Computer Networking Homework 4

Due date: 2018/12/18 13:00 PM

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The file should be in pdf format and named with YourSchoolNumber_hw#.pdf

1. (5%) Why is that voice and video traffic is often sent over TCP rather than UDP in today's Internet?(Hint: The answer we are looking for has nothing to do with TCP's congestion-control mechanism.)

2. (5%) Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how?

3. (5%) Suppose that the roundtrip delay between sender and receiver is constant and known to the sender. Would a timer still be necessary in protocol rdt 3.0, assuming that packets can be lost? Please explain your answer.

4. (5%) Suppose Host A sends two TCP segments back to back to Host B over a TCP connection. The first segment has sequence number 90; the second has sequence number 110.
   (a) (2.5%) How much data is in the first segment?
   (b) (2.5%) Suppose that the first segment is lost but the second segment arrives at B. In the acknowledgement that Host B sends to Host A, what will be the acknowledgment number?

5. (5%) UDP and TCP use 1s complement for their checksums. Is it possible that a 2-bit error will go undetected? Please briefly explain your answer or give a simple example.

6. (20%) Consider the Go-Back-N (GBN) protocol with a sender window size of 3 and a sequence number range of 1024. Suppose that at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium does not reorder messages. Answer the following questions:
   (a) (10%) What are the possible sets of sequence numbers inside the sender's window at time t? Justify your answer.
   (b) (10%) What are all possible values of the ACK field in all possible messages currently propagating back to the sender at time t? Justify your answer.

7. (15%) Consider two network entities, A and B, which are connected by a perfect bi-directional channel (i.e., any message sent will be received correctly; the channel will not corrupt, lose, or re-order packets). A and B are to deliver data messages to each other in an alternating manner: First, A must deliver a message to B, then B must deliver a message to A, then A must deliver a message to B and so on. If an entity is in a state where it should not attempt to deliver a message to the other side, and there is an event like `rdt_send(data)` call from above that attempts to pass data down for transmission to the other side, this call from above can simply be ignored with a call to `rdt_unable_to_send(data)`, which informs the higher layer that it is currently not able to send data. [Note: This simplifying assumption is made so you don't have to worry about buffering data.]

   Draw a FSM specification for this protocol (one FSM for A, and one FSM for B!). Note that you do not have to worry about a reliability mechanism here; the main point of this question is to create a FSM specification that reflects the synchronized behavior of the two entities. You should use the following events and actions that have the same meaning as protocol rdt1.0 in page 26 of the course slide: `rdt_send(data)`, packet = `make_pkt(data)`, `udt_send(packet)`, `rdt_rcv(packet)`, `extract(packet,data)`, `deliver_data(data)`. Make sure your protocol reflects the strict alternation of sending between A and B. Also, make sure to indicate the initial states for A and B in your FSM descriptions.

8. (40%) Consider the Figure below

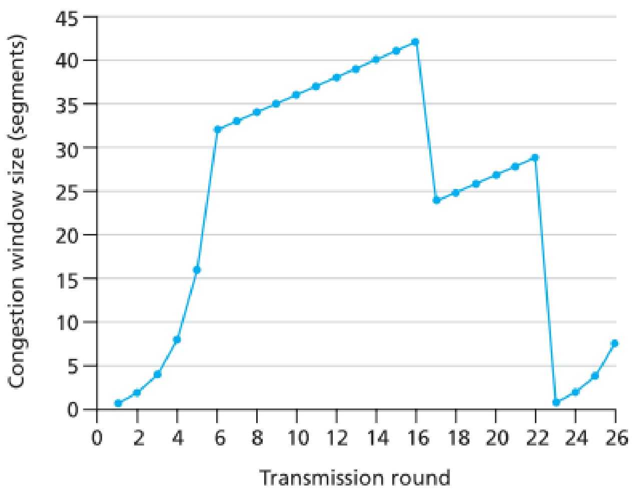

Figure 3.58 ♦ TCP window size as a function of time

Assuming TCP Reno is the protocol experiencing the behavior shown above, answer the following questions. In all cases, you should provide a short discussion justifying your answer.

(a) (3%) Identify the intervals of time when TCP slow start is operating.

(b) (3%) Identify the intervals of time when TCP congestion avoidance is operating.

(c) (3%) After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

(d) (3%) After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

(e) (4%) What is the initial value of *ssthresh* at the first transmission round?

(f) (4%) What is the value of *ssthresh* at the 18th transmission round?

(g) (4%) What is the value of *ssthresh* at the 24th transmission round?

(h) (4%) During what transmission round is the 70th segment sent?

(i) (4%) Assuming a packet loss is detected after the 26th round by the receipt of a triple duplicate ACK, what will be the values of the congestion window size and of *ssthresh*?

(j) (4%) Suppose TCP Tahoe is used (instead of TCP Reno), and assume that triple duplicate ACKs are received at the 16th round. What are the *ssthresh* and the congestion window size at the 19th round?

(k) (4%) Again suppose TCP Tahoe is used, and there is a timeout event at 22nd round. How many packets have been sent out from 17th round till 22nd round, inclusive?