

## Lecture 10 tutorial: Multimedia Networking

**During tutorials prepare a short report of your activities and show it to your tutor.**

Study the following **questions** and verify the correctness of the **answers** if given.

Be aware that the exam question might be directly related to the tutorial questions

**Additional Instructions:** Where a **group number** is indicated, please discuss this particular question with other members of your group, prepare a short written answer and email this to your tutor before the end of the class (you may wish to verify the correctness of your answer first). This will be used to produce a set of sample answers for study purposes.

**Note 1:** You should work through all of the questions **individually**, either during the tutorial itself or else later during revision, not just those assigned to your group.

**Note 2:** Questions marked **ALL** are considered higher priority and are for all students/groups; if you wish to include answers to these questions in an email report that is optional.

### Group 10

**Q1:** Construct a table showing bit rates for a specific activity and the number of bytes transferred in 67 mins when:

- Victor Video is watching a 4 Mbps video,
- Facebook Frank is looking at a new 100 Kbyte image every 20 seconds,
- Martha Music is listening to 200 kbps audio stream.

### ALL

**Q2:** There are two types of redundancy in video. Describe them, and discuss how they can be exploited for efficient compression.

### Group 9

**Q3:** Suppose an analog audio signal is sampled 16,000 times per second, and each sample is quantized into one of 1024 levels. What would be the resulting bit rate of the PCM digital audio signal?

### Group 8

**Q4:** Multimedia applications can be classified into three categories. Name and describe each category.

**ALL**

**Q5:** Streaming video systems can be classified into three categories. Name and briefly describe each of these categories.

**Group 7**

**Q6:** List three disadvantages of UDP streaming.

**Group 6**

**Q7:** With HTTP streaming, are the TCP receive buffer and the client's application buffer the same thing? If not, how do they interact?

**ALL**

**Q8:** Consider the simple model for HTTP streaming. Suppose the server sends bits at a constant rate of  $r = 2$  Mbps and playback begins when 8 million bits have been received. What is the initial buffering delay  $t_r$ ?

**Group 5**

**Q9:** CDNs typically adopt one of two different server placement philosophies. Name and briefly describe these two philosophies.

**Group 4**

**Q10:** Several cluster selection strategies were described in Slide 34. Which of these strategies finds a good cluster with respect to the client's LDNS? Which of these strategies finds a good cluster with respect to the client itself?

**Group 3**

**Q11:** Besides network-related considerations such as delay, loss, and bandwidth performance, there are many additional important factors that go into designing a cluster selection strategy. What are they?

**Group 2**

**Q12:** What is the difference between end-to-end delay and packet jitter? What are the causes of packet jitter?

### ALL - Important

**Q13:** Consider a Contents Delivery Network and simple contents access scenario. Assume that John gets URL for video <http://mycinema.com/7C62AB> from mycinema.com web page, but the video is stored at <http://KingCDN.com/myc7C6&2AB>.

**Sketch** the host/servers involved and **describe** the major steps in terms of the flow of protocol requests/responses until the video is streamed to John's PC. Itemize your description and show the steps in the sketch. (Refer to Slides 30-33).

### ALL

**Q14:** Describe VoIP playout technique with fixed playout delay

### Group 1

**Q15:** Describe Skype client operations

### Optional

**Q16: True or false** (refers to RTP)

- a. If stored video is streamed directly from a Web server to a media player, then the application is using TCP as an underlying transport protocol
- b. When using RTP, it is possible for a sender to change encoding in the middle of a session
- c. All applications that use RTP must use port 87.
- d. If an RTP session has a separate audio and video stream for each sender, then the audio and video streams use the same SSRC.
- e. In differentiated services, while per-hop behavior defines differences in performance among classes, it does not mandate any particular mechanism for achieving these performances.
- f. Suppose Alice wants to establish an SIP session with Bob. In her INVITE message she includes the line: m=audio 48753 RTP/AVP 3 (AVP 3 denotes GSM audio). Alice has therefore indicated in this message that she wishes to send GSM audio.
- g. Referring to the preceding statement, Alice has indicated in her INVITE message that she will send audio to port 48753.
- h. SIP messages are typically sent between SIP entities using a default SIP port number.
- i. In order to maintain registration, SIP clients must periodically send REGISTER messages.
- j. SIP mandates that all SIP clients support G.711 audio encoding.