FAST - National University of Computer & Emerging Sciences, Karachi

Computer Network (CS 3001)

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Practice questions from chapter 3

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1. Why is it 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. Consider Figure 3.5. What are the source and destination port values in the segments flowing from the server back to the clients' processes? What are the IP addresses in the network-layer datagrams carrying the transport-layer segments?

3. UDP and TCP use 1s complement for their checksums. Suppose you have the following three 8-bit bytes: 01010011, 01100110, 01110100. What is the 1s complement of the sum of these 8-bit bytes? (Note that although UDP and TCP use 16-bit words in computing the checksum, for this problem you are being asked to consider 8-bit sums.) Show all work. Why it that UDP takes the 1s is complement of the sum; that is, why not just use the sum? With the 1s complement scheme, how does the receiver detect errors? Is it possible that a 1-bit error will go undetected? How about a 2-bit error?

4. Suppose you have the following 2 bytes: 01011100 and 01100101. What is the 1s complement of the sum of these 2 bytes? b. Suppose you have the following 2 bytes: 11011010 and 01100101. What is the 1s complement of the sum of these 2 bytes? c. For the bytes in part (a), give an example where one bit is flipped in each of the 2 bytes and yet the 1s complement doesn't change.

5. 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?

6. Give a trace of the operation of protocol rdt3.0 when data packets and acknowledgment packets are garbled. Your trace should be similar to that used in Figure 3.16.

7. Consider a channel that can lose packets but has a maximum delay that is known. Modify protocol rdt2.1 to include sender timeout and retransmit. Informally argue why your protocol can communicate correctly over this channel.

8. Consider the rdt 3.0 protocol. Draw a diagram showing that if the network connection between the sender and receiver can reorder messages (that is, that two messages propagating in the medium between the sender and receiver can be reordered), then the alternating-bit protocol will not work correctly (make sure you clearly identify the sense in which it will not work correctly). Your diagram should have the sender on the left and the receiver on the right, with the time axis running down the page, showing data (D) and acknowledgment (A) message exchange. Make sure you indicate the sequence number associated with any data or acknowledgment segment.

9. Draw sequence diagram of GBN and SR with N=4.

10. Suppose that the five measured SampleRTT values (see Section 3.5.3) are 106 ms, 120 ms, 140 ms, 90 ms, and 115 ms. Compute the EstimatedRTT after each of these SampleRTT values is obtained, using a value of $\alpha = 0.125$ and assuming that the value of EstimatedRTT was 100 ms just before the first of these five samples were obtained. Compute also the DevRTT after each sample is obtained, assuming a value of $\beta = 0.25$ and assuming the value of DevRTT was 5 ms just before the first of these five samples was obtained.

11. Consider the TCP procedure for estimating RTT. Suppose that $\alpha = 0.1$. Let SampleRTT1 be the most recent sample RTT, let SampleRTT2 be the next most recent sample RTT, and so on.
    a. For a given TCP connection, suppose four acknowledgments have been returned with corresponding sample RTTs: SampleRTT4, SampleRTT3, SampleRTT2, and SampleRTT1. Express EstimatedRTT in terms of the four sample RTTs.
    b. Generalize your formula for n sample RTTs.
    c. For the formula in part (b) let n approach infinity. Comment on why this averaging procedure is called an exponential moving average.

12. Host A and B are communicating over a TCP connection, and Host B has already received from A all bytes up through byte 126. Suppose Host A then sends two segments to Host B back-to-back. The first and second segments contain 80 and 40 bytes of data, respectively. In the first segment, the sequence number is 127, the source port number is 302, and the destination port number is 80. Host B sends an acknowledgment whenever it receives a segment from Host A.

13. In our description of TCP in Figure 3.53, the value of the threshold, ssthresh, is set as ssthresh=cwnd/2 in several places and ssthresh value is referred to as being set to half the window size when a loss event occurred. Must the rate at which the sender is sending when the loss event occurred be approximately equal to cwnd segments per RTT? Explain your answer. If your answer is no, can you suggest a different manner in which ssthresh should be set?

14. Host A and B are directly connected with a 100 Mbps link. There is one TCP connection between the two hosts, and Host A is sending to Host B an enormous file over this connection. Host A can send its application data into its TCP socket at a rate as high as 120 Mbps but Host B can read out of its TCP receive buffer at a maximum rate of 50 Mbps. Describe the effect of TCP flow control.

15. Compare GBN, SR, and TCP (no delayed ACK). Assume that the timeout values for all three protocols are sufficiently long such that five consecutive data segments and their corresponding ACKs can be received (if not lost in the channel) by the receiving host (Host B) and the sending host (Host A) respectively. Suppose Host A sends five data segments to Host B, and the second segment (sent from A) is lost. In the end, all five data segments have been correctly received by Host B.
    a. How many segments has Host A sent in total and how many ACKs has Host B sent in total? What are their sequence numbers? Answer this question for all three protocols.
    b. If the timeout values for all three protocol are much longer than 5 RTT, then which protocol successfully delivers all five data segments in shortest time interval?

16. In our description of TCP in Figure 3.53, the value of the threshold, ssthresh, is set as ssthresh=cwnd/2 in several places and ssthresh value is referred to as being set to half the window size when a loss event occurred. Must the rate at which the sender is sending when the loss event occurred be approximately equal to cwnd segments per RTT? Explain your answer. If your answer is no, can you suggest a different manner in which ssthresh should be set?

17. Consider Figure 3.61. 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?

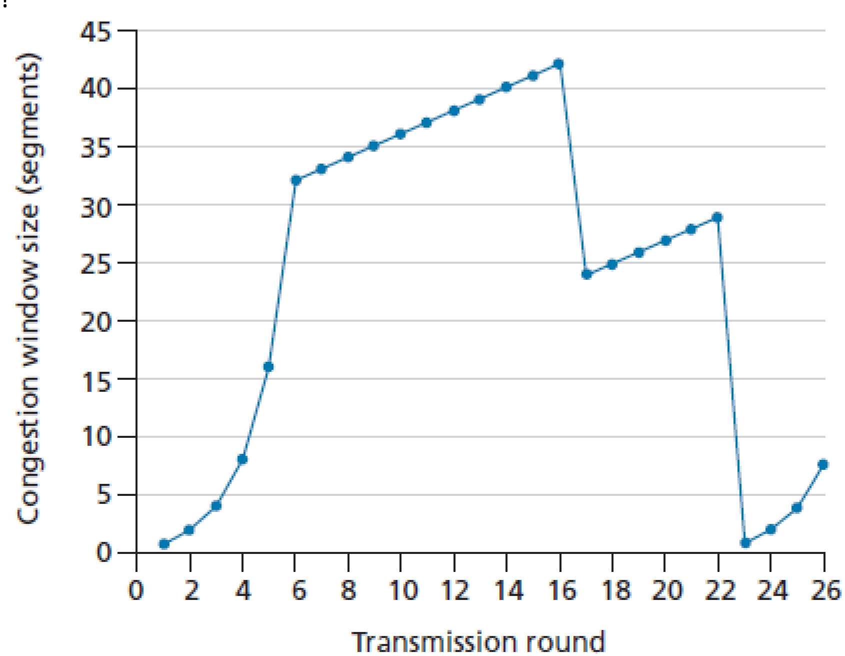

Figure 3.61 ♦ TCP window size as a function of time

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?

18. Consider sending a large file from a host to another over a TCP connection that has no loss. a. Suppose TCP uses AIMD for its congestion control without slow start. Assuming cwnd increases by 1 MSS every time a batch of ACKs is received and assuming approximately constant round-trip times, how long does it take for cwnd increase from 6 MSS to 12 MSS (assuming no loss events)? b. What is the average throughput (in terms of MSS and RTT) for this connection up through time = 6 RTT?