The University of Queensland
School of Information Technology and Electrical Engineering

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COMS3200 – Tutorial 10 - Solutions

Question 1

What is meant by interactivity for streaming stored audio/video? What is meant by interactivity for real-time interactive video?

The term "interactivity" for streaming stored audio/video means VCR-like operations (Pause, Rewind, FF, etc). In case of real-time interactive video, "interactivity" means communication interactions between distributed entities participating in real-time video, e.g. videoconference

Question 2

What is the difference between end-to-end delay and packet jitter?

Jitter is fluctuation of packet delay (often defined as the standard deviation of the delays experienced by packets).

Question 3

A compact disc holds 650 MB of data. Is compression used for audio CDs? Explain you reasoning.

Audio needs 1.4 Mbps, which is 175 KB/sec. On a 650MB device, there is room for 3714 sec of audio, which is just over an hour. CDs are never more than an hour long, so there is no need for compression and it is not used.

Question 4

An audio streaming server has a one-way distance of 50 msec with a media player. It outputs at 1 Mbps. If the media player has a 1MB buffer, what can you say about the position of the low-water mark and the high-water mark?

It takes 50 msec to get a pause command to the server, in which time 6250 bytes will arrive, so the low-water mark should be way above 6250, probably 50,000 to be safe. Similarly, the high-water mark should be at least 6250 bytes from the top, but, say, 50,000 would be safer.

Question 5

The H.323 and SIP protocols are two most popular protocols for Voice over IP (VoIP). Both of them are used by the networking vendors. For example the Cisco voice gateway supports both H.323 and SIP. Compare the H.323 and SIP protocols. Explain why H.323 prevails in the Internet backbone but is less popular than SIP in the "local loop" (domestic VoIP)?

Look into the Lecture Notes for a full comparison. Briefly: H.323 is a whole architecture of protocols handling call establishment, negotiations, encoding, etc., while SIP is a very simple protocol (HTTP like) which deals only with call establishment. SIP can however work with the encoding and control protocols listed in H.323.

As for the growing popularity of SIP - H.323 has been used in the Internet backbone for long time for voice and video conferencing. However H.323 is a very heavy suite of protocols and therefore it is very difficult to make it work in networks which use firewalls and/or NAT. Moreover, SIP is not only simpler but also similar to other Internet protocols (e.g. HTTP).

Many vendors equip their voice gateways with both H.323 and SIP (e.g. Cisco VoIP gateway).

Question 6

When audio is sent over the Internet it is often sent as "interleaved audio", i.e. odd numbered audio samples are sent in one packet and even numbered samples are sent in another packet (adjacent samples are not sent together). If one packet is lost it results in several small gaps in the reconstructed audio instead of one large gap. This allows the audio application to survive an occasional lost packet without introducing a gap in the playback. However, when *used for Internet telephony, it also has a small disadvantage. What is that?*

It introduces extra delay. Twice the number of samples have to be taken (compared to audio without sample interleaving) before a packet can be sent.

Question 7

Does voice over IP have the same problems with firewalls that streaming audio does? Discuss your answer.

It depends. If the caller is not behind a firewall and the callee is at a regular telephone, there are no problems at all. If the caller is behind a firewall and the firewall is not picky about what leaves the site, it will also work. If the callee is behind a firewall that will not let UDP/RTP packets in/out, it will not work.

Question 8

What is the bit rate for transmitting uncompressed 800 x 600 pixel colour frames with 8 bits/pixel at 40 frames/sec?

The number of bits/sec is just 800 * 600 * 40 * 8 or 153.6 Mbps.

Question 9

Consider a 100.000-customer video server, where each customer watches two movies per month. Half the movies are served at 8 pm. How many movies does the server have to transmit at once during this time period? If each movie requires 4 Mbps, how manu OC-12 connections (622 Mbps each) does the server need to do the work?

With 100,000 customers each getting two movies per month, the server outputs 200,000 movies per month or about 6600 per day. If half of these are at 8 pm, the server must handle about 3300 movies at once. If the server has to transmit 3300 movies at 4 Mbps each, the required bandwidth is 13.2 Gbps. Using OC-12 connections, with a capacity of 622 Mbps each, at least 22 connections will be needed. A machine serving 3300 movies simultaneously over 22 OC-12 connections has to be a large machine.

Question 10

Assume that real-time video is sent over TCP. What effect will the TCP congestion avoidance have on video delivery?

The TCP congestion avoidance uses slow start and also doubles timeout if a packet is lost. The timeout change will not have much impact (except for creating an additional network load) as retransmitted packets usually arrive too late for playback deadline. However the slow start will throttle video transmission.