**BCS THE CHARTERED INSTITUTE FOR IT**

BCS HIGHER EDUCATION QUALIFICATIONS

BCS Level 5 Diploma in IT

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**COMPUTER NETWORKS**

**EXAMINERS' REPORT**

## General Comments

This session has seen one of the worst overall percentage of pass in recent years. Except in one centre, candidates sitting in the rest of the centres turned out a very poor overall performance. In the case of the latter, the poor performance is attributed to poor responses toquestions in both sections, indicative of poor preparation. Given the choices of questions to answer, a poor performance hence indicates poor preparation. It is worth repeating this year too that candidates need significantly better preparation based on good understanding of concepts to have a realistic chance of passing the paper, or getting better marks. The candidates should also ensure that they are thoroughly familiar with the syllabus as well as relevant chapters in one of the primary texts. The examiners reports such as this in addition will help in examination preparation process for this paper. Having said this, in the one centre mentioned above, the overall performance was excellent, which shows that the paper was not hard, and what can be achieved with good preparation.

Candidates still are not reading the questions carefully and well, which leads to poor understanding of what is expected as answers to those questions. It is essential that candidates should spend some time in reading the questions before answering them. There were a number of borderline cases in this session. The number of blank answer books returned was less.

## Section A

A1. This question is about broadband Internet access.

a. A common method for providing broadband Internet access over existing telephone lines is Asymmetrical Digital Subscriber Line (ADSL).

i. Why is it Asymmetrical?

**(4 marks)**

ii. Explain how ADSL is able to transmit both data and telephone calls over the same twisted pair cable which connects a house to a local exchange without the two signals interfering with each other?

**(8 marks)**

b. A typical ADSL network for domestic (home) users would offer a service based upon a contention ratio of 50:1. What is meant by the contention ratio and how does it affect the download speeds that users experience when accessing the Internet?

**(8 marks)**

c. Many countries are now upgrading broadband access networks to offer customers high speed Internet access. One such technology is called fibre to the cabinet or fibre to the curb (FTTC). Briefly explain how this technology differs from ADSL and hence, is able to offer higher bandwidths than ADSL.

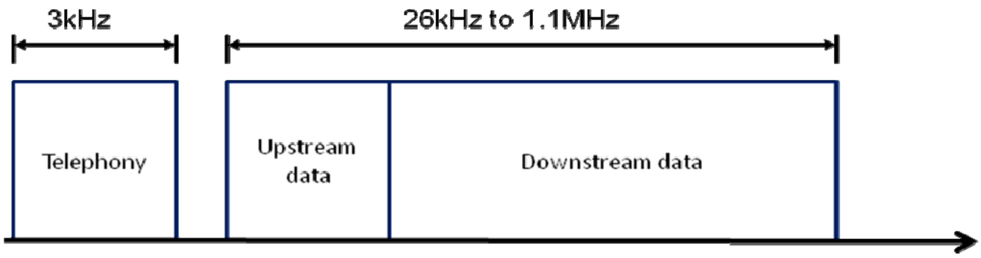
**(5 marks)**

**Answer pointers**

a) i. ADSL is optimised for accessing the web where there is a requirement for greater download than upload rates. This is what makes ADSL, asymmetric where there is a greater data carrying capacity from the exchange to a home than from the home back to the exchange.

2 marks for noting that ADSL was optimised for accessing the web; 2 marks for there being a greater download than upload speed.

ii. 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. Within a home it is important to keep data and telephony separate which requires the installation of micro-filters to achieve this.

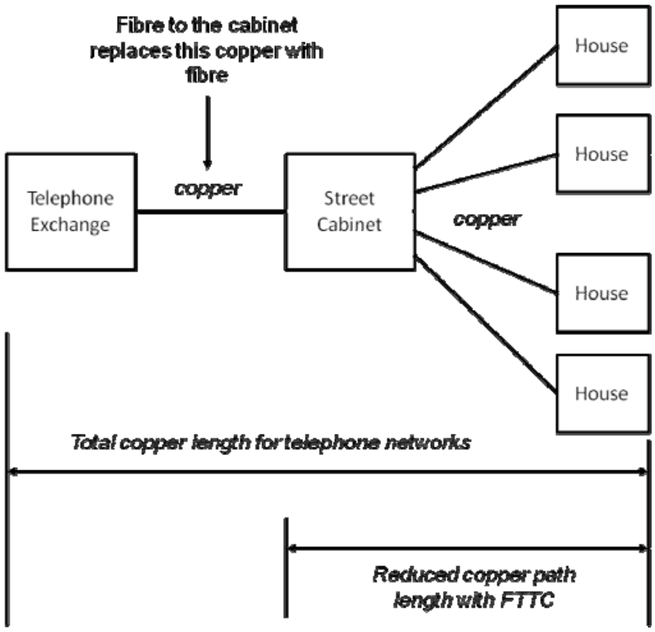



2 marks for telephony being within 3kHz band; 2 marks for ADSL using higher frequencies for data; 2 marks for dividing the data frequencies into upstream and downstream; 1 mark for appreciating that adaptive coding is used and 1 mark for the need to local micro-filters

b) The capacity from a telephone exchange to the core network (backhaul) and hence the Internet is fixed. Therefore, all of the customers connected to the exchange need to share this fixed capacity. This is where contention applies. The capacity of the link from the exchange to the Internet is, in this example, shared across 50 users on the basis that not all of those users will be generating data at the same time. However, if they do, then they will compete for access to a fixed capacity and as such demand will exceed capacity to deliver and the net effect is that each user will observe a reduction in actual Internet bandwidth.

2 marks for knowing that the connection from the exchange to the Internet (backhaul) is fixed; 2 marks for knowing that users are competing for access to this fixed link; 2 marks for recognising that problems exist when users access the Internet at the same time; 2 marks for noting the net effect of a slowing down in Internet access rate.

c) Existing telephony networks provide copper cable from the exchange to the roadside cabinet and then onwards from there to each home. The copper path is therefore from the home to the exchange. There are two key problems which limit bandwidth, the length of the copper path and cross-talk caused from the interference from each copper pair when placed in close proximity to each other. Fibre to the cabinet replaces the copper path from the exchange to the cabinet but leaves the copper from the cabinet to homes. However, this does mean that higher capacity data can be delivered to the cabinet and since the copper path has been reduced, increased capacity can also be achieved over the remaining copper from the cabinet to the home.




1 mark for knowing that the copper path extends from the exchange to the home; 1 mark for understanding the performance limits of this; 1 mark for knowing that fibre will replace copper from the exchange to the cabinet; 2 marks for performance improvement due to reduced copper path length and higher capacity to cabinet through fibre.

**Examiner's Guidance Notes**

This was a poorly answered question. Although it was attempted by 182 candidates (most popular) question, the average mark was only 7.6 out of 25, clearly indicating lack of preparation and understanding of ADSL technology. The majority of the marks were obtained from parts A and B. Most of the candidates lost marks in part C. As with previous years, at some centres candidates attempted to gain marks by re-writing the question in their answer. Many of the answers in this question were creative and generic.

A2. This question is about Asynchronous Transfer Mode (ATM) networks.

a. Show by means of a diagram, the cell format using within an ATM network.

**(5 marks)**

b. What is the difference between a Virtual Path and a Virtual Channel?

**(5 marks)**

c. Explain the purpose and function of the ATM Adaptation Layer (AAL) protocol.

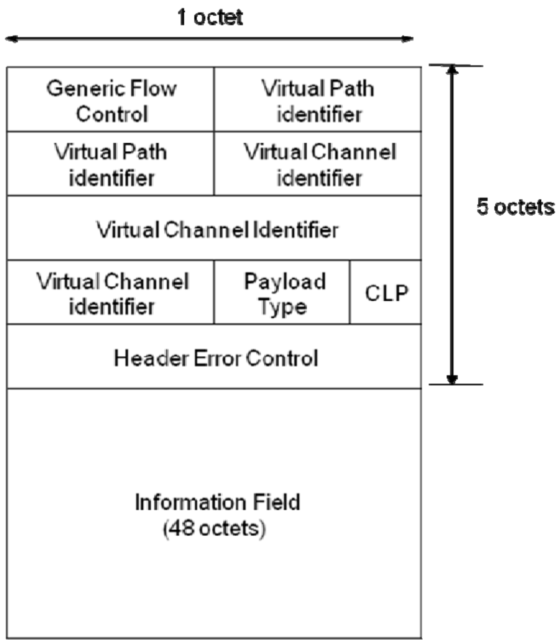
**(9 marks)**

d. Within the context of an ATM network, explain what the different is between constant bit rate (CBR) and available bit rate (ABR) traffic.

**(10 marks)**

**Answer pointers**

a)




1 mark for a 5 octet header; 1 mark for 48 octet payload; 1 mark for Virtual Path identified; 1 mark for Virtual Channel identifier; 1 mark for Header Error Control.

b) A virtual channel defines a single point to point connection, identified by its virtual channel identifier (VCI).

A virtual path however, is a bundle of virtual channels that share the same end-point. Hence, a virtual path can be considered as a container that contains several virtual channels. Each virtual path is identified by its unique virtual path identifier (VPI). 2 marks for the VC, 3 marks for the VP.

c) The lower layers of the ATM protocol suite are responsible for the transmission of the 53 octet ATM cells. At the higher layer we have the applications between transported over the network. Hence, there is a requirement in the middle to convert the application to and from an ATM cell stream. This is the function of the AAL layer.

The AAL protocols are end to end protocols and hence, only present in the end-stations. The basic function of AAL is to segment data from the higher layer protocols into cells and to reassemble a received cell stream into data structures acceptable by the higher layer protocols. Where an application requires a strict timing relationship to be maintained between communicating end-stations then it is the responsibility of the AAL protocol to achieve and maintain this. Equally, the AAL must overcome the problem of lost cells and provide flow and timing control. In this way, it is the AAL that provides the required Quality of Service demanded by the application.

2 marks for noting that AAL sits between the ATM lower layers and application; 2 marks for noting that AAL is only in the end-stations; 3 marks for mapping higher layer protocols onto ATM cells; 2 marks for recovering lost cells/flow control/timing control.

d) The Available Bit Rate service is intended for applications that require a variable bandwidth from the network. Nodes are allowed to transmit cells and providing that capacity exists within the network, these will be carried. However, if the network is unable to carry the traffic being submitted to it, then it will provide feedback to force ABR traffic sources to modify their cell generation rates.

Constant Bit Rate (CBR) is designed for applications which have a known and constant bandwidth requirement. Capacity is reserved within the network and data is transmitted at a fixed rate throughout a connection.

ABR – 2 marks for variable bandwidth; 1 mark for including active feedback to regulate the flow; CBR – 2 marks for constant and fixed bandwidth, 1 mark for cells being sent at the same rate throughout the connection.

**Examiner's Guidance Notes**

Many of the 139 candidates that attempted this question failed to even attract a pass mark. The average mark was 8.4.

Part a), was correctly answered by the majority of candidates. However, there was confusion when answering the remaining parts.

Candidates from many centres seemed to be totally unprepared for ATM.

A3. This question is about providing global network services.

a. Telecommunication companies (Telcos) that provide global network services, define the services they offer to customers within a Service Level Agreement (SLA). What is the purpose of a SLA and give examples of what it might contain.

**(7 marks)**

b. Companies that have offices in several countries around the globe need to create a private corporate network that is able to connect these sites together and transport traffic of different types between them. An increasingly popular way of providing such a network is to use Multiprotocol Label Switching (MPLS) data services. Briefly explain how MPLS works and how it is able to support different traffic types.

**(12 marks)**

c. An alternative to MPLS might be to consider using the Internet. What are the main disadvantages of the Internet that mean it would offer a worse solution than MPLS?

**(6 marks)**

**Answer pointers**

a) The SLA is an agreement between a customer and a Telco and defines what service is being provided to the customer and what the customer is allowed to do with that service.

The SLA would define:

- The data rate being delivered
- QoS parameters such as expected error rates, lost packets, time delays
- availability of the service (including any downtime for maintenance)
- how the service is being charged to the customer
- what the customer is allowed to connect to and use the service (acceptable use policy)
- how the customer can complain if they are unhappy with the service being provided

2 marks agreement between Telco and customer; 1 mark for defining service being offered; 1 mark for a relevant example of what is contained within the SLA up to a maximum of 4 marks.

b) MPLS operates as follows:

- A label switching protocol; a 'label' which is typically a 32 bit number
- Labels are added to data packets by an ingress router based on a classification of the incoming traffic
- That virtual circuits are established through the network for different classes of service using RSVP. This ensures that each virtual circuit is able to support the QoS demands of a particular traffic class or type.
- Information about which label identifiers have been assigned to which data flow is exchanged between routers using a Label Distribution Protocol. The label that is added by the ingress router identifies the QoS virtual circuit to use and all routers within the MPLS network route based on this label field.
- All intermediate routers are termed label switching routers and route this data based on its label only.
- On reaching the far end of the connection, the final or egress router, removes the label.

2 marks for label switching protocol; 2 marks that virtual circuits are established through the network for different classes of traffic; 2 marks for the mentioning that labels assigned to these circuits are distributed by a label distribution protocol; 2 marks for incoming traffic being classified and assigned to a specific virtual circuit; 2 marks for labels added and intermediate routers route using only the labels; 2 marks for label being removed by egress router.

c) Problems with the Internet which make it worse than MPLS are:

- The Internet is a best effort service which means that there is no guarantee that data will be delivered and if it is delivered, there is no way of predicting how long it will take.

- No prioritisation – all traffic types handled equally which means that you cannot differentiate time critical applications from non time critical ones.

- There is no inherent security which means that data can be read by others.

- The volume of traffic on the Internet cannot be predicted which means that the bandwidth being delivered for a service cannot be predicted either.

2 marks for each of the following valid reasons – no delivery guarantee, no traffic differentiation, no security and no bandwidth guarantee - up to a maximum of 6.

**Examiner's Guidance Notes**

Much better responses in this question compared to questions 1 and 2. Although this question was answered by the least number of candidates (118), the average mark was 11 - higher compared to the other two questions in section A.

Part a), was a relatively easy question and attracted many marks by the majority of the candidates. In part b), many candidates correctly identified the 32-bit length of the label but did know the role of the different routers within MPLS. Surprisingly, in part c), almost all of the candidates correctly identified the internet as a best effort service but the rest of their comments were creative. This was a good and challenging question.

## Section B

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 OSPF.

a) This 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 link-state protocols.

**(4 marks)**

ii) In what ways are link-state protocols often considered to be superior to distance-vector routing protocols?

**(4 marks)**

iii) Briefly describe the link-state protocol known as OSPF and explain how it copes with routing inside a large and complex autonomous system.

**(9 marks)**

**Answer Pointers**

a) i. The key demand here is that of capacity (1) and speed of processing (1). The forwarding decisions have to be taken for each and every packet (1) and need to be fast (1). Other relevant remarks rewarded with the total mark capped to 4.

ii. i ). noted the requirement for forwarding decisions to be fast and thus routers need to take such decisions based on information that is immediately available (1). Other routers are NOT contacted at this stage in terms of the current router taking its forwarding decisions (1). 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 (1). When a packet leaves the current router it will either be delivered to a directly connected destination (1) or handed on to another router (1). Other relevant remarks rewarded with the total mark capped to 4.

b) i. Link-state protocols operate via routers announcing information about directly connected links (1) and also forward onwards such announcement so that the knowledge is received by all routers within a given network (1). The routers thus gain complete knowledge of the network topology (1) and from that calculate their own routing tables (1). After initial announcements, further announcements are typically only made when changes take place. Other relevant remarks rewarded with the total mark capped to 4.

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). Other relevant remarks rewarded with the total mark capped to 4.

iii. OSPF splits the network in which it is being used into subsets it calls areas (1) one of which is called the backbone area or area zero (1). The routers are considered to be of different types (1). A router with all links inside a single area is called an interior router (1) and only gains detailed knowledge of the topology of its own area (1). A router with some interfaces in the backbone area is referred to as a backbone router (1). A router with interfaces in more than one area is called an Area Border Router (1) and gains detailed topological information about all those areas (1) which it keeps in separate link-state databases (1) and it inserts summaries of one area into another (1). A router with an interface to another external network is known as an Autonomous System Boundary Router (1). Other relevant remarks rewarded with the total mark capped to 9.

**Examiners' Guidance Notes**

This question was attempted by about 78% of the candidates of whom only a small proportion (17%) achieved a pass mark.

While a small numbers of good quality answers were submitted, many answers were fairly weak. Many candidates did not understand the separation between packet forwarding and the establishment of routing tables. Knowledge of OSPF was very weak.

B5. This question concerns the provision of "quality of service" (QoS) within networks that use the Internet Protocol (IP).

a) The Internet is often described as being a "best effort network". Briefly explain what is meant by the term "best effort network".

**(4 marks)**

b) Identify, and briefly describe, three QoS parameters that are often measured to characterise the behaviour of a network or network connection.

**(6 marks)**

c) Discuss the quality of service requirements of a Voice-over-IP (VoIP) application and how they differ from those of a file transfer application.

**(8 marks)**

d) Briefly describe the Diff-Serv approach to the provision of QoS.

**(7 marks)**

**Answer Pointers**

a) A best effort network is one in which no traffic is given preference over any other (1) and where a guarantee of in-order delivery is not promised (1). Indeed, the network may not guarantee delivery at all (1). The standard Internet we have today has exactly these characteristics and is thus correctly termed a best efforts network (1). Other relevant remarks rewarded with the total mark capped to 4.

b) One commonly used parameter is the usable bit rate (1), which is the rate with which data can be transferred through the network (1). A second parameter is the error rate (1) which would include both corrupted data and data lost (1). A third parameter is transmission delay or latency (1), that is, the time it takes from when a single bit (or perhaps packet) enters the network to when it emerges at its detonation (1). Candidates might also refer to jitter (1) which is a measure of the variation in delay (1). Other relevant remarks rewarded with the total mark capped to 6.

c) Voice –over-IP is a relatively low data rate application (1). However, as it is an interactive inter-human application we require a relatively low delay (1). We need an "evenness" of delivery (1) and so also want jitter to be low (1). However, we can cope with a relatively high error rate (1) as we humans can easily ignore squeaks and crackles and still understand the conversation (1). On the other hand, file transfer is quite different. Whether or not we require a high data rate depends on the size of the file and the amount of time available for the transfer to complete (1). It is unlikely that we have any stringent demands on either of delay (1) or jitter (1). As goes error rates, while we will not be happy with a very high error rate, most file transfer applications have their own mechanism for error checking (1) or will depend on the error checking capability of protocols such as TCP (1). Other relevant remarks rewarded with the total mark capped to 8.

d) The Diff-Serv approach to QoS fundamentally depends on the classification of all traffic into one of a small set of traffic classes (2) rather than applying special treatment to each flow (1). Each class of traffic will be given a certain level of priority handling (1) and each class's traffic will be placed in different output queues by each router (1). The router will then process the traffic from each queue with different priorities (2). Other relevant remarks or extra detail will be rewarded appropriately with the total mark capped to 7.

**Examiners' Guidance Notes**

This question was attempted by only 81% of the candidates of whom only a small proportion (20%) achieved a pass mark.

While a small numbers of good quality answers were submitted, many answers were fairly weak. Candidate's knowledge of the real requirements of the type of application being discussed was confused. Very little knowledge of Diff-Serv was presented in the answers that were submitted.

B6. This question concerns codes to assist in the checking and correcting of communications errors including the use of Hamming codes.

a) What is meant by:

i) the Hamming Distance between two binary numbers?

**(3 marks)**

ii) the minimum Hamming Distance of a code?

**(3 marks)**

b) Imagine you have been asked to design a Hamming Code to transmit information where the original data consists of k-bit binary values and you have decided to use n-bit codes.

i) Using this code as an example, what is meant by the redundancy of a Hamming Code?

**(3 marks)**

ii) Using this code as an example, what is meant by the code rate of a Hamming Code?

**(3 marks)**

c) What must be the minimum Hamming distance of a code if you need to be able to:

i) detect (but not correct) all errors where t bits have been corrupted?

**(3 marks)**

ii) correct all errors where t bits have been corrupted?

**(3 marks)**

d) Design a simple Hamming code to be used to transmit 2-bit data such that all single bit errors can be detected and corrected.

**(7 marks)**

**Answer Pointers**

a) i. The Hamming Distance between two binaries numbers is simply the number of bits by which they differ (3).

ii. The minimum Hamming Distance of a code is the minimum Hamming Distance when considering every possible pair of valid binary numbers defined as being valid within the code (3).

b) i. The redundancy of a Hamming Code is the proportion of bits being transmitted in excess of the number of bits in the original data (1). Formally, this is defined as the value (n –k) / n or equivalently (1 – k/n) (2).

ii. The code rate of a Hamming Code is the proportion of bits being transmitted in which represent the original data (1). Formally, this is defined as the value k / n (2). The redundancy is thus = 1 – redundancy. Should students draw on that then appropriate marks will also be awarded.

c) i. The minimum Hamming Distance of a code that can detect all t-bit errors must be t + 1 (3).

ii. The minimum Hamming Distance of a code that can detect all t-bit errors must be (2 t + 1) (3).

d) If we can detect all single bit errors, then the minimum Hamming Distance of the code needs to be = ( 2 * 1 + 1) = 3.

The original data consists of the values 00, 01, 10 and 11.

One approach is to use a pattern like

54321

dcdcc

where the dd are the data bits and the ccc are extra bits added to make up the code. The ccc bits are calculated by considering the code to have columns numbered from 1 at the right to (in this case) 5 at the left. To calculate ccc you choose a data value for the dd bits and work out which of the numbered columns have a 1. You then "exclusive or" the binary values of the columns populated by 1s in the data and use that addition as the values for the ccc bits.

So, for instance, for 10 1c0cc and the only one is in the five column which as a binary number has value 101 so we put that where the ccc is thus coming to the code value of

11001

And so on for data values 00, 01 and 11.

*The candidates can derive the code by any means, one possible solution being as below, rows with an empty data column being unused.*

| *data* | *code* |
|---|---|
| *00* | *00000* |
| | *00001* |

| | |
|---|---|
| | *00010* |
| | *00011* |
| | *00100* |
| | *00101* |
| | *00110* |
| *01* | *00111* |
| | *01000* |
| | *01001* |
| | *01010* |
| | *01011* |
| | *01100* |
| | *01101* |
| | *01110* |
| | *01111* |
| | *10000* |
| | *10001* |
| | *10010* |
| | *10011* |
| | *10100* |
| | *10101* |
| | *10110* |
| | *10111* |
| | *11000* |
| *10* | *11001* |
| | *11010* |
| | *11011* |
| | *11100* |
| | *11101* |
| *11* | *11110* |
| | *11111* |

*A mark from 0 -> 7 will be awarded depending on how much of the above the candidates manage to complete or show knowledge of.*

**Examiners' Guidance Notes**

This question was attempted by only 20% of the candidates. However, it did acquire the best mark in section B and 26% achieved a pass mark.

A small numbers of good quality answers were submitted, but many other answers were weak. Many candidates' knowledge of Hamming codes was very superficial. The "problem" in part d was about as simple as one can make yet many people did not answer this part and others showed a lack of understanding.