

Student ID: _____

CS457: Computer Networking

Date: 3/20/2007

Name: _____

Instructions:

1. Be sure that you have 8 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.

<i>Question</i>	<i>Points</i>	<i>Score</i>
1	10	
2	10	
3	15	
4	10	
5	10	
6	15	
7	10	
8	20	
total	100	

Student ID: _____

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

1. The Internet Protocol provides no delivery guarantees T F
2. Persistent and non-persistent connections are equivalent for getting only a single object
T F
3. P2P networks hide your identity from the authorities T F
4. Cookies allow one to maintain state across HTTP sessions T F
5. FTP is said to use “out of band” communication because it does not conform to TCP standards
T F
6. Email messages will go through at least 2 SMTP servers T F
7. The Kazaa network is like a hybrid of the Napster and Gnutella networks T F
8. A minimal transport layer does nothing more than multiplexing/de-multiplexing T F
9. GBN uses cumulative acknowledgments T F
10. RIP runs over TCP T F

2. IP Addresses

a. Describe how CIDR introduces a trade-off between the size of our routing tables and the number of wasted addresses in our address space?

CIDRized addresses allow sets of nodes to be defined in terms of an arbitrary length prefix. Compared to flat routing, this allows shorter routing tables, but subnets may waste addresses if the number of machines are smaller than the assigned address space. Compared to classful routing (e.g. prefix lengths can only be a multiple of eight), more prefixes need to be stored in the routing tables but subnets can be smaller, allowing for fewer wasted addresses.

b. Describe two technologies that address the problems caused by the number of devices on the Internet approaching or exceeding our 32-bit address space? What problems do these cause?

1. DHCP allows an ISP to reuse IP addresses for different users when only a small percentage of the subscribers are online at a time
2. NAT allows a network to connect multiple machines using a single IP address

Both of these techniques make it difficult to run a server because the server will not have a known, publicly addressable IP address.

c. Will these technologies become obsolete with Ipv6, which uses a 64-bit address space?

No. DHCP will still be useful for mobile devices and WiFi hotspots.

d. Besides increasing the size of the address space, describe 3 things that Ipv6 introduces that should make routing simpler and faster.

1. fixed length headers
2. no fragmentation allowed at the routing layer
3. no checksums at the routing layer

3. Persistent and Non-persistent connections

Assume that you want to retrieve a web page that has 6 images and 1 Java applet. The Java applet retrieves 3 more images before it can run.

a. How many messages must be sent when using non-persistent HTTP before this web page can be viewed ? How many RTTs?

A total of 11 objects must be retrieved, each requiring a new TCP connection. Thus, 44 messages will be sent, requiring 22 RTTs.

Answers indicating that FIN packets must be sent were given full credit, although this was not required for the answer.

b. How many messages must be sent when using persistent HTTP with no pipelining? How many RTTs?

Only one connection must be opened, so 24 messages will be sent requiring 12 RTTs.

c. How many messages must be sent when using persistent HTTP with pipelining? How many RTTs?

The first object, the next 7 objects, and the last 3 objects will each require their own RTTs. Thus, 24 messages will be sent, requiring 4 RTTs.

d. Would you expect persistent connections and pipelining to give you a bigger benefit over non-persistent connections in a high bit-rate network or a low bit-rate network? Why?

In a high bit-rate network. In a low bit-rate network, the transfer times are dominated by the time it takes to transmit the data.

e. Would you expect persistent connections and pipelining to give you a bigger benefit over non-persistent connections in a high latency network or a low latency network? Why?

In a high latency network, where the hosts will frequently stall while waiting for connection establishment.

4. DNS

a. Describe all of the DNS messages that must be sent in order to retrieve a URL such as <http://www.google.com>. Assume no cache hits and assume iterative queries.

1. source machine would send a DNS request to the local default name server.
2. The default name server would send a DNS request to a root server, and get a response
3. The default name server would send a DNS request to a TLD server, and get a response
4. The default name server would send a DNS request to an authoritative server, and get a response
5. The default name server would send the responses to the original request

b. Now assume recursive queries.

1. the source machine would send a DNS request to the local default name server
2. the default name server would send a request to the root server
3. the root server would send a request to the TLD server
4. the TLD server would send a request to the authoritative server, and get a response
5. the TLD server would respond to the root server
6. the root server would respond to the local default name server
7. the default name server would respond to the original request

c. Using iterative queries again, now assume your local default name server has the entry for the appropriate TLD server cached.

1. the source machine would send a DNS request to the local default name server.
2. The default name server would get a cache hit for the TLD server
3. The default name server would send a DNS request to a TLD server, and get a response
4. The default name server would send a DNS request to an authoritative server, and get a response
5. The default name server would send the responses to the original request

5. Reliable Transport

If you had a completely reliable communication layer, your reliable transport layer would not need to do much: it would simply send each packet and, upon reception, deliver it to the application layer.

a. What reliability mechanisms would you need to add if your channel introduced bit errors?

1. Checksums
2. Acknowledgements (ACKs) and/or Negative Acknowledgements (NACKs)
3. Sequence numbers (not required)

b. What reliability mechanisms would you need to add if your channel also lost packets?

Timeouts

6. TCP Acknowledgements

Assume a TCP host is expecting sequence number 2847. Describe what the TCP host does in each of the following scenarios:

a. The last packet received was already acknowledged. A new packet arrives with sequence number 2847 and 253 bytes of data in the message payload.

Wait up to 500ms for next segment. If no next segment, send ACK with sequence number 3100

b. The last packet received not yet been acknowledged. A new packet arrives with sequence number 2847 and 253 bytes of data in the message payload.

Immediately send ACK with sequence number 3100

c. The last packet received was just acknowledged. A new packet arrives with sequence number 3100 and 177 bytes of data in the message payload.

Buffer packet. Immediately send a duplicate ACK with sequence number 2847

d. The last packet received had sequence number 3100 and 177 bytes of data in the message payload. A new packet arrives with sequence number 2847 and 253 bytes of data in the message payload.

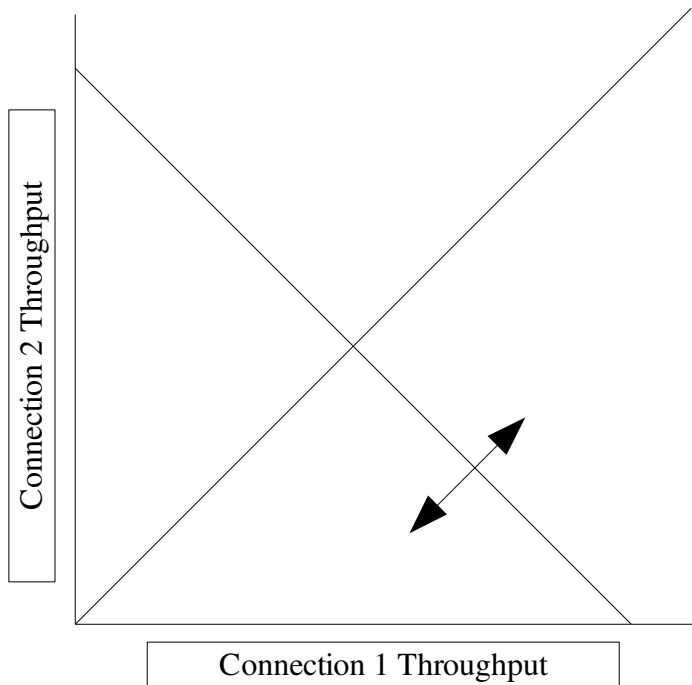
Immediately send ACK with sequence number 3277

7. TCP Fairness

a. We showed in class that TCP congestion control also provides fair utilization to two competing session. What does this mean about whether or not you can hog all of the bandwidth at a WiFi hotspot?

Nothing. You can simply open multiple TCP connections to get a larger percentage of the bandwidth. Furthermore, you can simply use UDP, which does not have congestion control.

b. If TCP decreased the congestion window linearly instead of multiplicatively, would it still converge to fair utilization of a link when shared between two TCP connections? Argue why or why not, using the graph below.



TCP would no longer converge to fair utilization because the additive increase and additive decrease mechanisms would keep the fairness ratio moving along the slope of 1 or -1, respectively; ie. if it did not start on the line $x=y$, it would never approach that line.

8. Distance Vector routing

a. Fill in the route calculations below for the Distance Vector algorithm, using the topology on the right. Then, use the results to fill in the routing table for node *x*.

node x table

		cost to		
		x	y	z
from	x	0	1	3
	y	∞	∞	∞
	z	∞	∞	∞

node y table

		cost to		
		x	y	z
from	x	∞	∞	∞
	y	1	0	5
	z	∞	∞	∞

node z table

		cost to		
		x	y	z
from	x	∞	∞	∞
	y	∞	∞	∞
	z	3	5	0

		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	5
	z	3	5	0

		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	4
	z	3	5	0

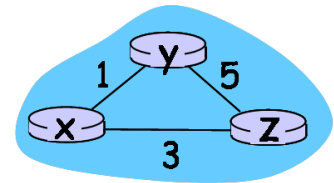
		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	5
	z	3	4	0

		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	4
	z	3	4	0

		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	4
	z	3	4	0

		cost to		
		x	y	z
from	x	0	1	3
	y	1	0	4
	z	3	4	0

time →



destination	link
x	--
y	(x,y)
z	(x,z)

b. Name 2 differences between generic Distance Vector routing and the RIP algorithm.

- 1) RIP has a 15 link maximum
- 2) RIP sends a distance vector with at most 25 destinations
- 3) RIP sends advertisement messages every 30 seconds
- 4) RIP tables send the destination, path length, *and* the next node
- 5) RIP times out a link after 180 seconds