COMSATS UNIVERSITY ISLAMABAD
DEPARTMENT OF COMPUTER SCIENCE
Mid Term Examination, Fall 2023

Class/Section: -BSCS – 5A/B, BDS-5, BCT-5, BAI-5
Subject: -Design and Analysis of Algorithm
Instructor: -Tanveer Ahmed /Dr. Qurat ul Ain

Marks:- 25
Time:- 80 minutes
Dated:- November 7,2023

- *Students are required to display their ID cards.*
- *Attempt all questions. Return this question paper with your answer sheet.*
- *Use of Mobile Phones and Digital Diaries not allowed*

CLO-1: Demonstrate fundamental concepts of algorithm along with an algorithmic approach to a given problem.

Question # 01: ---------- CLO-1 ---------- Marks = [5]

a) Trace the following algorithm for the input [2,5,7,9,1] and describe what this algorithm computes.

ALGORITHM *Something* (A[1....n])
// **Input:** An array A[n] of n (n ≥ 2) numbers in increasing order
// **Output:**

```
1. m ← (A[1] + A[n])/2
2. i ← 1
3. while (A[i] < m) do
4.     i ← i + 1
5.  if ((A[i] - A[1]) < (A[n] - A[i-1])) then
6.      return A[i]
7.  else return A[i-1]
```

CLO-2: Design new algorithms for different computational problems

Question # 02: ---------- CLO-2 ---------- Marks = [5]

Consider the problem of counting, in a given text, the number of substrings that start with an A and end with a B. For example, there are four such substrings in CABAAXBYA. Design an algorithm for this problem using Brute-Force approach.

CLO-4 Analyze best, average, and worst-case behaviors of an algorithm.

Question # 03: ---------- CLO-4 ---------- Marks = [5*3=15]

a) Let $C_{BubbleSort}(n)$ and $C_{InsertionSort}(n)$ denotes the time complexites of Bubble sort and Insertion sort respectively. Determine how the number of comparisons used in the ***best case***, ***average case***, and ***worst case*** changes when the size of the list to be sorted doubles from n to $2n$, where n is a positive integer. Justify your answer.

b) Suppose there are two algorithms A and B that solve the same computational problem P. The time complexity function of Algorithm A is $f_A(n) = n^2 + 4n$ whereas $g_B(n) = 29n + 3$ be the time complexity function of Algorithm B in the worst case, for inputs of size n. For what input sizes is Algorithm A faster than Algorithm B (in worst case).

CamScanner

COMSATS University Islamabad, Lahore Campus

Terminal Examination – Semester Fall 2019

Course Title:	Data Communications and Computer Networks		Course Code:	CSC339	Credit Hours: 3(2,1)
Course Instructor/s:	Mr. Imran Raza, Dr. Tariq Umer, Dr. M. Hasanain Ch., Mr. M. Ali Khan, Mr. Shuja		Program Name:	BS Computer Science, BS Software Engineering	
Semester:	5th & 6th	Batch: Spring 17, Fall 17 Section:	All sections	Date:	
Time Allowed:	**3 Hours**		**Maximum Marks:**	**100**	

Important Instructions / Guidelines:

- All questions are compulsory
- **Return the question paper along with answer sheet**

1. Answer the following short questions: [30]

   a. If the TCP round-trip time, RTT, is currently ***30 msec*** and the following acknowledgements come in after ***26***, ***32***, and ***24 msec***, respectively, what is the new RTT estimate using the Van Jacobson algorithm? Use $\alpha = 0.9$. Now, if a retransmission is triggered, what will be the new Retransmission Time-Out (RTO)?(5)

   b. From an ISP's point of view, POP3 and IMAP differ in an important way. POP3 users generally empty their mailboxes every day. IMAP users keep their mail on the server indefinitely. Imagine that you were called in to advise an ISP on which protocol it should support. What considerations would you bring up? (5)

   c. Consider a simple application-level protocol built on top of UDP that allows a client to retrieve a file from a remote server residing at a well-known address. The client first sends a request with file name, and the server responds with a sequence of data packets containing different parts of the requested file. To ensure reliability and sequenced delivery, client and server use a stop-and-wait protocol. Ignoring the obvious performance issue, do you see a problem with this protocol? Think carefully about the possibility of processes crashing. (5)

   d. In a TCP connection if receiver announces a window size of 0, sender will stop sending and wait for an ACK with non-zero size. If this ACK gets lost, a deadlock occurs where sender waits for ACK and receiver waits for data. How TCP resolves this deadlock? (5)

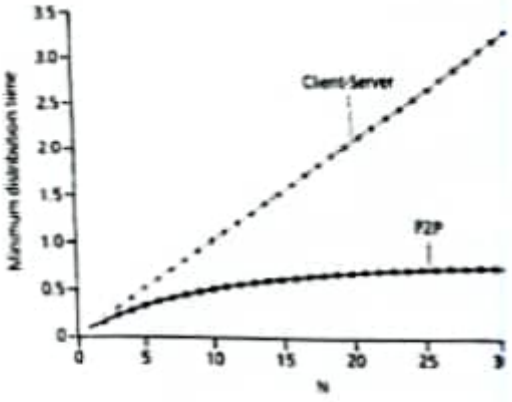
   e. Suppose Bob joins a BitTorrent torrent, but he does not want to upload any data to any other peers (so called free-riding). (5)

      a. Bob claims that he can receive a complete copy of the file that is shared by the swarm. Is Bob's claim possible? Why or why not?

      b. Bob further claims that he can further make his "free-riding" more efficient by using a collection of multiple computers (with distinct IP addresses) in the computer lab in his department. How can he do that?

   f. Most IP datagram reassembly algorithms have a timer to avoid having a lost fragment tie up reassembly buffers forever. Suppose that a datagram is fragmented into four fragments. The first three fragments arrive, but the last one is delayed. Eventually, the timer goes off and the three fragments in the receiver's memory are discarded. A little later, the last fragment stumbles in. What should be done with it? (5)

2. Consider distributing a file of $F = 15$ Gbits to N peers. The server has an upload rate of u_s = 30 Mbps, and each peer has a download rate of d_i =2 Mbps and an upload rate of u. For N =10, 100, and 1,000 and u = 300 Kbps, 700 Kbps, and 2 Mbps, prepare a chart, similar to the one given below, giving the minimum distribution time for each of the combinations of N and u for both client-server distribution and P2P distribution. [10]



3. Host A and B are communicating over a TCP connection, and Host B has already received from A all bytes up through byte

126. Suppose Host A then sends two segments to Host B back-to-back. The first and second segments contain 70 and 50 bytes of data, respectively. In the first segment, the sequence number is 127, the source port number is 302, and the destination port number is 80. Host B sends an acknowledgement whenever it receives a segment from Host A. [10]

a. In the second segment sent from Host A to B, what are the sequence number, source port number, and destination port number? (3)

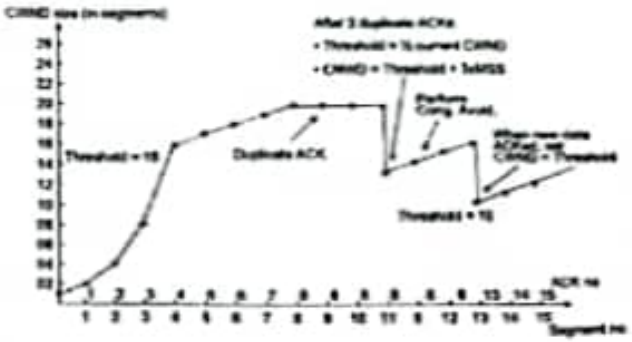
b. If the first segment arrives before the second segment, in the acknowledgement of the first arriving segment, what is the acknowledgment number, the source port number, and the destination port number? (3)

c. If the second segment arrives before the first segment, in the acknowledgement of the first arriving segment, what is the acknowledgment number? (4)

4. Consider the scenario given below and answer the following questions: [10]

a. Explain the convergence of TCP's Fast Retransmit and Fast Recovery (FRR) algorithm. What could be the cause of duplicate ACK and Why does it perform congestion avoidance rather than slow start as a part of Fast Recovery? (5)

b. How this modification to the congestion avoidance algorithm improves TCP's performance? (5)

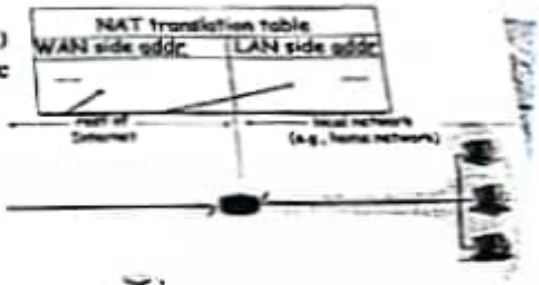


5. Explain CSMA/CD and CSMA/CA in detail with the help of an example. Why CSMA/CD cannot be used in wireless environment? Also discuss how collision is avoided in a wireless environment? [10]

6. Consider the given network setup, suppose that the ISP assigns the router the address 24.34.112.235 and that the network address of the home network is 192.168. 11.0/24. [10]

a. Assign addresses to all interfaces in the home network. (3)

b. Suppose each host has two ongoing TCP connections, all to port 80 at host 128.119.40.86. Provide the six corresponding entries in the NAT translation table. (7)

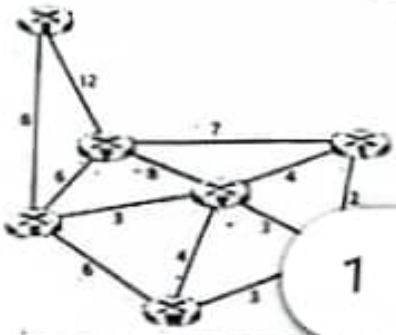


7. Consider the network shown below: [10]

a. Show the operation of Bellman Ford's (Distance Vector) algorithm for computing the least cost path from V to all destinations. Only compute the table of V. (5)

b. Identify the path with "count to infinity" problem and discuss its impact on the convergence of the algorithm in case of change in the link cost. (2)

c. Define Spanning Tree formally? Build a Minimum Spanning Tree (MST) using Reverse Path Forwarding (RPF) at node V to all nodes in the given network. (3)



8. Develop a subnetting scheme for the network requirements given below. [10]

d. Would you prefer using VLSM or FLSM for the given scenario? (2)

e. Identify the total number of required subnets? (2)

f. What is the subnet mask of each subnet? (3)

g. List down the subnet ids and host address range of all the subnets. (3)

IP Address 210.168.1.0/24	
Segment	Hosts
Computer Science	84
Electrical Engineering	52
Physics	30
WAN link1	2
WAN link2	2
WAN link3	2

CamScanner

COMSATS University Islamabad
Park Road, Tarlai Kalan, Islamabad 45550, Pakistan.
Department of Computer Science
Final Examination FALL-2021

Class/Section: BSCS, BSSE – 5th Semester **Dated:** Jan, 2022
Subject: Computer Communications & Networks **Marks:** 50
Instructor: Dr. Farhana Jabeen, Dr. Ashfaq Farooqi, Dr. Muhammad Waqar, Dr. Fakhar, Zulfiqar Ali **Time:** Three hour (3 Hour)

1. The question paper consists of 9 questions. Attempt all questions.

CLO-01(15 Marks)

a) [1.5 point] Briefly describe why the actual bandwidth on your cable access network connection on, e.g., a Saturday evening, could be significantly lower than what has been advertise by the provider (Comcast, Charter, etc.)? Would you see the same with a DSL-based provider?

Ans: The actual bandwidth on a cable access network connection could be significantly lower than what have been advertised by the provider because the cable access network is shared among multiple users. Bandwidth attached to each user will reduce to provider bandwidth to all users. We will not see the same with DSL-based provider because DSL connection is dedicated connection to the DSL office from the users.

b) [1.5 point] What is the purpose of the HTTP "COOKIE:" field? Are the values in the HTTP message's cookie field stored at the client or server or both? Explain briefly.

Ans: The purpose of the HTTP Cookie field is to send data of the stored cookie along with HTTP request. Cookie is used to store the state of a user. So, cookie field is set up in HTTO request message to server to maintain user state across sessions. The value in the HTTP messages cookie field is stored in the client.

c) [1.5 point] Suppose that we want to change the IP address of gaia.cs.umass.edu from 128.119.40.186 to 128.119.40.187 and change this mapping in the DNS authoritative name server for gaia.cs.umass.edu. Once this mapping is changed in the authoritative name server, will all future references (generated anywhere in the Internet) to gaia.cs.umass.edu then be sent to 128.119.40.187? Explain briefly (in two or three sentences).

Ans: If we change the Ip address of gaia.cs.umass.edu from 128.119.40.86 to 128.11.40.187 this mapping will not be known anywhere in the world at beginning. The reason for the local DNS server. Each dns mapping is stored for TTL which will expire. This this mapping will be known to all over the internet once cached entries are expired. Thus, all the future reference will be sent to 128.119.40.187

d) [03 point] Consider the scenario shown in Figure 2 in which a server is connected to a router by a 100Mbps link with a 50ms propagation delay. Initially this router is also connected to two routers, each over a 50Mbps link with a 200ms propagation delay. A 1Gbps link connects a host and a cache (if present) to each of these routers and we assume that this link has 0 propagation delay. All packets in the network are 20,000 bits long.

CamScanner

I. (02 Points) **What is the end--to--end delay from when a packet is transmitted by the server to when it is received by the client? In this case, we assume there are no caches, there's no queuing delay at the routers, and the packet processing delays at routers and nodes are all 0.**

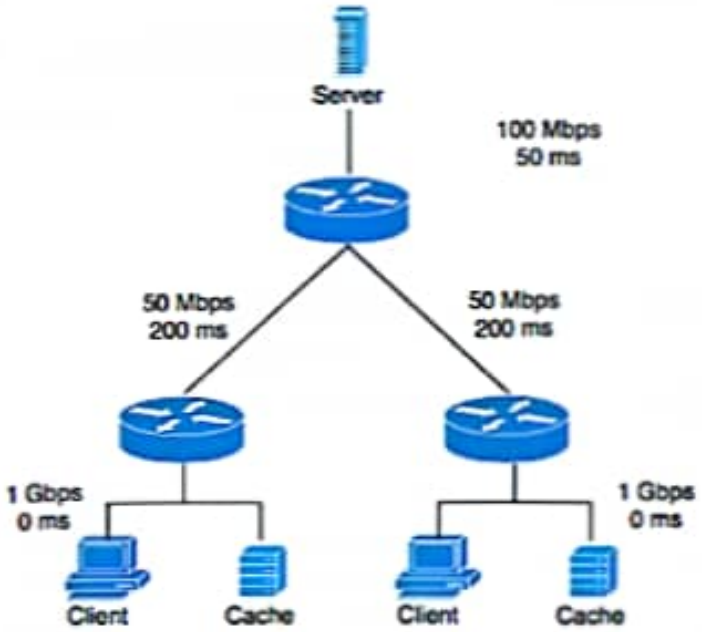


Figure 2

Ans: The

$$d_{proc} = 0,\ d_q = 0$$

$$d_{trans1} = \frac{L}{R_1} = \frac{20000}{100\times10^6} = 0.2ms$$

$$d_{trans2} = \frac{L}{R_2} = \frac{20000}{50\times10^6} = 0.4ms$$

$$d_{trans3} = \frac{L}{R_3} = \frac{20000}{1\times10^9} = 0.02ms$$

$$d_{nodal1} = d_{prop1} + d_{trans1} + d_{proc} + d_q = 50.2ms$$

$$d_{nodal2} = d_{prop2} + d_{trans2} + d_{proc} + d_q = 200.4ms$$

$$d_{nodal3} = d_{prop3} + d_{trans3} + d_{proc} + d_q = 0.02ms$$

Thus total delay is sum of all nodal delays

Total delay= $= d_{nodal1} + d_{nodal2} + d_{nodal3} = 250.62ms$

CamScanner

Hence the total end to end delay is 250.62ms

II. **(01 Points) Here we assume that client hosts send requests for files directly to the server (caches are not used or off in this case). What is the maximum rate at which the server can deliver data to a single client if we assume no other clients are making requests?**

Ans: If the host sends request for files to the server and server starts to deliver the data to the single client. Then the maximum rate at which server can deliver data to a single client will be 50Mbps. This is because the least rate of all three paths and it is causing bottle neck.

e) **[1.5 points] Consider a server-side socket that is used to communicate from server to client. In the case of a TCP socket, can data being read from the server-side socket have been sent by more than one client? Explain briefly. In the case of a UDP socket, can data being read from the server-side socket have been sent by more than one client? Explain briefly.**

Ans: In case of a server-side TCP socket for each client there will be a separate socket. This is because a tcp socket is identified by source Ip destination Ip, source port# and destination port#. Hence a specific TCP socket we will only receive data from single client. In this case UDP socket, a socket is identified by destination port# only. Because there are more than one client sending data to server with same destination port#. All the packet or data will be read from a same server-side socket.

f) **[1.5 points] Does the Internet checksum always detect errors in the transmitted segment?**

Ans: No, internet checksum cannot always detect errors for which the checksum remains same. Thus, error cannot be detected.

g) **[1.5 points] Consider two TCP sessions that must share a link's bandwidth. One of the TCP connections has been running for quite some time and has built up a large TCP sending window. The second connection then starts up with an initially small window. Long term, what will be the relative throughput achieved by these two TCP sessions? Explain.**

Ans: As there exists a tcp connection that has built a large TCP sending window and a new TCP connection with initially small window. Hence in order to achieve better performance both the TCP connections can be given equal bandwidth regardless of their starting time. In this case both will use half bandwidth of the connection which resulting an equal throughput for both connections.

h) **[1.5 points] Suppose nodes A, B, and C each attach to the same broadcast LAN (through their adapters). If A sends thousands of IP datagrams to B with each encapsulating frame addressed to the MAC address of B, will C's adapter process these frames? If so, will C's**

CamScanner

adapter pass the IP datagrams in these frames to the network layer in C? How would your answer change if A sends frames with the MAC broadcast address?

Ans: If A is sending thousands of IP datagrams to B with each encapsulated in frame addressed to the MAC address of B, then C will adapt never be able to process these frames because a switch learns about which device is connected to which interface and then sends data directl to that device. Hence C adapter will not process these frames. If A sends data in broadcas frame, then all noes will receive frame and can process the frame.

3

i) [1.5 points] Briefly describe how NAT works.

Ans: NAT stands for network address translation. The basic purpose of NAT is to use a single public address and have as many as required private address. NAT replaces the source IP with NAT IP and source port# with new port # for any outgoing packet and it remember packet reaches the router it source ip and port# with mapping.

CLO-02 (15 Marks)

---

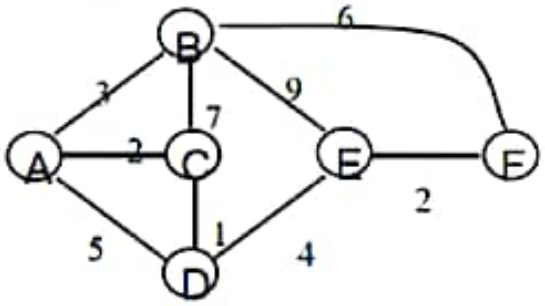
i. (3 Marks) Given a subnet 223.1.17.0/24, you are required to create three subnets from this such that Subnet 1 has at least 60 hosts, Subnet 2 has at least 40 hosts, and Subnet 3 has at least 12 hosts. You are required to provide the network addresses (of the form a.b.c.d/x) of these three subnets.

Ans: Three net- work addresses (of the form a.b.c.d/x) that satisfy given constraints as follows:

- 223.1.17.0/25
- 223.1.17.128/26
- 223.1.17.192/26

ii. (3 pts) Consider the network shown below. Show the operation of Dijkstra's algorithm for computing the least cost path from A to all destinations. Also draw the minimum spanning tree from A to all destinations.

Ans:

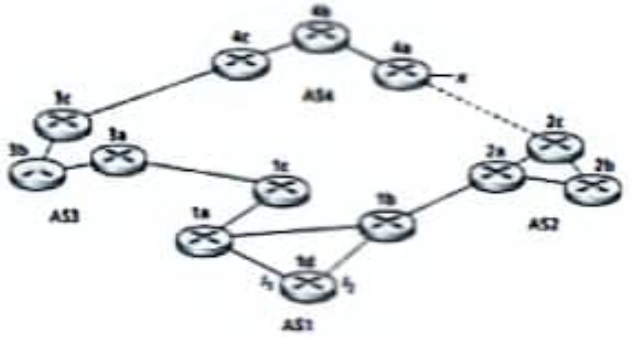


Step	N'	D(B), p(B)	D(C), p(C)	D(D), p(D)	D(E), p(E)	D(F), p(F)
0	A	3A	2A	5A	∞	∞
1	AC	3A	-	3C	∞	∞
2	ACD	3A	-	-	-	∞
3	ACDB	-	-	-	-	9B

CamScanner

4	ACDBE	-	-	-	-	9B
5	ACDBEF	-	-	-	-	

iii. **Consider the network shown in Figure 2. Suppose AS3 and AS2 are running OSPF for their intra-AS routing protocol. Suppose AS1 and AS4 are running RIP for their intra-AS routing protocol. Suppose eBGP and iBGP are used for the inter-AS routing protocol. Initially suppose there is no physical link between AS2 and AS4. (4 Marks)**



a. **Router 3c learns about prefix x from which routing protocol: OSPF, RIP, eBGP, or iBGP?**
eBGP: Router 3c learns about x from eBGP

b. **Router 3a learns about x from which routing protocol?**
iBGP: Router 3a learns about x from iBGP

c. **Router 1c learns about x from which routing protocol?**
eBGP: Router 1c learns about x from eBGP

d. **Router 1d learns about x from which routing protocol?**
iBGP: Router 1d learns about x from iBGP

iv. **A digital signal has eight levels. How many bits are needed per level? (1 Marks)**
Number of bits per level = log2(8) =3

v. **Assume we need to download text documents at the rate of 100 pages per second. What is the required bit rate of the channel? (1 Mark)**
A page is an average of 24 lines with 80 characters in each line.

1 page =24x80x8= 15360 bits for each page.

Bit rate = 100x15360= 1.536Mbps

vi. **The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNRdB? (1 Mark)**
The values of SNR and SNR_{dB} can be calculated as follows:
SNR = 10mW / 1 μW
SNR = 10 × 10 ^ -3 / 10 ^ -6 = 10,000
SNRdB = 10 log10 (10,000) = 40 dB

CamScanner

vii. **What are the differences between message confidentiality and message integrity? Can you have confidentiality without integrity? Can you have integrity without confidentiality? Justify your answer. (2 Marks)**

Ans: Confidentiality is the property that the original plaintext message cannot be determined by an attacker who intercepts the ciphertext-encryption of the original plaintext message. Message integrity is the property that the receiver can detect whether the message sent (whether encrypted or not) was altered in transit. The two are thus different concepts, and one can have one without the other. An encrypted message that is altered in transmit may still be confidential (the attacker can not determine the original plaintext) but will not have message integrity if the error is undetected. Similarly, a message that is altered in transit (and detected) could have been sent in plaintext and thus would not be confidential.

CLO-03 (15 Marks)

---

i. **Suppose you purchase a wireless router and connect it to your cable modem. Also suppose that your ISP dynamically assigns your connected device (that is, your wireless router) one IP address. Also suppose that you have five PCs at home that use 802.11 to wirelessly connect to your wireless router.**

(a) How are IP addresses assigned to the five PCs?

(b) Does the wireless router use NAT? Why or why not? (3 Marks)

Answer: Typically, the wireless router includes a Dynamic Host Configuration Protocol (DHCP) server. DHCP is used to assign IP addresses to the 5 PCs and to the router interface. Yes, the wireless router also uses NAT as it obtains only one IP address from the ISP.

(c) IP addresses are assigned from the range of private addresses.

(d) Wireless router uses NAT for it.

5

ii. **What is the bit rate for high-definition TV (HDTV)? (2 Marks)**

Answer: The HDTV screen is normally a ratio of 16: 9 (in contrast to 4: 3 for regular TV) There are 1920 by 1080 pixels per screen, and. Screen is renewed 30 times per second. 24-bits represent one color pixel.

Therefore,

Bit Rate = $1920 \times 1080 \times 30 \times 24 = 1.492992.000 =$ **1.5 Gbps.**

iii. **The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km? (1 Mark)**

The loss in the cable in decibels is 5 x (-0.3) = -1.5 dB.

We can calculate the power as

dB =10 log10 (p2/p1) = -1.5

(p2/p1) = 10 -1.5 = 0.71

P2 = 0.71*p1 = 0.71*2 = 1.4mw

iv. **We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need? (1 Marks)**

We can use the Nyquist formula as shown:

Bit Rate = 2 x bandwidth x log2 L

265,000 =2 * 20,000 * log2 L

CamScanner

log2 L =6.625
L =2^6.625 = 98.7 levels
Since this result is not a power of 2, we need to either increase the number of levels or reduce the bit rate. If we have 128 levels, the bit rate is 280 kbps. If we have 64 levels, the bit rate is 240 kbps. So that it is 128 level

v. **We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level? (1 Mark)**
First, we use the Shannon formula to find the upper limit.
C =B log2 (1 + SNR)
=10^6 log2 (1 + 63)
=10^6 1og2 64
=6 Mbps

Then we use the Nyquist formula to find the number of signal levels.
Bit Rate = 2 x bandwidth x log2 L
4Mbps=2* 1 MHz * log2 L
L=4

vi. **What are the propagation time and the transmission time for a 5-MB (megabyte) message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4 A~ 108 m/s. (1 Mark)** 6

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5{,}000{,}000 \times 8}{10^6} = 40 \text{ s}$$

Note that in this case, because the message is very long and the bandwidth is not very high, the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

vii. **From a service perspective, what is an important difference between a symmetric-key system and a public-key system? (1 Mark)**
Answer: One important difference between symmetric and public key systems is that in symmetric key systems both the sender and receiver must know the same (secret) key. In public key systems, the encryption and decryption keys are distinct. The encryption key is known by the entire world (including the sender), but the decryption key is known only by the receiver.

viii. **What does it mean for a signed document to be verifiable and nonforgeable? (1 Mark)**
Answer: Suppose Bob sends an encrypted document to Alice. To be verifiable, Alice must be able to convince herself that Bob sent the encrypted document. To be non-forgeable, Alice must be able to convince herself that only Bob could have sent the encrypted

CamScanner

document (e.g., non-one else could have guessed a key and encrypted/sent the document) To be nonreportable, Alice must be able to convince someone else that only Bob could have sent the document. To illustrate the latter distinction, suppose Bob and Alice share a secret key, and they are the only ones in the world who know the key. If Alice receives a document that was encrypted with the key and knows that she did not encrypt the document herself, then the document is known to be verifiable and non-forgeable (assuming a suitably strong encryption system was used). However, Alice can not convince someone else that Bob must have sent the document, since in fact Alice knew the key herself and could have encrypted/sent the document.

ix. **Suppose that Bob receives a PGP message from Alice. How does Bob know for sure that Alice created the message (rather than, say, Trudy)? Does PGP use a MAC for message integrity? (3 Marks)**
Answer:
- Bob receives a PGP message from Alice"

PGP stands for pretty good privacy. It is a signed message with encrypted e-mail scheme.

- Assume Alice encrypts the message is ($A(m)$) with her private key (P_A^-) and sends (m, $P_A^-(A(m))$) and decrypts public-key (as $P_A^+(P_A^-(A(m)))$
- Computer message digest ($A(m)$).
- $P_A^+(P_A^-(A(m))) = A(m)$
- If both the outcomes are same then *Bob* knows that the message Alice's message.

PGP does not PGP use a MAC for message integrity. The reason is that uses only digital signatures.

x. **What does it mean to say that a nonce is a once-in-a-lifetime value? In whose lifetime? (1 Mark)**
Answer: Once in a lifetimes means that the entity sending the nonce will never again use that value to check whether another entity is "live".

CLO-04 (10 Marks)

---

a. **Wireless channel has many impairments. Differentiate between multipath propagation and path loss. (2 Marks)**
Answer: Path loss is due to the attenuation of the electromagnetic signal when it travels through matter. Multipath propagation results in blurring of the received signal at the receiver and occurs when portions of the electromagnetic wave reflect off objects and ground, taking paths of different lengths between a sender and receiver. Interference from other sources occurs when the other source is also transmitting in the same frequency range as the wireless network.

CamScanner

b. Let's assume that RTS and CTS frames were of the size of the standard DATA and ACK frames. Would it be having any advantage? Why //why not? (2 Marks)

Answer:

No, there wouldn't be any advantage. Suppose there are two stations that want to transmit at the sometime and they both use RTS/CTS. If the RTS frame is as long as a DATA frames, the channel would be wasted for as long as it would have been wasted fortwo colliding DATA frames. Thus, the RTS/CTS exchange is only useful when the RTS/CTS frames are significantly smaller than the DATA frames.

c. What is the difference between direct and indirect routing from a mobile host? What is triangular routing problem? (2 Marks)

d. What is role of beacon frame in 802.11? how active scanning differs from passive scanning? (1 Mark)

Answer: Access Points (APs) transmit beacon frames. An AP"s beacon frames will be transmitted over one of the 11 channels. The beacon frames perm it nearby wireless stations to discover and identify the AP.

During an active scan, the client radio transmits a probe request and listens for a probe response from an AP. With a passive scan, the client radio listens on each channel for beacons sent periodically by an AP. A passive scan generally takes more time, since the client must listen and wait for a beacon versus actively probing to find an AP. Another limitation with a passive scan is that if the client does not wait long enough on a channel, then the client may miss an AP beacon.

e. In wired network, CSMA/CD is used to coordinate access among multiple nodes, what protocol is used in wireless network. How is that protocol different from CSMA/CD. (2 Marks)

8

Answer: CSMA/CA

CSMA/CA stands for Carrier Sense Multiple Access/ Collision Avoidance. It is a network protocol for transmission. It operates in the Medium Access Control Layer. This protocol is effective before the collision.

CSMA/CD

CSMA/CD stands for Carrier Sense Multiple Access/ Collision Detection. It is also a network protocol for transmission and operates in the Medium Access Control Layer. In this protocol, each station senses the collision by broadcast sensing. In case of collision, the transmission is stopped, and they send a jam signal and then wait for a random time context before retransmission.

CamScanner

The following are some of the important differences between CSMA/CA and CSMA/CD.

Sr. No.	Key	CSMA/CA	CSMA/CD
1	Effectiveness	CSMA/CA is effective before a collision.	CSMA/CD is effective after a collision.
2	Network Type	CSMA/CA is generally used in wireless networks.	CSMA/CD is generally used in wired networks.
3	Recovery Time	CSMA/CA minimizes the risk of collision.	CSMA/CD reduces recovery time.
4	Conflict Management	CSMA/CA initially transmits the intent to send the data, once an acknowledgment is received, the sender sends the data.	CSMA/CD resends the data frame in case a conflict occurs during transmission.
5	IEEE Standards	CSMA/CA is part of the IEEE 802.11 standard.	CSMA/CD is part of the IEEE 802.3 standard.
6	Efficiency	CSMA/CA is similar in efficiency as CSMA.	CSMA/CD is more efficient than CSMA.

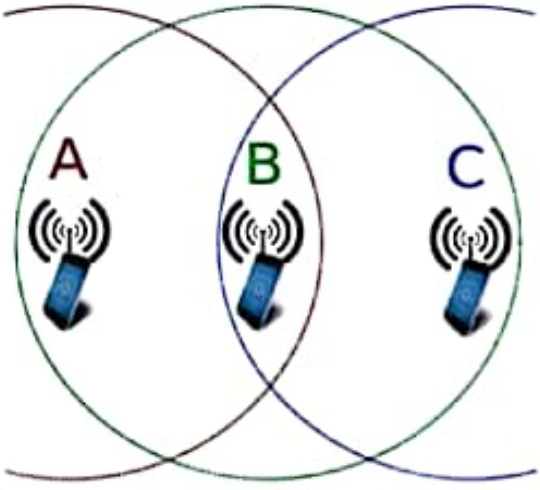
f. Explain Hidden node/terminal problem and its solution. (1 Mark)

Answer: A wireless network with lack of centralized control entity, sharing of wireless bandwidth among network access nodes i.e., medium access control (MAC) nodes must be organized in decentralized manner.

The hidden terminal problem occurs when a terminal is visible from a wireless access point (APs), but not from other nodes communicating with AP. This situation leads the difficulties in medium access control sub-layer over wireless networking.

In a formal way hidden terminals are nodes in a wireless network that are out of range of other node or a collection of nodes.

Consider the scenario of wireless networking with three wireless devices (e.g., mobile phones) as shown below.



The transmission range of access point A reaches at B, but not at access point C, similarly transmission range of access point C reaches B, but not at A. These nodes are known as hidden

CamScanner

terminals. The problem occurs when nodes A and C start to send data packets simultaneously to the access point B. Because the access points A and c are out of range of each other and resultant they cannot detect a collision while transmitting. Carrier sense multiple access with collision detection (CSMA/CD) does not work, and collisions occur, which then corrupt the data received by the access point B due to the hidden terminal problem.
The hidden terminal analogy is determined as follows:

- Terminal C wants to send data to B, terminal C senses a "free" medium (CS fails) and starts transmitting
- Collision at B occurs, A cannot detect this collision (CD fails) and continues with its transmission at B
- Terminal A is "hidden" from C and vice versa

Solution: Some other technology that can be employed to solve hidden node problem are: Increase Transmitting Power from the Nodes. With the enhancement of the transmission power of access point can solve the hidden terminal problem by allowing the cell around each mode to increase in size, encompassing all of the the nodes.
Use Omni directional antennas: Since nodes using directional antennas are nearly invisible to nodes that are not positioned in the direction the antenna is aimed at, directional antennas should be used only for very small networks.
Use protocol enhancement software: Pooling and token passing strategy should be used before start data transformation.

Past Paper: CLO-02 (15 Marks)

Q2 [15 pts]

a. (4.5 points) Consider the three-node topology: the link costs are c(x,y)=6, c(y,z)=2, and c(z,x)=50. We focus here only on y's and z's distance table entries to destination x. Please explain the details of how the DV algorithm could reach a quiescent (or stable) state when the following cases happen.

i. (1.5 points) When c(x,y) decreases from 6 to 1.

10

Y=

x	y	z
6	0	2

Z=

x	y	z
6	0	2

When c (x, y) decreases to 1 y and x will see that local link cost changed. Hence y will send advertisement to its neighbor as well as x will also send advertisement for. The Z will learn

CamScanner

that its distance through y to x is only 2. Thus, distance table

ii. (1.5 points) When c (x, y) increases from 6 to 5.

Y=

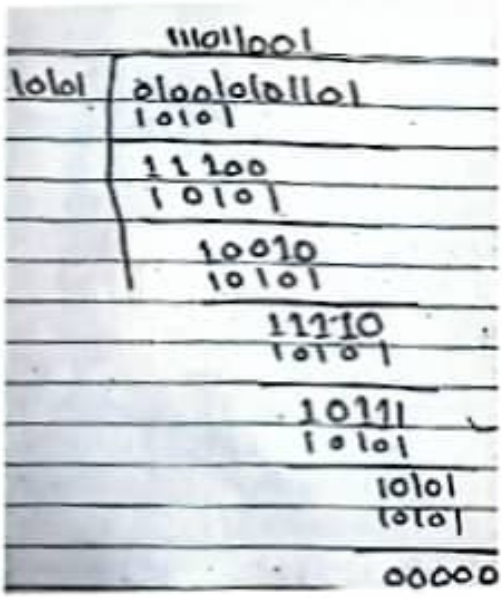
x	y	z
5	0	2

Z=

x	y	z
7	2	0

Q3. [4 points]

Suppose a receiver using the CRC generator polynomial x 4 + x 2 + 1 (10101) receives the sequence of bits 010010101101. Were they received correctly? If so, what was the transmitted message? If not, can you tell how many bit errors occurred? Show your work.



As the result or remainder is 0000. Hence there are not bit errors in the message. The transmitted message will be 01001010000 and with CRC bits it will be 01001010

Q4. [03 pts]

a) (1.5 pts) In Figure 5, A is sending an ARP request to determine the MAC address it should use for a frame destined to B. What MAC address will be returned to A in the ARP reply. Briefly explain your answer.

Ans: A us sending an ARP query to get the MAC address of B. This query is directed to B via its IP address in the ARP frame. Once the ARP frame reaches B. B sends its MAC address in reply. Thus MAC address in reply will be BB:89:34:E7:01:3B

CamScanner

b) (1.5 pts) How many interfaces of all the ones shown in Figure 5 receive the original ARP request sent by A and what is the destination MAC address on the frame containing the ARP request?

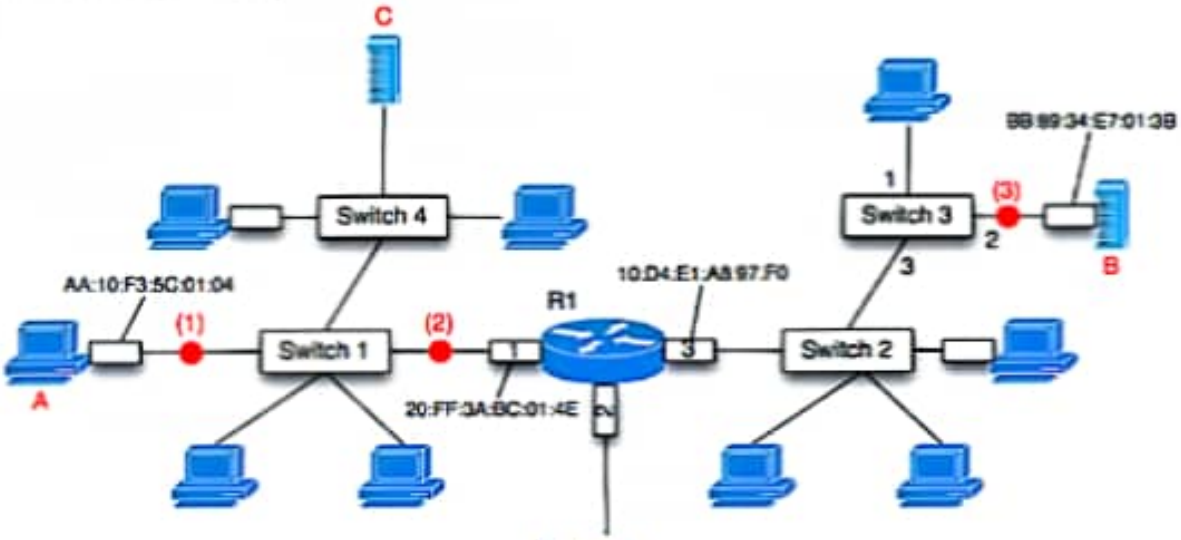


Figure 5

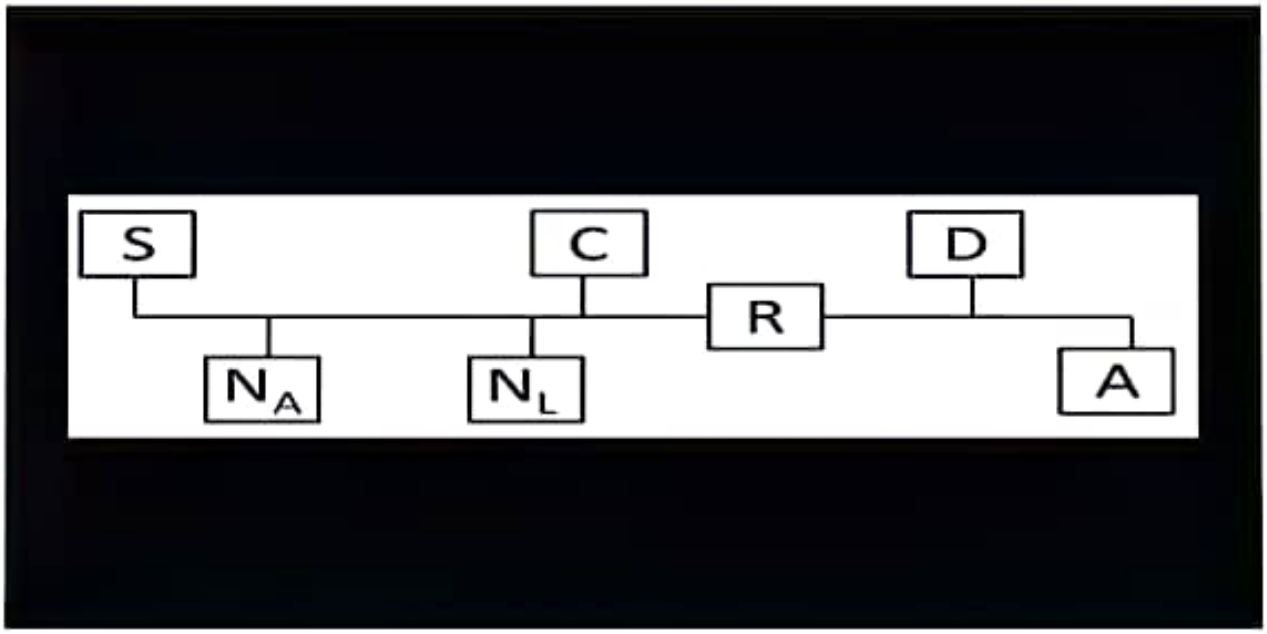
Ans: 20:FF:3A:BC:01:4E

Q5. [04 pts]

In the topology shown below, machine A is a desktop client, D is a local DHCP server, R is A's gateway router, S is a Web server, and C is a Web cache. NL is C's local nameserver, while NA is the nameserver authoritative for S. Client A is configured to use Web cache C for all requests (assume that the Web cache resolves the name for any Web server and that the client is configured with the IP address of the cache). All wires/links are Ethernet segments. Assume the following:

- **All the machines were just booted and their associated caches (ARP, DNS, Web, persistent connections) are all empty**
- **http://S/index.html fits in a single packet**

CamScanner



- **Persistent HTTP connections are used among A, C, and S (i.e. you should assume that once any connection between these hosts is established it is never closed)**
- **Web caches respond to TCP requests that look like packet two in table 1 below (e.g., GET http://foo/bar/). They reply with the normal web cache contents.**

i. **(01 point)To connect machine A, it needs to get its own IP address, addr of first-hop router, addr of DNS server. Which protocol should be used among DNS, DHCP, ARP.**

Ans: To get its own IP address, address of first hop router and address of DNS server it will use DHCP protocol.

ii. **(2 points) In which scenarios, TCP applies slow start, Tahoe, Reno and AIMD principles? Explain each with a proper example?**

Ans: TCP applies slow start when the device CWND size is less than ssthreash. As in the above scenario devices have just connected and there is no traffic on links. Hence slow start will be used to allow devices to send at faster rate.

AMID is used once cwnd increase from ssthreash. This is used to prevent congestion. If slowly increases the cwnd size so that network do not directly go into congestion state. TCP will use AIMD to prevent congestion.

iii. **(1 points) Why CSMA/CA is preferred over CSMA/CD for wireless channel?**

Ans: CSMA/CA is preferred over CSMA/CD for wireless channel because it is very difficult to detect collision in wireless network. Hence to remove this complexity and CTS packets to ensure that collision can avoided

Q6. [6 pts]

a. **(1.5 pts) What is the minimum-width channel that could possibly transmit 80 Kbps in the presence of a signal-to-noise ratio of 15?**

CamScanner

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

$$B = \frac{C}{\log_2(1+SNR)} = \frac{80\times1000}{\log_2(1+15)}$$

$$B = 20khz$$

b. (1.5 pts) We need to send 265 kbps over a noiseless channel with a bandwidth of 30 kHz. How many signal levels do we need?

$$C = 2B\log_2 L$$

$$\text{Log}_2 L = \frac{C}{2B} = \frac{256\times1000}{2\times30\times1000}$$

$$L = 19 \text{ signal levels}$$

c. (1.5 pts) The period of a signal is 100 ms. What is its frequency in kilohertz?

$$f\frac{1}{T} = \frac{1}{100ms} = 0.01KHz$$

CLO-04

Q7.

a. (2 pts) What is the hidden node problem in wireless networks? Explain how CSMA/CA is realized in the case of the IEEE 802.11 protocol.

Ans: Hidden node problem is that in wireless network two nodes might be located in a way that they could not receive signal of each other. Hence both nodes will never know each other

b. (2 pts) Explain why 802.11 uses Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) rather than CSMA/CD. Be specific.

Ans: CSMA/CA is used in 802.11 rather then CSMA/CD because in wireless network is very difficult to detect collision. Due to this reason CSMA/CA could not be used

Q8.

a. (2 pts) Use the Caesar cipher to encrypt the message "JAVATPOINT," and the key (shift) value of this message is 3?

Ans: MDYDWSRLQW

b. (2 pts) Alice has sent messages to Bob. These messages are in fact related to ordering an expensive book. Bob wants to be able to prove in court if necessary that Alice actually sent these messages to Bob containing the order for textbooks. What does Alice need to do/add to the messages, in order for Bob to be able to prove that Alice is the one who sent it originally?

Ans: In order to achieve this functionality asymmetric keys can be used. Hence when Alice is sending message, she should encrypt the message with her private key. Once bob receives the

CamScanner

message bob can use the Alice public key to read the data and to ensure that data was sent by Alice.

Q9. **[3 pts]**

a. (1 pts). What are the possible multiple access schemes for cellular networks?

Ans: CSMA/CA, TDM, FDM, OFDM

b. (2 pts). Explain the following terms related to cellular networks briefly:

i. Handoff

Handoff usually occurs when a mobile moves from one cell to another. Hence the base station has to handover the mobile device to another BS.

ii. Frequency Reuse

Frequency Reuse is used to use same set of frequencies against cells use same frequency because it can be harmful and can cause interference. But two cells which are away from each other can reuse the same frequency.

CamScanner

COMSATS University Islamabad
Department of Computer Science
Terminal Examination FALL-2022
Instructor: Dr. Mariam Akbar, Dr. Inam Ullah Khan, Mr. Saleem Raza

Course Title:	***CSC339 Data Comm. & Computer Networks***	**Maximum Marks:**	**50**
Dated:	**6th June, 2022**	**Time Allowed:**	**3 Hours**

Note: Solve all questions

CLO-01 (5 Marks)

Question 1:

i. Consider sending a 2400-byte datagram into a link that has an MTU of 700 bytes. Suppose the original datagram is stamped with the identification number 422. How many fragments are generated? What are the values in the various fields in the IP datagram(s) generated related to fragmentation? **(3 Marks)**

Length	Id	More Fragment Flag	Offset

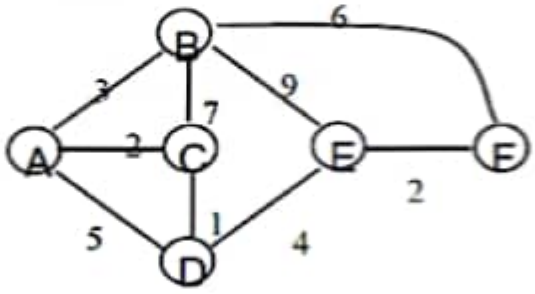
ii. Do routers have IP addresses? If so, how many IP addresses and does each IP address have to belong to the same or a different subnet? **(2 Marks)**

CLO-02 (15 Marks)

Question 2:

i. Given a subnet 223.1.17.0/24, you are required to create three subnets from this such that Subnet 1 has at least 60 hosts, Subnet 2 has at least 40 hosts, and Subnet 3 has at least 12 hosts. You are required to provide the network addresses (of the form a.b.c.d/x) of these three subnets. **(3 Marks)**

ii. Consider the network shown below in **Figure 1.** Show the operation of Dijkstra's algorithm for computing the least cost path from A to all destinations. Also draw the minimum spanning tree from A to all destinations. **(2 Marks)**



CamScanner

Figure 1

iii. Consider the network shown in **Figure 2**. Suppose AS3 and AS2 are running OSPF for their intra-AS routing protocol. Suppose AS1 and AS4 are running RIP for their intra-AS routing protocol. Suppose eBGP and iBGP are used for the inter-AS routing protocol. Initially suppose there is no physical link between AS2 and AS4. **(2 Marks)**

a. Router 3c learns about prefix x from which routing protocol: OSPF, RIP, eBGP, or iBGP?
b. Router 3a learns about x from which routing protocol?
c. Router 1c learns about x from which routing protocol?
d. Router 1d learns about x from which routing protocol?

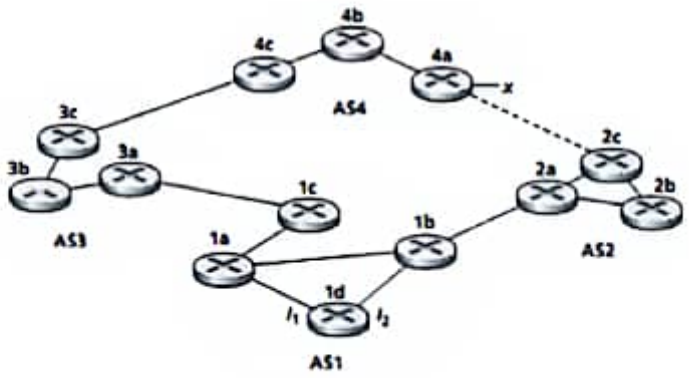


Figure 2

iv. A digital signal has eight levels. How many bits are needed per level? **(2 Marks)**

v. Assume we need to download text documents at the rate of 100 pages (20 lines/page and 60 characters/line) per second. What is the required bit rate of the channel? **(2 Marks)**

vi. The power of a signal is 20 mW and the power of the noise is 2 μW; what are the values of SNR and SNRdB? **(2 Marks)**

vii. What are the differences between message confidentiality and message integrity? Can you have confidentiality without integrity? Can you have integrity without confidentiality? Justify your answer. **(2 Marks)**

CLO-03 (16 Marks)

Question 3:

i. The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with −0.3 dB/km has a power of 5 mW, what is the power of the signal at 5 km? **(2 Marks)**

ii. We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need? **(2 Marks)**

iii. We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level? **(2 Marks)**

iv. What are the propagation time and the transmission time for a 5-MB (megabyte) message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s. **(2 Marks)**

v. From a service perspective, what is an important difference between a symmetric-key system and a public-key system? **(2 Marks)**

CamScanner

vi. What does it mean for a signed document to be verifiable and nonforgeable? **(2 Marks)**

vii. Suppose that Bob receives a PGP message from Alice. How does Bob know for sure that Alice created the message (rather than, say, Trudy)? Does PGP use a MAC for message integrity? **(2 Marks)**

viii. What does it mean to say that a nonce is a once-in-a-lifetime value? In whose lifetime? **(2 Marks)**

CLO-04 (14 Marks)

Question 4:

i. Suppose you purchase a wireless router and connect it to your cable modem. Also suppose that your ISP dynamically assigns your connected device (that is, your wireless router) one IP address. Also suppose that you have five PCs at home that use 802.11 to wirelessly connect to your wireless router. How are IP addresses assigned to the five PCs? Does the wireless router use NAT? Why or why not? **(4 Marks)**

ii. Wireless channel has many impairments. Differentiate between multipath propagation and path loss. **(2 Marks)**

iii. Let's assume that RTS and CTS frames were of the size of the standard DATA and ACK frames. Would it be having any advantage? Why //why not? **(2 Marks)**

iv. What is role of beacon frame in 802.11? how active scanning differs from passive scanning? **(2 Mark)**

v. In wired network, CSMA/CD is used to coordinate access among multiple nodes, what protocol is used in wireless network. How is that protocol different from CSMA/CD. **(2 Marks)**

vi. Explain Hidden node/terminal problem and its solution. **(2 Mark)**

CamScanner

COMSAT Lancaster Dual Degree Programme

COMSATS Institute of Information Technology Lahore

Lancaster University

Terminal Examination – Semester Fall 2016

Course Title:	Computer Communication and Networks				Course Code:	CSC352	Credit Hours: 3(3,0)
Course Instructor/s:	Dr. M. Hasanain Ch., Mr. Imran Raza & Dr. Shahbaz Akhtar Abid				Program Name:	BS Computer Science, BS Software Engineering	
Semester:	5th	Batch:	Fall-14	Section:	A, B, C	Date:	
Time Allowed:	3 Hours				Maximum Marks:	100	
Student's Name:					Reg. No.		

Important Instructions / Guidelines:

- All questions are compulsory
- Write answers for all the questions in your answer script

1. Answer following short questions: [30]

a. In the simplest version of the Chord algorithm for peer-to-peer lookup, searches do not use the finger table. Instead, they are linear around the circle, in either direction. Can a node accurately predict which direction it should search? Discuss your answer. (5)

b. Explain Point Coordination Function (PCF) and Distributed Coordination Function (DCF) in detail. Also, discuss that both of these coordination functions can work together or not, if yes, explain your answer with the help of a suitable example. (5)

c. Consider a scenario in which HTTP clients need to retrieve a web page at a known URL. Initially the IP address of the HTTP server is not known. The web page at the specified URL is having one JPEG image which is saved on the same HTTP server as the original document. What types of application layer and transport layer protocols besides HTTP arerequired? (5)

d. Suppose a server is utilizing TCP SYN cookies, then why it is essential for the server to use a special initial sequence number? (5)

e. Consider a simple application-level protocol built on top of UDP that allows a client to retrieve a file from a remote server residing at a well-known address. The client first sends a request with file name, and the server responds with a sequence of data packets containing different parts of the requested file. To ensure reliability and sequenced delivery, client and server use a stop-and-wait protocol. Ignoring the obvious performance issue, do you see a problem with this protocol? (5)

f. Consider a scenario in which two different TCP senders, sending their network traffic from thei respective hosts to different destinations, but pass through a common bottleneck link (that is the onl bottleneck link on either of their paths). What does it mean to say that TCP provides fair sharing bandwidth at the bottleneck link? Assume the RTTs of the two connections are altogether diff TCP "fair" for this situation? Legitimize your answer.

2. Explain how Peer-to-Peer (P2P) applications enhance following network functions.

a. Real Time Communications (RTC) (2.5)

b. Collaberation (2.5)

c. Content Distribution (2.5)

d. Distributed Processing (2.5)

3. Suppose users share a 3 Mbps link. Also suppose each user requires 150 kbpswhen transmitting each user transmits only 10 percent of the time. [15]

a. When circuit switching is used, how many users can be supported? (3)

b. For the remainder of this problem, suppose packet switching is used. Find the probability that a user is transmitting. (4)

c. Suppose there are 120 users. Find the probability that at any given time, exactly *n* use transmitting simultaneously. (4)

CamScanner

COMSAT Lancaster Dual Degree Programme

COMSATS Institute of Information Technology Lahore

Lancaster University

Terminal Examination – Semester Fall 2016

Course Title:	Computer Communication and Networks				Course Code:	CSC352	Credit Hours: 3(3,0)
Course Instructor/s:	Dr. M. Hasanain Ch., Mr. Imran Raza & Dr. Shahbaz Akhtar Abid				Program Name:	BS Computer Science, BS Software Engineering	
Semester:	5th	Batch:	Fall-14	Section:	A, B, C	Date:	
Time Allowed:	**3 Hours**				**Maximum Marks:**	**100**	
Student's Name:					Reg. No.		

Important Instructions / Guidelines:

- All questions are compulsory
- Write answers for all the questions in your answer script

1. **Answer following short questions:** [30]
   a. In the simplest version of the Chord algorithm for peer-to-peer lookup, searches do not use the finge table. Instead, they are linear around the circle, in either direction. Can a node accurately predict whic direction it should search? Discuss your answer. (5)
   b. Explain Point Coordination Function (PCF) and Distributed Coordination Function (DCF) in deta Also, discuss that both of these coordination functions can work together or not, if yes, explain yo answer with the help of a suitable example. (5)
   c. Consider a scenario in which HTTP clients need to retrieve a web page at a known URL. Initially t IP address of the HTTP server is not known. The web page at the specified URL is having one JPE image which is saved on the same HTTP server as the original document. What types of applicati layer and transport layer protocols besides HTTP arerequired? (5)
   d. Suppose a server is utilizing TCP SYN cookies, then why it is essential for the server to use a spec initial sequence number? (5)
   e. Consider a simple application-level protocol built on top of UDP that allows a client to retrieve a f from a remote server residing at a well-known address. The client first sends a request with file na and the server responds with a sequence of data packets containing different parts of the requested f To ensure reliability and sequenced delivery, client and server use a stop-and-wait protocol. Ignor the obvious performance issue, do you see a problem with this protocol? (5)
   f. Consider a scenario in which two different TCP senders, sending their network traffic from t respective hosts to different destinations, but pass through a common bottleneck link (that is the bottleneck link on either of their paths). What does it mean to say that TCP provides fair sharin bandwidth at the bottleneck link? Assume the RTTs of the two connections are altogether differer TCP "fair" for this situation? Legitimize your answer. (5)
2. **Explain how Peer-to-Peer (P2P) applications enhance following network functions.** [10]
   a. Real Time Communications (RTC) (2.5)
   b. Collaboration (2.5)
   c. Content Distribution (2.5)
   d. Distributed Processing (2.5)
3. **Suppose users share a 3 Mbps link. Also suppose each user requires 150 kbpswhen transmitting each user transmits only 10 percent of the time.** [15]
   a. When circuit switching is used, how many users can be supported? (3)
   b. For the remainder of this problem, suppose packet switching is used. Find the probability that a user is transmitting. (4)
   c. Suppose there are 120 users. Find the probability that at any given time, exactly n user transmitting simultaneously. (4)

SECRECY BRANCH DUAL DEGREE PROGRAMME

CamScanner

d. Find the probability that there are 21 or more users transmitting simultaneously. (4)

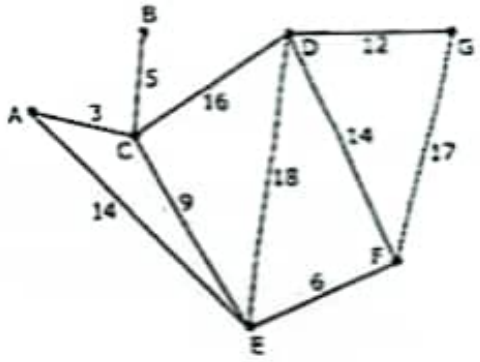
4. **Consider sending a 16000-byte datagram into a link that has an MTU of 900 bytes. Suppose the original datagram has the identification number 111. Answer following questions.** [10]
   a. How many segments are generated? (5)
   b. What are the values of Identification, DF, MF and fragment offset fields? (5)

5. **Consider the following scenario, in which user 1 from Thailand and user 2 from Japan are connected to the internet. The Web Server is situated in Korea and the DNS Server is situated in Hong Kong.**

[15]

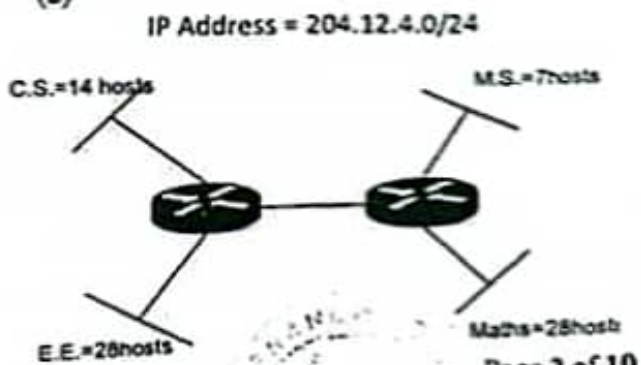
   a. User 2 from Japan wants to ping the www.google.com website. Do we need the DNS Server for the ping to be successful? (3)
   b. What has to be established before HTTP data can be sent or received from user 1 to the Web Server? (3)
   c. At the same time if both users (user 1 from Thailand and user 2 from Japan) types www.google.com on the browser of the respective computers. How does the server differentiate between the connections? (3)
   d. Will the communication be disrupted between the Web Server and the users if the DNS Server goes down during the data transfer between the web server and the user?(3)
   e. User 1 from Thailand opens two instances of www.google.com website on his computer. How does the Web Server differentiate between the communications? (3)

6. **Consider the network shown below:** [10]



   a. Assume that each node initially knows the costs to each of its neighbours. Use the distance vector algorithm and show the distance table entries at node 'F'. (5)
   b. Consider the operations of the Reverse Path Forwarding (RPF) algorithm for the given scenario. Using the same topology and least costs calculated in Part (a), find a set of paths from all nodes to the source node F (and include these paths in a graph using thicker-shaded lines) such that all other nodes will receive a copy of F's broadcast message from shortest path. (5)

7. **Use Variable-Length Subnet Masking (VLSM) to develop a subnetting scheme for the network scenario given below.** [10]
   a. Identify the total number of required subnets? (2)
   b. What is the subnet mask of each subnet? (3)
   c. List down the subnet ids and host address range of all the subnets. (5)



CamScanner

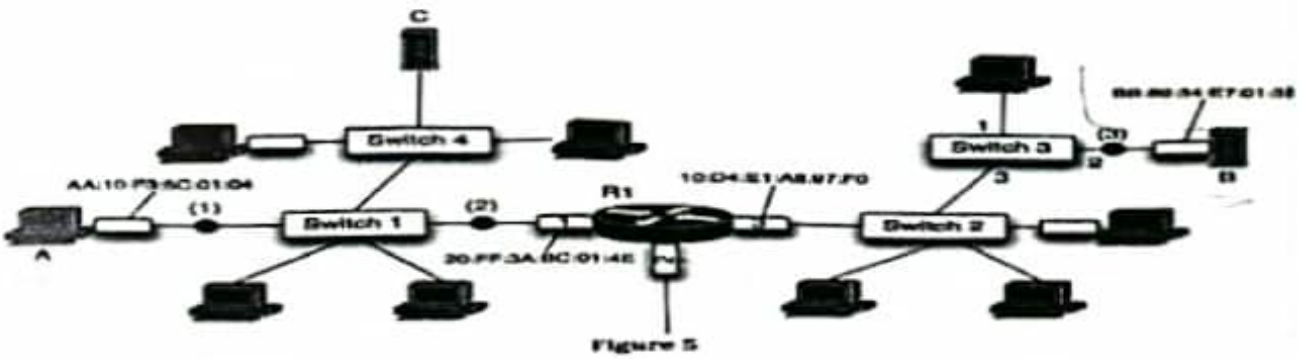
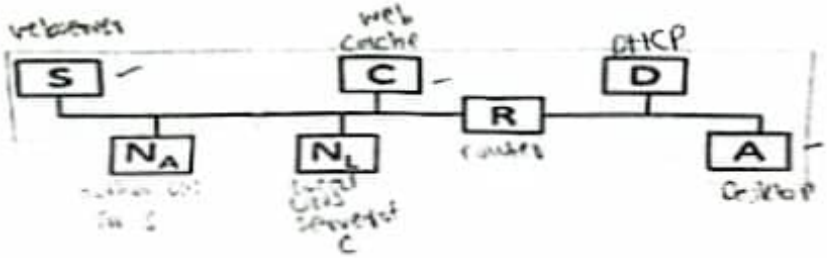


Figure 5

Q5.[04 pts]

In the topology shown below, machine A is a desktop client, D is a local DHCP server, R is A's gateway router, S is a Web server, and C is a Web cache. N_L is C's local nameserver, while N_A is the nameserver authoritative for S. Client A is configured to use Web cache C for all requests (assume that the Web cache resolves the name for any Web server and that the client is configured with the IP address of the cache). All wires/links are Ethernet segments. Assume the following:

- All the machines were just booted and their associated caches (ARP, DNS, Web, persistent connections) are all empty
- http://S/index.html fits in a single packet



- Persistent HTTP connections are used among A, C, and S (i.e. you should assume that once any connection between these hosts is established it is never closed)
  - Web caches respond to TCP requests that look like packet two in table 1 below (e.g., GET http://foo/bar/). They reply with the normal web cache contents.

i. (01 point)To connect machine A, it needs to get its own IP address, addr of first-hop router, addr of DNS server. Which protocol should be used among DNS, DHCP, ARP.

ii. (2 points) In which scenarios, TCP applies slow start, ~~Fehoe/Reno~~ and AIMD principles? Explain each with a proper example?

iii. (1 points) Why CSMA/CA is preferred over CSMA/CD for wireless channel?

Q6.[6 pts]

a. (1.5 pts) Assume we need to download text documents at the rate of 100 pages per second. What is the required bit rate of the channel? A page is an average of 24 lines with 80 characters in each line.

b. (1.5 pts) What is the minimum-width channel that could possibly transmit 80 Kbps in the presence of a signal-to-noise ratio of 15?

c. (1.5 pts) We need to send 265 kbps over a noiseless channel with a bandwidth of 30 kHz. How many signal levels do we need?

d. (1.5 pts) The period of a signal is 100 ms. What is its frequency in kilohertz?

CamScanner

COMSATS University Islamabad

Park Road, Tarlai Kalan, Islamabad 45550, Pakistan.
Department of Computer Science
Final Examination FALL-2021

Class/Section: BSCS, BSSE – 5th Semester
Subject: Computer Communications & Networks
Instructor: Dr. Farhana Jabeen, Dr. Ashfaq Farooqi, Dr. Muhammad Waqar, Dr. Fakhar, Zulfiqar Ali
Dated: Jan, 2022
Marks: 50
Time: Three hour (3 Hour)

1. The question paper consists of 9 questions. Attempt all questions.

CLO-01

Q1. [15 points]

a) [1.5 point] Briefly describe why the actual bandwidth on your cable access network connection on, e.g., a Saturday evening, could be significantly lower than what has been advertise by the provider (Comcast, Charter, etc.)? Would you see the same with a DSL-based provider?

b) [1.5 point] What is the purpose of the HTTP "COOKIE:" field? Are the values in the HTTP message's cookie field stored at the client or server or both? Explain briefly.

c) [1.5 point] Suppose that we want to change the IP address of gaia.cs.umass.edu from 128.119.40.186 to 128.119.40.187 and change this mapping in the DNS authoritative name server for gaia.cs.umass.edu. Once this mapping is changed in the authoritative name server, will all future references (generated anywhere in the Internet) to gaia.cs.umass.edu then be sent to 128.119.40.187? Explain briefly (in two or three sentences).

d) [03 point] Consider the scenario shown in Figure 2 in which a server is connected to a router by a 100Mbps link with a 50ms propagation delay. Initially this router is also connected to two routers, each over a 50Mbps link with a 200ms propagation delay. A 1Gbps link connects a host and a cache (if present) to each of these routers and we assume that this link has 0 propagation delay. All packets in the network are 20,000 bits long.

I. (02 Points) What is the end-to-end delay from when a packet is transmitted by the server to when it is received by the client? In this case, we assume there are no caches, there's no queuing delay at the routers, and the packet processing delays at routers and nodes are all 0.

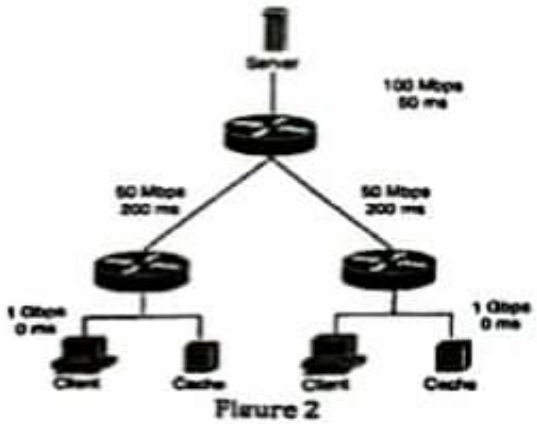


Figure 2

II. (01 Points) Here we assume that client hosts send requests for files directly to the server (caches are not used or off in this case). What is the maximum rate at which the server can deliver data to a single client if we assume no other clients are making requests?

e) [1.5 points] Consider a server-side socket that is used to communicate from server to client. In the case

CamScanner

CLO-04

Q7.[4 pts]

a. (2 pts) What is the hidden node problem in wireless networks? Explain how CSMA/CA is realized in the case of the IEEE 802.11 protocol.

b. (2 pts) Explain why 802.11 uses Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) rather than CSMA/CD. Be specific.

Q8.[4 pts]

a. (2 pts) Use the Caesar cipher to encrypt the message "JAVATPOINT," and the key (shift) value of this message is 3?

b. (2 pts) Alice has sent messages to BoB. These messages are in fact related to ordering an expensive book. Bob wants to be able to prove in court if necessary that Alice actually sent these messages to Bob containing the order for textbooks. What does Alice need to do/add to the messages, in order for Bob to be able to prove that Alice is the one who sent it originally?

Q9.[3 pts]

a. (1 pts). What are the possible multiple access schemes for cellular networks?

b. (2 pts). Explain the following terms related to cellular networks briefly:

i. *Handoff*

ii. Frequency Reuse

CamScanner

of a TCP socket, can data being read from the server-side socket have been sent by more than one client? Explain briefly. In the case of a UDP socket, can data being read from the server-side socket have been sent by more than one client? Explain briefly.

f) [1.5 points] Does the Internet checksum always detect errors in the transmitted segment?

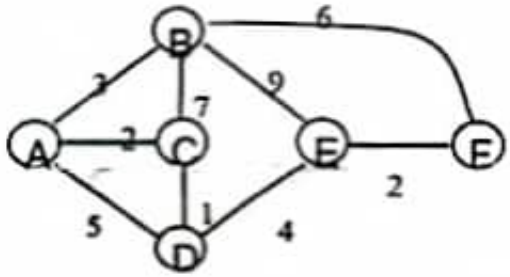
g) [1.5 points] Consider two TCP sessions that must share a link's bandwidth. One of the TCP connections has been running for quite some time and has built up a large TCP sending window. The second connection then starts up with an initially small window. Long term, what will be the relative throughput achieved by these two TCP sessions? Explain.

h) [1.5 points] Suppose nodes A, B, and C each attach to the same broadcast LAN (through their adapters). If A sends thousands of IP datagrams to B with each encapsulating frame addressed to the MAC address of B, will C's adapter process these frames? If so, will C's adapter pass the IP datagrams in these frames to the network layer in C? How would your answer change if A sends frames with the MAC broadcast address?

i) [1.5 points] Briefly describe how NAT works.

Q2. [07 points]

a. **(4 pts)** Consider the network shown below. Show the operation of Dijkstra's algorithm for computing the least cost path from A to all destinations. Also draw the minimum spanning tree from A to all destinations.



St ep	N'	D(B), p(B)	D(C) ,p(C)	D(D), p(D)	D(E), p(E)	D(F), p(F)
0						
1						
2						
3						
4						
5						

b. (4.5 points) Consider the three-node topology: the link costs are $c(x,y)=6$, $c(y,x)=2$, and $c(z,x)=50$. We focus here only on y's and z's distance table entries to destination x. Please explain the details of how the DV algorithm could reach a quiescent (or stable) state when the following cases happen.

i. (1.5 points) When c(x,y) decreases from 6 to 1.

ii. (1.5 points) When c(x,y) increases from 6 to 5.

Q3. [04 pts]

Suppose a receiver using the CRC generator polynomial $x^4 + x^2 + 1$ (10101) receives the sequence of bits 010010101101. Were they received correctly? If so, what was the transmitted message? If not, can you tell how many bit errors occurred? Show your work.

Q4.[03 pts]

a) **(1.5 pts)** In Figure 5, A is sending an ARP request to determine the MAC address it should use for a frame destined to B. What MAC address will be returned to A in the ARP reply. Briefly explain your answer.

b) **(1.5 pts)** How many interfaces of all the ones shown in Figure 5 receive the original ARP request sent by A and what is the destination MAC address on the frame containing the ARP request?

CamScanner

COMSATS University Islamabad

Department of Computer Science

Mid-Term Examination BS(CS) SPRING-2023

Instructor: Dr. Mariam Akbar, Dr. Sheneela Naz, Dr. Zulfiqar Ali, Dr. Abdul Rehman Salehi

Course Title:	CSC340 Data Computer Networks	Maximum Marks:	50
Dated:	2023	Time Allowed:	80 Min

Name:_____ Reg no._____

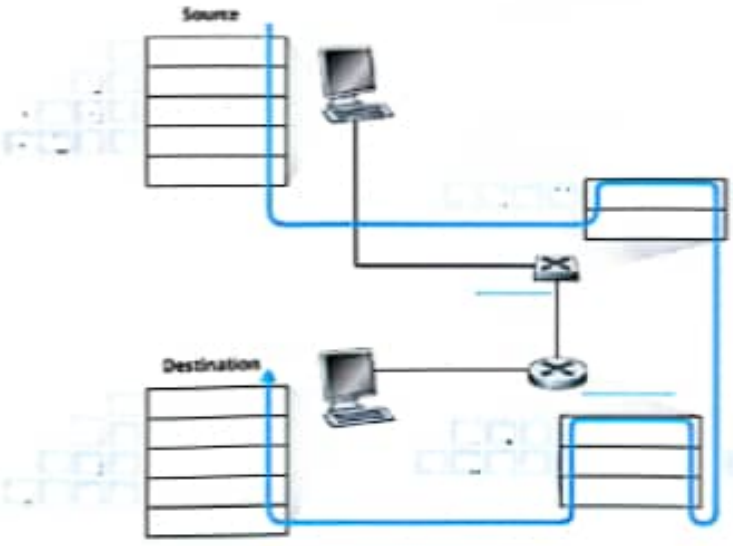
CLO-1

(Summarize the fundamental concepts of computer networks)

Q (1) = 15 marks total

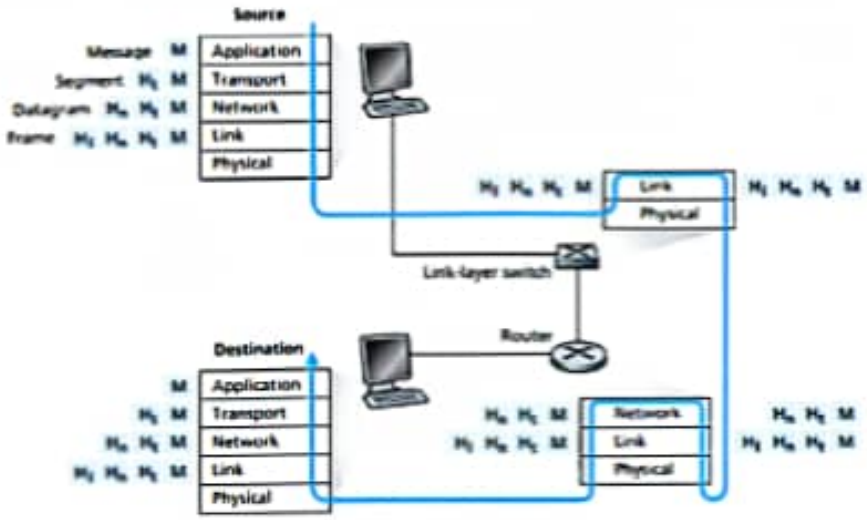
Q (1) Part A (marks=1+2+3=6)

The following Figure shows Hosts, routers, and switches; each contains a different set of layers, reflecting their differences in functionality. Using knowledge of protocol stack, label the figure and message transformation at different layer throughout the transmission path.



Solution:

CamScanner



Hosts, routers, and link-layer switches; each contains a different set of layers, reflecting their differences in functionality

Q (1) Part B (marks=3+3=6)

Consider the car-caravan analogy and assume a propagation speed of 100 km/hour.

a. Suppose the caravan travels 175 km, beginning in front of one tollbooth, passing through a second tollbooth, and finishing just after a third tollbooth. What is the end-to-end delay?

Given:

Distance traveled (D) = 175 km

Propagation speed (S) = 100 km/hour

Propagation Delay=175 km100 km/hourPropagation Delay=100 km/hour175 k
Propagation Delay=1.75 hours

b. Repeat (a), now assuming that there are eight cars in the caravan instead of ten.

As discussed in part (a), the propagation delay is 1.75 hours.

The number of cars in the caravan doesn't affects this calculation. The end-to-end delay is still the propagation delay because we are assuming negligible transmission, queuing, and processing delays.

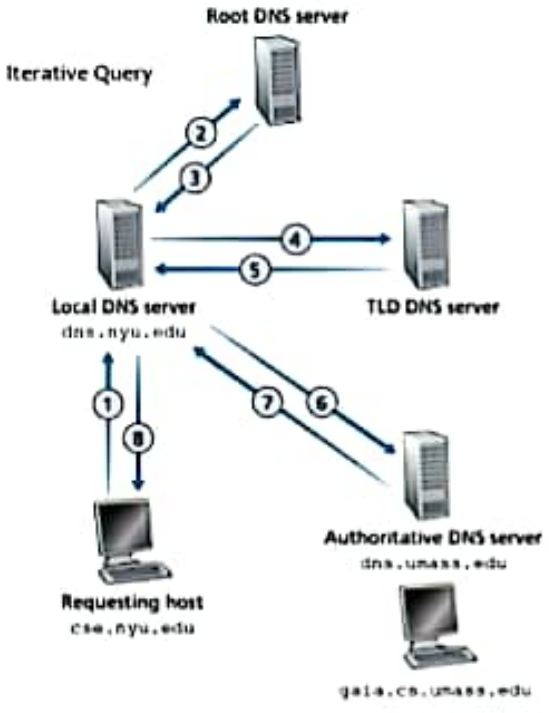
End-to-End Delay=1.75 hours

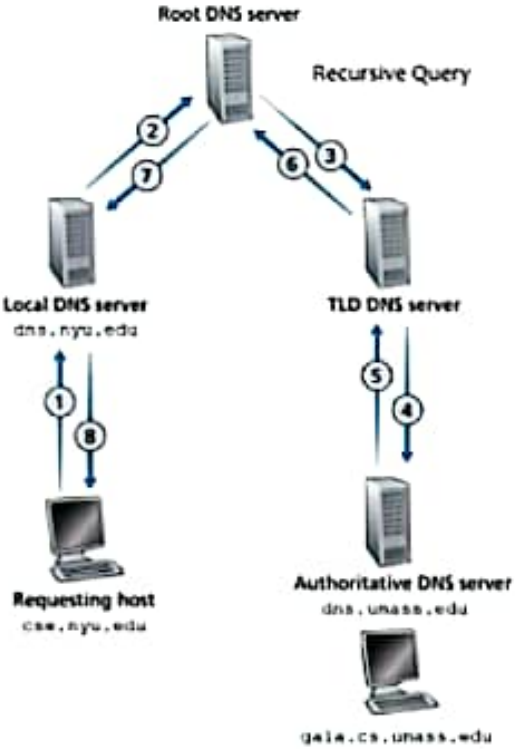
So, even if there are eight cars instead of ten in the caravan, the end-to-end delay remains 1.75 hours.

Q (1) Part C (marks=3)

CamScanner

Suppose a client cis.poly.edu wants to open www.csmit.edu on his web browser for the first time. For the given figure, draw the path showing the sequence of DNS queries and replies for both the recursive and iterative methods. Moreover, explain which query will be more suitable iterative or recursive. Why?





CLO-2

CamScanner

(Apply the concepts of computer networks to solve their related problems)

Q (2) = 16 marks total

Q (2) Part B (2+2+3=7)

1. If a channel operates with a bit rate of 4 Kbps and a one-way propagation delay of 20 msec, while employing the stop-and-wait protocol and assuming negligible transmission time for acknowledgment frames, what is the minimum frame size in bits required to achieve a channel efficiency of at least 50%.

2. In a Satellite Communication system with an average source-to-destination distance of 15,000 kilometers and a one-way delay of 250 milliseconds, what data rate would be required for the round-trip delay to be equal to the transmission delay for a 1 KB packet?

Solution-

Given-

- Distance = 15,000 Km

Propagation delay (T_p) = 250 milliseconds

-
- Packet size = 1 KB

We need to have-

Round Trip Time = Transmission delay

2 x Propagation delay = Transmission delay

2 x 250 milliseconds = 1 KB / Bandwidth

Bandwidth = 1 KB / 500 milliseconds

Bandwidth = 16000 bps

3. Consider a real-time scenario where you're streaming high-definition video content over a long-distance network connection. The video content is transmitted at a rate of 10 Mbps, and it needs to travel over a network link that spans 2,000 kilometers with a propagation speed of 2 x 10^8 meters per second.

$$T_{prop} = \frac{D}{S}$$

Where:

- T_{prop} is the Time of Propagation.
- D is the distance the signal must travel.
- S is the speed of signal propagation.

CamScanner

Let's calculate it:

$$T_{prop} = \frac{2000 \times 10^3\ m}{2 \times 10^8\ m/s}$$
$$T_{prop} = \frac{2000 \times 10^3}{2 \times 10^8}$$
$$T_{prop} = \frac{2000}{200}$$
$$T_{prop} = 10 \times 10^{-3}\ s$$
$$T_{prop} = 0.01\ s$$
$$T_{prop} = 10\ ms$$

So, the Time of Propagation for the signal to travel over a network link of 2000 kilometers at a propagation speed of 2×10^8 meters per second is 10 milliseconds (ms).

b. Explain the Delay Dependencies

Propagation Delay: This is dependent on the physical length of the link and the propagation speed of the signal in the medium (e.g., copper wire, fiber optic, wireless). In this scenario, the propagation delay is the time it takes for a bit to travel from the source to the destination.

Transmission Delay: This is the time it takes to push all the bits of a packet into the link. Transmission delay depends on the packet's length and the data rate of the link. In this scenario, transmission delay is not relevant to the calculation, but it is a factor in the overall delay.

Queueing Delay: This occurs when packets are held in a queue due to a busy link. Queueing delay varies dynamically with the network's state, depending on the router's queue length and the traffic intensity.

Processing Delay: This is the time routers take to process the packet header, check for bit-level errors in the packet, and determine the packet's outbound link. Processing delay is typically very small compared to other delays.

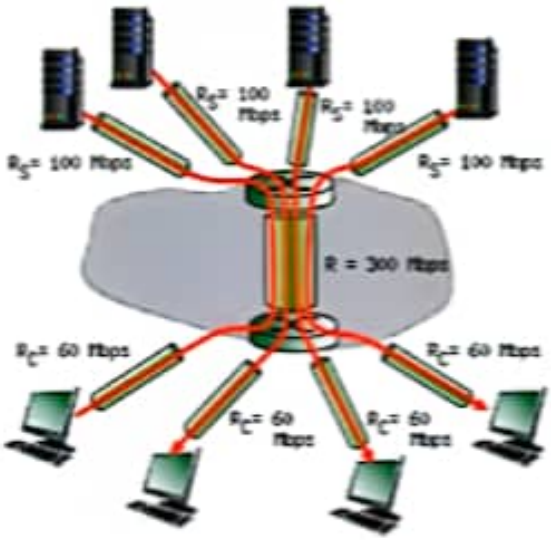
Jitter: This is the variation in packet arrival times and is critical for video content. Jitter can be caused by queueing delays, processing delays, and variations in propagation delays.

Q (2) Part A (marks=1+1=2)

CamScanner

Consider the scenario shown below, with four different servers connected to four different clients over four three-hop paths.

1. What is the maximum achievable end-end throughput (in Mbps) for each of the four client-to-server pairs, assuming that the middle link is fair-shared?
2. Which link is the bottleneck link for each session?



Solution:

a) Throughput=min(100, 300/4, 60)=min(100, 75,60)= 60mbps

b) Rc

5

Q (2) Part C (3)

Suppose that the five measured SampleRTT values are 106 ms, 120 ms. Compute the EstimatedRTT after each of these SampleRTT values is obtained, using a value of α = 0.125 and assuming that the value of EstimatedRTT was 100 ms just before the first of these five samples were obtained. Compute also the DevRTT after each sample is obtained, assuming a value of β = 0.25 and assuming the value of DevRTT was 5 ms just before the first of these five samples was obtained. Last, compute the TCP TimeoutInterval after each of these samples is obtained.

Calculate the EstimatedRTT after obtaining the first sample RTT=106*ms*,

*EstimatedRTT = α * SampleRTT+(1- α) * EstimatedRTT*

EstimatedRTT =0.125 * 106 + (1-0.125) * 100

=0.125* 106 + 0.875 * 100

=13.25 + 87.5

=100.75*ms*

CamScanner

Calculate the DevRTT after obtaining the first sample RTT:

DevRTT = β * | *SampleRTT- EstimatedRTT*|+(1- β)* *DevRTT*

=0.25 * |106-100.75| + (1-0.25) *5

=0.25 *5.25 + 0.75 * 5

=1.3125 + 3.75

=5.0625*ms*

Calculate the Timeout Interval after obtaining the first sample RTT:

TimeoutInterval = *EstimatedRTT* +4* *DevRTT*

= 100.75 + 4 *5.0625

=121*ms*

Calculate the EstimatedRTT after obtaining the second sample RTT=120*ms*,

EstimatedRTT = α * *SampleRTT*+(1- α) * *EstimatedRTT*

EstimatedRTT =0.125 * 120 + (1-0.125) * 100.75

=0.125* 120 + 0.875 * 100.75

=15 + 88.15625

=103.15625*ms*

Calculate the DevRTT after obtaining the second sample RTT:

DevRTT = β * | *SampleRTT- EstimatedRTT*|+(1- β)* *DevRTT*

=0.25 * |120-103.15625| + (1-0.25) *5.0625

=0.25 *16.84 + 0.75 * 5.0625

=4.21 + 3.79

=8*ms*

Calculate the Timeout Interval after obtaining the second sample RTT:

TimeoutInterval = *EstimatedRTT* +4* *DevRTT*

= 103.15 + 4 *8

=135.15*ms*

Calculate the EstimatedRTT after obtaining the third sample RTT=140*ms*:

EstimatedRTT = α * *SampleRTT*+(1- α) * *EstimatedRTT*

EstimatedRTT =0.125 * 140 + (1-0.125) * 103.15

=0.125* 140 + 0.875 * 103.15

=17.5 +90.26

=107.75*ms*

Calculate the DevRTT after obtaining the third sample RTT:

DevRTT = β * | *SampleRTT- EstimatedRTT*|+(1- β)* *DevRTT*

=0.25 * |140-107.75| + (1-0.25) *8

=0.25 *32.25 + 0.75 * 8

=8.06 + 6

=14.06*ms*

CamScanner

Calculate the Timeout Interval after obtaining the third sample RTT:

TimeoutInterval = EstimatedRTT +4 DevRTT*

= 107.75 + 4 *14.06

=164*ms*

Calculate the EstimatedRTT after obtaining the fourth sample RTT=90*ms*:

*EstimatedRTT = α * SampleRTT+(1- α) * EstimatedRTT*

EstimatedRTT =0.125 * 90 + (1-0.125) * 107.75

=0.125* 90 + 0.875 * 107.75

=11.25 +94.28

=105.53*ms*

Calculate the DevRTT after obtaining the fourth sample RTT:

*DevRTT = β * | SampleRTT- EstimatedRTT|+(1- β)* DevRTT*

=0.25 * |90-105.53| + (1-0.25) *14.06

=0.25 *15.53 + 0.75 * 14.06

=3.88 + 10.545

=14.42*ms*

Calculate the Timeout Interval after obtaining the fourth sample RTT:

TimeoutInterval = EstimatedRTT +4 DevRTT*

= 105.53 + 4 *14.42

=163.21*ms*

Calculate the EstimatedRTT after obtaining the fifth sample RTT=115*ms*:

*EstimatedRTT = α * SampleRTT+(1- α) * EstimatedRTT*

EstimatedRTT =0.125 * 115 + (1-0.125) * 105.53

=0.125* 115 + 0.875 * 105.53

=14.375 +92.34

=106.715*ms*

Calculate the DevRTT after obtaining the fifth sample RTT:

*DevRTT = β * | SampleRTT- EstimatedRTT|+(1- β)* DevRTT*

=0.25 * |115-106.715| + (1-0.25) *14.42

=0.25 *8.285 + 0.75 * 14.42

=2.07 + 10.815

=12.885*ms*

Calculate the Timeout Interval after obtaining the fifth sample RTT:

TimeoutInterval = EstimatedRTT +4 DevRTT*

= 106.715 + 4 *12.885

=158.255*ms*

CamScanner

Q (2) Part D (4)

Assign IP addresses for your new MAN (Metropolitan Area Network), which consists of 8 different buildings, each building will have 255 workstations. Your administrator tells you to only use as much of the 164.10.0.0 network as you need. Your administrator will assign the IP addresses to the serial interfaces using a different network. You will need to determine the following four items for each of the eight buildings:

(a) Subnet masks
(b) Network addresses
(c) Broadcast address for each subnet
(d) Valid host ranges on each subnet

Solution:

To assign IP addresses for the new MAN, we need to first determine the number of bits required for the network and host portions of the address. Since each building has 255 workstations, we need at least 8 bits to represent the host portion (2^8 - 2 = 254, where 2 is subtracted for the network and broadcast addresses). We can allocate the remaining bits to the network portion.

Since we are only allowed to use as much of the 164.10.0.0 network as we need, we can use the Class B network address 164.10.0.0/16. We can further divide this network into eight subnets, one for each building, using the following subnet mask:

(a) Subnet masks: 255.255.255.0 (/24)

Each building will have a subnet mask of 255.255.255.0, which allows for 256 IP addresses per subnet, with 254 usable for hosts.

(b) Network addresses:

Building 1: 164.10.1.0

Building 2: 164.10.2.0

Building 3: 164.10.3.0

Building 4: 164.10.4.0

Building 5: 164.10.5.0

Building 6: 164.10.6.0

Building 7: 164.10.7.0

Building 8: 164.10.8.0

Each building will have a unique network address, obtained by setting the host portion of the address to 0.

(c) Broadcast addresses:

CamScanner

Building 1: 164.10.1.255

Building 2: 164.10.2.255

Building 3: 164.10.3.255

Building 4: 164.10.4.255

Building 5: 164.10.5.255

Building 6: 164.10.6.255

Building 7: 164.10.7.255

Building 8: 164.10.8.255

The broadcast address for each subnet is obtained by setting the host portion of the address to all 1s.

(d) Valid host ranges:

Building 1: 164.10.1.1 to 164.10.1.254

Building 2: 164.10.2.1 to 164.10.2.254

Building 3: 164.10.3.1 to 164.10.3.254

Building 4: 164.10.4.1 to 164.10.4.254

Building 5: 164.10.5.1 to 164.10.5.254

Building 6: 164.10.6.1 to 164.10.6.254

Building 7: 164.10.7.1 to 164.10.7.254

Building 8: 164.10.8.1 to 164.10.8.254

The valid host range for each subnet is obtained by setting the first and last addresses of the subnet for the network and broadcast addresses, respectively.

CLO-3

(Analyze various protocols, algorithms, and their functions in a network)

Q (3) = 19 marks total

Q (3) Part A (marks=10)

CamScanner

What will be the output if checksum algorithm is applied on given data: 1110011001100110 and 1101010101010101?

0100 0100 0100 0011 (Check sum calculated)

Q (3) Part B (9)

1. Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how?
   Yes, it is possible for an application to achieve reliable data transfer even when it runs over the User Datagram Protocol (UDP), which is traditionally considered an unreliable, connectionless transport protocol. The reliability would need to be implemented at the application layer since UDP itself does not provide mechanisms for reliable data delivery, error recovery, or flow control.
2. 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 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?

In a UDP-based communication model, the destination port number is used to direct incoming UDP segments to the appropriate process or socket on the receiving host. In this scenario, both Host A and Host B are sending UDP segments to Host C with the same destination port number, 6789.

Yes, both UDP segments will be directed to the same UDP socket at Host C because the destination port number is what identifies the receiving socket. The operating system at Host C uses the destination port number to determine which process or application should receive the incoming UDP segments.

Now, to distinguish between the UDP segments originated from Host A and Host B, the information about the source IP address and source port number is used. Each UDP segment contains source and destination port numbers, as well as source and destination IP addresses.

The combination of source IP address and source port number uniquely identifies the sender of each UDP segment. Therefore, even though both UDP segments are destined for the same port (6789) on Host C, the process at Host C can examine the source IP address and source port number in each received UDP segment to differentiate between the segments sent by Host A and Host B.

In summary, both UDP segments will be directed to the same socket at Host C based on the destination port number. The process at Host C can distinguish between the segments from Host A and Host B by examining the source IP address and source port number in each UDP segment.

CamScanner

COMSATS University Islamabad (CUI)

Department of Computer Science
Terminal Examination Fall - 2022
BSE-5A/B,BCS-5A/B,BDS-4A,BAI-4A SEMESTER

Course: CSC339 Data Comm. & Computer Networks
Instructor: Dr. Mariam Akbar, Dr. Sheneela Naz, Mr. Saleem Raza, Mr. Zulfikar
Maximum Marks: 50 **Dated:** January 21, 2023 **Time Allowed:** 180 Minutes

- *All questions are self-explanatory and require no further explanations during exam time.*
- *All questions must be answered at the question paper (in space provided)*
- *Make sure that you have signed the attendance sheet before leaving the examination room.*
- *Attempt all questions.*

(CLO – 1: Summarize the fundamental concepts of computer networks)

Q.# 1. Consider the scenario shown in Figure in which a server is connected to a router by a 100Mbps link with a 50ms propagation delay. Initially, this router is also connected to two routers, each over a 50Mbps link with a 200ms propagation delay. A 1Gbps link connects a host and a cache (if present) to each of these routers and we assume that this link has 0 propagation delay. All packets in the network are 20,000 bits long. **[10 marks: 2+2+2+2+2]**

1.1) What is the end---to---end delay from when a packet is transmitted by the server to when it is received by the client? In this case, we assume there are no caches, there's no queuing delay at the routers, and the packet processing delays at routers and nodes are all 0.

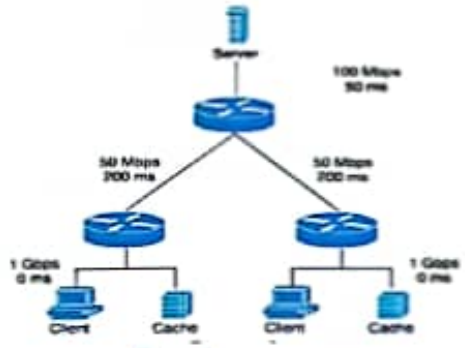


Figure 1

1.2) Here we assume that client hosts send requests for files directly to the server (caches are not used or off in this case). What is the maximum rate at which the server can deliver data to a single client if we assume no other clients are making requests?

CamScanner

1.3) Suppose two hosts, A, and B, are separated by 20,000 kilometers and are connected by a direct link of R = 2 Mbps. Suppose the propagation speed over the link is 2.5×10^8 meters/sec. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one large message. What is the maximum number of bits that will be in the link at any given time?

1.4) Consider a server-side socket that is used to communicate from server to client. In the case of a TCP socket, can data be read from the server-side socket have been sent by more than one client? Explain briefly. In the case of a UDP socket, can data be read from the server-side socket have been sent by more than one client? Explain briefly

1.5) Does the Internet checksum always detect errors in the transmitted segment?

1.6) [Through NAT a private network can communicate with the public network. In the following figure write the source and destination IP addresses in the blanks.

CamScanner

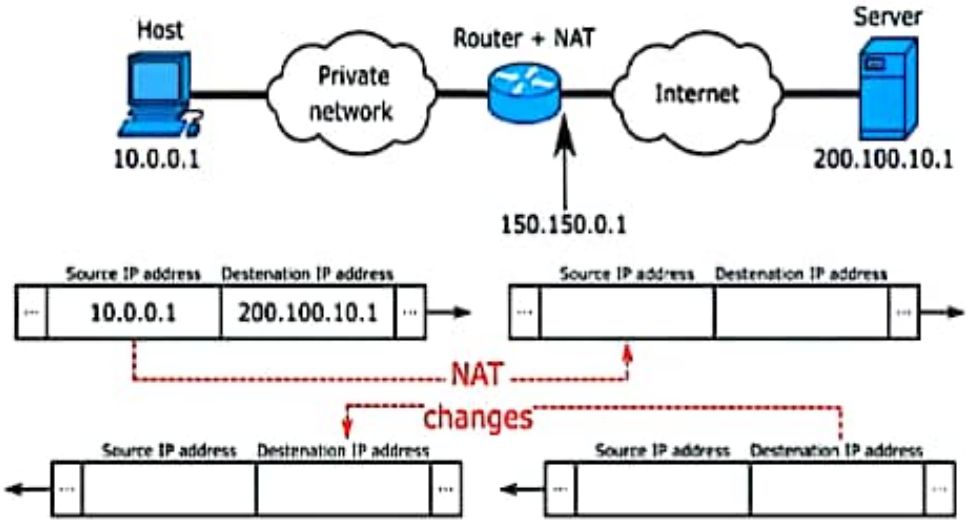


Figure 2

(CLO – 2: Apply the concepts of computer networks to solve its related problems)

Q.# 2. Calculate and sketch (where asked) **[10marks:2+2+2+2+2]**

2.1) The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with –0.3 dB/km has a power of 5 mW, what is the power of the signal at 5 km?

2.2) A slotted ALOHA network transmits 200-bit frames using a shared channel with a 200-kbps bandwidth. Find the throughput if the system (all stations together) produces

a. 1000 frames per second
b. 500 frames per second
c. 250 frames per second

CamScanner

3

2.3) A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal.

2.4) We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

2.5) What are the propagation time and the transmission time for a 5-MB (megabyte) message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

(CLO – 3: Analyze various protocols, algorithms and there functions in a network)

Q.# 3. Complete the following scenarios and answer relevant questions. **[14marks:5+3+3+2+2]**

CamScanner

3.1) Let's assume that a user connects a laptop to his university's Ethernet switch and downloads a Web page (say the home page of www.google.com).As we now know, there's a lot going on "under the hood" to satisfy this seemingly simple request.

Briefly list down the protocols involved along with their function in each step / layer till the web page appears on user's screen.

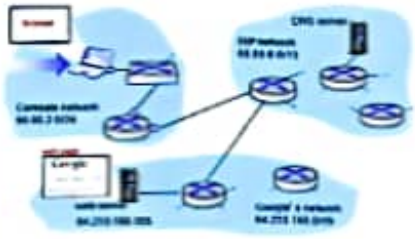
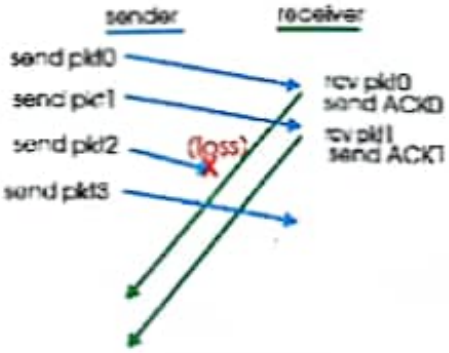
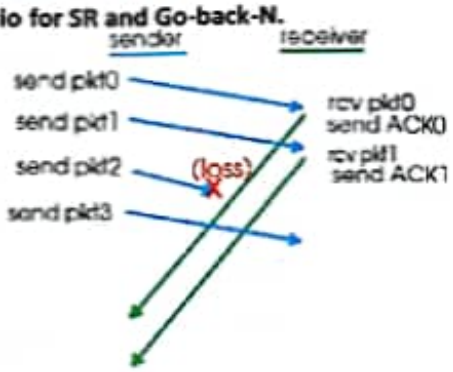


Figure 3

3.2) Complete the following scenario for SR and Go-back-N.





4

Go-Back-N in operation

Selective Repeat

Figure 4

3.3) Consider the network shown below in **Figure 5.** Show the operation of Dijkstra's

CamScanner

algorithm for computing the least cost path from A to all destinations. Also, draw the minimum spanning tree from A to all destinations.

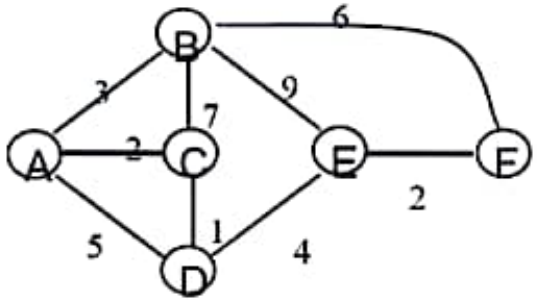


Figure 5

3.4) A network using *CSMA/CD* has a bandwidth of 10 Mbps. If the maximum propagation time (including the delays in the devices and ignoring the time needed to send a jamming signal is 25.611S, what is the minimum size of the frame?

3.5) Consider the network shown in **Figure 6**. Suppose AS3 and AS2 are running OSPF for their intra-AS routing protocol. Suppose AS1 and AS4 are running RIP for their intraAS routing protocol. Suppose eBGP and iBGP are used for the Inter-AS routing protocol. Initially suppose there is no physical link between AS2 and AS4.

3.5.1) Router 3c learns about prefix x from which routing protocol: OSPF, RIP, eBGP, or iBGP?

5

3.5.2) Router 3a learns about x from which routing protocol?

CamScanner

3.5.3) Router 1c learns about x from which routing protocol?

3.5.4) Router 1d learns about x from which routing protocol?

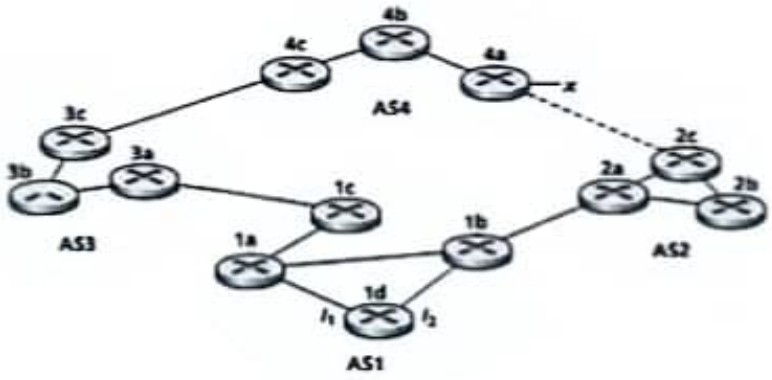


Figure 6

(CLO – 4: Comprehend the fundamentals of wireless and mobile networks along with the security issues)

Q.# 4. Answer the followings. **[16marks:2+2+2+2+1+2+2+3]**

4.1) Let's you want to share a file confidentially with your friend via a symmetric key cryptosystem. How Public Key cryptosystem can be used to securely transmit the key along with determining the integrity of the file.

4.2) In what way does a hash provide a better message integrity check than a checksum (such as the Internet checksum)?

6

CamScanner

4.3) It is claimed that without Authentication, Message Integrity becomes useless and vice versa. Do you agree with it? Why /why not?

4.4) To enable secure communication, the website presents a certificate mentioning its public key. Is it secure enough? Why / why not?

4.5) What does "triangle routing" mean in mobile and wireless networks?

4.6) Suppose the IEEE 802.11 RTS and CTS frames were as long as the standard DATA and ACK frames. Would there be any advantage to using the CTS and RTS frames? Why or why not?

4.7) As a mobile node gets farther and farther away from a base station, what are two actions that a base station could take to ensure that the loss probability of a transmitted frame does not increase?

4.8) Differentiate between

CamScanner

4.8.1) Path loss vs. Multipath propagation

4.8.2) Active vs. passive scanning

4.8.3) Visited vs. home networks

•••••BEST OF LUCK•••••

CamScanner

COMSATS University Islamabad

Department of Computer Science

Mid-Term Examination BS(CS) SPRING-2024

Instructor: Dr. Mariam Akbar, Dr. Zulfiqar Ali, Dr. AbdulRahman Al-Salehi, Mr. Shakeel

Course Title:	*CSC340 Computer Networks*	**Maximum Marks:**	50
Dated:	17-04-2024	**Time Allowed:**	90 Min

Name: _____ ***Reg no.*** _____

----------CLO-1----------

--(Summarize the fundamental concepts of computer networks)

Q (1) = (6+6+3=15 Marks)

Part A.

There is a user of Facebook who wants connectivity to the nearest Facebook server which has15 routers on the way. The entire length of optical fiber cable is 6543 Km between two host entities. The average processing delay at each router is 2 msec and there is no congestion on the link, The propagation delay is subject to refractive index of optical fiber equal to 1.5, The speed of light is universal constant of 3x10^8 m/sec. Calculate the end-to-end delay of data in one direction. Ignore transmission delays.

Part B.

Consider the car-caravan analogy and assume a propagation speed of 100 km/hour.

1. Suppose the caravan travels 175 km, beginning in front of one tollbooth, passing through a second tollbooth and finishing just after a third tollbooth. What is the end-to-end delay?
2. Repeat (a), now assuming that there are eight cars in the caravan instead of ten.

Part C.

In wireless system, user can be located anywhere in coverage area and the behavior of system is to reduce the user bandwidth to half after going away from tower for each 1 Km due to signa weakening. The bandwidth at the tower is 10 Mbits per second. How much time is required t transfer 1 GByte data at 2.9 Km location from tower.

----------CLO-2----------

sing a different network. You will need to determine the following four items for each of the eight buildings:

1. Subnet masks
2. Network addresses.
3. Broadcast address for each subnet.
4. Valid host ranges on each subnet

CLO-3

(Analyze various protocols, algorithms, and their functions in a network)

Q (3) = (6+ 5+ 4= 15 Marks)

Part A.

Consider sending a 2600-byte datagram into a link that has an MTU of 700 bytes. Suppose the original datagram is stamped with the identification number 422. How many fragments are generated? What are the values in the various fields in the IP datagram(s) generated related to fragmentation?

Length	Id	More Fragment Flag	Offset
700		1	
700		1	
		0	
160		1	

Part B.

Host A and B are directly connected with a 100 Mbps link. There is one TCP connection between the two hosts, and Host A is sending to Host B an enormous file over this connection. Host A can send its application data into its TCP socket at a rate as high as 120 Mbps but Host B can read out of its TCP receive buffer at a maximum rate of 50 Mbps. Describe the effect of TCP flow control.

Part C.

Compare GBN, SR, and TCP (no delayed ACK). Assume that the timeout values for all three protocols are sufficiently long such that 5 consecutive data segments and their corresponding ACKs can be received (if not lost in the channel) by the receiving host (Host B) and the sending host (Host A) respectively. Suppose Host A sends 5 data segments to Host B, and the 2nd segment (sent from A) is lost. In the end, all 5 data segments have been correctly received by Host B.

1. How many segments has Host A sent in total and how many ACKs has Host B sent in total? What are their sequence numbers? Answer this question for all three protocols.
2. If the timeout values for all three protocols are much longer than 5 RTT, then which protocol successfully delivers all five data segments in the shortest time interval?

CamScanner

(Apply the concepts of computer networks to solve their related problems)

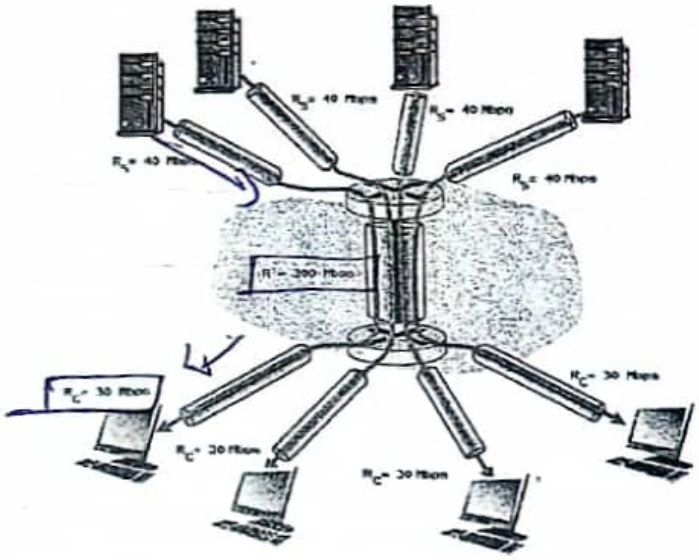
Q (2) = (7+7+6= 20 Marks)

Part A.

Consider the scenario shown below, with four different servers connected to four different clients over four three-hop paths. The four pairs share a common middle hop with a transmission capacity of R = 300 Mbps. The four links from the servers to the shared link have a transmission capacity of RS = 40 Mbps. Each of the four links from the shared middle link to a client has a transmission capacity of RC = 30 Mbps.

1. What is the maximum achievable end-end throughput (in Mbps) for each of the four client-to-server pairs, assuming that the middle link is fairly shared?
2. Which link is the bottleneck link for each session?

$\min(Rs, Rc, \frac{R}{10})$



Part B.

Suppose that the five measured SampleRTT values are 106 ms, 120 ms. Compute the EstimatedRTT after each of these SampleRTT values is obtained, using a value of $\alpha = 0.125$ and assuming that the value of EstimatedRTT was 100 ms just before the first of these five samples were obtained. Also compute the DevRTT after each sample is obtained, assuming a value of $\beta = 0.25$ and assuming the value of DevRTT was 5 ms just before the first of these five samples was obtained. Last, compute the TCP TimeoutInterval after each of these samples is obtained.

Part C.

Assign IP addresses for your new MAN (Metropolitan Area Network), which consists of 8 different buildings, each building will have 255 workstations. Your administrator tells you to only use as much of the 164.10.0.0 network as you need. Your administrator will assign the IP addresses to the serial interfaces