Question Number 1

a. SNR= 100
Bandwidth =400 KHz
C= B* $\log_2$ (1+SNR)
By putting the values:
C=400*10^3 * $\log_2$ (1+100) = 26.57 Mbps

- 0.5 marks if calculation mistake.
-0.5 marks if error in units.
-1 marks if only formula is correct.
-2 marks if formula used is incorrect.

b. Maximum data rate= 10Mbps
No. of frames sent in one sec=12000/60= 200 f/s
Size of each frame= 10,000 bits
Throughput= data sent in one second= 10,000*200 = 20000000bps =20 Mbps

-0.5 marks if calculation mistake.
-0.5 marks if error in units.
-1 marks if only formula is correct.
-2 marks if formula used is incorrect.

c. 00111110110001111011110001111101

-1 if only single stuffed bit is correct.
-2 marks if both the stuffed bits are incorrect.

d. i. 10 Kbps *5 + 2* 5Kbps = 60 kbps

-1 marks if data rate is incorrect.

ii. In synchronous TDM, each sender is assigned a fixed time slot in a TDM frame. The frame size is determined by the total number of senders and the time required by each sender to transmit their data.

Given that there are 7 senders and each sender wants to transmit 5 KB of data, the total amount of data to be transmitted in one frame is:
Total data = 7 x 5 KB = 35 KB = 280 Kb

In multiplexing, we are using 1 slot for each sender having data rate 10 Kbps and 2 senders with 5kbps are multiplexed into a single slot. **(2 marks)**

Since no information is given regarding the slot size so by default slot size is taken as 1 bit.

Data carried by each frame= 6 bits from each slot+1 synchronization frame
Total number of frames= =Total data/data bits of frame= 280 Kb/6 = 46.66 = 47K frames **(1 mark)**

This implies that the total number of framing bits will be equal to 47Kb (47K * 1b) (given in the question that synchronization bits are added for each frame)

Total data to be sent = 280 Kb+ 47Kb= 327 Kb **(1 mark)**

Efficiency = Data size/ Total data size = 280 Kb/ 327 Kb = 85.62% **(1 mark)**

OR

ii. In synchronous TDM, each sender is assigned a fixed time slot in a TDM frame. The frame size is determined by the total number of senders and the time required by each sender to transmit their data.

Given that there are 7 senders and each sender wants to transmit 5 KB of data, the total amount of data to be transmitted in one frame is:
Total data = 7 x 5 KB = 35 KB = 280 Kb

In multiplexing, we are 2 slots for each sender having data rate 10 Kbps and 1 slot for senders with 5kbps. **(2 mark)**

Since no information is given regarding the slot size so by default slot size is taken as 1 bit.

Data carried by each frame= 12 bits from each slot+1 synchronization frame= 13 bits
Total number of frames= =Total data/data bits of frame= 280 Kb/12 = 23.33 = 24K frames **(1 mark)**

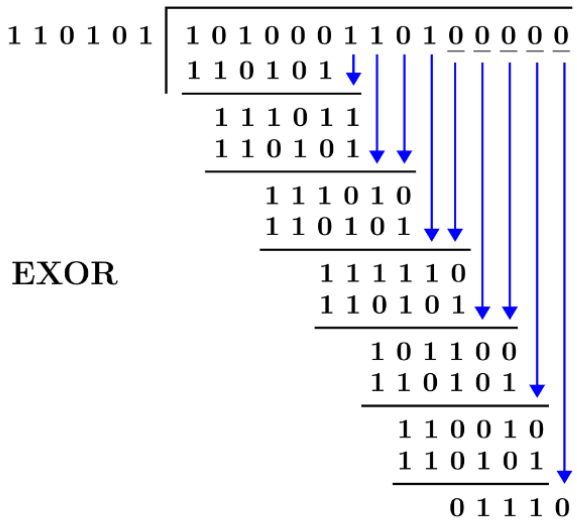
This implies that the total number of framing bits will be equal to 24Kb (24K * 1b) (given in the question that synchronization bits are added for each frame)

Total data to be sent = 280 Kb+ 24Kb= 304 Kb **(1 mark)**

Efficiency = Data size/ Total data size = 280 Kb/ 304 Kb = 92.10% **(1 mark)**

Question Number 2

a. M = 1010001101
Divisor polynomial bit= 110101
Bits to be appended to message= (divisor polynomial bits – 1) = 5
Append 5 zeros to message bits, modified message: 101000110100000



M'=1 0 1 0 0 0 1 1 0 1 0 1 1 1 0

-1 mark if final message is incorrect. (If there is error in the final message due to single incorrect EXOR operation).
-1 mark for each incorrect EXOR operation.
-2 marks if number of bits appended are incorrect.
- 6 marks if bits are not appended.

b. If two 16-bit words are swapped in the bit stream during transmission, the checksum will not detect this error.

The checksum algorithm is designed to detect errors that occur within a single 16-bit word. It works by adding up all the 16-bit words in the data, and then taking the one's complement of the sum. The result is then appended to the end of the data and sent along with it.

When the receiver receives the data, it performs the same checksum algorithm on the data (including the appended checksum), and compares the result with the checksum that was sent. If the two checksums match, the receiver assumes that the

data was transmitted without errors. If the checksums don't match, the receiver knows that an error has occurred.

Now, let's consider the scenario where two 16-bit words are swapped during transmission. In this case, the data that was received by the receiver will be different from the data that was transmitted by the sender. However, the sum of the 16-bit words will remain the same, as only the order of the words has been changed. This means that the calculated checksum will also be the same as the checksum that was transmitted by the sender.

As a result, the receiver will not detect the error and will assume that the data is error-free. This is because the checksum algorithm is designed to detect errors that occur within a single 16-bit word, but not errors that occur due to the order of the words being swapped.

-2 marks if it is mentioned that checksum may or may not detect.
-2 marks if it is mentioned that checksum will not detect.
-1 mark if reason is incorrect

Question Number 3

a. Selective Repeat ARQ (Automatic Repeat Request) is a type of error control mechanism used in data communication systems. It allows the receiver to selectively request retransmission of only those packets that are lost or corrupted during transmission, rather than asking for the entire set of packets to be retransmitted.
In selective repeat ARQ, each packet is assigned a sequence number, which is used by the receiver to acknowledge the successful reception of the packet. The sender maintains a window of unacknowledged packets, which can be transmitted without waiting for an acknowledgement.

Now, let's consider a selective repeat ARQ scheme with 3-bit sequence numbers. This means that the sequence numbers range from 0 to 7 (2^3 = 8 possible values).
The sender maintains a window of unacknowledged packets, and the size of this window determines the maximum number of packets that can be transmitted without waiting for an acknowledgement. The window size is limited by the number of available sequence numbers.
In this case, we have 8 possible sequence numbers (0 to 7), but one of these sequence numbers (e.g., 0) is used for the initial transmission of packets and is not available for retransmission. This leaves us with only 7 available sequence numbers for retransmission. If the window size is larger than the number of available sequence numbers, there is a risk that the same sequence number may be used for two different

packets, which can result in errors in the received data. For example, suppose we have a window size of 5 and the following packets are sent:

Packet 0, 1, 2, 3, 4 are sent and are all received successfully.
Packets 0 and 1 are acknowledged, but the acknowledgement for packet 2 is lost.
The sender retransmits packet 2, but the acknowledgement for packet 2 is again lost.
The sender then transmits packets 5, 6 and 7, which are all received successfully.
The receiver now sends an acknowledgement for packet 3, which includes sequence numbers 0, 1, 2 and 4.
The sender incorrectly assumes that packets 0, 1 and 4 have been acknowledged and removes them from the window.
However, packet 2 has not been acknowledged and is still in the window.
If the sender transmits a new packet with sequence number 2, it will overwrite the unacknowledged packet 2, resulting in an error in the received data.

This problem can be avoided by limiting the window size to the number of available sequence numbers for retransmission, which in this case is 7-1=6. Therefore, the maximum window size for a selective repeat ARQ scheme with 3-bit sequence numbers is 6, which means that at most 6 unacknowledged packets can be sent at a time without waiting for an acknowledgement. A window size of 4 is also safe, as it is less than the maximum of 6.

****************************Example may vary******************************

-2 marks if example is missing.
-2 marks if example is partially correct.
- 4 marks if it is not mentioned that frames are erroneously accepted.

b. For calculating efficiency: **5 marks**
Window size= 6 frames
Size of data frame = 64 bytes
Size of acknowledgement frame = 8 bytes
Transmission rate= 100Mbps
Length of the link= 10km
Propagation speed=$2*10^8$ m/sec **1 mark**
Propagation delay =$(10*1000)/(2*10^8)$ m/sec= $50 *10^{-6}$ sec
Round Trip Time (RTT)= 2* Propagation delay= $100*10^{-6}$ sec
Transmission Time of sender= Length of Frame/ Bandwidth = $64*8/ 100 * 10^6$ = $5.12 * 10^{-6}$ sec **1 mark**
Transmission Time of receiver= Length of acknowledgement/ Bandwidth = $8*8/ 100 * 10^6 = 0.64 * 10^{-6}$ sec

Total time= RTT+ Transmission Time of sender + Transmission Time of receiver = 105.76* 10^{-6} sec **1 mark**

Efficiency =((6* Transmission Time of sender)/ Total time) *100= 30.72/105.76 29.04% **2 marks (-1 marks if calculation mistake)**

For maximum efficiency: **3 marks**
(N* Transmission Time of sender)/ Total time >=1 **2 marks**
N =105.76/5.12 =20.65
N=21 **1 mark**

OR

Window size= 6 frames
Size of data frame = 64 bytes
Size of acknowledgement frame = 8 bytes
Transmission rate= 100Mbps
Length of the link= 10km
Propagation speed=2*10^8 m/sec
Propagation delay =(10*1000)/(2*10^8) m/sec= 50 *10^{-6} sec **1 mark**
Round Trip Time (RTT)= 2* Propagation delay= 100*10^{-6} sec
Transmission Time of sender= Length of Frame/ Bandwidth = 64*8/ 100 * 10^6= 5.12 * 10^{-6} sec **1 mark**

Efficiency =N/ 1+2a **1 mark**
a= 9.76 **1 mark**
Efficiency= (6/ 1+2*9.76)*100 =29.22% **1 mark**

For efficiency=100% **3 marks**
N/ (1+2a) =1 **2 marks**
N= 1+2a
N =20.52
N=21 **1 mark**

Indian Institute of Technology Kharagpur
Dept. of Computer Science & Engineering

Mid-Semester Examination, Spring 2022-23

Subject No.: CS31204/CS31006 **Subject Name: Computer Networks**
Time: 2 Hours **Full Marks: 60**

Answer ALL five questions

4. (a) Explain how a frame is sent by a node in a system using CSMA/CA system, including how collisions are detected and handled. (6)

2 marks for checking for line idle/busy, waiting for IFS etc.
1 mark for waiting for random time even after waiting for IFS time
1 mark for stopping wait timer during the random time wait if line is sensed busy
1 mark for collision detection (given only if you have written that detected by no acknowledgment received)
1 mark for collision handling (given only if you have said anything about changing random time to wait for the next time using exponential backoff)

RTS/CTS are optional and not in basic CSMA/CA. Ok if you have written, no marks deducted, but no marks for that.
Marks for collision handling is given only if you have explicitly said the above; just saying earlier binary exponential random time wait is not sufficient as it does not say where is the time changed.

(b) Suppose that two machines A and B want to communicate with each other in a CSMA/CD scheme using p-persistent CSMA, with p = 0.4. If both machines sense the medium to be free at the same time at beginning of time slot 0, what is the probability that exactly one process will transmit in time slot 0? Assume that a machine gives up after 3 collisions and the time for sensing the channel, detecting collision, processing etc. is negligible compared to the slot time. Show all calculations with justifications. (8)

Lets take only A transmits case. Case for B is symmetrical so can be just doubled.

- *No collision case: A transmits at slot 0, B does not, so 0.4x0.6 = 0.24*

- *One collision case: Here, after collision, machines will choose from {0, 1) slots to wait (binary exponential backoff). So A transmits if: both A and B transmits in slot 0 (so collision), and ((A chooses 0 slots to wait and B chooses 1 slot to wait and A transmits) OR (A chooses 0 slots to wait and B chooses 0 slot to wait and A transmits and B does not transmit)) = 0.4x0.4x (0.5x0.5x0.4) + (0.5x0.5x0.4x0.6))*
- *Two collision case: Similar, just that no. of slots to wait for will be chosen from {0, 1, 2, 3}, so A still has to choose 0 but B can choose any of 1, 2, 3.*

4 marks for no collision case (3 if you forgot to multiply 0.24 by 2)
2 marks for handling collision cases (even if you do not consider that slots to wait for will change with number of collisions; given 1 or 2 depending on how clearly written)
2 marks for handling changing the number of slots to wait for with different number of collisions

5. (a) Explain clearly why there is no collision in a full-duplex switched Ethernet. (4)

2 marks for writing how collision between distinct pairs (say A to B and C to D) are prevented (basically writing how a switch works, 1 or 2 given based on what you wrote), 1 mark for noting 2 separate cables because of full duplex (prevents collision when A has to both transmit and receive), 1 mark for saying buffer inside switch (prevents collision when both A and B send to C).

(b) Explain step-by-step how an Ethernet packet is processed by a receiver in a TCP/IP network. (5)

The purpose of mentioning TCP/IP here is just to see if you remember that in TCP/IP, Ethernet always means Ethernet-2, not 802.3. An Ethernet packet is processed only at the Ethernet layer, rest of the layers process the data part of the Ethernet packet, not relevant. Still given 1 if you wrote things about layering and IP processing etc. only.

1 marks for talking about all fields in Ethernet header (showing or implicitly while discussing)
1 mark for CRC check (this should be done first or 0.5 deduced. If CRC is wrong, no point checking anything else at all)
2 marks for checking destination address (DA) field: 1 deducted if you have not explicitly said broadcast address is also accepted. No marks if you brought in DSAP etc. of LLC which is not there in Ethernet-2.
1 mark for processing Type field. No marks if you said Length etc.

As stated in class, preamble handling is not considered a part of Ethernet frame processing, as you us it to get out the frame as a frame. Ok if you have written it, no marks deducted, just no marks for that.

(c) Suppose you are given five 8-port switches. What is the maximum number of machines you can connect with them? Justify briefly. (3)

2 marks for stating the correct answer (32), 1 mark for arguing why is this maximum. For example, if you have shown one configuration to show 32 machines can be connected, you get 2 because you have not justified why no other configuration can give you more. The max. comes easily from the simple graph fact that to connect 5 nodes you need at least 4 edges, now each edge takes away 2 ports (one at each end) from the total ports (over all switches), so ports remaining free to connect machines is 5x8 – 4x2 = 32.

(d) Can you do reliable communication across a link using Ethernet? Justify. (2)

2 marks for saying no, 0 for saying yes. Ethernet does not have any error control, only error detection, frames can still get dropped due to corruption or switch buffer overflow.

Many of you have written Yes with justification that higher layers can make it reliable, still get 0, that is not the issue at all, you can always add layers above to make any unreliable protocol reliable, that will not be Ethernet anymore. Also, the question asks for communication across a link, so not relevant for end-to-end communication like IP, TCP etc. anyway.

An exception where you got 2 even with saying yes is if you have explicitly said that by Ethernet, you are considering 802.3 and so 802.2 LLC can do this.