Indian Institute of Technology Kharagpur

EXAMINATION ANSWERSCRIPT

Stamp/Signature of the Invigilator

Spring Semester 2023-24 – End Sem Exam	**SEMESTER (Spring)**

Roll Number											**Section**		**Name**	

Subject Number	C	S	3	1	2	0	4		**Subject Name**	Computer Networks

Department/Centre/Schoo		**Additional Sheets**	

Same answerscript to be used for one student of CS31006 (old curriculum course no. of same course)

Important Instructions and Guidelines for Students

1. You must occupy your seat as per the Examination Schedule/Sitting Plan.

2. Do not keep mobile phones or any similar electronic gadgets with you even in the switched off mode.

3. Loose papers, class notes, books or any such materials must not be in your possession; even if they are irrelevant to the subject you are taking examination.

4. Data book, codes, graph papers, relevant standard tables/charts or any other materials are allowed only when instructed by the paper-setter.

5. Use of instrument box, pencil box and non-programmable calculator is allowed during the examination. However, the exchange of these items or any other papers (including question papers) is not permitted.

6. Write on both sides of the answer-script and do not tear off any page. **Use last page(s) of the answer-script for rough work**. Report to the invigilator if the answer-script has torn or distorted page(s).

7. It is your responsibility to ensure that you have signed the Attendance Sheet. Keep your Admit Card/Identity Card on the desk for checking by the invigilator.

8. You may leave the Examination Hall for wash room or for drinking water for a very short period. Record your absence from the Examination Hall in the register provided. Smoking and the consumption of any kind of beverages are strictly prohibited inside the Examination Hall.

9. Do not leave the Examination Hall without submitting your answer-script to the invigilator. **In any case, you are not allowed to take away the answer-script with you**. After the completion of the examination, do not leave your seat until the invigilators collect all the answer-scripts.

10. During the examination, either inside or outside the Examination Hall, gathering information from any kind of sources or exchanging information with others or any such attempt will be treated as '**unfair means**'. Don't adopt unfair means and also don't indulge in unseemly behavior.

Violation of any of the above instructions may lead to severe punishment.

Signature of the Student

To be Filled by the Examiner

Question Number	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**Total**
Marks Obtained											

Marks Obtained (in words)	**Signature of the Examiner**	**Signature of the Scrutineer**

INDIAN INSTITUTE OF TECHNOLOGY KHARAGPUR

End-Spring Semester Examination 2023-24

Date of Examination: 22.04.2024 **Session: (FN/AN) AN** **Duration: 3 hrs.** **Full Marks: 100**
Subject No.: CS31204/31006 **Subject: Computer Networks**
Department: Computer Science and Engineering
Specific charts, graph paper, log book etc., required: NONE
Special Instructions (if any): No clarification will be provided during the exam. State any assumption made. ONLY answer written in the space below each question will be evaluated.

ANSWER ALL QUESTIONS

1. (a) What is a virtual circuit? State one advantage and one disadvantage of the virtual circuit approach over the datagram approach. (1 + 2)

1 mark for virtual circuit definition (0.5 for saying fixed path, 0.5 for saying path can be shared)
1 mark for advantage and 1 mark for disadvantage (any reasonable ones accepted)

(b) State one advantage and one disadvantage of distance vector routing protocols over link state routing protocols. One sentence each only. (2)

1 mark for advantage, 1 mark for disadvantage (any reasonable one accepted)

(c) What is the primary difference between a Layer-2 switch and a Layer-3 switch? 1-2 sentence only. (2)

Full marks given if you have mentioned that a Layer-2 switch chooses the next node to send to based on MAC address, Layer-3 based on IP address. 1 mark given if you have only mentioned that Layer-2 operates at Ethernet layer and Layer-3 at IP layer.

Many of you have written long unnecessary things.

(d) What is the use of the SYN and FIN flags in a TCP segment? One sentence each only. (2)

1 mark for each. Anything that says or indicates connection establishment for SYN or connection termination for FIN has been given full marks

(e) Consider that a company X gets the network 203.110.245.20/22 from an ISP. It gives the first 256 IP addresses to its Engineering section and the last 128 IP addresses to its Administration section, for each to create its own subnet. The rest of the addresses are kept for future use. What are the subnets for the Engineering section and the Administration section? No explanation is needed. (2)

Engineering: *203.110.244.0/24* *Administration*: *203.110.247.128/25*

1 mark for each, only if fully correct

(f) Do you think it is a good idea for a routing protocol to use path delay (to destination) as a cost metric instead of number of hops? Justify briefly. (2)

Path delay usually has more variation than no. of hops due to sharing of paths by different connection and dynamic traffic variations in the path. This can cause more routing protocol overhead to propagate changes.

Also can cause oscillations as explained in class.

Any one reason of the above stated clearly has been given full marks. Partial marks given for many other answers. Some other answers also given marks.

(g) Name 4 parameters that a DHCP server can provide to a machine. At least 2 of the parameters must be something not needed for network access. No explanation is needed. (2)

0.5 marks for each. Note that IP, subnet etc. are all for network access, so only two accepted from everything specific to network access.

(h) Do two NICs manufactured by two different manufacturers guaranteed to have different MAC addresses? If yes, how is it ensured? If no, why not? 1-2 sentence only. (2)

Full marks given if the idea clearly comes out that part of the MAC address (vendor code, ok if you forgot the exact name) is a unique manufacturer specific address given by a central authority.

2. (a) Write a pseudocode to show how an IP packet received from the Ethernet layer is processed by the IP layer software. Assume that the receiver has only one NIC and no IP options are there in the packet. (10)

Note that the receiver has only one NIC, so cannot be configured as a router. Many of you have written too much unnecessarily on forwarding the packet including TTL, ARP etc. for some. No marks given for that.

2 marks for checksum checking (should be done first)
6 marks if destination IP matches

- *3 marks if it is a fragment, should check cases like if first fragment to arrive, last fragment to arrive that completes a datagram, fragment that does not complete datagram etc. Note that discarding packet when reassembly timer is over is not part of IP packet receive processing, it is a timeout event processing*
- *3 marks if it is a datagram (not a fragment), should state removing padding if any, checking protocol field etc.*

2 marks if destination IP does not match (drop and ICMP send)

In all parts, when you drop a packet, ICMP to be sent (ok if you forgot exact what packet). Descriptions should refer to specific header fields wherever appropriate and not just talk in general.

(b) Consider an IP packet P being sent from a host A with IP 200.100.23.34 to a host B with IP 225.200.194.7. A has an Ethernet interface eth0 with MAC 0x010203040506 and B has an Ethernet interface also named eth0 with MAC 0x0A0B0C0D0E0F. Subnet mask in all networks in the system are 255.255.255.0. The two networks the two hosts are in are connected with a single gateway/router R that connects only to these two networks, with the interface eth1 (with MAC address 0x112233445566) connecting to A's network and the interface eth2 (with MAC address 0x99AABBCCDDEE) connecting to B's network. Gateway IP address in the two networks is the 1st IP address available for allocation to a host in that network. Assume that there are no other networks these two networks will connect with. Ignore loopback addresses.

(i) Write the essential routing table entries in A, clearly identifying the fields. No explanation is needed. (4)

Answers varied but overall:

One entry for machines in its own network, so next hop will be locallink or same IP as m/c or any other word you used that is reasonable. Cannot be R's IP (200.100.23.1) (2 marks)
One entry for the other network, this should have R's IP as next hop. Default is also accepted though unnecessary given that there are only 2 networks. (2 marks)

Exact value of cost is ignored but the field must be there. Same goes for interface field

(ii) Write the routing table entries for the router R. No explanation is needed. (2)

One entry for A's network, interface should be eth1 (ok if IP200.100.23.1 specified instead) (1 mark)
One entry for B's network, interface should be eth2 (ok if IP200.200.194.1 specified instead) (1 mark)

(iii) Explain the steps by which the next hop for the IP packet P is decided at A referring to the routing table. (2)

Should at least mention bitwise and with subnet mask, computations not expected to be shown. Most of you have written in general without referring to packet P or m/c A's table. Marks still given if answer is clear.

(iii) Show the source and destination IP of all IP packets that are sent. No explanation is needed. (2)

Same for A to R and R to B. Both source = 200.100.23.34, destination = 225.200.194.7

2 marks if full correct, 0 otherwise

(iv) Show the source and destination MAC of all Ethernet packets that are sent. Assume that ARP cache contains mappings for all IPs in the respective networks. No explanation is needed (2)

A to R: source is A's MAC (eth0's), destination is eth1's MAC
R to B: source is eth2's MAC, destination is B's MAC (eth0's)

1 mark each

(d) Consider an implementation of the ping tool where a sequence of 10 Echo Request packets are sent in succession without waiting for the replies of any earlier packet sent. The RTTs of each of these packets will be averaged to be the RTT reported to the user. However, a naïve implementation of this requires starting one timer for each request at the sender, and using it when the corresponding reply comes back to compute the RTT of the corresponding packet. Can you give an implementation of the tool that will do this without maintaining individual timers for each request packet at the sender if no individual packet timeouts have to be reported? Be specific. (5)

What is done usually is to put the send time as data in ECHO REQUEST. ECHO REPLY copies the data back always, so when it comes back, you know the send time, the current time, so compute the difference.

If you used ICMP timestamp request/reply, you got 4 out of 5 if otherwise clearly stated. Computing RTT with Timestamp request/reply will need clocks to be synchronized.

Some of you used 2 timers, one to add up send times, and one to add up receives and then take difference (or variation that achieves the same thing). Though my intention was not to use any timers, gave full marks as didn't mention clearly and will work in this case. There is a boundary case I ignored, the simple method you specified will not work if the request or reply is lost.

Partial marks (1-2 depending on what you wrote) given for other answers like if you assume all sends are done at the same time, or sent at regular intervals etc. These cannot be ensured and should not be assumed, more so because there is an easy solution available.

3. (a) How does a TCP sender know what MSS size and initial receiver window size to use for a connection? One sentence each. (2 + 2)

MSS size can be negotiated using TCP options at connection establishment time. If not negotiated, default MSS size is used (1 mark for each)

Initial receiver window size of the receiver is known by the sender from the advertised window field of the SYN+ACK field received from the receiver during connection establishment. Ok as long as you mentioned using the window field in header of any packet during connection establishment.

(c) Name two problems of using packet loss as an indicator of congestion in TCP in the internet. Do you think delay (increased RTT) would be a better congestion signal to use for internet traffic? Justify briefly. (4)

1. *Loss need not be due to congestion only, so loss may not always indicate congestion. (1.5 marks)*
2. *When loss happens, congestion has already set in and packets have started getting dropped already. A drop will cause retransmissions thereby increasing congestion and further drops, and wasting bandwidth. Ideally, congestion should be known before packets start getting dropped so that send rate can adjusted. (1.5 marks)*

Delay can give early indication of congestion before drops actually happen and so is ideally a good method. However, delays can also be caused by different things, it is hard to measure without noise, and adjusting sending rates based on delay feedbacks are more complex.
(This had 1 mark. Just wanted to see what you think. 0.5 given to everyone who wrote good or bad. Full 1 marks given to few who showed they understood both aspects.)

(b) List clearly the events that happen when a timeout occurs for a TCP segment in TCP Reno. (6)

2 marks each for mentioning basics of change in timeout (multiply by delta etc. by Karn's algorithm), congestion control (at last mention slow star, cwnd, ssthresh), and retransmission.

Many of you have just written whatever you know of congestion control, including 3 duplicate acks etc. etc. No marks given. Note that event of receiving 3 duplicate acks is not timeout event, so congestion control will not do all that on a timeout event.

(d) Consider a TCP connection between two hosts A and B. The current send and receive window sizes at A are 5000 bytes and 6000 bytes respectively. The current receive window size at B is 4000 bytes (known to A from previous segments). There are no unacknowledged segments currently. TCP segments of size 980 bytes, 860 bytes, 740 bytes, 780 bytes, and 660 bytes (including headers) are sent in sequence at times t = 0, 1, 2, 3, 4, containing consecutive bytes of data (do not worry whether TCP can actually send these segments as per sending rules, just assume they are sent). Let the segments be numbered from 1 to 5 for reference. Segments 1 to 5 reach at t = 2, t = 8.5, t = 5, t = 6, and t = 7 respectively. Any other segment that may get sent by A or B takes 1 time unit to reach the other side. The user at B makes one recv() call at t = 3, and one recv() call at t = 6, each with a buffer size of 600 bytes. No data is transmitted in the other direction from B to A ever. TCP acknowledgment wait timer is 2 time units. Retransmission timeout is a fixed 10 time units. MSS is 1000 bytes. Sequence no. of first byte of Segment 1 is 1250. User receive buffer is the only buffer used by receiver.

(i) Show the following values for <u>all</u> TCP segments that are transmitted between A and B. Number of rows are just shown for formatting, add/delete as you need. No explanation is needed. (12)

5 marks for the 5 data packets sent at t=0 to t=4 as mentioned, sequence no.s just add up starting from 1250, window is 6000 for all. 1 mark total deducted if you did not consider that the sizes given are including header. 1 mark deducted if you didn't put window size (all same at 6000).

First ack will be sent at t = 4 (segment 1 reaches at t = 2, ack timeout is 2). Then a duplicate ack at each of t = 5, 6, 7 as segments 3, 4, 5 reach out of order. (4 marks for the 4 acks.)

The ack at t = 7 reaches A at t = 8, this is 3rd duplicate ack, so missing segment is retransmitted due to fast retransmit. (2 marks)

Ack sent for the entire length of bytes when segment 2 reaches at t = 8.5. (1 mark)

Marked upto this, there will be another ack after the data sent at t=8 reaches at t=9.

Ignored window field values in Acks mostly as long as you have something reasonable as many of you made different assumptions for what happens to out of order values (or just wrote random values maybe, for all I know ☺). Since there is one buffer only, out of order packets will have to be stored there only, though user recv() call will get only whatever inorder data is available. But accepted almost anything.

Partial marks given for many other answers as long as I could relate with the above.

(ii) if the congestion window at A just before t = 5 was 8, and A was in congestion avoidance phase, what would be the congestion window size at A at t = 9? Assume TCP Reno, with the formula shown in class. Justify briefly. (2)

Acks received after t=5 and before t=9 are all duplicate acks, so nothing happens. At 3 duplicate acks, congestion window gets set to 8/2 + 3 = 7.

Partial marks given for many variations depending on answer.

4. (a) State one difference between a recursive query and an iterative query in DNS. (2)

Most of you have written definitions of both, could have written less. Anyway, given full marks.

(b) How can you at IIT Kgp find out the IP address of the SMTP server of a company X.com? 1 sentence only. (2)

Make a DNS query (program/website/whatever, ok if you did not mention also) for the MX record of X.com. Keyword was the MX record, no marks given without that.

(c) Suppose that an organization X gets only one public IP 200.150.100.50 from its ISP, and uses NAPT (Network Address and Port Translation) using private IPs inside. A TCP client P inside X (IP address 10.5.20.20, port 30000) connects to a server S1 with IP address 172.217.166.68 and port 80. While it is communicating, another UDP client Q from inside X (IP address 10.7.30.30, port 400000) sends a message to a server S2 with IP address 108.158.58.146 and port 100 and waits for a reply.

(i) Show the content of the NAT table (show the fields as columns, and the values in rows) at the NAT-capable router of X when both P and Q are active. Assume arbitrary values for anything not specified that you may need. No explanation is needed. (2)

0.5 marks deducted if Protocol field is not shown.

(ii) Using the table, explain how a packet sent from the server S2 to Q reaches Q, referring to relevant header fields. Be specific. (2)

1 mark deducted if you did not specifically mention that the destination IP and port are <u>changed</u> at the NAT-router. Just saying routers sends/routes/forwards it to Q is not enough, there are other possible ways to do that such as tunneling etc. by building your own protocol.

(d) Clearly describe how a DHCP client can renew an IP lease (before lease expiry) from a DHCP server in a different subnet. Mention the message names, source-destination IP-port values for all messages, and relevant DHCP header field values (assume arbitrary machine IPs if you need, but clearly mention exactly what values you are assuming for what machine before describing the renewal process). (4)

Many of you missed that it is a renew, so the client is fully configured and no relay agent function is needed anywhere for the UDP packet sent. This is just simple UDP communication. Client has an IP, and DHCP server IP is known. No marks given for your long descriptions of relaying.

1 mark for starting from DHCP_Request, 1 mark for IP/Ports of request and Ack, 1 mark for other header fields (client_ip etc.), 1 mark for overall description.

Description for only the first step (before 87.5% lease time over) is ok.

5. (a) Taking an example of a network with exactly 5 nodes and 5 edges, show how a routing loop can form in a distance vector routing protocol even when split horizon is used. You should clearly list the sequence of events (in time order) and show relevant routing table entries at the nodes at each step (no need to show full routing table). . (6)

My intention was to ask you for a 4-length loop. But since I did not specify it, accepted anything as long as it has 5 nodes and 5 links, most of you showed 3 node loop which is fine.

Marks given depending on answer. You need to show the routing entries at each step clearly, handwaiving description only is not enough.

(b) Consider the distance vector routing protocol done in class. The following changes are made to the protocol:

(i) if a routing table entry is not refreshed for 4 consecutive update periods, it is deleted from the table. Note that "refresh" does not necessarily mean changed, it just means that no entry is received with that same destination address from the same next hop in any update message.

(ii) if a routing table entry's cost is changed to ∞, no update is accepted for that destination network from anywhere for the next 4 150 seconds (assume you have a time() call you can call to get system time in seconds).

Write the pseudocode executed when an update is received. If you need any additional field in the table or in the message header, please specify clearly first before writing the pseudocode. (10)

2 marks for additional header fields (one for last refresh time, one for keeping time set to infinity). You can do with one field also, code gets a bit complicated, given full marks if correct.

2 marks for adding new entry received with proper fields (including the new fields added, many of you missed that)
2 marks for changing if cost not infinity (same next hop or lower cost, 1 mark each)
2 marks for handling infinity setting and not changing for a period correctly
2 marks for refresh and delete.

*Any value of update period assumed (or just keeping it as a variable) is accepted. The 4 150 seconds was a small typo left while changing from 4 update periods to 150 seconds, any interpretation (4150, 4*150, only 4, only 150,…) is accepted.*

Some of you are confused between refresh and update. An entry may come from y to x in the next period with same dest and cost, and may not change an existing entry with same dest and next hop y, that is still refresh.
Also, for counting periods without refresh, simply doing +1 on processing an update message from same next hop is not enough, as one update in the middle may get lost fully. Need to check in terms of time