Student Number

THE UNIVERSITY OF MELBOURNE

SCHOOL OF COMPUTING AND INFORMATION SYSTEMS

PRACTICE EXAM – Semester 2, 2018

COMP90007 Internet Technologies

Exam Duration: 3 hours **Total marks for this Exam:** --
Reading Time: 15 minutes

Length: This paper has 18 pages including this cover page.

Authorized materials: Writing materials (e.g. pens, pencils). Calculators and all other books are not allowed.

The exam paper must remain in the exam room and be returned to the subject coordinator.

Instructions to Students:

- This paper contains 35 questions.
- Answer questions in this exam booklet using pen only in the **space** provided after the questions. All even pages are intentionally left blank, which you can use for rough work. Note that only your answers within the given space on odd numbered pages will be marked.
- For multiple choice questions, choose the best answer.
- For short answer questions, two or three sentences should be sufficient to answer each question.
- Bullet points are acceptable in answering descriptive questions.
- *Any unreadable answers will be considered wrong*.

Multiple choice questions [choose only one answer for each question]: [1 mark each]

Q1. The Internet Protocol (IP) address of the sending and receiving hosts is identified at which layer of the OSI model?

A) Layer 7
B) Layer 2
C) Layer 4
D) Layer 3
E) All of the above
F) None of the above

Q2. Which of the following statements is correct regarding Services and Protocols?

A) Service is a set of primitives that a layer provides to a layer above it
B) Service defines what operations the layer is prepared to perform on behalf of it's users
C) Protocol is a set of rules governing the format and meaning of packets exchanged by peers within a layer
D) All of the above
E) None of the above

Q3. Message latency, which is the time delay associated with sending a message over a link is made up of ___

A) Jitter
B) Transmission delay (message in bits / transmission rate)
C) Propagation delay (length of channel / signal speed)
D) Both transmission delay and propagation delay
E) All of the above
F) None of the above

Q4. Which of the following statements is correct?

- **A)** Attenuation is the loss of reduction in amplitude of a signal as it passes through a medium
- **B)** Time division multiplexing is used to give access to users with a fixed schedule
- **C)** Frequency multiplexing allocates specific frequency range for each user to send their data
- **D)** All of the above
- **E)** None of the above

Q5. In Data Link layer, which framing methods can be used?

- **A)** Character (Byte) count
- **B)** Flag bytes with byte stuffing
- **C)** Start and end flags with bit stuffing
- **D)** All of the above
- **E)** None of the above

Q6. Hamming distance, *d*, is the minimum bit flips to turn a valid codeword into any other valid one. Therefore, a Hamming distance of 4 can correct how many errors?

- **A)** 1
- **B)** 2
- **C)** 3
- **D)** 4
- **E)** All of the above
- **F)** None of the above

Q7. In MAC Sublayer, which of the following protocols could provide greater throughput under high network traffic load?

A) Pure ALOHA
B) Slotted ALOHA
C) 1-persistent CSMA
D) 0.1-persistent CSMA
E) All of the above
F) None of the above

Q8. Which of the following protocols is a limited contention protocol?

A) Binary Countdown protocol
B) Bit Map protocol
C) Carrier Sense Multiple Access (CSMA)
D) Adaptive Tree Walk Protocol
E) All of the above
F) None of the above

Q9. In Network layer, which of the following statements is correct?

A) Services provided by a Network layer protocol depend on the router technology
B) 20-bit label in MPLS network is used to determine where in the datagram the current fragment belongs to
C) Subnetting is used to assign local private IP addresses to hosts
D) NAT is a process used to join multiple IP prefixes into a single layer prefix to reduce the routing table size
E) All of the above
F) None of the above

Q10. What are the approaches used in establishing a reliable connection?

- **A)** Don't reuse Maximum Segment Lifetime (MSL) sequence numbers within twice the MSL?
- **B)** 3-way handshake for establishing connection
- **C)** Use a sequence number space large enough that it will not wrap even when packets are sent at a high transmission rate
- **D)** All of the above
- **E)** None of the above

Q11. Which of the following statements is correct?

- **A)** UDP is a connection-oriented protocol
- **B)** TCP is a connectionless protocol
- **C)** UDP supports flow control and error control through retransmission of bad segments
- **D)** All of the above
- **E)** None of the above

Q12. Which of the following techniques is/are used for supporting good quality-of-service (QoS)?

- **A)** Over-provisioning
- **B)** Traffic shaping
- **C)** Resource reservation
- **D)** Packet scheduling
- **E)** All of the above
- **F)** None of the above

Q13. In the lectures, we learnt that the Domain Name System is a distributed database implemented in a hierarchy of many name servers. Which of the following reasons is correct regarding why DNS is not centralised?

A) Single point of failure
B) Huge traffic volume will be going to a centralised location
C) Distant centralized database for many users
D) Difficult or hard to maintain a centralised system
E) All of the above
F) None of the above

Q14. In security, which of the following statements is correct?

A) Symmetric Key Algorithms uses the same key for both encryption and decryption
B) Asymmetric Key Algorithms allow different keys to be used for encryption and decryption
C) Public Key Infrastructure is important to ensure safe distribution of public keys
D) All of the above
E) None of the above

Q15. Which of the following statements is correct about MD?

A) MD uses a one-way hash function to take an arbitrary length of plaintext and compute a fixed-length bit string
B) A MD from plaintext is much faster than encrypting plaintext
C) MD could be used to speed up the derivation of a digital signature
D) All of the above
E) None of the above

Q16. a. Describe the OSI layer division principles (name 3). [1.5 marks]

• A layer should be created where a different abstraction is needed.
• Each layer should perform a well-defined function.
• The function of each layer should be chosen with a view toward defining internationally standardised protocols.
Also
• The layer boundaries should be chosen to minimise the information flow across the interfaces.
• The number of layers should be large enough that distinct functions need not to be thrown together in the same layer out of necessity, and small enough that the architecture does not become unwieldy.

b. Briefly explain the relative advantages and disadvantages of the OSI Reference Model versus the TCP/IP Reference Model. [1.5 marks]

OSI Negatives:
• Timing – TCP/IP adopted before OSI was even formalised. Vendors did not want to support 2nd standard.
• Although there's 7 layers, 2 of these (session, presentation) are almost empty and 2 others (data link, network) are cramped.
• Additionally some functions such as addressing, flow control and error control are recurring at each OSI layer.
• OSI implementations were inefficient compared to TCP/IP.
• OSI was widely perceived as the product of quasi-government standards processes rather than driven by good design processes
• TCP/IP Negatives:
• Lack of distinction between concepts doesn't clearly distinguish between service, interface and protocol
• Not adaptable not a general model, and hence poorly adapted to other protocol stacks
• Ambiguous layers Host-to-network is not really a layer, but an interface between network and data link layers
• Omitted layers Physical and data link layers are not present
• Early implementations were fragile

Q17. The following data is the output of traceroute on a computer in the laboratory.

traceroute to cis.unimelb.edu.au (128.250.37.164), 64 hops max, 52 byte packets
1 10.9.152.1 3.304 ms 3.304 ms 3.304 ms
2 172.18.68.81 1.146 ms 1.099 ms 1.076 ms
3 172.18.68.83 1.133 ms 1.144 ms 1.115 ms
4 172.18.68.33 2.175 ms 1.931 ms 2.149 ms
5 172.18.66.133 9.724 ms 1.688 ms 1.989 ms
6 128.250.37.130 1.246 ms 1.205 ms 1.381 ms
7 128.250.37.164 1.988 ms 2.035 ms 1.848 ms

a. What is the IP address of the router connected to the destination? [1 mark]

128.250.37.130

b. Explain how Traceroute uses ICMP (Internet Control Message Protocol) in its operation. [2 marks]

One suggested answer:
Traceroute sends packets to measure the RTT of each intermediate router in the path to reach the destination host.

It does this by sending packets with different TTL (hop limit) values to reach the intermediate routers. The packets Traceroute uses can be ICMP 'echo request' packets, UDP packets or TCP SYN packets.

Upon reaching a router, the TTL value is decremented by one, and is dropped if the TTL value is zero. If the packet is dropped, an ICMP 'time exceeded' packet is sent back to the sender. Traceroute measures the timestamp of this packet to determine the RTT.

Q18. a. Give three main characteristics that affect the performance of applications on networks. [1.5 marks]

Bandwidth Latency Jitter

b. Give an example of application that has stringent requirements on each of the three main characteristics you have given in Q18. a. [1.5 marks]

Bandwidth – HD video Latency – Interactive gaming Jitter – Real-time video

Q19. Briefly explain the relative advantages and disadvantages of using Fibre Optics versus Copper Wire. [3 marks]

Fibre – run over longer distances, higher bandwidth, scalable, rejects noise, thin, however it is expensive, requires specialists to deploy, is difficult to tape and is fragile. Copper is cheaper, no specialist skill required.

Q20. a. How can you increase the bit rate of a 1200 baud line from 1200 bit/s to 3600 bit/s? [1.5 marks]

Increase the number of bits per symbol from 1 bit/symbol to 3 bits/symbol, for example, by increasing from a two-level code to a 8-level code.

b. Consider a telephone signal that is bandwidth limited to 4 kHz. (i) At what rate should you sample the signal so that you can completely reconstruct the signal? (ii) If each sample of the signal is to be encoded at 256 levels, how many bits/symbol are required for each sample? (iii) What is the minimum bit rate required to transmit this signal? [1.5 marks]

This is a typical question from physical layer: i) By Nyquist's Theorem: min sampling rate = 2 × 4 × 103 samples/s = 8×103 samples/s

(ii) 256 possible values per sample requires
log2 256 = 8 bits/sample

(iii) Simple exercise.

Q21. a. The following binary data fragment occurs in the middle of a data stream for which the bit-stuffing algorithm described in the lectures is to be applied:

0001111101111111111001

Show the output binary data stream after the bit-stuffing algorithm has been applied. (Note that you do not need to add any flag bytes) [1 mark]

Study the relevant material from the book. Only answer is shown below:
000111110011111011111000 1

b. The following data fragment occurs in the middle of a data stream for which the byte-stuffing algorithm described in the lectures is to be applied: [1 mark]

A B ESC D FLAG FLAG ESC C

Show the output data stream after the byte-stuffing algorithm has been applied.

A B ESC ESC D ESC FLAG ESC FLAG ESC ESC C

c. What is the maximum overhead in the byte-stuffing algorithm in general? [1 mark]

100% when the payload consists of only ESC and FLAG bytes.

Q22. a. Briefly explain the difference in operation and philosophy of two approaches to error handling on the data link layer; error-correcting and error-detecting. [1.5 marks]

Include enough information in frames to allow reconstruction/deduction of original content (error-correcting) Include enough redundancy to allow receiver to determine an error occurred and request retransmission (error-detecting)

b. Data link protocols almost always put the CRC in a trailer rather than in a header. Why? [1.5 marks]

The CRC is computed during transmission and appended to the output stream as soon as the last bit goes out onto the wire. If the CRC were in the header, it would be necessary to make a pass over the frame to compute the CRC before transmitting. This would require each byte to be handled twice – once for checksumming and once for transmitting. Using the trailer cuts the work in half.

Q23. If a LAN is under high load, would it be more efficient to use a contention protocol or a collision free protocol in the MAC Sub-layer? Briefly explain your answer. [3 marks]

Under high load a contention protocol would cause many collisions and not be effective, where a collision free protocol allows each source to use the network in turn. Therefore a collision free protocol should be used.

Q24. Consider a client program that needs to run the following operations on a remote file server:
a. List the contents of a directory
b. Open a file
c. Read a text file
d. Display the attributes of a file.

For each of the above operations, indicate whether they are more likely to be delay sensitive or bandwidth sensitive. Justify your answer? [3 marks]

a. Delay-sensitive; directories are typically of modest size. b. Delay-sensitive; the messages exchanged are short. c. Bandwidth-sensitive, particularly for large files. (Technically this does presume that the underlying protocol uses a large message size or window size; stop-and-wait transmission with a small message size would be delay-sensitive.)

d. Delay-sensitive; a file's attributes are typically much smaller than the file itself (even on NT filesystems).

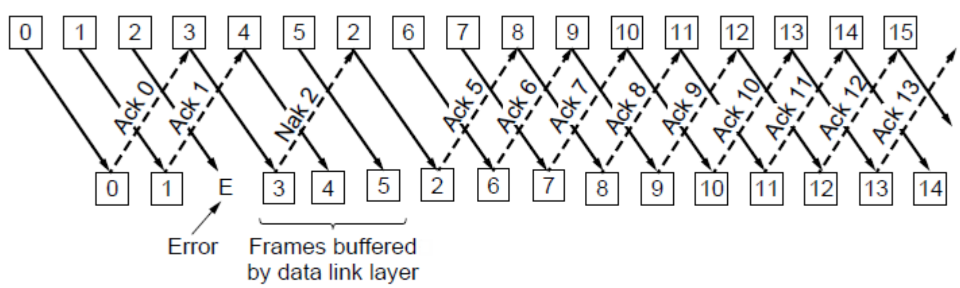
Q25. Give the names of two sliding window protocols and briefly explain how each protocol works. [3 marks]

Go-Back-N

Briefly explain the following diagram

Selective Repeat

Briefly explain the following diagram



Go-Back-N: receiver discards all subsequent frames from error point, sending no acknowledgement, until the next frame in sequence

Selective Repeat: receiver buffers good frames after an error point, and relies on sender to resend oldest unacknowledged frames

Trade-off between efficient use of bandwidth and data link layer buffer space

Q26. A router has built the following routing table. The router can directly deliver packets over Interface 0 and Interface 1 or it can forward to routers R2, R3 and R4.

Subnet Number	Subnet Mask	Next Hop
148.96.39.0	255.255.255.0	Interface 0
148.96.39.128	255.255.255.128	interface 1
148.96.40.0	255.255.255.128	R2
196.4.153.0	255.255.255.192	R3
Default		R4

Describe what the router does if a packet addressed to each of the following destinations is received.
(a) 148.96.40.12
(b) 148.96.39.193
(e) 196.4.153.90
[3 marks]

item(a) 148.96.40.12 -forwards to R2;

item(b) 148.96.39.193
Matches to 1) and 2) entries chooses the longest match; forwards it to Interface 1

item(c) 196.4.153.90 -forwards to R4 as it does not match to any entries from 1 to 4.

Q27. With respect to routing packets in the Network Layer, explain the difference between a connectionless and connection-oriented service? [3 marks]

Connectionless: packets are injected into the network individually and routed independently of each other. No advance setup is needed

Connection-oriented: If connection-oriented service is used, a path from the source router all the way to the destination router must be established before any data packets can be sent. This connection is called a VC (virtual circuit), in analogy with the physical circuits set up by the telephone system

Q28. Explain the purpose of subnetting and Classless Inter-Domain Routing (CIDR) for logically partitioning the IP Address space. [3 marks]

Subnetting allows networks to be split into several parts for internal uses whilst acting like a single network for external use Subnet masks can be written using: - “dotted decimal”(eg 255.255.255.128 indicates 2 internal networks) or “slash”notation (eg /25) Historically, classful addressing required allocation of complete classes (became very inefficient) - Currently CIDR (RFC1519) allows allocation of variable sized blocks of IP address space regardless of classes

Q29. a. Give three requirements that are needed to ensure reliable connection establishment in the transport layer. [1.5 marks]

We require that: (1) Packets in the network have a maximum lifetime (2) Sequence numbers are used by each end of the connection, such that they will not be reused during the maximum packet lifetime in the network, and (3) Use a 3-way handshake to exchange initial sequence numbers between both ends of the connection.

b. Give three types of policy choices at the Transport layer that can affect network congestion. In each case, briefly explain why the policy choice affects network congestion. [1.5 marks]

Retransmission policy - increases congestion if packets retransmitted too soon. Out-of-order caching policy - increases congestion if packets that arrive out-of-order cannot be buffered, and hence need to be retransmitted. Acknowledgement policy – negative acknowledgements can be used to avoid timeouts Flow control policy – small congestion windows reduce the data rate, and avoid congestion. Timeout determination – if too short, unnecessary retransmissions may result.

Q30. a. A common approach to removing jitter in streaming audio is to buffer incoming packets at the receiver. Briefly explain the main problem with using this approach for video conferencing. [1 mark]

a) Videoconferencing is a 2-way interactive service. Buffering introduces delay into the service, which is a nuisance for interactive services. b) Bringing content closer to the user (reducing overall network traffic)

b. Briefly explain what is bandwidth delay product? [1 mark]

Bandwidth Delay product is an important measure of a link or network path. It is defined as the product of the bandwidth of a link and the delay of the link. By "bandwidth" we mean the rate of the link or network path, R. For "delay", we will use the propagation delay, D. With this simplification we have: BD = R x D

c. Briefly explain what do we need low water mark and high water mark in media player buffer management? [1 mark]

Low water mark: Safety margin, to avoid a stall. When this threshold is reached, media player requests media server to resume data transmission. High water mark: Can pause server (or go ahead and save to disk). When this threshold is reached, media player requests media server to stop data transmission temporarily to avoid buffer overflow.

Q31. a. Give 3 reasons for the emergence of Voice-over-IP telephony as an alternative to the PSTN. [3 marks]

Data has overtaken voice as the primary traffic on many networks originally built for voice PSTN infrastructure is not flexible enough for the rapid deployment of new features PSTN technologies are largely incompatible with the convergence of data/voice/video The architecture built primarily for voice is not flexible enough to carry data Network providers are increasingly looking to leverage investment in network infrastructure by bring new services to data networks

Q32. a. Name two services that DNS provides and give one reason why DNS is not implemented centrally? [2 marks]

Services: Hostname to IP address translation Host Aliasing Load Balancing A centralized implementation results in a single point of failure and large amount of traffic to a single location

b. Is a DNS server a client, a server, or both? Briefly justify your answer. [1 marks]

a. Both, since it can act as a server if the requested domain name is in its database, or as a client if it needs to ask another server to resolve the name.

Q33. a. An encrypted file needs to be accessed in non-sequential order. Which cipher mode is best suited to encrypting this file, and briefly explain why [1.5 marks]

Counter mode is the best option, since each block can be encrypted or decrypted based on its location in the files using a counter. Other techniques required decryption of all preceding blocks in the file.

b. Give three important properties of a message digest [1.5 marks]

easy to compute MD(P) give P impractical to compute P given MD(P) given P, impractical to find P' such that MD(P') = MD(P) Also, a single bit change in P creates a very different message digest.

Q34. a. Briefly explain Dijkstra's algorithm in computing a set of optimal routes from all sources to a given destination. [1.5 mark]

Dijkstra's algorithm computes a sink tree on the graph: Each link is assigned a non-negative weight/distance Shortest path is the one with lowest total weight Using weights of 1 gives paths with fewest hops *Algorithm:* Start with sink, set distance at other nodes to infinity Relax distance to other nodes

Pick the lowest distance node, add it to sink tree
Repeat until all nodes are in the sink tree

b. Briefly explain Flooding and how it is used in any of the routing protocol. [1.5 mark]

- A non-adaptive algorithm
- Every incoming packet is sent out on every outgoing line except the one on which it arrived
- Generates a large number of duplicate packets - inefficient
- Selective flooding (where routers send packets only on links which are in approximately the right direction) is an improved variation

Used in LSR
In LSR:
Each router has to do 5 steps:
1. Discover neighbours and learn network addresses
2. Measure delay or cost to each neighbour
3. Construct packet resulting from previous steps
4. Send this packet to all other routers
5. Compute the shortest path to every other router

“Local information shared globally” using flooding

Q35. Considering the transport layer, briefly explain the key challenge in reliable connection establishment and 3 approaches in dealing with this challenge. [3 marks]

Key challenge is to ensure reliability even though packets may be lost, corrupted, delayed, and duplicated
- Don’t treat an old or duplicate packet as new
- (Use ARQ and checksums for loss/corruption)
-

Approaches:
1. Don’t reuse Maximum Segment Lifetime sequence numbers within twice the MSL (2min)
2. Three-way handshake for establishing connection
3. Use a sequence number space large enough that it will not wrap, even when sending at full rate