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SCHOOL OF ELECTRICAL AND ELECTRONICS ENGINEERING DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

UNIT – I COMPUTER NETWORK – SECA1604

UNIT I DATA COMMUNICATION- BASICS

I. INTRODUCTION

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

Digital data is information stored on a computer system as a series of 0's and 1's in a binary language. Information is stored on computer disks and drives as a magnetically charged switch which is in either a 0 or 1 state.

A digital signal refers to an electrical signal that is converted into a pattern of bits. Unlike an analog signal, which is a continuous signal that contains time-varying quantities, a digital signal has a discrete value at each sampling point. The precision of the signal is determined by how many samples are recorded per unit of time.

Bit rate describes the rate at which bits are transferred from one location to another. In other words, it measures how much data is transmitted in a given amount of time. Bit rate is commonly measured in bits per second (bps), kilobits per second (Kbps), or megabits per second (Mbps).

Bit length is the distance one bit occupies on the transmission medium

$$\textbf{Bit Length=Propagation Speed} \times \textbf{Bit Duration} \qquad (1.1)$$

1.1 EFFECTIVENESS OF A DATA COMMUNICATIONS

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

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.

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

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.

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 3D ms. If some of the packets arrive with 3D-ms delay and others with 4D-ms delay, an uneven quality in the video is the result.

1.2 DATA REPRESENTATION

Information today comes in different forms such as text, numbers, images, audio, and video.

Text: In data communications, text is represented as a bit pattern, a sequence of bits (Os or Is). Different sets of bit patterns have been designed to represent text symbols. Each set is called a code, and the process of representing symbols is called coding. Today, the prevalent coding system is called Unicode, which uses 32 bits to represent a symbol or character used in any language in the world. The American Standard Code for Information Interchange (ASCII), developed some decades ago in the United States, now constitutes the first 127 characters in Unicode and is also referred to as Basic Latin. Appendix A includes part of the Unicode.

Numbers: Numbers are also represented by bit patterns. However, a code such as ASCII is not used to represent numbers; the number is directly converted to a binary number to simplify mathematical operations.

Images: Images are also represented by bit patterns. In its simplest form, an image is composed of a matrix of pixels (picture elements), where each pixel is a small dot. The size of the pixel depends on the resolution. For example, an image can be divided into 1000 pixels or 10,000 pixels. In the second case, there is a better representation of the image (better resolution), but more memory is needed to store the image. After an image is divided into pixels, each pixel is assigned a bit pattern. The size and the value of the pattern depend on the image. For an image made of only blackand-white dots (e.g., a chessboard), a I-bit pattern is enough to represent a pixel. If an image is not made of pure white and pure black pixels, you can increase the size of the bit pattern to include gray scale. For example, to show four levels of gray scale, you can use 2-bit patterns. A black pixel can be represented by 00, a dark gray pixel by 01, a light gray pixel by 10, and a white pixel by 11. There are several methods to represent color images. One method is called RGB, so called because each color is made of a combination of three primary colors: red, green, and blue. The intensity of each color is measured, and a bit pattern is assigned to it. Another method is called YCM, in which a color is made of a combination of three other primary colors: yellow, cyan, and magenta.

Audio: Audio refers to the recording or broadcasting of sound or music. Audio is by nature different from text, numbers, or images. It is continuous, not discrete.

Video: Video refers to the recording or broadcasting of a picture or movie. Video can either be produced as a continuous entity (e.g., by a TV camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion.

1.3 NETWORKS

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.

1.3.1 Distributed Processing

Most networks use distributed processing, in which a task is divided among multiple computers. Instead of one single large machine being responsible for all aspects of a process, separate computer (usually a personal computer or workstation) handle a subset.

1.3.2 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 the software.

Performance is often evaluated by two networking metrics: throughput and delay. We often need more throughput 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 a catastrophe.

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.

1.4 ANALOG AND DIGITAL

1.4.1 Analog and Digital Data

Both data and the signals that represent them can be either analog or digital in form. Analog

and Digital Data. Data can be analog or digital. The term analog data refers to information that is continuous; digital data refers to information that has discrete states. For example, an analog clock that has hour, minute, and second hands gives information in a continuous form; the movements of the hands are continuous. On the other hand, a digital clock that reports the hours and the minutes will change suddenly from 8:05 to 8:06. Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal. Digital data take on discrete values. For example, data are stored in computer memory in the form of Os and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

1.4.2 Analog and Digital Signals

Like the data they represent, signals can be either analog or digital. An analog signal has infinitely many levels of intensity over a period of time. As the wave moves from value A to value B, it passes through and includes an infinite number of values along its path. A digital signal, on the other hand, can have only a limited number of defined values. Although each value can be any number, it is often as simple as 1 and O. The simplest way to show signals is by plotting them on a pair of perpendicular axes. The vertical axis represents the value or strength of a signal. The horizontal axis represents time. Figure 1.1 illustrates an analog signal and a digital signal. The curve representing the analog signal passes through an infinite number of points. The vertical lines of the digital signal, however, demonstrate the sudden jump that the signal makes from value to value.

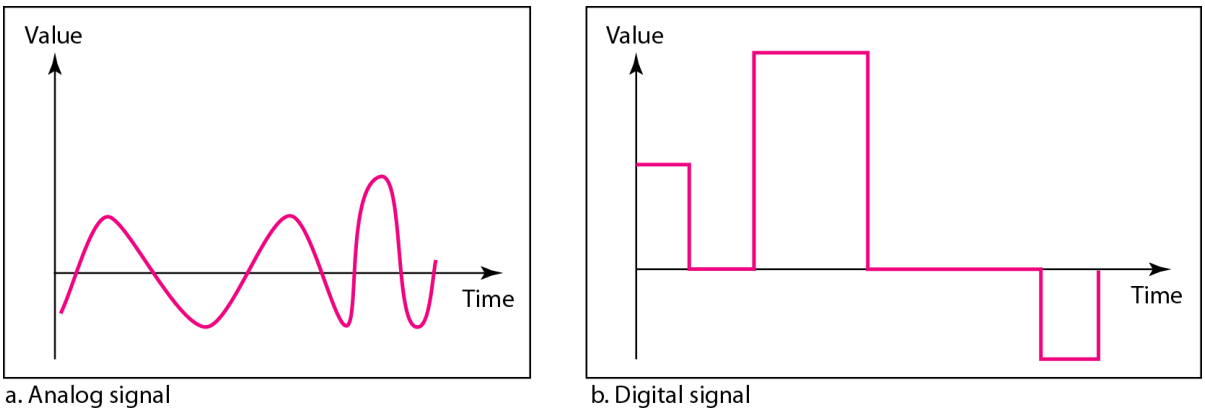


Fig. 1.1 ***Comparison of analog and digital signals***

1.5 TIME DOMAIN CONCEPTS

Continuous signal - Infinite number of points at any given time

Discrete signal - Finite number of points at any given time; maintains a constant level then changes to another constant level

Periodic signal - Pattern repeated over time

Aperiodic (non-periodic) signal - Pattern not repeated over time

✓ In data communications, we commonly use periodic analog signals and nonperiodic digital signals.

✓ Periodic analog signals can be classified as simple or composite.

✓ A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals.

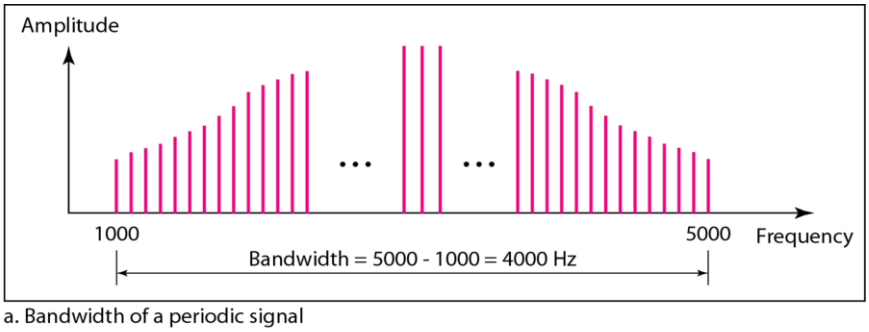
✓ A composite periodic analog signal is composed of multiple sine waves.

1.6 BANWIDTH

The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

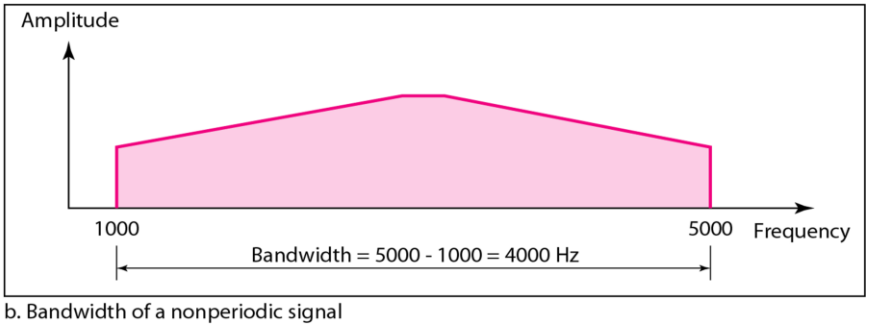
For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.

Note: each frequency is identifiable



a. Bandwidth of a periodic signal

Note: frequencies are all over the place



b. Bandwidth of a nonperiodic signal

Fig. 1.2 The bandwidth of periodic and non-periodic composite signals

EXAMPLE 1.1

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, ar $$B = f_h - f_l = 900 - 100 = 800 \text{ Hz}$$ ***equencies), assuming all co***

Solution

Let fh ***ie spectru***

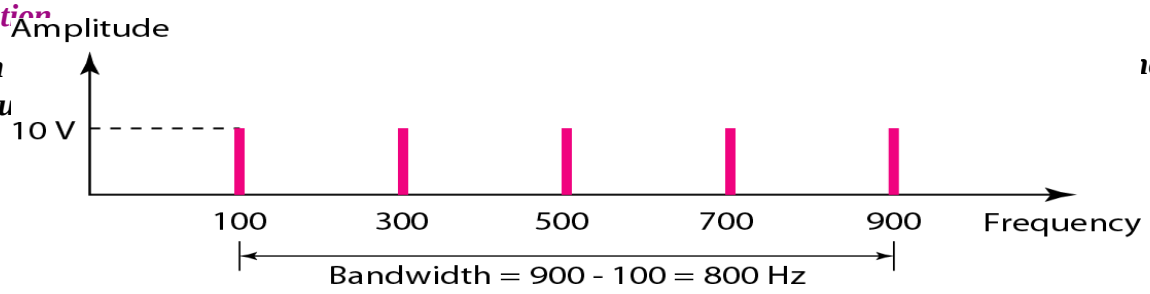


Fig. 1.3 The bandwidth for Example 1.1

DIGITAL SIGNALS

In addition to being represented by an analog signal, information can also be represented by a

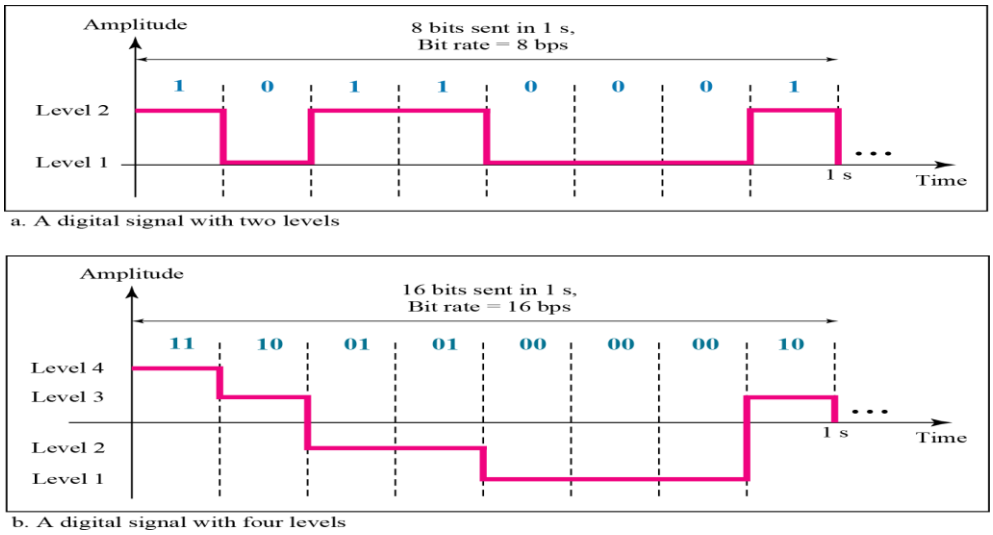


Fig 1.4 Two digital signals: one with two signal levels and the other **with four signal levels**

1.7.1 Bit rate

✓ Rate at which bits are transferred from one location to another.

- ✓ Measures how much data is transmitted in a given amount of time
- ✓ Measured in bits per second (bps), kilobits per second (Kbps), or megabits per second (Mbps).
- ✓ A good rule of thumb is for the bitrate of your stream to use no more than 50% of your available upload bandwidth capacity on a dedicate d line.
- ✓ For example, if the result you get from a speed test shows that you have 2Mbps of upload speed available, your combined audio and video bit rate should not exceed 1Mbps.

1.7.2 Bit rate, Bit Length, Baud rate Estimation of Bit rate

Frequency × bit depth × channels = bit rate.

44,100 samples per second × 16 bits per sample × 2 channels = 1,411,200 bits per second (or 1,411.2 kbps)

Baud Rate: Baud Rate is the number of signal unit transmitted per second. Thus Baud Rate is always less than or equal to bit rate.

Bit Length: The Bit Length is the distance of one Bit occupies on the transmission medium. Bit Length = Propagation speed * Bit duration

EXAMPLE 1.2

A digital signal has eight levels. How many bits are needed per level? We calculate the number of bits from the formula

Solution

$$\text{Number of bits per level} = \log_2 8 = 3$$

Each signal level is represented by 3 bits.

EXAMPLE 1.3

Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?

Solution

A page is an average of 24 lines with 80 characters in each line. If we assume that one

$$100 \times 24 \times 80 \times 8 = 1{,}636{,}000 \text{ bps} = 1.636 \text{ Mbps}$$

character requires 8 bits, the bit rate is

EXAMPLE 1.4

A digitized voice channel, as we will see in Chapter 4, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?

Solution

$$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

EXAMPLE 1.5

What is the bit rate for high-definition TV (HDTV)? Solution

HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16 : 9. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represents one color pixel. The TV stations reduce this rate to 20 to 40 Mbps through compression.

$$1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \text{ or } 1.5 \text{ Gbps}$$

1.7.3 BANDWIDTH REQUIREMENTS FOR VARIOUS BITRATE

A good rule of thumb is for the bit rate of your stream to use no more than 50% of you r available upload bandwidth capacity on a dedicated line. For example, if the result you get from a speed test shows that you have 2Mbps of upload speed available, your combined audio and video bit rate should not exceed 1Mbps.

Table 1.1 Bandwidth Requirements

Bit Rate	*Harmonic 1*	*Harmonics 1, 3*	*Harmonics 1, 3, 5*
$n = 1$ kbps	$B = 500$ Hz	$B = 1.5$ kHz	$B = 2.5$ kHz
$n = 10$ kbps	$B = 5$ kHz	$B = 15$ kHz	$B = 25$ kHz
$n = 100$ kbps	$B = 50$ kHz	$B = 150$ kHz	$B = 250$ kHz

EXAMPLE 1.6

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth?

Solution

Let f h be the highest frequency, f l the lowest frequency, and B the bandwidth. Then B = f h - f l = 900 - 100 =800 Hz.

EXAMPLE 1.7

A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency?

Solution

Let f h be the highest frequency, f l the lowest frequency, and B the bandwidth. Then B = f h - f l = 20 ; 20 = 60 - f l; f l = 40 Hz.

EXAMPLE 1.8

A non-periodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20V. Find the lowest and highest frequency.

Solution

Bandwidth B = 200 kHz Middle Frequency is 140 kHz fh = middle frequency + (B/2) = 140 + 100 = 240 kHz fl = middle frequency - (B/2) = 140 -100 = 40 kHz

1.7 TRANSMISSION CHANNEL

Physical Transmission Medium - Connection over a multiplexed channel - Radio Channel Types of Transmission Media

Guided Media - Twisted Pair Cable Coaxial Cable Optical Fibre Cable Unguided Media - Radiowaves Microwaves Infrared

1.7.1 Transmission of Digital Signals Baseband Transmission

- ✓ Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal.
- ✓ In baseband transmission, the required bandwidth is proportional to the bit rate; if we need to send bits faster, we need more bandwidth.

Note: Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.

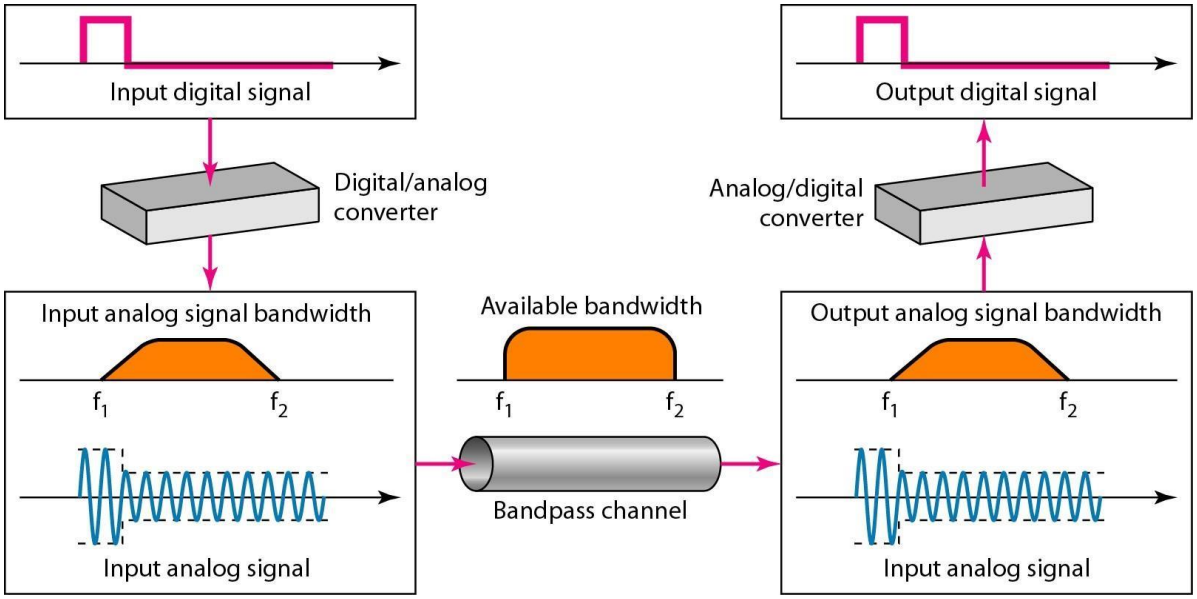


Fig. 1.5 Modulation of a digital signal for transmission on a bandpass channel

Broadband Transmission (or) Modulation

- Broadband transmission or modulation means changing the digital to analog signal for transmission
- It uses a band pass channel - a channel with a bandwidth that does not start with zero

Examples: internet data through telephone line using MODEMs, Modulation in cellular mobile networks, etc.,

1.8 TRANSMISSION IMPAIRMENT

Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium. What is sent is not what is received. Three causes of impairment are attenuation, distortion, and noise.

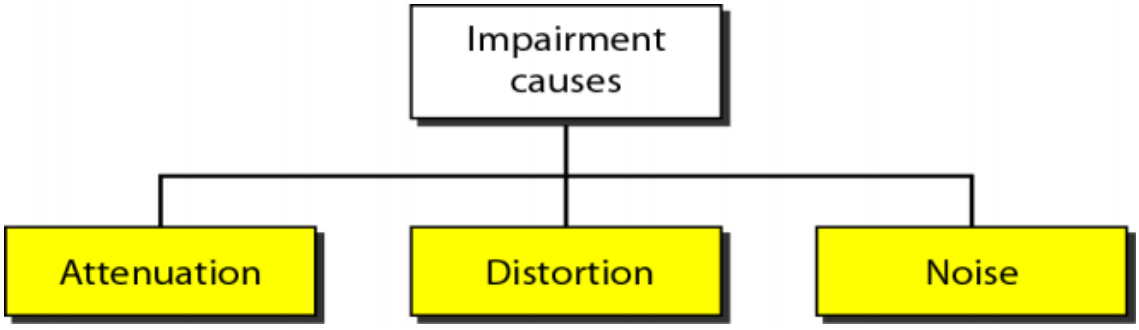


Fig. 1.6 Various causes for Impairment

Attenuation

- Attenuation means a loss of energy.
- When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium.
- Some of the electrical energy in the signal is converted to heat.
- To compensate for this loss, amplifiers are used to amplify the signal.

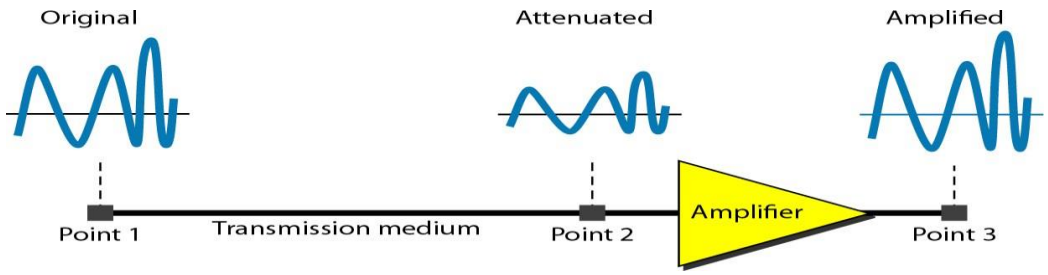


Fig. 1.7 Attenuation

Decibel

- To show that a signal has lost or gained strength, engineers use the unit of the decibel
- The decibel (dB) measures the relative strengths of two signals or one signal at two different points.
- Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.
- Variables PI and P2 are the powers of a signal at points 1 and 2, respectively.

$$N_{dB} = 10 \times \log_{10} (P2 / P1)$$

P2 = ending power level in watts P1 = beginning power level in watts

Distortion

- Distortion means that the signal changes its form or shape.
- Distortion can occur in a composite signal made of different frequencies.
- Each signal component has its own propagation speed through a mediumand, therefore,

its own delay in arriving at the final destination.

- Differences in delay may create a difference in phase if the delay is not exactly the same as the period duration.
- In other words, signal components at the receiver have phases different fromwhat they had at the sender.
- The shape of the composite signal is therefore not the same.

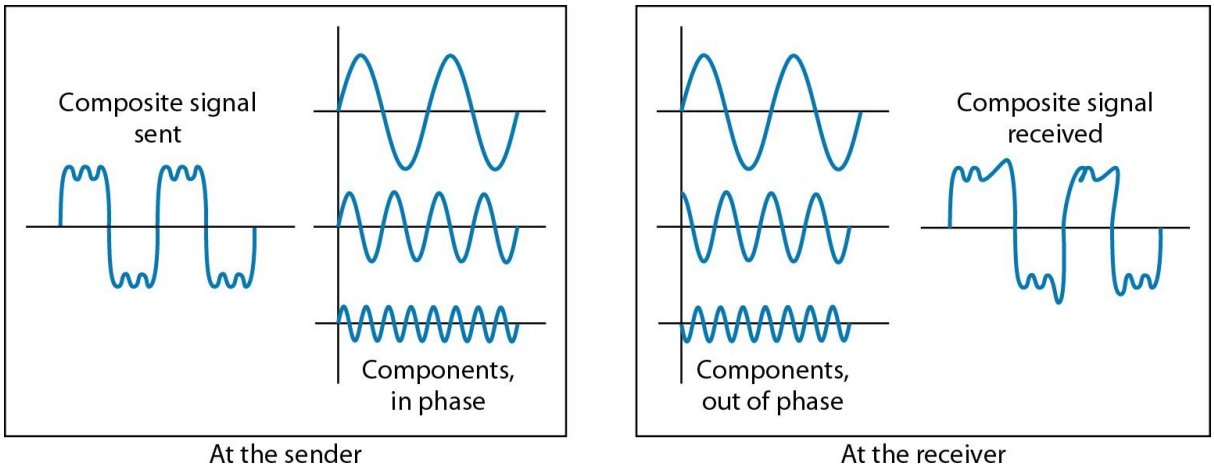


Fig. 1.8 Distortion

Noise

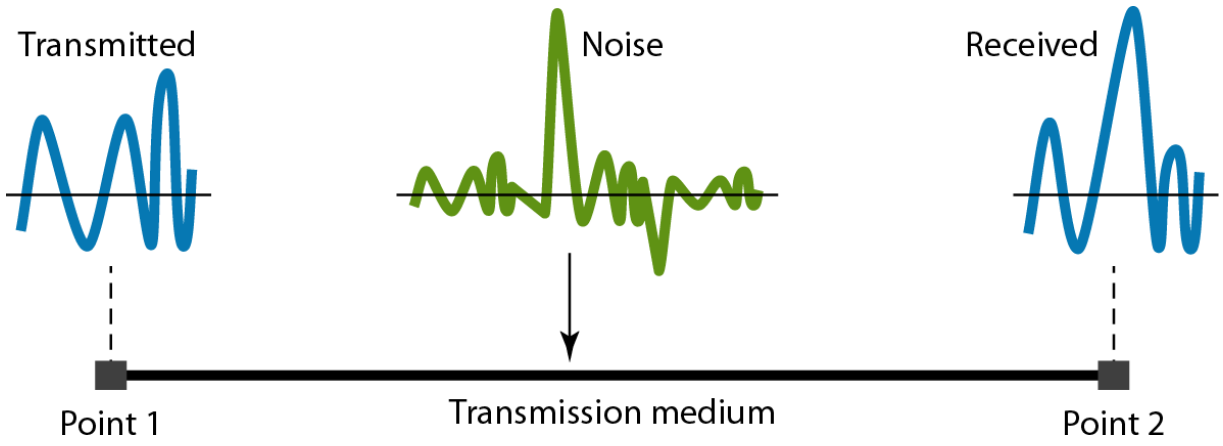


Fig. 1.9 noise

1.9 SWITCHING TECHNIQUES

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. Other topologies employing multipoint connections, such as a bus, areruled out because the distances between devices and the total number of devices increase beyond the capacities of the media and equipment.

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 1.10 shows a switched network.

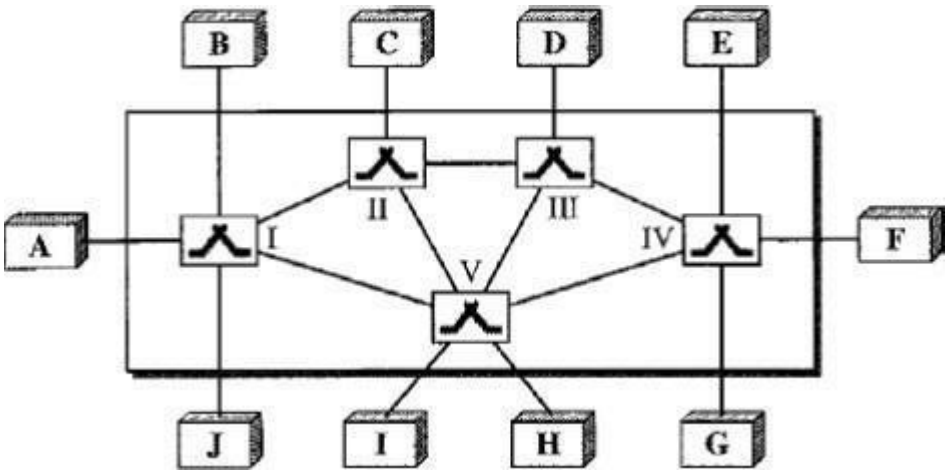


Fig.1.10 switched network

The end systems (communicating devices) are labeled A, B, C, D, and so on, and the switches are labeled I, II, III, IV, and V. Each switch is connected to multiple links.

Taxonomy of switched networks

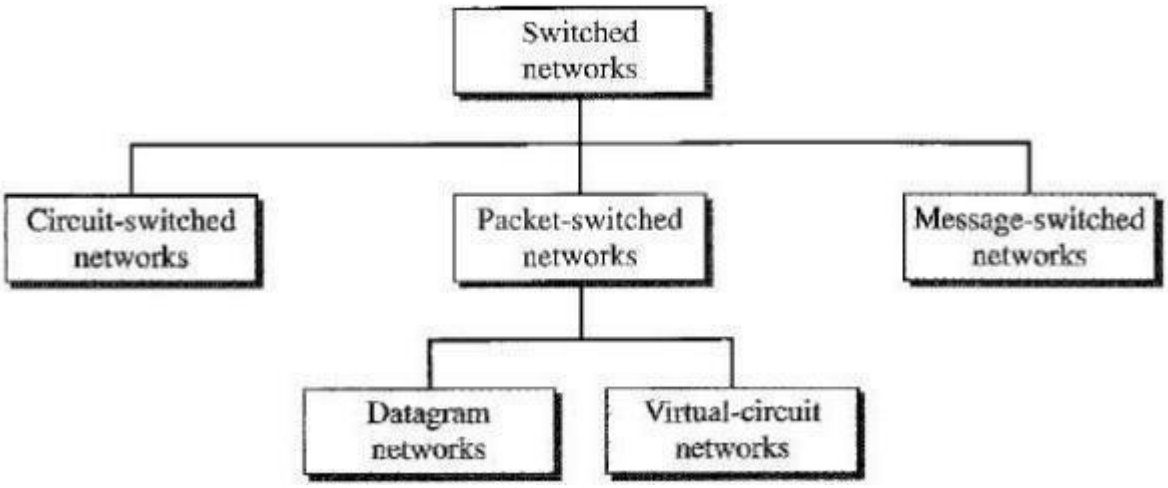


Fig.1.11 Taxonomy of Switched Networks

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

Figure 1.12 shows a trivial circuit-switched network with four switches and four links. Each link is divided into n (n is 3 in the figure) channels by using FDM or TDM.

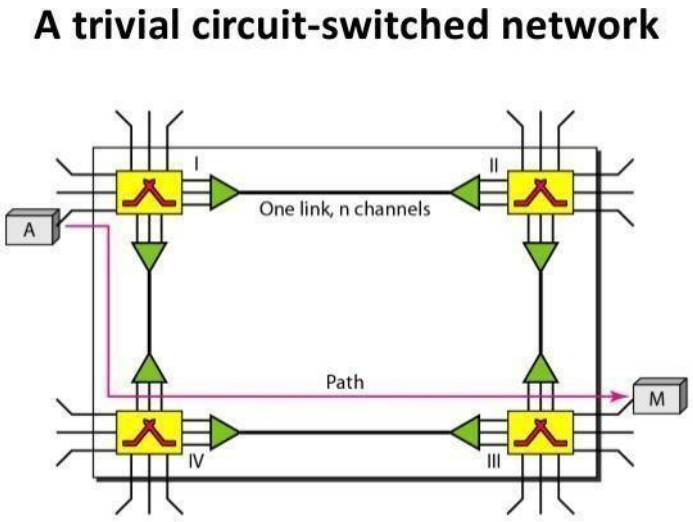


Fig 1.12 trivial circuit-switched network

Three Phases

The actual communication in a circuit-switched network requires three phases:

1. Connection setup
2. Data transfer,
3. Connection teardown

1. 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. The end systems are normally connected through dedicated lines to the switches, so connection setup means creating dedicated channels between the switches. For example, in Figure 1.2, 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. Note that end-to-end addressing is required for creating a connection between the two end systems. These can be, for example, the addresses of the computers assigned by the administrator in a TDM network, or telephone numbers in an FDM network.

2. Data Transfer Phase:

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

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

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. In a telephone network, people normally terminate the communication when they have finished their conversation. However, in computer networks, a computer can be connected to another computer even if there is no activity for a long time. In this case, allowing resources to be dedicated means that other connections are deprived.

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.

Figure 1.13 shows the idea of delay in a circuit switched network when only two switches are involved. As Figure shows, there is no waiting time at each switch. The total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit.

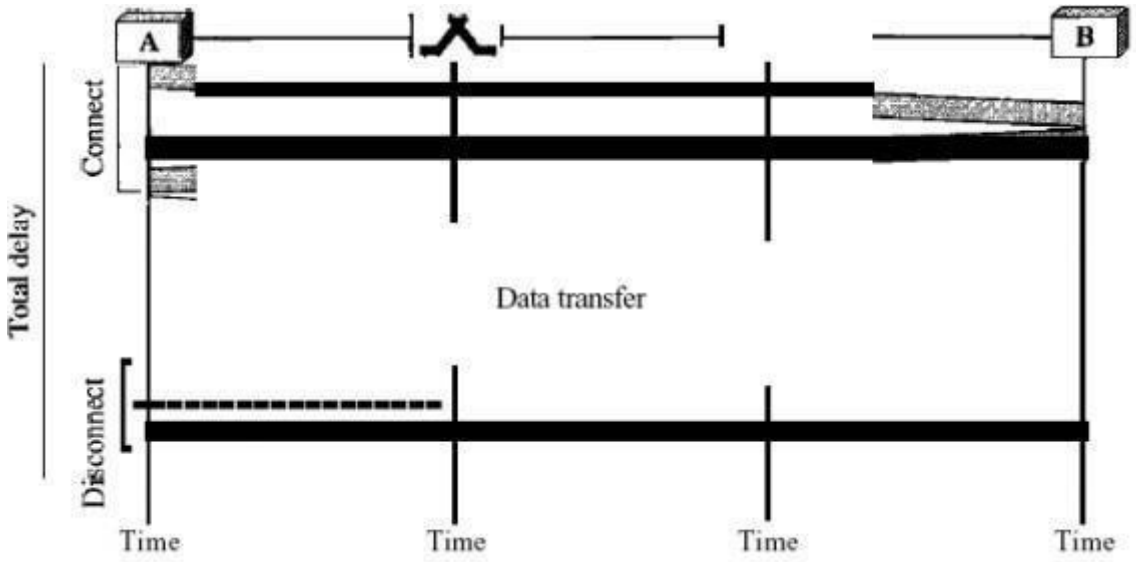


Fig 1.13 delay in a circuit switched network

The delay caused by the setup is the sum of four parts: the propagation time of the source computer request (slope of the first gray box), the request signal transfer time (height of the first gray box), the propagation time of the acknowledgment from the destination computer (slope of the second gray box), and the signal transfer time of the acknowledgment (height of the second gray box).

The delay due to data transfer is the sum of two parts: the propagation time (slope of the colored box) and data transfer time (height of the colored box), which can be very long. The third box shows the time needed to tear down the circuit. We have shown the case in which the receiver requests disconnection, which creates the maximum delay.

DATAGRAM NETWORKS

In a datagram network, each packet is treated independently of all others. Even if a packet is part of a multipacket transmission, the network treats it as though it existed alone. Packets in this approach are referred to as datagrams. Datagram switching is normally done at the network layer. Figure 1.14 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. That is why we use a different symbol for the switches in the figure.

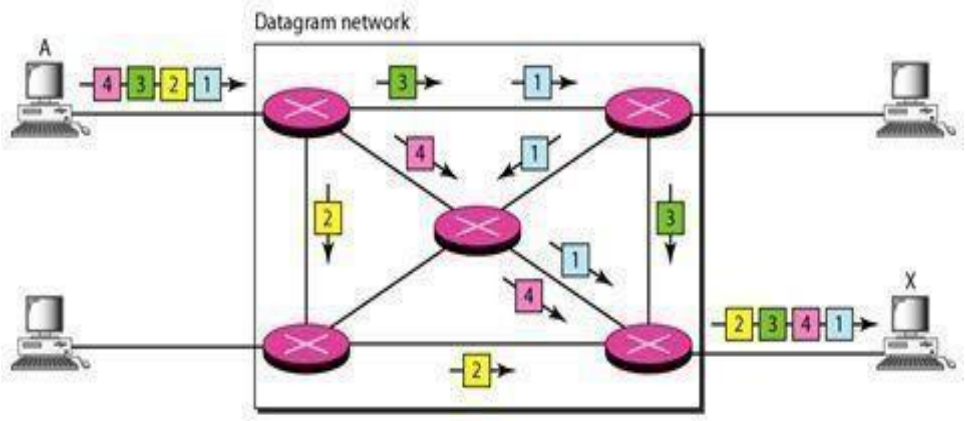


Fig 1.14 datagram network

In this example, all four packets (or datagrams) belong to the same message, but may travel different paths to reach their destination. This is so because the links may be involved in carrying packets from other sources and do not have the necessary bandwidth available to carry all the packets from A to X. This approach can cause the datagrams of a transmission to arrive at their destination out of order with different delays between the packets. Packets may also be lost or dropped because of a lack of resources.

In most protocols, it is the responsibility of an upperlayer protocol to reorder the datagrams or ask for lost datagrams before passing them on to the application. 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 or destination.

Routing Table:

If there are no setup or teardown phases, how are the packets routed to their destinations in a datagram network? In this type of network, each switch (or packet switch) has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables. This is different from the table of a circuit switched network in which each entry is created when the setup phase is completed and deleted when the teardown phase is over. Figure 1.15 shows the routing table for a switch.

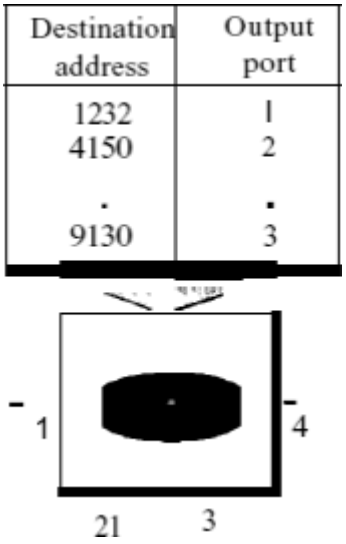


Fig 1.15 Routing table in a datagram network

Destination address:

Every packet in a datagram network carries a header that contains, among other information, the destination address of the packet. When the switch receives the packet, this destination address is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded. This address, unlike the address in a virtual circuit-switched 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. If a source sends a packet and there is a delay of a few minutes before another packet can be sent, the resources can be reallocated during these minutes for other packets from other sources.

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.

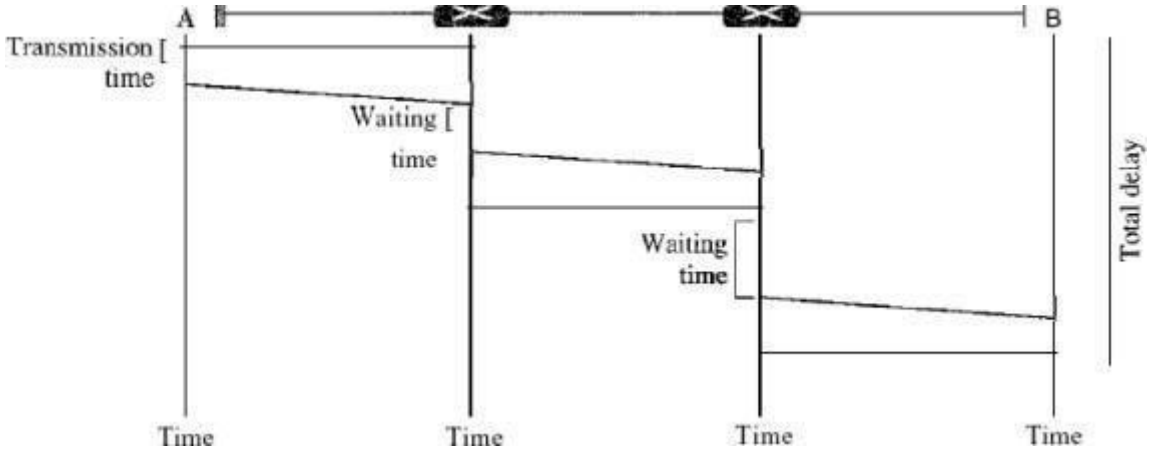


Fig 1.16 Delay in a datagram network

The packet travels through two switches. There are three transmission times (3T), three propagation delays (slopes 3't of the lines), and two waiting times (WI + w2)' we ignore the processing time in each switch.

The total delay is Total delay =3T + 3t + WI + W2

VIRTUAL-CIRCUIT NETWORKS:

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

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 canied), not end-to-end jurisdiction. The reader may ask how the intermediate switches know where to send the

packet if there is no final destination address carried by a packet. The answer will be clear when we discuss virtual circuit identifiers in the next section.

4. As in a circuit-switched network, all packets follow the same path established duringthe 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. But this may change in the future. Figure 1.17 is an example of a virtual-circuit network.

The network has switches that allow traffic from sources to destinations. A source or destination can be a computer, packet switch, bridge, or any other device that connects other networks.

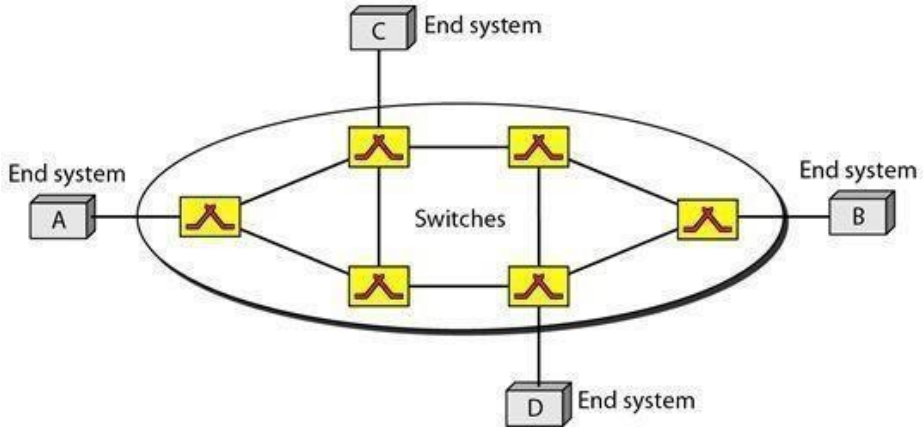


Fig 1.17 Virtual-circuit network

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 or internationally if the network is part of an international network. However, we will see that a global address in virtual-circuit networks is used only to create a virtual-circuit identifier, as discussed next.

Virtual-Circuit Identifier: The identifier that is actually used for data transfer is called the virtual circuit identifier (Vel). A vel, 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 1.18 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.

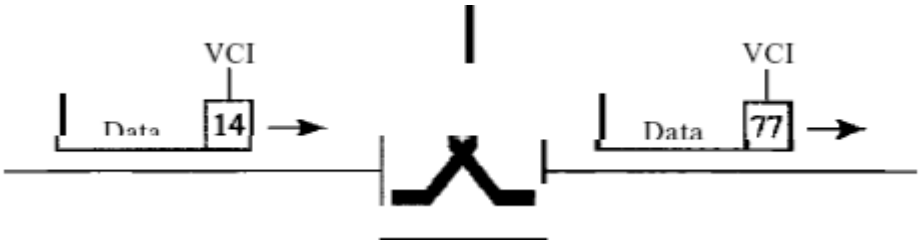


Fig 1.18 Virtual-circuit identifier

Three Phases

As in a circuit-switched network, a source and destination need to go through three phases in a virtual-circuit network: setup, data transfer, and teardown.

In the setup phase,

The source and destination use their global addresses to help switches make table entries for the connection. In the teardown phase, the source and destination inform the switches to delete the corresponding entry. Data transfer occurs between these two phases. We first discuss the data transfer phase, which is more straightforward; we then talk about the setup and teardown phases.

Data Transfer Phase:

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. This means that the switch holds four pieces of information for each virtual circuit that is already set up. 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. 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

Efficiency

As we said before, resource reservation in a virtual-circuit network can be made during the

setup or can be on demand during the data transfer phase. In the first case, the delay for each packet is the same; in the second case, each packet may encounter different delays. There is one big advantage in a virtual-circuit network even if resource allocation is on demand. The source can check the availability of the resources, without actually reserving it. Consider a family that wants to dine at a restaurant. Although the restaurant may not accept reservations (allocation of the tables is on demand), the family can call and find out the waiting time. This can save the family time and effort.

Delay in Virtual-Circuit Networks

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. Below Figure 1.19 shows the delay for a packet traveling through two switches in a virtual circuit network.

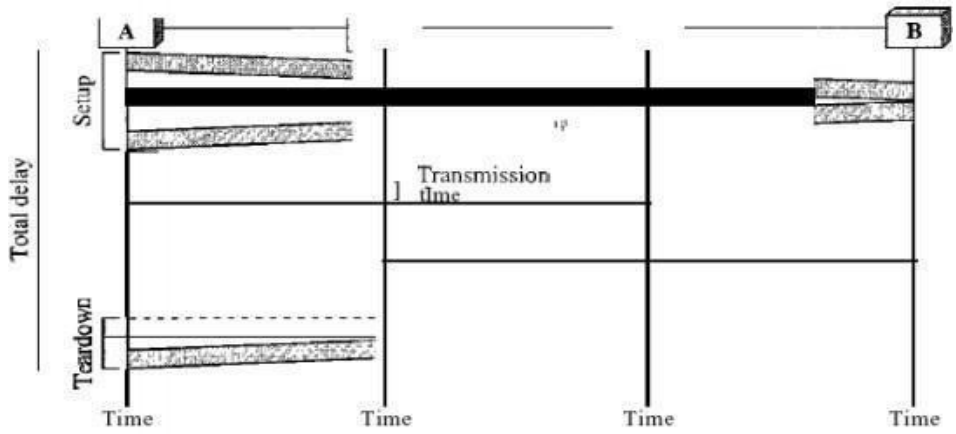


Fig 1.19 Delay in a virtual-circuit network

The packet is traveling through two switches (routers). There are three transmission times (3T), three propagation times (3't), data transfer depicted by the sloping lines, a setup delay (which includes transmission and propagation in two directions), and a teardown delay (which includes transmission and propagation in one direction). We ignore the processing time in each switch.

The total delay time is Total delay = 3T+ 3't + setup delay + teardown delay

Differences between Virtual Circuits & Datagram Networks Virtual Circuits

1. It is connection-oriented simply meaning that there is a reservation of resources like buffers, CPU, bandwidth,etc. for the time in which the newly setup VC is going to be used by a data transfer session.

2. First packet goes and reserves resources for the subsequent packets which as a result follow the same path for the whole connection time.

3. Since all the packets are going to follow the same path, a global header is required only for the first packet of the connection and other packets generally don't require global headers.

4. Since data follows a particular dedicated path, packets reach inorder to the destination.

5. From above points, it can be concluded that Virtual Circuits are highly reliable means of transfer.

6. Since each time a new connection has to be setup with reservation of resources and extra information handling at routers, its simply costly to implement Virtual Circuits.

Datagram Networks:

1. It is connectionless service. There is no need of reservation of resources as there is no dedicated path for a connection session.

2. All packets are free to go to any path on any intermediate router which is decided on the go by dynamically changing routing tables on routers.

3. Since every packet is free to choose any path, all packets must be associated with a header with proper information about source and the upper layer data.

4. The connectionless property makes data packets reach destination in any order, means they need not reach in the order in which they were sent.

5. Datagram networks are not reliable as Virtual Circuits.

6. But it is always easy and cost efficient to implement datagram networks as there is no extra headache of reserving resources and making a dedicated each time an application has to communicate.

ISO / OSI MODEL:

ISO refers International Standards Organization was established in 1947, it is a multinational body dedicated to worldwide agreement on international standards. OSI refers to Open System Interconnection that covers all aspects of network communication. It is a standard of ISO. Here open system is a model that allows any two different systems to communicate regardless of their underlying

architecture. Mainly, it is not a protocol it is just a model.

1.10 OSI MODEL

The open system interconnection model is a layered framework. It has seven separate but interrelated layers. Each layer having unique responsibilities.

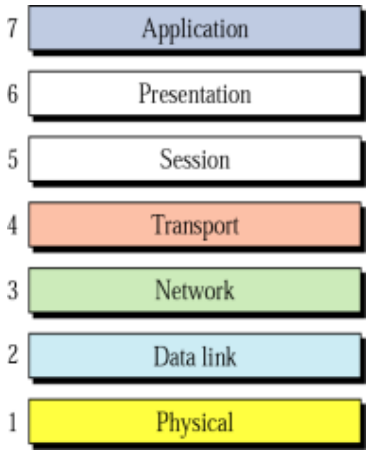


Fig 1.20 OSI Model

The OSI model shown in figure 1.20 is based on the proposal developed by the International Standards Organization (ISO) as a first step towards international standardization of the protocols used in the various layers. The model is called the OSI (Open System Interconnection) reference model because it deals with connecting open systems, i.e., systems that are open for communication with other systems. The purpose of the OSI model is to show how to facilitate communication between different systems without requiring changes to the logic of the underlying hardware and software.

The OSI model is not a protocol; it is a model for understanding and designing a network architecture that is flexible, robust and interoperable. The OSI model is a layered framework for the design of network systems that allows communication between all types of computer systems. It consists of seven separate but related layers, each of which defines a part of the process of moving information across a network.

The principles that were applied to arrive at the seven layers are as follows:

* A layer should be created where a different level of abstraction is needed. * Each layer should perform a well-defined function.

* The function of each layer should be chosen with an eye toward defining internationally standardized protocols.

* The layer boundaries should be chosen to minimize the information flow across the interfaces.

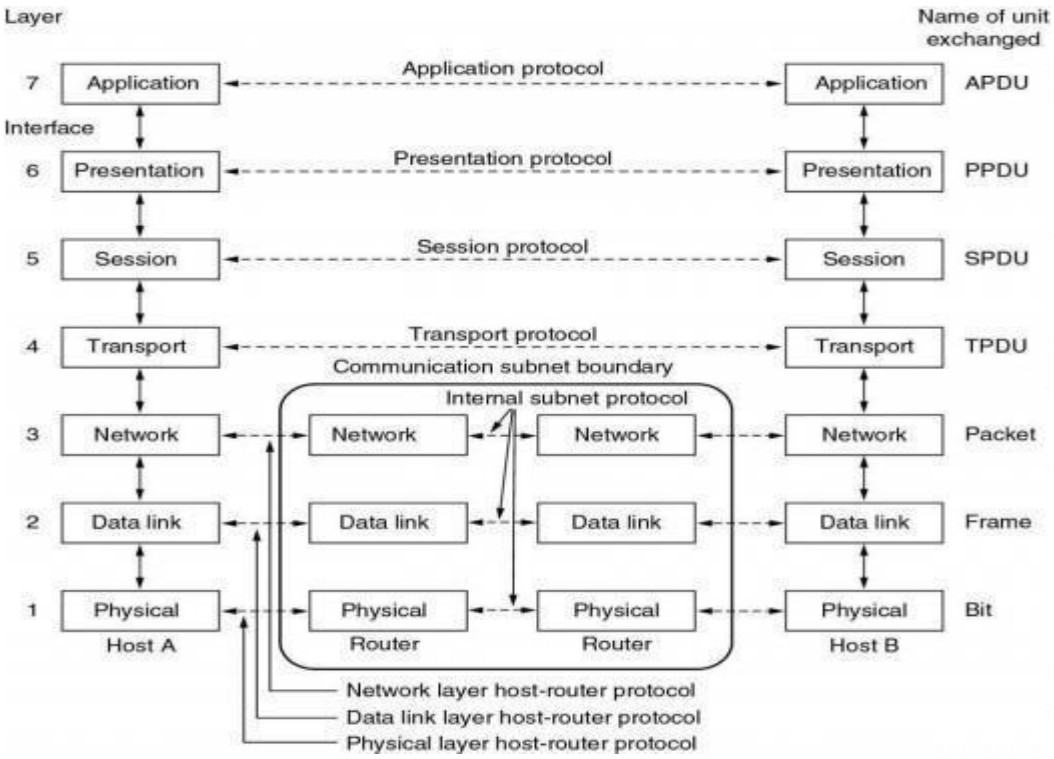
* The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that the architecture does not become unwieldy.

Layered Architecture:

The OSI model is composed of seven layers: Physical, Data link, Network, Transport, Session, Presentation, Application layers.

Figure 1.21 shows the layers involved when a message travels from A to B, it may pass through many intermediate nodes. These intermediate nodes involve only the first 3 layers of the OSI model. Within a single machine, each layer calls upon the services of the layer just below it, layer 3 for ex. Uses the services provided by layer 2 & provides services for layer 4. Between machines, layer X on one machine communicates with layer X on another machine. This communication is governed by an agreed upon series of rules & Conventions called protocols. The processes on each machine that communicate at a given layer are called peer – to – peer processes. Communication between machines is therefore a peer – to –peer process using the protocols appropriate to a given layer.

Fig 1.21 interaction between layers in the OSI model



ORGANIZATION OF LAYERS

The seven layers are arranged by three sub groups.

1. Network Support Layers
2. User Support Layers
3. Intermediate Layer

Network Support Layers:

Physical, Datalink and Network layers come under the group. They deal with the physical aspects of the data such as electrical specifications, physical connections, physical addressing, and transport timing and reliability.

User Support Layers:

Session, Presentation and Application layers comes under the group. They deal with the interoperability between the software systems.

Intermediate Layer:

The transport layer is the intermediate layer between the network support and the user support layers.

FUNCTIONS OF THE LAYERS PHYSICAL LAYER

The physical layer coordinates the functions required to transmit a bit stream over a physical medium. It deals with the mechanical and electrical specifications of the interface and the transmission medium.

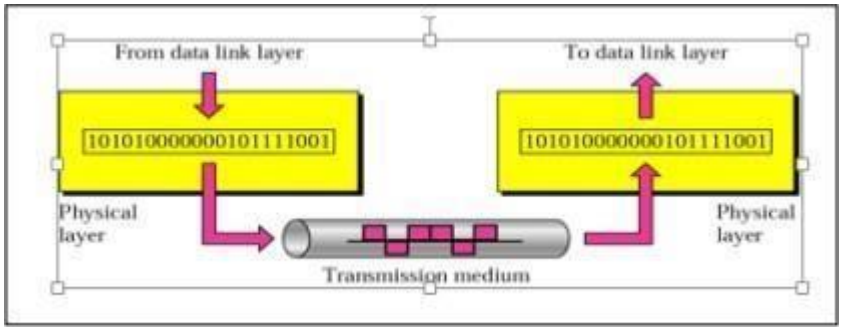


Fig 1.22 Physical Layer

The functions are,

1. Physical Characteristics of Interfaces and Media: It defines the electrical andmechanical characteristics of the interface and the media. It defines the types of transmission medium

2. Representation of Bits: To transmit the stream of bits they must be encoded into signal. It defines the type of encoding weather electrical or optical.

3. Data Rate: It defines the transmission rate i.e. the number of bits sent per second.

4. Synchronization of Bits: The sender and receiver must be synchronized at bit level.

5. Line Configuration: It defines the type of connection between the devices.
   Two types of connection are 1. Point to point 2. Multipoint
6. Physical Topology: It defines how devices are connected to make a network.
   Five topologies are, 1. mesh 2. star 3. tree 4. bus 5. ring

7. Transmission Mode It defines the direction of transmission between devices.
   Three types of transmission are, 1. simplex 2. half duplex3. full duplex

DATALINK LAYER

Datalink layer responsible for node-to-node delivery The responsibilities of Datalink layer are,

1. Framing: It divides the stream of bits received from network layer into manageable data units called frames.

2. Physical Addressing: It adds a header that defines the physical address of the sender and the receiver. If the sender and the receiver are in different networks, then the receiver address is the address of the device which connects the two networks.

3. Flow Control: It imposes a flow control mechanism used to ensure the data rate at the sender and the receiver should be same.

4. Error Control: To improve the reliability the Datalink layer adds a trailer which contains the error control mechanism like CRC, Checksum etc

5. Access Control: When two or more devices connected at the same link, then the Datalink layer used to determine which device has control over the link at any given time.

NETWORK LAYER

When the sender is in one network and the receiver is in some other network then the network layer has the responsibility for the source to destination delivery.

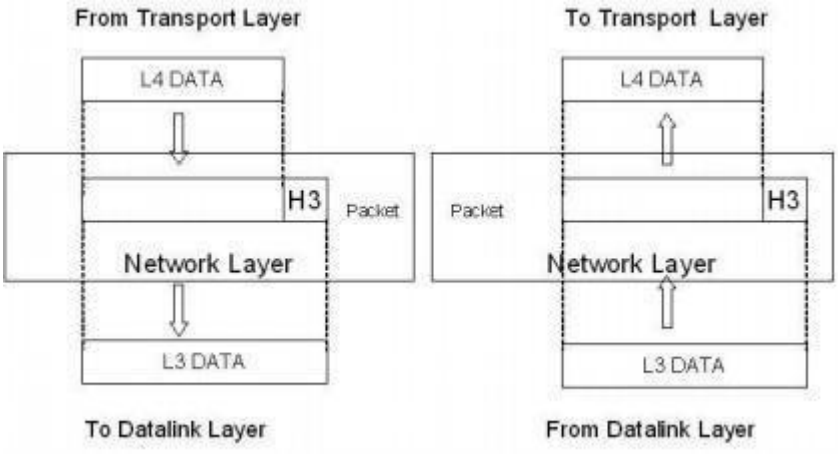


Fig 1.23 Network Layer

The responsibilities are,

1. Logical Addressing: If a packet passes the network boundary that is when the sender and receiver are places in different network then the network layer adds a header that defines the logical address of the devices.

2. Routing: When more than one networks connected and to form an internetwork, the connecting devices route the packet to its final destination. Network layer provides this mechanism.

TRANSPORT LAYER

The network layer is responsible for the end to end delivery of the entire message. It ensures that the whole message arrives in order and intact. It ensures the error control and flow control at source to destination level.

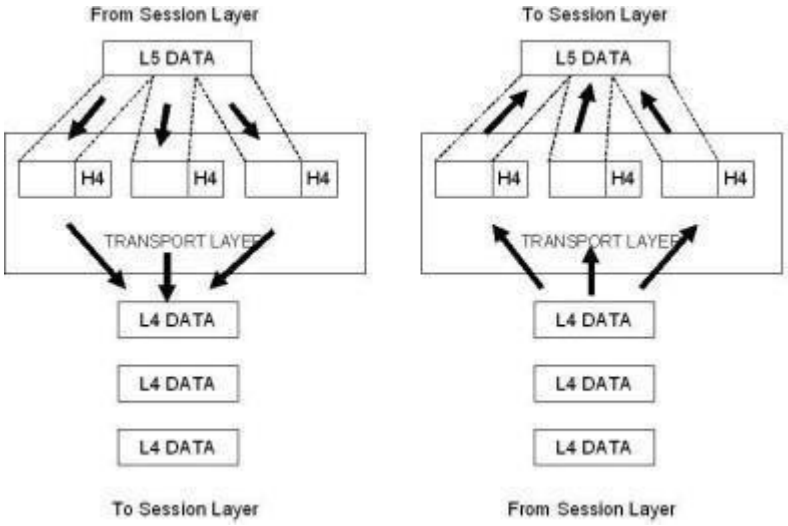


Fig 1.24 Transport Layer

The responsibilities are,

1. Service point Addressing: A single computer can often run several programs at the same time. The transport layer gets the entire message to the correct process on that computer. It adds a header that defines the port address which used to identify the exact process on the receiver.

2. Segmentation and Reassembly: A message is divided into manageable units called as segments. Each segment is reassembled after received that information at the receiver end. To make this efficient each segment contains a sequence number.

3. Connection Control: The transport layer creates a connection between the two end ports. It involves three steps. They are,

    1. Connection establishment

    2. Data transmission
    3. Connection discard

    4. Flow Control Flow control is performed at end to end level

    5. Error Control Error control is performed at end to end level.

SESSION LAYER

It acts as a dialog controller. It establishes, maintains and synchronizes the interaction between the communication devices.

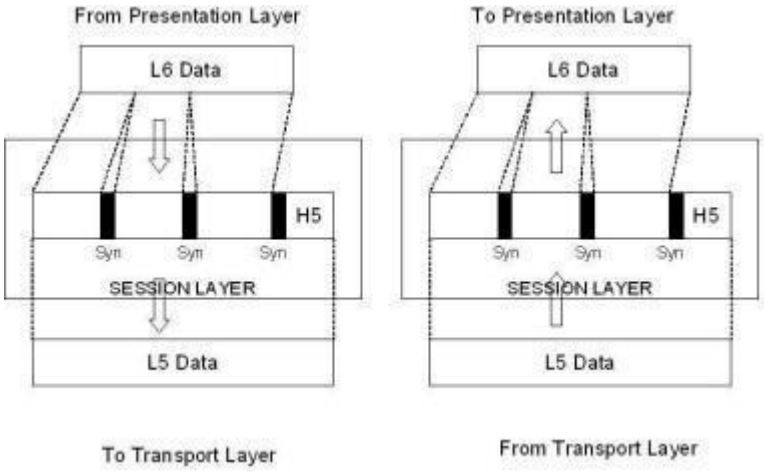


Fig 1.25 Session Layer

The responsibilities are,

1. Dialog Control: The session layer allows two systems to enter into a dialog. It allows the communication between the devices.

2. Synchronization: It adds a synchronization points into a stream of bits.

PRESENTATION LAYER

The presentation layer is responsible for the semantics and the syntax of the information exchanged.

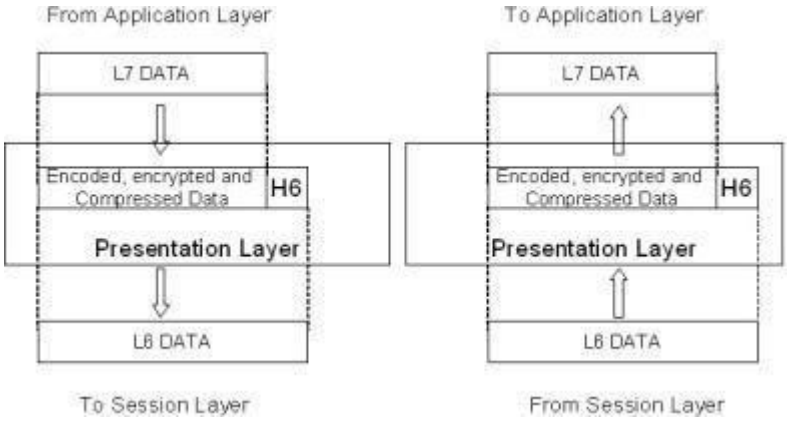


Fig 1.26 Presentation Layer

1. Translation: Different systems use different encoding systems. The presentation layer is responsible for interoperability between different systems. The presentation layer t the

sender side translates the information from the sender dependent format to a common format. Likewise, at the receiver side presentation layer translate the information from common format to receiver dependent format.

2. Encryption: To ensure security encryption/decryption is used. Encryption means transforms the original information to another form. Decryption means retrieve the original information from the encrypted data

3. Compression: It used to reduce the number of bits to be transmitted.

APPLICATION LAYER

The application layer enables the user to access the network. It provides interfaces between the users to the network.

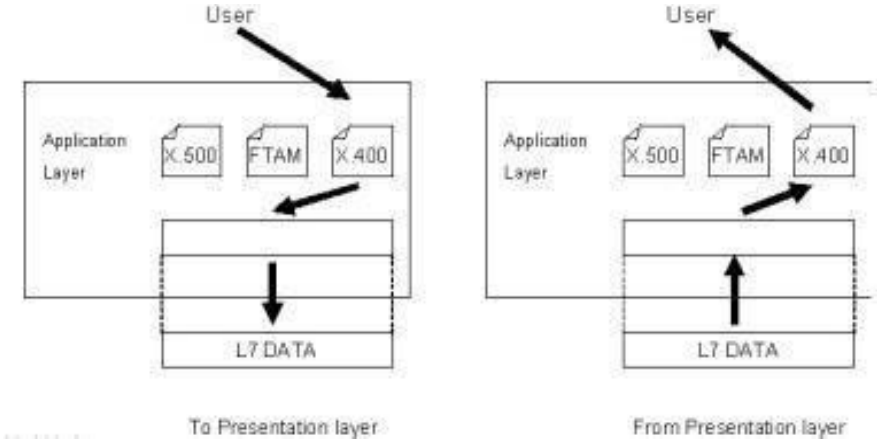


Fig 1.27 Application Layer

The responsibilities are,

1. Network Virtual Terminal: It is a software version of a physical terminal and allows a user to log on to a remote host.

2. File Transfer, Access, and Management: It allows a user to access files in a remote computer, retrieve files, and manage or control files in a remote computer.

3. Mail Services: It provides the basis for e-mail forwarding and storage.

4. Directory Services: It provides distributed database sources and access for global

information about various objects and services.

1.11 TCP/IP REFERENCE MODEL

TCP/IP is a set of protocols developed to allow cooperating computers to share resources across the network.

Layer 1: Host-to-network Layer

1. Lowest layer of the all.
2. Protocol is used to connect to the host, so that the packets can be sent over it. Varies from host to host and network to network.

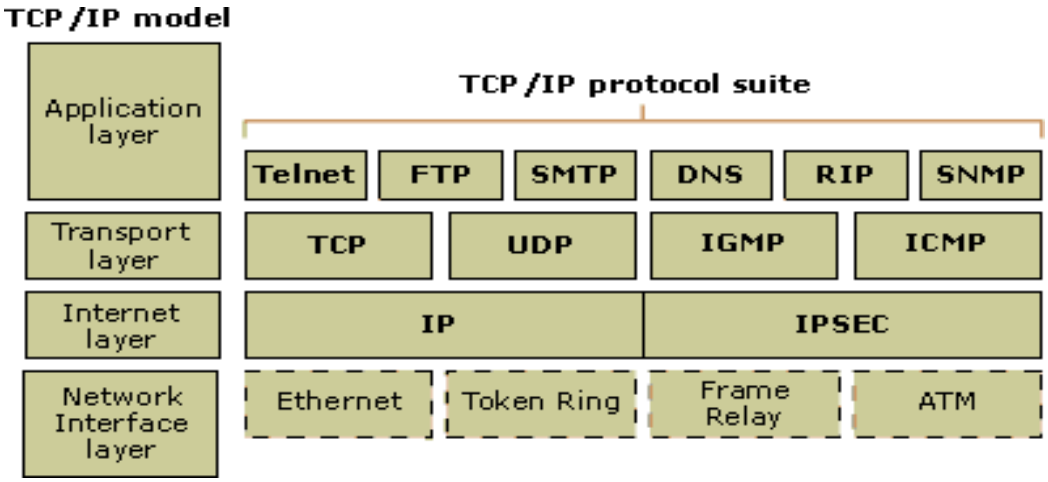


Fig 1.28 TCP/IP Model

Layer 2: Internet layer

1. Selection of a packet switching network which is based on a connectionless internetwork layer is called a internet layer.
2. It is the layer which holds the whole architecture together.
3. It helps the packet to travel independently to the destination.
4. Order in which packets are received is different from the way they are sent.
5. IP (Internet Protocol) is used in this layer.
6. The various functions performed by the Internet Layer are:
   - Delivering IP packets

- Performing routing
- Avoiding congestion

Layer 3: Transport Layer

1. It decides if data transmission should be on parallel path or singlepath.
2. Functions such as multiplexing, segmenting or splitting on the data is done by transport layer.
3. The applications can read and write to the transport layer.
4. Transport layer adds header information to the data.
5. Transport layer breaks the message (data) into small units so that they are handled more efficiently by the network layer.
6. Transport layer also arrange the packets to be sent, in sequence.

Layer 4: Application Layer

The TCP/IP specifications described a lot of applications that were at the top of the protocol stack. Some of them were TELNET, FTP, SMTP, DNS etc.

1. TELNET is a two-way communication protocol which allows connecting to a remote machine and run applications on it.
2. FTP (File Transfer Protocol) is a protocol, that allows File transfer amongst computer users connected over a network. It is reliable, simple and efficient.
3. SMTP (Simple Mail Transport Protocol) is a protocol, which is used to transport electronic mail between a source and destination, directed via a route.
4. DNS (Domain Name Server) resolves an IP address into a textual address for Hosts connected over a network.
5. It allows peer entities to carry conversation.
6. It defines two end-to-end protocols: TCP and UDP

TCP (Transmission Control Protocol): It is a reliable connection-oriented protocol which handles byte-stream from source to destination without error and flow control.

UDP (User-Datagram Protocol): It is an unreliable connection-less protocol that do not want TCPs, sequencing and flow control. Eg: One-shot request-reply kind of service

Merits of TCP/IP model

1. It operated independently.
2. It is scalable.
3. Client/server architecture.
4. Supports a number of routing protocols.
5. Can be used to establish a connection between two computers.

Demerits of TCP/IP

1. In this, the transport layer does not guarantee delivery of packets.
2. The model cannot be used in any other application.
3. Replacing protocol is not easy.
4. It has not clearly separated its services, interfaces and protocols.

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UNIT – II COMPUTER NETWORKS – SECA1604

UNIT II NETWORKING

2.1 NETWORK TOPOLOGIES

The term topology refers to the way a network is laid out, either physically or logically.

Two or more devices connect to a link; two or more links form topology.

The topology of a network is the geometric representation of the relationship of all the links and linking devices to each other.

There are five basic topologies possible :

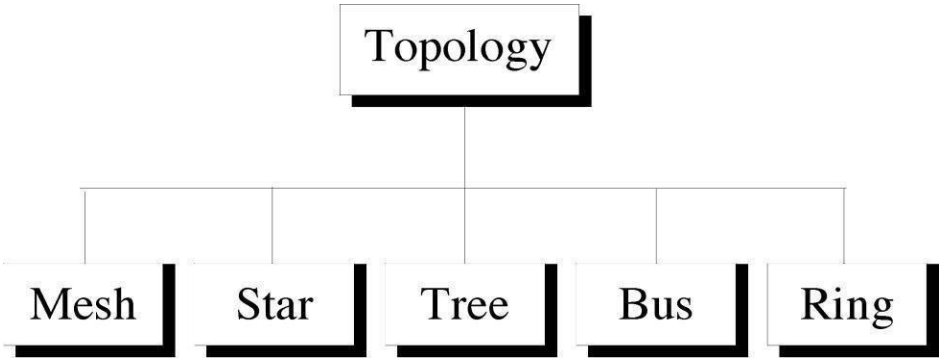

Fig 2.1 Types of Topologies

Mesh topology

In this type of topology, a host is connected to one or multiple hosts. This topology has hosts in point-to-point connection with every other host or may also have hosts which are in point-to-point connection to few hosts only

Hosts in Mesh topology also work as relay for other hosts which do not have direct point-to-point links. Mesh technology comes into two types:

Full Mesh: All hosts have a point-to-point connection to every other host in the network. Thus for every new host n(n-1)/2 connections are required.

It provides the most reliable network structure among all network topologies.

Partially Mesh: Not all hosts have point-to-point connection to every other host. Hosts connect to each other in some arbitrarily fashion. This topology exists where we need to provide reliability to some hosts out of all.

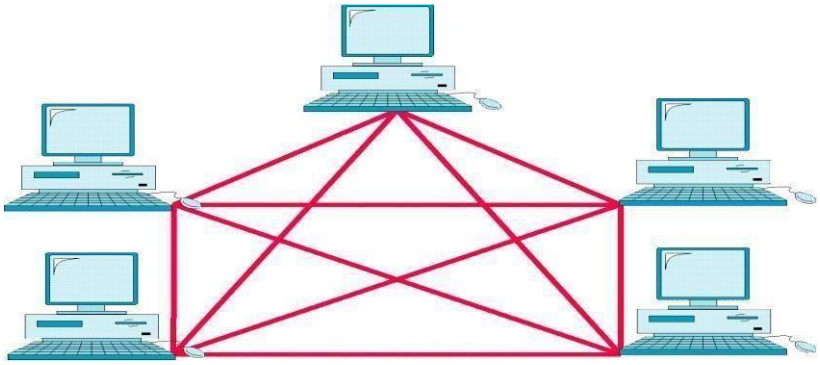

Fig 2.2 Partial Mesh Topology

Star topology

All hosts in Star topology are connected to a central device, known as hub device, using a point-to-point connection. That is, there exists a point to point connection between hosts and hub. The hub device can be any of the following:

Layer-1 device such as hub or repeater Layer-2 device such as switch or bridge Layer 3 device such as router or gateway

As in Bus topology, hub acts as single point of failure. If hub fails, connectivity of all hosts to all other hosts fails.

Every communication between hosts, takes place through only the hub.

Star topology is not expensive as to connect one more host, only one cable is required and configuration is simple.

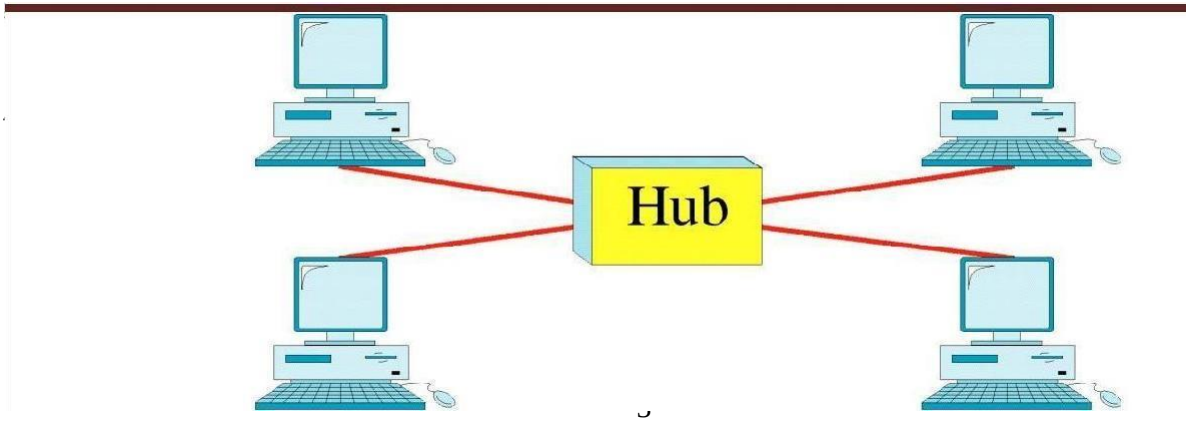

Fig 2.3 Star Topology

Tree Topology

Nodes in a tree are linked to central hub that controls the traffic to the network. Not every device plugs directly to the central hub

Majority of devices connected to secondary hub, that in turns connect to the central hub.

The central hub in the tree is an active hub An active hub contains repeater

The secondary hub may be active or passive

A passive hub provides a simple physical connection between two attached devices.

Repeater which is a hardware device that regenerates the received bit pattern before sending them out

Repeating strengthens transmission and increases the distance a signal can travel.

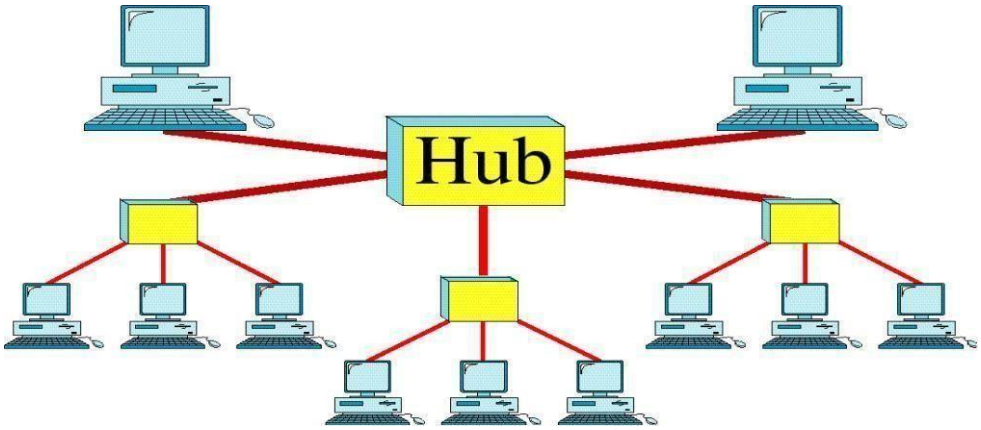

Fig 2.4 Tree Topology

Bus topology

The bus topology is an example of multipoint configurations. One long cable acts as backbone, links all devices in the network. Nodes are connected to the bus cable by drop line and taps.

A drop line is a connection running between the devices and the main cable.

A tap is a connector that either splices in to the main cable or punctures the sheathing of a cable to create a contact with the metallic core

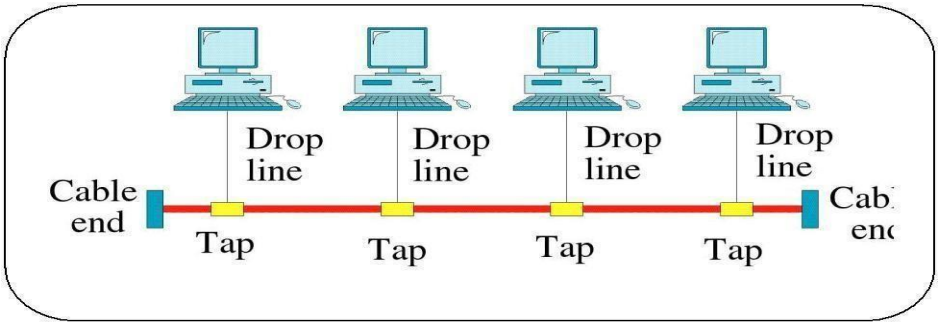

Fig 2.5 Bus Topology

Ring topology

In a ring topology, each device has a dedicated point-to-point line configuration only with the two devices on either side of it.

A signal is passed along the ring in one direction, from a device to device, until it reaches its destination

Each device in the ring incorporates a repeater .when a device receives a signal intended for another device ,its repeater regenerates the bits and passes them along

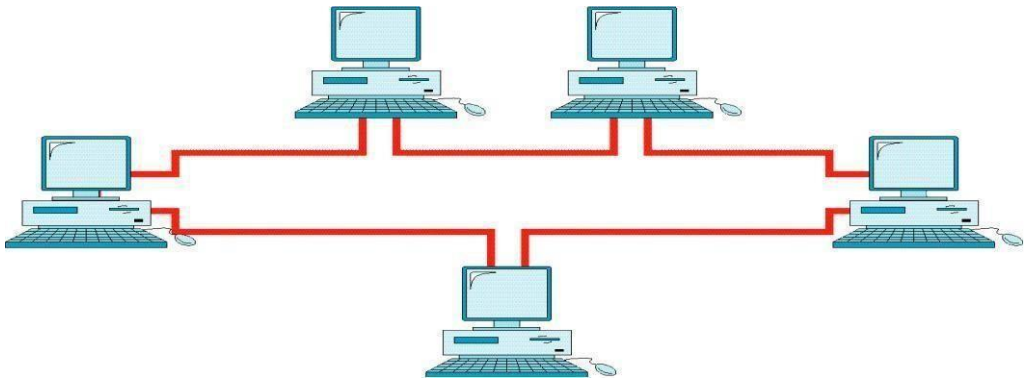

Fig 2.6 Ring Topology

2.2 STANDARDS IN NETWORKING

Standards are necessary in networking to ensure interconnectivity and interoperability between various networking hardware and software components. Without standards we would have proprietary products creating isolated islands of users which cannot interconnect.

Concept of Standard

Standards provide guidelines to product manufacturers and vendors to ensurenational and international interconnectivity. Data communications standards are classified into two categories:

De facto Standard

These are the standards that have been traditionally used and mean **by fact** or **by**

convention

These standards are not approved by any organized body but are adopted by widespread use.

De jure standard

It means by **law** or **by regulation.** These standards are legislated and approved by an body that is officially recognized.

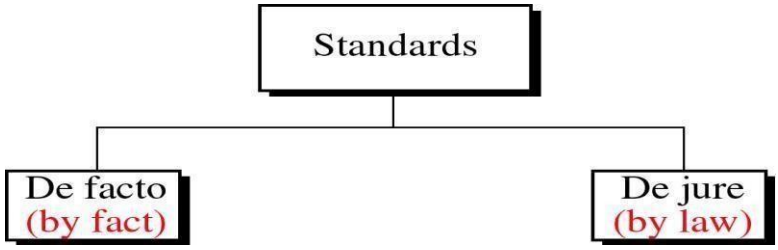

Fig 2.7 various standards

Standard Organizations in field of Networking

Standards are created by standards creation committees, forums, and government regulatory agencies.

Examples of Standard Creation Committees:

1. International Organization for Standardization(ISO)
2. International Telecommunications
3. Union Telecommunications Standard (ITU-T)
4. American National Standards Institute (ANSI)
5. Institute of Electrical & Electronics Engineers(IEEE)
6. Electronic Industries Associates (EIA)

Examples of Forums

1. ATM Forum
2. MPLS Forum
3. Frame Relay Forum

Examples of Regulatory Agencies:

1. Federal Communications Committee (FCC)

IEEE 802 is a family of IEEE standards dealing with local area networks and metropolitan area networks. More specifically, the IEEE 802 standards are restricted to networks carrying variable- size packets. By contrast, in cell relay networks data is transmitted in short, uniformly sized units called cells. Isochronous networks, where data is transmitted as a steady stream of octets, or groups of octets, at regular time intervals, are also out of the scope of this standard. The number 802 was simply the next free number IEEE could assign, though ‑802 is sometimes associated with the date the first meeting was held February 1980. The services and protocols specified in IEEE 802 map to the lower two layers (Data Link and Physical) of the seven-layer OSI networking reference model. In fact, IEEE 802 splits the OSI Data Link Layer into two sub- layers named logical link control (LLC) and media access control (MAC), so the layers can be listed like this:

- Data link layer
- LLC sublayer
- MAC sublayer
- Physical layer

The IEEE 802 family of standards is maintained by the IEEE 802 LAN/MAN Standards Committee (LMSC). The most widely used standards are for the Ethernet family, Token Ring, RFID, Bridging and Virtual Bridged LANs. An individual working group provides the focus for each area.

Wireless LAN and IEEE 802.11

A wireless LAN (WLAN or WiFi) is a data transmission system designed to provide location-independent network access between computing devices by using radio waves rather than a cable infrastructure In the corporate enterprise, wireless LANs are usually implemented as the final link between the existing wired network and a group of client computers, giving these users wireless access to the full resources and services of the corporate network across a building or campus setting. The widespread acceptance of WLANs depends on industry standardization to ensure product compatibility and reliability among the various manufacturers.

The 802.11 specification as a standard for wireless LANS was ratified by the Institute of Electrical and Electronics Engineers (IEEE) in the year 1997. This version of 802.11 provides for 1 Mbps and 2 Mbps data rates and a set of fundamental signaling methods and other services. Like all IEEE 802 standards, the 802.11 standards focus on the bottom two levels the ISO model, the physical layer and link layer (see figure below). Any LAN application, network operating system, protocol, including TCP/IP and Novell NetWare, will run on an 802.11-compliant WLAN as easily as they run over Ethernet.

The major motivation and benefit from Wireless LANs is increased mobility. Untethered from conventional network connections, network users can move about almost without restriction and access

LANs from nearly anywhere.The other advantages for WLAN include cost-effective network setup for hard-to-wire locations such as older buildings and solid-wall structures and reduced cost of ownership-particularly in dynamic environments requiring frequent modifications, thanks to minimal wiring and installation costs per device and user. WLANs liberate users from dependence on hard-wired access to the network backbone, giving them anytime, anywhere network access. This freedom to roam offers numerous user benefits for a variety of work environments, such as:

- Immediate bedside access to patient information for doctors and hospital staff
- Easy, real-time network access for on-site consultants or auditors
- Improved database access for roving supervisors such as production line managers, warehouse auditors, or construction engineers
- Simplified network configuration with minimal MIS involvement for temporarysetups such as trade shows or conference rooms

- Faster access to customer information for service vendors and retailers, resulting in better service and improved customer satisfaction
- Location-independent access for network administrators, for easier on-site troubleshooting and support
- Real-time access to study group meetings and research links for students

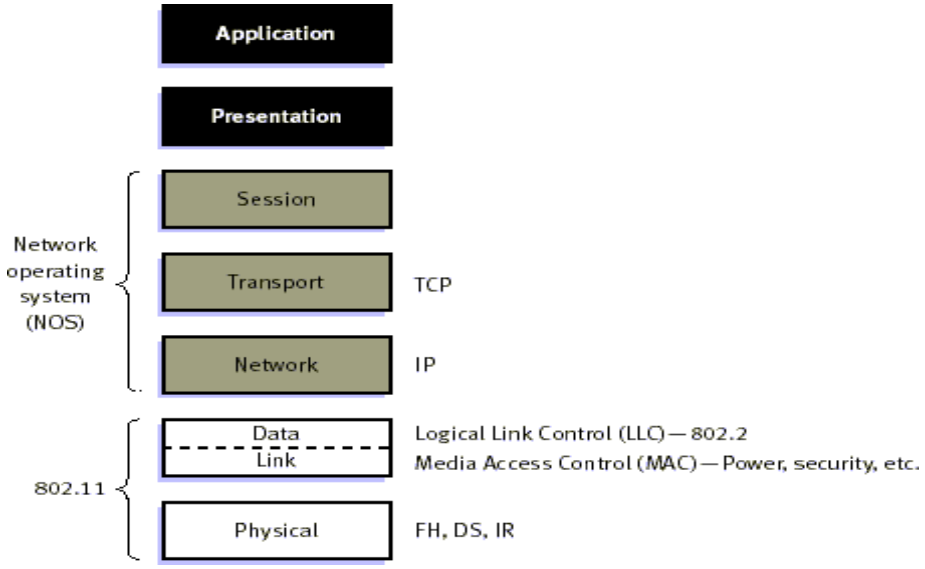

Fig 2.8 IEEE 802.11 and the ISO Model

Wireless sensor network (**WSN**) refers to a group of spatially dispersed and dedicated sensors for monitoring and recording the physical conditions of the environment and organizing the collected data at a central location. WSNs measure environmental conditions like temperature, sound, pollution levels, humidity, wind, and so on.

These are similar to **wireless ad hoc networks** in the sense that they rely on wireless connectivity and spontaneous formation of networks so that sensor data can be transported wirelessly. Sometimes they are called **dust networks**, referring to minute sensors as small as dust. **Smart dust** is a U C Berkeley project sponsored by DARPA. Dust Networks Inc., is one of the early companies that produced wireless sensor network products. WSNs are spatially distributed autonomous sensors to *monitor* physical or environmental conditions, such as temperature, sound, pressure, etc. and to cooperatively pass their data through the network to a main locations. The more modern networks are bi-directional, also enabling *control* of sensor activity. The development of wireless sensor networks was motivated by military applications such as battlefield surveillance; today such networks are used in many industrial and consumer applications, such as industrial process monitoring and control, machine health monitoring, and so on.

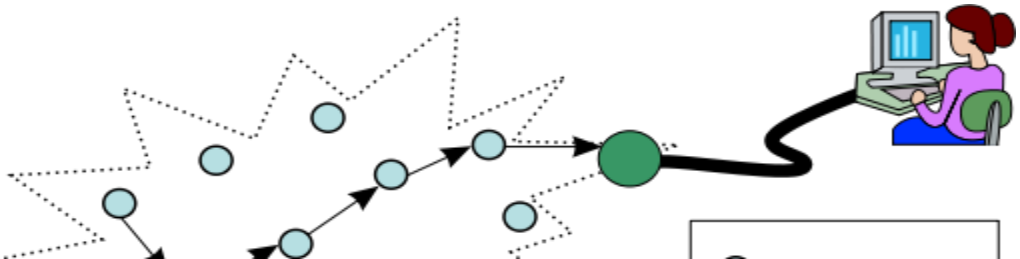

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Fig 2.9 Layout WSN

The WSN is built of "nodes" – from a few to several hundreds or even thousands, where each node is connected to one (or sometimes several) sensors. Each such sensor network node has typically several parts: a radio transceiver with an internal antenna or connection to an external antenna, a microcontroller, an electronic circuit for interfacing with the sensors and an energy source, usually a battery or an embedded form of energy harvesting. A sensor node might vary in size from that of a shoebox down to the size of a grain of dust, although functioning "motes" of genuine microscopic dimensions have yet to be created. The cost of sensor nodes is similarly variable, ranging from a few to hundreds of dollars, depending on the complexity of the individual sensor nodes. Size and cost constraints on sensor nodes result in corresponding constraints on resources such as energy, memory, computational speed and communications bandwidth. The topology of the WSNs can vary from a simple star network to an advanced multi-hop wireless mesh network. The propagation technique between the hops of the network can be routing or flooding. In computer science and telecommunications, wireless sensor networks are an active research area with numerous workshops and conferences arranged each year, for example IPSN, SenSys, and EWSN.

Wireless Sensor Networks (WSNs)

A Wireless sensor network can be defined as a network of devices that can communicate the information gathered from a monitored field through wireless links. The data is forwarded through multiple nodes, and with a gateway, the data is connected to other networks like wireless Ethernet.WSN is a wireless network that consists of base stations and numbers of nodes (wireless sensors).These networks are used to monitor physical or environmental conditions like sound, pressure, temperature and co-operatively pass data through the network to a main location as shown in the figure.

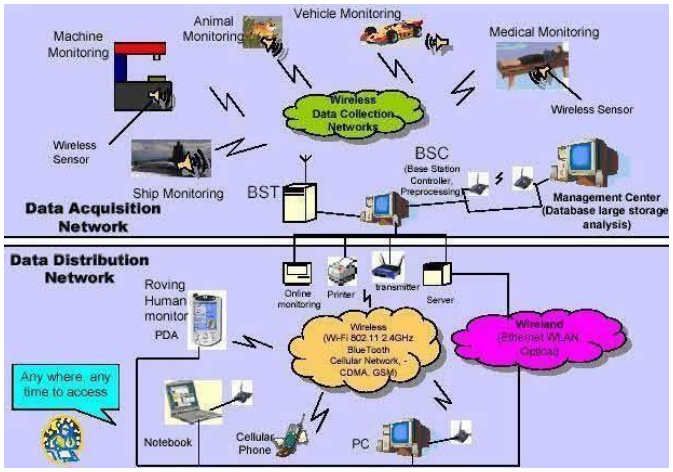

Fig 2.10 Applications

On the environment, the types of networks are decided so that those can be deployed underwater, underground, on land, and so on. Different types of WSNs include:

1. Terrestrial WSNs
2. Underground WSNs
3. Underwater WSNs
4. Multimedia WSNs
5. Mobile WSNs

1. Terrestrial WSNs

Terrestrial WSNs are capable of communicating base stations efficiently, and consist of hundreds to thousands of wireless sensor nodes deployed either in unstructured (ad hoc) or structured (Preplanned) manner. In an unstructured mode, the sensor nodes are randomly distributed within the target area that is dropped from a fixed plane. The preplanned or structured mode considers optimal placement, grid placement, and 2D, 3D placement models.

In this WSN, the battery power is limited; however, the battery is equipped with solar cells as a secondary power source. The Energy conservation of these WSNs is achieved by using low duty cycle operations, minimizing delays, and optimal routing, and so on.

2. Underground WSNs

The underground wireless sensor networks are more expensive than the terrestrial WSNs in terms of deployment, maintenance, and equipment cost considerations and careful planning. The WSNs networks consist of a number of sensor nodes that are hidden in the ground to monitor underground conditions. To relay information from the sensor nodes to the base station, additional sink nodes are located above the ground.

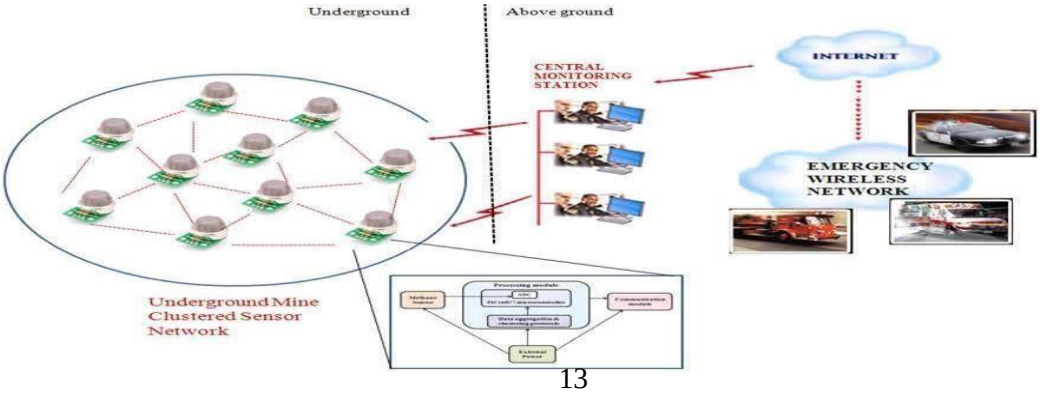

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Fig2 .10 Underground WSN

The underground wireless sensor networks deployed into the ground are difficult to recharge. The sensor battery nodes equipped with a limited battery power are difficult to recharge. In addition to this, the underground environment makes wireless communication a challenge due to high level of attenuation and signal loss.

3. Under Water WSNs

More than 70% of the earth is occupied with water. These networks consist of a number of sensor nodes and vehicles deployed under water. Autonomous underwater vehicles are used for gathering data from these sensor nodes. A challenge of underwater communication is a long propagation delay, and bandwidth and sensor failures.

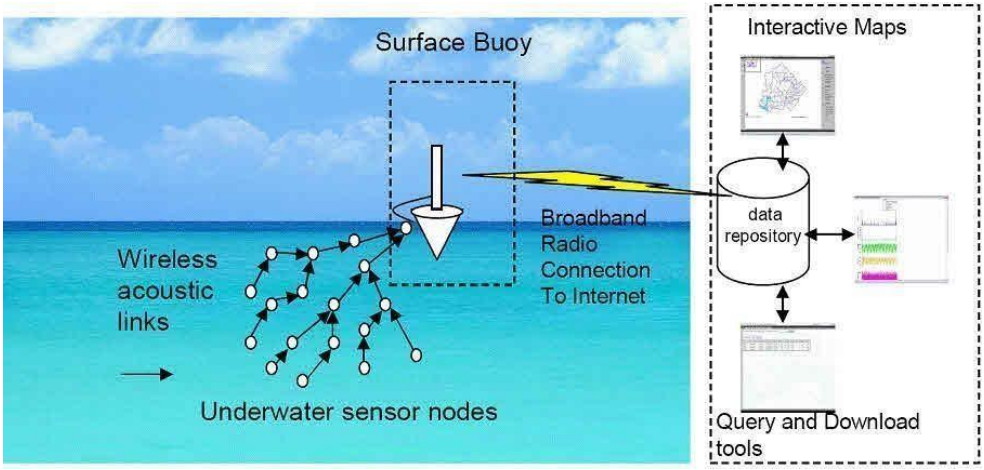

Fig 2.10 Underwater WSN

Under water WSNs are equipped with a limited battery that cannot be recharged or replaced. The issue of energy conservation for under water WSNs involves the development of underwater communication and networking techniques.

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4. Multimedia WSNs

Muttimedia wireless sensor networks have been proposed to enable tracking and monitoring of events in the form of multimedia, such as imaging, video, and audio. These networks consist of low-cost sensor nodes equipped with micrpphones and cameras.These nodes are interconnected with each other over a wireless connection for data compression, data retrieval and correlation.

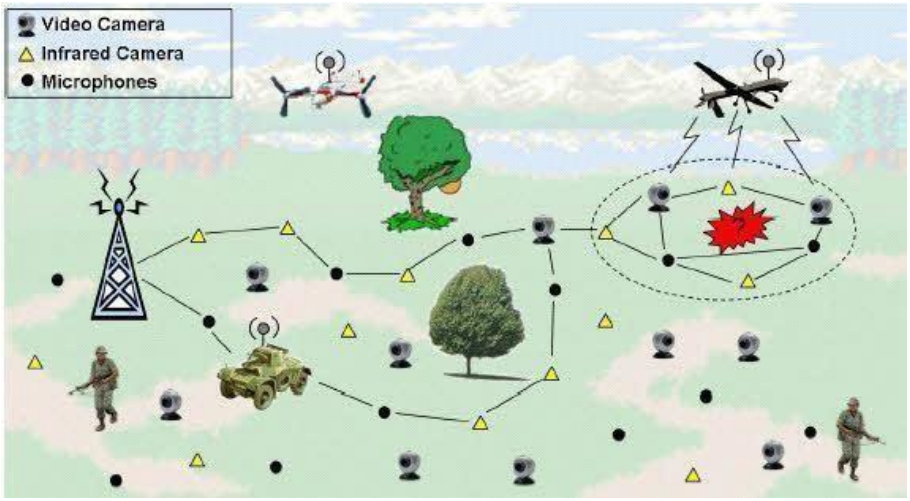


Fig 2.11 MultimediaWSN

The challenges with the multimedia WSN include high energy consumption, high bandwidth requirements, data processing and compressing techniques. In addition to this, multimedia contents require high bandwidth for the contents to be delivered properly and easily.

5. Mobile WSNs

These networks consist of a collection of sensor nodes that can be moved on their own and can be interacted with the physical environment. The mobile nodes have the ability to compute sense and communicate.

The mobile wireless sensor networks are much more versatile than the static sensor networks. The advantages of MWSN over the static wireless sensor networks include better and improved coverage, better energy efficiency, superior channel capacity, and so on.

Limitations of Wireless Sensor Networks

1. Possess very little storage capacity – a few hundred kilobytes

2. Possess modest processing power-8MHz

3. Works in short communication range – consumes a lot of power

4. Requires minimal energy – constrains protocols

5. Have batteries with a finite life time

6. Passive devices provide little energy

2.3 UMTS ARCHITECTURE

The UMTS architecture is required to provide a greater level of performance to that of the original GSM network. However as many networks had migrated through the use of GPRS and EDGE, they already had the ability to carry data. Accordingly many of the elementsrequired for the WCDMA / UMTS network architecture were seen as a migration. This considerably reduced the cost of implementing the UMTS network as many elements were in place or needed upgrading.

With one of the major aims of UMTS being to be able to carry data, the UMTS network architecture was designed to enable a considerable improvement in data performance over that provided for GSM.

UMTS network constituents

The UMTS network architecture can be divided into three main elements:

1. ***User Equipment (UE):*** The User Equipment or UE is the name given to what was previous termed the mobile, or cellphone. The new name was chosen because the considerably greater functionality that the UE could have. It could also be anything between a mobile phone used for talking to a data terminal attached to a computer with no voice capability.

2. ***Radio Network Subsystem (RNS):*** The RNS also known as the UMTS Radio Access Network, UTRAN, is the equivalent of the previous Base Station Subsystem or BSS in GSM. It provides and manages the air interface for the overall network.

3. ***Core Network:*** The core network provides all the central processing and management for the system. It is the equivalent of the GSM Network Switching Subsystem or NSS.

The core network is then the overall entity that interfaces to external networks including the public phone network and other cellular telecommunications networks.

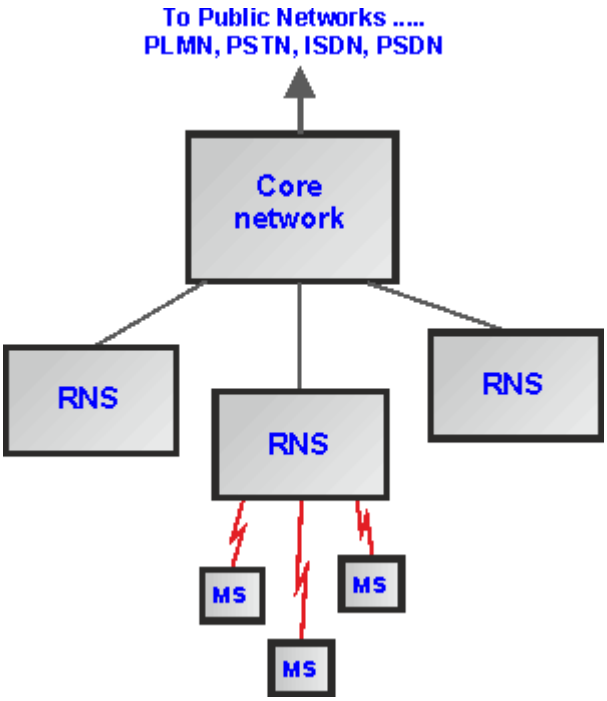


Fig 2.12 UMTS Network Architecture Overview

User Equipment, UE

The USER Equipment or UE is a major element of the overall 3G UMTS network architecture. It forms the final interface with the user. In view of the far greater number of applications and facilities that it can perform, the decision was made to call it a user equipment rather than a mobile. However it is essentially the handset (in the broadest terminology), although having access to much higher speed data communications, it can be much more versatile, containing many more applications. It consists of a variety of different elements including RF circuitry, processing, antenna, battery, etc.

There are a number of elements within the UE that can be described separately:

- ***UE RF circuitry:*** The RF areas handle all elements of the signal, both for the receiver and for the transmitter. One of the major challenges for the RF power amplifier was to reduce the power consumption. The form of modulation used for W-CDMA requires the use of a linear amplifier. These inherently take more current than non linear amplifiers which can be used for the form of modulation used on GSM. Accordingly to maintain battery life, measures were introduced into many of the designs to ensure the optimum efficiency.

- ***Baseband processing:*** The base-band signal processing consists mainly of digital circuitry. This is considerably more complicated than that used in phones for previous generations. Again this has been optimised to reduce the current consumption as far as possible.

- ***Battery:*** While current consumption has been minimised as far as possible within the circuitry of the phone, there has been an increase in current drain on the battery. With users expecting the same lifetime between charging batteries as experienced on the previous generation phones, this has necessitated the use of new and improved battery technology. Now Lithium Ion (Li-ion) batteries are used. These phones to remain small and relatively light while still retaining or even improving the overall life between charges.
- ***Universal Subscriber Identity Module, USIM:*** The UE also contains a SIM card, although in the case of UMTS it is termed a USIM (Universal Subscriber Identity Module). This is a more advanced version of the SIM card used in GSM and other systems, but embodies the same types of information. It contains the International Mobile Subscriber Identity number (IMSI) as well as the Mobile Station International ISDN Number (MSISDN). Other information that the USIM holds includes the preferred language to enable the correct language information to be displayed, especially when roaming, and a list of preferred and prohibited Public Land Mobile Networks(PLMN).

The USIM also contains a short message storage area that allows messages to stay with the user even when the phone is changed. Similarly "phone book" numbers and call information of the numbers of incoming and outgoing calls are stored.

The UE can take a variety of forms, although the most common format is still a version of a "mobile phone" although having many data capabilities. Other broadband dongles are also being widely used.

UMTS Radio Network Subsystem

This is the section of the 3G UMTS / WCDMA network that interfaces to both the UE and the core network. The overall radio access network, i.e. collectively all the Radio Network Subsystem is known as the UTRAN UMTS Radio Access Network.

The radio network subsystem is also known as the UMTS Radio Access Network or UTRAN.

3G UMTS Core Network

The 3G UMTS core network architecture is a migration of that used for GSM with further elements overlaid to enable the additional functionality demanded by UMTS.

In view of the different ways in which data may be carried, the UMTS core network may be split into two different areas:

- ***Circuit switched elements:*** These elements are primarily based on the GSM network entities and carry data in a circuit switched manner, i.e. a permanent channel for the duration of the call.

- ***Packet switched elements:*** These network entities are designed to carry packet data. This enables much higher network usage as the capacity can be shared and data is carried as packets which are routed according to their destination.

Some network elements, particularly those that are associated with registration are sharedby both domains and operate in the same way that they did with GSM.

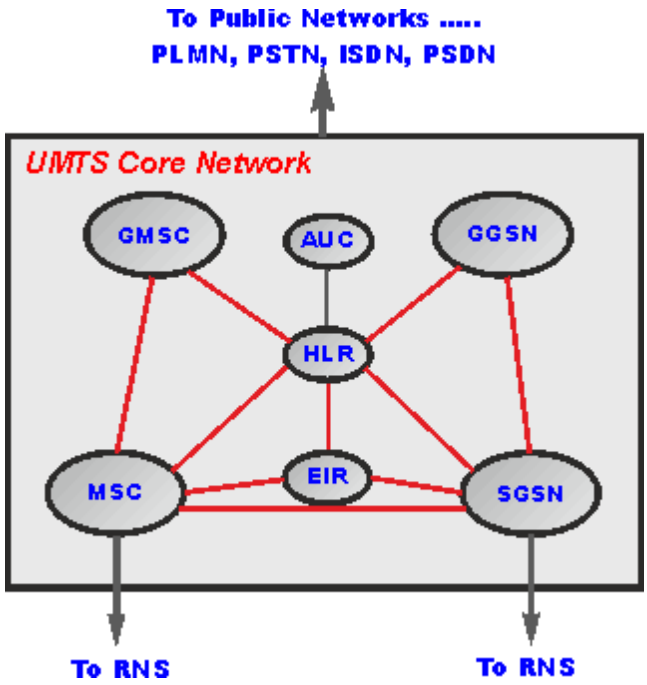


Fig 2.13 UMTS Core Network

Circuits witched elements

The circuit switched elements of the UMTS core network architecture include the following network entities:

- ***Mobile switching centre (MSC):*** This is essentially the same as that within GSM, and it manages the circuit switched calls under way.
- ***Gateway MSC (GMSC):*** This is effectively the interface to the external networks.

Packet switched elements The packet switched elements of the 3G UMTS core network architecture include the following network entities:

- ***Serving GPRS Support Node (SGSN):*** As the name implies, this entity was first developed when GPRS was introduced, and its use has been carried over into the UMTS

network architecture. The SGSN provides a number of functions within the UMTS network architecture.

- Mobility management When a UE attaches to the Packet Switched domain of the UMTS Core Network, the SGSN generates MM information based on the mobile's current location.

- Session management: The SGSN manages the data sessions providing the required quality of service and also managing what are termed the PDP (Packet data Protocol) contexts, i.e. the pipes over which the data is sent.

- Interaction with other areas of the network: The SGSN is able to manage its elements within the network only by communicating with other areas of the network, e.g. MSC and other circuit switched areas.

- Billing: The SGSN is also responsible billing. It achieves this by monitoring the flow of user data across the GPRS network. CDRs (Call Detail Records) are generated by the SGSN before being transferred to the charging entities (Charging Gateway Function, CGF).

- ***Gateway GPRS Support Node (GGSN):*** Like the SGSN, this entity was also first introduced into the GPRS network. The Gateway GPRS Support Node (GGSN) is the central element within the UMTS packet switched network. It handles inter-working between the UMTS packet switched network and external packet switched networks, and can be considered as a very sophisticated router. In operation, when the GGSN receives data addressed to a specific user, it checks if the user is active and then forwards the data to the SGSN serving the particular UE.
- **Shared elements** The shared elements of the UMTS core network architecture include the following network entities:
- **Home location register (HLR):** This database contains all the administrative information about each subscriber along with their last known location. In this way, the UMTS network is able to route calls to the relevant RNC / Node B. When a user switches on their UE, it registers with the network and from this it is possible to determine which Node B it communicates with so that incoming calls can be routed appropriately. Even when the UE is not active (but switched on) it re-registers periodically to ensure that the network (HLR) is aware of its latest position with their current or last known location on the network.

- **Equipment identity register (EIR):** The EIR is the entity that decides whether a given UE equipment may be allowed onto the network. Each UE equipment has a number known as

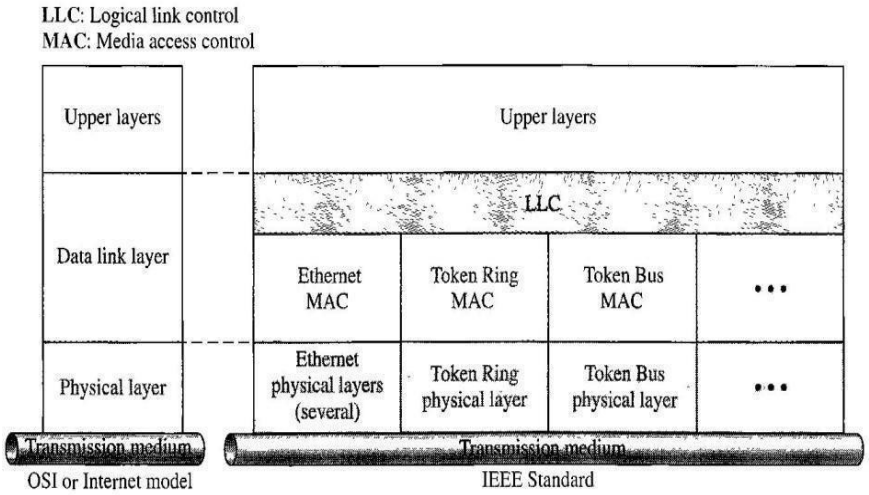
the International Mobile Equipment Identity. This number, as mentioned above, is installed in the equipment and is checked by the network during registration.

- **Authentication centre (AuC) :** The AuC is a protected database that contains the secret key also contained in the user's USIM card.

IEEE STANDARDS

The relationship of the 802 Standard to the traditional OSI model is shown in the figure. The IEEE has subdivided the data link layer into two sublayers: logical link control (LLC) and media access control (MAC). IEEE has also created several physical layer standards for different LAN protocols.

Fig 2.14 IEEE Standards



Data Link Layer

The data link layer in the IEEE standard is divided into two sublayers: LLC and MAC.

Logical Link Control (LLC)

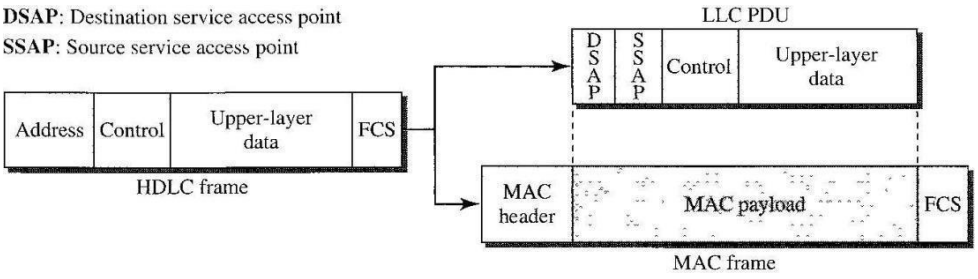
In IEEE Project 802, flow control, error control, and part of the framing duties are collected into one sublayer called the logical link control. Framing is handled in both the LLC sublayer and the MACsublayer.

The LLC provides one single data link control protocol for all IEEE LANs. In this way, the LLC is different from the media access control sublayer, which provides different protocols for different LANs. A single LLC protocol can provide interconnectivity between different LANs because it makes the MAC sublayer transparent.

Framing LLC defines a protocol data unit (PDU) that is somewhat similar to that of HDLC. The header contains a control field like the one in HDLC; this field is used for flow and error control. The two other header fields define the upper-layer protocol at the source and destination that uses LLC.

These fields are called the destination service access point (DSAP) and the source Gif service access point (SSAP). The other fields defined in a typical data link control protocol such as HDLC are moved to the MAC sublayer. In other words, a frame defined in HDLC is divided into a PDU at the LLC sublayer and a frame at the MAC sublayer, as shown in figure.

Fig 2.14 Framing



Need for LLC The purpose of the LLC is to provide flow and error control for the upper-layer protocols that actually demand these services. For example, if a LAN or several LANs are used in an isolated system, LLC may be needed to provide flow and error control for the application layer protocols. However, most upper-layer protocols such as IP, do not use the services of LLC. ***Media Access Control (MAC)***

IEEE Project 802 has created a sublayer called media access control that defines the specific access method for each LAN. For example, it defines CSMA/CD as the media access method for Ethernet LANs and the token-passing method for Token Ring and Token Bus LANs. Part of the framing function is also handled by the MAC layer.In contrast to the LLC sublayer, the MAC sublayer contains a number of distinct modules; each defines the access method and the framing format specific to the corresponding LAN protocol.

Physical Layer

The physical layer is dependent on the implementation and type of physical media used. IEEE defines detailed specifications for each LAN implementation. For example, although there is only one MAC sublayer for Standard Ethernet, there is a different physical layer specification for each Ethernet implementations.

Key features of LANs are summarized below:

- Limited geographical area – which is usually less than 10 Km and more than 1 m. High Speed – 10 Mbps to 1000 Mbps (1 Gbps) and more
- High Reliability – 1 bit error in 10^{11} bits.
- Transmission Media – Guided and unguided media, mainly guided media is used; except in a situation where infrared is used to make a wireless LAN in a room.
- Topology – It refers to the ways in which the nodes are connected. There are various topologies used.
- Medium-Access Control Techniques –Some access control mechanism is needed
- to decide which station will use the shared medium at a particular point in time. In this lesson we shall discuss various LAN standards proposed by the IEEE 8.2 committee with the following goals in mind:

- To promote compatibility

- Implementation with minimum efforts Accommodate the need for diverse applications

- For the fulfillment of the abovementioned goals, the committee came up with a bunch of LAN standards collectively known as IEEE 802 LANs as shown in Fig. 5.3.1. To satisfy diverse requirements, the standard includes CSMA/CD, Token bus, Token
- Ring medium access control techniques along with different topologies. All these standards differ at the physical layer and MAC sublayer, but are compatible at the data link layer.

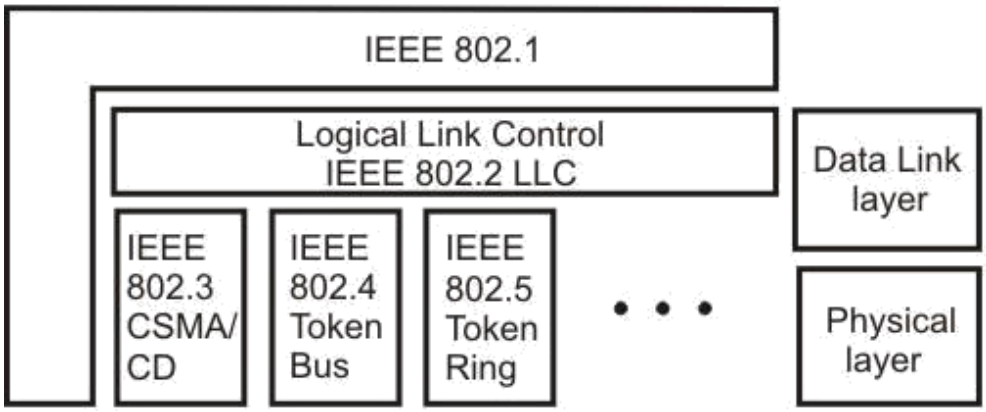


Figure 2.15 IEEE 802 Legacy LANs

The **802.1** sublayer gives an introduction to set of standards and gives the details of the interface primitives. It provides relationship between the OSI model and the 802 standards. The **802.2** sublayer describes the **LLC** (logical link layer), which is the upper part of the data link layer. LLC facilitate error control and flow control for reliable communication. It appends a header containing sequence number and acknowledgement number. And offers the following three types of services:

- Unreliable datagram service Acknowledged datagram service
- Reliable connection oriental service

The standards 802.3, 802.4 and 802.5 describe three LAN standards based on the CSMA/CD, token bus and token ring, respectively. Each standard covers the physical layer and MAC sublayer protocols. In the following sections we shall focus on these three LAN standards.

IEEE 802.3 and Ethernet

Ethernet - A Brief History

The original Ethernet was developed as an experimental coaxial cable network in the 1970s by Xerox Corporation to operate with a data rate of 3 Mbps using a carrier sense multiple access collision detection (CSMA/CD) protocol for LANs with sporadic traffic requirements. Success with that project attracted early attention and led to the 1980 joint development of the 10-Mbps Ethernet Version 1.0 specification by the three-company consortium: Digital Equipment Corporation, Intel Corporation, and Xerox Corporation.

The original IEEE 802.3 standard was based on, and was very similar to, the Ethernet Version 1.0 specification. The draft standard was approved by the 802.3 working group in 1983 and was subsequently published as an official standard in 1985 (ANSI/IEEE Std. 802.3-1985). Since then, a number of supplements to the standard have been defined to take advantage of improvements in the technologies and to support additional network media and higher data rate capabilities, plus several new optional network access control features. From then onwards, the term *Ethernet* refers to the family of local-area network (LAN) products covered by the IEEE 802.3 standard that defines what is commonly known as the CSMA/CD protocol. Three data rates are currently defined for operation over optical fiber and twisted-pair cables:

- 10 Mbps—10Base-T Ethernet
- 100 Mbps—Fast Ethernet
- 1000 Mbps—Gigabit Ethernet

Ethernet has survived as the major LAN technology (it is currently used for approximately 85 percent of the world's LAN-connected PCs and workstations) because its protocol has the following characteristics:

- It is easy to understand, implement, manage, and maintain It allows low-cost network implementations
- It provides extensive topological flexibility for network installation
- It guarantees successful interconnection and operation of standards-compliant products, regardless of manufacturer

Ethernet Architecture

Ethernet architecture can be divided into two layers:

- **Physical layer:** this layer takes care of following functions.

- Encoding and decoding Collision detection Carrier sensing Transmission and receipt

Data link layer: Following are the major functions of this layer. Station interface

Data Encapsulation /Decapsulation Link management

Collision Management

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 (10 t Mbps), Fast Ethernet (100 Mbps), Gigabit Ethernet (1 Gbps), and Ten-Gigabit Ethernet (10 Gbps), as shown in the figure:

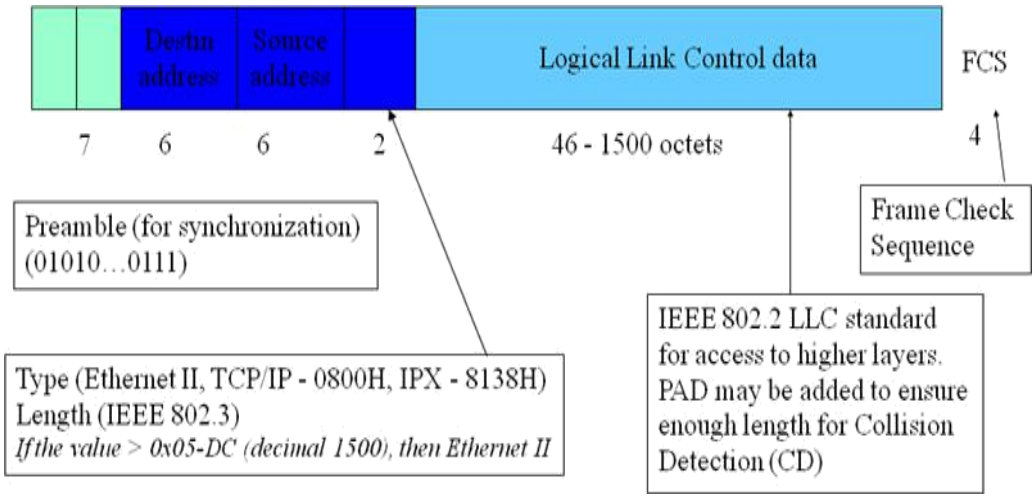


Figure 2.16 IEEE 802 Legacy LANs

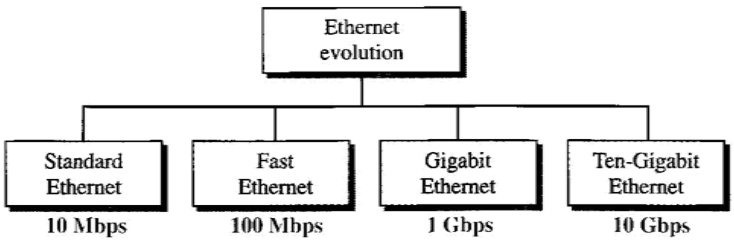


Fig 2.17 Ethernet Evolution

MAC Sublayer

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.

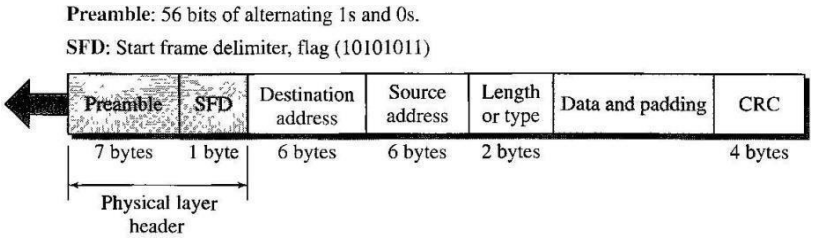


Fig 2.18 Frame Format

Acknowledgments must be implemented at the higher layers. The format of the MAC frame is shown in the figure.

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 (1 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 ofthe sender of the packet.

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.

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 figure.

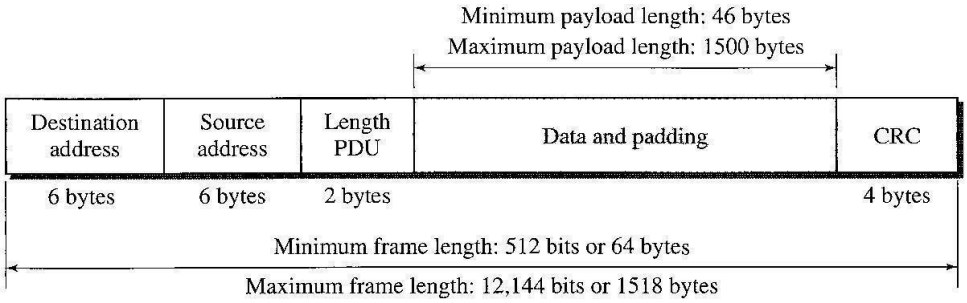


Fig 2.18 Frame Length

The minimum length restriction is required for the correct operation of CSMA/CD. 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

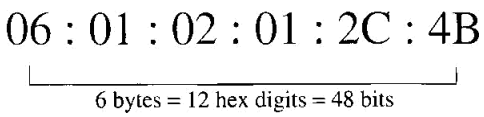


Fig 2.19 Addressing

Each station on an Ethernet network (such as a PC, workstation, or printer) has its own network interface card (NIC). The NIC fits inside the station and provides the station with a 6- byte physical address. As shown in the figure, the Ethernet address is 6 bytes (48 bits), normally written in hexadecimal notation, with a colon between the bytes26

Unicast, Multicast, and Broadcast Addresses 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. The following 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.

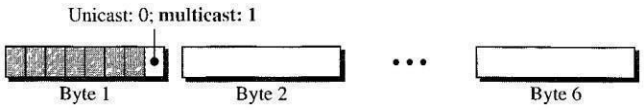


Fig 2.20 Frame Format

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 one4o-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

Access Method: CSMA/CD

Standard Ethernet uses 1-persistent CSMA/CD

Slot Time In an Ethernet network, the round-trip time required for a frame to travel from one end of a maximum-length network to the other plus the time needed to send the jam sequence is called the slot time.

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 51.2μs.

Slot Time and Collision The choice of a 512-bit slot time was not accidental. It was chosen to allow the proper functioning of CSMA/CD. To understand the situation, let us consider two cases.

In the first case, we assume that the sender sends a minimum-size packet of 512 bits. Before the sender can send the entire packet out, the signal travels through the network and reaches the end of the network. If there is another signal at the end of the network (worst case), a collision occurs. The sender has the opportunity to abort the sending of the frame and to send a jam sequence to inform other stations of the collision. The roundtrip time plus the time required to send the jam sequence should be less than the time needed for the sender to send the minimum frame, 512 bits. The sender needs to be aware of the collision before it is too late, that is, before it has sent the entire frame.

In the second case, the sender sends a frame larger than the minimum size (between 512 and

1518 bits). In this case, if the station has sent out the first 512 bits and has not heard a collision, it is

$$\text{MaxLength} = \text{PropagationSpeed} \times \frac{\text{SlotTime}}{2}$$

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

guaranteed that collision will never occur during the transmission of this frame. The reason is that the signal will reach the end of the network in less than one-half the slot time. If all stations follow the CSMA/CD protocol, they have already sensed the existence of the signal (carrier) on the line and have refrained from sending. If they sent a signal on the line before one- half of the slot time expired, a collision has occurred and the sender has sensed the collision. In other words, collision can only occur during the first half of the slot time, and if it does, it can be sensed by the sender during the slot time. This means that after the sender sends the first 512 bits, it is guaranteed that collision will not occur during the transmission of this frame. The medium belongs to the sender, and no other station will use it. In other words, the sender needs to listen for a collision only during the time the first 512 bits are sent.

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 x108 m/s (two-thirds of the rate for propagation in air). For traditional Ethernet, we calculate 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.

$$\text{MaxLength} = 2500 \text{ m}$$

Physical Layer

The Standard Ethernet defines several physical layer implementations; four of the most common, are shown in figure.Because Ethernet devices implement only the bottom two layers of the OSI protocol stack, they are typically implemented as network interface cards (NICs) that plug into the host device's motherboard, or presently built-in in the motherboard. Various types cabling supported by the standard are shown in Fig. 5.3.2. The naming convention is a concatenation of three terms indicating the transmission rate, the transmission method, and the media type/signal encoding. Consider for example, 10Base-T. where 10 implies transmission rate of 10 Mbps, Base represents that it uses baseband signaling, and T refers to twisted-pair cables as transmission media. Various standards are discussed below:

Encoding and Decoding

All standard implementations use digital signaling (baseband) at 10 Mbps. At the sender, data are converted to a digital signal using the Manchester scheme; at the receiver, the received signal is interpreted as Manchester and decoded into data. The figure shows the encoding scheme for Standard

Ethernet.

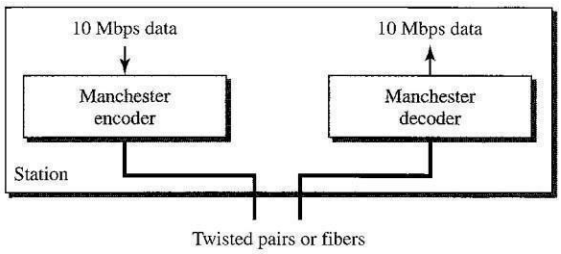


Fig 2.21 Encoding and Decoding

10Base5: Thick Ethernet

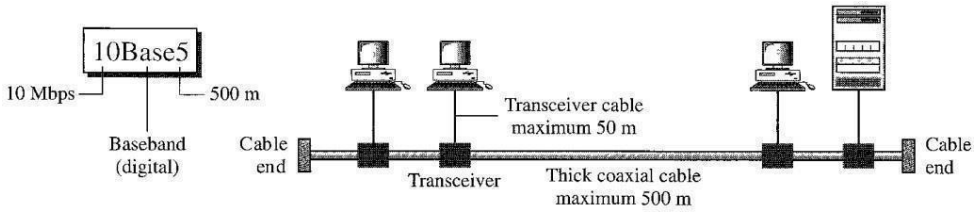


Fig 2.22 Thick Ethernet

10Base5 was the first Ethernet specification to use a bus topology with an external transceiver (transmitter/receiver) connected via a tap to a thick coaxial cable. The transceiver is responsible for transmitting, receiving, and detecting collisions.The transceiver is connected to the station via a transceiver cable that provides separate paths for sending and receiving. This means that collision can only happen in the coaxial cable.The maximum length of the coaxial cable must not exceed 500 m, otherwise, there is excessive degradation of the signal. If a length of more than 500 m is needed, up to five segments, each a maximum of 500-meter, can be connected using repeaters.

10Base2: Thin Ethernet

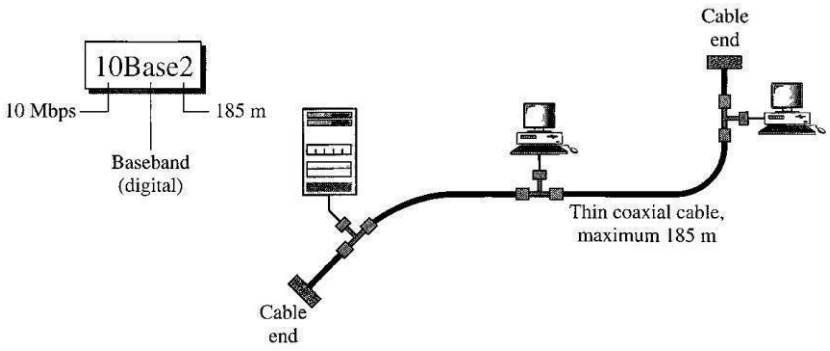


Fig 2.23 Thin Ethernet

10Base2 also uses a bus topology, but the cable is much thinner and more flexible. The cable can be bent to pass very close to the stations. In this case, the transceiver is normally part of the network interface card (NIC), which is installed inside the station.

Note that the collision here occurs in the thin coaxial cable. This implementation is more cost effective than 10Base5 because thin coaxial cable is less expensive than thick coaxial and the tee connections are much cheaper than taps. Installation is simpler because the thin coaxial cable is very flexible. However, the length of each segment cannot exceed 185 m (close to 200 m) due to the high level of attenuation in thin coaxial cable.

10Base- T: Twisted-Pair Ethernet

The third implementation is called 10Base-T or twisted-pair Ethernet. 10Base-T uses a physical star topology. The stations are connected to a hub via two pairs of twisted cable.

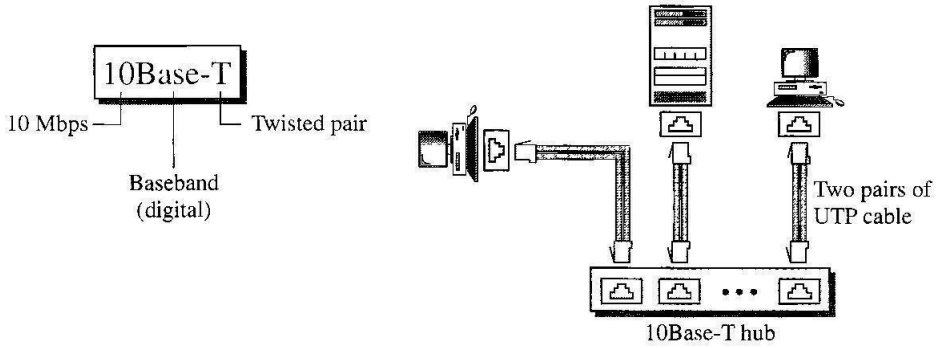


Fig 2.24 Twisted Pair Ethernet

Note that two pairs of twisted cable create two paths (one for sending and one for receiving) between the station and the hub. Any collision here happens in the hub. Compared to 10Base5 or 10Base2, we can see that the hub actually replaces the coaxial cable as far as a collision is concerned. The maximum length of the twisted cable here is defined as 100 m, to minimize the effect of attenuation in the twisted cable.

10Base-F: Fiber Ethernet

10Base-F uses a star topology to connect stations to a hub. The stations are connected to the hub using two fiber-optic cables.

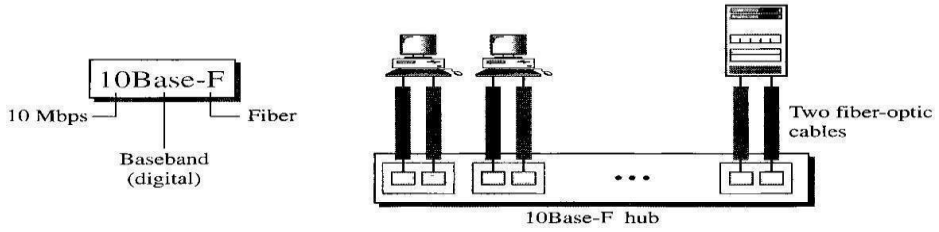


Fig 2.25 Fibre Ethernet

No Need for CSMA/CD

In full-duplex switched Ethernet, there is no need for the CSMA/CD method. In a full- duplex switched Ethernet, each station is connected to the switch via two separate links. Each station or switch can send and receive independently without worrying about collision. Each link is a point- to-point dedicated path between the station and the switch. There is no longer a need for carrier sensing; there is no longer a need for collision detection. The job of the MAC layer becomes much easier. The carrier sensing and collision detection functionalities of the MAC sublayer can be turned off.

MAC Control Layer

Standard Ethernet was designed as a connectionless protocol at the MAC sublayer. There is no explicit flow control or error control to inform the sender that the
frame has arrived at the destination without error. When the receiver receives the frame, it does not send any positive or negative acknowledgment.

To provide for flow and error control in full-duplex switched Ethernet, a new sublayer, called the MAC control, is added between the LLC sublayer and the MAC sublayer.

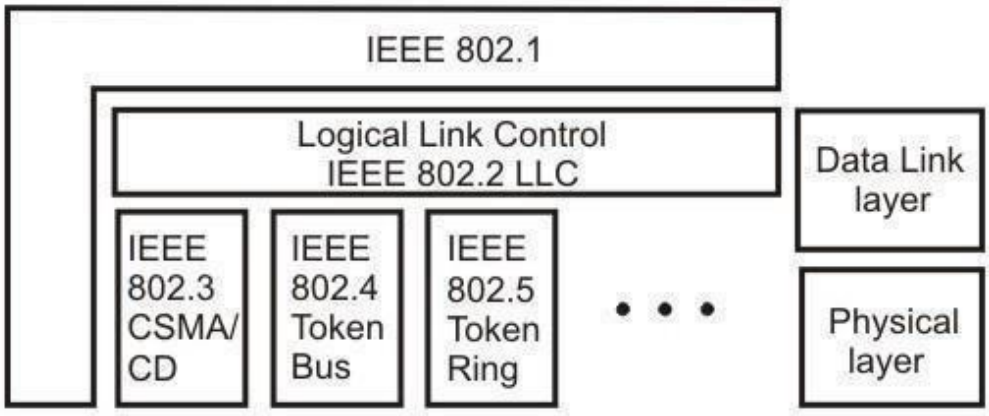


Fig2.26 IEEE 802 Legacy LANs

Token Ring (IEEE 802.5)

Token Ring: A Brief History

Originally, IBM developed Token Ring network in the 1970s. It is still IBM's primary local-area network (LAN) technology. The related IEEE 802.5 specification is almost identical to and completely compatible with IBM's Token Ring network. In fact, the IEEE 802.5 specification was modeled after IBM Token Ring, and on the same lines. The term *Token Ring* is generally used to refer

to both IBM's Token Ring network and IEEE 802.5 networks.

Before going into the details of the Token Ring protocol, let's first discuss the motivation behind it. As already discussed, the medium access mechanism used by Ethernet (CSMA/CD) may results in collision. Nodes attempt to a number of times before they can actually transmit, and even when they start transmitting there are chances to encounter collisions and entire transmission need to be repeated. And all this become worse one the traffic is heavy i.e. all nodes have some data to transmit. Apart from this there is no way to predict either the occurrence of collision or delays produced by multiple stations attempting to capture the link at the same time. So all these problems with the Ethernet gives way to an alternate LAN technology, Token Ring. Token Ring and IEEE802.5 are based on token passing MAC protocol with ring topology. They resolve the uncertainty by giving each station a turn on by one. Each node takes turns sending the data; each station may transmit data during its turn. The technique that coordinates this turn mechanism is called Token passing; as a Token is passed in the network and the station that gets the token can only transmit. As one node transmits at a time, there is no chance of collision. We shall discuss the detailed operation in next section.

Stations are connected by point-to-point links using repeaters. Mainly these links are of shielded twisted-pair cables. The repeaters function in two basic modes: Listen mode, Transmit mode. A disadvantage of this topology is that it is vulnerable to link or station failure. But a few measures can be taken to take care of it.

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UNIT – III COMPUTER NETWORKS- SECA1604

UNIT III PHYSICAL LAYER AND DATALINK LAYER

Design Factors for Transmission Media

Bandwidth: All other factors remaining constant, the greater the band-width of a signal, the higher the data rate that can be achieved.

Transmission impairments: Limit the distance a signal can travel.

Interference: Competing signals in overlapping frequency bands can distort or wipe out a signal.

Number of receivers: Each attachment introduces some attenuation and distortion, limiting distance and/or data rate.

TYPES OF TRANSMISSION MEDIA

1. Conducted or Guided Media :

Use a conductor such as a wire or a fiber optic cable to move the signal from sender to receiver.

2. Wireless or Unguided Media:

Use radio waves of different frequencies and do not need a wire or cable conductor to transmit signals.

Guided Transmission Media

Guided media includes everything that 'guides' the transmission. That usually takes the form of some sort of a wire. Usually copper, but can also be an optical fiber.

Transmission capacity depends on the distance and on whether the medium is point-to-point or multipoint. Ex: twisted pair wires, coaxial cables, optical fiber

Twisted Pair Wires

A transmission medium consisting of pairs of twisted copper wires arranged in a regular spiral pattern to minimize the electromagnetic interference between adjacent pairs .Often used at customer facilities and also over distances to carry voice as well as data communications .Low frequency transmission medium.

We can transmit 1 Mbps over short distances (less than 100m). They are mainly used to transmit analog signals, but they can be used for digital signals.

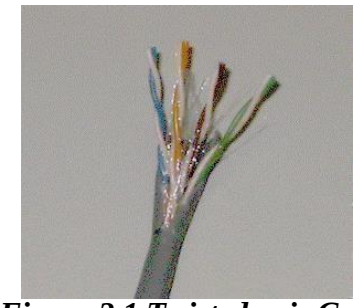


Figure 3.1 Twisted pair Cable

Twisted Pair Advantages

- Inexpensive and readily available Flexible and light weight
- Easy to work with and install

Twisted Pair Disadvantages

- Susceptibility to interference and noise Attenuation problem
- For analog, repeaters needed every 5-6km
- For digital, repeaters needed every 2-3km relatively low bandwidth (3000Hz)

Applications

They are used in telephone lines to provide voice and data channels. Local area networks, such as 10 Base-T and 100 Base-T also use twisted-pair cables.

Coaxial Cable (or Coax)

- In its simplest form, coaxial consists of a core made of solid copper surrounded by insulation, a braided metal shielding, and an outer cover.
- A transmission medium consisting of thickly insulated copper wire, which can transmit a large volume of data than twisted wire.

Coax Advantages

Higher bandwidth

- to 600MHz

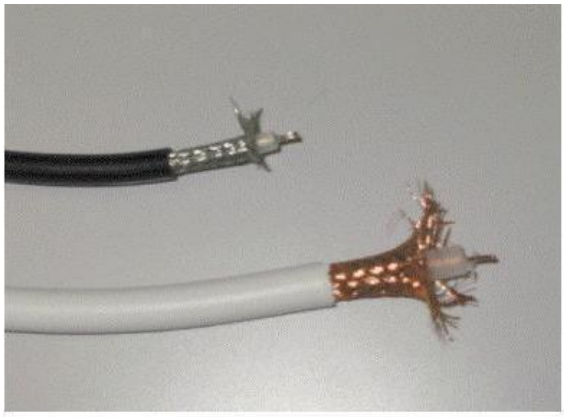
- up to 10,800 voice conversations

Much less susceptible to interference than twisted pair

Coax Disadvantages

High attenuation rate makes it expensive over long distance Bulky

Figure 3.2 Coaxial Cable



Applications

- It is used in cable TV networks
- It is used in traditional Ethernet LANs.

Fiber Optic Cable

Relatively new transmission medium used by telephone companies in place of long- distance trunk lines also used by private companies in implementing local data communications networks

Require a light source with injection laser diode (ILD) or light-emitting diodes (LED) Optical fiber consists of a glass core, surrounded by a glass cladding with slightly lower refractive index.

In most networks fiber-optic cable is used as the high-speed backbone, and twisted wire and coaxial cable are used to connect the backbone to individual devices.

Fiber Optic Advantages

- Greater capacity (bandwidth of up to 2 Gbps). Smaller size and lighter weight.
- Lower attenuation.
- Immunity to environmental interference.
- Highly secure due to tap difficulty and lack of signal radiation.

Fiber Optic Disadvantages

- expensive over short distance
- requires highly skilled installers
- adding additional nodes is difficult

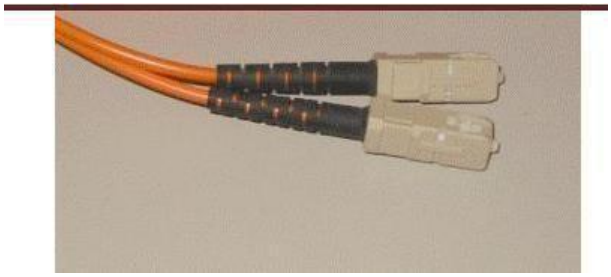


Figure 3.3 Fiber Optic Cable

Applications

- The fiber optic cable is often found in backbone networks because its bandwidth is cost effective.
- Used in TV companies.
- LAN such as 100 Base-FX Network

Wireless (Unguided Media) Transmission

Transmission and reception are achieved by means of an antenna directional transmitting antenna put$ out focused beam transmitter and receiver must be aligned omnidirectional signal spreads out in all directions can be received by many antennas

Wireless Examples

Terrestrial microwave, satellite microwave, broadcast radio ,infrared

Microwaves

Electromagnetic waves having frequency between 1 and 300 GHz are called as Micro waves.

- Micro waves are unidirectional.

- Microwave propagation is line of sight.
- Very high frequency Micro waves cannot penetrate walls.
- The microwave band is relatively wide, almost 299 GHz

Terrestrial Microwave

- Used for long-distance telephone service.
- Uses radio frequency spectrum, from 2 to 40 Ghz.
- Parabolic dish transmitter, mounted high.
- Used by common carriers as well as private networks.
- Requires unobstructed line of sight between source and receiver.
- Curvature of the earth requires stations (repeaters) ~30 miles apart.

Satellite Microwave

A microwave relay station in space can relay signals over long distances geostationary satellites

- remain above the equator at a height of 22,300 miles (geosynchronous orbit)
- travel around the earth in exactly the time the earth takes to rotate

Applications

- They are used in Cellular phones.
- They are used in satellite networks.

- They are used in wireless LANs.

Radio Waves Application

1. The omnidirectional characteristics of Radio waves make them useful for multicasting, in which there is one sender but many receivers.
2. AM and FM Radio, television, maritime radio, cordless phone, and paging are examples of multicasting.

PSTN

The public switched telephone network (PSTN) is the aggregate of the world's circuit- switched telephone networks that are operated by national, regional, or local telephony operators, providing infrastructure and services for public telecommunication. The PSTN consists of telephone lines, fiber optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all interconnected by switching centers, thus allowing most telephones to communicate with each other. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital in its core network and includes mobile and other networks, as well as fixed telephones. The technical operation of the PSTN adheres to the standards created by the ITU-T. These standards allow different networks in different countries to interconnect seamlessly. The E.163 and E.164 standards provide a single global address space for telephone numbers. The combination of the interconnected networks and the single numbering plan allow telephones around the world to dial each other.

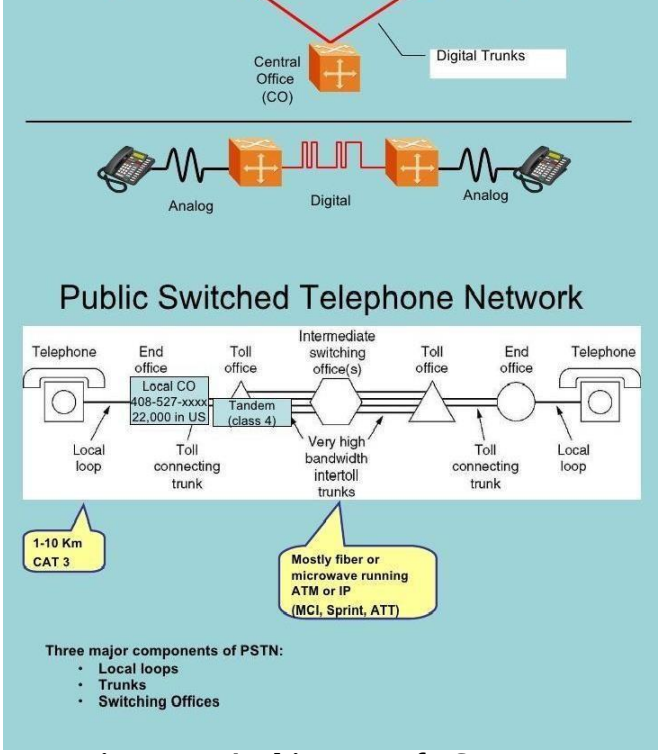


Figure 3.4 Architecture of PSTN

Data-Link Protocols

Data-Link Protocol Functions

Line discipline – coordinates hop-to-hop data delivery where a hop is a computer, a network controller, or some type of network-connecting device, such as router.It determines which device is transmitting and which is receiving at any point in time

Flow control – the rate at which data is transported over a link.

- It provides an acknowledgement mechanism that data is received at the destination.
- It regulate flow of data from sender to receiver

Error control – detects and corrects transmission errors

Framing – recognizing beginning and end of frames (blocks, packets).Communications requires at least two devices, one to send and one to receive. If both devices are ready to send some information and put their signals on the link then the two signals collides each other and became nothing. To avoid such a situation the data link layer use a mechanism called line discipline.

Line discipline coordinates the link system. It determines which device can send and when it can send. It answers the question, who should send now? Line discipline can serve in two ways:

1. enquiry / acknowledgement (ENQ / ACK)
2. poll / select (POLL / SELECT)

ENQ / ACK:

This method is used in peer to peer communications. That is where there is a dedicated link between two devices. The initiator first transmits a frame called an enquiry (ENQ) asking I the receiver is available to receive data. The receiver must answer either with an acknowledgement (ACK) frame if it ready to accept or with a negative acknowledgement (NAK) frame if it is not ready. If the response is positive, the initiator is free to send its data. Otherwise it waits, and try again. Once all its data have been transmitted, the sending system finishes with an end of transmission (EOT) frame.

Figure 3.5 Line discipline

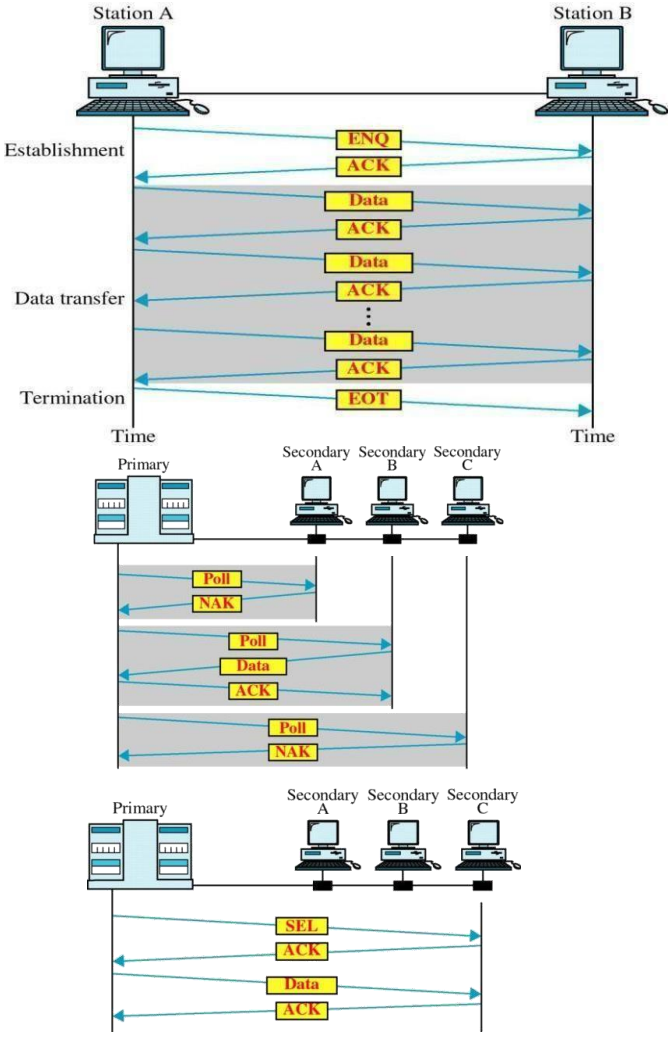


Figure 3.6 Poll/Select

Flow control

Stop-and-Wait flow control

- Source transmits frame
- Destination receives frame and replies with acknowledgement
- Source waits for ACK before sending next frame
- Destination can stop flow by not send ACK

- Works well for a few largeframes

It refers to a set of procedures used to restrict the amount of data flow between sending and receiving stations. It tells the sender how much data it can transmit before it must wait for an acknowledgement from the receiver. There are two methods are used. They are,

1. stop and wait
2. sliding window

STOP AND WAIT

In this method the sender waits or acknowledgment after every frame it sends. Only after an acknowledgment has been received, then the sender sends the next frame.

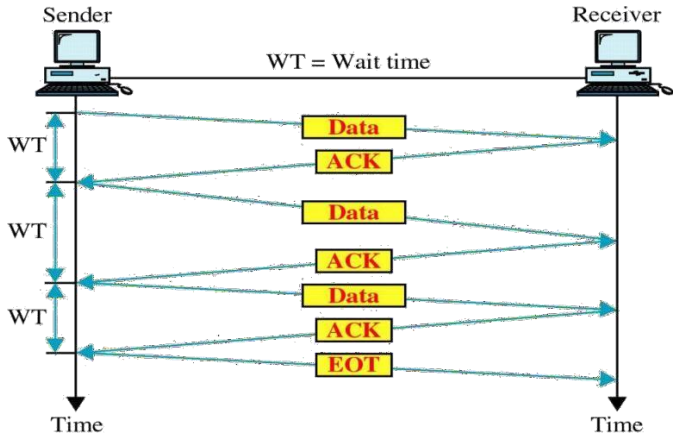


Figure 3.7 Stop-and-Wait flow control

The advantage is simplicity. The disadvantage is inefficiency

SLIDING WINDOW

In this method, the sender can transmit several frames before needing an acknowledgment. The receiver acknowledges only some of the frames, using a single ACK to confirm the receipt of multiple data frames.

The sliding window refers to imaginary boxes at both the sender and receiver. This window provides the upper limit on the number of frames that can be transmitted before requiring an acknowledgement. To identify each frame the sliding window scheme introduces the sequence number.

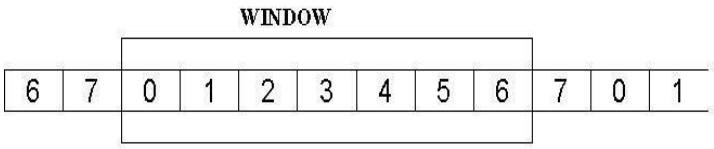


Figure 3.8 Sliding window flow control

The frames are numbered as 0 to n-1. And the size of the window is n-1. Here the size of the window is 7 and the frames are numbered as 0,1,2,3,4,5,6,7.

SENDER WINDOW

At the beginning the sender's window contains n-1 frames. As frames are sent out the left boundary of the window moves inward, shrinking the size of the window. Once an ACK receives the window expands at the right side boundary to allow in a number of new frames equal to number of frames acknowledged by that ACK.

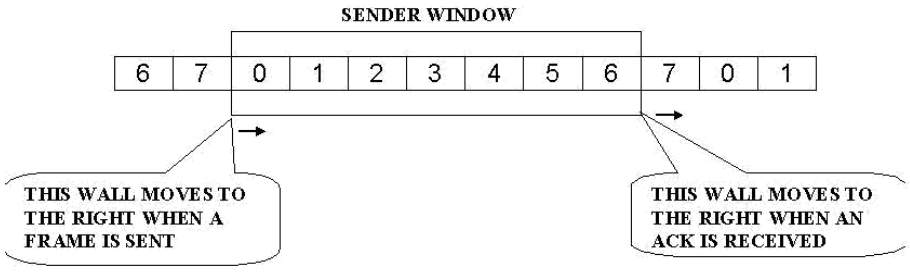
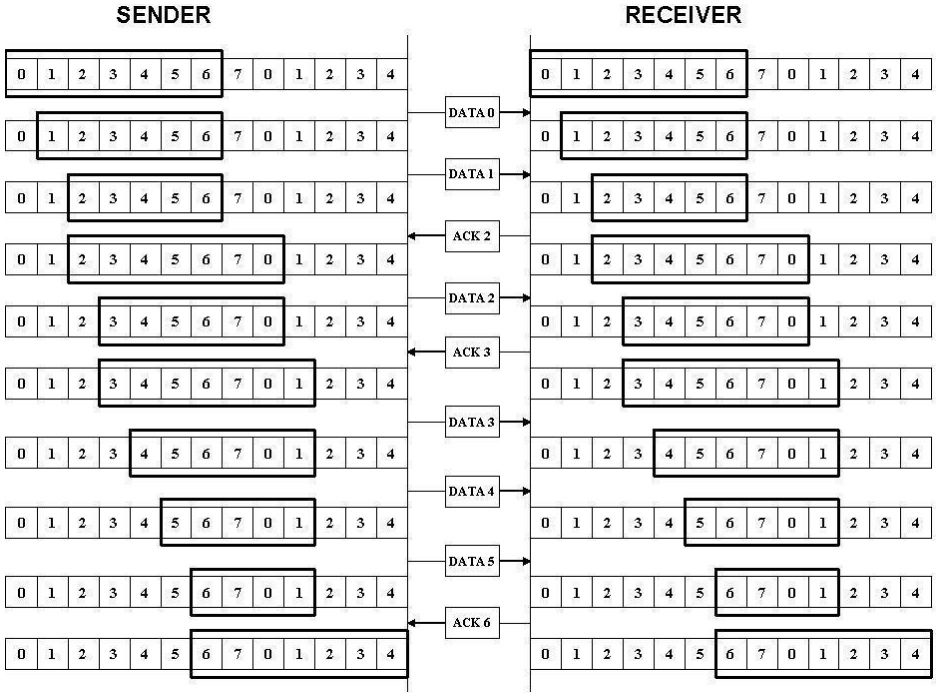


Figure 3.9 Sender Window

Figure 3.10 Sender and Receiver Window



ERROR CONTROL

Detection and correction of errors Lost frames Damaged frames Techniques for error control (Automatic repeat request)

--Error detection

--Positive acknowledgment

--Retransmission after timeout

--Negative acknowledgement and retransmission

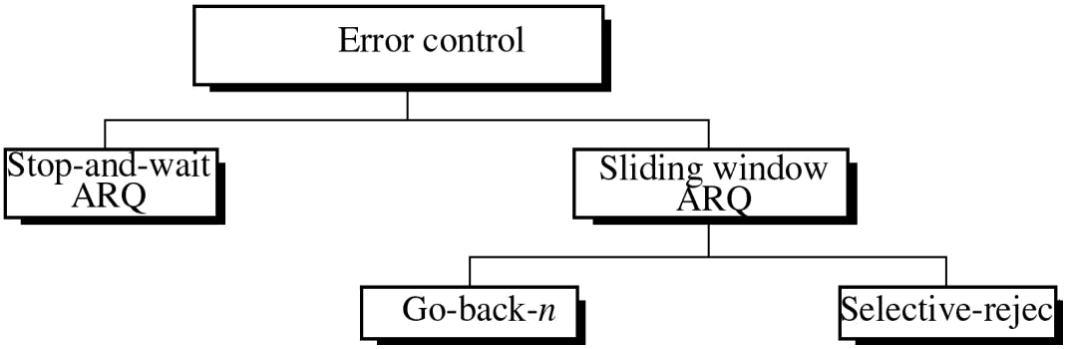


Figure 3.11 Classification of Error Control

Error control is implemented in such a way that every time an error is detected, a negative acknowledgement is returned and the specified frame is retransmitted. This process is called **automatic repeat request (ARQ).**The error control is implemented with the flow control mechanism. So there are two types in error control. They are,

1. stop and wait ARQ
2. sliding window ARQ

STOP AND WAIT ARQ:

It is a form of stop and wait flow control, extended to include retransmission of data in case of lost or damaged frames.

DAMAGED FRAME:

When a frame is discovered by the receiver to contain an error, it returns a NAK frame and the sender retransmits the last frame.

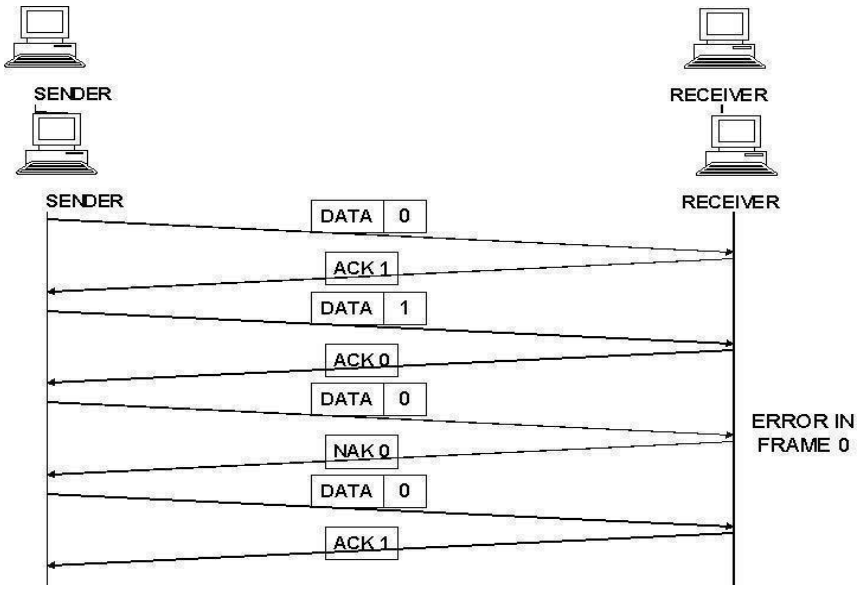


Figure 3.12 Damaged Frame in stop and wait ARQ

LOST DATA FRAME:

The sender is equipped with a timer that starts every time a data frame is transmitted. If the frame lost in transmission the receiver can never acknowledge it. The sending device waits for an ACK or NAK frame until its timer goes off, then it tries again. It retransmits the last data frame.

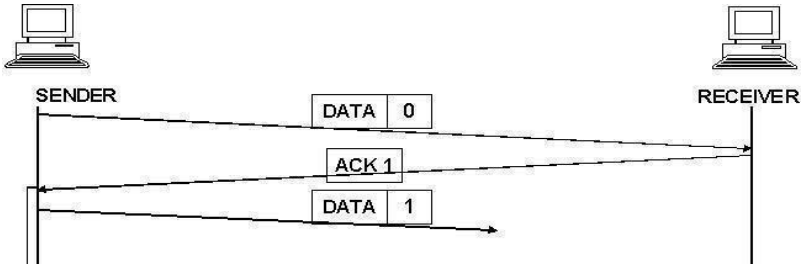


Figure 3.13 Lost Data Frame in stop and wait ARQ

LOST ACKNOWLEDGEMENT:

The data frame was received by the receiver but the acknowledgement was lost in transmission. The sender waits until the timer goes off, then it retransmits the data frame. The receiver gets a duplicated copy of the data frame. So it knows the acknowledgement was lost so it discards the second copy.

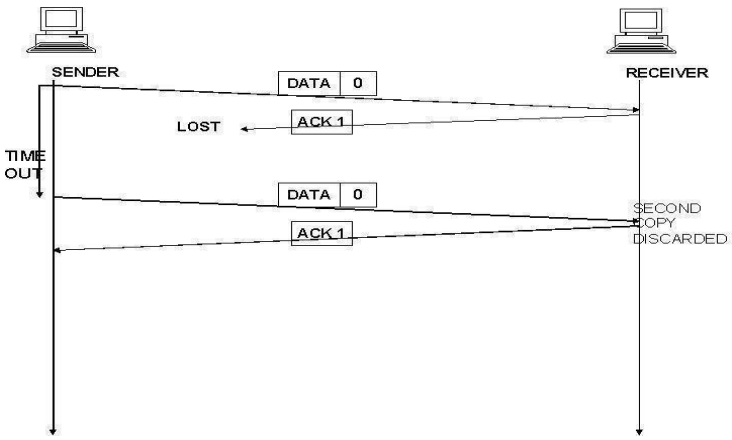


Figure 3.14 Lost Acknowledgement in stop and wait ARQ

SLIDING WINDOW ARQ

It is used to send multiple frames per time. The number of frame is according to the window size. The sliding window is an imaginary box which is reside on both sender and receiver side. It has two types. They are,

1. go-back-n ARQ
2. selective reject ARQ

GO-BACK-N ARQ:

In this method, if one frame is lost or damaged, all frames sent since the last frame acknowledged or retransmitted.

DAMAGED FRAME:

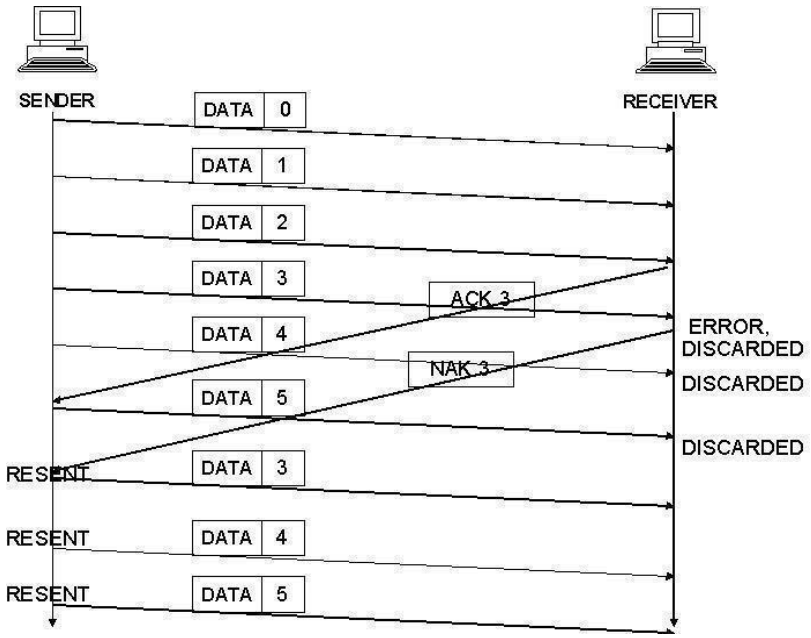


Figure 3.15 Damaged Frame in go back N ARQ

LOST FRAME:

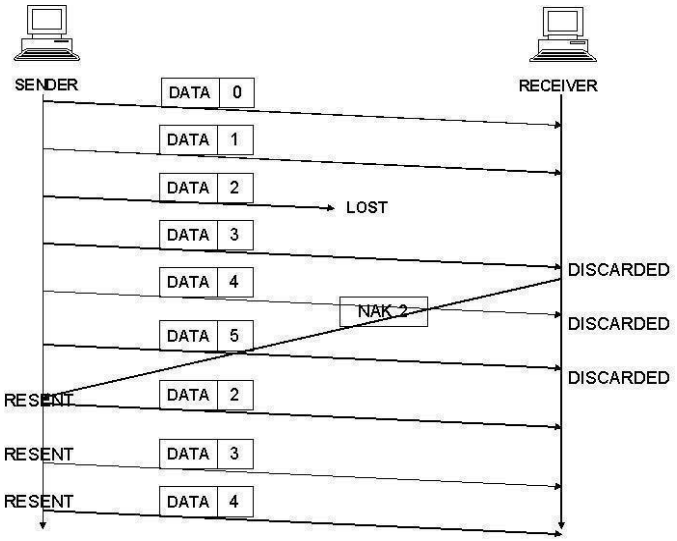


Figure 3.15 Lost Frame in go back N ARQ

LOST ACK:

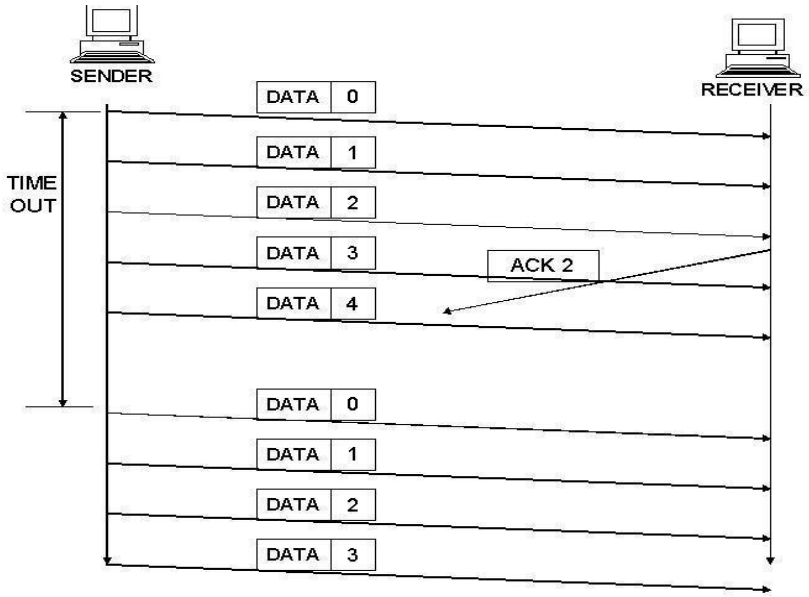


Figure 3.16 Lost Acknowledgement in go back N ARQ

SELECTIVE REPEAT ARQ

Selective repeat ARQ re transmits only the damaged or lost frames instead of sending multiple frames. The selective transmission increases the efficiency of transmission and is more suitable for noisy link. The receiver should have sorting mechanism.

DAMAGED FRAME

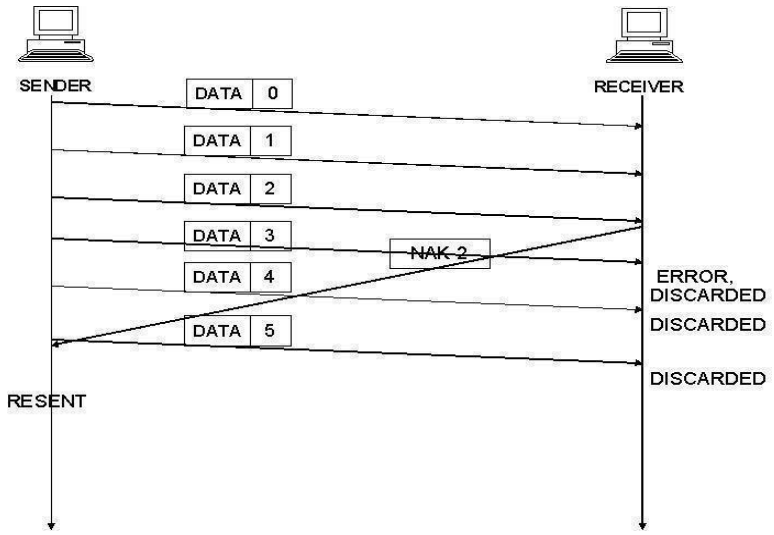


Figure 3.17 Damaged Frame in Selective Repeat ARQ

LOST FRAME

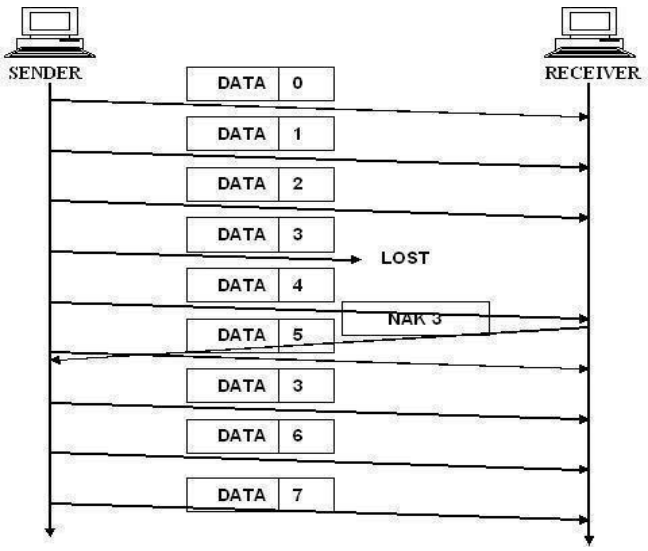
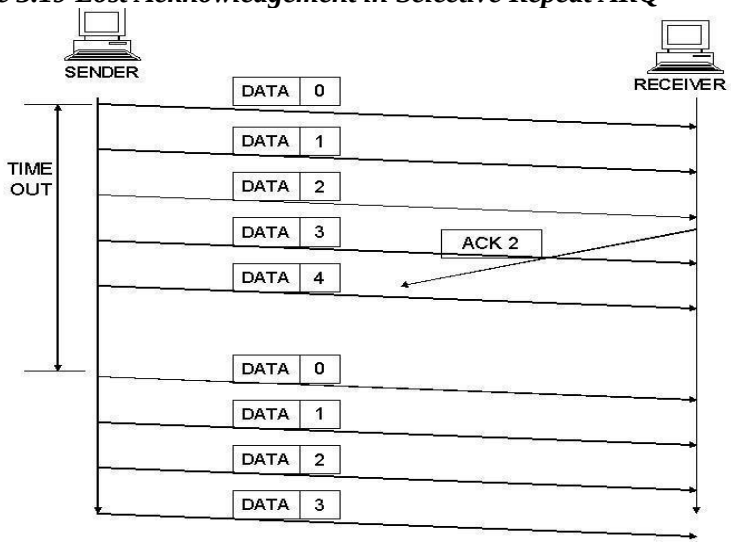


Figure 3.18 Lost Frame in Selective Repeat ARQ

LOST ACK

Figure 3.19 Lost Acknowledgement in Selective Repeat ARQ



HDLC

HDLC is a bit-oriented protocol. It was developed by the International Organization for Standardization (ISO). It falls under the ISO standards ISO 3309 and ISO 4335. It specifies a packitization standard for serial links. It has found itself being used throughout the world. It has been so widely implemented because it supports both half-duplex and full-duplex communication lines, point-to-point (peer to peer) and multi-point networks, and switched or non-switched channels. HDLC supports several modes of operation, including a simple sliding- window mode for reliable delivery. Since Internet provides retransmission at higher levels (i.e., TCP), most Internet applications use HDLC's unreliable delivery mode, Unnumbered Information.

Other benefits of HDLC are that the control information is always in the same position, and specific bit patterns used for control differ dramatically from those in representing data, which reduces the chance of errors. It has also led to many subsets. Two subsets widely in use are Synchronous Data Link Control (SDLC) and Link Access Procedure-Balanced (LAP-B).

HDLC Stations and Configurations

HDLC specifies the following three types of stations for data link control:

- Primary Station
- Secondary Station
- Combined Station

Primary Station

Within a network using HDLC as its data link protocol, if a configuration is used in which there is a primary station, it is used as the controlling station on the link. It has the responsibility of controlling all other stations on the link (usually secondary stations). A primary issues *commands* and secondary issues *responses*. Despite this important aspect of being on the link, the primary station is also responsible for the organization of data flow on the link. It also takes care of error recovery at the data link level (layer 2 of the OSI model).

Secondary Station

If the data link protocol being used is HDLC, and a primary station is present, a secondary station must also be present on the data link. The secondary station is under the control of the primary

station. It has no ability, or direct responsibility for controlling the link. It is only activated when requested by the primary station. It only responds to the primary station. The secondary station's frames are called responses. It can only send response frames when requested by the primary station. A primary station maintains a separate logical link with each secondary station.

Combined Station

A combined station is a combination of a primary and secondary station. On the link, all combined stations are able to send and receive commands and responses without any permission from any other stations on the link. Each combined station is in full control of itself, and does not rely on any other stations on the link. No other stations can control any combined station. May issue both commands and responses. HDLC also defines three types of configurations for the three types of stations. The word configuration refers to the relationship between the hardware devices on a link. Following are the three configurations defined by HDLC:

- Unbalanced Configuration
- Balanced Configuration
- Symmetrical Configuration

Unbalanced Configuration

The unbalanced configuration in an HDLC link consists of a primary station and one or more secondary stations. The unbalanced condition arises because one station controls the other stations. In an unbalanced configuration, any of the following can be used:

- Full-Duplex or Half-Duplex operation
- Point to Point or Multi-point networks

An example of an unbalanced configuration can be found below

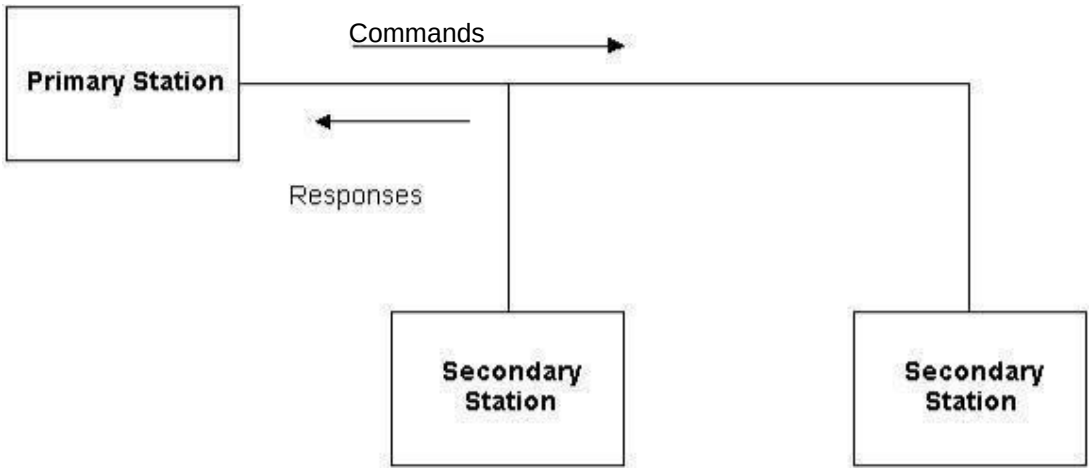


Figure 3.20 Unbalanced configuration

Balanced Configuration

The balanced configuration in an HDLC link consists of two or more combined stations.Each of the stations has equal and complimentary responsibility compared to each other.Balanced configurations can use only the following:

- Full - Duplex or Half - Duplex operation
- Point to Point networks

An example of a balanced configuration can be found below.

Commands/ responses

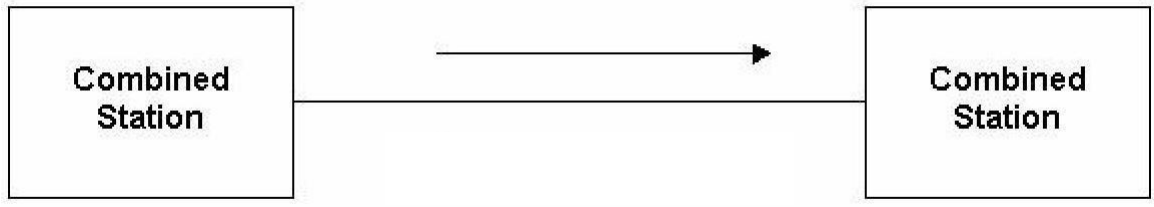


Figure 3.21 Balanced configuration

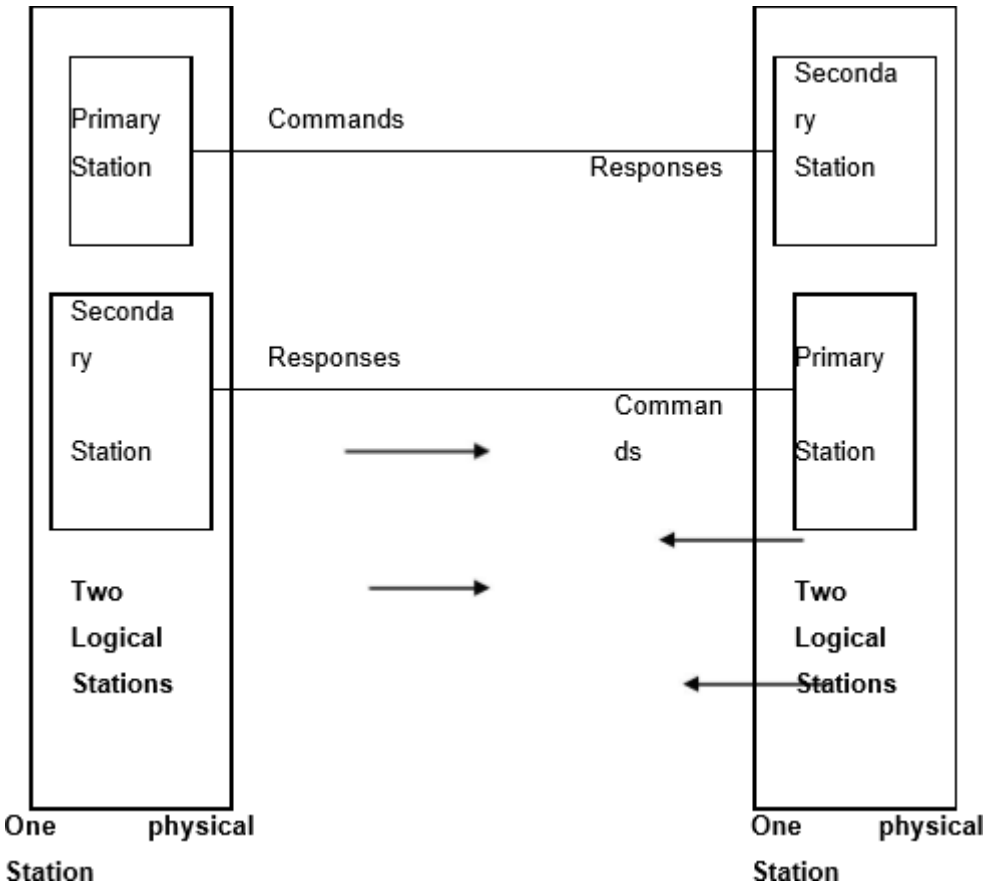


Figure 3.22 Symmetric configuration

Symmetrical Configuration

This third type of configuration is not widely in use today. It consists of two independent point-to- point, unbalanced station configurations as shown in Figure. In this configuration, each station has a primary and secondary status. Each station is logically considered as two stations **HDLC Operational Modes**

A mode in HDLC is the relationship between two devices involved in an exchange; the mode describes who controls the link. Exchanges over unbalanced configurations are always conducted in normal response mode. Exchanges over symmetric or balanced configurations can be set to specific mode using a frame design to deliver the command. HDLC offers three different modes of operation. These three modes of operations are:

- Normal Response Mode (NRM)
- Asynchronous Response Mode (ARM)
- Asynchronous Balanced Mode (ABM)

Normal Response Mode

This is the mode in which the primary station initiates transfers to the secondary station. The secondary station can only transmit a response when, and only when, it is instructed to do so by the primary station. In other words, the secondary station must receive explicit permission from the primary station to transfer a response. After receiving permission from the primary station, the secondary station initiates its transmission. This transmission from the secondary station to the primary station may be much more than just an acknowledgment of a frame. It may in fact be more than one information frame. Once the last frame is transmitted by the secondary station, it must wait once again from explicit permission to transfer anything, from the primary station. Normal Response Mode is only used within an unbalanced configuration.

Asynchronous Response Mode

In this mode, the primary station doesn't initiate transfers to the secondary station. In fact, the secondary station does not have to wait to receive explicit permission from the primary station to transfer any frames. The frames may be more than just acknowledgment frames. They may contain data, or control information regarding the status of the secondary station. This mode can reduce overhead on the link, as no frames need to be transferred in order to give the secondary station permission to initiate a transfer. However, some limitations do exist. Due to the fact that this mode is asynchronous, the secondary station must wait until it detects and idle channel before it can transfer any frames. This is when the ARM link is operating at half-duplex. If the ARM link is operating at full duplex, the secondary station can transmit at any time. In this mode, the primary station still retains responsibility for error recovery, link setup, and link disconnection.

Synchronous Balanced Mode

This mode is used in case of combined stations. There is no need for permission on the part of any station in this mode. This is because combined stations do not require any sort of instructions to perform any task on the link.Normal Response Mode is used most frequently in multi- pointlines, where the primary station controls the link. Asynchronous Response Mode is better for point-to- point links, as it reduces overhead. Asynchronous Balanced Mode is not used widely today. The "asynchronous" in both ARM and ABM does not refer to the format of the data on the link. It refers to the fact that any given station can transfer frames without explicit permission or instruction from any other station.

HDLC Non-Operational Modes

HDLC also defines three non-operational modes. These three non-operational modes are:

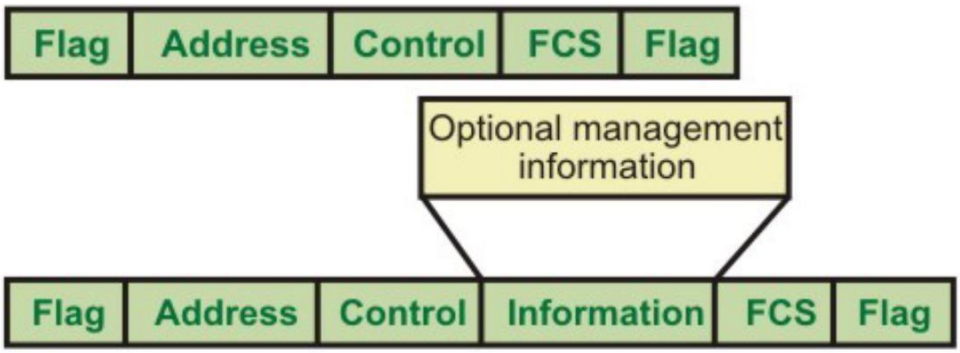
- Normal Disconnected Mode (NDM)
- Asynchronous Disconnected Mode (ADM)
- Initialization Mode (IM)

The two disconnected modes (NDM and ADM) differ from the operational modes in that the secondary station is logically disconnected from the link (note the secondary station is not physically disconnected from the link). The IM mode is different from the operations modes in that the secondary station's data link control program is in need of regeneration or it is in need of an exchange of parameters to be used in an operational mode.

HDLC Frame Structure

There are three different types of frames as shown in Fig. and the size of different fields are shown Table

I-Frame



U-Frame

Figure 3.23 Different types of Frames in HDLC

Table 3.1 Size of different fields	
Field Name	**Size(in bits)**
[redacted]	[redacted]
Flag Field(F)	8 bits
Address Field(A)	8 bits
Control Field(C)	8 or 16 bits
Information Field(I) OR Data	Variable; Not used in some frames
Frame Check Sequence(FCS)	16 or 32 bits
Closing Flag Field(F)	8 bits

The Flag field

Every frame on the link must begin and end with a flag sequence field (F). Stations attached to the data link must continually listen for a flag sequence. The flag sequence is an octet looking like 01111110. Flags are continuously transmitted on the link between frames to keep the link active. Two other bit sequences are used in HDLC as signals for the stations on the link. These two bit sequences are:

- Seven 1's, but less than 15 signal an abort signal. The stations on the link know there is a problem on the link.
- 15 or more 1's indicate that the channel is in an idle state.

The time between the transmissions of actual frames is called the **interframe time fill**. The interframe time fill is accomplished by transmitting continuous flags between frames. The flags may be in 8 bit multiples.

HDLC is a code-transparent protocol. It does not rely on a specific code for interpretation of line control. This means that if a bit at position N in an octet has a specific meaning, regardless of the other bits in the same octet. If an octet has a bit sequence of 01111110, but is not a flag field, HLDC uses a technique called bit-stuffing to differentiate this bit sequence from a flag field as we have discussed in the previous lesson.

At the receiving end, the receiving station inspects the incoming frame. If it detects 5 consecutive 1's it looks at the next bit. If it is a 0, it pulls it out. If it is a 1, it looks at the 8th bit. If the 8th bit is a 0, it knows an abort or idle signal has been sent. It then proceeds to inspect the following bits to

determine appropriate action. This is the manner in which HDLC achieves code- transparency. HDLC is not concerned with any specific bit code inside the data stream. It is only concerned with keeping flags unique.

The Address field

The address field (A) identifies the primary or secondary stations involvement in the frame transmission or reception. Each station on the link has a unique address. In an unbalanced configuration, the A field in both commands and responses refer to the secondary station. In a balanced configuration, the command frame contains the destination station address and the response frame has the sending station's address.

The Control field

HDLC uses the control field (C) to determine how to control the communications process. This field contains the commands, responses and sequences numbers used to maintain the data flow accountability of the link, defines the functions of the frame and initiates the logic to control the movement of traffic between sending and receiving stations. There three control field formats:

- **Information Transfer Format**: The frame is used to transmit end-user data between two devices.
- **Supervisory Format**: The control field performs control functions such as acknowledgment of frames, requests for re-transmission, and requests for temporary suspension of frames being transmitted. Its use depends on the operational mode being used.
- **Unnumbered Format**: This control field format is also used for control purposes. It is used to perform link initialization, link disconnection and other link control functions.

The Poll/Final Bit (P/F)

The 5^{th} bit position in the control field is called the **poll/final bit, or P/F bit**. It can only be recognized when it is set to 1. If it is set to 0, it is ignored. The poll/final bit is used to provide dialogue between the primary station and secondary station. The primary station uses P=1 to acquire a status response from the secondary station. The P bit signifies a poll. The secondary station responds to the P bit by transmitting a data or status frame to the primary station with the P/F bit set to F=1. The F bit can also be used to signal the end of a transmission from the secondary station under Normal Response Mode.

The Information field or Data field

This field is not always present in a HDLC frame. It is only present when the Information Transfer Format is being used in the control field. The information field contains the actually data the sender is transmitting to the receiver in an I-Frame and network management information in U- Frame.

The Frame check Sequence field

This field contains a 16-bit, or 32-bit cyclic redundancy check bits. It is used for error detection

as discussed in the previous lesson.

Error Detecting Codes

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. Vertical Redundancy Check(VRC)
2. Longitudinal Redundancy Check(VRC)
3. Checksum
4. Cyclic redundancy check

1. Vertical Redundancy Check (VRC)

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.

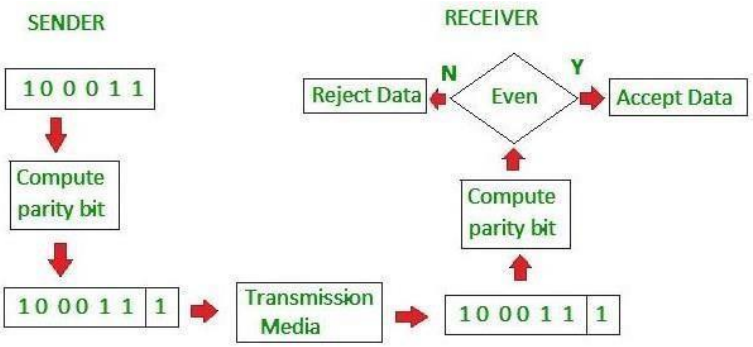


Figure 3.24 Vertical Redundancy Check

2. Longitudinal Redundancy Check (VRC)

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.

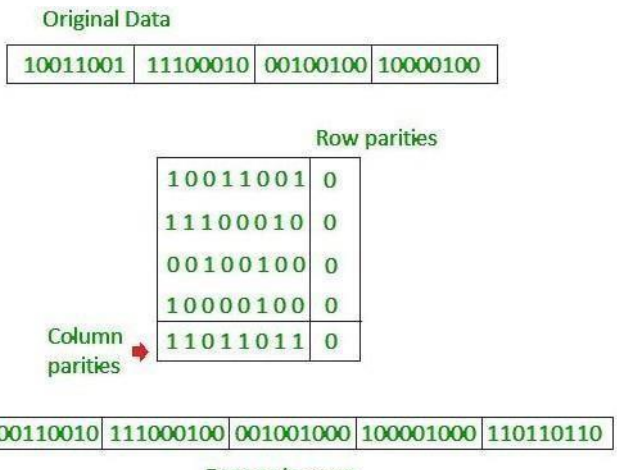


Figure 3.25 Longitudinal Redundancy Check

3. 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.

4. 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.

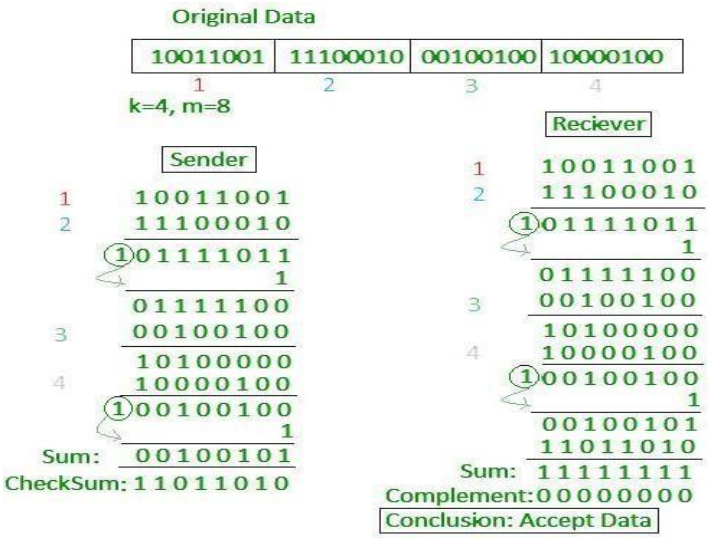


Figure 3.26 Checksum

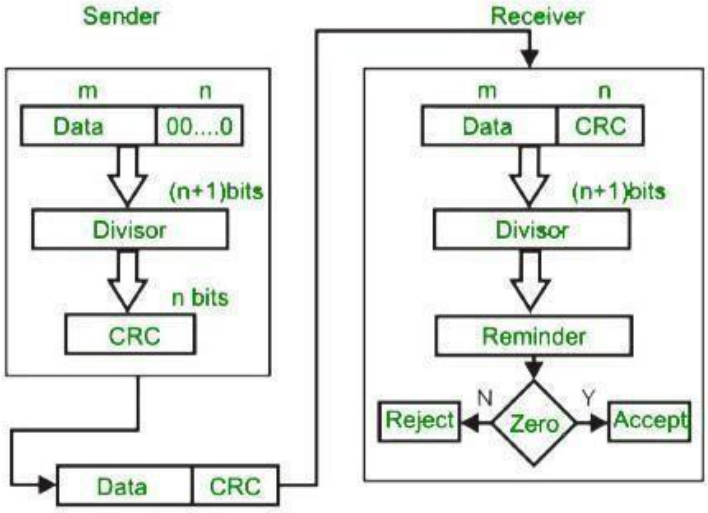


Figure 3.27 Algorithm of Cyclic redundancy check

Example :

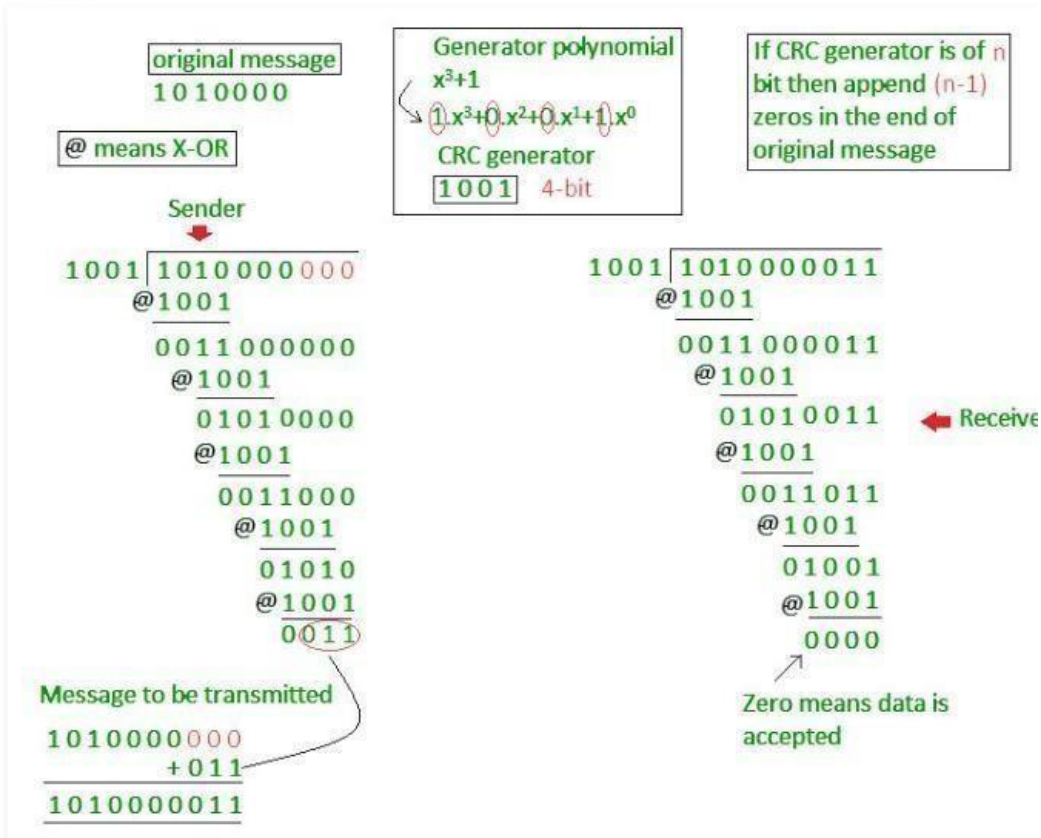


Figure 3.27 Example of Cyclic redundancy check

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SCHOOL OF ELECTRICAL AND ELECTRONICS ENGINEERING

DEPARTMENT OF ELECTRONICS AND COMMUNICATON ENGINEERING

UNIT IV COMPUTER NETWORKS
SECA1604

UNIT 4 - MAC SUB LAYER AND NETWORK LAYER

MAC sub layer for Standard Ethernet, Fast Ethernet, Wireless LAN and broadband wireless. Design issues of network layer - Routing algorithm - shortest path routing - Distance vector routing - Broadcast routing –Inter domain routing, Congestion control algorithm - Congestion control in virtual circuit and datagram switches - The network layer in the internet - The IP protocol-IP Addresses - IPv6, ARP,DHCP,ICMP, Classless Addressing, Network Address Translation.

MAC sub layer for Standard

Ethernet operates in the data link layer and the physical layer. It is a family of networking technologies that are defined in the IEEE 802.2 and 802.3 standards. Ethernet supports data bandwidths of

10 Mb/s
100 Mb/s
1000 Mb/s (1 Gb/s)
10,000 Mb/s (10 Gb/s)
40,000 Mb/s (40 Gb/s)
100,000 Mb/s (100 Gb/s)

Ethernet standards define both the Layer 2 protocols and the Layer 1 technologies. For the Layer 2 protocols, as with all 802 IEEE standards, Ethernet relies on the two separate sublayers of the data link layer to operate, the Logical Link Control (LLC) and the MAC sublayers.

LLC sublayer

The Ethernet LLC sublayer handles the communication between the upper layers and the lower layers. This is typically between the networking software and the device hardware. The LLC sublayertakes the network protocol data, which is typically an IPv4 packet, and adds control information to help deliver the packet to the destination node. The LLC is used to communicate

with the upper layers of the application, and transition the packet to the lower layers for delivery. LLC is implemented in software, and its implementation is independent of the hardware. In a computer, the LLC can be considered the driver software for the NIC. The NIC driver is a program that interacts directly with the hardware on the NIC to pass the data between the MAC sublayer and the physical media.

MAC sublayer

MAC constitutes the lower sublayer of the data link layer. MAC is implemented by hardware, typically in the computer NIC. The specifics are specified in the IEEE 802.3 standards. Figure 4.1 lists common IEEE Ethernet standards.

IEEE 802.3 and Ethernet

Very popular LAN standard.
Ethernet and IEEE 802.3 are distinct standards but as they are very similar to one another these words are used interchangeably.
A standard for a 1-persistent CSMA/CD LAN.
It covers the physical layer and MAC sublayer protocol.

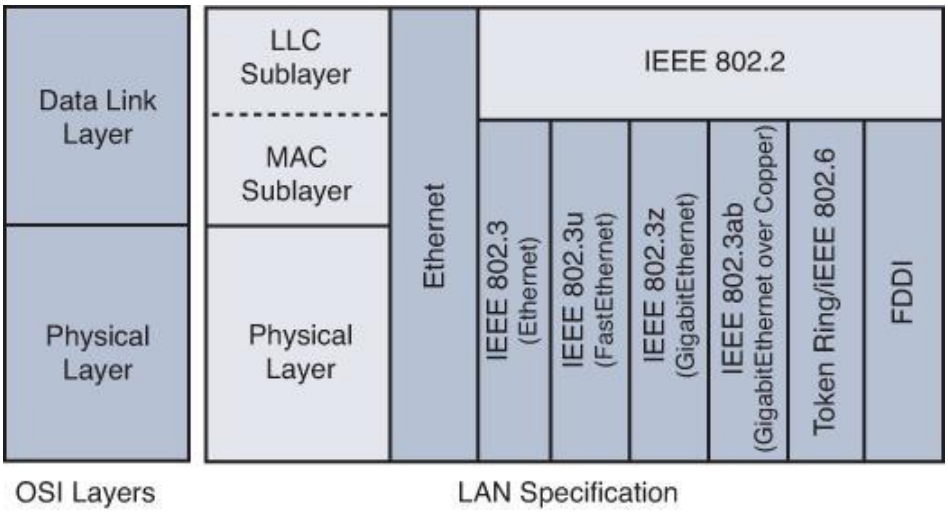


Figure 4.1 Common IEEE Ethernet Standards

Fast Ethernet

Fast Ethernet is a collective term for a number of **Ethernet** standards that carry traffic at the nominal rate of 100 Mbit/s (the earlier Ethernet speed was 10 Mbit/s). Of the Fast Ethernet standards, 100BASE-TX is by far the most common.
Fast Ethernet was introduced in 1995 as the IEEE 802.3u standard and remained the fastest version of Ethernet for three years before the introduction of **Gigabit Ethernet**.

wireless local area network (**WLAN**)

A wireless local area network (WLAN) is a **wireless computer network** that links two or more devices using **wireless communication** within a limited area such as a home, school, computer laboratory, or officebuilding. This gives users the ability to move around within a local coverage area and yet still be connected to the network. Through a **gateway,** a WLAN can also provide a connection to the wider **Internet**.
Most modern WLANs are based on **IEEE 802.11** standards and are marketed under the **Wi-Fi** brand name.

4.3.1 Wireless broadband

It is technology that provides high-speed **wireless Internet access** or **computer networking** access over a wide area. broadband means "having instantaneous **bandwidths** greater than 1 **MHz** and supporting **data rates** greater than about 1.5 **Mbit**/s.

The network layer design issues :

- Store and formed packet switching.
- Service provided to the transport layer.
- Implementation of connectionless service.
- Implementation of connection-oriented source.
- Comparison of virtual circuit and datagram submits.

- Store and formed packet switching :
- Store and forward operation : -
- Host transmits packet to router across LAN or oval point to point link.
- Packet is stored on router until fully arrived and processed.
- Packet is forward to next router.
- Service provide to transport layer

The network layer services have been designed with the goals : -

- The advice should independent of router telenet
- The transport layer should be shielded from the number type and topology of the router present.
- The network addresses maid available to transport

Implementation of connectionless service :

Connectionless service is offered packets are injected into the subnet individually and routed independently of each other. Each packet is transmitted independently.
Connectionless service used in network layer ID and transport layer.
Packet are frequently called datagram connectionless service is largely for data communication theinternet.

Implementation of connection-oriented service : -

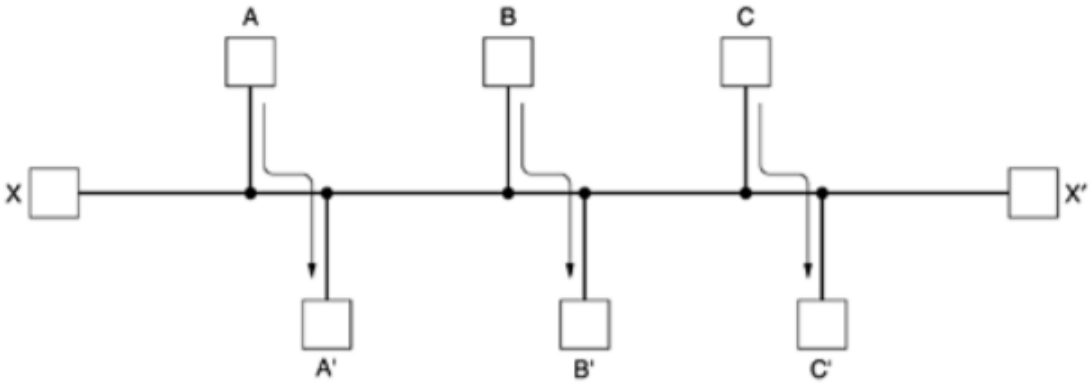
Connection-oriented service is used a path from the source router to the destination router must beestablished before any data packet can be sent.
Connection oriented service also called virtual circuit service. This service used network layer for ATM. It also used in transport layer for TCP.
A connection must be established before any can be sent packets order preserved logical connection isalso established here.

Routing Algorithm

A **Routing Algorithm** is a method for determining the routing of packets in a node. For each node of a network, the algorithm determines a routing table, which in each destination, matches an output line. The algorithm should lead to a consistent routing, that is to say without loop.
The routing algorithm is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on.

PROPERTIES OF ROUTING ALGORITHM:
Correctness, simplicity, robustness, stability, fairness, and optimality

FAIRNESS AND OPTIMALITY.



Fairness and optimality may sound obvious, but as it turns out, they are often contradictory goals. There is enough traffic between A and A', between B and B', and between C and C' to saturate the horizontal links. To maximize the total flow, the X to X' traffic should be shut off altogether. Unfortunately, X and X' may not see it that way. Evidently, some compromise between global efficiency and fairness to individual connections is needed.

CATEGORY OF ALGORITHM

Routing algorithms can be grouped into two major classes: **nonadaptive and adaptive.**

Nonadaptive algorithms do not base their routing decisions on measurements or estimates of the current traffic and topology. Instead, the choice of the route to use to get from I to J is computed in advance, off-line, and downloaded to the routers when the network is booted.

This procedure is sometimes called **Static routing.**

Adaptive algorithms, in contrast, change their routing decisions to reflect changes in the topology, and usually the traffic as well

This procedure is sometimes called **dynamic routing**

THE OPTIMALITY PRINCIPLE

If router J is on the optimal path from router I to router K, then the optimal path from J to K also falls along the same route.

The set of optimal routes from all sources to a given destination form a tree rooted at the destination. Such a tree is called a sink tree.

As a direct consequence of the optimality principle, we can see that the set of optimal routes from all sources to a given destination form a tree rooted at the destination.

Such a tree is called a sink tree where the distance metric is the number of hops. Note that a sink tree is not necessarily unique; other trees with the same path lengths may exist.

The goal of all routing algorithms is to discover and use the sink trees for all routers.

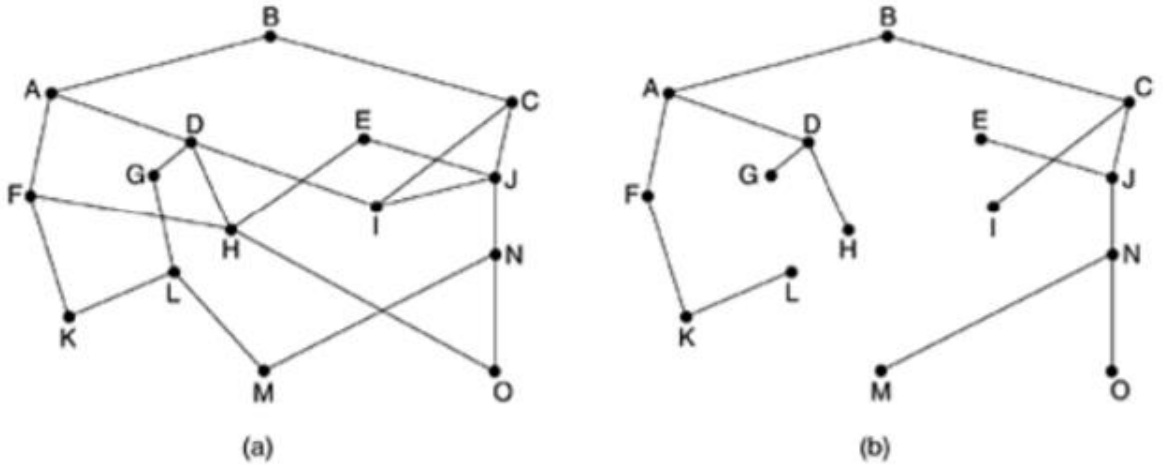


Fig 4.2 (a) A Sub Net, (b) A Sink tree for Router B

Shortest path routing

A technique to study routing algorithms: The idea is to build a graph of the subnet, with each node of the graph representing a router and each arc of the graph representing a communication line (often called a link).

To choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph.

One way of measuring path length is the number of hops. Another metric is the geographic distance in kilometers. Many other metrics are also possible. For example, each arc could be labeled with the mean queuing and transmission delay for some standard test packet as determined by hourly test runs.

In the general case, the labels on the arcs could be computed as a function of the distance, bandwidth, average traffic, communication cost, mean queue length, measured delay, and other factors. By changing the weighting function, the algorithm would then compute the "shortest" path measured according to any one of a number of criteria or to a combination of criteria.

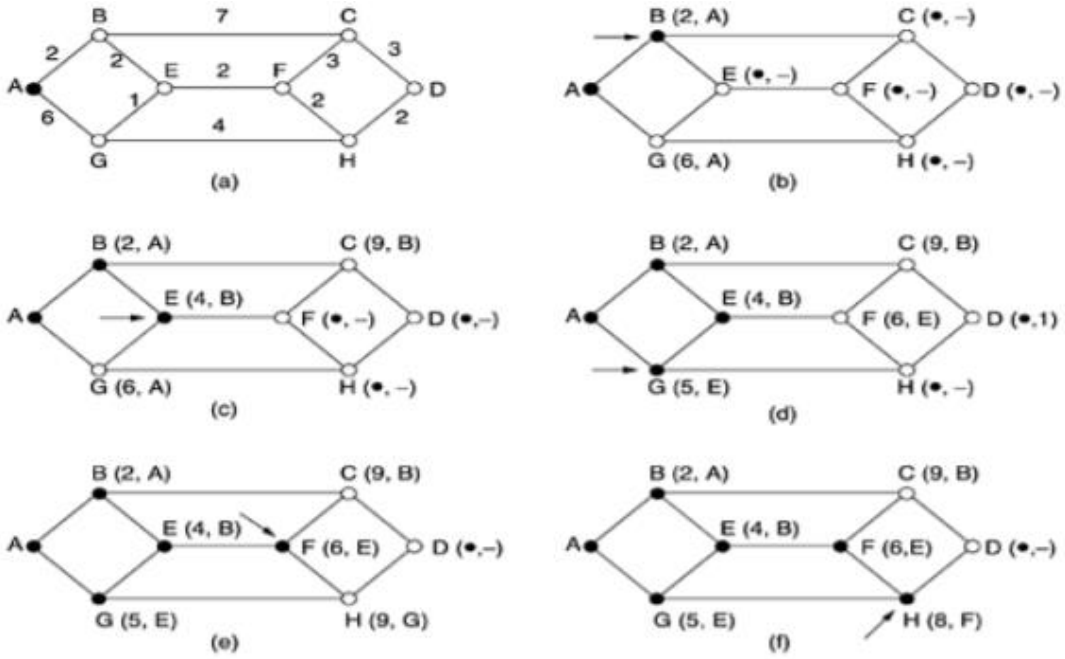


Figure 4.3 The first five steps used in computing the shortest path from A to D. The arrows indicate the working node.

To illustrate how the labelling algorithm works, look at the weighted, undirected graph of Fig. 4.3 (a), where the weights represent, for example, distance.

We want to find the shortest path from A to D. We start out by marking node A as permanent, indicated by a filled-in circle.

Then we examine, in turn, each of the nodes adjacent to A (the working node), relabeling each one with the distance to A.

Whenever a node is relabelled, we also label it with the node from which the probe was made so that we can reconstruct the final path later.

Having examined each of the nodes adjacent to A, we examine all the tentatively labelled nodes in the whole graph and make the one with the smallest label permanent, as shown in Fig. 4.3 (b). This one becomes the new working node.

We now start at B and examine all nodes adjacent to it. If the sum of the label on B and the distance from B to the node being considered is less than the label on that node, we have a shorter path, so

the node is relabeled

After all the nodes adjacent to the working node have been inspected and the tentative labels changed if possible, the entire graph is searched for the tentatively-labelled node with the smallest value. This node is made permanent and becomes the working node for the next round. Figure 4.3 shows the first five steps of the algorithm.

Another example using Dijkstra's algorithm to compute the shortest paths from a given source node to all other nodes in a network. Links are bi-directional, with the same distance in either direction. Distance can be any measure of cost.

Example with 8 nodes and 11 links

nodeset = {'A', 'B', 'C', 'D', 'E', 'F', 'G', 'H'}

linklist = [('A', 'B', 2), ('B', 'C', 7), ('C', 'D', 3),

(node,node,distance) ('B', 'E', 2), ('E', 'F', 2), ('F', 'C', 3), ('A', 'G', 6), ('G', 'E', 1), ('G', 'H', 4), ('F', 'H', 2), ('H', 'D', 2),

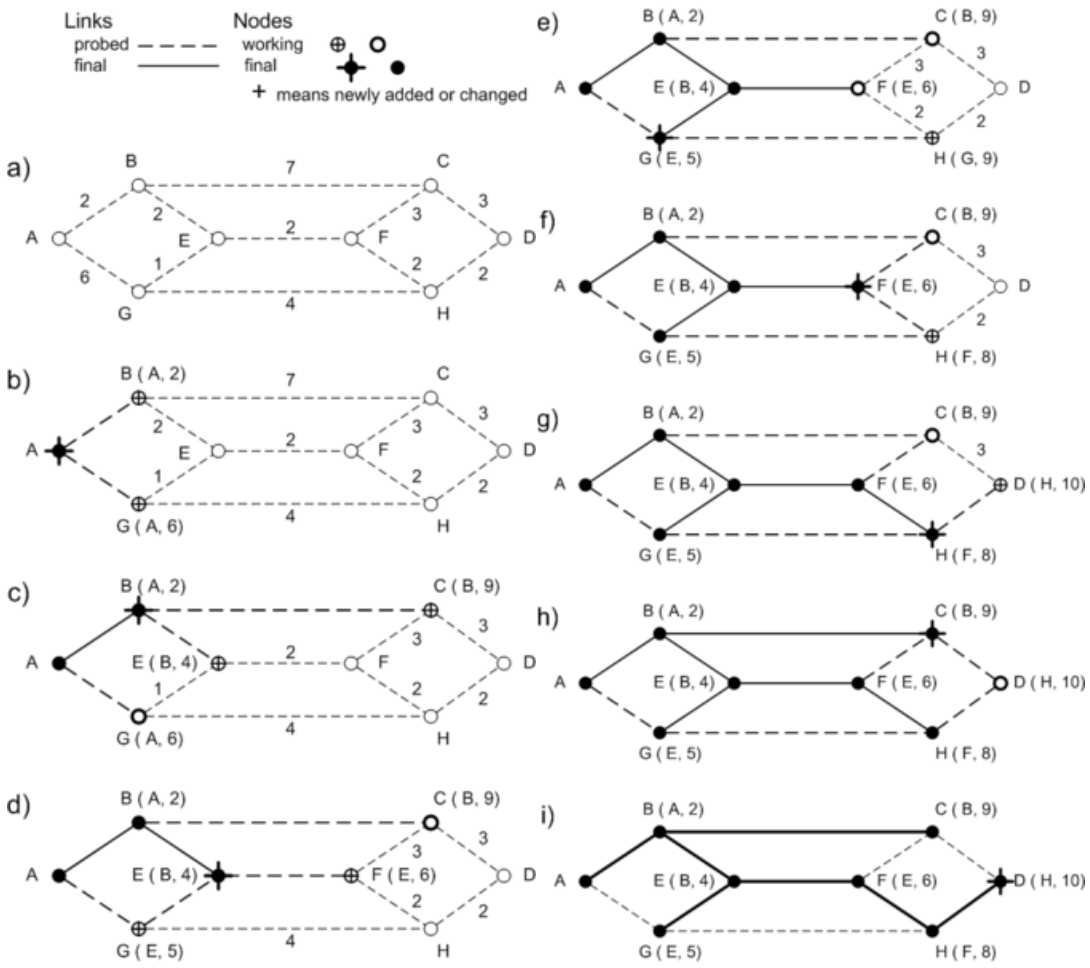


Figure 4.4 Dijkstra's algorithm

The strategy is to start at the source node, send probes to each of its adjacent nodes, pick the node with the shortest path from the source, and make that the new working node. Send probes from the new working node, pick the next shortest path, and make that the next working node. Continue selecting the shortest possible path until every every node in the network has been selected.

Figure 4.4 shows the first few steps in our example network. Labels on each node show its distance from the source, and the previous node on the path from which that distance was computed.

As new nodes are first probed, they are added to a working set, shown with a darkened open circle. After each probe cycle, we look at the entire set of working nodes. The node with the shortest path is moved to a final set, shown with a solid circle.

The light dotted lines are links not used in any shortest path from node A. They might be used in another tree, however. Each node in a network can compute its own shortest path tree, given the linklist for the entire network.

FLOODING

- Another static algorithm is flooding, in which every incoming packet is sent out on every outgoing line except the one it arrived on.
- Flooding obviously generates vast numbers of duplicate packets, in fact, an infinite number unless some measures are taken to damp the process.
- One such measure is to have a hop counter contained in the header of each packet, which is decremented at each hop, with the packet being discarded when the counter reaches zero.
- Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the subnet.

4.7 Distance-Vector Routing

Distance vector routing algorithms operate by having each router maintain a table (i.e, a vector) giving the best known distance to each destination and which line to use to get there.
These tables are updated by exchanging information with the neighbors.
The distance vector routing algorithm is sometimes called by other names, most commonly the distributed Bellman-Ford routing algorithm and the Ford-Fulkerson algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962).
It was the original ARPANET routing algorithm and was also used in the Internet under the name RIP.

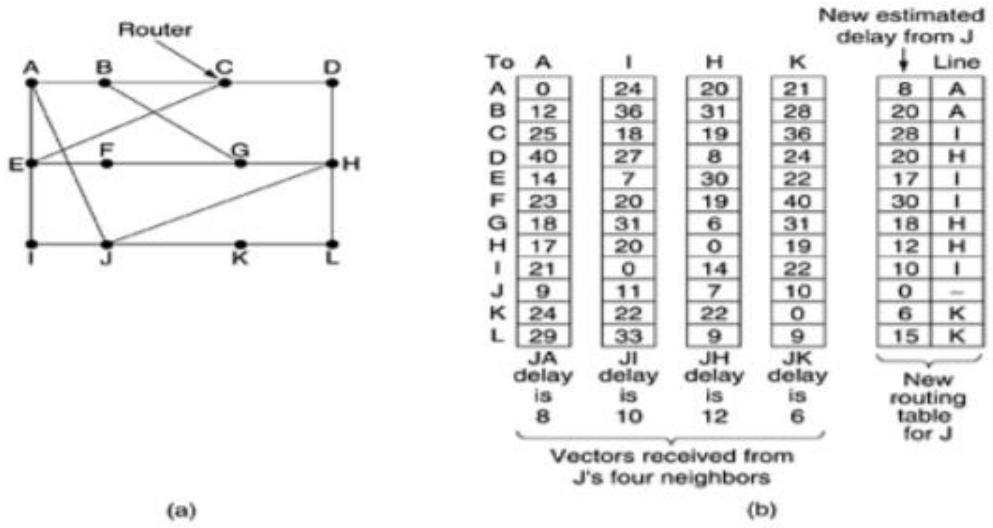


Figure 4.5 (a) A subnet. (b) Input from A, I, H, K, and the new routing table for J.

Part (a) shows a subnet. The first four columns of part (b) show the delay vectors received from the neighbours of router J.

A claims to have a 12-msec delay to B, a 25-msec delay to C, a 40-msec delay to D, etc. Suppose that J has measured or estimated its delay to its neighbours, A, I, H, and K as 8, 10, 12, and 6 msec, respectively.

Each node constructs a one-dimensional array containing the "distances"(costs) to all other nodes anddistributes that vector to its immediate neighbors.

The starting assumption for distance-vector routing is that each node knows the cost of the link to each of its directly connected neighbors.

A link that is down is assigned an infinite cost.

Example.

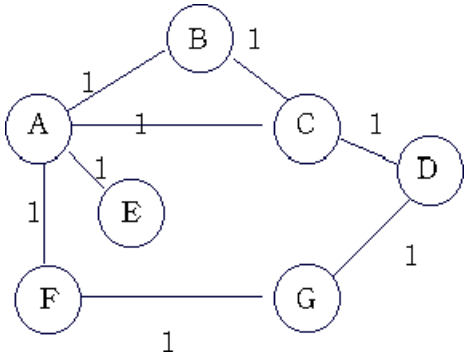


Figure 4.6 Distance-Vector Routing

Information Stored at Node	**Distance to Reach Node**						
	A	**B**	**C**	**D**	**E**	**F**	**G**
A	0	1	1	**?**	1	1	**?**
B	1	0	1	**?**	**?**	**?**	**?**
C	1	1	0	1	**?**	**?**	**?**
D	**?**	**?**	1	0	**?**	**?**	1
E	1	**?**	**?**	**?**	0	**?**	**?**
F	1	**?**	**?**	**?**	**?**	0	1
G	**?**	**?**	**?**	1	**?**	1	0

Table 1. Initial distances stored at each node (global view)

We can represent each node's knowledge about the distances to all other nodes as a table like the onegiven in Table 1.
Note that each node only knows the information in one row of the table.

Every node sends a message to its directly connected neighbors containing its personal list of distance. (for example, **A** sends its information to its neighbors **B,C,E**, and **F**.)

If any of the recipients of the information from **A** find that **A** is advertising a path shorter than the one they currently know about, they update their list to give the new path length and note that they should send packets for that destination through **A**. (node **B** learns from **A** that node **E** canbe reached at a cost of 1; **B** also knows it can reach **A** at a cost of 1, so it adds these to get the cost of reaching **E** by means of **A**. **B** records that it can reach **E** at a cost of 2 by going through **A**.)

After every node has exchanged a few updates with its directly connected neighbors, all nodeswill know the least-cost path to all the other nodes.

In addition to updating their list of distances when they receive updates, the nodes need to keep

track of which node told them about the path that they used to calculate the cost, so that they can create their forwarding table. (for example, **B** knows that it was **A** who said " I can reach **E** in one hop" and so **B** puts an entry in its table that says " To reach **E**, use the link to **A**.)

Information Stored at Node	**Distance to Reach Node**						
	A	**B**	**C**	**D**	**E**	**F**	**G**
A	0	1	1	2	1	1	2
B	1	0	1	2	2	2	3
C	1	1	0	1	2	2	2
D	2	2	1	0	3	2	1
E	1	2	2	3	0	2	3
F	1	2	2	2	2	0	1
G	2	3	2	1	3	1	0

Table 2. final distances stored at each node (global view).

In practice, each node's forwarding table consists of a set of triples of the form: (Destination, Cost, Next Hop).

For example, Table 3 shows the complete routing table maintained at node B for the network in figure 4.3.

Destination	**Cost**	**Next Hop**
A	1	A
C	1	C
D	2	C
E	2	A

F	2	A
G	3	A

Table 3. Routing table maintained at node B.

Broadcast routing

Sending a packet to all destinations simultaneously is called broadcasting.

1) The source simply sends a distinct packet to each destination. Not only is the method wasteful of bandwidth, but it also requires the source to have a complete list of all destinations.

2) Flooding. - The problem with flooding as a broadcast technique is that it generates too many packets and consumes too much bandwidth.

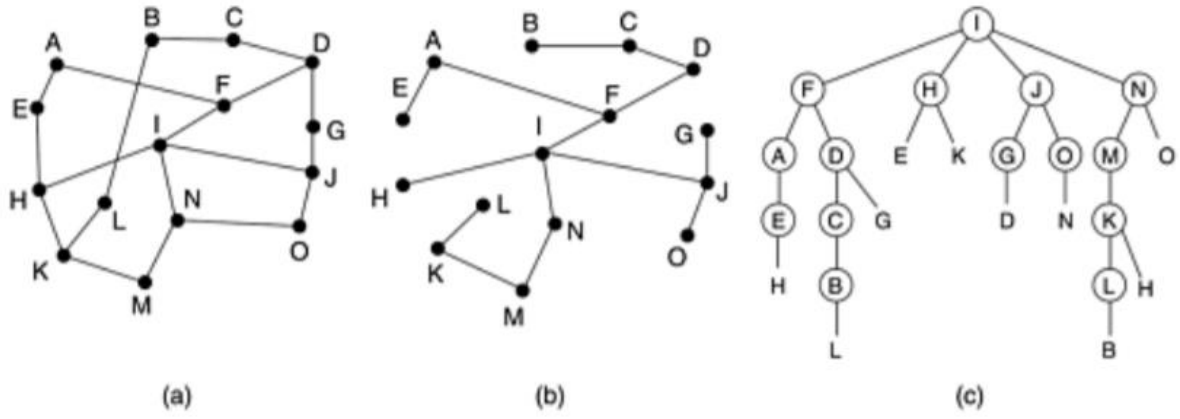


Figure 4.7 Reverse path forwarding. (a) A subnet. (b) A sink tree. (c) The tree built by reverse path forwarding.

Part (a) shows a subnet, part (b) shows a sink tree for router I of that subnet, and part (c) shows how the reverse path algorithm works.

- When a broadcast packet arrives at a router, the router checks to see if the packet arrived on the

line that is normally used for sending packets to the source of the broadcast. If so, there is an excellent chance that the broadcast packet itself followed the best route from the router and is therefore the first copy to arrive at the router.

- This being the case, the router forwards copies of it onto all lines except the one it arrived on. If, however, the broadcast packet arrived on a line other than the preferred one for reaching the source, the packet is discarded as a likely duplicate.

Congestion control algorithms

When too many packets are present in (a part of) the subnet, performance degrades. This situation is called congestion.

- Figure 4.8 depicts the symptom. When the number of packets dumped into the subnet by the hosts is within its carrying capacity, they are all delivered (except for a few that are afflicted with transmission errors) and the number delivered is proportional to the number sent.
- However, as traffic increases too far, the routers are no longer able to cope and they begin losing packets. This tends to make matters worse. At very high traffic, performance collapses completely and almost no packets are delivered.

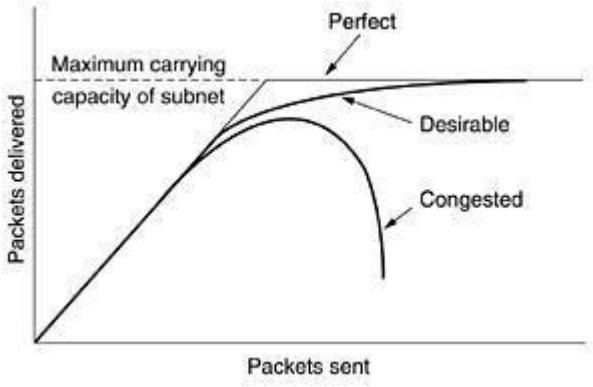


Figure 4.8. When too much traffic is offered, congestion sets in and performance degrades sharply.

Congestion can be brought on by several factors. If all of a sudden, streams of packets begin

arriving on three or four input lines and all need the same output line, a queue will build up.
If there is insufficient memory to hold all of them, packets will be lost.
Slow processors can also cause congestion. If the routers' CPUs are slow at performing the bookkeeping tasks required of them (queuing buffers, updating tables, etc.), queues can build up, even though there is excess line capacity. Similarly, low-bandwidth lines can also cause congestion.

APPROACHES TO CONGESTION CONTROL

Many problems in complex systems, such as computer networks, can be viewed from a control theory point of view. This approach leads to dividing all solutions into two groups: open loop and closed loop. Open loop solutions attempt to solve the problem by good design.
Tools for doing open-loop control include deciding when to accept new traffic, deciding when to discard packets and which ones, and making scheduling decisions at various points in the network.
Closed loop solutions are based on the concept of a feedback loop.
This approach has three parts when applied to congestion control: 1. Monitor the system to detect when and where congestion occurs. 2. Pass this information to places where action can be taken. 3. Adjust system operation to correct the problem.
A variety of metrics can be used to monitor the subnet for congestion. Chief among these are the percentage of all packets discarded for lack of buffer space, the average queue lengths, the number of packets that time out and are retransmitted, the average packet delay, and the standard deviation of packet delay. In all cases, rising numbers indicate growing congestion.
The second step in the feedback loop is to transfer the information about the congestion from the point where it is detected to the point where something can be done about it. In all feedback schemes, the hope is that knowledge of congestion will cause the hosts to take appropriate action to reduce the congestion.
The presence of congestion means that the load is (temporarily) greater than the resources (in part of the system) can handle. Two solutions come to mind: increase the resources or decrease the load.

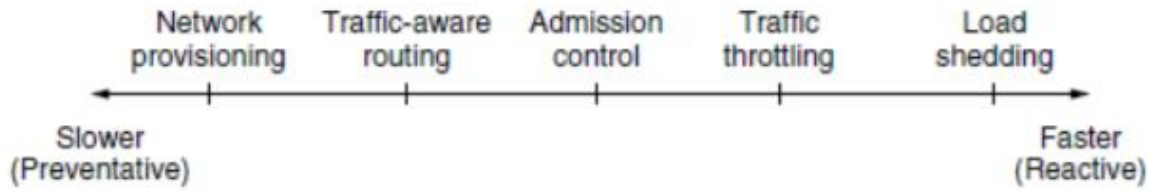


Figure 4.9 Timescales of Approaches to Congestion Control

Leaky Bucket Algorithm

Let us consider an example

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 sides and is lost.

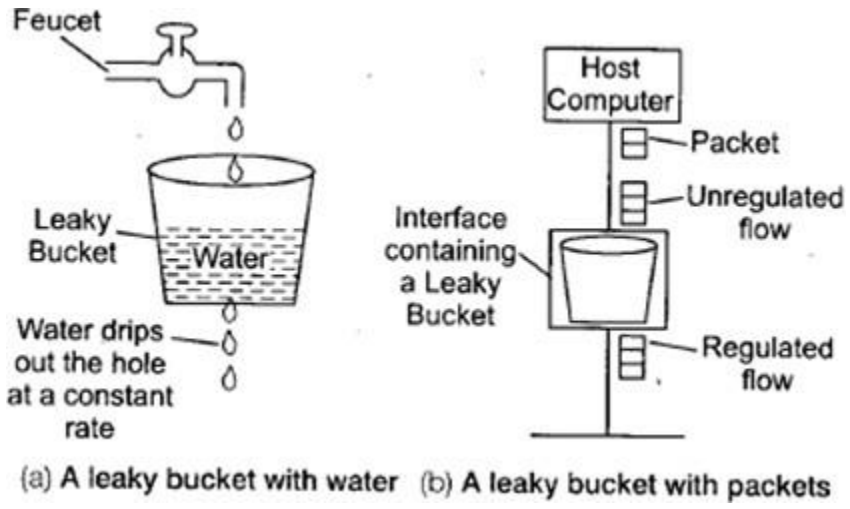


Figure 4.10 Leaky Bucket Algorithm

Similarly, each network interface contains a leaky bucket and the following **steps** are involved inleaky 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 aconstant 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. *f*

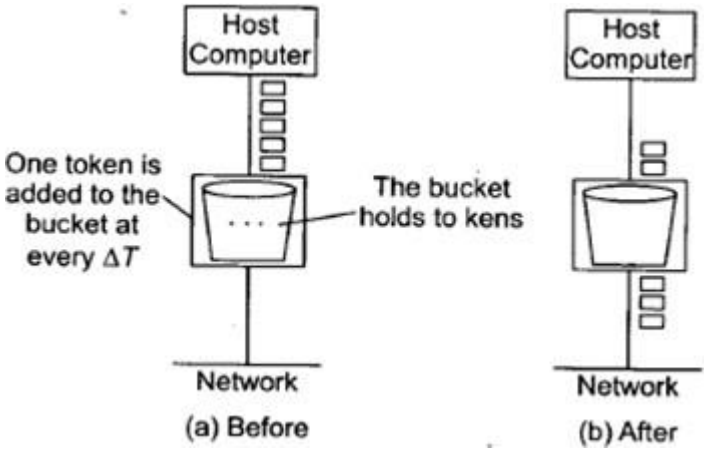
The bucket has a maximum capacity. *f*

If there is a ready packet, a token is removed from the bucket, and the packet is send.

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

Let's understand with an example,

In figure (A) 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.



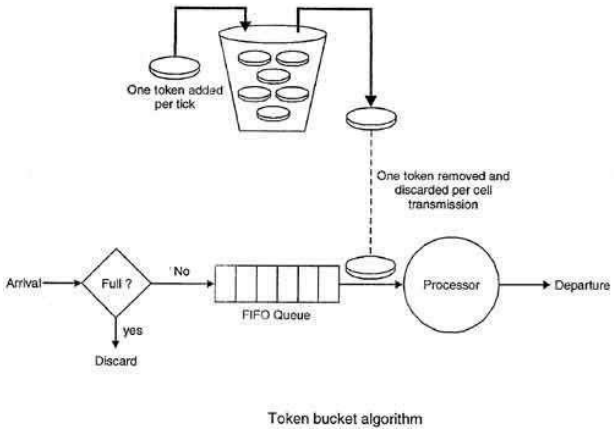


Figure 4.11 Token Bucket Algorithm

Congestion control in virtual Circuit

Different approaches are used to control the congestion in virtual-circuit network. Some of them are as follows:

Admission control: In this approach, once the congestion is signaled, no new connections are set up until the problem is solved. This type of approach is often used in normal telephone networks. When the exchange is overloaded, then no new calls are established. Allow new virtual connections other than the congested area.

Negotiate an agreement between the host and the network when the connection is setup. This agreement specifies the volume and shape of traffic, quality of service, maximum delay and other parameters. The network will reserve resources (Buffer space, Bandwidth and CPU cycle) along the path when the connection is set up. Now congestion is unlikely to occur on the new connections because all the necessary resources are guaranteed to be available. The disadvantage of this approach is that it may leads to wasted bandwidth because of the some idle connection.

Congestion control in Datagram Subnets

Congestion control in Datagram Subnets is achieved by sending warning to sender in advance. Each router can easily monitor the utilization of its output lines. If utilization is greater than threshold value then output line may be congested in future so mark it as warning state. Each newly arriving packet is checkedto see if its output line is in warning state. If it is, some action is taken.

The actions are:
The warning bit
Choke packets
Hop-by-hop choke packet

The warning bit
When a new packet is to be transmitted on the output line marked as warning state, a special bit is added in header to signal this state. At the destination, this information is sent back with ACK to the sender so that it could cut the traffic. When warning bit is absent, sender increases its transmitting rate.
Note: It uses a whole trip (source to destination to source) to tell the source to slow down

Choke Packet Technique
In this approach, the router sends a choke packet back to the source host. The original packet is marked so that it would not generate any more choke packets further along the path and is then forwarded in the usual way. When the source gets the choke packet, it is required to reduce the traffic by X packets.

Hop-by Hop Choke Packets

In this approach, unlike choke packet, reduction of flow starts from intermediate node rather than source node. To understand this, let us refer the figure 2. When the choke packet reaches the nearest router (say R) from router Q, it reduces the flow. However, router R now requires devoting more buffers to the flow since the source is still sending at full blast but it gives router Q immediate relief. In the next step, the choke packet reaches P and flow genuinely slow down. The net effect of hop-by-hop scheme is to provide quick relief at the point of congestion at the price of using up more buffers upstream.

The network layer in the internet

The transport layer enables the applications to efficiently and reliably exchange data. Transport layer entities expect to be able to send segment to any destination without having to understand anything aboutthe underlying subnetwork technologies. Many subnetwork technologies exist. Most of them differ in subtle details (frame size, addressing, ...). The network layer is the glue between these subnetworks and the transport layer. It hides to the transport layer all the complexity of the underlying subnetworks and ensures that information can be exchanged between hosts connected to different types of subnetworks.

Principles :

The main objective of the network layer in is to allow end systems, connected to different networks, to exchange information through intermediate systems called router. The unit of information in the network layer is called a packet**.**

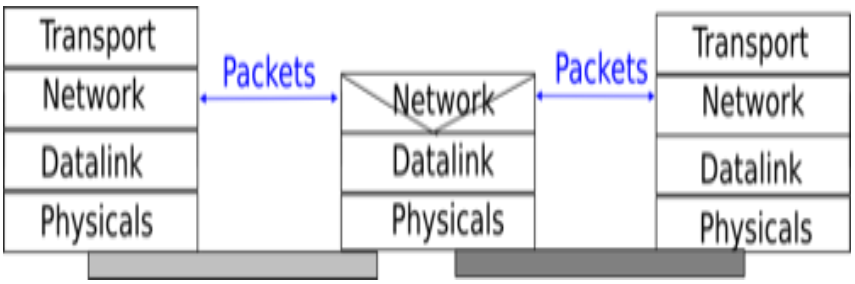


Figure 4.12 Representation of network layer

OSI model vs. TCP/IP model

The TCP/IP model is an alternative model of how the Internet works. It divides the processes involved into four layers instead of seven. Some would argue that the TCP/IP model better reflects the way the Internet functions today, but the OSI model is still widely referenced for understanding the Internet, and both models have their strengths and weaknesses.

In the TCP/IP model, the four layers are:

4. Application layer: This corresponds, approximately, to layer 7 in the OSI model.

3. Transport layer: Corresponds to layer 4 in the OSI model.

2. Internet layer: Corresponds to layer 3 in the OSI model.

1. Network access layer: Combines the processes of layers 1 and 2 in the OSI model.

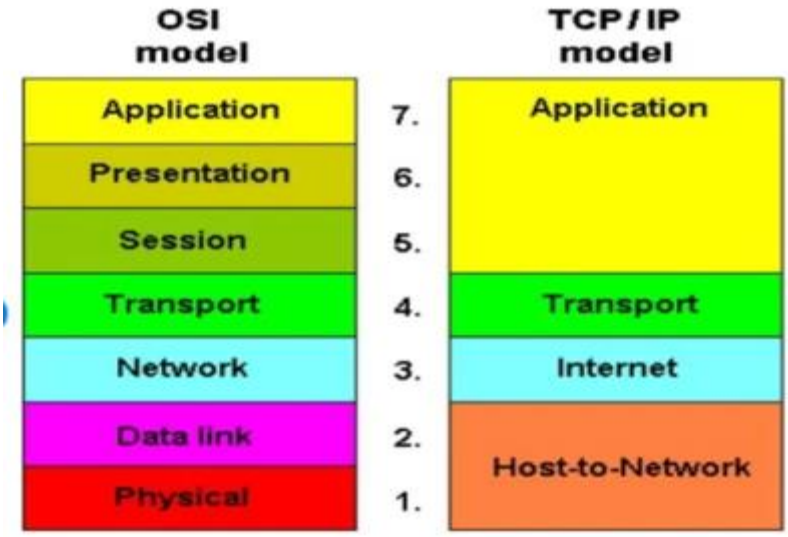


Figure 4.13 OSI Model Vs TCP/IP Model

The Internet Protocol (IP) is a network layer protocol.

Hosts and gateways process packets called Internet datagrams (IP datagrams).

IP provides connectionless, best-effort delivery service.

The Transmission Control Protocol (TCP) is a transport layer protocol that provides reliable stream service between processes on two machines. It is a sliding window protocol that uses acknowledgments and retransmissions to overcome the unreliability of IP.

The Universal Datagram Protocol (UDP) provides connectionless datagram service between machines.

An **Internet Protocol address** (**IP address**) is a numerical label assigned to each device connected toa **computer network** that uses the **Internet Protocol** for communication**.** An IP address serves two principal functions: host or network interface **identification** and location **addressing.**

Internet Addressing

Host identifiers are classified as names, addresses, or routes, where:

A name suggests what object we want.

An address specifies where the object is.

A route tells us how to get to the object.

In the Internet, names consist of human-readable strings such as eve, percival, or gwen.cs.purdue.edu.

Addresses consist of compact, 32-bit identifiers. Internet software translates names into addresses; lower protocol layers always uses addresses rather than names.

Internet addresses are hierarchical, consisting of two parts:

Network:

The network part of an address identifies which network a host is on. Conceptually, each LAN has its own unique IP network number.

Local:

The local part of an address identifies which host on that network.

Address Classes

The Internet designers were unsure whether the world would evolve into a few networks with many hosts (e.g., large networks), or many networks each supporting only a few hosts (e.g., small networks). Thus, Internet addresses handle both large and small networks. Internet address are four bytes in size, where:

Class A addresses start with a ``0" in the most significant bit, followed by a 7-bit network address and a 24-bit local part.

Class B addresses start with a ``10" in the two most significant bits, followed by a 14-bit network number and a 16-bit local part.

Class C addresses start with a ``110" in the three most significant bits, followed by a 22-bit network number and an 8-bit local part.

Class D addresses start with a ``1110" in the four most significant bits, followed by a 28-bit group number.

Note: The use of fixed-sized addresses makes the routing operation efficient. In the ISO world, addresses are of varying format and length and just extracting the address from the packet may not be straightforward.

Internet addresses can also refer to broadcast addresses. The all 1's address is used to mean ``broadcast on this network". Of course, if the underlying network technology doesn't support broadcasting, one can't broadcast Internet datagrams either.

Network addresses are written using dotted decimal notation. Each address consists of 4 bytes,

and each byte is written in decimal form. Sample addresses:

wpi.wpi.edu: 130.215.24.6 (class B)

owl.wpi.edu: 130.215.8.139 (class B)

wpi.edu: 130.215 (a network address)

rialto.mcs.vuw.ac.nz: 130.195.5.15 (class B)

gwen.cs.purdue.edu: 128.10.2.3 (class B)

c.nyser.net: 192.33.4.12 (Class C)

pescadero.stanford.edu: 36.8.0.8 (class A)

su-net-temp: 36 (network address)

Note: Internet addresses refer to network connections rather than hosts. Gateways, for instance, have two or more network connections and each interface has its own IP address. Thus, there is not a one-to-one mapping between host names and IP addresses.

4.11 IPv6

IPv6 is the replacement Internet protocol for IPv4. It corrects some of the deficiencies of IPv4 and simplifies the way that addresses are configured and how they are handled by Internet hosts. IPv4 has proven to be robust, easily implemented, and interoperable, and has stood the test of scaling an internetwork to a global utility the size of the Internet. However, the initial design did not anticipate the following conditions:

• Recent exponential growth of the Internet and the impending exhaustion of the IPv4 address space

• The ability of Internet backbone routers to maintain large routing tables

• Need for simpler auto configuration and renumbering

• Requirement for security at the IP level (IPSec)

• Need for better support for real-time delivery of data, known as quality of service (QoS)

Need for IPv6

With 32-bit address format, IPv4 can handle a maximum 4.3 billion unique IP addresses.

While this number may seem very large, it is not enough to sustain and scale the rapidly rising growth of the Internet. Although improvements to IPv4, including the use of NAT, have allowed the extended use of the protocol, address exhaustion is inevitable and could happen as soon as 2012. With its 128-bit address format, IPv6 can support 3.4 x 1038 or 340, 282, 366, 920, 938,463,463,374,607,431,768,211,456 unique IP addresses.

This number of addresses is large enough to configure a unique address on every node in the Internet and still have plenty of addresses left over. It is also large enough to eliminate the need

for NAT, which has its own inherent problems.

A few countries, governmental agencies, and multinational corporations have either already deployed or mandated deployment of IPv6 in their networks and software products. Some emerging nations have no choice but to deploy IPv6 because of the unavailability of new IPv4 addresses.

Advantages of IPv6

Besides providing an almost limitless number of unique IP addresses for global end to-end reachability and scalability, IPv6 has the following additional advantages:

• Simplified header format for efficient packet handling

• Larger payload for increased throughput and transport efficiency

• Hierarchical network architecture for routing efficiency

• Support for widely deployed routing protocols (OSPF, BGP, etc.)

• Auto configuration and plug-and-play support

• Elimination of need for network address translation (NAT) and application layered gateway (ALG).

• Increased number of multicast addresses.

IPv6 Simplifications

Fixed format headers – Use extension headers instead of options

• Remove header checksum – Rely on link layer and higher layers to check integrity of data

• Remove hop-by-hop segmentation – Fragmentation only by sender due to path MTU discovery

IPv6 Header Format

A side-by-side comparison of the IPv4 header and the IPv6 header in figure shows that the IPv6 header is more streamlined and efficient than the IPv4 header.

Fixed Header

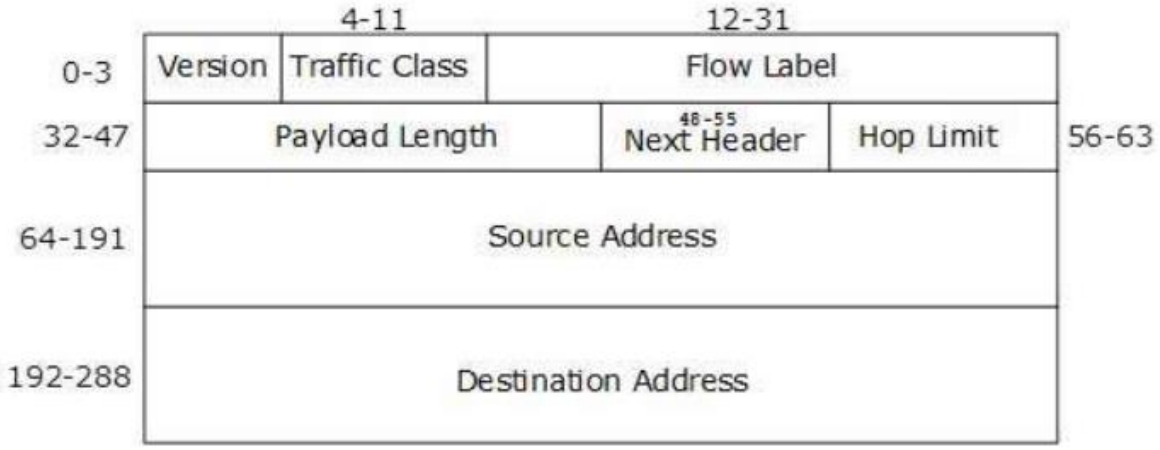


Figure 4.14 IPV6 Fixed Header

An IPv6 address is 4 times larger than IPv4, but surprisingly, the header of an IPv6 address is only 2 times larger than that of IPv4. IPv6 headers have one Fixed Header and zero or more Optional (Extension) Headers. All the necessary information that is essential for a router is kept in the Fixed Header. The Extension Header contains optional information that helps routers to understand how to handle a packet/flow.

Version (4-bits): It represents the version of Internet Protocol, i.e. 0110.

Traffic Class (8-bits): These 8 bits are divided into two parts. The most significant 6 bits are used for Type of Service to let the Router Known what services should be provided to this packet. The least significant 2 bits are used for Explicit Congestion Notification (ECN).

Flow Label (20-bits): This label is used to maintain the sequential flow of the packets belonging to a communication. The source labels the sequence to help the router identify that a particular packet belongs to a specific flow of information. This field helps avoid re-ordering of data packets. It is designed for streaming/real-time media.

Payload Length (16-bits): This field is used to tell the routers how much information a particular packet contains in its payload. Payload is composed of Extension Headers and Upper Layer data. With 16 bits, up to 65535 bytes can be indicated; but if the Extension Headers contain Hop-by-Hop Extension Header, then the payload may exceed 65535 bytes and this field is set to 0.

Next Header (8-bits): This field is used to indicate either the type of Extension Header, or if the Extension Header is not present then it indicates the Upper Layer PDU. The values for the type of Upper Layer PDU are same as IPv4's.

Hop Limit (8-bits): This field is used to stop packet to loop in the network infinitely. This is same as TTL in IPv4. The value of Hop Limit field is decremented by 1 as it passes a link (router/hop). When the field reaches 0 the packet is discarded.

Source Address (128-bits): This field indicates the address of originator of the packet.

Destination Address (128-bits): This field provides the address of intended recipient of the packet.

Extension Headers

In IPv6, the Fixed Header contains only that much information which is necessary, avoiding those information which is either not required or is rarely used. All such information is put between the Fixed Header and the Upper layer header in the form of Extension Headers. Each Extension Header is identified by a distinct value.

When Extension Headers are used, IPv6 Fixed Header's Next Header field points to the first Extension Header. If there is one more Extension Header, then the first Extension Header's 'Next-Header' field points to the second one, and so on. The last Extension Header's 'Next-Header' field points to the Upper Layer Header. Thus, all the headers points to the next one in a linked list manner.

If the Next Header field contains the value 59, it indicates that there are no headers after this header, not even Upper Layer Header.

The following Extension Headers must be supported as per RFC 2460:

Extension Header	Next Header Value	Description
Hop-by-Hop Options header	0	read by all devices in transit network
Routing header	43	contains methods to support making routing decision
Fragment header	44	contains parameters of datagram fragmentation
Destination Options header	60	read by destination devices
Authentication header	51	information regarding authenticity
Encapsulating Security Payload header	50	encryption information

Figure 4.15 Extension Header

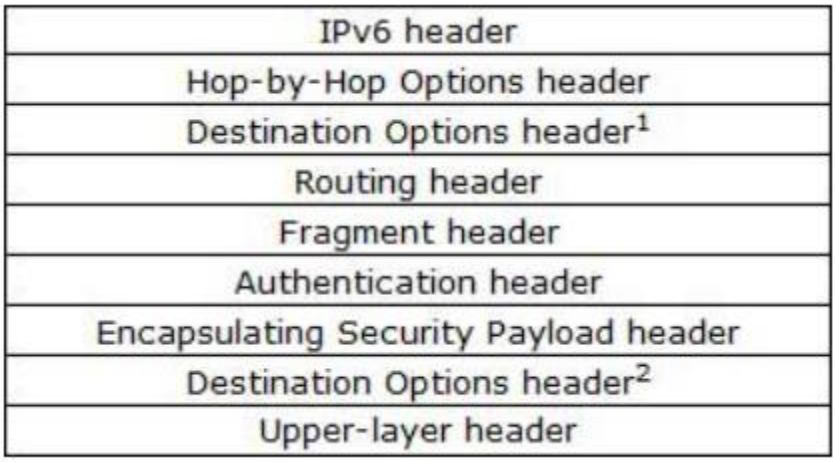


Figure 4.16 Sequence of Extension Header

These headers:

1. should be processed by First and subsequent destinations.

2. should be processed by Final Destination.

Extension Headers are arranged one after another in a linked list manner, as depicted in the following diagram:

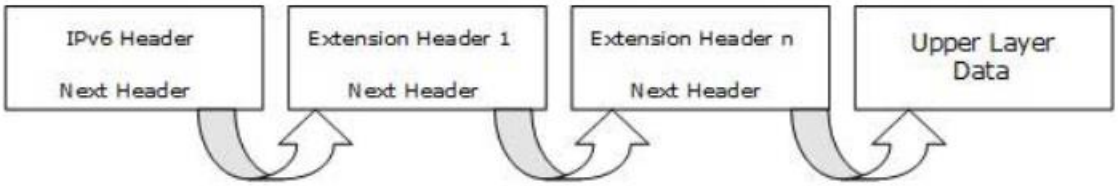


Figure 4.17 Extension Headers Connected Format

4.12 Address resolution protocol

While communicating, a host needs Layer-2 (MAC) address of the destination machine which belongs to the same broadcast domain or network. A MAC address is physically burnt into the Network Interface Card (NIC) of a machine and it never changes. On the other hand, IP address on the public domain is rarely changed. If the NIC is changed in case of some fault, the MAC address also changes. This way, for Layer-2 communication to take place, a mapping between the two is required.

Dynamically builds table of IP to physical address bindings for a local network

Broadcast request if IP address not in table

All learn IP address of requesting node (broadcast)

Target machine responds with its physical address

Table entries are discarded if not refreshed Reverse Address resolution protocol

ARP Ethernet frame format

The address resolution protocol (ARP) uses a basic message format that contains either address resolution request or address resolution response. The ARP message size depends on the address size of the link layer and the network layer. The message header describes the network type used at each layer and the address size of each layer. The message header is complete with the help of the operation code, which is 1 for request and 2 for the response. The payload of the packet has four addresses, these are:

Hardware address of the sender hosts

Hardware address of the receiver hosts

Protocol address of the sender hosts

Protocol address of the receiver hosts

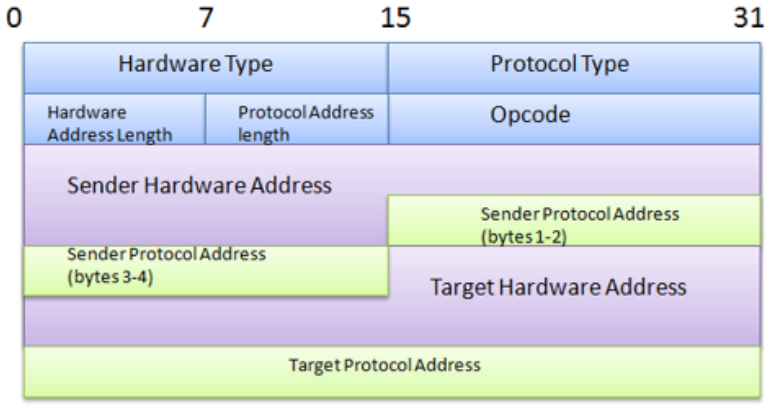


Figure 4.18 ARP Header

HTYPE (Hardware Type) - The size of the hardware type field is 16 bit. This field defines the network type that the local network needs to transmit the ARP message. There are some typical values for this field, which are given below:

Hardware Type (HTYPE)	**Value**
Ethernet	1
IEEE 802 Networks	6
ARCNET	7
Frame Relay	15
Asynchronous Transfer Mode (ATM)	16
HDLC	17
Fibre Channel	18

Asynchronous Transfer Mode (ATM)	19
Serial Line	20

Table.4 Hardware Type

PTYPE (Protocol Type) - The protocol type is a 16-bit field used to specify the type of protocol.

HLEN (Hardware Length) - The size of the hardware length field is 8-bit. This field specifies the length of the physical address in bytes.

Example: For this, the address length of Ethernet is 6.

PLEN (Protocol Length) - The size of the protocol length field is 8-bit long. It defines the length of the IP address in bytes.

OPER (Operation) - It is a 16-bit field that determines the type of ARP packet. There are two types of ARP packets, i.e., ARP request and ARP Reply. In the given table, the first two values are used for the ARP request and reply. The values for the other ARP frame format such as RARP, DRARP, etc. are also specified in this table.

ARP Message Type	**Opcode (Operation Code)**
ARP Request	1
ARP Reply	2
RARP Request	3
RARP Reply	4
DRARP Request	5
DRARP Reply	6
DRARP Error	7
InARP Request	8
InARP Reply	9

Table.5 Message Type

SHA (Sender Hardware Address) - This field specifies the physical address of the sender, and the length of this field is not fixed.

SPA (Sender Protocol Address) - This field is used to determine the logical address of the sender, and the length of this field is not fixed.

THA (Target Hardware Address) - The target hardware address specifies the physical address of the target. It is a variable-length field. For the ARP request packet, this field contains all zeros because the sender does not know the physical address of the receiver.

TPA (Target Protocol Address) - This field determines the logical address of the target. TPA

is a variable-length field.

4.13 Dynamic Host Configuration Protocol (DHCP)

Is an application layer protocol which is used to provide:

1. Subnet Mask (Option 1 – e.g., 255.255.255.0)

Router Address (Option 3 – e.g., 192.168.1.1)

DNS Address (Option 6 – e.g., 8.8.8.8)

Vendor Class Identifier (Option 43 – e.g., 'unifi' = 192.168.1.9 ##where unifi = controller)

DHCP is based on a client-server model and based on discovery, offer, request, and ACK. DHCP **port number** for server is 67 and for the client is 68. It is a Client server protocol which uses UDP services. IP address is assigned from a pool of addresses. In DHCP, the client and the server exchange mainly 4 DHCP messages in order to make a connection, also called DORA process, but there are 8 DHCP messages in the process. These messages are given as below:

DHCP discover message –

This is a first message generated in the communication process between server and client. This message is generated by Client host in order to discover if there is any DHCP server/servers are present in a network or not. This message is broadcasted to all devices present in a network to find the DHCP server. This message is 342 or 576 bytes long.

DHCP offer message –

The server will respond to host in this message specifying the unleased IP address and other TCP configuration information. This message is broadcasted by server. Size of message is 342 bytes. If there are more than one DHCP servers present in the network then client host will accept the first DHCP OFFER message it receives. Also a server ID is specified in the packet in order to identify the server.

DHCP request message –

When a client receives a offer message, it responds by broadcasting a DHCP request message. The client will produce a gratitutous ARP in order to find if there is any other host present in the network with same IP address. If there is no reply by other host, then there is no host with same TCP configuration in the network and the message is broadcasted to server showing the acceptance of IP address .A Client ID is also added in this message.

DHCP acknowledgement message –

In response to the request message received, the server will make an entry with specified client ID and bind the IP address offered with lease time. Now, the client will have the IP address provided by server.

DHCP negative acknowledgement message –

Whenever a DHCP server receives a request for IP address that is invalid according to the scopes that is configured with, it send DHCP Nak message to client. Eg-when the server has no IP address unused or the pool is empty, then this message is sent by the server to client.

DHCP decline –

If DHCP client determines the offered configuration parameters are different or invalid, it sends DHCP decline message to the server .When there is a reply to the gratuitous ARP by any host to the client, the client sends DHCP decline message to the server showing the offered IP address is already in use.

DHCP release –

A DHCP client sends DHCP release packet to server to release IP address and cancel any remaining lease time.

DHCP inform –

If a client address has obtained IP address manually then the client uses a DHCP inform to obtain other local configuration parameters, such as domain name. In reply to the dhcp inform message, DHCP server generates DHCP ack message with local configuration suitable for the client without allocating a new IP address. This DHCP ack message is unicast to the client.

Note – All the messages can be unicast also by dhcp relay agent if the server is present in different network.

Advantages – The advantages of using DHCP include:

centralized management of IP addresses

ease of adding new clients to a network

reuse of IP addresses reducing the total number of IP addresses that are required

simple reconfiguration of the IP address space on the DHCP server without needing to reconfigure each client

The DHCP protocol gives the network administrator a method to configure the network from a centralized area. With the help of DHCP, easy handling of new users and reuse of IP address can be achieved.

Disadvantages – Disadvantage of using DHCP is: IP conflict can occur

4.13 INTERNET CONTROL MESSAGE PROTOCOL (ICMP)

IP provides unreliable connectionless datagram service, original aim being efficient use of

network resources.

Types of messages ICMP messages are divided into two broad categories:

Error reporting Messages. 2. Query Messages

Error reporting: ICMP was designed to compensate the shortcoming of unreliability in IP. However ICMP does not correct errors, but only reports them. Error reporting messages are always sent to the original source. Five types of errors are handled:

Destination unreachable—In situations where a router cannot route a datagram or a host cannot deliver a datagram, the datagram is discarded and the router or host sends a destination unreachable message back to the source.

Source Quench—IP being a connectionless protocol, there is no communication between the source host, the router and the destination host. The resulting lack of flow control is a major hazard in the operation of source-destination delivery. And the lack of congestion control causes major problems n the routers. The source quench message in ICMP adds some flow control and congestion control to IP by notifying the source of a datagram being discarded and forcing it to slow down its transmission.

Time Exceeded—It is generated in two cases

A router receives a datagram with a zero value in the TTL field

All fragments that make up a message do not arrive at the destination host within a certain time limit

Parameter Problem—If a router or a destination host discovers an ambiguous or missing value in a any field of the datagram, it discards the datagram and sends a parameter problem message back to the source.

Redirection—When a host comes up, its routing table has a limited number of entries. It usually knows the IP address of a single default router. For this reason the host may send a datagram to the wrong router. The router that receives the datagram will forward it to the correct router and will send a redirection message back to the host for routing table updating.

Query Messages:

Query messages are used to diagnose some network problems. There are four different pairs of messages.

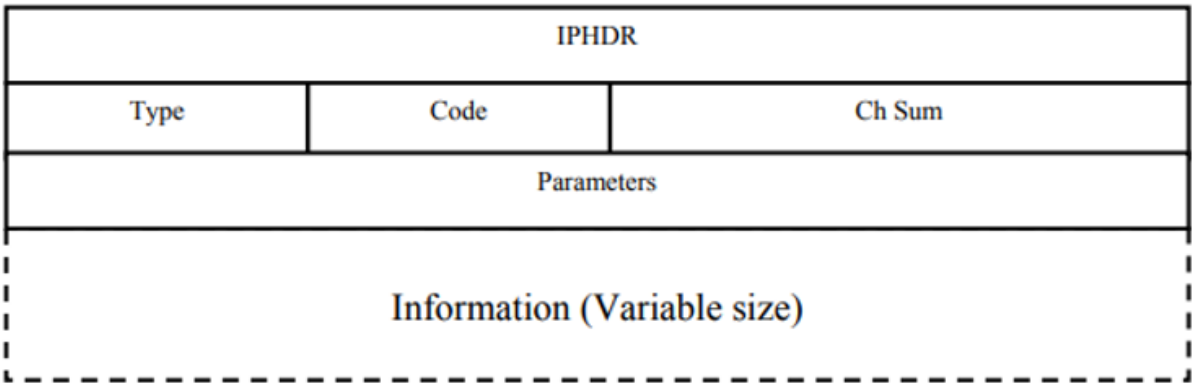
Echo Request/Reply messages—are designed for diagnostic purposes. Their combination determines whether two systems can communicate with each other.

Time stamp Request/Reply messages—can be used to determine the round trip time for an

IP datagram to travel between two machines and also to synchronize the clocks in them.

Address mask Request/Reply message—are used between the host and the router to indicate which part of the address defines the network and the sub-network address and which part corresponds to the host identifier.

Router Solicitation and Advertisement—are useful to inform a host that wants to send data to a host on another network, the address of routers connected to its own network and also their status and functioning.



Type	Specifies the types of errors, generally 256 types of errors may occur
: Code	Parameters that can be coded in a few bits.
Checksum	Checksum of entire IP message
Parameters	Specifies more lengthy parameters.

Figure 4.19 Header of ICMP

4.14 Classless addressing (CIDR - Classless Inter-Domain Routing)

Classless Inter-Domain Routing (CIDR) is another name for classless addressing. This addressing type aids in the more efficient allocation of IP addresses. This technique assigns a block of IP addresses based on specified conditions when the user demands a specific amount of IP addresses. This block is known as a "CIDR block", and it contains the necessary number of IP addresses.

When allocating a block, classless addressing is concerned with the following three rules.

Rule 1 − The CIDR block's IP addresses must all be contiguous.

Rule 2 − The block size must be a power of two to be attractive. Furthermore, the block's size is equal to the number of IP addresses in the block.

Rule 3 − The block's first IP address must be divisible by the block size.

For example, assume the classless address is 192.168.1.35/27.

The network component has a bit count of 27, whereas the host portion has a bit count of 5. (32-27)

The binary representation of the address is: (00100011 . 11000000 . 10101000 . 00000001).

(11000000.10101000.00000001.00100000) is the first IP address (assigns 0 to all host bits), that is, 192.168.1.32

(11000000.10101000.00000001.00111111) is the most recent IP address (assigns 1 to all host bits), that is, 192.168.1.63

The IP address range is 192.168.1.32 to 192.168.1.63.

Difference Between Classful and Classless Addressing

Classful addressing is a technique of allocating IP addresses that divides them into five categories. Classless addressing is a technique of allocating IP addresses that is intended to replace classful addressing in order to reduce IP address depletion.

The utility of classful and classless addressing is another distinction. Addressing without a class is more practical and helpful than addressing with a class.

The network ID and host ID change based on the classes in classful addressing. In classless addressing, however, there is no distinction between network ID and host ID. As a result, another distinction between classful and classless addressing may be made.

It was introduced in 1993 (**RCF 1517**) replacing the previous generation of IP address syntax – classful networks. CIDR introduction allowed for:

More efficient use of **IPv4 address** space

Prefix aggregation, which reduced the size of routing tables

CIDR allows routers to group together to reduce the bulk of routing information carried by the core routers. With CIDR, IP addresses and their subnet masks are written as four octets, separated by periods, followed by a forward slash and a two-digit number that represents the subnet mask e.g. 10.1.1.0/30, 172.16.1.16/28 and 192.168.1.32/27 etc.

CIDR / VLSM Network addressing topology example

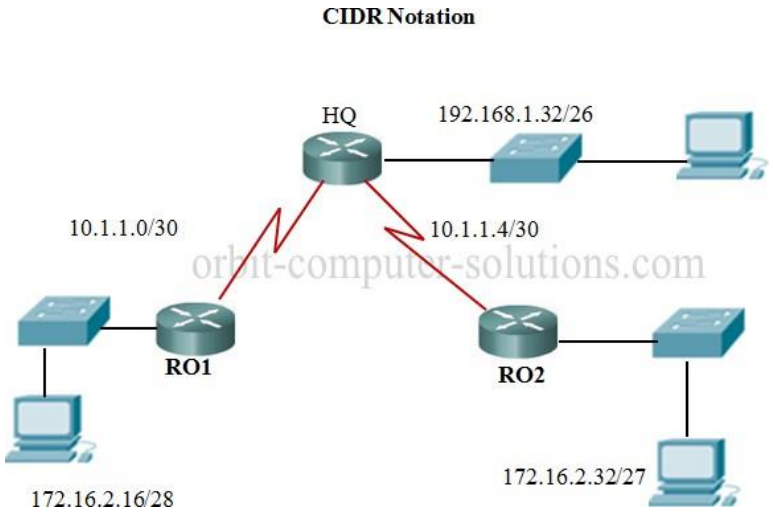


Figure 4.20 CIDR

CIDR uses VLSM (Variable Lenght Subnet Masks) to allocate IP addresses to subnetworks according to need rather than class. VLSM allows for subnets to be further divided or subnetted into even smaller subnets. With CIDR, address classes (Class A, B, and C) became meaningless. The network address was no longer determined by the value of the first octet, but assigned prefix length (subnet mask) address space. The number of hosts on a network, could now be assigned a specific prefix depending upon the number of hosts needed for that network. Propagating CIDR supernets or VLSM subnets require a classless **Routing Protocols** – A classless routing protocol includes the subnet mask along with the network address in the routing update.

Summary routes determination

Determining the summary route and subnet mask for a group of networks can be done in three easy steps:

To list the networks in **binary** format.

To count the number of left-most matching bits. This will give you the prefix length or subnet mask for the summarized route.

To copy the matching bits and then add zero bits to the rest of the address to determine the summarized network address.

CIDR Advantages

With the introduction of CIDR and VLSM, ISPs could now assign one part of a classful

network to one customer and different part to another customer. With the introduction of VLSM and CIDR, network administrators had to use additional **subnetting skills.**

4.15 Network Address Translation (NAT)

To access the Internet, one public IP address is needed, but we can use a private IP address in our private network. The idea of NAT is to allow multiple devices to access the Internet through a single public address. To achieve this, the translation of a private IP address to a public IP address is required. Network Address Translation (NAT) is a process in which one or more local IP address is translated into one or more Global IP address and vice versa in order to provide Internet access to the local hosts. Also, it does the translation of port numbers i.e. masks the port number of the host with another port number, in the packet that will be routed to the destination. It then makes the corresponding entries of IP address and port number in the NAT table. NAT generally operates on a router or firewall.

Network Address Translation (NAT) working –

Generally, the border router is configured for NAT i.e the router which has one interface in the local (inside) network and one interface in the global (outside) network. When a packet traverse outside the local (inside) network, then NAT converts that local (private) IP address to a global (public) IP address. When a packet enters the local network, the global (public) IP address is converted to a local (private) IP address. If NAT runs out of addresses, i.e., no address is left in the pool configured then the packets will be dropped and an Internet Control Message Protocol (ICMP) host unreachable packet to the destination is sent.

Why mask port numbers ?

Suppose, in a network, two hosts A and B are connected. Now, both of them request for the same destination, on the same port number, say 1000, on the host side, at the same time. If NAT does only translation of IP addresses, then when their packets will arrive at the NAT, both of their IP addresses would be masked by the public IP address of the network and sent to the destination. Destination will send replies to the public IP address of the router. Thus, on receiving a reply, it will be unclear to NAT as to which reply belongs to which host (because source port numbers for both A and B are the same). Hence, to avoid such a problem, NAT masks the source port number as well and makes an entry in the NAT table.

NAT inside and outside addresses –

Inside refers to the addresses which must be translated. Outside refers to the addresses which are not in control of an organization. These are the network Addresses in which the translation of the addresses will be done.

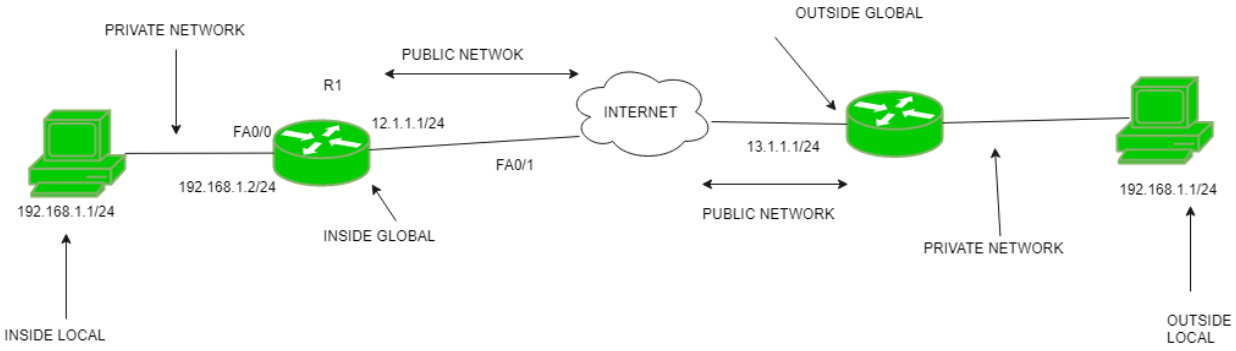


Figure 4.21 Network Address Translation (NAT)

Inside local address – An IP address that is assigned to a host on the Inside (local) network. The address is probably not an IP address assigned by the service provider i.e., these are private IP addresses. This is the inside host seen from the inside network.

Inside global address – IP address that represents one or more inside local IP addresses to the outside world. This is the inside host as seen from the outside network.

Outside local address – This is the actual IP address of the destination host in the local network after translation.

Outside global address – This is the outside host as seen from the outside network. It is the IP address of the outside destination host before translation.

Network Address Translation (NAT) Types – There are 3 ways to configure NAT:

Static NAT – In this, a single unregistered (Private) IP address is mapped with a legally registered (Public) IP address i.e one-to-one mapping between local and global addresses. This is generally used for Web hosting. These are not used in organizations as there are many

devices that will need Internet access and to provide Internet access, a public IP address is needed. Suppose, if there are 3000 devices that need access to the Internet, the organization has to buy 3000 public addresses that will be very costly.

Dynamic NAT – In this type of NAT, an unregistered IP address is translated into a registered (Public) IP address from a pool of public IP addresses. If the IP address of the pool is not free, then the packet will be dropped as only a fixed number of private IP addresses can be translated to public addresses.
Suppose, if there is a pool of 2 public IP addresses then only 2 private IP addresses can be translated at a given time. If 3rd private IP address wants to access the Internet then the packet will be dropped therefore many private IP addresses are mapped to a pool of public IP addresses. NAT is used when the number of users who want to access the Internet is fixed. This is also very costly as the organization has to buy many global IP addresses to make a pool.

Port Address Translation (PAT) – This is also known as NAT overload. In this, many local (private) IP addresses can be translated to a single registered IP address. Port numbers are used to distinguish the traffic i.e., which traffic belongs to which IP address. This is most frequently used as it is cost-effective as thousands of users can be connected to the Internet by using only one real global (public)

Advantages of NAT –

NAT conserves legally registered IP addresses.
It provides privacy as the device's IP address, sending and receiving the traffic, will be hidden.
Eliminates address renumbering when a network evolves.

Disadvantage of NAT –

Translation results in switching path delays.
Certain applications will not function while NAT is enabled.
Complicates tunneling protocols such as IPsec.
Also, the router being a network layer device, should not tamper with port numbers (transport layer) but it has to do so because of NAT.

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UNIT –V COMPUTER NETWORKS – SECA1604

UNIT V TRANSPORT LAYER AND APPLICATION LAYER

TRANSPORT LAYER AND APPLICATION LAYER

The transport layer:

The TCP/IP protocol suite does not specify nor require specific networking hardware. The only assumption that the TCP/IP protocol suite makes is that the networking hardware, also called physical network, supports the transmission of an IP datagram. The lowest layer of the TCP/IP protocol suite, the data link layer, is concerned with interfacing to a physical network. Whenever a new network hardware technology emerges all that is needed to run Internet applications over the new hardware is an interface at the data link layer that supports the transmission of an IP datagram packet over the new hardware. This minimal dependence of the TCP/IP protocol suite on the underlying networking hardware contributes significantly to the ability of the Internet protocols to adapt to advances in networking hardware.

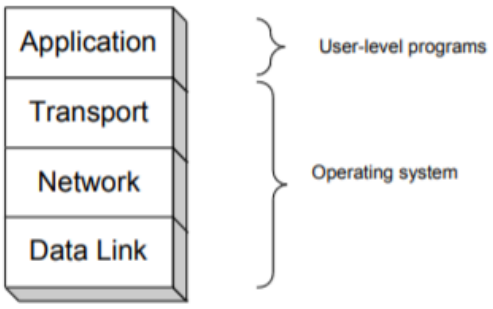

Fig.1. The Internet Protocol Stack

Services provided by Transport layer to the upper layer:

Before looking into the services offered by transport layer , a quick comparison about OSI and TCP/IP framework gives better understanding.

The main similarities between the OSI and TCP/IP models and services include the following:

• They share similar architecture. - Both of the models share a similar architecture. This can be illustrated by the fact that both of them are constructed with layers.

• They share a common application layer.- Both of the models share a common "application layer". However in practice this layer includes different services depending upon each model.

• Both models have comparable transport and network layers.- This can be illustrated by the fact that whatever functions are performed between the presentation and network layer of the

OSI model similar functions are performed at the Transport layer of the TCP/IP model.

• Both models assume that packets are switched.- Basically this means that individual packets may take differing paths in order to reach the same destination

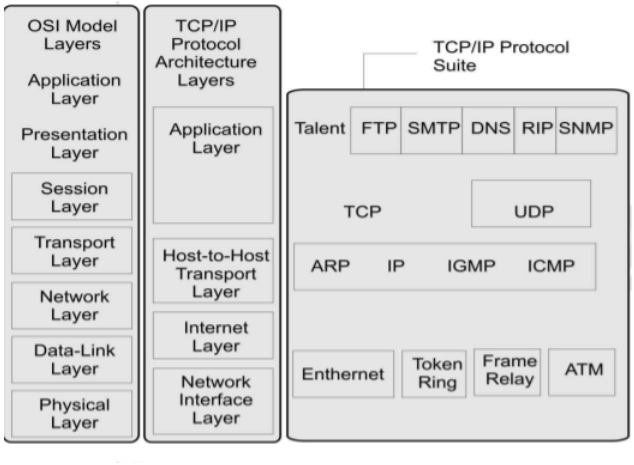

Fig.2. Comparison of OSI and TCP/IP model – Services

The major functions of Transport Layer are:

• It sets up and maintains a Chapter connection between two devices.

• It can provide for the reliable or unreliable delivery of data across the connection.

• It can implement flow control through ready/not ready signals or Windowing to ensure that the sender do not overwhelm the receiver with too many segments.

• It multiplexes the connections, allowing multiple applications to simultaneously send and receive data through port or socket numbers

The Most common Transport Layer Protocols are:

• T.C.P (Transmission Control Protocol)

• U.D.P (User Datagram Protocol)

The transport layer provides transparent transfer of data between end users, providing reliable data transfer services to the upper layers. The transport layer controls the reliability of a given link through flow control, segmentation/de-segmentation, and error control. This layer manages the end-to-end control (for example, determining whether all packets have arrived). It ensures complete data transfer. The Basic Transport Layer Services are:

• Resource Utilization (multiplexing): Multiple applications run on the same machine but use different ports.

• Connection Management (establishing & terminating): The second major task of Transport

Layer is establishing connection between sender & the receiver before data transmission starts & terminating the connection once the data transmission is finished

• Flow Control (Buffering / Windowing): Once the connection has occurred and transfer is in progress, congestion of the data flow can occur at a destination for a variety of reasons. Possible options include: The destination can become overwhelmed if multiple devices are trying to send it data at the same time. The destination can become overwhelmed if the source is sending faster than it can physically receive. The Transport Layer is responsible for providing flow control to alleviate the issue of congestion in the data transfer. Two main methods for flow control include:

Buffering:

This is a form of data flow control regulated by the Transport Layer. It is responsible for ensuring that sufficient buffers (Temporary Memory) are available at the destination for the processing of data and that the data is transmitted at a rate that does not exceed what the buffer can handle.

Windowing:

This is a flow control scheme in which the source computer will monitor and make adjustments to the amount of information sent based on successful.

Reliable Transport:

Transport layer provides reliable transport of data by sending positive acknowledgements back to the sender once the data has reached the receiving side, if the data is lost or is corrupted, a negative acknowledgement is sent.

Elements of transport protocols:

Types of Service. The transport layer also determines the type of service provided to the users from the session layer. ...

Error Control. ...

Flow Control. ...

Connection Establishment/Release

Multiplexing/De multiplexing

Fragmentation and re-assembly

Addressing

Transport protocol similar to data link protocols, both do error control and flow control.

The Internet protocol suite supports a connectionless transport protocol called UDP (User Datagram Protocol). UDP provides a way for applications to send encapsulated IP datagrams

without having to establish a connection.

The User Datagram Protocol (UDP) is a transport layer protocol defined for use with the IP network layer protocol. It is defined by RFC 768 written by John Postel. It provides a best-effort datagram service to an End System (IP host).

UDP transmits segments consisting of an 8-byte header followed by the pay-load. The two ports serve to identify the end-points within the source and destination machines.

When a UDP packet arrives, its payload is handed to the process attached to the destination port.

This attachment occurs when the BIND primitive. Without the port fields, the transport layer would not know what to do with each incoming packet. With them, it delivers the embedded segment to the correct application.

The different issues to be considered are:

(i). The TCP Service Model (ii). The TCP Protocol (iii). The TCP Segment Header (iv). The Connection Management (v). TCP Transmission Policy (vi). TCP Congestion Control (vii). TCP Timer Management

Addressing:

TCP/IP includes an Internet addressing scheme that allows users and applications to identify a specific network or host with which to communicate.

Four levels of addresses are used in an internet employing the TCP/IP protocols: physical address, logical address, port address, and application-specific address.

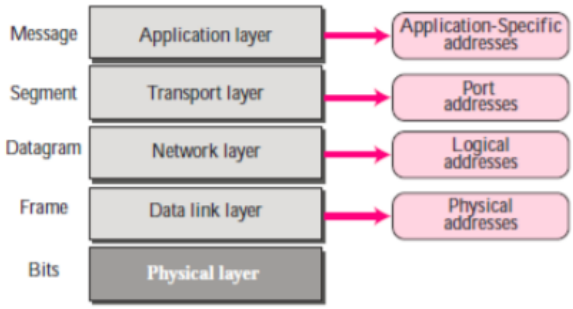

Fig.3. Addressing

Each address is related to a one layer in the TCP/IP architecture, as shown in the above Fig.3.

TCP Connection establishment:

The "three-way handshake" is the procedure used to establish a connection. This procedure normally is initiated by one TCP and responded to by another TCP. The procedure also works if two TCP simultaneously initiate the procedure. When simultaneous attempt occurs, each

TCP receives a "SYN" segment which carries no acknowledgment after it has sent a "SYN". Of course, the arrival of an old duplicate "SYN" segment can potentially make it appear, to the recipient that a simultaneous connection initiation is in progress. Proper use of "reset" segments can disambiguate these cases. The three-way handshake reduces the possibility of false connections. It is the implementation of a trade-off between memory and messages to provide information for this checking.

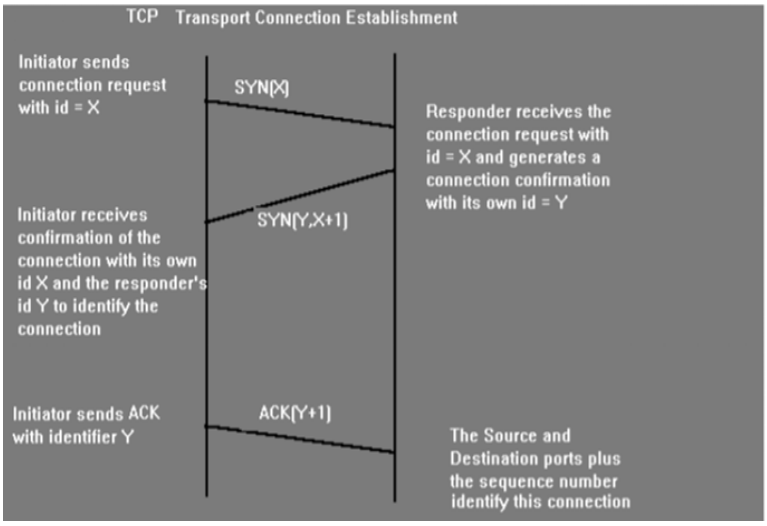

Fig.4. TCP connection Establishment

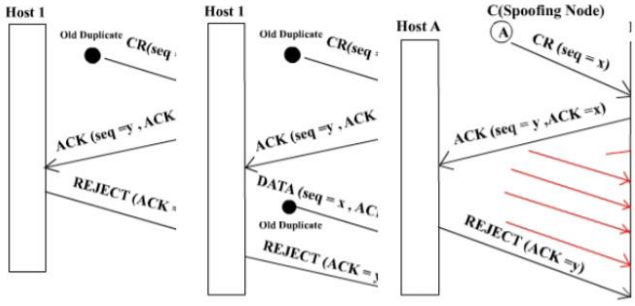

Fig.5. Handshaking

The first two figures show how a three way handshake deals with problems of duplicate/delayed connection requests and duplicate/delayed connection acknowledgements in the network. The third figure highlights the problem of spoofing associated with a two way handshake. Some Conventions 1. The ACK contains 'x+1' if the sequence number received is 'x'. 2. If 'ISN' is the sequence number of the connection packet then 1st data packet has the seq number 'ISN+1' 3. Seq numbers are 32 bit. They are byte seq number(every byte has a seq number).With a packet 1st seq number and length of the packet is sent. 4. Acknowledgements

are cumulative. 5. Acknowledgements have a seq number of their own but with a length 0.So the next data packet have the seq number same as ACK.

TCP Connection Release:

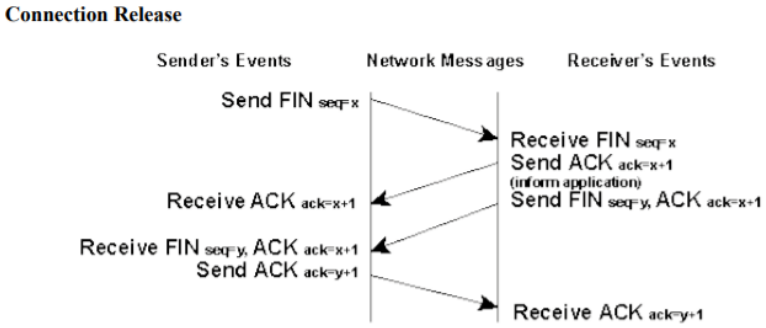

Fig.6. Connection Release

The initiator sends a FIN with the current sequence and acknowledgement number.

• The responder on receiving this informs the application program that it will receive no more data and sends an acknowledgement of the packet. The connection is now closed from one side.

• Now the responder will follow similar steps to close the connection from its side. Once this is done the connection will be fully closed. TCP connection is a duplex connection. That means there is no difference between two sides once the connection is established.

UDP–TCP

UDP -- like its cousin the Transmission Control Protocol (TCP) -- sits directly on top of the base Internet Protocol (IP). In general, UDP implements a fairly "lightweight" layer above the Internet Protocol. It seems at first site that similar service is provided by both UDP and IP, namely transfer of data. But we need UDP for multiplexing/demultiplexing of addresses.

UDP's main purpose is to abstract network traffic in the form of datagrams. A datagram comprises one single "unit" of binary data; the first eight (8) bytes of a datagram contain the header information and the remaining bytes contain the data itself.

UDP Headers

The UDP header consists of four (4) fields of two bytes each:

Source Port	Destination Port
length	checksum

Fig. 7. UDP

UDP port numbers allow different applications to maintain their own "channels" for data; both UDP and TCP use this mechanism to support multiple applications sending and receiving data concurrently. The sending application (that could be a client or a server) sends UDP datagrams through the source port, and the recipient of the packet accepts this datagram through the destination port. Some applications use static port numbers that are reserved for or registered to the application. Other applications use dynamic (unregistered) port numbers. Because the UDP port headers are two bytes long, valid port numbers range from 0 to 65535; by convention, values above 49151 represent dynamic ports.

The datagram size is a simple count of the number of bytes contained in the header and data sections . Because the header length is a fixed size, this field essentially refers to the length of the variable-sized data portion (sometimes called the payload). The maximum size of a datagram varies depending on the operating environment. With a two-byte size field, the theoretical maximum size is 65535 bytes. However, some implementations of UDP restrict the datagram to a smaller number -- sometimes as low as 8192 bytes.

UDP checksums work as a safety feature. The checksum value represents an encoding of the datagram data that is calculated first by the sender and later by the receiver. Should an individual datagram be tampered with (due to a hacker) or get corrupted during transmission (due to line noise, for example), the calculations of the sender and receiver will not match, and the UDP protocol will detect this error. The algorithm is not fool-proof, but it is effective in many cases. In UDP, check summing is optional -- turning it off squeezes a little extra performance from the system -- as opposed to TCP where checksums are mandatory. It should be remembered that check summing is optional only for the sender, not the receiver. If the sender has used checksum then it is mandatory for the receiver to do so.

Usage of the Checksum in UDP is optional. In case the sender does not use it, it sets the checksum field to all 0's. Now if the sender computes the checksum then the recipient must also compute the checksum an set the field accordingly. If the checksum is calculated and turns out to be all 1's then the sender sends all 1's instead of all 0's. This is since in the algorithm for checksum computation used by UDP, a checksum of all 1's if equivalent to a checksum of all 0's. Now the checksum field is unambiguous for the recipient, if it is all 0's then checksum has not been used, in any other case the checksum has to be computed.

TCP:

TCP/IP is the protocol suite upon which all Internet communication is based. Different vendors have developed other networking protocols, but even most network operating systems with their

own protocols, such as Netware, support TCP/IP. It has become the de facto standard.

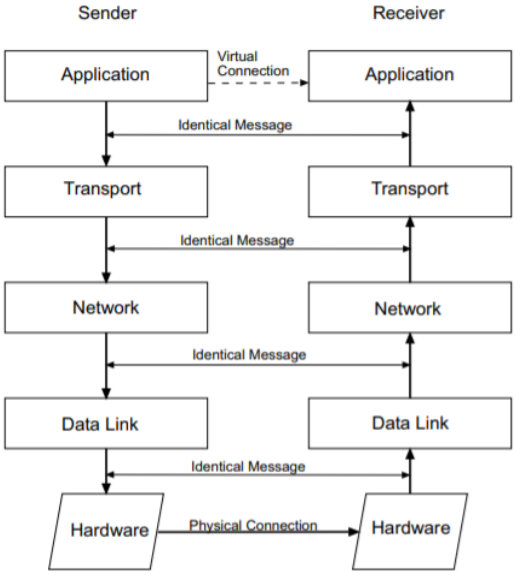

Fig.8. Flow of data between two computers using TCP/IP stacks

Each host or router in the internet must run a protocol stack. The details of the underlying physical connections are hidden by the software. The sending software at each layer communicates with the corresponding layer at the receiving side through information stored in headers. Each layer adds its header to the front of the message from the next higher layer. The header is removed by the corresponding layer on the receiving side.

Internet Protocol (IP) uses the Address Resolution Protocol (ARP), but is shown here at the same layer in the stack as shown in the Fig.9.

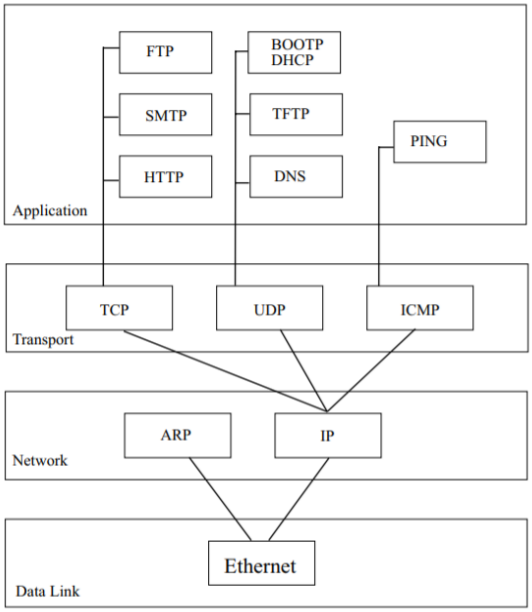

Fig.9. TCP/IP Protocol Flow

Application Layer-Introduction, providing services:

The application layer is the topmost layer of the protocol hierarchy. It is the layer where actual communication is initiated. It uses the services of the transport layer, the network layer, the data link layer, and the physical layer to transfer data to a remote host.

The application layer is closest to the end user.

• Network applications enable users to send and receive data with ease.

• The application layer acts as interface between the applications and the underlying network.

• Application layer protocols help exchange data between programs running on the source and destination hosts.

• The TCP/IP application layer performs the functions of the upper three layers of the OSI model.

• Common application layer protocols include: HTTP, FTP, TFTP, DNS.

Network Virtual terminal: An application layer allows a user to log on to a remote host. To do so, the application creates a software emulation of a terminal at the remote host. The user's computer talks to the software terminal, which in turn, talks to the host. The remote host thinks that it is communicating with one of its own terminals, so it allows the user to log on.

File Transfer, Access, and Management (FTAM): An application allows a user to access files in a remote computer, to retrieve files from a computer and to manage files in a remote computer. FTAM defines a hierarchical virtual file in terms of file structure, file attributes and the kind of operations performed on the files and their attributes.

Addressing: To obtain communication between client and server, there is a need for addressing. When a client made a request to the server, the request contains the server address and its own address. The server response to the client request, the request contains the destination address, i.e., client address. To achieve this kind of addressing, DNS is used.

Mail Services: An application layer provides Email forwarding and storage.

Directory Services: An application contains a distributed database that provides access for global information about various objects and services.

Authentication: It authenticates the sender or receiver's message or both.

Applications layer paradigms:

In this paradigm, communication at the application layer is between two running application programs called processes: a client and a server. A client is a running program that initializes the communication by sending a request; a server is another application program that waits for a request from a client.

Client-Server Paradigm:

The service provider is an application program, called the server process;

• Server process runs continuously, waiting for another application program, called the client process, to make a connection through the Internet and ask for service.

• The server process must be running all the time;

• The client process starts when the client needs to receive service.

• Several traditional services are still using this paradigm, e.g., WWW, HTTP, FTP, SSH, E-mail, and so on. • Problems: – the server should be a powerful computer – there should be a service provider willing to accept the cost and create a powerful server for a specific service.

Peer-to-Peer Paradigm:

There is no need for a server process to be running all the time and waiting for the client processes to connect.

• The responsibility is shared between peers.

• E.g.: Internet telephony, Bit-Torrent, Skype, IPTV

• Advantages: – easily scalable and cost-effective in eliminating the need for expensive servers to be running and maintained all the time

• Challenges: – more difficult to create secure communication between distributed services

• Few instant messaging application use both client-server and P2P

Client server model, Standard client-server application:

The client-server model, or client-server architecture, is a distributed application framework dividing tasks between servers and clients, which either reside in the same system or communicate through a computer network or the Internet.

Client-server model is a distributed application structure that partitions tasks or workloads between the providers of a resource or service, called servers, and service requesters, called clients. Often clients and servers communicate over a computer network on separate hardware, but both client and server may reside in the same system. A server host runs one or more server programs, which share their resources with clients. A client usually does not share any of its resources, but it requests content or service from a server. Clients, therefore, initiate communication sessions with servers, which await incoming requests. Examples of computer applications that use the client-server model are email, network printing, and the

World Wide Web.

The "client-server" characteristic describes the relationship of cooperating programs in an application. The server component provides a function or service to one or many clients, which initiate requests for such services. Servers are classified by the services they provide. For example, a web server serves web pages and a file server serves computer files. A shared resource may be any of the server computer's software and electronic components, from programs and data to processors and storage devices. The sharing of resources of a server constitutes a service.

Clients and servers exchange messages in a request–response messaging pattern. The client sends a request, and the server returns a response. This exchange of messages is an example of inter-process communication. To communicate, the computers must have a common language, and they must follow rules so that both the client and the server know what to expect. The language and rules of communication are defined in a communications protocol. All protocols operate in the application layer. The application layer protocol defines the basic patterns of the dialogue. To formalize the data exchange even further, the server may implement an application programming interface

When a bank customer accesses online banking services with a web browser (the client), the client initiates a request to the bank's web server. The customer's login credentials may be stored in a database, and the webserver accesses the database server as a client. An application server interprets the returned data by applying the bank's business logic and provides the output to the webserver. Finally, the webserver returns the result to the client web browser for display.

In each step of this sequence of client-server message exchanges, a computer processes a request and returns data. This is the request-response messaging pattern. When all the requests are met, the sequence is complete and the web browser presents the data to the customer. This example illustrates a design pattern applicable to the client–server model

E mail - The user agent - Message transfer agent:

Electronic mail is a method of exchanging messages between people using electronic devices. A message transfer agent (MTA) routes a mail message towards its final destination by sending the message to another MTA. A user agent (UA) interacts with an end-user and allows the user to send and receive mail messages. On the Internet, MTAs communicate with each other using the Simple Mail Transfer Protocol (SMTP).

Messages exchanged across networks are passed between mail servers, including any attached data files. These servers also often keep mailboxes for email. Access to this email by end users is typically either via webmail or an email client.

A message transfer agent receives mail from either another MTA, a mail submission agent (MSA), or a mail user agent (MUA). The transmission details are specified by the Simple Mail Transfer Protocol (SMTP). When a recipient mailbox of a message is not hosted locally, the message is relayed, that is, forwarded to another MTA. Every time an MTA receives an email message, it adds a Received trace header field to the top of the header of the message, thereby building a sequential record of MTAs handling the message. The process of choosing a target MTA for the next hop is also described in SMTP, but can usually be overridden by configuring the MTA software with specific routes.

A message transfer agent (MTA) is a software application used within an Internet message handling system (MHS). It is responsible for transferring and routing an electronic mail message from the sender’s computer to the recipient’s computer. The basic platform for an MTA is an exchange system with client/server architecture.

SMTP:

SMTP stands for Simple Mail Transfer Protocol. SMTP is a set of communication guidelines that allow software to transmit an electronic mail over the internet is called Simple Mail Transfer Protocol. It is a program used for sending messages to other computer users based on e-mail addresses.

SMTP or Simple Mail Transfer Protocol is an application that is used to send, receive, and relay outgoing emails between senders and receivers. When an email is sent, it's transferred over the internet from one server to another using SMTP. In simple terms, an SMTP email is

just an email sent using the SMTP server.

The Simple Mail Transfer Protocol (SMTP) is an internet standard communication protocol for electronic mail transmission. Mail servers and other message transfer agents use SMTP to send and receive mail messages.

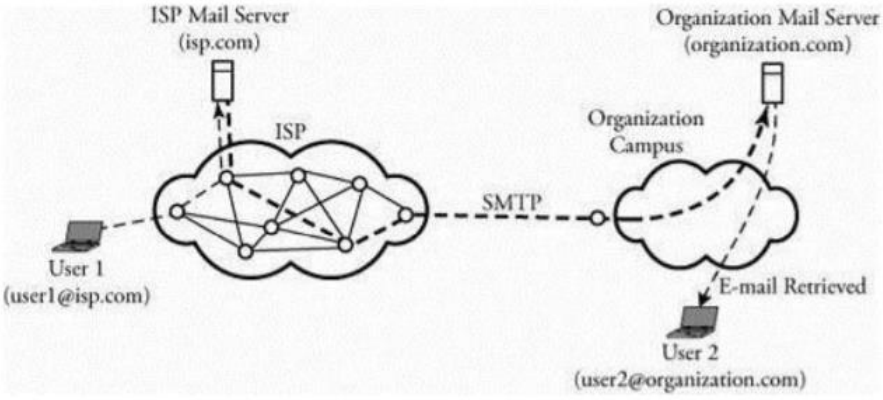

Fig.10. Two users exchanging email through SMTP

In Fig.10, user 1 is in a residential area, has an Internet service provider (ISP), and is sending an e-mail to user 2, working in an organization. Suppose that the mail servers are isp.com and organization.com, respectively. Thus, user 1 and user 2 have e-mail addresses of user1@isp.com and user2@organization.com, respectively. The procedure for an e-mail exchange between user 1 and user 2 is as follows

Begin SMTP Between Two Users:

1. User 1 provides user 2's e-mail address (user2@organization.com) and composes its message.
2. User 1 sends the message to its mail server (isp.com).
3. Server isp.com places the message in its queue.
4. SMTP on user 1's mail server notices the message in the queue and opens a TCP connection with the organization mail server (organization.com).
5. Initial SMTP handshaking takes place between the two servers.
6. The message is sent to organization.com's mail server, using the established TCP connection.
7. User 2's mail server receives the message and then puts it in user 2's mailbox, ready to be retrieved by user 2.

Message access agent: POP and IMAP:

Message Access Agent: POP and IMAP. MAA is a pull protocol; the client must pull

messages i.e., accesses from the server. Currently two message access protocols are available: Post Office Protocol, version 3(POP3) and Internet Mail Access Protocol, version 4 (IMAP4).
POP:

Post Office Protocol, version 3 (POP3) is simple and limited in functionality. The clientPOP3 software is installed on the recipient computer; the server POP3 software is installed on the mail server.

Message Access Agent (MAA) As we know to send and receive a mail two agents, message transfer agent and a message access agent are required. The message transfer agent transfers the message from client computer to the recipient's mail server. Now, it's the work of message access agent to pull the message from the mailbox present on the mail server at recipient's side to the recipient's computer, message access agent are also the final delivery at the recipient side. We have one message transfer agent i.e. SMTP (Simple Mail Transfer Agent), and we have two message access agents POP (Post Office Protocol) and IMAP (Internet Mail Access Protocol).

POP3 — Post Office Protocol version 3 The POP3 has client and server MAA; client MAA software is installed on the recipient computer whereas, the server MAA is installed on the recipient's mail server. To access/read the mail the user has to first download the mail from mailbox on mail server to its computer. To access mail from the mail box present at the mail server the client MAA at recipient computer establishes the connection with the mail server using TCP port 110. For establishing connection client MAA at recipient's computer sends username and password to the mailbox. Then the user is authenticated to retrieve mail messages one by one.

IMAP — The Internet Message Access Protocol One of the main protocols that is used for final delivery is IMAP (Internet Message Access Protocol). To use IMAP, the mail server runs an IMAP server that listens to port 143. The user agent runs an IMAP client. First, the client will start a secure transport if one is to be used (in order to keep the messages and commands confidential), and then log in or otherwise authenticate itself to the server. Once logged in, there are many commands to list folders and messages, fetch messages or even parts of messages, mark messages with flags for later deletion, and organize messages into folders. IMAP has many other features, too. It has the ability to address mail not by message number, but by using attributes (e.g., give me the first message from Alice). Searches can be performed on the server to find the messages that satisfy certain criteria so that only those messages are fetched by the client. IMAP is an improvement over an earlier final delivery protocol, POP3 (Post Office Protocol, version 3).

POP3 is a simpler protocol but supports fewer features and is less secure in typical usage. It is not easy to read mail on multiple computers, plus if the user agent computer breaks, all email may be lost permanently. Nonetheless, POP3 is still in use.

File Transfer Protocol - HTTP - SNMP – VOIP:

File Transfer and FTP File transfer is another computer networking application. It is always essential that files and information geographically distributed over different locations be shared among the members of a working group. In a certain application, files are typically saved in a server. A user then uses a file transfer protocol to access the server and transfer the desired file. Two file transfer protocols are FTP and SCP

File Transfer Protocol (FTP) File Transfer Protocol (FTP) is part of the TCP/IP suite and is very similar to TELNET. Both FTP and TELNET are built on the client/server paradigm, and both allow a user to establish a remote connection. However, TELNET provides a broader access to a user, whereas FTP allows access only to certain files. The essence of this protocol is as follows. Begin File Transfer Protocol 1. A user requests a connection to a remote server. 2. The user waits for an acknowledgment. 3. Once connected, the user must enter a user ID, followed by a password. 4. The connection is established over a TCP session. 5. The desired file is transferred. 6. The user closes the FTP connection. FTP can also run through a Web browser. Secure Copy Protocol (SCP) The Secure Copy Protocol (SCP) is similar to TELNET but is secure. Incorporated in the SCP structure are a number of encryption and authentication features that are similar to those in SSH. Also similar is the exchange of commands between local and remote hosts. SCP commands automatically prompt the user for the password information when it is time to access a remote machine. SCP cannot handle file transfer between machines of significantly different architectures.

Network monitoring is an important tool for pro-active reaction on failures. When moving the voice service to data networks, it is interesting to manage this service using existing tools for data services. The Simple Network Management Protocol (SNMP) is the standard protocol for monitoring TCP/IP networks. SNMP is not only used for network equipment management, but it is also possible to use it to manage services such as hardware features, databases and many others. As this protocol may be used for monitoring services, voice services might be monitored using SNMP.

Voice over Internet Protocol (VoIP), also called IP telephony, is a method and group of technologies for the delivery of voice communications and multimedia sessions over Internet

Protocol (IP) networks, such as the Internet. The terms Internet telephony, broadband telephony, and broadband phone service specifically refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the Internet, rather than via the public switched telephone network (PSTN), also known as plain old telephone service (POTS).