Qu No.	Solution	Page … of …….
Marks		

Please note that both exams have identical solutions, however the level of detail expected in ES is less, and the questions are phrased to provide more guidance on how to provide the solution.

5 1a

An Ethernet frame (represented below in hexadecimal) was recorded by network monitor. Explain how this frame may be decoded to show the transport protocol that was used and the expected application.

The answer is that the packet has an IP dest address of 139.133.207.5 (`8b85 cf05`) and a protocol ID of 0x11 or decimal 17 (SAP to Transport Layer), indicating UDP.

Hexadecimal data:
MAC, Type Field (0x800) indicates this ia an IP packet::

```
001f 5b38 7354 001a 2f52 4841 0800

IP:
                                   4500
0089 bba8 4000 fe11 0e58 8b85 cc52 8b85
cf05

UDP:
0035 ccbf 0075 20df 9b49 8583 0001
0000 0001 0000 0233 3203 3230 3403 3133
3303 3133 3907 696e 2d61 6464 7204 6172
7061 0000 0c00 01c0 0f00 0600 0100 0151
8000 3403 6465 6503 6572 6704 6162 646e
0261 6302 756b 0009 646e 736d 6173 7465
72c0 3d77 c09e ed00 0054 6000 000e 1000
093a 8000 0151 80fa 8762 5a
```

Decode:

139.133.204.82.53 > 139.133.207.5.52415: [udp sum ok] 39753 NXDomain* q: PTR? 32.204.133.139.in-addr.arpa. 0/1/0 ns: 204.133.139.in-addr.arpa. [1d] SOA dee.erg.abdn.ac.uk. dnsmaster.erg.abdn.ac.uk. 2009112301 21600 3600 604800 86400 (109)

Decode at MAC layer (fixed 14 bytes):

Destination `001f 5b38 7354` (unicast)
Source `001a 2f52 4841`
Ethertype IPv4 (0x0800) --- SAP to network layer <- This MUST be known.

Decode at IP layer (variable, in this case 20 bytes):

Src IP: 139.133.204.82
Dest 139.133.207.5
IP proto UDP (17) --- SAP to Transport Layer <- This MUST be known.

UDP port = DNS (53), 0x`0035` – not required in this answer

Qu No.1	Solution	Page … of …….

Marks

8 1b

The transmitter initialises the number of transmissions of the current frame (n) to zero, and starts listening to the cable (using the **carrier sense logic (CS)** - e.g., by observing the Rx signal at transceiver to see if any bits are being sent). If the cable is not idle, it waits (defers) until the cable is idle. **It then waits for a small Inter-Frame Gap (IFG)** (e.g., 9.6 microseconds) to allow to time for all receiving nodes to return to prepare themselves for the next transmission.

Transmission then starts with the **preamble**, followed by the frame data and finally the **CRC-32**. After this time, the transceiver transmit logic is turned off and the transceiver returns to passively monitoring the cable for other transmissions. During this process, a transmitter must also continuously monitor the **collision detection logic (CD)** in the transceiver to detect if a collision occurs. If it does, the transmitter aborts the transmission (stops sending bits) within a few bit periods, and starts the collision procedure, by **sending a Jam Signa**l to the transceiver transmit logic. It then calculates a retransmission time.

If all nodes attempted to retransmit immediately following a collision, then this would certainly result in another collision. Therefore a procedure is required to ensure that there is only a low probability of simultaneous retransmission. The scheme adopted by Ethernet uses a random **back-off period**, where each node selects a random number, multiplies this by the slot time (minimum frame period, 51.2 µS) and waits for this random period before attempting retransmission. The small Inter-Frame Gap (IFG) (e.g., 9.6 microseconds) is also added.

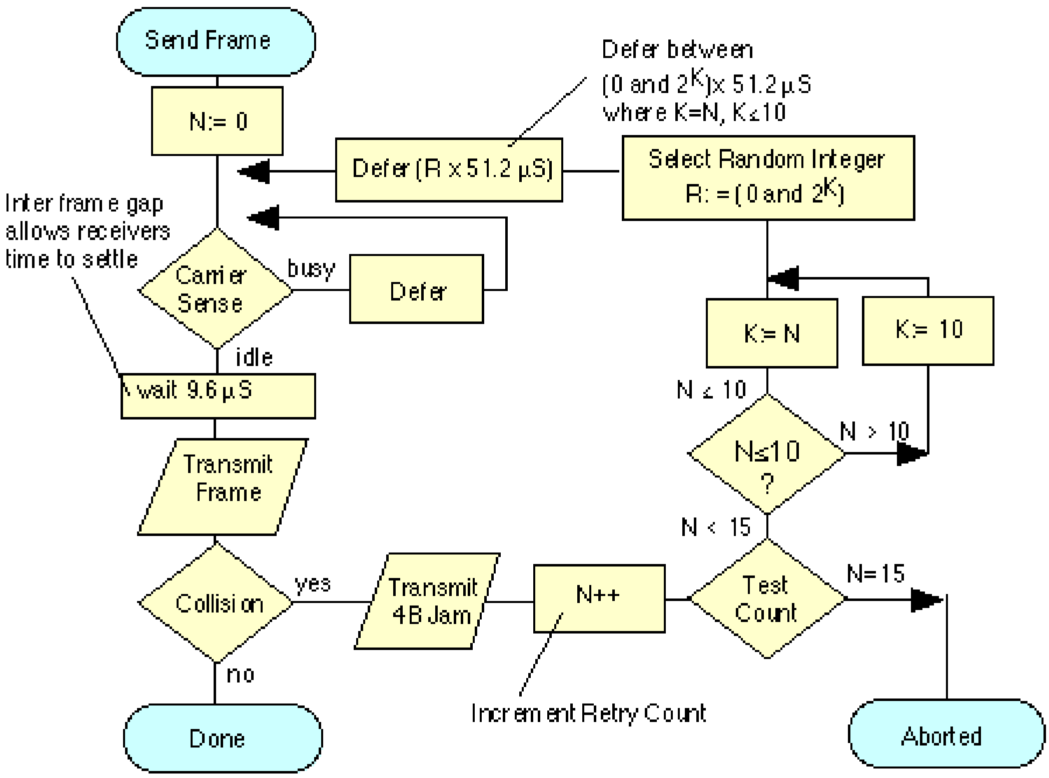
On a busy network, a retransmission may still collide with another retransmission (or possibly new data being sent for the first time by another node). The protocol therefore counts the number of retransmission attempts (using a variable N in the above figure) and attempts to retransmit the same frame up to 15 times. For each retransmission, the transmitter constructs a set of numbers:
{0, 1, 2, 3, 4, 5, ... L} where L is ([2 to the power (K)]-1) and where K=N; K<= 10;
A random value R is picked from this set, and the transmitter waits (defers) for a period
R x (slot time) i.e. R x 51.2 Micro Seconds

The scaling is performed by multiplication and is known as **exponential back-off**. This is what lets CSMA/CD scale to large numbers of nodes - even when collisions may occur. The first ten times, the back-off waiting time for the transmitter suffering collision is scaled to a larger value. The algorithm includes a threshold of 1024. The reasoning is that the more attempts that are required, the more greater the number of computers which are trying to send at the same time, and therefore the longer the period which needs to be deferred. Since a set of numbers {0,1,...,1023} is a large set of numbers, there is very little advantage from further increasing the set size.

Each transmitter also limits the maximum number of retransmissions of a single frame to 1 (N=15). After this number of attempts, the transmitter gives up transmission and discards t logging an error. In practice, a network that is not overloaded should never discard frames can effectively share the available capacity of an Ethernet segment.

Qu No.1	Solution	Page … of …….

Marks



2 1c There are two issues that are important for WiFi access:

•Some nodes in the network may be hidden from other nodes.
This can be alleviated by use of a CTS/RTS handshake.

• The WiFi base station can become a dominant source of data, and hence methods may be needed to prevent CSMA/CD capture.

Answers must at least identify the first and may also indentify the second point.

5 1d A node starts transmission by sending an 8 byte (64 bit) preamble sequence. This consists of 62 alternating 1's and 0's followed by the pattern 11. Strictly speaking the last byte which finished with the '11' is known as the "**Start of Frame Delimiter**". When encoded using Manchester encoding, at 10 Mbps, the 62 alternating bits produce a 5 MHz **square wave (an easy signal for the DPLL to gain lock at the receiver)**. When the special pattern (11), is received, the Ethernet receive interface starts collecting the bits into bytes for processing by the MAC layer.

The start of frame delimiter allows time for the receiver in each node to achieve lock of the receiver Digital Phase Lock Loop which is used to synchronise the receive data clock to the transmit data clock. At the point when the first bit of the preamble is received, each receiver may be in an arbitrary state (i.e. have an arbitrary phase for its local clock). **During the preamble it learns the correct phase, but in so doing it may miss (or gain) a number of bits.**

Students should provide appropriate diagrams in answers to get full marks (or provide detailed explanation instead).

5 1e Manchester encoding is a synchronous clock encoding technique used by the physical layer to encode the clock and data of a synchronous bit stream. In the Manchester encoding shown, a logic 0 is indicated by a 0 to 1 transition at the centre of the bit and a logic 1 is indicated by a 1 to 0 transition at the centre of the bit. Note that signal transitions do not always occur at the 'bit boundaries' (the division between one bit and another), but that there is always a transition at the centre of each bit. The Manchester encoding rules are summarised below:

Original Data	Value Sent
Logic 0	0 to 1 (upward transition at bit centre)
Logic 1	1 to 0 (downward transition at bit centre)

In some cases you will see the encoding reversed, with 0 being represented as a 0 to 1 transition. The two definitions have co-existed for many years. The Ethernet IEEE standards (10 Mbps) describe a Logic 0 is sent as 0 to 1 transition, and a Logic 1 as a one to zero transition (where a zero is represented by a less negative voltage on the cable). Note that because many physical layers employ an inverting line driver to convert the binary digits into an electrical signal, the signal on the wire is the exact opposite of that output by the encoder. Differential physical layer transmission, (e.g. 10BT) does not suffer this inversion.

A Manchester encoded signal contains frequent level transitions which allow the receiver to extract the clock signal using a Digital Phase Locked Loop (DPLL) and correctly decode the value and timing of each bit. To allow reliable operation using a DPLL, the transmitted bit stream must contain a high density of bit transitions. Manchester encoding ensures this, allowing the receiving DPLL to correctly extract the clock signal.

{0 1 1 0}, therefore encodes to HI-LO-LO-HI-LO-HI-LO-HI-HI-LO or the exact inverse of this. Students must explain the reasons for their answers for full marks.

10 2a The bridge learns which MAC addresses belong to the computers on each connected subnetwork by **observing the source address** values which originate on each side of the bridge. This is called “learning”. The learned addresses are stored in the corresponding interface address table. Once this table has been setup, the bridge examines the **destination address** of all packet, and forwards them only if the address does not correspond to the source address of a computer on the local subnetwork (it may **flood** an unknown address). A system administrator may override the normal forwarding by inserting entries in a **filter table** to inhibit forwarding between different workgroups (for example to provide security). A good answer should indicate that the cache is **aged**.

Summary:

MAC Sources address observed for learning
Associated with a port in the address table
MAC Destination address observed for forwarding
Learned addresses -> forward only to specified port
Discard frames to own address
Flood frames with unknown addresses to all ports
Aging is required and re-learning when computers change the connected port.

5 marks for a basic answer.
Labelled diagrams (or very detailed explanations) should be used to illustrate the answer for full marks.

3 2b

Explain the role of a frame Cyclic Redundancy Check (CRC).

Cyclic Redundancy Check (also known as a Frame Check Sequence). The link layer CRC protects the frame from corruption while being transmitted over the physical mediuym (cable). The CRC is removed by routers - as partr of the processing. A new CRC is added if the packet is forwarded by the router on another Ethernet link. While the packet is being processed by the router the packet data is protected by the CRC. Router processing errors may otherwise pass undetected.

The 32-bit CRC provides error detection in the case where line errors (or transmission collisions in Ethernet) result in corruption of the MAC frame. Any frame with an invalid CRC is discarded by the MAC receiver without further processing. The MAC protocol does not provide any indication that a frame has been discarded due to an invalid CRC.

The CRC is also used to verify correct bit-timing, i.e. to identify the DPLL has not missed or added bistream following a timing error.

6 2c ***Unshielded Twisted Pair* (UTP) cabling was originally used as the physical layer for 10BT LANs. What challenges were faced when using this links operating at 100 Mbps?**

Issues faced - Cross talk and restricted bandwidth, requiring new PHY technology.

100BASE-T is the provides 100 Mbit/s Ethernet in either half-duplex (using CSM/CD) or full-duplex forms. 100BASE-TX runs over two pairs of wires in category 5 unshielded twisted pair cable. Like 10BASE-T, the normal pairs are coloured orange and green pairs (using pins 1, 2, 3 and 6 of the RJ-45 connector). This cable has a bandwidth of less than 100 MHz. Diagrams showing frequency response of typical cable may be helpful.

The bi-phase Manchester encoding can consume up to approximately twice the bandwidth of the original signal (20 MHz). While this was of little concern in coaxial cable transmission, the limited bandwidth of necessitated a more efficient encoding method 100 Mbps using a 4b/5b MLT code. A scheme using 4B5B binary encoding therefore generates a series of 0 and 1 bits clocked at 125 MHz; the 4B5B encoding provides DC equalisation and spectrum shaping. 4B5B works by mapping each group of four bits (a 1/2 of a byte) to one group of 5 bits.

4B/5B encoding is a type of 'Block coding'. This processes groups of bits rather than outputting a signal for each individual bit (as in Manchester encoding). A group of 4 bits is encoded so that an extra 5th bit is added. Since the input data is taken 4-bits at a time, there are 2^4, or 16 different bit patterns. The encoded bits use 5-bit, and hence have 2^5 or 32 different bit patterns. As a result, the 5-bit patterns can always have two '1's in them even if the data is all '0's a translation occurs to another of the bit patterns. This enables clock synchronisation, required for reliable data transfer.

Since there are (2^5) 32 possible combinations of 5 bits, and there are only (2^4) 16 combinations of 4 bits one half the patterns are unused. The chosen set of 16 5-bit patterns are those with the most transitions, this ensures clocking information is present in the signal (for locking the receiver DPLL). This results in a bandwidth increased of 25%.

Cross-Talk requirements / RF Emission led to the need for a scrambler. The data is finally sent as a 3-level physical waveform known as MLT-3. MLT-3 cycles through a set of voltage levels {-1, 0, +1}, to indicate a 1-bit. The signal stays the same when transmitting a 0 bit. It takes four 1 bits to generate a complete cycle, this the maximum fundamental frequency is reduced to one fourth of the baud rate.

This combined scheme of 4b/5b with MLT-3 encoding leads to a waveform of 31.25 MHz, well within the specification for Unshielded Twisted Pair Cabling. Diagrams may be useful.

Answer could also highlight the issue of inter-frame gaps, identifying how the 10B2 specification a
technologies. Answer may also wish to discuss half and full duplex operation and design constraints
1000BT. No attempt was made to support the older co-axial cable installations, instead these must b
UTP or Fibre.

6 2d **What new techniques were introduced in the Gigabit Ethernet over UTP standard to allow an order of magnitude increase in the capacity over that offered by *Fast Ethernet*?**

Gigabit Ethernet utilises five levels and 8b/10b encoding, to provide even more efficient use of the limited cable bandwidth, sending 1 Gbps within approx 100 MHz of bandwidth (i.e. the capacity of a UTP Cat5e cable.

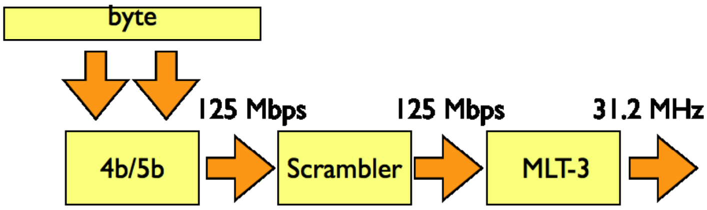
Using 8b/10b encoding each byte of data is assigned a 10 bit code. The byte is split up into the 3 most significant bits and the 5 least significant bits. This is then represented as two decimal numbers with the least significant bits first e.g. for the octet 101 00110 the result is the decimal 6.5. 10 bits are used to create this code and the naming convention follows the format /D6.5/. There are also 12 special codes which follow the naming convention /Kx.y/.

The 10 bit code must contain either five ones and five zeros, or four ones and six zeros, or six ones and four zeros. This prevents a sequence of too many consecutive ones and zeros, assisting clock synchronisation. Two 'commas' are used to aid in bit synchronisation, these 'commas' are the 7 bit patterns 0011111 (+comma) and 1100000 (-comma).

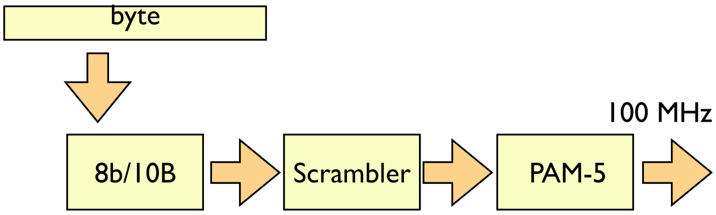
To maintain a DC balance, a calculation called the Running Disparity calculation is used to keep the number of '0's transmitted the same as the number of '1's transmitted. The method uses 10 bits for each 8 bits of data (byte) and therefore increases the rate required to send the data.

A 1Gbps the line speed results in a transmission rate of 10/8 x 1 = 1.25Gbps. In Gigabit Ethernet this rate is then reduced using PAM-5 a 5-level code (achieving less bandwidth than possible with a 3 level code).

The interface encoded byte of data generates a 10 bit code that is scrambled and converted into a physical layer signal by mapping pairs of bits using a 5-level Pulse-Amplitude-Modulation (PAM). The diagram below may be useful in this explanation:



Fast Ethernet Line Interface for 100 BT



Gigabit Ethernet Line Interface for 1000 BT

8	3a	The set of fields that are modified in the Layer 2 and Layer 3 protocol headers as a router processes and forwards an IP packet are: ETHER: Destination - replaced by next hop Layer 2 address ETHER: Source - replaced by routers outgoing interface address IP: Total length = 40 bytes (unchanged – unless options were updated) IP: Flags = (unchanged – unless DF=0 and fragmentation applied) IP: Fragment offset = (unchanged – unless fragmentation applied) IP: Time to live = reduced by one (unless 1 already, then discarded) IP: Header checksum = always updated , since TTL decremented A more complete analysis is given below: ETHER: ----- Ether Header ----- ETHER: ETHER: Destination - replaced by next hop Layer 2 address ETHER: Source - replaced by routers outgoing interface address ETHER: Ethertype - 0800 (IP) – as in received frame ETHER: IP: ----- IP Header ----- IP: IP: Version = 4 (unchanged) IP: Header length = 20 bytes (unchanged) IP: Type of service = 0x00 (unchanged, unless ECN marked) IP: Total length = 40 bytes (unchanged – unless options were updated) IP: Identification = 43773 (unchanged) IP: Flags = (unchanged – unless DF=0 and fragmentation applied) IP: Fragment offset = (unchanged – unless fragmentation applied) IP: Time to live = reduced by one (unless 1 already, then discarded) IP: Protocol = (unchanged) IP: Header checksum = always updated , since TTL decremented IP: Source address = (unchanged) IP: Destination address = (unchanged) IP: Options not normally present. Transport: (unchanged, operates end-to-end) - Students may supply diagrams.

5 3b **A *router* can forward 100,000 packets in each second. What is the maximum *Utilisation* that may be achieved when it sends the smallest allowed size of Ethernet frame over a 10 Mbps Ethernet interface?**

Smallest sized Ethernet frame is 64B (incl CRC) + 8B preamble = 72 B (2 marks)
(neglecting the small Inter-Frame Gap (IFG) (e.g., 9.6 microseconds – answers may include this).

multiply by 100,000
Divide by physical layer rate 100E6 and multiply by 100 to return a percentage (2 marks)

=570% <- Student should spot this is more than 100% (1 mark), and that the media is therefore fully utilised.

5 3c

A system sends one Internet Control Message Protocol (ICMP) message with 1000 bytes of data each second over a 10 Mbps cable segment. What is the resulting utilisation?

Calculate total size of frame = 8+14+20+8+1000+4 =1054 bytes
(neglecting the small Inter-Frame Gap (IFG) (e.g., 9.6 microseconds – answers may include this).

Convert to bits (x8)

Divide by physical layer rate 10E6 and multiply by 100 to return a percentage

Result is 0.08%

7 3d

The term address resolution refers to the process of finding an address of a computer in a network. The address is "resolved" using a protocol in which a piece of information is sent by a client process executing on the local computer to a server process executing on a remote computer. The information received by the server allows the server to uniquely identify the network system for which the address was required and therefore to provide the required address. The address resolution procedure is completed when the client receives a response from the server containing the required address.

The Domain Name Service is an example of a client/server system which is used by the Internet Protocol (IP) Suite to resolve the logical names of nodes in an IP network to an IP address (see also arp - which is used to resolve Ethernet addresses to IP addresses).

This example below considers a login from a computer X to a remote computer Y using a DNS server Z. The process is described in six steps:

A client program starts on the local computer (X) and attempts to resolve the network layer address of the remote computer from a known name using a known dns server (Z).

A dns query is sent to the server in an IP packet from X to Z.

The server (Z) processes the query and consults local dns entries and (possibly) the entries of other remote dns servers.

The dns server returns a response with the requested information (assuming success) in an IP packet from Z to X.

The local computer (X) then makes a direct connection to the remote computer (Y).

The remote computer starts a process (server) to handle the requested login. All further packets between X and Y are directed to the respective client and server processes.

Students should provide detailed notes and answers may include appropriate diagrams.

6 4a A switch works within the link layer (layer 2) of the OSI reference model. The format of PDUs at this layer in a LAN is defined by the Ethernet frame format (also known as MAC - Medium Access Control) consists of two 6 byte addresses and a one byte protocol ID / length field. The address field allows a frame to be sent to single and groups of stations. The MAC protocol is responsible for access to the medium and for the diagnosis of failure in either the hardware or the cabling.

The bridge learns which MAC addresses belong to the computers on each connected subnetwork by observing the source address values which originate on each side of the bridge. This is called "learning". The learned addresses are stored in the corresponding interface address table. Once this table has been setup, the bridge examines the destination address of all packet, and forwards them only if the address does not correspond to the source address of a computer on the local subnetwork. A system administrator may override the normal forwarding by inserting entries in a filter table to inhibit forwarding between different workgroups (for example to provide security).

Switch X **forwards** the frame:

MAC Source address of A observed by switch X for learning
Associated with left port in the address table (each frame)

MAC Source address of C observed by switch for learning

MAC Destination address (A) observed for forwarding
Learned addresses -> forward only to specified port connecting A (left)

Students need to explain why there answer is correct for full marks.

Switch Y learns and **discards** the frame:

MAC Source address of A observed by Y switch for learning
Associated with a port in the address table (left port)

MAC Source address of C observed by switch for learning

MAC Destination address (A) observed for forwarding
Learned addresses -> is on the same port as the received frame (left)
Discard frame (don't forward back to own interface)

Students need to explain why there answer is correct for full marks.

[8 marks]

Students could also mention:

Flood frames with unknown addresses to all ports (As broadcast L2)
Aging required and re-learning when computers change the port they are connected

[2 marks]

6 4b **Computer C sends a packet using the type value of 0x800 to computer A and one frame to computer D. For each of the two frames, sketch the MAC header of the *received* frame showing all MAC addresses.**

First frame Received frame: Dst:A **Src:C**, Type 0x800

The packet in this case is directed to the router, since D belongs to a different subnetwork. Router forwards the packet to D, using its own source address. Students must differentiate role of the router

Received frame: Dst:D **Src:Router** Type 0x800

The key issue is to identify the role of the addresses in routing!

8 4c

In the case of IP routers, forwarding is performed at the network layer, rather than the link layer. The key points here are that routers in the network have a perspective of the correct path to use to reach a destination. This is acquired via a L3 routing protocol (or by static configuration). This allows routers to take advantage of alternate paths. A topology including loops and parallel links is common, and traffic can flow over any link available to reach the required destination.

Routers connect two or more IP networks, or an IP network to an internet connection. A router consists of a computer with at least two network interface cards supporting the IP protocol. The router receives packets from each interface via a network interface and forwards the received packets to an appropriate output network interface. Received packets have all link protocol headers removed, and transmitted packets have a new link header added prior to transmission.

The router uses the information held in the IP header to decide whether to forward each received packet, and which network interface to use to send the packet.The packet is forwarded to an appropriate output interface, corresponding to the "best" path to the destination specified in the destination address of the IP packet header.

The routing and filter tables resemble similar tables in link layer bridges and switches. Except, that instead of specifying link hardware addresses (MAC addresses), the router table specify network (IP addresses). The routing table lists known IP destination addresses with the appropriate network interface to be used to reach that destination. A default entry may be specified to be used for all addresses not explicitly defined in the table. A filter table may also be used to ensure that unwanted packets are discarded. The filter may be used to deny access to particular protocols or to prevent unauthorised access from computers by discarding packets to specified destination addresses.

All IP packets contain a **TTL value** that determines the number of router hops that a packet may be routed, this is decremented by most routers (although routers can be configured to reduce the value by more than 1) It is used to prevent routing loops, and ensures topologies with loops do not result in packets that circulate indefinitely.

At the output interface, the packet (together with a new link layer header) is placed into a transmit queue until the **link layer processor** is ready to transmit the packet. This, like the receive queue. Each out-going packet requires a new link layer protocol header to be added (encapsulation) with the destination address set to the next system to the receive the packet. The link protocol controller also maintains the **hardware address table** associated with the interface. This usually involves using the Address Resolution Protocol (arp) to find out the hardware (MAC) addresses of other computers or routers directly connected to the same cable (or LAN). The packet is finally sent using the media interface with the hardware address set to the next hop system.

5 4d **Five Ethernet switches are connected according to the diagram. Provide notes commenting on the strengths and weaknesses of this design.**

The switches as shown form a loop. One advantage of this topology is that it provides **redundancy** that can protect the network from the **failure** of equipment or communications links.

Layer 2 switches will however lead to **looping** of packets (amplification). Unmanaged bridges must form a tree, and not a ring. That is, there must be only one path between any two computers. If more than one parallel path were to exist, a loop would be formed, resulting in endless circulation of frames over the loop. This would soon result in overload of the network.

To prevent this happening, the IEEE (in IEEE 802.1D) has defined the **Spanning Tree Algorithm** (STA) which automatically detects loops and disables one of the parallel paths. The Spanning Tree Algorithm may also be used to build fault-tolerant networks, since if the chosen path becomes invalid (e.g. due to a cable / bridge / switch fault), and an alternate path exists, the alternate path is enabled automatically.

8 5a **Use the LAN shown in Figure 3 and the information in Table 1 to explain the process by which computer A determines the *Medium Access Control (MAC)* address to be used to reach the computer B and a remote computer C connected via the Internet connection. Your answer should indicate the destination IP address and MAC address used in each case.**

Each IP address consists of two parts, the network part (identifying the network number, or LAN collision domain, to which the computer is attached) and the host part (which identifies the host within the local network).

The IP network ID is identified as the bit-wise logical AND of the 32-bit IP address with another 32-bit quantity, the net mask. All systems with the same network number share the same netmask (sometimes called a "subnet mask"). This has a bit with a logical '1' for each bit that is a part of the network number, and a logical '0' for each bit which is a part of the host number.

The forwarding method compares the source and destination network IDs at the received interface.

If the two match, use ARP to find a MAC address, and send directly.

If they do not match, send the packet to a router (this may require a route-lookup - if routing is used, otherwise the default route is used). If the router's MAC address is not in the ARP cache, then ARP is first used.

The address resolution protocol is used by the Internet Protocol (IP) to map IP network destination addresses to the hardware addresses used by a link protocol. The protocol operates below the network layer as a part of the OSI link layer, and is used when IP is used over Ethernet.

Hence in this question:

Dest MAC = 00:01:00:00:02:00
Dest IP = 140.0.2.3, apply 24 bit netmask, hence dest network id = 140.0.2. this is the same as the source, the destination is directly reachable. ARP is used to find the hardware address, and then frame is sent to this MAC address:
Destination 140.0.2.3, src 140.0.2.2:

Destination 201.77.188.166, src 140.0.2.2

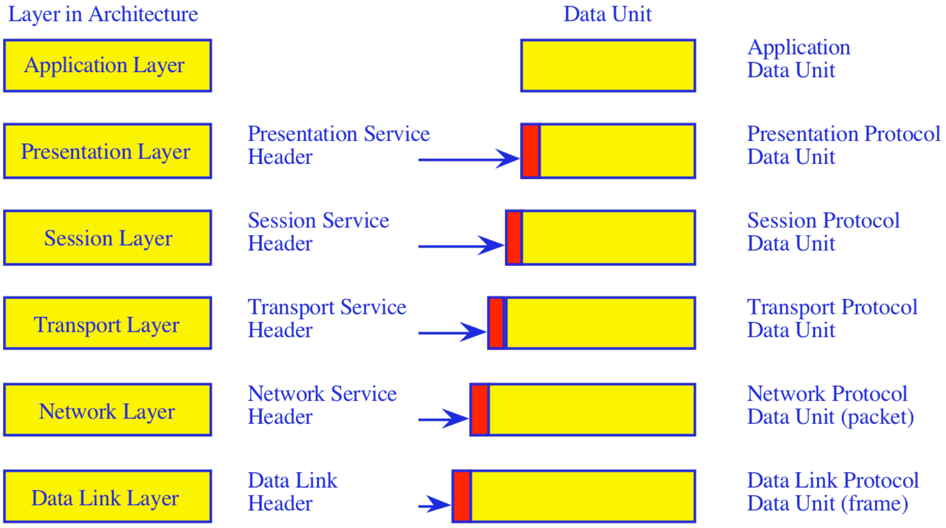
Dest MAC = 00:02:00:00:01:00 – note identify which interface is local!

Dest IP = 201.77.188.166

2	5b	The 12 hex digits of source address consist of the first/left 6 digits (which should match the vendor of the Ethernet network interface) and the last/right 6 digits which specify the interface serial number for that interface controller vendor (this gives 256 cubed addresses - or 16.78 million separate serial numbers). This allows each vendor to assign their own interface serial numbers (this is a flat addressing scheme), but also allows protocol monitors to examine the first 3 bytes of a frame address to determine the manufacturer of the interface card being used. The addresses associated with interface cards are source addresses, not multicast nor broadcast, so the second hex digit (reading from the left) will be even, not odd. Vendor MAC addresses (i.e. the first 3 bytes of a MAC source address, the OUI) are purchased from the IEEE.

15 5ci **Candidates must ANSWER one of 5Ci or 5cii**

4 marks for the diagram showing OSI Layers, correctly labeled., additional marks for detailed explanation.



When data (PDU) is passed by the application process to a layer, the layer processes it. It then prefixes a protocol header (PCI) and passes the packet (together with the new header) to the layer below (as an SDU). This layer in turn processes the data and prefixes an additional new header. Each
layer treats the assemblage of information from higher layers as data, and does not worry about its contents. This process continues until the packet reaches the Physical Layer, where they then become a sequence of bits that can be sent across a specific Physical Medium (cable, wireless link, etc).

The physical layer serialises the packet (i.e. converts it to a series of bits) and sends it across a cable or communications circuit to the destination. At the receiver, the remote system reassembles the series of bits to form a frame and forwards the frame for processing by the link layer. This removes the link
layer header, and passes it (up) to the next layer. The processing continues layer by layer as the packet rises up the remote protocol stack, until finally the original packet data is sent to the remote application. The upper three layers, or *middleware* (application, presentation, and session) are primarily concerned with ensuring that information is delivered in correct and understandable form.

The transport layer forms an interface between these groupings.

Key issues to be identified - the role of the network layer, it’s hop-by-hop forwarding and the way in which it routes traffic between systems.

15 5cii The “traceroute” program also contains a client interface to CGMP. Like the “ping” program, it may be used by a user to verify an end-to-end Internet Path is operational, but also provides information on each of the Intermediate Systems (i.e. IP routers) to be found along the IP Path from the sender to the receiver. Traceroute uses ICMPecho messages. These are addressed to the target IP address. The sender manipulates the TTL (hop count) value at the IP layer to force each hop in turn to return an error message.

The program starts by sending an ICMP Echo request message with an IP destination address of the system to be tested and with a Time To Live (TTL) value set to 1. The first system that receives this packet decrements the TTL and discards the message, since this now has a value of zero. Before it deletes the message, the system constructs an ICMP error message (with an ICMP message type of “TTL exceeded”) and returns this back to the sender. Receipt of this message allows the sender to identify which system is one link away along the path to the specified destination.

The sender repeats this two more times, each time reporting the system that received the packet. If all packets travel along the same path, each ICMP error message will be received from the same system. Where two or more alternate paths are being used, the results may vary.

If the system that responded was not the intended destination, the sender repeats the process by sending a set of three identical messages, but using a TTL value that is one larger than the previous attempt. The first system forwards the packet (decrementing the TTL value in the IP header), but a subsequent system that reduces the TTL value to zero, generates an ICMP error message with its own source address. In this way, the sender learns the identity of another system along the IP path to the destination.

This process repeats until the sender receives a response from the intended destination (or the maximum TTL value is reached).

Example:

```
>traceroute bbc.co.uk
traceroute to bbc.co.uk (212.58.224.131), 64 hops max, 40 byte packets
 1  10.10.10.1 (10.10.10.1)  51.940 ms  18.491 ms  1.260 ms
 2  lo0-plusnet.ptn-ag2.plus.net (195.166.128.53)  49.263 ms  55.061 ms  53.525 ms
 3  ge1-0-0-204.ptn-gw2.plus.net (84.92.3.106)  139.647 ms  52.525 ms  127.196 ms
 4  gi1-1-22.ptn-gw5.plus.net (212.159.4.6)  76.505 ms  57.524 ms  52.404 ms
 5  rt0.thdo.bbc.co.uk (212.58.239.25)  89.200 ms  49.666 ms  144.629 ms

6  212.58.238.133 (212.58.238.133)  48.786 ms  68.650 ms  51.599 ms
```

Diagrams may optionally be provided to show the effect of TTL scoping.

Course co-ordinator ……………………..

G Fairhurst

Scrutineer ……………………………….

J Watson

Examining/Exam Admin/2009-2010

Circuit switching is the most familiar technique used to build a communications network. It is used for ordinary telephone calls. It allows communications equipment and circuits, to be shared among users. Each user has sole access to a circuit (functionally equivalent to a pair of copper wires) during network use. Network use is initiated by a connection phase, during which a circuit is set up between source and destination, and terminated by a disconnect phase. After completion of the connection, a signal confirming circuit establishment (a connect signal in the diagram) is returned; this flows directly back to node A with no search delays since the circuit has been established. Transfer of the data in the message then begins. After data transfer, the circuit is disconnected; a simple disconnect phase is included after the end of the data transmission. Delays for setting up a circuit connection can be high, especially if ordinary telephone equipment is used. Call setup time with conventional equipment is typically on the order of 5 to 25 seconds after completion of dialling.

Message switching sends a complete message, forwarding it hop-by-hop through a network of nodes with internal storage. At each node the message is forwarded as the outbound link becomes available. Since the message may be competing with other messages for access to facilities, a queuing delay may be incurred while waiting for the link to become available. It repeats this process until it reaches its destination.

Circuit setup delays are replaced by queuing delays. Considerable extra delay may result from storage at individual nodes. A delay for putting the message on the communications link (message length in bits divided by link speed in bps) is also incurred at each node en route. Message lengths are slightly longer than they are in circuit switching, after establishment of the circuit, since header information must be included with each message; the header includes information identifying the destination as well as other types of information.

Packet switching is similar to message switching using short messages. Any message exceeding a network-defined maximum length is broken up into shorter units, known as packets, for transmission; the packets, each with an associated header, are then transmitted individually through the network. The fundamental difference in packet communication is that the data is formed into packets with a pre-defined header format (i.e. PCI), and well-known "idle" patterns which are used to occupy the link when there is no data to be communicated.

A packet network equipment discards the "idle" patterns between packets and processes the entire packet as one piece of data. The equipment examines the packet header information (PCI) and then either removes the header (in an end system) or forwards the packet to another system. If the out-going link is not available, then the packet is placed in a queue until the link becomes free. A packet network is formed by links which connect packet network equipment.

There are two important benefits from packet switching:

- The first and most important benefit is that since packets are short, the communication links between the nodes are only allocated to transferring a single message for a short period of time while transmitting each packet. Longer messages require a series of packets to be sent, but do not require the link to be dedicated between the transmission of each packet. The implication is that packets belonging to other messages may be sent between the packets of the message being sent from A to D. This provides a much fairer sharing of the resources of each of the links.
- Another benefit of packet switching is known as "pipelining". Pipelining is visible in the figure above. At the time packet 1 is sent from B to C, packet 2 is sent from A to B; packet 1 is sent from C to D while packet 2 is sent from B to C, and packet 3 is sent from A to B, and so forth. This simultaneous use of communications links represents a gain in efficiency, the total delay for transmission across a packet network may be considerably less than for message switching, despite the inclusion of a header in each packet rather than in each message.

Note: additional diagrams for all answers available on course web site.