✖ Exit

CS2105 AY19-20 Final Assessment

Objectives & Instructions:

By clicking the "Start The Quiz" button to take the exam, the student agrees to adhere to the NUS code of student conduct especially the following three paragraphs related to Academic, Professional, and Personal Integrity as follows.

3. The University is committed to nurturing an environment conducive for the exchange of ideas, advancement of knowledge and intellectual development. Academic honesty and integrity are essential conditions for the pursuit and acquisition of knowledge, and the University expects each student to maintain and uphold the highest standards of integrity and academic honesty at all times.

4. The University takes a strict view of cheating in any form, deceptive fabrication, plagiarism and violation of intellectual property and copyright laws. Any student who is found to have engaged in such misconduct will be subject to disciplinary action by the University.

5. It is important to note that all students share the responsibility of protecting the academic standards and reputation of the University. This responsibility can extend beyond each student's own conduct, and can include reporting incidents of suspected academic dishonesty through the appropriate channels. Students who have reasonable grounds to suspect academic dishonesty should raise their concerns directly to the relevant Head of Department, Dean of Faculty, Registrar, Vice Provost or Provost.

This quiz is worth **55 marks.**

You only have **1 attempt.**

You have **2 hours 0 minute** to complete this quiz - **the timer will continue to run if and when you choose to leave the page.**

Start The Quiz

Multiple Choice Questions

In this section, each question will have 5 choices and only 1 of them is correct.

You will get full marks if you choose the correct answer; otherwise, you get zero mark. The order of the 5 choices will be randomized.

1. In a stop-and-wait protocol, if m bits are used to represent the sequence number, the maximum send window size is ___ and the maximum receive window size is ___. (1 mark)

   4
   $2^3 2^2 2^1 2^0$
   $0 0 0 0$

   - $1; 2^m-1$
   - ✓ $2^m-1; 2^m-1$
   - $2^m; 2^m$
   - $1; 1$
   - $2^m-1; 1$

2. Consider the scenario where a sender and a receiver use the reliable stop-and-wait protocol rdt3.0 to communicate.

   Suppose the sender has received an acknowledgment ACK0 and sent a packet PKT1. What will the rdt3.0 sender do when it receives a duplicate ACK0?

   Suppose the receiver has received a packet PKT0 and sent ACK0 to acknowledge the reception of PKT0. What will the rdt3.0 receiver do when it receives a duplicate data packet PKT0?

   (1 mark)

   - The sender should do nothing and the receiver should send the acknowledgement ACK1.
   - The sender should do nothing and the receiver should resend the acknowledgement ACK0.
   - The sender should resend the packet PKT0 and the receiver should resend the acknowledgement ACK0.
   - The sender should resend the packet PKT1 and the receiver should send the acknowledgement ACK1.
   - The sender should resend the packet PKT1 and the receiver should do nothing.

3. A port number in TCP is ___ bytes long. (1 mark)

   bits.

   - ✓ 2
   - 16
   - 32
   - 8
   - 4

4. TCP is a type of _____ protocol.

   i) Go-back-N

i) Go back N

ii) Selective Repeat ✓   (1 mark)

iii) Pipelined ✓

iv) Reliable ✓

○ ii), iii) and iv) only.

○ i), iii) and iv) only.

○ i), ii) and iii) only.

○ iii) and iv) only.

◉ ✓ i), ii), iii) and iv). ✓

5. The behavior of a TCP sender can be influenced by a feedback packet received from a TCP receiver. Which of the following statement is FALSE? (1 mark)

○ It is possible that a TCP receiver's feedback packet triggers the TCP sender to transmit a new data packet. ? ✓

○ It is possible that a TCP receiver's feedback packet does not trigger any action taken by the TCP sender. ✓

○ It is possible that a TCP receiver's feedback packet triggers the TCP sender to transmit multiple new data packets. ? ✓

○ It is possible that a TCP receiver's feedback packet triggers the TCP sender to re-transmit an old data packet. ✓

◉ ✓ It is possible that a TCP receiver's feedback packet triggers the TCP sender to re-transmit multiple old data packets. ?

6. Assume that a computer on a LAN has a very small ARP table that can store only 10 IP-to-MAC mappings (because of a small memory size). However, the computer wants to communicate with 15 other computers on the same LAN.

Which of the following statement is TRUE?

(1 mark)

○ This will simply not work. ?.

◉ ✓ This will work because ARP can discover MAC addresses that are not in the ARP table, but it will be inefficient since some of the oldest mappings in the ARP table will constantly be replaced with the newest ones.

○ If the LAN uses a star-topology with a switch, then the ARP table size does not matter. maybe...

✗ In some cases a MAC address is not required to communicate with another computer.

✗ When the sender table is full, the sending computer has to wait until one of the TTLs expires and only then can it communicate with a new computer.

7. Suppose we want to design a stop-and-wait, reliable protocol for communication between a sender and a receiver over a channel with the following characteristics: data packets may be lost or corrupted, but will not be reordered. Feedback packets are always going to be received uncorrupted and in good order. Furthermore, the maximum RTT between the sender and the receiver is known.

Which of the following statement about the reliable protocol is TRUE?

(1 mark)

○ None of the other options is TRUE.

None of the other options is TRUE.

○ Sender must attach a sequence number to every data packet. ✗

○ In a feedback packet, the receiver must explicitly include the sequence number of the data packet being acknowledged. ✗

○ The receiver should discard any corrupted data packet but must acknowledge the sender about it. ✗ timer

✓ If the sender sets the timer properly, the receiver will not receive duplicate packets for sure.

8.

Which of the following technique/mechanism is essential in the design of reliable transmission protocols? (1 mark)

○ Encapsulation

✓ Error detection

○ Addressing

○ Layering

○ Multiplexing

```
128  64  32  16  8  4  2  1
 0    1   1   0  1  1  0  0      x  x  x  x  x x x x
                        13
                     8  8   3
```

9.

An IP address block 192.168.108/19 can be further divided into X subnets, each supporting a maximum of Y hosts. Which of the following is NOT a valid assignment? (1 mark)

○ X = 64; Y = 125.
   6 2^7

○ X = 128; Y = 60.
   7 2^6

✓ X = 32; Y = 260.
   5 2^8

○ X = 4; Y = 2000.
   2 2^{11}

○ X = 256; Y = 30.
   8 2^5

Multiple Response Questions

In this section, each question will also have 5 choices, but the number of correct answers may range from 1 to 5. The order of the 5 choices will also be randomized. Your final score will be calculated by the following formula:

Full Marks * (Number of Selected Correct Answers - Number of Selected Incorrect Answers) / Total Number of Correct Answers

10.

Many homes have a router with NAT capabilities that facilitate connecting many home devices to the Internet. Which of the following applications would NOT function properly (without special/advanced/not taught NAT router configurations) from a home computer/laptop connected to a NAT router?

(2.5 marks)

☑ Web server (e.g., Apache, Microsoft IIS).

✓ Messaging client (e.g., WhatsApp, WeChat).

☐ Networked printer (e.g., the printer is connected to WiFi and functions as print server).

✓ Email server (e.g., Postfix, Microsoft Exchange Server)

✓ Email client (e.g., Outlook, macOS Mail).

11. Some professional studio audio equipment is sampling audio at 48,000 samples per second. What is the data rate of such an audio stream, assuming a 16-bit representation (quantization) for each sample, and 2-channel (stereo) sound, and no compression?

→ 768000bps ·x2.
  = 1536000.

(2.5 marks)

☐ 92,000 bytes per second

☐ 88,200 bytes per second

☐ 192,000 bytes per second

✓ 1,536,000 bits per second

☐ 1,411,200 bits per second

12. Which of the following statements are TRUE for random access link layer protocols? (2.5 marks)

✓ Slotted ALOHA allows a higher throughput (bits per second over a certain, reasonably long time window) than Pure ALOHA.

☐ The hidden node problem can happen in every LAN.

✓ Acknowledgements (ACK) are never used in random access link layer protocols.

✓ CSMA/CD allows a higher throughput (bits per second over a certain, reasonably long time window) than just CSMA.

☐ Frames require a maximum size in order to detect collisions.

min.

128 64 32 16 8 4 2 1
1 0 1 0 0 0 0 0

13. Which of the following IP addresses belong to the subnet 192.168.160.0/20? (2.5 marks)

✓ 192.168.175.254

✓ 192.168.180.25

✓ 192.168.170.1

✗ 192.168.155.202

✓ 192.168.165.22

14. Which of the following properties are TRUE for the Slotted ALOHA link layer protocol? (2.5 marks)

☑ If all the nodes have a lot of data to send then still we can only utilize the channel capacity much less than 100% (approximately 37%).

☑ Every available transmission slot will be utilized as long as at least one sender has data (frames) to send.

☑ A time synchronization is required between the sender and the receiver. ?

☑ Every packet will eventually be transmitted.

☐ On average, some nodes have a higher probability to transmit than others. ✗

15.

At the link layer assume that a data frame is protected with two-dimensional bit parity. If a three-bit error occurs during the transmission of a frame, what is the result for the frame at the receiver?

Select the statements that are TRUE.

(2.5 marks)

☐ The problem is always detected and can be corrected.

☐ The problem may **not** be detected and the data is corrupted.

☐ The problem may be detected and the data is corrupted.

☐ The problem is always **not** detected.

☑ The problem is always detected.

16.

Assume we have a laptop connected to a home NAT router. The laptop is sending datagrams to an external web site. Assume further that w, x, y, z are port numbers and e, f, g, h represent full IP addresses (e.g., e = a.b.c.d). S: indicates the source, and D: the destination in an IP datagram. Each triplet [_: _,_] indicates a source or destination IP,port pair. Which of the following packet flows is correct? (If one of the variables is used twice, that means it represents the same IP or port number. Assume e is a private IP address; the others are public.)

(2.5 marks)

g, y.

e, w   f, z   f, z.

☑ Laptop to NAT router: [S: e, w], [D: f, z]
NAT router to web site: [S: g, y], [D: f, z]
Web site to NAT router: [S: f, z], [D: g, y]
NAT router to laptop: [S: f, z], [D: e, w]

✗ Laptop to NAT router: [S: e, w], [D: f, y]
NAT router to web site: [S: e, z], [D: f, y]
web site to NAT router: [S: f, y], [D: e, z]
NAT router to laptop: [S: f, y], [D: e, w]   same IP addr?

✓ Laptop to NAT router: [S: e, w], [D: f, y]
NAT router to web site: [S: g, z], [D: f, y]
web site to NAT router: [S: f, y], [D: g, z]
NAT router to laptop: [S: f, y], [D: e, w]

✗ Laptop to NAT router: [S: e, w], [D: f, x]
NAT router to web site: [S: g, z], [D: f, y]
web site to NAT router: [S: f, x], [D: g, z]
NAT router to laptop: [S: f, y], [D: e, w]

✗ Laptop to NAT router: [S: e, w], [D: f, y]
NAT router to web site: [S: g, z], [D: h, w]
Web site to NAT router: [S: h, w], [D: g, z]
NAT router to laptop: [S: h, w], [D: e, w]

$2^3 = 8$

17. Consider a sender and a receiver communicating using Selective Repeat protocol. Every packet embeds a 3-bit sequence number field. The sender has just sent a packet with sequence number 0. Sender's window size is 3.

Which of the following CANNOT possibly be the sequence number of the next packet transmitted by the sender?

(2.5 marks)

0 1 2 3 4 5 6 7

- [x] 4
- [ ] 6
- [ ] 7
- [x] 5
- [ ] 3

18. A sender S is sending packets to a receiver R using the Go-Back-N protocol. The sender's window is of size 4. After transmitting for a while, the first packet in the sender's window is p_k (assume k>1). Let a packet with sequence number i be p_i. Assume that packets may be lost or corrupted but won't be reordered.

Which of the following statements are TRUE?

(2.5 marks)

$P_k \ P_{k+1} \ P_{k+2} \ P_{k+3}$

- [x] It is possible that R has sent ACK for p_{k+3} already.
- [x] R might have received p_k already.
- [x] R must have received p_{k-1} already.
- [x] S must have received ACK for p_{k-1} already.
- [x] S must not have received ACK for p_{k+1}.

19. We have seen in Lecture 08 that there are two common channel partitioning protocols: TDMA and FDMA.

(2.5 marks)

Which of the following statements are TRUE?

- [ ] The average capacity (i.e., the total number of bits per second) of a channel utilizing TDMA is much higher than when using FDMA with the same channel when considered over a long time period. *mean throughput is R/N.* ?
- [ ] All senders and receivers require some kind of time synchronization for both TDMA and FDMA. *X no need only for TDMA.*
- [x] FDMA on average results in lower latency for short data transmissions. *? no need to wait.*
- [x] Consider a time slot in TDMA. The peak transmission capacity during a slot is higher for TDMA than it is for FDMA at any time. ?

FDMA peak is constant ∴ slower.

20. Assume there are two routers in an AS (autonomous systems) network, routers A and B. Assume the routers are using the Distance Vector (DV) algorithm and implement the RIP protocol for packet routing. Assume further that A and B are connected through 2 links: link X has 2 hops and link Y has 3 hops. Assume that 100 data packets arrive at A and need to be forwarded to B.

Which of the following statements are TRUE?

(2.5 marks)

- [ ] 50 packets are sent over X and 50 packets are sent over Y

☐ 50 packets are sent over X and 50 packets are sent over Y.

☑ ✗ 60 packets are sent over X and 40 packets are sent over Y.

☐ 40 packets are sent over X and 60 are sent over Y.

☑ If link X fails, then after several rounds of distance vector updates link Y is found as the best link and all 100 packets are transmitted over Y (let's assume the packets are not dropped in A while they wait).

☑ 100 packets are sent over X and 0 packets are sent over Y.

21.

Assume you are designing a VoIP system. Each data frame is 160 bytes, representing 20 msec of voice data. You know that your transmission channel may have some losses and you try to decide to either use 1-dimensional parity or 2-dimensional parity.

Which of the following statement are TRUE (and can help you to make your design decision)?

(2.5 marks)

☑ The parity data overhead (i.e., the parity data size as a fraction of the useful data size) is less with 1-dimensional parity.

☑ 3-dimensional parity would allow even more cases of data corruption to be detected, but it does not exist (i.e., there is no such thing as 3-dimensional parity).

☑ 2-dimensional parity will increase the latency (i.e., time delay) introduced at the sender side (until the data is sent into the network). ?

☑ 2-dimensional parity allows more cases of data corruption to be detected.

☑ 2-dimensional parity requires more computation than 1-dimensional parity at both the sender and the receiver side. not really.

Fill in the Blank

In this section, you are typically required to calculate some numerical values and put them in the blanks as your final answers.

If your answer matches the predefined correct answer value, you will get full marks.

For each question, a Response Rationale box is provided for you to input more derivation details. Although filling this box is optional, the contents of this box will be used by graders to possibly provide partial marks through manually evaluation, if we identify that you main logic of solving the question is correct, but the final answer is wrong due to careless calculation mistakes.

22.

Fill in the blanks (2 marks)

Suppose an intial value of EstimatedRTT is 100ms. Subsequently, the following four RTT samples are measured: 60ms, 55ms, 10ms and 200ms. After taking the four recent measurements, the most updated value of EstimatedRTT used by the TCP protocol is ___1___ ms.

Enter the correct answer below.

1 [95.] Please enter a number for this text box.

📩 **Response Rationale**
   Please provide a rationale for your answer.

Font ▾	Size ▾	▾	▾						▾	Source

$$1-\alpha\,(RTT_{est}) + \alpha\,(RTT_{sample}) = RTT_{est}.$$

23.

Fill in the blanks (2 marks)

Suppose an online service provider runs a UDP server and a TCP server to provide an online service to its users. If 100 flows are connected to the servers and at the server-side, a total of 50 sockets are created, __1__ of the 100 flows are TCP flows. If 20 of the 100 flows are TCP flows and 80 of the 100 flows are UDP flows, a total of __2__ sockets need to be created at the server-side.

Enter the correct answer below.

1 [48] Please enter a number for this text box.

2 [22] Please enter a number for this text box.

Response Rationale

Please provide a rationale for your answer.

| Font | Size | | | | | | | Source |

UDP socket = 1
TCP welcome socket = 1

24.

Fill in the blanks (2 marks)

The following diagram shows two hosts **X** and **Y** communicating over an ongoing TCP connection. **X** and **Y** are sending data to each other and each TCP segment contains 50 bytes of application data. The first segment send by **X** has sequence number 30 and ACK number 70. The second segment send by **X** is lost and retransmitted when the associated timer expires. Assume that no other the segments in the diagram are retransmitted packets, none of the segments are corrupted and receiver buffers out-of-order packets for eventual delivery to application.
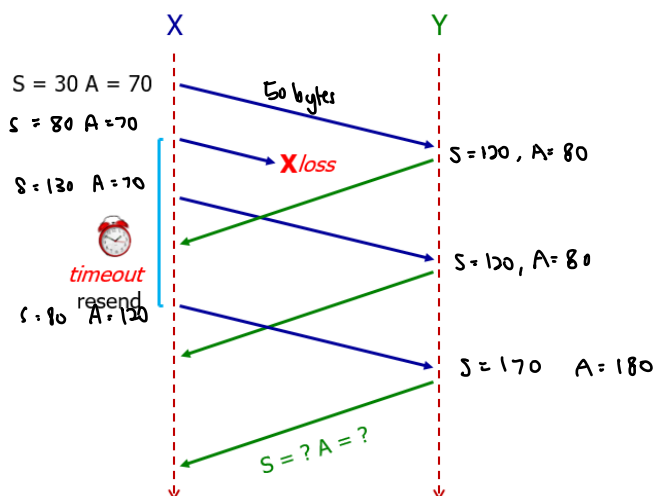
In the last TCP segment send by **Y**, the sequence number S = __1__ and the acknowledgement number A = __2__ .

Enter the correct answer below.

1 | 170 | Please enter a number for this text box.
2 | 180 - | Please enter a number for this text box.

Response Rationale

Please provide a rationale for your answer.

| Font ▾ | Size ▾ | ▾ ▾ | | | | | ▾ | Source |

25.

Fill in the blanks (2 marks)

With UDP streaming the media server __1__ the video data to the client (player), whereas with HTTP streaming the client __2__ the video data from the server.

Enter the correct answer below.

1 | push
2 | pull

Response Rationale

Please provide a rationale for your answer.

| Font ▾ | Size ▾ | ▾ ▾ | | | | | ▾ | Source |

26.

Fill in the blanks (2 marks)

In a VoIP application (or any media streaming application that requires a continuous, uninterrupted media data playout), then at the receiver side two types of data losses may occur: __1__ loss and __2__ loss.

Enter the correct answer below.

1 | Network
2 | Delay

Response Rationale

Please provide a rationale for your answer.

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27.

Fill in the blanks (2 marks)

Suppose a UDP server process with a port number 40,000 sends a UDP segment to a client whose port number is 40,001. If the segment does not contain any payload, the 16-bit binary value of the length filed is __1__ and the 16-bit binary value of the checksum filed is __2__. [The 16-bit binary representation of 40,000 is 1001 1100 0100 0000]

Enter the correct answer below.

1 | 0000000000001000 | Character Limit: 16
2 | 0011000100010010 | Character Limit: 16

Response Rationale

Please provide a rationale for your answer.

Font ▾ | Size ▾ | ▾ ▾ | | | | | ▾ Source

64 bits = 8 bytes

Source port + Dest port + length = checksum.

```
        1   1 1       1
     1001 1100 0100 0000
  +  1001 1100 0100 1001
   1 0011 1000 1000 1001
  +                    1
     0011 1000 1000 1010
  + 0000 0000 0000 1000
     0011 1000 1000 1010
```

28.

Fill in the blanks (2 marks)

Consider IP Datagram Fragmentation, Lecture 07.

An IP datagram needs to be transmitted at the link layer over a link that has an MTU of 1500 bytes. The IP datagram's "IP datagram length" field has a value of 3000 bytes. Therefore the IP datagram needs to be fragmented.

What is the value of the "IP datagram length" field of the **last** fragment that is transmitted? __1__   1480 data.   2980 data.

20 data

Enter the correct answer below.

1 | 40 | Please enter a number for this text box.

Response Rationale

Please provide a rationale for your answer.

Font ▾ | Size ▾ | ▾ ▾ | | | | | ▾ Source

[ ]

29.

Fill in the blanks (2 marks)

Suppose two hosts are connected by a ==direct link of 1 Mbps.== A ==stop-and-wait protocol== is used to transfer ==100 packets== from the sending host to the receiving host. Each packet is ==1000 bytes long.== RTT is 8 milliseconds. No packet is lost or corrupted during transmission and ACK packets are of negligible size.

The throughput of the transmission is ___1___ Mbps.

Enter the correct answer below.

1 [ ] Please enter a number for this text box.

Response Rationale
Please provide a rationale for your answer.

Font	Size	▼	▼					▼	Source

Total size = $100 \times 1000 \times 8 = 800\,000$ bits.

Total time = $8 \times 10^{-3} \times 100 = 0.8s$.

∴ Throughput = $\dfrac{800\,000}{0.8}$ = 1 Mbps.

[Finish Quiz] [Save For Later]