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KarpagamAcademy of Higher Education
(Established Under Section 3 of UGC Act 1956)
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YEAR : 2018-2019 (LATERAL ENTRY)

17CAP302 COMPUTER NETWORKS 4H - 4C

Instruction Hours / week : L: 4 T: 0 P: 0 **Marks:** Internal: **40** External: **60** Total: **100**
End Semester Exam: 3Hours

Scope: This course is to provide students the fundamentals of data communications networks, working knowledge of data transmission concepts, understanding the operation of all seven layers of OSI Model and the protocols used in each layer.

Course Objective: At the end of the course, students will:

- Become familiar with layered communication architectures (OSI and TCP/IP).
- Understand the client/server model and key application layer protocols.
- Learn sockets programming and how to implement client/server programs.
- Understand the concepts of reliable data transfer and how TCP implements these concepts.
- Know the principles of congestion control and trade-offs in fairness and efficiency.
- Learn the principles of routing and the semantics and syntax of IP.

UNIT I

Introduction to computer network

Advantages of networks - structure of the communications network - point-to-point and multidrop circuits - data flow and physical circuits - network topologies - topologies and design goals - Hierarchical topology - horizontal topology (Bus) - star topology - ring topology - mesh topology. The telephone network, switched and non-switched options - fundamentals of communications theory - channel speed and bit rate - voice communications and analog waveforms - bandwidth and the frequency spectrum - connecting the analog and digital worlds - digital worlds - digital signals- the modem, asynchronous and synchronous transmission.

Wide area and local networks, connection oriented and connectionless networks, classification of communications protocols, time division multiple access (TDMA), time division multiplexing (TDM), carrier sense (Collision) systems, token passing, peer-to-peer
priority systems; priority slot, carrier sense (collision free) systems, token passing (priority) systems.

UNIT II

Layered Protocols and the OSI model

Goals of Layered Protocols, network design problems" communication between layers, introduction to standard organizations and the OSI model, standards organizations, Layers of OSI, OSI status.

Local Area Networks

Way LANs?, Primary attributes of a LAN, Broadband and baseband and base LANs, IEEE LAN standards, e1ationship of the 802 standards to the ISO/CCITT model., connection options with LANs, LLC and MAC protocol data units, LAN topologies and protocols., token ring (Priority), token bus and IEEE 802.4, metropolitan area networks (MANs), ANSI fiber distributed data interface.

UNIT III

Network Protocols :TCP , UDP, IP, ICMP, SNMP,RMON
TCP/IP :TCP/IP and internetworking, example of TCP/IP operations, related protocols ports and sockets. The IP address structure, major features of IP, IP datagram. Major IP services. IP source routing, value of the transport layer, TCP, Major features of TCP, passive and active operation, the transmission control block (TCP), route discovery protocols, examples of route discovery protocols, application layer protocols.

UNIT IV

Polling/Selection Protocols

Character and bit protocols, binary synchronous control (BSC) HDLC; HOLC options, HDLC frame format, code transparency and synchronization, HDLC transmission process, HDLC subsets, SDLC; Protocol conversion.

Switching and Routing in Networks

Message switching, packet switching, when and when not to use packet switching, packet routing, packet switching support to circuit switching networks.

The X.25 Network and Supporting Protocols

Features of X.25, Layers of X.25 and the Physical layer, X.25 and the data link layer. Comparision standards to X.25, features of X.25, X.25 channel options, flow control principles, other packet types, X.25 logical channel states, packet formats. internetworking, connectionless mode networks, the frame relay and X.25 stacks.

UNIT V

Network Security: IP Security: Architecture, Authentication header, Encapsulating security payloads, is combining security associations, key management. Web Security: Secure socket layer and transport layer security, secure electronic transaction (SET). System Security: Intruders, Viruses and related threads, firewall design principals, trusted systems.

SUGGESTED READINGS

1. Black, V (1996), "Computer Networks Protocols, Standards and Interfaces", Prentice Hall of India.

2. Stallings, W (1993), "Computer Communication Networks", 4th edition, Prentice Hall of India.
3. Tanneabaum, A.S (1981), "Computer Networks", Prentice Hall of India.
4. B. Forouzan, “Cryptography and Network Security, TMH

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Eachanari (po), Coimbatore-21

DEPARTMENT OF COMPUTER APPLICATIONS
YEAR: 2018-2019 (PG LATERAL ENTRY)

COURSE: COMPUTER NETWORKS [17CAP302]

LECTURE PLAN - UNIT I

S.No.	Lecture Duration (Hr)	Topics to be Covered	Support Materials
1	1	**Introduction to computer network-** Advantages of networks - structure of the communications network - point-to-point and multidrop circuits - data flow and physical circuits.	T1:P(1-5), J1
2	1	Network topologies - topologies and design goals - Hierarchical topology - horizontal topology (Bus) - star topology - ring topology - mesh topology. The telephone network, switched and non-switched options	T1:P(6-12), W1
3	1	Fundamentals of communications theory - channel speed and bit rate - voice communications and analog waveforms - bandwidth and the frequency spectrum	T1:P(13-17)
4	1	Connecting the analog and digital worlds - digital worlds - digital signals- the modem, asynchronous and synchronous transmission	T1:P(18-23)
5	1	Wide area and local networks, connection oriented and connectionless networks - classification of communications protocols	T1:P(30-37)
6	1	Time division multiple access (TDMA), time division multiplexing (TDM)	T1:P(48-54)
7	1	carrier sense (Collision) systems, token passing, peer-to-peer priority systems;	T1:P(48-54)
8	1	Priority slot, carrier sense (collision free) systems, token passing (priority) systems	T1:P(54-55)
9	1	Recapitulation and important questions discussion	
Total no. of periods planned for Unit I : 9			
UNIT II			

S.No.	Lecture Duration (Hr)	Topics to be Covered	Support Materials
1	1	**Layered Protocols and the OSI model,** Goals of Layered Protocols, network design problems" communication between layers,	T1:P(56-63), J1
2	1	Introduction to standard organizations and the OSI model, standards organizations	T1:P(56-63), J1
3	1	Layers of OSI, OSI status	T1:P(64-66), W1

4	1	**Local Area Networks,** Way LANs?, Primary attributes of a LAN, Broadband and baseband and base LANs, IEEE LAN standards	T1:P(117-119)
5	1	Re1ationship of the 802 standards to the ISO/CCITT model, connection options with LANs, LLC and MAC protocol data units	T1:P(120-123)
6	1	LAN topologies and protocols	T1:P(125-126)
7	1	Token ring (Priority), token bus and IEEE 802.4	T1:P(130-136)
8	1	Metropolitan area networks (MANs), ANSI fiber distributed data interface.	T1:P(137-138, 145-147)
9	1	Recapitulation and important questions discussion	
Total no. of periods planned for Unit II : 9			
UNIT III			

S.No.	Lecture Duration (Hr)	Topics to be Covered	Support Materials
1	1	**Network Protocols :** TCP , UDP, IP, ICMP, SNMP,RMON	T1:P(261-263), J1,W1
2	1	**TCP/IP :** TCP/IP and internetworking	T1:P(261-263), J1,W1
3	1	Example of TCP/IP operations	T1:P(261-263), J1,W1
4	1	Related protocols ports and sockets.	T1:P(264-265)
5	1	The IP address structure, major features of IP, IP datagram. Major IP services. IP source routing	T1:P(265-270)
6	1	Value of the transport layer, TCP, Major features of TCP	T1:P(273-274)
7	1	Passive and active operation, the transmission control block (TCB)	T1:P(276-277)
8	1	Route discovery protocols, examples of route discovery protocols	T1:P(280-282)
9	1	Application layer protocols.	T1:P(285)
10	1	Recapitulation and important questions discussion	
Total no. of periods planned for Unit III : 10			
UNIT IV			

S.No.	Lecture Duration (Hr)	Topics to be Covered	Support Materials

1	1	**Polling/Selection Protocols** Character and bit protocols, binary synchronous control (BSC) HDLC; HOLC options, HDLC frame format	T1:P(72-80), W1,J1
2	1	Code transparency and synchronization, HDLC transmission process,	T1:P(81-100)
3	1	HDLC subsets, SDLC, Protocol conversion.	T1:P(81-100)
4	1	**Switching and Routing in Networks** Message switching, packet switching,	T1:P(149-159)
5	1	when and when not to use packet switching	T1:P(149-159)
6	1	Packet routing, packet switching support to circuit switching networks.	T1:P(161-170)
7	1	**The X.25 Network and Supporting Protocols** Features of X.25, Layers of X.25 and the Physical layer, X.25 and the data link layer. companion standards to X.25, features of X.25	T1:P(173-178)
8	1	X.25 channel options, flow control principles, other packet types, X.25 logical channel states, packet formats. internetworking	T1:P(178-225)
9	1	connectionless mode networks, the frame relay and X.25 stacks	T1:P(178-225)
10	1	Recapitulation and important questions discussion	
Total no. of periods planned for Unit IV : 10			
UNIT V			

o.	**Lecture Duration (Hr)**	**Topics to be Covered**	**Support Materials**
1.	1	Network Security: IP Security, Architecture	T4:P(961)
2.	1	Authentication header, Encapsulating security payloads	T4:P(962, 981-989)
3.	1	Combining security associations, key management.	T4:P(981-989)
4.	1	Secure electronic transaction (SET).	T4:P(1008-1013)
5.	1	System Security: Intruders, Viruses and related threads	W1,J1
6.	1	Firewall design principals, trusted systems	T4:P(1021-1023, W1,J1)
7.	1	Recapitulation and Discussion on important questions	
8.	1	Discussion of previous ESE question papers	

9.	1	Discussion of previous ESE question papers	
10.	1	Discussion of previous ESE question papers	
Total no. of periods planned for Unit V : 10			

SUGGESTED READINGS

1. Black, V (1996), "Computer Networks Protocols, Standards and Interfaces", Prentice Hall of India.
2. Stallings, W (1993), "Computer Communication Networks", 4th edition, Prentice Hall of India.
3. Tanneabaum, A.S (1981), "Computer Networks", Prentice Hall of India.
4. B. Forouzan, “Cryptography and Network Security, TMH

WEBSITES

1. W1: www.wikipedia.org/Topology/
2. W2 :www.wikipedia.org/OSI/
3. W3 : www.wikipedia.org/TCP/

JOURNALS

1. https://www.journals.elsevier.com/computer_networks/recent_articles
2. https://www.airccse.org/journal/ijcnc.html

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(Deemed University Established Under Section 3 of UGC Act 1956)
Coimbatore - 641021.
(For the candidates admitted from 2017 onwards)

DEPARTMENT OF CS, CA & IT

SYLLABUS:

UNIT I

Introduction to Computer Network

Advantages of networks - structure of the communications network - point-to-point and multidrop circuits - data flow and physical circuits - network topologies - topologies and design goals - Hierarchical topology - horizontal topology (Bus) - star topology - ring topology - mesh topology. The telephone network, switched and non-switched options - fundamentals of communications theory - channel speed and bit rate - voice communications and analog waveforms - bandwidth and the frequency spectrum - connecting the analog and digital worlds - digital worlds - digital signals- the modem, asynchronous and synchronous transmission.

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1.1 Advantages of networks

Computer networking has become one of the most successful ways of sharing information, where all computers are wirelessly linked together by a common network. Now, businesses and organizations heavily rely on it to get messages and information across to essential channels. Not only has that it benefited establishments, but also individuals, as they also need to share important information every day. But no matter how useful computer networking is, it does not come without drawbacks. Here are its advantages:

1. It enhances communication and availability of information.

Networking, especially with full access to the web, allows ways of communication that would simply be impossible before it was developed. Instant messaging can now allow users to talk in real time and send files to other people wherever they are in the world, which is a huge boon for businesses. Also, it allows access to a vast amount of useful information, including traditional reference materials and timely facts, such as news and current events.

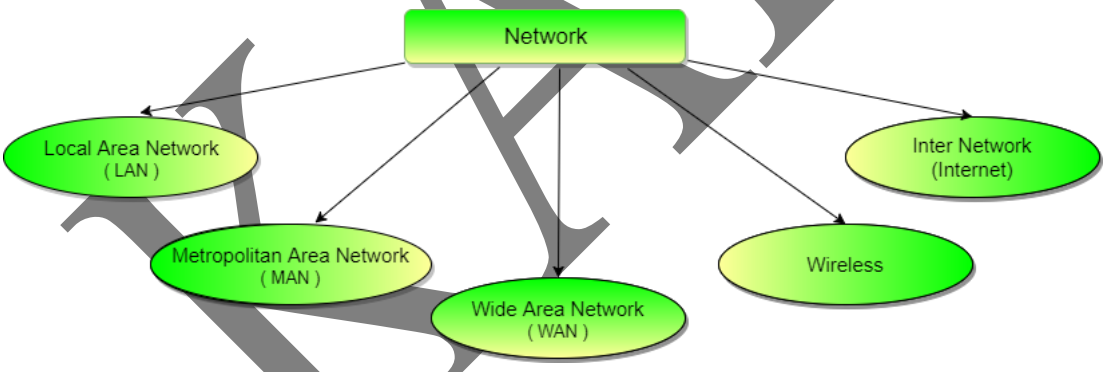
2. It allows for more convenient resource sharing.
This benefit is very important, particularly for larger companies that really need to produce huge numbers of resources to be shared to all the people. Since the technology involves computer-based work, it is assured that the resources they wanted to get across would be completely shared by connecting to a computer network which their audience is also using.

3. It makes file sharing easier.
Computer networking allows easier accessibility for people to share their files, which greatly helps them with saving more time and effort, since they could do file sharing more accordingly and effectively.

4. It is highly flexible.
This technology is known to be very flexible, as it gives users the opportunity to explore everything about essential things, such as software without affecting their functionality. Plus, people will have the accessibility to all information they need to get and share.

1.2 Structure of the communications network

Types of Communication Networks

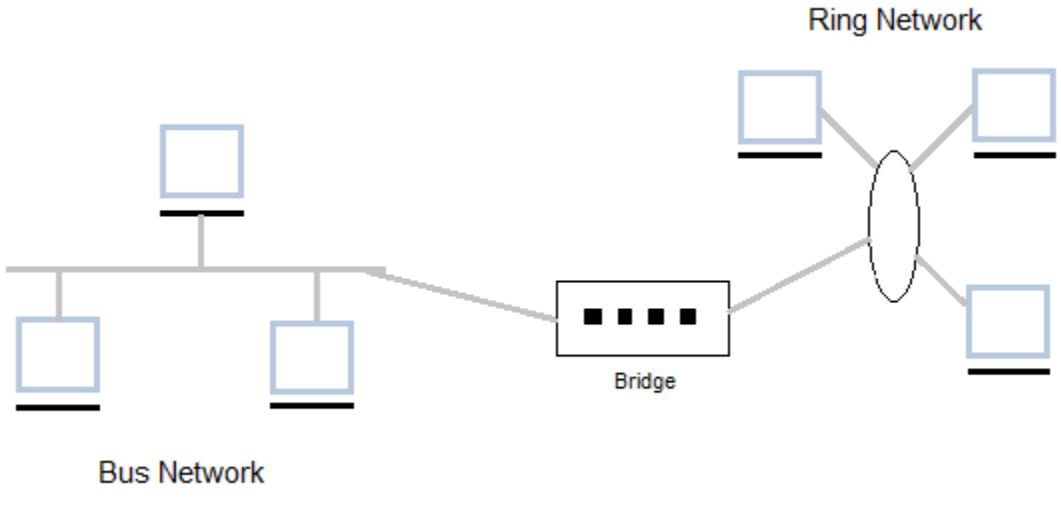


Local Area Network (LAN)

It is also called LAN and designed for small physical areas such as an office, group of buildings or a factory. LANs are used widely as it is easy to design and to troubleshoot. Personal computers and workstations are connected to each other through LANs. We can use different types of topologies through LAN, these are Star, Ring, Bus, Tree etc.

LAN can be a simple network like connecting two computers, to share files and network among each other while it can also be as complex as interconnecting an entire building.

LAN networks are also widely used to share resources like printers, shared hard-drive etc.



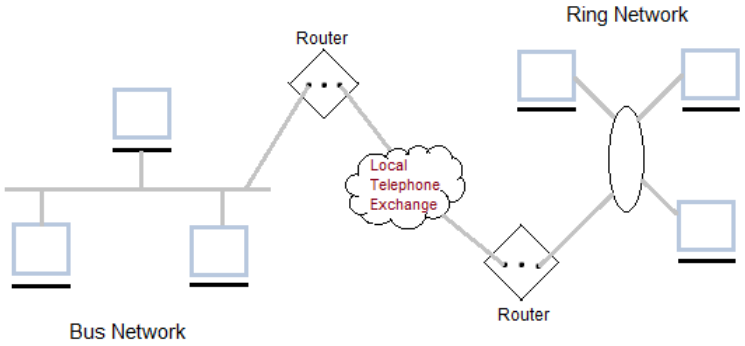
(Different Topologies interconnected in a Local Area Network)

Applications of LAN

- One of the computer in a network can become a server serving all the remaining computers called clients. Software can be stored on the server and it can be used by the remaining clients.
- Connecting Locally all the workstations in a building to let them communicate with each other locally without any internet access.
- Sharing common resources like printers etc are some common applications of LAN.

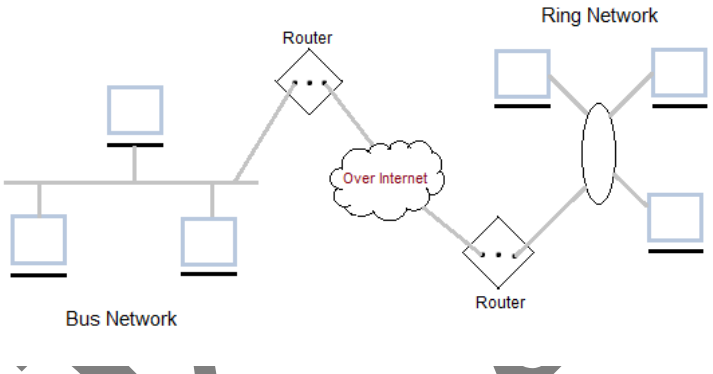
Metropolitan Area Network (MAN)

It is basically a bigger version of LAN. It is also called MAN and uses the similar technology as LAN. It is designed to extend over the entire city. It can be means to connecting a number of LANs into a larger network or it can be a single cable. It is mainly hold and operated by single private company or a public company.



Wide Area Network (WAN)

It is also called WAN. WAN can be private or it can be public leased network. It is used for the network that covers large distance such as cover states of a country. It is not easy to design and maintain. Communication medium used by WAN are PSTN or Satellite links. WAN operates on low data rates.



Wireless Network

Digital wireless communication is not a new idea. Earlier, **Morse code** was used to implement wireless networks. Modern digital wireless systems have better performance, but the basic idea is the same.

Wireless Networks can be divided into three main categories:

1. **System interconnection**
2. **Wireless LANs**
3. **Wireless WANs**

System Interconnection

System interconnection is all about interconnecting the components of a computer using **short-range radio**. Some companies got together to design a short-range wireless network called **Bluetooth** to connect various components such as monitor, keyboard, mouse and printer, to the main unit, without wires. Bluetooth also allows digital cameras, headsets, scanners and other devices to connect to a computer by merely being brought within range.

In simplest form, system interconnection networks use the master-slave concept. The system unit is normally the **master**, talking to the mouse, keyboard, etc. as **slaves**.

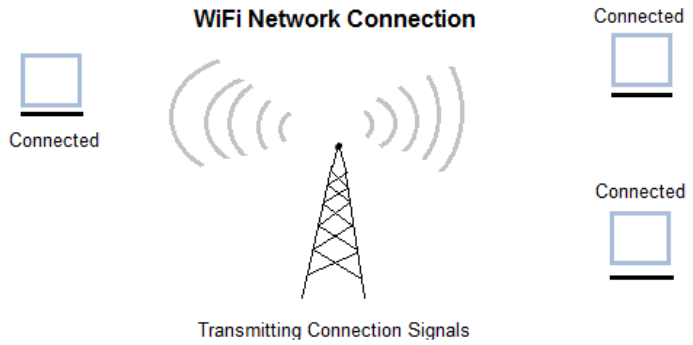
Wireless LANs

These are the systems in which every computer has a **radio modem** and **antenna** with which it can communicate with other systems. Wireless LANs are becoming increasingly common in small offices and homes, where installing **Ethernet** is considered too much trouble. There is a standard for wireless LANs called **IEEE 802.11**, which most systems implement and which is becoming very widespread.

Wireless WANs

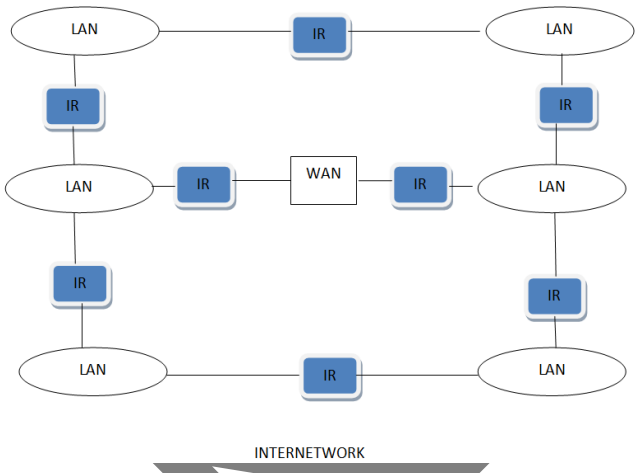
The radio network used for cellular telephones is an example of a low-bandwidth wireless WAN. This system has already gone through three generations.

- The first generation was analog and for voice only.
- The second generation was digital and for voice only.
- The third generation is digital and is for both voice and data.



Inter Network

Inter Network or Internet is a combination of two or more networks. Inter network can be formed by joining two or more individual networks by means of various devices such as routers, gateways and bridges.



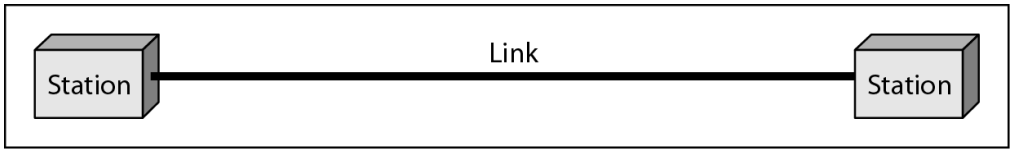
INTERNETWORK

1.3 Point-to-point and multidrop circuits

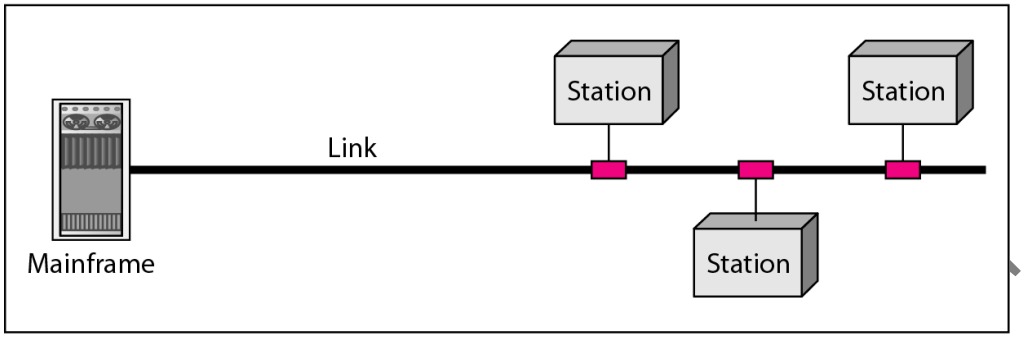
Physical Structures

Types of connections:

1. Point-to-Point
2. Multipoint



a. Point-to-point



b. Multipoint

a) **Point-to-Point**

A point-to-point connection provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links, are also possible. When you change television channels by infrared remote control, you are establishing a point-to-point connection between the remote control and the television's control system.

b) **Multipoint**

A multipoint (also called multi drop) connection is one in which more than two specific devices share a single link. In a multipoint environment, the capacity of the channel is shared, either spatially or temporally. If several devices can use the link simultaneously, it is a *spatially shared* connection. If users must take turns, it is a *timeshared* connection.

1.4 Data flow and physical circuits

Data Flow

Circuits can be designed to permit data to flow in one direction or in both directions. Actually, there are three ways to transmit: simplex, half-duplex, and full-duplex.

Simplex is one-way transmission, such as that with radios and TVs.

Half-duplex is two-way transmission, but you can transmit in only one direction at a time. A half-duplex communication link is similar to a walkie-talkie link; only one computer can transmit at a time.

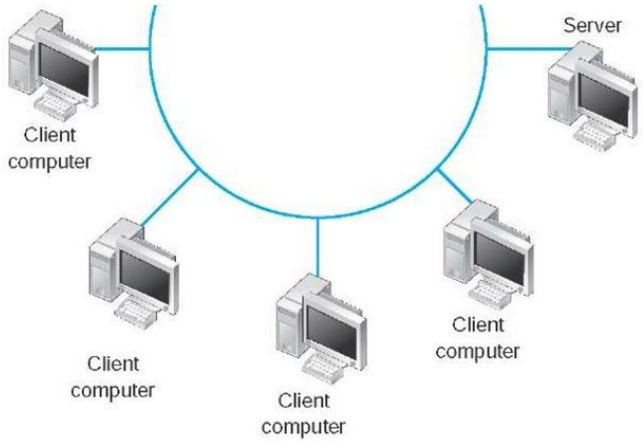


Figure 3.2 Multipoint circuit

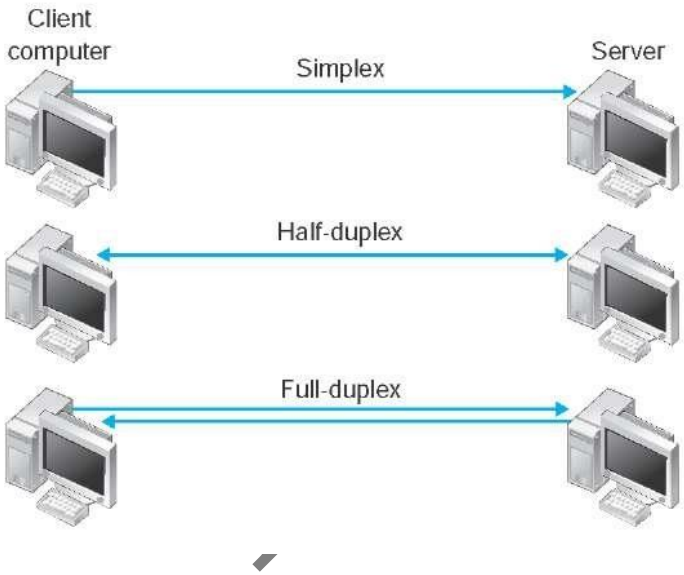


Figure 3.3 Simplex, half-duplex, and full-duplex transmissions

Computers use control signals to negotiate which will send and which will receive data. The amount of time half-duplex communication takes to switch between sending and receiving is called turnaround time (also called retrain time or reclocking time). The turnaround time for a

specific circuit can be obtained from its technical specifications (often between 20 and 50 milliseconds). Europeans sometimes use the term simplex circuit to mean a half-duplex circuit.

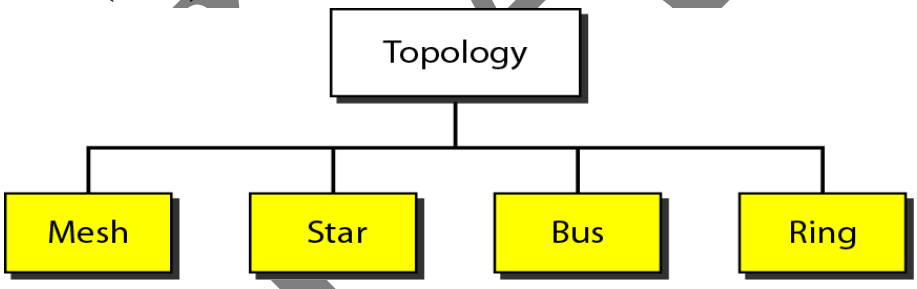
With full-duplex transmission, you can transmit in both directions simultaneously, with no turnaround time.

How do you choose which data flow method to use? Obviously, one factor is the application. If data always need to flow only in one direction (e.g., from a remote sensor to a host computer), then simplex is probably the best choice. In most cases, however, data must flow in both directions.

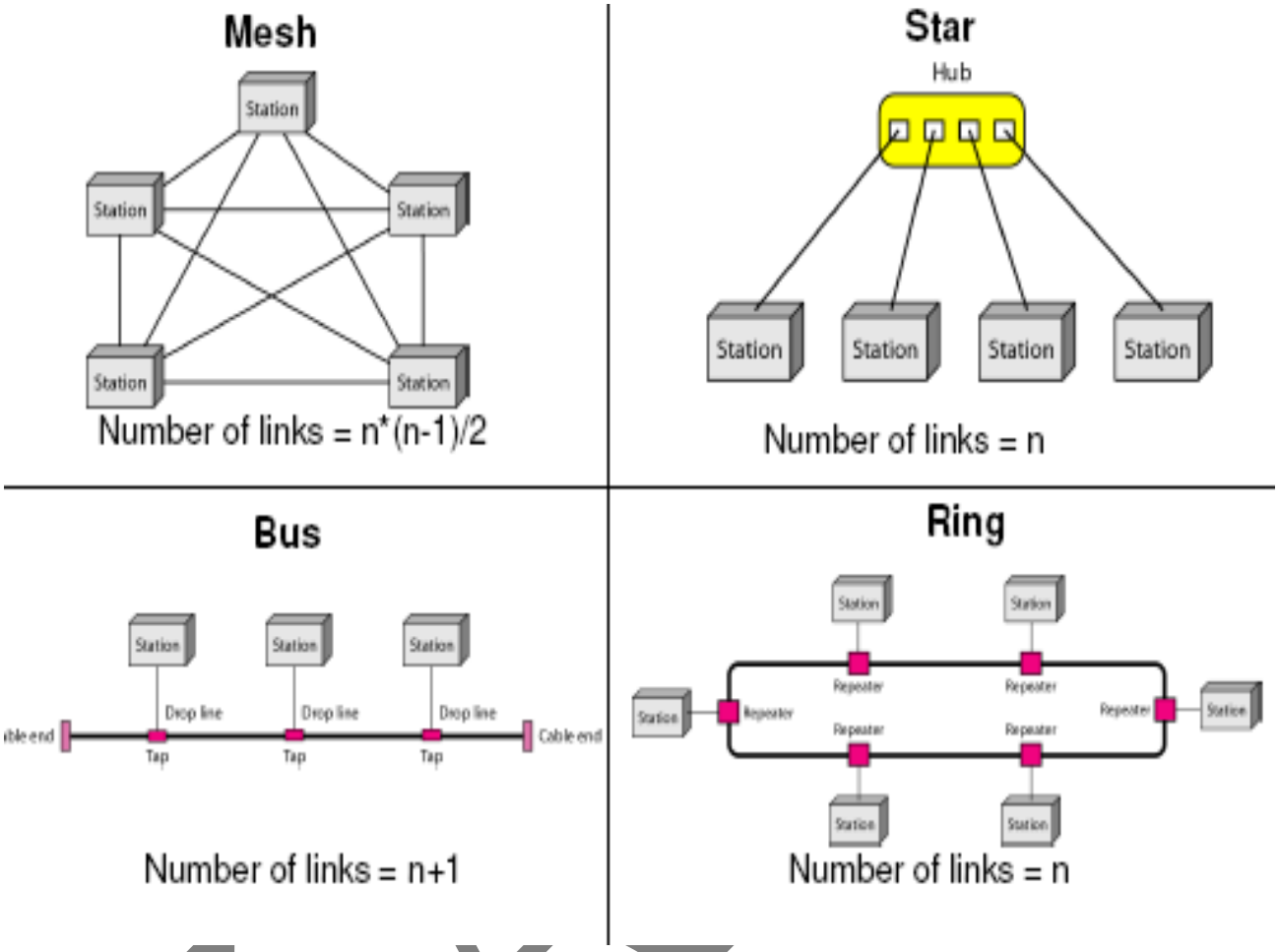
The initial temptation is to presume that a full-duplex channel is best; however, each circuit has only so much capacity to carry data. Creating a full-duplex circuit means that the available capacity in the circuit is divided—half in one direction and half in the other. In some cases, it makes more sense to build a set of simplex circuits in the same way a set of one-way streets can speed traffic. In other cases, a half-duplex circuit may work best. For example, terminals connected to mainframes often transmit data to the host, wait for a reply, transmit more data, and so on, in a turn-taking process; usually, traffic does not need to flow in both directions simultaneously. Such a traffic pattern is ideally suited to half-duplex circuits.

1.5 Network topologies

Network topology is the geometric representation of the relationship of all the links and linking devices (nodes)



Topology categories



Mesh Topology: In a mesh topology, every device has a dedicated point-to-point link to every other device. The term dedicated means that the link carries traffic only between the two devices it connects. To find the number of physical links in a fully connected mesh network with n nodes, we first consider that each node must be connected to every other node. Node 1 must be connected to n - I nodes, node 2 must be connected to n – 1 nodes, and finally node n must be connected to n - 1 nodes. We need n(n - 1) physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need ***n(n* -1) /2** duplex-mode links.

Advantages of Mesh Topology

A dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices.

A mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system.

There is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it. Physical boundaries prevent other users from gaining access to messages.

Finally, point-to-point links make fault identification and fault isolation easy. Traffic can be routed to avoid links with suspected problems. This facility enables the network manager to discover the precise location of the fault and aids in finding its cause and solution.

Disadvantages of Mesh Topology

The main disadvantages of a mesh are related to the amount of cabling and the number of I/O ports required.

- First, because every device must be connected to every other device, installation and reconnection are difficult.
- Second, the sheer bulk of the wiring can be greater than the available space (in walls, ceilings, or floors) can accommodate.
- Finally, the hardware required to connect each link (I/O ports and cable) can be prohibitively expensive.
- For these reasons a mesh topology is usually implemented in a limited fashion, for example, as a backbone connecting the main computers of a hybrid network that can include several other topologies.

One practical example of a mesh topology is the connection of telephone regional offices in which each regional office needs to be connected to every other regional office.

Star Topology

In a star topology, each device has a dedicated point-to-point link only to a central controller, usually called a hub. The devices are not directly linked to one another. Unlike a mesh topology, a star topology does not allow direct traffic between devices. The controller acts as an exchange: If one device wants to send data to another, it sends the data to the controller, which then relays the data to the other connected device.

Advantages of Star Topology

- A star topology is less expensive than a mesh topology. In a star, each device needs only one link and one I/O port to connect it to any number of others which also makes it easy to install and reconfigure.
- Other advantages include robustness. If one link fails, only that link is affected. All other links remain active. This factor also lends itself to easy fault identification and fault isolation.
- As long as the hub is working, it can be used to monitor link problems and bypass defective links.

Disadvantages of Star Topology

- One big disadvantage of a star topology is the dependency of the whole topology on one single point, the hub. If the hub goes down, the whole system is dead.
- Although a star requires far less cable than a mesh, each node must be linked to a central hub. For this reason, often more cabling is required in a star than in some other topologies (such as ring or bus).

The star topology is used in local-area networks (LANs), High-speed LANs often use a star topology with a central hub.

Bus Topology

The preceding examples all describe point-to-point connections. A **bus topology,** on the other hand, is multipoint. One long cable acts as a **backbone** to link all the devices in a network

Nodes are connected to the bus cable by drop lines and taps. A drop line is a connection running between the device and the main cable. A tap is a connector that either splices into the main cable or punctures the sheathing of a cable to create a contact with the metallic core. As a signal travels along the backbone, some of its energy is transformed into heat. Therefore, it becomes weaker and weaker as it travels farther and farther. For this reason there is a limit on the number of taps a bus can support and on the distance between those taps.

Advantages of Bus Topology

- Advantages of a bus topology include ease of installation. Backbone cable can be laid along the most efficient path, then connected to the nodes by drop lines of various lengths. In this way, a bus uses less cabling than mesh or star topologies.
- In a star, for example, four network devices in the same room require four lengths of cable reaching all the way to the hub. In a bus, this redundancy is eliminated. Only the backbone cable stretches through the entire facility. Each drop line has to reach only as far as the nearest point on the backbone.

Disadvantages

- Disadvantages include difficult reconnection and fault isolation.
- A bus is usually designed to be optimally efficient at installation. It can therefore be difficult to add new devices.
- Signal reflection at the taps can cause degradation in quality. This degradation can be controlled by limiting the number and spacing of devices connected to a given length of cable.
- Adding new devices may therefore require modification or replacement of the backbone.

- In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem.

Bus topology was the one of the first topologies used in the design of early local area networks. Ethernet LANs can use a bus topology, but they are less popular now.

Ring Topology

Ring Topology In a ring topology, each device has a dedicated point-to-point connection with only the two devices on either side of it. A signal is passed along the ring in one direction, from 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.

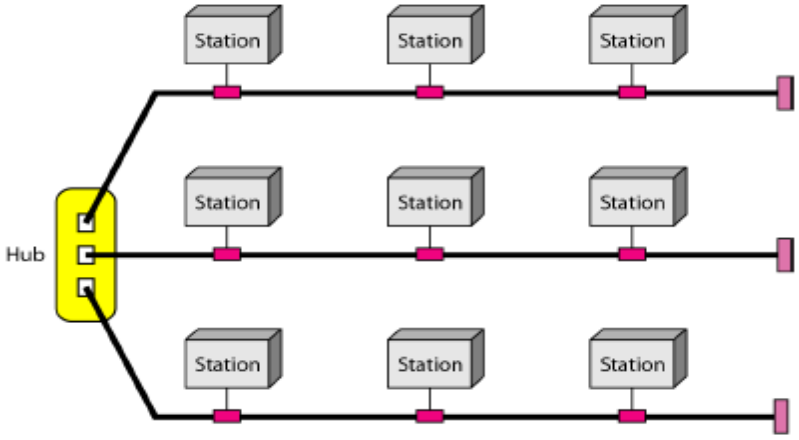
A ring is relatively easy to install and reconfigure. Each device is linked to only its immediate neighbors (either physically or logically). To add or delete a device requires changing only two connections. The only constraints are media and traffic considerations (maximum ring length and number of devices). In addition, fault isolation is simplified. Generally in a ring, a signal is circulating at all times. If one device does not receive a signal within a specified period, it can issue an alarm. The alarm alerts the network operator to the problem and its location. However, unidirectional traffic can be a disadvantage. In a simple ring, a break in the ring (such as a disabled station) can disable the entire network. This weakness can be solved by using a dual ring or a switch capable of closing off the break.

Ring topology was prevalent when IBM introduced its local-area network Token Ring. Today, the need for higher-speed LANs has made this topology less popular.

A hybrid topology

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

A star backbone with three bus networks



1.6 The telephone network, switched and non-switched options

The Public Switched Telephone Network (PSTN) is the global collection of interconnects originally designed to support circuit-switched voice communication. The PSTN provides traditional Plain Old Telephone Service (POTS) - also known as landline phone service - to residences and many other establishments. Parts of the PSTN are also utilized for Internet connectivity services including Digital Subscriber Line (DSL) and Voice over Internet Protocol (VoIP).

PSTN is one of the foundation technologies of telephony - electronic voice communications. While the original forms of telephony including PSTN all relied on analog signaling, modern telephony technologies employ digital signaling, work with digital data, and also support Internet connectivity. The rollout of Internet telephony allows both voice and data to share the same networks, a convergence that the worldwide telecommunications industry is moving toward (for largely financial reasons). A key challenge in Internet telephony is to achieve the same extremely high reliability and quality levels that traditional telephone systems achieved.

History of PSTN Technology

Telephone networks were expanded worldwide during the 1900s as telephones became a routine fixture in homes. Older telephone networks used analog signaling but were gradually upgraded to use digital infrastructure. Most people associate the PSTN with the copper wiring found in many homes although modern PSTN infrastructure also uses fiber optic cables and leaves copper only for the so-called "last mile" of wiring between home and the telecommunication provider's facilitates.The PSTN utilizes the SS7 signaling protocol.

Household PSTN telephones are plugged into wall jacks installed in homes using telephone cords with RJ11 connectors. Residences don't always have jacks in all the right locations, but homeowners can install their own telephone jacks with some basic knowledge of electrical wiring.

One PSTN link supports 64 kilobits per second (Kbps) of bandwidth for data.

The PSTN phone line can be used with traditional dial-up network modems for connecting a computer to the Internet. In the early days of the World Wide Web (WWW), this was the primary form of home Internet access but was made obsolete by broadband Internet services. Dial-up Internet connections supported up 56 Kbps.

PSTN vs. ISDN

Integrated Services Digital Network (ISDN) was developed as an alternative to PSTN that provides both telephone service and also digital data support. ISDN gained popularity in larger businesses due to its ability to support large numbers of phones with low installation costs. It was also offered to consumers as an alternative form of Internet access supporting 128 Kbps.

PSTN vs. VoIP

Voice over Internet Protocol (VoIP), sometimes also called IP telephony, was designed to replace the circuit-switched phone services of both PSTN and ISDN with a packet switched system based on Internet Protocol (IP). The first generations of VoIP services suffered from reliability and sound quality issues but have gradually improved over time.

1.7 Connecting the analog and digital worlds

Analog and Digital

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.

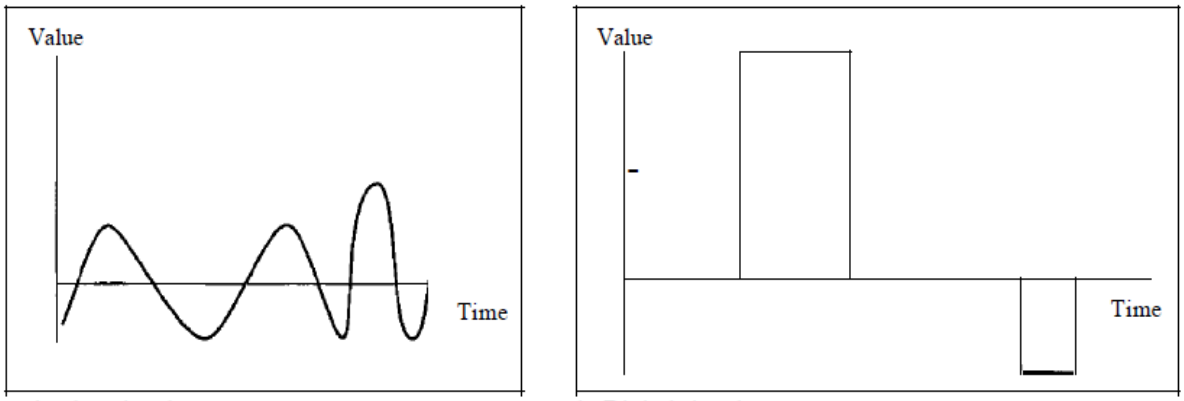
Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal.

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.

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. The following figure 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.



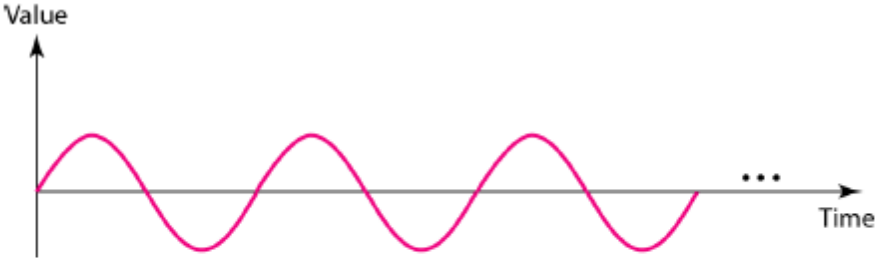
a. Analog signal b. Digital signal

Periodic and Non-periodic Signals

A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a cycle. A non-periodic signal changes without exhibiting a pattern or cycle that repeats over time. Both analog and digital signals can be periodic or non-periodic

PERIODIC ANALOG 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 A sine wave



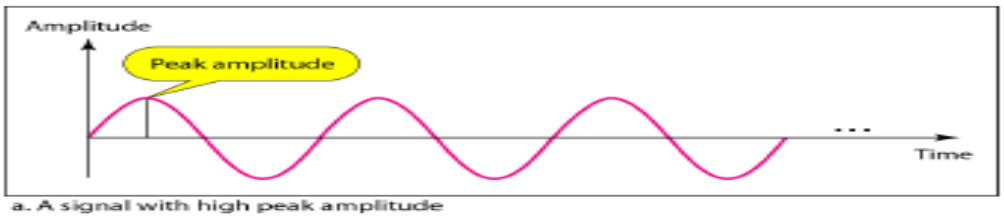
We can mathematically describe the sine wave as

$$s(t) = A\ sin(2\pi ft + \varphi)$$

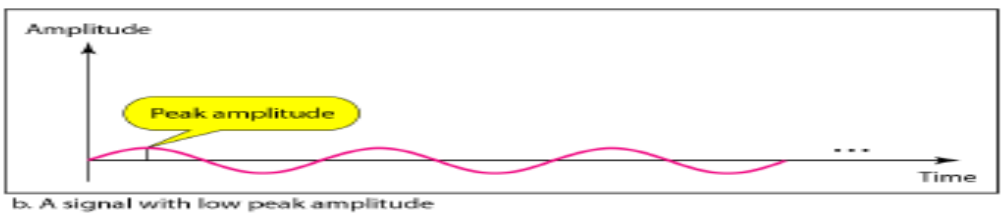
where, s is the instantaneous amplitude
A is the peak amplitude
f is the frequency
φ is the phase
t is the time
π is a constant (~3.14159)

Two signals

Same phase and frequency, but different amplitudes



a. A signal with high peak amplitude



b. A signal with low peak amplitude

Period and frequency

Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle.

• Denoted by ***T***, measured in seconds.

Frequency refers to the number of periods in one second

• Denoted by ***f***, measured in Hertz (Hz)

Note

Frequency and period are the inverse of each other.

$$f = \frac{1}{T} \quad \text{and} \quad T = \frac{1}{f}$$

Units of period and frequency

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

More about frequency

- **Frequency is the rate of change with respect to time.**
- **Change in a short span of time means high frequency.**
- **Change over a long span of time means low frequency.**

Two extremes

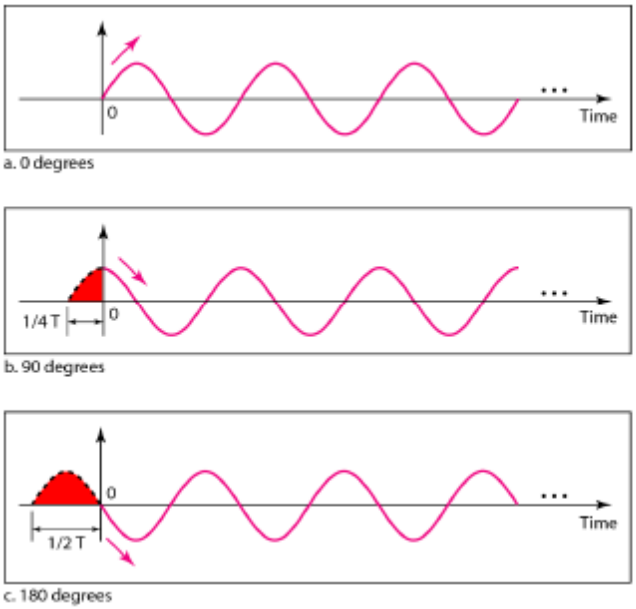
- **If a signal does not change at all, its frequency is zero.**
- **If a signal changes instantaneously, its frequency is infinite.**

Phase

Phase describes the position of the waveform relative to time 0.

Three sine waves

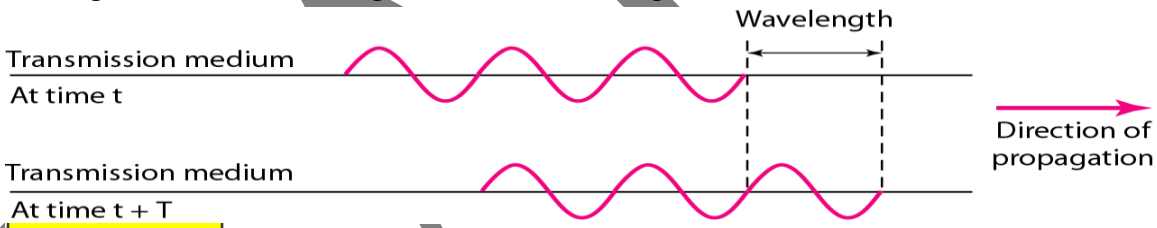
Same amplitude and frequency, but different phases



Wavelength and period

Wavelength is another characteristic of a signal traveling through a transmission medium.

- The wavelength depends on both the frequency and the medium.
- The wavelength is the distance a signal can travel in one period.



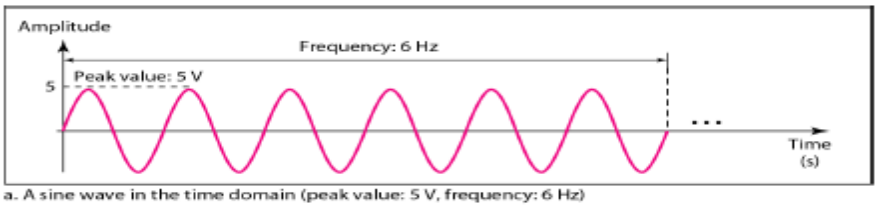
$$\lambda = c/f$$

where, λ is the wavelength

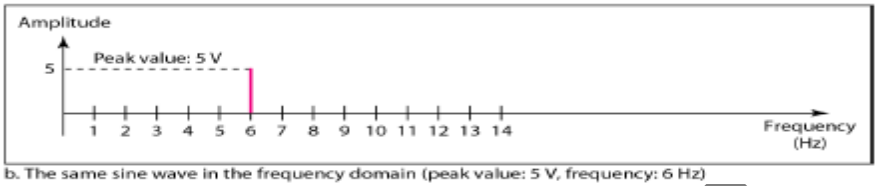
cis the speed of light (~3x108 m/s)

fis the frequency

Time-domain and frequency-domain plots of a sine wave



a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)

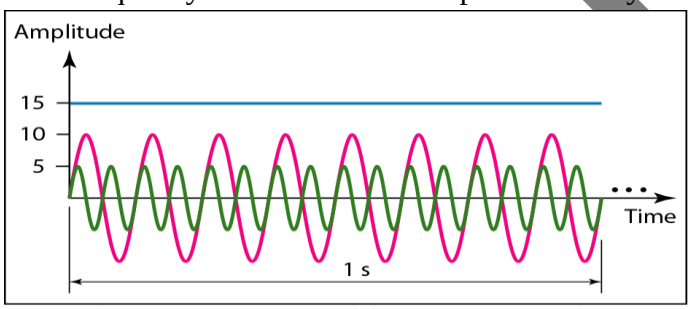


b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

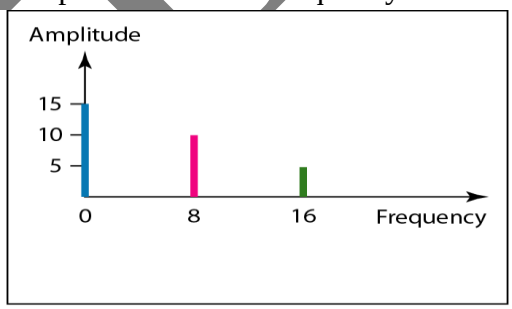
A complete sine wave in the time domain can be represented by one single spike in the frequency domain.

Example

The frequency domain is more compact and useful when we are dealing with more than one sine wave. For example, the following figure shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.



a. Time-domain representation of three sine waves with frequencies 0, 8, and 16



b. Frequency-domain representation of the same three signals

Composite signals

A single-frequency sine wave is not useful in data communications; we need to send a composite signal, a signal made of many simple sine waves**.**

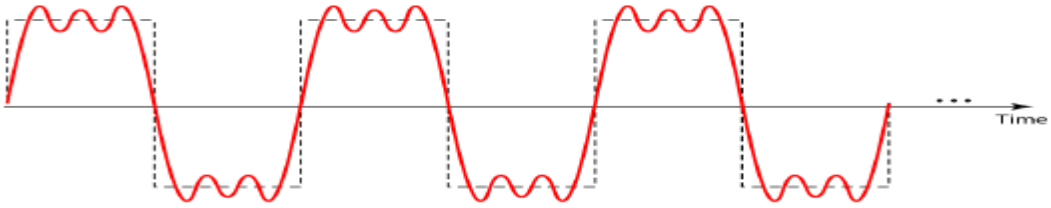
We can use a mathematical technique called **Fourier analysis** to show that any **periodic** signal is made up of an infinite series of sinusoidal frequency components.

If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies; if the composite signal is nonperiodic, the decomposition gives a combination of sine waves with continuous frequencies.

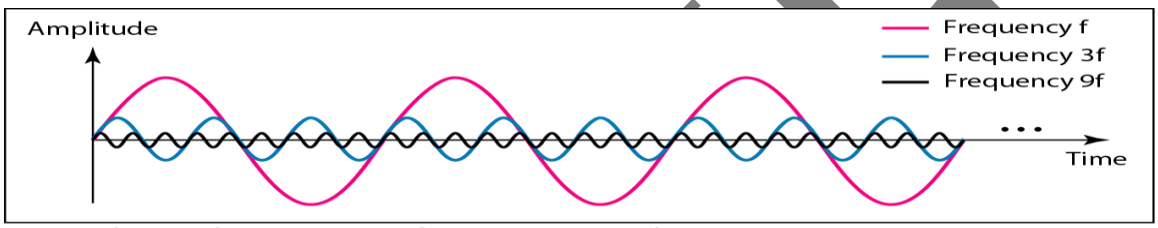
Example

The figure shows a periodic composite signal with frequency ***f***. This type of signal is not typical of those found in data communications. We can consider it to be three alarm systems, each with

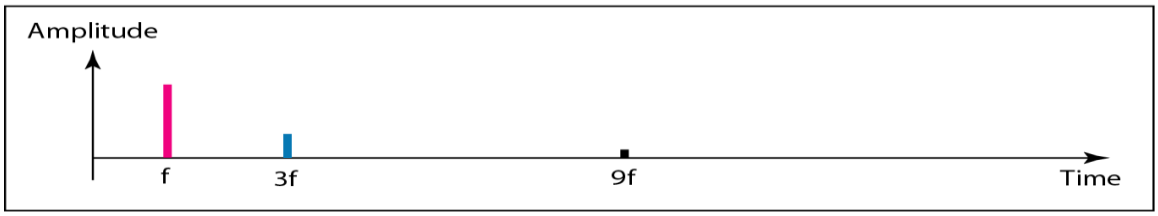
a different frequency. The analysis of this signal can give us a good understanding of how to decompose signals.



Decomposition of a composite periodic signal in the time and frequency domains



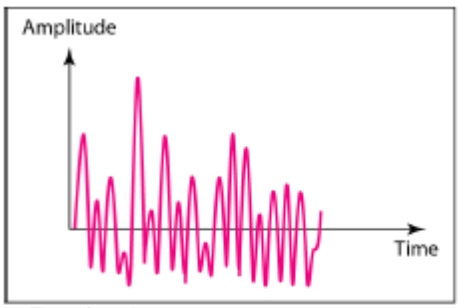
a. Time-domain decomposition of a composite signal



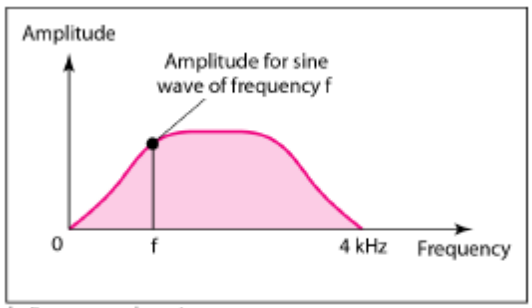
b. Frequency-domain decomposition of the composite signal

Example

The figure shows a nonperiodic composite signal. It can be the signal created by a microphone or a telephone set when a word or two is pronounced. In this case, the composite signal cannot be periodic, because that implies that we are repeating the same word or words with exactly the same tone.



a. Time domain

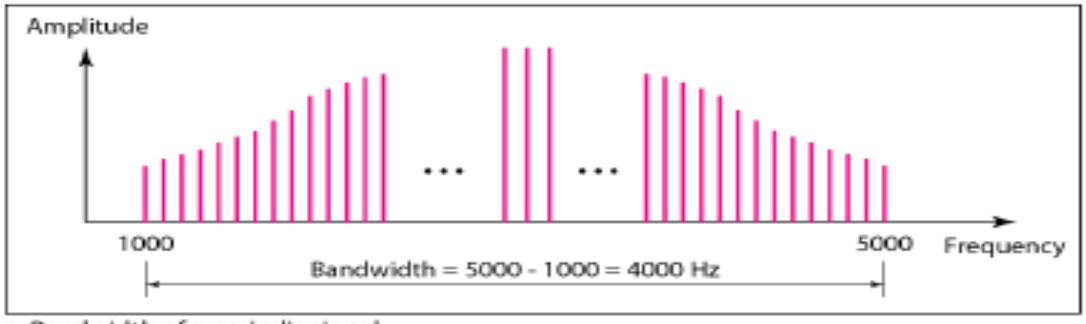


b. Frequency domain

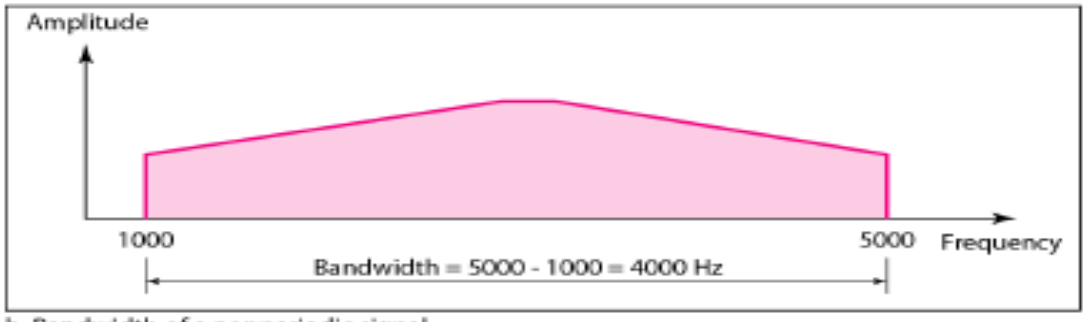
1.8 Bandwidth

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

The bandwidth of periodic and non periodic composite signals



a. Bandwidth of a periodic signal



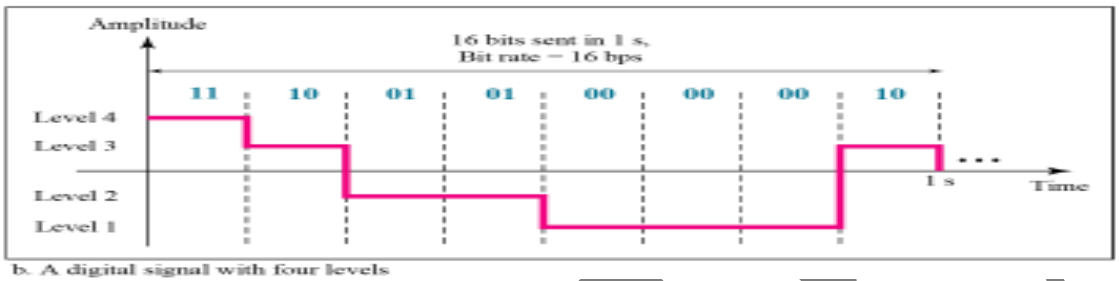
b. Bandwidth of a nonperiodic signal

1.9 DIGITAL SIGNALS

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

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

a. A digital signal with two levels

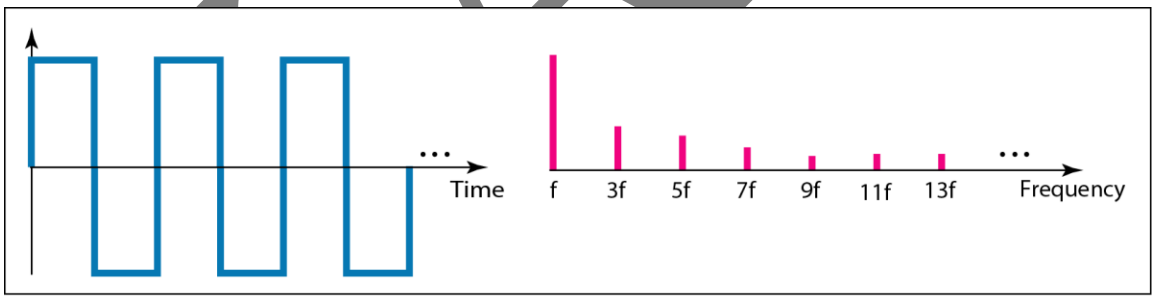


b. A digital signal with four levels

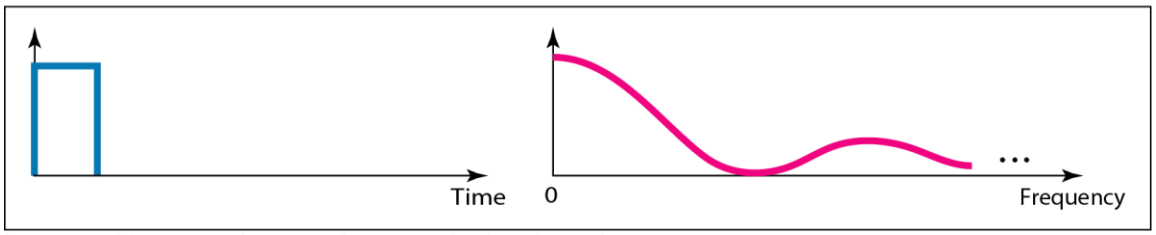
Bit Rate

Most digital signals are nonperiodic, and thus period and frequency are not appropriate characteristics. Another *term-bit rate* (instead *ofjrequency)-is* used to describe digital signals. The bit rate is the number of bits sent in Is, expressed in bits per second (bps). Figure 3.16 shows the bit rate for two signals.

The time and frequency domains of periodic and nonperiodic digital Signals

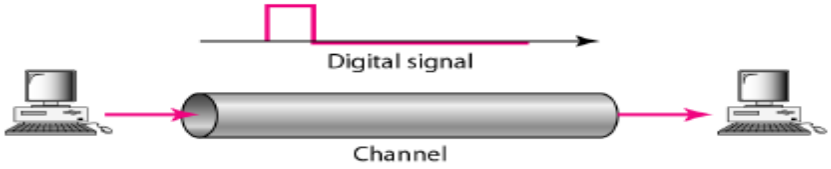


a. Time and frequency domains of periodic digital signal



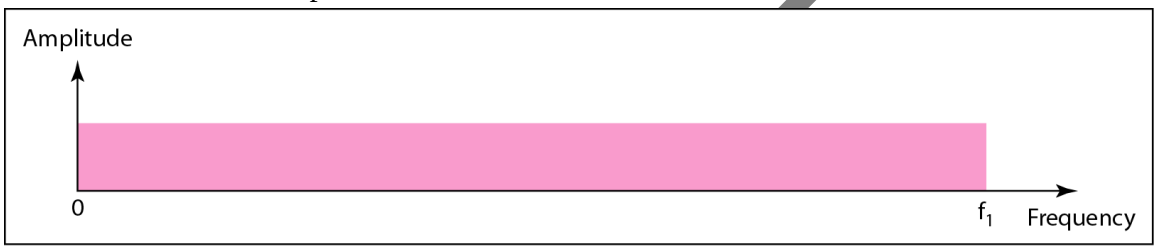
b. Time and frequency domains of nonperiodic digital signal

Baseband transmission



A digital signal is a composite analog signal with an infinite bandwidth.

Bandwidths of two low-pass channels

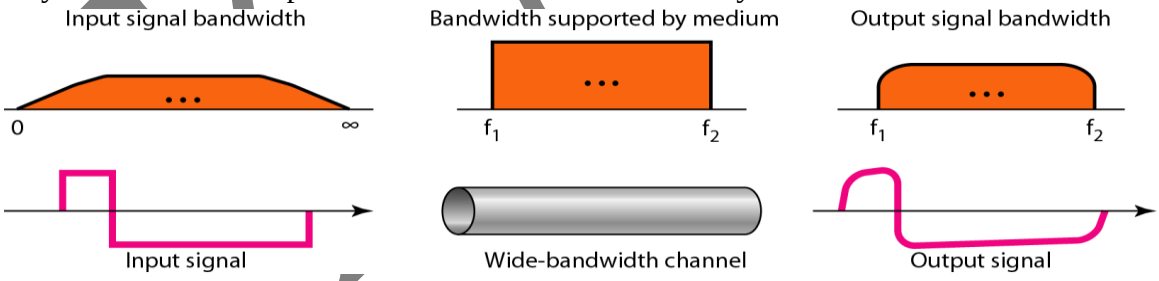


a. Low-pass channel, wide bandwidth

b. Low-pass channel, narrow bandwidth

Baseband transmission using a dedicated medium

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**.**

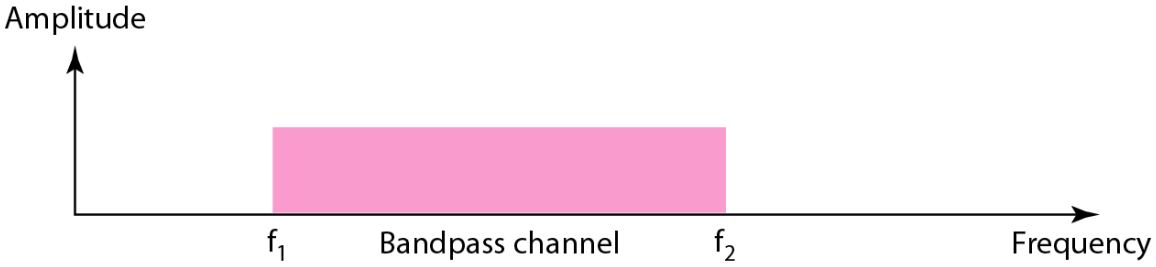


Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.

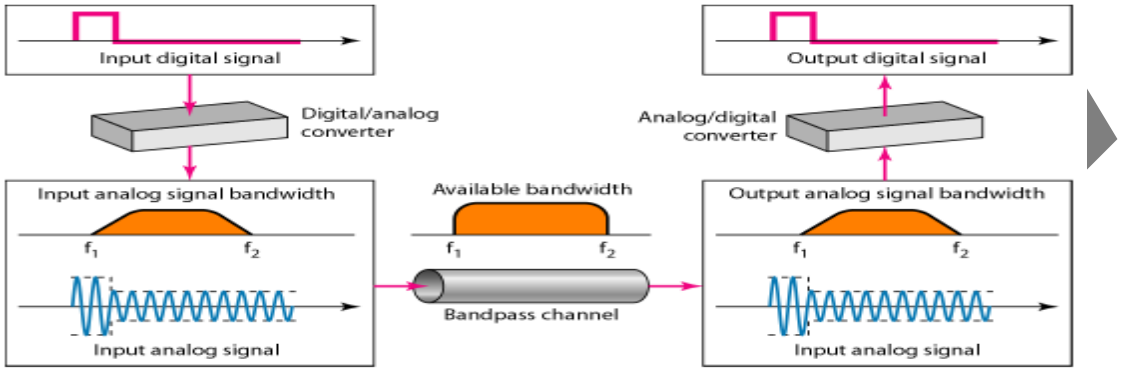
Broadband Transmission:

In broadband transmission the signal is converted to analog for transmission.
Bandwidth of a bandpass channel

If the available channel is a bandpass channel, we cannot send the digital signal directly to the channel; we need to convert the digital signal to an analog signal before transmission.



Modulation of a digital signal for transmission on a bandpass channel



Example

An example of broadband transmission using modulation is the sending of computer data through a telephone subscriber line, the line connecting a resident to the central telephone office. These lines are designed to carry voice with a limited bandwidth. The channel is considered a bandpass channel.

We convert the digital signal from the computer to an analog signal, and send the analog signal. We can install two converters to change the digital signal to analog and vice versa at the receiving end. The converter, in this case, is called a **modem**.

Data Rate Limits:

A very important consideration in data communications is how fast we can send data, in bits per second, over a channel. Data rate depends on three factors:

1. The bandwidth available
2. The level of the signals we use
3. The quality of the channel (the level of noise)

Noiseless channel: Nyquist bit rate

For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate

$$C = 2 B \log_2 L$$

where, C is the channel capacity or bit rate in bps
B is the bandwidth in Hz
L is the number of signal levels used to represent data

Noisy channel: Shannon capacity

In reality, we cannot have a noiseless channel; In this case, the Shannon capacity formula is used to determine the theoretical highest data rate for a noisy channel:

$$C = B \log_2 (1+SNR)$$

where, C is the capacity of the channel in bps
B is the bandwidth in Hz
SNR is the signal-to-noise ratio

PERFORMANCE

One important issue in networking is the performance of the network—how good is it? We discuss quality of service, an overall measurement of network performance

Bandwidth

In networking, we use the term bandwidth in two contexts.

- The first, bandwidth in hertz, refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.
- The second, bandwidth in bits per second, refers to the speed of bit transmission in a channel or link.

Throughput

The throughput is a measure of how fast we can actually send data through a network. Although, at first glance, bandwidth in bits per second and throughput seem the same, they are different. A link may have a bandwidth of *B* bps, but we can only send *T* bps through this link with *T* always less than *B*. In other words, the bandwidth is a potential measurement of a link; the throughput is an actual measurement of how fast we can send data. For example, we may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps. This means that we cannot send more than 200 kbps through this link.

Imagine a highway designed to transmit 1000 cars per minute from one point to another. However, if there is congestion on the road, this figure may be reduced to 100 cars per minute. The bandwidth is 1000 cars per minute; the throughput is 100 cars per minute.

Latency (Delay)

The latency or delay defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source. We can say that latency is made of four components: propagation time, transmission time, queuing time and processing delay.

Latency =propagation time +transmission time +queuing time + processing delay

Propagation Time

Propagation time measures the time required for a bit to travel from the source to the destination. The propagation time is calculated by dividing the distance by the propagation speed.

$$\text{Propagation time} = \frac{\text{Distance}}{\text{Propagation speed}}$$

The propagation speed of electromagnetic signals depends on the medium and on the frequency of the 28ignal For example, in a vacuum, light is propagated with a speed of 3 x 108 mfs. It is lower in air; it is much lower in cable.

Queuing Time

The third component in latency is the queuing time, the time needed for each intermediate or end device to hold the message before it can be processed. The queuing time is not a fixed factor; it changes with the load imposed on the network. When there is heavy traffic on the network, the queuing time increases. An intermediate device, such as a router, queues they arrived messages and processes them one by one. If there are many messages, each message will have to wait.

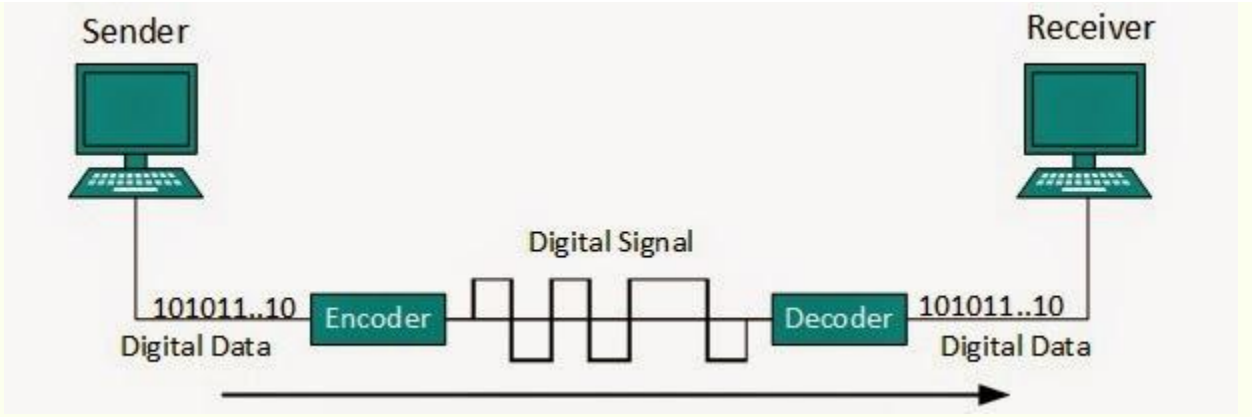
1.12 Digital worlds - digital signals

Digital to Digital Conversion:

In this section, we see how we can represent digital data by using digital signals. The conversion involves three techniques: line coding, block coding, and scrambling. Line coding is always needed; block coding and scrambling mayor may not be needed.

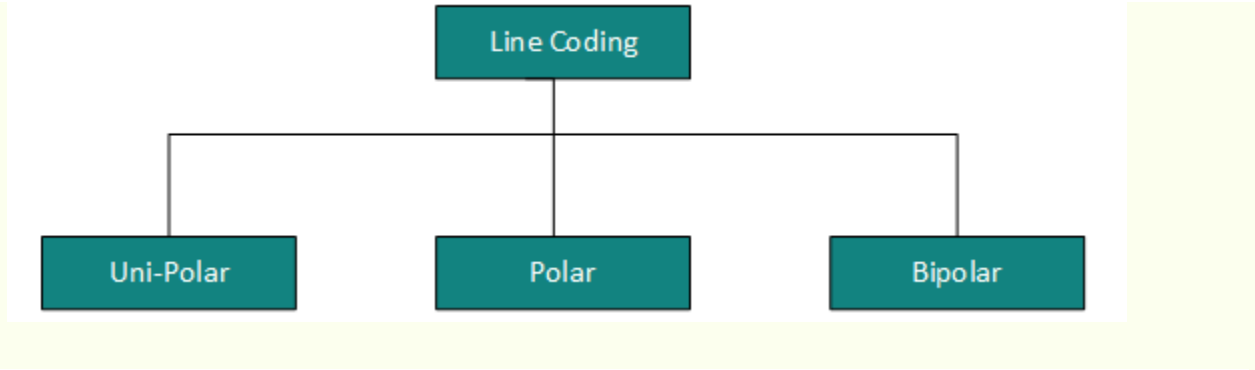
Line Coding

The process for converting digital data into digital signal is said to be Line Coding. Digital data is found in digital format, which is binary bits. It is represented (stored) internally as series of 1s and 0s.



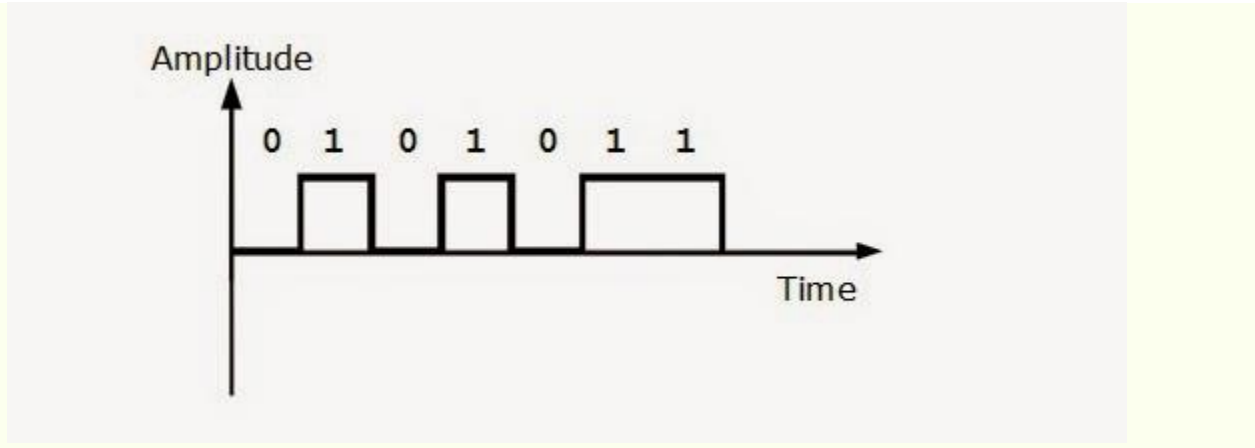
Line Coding

Digital signals which represents digital data, represented as discrete signals. There are three types of line coding schemes available:



Uni-Polar Encoding

Unipolar encoding schemes uses single voltage level to represent data. In this case, to represent binary 1 high voltage is transmitted and to represent 0 no voltage is transmitted. It is also called Unipolar-Non-return-to-zero, because there's no rest condition i.e. it either represents 1 or 0.

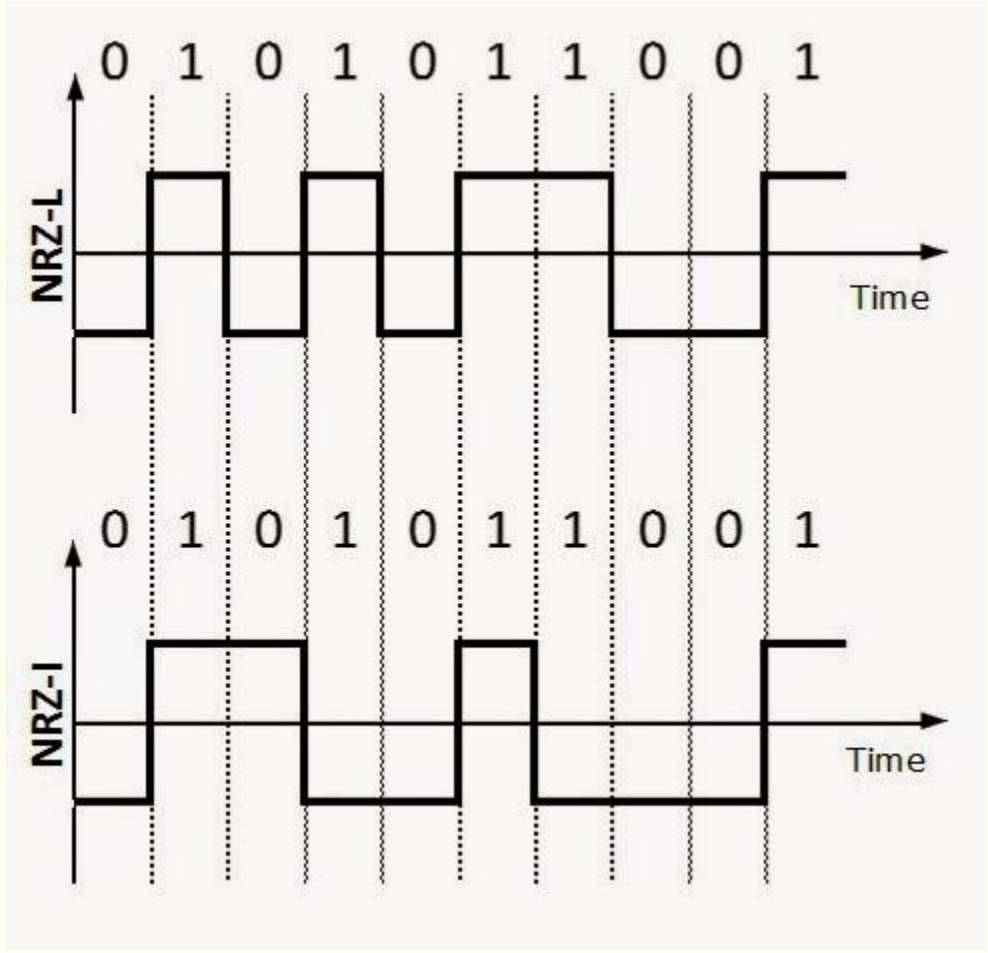


Polar Encoding

Polar encoding schemes multiple voltage levels are used to represent binary values. Polar encodings are available in four types:

Polar-NRZ (Non-Return To Zero)

It uses two different voltage levels to represent binary values, generally positive voltage represents 1 and negative value represents 0. It is also NRZ because there's no rest condition.
NRZ scheme has two variants: NRZ-L and NRZ-I.

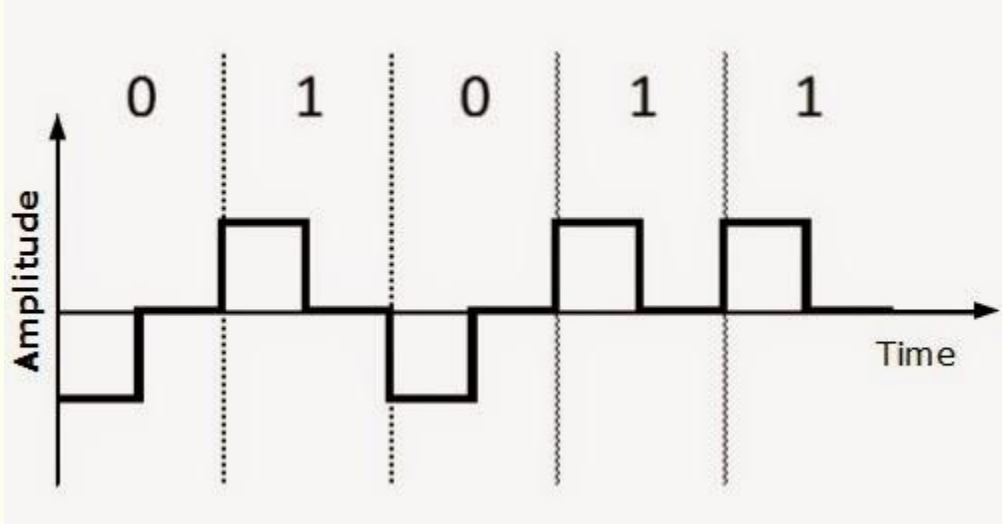


NRZ-L and NRZ-I

NRZ-L changes voltage level at when a different bit is encountered whereas NRZ-I changes voltage when a 1 is encountered.

RZ (Return to Zero)

Problem with NRZ was the receiver cannot conclude when a bit ended and when the next bit is started, in case when sender and receiver's clock are not synchronized.



Return-to-Zero Encoding

RZ uses three voltage levels, positive voltage to represent 1, negative voltage to represent 0 and zero voltage for none. Signals change during bits not between bits.

Manchester

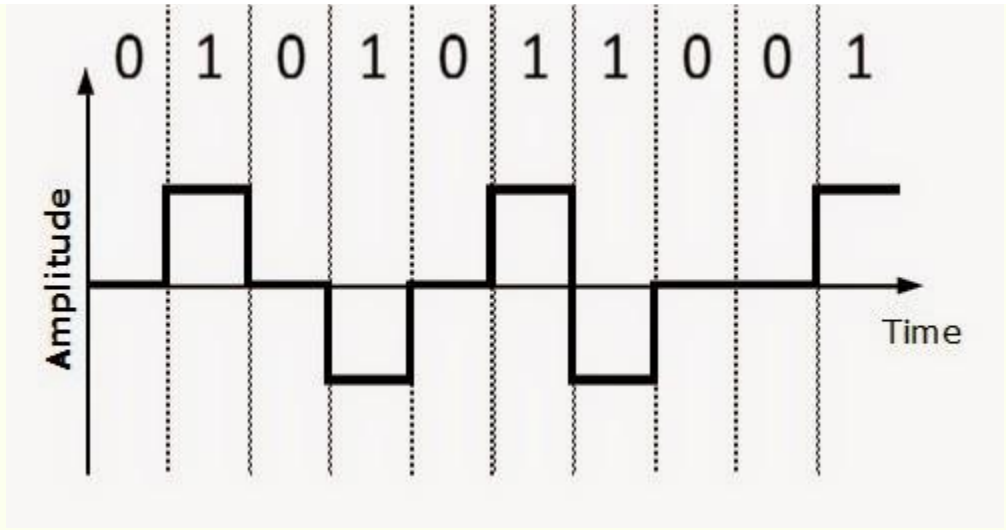
This encoding scheme is a combination of RZ and NRZ-L. Bit time is divided into two halves. It transitions at the middle of the bit and changes phase when a different bit is encountered.

Differential Manchester

This encoding scheme is a combination of RZ and NRZ-I. It also transitions at the middle of the bit but changes phase only when 1 is encountered.

Bipolar Encoding

Bipolar encoding uses three voltage levels, positive, negative and zero. Zero voltage represents binary 0 and bit 1 is represented by altering positive and negative voltages.



Bipolar Encoding

Block Coding

To ensure accuracy of data frame received, redundant bits are used. For example, in even parity one parity bit is added to make the count of 1s in the frame even. This way the original number of bits are increased. It is called Block Coding.

Block coding is represented by slash notation, mB/nB, that is m-bit block is substituted with n-bit block where n > m. Block coding involves three steps: division, substitution and combination. After block coding is done it is line coded for transmission.

Analog-to-digital conversion

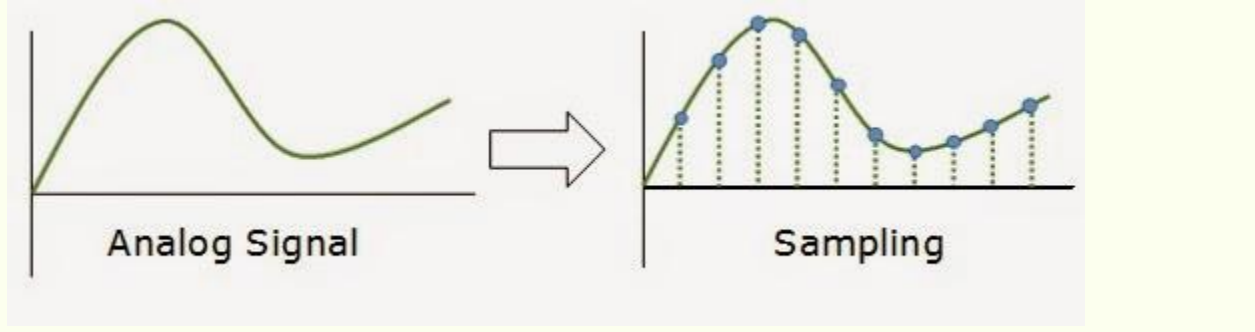
Microphones creates analog voice and camera creates analog videos, which here in our case is treated is analog data. To transmit this analog data over digital signals we need an analog to digital conversion.

Analog data is wave form continuous stream of data whereas digital data is discrete. To convert analog wave into digital data we use Pulse Code Modulation.

Pulse Code Modulation:

Pulse Code Modulation is one of the most commonly used method to convert analog data into digital form. It involves three steps: Sampling, Quantization and Encoding.

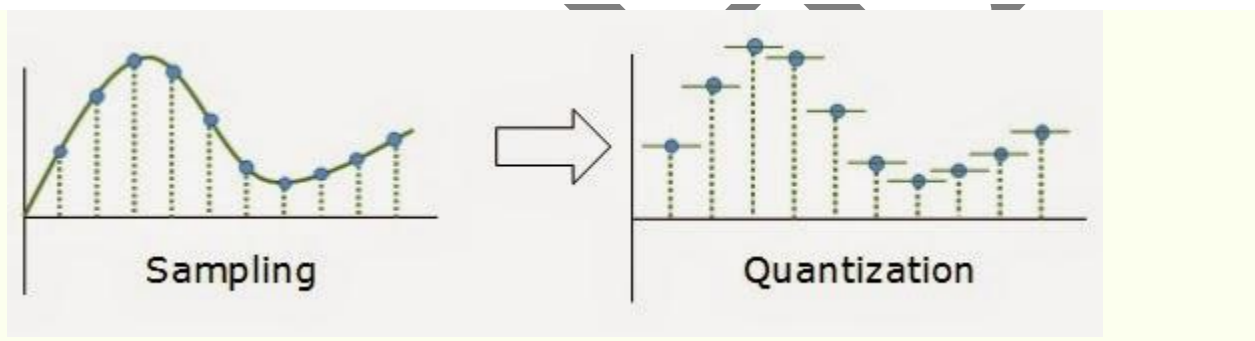
Sampling



Sampling of Analog Signal

The analog signal is sampled every T interval. Most important factor in sampling is the rate on which analog signal is sampled. According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.

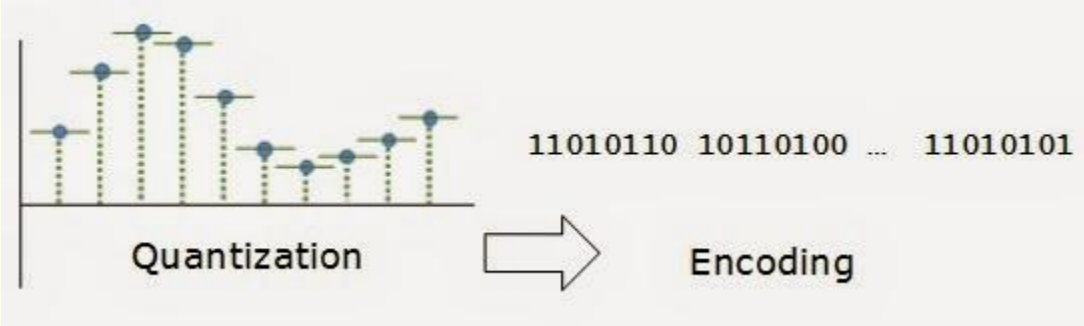
Quantization



Quantization of sampled analog signal

Sampling yields discrete form of continuous analog signal. Every discrete pattern shows the amplitude of the analog signal at that instance. The quantization is done between the maximum amplitude value and the minimum amplitude value. Quantization is approximation of the instantaneous analog value.

Encoding

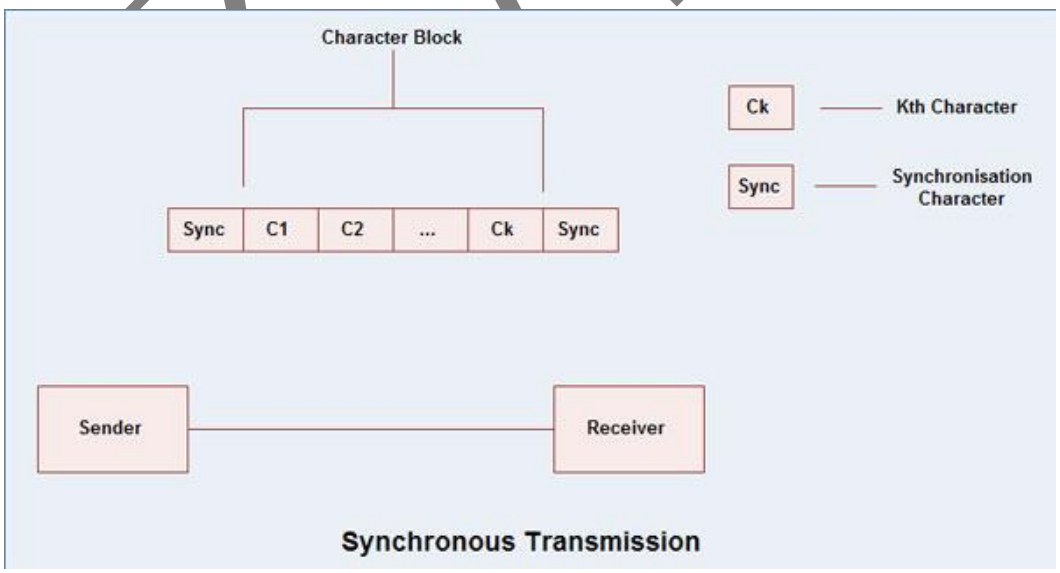


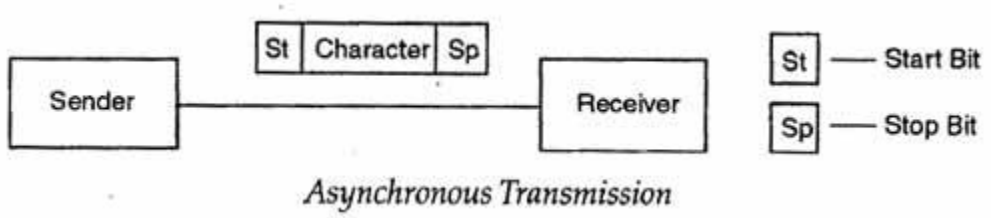
Encoding from quantization

In encoding, each approximated value is then converted into binary format.

1.13 The modem, asynchronous and synchronous transmission.

Data is transmitted between communication devices in multiples of fixed-length units, typically 8-bits. For example, if the computer is transferring a source program, the data will be made up of a block of 8-bit binary-encoded characters. On the other hand, if the data is in the form of a compiled object code of the program, the data will be made up of a block of 8-bit bytes. At the receiving end, the following parameters are determined to decode and interpret the message correctly.





Asynchronous Transmission

1.14 Wide area and local networks, connection oriented and connectionless networks

Connection Oriented Services

There is a sequence of operation to be followed by the users of connection oriented service. These are :

1. Connection is established
2. Information is sent
3. Connection is released

In connection oriented service we have to establish a connection before starting the communication. When connection is established we send the message or the information and then we release the connection.

Connection oriented service is more reliable than connectionless service. We can send the message in connection oriented service if there is an error at the receivers end. Example of connection oriented is TCP (Transmission Control Protocol) protocol.

Connection Less Services

It is similar to the postal services, as it carries the full address where the message (letter) is to be carried. Each message is routed independently from source to destination. The order of message sent can be different from the order received.

In connectionless the data is transferred in one direction from source to destination without checking that destination is still there or not or if it prepared to accept the message. Authentication is not needed in this. Example of Connectionless service is UDP (User Datagram Protocol) protocol.

Difference between Connection oriented service and Connectionless service

1. In connection oriented service authentication is needed while connectionless service does not need any authentication.

2. Connection oriented protocol makes a connection and checks whether message is received or not and sends again if an error occurs connectionless service protocol does not guarantees a delivery.
3. Connection oriented service is more reliable than connectionless service.
4. Connection oriented service interface is stream based and connectionless is message based.

Service Primitives

A service is formally specified by a set of primitives (operations) available to a user process to access the service. These primitives tell the service to perform some action or report on an action taken by a peer entity. If the protocol stack is located in the operating system, as it often is, the primitives are normally system calls. These calls cause a trap to kernel mode, which then turns control of the machine over to the operating system to send the necessary packets. The set of primitives available depends on the nature of the service being provided. The primitives for connection-oriented service are different from those of connection-less service. There are five types of service primitives :

1. **LISTEN :**When a server is ready to accept an incoming connection it executes the LISTEN primitive. It blocks waiting for an incoming connection.
2. **CONNECT :**It connects the server by establishing a connection. Response is awaited.
3. **RECIEVE:** Then the RECIEVE call blocks the server.
4. **SEND :**Then the client executes SEND primitive to transmit its request followed by the execution of RECIEVE to get the reply. Send the message.
5. **DISCONNECT :** This primitive is used for terminating the connection. After this primitive one can't send any message. When the client sends DISCONNECT packet then the server also sends the DISCONNECT packet to acknowledge the client. When the server package is received by client then the process is terminated.

Connection Oriented Service Primitives

There are 5 types of primitives for Connection Oriented Service :

LISTEN	Block waiting for an incoming connection
CONNECTION	Establish a connection with a waiting peer
RECEIVE	Block waiting for an incoming message

SEND	Sending a message to the peer
DISCONNECT	Terminate a connection

Connectionless Oriented Service Primitives

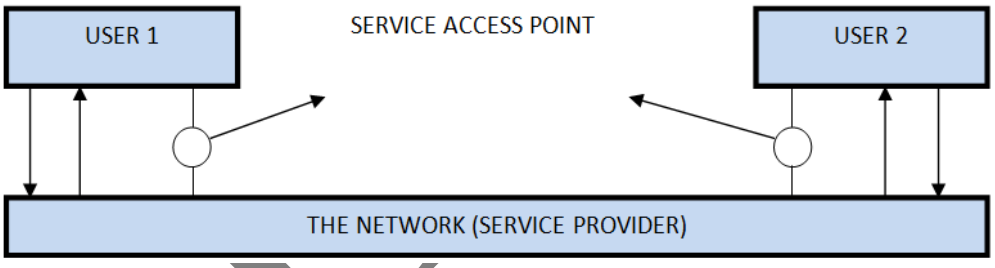
There are 4 types of primitives for Connectionless Oriented Service:

UNIDATA	This primitive sends a packet of data
FACILITY, REPORT	Primitive for enquiring about the performance of the network, like delivery statistics.

Relationship of Services to Protocol

Services

These are the operations that a layer can provide to the layer above it. It defines the operation and states a layer is ready to perform but it does not specify anything about the implementation of these operations.



Protocols

These are set of rules that govern the format and meaning of frames, messages or packets that are exchanged between the server and client.

1.15 Classification of communications protocols

Communication protocols are formal descriptions of digital message formats and rules. They are required to exchange messages in or between computing systems and are required in telecommunications.

Communications protocols cover authentication, error detection and correction, and signaling. They can also describe the syntax, semantics, and synchronization of analog and digital communications. Communications protocols are implemented in hardware and software. There are thousands of communications protocols that are used everywhere in analog and digital communications. Computer networks cannot exist without them.

Communications devices have to agree on many physical aspects of the data to be exchanged before successful transmission can take place. Rules defining transmissions are called protocols.

There are many properties of a transmission that a protocol can define. Common ones include: packet size, transmission speed, error correction types, handshaking and synchronization techniques, address mapping, acknowledgement processes, flow control, packet sequence controls, routing, address formatting

Popular protocols include:

1. File Transfer Protocol (FTP),
2. TCP/IP,
3. User Datagram Protocol (UDP),
4. Hypertext Transfer Protocol (HTTP),
5. Post Office Protocol (POP3),
6. Internet Message Access Protocol (IMAP),
7. Simple Mail Transfer Protocol (SMTP).

1.16 Time division multiple access (TDMA), time division multiplexing (TDM), carrier sense (Collision) systems, token passing, peer-to-peerpriority systems; priority slot, carrier sense (collision free) systems, token passing (priority) systems.

In TDMA, the bandwidth of channel is dividend amongst various stations on the basis of time.

• Each station is allocated a time slot during which it can sent its data *i.e.* each station can transmit its data in its allocated time slot only.

• Each station must know the beginning of its slot and the location of its slot.

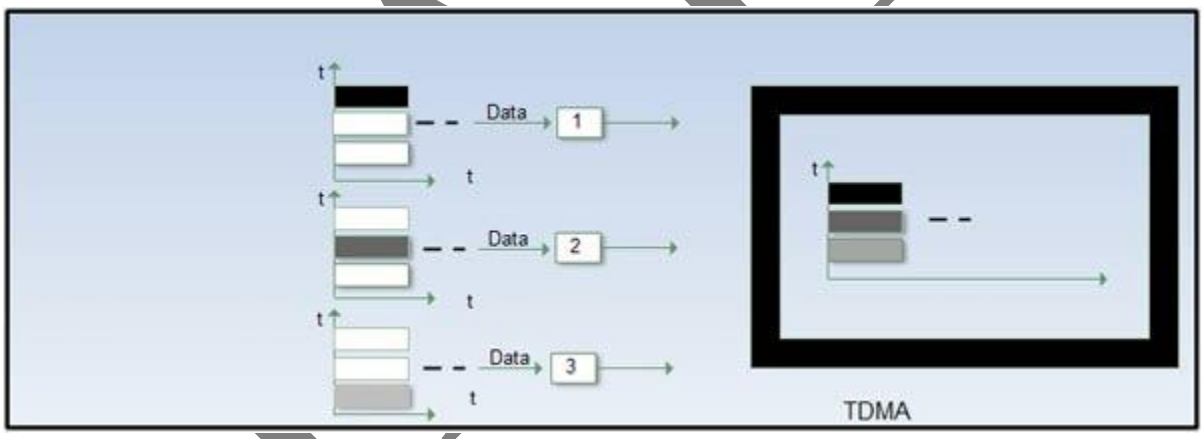
• TDMA requires synchronization between different stations.

• Synchronization is achieved by using some synchronization bits (preamble bits) at the beginning of each slot.

• TDMA is different from TDM, although they are conceptually same.

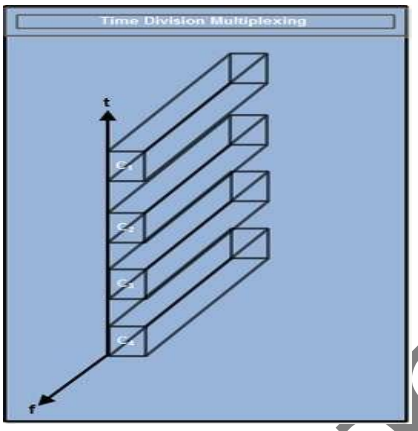
• TDM is a physical layer technique that combines the data from slower channels and transmits then by using a faster channel. This process uses physical multiplexer.

• TDMA, on other hand, is an access method in the data link layer. The data link layer in each station tells its physical layer to use the allocated time slot. There is no physical multiplexer at the physical layer.



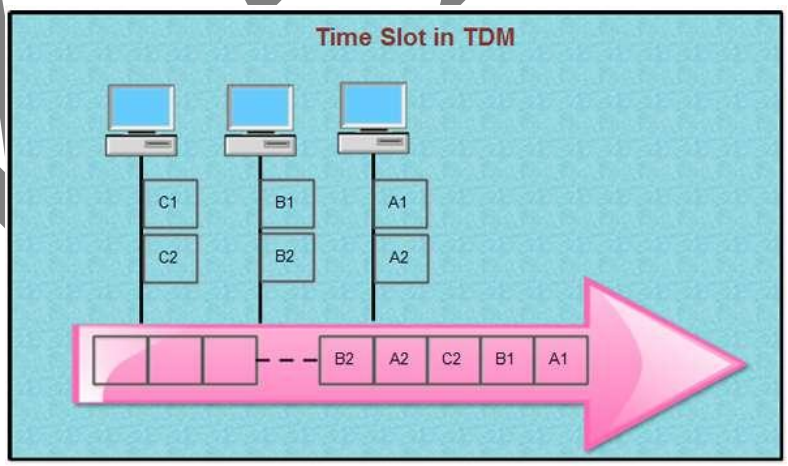
For digital sources, two alternative technologies have evolved for multiplexing. These are **Time Division Multiplexing** (*TDM*) and Code division Multiplexing (CDM). TDM provides a way to merge data from several sources into a single channel for communication over telephone lines, a microwave system or a satellite system. TDM can be implemented in two ways. These are synchronous TDM and asynchronous TDM. Asynchronous TDM is popularly known as Statistical TDM.

In synchronous TDM, a single channel is divided into time slots and each transmitting device is assigned at least one of the time slots for *its* transmission



Time slots are assigned in such a way that each transmitting device gets its required share of the available bandwidth. 'Because of this time-bandwidth multiplexing technique, TDMs are protocol insensitive and are capable of combining various protocols onto a single high-speed transmission link. In other words, we can say that multiplexer allocates exactly-the same time slot to each device at all times whether the device is active or idle.

Some devices such as voice and video systems may require more slots to ensure that data arrives at the distant link-end without becoming distorted from slower data rates. These different time slots are grouped into frames. A frame consists of one complete cycle of time slots.



Alternatively, Figure explains more clearly the concept of TDM in a data communication environment where three PCs are sharing the common circuit. The packets generated by each PC are multiplexed on the common line as A1, B1, and C1 and so on.

It is more flexible than the FDM unlike FDM. the whole bandwidth for a certain amount of time is provided to the user. All the users are using the same frequency but at a different time. This time allotment may be varied as per the requirement and priority of the users' services. In the Figure spaces between different time slots are shown, these are known as guard spaces in time dimension. These are used to eliminate co-channel interference.

The main disadvantage of this scheme is that a precise synchronization between different senders is necessary to avoid co-channel interference.

The **Token-Passing Protocol** relies on a control signal called the token. A token is a 24-bit packet that circulates throughout the network from NIC to NIC in an orderly fashion. If a workstation wants to transmit a message, first it must seize the token. At that point, the workstation has complete control over the communications channel. The existence of only one token eliminates the possibility of signal collisions. This means that only one station can speak at a time.

Logical Ring Physical Star topology for Token-Passing Standard.

It is sure that any break in the ring at any point will interrupt communications for all machines. To solve this problem, IBM developed a modified ring topology, which they called the logical ring physical star. The central point of the physical star configuration is Token Ring hub called the multi-station access unit (MSAU, pronounced as masow).

Workstations and servers attached to the MSAU through special STP adapter cables. IBM converted stars into a logical ring by connecting all MSAU hubs together through special ring-in (RI) and ring-out (RO) ports.

Advantages of Token Ring.

Here are Token ring's most useful advantages:

a. It offers excellent throughput under high-load conditions.

b. Token Ring facilitates LAN-to-LAN mainframe connections especially for interfacing with IBM's broader connectivity strategies.

c. It has built-in troubleshooting mechanisms such as beaconing and auto-reconfiguration and may now be used with UTP cabling.

d. It has the most reliable protocol (token-passing), the most trouble-free configuration (physical star) and the fastest connectivity scheme (r or 16 mb/s).

Disadvantages of Token Ring.

Few of the disadvantages of Token Ring are:

a. Token Ring is very expensive. All topology components cost much more than other more popular standards.

b. It is relatively proprietary. Token Ring's complexity is built into the hardware components. This means hat you need to choose a manufacturer and stick with it.

c. Engineers must have considerable expertise to manage and troubleshoot token ring components.

UNIT 1 - POSSIBLE QUESTIONS
PART A
(20 MARKS)
ONLINE EXAMINATION

PART B
(Each Question carries 2 marks)

1. Write about the advantages of networks.
2. Discuss with the structure of communication network.
3. Discuss about the network topologies.
4. Write about the telephone network.
5. Discuss about the fundamentals of communication theory.
6. Write a short notes channel speed and bit rate.
7. Discuss about the Polling and selection systems.
8. Write a short note sliding windows.
9. Write about the concept of sliding windows.
10. How carrier sense systems are utilized in the communication systems?

PART C

Compulsory Question

(Each Question Carries 8 marks)

1. Discuss about the classification of communication protocols.
2. Compare the OSI reference model with TCP/IP model.
3. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.
4. Explain in detail about the X.25 architecture with an example network.
5. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.

KARPAGAM ACADEMY OF HIGHER EDUCATION
DEPARTMENT OF COMPUTER APPLICATIONS
SUBJECT CODE/NAME: 17CAP302, COMPUTER NETWORKS
Degree : PG **Course: MCA** **Year: 2017**

UNIT 1						
S.NO	**QUESTIONS**	**OPTION 1**	**OPTION 2**	**OPTION 3**	**OPTION 4**	**ANSWER**
1	A _____ is a number of computers interconnected by one or more transmission paths.	Internet	Network	Group	ARPANET	**Network**
2	The _____is the end user application.	Application process	Application layer	Application terminal	Application software	**Application Process**
3	A network provides _____ and _____ communications for the computers and terminals to be connected.	Protocol and Internet	Application and Physical	Logical and physical	Logical and Internet	**Logical and physical**
4	The interfaces are specified and established through _____	Internet	Protocols	Network	Router	**Protocols**
5	_____ transmission is common in television and commericial radio.	Duplex	Simplex	Half duplex	Full duplex	**Simplex**
6	The bus topology is also called as _____	Hierarchial	Horizontal	Line	Sequence	**Horizontal**
7	Connection is provided to the CO through a pair of wires called _____	Local loop	Global loop	Star loop	Bus loop	**Local loop**
8	When one speaks oscillating waveforms of high and low air pressure are created is called _____	Digital	Analog	Physical	Logical	**Analog**
9	The Modem provides the _____ interface.	Digital/Analog	Digital	Analog	Logical	**Digital/Analog**
10	Short connections between machines often use a separate channel or line to provide the _____	Synchronization	Asynchronous	Analog	Digital	**Synchronization**

11	_____ have no signal below zero voltage or no signal above.	Unipolar code	Polar code	Bipolar code	AMI code	**Unipolar code**
12	Signal is above and below zero voltageis _____	Unipolar code	Polar code	Bipolar code	AMI code	**Polarcode**
13	The signal varies in all ways are _____	Unipolar code	Polar code	Bipolar code	AMI code	**Bipolar code**
14	Asynchronous transmission is widely used because the interfaces in the DTEs and DECs are _____	Expensive	Inexpensive	Flexible	Inflexible	**Inexpensive**
15	An element to check for a transmission error, typiclly called an _____	FCS Field	CRC field	DTE field	DCE field	**FCS field**
16	_____ is the basic unit of information transmitted across the communications channel.	Field	Frame	File	Code	**Frame**
17	The most common method used today for error checking is _____	CRC	FCS	DTE	DCE	**CRC**
18	The use of error checking techniques and _____ are necessary to ensure the integrity of user data.	CRC	ACK/NAK	DTE field	DCE field	**ACK/NAK**
19	PAD is meant by _____	Packet Assembly/Disassembly	Packet Allied Data	Packet Assigned Data	Packet Assembly Data	**Packet Assembly/Disassembly**
20	Links are relatively _____	Error prone	Error Detection	Error Correction	Error Sectors	**Error prone**
21	Links are usually owned by the _____	User Data	User Organization	User mail	Email	**User Organization**
22	A WAN topology has an _____	Regular shape	Irregualar shape	Star shape	Linear shape	**Irregular shape**
23	_____ network is the one in which no logical connection initially exists between the DTEs and the network.	Connectionless	Connecion oriented	Wireless	Satellite	**Connection oriented**

24	_____ network goes directly from an idle condition into a data transfer mode followed directly by the idle condition.	Connectionless	Connecion oriented	Wireless	Satellite	**Connectionless**
25	_____ networks are often compared conceptually to the telephone system.	Connectionless	Connecion oriented	Wireless	Satellite	**Connection oriented**
26	_____ have dominated computer wide area networks because of the inherent error prone nature of the telephone system.	Connecion oriented	Connectionless	Wireless	Satellite	**Connection oriented**
27	A LAN typically experiences an error rate of approximately _____	$1:10^{7}$	$1:10^{6}$	$1:10^{5}$	$1:10^{8}$	**$1:10^{8}$**
28	The majority of the protocols depicted in thecommunication protocols are _____	Line	Link	Data	Data hidden	**Line**
29	Data link protocols manage all communications traffic on a _____	Channel	Link	Line	Data Link	**Channel**
30	The _____ handshakes with the remote DLC logic to ensure both systems are ready to exchange user data.	DLC	DCE	DSE	DTE	**DCE**
31	Primary/Secondary Protocol is otherwise called as _____	Master/Slave Protocol	Hybrid Protocol	Polling Protocol	Nonpolling Protocol	**Master/Slave Protocol**
32	_____protocol technique has no primary station and typically provides for equal status to all stations on the channel	Hybrid	Peer to Peer	Master/Slave	Nonpolling	**Peer to Peer**
33	The purpose of the _____ command is to transmit data to the primary site.	Poll	Select	Poll and Select	ACK	**Poll**
34	The purpose of the _____ command is to transmit data to the secondary site.	Poll	Select	Poll and Select	ACK	**Select**
35	_____ are the principal commands neede to move data to any site on a channel or in the network.	Poll	Select	Poll and Select	ACK	**Poll and Select**
36	The primary site checks for errors and sends an ACK if the data are correct ot a NAK if they are _____	Correct	Incorrect	Mislead	Acknowledge	**Incorrect**

37	The _____station must then send an indicator that it has completed its transmission, such as the end-of-transmission.	Primary	Secondary	Master	Slave	**Secondary**
38	A disadvantage of a poling/selection system is the number of negative responses topolls, which can consume precious resources of the _____	Channel	Network	Devices	System	**Channel**
39	HDLC means _____	Hierarchial Data Link Control	High Data Link Control	High Level Data Link Control	Higher Data Link Control	**High Level Data Link Control**
40	SDLC means_____	Synchronous Data Link Control	System Data Link Control	Secure Data Link Control	Selective Data LinkControl	**Synchronous Data Link Control**
41	_____ is the technqiue and mechanism for multidrop communications links.	Secure Polling	Polling	Selective Polling	Non Polling	**Selective Polling**
42	_____ is more common on a ring or loop topology oron a linewith cluster controllers.	Secure Polling	Group Polling	Selective Polling	Non Polling	**Group Polling**
43	A station may _____ its transmission onto the data passing around the ring.	Piggyback	Error Detection	Error Correction	Error Sectors	**Piggyback**
44	_____ is widely used approach because it is relatively inexpensive in communication channels.	Polling	Stop and Wait	Correction	Error detection	**Stop and Wait**
45	A timeout means that after not receiving a reply to its transmissionwithin a given period,	retransmits the data	transmits the acknowledge	receives the data	stop the data to receive	**retransmits the data**
46	_____ is so named because a station is allowed to request automatically a retransmission from another station.	Continuous ARQ	Non Continuous ARQ	Reply data	Non replying data	**Continuous ARQ**
47	_____ does not use a polling selection system.	ARQ	DCE	TDMA	DTE	**TDMA**

48	_____ is probably the simplest example of peer to peer nonpriority systems.	TDM	TDMA	DCE	ARQ	**TDMA**
49	Carrier sense is an example for _____	Peer to peer nonpriority systems	Peer to peer nonpriority systems	Collision nonpriority systems	Peer to peer systems	**Peer to peer nonpriority systems**
50	_____ sense technique provides the facility for all stations to transmit immediately upon sensing the idle channel, with no arbitration before the transmission.	Nonpersistent	Persistant	Transmit	Non transmit	**Non persistent**
51	The _____ is a factor of the propagation delay of the signal and the diatance between two competing stations.	Continuous ARQ	Collision window	receives the data	Error Sectors	**Collision window**
52	_____ networks are usually implemented on LAN because the collison window lengthens with a longer WAN.	Carrier Sense	Continuous ARQ	Nonpersistent	Non Polling	**Carrier Sense**
53	The stations are connected to a concentric ring through _____	RIU	TCE	DTE	DSE	**RIU**
54	The technique called_____ system to any station is allowed to transmit data when it receives a free token.	explicit token	implicit token	token generation	token deletion	**implicit token**
55	The _____ eliminates the collisions found in the carrie sense (collision) sytems and allows the use of a nonring channel.	Protocol and Internet	Internet	Protocol	Network	**Protocol**
56	The protocol uses a control frame called an _____	Access	Access token	Token	Right	**Access token**
57	The stations receive the token through a cyclic sequence which forms a logical ring on the _____	Physical bus	Layered bus	Ring	Topology bus	**Physical bus**
58	_____ can be estblished without a master station.	Ring	Priority Slot	Slot	Ring Slot	**Priority Slot**

59	_____ have many similarities to the carrier sense networks.	Carrier sense (collison free)	Protocol and Internet	Non Polling	Collision nonpriority systems	**Carrier sense (collison free)**
60	_____ is an enhanced token passing scheme, in which priorities are added to a token passingsystem, usually a token ring.	Peer to peer nonpriority systems	Collision nonpriority systems	Peer to peer nonpriority systems	Peer to peer systems	**Peer to peer nonpriority systems**

KARPAGAM ACADEMY OF HIGHER EDUCATION
(Deemed University Established Under Section 3 of UGC Act 1956)
Coimbatore - 641021.
(For the candidates admitted from 2017 onwards)

DEPARTMENT OF CS, CA & IT

SYLLABUS:

UNIT II

Layered Protocols and the OSI model

Goals of Layered Protocols, network design problems, communication between layers, Introduction to standard organizations and the OSI model, standards organizations, Layers of OSI, OSI status.

Local Area Networks

Way LANs?, Primary attributes of a LAN, Broadband and baseband and base LANs, IEEE LAN standards, e1ationship of the 802 standards to the ISO/CCITT model., connection options with LANs, LLC and MAC protocol data units, LAN topologies and protocols., token ring (Priority), token bus and IEEE 802.4, metropolitan area networks (MANs), ANSI fiber distributed data interface.

2.1 Goals of Layered Protocols

The **Open Systems Interconnection model** (**OSI model**) is a conceptual model that characterizes and standardizes the communication functions of a telecommunication or computing system without regard to their underlying internal structure and technology. Its goal is the interoperability of diverse communication systems with standard protocols. The model partitions a communication system into abstraction layers. The original version of the model defined seven layers.

A layer serves the layer above it and is served by the layer below it. For example, a layer that provides error-free communications across a network provides the path needed by applications above it, while it calls the next lower layer to send and receive packets that comprise the contents of that path. Two instances at the same layer are visualized as connected by a *horizontal* connection in that layer.

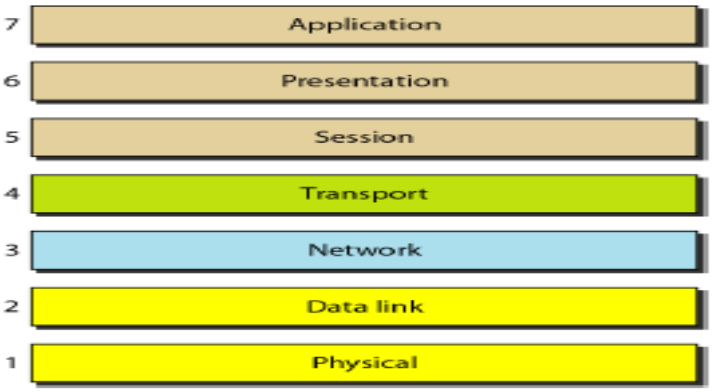
The model is a product of the Open Systems Interconnection project at the International Organization for Standardization (ISO), maintained by the identification ISO/IEC 7498-1.

2.2 The OSI model, standards organizations, Layers of OSI, OSI status.

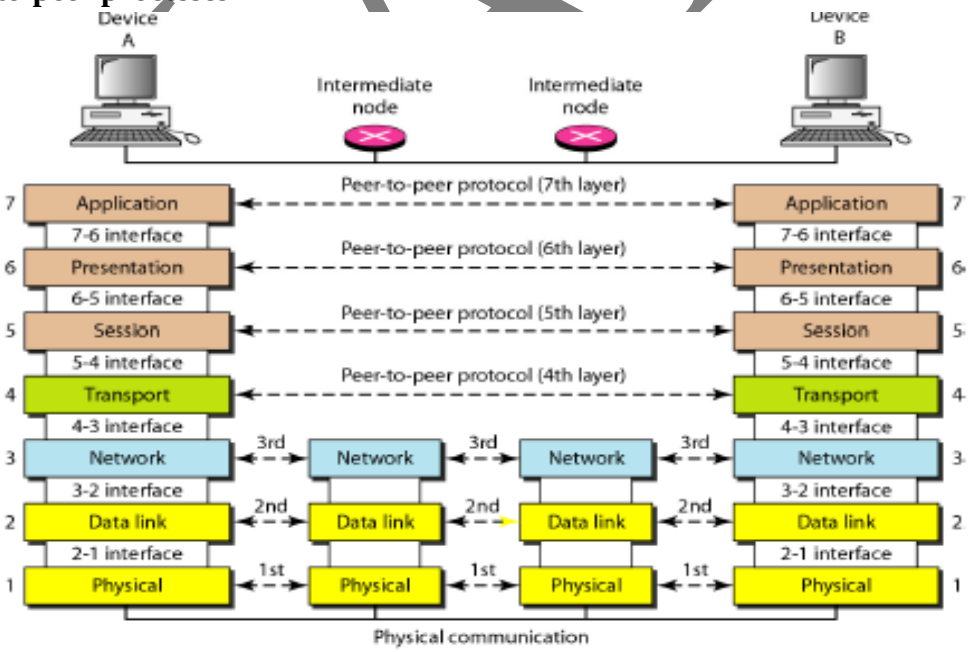
OSI Model

Established in 1947, the International Standards Organization (ISO) is a multinational body dedicated to worldwide agreement on international standards.
An ISO standard that covers all aspects of network communications is the Open Systems Interconnection (OSI) model. It was first introduced in the late 1970s.

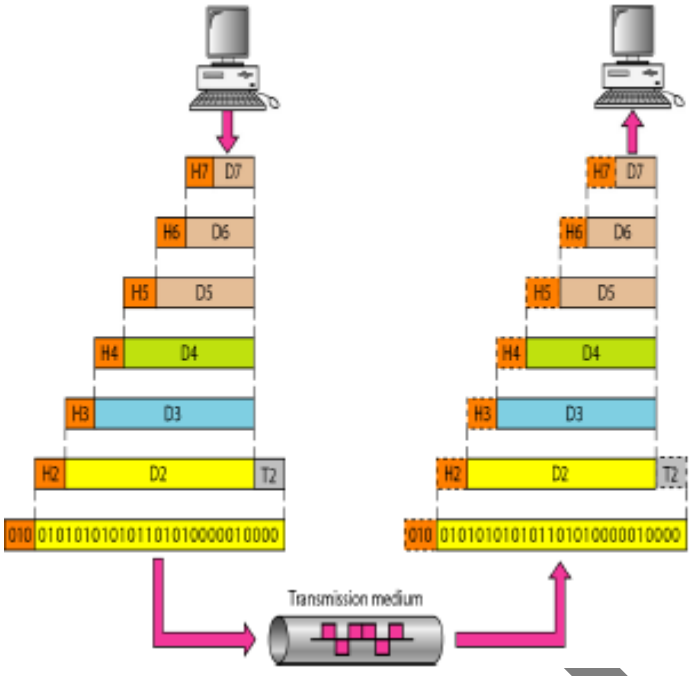
Seven layers of the OSI model



Peer-to-peer processes



Encapsulation

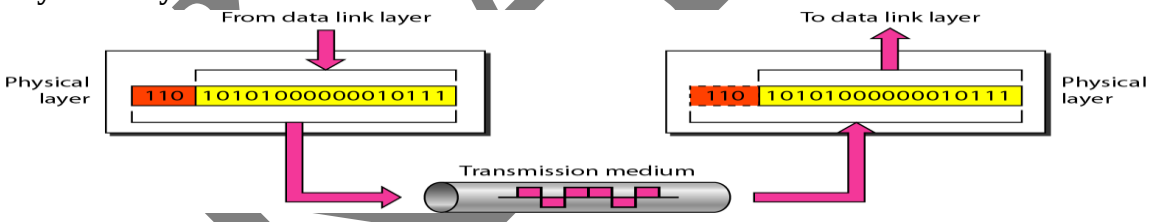


- The process starts at layer 7 (application layer), then moves from layer to layer in descending, sequential order.
- At each layer, a **header** is added to the data unit.
- At layer 2, a **trailer** is added as well.
- When the formatted data unit passes through the physical layer (layer 1) it is changed into an electromagnetic or optical signal and transported along a physical link
- At the destination the reverse process is performed

THE OSI MODEL AND LAYERS

In this section we briefly describe the functions of each layer in the OSI model.

Physical Layer



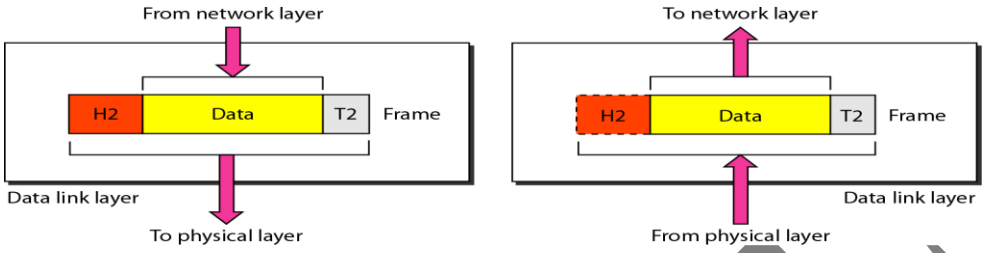
- The physical layer is responsible for movements of individual bits from one hop (node) to the next
- Mechanical and electrical specification, the procedures and functions

Duties:

- Physical characteristics of interfaces and media
- Representation of bits
- Data rate
- Synchronization of bits
- Line configuration
- Physical topology
- Transmission mode

Data link layer

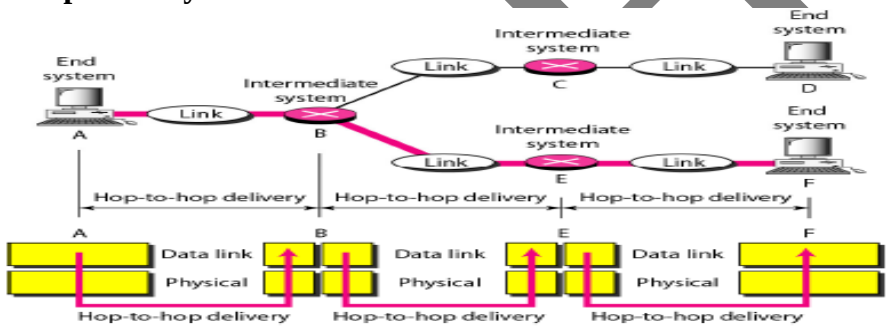
- The data link layer is responsible for moving frames from one hop (node) to the next
- Transform the physical layer to a reliable (error-free) link



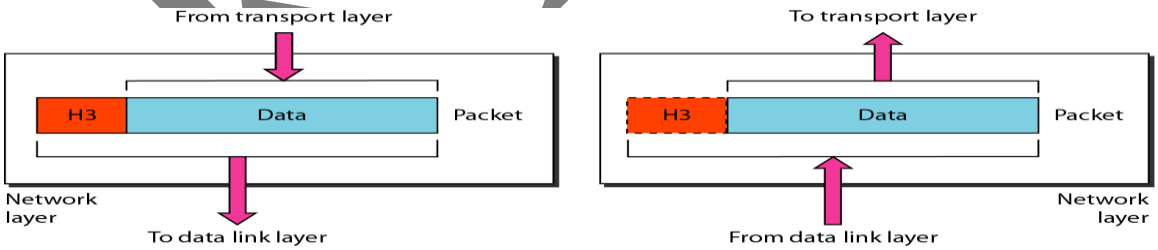
Duties:

- Framing
- Physical addressing
- Flow control
- Error control
- Access control

Hop-to-hop delivery



Network layer

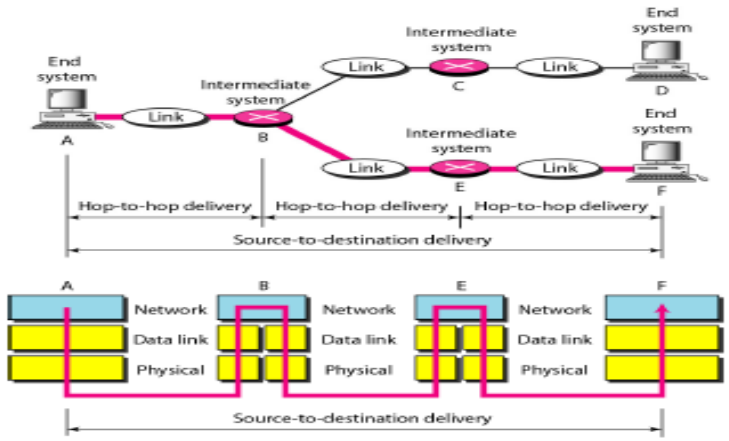


The network layer is responsible for the delivery of individual packets from the source host to the destination host.

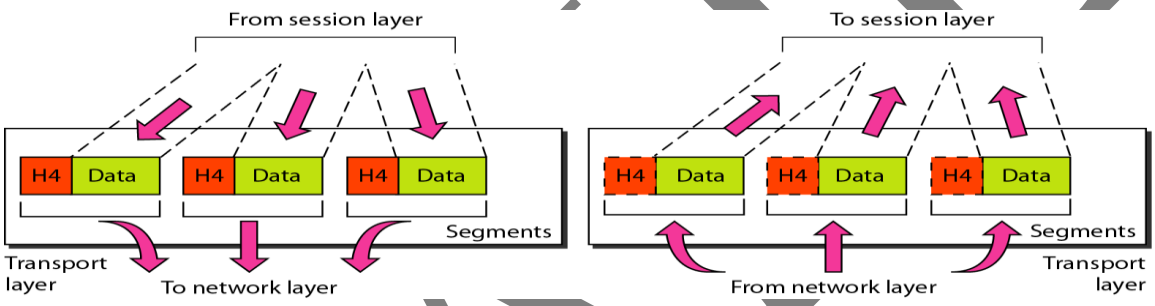
Duties:

- Logical addressing
- Routing

Source-to-destination delivery



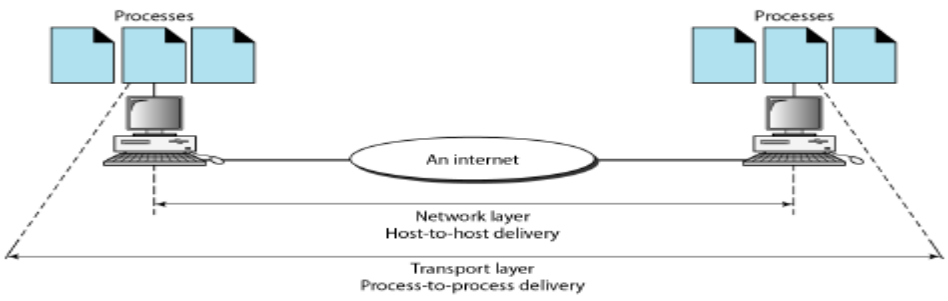
Transport layer



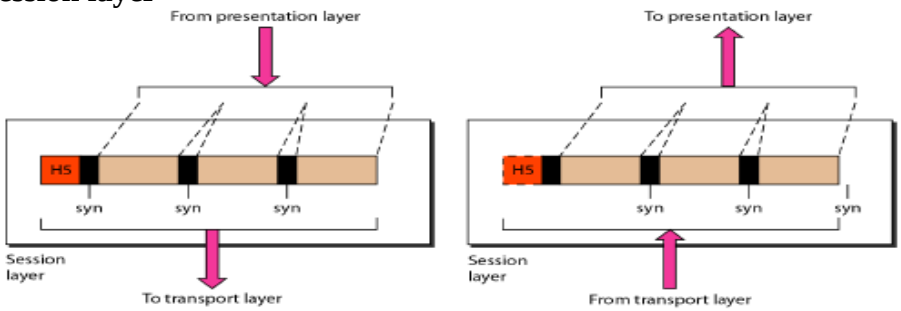
The transport layer is responsible for the delivery of a message from one process to another.
Duties:

- Service-point (port) addressing
- Segmentation and reassembly
- Connection control
- Flow control
- Error control

Reliable process-to-process delivery

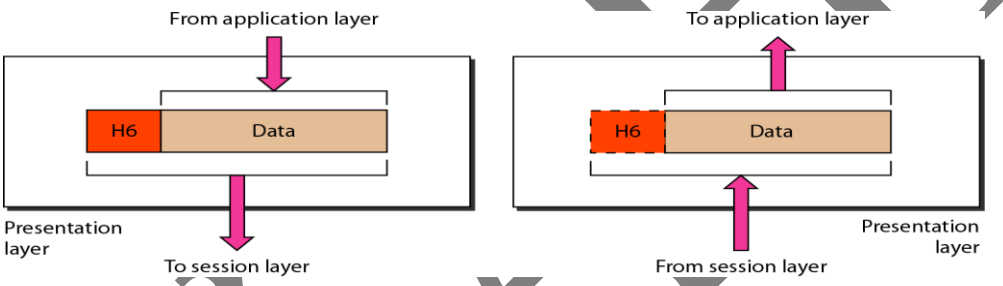


Session layer



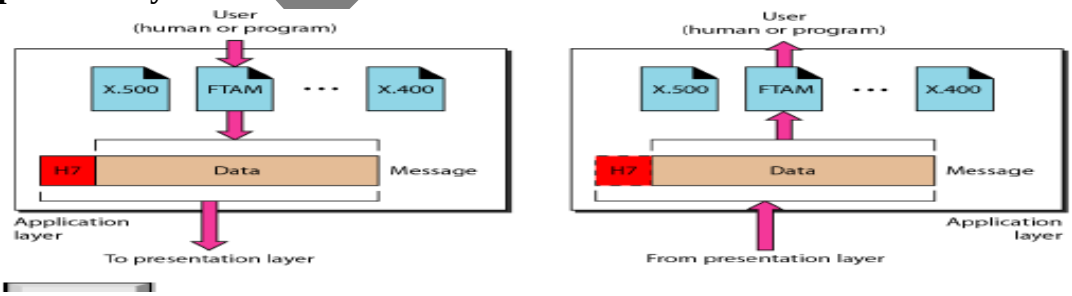
The session layer is responsible for dialog control and synchronization.

Presentation layer



The presentation layer is responsible for translation, compression, and encryption.

Application layer



The application layer is responsible for providing services to the user.

Services:

- Network virtual terminal
- Mail services
- File transfer, access, and management
- Directory services

The TCP/IP Reference Model:

The TCP/IP reference model was developed prior to OSI model. The major design goals of this model were,

1. To connect multiple networks together so that they appear as a single network.

2. To survive after partial subnet hardware failures.

3. To provide a flexible architecture.

Unlike OSI reference model, TCP/IP reference model has only 4 layers. They are,

1. Host-to-Network Layer

2. Internet Layer

3. Transport Layer

4. Application Layer
Application Layer
Transport Layer
Internet Layer Host-to-Network Layer

Host-to-Network Layer:

The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network to network.

Internet Layer:

This layer, called the internet layer, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have they travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The internet layer defines an official packet format and protocol called IP (Internet Protocol). The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to say that the TCP/IP internet layer is similar in functionality to the OSI network layer. Fig. shows this correspondence.

The Transport Layer:

The layer above the internet layer in the TCP/IP model is now usually called the transport layer. It is designed to allow peer entities on the source and destination hosts to carry on a conversation, just as in the OSI transport layer. Two end-to-end transport protocols have been defined here. The first one, TCP (Transmission Control Protocol), is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one on to the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

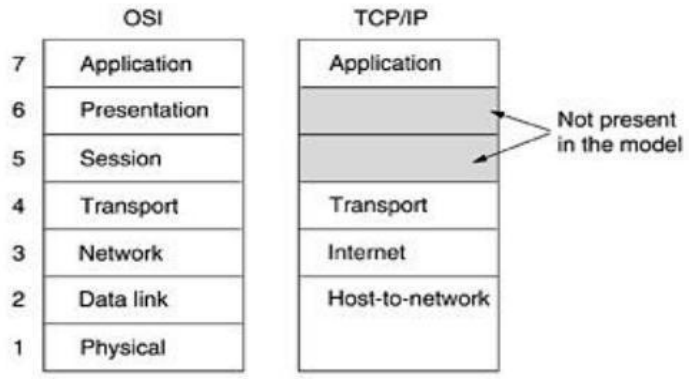


Fig.1: The TCP/IP reference model.

The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and

wish to provide their own. It is also widely used for one-shot, client-server-type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown in Fig.2. Since the model was developed, IP has been implemented on many other networks.

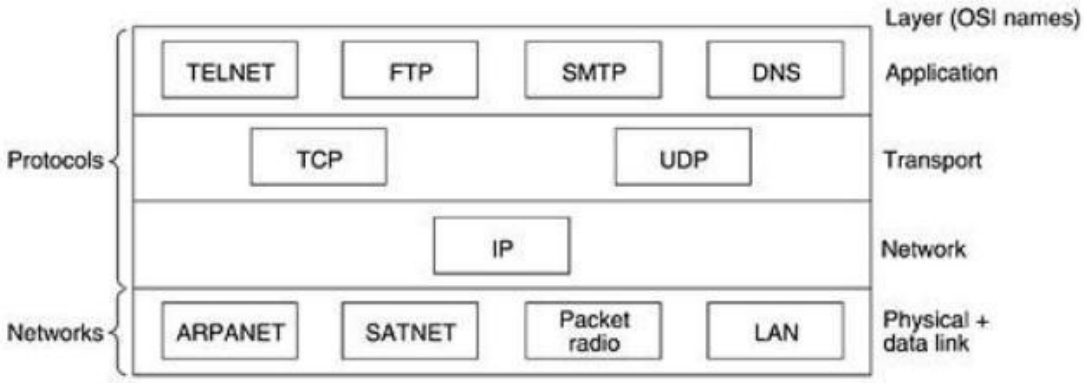


Fig.2: Protocols and networks in the TCP/IP model initially.

The Application Layer:

The TCP/IP model does not have session or presentation layers. On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig.6.2. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it. Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the

protocol for moving USENET news articles around, and HTTP, the protocol for fetching pages on the World Wide Web, and many others.

2.3 Local Area Networks

Why LANs?

In the modern office environment, each worker is equipped with a personal computer, containing its own disk drives and processor. Each of these computers can communicate with another by the way of a local area network (LAN), which is a computer network that covers a small area, usually a single building or group of buildings. In addition, the LAN may also connect the network of computers with a series of printers, a mainframe computer or file server with even greater processing power and memory storage, and with other devices that can send messages from the network over telephone lines to another location.

As the name suggests, a LAN is local, meaning that it is a proprietary system limited to a finite number of users. It generally serves an area of less than one mile. It is also a network, affording users both functional and communicative diversity through a distribution of resources. A LAN permits workers—isolated in separate offices—to operate off the same system, as if they were all sitting around a single computer.

One of the great attributes of a LAN is that it may be installed simply, upgraded or expanded with little difficulty, and moved or rearranged without disruption. LANs are also useful because they can transmit data quickly. Perhaps most importantly, anyone familiar with the use of a personal computer can be trained to communicate or perform work over a LAN. But despite their great potential and capabilities, LANs have yet to demonstrate an increase in office productivity. They have certainly eliminated paper and speeded the flow of information, but in many cases they have also created additional work in terms of organization, maintenance, and trouble-shooting.

THE HISTORY OF LANS

The advent of personal computers changed the type of information sent over office computer networks. Terminals were no longer "dumb," but contained the power to perform their own instructions and maintain their own memories. This took considerable pressure off mainframe devices, whose energies could now be devoted to more complex tasks.

LANs allowed for the transmission of data between workers. In turn, they enabled this shared data to be directed to a common printer, serving a larger group of users. This eliminated the need

for each worker to have a printer and ensured that the one printer provided was not underutilized. In addition, LANs allowed data to be called up directly on other workers' computers, providing immediate communication and eliminating the need for paper. The most common application was in interoffice communications, or electronic mail (e-mail). Messages could be directed to one or several people and copied to several more over the LAN. As a result, an e-mail system became something of an official record of communications between workers. Addressees became obligated to respond to e-mail messages in a timely manner because their failure to answer could be easily documented for supervisors.

Personal computers transformed LANs from mere shared processors to fully integrated communication devices. With processing power distributed among several computers, the mainframe's main role was eclipsed and complex processing, administrative functions, and data file storage became the job of a new device, the file server. Today, there are many different types of LANs. For example, many Macintosh computers use Appletalk, while IBM computers commonly use Ethernets.

PHYSICAL COMPONENTS OF LANS

The physical properties of a LAN include network access units (or interfaces) that connect the personal computer to the network. These units are actually interface cards installed on computer motherboards. Their job is to provide a connection, monitor availability of access to the LAN, set or buffer the data transmission speed, ensure against transmission errors and collisions, and assemble data from the LAN into usable form for the computer.

The next part of a LAN is the wiring, which provides the physical connection from one computer to another, and to printers and file servers. The properties of the wiring determine transmission speeds. The first LANs were connected with coaxial cable, the same type used to deliver cable television. These facilities are relatively inexpensive and simple to attach. More importantly, they provided great bandwidth (the system's rate of data transfer), enabling transmission speeds initially up to 20 megabits per second.

Another type of wiring, developed in the 1980s, used ordinary twisted wire pair (commonly used for telephones). The primary advantages of twisted wire pair are that it is very cheap, simpler to splice than coaxial, and is already installed in many buildings. The downside of this simplicity is that its bandwidth is more limited.

A more recent development in LAN wiring is optical fiber cable. This type of wiring uses thin strands of glass to transmit pulses of light between terminals. It provides tremendous bandwidth, allowing very high transmission speeds and because it is optical rather than electronic, it is impervious to electromagnetic interference. Still, splicing it can be difficult and requires a high

degree of skill. The primary application of fiber is not between terminals, but between LAN buses (terminals) located on different floors. As a result, fiber distributed data interface is used mainly in building risers. Within individual floors, LAN facilities remain coaxial or twisted wire pair.

When a physical connection cannot be made between two LANs, such as across a street or between buildings, microwave radio may be used. However, it is often difficult to secure frequencies for this medium. Another alternative in this application is light transceivers, which project a beam of light similar to fiber optic cable, but through the air rather than over cable. These systems do not have the frequency allocation or radiation problems associated with microwave, but they are susceptible to interference from fog and other natural obstructions.

LAN TOPOLOGIES

LANs are designed in several different topologies, or physical patterns, connecting terminals. These shapes can range from straight lines to a ring. Each terminal on the LAN contends with other terminals for access to the system. When it has secured access to the system, it broadcasts its message to all the terminals at once. The message is picked up by the one or group of terminal stations for which it is intended. The branching tree topology is an extension of the bus, providing a link between two or more buses.

A third topology, the star network, also works like a bus in terms of contention and broadcast. But in the star, stations are connected to a single, central node (individual computer) that administers access. Several of these nodes may be connected to one another. For example, a bus serving six stations may be connected to another bus serving 10 stations and a third bus connecting 12 stations. The star topology is most often used where the connecting facilities are coaxial or twisted wire pair.

The ring topology connects each station to its own node, and these nodes are connected in a circular fashion. Node 1 is connected to node 2, which is connected to node 3, and so on, and the final node is connected back to node 1. Messages sent over the LAN are regenerated by each node, but retained only by the addressees. Eventually, the message circulates back to the sending node, which removes it from the stream.

TRANSMISSION METHODS USED BY LANS

LANs function because their transmission capacity is greater than any single terminal on the system. As a result, each station terminal can be offered a certain amount of time on the LAN, like a timesharing arrangement. To economize on this small window of opportunity, stations organize their messages into compact packets that can be quickly distributed. When two

messages are sent simultaneously, they could collide on the LAN causing the system to be temporarily disrupted. Busier LANs usually utilize special software that virtually eliminates the problem of collisions by providing orderly, non-contention access.

The transmission methods used on LANs are either baseband or broadband. The baseband medium uses a high-speed digital signal consisting of square wave DC voltage. While it is fast, it can accommodate only one message at a time. As a result, it is suitable for smaller networks where contention is low. It also is very simple to use, requiring no tuning or frequency discretion circuits. This transmission medium may be connected directly to the network access unit and is suitable for use over twisted wire pair facilities.

By contrast, the broadband medium tunes signals to special frequencies, much like cable television. Stations are instructed by signaling information to tune to a specific channel to receive information. The information within each channel on a broadband medium may also be digital, but they are separated from other messages by frequency. As a result, the medium generally requires higher capacity facilities, such as coaxial cable. Suited for busier LANs, broadband systems require the use of tuning devices in the network access unit that can filter out all but the single channel it needs.

THE FILE SERVER

The administrative software of the LAN resides either in a dedicated file server; in a smaller, less busy LAN; or in a personal computer that acts as a file server. In addition to performing as a kind of traffic controller, the file server holds files for shared use in its hard drives, administers applications such as the operating system, and allocates functions.

When a single computer is used as both a workstation and a file server, response times may lag because its processors are forced to perform several duties at once. This system will store certain files on different computers on the LAN. As a result, if one machine is down, the entire system may be crippled. If the system were to crash due to undercapacity, some data may be lost or corrupted.

The addition of a dedicated file server may be costly, but it provides several advantages over a distributed system. In addition to ensuring access even when some machines are down, its only duties are to hold files and provide access.

OTHER LAN EQUIPMENT

LANs are generally limited in size because of the physical properties of the network including distance, impedance, and load. Some equipment, such as repeaters, can extend the range of a

LAN. Repeaters have no processing ability, but simply regenerate signals that are weakened by impedance. Other types of LAN equipment with processing ability include gateways, which enable LANs operating dissimilar protocols to pass information by translating it into a simpler code, such as ASCII. A bridge works like a gateway, but instead of using an intermediate code, it translates one protocol directly into another. A router performs essentially the same function as a bridge, except that it administers communications over alternate paths. Gateways, bridges, and routers can act as repeaters, boosting signals over greater distances. They also enable separate LANs located in different buildings to communicate with each other.

The connection of two or more LANs over any distance is referred to as a wide area network (WAN). WANs require the use of special software programs in the operating system to enable dial-up connections that may be performed by a telephone lines or radio waves. In some cases, separate LANs located in different cities—and even separate countries—may be linked over the public network.

LAN DIFFICULTIES

LANs are susceptible to many kinds of transmission errors. Electromagnetic interference from motors, power lines, and sources of static, as well as shorts from corrosion, can corrupt data. Software bugs and hardware failures can also introduce errors, as can irregularities in wiring and connections. LANs generally compensate for these errors by working off an uninterruptable power source, such as batteries, and using backup software to recall most recent activity and hold unsaved material. Some systems may be designed for redundancy, such as keeping two file servers and alternate wiring to route around failures.

Security problems can also be an issue with LANs. They can be difficult to manage and access because the data they use is often distributed between many different networked sources. In addition, many times this data is stored on several different workstations and servers. Most companies have specific LAN administrators who deal with these issues and are responsible for the use of LAN software. They also work to backup files and recover lost files.

Primary attributes of a LAN

AN's main function is to provide resource sharing and mutual communication, which provides the following main services:

- **The sharing of resources**, including hardware resource sharing, file sharing, and software inventory data sharing. Users can share a network system software and application software.

- **Data transfer and e-mail**: Data and network file transfer is an important feature of modern LANs not only transmit files, data, information, but also can send voice, images.
- **Improve the reliability** of the computer system. LAN computers can back each other, avoiding the stand-alone system without backup failure may occur when system failures, greatly improving the reliability of the system, particularly in the industrial process control, real-time data processing and other applications, is particularly important.
- **Easy to distributed processing**: Use of network technology you can have more than one computer connected to a high-performance computer system (Server) through a certain algorithm, the larger global issues points to a different computer to complete.

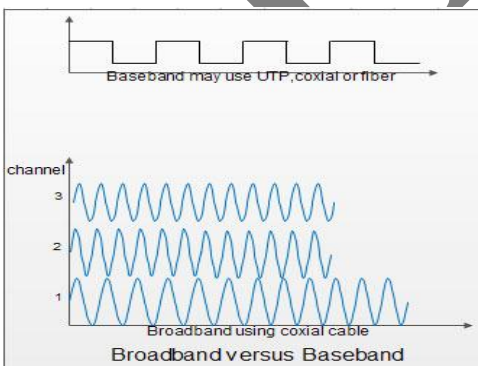
Broadband and baseband and base LANs

Broadband A communications medium that can carry a wide range of signal frequencies, typically from audio up to video frequencies. In telecommunications the significance of a broadband system is that it can carry television and videoconferencing data as well as voice calls. A broadband medium can be made to carry many signals at once by apportioning its total bandwidth into many independent channels, each of which carries only a specific range of frequencies. In contrast, a BASEBAND can carry only a single channel. ATM, ADSL and Cable TV are all broadband media, while standard ISDN barely qualifies.

There exist two LAN transmission options, Baseband and Broadband. Baseband LANs, which is the most prevalent by far, is a Single-channel system that supports a single transmission at any given time.

Broadband LANs, which are most unusual, support multiple transmissions via multiple frequency channels.

Broadband LANs



Broadband LANs are Multichannel, analog LANs as shown in Figure. typically based on coaxial cable as the transmission medium, although fiber optic cable is also used. Individual channels offer bandwidth of 1 to 5 Mbps, with 20 to 30 channels typically supported. Aggregate bandwidth is as much as 500 MHz. Its characteristics are:

• Digital signal modulated onto RF carrier (analog)

• Channel allocation based on FD M

• Head-End for bidirectional transmission

• Stations connected via RF modems, Le. radio modems accomplish the digital-to-analog conversion process, providing the transmitting device access to an analog channel.

Advantages

Data, voice and video can be accommodated on broadband channel

• Greater distances

• Greater bandwidth.

Disadvantages

• Cable design

• Alignment and maintenance

• High cost, requires modems

• Lack of well-developed standards.

Some broadband LANs are referred to as IOBroadband36 where 10stands for 10Mbps, Broadband for multichannel and 36 for 3600 meters maximum separation between devices.

Baseband LANs

Baseband LAN is single channel, supporting a single communication at a time as shown in Figure. They are digital in nature. Total bandwidth of I to 100Mbps is provided over coaxial cable, UTP, STP, or fiber optic cable. Distance limitations depend on the medium employed and

the specifics of the LAN protocol. Baseband LAN physical topologies including ring, bus, tree, and star.

Baseband LANs are by far the most popular and the most highly standardized. Ethernet, Token Passing, Token Ring and FDDI LANs are all baseband. They are intended only for data, as data communications is, after all, the primary reason for the existence of LANs. The characteristics of this system may be summarized as follows:

• Unmodulated digital signal

• Single channel

• Bidirectional propagation of signal

• Stations connected, via T connectors

• No need of modems - low cost installation.

Advantages

• Simplicity

• Low cost

• Ease of installation and maintenance

• High rates.

Disadvantages

• Limited distances

• Data and voice only.

IEEE LAN standards

Set of network standards developed by the IEEE. They include:

- **IEEE 802.1:** Standards related to network management.

- **IEEE 802.2:** General standard for the data link layer in the OSI Reference Model. The IEEE divides this layer into two sublayers -- the *logical link control (LLC) layer* and the *media access control (MAC) layer*.The MAC layer varies for different network types and is defined by standards IEEE 802.3 through IEEE 802.5.
- **IEEE 802.3:** Defines the MAC layer for bus networks that use CSMA/CD. This is the basis of the Ethernet standard. Also see the Ethernet Designations chart in the Quick Referencesection of Webopedia.
- **IEEE 802.4:** Defines the MAC layer for bus networks that use a token-passing mechanism (token bus networks).
- **IEEE 802.5:** Defines the MAC layer for token-ring networks.
- **IEEE 802.6:** Standard for Metropolitan Area Networks (MANs).

Re1ationship of the 802 standards to the ISO/CCITT model

The IEEE 802 LAN Standards Family

The IEEE 802 Local and Metropolitan Area Network Standards Committee is a major working group charted by IEEE to create, maintain, and encourage the use of IEEE and equivalent IEC/ISO standards. The IEEE formed the committee in February 1980, and this committee meets as a plenary body at least three times per year. The IEEE 802 committee produces the series of standards known as IEEE 802.x, and the JTC 1 series of equivalent standards is known as ISO 8802-nnn.

IEEE 802 includes a family of standards, as depicted in Figure 3.3. The MAC and Physical layers of the 802 standard were organized into a separate set of standards from the LLC because of the interdependence between medium access control, medium, and topology.

2.4 Connection options with LANs, LLC and MAC protocol data units

IEEE 802.2 LLC Overview

The LLC is the highest layer of the IEEE 802 Reference Model and provides functions similar to the traditional data link control protocol: HDLC (High-Level Data Link Control). ISO/IEC 8802-2 (ANSI/IEEE Standard 802.2), dated May 7, 1998, specifies the LLC. The purpose of the LLC is to exchange data between end users across a LAN using an 802-based MAC controlled link. The LLC provides addressing and data link control, and it is independent of the topology, transmission medium, and medium access control technique chosen.

Higher layers, such as TCP/IP, pass user data down to the LLC expecting error-free transmission across the network. The LLC in turn appends a control header, creating an LLC protocol data unit (PDU). The LLC uses the control information in the operation of the LLC protocol (see Figure 3.4). Before transmission, the LLC PDU is handed down through the MAC service access point (SAP) to the MAC layer, which appends control information at the beginning and end of the packet, forming a MAC frame. The control information in the frame is needed for the operation of the MAC protocol.

Figure 3.4 The LLC provides end-to-end link control over an 802.11-based wireless LAN.

IEEE 802.2 LLC Services

The LLC provides the following three services for a Network Layer protocol:

- Unacknowledged connectionless service
- Connection-oriented service
- Acknowledged connectionless service

These services apply to the communication between peer LLC layers--that is, one located on the source station and one located on the destination station. Typically, vendors will provide these services as options that the customer can select when purchasing the equipment.

All three LLC protocols employ the same PDU format that consists of four fields (see Figure 3.5). The Destination Service Access Point (DSAP) and Source Service Access Point (SSAP) fields each contain 7-bit addresses that specify the destination and source stations of the peer LLCs. One bit of the DSAP indicates whether the PDU is intended for an individual or group station(s). One bit of the SSAP indicates whether it is a command or response PDU. The format of the LLC Control field is identical to that of HDLC, using extended (7-bit) sequence numbers. The Data field contains the information from higher-layer protocols that the LLC is transporting to the destination.

Figure 3.5 The LLC PDU consists of data fields that provide the LLC functionality.

The Control field has bits that indicate whether the frame is one of the following types:

- **Information** Used to carry user data.
- **Supervisory** Used for flow control and error control.
- **Unnumbered** Various protocol control PDUs.

Unacknowledged Connectionless Service

The *unacknowledgedconnectionless service* is a datagram-style service that does not involve any error-control or flow-control mechanisms. This service does not involve the establishment of a data link layer connection (such as between peer LLCs). This service supports individual, multicast, and broadcast addressing. This service simply sends and receives LLC PDUs with no acknowledgement of delivery. Because the delivery of data is not guaranteed, a higher layer, such as TCP, must deal with reliability issues.

The unacknowledged connectionless service offers advantages in the following situations:

- If higher layers of the protocol stack provide the necessary reliability and flow-control mechanisms, then it would be inefficient to duplicate them in the LLC. In this case, the unacknowledged connectionless service would be appropriate. TCP and the ISO transport protocol, for example, already provide the mechanisms necessary for reliable delivery.
- It is not always necessary to provide feedback pertaining to successful delivery of information. The overhead of connection establishment and maintenance can be inefficient for applications involving the periodic sampling of data sources, such as monitoring sensors. The unacknowledged connectionless service would best satisfy these requirements.

Connection-Oriented Service

The *connection-oriented service* establishes a logical connection that provides flow control and error control between two stations needing to exchange data. This service does involve the establishment of a connection between peer LLCs by performing connection establishment, data transfer, and connection termination functions. The service can connect only two stations; therefore, it does not support multicast or broadcast modes. The connection-oriented service offers advantages mainly if higher layers of the protocol stack do not provide the necessary reliability and flow-control mechanisms, which is generally the case with terminal controllers.

Flow control is a protocol feature that ensures that a transmitting station does not overwhelm a receiving station with data. With flow control, each station allocates a finite amount of memory and buffer resources to store sent and received PDUs.

Networks, especially wireless networks, suffer from induced noise in the links between network stations that can cause transmission errors. If the noise is high enough in amplitude, it causes errors in digital transmission in the form of altered bits. This will lead to inaccuracy of the transmitted data, and the receiving network device may misinterpret the meaning of the information.

The noise that causes most problems with networks is usually Gaussian and impulse noise. Theoretically, the amplitude of Gaussian noise is uniform across the frequency spectrum, and it normally triggers random single-bit independent errors. Impulse noise, the most disastrous, is characterized by long quiet intervals of time followed by high amplitude bursts. This noise results from lightning and switching transients. Impulse noise is responsible for most errors in digital communication systems and generally provokes errors to occur in bursts.

To guard against transmission errors, the connection-oriented and acknowledged-connectionless LLCs use error control mechanisms that detect and correct errors that occur in the transmission of PDUs. The LLC ARQ mechanism recognizes the possibility of the following two types of errors:

- **Lost PDU** A PDU fails to arrive at the other end or is damaged beyond recognition.
- **Damaged PDU** A PDU has arrived, but some bits are altered.

When a frame arrives at a receiving station, the station checks whether there are any errors present by using a *Cyclic Redundancy Check(CRC)* error detection algorithm. In general, the receiving station will send back a positive or negative acknowledgement, depending on the outcome of the error detection process. In case the acknowledgement is lost in route to the sending station, the sending station will retransmit the frame after a certain period of time. This process is often referred to as *Automatic Repeat Request(ARQ)*.

Overall, ARQ is best for the correction of burst errors because this type of impairment occurs in a small percentage of frames, thus not invoking many retransmissions. Because of the feedback inherent in ARQ protocols, the transmission links must accommodate half-duplex or full-duplex transmissions. If only simplex links are available because of feasibility, then it is impossible to use the ARQ technique because the receiver would not be able to notify the transmitter of bad data frames.

The following are two approaches for retransmitting unsatisfactory blocks of data using ARQ:

- **Continuous ARQ** With this type of ARQ, often called a *sliding window protocol*, the sending station transmits frames continuously until the receiving station detects an error. The sending station is usually capable of transmitting a specific number of frames and maintains a table indicating which frames have been sent.

  The system implementor can set the number of frames sent before stopping via configuration parameters of the network device. If a receiver detects a bad frame, it will send a negative acknowledgement back to the sending station requesting that the bad frame be sent again. When the transmitting station gets the signal to retransmit the frame,

several subsequent frames may have already been sent (due to propagation delays between the sender and receiver); therefore, the transmitter must go back and retransmit the bad data frame.

There are a couple of ways the transmitting station can send frames again using continuous ARQ. One method is for the source to retrieve the bad frame from the transmit buffer and send it and all frames following it. This is called the *go-back*-n *technique*. A problem is that when *n* (the number of frames the transmitter sent after the bad frame plus one) becomes large, the method becomes inefficient. This is because the retransmission of just one frame means that a large number of possibly good frames will also be resent, thus decreasing throughput.

The go-back-*n* technique is useful in applications for which receiver buffer space is limited because all that is needed is a receiver window size of one (assuming frames are to be delivered in order). When the receive node rejects a bad frame (sends a negative acknowledgment), it does not need to buffer any subsequent frames for possible reordering while it is waiting for the retransmission, because all subsequent frames will also be sent.

An alternative to the continuous go-back-*n* technique is a method that selectively retransmits only the bad frame, then resumes normal transmission at the point just before getting the notification of a bad frame. This approach is called *selective repeat.* It is obviously better than continuous go-back-*n* in terms of throughput because only the bad frame needs retransmission. With this technique, however, the receiver must be capable of storing a number of frames if they are to be processed in order. The receiver needs to buffer data that has been received after a bad frame was requested for retransmission since only the damaged frame will be sent again.

- Stop-and-wait ARQ With this method, the sending station transmits a frame, then stops and waits for some type of acknowledgment from the receiver on whether a particular frame was acceptable or not. If the receiving station sends a negative acknowledgment, the frame will be sent again. The transmitter will send the next frame only after it receives a positive acknowledgment from the receiver.

  An advantage of stop-and-wait ARQ is that it does not require much buffer space at the sending or receiving station. The sending station needs to store only the current transmitted frame. However, stop-and-wait ARQ becomes inefficient as the propagation delay between source and destination becomes large. For example, data sent on satellite links normally experiences a round-trip delay of several hundred milliseconds; therefore, long block lengths are necessary to maintain a reasonably effective data rate. The trouble

is that with longer frames, the probability of an error occurring in a particular block is greater. Thus, retransmission will occur often, and the resulting throughput will be lower.

Acknowledged Connectionless Service

As with the The unacknowledged connectionless service, the *acknowledged connectionless service* does not involve the establishment of a logical connection with the distant station. But the receiving stations with the acknowledged version do confirm successful delivery of datagrams. Flow and error control is handled through use of the stop-and-wait ARQ method.

The acknowledged connectionless service is useful in several applications. The connection-oriented service must maintain a table for each active connection for tracking the status of the connection. If the application calls for guaranteed delivery, but there is a large number of destinations needing to receive the data, then the connection-oriented service may be impractical because of the large number of tables required. Examples that fit this scenario include process control and automated factory environments that require a central site to communicate with a large number of processors and programmable controllers. In addition, the handling of important and time-critical alarm or emergency control signals in a factory would also fit this case. In all these examples, the sending station needs an acknowledgment to ensure successful delivery of the data; however, an urgent transmission cannot wait for a connection to be established.

LLC/MAC Layer Service Primitives

Layers within the 802 architecture communicate with each other via service primitives having the following forms:

- **Request** A layer uses this type of primitive to request that another layer perform a specific service.
- **Confirm** A layer uses this type of primitive to convey the results of a previous service request primitive.
- **Indication** A layer uses this type of primitive to indicate to another layer that a significant event has occurred. This primitive could result from a service request or from some internally generated event.
- **Response** A layer uses this type of primitive to complete a procedure initiated by an indication primitive.

These primitives are an abstract way of defining the protocol, and they do not imply a specific physical implementation method. Each layer within the 802 model uses specific primitives. The LLC layer communicates with its associated MAC layer through the following specific set of service primitives:

- **MA-UNITDATA.request**The LLC layer sends this primitive to the MAC layer to request the transfer of a data frame from a local LLC entity to a specific peer LLC entity or group of peer entities on different stations. The data frame could be an information frame containing data from a higher layer or a control frame (such as a supervisory or unnumbered frame) that the LLC generates internally to communicate with its peer LLC.
- **MA-UNITDATA.indication**The MAC layer sends this primitive to the LLC layer to transfer a data frame from the MAC layer to the LLC. This occurs only if the MAC has found that a frame it receives from the Physical layer is valid and has no errors and the destination address indicates the correct MAC address of the station.

2.5 LAN topologies and protocols

Network topologies can take a bit of time to understand when you're all new to this kind of cool stuff, but it's very important to fully understand them as they are key elements to understanding and troubleshooting networks and will help you decide what actions to take when you're faced with network problems.

This article explains the different network topologies found in today's networks. We examine Bus Topology, Ring Topology, Star Topology, Mesh Topology, Hybrid Topology and many more.

Physical and Logical Topologies

There are two types of topologies: Physical and Logical. The physical topology of a network refers to the layout of cables, computers and other peripherals. Try to imagine yourself in a room with a small network, you can see network cables coming out of every computer that is part of the network, then those cables plug into a hub or switch. What you're looking at is the physical topology of that network !

Logical topology is the method used to pass the information between the computers. In other words, looking at that same room, if you were to try to see how the network works with all the computers talking (think of the computers generating traffic and packets of data going everywhere on the network) you would be looking at the logical part of the network. The way the computers will be talking to each other and the direction of the traffic is controlled by the various protocols (like Ethernet) or, if you like, rules.

If we used token ring, then the physical topology would have to change to meet the requirements of the way the token ring protocol works (logically).

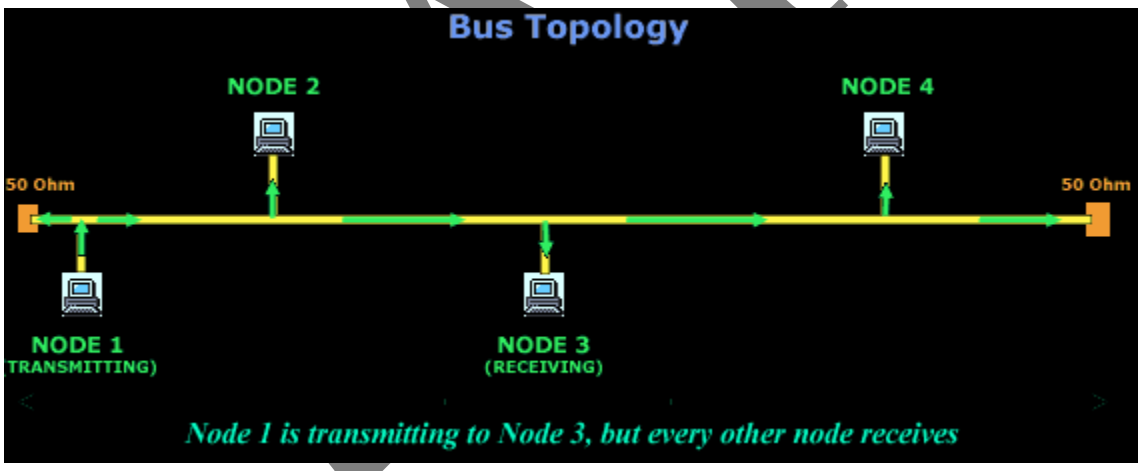
If it's all still confusing, consider this: The physical topology describes the layout of the network, just like a map shows the layout of various roads, and the logical topology describes how the data is sent accross the network or how the cars are able to travel (the direction and speed) at every road on the map.

The most common types of physical topologies, which we are going to analyse, are: Bus, Hub/Star and Ring

The Physical Bus Topology

Bus topology is fairly old news and you probably won't be seeing much of these around in any modern office or home.

With the Bus topology, all workstations are connect directly to the main backbone that carries the data. Traffic generated by any computer will travel across the backbone and be received by all workstations. This works well in a small network of 2-5 computers, but as the number of computers increases so will the network traffic and this can greatly decrease the performance and available bandwidth of your network.



As you can see in the above example, all computers are attached to a continuous cable which connects them in a straight line. The arrows clearly indicate that the packet generated by Node 1 is transmitted to all computers on the network, regardless the destination of this packet.

Also, because of the way the electrical signals are transmitted over this cable, its ends must be terminated by special terminators that work as "shock absorbers", absorbing the signal so it won't reflect back to where it came from. The value of 50Ohms has been selected after carefully taking

in consideration all the electrical characteristics of the cable used, the voltage that the signal which runs through the cables, the maximum and minimum length of the bus and a few more.

If the bus (the long yellow cable) is damaged anywhere in its path, then it will most certainly cause the network to stop working or, at the very least, cause big communication problems between the workstations.

Thinnet - 10 Base2, also known as coax cable (Black in colour) and Thicknet - 10 Base 5 (Yellow in colour) is used in these type of topologies.

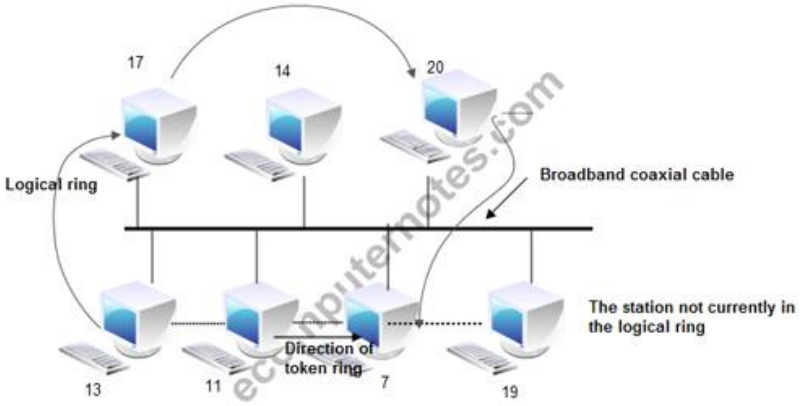
2.6 Token ring (Priority), token bus and IEEE 802.4

IEEE 802.4 Token Bus : In token bus Computer network station must have possession of a token before it can transmit on the computer network. The IEEE 802.4 Committee has defined **token bus** standards as broadband computer networks, as opposed to Ethernet's baseband transmission technique. Physically, the token bus is a linear or tree-shape cable to which the stations are attached

The topology of the computer network can include groups of workstations connected by long trunk cables. Logically, the stations are organized into a ring. These workstations branch from hubs in a star configuration, so the network has both a bus and star topology. Token bus topology is well suited to groups of users that are separated by some distance. **IEEE 802.4 token bus networks are constructed with 75-ohm coaxial cable using a bus topology**. The broadband characteristics of the 802.4 standard support transmission over several different channels simultaneously.

When the logical ring is initialized, the highest numbered station may send the first frame. The token and frames of data are passed from one station to another following the numeric sequence of the station addresses. Thus, the token follows a logical ring rather than a physical ring. The last station in numeric order passes the token back to the first station. The token does not follow the physical ordering of workstation attachment to the cable. Station 1 might be at one end of the cable and station 2 might be at the other, with station 3 in the middle.

In such a case, there is no collision as only one station possesses a token at any given time. In token bus, each station receives each frame; the station whose address is specified in the frame processes it and the other stations discard the frame.



A Token Bus

MAC Sublayer Function

• When the ring is initialized, stations are inserted into it in order of station address, from highest to lowest.

• Token passing is done from high to low address.

• Whenever a station acquires the token, it can transmit frames for a specific amount of time.

• If a station has no data, it passes the token immediately upon receiving it.

• The token bus defines four priority classes, 0, 2, 4, and 6 for traffic, with 0 the lowest and 6 the highest.

• Each station is internally divided into four substations, one at each priority level *i.e.* 0,2,4 and 6.

• As input comes in to the MAC sublayer from above, the data are checked for priority and routed to one of the four substations.

• Thus each station maintains its own queue of frames to be transmitted.

• When a token comes into the station over the cable, it is passed internally to the priority 6 substation, which can begin transmitting its frames, if it has any.

• When it is done or when its time expires, the token is passed to the priority 4 substation, which can then transmit frames until its timer expires. After this the token is then passed internally to priority 2 substation.

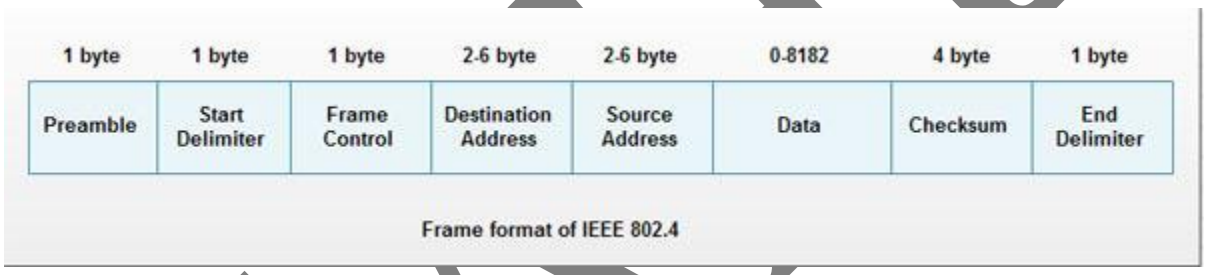
• This process continues until either the priority 0 substation has sent all its frames or its time expires.

• After this the token is passed to the next station in the ring.

Frame format of Token Bus

The various fields present in the frame format are

1. **Preamble**: This. Field is at least 1 byte long. It is used for bit synchronization.



Frame format of IEEE 802.4

2. **Start Delimiter**: This one byte field marks the beginning of frame.

3. **Frame Control:** This one byte field specifies the type of frame. It distinguishes data frame from control frames. For data frames it carries frame's priority. For control frames, it specifies the frame type. The control frame types include. token passing and various ring maintenance frames, including the mechanism for letting new station enter the ring, the mechanism for allowing stations to leave the ring.

4. **Destination address**: It specifies 2 to 6 bytes destination address.

5. **Source address**: It specifies 2 to 6 bytes source address.

6. **Data**: This field may be upto 8182 bytes long when 2 bytes addresses are used &upto 8174 bytes long when 6 bytes address is used.

7. **Checksum**: This 4 byte field detects transmission errors.

8. **End Delimiter**: This one byte field marks the end of frame.

The various control frames used in token bus are:

Farme Control Field	Name	Meaning
00000000	Claim_token	Claim token during ring initialization
00000001	Solicit successor_1	Allow station to enter the ring
00000010	Solicit successor_2	Allow stations to enter the ring
00000011	Who_follows	Recover from lost token.
00000100	Resolve_contention	Used when multiple stations want to enter.
00001000	Token	Pass the token
00001100	Set successor	Allow station to leave the ring.

2.7 Metropolitan area networks (MANs)

A metropolitan area network (MAN) is a network that interconnects users with computer resources in a geographic area or region larger than that covered by even a large local area network (LAN) but smaller than the area covered by a wide area network (WAN). The term is applied to the interconnection of networks in a city into a single larger network (which may then also offer efficient connection to a wide area network). It is also used to mean the interconnection of several local area networks by bridging them with backbone lines. The latter usage is also sometimes referred to as a campus network.

2.8 ANSI fiber distributed data interface.

A high-speed network technology, conforming to the Open Systems Interconnection (OSI) reference model for networking and the American National Standards Institute (ANSI) standard X3T9, which runs at 100 Mbps over fiber-optic cabling; often used for network backbones in a local area network (LAN) or metropolitan area network (MAN).

How FDDI Works

Fiber Distributed Data Interface (FDDI) is usually implemented as a dual token-passing ring within a ring topology (for campus networks) or star topology (within a building). The dual ring consists of a primary and secondary ring. The primary ring carries data. The counter-rotating secondary ring can carry data in the opposite direction, but is more commonly reserved as a

backup in case the primary ring goes down. This provides FDDI with the degree of fault tolerance necessary for network backbones. In the event of a failure on the primary ring, FDDI automatically reconfigures itself to use the secondary ring as shown in the illustration. Faults can be located and repaired using a fault isolation technique called beaconing. However, the secondary ring can also be configured for carrying data, extending the maximum potential bandwidth to 200 Mbps.

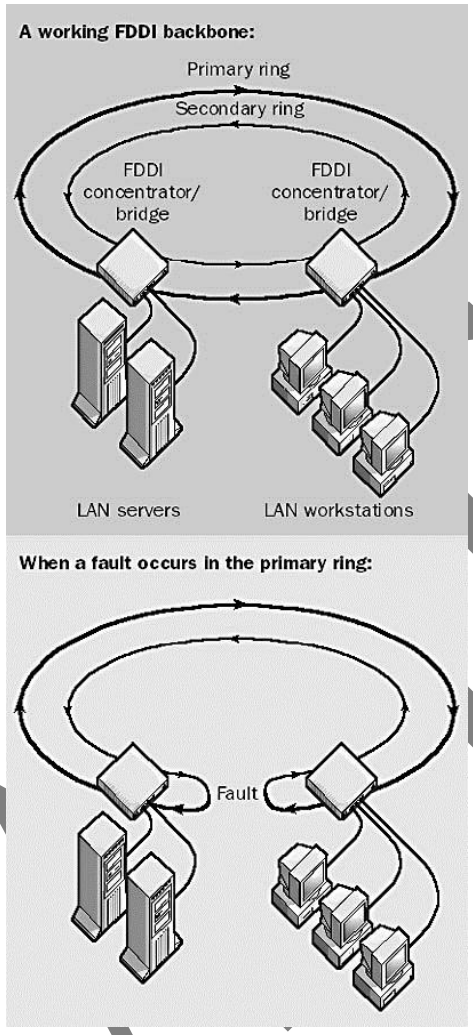
Stations connect to one (or both) rings using a media interface connector (MIC). Its two fiber ports can be either male or female, depending on the implementation. There are two different FDDI implementations, depending on whether stations are attached to one or both rings:

- **Single-attached stations (Class B stations):**

  Connect to either the primary or secondary ring using M ports. Single-attached FDDI uses only the primary ring and is not as commonly deployed for network backbones as dual-attached FDDI. Single-attached stations are used primarily to connect Ethernet LANs or individual servers to FDDI backbones.

- **Dual-attached stations (Class A stations):**

  Connect to both rings. The A port is the point at which the primary ring enters and the secondary ring leaves; the B port is the reverse. M ports provide attachment points for single-attached stations. Dual-attached FDDI uses both rings, with the secondary ring serving as a backup for the primary. Dual-attached FDDI is used primarily for network backbones that require fault tolerance. Single-attached stations can be connected to dual-attached FDDI backbones using a dual-attached device called a concentrator or multiplexer.



FDDI uses a timed token-passing technology similar to that of token ring networks as defined in the IEEE 802.5 standard. FDDI stations generate a token that controls the sequence in which other stations will gain access to the wire. The token passes around the ring, moving from one node to the next. When a station wants to transmit information, it captures the token, transmits as many frames of information as it wants (within the specified access period), and then releases the token. This feature of transmitting multiple data frames per token capture is known as a capacity allocation scheme, in contrast to the priority mechanism used in the IEEE 802.5 token ring standard. Every node on the ring checks the frames. The recipient station then reads the information from the frames, and when the frames return to the originating station, they are stripped from the ring.

There can be up to 500 stations on a dual-ring FDDI network. The maximum circumference for an FDDI ring is 100 kilometers (or 200 kilometers for both rings combined), and there must be a repeater every 2 kilometers or less. Bridges or routers are used to connect the FDDI backbone

network to Ethernet or token ring departmental LANs. For these reasons, FDDI is not often used as a wide area network (WAN) solution, but is more often implemented in campus-wide networks as a network backbone.

UNIT 2 - POSSIBLE QUESTIONS

PART A
(20 MARKS)
ONLINE EXAMIANTION

PART B
(Each Question carries 2 marks)

1. Mentionany four network design problems.
2. How many layers in the OSI model?
3. Write a status of OSI.
4. Mention about the primary attributes of LAN.
5. What are the connection options with LAN? Discuss it.
6. What is meant by the Token ring?
7. How to introduce standards organizations and the OSI model?
8. Write a short note on Wide area networks.
9. How to explain the LAN topologies and protocols in communication systems?
10. How the interface of ANSI Fiber Distributed data interface was made?

PART C

Compulsory Question

(Each Question Carries 8 marks)

1. Discuss about the classification of communication protocols.
2. Compare the OSI reference model with TCP/IP model.
3. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.
4. Explain in detail about the X.25 architecture with an example network.
5. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.

KARPAGAM ACADEMY OF HIGHER EDUCATION

DEPARTMENT OF COMPUTER APPLICATIONS

SUBJECT CODE/NAME: 17CAP302, COMPUTER NETWORKS

Degree : PG **Course: MCA** **Year: 2017**

S.NO	QUESTIONS	OPTION 1	OPTION 2	OPTION 3	OPTION 4	ANSWER
	UNIT 2					
1	The modern computer network is designed around the concept of layered protocols or _____	Procedures	Functions	Tasks	Layers	**Functions**
2	Networks require the writing of hundreds of software programs, entailing thousnads of _____	coding statements	Functions	Procedures	Packages	**Coding statements**
3	Certain major organizations have established _____ because of thier dominant position in the industry.	ISO Standards	Defacto standards	ARPA Standards	ISI standards	**Defacto Standards**
4	Each subsystems in the network may also be madeup of _____	Entities	Layers	Hosts	Tasks	**Entities**
5	_____ are invoked to and from the layer through identifiers called service access points.	Entities	Layers	Hosts	Primitives	**Primitives**
6	_____ user data transferred transparently by layer N+1 to layer N and subsequently to N-1.	SDU	PCI	PDU	ICI	**SDU**
7	_____ information excahnged by peer entities at different sites on the network to instruct an entity to performa a service function.	SDU	PCI	PDU	ICI	**PCI**
8	The combination of the SDU and PCI is _____	SDU	PCI	PDU	ICI	**PDU**
9	A temporary parameter passed between N and N-1 to invoke service functions_____	SDU	PCI	PDU	ICI	**ICI**
10	_____ is a fancy name of a header.	SDU	PCI	PDU	ICI	**PCI**
11	The total unit of information transferred across the layer boundaries _____	IDU	SDU	PDU	PCI	**IDU**
12	The ICI is used only between adjacent layers at the _____ node.	Same	Different	Enabled	disabled	**Same**
13	_____ is supported by the major standards organizations, telecommunications, and trade associations.	TCP	OSI	IP	TCP/IP	**OSI**
14	CCITT meant as _____	International Telegraph & Telephone Consultative Committee	Control Committe for Information Technology Team	Consulative Committee for International Telegraph & Telephone	Committee Consultaive for International Telegraph &	**International Telegraph & Telephone Consultative**
15	RPOA is meant by _____	Research Private Operating Agencies	Recognized Private Operating Agencies	Related Process Operating Agencies	Realtional Process Operating Agencies	**Recognized Private Operating Agencies**
16	ECMA is combination of _____	ISO & CCITT	ITU & ANSI	ANSI & ISO	ISO & ANSI	**ISO & CCITT**
17	The _____ is the primary US Organization on OSI.	ANSI	ISO	CCITT	ITU	**ANSI**
18	The _____ is deidcated to the development of standards applicable to computer and communication technology.	CCITT	ANSI	ECMA	ISO	**ECMA**
19	Several subcommittees within ECMA work actively with _____	CCITT & ISO	ITU & ANSI	ANSI & ISO	ISO & ANSI	**CCITT & ISO**

20	____ is active in developing standards for data communications for the OSI.	CCITT	ANSI	ECMA	ISO	**ANSI**
21	_____ is a national trade association which has been active for many years in the development of standards.	CCITT	ANSI	EIA	ISO	**EIA**
22	The _____ publishes its own standards and also submits proposals to ANSI for publication as US national standards.	EIA	ANSI	CCITT	ITU	**EIA**
23	The _____activity addresses local area network and many other standards as well.	EIA	ANSI	IEEE	ITU	**IEEE**
24	The _____is a consortium of federal agencies that have large telecommunications capabilities.	NCS	IEEE	EIA	ANSI	**NCS**
25	The lowest layer in the OSI model is ____	Application Layer	Physical Layer	Presentation Layer	Session Layer	**Physical Layer**
26	The _____ layer is resposible for the transfer of data over the channel.	Data Link	Physical	Application	Session	**Data Link**
27	The _____ specifies the interface of the user DTE into a packet switched network as well as interface of two DTEs with each other through a packet network.	Application Layer	Physical Layer	Network Layer	Session Layer	**Network Layer**
28	The _____ provides the interface between the data communications network and the upper three layers.	Application Layer	Transport Layer	Network Layer	Session Layer	**Transport Layer**
29	The _____ serves as user interface into the transport service layer.	Application Layer	Transport Layer	Network Layer	Session Layer	**Session Layer**
30	The _____ supports an end user application process.	Application Layer	Transport Layer	Network Layer	Session Layer	**Application Layer**
31	The _____industry has now matured and is relatively stable field.	WAN	LAN	MAN	Network	**WAN**
32	ECMA means _____	European Computer Manufacturers Association	Established Computer Manufacturers Association	Extended Computer Manufacturers Association	European Commerce Manufacturers Association	**European Computer Manufacturers Association**
33	The LAN transmission capacity is usually greater than that of a _____	WAN	Network	MAN	LAN	**WAN**
34	The ECMA voted to accept the 802.5_____ as its standard.	Network	Topology	Token ring	Layers	**Token ring**
35	The problem is particularly serve in the LAN with its high speed channels and _____	Low error rates	High error rates	Error rates	Moderate errors	**Low error rates**
36	The LLC and MAC sublayers use _____ to communicate.	TCE	PDU	DTE	DCE	**PDU**
37	The _____ field indicates the length of the LLC and data fields.	Data Numbers	Data Length	Data	Packet	**Data Length**
38	_____ corporation was instrumental in providing the research for CSMA/CD and in developing the first commercial products.	SUN	XEROX	MICROSOFT	Xenix	**XEROX**
39	_____ transmits the frame into the physical layer and receives the frame from the physical layer.	Medium access Management	CSMA/CD	XEROX	MAC	**Medium Access Management**
40	In a CSMA/CD network aeach station has both a transmit and receive side to provide the _____ flow of data.	I/O	OUTPUT	INPUT	Memory	**I/O**
41	The transmit data encapsulation entity receives the data and constructs the _____ frame.	MAC	TCP	IP	TCP/IP	**MAC**
42	The CSMA/CD frame is transmitted to all stations connected to the _____	I/O	Channel	Network	TCP/IP	**Channel**
43	A central aspect of collisions deals with the _____	Collision window	sliding window	protocol	Network	**Collision window**

44	An ethernet 10Mbit/s channel has a propagation delay of _____ bit times.	350	450	400	425	**450**
45	The CSMA/CD LAN requires that the jam be at atleast _____bits.	64	32	16	8	**32**
46	At the receiving station or stations the bits resulting fromthe collision are decoded by the _____ layer.	Physical	Network	Session	Presentation	**Physical**
47	Both Ethernet and _____ use a 1-persistent technique to manage collisions and channel contention.	802.4	802.3	802.1	802.2	**802.3**
48	CSMA/CD performs best under conditions when aggregate channel utilization is relatively _____	High	Low	Moderate	Very Low	**Low**
49	An asynchronous terminal based system should perform well with _____	Medium access Management	CSMA/CD	XEROX	MAC	**CSMA/CD**
50	The _____ uses a token to provide priority access to the network.	Network media	window	Token ring	Protocol	**Token ring**
51	With the token priority passing approach the stations have priority established for access to the _____	Physical	Network	Session	Presentation	**Network**
52	The _____ is the heart of the token bus system.	ACM	TxM	Interface Machine	RxM	**ACM**
53	_____ determines the logical ring of the physical bus by the numeric value of addresses.	ACM	IEEE802.4	IEEE 802.5	IEEE 802.1	**IEEE 802.4**
54	An _____ or LLC data unit provides the fcility for the lowest address to hand the token to the highest address.	MAC	Physical	Network	HDLC	**MAC**
55	_____ also forms the basis for the Switched Multi- megabit Data Service.	802.6	802.5	802.3	802.1	**802.6**
56	The ____ standard is organized around a topology and technique called DQDB.	WAN	MAN	LAN	Wireless	**MAN**
57	The implementation for MAN provides fro transfer rates from 34 to ___ Mbit/s.	100	150	125	110	**150**
58	_____ has developed a sepcification for LAN and optical fibres.	ANSI	CCITT	ISO	NIIT	**ANSI**
59	FDDI means_____	Fiber Data Distributed Interaction	Fiber Distributed Data Interface	Fiber Data Distributed Interface	Fiber Distributed Data Interaction	**Fiber Distributed Data Interface**
60	The _____ channel operates at 100 Mbit/s upto 10000 nodes can be placed on one optical fiber ring.	Network	Wireless	Wired	optical fiber	**Optical fiber**

KARPAGAM ACADEMY OF HIGHER EDUCATION
(Deemed University Established Under Section 3 of UGC Act 1956)
Coimbatore - 641021.
(For the candidates admitted from 2017 onwards)

DEPARTMENT OF CS, CA & IT

SYLLABUS:

UNIT III

Network Protocols :TCP , UDP, IP, ICMP, SNMP,RMON
TCP/IP :TCP/IP and internetworking, example of TCP/IP operations, related protocols ports and sockets. The IP address structure, major features of IP, IP datagram. Major IP services. IP source routing, value of the transport layer, TCP, Major features of TCP, passive and active operation, the transmission control block (TCP), route discovery protocols, examples of route discovery protocols, application layer protocols.

3.1 Network Protocols :TCP , UDP, IP, ICMP, SNMP,RMON

Protocols
In information technology, a protocol (from the Greek *protocollon*, which was a leaf of paper glued to a manuscript volume, describing its contents) is the special set of rules that end points in a telecommunication connection use when they communicate. Protocols exist at several levels in a telecommunication connection. For example, there are protocols for the data interchange at the hardware device level and protocols for data interchange at the application program level. In the standard model known as Open Systems Interconnection (OSI), there are one or more protocols attach layer in the telecommunication exchange that both ends of the exchange must recognize and observe. Protocols are often described in an industry or international standard.

Standards Organization

Standards creation Communities

IEEE (Institute of Electrical and Electronics Engineers)
IEEE's Constitution defines the purposes of the organization as "scientific and educational, directed toward the advancement of the theory and practice of Electrical, Electronics, Communications and Computer Engineering, as well as Computer Science, the allied branches of engineering and the related arts and sciences." The IEEE is incorporated under the Not-for-Profit Corporation Law of the state of New York, United States. It was formed in 1963 by the merger of the Institute of Radio Engineers (IRE, founded 1912) and the American Institute of Electrical

Engineers (AIEE, founded 1884). It has more than 400,000 members in more than 160 countries, 45% outside the United States.In pursuing these goals, the IEEE serves as a major publisher of scientific journals and a conference organizer. It is also a leading developer of industrial standards (having developed over 900 active industry standards) in a broad range of disciplines, including electric power and energy, biomedical technology and health care, information technology, information assurance, telecommunications, consumer electronics, transportation, aerospace, and nanotechnology. IEEE develops and participates in educational activities such as accreditation of electrical engineering programs in institutes of higher learning. IEEE is one of the leading standards-making organizations in the world. IEEE performs its standards making and maintaining functions through the IEEE Standards Association (IEEE-SA). IEEE standards affect a wide range of industries including: power and energy, biomedical and health care, Information Technology (IT), telecommunications, transportation, nanotechnology, information assurance, and many more. In 2005, IEEE had close to 900 active standards, with 500 standards under development. One of the more notable IEEE standards is the IEEE 802 LAN/MAN group of standards which includes the IEEE 802.3 Ethernet standard and the IEEE 802.11 Wireless Networking standard.

ANSI (American National Standards Institute)

Though ANSI itself does not develop standards, the Institute oversees the development and use of standards by accrediting the procedures of standards developing organizations. ANSI accreditation signifies that the procedures used by standards developing organizations meet the Institute's requirements for openness, balance, consensus, and due process. ANSI was originally formed in 1918, when five engineering societies and three government agencies founded the American Engineering Standards Committee (AESC). In 1928, the AESC became the American Standards Association (ASA). In 1966, the ASA was reorganized and became the United States of America Standards Institute (USASI). The present name was adopted in 1969.Prior to 1918, these five engineering societies:

- American Institute of Electrical Engineers (AIEE, now IEEE)
- American Society of Mechanical Engineers (ASME)
- American Society of Civil Engineers (ASCE)
- American Institute of Mining Engineers (AIME, now American Institute of Mining, Metallurgical, and Petroleum Engineers)
- American Society for Testing and Materials (now ASTM International)

ANSI also designates specific standards as American National Standards, or ANS, when the Institute determines that the standards were developed in an environment that is equitable, accessible and responsive to the requirements of various stakeholders.

The American National Standards process involves:

- consensus by a group that is open to representatives from all interested parties
- broad-based public review and comment on draft standards
- consideration of and response to comments
- incorporation of submitted changes that meet the same consensus requirements into a draft standard
- availability of an appeal by any participant alleging that these principles were not respected during the standards-development process.

ITU (International Telecommunications Union - formerly CCITT)

The **International Telecommunication Union** is the specialized agency of the United Nations which is responsible for information and communication technologies. ITU coordinates the shared global use of the radio spectrum, promotes international cooperation in assigning satellite orbits, works to improve telecommunication infrastructure in the developing world and establishes worldwide standards.

ITU coordinates the shared global use of the radio spectrum, promotes international cooperation in assigning satellite orbits, works to improve telecommunication infrastructure in the developing world and establishes worldwide standards.ITU also organizes worldwide and regional exhibitions and forums, such as ITU TELECOM WORLD, bringing together representatives of government and the telecommunications and ICT industry to exchange ideas, knowledge and technology.The ITU is active in areas including broadband Internet, latest-generation wireless technologies, aeronautical and maritime navigation, radio astronomy, satellite-based meteorology, convergence in fixed-mobile phone, Internet access, data, voice, TV broadcasting, and next-generation networks.

ISO (International Organization for Standards)

The **International Organization for Standardization** widely known as **ISO**, is an international standard-setting body composed of representatives from various national standards organizations. Founded on February 23, 1947, the organization promulgates worldwide proprietary industrial and commercial standards. It has its headquarters in Geneva, Switzerland. While ISO defines itself as a non-governmental organization, its ability to set standards that often become law, either through treaties or national standards, makes it more powerful than most non-governmental organizations. In practice, ISO acts as a consortium with strong links to governments

ISO, is an international standard-setting body composed of representatives from various national standards organizations the organization promulgates worldwide proprietary industrial and commercial standards.ISO's main products are the International Standards. ISO also publishes

Technical Reports, Technical Specifications, Publicly Available Specifications, Technical Corrigenda, and Guides.

EIA (Electronic Industries Association)

The **Electronic Industries Alliance** (EIA, until 1997 *Electronic Industries Association*) was a standards and trade organization composed as an alliance of trade associations for electronics manufacturers in the United States. They developed standards to ensure the equipment of different manufacturers was compatible and interchangable.In 1924 the *Associated Radio Manufacturers* alliance was formed, which was renamed to *Radio Manufacturers Association (RMA)* the same year. Upcoming new electronic technologies brought new members and further name changes: *Radio Television Manufacturers Association (RTMA)* (1950), *Radio Electronics Television Manufacturers (RETMA)* (1953) and *Electronics Industries Association (EIA)* (1957). The last renaming took place in 1997, when EIA became *Electronics Industries Alliance (EIA)*, reflecting the change away from a pure manufacturersassociationA standard defining serial communication between computers and modems e. g. was originally drafted by the radio sector as RS-232. Later it was taken over by the EIA as *EIA-232*. Later this standard was managed by the TIA and the name was changed to the current *TIA-232*. Because the EIA was accredited by ANSI to help develop standards in its areas, the standards are often described as e. g. *ANSI TIA-232* (or formerly as *ANSI EIA/TIA-232')*.

ETSI (European Telecommunications Standards Institute)

The **European Telecommunications Standards Institute** (**ETSI**) is an independent, non-profit, standardization organization in the telecommunications industry (equipment makers and network operators) in Europe, with worldwide projection. ETSI has been successful in standardizing the Low Power Radio, Short Range Device, GSM cell phone system and the TETRA professional mobile radio system.Significant ETSI standardisation bodies include TISPAN (for fixed networks and Internetmachine-to-machine communications). ETSI inspired the creation of, and is a partner in 3GPP.

ETSI was created by CEPT in 1988 and is officially recognized by the European Commission and the EFTA secretariat. Based in Sophia Antipolis (France), ETSI is officially responsible for standardization of Information and Communication Technologies (ICT) within Europe. These technologies include telecommunications, broadcasting and related areas such as intelligent transportation and medical electronics. ETSI has 740 members from 62 countries/provinces inside and outside Europe, including manufacturers, network operators, administrations, service providers, research bodies and users — in fact, all the key players in the ICT arena. convergence)

and M2M (for ETSI has been successful in standardizing the Low Power Radio, Short Range Device, GSMTETRA professional mobile radio system.ETSI was created by CEPT in 1988 and is officially recognized by the European Commission and the EFTASophia Antipolis (France), ETSI is officially responsible for standardization of Information and Communication Technologies (ICT) within Europe. These technologies include telecommunications, broadcasting and related areas such as intelligent transportation and medical electronics.

W3C - World Wide Web Consortium

The **World Wide Web Consortium** (**W3C**) is the main international standards organizationWorld Wide Web (abbreviated WWW or W3).

Founded and headed by Tim Berners-Lee,the consortium is made up of member organizations which maintain full-time staff for the purpose of working together in the development of standards for the World Wide Web. As of 18 February 2011, the World Wide Web Consortium (W3C) has 322 members.

W3C also engages in education and outreach, develops software and serves as an open forum for discussion about the Web.

W3C also engages in education and outreach, develops software and serves as an open forum for discussion about the Web.W3C was created to ensure compatibility and agreement among industry members in the adoption of new standards. Prior to its creation, incompatible versions of HTML were offered by different vendors, increasing the potential for inconsistency between web pages. The consortium was created to get all those vendors to agree on a set of core principles and components which would be supported by everyone.

3.2 TCP/IP :TCP/IP and internetworking

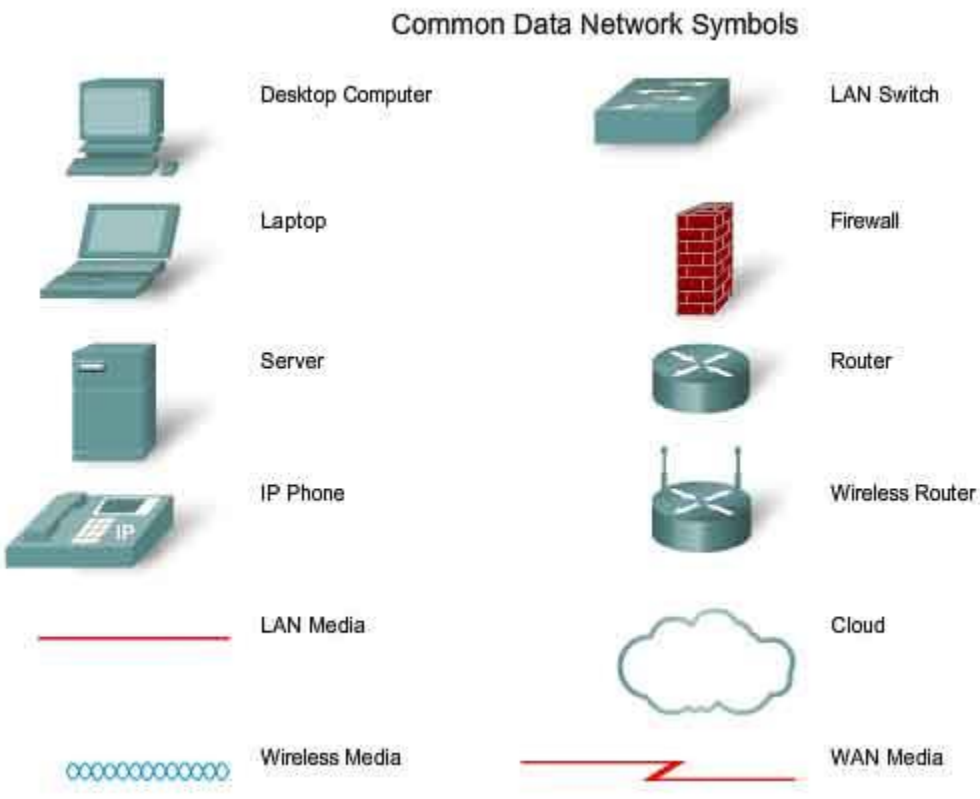
Internetworking elements:

The diagram shows elements of a typical network, including devices, media, and services, tied together by rules, that work together to send messages. We use the word *messages* as a term that encompasses web pages, e-mail, instant messages, telephone calls, and other forms of communication enabled by the Internet.

Four elements of a network:
- Rules
- Medium
- Messages
- Devices

Internetworking is a very graphically oriented subject, and icons are commonly used to represent networking devices. On the left side of the diagram are shown some common devices which often originate messages that comprise our communication. These include various types of computers (a PC and laptop icon are shown), servers, and IP phones. On local area networks these devices are typically connected by LAN media (wired or wireless). The right side of the figure shows some of the most common intermediate devices, used to direct and manage messages across the network, as well as other common **internetworking** symbols. Generic symbols are shown for:

- Switch - the most common device for interconnecting local area networks
- Firewall - provides security to networks
- Router - helps direct messages as they travel across a network
- Wireless Router - a specific type of router often found in home networks
- Cloud - used to summarize a group of networking devices, the details of which may be unimportant to the discussion at hand
- Serial Link - one form of WAN interconnection, represented by the lightning bolt-shaped line



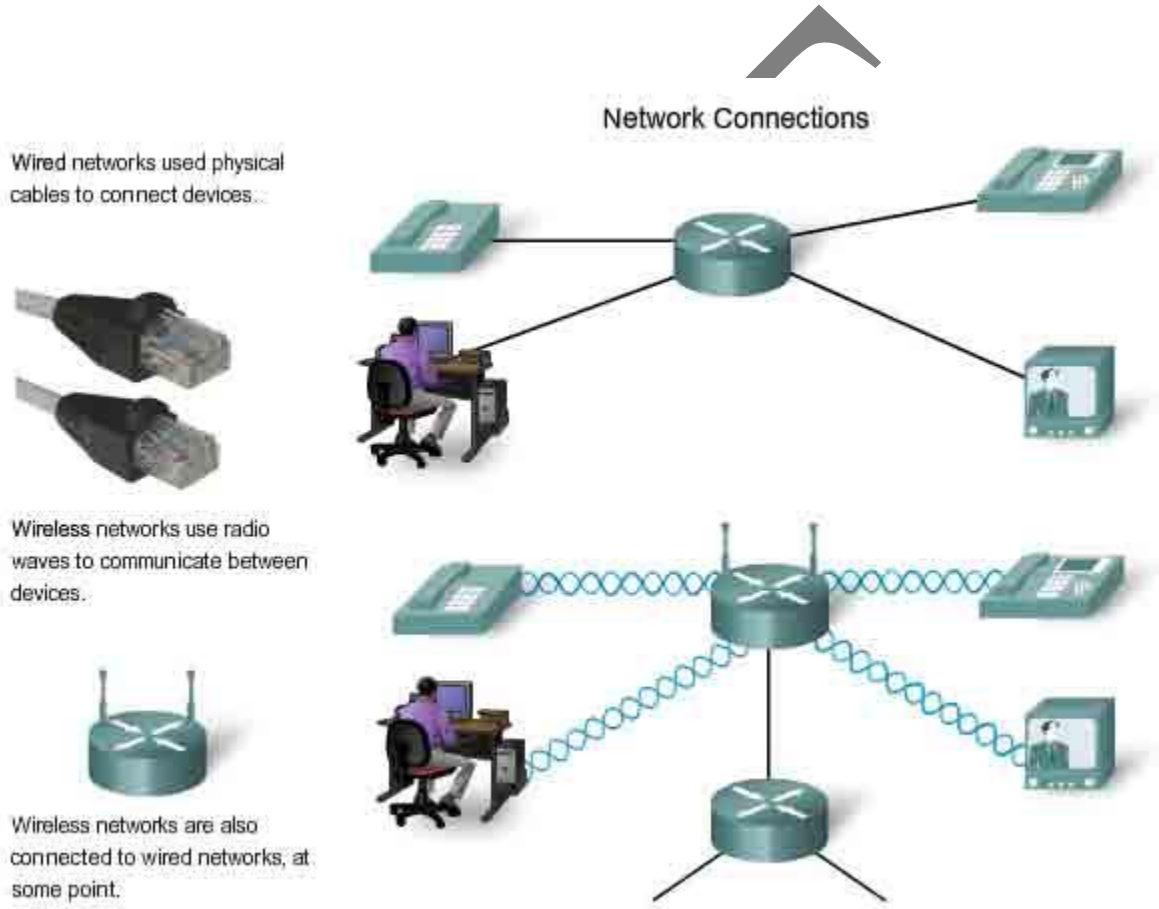
Network Interface Card - A NIC, or LAN adapter, provides the physical connection to the network at the PC or other host device. The media connecting the PC to the networking device plugs directly into the NIC.

Physical Port - A connector or outlet on a **internetworking** device where the media is connected to a host or other networking device.

Interface - Specialized ports on an **internetworking** device that connect to individual networks. Because routers are used to interconnect networks, the ports on a router are referred to network interfaces.

For a network to function, the devices must be interconnected. Network connections can be wired or wireless. In wired connections, the medium is either copper, which carries electrical signals, or optical fiber, which carries light signals. In wireless connections, the medium is the Earth's atmosphere, or space, and the signals are microwaves. Copper medium includes cables, such as twisted pair telephone wire, coaxial cable, or most commonly, what is known as Category 5 Unshielded Twisted Pair (UTP) cable. Optical fibers, thin strands of glass or plastic

that carry light signals, are another form of **internetworking** media. Wireless media may include the home wireless connection between a wireless router and a computer with a wireless network card, the terrestrial wireless connection between two ground stations, or the communication between devices on earth and satellites. In a typical journey across the Internet, a message may travel across a variety of media.



Human beings often seek to send and receive a variety of messages using computer applications; these applications require services to be provided by the network. Some of these services include the World Wide Web, e-mail, instant messaging, and IP Telephony. Devices interconnected by medium to provide services must be governed by rules, or protocols. In the chart, some common services and a protocol most directly associated with that service are listed. Protocols are the rules that the networked devices use to communicate with each other. The industry standard in **internetworking** today is a set of protocols called TCP/IP(Transmission Control

Protocol/Internet Protocol). TCP/IP is used in home and business networks, as well as being the primary protocol of the Internet. It is TCP/IP protocols that specify the formatting, addressing and routing mechanisms that ensure our messages are delivered to the correct recipient.

The Messages

In the first step of its journey from the computer to its destination, our instant message gets converted into a format that can be transmitted on the network. All types of messages must be converted to bits, binary coded digital signals, before being sent to their destinations. This is true no matter what the original message format was: text, video, voice, or computer data. Once our instant message is converted to bits, it is ready to be sent onto the network for delivery.

The Devices

To begin to understand the robustness and complexity of the interconnected networks that make up the Internet, it is necessary to start with the basics. Take the example of sending the text message using an instant messaging program on a computer. When we think of using network services, we usually think of using a computer to access them. But, a computer is only one type of device that can send and receive messages over a network. Many other types of devices can also be connected to the network to participate in network services. Among these devices are telephones, cameras, music systems, printers and game consoles.

In addition to the computer, there are numerous other components that make it possible for our instant message to be directed across the miles of wires, underground cables, airwaves and satellite stations that might exist between the source and destination devices. One of the critical components in any size network is the router. A router joins two or more networks, like a home network and the Internet, and passes information from one network to another. Routers in a network work to ensure that the message gets to its destination in the most efficient and quickest manner.

The Medium

To send our instant message to its destination, the computer must be connected to a wired or wireless local network. Local networks can be installed in homes or businesses, where they enable computers and other devices to share information with each other and to use a common connection to the Internet.

Wireless networks allow the use of networked devices anywhere in an office or home, even outdoors. Outside the office or home, wireless networking is available in public hotspots, such as coffee shops, businesses, hotel rooms, and airports.

Many installed networks use wires to provide connectivity. Ethernet is the most common wired **internetworking** technology found today. The wires, called cables, connect the computers and other devices that make up the networks. Wired networks are best for moving large amounts of data at high speeds, such as are required to support professional-quality multimedia.

The Services

Network services are computer programs that support the human network. Distributed on devices throughout the network, these services facilitate online communication tools such as e-mail, bulletin/discussion boards, chat rooms, and instant messaging. In the case of instant messaging, for example, an instant messaging service, provided by devices in the cloud, must be accessible to both the sender and recipient.

The Rules

Important aspects of networks that are neither devices nor media are rules, or protocols. These rules are the standards and protocols that specify how the messages are sent, how they are directed through the network, and how they are interpreted at the destination devices. For example, in the case of Jabber instant messaging, the XMPP, TCP, and IP protocols are all important sets of rules that enable our communication to occur.

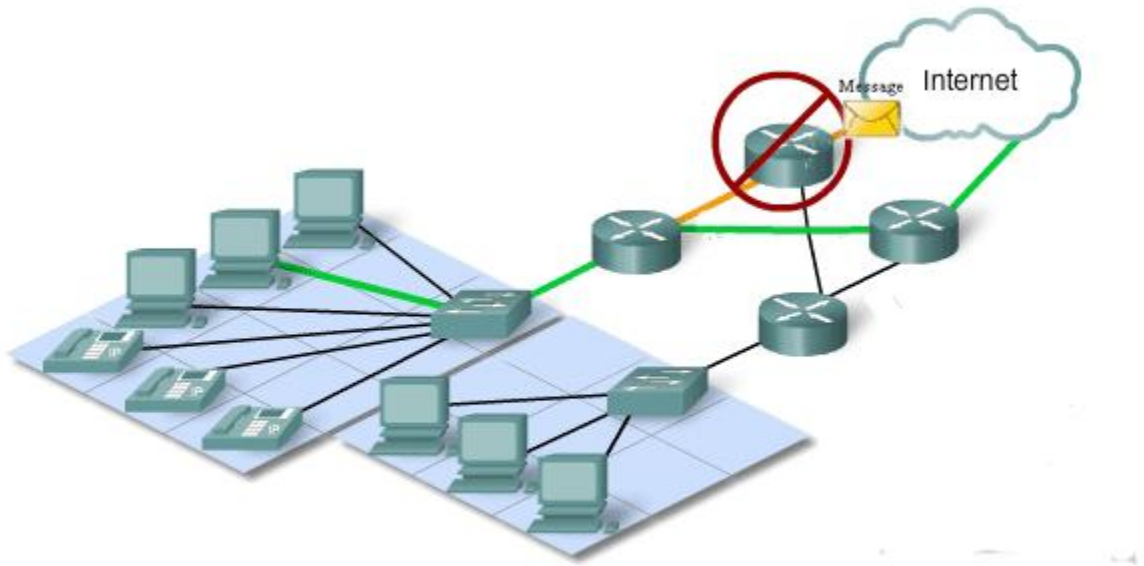
The Network architecture:

Networks must support a wide range of applications and services, as well as operate over many different types of physical infrastructures. The term network architecture, in this context, refers to both the technologies that support the infrastructure and the programmed services and protocols that move the messages across that infrastructure. As the Internet, and networks in general, evolve, we are discovering that there are four basic characteristics that the underlying architectures need to address in order to meet user expectations: fault tolerance, scalability, quality of service, and security.

Fault Tolerance

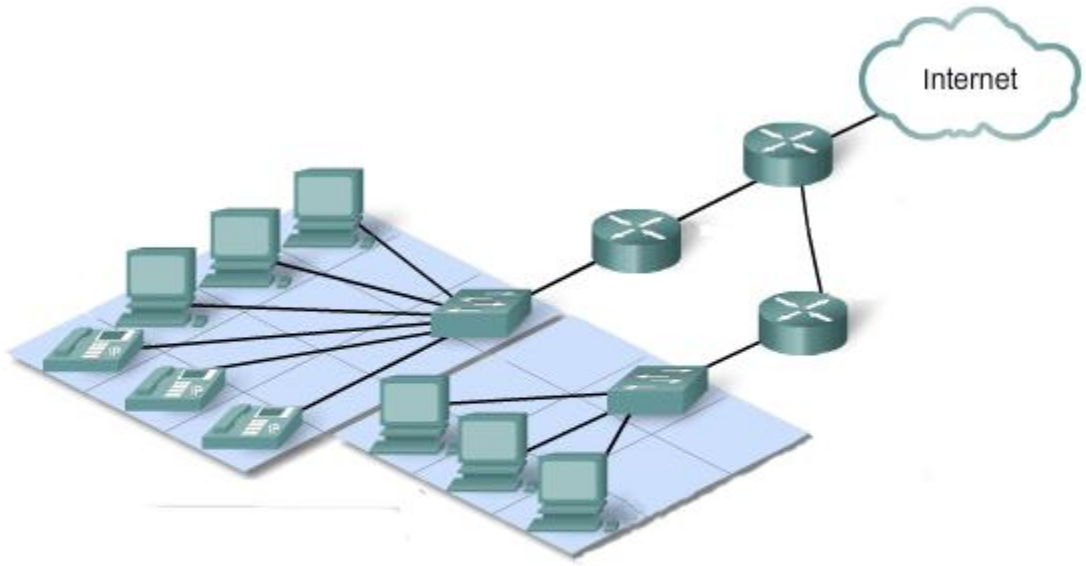
The expectation that the Internet is always available to the millions of users who rely on it requires a network architecture that is designed and built to be fault tolerant. A fault tolerant network is one that limits the impact of a hardware or software failure and can recover quickly when such a failure occurs. These networks depend on redundant links, or paths, between the source and destination of a message. If one link or path fails, processes ensure that messages can

be instantly routed over a different link transparent to the users on either end. Both the physical infrastructures and the logical processes that direct the messages through the network are designed to accommodate this redundancy. This is a basic premise of the architecture of current networks.



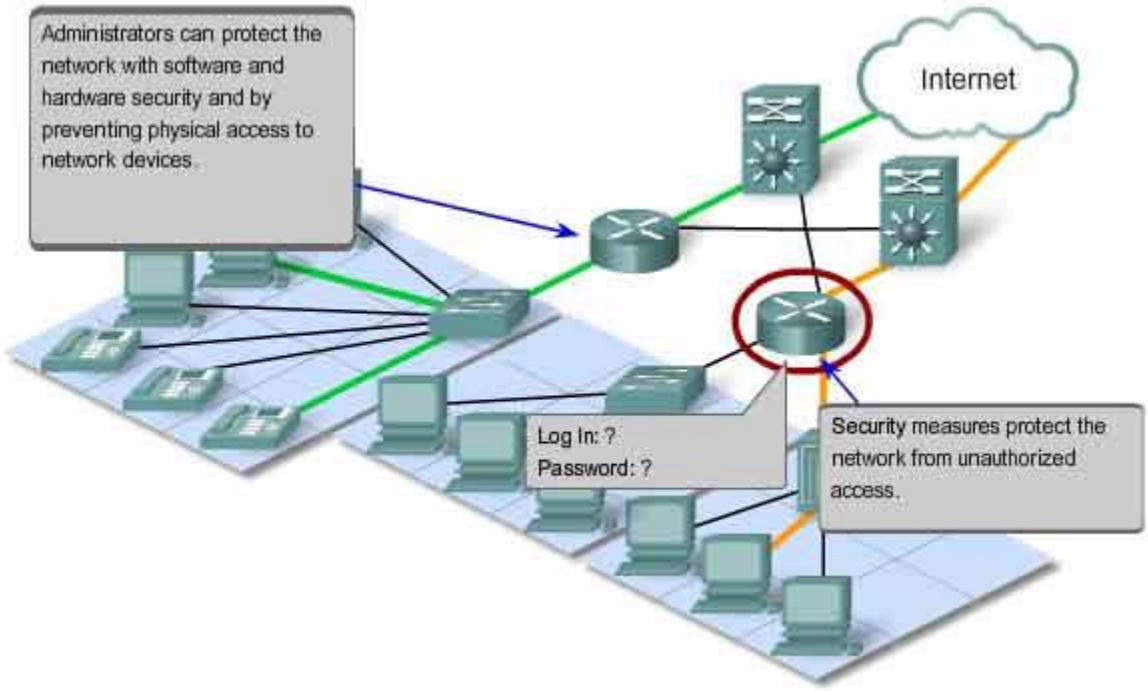
Scalability

A scalable network can expand quickly to support new users and applications without impacting the performance of the service being delivered to existing users. Thousands of new users and service providers connect to the Internet each week. The ability of the network to support these new interconnections depends on a hierarchical layered design for the underlying physical infrastructure and logical architecture. The operation at each layer enables users or service providers to be inserted without causing disruption to the entire network. Technology developments are constantly increasing the message carrying capabilities and performance of the physical infrastructure components at every layer. These developments, along with new methods to identify and locate individual users within an internetwork, are enabling the Internet to keep pace with user demand.



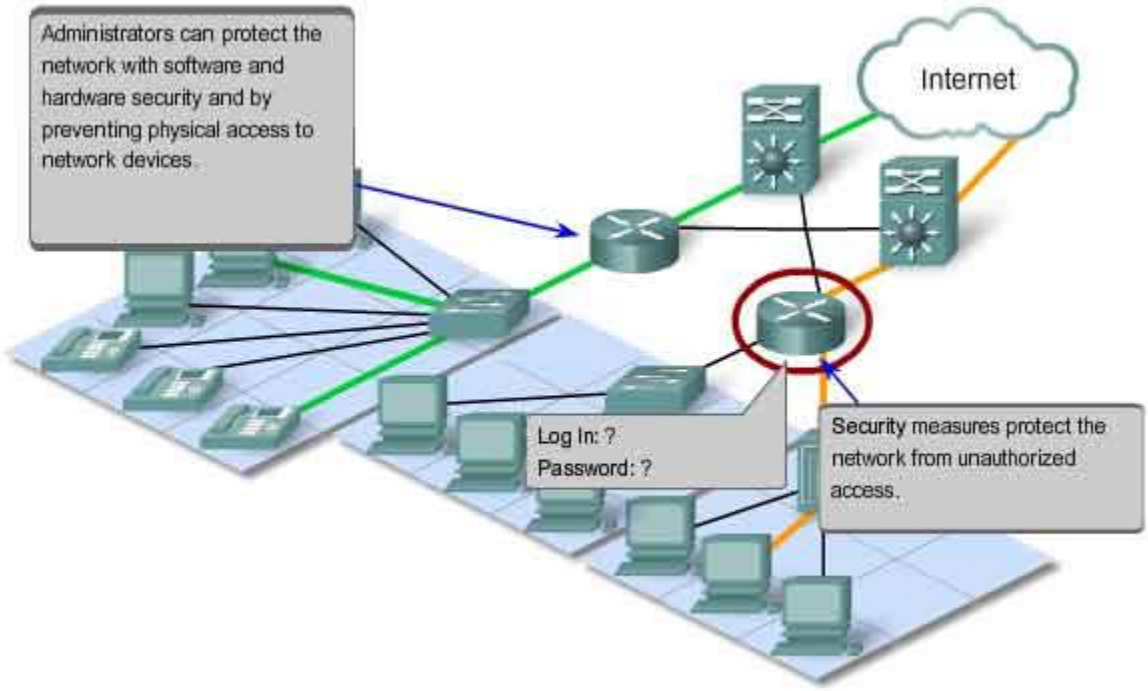
Quality of Service (QoS)

The Internet is currently providing an acceptable level of fault tolerance and scalability for its users. But new applications available to users over internetworks create higher expectations for the quality of the delivered services. Voice and live video transmissions require a level of consistent quality and uninterrupted delivery that was not necessary for traditional computer applications. Quality of these services is measured against the quality of experiencing the same audio or video presentation in person. Traditional voice and video networks are designed to support a single type of transmission, and are therefore able to produce an acceptable level of quality. New requirements to support this quality of service over a converged **internetworking** are changing the way network architectures are designed and implemented.



Security

The Internet has evolved from a tightly controlled internetwork of educational and government organizations to a widely accessible means for transmission of business and personal communications. As a result, the security requirements of the network have changed. The security and privacy expectations that result from the use of internetworks to exchange confidential and business critical information exceed what the current architecture can deliver. Rapid expansion in communication areas that were not served by traditional data networks is increasing the need to embed security into the network architecture. As a result, much effort is being devoted to this area of research and development. In the meantime, many tools and procedures are being implemented to combat inherent security flaws in the network architecture.



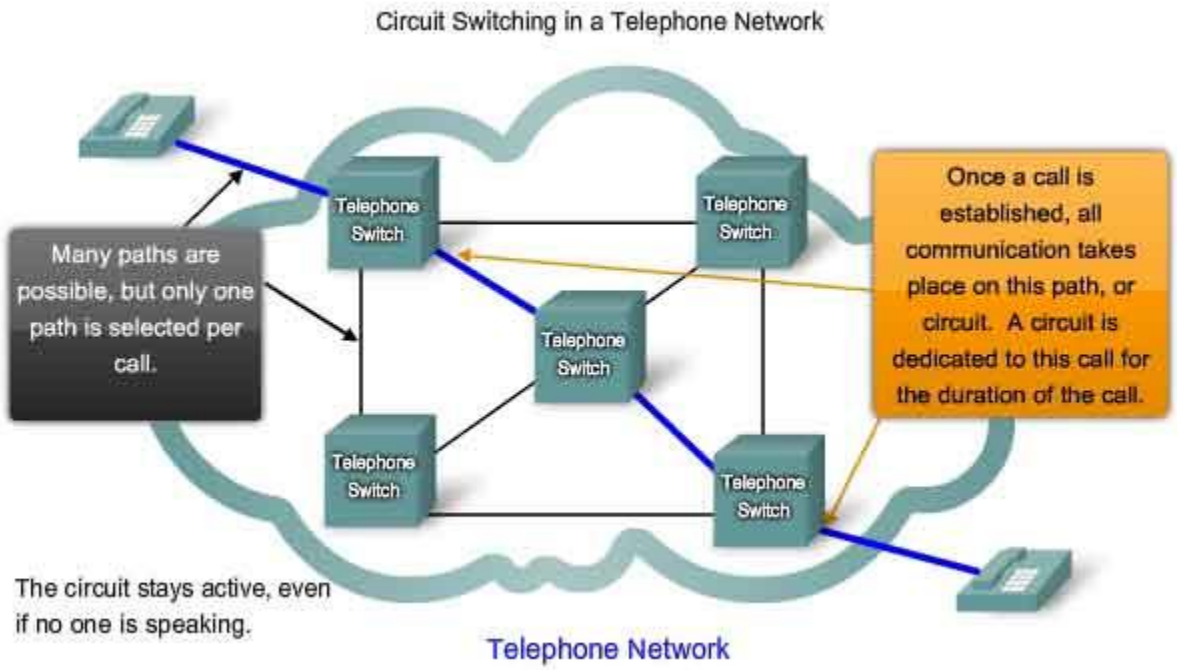
Fault Tolerant Network Architecture

The Internet, in its early inception, was the result of research funded by the United States Department of Defense (DoD). Its primary goal was to have a communications medium that could withstand the destruction of numerous sites and transmission facilities without disruption of service. It only follows that fault tolerance was the focus of the effort of the initial internetwork design work. Early network researchers looked at the existing communication networks, which were primarily for the transmission of voice traffic, to determine what could be done to improve the fault tolerance level.

Circuit Switched Connection-oriented Networks

To understand the challenge that the DoD researchers were faced with, it is necessary to look at how early telephone systems work. When a person makes a call using a traditional telephone set, the call first goes through a setup process, where all of the telephone switching locations between the person and the phone set that they are calling are identified. A temporary path, or circuit, is created through the various switching locations to use for the duration of the telephone call. If any link or device participating in the circuit fails, the call is dropped. To reconnect, a new call must be made, and a new circuit created between the source telephone set and the destination.

This type of connection-oriented network is called a circuit-switched network. Early circuit switched networks did not dynamically recreate dropped circuits. In order to recover from failure, new calls had to be initiated and new circuits built end-to-end. Many circuit switched networks give priority to maintaining existing circuit connections, at the expense of new circuit requests. In this type of connection-oriented network, once a circuit is established, even if no communication is occurring between the persons on either end of the call, the circuit remains connected and resources reserved until one of the parties disconnects the call. Since there is a finite capacity to create new circuits, it is possible to occasionally get a message that all circuits are busy and a call cannot be placed. The cost to create many alternate paths with enough capacity to support a large number of simultaneous circuits, and the technologies necessary to dynamically recreate dropped circuits in the event of a failure, led the DoD to consider other types of networks.



Packet Switched Connectionless Networks

In the search for a network that could withstand the loss of a significant amount of its transmission and switching facilities, the early Internet designers reevaluated early research regarding packet switched networks. The premise for this type of networks is that a single message can be broken into multiple message blocks. Individual blocks containing addressing information indicate both their origination point and their final destination. Using this embedded information, these message blocks, called packets, can be sent through the network along various paths, and can be reassembled into the original message upon reaching their destination.

Utilizing Packets

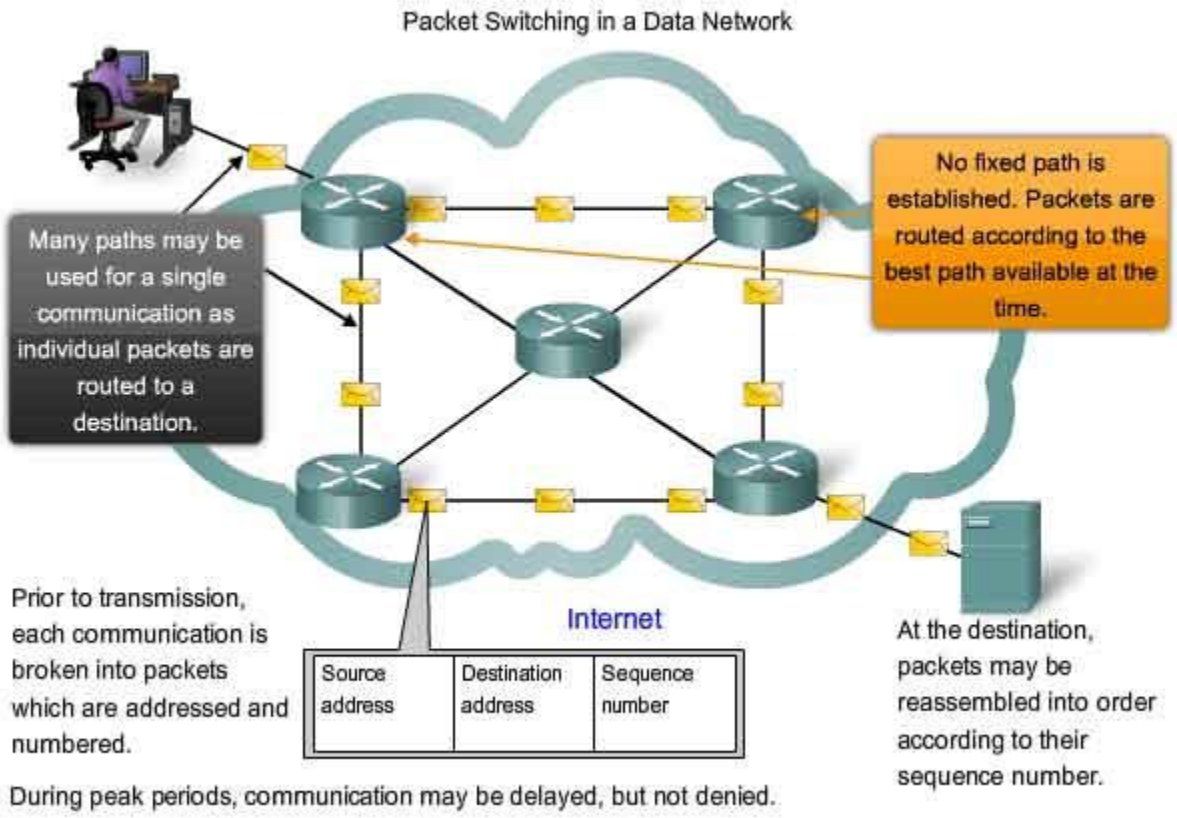
The devices within the network itself are unaware of the content of the individual packets, only visible is the address of the final destination and the next device in the path to that destination. No reserved circuit is built between sender and receiver. Each packet is sent independently from one switching location to another. At each location, a routing decision is made as to which path to use to forward the packet towards its final destination. If a previously used path is no longer available, the routing function can dynamically choose the next best available path. Because the messages are sent in pieces, rather than as a single complete message, the few packets that may be lost in the advent of a failure can be retransmitted to the destination along a different path. In many cases, the destination device is unaware that any failure or rerouting has occurred.

Packet-switched Connectionless Networks

The DoD researchers realized that a packet switched connectionless network had the features necessary to support a resilient, fault tolerant network architecture. The need for a single, reserved circuit from end-to-end does not exist in a packet switched network. Any piece of a message can be sent through the network using any available path. Packets containing pieces of messages from different sources can travel the network at the same time. The problem of underutilized or idle circuits is eliminated -- all available resources can be used at any time to deliver packets to their final destination. By providing a method to dynamically use redundant paths, without intervention by the user, the Internet has become a fault tolerant, scalable method of communications.

Connection-oriented Networks

Although packet-switched connectionless networks met the needs of the DoD, and continue to be the primary infrastructure for today's Internet, there are some benefits to a connection-oriented system like the circuit-switched telephone system. Because resources at the various switching locations are dedicated to providing a finite number of circuits, the quality and consistency of messages transmitted across a connection-oriented network can be guaranteed. Another benefit is that the provider of the service can charge the users of the network for the period of time that the connection is active. The ability to charge users for active connections through the network is a fundamental premise of the telecommunication service industry.



A Scalable Network Architecture

The fact that the Internet is able to expand at the rate that it is, without seriously impacting the performance experienced by individual users, is a function of the design of the protocols and underlying technologies on which it is built. The Internet, which is actually a collection of interconnected private and public networks, has a hierarchical layered structure for addressing, for naming and for connectivity services. At each level or layer of the hierarchy, individual network operators maintain peering relationships with other operators at the same level. As a result, network traffic that is destined for local or regional services does not need to traverse to a central point for distribution. Common services can be duplicated in different regions, thereby keeping traffic off the higher level backbone networks.Although there is no single organization that regulates the Internet, the operators of the many individual networks that provide Internet connectivity cooperate to follow accepted standards and protocols. The adherence to standards enables the manufacturers of hardware and software to concentrate on product improvements in the areas of performance and capacity, knowing that the new products can integrate with and enhance the existing infrastructure. The current Internet architecture, while highly scalable, may

not always be able to keep up with the pace of user demand. New protocols and addressing structures are under development to meet the increasing rate at which Internet applications and services are being added.

Providing Quality of Service

Networks must provide secure, predictable, measurable, and, at times, guaranteed services. The packet-switched network architecture does not guarantee that all packets that comprise a particular message will arrive on time, in their correct in order, or even that they will arrive at all. Networks also need mechanisms to manage congested network traffic. Congestion is caused when the demand on the network resources exceeds the available capacity. If all networks had infinite resources, there would not be a need to use QoS mechanisms to ensure quality of service. Unfortunately, that is not the case. There are some constraints on network resources that cannot be avoided. Constraints include technology limitations, costs, and the local availability of high-bandwidth service. Network bandwidth is the measure of the data carrying capacity of the network. When simultaneous communications are attempted across the network, the demand for network bandwidth can exceed its availability. The obvious fix for this situation is to increase the amount of available bandwidth. But, because of the previously stated constraints, this is not always possible. In most cases, when the volume of packets is greater than what can be transported across the network, devices queue the packets in memory until resources become available to transmit them. Queuing packets causes delay. If the number of packets to be queued continues to increase, the memory queues fill up and packets are dropped.

Achieving the required Quality of Service (QoS) by managing the delay and packet loss parameters on a network becomes the secret to a successful end-to-end application quality solution. Thus, ensuring QoS requires a set of techniques to manage the utilization of network resources. In order to maintain a high quality of service for applications that require it, it is necessary to prioritize which types of data packets must be delivered at the expense of other types of packets that can be delayed or dropped.

Classification

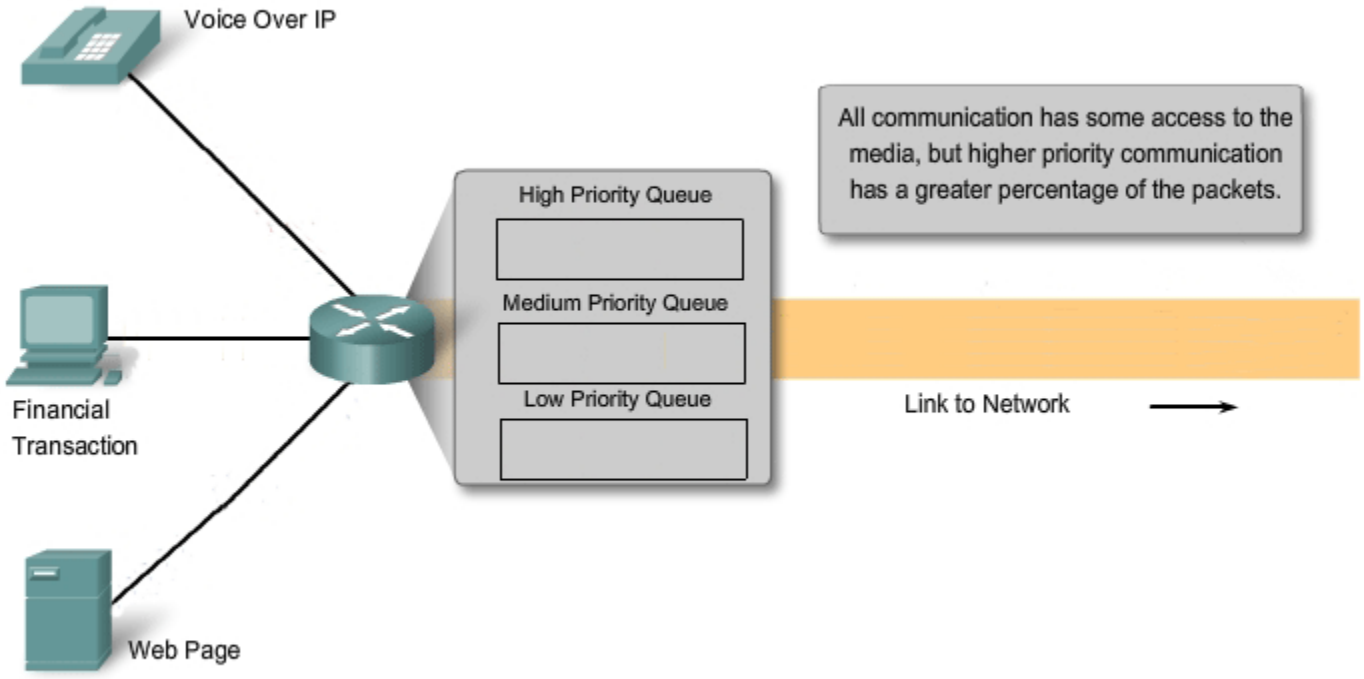
Ideally, we would like to assign a precise priority for each type of communication. Currently, this is neither practical nor possible. Therefore, we classify applications in categories based on specific quality of service requirements. To create QoS classifications of data, we use a combination of communication characteristics and the relative importance assigned to the application. We then treat all data within the same classification according to the same rules. For

example, communication that is time-sensitive or important would be classified differently from communication that can wait or is of lesser importance.

Assigning priorities

The characteristics of the information being communicated also affect its management. For example, the delivery of a movie uses a relatively large amount of network resources when it is delivered continuously without interruption. Other types of service - e-mail, for example - are not nearly as demanding on the network. In one company, an administrator might decide to allocate the greatest share of the network resources to the movie, believing that this is the priority for his customers. This administrator may decide that the impact will be minimal if e-mail users have to wait a few additional seconds for their e-mail to arrive. In another company, the quality of a video stream is not as important as critical process control information that operates the manufacturing machinery.

Using Queues to Prioritize Communication



QoS mechanisms enable the establishment of queue management strategies that enforce priorities for different classifications of application data. Without properly designed and implemented QoS mechanisms, data packets will be dropped without consideration of the application characteristics or priority. Examples of priority decisions for an organization might include:

- Time-sensitive communication - increase priority for services like telephony or video distribution.
- Non time-sensitive communication - decrease priority for web page retrieval or e-mail.
- High importance to organization - increase priority for production control or business transaction data.
- Undesirable communication - decrease priority or block unwanted activity, like peer-to-peer file sharing or live entertainment.

The Quality of Service a network can offer is a vital issue, and in some situations, it is crucial. Imagine the consequences of a dropped distress call to an emergency response center, or of a lost control signal to an automated piece of heavy machinery. A key responsibility for the network managers in an organization is to establish a Quality of Service policy and ensure that the mechanisms are in place to meet that goal.

Providing Network Security

The network infrastructure, services, and the data contained on network attached computers are crucial personal and business assets. Compromising the integrity of these assets could have serious business and financial repercussions. Consequences of a network security breach could include:

- Network outage that prevents communications and transactions occurring, with consequent loss of business
- Misdirection and loss of personal or business funds
- Company intellectual property (research ideas, patents or designs) that is stolen and used by a competitor
- Customer contract details that become known to competitors or made public, resulting in a loss of market confidence in the business

A lack of public trust in the business's privacy, confidentiality, and integrity levels may lead to loss of sales and eventual company failure. There are two types of network security concerns that must be addressed to prevent serious consequences: network infrastructure security and content

security. Securing a network infrastructure includes the physical securing of devices that provide network connectivity and preventing unauthorized access to the management software that resides on them. Content security refers to protecting the information contained within the packets being transmitted over the network and the information stored on network attached devices. When transmitting information over the Internet or other network, the content of the individual packets is not readily known to the devices and facilities through which the packets travel. Tools to provide security for the content of individual messages must be implemented on top of the underlying protocols which govern how packets are formatted, addressed and delivered. Because the reassembly and interpretation of the content is delegated to programs running on the individual source and destination systems, many of the security tools and protocols must be implemented on those systems as well.

Security measures taken in a network should:

- Prevent unauthorized disclosure or theft of information
- Prevent unauthorized modification of information
- Prevent Denial of Service

Means to achieve these goals include:

- Ensuring confidentiality
- Maintaining communication integrity
- Ensuring availability

Ensuring Confidentiality

Data privacy is maintained by allowing only the intended and authorized recipients - individuals, processes, or devices - to read the data. Having a strong system for user authentication, enforcing passwords that are difficult to guess, and requiring users to change them frequently helps restrict access to communications and to data stored on network attached devices. Where appropriate, encrypting content ensures confidentiality and minimizes unauthorized disclosure or theft of information.

Maintaining Communication Integrity

Data integrity means having the assurance that the information has not been altered in transmission, from origin to destination. Data integrity can be compromised when information has been corrupted - willfully or accidentally - before the intended recipient receives it. Source integrity is the assurance that the identity of the sender has been validated. Source integrity is compromised when a user or device fakes its identity and supplies incorrect information to a

recipient. The use ofdigital signatures, hashing algorithms and checksum mechanisms are ways to provide source and data integrity across a network to prevent unauthorized modification of information.

Ensuring Availability

Ensuring confidentiality and integrity are irrelevant if network resources become over burdened, or not available at all. Availability means having the assurance of timely and reliable access to data services for authorized users. Resources can be unavailable during a Denial of Service (DoS) attack or due to the spread of a computer virus. Network firewall devices, along with desktop and server anti-virus software can ensure system reliability and the robustness to detect, repel, and cope with such attacks. Building fully redundant network infrastructures, with few single points of failure, can reduce the impact of these threats. The result of the implementation of measures to improve both the quality of service and the security of network communications is an increase in the complexity of the underlying network platform. As the Internet continues to expand to offer more and more new services, its future depends on new, more robust **internetworking** architectures being developed that include all four characteristics: fault tolerance, scalability, quality of service, and security.

3.3 Example of TCP/IP operations, related protocols ports and sockets.

Operation of TCP/IP

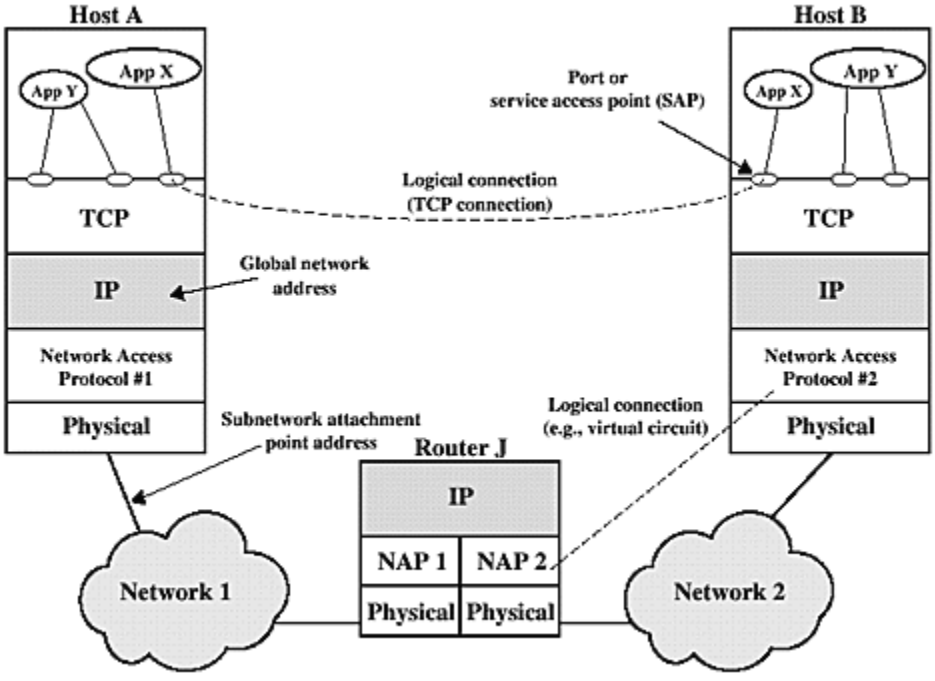


Figure 1 indicates how these protocols are configured for communications. To make clear that the total communications facility may consist of multiple networks, the constituent networks are usually referred to as *subnetworks*. Some sort of network access protocol, such as the Ethernet logic, is used to connect a computer to a subnetwork. This protocol enables the host to send data across the subnetwork to another host, or, in the case of a host on another subnetwork, to a router. IP is implemented in all of the end systems and the routers. It acts as a relay to move a block of data from one host, through one or more routers, to another host. TCP is implemented only in the end systems; it keeps track of the blocks of data to ensure that all are delivered reliably to the appropriate application.

Figure 1 TCP/IP concepts.

For successful communication, every entity in the overall system must have a unique address. Actually, two levels of addressing are needed. Each host on a subnetwork must have a unique global Internet address; this allows the data to be delivered to the proper host. This address is used by IP for routing and delivery. Each application within a host must have an address that's unique within the host; this allows the host-to-host protocol (TCP) to deliver data to the proper process. These latter addresses are known as *ports*.

Let's trace a simple operation. Suppose that an application, associated with port 1 at host A, wants to send a message to another application, associated with port 2 at host B. The application at A hands the message down to TCP with instructions to send it to host B, port 12. TCP hands the message down to IP with instructions to send it to host B. Note that IP need not be told the identity of the destination port. All it needs to know is that the data is intended for host B. Next, IP hands the message down to the network access layer (such as Ethernet logic) with instructions to send it to router J (the first hop on the way to B).

To control this operation, control information as well as user data must be transmitted, as suggested in Figure 2. Let's say that the sending process generates a block of data and passes this to TCP. TCP may break this block into smaller pieces to make it more manageable. To each of these pieces, TCP appends control information known as the *TCP header*, forming a *TCP segment*. The control information is to be used by the peer TCP protocol entity at host *B*. Examples of fields that are part of this header include the following:

- **Destination port:** When the TCP entity at B receives the segment, it must know to whom the data is to be delivered.
- **Sequence number:** TCP numbers the segments that it sends to a particular destination port sequentially, so that if they arrive out of order, the TCP entity at B can reorder them.

- **Checksum:** The sending TCP includes a code that's a function of the contents of the remainder of the segment. The receiving TCP performs the same calculation and compares the result with the incoming code. A discrepancy results if there has been some error in transmission.

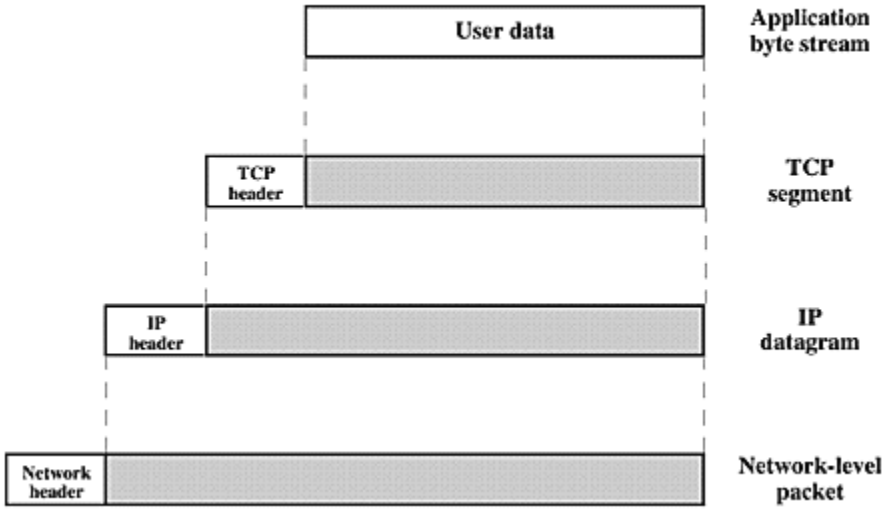


Figure 2 Protocol data units (PDUs) in the TCP/IP architecture.

Next, TCP hands each segment over to IP, with instructions to transmit it to B. These segments must be transmitted across one or more subnetworks and relayed through one or more intermediate routers. This operation, too, requires the use of control information. Thus, IP appends a header of control information to each segment to form an *IP datagram*. An example of an item stored in the IP header is the destination host address (in this example, B).

Finally, each IP datagram is presented to the network access layer for transmission across the first subnetwork in its journey to the destination. The network access layer appends its own header, creating a *packet* or *frame*. The packet is transmitted across the subnetwork to router J. The packet header contains the information that the subnetwork needs to transfer the data across the subnetwork. Examples of items that may be contained in this header include the following:

- **Destination subnetwork address:** The subnetwork must know to which attached device the packet is to be delivered.
- **Facilities requests:** The network access protocol might request the use of certain subnetwork facilities, such as priority.

At router J, the packet header is stripped off and the IP header examined. On the basis of the destination address information in the IP header, the IP module in the router directs the datagram out across subnetwork 2 to B. To do this, the datagram is again augmented with a network access header.

When the data is received at B, the reverse process occurs. At each layer, the corresponding header is removed, and the remainder is passed on to the next higher layer, until the original user data is delivered to the destination application.

3.4 The IP address structure

The IP address of an electronic device is its unique identifier in a network of many devices. The device can be a PC, a router, a server, or even an IP phone. This address is used to transfer data to the different devices over a network working on IP protocol system.

Each device connected to a network must have a unique IP address. Failure to do so can give rise to serious problems, as the data transfer management devices, such as, hubs, switches, and routers, would not know where to send data. This is known as IP conflict, and can even bring down an entire network in worst cases.

The basic structure of an IP address is like xxx.xxx.xxx.xxx, where each xxx can be any number between 0 and 255. Each of these parts is stored in 8 bits. So, the maximum number of possible combinations for each group of number is 256 (i.e. each group can have any one value from the range of 0 to 255).

Each address can be technically divided into two parts. One of these parts is the network part, which represents the class of IP address that is being used in the network. The other one is the host part, which represents the unique ID of the device in the network. Let us consider the IP address 192.168.10.14 – here 192.168.10 is the network part, and represents the network. The number 14 represents the unique ID of the device in the network.

You can find out the IP address of your PC when you are connected to a network. Here's how:

1. Click on Start button and then on Run.
2. In the box named Open, type "command" (without the quotes) and hit the Enter key.
3. In the window that appears, type "ipconfig" (minus the quotes) and hit Enter key.

It may take a while, depending on the type of connection your PC has. Wireless connections take longer to display the IP address.

Once the prompt displays information, you will see a line like:

IP Address : 192.168.10.14

This is the IP address of your PC in the network. The value given on the right side is a sample value. Your IP address will likely be different from this one, but the format should remain similar.

3.5 Major features of IP, IP datagram.

IP Functions

In the preceding topic I described the general operation of IP and boiled down its primary job as *internetwork datagram delivery*. I also explained the most important characteristics of how IP does this job. With that as a foundation, let's now look a bit deeper, at ***how*** IP "gets the job done". A good way to do this is to examine the various functions that the Internet Protocol includes.

The exact number of IP functions depends on where you "draw the line" between certain activities. For explanatory purposes, however, I view IP as having four basic functions (or more accurately, function sets):

- **Addressing:** In order to perform the job of delivering datagrams, IP must know where to deliver them to! For this reason, IP includes a mechanism for host addressing. Furthermore, since IP operates over internetworks, its system is designed to allow unique addressing of devices across arbitrarily large networks. It also contains a structure to facilitate the routing of datagrams to distant networks if that is required.

  Since most of the other TCP/IP protocols use IP, understanding the IP addressing scheme is of vital importance to comprehending much of what goes on in TCP/IP.

- **Data Encapsulation and Formatting/Packaging:** As the TCP/IP network layer protocol, IP accepts data from the transport layer protocols UDP and TCP. It then encapsulates this data into an IP datagram using a special format prior to transmission.

- **Fragmentation and Reassembly:** IP datagrams are passed down to the data link layer for transmission on the local network. However, the maximum frame size of each physical/data-link network using IP may be different. For this reason, IP includes the ability to *fragment* IP datagrams into pieces so they can each be carried on the local

network. The receiving device uses the *reassembly* function to recreate the whole IP datagram again.

- **Routing / Indirect Delivery:** When an IP datagram must be sent to a destination on the same local network, this can be done easily using the network's underlying LAN/WLAN/WAN protocol using what is sometimes called *direct delivery*. However, in many (if not most cases) the final destination is on a distant network not directly attached to the source. In this situation the datagram must be delivered *indirectly*. This is accomplished by routing the datagram through intermediate devices (shockingly called *routers*). IP accomplishes this in concert with support from the other protocols including ICMP and the TCP/IP gateway/routing protocols such as RIP and BGP.

3.6 The transmission control block (TCB)

Transmission Control Protocol (TCP) keeps track of different information about each connection. TCP set up a complex data structure known as Transmission Control Block (TCB) to do this, which maintains information about the local and remote socket numbers, the send and receive buffers, security and priority values, and the current segment in the queue. The Transmission Control Block (TCB) also manages send and receive sequence numbers.

TCP Sliding Window

The working of the TCP sliding window mechanism can be explained as below.

The sending device can send all packets within the TCP window size (as specified in the TCP header) without receiving an ACK, and should start a timeout timer for each of them.

The receiving device should acknowledge each packet it received, indicating the sequence number of the last well-received packet. After receiving the ACK from the receiving device, the sending device slides the window to right side.

In this case, the sending device can send up to 5 TCP Segments without receiving an acknowledgement from the receiving device. After receiving the acknowledgement for Segment 1 from the receiving device, the sending device can slide its window one TCP Segment to the right side and the sending device can transmit segment 6 also.

If any TCP Segment lost while its journey to the destination, the receiving device cannot acknowledge the sender. Consider while transmission, all other Segments reached the destination except Segment 3. The receiving device can acknowledge up to Segment 2. At the sending device, a timeout will occur and it will re-transmit the lost Segment 3. Now the receiving device has received all the Segments, since only Segment 3 was lost. Now the receiving device will send the ACK for Segment 5, because it has received all the Segments to Segment 5.

Acknowledgement (ACK) for Segment 5 ensures the sender the receiver has succesfully received all the Segments up to 5.

TCP uses a byte level numbering system for communication. If the sequence number for a TCP segment at any instance was 5000 and the Segment carry 500 bytes, the sequence number for the next Segment will be 5000+500+1. That means TCP segment only carries the sequence number of the first byte in the segment.

The Window size is expressed in number of bytes and is determined by the receiving device when the connection is established and can vary later. You might have noticed when transferring big files from one Windows machine to another, initially the time remaining calculation will show a large value and will come down later.

We have four catagories in above example.

1) Bytes already sent and acknowledged (upto Byte 20).

2) Bytes sent but not acknowledged (Bytes 21-24).

3) Bytes the receiver is ready to accept (Bytes 25-28).

4) Bytes the receiver is not ready to accept (Byte 29 onwards).

The Send Window is the sum of Bytes sent but not acknowledged and Bytes the receiver is ready to accept (Usable Window).

3.7 Route discovery protocols

To enable router discovery, the IRDP defines two kinds of ICMP messages:

- The **ICMP Router Solicitation Message** is sent from a computer host to any routers on the local area network to request that they advertise their presence on the network.
- The **ICMP Router Advertisement Message** is sent by a router on the local area network to announce its IP address as available for routing.

When a host boots up, it sends solicitation messages to IP multicast address 224.0.0.2.In response, one or more router may send advertisement messages. If there is more than one router, the host usually picks the first message it gets and adds that router to its routing table. Independently of a solicitation, a router may periodically send out advertisement messages. These messages are not considered a routing protocol, as they do not determine a routing path, just the presence of possible gateways.

UNIT 3 - POSSIBLE QUESTIONS

PART A
(20 MARKS)
ONLINE EXAMINATION

PART B
(Each Question carries 2 marks)

1. Sketch out the neat diagram for ports and sockets in TCP/IP.
2. Write about the TCP/IP internetworking.
3. Write about the IP address structure.
4. Discuss about the major features of IP.
5. Write a short note on major IP services.
6. With an example discuss about the ICMP message formatting
7. Discuss about the passive and active opens in TCP/IP.
8. Briefly explain the route discovery protocols in TCP/IP.
9. Write a short note on major IP services.
10. With an example discuss about the ICMP message formatting.

PART C

Compulsory Question

(Each Question Carries 8 marks)

1. Discuss about the classification of communication protocols.
2. Compare the OSI reference model with TCP/IP model.
3. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.
4. Explain in detail about the X.25 architecture with an example network.
5. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.

KARPAGAM ACADEMY OF HIGHER EDUCATION
DEPARTMENT OF COMPUTER APPLICATIONS
SUBJECT CODE/NAME: 17CAP302, COMPUTER NETWORKS
Degree : PG **Course: MCA** **Year: 2017**

S.NO	QUESTIONS	OPTION 1	OPTION 2	OPTION 3	OPTION 4	ANSWER
	UNIT 3					
1	Network _____ also were relucant to allow users to tap into their resources, due to concerns about security.	Security	Administrators	Applications	Layers	**Administrators**
2	It is also followed the applications such as electronic mail and file transfer should be standardized as well to permit interconnections of _____ applications	End users	Security	Administrators	Applications	**End users**
3	The terms gateway or _____ are used to describe a machine that performs relaying functions between networks.	Switches	Modem	router	Packets	**Router**
4	The term does not mean that they provide fewer functions than a conventional netwrok.	Subnetworks	Network	Protocols	Systems	**Subnetworks**
5	An _____ gateway is designed to remain transparent to the enduser application.	Interconnection	internetworking	internet	interaction	**internetworking**
6	The _____ can dedicate itself to fewer tasks, such as managing the traffic between networks.	End users	Gateway	Administrators	Applications	**Gateway**
7	The gateway considers the application message as nothing more than a transparent _____	PDU	TCP/IP	DTE	DCE	**PDU**
8	The procedures are usually proprietary in nature, and standards do not describe this interface between the gateway and the _____	Subnetworks	Network	Protocols	Systems	**subnetworks**
9	The _____to statement is the publication of IEEE,OSI, and Internet service definitions the host and gateway Protocols.	exception	Systems	Protocols	Networks	**exception**
10	The file transfer software performs a variety of functions and appends a file transfer header to the _____	user data	header file	file systems	procedures	**user data**
11	_____ performs several functions and adds a header to the PDU passed to it.	TCP/IP	IP	TCP	SMTP	**TCP**
12	The _____ from the upper layers is considered to be data to TCP.	TCP/IP	PDU	IP	SMTP	**PDU**
13	The TCP passes the _____ to the network layer which operates with IP.	layers	segment	token	ring	**segment**
14	A datagram is otherwise called as _____	IP	Sockets	TCP/IP	Protocols	**TCP/IP**
15	The frame is otherwise called as _____	datagram	dataunit	packets	source unit	**dataunit**
16	The network _____ is free to manipulate and manage the PDU ina ny manner necessary.	layers	packets	frames	manager	**manager**
17	The datagram is passed to the communications link that is connected to the appropriate subnetwork in _____	lower layers	higher layers	middle layers	top layers	**lower layers**
18	The datagram is re-encapsulated into the datalink layer and passed to the next subntwork is _____	packets	frames	units	TCP units	**frames**
19	The destination _____receives the trafiic through its lower layers and reverses the process that transpired at host A.	Host A	Host C	Host B	protocols	**Host B**
20	The _____ created by the file transfer application is passed to the file transfer application residing at host B.	IP	PDU	TCP/IP	TCP	**PDU**
21	Each application layer process using the TCP/IP protocols must identify itself with a _____	IP address	protocols	packets	port number	**port number**

22	The use of port numbers alsoprovides a multiplexing capability by allowing multiple user programs to communicate concurrently with one application program such as _____	TCP	IP	subprograms	TCP/IP	**TCP**
23	TCP/IP based protocols also use an abstract identifier called a _____	layers	socket	packets	sources	**socket**
24	The socket was derived from the network input/output operations of the BSD _____ system	EPX	LYNX	UNIX	Source protocols	**UNIX**
25	In TCP/IP a socket is the concatenation of a port number and the network address of the host that supports the _____	packet numbers	local address	physical address	port service	**port service**
26	The well known port numbers occupy values ranging from 0 to _____	255	126	64	32	**255**
27	TCP/IP networks use a 32 bit address to identify a host computer and the network to which the host is _____	semidetached	attached	separate	linked unit	**attached**
28	The IP address format is _____	255.255.16.3	local address + network address	Network address + host address	16.12.53.54	**Network address + host address**
29	The host address is also called as _____	local addess	base address	network address	IP address	**local address**
30	Class B addresses are used for networks of _____ size.	immediate	interface	predefined	intermediate	**intermediate**
31	Class C networks contain fewer than _____ hosts.	256	255	0	1	**256**
32	Class D addresses are reserved for multicasting , a form of broadcasting but within a _____ area.	unlimited	limited	infinte	numbered	**limted**
33	IP is an example of a _____ service.	connection oriented	wireless	connectionless	Wifi	**connecionless**
34	The IP gateway enforces a maximum queue length size, and if this queue length is violated the buffers will _____	underflow	limited	infinte	overflow.	**overflow**
35	Class A addresses provide for networks that have a large number of _____	hosts	mini hosts	systems	terminals	**hosts**
36	A higher level transport layer protocol called as _____	IP	TCP	UDP	SMTP	**TCP**
37	____ hides the underlying subnetwork from the enduser.	TCP	UDP	IP	TCP/IP	**IP**
38	IP is an _____ best effort, datagram type protocol, it has no reliability mechanisms.	reliable	not varying	varying	unreliable,	**unreliable**
39	The _____ field identifies the version of IP in use.	version	update	cummulative	summative	**version**
40	The _____ field contains four bits which are set to a value to indicate the length of teh datgram header.	bit address	header length	local address	host address	**header length**
41	The _____ field can be used to identify several QOS functions provided for in Internet.	BOS	COS	Terminal	TOS	**TOS**
42	The maximum possible lengthof a datgram is _____ octets.	124	1024	65536	16	**65536**
43	All _____ must accomodate datagrams of 576 octets in total length.	packets	terminals	hosts	gateways	**gateways**
44	The IP protocol use _____ fileds in the header to control datagram fragmentation and reassembly.	three	four	five	two	**three**
45	The _____ parameter is used to measure the time a datagram has been in the internet.	CMOS	SMTP	TTL	UDP	**TTL**
46	The _____ field is used to identify the most next level protocol above the IP tha is toreceive the datagram at the final host destination.	address	protocol	hosts	terminals	**protocol**
47	The _____ checksum is used to detect a distortium that may have occurred in the header.	host address	local address	header	files	**header length**
48	IP carries ____ addresses in the datagram.	three	four	no	Two	**Two**
49	The _____ field may be ised to make certain that the datagram header aligns on an exact 32 bit boundary.	padding	addressing	hosting	terminating	**padding**

50	IP uses a mechanism called _____ as part of its routing algorithm.	routing protocol	base address	source routing	local routing	**source routing**
51	_____ source routing requires that the datagram travel only through the networks whose addresses are indicated in the source list.	complex	strict	basics	liberal	**strict**
52	ICMP means _____	Internet Control Machine Protocol	Interchange Control Machine Protocol	Internet Control Message Protocol	Interchange Computer Message Protocol	**Internet Control Message Protocol**
53	TCP is a _____ protocol.	connectionless	importatnt	wireless	connection oriented	**connection oriented**
54	TCP is responsible for the _____ of each of the characters passed it to from an upper layer.	unreliable	reliable transfer	basic transfer	host transfering	**reliable transfer**
55	The ____ mode allows the ULP to tell the TCP.	passive open	active open	terminal open	IP hosting	**passive open**
56	TCP must remember several things about each virtual connection, it stores information in a _____	LAN	WAN	TCB	MAN	**TCB**
57	Interworking, gateways and routers play a very important role in _____ based networks.	IP	TCP	TCP/IP	TCB	**TCP/IP**
58	The _____ is used to provide network reachability information between neighbhouring gateways.	MGP	FGP	protocols	EGP	**EGP**
59	The _____ was developed based on research at the Xerox Palo Alto Research Center routing protocols.	RIP	RIU	FTP	SMTP	**RIP**
60	_____ forsimple file transfer services in the application layer.	FTP	TFTP	SMTP	HTTP	**TFTP**

KARPAGAM ACADEMY OF HIGHER EDUCATION
(Deemed University Established Under Section 3 of UGC Act 1956)
Coimbatore - 641021.
(For the candidates admitted from 2017 onwards)

DEPARTMENT OF CS, CA & IT

SYLLABUS:

UNIT IV

Polling/Selection Protocols

Character and bit protocols, binary synchronous control (BSC) HDLC; HOLC options, HDLC frame format, code transparency and synchronization, HDLC transmission process, HDLC subsets, SDLC; Protocol conversion.

Switching and Routing in Networks

Message switching, packet switching, when and when not to use packet switching, packet routing, packet switching support to circuit switching networks.

The X.25 Network and Supporting Protocols

Features of X.25, Layers of X.25 and the Physical layer, X.25 and the data link layer.companion standards to X.25, features of X.25, X.25 channel options, flow control principles, other packet types, X.25 logical channel states, packet formats. internetworking, connectionless mode networks, the frame relay and X.25 stacks.

Polling/Selection Protocols

4.1 Character and bit protocols

A **bit-oriented protocol** is a communications protocol that sees the transmitted data as an *opaque* stream of bits with no semantics, or meaning. Control codes are defined in terms of bit sequences instead of characters. Bit oriented protocol can transfer data frames regardless of frame contents. It can also be stated as "bit stuffing" this technique allows the data frames to contain an arbitrary number of bits and allows character codes with arbitrary number of bits per character.

Synchronous framing High-Level Data Link Control is a popular bit-oriented protocol. Synchronous framing High-Level Data Link Control may work like this:

- Each frame begins and ends with a special bit pattern 01111110, called a flag byte.
- A bit stuffing technique is used to prevent the receiver from detecting the special flag byte in user data e.g. whenever the sender's data link layer encounters 5 consecutive ones in the data, it automatically stuffs 0 into the outgoing stream.

Application Domains for Selected Data Link Protocols

	Circuit	Operation	Orientation	Low Bit Rate	High Bit Rate
Point-to-point	Local circuit eg Null Modem cable	End-to-End	Connection	Character Oriented Idle RQ eg Kermit	Bit Oriented Continuous RQ eg HDLC
	Analogue PSTN using Modems	End-to-End	Connection	Character Oriented Idle RQ eg Kermit	Bit Oriented Continuous RQ eg HDLC or LAP-M (V.32)
	Leased Digital Circuit	End-to-End	Connection	Character Oriented Idle RQ eg Kermit or Bit Oriented Continuous RQ eg TCP	Bit Oriented Continuous RQ eg HDLC or TCP
	Leased Digital Circuit	End-to-End	Connectionless	Bit Oriented Continuous RQ eg IP	Bit Oriented Continuous RQ eg IP
Multipoint /Multidrop	Leased Digital Circuit	Poll-Select	Connection	Character Idle RQ eg BSC	Bit Oriented Continuous RQ eg HDLC subset NRM (Normal Response Mode)
Switched WANs	Packet Switched	Local Significance DTE<>DCE	Connection (Pacnet) or connectionless (datagram)	Bit Oriented Continuous RQ eg HDLC subset LAPB (X.25)	
		End to end	Connectionless		Bit Oriented Continuous

			or connectionless (datagram)		RQ eg HDLC subset LAPD extended (Frame Relay)
	Circuit Switched	End-to-End	Connection		Bit Oriented Continuous RQ eg HDLC subset LAPD (ISDN)
LAN	Bus or Ring	End-to-End	Connectionless		CSMA/CD as in Ethernet

4.2 Binary synchronous control (BSC) HDLC

Binary Synchronous Communication (**BSC** or **Bisync**) is an IBM character-oriented, half-duplex link protocol, announced in 1967 after the introduction of System/360. It replaced the synchronous transmit-receive (STR) protocol used with second generation computers. The intent was that common link management rules could be used with three different character encodings for messages. Six-bit Transcode looked backwards to older systems; USASCII with 128 characters and EBCDIC with 256 characters looked forward. Transcode disappeared very quickly but the EBCDIC and USASCII dialects of Bisync continued in use.

Framing

Bisync differs from protocols that succeeded it in the complexity of message framing. Later protocols use a single framing scheme for all messages sent by the protocol. HDLC, Digital Data Communications Message Protocol (DDCMP), Point-to-Point Protocol (PPP), etc. each have different framing schemes but only one frame format exists within a specific protocol. Bisync has five different framing formats.[*citation needed*]

BSC Link Control Characters

Char	EBCDIC (hexadecimal)	USASCII (hexadecimal)	Transcode (hexadecimal)	Description
SYN	32	16	3A	Synchronous idle
SOH	01	01	00	Start of heading
STX	02	02	0A	Start of text
ETB	26	17	0F	End of transmission block

ETX 03	03	2E	End of text
EOT 37	04	1E	End of transmission
ENQ 2D	05	2D	Enquiry
NAK 3D	15	3D	Negative acknowledgement
DLE 10	10	1F	Data link escape
ITB 1F	1F (US)	1D (US)	Intermediate block check character

ACK0 and **ACK1** (even/odd affirmative acknowledgement) are encoded as two characters—DLE '70'x, and DLE / for EBCDIC, DLE 0 and DLE 1 for USASII, DLE - and DLE T for Transcode. **WABT** (wait before transmit) was encoded as DLE ", DLE ?, or DLE W.

All frame formats begin with at least two SYN bytes. The binary form of the SYN byte has the property that no rotation of the byte is equal to the original. This allows the receiver to find the beginning of a frame by searching the received bit stream for the SYN pattern. When this is found, tentative byte synchronization has been achieved. If the next character is also a SYN, character synchronization has been achieved. The receiver then searches for a character that can start a frame. Characters outside of this set are described as "leading graphics". They are sometimes used to identify the sender of a frame. Long messages have SYN bytes inserted approximately every second to maintain synchronization. These are ignored by the receiver.

A normal block ending character (ETB or ETX) is followed by a check sum (block check character or BCC). For USASCII, this is a one character longitudinal redundancy check (LRC); for Transcode and EBCDIC, the check sum is a two character cyclic redundancy check(CRC). A data frame may contain an intermediate check sum preceded by an ITB character. This ability to include intermediate check sums in a long data frame allows a considerable improvement of the error detection probability. USASCII characters are also transmitted using *odd parity* for additional checking.

Pad characters are required following a line turn-around—NAK, EOT, ENQ, ACK0, ACK1. If the transmission ends with EOT or ETX the pad follows the BCC. This pad is either all '1' bits or alternating '0' and '1' bits. The next transmission begins with a pad character which can be either of the above or a SYN.

An optional *heading* containing control information can precede data in a frame. The content of the heading is not defined by the protocol but is defined for each specific device. The heading, if present, is preceded by an SOH (start of heading) character and followed by an STX (start of text).[4]

Text data normally follows the heading, begun by the STX, and terminated by ETX (end of text) or ETB (end transmission block).

Normal data frames do not allow certain characters to appear in the data. These are the block ending characters: ETB, ETX and ENQ and the ITB and SYN characters. The number of unique characters that can be transmitted is therefore limited to 59 for Transcode, 123 for USASCII, or 251 for EBCDIC.

Transparent data framing provides an unrestricted alphabet of 64, 128 or 256 characters. In transparent mode block framing characters such as ETB, ETX, and SYN are preceded by a DLE character to indicate their control significance (The DLE character itself is represented by the sequence DLE DLE). This technique became known as character stuffing, by analogy with bit stuffing.

Link control

The link control protocol is similar to STR. The designers attempted to protect against simple transmission errors. The protocol requires that every message be acknowledged (ACK0/ACK1) or negatively acknowledged (NAK), so transmission of small packets has high transmission overhead. The protocol can recover from a corrupted data frame, a lost data frame, and a lost acknowledgment.

Error recovery is by retransmission of the corrupted frame. Since Bisync data packets are not serial-numbered, it's considered possible for a data frame to go missing without the receiver realizing it. Therefore, alternating ACK0s and ACK1s are deployed; if the transmitter receives the wrong ACK, it can assume a data packet (or an ACK) went missing. A potential flaw is that corruption of ACK0 into ACK1 could result in duplication of a data frame.

Error protection for ACK0 and ACK1 is weak. The Hamming distance between the two messages is only two bits.

The protocol is half-duplex (2-wire). In this environment, packets or frames of transmission are strictly unidirectional, necessitating 'turn-around' for even the simplest purposes, such as acknowledgments. Turn-around involves

- the reversal of transmission direction,
- quiescing of line echo,
- resyncing.

In a 2-wire environment, this causes a noticeable round-trip delay and reduces performance.

Some datasets support full-duplex operation, and full-duplex (4-wire) can be used in many circumstances to improve performance by eliminating the turn-around time, at the added expense of 4-wire installation and support. In typical full-duplex, data packets are transmitted along one wire pair while the acknowledgements are returned along the other.

HDLC options, HDLC frame format,

HDLC (High-level Data Link Control)

HDLC (High-level Data Link Control) is a group of protocols or rules for transmitting data between network points (sometimes called nodes). In HDLC, data is organized into a unit (called a *frame*) and sent across a network to a destination that verifies its successful arrival. The HDLC protocol also manages the flow or pacing at which data is sent. HDLC is one of the most commonly-used protocols in what is layer 2 of the industry communication reference model called Open Systems Interconnection (OSI). (Layer 1 is the detailed physical level that involves actually generating and receiving the electronic signals. Layer 3 is the higher level that has knowledge about the network, including access to router tables that indicate where to forward or send data. On sending, programming in layer 3 creates a frame that usually contains source and destination network addresses. HDLC (layer 2) encapsulates the layer 3 frame, adding data link control information to a new, larger frame.

Now an ISO standard, HDLC is based on IBM's SDLC protocol, which is widely used by IBM's large customer base in mainframe computer environments. In HDLC, the protocol that is essentially SDLC is known as Normal Response Mode (NRM). In Normal Response Mode, a primary station (usually at the mainframe computer) sends data to secondary stations that may be local or may be at remote locations on dedicated leased lines in what is called a multidrop or multipoint network. (This is not the network we usually think of; it's a nonpublic closed network. In this arrangement, although communication is usually half-duplex.)

Variations of HDLC are also used for the public networks that use the X.25 communications protocol and for frame relay, a protocol used in both and wide area network, public and private.

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In the X.25 version of HDLC, the data frame contains a packet. (An X.25 network is one in which packets of data are moved to their destination along routes determined by network conditions as perceived by routers and reassembled in the right order at the ultimate destination.) The X.25 version of HDLC uses peer-to-peer communication with both ends able to initiate communication on duplex links. This mode of HDLC is known as Link Access Procedure Balanced (LAPB).

The following table summarizes the HDLC variations and who uses them.

HDLC SUBSET	USES
NRM (Normal Response Mode)	Multipoint networks that typically use SDLC
LAP (Link Access Procedure)	Early X.25 implementations
LAPB (Link Access Procedure, Balanced)	Current X.25 implementations
LAPD (Link Access Procedure for the Integrated Services Digital Network D channel)	ISDN D channel and frame relay
LAPM (Link Access Procedure for Modems)	Error-correcting modems (specified as part of V.42)

.

Switching and Routing in Networks

There are a number of ways to perform switching:

Different types of switching techniques are employed to provide communication between two computers. These are: Circuit switching, message switching and packet switching.

CircuitSwitching

In this technique, first the complete physical connection between two computers is established and then data are transmitted from the source computer to the destination computer. That is, when a computer places a telephone call, the switching equipment within the telephone system seeks out a physical copperpath all the way from sender telephone to the receiver's

telephone. The important property of this switching technique is to setup an end-to-end path (connection) between computer before any data can be sent. This method involves the physical interconnection of two devices. A good example of circuit switching involves the Public phone network. A data example would be the classic A/B switch!

Message Switching

In this technique, the source computer sends data or the message to the switching office first, which stores the data in its buffer. It then looks for a free link to another switching office and then sends the data to this office. This process is continued until the data are delivered to the destination computers. Owing to its working principle, it is also known as store and forward. That is, store first (in switching office), forward later, one jump at a time.

Message Switching techniques were originally used in data communications. An example would be early "store and forward" paper tape relay systems. E-Mail delivery is another example of message switching. In voice systems, you can find Voice Mail delivery systems on the Internet. The classic "forward voice mail" capability in some voice mail systems is another example.

Packet Switching

With message switching, there is no limit on block size, in contrast, packet switching places a tight upper limit on block size. A fixed size of packet which can be transmitted across the networkis specified. Another point of its difference from message switching is that data packets are stored on the disk in message switching whereas in packet switching, all the packets of fixed size are stored in main memory. This improves the performance as the access time (time taken to access a data packet) is reduced, thus, the throughput (measure of performance) of the network is improved

Packet Switching techniques switch packets of data between destinations. Traditionally, this applied to X.25 techniques, but this also applies to TCP/IP and IPX/SPX routers also. Proprietary Frame Relay switches can switch voice signals.

Cell Switching

Cell Switching is similar to packet switching, except that the switching does not necessarily occur on packet boundaries. This is ideal for an integrated environment and is found within Cell-based networks, such as ATM. Cell-switching can handle both digital voice and data signals.

The X.25 Network and Supporting Protocols

X.25 is a standard suite of protocols used for packet-switched communications over a wide area network—a WAN. A protocol is an agreed-upon set of procedures and rules. Two devices that follow the same protocols can understand each other and exchange data.

History of X.25

X.25 was originally developed in the 1970s to carry voice over analog telephone lines—dial-up networks. Typical applications of X.25 included automatic teller machine networks and credit card verification networks.

X.25 also supported a variety of mainframe terminal and server applications. The 1980s were the heydays of X-25 technology when it was used by public data networks Compuserve, Tymnet, Telenet, and others. In the early 90s, many X.25 networks were replaced by Frame Relay in the U.S. Many older public networks outside the U.S. continued to use X.25 until just recently. Most networks that once required X.25 now use the less complex Internet Protocol. X-25 is still used in some ATMs and credit card verification networks.

X-25 Structure

Each X.25 packet contained up to 128 bytes of data. The X.25 network handled packet assembly at the source device, the delivery, and the reassembly at the destination. X.25 packet delivery technology included not only switching and network-layer routing but also error checking and retransmission logic should delivery failures occur. X.25 supported multiple simultaneous conversations by multiplexing packets and using virtual communication channels.

X-25 offered three basic layers of protocols:

- Physical layer
- Data link layer
- Packet layer

X-25 predates the OSI Reference Model, but the X-25 layers are analogous to the physical layer, data link layer and network layer of the standard OSI model.

With the widespread acceptance of Internet Protocol (IP) as a standard for corporate networks, X.25 applications have now migrated to cheaper solutions using IP as the network layer protocol and replacing the lower layers of X.25 with Ethernet or with new ATM hardware.

X.25 is a standard used by many older public networks specially outside the U.S.

• This was developed in *1970s* by CCITT for providing an interface between public packet-switched network and their customers.

• The packet switching networks use X.25 protocol. The X.25 recommendations were first prepared in 1976 and then revised in 1978, 1980 and 1984.

• **X.25** was developed for computer connections, used for terminal/timesharing connection.

• This protocol is based on the protocols used in early packet switching networks such as ARPANET, DATAPAC, and TRANSPAC etc.

• X.25 Packet Switched networks allows remote devices to communicate with each other across high speed digital links without the expense of individual leased lines.

• A protocol X.21 which is a physical layer protocol is used to specify the physical electrical and procedural interface between the host and network.

• The problem with this standard is that it needs digital signal rather than analog signals on telephone lines.

• So not many networks support this standard. Instead RS 232 standard is defined.

• The data link layer standard has a number of variations. It is designed for error detection and corrections.

• The network layer protocol performs the addressing, flow control, delivery confirmation etc.

• It allows the user to establish virtual circuits and send packets on them. These packets are delivered to the destination reliably and in order.

• X.25 is a connection oriented service. It supports switched virtual circuits as well as the permanent circuits.

• Packet Switching is a technique whereby the network routes individual packets of HDLC data between different destinations based on addressing within each packet.

• A switched virtual circuit is established between a computer and network when the computer sends a packet to the network requesting to make a call to other computer.

• Packets can then be sent over this connection from sender to receiver.

• X.25 provides the flow control, to avoid a fast sender overriding a slow or busy receiver.

• A permanent virtual circuit is analogous to-a leased line. It is set up in advance with a mutual agreement between the users.

• Since it is always present, no call set up is required for its use.

• **In** order to allow the computers which do not use the X.25 to communicate with the X.25 network a packet assembler disassembler (PAD) is used.

• PAD is required to be installed along with each computer which does not use X.25.

• X.25 defines the interface for exchange of packets between a **DTE** and switch data subnetwork node.

Three Layers of X.25:

The X.25 interface is defined at three levels:

The three levels are:

(i) Physical layer (level 1)

(ii) Data link layer (level 2)

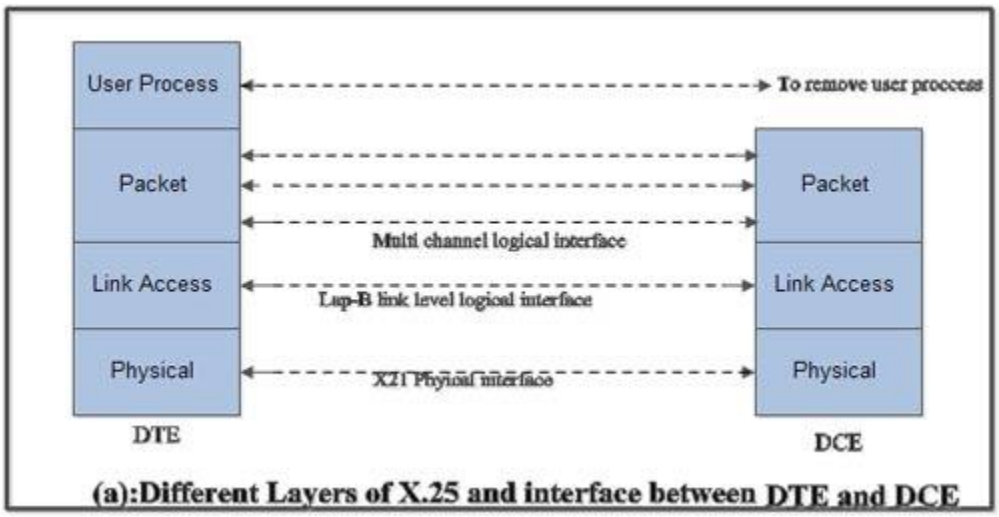
(iii) Packet layer (level 3).

• These three layers correspond to the three lower most layers of the ISO-OSI reference model. The physical layer takes care of the interface between a computer terminal and the link which attaches it to the packet switching node.

• The X.25 defines the interface for exchange of packets between the user's machine (DTE) and the packet switching node to which this DTE is attached which is called as DCE.

• The three layers of X.25 interface are as shown in Fig.(a).

• At the physical level X.21 physical interface is being used which is defined for circuit switched data network. At the data link level, X.25 specifies the link access procedure-B (LAP-B) protocol which is a subset of HDLC protocol.



(a):Different Layers of X.25 and interface between DTE and DCE

• At the network level (3rd level), X.25 defines a protocol for an access to packet data subnetwork.

• This protocol defines the format, content and procedures for exchange of control and data transfer packets. The packet layer provides an external virtual circuit service.

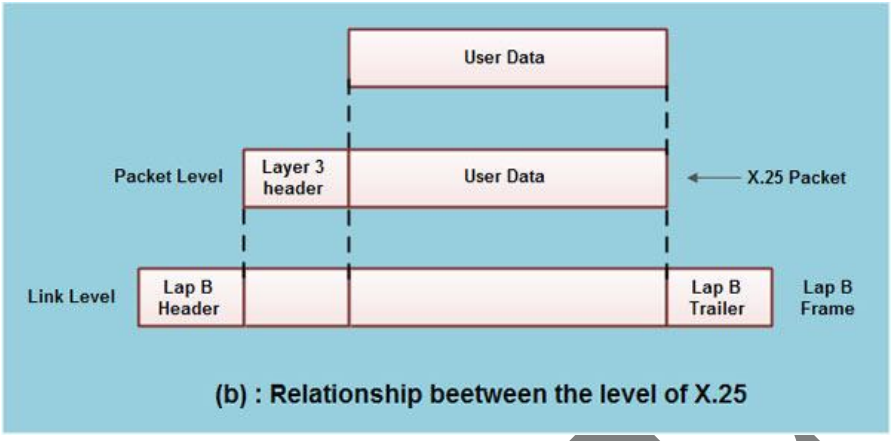
• Fig.(b) shows the relationship between the levels of x'25. User data is passed down to X.25 level 3.

• This data then appends the control information as a header to form a packet. This control .information is then used in the operation of the protocol.

• The entire X.25 packet formed at the packet level is then passed down to the second layer i.e. the data link layer.

• The control information is appended at the front and back of the packet forming a LAP-B frame. The control information in LAP-B frame is needed for the operation of the LAP-B protocol.

• This frame is then passed to the physical layer for transmission.



(b) : Relationship beetween the level of X.25

Virtual Circuit Service

• With the X25 packet layer, data are transmitted in packets over external virtual circuits, The virtual circuit service of X25 provides for two types of virtual circuits,

• The virtual circuit service of X25 provides for two types of virtual circuits i.e. "virtual call" and "permanent virtual circuit".

• A virtual call is a dynamically established virtual circuit using a call set up and call clearing procedure.

• A permanent virtual circuit is a fixed, network assigned virtual circuit. Data transfer takes place as with virtual calls, but no call set up or clearing is required.

Characteristics of X.25

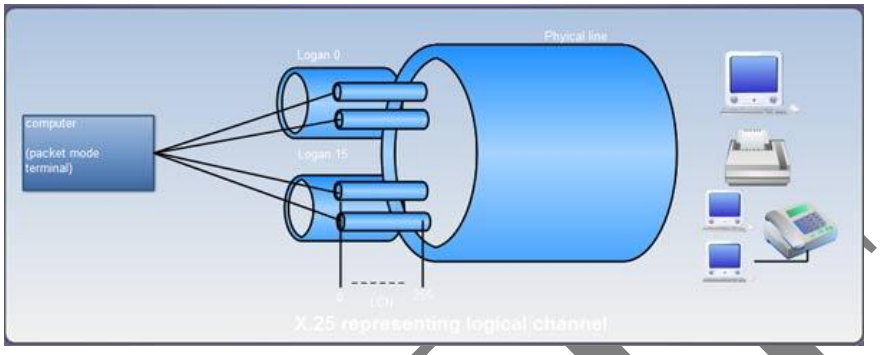
In addition to the characteristics of the packet switched network, X.25 has the following characteristics:

1. Multiple logical channels can be set on a single physical line
2. Terminals of different communication speeds can communicate
3. The procedure for transmission controls can be changed.

Multiple Logical Channels can be set on a Single Physical Line

The terminal connected to the packet switched network can communicate with multiple terminals at the same time using a single physical line. This makes it possible to set multiple logical paths

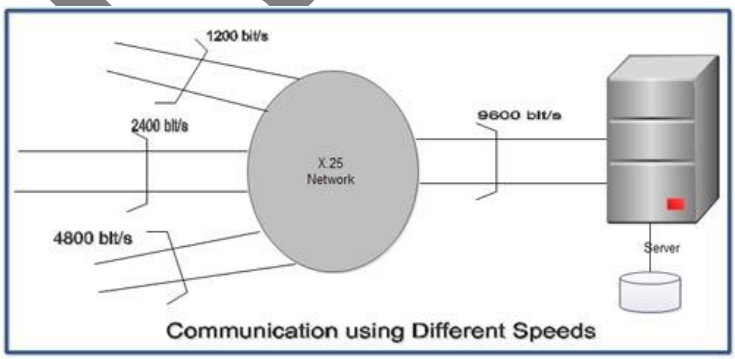
called logical channels on a single physical line. Multiple communications thus takes place through these logical channels. Based on the X.25 rules, 4096 logical channel can be set on a single physical line. To enable control of 4096 logical channels there are 16 logical channel groups. Each logical channel group is divided into 256 logical channels. These channel groups are known as LCGN (Logical Channel Group Number) and LCN (Logical Channel Number).



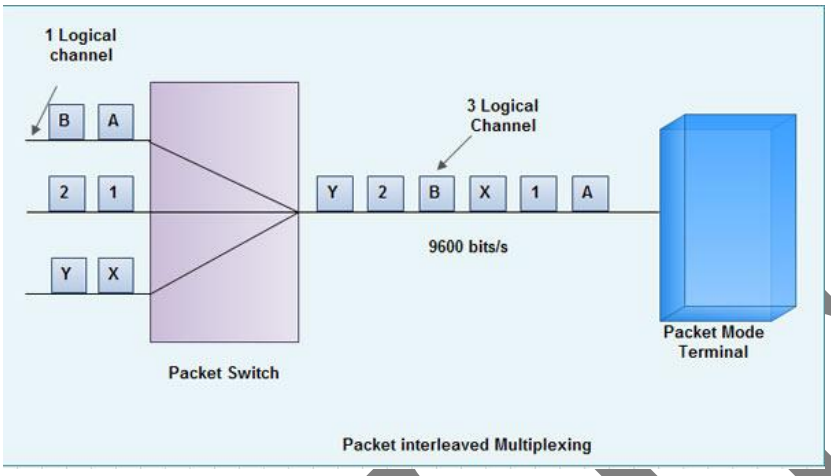
Terminals of Different Communication Speeds can communicate

As X.25 uses the store and forward method, therefore, the communication is possible. In other words, a terminal of 1.2 Kbits/s can communicate with a host computer at 9600 bits/s through the packet switched network. When the 'telephone network or a leased line is used, this type of communication cannot be established. In other words, in these environments, the transmission speed of the sender should be the same as that of the receiver.

The reason that communication between terminals with different communication speeds is possible is that the senders and the receivers are not physically connected. Data transmission from a 1.2 Kbits/s terminal is temporarily stored in the receiving buffer of the packet switched network and the data is then passed through the network and transmitted to the host computer at 9600 bits/s.

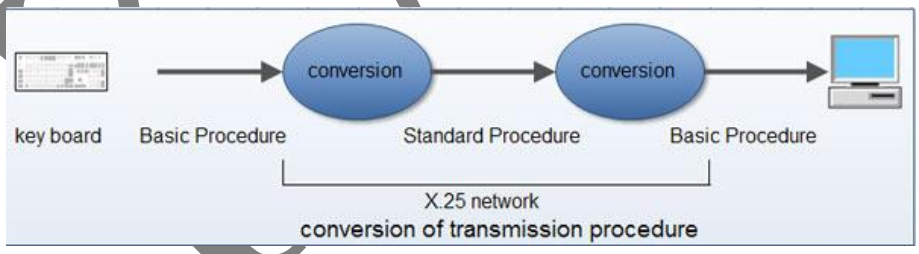


By using the above 2 features the network can be established. By applying a higher line speed to the host computer than the terminal and setting multiple logical channels, the number of lines at the computer can be reduced.



The Procedure for Transmission Controls can be changed

It is possible to change the procedure for transmission control. As we know that X.25 uses the store and forward method, therefore, all data must be once stored in the packet switched unit. By implementing a protocol conversion function to the packet switched unit can connect the devices with different transmission control (basic procedure and X.25 protocol).With the help of this method, any terminal that cannot handle packets cannot be connected to the packet switched network. A terminal that cannot handle packets is called an NPT (Non-packet mode terminal).



UNIT 4 - POSSIBLE QUESTIONS

PART A
(20 MARKS)
ONLINE EXAMIANTION

PART B
(Each Question carries 2 marks)

1. Discuss about the features of X.25 network.
2. Describe about the layers of X.25 network.
3. What are the channels options in X.25and discuss about it?
4. What is meant by flow control and time limits in X.25 Network?
5. What are the facilities available in X.25 network?
6. Write about the concept of code transparency and synchronization.
7. Explain the concept of packet switching and packet routing.
8. Explain the process of HDLC transmission system.
9. Briefly explain the SDLC options.

PART C

Compulsory Question

(Each Question Carries 8 marks)

1. Discuss about the classification of communication protocols.
2. Compare the OSI reference model with TCP/IP model.
3. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.
4. Explain in detail about the X.25 architecture with an example network.
5. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.

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DEPARTMENT OF COMPUTER APPLICATIONS

SUBJECT CODE/NAME: 17CAP302, COMPUTER NETWORKS

Degree : PG **Course: MCA** **Year: 2017**

S.NO	QUESTIONS	OPTION 1	OPTION 2	OPTION 3	OPTION 4	ANSWER
	UNIT 4					
1	The polling protocols in use today are either _____	character oriented	space oriented	network oriented	physically depended	**character oriented**
2	IBM introduced the first generalpurpose data link control to support multipoint and _____ configurations.	multipoint	Point to point	multidrop	single point	**Point to point**
3	Protocol is otherwise called as _____	unidirectional	bidirectional	Bisync	asynchronous	**Bisync**
4	BSC is a _____ protocol.	simplex	duplex	full duplex	half duplex	**half duplex**
5	_____ must be excluded fromt the text and header fileds.	Control codes	simple code	dual codes	multicodes	**Control codes**
6	The _____ places the line to transparent text mode which allows the transmission of any bit pattern.	DCE	DLE	DTE	SCMP	**DLE**
7	The _____ is used by a master station to control the operations on the link such as the transmissionof polling and selection frames.	Control codes	duplex	control modes	simpex	**control modes**
8	The _____ is used for the transmission of an information block or blocks to and from the stations.	network	packets	switches	message	**message**
9	BSC alsoprovides for _____ operation on a point topoint circuit.	intention	contention	dual codes	single codes	**contention**
10	The transmitting station knows the exact order of frames it transmits and it expects to receive ACKs to its _____	transmitted	non transmitted	transmission	transferred	**transmission**
11	The _____ is encouraged to check with specific vendors for their actual implementationof the HDLC structure.	header	files	data	reader	**reader**
12	The _____ disconnected state prohibits a station from transmitting or receiving information.	logically	physically	message	packets	**logically**
13	The _____ state is define by specific vendors and is outside the standards of HDLC.	declaration	initialization	duplexing	multiplexing	**initialization**
14	The _____transfer rate permits the secondary, primary adn combined stations to transmit and receive user information.	data	raw data	information	mutli user data	**information**
15	In HDLC frame the information foramt frame is used to transmit _____ between two devices.	multi user data	sinlge user data	no data	user data	**user data**
16	The _____ format frame performs control functions such as the acknowledgement of frames, the request for the retransmission of frames and the request for the temporary suspension of the transmission of frames.	unsupervisory	supervisory	declared data	unformatted data	**supervisory**
17	The _____ format frame is also used for control purposes in HDLC format.	numbered	formatted	unnumbered	unformatted	**unnumbered**
18	The _____ identifies the primary or secondary station involved in the particular frame trnsmission.	header field	physical address	logical address	address field	**address field**
19	HDLC does not rely on a specific code _____ for line control.	ASCII/IA5	ASCII	ANSI & ISO	ISO	**ASCII/IA5**

20	SNRM,SARM, and SABM are the commands of _____ commands	line setting	mode setting	channel setting	address setting	**mode setting**
21	HDLC standard has provided a cohesive foundation from which to implement subsets of the _____	HDLC network	HDLC address	HDLC protocol	HDLC base address	**HDLC protocol**
22	The HDLC superset structure provides for bit oriented protocols to recognize and use the same procedures among different types of _____	network	packets	routers	applications	**applications**
23	_____ is an earlier subset of HDLC.	LAP	LCP	TCP	IP	**LAP**
24	_____ is IBM's version or rendition of HDLC.	HDLC 7	SDLC	HDLC 5	HDLC 8	**SDLC**
25	The pervasive method for switching data communications traffic ws _____	packet switching	packets	random switching	message switching	**message swtiching**
26	Message swtiching is a store and forward technology because of the storage capability at the switch , usually in the form of _____	data units	performers	users	disk units	**disk units**
27	The message swtiching technology usually operates with a _____ relationship.	master/slave	slave	master	attached	**master/slave**
28	In the 1970s the industry began to move toward a different data communications switching structure known as _____	routers	packet swtiching	packets	message swtiching	**packet swtiching**
29	Packet swtiching reduces the vulnerability of _____	line failure	layer failure	network failure	address format	**network failure**
30	Packet switching is so named because user data are broken into _____ pieces	larger	very tiny	2	smaller	**smaller**
31	Port and channel multiplexing are referred toa s a _____ channel.	reality	virtual	packet	router	**virtual**
32	_____ entails the use of logic at the swithces to move the data packets through the network to the end destination.	networking routing	random routing	specific routing	routing	**networking routing**
33	A centralized _____ approach also entails considerable overhead, with the requirement for the network control centre to distribute routing information to its subordinate packet switches.	random routing	routing network	specific routing	routing	**routing network**
34	A centralised routing network provides for one network control centre to determine the routing of the packets through the _____	reality	virtual	network	router	**network**
35	_____ given packet generates additional identical packets.	TCE	TCP	IP	TME	**TME**
36	The _____ systems are further classified as partial path directory or full path directory.	directory	packets	routers	applications	**directory**
37	_____ every possible path is used between a transmitting and receiving DCE.	packet switching	Packet flooding	networking routing	specific routing	**Packet flooding**
38	_____ in networking where multipoint links, various LANs and packet switch topologies exist is more complex.	reality	virtual	Broadcasting	router	**Broadcasting**
39	_____ directory approach is illustrated by the department of Defense's ARPANET.	adaptive	files	data	reader	**adaptive**
40	_____ is another technique used for network packet swtiching.	routers	packet swtiching	packets	Random routing	**random routing**
41	_____ can best be introduced by an examination of the system services control point.	IP	SNA	TCP	TCE	**SNA**
42	_____ is responssible for the SNA network .	SMTP	HTTP	SSCP	FTP	**SSCP**

43	_____ was the first major packet swtiching system implemented by AT&T.	CCIS	SNA	TCP	TCE	**CCIS**
44	Call seta were called as _____	data units	sinlge units	memory	signal units.	**signal units**
45	X.25 is also called as ____	white book	Gray book	black book	blue book	**Gray book**
46	A _____ network and the user stations must have control mechanisms when they interface with each other.	messages	blogs	packet	switches	**packet**
47	The recommended physical layer interface between the _____ and _____ is X.21	DCE	DTE	TCP/IP	DCE and DTE	**DCE and DTE**
48	X.25 assumes the _____ layer to be LAPB	network	datalink	physical	application	**datalink**
49	X.25 operates on the premises of ____ circuit services	physical	logical	virtual	original	**virtual**
50	X,25 uses logical channel numbers to identify the DTE connections to the _____	network	messages	blogs	packet	**network**
51	A _____ resembles some of the procedures associated with telephone dialup lines.	video call	virtual call	audio call	calling	**virtual call**
52	The interrupt procedure allows a DTE to transmit one nonsequenced packet to _____ DTE.	same	different	another	nothing	**another**
53	The _____ and ____packets are used in a fashion quite similar to the same commands in the HDLC and LAPB subset.	RR	RR and RNR	RNR	TCP/IP	**RR and RNR**
54	_____ channel states provide the foundation for managing the DTE/DCE connection.	datalink	physical	application	logical	**logical**
55	Every packet transferred across the _____ interface to the network must contain atleast three octets.	DTE	DTE/DCE	DCE	TCP	**DTE/DCE**
56	The transport layer requires the end user to specify a _____ from the network.	CMOS	MOS	QOS	ECMP	**QOS**
57	Class 3 is called in QOS as _____	error recovery class	Error Detection	error correction	error recovery	**error recovery class**
58	Class 4 is called in QOS as _____	Error Detection	recovery class	error detection and recovery class	Failures	**error detection and recovery class**
59	At the top is the _____independence convergence fucntion which provides the relay and routing services for internetworking.	network	subnetwork	mask	packets	**subnetwork**
60	The _____ layer protocols are not affected by IP.	subnetworks	packets	network	subnetmask	**network**

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Coimbatore - 641021.
(For the candidates admitted from 2017 onwards)

DEPARTMENT OF CS, CA & IT

SYLLABUS:

UNIT V

Network Security: IP Security: Architecture, Authentication header, Encapsulating security payloads, is combining security associations, key management. Web Security: Secure socket layer and transport layer security, secure electronic transaction (SET). System Security: Intruders, Viruses and related threads, firewall design principals, trusted systems.

5.1 Network Security

Security can be defined as state of freedom from a danger, risk or attack. Information security can be defined as the task of guarding information which is processed by a server, stored on a storage device, and transmitted over a network like Local Area Network or the public Internet. Information security means protecting information and information systems from unauthorized access, use, disclosure, disruption, modification or destruction

Introduction to AAA

AAA stands for Authentication, Authorization and Accounting. AAA are a set of primary concepts that aid in understanding computer and network security as well as access control. These concepts are used daily to protect property, data, and systems from intentional or even unintentional damage. AAA is used to support the Confidentiality, Integrity, and Availability (CIA) security concept.

Confidentiality: The term confidentiality means that the data which is confidential should remain confidential. In other words, confidentiality means secret should stay secret.

Integrity: The term integrity means that the data being worked with is the correct data, which is not tampered or altered.

Availability: The term availability means that the data you need should always be available to you.

Authentication provides a way of identifying a user, typically requiring a Userid/Password combo before granting a session. Authentication process controls access by requiring valid user credentials. After the Authentication process is completed successfully, a user must be given authorization (permission) for carrying out tasks within the server. Authorization is the process that determines whether the user has the authority to carry out a specific task. Authorization controls access to the resources after the user has been authenticated. The last one is Accounting. Accounting keeps track of the activities the user has performed in the server.

5.2 IP Security

Internet protocol security (IPsec) is a set of protocols that provides security for Internet Protocol. It can use cryptography to provide security. IPsec can be used for the setting up of virtual private networks (VPNs) in a secure manner.

IPsec involves two security services:

- Authentication Header (AH): This authenticates the sender and it discovers any changes in data during transmission.
- Encapsulating Security Payload (ESP): This not only performs authentication for the sender but also encrypts the data being sent.

There are two modes of IPsec:

- Tunnel Mode: This will take the whole IP packet to form secure communication between two places, or gateways.

- Transport Mode: This only encapsulates the IP payload (not the entire IP packet as in tunnel mode) to ensure a secure channel of communication.

5.3 Architecture, Authentication header, Encapsulating security payloads, combining security associations, key management.

This lesson explains Encapsulating Security Payload (ESP), Encapsulating Security Payload (ESP) Header and trailer, Encapsulating Security Payload (ESP) Header and trailer fields description.

The Encapsulating Security Payload (ESP) protocol provides all the functions of Authentication Header (Authentication, Data Integrity, and anti-replay protection). The difference here is that the Encapsulating Security Payload (ESP) provides the most critical security function, data confidentiality. The benefits of using Encapsulating Security Payload (ESP) are listed below.

1) Confidentiality of Datagrams through encryption.

2) Traffic flow confidentiality using security gateways.

3) Authentication of data origin using digital certificates.

4) Anti-replay services using sequence number mechanism.

Encapsulating Security Payload (ESP) Header

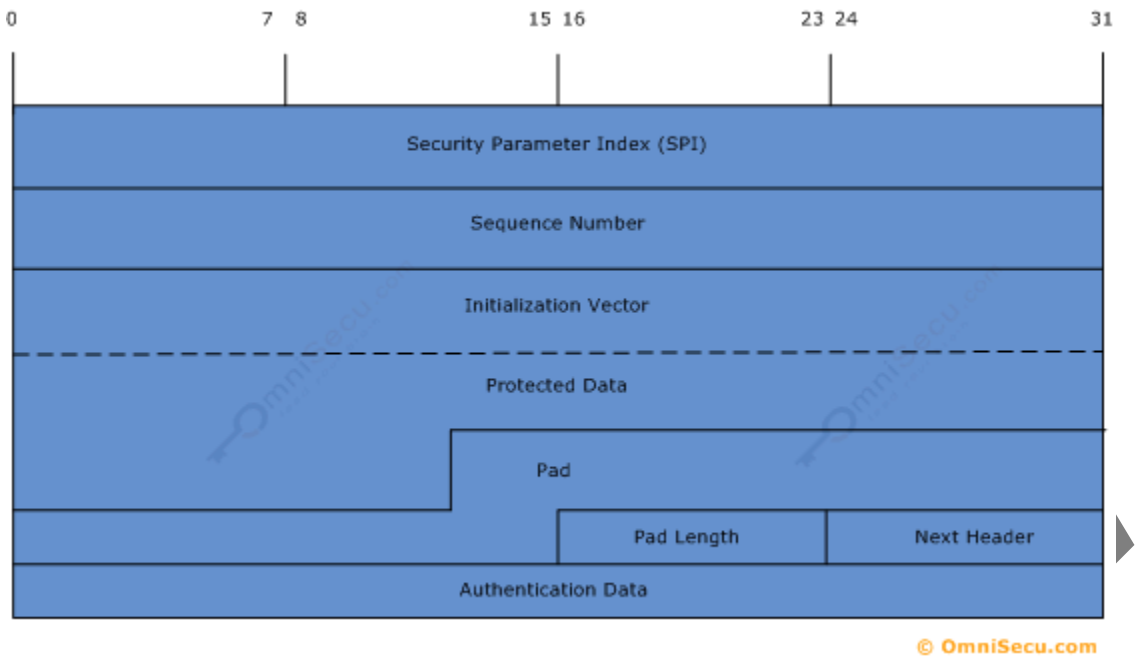

Figure 7: Encapsulating Security Payload (ESP) Header

Security Parameter Index (SPI): Security Parameter Index (SPI) field in the Encapsulating Security Payload (ESP) header along with the destination address, and the IPsec protocol are used to uniquely identify the SA that applies to this packet.

Sequence Number: The sequence number in the Encapsulating Security Payload (ESP) header provides antireplay services to Encapsulating Security Payload (ESP).

Protected Data: Protected Data field in the Encapsulating Security Payload (ESP) heade is the actual data being protected by Encapsulating Security Payload (ESP). The length of this field is variable and depends on the size of the data. The protected data field also contain any initialization vector for encryption algorithm.

Padding: Block ciphers require that plaintext be padded to a multiple of the block size. If any padding is needed, it follows immediately after the payload data in the padding field.

Pad Length: Pad Length specifies the length of the padding.

Next Header: The next header field indicates what type of data is in the payload data field.

Authentication Data: The authentication data field contains an integrity check value for the Encapsulating Security Payload (ESP) packet.

Authentication is the process which allows a sender and receiver of information to validate each other. If the sender and receiver of information cannot properly authenticate each other, there is no trust in the activities or information provided by either party. Authentication can involve highly complex and secure methods or can be very simple. The simplest form of authentication is the transmission of a shared password between entities wishing to authenticate each other. Today's authentication methods uses some of the below factors.

1) What you know

An example of this type of Authentication is a "Password". The simple logic here is that if you know the secret password for an account, then you must be the owner of that account. The problems associated with this type of Authentication is that the password can be stolen, someone might read it if you wrote it somewhere. If anyone came to know your password, he might tell someone else. If you have a simple dictionary password, it is easy to crack it by using password cracking software.

2) What you have

Examples of this type of Authentication are smart cards, tokens etc. The logic here is if you have the smart card with you, you must be the owner of the account. The problems associated with this type of authentication are you might lose the smart card, it can be stolen, or someone can duplicate the smart card etc.

3) What you are

Examples of this type of authentication are your fingerprint, handprint, retina pattern, voice, keystroke pattern etc. Problems associated with this type of authentication are that there is a

chance of false positives and false negatives. Chances are there that a valid user is rejected and an invalid user is accepted. Often people are not comfortable with this type of authentication.

Access control can be a policy, a software, or a hardware device which is used to allow or deny access to a resource. Access control can be by using devices like biometric device, switches, routers, Remote Access Service (RAS), virtual private networks (VPNs), etc. Access control can also be implemented on File System level like Microsoft's New Technology File System (NTFS), GNU/Linux's ext2/ext3/ext4 etc. The following are the three main concepts of Access Control.

• Discretionary access control (DAC)

• Mandatory access control (MAC)

• Role-based access control (RBAC)

A Virtual Private Network (VPN) can be viewed as a private network which is connected through a public network. Virtual Private Networks (VPNs) are used to connect LANs together across the Internet. Using Virtual Private Network (VPN) technologies, remote users can connect to enterprise network securely over the public internet as if their computers are physically connected to the network.

Virtual Private Network (VPN) connections use either Point-to-Point Tunneling Protocol (PPTP) or Layer Two Tunneling Protocol/Internet Protocol security (L2TP/IPSec) over internet. Internet connections are usually cheaper than leased line, Dial-up, ISDN or similar type of connections. Since Internet is the connection medium, Virtual Private Network (VPN) can save huge telecom costs.

Point-to-Point Tunneling Protocol (PPTP)

PPTP was created by Microsoft and available since Windows NT 4.0 Routing and Remote Access Services. Point-to-Point Tunneling Protocol (PPTP) encrypts the data it encapsulates, but

the header is not encrypted. Since the VPN header is not encrypted, an eavesdropper can read the VPN header but the data is somewhat secure since the contents are encrypted.

Layer 2 Tunneling Protocol (L2TP)

Layer 2 Tunneling Protocol (L2TP) is another VPN tunneling protocol which is used together with Internet Protocol Security (IPSec). IPSec encrypts the entire L2TP packet. A advantage of L2TP over PPTP is that eavesdroppers cannot identify that a VPN is in use, because IPSec encrypts the L2TP header information also. Hence L2TP/IPSec protocol is much more secure than Point-to-Point Tunneling Protocol (PPTP).

IPSec policies determines which IP traffic should be secured and which IP packets should be not be secured, what type of security should be appplied to the IP packets. IPSec polices contain IPSec rules and IPSec rules contain filter lists and filter actions.

There are three default IPSec policies in Windows Server 2003. We can assign only one policy at a time.

Client (Respond Only): Client (Respond Only) IPSec policy allows the computer to attempt unsecured communications first and switch to secured communications if requested.. This policy contains the default response rule, which creates dynamic IPSec filters for inbound and outbound traffic based on the requested protocol and port traffic for the communication the system is securing.

Server (Request Security): Server (Request Security) IPSec policy configured systems can request secure IP communications whenever possible but will fall back to clear-text IP communication if non IPSec-aware computers or systems not configured using a Client (Respond Only) policy request communication.

Secure Server (Require Security): Systems configured with a Secure Server (Require Security) IPSec policy require secure communications. The filters for this policy require all communication from the given system to be secure, with the exception of the initial inbound

communication request. Non IPSec aware devices and the devices which are not configured with Client (Respond Only) IPSec policy will not be able to communicate with a device configured with Server (Request Security) IPSec policy.

The default polices can be viewed at group policy editor. If you are working in a Windows 2003 Domain Controller, select Start > Programs > Administrative Tools > Domain Controller Security Policy.

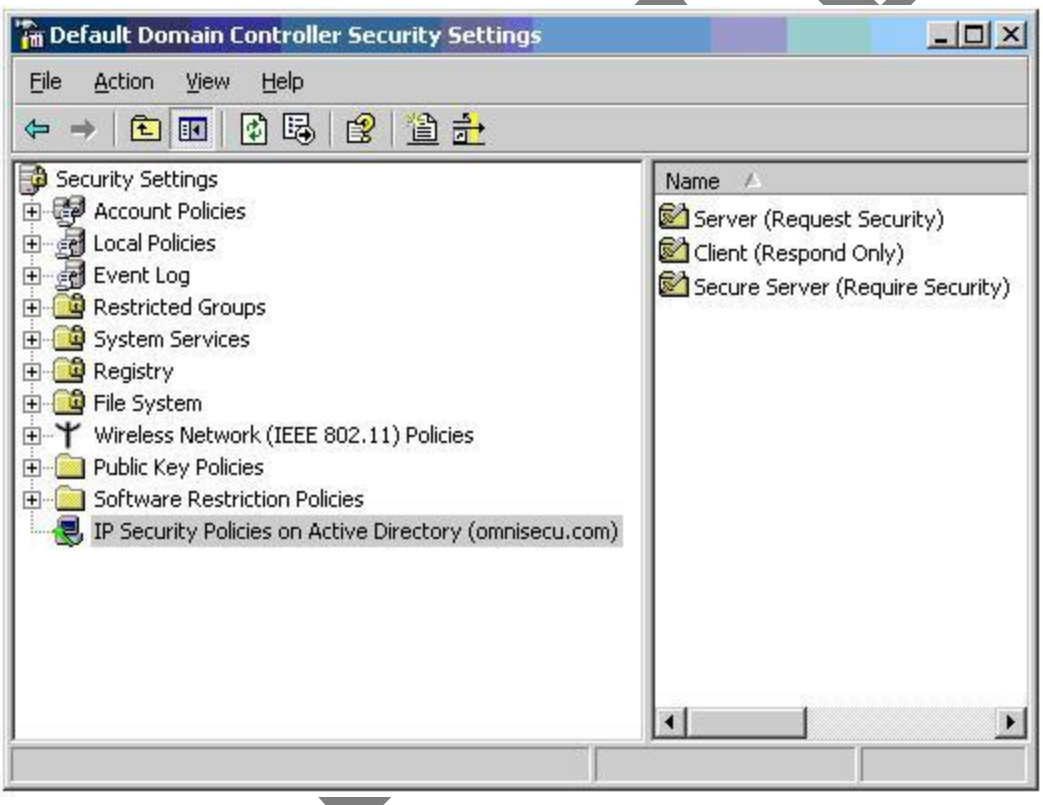

Figure 9: Domain Controller Security Policy MMC snap-in.

UNIT 5 - POSSIBLE QUESTIONS

PART A
(20 MARKS)
ONLINE EXAMINATION

PART B
(Each Question carries 2 marks)

1. Write about the network and IP security.
2. Discuss about the architecture of network security.
3. Write about the encapsulation security payloads in network security.
4. Discuss about the key management in network security.
5. How to combine security associations in network security?
6. Briefly explain the concept of viruses and related threads.
7. Explain the secure electronic transactions in detail.
8. How to explain the concept of firewall trusted systems?

PART C

Compulsory Question

(Each Question Carries 8 marks)

1. Discuss about the classification of communication protocols.
2. Compare the OSI reference model with TCP/IP model.
3. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.
4. Explain in detail about the X.25 architecture with an example network.
5. Discuss about the transmitting a message through a TCP/IP model and explain it with an example.

KARPAGAM ACADEMY OF HIGHER EDUCATION
DEPARTMENT OF COMPUTER APPLICATIONS

Degree : PG **Course: MCA** **Batch: 2017**

Subject Code/Name : 17CAP302/COMPUTER NETWORKS

S.NO	QUESTIONS	OPTION 1	OPTION 2	OPTION 3	OPTION 4	ANSWER
	UNIT 5					
1	A_____ replaces one symbol with another one.	substitution cipher	monoalphabetic cipher	traditional cipher	block cipher	**substitution cipher**
2	A_____ reorders (permutes) symbols in a block of symbols	traditional cipher	substitution cipher	transposition cipher	monoalphabetic cipher	**transposition cipher**
3	The _____ is sometimes referred to as the caesar cipher	monoalphabetic ciphet	shift cipher	traditional cipher	transposition cipher	**shift cipher**
4	A _____ is a techique that emplys the morden block ciphers such as DES and AES	modes	modes of operation	operating server	server	**modes of operation**
5	The most common public key algorithm is _____	RSA	RSSA	RSS	AES	**RSA**
6	_____means that the data can must arrive at the receiver axactly as they were sent	integrity	message integrity	authentication	message authentication	**message integrity**
7	_____ is the service beyond message integrity	message authentication	message integrity	integrity	authentication	**message authentication**
8	A digital signature needs a _____ system	private_key	primary_key	public_key	symmetric_key	**public_key**
9	A digital signature today provides _____	message authentication	integrity	message integrity	authentication	**message integrity**
10	a_____ keys between two parties is used only once	session	primary_key	private_key	public_key	**session**
11	_____ are qualitative values that represent a flow data	data traffic	data descriptor	data traffic and data descriptor	traffic data	**data descriptor**
12	The _____ define the maximum data rate of the traffic	peak data rate	maximum burst size	bandwith	effective bandwith	**peak data rate**
13	The _____ define the maximum length of time the traffic is generated in the peak rate	effective bandwith	constant rate	peak data rate	maximum burst size	**maximum burst size**
14	The _____ is the bandwith that the network needs to allocate for the flow of traffic	effective bandwith	peak data rate	maximum burst size	data descriptor	**effective bandwith**
15	A constant_bit_rate is also called as _____	fixed_rate	nonfixed rate	bit_rate	bit_constant	**fixed_rate**
16	In _____the rate of data flow changes in time, whith the change smooth instead of sudden and sharp	constant_bit_rate	variable_bit_rate	dynamic_bit_rate	bit rate	**variable_bit_rate**
17	In the _____the data rate changes suddenly in a very short times	variable_bit_rate	constant_bit_rate	bursty data	constant_bit_rate	**bursty data**
18	Congestion control is divided into _____ types	1	2	3	4	**2**
19	A _____is mechanism that can prevent before and after it happens	open_loop	closed_loop	congestion control	Flow control	**congestion control**
20	In _____ control ,policies are applied to prevent congestion before it happens	open_loop congestion	closed_loop congestion	closed_loop	congestion control	**open_loop congestion**
21	If the sender feels that a sent packets is lost ,the packet needs to _____	transmission	delete	retransmission	remove	**retransmission**
22	The type of _____ at the sender may also affect congestion	closed_loop	window	control	discarding	**window**

23	The _____ policy imposed by the receiver may also effect	acknowledgment	discarding	admission	window	**acknowledgment**
24	A good _____ policy by the routers may prevent congestion and the same time may not harm the integrity network	admission	window	discarding	acknowledgment	**discarding**
25	An_____ policy , which is a quality of service mechanism	acknowledgment	window	daiscarding	admission	**admission**
26	A _____is mechanism try to alleviate congestion after it happens	open_loop	closed_loop	congestion control	full_open	**closed_loop**
27	The technique of _____ refer to congestion control mechanism in which a congestion node stops receiving data from the immediate upstream nodes	choke packet	control	window	backpressure	**backpressure**
28	In_____ is anodeto node congestion control that start with a node and propagates	backpressure	choke packet	backtracking	window	**backpressure**
29	A_____ is apacket sent by anode to the source to inform it of congestion	control	choke packet	admission	backpressure	**choke packet**
30	In _____there is no communication between the congested nodes and source	explict signaling	left side	implicit signaling	right side	**implicit signaling**
31	Lack of reliablity means losing a _____	packet	control	data flow	admission	**packet**
32	A _____ in a file transfer or E_mail is less important	jitter	delay	reliablity	ability	**delay**
33	A _____ in the variation in delay for packets belonging to the same flow	jitter	reliablity	delay	ability	**jitter**
34	Different application need different _____	maximum burst size	effective bandwith	bandwith	peak data rate	**bandwith**
35	Packets from different flows arrive at _____	scheduling	fifo	bandwith	switch	**switch**
36	A good _____ technique treats the different flows in apair in appropriate manner	bandwith	scheduling	admission	window	**scheduling**
37	Several scheduling are designed to improve _____	quality of service	quality of data	quality of control	quality of data flow	**quality of service**
38	In _____ queuing , packets wait in a buffer(queue) until the node is ready to process them	lifo	linked	fifo	filo	**fifo**
39	In _____ queuing , packets are first assigned to a priorety class	fifo	lifo	circular	priority	**priority**
40	In priority the packets in a_____ priority queue are processed first	lowest	highest	medium	eaual	**highest**
41	A better scheduling method is_____ queuing, in this ,the packets are still assigned to different classes	weighted fair	priority	ciruclar	fair queuing	**weighted fair**
42	The_____ is amechanism to control the amount and rate of traffic sent to the network	priority	data descriptor	traffic shaping	weighted fair	**traffic shaping**
43	The _____ does not credit an idle host	token bucket	leak bucket	top bucket	empty bucket	**leak bucket**
44	A_____ algorithm shapes bursty traffic into fixed_rete traffic by averaging the data rate	leak bucket	token bucket	empty bucket	top bucket	**leak bucket**
45	The _____ bucket allows the bursty traffic at aregulated maximum rate	empty bucket	leak bucket	token bucket	top bucket	**token bucket**
46	The_____ can be combained to credit an idle host and at the same time regulate the traffic	leak bucket	top bucket	empty bucket	lowest bucket	**leak bucket**
47	_____allows us to send message include text,auido and video .	mail	internet	E_mail	internetwork	**E_mail**
48	The_____client established a connection with MTA server on the system	MTA	alice	UA	system server	**MTA**
49	The first component of an electrionic mail system is the_____	alice	server	user agent	services provider	**user agent**

50	_____is the example of user agents are mail,pine,and elm	user agent	command driven	GUI_based	E_mail	**command driven**
51	_____ define the names of aspecial files	local part	domain name	mime	header	**local part**
52	The second part of address is_____	system server	internet	domain name	local part	**domain name**
53	MIME is_____	multiple internet mail extensions	multipurpose interface mail extensions	multipurpose internet mail exchange	multipurpose internet mail extensions	**multipurpose internet mail extensions**
54	_____ has delete and keep mode	pop	pop2	pop3	pop1	**pop3**
55	_____is the mechanism provided by TCP/IP for copying a file from one host to another	FTP	MIME	UA	pop3	**FTP**
56	_____ is the default formate for transferring text files	image	ASCII	data structure	record structure	**ASCII**
57	_____ is the default formate for transferring binary files	image	data structure	record structure	ASCII	**image**
58	In the _____ formate, the file is a continuous stream of byte	file structure	record structure	data structure	image	**file structure**
59	The service provider is distrubuted over many location called _____	internet	sites	www	http	**sites**
60	The web page store at the _____	hard disk	disk	client	server	**server**
61	The _____ is the computer on which the information is located	path	sites	host	cookies	**host**
62	_____is the pathname of the file where the information is located	host	path	server	sites	**path**
63	_____ is language for creating web pages	HTML	C	C++	java	**HTML**
64	A_____ is created by a web server whenever a browser request the document	common gate way	dynamic document	script	static document	**dynamic document**
65	_____ is the protocol used mainly to access data on the world wide web	communication	network	WWW	HTTP	**HTTP**

Reg. No.....

[14CAP303]

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(For the candidates admitted from 2014 onwards)

MCA DEGREE EXAMINATION, NOVEMBER 2015

Third Semester

COMPUTER APPLICATIONS

COMPUTER NETWORKS

Time: 3 hours Maximum : 60 marks

PART – A (20 x 1 = 20 Marks) (30 Minutes)
(Question Nos. 1 to 20 Online Examinations)

PART B (5 x 8 = 40 Marks) (2 ½ Hours)
Answer ALL the Questions

21. a) Explain in detail about the OSI Reference model?

Or

b) Explain about the guided transmission media?

22. a) List out the data link layer design issues? Explain it.

Or

b) Briefly explain about the example data link protocols?

23. a) Write about the shortest path algorithm?

Or

b) Explain in detail about leaky and token buckets?

24. a) Briefly explain about Berkeley sockets?

Or

b) Explain about RPC?

25. a) Write about electronic mail?

Or

b) Explain in detail about the HTML?

Reg. No.....

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Reg. No.....

[13CAP303]

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MCA DEGREE EXAMINATION, NOVEMBER 2014

Third Semester

COMPUTER APPLICATIONS

COMPUTER NETWORKS

Time: 3 hours Maximum : 60 marks

PART – A (10 x 2 = 20 Marks)
Answer any TEN Questions

1. What are the advantages of FDDI over a basic token ring?
2. What is the purpose of the timer at the sender in systems using ARQ?
3. Give the two reasons for using layered protocols.
4. Write a short note on ISDN.
5. Why data link layer protocols position the checksum in the trailer and not in the header?
6. How the wired transmission media differ from wireless?
7. What are the functions of Network Layer?
8. List out the main responsibilities of the network layer.
9. Mention the two sub layers of data link layer.
10. What is the baud rate of a standard 10 mbps Ethernet LAN?
11. Write about distance vector routing algorithm.
12. Define 1-persistent CSMA.
13. Write a note on second generation mobile phones.
14. What are the advantages of RSA algorithm?
15. Differentiate between SMTP and SNMP.

PART B (5 X 8= 40 Marks)
Answer ALL the Questions

16. a. Explain in detail the seven layer architecture of Network?

Or

b. Discuss Public Switched Telephone Network.

17. a. Describe the various IP addressing methods.

Or

b. Explain about the Link State Routing.

18. a. Explain leaky bucket and token bucket algorithm.

Or

b. Discuss TCP.

19. a. Discuss domain name system.

Or

b. Explain about World Wide Web.

20. **Compulsory : -**

Analyze the techniques used in 4G mobile phones.

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MCA DEGREE EXAMINATION, NOVEMBER 2016
Third Semester

COMPUTER APPLICATIONS

COMPUTER NETWORKS

Time: 3 hours Maximum : 60 marks

PART – A (20 x 1 = 20 Marks) (30 Minutes)
(Question Nos. 1 to 20 Online Examinations)

PART B (5 x 6 = 30 Marks)
Answer ALL the Questions

21. a) Describe the layers of TCP/IP Reference model.

Or

b) Explain the functioning of Twisted Pairs and Coaxial Cable in Transmission media.

22. a) Discuss on working of Error-Correcting codes.

Or

b) Write short notes on Sliding Window Protocols.

23. a) Describe the Connection-oriented and Connection-less services of Network Layer.

Or

b) Explain the Shortest path and Distance Vector Routing algorithms in Network layer.

24. a) Discuss on Error Control and Flow Control mechanisms in Transport protocol.

Or

b) Describe the functioning of UDP in Transport layer.

25. a) Explain the Architecture, Services, Message formats and transfer in Electronic Mail.

Or

b) Describe the working of RSA algorithm.

PART C (1 x 10 = 10 Marks)
(Compulsory)

26. Discuss in detail on Symmetric-Key algorithms in Network security.

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[15CAP302]

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MCA DEGREE EXAMINATION, NOVEMBER 2016
Third Semester

COMPUTER APPLICATIONS

COMPUTER NETWORKS

Time: 3 hours Maximum : 60 marks

PART – A (20 x 1 = 20 Marks) (30 Minutes)
(Question Nos. 1 to 20 Online Examinations)

PART B (5 x 6 = 30 Marks)
Answer ALL the Questions

21. a) Describe the layers of TCP/IP Reference model.

Or

b) Explain the functioning of Twisted Pairs and Coaxial Cable in Transmission media.

22. a) Discuss on working of Error-Correcting codes.

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Or

b) Describe the working of RSA algorithm.

PART C (1 x 10 = 10 Marks)
(Compulsory)

26. Discuss in detail on Symmetric-Key algorithms in Network security.
