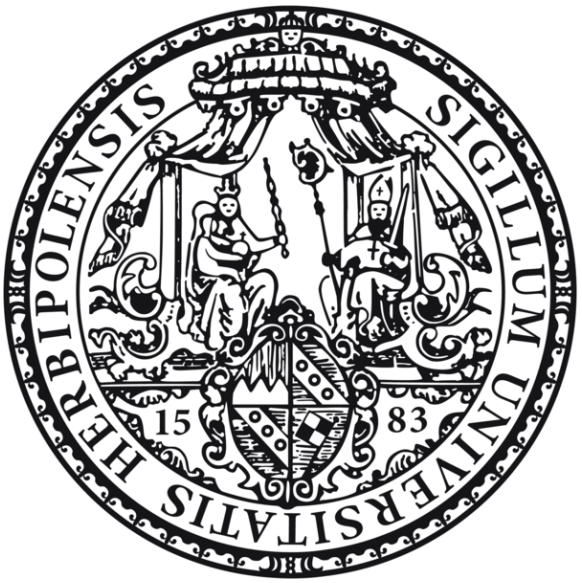
Hardwarepraktikum Internet-Technologien

Comprehension Questions of Task 7: Paquet-lost and Latency



Julius-Maximilians-Universität Würzburg

Chair of Computer Science III

A project report submitted by **group 11**

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Summer term 2022

May, 2022

Contents

1. Even with a perfect connection (no latency and no packet loss), the buffers of arecord and aplay are necessary to transmit the signal correctly. Why? Consider how the buffer content changes over time during playback. How does the content change during playback? What happens when a packet arrives? Under perfect conditions, how large a buffer be chosen?

Because, when the information is sent, the system that receives it must process it, therefore, and so that it does not collapse, the information received is stored in the buffer so that the system can access it when it has finished processing the information previously received.

The information that arrives first is the first to be removed (FIFO architecture, First In, First Out).Enough to saturate the link, increase throughput efficiency. Too much buffering can increase delays.

A good buffer size for recording is 128 samples, but you can also get away with raising the buffer size up to 256 samples without being able to detect much latency in the signal.

2. Which layer 4 protocol is used in reality for real-time audio transmission (voice chat, IP telephony), and why?

VoIP (Voice on Internet Protocol).

-VoIP end-to-end-delay requirements, needs to maintain “conversational” aspect.

-Includes application-level (packetization, playout), network delays.

-Value-added services: call forwarding, screening, recording.

Loss tolerance: packet loss rates between 1% and 10% can be tolerated.

3. Which characteristics do you think have a greater impact on audio quality? Average latency, latency variance, or packet loss?

Packet loss can cause information to be lost, but human hearing and understanding can approximate what the transmitter is saying, whereas too high a latency can cause the conversation to be out of sync and therefore not understood. (Excessive packet loss can also cause the conversation not to be understood at all). Finally, constant changes in latency can cause the sender to be heard too quickly (when packets are retrieved all at once) or to pause the conversation.

4. TCP ensures that all packets arrive at their destination by acknowledging each transmission with an ACK. Compared to UDP, what other property is guaranteed? Is this important for audio transmission?

Useful for constantly sending packets (e.g. avatar updates in a video game).

It does not require coordination between client and server (no handshake protocol), and is faster than TCP (UDP mailbox, TCP telephone).

In-Order delivery guarantee (TCP).

Also, TCP, being based on the handshake protocol, detects disconnection, unlike UDP.

TCP checks for dropped packets (UDP does not).

TCP transmits per stream (UDP per packet).

TCP only unicasting, UDP uni-, multi-, and broadcasting.

5. Transfer your impressions to other applications. What would you rather use TCP for, what would you rather use UDP for, and why?

a) YouTube and other pre-recorded videos, UDP because you need to download the video packets.

b) Live broadcast of a football match, TCP for button control (pause, volume), UDP for video packets.

c) Video telephony, TCP because it requires coordination between the sender and receiver.

d) First-person shooter games, UDP packets indicating movement and map update. TCP for player information (K/D scores).

e) Gaming in the cloud (e.g. Google Stadia), TCP for button control (virtualised controller), UDP for video packets (environment update).

f) Long-range communication (e.g. satellite). UDP for transmitting information packets.