Assignment 9

R1. Reconstruct Table 9.1 for when Victor Video is watching a 4 Mbps video, Facebook Frank is looking at a new 100 Kbyte image every 20 seconds, and Martha Music is listening to 200 kbps audio stream.

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| --- | --- | --- |
|  | Bit Rate | Bytes Transferred |
| Facebook Frank | 40 kbps | 20 Mbytes |
| Martha Music | 200 kbps | 100 Mbytes |
| Victor Video | 4 Mbps | 2Gbytes |

R2. There are two types of redundancy in video. Describe them and discuss how they can be exploited for efficient compression.

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 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 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.

R3. 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?

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

R4. Multimedia applications can be classified into three categories. Name and describe each category.

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 like 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.

R5. Streaming video systems can be classified into three categories. Name and briefly describe each of these categories.

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.

R6. List three disadvantages of UDP streaming.

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.

R7. With HTTP streaming, are the TCP receive buffer and the client’s application buffer the same thing? If not, how do they interact?

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

R8. Consider the simple model for HTTP streaming. Suppose the server sends bits at a constant rate of 2 Mbps and playback begins when 8 million bits have been received. What is the initial buffering delay tap?

The initial buffering delay is tap = Q/x = 4 seconds.

R9. What is the difference between end-to-end delay and packet jitter? What are the causes of packet jitter?

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.

R10. Why is a packet that is received after its scheduled playout time considered lost?

A packet that arrives after its scheduled play out time cannot be played out. Therefore, from the perspective of the application, the packet has been lost.

R11. Section 9.3 describes two FEC schemes. Briefly summarize them. Both schemes increase the transmission rate of the stream by adding overhead. Does interleaving also increase the

First scheme: send a redundant encoded chunk after every n chunk; the redundant chunk is obtained by exclusive OR-in the n original chunks.

Second scheme: send a lower-resolution low-bit rate scheme along with the original stream. Interleaving does not increase the bandwidth requirements of a stream.

R12. How are different RTP streams in different sessions identified by a receiver? How are different streams from within the same session identified?

RTP streams in different sessions: different multicast addresses; RTP streams in the same session: SSRC field; RTP packets are distinguished from RTCP packets by using distinct port numbers

R13. What is the role of a SIP registrar? How is the role of a SIP registrar different from that of a home agent in Mobile IP?

The role of a SIP registrar is to keep track of the users and their corresponding IP addresses which they are currently using. Each SIP registrar keeps track of the users that belong to its domain. It also forwards INVITE messages (for users in its domain) to the IP address which the user is currently using. In this regard, its role is like that of an authoritative name server in DNS.

**Work Cited**

https://usermanual.wiki/Document/Solutions20Manual20for20Computer20Networking20A20TopDown20Approach2C207th20Edition.609668037/view