**Department of Electrical and Computer Engineering**

# ECE 358 – Computer Networks

## Selected Problems from Previous Midterm Exams

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## Question 1 [5 Marks]:

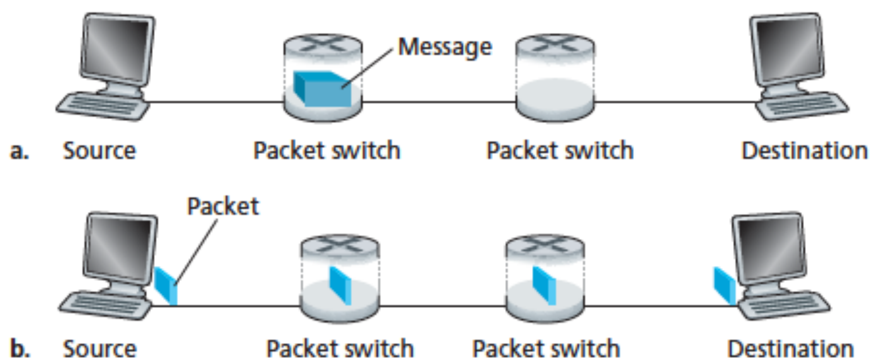


**Figure 1**. End-to-end message transport: (a) without message segmentation; (b) with message segmentation

In modern packet-switched networks, including the Internet, the source host segments long, application-layer messages (for example, an image or a music file) into smaller packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as message segmentation. Figure 1 illustrates the end-to-end transport of a message with and without message segmentation. Consider a message that is 8 M bits long that is to be sent from source to destination in Figure 1. Suppose each link in the figure is 2 Mbps. Ignore propagation, queuing, and processing delays.

**(a)** Consider sending the message from source to destination without message segmentation. How long does it take to move the message from the source host to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching, what is the total time to move the message from source host to destination host? **(2 marks)**

**(b)** Now suppose that the message is segmented into 800 packets, with each packet being 10,000 bits long. How long does it take to move the first packet from source host to the first switch? When the first packet is being sent from the first switch to the second switch, the second packet is being sent from the source host to the first switch. At what time will the second packet be fully received at the first switch? **(1 marks)**

**(c)** How long does it take to move the file from source host to destination host when message segmentation is used? Compare this result with your answer in part (a) and comment. **(2 marks)**

(a) Time to send message from source host to first packet switch = $\frac{8\times10^6}{2\times10^6}\sec = 4\sec$ With store-and-forward switching, the total time to move message from source host to destination host = $4\sec\times 3\ hops = 12\sec$

(b) Time to send $1^{st}$ packet from source host to first packet switch = . $\frac{1\times10^4}{2\times10^6}\sec = 5\ m\sec$. Time at which $2^{nd}$ packet is received at the first switch = time at which $1^{st}$ packet is received at the second switch = $2\times 5m\sec = 10\ m\sec$

(c) Time at which $1^{st}$ packet is received at the destination host = $5\ m\sec\times 3\ hops = 15\ m\sec$. After this, every 5msec one packet will be received; thus time at which last ($800^{th}$) packet is received = $15\ m\sec + 799 * 5m\sec = 4.01\sec$. It can be seen that delay in using message segmentation is significantly less (almost $1/3^{rd}$).

**Question 2 [7 marks]:**

**(a)** What is the difference in the service offered to applications by the TCP and UDP protocols? **(2 marks)**

**(b)** For each of the following applications determine whether you would use TCP or UDP and explain the reasons for your choice. **(2 marks)**

1. File transfer
2. Watching a real time streamed video
3. Web browsing
4. A Voice Over IP (VoIP) telephone conversation

**(c)** Both TCP and UDP use port numbers. What are these port numbers used for? **(1 mark)**

**(d)** Briefly explain the difference between persistent and non-persistent HTTP. **(1 mark)**

**(e)** Briefly explain the difference between Go-Back-N and Selective Repeat protocols. **(1 mark)**

(a) TCP operates above IP and provides a connection orientated service in which end to end data transfer is guaranteed with flow control and congestion avoidance capabilities. It is a reliable service. UDP operates above IP and provides a connectionless, unacknowledged service to the upper layers. There is no error detection or recovery and no flow control provided by UDP. It is a best effort service.

(b)

1. TCP, losing data in text is not acceptable, therefore the reliable data transfer of TCP is needed.
2. UDP, the reason is that when watching a movie, delay is critical and therefore there simply isn't any time to seek the retransmission of any errors. The simplicity of UDP is therefore required.
3. TCP, The reason is that web pages need to be delivered without error so that all content is properly formatted and presented. therefore the reliable data transfer of TCP is needed.
4. UDP, The reason is that a telephone conversation has strict timing requirements for the transfer of data and seeking the retransmission of any errors would introduce too much delay. Therefore, the simplicity of UDP is needed.

(c) Both TCP and UDP provide services to higher layer protocols however multiple higher layer protocols can be multiplexed onto a single UDP or TCP layer. Each of these higher layer protocols are then differentiated by means of UDP/TCP port numbers. Therefore the port number identifies the particular higher layer protocol to which a given data stream is destined.

(d) In persistent HTTP, only one TCP is created for multiple objects. In non-persistent HTTP, one TCP is required per object.

(e) <u>Go-Back-N</u>

- sender can have up to N unacked packets in pipeline
- receiver only sends cumulative ack
- doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
- when timer expires, retransmit all unacked packets

<u>Selective Repeat</u>

- sender can have up to N unack'ed packets in pipeline
- receiver sends individual ack for each packet

- sender maintains timer for each unacked packet
- when timer expires, retransmit only that unacked packet

**Question 3 [8 marks]:**

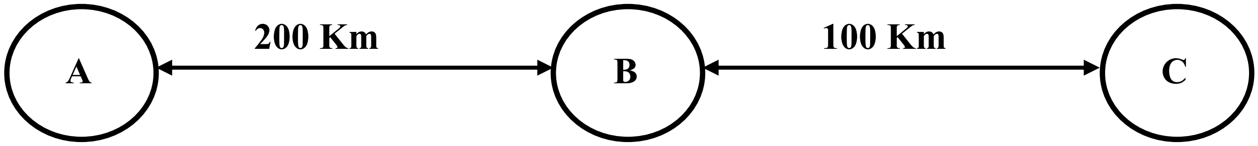


**Figure 2.** Question 3

In Figure 2, frames generated by node A are sent to node C through node B. Based on the following data:

- The data rate between A and B is 1 Mbps.
- The propagation speed is $10^8$ m/sec for both lines.
- There are full duplex lines between the nodes.
- All data frames are 1000 bits long; ACK frames are separate frames of negligible length.
- There are no errors.

Determine the minimum transmission rate required between node B and C so that the buffers of node B are not flooded for each of the following cases.

(a) A sliding-window protocol with window size of 2 is used between A and B and the stop-and-wait protocol is used between B and C. **(5 marks)**

(b) A sliding-window protocol with window size of 10 is used between A and B and the stop-and-wait protocol is used between B and C. **(3 marks)**

(a) For link AB

The propagation delay

$$d_{prop_{AB}} = \frac{200\ Km}{10^8\ m/sec} = 0.002\ sec$$

The transmission delay

$$d_{trans_{AB}} = \frac{1000\ bits}{10^6\ bps} = 1\ msec$$

The total delay

$$T_{AB} = d_{trans_{AB}} + 2\ d_{prop_{AB}}$$
$$= 0.001 + 2 * 0.002 = 0.005$$

$$Efficiency_{AB} = \min\left(\frac{W.d_{trans_{AB}}}{T_{AB}}, 1\right)$$
$$= \frac{2 * 0.001}{0.005} = 0.4$$

$$Throughput_{AB} = Efficieny * link\ rate$$
$$= 0.4 * 10^6 = 400\ Kbps = 400\ pkt/sec$$

<u>For link BC</u>
The propagation delay

$$d_{prop_{BC}} = \frac{100\ Km}{10^8\ m/sec} = 0.001\ sec$$

The transmission delay

$$d_{trans_{AB}} = \frac{1000\ bits}{R_{BC}}$$

The total delay

$$T_{BC} = d_{trans_{BC}} + 2\ d_{prop_{BC}}$$
$$= d_{trans_{BC}} + 2 * 0.001$$
$$= d_{trans_{BC}} + 0.002$$

$$Throughput_{BC} = \frac{1}{T_{BC}} = \frac{1}{d_{trans_{BC}} + 0.002}$$

In order not to flood B,

$$Throughput_{BC} \geq Throughput_{AB}$$

$$\frac{1}{d_{trans_{BC}} + 0.002} \geq 400$$

$$d_{trans_{BC}} \leq 0.0005$$

$$\frac{1000}{R_{BC}} \leq 0.0005$$

$$R_{BC} \geq 2\ Mbps$$

<u>(b) For link AB</u>
The propagation delay

$$d_{prop_{AB}} = \frac{200\ Km}{10^8\ m/sec} = 0.002\ sec$$

The transmission delay

$$d_{trans_{AB}} = \frac{1000\ bits}{10^6\ bps} = 1\ msec$$

The total delay

$$T_{AB} = d_{trans_{AB}} + 2\ d_{prop_{AB}}$$
$$= 0.001 + 2 * 0.002 = 0.005$$

$$Efficiency_{AB} = \min\left(\frac{W.d_{trans_{AB}}}{T_{AB}}, 1\right)$$
$$= \min\left(\frac{10 * 0.001}{0.005}, 1\right) = 1$$

$$Throughput_{AB} = Efficieny * link\ rate$$
$$= 1 * 10^6 = 1\ Mbps = 1000\ pkt/sec$$

<u>For link BC</u>

The propagation delay

$$d_{prop_{BC}} = \frac{100\ Km}{10^8\ m/sec} = 0.001\ sec$$

The transmission delay

$$d_{trans_{AB}} = \frac{1000\ bits}{R_{BC}}$$

The total delay

$$T_{BC} = d_{trans_{BC}} + 2\ d_{prop_{BC}}$$
$$= d_{trans_{BC}} + 2 * 0.001$$
$$= d_{trans_{BC}} + 0.002$$

$$Throughput_{BC} = \frac{1}{T_{BC}} = \frac{1}{d_{trans_{BC}} + 0.002}$$

The maximum $Throughput_{BC}$ happens when $d_{trans_{BC}}$ is negligibly small (i.e., the link rate is too fast). In this case

$$Throughput_{BC_{MAX}} = \frac{1}{0.002} = 500\ pkt/sec$$

It is clear that $Throughput_{BC_{MAX}}$ will be less than $Throughput_{AB}$ and in this case B will be flooded under any value for $R_{BC}$.

## <u>Question 4 [25 marks]</u>: Packet-Switching

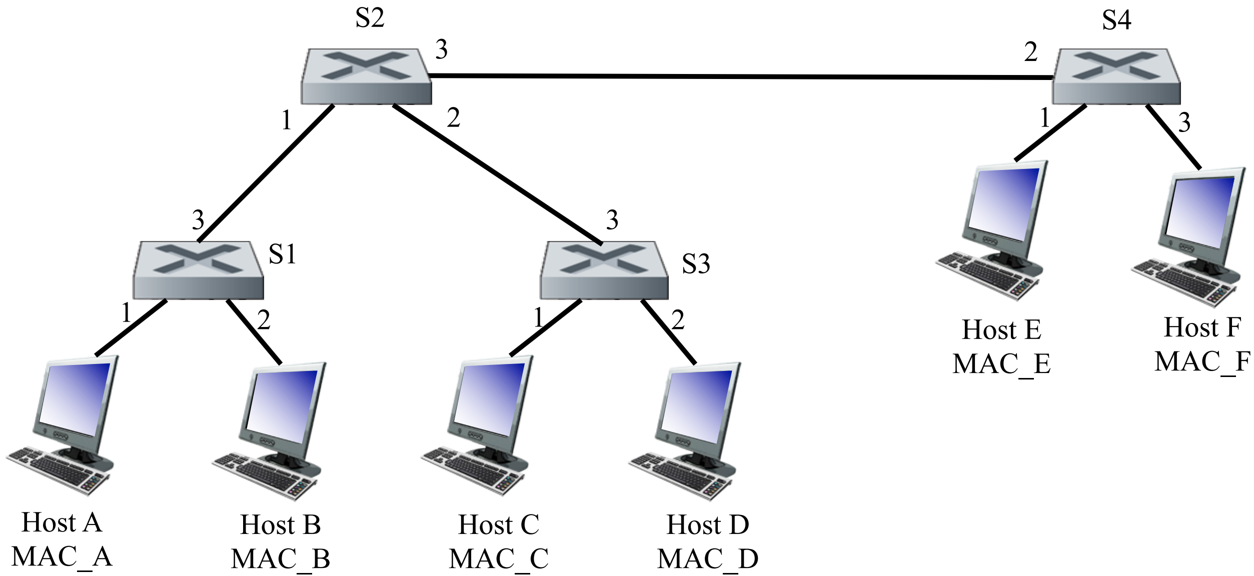


**Figure 1**. Question 2.

In the **packet-switched** network shown in Figure 1, all the links have an equal speed of **1 Mbps**. S1 - S4 are packet switches. Ignore the processing, propagation and queuing delays at all the switches.

**Host A** will **transmit** a file of size **1 MBytes** to **Host F**.
**Host A** has a fixed packet size of **1 KBytes**. Consider that the transmission of packet 1 will start at **time 0**.

(a) When will Host F completely receive the first packet? (3 marks)

There are 4 links of the same speed between A and F

$$T_{F-1st-packet} = 4 \times d_{trans-1M}$$

$$T_{F-1st-packet} = 4 \times \frac{1000 \times 8}{1 \times 10^6}$$

$$T_{F-1st-packet} = 32\ msec$$

(b) When will Host F completely receive all the packets? (3 marks)

$$Number\ of\ packets = \frac{1M}{1K} = 1000\ packets$$

$$T_{F-complete} = T_{F-1st-packet} + (number\ of\ packets - 1) \times d_{trans-1M}$$

$$T_{F-complete} = 32 + (1000 - 1) \times 8$$

$$T_{F-complete} = 8024\ msec$$

(c) What is average throughput for Host F? (2 marks)

$$Throughput_F = \frac{1\ Mbytes}{T_{F-complete}} = \frac{1M}{8024 \times 10^{-3}} = 124.63\ Kbytes/\sec = 997Kbps$$

(d) Using the bottleneck link concept, calculate the approximate time required for Host F to receive all the packets? (3 marks)

Since all the links are of equal speed, then the bottleneck link is either one of them which is 1 Mbps.

$$T_{F-complete-Approximate} = number\ of\ packets \times d_{trans-1M}$$

$$T_{F-complete-Approximate} = 1000 \times 8\ mesc$$

$$T_{F-complete-Approximate} = 8000\ msec$$

## <u>Question 5:</u> TCP

Let two nodes (computers and/or routers), A and B, be connected by a single physical link and run the SR protocol for reliable data transfer from A to B. Assume that K is the size of the sequence number space, Ws is the size of the sender's window, and Wr is the size of the receiver's window.

a) State the four kinds of errors that a reliable data transfer protocol must detect and fix.

b) Clearly explain by means of an example how a selective-repeat sliding-window protocol might deliver duplicate data to its upper layer.

c) Develop and explain a relationship between Ws and K to solve the problem in part b.

See lecture notes for the answer of this question

## Question 6: TCP

Consider a single Reno TCP connection between two hosts.
When a loss event occurs, the congestion window is reduced to W/2.

A loss event is an event that enables the sender to know that one segment did not reach the receiver, but the following segment reached the receiver. For example, a triple duplicate ACK received by the sender is a loss event.

Suppose that the sender never gets a timeout.
Denote by W the value of the congestion window in units of MSS when a loss event occurs.
In the period of time from when the connection's congestion window varies from W/2 to W, only one segment is lost (at the very end of the period).

Assume that RTT and W are roughly constant throughout a TCP connection.

Suppose that 1 MSS = 1500 bytes, W = 500 MSS, and RTT = 100 msec.

a) When a loss event occurs, what is the data rate (in Mbps) of the TCP connection?

The data rate when a loss event occurs = W/RTT = 8 x 1500 x 500 / 0.1 = 60 Mbps.

b) What is the average throughput (in Mbps) of the TCP connection for the given values of W, MSS, and RTT?

The average data rate = 0.75 x W x MSS/RTT = 0.75 x (8x500 x 1500)/(0.1) bps= 45 Mbps.

## **Question 7 [10 Marks]:**

Figure 1 shows the graph of a TCP RENO throughput (**NOT DRAWN TO SCALE**), where the y-axis describes the TCP window size in Bytes of the sender and the x-axis is the time.

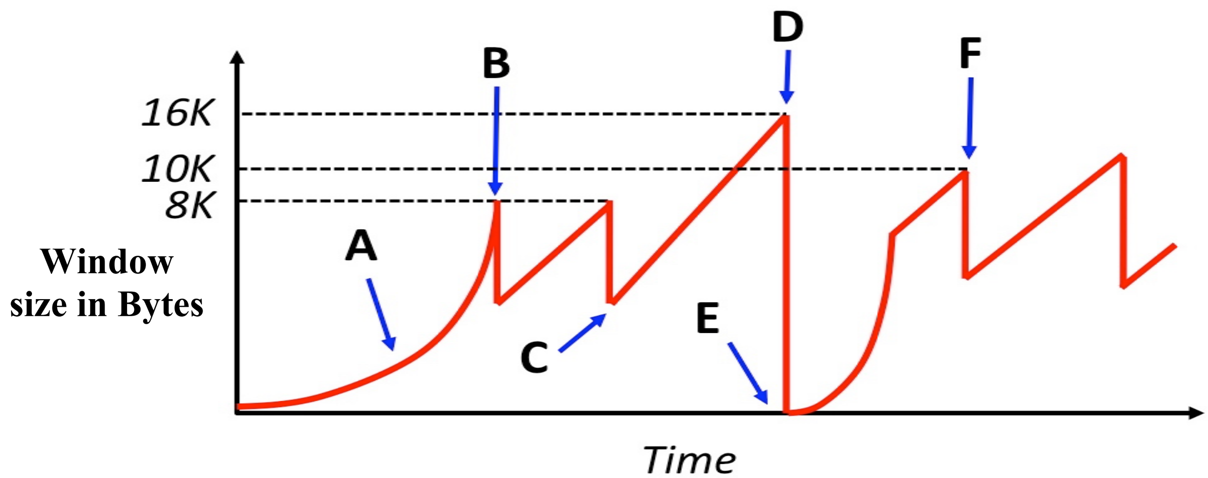


**Figure 1**. Question 3.

1) The window size of the TCP sender decreases at several points in the graph, including those marked by B and D.
   a) Name the event occurred at B that causes the sender to decrease its window. **(1 marks)**

**Triple duplicate ACK**

   b) "At event B, the network is fully congested and it cannot deliver any packet to the receiver". Is the previous statement "True" or "False"? If it is "False", change any wording to make the statement correct. Justify your answer. **(2 marks)**

**False. "At event B, the network is congested however it still can deliver some packets to the receiver".**
**The packets sent after the lost packet are received by the receiver as indicated by the triple ACK.**

   c) Name the event occurred at D that causes the sender to decrease its window. **(1 marks)**

**Timeout**

   d) "At event D, the network is fully congested and it cannot deliver any packet to the receiver". Is the previous statement "True" or "False"? If it is "False", change any wording to make the statement correct. Justify your answer. **(1 marks)**

**True.**

2) Consider the curved slope labeled by A. Why does the TCP window behave in such a manner, rather than having a linear slope? (In other words, why would it be bad if region A had a linear slope?) **(1 marks)**

**This "slow-start" period quickly discovers the maximum acceptable throughput that the path supports – otherwise, AI (additive increase) could take too long (each a full RTT).**

3) Assume that the network has an MSS of 1000 bytes and the round-trip-time between sender and receiver is 100 milliseconds. Assume at time 0 the sender starts its transmission. Ignore the transmission delay. The only latency you need to worry about is the propagation delay of the network.

a) How much time has progressed by point B? **(1 marks)**
**4 RTT in slow-start (1, 2, 4, (8) MSS)**
**= 4 RTT = 400 ms**

**b)** How much time has progressed between points C and D? **(1 marks)**

**4 MSS to 16 MSS = 13 periods of RTT = 1.3 s** (Timing is calculated from the beginning of event C till the end of event D)

c) How much time has progressed between points E and F? **(1 marks)**

**First: slow start to 8K window size (1, 2, 4, 8 MSS), then AI from 9 to 10 MSS window size (9, 10 MSS). Time = 6 RTT = 600 ms.** (Timing is calculated from the beginning of event E till the end of event F)

4) If the sender shares its network with other clients whose traffic traverses the same IP routers, give one explanation for why point D is higher than point B? **(1 marks)**

**Changing cross-traffic by other concurrent senders across same routers. In other words, other users may have stopped transmitting.**