Chapter-3 Assigned/ Practice - Problems

Problem 22

- a) Here we have a window size of N=3. Suppose the receiver has received packet k-1, and has ACKed that and all other preceding packets. If all of these ACK's have been received by sender, then sender's window is [k, k+N-1]. Suppose next that none of the ACKs have been received at the sender. In this second case, the sender's window contains k-1 and the N packets up to and including k-1. The sender's window is thus [k-N,k-1]. By these arguments, the senders window is of size 3 and begins somewhere in the range [k-N,k].
- b) If the receiver is waiting for packet k, then it has received (and ACKed) packet k-1 and the N-1 packets before that. If none of those N ACKs have been yet received by the sender, then ACK messages with values of [k-N,k-1] may still be propagating back.Because the sender has sent packets [k-N, k-1], it must be the case that the sender has already received an ACK for k-N-1. Once the receiver has sent an ACK for k-N-1 it will never send an ACK that is less that k-N-1. Thus the range of in-flight ACK values can range from k-N-1 to k-1.

Problem 23

In order to avoid the scenario of Figure 3.27, we want to avoid having the leading edge of the receiver's window (i.e., the one with the "highest" sequence number) wrap around in the sequence number space and overlap with the trailing edge (the one with the "lowest" sequence number in the sender's window). That is, the sequence number space must be large enough to fit the entire receiver window and the entire sender window without this overlap condition. So - we need to determine how large a range of sequence numbers can be covered at any given time by the receiver and sender windows.

Suppose that the lowest-sequence number that the receiver is waiting for is packet m. In this case, it's window is [m,m+w-1] and it has received (and ACKed) packet m-1 and the w-1 packets before that, where w is the size of the window. If none of those w ACKs have been yet received by the sender, then ACK messages with values of [m-w,m-1] may

still be propagating back. If no ACKs with these ACK numbers have been received by the sender, then the sender's window would be [m-w,m-1].

Thus, the lower edge of the sender's window is m-w, and the leading edge of the receivers window is m+w-1. In order for the leading edge of the receiver's window to not overlap with the trailing edge of the sender's window, the sequence number space must thus be big enough to accommodate 2w sequence numbers. That is, the sequence number space must be at least twice as large as the window size, $k \ge 2w$.

Problem 24

- a) True. Suppose the sender has a window size of 3 and sends packets 1, 2, 3 at t0. At t1 (t1 > t0) the receiver ACKS 1, 2, 3. At t2 (t2 > t1) the sender times out and resends 1, 2, 3. At t3 the receiver receives the duplicates and re-acknowledges 1, 2, 3. At t4 the sender receives the ACKs that the receiver sent at t1 and advances its window to 4, 5, 6. At t5 the sender receives the ACKs 1, 2, 3 the receiver sent at t2. These ACKs are outside its window.
- b) True. By essentially the same scenario as in (a).
- c) True.
- a) True. Note that with a window size of 1, SR, GBN, and the alternating bit protocol are functionally equivalent. The window size of 1 precludes the possibility of out-of-order packets (within the window). A cumulative ACK is just an ordinary ACK in this situation, since it can only refer to the single packet within the window.

Problem 25

- a) Consider sending an application message over a transport protocol. With TCP, the application writes data to the connection send buffer and TCP will grab bytes without necessarily putting a single message in the TCP segment; TCP may put more or less than a single message in a segment. UDP, on the other hand, encapsulates in a segment whatever the application gives it; so that, if the application gives UDP an application message, this message will be the payload of the UDP segment. Thus, with UDP, an application has more control of what data is sent in a segment.
- b) With TCP, due to flow control and congestion control, there may be significant delay from the time when an application writes data to its send buffer until when the data is given to the network layer. UDP does not have delays due to flow control and congestion control.

Problem 26

There are $2^{32} = 4,294,967,296$ possible sequence numbers.

a) The sequence number does not increment by one with each segment. Rather, it increments by the number of bytes of data sent. So the size of the MSS is irrelevant -- the maximum size file $2^{32} \approx 4.19$ Gbytes

that can be sent from A to B is simply the number of bytes representable by $2^{32} \approx 4.19$ Gbytes

$$\left[\frac{2^{32}}{536}\right] = 8,012,999$$

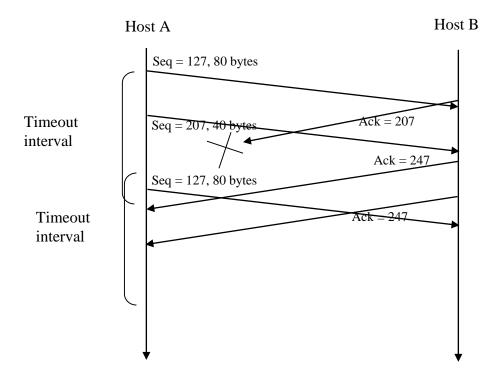
b) The number of segments is $\begin{vmatrix} 530 \end{vmatrix}$. 66 bytes of header get added to each segment giving a total of 528,857,934 bytes of header. The total number of bytes transmitted is $2^{32} + 528,857,934 = 4.824 \times 10^9$ bytes.

Thus it would take 249 seconds to transmit the file over a 155~Mbps link.

Problem 27

- a) In the second segment from Host A to B, the sequence number is 207, source port number is 302 and destination port number is 80.
- b) If the first segment arrives before the second, in the acknowledgement of the first arriving segment, the acknowledgement number is 207, the source port number is 80 and the destination port number is 302.
- c) If the second segment arrives before the first segment, in the acknowledgement of the first arriving segment, the acknowledgement number is 127, indicating that it is still waiting for bytes 127 and onwards.

d)



Problem 28

Since the link capacity is only 100 Mbps, so host A's sending rate can be at most 100Mbps. Still, host A sends data into the receive buffer faster than Host B can remove data from the buffer. The receive buffer fills up at a rate of roughly 40Mbps. When the buffer is full, Host B signals to Host A to stop sending data by setting RcvWindow = 0. Host A then stops sending until it receives a TCP segment with RcvWindow > 0. Host A will thus repeatedly stop and start sending as a function of the RcvWindow values it receives from Host B. On average, the long-term rate at which Host A sends data to Host B as part of this connection is no more than 60Mbps.

Problem 29

a) The server uses special initial sequence number (that is obtained from the hash of source and destination IPs and ports) in order to defend itself against SYN FLOOD attack.

- b) No, the attacker cannot create half-open or fully open connections by simply sending and ACK packet to the target. Half-open connections are not possible since a server using SYN cookies does not maintain connection variables and buffers for any connection before full connections are established. For establishing fully open connections, an attacker should know the special initial sequence number corresponding to the (spoofed) source IP address from the attacker. This sequence number requires the "secret" number that each server uses. Since the attacker does not know this secret number, she cannot guess the initial sequence number.
- c) No, the sever can simply add in a time stamp in computing those initial sequence numbers and choose a time to live value for those sequence numbers, and discard expired initial sequence numbers even if the attacker replay them.

Problem 30

- a) If timeout values are fixed, then the senders may timeout prematurely. Thus, some packets are re-transmitted even they are not lost.
- b) If timeout values are estimated (like what TCP does), then increasing the buffer size certainly helps to increase the throughput of that router. But there might be one potential problem. Queuing delay might be very large, similar to what is shown in Scenario 1.

Problem 32

a)

Denote *EstimatedRTT*⁽ⁿ⁾ for the estimate after the *n*th sample.

$$EstimatedRTT^{(4)} = xSampleRTT_{1} + (1-x)[xSampleRTT_{2} + (1-x)[xSampleRTT_{3} + (1-x)SampleRTT_{4}]]$$
$$= xSampleRTT_{1} + (1-x)xSampleRTT_{2} + (1-x)^{2}xSampleRTT_{3} + (1-x)^{3}SampleRTT_{4}$$

b)

EstimatedRTT⁽ⁿ⁾ =
$$x \sum_{j=1}^{n-1} (1-x)^{j-1} SampleRTT_j$$

 $+(1-x)^{n-1}SampleRTT_n$

c)

$$\begin{aligned} EstimatedRTT^{(\infty)} &= \frac{x}{1-x} \sum_{j=1}^{\infty} (1-x)^{j} SampleRTT_{j} \\ &= \frac{1}{9} \sum_{j=1}^{\infty} .9^{j} SampleRTT_{j} \end{aligned}$$

The weight given to past samples decays exponentially.

Problem 44

- a) It takes 1 RTT to increase CongWin to 6 MSS; 2 RTTs to increase to 7 MSS; 3 RTTs to increase to 8 MSS; 4 RTTs to increase to 9 MSS; 5 RTTs to increase to 10 MSS; 6 RTTs to increase to 11 MSS; and 7 RTTs to increase to 12MSS.
- b) In the first RTT 5 MSS was sent; in the second RTT 6 MSS was sent; in the third RTT 7 MSS was sent; in the fourth RTT 8 MSS was sent; in the fifth RTT, 9 MSS was sent; and in the sixth RTT, 10 MSS was sent. Thus, up to time 6 RTT, 5+6+7+8+9+10 = 45 MSS were sent (and acknowledged). Thus, we can say that the average throughput up to time 6 RTT was (45 MSS)/(6 RTT) = 7.5 MSS/RTT.

Problem 45

a) The loss rate, L, is the ratio of the number of packets lost over the number of packets sent. In a cycle, 1 packet is lost. The number of packets sent in a cycle is

$$\frac{W}{2} + \left(\frac{W}{2} + 1\right) + \dots + W = \sum_{n=0}^{W/2} \left(\frac{W}{2} + n\right)$$
$$= \left(\frac{W}{2} + 1\right) \frac{W}{2} + \sum_{n=0}^{W/2} n$$
$$= \left(\frac{W}{2} + 1\right) \frac{W}{2} + \frac{W/2(W/2 + 1)}{2}$$
$$= \frac{W^2}{4} + \frac{W}{2} + \frac{W^2}{8} + \frac{W}{4}$$
$$= \frac{3}{8}W^2 + \frac{3}{4}W$$
ate is

Thus the loss rate is

$$L = \frac{1}{\frac{3}{8}W^2 + \frac{3}{4}W}$$

b) For W large,
$$\frac{3}{8}W^2 >> \frac{3}{4}W$$
. Thus $L \approx 8/3W^2$ or $W \approx \sqrt{\frac{8}{3L}}$. From the text, we therefore have
average throughput $= \frac{3}{4}\sqrt{\frac{8}{3L}} \cdot \frac{MSS}{RTT}$
 $= \frac{1.22 \cdot MSS}{RTT \cdot \sqrt{L}}$