CS31006: Computer Networks
Long Test 1, Part - 2
February 22, 2022

Duration: 1 Hour (45 minutes for answering questions + 15 minutes for submission)

Answer ALL questions

All your answers MUST BE HANDWRITTEN on paper. Scan all papers with your answer in a SINGLE pdf and upload in the course page in Moodle in the appropriate link. The size of the final pdf must be less than 10MB. You must upload the pdf strictly by 9-50 am Moodle server time.

1. (a) Show the encoding of the bit stream 1100110010101100 using Pseudoternary encoding scheme assuming that the last signal level before the start of this bit stream has been positive. What is the total number of signals transitions? (3 marks)

   [ANS]

   (b) Suppose you want to send the bit stream 011110111010. Show the final bit stream transmitted if checksum is used for adding error detecting bits with 4-bit word size. (3 marks)

   [ANS]

   (c) Suppose a sender S is sending 12 frames (numbered 0 to 11) to a receiver R using sliding window flow control protocol. The window size used is 4. Consider a time at which the sender has sent frames 0 to 7, and has received acknowledgements for the frames 0 to 4. The receiver has received frames 0 to 5. Draw a diagram to clearly show the position of the sender and receiver windows at this time with brief (1-2 sentences max for each window) justification. Also show on the diagram which frames are in transit. Assume that there is no error or loss in transmitting any frame. (4 marks)

   [ANS]

2. (a) Consider a sender S sending 7 frames (numbered 0 to 6) to a receiver R. Go-back-N is used as the error control method with 3-bit sequence number and window size of 4. Processing time at the receiver is negligible, ACKs are sent immediately on receive (no piggybacked ACK), the time for any frame to travel over the link (either from S to R or from R to S) is 1 time unit (constant for all frames), and the timeout duration is 2 time units. Assume that when frame 2 and frame 4 are sent **for the first time**, the frames are lost. No other frames are lost. Show clearly with a diagram (as shown in class for describing ARQ protocols) the flow of the frames between S and R, clearly showing the data frame numbers, type of frame, and sequence numbers in each frame. No explanation is needed, just show the diagram. (7 marks)

[ANS]

(b) Suppose you want to do error control between a sender S and a receiver R. If the probability of frame loss/corruption is low and the total delay for a frame to reach from S to R is high, which of Go-back-N and Selective-Repeat ARQ methods would you use? Briefly justify your answer. (3 marks)

[ANS]

3. (a) Consider a transmitter T sending a frame to a receiver R at 10 Mbps over a shared medium using 1-persistent CSMA/CD. The propagation speed is 2 x 10^8 m/sec, and the maximum distance between two stations is 200 m. Processing delays at receiving nodes are negligible and can be ignored. Suppose that 4 microseconds have passed from the start of transmission and T is still transmitting the frame. Do you think it will be able to complete the transmission of the frame successfully? Justify your answer showing all calculations you use. (5 marks)

[ANS]
T will be able to transmit the frame successfully if twice the propagation delay is less than 4 microseconds (time for 1st bit to reach and the jamming signal to come back in case there is a collision). Propagation delay is 200/(2 x $10^{8)}$ sec = 1 microseconds. So twice the propagation delay is 2 microseconds, which is less than 4 microseconds. So T will be able to successfully transmit the frame.

Most of you got full marks in this.

(b) Show with an example why the RTS/CTS scheme may not fully solve the exposed terminal problem. (5 marks)

[ANS]
Consider 4 nodes A, B, C, D, with A in range of B only, B in range of A and C only, C in range of B and D only, and D in range of C only. B sends a RTS to A, but C does not receive it due to some error. A sends CTS to B (not received by C because it is not in range), and B starts transmitting to A. Now C will find the channel busy and not transmit to D.

Most of you did not attempt this.

4. (a) In a packet switched network, suppose that a total of 10000 bytes of data is to be transferred from a node A to a node B at distance of 2 hops away (i.e., one router in between A and B) using datagram packet switching over 200 kbps links. Each packet has a size of 500 bytes (not including header) and requires a header of size 50 bytes. The time for one packet (including header) to be completely received over one hop (i.e., the time from the transmitter transmitting the first bit till the time the receiver receives the last bit of the

packet) is 50 milliseconds. What would be the total time for B to receive all 5000 bytes? Show all your calculations. (5 marks)

[ANS]

Firstly, there was a small problem with this question, at the beginning it says 10000 bytes but at the end says "all 5000 bytes". Marks are given for both.

The main thing to see in this is that there will be lots of overlap in transmission of the packets. There will be overlap in propagation and transmission times on the same link, as well as overlap between packets flowing in the two links. This is somewhat like a pipeline.

Given the parameters, there will be 20 packets, each of size 550 bytes (500 bytes of data + 50 bytes of header). At 200 kbps, the time to transmit each packet is (550 x 8)/200 milliseconds = 22 milliseconds. Also the router will start sending a frame out on the second link as soon as that frame is completely received at the router.

Now just think of each packet. The time to transmit 1 bit is 0.005 millisecond. So for packet 1:

- *1st bit is put on link 1 (A to router) at t = 0.005 ms*
- *Complete frame received at router at 50.005 ms*
- *1st bit is put on link 2 (from router to B) at t = 50.01 ms*
- *Complete frame received at B at 100.01 ms*

Similarly, for packet 2;

- *1st bit is put on link 1 (A to router) at t = 22 (for first frame) + 0.005 = 22.005 ms*
- *Complete frame received at router at 72.005 ms*
- *1st bit is put on link 2 (from router to B) at t = 72.01 ms*
- *Complete frame received at B at 122.01 ms*

*You can continue like this. If you do one more, you should see the formula. Ignoring the negligible time to transmit 1 bit part (0.005 ms), for n packets, the time will be (n-1)*22 + 2*50. The 2* factor comes because the first frame travelling on link 1 is not overlapped with anything on link 2 and the last frame travelling on link 2 is not overlapped with anything on link 1. There is another simpler way to look at it also, ok if you got that.*

You got 1 out of 5 if you did not see the overlap at all, that was the main thing, otherwise it is trivial to compute the time for 10 packets.

(b) Consider that an Ethernet frame is received at the data link layer of a machine in a LAN. Draw the frame format and list step-by-step how is the frame processed in the data link layer. (5 marks)

[ANS]

Frame format you can see from text. (1 mark)

Processing:

- *Compute the CRC value for the frame and compare with the CRC in the FCS field. If it does not match, drop the frame. (1 mark)*
- *Check the destination address field. If it either matches the MAC address of this host, or is the broadcast MAC address, process the frame further, else drop it. (2 marks, 1 each for mentioning host MAC and broadcast MAC)*
- *Check the Type field. If it matches a valid protocol type installed in this host, strip the frame header and pass the data part of the frame to this protocol software. Else, drop the frame (1 mark)*

I have assumed the Ethernet-2 format (so Type field), but marks are given if you have assumed Length field as long as you can say how to process it.

Preamble processing is ok, but no marks for that. Preamble triggers h/w circuitry for synchronization and check to identify the beginning of a frame, and not really considered a part of frame processing.