CSE 473 – Introduction to Computer Networks *Jonathan Turner*

Quiz 6 Solutions

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1. (4 points). Consider a network with four hosts *A*, *B*, *C*, *D* and two routers *X* and *Y* with IP addresses 1.2.3.31, 1.2.3.23. Which of these is the IGMP querier?

   Y is the querier

   Assume that *A* is configured to receive packets sent to 229.1.1.1 and 231.1.1.1, *B* is configured to receive packets sent to 237.1.1.1, *C* is configured to receive packets sent to 231.1.1.1 and *D* is configured to receive packets sent to 231.1.1.1 and 237.1.1.1. If the querier sends out an IGMP general query packet, how many IGMP membership reports will be sent by the hosts, assuming that the switches in the network do not implement IGMP snooping?

   One report is sent for each multicast address for which there is a subscriber. So three reports will be sent.

   How many membership reports will be sent by the hosts if the switches do implement IGMP snooping?

   In this case, every host sends a report for each of the multicast addresses it is subscribed to, so six reports will be sent.

2. (6 points) Consider a real-time audio stream sent from host *A* to host *B*. Assume that at time 0, *A* starts sending audio packets at a rate of one packet every 10 ms, with the audio encoded using 64 Kb/s PCM.

   How many audio samples are contained in each packet?

   80 samples per packet

   Suppose that *B* uses adaptive playout with parameter *K*=4 and that before the start of the talk spurt, the average delay measured at the receiver is 100 ms, and the average delay variation is 10 ms. Suppose that at time 170 ms, *B* receives a packet that was sent by *A* at time 50 ms. At what time will *B* start to play out the audio data contained in this packet? At what time will it finish playing the audio data from this packet?

   It will start playing out the audio at time 190 ms and finish at 200 ms.

   Suppose that the audio packets are sent using UDP and RTP across an Ethernet network and that the RTP header of 12 bytes. Approximately what fraction of each packet is actual audio data (as opposed to protocol fields). Account for all protocol fields, but you may ignore the Ethernet preamble and flag fields.

   The number of protocol bytes is 18+20+8+12=58. So the fraction of each packet that is audio data is 80/138, or roughly 55%.

   Suppose that the audio data is sent using RTP and the timestamp in the first RTP packet of the talk spurt is 30. What is the value of the timestamp in the second RTP packet?

   30+80=110