Digital Communication I

Describe briefly both Synchronous and Asynchronous Time Division Multiplexing (TDM). [4 marks]

Describe four solutions to the problem of contention resolution in Asynchronous TDM. [12 marks]

Which solution is adopted by Ethernet and what measures are taken to ensure stability in circumstances of high load? [4 marks]
Digital Communication I

Describe the properties of a physical channel which need to be considered when the channel is used for communications. [7 marks]

Describe how the properties of a digital synchronous channel are related to the properties of the underlying physical channel. [7 marks]

Describe three ways in which a protocol entity can provide higher layer channels from lower layer channels. [6 marks]
Digital Communication I

Define the term *circuit* as used in “circuit switching”. \[4 \text{ marks}\]

Sketch the design of the in-band switching function of a circuit switch which switches \(4 	imes 2\) Mbps trunks each supporting \(32 	imes 64\) Kbps channels. \[8 \text{ marks}\]

Describe how you would augment this design to allow the set up and clearing of connections. You should invent your own simple protocol for this purpose. \[8 \text{ marks}\]
Define the term *flow control*. [5 marks]

How does it differ from *congestion control*? [3 marks]

What is meant by the terms *entry level*, *hop by hop* and *end to end* flow control? When is each appropriate? [8 marks]

Sketch the design of a simple flow control protocol. [4 marks]
Digital Communication I

Compare the functions of a *MAC level bridge* with an *IP router*. In what circumstances is it more appropriate to use one than the other? [6 marks]

Discuss the tables inside both bridges and routers used to control the acceptance and forwarding of packets. Indicate both how these tables are used and how information is put in the tables. How quickly can the tables be searched in each case? [10 marks]

What features might be added to a router or bridge to improve some aspects of network security? [4 marks]
Digital Communication I

What is the purpose of a signalling system in a digital telephone network? Describe the operations that the signalling system should perform. [6 marks]

Describe a possible implementation of a signalling system, including a discussion of how signalling information might be carried over the network. [8 marks]

What are the advantages/disadvantages of having a separate network for signalling traffic? [3 marks]

To what extent is a signalling system different from a general-purpose distributed computation? [3 marks]
Hosts X and Y are communicating through the data network provided by the switches A, B, C and D and the links interconnecting them as shown above. Initially all packets are travelling through switches A, C and D.

(a) A packet is corrupted on the link between C and D. Describe the events that take place to recover from the error when

(i) an end to end flow and error control protocol is in operation  [5 marks]

(ii) flow and error control are performed on a hop by hop basis  [5 marks]

(b) Switch C fails. Describe the events that follow to recover when

(i) the network is a datagram network  [5 marks]

(ii) the network is connection oriented  [5 marks]
Digital Communication I

Operations of similar functionality can be performed at different layers of a protocol stack. Discuss this in relation to

(a) routing [4 marks]

(b) multiplexing [4 marks]

(c) error recovery [4 marks]

(d) flow control [4 marks]

(e) synchronization [4 marks]
Digital Communication I

Compare the multiplexing aspects of packet switching and circuit switching. [5 marks]

ATM has been described as a compromise between circuit switching and packet switching. Explain this with respect to multiplexing. [5 marks]

Consider a generic switch with multiple inputs and outputs. Describe the functions that are performed in moving information from an input to an output in (a) a circuit switch and (b) a packet switch. How is contention for output transmission capacity resolved in each case? [10 marks]
Digital Communication I

Explain the terms ARQ protocol and window of an ARQ protocol. [5 marks]

An ARQ protocol uses a window of 1 kbyte. The protocol is used over a link whose capacity is 1 Mbps. In the absence of transmission errors (or any other loss) determine (a) for a link delay of 100 µs, and (b) for a link delay of 250 ms, the time required to transfer each of the following amounts of information over the link:

1 kbyte, 1 Mbyte and 1 Gbyte [12 marks]

State and explain in which of these cases moving to a larger window size will not significantly improve the transfer time. [3 marks]
Digital Communication I

How can packet loss occur in a network? [5 marks]

Outline a way in which packet loss can be reduced. Can it be eliminated completely? [5 marks]

How does an ARQ system deal with packet loss? [5 marks]

“An ARQ implementation should keep as much data in transit as the receiver is willing to receive.” Discuss. [5 marks]
You are required to design a topology discovery protocol for a network of switching nodes interconnected by links. There are \( n \) nodes, \( l \) links, the maximum degree of any node is \( k \) and there is a path between any two nodes of not more than \( d \) hops. All links are bi-directional.

Each node has a unique identifier of four bytes which it knows.

(a) Design a protocol (including message formats) for a node to learn about its immediate neighbours. [5 marks]

(b) Design a protocol (including message formats) for distributing this information across the network. [10 marks]

(c) Give a bound on the total amount of information which is transmitted to ensure that every node acquires complete topology information. [5 marks]
Digital Communication I

Compare packet switching and circuit switching with particular reference to the following issues:

(a) how multiplexing is performed in each

(b) how addressing is performed in each

(c) functions which must be performed by a switch in each case

(d) situations in which each is advantageous [15 marks]

“As communication bandwidth becomes less and less expensive, the efficiency of packet switching will become less important than the simplicity of implementation and guarantees offered by circuit switching.” Discuss. [5 marks]
Digital Communication I

What is a **hierarchical address space**? Give an example of an address space which is hierarchical and one which is not. [3 marks]

What is the Address Resolution Protocol? Describe its operation when used to resolve IP addresses to Ethernet addresses. Pay particular attention to the freshness of information. [7 marks]

Information is transferred via a long, error-prone communication link. The link has a data rate of 10 Mbps and a constant delay. The bit error rate on the link is 1 bit in $10^4$. A forward error correcting coder is available which can act in the following settings:

<table>
<thead>
<tr>
<th>Data rate</th>
<th>Error rate</th>
<th>Code rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>10 Mbps</td>
<td>$10^{-4}$</td>
<td>unity (no coding)</td>
</tr>
<tr>
<td>5 Mbps</td>
<td>$10^{-5}$</td>
<td>half</td>
</tr>
</tbody>
</table>

A simple ARQ protocol is used over the link. Packets have 32-bit CRCs. You may assume that the undetected error rate is less than 1 in $10^{20}$, that is, effectively zero.

Information is sent in 1000-bit packets, with a window of one packet. At what link delay would it be beneficial to use the FEC coder? [10 marks]
What is meant by the term flow control? [3 marks]

What is meant by the term credit-based flow control? [4 marks]

What is meant by start-stop (or XON–XOFF) flow control? [4 marks]

A start-stop system is used on a 10 kbps link with a constant delay of 5 ms. How much buffer must a receiver keep in reserve for “stopping time” in order to prevent information loss? [3 marks]

Which system is more appropriate to use across the Internet and why? [6 marks]
Digital Communication I

Compare circuit switching and packet switching, paying attention to channel characteristics and resource efficiency. [7 marks]

What is wave division multiplexing (WDM)? Is it more like circuit switching or packet switching and why? [7 marks]

Wave length conversion is the process, either optical or optical–electronic–optical, of receiving a signal on one wavelength and transmitting on another.

How does wave length conversion ease the problem of routing optical carriers in a network? [3 marks]

“The huge capacity of WDM systems will mean that IP becomes redundant.” Discuss. [3 marks]
Digital Communication I

Information is to be conveyed from A to B using automatic repeat request (ARQ), forward error correction (FEC), and lossless compression.

(a) Explain the terms ARQ, FEC and lossless compression. [5 marks]

(b) If we consider each of these functions to be operating at different protocol layers, what would be the most sensible ordering of the layers, and why? [5 marks]

(c) Suppose:

- The underlying bit channel has a capacity of B, a delay τ and error rate ϵ₀.
- The compression ratio is C < 1.
- The FEC has rate R < 1 and given an error rate ϵ₀ provides an error rate ϵ₁ (which is detected).
- The ARQ protocol has a window size of W.

At what rate can the information be conveyed? [Hint: Consider when retransmissions are made.] State any assumptions you make about the operation of the ARQ protocol. [10 marks]
Digital Communication I

(a) Define the terms circuit and packet in the context of communication systems. [5 marks]

(b) What sort of guarantee does circuit switching provide? [5 marks]

(c) What advantages does packet switching provide over circuit switching? [5 marks]

(d) Which of frequency division multiplexing, time division multiplexing and code division multiplexing lend themselves to circuit switching? Which to packet switching? Explain why or why not in each case. [5 marks]
Digital Communication I

Consider the real-time transport of audio across a network.

(a) What are the advantages of digitising the audio? [5 marks]

(b) What are the disadvantages and how can they be mitigated? [5 marks]

(c) What characteristics of the end-to-end channel across the network would be desirable, and how are these different from those which would be desirable for time-insensitive data? [5 marks]

(d) Discuss the applicability of asynchronous and synchronous multiplexing in carrying real-time audio traffic. [5 marks]
Define a *resource* in a digital communication system as anything whose use by one instance of communication prevents simultaneous use by another. Channel capacity is one example.

(a) Give two more examples of resource in digital communication systems. [4 marks]

(b) For the three resources, indicate how the amount of total resource can be increased. [6 marks]

(c) How are allocations of each of these resources to instances of communication performed? [10 marks]
Digital Communication I

(a) Define the terms capacity and latency as applied to a communications channel. [4 marks]

(b) How can variable latency cause problems? You may wish to consider

(i) XON/XOFF flow control;

(ii) streaming media;

(iii) protocol timeouts. [6 marks]

(c) Describe the operation of a simple ARQ protocol with a window of a single packet. [4 marks]

(d) A simple ARQ scheme is used to provide reliable transport over a link where 80% of packets other than short acknowledgements experience a 1 ms delay, 10% experience a 10 ms delay, and 10% are lost. Acknowledgements always experience a 1 ms delay and are never lost. What would be the expected throughput in packets/sec if the timeout was

(i) 10 ms?

(ii) 12 ms?

Assume that the transmitter always has information to send and that transmission time is negligible.

It may be helpful to note that

\[ \sum_{i=0}^{\infty} i x^i = \frac{x}{(1-x)^2} \] [6 marks]
Digital Communication I

(a) Define the term *multiplexing* as applied to communication systems. [4 marks]

(b) Describe *three* types of multiplexing, identifying in each case

(i) mechanisms by which symbols are associated with particular channels;

(ii) mechanisms by which transmitters are assigned channel resource;

(iii) characteristics of the multiplexed channels;

(iv) applications which are suited to the type of multiplexing. [16 marks]
Digital Communication I

(a) Define the terms flat and hierarchical as applied to address spaces. [2 marks]

(b) Give four examples of address spaces and state whether they are flat or hierarchical, and why. [4 marks]

(c) Describe class-based addresses as used in the Internet. You need not worry about precise field sizes or class names. [4 marks]

(d) Describe classless addresses as used in the Internet. [3 marks]

(e) Why were they introduced? [2 marks]

(f) What information must be held in a routing table when classless addresses are used? [5 marks]
Digital Communication I

(a) Define the terms *latency* and *capacity* as applied to communication channels. [2 marks]

(b) Is there a strict relation between the two? [1 mark]

(c) Show how the latency of a channel can have a direct effect on the capacity of a higher-layer channel which uses it. [10 marks]

(d) How can the capacity of the higher-layer channel be improved (keeping the characteristics of the underlying channel unchanged)? [4 marks]

(e) In what circumstances might these improvements have only limited benefit? [3 marks]
Digital Communication I

It is proposed to send information across a fixed delay channel using a simple (window of 1) ARQ protocol with a transmitter timeout of $T$. That is, if the transmitter does not receive an acknowledgement for a packet within time $T$ of sending the packet, it retransmits.

The delay of the underlying channel is $\tau$, the data rate is $B$ and the packet size is $p$ bits. Bit errors in the channel are independent and packets of size $p$ have a packet error rate of $e$. Errors in the small acknowledgement packets are rare enough to be discounted in this analysis.

(a) What is the expected throughput of the ARQ protocol if $e$ is zero? [4 marks]

(b) What is the expected throughput if $e$ is non-zero, but small enough that $e^2$ is negligibly small? [4 marks]

(c) How could a forward error code help the throughput of the ARQ scheme? [2 marks]

(d) What is meant by the term code rate of a forward error code? [2 marks]

(e) What code rate must a code which squared the error rate have in order to improve throughput of the ARQ scheme? [4 marks]

(f) If the forward error coder adds delay, how will this affect performance? [4 marks]
Digital Communication I

(a) Describe on–off flow control. In what circumstances is it appropriate? [4 marks]

(b) Describe the operation of window-based flow control. [4 marks]

(c) What happens if window-based flow control is used on a flow passing through a highly loaded resource (e.g. router) that is not participating in the flow control protocol? [4 marks]

(d) How is this addressed in the Internet? [4 marks]

(e) What are the advantages and disadvantages of having Internet routers participate in window-based flow control of every TCP connection? [4 marks]
Digital Communication I

(a) Describe the concepts of *circuit switching* and *packet switching*. [5 marks]

(b) What are the fundamental advantages of each over the other? [5 marks]

(c) What is the role of buffering and buffering policy in each approach? [5 marks]

(d) There is an expectation that in the near future telephony will move from circuit switching to packet switching. Why is this so in light of the advantages of each approach? [5 marks]
Digital Communication I

(a) Describe and contrast the processes of (i) forward error correction and (ii) error detection with retransmission. [5 marks]

(b) What properties should be considered when deciding which should be used to control errors? [10 marks]

(c) Are there circumstances when both should be used? Justify your answer. [5 marks]
Digital Communication I

Using four examples, explain how multiple higher-layer channels can be multiplexed onto a lower-layer channel. In each example consider

(i) how the individual higher-layer channels can be recognised;

(ii) what the mechanism is for allocation of lower-layer channel resources to the higher-layer channels; and

(iii) the characteristics of the higher-layer channels.

[4 × 5 marks]
(a) Define the term *flow control* as used in communication networks. [4 marks]

(b) Describe on-off flow control, window-based flow control, and flow control used in circuit switching. [9 marks]

(c) Consider a channel of capacity $b$ and delay $\tau$, over which packets of size $p$ are sent. Compare the performance of window-based flow control protocols having:

(i) a window size of one packet;

(ii) a window size of two packets; and

(iii) a window size of one packet, but with a packet size of $2p$. [7 marks]
Digital Communication I

(a) Describe five physical properties of a communications channel. [5 marks]

(b) Consider the figure below. Entities $N$ and $N'$ use an ARQ system.

(i) Explain how the latency of channel $n - 1$ can have a direct effect on the capacity of channel $n$. [6 marks]

(ii) Define windowing as it relates to an ARQ system and describe how the capacity of the ARQ system may be improved through its use. [4 marks]

(iii) If an ARQ system is used for an interactive session, the ARQ system can lead to many small packets, each under-full and perhaps sent with significant overhead. Design and describe an algorithm that overcomes the limitation of sending many mostly-empty packets for an interactive session. [5 marks]
Digital Communication I

(a) For each of these examples of addressing, state whether it is flat or hierarchical and why:

(i) postal;

(ii) telephone;

(iii) Ethernet (MAC) address;

(iv) Internet (IP) address. [4 marks]

(b) Compare class-based and classless addresses as used in the Internet. [4 marks]

(c) Why were classless addresses introduced? [1 mark]

(d) Consider a router of IP packets.

(i) What information must be held in a routing-table when classless addresses are used? [3 marks]

(ii) Describe Longest-Prefix Match, providing an example of its use. [3 marks]

(iii) Describe the process of routing-table lookup that leads to the default-route being used and comment on the circumstance in which an IP router does not have a default-route. [3 marks]

(e) Considering your answers to part (d), describe two challenges for router-vendors following the introduction of classless addressing. [2 marks]
Digital Communication I

(a) Compare flow control with congestion control. [2 marks]

(b) Describe what is meant by sliding-window protocol. [2 marks]

(c) Describe how a sliding-window protocol may be used to implement flow control. [2 marks]

(d) Explain why implementing flow control in this manner is not a good idea. [2 marks]

(e) Provide an alternative to a sliding-window protocol for the implementation of flow control. [2 marks]

(f) Consider a sliding-window protocol for a point-to-point link from the surface of the earth to a geostationary satellite. The link speed is 1Gbps and the one-way latency is 125ms.

(i) Assuming each packet of data is fixed to 1KByte in length, what is the minimum number of bits you need for the sequence number? [2 marks]

You have been asked to construct an emulation of the satellite link, replicating the behaviour (delay and speed), allowing others to test their applications without using the satellite system. A simple way to do this is to provide an artificial delay of packets, emulating their flight to and from the satellite.

(ii) For a simple packet length of 1KByte, how much memory is required to emulate the satellite link alone? Comment on other sources of memory utilisation. [2 marks]

(iii) What capabilities must a standard computer have to emulate a link with 1Gbps capacity? Consider the speed and delays in the CPU, memory, and PCI interconnect. Comment on the suitability of a standard PC platform for such a task. [6 marks]
Digital Communication I

(a) Define the following terms and illustrate with an example.

(i) Baud (sometimes referred to as the Baud rate), comparing it with bit rate. [1 mark]

(ii) Manchester line-encoding. [2 marks]

(iii) CRC (Cyclic-Redundancy Check) function. [3 marks]

(iv) Hamming distance. [3 marks]

(b) Digital data may be represented as 1’s and 0’s. On a communications link it may be difficult to differentiate an idle link (consecutive 0’s) from a broken link. A number of schemes are used to indicate that the link is idle but functioning; two examples include data scramblers and block codecs.

(i) Compare and contrast block codecs with scramblers, taking care to describe each fully. [4 marks]

(ii) We wish to achieve an encoded data rate of 1Gbps. Compute the required (symbol) line bit rate for a block codec (e.g. the 8b/10b block codec) and a fixed-length scrambler (e.g. as used in the 64b/66b codec) to achieve a 1Gbps data rate. Which method is more efficient? [2 marks]

(iii) Give an example where the 8b/10b block codec would be more desirable than the 64b/66b codec. [2 marks]

(iv) Scramblers such as that used in the 64b/66b codec are sometimes referred to as self-synchronising. Describe, using an example if required, what is meant by self-synchronising. [3 marks]
Digital Communication I

When Skype establishes an audio channel for telephony calls, it can do so in three ways:

- Direct connection, using UDP.
- Indirect connection, using UDP relayed via a Supernode.
- Indirect connection, using TCP to reach a Supernode, then UDP from there to the destination.

(a) Why does Skype provide these three modes? [2 + 2 + 2 marks]

(b) Describe the different audio problems you might encounter when the first and last modes are used. [8 marks]

(c) Which mode will normally provide the best audio experience? Why? [2 marks]

(d) Suggest two further techniques that an Internet telephony application such as Skype can use to minimise the effects of packet loss. Discuss their relative merits. [2 + 2 marks]
Digital Communication I

(a) The diagram below shows an abstraction of the modules involved in processing an incoming packet on an Internet host.

Explain how these modules process the header fields in the incoming packet so that the data is delivered to the correct process. [6 marks]

(b) The Transmission Control Protocol (TCP) utilises a 3-way handshake at the start of a connection. Explain, with reference to sequence numbers, how this operates and the purpose of the third packet in this exchange. [8 marks]

(c) What is meant by a TCP port? Make reference to how ports are used at client and server when a web browser opens a TCP connection of a web server. [6 marks]
Computer Networking

The popular press suggest that the Internet is a great success.

Based on the range of topics covered in the Computer Networking course, critique the technological success and failure of the Internet. Assertions alone will not constitute an answer to the question: please supply evidence by examples.

[20 marks]
Computer Networking

Consider two physically-separated entities A and B. B has been supplied messages that will be sent to A following these conventions:

- A gets a request from the layer above to retrieve the next data (D) message from B.
- A must send a request (R) message to B on the A-to-B channel.
- Upon receipt of an R, B will send D back to A on the B-to-A channel.
- A should deliver exactly one copy of each D message to the layer above.
- R messages may be lost (but will not be corrupted) in the A-to-B channel.
- D messages are always delivered correctly (no loss or corruption).
- The delay along each channel is unknown and variable.

Give the FSM describing a protocol employed by A and B.

This FSM must compensate for the loss-prone channel between A and B, as well as implementing message passing to the layer above at entity A. Your FSM must not use more mechanisms than is necessary.

[20 marks]
Computer Networking

(a) The diagram below shows a TCP connection between Hosts $H_A$ and $H_B$ passing through networks with different values of Maximum Transmission Unit (MTU) shown. Version 4 of the Internet Protocol (IPv4) is in use.

```
HA  MTU 4000 bytes  MTU 1500 bytes  MTU 4000 bytes  HB
```

$H_A$ chooses a TCP segment size of 3000 bytes of data (TCP and IP headers are each 20 bytes in size).

(i) Describe the way in which fragmentation takes place as $H_A$ sends data to $H_B$. Include the arithmetic used to calculate fragment sizes. Explain the saving that may be made by $H_A$ choosing an optimal TCP segment size when sending 60KBytes of data. [8 marks]

(ii) Briefly explain how the situation described in part (i) would be handled if Internet Protocol version 6 (IPv6) were used. [2 marks]

(b) The formulae below are used in TCP implementations to compute a value for the retransmission time-out ($R$). $R$ is an estimate of the round-trip time, $M$ is the most recently measured round-trip measurement, $\alpha = 0.875$ and $h = 0.25$.

\[
\begin{align*}
D &\leftarrow D + h(|M - R| - D) \\
R &\leftarrow \alpha R + (1 - \alpha)M \\
R &\leftarrow R + 4D
\end{align*}
\]

(i) How is $M$ measured? [2 marks]

(ii) Explain the principles behind the design of these formulae and the values $h$, $\alpha$ and $D$. [8 marks]
4 Computer Networking (AWM)

(a) In a data-center context, describe a straggler using two examples. [2 marks]

(b) (i) Describe the TCP incast problem. [2 marks]

(ii) Outline and critique a solution to the TCP incast problem. [3 marks]

(c) (i) Show that to achieve a steady-state throughput of 10 Gbps, a TCP session with a Round-Trip-Time (RTT) of 100 ms and a Maximum-Segment-Size (MSS) of 1500 bytes can tolerate a packet loss probability of less than $2 \times 10^{-10}$. [4 marks]

(ii) Compute the potential packet-memory requirement of either end-system implementing Selective-Acknowledgements (SACK). [3 marks]

(iii) What is the tolerable packet loss probability if this same network (same MSS and RTT) operated at 100 Gbps? [2 marks]

(d) Some experts say: “Many TCP transactions in the Internet never enter congestion-avoidance.” Discuss this claim.

[Hint: It has been measured that greater than 90% of web objects are less than 10 Kbytes.] [4 marks]
5 Computer Networking (AWM)

(a) (i) Define the terms capacity and latency as applied to communication channels; explaining whether there is a strict relationship between the capacity of a channel and its latency. [3 marks]

(ii) Using a clear example explain how the latency of a channel can have a direct effect on the capacity of a higher-layer channel which uses it. [8 marks]

(iii) Describe how the capacity of the higher-layer channel may be improved, without any change to the characteristics of the underlying channel. [3 marks]

(iv) Describe in what circumstances such changes would provide only limited benefit. [2 marks]

(b) The figure below illustrates, for a single TCP connection, changes in the advertised congestion window (CWND).

![Congestion Window (CWND)](image)

(i) Indicate which phase of congestion control the TCP connection is in at G.

(ii) Indicate which phase of congestion control the TCP connection is in at E.

(iii) Describe the event that has occurred at D.

(iv) Describe the event that has occurred at F. [1 mark each]
6 Computer Networking (AWM)

(a) Consider the host mine.ja.net, with a local DNS server dns1.ja.net. [Note: dns1.ja.net is configured to use recursive DNS by default.]

(i) Host mine.ja.net asks server dns1.ja.net to resolve the hostname yours.foobar.com. Assume there are no cached entries relevant to this request. Write down the steps taken to resolve yours.foobar.com and respond to mine.ja.net. [4 marks]

(ii) Describe the differences between this solution and one achieved using iterative DNS. [2 marks]

(iii) Compare and contrast DNS with ARP. [4 marks]

(b) An office has an (Internet) access link rated at 10 Mbps full-duplex. Each user requires 1 Mbps when transmitting and each user is active 10% of the time.

(i) Initially a static allocation of bandwidth is made for each user. How many users can the access link support? [1 mark]

(ii) The office opts for a pure packet-switched access link. What is the probability that a given user is transmitting? [1 mark]

(iii) The office supports 35 users on the packet-switched access link. What is the probability that exactly \( n \) users are transmitting simultaneously? [2 marks]

(iv) Find the probability that there are 11 or more users transmitting simultaneously. [3 marks]

(v) Describe an assumption about the nature of the traffic that underlies the answer to part (b)(iv) and give two examples of network traffic where this assumption is not valid. [3 marks]
4 Computer Networking (AWM)

(a) A spy elects to use a self-synchronizing scrambler to encode his secret message. Explain why this will not give him any privacy and why his self-synchronising approach would be better used by a communications engineer. [5 marks]

(b) With the assistance of annotated diagrams explain CSMA/CD and CSMA/CA. In your explanation, note the physical constraints on packets and networks that these approaches impose. [10 marks]

(c) Consider the network buffer sizing formula $B = 2T \times C$

(i) Explain this formula. [2 marks]

(ii) Discuss the network architecture and traffic assumptions made in the use of this formula. [3 marks]
5 Computer Networking (AWM)

Here are four options for improving web page performance.

Option 1: HTTP Caching with a Forward Proxy

Option 2: CDN using DNS

Option 3: CDN using anycast

Option 4: CDN based on rewriting HTML URLs

You have been asked to help reduce the costs for networking in the University.

(a) The University pays its service provider networks’r’us, based on the bandwidth it uses; bandwidth use is dominated by students downloading external web pages. Which, if any, of the above four options would reduce the bandwidth usage?

Explain your choice. [4 marks]

(b) The delivery of online courses has become a tremendous success – but this has led to a significant increase in network costs for the University.

You must select one of the options above to minimize load on the servers. Compare the operation of each option and justify a selection that provides the finest granularity of control over load to the content-servers and a selection that will serve each customer from the closest CDN server. [12 marks]

(c) You have looked up the IP address of your favourite search engine on the University network and noticed the answer is different from that given to your friend when he did the lookup in Newfoundland, Canada.

For each option above, indicate why it might, or might not, be used by your favourite search engine to improve web page performance. [4 marks]
6 Computer Networking (AWM)

(a) Considering either TCP/IP or UDP/IP, write a description of how server-port, client-port, source-port and destination-port relate to each other. You may wish to give examples and use diagrams as appropriate. [4 marks]

(b) What is a routing-loop? Include a diagram in your answer. [4 marks]

(c) Describe a mechanism that prevents routing-loops in Ethernet networks. [4 marks]

(d) (i) Describe and, with the aid of an example, illustrate the IP Time-To-Live (TTL) mechanism for minimising the impact of routing-loops. [2 marks]

(ii) Assuming, in part (d)(i), a perfect implementation, describe a disadvantage of the approach including the symptoms that might be experienced in a network subject to this disadvantage, and a test that may identify the problem. [2 marks]

(e) Explain the technical and architectural argument behind the decision in IPv6 to retain header TTL but not a header checksum. [2 marks]

(f) Explain why there is ambiguity about handling packets with TTL values of 1 and give a practical solution. [2 marks]
4 Computer Networking (AWM)

(a) What is the difference between routing and forwarding? [2 marks]

(b) Routing algorithms can be either link-state or distance-vector. Define these two terms and explain the trade-offs between them. [6 marks]

(c) You are required to design a topology discovery protocol for a network of switching nodes interconnected by links. There are $n$ nodes, $l$ links, the maximum degree of any node is $k$ and there is a path between any two nodes of not more than $d$ hops. All links are bi-directional.

Each node has a unique identifier of four bytes which it knows.

(i) Describe a protocol for a node to learn about its immediate neighbours. You should specify the format of your messages and the size of any message fields. [4 marks]

(ii) Using the characteristics of the network described above, design a protocol for distributing this information across the network. You should specify the format of your messages and the size of any message fields. [8 marks]
(a) Below is an excerpt from the DNS record for a fictitious corporation, Lemon:

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Value</th>
<th>TTL (seconds)</th>
</tr>
</thead>
<tbody>
<tr>
<td>lemon.co.uk</td>
<td>A</td>
<td>91.45.20.24</td>
<td>86400</td>
</tr>
<tr>
<td>lemon.co.uk</td>
<td>NS</td>
<td>grove.lemon.co.uk</td>
<td>86400</td>
</tr>
<tr>
<td>lemon.co.uk</td>
<td>NS</td>
<td>tree.lemon.co.uk</td>
<td>86400</td>
</tr>
<tr>
<td>lemon.co.uk</td>
<td>MX</td>
<td>stem.lemon.co.uk</td>
<td>60</td>
</tr>
<tr>
<td>grove.lemon.co.uk</td>
<td>A</td>
<td>91.45.23.22</td>
<td>86400</td>
</tr>
<tr>
<td>tree.lemon.co.uk</td>
<td>A</td>
<td>91.45.23.23</td>
<td>86400</td>
</tr>
<tr>
<td>orchard.lemon.co.uk</td>
<td>A</td>
<td>91.45.23.82</td>
<td>86400</td>
</tr>
<tr>
<td>stem.lemon.co.uk</td>
<td>A</td>
<td>91.45.23.85</td>
<td>86400</td>
</tr>
<tr>
<td><a href="http://www.lemon.co.uk">www.lemon.co.uk</a></td>
<td>CNAME</td>
<td>orchard.lemon.co.uk</td>
<td>86400</td>
</tr>
</tbody>
</table>

(i) If you type http://www.lemon.co.uk into your web browser, to which IP address will your web browser connect? [1 mark]

(ii) If you send email to support@lemon.co.uk, to which IP address will the message get delivered? [1 mark]

(iii) The TTL field refers to the maximum amount of time a DNS server can cache the record. Most of the TTLs in this record were chosen to be 86400 seconds (1 day). What is the trade-off between choosing a shorter or a longer time? Why was the MX record specifically chosen to have a 60 second TTL? [4 marks]

(iv) Explain why the Internet DNS uses caching. [2 marks]

(v) Comment on how the provision of name servers for lemon.co.uk affects the availability of the name service. [2 marks]

(vi) Outline two strategies to improve availability of the DNS server for the lemon.co.uk domain. [2 marks]
(b) Consider the scenario shown above. Host A is sending tiny packets to hosts B and C. R is a store-and-forward switch with an average arrival rate of 10Gb/s and a buffer that contains, on average, 8MBytes of packet data. Delays due to the packet size and packet-processing are negligible.

Little’s Law tells us that the average amount of buffered data equals the product of the arrival rate and the average delay experienced.

(i) What is the average delay that packets will incur going through the switch? [3 marks]

(ii) Compute the latency of the shortest path between each pair of end-nodes: A to B, A to C, and C to B. [3 marks]

(iii) Without changing the network propose a solution to decrease the delay between A and B. [2 marks]
6 Computer Networking (AWM)

(a) Consider an unreliable message service where messages of a fixed size are sent between known endpoints. Outline the minimum set of additional features offered by a reliable byte-stream delivery service. [3 marks]

(b) A researcher notes that the message service, fritter, resembles a datagram service. It is prone to delivery delays of up to 1 second, message re-ordering and message loss. Fritter permits a 140-byte message to be relayed between any two users and each message is delivered without data-corruption.

You are asked to implement a Stop-and-wait ARQ to provide a unidirectional reliable byte-stream delivery service between two fritter users. Assume this is the only service between the two fritter users.

(i) Provide a labelled diagram illustrating the format for a fritter message that could be used by a reliable, byte-stream, delivery service. Justify your answer. [3 marks]

(ii) Draw and label the Finite State Machine that implements the sender portion of the Stop-and-wait ARQ. Your function will be called as reliable_send() while the fritter message receive and message send functions are fritter_recv() and fritter_send() respectively. You may assume that the argument to the reliable_send() function does not exceed 100 bytes per function call. [8 marks]

(iii) Users assert that the performance using your Stop-and-wait ARQ is terrible for large transfers. Explain why they are correct. [2 marks]

(iv) Describe an appropriate enhancement to the ARQ that will improve performance. Given the constraints of a small fritter message size, justify why your particular ARQ enhancement is best suited to the fritter application. [4 marks]
The following equation provides a simple way to estimate the throughput of a TCP connection, as a function of the loss probability $p$, the round-trip time RTT, and the maximum segment size MSS.

$$\text{TCP Throughput} = \sqrt{\frac{3}{2}} \frac{\text{MSS}}{\text{RTT} \sqrt{p}}$$

(a) Alice wants to send a large amount of data to Bob over a network path with RTT = 100 ms, $p = 0.01$, and MSS = 10,000 bits. What is the expected throughput in Mbit/s? [2 marks]

(b) With the aid of a clearly labelled diagram showing window-size versus time, derive the above equation. [10 marks]

(c) Alice has two options to improve the throughput: halving either the RTT or the loss probability $p$. If both cost the same, which is more cost effective and why? [2 marks]

(d) Consider your derivation of the equation in part (b). State three assumptions that are made and describe when these assumptions may not hold in reality. [6 marks]
5 Computer Networking (AWM)

(a) In the above network, use Dijkstra’s shortest-path algorithm to compute the shortest path from E to all network nodes. Show your working in a table: each column indicating a destination node, each row indicating an iteration of the algorithm. [10 marks]

(b) In 2008 an ISP was reported to have hijacked traffic for YouTube causing traffic for YouTube to be diverted into the ISP’s network.

At the time, YouTube used only three IP addresses; 208.65.153.238, 208.65.153.251 and 208.65.153.253, announced as a single prefix 208.65.152.0/22.

Despite YouTube using only three addresses, each browser’s YouTube URL requests are ultimately routed to the closest of over a dozen data-centres Google operates world-wide.

(i) Describe two concepts from the course that make this possible. [2 marks]

(ii) State the smallest advertised netblock that would identify all YouTube addresses. [1 mark]

(iii) In an attempt to resolve the problem, YouTube advertised the netblock 208.65.153.0/24, but this was the same netblock as advertised by the rogue ISP. Why would this not solve the problem? [2 marks]

(iv) YouTube advertised two smaller netblocks, each one half of 208.65.153.0/24. Why should this now work? [2 marks]

(v) BGP networks may optionally filter netblocks that are below a given size. This filtering affected the YouTube fix in (b)(iv), but not that in (b)(iii). Estimate the size of the netblock filter. [1 mark]

(vi) Why does BGP implement such filtering? [2 marks]
6 Computer Networking (AWM)

(a) An older home-network router has an upload bandwidth of 1Mbit/s to the Internet, and a 100kbyte first-in first-out (FIFO) buffer for packets awaiting transmission. Packets have a maximum transmission unit (MTU) of 1500 bytes.

(i) If the buffer is completely full, how long does it take the router to transmit all of the bytes in the buffer? [2 marks]

(ii) Suppose the router supports two FIFO queues, one high-priority for interactive applications (like Voice over IP) and the other lower-priority for all remaining traffic. If a VoIP packet arrives when the queue for interactive applications is empty, what is the maximum time before the router starts transmitting the VoIP packet? (Assume that the router does not preempt any ongoing packet transmission.) [2 marks]

(b) Consider the delay at each node (router or switch) in a network given by this equation:

\[ d_{\text{node}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}} \]

For each term: \(d_{\text{proc}}, d_{\text{queue}}, d_{\text{trans}}, \text{and } d_{\text{prop}} \)

(i) explain what it represents;

(ii) state one way to reduce it; and

(iii) indicate a typical range of values in a 1Gbit/s local area network with link-length less than 500m.

State your assumptions throughout. [4 \times 3 marks]

(c) A lecturer remarks that “centralised multiplexing” offers potential gains in efficiency over non-centralised multiplexing.

Give two reasons why this could be so. In each, state clearly what feature must be centralised to achieve these gains. [4 marks]
4 Computer Networking (AWM)

Two objects are being retrieved from B by A using HTTP over a network where the TCP-style transport protocol has no maximum segment size and the network experiences no loss. Each object is 125 kByte (i.e., one Mbit) in size. We assume all connection setup packets and HTTP request packets are negligible in size; we ignore the connection tear-down.

(a) Suppose the Round-Trip-Time (RTT) between A and B is 10 milliseconds and the bandwidth between the sites is 10 Mbit/s. Illustrate with a simple diagram in each case how long it takes for A to retrieve both files under the following circumstances: [2 marks each]

(i) Sequential (one-at-a-time) requests with non-persistent connections.
(ii) Concurrent requests with non-persistent connections.
(iii) Sequential requests within a single persistent connection.
(iv) Pipelined requests within a single persistent connection.

(b) Suppose A is connected to a cache C by a link with 10 Gbit/s bandwidth and negligible RTT. C connects to B with the link originally specified in part (a): 10 Mbit/s and 10 millisecond RTT.

The cache operates as follows:

- If the object is not in the cache, the request is forwarded to B which responds with the object, which the cache stores with an object timeout and then forwards to A.
- If the object is in the cache, and the cache entry has not timed out (i.e., the cache Time To Live (TTL) has not expired), the object is returned to the client.
- If the object is in the cache, but the cache entry has timed out, the cache issues a conditional GET to the original server, asking if the object has changed since this object was cached. If the original server responds that it has not, the cache returns the cached object, otherwise the original server responds with the updated object which the cache forwards to the client.

Showing your working, how long does it take for A to retrieve one file under the following circumstances: [3 marks each]

(i) The file is not in the cache
(ii) The file is in the cache and the TTL has not expired
(iii) The file is in the cache, the TTL has expired, but the file has not changed
(iv) The file is in the cache, the TTL has expired, and the file has changed
5 Computer Networking (AWM)

(a) This question concerns IPv6 notation. [2 marks]

(i) Rewrite the following IPv6 address in the most compact form.
    3ffe:1944:0000:0000:0010:0000:0000:d00b

(ii) In IPv6 notation, what do ::/0 and ::1/128 represent?

(b) Describe five differences between IPv4 and IPv6. [5 marks]

(c) Describe, giving reasons, how an Internet Service Provider (ISP) may use
    their allocated IPv6 addresses to provide address blocks to their (residential)
    customers. [3 marks]

(d) What are the implications of IPv6 for the Network Address Translation (NAT)
    gateways and firewalls of residential customers? [4 marks]

(e) Give one advantage and one disadvantage that Stateless Address Autoconfigura-
    tion (SLAAC) has compared to DHCPv4. [2 marks]

(f) Suggest two features that would be desirable extensions of IPv6. Justify your
    answer. [4 marks]
6 Computer Networking (AWM)

(a) Consider packet switching and circuit switching. In simple terms, packet switching may allow more users to use the network.

We have \( N \) users sharing a 1 Mbit/s link. Each user consumes 100 kbit/s when transmitting but only transmits for 10% of the time, on average.

(i) How many users would a circuit-switching system support? State your assumptions. [1 mark]

(ii) For \( N = 40 \) users, the probability that more than 10 users are active at the same time is approximately 0.0015.

Show how to compute this probability. You are not required to calculate the actual figures. [6 marks]

(iii) State two assumptions that you made in Part (a)(ii) about the network users. In each case describe why the assumption may fail for current Internet traffic. [4 marks each]

(b) The formulae below are used in TCP implementations to compute a value for the retransmission time-out \( T \). \( R \) is an estimate of the round-trip time (RTT), \( D \) is an estimate of variance, \( M \) is the most recently measured round-trip measurement, \( \alpha = 0.875 \) and \( h = 0.25 \).

\[
\begin{align*}
D & \leftarrow D + h(|M - R| - D) \\
R & \leftarrow \alpha R + (1 - \alpha)M \\
T & = R + 4D
\end{align*}
\]

(i) Give an example of how the retransmission time-out \( T \) is used within TCP. [1 mark]

(ii) Describe why the computation of the retransmission time-out \( T \) incorporates a correction for deviation in the estimate of the RTT. [2 marks]

(iii) For each assumption you stated in Part (a)(iii), describe the impact on the estimate of the retransmission time-out \( T \). [3 marks each]
4 Computer Networking (AWM)

(a) Two network-architecture approaches are commonplace: hop-by-hop, and end-to-end.

Briefly compare these two approaches with reference to the encryption of a web-page and the use of WPA2 (or similar schemes) to secure WiFi. [4 marks]

(b) Consider four components that constitute delay for a packet network: queueing delay, processing delay, propagation delay, and transmission delay.

(i) Order these delays by magnitude, giving typical values for each and making clear your justification. [8 marks]

(ii) (A) Describe circumstances where processing delay for one packet type varies significantly from the mean processing delay of a packet. [2 marks]

(B) Estimate what such a difference in delay might be. State your assumptions and show your working. [2 marks]

(iii) Packet-network delays of a typical datacentre may vary significantly from those found in the Internet.

Discuss why this should be the case, with particular reference to the Bandwidth-Delay product and discuss the implications for network architectures in datacentres where we wish to minimise delay. [4 marks]
5 Computer Networking (AWM)

(a) A router vendor is interested in building an ultra low-cost router. Analysis reveals that the total build-costs are 90% for linecards and 10% for control processors. The router vendor focuses on reducing linecard costs.

Discuss which of the following strategies would help the vendor achieve this goal. Keep in mind that high-speed memory is expensive.

(i) Develop a faster implementation of Dijkstra’s algorithm. [2 marks]

(ii) Convince the IETF to eliminate IPv4 fragmentation support. [2 marks]

(iii) Convince the IETF to abolish multi-homing. [2 marks]

(iv) Convince operating systems vendors to implement packet “pacing” so end hosts will not generate bursts of back-to-back packets. [2 marks]

(b) You are running a TCP-based movie streaming service called MeTube. Several of your users are complaining about poor performance. You call in your team of three engineers to examine the problem. They all observe that the 10Gbps access link to the MeTube server is only 25% utilised, and that the server’s CPU is only 15% utilised. Each engineer then independently reports back with the following conclusions and suggestions for improvements.

Consider each of the engineers’ conclusions and discuss why it is correct or not.

(i) Engineer Phil notices that a number of packets are being retransmitted multiple times. Phil recommends the TCP implementation be modified reducing both the timeout period and the fast retransmit threshold from 3 to 2 dupAcks (duplicate Acknowledgments). While this may lead to even more retransmissions, Phil concludes this is OK as the server’s access link is only 25% utilised. With these changes, users will receive lost packets faster and hence will definitely see improved performance. [4 marks]

(ii) Engineer Leslie runs traceroute from the MeTube server to impacted users and finds that a router R1 along the path repeatedly drops her traceroute messages. Leslie concludes that the high packet drop rates at R1 are the cause of their performance problems. [4 marks]

(iii) Engineer Chris notes that users reporting problems have IP addresses belonging to the Classless Inter-Domain Routing (CIDR) block allocated to Tiny-Telco. She believes the problem may be due to high loss rates involving Tiny-Telco’s network. Chris recommends they contact Tiny-Telco and ask them to diagnose and fix the problem. [4 marks]
6 Computer Networking (AWM)

(a) Describe on-off flow control. In what circumstances is it appropriate? [4 marks]

(b) Describe the operation of window-based flow control. [4 marks]

(c) What happens if a flow using window-based flow control passes through a highly-loaded switch or router? You may assume the switch or router does not participate in the flow control protocol. [4 marks]

(d) How is this addressed in the TCP protocol? [4 marks]

(e) What are the advantages and disadvantages of having Internet routers participate in window-based flow control of every TCP connection? [4 marks]
Computer Networking (EK)

(a) Describe two drawbacks of layering. Provide an example for each. [4 marks]

(b) (i) Explain the single-bit parity error-detection code using a single byte of data. How many bit errors can this code detect?

(ii) Based on the single-bit parity error-detection code devise a new code to detect and correct a single 1-bit error in 4 bytes of data. How many parity bits do you require? You may assume that parity bits are error-free. [8 marks]

(c) Consider a wireless network. For each of the following cases, state whether the packet transmission would be successful; assume no collision avoidance. Explain your answers.

(i) Nodes A and B are in range of each other; no other node is within range. Node A sends a packet to B.

(ii) Nodes A and B are in range of each other; nodes B and C are in range of each other; A and C are not in range of each other. Both A and C send a packet to B simultaneously.

(iii) Nodes A and B are in range of each other; nodes B and C are in range of each other; A and C are not in range of each other. C is transmitting and A wants to send a packet to B.

(iv) Nodes A and B are in range of each other; nodes B and C are in range of each other; A and C are not in range of each other. A is transmitting and B wants to send a packet to C. [8 marks]
5 Computer Networking (EK)

(a) Consider two clusters A and B each hosting multiple applications. All applications send bursty traffic between A and B over a link E. Under what conditions is circuit switching more efficient to use as opposed to packet switching? [2 marks]

(b) Compare the link state and distance-vector protocols in terms of message complexity, processing complexity and robustness. [6 marks]

(c) Cambridge University is about to open a new School with three new departments A, B and C. The IPv4 address prefix of the new School is 128.232.1.0/24 and it is expecting each department to have the following number of hosts:

Department A: between 40 and 60 hosts
Department B: between 100 and 120 hosts
Department C: between 20 and 30 hosts

(i) The university wishes to allocate a subnet for each department. Give possible IPv4 subnet masks for each new department. [3 marks]

(ii) Later, the School opens a fourth department D with 30 hosts. Provide possible IPv4 subnet masks to accommodate all four departments. [2 marks]

(iii) Finally, the School opens a fifth department E of similar size to B. Provide possible IPv4 subnet masks to accommodate all five departments. [4 marks]

(iv) Are there any practical problems with your answer to Part (c)(iii)? Briefly discuss an alternative solution to accommodate all five departments. [3 marks]
6 Computer Networking (EK)

(a) Briefly describe the main differences and similarities between routers and switches. [4 marks]

(b) Consider the network shown in the figure below with four nodes. Cost links are shown in the diagram. Give the distance-vector routing tables for node C in the following two consecutive steps.

(i) Step 0: C knows the distances to its immediate neighbours and [1 mark]

(ii) Step 1: information from step 0 is exchanged as per the distance-vector algorithm. [3 marks]

(c) (i) What is the problem that the Karn-Partridge algorithm aims to solve? [2 marks]

(ii) Karn and Partridge proposed the use of exponential backoff in the TCP timeout value. Why is it that an exponential increase in the timeout value is more efficient than a linear increase for example? [2 marks]

(d) What is the difference between congestion control and flow control in TCP? [2 marks]

(e) Assume that three TCP flows \( f_1, f_2, f_3 \), share a single link. The bandwidth of the link is 200 Mbps. The desired bandwidth of each flow is: \( f_1 \) 60 Mbps, \( f_2 \) 80 Mbps, and \( f_3 \) 100 Mbps.

(i) What would be the max-min bandwidth allocation of the flows? [2 marks]

(ii) Propose a way to “cheat” max-min fairness so that \( f_3 \) increases its allocated bandwidth as computed in Part (e)(i). [4 marks]
4 Computer Networking (awm22)

The TCP transport protocol is an example of an ARQ.

(a) Provide a definition of an ARQ. [1 mark]

(b) Describe the design and operation of a simple ARQ for a lossy communication channel. [3 marks]

(c) When and why does a simple ARQ at the transport layer have a significant negative impact upon application performance? [3 marks]

(d) Describe what additions are required to a simple ARQ to support windowing. When and why will an ARQ which supports windowing provide better application performance than a simple ARQ? [5 marks]

(e) Describe two situations when the performance of a windowed ARQ is no better than a simple ARQ. [4 marks]

(f) Would the QUIC protocol, when based upon UDP rather than TCP, overcome the two situations you outlined when answering Part (e)? [4 marks]
5 Computer Networking (awm22)

(a) You need to urgently deliver 500 TByte of data from Zurich to London.

(i) NoWayNet is offering a 10 Gbit/s reliable delivery service between Zurich and London (about 776km). Should you use either NoWayNet, or an overnight package delivery? Why? [2 marks]

(ii) A new company, UnlikelyComms, is offering a 400 Gbit/s reliable delivery service between Zurich and London, but it takes a very indirect 2600 km path. Should you use UnlikelyComms? [2 marks]

(iii) Following your successful urgent delivery of 500 TBytes of data, this has become an hourly task. Alongside the need to regularly deliver 500 TByte data between Zurich and London, you have an interactive virtual reality system; it requires six displays each needing 50 Gbit/s and an end-to-end latency of less than 5ms.

Fortunately, a startup, FlyByNight, boasts an offering with an effective bandwidth of over 16 Tbit/s, using a special transport with no end-to-end latency at all. The downside is it can only transfer data in 500 TByte units, once every 4 minutes. Explain whether FlyByNight is, or is not, suited to your two workloads. [4 marks]

(b) Ethernet standards enable 1Gbit/s over 4 pairs of twisted cabling, yet the physical media has a bandwidth much less than 1GHz, e.g., 250MHz is common.

(i) Explain how such high data rates are achieved, and [4 marks]

(ii) explain how physical media errors are reduced or eliminated. [2 marks]

(c) Explain, with the aid of diagrams, how Code-Division Multiple Access permits two or more pairs of nodes to communicate over a common medium (e.g., wireless) simultaneously. [6 marks]
6 Computer Networking (awm22)

(a) Provide five examples of resource-multiplexing in computer networks. In each case state the layer of the network stack where this takes place, which resource is multiplexed and the multiplexing mechanism. [2 marks each]

(b) A relative has contacted you for help; they believe the Internet is broken. The fault turns out to be that a piece of network-protection software had installed a specific IP address entry for an (alternative) default DNS server.

(i) What should have provided the correct DNS entry? [1 mark]

(ii) By concentrating on how DNS is intended to work, describe why network applications may not work correctly. [3 marks]

(iii) How would you diagnose this problem? [2 marks]

(c) DNS substitution may also be used maliciously.

(i) Outline how spyware might use an (alternative) DNS entry. [2 marks]

(ii) Describe a network-centric method for overcoming such treachery. [2 marks]
The network illustrated above represents an Ethernet Layer-2 network that uses spanning-tree to compute forwarding tables. Assume all links have a link-weight of one.

[Note: Tie-breaking/leader-elections use the switch identifier from this diagram.]

(a) Compute the steady state routing/forwarding table for Switch 3.

[4 marks]

(b) Noting which switches recompute a solution, enumerate the changed forwarding tables in switches of this network resulting from the complete failure and removal of link D.

[9 marks]

(c) Following the removal of Link D, a new link H is added between Switch 1 and 4; however, this link fails frequently.

Denied access to monitor the network-traffic, outline a diagnostic strategy to identify the faulty Link H, making clear how the network-operator might use interrogation of network switch forwarding tables.

[4 marks]

(d) Now suppose the switches do not permit interrogation of switch forwarding tables and that the link status information is untrustworthy. Outline other techniques that might be used to identify the failed link.

[3 marks]
5 Computer Networking (awm22)

(a) Using a diagram, illustrate how long a packet of length \(L\) bytes will take to travel over an idle network from Host 4 to Host 1? The routers use a store-and-forward architecture.

(b) Computers in Calm building are often not getting allocated IP addresses and the performance is quite poor. The department router serves DHCP for the College network and is operating correctly. Residents in Blue report intermittent performance issues, but no one in Admin reports any problems. Network measurements reveal that the per-router packet loss for each switch under load can be as high as one packet in five thousand, but it is significantly worse for packets smaller than 1000 bytes, where as many as one packet in twenty are lost.

With these insights, explain the cause of the problems experienced. Make clear any simplifying assumptions you have made.

(c) Some students in Calm have found using IPv6 will ‘work’ (i.e., connecting to the wider University services is possible, but not to Internet services); although still not performing as well as when they are in the Admin building. Describe the steps by which IPv6 addresses are allocated without DHCP and consider why this service may be working more reliably than IPv4?

(d) Two approaches to improve the network performance are available: one is to upgrade the performance of the physical links between the buildings to 10Gbit/s. The alternative approach is to significantly change the topology of the network by adding an additional high performance router, but leaving the performance of the physical links unchanged.

Briefly give the advantages and disadvantages of each approach.
6 Computer Networking (awm22)

(a) Arriving at your college room, you plug into the wired Ethernet jack for the first time. The network admin has a record of your MAC address and your machine can join the network without further action on your part.

Assume: Your laptop’s Ethernet address is 0a:0b:0c:0d:0e:0f, DHCP server address is 131.111.7.3, your IPv4 address will be 131.111.7.121, the gateway’s IP address is 131.111.7.1, and Ethernet address is 00:01:02:03:04:05, the network netmask is 255.255.255.0

Write the series of protocol/packet exchanges that occur on the wired Ethernet link, up until you can send a single packet to 128.232.0.20. You do not need to describe packets after this packet has left the link. Include ARP and DHCP packets, stating the IP and Ethernet addresses of the packets where possible.

[10 marks]

(b) Consider two neighbours, Alice and Bob. Each have wireless IPv4 routers with integrated NAT. Each neighbour connects their laptop to their own wireless router, and each uses appropriate utilities to examine the IP address of each laptop. They realise the laptops have the same IP address.

(i) How is that possible? [2 marks]

(ii) Justify one reason that widespread deployment of IPv6 would remove the need for the NAT devices. [2 marks]

(iii) Justify one reason that an IPv6 user might want to continue using their NAT. [2 marks]

(iv) Further investigations show the two wireless routers are using the same wireless channel, although with different SSIDs. Detail what this situation means, why this situation is both possible and perfectly standards-compliant behaviour, and what implications there are for this situation — including how any negative effects can be made less severe. [4 marks]
4 Computer Networking (awm22)

(a) Consider the layer-2 switch topology shown below.

Assuming all switches start with empty switch-forwarding tables; Host X (with physical address X) sends a packet destined for Host Z. Enumerate in the style below, all packets sent across the network until the message arrives at Host Z. You may assume packet-processing, latency and transmission time are negligible. Additionally, indicate packets transmitted simultaneously. [5 marks]

\[
\begin{array}{c|c|c|c}
\text{Time Step} & \text{Sent by Device} & \text{Sent on Link} & \text{Frame Source} \\
\hline
0 & X & X-A1 & X \\
\vdots & \vdots & \vdots & \vdots \\
? & C & C7-Z & X \\
\end{array}
\]

(b) Enumerate in the style below, the forwarding table of Switch B at the end of Part (a).

\[
\begin{array}{c|c}
\text{Destination} & \text{Port} \\
\hline
\vdots & \vdots \\
\end{array}
\]

[2 marks]

(c) Consider a layer-2 network consisting of S+1 switches, S directly attached to H hosts. Each host runs V virtual machines, each with a single address. S switches are connected using a star topology with a single switch C at the centre. Each host exchanges data with a selected and stable subset of other hosts.

Estimate the worst case number of entries in the forwarding table for any typical switch S and the number of entries in the forwarding table for switch C. [3 marks]

(d) Users of this cluster of machines complain of occasional misbehaviour attributed to network timeouts and network slowdown.

Outline two plausible chains of events causing the problem. Describe two appropriate, cost-effective, strategies for overcoming the issues faced by the users. You may assume the switches are state-of-the-art and buying more hosts is not the answer. [10 marks]
5 Computer Networking (awm22)

(a) A user complains that their web application **Times Out** after successfully connecting to a remote host, yet, an investigation with **ping** indicates the remote host is alive.

Propose a cause of this fault and outline a test-strategy that could be used automatically to detect such a fault. [4 marks]

(b) A user would like to infer the path between two hosts; they suggest using the utility **ping**. Make your argument for why **traceroute** is the more appropriate utility, comparing and contrasting their operation. [4 marks]

(c) Consider the test network below; Y and Z are two unix IPv4 hosts, while nodes A through E represent fully-conformant IPv4 routers providing the only connectivity between the two hosts.

![Network Diagram]

Per-packet load balancing by A and E means packets may be sent on any valid path. No firewalls or packet filtering is used anywhere in the network.

(i) Using **traceroute** infers the following paths:

   Explain (with diagrams as appropriate) why these results have occurred and identify the two true and two false results; [8 marks]

(ii) A firmware upgrade for routers A and E means they now do per-flow and not per-packet load balancing. Despite the functions of the flow-based load balancing, **traceroute** results have not changed; two false paths are inferred along with two true paths.

   Explain this outcome and a strategy to identify only true paths. [4 marks]
6 Computer Networking (awm22)

(a) For IPv4 ISPs, each domestic installation typically gets a /32 network. You have a complicated configuration requiring NAT and multiple IPv4 subnets.

(i) Why would an IPv6 based provider allocate four /64 networks for your premises when each /64 represents $2^{64}$ addresses? [2 marks]

(ii) A colleague has IPv6 with another provider; they only allow one /64 for each domestic installation. In the past your colleague has used a NAT and many IPv4 private address blocks, but keenly adopted IPv6 permitting them to upgrade their home network. They are now using blocks of the allocated /64 and a router in their home to interconnect the subnets.

Not everything is working as they hoped; for example, sometimes IoT devices can’t connect to the Internet to update and your colleague can not connect to their front-door camera when at work.

Explain what sort of problems your colleague may face along with methods by which they could verify the root cause. [6 marks]

(iii) Explain to your colleague why you might not be able to lend them one of your /64 allocations, even though the /64 blocks (provided to you by your ISP) are each globally routable addresses. [2 marks]

(b) A local area network may carry several different LANs simultaneously; such a network would be designated for known sets of HomePlug devices.

Describe a physical line coding approach for the HomePlug devices that: allows two or more simultaneous virtual local area networks to fairly share the same physical channel, but does not permit trivial interception of network traffic.

Outline your approach along with its benefits and drawbacks, comparing it with the simplest use of VLAN tags in Ethernet. [10 marks]
Consider a satellite network consisting of a constellation of $S$ satellites placed at an altitude of 1,000km above the Earth (Earth’s diameter is $\approx 12,750$km) and $G$ ground stations connecting both customers and Internet exchange points. Each orbit of the constellation consists of 50 satellites (20 orbits total). Each satellite is equipped with five communications laser-receivers; one to communicate with the ground and four to communicate with the nearest neighbour satellites. The satellites are not geostationary. You may assume each satellite only communicates with its nearest neighbours at any time. Each node—satellite, or ground station—has a unique identifier of 16 bytes which it knows.

Design a topology-discovery protocol than can identify the shortest path among any two ground stations using the 1,000 satellites in orbit.

Symmetric paths may be presumed.

(a) Outline a protocol (including message formats) for a node to learn about its immediate neighbours. [3 marks]

(b) Design a protocol (including message formats) for distributing this information across the network. [7 marks]

(c) Give a bound on the total amount of non-redundant information which is transmitted to ensure that every node acquires complete topology information. [5 marks]

(d) The channel bandwidth is given as $B$, an approximation of $3 \times 10^8$ meters per second may be used for the speed of light, and per node packet processing time may be considered zero.

(i) Make an estimate of the worst-case total amount of time the exchange of this information will take to propagate across the network. [2 marks]

(ii) Outline one method to improve the time-period. Speculate on the improved upper bound of total time for the information exchange. [3 marks]
2 Computer Networking (awm22)

A networking enthusiast says, “TCP does not perform well for the very long, the very short, the very fast, or the very slow.”

(a) When and why are they correct? Explain your answer clearly. [8 marks]

(b) QUIC is a new transport protocol that offers many advantages over TCP, it may not solve the issues the enthusiast of the last question discusses. Prototyping new protocols to replace TCP has proven difficult in the past. Discuss the QUIC approach to this particular challenge. [2 marks]

(c) Considering the capacity and limits of end-system components (e.g., CPU(s), cache(s), main memory, peripheral interconnects, network adapters), propose a practical strategy for end-systems to deliver the next generation of Ethernet speeds (100Gbit/s and 400Gbit/s).

Alongside your strategy you may wish to also consider the particular challenges invoked by improved security in networking such as the wider use of IP header encryption (IPsec), the increasing use of full application-payload encryption, and so-forth.

Justify your approach by stating your assumptions throughout.

Hint: your approach need not be universal. It may help to consider aspects of Computer Architecture, Algorithms and Computer Networking in your approach. [10 marks]
3 Computer Networking (awm22)

(a) Using an explanation of the difference between flow-control and congestion-control, discuss the impact of a stable end-to-end latency. [5 marks]

(b) Mobile end-points present a challenge for any connection-based application on an IP network. Using an example, discuss how this challenge might be resolved without a complete reworking of the end-system application. [5 marks]

(c) A reverse proxy such as Varnish makes static snapshots of the content of a dynamic webpage. Outline a use-case for using such a reverse proxy, paying particular attention to a case when an otherwise useful system might need disabling. [5 marks]

(d) In 2021 a large social network had to reboot all its systems after systemic failure. Discussing the extensive use of image caching, outline the challenges that face an image serving system in the social media space (that is, one that uses popularity, hashtags, and so-forth). Discuss in particular why, for weeks after the reboot, many users complained of poor performance, and of being served irrelevant or old images.

Hint: A distinguishing feature of social networks has been a heavy-tailed distribution of interest among images and users.

For your information: a large image sharing site has $500 \times 10^6$ active daily users, $500 \times 10^6$ active posts (groups of images and interactions) and $100 \times 10^6$ new images each day. Sizes varies considerably but an average image size approaches 100 Kbytes. [5 marks]