CS 421 HW#1, due Nov. 8, 2010

1) Given the following parameters for a datagram packet switching network:

   N: number of hops between two given stations;
   L: total number of bits to be Xmitted;
   B: common data rate, in bits/second, on all links;
   H: number of overhead bits per packet;
   P: packet size, in bits (excluding overhead bits);
   D: propagation delay per hop, in seconds.

   Derive a general expression for end-to-end delay between the two stations. Assume L is a multiple of P, and that the processing and queueing delays in each node are negligible.

2) Suppose that you are required to use the Stop-and-Wait protocol for reliable communication over a 100 Mbps (100×10^6 bps) channel with an end-to-end distance of 6000 km. Each packet sent by the sender is 1000 bytes long. Assume that the speed of propagation is 3×10^5 km/s and ignore all other sources of delay other than the propagation delay. ACK packets have negligible transmission times. What is the utilization of the channel, U, which is defined as the fraction of time the sender is busy sending bits into the channel?

3) Consider a 1 Mbps channel with a 20 msec one-way propagation delay, i.e., 40 msec roundtrip propagation delay. We want to transfer a file of size 13500 Bytes. Each packet has a total size of 1625 Bytes including the 125 Bytes header, i.e., each packet contains 1500 Bytes of data. When there is data to be transmitted, each packet contains the maximum number of bytes. Assume that ACK packets are of 125 Bytes long and there is a processing delay of 1 msec after a packet is fully received at the receiver until the transmission of the corresponding ACK is started. **Selective Repeat** protocol is used with a window size of N = 4 packets. Assume that every 6^{th} packet crossing the forward channel is lost while ACKs are not lost or corrupted. Assume that the processing delay at the sender after an ACK is received is negligible. How much time is required to complete the transfer of the whole file and receive the final ACK at the sender? Assume that the timeout for each packet is set to 50 ms starting from the end of the transmission of the packet.

4) Redo Problem 3 assuming that **Go-Back-N** protocol is used again with a window size of N = 4 segments and the timeout for each window is set to 60 msec starting from the beginning of the sender's window (timeout for the window is restarted when a new ACK is received and the sender's window advances or when a timeout occurs, as discussed in class.)

5) Suppose a TCP connection experiences round-trip times (RTT) of 20 msec for 60% of its packets, 40 msec for 30% of its packets and 100 msec for 10% of its packets. Suppose no packets are actually lost. Assume that the estimated RTT (according to the exponential weighted moving average) is equal to the true (ensemble) average of RTT, i.e., EstimatedRTT = average value of RTT = E[SampleRTT].

   a) Assume that the timeout is set to **2 (two)** times the estimated RTT (as in the original version of TCP), i.e., TimeOut = 2 x EstimatedRTT. What fraction of the packets will be assumed lost by the TCP sender?
   b) Currently used versions of TCP estimate both the mean and the mean deviation as we discussed in the class (the mean deviation is the average absolute distance of RTT samples from the estimated RTT), and sets the timeout to the estimated mean (estimatedRTT) plus **4 (four)** times the estimated deviation (devRTT), i.e., TimeOut =

EstimatedRTT + 4 x devRTT. Assume that the TCP connection uses this new method, and the estimated RTT and deviation are equal to their true (ensemble) values, i.e., EstimatedRTT = E[SampleRTT] and devRTT = E[|SampleRTT-EstimatedRTT|]. What fraction of the packets will be assumed lost by the TCP sender?

6) Consider the RTT estimation algorithm for setting the retransmission TimeOut used by TCP as we discussed in the class:

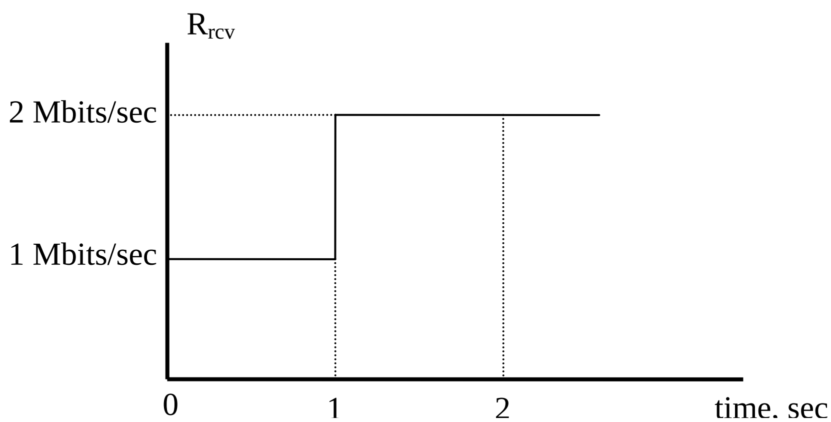
$$EstimatedRTT_{new} \leftarrow (1 - \alpha)\ EstimatedRTT_{old} + \alpha\ SampleRTT$$

$$DevRTT_{new} \leftarrow (1 - \beta)\ DevRTT_{old} + \beta\ |EstimatedRTT_{old} - SampleRTT|$$

$$TimeOut \leftarrow EstimatedRTT_{new} + \gamma\ DevRTT_{new}$$

One of the requirements of this algorithm, although not explicitly stated, is that the TimeOut should be at least equal to the most recent measured RTT, i.e., TimeOut ≥ SampleRTT. Obtain a relation among α, β and γ such that this requirement is always satisfied. The following values are used in TCP: $\alpha = 1/8$, $\beta = 1/4$ and $\gamma = 4$. Determine whether these values satisfy the expression you obtained. (*Hint*: Consider the case when we measure an RTT which is larger than the current value of the RTT estimate, i.e., SampleRTT ≥ $EstimatedRTT_{old}$.)

7) Suppose that R_{rcv}, the rate at which bits are arriving to a TCP receive buffer, is given in the following figure as a function of time. The application process at the receiver is removing bits from the receive buffer at the constant rate of 1 Mbits/sec (1×10^6 bits/sec). Assume that the receive buffer is initially empty and it has a fixed size of 150,000 Bytes. What is the value of the Receive Window advertised by the receiver at t = 2 sec?



8) Suppose that the TCP congestion window, CongWin, at a TCP sender is currently 7 KB and the slow start threshold, ssthresh, 10 KB (for simplicity assume that 1KB=1000 Bytes). Assume that the maximum segment size, MSS, is 1000 Bytes. After the sender sends 7 segments for a total of 7 KB of data and receives all the ACKs for the data transmitted, what will be the final value of CongWin in units of KB?