National University of Computer and Emerging Sciences, Lahore Campus

Course Name:	Computer Networks	Course Code:	CS307
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Section:	ALL	Page(s):	10
Exam Type:	Final		

Student : Name:_________________________ Roll No.______________ Section:______

Instruction/Notes: Attempt questions on this paper. You may use rough sheet but it should not be attached to this paper as it will not be marked.

1. Multiple choice questions. Please write your answer/section in the table given below. Answers outside the table will not be considered. [10 Marks]

1.1	(a)	1.6	(a)
1.2	(a)	1.7	(c)
1.3	(a)	1.8	(b)
1.4	(b)	1.9	(d)
1.5	(a)	1.10	(c)

1.1 ICMP is primarily used for
a) error and diagnostic functions
b) addressing
c) forwarding
d) none of the mentioned

1.2 User datagram protocol is called connectionless because
a) all UDP packets are treated independently by transport layer
b) it sends data as a stream of related packets
c) it is received in the same order as sent order
d) none of the mentioned

1.3 An endpoint of an inter-process communication flow across a computer network is called
a) socket
b) pipe
c) port
d) none of the mentioned

1.4 Transport layer protocols deals with
a) application to application communication
b) process to process communication
c) node to node communication
d) none of the mentioned

1.5 For a 10Mbps Ethernet link, if the length of the packet is 32bits, the transmission delay is (in milliseconds)
a) 3.2
b) 32
c) 0.32
d) None of the mentioned

1.6 The time required to examine the packet's header and determine where to direct the packet is part of
a) Processing delay
b) Queuing delay
c) Transmission delay
d) All of the mentioned

1.7 The computation of the shortest path in OSPF is usually done by
a) Bellman-ford algorithm
b) Routing information protocol
c) Dijkstra's algorithm
d) Distance vector routing

1.8 Correct order of the operations of OSPF
1. Hello packets
2. Propagation of link-state information and building of routing tables
3. Establishing adjacencies and synchronisation database

a) 1-2-3
b) 1-3-2
c) 3-2-1
d) 2-1-3

1.9 Which of this is not a class of IP address?
a) ClassE
b) ClassC
c) ClassD
d) ClassF

1.10 The TTL field in an IP datagram/packet has a value of 10. How many routers (max) can process this datagram?
a) 11
b) 5
c) 10
d) 1

2. Consider the network scenario given in Figure 1. Assume that we know the bottleneck link along the path from the server to the client is the first link with rate **$R_s$bits/sec**. Suppose we send a pair of packets back to back from the server to the client, where client is connected to the router with **R_c bits/sec** and there is no other traffic on this path. Assume each packet of size **L bits**, and both links have the same propagation delay given by **d_{prop}.** **[1+2+2=5 Marks]**

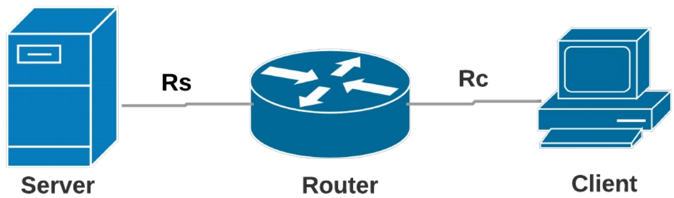

Figure. 1

2.1 What is the packet inter-arrival time at the destination (client)? Meaning how much time elapses from when the last bit of the first packet arrives until the last bit of the second packet arrives?

- *L/R_s*

2.2 Now assume a change in the above scenario where instead of first the second link is the bottleneck (i.e. R_c<R_s). Would it be possible that the second packet gets queued at the input queue of the second link? Explain.

- *Yes.*
- *Reason: Explanation for inequality $L/Rs + L/Rs < L/Rs + L/Rc$ is acceptable solution*

2.3 Now suppose that the server sends the second packet T seconds after sending the first packet. How large must T be to ensure no queuing before the second link? Explain. [Use the same bottleneck as 2.2 i.e. R_c <R_s.]

- *$T = L/Rc - L/Rs$*

3. Suppose the current TCP window size is 28000 bytes and the maximum segment (packet) size is 1400 bytes. The congestion window is maintained in bytes. In slow start, each incoming new ACK increments the congestion window by the maximum segment size. Suppose the sender previously received cumulative ACK 135801 (i.e., the receiver says it received all bytes sent to it, up to and including, sequence number 135800). The sender next receives cumulative ACK 140001. Assuming the sender is in slow start and has enough data in its transmit socket buffer, calculate the number of bytes it now sends. Show all of your calculations clearly. **[5 Marks]**

Bytes the sender can send: __28,000 + 1400 + 1400____________

Calculations:

4. Assume a general network scenario comprising a client, a server and three routers between them. Ignoring fragmentation, an IP datagram is sent from the client to the server. Answer the following short questions. **[1+2+2=5 Marks]**

4.1 How many interfaces in total will this IP datagram traverse (i.e. the datagram will travel through how many interfaces)?

- 8

4.2 How many forwarding tables will be indexed to successfully move the datagram from the client to the server?

- 3

4.3 Suppose TCP segments are being encapsulated in the IP datagrams being sent between the client and the server. When the server receives the datagram, how does the network layer on the server side know which upper layer protocol it should pass the segment to. How does the server-side determine that the payload of the datagram should be given to TCP rather than UDP or (something else)?

- Upper layer protocol

5. Consider the network scenario given in Figure 2 comprising four end-systems/hosts: A, B, C and D with MAC addresses: MAC-A, MAC-B, MAC-C, MAC-D and IP addresses: IP-A, IP-B, IP-C, IP-D, respectively. DHCP is enabled for the given network topology. S1 and S3 denote the switches while routers are represented by R1 and R3. Connecting lines depict the network connections (wired) between all the network nodes with individual digits on each end representing the interface number of the connected node.
Furthermore, for IP and MAC addresses of the interfaces, IP-(Interface number) and MAC-(Interface number) format is in use. For example, IP-9 and MAC-9 refers to the IP and MAC addresses of Router1's interface 9.

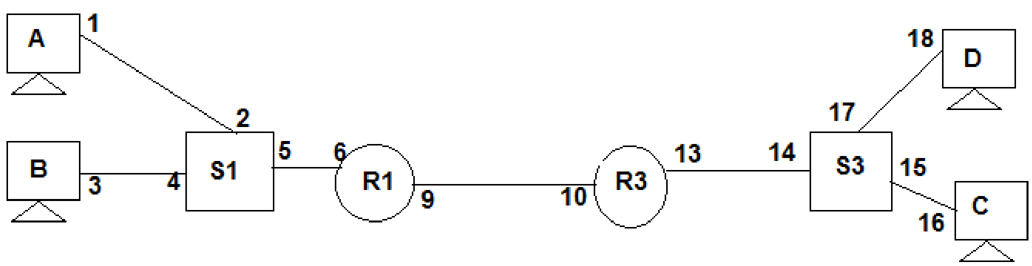

Figure. 2

Consider the following ARP table for interface 6 of the router R1 and specify in the entries given are correct or not. For each entry, please give a brief reason for your choice. [5 Marks]

IP Address	MAC Address	Correct Entry? (Yes/No)	Brief Reason
IP-A	**MAC-A**	**Yes**	**Same subnet**
IP-D	**MAC-D**	**No**	**Not on the same subnet**
IP-9	**MAC-9**	**No**	**Same router's second interface**
IP-5	**MAC-5**	**No**	**Switch interface**

IP-3	**MAC-B**	**Yes/No**	**Could be IP-B**

6. IP Subnetting. Please use the space provided to show your working. [2.5+2.5=5 Marks]

6.1 Consider a router that interconnects three subnets: Subnet 1, Subnet 2, and Subnet 3. Suppose all of the interfaces in each of these three subnets are required to have the prefix 223.15.17/24. Also suppose that Subnet 1 is required to support at least 58 interfaces, Subnet 2 is to support at least 90 interfaces, and Subnet 3 is to support at least 15 interfaces. Provide three network addresses (of the form a.b.c.d/x) that satisfy these constraints.

Subnet address 1: ________223.15.17.0/26______________________________

Subnet address 2: _______223.15.17.128/25___________________

Subnet address 3: ________223.15.17.192/28__________________

Rough work (6.1):

6.2 For the following IP address and subnet mask **10.x.y.z /30**. List down all the network prefixes possible, listing the difference/spacing between each subnetwork. [Hint: calculate all the available values/range of x, y and z for subnet: /30]

Range of second octet (x): _________________ **spacing between successive values:** ___________

Range of third octet (y): _________________ **spacing between successive values:** ___________

Range of fourth octet (z): _________________ **spacing between successive values:** ___________

Rough work (6.2):

/30 = 255.255.255.252

x = 0-255, spacing 1

y = 0-255, spacing 1
z = 0-252, spacing 4

7. Suppose that TCP's current estimated values for the round trip time (estimatedRTT) and deviation in the RTT (DevRTT) are 210 msec and 49 msec, respectively. Suppose that the next three measured values of the RTT are 280, 210, and 220 respectively.

Compute TCP's new value of estimatedRTT, DevRTT, and the TCP timeout value after each of these three measured RTT values is obtained.Use the values of $\alpha = 0.125$ and $\beta = 0.25$. **[1+2+2=5 Marks]**

After the first RTT estimate is made:

estimatedRTT = 0.875*210 + 0.125*280 = 218.75 msecs
DevRTT = 0.75*49 + 0.25*(abs(280 - 218.75)) = 52.0625 msecs
TimeoutInterval = 218.75 + 4*52.0625 = 427 msecs

After the second RTT estimate is made:

EestimatedRTT = 0.875*218.75 + 0.125*210 = 217.65625 msecs
DevRTT = 0.75*52.0625 + 0.25*(abs(210 - 217.65625)) = 40.9609375 msecs
TimeoutInterval = 217.65625 + 4*40.9609375 = 381.5 msecs

After the third RTT estimate is made:

EestimatedRTT = 0.875*217.65625 + 0.125*210 = 217.94921875 msecs
DevRTT = 0.75*40.9609375 + 0.25*(abs(220 - 217.94921875)) = 40.9609375 msecs
TimeoutInterval = 217.94921875 + 4*31.2333984375 = 342.8828125 msecs

Rough work:

8. Consider the Figure 3 below, which plots the evolution of TCP's congestion window at the beginning of each time unit (where the unit of time is equal to the RTT); TCP sends a "flight" of packets of size cwnd at the beginning of each time unit. The result of sending that flight of packets is that either *(i)* all packets are ACKed at the end of the time unit, *(ii)* there is a timeout for the first packet, or *(iii)* there is a triple duplicate ACK for the first packet. In this problem, you are asked to reconstruct the sequence of events (ACKs, losses) that resulted in the evolution of TCP's cwnd shown below.**[5 Marks]**

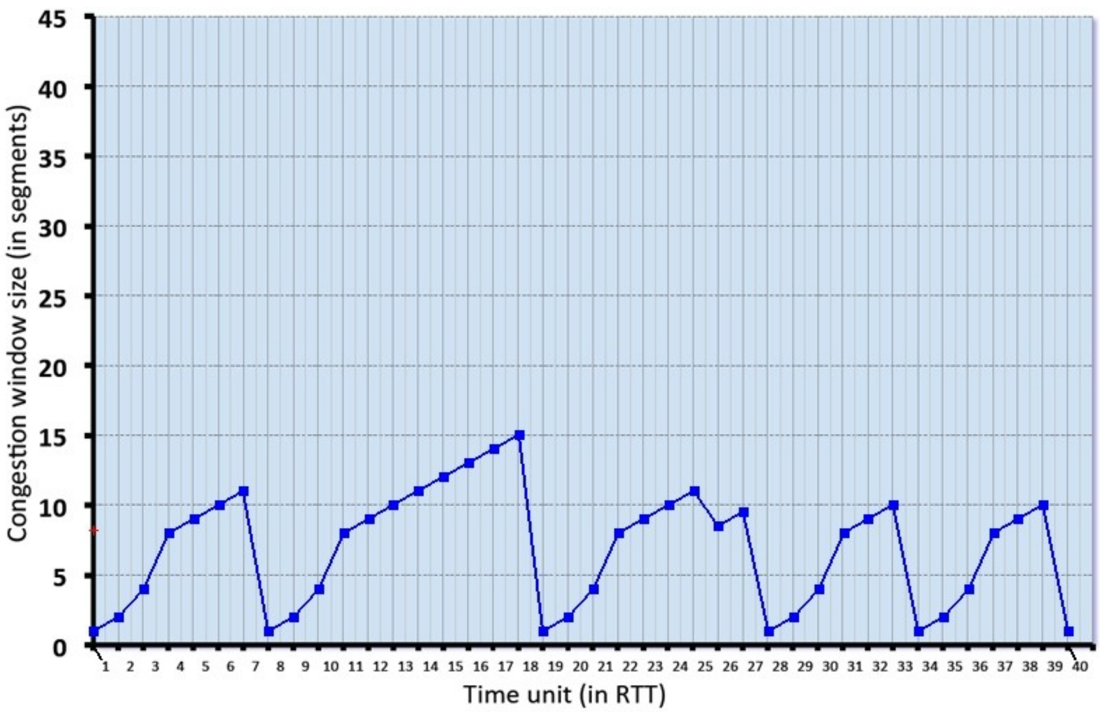

Figure. 3

The initial value of cwnd is 1 and the initial value of ssthresh is 8. Now answer the following questions:

8.1 Give the times at which TCP is in slow start and fast recovery at the start of a time slot, when the flight of packets is sent.

Slow-start:*format [a,b] [x,y] [z]*: [1,3][8,10] [19,21] [28,30] [34,36]
Fast recovery:[26]

8.2 Give the times at which the first packet in the sent flight of packets is lost, and indicate whether that packet loss is detected via timeout, ot by triple duplicate ACKs.

Timeout Loss Interval Points: [7] [18] [27] [33] [39]
3 Duplicate ACK Loss Points: [25]

9. Consider the scenario in Figure 4, in which three hosts, with private IP addresses 10.0.1.12, 10.0.1.16, 10.0.1.20 are in a local network behind a NATted router that sits between these three hosts and the larger Internet. IP datagrams being sent from, or destined to, these three hosts must pass through this NAT router. The router's interface on the LAN side has IP address 10.0.1.26, while the router's address on the Internet side has IP address 135.122.202.209. **[5 Marks]**

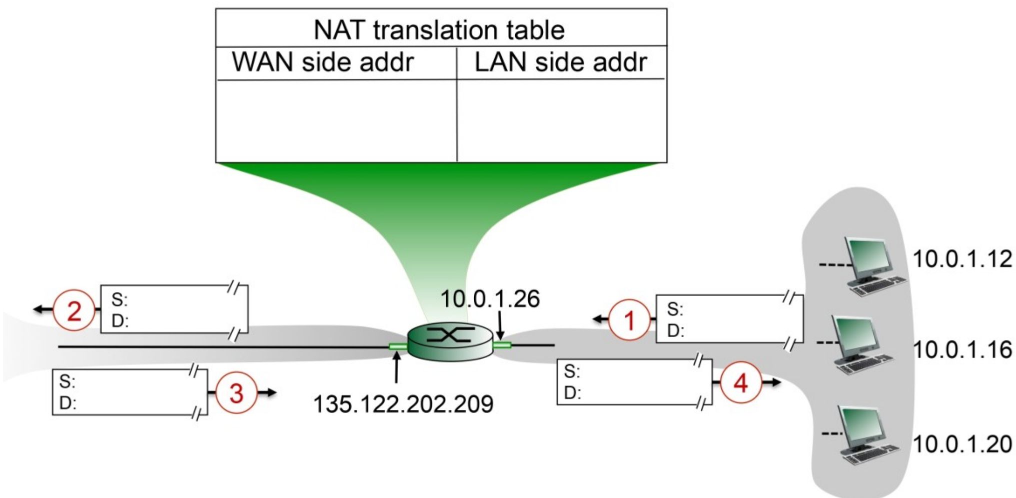

Figure. 4 Network Address Translation Scenario

Suppose that the host with IP address 10.0.1.16 sends an IP datagram destined to host 128.119.166.183. The source port is 3303, and the destination port is 80.

9.1 Consider the datagram at step 1, after it has been sent by the host but before it has reached the NATted router. What are the source and destination IP addresses for this datagram? What are the source and destination port numbers for the TCP segment in this IP datagram?

Source Socket:10.0.1.16:3303
Destination Socket:128.119.166.183:80

9.2 Now consider the datagram at step 2, Specify the entry that has been made in the router's NAT table. Make any assumptions as required.

Changes in Source Socket (if any):135.122.202.209:5251
Changes in Destination Socket (if any):128.119.166.183:80

9.3 Now consider the datagram at step 3, just before it is received back by the NATted router. What are the source and destination IP addresses for this datagram? What are the source and destination port numbers for the TCP segment in this IP datagram?

Source Socket:128.119.166.183:80
Destination Socket: 135.122.202.209:5251

9.4 consider the datagram at step 4, after it has been transmitted by the NATted router but before it has been received by the host. What are the source and destination IP address for this datagram? What are the source and destination port numbers for the TCP segment in this IP datagram?

Source Socket:128.119.166.183:80
Destination Socket: 10.0.1.16:3303

9.5 Identify the differences in datagram's IP addresses and port numbers between step 3 and step 4. Has a new entry been made in the router's NAT table, or removed from the NAT table? Explain your answer.

When this datagram arrived at NAT router's left port from the Internet, the router indexed the NAT translation table using the destination IP address and destination port number to obtain the appropriate IP address(10.0.1.16) and destination port (3303) for the destination host in the home network. The router then rewrites the datagram's destination address and destination port number, and forwards the datagram into the home network.