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Instructions:

Duration	3
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Requirements:	
Handout	
MCQ	
Statistical Tables	Y
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Log Graph Paper	
Other Material	
No. of Pages	5
Department(s)	Electronic Engineering

1. *(a)* Explain how the *class-based* addresses used in IP version 4 offers a *hierarchical* addressing scheme. Contrast the operation of routers using the traditional *class based* and the more contemporary *classless* routing approach in IP networks under the following headings:

- Address space usage efficiency,
- Complexity of route determination for an incoming IP packet,
- Scalability.

[8 marks]

(b) The header of an incoming IP packet arriving at an IP node contains a number of mandatory parameters. Describe the action of the IP node when analysing the following IP header fields:

- Checksum,
- Time To Live (TTL),
- Type of Service (ToS).

[6 marks]

(c) The IP packet header also supports *fragmentation and reassembly* of IP packets. Explain the role of the *Maximum Transmission Unit* in determining whether IP packet fragmentation is required at an IP node and, using the example context below, outline the role of the IP header fields in supporting such fragmentation.

Example Network Context

Network 1 has an MTU of 1000 bytes
Network 2 has an MTU of 500 bytes
A payload of 1600 bytes is to be transmitted from a host on Network 1 to a host on Network 2.

[6 marks]

2. *(a)* An application developer has to decide between the use of *TCP* and *UDP* as the transport mechanism for infrequently transferring a block of data from one computer system to another. Clearly explain how **TCP** implements a reliable transport mechanism while **UDP** offers an efficient transport mechanism particularly if the size of the block of data is small.

If the size of the block of data is 100 bytes, estimate the efficiency (as the ratio of data bytes transferred to the TOTAL number of bytes transmitted over the network) of both UDP and TCP for carrying out this exchange under the following set of assumptions:

- IPv4 packets with only mandatory header fields (20 bytes),
- TCP segment header only contain mandatory fields (20 bytes),
- UDP segment header size is 8 bytes,
- Additional fields added by data link layer onto all frames (10 bytes).

If the average round trip time (RTT) for the network is currently 250 ms, estimate the total time for which the TCP/UDP port is in use for this transfer (i) if no IP packet loss takes place and (ii) if the IP packet containing the block of data is lost in the IP network. In the latter case, you may assume that the TCP timeout period is 500 ms and that for a UDP implementation that the developer has configured a fixed 1 second acknowledgement timeout period.

Your answer should include a sketch showing the various IP packets transferred in completing the data transfer.

[10 marks]

(b) The Jacobson-Karels algorithm for determining the TCP timeout duration operates by estimating the *mean* and *standard deviation* of the round trip times (RTT) experienced by TCP segments. Explain what the disadvantage would be in having this time-out period set to be either (i) *too long* or (ii) *too short.*

State the equations used by this algorithm to:

(i) Estimate the mean and standard deviation of the RTT,
(ii) Determine the TCP timeout period from the values estimated in (ii).

Clearly state any restriction that exists regarding the use of a measured RTT in calculating the above.

A TCP implementation measures the round trip time (RTT) for a number of specific TCP segment as 120 ms, 110 ms and 100 ms, in that sequence. Determine the TCP timeout period at the end of this period of time using the Jacobson-Karels algorithm with (μ=1, ϕ=4, δ=0.2 and an initial estimated RTT of 100 ms and an initial deviation estimate of 10 ms)

An alternative approach for determining the TCP timeout duration is proposed for use in a network. In this approach, the **actual** mean and **actual** standard deviation of three sample RTT measurements is calculated and used in equation (ii) above. Determine, the TCP timeout period which would be calculated using this new algorithm after processing of the measured TCP segment RTTs outlined above.

Hence, if the true probability distribution function for the RTT was a normal distribution with mean 110 ms and standard deviation of 20 ms, determine what percentage of TCP segments would be unnecessarily re-transmitted for EACH of the two algorithms (assuming no packets are dropped by the network).

[10 marks]

3. *(a)* Connection oriented packet (cell) switched data networks are typically characterised as offering the following when compared to connectionless networks:

- Route determination prior to data transfer,
- Faster packet (cell) switching based on a *Virtual Circuit Identifier*,
- A guarantee of packet (cell) ordering at the recipient node with easy detection of packet (cell) loss,
- Little robustness to node or link failure.

For each of these points, clearly explain how the characterization is representative of the operation of an *Asynchronous Transfer Mode (ATM)* network.

[8 marks]

(b) Two different applications, one a real time (fixed bit rate) voice application and the second a non-real time data transfer application are to operate over an ATM network. Outline the specific support provided by the *ATM Adaptation Layers (AAL)* for each type of application under the following headings:

(i) Support for detection of single cell loss,
(ii) Support for detection of multiple consecutive cell losses,
(iii) Handling of jitter and cell payload loss.

[6 marks]

(c) IP packets carrying generic data to be stored in files on a remote computer system have an average size of 3200 bits. These files need to be transmitted over an ATM network as part of their route between source and destination hosts.

(i) Identify ALL ATM adaptation layers which could be used by this application and hence calculate the efficiency with which these AALs could transfer this service data unit,
(ii) If the 3200 bit packets contains a VoIP speech call, indicate which AAL might be used and the efficiency associated with the use of that AAL if the speech codec in use operates using a FIXED bit rate OR a VARIABLE bit rate.

Figure 1 located at the end of this paper may be of use in completing this question.

[6 marks]

4. *(a)* Explain what you understand by the term *flow* in a connectionless network, such as an IP network, and explain why flow detection is said to be implemented using *soft state* information in routers.

[4 marks]

(b) Describe the operation and clearly identify a relative advantage AND disadvantage of each of the following queuing paradigms.

- (i) Single FIFO,
- (ii) Priority based queues,
- (iii) Per flow based queues.

A router implementing a "per flow" queuing paradigm currently has three flows established. The table below outlines the arrival time (in system clock ticks), source flow and packet length of incoming packets to this router.

Packet	Source Flow	Arrival Time (Clock Ticks)	Frame Length (Bits)
A	1	0	200
B	1	40	100
C	2	120	100
D	3	150	100
E	1	190	110
F	3	210	150
G	2	240	200
H	3	270	150
I	2	310	100

Determine the order in which the packets will be transmitted if the following queue servicing algorithms are in operation:

- (i) Fair queuing,
- (ii) Weighted fairing queuing with weights of 1.25 1.45 and 1.3 for flows 1, 2 and 3 respectively.

You may assume that the system clock tick period of the router is one bit period in duration Your answer must clearly indicate the methodology used in determining your answer.

[10 marks]

(c) Two commonly used flow based metrics for the comparison of network performance are (i) *Power* and (ii) *Jain's Fairness Index*. Clear state the equations for calculating these two terms and explain the meaning of all terms in these equations.

Two alternative congestion control algorithms are being considered for deployment in an IP network. Initial tests are carried out on a test network by establishing a number of flows over the network and measuring the throughput and delay of each flow. The results of the tests are tabulated below:

	Throughput (kbps)	Delay (ms)
Congestion Control Algorithm 1		
Flow A	120	15
Flow B	100	40
Flow C	90	15
Flow D	150	10
Congestion Control Algorithm 2		
Flow A	100	10
Flow B	120	30
Flow C	130	20

Using the parameters identified previously, determine the more effective and fairer of the two algorithms outlined above.

[6 marks]

5. *(a)* Clearly distinguish between the aims and general operation of TCP *congestion control* and *flow control*. Your answer should discuss the various modes of operation of the TCP congestion control algorithm (e.g. slow start, multiplicative decrease, additive increase etc.) and the conditions under which each mode is invoked.

[8 marks]

(b) By describing the operation one example of each type of algorithm, clearly explain the difference between *host based* and *router based* congestion avoidance algorithms. Your answer should clearly outline one advantage and one disadvantage of each type of approach.

[6 marks]

(c) A RED algorithm is operating on an IP router and the current state of the RED algorithm is summarised as follows:

- Queue length averaging factor α=0.2,
- Actual Queue Length = 3000 bytes,
- Current Average Queue Length =2800 bytes,
- RED Maximum Queue Length Threshold = 4000 bytes,
- RED Minimum Queue Length Threshold = 2000 bytes,
- RED Maximum Probability Drop =0.5
- Number of packets processed since last packet drop = 2 packets

A packet of length 500 bytes is the next packet to arrive at this router. Determine the probability that the RED algorithm will NOT drop this packet. You should assume that the average queue length is recalculated after the arrival of each new incoming packet.

Hence, if the length of the next packet which arrives at this router after this 500 byte packet is 450 bytes and if the actual queue length of the queue just before each of this packets arrive is 3000 bytes, determine the probability that the 450 byte packet will actually be the next packet to be dropped as a result of the RED algorithm.

[8 marks]

6. *(a)* Explain, by outlining a sample application, why the traditional concept of *best-effort* handling of IP packets is not sufficient for many modern *real-time* or *loss intolerant* applications.

[4 marks]

(b) "An IP network supports *Quality of Service (QoS)* using the *Differentiated Services* model and implements two service levels using *expedited forwarding*". Clearly explain how packets in such a network are processed when:

(i) Entering the QoS enabled network,

(ii) Arriving at a QoS enabled router in the network.

Describe how the handling of these packets at the QoS enabled router would differ if the service levels were implemented using the *assured forwarding* approach.

[8 marks]

(c) Discuss the meaning of the following terms in the context of the *Integrated Services* model for Quality of Service (QoS):

(i) Guaranteed Service,

(ii) Controlled Load Service,

(iii) Reservation Resource Protocol,

(iv) Scalability.

[8 marks]

7. *(a)* Explain why *delay* is a serious concern in VoIP networks and, hence, describe four different sources of delay that commonly occur in a VoIP network.

[6 marks]

(b) Describe, with the aid of a diagram, the role and operation of the *jitter buffer* typically implemented in the receiver of a VoIP terminal and clearly distinguish between *inter-talkspurt* and *intra-talkspurt* playout delay adaptation algorithms.

A proposed inter-talkspurt playback delay adaptation algorithm calculates the playout delay to be applied to talkspurt N by estimating the probability distribution function of the inter-packet arrival times during talkspurt N-1. For a particular talkspurt N, the algorithm estimates that the inter-packet arrival times during the previous talkspurt followed a normal distribution with a mean of 100 ms and a standard deviation of 15 ms.

Estimate the playout delay which should be used for talkspurt N such that 5% of all packets will arrive too late for playback.

Using this delay setting, estimate the percentage of packets which will arrive too late for playback if the network delay probability distribution function changes abruptly, during the silence period between talkspurts N-1 and N, to a normal distribution function with a mean of 110 ms and a standard deviation of 5 ms.

[8 marks]

(c) Describe, with the use of a diagram illustrating the establishment\termination of the call, the operation of SIP and RTP in setting up a basic VoIP call directly between two terminals in a VoIP network. Hence, explain the rationale for the use, and the operation, of a *redirect server* and *proxy server* in such a SIP VoIP network.

[6 marks]

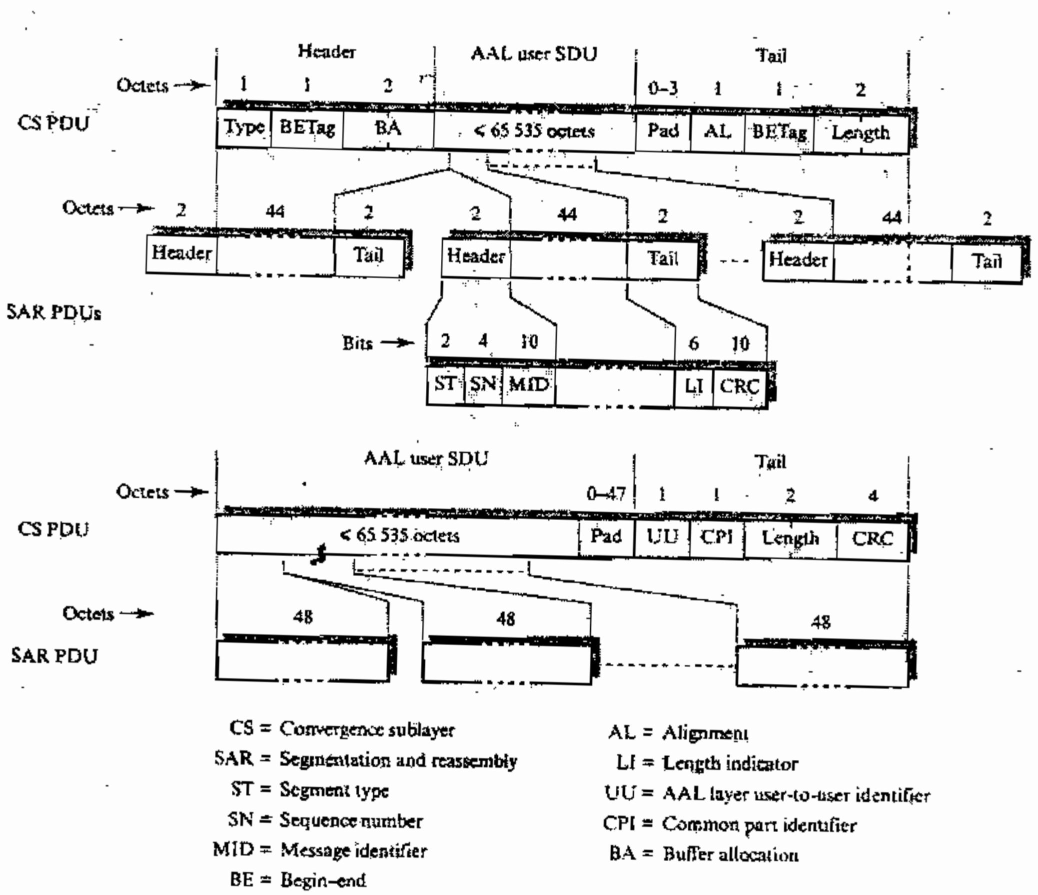

Figure 1 – Structure of AAL3\4 and AAL5 PDUs