THE UNIVERSITY OF MELBOURNE
DEPARTMENT OF COMPUTING AND INFORMATION SYSTEMS

Practice Examination – Semester 1, 2015

COMP90007: Internet Technologies
Suggested solutions

Exam Duration: 3 hours **Total marks for this Exam: 60**
Reading Time: 15 minutes
This exam has 2 pages.

Authorised materials:
The following items are authorized: writing materials (e.g. pens, pencils) and non-electronic dictionaries are allowed.
Calculators and all other books are *not* allowed.

Instructions to Invigilators:
Supply students with standard script book.

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

Instructions to Students:

- This paper contains 20 questions each worth 3 marks. Attempt all questions.
- Answer questions in the script book(s) provided. Do not write your answers on this paper.
- Clearly number your answers.
- Bullet points are acceptable in answering descriptive questions.
- As a guide, two or three sentences should be sufficient to answer each question. Marks may be deducted for overly long answers or irrelevant information.
- *Any unreadable answers will be considered wrong*.

Q1. The following data is the output of traceroute on a computer in the EDS 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?

One suggested answer:
128.250.37.130

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

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.

Q2. Briefly explain the relative advantages and disadvantages of the OSI Reference Model versus the TCP/IP Reference Model. [3 marks]

Some suggested answers:
OSI Negatives:

- *Timing – TCP/IP adopted before OSI was even formalised. Vendors did not want to support 2nd standard.*
- *Although there are 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 and duplicated 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 the Physical and Data Link layers*
- *Early implementations were fragile*

Q3. The bandwidth and latency are the main characteristics of the networks affecting the performance of applications on networks. Define bandwidth and latency as discussed in lectures. Give an example of a network that exhibits high bandwidth but also high latency. Then give an example of a network that has both low bandwidth and low latency. [3 marks]

One suggested answer:
Bandwidth: the rate of data transferred across two hosts
Latency: the delay experienced by each communicated bit
A network with high bandwidth and high latency: transcontinental fibre link
A network with low bandwidth and low latency: telephone modem

Q4. How can you increase the bit rate of a 1200 baud line from 1200 bit/s to 3600 bit/s? [3 marks]

One suggested answer:
Increase the number of bits encoded per symbol from 1 bit/symbol to 3 bits/symbol. For example, by increasing from a two-level (BPSK) code to an 8-level code (8-PSK).

Q5. Briefly explain the relative advantages and disadvantages of using Fibre Optics versus Copper Wire. [3 marks]

One suggested answer:
Fibre: higher bandwidth, scalable, immune to electromagnetic noise, thin, but it is expensive, requires specialists to deploy and is fragile.
Copper: cheaper, no specialist skill required, but lower bandwidth.

Q6. Consider a telephone signal that is bandwidth limited to 4 kHz. [3 marks]

a. At what rate should you sample the signal so that you can completely reconstruct the signal?

One suggested answer:
By Nyquist's theorem: min sampling rate $= 2 \times 4\times10^3$
$= 8\times10^3$ *samples/s*
= 8 kHz

b. If each sample of the signal is to be encoded at 256 levels, how many bits/symbol are required for each sample?

One suggested answer:
256 possible values per sample requires:
$\log_2(256) = 8$ *bits/sample*

c. What is the minimum bit rate required to transmit this signal?

One suggested answer:
8×10^3 *samples/s* × *8 bits/sample*
$= 64\times10^3$ *bits/s*

Q7. 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)

One suggested answer:
`0001111100111110111110001`

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:

`A B ESC D FLAG FLAG ESC C`

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

One suggested answer:
`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? [3 marks]

One suggested answer:
100% when the payload consists of only ESC and FLAG bytes.

Q8. Briefly explain the difference in operation and philosophy of two approaches to error handling on the data link layer; error-correcting and error-detecting. [3 marks]

One suggested answer:

Error detection includes adding extra information (e.g. a checksum) to the information to allow the receiver to determine if an errors has occurred and subsequently request a restransmission.

Error correction goes one step further and adds extra redundancy to the information so that even in the presence of errors, some of the information can be reconstructed/deduced to what was transmitted.

Q9. Data link protocols almost always put the CRC in a trailer rather than in a header. Why? [3 marks]

One suggested answer:
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.

Q10. 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]

One suggested answer:
Under high load a contention protocol would cause many collisions and not be effective, whereas a collision free protocol allows each source to use the network in turn, with no collisions. Therefore, a collision free protocol should be used at the cost of a higher overhead compared to contention protocols.

Q11. Explain the operational benefits of Switched Ethernet in terms of handling increased traffic loads and avoiding collisions. [3 marks]

Some suggested answers:

- *Hubs do not buffer traffic and are highly susceptible to collisions.*
- *Switching allows the handling of increased traffic loads on a network.*
- *When a station wants to transmit an Ethernet frame, it sends a standard frame to the switch. The switch determines if the frame is destined for another station connected to the same switch, and if so, the frame is copied there. If not, the switch transmits the frame across a high-speed backbone to the station (possibly via another switch.*

Q12. 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	Router 2
196.4.153.0	255.255.255.192	Router 3
Default		Router 4

Describe what the router does if a packet addressed to each of the following destinations is received. [3 marks]

a. 148.96.40.12

One suggested answer:
Router 2

b. 148.96.39.193

One suggested answer:
Interface 1 (also matches Interface 0, but longest-prefix matching)

c. 196.4.153.90

One suggested answer:
Router 4 (no matches, so packets gets forwarded to Default interface)

Q13. Explain the purpose of subnetting and Classless Inter-Domain Routing (CIDR) for logically partitioning the IP Address space. [3 marks]

One suggested answer:
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"(e.g. 255.255.255.128 indicates 2 internal networks)*
- *"slash" notation (e.g. /25)*

Historically, classful addressing required allocation of complete classes (became very inefficient). Currently CIDR (RFC 1519) allows allocation of variable sized blocks of IP address space regardless of classes.

Q14. The following round-trip times (RTT) were measured by connecting from a host in Melbourne to one residing in the Netherlands.

Trial i	x_i RTT (ms)
1	0

2	1000
3	2000

a. Determine the jitter in milliseconds. The jitter is defined as the square root of the variance of a normal distribution, as shown below:

$$\sigma = \sqrt{\frac{1}{N-1}\sum_{i=1}^{N}(x_i - \bar{x})^2}$$

where $\bar{x}$ is the mean of the samples.

One suggested answer:

$$\bar{x} = \frac{0 + 1000 + 2000}{3} = 1000\text{ms}$$

$$\sigma = \sqrt{\frac{1}{3-1} \times [(0-1000)^2 + (1000-1000)^2 + (2000-1000)^2]} = 1000\text{ms}$$

b. Explain how the value of jitter obtained above can be detrimental to certain network applications running on the host.

One suggested answer:
Teleconferencing, VoIP and media streaming are jitter sensitive applications. For example, a variance in the arrival time of video frames will cause the video to stutter.

c. How can this problem be alleviated and what are the corresponding trade-offs involved? [3 marks]

One suggested answer:
Can mitigate this issue by using a buffer at the cost of increasing the delay.

Q15. 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. [3 marks]

Some suggested answers:

- *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.*

Q16. 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. [3 marks]

One suggested answer:
Videoconferencing is a 2-way interactive service. Buffering introduces delay into the service, which is a nuisance for interactive services.

Q17. Briefly explain the architecture of the email system by describing key components and services. What are the basic steps of SMTP protocol? [3 marks]

One suggested answer:

- *User agents: fetch email from MTAs with POP3 or IMAP (delivery protocols) and allows users to read emails*
- *Mail transfer agents: transports emails to destination via SMTP (transfer protocols)*
- *SMTP uses ASCII human-readable commands to communicate with MTAs. Typical message transfer:*
    - *HELO [server name]*
    - *MAIL FROM [sender]*
    - *RCPT TO [recipient]*
    - *DATA*
    - *QUIT*

Q18. Give 3 reasons for the emergence of Voice-over-IP telephony as an alternative to the PSTN. [3 marks]

Some suggested answers:

- *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*

Q19. a. Is a DNS server a client, a server, or both? Briefly justify your answer.

One suggested answer:
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.

b. Give three important properties of a message digest. [3 marks]

Some suggested answers:
- *Easy to compute MD(P), given P*
- *Impractical to compute P, given MD(P)*
- *Given P, impractical to find P' such that MD(P') = MD(P)*
- *A single bit change in P creates a very different message digest.*

Q20. Briefly summarize the relative strengths and weaknesses of using UDP to implement remote procedure calls (RPC). [3 marks]

One suggested answer:
Strengths:
- *UDP is fast, a request and reply may only need one packet each in small messages*
- *Lost packets can be detected with timeouts and retransmitted, without a TCP 3-way handshake for every RPC call*

Weaknesses:
- *Repeated packets may cause repeated executions which change state (protocol is not idempotent, like DNS, where multiple calls are safe) – need semantics to detect this behaviour*
- *If message or parameters to deliver are greater than the UDP packet size, then other protocols need to be developed to account for this*

End of exam

Additional Practice Questions

Q1. The following data is the output of traceroute on a computer in the EDS laboratory. [3 marks]

```
traceroute to cis.unimelb.edu.au (128.250.37.164), 64 hops
max, 52 byte packets

1 10.0.0.254 33.100 ms 33.100 ms 33.100 ms
2 58.96.2.205 27.800 ms 27.800 ms 27.800 ms
3 58.96.2.129 28.000 ms 28.000 ms 28.000 ms
4 218.100.78.33 28.299 ms 28.469 ms 28.332 ms
5 202.158.200.9 29.626 ms 28.871 ms 29.841 ms
6 202.158.210.26 31.320 ms 28.722 ms 29.135 ms
7 202.158.200.250 29.668 ms 29.096 ms 28.660 ms
8 * * *
9 * * *
10 * * *
11 128.250.37.130 957.521 ms 33.475 ms 29.891 ms
12 128.250.37.164 29.940 ms 29.260 ms 30.020 ms
```

a. Describe two techniques that can be used in traceroute to speed up determining the number of hops to a host. Assume that all other information aside from the final hop count is irrelevant in the traceroute output for this particular application.

One suggested answer:

- *Don't resolve the IPs to hostname, saving the time required to make a DNS lookup*
- *Set a shorter timeout, since only the hop count is important, not the RTT*

b. Why is the eighth, ninth and tenth hops shown as stars?

One suggested answer:
That particular server did not respond to the packets used in traceroute; whether by configuration of the remote server or by firewall policies.

c. What is the round trip delay between the source and the router with IP address 58.96.2.129?

One suggested answer:
28 ms

Q2. With respect to routing packets in the Network Layer, explain the difference between a connectionless and connection-oriented service? [3 marks]

One suggested answer:
Connection-less: move packets in a potentially unreliable subnet, QoS is not easily implemented, e.g. the internet
Connection-oriented: Packets travelling between destinations all use the same route. Telco view: guarantee reliability of subnet – QoS is important

Q3. 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. [3 marks]

One suggested answer:
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.

Q4. Give three requirements that are needed to ensure reliable connection establishment in the transport layer. [3 marks]

One suggested answer:
We require that:

- *packets in the network have a maximum lifetime*
- *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*
- *use a 3 way handshake to exchange initial sequence numbers between both ends of the connection.*