Midterm Solution- CS 132/EECS 148
Computer Networks - Winter 2018
Prof. Marco Levorato

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Seat no.:	Date: February 13^{th}, 2018

- Points pre-assigned to review questions and problems are just an indication of the final score: partial and unsatisfactory answer/solution will receive a fraction of the points, whereas excellent answers may exceed the indicated points.
- One two sided **handwritten** A4 cheat sheet is allowed.
- Indicate name and UCI ID number in ALL the solution sheets.
- IF you use the extra sheet, clearly state it under each question that continues there.
- Time allowed: 70 minutes.

MULTIPLE CHOICE

Question 1. *(2.5 points)* In the OSI network layers model, which layer(s) does a router process?

☐ Layers 1-2.
☐ **Layers 1-3**
☐ Layer 3.
☐ Layer 2.

Reason: The routers perform routing at layer 3 and to support this it implements lower layers.

Question 2. *(2.5 points)* In order to provide a reliable message transfer service across an unreliable packet network, one must add:

☐ Packet reordering.
☐ **Packet retransmission.**
☐ Packet reordering and packet retransmission.
☐ Unique routing.

Reason: Only packet retransmission is required for a reliable message transfer.

Question 3. *(2.5 points)* The advantage of layered design is:

☐ Software can be written in the smallest number of lines of code.
☐ Competing network product designers use the same components.
☐ A protocol designer at one layer doesnt need to know about the interface used by a protocol at the next lower layer.
☐ **Interfaces are standardized.**

Reason: Because of the layered design, there are standard interfaces between the layers.

Question 4. *(2.5 points)* In telephone networks, which of the following performance measures has a firm limit in order to for the call to have acceptable performance?

☐ Packet loss.
☐ Blocking probability.
☐ **Delay.**
☐ Throughput.

Reason: In telephone networks or any other voice application over Internet, delayed packets must be discarded.

Question 5. *(2.5 points)* If the most popular application in a network is uncompressed video streaming, which network architecture is most appropriate?

☐ **Circuit-switching.**
☐ Packet-switching.
☐ Connection-oriented.
☐ Asynchronous.

Reason: For uncompressed video streaming, we must have consistent high throughput link all the time. Hence, circuit-switching would be the ideal approach.

Question 6. *(2.5 points)* In the Tahoe version of TCP, computer A does not decrease its window size when:

☐ A packet timeout occurs.
☐ A packet transmitted by computer A is dropped by a router.
☐ **A packet transmitted to computer A is dropped by a router.**
☐ A packet transmitted by computer A queues for a very long time in a router.

Reason: The congestion window size is decreased at the sender and not the receiver.

Question 7. *(2.5 points)* Consider a transmission in which the hosts are really far from each other (let's say more than $100,000$ km) and the message carried is very small (a few bytes), with link speed equal 80% of the speed of light. Which one of the following statements is certainly true?

☐ $d_{transmission} < 0$
☐ $d_{transmission} + d_{queue} > d_{propagation}$
☐ $\mathbf{d_{transmission} < d_{propagation}}$
☐ $d_{transmission} + d_{queue} > d_{propagation} + d_{processing}$

Reason: Since the message is very small and the link distance is very high, the transmission delay must be lower then the propagation delay.

Question 8. *(2.5 points)* Suppose there are two links between host A and host B. The first link has transmission rate 100 Mbps and the second link has transmission rate 10 Mbps. Assuming that the only traffic in the network comes from the source, what is the throughput for a large file transfer?

☐ 110 Mbps.
☐ 100 Mbps.
☐ **10 Mbps.**
☐ 1 Gbps.

Reason: The bottleneck link here is the second link with 10 Mbps, so the throughput is limited to 10Mbps.

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PROBLEMS

Question 9. *(20 points)* We are sending a 30 Mbit MP3 file from a source host S to a destination host D. All links in the path between source and destination have a transmission rate of 10 Mbps, unless otherwise noted (e.g., in question (g)). Packet switching is used unless otherwise stated (e.g., as in question (f)). Assume that the propagation speed is 2×10^8 meters/sec, and the distance between source and destination is $10,000km$.

1) Initially suppose there is only one link between source and destination. Also suppose that the entire MP3 file is sent as one packet. The transmission delay is:
- ☐ 50 milliseconds
- ☐ 3.05 seconds
- ☐ **3 seconds**
- ☐ none of the above

Reason: The transmission delay is $L/R = (30 \times 10^6)/(10 \times 10^6) = 3seconds$.

2) Referring to the above question, the end-to-end delay (defined from the time S starts transmitting the first bit, until D receives the last bit of the MP3 file) is
- ☐ 6 seconds
- ☐ 3 seconds
- ☐ **3.05 seconds**
- ☐ none of the above

Reason: The end-to-end delay is $d_{trans} + d_{prop} = 3 + 0.05 = 3.05seconds$

3) Referring to the above question, how many bits will the source have transmitted when the first bit arrives at the destination.
- ☐ 1 bit
- ☐ 30,000,000 bits
- ☐ **500,000 bits**
- ☐ none of the above

Reason: Basically it's the bandwidth-delay product, which is $(10 \times 10^6) \times 0.05 = 500,000bits$

4) Now suppose there are two links between source and destination, with one router connecting the two links. Each link is 5,000 km long. Again suppose the MP3 file is sent as one packet. Suppose there is no congestion, so that the packet is transmitted onto the second link as soon as the router receives the entire packet. The end-to-end delay is

☐ 6.1 seconds
☐ **6.05 seconds**
☐ 3.05 seconds
☐ none of the above

Reason: The transmission delay is same, but the propagation delay on each link is $(5000 \times 10^3)/(2 \times 10^8) = 0.025sec$. Thus, the end-to-end delay is $2(d_{trans} + d_{prop}) = 2(3 + 0.025) = 6.05sec$.

5) Assume again two links with the previous parameters, but now suppose that the MP3 file is broken into 3 packets, each of 10 Mbits. Ignore headers that may be added to these packets. Also ignore router processing delays. Assuming store and forward packet switching at the router, the total delay is

☐ 3.05 seconds
☐ 6.05 seconds
☐ **4.05 seconds**
☐ none of the above

Reason: The transmission delay for 1 packet = $(10 \times 10^6)/(10 \times 10^6) = 1seconds$. The last bit of the first packet arrives at receiver after $2(d_{trans} + d_{prop}) = 2(1 + 0.025) = 2.05sec$. The remaining two packets arrives after $2 \times d_{trans} = 2sec$. Hence, the end-to-end delay is $2.05sec + 2sec = 4.05seconds$

6) Now suppose there is only one link between source and destination, and there are 10 TDM channels in the link. The MP3 file is sent over one of the channels, using circuit switching. Ignore setup time, e.g., to allocate channels to connections. The end-to-end delay is

☐ **30.05 seconds**
☐ 30 seconds
☐ 300 microseconds
☐ none of the above

Reason: Because of TDM, each user will get transmission rate of 1 Mbps. Now, the transmission delay will be $(30 \times 10^6)/(1 \times 10^6) = 30seconds$ and the propagation delay stays same. Therefore, the end-to-end delay is $30.05seconds$.

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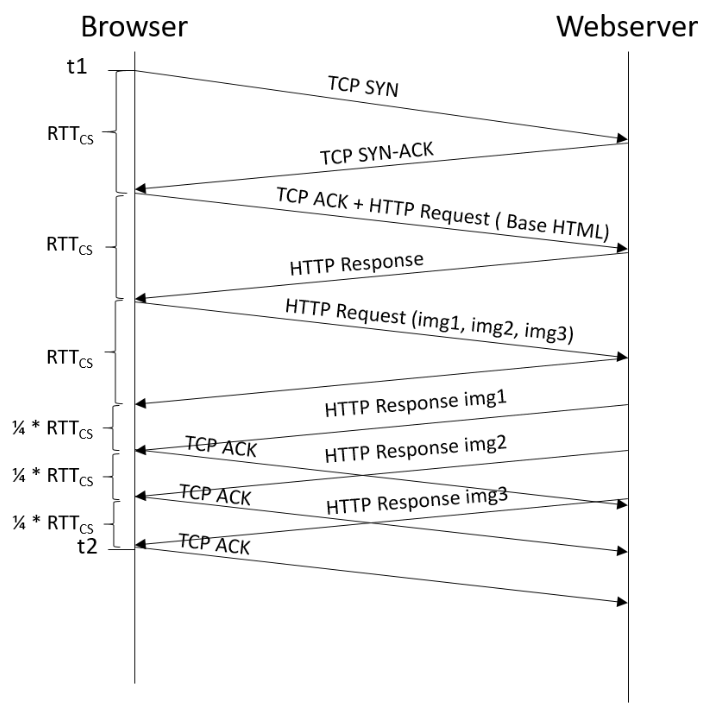
Question 10. *(20 points)* Assume that the MIT webpage consists of a base HTML file (http://www.mit.edu/index.html) and 3 images. Let RTT_{cs} be the RTT between my laptop and the MIT webserver. The base HTML file is small, with negligible transmission delay. However, the images are large: (an HTTP response for) each image fits in exactly one TCP segment (each with the maximum segment size, MSS Bytes) and transmission delay $1/4 \times RTT_{cs}$. The transmission delays of all other messages are negligible. (Additional simplifying assumptions: no packet is lost; ignore the TCP window effect; ignore any processing delays; all TCP segments have either negligible or maximum (MSS) size, as specified above.)
Consider that your browser uses persistent HTTP with pipelining. Complete the diagram below. Indicate the TCP connection establishment segments and the HTTP messages, exchanged right after DNS is resolved ($t1$) and before the webpage is ready to be displayed by the browser (time $t2$). How long is this delay $t2 - t1$?



Total time until the browser has all the information to display the page is: $t2 - t1 = (15/4) \times RTT_{CS}$. Please note that at that point the TCP connection is not complete (still ACKs are sent and the connection is not closed).
Since the protocol is persistent, the same TCP connection is used to get all the objects. Three different HTTP requests are sent, one for each of img1. img2, img3, serially, one after the other. However, because their transmission delay is assumed negligible, they are drawn one on top of the other on the figure. In contrast, the HTTP responses (with img1, img2, img3 respectively, and also send serially over the the same TCP connection) have non negligible transmission delay $RRT_{CS}/4$, thus they are easier to distinguish on the figure.
Finally notice that the third part of the TCP handshake (TCP ACK set to 1) also carries data (the HTTP Request for index.html) in the same packet.

Q&A

Question 11. *(10 points)* What is the main source of packet loss in packet switched wired networks?

Solution: Packets are **dropped** at routers with full queues.

Question 12. *(10 points)* List and describe the layers of the Internet stack.

Solution:

- Application
- Transport
- Network
- Link

Description and additional details are in book, slides, web, discussion,...

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Question 13. *(10 points)* Is in real real-time multimedia applications preferable to use Forward Error Correction or packet retransmission? Briefly explain why.

Solution: Multimedia applications have either strict (video calls) or fixed (video streaming) delay constraint. This implies a strict requirement over jitter: delay cannot vary too much.

Therefore FEC is better since retransmission of packets can increase delay and jitter in unpredictable ways. FEC decreases throughput BUT even slowed down, the connection is more reliable and setting N (see details of FEC).

Question 14. *(10 points)* For each of the following applications, describe loss, delay, jitter and throughput requirements (QoS characterization) as **HIGH, MEDIUM** or **LOW** (HIGH is strict requirement):
Solution: Many entries can be either one or another: we are accepting both, when stated. Example: MEDIUM (H): the more appropriate is MEDIUM, but we accept HIGH as well.

	File sharing	**Email**	**HTTP**	**Video call**
Loss	MEDIUM	MEDIUM (H)	MEDIUM (H)	LOW
Delay	LOW	LOW	MEDIUM	HIGH
Jitter	LOW	LOW	LOW (M)	HIGH
Throughput	HIGH	LOW(M)	MEDIUM	HIGH