Computer Network

R1. What is the difference between a host and an end system? List several different types of end systems. Is a Web server an end system?

- ➔ There is no difference. Throughout this text, the words “host” and “end system” are used interchangeably. End systems include PCs, workstations, Web servers, mail servers, PDAs, Internet-connected game consoles, etc.

R2. The word protocol is often used to describe diplomatic relations. How does Wikipedia describe diplomatic protocol?

- ➔ From Wikipedia: Diplomatic protocol is commonly described as a set of international courtesy rules. These well-established and time-honored rules have made it easier for nations and people to live and work together. Part of protocol has always been the acknowledgment of the hierarchical standing of all present. Protocol rules are based on the principles of civility.

R3. Why are standards important for protocols?

- ➔ Standards are important for protocols so that people can create networking systems and products that interoperate.

R4. List six access technologies. Classify each one as home access, enterprise access, or widearea wireless access

- ➔ 1. Dial-up modem over telephone line: home; 2. DSL over telephone line: home or small office; 3. Cable to HFC: home; 4. 100 Mbps switched Ethernet: enterprise; 5. Wifi (802.11): home and enterprise: 6. 3G and 4G: wide-area wireless.

R5. Is HFC transmission rate dedicated or shared among users? Are collisions possible in a downstream HFC channel? Why or why not?

- ➔ HFC bandwidth is shared among the users. On the downstream channel, all packets emanate from a single source, namely, the head end. Thus, there are no collisions in the downstream channel.

R6. List the available residential access technologies in your city. For each type of access, provide the advertised downstream rate, upstream rate, and monthly price.

- ➔ In most American cities, the current possibilities include: dial-up; DSL; cable modem; fiber-to-the-home.

R7. What is the transmission rate of Ethernet LANs?

- ➔ Ethernet LANs have transmission rates of 10 Mbps, 100 Mbps, 1 Gbps and 10 Gbps.

R8. What are some of the physical media that Ethernet can run over?

- ➔ Today, Ethernet most commonly runs over twisted-pair copper wire. It also can run over fibers optic links.

R9. Dial-up modems, HFC, DSL and FTTH are all used for residential access. For each of these access technologies, provide a range of transmission rates and comment on whether the transmission rate is shared or dedicated

- ➔ Dial up modems: up to 56 Kbps, bandwidth is dedicated; ADSL: up to 24 Mbps downstream and 2.5 Mbps upstream, bandwidth is dedicated; HFC, rates up to 42.8 Mbps and upstream rates of up to 30.7 Mbps, bandwidth is shared. FTTH: 2-10Mbps upload; 10-20 Mbps download; bandwidth is not shared.

R10. Describe the most popular wireless Internet access technologies today. Compare and contrast them

- ➔ There are two popular wireless Internet access technologies today:

  a) Wifi (802.11) In a wireless LAN, wireless users transmit/receive packets to/from an base station (i.e., wireless access point) within a radius of few tens of meters. The base station is typically connected to the wired Internet and thus serves to connect wireless users to the wired network.

  b) 3G and 4G wide-area wireless access networks. In these systems, packets are transmitted over the same wireless infrastructure used for cellular telephony, with the base station thus being managed by a telecommunications provider. This provides wireless access to users within a radius of tens of kilometers of the base station.

R11. 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 R and R , respectively. Assuming that the switch uses store-and-forward packet switching, what is the total end-to-end delay to send a packet of length L? (Ignore queuing, propagation delay, and processing delay.)

➔ At time t0 the sending host begins to transmit. At time t1 = L/R1, the sending host completes transmission and the entire packet is received at the router (no propagation delay). Because the router has the entire packet at time t1, it can begin to transmit the packet to the receiving host at time t1. At time t2 = t1 + L/R2, the router completes transmission and the entire packet is received at the receiving host (again, no propagation delay). Thus, the end-to-end delay is L/R1 + L/R2.

R12. What advantage does a circuit-switched network have over a packet-switched network? What advantages does TDM have over FDM in a circuit-switched network?

➔ A circuit-switched network can guarantee a certain amount of end-to-end bandwidth for the duration of a call. Most packet-switched networks today (including the Internet) cannot make any end-to-end guarantees for bandwidth. FDM requires sophisticated analog hardware to shift signal into appropriate frequency bands.

R13. 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. (See the discussion of statistical multiplexing in Section 1.3 .)

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

➔ a) 2 users can be supported because each user requires half of the link bandwidth.

b) Since each user requires 1Mbps when transmitting, if two or fewer users transmit simultaneously, a maximum of 2Mbps will be required. Since the available bandwidth of the shared link is 2Mbps, there will be no queuing delay before the link. Whereas, if three users transmit simultaneously, the bandwidth required will be 3Mbps which is more than the available bandwidth of the shared link. In this case, there will be queuing delay before the link.

c) Probability that a given user is transmitting = 0.2

d) Probability that all three users are transmitting simultaneously = (3/3)p^3 (1-p)^3-3 = (0.2)3 = 0.008. Since the queue grows when all the users are transmitting, the fraction of time during which the queue grows (which is equal to the probability that all three users are transmitting simultaneously) is 0.008.

R14. Why will two ISPs at the same level of the hierarchy often peer with each other? How does an IXP earn money?

➔ If the two ISPs do not peer with each other, then when they send traffic to each other they have to send the traffic through a provider ISP (intermediary), to which they have to pay for carrying the traffic. By peering with each other directly, the two ISPs can reduce their payments to their provider ISPs. An Internet Exchange Points (IXP) (typically in a standalone building with its own switches) is a meeting point where multiple ISPs can connect and/or peer together. An ISP earns its money by charging each of the the ISPs that connect to the IXP a relatively small fee, which may depend on the amount of traffic sent to or received from the IXP.

R15. Some content providers have created their own networks. Describe Google's network. What motivates content providers to create these networks?

➔ Google's private network connects together all its data centers, big and small. Traffic between the Google data centers passes over its private network rather than over the public Internet. Many of these data centers are located in, or close to, lower tier ISPs. Therefore, when Google delivers content to a user, it often can bypass higher tier ISPs. What motivates content providers to create these networks? First, the content provider has more control over the user experience, since it has to use few intermediary ISPs. Second, it can save money by sending less traffic into provider networks. Third, if ISPs decide to charge more money to highly profitable content providers (in countries where net neutrality doesn't apply), the content providers can avoid these extra payments.

R16. Consider sending a packet from a source host to a destination host over a fixed route. List the delay components in the end-to-end delay. Which of these delays are constant and which are variable?

➔ The delay components are processing delays, transmission delays, propagation delays, and queuing delays. All of these delays are fixed, except for the queuing delays, which are variable.

R17. Visit the Transmission Versus Propagation Delay applet at the companion Web site. Among the rates, propagation delay, and packet sizes available, find a combination for which the sender finishes transmitting before the first bit of the packet reaches the receiver. Find another combination for which the first bit of the packet reaches the receiver before the sender finishes transmitting

➔ a) 1000 km, 1 Mbps, 100 bytes
b) 100 km, 1 Mbps, 100 bytes

R18. How long does it take a packet of length 1,000 bytes to propagate over a link of distance 2,500 km, propagation speed m/s, and transmission rate 2 Mbps? More generally, how long does it take a packet of length L to propagate over a link of distance d, propagation speed s, and transmission rate R bps? Does this delay depend on packet length? Does this delay depend on transmission rate?

➔ 10msec; d/s; no; no

R19. 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 R reduced to 100 kbps.

➔ a) 500 kbps
b) 64 seconds
c) 100kbps; 320 seconds

R20. 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 router, what information in the packet does the router 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. It adds header to each chunk, thereby generating multiple packets from the file. The header in each packet includes the IP address of the destination (end system B). The packet switch uses the destination IP address in the packet to determine the outgoing link. Asking which road to take is analogous to a packet asking which outgoing link it should be forwarded on, given the packet's destination address.

R21. Visit the Queuing and Loss applet at the companion Web site. What is the maximum emission rate and the minimum transmission rate? With those rates, what is the traffic intensity? Run the applet with these rates and determine how long it takes for packet loss to occur. Then repeat the experiment a second time and determine again how long it takes for packet loss to occur. Are the values different? Why or why not?

- ➔ The maximum emission rate is 500 packets/sec and the maximum transmission rate is 350 packets/sec. The corresponding traffic intensity is 500/350 =1.43 > 1. Loss will eventually occur for each experiment; but the time when loss first occurs will be different from one experiment to the next due to the randomness in the emission process.

R22. List five tasks that a layer can perform. Is it possible that one (or more) of these tasks could be performed by two (or more) layers?

- ➔ Five generic tasks are error control, flow control, segmentation and reassembly, multiplexing, and connection setup. Yes, these tasks can be duplicated at different layers. For example, error control is often provided at more than one layer.

R23. What are the five layers in the Internet protocol stack? What are the principal responsibilities of each of these layers?

- ➔ The five layers in the Internet protocol stack are – from top to bottom – the application layer, the transport layer, the network layer, the link layer, and the physical layer. The principal responsibilities are outlined in Section 1.5.1

  Functinality : (1) performing certain actions within that layer (for example, at the gate layer, loading and unloading people from an airplane) and by (2) using the services of the layer directly below it (for example, in the gate layer, using the runway-to-runway passenger transfer service of the takeoff/landing layer)

R24. What is an application-layer message? A transport-layer segment? A network-layer datagram? A link-layer frame?

➔ Application-layer message: data which an application wants to send and passed onto the transport layer; transport-layer segment: generated by the transport layer and encapsulates application-layer message with transport layer header; network-layer datagram: encapsulates transport-layer segment with a network-layer header; linklayer frame: encapsulates network-layer datagram with a link-layer header.

R25. Which layers in the Internet protocol stack does a router process? Which layers does a link-layer switch process? Which layers does a host process?

➔ Routers process network, link and physical layers (layers 1 through 3). (This is a little bit of a white lie, as modern routers sometimes act as firewalls or caching components, and process Transport layer as well.) Link layer switches process link and physical layers (layers 1 through2). Hosts process all five layers.

R26. What is the difference between a virus and a worm?

➔ a) Virus

Requires some form of human interaction to spread. Classic example: E-mail viruses.

b) Worms

No user replication needed. Worm in infected host scans IP addresses and port numbers, looking for vulnerable processes to infect.

R27. Describe how a botnet can be created and how it can be used for a DDoS attack.

➔ Creation of a botnet requires an attacker to find vulnerability in some application or system (e.g. exploiting the buffer overflow vulnerability that might exist in an application). After finding the vulnerability, the attacker needs to scan for hosts that are vulnerable. The target is basically to compromise a series of systems by exploiting that particular vulnerability. Any system that is part of the botnet can automatically scan its environment and propagate by exploiting the vulnerability. An important property of such botnets is that the originator of the botnet can remotely control and issue commands to all the nodes in the botnet. Hence, it becomes possible for the attacker to issue a command to all the nodes, that target a single node (for example, all nodes in the botnet might be commanded by the attacker to send a TCP SYN message to the target, which might result in a TCP SYN flood attack at the target).

R28. Suppose Alice and Bob are sending packets to each other over a computer network. Suppose Trudy positions herself in the network so that she can capture all the packets sent by Alice and send whatever she wants to Bob; she can also capture all the packets sent by Bob and send whatever she wants to Alice. List some of the malicious things Trudy can do from this position.

- ➔ Trudy can pretend to be Bob to Alice (and vice-versa) and partially or completely modify the message(s) being sent from Bob to Alice. For example, she can easily change the phrase “Alice, I owe you $1000” to “Alice, I owe you $10,000”. Furthermore, Trudy can even drop the packets that are being sent by Bob to Alice (and vise-versa), even if the packets from Bob to Alice are encrypted.

Chapter – 2

R1. List five nonproprietary Internet applications and the application-layer protocols that they use

- ➔ The Web: HTTP; file transfer: FTP; remote login: Telnet; e-mail: SMTP; BitTorrent file sharing: BitTorrent protocol

R2. What is the difference between network architecture and application architecture?

- ➔ Network architecture refers to the organization of the communication process into layers (e.g., the five-layer Internet architecture). Application architecture, on the other hand, is designed by an application developer and dictates the broad structure of the application (e.g., client-server or P2P).

R3. For a communication session between a pair of processes, which process is the client and which is the server?

- ➔ The process which initiates the communication is the client; the process that waits to be contacted is the server.

R4. For a P2P file-sharing application, do you agree with the statement, “There is no notion of client and server sides of a communication session”? Why or why not?

- ➔ No. In a P2P file-sharing application, the peer that is receiving a file is typically the client and the peer that is sending the file is typically the server.

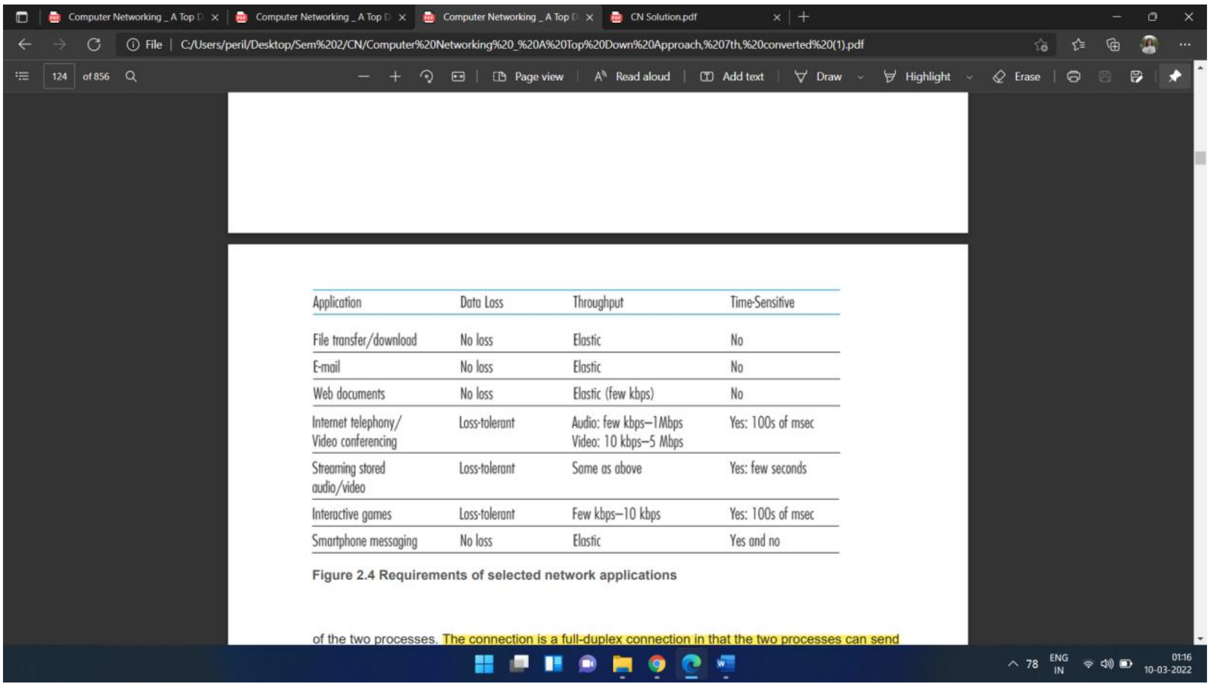
R5. What information is used by a process running on one host to identify a process running on another host?

- ➔ The IP address of the destination host and the port number of the socket in the destination process.

R6. Suppose you wanted to do a transaction from a remote client to a server as fast as possible. Would you use UDP or TCP? Why?

- ➔ You would use UDP. With UDP, the transaction can be completed in one roundtrip time (RTT) - the client sends the transaction request into a UDP socket, and the server sends the reply back to the client's UDP socket. With TCP, a minimum of two RTTs are needed - one to set-up the TCP connection, and another for the client to send the request, and for the server to send back the reply.

R7. Referring to Figure 2.4 , we see that none of the applications listed in Figure 2.4 requires both no data loss and timing. Can you conceive of an application that requires no data loss and that is also highly time-sensitive?



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124
of 856
Page view
Read aloud
Add text
Draw
Highlight
Erase
78
ENG
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10-03-2022

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
Smartphone messaging	No loss	Elastic	Yes and no

Figure 2.4 Requirements of selected network applications

of the two processes. The connection is a full-duplex connection in that the two processes can send

- ➔ One such example is remote word processing, for example, with Google docs. However, because Google docs runs over the Internet (using TCP), timing guarantees are not provided

R8. List the four broad classes of services that a transport protocol can provide. For each of the service classes, indicate if either UDP or TCP (or both) provides such a service.

Ans :

a) Reliable data transfer

TCP provides a reliable byte-stream between client and server but UDP does not.

b) A guarantee that a certain value for throughput will be maintained

Neither

c) A guarantee that data will be delivered within a specified amount of time

Neither

d) Confidentiality (via encryption)

Neither

R9. Recall that TCP can be enhanced with SSL to provide process-to-process security services, including encryption. Does SSL operate at the transport layer or the application layer? If the application developer wants TCP to be enhanced with SSL, what does the developer have to do?

- ➔ SSL operates at the application layer. The SSL socket takes unencrypted data from the application layer, encrypts it and then passes it to the TCP socket. If the application developer wants TCP to be enhanced with SSL, she has to include the SSL code in the application.

R10. What is meant by a handshaking protocol?

- ➔ A protocol uses handshaking if the two communicating entities first exchange control packets before sending data to each other. SMTP uses handshaking at the application layer whereas HTTP does not.

R11. Why do HTTP, SMTP, and POP3 run on top of TCP rather than on UDP?

- ➔ The applications associated with those protocols require that all application data be received in the correct order and without gaps. TCP provides this service whereas UDP does not.

R12. Consider an e-commerce site that wants to keep a purchase record for each of its customers. Describe how this can be done with cookies.

- ➔ When the user first visits the site, the server creates a unique identification number, creates an entry in its back-end database, and returns this identification number as a cookie number. This cookie number is stored on the user's host and is managed by the browser. During each subsequent visit (and purchase), the browser sends the cookie number back to the site. Thus the site knows when this user (more precisely, this browser) is visiting the site.

R13. Describe how Web caching can reduce the delay in receiving a requested object. Will Web caching reduce the delay for all objects requested by a user or for only some of the objects? Why?

- ➔ Web caching can bring the desired content "closer" to the user, possibly to the same LAN to which the user's host is connected. Web caching can reduce the delay for all objects, even objects that are not cached, since caching reduces the traffic on links.

R15. List several popular messaging apps. Do they use the same protocols as SMS?

- ➔ A list of several popular messaging apps: WhatsApp, Facebook Messenger, WeChat, and Snapchat. These apps use the different protocols than SMS.

R16. Suppose Alice, with a Web-based e-mail account (such as Hotmail or Gmail), sends a message to Bob, who accesses his mail from his mail server using POP3. Discuss how the message gets from Alice's host to Bob's host. Be sure to list the series of application-layer protocols that are used to move the message between the two hosts.

- ➔ The message is first sent from Alice's host to her mail server over HTTP. Alice's mail server then sends the message to Bob's mail server over SMTP. Bob then transfers the message from his mail server to his host over POP3.

R18. From a user's perspective, what is the difference between the download-and-delete mode and the download-and-keep mode in POP3?

- ➔ With download and delete, after a user retrieves its messages from a POP server, the messages are deleted. This poses a problem for the nomadic user, who may want to access the messages from many different machines (office PC, home PC, etc.). In the download and keep configuration, messages are not deleted after the user retrieves the messages. This can also be inconvenient, as each time the user retrieves the stored messages from a new machine, all of non-deleted messages will be transferred to the new machine (including very old messages).

R19. Is it possible for an organization's Web server and mail server to have exactly the same alias for a hostname (for example, foo.com)? What would be the type for the RR that contains the hostname of the mail server?

➔ Yes an organization's mail server and Web server can have the same alias for a host name. The MX record is used to map the mail server's host name to its IP address.

R20. Look over your received e-mails, and examine the header of a message sent from a user with a .edu e-mail address. Is it possible to determine from the header the IP address of the host from which the message was sent? Do the same for a message sent from a Gmail account

➔ You should be able to see the sender's IP address for a user with an .edu email address. But you will not be able to see the sender's IP address if the user uses a gmail account

R26. In Section 2.7, the UDP server described needed only one socket, whereas the TCP server needed two sockets. Why? If the TCP server were to support n simultaneous connections, each from a different client host, how many sockets would the TCP server need?

➔ With the UDP server, there is no welcoming socket, and all data from different clients enters the server through this one socket. With the TCP server, there is a welcoming socket, and each time a client initiates a connection to the server, a new socket is created. Thus, to support n simultaneous connections, the server would need n+1 sockets.

R27. For the client-server application over TCP described in Section 2.7 , why must the server program be executed before the client program? For the client-server application over UDP, why may the client program be executed before the server program?

➔ For the TCP application, as soon as the client is executed, it attempts to initiate a TCP connection with the server. If the TCP server is not running, then the client will fail to make a connection. For the UDP application, the client does not initiate connections (or attempt to communicate with the UDP server) immediately upon execution

P1. True or false?

a. A user requests a Web page that consists of some text and three images. For this page, the client will send one request message and receive four response messages. - False

b. Two distinct Web pages (for example, www.mit.edu/research.html and www.mit.edu/students.html) can be sent over the same persistent connection. - True

c. With nonpersistent connections between browser and origin server, it is possible for a single TCP segment to carry two distinct HTTP request messages. - False

d. The Date: header in the HTTP response message indicates when the object in the response was last modified. - False

e. HTTP response messages never have an empty message body. – False

P2. SMS, iMessage, and WhatsApp are all smartphone real-time messaging systems. After doing some research on the Internet, for each of these systems write one paragraph about the protocols they use. Then write a paragraph explaining how they differ.

SMS (Short Message Service) is a technology that allows the sending and receiving of text messages between mobile phones over cellular networks. One SMS message can contain data of 140 bytes and it supports languages internationally. The maximum size of a message can be 160 7-bit characters, 140 8-bit characters, or 70 16-bit characters. SMS is realized through the Mobile Application Part (MAP) of the SS#7 protocol, and the Short Message protocol is defined by 3GPP TS 23.040 and 3GPP TS 23.041. In addition, MMS (Multimedia Messaging Service) extends the capability of original text messages, and support sending photos, longer text messages, and other content.

iMessage is an instant messenger service developed by Apple. iMessage supports texts, photos, audios or videos that we send to iOS devices and Macs over cellular data network or WiFi. Apple's iMessage is based on a proprietary, binary protocol APNs (Apple Push Notification Service).

WhatsApp Messenger is an instant messenger service that supports many mobile platforms such as iOS, Android, Mobile Phone, and Blackberry. WhatsApp users can send each other unlimited images, texts, audios, or videos over cellular data network or WiFi. WhatsApp uses the XMPP protocol (Extensible Messaging and Presence Protocol).

iMessage and WhatsApp are different than SMS because they use data plan to send messages and they work on TCP/IP networks, but SMS use the text messaging plan we purchase from our wireless carrier. Moreover, iMessage and WhatsApp support sending photos, videos, files, etc., while the original SMS can only send text message. Finally, iMessage and WhatsApp can work via WiFi, but SMS cannot.

P3. Consider an HTTP client that wants to retrieve a Web document at a given URL. The IP address of the HTTP server is initially unknown. What transport and application-layer protocols besides HTTP are needed in this scenario?

- ➔ Application layer protocols: DNS and HTTP
- ➔ Transport layer protocols: UDP for DNS; TCP for HTTP

P11. Consider the scenario introduced in the previous problem. Now suppose that the link is shared by Bob with four other users. Bob uses parallel instances of non-persistent HTTP, and the other four users use non-persistent HTTP without parallel downloads.

a. Do Bob’s parallel connections help him get Web pages more quickly? Why or why not?

b. If all five users open five parallel instances of non-persistent HTTP, then would Bob’s parallel connections still be beneficial? Why or why not?

- ➔ a) Yes, because Bob has more connections, he can get a larger share of the link bandwidth.
- ➔ b) Yes, Bob still needs to perform parallel downloads; otherwise he will get less bandwidth than the other four users

P31. We have seen that Internet TCP sockets treat the data being sent as a byte stream but UDP sockets recognize message boundaries. What are one advantage and one disadvantage of byte-oriented API versus having the API explicitly recognize and preserve application-defined message boundaries?

- ➔ For an application such as remote login (telnet and ssh), a byte-stream oriented protocol is very natural since there is no notion of message boundaries in the application. When a user types a character, we simply drop the character into the TCP connection. In other applications, we may be sending a series of messages that have inherent boundaries between them. For example, when one SMTP mail server sends another SMTP mail server several email messages back to back. Since TCP does not have a mechanism to indicate the boundaries, the application must add the indications itself, so that receiving side of the application can distinguish one message from the next. If each message were instead put into a distinct UDP segment, the receiving end would be able to distinguish the various messages without any indications added by the sending side of the application.

Chapter – 3

R1. Suppose the network layer provides the following service. The network layer in the source host accepts a segment of maximum size 1,200 bytes and a destination host address from the transport layer. The network layer then guarantees to deliver the segment to the transport layer at the destination host. Suppose many network application processes can be running at the destination host.

a. Design the simplest possible transport-layer protocol that will get application data to the desired process at the destination host. Assume the operating system in the destination host has assigned a 4-byte port number to each running application process.

b. Modify this protocol so that it provides a “return address” to the destination process.

c. In your protocols, does the transport layer “have to do anything” in the core of the computer network?

Ans :

a) Call this protocol Simple Transport Protocol (STP). At the sender side, STP accepts from the sending process a chunk of data not exceeding 1196 bytes, a destination host address, and a destination port number. STP adds a four-byte header to each chunk and puts the port number of the destination process in this header. STP then gives the destination host address and the resulting segment to the network layer. The network layer delivers the segment to STP at the destination host. STP then examines the port number in the segment, extracts the data from the segment, and passes the data to the process identified by the port number.

b) The segment now has two header fields: a source port field and destination port field. At the sender side, STP accepts a chunk of data not exceeding 1192 bytes, a destination host address, a source port number, and a destination port number. STP creates a segment which contains the application data, source port number, and destination port number. It then gives the segment and the destination host address to the network layer. After receiving the segment, STP at the receiving host gives the application process the application data and the source port number.

c) No, the transport layer does not have to do anything in the core; the transport layer "lives" in the end systems.

R2. Consider a planet where everyone belongs to a family of six, every family lives in its own house, each house has a unique address, and each person in a given house has a unique name. Suppose this planet has a mail service that delivers letters from source house to destination house. The mail service requires that (1) the letter be in an envelope, and that (2) the address of the destination house (and nothing more) be clearly written on the envelope. Suppose each family has a delegate family member who collects and distributes letters for the other family members. The letters do not necessarily provide any indication of the recipients of the letters.

a. Using the solution to Problem R1 above as inspiration, describe a protocol that the delegates can use to deliver letters from a sending family member to a receiving family member.

b. In your protocol, does the mail service ever have to open the envelope and examine the letter in order to provide its service?

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b. In your protocol, does the mail service ever have to open the envelope and examine the letter in order to provide its service?

Ans :

a. For sending a letter, the family member is required to give the delegate the letter itself, the address of the destination house, and the name of the recipient. The delegate clearly writes the recipient’s name on the top of the letter. The delegate then puts the letter in an envelope and writes the address of the destination house on the envelope. The delegate then gives the letter to the planet’s mail service. At the receiving side, the delegate receives the letter from the mail service, takes the letter out of the envelope, and takes note of the recipient name written at the top of the letter. The delegate then gives the letter to the family member with this name.

b. No, the mail service does not have to open the envelope; it only examines the address on the envelope.

R3. Consider a TCP connection between Host A and Host B. Suppose that the TCP segments traveling from Host A to Host B have source port number x and destination port number y. What are the source and destination port numbers for the segments traveling from Host B to Host A?

➔ Source port number y and destination port number x.

R4. Describe why an application developer might choose to run an application over UDP rather than TCP.

➔ An application developer may not want its application to use TCP's congestion control, which can throttle the application's sending rate at times of congestion. Often, designers of IP telephony and IP videoconference applications choose to run their applications over UDP because they want to avoid TCP's congestion control. Also, some applications do not need the reliable data transfer provided by TCP.

R5. Why is it that voice and video traffic is often sent over TCP rather than UDP in today's Internet? (Hint: The answer we are looking for has nothing to do with TCP's congestion-control mechanism.)

➔ Since most firewalls are configured to block UDP traffic, using TCP for video and voice traffic lets the traffic though the firewalls.

R6. Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how?

➔ Yes. The application developer can put reliable data transfer into the application layer protocol. This would require a significant amount of work and debugging, however.

R7. Suppose a process in Host C has a UDP socket with port number 6789. Suppose both Host A and Host B each send a UDP segment to Host C with destination port number 6789. Will both of these segments be directed to the same socket at Host C? If so, how will the process at Host C know that these two segments originated from two different hosts?

➔ Yes, both segments will be directed to the same socket. For each received segment, at the socket interface, the operating system will provide the process with the IP addresses to determine the origins of the individual segments.

R8. Suppose that a Web server runs in Host C on port 80. Suppose this Web server uses persistent connections, and is currently receiving requests from two different Hosts, A and B. Are all of the requests being sent through the same socket at Host C? If they are being passed through different sockets, do both of the sockets have port 80? Discuss and explain.

➔ For each persistent connection, the Web server creates a separate “connection socket”. Each connection socket is identified with a four-tuple: (source IP address, source port number, destination IP address, destination port number). When host C receives and IP datagram, it examines these four fields in the datagram/segment to determine to which socket it should pass the payload of the TCP segment. Thus, the requests from A and B pass through different sockets. The identifier for both of these sockets has 80 for the destination port; however, the identifiers for these sockets have different values for source IP addresses. Unlike UDP, when the transport layer passes a TCP segment’s payload to the application process, it does not specify the source IP address, as this is implicitly specified by the socket identifier.

R14. True or false?

a. Host A is sending Host B a large file over a TCP connection. Assume Host B has no data to send Host A. Host B will not send acknowledgments to Host A because Host B cannot piggyback the acknowledgments on data. - false

b. The size of the TCP rwnd never changes throughout the duration of the connection. - false

c. Suppose Host A is sending Host B a large file over a TCP connection. The number of unacknowledged bytes that A sends cannot exceed the size of the receive buffer. - true

d. Suppose Host A is sending a large file to Host B over a TCP connection. If the sequence number for a segment of this connection is m, then the sequence number for the subsequent segment will necessarily be m + 1 - false

e. The TCP segment has a field in its header for rwnd . - true

f. Suppose that the last SampleRTT in a TCP connection is equal to 1 sec. The current value of TimeoutInterval for the connection will necessarily be sec. - false

g. Suppose Host A sends one segment with sequence number 38 and 4 bytes of data over a TCP connection to Host B. In this same segment the acknowledgment number is necessarily 42 – false

R15. Suppose Host A sends two TCP segments back to back to Host B over a TCP connection.

The first segment has sequence number 90; the second has sequence number 110.

a. How much data is in the first segment?

b. Suppose that the first segment is lost but the second segment arrives at B. In the

acknowledgment that Host B sends to Host A, what will be the acknowledgment number?

➔ a) 20 bytes b) ack number = 90

R18. True or false? Consider congestion control in TCP. When the timer expires at the sender, the value of ssthresh is set to one half of its previous value.

➔ False, it is set to half of the current value of the congestion window.