

NET ID \_\_\_\_\_

**CS519, Prelim (March 17, 2004)**

NAME: \_\_\_\_\_

You have 50 minutes to complete the test.

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**Q1. 2 points**

Write your NET ID at the top of every page of this test.

**Q2. X points**

Name 3 advantages of a circuit network over a packet network:

1 \_\_\_\_\_

2 \_\_\_\_\_

3 \_\_\_\_\_

**Q3. X points**

Name 3 advantages of a packet network over a circuit network:

1 \_\_\_\_\_

2 \_\_\_\_\_

3 \_\_\_\_\_

**Q4. X points**

Which of the following best characterizes a “long fat pipe” (i.e. high delay x bandwidth product)? Check one:

\_\_\_\_\_ OC12 link from between the east and west coasts of the USA

\_\_\_\_\_ 1Mbps link from New York City to Tokyo Japan

\_\_\_\_\_ 100 Gbps LAN

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**Q5. X points**

Two hosts with private addresses 10.9.8.7 and 10.9.8.8 respectively both have LAN interfaces. Do you know for sure that they are connected to the same LAN? (Assume that they are properly connected and addressed etc.)

\_\_\_\_\_ YES                      \_\_\_\_\_ NO

Why or why not? \_\_\_\_\_

\_\_\_\_\_

**Q6. X points**

Two hosts with global addresses 20.9.8.7 and 20.9.8.8 respectively both have LAN interfaces. Do you know for sure that they are connected to the same LAN? (Assume that they are properly connected and addressed etc.)

\_\_\_\_\_ YES                      \_\_\_\_\_ NO

Why or why not? \_\_\_\_\_

\_\_\_\_\_

**Q7. X points**

Why isn't Internet routing run as one huge unified routing domain (rather than the 2-tier intra-domain and inter-domain architecture we have today)? Check each reason that is true for the 2-tier architecture, but would not be true for a unified routing domain architecture.

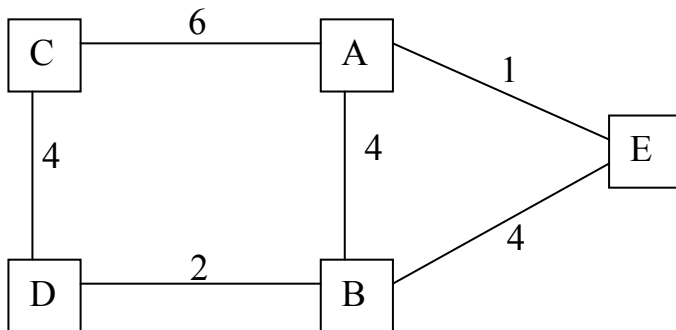
\_\_\_\_\_ Routing convergence would be too slow (i.e. after a change)

\_\_\_\_\_ Sites want to run different routing protocols

\_\_\_\_\_ External routing errors may effect routes within a site (i.e. between two hosts in the same site)

\_\_\_\_\_ ISPs do not want to receive routing updates showing the internals of sites (i.e. the subnets in a site)

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**Figure 1: A network with five routers, A - E. Assume that the costs shown are for both directions of the link.**

**Q8. X points**

In the space below, write two RIBs for router E in Figure 1, one of them assuming a Distance Vector (DV) protocol, and the other assuming a Link State (LS) protocol. Clearly label which is which.

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**Q9. X points**

Fill in the contents of the confirmed and tentative lists for each step of a shortest path spanning tree algorithm for Router E in Figure 1 using the (Dest, cost, next-hop) notation. Steps 1 and 2 already been filled in. Note there may be fewer than 12 steps.

Step	Confirmed	Tentative	Step	Confirmed	Tentative
1	(E,0,-)		2	(E,0,-)	(A,1,A) (B,4,B)
3			4		
5			6		
7			8		
9			10		
11			12		

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**Q10. X points**

Name 3 advantages of TCP over UDP:

1 \_\_\_\_\_

2 \_\_\_\_\_

3 \_\_\_\_\_

**Q11. X points**

Name 3 advantages of UDP over TCP:

1 \_\_\_\_\_

2 \_\_\_\_\_

3 \_\_\_\_\_

**Q12. X points**

In a sliding window protocol, normally the receive window is as big as or bigger than the send window. What happens if the send window is bigger than the receive window? Check one:

\_\_\_\_\_ It can no longer be guaranteed that all bytes will be received.

\_\_\_\_\_ It can no longer be guaranteed that all bytes will arrive in order.

\_\_\_\_\_ It can no longer be guaranteed that all bytes will be received or will arrive in order.

\_\_\_\_\_ None of the above

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**Q13. X points**

In class we saw that a small packet (and corresponding small ACK) can be introduced into the TCP “control loop” when the sending application does a send of only a few bytes (and the TCP layer subsequently sends the bytes after a short time waiting for another send). Using only the space provided below, describe another way in which a small packet can be introduced.

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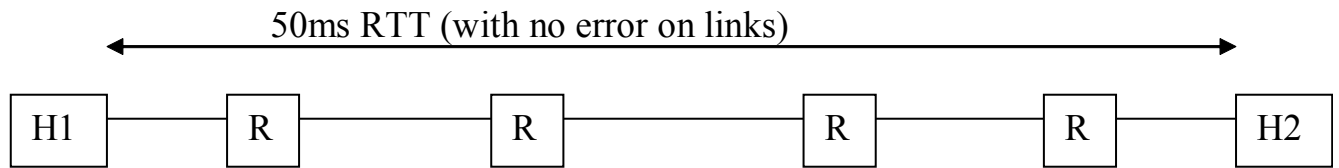
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**Q14. X points**

What are the three defining characteristics of “quality” or “performance” in packet networks?

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**Figure 2:**

**Q15. X points**

In the network of Figure 2, each of the five links exhibits a high error rate (5% packet loss). The RTT between the two hosts is 50ms. The two hosts are running TCP. Imagine that, by introducing a link-level reliable sliding window protocol on each link, you can reduce the error rate to 0% of each link, but in doing so you add 5ms of latency to each link, increasing the RTT to 75ms. Given that the two hosts are running TCP, does the introduction of the link level reliability protocols violate the End-to-end principle?

\_\_\_\_\_ YES

\_\_\_\_\_ NO

Why or why not? \_\_\_\_\_

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**Q16. X points**

From the point of view of a router in a backbone ISP (i.e. one that does not participate in any site routing), what is the structure of the IP address? Please check one of the following:

\_\_\_\_\_ | prefix | suffix |

\_\_\_\_\_ | ISP | Site | suffix |

\_\_\_\_\_ | ISP | Site | Subnet | Host |

\_\_\_\_\_ | Class A/B/C Network | Host |

**Q17. X points**

Image a very large network (thousands of destinations) where only a small fraction of destinations need to be reachable. For instance, this might be the case for a large sensor network where a few locations gather data from the sensors. As a result, the routing algorithm only needs to keep track of a few destinations. Given this, would it make more sense to use a link state routing algorithm or a distance vector routing algorithm?

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**Q18. X points**

What is the primary cause of large routing tables in BGP? Check one:

\_\_\_\_\_ Large address size

\_\_\_\_\_ Site multihoming

\_\_\_\_\_ Increasing number of home networks

\_\_\_\_\_ Increased NAT usage

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**Q19. X points**

When an application does a `send()` on a TCP socket, and the `send()` returns with no errors, what does the application know for sure about the sent data.

Check one:

\_\_\_\_\_ The data has been written into the local TCP's send buffer

\_\_\_\_\_ The data has been transmitted onto the network

\_\_\_\_\_ The data has been received by the remote host's TCP layer

\_\_\_\_\_ The data has been received by the remote host's application

**Q20. X points**

Of the following TCP socket calls, check those that are used by the client and/or by the server.

Used by client	Used by server	Socket call
		<code>connect()</code>
		<code>listen()</code>
		<code>accept()</code>
		<code>recv()</code>
		<code>send()</code>
		<code>close()</code>

**Q21. X points**

With that "fast retransmit" method of TCP retransmission (i.e. after three duplicate ACKs), is it possible for a retransmit to happen sooner than one RTT from the time the lost packet was sent?

\_\_\_\_\_ YES                      \_\_\_\_\_ NO

Why or why not? \_\_\_\_\_

\_\_\_\_\_

\_\_\_\_\_

\_\_\_\_\_

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### Q22. X points

With streaming media, the media is downloaded for a while into a local buffer (often for many seconds) before the media starts to play on the local player. This occurs even if the average throughput between the media server and the player is higher than that required for the media player. Why is this necessary? Check one only (the one which most drives the need for buffering).

\_\_\_\_\_ The buffer eliminates traffic jitter

\_\_\_\_\_ The buffer allows the player to read data in a uniform way regardless of how the stream was received (i.e. UDP, TCP, or even HTTP)

\_\_\_\_\_ The buffer hides the fact that throughput may vary considerably over the length of the stream

\_\_\_\_\_ The buffer allows the player to request retransmissions of lost packets and deliver them in order

### Q23. X points

Imagine that an IP layer sends two packets, one fragmented and the other not fragmented. Suppose also that two simultaneous bit errors on the unfragmented packet cause its ID field to match that of the fragmented packet in such a way that the checksum is still correct (admittedly a low probability event, but possible). Finally assume that at the receiver, the unfragmented packet is received after the first fragment of the fragmented packet, but before the last fragment of the fragmented packet (see below). Does the receiver receive the packets correctly or not?

\_\_\_\_\_ Received correctly

\_\_\_\_\_ Not received correctly

Why or why not? \_\_\_\_\_

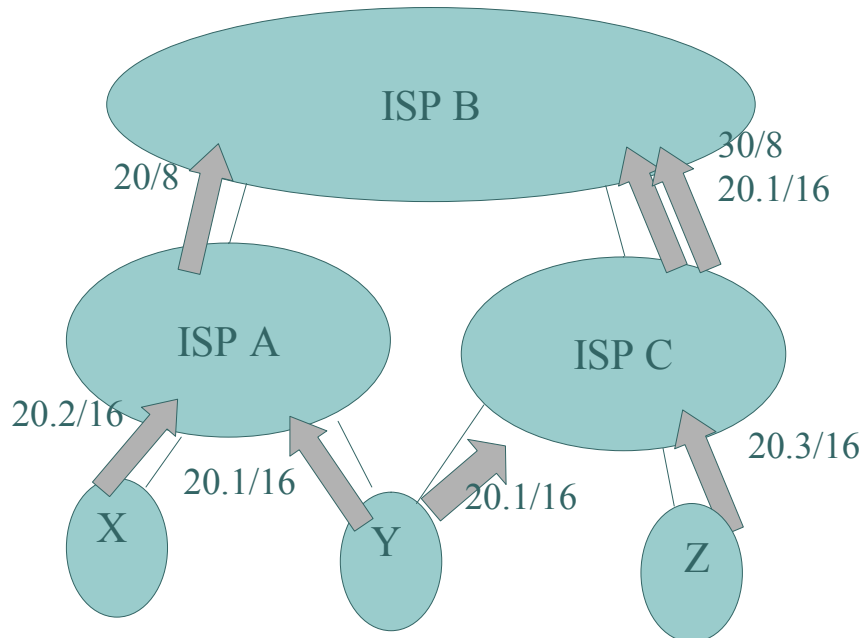
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**Q24. X points**



**Figure 3**

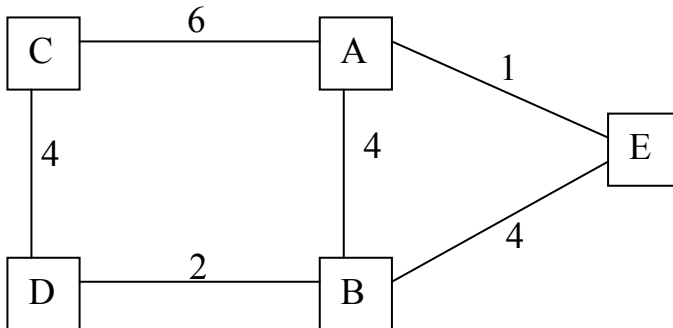
Describe the AS path (assume that all Sites X, Y, and Z, and ISPs are ASs) from each ISP to each Site, using the table below. If there is no path from the ISP to the Site, write “none”.

From-to	AS-path	From-to	AS-path
A→X		B→Z	
A→Y		C→X	
A→Z		C→Y	
B→X		C→Z	
B→Y			

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**Q25. X points**

Referring to the following figure (which is identical to Figure 1), imagine that there is a lot of traffic from router E to router D, so much so that the E-B and B-D links are saturated with traffic. Modify the link costs so that all traffic except the E to D traffic avoids the two saturated links.



**Q26. X points**

Given current IP address assignment policies, what is the theoretical smallest routing table size in the backbone ISP routers. Check one:

\_\_\_\_\_ Number of class A addresses (256)

\_\_\_\_\_ Number of large sites

\_\_\_\_\_ Number of backbone ISPs

\_\_\_\_\_ Number of ISPs of all types

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### Q27. X points

Consider the two traffic sequence diagrams in Figure 4 below. Assume identical time scales, and assume that the same sized data packets are being sent. Assume a send window of 4 packets. Assume that the sender is sending as fast as it can, within the limits of the send window.

Of these two sequences, which has higher bandwidth (select one)?

\_\_\_\_\_ Left sequence

\_\_\_\_\_ Right sequence

Of these two sequences, which has longer latency (select one)?

\_\_\_\_\_ Left sequence

\_\_\_\_\_ Right sequence

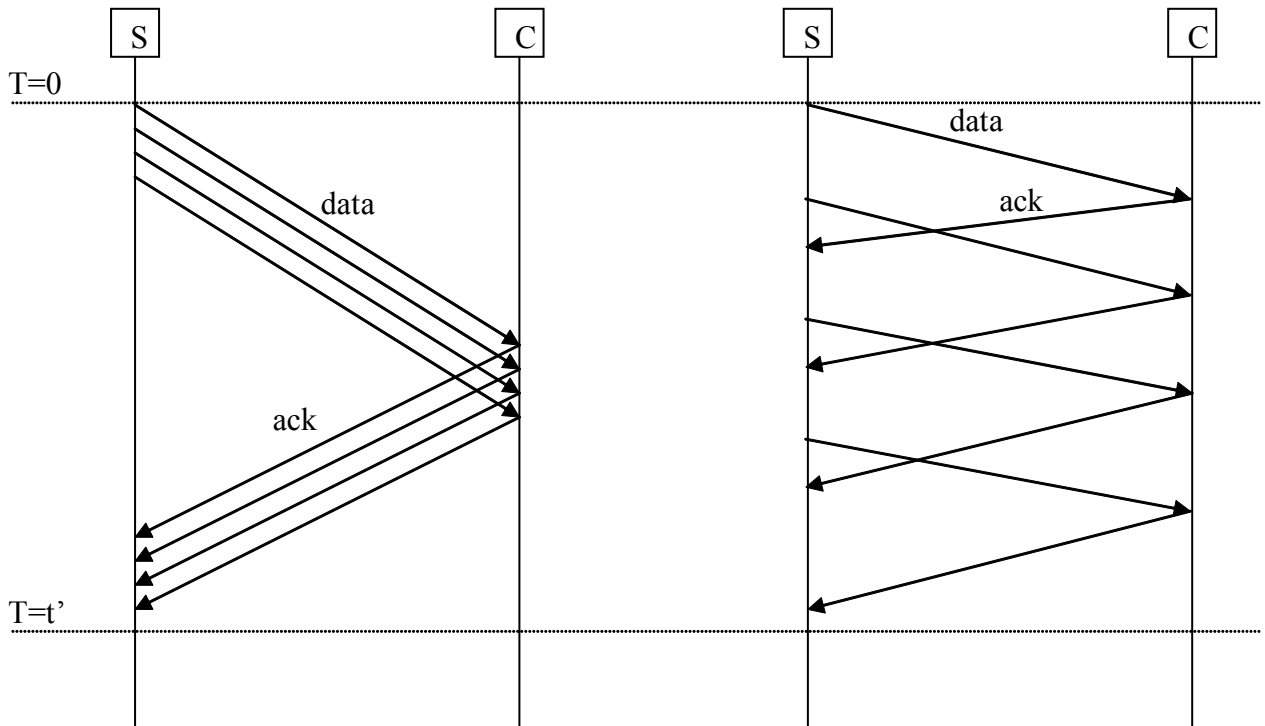


Figure 4

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**Q28. X points**

Describe up to five ways by which an application can either close an open TCP socket, or learn that the other end has closed the socket.

1. \_\_\_\_\_
2. \_\_\_\_\_
3. \_\_\_\_\_
4. \_\_\_\_\_
5. \_\_\_\_\_

**Q29. X points**

For each of the following two scenarios, what contributes most to the total delay, queuing, propagation, or transmit?

\_\_\_\_\_ A 1500 byte packet transmitted over a lightly loaded 10Mbps Ethernet

\_\_\_\_\_ A 10Gbps transmission from Boston to Los Angeles over a high-speed network

**Q30. X points**

To which of the following blocks of addresses does the address 64.65.24.7 belong? Check all that apply:

- |                     |                   |
|---------------------|-------------------|
| _____ 64.65.24.0/28 | _____ 64.65.16/21 |
| _____ 64.64/12      | _____ 1/0         |
| _____ 0/0           |                   |

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**Q31. X points**

The following question applies to Figure 1. Assuming a distance vector routing protocol, which node's RIBs *do not* change if link C-D goes down. Which node's FIBs *do not* change?

RIBs that don't change \_\_\_\_\_

FIBs that don't change \_\_\_\_\_

**Q32. X points**

What effect does BGP policies have on paths in the Internet? Check all that apply:

\_\_\_\_\_ Paths are longer than the shortest possible

\_\_\_\_\_ Paths have low latency

\_\_\_\_\_ Paths are asymmetric

\_\_\_\_\_ Paths cannot always be trace routed.

**Q33. X points**

Assuming a Go-Back-N acknowledgement scheme (with no selective acknowledgement), if a sender sends two packets (1 and 2), and receives back an ACK for the bytes in packet 2, but not for the bytes in packet 1, what does the sender know for sure? Check one:

\_\_\_\_\_ The receiver has not received packet 1

\_\_\_\_\_ Either packet 1 was dropped, or the ACK to packet 1 was dropped

\_\_\_\_\_ The ACK to packet 1 was dropped

\_\_\_\_\_ None of the above



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**Q34. X points**

Assume 1000 byte packets being send over TCP. Assume that the sender has sent 10 packets constituting bytes 10000 through 19999. Assume further that the sender receives a TCP packet with an acknowledgement of 13000, and with a SACK option indicating the ranges 15000-15999, and 18000-19999. Which bytes can the sender remove from its send buffer?

Ans: \_\_\_\_\_