CS 421: Computer Networks **FALL 2010**

MIDTERM
November 10, 2010
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 is circuit switching rather than packet switching used for traditional telephony service?

b) (5 pts) Suppose that an HTTP server is holding *N* connections to *N* different clients at a particular time. At least how many sockets are open at the server?

c) (5 pts) An FTP client connected to an FTP server and uploaded 5 files and then disconnected from the server. How many TCP connections, in total, have been established during this session between the server and the client?

d) (5 pts) Why doesn't an email user agent directly contact the recipient's SMTP server?

e) (7 pts) Suppose you are developing a Voice over IP (VOIP) application. Which of the two transport layer services provided by the Internet, namely TCP and UDP, will you prefer to use for your application. Fully justify your answer by discussing the advantages and disadvantages of using the two alternatives.

f) (8 pts) We have the following UDP sender program written in Java. The program works correctly. There are no syntax or run-time errors. Study the program very carefully, understand what it is doing, and answer the questions below based on that. Assume we are running this program once. Assume packets are not re-ordered in the outgoing link and in the network.

```
import java.io.*;
import java.net.*;

class client {
    public static void main (String args[]) throws Exception
    {
     byte[] sendData = {1,1,1,1};

     DatagramSocket s = new DatagramSocket(9000);
     for (int hostid = 1; hostid <= 100; ++hostid) {
           InetAddress IPaddr =
               InetAddress.getByName("139.179.15." + hostid);

           DatagramPacket sendPacket = new DatagramPacket(sendData,
                  sendData.length, IPaddr, 9500);

           s.send(sendPacket);
     }

     s.close();
    }
}
```

i) What are the source port number and destination port number in the packets sent by this program?

ii) What is the IP address of the destination machine to where the 3rd packet is sent?

iii) What is the size of data in each sent packet?

iv) How many DNS queries are generated for this program during its lifetime?

2)

a) Assume that there are 3 links on a path connecting nodes A and B, where each link has a distance of 200 km and a transmission rate of 10 Mbps. We are transmitting a file composed of three packets from node A to node B using datagram packet switching. Each packet has a length of 1000 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.

   i. (6 pts) Calculate the total delay incurred in transferring the file from node A to node B.

   ii. (4 pts) What is the average rate of data transfer, in bps, in the above file transfer assuming that each packet contains a 40 Byte header?

b) (7 pts) Suppose you use the Go-Back-N protocol for reliable data delivery over a connection with an end-to-end delay of 30 msec and with a window size of 2^{16} Bytes. The connection is perfectly reliable with a transmission rate of 100 Mbps. What is the maximum possible sender's utilization for this connection?

c) (10 pts) Consider a 1 Mbits/sec connection with a 14 msec one-way propagation delay, i.e., 28 msec roundtrip propagation delay. We want to transfer a file of size 9000 bytes. Each segment has a total size of 625 bytes including the 25-bytes header. When there is data to be transmitted, each segment contains the maximum number of bytes. Assume that ACK segments have a size of 125 bytes and there is a processing delay of 1 msec after a segment is fully received at the receiver until the transmission of the corresponding ACK is started. Assume 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 fully duplex, i.e., sender can send data segments while receiving an ACK segment. **Selective Repeat** protocol is used with a window size of N = 8 segments. Assume that **every 6th data packet is lost whereas ACK packets are fully reliable**. The timeout for each 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?

3)

a) Consider the RTT estimation algorithm for setting the retransmission TimeOut used by TCP as we discussed in the class:

$$\text{EstimatedRTT}(k+1) \leftarrow (1 - \alpha)\ \text{EstimatedRTT}(k) + \alpha\ \text{SampleRTT}(k+1)$$

for k = 1,2,…, where EstimatedRTT(1) = SampleRTT(1). Assume that $\alpha = 1/8$. Suppose this algorithm is used for two separate TCP connections for estimating the RTT.

i) (6 pts) For the first connection, RTT of the first segment in its first transmission attempt is 16 msec, and RTT of the second segment in its first transmission attempt is 8 msec. What is EstimatedRTT(2) generated by the RTT estimation algorithm?

ii) (6 pts) For the second connection, the first segment of the TCP connection times out in its first transmission attempt, RTT of this first segment in its second transmission attempt is 20 msec, RTT of the second segment in its first transmission attempt is 16 msec, and RTT of the third segment in its first transmission attempt is 8 msec. What is EstimatedRTT(2) generated by the RTT estimation algorithm?

b) (6 pts) Suppose two applications running at Host A and Host B are communicating using a TCP connection. Assume by time t, Host B has already received all the bytes up to and including byte 300 from Host A and the application running on Host B has read all these bytes from the TCP (socket) buffer at Host B (i.e., all bytes 1...300 are received and given to the application). After time t, Host A is sending 3 TCP segments back-to-back to Host B. Segment 1 has 70 bytes of data in it, segment 2 has 40 bytes of data and segment 3 has 50 bytes of data in it (these are the size of data, not including the headers). Assume, segment 2 arrives first to Host B, then segment 1, and then segment 3. After each segment is received, Host B sends an ACK immediately. Assume there are no packet/ACK losses and no timeouts happened. Assume the size of TCP receive-buffer at Host B for this connection is 2000 bytes. Assume, after time t, the application at Host B will not read data from the corresponding socket before all ACKs corresponding to these 3 segments are sent. What will the acknowledgement and receive-window field values in the three ACK packets sent from Host B to host A?

c) (6 pts) We learned in the class that during the "slow start" phase, TCP's congestion window size, CongWin, is increased by the maximum segment size (MSS) after each received ACK which acknowledges new data, independent of the number of bytes acknowledged in the incoming ACK. However, RFC 5681 recommends that upon receiving an ACK, CongWin is increased by min(MSS,N), where N is the number of previously unacknowledged bytes acknowledged in the incoming ACK. What might be a reason for this recommendation?

d) (8 pts) Assume that the congestion window of a TCP flow was 20 segments long when a timeout occurred. Assume that there are no segments or acknowledgments of this flow that were in transit when the timeout occurred. The round trip delay for the flow, i.e., the time from the completion of a segment transmission until the corresponding ACK is full received is equal to 30 msec and the transmission time for a segment is 1 msec. The receive window is fixed at 100 segments for the entire duration of the connection. What is the minimum time the flow will spend in the Slow Start phase after the timeout before reaching the Congestion Avoidance phase?

e) (6 pts) Suppose that there are two TCP connections sharing a bottleneck link which has a bandwidth of 10 Mbps, i.e., all other links that these two connections pass through have bandwidths significantly larger than 10 Mbps. The first connection has a round-trip delay of 5 msec, whereas the second connection has a round-trip delay of 20 msec. Calculate the average throughputs achieved by each connection.