Question 1 Packet switching and circuit switching: (8 points)

(a) Is the Internet a packet switching or circuit switching network? Justify your answer. (2 points)

Packet switching

(b) If your answer for (a) was "packet switching", provide an example of a network that uses circuit switching technology; if your answer for (a) was "circuit switching", provide an example of a packet-switching network. In both cases, justify your answer. (2 points)

The telephone service is an example of a network that uses circuit switching technology. A circuit is made between two phones before you can begin talking on the phone.

(c) Suppose you were recruited to take part in the group who is designing the Future Internet. To do that, you and your colleagues are considering that the main applications driving the Internet of the Future will be real-time services such as audio and video streaming, distributed games, etc. Would you propose a design based on packet switching or circuit switching? Justify your answer. (4 points)

I would recommend using packet-switching technology. Audio/Video streaming use large amounts of bandwidth, and the inability to make use of idle resources on a circuit switching design make it an impractical choice.

Question 2 Network performance: (8 points)

(a) One metric of network performance we covered in class is data loss. Explain how data loss can be used as an indicator of how the network is performing. (2 points)

Data loss typically indicates that come component of a network is improperly configured, broken, or needs to be upgraded to be able to handle the amount of traffic.

(b) In class, we discussed different ways loss can occur as data is transferred over the network. List and provide a brief explanation of the different types of data loss we discussed. (2 points)

Congestion – When more packets try to traverse a link than it can handle, data can be lost due to congestion. In this situation, the buffers on the network devices fill up, and packets will be dropped.

Errors – When there are errors during transit, packets will fail checksum tests, and be dropped.

(c) Latency is another way to measure network performance. One way to account for network latency is to measure the round-trip time (RTT). What is the round-trip time? (2 points)

The round trip time is the amount of time that it takes to travel to another host, and back again.

(d) Is the RTT constant or variable? Explain your answer. (2 points)

Variable. It will fluctuate based on the current utilization of the network.

Question 3 List one advantage and one disadvantage of: (8 points)

(a) Peer-to-peer model for networked applications (when compared to the client-server model). Explain. (2 points)

Benefits: No single point of failure, bandwidth use spread across entire network.

Disadvantages: Administration can be difficult due to no central server.

(b) DNS caching. Explain. (2 points)

Advantage: Potentially faster lookup times.

Disadvantage: Ensuring information is up to date.

(c) Layering. Explain. (2 points)

Advantage: Modularity and compatibility/interoperability between vendors.

Disadvantages: Increased complexity.

(d) DNS' distributed database (compared to a centralized name service). Explain. (2 points)

Advantages: Not overwhelming a single server

Disadvantages: Potentially longer lookup times (need to forward requests up and down the hierarchy)

Question 4 Alice was studying for her midterm exams in the library when her friend Bob calls to tell her he just posted a collection of her favorite video clips on his Web site hosted at www.coolsites.com. (18 points)

(a) Alice immediately tries to access Bob's Web site. Describe the steps that need to happen before Alice's machine at QueensAcademy.edu can issue a request for Bob's Web site hosted at www.coolsites.com. Assume that this is the 1st time content from coolsites.com is requested by a someone at QueensAcademy.edu. Explain your answer. (4 points)

Alice -> QueensAcademy.com

QueensAcademy.edu -> root (reply w/ com NS)

QueensAcademy.edu -> com (reply w/ coolsites.com NS)

QueensAcademy.edu -> coolsites.com (reply w/ requested resource records)

QueensAcademy.edu -> Alice (requested resource records)

Alice would make a DNS request to the DNS NS at QueensAcademy.edu. The QueensAcademy.edu DNS NS would check it's cache, and find that it had no cached entry for coolsites.com. It would then make a request to the root NS for coolsites.com, and would learn the DNS NS for .com.. It would then make the request to the .com DNS NS, and receive the DNS NS for coolsites.com. Finally, it would make the request to the coolsites.com DNS NS, and receive the DNS resource records requested.

(b) Soon thereafter, Bob calls his other friend Carla, who also goes to Queen's Academy. Carla quickly tries to watch Bob's video from her machine at QueensAcademy.edu by issuing a request to Bob's Web site at www.coolsites.com. What steps need to happen before Carla's request is sent to www.coolsites.com. Assume Alice's request has already been issued. Explain your answer. (4 points)

Carla -> QueensAcademy.edu

QueensAcademy.edu -> Carla (requested resource records)

In this situation, the record would be cached, so Carla would make the request, and the QueensAcademy.edu NS would reply right away without needing to refer to the root NS.

(c) Alice is finally able to download Bob's Web site. There are seven videos embedded in it. The total processing/service time within the network is 20ms and the one-way propagation delay is 120ms. Assume that transmission delay is negligible. What is the response time, i.e., the time between when the browser on Alice's machine requests the videos from www.coolsites.com and when they are delivered, assuming Alice's browser uses non-persistent HTTP? Explain your answer and show your work. (4 points)

Assuming non-persistent HTTP, each item would require 1 RTT to establish the TCP connect, then 1 RTT + transmission time to transfer the object. Therefore, each object would take 2RTT + Transfer time.

In this situation, there are 8 objects (1 HTML file, 7 videos). Therefore, the total time is:

8(2(240ms) + 20ms) = 4000ms = 4s

(d) What would be the response time if Alice's browser uses persistent HTTP? Explain and show your work. (2 points)

Using persistent HTTP, the time would be 1RTT to set up the connection, then 1RTT + Transfer time for each object. Therefore, the total time would be:

1RTT + 8(RTT + Transfer)

240ms + 8(240ms + 20ms) = 2320ms = 2.32s

(e) Now, it's Carla's turn to download Bob's videos. Assuming the steps in (b) have already been executed and that the hit ratio for QueensAcademy.edu's cache is 50%, what is the average response time Carla experiences for each object in Bob's Web page? Suppose that the delay to access an object from within Queen's Academy is 15ms (4 points)

On a question like this, please specify if you used persistant or non-persistant HTTP. Both examples will be shown:

Persistent:

240ms + 8(.5(240ms + 20ms) + .5(15ms)) = 1.34s

Non-Persistent:

8(240ms + .5(240ms + 20ms) + .5(15ms)) = 3.02s

Question 5 Reliable data delivery transport protocols employ several mechanisms including feedback, checksums, sequence numbers, retransmissions, and retransmission timers. (12 points)

(a) What is the specific problem retransmission timers try to address when used to accomplish reliable data transfer? Explain. (3 points)

Lost/dropped packets. This is the situation where the sender will never receive an ACK/NACK for a packet, so it needs to resend it.

(b) What is the retransmission timeout? (2 points)

The time that a sender will wait for an ACK before resending the packet.

(c) What is the trade-off in defining the value of the retransmission timeout? (2 points)

A higher retransmission timeout gives a packet more time to reach it’s destination, which could save bandwidth if the packet is not completely lost, but simply experiencing delay reaching it’s destination. This comes at the cost of higher latency.

(d) Protocols that employ negative acknowledgments (NACKs) should also use positive acknowledgments (ACKs). Why is that the case? (3 points)

If only NACKs were sent, the sender would never know if a packet was lost in transmission, since the receiver can only NACK packets that are received, but have errors.

(e) Why do reliable data transfer protocols need sequence numbers? (2 points)

Sequence numbers are needed to ensure in-order delivery, as well as to prevent duplicate packets from being sent.

Question 6 Suppose that Alice is sending Bob a message over the Internet. Alice’s computer is directly connected to router1 and Bob’s to router2. router1 and router2 are directly connected. Thus, Alice’s communication with Bob goes to router1, then to router2, and finally to Bob’s computer. (12 points) Assume the message is small enough that it does not need to be broken down into smaller units as it is processed by the lower layers. Illustrate your answer using a diagram.

(a) What happens to Alice’s message as it gets ready to be transmitted, i.e, as it goes down the protocol stack on Alice’s computer? Show what happens to the message at each protocol layer. Is this process called encapsulation or de-encapsulation? (3 points)

At each layer on Alice’s computer, packets are encapsulated, and headers are added to the packet. For example, source and destination port numbers are added at the Transport Layer. This process is called encapsulation.

(b) What happens to the message as it is processed by router1? Show what happens at each layer. (3 points)

Router1 will look at the destination IP address, which it will use to determine which interface to send the packet out of. It will then send the packet out the interface towards router2 by using information stored in it's routing table.

(c) And at router2? Show what happens at each layer. (3 points)

At router2, the router will again look at the headers, and see that the destination is a network attached locally. It will send the packet out the local interface towards Bob's PC.

(d) What happens to the message when it arrives at Bob's computer as it goes up the protocol stack? Is this process called encapsulation or de-encapsulation? Show what happens at each layer. (3 points)

When the message gets to Bob's computer, it will travel back up the protocol stack. The headers will be removed at each layer. When the packet arrives at the Transport layer, the PC will use the destination port to determine which application to deliver the data to.

Question 7 "Pipelined" protocols, also known as "sliding window" protocols, are a type of ARQ (Automatic Repeat Request) protocols that use a "window" to control the amount of data they inject into the network. (14 points)

(a) Stop-and-Wait is one type of ARQ protocol. What is the window size of Stop-and-Wait? Explain (2 points)

The window size of stop-and-wait is 1. This is because the protocol requires that we receive an ACK before we can send another packet.

(b) Based on your answer for (a), how many unique sequence numbers does Stop-and-Wait need? How many bits are needed to represent Stop-and-Wait's unique sequence numbers? Explain. (2 points)

The window size of stop-and-wait is 2. This is necessary to avoid issues of duplicate packets being sent and uncaught when an ACK is lost.

(c) What is the main advantage of requiring a smaller number of bits to represent the range of unique sequence numbers employed by a protocol? What is the main disadvantage? (3 points)

The advantage of a smaller number of bits is less overhead/data required in the header of each packet. The main disadvantage is that a small number of bits limits the size of the window that you can use (i.e. less packets "in flight").

(d) For a 100Mbits/sec channel with 100ms propagation delay, what is the channel utilization when sending 2KByte segments if Stop-and-Wait is used? Show your work. (3 points)

(L/R) / (L/R + Delay) = (16,000 / 100,000,000) / ((16,000 / 100,000,000) + .2s)

.075% utilization

(e) How can you increase channel utilization? Explain using a numeric example to support your solution. (3 points)

Pipelining by increasing the window size. Example of increasing to 8:

(8 * (L/R)) / (L/R + Delay) = (8 * (16,000 / 100,000,000)) / ((16,000 / 100,000,000) + .2s)

.6% utilization

(f) Describe the additional complexity of your solution compared to Stop-and-Wait. (3 points)

This solution would require additional bits used to sequence numbers, as well as additional timers and buffers to hold packets awaiting ACKs/etc.

Question 8 Suppose that a sender and a receiver are using ARQ to perform reliable data delivery. (20 points)

(a) How many sequence numbers are needed to implement Stop-and-Wait? (2 points)

2 sequence numbers, in order to handle the case when an ACK is lost.

(b) In a Go-Back-N ARQ protocol, the window size is 6. Segments with sequence numbers 1, 2, 3, 4 and 5 have been sent. The sender just received an ACK for segment 1. Segments 6, 7, 8, 9 and 10 are waiting to be sent. Draw the time diagram showing this scenario. (3 points)

(c) Which segment(s) can the sender send before it must wait for the next ACK from the receiver? Explain. (3 points)

It can send 6 and 7. The window size is 6, and we currently have packets 2, 3, 4, and 5 outstanding. Since there are only 4 outstanding packets, we have space to send 2 more.

(d) Some time later, the sender transmitted segments 20, 21, 22, 23, 24, and 26; however, segment 22 got lost. If Go-Back-N is used, what segment(s) would the sender have to retransmit? Explain. (3 points)

The sender would need to retransmit 22, 23, and 24. This is because we "go back" to segment 22 and retransmit everything beyond it as well.

(e) Suppose the same situation as above but sender and receiver use Selective-Repeat ARQ. What segment(s) would the sender need to retransmit? Explain. (3 points)

In selective repeat, we retransmit only segment 22. This is because every packet is ACKd, so we know specifically which packet(s) to retransmit.

(f) Can Selective-Repeat ARQ use cumulative ACKs? Explain. (3 points)

No, Selective-Repeat requires individual ACKs for each packet in order to know which packets to retransmit.

(g) What are the trade-offs between Go-Back-N ARQ and Selective-Repeat ARQ? (3 points)

Go-Back N is "pessimistic" about the fates of packets near any packets which are lost. It assumes that if a packet is lost, it's likely that additional packets near it will also be lost. It trades potentially lost bandwidth (since we retransmit everything after the lost packet automatically) for potential proactively resending lost packets. Selective Repeat is "optimistic" about the fates of the packets near the lost packet. It assumes that the packets will get there, and only resends them if required. It also requires some additional bandwidth, since every packet needs to be ACKd.