THE UNIVERSITY OF BRITISH COLUMBIA
Computer Science 317, Section 101
Quiz 3
November 19, 2018

Last Name: ______________________________

First Name: ______________________________

Student ID: ____________________

Date	November 19, 2018
Time Period	11:05 a.m - 11:50 a.m.
Duration of Exam	45 minutes
Number of Test Pages	8 pages (pages double-sided)
Total Possible Marks	50
Additional Materials Allowed	BASIC CALCULATOR

Instructions

1. Write your name and ID number at the top of this page.
2. Please note that there are questions on both sides of the page.
3. Answer the questions in the spaces provided. If you require additional space to answer a question, please use the second last page and refer to this page in your solutions. You may tear off the last page to use for rough work.
4. **Your grade will be influenced by how clearly you express your ideas, and how well you organize your solutions. Numerical answers should be in exact values.**
5. **DO NOT WRITE FORMULAS ON THIS COVER PAGE.**

1. (12 points) Answer the following questions in a sentence or two.

(a) (2 points) On a given link, with a fixed protocol, order the following rates, according to their relationship between each other, from largest to smallest: goodput, bandwidth, and throughput.

> **Solution:** BW $\geq$ throughput $\geq$ goodput

(b) (2 points) What is jitter?

> **Solution:** Varying inter-packet delay times

(c) (2 points) How is propagation delay affected if the length of the packet is increased?

> **Solution:** It is not affected at all.

(d) (2 points) With regards to bandwidth (either high or low bandwidth) and with regards to latency (either high latency or low latency), under which combination of bandwidth and latency is a sliding window protocol especially better. Why?

> **Solution:** The advantage of sliding window is its' ability to have more packets in flight to make full utilization of link. We can have more packets in flight when there is high bandwidth and high latency.

(e) (2 points) What two words best complete the following sentence about TCP. It is important for RTT estimation in TCP to use both the __________ and __________ in order to calculate a time-out value.

> **Solution:** average and variance (accepted estimatedRTT adn sampleRTT)

(f) (2 points) Four bits are used for sequence numbering in a sliding window protocol used in a computer network. To work properly the send window size must be less than or equal to what value?

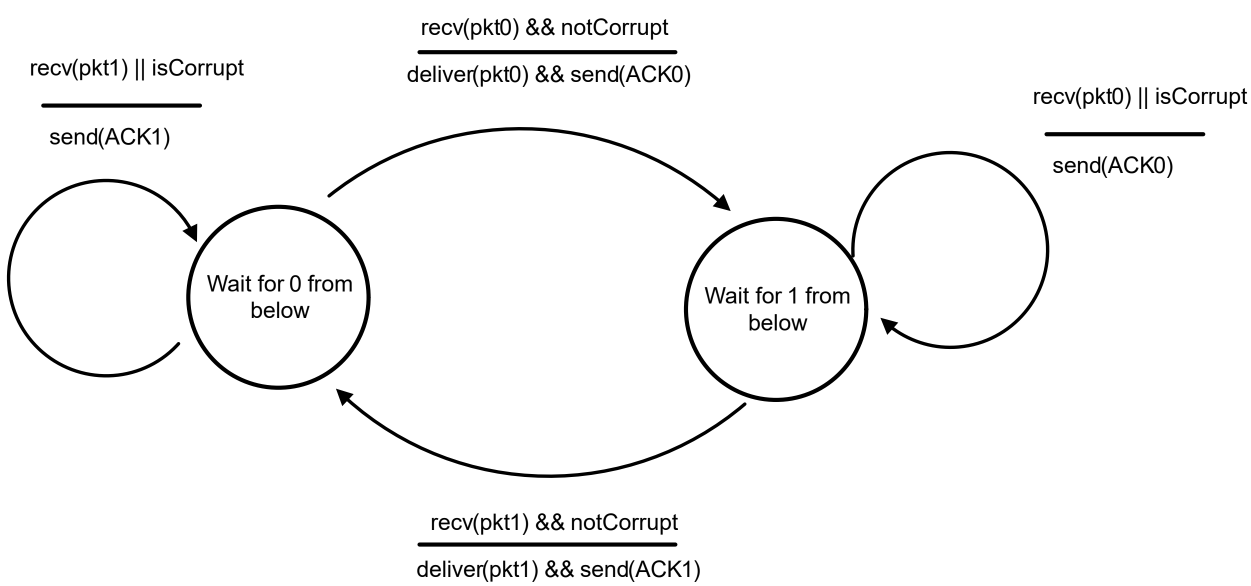
> **Solution:** Because 4 bits are used there are 16 sequences numbers (2^4). We know that $\#SeqNumbers \geq SWS + RWS$ and since the RWS is at least one this means the SWS is less than or equal to 15.

2. (8 points) The diagram below gives a partially completed finite state machine for the *receiving side* of the reliable protocol from Kurose and Ross that handles both corrupt packets and loss (rdt 3.0). Complete the diagram by labelling each arc (events/actions) with all appropriate actions and events from the list below:

`deliver(pkt0)`, `deliver(pkt1)`,
`recv(pkt0)`, `recv(pkt1)`, `recv(ACK0)`, `recv(ACK1)`,
`send(ACK0)`, `send(ACK1)`, `send(pkt0)`, `send(pkt1)`,
`timeout`, `settimer`, `isACK1`, `isACK0`, `isCorrupt`, `notCorrupt`

Not all events or actions from the above list are needed. Functions `send()` and `recv()` take a parameter indicating a 0 or 1 packet, or acknowledgment of 0 or 1 packet (e.g., `send(ACK0)` sends an acknowledgement for a packet with sequence number 0 to the sender side). Function `deliver()` delivers the packet to the upper layer. You can also use the symbols `&&` and `||`, as used in Kurose and Ross to denote "and" and "or" operators.

Solution: Remember rdt3.0 was the standard protocol that was the basis of our sliding window protocol where (a) there is NO timer on the receiver, (b) whenever we received the last packet again (waiting for zero but received ptk1) we needed to send ACK1 to notify the sender of the last packet received (this happens when an ACK is lost).



3. (10 points) Consider a router with a queue large enough to hold 2,000 packets. Assume that packets arrive at the queue with an average rate of 4,000 packets per second and the average packet length is 2,250 bits. The transfer rate out of the router is 10 Mbps.

(a) (2 points) What is the transmission time for an average length packet?

> **Solution:** This asks for transfer time (number of bits)/(number of bits/second transferred).
>
> The link speed is 10 bits per microsecond, so a 2,250 bit packet can be sent in 225 microseconds or 0.225 milliseconds $= 2250/10 \times 10^6$

(b) (2 points) What is the traffic intensity?

> **Solution:** From the formula for traffic intensity we have $I = \frac{La}{R}$ (bits in/bits out), where L is 2250 bits/packet and a is 4,000 packets/sec giving 9 Mbps. R is 10 Mbps, giving the traffic intensity to be $I = \frac{2250 \times 4000}{10 \times 10^6} = 0.9$

(c) (2 points) Assuming that packets arrive randomly, what is the average wait time in the system?

> **Solution:** Since packets are arriving randomly we need to use the formula $S/1 - U$ (remember U is just the I calculated above), which calculates the average delay. We have S, the packet service time, which in this case is simply the transmission delay, is $2250/10 \times 10^6$ or 225 microseconds(same as calculated in part (a)) which divided by $1 - 0.9$ is 2250 microseconds.

(d) (2 points) What is the average number of packets in the queue?

> **Solution:** Since the average delay is 2250 microseconds and each one is serviced in 225 microseconds this means there are 9 in the queue and 1 being serviced.

(e) (2 points) What is the average number in the queue, if the average arrival rate is 5,000 packets per second?

> **Solution:** By re-calculating the traffic intensity, $I = \frac{2250 \times 5000}{10 \times 10^6} = 1.125$
> We see the traffic intensity is 1.125, so the queue will be full all the time with packets being dropped. The answer is 2000
> Note: a very similar question was on the practice quiz.

4. (6 points) Two nodes A and B are 4,000KM apart on a fiber optics link where bits travel at 2×10^8 meters/second. The transmission rate for the link is 2 Mbps and the packet size is 1000 bytes. Answer the following questions.

   (a) (2 points) What is the transmission delay and the propagation delay for sending one packet from node A to node B.

   Solution:
   propagation delay: $4000 \times 1000/2 \times 10^8 = 2 \times 10^{-2} = 20$, 20 milliseconds
   transmission delay: $1000 \times 8/2 \times 10^6 = 4 \times 10^{-3} = 4$, 4 millisecond.

   (b) (2 points) Suppose A and B run a sliding window protocol to ensure reliable delivery. What is the minimum sending sliding window size to keep the link fully utilized?

   Solution:
   Draw a picture! From the picture you will see the total time is the transmission delay of one packet plus RTT time, which is 4 + 2*20 = 44.

   So in 44 milliseconds we are able to put 44/4 packets on the link to keep the link 100% utilized.

   Based on the above we did develop this formula in the tutorial $U = K \times \frac{(L/a)}{RTT+(L/a)}$, where $K \geq 1$ is the number of packets in flight. When $U = 1$ that gives us 44/4.

   (c) (2 points) Using the sliding window protocol from (b) that fully utilizes the link what is the total time to send a 5 Mbyte from node A to B? (Assume that 1 Megabyte is 10^6 bytes).

   Solution: We don't have to worry packets here because we are told that we are fully utilizing the link. This means the time to send 5 Mbyte is equal to the transmission delay plus propagation delay.

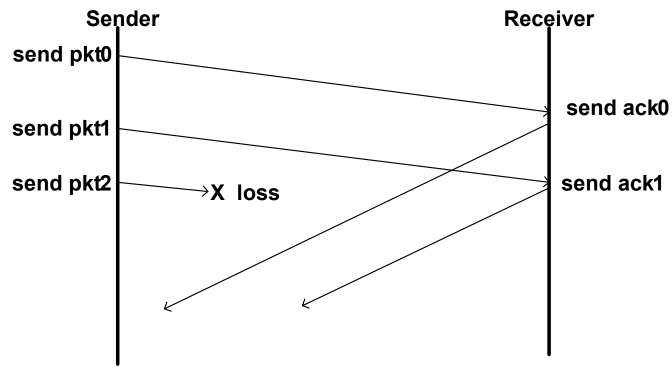
   This is $\frac{8\times(5\times10^6)}{2\times10^6} = 20$ seconds transmission time and 20 milliseconds for propagation. In total the time is 20.020 seconds.

   We accepted 20 seconds plus a couple of propagation delays. The "time to send" is ambiguous and could have included the propagation delay of a return ack.

5. (6 points) Sliding Window Protocol

(a) (3 points)

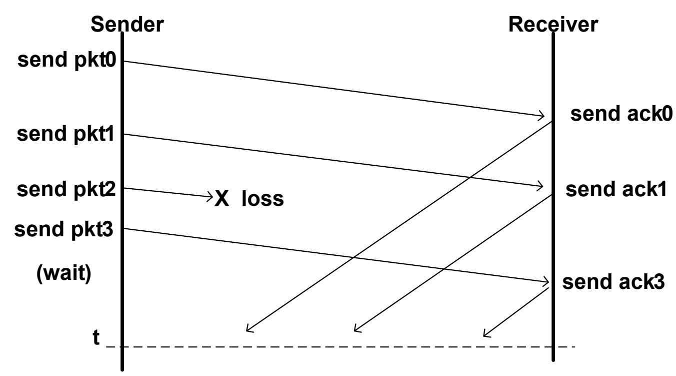
Consider the sliding window protocol in the figure to the right. Does this figure indicate that Go-Back-N (GBN) is being used, Selective Repeat (SR) is being used, or there is not enough information to tell? Explain your answer, why it can or cannot be SR or GBN.



Solution: There is not enough information to tell, since both GBN and SR will individually ACK each of the first two messages as they are received correctly. It can be either GBN or SR.

(b) (3 points)

Consider the sliding window protocol in the figure to the right. Does this figure indicate that Go-Back-N is being used, Selective Repeat is being used, or there is not enough information to tell? Explain your answer, why it can or cannot be SR or GBN.



Solution: It cannot be the GBN protocol because GBN would not have sent an ack3 when pkt3 was received with packet 2 missing. GBN would have discarded packet 3 and acked packet 1.

It can be the SR protocol. With a recevie window of 2 or more, SR will save pkt3 and ack pkt 3 and simply wait for the sender to time-out and re-transmist pkt 2

6. (8 points) Professor Bitwise decided to reduce the number of bits in the TCP header used to store sequence numbers and acknowledgements from 32 bits to only 16 bits so that sequence numbers now range from 0 to 65,535. Let's assume we are on a standard Ethernet link with maximum transmission unit of 1520 bytes with a 20 byte Ethernet header, 20 byte IP header, the newly modified 16 byte TCP header. Answer the following questions about Professor Bitwise's idea. (Answers within one decimal point are good).

   (a) (2 points) Calculate the protocol overhead for sending TCP data over the Ethernet link as described in the above.

   Solution: It is $(20 + 20 + 16)/1520 = 0.0368$, but fine to put 56 as well.

   (b) (2 points) Now assuming the link is 10 Mbps, calculate the time required for the Ethernet link to send the segments covering the entire range of sequence numbers 0 to 65,535.

   Solution: The segments covering the entire range of sequence numbers means the time to send 65,536 bytes, $65536 \times 8/10 \times 10^6 = 2^{19}/10^7 = 0.0524288$ around 52.4 milliseconds. You should have accounted for overhead so it is $52.4 + 52.4 \times 0.0368 = 54.3$ milliseconds

   (c) (2 points) Suppose now the propagation delay on the 10 Mbps link is 100 milliseconds. Calculate the maximum throughput of TCP on this link?

   Solution: The important thing to note is that it is NOT possible to send more than one window full of bytes at a time. Therefore, like question 6(b), we calculate the transmission time for one window (part (b)) over the RTT plus transmission time. The answer is $(54.3/254.3) \times 10 \times 10^6$ or around 2Mbps. It was not necessary to be very exact with the answers.

   (d) (2 points) Can you suggest any problems with Professor Bitwise's idea? Was the savings worth it?

   Solution: What I hoped you would see is that it would **not** be possible to be able to fully utilize any link because after every 50 milliseconds (approximately) you need to wait for an ack. It just isn't possible to fill a pipe for reasonably sized network.

   It is also correct to say, that it won't work because of out-of-order packets may arrive with sequence numbers inside the small sequence number range we have. On the Internet the assumption is 2 minutes (far greater than 50 milliseconds).
   This is a real problem because of current speed of networks and why the advertised window size (which is 16 bits) is auto-scaled.