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Internal Assesment Test – III Answer Key

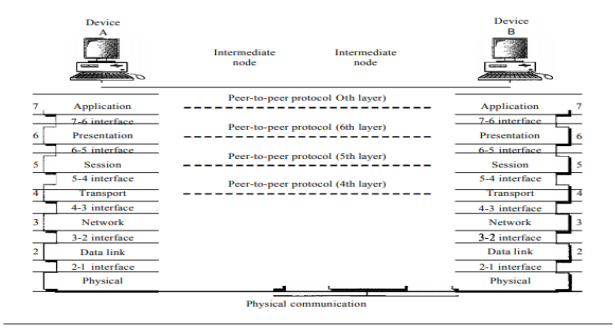
Subject : Computer Networks				Code : 17MCA31
Date : 19/11/2018	Duration : 90 mins	Max Marks : 50	Sem : III	Branch : MCA

Answer FIVE FULL Questions,choosing ONE Full Question From Each Module

		Marks	OBE CO	OBE RBT
1	Explain OSI network architecture. Explain each layer in detail.	[10]	CO1	L1, L2

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 model. It was first introduced in the late 1970s.

The OSI model is a layered framework for the design of network systems that allows communication between all types of computer systems. It consists of seven separate but related layers, each of which defines a part of the process of moving information across a network



LAYERS IN THE OSI MODEL

1) Physical Layer

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

It also defines the procedures and functions that physical devices and interfaces have to perform for transmission to Occur.

Representation of bits. The physical layer data consists of a stream of bits (sequence of Os or 1s) with no interpretation. To be transmitted, bits must be encoded into signals--electrical or optical. The physical layer defines the type of encoding (how Os and Is are changed to signals).

Data rate. The transmission rate-the number of bits sent each second-is also defined by the physical layer. In other words, the physical layer defines the duration of a bit, which is how long it lasts.

Synchronization of bits. The sender and receiver not only must use the same bit rate but also must be synchronized at the bit level. In other words, the sender and the receiver clocks must be synchronized.

Physical topology. The physical topology defines how devices are connected to make a network.

Devices can be connected by using a mesh topology (every device is connected to every other device), a star topology (devices are connected through a central device), a ring topology (each device is connected to the next, forming a ring), a bus topology (every device is on a common link), or a hybrid topology (this is a combination of two or more topologies).

Transmission mode. The physical layer also defines the direction of transmission between two devices: simplex, half-duplex, or full-duplex. In simplex mode, only one device can send; the other can only receive. The simplex mode is a one-way communication. In the half-duplex mode, two devices can send and receive, but not at the same time. In a full-duplex (or simply duplex) mode, two devices can send and receive at the same time.

2) Data Link Layer

The data link layer transforms the physical layer, a raw transmission facility, to a reliable link. It makes the physical layer appear error-free to the upper layer (network layer).

Other responsibilities of the data link layer include the following:

Framing. The data link layer divides the stream of bits received from the network layer into manageable data units called frames. The data link layer is responsible for moving frames from one hop (node) to the next.

Physical addressing. If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the sender and/or receiver of the frame. If the frame is intended for a system outside the sender's network, the receiver address is the address of the device that connects the network to the next one.

Flow control. If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver.

Error control. The data link layer adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged or lost frames. It also uses a mechanism to recognize duplicate frames. Error control is normally achieved through a trailer added to the end of the frame.

Access control. When two or more devices are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time

3)Network Layer

The network layer is responsible for the source-to-destination delivery of a packet, possibly across multiple networks (links). Whereas the data link layer oversees the delivery of the packet between two systems on the same network (links), the network layer ensures that each packet gets from its point of origin to its final destination.

If two systems are connected to the same link, there is usually no need for a network layer. However, if the two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source-to-destination delivery. Figure shows the relationship of the network layer to the data link and transport layers.

Logical addressing. The physical addressing implemented by the data link layer handles the addressing problem locally. If a packet passes the network boundary, we need another addressing system to help distinguish the source and destination systems. The network layer adds a header to the packet coming from the upper layer that, among other things, includes the logical addresses of the sender and receiver.

Routing. When independent networks or links are connected to create internetworks (network of networks) or a large network, the connecting devices (called routers or switches) route or switch

the packets to their final destination. One of the functions of the network layer is to provide this mechanism.

4)Transport Layer

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level.

Service-point addressing.

Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process (running program) on one computer to a specific process (running program) on the other. The transport layer header must therefore include a type of address called a service-point address (or port address).

The network layer gets each packet to the correct computer; the transport layer gets the entire message to the correct process on that computer.

Segmentation and reassembly.

A message is divided into transmittable segments, with each segment containing a sequence number. These numbers enable the transport layer to reassemble the message correctly upon arriving at the destination and to identify and replace packets that were lost in transmission.

Connection control.

The transport layer can be either connectionless or connection oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data are transferred, the connection is terminated.

flow control.

at this layer is performed end to end rather than across a single link.

error control

at this layer is performed process-to process rather than across a single link. The sending transport layer makes sure that the entire message arrives at the receiving transport layer without error (damage, loss, or duplication). Error correction is usually achieved through retransmission.

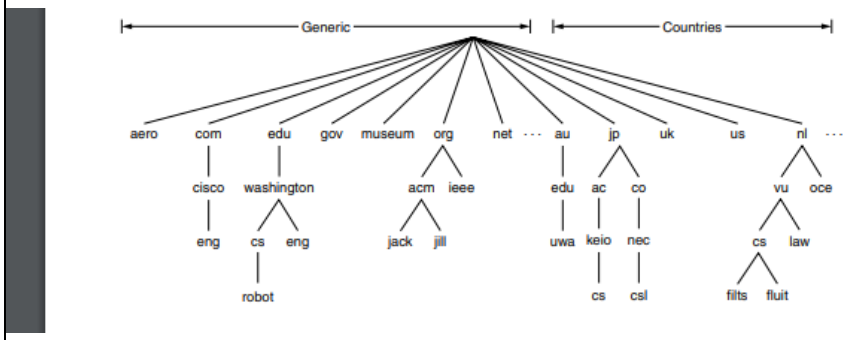
5)Session Layer

The services provided by the first three layers (physical, data link, and network) are not sufficient for some processes. The session layer is the network dialog controller. It establishes, maintains, and synchronizes the interaction among communicating systems. The session layer is responsible for dialog control and synchronization.

Specific responsibilities of the session layer include the following:

Dialog control. The session layer allows two systems to enter into a dialog. It allows the communication between two processes to take place in either half duplex (one way at a time) or

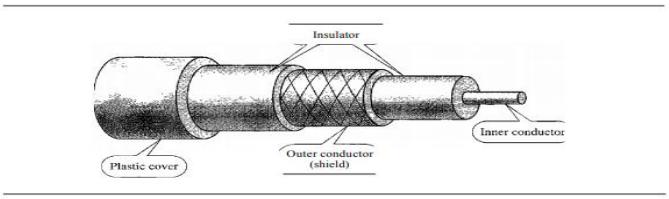
	full-duplex (two ways at a time) mode. Synchronization. The session layer allows a process to add checkpoints, or synchronization points, to a stream of data. For example, if a system is sending a file of 2000 pages, it is advisable to insert checkpoints after every 100 pages to ensure that each 100-page unit is received and acknowledged independently. In this case, if a crash happens during the transmission of page 523, the only pages that need to be resent after system recovery are pages 501 to 523. Pages previous to 501 need not be resent. 6) Presentation Layer The presentation layer is concerned with the syntax and semantics of the information exchanged between two systems. Translation. The processes (running programs) in two systems are usually exchanging information in the form of character strings, numbers, and so on. The information must be changed to bit streams before being transmitted. Because different computers use different encoding systems, the presentation layer is responsible for interoperability between these different encoding methods. Encryption. To carry sensitive information, a system must be able to ensure privacy. Encryption means that the sender transforms the original information to another form and sends the resulting message out over the network. Decryption reverses the original process to transform the message back to its original form. Compression. Data compression reduces the number of bits contained in the information. Data compression becomes particularly important in the transmission of multimedia such as text, audio, and video. 7). Application Layer The application layer enables the user, whether human or software, to access the network. It provides user interfaces and support for services such as electronic mail, remote file access and transfer, shared database management, and other types of distributed information services. In Figure shows the relationship of the application layer to the user and the presentation layer. Of the many application services available, the figure shows only three: XAOO (message-handling services), X.500 (directory services), and file transfer, access, and management (FTAM). The user in this example employs XAOO to send an e-mail message. Specific services provided by the application layer include the following: Network virtual terminal. A network virtual terminal is a software version of a physical terminal, and it allows a user to log on to a remote host. To do so, the application creates a software emulation of a terminal at the remote host. The user's computer talks to the software terminal which, in turn, talks to the host, and vice versa. The remote host believes it is communicating with one of its own terminals and allows the user to log on. File transfer, access, and management. This application allows a user to access files in a remote host (to make changes or read data), to retrieve files from a remote computer for use in the local computer, and to manage or control files in a remote computer locally. Mail services. This application provides the basis for e-mail forwarding and storage. Directory services. This application provides distributed database sources and access for global information about various objects and services.			
2(a)	Define the terms:(i) Switch (ii) Router (iii) Hub.	[5]	CO2	L2

2(b)	Write a short note on DNS. DNS (Domain Name System) was invented in 1983. It has been a key part of the Internet ever since. The essence of DNS is the invention of a hierarchical, domain-based naming scheme and a distributed database system for implementing this naming scheme. It is primarily used for mapping host names to IP addresses but can also be used for other purposes. DNS is defined in RFCs 1034, 1035, 2181, and further elaborated in many others. Very briefly, the way DNS is used is as follows. To map a name onto an IP address, an application program calls a library procedure called the resolver, passing it the name as a parameter. We saw an example of a resolver, gethostbyname, in Fig. 6-6. The resolver sends a query containing the name to a local DNS server, which looks up the name and returns a response containing the IP address to the resolver, which then returns it to the caller. The query and response messages are sent as UDP packets. Armed with the IP address, the program can then establish a TCP connection with the host or send it UDP packets. The DNS Name Space For the Internet, the top of the naming hierarchy is managed by an organization called ICANN (Internet Corporation for Assigned Names and Numbers). ICANN was created for this purpose in 1998, as part of the maturing of the Internet to a worldwide, economic concern. Conceptually, the Internet is divided into over 250 top-level domains, where each domain covers many hosts. Each domain is partitioned into subdomains, and these are further partitioned, and so on. All these domains can be represented by a tree. The leaves of the tree represent domains that have no subdomains (but do contain machines, of course). A leaf domain may contain a single host, or it may represent a company and contain thousands of hosts.  The top-level domains come in two flavors: generic and countries. The generic domains, listed in Fig. 7-2, include original domains from the 1980s and domains introduced via applications to ICANN. Other generic top-level domains will be added in the future. The country domains include one entry for every country, as defined in ISO 3166. Internationalized country domain names that use non-Latin alphabets were introduced in 2010. These domains let people name hosts in Arabic, Cyrillic, Chinese, or other languages. Getting a second-level domain, such as name-of-company.com, is easy. The top-level domains are run by registrars appointed by ICANN. Getting a name merely requires going to a corresponding registrar (for com in this case) to check if the desired name is available and not somebody else's trademark. If there are no problems, the requester pays the registrar a small annual fee and gets the	[5]	CO5	L1, L2

	name.			
3	Explain the co-axial cable and optical fiber with a neat diagram and with their applications.	[10]	CO2	L2

Coaxial Cable

Coaxial cable (or coax) carries signals of higher frequency ranges than those in twistedpair cable, in part because the two media are constructed quite differently. Instead of having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover.



Coaxial Cable Standards

Coaxial cables are categorized by their radio government (RG) ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the inner conductor, the thickness and type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function.

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the Bayone-Neill-Concelman (BNC), connector. Figure three popular types of these connectors: the BNC connector, the BNC T connector, and the BNC terminator. The BNC connector is used to connect the end of the cable to a device, such as a TV set. The BNC T connector is used in Ethernet networks to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

Performance

As we did with twisted-pair cables, we can measure the performance of a coaxial cable. We notice in Figure 7.9 that the attenuation is much higher in coaxial cables than in twisted-pair cable. In other words, although coaxial cable has a much higher bandwidth, the signal weakens rapidly and requires the frequent use of repeaters.

Applications

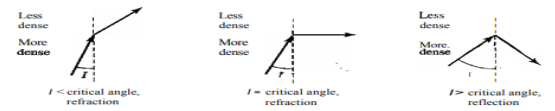
Coaxial cable was widely used in analog telephone networks where a single coaxial network could carry 10,000 voice signals. Later it was used in digital telephone networks where a single coaxial cable could carry digital data up to 600 Mbps. However, coaxial cable in telephone networks has largely been replaced today with fiber-optic cable. Cable TV networks also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Later, however, cable TV providers replaced most of the media with fiber-optic cable; hybrid networks use coaxial cable only at the network boundaries, near the consumer premises. Cable TV uses RG-59 coaxial cable.

Another common application of coaxial cable is in traditional Ethernet LANs . Because of its high bandwidth, and consequently high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The 10Base-2, or Thin Ethernet, uses RG-58 coaxial cable with BNE connectors to transmit data at 10 Mbps with a range of 185 m. The lOBase5, or Thick Ethernet, uses RG-11 (thick coaxial cable) to transmit 10 Mbps with a

range of 5000 m. Thick Ethernet has specialized connectors.

Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light. Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling through one substance suddenly enters another substance (of a different density), the ray changes direction. Figure 7.10 shows how a ray of light changes direction when going from a more dense to a less dense substance.



As the figure shows, if the angle of **incidence I** (the angle the ray makes with the line perpendicular to the interface between the two substances) is less than the critical angle, the ray refracts and moves closer to the surface.

If the angle of incidence is equal to the critical angle, the light bends along the interface. If the angle is greater than the critical angle, the ray reflects (makes a turn) and travels again in the denser substance.

Note that the critical angle is a property of the substance, and its value differs from one substance to another. Optical fibers use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it.

Propagation Modes Current technology supports two modes (multimode and single mode) for propagating light along optical channels, each requiring fiber with different physical characteristics. Multimode can be implemented in two forms: step-index or graded-index (

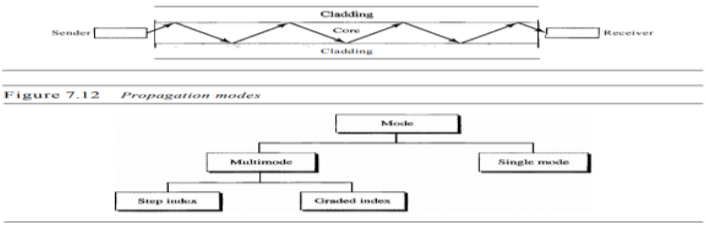


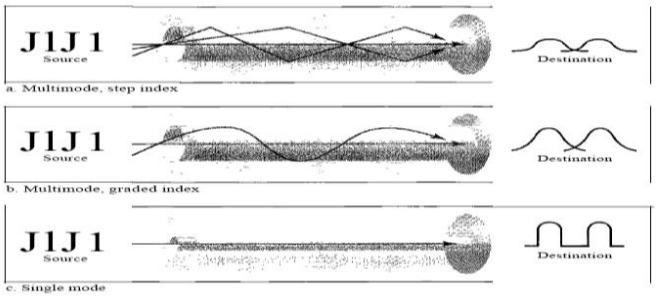
Figure 7.12 *Propagation modes*

Multimode:-

Multimode is so named because multiple beams from a light source move through the core in different paths. How these beams move within the cable depends on the structure of the core, as shown in Figure 7.13. In multimode step-index fiber, the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. At the interface, there is an abrupt change due to a lower density; this alters the angle of the beam's motion. The term step index refers to the suddenness of this change, which contributes to the distortion of the signal as it passes through the fiber.

A second type of fiber, called multimode graded-index fiber, decreases this distortion of the signal through the cable. The word index here refers to the index of refraction. As we saw above, the index of refraction is related to density. A graded-index fiber, therefore, is one with varying densities. Density is highest at the center of the core and decreases gradually to its lowest at the edge. Figure shows the impact of this variable density on the propagation of light beams.

Figure 7.13 *Modes*



Single-Mode:-

Single-mode uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal. The single mode fiber itself is manufactured with a much smaller diameter than that of multimode fiber, and with substantiallY lower density (index of refraction). The decrease in density results in a critical angle that is close enough to 90° to make the propagation of beams almost horizontal. In this case, propagation of different beams is almost identical, and delays are negligible. All the beams arrive at the destination "together" and can be recombined with little distortion to the signal.

Fiber Sizes

Optical fibers are defined by the ratio of the diameter of their core to the diameter of their cladding, both expressed in micrometers. The common sizes are shown in Table Note that the last size listed is for single-mode only.

Type	*Core (μm)*	*Cladding (μm)*	*Mode*
501125	50.0	125	Multimode, graded index
62.51125	62.5	125	Multimode, graded index
100/125	100.0	125	Multimode, graded index
7/125	7.0	125	Single mode

Cable Composition

Figure 7.14 shows the composition of a typical fiber-optic cable. The outer jacket is made of either PVC or Teflon. Inside the jacket are Kevlar strands to strengthen the cable. Kevlar is a strong material used in the fabrication of bulletproof vests. Below the Kevlar is another plastic coating to cushion the fiber. The fiber is at the center of the cable, and it consists of cladding and core.

Fiber-Optic Cable Connectors

The subscriber channel (SC) connector is used for cable TV. It uses a push/pull locking system. The straight-tip (ST) connector is used for connecting cable to networking devices. It uses a bayonet locking system and is more reliable than SC. MT-RJ is a connector that is the same size as RJ45.

Performance

The plot of attenuation versus wavelength in Figure 7.16 shows a very interesting phenomenon in fiber-optic cable. Attenuation is flatter than in the case of twisted-pair cable and coaxial cable. The performance is such that we need fewer (actually 10 times less) repeaters when we use fiber-optic cable.

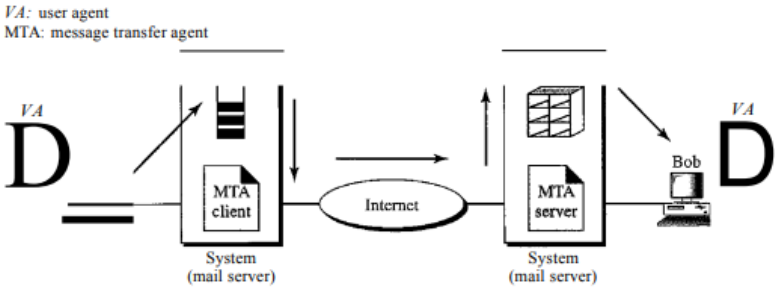
Applications

Fiber-optic cable is often found in backbone networks because its wide bandwidth is cost-effective. Today, with wavelength-division multiplexing (WDM), we can transfer data at a rate of 1600 Gbps. The SONET network that we discuss in Chapter 17 provides such a backbone. Some cable TV companies use a combination of optical fiber and coaxial cable,

	thus creating a hybrid network. Optical fiber provides the backbone structure while coaxial cable provides the connection to the user premises. This is a cost-effective configuration since the narrow bandwidth requirement at the user end does not justify the use of optical fiber. Local-area networks such as 100Base-FX network (Fast Ethernet) and 1000Base-X also use fiber-optic cable **Advantages and Disadvantages of Optical Fiber:-** Advantages Fiber-optic cable has several advantages over metallic cable (twistedpair or coaxial). Higher bandwidth. Fiber-optic cable can support dramatically higher bandwidths (and hence data rates) than either twisted-pair or coaxial cable. Currently, data rates and bandwidth utilization over fiber-optic cable are limited not by the medium but by the signal generation and reception technology available. Less signal attenuation. Fiber-optic transmission distance is significantly greater than that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted-pair cable. Immunity to electromagnetic interference. Electromagnetic noise cannot affect fiber-optic cables. Resistance to corrosive materials. Glass is more resistant to corrosive materials than copper. Light weight. Fiber-optic cables are much lighter than copper cables. Greater immunity to tapping. Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped. **Disadvantages There are some disadvantages in the use of optical fiber:-** Installation and maintenance. Fiber-optic cable is a relatively new technology. Its installation and maintenance require expertise that is not yet available everywhere. Unidirectional light propagation. Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed. Cost. The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.			
4	Explain the architecture of the email system. Architecture To explain the architecture of e-mail, we give four scenarios. We begin with the simplest situation and add complexity as we proceed. The fourth scenario is the most common in the exchange of email. First Scenario In the first scenario, the sender and the receiver of the e-mail are users (or application programs) on the same system; they are directly connected to a shared system. The administrator has created one mailbox for each user where the received messages are stored. A mailbox is part of a local hard drive, a special file with permission restrictions. Only the owner of the mailbox has access to it. When Alice, a user, needs to send a message to Bob, another user, Alice runs a user agent (VA) program to prepare the message and store it in Bob's mailbox. The message has the sender and recipient mailbox addresses (names of files). Bob can retrieve and read the contents of his mailbox at his convenience, using a user agent. Figure 26.6 shows the concept. This is similar to the traditional memo exchange between employees in an office. There is	[10]	CO4	L2

a mailroom where each employee has a mailbox with his or her name on it.

Second Scenario In the second scenario, the sender and the receiver of the e-mail are users (or application programs) on two different systems.The message needs to be sent over the Internet. Here we need user agents (VAs) and message transfer agents (MTAs),

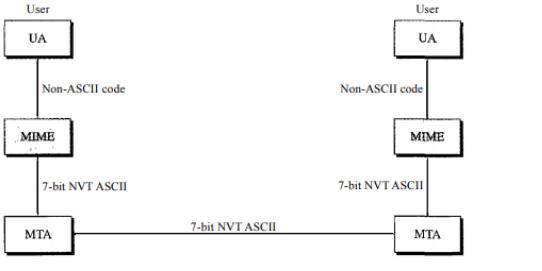


Alice needs to use a user agent program to send her message to the system at her own site. The system (sometimes called the mail server) at her site uses a queue to store messages waiting to be sent. Bob also needs a user agent program to retrieve messages stored in the mailbox of the system at his site. The message, however, needs to be sent through the Internet from Alice's site to Bob's site. Here two message transfer agents are needed: one 'client and one server. Like most client/server programs on the Internet, the server needs to run all the time because it does not know when a client will ask for a connection. The client, on the other hand, can be alerted by the system when there is a message in the queue to be sent.

Third Scenario In the third scenario, Bob, as in the second scenario, is directly connected to his system. Alice, however, is separated from her system. Either Alice is connected to the system via a point-to-point WAN, such as a dial-up modem, a DSL, or a cable modem; or she is connected to a LAN in an organization that uses one mail server for handling e-mails-all users need to send their messages to this mail server.

MIME Electronic mail has a simple structure. Its simplicity, however, comes at a price. It can send messages only in NVT 7-bit ASCII format. In other words, it has some limitations. For example, it cannot be used for languages that are not supported by 7-bit ASCII characters (such as French, German, Hebrew, Russian, Chinese, and Japanese). Also, it cannot be used to send binary files or video or audio data. Multipurpose Internet Mail Extensions (MIME) is a supplementary protocol that allows non-ASCII data to be sent through e-mail. MIME transforms non-ASCII data at the sender site to NVT ASCII data and delivers them to the client MTA to be sent through the Internet. The message at the receiving side is transformed back to the original data. We can think of MIME as a set of software functions that transforms non-ASCII data (stream of bits) to ASCII data and vice versa.

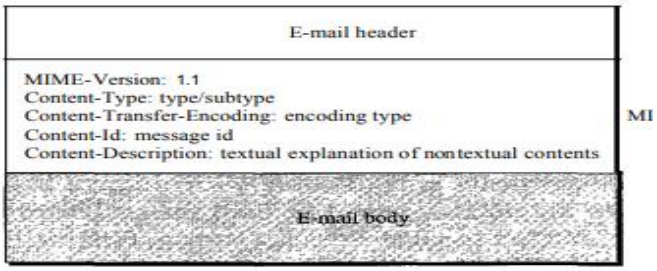
Figure 26.14 *MIME*



MIME defines five headers that can be added to the original e-mail header section to define the

transformation parameters: 1. MIME-Version 2. Content-Type 3. Content-Transfer-Encoding 4. Content-Id 5. Content-Description

MIME header

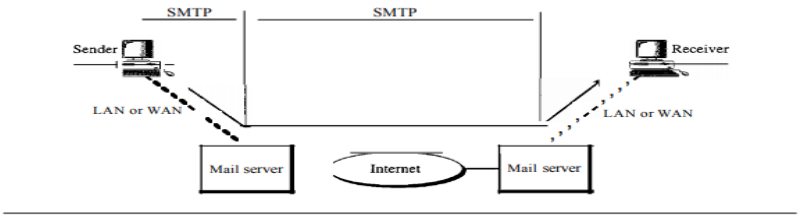


MIME-Version This header defines the version of MIME used. The current version is 1.1. Content-Type This header defines the type of data used in the body of the message. The content type and the content subtype are separated by a slash. Depending on the subtype, the header may contain other parameters.

MIME allows seven different types of data. These are listed in Table

Type	*Subtype*	*Description*
Text	Plain	Unformatted
	HTML	HTML format (see Chapter 27)
Multipart	Mixed	Body contains ordered parts of different data types
	Parallel	Same as above, but no order
	Digest	Similar to mixed subtypes, but the default is message/ RFC822
	Alternative	Parts are different versions of the same message
Message	RFC822	Body is an encapsulated message
	Partial	Body is a fragment of a bigger message
	External-Body	Body is a reference to another message
Image	IPEG	Image is in IPEG format
	GIF	Image is in GIF format
Video	MPEG	Video is in MPEG format
Audio	Basic	Single-channel encoding of voice at 8 kHz
Application	PostScript	Adobe PostScript
	Octet-stream	General binary data (8-bit bytes)

Message Transfer Agent: SMTP The actual mail transfer is done through message transfer agents. To send mail, a system must have the client MTA, and to receive mail, a system must have a server MTA. The formal protocol that defines the MTA client and server in the Internet is called the Simple Mail Transfer Protocol (SMTP). As we said before, two pairs of MTA client/server programs are used in the most common situation (fourth scenario). Figure 26.16 shows the range of the SMTP protocol in this scenario.



SMTP is used two times, between the sender and the sender's mail server and between the two mail servers. As we will see shortly, another protocol is needed between the mail server and the receiver. SMTP simply defines how commands and responses must be sent back and forth. Each network is free to choose a software package for implementation. We discuss the mechanism of

mail transfer by SMTP in the remainder of the section.

Commands Commands are sent from the client to the server. The fonnat of a command is shown in Figure 26.18. It consists of a keyword followed by zero or more arguments. SMTP defines 14 commands. The first five are mandatory; every implementation must support these five commands. The next three are often used and highly recommended. The last six are seldom used.

Table 26.7 *Commands*

Keyword	*Argument(s)*
HELO	Sender's host name
MAIL FROM	Sender of the message
RCPTTO	Intended recipient of the message
DATA	Body of the mail
QUIT	
RSET	
VRFY	Name of recipient to be verified
NOOP	
TURN	
EXPN	Mailing list to be expanded
HELP	Command name

Table 26.7 *Commands (continued)*

Keyword	*Argument(s)*
SEND FROM	Intended recipient of the message
SMOLFROM	Intended recipient of the message
SMALFROM	Intended recipient of the message

Responses Responses are sent from the server to the client. A response is a threedigit code that may be followed by additional textual information. Table 26.8 lists some of the responses.

Code	*Description*
	Positive Completion Reply
211	System status or help reply
214	Help message
220	Service ready
221	Service closing transmission channel
250	Request command completed
251	User not local; the message will be forwarded
	Positive Intermediate Reply
354	Start mail input
	Transient Negative Completion Reply
421	Service not available
450	Mailbox not available
451	Command aborted: local error
452	Command aborted: insufficient storage
	Permanent Negative Completion Reply
500	Syntax error; unrecognized command
501	Syntax error in parameters or arguments
502	Command not implemented
503	Bad sequence of commands
504	Command temporarily not implemented
550	Command is not executed; mailbox unavailable
551	User not local
552	Requested action aborted; exceeded storage location
553	Requested action not taken; mailbox name not allowed
554	Transaction failed

5 Explain Distance vector routing algorithm with an example.

Computer networks generally use dynamic routing algorithms that are more complex than flooding, but more efficient because they find shortest paths for the current topology. Two

dynamic algorithms in particular, distance vector routing and link state routing, are the most popular. In this section, we will look at the former algorithm. In the following section, we will study the latter algorithm.

A distance vector routing algorithm operates by having each router maintain a table (i.e., a vector) giving the best known distance to each destination and which link to use to get there. These tables are updated by exchanging information with the neighbors.

Eventually, every router knows the best link to reach each destination. The distance vector routing algorithm is sometimes called by other names, most commonly the distributed Bellman-Ford routing algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962).

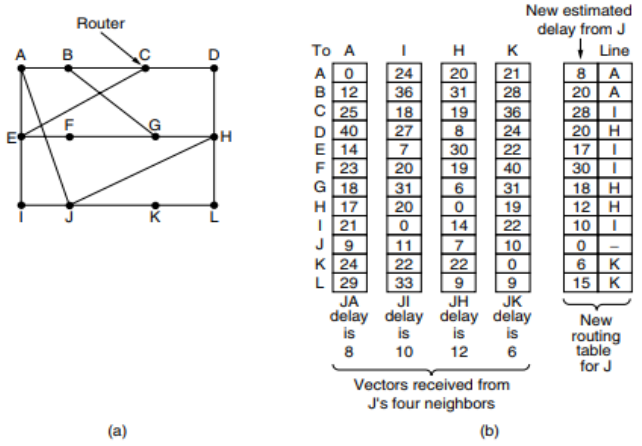
It was the original ARPANET routing algorithm and was also used in the Internet under the name RIP. In distance vector routing, each router maintains a routing table indexed by, and containing one entry for each router in the network. This entry has two parts: the preferred outgoing line to use for that destination and an estimate of the distance to that destination. The distance might be measured as the number of hops or using another metric, as we discussed for computing shortest paths.

As an example, assume that delay is used as a metric and that the router knows the delay to each of its neighbors. Once every T msec, each router sends to each neighbor a list of its estimated delays to each destination. It also receives a similar list from each neighbor.

Imagine that one of these tables has just come in from neighbor X, with Xi being X's estimate of how long it takes to get to router i. If the router knows that the delay to X is m msec, it also knows that it can reach router i via X in Xi + m msec.

By performing this calculation for each neighbor, a router can find out which estimate seems the best and use that estimate and the corresponding link in its new routing table. Note that the old routing table is not used in the calculation.

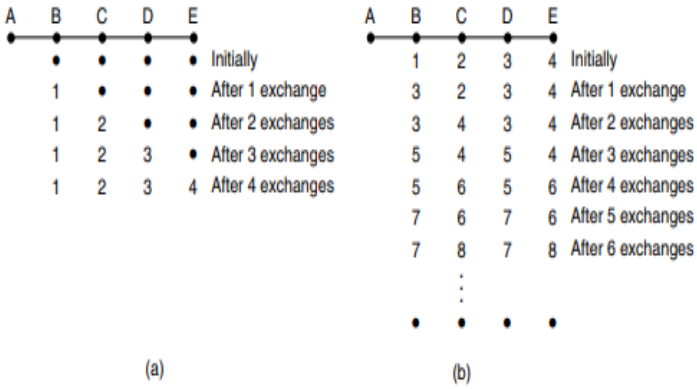
This updating process is illustrated in shows a network. The first four columns of part (b) show the delay vectors received from the neighbors of router J. A claims to have a 12-msec delay to B, a 25-msec delay to C, a 40- msec delay to D, etc. Suppose that J has measured or estimated its delay to its neighbors, A, I, H, and K, as 8, 10, 12, and 6 msec, respectively.



Consider how J computes its new route to router G. It knows that it can get to A in 8 msec, and furthermore A claims to be able to get to G in 18 msec, so J knows it can count on a delay of 26 msec to G if it forwards packets bound for G to A. Similarly, it computes the delay to G via I, H, and K as 41 (31 + 10), 18 (6 + 12), and 37 (31 + 6) msec, respectively. The best of these values is 18, so it makes an entry in its routing table that the delay to G is 18 msec and that the route to use is via H. The same calculation is performed for all the other destinations, with the new routing table shown in the last column of the figure.

The Count-to-Infinity Problem The settling of routes to best paths across the network is called convergence. Distance vector routing is useful as a simple technique by which routers can collectively compute shortest paths, but it has a serious drawback in practice: although it converges to the correct answer, it may do so slowly. In particular, it reacts rapidly to good news, but leisurely to bad news.

Consider a router whose best route to destination X is long. If, on the next exchange, neighbor A suddenly reports a short delay to X, the router just switches over to using the line to A to send traffic to X. In one vector exchange, the good news is processed. To see how fast good news propagates, consider the five-node (linear) network of Fig. 5-10, where the delay metric is the number of hops. Suppose A is down initially and all the other routers know this. In other words, they have all recorded the delay to A as infinity.



When A comes up, the other routers learn about it via the vector exchanges. For simplicity, we will assume that there is a gigantic gong somewhere that is struck periodically to initiate a vector exchange at all routers simultaneously. At the time of the first exchange, B learns that its left-hand neighbor has zero delay to A. B now makes an entry in its routing table indicating that A is one hop away to the left.

All the other routers still think that A is down. At this point, the routing table entries for A are as shown in the second row of Fig. On the next exchange, C learns that B has a path of length 1 to A, so it updates its routing table to indicate a path of length 2, but D and E do not hear the good news until later. Clearly, the good news is spreading at the rate of one hop per exchange.

In a network whose longest path is of length N hops, within N exchanges everyone will know about newly revived links and routers. Now let us consider the situation of Fig. 5-10(b), in which all the links and routers are initially up. Routers B, C, D, and E have distances to A of 1, 2, 3, and 4 hops, respectively. Suddenly, either A goes down or the link between A and B is cut (which is effectively the same thing from B's point of view).

6	Illustrate Nyquist bandwidth and Shannon capacity formula. **Channel capacity.** there are four concepts here that we are trying to relate to one another. • **Data rate:-** This is the rate, in bits per second(bps), at which data can be communicated. • **Bandwidth:-** This is the bandthwidth of the transmitted signal as constrained by the transmitter and the nature of the transmission medium, expressed in cycles per second, or Hertz. • **Noise:** This is the average level of noise over the communications path. • **Error rate:-** this is the rate at which error occur, where an error is the reception of a 1 when a 0 was transmitted or the reception of a 0 when 1 was transmitted. • **Methods to Calculate Data Rate:-** **1)Nyquist Bandwidth** Nyquist Theorem says that there is a limit to the amount of data that can be put on a line at a time. that limit is related to bandwidth(in HZ) and bits per signal change. for a noiseless channel, the	[10]	CO2	L2

Nyquist bit rate formula defines the theoretical maximum bit rate:

Bit Rate=2 * Bandwidth(B)*log2L

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data and Bit Rate is the bit rate in bits per second. According to the formula , given a specific bandwidth, one can have any bit rate by increasing the number of signal levels. although the idea is theoretically correct, practically there is a limit.

When the number of signal levels are increased, a burden is imposed on the receiver. if the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1.iF the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels. i other words, increasing the levels of a signal reduces the reliability of the system. increasing the levels of a signal may reduce the reliability of the system.

For Example:- Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. the maximum bit rate can be calculated as,

Bit RATE=2*3000*$\log_2$ 2=6000bps

For Example:- consider the same noiseless channel transmitting a signal with four signal levels(for each level, we send 2 bits). the maximum bit rate can be calculated as,

Bit Rate=2*3000*$\log_2$4=12000bps

2)Shannon Capacity Formula/Channel Capacity for Noisy Channel

in reality, one cannot have a noiseless channel because channels are always noisy. in 1944, claude shannon introduced a formula, called the shannon capa city, to determine the theoretical highest data rate for a noisy channel:

Capacity=Bandwidth(B)*log2(1+SNR)

where,

1)Bandwidth(B): bandwidth of the channel,

2) S/N(or SNR): signal-to -noise ratio and

3) Capacity: Capacity of channel in bits per second.

shannon formula there is no indication of the signal level, which means that no matter how many levels one have, we cannot achieve a data rate higher than the capacity of the channel. in other words, the formula defines a characteristic of the channel, not the method of transmission.

For **Example**:- Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. in other words, the noise is so strong that the signal is faint. for this channel the capacity C is calculated as:

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

$=B\log_2(1+0)$

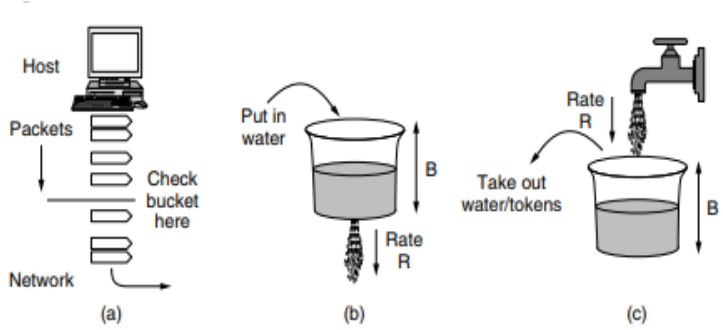
$=B\log_2 1$

$=B*0$

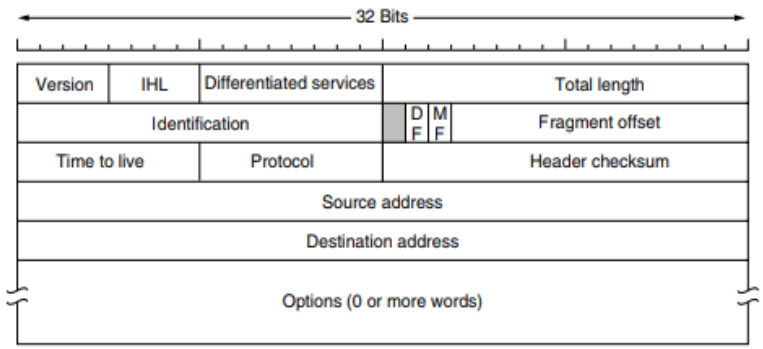
$=0$

For Example

A telephone line normally has a bandwidth of 3000Hz assigned for data communications. the signal-to-noise ratio is usually 3162. Calculate the channel capacity for this channel.

	Solution: Given bandwidth=3000Hz, SNR=3162. signal-to-noise ratio is usually 3162. for this channel the capacity is calculated as C=Blog2(1+SNR) =3000log2(1+3162) =3000*11.62 =34,860bps this means that the highest bit rate for a telephone line is 34,860kbps. if user wants to send data faster than this, then increase the bandwidth of the line or improve the signal-to-noise ratio.			
7	What is traffic shaping? Explain leaky bucket algorithm. Traffic shaping is a technique for regulating the average rate and burstiness of a flow of data that enters the network. The goal is to allow applications to transmit a wide variety of traffic that suits their needs, including some bursts, yet have a simple and useful way to describe the possible traffic patterns to the network. When a flow is set up, the user and the network (i.e., the customer and the provider) agree on a certain traffic pattern (i.e., shape) for that flow. Leaky Bucket Algorithm:- We have already seen one way to limit the amount of data an application sends: the sliding window, which uses one parameter to limit how much data is in transit at any given time, which indirectly limits the rate. Now we will look at a more general way to characterize traffic, with the leaky bucket and token bucket algorithms. The formulations are slightly different but give an equivalent result. Try to imagine a bucket with a small hole in the bottom, as illustrated in Fig. 5-28(b). No matter the rate at which water enters the bucket, the outflow is at a constant rate, R, when there is any water in the bucket and zero when the bucket is empty. Also, once the bucket is full to capacity B, any additional water entering it spills over the sides and is lost.  This bucket can be used to shape or police packets entering the network, as shown in Fig. Conceptually, each host is connected to the network by an interface containing a leaky bucket. To send a packet into the network, it must be possible to put more water into the bucket. If a packet arrives when the bucket is full, the packet must either be queued until enough water leaks out to hold it or be discarded. The former might happen at a host shaping its traffic for the network as part of the operating system. The latter might happen in hardware at a provider network interface that is policing traffic entering the network. This technique was proposed by Turner (1986) and is called the leaky bucket algorithm.			
8	Explain IPv4 header along with a neat diagram. An appropriate place to start our study of the network layer in the Internet is with the format of the IP datagrams themselves. An IPv4 datagram consists of a header part and a body or payload	[10]	CO3	L2

part. The header has a 20-byte fixed part and a variable-length optional part. The header format is shown in Fig. 5-46. The bits are transmitted from left to right and top to bottom, with the high-order bit of the Version field going first. (This is a ‘‘big-endian’’ network byte order. On littleendian machines, such as Intel x86 computers, a software conversion is required on both transmission and reception.) In retrospect, little endian would have been a better choice, but at the time IP was designed, no one knew it would come to dominate computing.



The Version field keeps track of which version of the protocol the datagram belongs to. Version 4 dominates the Internet today, and that is where we have started our discussion. By including the version at the start of each datagram, it becomes possible to have a transition between versions over a long period of time.

In fact, IPv6, the next version of IP, was defined more than a decade ago, yet is only just beginning to be deployed. We will describe it later in this section. Its use will eventually be forced when each of China’s almost 231 people has a desktop PC, a laptop, and an IP phone. As an aside on numbering, IPv5 was an experimental real-time stream protocol that was never widely used.

IHL, is provided to tell how long the header is, in 32-bit words

The maximum value of this 4-bit field is 15, which limits the header to 60 bytes, and thus the Options field to 40 bytes. For some options, such as one that records the route a packet has taken, 40 bytes is far too small, making those options useless.

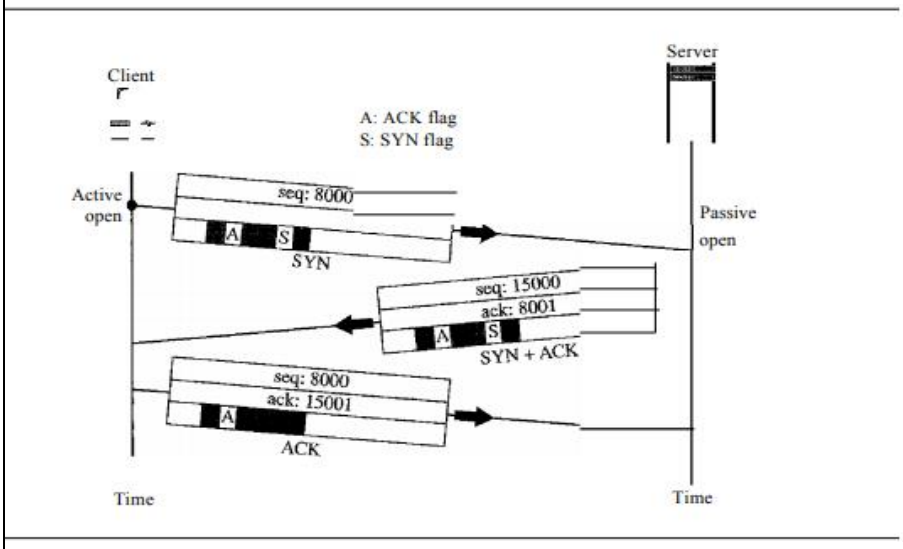
The Differentiated services field is one of the few fields that has changed its meaning (slightly) over the years. Originally, it was called the Type of service field. It was and still is intended to distinguish between different classes of service. Various combinations of reliability and speed are possible. For digitized voice, fast delivery beats accurate delivery.

For file transfer, error-free transmission is more important than fast transmission. The Type of service field provided 3 bits to signal priority and 3 bits to signal whether a host cared more about delay, throughput, or reliability. However, no one really knew what to do with these bits at routers, so they were left unused for many years. When differentiated services were designed, IETF threw in the towel and reused this field. Now, the top 6 bits are used to mark the packet with its service class;

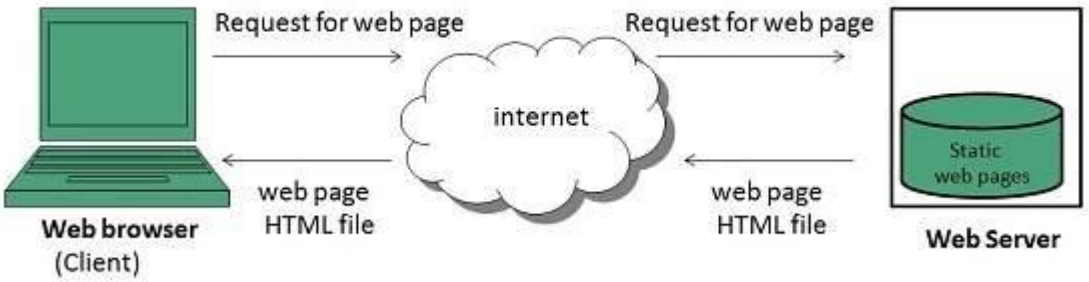
we described the expedited and assured services earlier in this chapter. The bottom 2 bits are used to carry explicit congestion notification information, such as whether the packet has experienced congestion; we described explicit congestion notification as part of congestion control earlier in this chapter. The Total length includes everything in the datagram—both header and data.

The maximum length is 65,535 bytes. At present, this upper limit is tolerable, but with future networks, larger datagrams may be needed. The Identification field is needed to allow the destination host to determine which packet a newly arrived fragment belongs to. All the fragments of a packet contain the same Identification value.

MF stands for More Fragments. All fragments except the last one have this bit set. It is needed to

	know when all fragments of a datagram have arrived. The Fragment offset tells where in the current packet this fragment belongs. The TtL (Time to live) field is a counter used to limit packet lifetimes. It was originally supposed to count time in seconds, allowing a maximum lifetime of 255 sec.			
9	Explain three way hand shake with the help of a neat diagram. Three-Way Handshaking The connection establishment in TCP is called threeway handshaking. In our example, an application program, called the client, wants to make a connection with another application program, called the server, using TCP as the transport layer protocol. The process starts with the server. The server program tells its TCP that it is ready to accept a connection. This is called a request for a passive open. Although the server TCP is ready to accept any connection from any machine in the world, it cannot make the connection itself. The client program issues a request for an active open. A client that wishes to connect to an open server tells its TCP that it needs to be connected to that particular server. TCP can now start the three-way handshaking process as shown in Figure 23.18. To show the process, we use two time lines: one at each site. Each segment has values for all its header fields and perhaps for some of its option fields, too. However, we show only the few fields necessary to understand each phase. We show the sequence number,  the acknowledgment number, the control flags (only those that are set), and the window size, if not empty. The three steps in this phase are as follows. 1. The client sends the first segment, a SYN segment, in which only the SYN flag is set. This segment is for synchronization of sequence numbers. It consumes one sequence number. When the data transfer starts, the sequence number is incremented by 1 We can say that the SYN segment carries no real data, but we can think of it as containing 1 imaginary byte. A SYN segment cannot carry data, but it consumes one sequence number. 2. The server sends the second segment, a SYN + ACK segment, with 2 flag bits set: SYN and ACK. This segment has a dual purpose. It is a SYN segment for communication in the other direction and serves as the acknowledgment for the SYN segment. It consumes one sequence number. A SYN + ACK segment cannot carry data, but does consume one sequence number.	[10]	CO4	L2

	3. The client sends the third segment. This is just an ACK segment. It acknowledges the receipt of the second segment with the ACK flag and acknowledgment number field. Note that the sequence number in this segment is the same as the one in the SYN segment; the ACK segment does not consume any sequence numbers. An ACK segment, if carrying no data, consumes no sequence number.			
10(a)	Explain WWW Architecture. **WWW** stands for **World Wide Web.** A technical definition of the World Wide Web is : all the resources and users on the Internet that are using the Hypertext Transfer Protocol (HTTP). A broader definition comes from the organization that Web inventor **Tim Berners-Lee** helped found, the **World Wide Web Consortium (W3C).** The World Wide Web is the universe of network-accessible information, an embodiment of human knowledge. In simple terms, The World Wide Web is a way of exchanging information between computers on the Internet, tying them together into a vast collection of interactive multimedia resources. *Web Page* **web page** is a document available on world wide web. Web Pages are stored on web server and can be viewed using a web browser. A web page can cotain huge information including text, graphics, audio, video and hyper links. These hyper links are the link to other web pages. Collection of linked web pages on a web server is known as **website.**There is unique **Uniform Resource Locator (URL)** is associated with each web page. Static Web page **Static web pages** are also known as flat or stationary web page. They are loaded on the client's browser as exactly they are stored on the web server. Such web pages contain only static information. User can only read the information but can't do any modification or interact with the information. Static web pages are created using only HTML. Static web pages are only used when the information is no more required to be modified.	[5]	CO5	L2



Dynamic Web page

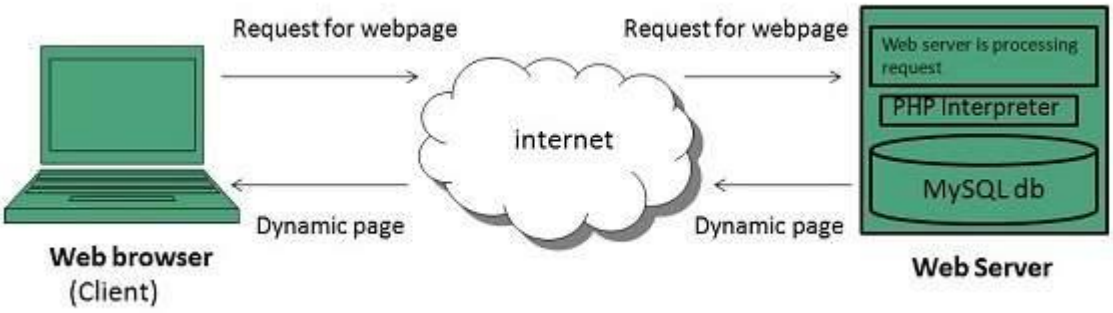
Dynamic web page shows different information at different point of time. It is possible to change a portaion of a web page without loading the entire web page. It has been made possible using **Ajax** technology.

SERVER-SIDE DYNAMIC WEB PAGE

It is created by using server-side scripting. There are server-side scripting parameters that determine how to assemble a new web page which also include setting up of more client-side processing.

CLIENT-SIDE DYNAMIC WEB PAGE

It is processed using client side scripting such as JavaScript. And then passed in to **Document Object Model (DOM).**



Web Browser

web Browser is an application software that allows us to view and explore information on the web. User can request for any web page by just entering a URL into address bar.

Web browser can show text, audio, video, animation and more. It is the responsibility of a web browser to interpret text and commands contained in the web page.

Earlier the web browsers were text-based while now a days graphical-based or voice-based web browsers are also available.

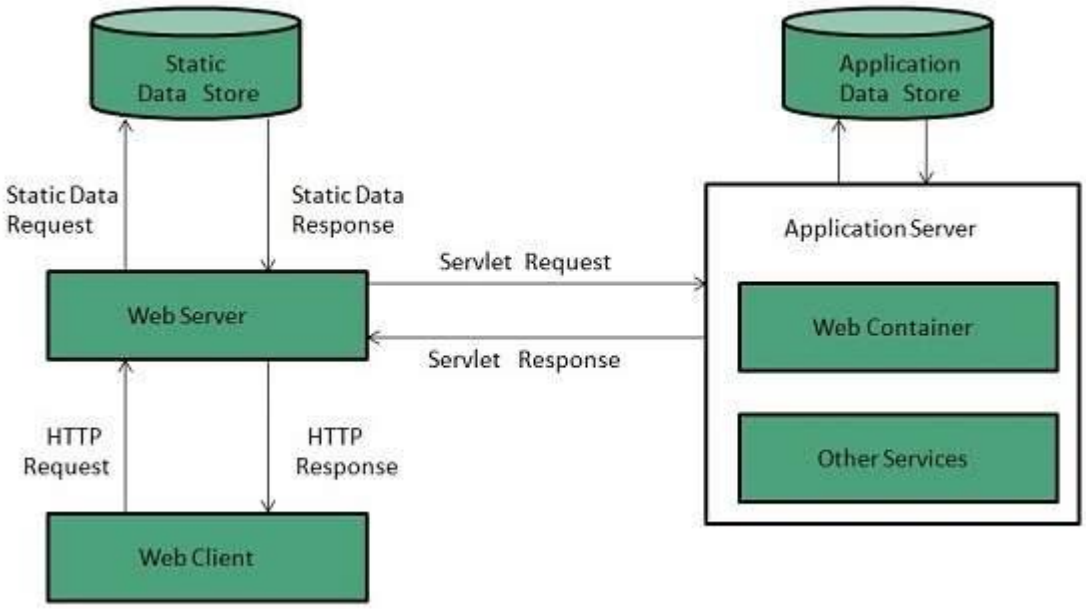
Web server is a computer where the web content is stored. Basically web server is used to host the web sites but there exists other web servers also such as gaming, storage, FTP, email etc.

Web site is collection of web pages whileweb server is a software that respond to the request for web resources.

Web Server Working

Web server respond to the client request in either of the following two ways:

- Sending the file to the client associated with the requested URL.
- Generating response by invoking a script and communicating with database



Key Points

- When client sends request for a web page, the web server search for the requested page if requested page is found then it will send it to client with an HTTP response.
- If the requested web page is not found, web server will the send an **HTTP response:Error 404 Not found.**
- If client has requested for some other resources then the web server will contact to the application server and data store to construct the HTTP response.

10(b)	Explain Real time transport protocol. The Real-Time Transport Protocol (RTP) is an Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over either unicast or multicast network services. Originally specified in Internet Engineering Task Force (IETF) Request for Comments (RFC) 1889, RTP was designed by the IETF's Audio-Video Transport Working Group to support video conferences with multiple, geographically dispersed participants. RTP is commonly used in Internet telephony applications. RTP does not in itself guarantee real-time delivery of multimedia data (since this is dependent on network characteristics); it does, however, provide the wherewithal to manage the data as it arrives to best effect.	[5]	CO4	L2

RTP combines its data transport with a control protocol (RTCP), which makes it possible to monitor data delivery for large multicast networks. Monitoring allows the receiver to detect if there is any packet loss and to compensate for any delay jitter. Both protocols work independently of the underlying Transport layer and Network layerprotocols. Information in the RTP header tells the receiver how to reconstruct the data and describes how the codec bit streams are packetized. As a rule, RTP runs on top of the User Datagram Protocol (UDP), although it can use other transport protocols. Both the Session Initiation Protocol (SIP) and H.323 use RTP.

RTP components include: a *sequence number*, which is used to detect lost packets; *payload identification*, which describes the specific media encoding so that it can be changed if it has to adapt to a variation in bandwidth; *frameindication*, which marks the beginning and end of each frame; *source identification*, which identifies the originator of the frame; and *intramedia synchronization*, which uses timestamps to detect different delay jitter within a single stream and compensate for it.

RTPC components include: *quality of service (QoS) feedback*, which includes the numbers of lost packets, round-trip time, and jitter, so that the sources can adjust their data rates accordingly; *session control*, which uses the RTCP BYE packet to allow participants to indicate that they are leaving a session; *identification*, which includes a participant's name, e-mail address, and telephone number for the information of other participants; and *intermedia synchronization*, which enables the synchronization of separately transmitted audio and video streams.