Problem 1

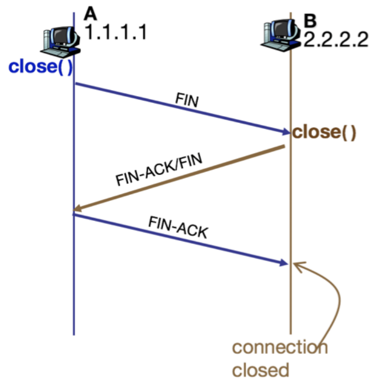
Host A and B are directly connected with a 100 Mbps link. There is one TCP connection between the two hosts, and Host A is sending to Host B an enormous file over this connection. Host A can send its application data into its TCP socket at a rate as high as 120 Mbps but Host B can read out of its TCP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

TCP flow control becomes important because Host B can't remove data from the buffer as quickly as host A transmits data into the receiver's buffer. Host A's sending rate is limited to 100Mb per second of link capacity, but B's buffer will still overflow without flow control.
In TCP flow control, the receiver controls the sender. So, when host B's buffer is full, it sends rwnd = 0 in the TCP header of receiver-to-sender segments. This signals host A to stop sending data. Host A temporarily stops sending data until it receives a segment from Host B with rwnd > 0. The process may repeat if Host B's buffer fills up again. Therefore, Host B's buffer will never fill up.

Problem 2

A sends a TCP FIN message to B to close the TCP connection with B, the TCP header of A's FIN message is shown below. When B receives A's TCP FIN, it also decides to close the connection, so B sends a combined FIN and FIN-ACK message, whose TCP header is also shown below. Please fill in all the fields with a question mark in this TCP header.



<table>
<tr><td colspan="8">... src 1.1.1.1, dst: 2.2.2.2</td></tr>
<tr><td colspan="4">s_port: 2008</td><td colspan="4">d_port: 5670</td></tr>
<tr><td colspan="8">seq_no: 980</td></tr>
<tr><td colspan="8">ack_no: 3120</td></tr>
<tr><td>header length</td><td>not used</td><td>0</td><td>1</td><td>0</td><td>0</td><td>0</td><td>1</td><td>rcv_window: 200</td></tr>
<tr><td colspan="8">checksum: ...</td><td>0 (ignore this field)</td></tr>
</table>

A to B

<table>
<tr><td colspan="8">... src 2.2.2.2, dst: 1.1.1.1</td></tr>
<tr><td colspan="4">s_port: ?</td><td colspan="4">d_port: ?</td></tr>
<tr><td colspan="8">seq_no: ?</td></tr>
<tr><td colspan="8">ack_no: ?</td></tr>
<tr><td>header length</td><td>not used</td><td>0</td><td>?</td><td>0</td><td>0</td><td>?</td><td>?</td><td>rcv_window: 400</td></tr>
<tr><td colspan="8">checksum: ...</td><td>0 (ignore this field)</td></tr>
</table>

B to A

Write your solution to Problem 2 in this box

From B to A

<table>
<tr><td colspan="8">... src 2.2.2.2, dst: 1.1.1.1</td></tr>
<tr><td colspan="8">s port: 5670</td><td>d port: 2008</td></tr>
<tr><td colspan="9">seq no: 3120</td></tr>
<tr><td colspan="9">ack no: 981</td></tr>
<tr><td>header length</td><td>not used</td><td>0</td><td>1</td><td>0</td><td>0</td><td>0</td><td>1</td><td>rcv window: 400</td></tr>
<tr><td colspan="8">checksum: ...</td><td>0 (ignore this field)</td></tr>
</table>

Problem 3

Recall the macroscopic description of TCP throughput (Slide 134), 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 $\simeq 1.22 \times MSS/(RTT \times \sqrt{L})$. (*Hint*: average rate = $\frac{3}{4} \cdot \frac{W}{RTT}$)

Write your solution to Problem 3 in this box

(a) Packets sent in cycle: $\left(\frac{W}{2}\right) + \left(\frac{W}{2}+1\right) + \cdots + W$

$$= \sum_{n=0}^{W/2} \frac{W}{2} + n = \sum_{n=0}^{W/2} \frac{W}{2} + \sum_{n=0}^{W/2} n = \left(\frac{W}{2}+1\right)\frac{W}{2} + \frac{\frac{W}{2}\left(\frac{W}{2}+1\right)}{2}$$

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

$$L = \frac{\text{No. of lost packets}}{\text{No. of packets sent}} = \boxed{\frac{1}{\left(\frac{3W^2}{8} + \frac{3W}{4}\right)}}$$

(b) If W is large, $\frac{3W^2}{8} \cancel{\gg} \frac{3}{4}$

$$\therefore L = \frac{8}{3W^2} \Longrightarrow W = \sqrt{\frac{8}{3L}}$$

$$\text{Avg. rate} = \frac{3}{4}\sqrt{\frac{8}{3L}} \cdot \frac{MSS}{RTT} \approx \boxed{\frac{1.22\, MSS}{RTT\sqrt{L}}}$$

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 800 Mbps and the RTT is 400 ms. Assume the maximum segment lifetime is 25 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) Receive Window must be larger for the connection to be full.

$\therefore$ Receive Window $\geq$ RTT $*$ Bandwidth

$\geq 4 \times 10^7$ bytes

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

$2^{26} = 67,108,864$ $\therefore$ Receive Window = $\boxed{26 \text{ bits}}$

SequenceNum $\geq$ MaxSegLifetime $*$ Bandwidth

$\geq 2.5 \times 10^9$ bytes

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

$2^{32} = 4,294,967,296$ $\therefore$ SequenceNum = $\boxed{32 \text{ bits}}$

(b) If receive window is full, we can limit the amount of data transfer to

$2^{16}/400ms \cong \boxed{163.84 \text{ Kb/s}}$

Problem 5

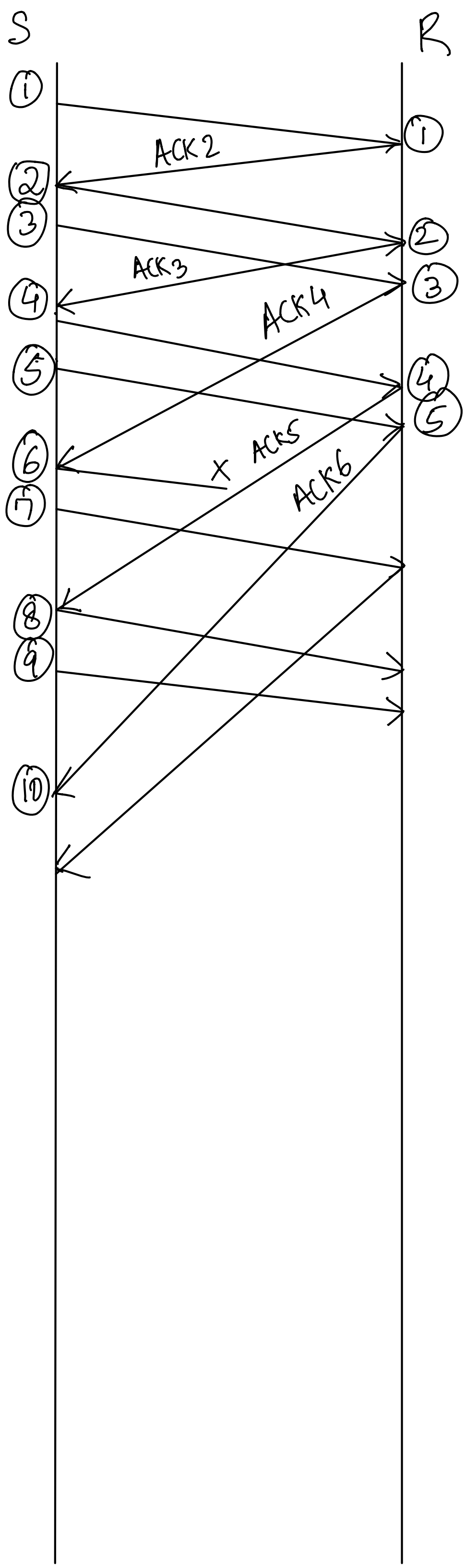
Consider the evolution of a TCP connection with the following characteristics. Assume that all the following algorithms are implemented in TCP congestion control: slow start, congestions avoidance, fast retransmit and fast recovery, and retransmission upon timeout. If ssthresh equals to cwnd, use the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

- The TCP receiver acknowledges every segment in cumulative way, 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



Cwnd	ssthresh	State
1	5	slow start
2		
3		
4		