| Name | Student |
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# 2nd Examination

Introduction to Computer Networks (Online) Class#: EE 4020, Class-ID: 901E31110 Spring 2024

> 10:20-12:10 Wednesday May 1, 2024

### Cautions

- 1. There are in total 100 points to earn. You have 90 minutes to answer the questions. Skim through all questions and start from the questions you are more confident with.
- 2. Use only English to answer the questions. Misspelling and grammar errors will be tolerated, but you want to make sure with these errors your answers will still make sense.

 (ch26, 4pt) A viewer is watching a video on Netflix. Assume Netflix streams videos via DASH and the videos are H.264-encoded in a number of resolutions as follows.

| Resolution | Bitrate  |
|------------|----------|
| 2160p (4K) | 16 Mbps  |
| 1440p (2K) | 9.6 Mbps |
| 1080p      | 5 Mbps   |
| 720p       | 2.5 Mbps |
| 480p       | 1.2 Mbps |
| 360p       | 800 Kbps |

Suppose server A and B both hold a copy of the video. The available bandwidth from these servers to the viewer change over time. The table below shows the measured available bandwidth at time t1, t2, and t3.

|          | t1     | t2    | t3    |
|----------|--------|-------|-------|
| Server A | 8Mbps  | 6Mbps | 7Mbps |
| Server B | 20Mbps | 2Mbps | 2Mbps |

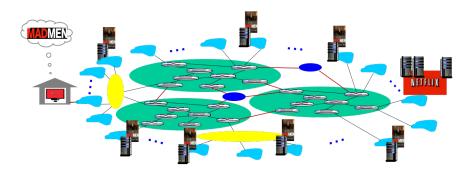
Available Bandwidth

- (1) Suppose Netflix assigns server A to the viewer. Tell the resolution the viewer is watching at t1. (1pt)
- (2) Suppose Netflix assigns the server with the higher available bandwidth to the viewer. Tell the resolution of the viewer is watching at t2. (1pt)
- (3) Suppose Netflix assigns the server with the higher average available bandwidth over t1-t3 to the viewer. Tell the resolution the viewer is watching at t3. (1pt)
- (4) Suppose Netflix assigns the server with the lower available bandwidth variation over t1-t3 to the viewer. Tell the resolution the viewer is watching at t3. (1pt)

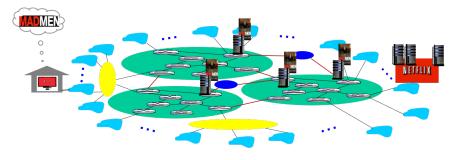
## Sample Solution:

(1080p, 1080p, 480p, 1080p)

 (ch26, 4pt) Suppose Netflix's videos are distributed by two content distribution network (CDN) providers. Depicted below are where CDN (a) and CDN (b) deploy their video servers. The original video server is on the right and the client on the left.









- (1) Which CDN is the "enter deep" type (1pt)?
- (2) Which CDN is the "bring home" type (1pt)?
- (3) Which CDN avoids network congestion better (1pt)?
- (4) Which CDN tends to cost less (1pt)?

(a, b, a, b)

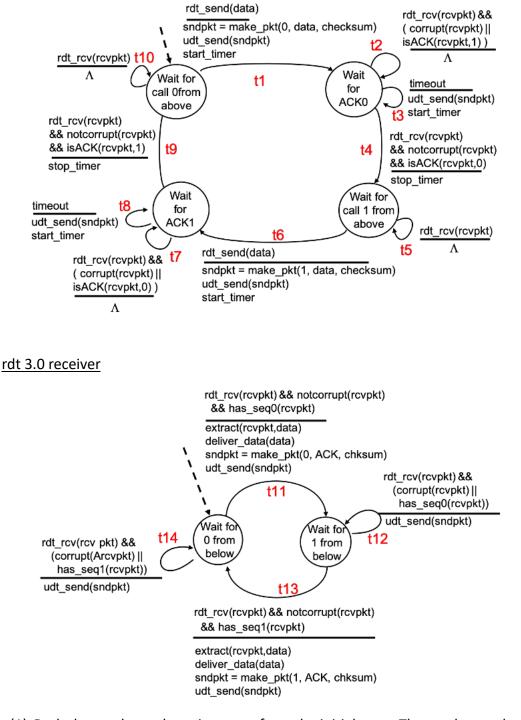
- 3. (ch3, 5pt) There are a lot more fields in the TCP packet header than UDP's. Tell which header(s) each of the following fields belongs to, (a) TCP, (b) UDP, (c) both, or (d) none.
  - (1) checksum (1pt)
  - (2) sequence number (1pt)
  - (3) port number (1pt)
  - (4) IP address (1pt)
  - (5) packet length (1pt)

### Sample Solution:

(c, a, c, d, b)

4. (ch34, 5pt) Below is the FSM of rdt 3.0. Tell the sequence of transitions for the following scenarios.

### rdt 3.0 sender



- Both the sender and receiver start from the initial state. The sender sends one packet. The packet is lost and no more bit errors or packet losses afterwards. (1pt)
- (2) Continue from (1). The sender sends one more packet. The packet arrives at the receiver but "ACK 0" returning to the sender has a bit error. There are no bit errors or packet losses afterwards. (1pt)
- (3) Continue from (2). The sender sends one more packet. There is a bit error in the "ACK 0" packet coming back and no more bit errors or packet losses afterwards. (1pt)

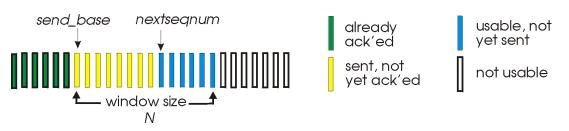
- (4) Continue from (3). The sender sends one more packet. The "ACK 0" packet coming back is lost and no more bit errors or packet losses afterwards. (1pt)
- (5) In rdt 3.0 sender FSM, t2 does not do anything when the ACK packet received is corrupted or a duplicate. Do you like this? (0pt) Why or why not? (1pt)

- (1) t1, t3, t11, t4
- (2) t6, t12, t7, t8, t13, t9
- (3) t1, t11, t2, t3, t12, t4
- (4) t6, t12, t8, t13, t9
- (5) take your pick and justify your answer
- 5. (ch34, 4pt) Between GBN and SR, which one would you use under the following conditions (0pt) and why (4pt).
  - (1) limited network bandwidth (1pt)
  - (2) limited memory space at the receiver (1pt)
  - (3) packet drops are always back-to-back (1pt)
  - (4) packet drops are sparse or random (1pt)

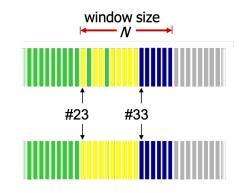
## Sample Solution:

Take your pick and justify your answer

6. (ch34, 4pt) Recall the color scheme of the packets at the sender side of GBN and SR.



Below are sending windows of two pipelined rdt connections. We know that one of them is a GBN connection and the other an SR connection. Tell the value of send\_base when an ACK packet arrives back at the sender.



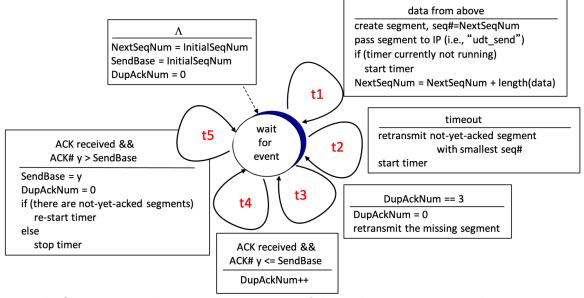
- (1) ACK #23 arriving back at the GBN sender
- (2) ACK #23 arriving back at the SR sender
- (3) ACK #26 arriving back at the GBN sender
- (4) ACK #26 arriving back at the SR sender

- (1) 24
- (2) 25
- (3) 27
- (4) 23
- 7. (ch35, 3pt) TCP's rdt is based on GBN. Tell which of the following GBN characteristics are no longer implemented in TCP. Grading policy: -1pt per error till 0 pt.
  - (a) pipelined transfer
  - (b) fixed window size
  - (c) cumulative ack
  - (d) one timer per flight of packets
  - (e) discarding out-of-order arrival
  - (f) restarting timer at duplicate acks

Sample Solution:

(b, e, f)

 (ch35, 8pt) Below is the FSM of the TCP sender. The mechanism to derive the retransmission timeout interval is missing. The timeout interval derivation starts by getting the SampleRTT. Let's try if we can find places in the FSM where we add the mechanism to obtain the SampleRTTs.



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A SampleRTT is calculated by taking the time difference between transmitting a data packet and receiving the corresponding ack. We need to be careful though. The SampleRTT of a data packet that has been retransmitted should be ignored because the time difference can potentially be longer than the actual RTT (including the retransmission timeout interval for example). Grading policy: -1pt per error till 0 pt.

- (1) Which transition(s) to record the time transmitting a new data packet? (1pt)
- (2) Why is it necessary to record the transmission time of a new data packet? (1pt)
- (3) Which transition(s) to record whether a data packet has been retransmitted? (1pt)
- (4) Why is it necessary to record whether a data packet has been retransmitted? (1pt)
- (5) Which transition(s) to record the time receiving a new ACK packet? (1pt)
- (6) Why is it necessary to record the time receiving a new ACK packet? (1pt)
- (7) Which transition(s) to discard the SampleRTT that has been calculated? (1pt)

(8) Why is it necessary to discard the SampleRTT that has been calculated? (1pt) Sample Solution:

(1) t1

- (2) need to record the time sending a data packet the first time.
- (3) t2, t3 (t2, t3, t4 or just t4)
- (4) Both transitions retransmit data packets. need to record the data packets that have been retransmitted. Therefore, when ACKs are received, the sender can check whether the SampleRTT calculated need to be removed.

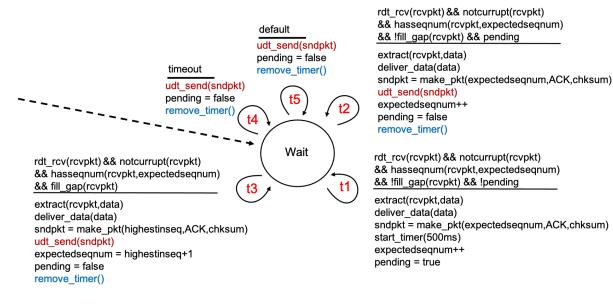
One might select t4 in addition. Note that a dup ACK is for the receiver to tell that there is likely a packet lost. Think variation of delay on the Internet. The data packet might be retransmitted, but not a sure thing yet. Noting/discarding the SampleRTTs when we know for sure the data packets are retransmitted will be exact and robust.

- (5) t5
- (6) need to take the timestamp when the first ack of a data packet is received, so one can calculate the sampleRTT. Be careful though. One wants to check whether the packet has been retransmitted as well. If so, discard the SampleRTT.
- (7) t4, t5 (or just t4)

Just t4 is OK as one can check at t5 if a packet is a retransmission before calculating the SampleRTT.

- (8) t4. The SampleRTT of the dup ACK has been calculated. One checks whether the packet has been retransmitted. If so, discard the SampleRTT. This is for the cases when unnecessary retransmissions occur.
  - t5. See the explanation in (6)

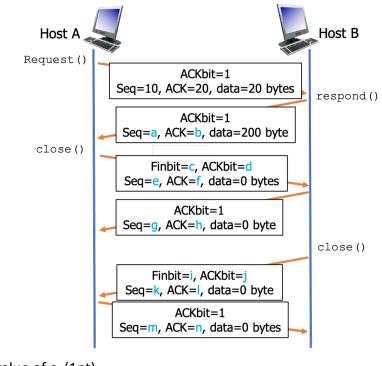
### 9. (ch35, 4pt) Below is the FSM of the TCP receiver.



- (1) Which transition(s) is delaying ACKs (0pt) and why (1pt)?
- (2) Which transition(s) sends duplicate ACKs (0pt) and why (1pt)?
- (3) Which transition(s) sends new ACKs (0pt) and why (2pt)?

### Sample Solution:

- (1) t1, the 500ms timer is started in the transition
- (2) t5, packet arriving out of order will trigger t5
- (3) t2, t3, t4,
  - t2 sends the 2<sup>nd</sup> ack of a pair
  - t3 sends a new ack after filling a gap
  - t4 send a new ack after the 500ms timer times out.
- 10. (ch35, 8pt) Depicted below is a part of packet exchange in a TCP connection between Host A and B. Host A sends a request. Host B sends back a response. After receiving B's response, A initiates closing of the connection by sending a FIN packet. No long after, B closes the connection by sending a FIN packet as well.



- (1) Tell the value of c. (1pt)
- (2) Tell the value of d. (1pt)
- (3) Tell the value of g. (1pt)
- (4) Tell the value of h. (1pt)
- (5) Tell the value of i. (1pt)
- (6) Tell the value of j. (1pt)
- (7) Tell the value of m. (1pt)
- (8) Tell the value of n. (1pt)

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(c, d, g, h, i, j, m, n) = (1, 1, 220, 31, 1, 0, 31, 221)
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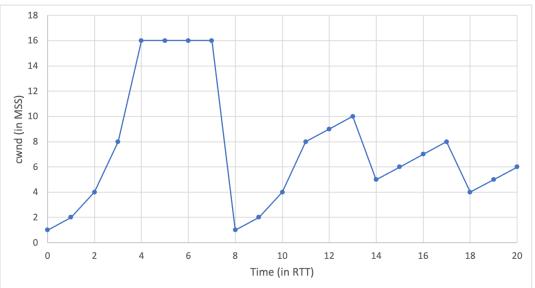
- 11. (ch36, 6pt) There are two general approaches to congestion control end to end and network assisted.
  - (1) Describe how the end-to-end approach works. (1pt)
  - (2) Describe how the network-assisted approach works. (1pt)
  - (3) Tell two advantages of taking the end-to-end approach. (2pt)
  - (4) Tell two advantages of taking the network-assisted approach. (2pt)

## Sample Solution:

- (1) Having the end hosts to detect packet loss or delay and inform the traffic source to set the sending rate accordingly
- (2) Having routers to track the free buffer space and inform the traffic source to set the sending rate accordingly
- (3) Ease of deployment

Ease of making changes (complexity at the edge) Lower router load

- (4) Precise estimation of available bandwidthQuick detection of congestion (right at the router where the congestion happens)Lower end-system load
- 12. (ch37, 7pt) Drawn in the plot below is the progress of cwnd over time in a TCP connection.

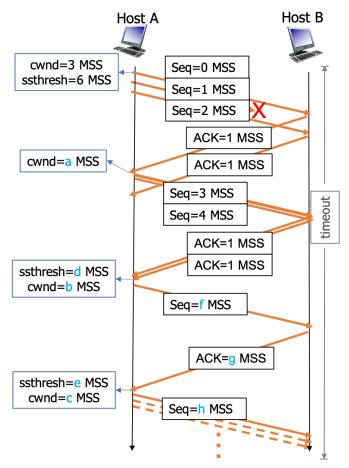


- (1) What is the state of the TCP connection during Time = 0-4? (1pt)
- (2) What is the event that triggers the cwnd reduction at Time = 8? (1pt)
- (3) What is the state of the TCP connection during Time = 8-11? (1pt)
- (4) What is the state of the TCP connection during Time = 11-13? (1pt)
- (5) What is the event that triggers the cwnd reduction at Time = 14? (1pt)
- (6) What is the ssthresh at Time = 14? (1pt)
- (7) Is this a TCP connection with Fast Recovery (0pt) and why (1pt)?

Sample Solution:

- (1) slow start state
- (2) retransmission timeout
- (3) slow start state
- (4) congestion avoidance state
- (5) 3 duplicate ACKs
- (6) 5 MSS
- (7) TCP without fast recovery
- 13. (ch37, 8pt) The figure below illustrates a part of the data and ACK packet exchange in a TCP connection. Assume that (1) there are always data from above, (2) the packets are

always MSS bytes large. If there remains a partial MSS in the cwnd, the sender does not generate a smaller packet to send. That is, if the cwnd = 2.5 MSS, the sender sends only 2 packets or allows only 2 packets in flight.



- (1) Tell the value of a. (1pt)
- (2) Tell the value of b without fast recovery. (1pt)
- (3) Tell the value of c without fast recovery. (1pt)
- (4) Tell the value of b with fast recovery. (1pt)
- (5) Tell the value of c with fast recovery. (1pt)
- (6) Tell the value of f. (1pt)
- (7) Tell the value of g. (1pt)
- (8) Tell the value of h. (1pt)

### Sample Solution:

(a, b, c, b, c f, g, h)=(4, 2, 2.5, 5, 2, 1, 5, 5)

- 14. (PA, 14pt) Please go on the PA workstation and work under the exam2 directory for this problem set. Create the exam2 directory if you have not yet done so. Grading policy: pts for later problems will be given only when the former ones are completed.
  - (1) Develop exam2-p14-1.go such that it connects to the server running on port 12000 and then closes the connection. (2pt)
  - (2) Develop exam2-p14-2.go such that it connects to the server running on port 12000, sends "PLAY\n" and then closes the connection. (2pt)
  - (3) Develop exam2-p14-3.go such that it connects to the server running on port 12000, sends "PLAY\n", receives a line of message from the server, prints the message on the screen, and then closes the connection. (3pt)

You should see the response from the server on port 12000. It says it is the "Hangman Not Quite" game engine. In this simplified Hangman game, a player is asked to figure out an English word the game engine has in mind. Given a character, the game engine responses with a string showing where the character appears in the word.

- (4) Develop exam2-p14-4.go such that it connects to the server running on port 12000, sends "PLAY\n", receives a line of message from the server, prints the message on the screen. Then, prompts the user for a guess, sends the character in one line (+'\n' at the end of the character), and then closes the connection. (3pt)
- (5) Develop exam2-p14-5.go such that it connects to the server running on port 12000, sends "PLAY\n", receives a line of message from the server, prints the message on the screen. Then, prompts the user for a guess, sends the character in one line (+'\n' at the end of the character), receives a line of message from the server, prints the line of message on the screen, and then closes the connection. (3pt)
- (6) Execute exam2-p14-5.go multiple times. Guess a different character each time until you figure out the word the game engine has in mind. Tell what the word is. (1pt) Sample Solution:

Whatever that works.

- 15. (PA, 16pt) Please go on the PA workstation and work under the exam2 directory for this problem set. Create the exam2 directory if you have not yet done so. Grading policy: pts for later problems will be given only when the former ones are completed.
  - (1) Develop exam2-p15-1.go running on your exam port # such that it allows multiple clients such as your exam2-p14-1.go to connect. (2pt)
  - (2) Develop exam2-p15-2.go such that it allows multiple clients such as your exam2-p14-2.go to connect. It allows the client to send a message such as "PLAY\n" and prints the message on screen. (3pt)
  - (3) Develop exam2-p15-3.go such that it allows multiple clients such as your exam2-p14-3.go to connect, and allows the client o send a message such as "PLAY\n". After receiving "PLAY\n", it sends "Welcome to Hangman Not Quite! I have an English word for you to figure out. Give me a character. I will show where the character appears in the word.\n" back to the client. (4pt)
  - (4) Develop exam2-p15-4.go such that it allows multiple clients such as your exam2-p14-4.go to connect, and allows the client to send "PLAY\n". After receiving "PLAY\n", it sends "Welcome to Hangman Not Quite! I have an English word for you to figure out. Give me a character. I will show where the character appears in the word.\n" back to the client. Afterwards, allow the client to send a character followed by "\n" and prints the character on screen. (3pt)
  - (5) Develop exam2-p15-5.go such that it allows multiple clients such as your exam2-p14-5.go to connect, and allows the client to send "PLAY\n". After receiving "PLAY\n", it sends "Welcome to Hangman Not Quite! I have an English word for you to figure out. Give me a character. I will show where the character appears in the word.\n" back to the client. Afterwards, allow the client to send a character followed by "\n". After receiving the character, send a response showing the character in the word according to your answer for Problem 14-6. (4pt)

Whatever that works.