Student ID: ____________________

CS457: Computer Networking
Date: 3/20/2007
Name: ________________________

Instructions:
1. Be sure that you have 9 questions
2. Be sure your answers are legible.
3. Write your Student ID at the top of every page
4. This is a closed book exam
5. Answer each question clearly and to the point. Show all work and assumptions, but do not define or describe concepts unless asked to do so; assume that the graders are familiar with the concepts.

Question	***Points***	***Score***
1	10	
2	10	
3	10	
4	15	
5	10	
6	10	
7	15	
8	10	
9	10	
total	**100**	

Student ID: ____________________

1. Answer the following True/False questions by circling either **T** or **F.**

    1. Transport layers can provide guarantees about bandwidth, reliability, and latency
       T F
    2. FTP is a stateless protocol T F
    3. DNS is an application level protocol that runs over TCP T F
    4. HTTP's conditional GET prevents any messages from being sent over the Internet on a cache hit
       T F
    5. POP3 is a stateless protocol T F
    6. ICANN must authorize the creation of every new domain name. T F
    7. Transport protocols are used by every node on the Internet T F
    8. GBN does not buffer out of order packets, so a receive buffer is not needed T F
    9. Selective Repeat allows the sender and receiver windows to be unsynchronized T F
    10. OSPF runs over TCP T F

Student ID: ____________________

2. Packet Switching and Circuit Switching

a. Describe the difference between a packet switched network and a circuit switched network, and indicate when it is more advantageous to use each.

A packet switched network sends packets on-demand, with no reservations. Each packet is addressed to a destination host.

A circuit switched network requires a connection to be established before two hosts can send messages to each other. Each packet is addressed by the circuit number.

Circuit switched networks are preferred when bandwidth constraints must be met, or when traffic will be constraint. Packet switched networks are preferred when low initial latency is required, or when traffic will be bursty.

b. Calculate how long it takes to send a 1Mb file over a 1.5Mbps link with

-- a circuit switched network that uses 24 different frequency slots and a 500msec connection establishment time

latency = 1/ (1.5/24) + .5 = 16.5 seconds

-- a 10-hop packet switched network where each link is a 1.5Mbps link. Assume no congestion and no packet segmentation.

latency = 1/ (1.5) * 10 = 6.66 seconds

c. Redo your calculations for the packet switched network above assuming your maximum segment size (MSS) is 1KB.

latency = time to send pipeline + forwarding delay
= 1 / 1.5 + .001/1.5*10 = .673 seconds

Student ID: ___________________

3. Sockets

a. When one host sends a UDP packet to another host, what values will the receiving host use from the packet headers to direct the segment to the appropriate socket?

The destination address and the destination port

b. When one host sends a TDP packet to another host, what values will the receiving host use from the packet headers to direct the segment to the appropriate socket?

The source address, the source port, the destination address, and the destination port

c. Can a client process open multiple TCP connections to the same server process? Why or why not?

Multiple TCP connections can be opened because the client machine automatically assigns a new, unique source port for every new out going connection, giving that connection a unique 4-tuple.

Student ID: ___________________

4. Web caches

Assume that a local area network sees about 250 web requests per second, with an average object size of 10Kb. The 10Mbps LAN is connected to the Internet over a 3Mbps link. The average web server on the Internet takes 1.5 seconds to respond to a web request, and the router on the link to the Internet has an average 15 second queuing delay for incoming traffic due to its high utilization.

a. Calculate the average latency for each web request.

latency = internet latency + queing latency + link transport time + LAN transport time
= 1.5 + 15 + 10/3000 + 10/10000 = 16.5 seconds

b. Calculate the average latency for each web request if the link to the Internet were upgraded to a 10Mbps link.

This upgrade would bring the link utilization down from 83% to about 25%, eliminating the queuing delay. Thus:

latency = internet latency + queing latency + link transport time + LAN transport time
= 1.5 + 10/10000 + 10/10000 = 1.5 seconds

c. Calculate the average latency for each web request if a web cache were added to the LAN and had a 40% hit rate.

This upgrade would reduce incoming traffic by 40%, bringing the link utilization down from 83% to about 50%, eliminating the queuing delay. Furthermore, the latency due to HTTP's conditional GET will be minimal since it will only be paid once per connection, not once per object. Thus:

latency >= .6 * 1.5 seconds +
.4 * (10/10000) = .9 seconds

Student ID: ____________________

5. Checksums

a. Fill in the checksum field in the header of this UDP packet:

Source Port #: 1111111111111110
Destination Port#: 0000000000000010
Length: 0000000000001100
Checksum: ??
Data: 0000000000000010 0000000000010000

```
          1111111111111110
+         0000000000000010
--------------------------------------------------------------------
          10000000000000000 --> wraparound -->   0000000000000001
                                                 0000000000001100
                                                 0000000000000010
+                                                0000000000010000
--------------------------------------------------------------------
                                                 0000000000011111
```

Checksum is 1's complement = **1111111111100000**

b. When the receiving host receives this packet, how would it check the checksum? What value would it need to compute in order to assume that the packet has no bit errors?

It would add all 16-bit integers in the packet, including the checksum. It would need to computer a 16-bit integer with all 1's in order to assume the packet has no bit errors.

Student ID: ____________________

6. Pipelining

a. With no pipelining, what is the sender utilization (the percentage of time that the sender is busy sending) when using a 100Mbps link with a maximum segment size (MSS) of 2.5KB and a round trip time (RTT) of 10ms?

(L/R) / (RTT + (L/R)) = (2.5*8/100) / (10 + (2.5*8/100) ~= 1.96% utilization

b. What is the sender utilization when using a pipeline of 30 packets? 60 packets?

Pipelines of size N increase utilization by a factor of N.
30 packet pipeline --> 58.8% utilization
60 packet pipeline --> 100% utilization

c. Name 3 factors that would make you want to limit the size of the sender window when using the Go-back-N algorithm? When using the Selective Repeat algorithm?

GBN:

1. With larger windows, more messages need to be retransmit every time a packet is lost
2. The window size can be no larger than the sequence number
3. A window size longer than (RTT + L/R) / (L/R) does not improve performance

SR:

4. With larger windows, the transmitter must have a larger receive buffer
5. The window size can be no larger than the sequence number
6. A window size longer than (RTT + L/R) / (L/R) does not improve performance

Student ID: ____________________

7. TCP Congestion Control

Assume a host running TCP Reno (which came after TCP Tahoe) is connected to a link with maximum segment size (MSS) of 2.5KB. The host initializes *CongWin* to 1 segment and *Threshold* to 16 segments. The round trip time RTT on this link is 50ms.

a. What is the value of CongWin after 300msecs?

18 MSS. CongWin grows to threshold in 4 RTT = 200msecs, and then converts to congestion-avoidance for 2RTT =100msecs.

b. Assume a packet is sent at time=300msecs and is lost and a triple ACK occurs. What is the new value of CongWin? The value of threshold?

9MSS for both. They are divided in half when a triple ack occurs.

c. Assume a packet is sent at time=300msecs, is lost, and a timeout occurs. What is the new value of CongWin? The value of threshold?

CongWin is set to 1MSS, but threshold is set to 9MSS.

d. Why might it be easier to perform congestion control on a virtual circuit network such as ATM than a packet switched network?

A virtual circuit network is based on *reserved paths*, so it has more information about what kind of traffic it might see in the future, and it also has the ability to reject requests for bandwidth.

Student ID: ____________________

8. Switching

A router needs to get packets from the input port buffers to the output port buffers through a switching fabric.

a. Why is it faster to use a bus to interconnect the input buffers with output buffers instead of connecting them through a central memory store? Assuming the memory store is connected to the same type of bus, how much faster is it?

It should be twice as fast because the bus is only used once per packet transfer instead of twice for both a read and a write into memory.

b. Assume you have N input ports and N output ports connected by a bus that can transfer a packet in 1/Nth the time it takes to transfer a packet on the input/output links. Will your input buffers ever overflow? Your output buffers? Why or why not?

The input buffers will never overflow because any packet that is received can always be sent to the output buffers before the next packet arrives. However, if all packets from all input ports are destined for the same output port, that output port buffer could overflow.

c. Assume you have N input ports and N output ports connected by a bus that can transfer a packet in the same time it takes to transfer a packet on the input/output links, but it can transfer N packets simultaneously. Will your input buffers ever overflow? Your output buffers? Why or why not?

The input buffers can overflow due to "Head-of-the-Line" blocking. The output buffers can overflow in the same circumstances as (b).

9. Inter-AS routing

a. Why do we have different routing algorithms for inter-AS and intra-AS routing?

Intra-AS routing is oriented towards efficiency while inter-AS routing is oriented towards scalability and providing autonomy of routing policies.

b. When BGP sends a routing advertisement, it appends an AS-PATH, which contains all of the ASs through which this advertisement has passed. Name two ways the AS-PATH is used?

1. The AS-PATH is used for route optimization: routes with long AS-PATHS are discarded
2. The AS-PATH is used for policy enforcement: routes through or from non-customer ASs may not be forwarded.