BCS HIGHER EDUCATION QUALIFICATIONS
BCS level 5 Diploma in IT

April 2010

EXAMINERS' REPORT

COMPUTER NETWORKS

General Comments

This time the responses to questions were as before in April 2009 examinations, not as high a quality as expected. As in the last examination, again a number of students were on the borderline or near borderline pass category. The number of passes was about the same as in the previous examination. There were a few good answers, but not in the numbers expected. In general, as said in the previous examiners' report, students need better preparation based on good understanding of concepts which alone ensures good performance. The students are strongly advised to read examiners reports such as this as part of their preparation for the examination besides preparing answers for questions..

Section A

A1

a. What is the difference between a *connection orientated* and *connectionless* communications protocol?

(4 marks)

b. Explain what Quality of Service (QoS) is offered by:

   i. IP

   (1 mark)

   ii. UDP

   (3 marks)

   iii. TCP

   (4 marks)

c. What function do TCP and UDP port numbers provide and what is meant by a *well known* port?

(5 marks)

d. Briefly explain how TCP achieves reliable data transfer over a network even if errors occur during the transmission.

(8 marks)

Answer pointers

a)

Connection orientated:	Before data is transferred, a connection needs to be opened and acknowledged Once established data is transferred and acknowledged to ensure reliable transfer

Connectionless:
Data is simply transmitted when ready (no connection establishment phase)
However, data is unacknowledged and therefore unreliable

(Marking: 2 for the need for a connection establishment phase and 2 marks for noting that connection orientated protocols provide reliable data transfer with acknowledgements. 2 marks for the fact that no connection establishment is needed and that data is sent when ready, 2 marks for noting that data is unacknowledged and therefore unreliable transfer)

b) IP best effort.

<u>UDP</u>
UDP provides a connectionless, unacknowledged service to the upper layers. There is no error detection or recovery and no flow control provided by UDP.

(Marking: 1 mark for connectionless, 1 mark for no error control, 1 mark for no flow control)

<u>TCP</u>
TCP provides a connection orientated service in which end to end data transfer is guaranteed through acknowledgements and repeated re-transmissions. The protocol also provides flow control and congestion avoidance. It is a reliable service.

Marking: 1 mark for connection orientated, 1 mark for reliable data transfer including mention of acknowledgements, 1 mark for flow control/congestion avoidance, 1 mark for reliable service.).

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 and are carried in the TCP/UDP datagrams when transmitted over a network. Therefore the port number identifies the particular higher layer protocol to which a given data stream is destined.
Some of these port numbers have been defined by the IETF to signify particular protocols – for example port 80 refers to http. Such port numbers are called well known ports.
(Marking: 1 mark for multiplexing of higher layer protocols onto TCP/UDP, 1 mark for identifying higher layer protocol to which data belongs, 1 mark for 16 bit length, 1 mark for the fact that port numbers are included within TCP/UDP datagrams, 1 mark for well known port = protocol id defined by a standard.).

d) 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 data is corrupted or lost in transit then this must be detected by the transmitter. If 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.

Examiner's comments:

a) Some students answered this in the manner set out in the marking scheme, but there were many confusions amongst the rest. The principal confusion was to regard the setting up of a distinct route as the key to connection-oriented. This may happen in some connection-oriented systems, but is not a necessary part of the concept – in particular this view overlooks how TCP works over IP. It was notable that quite a few students in fact thought that TCP does indeed select a path through the network in setting up a connection – this shows a clear weakness amongst the students on basic point concerning TCP/IP (a point reflected in later parts of the question as well).

b) Most students found the first part of this question the hardest – the quality of service offered by IP. They did better on UDP and TCP.
i) a surprising number of students thought that IP offered a reliable service. Many others were simply not sure what to say in relation to this question. There was a tendency to talk about IP addressing.
ii) and (iii) were answered fairly well by a fair proportion of the students.
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 and are carried in the TCP/UDP datagrams when transmitted over a network. Therefore the port number identifies the particular higher layer protocol to which a given data stream is destined.
Some of these port numbers have been defined by the IETF to signify particular protocols – for example port 80 refers to http. Such port numbers are called well known ports.
(Marking: 1 mark for multiplexing of higher layer protocols onto TCP/UDP, 1 mark for identifying higher layer protocol to which data belongs, 1 mark for 16 bit length, 1 mark for the fact that port numbers are included within TCP/UDP datagrams, 1 mark for well known port = protocol id defined by a standard.).

d) Surprisingly the students did not score well off this question on the whole. It is not clear why since the question is straightforward and could be anticipated as a likely question. Many students (as always) thought that the datagrams were numbered (rather than the octets). If the students said that a 3 way handshake was involved some marks were given for this, although it was not mentioned in the marking scheme.

A2

a. A telephone line is designed to provide a channel bandwidth of approximately 3kHz for the transmission of voice. Explain the basic principles of a modem (modulator/demodulator) which allows for computer data to be transmitted over a telephone voice channel.

(6 marks)

b. Explain how asynchronous digital subscriber line (ADSL) technology is able to deliver very high bandwidths for downloading data from the world-wide-web over voice grade telephone lines.

(8 marks)

c. A user is in their home and accessing the Internet from a laptop computer connected via a wireless LAN (WiFi) to an ADSL broadband router which in turn is connected to their telephone line. The user also has a conventional (analogue) telephone.

i. Why is it necessary to use a micro-filter in this network?

(2 marks)

ii. How is the user's data separated from their telephone calls within the local exchange?

(3 marks)

d. Although ADSL is advertised to provide data rates of up to 8MBps download, give <u>three</u> reasons why a user may actually experience a data rate much lower than this.

(6 marks)

Answer pointers

a) Data bits are transmitted as a sine wave within the 3,000 Hz bandwidth limit; logic 1 and 0 each at a different frequency
Two sets of frequency are used; one for upstream and one for downstream
Data is then encoded through a combination of the amplitude of the sine wave and also its phase angle. For example, 2 amplitudes and 4 phase angles gives 8 combinations thereby allowing 3 bits to be represented by each combination. In this way, higher data rates can be transmitted.

(Marking: 1 mark for using a sine wave within 3kHz, 1 mark for knowing that logic 0 and 1 are sent at different frequencies, 1 mark for knowing that upstream and downstream use different frequencies, 3 marks for amplitude and phase changes and how this results in higher data rates.).

b)Unlike modems which use frequencies in the 3KHz audio band, ADSL uses frequencies between 26kHz and 1.1MHz.

This bandwidth is then divided into 256 channels, each of 4.3kHz.
Within these 256 channels, adaptive coding is used (QPSK, QAM) to encode up to 64 kbps per channel.

The modem automatically determines the quality of signal it is obtained on each channel and reduces or increases its encoding scheme accordingly to reduce errors at the expense of a lower data rate.

Marking: 2 marks for knowing that ADSL uses frequencies well above the 3kHz audio band; 2 marks for knowing that this bandwidth is divided into channels (256) and 2 marks for knowing that data is then encoded on these channels; 2 marks for knowing that the modem auto-adjusts the encoding to match the quality of signal)

c) Micro-filter

The micro-filter separates in frequency, the analogue telephone (POTS) from the data stream generated by the ADSL router.

At the exchange, the analogue telephone is separated from the data stream by filtering. The telephone signals are routed to the telephone network whereas the data passes to a Digital Subscriber Line Access Multiplexer (DSLAM). This multiplexers the local data streams into a single data stream from the exchange to the Internet.

(Marking: 2 marks for knowing that data is passed to a DSLAM at the exchange and 1 mark for knowing that the DSLAM multiplexes local data traffic onto a single external line from the exchange)

d)

1. The quality of the telephone cable to your house will affect the data rate achievable;
2. The distance from the exchange will cause signal attenuation which again, reduces the data rate;
3. Contention at the DSLAM within the exchange means that when more people are using their ADSL lines, so each will get proportionally less.

(Marking: 1 mark for each of the above. If the above three points are not covered, allow 1 mark for suggesting that the performance of the user's computer or WiFi network could be a factor)

Examiner's comments:

(a) in the marking scheme a lot of detail was called for. If the students simply gave a basic account of the operation of a modem (digital to analogue and analogue to digital) then I gave 3 marks.
(b) a fair proportion of students gave quite good answers here – even if not quite to the level of detail in the marking scheme. One mark was given for saying that the rate of transmission was different in the 2 directions, since the question was how ADSL was able to deliver very high bandwidths for downloading.
(c) A good proportion of the students knew the function of a micro-filter and a smaller proportion were able to shed light on the set up at the exchange end of the communication and the role of the DSLAM.
(d) this question was well done – many students gave all or some of the precise answers required and others made sensible points that were relevant.

A3.

a. Produce a sketch of the ISO Reference Model in which you clearly label each layer with its name.

(7 marks)

b. Explain what is meant by the term *peer to peer protocol*.

(2 marks)

c. MAC and IP addresses belong to different layers within the ISO Reference Model. Which layer do they belong to and what function do they perform?

(6 marks)

d. If you know an IP address of a computer that is within the same sub-network as you, explain how the Address Resolution Protocol (ARP) is able to obtain that computer's MAC address.

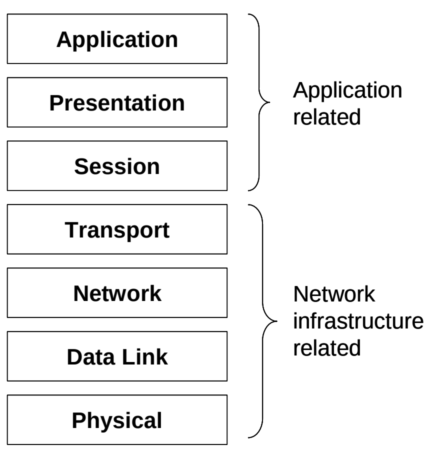
(8 marks)

e. If you were trying to connect to a computer that is on a different sub-network to your own, what MAC address would you need to use?

(2 marks)

Answer pointers

a)



Marking: 1 mark for correctly labelling each layer in sequence

b) In a layered protocol architecture data flows vertically through the layers in each end station and horizontally over the network medium between end-stations. A peer to peer protocol is defined as the communication that takes place between two layers at the same level in different end-stations.

Marking: 2 marks for communication between two layers at the same level.

c)

MAC:
Belongs to layer 2
This is a hardware address, belonging to a particular network interface. It has local, not global significance.

IP:
Belongs to layer 3
This is a global address and must therefore be unique within the whole network to wich it belongs.

Marking: 1 mark for the correct layer, 2 marks for the definition – MAC being hardware/local and IP being global/unique

d) You issue an ARP request. This will be carried in a LAN frame with the destination address set to broadcast. The data field of this LAN frame will contain the ARP request PDU. This PDU contains the IP address of the other computer, your IP address and your MAC address and a blank field to indicate that the MAC address of the other computer.

All devices on the LAN will receive this ARP request (MAC broadcast). They will each examine the ARP PDU and check the destination IP address. Only the device that has that address will respond.

The computer will therefore issue an ARP reply. This will be transported within a LAN frame that has a destination address equal to your MAC address. This PDU will contain all of the addresses that were in the original ARP request, however, the other computer's MAC address will now be included in what was previously the blank field.

Marking: ARP request: 2 marks for noting that it is sent with the MAC destination set to broadcast, 2 marks for the IP addresses of the ARP request; 1 mark for the blank MAC address field; 1 mark for determining which computer responds to the ARP request; 2 marks for the ARP reply containing the MAC address required)

In this case, the MAC address would be that of your default router.

Examiner's comments:
a) very well done – most students were able to score all the marks available.
b) well done, although some students always think that peer to peer means networking without a server (which it does in a different context).
c) most students got at least some of the points right here and quite a lot got all the points.
d) There was a lot of confusion overall about the workings of ARP, although some students understood perfectly how it works even to the level of the blank field in the ARP request. lots of students scored quite well on this, knowing that the ARP request is broadcasted and that the computer with the relevant IP address responds with its MAC address.
e) aAgood proportion of students knew that the relevant MAC address was the one on the default gateway (router).

Section B

B4

The work conducted by routers used within TCP/IP inter-networks can be broken into two separate tasks that could be named as ***packet forwarding*** and ***routing table determination.***

a)

i) Describe the actions involved in the task known as ***packet forwarding***. Include within your answer a discussion of the frequency that packet forwarding occurs and an important objective to be met by the packet forwarding task.

(9 marks)

ii) Discuss the extent to which the ***packet forwarding*** task is an activity within a single router or conducted as an activity involving multiple routers.

(3 marks)

b)

i) Describe the actions involved in the task known as ***routing table determination***. Include within your answer a discussion of the frequency that route table determination occurs and two important objectives to be met by the route table determination subtask.

(9 marks)

ii) Discuss the extent to which the ***routing table determination*** task is an activity within a single router or conducted as an activity involving multiple routers.

(4 marks)

Answer Pointers

a) i) The packet forwarding task is concerned with examining individual packets, comparing their header information with the routing tables that already exist and trying to find the best match to the destination address in the header [2 marks], and then moving the packet onto the outgoing queue for the appropriate interface [2 marks]. The packet would be consumed (or discarded) by the router it if the destination is the router itself. The packet would also be discarded if the destination is not present in the routing tables and no default route exists [1 mark]. This happens for every packet [2 marks] and thus an important objective is that the forwarding must happen very quickly [2 marks]. If the candidate misses some of the above, but provides other relevant information marks will be awarded appropriately.

a) ii) The packet forwarding subtask itself takes place in a single router using information that the router already has. Of course, the router may indeed pass a packet on to a further router if it can't deliver it directly to the final destination. [3 marks].

b) i) The routing table determination task is concerned with creating the routing tables [2 marks]. These tables will hold a set of destinations and the interfaces that lead towards them. There will also be information to note whether the destination is a network directly connected to the present router or whether it is beyond another intermediate router [3 marks]. This activity happens **MUCH** less often than does packet forwarding, it is **NOT** conducted for every packet [2 marks]! Routing table determination will probably only take place when the router has reason to believe that connectivity of the network has changed [2 marks]. If the candidate misses some of the above, but provides other relevant information marks will be awarded appropriately.

b) ii) The routing determination task is really an activity involving multiple routers [2 marks]. A router will receive information from neighbours detailing connectivity changes of which they are aware and similarly, will send them details of changes of which it is aware [2 marks].

Examiner's Comments

This question was attempted by about 31% of the candidates. Some candidates confused the activity of the creation and updating of routing tables with the activity of actual packet forwarding. The creation and updating of routing tables only happens only relatively rarely, whereas packet forwarding happens on every packet. Some candidates seem to think that when a router is about to forward a packet, it will, at that time, start to ask other routers where it should send it. This is completely wrong. As I note above, at the time of deciding how to forward a given packet, the router will ONLY use information that it already has in its routing tables.

B5

Many wire based local area networks comply with the twisted pair Ethernet (IEEE802.3) standards normally known as 10base-T or 100base-T.

a) Describe the main differences between the two types of device known as hubs (sometimes called repeating hubs) and switches (sometimes called switching hubs). Include in your answer a discussion of the devices' handling of unicast Ethernet frames and broadcast Ethernet frames. Also discuss the extent to which hubs and switches be compared to the devices known as repeaters and bridges on co-axial cable Ethernets?

(10 marks)

c) Describe the frame format used by the Ethernet. Include in your answer a discussion of at least five of the seven fields that make up an Ethernet frame.

(15 marks)

Answer Pointers

a) A repeating hub will simply take in frames arriving on its interfaces and forward it to all other interfaces [2 marks] without filtering them in any way [2 marks]. On the other hand, switches will examine the destination addresses in all arriving frames [1 mark], will forward frames only to the interface where the destination is believed to be [1 mark]. Frames with unknown destinations will be issued out of all interfaces other than the one on which they arrive [1 mark]. Broadcast frames will be issued out of all interfaces other than the one on which they arrive [1 mark]. Hubs are in many ways somewhat similar to co-axial network repeaters and switches are similar to bridges [2 marks]

b) Marks will be awarded as 3 marks for the discussions of any five fields. Should a student give a poor or incomplete description of any of their chosen five fields, but in fact mention more than five, marks will be awarded as appropriate.
The fields, in order are...

1. preamble – a sequence of repeated bytes (7) used to synchronize the clock of receivers to that being used by a transmitter
2. start of frame delimiter – a single byte field used to mark the end of the preamble and indicate that address information is about to arrive next
3. Destination address – a six byte field (although there are some circumstances when it could be two bytes) used to indicate the destination to which a packet is being sent. Also includes group destinations.
4. Source address – a six byte field (although there are some circumstances when it could be two bytes) used to indicate the source of the packet.
5. Type or length field – a two byte field. In the original Ethernet spec (and in reality mostly today) this is used to carry a packet type code. In the IEEE 802.3 standardized versions of Ethernet this carries the packet length.
6. Data field – between 46 and 1500 bytes of data. The contents could be any sort of information. It may include padding if necessary to reach the minimum 46 byte size.
7. Frame check sequence – a four byte field holding a 32 bit CRC used for error detection.

Candidates should note that the above discussion does NOT consider the modifications that would occur if virtual LANs (VLANs) were in use. VLANs are included on the syllabus for this module and could thus be examined in future papers.

Examiner's Comments

This question was attempted by about 75% of the candidates. Many answers were of quite good quality. However, in the answers to 5) a) a lot of candidates used terms such as "intelligent" or "efficient" without any other words or phrases to explain to us what they meant by the use of such a term. Students should always make sure that they justify their answers so that we, as examiners, can be confident that the student has real knowledge and understanding and is not just quoting a term they have read elsewhere. The question specifically required candidates to "discuss the devices' handling of unicast Ethernet frames and broadcast Ethernet frames", many candidates ignored this. Several candidates also said things like "switches cannot handle broadcast frames", of course, this is completely false.

B6

Error detection and correction is an important issue when we are moving information across network links.

a) Briefly describe the technique known as single bit parity checking. Include is your answer an explanation of what is meant by even parity and odd parity.

(6 marks)

b) Describe what is meant by longitudinal parity and transverse parity and how both can be used together as an error correction technique under some circumstances.

(6 marks)

d) In many circumstances, parity checking is inadequate and other techniques have to be used. Produce an outline of the technique known as Frame Check Sequence (FCS) or Cyclic Redundancy Check (CRC).

(10 marks)

e) Identify the circumstances when the use of Frame Check Sequence (FCS) would be more suitable than parity checking.

(3 marks)

Answer Pointers

a) Single parity bit checking involves the insertion of a single extra bit per chunk of data (usually a character) in such a way as to meet a defined rule [2 marks]. With even parity, the rule is that the total number set of `ones', including the data and the parity bit is an even number [2 marks]. With even parity, the rule is that the total number set of `ones', including the data and the parity bit is an odd number [2 marks]. Of course, the receiver has to know in advance what type of parity is in use so as it can decide if data has arrived correctly or has been corrupted.

b) Transverse parity is the term normally used when the chunk of data to which the parity bit applies in indeed a single character [2 marks]. Longitudinal parity is the term used when the chunk of data to which a parity bit applies is a single bit position of a sequence of characters that make up a block [2 marks]. Thus, effectively a parity byte is added to a block of characters. The students could (and perhaps probably will) give a diagram to show this idea instead of words and if they do, those first four marks would be awarded. The technique can be used as an error correcting technique if one assumes that only a single bit can become corrupt in a block in which case the single bit in the block that matches both a transverse parity check failure and a longitudinal parity bit failure would be the corrupt bit. [2 marks].

This represents a transmission with d4 of character three corrupt

pbit	**d0**	**d1**	**d2**	**d3**	**d4**	**d5**	**d6**	
1	**0**	**1**	**0**	**1**	**0**	**1**	**0**	**character one**
1	**0**	**0**	**0**	**0**	**1**	**1**	**1**	**character two**
1	**1**	**1**	**1**	**0**	**<u>1</u>**	**0**	**0**	**character three**
0	**1**	**0**	**1**	**0**	**1**	**0**	**1**	**character four**
0	**1**	**1**	**1**	**0**	**1**	**1**	**1**	**long. parity**

c) The students could choose to answer this part of the question in a fairly formal mathematical way or a more textual descriptive manner. The students could also choose to give a description more by describing how it might be implemented. Marks will be awarded for any of the possible approaches.

1. The basic idea is that the data to be transmitted is considered to be a k-bit binary number (let's call in M(x)). [2 marks]
2. This is then divided by a second binary number (the pre-agreed generator) with n+1 bits (where k > n) [2 marks].
3. This is done using the quotient/remainder approach using modulo-2 arithmetic [2 marks].
4. The original message, with the remainder (and n-bit number) attached is transmitted [2 marks].
5. The receiver divides the complete received sequence by the pre-agreed generator and the result should be zero. If so, the data has been transmitted correctly [2 marks].

d) FCS is the more appropriate technique if you believe burst errors are more likely than isolated single bit corruption errors [3 marks].

Examiner's Comments

This question was attempted by about 40% of the candidates. Some answers to part a) seemed to show a lack of understanding by some candidates. For instance, some said that "odd parity" meant adding a single bit set to 1, rather than giving the sort of details that is illustrated in the answer pointer given above. A small number of students gave good answers to part b), largely following the style of answer as given above. Part c) was ignored by many students, but other students gave really good answers. While some answers to part d) were good, others seems to imply that you could choose your error checking technique after you had received the data, clearly this is not true.