Unit 1

Syllabus
(*internet basics, internet components, layering, tcpip model, osi model, CKT SWITCHING, PACKET SWITCHING)*

Short Ans

1.What constitutes edge of internet give an example ? Name one unconventional end system?

Millions of connected hosts (or end systems) running network apps eg . laptop, desktop, smartphone etc ii. communication links - eg. twisted copper, fiber,radio, satellite etc

Some **of unconventional hosts** are ATM machine , smart watch, smart tv, card swiping machine in flight,railway and env.monitors etc.

More Info (The Internet's end systems include desktop computers (e.g., desktop PCs, Macs, and Linux boxes), servers(e.g., Web and e-mail servers), and mobile computers (e.g., laptops, smartphones, and tablets). Furthermore, an increasing number of non-traditional devices are being attached to the Internet as end systems . End systems are also referred to as hosts because they host (that is, run) application programs such as a Web browser program, a Web server program, an e-mail client program, or an e-mail server program.

(**Beginning in the late 1990s and continuing today, a wide range of interesting devices are being connected to the Internet, leveraging their ability to send and receive digital data. Given the Internet's ubiquity, its well-defined (standardized) protocols, and the availability of Internet-ready commodity hardware, it's natural to use Internet technology to network these devices together and to Internet-connected servers. Many of these devices are based in the home—video game consoles (e.g., Microsoft's Xbox), Internet-ready televisions, digital picture frames that download and display digital pictures, washing machines, refrigerators, and even a toaster that downloads meteorological information and burns an image of the day's forecast (e.g., mixed clouds and sun) on your morning toast [BBC 2001]. IP-enabled phones with GPS capabilities put location-dependent services (maps, information about nearby services or people) at your fingertips. Networked sensors embedded into the physical environment allow monitoring of buildings, bridges, seismic activity, wildlife habitats, river estuaries, and the weather.Biomedical devices can be embedded and networked in a body-area network.)**

2.**How are hosts connected to each other name two types of connections give examples?**

End systems are connected to each other used wired or wireless networks containing many devices like routers, switches, access networks etc . examples of wired connections are twisted copper pair, fiber optic cables etc. examples of wireless connections are wifi, radio connections, bluetooth, satellite connections etc

3. **Name different types of applications offered by internet**

Web, Transfer, Email like GMail, Yahoo mail, video conferencing, video streaming, live video, , gaming etc

4.**What is VOIP? Is it same as cellular phone call ?**

Real-time conversational voice over the Internet is often referred to as Internet telephony, since, from the user's perspective, it is similar to the traditional circuit switched telephone service except for the fact that here the telephonic conversation i.e relay of your audio voice or even video happens via internet . It is also commonly called Voice-over-IP (VoIP). Conversational video is similar, except that it includes the video of the participants as well as their voices. Most of today's voice and video conversational systems allow users to create conferences with three or more participants. Conversational voice and video are widely used in the Internet today, with the Internet companies Skype, whats app, QQ, and Google Talk boasting hundreds of millions of daily user

5. **Name Components of Internet and define the role played by them**.

network edge

i. millions of connected **hosts** (or **end systems**) running network apps eg . laptop, desktop, smartphone etc

ii. communication links - eg. twisted copper, fiber,radio, satellite etc

network core :

iii. roiuters or switches : finds/ searches addresses and forward packets (chunks of data/ small parts of files)

protocols eg. HTTP, TCP/IP SMTP etc.

6. **what is a packet**

When one end system has data to send to another end system, the sending end system segments the data and adds header bytes to each segment. The resulting packages of information, is known as packet

7. **. what are the two types of services offered by internet explain with post office example**

reliable and best effort eg. in post office we have speed post, registered post, ordinary post etc

8 **what is an isp and how does he provide internet connections to many people**

End systems or hosts access the Internet through Internet Service Providers (ISPs) eg . airtel, BSNL, VI, JIO etc

9. **what is information exchange why is it needed**

information exchange is transfer of data from one end system to another eg. email, web pages, videos, files etc. Internet Messages are of two types control messages, messages containing data.

10. **What is a circuit**
if one person wants to send information (voice or facsimile) to another over **telephone network.** Before the sender can send the information, the network must **establish a connection between** the sender and the **receiver so that it reserves needed resources (frequenc or bandwidth or time etc) to send message** This is a **bonafide** connection for which the switches on the path between the sender and receiver maintain connection **state for that connection**. **In the jargon of telephony, this connection is called a circuit.** When the network establishes the circuit, it also reserves a constant transmission rate in the network's links (representing a fraction of each link's ransmission capacity) for the duration of the connection. Since a given transmission rate has been **reserved for this sender-to-receiver connection**, the sender can transfer the data to the receiver at the **guaranteed constant rate.**

Long Ans

1. What do you mean by network of networks explain ?

2**. If a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec. How much time does the packet take to reach to destination**
If a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec, then the time to transmit the packet is L/R seconds**.**

3. **What is circuit switching and what is packet switching. What kind of resources are needed for a source to send a message to destination.**
There are two fundamental approaches to moving data through a network of links and switches: **circuit switching and packet switching**

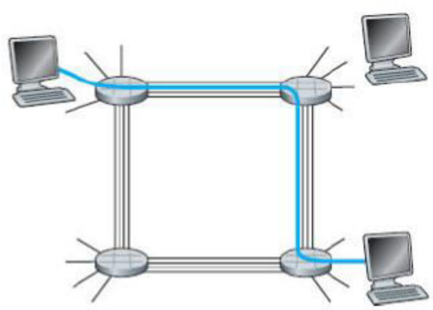
Packet Switching

In a network application, **end systems exchange messages with each other**. **Messages (control, data) can contain anything the application designer want**s. To send a message from a source end system to a destination end system, **the source breaks long messages into smaller chunks of data known as packets**. Between source and destination, each packet travels through communication links and **packet switches** (**for which there are two predominant types, routers and link_layer switches**). Packets Packets are transmitted over each communication link at a rate equal to the **full transmission rate of the link**. A network that switches packets from a source to destination randomly without reserving any resources (guarenteed transmission rate, place in output buffer) is called packet switching.

Circuit Switching

In **circuit-switched networks**, the **resources needed along a path buffers, link transmission rate** to provide for communication between the end systems are *reserved* for the duration of the communication session between the end systems.
In **packet-switched networks**, these **resources are not reserved**; a session's messages **use the resources on demand**, and as a consequence, may have to **wait** (**that is,queue**) for **access to a communication link.**
Eg. a simple analogy, consider two restaurants, a doctors appointment
Case1 : One that requires reservations .... For this restaurant we go through the hassle of

calling before we leave home......... When we arrive at the restaurant we, immediately be seated and order our meal (table is booked)
Case 2 : Here reservations are accepted..... here don't need to bother to reserve a table. But when we arrive at the restaurant, we may have to wait for a table before we can be seated Eg Traditional telephone networks are circuit-switched networks ....
if one person wants to send information (voice or facsimile) to another over **telephone network.**Before the sender can send the information, the network must **establish a connection between** the sender and the **receiver.**
This is a **bonafide** connection for which the switches on the path between the sender and receiver maintain connection **state for that connection**. **In the jargon of telephony, this connection is called a circuit.** When the network establishes the circuit, it also reserves a constant transmission rate in the network's links (representing a fraction of each link's transmission capacity) for the duration of the connection. Since a given transmission rate has been **reserved for this sender-to-receiver connection**, the sender can transfer the data to the receiver at the **guaranteed constant rate.**
In the above figure When two hosts want to communicate**, the network establishes a dedicated end-to-end connection between the two hosts.** Thus, **in order for Host A to communicate with Host B, the network must first reserve one circuit on each of two links** A circuit in a link is implemented with either **frequency-division multiplexing (FDM)** or **time-division multiplexing (TDM)**.



4. **How FDM and TDM help in ckt switching expalin with an example**
5. **Suppose users share a 2 Mbps link. Also suppose each user transmits continuously at 1 Mbps when transmitting, but each user transmits only 20 percent of the time**
a. **When circuit switching is used, how many users can be supported?**
b. For the remainder of this problem, suppose packet switching is used. Why will there be essentially no queuing delay before the link if two or fewer users transmit at the same time? Why will there be a queuing delay if three users transmit at the same time?
c. Find the probability that a given user is transmitting.
d. Suppose now there are three users. Find the probability that at any given time, all three users are transmitting simultaneously. Find the fraction of time during which the queue grows
6. **Consider the circuit-switched network in Figure 1.13. Recall that there are 4 circuits on each link. Label the four switches A, B, C and D, going in the clockwise direction.**
a. What is the maximum number of simultaneous connections that can progress at any one time in this network?
b. Suppose that all connections are between switches A and C. What is the maximum number of simultaneous connections that can be in progress?

c. Suppose we want to make four connections between switches A and C, and another four connections between switches B and D. Can we route these calls through the four links to accommodate all eight connections?

7. **What is internet ? Give Nuts and Bolts (components) and services description**

8 **What is difference between connectionless and connection oriented service?. What are disadvantages of persistent connections**

Unit 2

Syllabus

(*physical layer access networks,delay throughput, transmission rate, DSL,cABLE, TWISTED COPPER, COXIAL CABLE, FIBER OPTICS, RADIO CHANNELS , WIFI, ETHERNET, BLUETOOTH)*

Short Ans

1**. What is delay throughput and transmission rate ?**

A packet starts in a host (the source), passes through a series of routers, and ends its journey in another host (the destination). As a packet travels from one node (host or router) to the subsequent node (host or router) along this path, the packet suffers from several types of delays at each node along the path. The most important of these delays are the nodal processing delay, queuing delay, transmission delay, and propagation delay; together, these delays accumulate to give a total nodal delay. Sme of the important delays are processing delay, transmission delay, queuing delay, propagation delay etc. they together accumulate to give end to end delay. In addition to delay and packet loss,

Another critical performance measure in computer networks is **end-to-end throughput**. To define throughput, consider transferring a large file from Host A to Host B across a computer network. This transfer might be, for example, a large video clip from one peer to another in a P2P file sharing system. The instantaneous throughput at any instant of time is the rate (in bits/sec) at which Host B is receiving the file

If the file consists of F bits and the transfer takes T seconds for Host B to receive all F bits, then the average throughput of the file transfer is F/T bits/sec Figure 1.19(a) shows two end systems, a server and a client, connected by two communication links and a router. Consider the throughput for a file transfer from the server to the client. Let Rs denote the rate of the link between the server and the router; and Rc denote the rate of the link between the router and the client. If Rs < Rc, then the bits pumped by the server will "flow" right through the router and arrive at the client at a rate of Rs bps, giving a throughput of Rs bps. If, on the other hand, Rc < Rs , then the router will not be able to forward bits as quickly as it receives them. In this case, bits will only leave the router at rate Rc , giving an end-to-end throughput of Rc. Thus, for this simple two-link network, the throughput is min{Rc , Rs }. a network with N links between the server and the client, with the transmission rates of the N links being R1, R2,..., RN. Applying the same analysis as for the two-link network, we find that the throughput for a file transfer from server to client is min{R1, R2,..., RN}

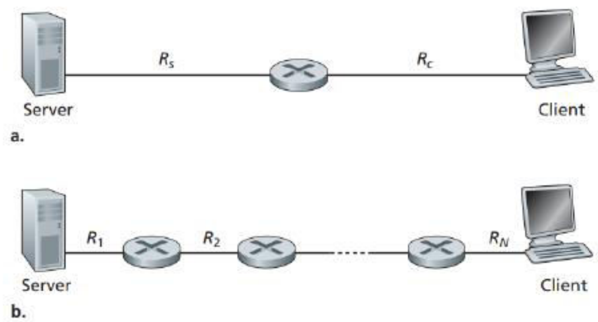


Figure 1.19 ♦ Throughput for a file transfer from server to client

Different links can transmit data at different rates, with the **transmission rate** of a link measured in bits/second.,
kbps mbps or even gbps

2. **What is store and forward delay?**

Store-and-forward transmission means that the packet switch must receive the entire packet before it can begin to transmit the first bit of the packet onto the outbound link. So, if a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec, then the time to transmit the packet is L/R seconds. Thus the router recieves entire packet only after L/R seconds Because the router employs store-and-forwarding, the router cannot transmit the bits it has received; instead it must first buffer (i.e., "store") the packet's bits. Only after the router has received all of the packet's bits can it begin to transmit (i.e., "forward") the packet onto the outbound link i.e at L/R sec it stats transmitting thus destination recieves the entire packet at 2L/R seconds the time elapsed in this storing and forwarding is called store and forward delay

3. **What is Communication Infrastructure ? Presently in India what kind of Infrastructure is internet dependent upon**

4. **Nane some wired and wireless technologies used for connecting end hosts in internet (atleast 3 each) and tabulate them to list down the transmission rates they offer**.

5.**What do u mean by assymetric transmission**

6. **What is Packet Switching**

In a network application, end systems exchange messages with each other. Mes_sages can contain anything the application designer wants. Messages may perform a control function. or can contain data, such as an email message, a JPEG image, or an MP3 audio file. To send a message from a source end system to a destination end system, the source breaks long messages into smaller chunks of data known as packets. Such a process of sending packets from one host to another is called packet switching and such a network is called packet switched network.

7. **What is a packet switch**

Packet switch is a device that switches packets from one communication link to the other . Packet switches are two predominant types, routers and link_layer switches. Most packet switches use store-and-forward transmission at the inputs to the links. A router takes a packet arriving on one of its attached communication links and forwards that packet onto another one of its attached communication links. When a packet arrives at a router in the network, the router examines a portion of the packet's destination address and for_wards the packet to an adjacent router. More specifically, each router has a forwarding table that maps destination addresses (or portions of the destination addresses) to that router's outbound links.

8 **If a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec. How much time does the packet take to reach to destination**
If a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec, then the time to transmit the packet is L/R seconds

9.**What is a protocol give examples**

10. Skype offers a service that allows you to make a phone call from a PC to an ordinary phone. This means that the voice call must pass through both the Internet and through a telephone network. Discuss how this might be done. Explain with a diagram

Long Ans

1. **Why layered approach is used in internet ? List down its advantages and disadvantages?**

Layering is all about simplifying a complex job into a simple steps by associating a layer to every step/simple service. (Eg. airline travel broken down to ticketing, baggage drop, finding the gate, travelling on flight, baggage cllection, complaining etc)

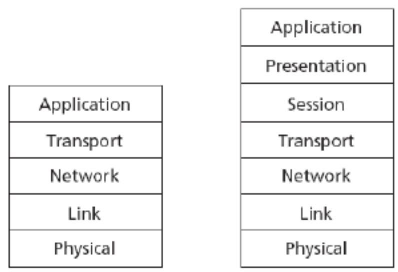
Advantages

- 1. A layered architecture simplifies task of dealing with a large complex system like Internet/Any Computer Network (Internet is a very complex system involving millions of hosts connected to each other number of varieties of communication links, routers etc performing many man functionalities like message/packet creation, route identification, error management, traffic management, congestion control, real time delivery etc)
- 2. it aids in explicit structure with every service offered is associated with different layer. (eg. reliability and congestion control is offered by transport layer, message creation by application layer, route fixation by network layer etc.
- 3. It allows us to discuss a well-defined, specific part of a large and complex system.
- 4. This simplification itself is of considerable value by providing modularity (independence)
- 5. It makies implementation any service indepently or by taking the help of other layers(eg. reliable delivery, destination identification, in analogy with airline travel the services could be baggage checking, gate identification and transfer, flying to destination airport etc)it much easier to change the implementation of the service provided by the

laye

- 5. Change any service while not disturbing other layers e.g., change in gate procedure like multiple queue at gate or even a single queue doesn't affect rest of system

2. **What are two famous layered Architectures used in Computer Networks ? Describe them?** The two famous architectures used in computer networks are the TCPIP Model which divides computer network into 5 layers i.e The Application Layer, the transport layer, the network layer, the MAC or link layer and the physical layer. The other architecture is the OSI model. It associates computer networks to seven they include e The Application Layer, Presentation, Session, , the Transport layer, the network layer, the MAC or link layer and the physical layer.



a. Five-layer Internet protocol stack b. Seven-layer ISO OSI reference model

Functions of various layers

3. **Name different types of delay and define them ?**

End-to-end route between source and destination, a packet is sent from the upstream node through router A to router B. Our goal is to characterize the nodal delay at router A. There are various types of delay taht might occur before the packet arrives at host B. they are

Processing Delay The time required to examine the packet's header and determine where to direct the packet is part of the processing delay. The processing delay can also include factors, such as the time needed to check for bit-level errors in the packet that occurred in transmitting the packet's bits

Queuing Delay At the queue, the packet experiences a queuing delay as it waits to be transmitted onto the link. The length of the queuing delay of a specific packet will depend on the number of earlier-arriving packets that are queued and waiting for transmission onto the link. If the queue is empty and no other packet is currently being transmitted, then our packet's queuing delay will be zero. On the other hand, if the traffic is heavy and many other packets are also waiting to be transmitted, the queuing delay will be long

Transmission Delay If the length of the packet by L bits, and the transmission rate of the link from router A to router B by R bits/sec. For example, for a 10 Mbps Ethernet link, the rate is R = 10 Mbps; for a 100 Mbps Ethernet link, the rate is R = 100 Mbps. The transmission delay is L/R.

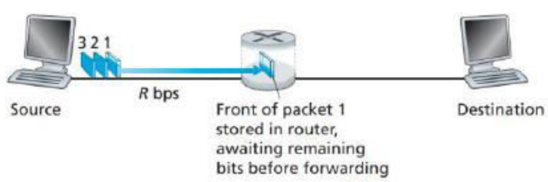
This is the amount of time required to push (that is, transmit) all of the packet's bits into the link

Propagation Delay Once a bit is pushed into the link, it needs to propagate to router B. The time required to propagate from the beginning of the link to router B is the propagation delay

If we let dproc, dqueue, dtrans, and dprop denote the processing, queuing, transmission, and propagation delays, then the **total nodal delay** is given by
dnodal = dproc + dqueue + dtrans + dprop
Suppose there are N -1 routers between the source host and the destination host. Let's also suppose for the moment that the network is uncongested (so that queuing delays are negligible), the processing delay at each router and at the source host is dproc, the transmission rate out of each router and out of the source host is R bits/sec, and the propagation on each link is dprop. The nodal delays accumulate and give an end-to-end delay, dend-end = N (dproc + dtrans + dprop) (1.2) where, once again, dtrans = L/R

4. **What is multiplexing . How many types of signals travel through DSL line explain with a diagram**

5. **Consider a simple network consisting of two end systems connected by a single router, as shown in Figure below.Suppose the source has transmitted some of packet1 and transmit 3 packets to destination how long will it take**.



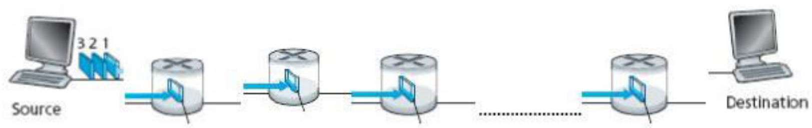
Because the router employs **store-and-forwarding**, at this instant of time, the router cannot transmit the bits it has received; instead **it must first buffer (i.e., "store") the packet's bits.**

- Only **after the router has received all of the packet's bits** can **it begin to transmit** (i.e., "forward") the packet onto the outbound link.
- The source **begins to transmit at time 0; at time L/R seconds, the source has transmitted the entire packe**t, and the entire packet has been received and stored at the router.
- **At time L/R seconds, since the router has just received the entire packet, it can begin to transmit the packet onto the outbound link** towards the destination;
- **At time 2L/R, the router has transmitted the entire packet**, and the entire packet has been received **by the destination**.
- Thus, **the total delay is 2L/R.**
- If the switch instead forwarded bits as soon as they arrive (without first receiving the entire packet), then the total delay would be L/R since bits are not held up at the router. .

For three packets this is the scenario
At time 0 ...... source **begins to transmit at time 0**
at time L/R seconds ...... the source has transmitted the entire packet , the router has recieved the entire first packet and hence the router begins to forward the first packet. But also **at time L/R** the source will begin to send the second packet at **2L/R ....the destination has received the first packet and the router has received the**

second packet, the source will begin to send the **third packet**
at 3L/R, the destination has received the first two packets and the router has received the third packet. Since the source has nothing to send it will not send any
at time 4L/R ....the destination has received all three packets

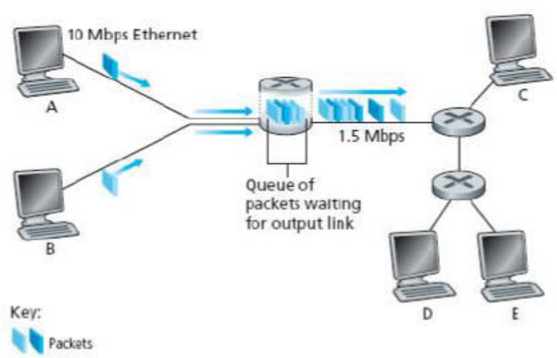
6. **Consider the general case of sending one packet from source to destination over a path consisting of N links each of rate R (thus, there are N-1 outersbetween source and destination). How long will it take.**
ii. Determine what the delay would be for P packets sent over a series of N links



(Source) Router R1 R2 R3............................RN-1 (Destination)

First Packet : at 0 it it is at Source . at L/R at R1, at 2L/R at R2 at 3L/R at R3 at N-1L/R at (N-1)L/R at NL/R it recived by Destination
Thus 1 packet takes NL/R seconds to reach to destination over N links

7. **What is queuing delay and what is packet loss. How much buffer space does routers have is it limited or unlimited**
Each packet switch has multiple links attached to it. For each attached link, the packet switch has an output buffer (also called an output queue), which stores packets that the router is about to send into that link. If an arriving packet needs to be transmitted onto a link but finds the link busy with the transmission of another packet, the arriving packet must wait in the output buffer.Thus, in addition to the store-and-forward delays, packets suffer output buffer queuing delays These delays are variable and depend on the level of congestion in the network. Since the amount of buffer space is finite, an arriving packet may find that the buffer is completely full with other packets wait_ing for transmission. In this case, packet loss will occur—either the arriving packet or one of the already-queued packets will be dropped. Suppose Hosts A and B are sending packets to Host E. Hosts A and B first send their packets along 10 Mbps Ethernet links to the first router. The router then directs these packets to the 1.5 Mbps link as shown in fig below



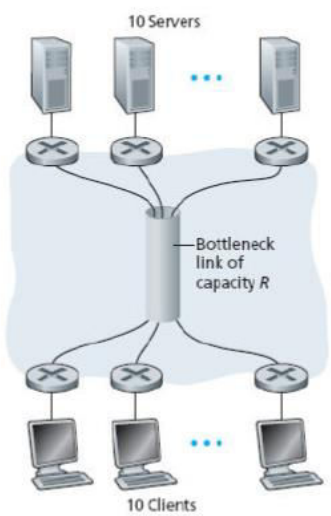
If, during a short interval of time, the arrival rate of packets to the router (when converted to bits per second) exceeds 1.5 Mbps, congestion will occur at the router as packets queue in the link's output buffer before being transmitted onto the link. For example, if Host A and B each send a burst of five packets back-to-back at the same time, then most of these packets will spend some time waiting in the queue

If **La/R > 1, (traffic intensity**) then the average rate at which bits arrive at the queue exceeds the rate at which the bits can be transmitted from the queue. In this unfortunate situation, the queue will tend to increase without bound and the queuing delay will approach infinity!

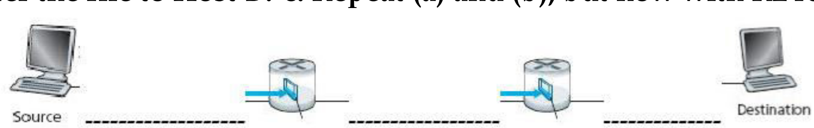
8 **Explain what will happen to packets in the above diagram . Where will they spend most of their time. What do you mean by a bottle neck in network**

If, during a short interval of time, the arrival rate of packets to the router (when converted to bits per second) exceeds 1.5 Mbps, congestion will occur at the router as packets queue in the link's output buffer before being transmitted onto the link. For example, if Host A and B each send a burst of five packets back-to-back at the same time, then most of these **packets will spend some time waiting in the queue.**

A bottleneck node is one whose arrival link or incoming link offers greater rate than transmission or outgoing link capacity . In the above diagram the arrival rate at router is (10+10) 20Mbps if the sources ae continuosly transmitting and the transmission rate via outbound link is 1.5Mbps. (its like recieving water from big pipe and tranfering via very small diameter pipe



10. **Suppose Host A wants to send a large file to Host B. The path from Host A to Host B has three links, of rates R1 = 500 kbps, R2 = 2 Mbps, and R3 = 1 Mbps. a. Assuming no other traffic in the network, what is the throughput for the file transfer? b. Suppose the file is 4 million bytes. Dividing the file size by the throughput, roughly how long will it take to transfer the file to Host B? c. Repeat (a) and (b), but now with R2 reduced to 100 kbps.**



(Source) R1 (500kbps) Switch R2(2 Mbps) R31 Mbps) Destination

(a) Throughput in the above scenario is min(R1,R2,R3)

Throughput = min(R1,R2,R3) = min(500kbps, 2Mbps, 1Mbps)
= 500kbps (*At the end of the day effective flowrate through 3 pipes will be min of the 3 pipes flow rate*)
(b) Size of the file 4 million bytes Throughput is 500kbps the time taken by 4 million bytes (4x8x106) file to be transmitted through a series of links with throughput 500 kbps = 4 x 106x8/500 x 103 = (4x8)/5 x 10 = 320/5 = 64sec
(c) Repeat at (a) and (b), but now with R2 reduced to 100 kbps.
Throughput = min(R1,R2,R3) = min(500kbps, 100kbps, 1Mbps)
= 100kbps
the time taken by 4 million bytes file to be transmitted through a series
of links with throughput 100 kbps = 4 x 106x8/100 x 103 = 320sec

Unit 3

Syllabus

LINK LAYER (SERVICE MODEL, DESIGN OPTIONS, flow control, error detection, checksum, crc. STOP AN WAIT , GOBACK N, SELECTIVE REPEAT, CHANNEL PARTITIONING, LINK LAYER PROTOCOLS, ALOHA, CSMA, CDMA

1. **What is a packet switch**

Packet switch is a device that switches packets from one communication link to the other .Packet switches are two predominant types, routers and link_layer switches. Most packet switches use store-and-forward transmission at the inputs to the links. A router takes a packet arriving on one of its attached communication links and forwards that packet onto another one of its attached communication links. When a packet arrives at a router in the network, the router examines a portion of the packet's destination address and for_wards the packet to an adjacent router. More specifically, each router has a forwarding table that maps destination addresses (or portions of the destination addresses) to that router's outbound links.

2**. What is the role played by a MAC protocol**

Amedium access control (MAC) protocol specifies the rules by which a frame is transmitted onto the link. For point-to-point links that have a single sender at one end of the link and a single receiver at the other end of the link, the MAC protocol is simple (or nonexistent)—the sender can send a frame whenever the link is idle. The more interesting case is when multiple nodes share a single broadcast link—the so-called multiple access problem

3. **In the world of ipaddresses identifying host uniquwly why do we need MAC addresses**.

Imagine a datagram will actually pass through six links: a WiFi link between sending host and WiFi access point, an Ethernet link between the access point and a link-layer switch; a link between the link-layer switch and the router, a link between the two routers; an Ethernet link between the router and a link-layer switch; and finally an Ethernet link between the switch and the server. Over a given link, a transmitting node encapsulates the datagram in a link-layer frame(which that link can read) and transmits the frame into the link.

Consider a travel agent who is planning a trip for a tourist traveling from Princeton, New Jersey, to Lausanne, Switzerland. The travel agent decides that it is most convenient for the tourist to take a limousine from Princeton to JFK airport, then a plane from JFK airport to Geneva's airport, and finally a train from Geneva's airport to Lausanne's train station. Once the travel agent makes the three reservations, it is the responsibility of the Princeton limousine company to get the tourist from Princeton to JFK; it is the responsibility of the airline company to get the tourist from JFK to Geneva; and it is the responsibility of the Swiss train service to get the tourist from Geneva to Lausanne. Each of the three segments of the trip is "direct" between two "adjacent" locations. Note that the three transportation segments are managed by different companies and use entirely different transportation modes (limousine, plane, and train). Although the transportation modes are different, they each provide the basic service of moving passengers from one location to an adjacent location. In this transportation analogy, the tourist is a datagram, each transportation segment is a link, the transportation mode is a link-layer protocol, and the travel agent is a routing protocol

4. **What kimd of links provide low bit level errors. For what kind of arw reliable link layer protocols recommended**

Wred links usually provide low bit errors. Wireless links usually proxide high bit level errors So usually reliability in link layer protocols is implemented in wireless links. because they demand great resources(imagine TCP causing lots of retransmissions demanding greater bandwidth, memory, buffer space etc)

5. **Where is link layer protocol implemented**

most part, the link layer is implemented in a network adapter, also sometimes known as a network interface card (NIC). At the heart of the network adapter is the link-layer controller, usually a single, special-purpose chip that implements many of the link-layer services (framing, link access, error detection, and so on)

On the sending side, the controller takes a datagram encapsulates the datagram in a link-layer frame and then transmits the frame into the communication link, following the link-access protocol. On the receiving side, a controller receives the entire frame, and extracts the network-layer datagram does error detection by checking error bits. while most of the link layer is implemented in hardware, part of the link layer is implemented in software that runs on the host's CPU.

software components of the link layer implement higher-level link layer functionality such as assembling link-layer addressing information and activating the controller hardware

6. **What are services offered by link layer**

Framing.

Link access.

Reliable delivery.

Error detection and correction.

7. **What is channel partitioning Name some channel partioning protocols**

8 Name different categories of MAC layer protocols

Multiple access protocol are classified as belonging to one of three categories: channel

partitioning protocols, random access protocols, and taking-turns protocols.

9. **What is the main role of ARP protocol Expalin What kind of casting is used by ARP is it same as taht used by TCP packet**

10. what is collision in the world of internet ? Why detection or avoidance is needed

Long Ans

1. **What are 4 important properties of internet based applications and classify different application categories in terms of their property needs**
We can broadly classify the possible services along four dimensions: reliable data transfer, throughput, timing, and security. Reliable Data Transfer For example, a packet can overflow a buffer in a router, or can be discarded by a host or router after having some of its bits corrupted. For many applications—such as electronic mail, file transfer, remote host access, Web document transfers, and financial applications—data loss can have devastating consequences. Thus, to support these applications, something has to be done to guarantee that the data sent by one end of the application is delivered correctly and completely to the other end of the application. If a protocol provides such a guaranteed data delivery service, it is said to provide reliable data transfer. When a transport protocol provides this service, the sending process can just pass its data into the socket and know with complete confidence that the data will arrive without errors at the receiving process. When a transport-layer protocol doesn't provide reliable data transfer, some of the data sent by the sending process may never arrive at the receiving process. This may be acceptable for loss-tolerant applications, most notably multimedia applications such as conversational audio/video
Throughput

in the context of a communication session between two processes along a network path, it is the rate at which the sending process can deliver bits to the receiving process. Because other sessions will be sharing the bandwidth along the network path, and because these other sessions will be coming and going, the available throughput can fluctuate with time. These observations lead to another natural service that a transport-layer protocol could provide, namely, guaranteed available throughput at some specified rate. With such a service, the application could request a guaranteed throughput of r bits/sec, end delay. Applications that have throughput requirements are said to be bandwidth-sensitive applications. Many current multimedia applications are bandwidth sensitive, although some multimedia applications may use adaptive coding techniques to encode digitized voice, elastic applications can make use of as much, or as little, throughput as happens to be available. Electronic mail, file transfer, and Web transfers are all elastic applications
Timing

A transport-layer protocol can also provide timing guarantees. As with throughput guarantees, timing guarantees can come in many shapes and forms. An example guarantee might be that every bit that the sender pumps into the socket arrives at the receiver's socket no more than 100 msec later. Such a service would be appealing to interactive real-time applications, such as Internet telephony, virtual environments, teleconferencing, and multiplayer games, all of which require tight timing constraints on data delivery in order to be effective. Long delays in Internet telephony, for example, tend to result in unnatural pauses in the conversation; in a multiplayer game or virtual interactive environment. a long delay between taking an action and seeing the response from the environment makes the application feel less realistic. For non-real-time applications, lower delay is always preferable to higher delay, but no tight constraint is placed on the end-to-end delays.

Security Finally, a transport protocol can provide an application with one or more security services.For example, in the sending host, a transport protocol can encrypt all data transmitted by the sending process, and in the receiving host, the transport-layer protocol can decrypt the data before delivering the data to the receiving process. Such a service would provide confidentiality between the two processes in addition to confidentiality, including data integrity and end-point authentication Neither TCP nor UDP provide any encryption—the data that the sending process passes into its socket is the same data that travels over the network to the destination process. Because privacy and other security issues have become critical for many applications, the Internet community has developed an enhancement for TCP, called Secure Sockets Layer (SSL). TCP-enhanced-with-SSL not only does everything that traditional TCP does but also provides critical process-to-process security services, including encryption, data integrity, and end-point authentication

Application	Data Loss	Throughput	Time-Sensitive
File transfer/download	No loss	Elastic	No
E-mail	No loss	Elastic	No
Web documents	No loss	Elastic (few kbps)	No
Internet telephony/ Video conferencing	Loss-tolerant	Audio: few kbps–1Mbps Video: 10 kbps–5 Mbps	Yes: 100s of msec
Streaming stored audio/video	Loss-tolerant	Same as above	Yes: few seconds
Interactive games	Loss-tolerant	Few kbps–10 kbps	Yes: 100s of msec
Instant messaging	No loss	Elastic	Yes and no

Figure 2.4 ♦ Requirements of selected network applications

2. i. **What is congestion control and flow control in traffic engineering?**
ii.Name two protocols ttaht offer such controls and Explain them in detail

2. **What are channel partitioning protocols. Name one channel partitioning protocol and explain it in detail** .

3. **Explain CSMA/CD in detail . How is it different from CSMA/CA**

4. **Name different error control techniques offered by link layer ? Explain each of them in detail?**

Checksum, CRC, parity checking,

5. **What is a link layer switch ? Explain the role played by it in detail and differentiate it with router ?**

6. List down some of the link layer devices and explain them in detail.

7. **What is PPP Expalin**

8 What is link virtualization Expalain

9. Expalin Link-Layer Addressing and ARP protocol

10. Explain in detail the following terms

i. Traffic Engineering

ii. load balancing

iii. Switch poisining

iv. Ethernet Frame

v. MAC address

Unit 4

Syllabus

NETWORK LAYER - Unicast, Multicast, Broadcast, ROUTE, ROUTING TABLE, IPADDRESSING, CLASSFUL, CLASSLESS, SUBNETTING, NAT, dykstras, distance vector

1. **What is route or path in world of internet who finds routes in internet for information exchange how does it do so?**

The sequence of communication links and packet switches traversed by a packet from the sending end system to the receiving end system is known as a route or path through the network. A packet switch takes a packet arriving on one of its incoming communication links and forwards that packet on one of its outgoing communication links. Packet switches come in many shapes and flavors, but the two most prominent types in today's Internet are routers and link-layer switches protocols in routers determine a packet's path from source to destination. In the Internet, every end system has an 32 bit address called an IP address. When a source end system wants to send a packet to a destination end system, the source includes the destination's IP address in the packet's header. As with postal addresses, this address has a hierarchical structure. When a packet arrives at a router in the network, the router examines a portion of the packet's destination address and forwards the packet to an adjacent router. More specifically, each router has a forwarding table that maps destination addresses (or portions of the destination addresses) to that router's outbound links. When a packet arrives at a router, the router examines the address and searches its forwarding table, using this destination address, to find the appropriate outbound link. The router then directs the packet to this outbound link.

2.**How are these routing tables filled.**
The Internet has a number of special **routing protocols** that are used to automatically set the forwarding tables. A routing protocol may, for example, determine the shortest path from each router to each destination and use the shortest path results to configure the forwarding tables in the routers.
(Simply visit the site www.traceroute.org, choose a source in a particular country, and trace the route from that source to your computer you will understand how routes

3. **Suppose end system A wants to send a large file to end system B. At a very high level, describe how end system A creates packets from the file. When one of these packets arrives to a packet switch, what information in the packet does the switch use to determine the link onto which the packet is forwarded? Why is packet switching in the Internet analogous to driving from one city to another and asking directions along the way?**
End system A breaks the large file into chunks called packet size depends on the nature of the link. It adds header to each chunk, thereby generating multiple packets from the file. The header in each packet includes the IP address of both source and destination (end system B). The ip address consists of the heirarchical structure (eg. 128.34.108.63) The packet switch uses full destination IP address or part of it in the header of packet to determine the outgoing link by matching it t its routing table. Asking traffic police or people surrounding while travelling from

one city to other , which road to take is analogous to a packet asking which outgoing link it should be forwarded on, given the packet's destination address. Thus **is packet switching in the Internet analogous to driving from one city to another and asking directions along the way.**

4. Explain the difference between unicast, multicast and broabcast ?

5. Which casting does ARP use and which one does TCP use ? Why

6. Define RIP, OSPF ,and BGP protocol ? Which one is an internetwork protocol

7. Where do we find network layer protocols in the end host or in routers ? Explain with a neat diagram

8 Explain the difference between routing and forwarding

9. What is NAT ? Explain

10. What is difference between IPV6 and IPV4.

Long Ans

1**. How are hosts identified in the world of internet? How does a router traces out routes ?**
In the Internet, every end system has an 32 bit address called an IP address. When a source end system wants to send a packet to a destination end system, the source includes the destination's IP address in the packet's header. As with postal addresses, this address has a hierarchical structure. The sequence of communication links and packet switches traversed by a packet from the sending end system to the receiving end system is known as a route or path through the network. A packet switch takes a packet arriving on one of its incoming communication links and forwards that packet on one of its outgoing communication links. Packet switches come in many shapes and flavors, but the two most prominent types in today's Internet are routers and link-layer switches protocols in routers determine a packet's path from source to destination. When a packet arrives at a router in the network, the router examines a portion of the packet's destination address and forwards the packet to an adjacent router. More specifically, each router has a forwarding table that maps destination addresses (or portions of the destination addresses) to that router's outbound links. When a packet arrives at a router, the router examines the address and searches its forwarding table, using this destination address, to find the appropriate outbound link. The router then directs the packet to this outbound link.
2. **What is forwarding table. How different is it from routing table. By the way how are packets forwarded to destination address by the router**
A router takes a packet arriving on one of its attached communication links and forwards that packet onto another one of its attached communication links. Packet forwarding is actually done in different ways in different types of computer networks
In the Internet, every end system **has an address called an IP address**. When a source end system wants to send a packet to a destination end system, **the source includes the destination's IP address** in the packet's header. As with postal addresses, this address has a hierarchical structure. When a packet arrives at a router in the network, the router examines a portion of the packet's destination address and for_wards the packet to an adjacent router. More specifically, each router has a **forwarding table that maps destination addresses** (**or portions of the destination addresses**) to that router's outbound links. When a packet arrives at a router,

the **router examines the address and searches its forwarding table,** using this destination address, to find the ppropriate outbound link. The router then directs the packet to this outbound link.
Fig below shows typical routing table.

Destination Address Range	Link Interface
11100000 00000000 00000000 00000000 through 11100000 00111111 11111111 11111111	0
11100000 01000000 00000000 00000000 through 11100000 01000000 11111111 11111111	1
11100000 01000001 00000000 00000000 through 11100001 01111111 11111111 11111111	2
otherwise	3

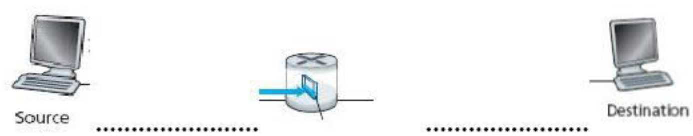
3. What is trace route utility in internet. Explain its uses with the help of online libraries.
Traceroute is a simple program that can run in any Internet host. When the user specifies **a destination hostname**, **the program in the source host sends multiple, special packets toward that destination.** As these packets work their way toward the destination, they pass through a series of routers. When a router receives one of these special packets, it sends back to the source a short message that contains the name and address of the router The source records the time that elapses between when it sends a packet and when it receives the corresponding return message; it also records the name and address of the router (or the destination host) that returns the message. In this manner, **the source can reconstruct the route** taken by packets flowing from source to destination, and the source can determine the **round-trip delays(The famous network parameter RTT ..The Round Trip Time)** intervening routers.Usually Trace_route actually repeats the experiment just described three times sending
3N packets

4. **What is packet sniffing**
Many users today access the Internet via wireless devices, such as WiFi-connected laptops or handheld

devices with cellular Internet connections. While ubiquitous Internet access is extremely convenient and enables marvelous new applications for mobile users, it also creates a major security vulnerability—by placing a passive receiver in the vicinity of the wireless transmitter, that receiver can obtain a copy of every packet that is transmitted! These packets can contain all kinds of sensitive information, including passwords, social security numbers, trade secrets, and private personal messages. A passive receiver that records a copy of every packet that flies by is called a **packet sniffer. The process of obtaining copies of packets being trnsmitted over the network is called packet sniffing.** Sniffers can be deployed in wired environments as well. In wired broadcast environments,as in many Ethernet LANs, a packet sniffer can obtain copies of broadcast packets sent over the LAN. As described in Section 1.2, cable access technologiesalso broadcast packets and are thus vulnerable to sniffing. Furthermore, a bad guy who gains access to an institution's access rou ter or access link to the
Internet may

5. **Suppose there is exactly one packet switch between a sending host and a receiving host. The transmission rates between the sending host and the switch and between the switch and the receiving host are R1 and R2, respectively. Assuming that the switch uses store-and-forward packet switching,what is the total end-to-end delay to send a packet is of length L?**

Since there is just one packet of length L to send between a sending host and a recieving host



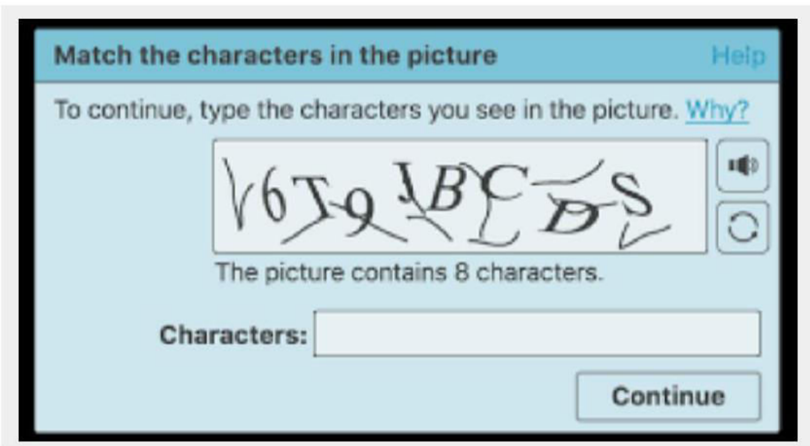
(Source) R1 Switch R2 (Destination)

At time 0 source starts tansmitting bits to switch. Time taken to transmit all the bits of packet from Source to Switch is L/R1 . Since the switch employs stoe and forward mechanism it waits until all the bits are collected before it starts transferring bits onto outbound link.At L/R1 all bits completely arrive at Switch. At time L/R1 the Switch starts transmitting bits onto outbound link of capacity R2. The time taken by the switch to transmit all bits to destination host is L/R2 . Total end to end delay from Switch to Destination is
L/R1 + L/R2

6. **What is a botnet. Why do we see captcha in websites**

We attach devices to the Internet because we want to receive/send data from/to the Internet. This includes all kinds of good stuff, including Web pages, e-mail messages, MP3s, telephone calls, live video, search engine results, and so on.But, unfortunately, along with all that good stuff comes malicious stuff—collectively known as **malware**—that can also enter and infect our devices. Once malware infects our device it can do all kinds of devious things, including deleting our files; installing spyware that collects our private information, such as social security numbers, passwords, and keystrokes, and then sends this (over the Internet, of course!) back to the bad guys. Our compromised host may also be enrolled in a network of thousands of similarly compromised devices, collectively known as a **botnet**, which the bad guys control and leverage for spam email distribution or distributed denial-of-service attacks against targeted hosts. A distributed denial-of-service attack i.e. DDoS attacks leveraging botnets with thousands of comprised hosts are a common occurrence today **Such a large-scale DDoS attack against DNS root servers actually took place on October 21, 2002. In this attack, the attackers**

leveraged a botnet to send truck loads of ICMP ping messages to each of the 13 DNS root servers With cybersecurity threats on the rise, organizations need to protect all areas of their business. This includes defending their websites and web applications from bots, spam, and abuse. In particular, web interactions such as logins, registrations, and online forms are increasingly under attack. Preventing and mitigating botnet attacks can be challenging, due to the complexity and resilience of botnets. However, there are several strategies that can be effective, including maintaining good cybersecurity hygiene, using advanced detection and response tools, and participating in collaborative efforts to dismantle botnets. one of the versatile and most used tecniques include use of a Friendly Captcha which offers a secure and invisible
alternative to traditional captchas. It is used successfully by large corporations, governments and startups worldwide.



7. **Explain the diferrence of roles played by network and transport layer using an analogy**

Transport layer lies just above the network layer in the protocol stack. While Network layer offers Logical communication between hosts whereas Transport layer: offers Logical communication between processes

relies on, enhances, network layer services

Several processes running on each host.(web, whatsapp, gmail, yahoomail , windows update, classroom , skype, ftp, google drive , railway, amazon etc

Analogy to understand diferrence of roles played by network and transport layer.

Consider two homes one in kashmir and other in kanyakumari. Each house having many siblings. Siblings of Kashmir are cousins of Kanya… One Ann in Kashmir and Bill in Kanya are responsible for mail delivery in their respective homes and also to pass it on to Postal Service. Postal service collects mail from kashmir and delivers to kanyakumai. Ann and bill are like transport layer does work only in their home and has no knowledge about how mail is transported from kashmir to kanyakumari. Postal Service is like Network layer which finds routes and delivers mail from kashmir to kanyakumari (these two homes are like hosts). Bill and Ann collect the mail and pass on to their siblings (other processes in network world). They does work only in their house. They donot step out of their house. Thus we donot find transport layer in networkcore i.e. routers and switches. howeve r we find network layer in routers and switches. We also have other cousins susan and harvey chosen instead of bill and ann . They dont do work dedicatedly like Ann and Bill. They only offer best effort. No assurance is given that mail will be delivell. They are like UDP and Bill, Ann are like TCP who offer reliable service withou the suport of network layer.

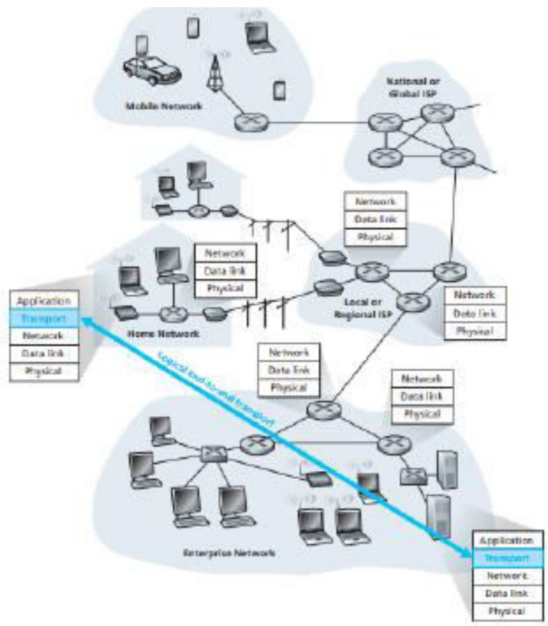


Figure 3.1 ♦ The transport layer provides logical rather than physical communication between application processes

8 **Describe two basic functions of router ? List down the three switch fabric technologies used by router and explain**

9. Explain IPV4 addressing and subnetting. Which part of the address will be subnet part and which part host part ? Explain with an example?

10. Clearly explain CLASSFUL and class less addressing ?

ii. Which protocols are use inter domain routing ? Explain

unit 5

Syllabus

Application layer - HTTP, FTP, SMTP, DNS, Transport Connection Management TCP connection based service, 3 way handshake, features of TCP, UDP connectionless service, features of UDP, telnet

1.**What is video streaming how is it different from downloading a video, give examples? Is continuous internet needed for video streaming ?**
Multimedia applications are classified into three broad categories: (i) streaming stored audio/video, (ii) conversational voice/video-over-IP, and (iii) streaming live audio/video. Many Internet companies today provide streaming video, including YouTube (Google), Netflix, and Hulu. By some estimates, streaming stored video makes up over 50 percent of the dwnstream traffic in the Internet access networks today [Cisco 2011]. Streaming stored video has three key distinguishing feature

In a streaming stored video application, the client typically begins video playout within a few seconds after it begins receiving the video from the server. This means that the client will be playing out from one location in the video while at the same time receiving later parts of the video from the server. This technique, known as streaming, avoids having to download the entire video file (and incurring a potentially long delay) before playout begins. it provides interactivity Because the media is prerecorded, the user may pause, reposition forward, reposition backward, fast-forward, and so on through the video content. The time from when the user makes such a request until the action manifests itself at the client should be less than a few seconds for acceptable responsiveness. Continuous playout. Once playout of the video begins, it should proceed according to the original timing of the recording. Therefore, data must be received from the server in time for its playout at the client; important performance measure for streaming video is average throughput For many streaming video applications, prerecorded video is stored on, and streamed from, a CDN rather than from a single data center. There we can also deliver stored video by P2P video streaming applications.Streaming is different from downloaded video because it does not require downloading complete video
only a small amount of video is downloaded at a time and discarded later. it requires a continuous internet connection for playing complete video. But a downloaded video can played offline and does not require any internet connection, because it is existent in our local harddrive. the streaming video on the other hand is existent on a server which is far away from our local computer.
2. **Name some application layer protocols ? Classify them according to architectures they use**
HTTP, FTP, SMTP all use client server architecture, VOIP or internet telephony such as Skype, Whatsapp, FTP using bittorrent, some times video streaming , internet tv use peer to peer architecture
3. **Name some famous port ids assocations like HTTP, FTP, SMTP etc. what is this port is this a software port or a hardware port**.

4.What is a socket ? What kind of transport parameters does socket programmer can control ?

5. How TCP and UDP use multiplexing and demux to transfer messages

6. How TCP ensures reliability, inorder delivery and error control strategy over extremely lossy channel. Explain the design.

7. Define the role played by SSL What kind of security features deoes TCP ensure on its own

8 Explain the connectionless services by UDP

9. Explain TCP 3 way handshake

10. TCP is a stateless protocol ? How does it maintain state explain

Long Ans

1. **What is the difference between client Server and Peer to Peer Architecture ?**
In a client-server architecture, there is an always-on host, called the server, which services requests from many other hosts, called clients. A classic example is the Web application for which an always-on Web server services requests from browsers running on client hosts. When a Web server receives a request for an object from a client host, it responds by sending the requested object to the client host. Note that with the client-server architecture, clients do not directly communicate with each other. for example, in the Web application, two browsers do not directly communicate. Another characteristic of the client-server architecture is that the server has a fixed, well-known address, called an IP address . Because the server has a fixed, well-known address, and because the server is always on, a client can always contact the server by sending a packet to the server's IP address. Some of the better-known applications with a client-server architecture include the Web, FTP, Telnet, and e-mail In a P2P architecture, there is minimal (or no) reliance on dedicated servers in data centers. Instead the application exploits direct communication between pairs of intermittently connected hosts, called peers. The peers are not owned by the service provider, but are instead desktops and laptops controlled by users, with most of the peers residing in homes, universities, and offices. Because the peers communicate without passing through a dedicated server, the architecture is called peer-to-peer . . These applications include file sharing (e.g.,BitTorrent), peer-assisted download acceleration (e.g., Xunlei), Internet Telephony (e.g., Skype), and IPTV (e.g.,Kankan and PPstream). One of the most compelling features of P2P architectures is their self-scalability

2. **Differentiate persistent and non persistent http protocols?**
When this client-server interaction is taking place over TCP, the application developer needs to make an important decision––should each request/response pair be sent over a separate TCP connection, or should all of the requests and their corresponding responses be sent over the same TCP connection? In the former approach, the application is said to use non-persistent connections; and in the latter approach, persistent connections. Connection establishment establishes one RTT , HTTP request/response eats up another RTT. Thus,roughly, the total response time is two RTTs plus the transmission time at the server of the HTML file. Thus non persistent connectios cause longer delay. When it needs to send a webpage containing 10 objects it opens 10 parallel connections enough overload. Non-persistent connections have some shortcomings. First, a brand-new connection must be established and maintained for each requested object. For each of these connections, TCP buffers must be allocated and TCP variables must be kept in both the client and server. This can place a significant burden on the Web server, With persistent connections, the server leaves the TCP connection open after sending a response.Subsequent requests and responses between the same client and server can be sent over the same connection. In particular, an entire Web page (in the example above, the base HTML file and the 10 images) can be sent over a single persistent TCP connection. Moreover, multiple Web pages residing on the same server can be sent from the server to the same client over a single persistent TCP connection. These requests for objects can be made back-to-back, without waiting for replies to pending requests (pipelining). These persistent

connections can be a potential security threat to webservers as half open TCP connection can lead to denial of service attack. But Typically, the HTTP server closes a connection when it isn’t used for a certain time (a configurable timeout interval).

3. **What is a data center ? How applications like google, youtube, railway etc manages its traffic load ?**
Often in a client-server application, a single-server host is incapable of keeping up with all the requests from clients. For example, a popular social-networking site can quickly become overwhelmed if it has only one server handling all of its requests. For this reason, a data center, housing a large number of hosts, is often used to create a powerful virtual server. The most popular Internet services—such as search engines (e.g., Google and Bing), Internet commerce (e.g., Amazon and e-Bay), Web-based email (e.g., Gmail and Yahoo Mail), social networking (e.g., Facebook and Twitter)— employ one or more data centers. Google has 30 to 50 data centers distributed around the world, which collectively handle search, YouTube, Gmail, and other services. A data center can have hundreds of thousands of servers, which must be powered and maintained. Additionally, the service providers must pay recurring interconnection and bandwidth costs for sending data from their data centers.

4. **Describe the features of HTTP protocol in detail ?**

A web page is a Collection of Objects(links, images, videos, text etc) uses Hyper text transfer protocols(HTTP) for transferring messages from client to server and server to client Is based on Client Server architecture HTTP is stateless(remembers no information about past) protocol HTTP uses TCP’s 3 way handshake to communicate with and get needed info

- Two types of HTTP are prevalent namely persistent (remain connected always) and non persistent http connections. Non persistent connections involve greater RTT (Round trip time) to get respons it uses one RTT for establishing connection, one more for request and response. It tears the connection after every transfer so long delays are observed in these HTTP connections
- Sends two types of messages request and response managed by the GET and POST methods of HTML. HTTP response message includes status code that lets you know the status of your request bad good, ok etc
- Although HTTP is a stateless protocol it maintains some state information using user based cookies which go along with every http message Neither TCP nor UDP provide any encryption—the data that the sending process passes into its socket is the same data that travels over the network to the destination process. the Internet community has developed an enhancement for TCP, called Secure Sockets Layer (SSL).
- TCP-enhanced-with-SSL not only does everything that traditional TCP does but also provides critical process-to-process security services, including encryption, data integrity, and end-point authentication

5. What is DNS ? How a servers ipaddress is searched on internet ? What kind of resorce

is DNS ?

6. There are multiple internet based processes/applicatiions running in your browser or end system like laptop ? Name some of the applications that run parallely inyour laptop which require external communication. How your system identifies the messages intended for different applications when it recieves them from external source. Explain?

7. Name some of important features of FTP protocol or web based application. What kind of transport protocol does it use and why

9. Name some of important features of SMTP protocol or web based application. What kind of transport protocol does it use and why

10 Name some mail access protocols and define their characteristics. Which category does gmail, hotmail fall into