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

Do not miss Q13!

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.

A1:

	Bit rate	Bytes transferred in 67 mins
Facebook Frank	40 kbps	20 Mbytes
Martha Music	200 kbps	100 Mbytes
Victor Video	4 Mbps	2 Gbytes

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

A2:

- Spatial Redundancy**:** It is the redundancy within a given image. Intuitively, an image consists of mostly white space has a high degree of redundancy and can be efficiently compressed without significantly sacrificing image quality.
- Temporal Redundancy reflects repetition from image to subsequent image. If, for example, an image and the subsequent image are exactly the same, there is no reason to re-encode the subsequent image; it is instead more efficient simply to indicate during encoding that the subsequent image is exactly the same. If the two images are very similar, it may be not efficient to indicate how the second image differs from the first, rather than re-encode the second image.

Q3: Suppose an analog audio signal is sampled 16,000 times per second, and each sample is quantized into one of l0241evels. What would be the resulting bit rate of the PCM digital audio signal?

A2:
Quantizing a sample into 1024 levels means 10 bits per sample. The resulting rate of the PCM digital audio signal is 160 Kbps.

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

A4:

- Streaming stored audio/video: In this class of applications, the underlying medium is prerecorded video, such as a movie, a television show, or a prerecorded sporting event. These prerecorded videos are played on servers, and users send requests to the servers to view the videos on demand. Many internet companies today provide streaming video, including YouTube, Netflix, and Hulu.

- Conversational Voice- and Video-over-IP**:** Real-time conversational voice over the Internet is often referred to as Internet telephony, since, from the user's perspective, it is similar to the traditional circuit-switched telephone service. It is also commonly called Voice-over-IP (VOIP).Conversational video is similar except that it includes the video of the participants as well as their voices. Conversational voice and video are widely used in the Internet today, with the Internet companies like Skype and Google Talk boasting hundreds of millions of daily users.

- Streaming Live Audio and Video: These applications allow users to receive a live radio or television transmission over the Internet. Today, thousands of radio and television stations around the world are broadcasting content over the internet.

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

A5:

- UDP Streaming: With UDP streaming, the server transmits video at a rate that matches the client's video consumption rate by clocking out the video chunks over UDP at a steady rate.

- HTTP Streaming: In HTTP streaming, the video simply stored in an HTTP server as ordinary file with a specific URL. When a user wants to see the video, the client establishes a TCP connection with the server and issues an HTTP GET request for that URL. The server then sends the video file, within an HTTP response message, as quickly as possible, that is, as quickly as TCP congestion control and flow control will allow.

- Adaptive HTTP Streaming (DASH): In Dynamic Adaptive Streaming over HTTP, the video is encoded several different versions, with each version having a different bit rate and, correspondingly, a different quality level. The client dynamically requests the chunks of video segments of a few seconds in length from the different versions. When the amount of available bandwidth is high, the client naturally selects chunks from a high-rate version; and when the available bandwidth is low, it naturally selects from a low-rate version.

Q6: List three disadvantages of UDP streaming.

A6:
The three significant drawbacks of UDP Streaming are:

1. Due to unpredictable and varying amount of available bandwidth between server and client, constant-rate UDP streaming can fail to provide continuous play out.
2. It requires a media control server, such as an RTSP server, to process client-to-server interactivity requests and to track client state for each ongoing client session.
3. Many firewalls are configured to block UDP traffic, preventing users behind these firewalls from receiving UDP video.

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

A7:
No. On the client side, the client application reads bytes from the TCP receive buffer and places the bytes in the client application buffer.

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?

A8:
The initial buffering delay is $t_p = Q/r = 4$ seconds.

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

A9:

- Enter Deep: This philosophy is to enter deep into the access networks of Internet Service Providers, by deploying server clusters in access ISPs all over the world.

- Bring Home: A second design philosophy is to bring the ISP's home by building large clusters at a smaller number of key locations and connecting these clusters using a private high-speed network.

Q10: Several cluster selection strategies were described in Slide 27. 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?

A10:
Geographically closest cluster selection and real time measurements selection can find a good cluster with respect to LDNS. IP anycast chooses good cluster with respect to client itself.

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?

A11:

- Load on the cluster – clients should not be directed to overload clusters.

- ISP delivery cost – the clusters may be chosen so that specific ISPs are used to carry CDN-to-client traffic, taking into account the different cost structures in the contractual relationships between ISPs and cluster operators.

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

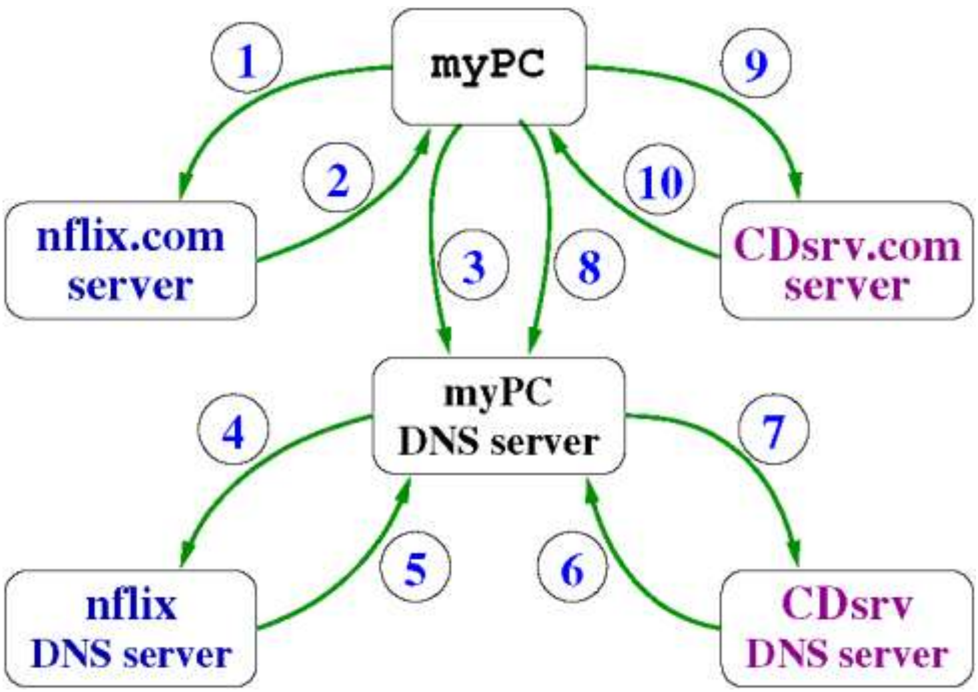
A12:

- End-to-end delay is the time it takes a packet to travel across the network from source to destination.
- Delay jitter is the fluctuation of end-to-end delay from packet to the next packet.

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

A13: Slides 28, 29

Q13a: Another version of the **contents access scenario** example:



In the example **myPC** is searching for a movie on the **nflix.com** server. After the movie is selected and allowed to be screened (e.g. after paying the price of access to the movie) a PLAY app in myPC use DNS (Domain Name Services) to obtain the IP address of the selected movie to be screened from the **CDsrv.com** server. In the example the **DNS servers** use the iterative method of obtaining the IP address.
Describe activities in each step as marked in the above diagram. (Slides 28, 29)

Q14: Describe VoIP playout technique with fixed playout delay

A14: Slides 37, 38

Q15: Describe Skype client operations

A15: Slide 43

Q16: **True or false** (**optional**, 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.

A 16

a) True
b) True
c) No, RTP streams can be sent to/from any port number.
d) No, typically they are assigned different SSRC values.
e) True
f) False, she is indicating that she wishes to *receive* GSM audio
g) False, she is indicating that she wishes to *receive* audio on port 48753
h) True, 5060 for both source and destination port numbers
i) True
j) False, this is a requirement of H.323 and not SIP.