

Q1

Assumption: This is a TCP zero
based graph.

- (a) At event B, the decrease in window is due to ~~to³ duplicate acks that the sender received~~.
- (b) No, because it might be possible that the packet is only delayed. It should be noted that even the timeout set for the packet hasn't been completed. However, the packet under discussion will necessarily be retransmitted.
- (c) Since the window changed to zero, this is a ~~packet~~ timeout event.
- (d) Since timeout has occurred, there is a higher probability that the packet has been lost and will be ~~retransmitted~~. However, it may be possible that the packet was only delayed. Again, it will necessarily be retransmitted.

(e) ~~B~~ is more likely. Since, in a highly light network, time out will be same but with multiple packets in flight, one of them might be only slightly delayed will lead to event B. However, D is not likely for the aforesaid reason.

(f) If A & B were linear, the growth will be very slow even for light networks. For example, if network bandwidth allowed speed of 256 MSS without any congestion, Linear will be at 256 after 256 transmission while A with exponential growth it will get there after ~~8~~ transmissions.

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Q2

1- data received::

If data is received, increment length data with $\text{seq} = \text{sendbase}$ and increment the $\text{sendbaseNextSeqNum}$ by the length of data.

timer

2- timeout::

If a timeout occurs, send oldest packet that hasn't been acknowledged for yet.

3- ACK received ::

If $y > \text{send base}$, then the sender is acknowledging too unacked bytes so we set $\text{send base} = y$.

⑥

In addition to it, if timer was not running in ①, timer is started in ③ if there are data packets for which ACK hasn't arrived, we restart timer after setting $\text{send base} = y$.

Q3

$$(a) 72 - 61 = 11$$

$$(b) 61$$

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Q4 $RTT = 30$ Estimated $RTT = 30$ $devRTT = 3 \text{ ms}$

$$\begin{aligned} \text{Estimated } RTT &= 0.8125 \times 30 \\ &+ 0.125 \times 28 \end{aligned}$$

$$\text{Estimated } RTT = 29.75$$

$$\begin{aligned} devRTT &= 0.75(3) + 0.25(2) \\ Time_{devRTT} &= 2.75 \end{aligned}$$

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$$\begin{aligned} \text{Timeout} &= \text{Estimated } RTT + 4(\text{devRTT}) \\ &= 29.75 + 4(2.75) \end{aligned}$$

$$\boxed{\text{Timeout} = 40.75}$$

assuming Sample RTT = 28,

$$\alpha = \frac{1}{8}, \beta = 0.25$$

Q5

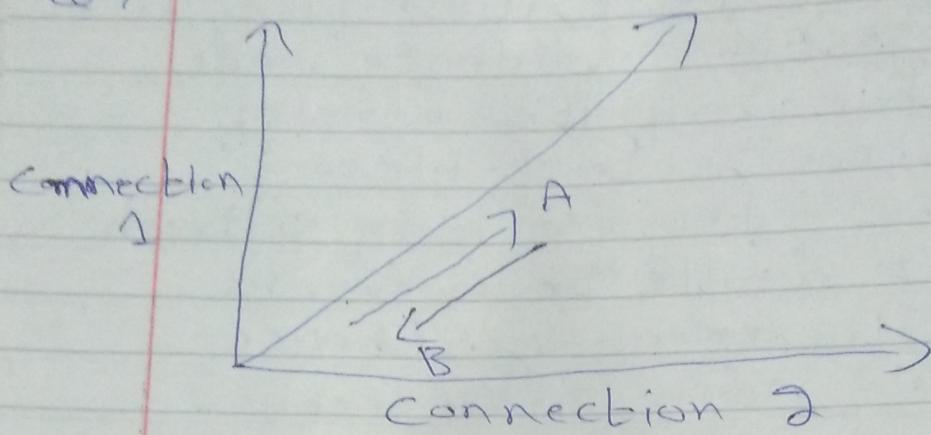
When TCP congestion control algorithm reached a threshold value (ssthresh) while in slow start mode, it shifts to congestion avoidance mode, ssthresh is half the value of cwnd when the last timeout occurs thus after reaching ssthresh we can't double the window, we instead increment by 1 (hence called congestion avoidance)

Yes, it can. Every time a timeout occurs, the algo shifts to slow-start irrespective of the current state.

Q6

Ars No. Why would it? UDP doesn't care about timeout or packet loss or congestion control. So, it ~~requires~~ ^{has} no use of RTT. It simply sends whatever is being sent to it.

Q7



This can be best explained if there are only two connections in the network. In this case if both start at the same time, both will incrementally increase their window size, as shown in arrow A. Uptill, a point reaches where the packet losses start and TCP would keep decreasing ssthresh to de-congest the network. Then, both will go down in the graph till the network is de-congested and then the same process may repeat.

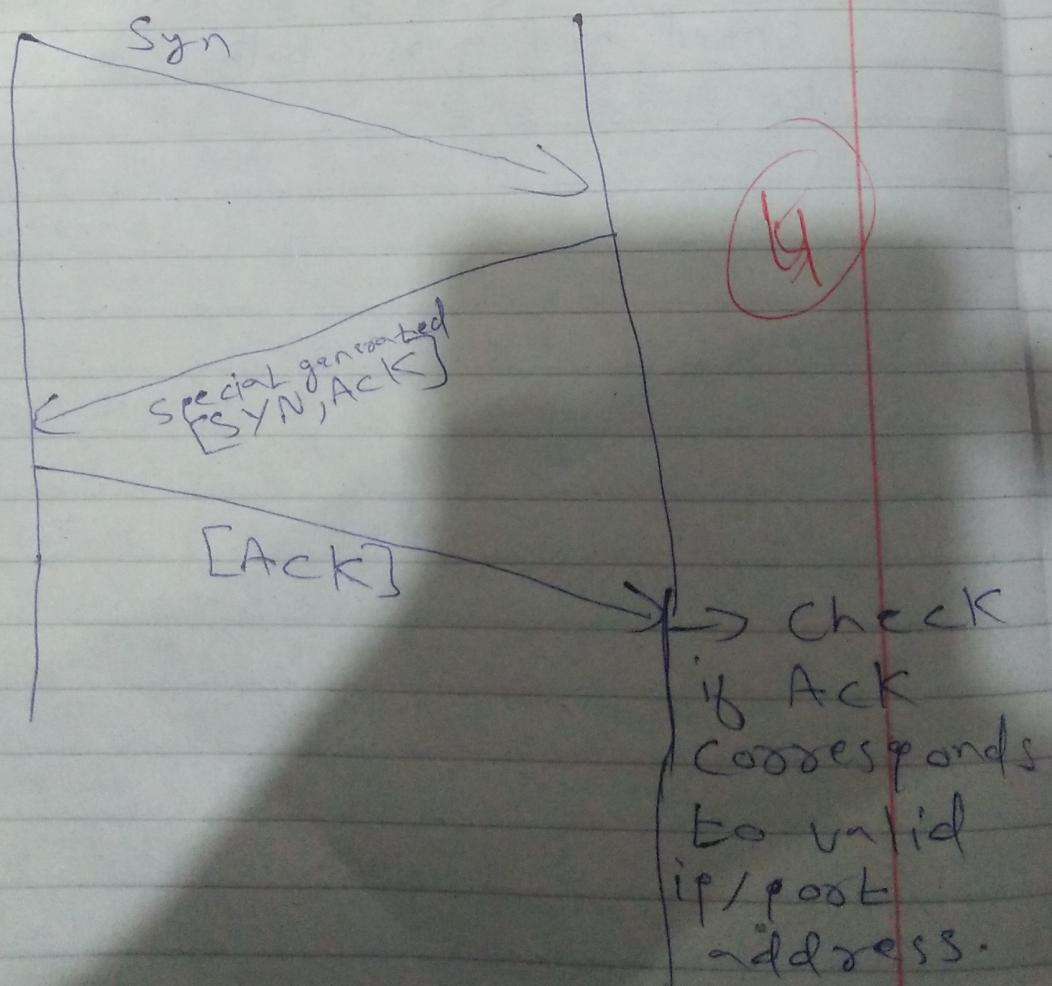
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~~FALL~~ This is because the slow start algorithm allows a client to keep increasing speed until packet loss occurs and then it halves the value to ensure stabilization. So, both clients will eventually converge to similar speeds.

Even if both do not start at same time, say 1 starts before 2. If 1 keeps increasing its size, at one point it loss will occur and its speed will be halved, during that time 2 can get up to speed and again both converge to same rate.

Ans

SYN cookies are special syn sequence numbers that are generated from the server by using the source ip address and source ports. A number is generated using which is a function of the source port or ip and then this number is stored using a hash function. Once the ack is received, the track number is passed to the hash function to see if it is from a valid client ip/post.



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This ensures that if a connection is accepted, it is accepted by from a valid client. For a attack, the ack will not correspond to valid ip.