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 overall pass rate for this module has declined again and considerably. The level of answers provided by the candidates were of poor quality and although candidates attempted most of the questions it was clear that it was merely a desperate attempt to get marks. The highest mark was 59 but there were candidates with a mark of 0.

Question 3 was the most popular question however only 28% of the candidates obtained a favourable mark. It is the examiners view that candidates were not prepared for the question although it is an extremely basic networking topic. The least popular question was question 2 however it was the one with a higher pass rate.

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 prepare properly for the topics indicated in 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;
- do not prepare candidates to answer questions relating to specific scenarios.

Ten per cent of the candidates who registered for the module failed to present themselves for the examination. The centre with the largest number of candidates also had the highest proportion (26%) of registered candidates failing to appear.

Question A1

A1. Explain briefly the following terms:

a) Integrated Services Digital Network (ISDN) **(5 marks)**

b) Session Initiation Protocol (SIP) **(5 marks)**

c) Asymmetric Digital Subscriber Line (ADSL) **(5 marks)**

d) Asynchronous Transfer Mode (ATM) **(5 marks)**

e) ITU-T (formerly CCITT) **(5 marks)**

Answer Pointers

Part (a)

ISDN is a set of communication standards defined in the late 1980s. It provides for the simultaneous transmission of voice, video and data over the public switched telephone network using digital techniques. It is a circuit-switched system, which also provides access to packet switched networks. It was designed to allow digital transmission of voice and data over ordinary telephone copper wires.

Part (b)

SIP is a protocol for controlling multimedia communications sessions. It is widely used in internet telephony and in instant messaging services over IP networks. It defines the messages that are sent between endpoints and which govern the setting up, termination and other elements of a call and can be used for managing sessions consisting of several media streams. It is an application layer protocol designed to be independent of the underlying transport layer.

Part (c)

ADSL delivers data and telephony to the subscriber over the same copper wires as were used for simple telephony. It does this by using higher frequencies for the data services; a micro filter is used within the home to ensure that the higher frequencies cannot reach the telephone. Similarly at the telephone exchange filters separate out the voice and data frequencies. Because ADSL is usually marketed as a means of providing Internet access, greater bandwidth and bit rate are provided downstream than upstream. Hence the use of the term ‘asymmetric’.

Part (d)

ATM is a protocol that maps approximately on to the physical, data link and network layers of the OSI reference model. It was designed to handle both high-throughput data traffic and low-latency, real-time traffic such as voice and video. It uses asynchronous time-division multiplexing and data is transmitted in small fixed-size packets, in contrast to IP or Ethernet systems that use variable length packets. It uses a connection-oriented model in which a virtual circuit must be set up before data exchange begins.

Part (e)

The ITU Telecommunication Standardization Sector (ITU-T) is one of the three divisions of the International Telecommunication Union (ITU); it coordinates standards for telecommunications. Its mission is to ensure the efficient and timely production of standards covering all fields of telecommunications on a worldwide basis, as well as defining tariff and accounting principles for international telecommunication services.

Since the ITU-T is part of the ITU, which is a United Nations specialized agency, its standards carry more formal international weight than those of most other standards development organizations that publish technical specifications of a similar form.

Examiner’s Comments

Attempted by 77% of candidates, 18% of who attained a pass mark. Average mark for the question was 6 (out of 20) with two candidates achieving a mark of 19.

Many of the candidates that attempted this question confused ISDN with ADSL and vice versa, indicating that they had not understood the differences between these two technologies.

The vast majority of the candidates did not relate SIP as a communications protocol; instead they defined it as a general session connection protocol, e.g.

creating a session to transfer data between two computers. This demonstrated lack of understanding of the purpose and main functions of SIP.

In part D (ATM), many of the candidates only provided a simple diagram of ATM rather than explaining its function.

Overall, the question was poorly attempted, indicating lack of preparation for the topic.

Question A2

A2. This question is about IPv6.

a) Explain the terms **global unicast address** and link local address and the difference between them. **(6 marks)**

b) Explain the compressed format for writing IPv6 addresses and write the following IPv6 addresses in their shortest compressed form:

2001:0DB8:0000:1470:0000:0000:0000:0200

(4 marks)

F380:0000:0000:0000:0123:4567:89AB:CDEF

(3 marks)

c) Global unicast IPv6 addresses can be assigned dynamically in two different ways: stateless address autoconfiguration (SLAAC) and dynamic host configuration protocol v6 (DHCPv6). Describe the differences between the two methods. **(12 marks)**

Answer Pointers

Part (a)

Both terms are unicast addresses. A global unicast IPv6 address is globally unique; it is an Internet routable address and can be configured statically or assigned dynamically. (3 marks)

A link-local address is used to communicate with other devices on the same local link; it is not routable beyond the link and it is automatically configured on the interface. (3 marks)

Part (b)

A double colon (::) can replace any single, contiguous string of one or more 16-bit segments (hextets) consisting of all 0s, and can only be used once per IPv6 address. Any leading 0s (zeros) in any 16-bit section or hextet can be omitted.

2001:DB8:0:1470::200	(3 marks)
F380::123:4567:89AB:CDEF	(4 marks)

Part (c)

SLAAC allows a device to obtain its prefix, prefix length and default gateway from an IPv6 router; no DHCPv6 server is needed. It relies on ICMPv6 (Internet Control Message Protocol v6) Router Advertisement (RA) messages. When first connected to a network, a host sends a link-local router solicitation multicast request for its configuration parameters; routers respond to such a request with a router advertisement packet that contains Internet layer configuration parameters. (6 marks)

DHCPv6 works as in IPv4. It automatically receives addressing information, including a global unicast address, prefix length, default gateway address and the addresses of DNS servers using the services of a DHCPv6 server. It may receive all or some of its IPv6 addressing information from a DHCPv6 server depending upon whether option 2 (SLAAC and DHCPv6) or option 3 (DHCPv6 only) is specified in the ICMPv6 RA message. Host may choose to ignore whatever is in the router's RA message and obtain its IPv6 address and other information directly from a DHCPv6 server. (6 marks)

Examiner's Comments

This question was attempted by 48% of the candidates with only 28% of them achieved a pass mark. The average mark for this question was 10 with a maximum mark of 23.

Most of the candidates were familiar with IPv6 but only a few had understood the process for producing the compressed format of an IPv6 address. Many candidates did not know the function of SLAAC.

Question A3

A3. This question is concerned with local area networks using Ethernet technology.

a) Explain the difference between a switch and a bridge. **(6 marks)**

b) Briefly explain the difference between cut-through, fragment-free and store-and-forward switches. **(9 marks)**

c) Why are switches generally preferred to bridges except for the smallest networks? **(6 marks)**

d) How do routers operate differently from switches? **(4 marks)**

Answer Pointers

Part (a)

Bridges and switches are both devices that operate at the data link layer of the OSI reference model. A bridge connects a local area network (LAN) to another local area network that uses the same protocol. It has a one incoming and one outgoing port and filters traffic on the LAN by looking at the MAC address of the destination of the packet and forwarding it only if its destination is on the opposite side of the bridge from the sender. (3 marks)

In contrast, a switch has multiple ports. When a packet arrives at a switch it is read to determine which destination to send the data to (either a specific destination or another switch). (3 marks)

Part (b)

A cut-through switch starts forwarding a frame (or packet) before the whole frame has been received, normally as soon as the destination address is processed. This reduces the latency through the switch but means that corrupt packages may be forwarded. (3 marks)

In a fragment-free switch, forwarding doesn't start until the first 64 bytes of the frame have been received. Since many errors result in a packet being shorter than the minimum Ethernet packet length of 64 bytes, this means that a lot of corrupt packets will be detected and discarded; some, however, will still get through and the destination device must still carry out error detection checks. (3 marks)

In a store and forward switch, forwarding does not start until the whole packet has been received, meaning that the switch can check the integrity of of the packet, at the cost of more time being required for the switch to process the packet. (3 marks)

Part (c)

Switches support full duplex communication while bridges only support half-duplex. (2 marks)

Switches introduce less delay than bridges because packet forwarding is usually done by ASICs while in bridges it is usually done by software. (2 marks)

Switches have one broadcast domain per VLAN. (2 marks)

Part (d)

A router is similar in a switch in that it forwards packets based on address. But, instead of the MAC address that a switch uses, a router can use the IP address. (2 marks)

Before forwarding a packet the router will review the destination IP address. This allows the network to go across different protocols. (2 marks)

Examiner's Comments

A reasonably popular question attempted by 92% of the candidates. However, only 28% of the candidates achieved a passing mark.

Although the vast majority of candidates attempted this question, the responses were very poor, clearly indicating very little preparation and understanding of the function of switches, bridges and routers within an Ethernet network. Most of the candidates used examples from residential/home settings rather than business/corporate to describe and discuss these technologies. Almost all of the candidates that attempted the question did not understand the function of cut-through, fragment-free and store-and-forward switches.

Question B4

This question is about the network layer and its functionality.

a) Briefly describe the concepts of: i) routing information, ii) routing algorithm and, iii) autonomous system, and indicate their involvement in the routing process.

(8 marks)

b) There are three approaches to gathering and using routing information:

i. Distance-vector routing
ii. Link-state routing
iii. Path-vector routing

Briefly compare the three approaches by describing the routing information used and the way the routing algorithm works.

(9 marks)

c) Describe three main differences between RIPv1 and OSPF. **(6 marks)**

d) Summarise the difference between multicast and broadcast communication.

(2 marks)

Answer Pointers

Part (a)

Routing information: information about the topology and delays of the internet. (2 marks) Routing algorithm: The algorithm used to make a routing decision for a particular datagram, based on current routing information. (2 marks)

Autonomous system: A set of routers and networks managed by a single organisation. (2 marks)

Routers within the same autonomous system will exchange routing information, which then will be used to make routing decisions via the routing algorithm. (2 marks)

Part (b)

Distance-vector routing requires that each node exchange information with its neighbouring nodes. Each node maintains a vector of link costs for each directly attached network and distance and next-hop vectors for each destination. Each router must send a distance vector to all of its neighbours, and that vector contains the estimated path cost to all networks in the configuration. (3 marks)

In Link-state routing, when a router is initialised, it determines the link cost on each of its network interfaces. The router then advertises this set of link costs to all other routers in the internet topology. From then on, the router monitors its links costs. Whenever there is a significant change, the router again advertises its set of link costs to all other routers in the configuration. (3 marks)

Path-vector routing is used to exchange routing information between two ASs. The protocol dispenses with routing metrics and simply provides information about which networks can be reached by a given router and the ASs that must be crossed to get there. (3 marks)

Part (c)

RIPv1 is a distance-vector protocol, whilst OSPF is a link-state protocol (2 marks)

RIPv1 is a classful routing protocol, whilst OSPF is a classless routing protocol (2 marks)

RIPv1 uses hop count as metric, whilst OSPF uses bandwidth (2 marks)

Part (d)

Multicast is a one-to-many communication whilst broadcast is one-to-all.

Examiner's Comments

This was the fourth more popular question, being attempted by 60% of the candidates. Only 18% of the candidates that attempted the question obtained a pass mark. The highest mark for this question was 19 with an average mark of 6.

In the examiners' opinion, candidates failed to address in detail what was contained within routing information, just stating information for a router. Many candidates failed to address that the routing algorithm was needed to indicate how routing information was handed by different routing protocols. Many just quoted different routing protocols rather than examples of the algorithm used. Very few candidates answered what an Autonomous System was let alone how it was used.
Distance vector routing was often poorly defined with marks only usually obtained for mention of things like hop count, the trouble was that many then said it was OSPF that used the information. Main characteristic identified was periodic updates. Link state routing was again poorly described. The main characteristic identified was updates

issued when something changed rather than periodic flooding of routing tables. Path vector routing was shambolic in its description with almost no one knowing the correct details.
Section c was generally better answered but some candidates got confused with these being the wrong way round, candidates typically identified hop count and bandwidth as metrics and distance-vector vs. link state. A few managed classful versus classless. One of the common mistakes was candidates identified broadcast in its TV capacity and multicast was confusing to them.

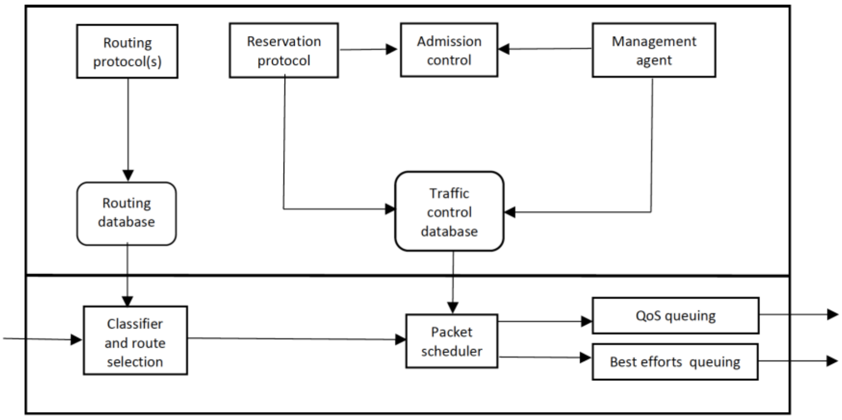
Question B5

This question is about the management of Quality of Service (QoS) in a network.

a) Briefly indicate two reasons why QoS management is required within an IP network. **(4 marks)**

b) The diagram below depicts the Integrated Services Architecture implemented in a router. Briefly indicate the function of at least two components (either from the foreground or background).

(6 marks)



c) Describe FOUR key characteristics of the differentiated services mechanism for classifying and managing network traffic that contribute to its efficiency and ease of deployment.

(8 marks)

d) Describe THREE IP metrics, defined by the IETF, that relate to the quality, performance, and reliability of Internet data delivery. Which one is used to measure jitter? **(7 marks)**

Answer Pointers

Part (a)

Voice, video and data transmission have different requirements and, furthermore, different types of data transmission may have different requirements. High latency, for example, is unacceptable for voice telephony but is not a problem for most types of data transmission.

Part (b)

Reservation protocol: it is responsible for maintaining flow-specific state information at the end systems and at the routers along the path of the flow. (2 marks)
Admission control: it determines if sufficient resources are available for the flow at the requested QoS. The determination is based on the current level of commitment to other reservations and/or on the current load on the network (2 marks)
Management agent: a network management agent is able to modify the traffic control database and to direct the admission control module in order to set admission control policies. (2 marks)
Classifier and router selection: this function determines the next-hope address for a packet based on the packet's class and its destination IP. (2 marks)
Packet Scheduler: manages one or more queues for each output port.
This question should be capped to 6 marks.

Part (c)

IP packets are labelled for differing QoS treatment using the existing IPv4 or IPv6 DS fields. Thus, no change is required to IP (2 marks)
A service level agreement (SLA) is established between the service provider and the customer prior to the use of DS. (2 marks)
DS provides a built-in aggregation mechanism. All traffic with the same DS octet is treated the same by the network service. (2 marks)
DS is implemented in individual routers by queuing and forwarding packets based on the DS octet. Routers deal with each packet individually and do not have to save state information on packet flows. (2 marks)

Part (d)

One-way delay - time taken for a packet to be transmitted across a network from source to destination (2 marks)
Round trip delay – time required by a packet to travel from a specific source to a specific destination and back again (2 marks)
Packet delay variation – is the difference in end-to-end one-way delay between selected packets in a flow with any lost packets being ignored (2 marks)
Packet delay variation is used to measure jitter (1 mark)

Examiner's Comments

Question 5 was attempted by 64% of candidates, of whom only 3% achieved a pass mark. This was by far the question with more candidates failing to answer correctly.

Many candidates had no idea about QoS about prioritising traffic or even what traffic types benefited. Many seemed to think it was a TCP issue and flow control. Candidates generally picked a few marks up here but might only identify one or two components and scored poorly. They seemed to pick components identified in question B4 as that seemed easier for them. Very poor knowledge of QoS displayed. Very poor recognition of DS characteristics or even any mention of DS/CoS/ToS bits. In line with previous comments about poor knowledge of QoS. Very poor answers to metrics, a few seemed to know that jitter was variation in delay but almost done picked up marks for any other metrics here.

Question B6

This question is about error detection and correction in data communication.

a) Consider the following example, transmitter A is transmitting character G (1110001) and using odd parity for error detection. Briefly explain what the transmitter will transmit and what will the receiver do to detect any error.

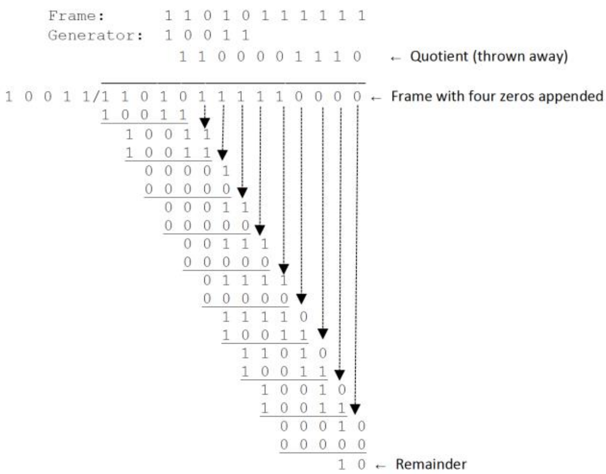
(4 marks)

b) One of the most common error-detecting codes is Cyclic Redundancy Check (CRC).

i. Briefly describes how CRC performs error-detecting. **(4 marks)**

ii. Given the frame (message), generator and remainder shown below, what would be the transmitted frame?

(3 marks)



c) A 1024-bit message is sent that contains 992 data bits and 32 CRC bits. CRC is computed using the IEEE 802, standardized, 32-degree CRC polynomial. For each of the following, explain whether the errors during message transmission will be detected by the receiver: **(10 marks)**

i. There was a single-bit error.
ii. There were two isolated bit errors.
iii. There were 18 isolated bit errors.
iv. There was a 24-bit long burst error.
v. There was a 35-bit long burst error.

d) Describe how Hamming Codes are used to perform error correction.

(4 marks)

Answer Pointers

Part (a)

The transmitter will send 11110001 (to make an odd number of 1s) (2 marks for the code). The receiver examines the received character and, if the total number of 1s is odd, assumes no error has occurred (2 marks for the explanation).

Part (b)

(i) Blocks of data entering these systems get a short check value attached (2 marks), based on the remainder of a polynomial division (2 marks) of their contents. On retrieval, the calculation is repeated and, in the event the check values do not match, corrective action can be taken against data corruption (2 marks).

(ii) The polynomial used is degree 4 therefore the message will be the frame, plus 4 zeroes minus the remainder: 11010111110010. (3 marks)

Part (c)

i. Yes. CRC catches all single-bit errors.
ii. Yes. CRC catches all double-bit errors for any reasonably long message.
iii. No. CRC may not be able to catch all even numbers of isolated bit errors.
iv. Yes. CRC catches all burst errors with burst lengths less than or equal to 32.
v. No. CRC may not be able to catch a burst error with burst length greater than 32.

Part (d)

The Hamming distance between two base-2 values is the number of bits at which they differ. If a computer transmits A, but B is received, then the number of bits that must have been switched in transmission is the Hamming distance between A and B.

Examiner's Comments

This question was attempted by 51% of the candidates with a 22% of them achieving a passing mark. The highest mark was 16 and the average 6.

Many candidates attempted this question and many knew the principle of inserting an additional bit to make the odd parity but quite a few put it in the wrong place. Many answers tried to provide a complicated definition. A few Candidates knew something about how CRC worked for error detection about the polynomial division and remainder and calculating after transmission when the signal is detected. Very few had any idea about the calculation or any working, 2-3 had the correct answer. Many candidates picked up marks here by essentially guessing the answers i.e. yes or no, not many picked up marks with any explanation/justification. Some candidates failed to give coherent answers even in a simple yes or no form. Very few candidates could explain the hamming distance or explain properly how it worked. Examples given were very poor and confusing, as they didn't seem to relate to the question.