BCS THE CHARTERED INSTITUTE FOR IT

BCS Higher Education Qualifications
BCS Level 5 Diploma in IT

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EXAMINERS' REPORT

Computer Networks

General comments on candidates' performance

The pass rate this sitting was the best for seven sittings. The top 25% of candidates performed very well and every question had a few candidates gaining close to full marks. There was a welcome reduction in the number of very badly prepared candidates, with the proportion gaining less than 20% dropping to 11%. If such candidates are removed from the figures, the mark distribution is very similar to what would be expected in a similar module at a British university.

It was still the case that some 25% of candidates gained a mark of less than 25%. While some of these candidates appear to have such poor skills in written English that they cannot express the knowledge, the majority seem to be inadequately prepared. In particular, there is no sign that they have studied previous exam papers and the answer pointers provided in the examiners' reports.

While they have no firm evidence, the examiners have the impression that some of those teaching courses to prepare candidates for this examination:

- do not cover the whole course but concentrate on what they believe to be the easier parts of the syllabus;
- do not draw candidates' attention to the existence of the examiners' report nor encourage them to read them;
- do not themselves read and take notice of the examiners' reports.

Nineteen per cent of the candidates who registered for the module failed to present themselves for the examination. Furthermore, this proportion varies considerably from centre to centre: the three largest centres had proportions of 8%, 16% and 25%.

Question A1

This question is about the frames used by the Local Area Network (LAN) technology known as Ethernet /IEEE 802.3.

a) Produce a sketch diagram to show the fields of a frame as used by Ethernet / IEEE 802.3. **(6 marks)**

b) Explain the role of the preamble and start of frame delimiter fields in an Ethernet / IEEE 802.3 frame. **(6 marks)**

c) How was the original definition of an Ethernet / IEEE 802.3 frame updated by IEEE 802.1Q to permit the use of Virtual Local Area Networks (VLANs)? **(4 marks)**

d) What range of sizes is permitted for the data/payload field of an Ethernet / IEEE802.3 frame? **(3 marks)**

e) Why was it necessary to specify both a minimum length and a maximum length for the data/payload field of an IEEE802.3 frame? **(6 marks)**

Syllabus section: Local Area Networks

Answer Pointers

Part (a)

Preamble	Start of frame delimiter	MAC destination	MAC source	802.1Q tag (optional)	Ethertype (Ethernet II) or length (IEEE 802.3)	Payload	Frame check sequence (32-bit CRC)
7 octets	1 octet	6 octets	6 octets	(4 octets)	2 octets	46(42)[b]–1500 octets	4 octets
		← 64–1518(1522) octets →					

(Marking scheme: 6 marks for the correct frame)

Part (b)

The preamble is an alternating sequence of 1s and 0s and is largely present to enable receivers' clocks to be tuned to the precise bit rate being used as the transmit bit rate. The start of frame delimiter, which is 8 bits, is 01010111, that is, it ends with two 1s and its purpose is to enable the receiver to synchronize to the octet boundaries and know that the real packet information follows next.

(Marking scheme: 3 marks per correct explanation)

Part (c)

The type/length field and the 802.1Q tag field were added to hold VLAN ID and priority information.

(Marking scheme: 2 marks per field)

Part (d)

The data/payload field has a "traditional" minimum size of 46 octets and a maximum size of 1500 octets. The presence of an 802.1Q tag field changes the minimum size to 42 octets.

(Marking scheme: 1 mark for indicating original minimum size; 1 mark for maximum size and 1 mark for minimum size when 802.1Q tag was included)

Part (e)

The origin of the sizes are related to the use of the CSMA/CD algorithm, in shared co-axial cable Ethernets and the maximum permitted network path length. The essence of the minimum size is that the time to transmit even the smallest permitted packet should be at least the round trip time of a 10mbps co-axial Ethernet at maximum permitted length of 2.5km including four repeaters. The maximum time is based on not allowing any one station to be greedy and taking too high a share of the available capacity

(Marking scheme: 2 marks for relating it to CSMA/CD, 2 marks for specifying the relationship; 2 marks for the purpose)

Examiner's Comments

This was not a popular question, being attempted by only 36% of candidates. Of those who attempted the question, about a third showed complete ignorance of the topic. A further third were only attempted one or two sections. The majority of candidates from all centres appeared to have little understanding of IEEE 802.3.

Question A2

This question is about the service provided by the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).

a) What is the difference in the service offered to applications by the TCP and UDP protocols? **(8 marks)**

b) For each of the following applications determine whether you would use TCP or UDP and explain the reasons for your choice.

   i. File transfer **(3 marks)**

   ii. Watching a real time streamed video **(3 marks)**

   iii. Web browsing **(3 marks)**

   iv. A Voice Over IP (VoIP) telephone conversation **(3 marks)**

c) Both TCP and UDP use port numbers. What are these port numbers used for? **(5 marks)**

Syllabus section: Inter Networks

Answer Pointers

Part (a)

TCP operates above IP and provides a connection orientated service in which end to end data transfer is guaranteed with flow control and congestion avoidance capabilities. It is a reliable service.

UDP operates above IP and provides a connectionless, unacknowledged service to the upper layers. There is no error detection or recovery and no flow control provided by UDP. It is a best effort service.

(Marking scheme: 4 marks for each correct definition)

Part (b)

i. TCP, losing date in text is not acceptable, therefore the error detection and correction mechanisms of TCP are needed.

ii. TCP, the reason is that when watching a movie, delay is critical and therefore there simply isn't any time to seek the retransmission of any errors. The simplicity of UDP is therefore required.

iii. TCP, The reason is that web pages need to be delivered without error so that all content is properly formatted and presented. Therefore the error detection and correction properties of TCP are needed.

iv. UDP, The reason is that a telephone conversation has strict timing requirements for the transfer of data and seeking the retransmission of any errors would introduce too much delay. Therefore the simplicity of UDP is needed.

(Marking scheme: 3 marks for each correct answer)

Part (c)

Both TCP and UDP provide services to higher layer protocols however multiple higher layer protocols can be multiplexed onto a single UDP or TCP layer. Each of these higher layer protocols are then differentiated by means of UDP/TCP port numbers. Port numbers are 16 bits in length. Therefore the port number identifies the particular higher layer protocol to which a given data stream is destined.

(Marking scheme: 2 marks for explaining the multiplex nature of the protocols; 2 marks for indicating the purpose of the ports; 1 mark for specifying the length of the port)

Examiner's Comments

This question was attempted by all the candidates and 78% of them achieved a pass mark. One candidate got full marks and 9% got marks of more than 90% for the question. A healthy majority of the candidates demonstrated a good understanding of how TCP and UDP work.

Question A3

This question is about physical layer transmission systems and Asymmetric Digital Subscriber Line (ADSL) broadband.

a) The twisted pair telephone line which connects a house to the local telephone exchange was originally designed for the transmission of analogue voice. What is the typical frequency range used by analogue voice signals? **(2 marks)**

b) Nowadays the twisted pair telephone line is additionally able to support Internet access with data rates measured in Mega-bits-per-second (Mbps) using Asymmetric Digital Subscriber Line (ADSL) technology. Explain how ADSL is able to transmit data over the same line as analogue voice and without the two interfering with one another. Complement your answer with an image that shows how ADSL works. **(9 marks)**

c) With reference to ADSL, explain the function performed by the Digital Subscriber Line Access Multiplexer (DSLAM). **(8 marks)**

d) Two customers are connected to the same local telephone exchange and both receive their Internet connection via ADSL. However, one customer has measured their Internet connection download data rate to be 5Mbps and the other 2Mbps. Thinking about these two customer's telephone lines, suggest at least two reasons to explain why their actual data rates are so different.

(6 marks)

Syllabus section: Digital Communication

Answer Pointers

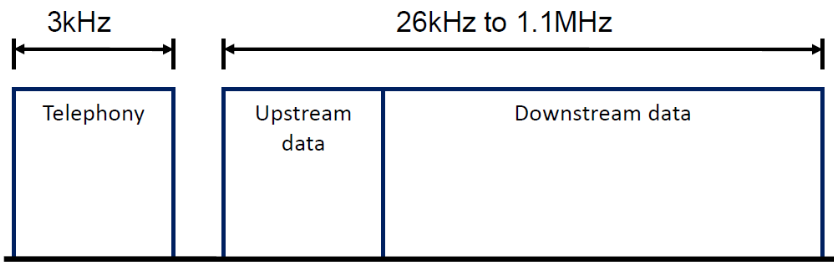
Part (a)

Frequency range is 300Hz to 3400Hz (also accept 0 to 3kHz or 0 to 3.5kHz).

(Marking scheme: 2 marks for the correct frequency range)

Part (b)

ADSL delivers data and telephony over the same twisted pair by using higher frequencies for the data services. Telephony occupies the first 3kHz of a line's bandwidth and ADSL uses frequencies in the range 26kHz to 1.1MHz for data. This bandwidth is then divided into 256 channels, each of 4.3kHz and within these 256 channels, adaptive coding is used (QPSK, QAM) to encode up to 64 kbps per channel. In order to maintain the separation of the data and voice signals, a micro filter is used within the home to ensure that frequencies above 3kHz cannot reach the telephone. Similarly at the telephone exchange filters separate out the voice and data frequencies.



(Marking scheme: 2 marks for explaining the use of different frequencies; 2 marks for indicating the frequencies used by telephony; 2 marks for indicating the frequencies used by data; 2 marks for indicating the use of microfilter; 1 mark for the diagram)

Part (c)

The DSLAM is located within the telephone exchange. Each telephone line is filtered to extract the data component which is passed through the DSLAM. The DSLAM therefore accepts data from all locally connected telephone lines and multiplexes these onto a single data connection from the telephone exchange to the backbone network and from there onwards to the Internet. The DSLAM in effect functions as a network switch. Since the backbone network connection from the exchange has a finite capacity, a contention ratio exists between the amount of data that could be generated by locally connected users and the capacity of this connection.

(Marking scheme: 2 marks for indicating the location; 2 marks for indicating the purpose of DSLAM; 2 marks for describing how DSLAM works; 2 marks for indicating how DSLAM affects transmission speeds)

Part (d)

Possible reasons are:

- That the customer receiving 5Mbps is physically closer to the exchange than the user receiving 2Mbps.
- The quality of the copper line for the user receiving 2Mbps is poorer (older, lower quality) than the user receiving 5Mbps)
- The line for the user receiving 5Mbps is subject to less interference (cross talk in the local loop) compare to the user receiving 2Mbps.

(Marking scheme: 3 marks for each correct reason)

Examiner’s Comments

A reasonably popular question attempted by 84% of the candidates but only 47% attained a pass mark.

The main weakness in the answers was that they were vague and imprecise. Most candidates clearly knew something about the topics but were able to demonstrate only superficial knowledge. Some candidates confused the DSLAM technology withfibre optic technology.

Question B4

The two main functions of an Internet Protocol router are the forwarding of individual packets (“switching”) and the maintenance of routing tables (“routing”). This question considers router behaviour and routing protocols such as EIGRP.

a) The first part of this question concerns the forwarding of individual packets (“switching”).

   i. What are the main demands on the internal resources of a router when it is taking forwarding decisions for individual packets? **(4 marks)**

   ii. To what extent are other routers involved in the internal forwarding decisions and actions of a router when it processes an individual packet? **(4 marks)**

b) This second part of this question concerns the maintenance of routing tables (“routing”).

   i. Briefly explain the behaviour of the class of routing protocols normally described as distance vector protocols. **(4 marks)**

   ii. In what ways are link-state protocols often considered to be superior to distance-vector routing protocols? **(6 marks)**

   iii. Briefly describe the routing protocol known as EIGRP and explain how it copes with routing inside a large and complex autonomous system. **(7 marks)**

Syllabus section: Wide Area Networks

Answer Pointers

Part (a)

i. The key demand here is that of capacity and speed of processing. The forwarding decisions have to be taken for each and every packet and need to be fast.
(Marking scheme: 1 mark per resource; 1 mark for indicating the process per packet and 1 mark for indicating the requirement)

ii. The requirement for forwarding decisions is to be fast and thus routers need to take such decisions based on information that is immediately available. Other routers are NOT contacted at this stage in terms of the current router taking its forwarding decisions. The routing tables will of course be based on information received from other routers, but the creation of those tables is done on another timescale and NOT at the time of actual forwarding of packets. When a packet leaves the current router it will either be delivered to a directly connected destination or handed on to another router.
(Marking scheme: 2 marks for indicating the involvement of routers when taking decisions on forwarding and 2 marks for the involvement of other routers in the information available for routing.)

Part (b)

i. Distance vectors protocols operate via routers announcing information about directly connected links and forwarding their routing tables to directly connected neighbours (routers). The neighbours calculate their own routing tables from the information received and repeat the process. In distance vectors protocols, routers send their entire routing table information periodically regardless of an update occurring or not.

(Marking scheme: 2 marks for initial explanation of how they work; 2 marks for indicating the updates)

ii. Link-state protocols typically generate less network traffic than distance vector protocols (1). Link-state protocols converge quicker after topology changes (1) and are very stable when the network is stable (1). Link-state protocols are much less likely to suffer from routing loops (1). Link-state protocols can share traffic over multiple valid paths (1).

(Marking scheme: 1 mark per characteristic)

iii. Fast convergence—EIGRP uses DUAL to achieve rapid convergence. A router running EIGRP stores its neighbours' routing tables so that it can quickly adapt to changes in the network. If no appropriate route or backup route exists in the local routing table, EIGRP queries its neighbours to discover an alternative route. These queries are propagated until an alternative route is found, or it is determined that no alternative route exists.

Partial updates—EIGRP sends partial triggered updates instead of periodic updates. These updates are sent only when the path or the metric for a route changes; they contain information about only that changed link rather than the entire routing table. Propagation of these partial updates is automatically bounded so that only those routers that require the information are updated. As a result, EIGRP consumes significantly less bandwidth than IGRP. This behaviour is also different than link-state protocol operation, which sends a change update to all routers within an area.

Successor and feasible successor—EIGRP allows to have feasible successors or secondary routers stored in the topology table and they can become active as soon as the primary link goes that.

(Marking scheme: 3 marks per explanation)

Examiner's Comments

This was one of the least popular questions, with only 43% of the candidates attempting it. Of those attempting it, only 23% achieved a pass mark.

It was clear that some students didn't understand the concept of "router resources" and did not have a clear view on how routing information is exchanged with neighbouring routers. Most of the students knew the difference between distance-vector and link-state protocols but failed to explain how EIGRP works.

Question B5

This question is about error control in communications systems.

a) Briefly explain the difference between single-bit errors and burst errors.

(3 marks)

b) Imagine that a noise event causes a burst error to occur that lasts for 0.1 ms (millisecond).

i. If data is being transmitted at 10Mbps. If how many data bits will be affected?

(3 marks)

ii. If data is being transmitted at 100Mbps. If how many data bits will be affected?

(3 marks)

c) Under what circumstances is the use of parity bits an appropriate error control technique? **(3 marks)**

d) Explain the meaning of the term "residual error rate" in the context of error detection schemes. **(3 marks)**

e) Under what circumstances is the use of cyclic redundancy checks (CRC) an appropriate error control technique? **(3 marks)**

f) Very briefly outline how the CRC method functions. **(7 marks)**

Syllabus section: Errors

Answer Pointers

Part (a)

Single bit errors are errors which only affect isolated bits. Burst errors are errors which affect many bits within a block of bits.

(Marking scheme: 1 mark for explaining single bit errors; 2 marks for explaining burst errors)

Part (b)

i. A time period of 0.1ms is the same as 1/10000, 0.0001 of a second. If we are transmitting at the rate of 10 Mbps, then in 0.1ms 1,000 bits will have been transmitted and thus affected by the burst.

   (Marking scheme: 3 marks for the correct calculation)

ii. If we are transmitting at the rate of 100 Mbps, then in 0.1 ms 10,000 bits will have been transmitted and thus affected by the burst.

   (Marking scheme: 3 marks for the correct calculation)

Part (c)

The use of parity bits is really only appropriate if just single bit errors are expected. (Marking Scheme: 3 marks for the correct answer)

Part (d)

The residual error rate is a measure of the proportion of bits that remain corrupted and undetected, even though some form of error control system is in use.

(Marking scheme: 3 marks for correct answer)

Part (e)

The use of cyclic redundancy counts (CRC) is appropriate if burst errors are expected.

(Marking scheme: 3 marks for correct answer)

Part (f)

The CRC method works by taking a sequence of bits and dividing this by a predefined polynomial and then transmitting the remainder along with the data. On reception, the receiver conducts the same arithmetic and if the answers match it assumes successful transmission otherwise it notes an error.

(Marking scheme: 4 marks for the sender[s operation and 3 marks for the receiver's operation)

Examiner's Comments

Question 5 was attempted by only 49% of candidates, of whom 55% achieving a pass mark.

More or less every sitting of this module contains a straightforward question on error control. However, the question never proves popular and is never well answered – or, more precisely, some candidates produce excellent answers while others display almost complete ignorance.

Question B6

This question is about the ISO Reference Model and the TCP/IP protocol stack.

a) The ISO Reference Model defines seven protocol layers, each of which is responsible for a specific range of functions. By considering this model, mention two main functions performed by a protocol operating at the network layer.
(2 Marks)

b) Give the names of the seven layers of the ISO Reference Model and the names of the four corresponding layers in the TCP/IP protocol stack, showing the correspondence explicitly. **(11 marks)**

c) Figure 1 shows a small scale network comprising one switch and one router. A personal computer is connected to the switch and a server is connected to the router. All switch and router ports are IEEE 802.3 CSMA/CD.

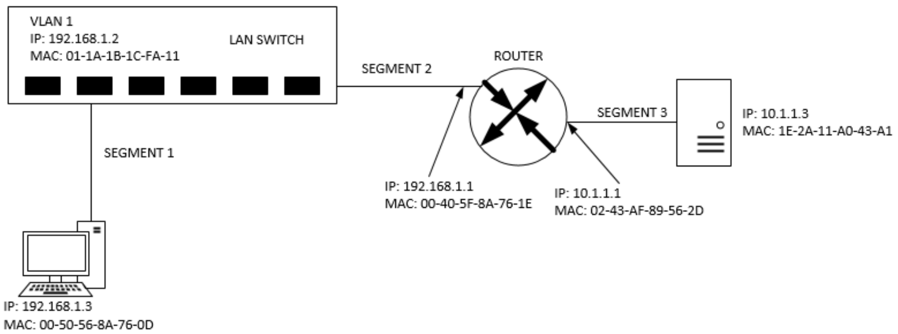


Figure 1

For this network, consider data being sent from the personal computer to the server and indicate the values of the source and destination address fields of the frame and the IP header for each segment. **(12 marks)**

Syllabus section: Introduction

Answer Pointers

Part (a)

The network layer is responsible for packet forwarding, including routing, through intermediate routers. The functions include:

- Connectionless communication
- Host addressing
- Message forwarding

(Marking scheme: 1 mark per function up to 2 marks for the question)

Part (b)

Layer 7	Application	Application
Layer 6	Presentation	
Layer 5	Session	
Layer 4	Transport	Transport
Layer 3	Network	Internet
Layer 2	Data link	Network Access
Layer 1	Physical	
	ISO Reference model	**TCP/IP**

Part (c)

	Segment 1	**Segment 2**	**Segment 3**
Source IP address	192.168.1.3	192.168.1.3	192.168.1.3
Destination IP address	10.1.1.3	10.1.1.3	10.1.1.3
Source MAC address	00-50-56-8A-76-0D	00-50-56-8A-76-0D	02-43-AF-89-56-1D
Destination MAC address	00-40-5F-8A-76-1E	00-40-5F-8A-76-1E	1E-2A-11-A0-43-A1

Examiner's Comments

This was the second most popular question on the paper, being attempted by 90% of the candidates, 72% of whom passed.

Students seem to have a clear understanding of the OSI and TCP/IP models but there seems to be a lack of understanding on how this applies during network communication. Many of the students failed to differentiate between IP address and MAC address and segment-to-segment communication.