilizes addressing protocols and can affect the flow control of a single LAN. Bridges are more active at the data link layer. It receives the signal and it can check the physical (MAC) address of source and destination stored in the frame.

Generally 2 types of bridges exist namely,

- Transparent bridge
- Routing bridge

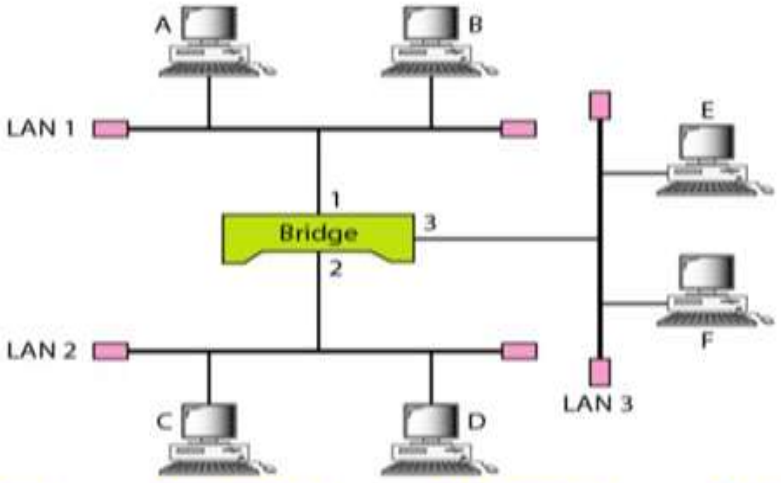
2.5.2.1. Transparent Bridge

In this bridge, the stations are unaware of bridge connectivity. Transparent bridges keep a table of address in memory to regulate the forwarding of data.

The duties of transparent bridge are:

1. Filtering flames
2. Forwarding
3. Blocking

Each packet traces the destination and source addresses. It checks the destination to decide where to send the packet. If it does not recognize the destination address it relays the packet to all of the stations on both segments. When a frame arrives at a bridge, it must choose to **either** discard or forward it and if the latter is true, then decide on which LAN to put the frame.



Address	Port

a. Original

Address	Port
A	1

b. After A sends a frame to D

Address	Port
A	1
E	3

c. After E sends a frame to A

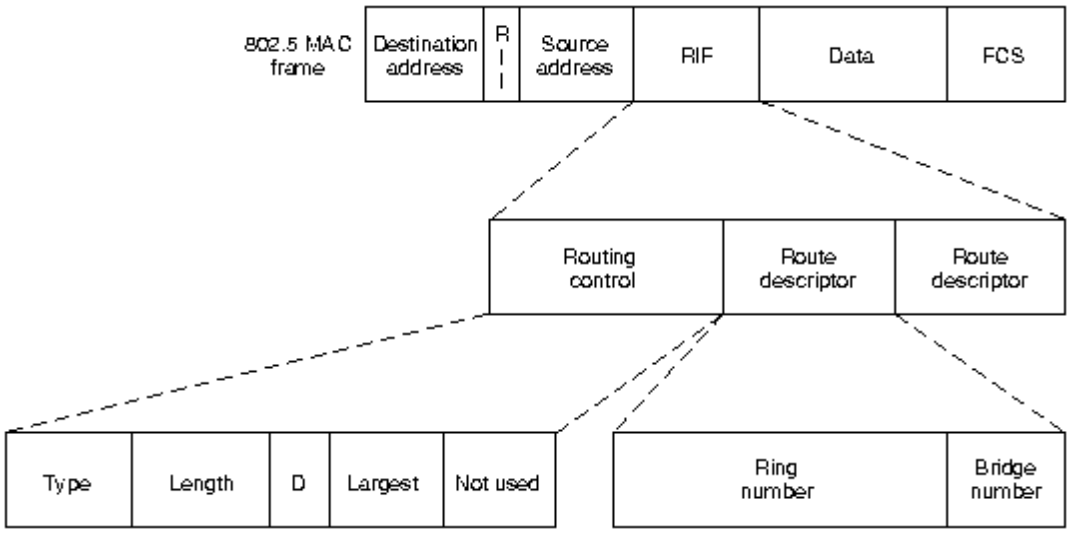
Address	Port
A	1
E	3
B	1

d. After B sends a frame to C

- A sends frame to D:**flooding**
- E sends a frame to A: **Forwarding**
- B sends a frame to C :**flooding**

2.5.2.2. Routing Bridge

The routing bridge is used to interconnect token ring networks. The header holds the route to destined node, during its travel. This information is present only if the communication is between varied LANs.



How to Discover a Route?

- The station who wants to discover a route first broadcasts a special frame.
- The frame visits every LAN exactly once and eventually reaches the destination.
- Next, a special frame named all routes special frame is responded from the destination station which produces all probable routes to the source station.
- Finally, the best route is chosen at the source from the all collected routes and is saved.

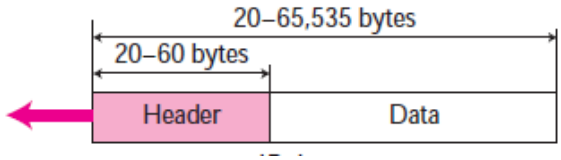
2.6. Basic Internetworking

2.6.1. Internet Protocol (IP)

IP is network layer protocol designed as connectionless delivery between hosts for the internet that guarantees least reliability. This is due to the fact that it supplies no error control or flow control. Also, the error alone will be detected and the corrupted packet will be discarded. Each IP packet is handled independently and it can follow a different route to the destination.

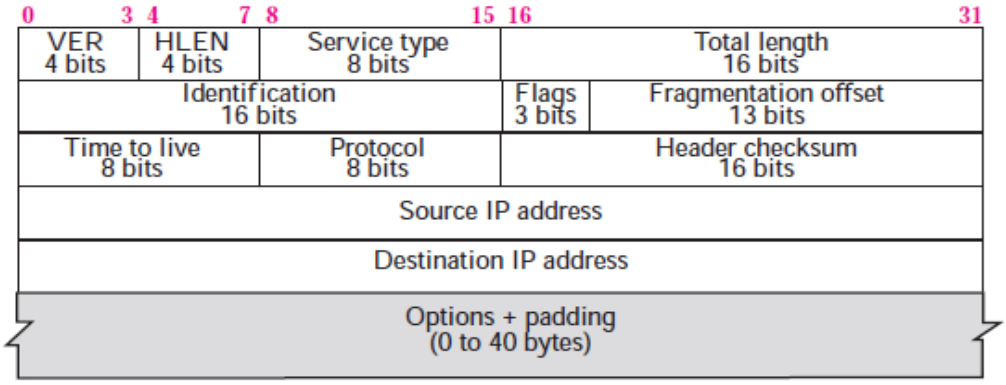
Datagram

- Packets in IP layer is called as datagrams.
- A datagram is a variable length with two parts; header and data.
- The header is 20 to 60 bytes in length it contains details for routing and delivery.
- The field width of the data is not of fixed length.



Structure of IP Frame Header

- The routing details are held at the IP frame header that is linked with diagram delivery.



Various Fields in IP Header

VER (Version)

- IP version is defined.
- IPV4 and IPV6 are the present versions.
- It is four-bit long field.

HLEN (Header Length)

- This field defines a 4-byte word as the length of the datagram header.
- The value is obtained by multiplying the length by 4 in bytes.

Differential Service (DS)

- The quality of service is shown by the class of the packet in this field.
- They accept traffic only above certain precedence at time of high load.
- The trade-off between low delay, high reliability and throughput.

Total Length

- The entire length of IP packet is defined as the complete length.
- The header and data field comprises the actual total length.
- The length of the field is 16 bits which equals 65535 bytes.

Identification; Flag; Offset

(i) Identification

- The identifying the datagram that originates from the source host is held in this field.
- The copier is held in all fragments, with its identification.
- The identification member helps destination in reassembling fragments of datagram.

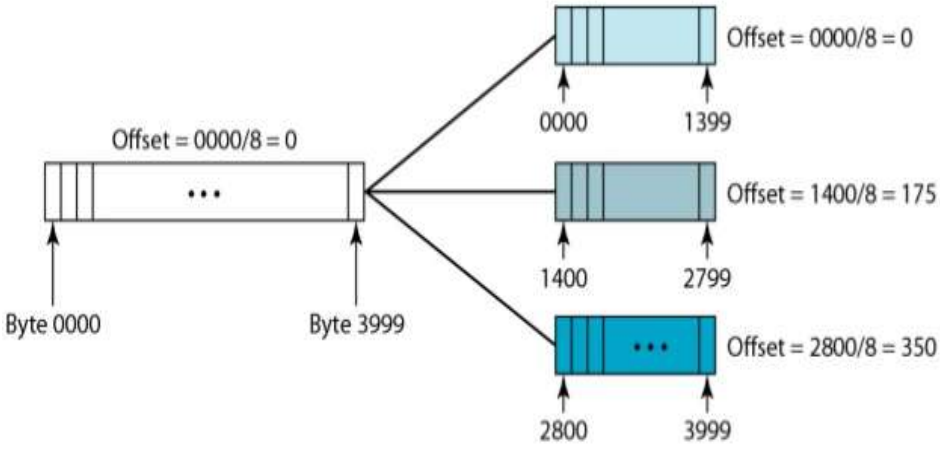
(ii) Flag

- The field has 3-bits.

- (0) is reserved at the first bit, whereas the second bit is referred by Do not fragment bit and finally, the third bit is called to be more fragment bit.

(iii) Fragmentation Offset

- A 13-bit field that indicates relative position of the fragmentation in the full datagram.
- An 8 bytes of offset is left in the datagram.
- 4000 bytes are held by IP packets ranging from 0 to 3999.
- It is segregated into 3.



Time to live

- The field has a length of eight bits and also holds a control over most of the routers which the datagram has made a visit.

Protocol

- Higher level protocol is defined by this field that uses various services provided by the IP layer.
- A protocol of higher level which summarizes data in IP datagram is done by TCP.

Header Checksum

- For an IP packet, the header alone is covered in this checksum.
- At each point, the field will be recomputed and verified, when the fields at the header attempts a change.

Source and Destination Address

- These are used in defining the IP addresses of source and destination fields.

Options

- They are used for network testing and debugging IP that provides several options of allowing a packets sender of requisites over the path which is followed on a network, identifies the route taken by packets which further label them with preserving attributes of secure transmission.

Services Provided

The following services are offered by IP:

a) Addressing

The header of IP comprehends 32-bit addresses to recognize the hosts of the sender and receiver. Intermediate routers employ the usage of these addresses in order to choose the path for a packet via a network.

b) Fragmentation

It splits larger packets into smaller fragments that helps networks that can handle only the smaller packets. These fragments are transparent as well.

c) Packet Timeout

Time to live (TTL) field is found in all IP packet that gets decremented whenever a router handles a packet and is discarded, when reaches zero.

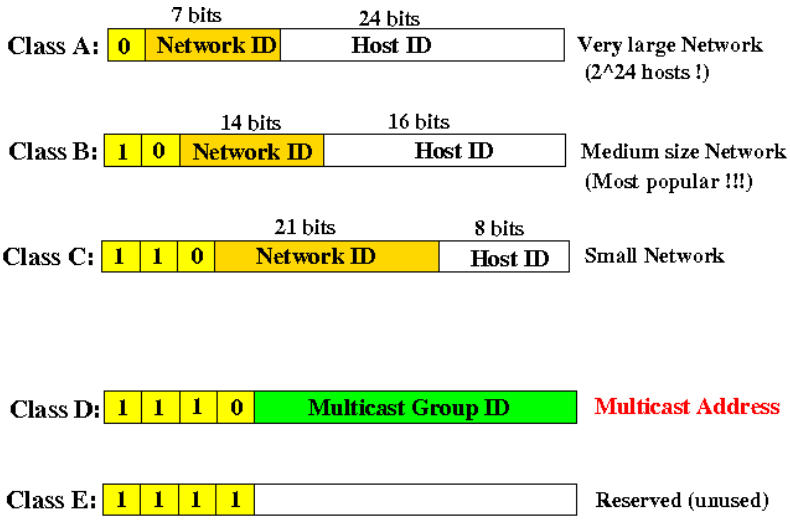
d) Types of Services

IP supports traffic **prioritization** by allowing packets to be labelled with an abstract type service.

IP Addresses

- The IP address for hosts is assigned by the network administrator.
- An IP address consist of two part.

1. **Address, called as network number which identifies a network.**
2. **Host ID, refers to each host of the network.**

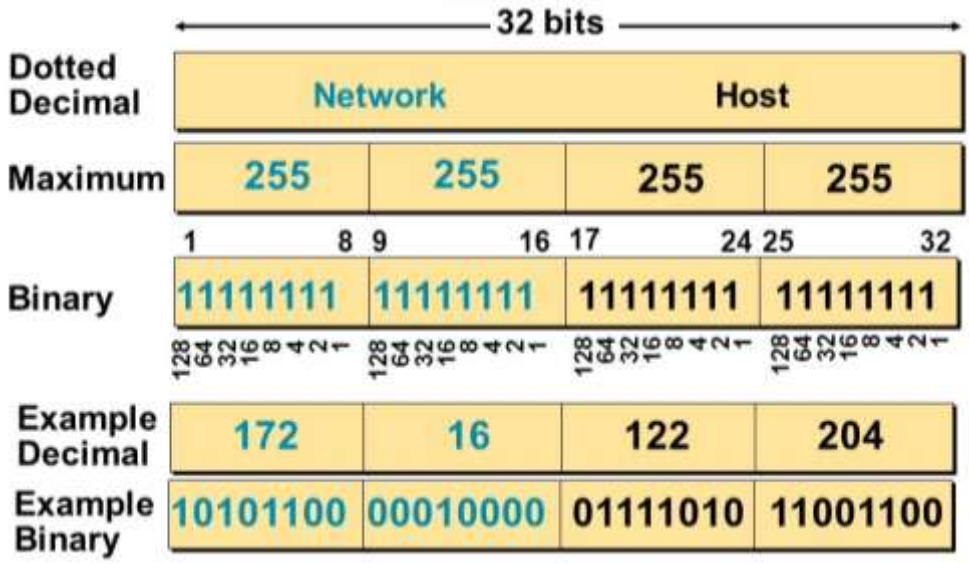


Network Address

- This address helps in defining the network and is not possible to be allotted to a host.

Class	Address range	Supports
Class A	1.0.0.1 to 126.255.255.254	Supports 16 million hosts on each of 127 networks.
Class B	128.1.0.1 to 191.255.255.254	Supports 65,000 hosts on each of 16,000 networks.
Class C	192.0.1.1 to 223.255.254.254	Supports 254 hosts on each of 2 million networks.
Class D	224.0.0.0 to 239.255.255.255	Reserved for multicast groups.
Class E	240.0.0.0 to 254.255.255.254	Reserved for future use, or research and development purposes.

IP Address (Decimal and Binary Form)



Subnetting

Identical network number is provided to every hosts present in a network. IP addressing can be problematic as the network size increases. Subnetting allows an additional level of hierarchy in IP addressing. Network is split into several smaller networks internally but it acts like a single network. The smaller parts of a networks are called as subnets.

Subnet Mark

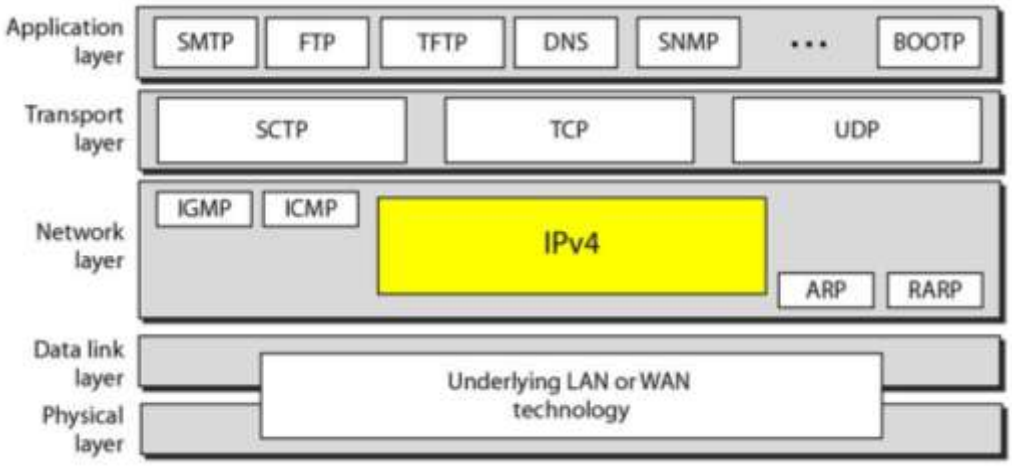
The number of 1s in the subnet mask is more than the number of 1s in the corresponding default mask. Leftmost 0s are default mask to make a subnet mask.

Super Netting

The addresses of class A and B are most depleted but class C addresses are marked until available. The prerequisite of the organisation is left unsatisfied by the C address as its size maximizes to 256 only. Therefore, it requires more addresses which is attained by combining several networks to form a super network.

2.6.1.1. IPV4

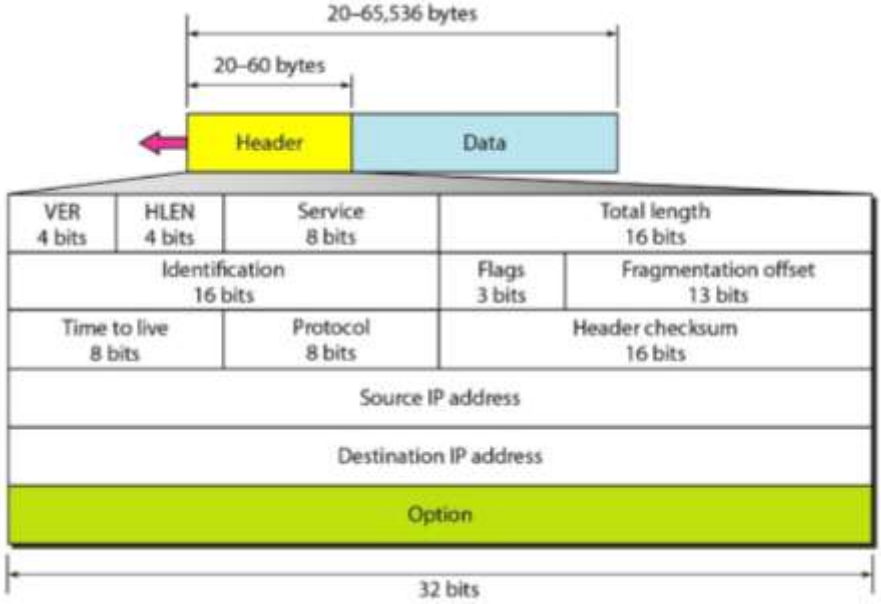
TCP/IP protocols uses internet protocol version 4 (IPV4) as the procedure to perform delivery of packets.



IPV4 is considered to be a datagram protocol which is unreliable and connectionless. It is a best effort delivery service. Best effort means IPV4 provides no error control or flow control. This protocol that utilizes packet switching technology follows the datagram approach. Each datagram is routed independently and different routes to the destination are followed through the network.

Datagram

- In IPV4 layer, a datagram is referred as the packets.
- Each datagram is of variable size and consist two parts, namely, the header and data. Each header is about 20 to 60 bytes long.



VER (Version)

The version of 4 is held in 4-bit field which states the version of the IP*4 protocol.

Header Length (HLEN)

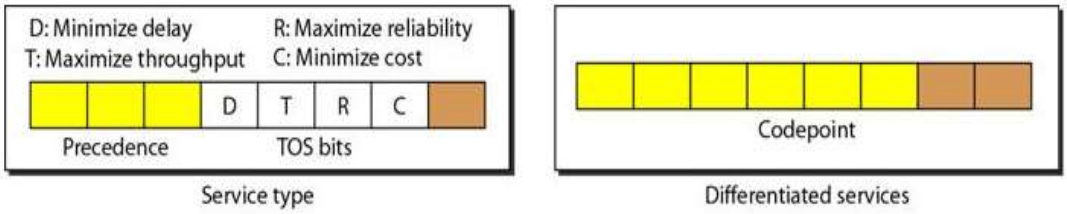
This 4-bit field defines the total length of the datagram header in 4-byte words.

Services

Service type is 8-bit field that offers differentiated services.

(a) Service Type

Precedence bits are the first three as shown in the figure and type of service (TOS) bits occupy the next 4 bits.



D **-** Minimize delay

T **-** Maximize throughput

R **-** Maximize reliability

C **-** Minimize cost

Precedence - The precedence gives a priority to the datagram during congestion, which is a 3-bit field.

TOS bits - It is a 4-bit field with each bit having a special meaning.

(b) Differentiated Services

The first 6 bits make up the code point subfield and last 2 bits are not used.

Total Length

This is a 16-bit field that defines the total length of the IPV4 datagram in bytes. Length of data = total length-header length.

Identification

This field is used in fragmentation.

Flags

This field is used in fragmentation offset and it has offset value. As a datagram travels across various networks, each network route decapsulates the frame it receives, processes the frame and then encapsulates it with another frame. The format and size of each frame is dependent on the protocols of the physical network.

Time to Live

The travel lifetime is limited for a datagram via internet. This field is designated for timestamp and gets decremented once a route is visited. Also, a datagram can travel without being delivered at its host for a relatively longer time. Therefore, this field helps in limiting the lifetime of a datagram as well as in limiting the journey of the **packet**.

Protocol

The functionalities of the IPv4 layer is given by a higher-level protocol and uses 8-bit field**.** It encapsulates data from different protocols of higher-levels namely TCP, UDP, ICMP, IGMP. It notifies the final destination protocol to which the IPV4 datagram is delivered.

Checksum

The concept of checksum and its calculations were checked for error detection.

Source Address

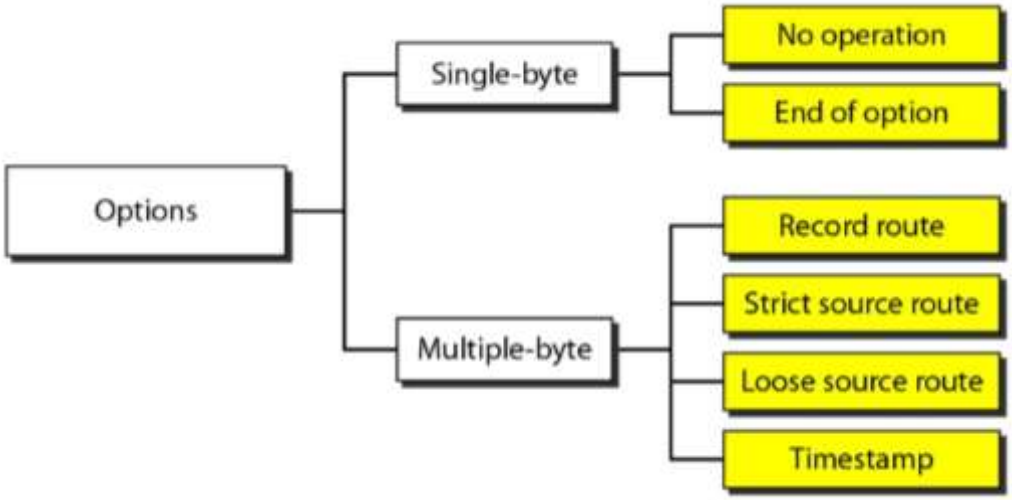
It is a field for the source that is used to define the address of IPV4 and contains 32 bits. It is expected to be the same without any chances to the datagram during the source to destination travel of the host.

Destination Address

The destination address of the IPV4 is defined by 32 bits and is expected to be the same without modifications.

Options

A fixed and a variable part is found at the IPv4 datagram's header, with 20 and 40 bytes of length respectively and utilized for the process of testing and debugging in a network.



No Operation

It is just a single-byte operation which is often utilized for filtering options.

End of Option

This is also a single-byte option utilized especially at the end of the option for padding.

Strict Source Route

In order to opt for a specific service, for e.g., the service can be a minimum delay or some other services like maximum throughput, the sender has got a privilege in selecting a route. This is offered by strict source route option that enables a predetermined datagram route that travels via internet.

Loose Source Route

Even though loose source route appears to be similar to the strict source route, nevertheless it's rigidity is less. Various routers can be visited by the datagram.

Timestamp

This option is utilized when datagram's time processed by routes needs to be recorded. The routing time of a datagram is given here.

2.6.2. Address Resolution Protocol (ARP)

The internet comprises of numerous networks and routers. In general, a packet visits various physical networks while beginning from the source host, reaching the destination host. At the network level, the addresses are used in recognising the hosts and routers.

1. IP Address

An IP address is a unique internetwork address and every protocol requires IP address for internetworking.

2. MAC Addresses

As discussed above, packets pass via physical network and at the physical level, the IP addresses alone are not sufficient but MAC addresses are required to recognize the hosts and routes. A MAC address is a unique local address and it is also passed through different physical network. The IP and MAC addresses are two different identifies and both are required to be available at the same time in the network layer. To deliver a packet to a host or a route, bi-level addressing is required as IP addressing and MAC addressing.

Mapping can be done by using static mapping or dynamic mapping.

1. Static Mapping

A table is maintained to associate IP address with MAC address. Whenever a machine wants to communicate with another machine knowing its IP address, then the MAC address can be found in the table. The table of the static mapping has to be updated periodically. The Address Resolution Protocol (ARP) associates an IP address with the physical address. On a typical physical network LAN, each device on a link is identified by a physical or station address usually imprinted on the Network Interface Card (NIC).

2. Dynamic Mapping

A protocol is set to discover the address of the other machine, given a known address in dynamic mapping. The performance of dynamic mapping is defined by the following two protocols,

- Address Resolution Protocol (ARP)
- Reverse Address Resolution Protocol (RARP)

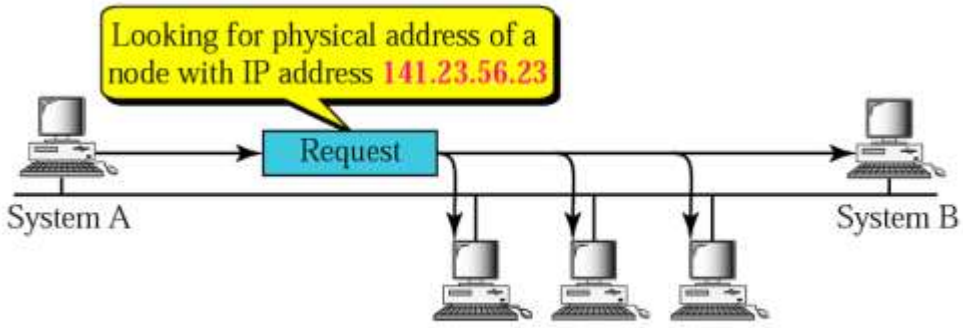
ARP Operation

The association of an IP address and the appropriate MAC address is done by ARP. Every LAN has its own physical or station address as its identification which is imprinted on the NIC.

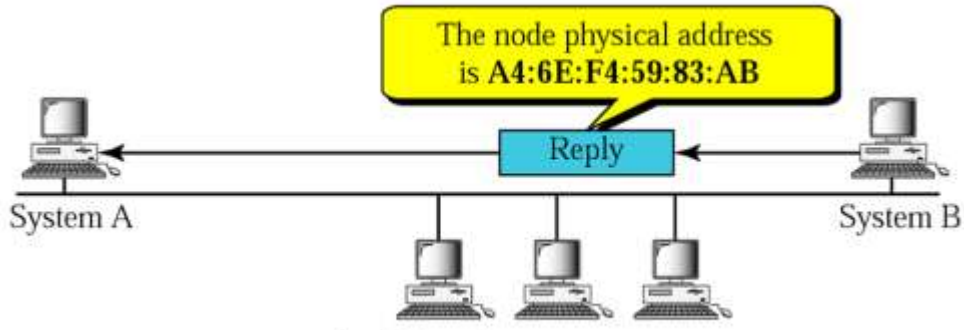
How to Find the MAC Address?

The discovery of MAC addresses of other hosts or network requires the following steps.

- An ARP request packet is sent when the router or host requests to discover the MAC address of another router.
- This packet includes both IP along with MAC address of transitter and the IP address of receiver.
- Although all router and host throughout the network receives and processes the ARP request packet, only specific receiver (B) recognizes its IP address in the request packet and reverts an ARP response packet containing IP and physical addresses of the receiver (B).
- This packet is delivered only to A (unicast) using A's physical address in the ARP request packet.



a. ARP request is broadcast

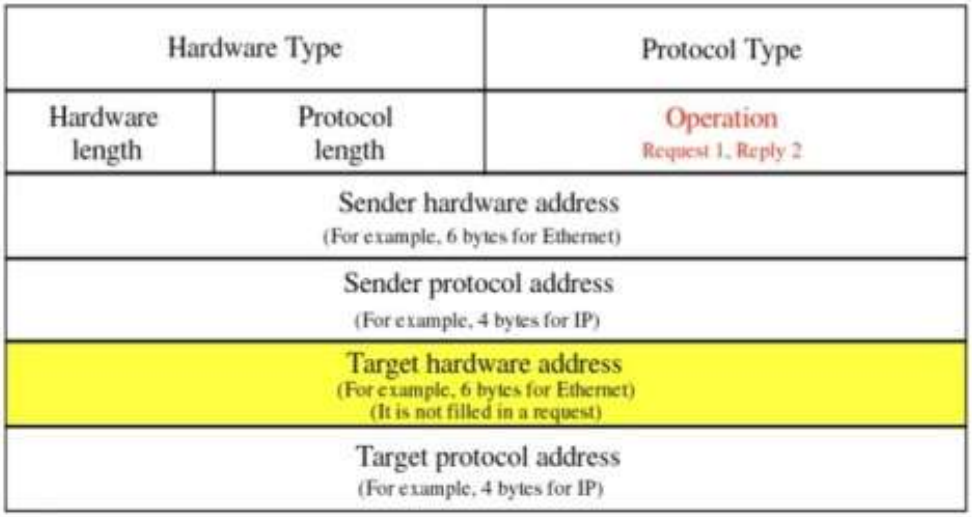


b. ARP reply is unicast

ARP Packet Format

The format of ARP packet that includes a variety of fields that are listed below:

- Having been aware of the internet address of any node, ARP can find the physical address.



1. Hardware Type (HTYPE)

ARP has the potential to run on any physical network and is given by a 16-bit field.

2. Protocol Type (PTYPE)

This is also a 16-bit field that can be used along with higher-level protocols namely IPV4.

3. Hardware Length (HLEN)

This is an 8-bit space and is designated in obtaining the physical addresses' length which is given in bytes.

4. Protocol Length (PLEN)

How long the IP address is stored in bytes is preserved in this 8-bit field.

5. Operation (OPER)

The category of packet is preserved in this 16-bit field. ARP request and ARP reply are the two types of packets.

6. Sender Hardware Address (SHA)

The sender's logical address is defined by SHA and its length is variable.

7. Target Hardware Length (THL)

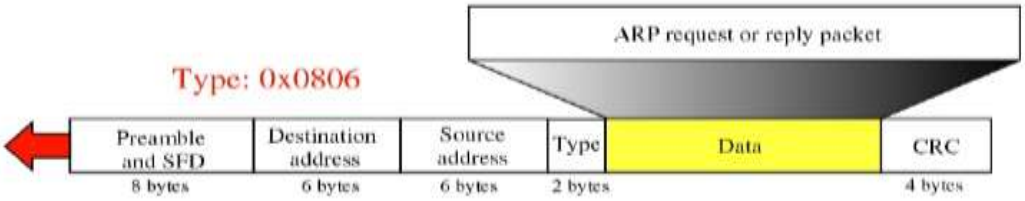
The target's physical address is defined by THL which is of variable length. At the time of ARP request packet, the field is set to zeros as the sender is unaware of the receiver's physical address.

8. Target Protocol Address (TPA)

It is another field with a length that varies and defines the target's logical address.

Encapsulation

The datalink layer holds the ARP packet (request or reply) condensed frame. The indication of the data is specified in the type field as for ARP request or reply packet.



ARP Operations

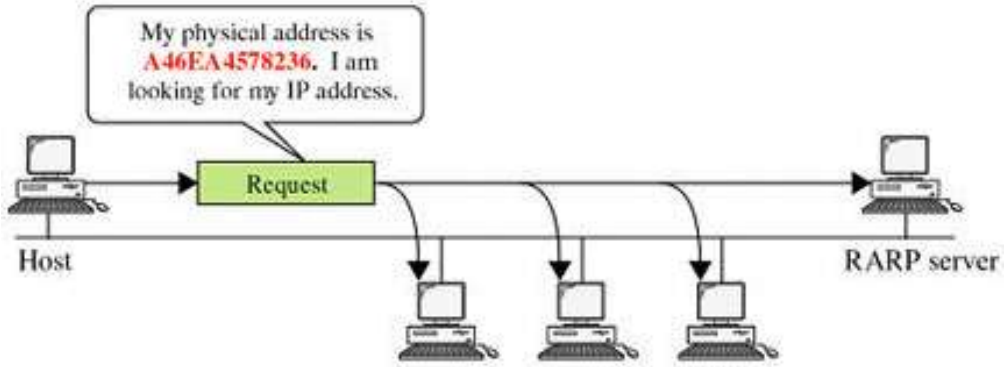
It can be used under the following conditions when it is being operated on internet.

1. Intra network transmission between hosts.
2. Inter network transmission between hosts.
3. Inter network transmission from a router to a host.
4. Inter network transmission received by a router destined to another host.

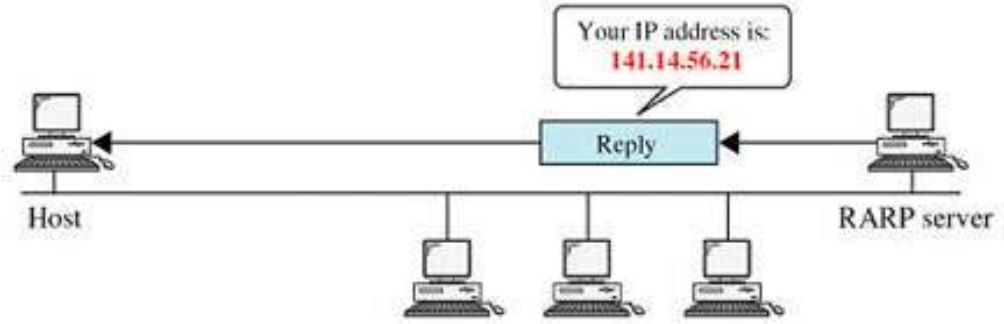
The address of the IP is known to the sender. IP address uses ARP to create an ARP request message. The request packet consists of physical and IP address of the sender along with IP address of the target. In DL layer, the ARP request packet is composed by frame and the broadcast of it is received by all routers and hosts, while the data packet is accessed only by the destined node. The destiny acknowledges with ARP reply holding the physical address of the target. This process is unicast. The frame containing IP datagram with data is sent as unicast to the destination.

2.6.3. Reverse Address Resolution Protocol (RARP)

A part of the TCP / IP protocol suite is RARP. This enables a machine or a diskless workstation, to obtain an IP address from a server. It broadcasts a RARP request packet on the network when a diskless TCP / IP workstation is enabled on a network. This address packet is transmitted for everyone to receive on the network since the workstation does not know the server's IP address which provides an address. The reply will be done using the physical network address or MAC address in request packet. The request pack in the server access the table content for a comparison with the IP address of MAC. Further it returns the IP address to the workstation that is diskless.



a. RARP request is broadcast



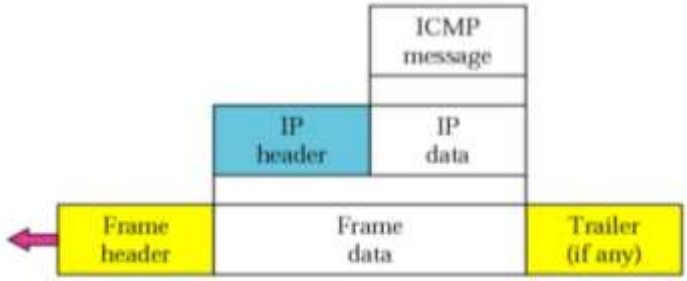
b. RARP reply is unicast

2.6.4. Internet Control Message Protocol (ICMP)

The ICMP intelligences error as well as control messages and they are sent to IP. ICMP never makes IP reliable instead reports error and a feedback is provided for certain conditions. ICMP designed to compensate these two deficiencies.

1. A host sometimes need to determine if a router host is alive.
2. Sometimes a network manager needs information from another host to router.

ICMP is a network layer protocol, its messages are not passed directly to the data link layer as expected.



The encapsulation of messages inside IP datagrams is done before travesring to the lower layer. The value of the protocol field in the IP datagram is 1 to indicate that the IP data is an ICMP message. The ping command is used to test whether station is reachable. Ping packages an ICMP echo request message in a datagram and sends it to a selected destination.

Ping 100.50.25.1

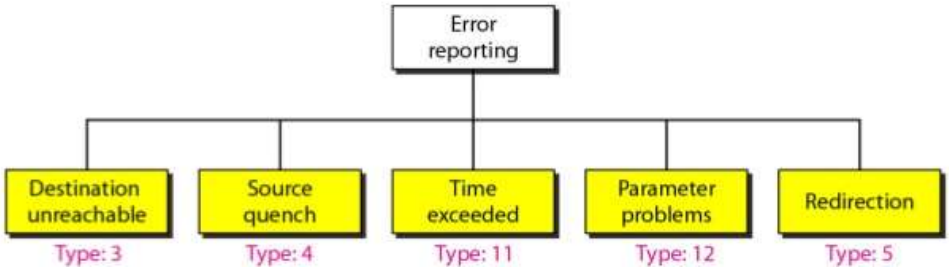
When the destination receives the echo request messages, it responds Y sending an ICMP echo reply messages. If a reply is not returned within a set time, ping resends the echo request several more times or it indicates destination is unreachable. ICMP traces routes which provide a list of all the routers along the path to the specified destination.

Types of Messages

The two types of ICMP messages are error repeating messages and query messages.

Error Repeating Messages

Error reporting has been the prime responsibility of ICMP. ICMP without correcting the errors it simply reports them and error correction is at the discretion to the higher-level protocols. ICMP sends the error reporting messages back to original source. It holds 5 types of errors.



1. Destination Unreachable

The not reachable destiny of IP packet crossing any router, signals the sender with this message.

2. Source Quench Message

A host or router uses source quench messages to report congestion to the original source and to request it to reduce its current state of packet transmission. IP does not support flow control or congestion control. There is no flow control or congestion control mechanism in IP. This type of message is ICMP which is defined to append the flow as well as congestion control

to IP. The source will be informed about the destruction of datagram. The source is further instructed to quench its flow, as a congestion is identified at some place.

3. Time Exceeded Message

Two different cases are discussed under this type of message.

(1) The datagram with TTL 0 is destroyed and acknowledges with message that intimates the source with exceed of time.

(2) The non-reception of fragments at the destiny in the stipulated time, sends a time exceeded message to the source.

4. Parameter Problem Message

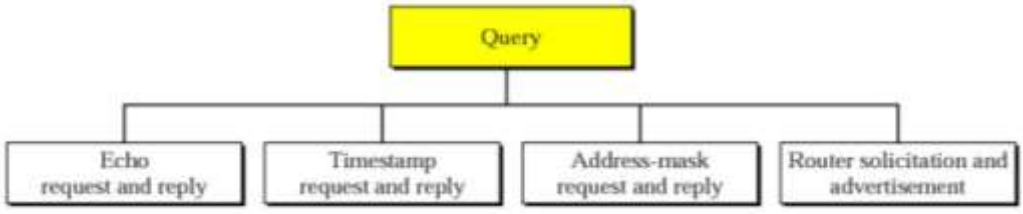
Header section of any datagram should be devoid of ambiguity. If any such indecision or lost value is identified by a router or destination host, further the message is discarded and intimated to sender.

5. Redirection Message

When a packet is transferred to another network, the next router's IP address should be known. In order to find the address of the next router, an entry in routing table has to be preserved in the router and also in the hosts, thereby routing table are continuously modified. For each update, a redirection message is sent back to its host by ICMP.

Query Messages

The ICMP message can diagnose some of the network problems and such a diagnosis is accomplished through the query messages. These messages are categorised as follows.



- **Echo Request and Reply**

The prime focus of this message is diagnosis. They determine if two systems (hosts or routers) communicates with one another.

- **Timestamp Request and Reply**

These IP datagram holds the control over the circuit of the message visiting nodes. Clock synchronization is also accomplished.

- **Address Mask Request and Reply**

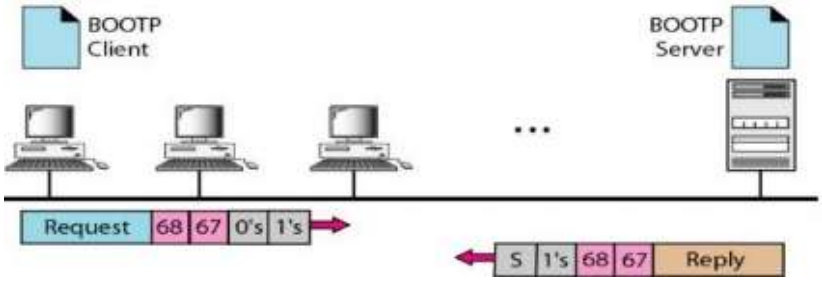
A network address, subnet address and host identifier are contained in an IP address. IP address can be held by host and cannot hold its bifurcation. Address mask reply message is transmitted further.

1. Router Solicitation and Advertisement

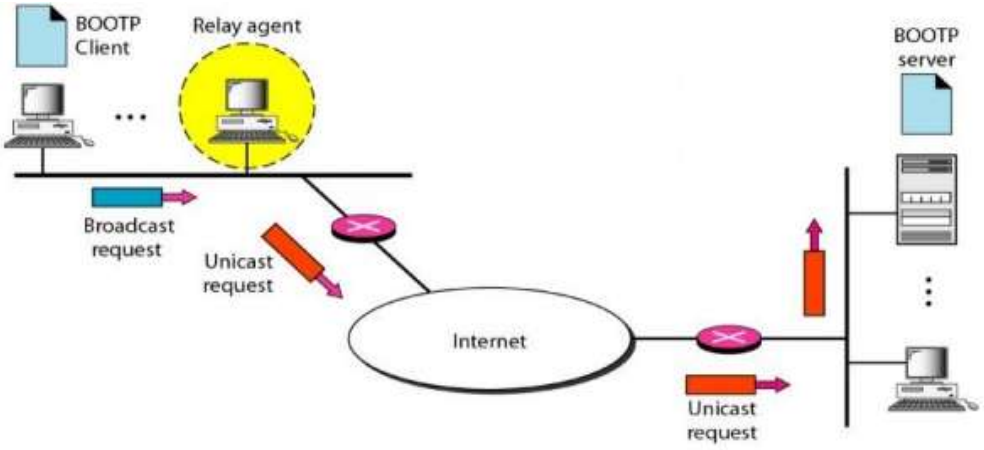
This message makes the routers to broadcast their routing information.

2.6.5. Bootstrap Protocol (BOOTP)

The mapping process of physical address with logical address mapping is facilitated by the client/server protocol called BOOTP. BOOTP is considered to be application layer protocol. The client and server may be available as intra or inter networking, as mentioned by administrator. UDP packet holds BOOTP messages encapsulated in an IP packet.



Without knowing the self-address or the server's IP address, it is possible for a client to transfer an IP datagram. BOOTP is client and server application process. The existence of client and server might be in different networks. Broadcasting from the client is initiated having been unaware of the server's IP address, but router does not allow an IP datagram which is broadcasted.



Relay agent is the host employed to know the unicast address of a BOOTP server. The packet received is encapsulated in unicast datagram and sent to the BOOTP server with request. The packet with unicast address reaches the BOOTP server. The BOOTP server knows the message coming from a relay agent because one of the fields in the request message defines the IP address of the relay agent. The reply is received from the relay agent and forwarded to BOOTP client.

2.6.6. Dynamic Host Configuration Protocol (DHCP)

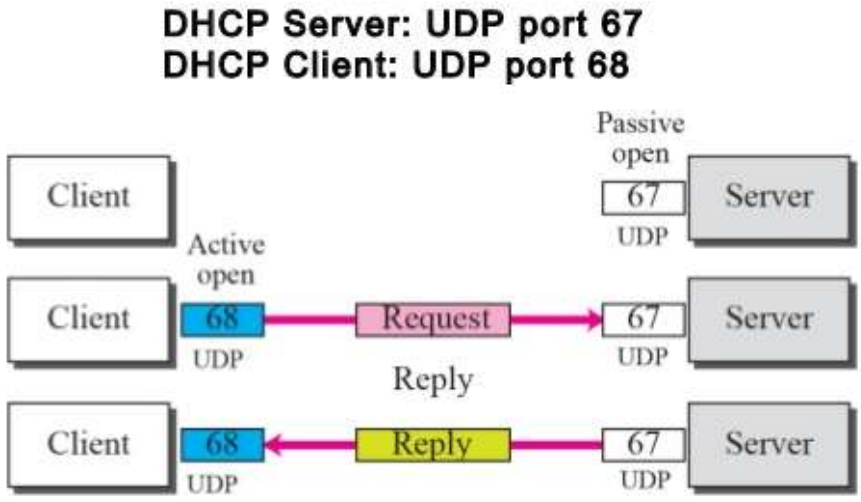
The dynamic address allocation and also static address allocation are devised by DHCP either manually or automatically.

Static Address Allocation

A host BOOTP client executing host requests for a static address from DHCP server, where both are backward compatible. The Physical address and IP address are bound statically in a database of DHCP server.

Dynamic Address Allocation

This DHCP client posts a request for temporary IP address, then the server checks with the unused IP addresses in the pool and allocates an appropriate IP address for a specified duration of time. When DHCP requests its server, it checks for static database. When the address requested is present in the static database, then it's permanent IP address is the response message or if one such address is not present, it selects any available IP address and assigns to client initially and appends this new notification in the dynamic database. The address assigned from the pool is temporary address.



Manual and Automatic Configuration

- Owing to change of physical address and IP address, manual configuration as well as automatic configuration are used.
- The manual config creates static address.
- The automatic config creates dynamic address.

CHAPTER 3

3. Routing

Objectives

- To understand about routing and its types.
- To know about global internet IPv6.
- To explain about unicast routing – RIP, OSPF, BGP.
- To explain about multicast routing – MOSPF, DVMRP, RPF, RPB, RPM, CBT, PIM.

3. Routing

The means of creating routing tables to support forwarding is referred to as routing and forwarding a packet means, it is supplied to station of its destiny. These protocols constantly update the tables which holds the routing information for forwarding and routing consulted.

Routing Table

A host or path for every destined station, even it might be a mixture of destination to path IP packets constitutes routing table. The routing table might be static otherwise it might be dynamic.

Static Routing Table

Here, the details are entered manually. For each destination, the manager marks the route in the table. If any table is built, it cannot be changed automatically when the internet change. Administration has to manually alter the table.

Dynamic Routing Protocols

These protocols use a table that is updated occasionally by protocols such as RIP, OSPF or BGP. The tables of the routers are updated automatically by dynamic routing protocols when there is a shift in the Internet, such router shutdown or the infringement of connection.

3.1. Unicast Routing Protocols

- A routing table can be made static or dynamic.
- A static table holds manual entries.
- A dynamic table holds information that is routinely changed anywhere on the internet if there is a change.
- Routing protocols are developed based on the response to the demand for expressing dynamism.
- The routing protocol combines the rules and procedures which allow routers to inform each other on the Internet.

Optimization

In general, a packet is received at the router from the network and then transferred to a different one. Metric stays organised in order that cost is assigned to pass through the network. The type of protocol decides the metric that is assigned to each network.

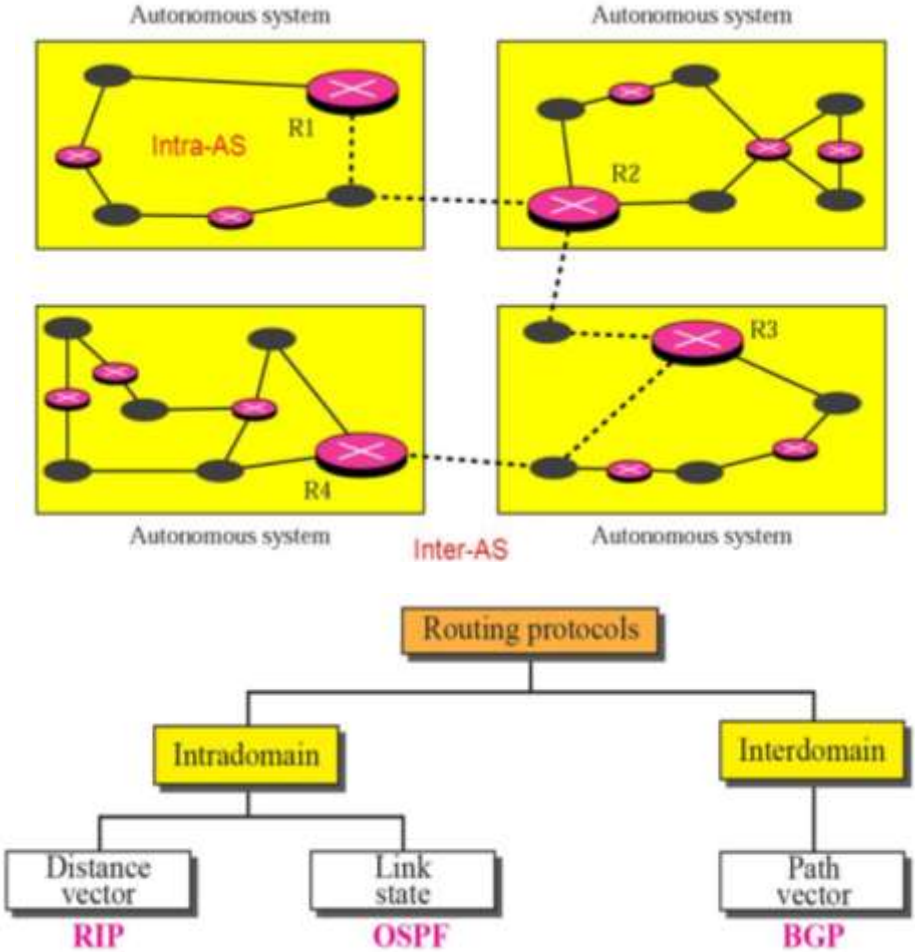
Intra and Inter-domain Routing

From a network, a router receives a packet and transfers it to another network. Typically, a single router has connections to many networks. The metric is thrown into assigning the passage of network expense. Depending on the type of protocol, metrics are allocated to each network.

The internet is divided into autonomous structures. Under the influence of a single administration, a set of networks and routers forms the autonomous appliance.

In the autonomous system, routing table is called inter-domain routing. Intra-domain routing is known as routing between autonomous systems. One or more intra-domain routing protocols can be chosen by each independent system to handle routing within system possessing autonomy.

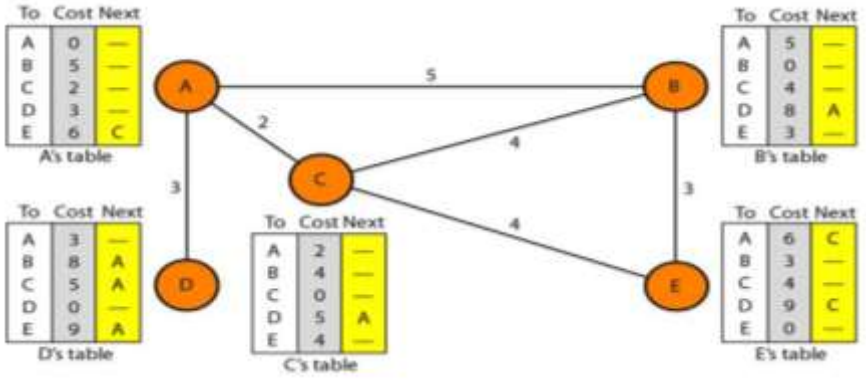
Only one inter-domain routing protocol does routing between autonomous systems.



3.1.1. Distance Vector Routing

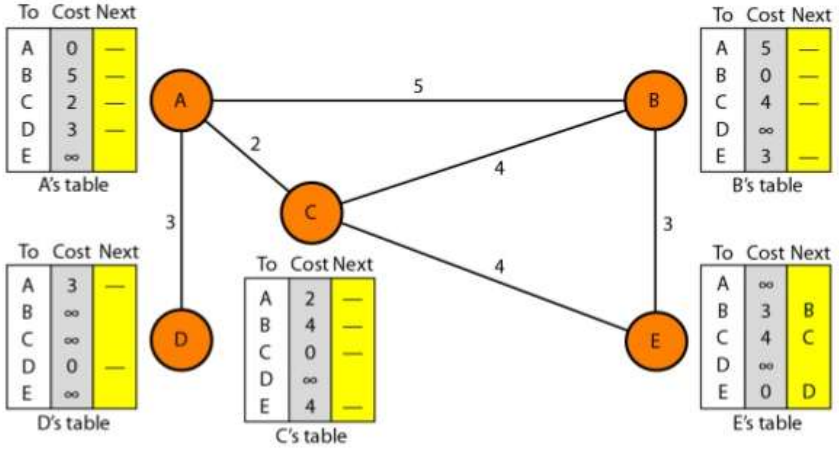
The minimum weighted route is the minimum cost path between the nodes in distance vector routing.

- A table of vectors with least distance to each node is maintained at each node.
- The packets to the desired node are also guided by the table of each node by displaying further hop routing on the path.



Initialization

- The cost or way to reach a node is known by every node. Any node knows the distance from self to its adjacent node that are directly connected to it.
- In order to find the distance between their neighbours and themselves, each node will send a message to their immediate neighbours.
- For any entry that is not a neighbour, then the distance noted to be infinite or unreachable.

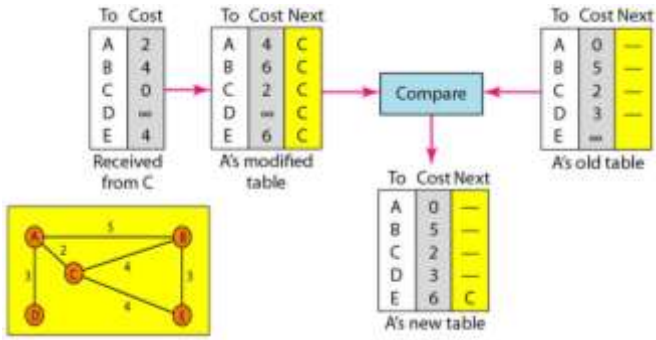


Sharing

- In the routing of distance vectors, whenever there is a shift, the adjacent node of each one can get their table entry regularly.
- Information sharing between neighbours is the whole principal of vector distance routing.
- If Node A is unaware of the E node but node C can and this C node can share its routing table with A, therefore node A can reach node E as well.
- If they render support each other, then the nodes A and C will boost their routing tables as immediate neighbours.

Updating

- If the two nodes receive the neighbouring two-column table, its routing table needs to be modified.
- There are 3 steps while updating.
- In the second column, the receiver must append the cost of any value between itself and the transmitting node.
- If the receiving node uses information from another row, the name of the sender is added to the third column of every row by the receiver.
- The send node will be the next node in the path.
- Each row of the previous table has been compared with the appropriate row of the amended table edition by the receiving node.
- The receiver selects the row with the smallest cost if the next node entry is different. If both remains, the previous node will be held.
- A new row is chosen by the receiver, while the forthcoming station has the same marking.
- By using the tables obtained from other nodes, each node will update its table.



When to Share

Periodic update: Routing table is sent by A node through a constant update of 30's. The duration depends on the protocol which uses the routing of distance vectors.

Triggered update: Whenever a shift is encountered in its routing table, the neighbours receive a routing table with 2 columns from a node. This upholds the update trigger process.

3.1.1.1. Routing Information Protocol (RIP)

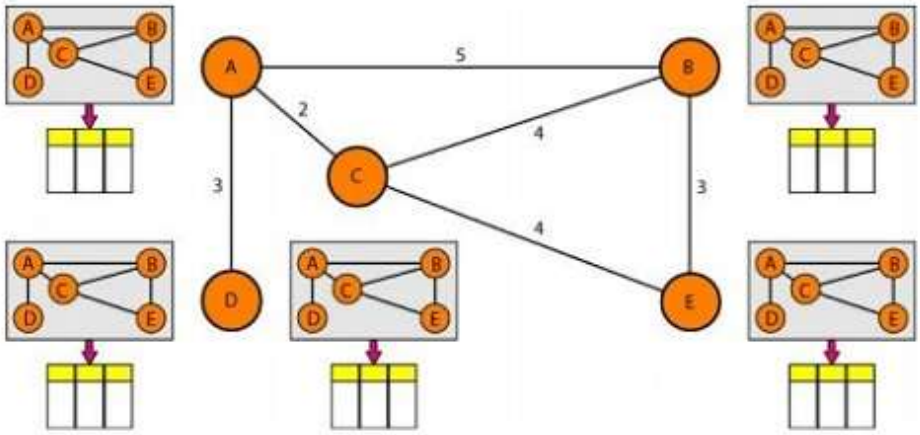
This follows a methodology that implements routing within an autonomous system.

Centred around distance vector routing, this seems to be a very simple protocol. It specifically incorporates distance vector routing with the same specifications:

- Routers as well as networks are dealt in an autonomous system. Routing tables are found in routers where in networks it is missing.
- Routing table refers a network address that is specified in the first column.

3.1.2. Link State Routing

- The RIP uses a very simple metric, the number of links (networks) represent the distance to reach the destination. In RIP, this metric is referred as count of hops.
- 16 is specified instead of infinity, meaning which chosen route may not have more than 15 hops in an autonomous system using RIP.
- The destination address is specified by the next node column.
- The routing table of a node can be constructed by the Dijkstra algorithm when each node in a domain contains the complete domain's topology, the list of nodes, links and the connection types, the metrics used, and link condition.
- This table is unique for every node even though it uses similar topology, because dissimilar understandings are used in calculating the topology.
- The topology of each node and connexion must be dynamic, reflecting the latest state.
- The topology must be changed for each node if there are adjustments at some point in the network.



Somewhere in the network, the understanding of the topology at the start or post transit is not possible.

Each node has partial knowledge of its relationship to the state in terms of type, cost and condition.

Building Routing Tables

The 4 action sets are the requisites of link state routing in order that the routing table.

Link state routing requires four sets of actions in order that the table containing routing data of every part indicates the least costly node for any other node.

1. Link state packet (LSP) is the creation of states by the links of each node.
2. Effectively and efficiently disseminate LSP to any other router named flood.
3. Forms the shortest tree path for every node.
4. Routing table calculation is determined by the tree which holds the quick routes.

1. Creation of LSP

- The LSP can carry large amount of information.
- Example: It carries minimum amount of data, a list of links, a sequence number and age, the node identity 1.
- The identity of the first two nodes and list of links are required to form a topology.
- A new LSP is set apart from older ones, because flooding is encouraged by the 3rd sequence number.
- The old LSP is avoided staying long in the domain due to the 4th era.

Created LSP on 2 occasions:

1. Where the topology of the domain shifts.
2. Periodically dependent.

2. Flooding of LSP

When the LSP is ready at a node, it must not be disseminated to neighbours alone, but to all other nodes.

This procedure is often referred as flooding, and is built on the following:

1. The node created sends out each interface from a copy of the LSP.
2. A node receiving an the LSP received at a node is compared with the copy that is already available.
   a) The new LSP is preserved, whereas the previous is discarded.
   b) Transmits replica of it from every boundary to presume the one that the interface is from the packet arrived. This means flooding will stop somewhere inside the domain.

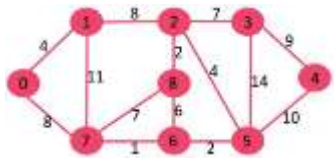
3. Formation of the Shortest Path Tree

Dijkstra's Algorithm

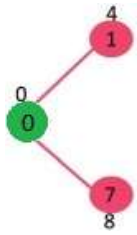
Every other node will have a copy of the entire topology until all LSPs have been sent. A shortest path tree is important for finding the shortest path to another node.

- The tree is a node and a connexion graph, and a single node is known as the root.
- All the other nodes are accessed via a single route from the root.
- If the path from the root to another node is the minimal path, then the path is declared as the shortest path of the tree.
- Algorithm Dijkstra generates the minimal tree path of graphs.
- The split up of nodes as two by algorithm are given as tentative & pentative.
- Neighbours to the new one believes that it alerts, testing it & making them permanent holding a satisfied constraint.

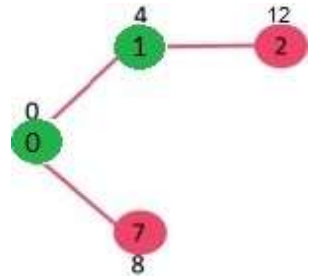
Example



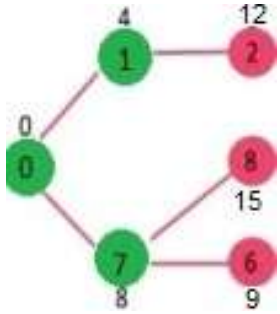
Pass 1:



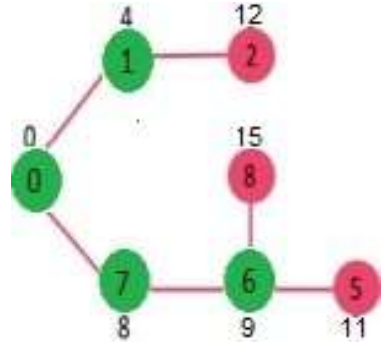
Pass 2:



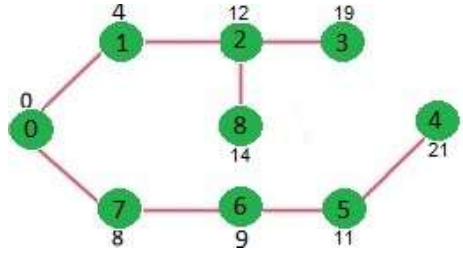
Pass 3:



Pass 4:



Pass 5:



Routing Table Calculation

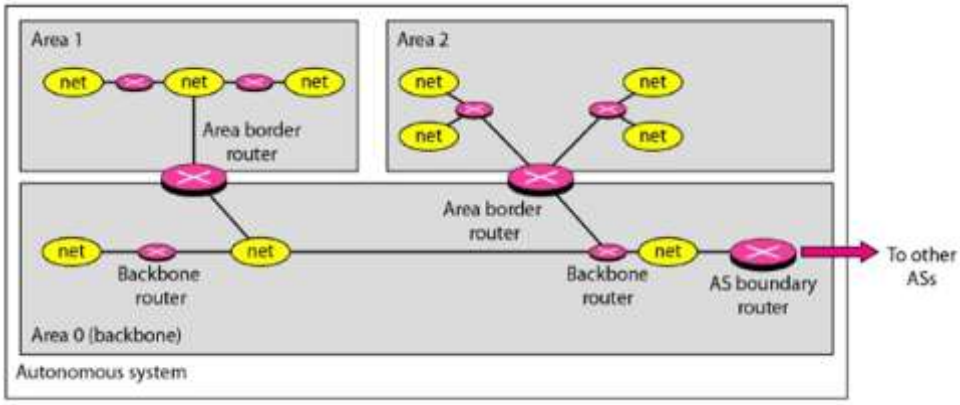
- Each node builds its table using the shortest tree path protocol.
- The routing table indicates the cost from the root to reach each node.

Node	***Cost***	***Next Router***
A	0	-
B	5	-
C	2	-
D	3	-
E	6	C

3.1.2.1. Open Shortest Path First (OSPF)

This routing protocol utilizes LSR method in intra-domain routing-based connection. It also has an autonomous structure as its domain. OSPF breaks an individual system into areas to make it functional. A system is autonomous when it comprises of hosts, different networks and routers. It is possible to split an autonomous system into several different fields. Both networks must be linked to within a network. With routing information, routers within the area cause flood. Special routers called the area boundary routers summarise the area 's details at the boundary of a country and forward it to other regions. Since there is a special backbone, it is important to link all areas within an autonomous system with the backbone. Backbone routers are known as the router within the backbone.

When the backbone with area's connection gets wrecked, an administrator should establish a virtual connexion between routers to allow the backbone functions to be linked as a primary area.



Metric

The OSPF protocol which enables the cost allocation to each path by the administrator, it is termed as the metric.

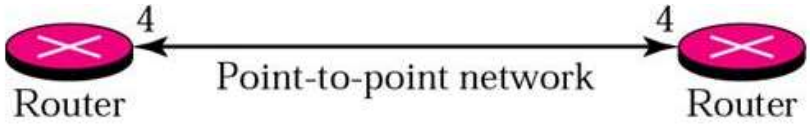
A minimum delay form of operation, the maximum throughput, may be dependent on the metric.

Types of Link

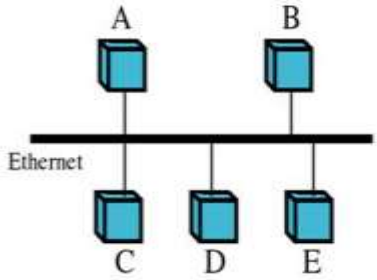
In OSPF, a relation is termed as a connexion.

There are 4 types of links namely Point to point, Transient, Stub and virtual.

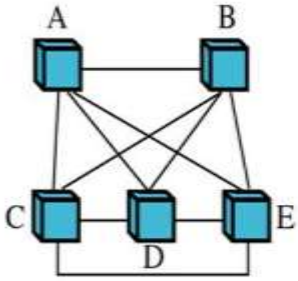
- A connection is established between any routers without the presence of extra host or router using direct point link.



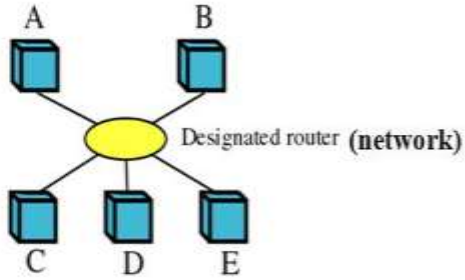
- A network getting attached with several routers form a transient link.
- The data is allowed to enter or move off from a routing device.



a. Transient network

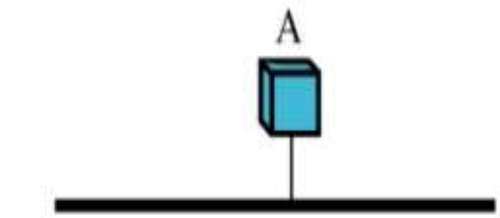


b. Unrealistic representation



c. Realistic representation

- If a network has no other router connector except one, that network is referred as a stub connection.
- Entry and exit of data packets from the network are done using only one router.

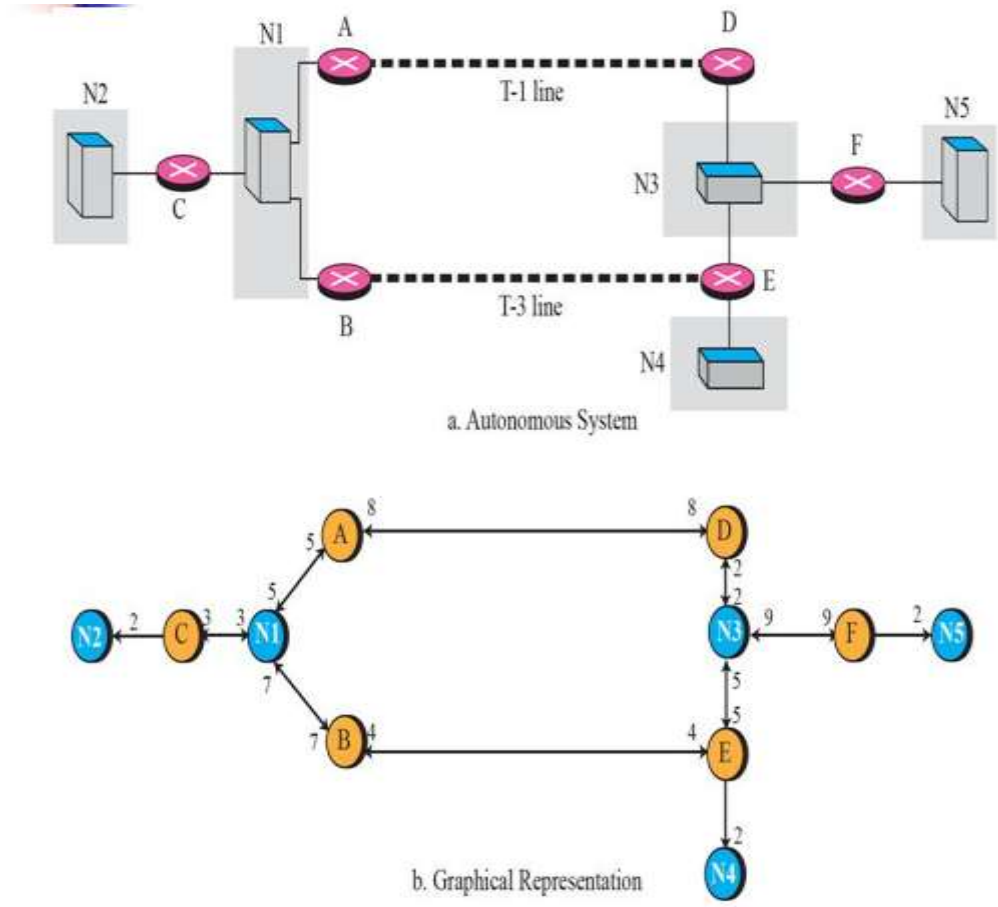


a. Stub network

b. Representation

- The administration can establish a virtual link when the connection with two routers get damaged. It uses the longest path that possibly visits many other routers.

Autonomous System in its Nodal Manipulative Imagery



3.1.3. Path Vector Routing

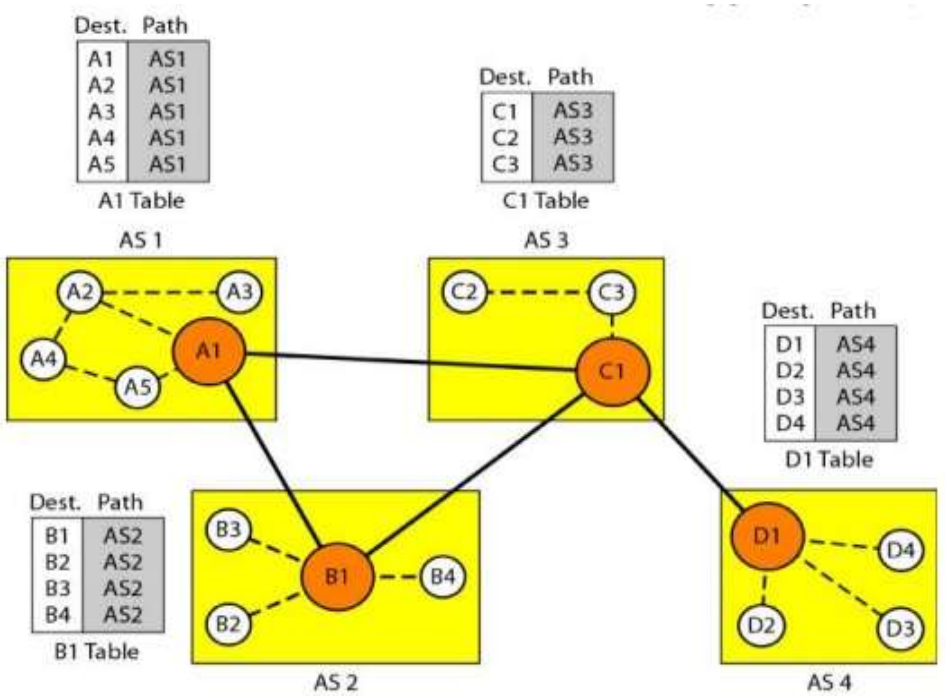
It takes place within the domain and is almost identical with routing over distance vector.

If in any autonomous system, the one node functioning on behalf of the whole autonomous system is the speaker node. The speaker node in an AS generates the routing table and advertises the neighbouring AS t speaker nodes.

The path alone is advertised ignoring the metrics of the nodes.

Initialization

The nodes' reachability is made known only at the beginning of every speaker node within an independent framework.



The speaker nodes of i = 1 to n (here n = 4) for AS1 is A1, B1, C1 and D1 respectively. A1 that creates a starting table showing the location A1 to A5 in AS1 might be reached accordingly.

B1 to B4 is advertised in node B1 and reached via. B1 and so on.

Sharing

With immediate neighbours, the table is shared by a speaker in an autonomous system.

The table of node A1 is shared with nodes B1 and C1. Similarly, the node C1 shares its table with nodes D1, B1 and C1. Also, node B1 shares with C1 also with A1 and finally, the D1 connects with C1 as shown in the figure.

Updating

After showing a while the tables are stabilised for each speaker node after the device.

When a neighbour's two column table is obtained by a speaker node, its table is updated by appending other nodes. After showing a while the tables are stabilised for each speaker node after the device.

The routing table fully shows the route.

Suppose a packet for node A is received by node D1 in AS4, it recognizes that a travel can be made via. AS4 then AS3 and also AS1.

Loop Prevention

Path vector routing helps in avoiding the loops formation as well the unpredictability caused by the distance vector routing. A monitor is performed to all packets received to verify whether the autonomous system is found at the destination path. The message will be overlooked if looping is required.

Policy Routing

It is easy to implement based on path vector routing. It can check the path when a router receives a message. A path and its destination are ignored only if the path violates the policy. With this route, it does not both update its routings and no message is sent to its neighbour.

Optimum Path

Optimum path always suits a company. Every autonomous system can have many routes to reach the destination.

3.1.3.1. Border Gateway Protocol (BGP)

It is an inter-domain routing protocol which uses path vector routing.

Types

The internet canbe divided into autonomous structures called hierarchical domains. An autonomous system is the local ISP which is responsible for providing services to local customers.

It is categorised into 3 types namely Stub, Multi-homed and Transit.

Stub

A separate AS relates to a stub in inter domain traffic which is created otherwise completed in stub.

The traffic is sent from hosts of AS to next ASs for data. Data traffic won't be able to pass through a stub AS. The AS stub can either be a source or sink.

Multi-homed

A multi-homed AS possess more connectivity with another.

AS without data traffic is termed just as a sink or otherwise source. The data traffic observed from other AS is received and it transmits data traffic over to other ASs, it shows non-existence of transient data. It does not allow data to move through from one AS and go to another AS.

Transient

Transient traffic is also allowed in a multi-homed AS which forms a transient AS.

Path Attribute

The route shows a list of autonomous systems, or attributes. Each attribute sets out some path detail. In enforcing its regulation, the attribute list helps the receiving router to make a more conscious division.

The 2 different categories of attributes are given as Well known or Optional.

An attribute is known to be a well-known attribute when it can be recognised by any BGP router. An optional attribute is one that each router does not need to know.

It has 2 categories namely well known mandatory attributes and well known discretionary attributes.

An optional attribute is divided into 2 categories namely, Optional transitive attributes and optional non transitive attributes.

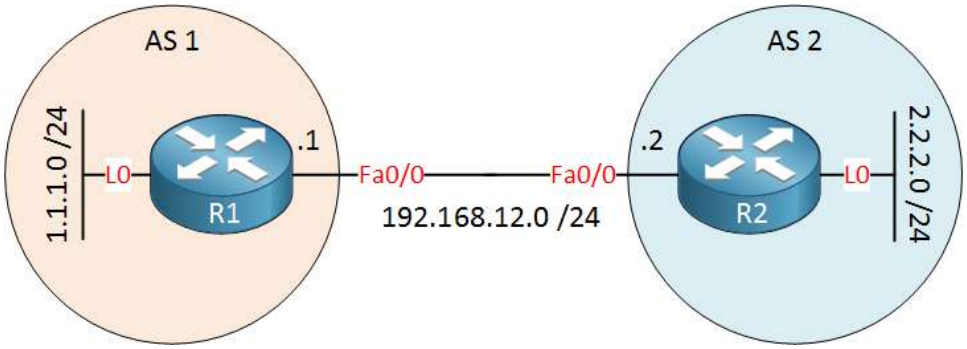
A well-known compulsory feature is the one that appears in the route definition. Each router must recognise one well-known discretionary attribute, but it is not necessary to be included in every update message.

An optional transitive attribute is one that the router who has not implemented this attribute must transfer to the next router, whereas, an optional non-transitive attribute discards when it is not realized by the receiving router.

BGP Sessions

- In BGP sessions, transfer of information happens between the routers.
- The exchange of router details with 2 BGP routers is done by a session.
- BGP is divided into two sessions:
  - External BGP (E-BGP)
  - Internal BGP (I-BGP)
- The E-BGP session is used in data transfer between speaker nodes pertaining to distinct independent systems.
- The I-BGP session is employed in information transmission within an autonomous device between two routers.

Example



- E-BGP session is a session between AS1 and AS2.
- Network data is shared between two speaker routers.
- These two routers in the autonomous system need to gather information from other routers and it happens via I-BGP sessions.

3.2. Global Internet - IPV6

Version 6 of the Internetworking Protocol is often referred to as IPng (internetworking protocol, next generation). It overcomes IPV4 deficiencies. The packet has a stipulated format and duration.

Advantages

- Greater address space.
- Efficient header format.
- Advanced options.
- Life allowance.
- Support for resource allocation and security.

Packet Format

A mandatory base header and a payload forms a packet.

Two sections that forms the payload:

1. Headers for optional extensions
2. Data at the top sectional layer.

The base header covers 40 bytes, while the extension header and the higher layer data comprise to 65,535 data in bytes.

Base Header

Version

The version number of IP is given in 4 bits field. The value of IPV6 is 6.

Priority: The packet's priority in relation to congestion from traffic is given in the 4-bit priority area.

Flow label: The flow mark is a 3-byte field designed for a defined data flow to provide special handling.

Payload data length: The header followed by base of header is given in the next header as 8 bit field. It is an optional IP header with extension otherwise header of an encapsulated packet such as the UDP or TCP.

Also contains the following field in each of the extension header:

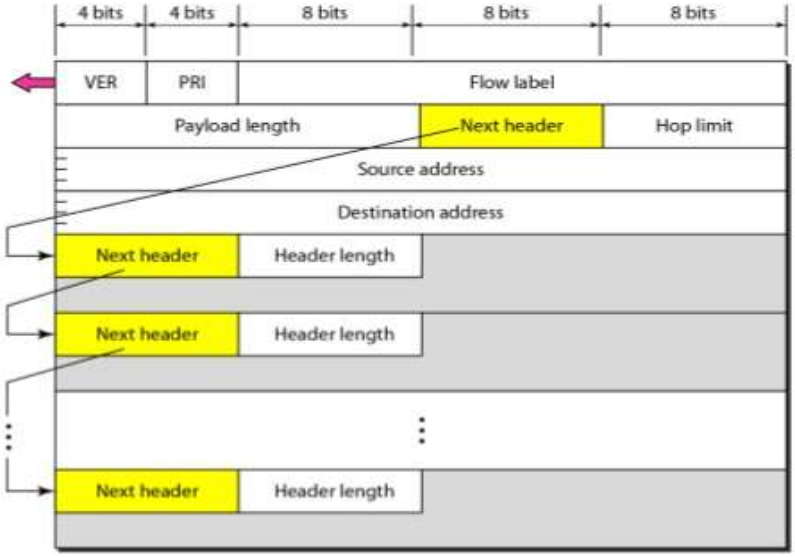
Hop limit: The 8-hop limit area serves the same function as IPV4's TTL area.

Source address: The field whose internet address is of 16 bytes (128 bit) which identifies the original datagram source.

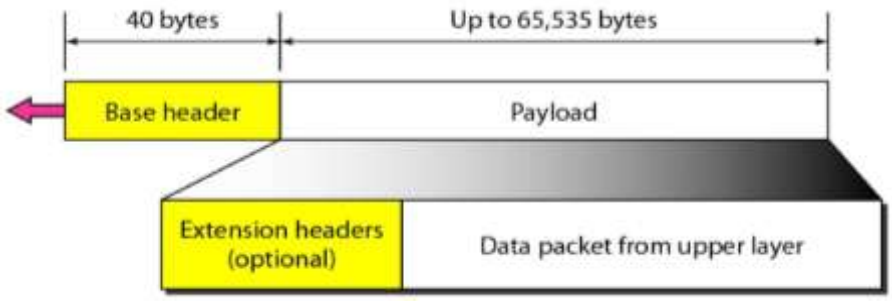
Destination Address

A 16 bytes (128 bit) internet address, which typically specifies the datagram's definite destination.

Datagram Format



Base Header and Payload Format



Priority

The IPV6 packet's priority field determines each packet's priority over other packets comin g from the same source.

There are 2 types of traffics are given below:

a. Congestion controlled traffic
b. Non-congestion controlled traffic

a) Congestion Controlled Traffic

- Congestion controlled traffic is referred as whenever a congestion is encountered and the source can adapt to slowdown of traffic.
- TCP protocol too could respond quickly to traffic by means of the sliding window protocol.
- It is easy to understand that delayed, misplaced or out of order packets will arrive.

Priority Meaning

1. No specific traffic
2. Background data
3. Unattended data traffic
4. Reserved
5. Attended bulk data traffic
6. Reserved
7. Interactive traffic
8. Control traffic

b) Non Congestion Controlled Traffic

This can be applied to the traffic predicted as causing minimal delayed.

Usually, the packets are discarded based on priorities and consistency of the data obtained.

Eg: video and audio.

Data with less redundancy can have a higher priority (15) and a lower priority (8) can be provided by more redundancy.

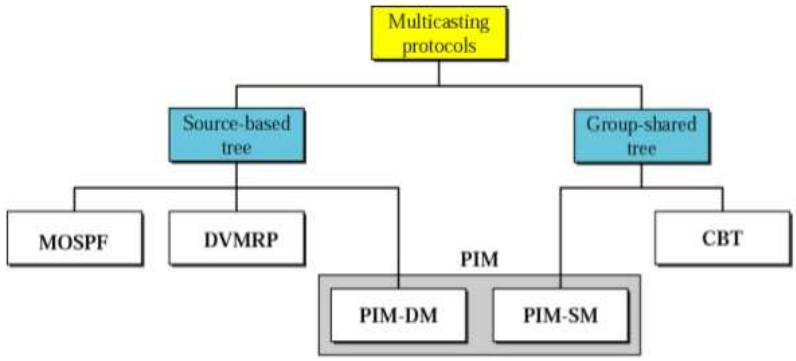
Flow Label

A packet flow is called a series of data packs transmitted between a specific source and destined node which requires a peculiar management of router.

The packet flow is uniquely determined by the combination of source address and flow label value.

3.3. Multicast Link State Routing

Multicast routing are emerged with the extensions of unicast routing protocols.



3.3.1. Multicast Open Shortest Path First (MOSPF)

The source-oriented tree approach is employed in multicast link state routing. Multicast open shortest path first (MOSpF) protocol is an extension of OSPF protocol that uses the routing of multicast link state to create source based trees.

To attach a host's unicast address with the address of the group or addresses that the host supports, the protocol needs a new state of update packet. This is called the group membership LSA. Only the hosts that belong to a specific group are included in the tree. The efficiency of the trees shortest path is calculated on request by router.

The tree can be saved by the same source / group pair for future use in cache memory. MOSPF is a protocol based on data. The datagram is built by the router, revealing the source and group address for the first time that a MOSPF router sees a datagram. the router constructs the shortest path tree for the Dijkstra.

3.3.2. Multicast Distance Vector (DVMRP)

Multicast routing can be supported by easily expanding the unicast vector distance routing. It prevents the routing table to be shared by the router.

A table can be built from start using unicast distance vector table.

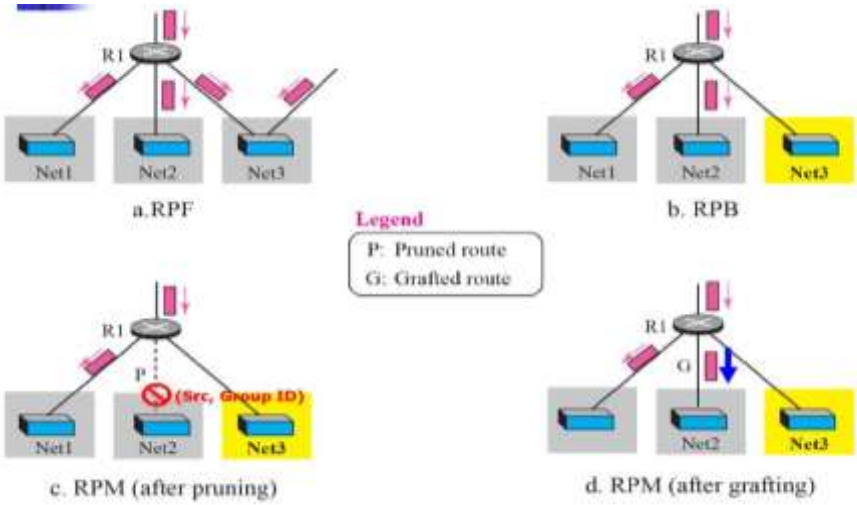
The tree-oriented source is employed for a multicast routing where the router cannot produce routing table. Whenever a multicast packet is received, the packet is forwarded through a routing table.

Flooding

- Broadcast of packets are done by flooding, which also formulates looping in the systems.
- A router receives a packet and sends it out from every interface except the one from which it was sent, leaving the destination group address unnoticed.
- A packet that has left the router can return from another interface again or it is possible to forward the same interface again.

3.3.3. Reverse Path Forwarding (RPF)

The looping caused during flooding is eradicated using RPF.



3.3.4. Reverse Path Broadcasting (RPB)

RPB establishes the shortest path to each destination broadcast tree from the source and ensures that only a single copy of a packet is received at the destiny.

3.3.5. Reverse Path Multicasting (RPM)

Pruning along with grafting is added to RPB by RPM which forms a multicast tree with shortest path as and when dynamic association changes.

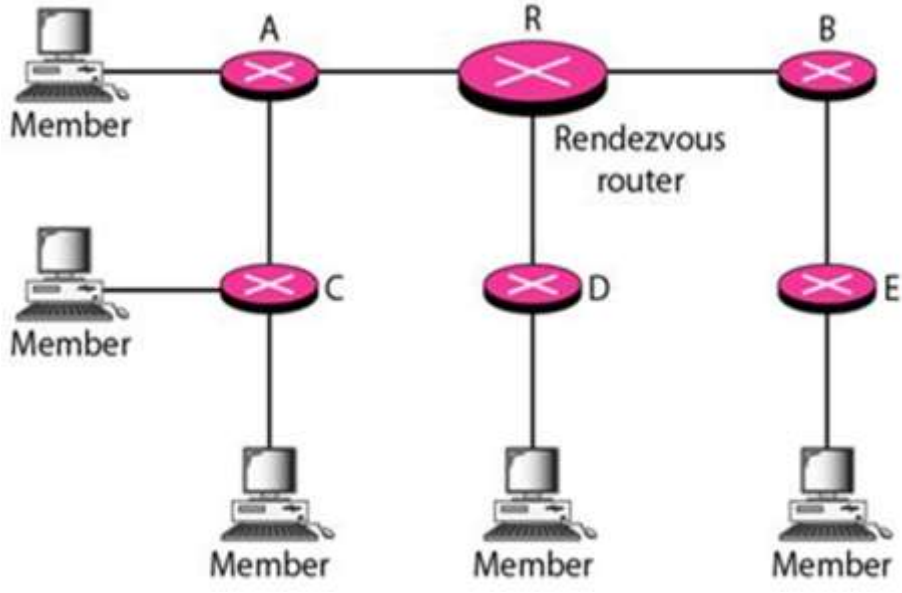
3.3.6. Core Based Tree (CBT)

The CBT is a protocol that is shared by a crew with a tree's base as a core. Each region is chosen as a centre by segregating the autonomous system into regions.

Formation of the Tree

After the core is selected, each router is defined to the unicast address of the selected router.

Each router then sends a unicast join message to indicate that the community wants to be joined.



- The required information is extracted from the message by the intermediate router such as the sender's unicast address along with interface and the successive router receives the message.
- The router can leave from a party by sending a message to its flow routes.

Sending Multicast Packets

- Any source can send a multicast packet to all members of the community after the creation of the tree.
- The packet is then sent to the rendezvous router via the rendezvous unicast address; then distributes the packets to all community members.
- Any hosts within or outside the shared tree may act as the source host.

Selecting the Appropriate Router

The following is the procedure to send packet from the source to the community members:

1. In multicast encapsulation, the part of the tree can either contain the root or not. With unicast destination address, the packet can be sent to the centre of the core. Using the unicast address, this part of distribution is done; the core router is the only re ceiver.
2. Decapsulation of the unicast packet is done at the centre and interfaces that are concerned will have or may receive.
3. The routers receiving the multicast packet, forwards them all to the interested interfaces**.**

3.3.7. Protocol Independent Multicast (PIM)

The PIM has 2 independent multicast routing protocols:

a. PIM DM - Protocol Independent Multicast, the Dense Mode.
b. PIM SM - Protocol Independent Multicast, the Sparse Mode.

a) PIM DM

- PIM-DM is employed when each router needs to engage in multicasting (dense mode).
- It is a type of protocol that prefers tree routing using strategies such as multicasting RPF with pruning and grafting.
- DVMRP's service is almost identical with PIMDM.
- Unicast protocol is used by independent system. A protocol and a table are contained in each router that helps in identifying or seeking an ideal path to destination for the outgoing interface.
- This unicast protocol can be either a RIP or OSPF.

b) PIM SM

- The usage of PIM-SM is where there is very less multicasting (sparse mode), such as WAN, for each router.
- The CBT protocol that uses a group-shared tree that is not justified by the protocol by which the packet is broadcasted.

CHAPTER 4

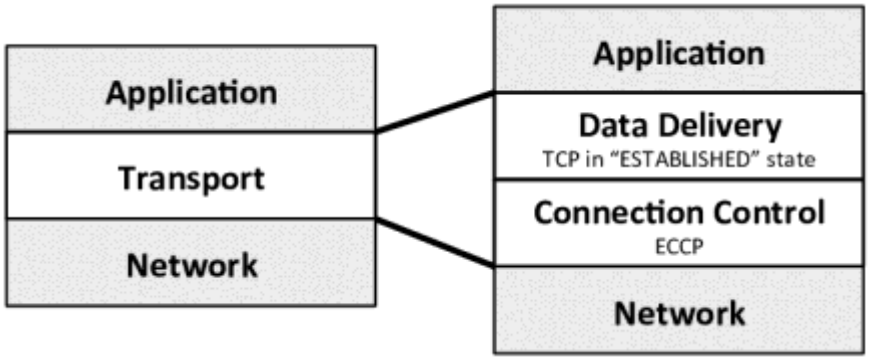
4. Transport Layer

Objectives

- To recognize the transport layer.
- Illustrating the definition of TCP, UDP and SCTP.
- Learning about congestion management and its forms in order to prevent congestion.
- To consider the standard of service (QOS) and its characteristics.

4.1. Overview of Transport Layer

In OSI model, the transport layer is the center layer. Transport layer is mainly used for end to end process delivery, concatenation and segmentation. This layer offers services to the application layer and eliminates services from the network layer.



4.1.1. Duties of Transport Layer

The **main duties of transport layer** are the process to process delivery and by performing a variety of functions like packetizing, connection control, error control, addressing, flow control, providing reliability, congestion control and QoS.

1. Packetizing

The transport layer generates packets from application layer where messages obtained. The method of splitting a long message into smaller messages is said to be as Packetizing. These packets are encapsulated in the transport layer packet data field and the headers are attached. The length of the message is split into smaller parts. Each segment is encapsulated in a separate packet. Header is applied to each packet so that the layer can perform its other functions.

2. Connection Control

In connection control, there are two types:

1. Connection oriented
2. Connectionless

Connection oriented creates a connection (i.e. a virtual route between sender and receiver, where packets are numbered consecutively and communicated bi-directionally. The Connectionless Protocol will handle each packet independently. There won't be a link between them. Each packet may take a different route of its own.

3. Addressing

The client needs the address of the remote machine that the client wants to connect with. Here, the remote computer has a unique address so that it can be distinguished from the ability to connect with the remote computer.

4. Providing Reliability

Flow management and error detection should be implemented for high reliability.

5. Flow Control

Flow control always happened from end to end delivery rather than via a single connection.

6. Error Control

Error correction can be accomplished by retransmission.

7. Congestion Control and QOS

The transport layer may allow the user to define the desired, appropriate and minimum value of the different service parameters when setting up a connection.

- Connection establishment relay
- Connection establishment failure
- Throughput
- Transit delay
- Protection
- Resident error ratio
- Priority
- Residence

4.1.2. Quality of Service (QOS)

The QOS parameters are:

1. Connection Establishment Delay

The delay time between the instant at which a transport connection is sought and the instant at which it is accepted is called the connection establishment delay. The shorter the gap, the better the service is. It's the likelihood of a connection.

2. Connection Establishment Failure Probability

It is probable that the link will not be formed even after the maximum link has been delayed. This could be due to network congestion, lack of table space, or any other issues.

3. Throughput

It measures the number of bytes of user data transmitted per second, calculated over a period of time. It is calculated for each direction separately.

4. Transit Delay

It is time for a message to be sent to the source computer by the transport user and received by the transport layer.

5. Residual Error Ratio

It calculates the amount of lost or clogged messages as a function of the total number of messages sent. The value of this ratio should be zero and as small as possible.

6. Protection

This parameter provides a way to protect the transmitted data from being read or changed by unauthorized parties.

7. Priority

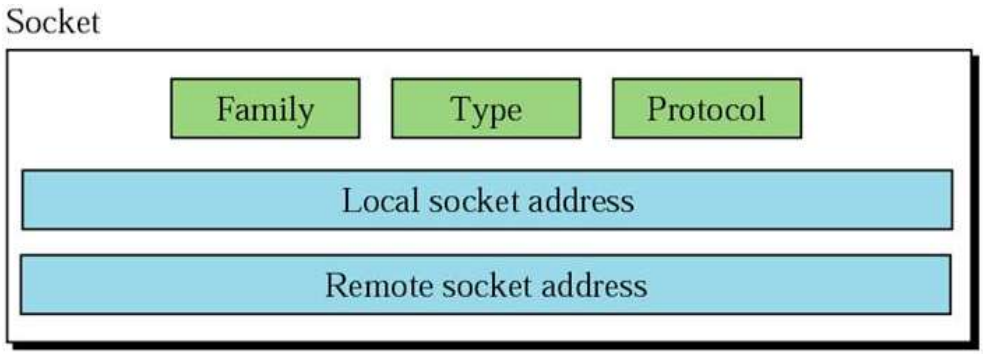
It provides a way for the user to demonstrate that some of its connections are more important than others, while managing congestion. Since service should be provided higher priority than low priority connections.

8. Resilience

The transport layer spontaneously terminates the connection due to internal problems or congestion. The resilience parameter lowers the chances of such termination.

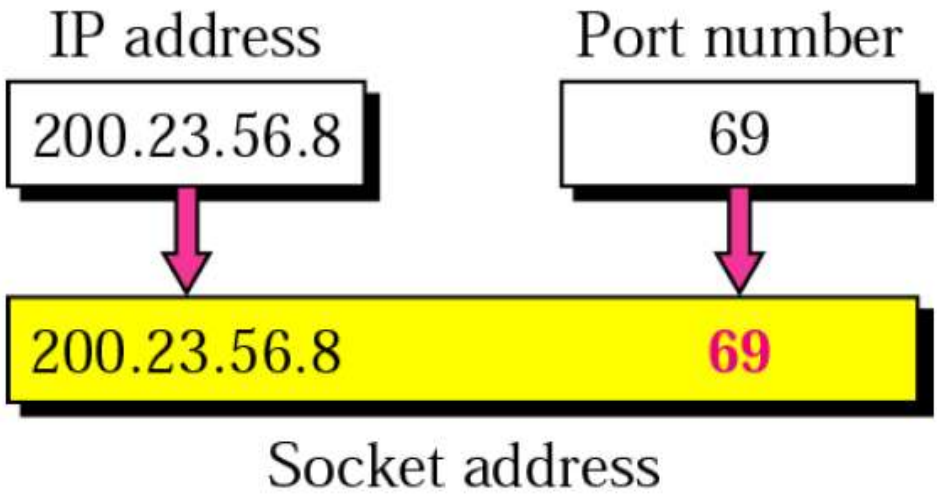
4.1.3. Sockets

The socket specifies a series of system calls or procedures that function as an interface. The socket also serves as an end point where two processes will interact if and only if and only if they have a socket at either end.



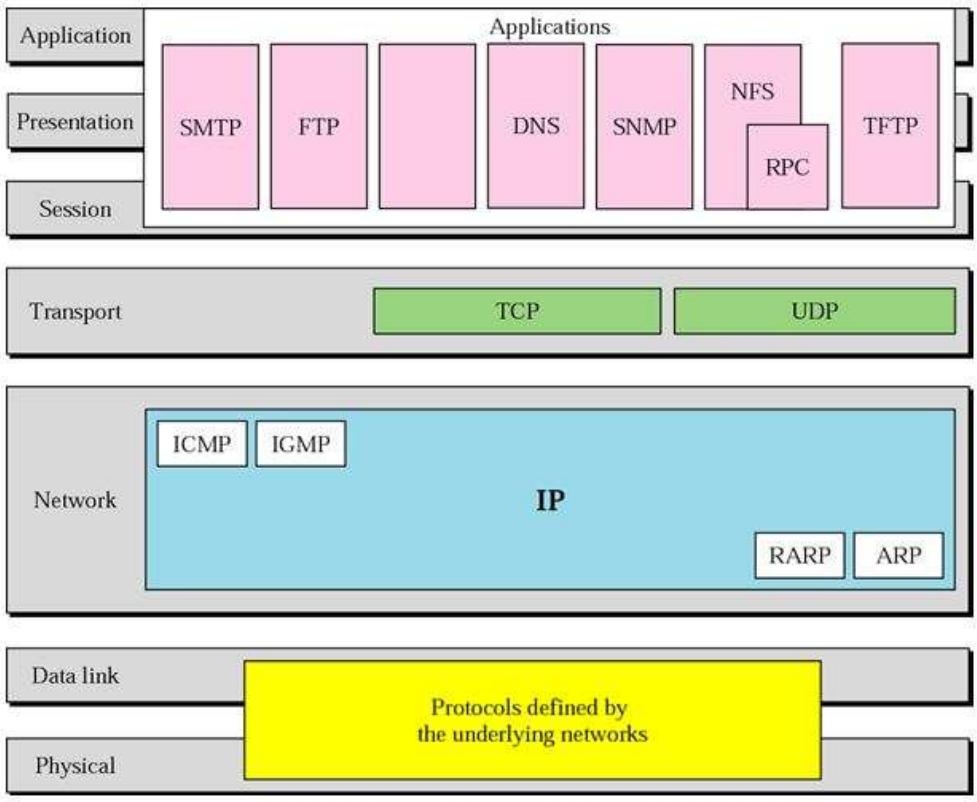
Socket Types

- There are 3 types of sockets. They are stream socket, packet socket and raw socket. In this process to process delivery mechanism, it needs two identifiers, one is IP address and another one is Port number.
- Socket address is defined as the combination of port number and IP address.
- It defines the client and server process uniquely just as the sever socket address defines.
- There is pair of socket addresses used. They are client socket address and server socket address.



Connectionless service - UDP

Connection – oriented service - TCP, SCTP



4.2. User Datagram Protocol (UDP)

The User Datagram Protocol (UDP) is a connectionless, unstable transport protocol. It is a process to process communication where a data unit sent by UDP is called a datagram. UDP has four 16-bit header fields (8 bytes) to every data sent.

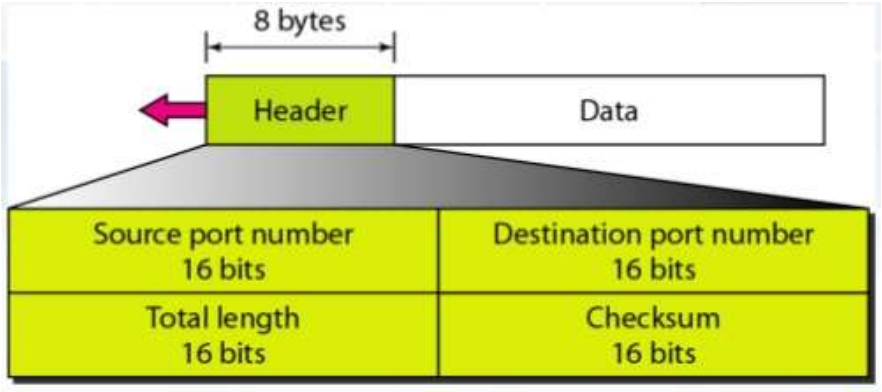
- A length field
- A checksum field
- Source port number
- Destination port number

The port number is used to identify the protocol module which has sent or to receive the data. The use of the standard port number makes it possible for clients to connect with a server without having to specify which port to use.

4.2.1. Purpose of UDP

UDP offers a connectionless packet service that represents the most unreliable attempt to deliver. The delivery of the packets, the right sequence of the packets sent, is not guaranteed. UDP is often used for applications that usually transmit small quantities of data at one time.

It also distinguishes between multiple programs running on a single computer. Each UDP message includes both a destination port number and a source port number. This helps to make UDP applications accessible at the destination to send a message to the current program and to send a reply to the application program. The UDP header is divided into four 16 bits.



1. Source Port

This port is an optional field that indicates the sending process port and the reply to the address of absence portion. It is said to be not interested when 0 is not used.

2. Destination Port

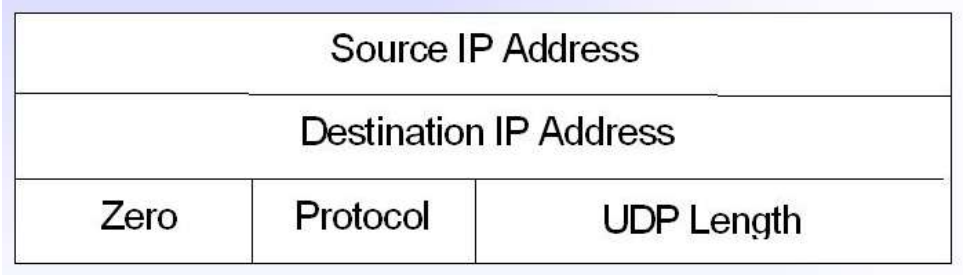
The destination port has significance of a specific destination IP address.

3. Length

This field denotes the size of the UDP packet bytes; it includes the header and the data. The minimum length of the header is 8 bytes.

4. UDP Checksum

It is used to check the consistency of the UDP header. The pseudo code header is used to carry the checksum. It consists of data from source and destination with IP header and the UDP header.



4.2.2. UDP Operation

UDP operations are similar as the transport layer.

Connectionless Services

User Datagram Protocol (UDP) is a connectionless service to every user datagram that UDP delivers. So it is also said to be an independent datagram. In this service, the data comes from same source and delivers to the same destination without any relationship. It does not count the user datagrams and also no link formed for the termination process. Each datagram of the user will follow a different route. UDP cannot send a data in to the stream and hacked into various related datagrams.

Flow Control and Error Control

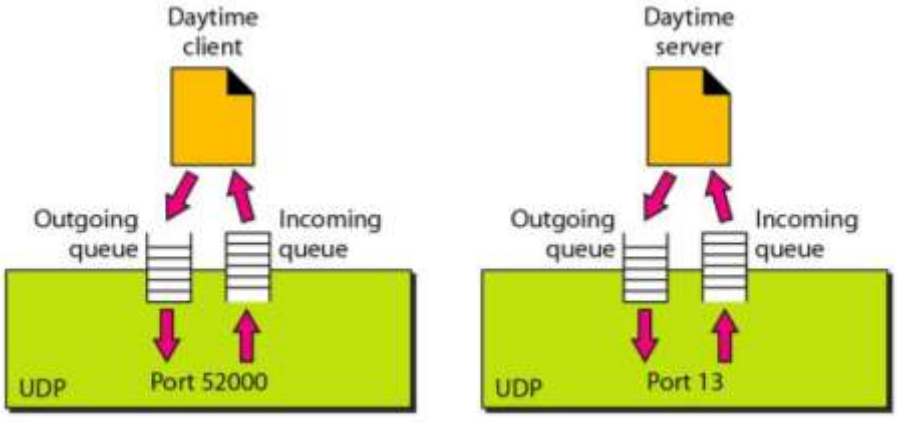
User Datagram Protocol (UDP) is a connectionless transport protocol. There is no window mechanism because of less error and flow control. The received messages are filled in the receivers. In UDP, only in the checksum there is error management. When a message is duplicated or lost, the sender is not able to find. The user datagram will be terminated when receiver detects an error in the checksum. The UDP process provided the mechanism of controlling flow and errors.

Encapsulation and Decapsulation

Both encapsulation and decapsulation helps to send a message from one process to another in an IP datagram.

Queuing

When a process begins, the client demands a port number from the operating system.



The incoming and outgoing queue is generated for each operation. When several number of systems are connected with a process: port number, one incoming and one outgoing queues will be generated. The queues will be killed, once process ended. The outgoing queue is used to send message to the client using source port numbers. One by one the messages will be removed by UDP after delivering them to IP after inserting the UDP header. The outgoing queue can overflow. When a message receives at a client side, UDP immediately checks the incoming queue whether the port number has been generated in the field of destination port number of the user datagram. Once UDP receives the user datagram, the unreachable port message will be send to the server from ISMP protocol. All the incoming messages from same or different server are sent to the queue. After receiving the message at UDP server, it verifies the port number of user datagram destination port number at incoming queue. At the end of the queue, UDP sends an acknowledgement of receiving datagram. UDP terminates the user datagram if there is no queue. Finally ICMP sends an unreachable request to the client. All the incoming messages from same or different client are sent to the same queue. UDP removes the message one by one and delivers those messages to the IP after inserting the UDP header.

4.2.3. Advantages of UDP

- The process like simple request, flow control and error management are supported by UDP.
- Multicasting is also supported by UDP transport layer.
- It also supports controlling process like SNMP.
- It uses updating protocols for certain routes like RIP.
- It is also suitable for internal flow and error management processes.

4.3. Transmission Control Protocol (TCP)

Transmission Control Protocol (TCP) is a process to process or program to program protocol. It is a link oriented protocol where port numbers are used in TCP. Virtual connection is established for data transmission between two TCPs. It uses an error management system and flow control in the transport layer. This TCP protocol is always a secure transport protocol because of connectivity-oriented and it provides reliable features to IP services.

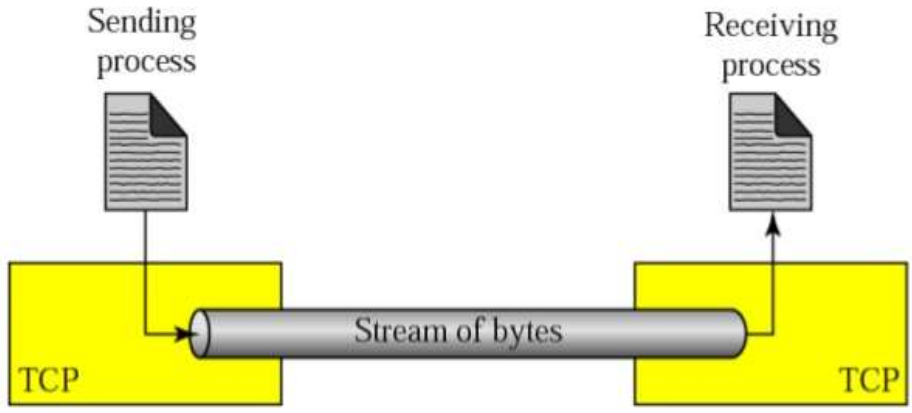
4.3.1. TCP Services

Process to process communication is provided by port numbers in TCP. These are list of well-known port numbers used by TCP.

Port	Protocol	Description
7	echo	echoes a received/send back
9	discard	discards received datagram
11	users	active users
13	daytime	returns day/ time
17	quote	returns quote
19	chargen	returns string of character
20	FTP, data	File Transfer Protocol/ data
21	FTP, control	File Transfer Protocol/control
23	TELNET	Terminal Network
25	SMTP	Simple Mail Transfer Protocol
53	DNS	Domain Name Server
67	BOOTP	Bootstrap Protocol
79	Finger	Finger
80	HTTP	HyperText Transfer Protocol
111	RPC	Remote Procedure Call

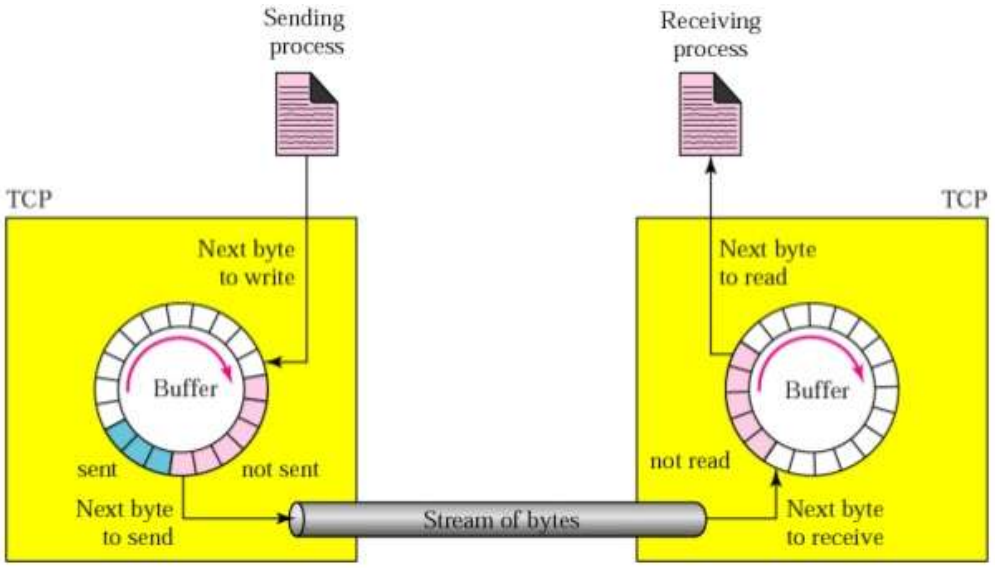
Stream Delivery Service

Transmission Control Protocol (TCP) is a streaming protocol which helps to send the data to convey using stream of bytes and enables to receive data as a stream of bytes. An imaginary tube is used to carry the data between two processes over the internet. This imaginary environment is used to send a mechanism that generates to write or read the stream of bytes over the sending/receiving device.



Sending and Receiving Buffers

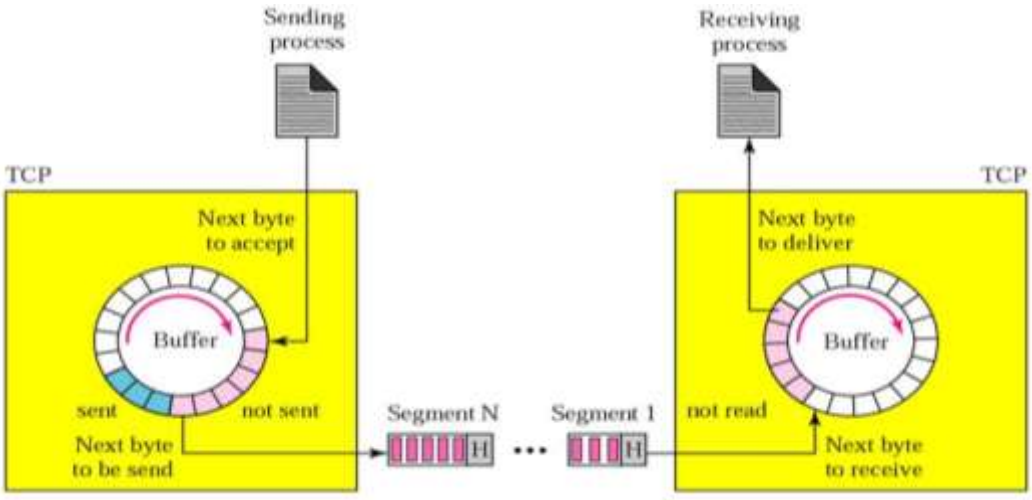
The write or read operation over data cannot be done at the same speed in sending and receiving process. Buffers are using in TCP for storage. There are mainly 2 buffers used in TCP, one is used for sending buffer and the receiving buffer and another is used to implement a buffer is to use a 1 byte location circular array.



There are 3 chambers of buffer used at the sending side; they are white section, gray section, and colored section. Empty chambers are filled by sending process in white section. The sent data which is not recognized are presented in grey area by bytes. TCP holds these bytes in the buffer till the acknowledgment receives. The sent data by the sender are presented in the colored area. At the receiver side the operation will be completed easily. There are 2 areas in the circular buffer; they are white and colored. The empty chambers are filled by bytes in the white buffer which is obtained by the network. The received bytes are read by the receiver side and filled in the empty chamber in colored buffer area. The chamber is returned and recycled to the empty chamber area, once the byte is read at the receiving process.

Segments

An empty chamber which is obtained from network is filled by bytes in the white buffer. All the received bytes in receiving methods are read in colored segment. The chamber is returned and recycled to the empty chamber area, once the byte is read at the receiving process.



Full Duplex Communication

In this communication, TCP offers data flow in both direction and at also at the same time. In both the direction TCP buffer sends and receives the segments.

Connection Oriented Service

In a TCP, if one process needs to send and receive the data from the next process, there establishes a two connection between them. They are,

1. In both directions, data to be exchanged.
2. Connection termination.

It is a stream oriented environment and it acknowledges the distributing bytes to its destination. In an IP datagram, if the TCP segment is encapsulated the data can be misplaced, corrupted or sent out of order and then it should be resented. Different paths can be used to reach the destination.

Reliable Service

TCP is always said to be a secured protocol for transporting services. It detects and verifies the arrival of data and its safety.

4.3.2. TCP Features

Number System

TCP is used to track the segment number in the header for both the sent and received segments. Acknowledgment number and Sequence number are the two fields in the segment header.

Byte Number

TCP numbers the transmitted bytes of data to each link. The numbering begins with a number generated at random. They're stored in the send buffer and counted.

Sequence Number

Once the bytes numbered, sequence number will be assigned to each segment by the TCP. In that segment, sequence number is the first byte number for each segment. Combination of both control and data information uses the sequence number in each section. When there is no data in the segment, sequence number will not be defined. The receiver should have a sequence number when it has control information to acknowledge it.

Acknowledgement Number

The acknowledgement number will be used by the sender and also receiver site to confirm the received bytes. The acknowledgment number should be given to the next byte information to be received. Cumulative acknowledgement gives the received number of last byte.

Flow Control

Flow control uses byte-oriented. The sum of data which is transmitted by the sender is monitored by data receiver.

Error Control

Error control is byte-oriented. Errors are detected by data unit segments.

Congestion Method

The sender's data will be governed by the receiver and in network it is also measured by the degree of congestion.

Source Port Address

This address is the 16-bit field. It is used to specify the port number where the host sends the application in the section.

Destination Port Address

This address is the 16-bit field. It is used to determine the port number where the host receives the application in the section.

Sequence Number

It is the 32-bit field. It is used to determine the first data byte and also the destination. Initial Sequence Number (ISN) is generated by random number generator for each parity in different direction.

Acknowledgement Number

It is the 32-bit field. The byte number is determined for the receiver which receives from another entity.

Header Length

This is 4 bit field. It shows the number in TCP header of 4 byte words.

Reserved

This is a 6 bit field. It is mainly reserved for future.

Control

This is 6 bit field. It is used to identify 6 separate flags or control bits and also more than one can be set at a time.

URG – Urgent Pointer field value

ACK – Acknowledgement value

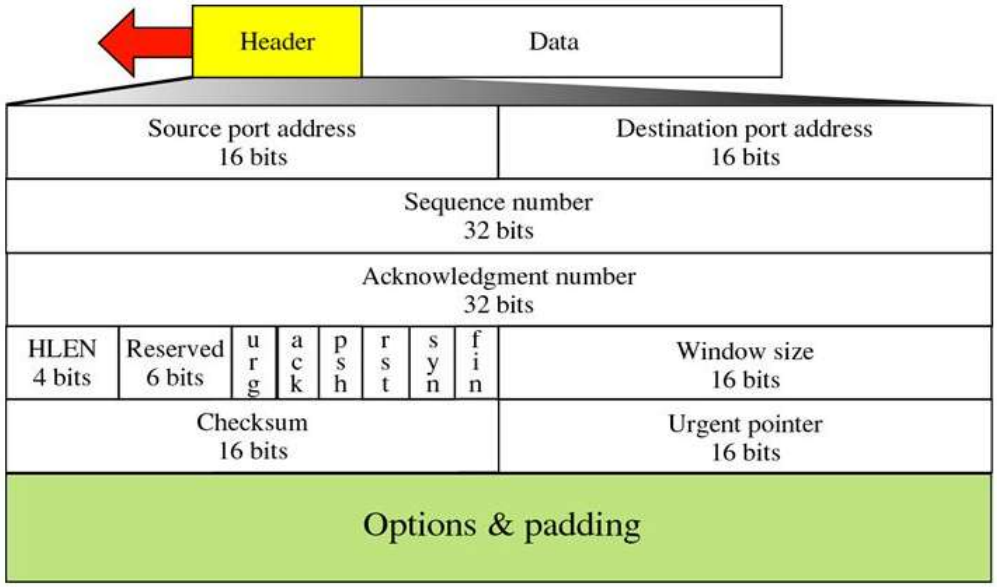
PSH – Data Push

RST – Connection resetting

SYN – Synchronization

FIN – Connection termination

Segment Format



Header Segment Consists of 20-60 Bytes

4.3.3. TCP Connection

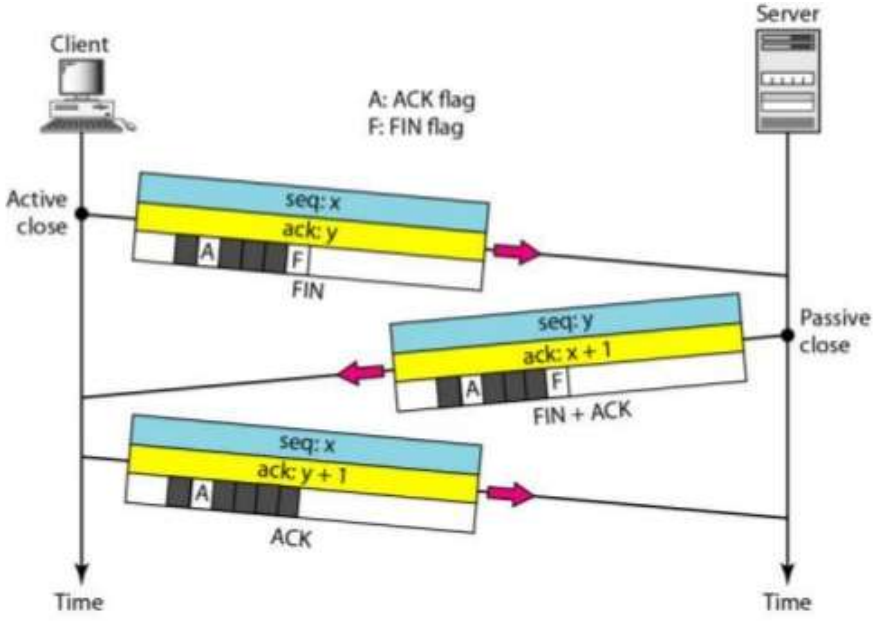
Transmission Control Protocol (TCP) makes a virtual path between the source and the destination so it is said to be a connection-oriented network. Through this virtual path all the segment messages are sent. It is made by these 3 phases. One is connection establishment; two is data transfer and last is connection transfer.

1. Connection Establishment

In this connection establishment, full duplex method is used to transmit the data. The segments are sent concurrently by linking two TCPs along with two devices. Before the data transfer each device must initialize and also should able to receive the data from another device.

Three – Way Handshaking

- Three-way Handshaking method is linked with TCP system.
- Using TCP transport layer the client sends the link to the server.
- Server will start the program.
- An passive open request is an request accepts the connection which is ready and it is informed to TCP by server program.
- An active open request is issued by client program.
- If any client wants to be connected to open server it will be informed by the any server to its TCP.



There are 3 steps in this phase,

- SYN segment – It consumes one sequence number and it does not carry any data.
- SYN +ACK – It does consume one sequence number and it does not carry any data.
- ACK segment - It does not consume one sequence number and data.

2. Data Transfer

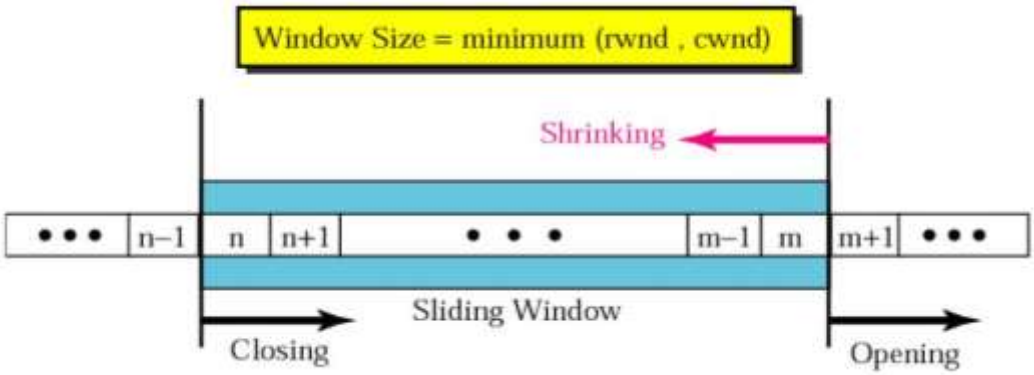
Bi-directional data transfer can take place after the link has been formed. Both the client and the server will submit data and acknowledgements. The receipts are piggybacked with the results.

3. Connection Termination

Link termination uses their way or four-way handshaking. The FIN segment consumes one sequencing number if it does not carry data. The FIN+ACK segment consumes one sequencing number if it does not carry data.

4.3.4. Flow Control

TCP uses a sliding window to control the movement. The sliding window protocol used by TCP is between Go-back N and selective sliding repeat window. It is byte-oriented and variable size fixed.



The window covers a portion of the buffer containing the bytes obtained from the method. Bytes in the window can transit and also sent without acknowledgement. The two wall on the left and right side acts like an imaginary window. It carries out three activities: opened, closed, and reduced. In these operations, the receiver instructions should be followed by the sender. The windows opened by shifting the right wall to the right side. The windows closed by shifting the left wall to the right. The window shrinks by shifting the right wall to the left. The

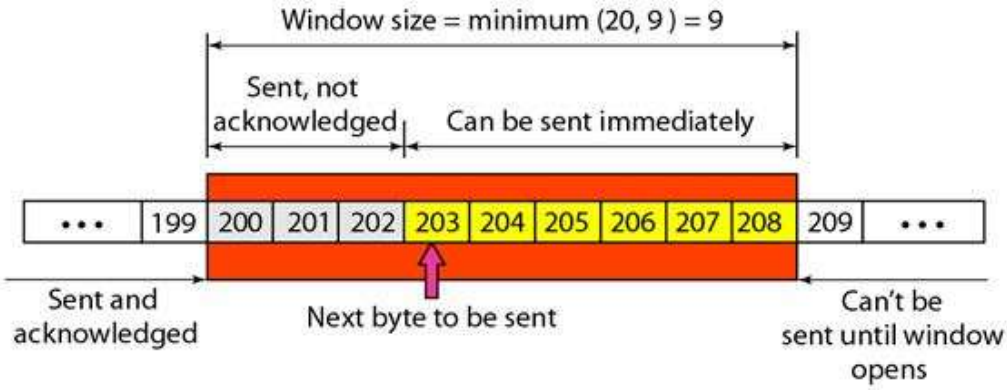
left wall moves only if it receives an acknowledgement from the previous one. The size of the window is determined at the one end and it should be lesser than two values.

1. Receiver window (rwnd)
2. Congestion window (cwnd)

Acknowledgement section at the opposite ends value is marked at the receiver window. The value which is calculated by the network is to prevent congestion at congestion window. By this method the flow of data is controlled and transmission is also more effective, so the data in not overloaded in the destination. This method is said to be as byte-oriented.

Note

- Window size is small (rwnd & cwnd).
- In this window the source will not submit the complete data.
- The receiver can open and close the window, but not in narrow.
- At the any time, destination can submit the acknowledgment at any time.
- Shrinking window does not result in destination.
- The sender window is shut downed by the receiver temporarily.
- After the window shut downs, it can send segment of 1 byte.



4.3.5. Error Control (Retransmission)

Transmission Control Protocol (TCP) is a stable protocol for the transport layer. A data stream is delivered to TCP to transmit the overall stream to the application. An application program will not lose or duplicated. It provides reliability through error management. A system identifies the corrupted, lost, duplicated or out of ordered segments by error management. Checksum is used to identify the error and its correction and also the time recognition.

Checksum

- The damages segment will be checked by the checksum in each segment.
- The corrupted segment is discarded by TCP and it is said as missing.
- 16 bit checksum is mandatory for each segment in TCP.

Acknowledgement

- Acknowledgement is used to validate the stream of data fragments in TCP.
- It is not acknowledged by ACK segments.
- The sequence number is acknowledged and does not hold any data in control segments.

Retransmission

- The retransmission is the center of error control.
- It is retransmitted once the section delays, corrupted or skipped.
- The retransmission is twice the section.
    1. When the timer of retransmission expires.
    2. The 3 duplicate ACKs received at the sender side.

1. When the Timer of Retransmission Expires

- For all sent segments one timer Retransmission Time-Out (RTO) is retained by TCP.
- The segment which sent earlier is also retransmitted because delay and lack of ACK received or missed acknowledgement.
- In TCP, the RTO value is dynamic in nature and modification for the segments is done by Round Trip Time (RTT).
- The acknowledgement and time taken by the segment to reach the destination is said to be as Round Trip Time (RTT).

2. The 3 Duplicate ACKs Received at the Sender Side

- The receiver receives more than one segment at the time of segment loss, because it is not able to save.
- When one segment is lost and the receiver receives so many segments out of order that they cannot be saved (limited buffer size). It follows three duplicate ACK rules and immediately retransmits the missing segments.
- This happens because of limited buffer size and it is said to be as quick retransmission.

Out of Order Segments

- The segment may be out of order because of delay, skip or discard of the segments.

- The missing of segments may flag it as out of order.
- Most of the implementation is not able to terminate as out of order because they stored as temporary.
- Data which is received by TCP is stored as temporary when it is out of order.
- Out of order segments should not be delivered and it should be verified by TCP.

4.4. Stream Control Transmission Protocol (SCTP)

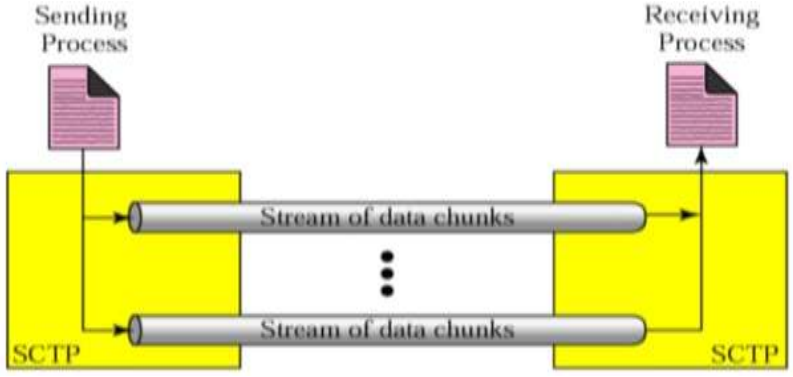
Stream Control Transmission Protocol (SCTP) is a message-oriented, modern, efficient transport layer protocol. It provides better efficiency and reliability. SCTP incorporates the best User Datagram Protocol and Transmission Control Protocol functionality. This protocol maintains the boundary for message and also able to detect the duplicates/missing data on out-of-order segment. This protocol has a system for managing congestion and flow.

4.4.1. SCTP Services

In STCP process-to-process communication uses all the well-known parts in TCP.

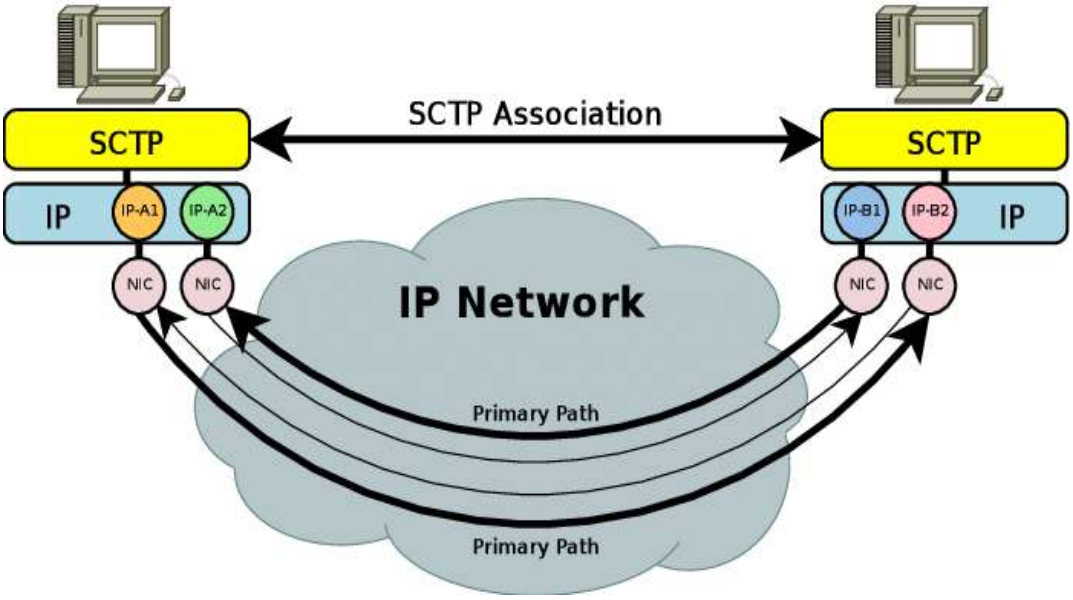
Multiple Streams

Both the TCP client and server have a connection as a single stream. This protocol enables multiple stream connection to be established that is referred to as Stream Control Transmission Protocol association. If one stream has been blocked, the other streams will deliver the data. Multiple streams is always allowed in Stream Control Transmission Protocol association.



Multihoming

Multihoming service sends and receives a host that can specify several IP addresses for each and every association. Alternate interface will be used without any interruption when one route fails. SCTP allows multiple IP address at both the end.



Full Duplex Communication

Full duplex is used when data flows in both directions at the same time. In both connections, it has buffer sender and receiver to send/receive the packets at both directions.

Connection Oriented Service

When process A needs to send the data and receives another data from process B. The following shall occur:

1. An association should be created between two SCTPs.
2. In both the directions, data can be send/receive.
3. Then association can be terminated.

4.4.2. SCTP Features

Transmission Sequence Number

In Stream Control Transmission Protocol (SCTP), the one to one communication may be used to transfer the message due to fragmentation. By numbering data chunks in SCTP, data transmission is managed. Transmission Sequence Number (TSN) is used to number the data chunks. TSN is 32 bit long in Stream Control Transmission Protocol (0 & 2-1).

Stream Identifier

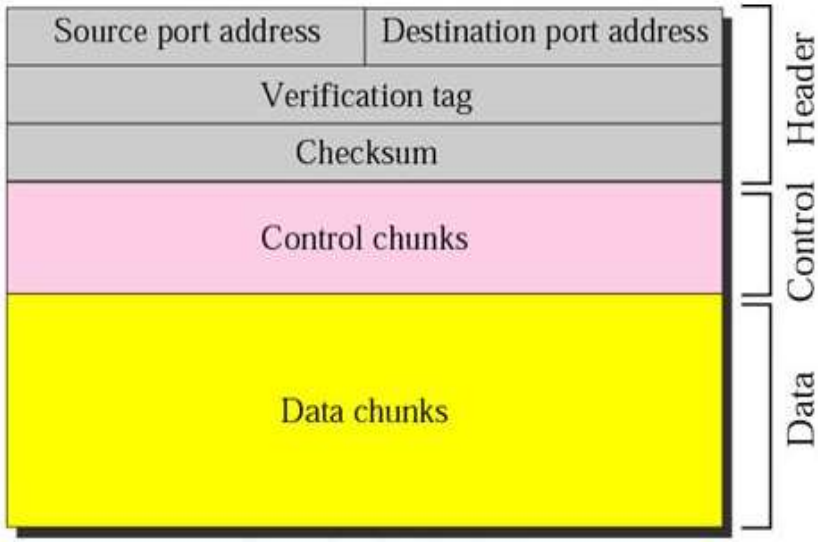
Stream Identifier (SI) is used to identify the stream in SCTP. Before arriving to the destination each data should have its SI in header.

Stream Sequence Number

In SCTP destination, the data packets will be forwarded to their stream once it arrives. Stream Sequence Number (SSN) is used to define the data packets in their stream.

Packets

Information is transmitted as data packets and control information is transmitted as control packets. A lot of control and data packets can be bundled in a single packet. The Stream Control Transmission Protocol packet also has a same mechanism as the Transmission Control Protocol portion. TCP has parts, SCTP has packets.



Acknowledgement Number

The ICP acknowledgement numbers are applied to sequence it. It is byte-oriented. However the Stream Control Transmission Protocol acknowledgements are always packet oriented by reference TSN.

Flow Control

Flow control is used to prevent exhausting the receiver.

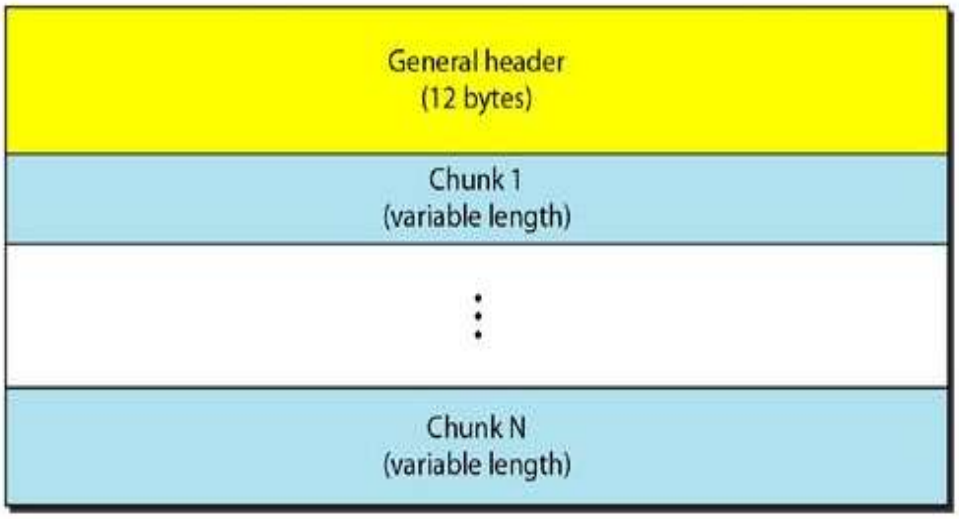
Error Control

It implements error management to provide reliability.

Congestion Control

It is used to determine how many data chunks can be rejected on the network.

Packet Format



General Header

Source Port Address

It is 16-bit field address. Port number is determined for sending the packet.

Destination Port Address

It is 12-bit field address. Port number is determined for receiving the packet.

Verification Tag

Packets are connected by using this verification number. It is used to describe the process that is replicated in every packet during the process.

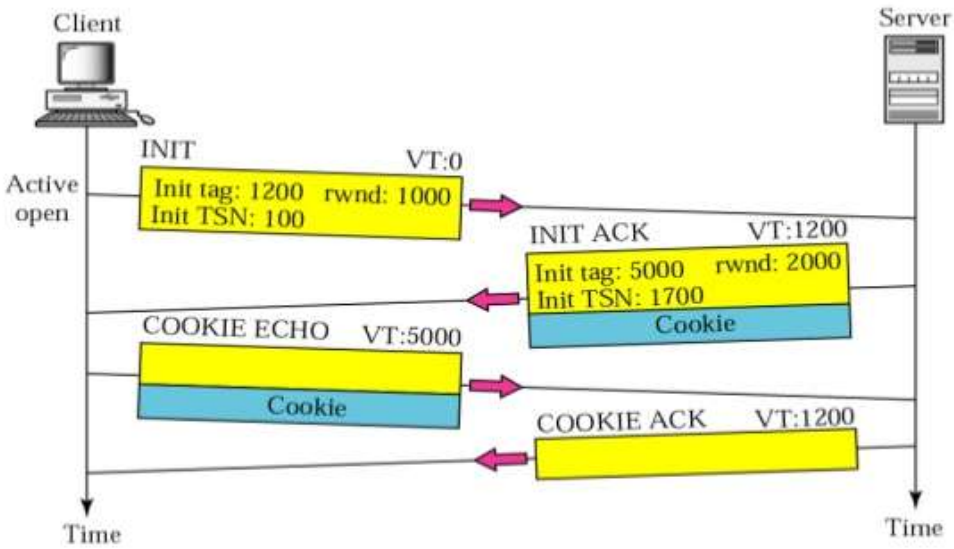
Checksum

It is 32-bit field. It includes the CRC-32 checksum.

4.4.3. SCTP Connection

1. Association Establishment

- Four Way Handshaking process is used in the SCTP.
- The server uses SCTP as a transport layer and client creates a link with the process.
- The client initiates the link (active open) and server must be ready to accept the link (passive open).
- The 4 way handshaking steps are:
  1. The INIT chunk is the first packet send by the client.
  2. The INIT ACK chunk is the second packet.
  3. The COOKIE ECHO chunk is the third packet send by the client. It includes a simple chunk that echoes without any change.
  4. The COOKIE ACK chunk is the fourth packet send by the server. It includes acknowledgement receipt of the cookie echo chunk.
- The solution is to pack the information and send it back to the client. This is called generating a cookie.



2. Data Transfer

- Bi-directional data transfer can take place after the association has been formed.
- The boundaries are established and recognized.
- After processing, the message will be considered as a single unit and it mage as data chunk process it till it fragments.

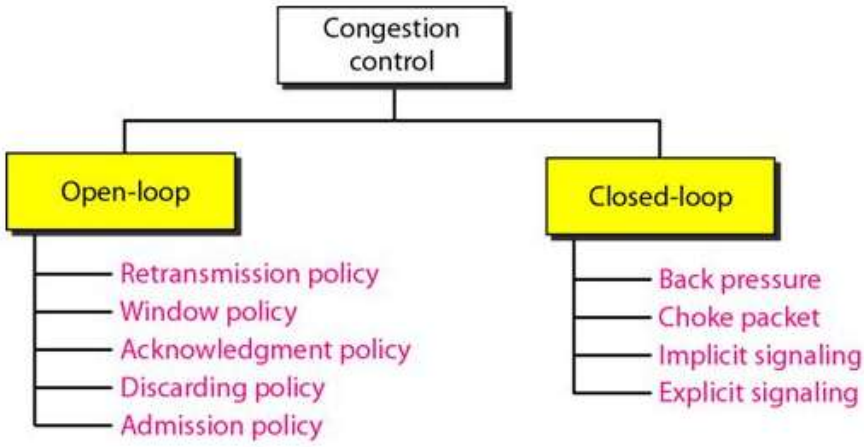
- Then data chunks will be broken down by adding header to the message.
- A single TSN is given for the message or fragments which has set of information.

3. Connection Termination

- The process should avoid sending the new data once the process has been terminated.
- The queued data should be submitted or closed before the request sends for termination.
- The three packets are used for process termination; they are shutdown, ACK shutdown, and full shutdown.

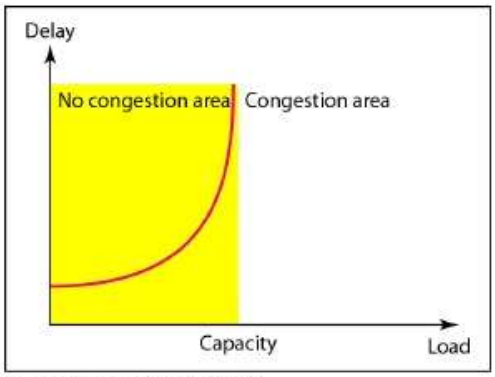
4.5. Congestion Control

The process of avoiding congestion before it occurs or eradicate congestion after it has been occurs is said to be as congestion control. It is divided into two wide categories.

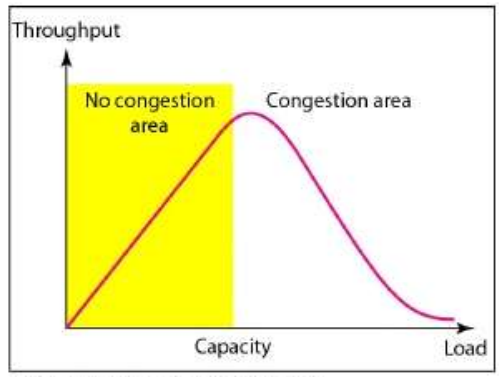


4.5.1. Congestion - Network Performance

- When network is loading the number of packets sent to the network is greater than the ability of the network. Here the number of packets the network can handle is said to be a network congestion.
- The mechanism of managing congestion and holding the load below capacity is congestion control.
- The routers and switches have buffer queues to carry both the before and after processing packets.
- It has 2 factors to calculate the network efficiency. They are throughput and delay.



a. Delay as a function of load



b. Throughput as a function of load

4.5.2. Open Loop Congestion Control

In the form of open loop congestion management, policies are implemented to avoid congestion before it occurs. This can be happened by source or destination.

1. Retransmission Policy

- This policy is also unnecessary.
- The packets are retransmitted when the sender send the packet as misplaced or identical.
- Network congestion occurs by retransmission.
- A strong transport policy will avoid congestion.
- The policy on retransmission and timers must be structured to maximize performance and also avoid congestion.

2. Window Policy

- Congestion can cause on the sender windows.
- Instead of using back N window, the selective repeat window manages congestion.
- The packets will be mistrustful when timer times out. Few packets may receive safe at the receiver in back N window.
- The lost and corrupted unique packets are tried to send in the selective repeat window.

3. Acknowledgement Policy

- The acknowledgement policy can also have an effect on congestion.
- If the receiver does not accept any packets it receives, it can slow down the sender and help avoid congestion.

- A receiver can send an acknowledgment only if a packet has to be sent or a special timer has expired.
- Only N packets can be acknowledged at a time by the receiver.
- Sending fewer acknowledgements in the network.

4. Discarding Policy

- Congestion and transmission integrity is avoided by discarding policy.

5. Admission Policy

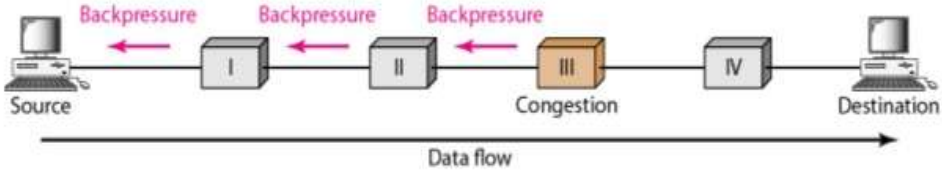
- It is also said to be a quality of service mechanism.
- In this policy, congestion is avoided in virtual circuit networks.
- In flow requirement switches are first verified before moving to the network.
- A virtual circuit link is refused by the router when there is congestion.

4.5.3. Closed Loop Congestion Control

In this control, congestion is minimized only after it occurs.

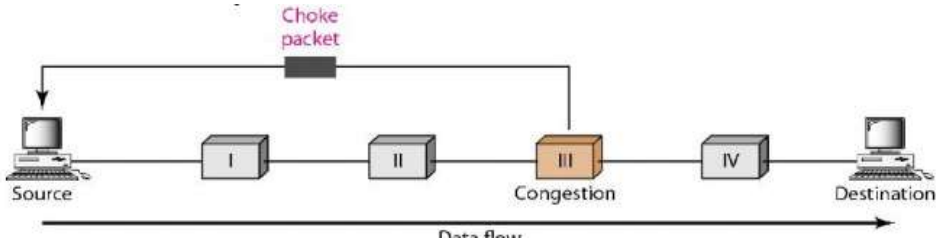
1. Back Pressure

- The technique of back pressure refers to the congestion management system under which the congestion node avoids receiving data from the immediate upstream node or node.
- Congestion occurs in the upstream node and data's also refused from its node.
- Back pressure is a also said to be a node-to-node congestion control.
- This control starts by beginning and flows till the opposite side to the source.
- It happens only in virtual circuit networks and data flows of each node are identified only by upstream node.



2. Choke Packet

- A choke packet sends packet to the source to identify congestion.
- It sends congestion warning directly to the source station from router.
- The intermediate nodes are not able to alert.



3. Implicit Signaling

- In this signaling, there will be no contact between source and congested node.
- If there is any congestion in the system the source will inform to the network.

Eg:- If a source needs to send multiple data streams and if there are no acknowledgements, there may be a congestion. A network congestion may cause by delay in receiving the acknowledgement or it may slows down the source.

4. Explicit Signaling

- In this signaling, the signal will be sent to the source or destination to find the congestion.
- The method of choke packets is not as same as signaling.
- For this reason, a separate packet is used, and this signal is included in the packets that carry the data.
- In either forward or backward directions, the congestion control of the frame relay occurs.

a) Background Signaling

- A bit can send through the packet of data by travelling in opposite direction of the congestion window. In case of discarding, the bit will send an alert to the source to slow down the process.

b) Forward Signaling

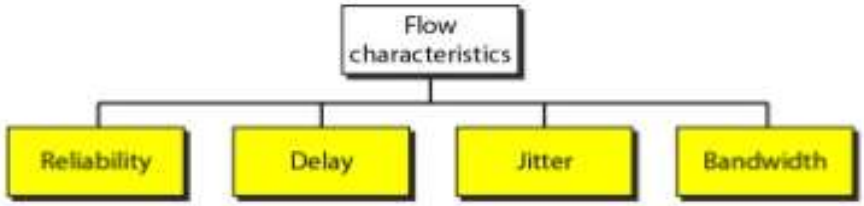
- In this signaling a set of packets can travel in a congestion window as bits. When congestion occurs in particular bit the alert will be sent.
- Congestion is reduced by slowing down the acknowledgement.

4.6. Quality of Service (QOS)

A problem which occurs in internetworking to flow seeks to be attained is defined as Quality of Services.

4.6.1. Flow Characteristics

There are 4 types of characteristics are attributed to a flow.



1. Reliability

Reliability is the characteristics that a flow requires Lack of reliability implies the failure of a packet or an acknowledgment that includes retransmission.

Eg:- Email, transferring a file and internet connectivity.

2. Delay

This is used get delay from source to destination.

Eg:- Audio/video conference, telephony, remote access.

3. Jitter

Jitter is a difference in the delay of packets belonging to the same flow.

Eg:- If 4 packets depart in time 0,1,2,3 and arrive at 20,21,22,23 all have the same delays, 20 units of time.

If 4 packets arrive at 21,23,21,28, they will have different delays: 21,22,19,24.

Jitter is known as the variation in the delay of the packet. High jitter means that the difference between delays is large, low jitter means that the variance is small.

4. Bandwidth

There is a variable bandwidth depending on the different applications. In video conferencing, millions of bits per second are required to refresh the color screen while the total number of bits in email may not even exceed a million.

CHAPTER 5

5. Application Layer

Objectives

- Understanding Traditional Applications.
- Illustrating the basics of e-mail, POP3, SMTP, MIME, IMAP.
- Illustrating HTTP design and online documents.
- Explaining FTP.
- Learning the fundamentals of web services.
- To understand the DNS and how it is distributed.
- Learning about the SNMP.

5.1. Traditional Applications

Among the common global technologies, the World Wide Web (WWW) and E-mail, facilitates the model of request / reply methodology. The request is submitted to server and the response is received accordingly. Such applications come under the category of "traditional" applications, as they describe the kind of applications which have been in existence from the very start of the computer age. (Web plays an advanced role than simple text application, yet the file transfers that predated it have their roots). On the other hand, the group of applications which are popular in the current era ranges as applications based on streaming and imaging.

Generally, an important point has to be taken a close look with priority before exploring the details of these applications. It is the discrepancy of application programs with that of application protocols which has to be understood. The Hyper Text Transport Protocol (HTTP), for instance, is an application protocol used to obtain Web pages from remote servers. Other application programs like web clients - Internet Explorer, Chrome, Firefox, and Safari deliver an appealing window environment to users, even though they use the HTTP protocol in common, to connect over the Internet with web servers. It is a known truth that the protocol is published and standardized which allows interoperation of application programs created by various companies and individuals. That's how many browsers can communicate with all of the web servers.

5.2. Email

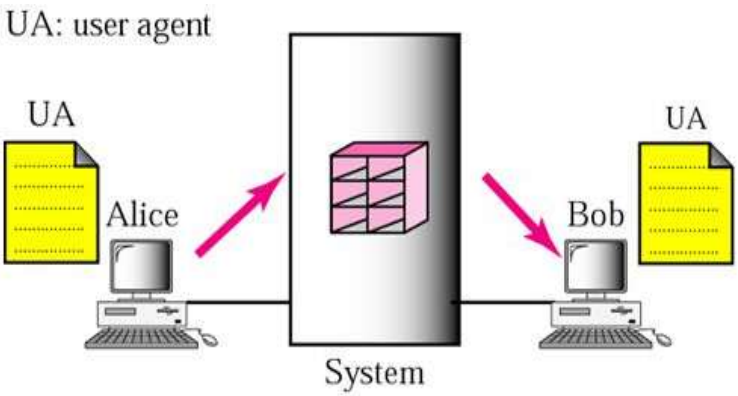
Electronic mail (Email) is among the most popular internet services. The message sent via electronic mail was brief and contains only text. It allows text, audio and video to be included into a post. This enables one message to be sent to one additional recipient.

5.2.1. Architecture

The email architecture is explained in 4 scenarios.

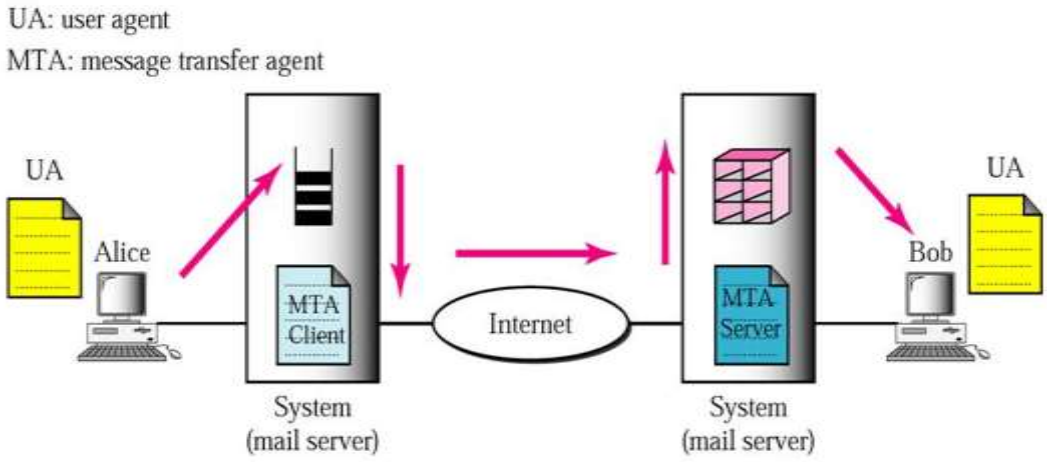
1. First Scenario

The Email's sender and receiver are uses on the same device that a common device directly links. A mailbox is a particular file with permission restrictions that is part of a local storage device. It is used to receive and store messages.



A user wants to send a message to Bob, while Alice is operating. Alice runs a User Agent (UA) program to prepare and store the message in Bob's mailbox. The message contains addresses of mailbox of sender and receiver. Using a user agent the contents of his mailbox will be capable of being retrieved and read at their convenience. If the sender and the email recipient dwell on the same platform, we just require user agents.

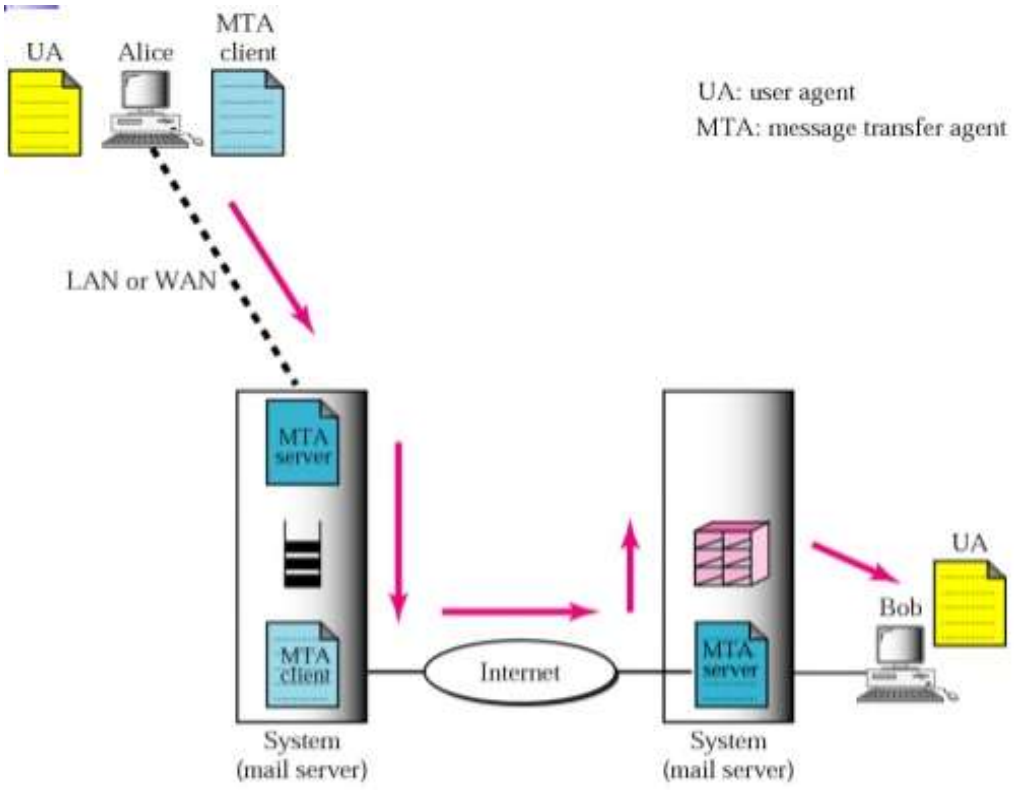
2. Second Scenario



To deliver her message to the device at her own place, an agent programme is necessitated for Alice. The mail server uses a queue to store message which is waiting for its turn for transmission. In order to retrieve messages saved in the system's mailbox at his location, Bob also requires a user agent application. The receiving message has two client and server message transfer agents. It obviously requires 2 UA and 2 MTA (client and server) where the sender and the recipient of an email exist in separate systems.

3. Third Scenario

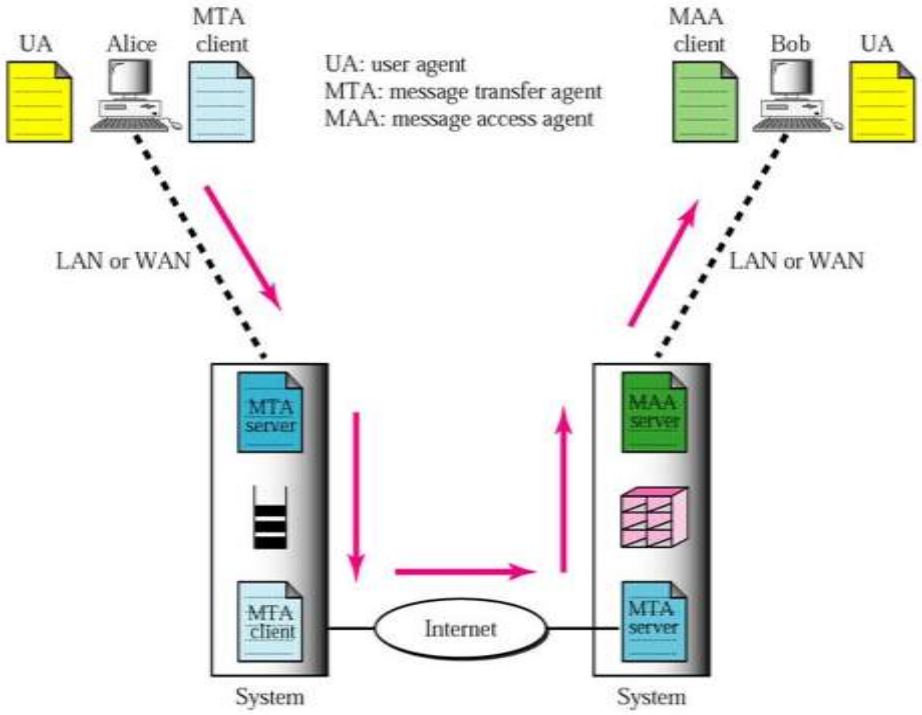
Bob is specifically related to a system of lies. Alice is connected using a dial-up modem, DSL or cable modem though WAN, to the system.



Alice requires a UA to prepare its message and send the same via the LAN / WAN. The UA is called, which further proceeds to call MTA client, when Alice has to send a request. The MTA client creates a link that is running all the time with the MTA server on the device. The machine at the Alice site queues all received messages and the MTA client sends the messages to the Bob site. It requires 2 UA and 2 MTA pairs when the sender is attached via LAN / WAN to the mail server.

4. Fourth Scenario

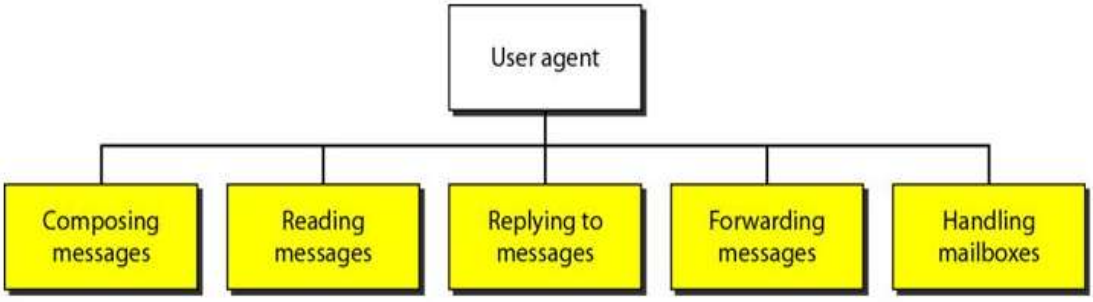
Here also a WAN / LAN links Bob to his mail server. The mail's reception at Bob's mail server makes Bob to retrieve it. The Message Access Agent (MAA) retrieves Bob's messages to the client. Then a request is initiates by the client to MAA server which is active always to facilitate message transfer.



When both sender and receiver are linked via a LAN / WAN to the mail server, 2 UA, 2 MTA pairs and 2 MAA are required and this turns to be the general issue.

5.2.2. User Agent

UA provides different uses with service to promote the sending and receiving of messages. The services provided by a UA were,



1. Composing Messages

The email address that has to be sent will be composed by the user and assisted by agent.

2. Reading Messages

A user agent processing the incoming messages.

3. Replying to Messages

A user will then use UA to respond to a message after reading the message. Typically, a UA agent helps in enabling the user in responding the actual sender in addressing all message recipients.

4. Forwarding Messages

Forwarding is described as the sending to a third party of an email.

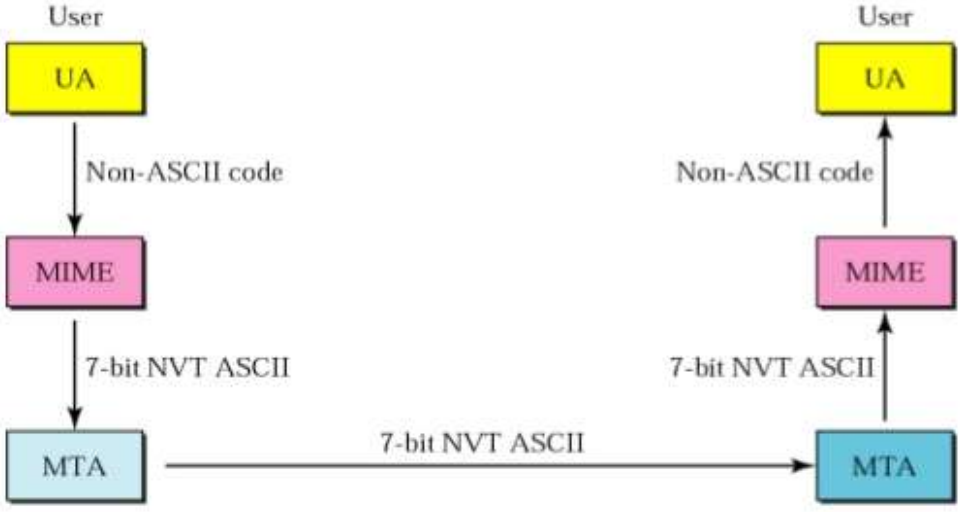
5. Handling Messages (Inbox & Outbox)

The agent utilizes a special format for managing a file that is a box.

5.2.3. Message Transfer Agent

A) MIME-Multipurpose Internet Mail Extensions (MIME)

MIME is a supplementary protocol that allows e-mail sending of non-ASCII data. It converts non-ASCII data to NVT ASCII data at the sending zone and delivers at MTA client that is to be sent over the internet. Further, the message is converted back to the original data at the receiving side.



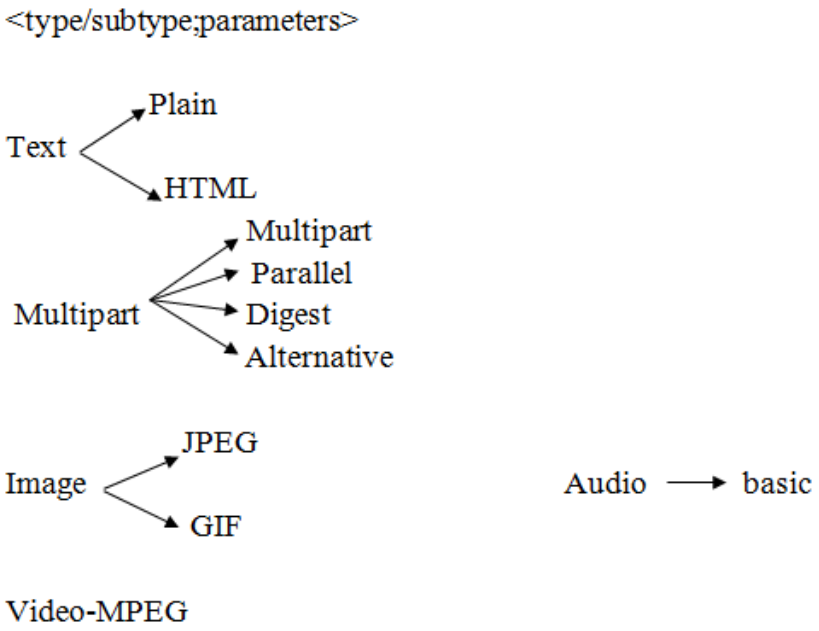
MIME defines 5 headers to describe the parameters that were applied to the original email header portion. They were MIME version, Content version, Content transfer encoding, Content ID, Content description.

MIME Version

The header specifies the version used in MIME. The new update is version 1.1.

Content Type

Header specifies the data's type which is in the message's body. The subtype of content and material is divided by a slash.



Content Transfer Encoding

This header defines the entire message uniquely within a multiple message setting.

ID=<content-id>

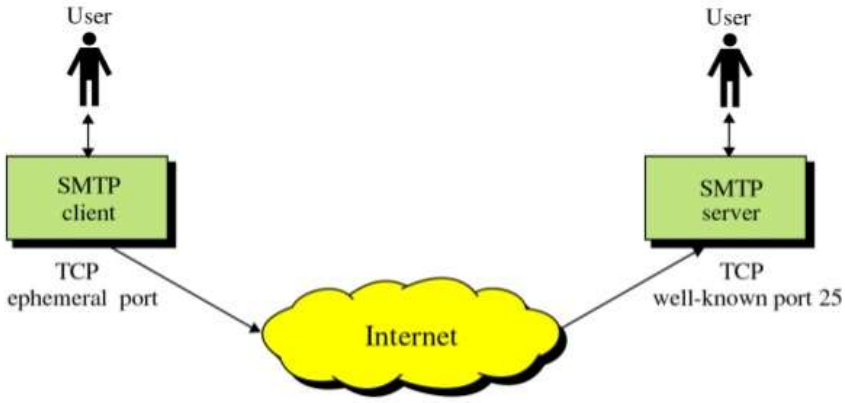
Content Description

The nature of the message be it an imagery or audio wave or visual, this header defines it.

<description>

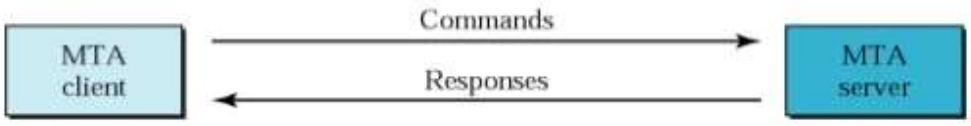
B) Simple Mail Transfer Protocol (SMTP)

The mail is transferred and gets completed through message transfer agents. If a mail has to be sent the system requires an MTA client, and MTA server is required for receiving a mail. The simple mail transfer protocol (SMTP) is considered the formal protocol that describes the MTA client and server on the internet. Between the sender and the sender's mail server and between two mail servers, SMTP is used twice. Simply it describes how to give back and forth requests and responses. At the time of execution, each and every network has the freedom to choose a software.



Commands / Responses

All commands or responses are concluded with delimiter token of two-character end of line (carriage return and line feed). Commands and responses are used by SMTP to transmit messages from MTA client to MTA server.



Commands

Commands are sent to the server from the client. The format and commands contain a keyword which is followed by a zero argument or more. It brings out fourteen instructions. The first 5 are necessary that implication of these five commands must help. Sometimes the following 3 are used and highly recommended. Rarely are the last six included.

Keyword: argument(S)		
HELLO	**NOOP**	**211-reply**
MAIL FROM	**TURN**	**214-help**
RCPT TO	**EXPN**	**220-service ready**
DATA	**HELP**	**221-service closing**
QUIT	**SEND FROM**	**250-request**
RSET	**SMOL FROM**	**251-forwarded**
VRFY	**SMAL FROM**	**354-start i/p**
		421-service unavailable
		450-mailbox unavailable
		451-local error

Responses

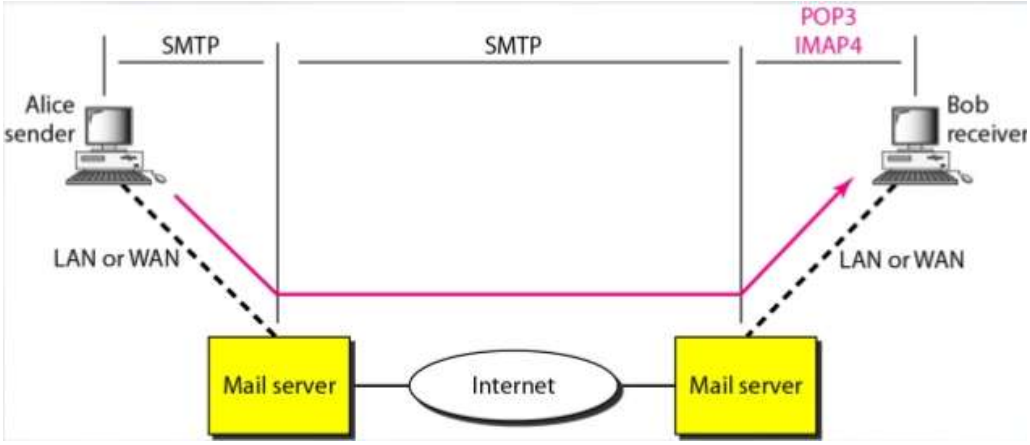
Responses are transmitted to client from server. It is a code of about 3-digit length which can append with additional text.

Mail Transfer Phases

The method of e-mail transmission takes place in 3 phases. They start with connection establishment followed by a mail transfer and ends with connection termination.

C) POP & IMAP: (MAA)

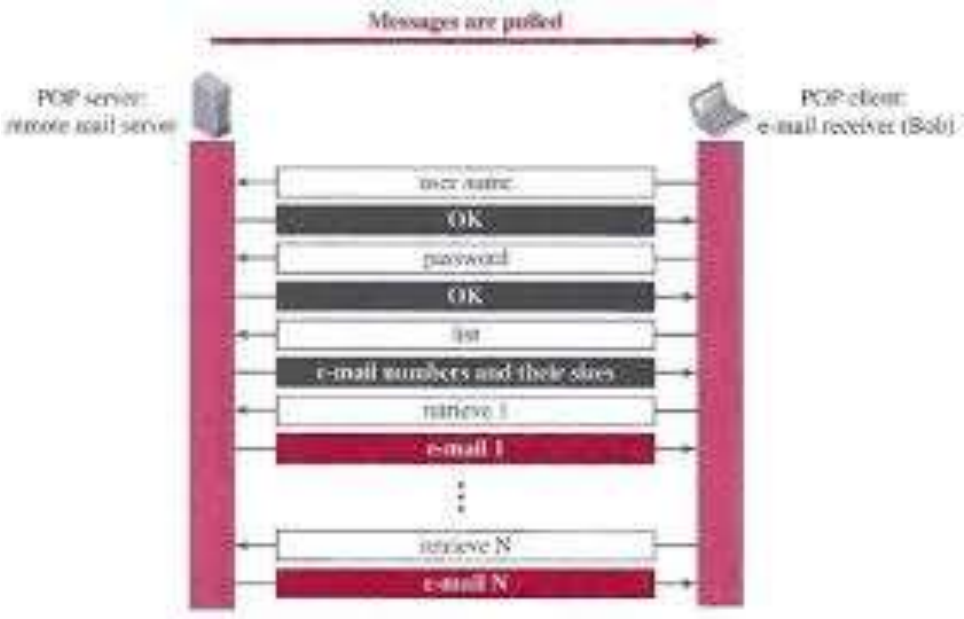
There are 2 messages access protocol. They were Post Office Protocol version: 3 (POP 3) & Internet, Mail Access Protocol version: 4 (IMAP4).



Pop 3

In accessibility, POP 3 is easy & restricted. The POP 3 client software is installed on the receiving machine and the POP3 application software is installed on the mail server. When the user has to retrieve email from the mailbox on the mail server, mail access begins with the client. On TCP port 110 the client opens a connection to the server. If the username and password are then sent to enter the mailbox. The user will then list the mail messages one by one and retrieve them.

There are 2 modes: one mode is delete and the other is keep. After each retrieval, the mailbox's mail will be removed when the mode delete is in progress and widely used while a permanent machine is used by the user. The mail of the mail box stays after retrieval is invoked in keep mode and is preferred while the user is away to access in order to access their initial device. Although the mail gets read, it stays in the organising and retrieval store for future use.



IMAP4

IMAP4 is more active and complex. Prior to uploading, a user should review the email header. Scanning of contents of the email is permissible until it is downloaded by the user. In part, downloading an email by user is possible. If bandwidth stands small and the mail includes multimedia content which demand high bandwidth as its requirements, this is generally advantageous. A user can build mailboxes, remove or rename them. Mailboxes may be created, removed or renamed by a user. In an email storage folder, the mailboxes' hierarchy can be constructed by the user.

5.3. Hypertext Transfer Protocol (HTTP)

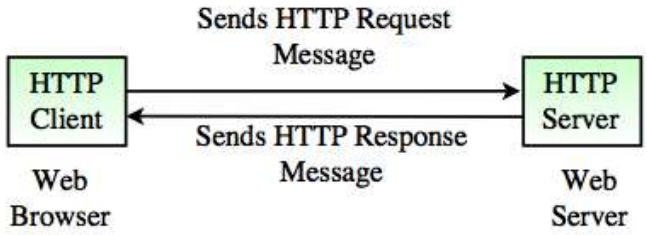
The HTTP is employed mostly for data access on the www. As a mix of FTP and SMTP, HTTP works. This technology reflects the core of FTP, since it shares files along with the utilization of TCP services. HTTP is twinned with SMTP, where the content delivery from client to server is done. The message format is regulated by the MIME-like headers. HTTP utilizes TCP services in a well-known port namely, port 80.

HTTP Transaction

- HTTP transfers between server and client
- HTTP utilizes TCP services; HTTP is considered as a stateless protocol itself.
- The transaction is initialized by the client by transmitting a request message.
- The server responds with a reply message.

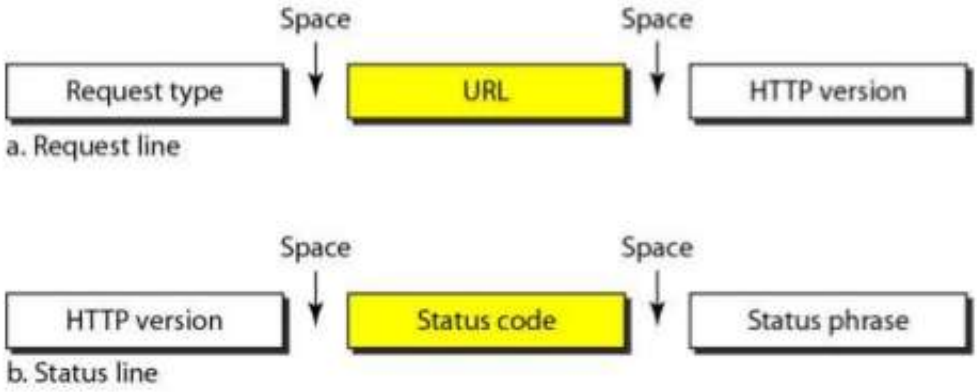
Messages

The request message is a request line, a header with a body whereas a response message is a status line, a header which seems to be a body.



Request and Status Line

A request line is considered the first line of request message. The request line is termed as a status line of response message.



Request Type

The request type is categorized into methods and this field is useful for request message.

GET	requests a document from source
HEAD	requests info about document
POST	sends some information
PUT	sends a document
TRACE	echoes the incoming request
CONNECT	reserved
OPTION	inquiries about available options

Version

The version 1.1 is observed to be the recent one of HTTP.

Status Code

It is a part of the message sequence which is identical to the one in FTP as well as SMTP protocols. It contains a 3-digit code of range. They are,

100 ranges - informational
200 ranges - successful request
300 ranges - connects to another URL
400 ranges - error at client site
500 ranges - error at server site

Status Phrases

Response message utilizes by the field and status code is described in the text format.

Header

The client and server get some additional information that will be shared by header. One or extra lines of header is found at the header. Each header line has a name for the header, a column, and a space and a value for the header. The header line may go in with any of the mentioned 4 groups under General or Response or Request or Entity.

A request header consists of general request and an entity header. A response header consists of general, response and an entity header.

Header Name: Header Value

General Header

General information is available for the message and can be found in requests and response, is provided in the general header.

Connection	- Shows if the connection is to be terminated or not
Date	- Displays the current date
MIME-version	- Shows MIME version
Upgrade	- Notifies which communication protocol is preferred

Request Header

The request header is blocked only when a request message is invoked and clients' configuration and the format of the document is specified.

Accept	Shows the client accepts.
Authorization	Shows permission held by the client
From	Shows email address
Host	Shows host and port number
User-agent	Identifies the client program.

Response Header

The response header can only be blocked by a reply message and also specifies the configuration of the server and the basic details on the request.

Age	Shows age of the document
Server	Shows the server name & version
Public	Shows the supported list

Entity Header

The information regarding the body of text is held in this header, which is represented in the methods of response messages or request messages, namely POST or PUT along with a body.

Body

The body might be found in the request or reply message. It demands a request to either transmit or retrieve.

Proxy Server

HTTP is supported by proxy servers which is a machine holding the response copy that is a recent one. Proxy server will receive a request from the HTTP client and its cache will be reviewed. When the cache contains the response, the response would be sent by the proxy server to the corresponding server. It receives the incoming response and process to align with the future requests of other clients.

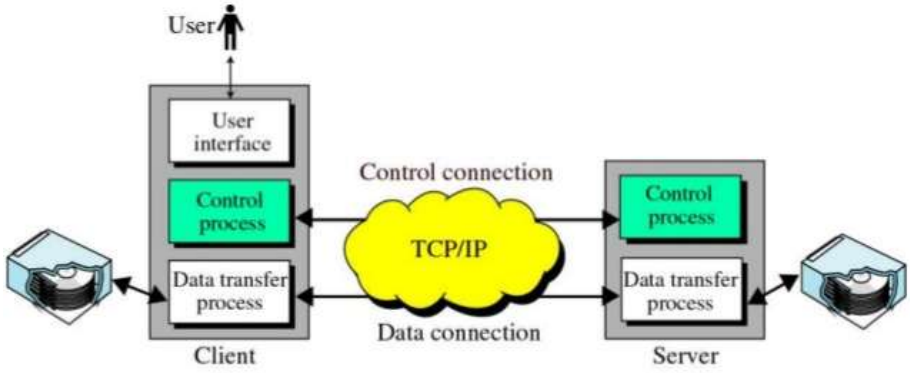
File Transfer

File transfer is an essential and a very important task required during the transfer of files from a device to another, be it in intranet or internet.

5.4. File Transfer Protocol (FTP)

File Transfer Protocol (FTP) is used to copy a file from one host to another host. It is a standard mechanism which is provided by TCP/IP. It separates commands and transferring

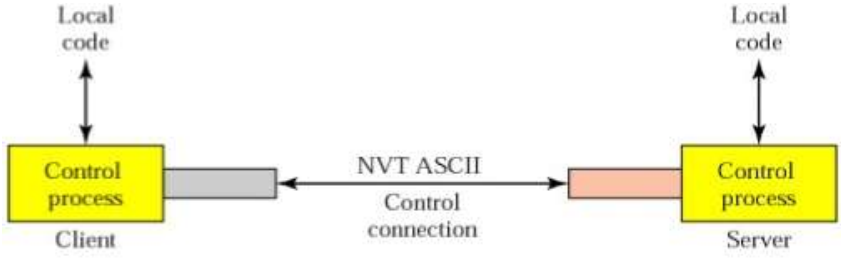
data makes FTP more effective. FTP uses the TCP operation. It require two connectivity to the TCP. The well-known port 21 is used for control connection and the well-known section 20 for network connectivity. There are three components in the client: the user interface, the device control and the data transfer. There are two components in the server: the server control process and the server transfer process. The control is done by the control communication and the data transfer is done by data link.



The connection control is connected throughout the FTP session. For each file transfer the data connection is opened and closed. It opens when the commands that require moving files are to be used and when the file is moved, it will be closed.

Communication Over Control Connection

FTP uses the same technique as SMTP to communicate over a control link. It uses a set of 7 bit ASCII characters. Communication is carried out by each command or response is a short line. The termination for each line is done by two characters that are line feed and carriage return at the end-of-line token.

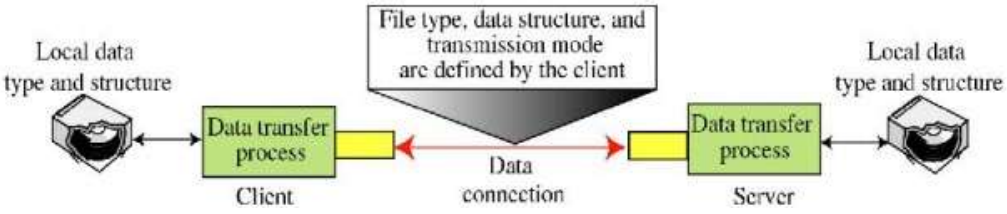


Communication Over Data Communication

It is different from the purpose of the control connection. The transfer of file takes place over the data connection link under the supervision of the commands sent over the control connection.

There are 3 things to transfer in FTP:

1. The file which is copied from the server to the client is said as retrieving a file and it is performed by the command RETR.
2. The file which is copied from the client to the server is said as storage of a file and it is performed by the command STOR.
3. A list file names or directory must be sent from the server to the client by the command LIST.



The client must specify the type of file to be transmitted, the configuration of the files and the mode of transmission. Before sending a file through a data connection, forward it through a control connection. The heterogeneity problem is overcome by specifying three contact attributes:

- File type
- Data structure
- Transmission mode

1. File Type

File type is used to transfer any one of the following over a connection of data; an ASCII file, an EBCDIC file or an image file. The ASCII file is used to transfer text files; it is said as the default format. The EBCDIC file format is used to transfer EBCDIC encoding and it is used by IBM. The image file is used to transfer binary data it is said as the default format.

2. Data Structures

A file which is transferred as data link through one of the structure of data interruptions:

- File structure
- Record structure
- Page structure

A continuous stream of bytes was used in this file structure format. It is divided into records (file: text) in the record structure. In the page structure, the files are divided into pages with a page number and a page header for each page.

3. Transmission Mode

A file which is transferred as data link through FTP by anyone of the 3 following modes of transmission:

- Stream mode
- Block mode
- Compressed mode

The default mode of data is provided from File Transfer Protocol to Transmission Control Protocol as a continuous stream of bytes in stream mode. Data can be delivered from FTP to TCP in block mode. In compressed mode, if the file is large, the data can be compressed.

Anonymous FTP

A user should have an account that is user name and password on a remote server to use FTP. Some pages will provide a collection of information that may be available for public access so it is said to be as anonymous FTP access. You don't need an account or password to access these files. An anonymous user can be used as a username and a guest as a password.

5.5. World Wide Web (WWW)

The World Wide Web (WWW) is a repository of knowledge linked from all over the world. The WWW has a special combination of versatility, portability and user-friendly features that differentiate it from other internet services. The www project was initiated by CERN.

5.5.1. Architecture

World Wide Web is a browser based client which can access through a server so it is said to be a distributed client/server. The server supports through a variety of locations called sites.

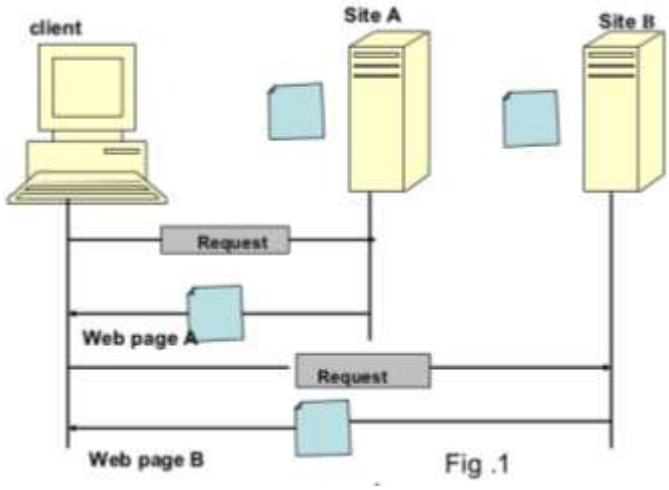


Fig .1

One or more documents in each site is said to be as web pages. Each page links with another page on the same or on other pages. Each site contains one or more documents referred to as web pages. Each page can contain a link to another page on the same site or on other pages. Pages can be retrieved and accessed using a browser.

5.5.2. Client (Browser)

A number of vendors provide commercial browsers that interpret and display a web document and all use almost the same architecture. Each browser is normally made up of three sections.

- Controller
- Client protocol
- Interpreters

The controller will receive input from the keyboard or mouse and will use the client programmes to access the text. The controller will use one of the interpreters to view the text on the screen. The client protocol can be one of the mentioned protocols (FTP / HTTP). The interpreter can be an HTML, Java or Java script, depending on the type of document.

5.5.3. Server

The website is stored in the server. Every time a client request arrives, the corresponding document is forwarded to the client. To improve efficiency, servers usually store the requested files in the memory cache; the memory is faster to access than the disk. The server can also be made more effective by multi-threading or multi-processing. The server can respond to more than one request at a time.

5.5.4. Uniform Resource Locator (URL)

A client who wants to access a website requires an address. HTTP uses locators to allow access to documents scattered around the world. The URL is a standard for specifying any type of information on the Internet. The URL describes four things:-

- Protocol
- Host computer
- Port
- Path

Protocol:// host : port / path

The protocol is a client / server program that is used to retrieve a document. A document can be obtained by several different protocols, including HTTP/FTP. The information is stored

on the host; the name of the machine can differ. The web pages are mainly stored in the computers and the alias names were given to computers that usually start with "www" characters. The URL has the port number of the server. When it is in usage it can be inserted between the path and host. A colon is used to separate from the host. The path name of the file is used to store the information.

Cookies

The www was actually designed to be a stateless body. The client is sending a request; the server is responding. It retrieves the accessible documents publically which suits this function.

1. The registered clients will have access for particular websites.
2. Websites can also use as electronic shop.
3. It allow users to search and select the content what they want, add them in cart and pay at the end.
4. Users can choose the website as portals when they want to visit the website.
5. Some of the blogs are all ads.

Creation and Storage of Cookies

The production and storage of cookies depends on the implementation, but the concept is the same.

1. When a server receives a request from a client, it stores the client information in a file or in a strong file. The details may include the client's domain name, cookie content, time stamp and on the development.
2. Client response side also has a cookie which is sent by the server.
3. The browser stores cookies in the cookie directory once client receives a response and that is aligned in the DNS.

How to Use Cookies

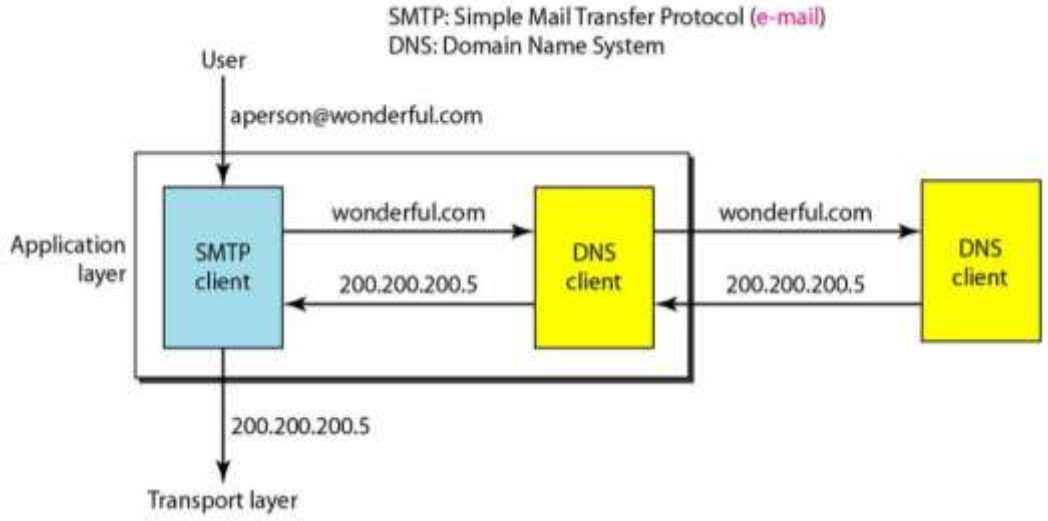
When a server receives a request from the client, the web page will search the directory of cookies to check whether server received cookies. It includes the request when the cookie is found. The server should understand whether it is an old client or new client when it receives a request. It's a cookie made by the server and eaten by the server.

1. The registered clients have a limits access to send a cookie to the client when it is registered first.
2. Client shoppers use an electronic store to use a cookie.

3. The web portal uses cookies to submit the server to reveal what the client is searching for.
4. Advertising companies often use a cookie.

5.6. Domain Name Systems (DNS)

Domain Name Systems (DNS) is support software used for other applications such as e-mail. The recipient's email address may be known to the user of the email program. The IP address was required in the IP protocol. The DNS server receives the request to the DNS client which helps to map the e-mail address with its corresponding IP address.



To define an entity TCP / IP protocol, use an IP address that uniquely identifies the connection of a host to the Internet. The host that needs mapping can contact the lost computer holding the information needed for this method to be used by the DNS.

Namespace

A unique name maps the each address in namespace and it can be arranged in two ways they are,

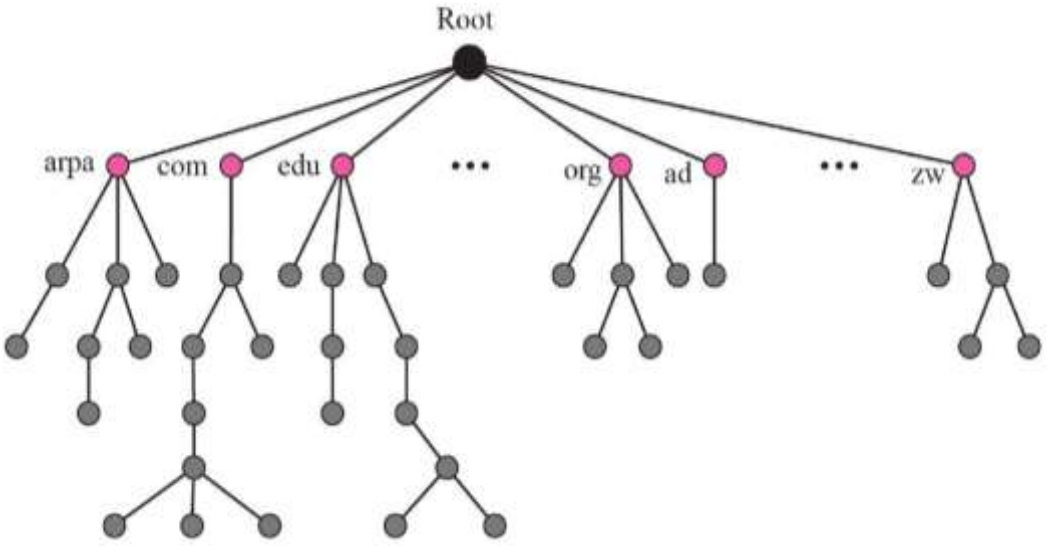
- Flat namespace
- Hierarchical namespace

5.6.1. Flat Name Space

In name space, the name is allocated to an address. Sequence of characters without any structure is said to be a name. they does not have any meaning to have a common section.

5.6.2. Hierarchical Name Space

Each name is made up of many parts in the hierarchical name space. The first part can define the meaning of the organization, the second part can define the name of the organization, and the third part can define the sections of the organization and so on. A central authority may assign a part of the name that defines the goal of the system and the organization name.



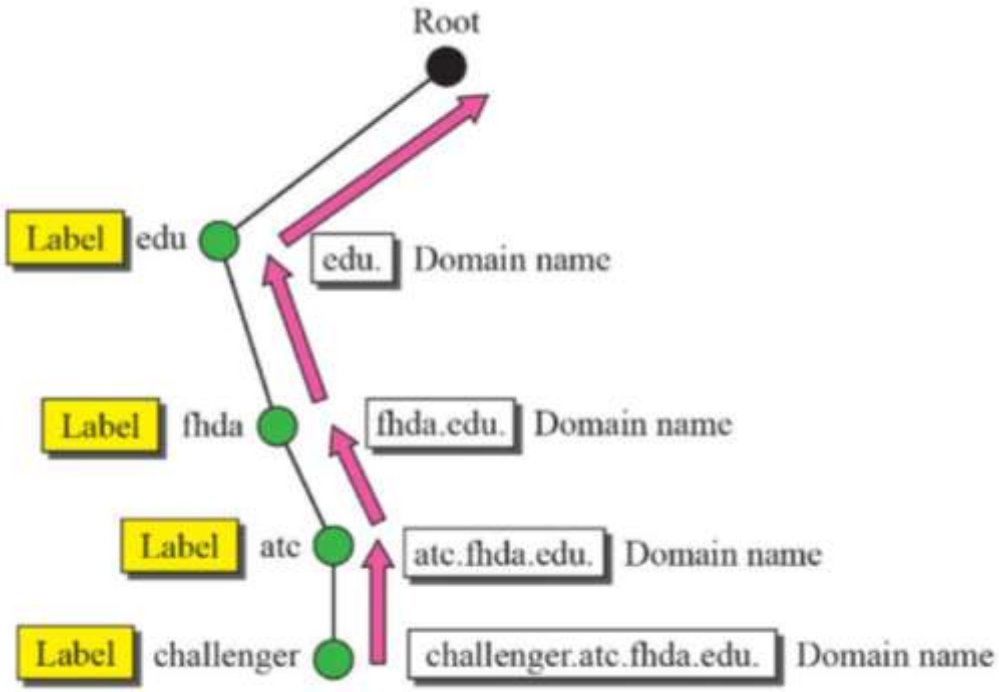
A domain name space has been designed to have a hierarchical name space. From the top level of the inverted tree it is said to be as names. The tree can only have 128 levels and it starts from level 0 (root) to level 127 (nodes).

Label

A node in the tree has a name, a string with a maximum of 63 characters. The root label is a null string or an empty string. DNS requires that node children (nodes that branch from the same node) have separate labels that ensure the uniqueness of domain names.

Domain Name

Each node in the tree has a domain name. The full domain name is separated by dot (.) for list of labels. Domain names are read from the node to the root. The last label is the root mark said to be as null, which means that the full domain name always ends in the null mark. The last character is a dot which means the null string is nothing.



Fully Qualified Domain Name (FQDN)

If a label is terminated by a null string, it will be considered a fully qualified domain name.

Eg:- challenger.atc.fhda.edu

It includes the entire host name which contains all the labels. It specifies the most general and uniquely defines the host name. Only a FQDN can match a DNS server to an address. The name must end with a null mark and since null means nothing the label will end with a dot (.)

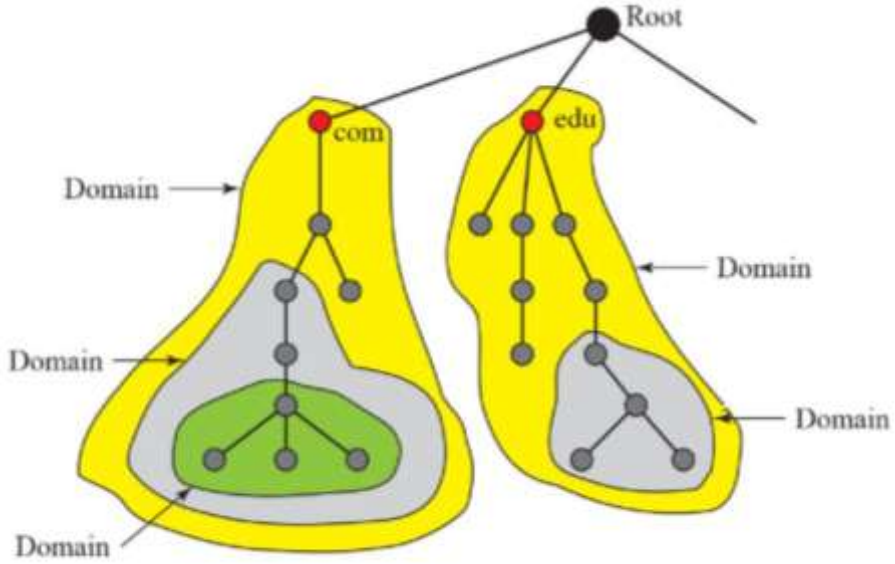
Partially Qualified Domain Name (PQDN)

Partially Qualified Domain Name (PQDN) is said when a label does not transmit a null string. It starts from the node, and doesn't reach the root. When the name is fixed on same site as client PQDN is used. The missing element is said to be as suffix to generate FQDN.

Eg:-challenger User->fhda.edu

Domain

In DNS, the domain is a sub-tree. The domain name is the name given for the node at the top level. It divides domain into subdomain by itself.

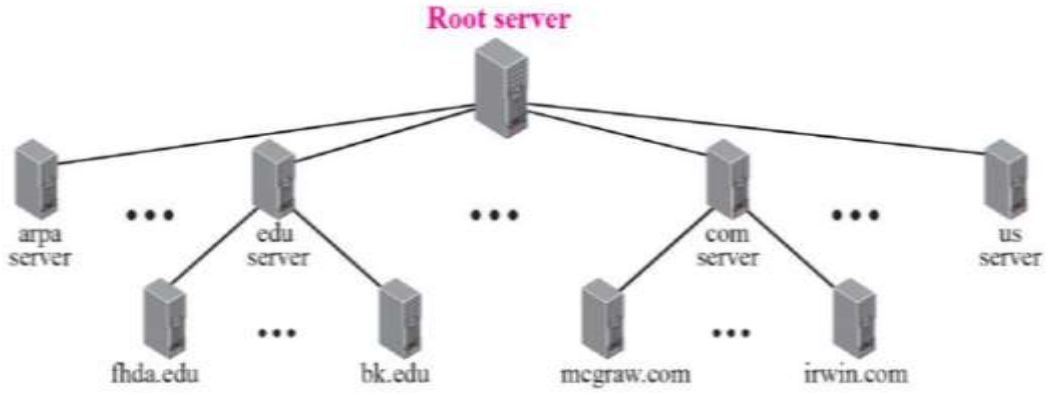


5.6.3. Namespace Distribution

The solution to these problems is to transmit information to a number of computers called the DNS service. It divides the entire space into a number of domains centered on the first level.

Name Server Hierarchy

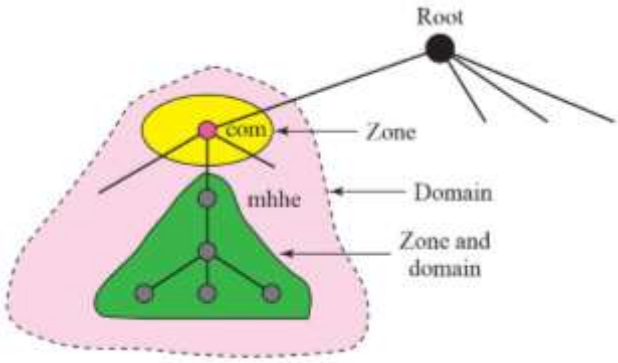
The information which found must be processed in the DNS. This is not accurate and quite expensive to store such a large amount of knowledge in one computer.



In DNS, domains are divided into subdomains. Each server is responsible for each domain either it is large or small.

Zone

The entire hierarchy cannot able be store on a single server, so it has been splited between multiple servers. What a server is responsible for or has authority over is referred to as a zone. If the server takes responsibility for the domain and does not break the domain into smaller domains, the domain and the zone will refer to the same thing.



Root Server

The root server is holds the entire tree. It does not store any information from domain, but it assigns permission to another server by referring other servers. The root servers cover the entire domain name space. The server is spread all over the world.

Primary and Secondary Server

DNS server was divided in to 2 types: one is primary and another is secondary.

A server stores a file for authority is said to be as the primary server. It is mainly used for developing and managing the zone. It is saved in the local drive.

A server which is used to transfers the complete information about the zone to another server which can either be a primary or secondary and it stores the data on the local drive. This type of server does not build or change zone files. The primary server loads all information from the secondary server disk file and loads all information from the primary server. When secondary information is downloaded from the original, it is called zone transfer.

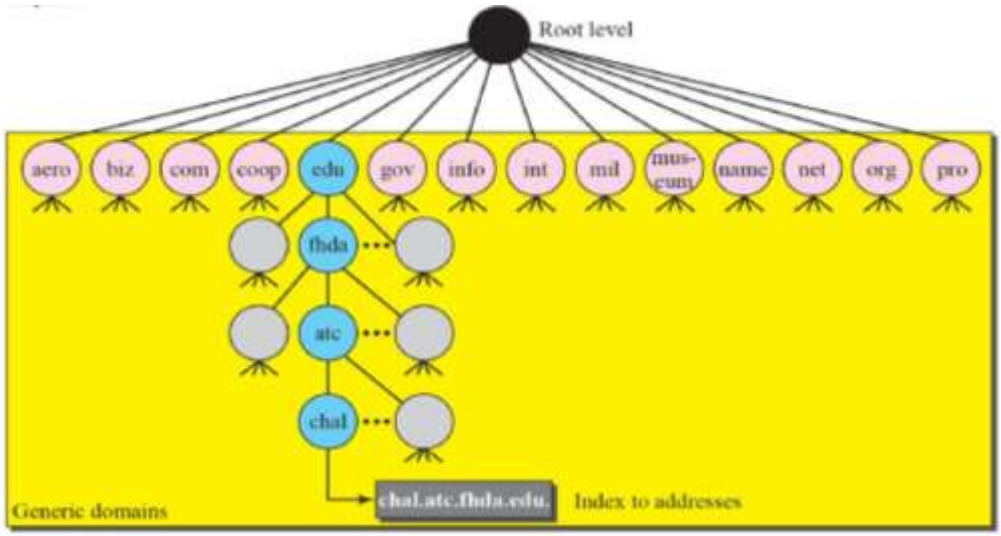
5.6.4. DNS in the Internet

DNS protocol is used in various platforms. It is divided as 3 sections in the internet:

- Generic domain
- Country domain
- Inverse domain

1. Generic Domains

Generic domains identify registered hosts by their generic behavior. Each node in the tree defines a domain that is an index to the domain name space database.



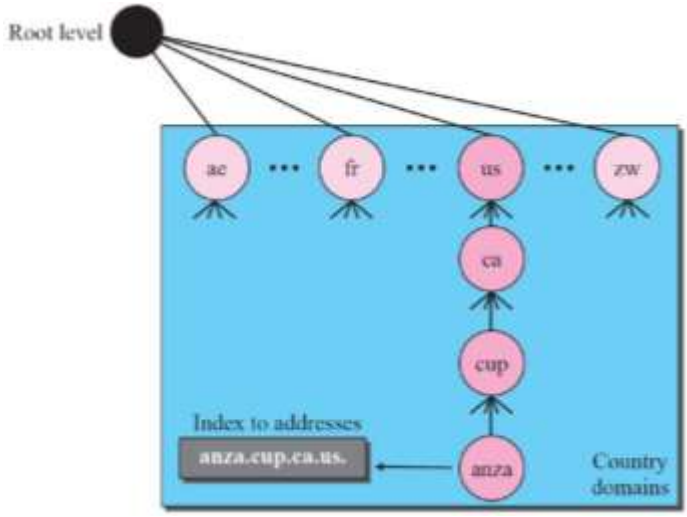
Label	*Description*
aero	Airlines and aerospace companies
biz	Businesses or firms (similar to "com")
com	Commercial organizations
coop	Cooperative business organizations
edu	Educational institutions
gov	Government institutions
info	Information service providers
int	International organizations
mil	Military groups
museum	Museums and other non-profit organizations
name	Personal names (individuals)
net	Network support centers
org	Nonprofit organizations
pro	Professional individual organizations

2. Country Domains

In this section, two characters were used as abbreviation to display a country.

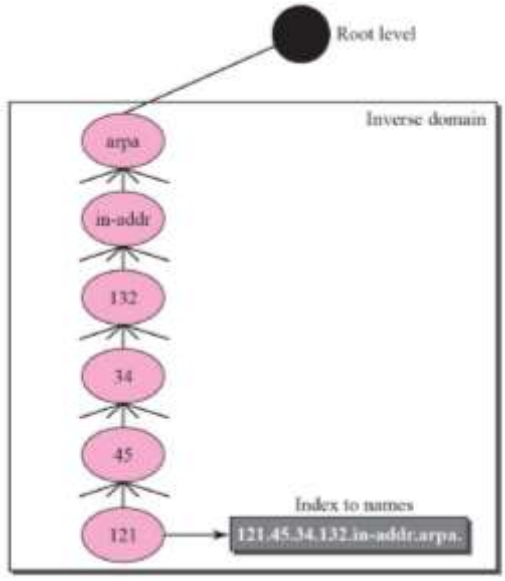
Eg:- US for United States

For organization or national destination, second labels were used.



3. Inverse Domains

Inverse domain is used to map a name address. (e.g) when the server has received a request from the client to perform an operation. The server has a file containing a list of authorised clients with only the IP address of the client. The server asks to send a request to the DNS server to map the address to the name to decide whether the client is on the authorised list. This type of query is called the inverse or pointer (PTR) query. To manage a pointer query, the inverse domain is applied to the domain namespace with the first level node called arpa and the second level is just a single node named in-addr. The rest of the domain will determine the IP address.



Mapping

Name address resolution is mapping a name to an address or a name address.

Resolver

A client/server program is designed by DNS. A host which maps an address to a name or a name for an address can call a DNS client called a resolver.

Mapping Name to Addresses

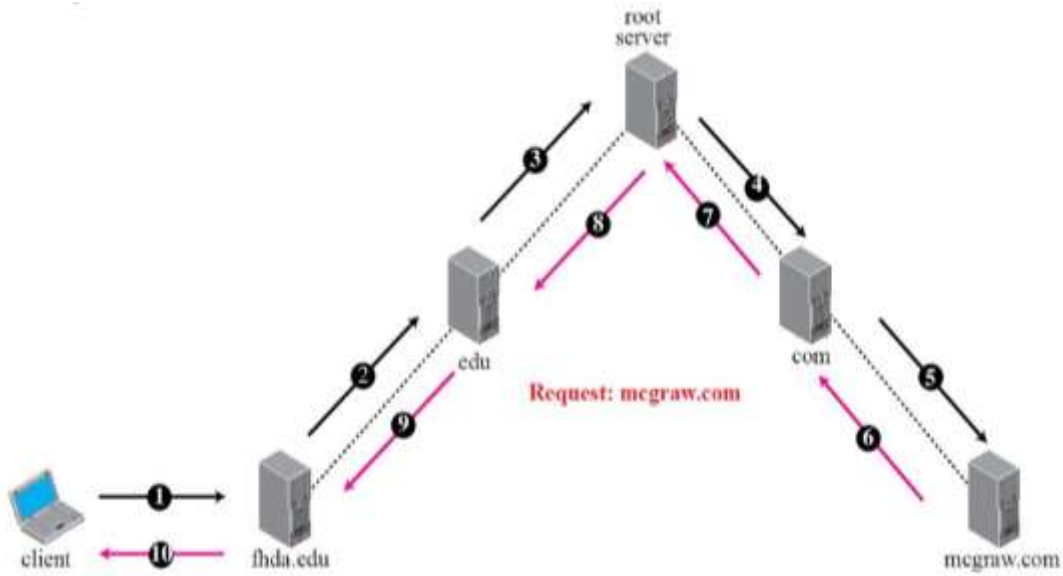
Resolve gives the server a domain name and asks for the corresponding address. For mapping, the server checks the generic domain or the country domain.

Mapping Addresses to Names

An IP address is sent by the client which is mapped to domain name to the server.

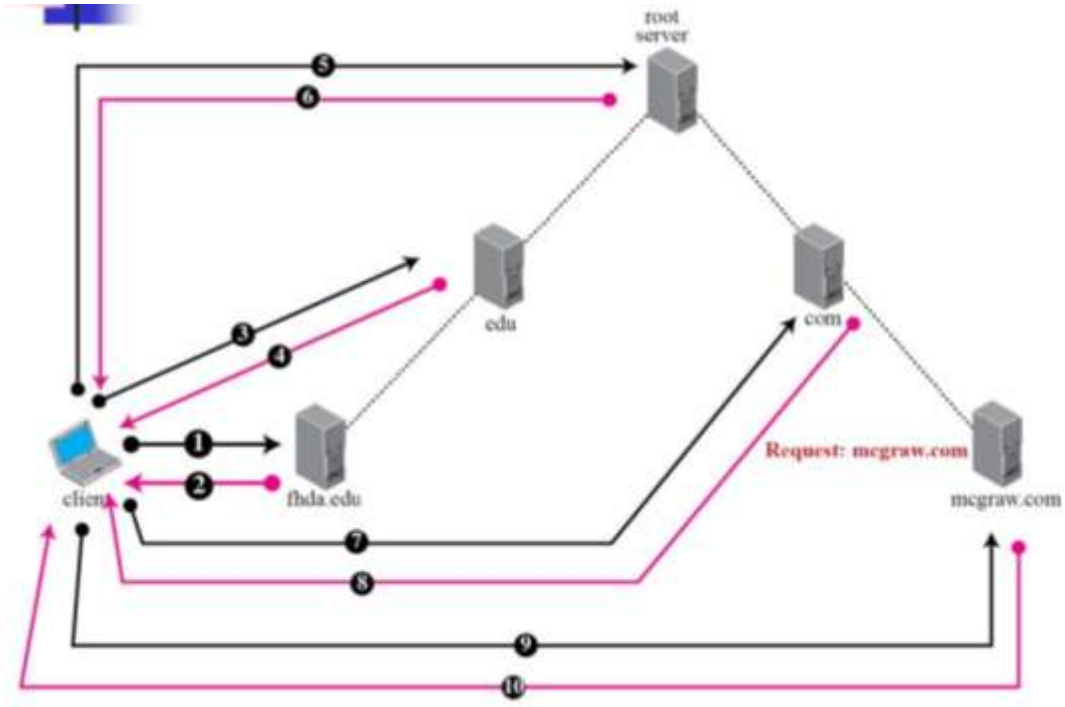
Recursive Resolution

A recursive response will be requested by the client from the name server. Server waits for the final response to be applied. If the server is the domain name authority, it checks the database and responds. If the server is not an authority, it sends the request to another server (usually the parent) and waits for the response If the parent is the authority that responds otherwise, the request would be sent to another server. When the request is eventually resolved, the answer will move back before it finally reaches the requesting client. This is called a recursive resolution.



Iterative Recursive

If the client does not ask for a recursive answer, mapping can be done iteratively. If the server is the name authority, it sends the response or returns the IP addresses of the server that it thinks can resolve the query. It is the duty of the client to repeat the query to this second server. If the newly addressed server is able to resolve the query, it will answer the query with the IP address to return the new server's IP address to the client. The client must now repeat the query to the third server. This method is called a recursive iterative.



Caching

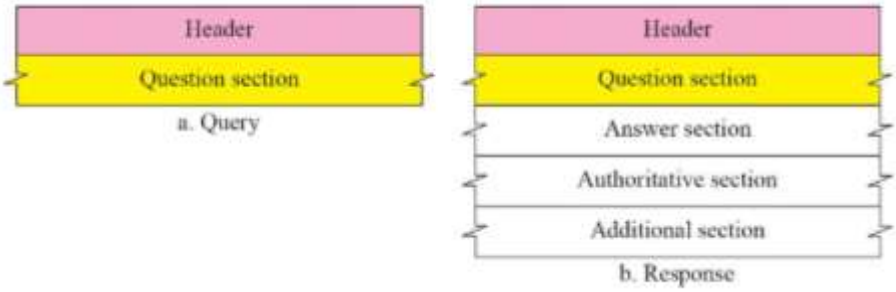
When a query is received at server side which is not in domain, a server IP address needs to be searched for in its database. Reducing this search time will improve productivity. This process is called caching.

5.6.5. Messages in DNS

There are two types of DNS message: request and response. In request message, the header and query records are placed and in response message, the header, query records, reply records, authoritative records and additional records are found.

Header

In same header, both response and request messages filled as zero for query message in certain fields.



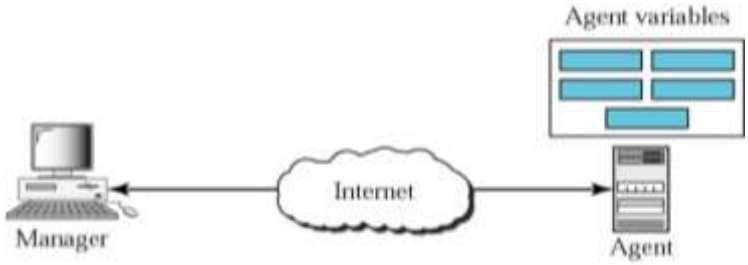
The client uses identification to fit the response to the query. Every time question is sent, the client uses a different identification number. The flags determine the form of the message the form of desired resolution. In sub field, the count of request records includes the count of response records of the message. The count of response records in the sub-field includes the count of response records in the reply portion of the reply letter. The count of authoritative parts includes the count of authoritative records in the authority's part of the responses. The count of additional ones in this includes the count of additional records in the additional portion of the responses.

5.7. Simple Network Management Protocol (SNMP)

A mechanism which is used for controlling devices using Transmission Control Protocol/Internet Protocol on the Internet is said to be as Simple Network Management Protocol (SNMP). Mainly collection of basic operations are monitored and maintained in the Internet.

5.7.1. Concept

The concept of manager and agent is used by SNMP. The host manager normally manages and tracks a group of agents, normally the header.



An application level protocol which has a few management stations to manage the number of users/agents. It is structured at the stage of the application; it can control the system which is created by various mechanisms. It is mounted on different physical networks. It releases control functions from both the physical features of managed devices and the underlying networking technologies. Heterogeneous connection is done at different LANs and WANs network which connected by routers in SNMP.

Managers and Agents

The manager acts as a host who running the SNMP client program. An agent called management station where a router running the SNMP server program. Management is accomplished by clear contact between the agent and manager. The output details are stored in the database by the agent. The value in the database and router navigation is done by manager. Agents can also contribute to management process.

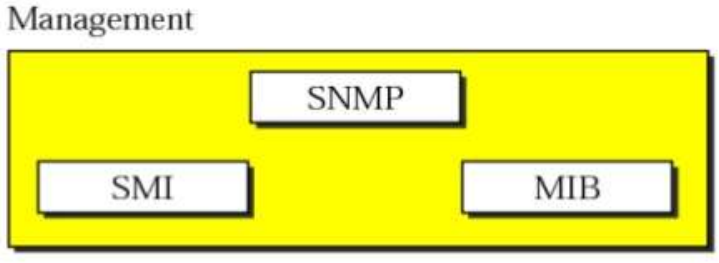
The process status will be checked by agent who running the server program and, if anything went wrong, an alert message will be send called a trap to the manager. SNMP is based on three simple concepts.

1. The manager checks the agent by requesting information that reflects the behavior of the agent.
2. The manager requires the agent to perform the assignment by resetting the value in the agent database.
3. The agent contributes directly to the management process by warning the manager of an unusual situation.

5.7.2. Management Components

SNMP uses 2 protocols to perform management tasks. They are,

- MIB - Management Information Base
- SMI - Structure of Management Information



Role of SNMP

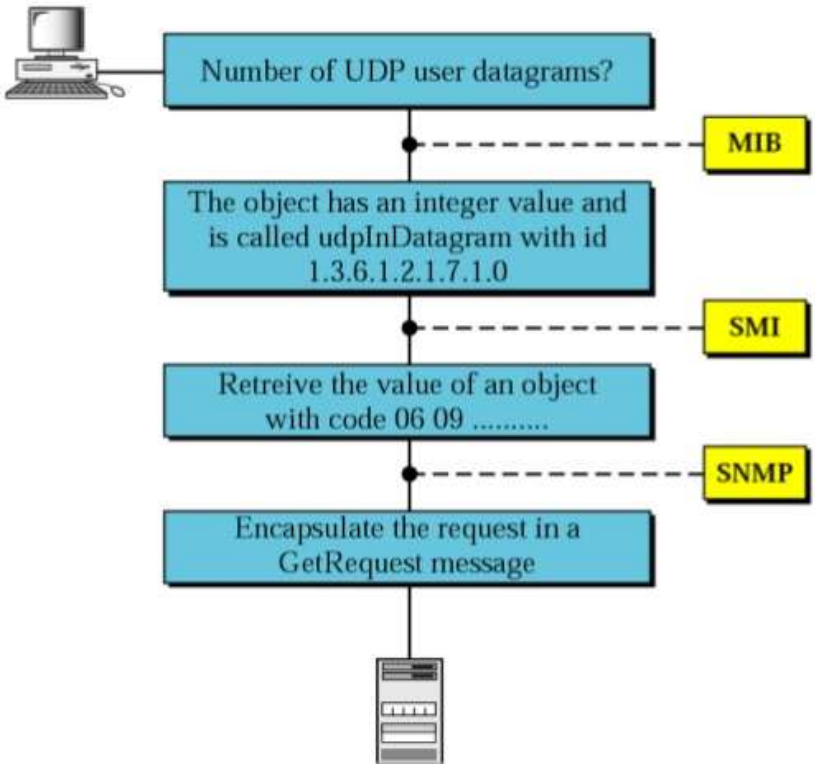
- Reading and altering is carried out by SNMP.
- Exchanged packets have object name and its status. It interprets the statistics which produced and outcomes.
- It has many important functions in the network management.
- The packet format which sent from the manager to the agent is determined, and also vice versa.

Role of SMI

- SMI specifies the common rules to name types of entity range and its length and demonstrates the values and objects production.
- It does not specify objects count where the entity can control or managed by the objects name and establishes the relation between values and object.

Role of MIB

- A list of named objects, forms and its relationship between the entity are generated and controlled by MIB.



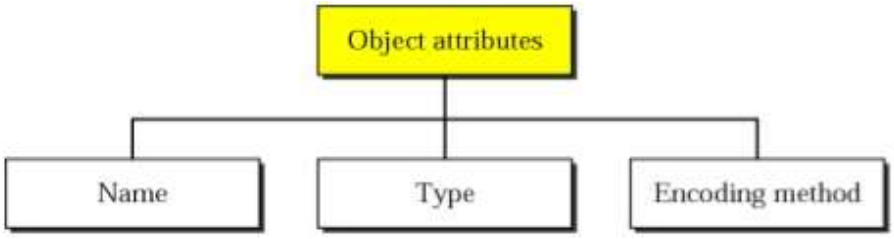
5.7.3. Structure of Management Information (SMI)

The structure of management information version 2 (SMIV2) is a network management. The properties include:

1. Object name.
2. Data type should be defined to store in the object.
3. Data transmission over the network.

SNMP guides the structure of management information and it has 3 attributes to treat it as an object,

- Object name
- Data type
- encoding method



Name

A hierarchical identifier is used to identify the objects in SMI like a tree structure to label objects globally. It always starts with 1.3.6.1.2.1.

Type

Abstract Syntax Notation 1 (ASN.1) is used to define the data type in SMI. It acts as a subset and also a superset, and two large categories of data types were used.

- Simple
- Structured.

Simple

Efficient data types all types of atomic data. The remaining types are taken directly from ASN.1 and added by SMI.

5: ASN.1 & 7: SMI

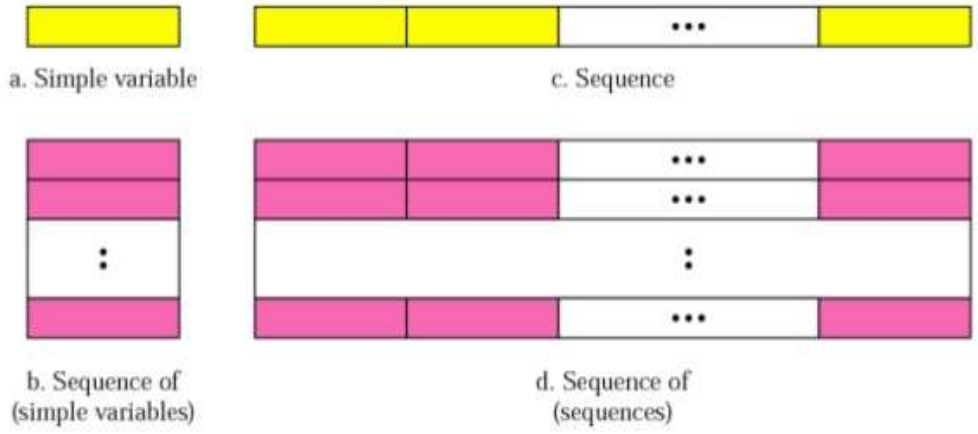
Integer	4 bytes
Integer 32	4 bytes

Unsigned 32	4 bytes
Octet string	variable
Object identifier	variable
IP address	4 bytes
Counter 32	4 bytes
Counter 64	8 bytes
Gauge32	4 bytes
Time ticks	4 bytes
BITS	bits
Opaque	variable

Structured Type

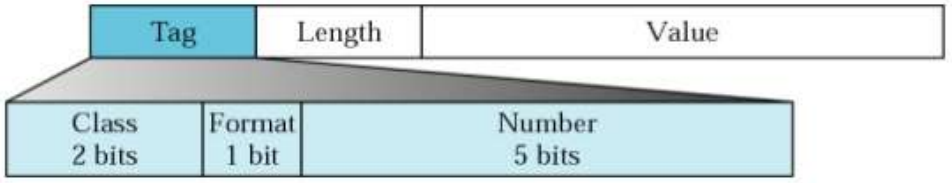
There are two types in SMI structured data. They are:

1. **Sequence:** It is a collection of basic data forms.
2. **Sequence of**: It is a combination of a single/sequence data type of the same type.



Encoding Format

Basic Encoding Rules (BER) is used to encode data which is going to be transmitted over the network. It indicates all the data to be encoded in a triplet format as follows:



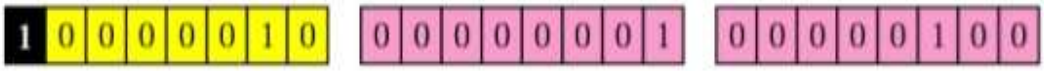
Tag

This field is used to specify the data type. Its field is 1 byte and consists of three sub-fields. They are 2 bits for Class, 1 bit for format and 5 bits for number. These subfield are used to indicate whether the data is simple (n) or structured (i).

Length

In this field, the length may be 1 or more bytes. The MSB will be 0 when it is 1 byte and the data length is defined by other remaining 7 bits.

a. The colored part defines the length (2)



b. The yellow part defines the length of the length (2 bytes); the pink bytes define the length (260 bytes)

Value

In this field, the data values are coded by BER rules.

5.7.4. Management Information Base (MIB)

In network management MIB 2 that is Version 2 of the Management Information Base is the second aspect. The manager is able to handle each agent with its own MIB objects. These objects are classified by 10 groups. They were interface, ip, udp, system, tcp, address translation, icmp, egp, snmp tables, variables and transmission are specified in each category.

sys	system
if	interface
at	address translation
ip	IP address
icmp	ICMP
tcp	TCP
udp	UDP
snmp	SNMP

MIB Variable Access

Simple Variables

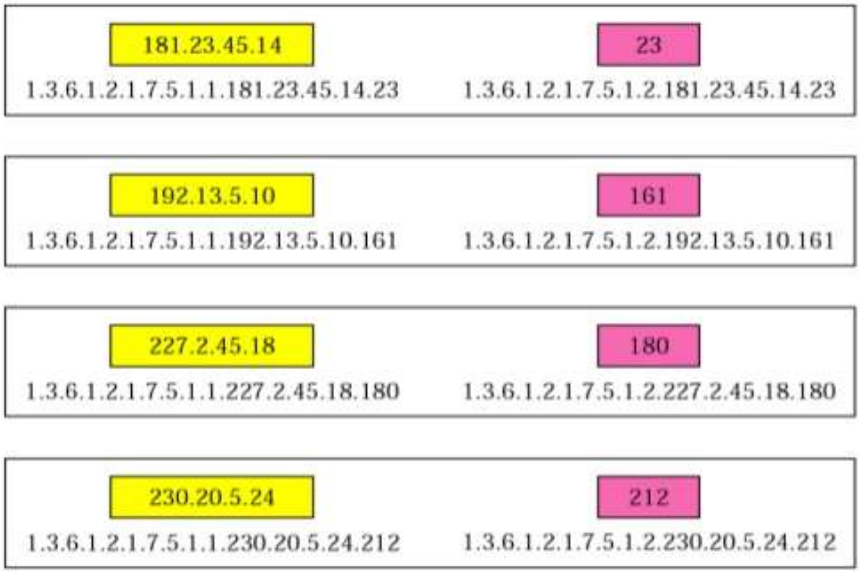
Simple variables are accessed by using the group ID (1.3.6.1.2.1.7) followed by the variable ID.

- Udp in datagrams ->1.3.6.1.2.1.7.1
- Udp no ports->1.3.6.1.2.1.7.2
- Udp in errors->1.3.6.1.2.1.7.3
- Udp out datagrams->1.3.6.1.2.1.7.4

Tables

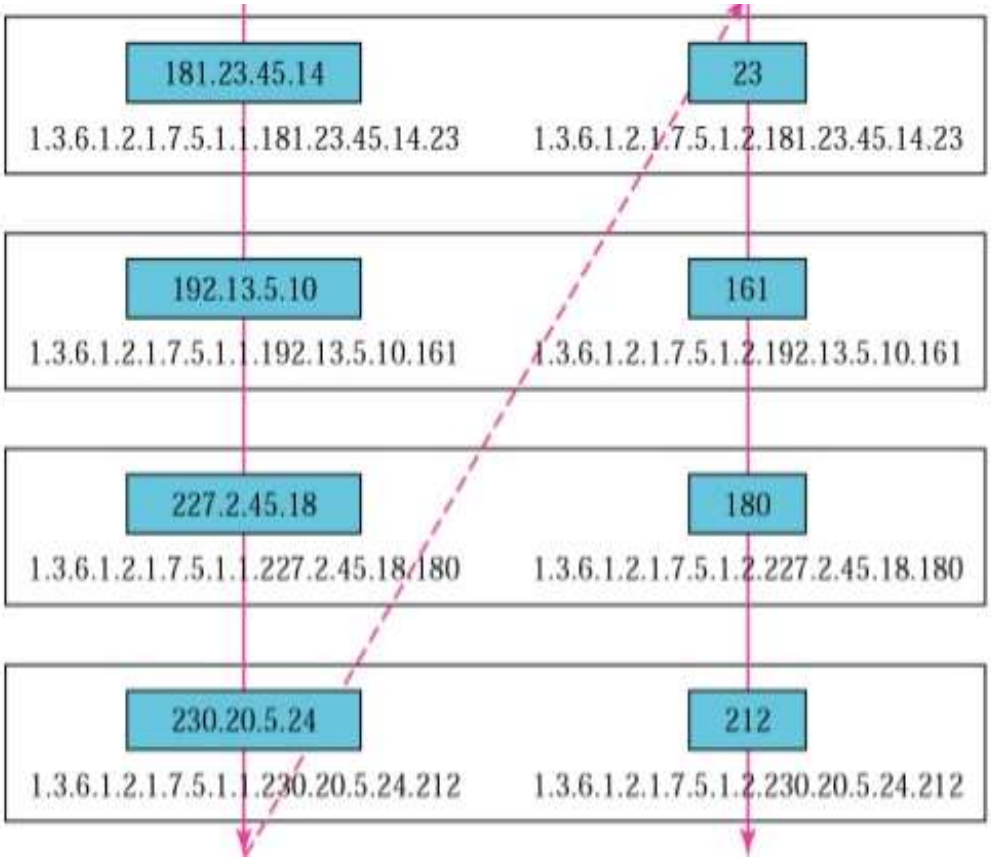
The udp creates tables with table id.

- Udp table->1.3.6.1.2.1.7.2
- Udp entry->1.3.6.1.2.1.7.5.1
- Udp local address->1.3.6.1.2.1.7.5.1.7.
- Udp local port->1.3.6.1.2.1.7.5.1.2



Lexicographic Ordering

An object identifier over the variables of the MIB is ordered by lexicography. It will be sorted by column row table manner, which means top to bottom of the column. One should go from top to bottom in each column. The manager should have access to set a variable after the first variable over the next by lexicographic ordering.



SNMP

For Internet network control in SNMP, it uses both MBI and SMI. SNMP is application software where,

1. An object is identified by the agent to manager to retrieve the value.
2. A value is stored in agent specified object by manager.
3. A warning on irregular process is given by agent to the manager.

PDUs

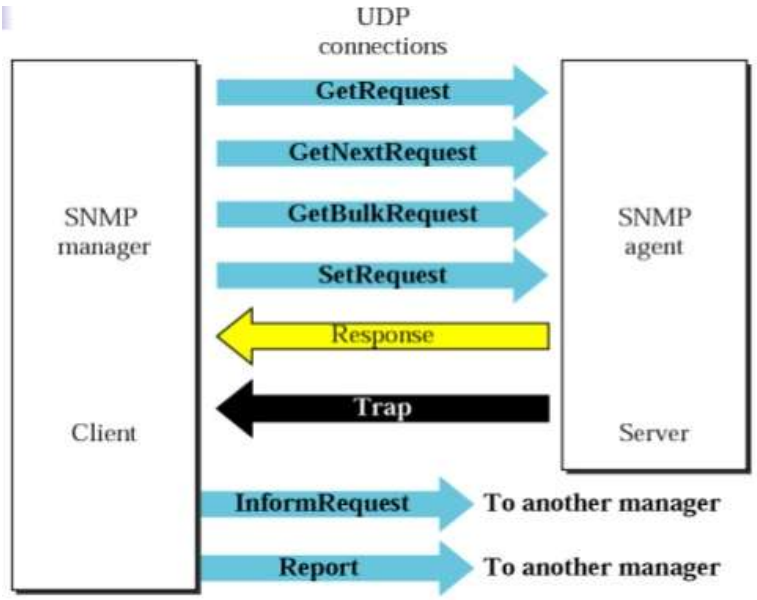
There are 8 packet types in SNMP Version 3. They are,

-> get request -> response

-> get bulk request -> trap

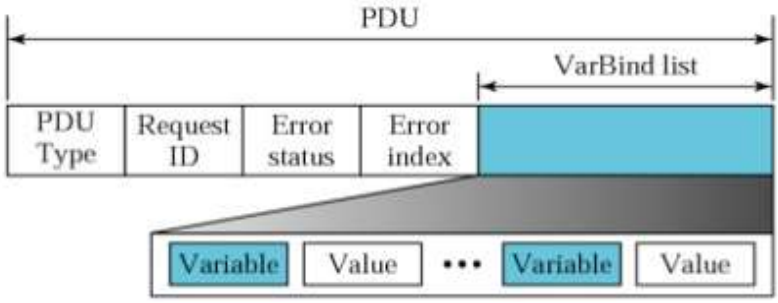
-> get next request -> information request

-> set request -> report



Format

This format is for the 8 SNMP PDU**.** The get bulk request PDU differs from the others in 2 areas.

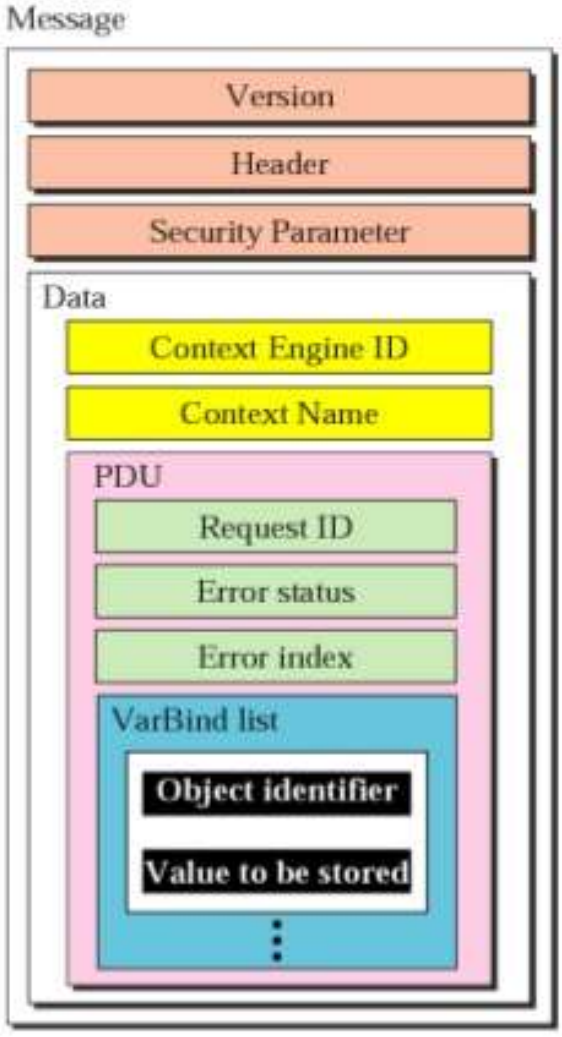


The status and error index values are zero for all request messages except to get a bulk response. Error status field is replaced by a non-repeater field and the error index field is replaced by a max-repeater field in get bulk request. The fields are listed.

- **PDU type -** Says the type of PDU.
- **Request ID -** Denotes sequence number and agent response over the PDU request.
- **Error status** - Reports the error type by the agent.
- **Non repeaters -** Replaces the error by using GetBulkRequest.
- **Error index -** Indicates the index of the error to the manager.
- **Max-repetition -** Replaces the error which is empty by using GetBulkRequest.
- **Var bind list -** Variables with values can be set or retrieved when manager wants.

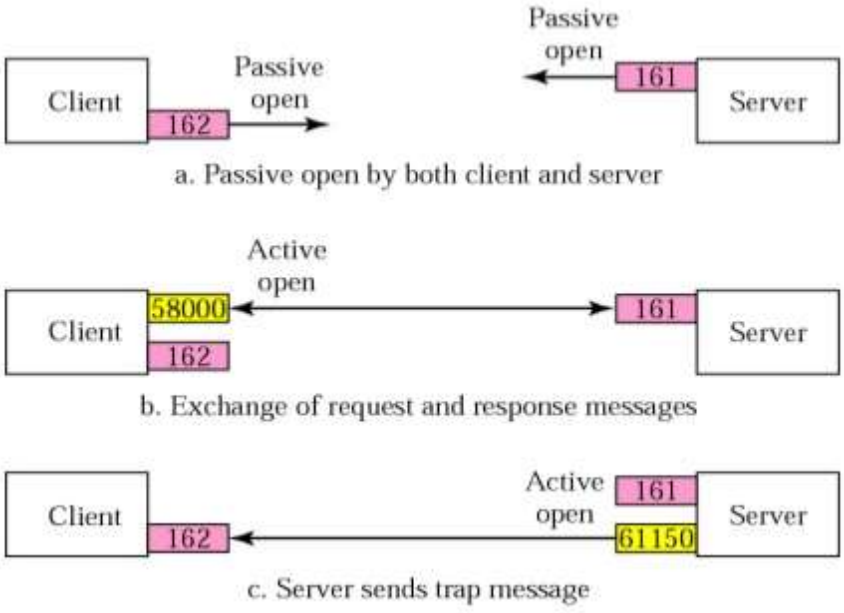
Messages

The SNMP encloses the PDU in a message. The SNMP Version 3 message consists of version which describes the new version as 3, the header includes the message identification values of the message size and how message protects, and the security message parameter is used to build a message digest. The data contains the PDU. SNMP uses a tag to describe the form of PDU. The class is sensitive to context (10).



UDP Ports

161 - Server (as agent) and 162 - Client (as manager) are the two ports used by SNMP. The port 761 is a server's passive open and port 162 is a client's active open.



5.7.5. SNMP Securities

The two securities are given by SNMP version 3. They are specific and general. SNMP Version 3 offers authentication of messages, anonymity and management authorization. It allows the manager to change the security configuration remotely. It also ensures that the manager does not have to be physically present at the manager station.