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 that can be sent from A to B is simply the number of bytes representable by $2^{32} \approx 4.29$ Gbytes (4Gbytes is also correct)

b) The number of segments is $\left\lceil \frac{2^{32}}{536} \right\rceil = 8{,}012{,}999$. 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.

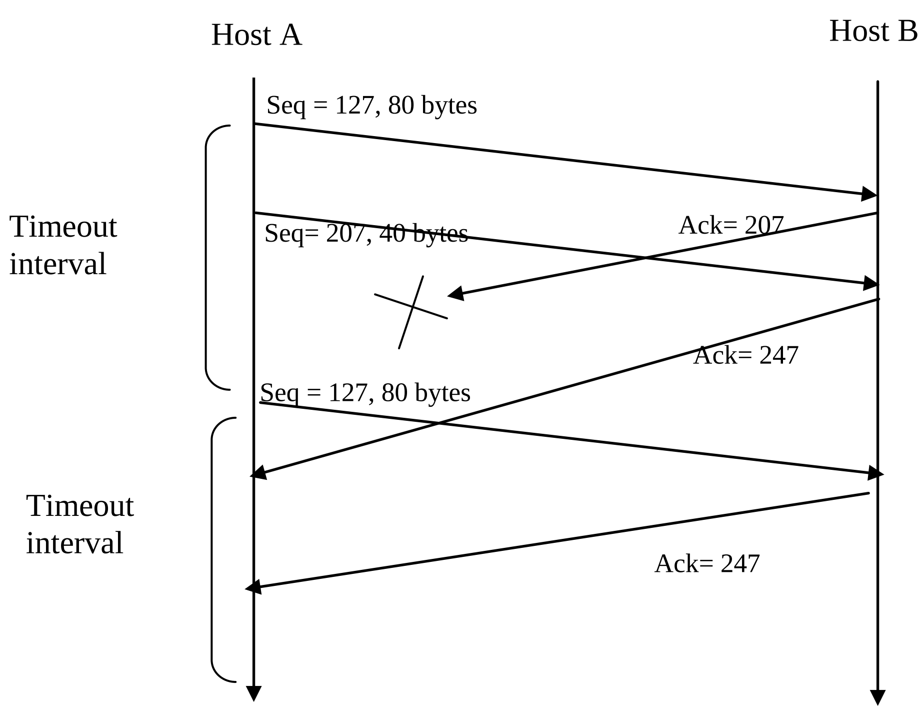
Problem 27

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 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)

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

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

Problem 33

Let's look at what could wrong if TCP measures `SampleRTT` for a retransmitted segment. Suppose the source sends packet P1, the timer for P1 expires, and the source then sends P2, a new copy of the same packet. Further suppose the source measures `SampleRTT` for P2 (the retransmitted packet). Finally suppose that shortly after transmitting P2 an acknowledgment for P1 arrives. The source will mistakenly take this acknowledgment as an acknowledgment for P2 and calculate an incorrect value of `SampleRTT`.

Problem 39

If the arrival rate increases beyond R/2 in Figure 3.46(b), then the total arrival rate to the queue exceeds the queue's capacity, resulting in increasing loss as the arrival rate increases. When the arrival rate equals R/2, 1 out of every three packets that leaves the queue is a retransmission. With increased loss, even a larger fraction of the packets leaving the queue will be retransmissions. Given that the maximum departure rate from the queue for one of the sessions is R/2, and given that a third or more will be transmissions as the arrival rate increases, the throughput of successfully deliver data can not increase beyond λ_{out}. Following similar reasoning, if half of the packets leaving the queue are retransmissions, and the maximum rate of output packets per session is R/2, then the maximum value of λ_{out} is (R/2)/2 or R/4.

Problem 40

a) TCP slowstart is operating in the intervals [1,6] and [23,26]
b) TCP congestion avoidance is operating in the intervals [6,16] and [17,22]
c) After the 16th transmission round, packet loss is recognized by a triple duplicate ACK. If there was a timeout, the congestion window size would have dropped to 1.
d) After the 22nd transmission round, segment loss is detected due to timeout, and hence the congestion window size is set to 1.
e) The threshold is initially 32, since it is at this window size that slow start stops and congestion avoidance begins.
f) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 16, the congestion windows size is 42. Hence the threshold is 21 during the 18th transmission round.
g) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 22, the congestion windows size is 29. Hence the threshold is 14 (taking lower floor of 14.5) during the 24th transmission round.
h) During the 1st transmission round, packet 1 is sent; packet 2-3 are sent in the 2nd transmission round; packets 4-7 are sent in the 3rd transmission round; packets 8-15 are sent in the 4th transmission round; packets 16-31 are sent in the 5th transmission round; packets 32-63 are sent in the 6th transmission round; packets 64 – 96 are sent in the 7th transmission round. Thus packet 70 is sent in the 7th transmission round.

i) The threshold will be set to half the current value of the congestion window (8) when the loss occurred and congestion window will be set to the new threshold value + 3 MSS . Thus the new values of the threshold and window will be 4 and 7 respectively.
j) threshold is 21, and congestion window size is 4.
k) round 17, 1 packet; round 18, 2 packets; round 19, 4 packets; round 20, 8 packets; round 21, 16 packets; round 22, 21 packets. So, the total number is 52.

Problem 41

Refer to Figure 5. In Figure 5(a), the ratio of the linear decrease on loss between connection 1 and connection 2 is the same - as ratio of the linear increases: unity. In this case, the throughputs never move off of the AB line segment. In Figure 5(b), the ratio of the linear decrease on loss between connection 1 and connection 2 is 2:1. That is, whenever there is a loss, connection 1 decreases its window by twice the amount of connection 2. We see that eventually, after enough losses, and subsequent increases, that connection 1's throughput will go to 0, and the full link bandwidth will be allocated to connection 2.

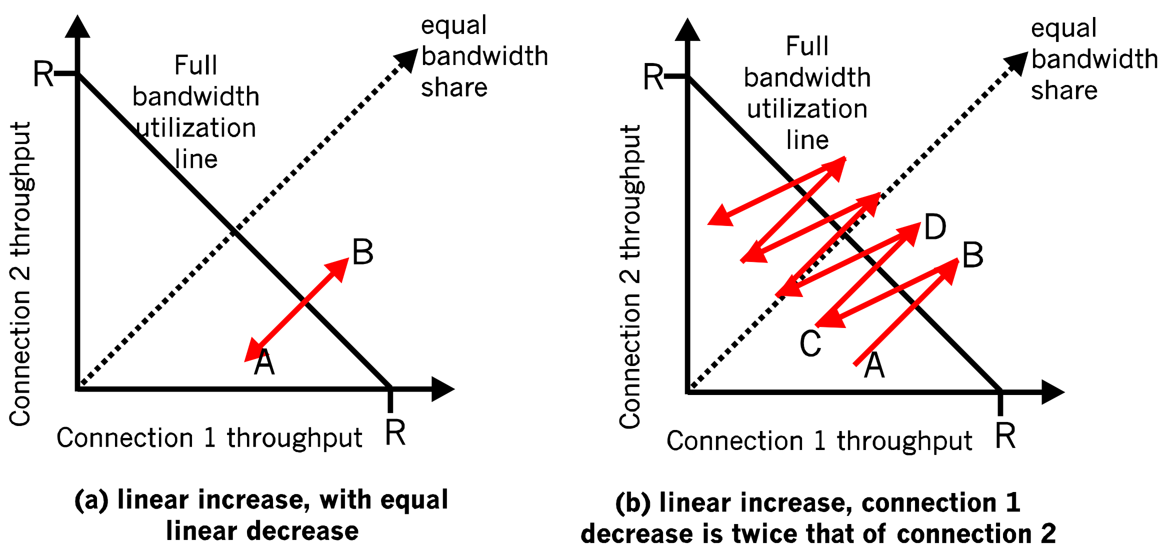


Figure 5: Lack of TCP convergence with linear increase, linear decrease

Problem 42

If TCP were a stop-and-wait protocol, then the doubling of the time out interval would suffice as a congestion control mechanism. However, TCP uses pipelining (and is therefore not a stop-and-wait protocol), which allows the sender to have multiple outstanding unacknowledged segments. The doubling of the timeout interval does not prevent a TCP sender from sending a large number of first-time-transmitted packets into the network, even when the end-to-end path is highly congested. Therefore a congestion-control mechanism is needed to stem the flow of “data received from the application above” when there are signs of network congestion.

Problem 43

In this problem, there is no danger in overflowing the receiver since the receiver's receive buffer can hold the entire file. Also, because there is no loss and acknowledgements are returned before timers expire, TCP congestion control does not throttle the sender. However, the process in host A will not continuously pass data to the socket because the send buffer will quickly fill up. Once the send buffer becomes full, the process will pass data at an average rate or R << S.