CS 421: Computer Networks **FALL 2013**

MIDTERM
November 20, 2013
120 minutes

Name: ______________________________

Student No: ______________________________

Show all your work very clearly. Partial credits will only be given if you carefully state your answer with a reasonable justification.

Q1	
Q2	
Q3	
TOT	

1)

a) (5 pts) Why does DNS prefer UDP rather than TCP?

b) (5 pts) Suppose you want to transfer a large file to a very large number of users. Which of the two application architectures will you prefer: client-server or peer-to-peer? Why?

c) (5 pts) Considering the two services provided by the Internet, namely TCP and UDP, which one is more suitable for applications that require minimum bandwidth guarantee, e.g., Internet telephony or video conferencing? Why?

d) Assume that there are 10 established TCP connections between a given client host and a given Web server host.

   i) (3 pts) How many sockets are needed at the client host to support these connections? Explain your answer.

   ii) (3 pts) How many sockets are needed at the Web server host to support these connections? Explain your answer.

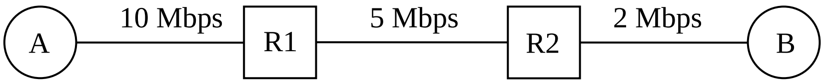
e) (5 pts) Describe a networking scenario where you will prefer Go-Back-N rather than Selective Repeat. Explain your reasoning.

f) (5 pts) What is the difference between congestion control and flow control?

g) (5 pts) Why does TCP consider DevRTT, i.e., standard deviation of RTT measurements, in the calculation of the retransmission timeout?

2)

a) (8 pts) Assume that there are 3 links on a path connecting hosts A and B passing through routers R1 and R2 as shown in the following figure. Each link has a distance of 600 km and the transmission rate of each link is shown in the figure. We are transmitting a file composed of four packets from node A to node B using datagram packet switching. **Packets are transmitted from A every 5msec**, i.e., packet transmissions at A start at 0,5,10,15 msec. Each packet has a length of 1250 Bytes including all headers. Assume that the processing and queuing delays in each intermediate node are negligible and the propagation speed is 2×10^5 km/s. Calculate the total delay incurred in transferring the file from host A to host B.



b) (10 pts) Consider a connection with a **30 msec** roundtrip, delay (including all delays incurred within the network, but excluding the packet transmission time of the sender). We want to transfer a file composed of **20 segments** (with sequence numbers from 1 to 20), where each segment has a transmission time of **1 msec**. Assume that ACK segments have negligibly small size and there is no processing delay at the receiver. Assume also that the processing delay at the sender after an ACK is received is negligible. We assume that the communication between the sender and receiver is full duplex, i.e., sender can send data segments while receiving an ACK segment. **Selective Repeat** protocol is used with a window size of **N = 12** segments. Assume that all data segments are received correctly while the **first transmissions** of the **data segments** with sequence numbers **4** and **14,** and **ACK segment** with acknowledgment number **19** are errored, whereas **all other data** and **ACK segments are fully reliable**. The timeout for each data segment is set to **40 msec starting from the end of the transmission of the segment**. How much time is required to complete the transfer of the whole file and receive the **final ACK** at the sender?

c) 1000 Byte segments are transmitted over a 10 Mbps (10×10^6 bits/sec) connection which has a 100 msec round-trip delay.

   i) (5 pts) What is the minimum window size, **in Bytes**, necessary for this connection to achieve 100% sender utilization, i.e., $U_{sender}=1$?

   ii) (5 pts) Is it possible for this connection to achieve 100% sender utilization if it uses TCP without window scaling? Explain your reasoning.

3)

a) (8 pts) SampleRTT values measured by a TCP connection have the following probability distribution:

$$\text{SampleRTT}=\begin{cases} 10\text{msec} & \text{with probability } 0.8 \\ 100\text{msec} & \text{with probability } 0.2 \end{cases}$$

Assume that no packets are lost for this connection. Recall that the retransmission timeout is given by TimeOut = EstimatedRTT+4xDevRTT. Assume that Estimated RTT and DevRTT values computed by TCP according to the exponential weight moving averages are such that they are equal to their true (ensemble) averages, i.e., EstimatedRTT = E[SampleRTT] and DevRTT = E[|SampleRTT-EstimatedRTT|]. Do you expect any timeouts for this connection? Justify your answer.

b) Assume that there is a TCP connection between two processes, one running at Host A and the other at Host B in order to transfer a very large file from Host A to Host B. The slowest link in the network along this connection has a transmission rate of 20 Mbps. The application running at Host A can write data to its TCP send buffer at 50 Mbps, whereas the application running at Host B can read data from its TCP receive buffer at 30 Mbps.

   i) (4 pts) What is the average rate of data transfer from the application at Host A to the application at Host B? Justify your answer.

   ii) (4 pts) Which of the TCP flow control and congestion control algorithms limits the data transfer rate in your answer above? How does this algorithm limit the rate?

c) Suppose that a file composed of 9 segments each with size 1000Bytes will be transferred over a TCP connection with a round-trip delay of 10msec and bandwidth of 8Mbps, i.e., $8\text{x}10^6$ bits/second. The retransmission timeout for TCP is set to 20 msec.

   i) (4 pts) Assume that the initial sequence number for the TCP connection is 5000. What are the sequence numbers of the nine segments?

   ii) (8 pts) Assume that the **fourth data segment transmitted is lost** in its first transmission and no other data or ACK segments are lost or errored during the file transfer. We assume that out-of-order segments are buffered at the receiver. Further assume that the slow start threshold (ssthresh) at the beginning of the TCP connection is infinitely large. How long does it take to transmit the entire file starting from the connection establishment until the final ACK is received?

d) (8 pts) Suppose that you are given the task of building a reliable communication link between Earth and Mars, where TCP is going to be used as the transport layer protocol. Assume that layers 1 and 2 (the physical and link layers) have already been designed so you do not need to worry about how to physically get a packet from Earth to Mars. In this task, you have the flexibility of changing TCP's congestion control algorithm, e.g., how you manage the congestion window, how you handle duplicate ACKs, how you set the timeout, etc. However, you cannot change other things in TCP outside the congestion control algorithm, e.g., you cannot change the TCP header or the flow control algorithm. Propose **two modifications** to TCP's congestion control algorithm so that you can obtain a higher throughput over the Earth-Mars connection. You need to justify each proposal. Make sure that your proposals do not harm the reliability of TCP. ***Hint:*** The distance between Earth and Mars can be as large as 380 million kilometers, which corresponds to a RTT of more than 40 minutes.