Dashboard / My courses / COMS2014A-CN-2021 / Large Test 2 / Large Test 2

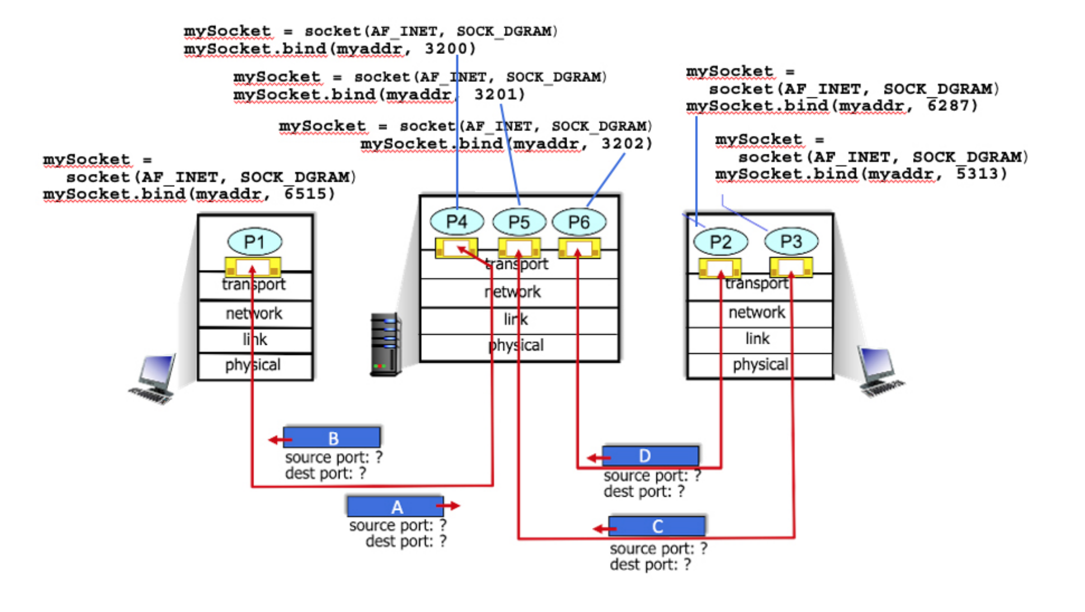
Started on	Friday, 15 October 2021, 4:45 PM
State	Finished
Completed on	Friday, 15 October 2021, 7:15 PM
Time taken	2 hours 30 mins
Grade	**34.03** out of 45.00 (**76**%)

Question **1**

Correct

Mark 1.50 out of 1.50

Consider the figure below, with 6 sockets shown across the network, and the corresponding Python code at each host. There are four UDP segments in flight. Match the source and destination port numbers for each segment with a value below.



Segment C source port #	5313 ✔
Segment B source port #	3200 ✔
Segment D source port #	6287 ✔
Segment A source port #	6515 ✔
Segment C destination port #	3201 ✔
Segment D destination port #	3202 ✔

Question **2**

Incorrect

Mark 0.00 out of 0.50

What is meant by transport-layer multiplexing?

- a. Taking data from multiple sockets, all associated with the same destination IP address, adding destination port numbers to each piece of data, and then concatenating these to form a transport-layer segment, and eventually passing this segment to the network layer.
- b. Receiving a transport-layer segment from the network layer, extracting the payload (data) and delivering the data to the correct socket. ✖
- c. Receiving a transport-layer segment from the network layer, extracting the payload, determining the destination IP address for the data, and then passing the segment and the IP address back down to the network layer.
- d. Taking data from one socket (one of possibly many sockets), encapsulating a data chuck with header information – thereby creating a transport layer segment – and eventually passing this segment to the network layer.

Question **3**

Correct

Mark 0.50 out of 0.50

True or False: When multiple TCP clients send TCP segments to the same destination port number at a receiving host, those segments (from different senders) will always be directed to the same socket at the receiving host.

- a. False ✔
- b. True

Question **4**

Correct

Mark 0.50 out of 0.50

True or False: It is possible for two TCP segments with source port 80 to be sent by the sending host to different clients.

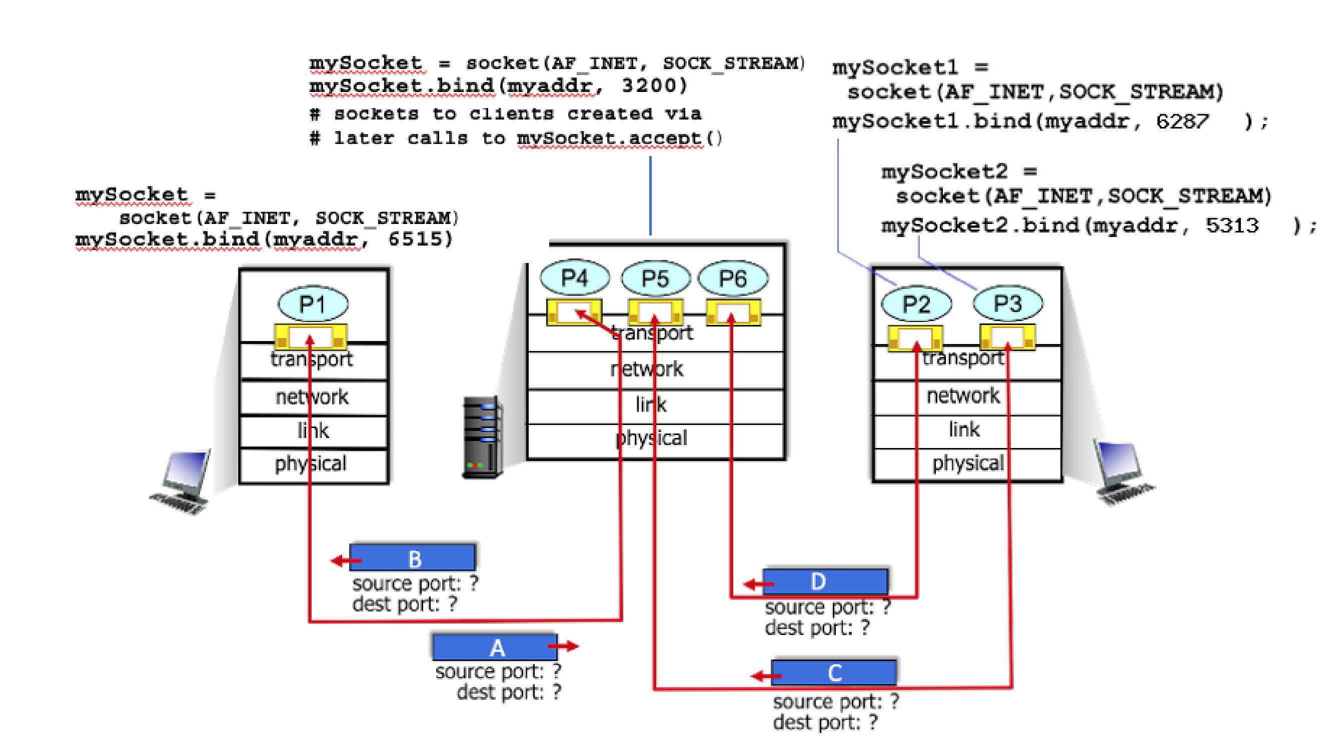
- a. False
- b. True ✔

Question 5

Correct

Mark 1.00 out of 1.00

Consider the figure below, with 6 sockets shown across the network, and the corresponding Python code at each host. There are four TCP segments in flight. Match the source and destination port numbers for each segment with a value below.



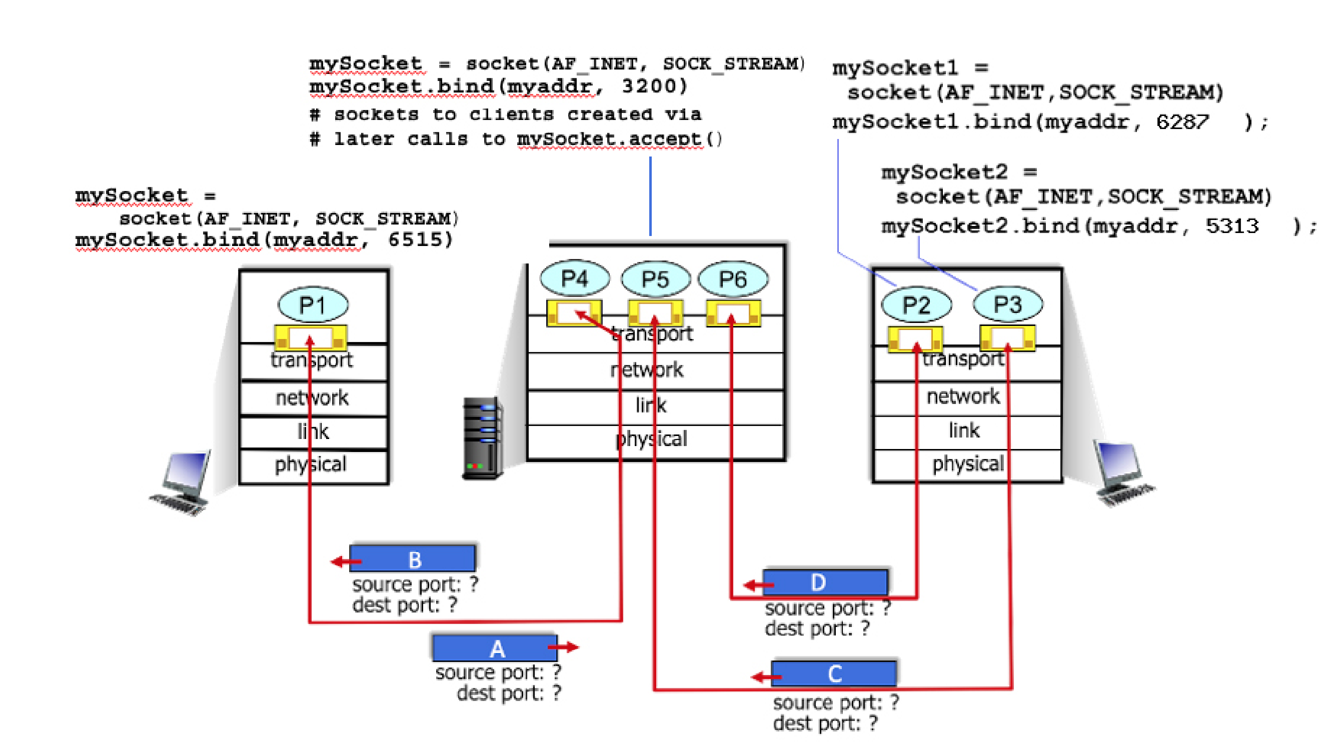
Segment D source port #	6287 ✔
Segment C source port #	5313 ✔
Segment A destination port #	3200 ✔
Segment A source port #	6515 ✔

Question 6

Correct

Mark 1.50 out of 1.50

Consider the figure below, with 6 sockets shown across the network, and the corresponding Python code at each host. There are four TCP segments in flight.



Which of the following source *or* destination port number fields in these four segments has the value of 3200?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. Segment A source port #
- [ ] b. Segment B destination port #
- [ ] c. Segment C source port #
- [x] d. Segment D destination port # ✔
- [x] e. Segment C destination port # ✔
- [x] f. Segment B source port # ✔
- [ ] g. Segment D source port #
- [x] h. Segment A destination port # ✔

Question **7**

Correct

Mark 0.50 out of 0.50

Which of the following datagram and segment header fields are used, when demultiplexing data up to a TCP socket?

- a. None of the other answers is correct.
- b. Destination IP address and port number only.
- c. Destination port number only.
- d. Source and destination IP addresses, and source and destination port numbers. ✔

Question **8**

Correct

Mark 1.00 out of 1.00

Which of the fields below are in a UDP segment header?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. Upper layer protocol
- [x] b. Internet checksum ✔
- [x] c. Source port number ✔
- [x] d. Destination port number ✔
- [ ] e. Source IP address
- [x] f. Length (of UDP header plus payload) ✔
- [ ] g. Data (payload)
- [ ] h. Sequence number

Question **9**

Correct

Mark 0.50 out of 0.50

True or False: When computing the Internet checksum for two numbers, a single flipped bit (i.e., in just one of the two numbers) will always result in a changed checksum.

- a. True ✔
- b. False

Question **10**

Correct

Mark 1.50 out of 1.50

Consider the two sixteen bit numbers:

```
10110100 01000110
11001000 01101110
```

Compute the Internet Checksum of these two values

Enter the 2 bytes each as an 8-bit number with only 0's and 1's, and make a single blank space between the two 8-bit numbers (e.g., 01010101 00101000).

Answer: 10000011 01001010

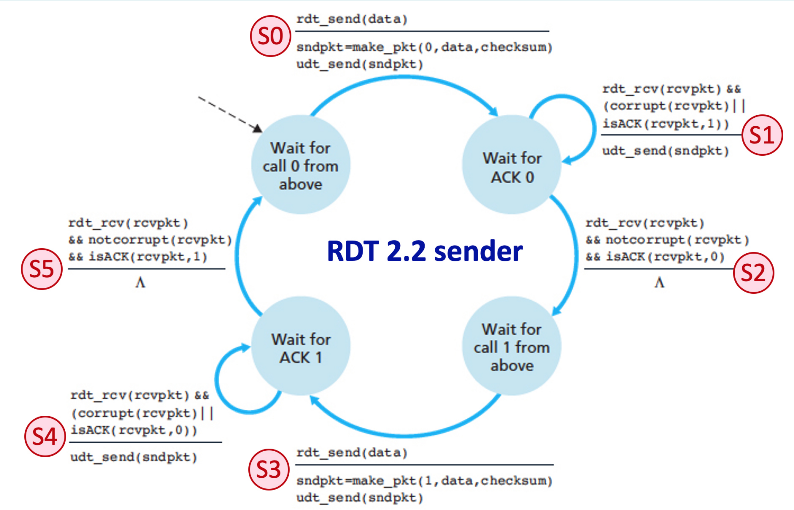
Question **11**

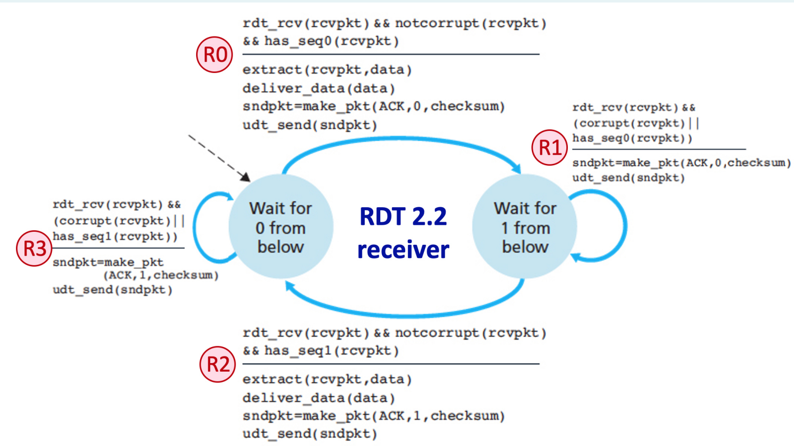
Partially correct

Mark 1.13 out of 1.50

Consider the rdt 2.2 sender and receiver below, with FSM transitions labeled in red.

Which of the following sequences of transitions could possibly occur as a result of an initial rdt_send() call at the sender (with no messages initially in the channel) , and possible later message corruption and subsequent error recovery?





(Select all that apply, there **is** negative marking for incorrect choices!)

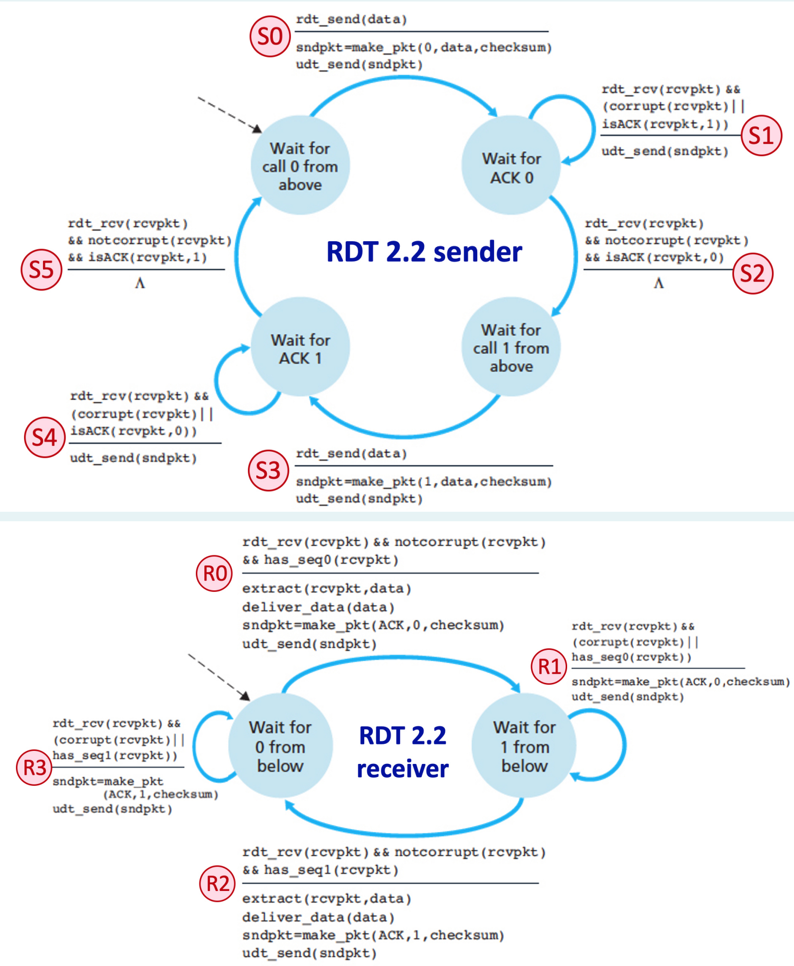
- [ ] a. S0, R0, R2, S2
- [ ] b. S0, R0, S2, R1
- [x] c. S0, R0, S2, S3, R2, S5 ✔
- [ ] d. S0, R0, S1, R2
- [ ] e. S0, R0, S2, R2
- [x] f. S0, R0, S1, R1, S1 ✔
- [x] g. S0, R0, S1, R1 ✔

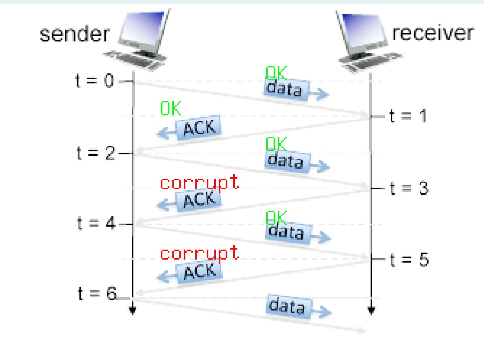
Question **12**

Correct

Mark 1.50 out of 1.50

Consider the rdt 2.2 sender and receiver below, with FSM transitions labeled in red. Also consider the sender and receiver timeline following the FSMs, where a green OK label indicates a message that is not corrupted, and a red corrupt label indicates a message that is corrupted. We are interested in the sequence number (0 or 1) of a data message, and the ACK number (0 or 1) of an ACK message for the messages sent at t= 2, 3, 4, 5 in the figure below.





Which of the following sequences of interleaved data sequence numbers and ACK numbers corresponds to those in the messages sent at t=2 (data sequence number); t=3 (ACK number), t=4 (data sequence number), and t=5 (ACK number)? Only one sequence is correct.

- a. 1, 1, 1, 1 ✔
- b. 1, 1, 1, 0

c. 1, 1, 0, 1

d. 1, 0, 0, 1

e. 0, 1, 1, 0

f. 1, 0, 1, 1

g. 0, 0, 0, 0

h. 1, 0, 1, 0

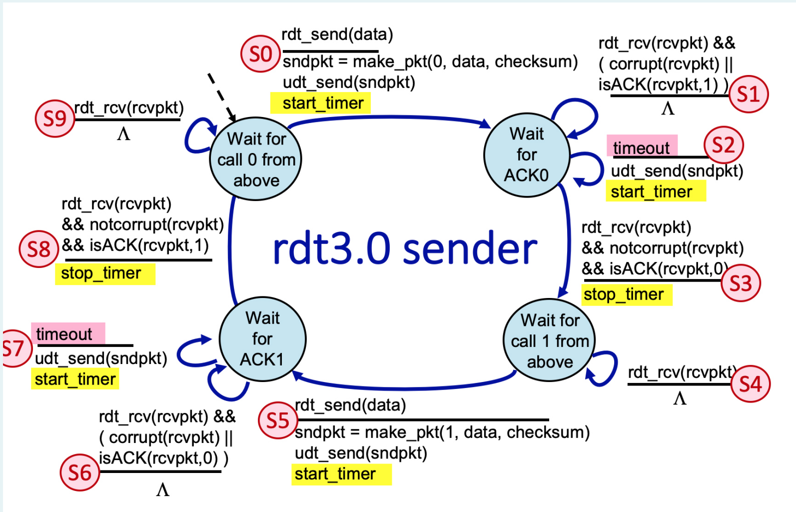
Question **13**

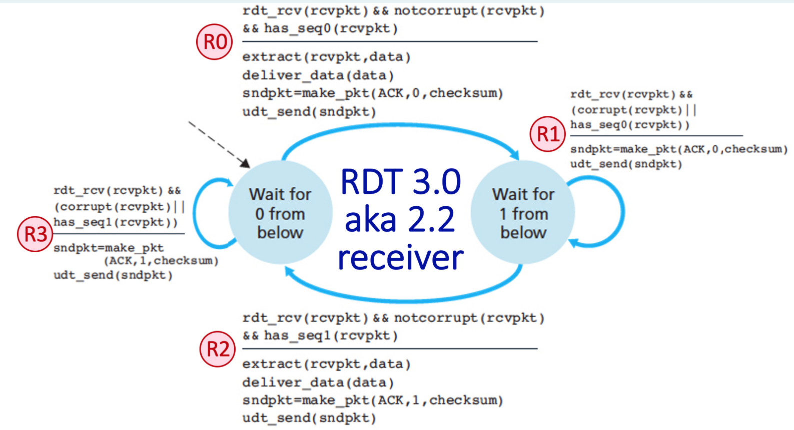
Partially correct

Mark 0.40 out of 1.00

The rdt3.0 protocol (also known as the alternating bit protocol) reliably transfers data from sender to receiver over a channel that can corrupt, lose and delay (but not reorder) packets. The rdt 3.0 sender is shown below, with the new timer operations (the only addition beyond rdt 2.2) highlighted. If you think about it, you'll hopefully realize that the rdt2.2 receiver works also in the rdt 3.0 setting (since it can already handle bit error and duplicates); so we don't need a new rdt 3.0 receiver. The rdt 2.2 receiver (also now the rdt 3.0 receiver) is also shown below.

Which of the following sequences of transitions could possibly occur as a result of an initial rdt_send() call at the sender (with no messages initially in the channel) , and possible later message corruption and subsequent error recovery?





(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. S0, R0, S3
- [ ] b. S0, R0, S3, R2
- [x] c. S0, R0, S1, R1 ✔
- [ ] d. S0, R0, S3, S5, R1
- [ ] e. S0, R0, S2, R1
- [ ] f. S0, R0, S3, R3
- [x] g. S0, R0, S3, S5, R2 ✔

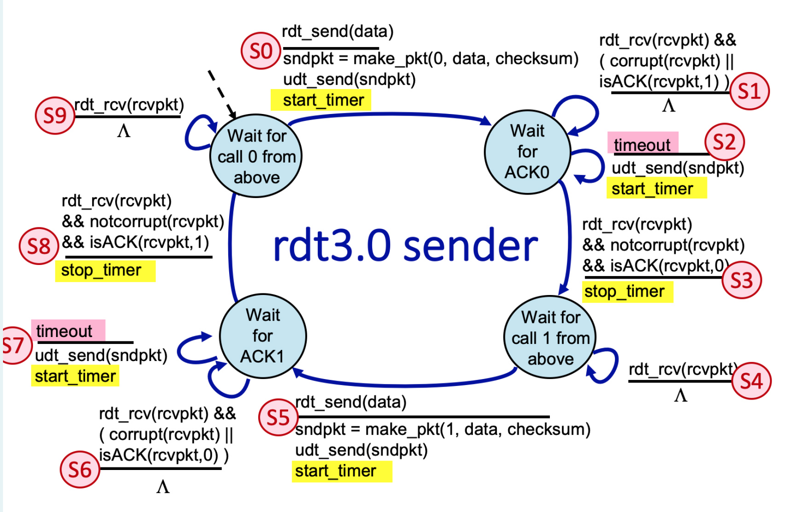
Question **14**

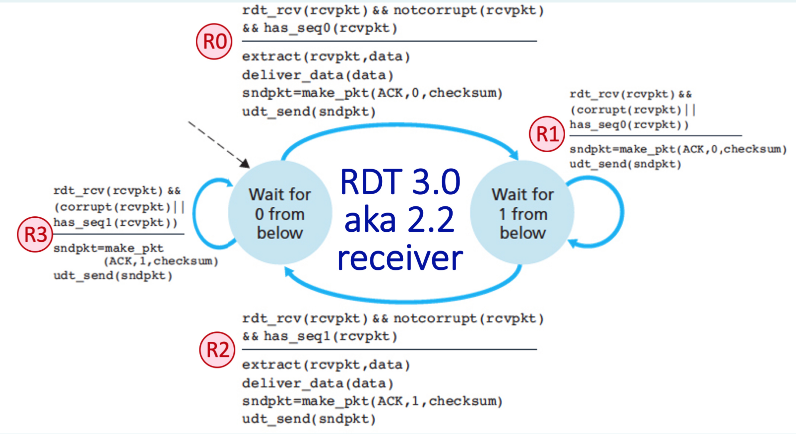
Correct

Mark 1.00 out of 1.00

Consider the rdt3.0 sender and receiver shown below, with FSM transitions labeled in red. Complete the sequence of transitions that the sender and receiver FSMs would make, in an global order, to deliver two messages from sender to receiver, assuming no errors occur (including the ACK received at the sender for the second packet).

The transition sequence is: S0, R0, *t1, t2, t3, t4*. Match unspecified transitions *t1, t2, t3* and *t4* with a labeled transition from the figures below.





Transition *t4* is: S8 ✔

Transition *t1* is: S3 ✔

Transition *t2* is: S5 ✔

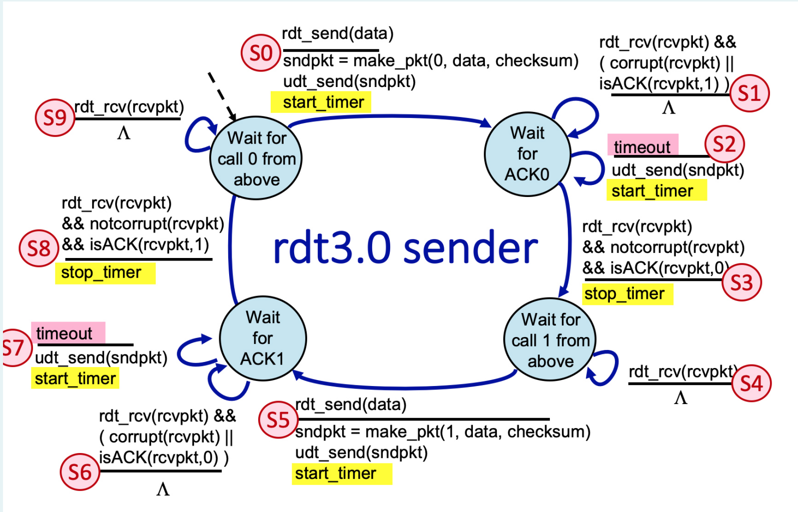
Transition *t3* is: R2 ✔

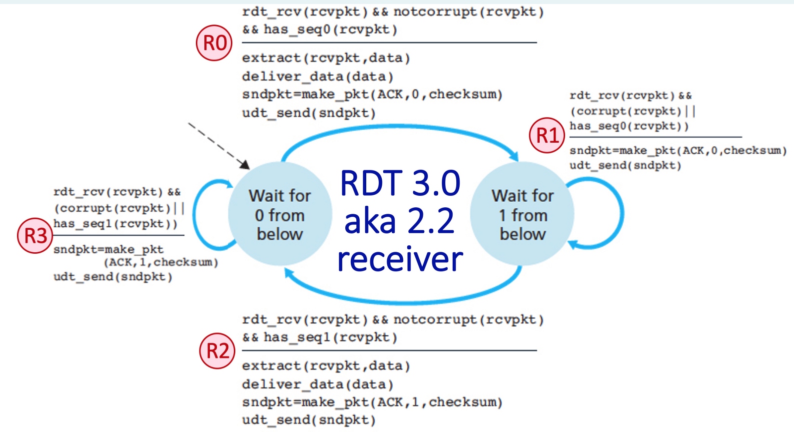
Question **15**

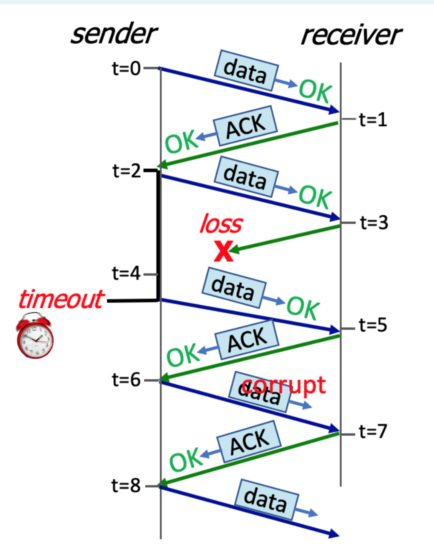
Incorrect

Mark 0.00 out of 1.50

Consider the rdt 3.0 sender and receiver below, with FSM transitions labeled in red. Also consider the sender and receiver timeline following the FSMs, where a green OK label indicates a message that is not corrupted, and a red corrupt label indicates a message that is corrupted. We are interested in the sequence number (0 or 1) of a data message, and the ACK number (0 or 1) of an ACK message for the messages sent at t= 2, 3, 4, 5 in the figure below.







Which of the following sequences of interleaved data sequence numbers and ACK numbers corresponds to those in the messages sent at t=2 (data sequence number); t=3 (ACK number), t=timeout, just after t=4 (data sequence number), and t=5 (ACK number), t=6 (data sequence number), t=7 (ACK number)? Only one sequence is correct.

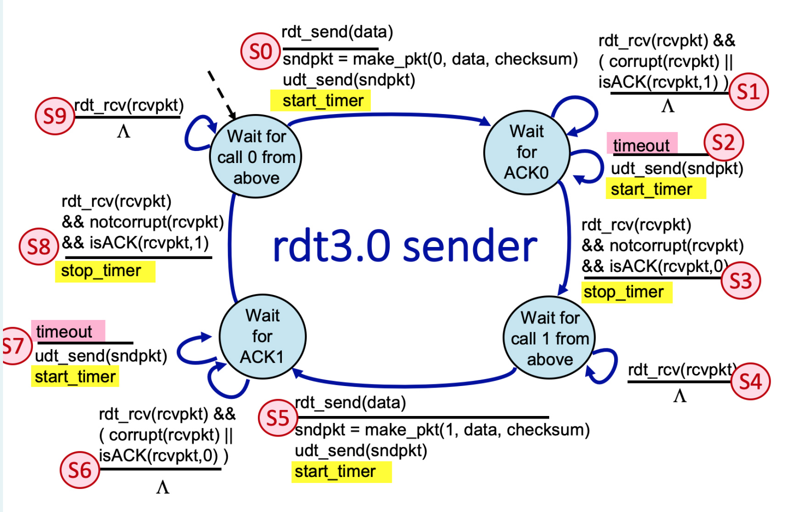
- a. 0, 0, 0, 0, 0, 1
- b. 1, 0, 1, 0, 0, 0
- c. 1, 0, 1, 0, 0, 1
- d. 0, 1, 1, 0, 1, 0
- e. 1, 0, 0, 1, 0, 1
- f. 1, 1, 1, 0, 0, 1
- g. 0, 1, 1, 0, 0, 1
- h. 1, 1, 0, 1, 0, 1
- i. 1, 0, 1, 0, 1, 0
- j. 1, 1, 1, 1, 0, 1
- k. 1, 0, 1, 1, 0, 1
- l. 0, 1, 1, 0, 0, 0

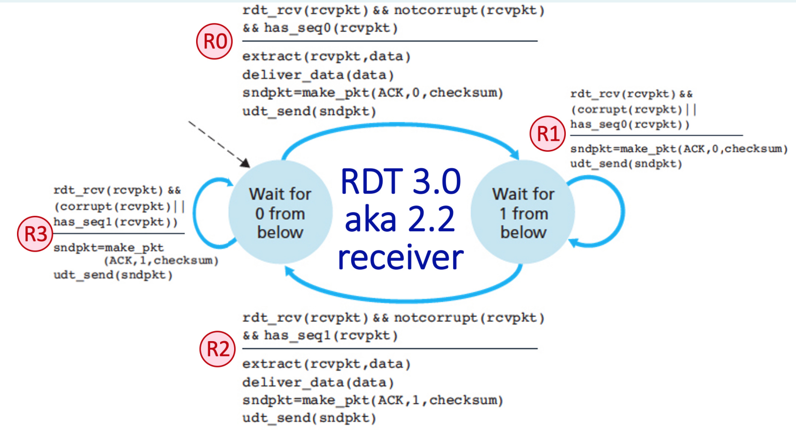
Question **16**

Correct

Mark 1.00 out of 1.00

Consider the rdt 3.0 sender and receiver below, with FSM transitions labeled in red. What are the interleaved sequence of sender and receiver transitions that occur when *(i)* the first message is lost, *(ii)* a timeout/retransmission then occurs, and *(iii)* the retransmitted packet is correctly received and an ACKnowledgement is then correctly sent?





- a. S0, R0, S2
- b. S0, S2, S3
- c. S0, S2, R0 ✔
- d. S0, R0, S2

Question **17**

Correct

Mark 0.50 out of 0.50

We now want to consider changing some of the channel assumptions underlying our design of RDT 3.0. In particular:

1. Assume that the *receiver-to-sender* channel is perfect - it will neither corrupt nor lose ACK messages.
2. Assume that the maximum delay between the sender and receiver is known (in both directions).
3. As in the case of RDT 3.0, we will continue to assume that sender-to-receiver messages may *still* be lost or corrupted.

We now want to consider what functionality in RDT 3.0 is no longer needed given these new assumptions (1 and 2 above). Are some form of ACKnowledgment messages from receiver to sender still needed?

- a. No
- b. Yes ✔

Information

For the next three questions consider the following:

Suppose that TCP's current estimated values for the round trip time (*estimatedRTT*) and deviation in the RTT (*DevRTT*) are 300 msec and 13 msec, respectively. Suppose that the next two measured RTTs are 330 msec and 240 msec respectively. We want to calculate TCP's RTT estimate, and the value of TCP's timeout interval. Note that given a new measured RTT, you should first compute *devRTT*, then *estimatedRTT* (the textbook incorrectly reverses those computations), and then (lastly) the timeout interval. Use the values of $\alpha = 0.125$, $\beta = 0.25$.

Question **18**

Correct

Mark 0.75 out of 0.75

Following the newly measured RTT of 330 msec, what is the new value for *devRTT* in msec? [Note: round your answer to the nearest msec - enter an integer value, do not include any decimal places/point or any leading zeros, but keep your full resolution value of *devRTT* handy and use it in any later calculations you perform using *devRTT*.

Answer: 17 ✔

Question **19**

Correct

Mark 0.75 out of 0.75

Following the newly measured RTT of 330 msec, what is the new value for *estimatedRTT* in msec? Note you should use the new value of *DevRTT* calculated in the previous question. [Note: round your answer to the nearest msec - enter an integer value, do not include any decimal places/point or any leading zeros, but keep your full resolution value of *estimatedRTT* handy and use it in any later calculations you perform using *estimatedRTT*. Remember also to use the new value of *devRTT* you calculated in the preceding question].

Answer: 304 ✔

Question **20**

Correct

Mark 0.75 out of 0.75

Following the newly measured RTT of 330 msec, what is the new value for TCP's timeout value in msec? Remember to use the new values for *devRTT* and *estimatedRTT* that you calculated in the previous questions and use the full 6-digit precision values you calclated (!). [Note: round your answer to this question, however, to the nearest msec - enter an integer value, do not include any decimal places/point or any leading zeros].

Answer: 373

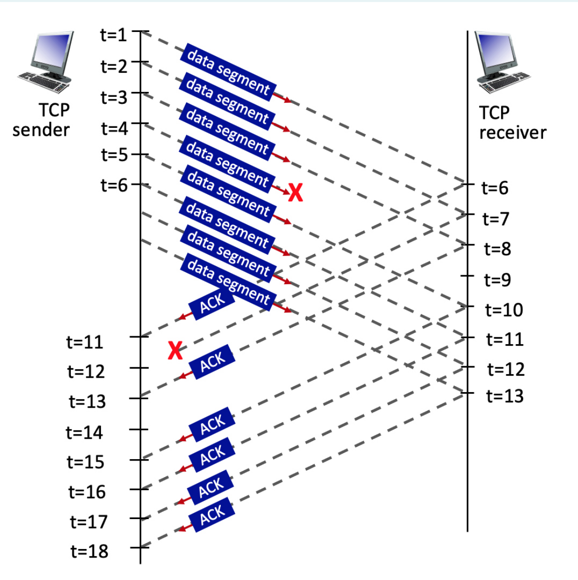
Question **21**

Correct

Mark 1.00 out of 1.00

Consider the figure below, where a TCP sender sends 8 TCP segments at *t = 1, 2, 3, 4, 5, 6, 7, 8.* Suppose the initial value of the sequence number is 0 and every segment sent to the receiver each contains 100 bytes. The delay between the sender and receiver is 5 time units, and so the first segment arrives at the receiver at t = 6. The ACKs sent by the receiver at *t = 6, 7, 8, 10, 11, 12* are shown. The TCP segments (if any) sent by the sender at t = 11, 13, 15, 16, 17, 18 are *not* shown.

The segment sent at t=4 is lost, as is the ACK segment sent at t=7.



What is the sequence number of the segment sent at t=2?

- a. 1
- b. 100
- c. 200
- d. 2

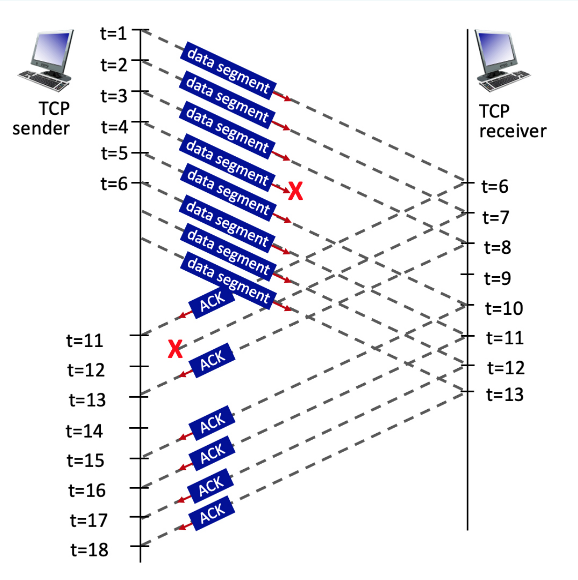
Question **22**

Incorrect

Mark 0.00 out of 1.00

Consider the figure below, where a TCP sender sends 8 TCP segments at *t = 1, 2, 3, 4, 5, 6, 7, 8.* Suppose the initial value of the sequence number is 0 and every segment sent to the receiver each contains 100 bytes. The delay between the sender and receiver is 5 time units, and so the first segment arrives at the receiver at t = 6. The ACKs sent by the receiver at *t = 6, 7, 8, 10, 11, 12* are shown. The TCP segments (if any) sent by the sender at t = 11, 13, 15, 16, 17, 18 are *not* shown.

The segment sent at t=4 is lost, as is the ACK segment sent at t=7.



What is the ACK value carried in the receiver-to-sender ACK sent at t = 10?

- a. None of these other answers.
- b. 100
- c. 400
- d. 3
- e. 200
- f. 300

Question **23**

Correct

Mark 1.00 out of 1.00

Which of the following actions are associated with end-end congestion control (say versus network-assisted congestion control).

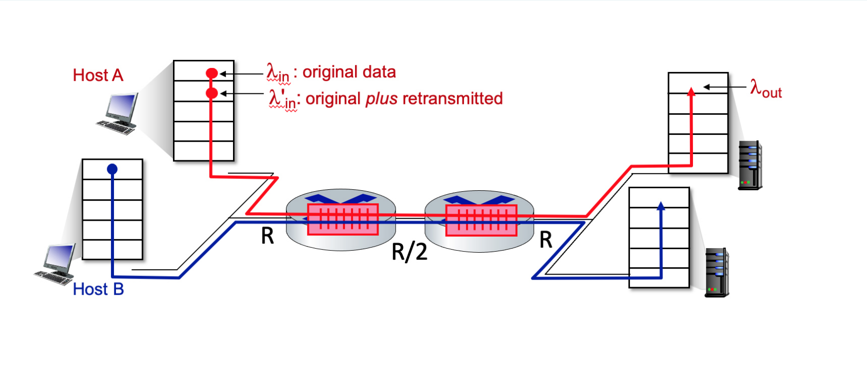
(Select all that apply, there **is** negative marking for incorrect choices!)

- a. The transport-layer receiver informs sender of the size of its (transport-payer receiver) receive window.
- b. A sender decreases its sending rate in response to packet loss detected via its transport-layer ACKing.
- c. A router marks a field in the datagram header at a congested router.
- d. The transport-layer sender decreases its sending rate in response to a measured increase in the RTT.
- e. A router drops a packet at a congested router, which causes the transport-layer sender to infer that there is congestion due to the missing ACK for the lost packet.
- f. A router sends an ICMP message to a host telling it to slow down its sending rate.
- g. A datagram experiences delay at a congested network router, which is then measured by the sender and used to decrease the sending rate.

Information

We learned about end-to-end and network-assisted congestion control. A third type of congestion control (mostly deployed only in internal data-center networks) is hop-by-hop congestion control. In this approach a downstream router/host explicitly informs the previous hop upstream router/host about the amount of free buffer space it has available, and the upstream router will only forward a packet when it knows the downstream router has buffer space.

Consider the three-hop scenario shown below, where the first and third links have capacity R, and the link between the routers has capacity R/2 for the next three questions.



Question **24**

Incorrect

Mark 0.00 out of 0.50

What is the maximum throughput of the red connection, given that the red and blue connections are similarly controlled, and therefore have the same maximum throughput?

- a. R/4
- b. R/2 ✖
- c. R
- d. 0

Question **25**

Incorrect

Mark 0.00 out of 0.50

With hop-by-hop congestion control, will the sender need to retransmit due to buffer overflow?

- a. No
- b. Yes ✖

Question **26**

Incorrect

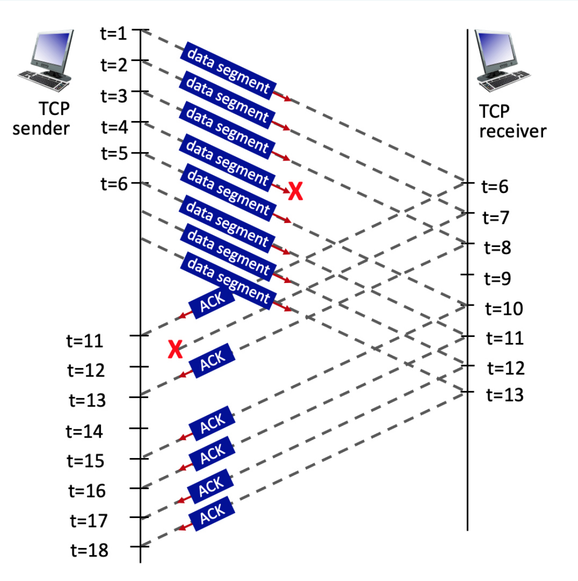
Mark 0.00 out of 0.50

With hop-by-hop congestion control and no congestion loss, might the sender still have to sometimes perform retransmission?

- a. Yes
- b. No

Information

Consider the below figure and setup for the next 4 questions, where a TCP sender sends 8 TCP segments at *t = 1, 2, 3, 4, 5, 6, 7, 8.* Suppose the initial value of the sequence number is 0 and every segment sent to the receiver each contains 100 bytes. The delay between the sender and receiver is 5 time units, and so the first segment arrives at the receiver at t = 6. The ACKs sent by the receiver at *t = 6, 7, 8, 10, 11, 12* are shown. The TCP segments (if any) sent by the sender at t = 11, 13, 15, 16, 17, 18 are *not* shown.



Question **27**

Correct

Mark 0.75 out of 0.75

The segment sent at t=4 is lost, as is the ACK segment sent at t=7. What is the sender action at t = 11 upon receipt of the ACK?

- a. Send an ACK to the ACK.
- b. Do nothing.
- c. Keep the congestion window size the same but send new segments, as available and as allowed by the congestion window.
- d. Increase the congestion window size, move the window base forward by 2, and send new segments, as available and as allowed by the congestion window
- e. Increase the congestion window size, move the window base forward by 1, and send new segments, as available and as allowed by the congestion window

Question **28**

Incorrect

Mark 0.00 out of 0.75

The segment sent at t=4 is lost, as is the ACK segment sent at t=7. What is the sender action at t = 13 upon receipt of the ACK?

- a. Increase the congestion window size, move the window base forward by 1, and send new segments, as available and as allowed by the congestion window
- b. Increase the congestion window size, move the window base forward by 2, and send new segments, as available and as allowed by the congestion window
- c. Send an ACK to the ACK.
- d. Keep the congestion window size the same. Increment the duplicate ACK count by 1. ✖
- e. Do nothing.

Question **29**

Incorrect

Mark 0.00 out of 0.75

The segment sent at t=4 is lost, as is the ACK segment sent at t=7. What does the sender do at t=16? You can assume for this question that no timeouts have occurred.

- a. Cut its value of `cwnd` in half, and retransmit the segment with sequence number 300
- b. Do nothing except increment the number of duplicate ACKs received by 1.
- c. Sets its `cwnd` window value to 1, and retransmit the segment with sequence number 300 ✖
- d. Inform the upper layer that the connection is terminated, and close the socket.

Question **30**

Incorrect

Mark 0.00 out of 0.75

The segment sent at t=4 is lost, as is the ACK segment sent at t=7. What does the sender do at t=17? You can assume for this question that no timeouts have occurred.

- a. Inform the upper layer that the connection is terminated, and close the socket.
- b. Sets its `cwnd` window value to 1, and retransmit the segment with sequence number 300
- c. Do nothing except increment the number of duplicate ACKs by 1. ✖
- d. Cut its value of `cwnd` in half, and retransmit the segment with sequence number 300

Question **31**

Incorrect

Mark 0.00 out of 0.75

The segment sent at t=4 is lost, as is the ACK segment sent at t=7. Suppose that the next event after t=17 is a timeout event. What does the sender do?

a. Do nothing.

b. Cut its cwnd window value in half, and retransmit the segment with sequence number 300

c. Sets it its cwnd window value to 1, a, and retransmit the segment with sequence number 100

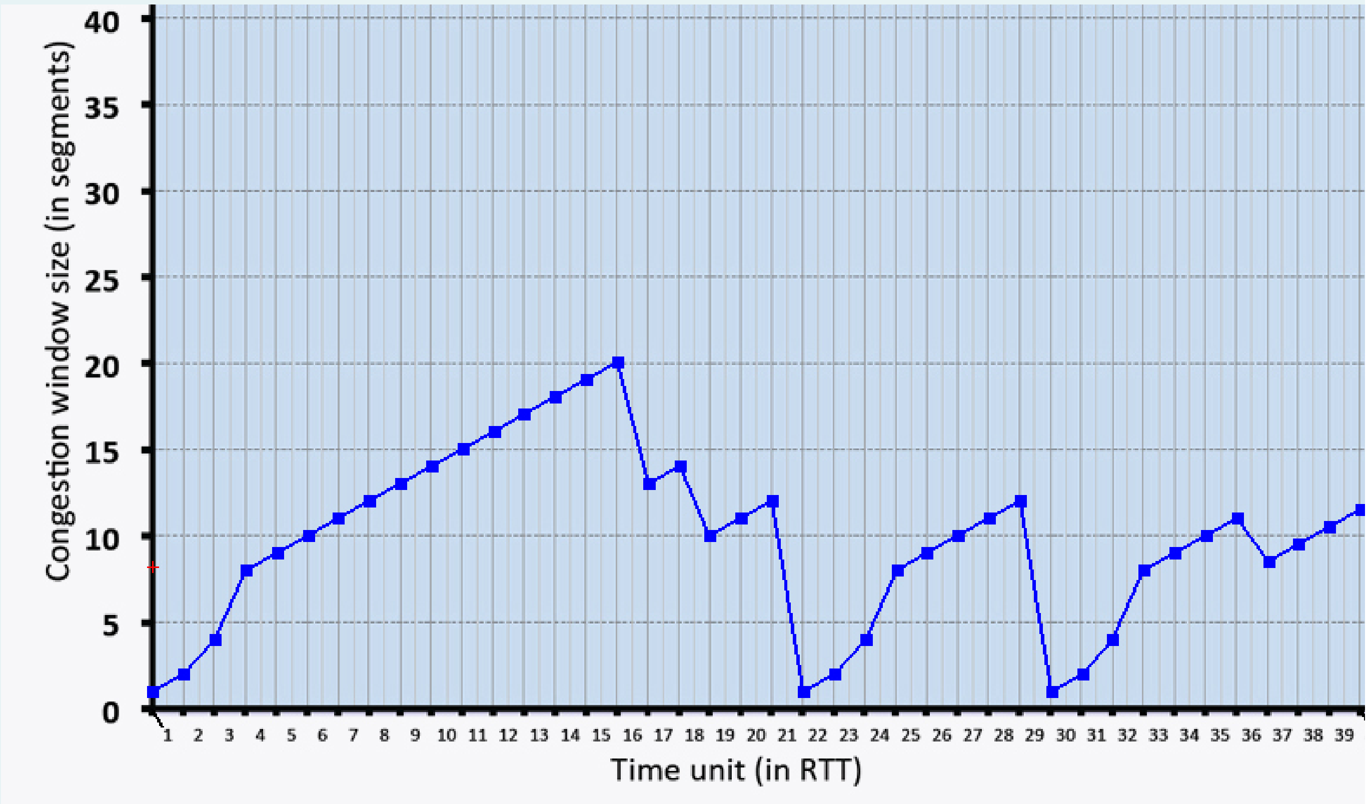
d. Sets it its cwnd window value to 1, and retransmit the segment with sequence number 300

e. Inform the upper layer that the connection is terminated, and close the socket.

Information

The following setup is for the next four questions

Consider the figure below, which plots the evolution of TCP's congestion window at the beginning of each time unit (where the unit of time is equal to the RTT). In the abstract model for this problem, TCP sends a "flight" of packets of size cwnd at the beginning of each time unit. The result of sending that flight of packets is that either (i) all packets are ACKed at the end of the time unit, (ii) there is a timeout for the first packet, or (iii) there is a triple duplicate ACK for the first packet.



Question **32**

Correct

Mark 0.75 out of 0.75

During which of the following intervals of time is TCP performing slow start?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [x] a. [1,3] ✔
- [ ] b. 18
- [ ] c. [19,20]
- [ ] d. 17
- [x] e. [22,24] ✔
- [ ] f. 21
- [ ] g. [4,15]
- [ ] h. 16

Question **33**

Partially correct

Mark 0.25 out of 0.75

During which of the following intervals of time is TCP performing congestion avoidance?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. 17
- [ ] b. 18
- [ ] c. 16
- [ ] d. [1,3]
- [ ] e. 21
- [x] f. [4,15] ✔
- [ ] g. [19,20]
- [ ] h. [22,24]

Question **34**

Correct

Mark 0.75 out of 0.75

At the end of which units of time does TCP detect a triple-duplicate-ACK?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. 21
- [ ] b. [4,15]
- [ ] c. [19,20]
- [ ] d. [22,24]
- [ ] e. [1,3]
- [x] f. 16 ✔
- [ ] g. 17
- [x] h. 18 ✔

Question **35**

Correct

Mark 0.75 out of 0.75

At the end of which unit(s) of time does TCP detect a loss via timeout?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. 17
- [ ] b. 16
- [ ] c. 18
- [ ] d. [1,3]
- [ ] e. [22,24]
- [ ] f. [4,15]
- [ ] g. [19,20]
- [x] h. 21 ✔

Question **36**

Correct

Mark 1.00 out of 1.00

For each of the actions below, select those actions below that are primarily in the network-layer data plane.

(Select all that apply, there **is** negative marking for incorrect choices!)

- [x] a. Dropping a datagram due to a congested (full) output buffer. ✔
- [ ] b. Monitoring and managing the configuration and performance of an network device.
- [ ] c. Computing the contents of the forwarding table.
- [x] d. Moving an arriving datagram from a router’s input port to output port ✔
- [x] e. Looking up address bits in an arriving datagram header in the forwarding table. ✔

Question **37**

Correct

Mark 1.00 out of 1.00

Which of the following quality-of-service guarantees are part of the Internet's best-effort service model?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. In-order datagram payload delivery to the transport layer of those datagrams arriving to the receiving host.
- [ ] b. A guaranteed minimum bandwidth is provided to a source-to-destination flow of packets
- [ ] c. Guaranteed delivery time from sending host to receiving host.
- [ ] d. Guaranteed delivery from sending host to receiving host.
- [x] e. *None* of the other services listed here are part of the best-effort service model. Evidently, best-effort service really means no *guarantees* at all! ✔

Question **38**

Correct

Mark 1.00 out of 1.00

Where in a router is the destination IP address looked up in a forwarding table to determine the appropriate output port to which the datagram should be directed?

- ○ a. Within the switching fabric.
- ○ b. Within the routing processor.
- ◉ c. At the input port where a packet arrives. ✔
- ○ d. At the output port leading to the next hop towards the destination.

Question **39**

Correct

Mark 0.50 out of 0.50

DHCP client and servers on the same subnet communicate via?

- ○ a. UDP connections
- ○ b. TCP connections
- ○ c. TCP broadcast
- ◉ d. UDP broadcast ✔

Question **40**

Correct

Mark 0.50 out of 0.50

IP assigned for a client by DHCP server is?

- a. For a limited period.
- b. Not time dependent,
- c. For an unlimited period.
- d. None of the above.
- e. A requested duration.

Information

For the next four questions use the following routing table and longest prefix matching.

Destination Address Range				Link interface
1010 ****	**** ****	**** ****	**** ****	0
1010 0000	1010****	**** ****	**** ****	1
1010 0000	1010 1111	**** ****	**** ****	2
**** ****	**** ****	**** ****	**** ****	3
1010 0000	1010 1111	1010 1111	**** ****	4

Question **41**

Correct

Mark 0.50 out of 0.50

What link interface would the following binary address map to?

1010 0000 10101111 10101111 10101111

Answer: 4

Question **42**

Correct

Mark 0.50 out of 0.50

What link interface would the following binary address map to?

1010 0000 00000000 1111 1111 0000 0000

Answer: 0

Question **43**

Correct

Mark 0.50 out of 0.50

What link interface would the following binary address map to?

1011 0000 00000000 1111 1111 0000 0000

Answer: 3

Question **44**

Correct

Mark 0.75 out of 0.75

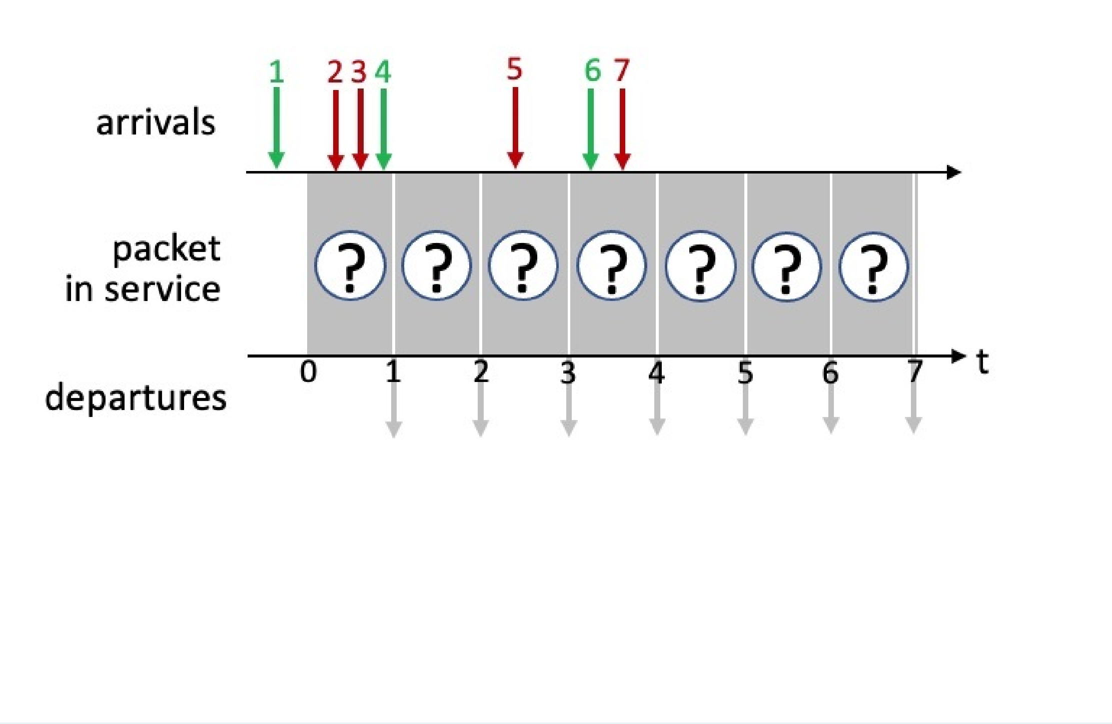
What link interface would the following IPV4 address map to?

160.160.5.5

Answer: 1

Information

Consider the pattern of red and green packet arrivals to a router's output port queue, shown below for the next for questions. Suppose each packet takes one time slot to be transmitted, and can only begin transmission at the beginning of a time slot after its arrival.



Question **45**

Correct

Mark 1.25 out of 1.25

Indicate the sequence of departing packet numbers (at t = 1, 2, 3, 4, 5, 6, 7) under **FCFS** scheduling.

Give your answer as 7 ordered digits (each corresponding to the packet number of a departing packet), with a single space between each digit, and no spaces before the first or after the last digit, e.g., in a form like 7 6 5 4 3 2 1).

Answer: 1 2 3 4 5 6 7 ✔

Question **46**

Correct

Mark 1.25 out of 1.25

Indicate the sequence of departing packet numbers (at t = 1, 2, 3, 4, 5, 6, 7) under **priority** scheduling (where red packets have priority over green packets).

Give your answer as 7 ordered digits (each corresponding to the packet number of a departing packet), with a single space between each digit, and no spaces before the first or after the last digit, e.g., in a form like 7 6 5 4 3 2 1).

Answer: 1 2 3 5 7 4 6 ✔

Question **47**

Correct

Mark 1.25 out of 1.25

Indicate the sequence of departing packet numbers (at t = 1, 2, 3, 4, 5, 6, 7) under **round robin** scheduling. Assume a round-robin scheduling cycle begins with green packets.

Give your answer as 7 ordered digits (each corresponding to the packet number of a departing packet), with a single space between each digit, and no spaces before the first or after the last digit, e.g., in a form like 7 6 5 4 3 2 1).

Answer: 1 2 4 3 6 5 7 ✔

Question **48**

Correct

Mark 0.50 out of 0.50

Suppose that the output port has enough buffer space for only three packets, including the packet being transmitted (in practice, routers have a *lot* more buffer space!). What happens to packet 4 in the example above, when it arrives to find the output buffers full?

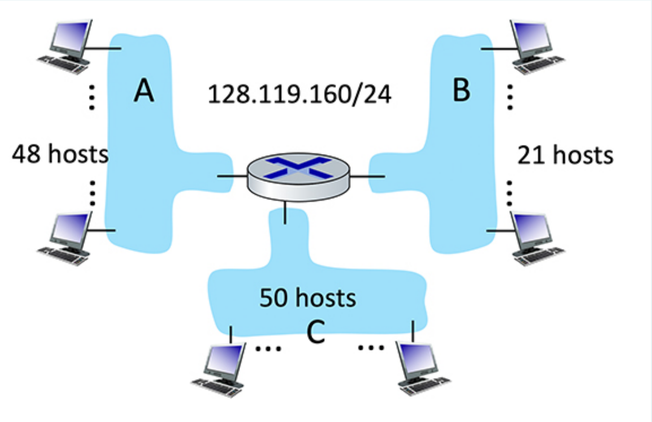
- a. Packet 4 is held in the switch fabric and released to the output port when there is free buffer space.
- b. Packet 4 is dropped or a queued packet is dropped from the buffer to make space for packet 4. ✔
- c. Packet 4 is recirculated back to input port and buffered there.

Question **49**

Partially correct

Mark 1.00 out of 3.00

Consider the three subnets below, each in the larger 128.119.160/24 network. The following questions are concerned with subnet addressing. Answer each question by selecting a matching answer. Each answer can be used to answer only *one* question.



What is the maximum number of hosts possible in the larger 128.119.160/24 network? 256 ✔

Suppose that subnet A has a CIDRized subnet address range of 128.119.160.128/26 (hint: 128 is 1000 0000 in binary); Subnet B has an CIDRied subnet address range of 128.119.160.64/26. We now want a valid CIDRized IP subnet address range for subnet C of the form 128.119.160.*x*/26. What is a valid value of *x*? 64 ✘

How many bits are needed to be able to address all of the host in subnet A? 96 ✘

Question **50**

Correct

Mark 1.00 out of 1.00

What are the principal components of the IPv4 protocol?

(Select all that apply, there **is** negative marking for incorrect choices!)

- [ ] a. Routing algorithms and protocols like OSPF and BGP.
- [ ] b. SDN controller protocols.
- [x] c. IPv4 addressing conventions. ✔
- [x] d. IPv4 datagram format. ✔
- [x] e. Packet handling conventions at routers (e.g., segmentation/reassembly) ✔
- [ ] f. ICMP (Internet Control Message Protocol)

◄ Lecture 20 Slides: Chapter 5.1-5.2.1

Jump to...

Large Test 2: Control ►