Athlone Institute of Technology

Faculty of Engineering & Informatics

Bachelor of Engineering (Hons) in Software Engineering
Bachelor of Science (Hons) in Software Design with Mobile Apps & Connected Devices

Networks 4

AUTUMN REPEAT – ALTERNATIVE ASSESSMENT

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Instructions:

- Answer **ALL** questions.
- All questions carry the same marks.
- Show all calculations in full.
- A pdf file of the completed assessment must be submitted through Moodle.

Q1 An application on a device, which is connected to the Internet via a 100Mbit/s Ethernet connection, generates a 1,200-byte block of data. The service provided by the transport layer is reliable. How long does it take to transmit the Ethernet frame? Explain clearly any assumptions made.

Q2 Interleaving is used in a VoIP application to conceal burst loss. Consider that 24 packets (numbered 1 to 24) of speech are to be transmitted and the interleaving depth used is 8. All the packets are 225 bytes and the transmit rate is 10 Mbit/s.

i) If transmission starts at t = 0 seconds, at what time is transmission of packet number 4 complete?

ii) Supporting your answer using the information given, explain a disadvantage of the interleaving process.

Q3 An application is running on top of UDP at the transport layer and IP at the network layer.

i) What type of service do UDP and IP provide and what are the characteristics of such a service?

ii) What is the disadvantage of using such a service at the network layer?

iii) What is the advantage of using such a service at the network layer?

iv) Giving reasons, explain what needs to be done at the data link layer and the application layer to ensure that the sending application's data is replicated at the receiver.

v) Why is there a need for UDP if it provides the same service as IP?

Q4 A company is implementing VoIP to carry voice calls between sites. WAN connections between sites will carry voice and data. G.711 CODECs are used , generating 8,000, 8-bit samples per second. The speech sample size is 20ms and the additional overhead required per packet is 25%.

i) How many bytes are there per packet?

ii) What is the total bandwidth required for 50 concurrent calls?

iii) The network administrator decides to change to a G.728 codec which operates at 16kbps and has the same sampling period as the G.711 codec. Explain one advantage and one disadvantage of this change.

Q5 Two nodes, A and B, are attached to opposite ends of a 2km cable. Node A has one frame of 9,744 bits (including all header and trailer bits) to send to node B. The signal propagation speed on the cable is 0.5×10^{-8} s/m and the transmission rate is 100 Mbps. There are five routers between A and B, each inserting a 20-bit delay. If A starts transmitting the frame at time t = 0s, and assuming the same link layer protocol operates across all links, at what time does the last bit of A's frame arrive at B?

Q6 Figure Q6 shows the variation of a TCP congestion window as a function of time (transmission round) when using the TCP Reno algorithm. Using Figure Q6 to explain the operation of the TCP Reno algorithm, highlighting the four key elements of the algorithm.

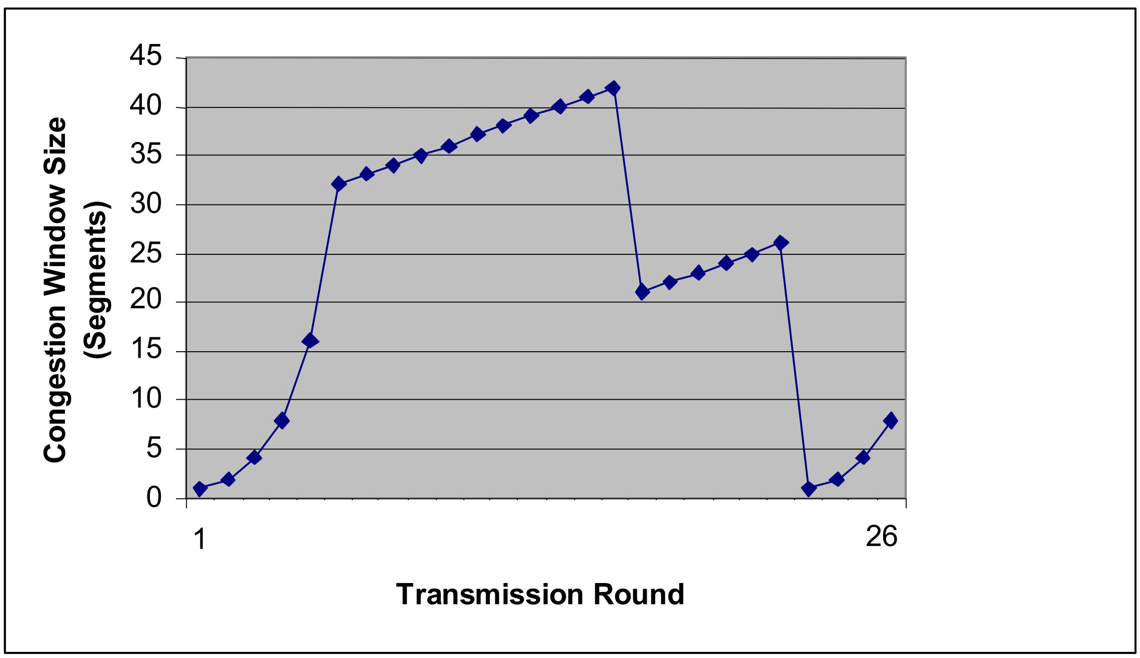

Figure Q6

Q7 Using a frame sequence diagram, give an example of an exchange of frames between X and Y that involves the three different HDLC frame types. Give a detailed description of what is happening in the exchange. Include, where appropriate, frame sequence numbers in the form [N(S), N(R)].

Q8 Packets of four different data classes D1, D2, D3 and D4 are queued for transmission at a router. All packets are 500 bytes in size and the transmit rate is 10Mbps. At t = 0 seconds all queues are full, each containing 10 packets.

i) Assuming priority queueing, where D1 is the highest priority and D4 the lowest priority, at what times will transmission of the last D1 and D4 packets be complete?

ii) Assuming round-robin queueing, with the same assigned priorities as in (i), at what times will transmission of the last D1 and D4 packets be complete?

iii) If weighted fair queueing is used and the weights assigned to D1, D2, D3 and D4 are 0.2, 0.3, 0.1 and 0.4 respectively, at what times will transmission of the last D1 and D4 packets be complete?

Q9 Two stations are communicating across a data link using the HDLC protocol and both stations are busy generating frames. The data rate is 400Mbit/s, the end-to-end delay is 20μs and the fixed frame size is 250 bytes of which 8 are overhead. A window size of 4 is used and the ARQ mechanism is go-back-N. The protocol is set up in such a way that for information frames all acknowledgements are piggybacked. Assume that there are no frame errors.

i) Calculate the efficiency of the given data link as described above.

ii) What impact will increasing the window size to 12 have on the efficiency of the link?

Q10 Using a diagram to help illustrate your answer, compare the role and scope of operation of layer 2 source/destination addresses with that of layer 3 and layer 4 source/destination addresses in the exchange of messages between two end-points in the Internet.

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Question 1.

Preamble	SFD	Dest.address	Src add	Typefield	Data	CRC
7 bytes	1 byte	6 bytes	6 bytes	6 bytes	46-1500 bytes	4 bytes

Ethernet Frame

Given,

R(Rate) = 100Mbit/s

L(length of data) = 1200 Byte

H(Header) = 18

Transmit Time (T) = Number of bits / bits rate = L/R = (Payload + header) / R

$T = 1218*8 / 100 * 10^6$

$T = 9.744 * 10^{-5}$ s

Assumption made: We are transmitting frame using Ethernet II protocol. But we inserted 14 byte MAC header + 4 byte CRC checksum amounting to 18 bytes of total header over the payload.

Question 2.

Given,

Packet = 24

Depth = 8

L = 225 bytes

R = 10Mbit/s = 10^7 bit/s

No of Interleaving Stream = no of packets / interleaving depth

= 24/ 8 = 3

Because of Interleaving process, packet 4 will send with first interleaving stream.

i. The time taken to transmit the first package is going to be when first group gets transmitted.

Since, Transmit Time (T) = Number of bits / bits rate = L/R

L = 225 * D * 8 = 225 *8*8 bit

$R = 10*10^6$ bit/s

$T = 14400/10^7 = 1.44* 10^{-3}$ s

ii. First in the interleaving phase, we interleave after packets are created during the talk spurt after the transmission, so we also need to arrange back in the specified order, so this increases the delay in the playout.

Question 3.

i. UDP and IP both provide connectionless and unreliable service.

ii. The downside to using unreliable and communication service in network layer is that there are no error correction and recovery functionality for datagrams that are either duplicated, lost or delivered in a different sequence to the remote host than the one received.

iii. The advantage of using this service on a network layer is that you don't need to create and manage a bridge to transmit data that requires overhead and it doesn't take time for low latency circuit setups either.

iv. Since the program we are using runs UDP and IP protocols that are both connectionless and insecure, when frames or datagrams are duplicated, missing or out of order, there is uncertainty and inaccuracy in data transmission, so we need to implement error checking and error management procedures on data link layer and application layer to ensure smooth and secure data transfer.

v. UDP and IP both provide the same type of service, but it does have something that IP does not have which is port number to help distinguish user requests and, optionally, a checksum to ensure that the data has arrived intact.

Question 4.

Given,

Sample = 8 bit/s * 8000

Total samples = 64000 bit/s

Speech sample = 20 ms

i. Total bytes = Total sample rate * Speech sample size

$= 64000 * 20 * 10^{-3}$

= 1280 bits = 160 bytes

Plus,Additional overhead = 25%

Total Bytes = 160 + 25/100(160)

Bytes = 200

ii. Bandwidth = calls * bitrate

= 50 * byte/ sizeof sample

$= 50 * 200 * 8/ 20 * 10^{-3}$ bit/s

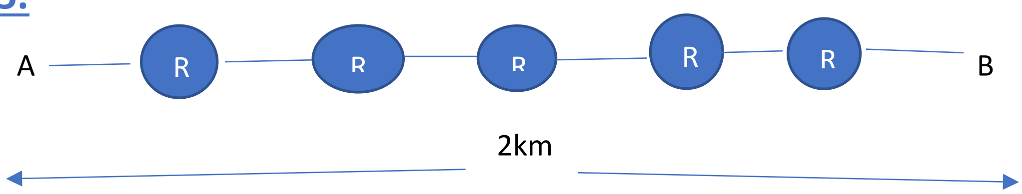
$= 4* 10^{6}$ bit/s

=4 Mbps

iii. The advantage would be that we will make more calls as it will hold more data as a result of increasing bit rate.

The disadvantage is that the calls are not reliable and are of poor quality.

Question 5.



Given,

L = 9,744 bits

Speed = $0.5 * 10^{-8}$ m/s

$= 2*10^{8}$ m/s

R = 100 Mbps

Processing delay = 20 bit

Time to arrive last bit of A at B = (Transmit time * 6) + cable delay + Processing delay * 5

$= 9744 * 6/ 10*10^{6} + 2000/2*10^{8} + 200/100*10^{6} * 5$

$= 9744 * 10^{-8} * 6 + 10 * 10^{-6} + 10^{-6}$

$= 58464 * 10^{-6} + 10\ \mu s + 1\ \mu s$
$= 584.64\ \mu s + 11\ \mu s$
$= 595.6\ \mu s$

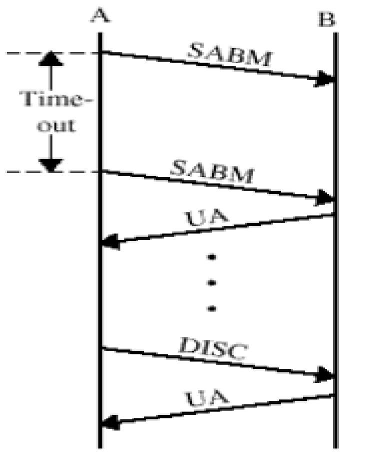
Question 6.

When using the TCP Reno congestion control algorithm if a packet loss happens (three duplicate ACKs), Reno carries out a fast retransmission and skips the slow start phase by halving the congestion window, setting the slow start threshold equal to the new congestion window.

Four key elements of algorithm are:

1. Slow start in case of ACK timeout (RTO timeout) Congestion window is cut in half window then grows linearly
2. Fast retransmit
3. Fast Recovery
4. Loss indicated by 3 duplicates

Question 7.

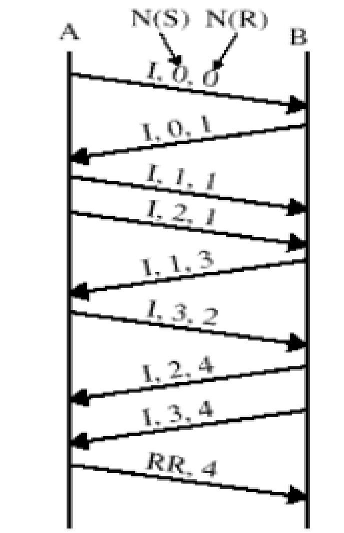


(a) Link setup and disconnect

Src: Moodle

The diagram shown above shows link setup and separation using frame exchange between A and B with U-frames (Unnumbered), they are mainly for control purposes , i.e. setting up and terminating connections, unnumbered ACK, etc., but can also be used to bring data from unrecognized connectionless networks, and the process is represented in M-bits.

The diagram A above uses the SABM frame (Set Asynchronous Balanced Mode) to reset the connection, then the B sends a UA (unnumbered acknowledgement), and A sends a DISC (DISConnect) to break the logical connection.



b) Two-way data exchange

Src: Moodle

The diagram above shows a two-way data exchange between A and B, in which information is transmitted and received from both sides. Sent and obtained frames are counted using N(S) and N(R), respectively. I- frames are sent from both ends, containing user data. In addition, the flow and error management (ACKs) can be piggybacked within an I-frame 's control field using N(R). S-frame that has RR (Receive Ready) should be sent close to the end of data exchange. S-frames have flow / error management while piggybacking is not used.

Question 8.

i. Priority Queuing

First all D1 are transmitted

Last all D4 are transmitted

Transmitted for last D1 = L/R * 10

$= 500 * 8/ 10 * 10^6 * 10$

$= 4000 * 10^{-6}$

$= 4 * 10^{-3}$ s

Transmitted for last D4 = L/R * 40

$= 500 * 8/ 10 * 10^6 * 40$

$= 16000 * 10^{-6}$

$= 16 * 10^{-3}$ s

ii. Round Robin

Transmit time for last D1 = L/R * 37

$= 4000/10 * 10^6 * 37$

= 0.0148 s

Transmit time for last D4 = L/R * 40

$= 4000/10 * 10^6 * 40$

$= 16000 * 10^{-6}$

$= 16 * 10^{-3}$ s

iii. Weighted Fair Queuing

0.2 0.3 0.1 0.4

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01	02	03	04

D1	D1	D2	D2	D2	D3
D4	D4	D4	D4	D1	D1
D2	D2	D2	D3	D4	D4
D4	D4	D1	D1	D2	D2
D2	D3	D4	D4	D1	D1
D2	D3	D1	D3	D3	D3
D3	D3	D3			

D1 last = 34

Transmit time for last D1 = L/R* 34

= 4000/10 * 10^6 * 34

= 0.0136 s

D4 last = 28

Transmit time for last D4 = L/R* 28

= 4000/10 * 10^6 * 28

= 0.0112 s

Question 9.

i. Determine window size

Window Size = 1 + A/F + 2CI/F

=1+250/250+ 2 *400*10^6 20 * 10^{-6}/ 250 *8

= 1+1+8

= 10

Since, 10>4 (Given)

Small window

Efficiency = WD/ F+ A+ 2CI

$E = 4 * 242 * 8/ 250 *8 +250*8+2*400*10^6 * 20 * 10^{-6}$

= 7744/20000

= 0.3872

Efficiency = 38 %

ii. If window size 12 < 10 (Max)

Large window efficiency = D/D+H*

= 242 * 8/ 242*8 + 8*8

= 0.968

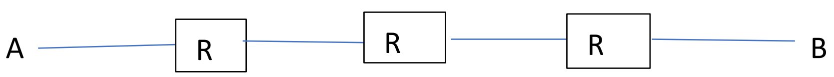
= 96%

Efficiency increases due to utilization

Question 10.

Port X PORT Y

IP – A.B.C.D IP – E.F.G.H



The addresses in Layer 2 are used for each connection to address the source and destination. For an interface between two endpoints there may be several routers and links in between. Thus it produces frame and eliminates frame in layer 2 for each connection, thus modifying the MAC address by using port number respectively the Layer 3 and Layer 4 addresses are used to address end devices using IP address and end processes, which remain constant and do not change.