COMP 3331/9331: Computer Networks and Applications

Answers to Review Questions
Practice Problem (1)

• When the traffic load on the LAN is very light, which of the following two MAC protocols: Token-passing or Ethernet has a smaller delay? Explain briefly.

• Answer:

   Ethernet will have a smaller delay since the nodes that want to transmit will almost always be able to transmit instantly (without any delays). Further the # of collisions will be very small. On the contrary, token-passing incurs the additional delay of circulating the token along the network after each transmission.
Practice Problem (2)

• Consider two hosts A and B that are connected via a direct link. Assuming that
  • Host A is the transmitter and host B is the receiver
  • The transport layer for hosts A and B uses selective repeat with a window size of \( t \) for reliable delivery
  • The link connecting A and B uses selective repeat with a window size \( m \) for reliable delivery
  • The link layer has sufficient memory to buffer at least \( \max(t,m) \) frames
  • The receiver at link layer will pass all received frames (provided that they are not corrupted) to the transport layer, even if they are out of order
  • One segment will generate one packet and will in turn generate one frame. There is no packet fragmentation.
  • Congestion control and flow control are not used at both the transport and link layers.
Practice Problem (2)

- If \( t = 10 \) and \( m = 5 \), what is the maximum number of out of order buffered segments at the transport layer of host B?
- If \( t = 10 \) and \( m = 20 \), what is the maximum number of out of order buffered segments at the transport layer of host B?
Practice Problem (2)

• Since the link layer will pass all frames to the transport layer, the number of out of order buffered frames = number of out of order buffered segments.

• (t = 10, m = 5) If t = 10, the transport layer can send 10 segments at most but the link layer is limited to send 5 frames, so the maximum number of out of order buffered frames is 4, so is the number of out of order buffered segments.

• (t = 10, m = 20) If t = 10, the transport layer can send 10 segments. However, with m = 20, the link layer can send up to 20 frames but will only get a maximum of 10 segments at a time. So, the maximum number of out of order buffered frames is 9, so is the number of out of order buffered segments.
Practice Problem (3)

- Consider the following 3 applications over a wireless network
  - Voice-over-IP, where packets are very small and the send rate is constant
  - MPEG movie streaming, where the packet size is large and the send rate is variable
  - Instant messenger chat, where packet size is small and send rate is variable
- For each application, list and explain which of the following MAC protocols you would use – (i) TDMA, (ii) plain CSMA/CA (iii) CSMA/CA + RTS/CTS
Practice Problem (3)

• Answer:

• VoIP: TDMA, because we can easily split up each constant rate flow into constant size slots.

• Movies: CSMA/CA + RTS/CTS because collisions of large packets are expensive and we want to avoid them. TDMA would not allow efficient use of the medium because the send rate is variable.

• Messaging: plain CSMA/CA since the overhead of RTS/CTS is not worth it for small packets. TDMA is not suitable for the same reason as above.
Practice Problem (4)

• Each node in a network runs Dijkstra’s algorithm to compute its shortest paths to all the other nodes (assume that every node gathers all the needed information correctly). Node A determines that its shortest path to node B contains 5 links and has a cost of 60. Node A also determines that its shortest path to node D contains 11 links, has a cost of 90, and passes first through node B and then through node C. Node B determines that its shortest path to node D contains 7 links, has a cost of 21, and does not pass through node C. Is such an outcome possible? Why?

• Answer:

• It is not possible. Otherwise, there would be a path from A to D with a cost of $60 + 21 = 81$ which is less than 90, the cost of the shortest path from A to D.
Practice Problem (5)

- Host A uses TCP Reno to transfer a file to host B. The file contains 32 MSS of data. During the first transmission round, the congestion window is equal to 1 MSS. During the fourth round when the connection is still in the slow-start mode, all the transmitted packets are lost (and, therefore, host A transmits less during the fifth round). There is no packet loss during any other round. During what round does host B receive the complete file? Explain.
Practice Problem (5)

Answer: In the slow-start mode, the congestion window doubles in size each transmission round. During the fourth round when the congestion window equals to 8 MSS, all 8 transmitted packets are lost. Hence, the sender determines the losses via a timeout, sets the threshold to 4 MSS, and reduces the congestion window to 1 MSS. Staring from the fifth round, the sender retransmits the lost data and then transmits the remaining portion of the file. The congestion window doubles until it becomes equal to the threshold (i.e., 4 MSS) during the seventh transmission round. Then, the TCP connection switches to the congestion-avoidance mode and increases the congestion window by 1 MSS per transmission round. Host A finishes the delivery of the 32-MSS file during the tenth round:

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<th>Round</th>
<th>Congestion window (MSS)</th>
<th>Transmitted data (MSS)</th>
<th>Delivered data (MSS)</th>
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Practice Problem (6)

• Consider a TCP Reno connection between two hosts with a large amount of data packets flowing in only one direction. Assume a malicious router along the path drops every other data segment of this connection (i.e. it drops 2, 4, 6, etc.). Assume that the receiver’s advertised window is 256 MSS. Also assume that there is plenty of bandwidth available on the path, so as to accommodate a TCP window size of 64 MSS in the absence of the malicious router.

  • (a) What is the maximum window size achieved on this connection?
  • (b) Repeat (a) for the case when the malicious drops every eight packet (say 8, 16, 24, etc.)
Practice Problem (6)

- Answer:
  (a) Maximum window size = 2. This is because the transmission of every second packet causes a timeout followed by slow start. In other words, the window size will oscillate between 1 and 2.
Practice Problem (6)

(b) Window size will grow up to 8. When packet # 8 is dropped, the congestion window = 8. Several duplicate ACKs are returned which would trigger fast retransmit and the window would be cut to 4 (threshold also would be 4). TCP will now enter congestion avoidance and increase the window linearly to 5. Since every 8\textsuperscript{th} packet is dropped, in the back-to-back rounds when the window size is 4 and 5 respectively, there will always be a packet loss event. As such the window will never increase beyond 5. So the maximum window size achieved will be 8 during the initial slow start phase.
Practice Problem (7)

• Consider a network architecture that has $n$ layer protocol hierarchy. Applications generate messages of length $M$ bytes. At each of the layer, an $h$-byte header is added. What fraction of the network bandwidth used by the application is wasted on headers?

• Answer:

Packet size of the transmitted packet = $M + n*h$.  
The fraction of bandwidth wasted on headers = $nh/(M+nh)$. 
Practice Problem (8)

• Assume that a group of 10 people wishes to communicate securely with each other. Each member of the group needs to send secret data to the other 9 people within the group. All communication between any two people $p$ and $q$ is visible to all other people in this group and no other person in the group should be able to decode their communication.
  
  • (a) If the group decides to use symmetric key encryption, how many keys are required in the system as a whole?
  
  • (b) Instead if public key encryption is chosen, how many keys would be required?
Practice Problem (8)

• Answer:

(a) if symmetric key encryption is used, then each pair of people communicating would require their own unique key. For N people this comes out to N(N-1)/2. Hence, for N=10, we have 45 (= 1 + 2 + 3 + ....+ 9)

(b) For public key encryption, each user needs its own public private key pair. All the other users to send data to him can use the public key. So in this case, 10 pairs of public and private keys will be needed