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 is again like the last September session has recorded declining percentage of pass. Except in one centre, where the performance is above this average, candidates sitting in the rest of the centres turned out a relatively poor overall performance which is attributed to poor responses to questions in both sections, indicative of poor preparation indicative of poor preparation. It is worth repeating this message again 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 examiners' reports such as this which are available for 3 years of past sessions will help in examination preparation process for this paper.
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Again this year too it is worth saying that candidates still are not reading the questions carefully and well, which leads to poor understanding of what is expected as answers to those questions. There were a good number of borderline cases in this session.
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Section A

A1. This question is about physical layer transmission systems and Asynchronous 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. This is often now referred to as the Plain Old Telephone Service (POTS). What is the typical frequency range used by analogue voice signals?

(2 marks)

b. Today the twisted pair telephone line is additionally able to support Internet access with data rates measured in Mega-bits-per-second using Asymmetric Digital Subscriber Line (ADSL) technology. Briefly explain how ADSL is able to transmit data over the same line as analogue voice and without the two interfering with one another.

(10 marks)

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

(7 marks)

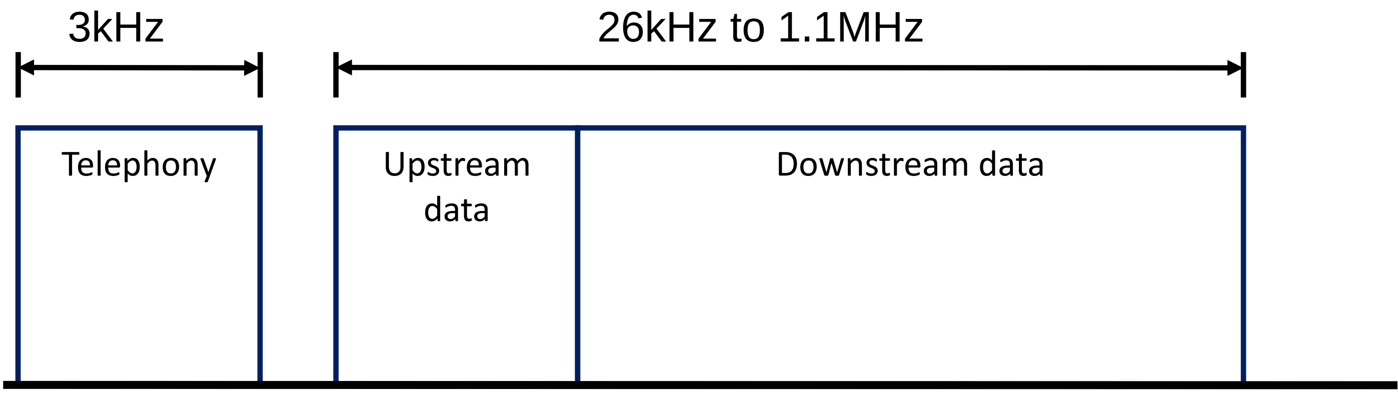
d. Two customers are connected to the same local telephone exchange and both receive their Internet connection via ADSL. However, one customer has measured their Internet connection download data rate to be 8 Mbps and the other 3 Mbps. Thinking about these two customer's telephone lines, suggest reasons to explain why their actual data rates are so different.

(6 marks)

Answer pointers

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

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.



10 marks

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.

7 marks

d) Possible reasons are:
That the customer receiving 8Mbps is physically closer to the exchange than the user receiving 3Mbps.

The quality of the copper line for the user receiving 3Mbps is poorer (older, lower quality) than the user receiving 8Mbps)

The line for the user receiving 8Mbps is subject to less interference (cross talk in the local loop) compare to the user receiving 3Mbps.

6 marks

Examiners' Guidance Notes

Question A1 was attempted by almost all candidates (89%). The candidates were familiar with parts of ADSL technology. However, the pass rate was not high, only at 41% indicating that their understanding was not complete. For example, many of the responses in part A provided only a frequency rather than a frequency range. Responses to parts B and D were okay. Responses to part C were very weak. The majority of the candidates did not seem to have an understanding of the DSLAM nor of its operation.

A2. This question is about virtual circuits and the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).

a. What is meant by a virtual circuit?

(4 marks).

b. A user wishes to establish a connection from their laptop to a server via the Internet using the TCP protocol. By considering the protocol messages that TCP generates, explain how a virtual circuit is established between the laptop and server using TCP.

(9 marks)

c. Whilst transmitting data over a TCP virtual circuit, if errors occurred and data was lost, explain how the TCP protocol would be able to detect this has happened and how it would ensure that the lost data is re-transmitted.

(8 marks)

d. In contrast to TCP, UDP is described as a connectionless protocol. Briefly explain how data is transmitted between two computers using the UDP protocol.

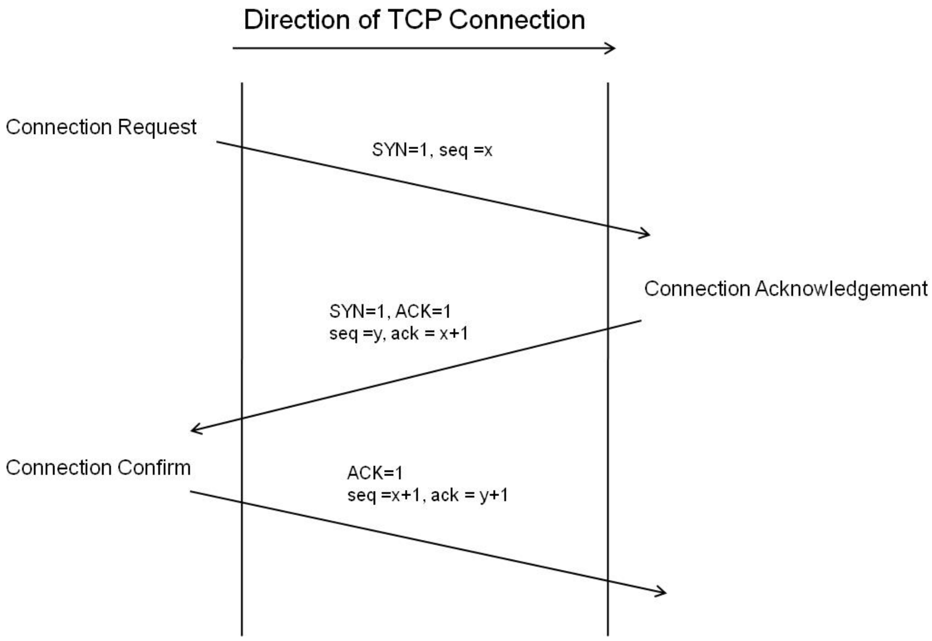
(4 marks)

Answer pointers

a) A virtual circuit is a means of establishing a connection between two points on a packet switched network. It is virtual in the sense that it appears as though there is a dedicated link between these two points.
4 marks

b) A TCP connection is uni-directional which means that for two way communications, a connection must be established from each side. Each process does however, follow the same three way handshake procedure.

The end-station requesting the connection will issue a TCP segment with the SYN flag set and the sequence number equal to some initial value – say x. The receiving end-station – if it wants to accept the connection request – will return a TCP segment with both the SYN and ACK flags set. It will also choose a sequence number starting value – say y. The acknowledgement field of this segment will be set to x+1 to acknowledge receipt of the connection request segment. When the requesting end-station receives this response it will issue one further TCP segment with the ACK field set. The acknowledgement in this segment will be y+1 and the sequence number will be x+1



9 marks

c) The following key points should be noted:

- Every octet transmitted through TCP is uniquely identified by a 32 bit sequence number which increases by one for each new octet.

- Data is transmitted and acknowledged by the receiving end station. Acknowledgements are identified by means of the ACK bit and acknowledgement number within the TCP header.

- A positive acknowledgement is indicated by virtue of the fact that the ACK bit is set and then the acknowledgement number will indicate the number of the first non-acknowledged octet. In other words all octets up to an including acknowledgement number -1 have been successfully received.

- If errors occur in transmission and data is lost, this will be detected by virtue of the fact that an acknowledgement has not been received within a given time – determined by a timer – then the transmitter simply sends the data again. It is the responsibility of the receiver to ignore any duplicates it receives. Hence, the transmitter will continue re-sending data until a positive acknowledgement is received.
  8 marks

d) With UDP there is no need to establish a connection before sending data. Data is simply packaged into a UDP datagram and transmitted to the destination. There are no sequence numbers or acknowledgements. Hence, there is no way for the transmitter to know whether or not data has been correctly received by the receiver.

4 marks

Examiners' Guidance Notes

This question was attempted by 89.5 per cent of the candidates. This question had the highest average mark (10.5 out of 25). The pass mark was 54%, indicating that the candidates had develop a good understanding of virtual circuits and TCP/UPD. However, many candidates did not attract high marks in parts C and D. However, parts A and B were answered correctly by many of the candidates. Many candidates included a diagram in part B but without a description or explanation. Responses to part C were weak. Many of the candidates did not indicate that octets are identified by 32-bit sequence number.

A3. This question is about the Internet and Multi Protocol Label Switching (MPLS)

a. In terms of Quality of Service (QoS), the Internet is described as offering a best effort service. What is meant by this description?

(6 marks)

b. Many telecommunications companies now offer Multi Protocol Label Switching (MPLS) services. How does the Quality of Service (QoS) offered by MPLS differ from that offered by the Internet?

(7 marks)

c. A company has an office in London and another in Brussels and wishes to connect them in order to transmit both data and telephone calls between the two sites. To provide this interconnection, the company is using a Multi Protocol label Switching (MPLS) service. Briefly describe the basic principles of how MPLS functions and explain how it would be able to offer a different Quality of Service (QoS) to the telephone calls compared to the data being transmitted between the sites.

(12 marks)

Answer pointers

a) Best effort means that:

- There is no guarantee that data will be delivered to its destination.
- If data is delivered, there is no guarantee as to how long it will take to be delivered.
- All data is regarded as equal with no prioritisation of traffic types being possible.

6 marks

b) Key points are:

- That MPLS networks seek to differentiate traffic types and ensure that each type is given an appropriate QoS.
- Traffic entering an MPLS network is classified according to its QoS requirements.
- All routers within the MPLS network and along each traffic route apply traffic policies to ensure that each traffic class is given an appropriate QoS (transmission priority)
- In contrast the Internet offers none of these controls.

7 marks

c) An MPLS network comprises a series of ‘label’ switching routers. When traffic arrives at the first ingress router of an MPLS network (Label Edge Router), the traffic is classified and a label added that reflects its destination and QoS requirements. Virtual circuits can either be pre-configured through the network or established dynamically using RSVP.

All intermediate routers are termed label switching routers and process/route each packet based on the value of its label only. This label identifies the QoS requirement of each packet. Information about which label identifiers have been assigned to which data flow is exchanged between routers using a Label Distribution Protocol. A given label is therefore linked to a specific QoS requirement and it is up to each router to ensure that packets are given the appropriate priority when being processed.

On arrival at the final router in the route (egress router), the label is removed and the packet transmitted to the destination.

Telephone traffic has strict requirements in terms of the maximum transmission delay it can endure whereas data has no such restrictions. Therefore it is important for the telephone traffic to be differentiated from the data by defining two classes of service. Labels will be assigned for each class of service and these will be allocated to each data packet as it enters the MPLS network and is classified.

12 marks

Examiners' Guidance Notes

This question was the least popular in section A, with 67% of the candidates attempting it. The pass mark was very low at 16%. This clearly indicated that the candidates were not correctly prepared for MPLS and QoS. Responses to part A varied. Responses to parts B and C were very weak.

Section B

B4. A key component of all internet networks is the router. The two main tasks conducted by a router are packet forwarding and routing table construction. This question is about router behaviour and distance vector routing protocols such as Routing Information Protocol (RIP).

a) This part of this question is about packet forwarding. Assume a router receives an IP packet. Describe the actions the router will perform when forwarding the packet and how this differs for local or remote destinations.

(10 marks)

b) This 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 and illustrate your answer by reference to Routing Information Protocol version 1 (RIPv1).

(9 marks)

ii. What were the main difference between RIPv1 and RIPv2?

(6 marks)

Answer Pointers

a). As noted at the start of the question, the actions of routers can be split into two groups. This part of the question is only about the actions performed when a router tries to forward an individual IP packet. The initial actions of the router will be concerned with comparing the destination IP address in the packet (1) with the (already existing) routing tables (1). This may show the destination matches one of the router's own addresses, if which case the packet will be passed to the appropriate internal module (2). If the address is shown to match a device on a network directly connected to the router, then the router will need to locate a link-level (MAC) address for the destination device (2) and will then deliver the packet via the appropriate interface to the appropriate destination (1). If the destination is seen to be reached via another router (explicitly or via a default route), then the router will need to locate a link-level (MAC) address for the next router (not the final destination of course) (2) and will then deliver the packet via the appropriate interface to that next router (1). Any other relevant information rewarded with total mark for this part capped at 10. Note: the router does *not* construct the routing table at this stage.

b). i. This part of the question is about the Distance Vector routing protocol known as RIPv1. The first point to be made is that the DV protocol does *not* get run for every IP packet (1). The DV protocol is used on a regular basis (1), typically once every 30 seconds or so (1), to announce a router's complete current routing table (1) by broadcast (1) to all other devices and routers (1) on directly connected networks (1). On receiving such RIPv1 packets the router compares all entries in the received announcements, based on the metric (hop count) (1) with the current entries in its own routing table (1) and replaces entries if the new entry has a lower metric (1). Any other relevant information also rewarded with total mark for this part capped at 9.

ii. This part of the question is about updates made when RIPv2 was defined. RIPv1 had many restrictions, some of which are cured in RIPv2. Firstly, RIPv1 announcements do not include network masks (1) and so cannot support classless internet domain routing (CIDR) (1), this information is included in RIPv2 (1). RIPv2 uses multicast rather than broadcast for its announcements (1) thus they will only enter the IP processing stack of devices if those devices subscribe to the appropriate multicast IP address (1). RIPv2 also includes support for a variety of simple forms of authentication (1). Any other relevant information also rewarded with total mark for this part capped at 6.

Examiners' Guidance Notes

This question was attempted by about 76% of the candidates of whom 25% achieved a pass mark. While a small number of reasonable quality answers were submitted, many answers were fairly weak.

While some candidates missed the later point about appropriate selection of MAC address in answers to part a), other candidates included this and indeed went on to discuss the use of ARP to acquire this information if required. Some candidates also chose to discuss QoS as part of their answer. All such valid extra information was awarded marks when appropriate. Some answers to part a) showed one particular misunderstanding which has featured in answers to similar questions in past years. Some candidates clearly, but falsely believe that routers spend a lot of time and effort, per data packet, in actually creating routing tables – this is false. For part b) many answers lacked detail on what routers do with RIPv1 packets when they arrive, when they cause routing tables to be updated and when they might be discarded. Answers to part c) showed a range of understanding and knowledge.

B5. 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)

Answer Pointers

a). Standard diagram expected similar to that shown on the wikipedia page at http://en.wikipedia.org/wiki/Ethernet_frame or any similar presentation, but it clearly says the marks here are for a diagram and not lots of text. (6) However, we note that candidates are likely to show the version without the optional 802.1Q tag field and also show the payload area as being a minimum of 46 octets. Either form will be permitted full marks for this section of the question.

b). This part of the question asks for an “explanation” and so should not be just bullet points. The students should explain that the preamble is an alternating sequence of 1s and 0s (1) and is largely present to enable receivers' clocks to be tuned to the precise bit rate being used as the transmit bit rate (2). The start of frame delimiter, which is 8 bits, is 01010111, that is, it ends with two 1s (1) and its purpose is to enable the receiver to synchronize to the octet boundaries and know that the real packet information follows next (2). Other relevant information also rewarded with total mark capped at 6.

c). For this part of the answer, candidates should now discuss the specification of s specific value for the type/length field (1) and the addition of the 802.1Q tag field (2), which is used to hold VLAN id and priority information (1). Any other relevant information also rewarded with total mark for this part capped at 4.

d). The data/payload field has a “traditional” minimum size of 46 octets (1) and a maximum size of 1500 octets (1). The presence of an 802.1Q tag field changes the minimum size to 42 octets (1). Any other relevant information also rewarded with total mark for this part capped at 3.

e). The candidates should now discuss the reason for data/payload size limits. They should discuss that the origin of the sizes are related to the use of the CSMA/CD algorithm (1), in shared co-axial cable Ethernets and the maximum permitted network path length (1). 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 (2). The maximum time is based on not allowing any one station to be greedy and taking too high a share of the available capacity (2). There is lots more detail candidates could provide, but that is not required. Any other relevant information that is given will also be rewarded with total mark for this part capped at 6.

Examiners' Guidance Notes

This question was attempted by about 35% of the candidates of whom 35% achieved a pass mark.

While a small number of reasonable quality answers were submitted, many answers were fairly weak. Some candidates were very aware of the changes made to accommodate VLANs and gave good answers to part c) (and indeed made appropriate reference in other parts) whereas some candidates appeared to have no knowledge of this issue even though it now has very practical relevance in many network deployments.

B6. This question is about various error checking and correction techniques that are often used in data communications.

a) A common simple error checking system is known as parity checking.

i. Assume we are told that an odd parity system is in use and the following binary value is received.

01001011

Explain what we can say about whether or not this value has been corrupted.

(4 marks)

ii. Imagine a block of eight characters, each of 7 bits, needs to be transmitted. Explain how a combination of vertical and horizontal parity could be used to both detect, and correct, a single bit error.

(8 marks)

b) Another commonly used form of error checking is known as Cyclic Redundancy Codes (CRCs).

i. Briefly explain why the use of simple parity bits is not appropriate when trying to detect error bursts.

(4 marks)

ii. Explain how the CRC system operates and illustrate your answer by referring to the following example.

We wish to transmit 6-bit data values.
We wish to use a 3-bit Frame Check Sequence (FCS).
Overall, we thus transmit a 9-bit sequence.

(9 marks)

Answer Pointers

a). i. The candidates are expected to demonstrate that they understand that “odd parity” means that the overall set of bits, including data and parity bits, should include an odd number of 1s (2). Thus, considering the example given, 01001011, it is clear we have an even number of 1s and thus, no matter which bit or bits are parity bits, this value must have been corrupted (2). Any other relevant information rewarded with total mark for this part capped at 4.

ii. The candidates should demonstrate that they understand that parity can be applied as a bit per character in a transmission (1) or as an extra parity character per transmitted block of characters (1). Given we are told that 7 bit characters are in use, this, together with the parity bit, means that the items transmitted will be eight bits each (1). We are then told that eight characters are transmitted per block, so with the extra parity character, 9 items will be transmitted per block (1). We are now asked about how we can detect and correct a single bit error. Candidates should note that a single bit error will cause the parity be corrupt and we can identify that. The single bit error will also cause the parity to be wrong for one of the columns of bits in the block and we can also identify which column (2). Thus, we know which character and which bit has been corrupted so we can correct that by changing the bit in question from 1 to 0 or 0 to 1 as appropriate (2). Candidates might choose to provide a diagram as part of this answer, that, and any other relevant information will be rewarded with total mark for this part capped at 8.

b). i. Candidates should demonstrate that they understand that multiple corrupted bits cloak each other (2). Thus, if multiple bits are corrupted, as happens with burst errors, then the use of parity bits will be likely to not detect the corruption (2). Any other relevant information also rewarded with total mark for this part capped at 4.

ii. CRC operates by adding an error control to sequences of data [6-bit data values in our example] (1). The original data is first augmented by a small set of zeros the length of our intended FCS sequence [thus 3 zero bits in our example] (2). The augmented sequence is then divided [using modulo 2 arithmetic] by a pre-agreed divisor [not specified in our example] (2) and the remainder is then added to the original data which is then transmitted (2). The receiver then divides by the same pre-agreed divisor and if all is o.k. there should then be no remainder (2). There are alternative ways to describe this and all will be awarded marks as appropriate.
Any other relevant information also rewarded with total mark for this part capped at 9.

Examiners' Guidance Notes

This question was attempted by about 43% of the candidates of whom 38% achieved a pass mark.

While a small numbers of good quality answers were submitted, many answers were fairly weak. Many answers contained a common misunderstanding with regard to "parity bits". Some answers, quite incorrectly, said that odd parity implied that the parity bit would be set to a 1. Candidates should realise that the parity bit will be set so that the selected rule applies across all the bits being considered including the parity bit. This is further explained in the answer pointers given above. In many answers to part bi, many answers completely lacked detail and did little above repeating our question. A number of good answers were given for part c) but other students were clearly cleared and talked about hamming codes rather than the question we had asked.