Notes: There are five questions in this assignment. Each question has 10 points.

1. (10pt.) Please describe the TCP closing sequence. Please also justify why these steps are required.

**Answer:**

- **Step 1** Client end system sends TCP FIN control segment to server
  - This step is required to signal that the client wants to close.

- **Step 2** Server receives FIN, replies with ACK. half closed
  - This step tells the client that the server knows it wants to close.

- **Step 3** Client receives FIN, half closed, wait for server to close. Server finishes sending data, also ready to close:
  - The server is deallocating resources to start closing.

- **Step 4** Server sends FIN.
  - This tells the client that the server is about to close this TCP connection.

- **Step 5** Client receives FIN, replies with ACK. connection fully closed
  - Clients knows server has finished deallocating resources.

- **Step 5** Server, receives ACK, connection fully closed
  - This is required so the server can know that the clients is aware that this TCP connection will be closed.
2. (10pt.) Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how? Hint: Consider the reliable in-order delivering data in application layer. What do you need to include the application layer protocol?

**Answer:**

Yes.
The application developer may not want his application to use TCP’s congestion control. This is because TCP congestion control can throttle the application’s sending rate during congestion. So, the application developer can put reliable data transfer into the application layer protocol which will require significant amount of work and debugging on the developer’s part.

To ensure reliability, the developer will need to add at the application layer:
- Ack functions to help make sure data segments are received.
- Sequencing to help order the data segments.
- Flow control to help throttle the flow of data.

Grading:
- 2 point for Yes.
- 2 if you have not shown how the reliability could be achieve in UDP.
- 2 if a reason for using UDP over TCP is not mentioned.

3. (10pt.) Consider two hosts are linked by a 4Mbps ($4 \times 10^6$) channel and RTT is 0.04 sec. Assume the size of each packet is 8000 bits. Answer the following questions for ARQ schemes:

a) Assume that the link is error-free, what is the possible maximum rate of transmission for Stop-and-wait, GBN, and SR, respectively? Why?

b) For GBN and SR, in order to allow sender to continuously send packets without any waiting, what is the minimum window size in terms of the number of packets?

c) For b), what should be the minimum number of bits for the sequence number for GBN and SR, respectively?

d) Suppose that we are continuously transmitting packets end-to-end start from the 3rd packet, and the 7th packet is lost. Assume there is no other packet lost or ACK lost. For stop-and-wait, GBN, and SR, which packets need to be retransmitted?

**Answer:**

a) Stop-and-wait
Since Stop-and-wait need to wait for the ACK of each packet before transmitting the next one. The maximum rate of transmission will be the time taken to transmit 1 packet and receive the ACK. i.e

$$Rate_{transmission} = \frac{8000}{4000000} + 0.04 = 0.042$$

GBN:
If the window size is large enough, we can find the maximum number of packets that can be transmitted in that window size:

$$Max\_packets = \frac{4000000 \times 0.04}{8000} = 20\_packets$$

Therefore the maximum rate can be 4Mbps.
If the window size is not larger than 20 packets:
Max rate = \frac{4000000 \times \text{window size}}{20}

SR
Max rate of transmission is same as GBN
b) Since the RTT is 0.04 sec, the windows size will be the maximum number of packets.

\[ \text{Max packets} = \frac{4000000 \times 0.04}{8000} = 20 \text{ packets} \]

c) The minimum number of bits for the sequence number for GBN should be 5 where \(2^5 = 32\), which is greater than 20.
In SR, the sequence number should be at least twice the window size to avoid possible out of order transmissions. The number of sequence number should be at least 40, where \(2^6 = 64\) is greater than 40.

d) • In stop-and-wait, only the 7th packet needs to be re-transmitted.
• In GBN, the 7th packet and all the following packets being transmitted in the window will have to be re-transmitted.
• For SR, only the 7th packet needs to be re-transmitted.

Grading:
a) 2 points for Stop-and-wait, 1 point for GBN and 1 point for SR.
b) 1 point
c) 2 points
d) 3 points
-1 if rate is not calculated correctly.
-1 if calculation errors.

4. (10pt.) (a) What happened during round 4, 6-7, 10-11, 13?
(b) Can you write down the CongWin & Threshold values at each round?

Answer:
a) At round 4, the congestion window has reached a threshold, and hence moving into congestion avoidance phase.
At rounds 6 - 7, packet gets lost indicated by three duplicate ACKs.
At round 10 - 11, a timeout occurs going into a re-transmission timeout phase.
At round 13, slow start and the congestion window grows exponentially.

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Grading:
4 points for section a and 6 points for section b.
-1 if minor errors in the table.
-2 if lost events are not indicated.

5 (10pt) Can you complete the finite state machine of client for 3-way handshake as listed in Slide 21 of Transport Layer: Part I?

Answer:
Grading:
Each event and action is 1 point.
-1 if event or action is not correct.