HW4

1. Determine the following statements are true or false. (Use enough reasons and explanation to support your answer)

A) Single packet error can, cause GBN to retransmit a large number of packets.

B)When ,SR receiver, receives packet "n", delivers it to application layer even if it packet "n-1" is not received.

C) In the receiver side of the SR if the received packet is *in* [rcv_base-N, rcv_base-1], the receiver simply ignore it because it has previously acknowledged that packet.

D) TCP is said to be **connection-oriented** which means it acts exactly in the same way as protocols in circuit switched network.

E) "multicasting" —the transfer of data from one sender to many receivers in a single send operation—is possible with TCP.

F) In the TCP 3-way handshaking, The first two segments carry no payload, the third of these segments may carry a payload.

G) MSS which is the maximum size of TCP segment, is determined by TCP receiver in 3-way-hand shaking.
H) The maximum amount of TCP length field equals to 15 bytes.

2. Answer the following questions:

a)In the GBN approach which is discussed in the book it is assumed that the receiver discards out-of-order packets, what are the advantageous and disadvantageous of such an approach?

b) It is assumed that a packet cannot "live" in the network for longer than some fixed maximum amount of time, what is the reason for such an assumption?

c) In practice, both sides of a TCP connection randomly choose an initial sequence number, what is the reason for such a choice?

D) What is the main difference between congestion and flow control mechanisms? (in terms of cause of existence)

3.consider we have proposed a new UDP checksum method in which the UDP at the sender side performs the 1s complement of the XOR of all the 16-bit words in the segment and At the receiver, XOR is performed on all 16-bit words in segment , including the checksum. If no errors are introduced into the packet, then clearly the sum at the receiver will be 1111111111111111. Now assume we have n bytes of data in the message section of the UDP segment and the probability of bit corruption to be equal to "p"; if we know that one of the bits is corrupted, for sure, what is the probability for the receiver not to be able to detect the corruption? (hint: it is known that one of the bits is corrupted but we don't know if other bits are corrupted or not)

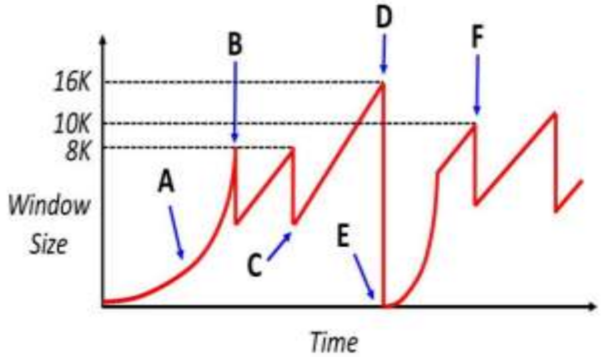
4.In stop-and-wait transmission, suppose that both sender and receiver retransmit their last frame immediately on receipt of a duplicate ACK or data frame; such a strategy is superficially reasonable because receipt of such a duplicate is most likely to mean the other side has experienced a timeout.

(a) Draw a timeline showing what will happen if the first data frame is somehow duplicated, but no frame is lost. How long will the duplications continue? This situation is known as the Sorcerer's Apprentice bug.

b) Suppose that, like data, ACKs are retransmitted if there is no response within the timeout period. Suppose also that both sides use the same timeout interval. Identify a reasonably likely scenario for triggering the Sorcerer's Apprentice bug.

5-Give an FSM description for TCP, for both sides (sender and receiver), considering congestion control (TCP Reno) and flow control mechanisms.(assume that RTT is fixed)

6. consider the below scheme, in which Reno TCP is used, and answer the questions.(assume MSS=1k ,RTT=100ms and in time 0 window size equals to 1k)



a)Name the event at specified points in the above scheme.

b)Consider the curved slope labeled by point A. Why does the TCP window behave in such a manner, rather than have a linear slope? (Put another way, why would it be bad if region A had a linear slope?)

c) Assume that the network has an MSS of 1000 bytes and the roundtrip-time between sender and receiver of 100 milliseconds. Assume at time 0 the sender attempts to open the connection. Also assume that the sender can "write" a full window's worth of data instantaneously, so the only latency you need to worry about is the actual propagation delay of the network.

C1) How much time has progressed by point c ?

C2) How much time has progressed between points E and F?

7.Consider a simplified TCP's AIMD algorithm where the congestion window size is measured in number of segments, not in bytes. In additive increase, the congestion window size increases by one segment in each RTT. In multiplicative decrease, the congestion window size decreases by half (if the result is not an integer, round down to the nearest integer). Suppose that two TCP connections, C_1 and C_2, share a single congested link of speed 30 segments per second. Assume that both C_1 and C_2 are in the congestion avoidance phase. Connection C_1's RTT is 50 msec and connection C_2's RTT is 100 msec. Assume that when the data rate in the link exceeds the link's speed, all TCP connections experience data segment loss.

a. If both C_1 and C_2 at time t_0 have a congestion window of 10 segments, what are their congestion window sizes after 1000 msec?

b. In the long run, will these two connections get the same share of the bandwidth of the congested link? Explain.

8. Consider the TCP procedure for estimating RTT. Suppose that $\alpha = 0.1$. Let $SampleRTT_1$ be the most recent sample RTT, let $SampleRTT_2$ be the next most recent sample RTT, and so on.

a. For a given TCP connection, suppose four acknowledgments have been returned with corresponding sample RTTs: $SampleRTT_4$, $SampleRTT_3$, $SampleRTT_2$, and $SampleRTT_1$. Express EstimatedRTT in terms of the four sample RTTs.

b. Generalize your formula for *n* sample RTTs.

c. For the formula in part (b) let *n* approach infinity. Comment on why this averaging procedure is called an exponential moving average.

9. Consider a scenario with two hosts, Alice and Bob. A web server running on Alice is trying to send data to a browser on Bob. For each TCP connection, Alice's TCP stack maintains a buffer of 512 bytes and Bob's TCP stack maintains a buffer of 1024 bytes. For simplicity, assume TCP sequence numbers began at 1 in this problem.

2.A) Bob's stack received up to byte 560 in order from Alice, although its browser has only read up to the first 60 bytes. What will be the ACK and window in the TCP headers that Bob next sends to Alice?

2.B) Later in the same connection, Alice's congestion window is set to 1 MSS = 536 bytes and the advertised receive window from Bob is 560 bytes. The last ACK that Alice received from Bob is byte 700, and the last byte that Alice sends to Bob is byte 900.

2.B.i) What is the smallest byte number that Bob will not accept?

2.B.ii) Assuming that Alice doesn't receive any more ACKs and her window does not change, what is the greatest byte number that Alice can send?

2.B.iii) Again assuming that Alice doesn't receive any additional ACKs, how many more bytes can the web server running on Alice write to its network socket before blocking?

10. Consider the following behavior of a TCP connection (using the congestion control algorithm we learned in class). A time 0, a TCP sender initiates a connection. As soon as the connection is established, the TCP sender will begin sending data. The MSS is 1KB and RTT is 100 ms.

A) Assuming the connection does not lose any data or experience any timeouts, at what time will the sender's congestion window be 16KB? Right after the sender's congestion window has reached a size of

16KB, a timeout occurs. After the timeout is detected, the sender continues sending more data over the established connection.

B) Assuming no additional packet loss or timeouts, how long (since the observed timeout) will it take for the congestion window to build to size 14KB?

C) While its congestion window is at 14KB, the sender receives three acknowledgements for the same sequence number. How long after receiving the third acknowledgement will it take for the sender's congestion window to be at least 9KB again?