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.

Write your solution to Problem 1 in this box

The receive buffer will begin to fill up because host A will be sending data faster than B can read the data from it. Once the buffer is full, B will send a message to A that the maximum buffer reached and will tell A to stop sending data. Once space in buffer, B will send message to A to send TCP connection back to A. Process will continue till all data is transferred.

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.

Write your solution to Problem 2 in this box

The graph in the textbook is the best description to this problem.
It would be an equal share algorithm.

The ratio is 2:1. Whenever there is a loss, connection 1 decreased its window size by twice the amount of connection 2.

Eventually, connection 1will have a throughput of 0, and the full bandwidth will be allocated to connection 2.

Problem 3

Recall the macroscopic description of TCP throughput, in the period of time from when the connection's rate varies from W/(2 RTT) to W/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=lossrate= $1/(3/8W^2 + 3/4W)$

(b) Use the result above to show that if a connection has loss rate L, then its average rate is approximately given by $\simeq 1.22 \times MSS/(RTT \times \sqrt{L})$

Write your solution to Problem 3 in this box

a. Only one packet will be lost when rate changes from 2RTT to 1RTT. lost/sent

(W/2) + (W/2 + 1) + (W/2+2)….. + (W)

Need to find Sn:

(W/2 + 1)W/2 + (W/2)(W/2 + 1)/2
W^2/4 + W/2 + W^2/8 + W/4

3/8W^2 + 3/4W

Thus becomes 1/ (ANS)

b. L = 8/2 W^2

W^2 = 8/(3L)
W = sqrt (8/(3l))

then if replace W with sqrt(8/(3L)), and to algebra, will get 1.2247

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?

Write your solution to Problem 4 in this box

a.
AdvertisedWindow needs to be bigger than delay*rate:
300ms * 1GBPS = 300 Mbit

SequenceNum needs to be bigger than TTL*rate

30s* 1Gbps = 30Gbit

b. 16 bit is smaller than an effective bandwidth. advertisedwindow is the limiting factor.

which means only 2^16 / 300ms could be transferred in one RTT.

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, congestions 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 ssthreshold when the sender receives the TCP ACK with number 15? Show your intermediate steps or your diagram in your solution.

Problem 5 continued on next page. . .

Write your solution to Problem 5 in this box