Capítulo 3 – Kurose – Exercícios

1) Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how?

2) Suppose that a Web server runs in Host C on port 80. Suppose this Web server uses persistent connections, and is currently receiving requests from two different Hosts, A and B. Are all of the requests being sent through the same socket at Host C? If they are being passed through different sockets, do both of the sockets have port 80? Discuss and explain.

3) 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? Explain.

4) 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. How much data is in the first segment?
b. Suppose that the first segment is lost but the second segment arrives at B. In the acknowledgment that Host B sends to Host A, what will be the acknowledgment number?

5) Suppose Client A initiates a Telnet session with Server S. At about the same time, Client B also initiates a Telnet session with Server S. Provide possible source and destination port numbers for
a. The segments sent from A to S.
b. The segments sent from B to S.
c. The segments sent from S to A.
d. The segments sent from S to B.
e. If A and B are different hosts, is it possible that the source port number in the segments from A to S is the same as that from B to S?
f. How about if they are the same host?

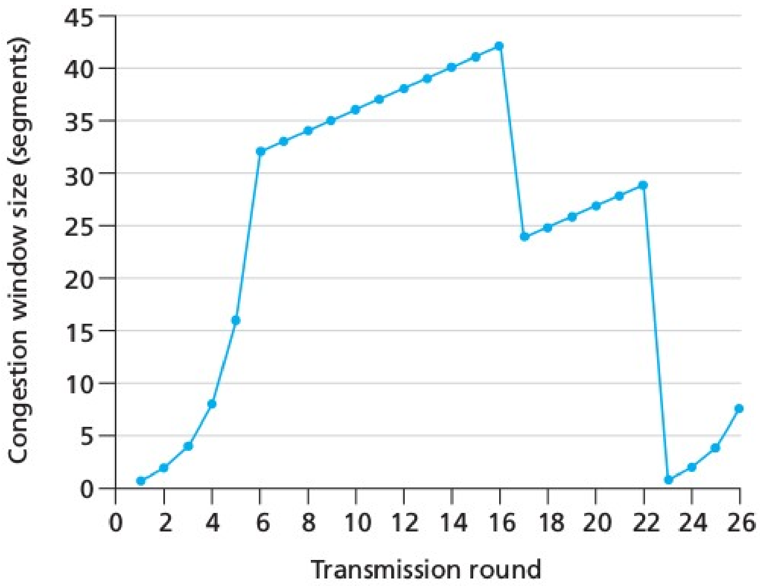
6) In protocol rdt3.0, the ACK packets flowing from the receiver to the sender do not have sequence numbers (although they do have an ACK field that contains the sequence number of the packet they are acknowledging). Why is it that our ACK packets do not require sequence numbers?

7) Suppose an application uses rdt 3.0 as its transport layer protocol. As the stop-and-wait protocol has very low channel utilization (shown in the cross-country example), the designers of this application let the receiver keep sending back a number (more than two) of alternating ACK 0 and ACK 1 even if the corresponding data have not arrived at the receiver. Would this application design increase the channel utilization? Why? Are there any potential problems with this approach? Explain.

8) In Section 3.5.4, we saw that TCP waits until it has received three duplicate ACKs before performing a fast retransmit. Why do you think the TCP designers chose not to perform a fast retransmit after the first duplicate ACK for a segment is received?

9) Consider Figure 1 bellow. 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. Identify the intervals of time when TCP slow start is operating.
b. Identify the intervals of time when TCP congestion avoidance is operating.
c. After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

d. After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
e. What is the initial value of ssthresh at the first transmission round?
f. What is the value of ssthresh at the 18th transmission round?
g. What is the value of ssthresh at the 24th transmission round?
h. During what transmission round is the 70th segment sent?
i. 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. 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. 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?



10) Host A is sending an enormous file to Host B over a TCP connection. Over this connection there is never any packet loss and the timers never expire. Denote the transmission rate of the link connecting Host A to the Internet by R bps. Suppose that the process in Host A is capable of sending data into its TCP socket at a rate S bps, where $S = 10 \cdot R$. Further suppose that the TCP receive buffer is large enough to hold the entire file, and the send buffer can hold only one percent of the file. What would prevent the process in Host A from continuously passing data to its TCP socket at rate S bps? TCP flow control? TCP congestion control? Or something else? Elaborate.