

## Problem 1

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.

The receive buffer will begin to fill up, due to the fact that Host A will be sending data into it faster than Host B can remove the data from it. Once the buffer has completely filled, Host B will send a message to Host A to stop sending data until Host B can remove data from the buffer. Host B will then send a TCP segment to Host A, informing it to continue sending data. The buffer will fill up again and this process will repeat until all of the data has been sent from Host A to Host B.

## Problem 2

Suppose that instead of a multiplicative decrease, TCP decreased the window size by a constant amount. Would the resulting AIAD algorithm converge to an equal share algorithm? Justify your answer using a graphical diagram similar to Slide 100 of the lecture.

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.

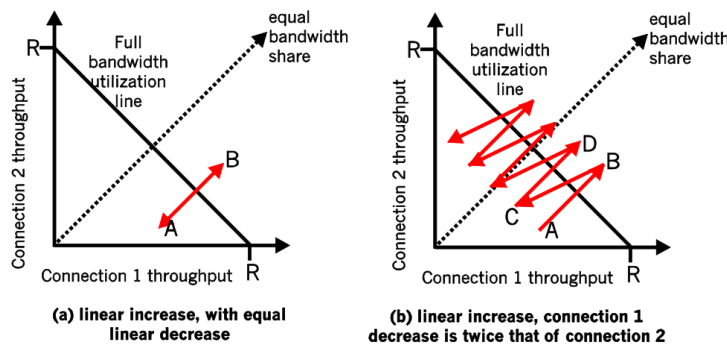


Figure 5: Lack of TCP convergence with linear increase, linear decrease

### Problem 3

Recall the macroscopic description of TCP throughput, in the period of time from when the connection's rate varies from  $W/(2 \text{ RTT})$  to  $W/\text{RTT}$ , only one packet is lost (at the very end of the period).

- (a) Show that the loss rate (fraction of packets lost) is equal to  $L = \text{lossrate} = 1/(3/8 W^2 + 3/4 W)$
- (b) Use the result above to show that if a connection has loss rate  $L$ , then its average rate is approximately given by  $\simeq 1.22 \times MSS/(\text{RTT} \times \sqrt{L})$

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 cycle is

$$\begin{aligned}
 \frac{W}{2} + \left(\frac{W}{2} + 1\right) + \dots + W &= \sum_{n=0}^{W/2} \left(\frac{W}{2} + n\right) \\
 &= \left(\frac{W}{2} + 1\right) \frac{W}{2} + \sum_{n=0}^{W/2} n \\
 &= \left(\frac{W}{2} + 1\right) \frac{W}{2} + \frac{W/2 \cdot (W/2 + 1)}{2} \\
 &= \frac{W^2}{4} + \frac{W}{2} + \frac{W^2}{8} + \frac{W}{4} \\
 &= \frac{3}{8} W^2 + \frac{3}{4} W
 \end{aligned} \tag{1}$$

Thus the loss rate is

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

b) For  $W$  large,  $\frac{3}{8} W^2 \gg \frac{3}{4} W$ . Thus  $L \approx \frac{8}{3W^2}$  or  $W \approx \sqrt{\frac{8}{3L}}$ . From the text, we therefore have

$$\begin{aligned}
 \text{averagethroughput} &= \frac{3}{4} \sqrt{\frac{8}{3L}} \frac{MSS}{\text{RTT}} \\
 &= \frac{1.22 MSS}{\text{RTT} \sqrt{L}}
 \end{aligned} \tag{3}$$

## Problem 4

You are designing a reliable, sliding window, byte-stream protocol similar to TCP. It will be used for communication with a geosynchronous satellite network, for which the bandwidth is 1 Gbps and the RTT is 300 ms. Assume the maximum segment lifetime is 30 seconds.

- (a) How many bits wide should you make the `ReceiveWindow` and `SequenceNum` fields? (`ReceiveWindow` is also called “Advertised Window” in some other textbooks.)
- (b) If `ReceiveWindow` is 16 bits, what upper bound would that impose on the effective bandwidth?

- (a) To fully utilize the network, `ReceiveWindow` needs to be larger than  $(\text{Delay} \times \text{Bandwidth})$ .  
 $(\text{ReceiveWindow}) \geq (\text{Delay}) \times (\text{Bandwidth}) = 300 \text{ ms} \times 1 \text{ Gbps} = 300 \text{ Mbit} = 37.5 \text{ MB}$

$$2^{25} = 33,554,432$$

$$2^{26} = 67,108,864$$

Therefore, 26 bits are needed for `ReceiveWindow`.

`SequenceNum` needs be large enough that the sequence number does not wrap around before any delayed segments have left the network, which is presumed to occur within the maximum segment lifetime.

$$\text{SequenceNum} \geq (\text{Maximum Segment Lifetime}) \times (\text{Bandwidth}) = 30 \text{ s} \times 1 \text{ Gbps} = 30 \text{ Gbit} = 3.75 \text{ GB}$$

$$2^{31} = 2,147,483,648$$

$$2^{32} = 4,294,967,296$$

Therefore, 32 bits are needed for `SequenceNum`.

- (b) If `ReceiveWindow` is 16 bits, it is smaller than  $(\text{Delay} \times \text{Bandwidth})$  product. Therefore, `ReceiveWindow` is the limiting factor of the effective bandwidth. During a single RTT, only `ReceiveWindow` amount of data can be transferred.

$$\text{Effective Bandwidth} = \text{ReceiveWindow} / \text{RTT} = 2^{16} \text{ B} / 300 \text{ ms} = 218.3 \text{ KB/s}$$

## Problem 5

Consider the evolution of a TCP connection with the following characteristics. Assume that all the following algorithms are implemented in TCP congestion control: slow start, congestion avoidance, fast retransmit and fast recovery, and retransmission upon timeout. If `ssthresh` equals to `cwnd`, use the slow start algorithm in your calculation.

- The TCP receiver acknowledges every segment, and the sender always has data segments available for transmission.
- The RTT is 100 ms for all transmissions, consists of the network latency of 60 ms in sending a segment (header and payload) from the sender to the receiver and 40 ms in sending an acknowledgment (header only) from the receiver to the sender. Ignore packet-processing delays at the sender and the receiver.
- Initially `ssthresh` at the sender is set to 5. Assume `cwnd` and `ssthresh` are measured in segments, and the transmission time for each segment is negligible.
- Retransmission timeout (RTO) is initially set to 500ms at the sender and is unchanged during the connection lifetime.
- The connection starts to transmit data at time  $t = 0$ , and the initial sequence number starts from 1. TCP segment with sequence number 6 is lost once (i.e., it sees segment loss during its first transmission). No other segments are lost during transmissions.

What are the values for `cwnd` and `ssthresh` when the sender receives the TCP ACK with number 15? Show your intermediate steps or your diagram in your solution.

