1993 Paper 5 Question 3

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]

1993 Paper 6 Question 3

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]

1994 Paper 5 Question 3

Digital Communication I

Define the term *circuit* as used in "circuit switching". [4 marks]

Sketch the design of the in-band switching function of a circuit switch which switches 4×2 Mbps trunks each supporting 32×64 Kbps channels. [8 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 marks]

1994 Paper 6 Question 3

Digital Communication I

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]

1995 Paper 5 Question 3

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]

1995 Paper 6 Question 3

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]

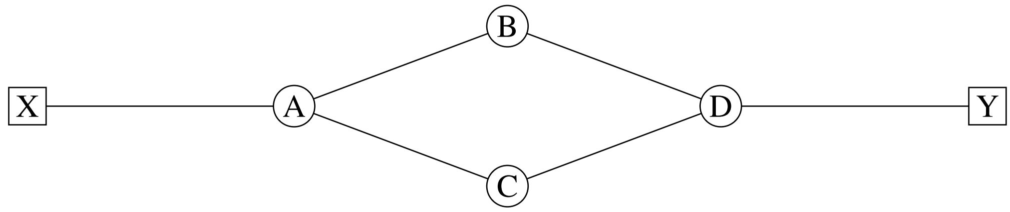
Describe a possible implementation of a signalling system, including a discussion of how signalling information might be carried over the network. [8 marks]

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]

1996 Paper 5 Question 3

Digital Communication I



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]

1996 Paper 6 Question 3

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]

1997 Paper 5 Question 1

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]

1997 Paper 6 Question 3

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]

1998 Paper 5 Question 3

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]

1998 Paper 6 Question 3

Digital Communication I

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]

1999 Paper 5 Question 3

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]

1999 Paper 6 Question 3

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:

Data rate	Error rate	Code rate
10 Mbps	10^{-4}	unity (no coding)
5 Mbps	10^{-5}	half

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]

2000 Paper 5 Question 3

Digital Communication I

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]

2000 Paper 6 Question 3

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]

2001 Paper 5 Question 3

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 ϵ_0.
- The compression ratio is $C < 1$.
- The FEC has rate $R < 1$ and given an error rate ϵ_0 provides an error rate ϵ_1 (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]

2001 Paper 6 Question 3

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]

2002 Paper 5 Question 3

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]

2002 Paper 6 Question 3

Digital Communication I

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]

2003 Paper 5 Question 3

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]

2003 Paper 6 Question 3

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]

2004 Paper 5 Question 3

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]

2004 Paper 6 Question 3

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]

2005 Paper 5 Question 3

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 τ, 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]

2005 Paper 6 Question 3

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]

2006 Paper 5 Question 3

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]

2006 Paper 6 Question 3

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]

2007 Paper 5 Question 3

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]

2007 Paper 6 Question 3

Digital Communication I

(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 τ, 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 $2p$.

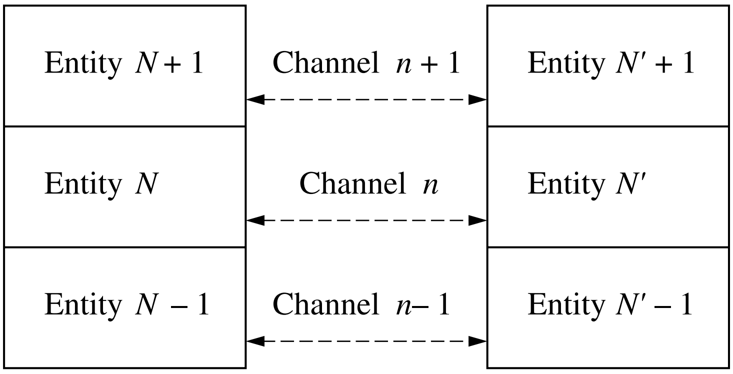
[7 marks]

2008 Paper 5 Question 3

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]

2008 Paper 6 Question 3

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]

2009 Paper 5 Question 7

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]

2009 Paper 5 Question 8

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]

2010 Paper 5 Question 7

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]

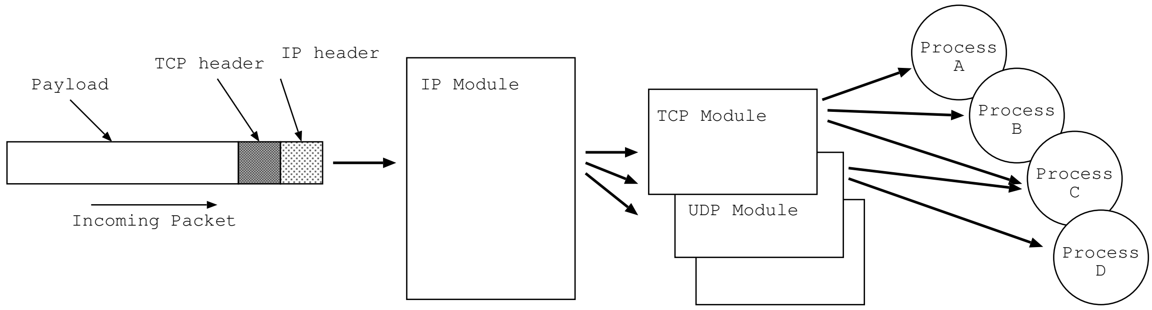
(*c*) Which mode will normally provide the best audio experience? Why? [2 marks]

(*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]

2010 Paper 5 Question 8

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]

2011 Paper 5 Question 4

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]

2011 Paper 5 Question 5

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

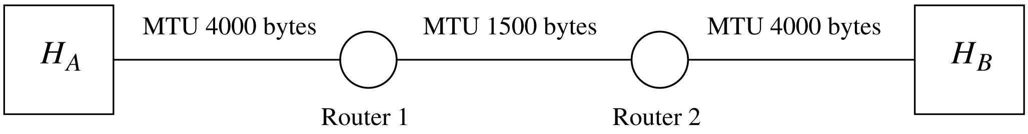
Give the FSM describing a protocol employed by **A** and **B**.

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.



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 ($\mathcal{R}$). R is an estimate of the round-trip time, M is the most recently measured round-trip measurement, $\alpha = 0.875$ and $h = 0.25$.

$$\begin{aligned} D &\leftarrow D + h(|M - R| - D) \\ R &\leftarrow \alpha R + (a - \alpha)M \\ \mathcal{R} &= R + 4D \end{aligned}$$

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

(*ii*) Explain the principles behind the design of these formulae and the values h, α 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 2×10^{-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)

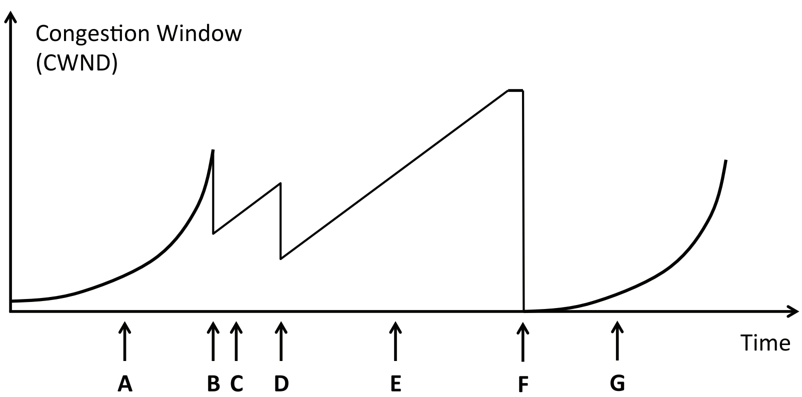
(*a*) (*i*) Define the terms *capacity* and *latency* as applied to communication channels; explaining whether there is a **strict** relationship between the capacity of a channel and its latency. [3 marks]

(*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).



(*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]

5 Computer Networking (AWM)

(*a*) Below is an excerpt from the DNS record for a fictitious corporation, Lemon:

Name	Type	Value	TTL (seconds)
`lemon.co.uk`	A	91.45.20.24	86400
`lemon.co.uk`	NS	`grove.lemon.co.uk`	86400
`lemon.co.uk`	NS	`tree.lemon.co.uk`	86400
`lemon.co.uk`	MX	`stem.lemon.co.uk`	60
`grove.lemon.co.uk`	A	91.45.23.22	86400
`tree.lemon.co.uk`	A	91.45.23.23	86400
`orchard.lemon.co.uk`	A	91.45.23.82	86400
`stem.lemon.co.uk`	A	91.45.23.85	86400
`www.lemon.co.uk`	CNAME	`orchard.lemon.co.uk`	86400

(*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]

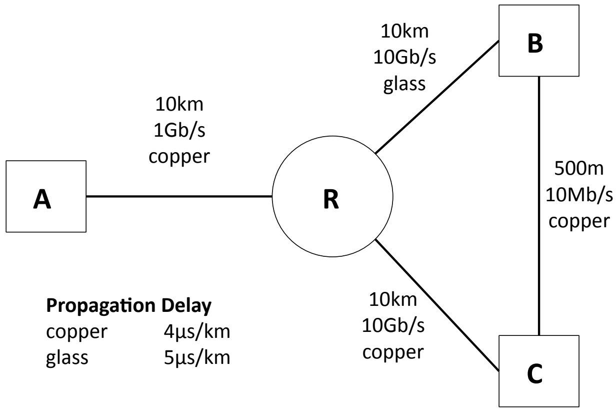
(*iii*) The TTL field refers to the maximum amount of time a DNS server can cache the record. Most of the TTLs in this record were chosen to be 86400 seconds (1 day). What is the trade-off between choosing a shorter or a longer time? Why was the MX record specifically chosen to have a 60 second TTL?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

[continued . . .]



(*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_rcv()* 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]