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SCHOOL OF COMPUTING

DEPARTMENT OF INFORMATION TECHNOLOGY

UNIT – I – Data communication and Computer networks – SCS1314

DATA COMMUNICATION

Introduction to data communication - Network protocols & standards - Line configuration - Topology - Transmission mode - Categories of networks - OSI model - Layers of OSI model - TCP/IP Model - Transmission media - Guided media - Unguided media

INTRODUCTION

A network is a set of devices (often referred to as *nodes)* connected by communication links. A node can be a computer, printer, or any other device capable of sending and/or receiving data generated by other nodes on the network.

"Computer network'' to mean a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange information.

The connection need not be via a copper wire; fiber optics, microwaves, infrared, and communication satellites can also be used.

Networks come in many sizes, shapes and forms, as we will see later. They are usually connected together to make larger networks, with the **Internet** being the most well-known example of a network of networks.

There is considerable confusion in the literature between a **computer network** and a **distributed system**. The key distinction is that in a distributed system, a collection of independent computers appears to its users as a single coherent system. Usually, it has a single model or paradigm that it presents to the users. Often a layer of software on top of the operating system, called **middleware**, is responsible for implementing this model. A well-known example of a distributed system is the **World Wide Web**. It runs on top of the Internet and presents a model in which everything looks like a document (Web page).

USES OF COMPUTER NETWORKS

1. Business Applications

- to distribute information throughout the company (**resource sharing).** sharing physical resources such as printers, and tape backup systems, is sharing information
- **client-server model**. It is widely used and forms the basis of muchnetwork usage.
- **communication medium** among employees.**email** (**electronic mail**), which employees generally use for a great deal of daily communication.
- Telephone calls between employees may be carried by the computer network instead of by the phone company. This technology is called **IP telephony** or **Voice over IP** (**VoIP**) when Internet technology is used.
- **Desktop sharing** lets remote workers see and interact with a graphical computer

screen

- doing business electronically, especially with customers and suppliers. This new model is called **e-commerce** (**electronic commerce**) and it has grown rapidly in recent years.

2 Home Applications

- **peer-to-peer** communication
- person-to-person communication
- ectronic commerce
- entertainment.(game playing,)

3 Mobile Users

- Text messaging or texting
- Smart phones,
- GPS (Global Positioning System)
- m-commerce
- NFC (Near Field Communication)

4 Social Issues

With the good comes the bad, as this new-found freedom brings with it many unsolved social, political, and ethical issues.

Social networks, message boards, content sharing sites, and a host of other applications allow people to share their views with like-minded individuals. As long as the subjects are restricted to technical topics or hobbies like gardening, not too many problems will arise.

The trouble comes with topics that people actually care about, like politics, religion, or sex. Views that are publicly posted may be deeply offensive to some people. Worse yet, they may not be politically correct. Furthermore, opinions need not be limited to text; high-resolution color photographs and video clips are easily shared over computer networks. Some people take a live-and-let-live view, but others feel that posting certain material (e.g., verbal attacks on particular countries or religions, pornography, etc.) is simply unacceptable and that such content must be censored. Different countries have different and conflicting laws in this area. Thus, the debate rages.

Computer networks make it very easy to communicate. They also make it easy for the people who run the network to snoop on the traffic. This sets up conflicts over issues such as **employee rights versus employer rights**. Many people read and write email at work. Many employers have claimed the right to read and possibly censor employee messages, including messages sent from a home computer outside working hours. Not all employees agree with this, especially the latter part.

Another conflict is centered around government versus citizen's rights.

A new twist with mobile devices is location privacy. As part of the process of providing service to your mobile device the network operators learn where you are at different times of day. This allows them to track your movements. They may know which nightclub you frequent and which medical center you visit.

Phishing ATTACK: ***Phishing*** is a type of social engineering ***attack*** often used to steal user data, including login credentials and credit card numbers. It occurs when an attacker, masquerading as a trusted entity, dupes a victim into opening an email, instant message, or text message.

BOTNET ATTACK: Botnets can be used to perform distributed denial-of-service attack (DDoS attack), steal data, send spam, and allows the attacker to access the device and its connection.

The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

I. **Delivery.** The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.

2 **Accuracy.** The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.

3. **Timeliness**. The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called *real-time* transmission.

4. **Jitter**. Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30 ms. If some of the packets arrive with 30-ms delay and others with 40-ms delay, an uneven quality in the video is the result.

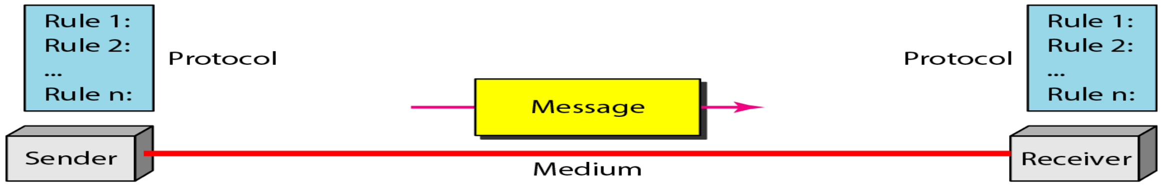
A data communications system has five components

I. **Message**. The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video. 2 **Sender**. The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.

3. **Receiver.** The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.

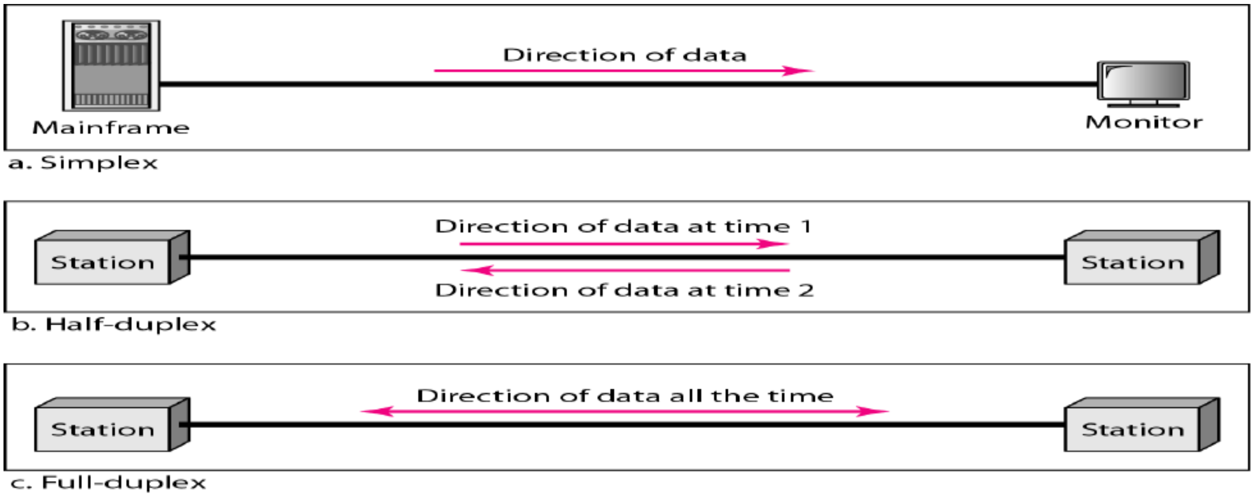
4. **Transmission medium**. The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.

5. **Protocol.** A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.



Data Flow

Communication between two devices can be simplex, half-duplex, or full-duplex as shown in Figure.


Direction of data
Mainframe
Monitor
Direction of data at time 1
Station
Station
Direction of data at time 2
Direction of data all the time
Station
Station

a. Simplex

b. Half-duplex

c. Full-duplex

Simplex In simplex mode, the communication is unidirectional, as on a one- way street. Only one of the two devices on a link can transmit; the other can only receive (Figure a). Keyboards and traditional monitors are examples of simplex devices.

Half-Duplex

In half-duplex mode, each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa (Figure b). Walkie-talkies and CB (citizens band) radios are both half- duplex systems.

Full-Duplex

In full-duplex, both stations can transmit and receive simultaneously (Figure c). One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time. The full-duplex mode is used when communication in both directions is required all the time.

Network Criteria

A network must be able to meet a certain number of criteria. The most important of these are performance, reliability, and security.

Performance

Performance can be measured in many ways, including transit time and response time. Transit time is the amount of time required for a message to travel from one device to another. Response time is the elapsed time between an inquiry and a response. The performance of a network depends on a number of factors, including the number of users, the type of transmission medium, the capabilities of the connected hardware, and the efficiency of thesoftware.

Performance is often evaluated by two networking metrics: throughput and delay. We often need more throughputs and less delay. However, these two criteria are often contradictory. If we try to send more data to the network, we may increase throughput but we increase the delay because of traffic congestion in the network.

Reliability: In addition to accuracy of delivery, network reliability is measured by the frequency of failure, the time it takes a link to recover from a failure, and the network's robustness in acatastrophe.

Security: Network security issues include protecting data from unauthorized access, protecting data from damage and development, and implementing policies and procedures for recovery from breaches and data losses.

Physical Structures

Before discussing networks, we need to define some network attributes.

Type of Connection

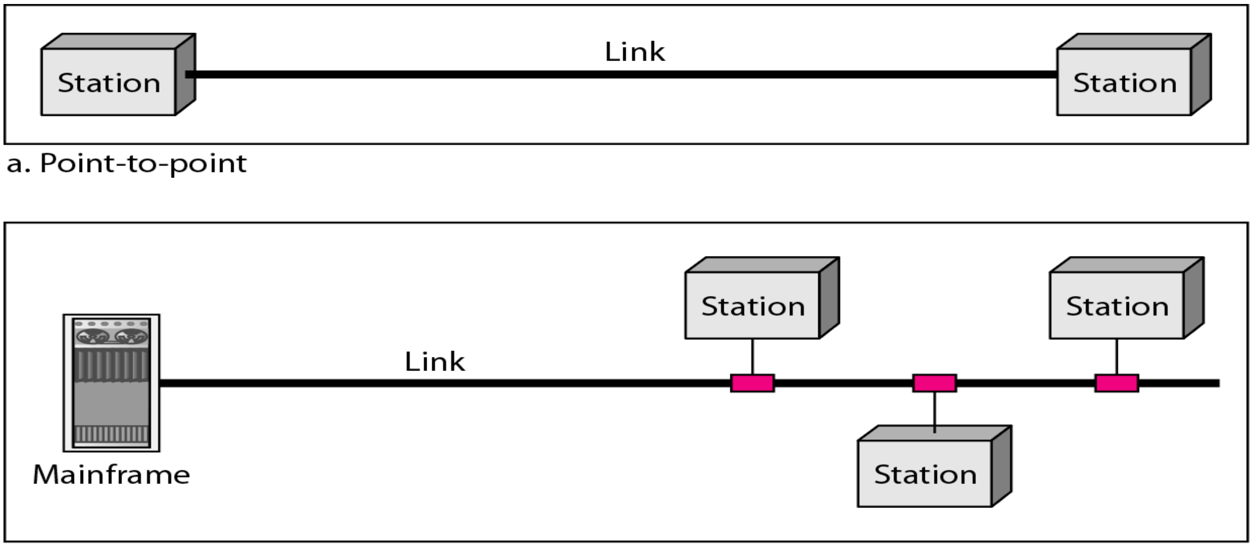
A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another.

There are two possible types of connections: point-to-point and multipoint. **Point-to-Point** A point-to-point connection provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links, are also possible

When you change television channels by infrared remote control, you are establishing a point-to-point connection between the remote control and the television's control system.

Multipoint A multipoint (also called multi-drop) connection is one in which more than two specific devices share a single link

In a multipoint environment, the capacity of the channel is shared, either spatially or temporally. If several devices can use the link simultaneously, it is a *spatially shared* connection. If users must take turns, it is a *timeshared* connection.



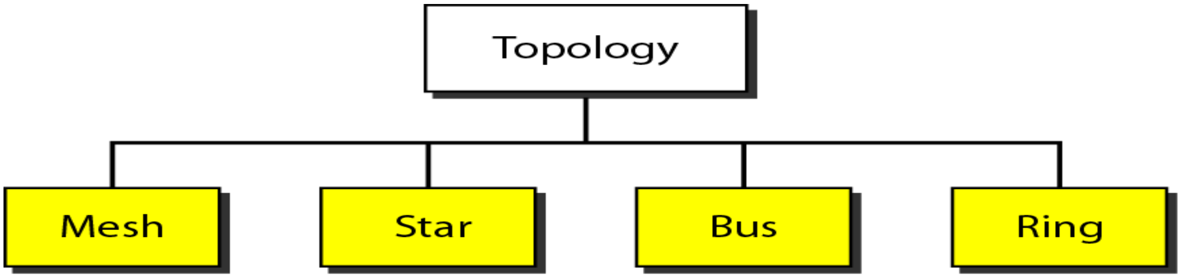
a. Point-to-point

b. Multipoint

Physical Topology

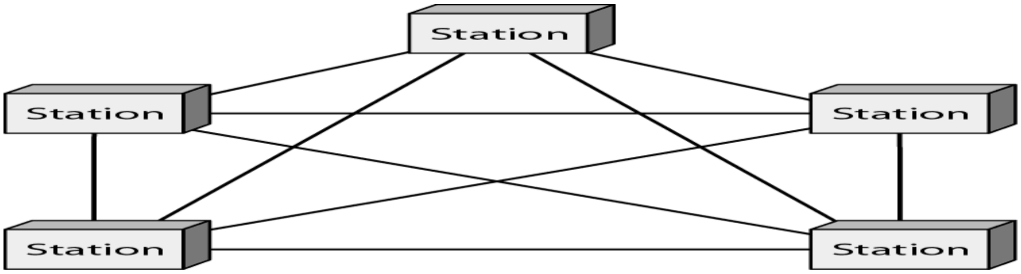
The term *physical topology* refers to the way in which a network is laid out physically.

Two or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called nodes) to one another.
There are four basic topologies possible: mesh, star, bus, and ring



MESH:

A mesh topology is the one where every node is connected to every other node in the network.



A mesh topology can be a **full mesh topology** or a **partially connected mesh topology**.

In a *full mesh topology*, every computer in the network has a connection to each of the other computers in that network. The number of connections in this network can be calculated using the following formula (***n*** is the number of computers in the network): **n(n-1)/2**

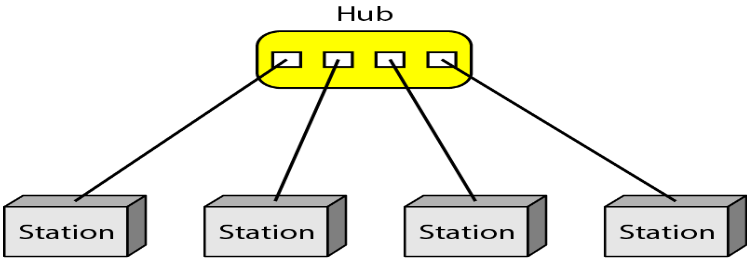
In a *partially connected mesh topology*, at least two of the computers in the network have connections to multiple other computers in that network. It is an inexpensive way to implement redundancy in a network. In the event that one of the primary computers or connections in the network fails, the rest of the network continues to operatenormally.

Advantages of a mesh topology

- Can handle high amounts of traffic, because multiple devices can transmit data simultaneously.
- A failure of one device does not cause a break in the network or transmission of data.
- Adding additional devices does not disrupt data transmission between other devices.

Disadvantages of a mesh topology

- The cost to implement is higher than other network topologies, making it a less desirable option.
- Building and maintaining the topology is difficult and time consuming.
- The chance of redundant connections is high, which adds to the high costs and potential for reduced efficiency.



STAR:

A star network, **star topology** is one of the most common network setups. In this configuration, every node connects to a central network device, like a hub, switch, or computer. The central network device acts as a server and the peripheral devices act as clients. Depending on the type of network card used in each computer of the star topology, a coaxial cable or a RJ-45 network cable is used to connect computers together.

Advantages of star topology

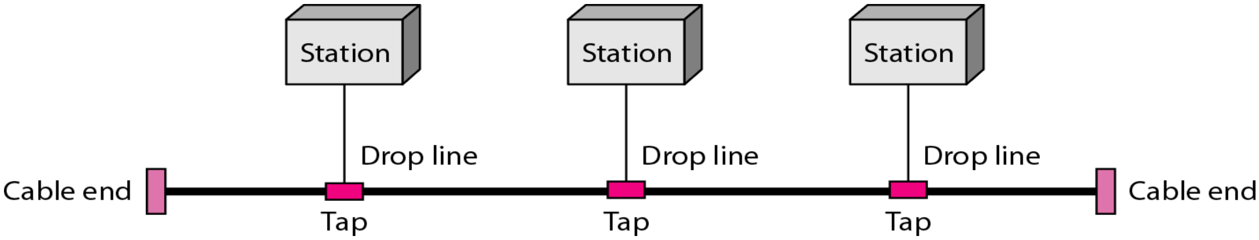
- Centralized management of the network, through the use of the central computer, hub, or switch.

- Easy to add another computer to the network.
- If one computer on the network fails, the rest of the network continues to function normally.
- The star topology is used in local-area networks (LANs), High-speed LANs often use a star topology with a central hub.

Disadvantages of star topology

- Can have a higher cost to implement, especially when using a switch or router as the central network device.
- The central network device determines the performance and number of nodes the network can handle.
- If the central computer, hub, or switch fails, the entire network goes down and all computers are disconnected from the network

BUS:



a **line topology**, a **bus topology** is a network setup in which each computer and network device are connected to a single cable or backbone.

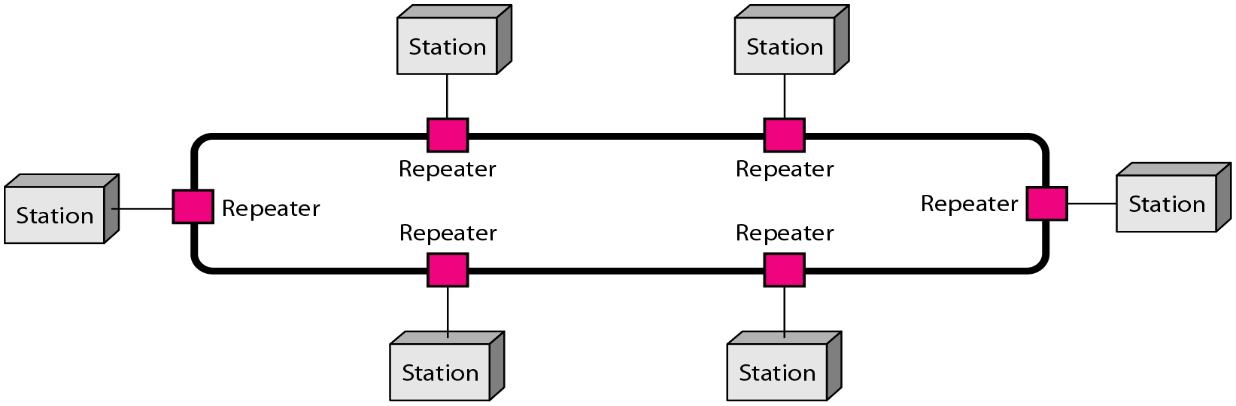
Advantages of bus topology

- It works well when you have a small network.
- It's the easiest network topology for connecting computers or peripherals in a linear fashion.
- It requires less cable length than a star topology.

Disadvantages of bus topology

- It can be difficult to identify the problems if the whole network goes down.
- It can be hard to troubleshoot individual device issues.
- Bus topology is not great for large networks.
- Terminators are required for both ends of the main cable.
- Additional devices slow the network down.
- If a main cable is damaged, the network fails or splits into two.

RING:



ring topology is a network configuration in which device connections create a circular data path. In a ring network, packets of data travel from one device to the next until they reach their destination. Most ring topologies allow packets to travel only in one direction, called a **unidirectional** ring network. Others permit data to move in either direction, called **bidirectional**.

The major disadvantage of a ring topology is that if any individual connection in the ring is broken, the entire network is affected.

Ring topologies may be used in either local area networks (LANs) or wide area networks (WANs).

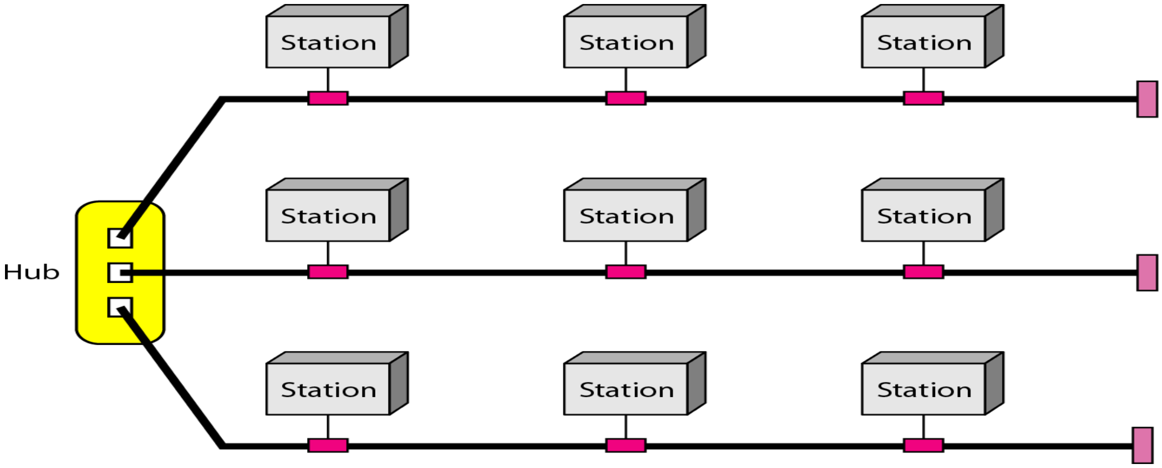
Advantages of ring topology

- All data flows in one direction, reducing the chance of packet collisions.
- A network server is not needed to control network connectivity between each workstation.
- Data can transfer between workstations at high speeds.
- Additional workstations can be added without impacting performance of the network.

Disadvantages of ring topology

- All data being transferred over the network must pass through each workstation on the network, which can make it slower than a star topology.
- The entire network will be impacted if one workstation shuts down.
- The hardware needed to connect each workstation to the network is more expensive than Ethernet cards and hubs/switches.

Hybrid Topology A network can be hybrid. For example, we can have a main star topology with each branch connecting several stations in a bus topology as shown in Figure



Types of Network based on size

The types of network are classified based upon the size, the area it covers and its physical architecture. The three primary network categories are LAN, WAN and MAN. Each network differs in their characteristics such as distance, transmission speed, cables and cost.

Basic types

LAN (Local Area Network)

Group of interconnected computers within a small area.(room,building, campus)
Two or more pc's can from a LAN to share files, folders, printers, applications and other devices.
Coaxial or CAT 5 cables are normally used for connections.
Due to short distances, errors and noise are minimum.
Data transfer rate is 10 to 100 mbps.
Example: A computer lab in a school.

MAN (Metropolitan Area Network)

Design to extend over a large area.

Connecting number of LAN's to form larger network, so that resources can be shared.
Networks can be up to 5 to 50 km.
Owned by organization or individual.
Data transfer rate is low compare to
LAN.
Example: Organization with different branches located in the city.

WAN (Wide Area Network)

Are country and worldwide
network. Contains multiple
LAN's and MAN's.
Distinguished in terms of geographical range.
Uses satellites and microwave relays.

Data transfer rate depends upon the ISP provider and varies over the location. Best example is the internet.

Other types

WLAN (Wireless LAN)

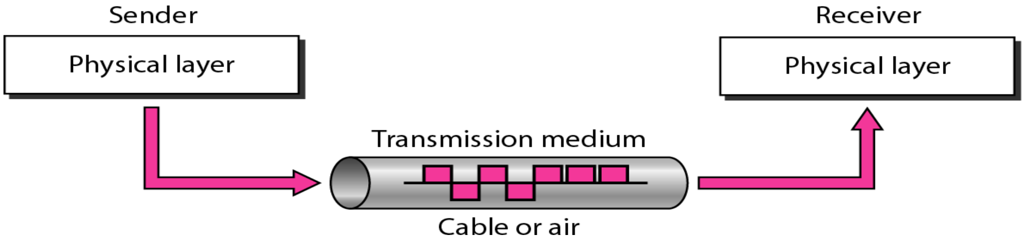
A LAN that uses high frequency radio waves for communication. Provides short range connectivity with high speed data transmission. **PAN (Personal Area Network)**

Network organized by the individual user for its personal use.

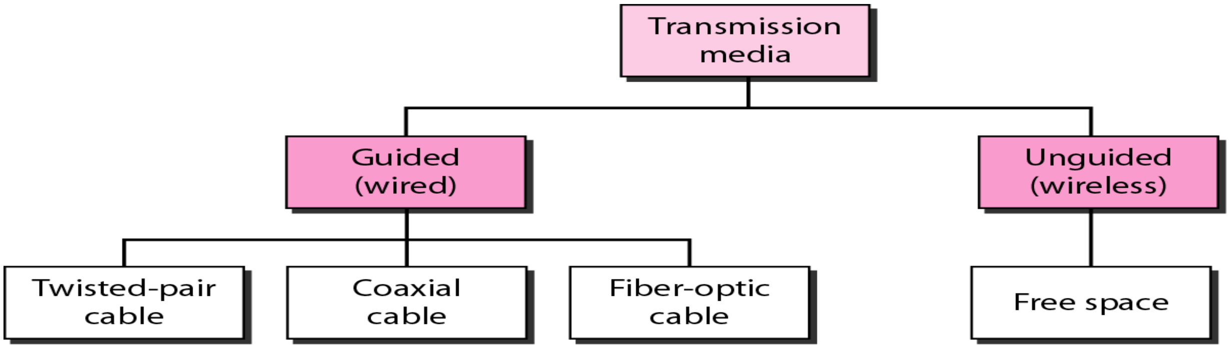
SAN (Storage Area Network)

Connects servers to data storage devices via fiber-optic cables.
E.g.: Used for daily backup of organization or a mirror copy

A **transmission medium** can be broadly defined as anything that can carry information from a source to a destination.



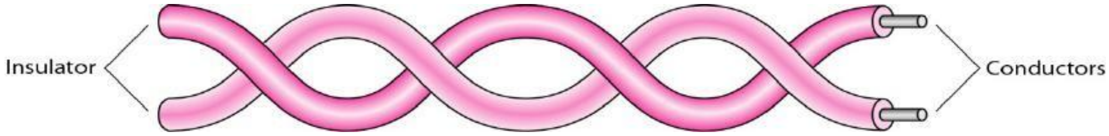
<u>Classes of transmission media</u>

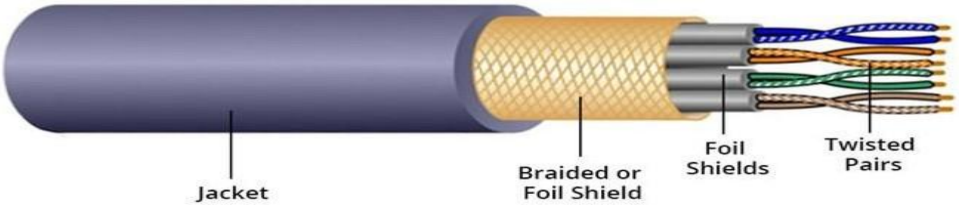


<u>**Guided Media**</u>: Guided media, which are those that provide a medium from one device to another, include twisted-pair cable, coaxial cable, and fiber-optic cable.

<u>**Twisted-Pair Cable**</u>: A twisted pair consists of two conductors (normally copper),

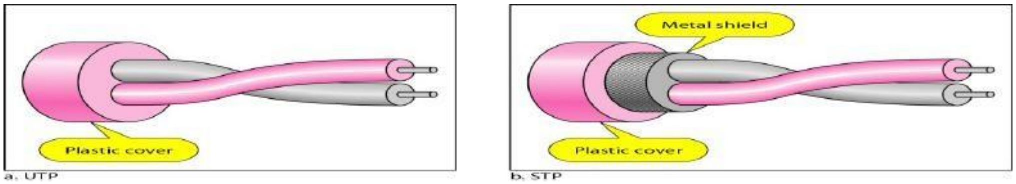
each with its own plastic insulation, twisted together. One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference.





Unshielded Versus Shielded Twisted-Pair Cable

The most common twisted-pair cable used in communications is referred to as unshielded twisted-pair (UTP). STP cable has a metal foil or braided mesh covering that encases each pair of insulated conductors. Although metal casing improves the quality of cable by preventing the penetration of noise or crosstalk, it is bulkier and moreexpensive.



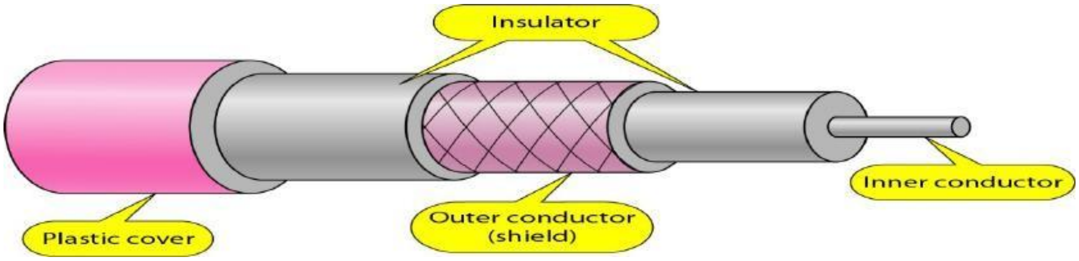
The most common UTP connector is RJ45 (RJ stands for registered jack)

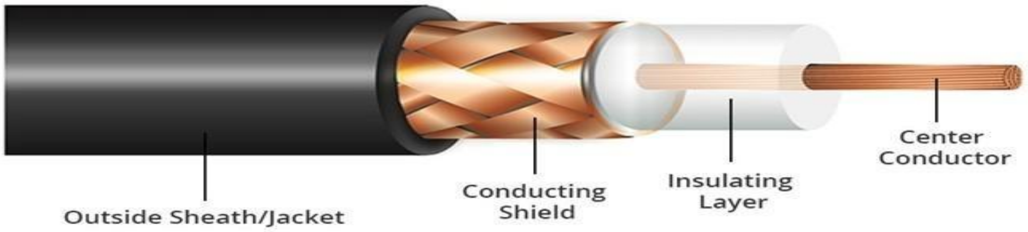
Applications

Twisted-pair cables are used in telephone lines to provide voice and data channels. Local-area networks, such as l0Base-T and l00Base-T, also use twisted-pair cables.

Coaxial Cable

Coaxial cable (or *coax)* carries signals of higher frequency ranges than those in twisted pair cable. coax has a central core conductor of solid or stranded wire (usuallycopper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit.This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover.





The most common type of connector used today is the Bayone-Neill-Concelman (BNe), connector.

Applications

Coaxial cable was widely used in analog telephone networks,digital telephone networks Cable TV networks also use coaxialcables.

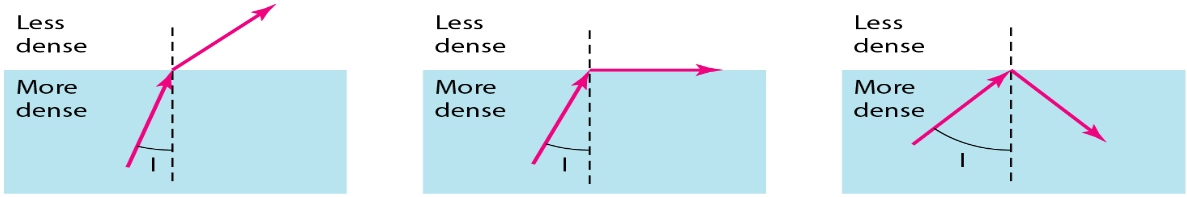
Another common application of coaxial cable is in traditional Ethernet LANs

Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. Light travels in a straight line as long as it is moving through a single uniform substance.

If a ray of light traveling through one substance suddenly enters another substance(of a different density), the ray changes direction.

Bending of lightray

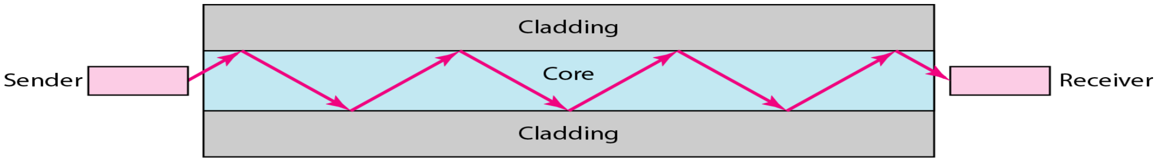


I < critical angle, refraction

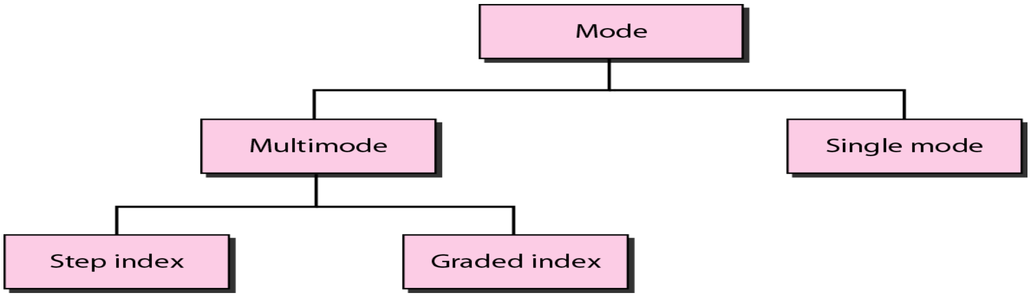
I = critical angle, refraction

I > critical angle, reflection

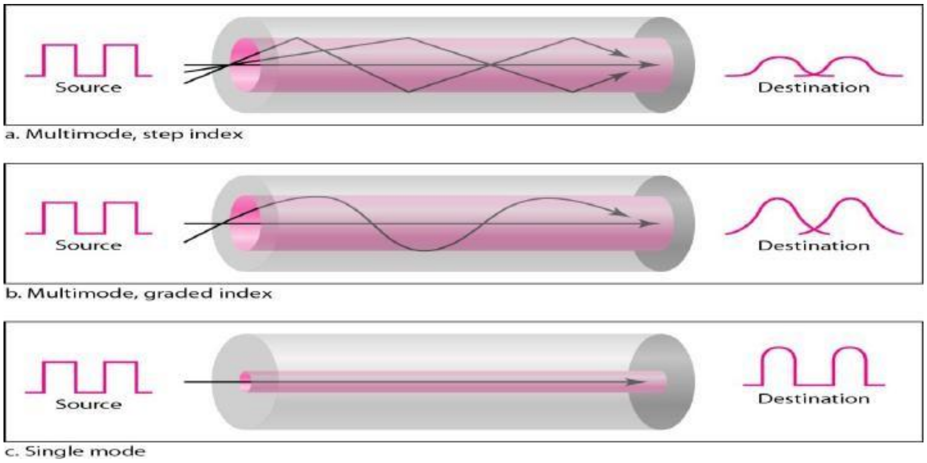
Optical fibers use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic.



Propagation Modes



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



a. Multimode, step index

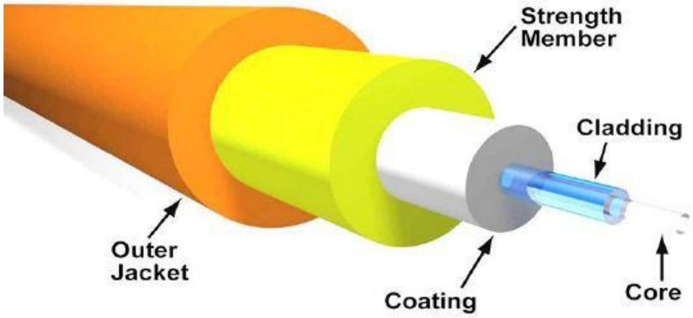
b. Multimode, graded index

c. Single mode

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

A second type of fiber, called **multimode graded-index fiber**, decreases this distortion of the signal through the cable. The word *index* here refers to the index of refraction.

Single-Mode: Single-mode uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal.Fiber Construction



The **subscriber channel** (SC) **connector,** The **straight-tip** (ST) **connector, MT-RJ(mechanical transfer registered jack)** is a connector

Applications

Fiber-optic cable is often found in backbone networks because its wide

Some cable TV companies use a combination of optical fiber and coaxial cable,thus creating a hybrid network.

Local-area networks such as 100Base-FX network (Fast

Ethernet) and 1000Base-X also use fiber-optic cable

Advantages and Disadvantages of Optical Fiber

Advantages Fiber-optic cable has several advantages over metallic cable (twisted pair or coaxial).

1 Higher bandwidth.

2 Less signal attenuation. Fiber-optic transmission distance is significantly greaterthan that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted- pair cable.

3 Immunity to electromagnetic interference. Electromagnetic noise cannot affect fiber-optic cables.

4 Resistance to corrosive materials. Glass is more resistant to corrosive materials than copper.

5 Light weight. Fiber-optic cables are much lighter than copper cables.

6 Greater immunity to tapping. Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped.

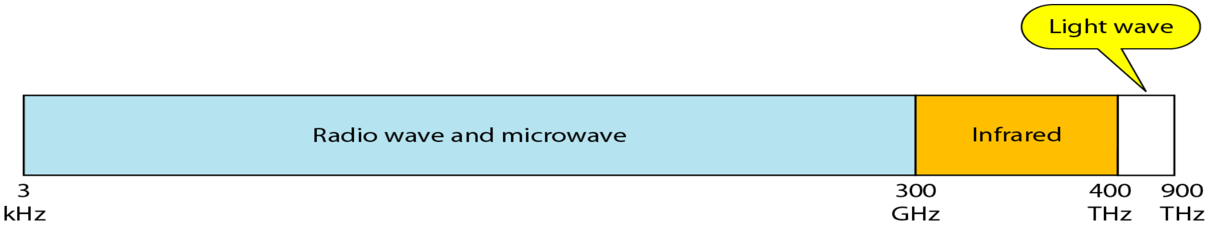
Disadvantages There are some disadvantages in the use of optical fiber. 1Installation and maintenance

2 Unidirectional light propagation. Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed.

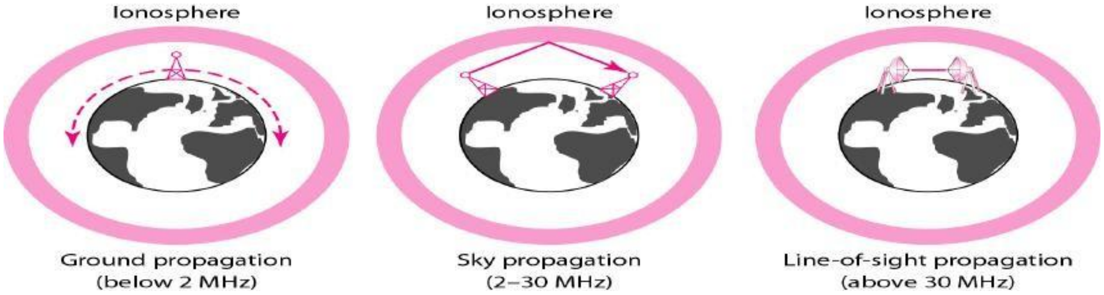
3 Cost. The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.

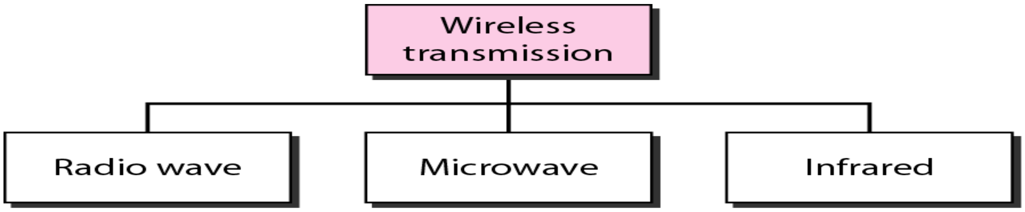
UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication.



Unguided signals can travel from the source to destination in several ways: ground propagation, sky propagation, and line-of-sight propagation, as shown in Figure



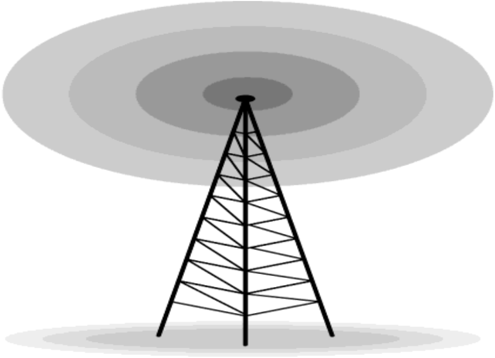


Radio Waves

Electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called radio waves. Radio waves are omni directional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The omni directional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Omni directional Antenna

Radio waves use omnidirectional antennas that send out signals in all directions. Based on the wavelength, strength, and the purpose of transmission, we can have several types of antennas. Figure shows an omnidirectional antenna.



Applications

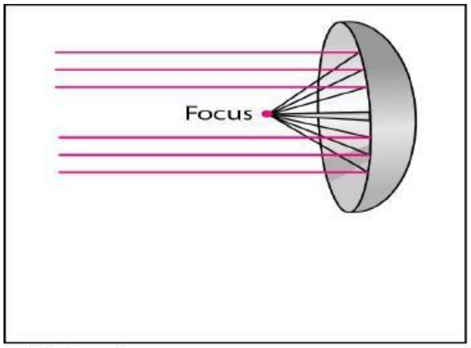
The Omni directional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.

Microwaves

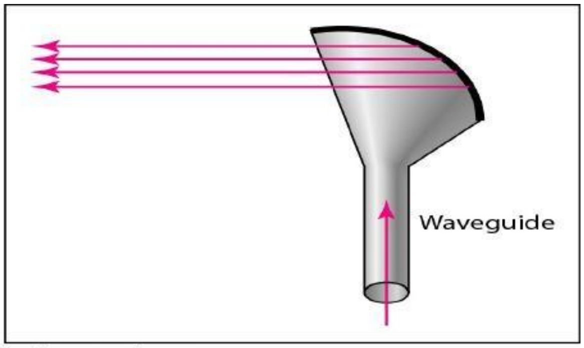
Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves. Microwaves are unidirectional. The sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas

Unidirectional Antenna

Microwaves need unidirectional antennas that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the horn



a. Dish antenna



b. Horn antenna

Applications:

Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous

characteristic prevents interference between one system and another; a short- range communication system in one room cannot be affected by another system in the next room.

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

Applications:

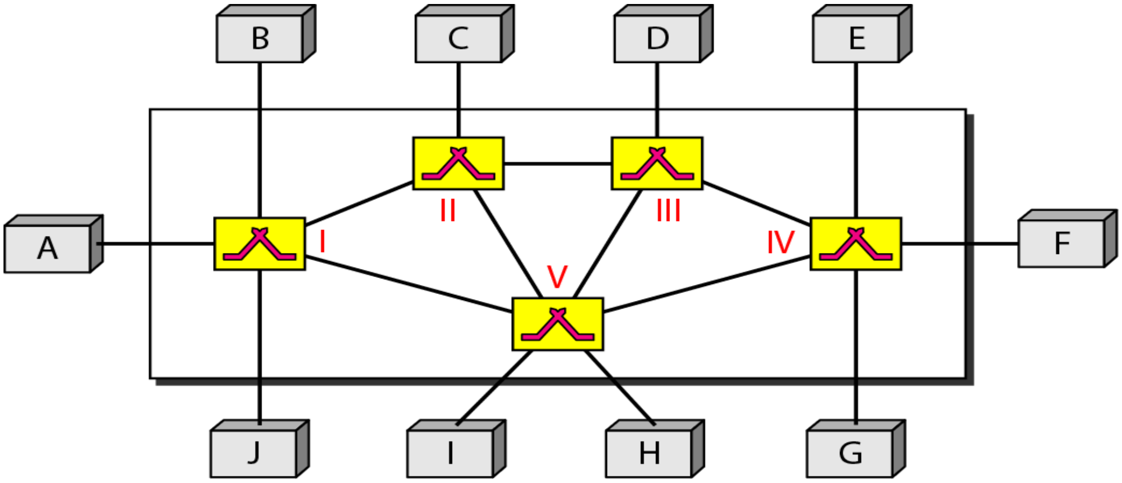
Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

Switching

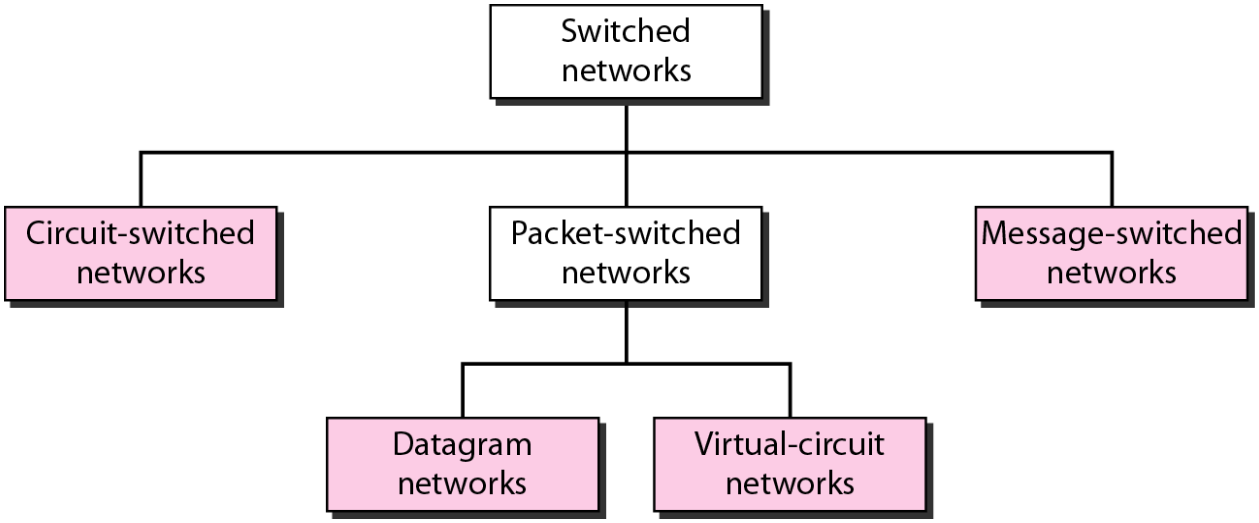
A network is a set of connected devices. Whenever we have multiple devices, we have the problem of how to connect them to make one-to-one communication possible. One solution is to make a point-to-point connection between each pair of devices (a mesh topology) or between a central device and every other device (a star topology). These methods, however, are impractical and wasteful when applied to very large networks.

The number and length of the links require too much infrastructure to be cost-efficient, and the majority of those links would be idle most of the time.

A better solution is switching. A switched network consists of a series of interlinked nodes, called switches. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (computers or telephones, for example). Others are used only for routing. Figure shows a switched network.



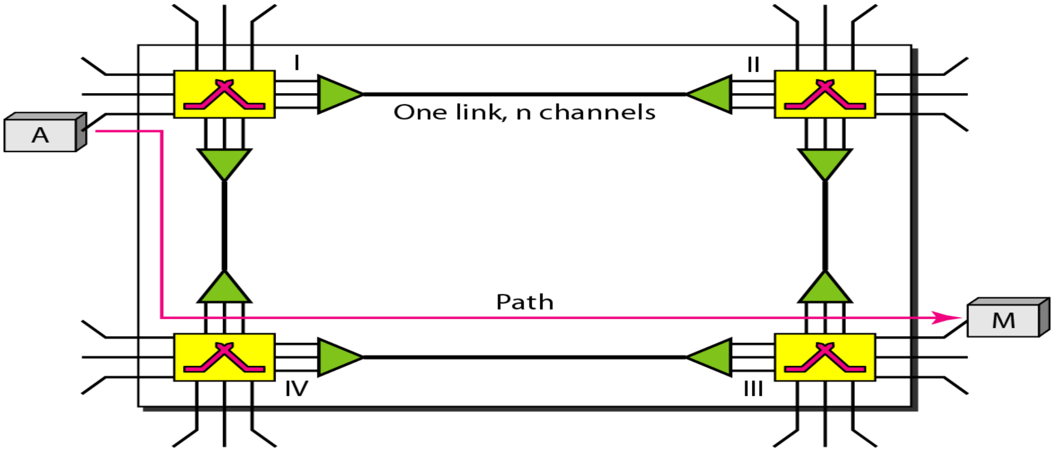
We can then divide today's networks into three broad categories: circuit- switched networks, packet-switched networks, and message-switched. Packet- switched networks can further be divided into two subcategories-virtual-circuit networks and datagram networks as shown in Figure.



CIRCUIT-SWITCHED NETWORKS

A circuit-switched network consists of a set of switches connected by physical links. A connection between two stations is a dedicated path made of one or more links. However, each connection uses only one dedicated channel on each link. Each link is normally divided into n channels by using FDM or TDM.

In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of data transfer until the teardown phase



Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, data transfer, and connection teardown.

Setup Phase

Before the two parties (or multiple parties in a conference call) can communicate, a dedicated circuit (combination of channels in links) needs to be established. Connection setup means creating dedicated channels between the switches. For example, in Figure, when system A needs to connect to system M, it sends a setup request that includes the address of system M, to switch I. Switch I finds a channel between itself and switch IV that can be dedicated for this purpose. Switch I then sends the request to switch IV, which finds a

dedicated channel between itself and switch III. Switch III informs system M of system A's intention at this time.

In the next step to making a connection, an acknowledgment from system M needs to be sent in the opposite direction to system A. Only after system A receives this acknowledgment is the connection established.

Data Transfer Phase

After the establishment of the dedicated circuit (channels), the two parties can transfer data.

Teardown Phase

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

Efficiency

It can be argued that circuit-switched networks are not as efficient as the other two types of networks because resources are allocated during the entire duration of the connection. These resources are unavailable to other connections.

Delay

Although a circuit-switched network normally has low efficiency, the delay in this type of network is minimal. During data transfer the data are not delayed at each switch; the resources are allocated for the duration of the connection.

The total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit.

Switching at the physical layer **in the traditional telephone network** uses **the circuit-switching**

DATAGRAM NETWORKS

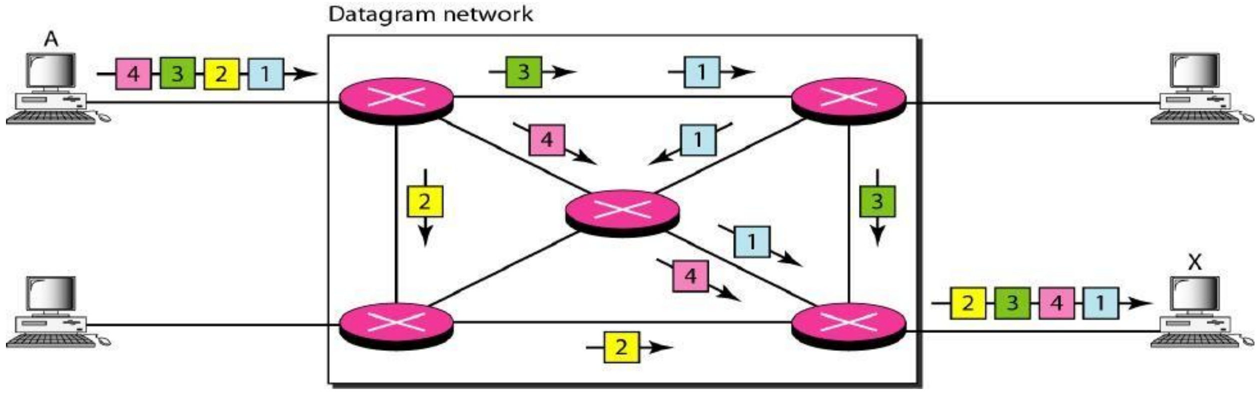
In a packet-switched network, there is no resource reservation; resources are allocated on demand. The allocation is done on a first come, first-served basis. When a switch receives a packet, no matter what is the source or destination, the packet must wait if there are other packets being processed. This lack of reservation may create delay. For example, if we do not have a reservation at a restaurant, we might have to wait.

In a datagram network, each packet is treated independently of all others. Packets in this approach are referred to as datagrams. Datagram switching is normally done at the network layer.

Figure shows how the datagram approach is used to deliver four packets from station A to station X. The switches in a datagram network are traditionally referred to as routers.

The datagram networks are sometimes referred to as connectionless networks. The term ***connectionless*** here means that the switch (packet switch) does not keep information about the connection state. There are no setup or teardown phases. Each packet is treated the same by a switch regardless of its source ordestination.

A switch in a datagram network uses a routing table that is based on the destination address. The destination address in the header of a packet in a datagram network remains the same during the entire journey of the packet.



Efficiency

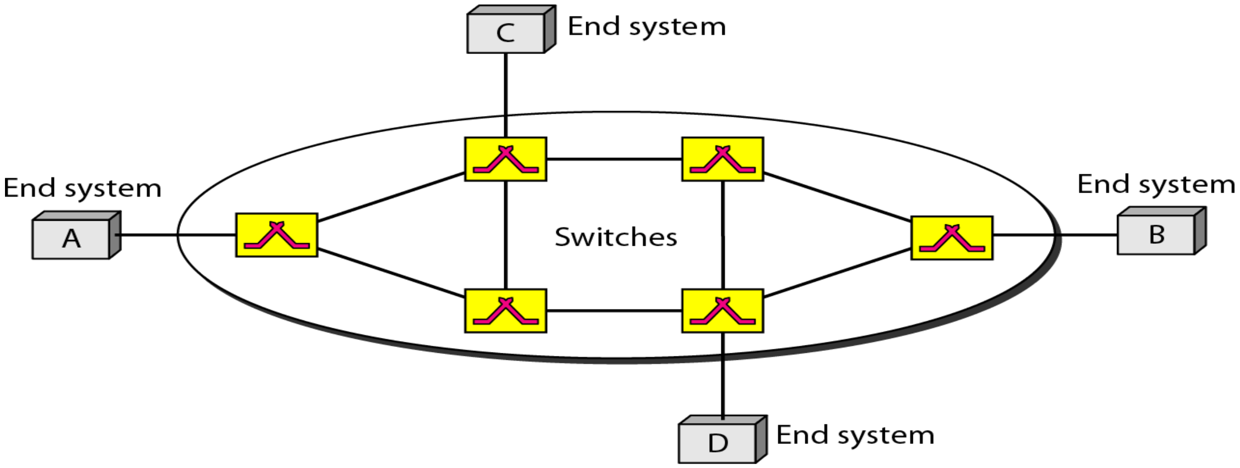
The efficiency of a datagram network is better than that of a circuit-switched network; resources are allocated only when there are packets to be transferred. **Delay**

There may be greater delay in a datagram network than in a virtual-circuit network. Although there are no setup and teardown phases, each packet may experience a wait at a switch before it is forwarded. In addition, since not all packets in a message necessarily travel through the same switches, the delay is not uniform for the packets of a message.

Switching in the Internet is done by using the datagram approach to packet switching at the network layer.

VIRTUAL-CIRCUIT NETWORKS

A virtual-circuit network is a cross between a circuit-switched network and a datagram network. It has some characteristics ofboth.



1. As in a circuit-switched network, there are setup and teardown phases in addition to the data transfer phase.

2. Resources can be allocated during the setup phase, as in a circuit-switched network, or on demand, as in a datagram network.

3. As in a datagram network, data are packetized and each packet carries an address in the header. However, the address in the header has local jurisdiction (it defines what should be the next switch and the channel on which the packet is being carried), not end-to-end jurisdiction.

4. As in a circuit-switched network, all packets follow the same path established during the connection.

5. A virtual-circuit network is normally implemented in the data link layer, while a circuit-switched network is implemented in the physical layer and a datagram network in the network layer.

Addressing

In a virtual-circuit network, two types of addressing are involved: global and local (virtual-circuit identifier).

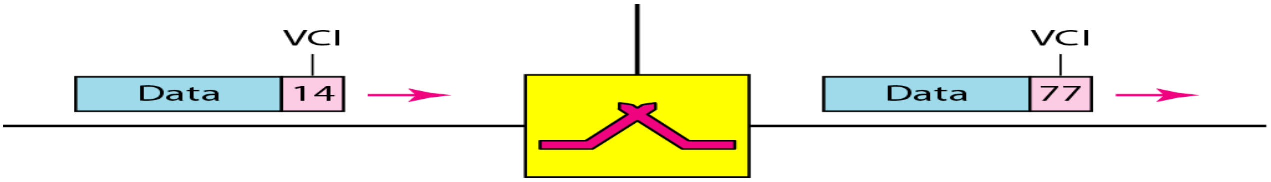
Global Addressing

A source or a destination needs to have a global address-an address that can be unique in the scope of the network.

Virtual-Circuit Identifier

The identifier that is actually used for data transfer is called the virtual-circuit identifier (VCI). A VCI, unlike a global address, is a small number that has only switch scope; it is used by a frame between two switches. When a frame arrives at a switch, it has a VCI; when it leaves, it has a different VCl.

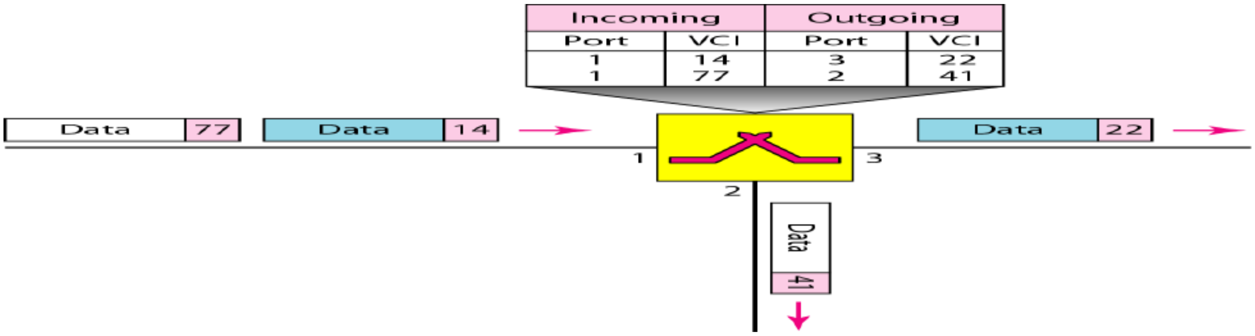
Figure shows how the VCI in a data frame changes from one switch to another. Note that a VCI does not need to be a large number since each switch can use its own unique set of VCls.



Three Phases

Three phases in a virtual-circuit network: setup, data transfer, and teardown. We first discuss the data transfer phase, which is more straightforward; we then talk about the setup and teardown phases.

Data Transfer Phase



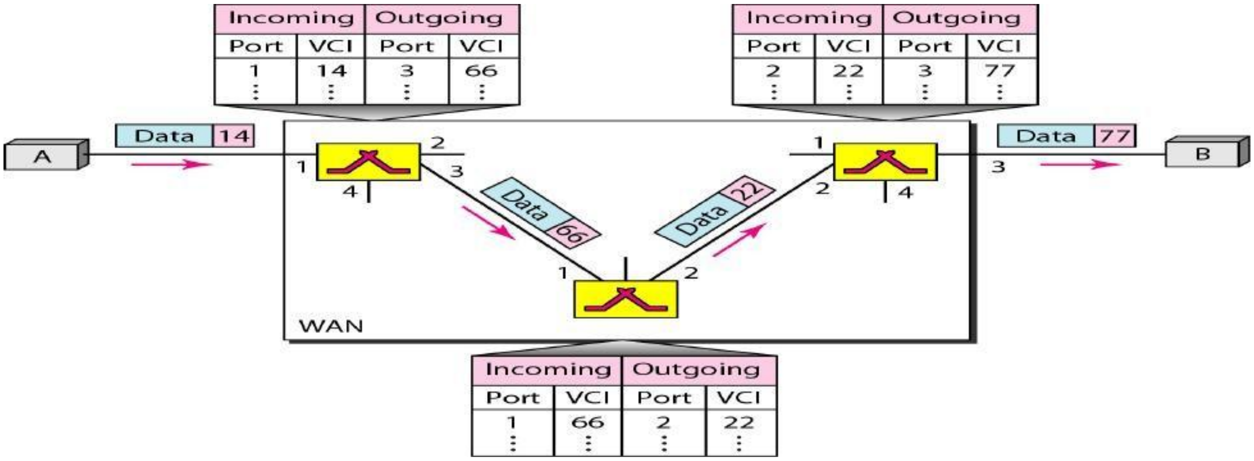
Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	22
1	77	2	41

To transfer a frame from a source to its destination, all switches need to have a table entry for this virtual circuit. The table, in its simplest form, has four columns.

We show later how the switches make their table entries, but for the moment we assume that each switch has a table with entries for all active virtual circuits. Figure shows such a switch and its corresponding table.

Figure shows a frame arriving at port 1 with a VCI of 14. When the frame arrives, the switch looks in its table to find port 1 and a VCI of 14. When it is found, the switch knows to change the VCI to 22 and send out the frame from port 3.

Figure shows how a frame from source A reaches destination B and how its VCI changes during the trip.



Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	66
⋮	⋮	⋮	⋮

Incoming		Outgoing	
Port	VCI	Port	VCI
2	22	3	77
⋮	⋮	⋮	⋮

Incoming		Outgoing	
Port	VCI	Port	VCI
1	66	2	22
⋮	⋮	⋮	⋮

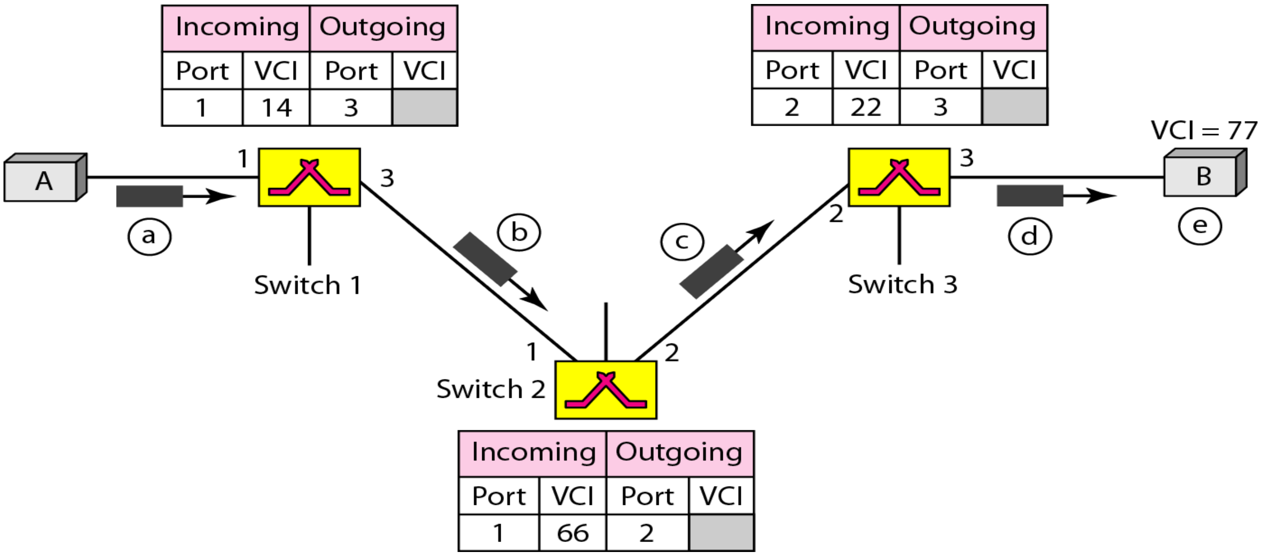
Each switch changes the VCI and routes the frame.

The data transfer phase is active until the source sends all its frames to the destination. The procedure at the switch is the same for each frame of a message. The process creates a virtual circuit, not a real circuit, between the source and destination.

Setup Phase

In the setup phase, a switch creates an entry for a virtual circuit. For example, suppose source A needs to create a virtual circuit to B. Two steps are required: the setup request and the acknowledgment.

Setup Request A setup request frame is sent from the source to the destination. Figure shows the process.



Switch 1

Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	

Switch 3

Incoming		Outgoing	
Port	VCI	Port	VCI
2	22	3	

Switch 2

Incoming		Outgoing	
Port	VCI	Port	VCI
1	66	2	

a. Source A sends a setup frame to switch 1.

b. Switch 1 receives the setup request frame. It knows that a frame going from A to B goes out through port 3. For the moment, assume that it knows the output port. The switch creates an entry in its table for this virtual circuit, but it is only able to fill three of the four columns. The switch assigns the incoming port (1) and chooses an available incoming VCI (14) and the outgoing port (3). It does not yet know the outgoing VCI, which will be found during the acknowledgment step. The switch then forwards the frame through port 3 to switch 2.

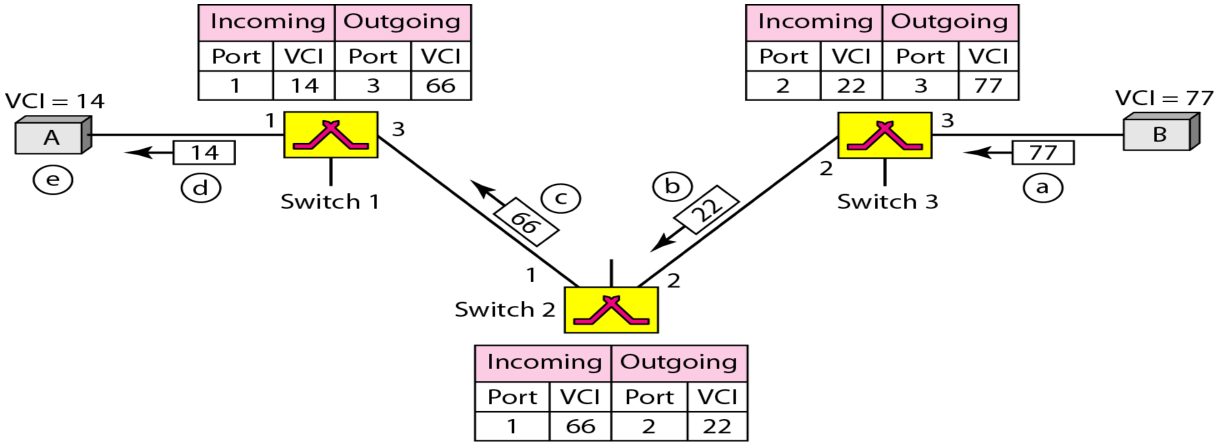
c. Switch 2 receives the setup request frame. The same events happen here as at switch 1; three columns of the table are completed: in this case, incoming port (l), incoming VCI (66), and outgoing port (2).

d. Switch 3 receives the setup request frame. Again, three columns are completed: incoming port (2), incoming VCI (22), and outgoing port (3).

e. Destination B receives the setup frame, and if it is ready to receive frames from A, it assigns a VCI to the incoming frames that come from A, in this case 77. This VCI lets the destination know that the frames come from A, and not other sources.

Acknowledgment A special frame, called the acknowledgment frame, completes the entries in the switching tables.

Figure shos the process.



Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	66

Incoming		Outgoing	
Port	VCI	Port	VCI
2	22	3	77

Incoming		Outgoing	
Port	VCI	Port	VCI
1	66	2	22

a. The destination sends an acknowledgment to switch 3. The acknowledgment carries the global source and destination addresses so the switch knows which entry in the table is to be completed. The frame also carries VCI 77, chosen by the destination as the incoming VCI for frames from A. Switch 3 uses this VCI to complete the outgoing VCI column for this entry. Note that 77 is the incoming VCI for destination B, but the outgoing VCI for switch 3.

b. Switch 3 sends an acknowledgment to switch 2 that contains its incoming VCI in the table, chosen in the previous step. Switch 2 uses this as the outgoing VCI in the table.

c. Switch 2 sends an acknowledgment to switch 1 that contains its incoming VCI in the table, chosen in the previous step. Switch 1 uses this as the outgoing VCI in the table.

d. Finally switch 1 sends an acknowledgment to source A that contains its incoming VCI in the table, chosen in the previous step.

e. The source uses this as the outgoing VCI for the data frames to be sent to destination B.

Teardown Phase

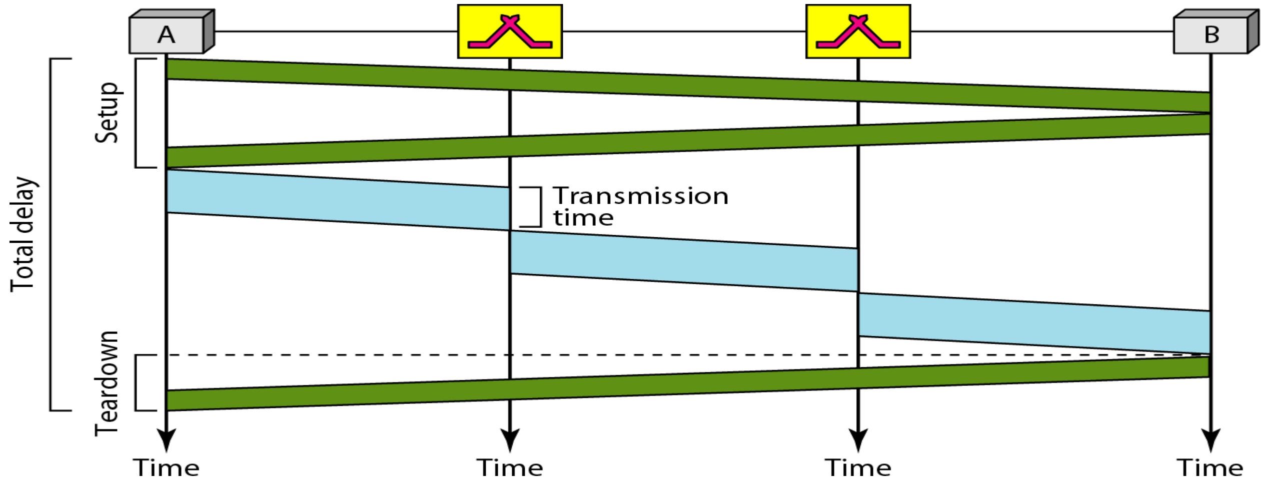
In this phase, source A, after sending all frames to B, sends a special frame called a *teardown request*. Destination B responds with a teardown confirmation frame. All switches delete the corresponding entry from their tables.

Efficiency

In virtual-circuit switching, all packets belonging to the same source and destination travel the same path; but the packets may arrive at the destination with different delays if resource allocation is on demand.

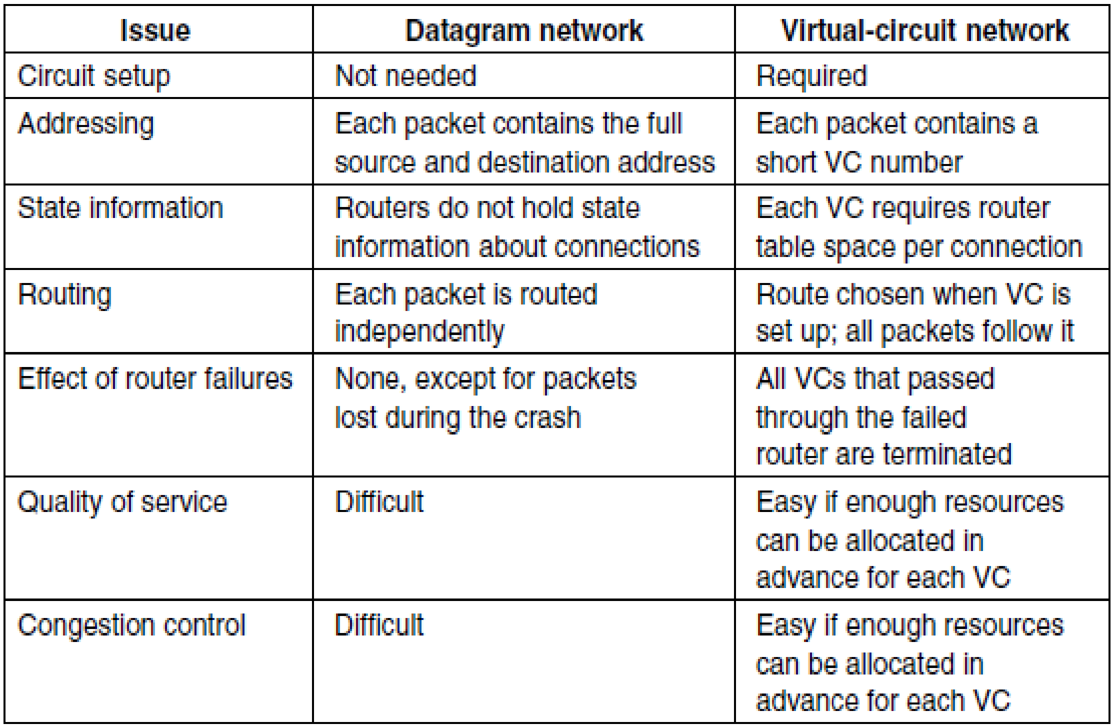
Delay

In a virtual-circuit network, there is a one-time delay for setup and a one-time delay for teardown. If resources are allocated during the setup phase, there is no wait time for individual packets. Figure shows the delay for a packet traveling through two switches in a virtual-circuit network



Switching at the data link layer in a switched WAN is normally implemented by using virtual-circuit techniques.

Comparison



Issue	Datagram network	Virtual-circuit network
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Routers do not hold state information about connections	Each VC requires router table space per connection
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow it
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Quality of service	Difficult	Easy if enough resources can be allocated in advance for each VC
Congestion control	Difficult	Easy if enough resources can be allocated in advance for each VC

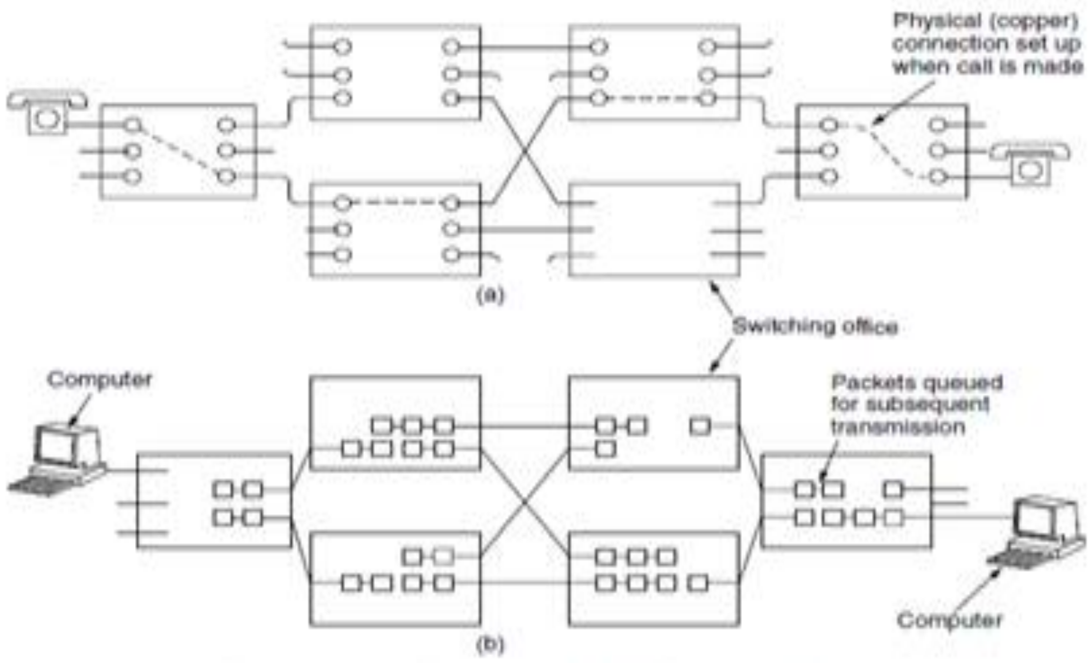

Physical (copper) connection set up when call is made
(a)
Switching office
Computer
Packets queued for subsequent transmission
Computer
(b)

Figure (a) Circuit switching. (b) Packet switching.

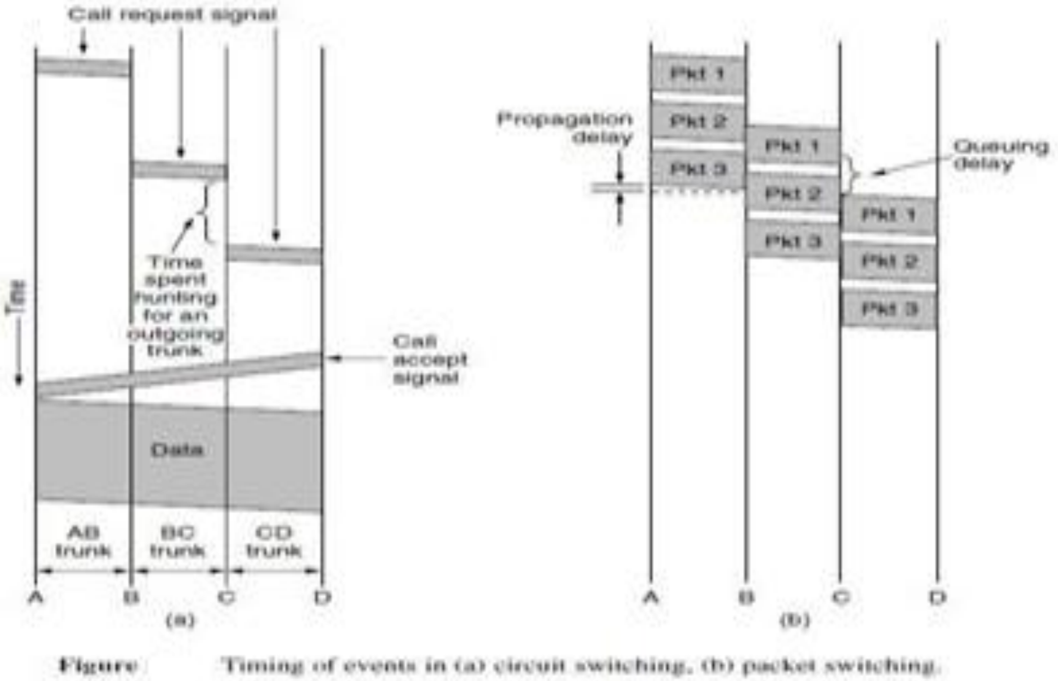
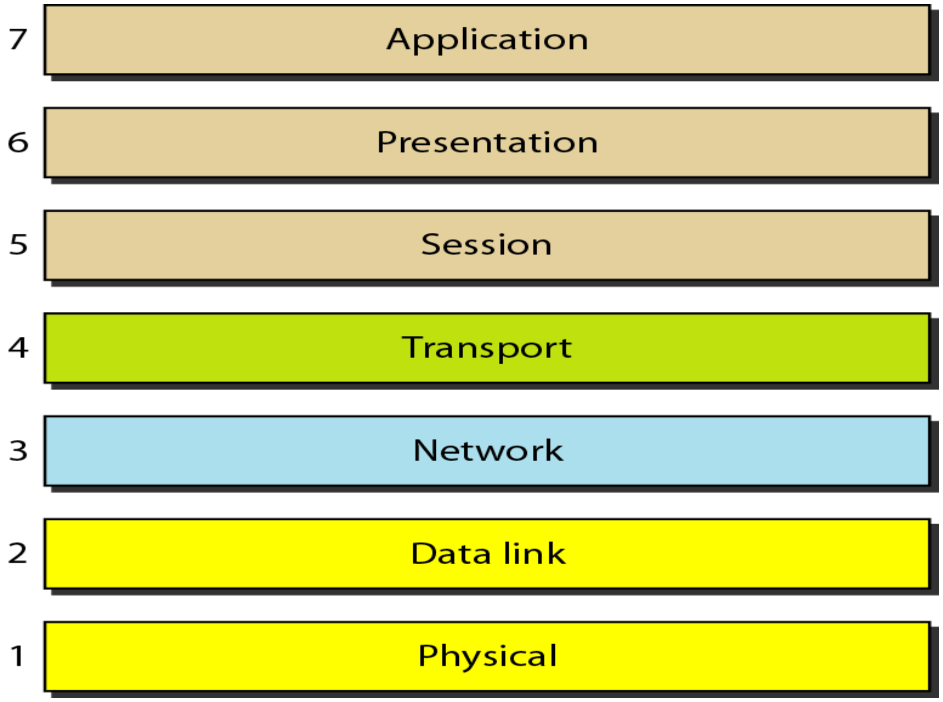


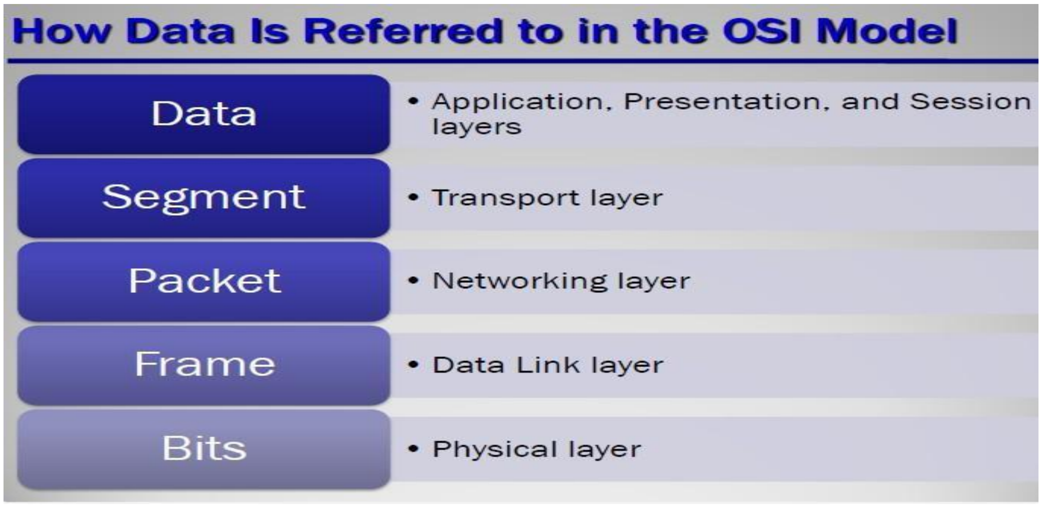
Figure Timing of events in (a) circuit switching, (b) packet switching.

OSI

- OSI stands for Open Systems Interconnection
- Created by International Standards Organization (ISO)
- Was created as a framework and reference model to explain how different networking technologies work together andinteract
- It is not a standard that networking protocols must follow
- Each layer has specific functions it is responsible for
- All layers work together in the correct order to move data around a network



Top to bottom
–All People Seem To Need Data
Processing Bottom to top
–Please Do Not Throw Sausage Pizza Away



How Data Is Referred to in the OSI Model

Data	• Application, Presentation, and Session layers
Segment	• Transport layer
Packet	• Networking layer
Frame	• Data Link layer
Bits	• Physical layer

Physical Layer

- Deals with all aspects of physically moving data from one computer to the next
- Converts data from the upper layers into 1s and 0s for transmission over media
- Defines how data is encoded onto the media to transmit the data
- Defined on this layer: Cable standards, wireless standards, and fiber optic standards. Copper wiring, fiber optic cable, radio frequencies, anything that can be used to transmit data is defined on the Physical layer of the OSI Model
- Device example: Hub
- Used to transmit data

Data Link Layer

- Is responsible for moving frames from node to node or computer to computer
- Can move frames from one adjacent computer to another, cannot move frames across routers
- Encapsulation = frame
- Requires MAC address or *physical address*
- Protocols defined include Ethernet Protocol and Point-to-Point Protocol (PPP)
- Device example: Switch
- Two sublayers: Logical Link Control (LLC) and the Media Access Control (MAC)
  - Logical Link Control (LLC)
    - –Data Link layer addressing, flow control, address notification, error control
  - Media Access Control (MAC)
    - –Determines which computer has access to the network media at any given time
    - –Determines where one frame ends and the next one starts, called frame synchronization

Network Layer

- Responsible for moving packets (data) from one end of the network to the other, called *end-to-end communications*

- Requires *logical addresses* such as IP addresses
- Device example: Router
- –Routing is the ability of various network devices and their related software to move data packets from source to destination

Transport Layer

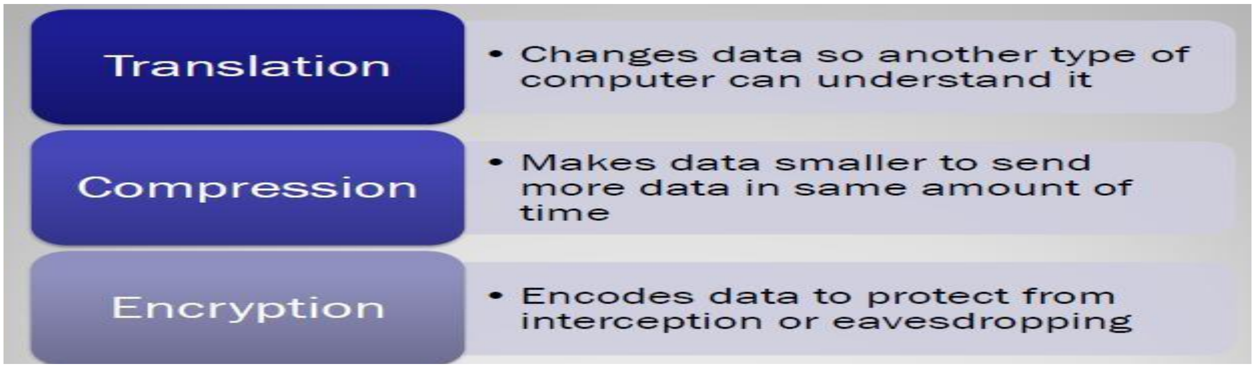
- Takes data from higher levels of OSI Model and breaks it into segments that can be sent to lower-level layers for data transmission
- Conversely, reassembles data segments into data that higher-level protocols and applications can use
- Also puts segments in correct order (called sequencing) so they can be reassembled in correct order at destination
- Concerned with the reliability of the transport of sent data
- May use a *connection-oriented protocol* such as TCP to ensure destination received segments
- May use a *connectionless protocol* such as UDP to send segments without assurance of delivery
- Uses port addressing

Session Layer

- Responsible for managing the dialog between networked devices
- Establishes, manages, and terminates connections
- Provides duplex, half-duplex, or simplex communications between devices
- Provides procedures for establishing checkpoints, adjournment, termination, and restart or recovery procedures

Presentation Layer

- Concerned with how data is presented to the network
- Handles three primary tasks: –Translation , –Compression , –Encryption



Translation
- Changes data so another type of computer can understand it

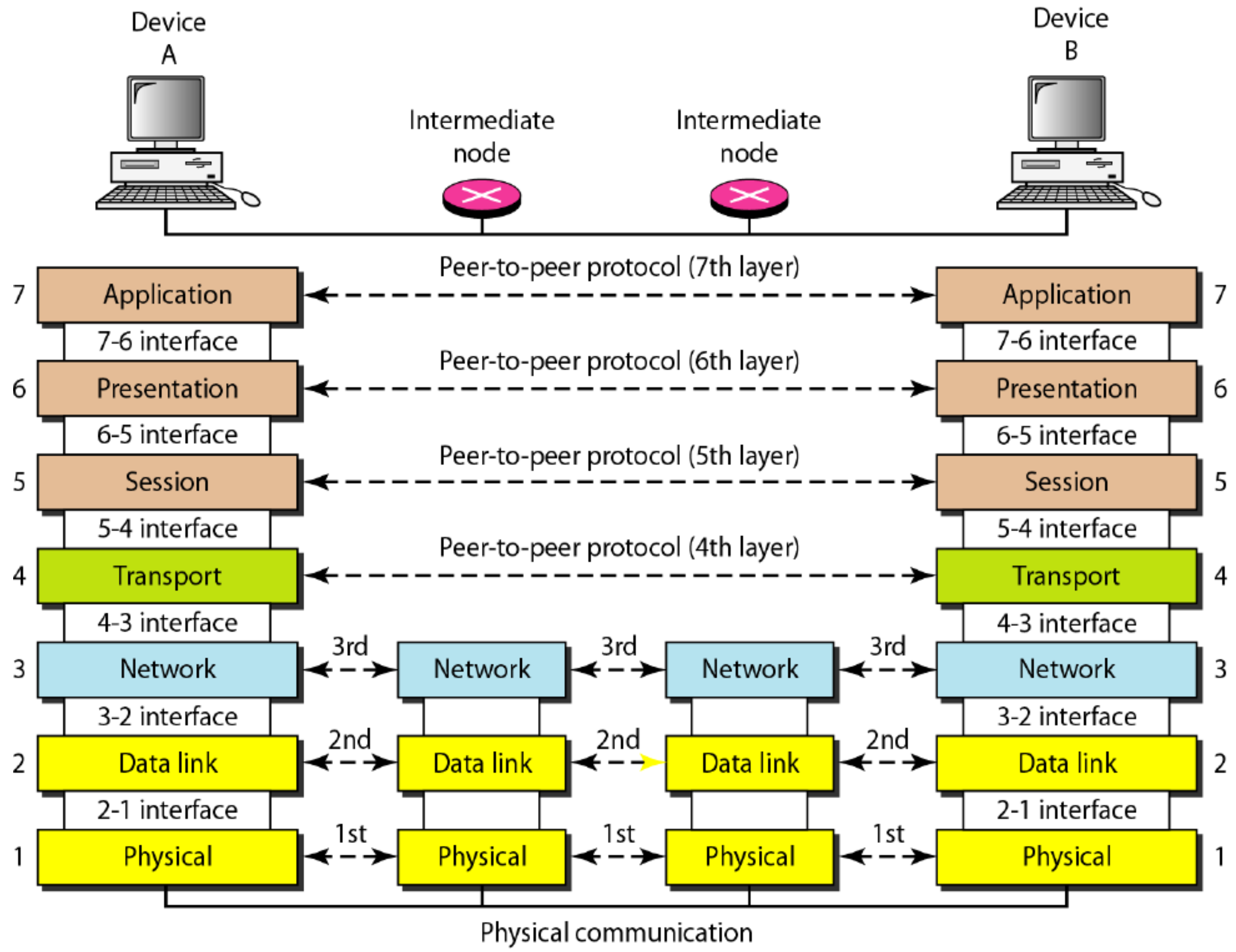
Compression
- Makes data smaller to send more data in same amount of time

Encryption
- Encodes data to protect from interception or eavesdropping

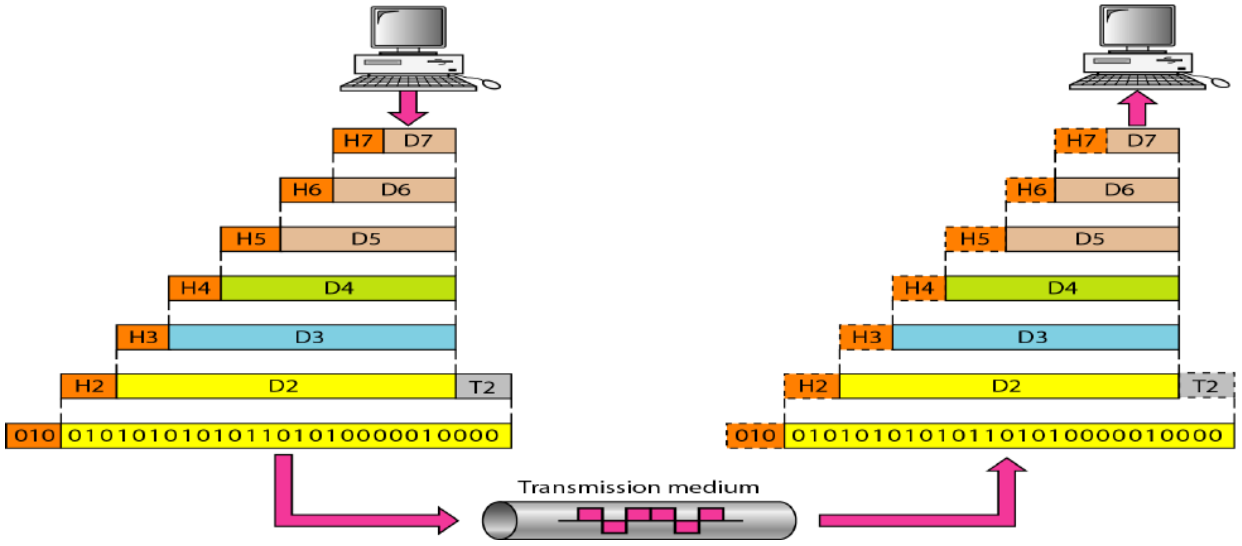
Application Layer

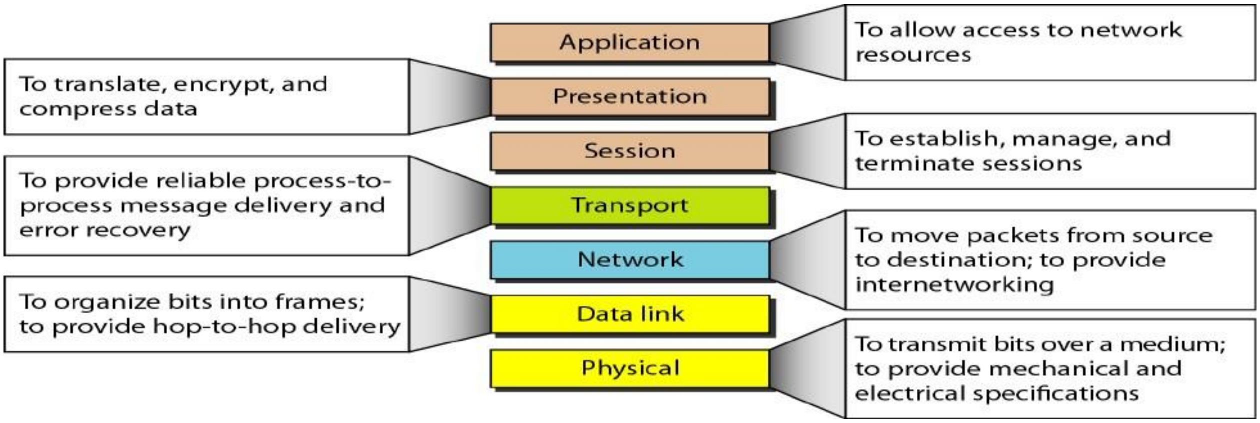
- Contains all services or protocols needed by application software or operating system to communicate on the network
- Examples
  - –Firefox web browser uses HTTP (Hyper-Text Transport Protocol)
  - –E-mail program may use POP3 (Post Office Protocol version 3) to read e-mails and SMTP (Simple Mail Transport Protocol) to send e-mails

The interaction between layers in the OSI model



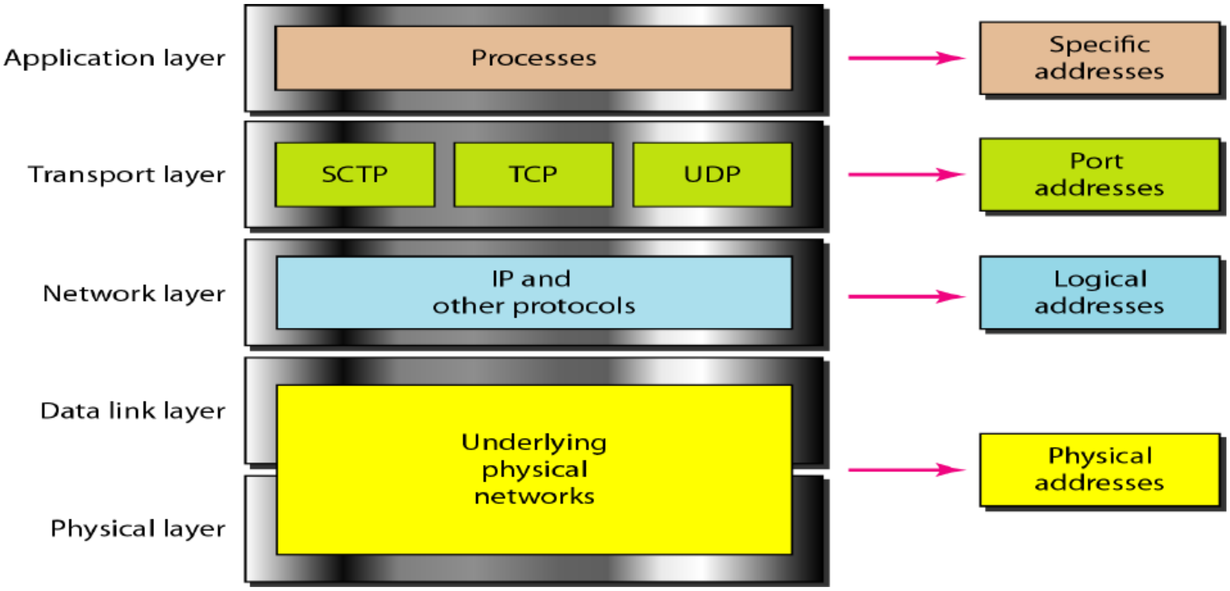
An exchange using the OSI model




Application
To allow access to network resources
To translate, encrypt, and compress data
Presentation
Session
To establish, manage, and terminate sessions
To provide reliable process-to-process message delivery and error recovery
Transport
Network
To move packets from source to destination; to provide internetworking
To organize bits into frames; to provide hop-to-hop delivery
Data link
Physical
To transmit bits over a medium; to provide mechanical and electrical specifications

TCP/IP Model (Transmission Control Protocol/Internet Protocol)

–A *protocol suite* is a large number of related protocols that work together to allow networked computers to communicate



Relationship of layers and addresses in TCP/IP

Application Layer

- Application layer protocols define the rules when implementing specific network applications
- Rely on the underlying layers to provide accurate and efficient data delivery
- Typical protocols:
  - FTP – File Transfer Protocol
    - For file transfer
  - Telnet – Remote terminal protocol
    - For remote login on any other computer on the network
  - SMTP – Simple Mail Transfer Protocol
    - For mail transfer
  - HTTP – Hypertext Transfer Protocol
    - For Web browsing

- Encompasses same functions as these OSI Model layers Application Presentation Session

- **Transport Layer**
- TCP is a connection-oriented protocol
  - Does not mean it has a physical connection between sender and receiver
  - TCP provides the function to allow a connection virtually exists – also called virtual circuit
- UDP provides the functions:
  - Dividing a chunk of data intosegments
  - Reassembly segments into the originalchunk
  - Provide further the functions such as reordering and data resend
- Offering a reliable byte-stream delivery service
- Functions the same as the Transport layer inOSI

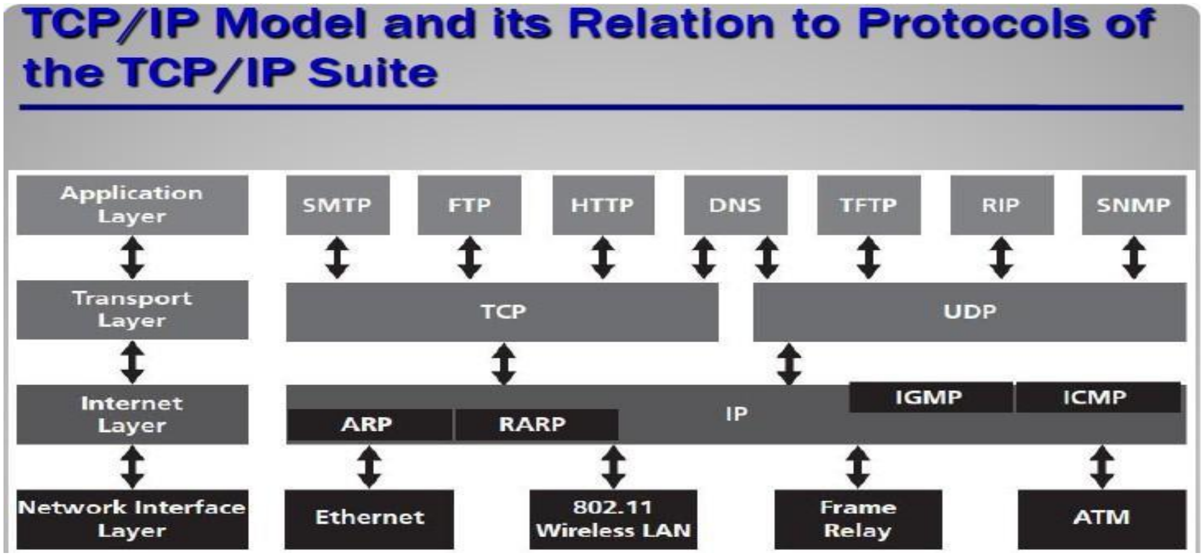
- Synchronize source and destination computers to set up the session between the respective computers

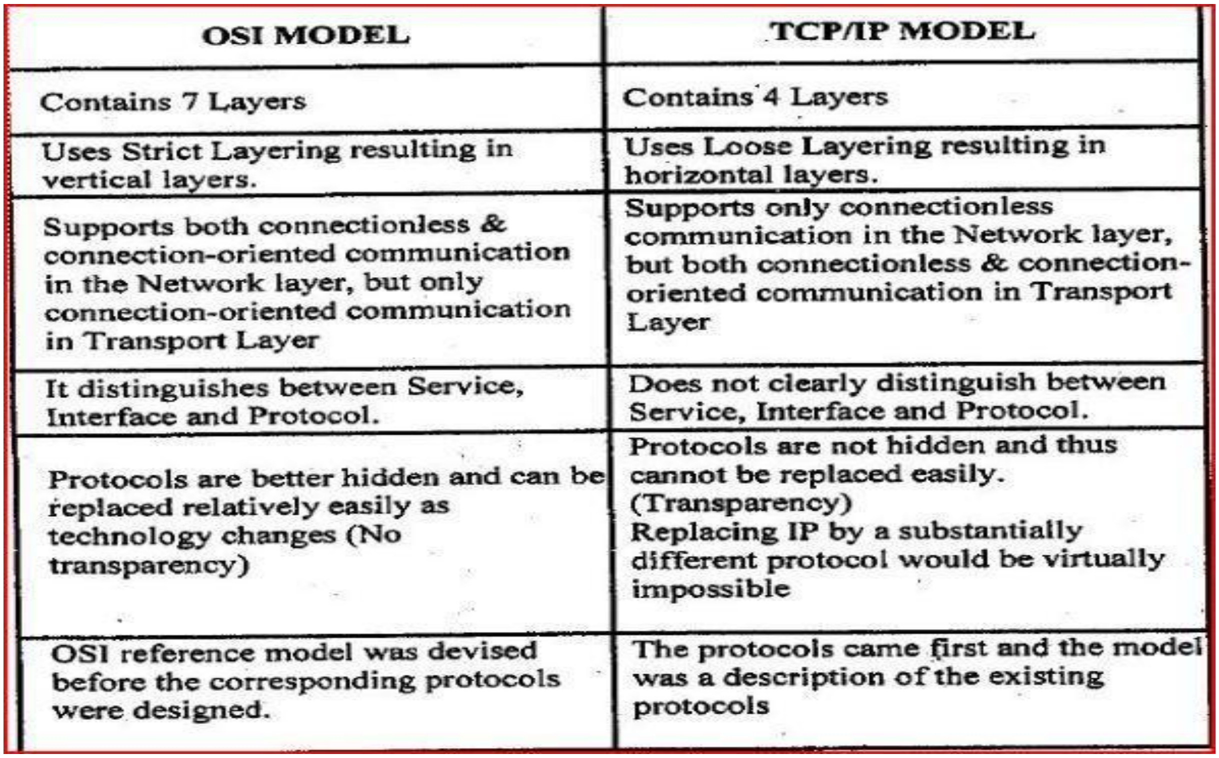
Internet Layer

- The network layer, also called the internet layer, deals with packets and connects independent networks to transport the packets across network boundaries. The network layer protocols are the IP and the Internet Control Message Protocol (ICMP), which is used for error reporting.

Host-to-network layer

The **Host-to-network layer** is the lowest **layer** of the **TCP**/**IP** reference model. It combines the link **layer** and the physical **layer** of the ISO/OSI model. At this **layer**, data is transferred between adjacent **network** nodes in a WAN or between nodes on the same LAN.





OSI MODEL	TCP/IP MODEL
Contains 7 Layers	Contains 4 Layers
Uses Strict Layering resulting in vertical layers.	Uses Loose Layering resulting in horizontal layers.
Supports both connectionless & connection-oriented communication in the Network layer, but only connection-oriented communication in Transport Layer	Supports only connectionless communication in the Network layer, but both connectionless & connection-oriented communication in Transport Layer
It distinguishes between Service, Interface and Protocol.	Does not clearly distinguish between Service, Interface and Protocol.
Protocols are better hidden and can be replaced relatively easily as technology changes (No transparency)	Protocols are not hidden and thus cannot be replaced easily. (Transparency) Replacing IP by a substantially different protocol would be virtually impossible
OSI reference model was devised before the corresponding protocols were designed.	The protocols came first and the model was a description of the existing protocols

THE INTERNET

The Internet has revolutionized many aspects of our daily lives. It has affected the way we do business as well as the way we spend our leisure time. Count the ways you've used the Internet recently. Perhaps you've sent electronic mail (e-mail) to a business associate, paid a utility bill, read a newspaper from a distant city, or looked up a local movie schedule-all by using the Internet. Or maybe you researched a medical topic, booked a hotel reservation, chatted with a fellow Trekkie, or comparison-shopped for a car. The Internet is a communication system that has brought a wealth of information to our fingertips and organized it for ouruse.

A Brief History

A network is a group of connected communicating devices such as computers and printers. An internet (note the lowercase letter i) is two or more networks that can communicate with each other. The most notable internet is called the Internet (uppercase letter I), a collaboration of more than hundreds of thousands of interconnected networks. Private individuals as well as various organizations such as government agencies, schools, research facilities, corporations, and libraries in more than 100 countries use the Internet. Millions of people are users. Yet this extraordinary communication system only came into being in 1969.

In the mid-1960s, mainframe computers in research organizations were standalone devices. Computers from different manufacturers were unable to communicate with one another. The Advanced Research Projects Agency

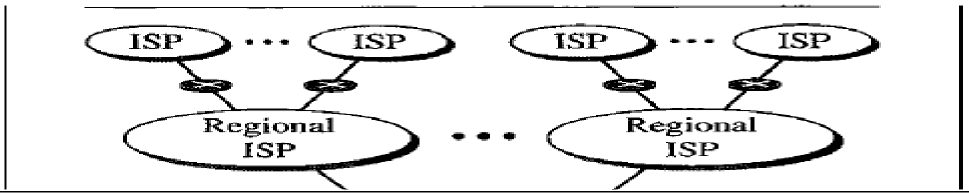
(ARPA) in the Department of Defense (DoD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort.

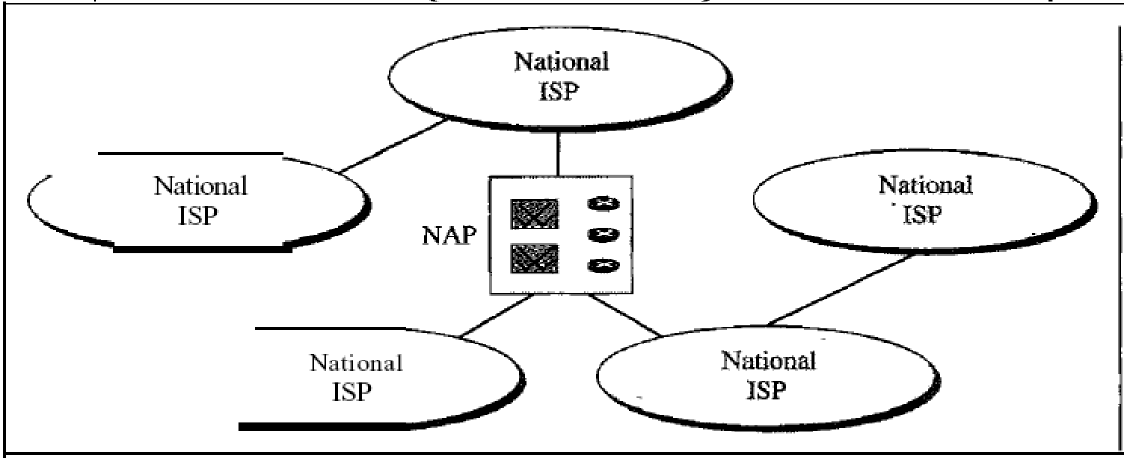
In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for ARPANET, a small network of connected computers. The idea was that each host computer (not necessarily from the same manufacturer) would be attached to a specialized computer, called an *inteiface message processor* (IMP). The IMPs, in tum, would be connected to one another. Each IMP had to be able to communicate with other IMPs as well as with its own attached host. By 1969, ARPANET was a reality. Four nodes, at the University of California at Los Angeles (UCLA), the University of California at Santa Barbara (UCSB), Stanford Research Institute (SRI), and the University of Utah, were connected via the IMPs to form a network. Software called the *Network Control Protocol* (NCP) provided communication between the hosts.

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetting Projec1.* Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end- to-end delivery of packets. This paper on Transmission Control Protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway. Shortly thereafter, authorities made a decision to split TCP into two protocols: Transmission Control Protocol (TCP) and Internetworking Protocol (lP). IP would handle datagram routing while TCP would be responsible for higher-level functions such as segmentation, reassembly, and error detection. The internetworking protocol became known as TCPIIP.

The Internet Today

The Internet has come a long way since the 1960s. The Internet today is not a simple hierarchical structure. It is made up of many wide- and local-area networks joined by connecting devices and switching stations. It is difficult to give an accurate representation of the Internet because it is continually changing-new networks are being added, existing networks are adding addresses, and networks of defunct companies are being removed. Today most end users who want Internet connection use the services of Internet service providers (lSPs). There are international service providers, national service providers, regional service providers, and local service providers. The Internet today is run by private companies, not the government. Figure 1.13 shows a conceptual (not geographic) view of the Internet.





b. Interconnection of national ISPs

International Internet Service Providers:

At the top of the hierarchy are the international service providers that connect nations together.

National Internet Service Providers:

The national Internet service providers are backbone networks created and maintained by specialized companies. There are many national ISPs operating in North America; some of the most well known are SprintLink, PSINet, UUNet Technology, AGIS, and internet Mel. To provide connectivity between the end users, these backbone networks are connected by complex switching stations (normally run by a third party) called network access points (NAPs). Some national ISP networks are also connected to one another by private switching stations called *peering points.* These normally operate at a high data rate (up to 600 Mbps).

Regional Internet Service Providers:

Regional internet service providers or regional ISPs are smaller ISPs that are connected to one or more national ISPs. They are at the third level of the hierarchy with a smaller data rate. ***Local Internet Service Providers:***

Local Internet service providers provide direct service to the end users. The local ISPs can be connected to regional ISPs or directly to national ISPs. Most end users are connected to the local ISPs. Note that in this sense, a local ISP can be a company that just provides Internet services, a corporation with a network that supplies services to its own employees, or a nonprofit organization, such as a college or a university, that runs its own network. Each of these local ISPs can be connected to a regional or national service provider.

TEXT / REFERENCE BOOKS

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SCHOOL OF COMPUTING
DEPARTMENT OF INFORMATION TECHNOLOGY

UNIT – II – Data communication and Computer networks – SCS1314

DATA LINK LAYER

Link layer services - Framing - Flow Control - Error control- Medium Access Control - Ethernet CSMA/CD - Token Ring - FDDI - Token Passing- Wireless LAN - CSMA/CA

1.**Providing services to the network layer:**

1 <u>Unacknowledged connectionless service</u>.

Appropriate for low error rate and real-time traffic. Ex:Ethernet

1. <u>Acknowledged connectionless service</u>.

Useful in unreliable channels, WiFi. Ack/Timer/Resend

2. <u>Acknowledged connection-oriented service</u>.

Guarantee frames are received exactly once and in the right order. Appropriate over long, unreliable links such as a satellite channel or a long- distance telephone circuit

2. **Framing:** Frames are the streams of bits received from the network layer into manageable data units. This division of stream of bits is done by Data Link Layer.
3. **Physical Addressing**: The Data Link layer adds a header to the frame in order to define physical address of the sender or receiver of the frame, if the frames are to be distributed to different systems on the network.
4. **Flow Control:** A receiving node can receive the frames at a faster rate than it can process the frame. Without flow control, the receiver's buffer can overflow, and frames can get lost. To overcome this problem, the data link layer uses the flow control to prevent the sending node on one side of the link from overwhelming the receiving node on another side of the link. This prevents traffic jam at the receiver side.
5. **Error Control:** Error control is achieved by adding a trailer at the end of the frame. Duplication of frames are also prevented by using this mechanism. Data Link Layers adds mechanism to prevent duplication of frames.

   Error detection: Errors can be introduced by signal attenuation and noise. Data Link Layer protocol provides a mechanism to detect one or more errors. This is achieved by adding error detection bits in the frame and then receiving node can perform an error check.

   Error correction: Error correction is similar to the Error detection, except that receiving node not only detects the errors but also determine where the errors have occurred in the frame.
6. **Access Control**: Protocols of this layer determine which of the devices has control over the link at any given time, when two or more devices are connected to the same link**.**
7. **Reliable delivery**: Data Link Layer provides a reliable delivery service, i.e., transmits the network layer datagram without any error. A reliable

delivery service is accomplished with transmissions and acknowledgements. A data link layer mainly provides the reliable delivery

service over the links as they have higher error rates and they can be corrected locally, link at which an error occurs rather than forcing to retransmit the data.

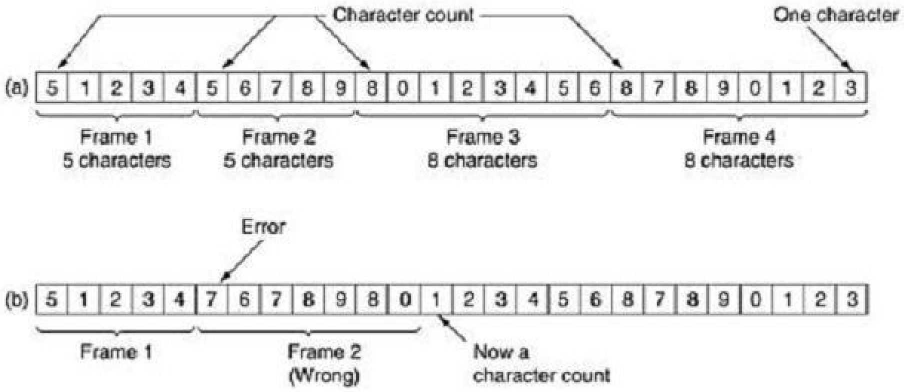
8. **Half-Duplex & Full-Duplex**: In a Full-Duplex mode, both the nodes can transmit the data at the same time. In a Half-Duplex mode, only one node can transmit the data at the same time.

FRAMING:

To provide service to the network layer, the data link layer must use the service provided to it by the physical layer. What the physical layer does is accept a raw bit stream and attempt to deliver it to the destination. This bit stream is not guaranteed to be error free. The number of bits received may be less than, equal to, or more than the number of bits transmitted, and they may have different values. It is up to the data link layer to **detect and, if necessary, correct errors**. The usual approach is for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame **(framing)**. When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it (e.g., discarding the bad frame and possibly also sending back an error report).We will look at four framing methods:

1. Character count.
2. Flag bytes with byte stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

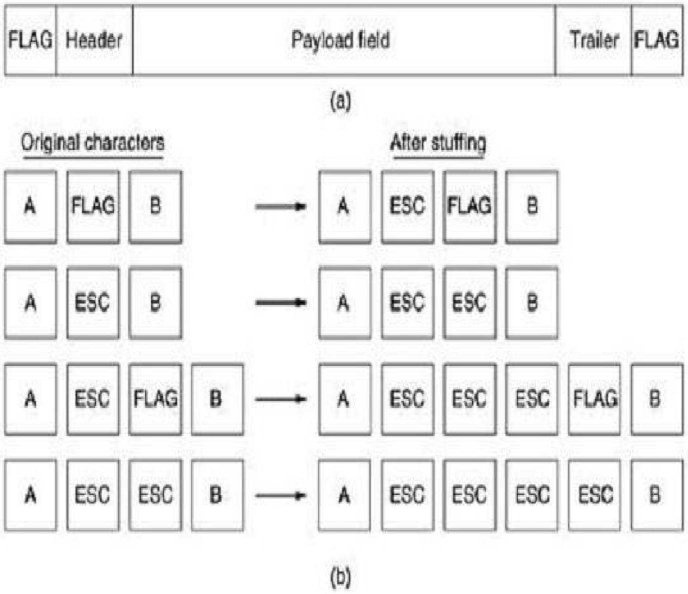
Character count method uses a field in the header to specify the number of characters in the frame. When the data link layer at the destination sees the character count, it knows how many characters follow and hence where the end of the frame is. This technique is shown in Fig. (a) For four frames of sizes 5, 5, 8, and 8 characters, respectively.



A character stream. (a) Without errors. (b) With one error

The trouble with this algorithm is that the count can be garbled by a transmission error. For example, if the character count of 5 in the second frame of Fig. (b) becomes a 7, the destination will get out of synchronization and will be unable to locate the start of the next frame. Even if the checksum is incorrect so the destination knows that the frame is bad, it still has no way of telling where the next frame starts. Sending a frame back to the source asking for a retransmission does not help either, since the destination does not know how many characters to skip over to get to the start of the retransmission. For this reason, the character count method is rarely used anymore.

Flag bytes with byte stuffing method gets around the problem of resynchronization after an error by having each frame start and end with special bytes. In the past, the starting and ending bytes were different, but in recent years most protocols have used the same byte, called a flag byte, as both the starting and ending delimiter, as shown in Fig. (a) as FLAG. In this way, if the receiver ever loses synchronization, it can just search for the flag byte to find the end of the current frame. Two consecutive flag bytes indicate the end of one frame and start of the next one.



(a) A frame delimited by flag bytes (b) Four examples of byte sequences before and after byte stuffing

It may easily happen that the flag byte's bit pattern occurs in the data. This situation will usually interfere with the framing. One way to solve this problem is to have the sender's data link layer insert a special escape byte (ESC) just before each "accidental" flag byte in the data. The data link layer on the receiving end removes the escape byte before the data are given to the network layer. This technique is called byte stuffing or character stuffing.

Thus, a framing flag byte can be distinguished from one in the data by the absence or presence of an escape byte before it.

What happens if an escape byte occurs in the middle of the data? The answer is that, it too is stuffed with an escape byte. Thus, any single escape byte is part of an escape sequence, whereas a doubled one indicates that a single escape occurred naturally in the data. Some examples are shown in Fig. (b). In all cases, the byte sequence delivered after de stuffing is exactly the same as the original byte sequence.

A major disadvantage of using this framing method is that it is closely tied to the use of 8-bit characters. Not all character codes use 8-bit characters. For example UNICODE uses 16-bit characters, so a new technique had to be developed to allow arbitrary sized characters

Starting and ending flags, with bit stuffing allows data frames to contain an arbitrary number of bits and allows character codes with an arbitrary number of bits per character. It works like this. Each frame begins and ends with a special bit pattern, 01111110 (in fact, a flag byte). Whenever the sender's data link layer encounters five consecutive 1s in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This bit stuffing is analogous to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically de- stuffs (i.e., deletes) the 0 bit. Just as byte stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contain the flag pattern, 01111110, this flag is transmitted as 011111010 but stored in the receiver's memory as 01111110.

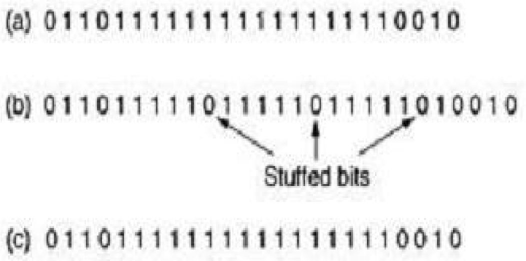
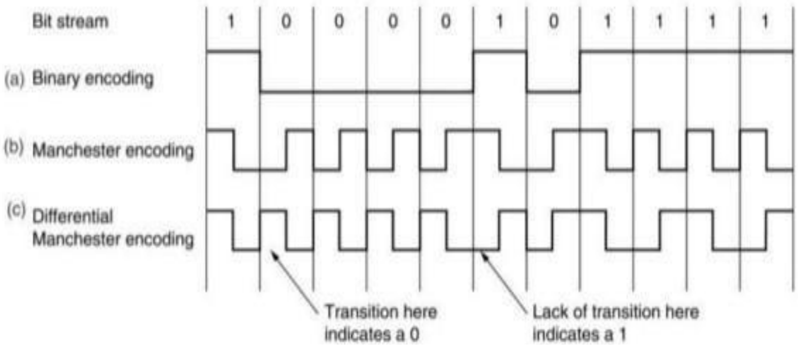

(a) 011011111111111111110010
(b) 011011111011111011111010010
Stuffed bits
(c) 011011111111111111110010

Fig:Bit stuffing. (a) The original data. (b) The data as they appear on the line. (c) The data as they are stored in the receiver's memory after destuffing.

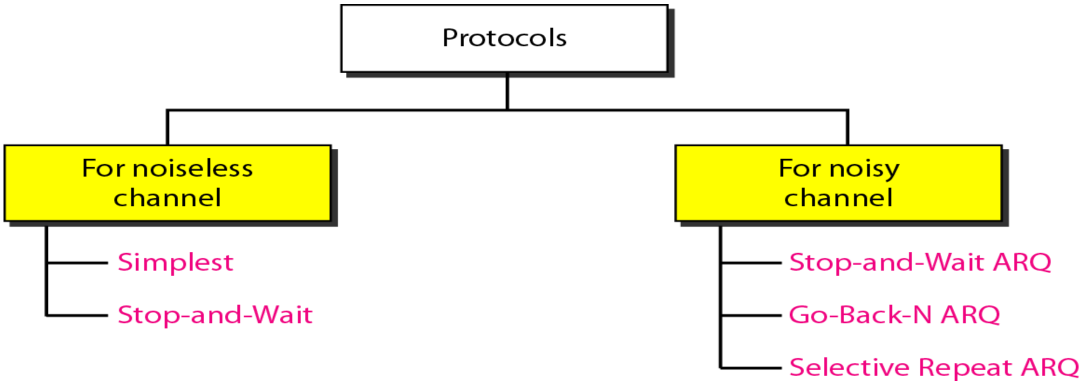
With bit stuffing, the boundary between two frames can be unambiguously recognized by the flag pattern. Thus, if the receiver loses track of where it is, all it has to do is scan the input for flag sequences, since they can only occur at frame boundaries and never within the data.

Physical layer coding violations method of framing is only applicable to networks in which the encoding on the physical medium contains some redundancy. For example, some LANs encode 1 bit of data by using 2 physical bits. Normally, a 1 bit is a high-low pair and a 0 bit is a low-high pair. The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. The combinations high-

high and low-low are not used for data but are used for delimiting frames in some protocols.

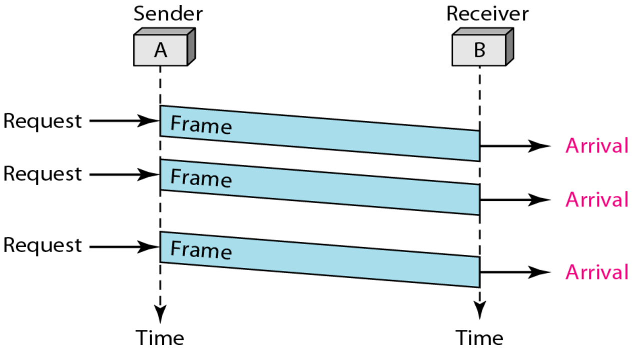

Bit stream
1 0 0 0 0 1 0 1 1 1 1
(a) Binary encoding
(b) Manchester encoding
(c) Differential Manchester encoding
Transition here indicates a 0
Lack of transition here indicates a 1

As a final note on framing, many data link protocols use combination of a character count with one of the other methods for extra safety. When a frame arrives, the count field is used to locate the end of the frame. Only if the appropriate delimiter is present at that position and the checksum is correct is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter



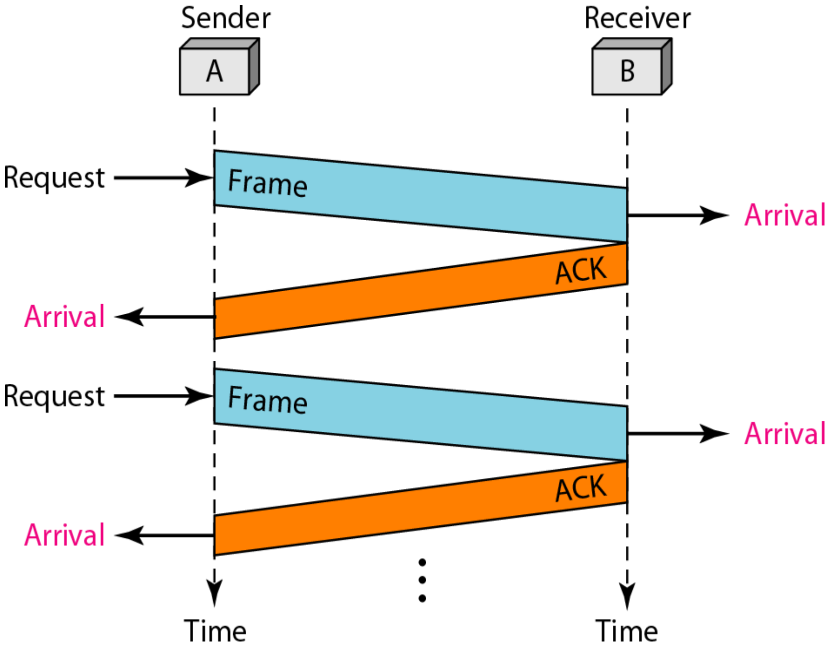
ELEMENTARY DATA LINK PROTOCOLS

Simplest Protocol



It is very simple. The sender sends a sequence of frames without even thinking about the receiver. Data are transmitted in one direction only. Both sender & receiver always ready. Processing time can be ignored. Infinite buffer space is available. And best of all, the communication channel between the data link layers never damages or loses frames. This thoroughly unrealistic protocol, which we will nickname ‘‘Utopia,’’ .The utopia protocol is unrealistic because **it does not handle either flow control or error correction**

Stop-and-wait Protocol



It is still very simple. The sender sends one frame and waits for feedback from the receiver. When the ACK arrives, the sender sends the next frame

It is Stop-and-Wait Protocol because the sender sends one frame, stops until it receives confirmation from the receiver (okay to go ahead), and then sends the next frame. We still have unidirectional communication for data frames, but auxiliary ACK frames (simple tokens of acknowledgment) travel from the other direction. We add flow control to our previous protocol.

NOISY CHANNELS

Although the Stop-and-Wait Protocol gives us an idea of how to add flow control to its predecessor, noiseless channels are nonexistent. We can ignore the error (as we sometimes do), or we need to add error control to our protocols. We discuss three protocols in this section that use error control.

Sliding Window Protocols:

1 Stop-and-Wait Automatic Repeat Request

2 Go-Back-N Automatic Repeat Request

3 Selective Repeat Automatic Repeat Request

1 Stop-and-Wait Automatic Repeat Request

To detect and correct corrupted frames, we need to add redundancy bits to our data frame. When the frame arrives at the receiver site, it is checked and if it is corrupted, it is silently discarded. The detection of errors in this protocol is manifested by the silence of the receiver.

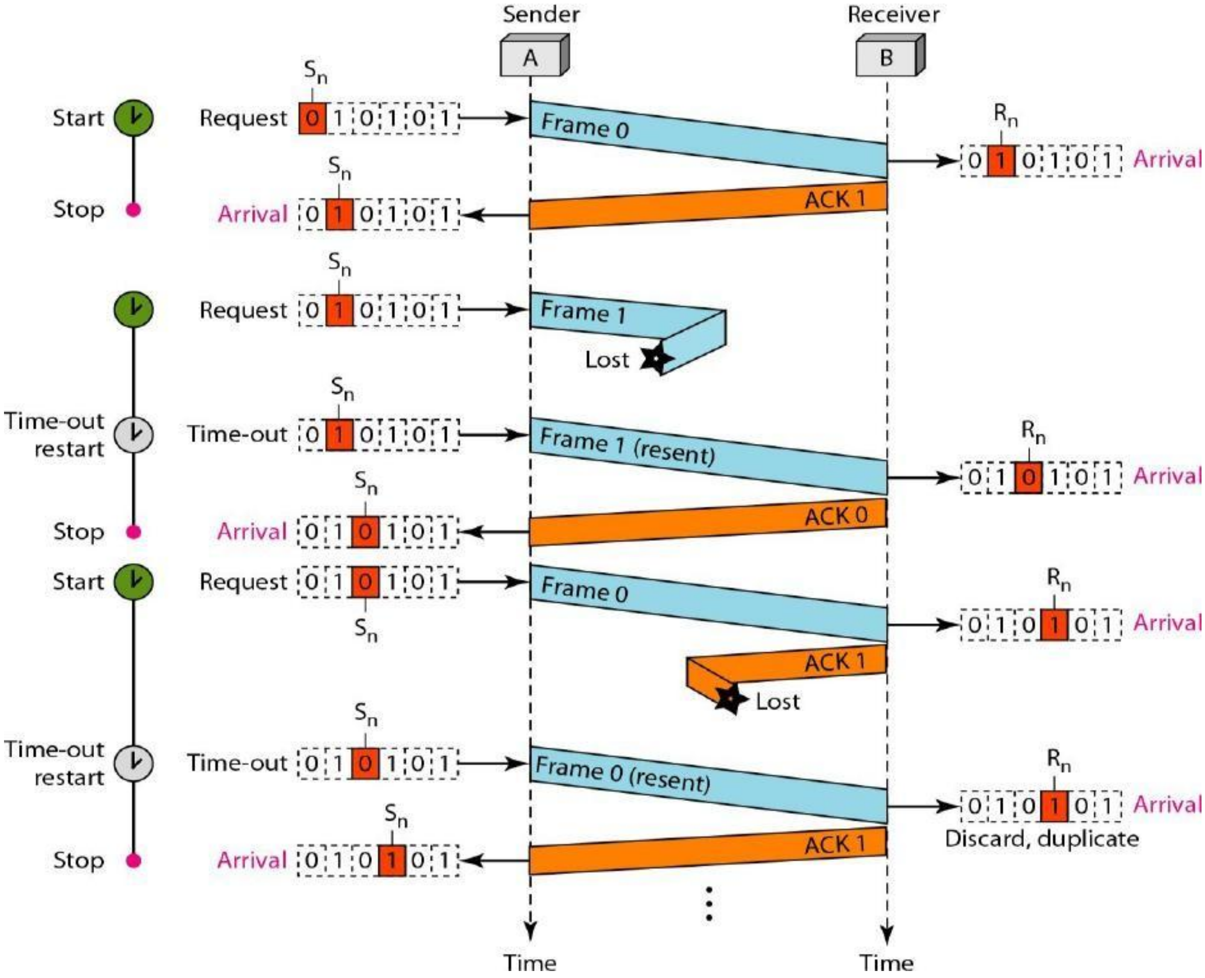
Lost frames are more difficult to handle than corrupted ones. In our previous protocols, there was no way to identify a frame. The received frame could be the correct one, or a duplicate, or a frame out of order. The solution is to number the frames. When the receiver receives a data frame that is out of order, this means that frames were either lost or duplicated

The lost frames need to be resent in this protocol. If the receiver does not respond when there is an error, how can the sender know which frame to resend? To remedy this problem, the sender keeps a copy of the sent frame. At the same time, it starts a timer. If the timer expires and there is no ACK for the sent frame, the frame is resent, the copy is held, and the timer is restarted. Since the protocol uses the stop-and-wait mechanism, there is only one specific frame that needs an ACK

Error correction in Stop-and-Wait ARQ is done by keeping a copy of the sent frame and retransmitting of the frame when the timer expires

In Stop-and-Wait ARQ, we use sequence numbers to number the frames. The sequence numbers are based on modulo-2 arithmetic.

In Stop-and-Wait ARQ, the acknowledgment number always announces in modulo-2 arithmetic the sequence number of the next frame expected.



Bandwidth Delay Product:

Assume that, in a Stop-and-Wait ARQ system, the bandwidth of the line is 1 Mbps, and 1 bit takes 20 ms to make a round trip. What is the bandwidth-delay product? If the system data frames are 1000 bits in length, what is the utilization percentage of the link?

$$\boxed{(1 \times 10^6) \times (20 \times 10^{-3}) = 20{,}000 \text{ bits}}$$

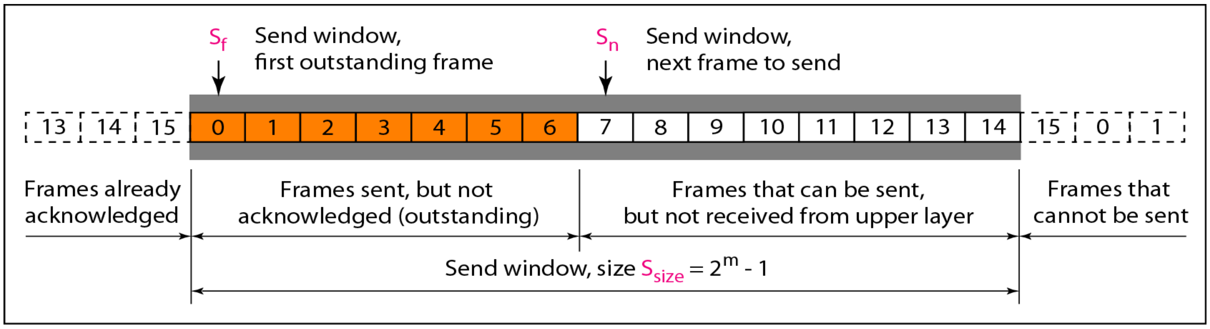
The link utilization is only 1000/20,000, or 5 percent. For this reason, for a link with a high bandwidth or long delay, the use of Stop-and-Wait ARQ wastes the capacity of the link.

2. Go-Back-N Automatic Repeat Request

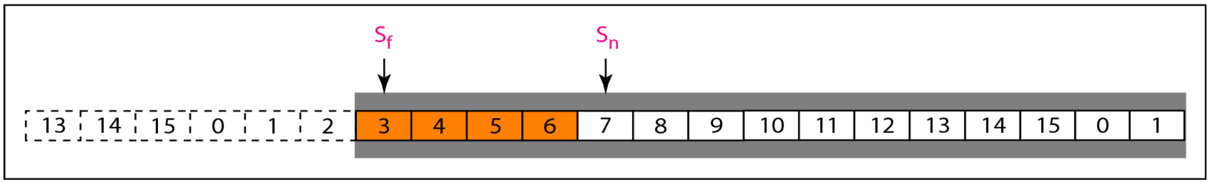
To improve the efficiency of transmission (filling the pipe), multiple frames must be in transition while waiting for acknowledgment. In other words, we need to let more than one frame be outstanding to keep the channel busy while the sender is waiting for acknowledgment.

The first is called Go-Back-N Automatic Repeat. In this protocol we can send several frames before receiving acknowledgments; we keep a copy of these frames until the acknowledgments arrive.

In the Go-Back-N Protocol, the sequence numbers are modulo 2^m, where m is the size of the sequence number field in bits. The sequence numbers range from 0 to *2 power m*- 1. For example, if *m* is 4, the only sequence numbers are 0 through 15inclusive.



a. Send window before sliding



b. Send window after sliding

The **sender window** at any time divides the possible sequence numbers into four regions.

The first region, from the far left to the left wall of the window, defines the sequence numbers belonging to frames that are already acknowledged. The sender does not worry about these frames and keeps no copies of them.

The second region, colored in Figure (a), defines the range of sequence numbers belonging to the frames that are sent and have an unknown status. The sender needs to wait to find out if these frames have been received or were lost. We call these outstandingframes.

The third range, white in the figure, defines the range of sequence numbers for frames that can be sent; however, the corresponding data packets have not yet been received from the network layer.

Finally, the fourth region defines sequence numbers that cannot be used until the window slides

The send window is an abstract concept defining an imaginary box of size $2^m - 1$ with three variables: S_f, S_n, and S_{size}. The variable *Sf* defines the sequence number of the first (oldest) outstanding frame. The variable *Sn* holds the sequence number that will be assigned to the next frame to be sent. Finally, the variable Ssize defines the size of the window.

Figure (b) shows how a send window can slide one or more slots to the right when an acknowledgment arrives from the other end. The acknowledgments in this protocol are cumulative, meaning that more than one frame can be acknowledged by an ACK frame. In Figure, frames 0, I, and 2 are acknowledged, so the window has slide to the right three slots. Note that the value of *Sf* is 3 because frame 3 is now the first outstanding frame.**The send window can slide one or more slots when a valid acknowledgment arrives.**

<u>Receiver window:</u> variable *Rn* (receive window, next frame expected) .

The sequence numbers to the left of the window belong to the frames already received and acknowledged; the sequence numbers to the right of this window define the frames that cannot be received. Any received frame with a sequence number in these two regions is discarded. Only a frame with a sequence number matching the value of *Rn* is accepted and acknowledged. The receive window also slides, but only one slot at a time. When a correct frame is received (and a frame is received only one at a time), the window slides.(see below figure for receiving window)

The receive window is an abstract concept defining an imaginary box of size 1 with one single variable Rn. The window slides when a correct frame has arrived; sliding occurs one slot at a time

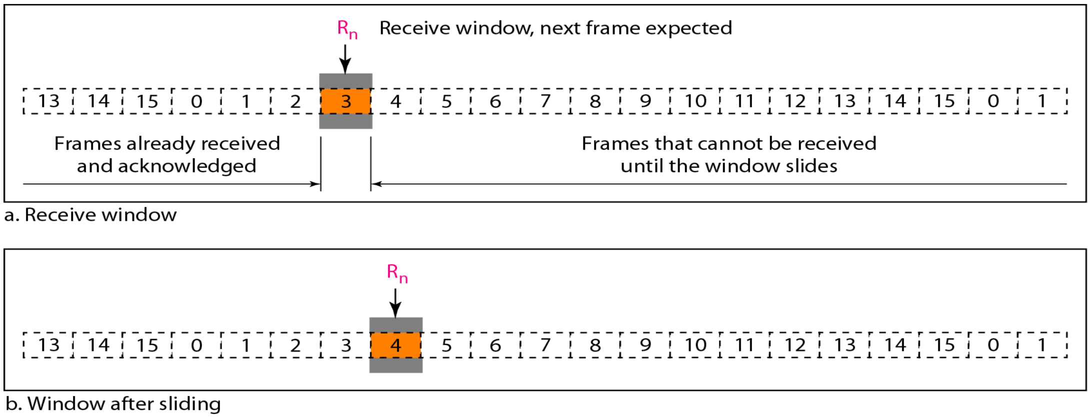


Fig: Receiver window (before sliding (a), After sliding (b))

Timers

Although there can be a timer for each frame that is sent, in our protocol we use only one. The reason is that the timer for the first outstanding frame always expires first; we send all outstanding frames when this timer expires.

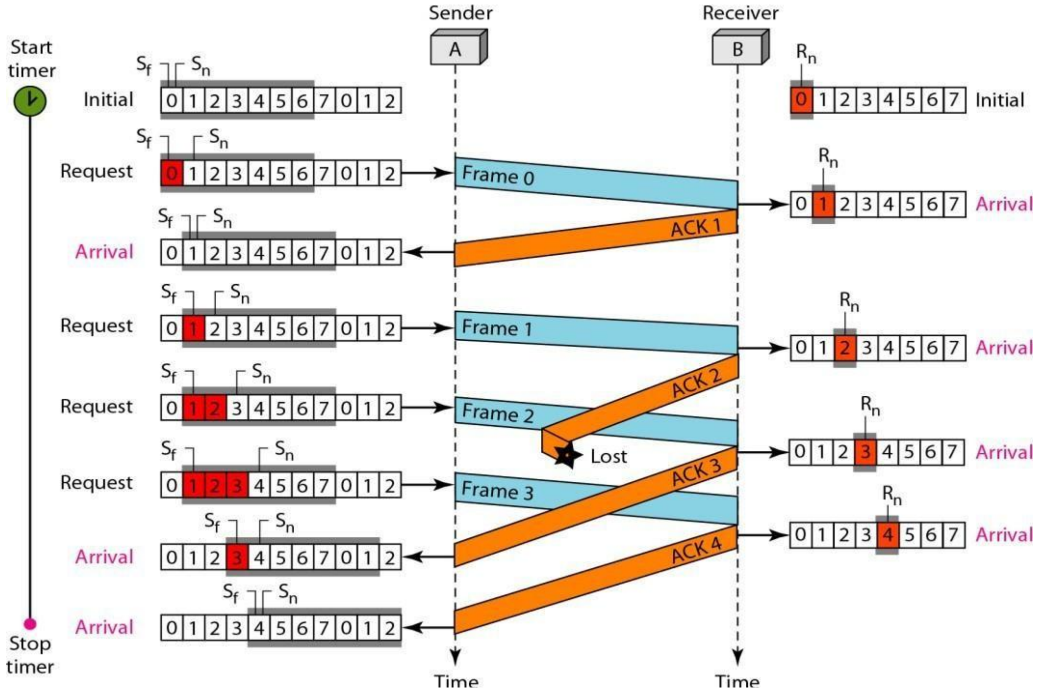
Acknowledgment

The receiver sends a positive acknowledgment if a frame has arrived safe and sound and in order. If a frame is damaged or is received out of order, the receiver is silent and will discard all subsequent frames until it receives the one it is expecting. The silence of the receiver causes the timer of the unacknowledged frame at the sender side to expire. This, in turn, causes the sender to go back and resend all frames, beginning with the one with the expired timer. The receiver does not have to acknowledge each frame received. It can send one cumulative acknowledgment for several frames.

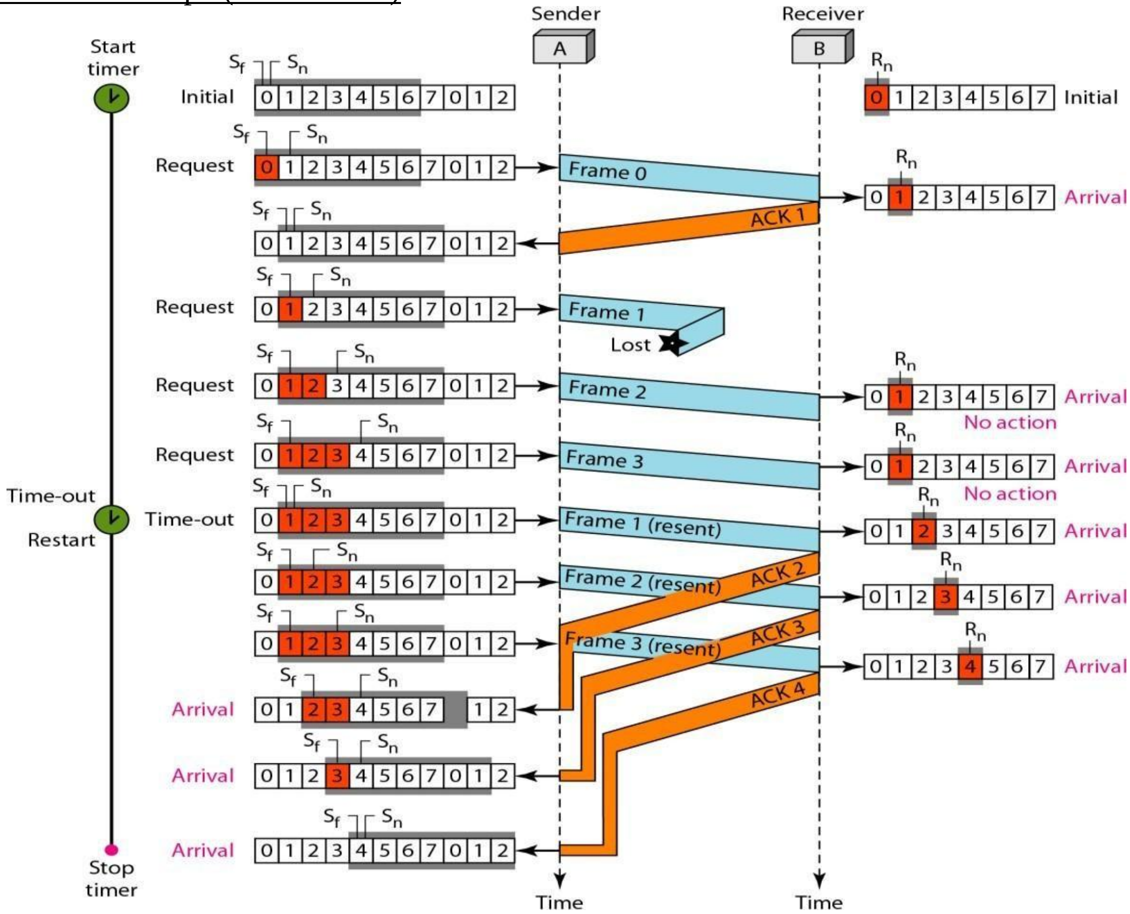
Resending a Frame

When the timer expires, the sender resends all outstanding frames. For example, suppose the sender has already sent frame 6, but the timer for frame 3 expires. This means that frame 3 has not been acknowledged; the sender goes back and sends frames 3,4,5, and 6 again. That is why the protocol is called *Go-Back-N* ARQ.

Below figure is an example(if ack lost) of a case where the forward channel is reliable, but the reverse is not. No data frames are lost, but some ACKs are delayed and one is lost. The example also shows how cumulative acknowledgments can help if acknowledgments are delayed or lost



Below figure is an example(if frame lost)



Stop-and-Wait ARQ is a special case of Go-Back-N ARQ in which the size of the send window is 1.

3 Selective Repeat Automatic Repeat Request

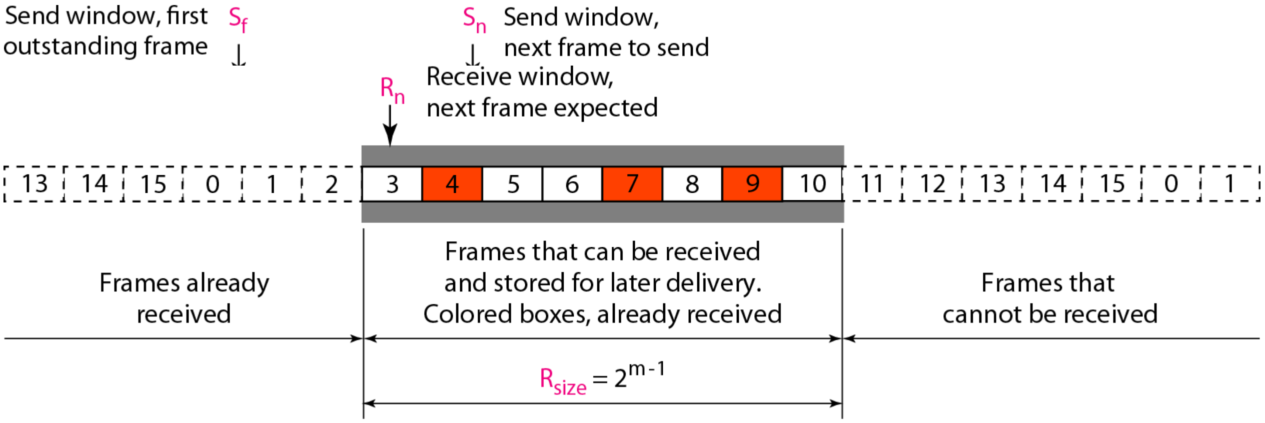
In Go-Back-N ARQ, The receiver keeps track of only one variable, and there is no need to buffer out-of- order frames; they are simply discarded. However, this protocol is very inefficient for a noisy link.

In a noisy link a frame has a higher probability of damage, which means the resending of multiple frames. This resending uses up the bandwidth and slows down the transmission.

For noisy links, there is another mechanism that does not resend *N* frames when just one frame is damaged; only the damaged frame is resent. This mechanism is called Selective Repeat ARQ.

It is more efficient for noisy links, but the processing at the receiver is more complex.

Sender Window (explain go-back N sender window concept (before & after sliding.) The only difference in sender window between Go-back N and Selective Repeat is Window size)


Send window, first outstanding frame
S_f
S_n
Send window, next frame to send
R_n
Receive window, next frame expected
13 14 15 0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 0 1
Frames already received
Frames that can be received and stored for later delivery. Colored boxes, already received
Frames that cannot be received
$R_{size} = 2^{m-1}$

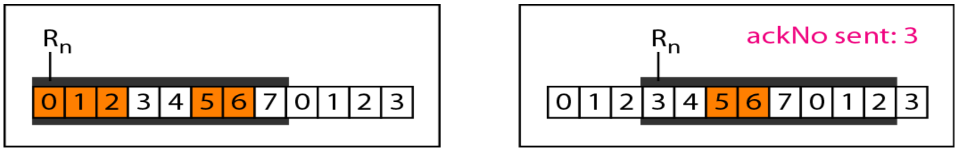
Receiver window

The receiver window in Selective Repeat is totally different from the one in Go Back-N. First, the size of the receive window is the same as the size of the send window *(2^{m-1}).*

The Selective Repeat Protocol allows as many frames as the size of the receiver window to arrive out of order and be kept until there is a set of in-order frames to be delivered to the network layer. Because the sizes of the send window and receive window are the same, all the frames in the send frame can arrive out of order and be stored until they can be delivered. However the receiver never delivers packets out of order to the network layer. Above Figure shows the receive window. Those slots inside the window that are colored define frames that have arrived out of order and are waiting for their neighbors to arrive before delivery to the network layer.

In Selective Repeat ARQ, the size of the sender and receiver window must be at most one-half of 2^m

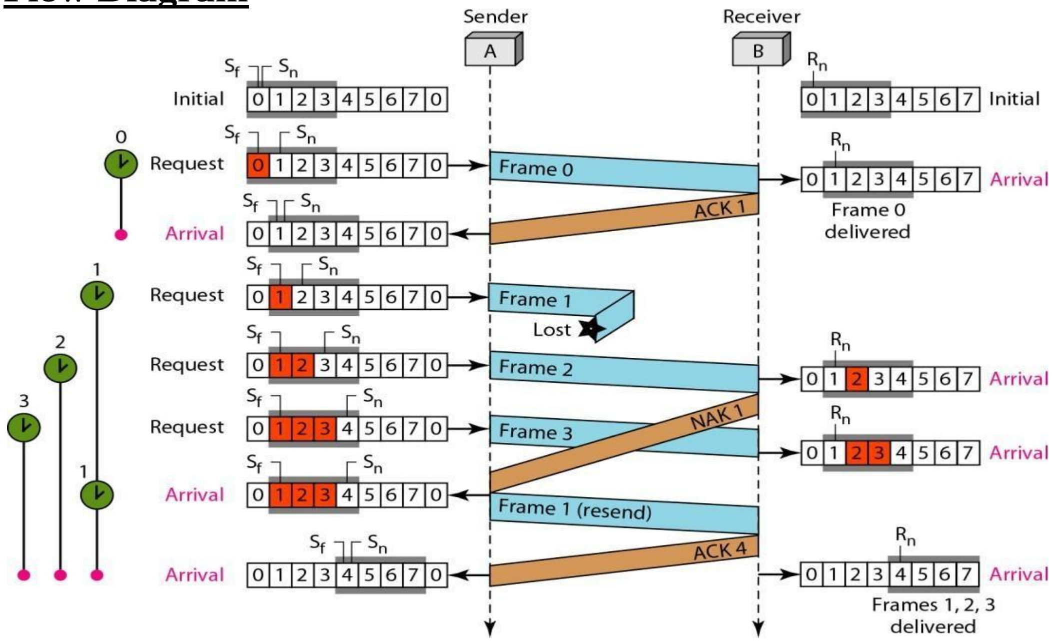
Delivery of Data in Selective Repeat ARQ:



a. Before delivery

b. After delivery

Flow Diagram



Differences between Go-Back N & Selective Repeat

One main difference is the number of timers. Here, each frame sent or resent needs a timer, which means that the timers need to be numbered (0, 1,2, and 3). The timer for frame 0 starts at the first request, but stops when the ACK for this frame arrives.

There are two conditions for the delivery of frames to the network layer: First, a set of consecutive frames must have arrived. Second, the set starts from the beginning of the window. After the first arrival, there was only one frame and it started from the beginning of the window. After the last arrival, there are three frames and the first one starts from the beginning of the window.

Another important point is that a NAK is sent.

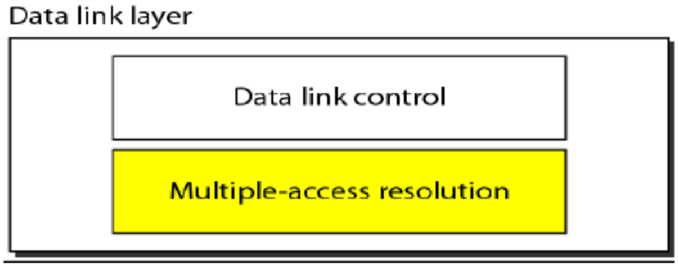
The next point is about the ACKs. Notice that only two ACKs are sent here. The first one acknowledges only the first frame; the second one acknowledges three frames. In Selective Repeat, ACKs are sent when data are delivered to the network layer. If the data belonging to *n* frames are delivered in one shot, only one ACK is sent for all of them.

Piggybacking

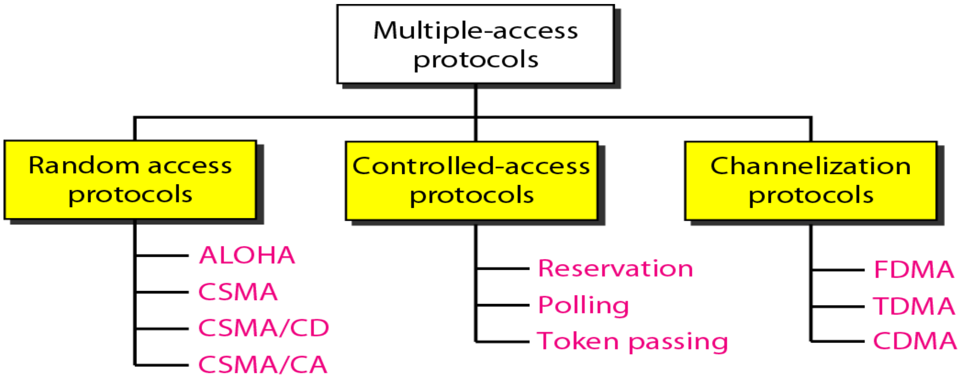
A technique called **piggybacking** is used to improve the efficiency of the bidirectional protocols. When a frame is carrying data from A to B, it can also carry control information about arrived (or lost) frames from B; when a frame is carrying data from B to A, it can also carry control information about the arrived (or lost) frames from A.

RANDOM ACCESS PROTOCOLS

We can consider the data link layer as two sub layers. The upper sub layer is responsible for data link control, and the lower sub layer is responsible for resolving access to the shared media



The upper sub layer that is responsible for flow and error control is called the logical link control (LLC) layer; the lower sub layer that is mostly responsible for multiple access resolution is called the media access control (MAC) layer. When nodes or stations are connected and use a common link, called a multipoint or broadcast link, we need a multiple-access protocol to coordinate access to the link.



Taxonomy of multiple-access protocols

RANDOM ACCESS

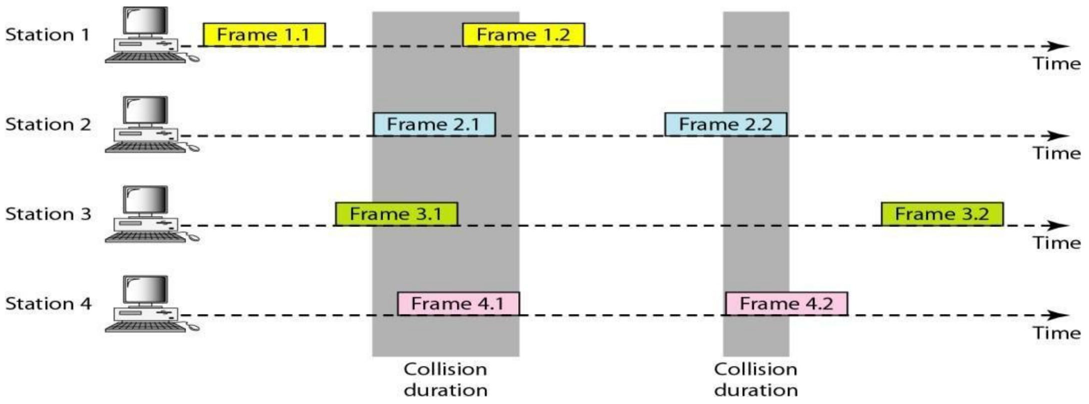
In random access or contention methods, no station is superior to another station and none is assigned the control over another.

Two features give this method its name. First, there is no scheduled time for a station to transmit. Transmission is random among the stations. That is why these methods are called *random access*. Second, no rules specify which station should send next. Stations compete with one another to access the medium. That is why these methods are also called *contention* methods.

ALOHA

1 Pure ALOHA

The original ALOHA protocol is called pure ALOHA. This is a simple, but elegant protocol. The idea is that each station sends a frame whenever it has a frame to send. However, since there is only one channel to share, there is the possibility of collision between frames from different stations. Below Figure shows an example of frame collisions in pure ALOHA.



Frames in a pure ALOHA network

In pure ALOHA, the stations transmit frames whenever they have data to send.

□ When two or more stations transmit simultaneously, there is collision and the frames are destroyed.

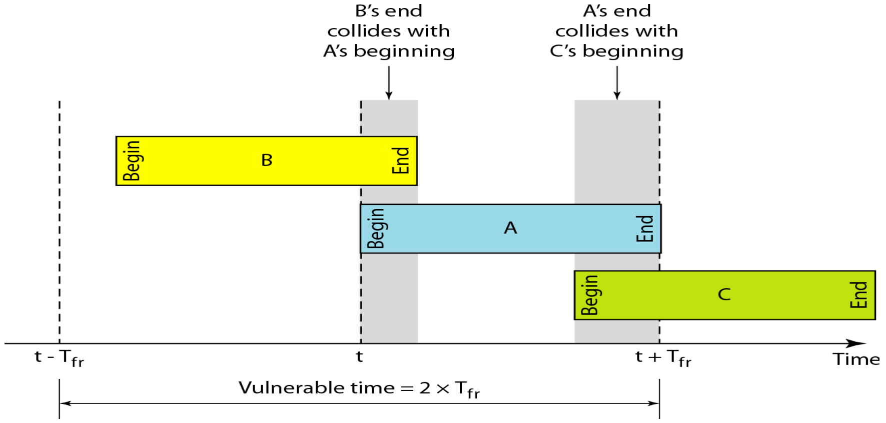
□ In pure ALOHA, whenever any station transmits a frame, it expects the acknowledgement from the receiver.

□ If acknowledgement is not received within specified time, the station assumes that the frame (or acknowledgement) has been destroyed.

□ If the frame is destroyed because of collision the station waits for a random amount of time and sends it again. This waiting time must be random otherwise same frames will collide again and again.

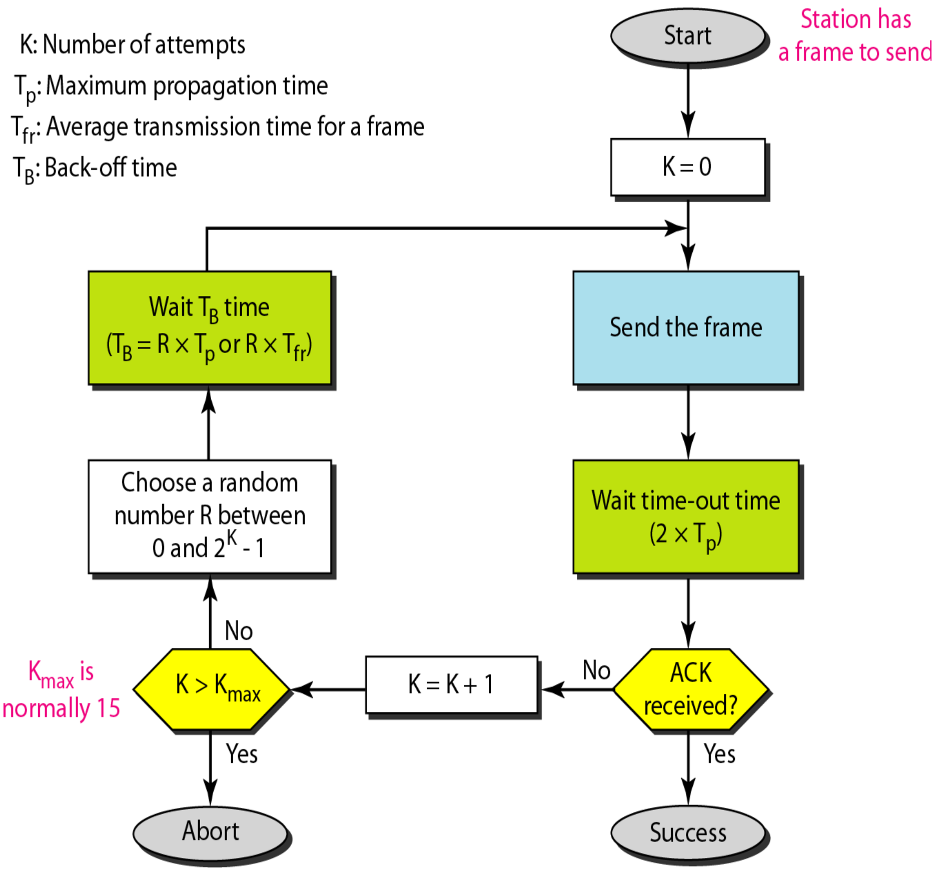
□ Therefore pure ALOHA dictates that when time-out period passes, each station must wait for a random amount of time before resending its frame. This randomness will help avoid more collisions.

Vulnerable time Let us find the length of time, the **vulnerable time,** in which there is a possibility of collision. We assume that the stations send fixed- length frames with each frame taking *Tfr* S to send. Below Figure shows the vulnerable time for station A.


B's end collides with A's beginning
A's end collides with C's beginning
Begin
B
End
Begin
A
End
Begin
C
End
$t - T_{fr}$
t
$t + T_{fr}$
Time
Vulnerable time = $2 \times T_{fr}$

Station A sends a frame at time *t.* Now imagine station B has already sent a frame between *t* - *T*fr and *t.* This leads to a collision between the frames from station A and station B. The end of B's frame collides with the beginning of A's frame. On the other hand, suppose that station C sends a frame between *t* and *t* + *Tfr* . Here, there is a collision between frames from station A and station C. The beginning of C's frame collides with the end of A's frame

Looking at Figure, we see that the vulnerable time, during which a collision may occur in pure ALOHA, is 2 times the frame transmission time. Pure ALOHA vulnerable time = 2 xTfr


K: Number of attempts
T_p: Maximum propagation time
T_{fr}: Average transmission time for a frame
T_B: Back-off time
Start
Station has a frame to send
K = 0
Wait T_B time ($T_B = R \times T_p$ or $R \times T_{fr}$)
Send the frame
Choose a random number R between 0 and $2^K - 1$
Wait time-out time ($2 \times T_p$)
No
$K > K_{max}$
K_{max} is normally 15
K = K + 1
No
ACK received?
Yes
Yes
Abort
Success

Procedure for pure ALOHA protocol

Example

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the requirement to make this frame collision-free?

Solution

Average frame transmission time *Tfr* is 200 bits/200 kbps or 1 ms. The vulnerable time is 2 x 1 ms =2 ms. This means no station should send later than 1 ms before this station starts transmission and no station should start sending during the one I-ms period that this station is sending.

The throughput for pure ALOHA is S = G × e −2G . The maximum throughput Smax = 0.184 when G=(1/2).

PROBLEM

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the throughput if the system (all stations together) produces a.

1000 frames per second b. 500 frames per second c. 250 frames per second. The frame transmission time is 200/200 kbps or 1 ms.

a. If the system creates 1000 frames per second, this is 1 frame per millisecond. The load is 1. In this case S = G× $e^{-2\ G}$ or S = 0.135 (13.5 percent). This means that the throughput is 1000 × 0.135 = 135 frames. Only 135 frames out of 1000 will probably survive.

b. If the system creates 500 frames per second, this is (1/2) frame per millisecond. The load is (1/2). In this case S = G × e^{-2G} or S = 0.184 (18.4 percent). Thismeans that the throughput is 500 ×0.184 = 92 and that only 92 frames out of 500 will probably survive. Note that this is the maximum throughput case, percentage wise.

c. If the system creates 250 frames per second, this is (1/4) frame per millisecond. The load is (1/4). In this case S = G × e $-^{2G}$ or S = 0.152 (15.2 percent). This means that the throughput is 250 × 0.152 = 38. Only 38 frames out of 250 will probably survive.

2 Slotted ALOHA

Pure ALOHA has a vulnerable time of 2 x *Tfr* . This is so because there is no rule that defines when the station can send. A station may send soon after another station has started or soon before another station has finished. Slotted ALOHA was invented to improve the efficiency of pure ALOHA.

In slotted ALOHA we divide the time into slots of Tfr s and force the station to send only at the beginning of the time slot. Figure 3 shows an example of frame collisions in slotted ALOHA

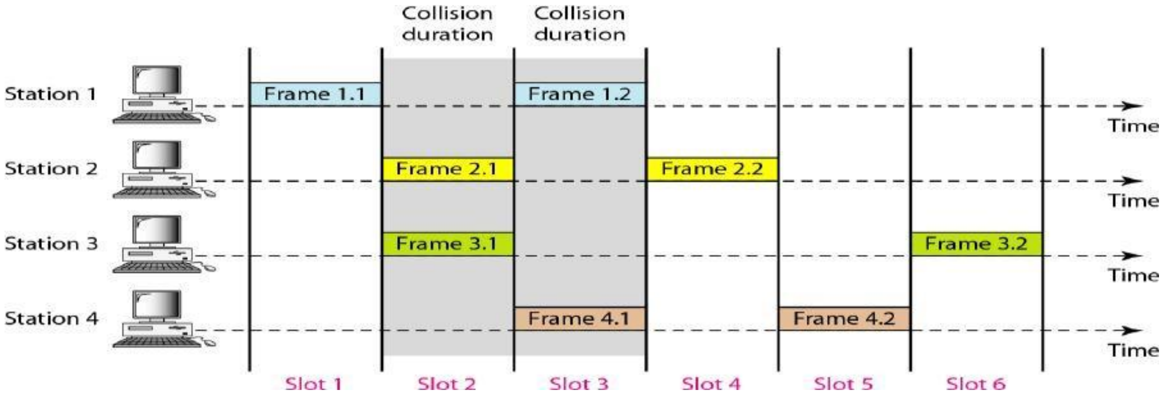
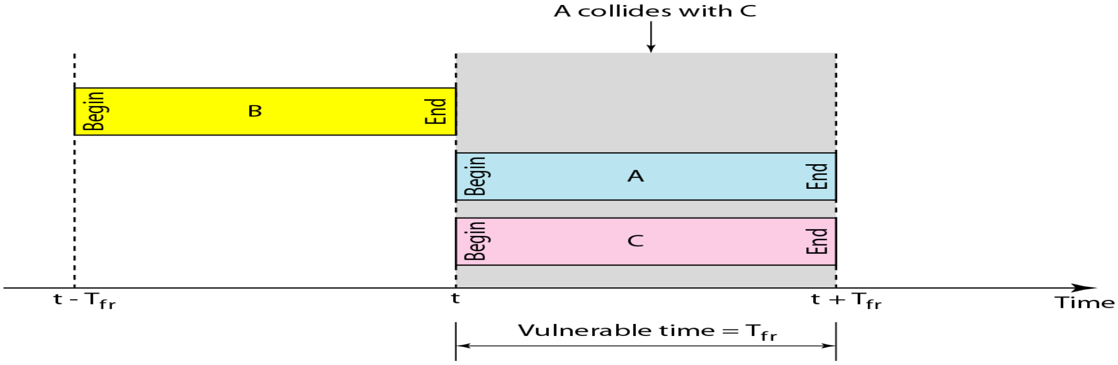


FIG:3

Because a station is allowed to send only at the beginning of the synchronized time slot, if a station misses this moment, it must wait until the beginning of the next time slot. This means that the station which started at the beginning of this slot has already finished sending its frame. Of course, there is still the possibility of collision if two stations try to send at the beginning of the same time slot. However, the vulnerable time is now reduced to one-half, equal to *Tfr* Figure 4 shows the situation

Below fig shows that the vulnerable time for slotted ALOHA is one-half that of pure ALOHA. Slotted ALOHA vulnerable time = Tfr



The throughput for slotted ALOHA is S = G × e−G . The maximum throughput Smax = 0.368 when G =1.

A slotted ALOHA network transmits 200-bit frames using a shared channel with a 200- Kbps bandwidth. Find the throughput if the system (all stations together) produces

a. 1000 frames per second b. 500 frames per second c. 250 frames per second

Solution

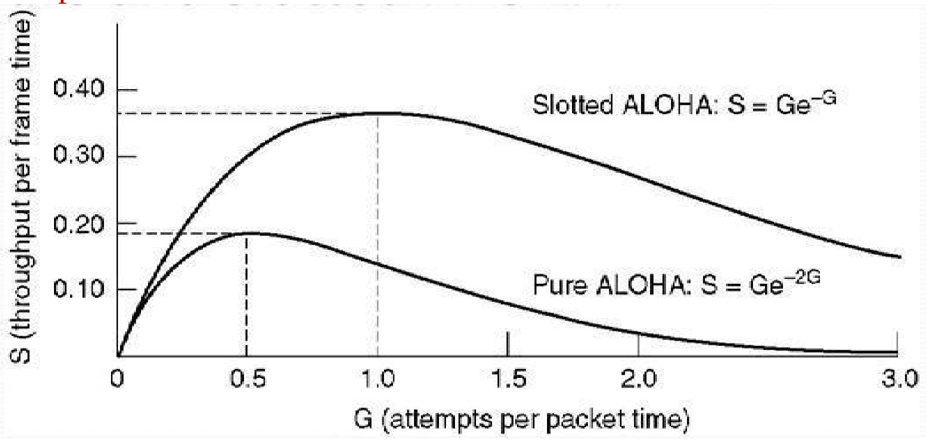
This situation is similar to the previous exercise except that the network is using slotted ALOHA instead of pure ALOHA. The frame transmission time is *200/200* kbps or 1 ms.

a. In this case G is 1. So S =G x *e-G* or *S* =0.368 (36.8 percent). This means that the throughput is 1000 x 0.0368 =368 frames. Only 368 out of 1000 frames will probably survive. Note that this is the maximum throughput case, percentagewise.

b. Here G is 1/2 In this case S =G x *e-G* or S =0.303 (30.3 percent). This means that the throughput is 500 x 0.0303 =151. Only 151 frames out of 500 will probably survive.

c. Now G is 1/4. In this case S =G x *e-G* or S =0.195 (19.5 percent). This means that the throughput is 250 x 0.195 = 49. Only 49 frames out of 250 will probably survive

Comparison between Pure Aloha & Slotted Aloha



Carrier Sense Multiple Access (CSMA)

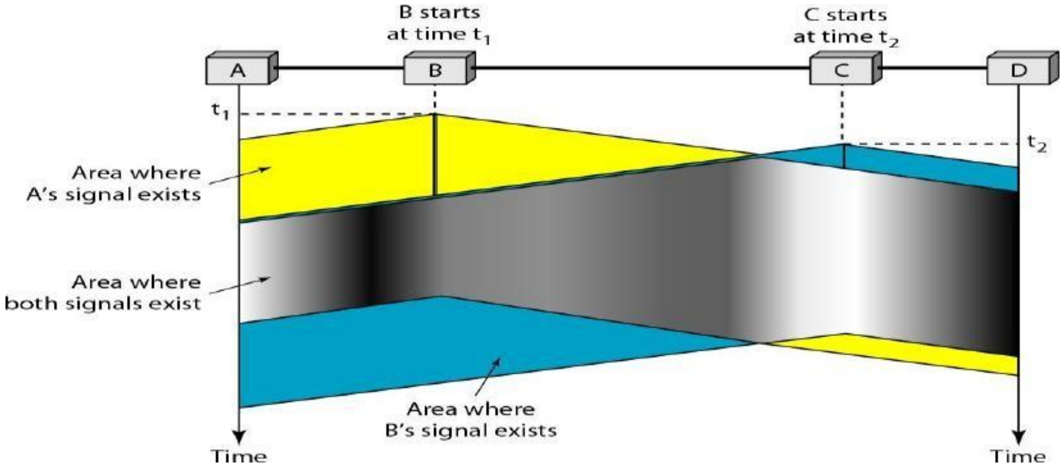
To minimize the chance of collision and, therefore, increase the performance, the CSMA method was developed. The chance of collision can be reduced if a station senses the medium before trying to use it. Carrier sense multiple access (CSMA) requires that each station first listen to the medium (or check the state of the medium) before sending. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk."

CSMA can reduce the possibility of collision, but it cannot eliminate it. The reason for this is shown in below Figure. Stations are connected to a shared channel (usually a dedicated medium).

The possibility of collision still exists because of propagation delay; station may sense the medium and find it idle, only because the first bit sent by another station has not yet been received.

At time *tI'* station B senses the medium and finds it idle, so it sends a frame. At time *t2 (t2> tI)'* station C senses the medium and finds it idle because, at this time, the first bits from station B

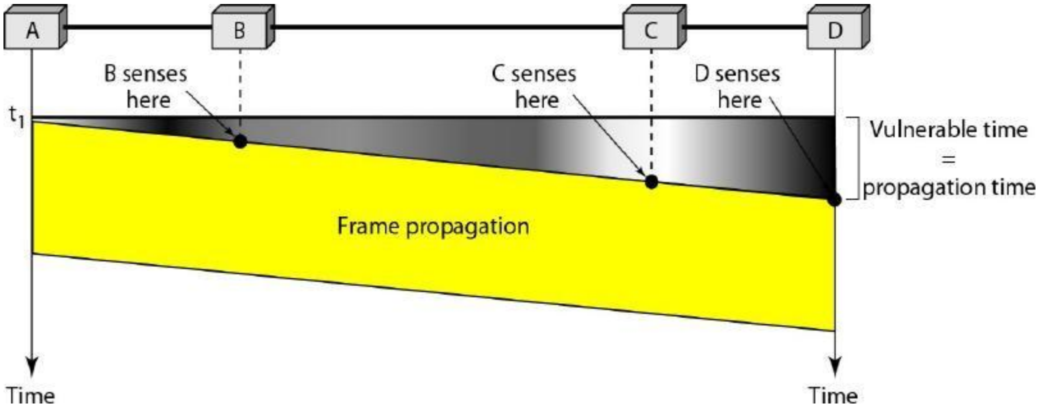
have not reached station C. Station C also sends a frame. The two signals collide and both frames are destroyed.



Space/time model of the collision in CSMA

Vulnerable Time

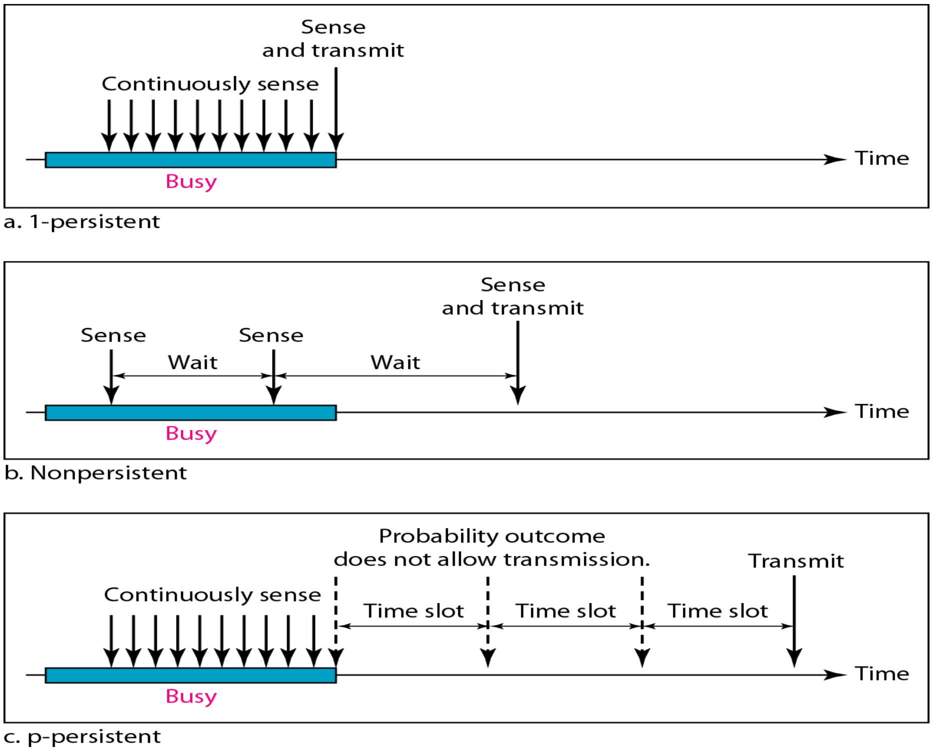
The vulnerable time for CSMA is the propagation time Tp . This is the time needed for a signal to propagate from one end of the medium to the other. When a station sends a frame, and any other station tries to send a frame during this time, a collision will result. But if the first bit of the frame reaches the end of the medium, every station will already have heard the bit and will refrain from sending



Vulnerable time in CSMA

Persistence Methods

What should a station do if the channel is busy? What should a station do if the channel is idle? Three methods have been devised to answer these questions: the 1-persistent method, the non-persistent method, and the p-persistent method



a. 1-persistent

b. Nonpersistent

c. p-persistent

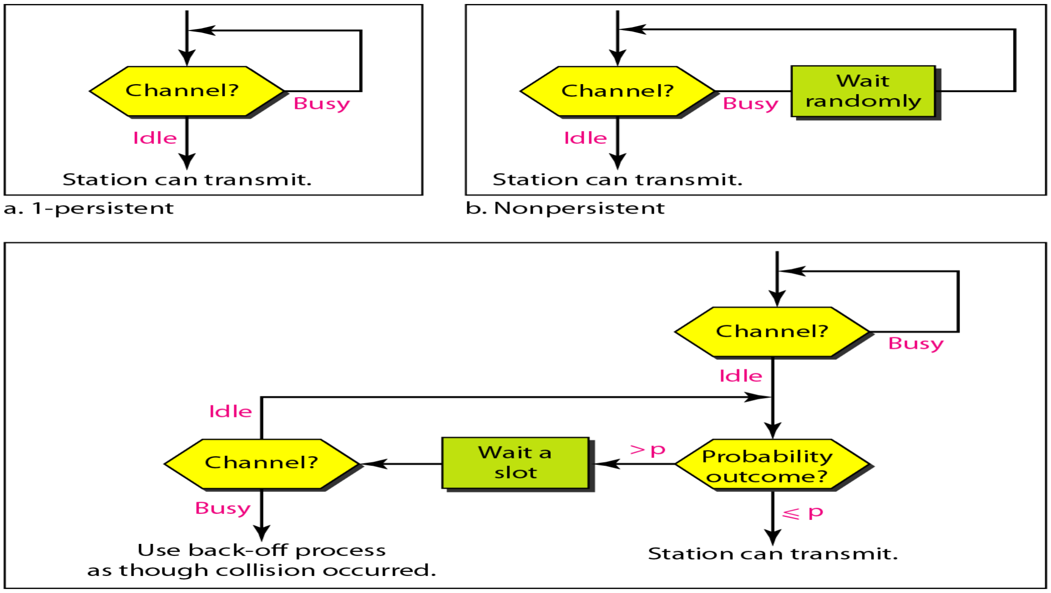
1-Persistent: In this method, after the station finds the line idle, it sends its frame immediately (with probability 1). This method has the highest chance of collision because two or more stations may find the line idle and send their frames immediately.

Non-persistent: a station that has a frame to send senses the line. If the line is idle, it sends immediately. If the line is not idle, it waits a random amount of time and then senses the line again. This approach reduces the chance of collision because it is unlikely that two or more stations will wait the same amount of time and retry to send simultaneously. However, this method reduces the efficiency of the network because the medium remains idle when there may be stations with frames to send.

p-Persistent: This is used if the channel has time slots with a slot duration equal to or greater than the maximum propagation time. The p-persistent approach combines the advantages of the other two strategies. It reduces the chance of collision and improves efficiency.

In this method, after the station finds the line idle it follows thesesteps:

1. With probability *p*, the station sends itsframe.
2. With probability *q* = 1 - *p*, the station waits for the beginning of the next time slot and checks the line again.
   a. If the line is idle, it goes to step 1.
   b. If the line is busy, it acts as though a collision has occurred and uses the backoff procedure.



a. 1-persistent

b. Nonpersistent

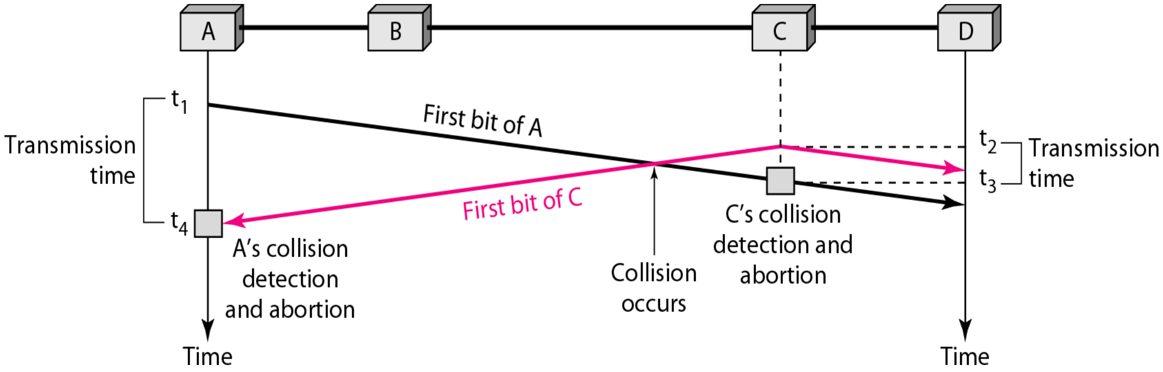
c. p-persistent

Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

The CSMA method does not specify the procedure following a collision. Carrier sense multiple access with collision detection (CSMA/CD) augments the algorithm to handle the collision.

In this method, a station monitors the medium after it sends a frame to see if the transmission was successful. If so, the station is finished. If, however, there is a collision, the frame is sent again.

To better understand CSMA/CD, let us look at the first bits transmitted by the two stations involved in the collision. Although each station continues to send bits in the frame until it detects the collision, we show what happens as the first bits collide. In below Figure, stations A and C are involved in the collision.



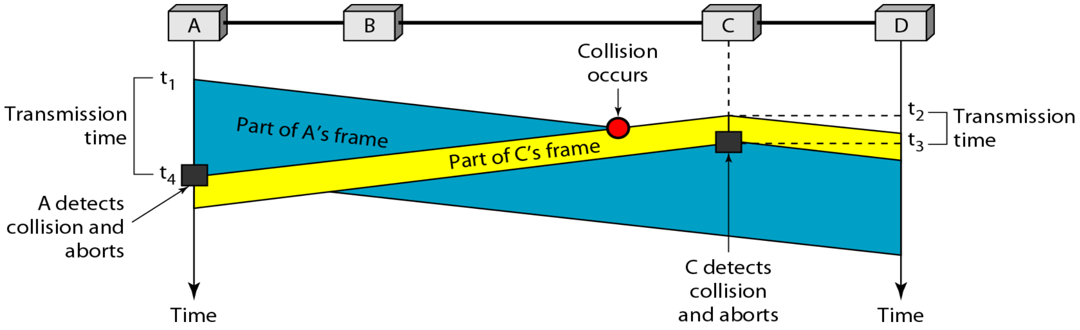
Collision of the first bit in CSMA/CD

At time *t* 1, station A has executed its persistence procedure and starts sending the bits of its frame. At time *t2*, station C has not yet sensed the first bit sent by A. Station C executes its persistence procedure and starts sending the bits in its frame, which propagate both to the left and to the right. The collision occurs sometime after time *t2*.Station C detects a collision at time *t3* when it receives the first bit of A's frame. Station C immediately (or after a short time, but we assume immediately) aborts transmission.

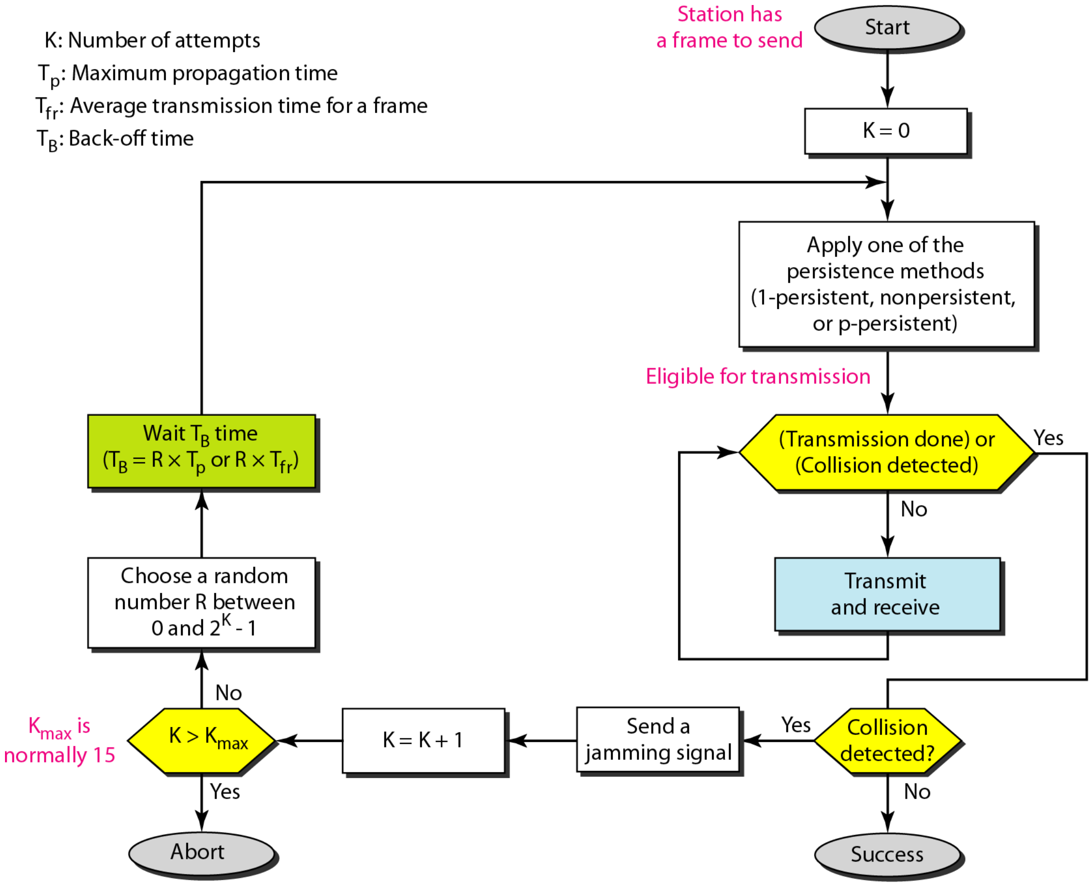
Station A detects collision at time *t4* when it receives the first bit of C's frame; it also immediately aborts transmission. Looking at the figure, we see that A transmits for the duration *t4* - *tl;* C transmits for the duration *t3* - *t2*.

Minimum Frame Size

For *CSMAlCD* to work, we need a restriction on the frame size. Before sending the last bit of the frame, the sending station must detect a collision, if any, and abort the transmission. This is so because the station, once the entire frame is sent, does not keep a copy of the frame and does not monitor the line for collision detection. Therefore, the frame transmission time *T*fr must be at least two times the maximum propagation time *Tp*. To understand the reason, let us think about the worst-case scenario. If the two stations involved in a collision are the maximum distance apart, the signal from the first takes time *Tp* to reach the second, and the effect of the collision takes another time *Tp* to reach the first. So the requirement is that the first station must still be transmitting after *2Tp* .



Collision and abortion in CSMA/CD


K: Number of attempts
T_p: Maximum propagation time
T_{fr}: Average transmission time for a frame
T_B: Back-off time
Station has a frame to send
Start
K = 0
Apply one of the persistence methods (1-persistent, nonpersistent, or p-persistent)
Eligible for transmission
(Transmission done) or (Collision detected)
Yes
No
Transmit and receive
Collision detected?
Yes
No
Success
Send a jamming signal
K = K + 1
$K > K_{max}$
K_{max} is normally 15
No
Yes
Abort
Choose a random number R between 0 and $2^K - 1$
Wait T_B time ($T_B = R \times T_p$ or $R \times T_{fr}$)

Flow diagram for the CSMA/CD

PROBLEM

A network using CSMA/CD has a bandwidth of 10 Mbps. If the maximum propagation time (including the delays in the devices and ignoring the time needed to send a jamming signal, as we see later) is 25.6 μs, what is the minimum size of the frame?

SOL

The frame transmission time is Tfr = 2 × Tp = 51.2 μs. This means, in the worst case, a station needs to transmit for a period of 51.2 μs to detect the collision. The minimum size of the frame is 10 Mbps × 51.2 μs = 512 bits or 64 bytes. This is actually the minimum size of the frame for Standard Ethernet.

DIFFERENCES BETWEEN ALOHA & CSMA/CD

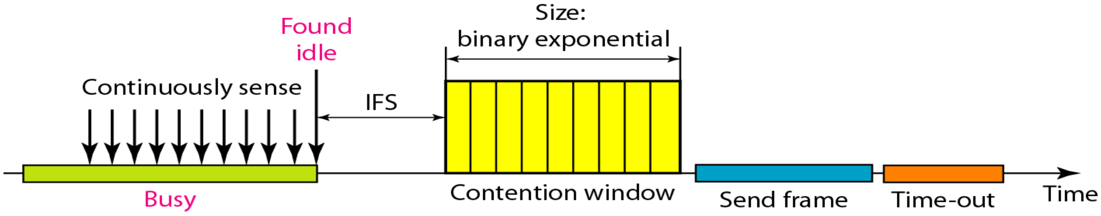
The first difference is the addition of the persistence process. We need to sense the channel before we start sending the frame by using one of the persistence processes

The second difference is the frame transmission. In ALOHA, we first transmit the entire frame and then wait for an acknowledgment. In *CSMA/CD,* transmission and collision detection is a continuous process. We do not send the entire frame and then look for a collision. The station transmits and receives continuously and simultaneously

The third difference is the sending of a short jamming signal that enforces the collision in case other stations have not yet sensed the collision.

Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA)

We need to avoid collisions on wireless networks because they cannot be detected. Carrier sense multiple access with collision avoidance *(CSMAlCA)* was invented for wirelesss network. Collisions are avoided through the use of CSMA/CA's three strategies: the inter frame space, the contention window, and acknowledgments, as shown in Figure



Timing in CSMA/CA

Inter frame Space (IFS)

First, collisions are avoided by deferring transmission even if the channel is found idle. When an idle channel is found, the station does not send immediately. It waits for a period of time called the inter frame space or IFS.

Even though the channel may appear idle when it is sensed, a distant station may have already started transmitting. The distant station's signal has not yet reached this station. The IFS time allows the front of the transmitted signal by the distant station to reach this station. If after the IFS time the channel is still idle, the station can send, but it still needs to wait a time equal to the contention time. The IFS variable can also be used to prioritize stations or frame types. For example, a station that is assigned shorter IFS has a higher priority.

In CSMA/CA, the IFS can also be used to define the priority of a station or a frame.

Contention Window

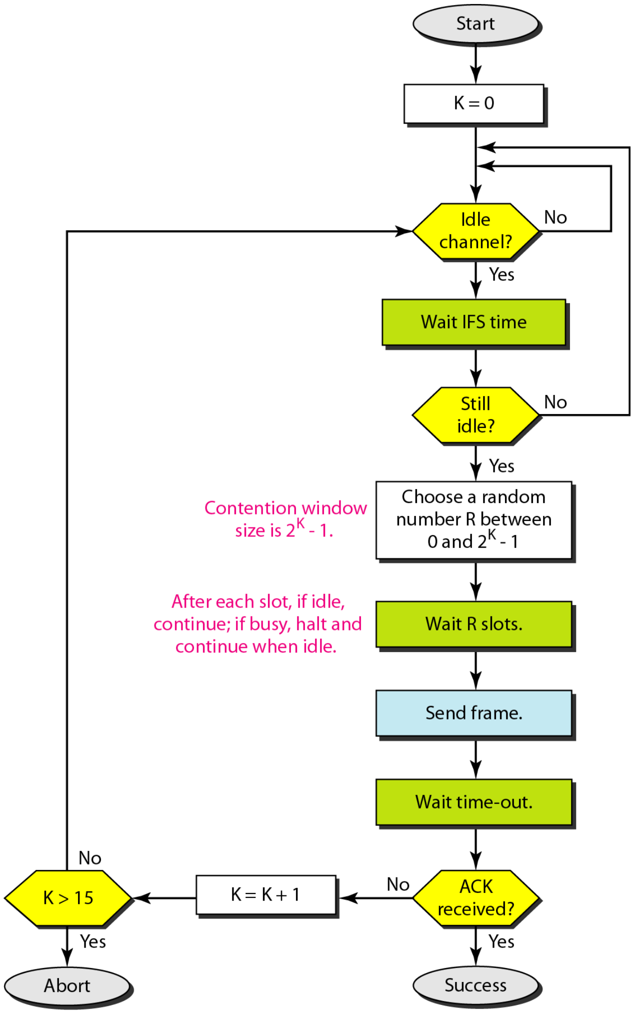
The contention window is an amount of time divided into slots. A station that is ready to send chooses a random number of slots as its wait time. The number of slots in the window changes according to the binary exponential back-off strategy. This means that it is set to one slot the first time and then doubles each time the station cannot detect an idle channel after the IFS time. This is very similar to the p-persistent method except that a random outcome defines the number of slots taken by the waiting station.

One interesting point about the contention window is that the station needs to sense the channel after each time slot. However, if the station finds the channel busy, it does not restart the process; it just stops the timer and restarts it when the channel is sensed as idle. This gives priority to the station with the longest waiting time.

In CSMA/CA, if the station finds the channel busy, it does not restart the timer of the contention window; it stops the timer and restarts it when the channel becomes idle.

Acknowledgment

With all these precautions, there still may be a collision resulting in destroyed data. In addition, the data may be corrupted during the transmission. The positive acknowledgment and the time-out timer can help guarantee that the receiver has received the frame.


Start
K = 0
Idle channel?
No
Yes
Wait IFS time
Still idle?
No
Yes
Choose a random number R between 0 and 2^K - 1
Contention window size is 2^K - 1.
After each slot, if idle, continue; if busy, halt and continue when idle.
Wait R slots.
Send frame.
Wait time-out.
ACK received?
No
Yes
K = K + 1
K > 15
No
Yes
Abort
Success

This is the CSMA protocol with collision avoidance.

- The station ready to transmit, senses the line by using one of the persistent strategies.
- As soon as it finds the line to be idle, the station waits for an IFS (Inter frame

space) amount of time.

- If then waits for some random time and sends the frame.
- After sending the frame, it sets a timer and waits for the acknowledgement from the receiver.
- If the acknowledgement is received before expiry of the timer, then the transmission is successful.
- But if the transmitting station does not receive the expected acknowledgement before the timer expiry then it increments the back off parameter, waits for the back off time and re senses the line

Controlled Access Protocols

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

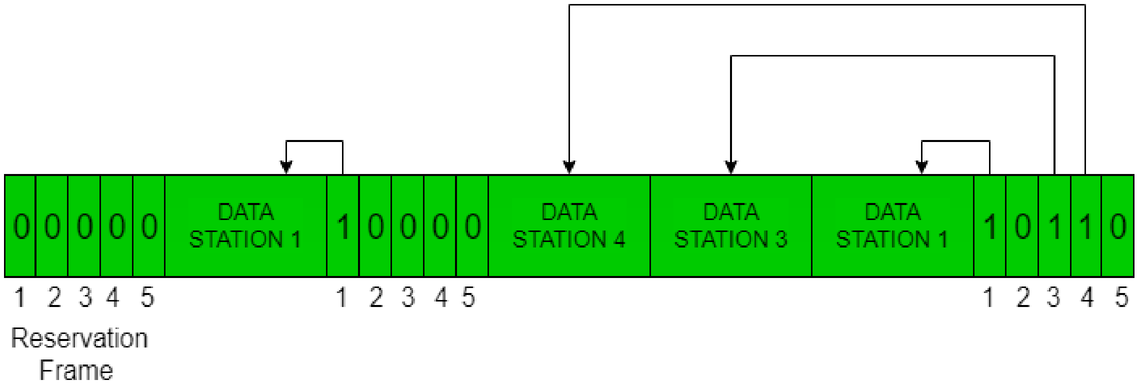
1 Reservation 2 Polling 3 Token Passing

Reservation

- In the reservation method, a station needs to make a reservation before sending data.
- The time line has two kinds of periods:
  1. Reservation interval of fixed time length
  2. Data transmission period of variable frames.
- If there are M stations, the reservation interval is divided into M slots, and each station has one slot.
- Suppose if station 1 has a frame to send, it transmits 1 bit during the slot

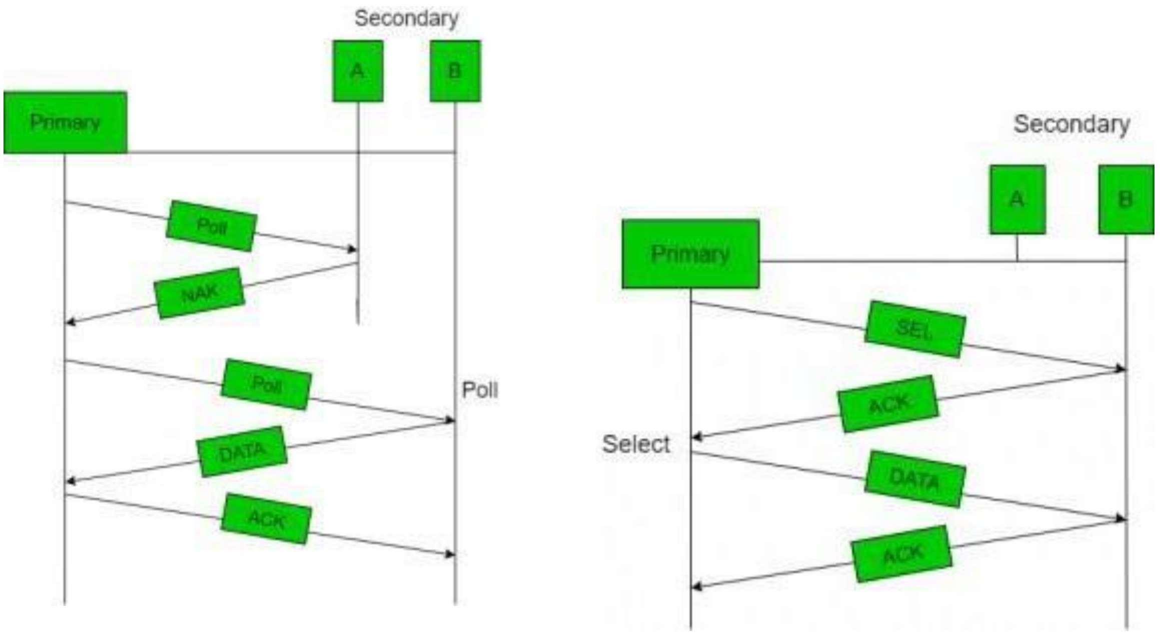
  1. No other station is allowed to transmit during this slot.
- In general, i th station may announce that it has a frame to send by inserting a 1 bit into i th slot. After all N slots have been checked, each station knows which stations wish to transmit.
- The stations which have reserved their slots transfer their frames in that order.
- After data transmission period, next reservation interval begins.
- Since everyone agrees on who goes next, there will never be any collisions.

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



Polling

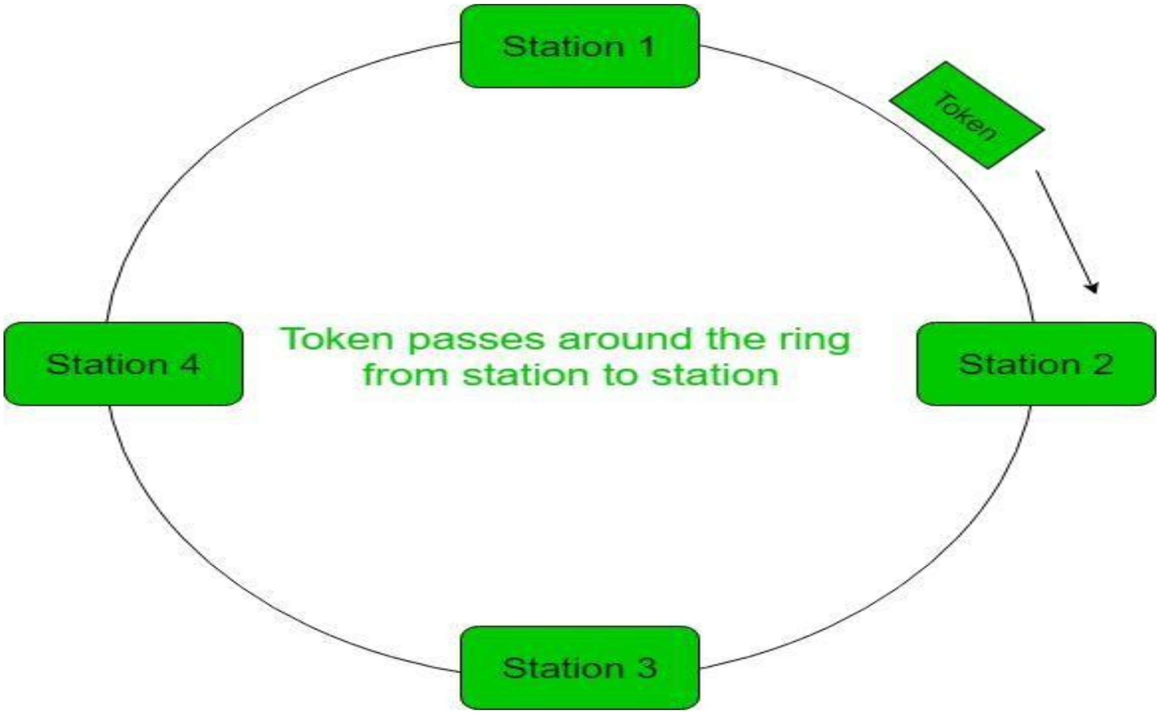
- Polling process is similar to the roll-call performed in class. Just like the teacher, a controller sends a message to each node in turn.
- In this, one acts as a primary station(controller) and the others are secondary stations. All data exchanges must be made through the controller.
- The message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



Token Passing

- In token passing scheme, the stations are connected logically to each other in form of ring and access of stations is governed by tokens.

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



Error Detection

Error

A condition when the receiver’s information does not matches with the sender’s information. During transmission, digital signals suffer from noise that can introduce errors in the binary bits travelling from sender to receiver. That means a 0 bit may change to 1 or a 1 bit may change to 0.

Error Detecting Codes (Implemented either at Data link layer or Transport Layer of OSI Model) Whenever a message is transmitted, it may get scrambled by noise or data may get corrupted. To avoid this, we use error-detecting codes which are additional data added to a given digital message to help us detect if any error has occurred during transmission of the message.

Basic approach used for error detection is the use of redundancy bits, where additional bits are added to facilitate detection of errors. Some popular techniques for error detection are:

1. Simple Parity check

2. Two-dimensional Parity check

3. Checksum

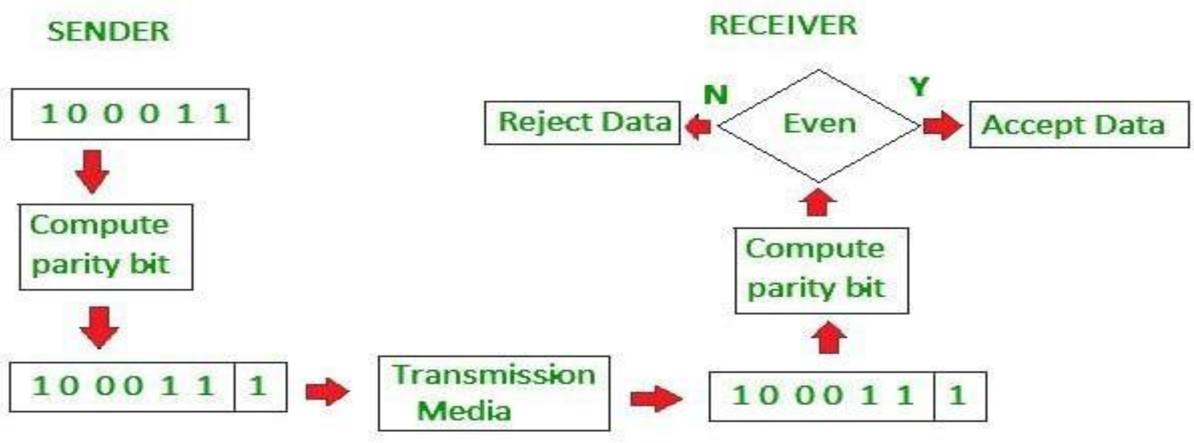
4. Cyclic redundancy check

Simple Parity check

Blocks of data from the source are subjected to a check bit or parity bit generator form, where a parity of : 1 is added to the block if it contains odd number of 1's, and

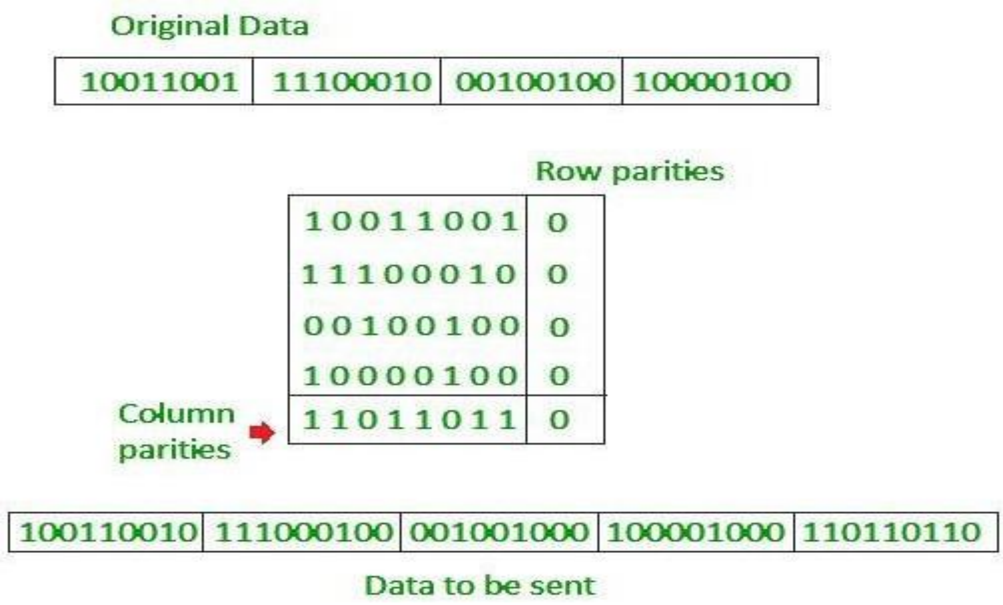
0 is added if it contains even number of 1's

This scheme makes the total number of 1's even, that is why it is called even parity checking.



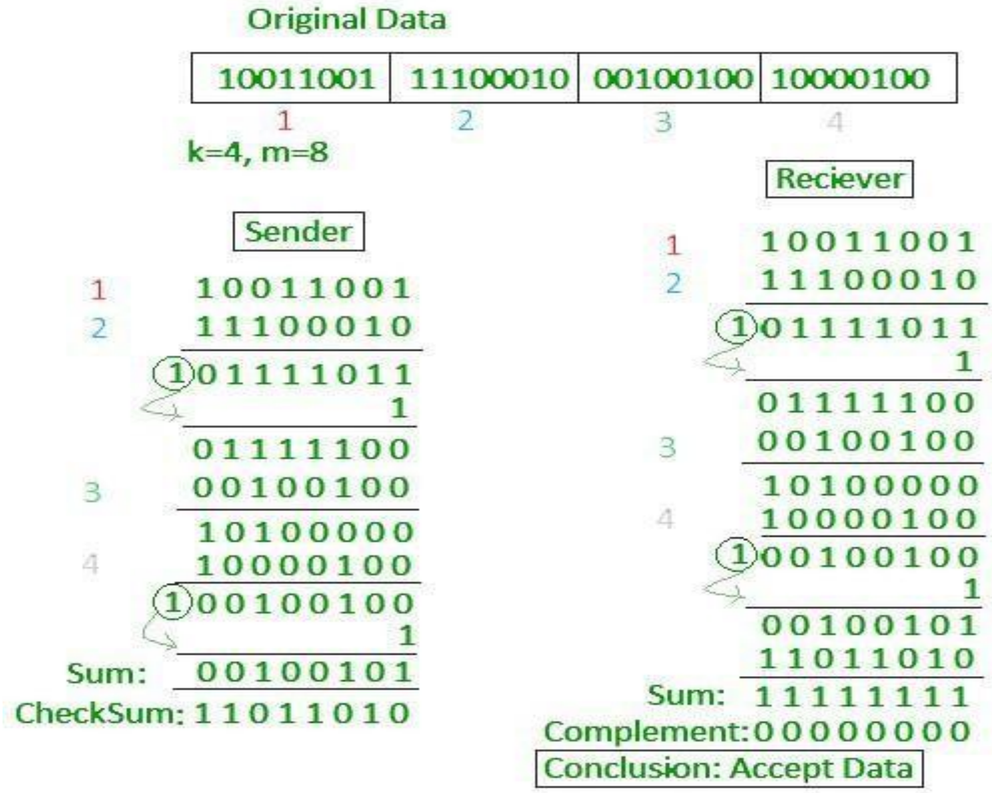
Two-dimensional Parity check

Parity check bits are calculated for each row, which is equivalent to a simple parity check bit. Parity check bits are also calculated for all columns, then both are sent along with the data. At the receiving end these are compared with the parity bits calculated on the received data.

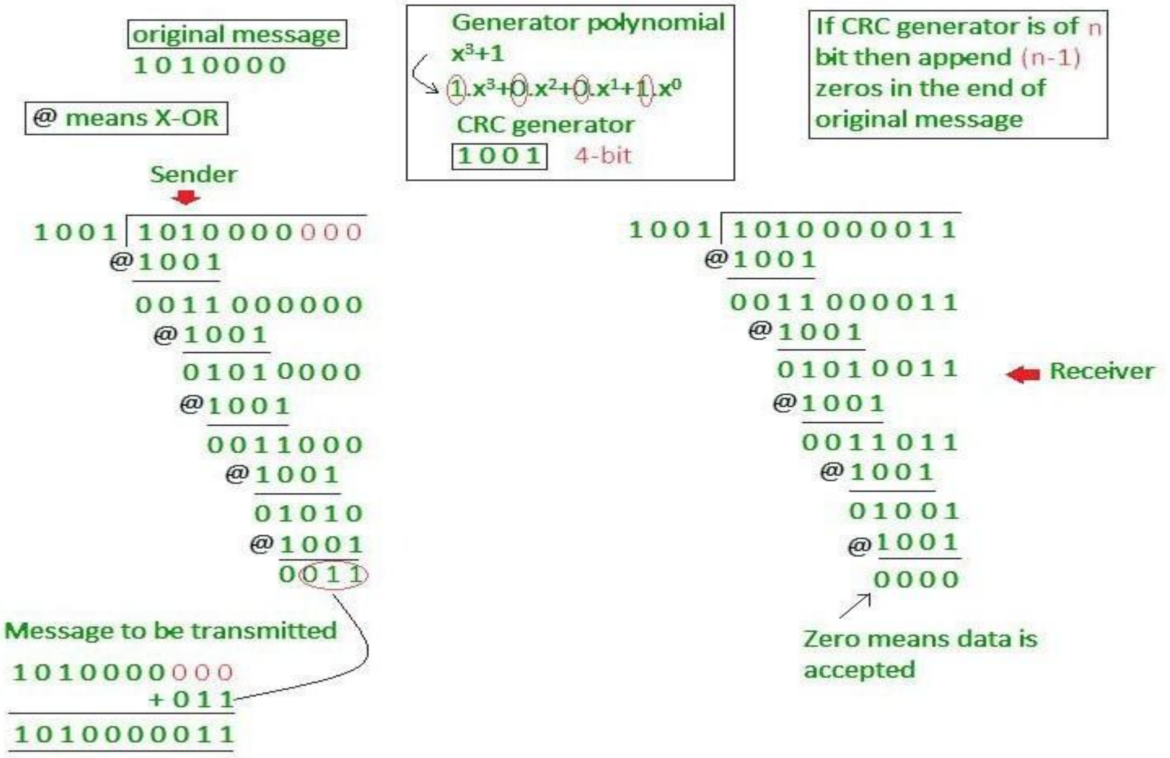


Checksum

- In checksum error detection scheme, the data is divided into k segments each of m bits.
- In the sender's end the segments are added using 1's complement arithmetic to get the sum. The sum is complemented to get the checksum.
- The checksum segment is sent along with the data segments.
- At the receiver's end, all received segments are added using 1's complement arithmetic to get the sum. The sum is complemented.
- If the result is zero, the received data is accepted; otherwise discarded.



Cyclic redundancy check (CRC)



- Unlike checksum scheme, which is based on addition, CRC is based on binary division.
- In CRC, a sequence of redundant bits, called cyclic redundancy check bits, are appended to the end of data unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number.
- At the destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit is assumed to be correct and is therefore accepted.
- A remainder indicates that the data unit has been damaged in transit and therefore must be rejected.

Error Correction

Error Correction codes are used to detect and correct the errors when data is transmitted from the sender to the receiver.

Error Correction can be handled in two ways:

Backward error correction: Once the error is discovered, the receiver requests the sender to retransmit the entire data unit.

Forward error correction: In this case, the receiver uses the error-correcting code which automatically corrects the errors.

A single additional bit can detect the error, but cannot correct it.

For correcting the errors, one has to know the exact position of the error. For example, If we want to calculate a single-bit error, the error correction code will determine which one of seven bits is in error. To achieve this, we have to add some additional redundant bits.

Suppose r is the number of redundant bits and d is the total number of the data bits. The number of redundant bits r can be calculated by using the formula:

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

The value of r is calculated by using the above formula. For example, if the value of d is 4, then the possible smallest value that satisfies the above relation would be 3.

To determine the position of the bit which is in error, a technique developed by

R.W Hamming is Hamming code which can be applied to any length of the data unit and uses the relationship between data units and redundant units.

Hamming Code

Parity bits: The bit which is appended to the original data of binary bits so that the total number of 1s is even or odd.

Even parity: To check for even parity, if the total number of 1s is even, then the

value of the parity bit is 0. If the total number of 1s occurrences is odd, then the value of the parity bit is 1.

Odd Parity: To check for odd parity, if the total number of 1s is even, then the

value of parity bit is 1. If the total number of 1s is odd, then the value of parity bit is 0.

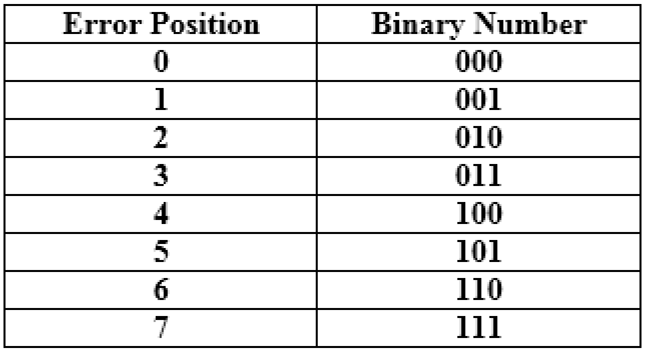
Algorithm of Hamming code:

An information of 'd' bits are added to the redundant bits 'r' to form d+r. The location of each of the (d+r) digits is assigned a decimal value.

The 'r' bits are placed in the positions 1,2, ... 2^{k-1}

At the receiving end, the parity bits are recalculated. The decimal value of the parity bits determines the position of an error.

Relationship b/w Error position & binary number.



Error Position	Binary Number
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Let's understand the concept of Hamming code through an example:

Suppose the original data is 1010 which is to be sent.

Total number of data bits 'd' = 4

Number of redundant bits r : 2^r >=d+r+1

$$2^r >= 4+r+1$$

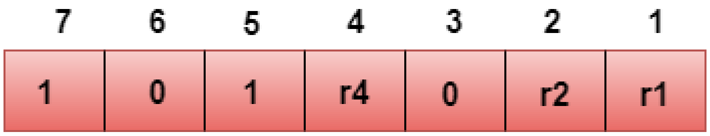
Therefore, the value of r is 3 that satisfies the above relation. Total number of bits = d+r = 4+3 = 7;

Determining the position of the redundant bits

The number of redundant bits is 3. The three bits are represented by r1, r2, r4. The position of the redundant bits is calculated with corresponds to the raised power of 2. Therefore, their corresponding positions are 1, 2^1, 2^2.

The position of r1 = 1, The position of r2 = 2 , The position of r4 = 4

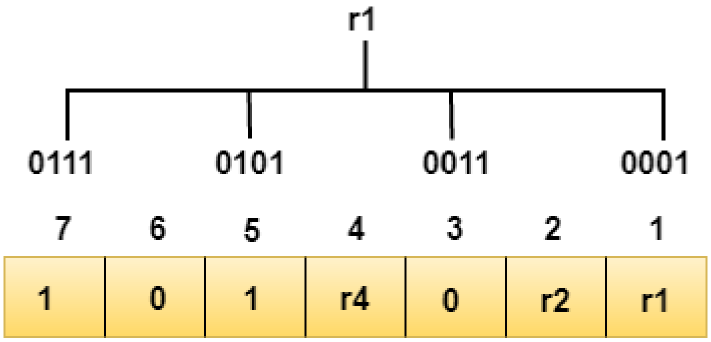
Representation of Data on the addition of parity bits:



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

Determining the Parity bits

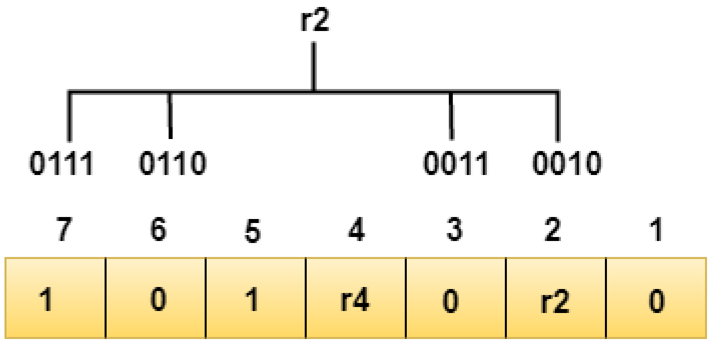
Determining the r1 bit: The r1 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the first position.



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

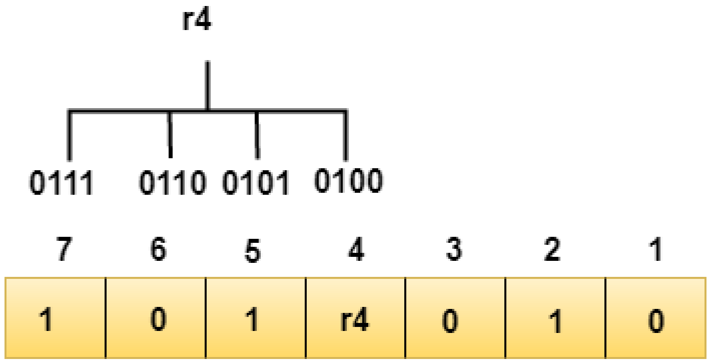
We observe from the above figure that the bit position that includes 1 in the first position are 1, 3, 5, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r1 is even, therefore, the value of the r1 bit is 0.

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



We observe from the above figure that the bit positions that includes 1 in the second position are 2, 3, 6, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r2 is odd, therefore, the value of the r2 bit is 1.

Determining r4 bit: The r4 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the third position.



We observe from the above figure that the bit positions that includes 1 in the third position are 4, 5, 6, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r4 is even, therefore, the value of the r4 bit is 0.

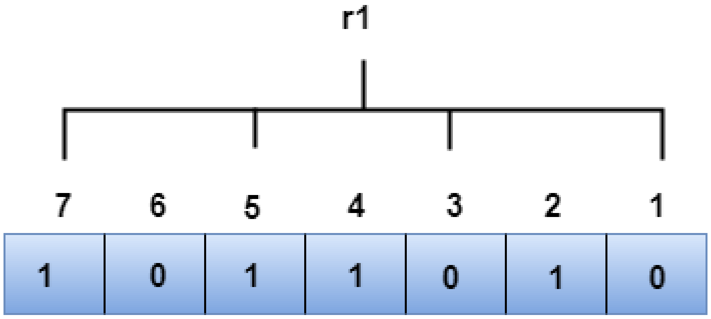
Data transferred is given below:

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

Suppose the 4th bit is changed from 0 to 1 at the receiving end, then parity bits are recalculated.

R1 bit

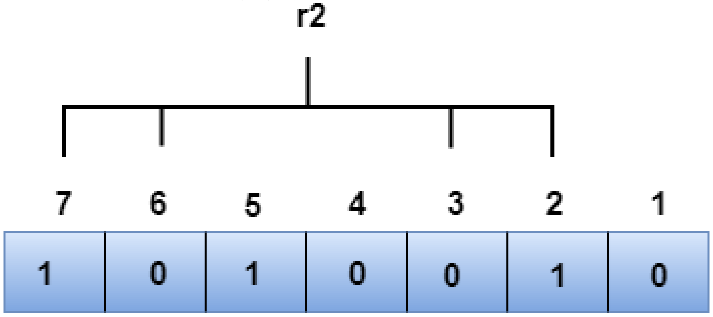
The bit positions of the r1 bit are 1,3,5,7



We observe from the above figure that the binary representation of r1 is 1100. Now, we perform the even-parity check, the total number of 1s appearing in the r1 bit is an even number. Therefore, the value of r1 is 0.

R2 bit

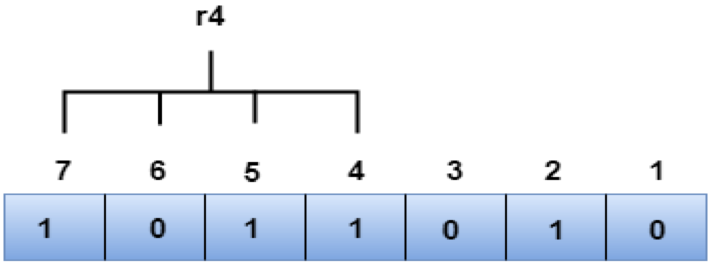
The bit positions of r2 bit are 2,3,6,7.



We observe from the above figure that the binary representation of r2 is 1001. Now, we perform the even-parity check, the total number of 1s appearing in the r2 bit is an even number. Therefore, the value of r2 is 0.

R4 bit

The bit positions of r4 bit are 4,5,6,7.



We observe from the above figure that the binary representation of r4 is 1011. Now, we perform the even-parity check, the total number of 1s appearing in the r4 bit is an odd number. Therefore, the value of r4 is 1.

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

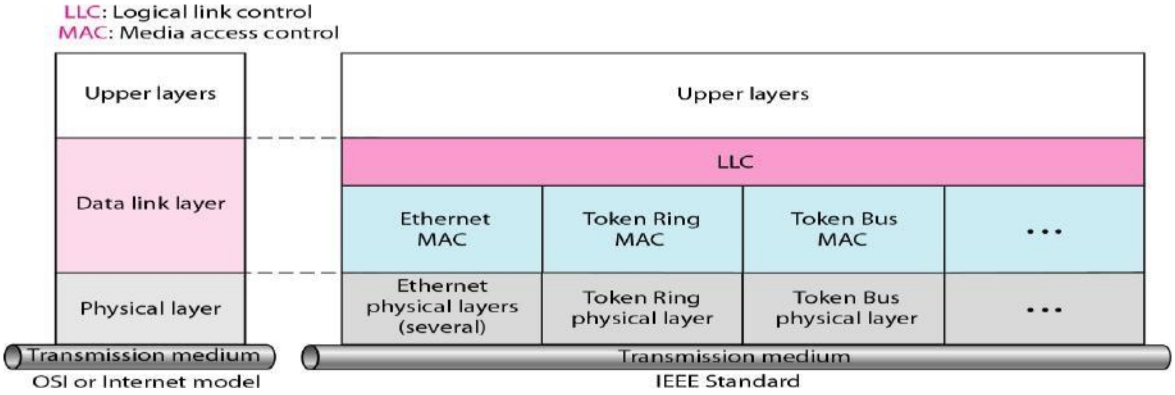
Wired LANs: Ethernet

In 1985, the Computer Society of the IEEE started a project, called Project

802, to set standards to enable intercommunication among equipment from a variety of manufacturers. Project 802 is a way of specifying functions of the physical layer and the data link layer of major LAN protocols.

The relationship of the 802 Standard to the traditional OSI model is shown in below Figure. The IEEE has subdivided the data link layer into two sub layers: logical link control (LLC) and media access control).

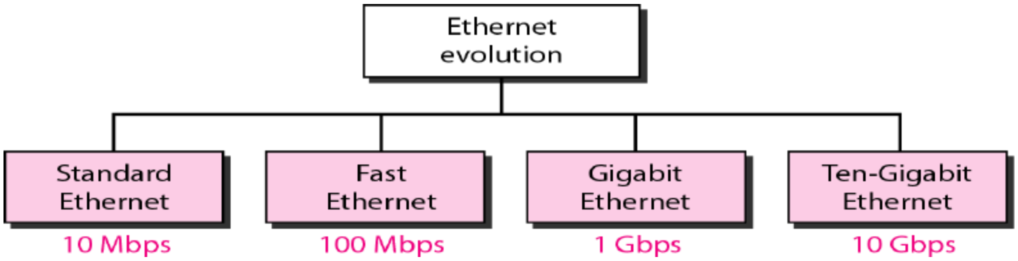
IEEE has also created several physical layer standards for different LAN protocols



IEEE standard for LANs

STANDARD ETHERNET

The original Ethernet was created in 1976 at Xerox’s Palo Alto Research Center (PARC). Since then, it has gone through four generations.
Standard Ethernet (l0 Mbps), Fast Ethernet (100 Mbps), Gigabit Ethernet (l Gbps), and Ten-Gigabit Ethernet (l0 Gbps),
We briefly discuss the Standard (or traditional) Ethernet in this section



Ethernet evolution through four generations

<u>MAC Sublayer</u>

In Standard Ethernet, the MAC sublayer governs the operation of the access method. It also frames data received from the upper layer and passes them to the physical layer.

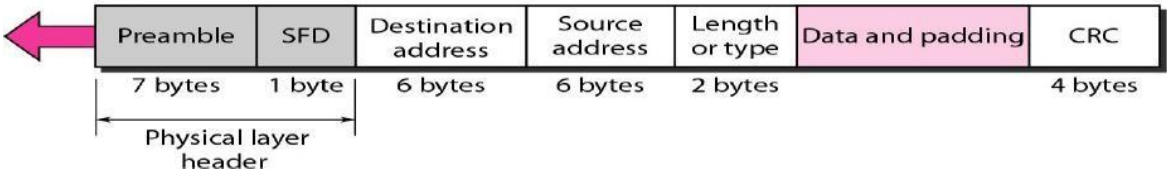
Frame Format

The Ethernet frame contains seven fields: preamble, SFD, DA, SA, length or type of protocol data unit (PDU), upper-layer data, and the CRC. Ethernet does not provide any mechanism for acknowledging received frames, making it what is known as an unreliable medium. Acknowledgments must be implemented at the higher layers. The format of the MAC frame is shown in below figure

802.3 MAC frame

Preamble: 56 bits of alternating 1s and 0s.

SFD: Start frame delimiter, flag (10101011)



Preamble. The first field of the 802.3 frame contains 7 bytes (56 bits) of alternating 0s and 1s that alerts the receiving system to the coming frame and enables it to synchronize its input timing. The pattern provides only an alert and a timing pulse. The 56-bit pattern allows the stations to miss some bits at the beginning of the frame. The preamble is actually added at the physical layer and is not (formally) part of the frame.

Start frame delimiter (SFD). The second field (l byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits is 11 and alerts the receiver that the next field is the destination address.

Destination address (DA). The DA field is 6 bytes and contains the physical address of the destination station or stations to receive the packet.

Source address (SA). The SA field is also 6 bytes and contains the physical address of the sender of thepacket.

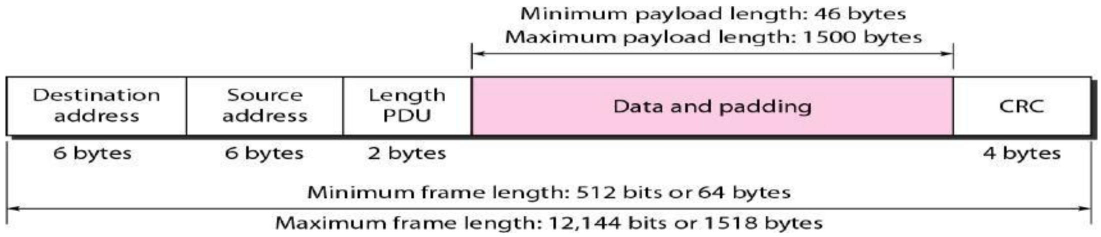
Length or type. This field is defined as a type field or length field. The original Ethernet used this field as the type field to define the upper-layer protocol using the MAC frame. The IEEE standard used it as the length field to define the number of bytes in the data field. Both uses are common today.

Data. This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes.

CRC. The last field contains error detection information, in this case a CRC-32

Frame Length

Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame, as shown in below Figure


Minimum payload length: 46 bytes
Maximum payload length: 1500 bytes
Destination address
Source address
Length PDU
Data and padding
CRC
6 bytes
6 bytes
2 bytes
4 bytes
Minimum frame length: 512 bits or 64 bytes
Maximum frame length: 12,144 bits or 1518 bytes

Minimum and maximum lengths

An Ethernet frame needs to have a minimum length of 512 bits or 64 bytes. Part of this length is the header and the trailer. If we count 18 bytes of header and trailer (6 bytes of source address, 6 bytes of destination address, 2 bytes of length or type, and 4 bytes of CRC), then the minimum length of data from the upper layer is 64 - 18 = 46 bytes. If the upper-layer packet is less than 46 bytes, padding is added to make up the difference

The standard defines the maximum length of a frame (without preamble and SFD field) as 1518 bytes. If we subtract the 18 bytes of header and trailer,the maximum length of the payload is 1500 bytes.

The maximum length restriction has two historical reasons.

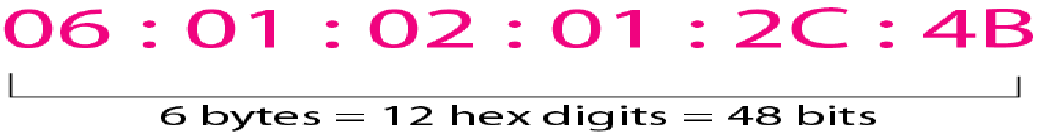
First, memory was very expensive when Ethernet was designed: a maximum length restriction helped to reduce the size of the buffer.

Second, the maximum length restriction prevents one station from monopolizing the shared medium, blocking other stations that have data to send.

Addressing

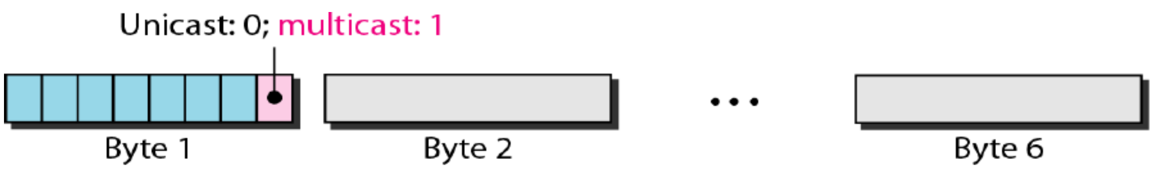
The Ethernet address is 6 bytes (48 bits), normally written in hexadecimal notation, with a colon between the bytes.

Example of an Ethernet address in hexadecimal notation


06 : 01 : 02 : 01 : 2C : 4B
6 bytes = 12 hex digits = 48 bits

<u>Unicast, Multicast, and Broadcast Addresses</u> A source address is always a unicast address-the frame comes from only one station. The destination address, however, can be **unicast, multicast, or broadcast**. Below Figure shows how to distinguish a unicast address from a multicast address.

If the least significant bit of the first byte in a destination address is 0, the address is unicast; otherwise, it is multicast.



Unicast and multicast addresses

A unicast destination address defines only one recipient; the relationship between the sender and the receiver is one-to-one.

A multicast destination address defines a group of addresses; the relationship between the sender and the receivers is one-to-many.

The broadcast address is a special case of the multicast address; the recipients are all the stations on the LAN. A broadcast destination address is forty-eight 1s.

Access Method: CSMA/CD

Standard Ethernet uses I-persistent CSMA/CD Slot Time In an Ethernet network.

Slot time =round-trip time + time required to send the jam sequence

The slot time in Ethernet is defined in bits. It is the time required for a station to send 512 bits. This means that the actual slot time depends on the data rate; for traditional 10-Mbps Ethernet it is 51.2 micro sec.

Slot Time and Maximum Network Length There is a relationship between the slot time and the maximum length of the network (collision domain). It is dependent on the propagation speed of the signal in the particular medium.

In most transmission media, the signal propagates at 2×10^{8} m/s (two-thirds of the rate for propagation inair).

For traditional Ethernet, we calculate

MaxLength =PropagationSpeedx (SlotTime/2)

MaxLength= (2×10^{8}) X$(51.2 \text{ X}10^{-6})/2$= 5120m

Of course, we need to consider the delay times in repeaters and interfaces, and the time required to send the jam sequence. These reduce the maximum-length of a traditional Ethernet network to 2500 m, just 48 percent of the theoretical calculation. MaxLength=2500 m

TEXT / REFERENCE BOOKS

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DEPARTMENT OF INFORMATION TECHNOLOGY

UNIT – III – Data communication and Computer networks – SCS1314

NETWORK LAYER

Circuit Switching - Packet Switching - Routing - Distance Vector Routing – Link StateRoutingAddressing-Subnetting - IPV4- IPV6- ARP - RARP - ICMP - IGMP - DHCP

Functions of Net Work layer

1. Routing 2. Congestion Control

Routing algorithms

The main function of the network layer is routing packets from the source machine to the destination machine. Routing algorithm can be grouped into two major classes. Nonadaptive and Adaptive algorithms.

Non adaptive	**Adaptive**
1) Routing decisions are not based on measurements or estimates of the current traffic and topology.	1) Routing decisions are based on measurements of the current traffic and topology.
2) The route is computed well in advance.	2) The route is computed depends on situation.
3) When the network is booted the routers are downloaded.	3) The routers are not downloaded.
.	4) This is a dynamic routing.

Classification of a Routing algorithm

Distance Vector Routing Algorithm

- The Distance vector algorithm is iterative, asynchronous and distributed.

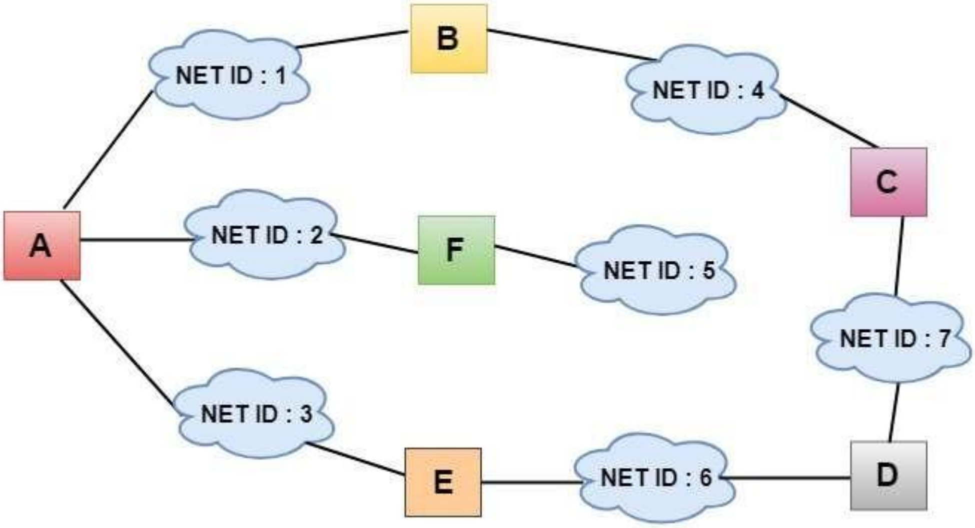
- **Distributed:** It is distributed in that each node receives information from one or more of its directly attached neighbors, performs calculation and then distributes the result back to its neighbors.
- **Iterative:** It is iterative in that its process continues until no more
  - information is available to be exchanged between neighbors.
- **Asynchronous:** It does not require that all of its nodes operate
  - in the lock step with each other.

- The Distance vector algorithm is a dynamic algorithm.
- It is mainly used in ARPANET, and RIP.
- Each router maintains a distance table known as **Vector**.

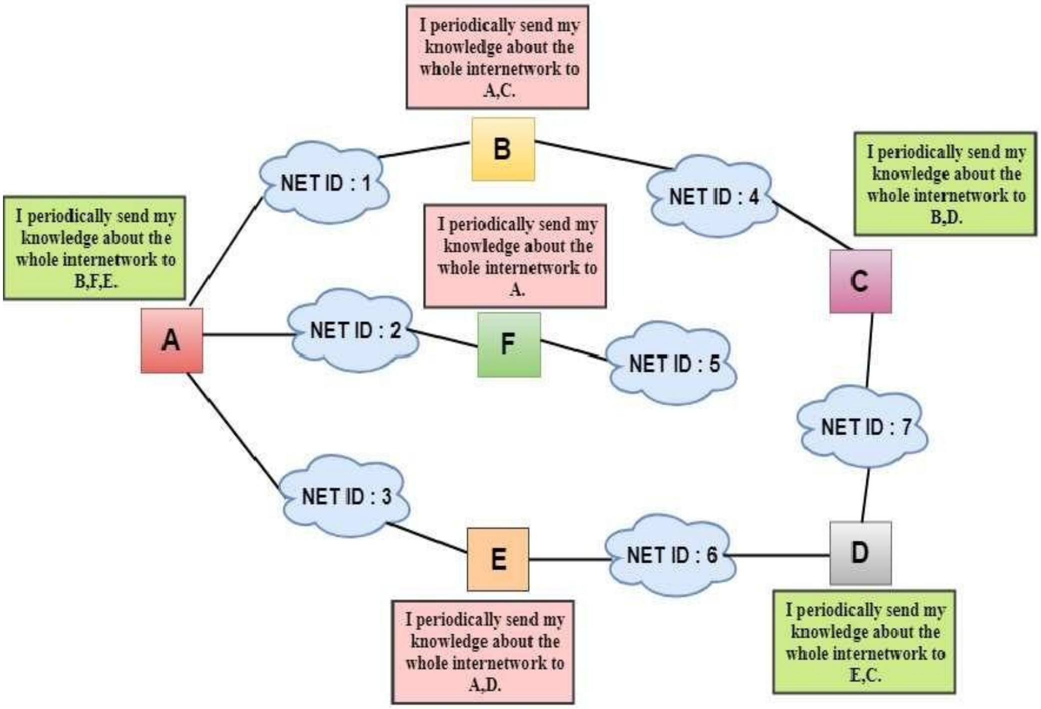
Three Keys to understand the working of Distance Vector Routing Algorithm

- **Knowledge about the whole network:** Each router shares its knowledge through the entire network. The Router sends its collected knowledge about the network to its neighbors.
- **Routing only to neighbors:** The router sends its knowledge about the network to only those routers which have direct links. The router sends whatever it has about the network through the ports. The information is received by the router and uses the information to update its own routing table.
- **Information sharing at regular intervals:** Within 30 seconds, the router sends the information to the neighboring routers.

Let's understand through an example



- In the above figure, each cloud represents the network, and the number inside the cloud represents the network ID.
- All the LANs are connected by routers, and they are represented in boxes labeled as A, B, C, D, E, F.
- Distance vector routing algorithm simplifies the routing process by assuming the cost of every link is one unit. Therefore, the efficiency of transmission can be measured by the number of links to reach the destination.
- In Distance vector routing, the cost is based on hop count.


I periodically send my knowledge about the whole internetwork to A,C.
B
NET ID : 1
NET ID : 4
I periodically send my knowledge about the whole internetwork to B,D.
I periodically send my knowledge about the whole internetwork to B,F,E.
I periodically send my knowledge about the whole internetwork to A.
C
A
NET ID : 2
F
NET ID : 5
NET ID : 7
NET ID : 3
E
NET ID : 6
D
I periodically send my knowledge about the whole internetwork to A,D.
I periodically send my knowledge about the whole internetwork to E,C.

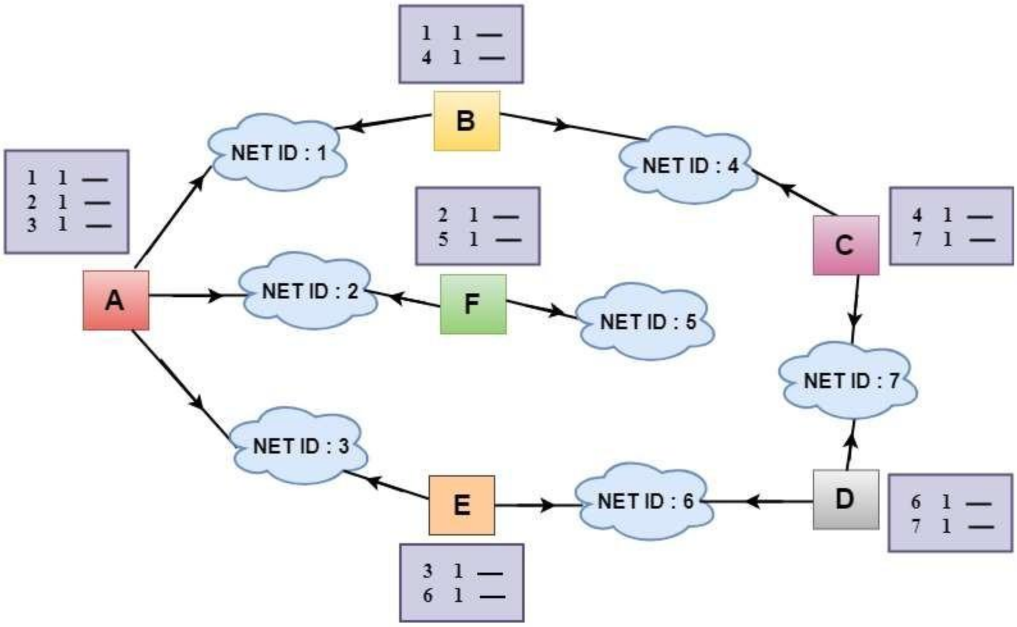
- In the above figure, we observe that the router sends the knowledge to the immediate neighbors.

- The neighbors add this knowledge to their own knowledge and sends the updated table to their own neighbors.

- In this way, routers get its own information plus the new information about the neighbors.

Routing Table

- Two process occurs:
- Creating the Table
- Updating the Table
- Initially, the routing table is created for each router that contains atleast three types of information such as Network ID, the cost and the next hop.
- **NET ID:** The Network ID defines the final destination of the packet.
- **Cost:** The cost is the number of hops that packet



- In the above figure, the original routing tables are shown of all the routers. In a routing table, the first column represents the network ID, the second column represents the cost of the link,

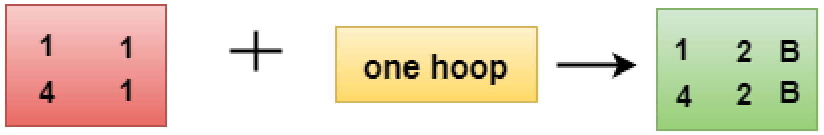
  and the third column is empty.

  For Example:

- A sends its routing table to B, F & E.
- B sends its routing table to A & C.
- C sends its routing table to B & D.
- D sends its routing table to E & C.
- E sends its routing table to A & D.
- F sends its routing table to A.

Updating the Table

- When A receives a routing table from B, then it uses its information to update the table.
- The routing table of B shows how the packets can move to the networks 1 and 4.
- The B is a neighbor to the A router, the packets from A to B can reach in one hop. So, 1 is added to all the costs given in the B's table and the sum will be the cost to reach a particular network.



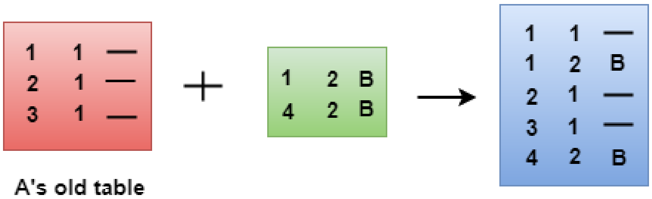
1	1
4	1

Received from B

1	2	B
4	2	B

After adjustment

After adjustment, A then combines this table with its own table to create a combined table.



1	1	—
2	1	—
3	1	—

A's old table

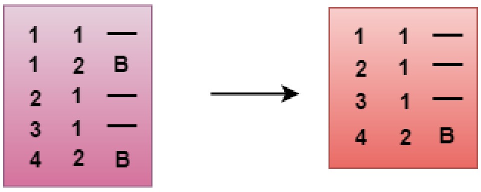
1	2	B
4	2	B

1	1	—
1	2	B
2	1	—
3	1	—
4	2	B

Combined

- The combined table may contain some duplicate data. In the above figure, the combined table of router A contains the duplicate data, so it keeps only those data which has the lowest cost. For example, A can send the data to network 1 in two ways. The first, which uses no next router, so it costs one hop. The second requires two hops (A to B, then B to Network 1). The first option has the lowest cost, therefore it is kept and the second one is dropped.

- The process of creating the routing table continues for all routers. Every router receives the information from the neighbors, and update the routing table.



1	1	—
1	2	B
2	1	—
3	1	—
4	2	B

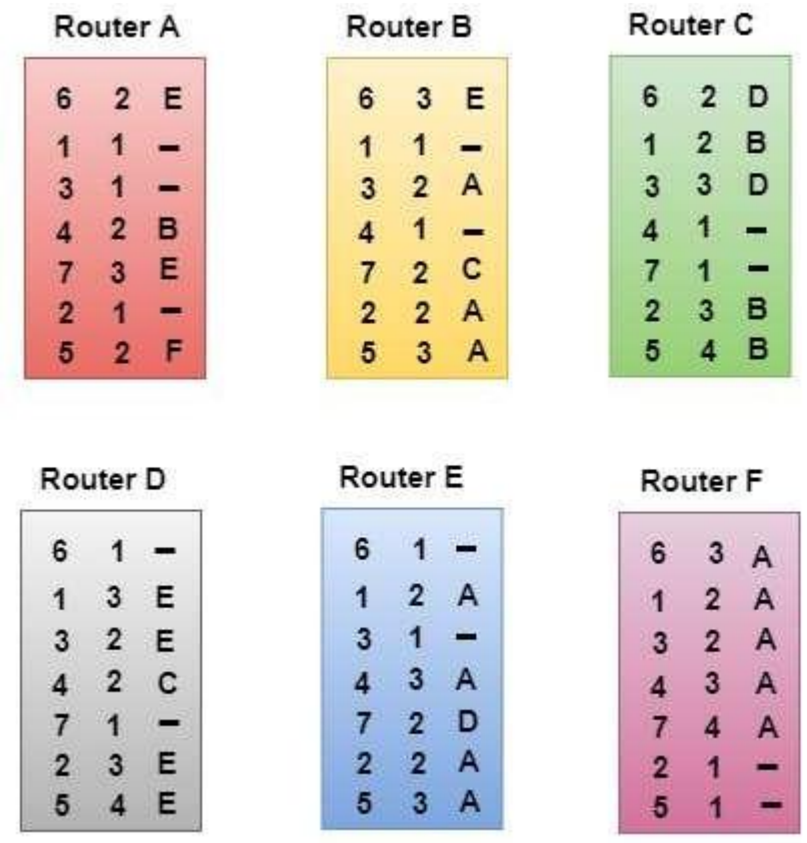
Combined

1	1	—
2	1	—
3	1	—
4	2	B

A's new table

Final routing tables of all the

routers are given below



Router A

6	2	E
1	1	–
3	1	–
4	2	B
7	3	E
2	1	–
5	2	F

Router B

6	3	E
1	1	–
3	2	A
4	1	–
7	2	C
2	2	A
5	3	A

Router C

6	2	D
1	2	B
3	3	D
4	1	–
7	1	–
2	3	B
5	4	B

Router D

6	1	–
1	3	E
3	2	E
4	2	C
7	1	–
2	3	E
5	4	E

Router E

6	1	–
1	2	A
3	1	–
4	3	A
7	2	D
2	2	A
5	3	A

Router F

6	3	A
1	2	A
3	2	A
4	3	A
7	4	A
2	1	–
5	1	–

Link State Routing

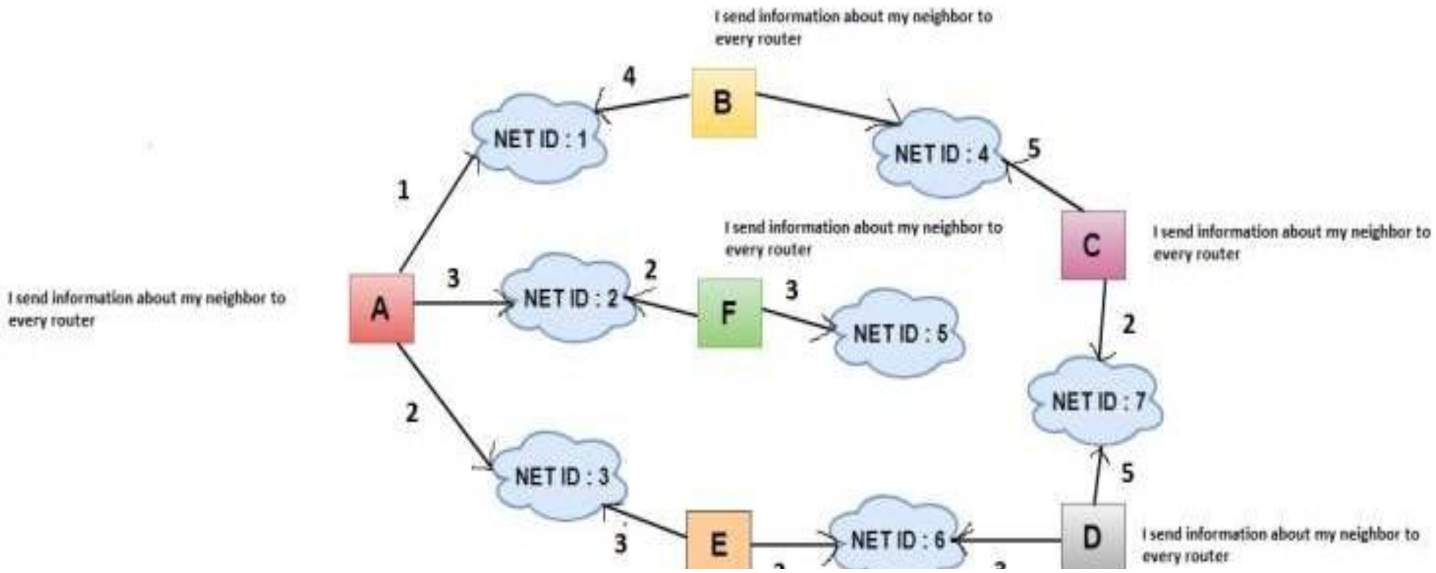
- Link state routing is a technique in which each router shares the knowledge of its neighborhood with every other router in the internetwork.
- **The three keys to understand the Link State Routing algorithm:**
- **Knowledge about the neighborhood:** Instead of sending its routing table, a router sends the information about its neighborhood only. A router broadcast its identities and cost of the directly attached links to other routers.
- **Flooding:** Each router sends the information to every other router on the internetwork except its neighbors. This process is known as Flooding. Every router that receives the packet sends the copies to all its neighbors. Finally, each and every router receives a copy of the same information.

- **Information sharing:** A router sends the information to every other router only when the change occurs in the

  Link State Routing has two phases

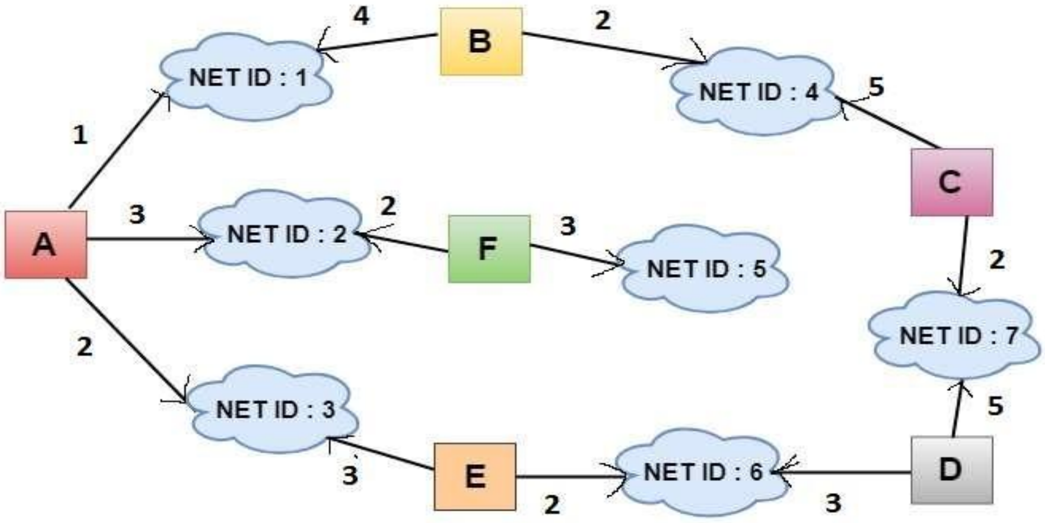
- Reliable Flooding
- **Initial state:** Each node knows the cost of its neighbors.
- **Final state:** Each node knows the entire graph.
- Route Calculation
- Each node uses Dijkstra's algorithm on the graph to calculate the optimal routes to all nodes.
- The Link state routing algorithm is also known as Dijkstra's algorithm which is used to find the shortest path from one node to every other node in the network.
- The Dijkstra's algorithm is an iterative, and it has

INFORMATION SHARING

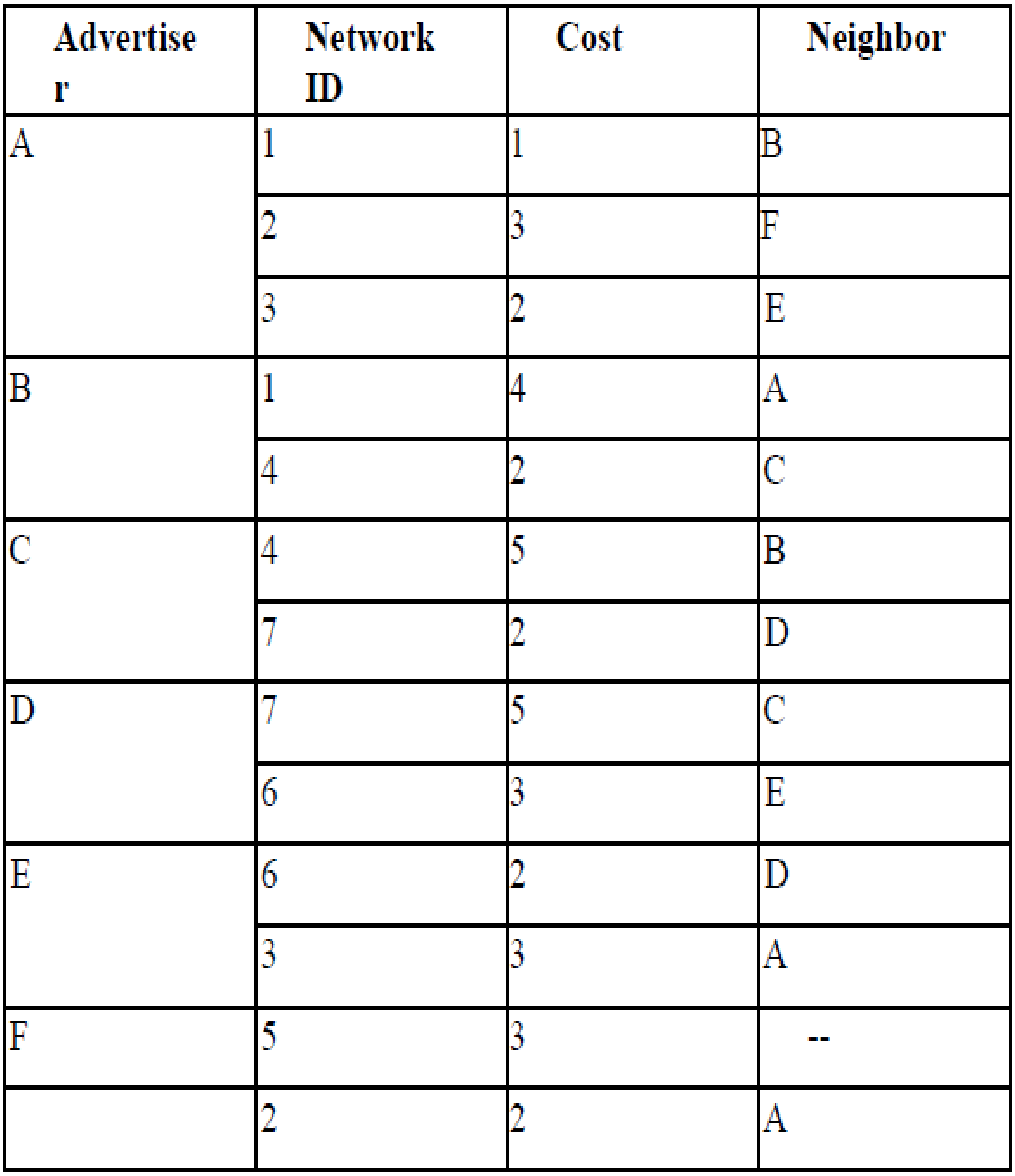

I send information about my neighbor to every router
4
B
NET ID : 1
NET ID : 4
5
1
I send information about my neighbor to every router
C
I send information about my neighbor to every router
I send information about my neighbor to every router
A
3
NET ID : 2
2
F
3
NET ID : 5
2
2
NET ID : 7
5
NET ID : 3
3
E
NET ID : 6
D
I send information about my neighbor to every router

- In link state cost is a weighted value based on factors such as security levels, traffic or state of the link.
- The cost from router A to network ID 1 is different from router A to network ID 2 .

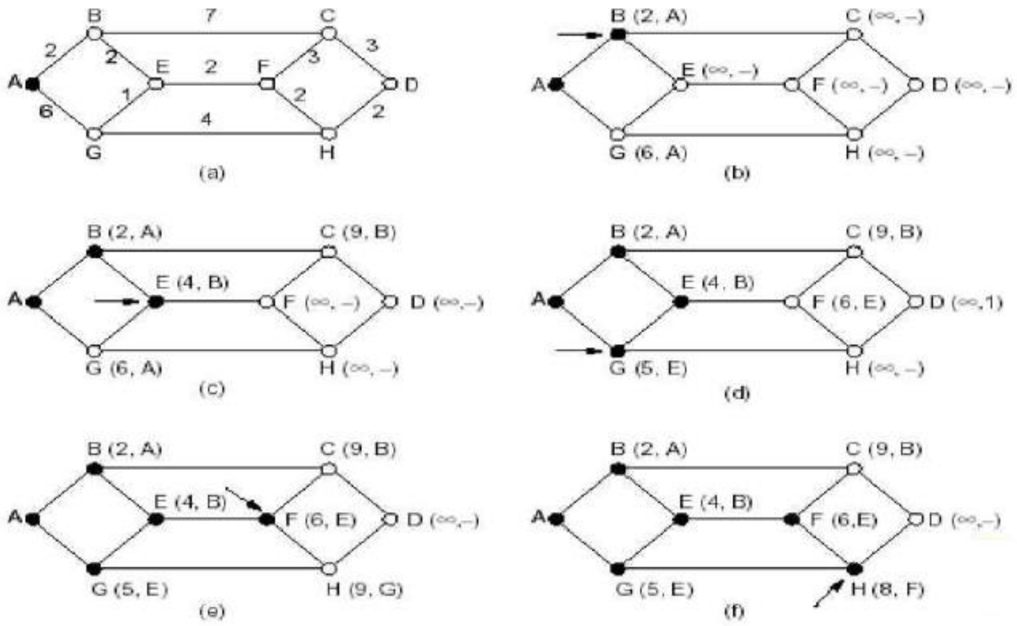
- Two factors how cost is applied
- 1.Cost is applied only by routers not by any other stations on a network.
- 2. Cost is applied as a packet cost



- When a router floods the network with information about its neighbored it is said to be advertising.
- The basis of this advertising is a short packet called link state packet (LSP).
- It contains 4 fields
- 1.ID of advertiser
- 2. ID of destination network
- 3. cost



Advertiser	Network ID	Cost	Neighbor
A	1	1	B
	2	3	F
	3	2	E
B	1	4	A
	4	2	C
C	4	5	B
	7	2	D
D	7	5	C
	6	3	E
E	6	2	D
	3	3	A
F	5	3	--
	2	2	A



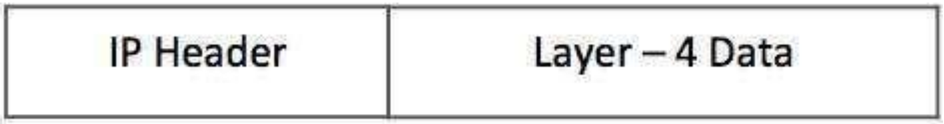
The shortest path from A to D
is:ABEFHD

Internet Protocol Version 4 (IPv4)

- Internet Protocol is one of the major protocols in the TCP/IP protocols suite.
- This protocol works at the network layer of the OSI model and at the Internet layer of the TCP/IP model.
- Thus this protocol has the responsibility of identifying hosts based upon their logical addresses and to route data among them over the underlying network.
- IP provides a mechanism to uniquely identify hosts by an IP addressing scheme. IP uses best effort delivery, i.e. it does not guarantee that packets would be delivered to the destined host, but it will do its best to reach the destination.
- Internet Protocol version 4 uses 32-bit logical address.

IPv4 - Packet Structure

- Internet Protocol being a layer-3 protocol (OSI) takes data Segments from layer-4 (Transport) and divides it into packets.
- IP packet encapsulates data unit received from above layer and add to its own header information.

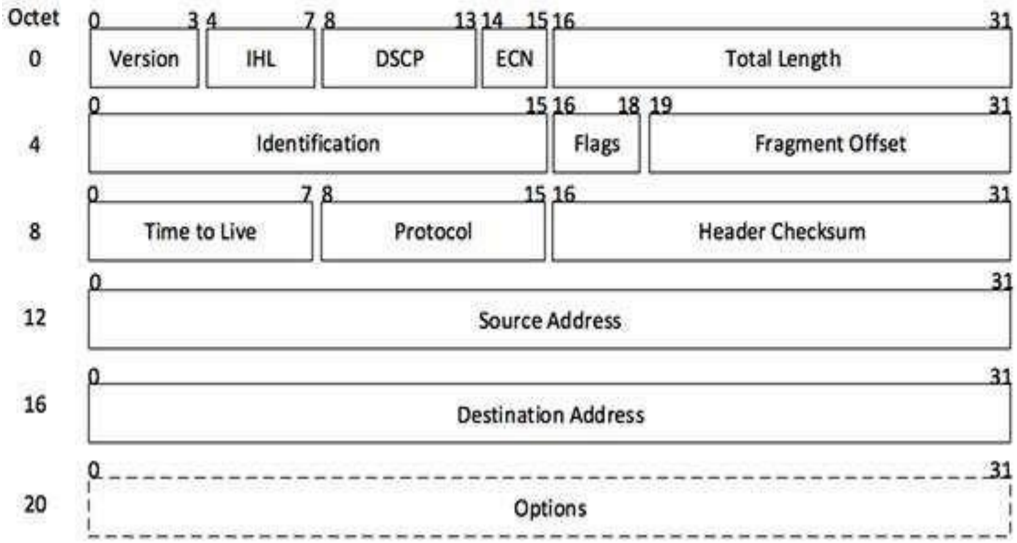


(IP Encapsulation)

- The encapsulated data is referred to as IP Payload. IP header contains all the necessary information to deliver the packet at the other end.

IP header

- IP header includes many relevant information including Version Number, which, in this context, is 4.

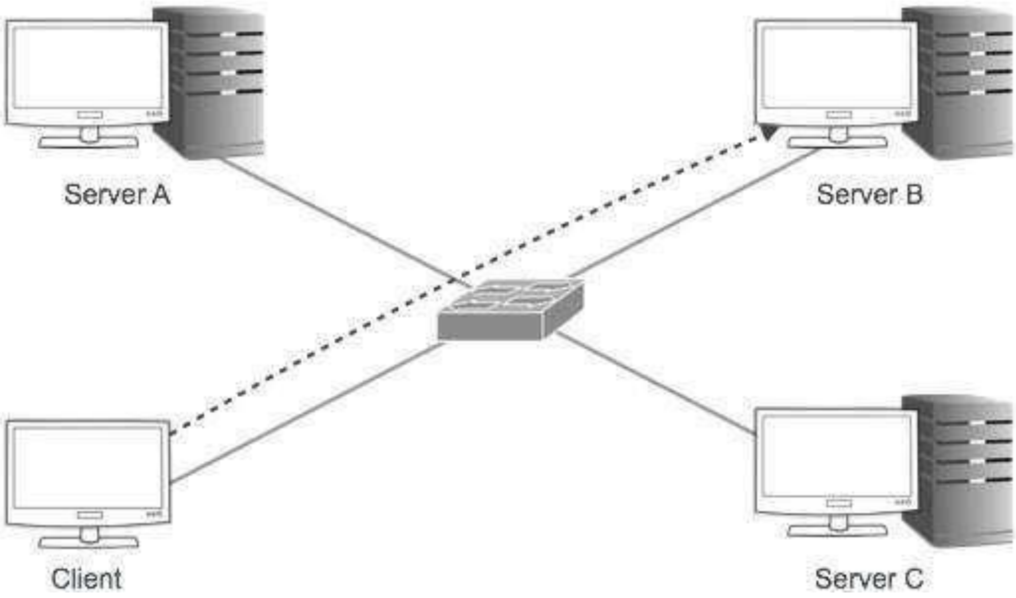


[Image: IP Header]

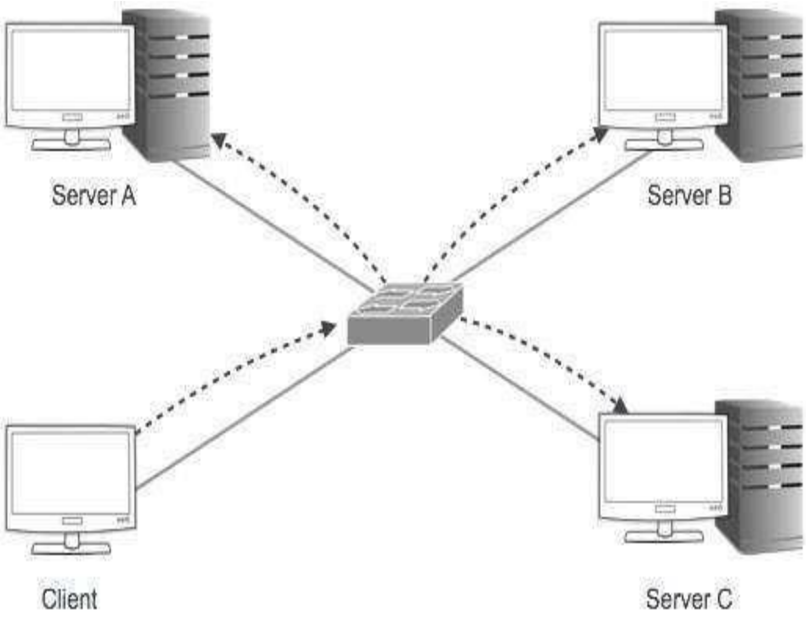
- **Version** − Version no. of Internet Protocol used (e.g. IPv4).
- **IHL** − Internet Header Length; Length of entire IP header.
- **DSCP** − Differentiated Services Code Point; this is Type of Service.
- **ECN** − Explicit Congestion Notification; It carries information about the congestion seen in the route.
- **Total Length** − Length of entire IP Packet (including IP header and IP Payload).
- **Identification** − If IP packet is fragmented during the transmission, all the fragments contain same identification number. to identify original IP packet they belong to.
- **Flags** − As required by the network resources, if IP Packet is too large to handle, these 'flags' tells if they can be fragmented or not. In this 3- bit flag, the MSB is always set to '0'.
- **Fragment Offset** − This offset tells the exact position of the fragment in the original IP Packet.
- **Time to Live** − To avoid looping in the network, every packet is sent with some TTL value set, which tells the network how many routers (hops) this packet can cross. At each hop, its value is decremented by one and when the value reaches zero, the packet is discarded.
- **Protocol** − Tells the Network layer at the destination host, to which Protocol this packet belongs to, i.e. the next level Protocol. For example protocol number of ICMP is 1, TCP is 6 and UDP is 17.
- **Header Checksum** − This field is used to keep checksum value of entire header which is then used to check if the packet is received error-free.
- **Source Address** − 32-bit address of the Sender (or source) of the packet.
- **Destination Address** − 32-bit address of the Receiver (or destination) of the packet.
- **Options** − This is optional field, which is used if the value of IHL is greater than 5. These

- options may contain values for options such as Security, Record Route, Time Stamp, etc. IPv4 - Addressing

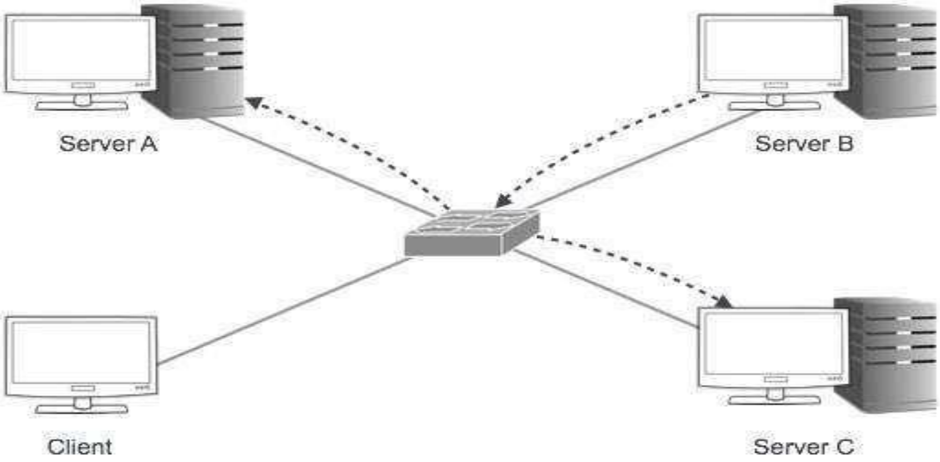
- IPv4 supports three different types of addressing modes
- Unicast Addressing Mode:
- In this mode, data is sent only to one destined host. The Destination Address field contains 32- bit IP address of the destination host. Here the client sends data to the targeted server



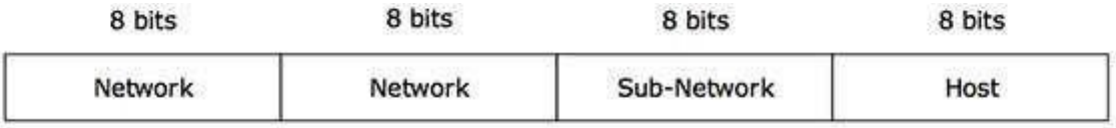
- Broadcast Addressing Mode:
- In this mode, the packet is addressed to all the hosts in a network segment. The Destination Address field contains a special broadcast address, i.e. **255.255.255.255**. When a host sees this packet on the network, it is bound to process it. Here the client sends a packet,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



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



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



8 bits	8 bits	8 bits	8 bits
Network	Network	Sub-Network	Host

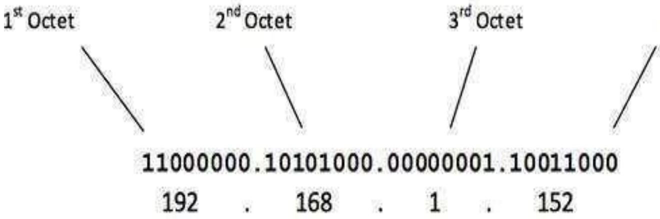
- A single IP address can contain information about the network and its sub-network and ultimately the host. This scheme enables the IP Address to be hierarchical where a network can have many sub-networks which in turn can have

IPv4 - Address Classes

- Internet Protocol hierarchy contains several classes of IP Addresses to be used efficiently in various situations as per the requirement of hosts per network.

- Broadly, the IPv4 Addressing system is divided into five classes of IP Addresses. All the five classes are identified by the first octet of IP Address.

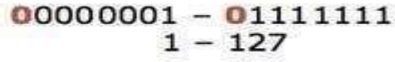
- The first octet referred here is the left most of all. The octets numbered as follows depicting dotted decimal notation of IP Address−



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

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

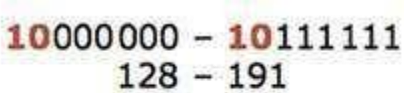
- Class AAddress
- The first bit of the first octet is always set to 0 (zero). Thus the first octet ranges from 1 – 127, i.e.


00000001 – 01111111
1 – 127

- Class A addresses only include IP starting from 1.x.x.x to 126.x.x.x only. The IP range 127.x.x.x is reserved for loopback IP addresses.
- The default subnet mask for Class A IP address is 255.0.0.0 which implies that Class A addressing can have 126 networks (2^7-2) and 16777214 hosts (2^{24}-2).
- Class A IP address formatis thus: **0NNNNNNN**.HHHHHHHH.HHHHHHHH.HHHHHHHH

- Class B Address

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



10000000 - 10111111
128 - 191

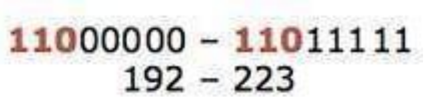
- Class B IP Addresses range from 128.0.x.x to 191.255.x.x. The default subnet mask for Class B is 255.255.x.x.

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

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

- Class C Address

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



11000000 - 11011111
192 - 223

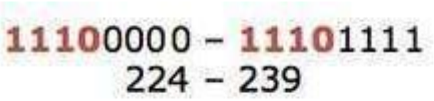
- Class C IP addresses range from 192.0.0.x to 223.255.255.x. The default subnet mask for Class C is 255.255.255.x.

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

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

- Class D Address

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



11100000 - 11101111
224 - 239

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

- Class E Address

- This IP Class is reserved for experimental purposes only for R&D or Study. IP addresses

- in this class ranges from 240.0.0.0 to 255.255.255.254. Like Class D, this class too is

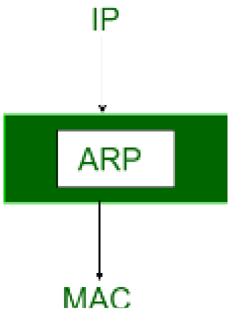
- not equipped with any subnet mask.

Internet Protocol v6 (IPv6)

- IETF(InternetEngineeringTaskForce)has redesigned IP addresses to mitigate the drawbacks of IPv4.
- The new IP address is version 6 which is 128-bit address, by which every single inch of the earth can be given millions of IP addresses.
- Today majority of devices running on Internet are using IPv4 and it is not possible to shift them to IPv6 in the coming days.
- There are mechanisms provided by IPv6, by which IPv4 and IPv6 can co-exist unless the Internet entirely shifts to IPv6 −
- Dual IP Stack
- Tunneling (6to4 and 4to6)
- NAT Protocol Translation

Address Resolution Protocol (ARP)

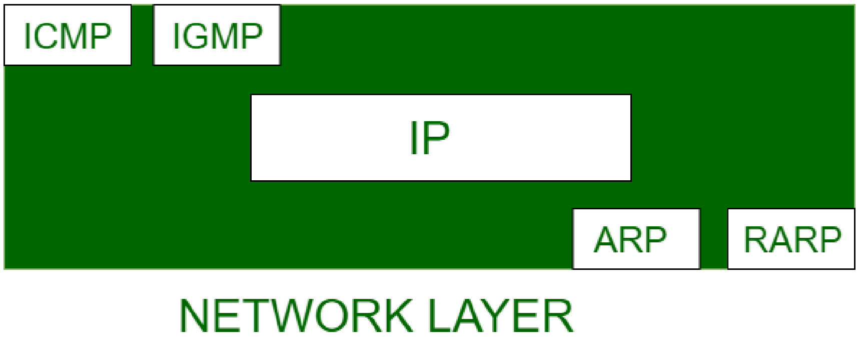
- Most of the computer programs/applications use **logical address (IP address)** to send/receive messages, however the actual communication happens over the **physical address (MAC address)** i.e from layer 2 of OSI model.
- So our mission is to get the destination MAC address which helps in communicating with other devices.
- This is where ARP comes into the picture, its functionality is to translate IP address to physical address.



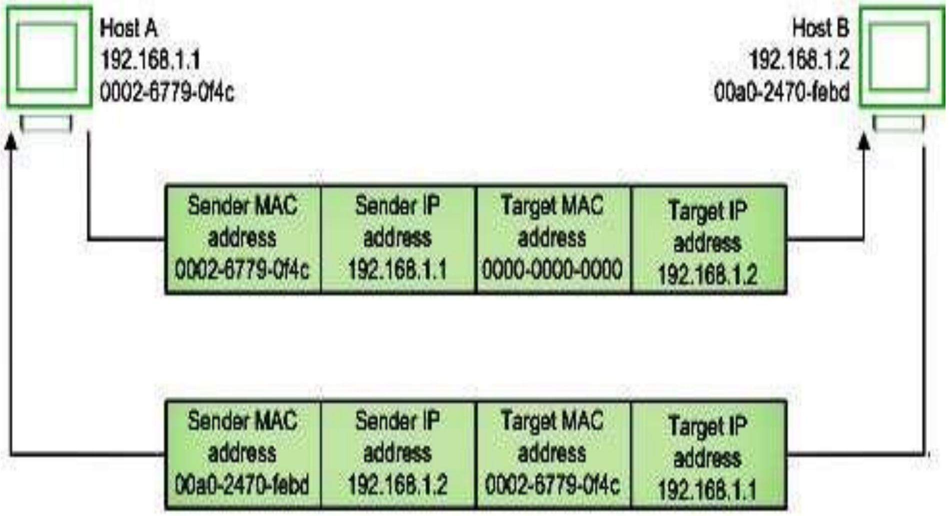
- T

he acronym ARP stands for **Address Resolution Protocol** which is one of the most important protocols of the Network layer in the OSImodel.

Note: ARP finds the hardware address, also known as Media Access Control (MAC) address, of a host from its known IP address.



- Address Resolution Protocol is a communication protocol used for discovering physical address associated with given network address.
- Typically, ARP is a network layer to data link layer mapping process, which is used to discover MAC address for given Internet Protocol Address.
- In order to send the data to destination, having IP address is necessary but not sufficient;
- we also need the physical address of the destination machine. ARP is used to get the physical address (MAC address) of destination machine.



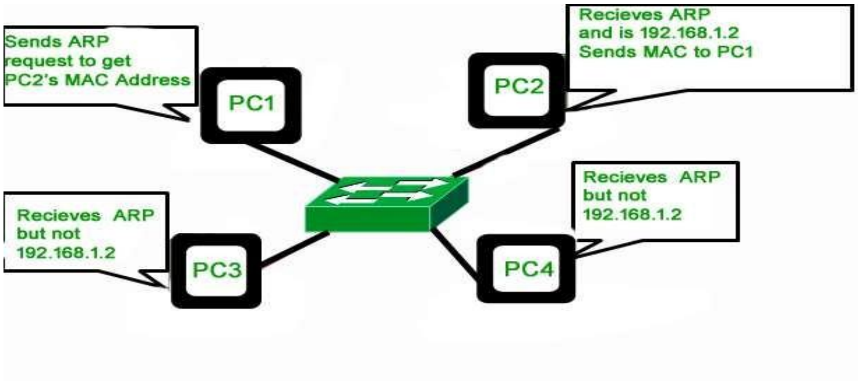
Sender MAC address	Sender IP address	Target MAC address	Target IP address
0002-6779-0f4c	192.168.1.1	0000-0000-0000	192.168.1.2

Sender MAC address	Sender IP address	Target MAC address	Target IP address
00a0-2470-febd	192.168.1.2	0002-6779-0f4c	192.168.1.1

Let’s look at how ARP works.

- Imagine a device wants to communicate with the other over the internet. What ARPdoes?
- It broadcast a packet to all the devices of the source network.

- The devices of the network peel the header of the data link layer from the **protocol data unit (PDU)** called frame and transfers the packet to the network layer (layer 3 of OSI) where the network ID of the packet is validated with the destination IP's network ID of the packet and if it's equal then it responds to the source with the MAC address of the destination, else the packet reaches the gateway of the network and broadcasts packet to the devices it is connected with and validates their network ID
- The above process continues till the second last network device in the path to reach the destination where it gets validated and ARP, in turn, responds with the destination MAC address.
- Before sending the IP packet, the MAC address of destination must be known.
- If not so, then sender broadcasts the ARP-discovery packet requesting the MAC address of intended destination.
- Since ARP-discovery is broadcast, every host inside that network will get this message but the packet will be discarded by everyone except that intended receiver host whose IP is associated.
- Now, this receiver will send a unicast packet with its MAC address (ARP-reply) to the sender of ARP-discovery packet. After the original sender receives the ARP-reply, it updates ARP-cache and start sending unicast message to the destination.


Sends ARP request to get PC2's MAC Address
PC1
Recieves ARP and is 192.168.1.2 Sends MAC to PC1
PC2
Recieves ARP but not 192.168.1.2
PC3
Recieves ARP but not 192.168.1.2
PC4

The important terms associated withARP are

- **ARP Cache:** After resolving MAC address, the ARP sends it to the source where it stores in a table for future reference. The subsequent communications can use the MAC address from the table
- **ARP Cache Timeout:** It indicates the time for which the MAC address in the ARP cache can reside
- **ARP request:** This is nothing but broadcasting a packet over the network to validate whether we came across destination MAC address or not.

  - The physical address of the sender.
  - The IP address of the sender.
  - The physical address of the receiver

– The IP address of the receiver

- **ARP response/reply:** It is the MAC address response that the source receives from the destination which aids in further communication of the data.

- **CASE-1:** The sender is a host and wants to send a packet to another host on the same network.

– Use ARP to find another host's physical address

- **CASE-2:** The sender is a host and wants to send a packet to another host on another network.

– Sender looks at its routing table.

– Find the IP address of the next hop (router) for this destination.

– Use ARP to find the router's physical address

- **CASE-3:** the sender is a router and received a datagram destined for a host on another network.

– Router check its routing table.

– Find the IP address of the next router.

– Use ARP to find the next router's physical address.

- **CASE-4:** The sender is a router that has received a datagram destined for a host in the same network.

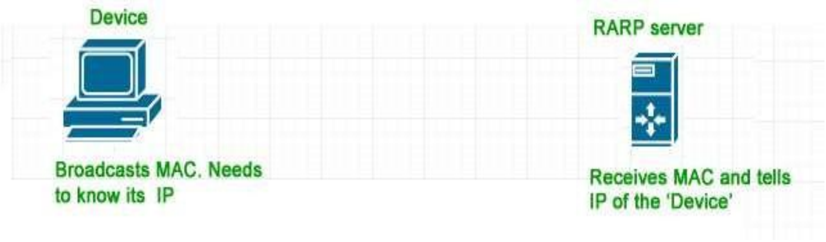
– Use ARP to find this host's physical address.

- **NOTE:** An ARP request is a broadcast, and an ARP response is a Unicast.

Reverse Address Resolution Protocol (RARP)

- Reverse ARP is a networking protocol used by a client machine in a local area network to request its Internet Protocol address (IPv4) from the gateway-router's ARP table.

- The network administrator creates a table in gateway-router, which is used to map the MAC address to corresponding IP address. When a new machine is setup or any machine which don't have memory to store IP address, needs an IP address for its own use. So the machine sends a RARP broadcast packet which contains its own MAC address in both sender and receiver hardware address field.


Device
RARP server
Broadcasts MAC. Needs to know its IP
Receives MAC and tells IP of the 'Device'

- A special host configured inside the local area network, called as RARP-server is responsible to reply for these kind of broadcast packets. Now the RARP server attempt to find out the entry in IP to MAC address mapping table. If any entry matches in table, RARP server send the response packet to the requesting device along with IP address.
- LAN technologies like Ethernet, Ethernet II, Token Ring and Fiber Distributed Data Interface (FDDI) support the Address Resolution Protocol.
- RARP is not being used in today's networks. Because we have much great featured protocols like BOOTP (Bootstrap Protocol) and DHCP (Dynamic Host Configuration Protocol).

TEXT / REFERENCE BOOKS

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SCHOOL OF COMPUTING

DEPARTMENT OF INFORMATION TECHNOLOGY

UNIT – IV – Data communication and Computer networks – SCS1314

TRANSPORT LAYER

TCP- UDP - Connection Management- Flow Control - Retransmission - Congestion Control - Detection and Avoidance

The Transmission Control Protocol

The Transmission Control Protocol (TCP) is one of the most important protocols of Internet Protocols suite. It is most widely used protocol for data transmission in communication network such as internet

FEATURES

- TCP is reliable protocol. That is, the receiver always sends either positive or negative acknowledgement about the data packet to the sender, so that the sender always has bright clue about whether the data packet is reached the destination or it needs to resend it.
- TCP ensures that the data reaches intended destination in the same order it was sent.
- TCP is connection oriented. TCP requires that connection between two remote points be established before sending actual data.
- TCP provides error-checking and recovery mechanism.
- TCP provides end-to-end communication.
- TCP provides flow control and quality of service.
- TCP operates in Client/Server point-to-point mode.

Transmission Control Protocol (TCP)

- Connection oriented
  - Explicit set-up and tear-down of TCP session
- Stream-of-bytes service
  - Sends and receives a stream of bytes, not messages
- Reliable, in-order delivery
  - Checksums to detect corrupted data
  - Acknowledgments & retransmissions for reliable delivery
  - Sequence numbers to detect losses and reorder data
- Flow control
  - Prevent overflow of the receiver’s buffer space
- Congestion control
  - Adapt to network congestion for the greater good

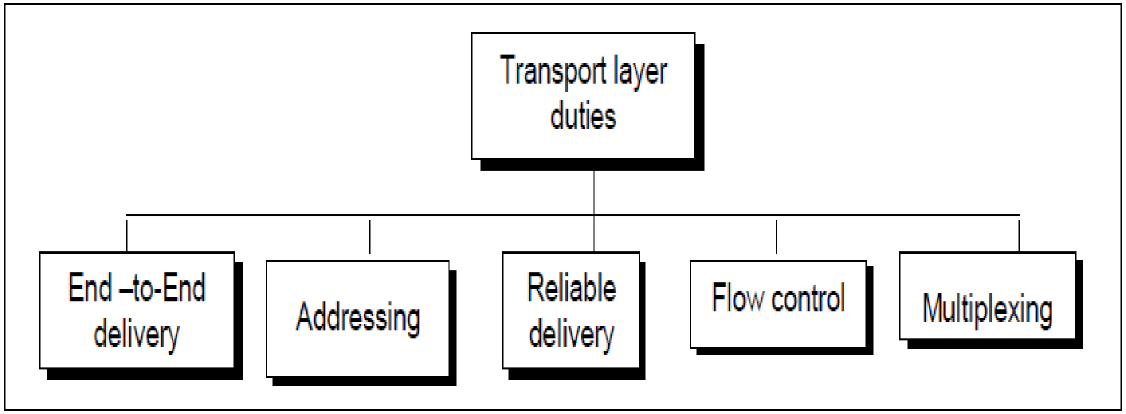


Fig 3.1:TCP features

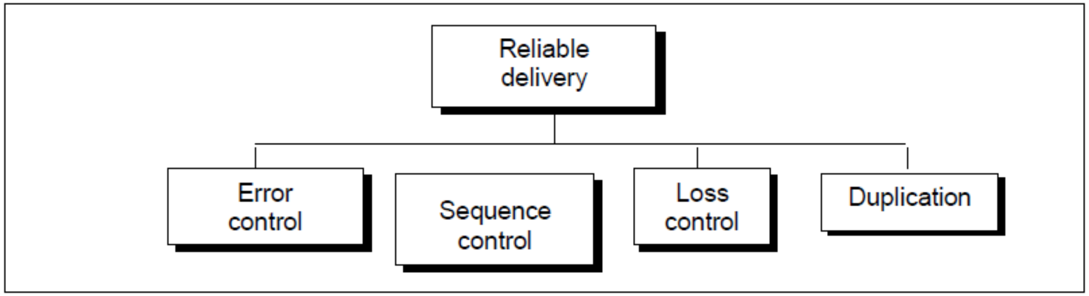


Fig 3.2 TCP reliable delivery

Header

- The length of TCP header is minimum 20 bytes long and maximum 60 bytes.

<table>
<tr><td colspan="5">Source Port</td><td colspan="4">Dest Port</td></tr>
<tr><td colspan="9">Seq No</td></tr>
<tr><td colspan="9">Ack No</td></tr>
<tr><td>Data
Offset</td><td>Resvd</td><td>U</td><td>A</td><td>P</td><td>R</td><td>S</td><td>F</td><td>Window</td></tr>
<tr><td colspan="8">Checksum</td><td>Urgent</td></tr>
<tr><td colspan="9">Options</td></tr>
<tr><td colspan="6"></td><td colspan="3">Pad</td></tr>
<tr><td colspan="9">Data</td></tr>
</table>

Fig3.3:TCPHeader

- **Source Port (16-bits)**- It identifies source port of the application process on the sending device.

- **Destination Port (16-bits)** - It identifies destination port of the application process on the receiving device.

- **Sequence Number (32-bits)** - Sequence number of data bytes of a segment in a session.

- **Acknowledgement Number (32-bits)**- When ACK flag is set, this number contains the next sequence number of the data byte expected and works as acknowledgement of the previous data received.

- **Data Offset (4-bits)**- This field implies both, the size of TCP header (32-bit words) and the offset of data in current packet in the whole TCP segment.

- **Reserved (3-bits)** - Reserved for future use and all are set zero by default.

- **Flags (1-bit each)**

– **NS** - The nonce sum flag is still an experimental flag used to help protect

against accidental malicious concealment of packets from the sender.

– **CWR** - When a host receives packet with ECE bit set, it sets **Congestion**

Windows Reduced to acknowledge that ECE received.

– **ECE** -The first, ECN-Echo (**ECE**) is used to echo back the congestion indication (i.e. signal the sender to reduce the amount of information it sends). The second, Congestion Window Reduced (**CWR**), to acknowledge that the congestion-indication echoing was received.

– **URG** The urgent flag is used to notify the receiver to process the urgent packets before processing all other packets. The receiver will be notified when all known urgent data has been received

- **ACK** - The acknowledgment flag is used to acknowledge the successful receipt of a packet. As we can see from the diagram above, the receiver sends an ACK as well as a SYN in the second step of the three way handshake process to tell the sender that it received its initial packet..
- **PSH** - The push flag is somewhat similar to the URG flag When set, it is a request to the receiving station to PUSH data (as soon as it comes) to the receiving application without buffering it.

- **RST** - Reset flag has the following features:

- It is used to refuse an incoming connection.
- It is used to reject a segment.
- It is used to restart a connection.

- **SYN** - The synchronization **flag** is used as a first step in establishing a three way handshake between two hosts. Only the first packet from both the sender and receiver should have this **flag** set.

- **FIN** - The finished flag means there is no more data from the sender. Therefore, it is used in the last packet sent from the sender

- **Windows Size** - This field is used for flow control between two stations and indicates the amount of buffer (in bytes) the receiver has allocated for a segment, i.e. how much data is the receiverexpecting.

- **Checksum** - This field contains the checksum of Header, Data and Pseudo Headers.

- **Urgent Pointer** - It points to the urgent data byte if URG

  flag is set to 1.

- **Options** - It facilitates additional options which are not covered by the regular header. Option field is always described in 32-bit words. If this field contains data less than 32-bit, padding is used to cover the remaining bits to reach 32-bit boundary.

Connection Management

TCP communication works in Server/Client model. The client initiates the connection and the server either accepts or rejects it. Three-way handshaking is used for connection management.

- **Step 1 (SYN) :** In the first step, client wants to establish a connection with server, so it sends a segment with SYN(Synchronize Sequence Number) which informs server that client is likely to start communication and with what sequence number it starts segments with
- **Step 2 (SYN + ACK):** Server responds to the client request with SYN-ACK signal bits set. Acknowledgement(ACK) signifies the response of segment it received and SYN signifies with what sequence number it is likely to start the segments with

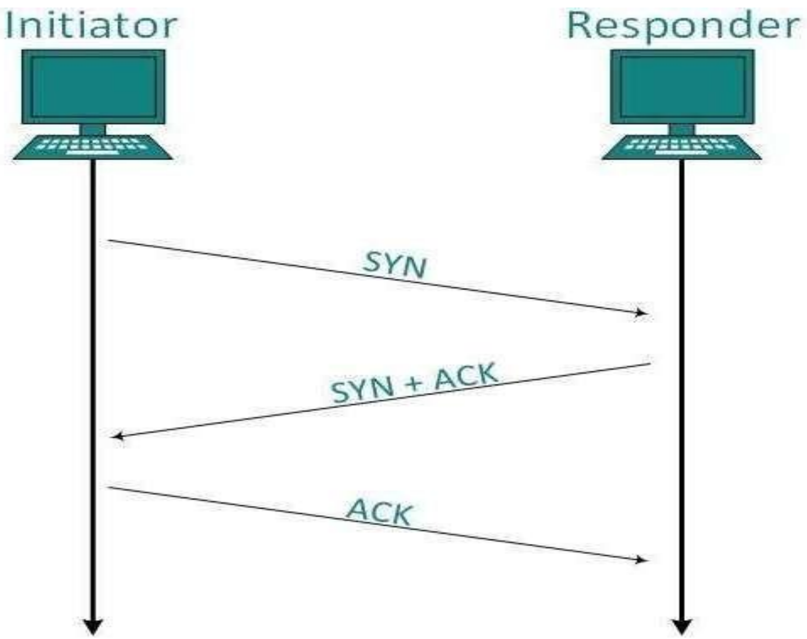


Fig3.4:Threewayhandshake

- **Step 3 (ACK) :** In the final part client acknowledges the response of server and they both establish a reliable connection with which they will start the actual data transfer
- The steps 1, 2 establish the connection parameter (sequence number) for one direction and it is acknowledged. The steps 2, 3 establish the connection parameter (sequence number) for the other direction and it is acknowledged. With these, a full-duplex communication is established.

Bandwidth Management

- TCP uses the concept of window size to accommodate the need of Bandwidth management. Window size tells the sender at the remote end, the number of data byte segments the receiver at this end can receive. TCP uses slow start phase by using window size 1 and increases the window size exponentially after each successful communication.
- For example, the client uses windows size 2 and sends 2 bytes of data. When the acknowledgement of this segment received the windows size is doubled to 4 and next sent the segment sent will be 4 data bytes long. When the acknowledgement of 4-byte data segment is received, the client sets windows size to 8 and so on.
- If an acknowledgement is missed, i.e. data lost in transit network or it received NACK, then the window size is reduced to half and starts again.

Error Control &and Flow Control

- TCP uses sequence numbers to synchronize itself with the remote host. All data segments are sent and received with sequence numbers. The Sender knows which last data segment was received by the Receiver when it gets ACK. The Receiver knows about the last segment sent by the Sender by referring to the sequence number of recently received packet.

- If the sequence number of a segment recently received does not match with the sequence number the receiver was expecting, then it is discarded and NACK is sent back. If two segments arrive with the same sequence number, the TCP timestamp value is compared to make a decision.

Congestion Control

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

Timer Management

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

- **Keep-alive timer**:
- This timer is used to check the integrity and validity of a connection.
- When keep-alive time expires, the host sends a probe to check if the connection still exists.
- **Retransmission timer:**
- This timer maintains stateful session of data sent.
- If the acknowledgement of sent data does not receive within the Retransmission time, the data segment is sent again.

- **Persist timer:**

- TCP session can be paused by either host by sending

  Window Size 0.
- Toresume the session a host needs to send Window Size with some larger value.
- If this segment never reaches the other end, both ends may wait for each other for infinite time.

- When the Persist timer expires, the host re-sends its

  window size to let the other end know.
- Persist Timer helps avoid deadlocks in communication.

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

Error Control in TCP

TCP protocol has methods for finding out corrupted segments, missing segments, out-of-order segments and duplicated segments.

Error control in TCP is mainly done through use of **three simple techniques** :

Checksum – Every segment contains a checksum field which is used to find corrupted segment. If the segment is corrupted, then that segment is discarded by the destination TCP and is considered as lost

Acknowledgement – TCP has another mechanism called acknowledgement to affirm that the data segments have been delivered.

- **Retransmission –** When a segment is missing, delayed to deliver to receiver, corrupted when it is checked by receiver then that segment is retransmitted again. Segments are retransmitted only during two events: when the sender receives three duplicate acknowledgements (ACK) or when a retransmission timer expires.

  - **Retransmission after RTO :** TCP always preserve one retransmission time-out (RTO) timer for all sent but not acknowledged segments. When the timer runs out of time, the earliest segment is retransmitted. Here no timer is set for acknowledgement. In TCP, RTO value is dynamic in nature and it is updated using round trip time (RTT) of segments.
  - **RTT(round trip time)** is the time duration needed for a segment to reach receiver and an acknowledgement to be received to the sender.

- **Retransmission after Three duplicate ACK segments :** RTO method works well when the value of RTO is small. If it is large, more time is needed to get confirmation about whether a segment has delivered or not. Sometimes one segment is lost and the receiver receives so many out-of-order segments that they cannot be saved. In order to solve this situation, three duplicate acknowledgement method is used and missing segment is retransmitted immediately instead of retransmitting already delivered segment. This is a fast retransmission because it makes it possible to quickly retransmit lost segments instead of waiting for timer to end.

User Datagram Protocol (UDP)

- **User Datagram Protocol (UDP)** is a Transport Layer protocol. UDP is a part of Internet Protocol suite, referred as UDP/IP suite. Unlike TCP, it is **unreliable and connectionless protocol.** So, there is no need to establish connection prior to data transfer.
- Though Transmission Control Protocol (TCP) is the dominant transport layer protocol used with most of Internet services; provides assured delivery, reliability and much more but all these services cost us with additional overhead and latency. Here, UDP comes into picture. For the realtime services like computer gaming, voice or video communication, live conferences; we need UDP. Since high performance is needed, UDP permits packets to be dropped instead of processing delayed packets. There is no error

- checking in UDP, so it also save bandwidth. User Datagram Protocol (UDP) is more
- efficient in terms of both latency and bandwidth.

- The User Datagram Protocol (UDP) is simplest Transport Layer communication protocol available of the TCP/IP protocol suite. It involves minimum amount of communication mechanism. UDP is said to be an unreliable transport protocol but it uses IP services which provides best effort delivery mechanism.
- In UDP, the receiver does not generate an acknowledgement of packet received and in turn, the sender does not wait for any acknowledgement of packet sent. This shortcoming makes this protocol unreliable as well as easier on processing.

Features

UDP is used when acknowledgement of data does not hold any significance.

- UDP is good protocol for data flowing in one direction.
- UDP is simple and suitable for query based communications.
- UDP is not connection oriented.
- UDP does not provide congestion control mechanism.
- UDP does not guarantee ordered delivery of data.
- UDP is stateless.
- UDP is suitable protocol for streaming applicationssuch as VoIP, multimedia streaming.

UDP Header

- UDP header is **8-bytes** fixed and simple header, while for TCP it may vary from 20 bytes to 60 bytes. First 8 Bytes contains all necessary header information and remaining part consist of data. UDP port number fields are each 16 bits long, therefore range for port numbers defined from 0 to 65535; port number 0 is reserved. Port numbers help to distinguish different user requests or process.

- **Source Port :** Source Port is 2 Byte long field used to

  identify port number of source.

- **Destination Port :** It is 2 Byte long field, used to identify the port of destined packet.
- **Length :** Length is the length of UDP including header and the data. It is 16-bits field.
- **Checksum :** Checksum is 2 Bytes long field. It is the 16-bit one's complement of the one's complement sum of the UDP header, pseudo header of information from the IP header and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.

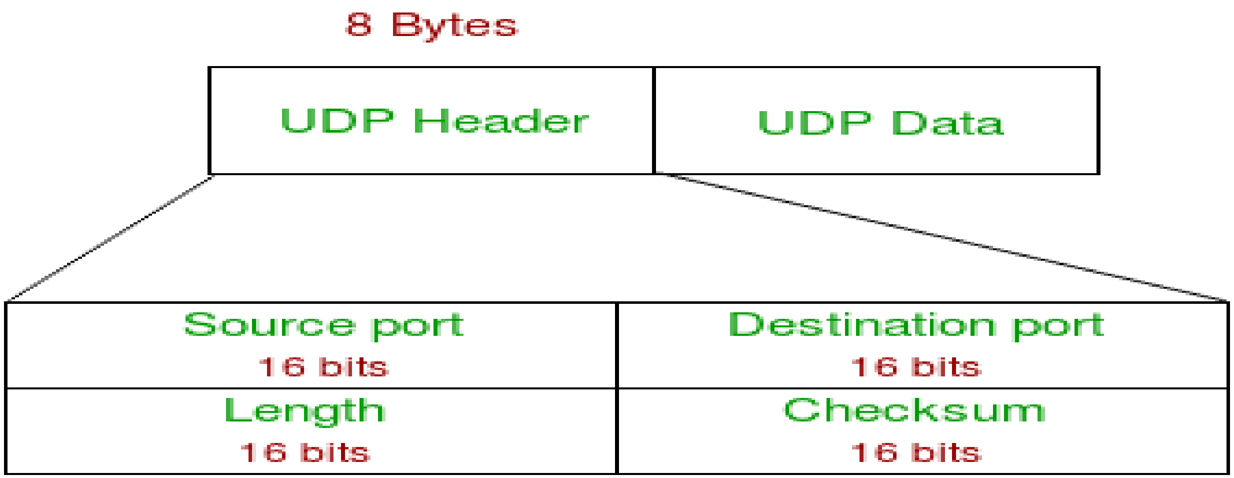


Fig 3.5:UDP Header

Applications of UDP:

- Used for simple request response communication when size of data is less and hence there is lesser concern about flow and error control.
- It is suitable protocol for multicasting as UDP supports packet switching.

- UDP is used for some routing update protocols like RIP(Routing Information Protocol).
- Normally used for real time applications which can not tolerate uneven delays between sections of a received message.
- Following implementations uses UDP as a transport layer protocol:
  - NTP (Network Time Protocol)
  - DNS (Domain Name Service)
  - BOOTP, DHCP.
  - NNP (Network News Protocol)
  - Quote of the day protocol
  - TFTP, RTSP, RIP, OSPF.

- Application layer can do some of the tasks through UDP-
  - Trace Route
  - Record Route
  - Time stamp

- UDP takes datagram from Network Layer, attach its header and send it to the user. So, it works fast.
- Actually UDP is null protocol if you remove checksum field.

What is **congestion**

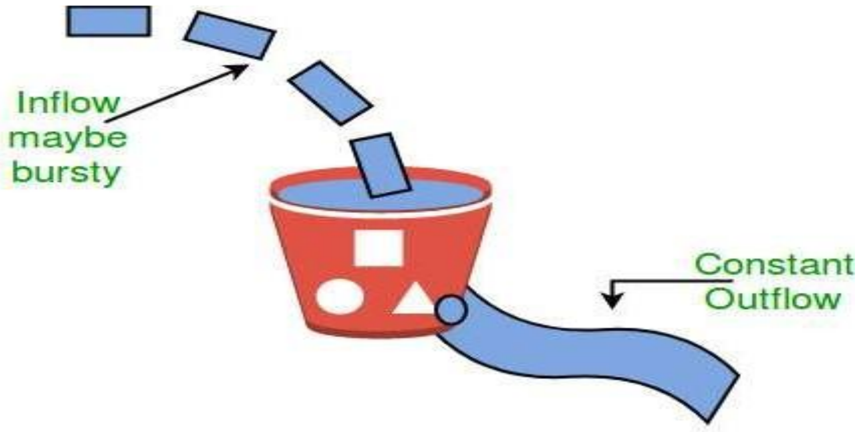
- A state occurring in network layer when the message traffic is so heavy that it slows down network response time.**Effects** of Congestion
- As delay increases, performance decreases.If delay increases, retransmission occurs, making situation worse.

Congestion control algorithms

- **Leaky Bucket Algorithm**

Let us consider an example to understand

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



and is lost.

Fig3.6: Leaky Bucket Algorithm

- Similarly, each network interface contains a leaky bucket and the following **steps** are involved in leaky bucket algorithm:
- When host wants to send packet, packet is thrown into the bucket.
- The bucket leaks at a constant rate, meaning the network interface transmits packets at a constant rate.
  - Bursty traffic is converted to a uniform traffic by the leaky bucket.
  - In practice the bucket is a finite queue that outputs at a finite rate.
  - **Token bucket Algorithm**
    - **Need** of token bucket Algorithm:-

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

- **Steps** of this algorithm can be described as follows:
- In regular intervals tokens are thrown into the bucket. *ƒ*
- The bucket has a maximum capacity. *ƒ*
- If there is a ready packet, a token is removed from the bucket, and the packet is sent.

- If there is no token in the bucket, the packet cannot be sent.

EXAMPLE

- Let’s understand with an example,

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

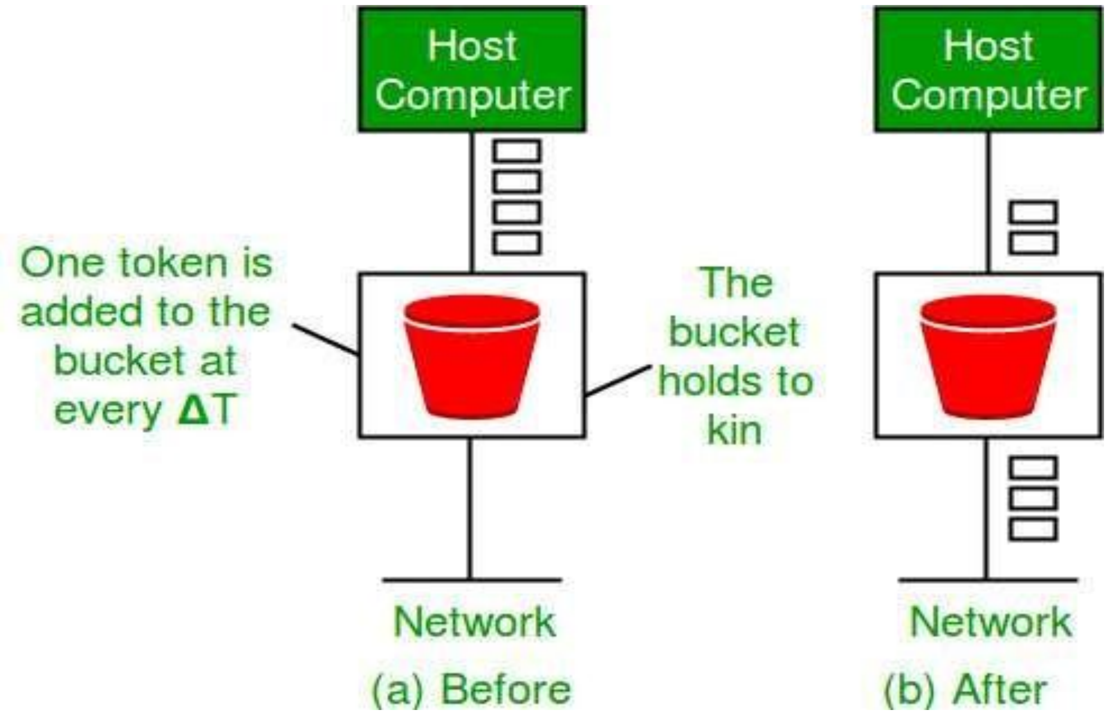


Fig3.6: Token Bucket Algorithm

Ways in which token bucket is superior to leaky bucket

The leaky bucket algorithm controls the rate at which the packets are introduced in the network, but it is very conservative in nature.

- Some flexibility is introduced in the token bucket algorithm. In the token bucket, algorithm tokens are generated at each tick (up to a certain limit).

- For an incoming packet to be transmitted, it must capture a token and the transmission takes place at the same rate.

- Hence some of the busty packets are transmitted at the same rate if tokens are available and thus introduces some amount of flexibility in the system.

Formula:

$M * s = C + \rho * s$ where S – is time taken

M – Maximum output rate ρ – Token arrival rate , C – Capacity of the token bucket in byte

Leaky Bucket vs Token Bucket

- LB discards packets; TB does not. TB discards tokens.
- With TB, a packet can only be transmitted if there are enough tokens
- LB sends packets at an average rate. TB allows for large bursts to be sent faster by speeding up the output.
- TB allows saving up tokens (permissions) to send large bursts. LB does not allow saving.

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SCHOOL OF COMPUTING

DEPARTMENT OF INFORMATION TECHNOLOGY

UNIT – V – Data communication and Computer networks – SCS1314

APPLICATION LAYER

Networking Devices - Repeaters - Switches - Bridges - Routers - Gateways- Domain Name System - FTP - WWW and HTTP - SNMP - SMTP - POP3 - IMAP - MIME.

Networking Devices

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

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

Types of Hub

a. Active Hub:- These are the hubs which have their own power supply and can clean, boost and relay the signal along with the network. It serves both as a repeater as well as wiring centre. These are used to extend the maximum distance between nodes.

b. Passive Hub :- These are the hubs which collect wiring from nodes and power supply from active hub. These hubs relay signals onto the network without cleaning and boosting them and can’t be used to extend the distance between nodes.

c. Intelligent Hub :- It work like active hubs and include remote management capabilities. They also provide flexible data rates to network devices. It also enables an administrator to monitor the traffic passing through the hub and to configure each port in the hub.

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

Types of Bridges

a. Transparent Bridges:- These are the bridge in which the stations are completely unaware of the bridge’s existence i.e. whether or not a bridge is added or deleted from the network, reconfiguration of the stations is unnecessary. These bridges make use of two processes i.e. bridge forwarding and bridge learning.

b. Source Routing Bridges:- In these bridges, routing operation is performed by source station and the frame specifies which route to follow. The hot can discover frame by sending a special frame called discovery frame, which spreads through the entire network using all possible paths to destination.

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

5. Routers – A router is a device like a switch that routes data packets based on their IP addresses. Router is mainly a Network Layer device. Routers normally connect LANs and WANs together and have a dynamically updating routing table based on which they make decisions on routing the data packets. Router divide broadcast domains of hosts connected through it.

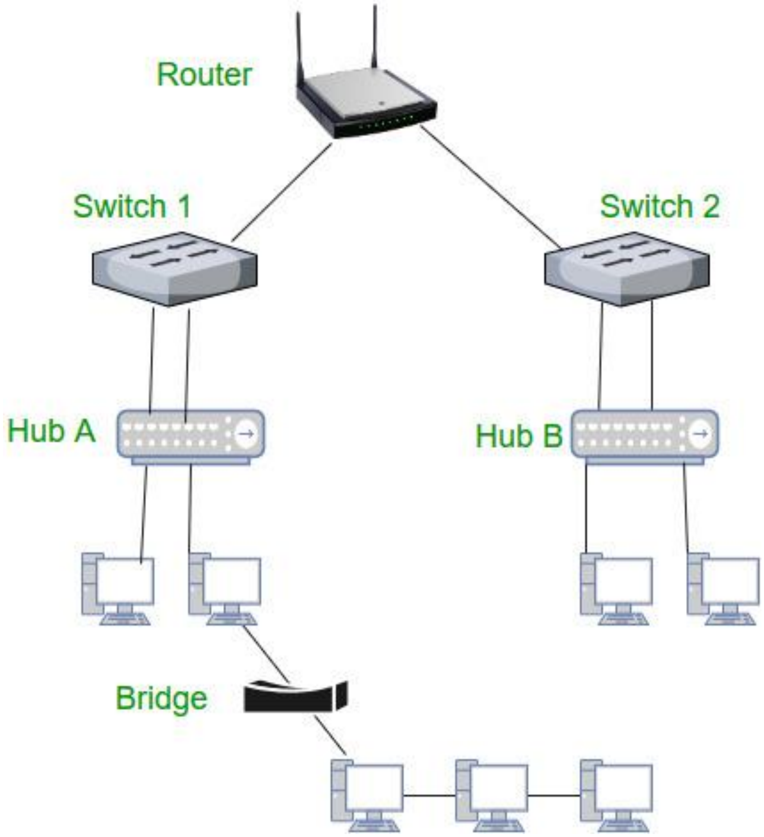

Figure 5.1. Networking Devices

6. Gateway – A gateway, as the name suggests, is a passage to connect two networks together that may work upon different networking models. They basically work as the messenger agents that take data from one system, interpret it, and transfer it to another system. Gateways are also called protocol converters and can operate at any network layer. Gateways are generally more complex than switch or router.

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

DOMAIN NAME SYSTEM

Generally host names, mailboxes and other resources are represented by using ASCII sting such as rgm@vsnl.net.in.But the network itself only understands binary address i.e., the address written in the binary form. So we need some mechanism to convert the ASCII strings to network addresses in binary. It is easy to maintain the host names and their IP addresses in file for a network of few hundred hosts. For a network of thousand hosts it is very difficult.

The Domain Name System, DNS is a distributes data that is used by TCP/IP application to map between host names and IP addresses, and to provide electronic mail routing information. We use the term distributed because no single site on the Internet knows all the information. Each site maintains its own data base information and runs a server program that other systems (clients) across the Internet can query. It is a good example of a TCP/IP client-server application.
The DNS provides the protocol that allows client and server to communicate with each other. DNS is defined in RFC's 1034 and 1035.

The DNS identifies each host on the internet with a unique name that identifies it as unambiguously as its IP address as follows. To map a name onto an IP address, an application program calls a library procedure called the resolver, passing it the name as a parameter. The 'resolver' sends a UDP packet to a local DNS server, which then looks up the name and returns the IP address to the resolver, which then returns it to the caller. To create names that are unique and at the same time decentralized and easy to change, the TCP/IP designers have chosen a hierarchical system made up of a number of labels separated by dots.

THE DNS NAME SPACE

Internet is divided it several hundred top level domains, where each domain covers many hosts. Each domain is partitioned into sub domains, these are further partitioned and so on. Thus DNS is implemented using a tree in which each node represents one possible label of up to 63 characters. The root of the tree is a special node with new label as shown in fig. Any comparison of label considers uppercase and lower-case characters the same i.e., Domain names are case insensitive. The leaves of the tree represent a company/organization and contain thousands of hosts.

Each domain is named by the path from it to the unnamed root. The components in the name are separated by periods (dots), that is domain name of any node in the tree is the list of labels starting at the node, working up to the root using the period (dot) separate the labels.
The domain names that ends with a period is called an absolute domain name or fully qualified domain name(FQDN).An example is vax.ugc,central.edu.

If domain does not end with a period, it is assumed that the name needs to be completed. How the name is completed on the DNS software being used. If the incomplete names consist of two or more labels, it might be considered to be complete. Otherwise, local addition might be added to the right of the name. The name vax might be completed by adding the local suffix.ugc.central.edu.

The right most label in the name corresponds to the level of the tree closest to the root (lowest), and left-most to the level farthest from the root(highest).The tree is divided into three domains: generic, country and reverse as shown in fig 5.2.

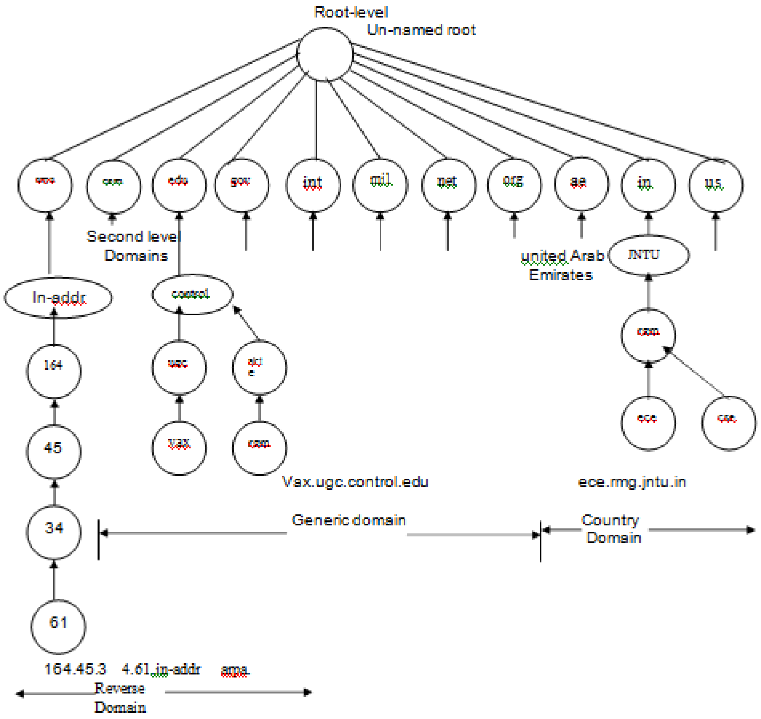

Figure 5.2. Domain Name System

DOMAIN NAME SYSTEM

Generic Domain: The generic domain is also called the organization domain, divides registered hosts according to their generic behaviour. Generic domain names, read left to the right , start with the most specific information about the host(e.g. the name of the workstation) and become more and more general with each label until they reach the rightmost label, which describes the broadcast affiliation of the normal host i.e., the nature of the organization.

The first level of the generic domain convention allows seven possible three character labels describing organization type.

1. Com. commercial organization.

2. edu.: educational institution .
3. gov.: government institution.
4. int.: international organization.
5. mil.: military group.
6. net.: Network support center.
7. org. organizations other than listed above.

Each domain name corresponds to a particular IP address. To find the address, the resolution application begins searching with the first level. As a much is found, a pointer leads to the next level and finally to the associated IP address.

Country Domain: The country domain convention follows the same format as generic domain, but uses two character country abbreviation in place of three character organizational abbreviations at the first level shown in table. Second level labels can be organizational or they can be more specific national designations.

Table: SOME DOMAIN NAME SYSTEM COUNTRY CODE

Country Code	Country Name	Country Code	Country Name
AE AU BE CA CH DE DK ES FI	Arubeme rate Australia Belgium Canada Switzerland Germany Denmark Spain Finland	IN IT JP KW NL NO NZ SE US	India Italy Japan Kuwait Netherlands Norway Newzeland Sweden United States of America
GR	Greece		

Reverse Domain: If we have the IP address and need the domain name, you can reverse domain the functions of DNS.
The domain can be inserted onto the tree in two ways. For example ugc.control.edu could equally be listed under the country domain as cs.yale.ct.us.

To create a new domain, permission is required of the domain in which it will be included. For example, rgm group was started under aicte and is known as rgm.aicte.control.edu. It needs permission from which use manages aicte.control.edu. Naming follows organizational boundaries, not physical networks.

RESOURCE RECORDS
Every domain in the DNS tree maintains a set of Resource Records, which are connected to it. For

a leaf node i.e., single host, the most common resource record is its IP address. When a resolver gives a name to DNS, it gets back called as resource records associated with that name.
The original function of a DNS is to map domain names on to the resource records.

A resource record is a five tuple, in ASCII text they are represented as Domain-name Time-to live type class value.

The domain-name tells the domain to which this record belongs. This is the primary search key used to satisfy queries.
The time-to live field gives information regarding the stability of the record. A large value such as 86-400(number of seconds in one day) indicates that the information is highly stable. The small value such as 60(1 minute) indicates that the information is highly volatile.
The type of field tells what kind of record it is, some of the type records are listed in table 5.3.

S.No	Type	Meaning	Value
1.	SoA A	Start of Authority	Parameter for this zone 32 bit integer
2.	Mx NS	IP address of a hos	Priority
3.	CNAME	Mail Exchange Nam	Name of the server for this domai
4.	PTR TXT	Server Canonica	Domain Name
5.		name Pointer	Alias for an IP address Uninterprete
6.		Text	ASCII text
7.			

1. The SOA record provides name of the primary source of information about (a) name servers zone (b) e-mail address of its administration (c) various flags and (d) various time outs.
2. The record A, holds a 32 bit IP address of the host. If a host connects two or more networks, each case it has one type of a resource record per network connection.
3. The MX record specifies the name of domain prepared to accept e-mail for the specified domain. It allows the host that is not on the internet to receive e-mail from internet sites.
4. NS record specifies Name server.
5. CNAME record specifies allows the aliases to be created.
6. PTR is a regular DNS data type whose interpretation depends on the context on which it is found.
7. The TXT record allows domains to identify themselves in arbitrary way i.e., it is for user convenience.

The fourth field in the general structure of resource record is the class. It may be Internet information, used IN and for non-internet information, other codes are used.
The value field can be number, domain name or an ASCII string.

NAME SERVERS

The Inter network Information center (Inter NIC) manages the top level domain names. The Inter NIC delegates responsibility for assigning names to different organizations. Each organization is responsible for a specific portion of the DNS tree structure. Internet professionals refer to these areas of responsibilities as zones.

Alternatively, the Inter NIC delegates responsibility for assigning names with in a specific zone to specific organizations. Each zone contains some part of the tree and also contains name servers holding the authoritative information about the zone. Each zone contains one primary name server and one or more secondary name servers. Primary name server and one or more secondary name servers. Primary name server gets its information from a file on its disk, the secondary name server and get their information from the primary name server. One or more servers are located outside the zone, for each zone, for reliability. The number of name servers needed in a zone depends on the zone boundaries.

Let us consider an example shown in fig connected with another domain. here a resolver on “ece.rgm.jntu.in” wants to know the IP address of the host “rgm.aicte.control.edu” can be explained in 8 steps.

Step 1: It sends a query to the local name server rgm.jntu.in.This query asks a record of type A and the class IN.
Step 2: If the local name server had no such domain and knows nothing about it, it may ask a few other near by name servers if none of them know, it sends a UDP packet to the server for “edu” given in its database (see fig) edu.server.net.
Step 3: It forwards the request to the name server control.edu.
Step 4: And in turn this forwards the request aicte.control.edu, which has authoritative resource records.

This is the request from client to a server, the resource record requested will work its way back in step 5 to step 8.Once these records get back to rgm.jntu.in name server, they will be entered into a cache/memory. However this information is not authoritative, since changes made at aicte.control.edu will not be propagated to all the memories in the world. For this reason cache should not live too long, so time-to-live field is used in each resource record. It tells the name server how long to cache records.

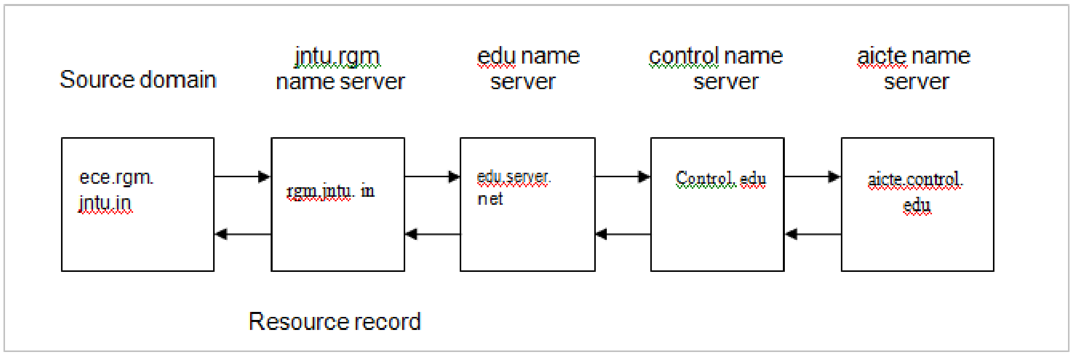

Figure 5.3. Domain Name System Example

FTP

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

Objectives of FTP

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

Why FTP?

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

Mechanism of FTP

The figure 5.4.shows the basic model of the FTP. The FTP client has three components: the user interface, control process, and data transfer process. The server has two components: the server control process and the server data transfer process.

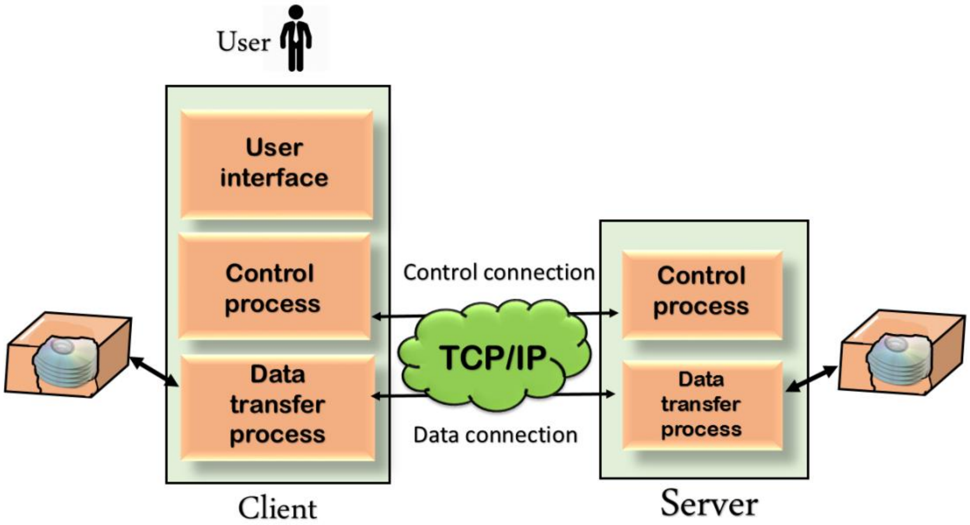

Figure 5.4. The Basic Model Of The FTP

There are two types of connections in FTP:

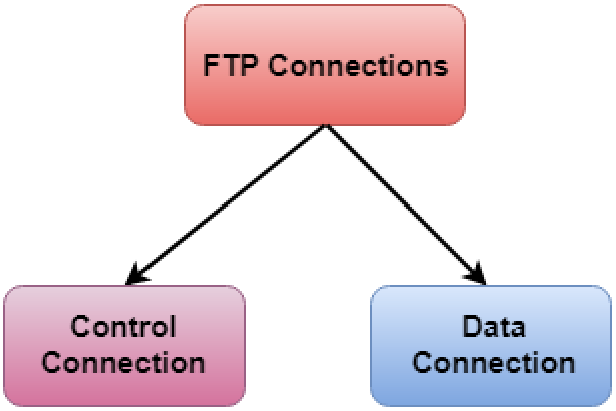

Figure 5.5. The connections in FTP

- ❖ **Control Connection:** The control connection uses very simple rules for communication. Through control connection, we can transfer a line of command or line of response at a time. The control connection is made between the control processes. The control connection remains connected during the entire interactive FTP session.
- ❖ **Data Connection:** The Data Connection uses very complex rules as data types may vary. The data connection is made between data transfer processes. The data connection opens when a command comes for transferring the files and closes when the file is transferred.

FTP Clients

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

Advantages of FTP:

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

Disadvantages of FTP:

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

HTTP

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

Features of HTTP:

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

HTTP Transactions

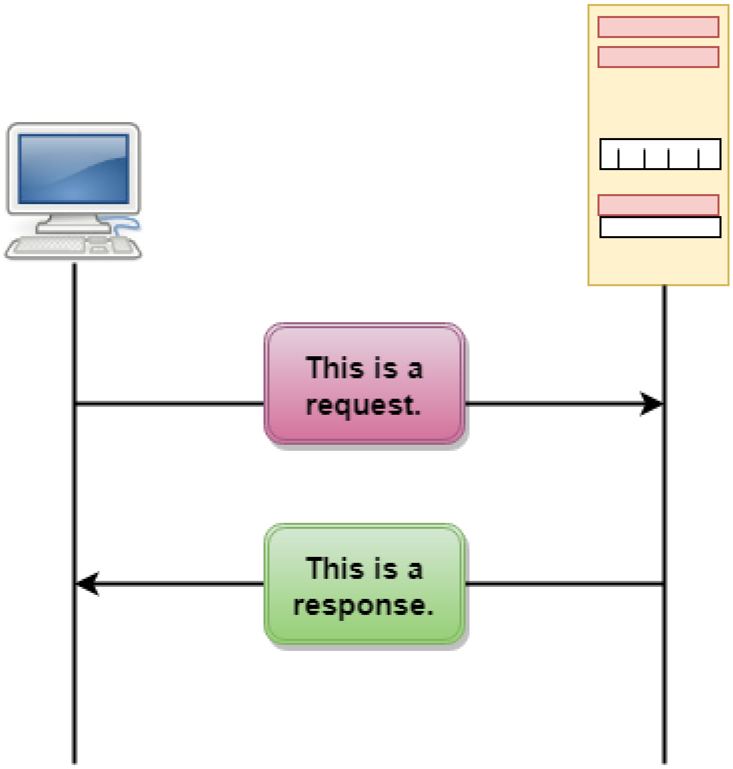

Figure 5.5 HTTP transaction between client and server

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

Messages

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

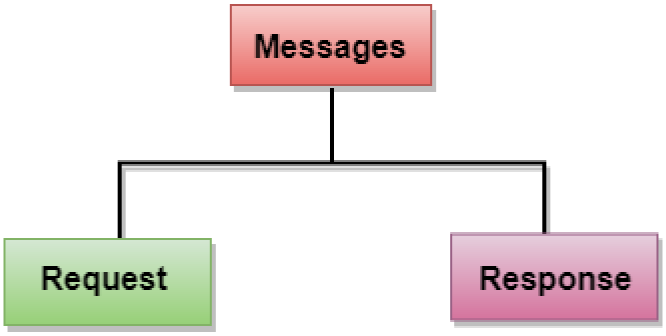

Figure 5.5 HTTP Types

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

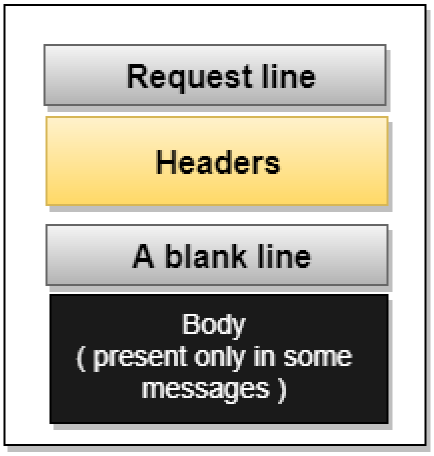

Figure 5.6 Request Message

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

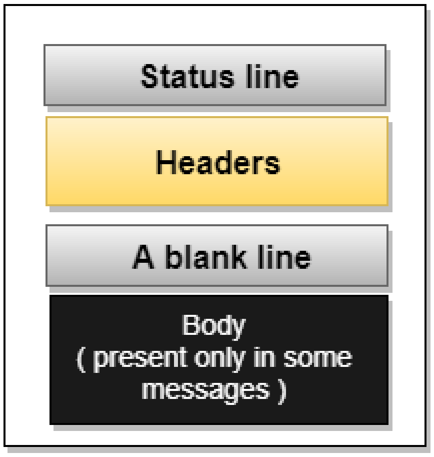

Figure 5.7 Response Message

Uniform Resource Locator (URL)

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

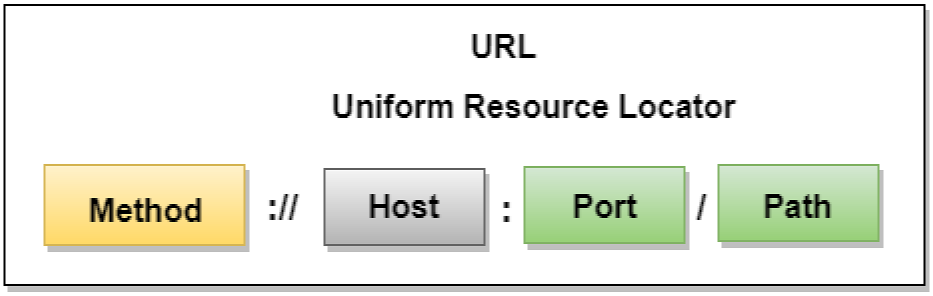

Figure 5.8 Uniform Resource Locator

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

SNMP

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

SNMP Concept

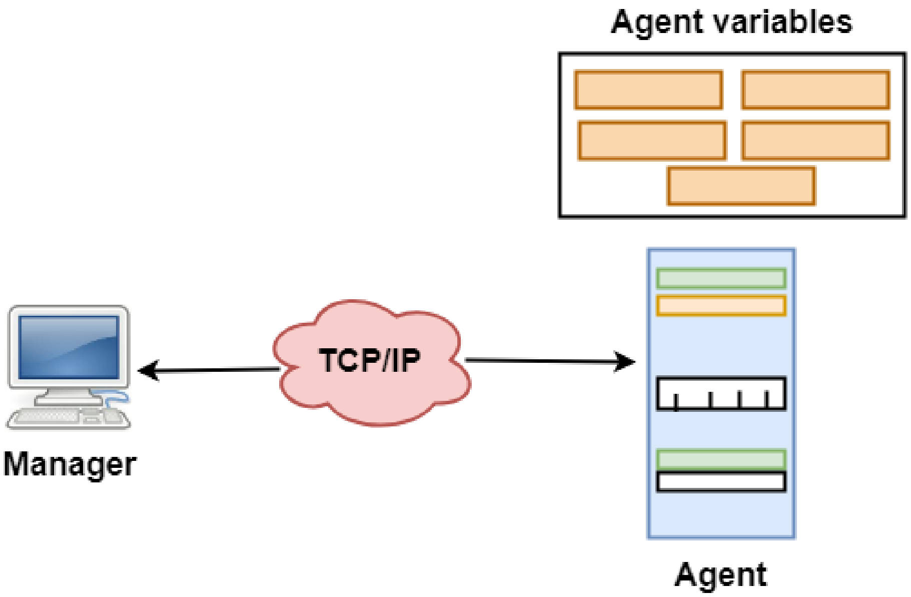

Figure 5.9 SNMP

SNMP has two components Manager and agent.

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

Managers & Agents

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

Management with SNMP has three basic ideas:

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

Management Components

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

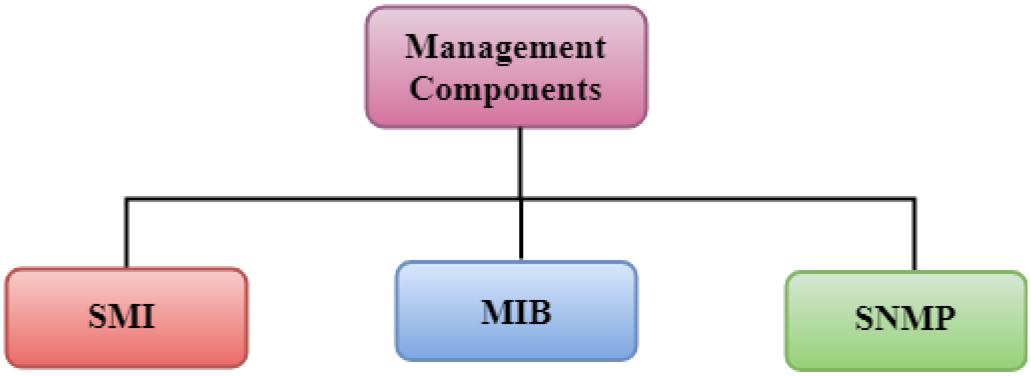

Figure 5.10 Management Components

SMI

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

MIB

The MIB (Management information base) is a second component for the network management.
○ Each agent has its own MIB, which is a collection of all the objects that the manager can manage. MIB is categorized into eight groups: system, interface, address translation, ip, icmp, tcp, udp, and egp. These groups are under the mib object.

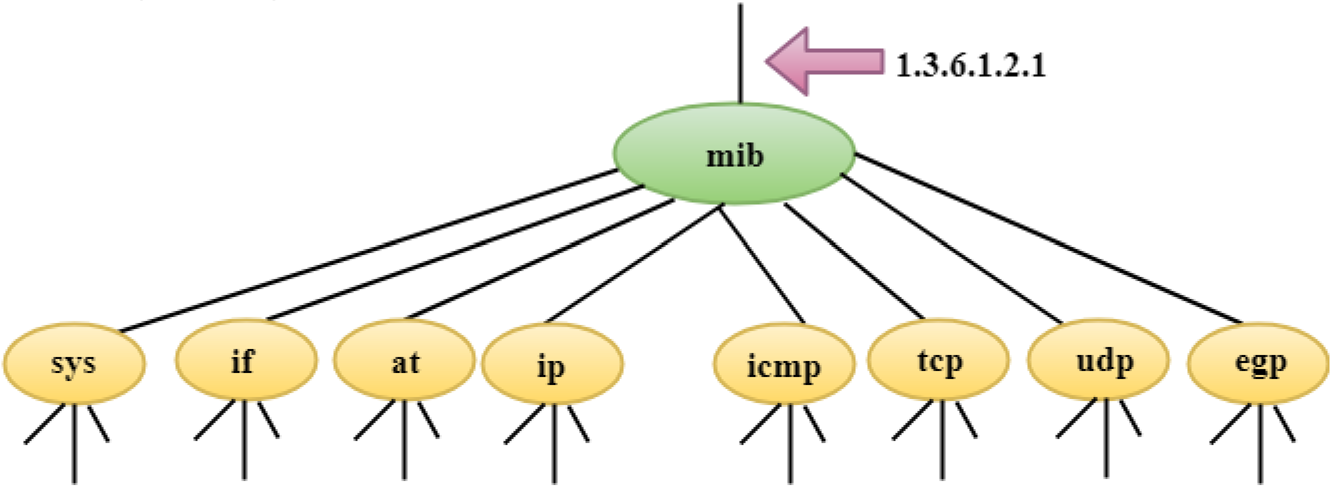

Figure 5.11 MIB

SNMP

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

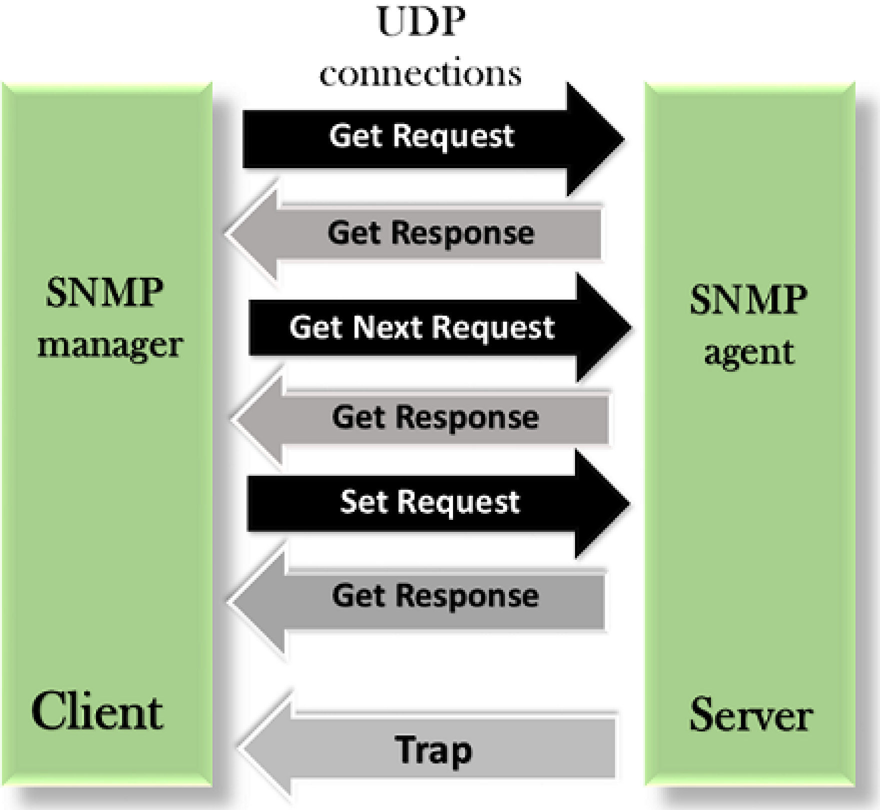

Figure 5.12 SNMP

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

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

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

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

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

POP Protocol

The POP protocol stands for Post Office Protocol. As we know that SMTP is used as a message transfer agent. When the message is sent, then SMPT is used to deliver the message from the client to the server and then to the recipient server. But the message is sent from the recipient server to the actual server with the help of the Message Access Agent. The Message Access Agent contains two types of protocols, i.e., POP3 and IMAP.

How is mail transmitted?

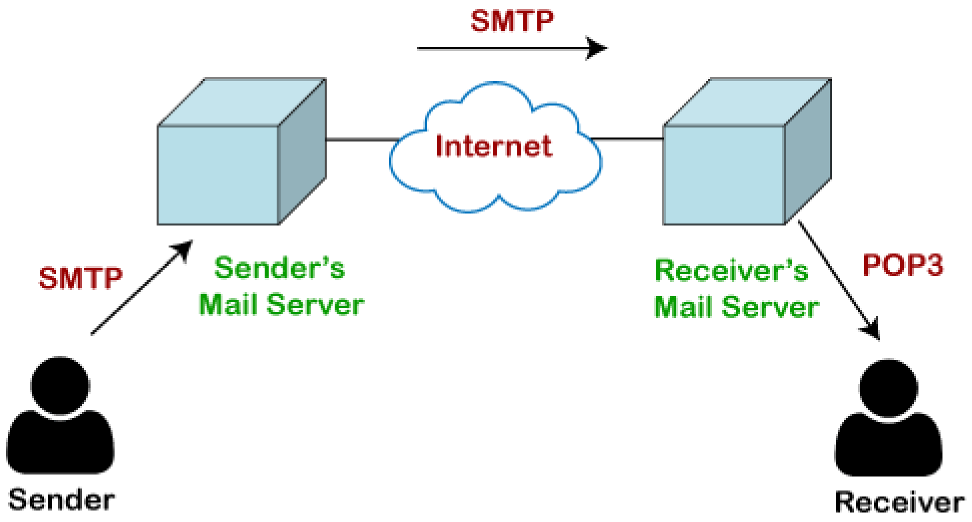

Figure 5.13 How is mail transmitted

Suppose sender wants to send the mail to receiver. First mail is transmitted to the sender's mail server. Then, the mail is transmitted from the sender's mail server to the receiver's mail server over the internet. On receiving the mail at the receiver's mail server, the mail is then sent to the user. The whole process is done with the help of Email protocols. The transmission of mail from the sender to the sender's mail server and then to the receiver's mail server is done with the help of the SMTP protocol. At the receiver's mail server, the POP or IMAP protocol takes the data and transmits to the actual user.

Since SMTP is a push protocol so it pushes the message from the client to the server. As we can observe in the above figure that SMTP pushes the message from the client to the recipient's mail server. The third stage of email communication requires a pull protocol, and POP is a pull protocol. When the mail is transmitted from the recipient mail server to the client which means that the client is pulling the mail from the server.

What is POP3?

The POP3 is a simple protocol and having very limited functionalities. In the case of the POP3 protocol, the POP3 client is installed on the recipient system while the POP3 server is installed on the recipient's mail server.

History of POP3 protocol

The first version of post office protocol was first introduced in 1984 as RFC 918 by the internet engineering task force. The developers developed a simple and effective email protocol known as the POP3 protocol, which is used for retrieving the emails from the server. This provides the facility for accessing the mails offline rather than accessing the mailbox offline. In 1985, the post office protocol version 2 was introduced in RFC 937, but it was replaced with the post office protocol version 3 in 1988 with the publication of RFC 1081. Then, POP3 was revised for the next 10 years before it was published. Once it was refined completely, it got published on 1996.

Although the POP3 protocol has undergone various enhancements, the developers maintained a basic principle that it follows a three-stage process at the time of mail retrieval between the client and the server. They tried to make this protocol very simple, and this simplicity makes this protocol very popular today.

Let's understand the working of the POP3 protocol.

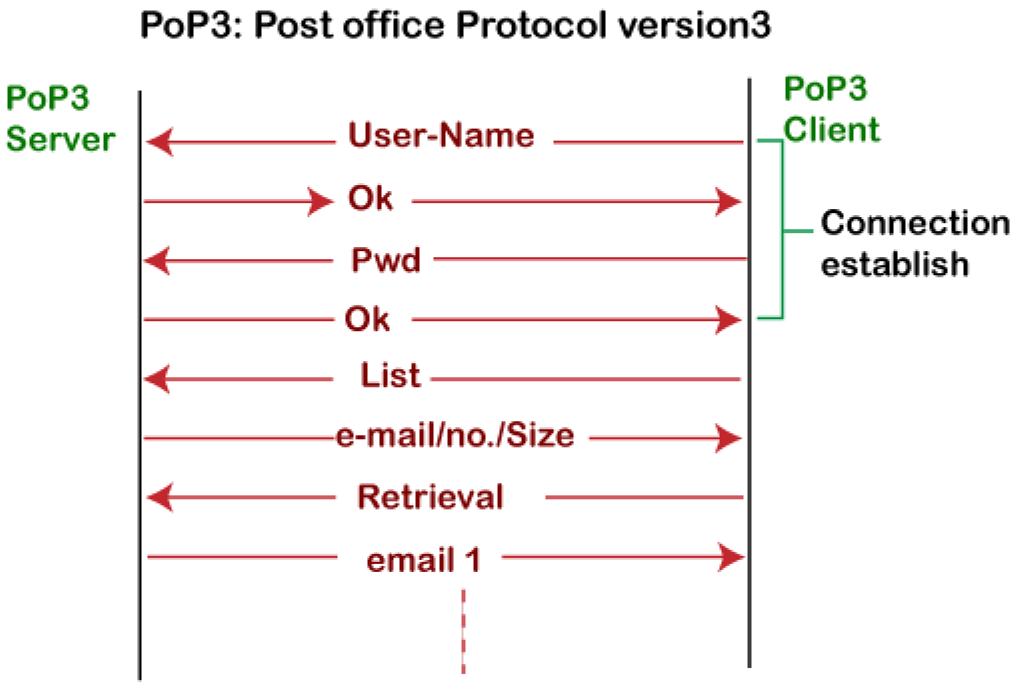

Figure 5.14 POP3

To establish the connection between the POP3 server and the POP3 client, the POP3 server asks for the user name to the POP3 client. If the username is found in the POP3 server, then it sends the ok message. It then asks for the password from the POP3 client; then the POP3 client sends the password to the POP3 server. If the password is matched, then the POP3 server sends the OK message, and the connection gets established. After the establishment of a connection, the client can see the list of mails on the POP3 mail server. In the list of mails, the user will get the email numbers and sizes from the server. Out of this list, the user can start the retrieval of mail.

Once the client retrieves all the emails from the server, all the emails from the server are deleted. Therefore, we can say that the emails are restricted to a particular machine, so it would not be possible to access the same mails on another machine. This situation can be overcome by configuring the email settings to leave a copy of mail on the mail server.

Advantages of POP3 protocol

The following are the advantages of a POP3 protocol:

- It allows the users to read the email offline. It requires an internet connection only at the time of downloading emails from the server. Once the mails are downloaded from the server, then all the downloaded mails reside on our PC or hard disk of our computer, which can be accessed without the internet. Therefore, we can say that the POP3 protocol does not require permanent internet connectivity.
- It provides easy and fast access to the emails as they are already stored on our PC.
- There is no limit on the size of the email which we receive or send.
- It requires less server storage space as all the mails are stored on the local machine.
- There is maximum size on the mailbox, but it is limited by the size of the hard disk.
- It is a simple protocol so it is one of the most popular protocols used today.
- It is easy to configure and use.

Disadvantages of POP3 protocol

The following are the advantages of a POP3 protocol:

- If the emails are downloaded from the server, then all the mails are deleted from the server by default. So, mails cannot be accessed from other machines unless they are configured to leave a copy of the mail on the server.
- Transferring the mail folder from the local machine to another machine can be difficult.
- Since all the attachments are stored on your local machine, there is a high risk of a virus attack if the virus scanner does not scan them. The virus attack can harm the computer.
- The email folder which is downloaded from the mail server can also become corrupted.
- The mails are stored on the local machine, so anyone who sits on your machine can access the email folder.

IMAP Protocol

IMAP stands for Internet Message Access Protocol. It is an application layer protocol which is used to receive the emails from the mail server. It is the most commonly used protocols like POP3 for retrieving the emails.

It also follows the client/server model. On one side, we have an IMAP client, which is a process running on a computer. On the other side, we have an IMAP server, which is also a process running on another computer. Both computers are connected through a network.

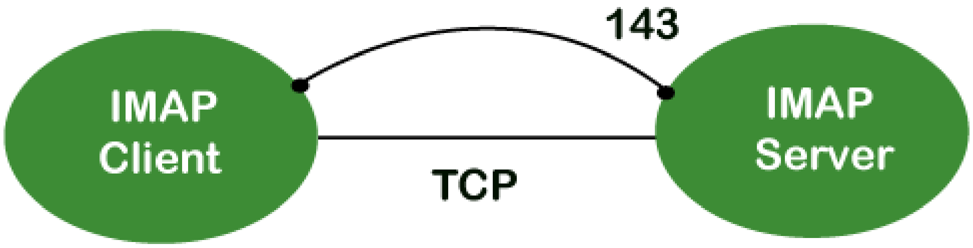

Figure 5.15 IMAP Protocol

The IMAP protocol resides on the TCP/IP transport layer which means that it implicitly uses the reliability of the protocol. Once the TCP connection is established between the IMAP client and IMAP server, the IMAP server listens to the port 143 by default, but this port number can also be changed.

By default, there are two ports used by IMAP:

- Port 143: It is a non-encrypted IMAP port.
- Port 993: This port is used when IMAP client wants to connect through IMAP securely.

Why should we use IMAP instead of POP3 protocol?

POP3 is becoming the most popular protocol for accessing the TCP/IP mailboxes. It implements the offline mail access model, which means that the mails are retrieved from the mail server on the local machine, and then deleted from the mail server. Nowadays, millions of users use the POP3 protocol to access the incoming mails. Due to the offline mail access model, it cannot be used as much. The online model we would prefer in the ideal world. In the online model, we need to be connected to the internet always. The biggest problem with the offline access using POP3 is that the mails are permanently removed from the server, so multiple computers cannot access the mails. The solution to this problem is to store the mails at the remote server rather than on the local server.

The POP3 also faces another issue, i.e., data security and safety. The solution to this problem is to use the disconnected access model, which provides the benefits of both online and offline access. In the disconnected access model, the user can retrieve the mail for local use as in the POP3 protocol, and the user does not need to be connected to the internet continuously. However, the changes made to the mailboxes are synchronized between the client and the server. The mail remains on the server so different applications in the future can access it. When developers recognized these benefits, they made some attempts to implement the disconnected access model. This is implemented by using the POP3 commands that provide the option to leave the mails on the server. This works, but only to a limited extent, for example, keeping track of which messages are new or old become an issue when both are retrieved and left on the server. So, the POP3 lacks some features which are required for the proper disconnected access model.

In the mid-1980s, the development began at Stanford University on a new protocol that would provide a more capable way of accessing the user mailboxes. The result was the development of the interactive mail access protocol, which was later renamed as Internet Message Access Protocol.

IMAP History and Standards

The first version of IMAP was formally documented as an internet standard was IMAP version 2, and in RFC 1064, and was published in July 1988. It was updated in RFC 1176, August 1990, retaining the same version. So they created a new document of version 3 known as IMAP3. In RFC 1203, which was published in February 1991. However, IMAP3 was never accepted by the market place, so people kept using IMAP2. The extension to the protocol was later created called IMAPbis, which added support for Multipurpose Internet Mail Extensions (MIME) to IMAP. This was a very important development due to the usefulness of MIME. Despite this, IMAPbis was never published as an RFC. This may be due to the problems associated with the IMAP3. In December 1994, IMAP version 4, i.e., IMAP4 was published in two RFCs, i.e., RFC 1730 describing the main protocol and RFC 1731 describing the authentication mechanism for IMAP 4. IMAP 4 is the current version of IMAP, which is widely used today. It continues to be refined, and its latest version is actually known as IMAP4rev1 and is defined in RFC 2060. It is most recently updated in RFC 3501.

IMAP Features

IMAP was designed for a specific purpose that provides a more flexible way of how the user accesses the mailbox. It can operate in any of the three modes, i.e., online, offline, and disconnected mode. Out of these, offline and disconnected modes are of interest to most users of the protocol.

The following are the features of an IMAP protocol:

- Access and retrieve mail from remote server: The user can access the mail from the remote server while retaining the mails in the remote server.
- Set message flags: The message flag is set so that the user can keep track of which message he has already seen.
- Manage multiple mailboxes: The user can manage multiple mailboxes and transfer messages from one mailbox to another. The user can organize them into various categories for those who are working on various projects.
- Determine information prior to downloading: It decides whether to retrieve or not before downloading the mail from the mail server.
- Downloads a portion of a message: It allows you to download the portion of a message, such as one body part from the mime-multi part. This can be useful when there are large multimedia files in a short-text element of a message.
- Organize mails on the server: In case of POP3, the user is not allowed to manage the mails on the server. On the other hand, the users can organize the mails on the server according to their requirements like they can create, delete or rename the mailbox on the server.
- Search: Users can search for the contents of the emails.
- Check email-header: Users can also check the email-header prior to downloading.
- Create hierarchy: Users can also create the folders to organize the mails in a hierarchy.

IMAP General Operation

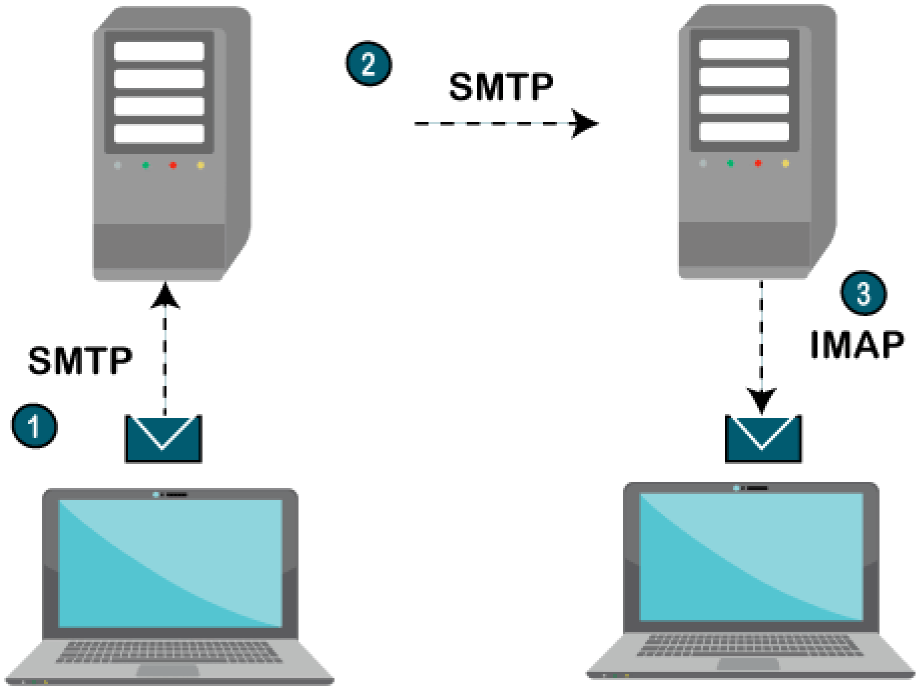

Figure 5.16 IMAP Protocol

1. The IMAP is a client-server protocol like POP3 and most other TCP/IP application protocols. The IMAP4 protocol functions only when the IMAP4 must reside on the server where the user mailboxes are located. In c the POP3 does not necessarily require the same physical server that provides the SMTP services. Therefore, in the case of the IMAP protocol, the mailbox must be accessible to both SMTP for incoming mails and IMAP for retrieval and modifications.
2. The IMAP uses the Transmission Control Protocol (TCP) for communication to ensure the delivery of data and also received in the order.
3. The IMAP4 listens on a well-known port, i.e., port number 143, for an incoming connection request from the IMAP4 client.

Let's understand the IMAP protocol through a simple example.

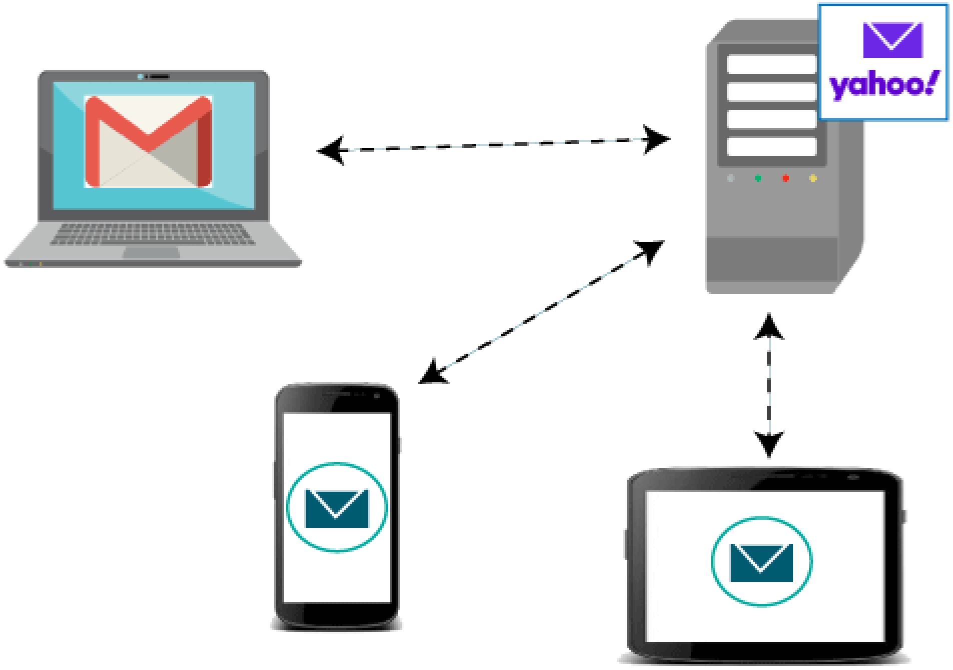

Figure 5.16 IMAP Protocol Example

The IMAP protocol synchronizes all the devices with the main server. Let's suppose we have three devices desktop, mobile, and laptop as shown in the above figure. If all these devices are accessing the same mailbox, then it will be synchronized with all the devices. Here, synchronization means that when mail is opened by one device, then it will be marked as opened in all the other devices, if we delete the mail, then the mail will also be deleted from all the other devices. So, we have synchronization between all the devices. In IMAP, we can see all the folders like spam, inbox, sent, etc. We can also create our own folder known as a custom folder that will be visible in all the other devices.

Multipurpose Internet Mail Extensions(MIME):

This is the solution defined in 1341 and updated in 1521 for the following problems.

- Messages in languages with accents.
- Messages in non Latin alphabets.
- Messages in languages with out alphabets.
- Messages not containing text at all.

The basic idea of MIME is to continue the use of RFC 822 format, but to add structure to the message body defined encoding rules for non ASCII formats. The MIME messages can be sent using the existing mail programs, and protocols.

The MIME defines five new message header

MIME-Version: It tells the use agent receiving the message that it is dealing with a MIME message, and which version of MIME it uses.

Content-Description: It tells what is there in the message, this header helps the recipient whether it is worth decoding and reading the message.

Content-Transfer Encoding: It tells how the body is wrapped for transmission through a network that may object to most characters other than letters, numbers and punctuation marks.

Content-Type: It specifies the nature of the message body. Seven types are defined in RFC 1521, each of which has one or more sub types. The type and sub type are separated by a slash. The sub type must be given explicitly in the header, no defaults are provided. Table shows the list of types and sub types.

TYPE AND SUB TYPE FIELDS DEFINED IN RFC 1521

S.No	Type	Sub Type	Meaning
1.	Text	Plain HTML Rich text	Unformated text Hyper text mark up language Allows a simple mark up language to included in the text (standardized general ma up language (SGML)
2.	Image	GIF JPEG PNG	To transmit still pictures in GIF format To transmit still pictures in JPEG format To transmit still pictures in portable network graphics
3.	Audio	au Basic aiff	Sun micro systems sound Audiable sound Apple sound
4.	Video	sgi.movie MPEG avi	Silicon graphics movie Visual information, the video format is moving picture experts group MPEG Microsoft audio video interleaved
5.	Application	Octet stream Post Script tex	It is a sequence of uninterrupted bytes Which refers the postscript language produced by Adobe systems and widely used for describing printed pages. TEX document.
6.	Message	RFC822 Partial External	A MIME RFC-822 message (ASCII characters message) Break and encapsulated message up into pieces and send them separately. Used for very long message (i.e., video films)
7.	Multiport	Mixed Alternative Parallel Digest	Each part to be different with no additional structure imposed Each part must contain the same message but expressed in a different medium or encoding. All parts must be viewed simultaneously Many messages are packed together into composite message.

MESSAGE TRANSFER

The message Transfer system, MTS is concerned with relaying messages from originator to the recipent.The simplest way to do this is to establish a transport connection from source machine to the destination machine and just transfer the message.

Mail servers are from the core of the e-mail infrastructure.Each recipient has a mail box, located in one of the mail servers.A typical message starts its journey in the sender's user agent, travels to the sender's main server, and then travels to the recipient mail server where it is deposited in the recipient mail box.

A mail server needs to be running all the time, waiting for e-mail messages and routing them approximately.If a mail server crashes or down for an extended period(3-4 days), e-mail can be lost.There may be a limitation on the size of mail box.Generally once this limit is reached, new incoming messages are refused until you free up space by deleting some messages.

SIMPLE MAIL TRANSFER PROTOCOL-SMTP

The simple mail transfer protocol (SMTP) is the principal application layer protocol for internet e- mail. It is simple ASCII protocol. It uses the reliable data transfer service of TCP to transfer mail from the sender's mail server to the recipient's mail server. In most application protocols SMTP has two sides: a client side, which executes on the sender's mail server and a server side-which executes on the recipient mail server. When a mail server sends a mail (to other mail server), it acts as a client SMTP.When a mail server receives a mail (from other mail server), it acts as an SMTP server.

The SMTP defined in RF821, is at the heart of Internet e-mail.SMTP is much older than HTTP.To illustrate the basic operation of SMTP, let's walkthrough a common scenario. Suppose Ramu wants to send Raju a simple ASCII message.

- ❖ Ramu invokes his user agent for e-mail, provides Raju's e-mail address(example Raju@some school.edu) composes a message, and instructs the user agent to send the message.
- ❖ Ramu's user agent sends the message to his mail server, where it is placed in a message queue.
- ❖ The client side of SMTP, running on Ramu's mail server, sees the message in the message queue.It opens a TCP connection to a SMTP running Raju's mail server.
- ❖ After some initial SMTP hand shaking, the SMTP client sends Ramu's message into the TCP connection.
- ❖ At Raju's mail server host, the server side of SMTP recives the message.Raju's mail server then places the message in Raju's mail box.
- ❖ Raju invokes his user agent to read the message at his convenience. The scenario is summarized in fig.5.29

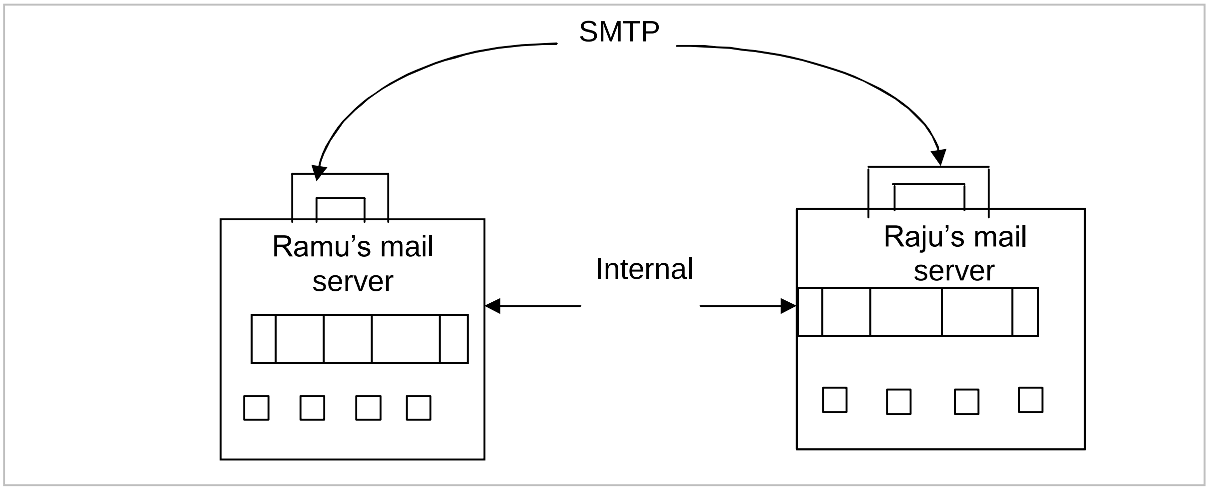

Figure 5.16 SMTP

RAMU'S MAIL SERVER TRANSFERS RAMU'S MESSAGE TO RAJU'S MAIL SERVER

Let us now take closer look at how SMTP transfers a message from a sending mail server to a receiving mail server.

We will see that the SMTP protocol has many similarities with protocols that are used for face-to-face human interaction.

- ❖ The client SMTP has TCP to establish a connection on port 25 to server SMTP.If server is down, the clients tries again later. Once the connection is established, the server and client perform some application layer handshaking. During this SMTP handshaking phase, the SMTP client indicates the e-mail address of the sender and the e-mail address of the recipient. Once the SMTP client and server have introduced themselves to each other, the client sends the message, SMTP can count on the reliable data transfer service of TCP to get the message to the server without errors. The client then repeats this process over the same TCP connection if it has other message to send to the server; otherwise it instructs TCP to close the connection.
- ❖ Even though the SMTP protocol is well defined, a few problems can still arise. These are.
- ❖ **Related to the Message Length :** Some older implementations cannot handle messages exceeding 64kB.
- ❖ **Related to Time Outs :** If the client and server have different time-outs, one of them may give up while the other is still busy, unexpectedly terminating the connection.
- ❖ Infinite mail storms can be triggered .

To get around some of these problems, extended SMTP (ESMTP) has been defined in RFC1425.

E-mail Gateways: E-mail using SMTP works best when both the sender and receiver on the internet and can support TCP connections between sender and receiver.However many machines that are not on the internet)because of security problem) still want to send and recive e- mail from internet sites.
Another problem occurs when the sender speaks only RFC822 and the receiver speaks only X.400 or some proprietary vendor specific mail protocol.
Here Host1 speaks only TCP/IP and RFC822, where as host 2 speaks only OSITP$_4$ ans X.400. They can exchange e-mail using an e-mail gateway.

Procedure:

- ❖ Host 1 establishes a TCP connection to gateway and then use SMTP to transfer message there.
- ❖ The gateway then puts the message in a buffer of messages destined to host 2.
- ❖ A TP$_4$ connection is established between host 2 an the gateway.
- ❖ The message is transferred using OSI equivalent of SMTA.

The problems here are

1) The Internet address and X.400 address are totally different. Need of elaborating mapping mechanism between them.
2) Envelope and header fields are present in one system and are not present in the other.

MAIL ACCESS PROTOCOL

Till now we have assumed that an users work on machines that are capable of sending and receiving e-mail. Sometimes this situation is false. For example in an organization, users work on desktop PCs that are no in the internet and are capable of sending and receiving e-mail from outside. Instead the organization has one or more e-mail servers that can send and receive e-mail. To sned and receive e-mails, a PC must talk to an e-mail server using some kind of delivery protocol.

There are currently two popular mail access protocols:POP$_3$(Post office Protocol version3) and 1 MAP (internet mail access protocl)

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