

Computer Networks

Homework #9

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Classes of Multimedia Applications

- Q: Name and characterize three classes of multimedia applications. Add one example for each class.

Classes of Multimedia Applications (cont'd)

- **Stored streaming**
 - Media already present (stored) at the source
 - Streaming: Media is transmitted to the client in time for playing out
 - Client can already begin playing out before all the data is transmitted
 - Interactivity: VCR-like functionality (e.g. pause, rewind, forward, ...)
 - Example: YouTube, Hulu

Classes of Multimedia Applications (cont'd)

- **Live streaming**
 - Media is streamed while it is being produced
 - Streaming: Media is transmitted while being recorded, small gaps (tens of seconds) possible
 - Client can begin playing out as soon as enough buffered data is available
 - Interactivity: Forward skipping not possible, other functions (pause, rewind, ...) possible
 - Example: ffn.de, zattoo.com

Classes of Multimedia Applications (cont'd)

- **Real-time interactive streaming**
 - Media is streamed while it is being produced (at multiple sources)
 - Streaming: Media needs to be transmitted immediately (real-time)
 - Client must playout media as soon as possible (e.g. audio should have < 400ms end-to-end delay)
 - Interactivity: Live audio/video interaction
 - Example: Skype, Google Talk

UDP vs. TCP

- Q: Discuss the usage of UDP vs. TCP to stream multimedia.
- UDP
 - Unreliable, no retransmissions
 - Error recovery has to be handled on application level (if time permits)
 - Oblivious to network congestion
 - Sending rate = encoding rate
 - Difficulties to pass firewalls/NATs

UDP vs. TCP (cont'd)

- TCP
 - Reliable data transfer
 - Retransmissions occur (and introduce delay) whether application likes it or not
 - Yields to network congestion
 - Sending rate \leq encoding rate
 - Passes more easily through firewalls/NATs

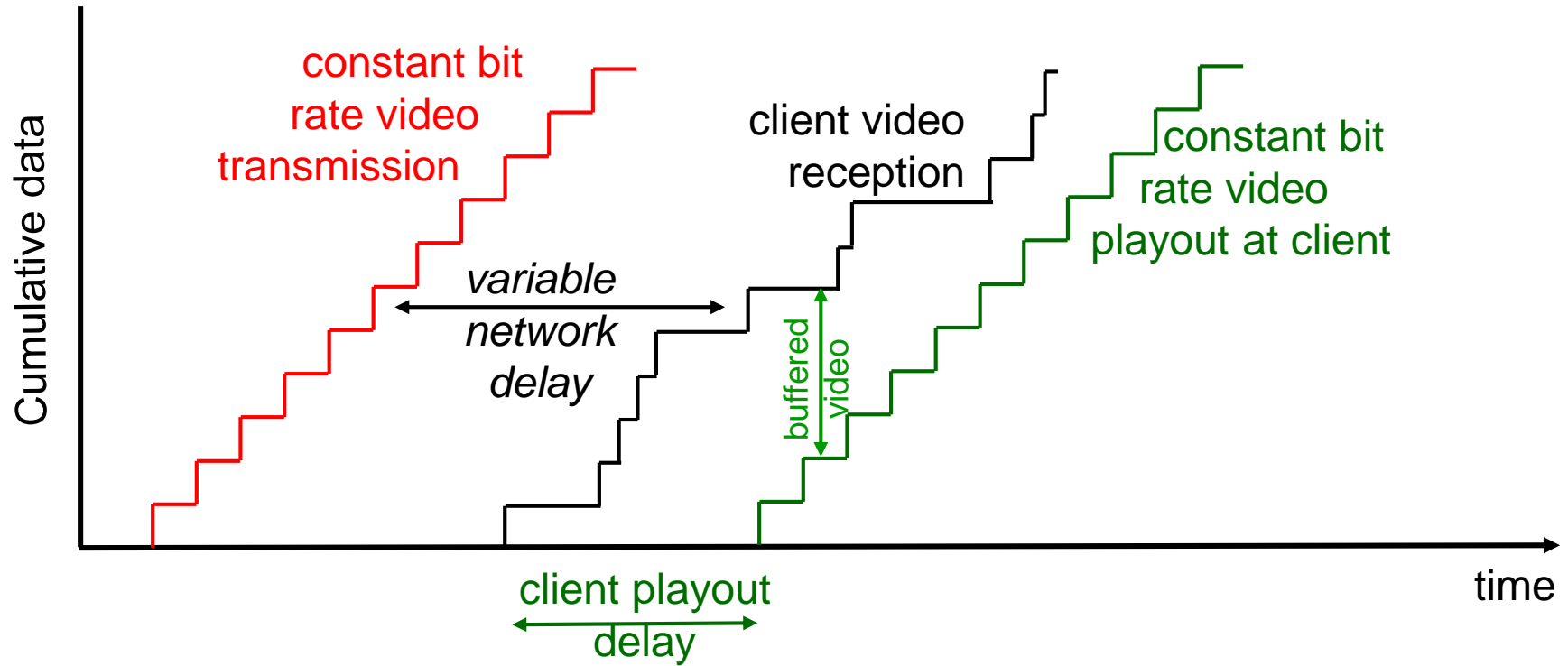
UDP vs. TCP (cont'd)

- UDP
 - Well suited for media with short playout delay (no time for retransmissions anyway)
 - Not well suited if reliable data transfer is important
- TCP
 - Suited for media with long playout delay
 - Easier to pass firewalls/NATs with

Jitter

- Q: What is jitter and how does it impact the playback of multimedia content? Which client-side mechanism can be used to compensate for jitter?
- Jitter is the variability over time of the packet latency across a network.
- Client-side buffering is commonly used to compensate for jitter.

Jitter (cont'd)



Buffering

- Q: Assume a multimedia application streams data with a constant bit rate of 2 Mbps. The network delay is between 0 and 2 seconds. How long does a client have to wait before it can begin to playback the stream and how much data does it have to buffer.
- The client has to wait 2 seconds and buffer 4Mb at most (worst case)

Adaptive Playout Delay

- What is the goal of adaptive playout delay? Give a brief overview of the steps involved.
- In voice over IP applications, adaptive playout delay is used to minimize the playout delay by dynamically adjusting the playout delay to the current network conditions.

Adaptive Playout Delay - Steps

- Every packet is time stamped
- Receiver maintains moving average of delay:

$$d_i = (1 - u)d_{i-1} + u(r_i - t_i)$$

with

t_i = time packet i is sent

r_i = time packet i received

- Receiver dynamically adjusts playout delay at the beginning of each talk spurt

Adaptive Playout Delay - Steps (cont'd)

- Receiver dynamically adjusts playout delay at the beginning of each talk spurt:

$$p_i = t_i + d_i + Kv_i$$

- Talk spurts can be identified by

$$t_i - t_{i-1} > 20\text{msec}$$

(given that the sending interval is 20 msec)

RTCP

- Q: What is the role of RTCP? How can it be used to synchronize RTP streams?
- The Real Time Control Protocol (RTCP) is used to periodically send control packets to all participants in a RTP session
- It primarily provides feedback on the quality of the RTP data distribution
- It *does not* control the RTP media streams (e.g. play, pause, ff) => RTSP

SIP

- Q: Illustrate how a voice call is established using the Session Initiation Protocol.

Setting up a call to known IP address

Alice

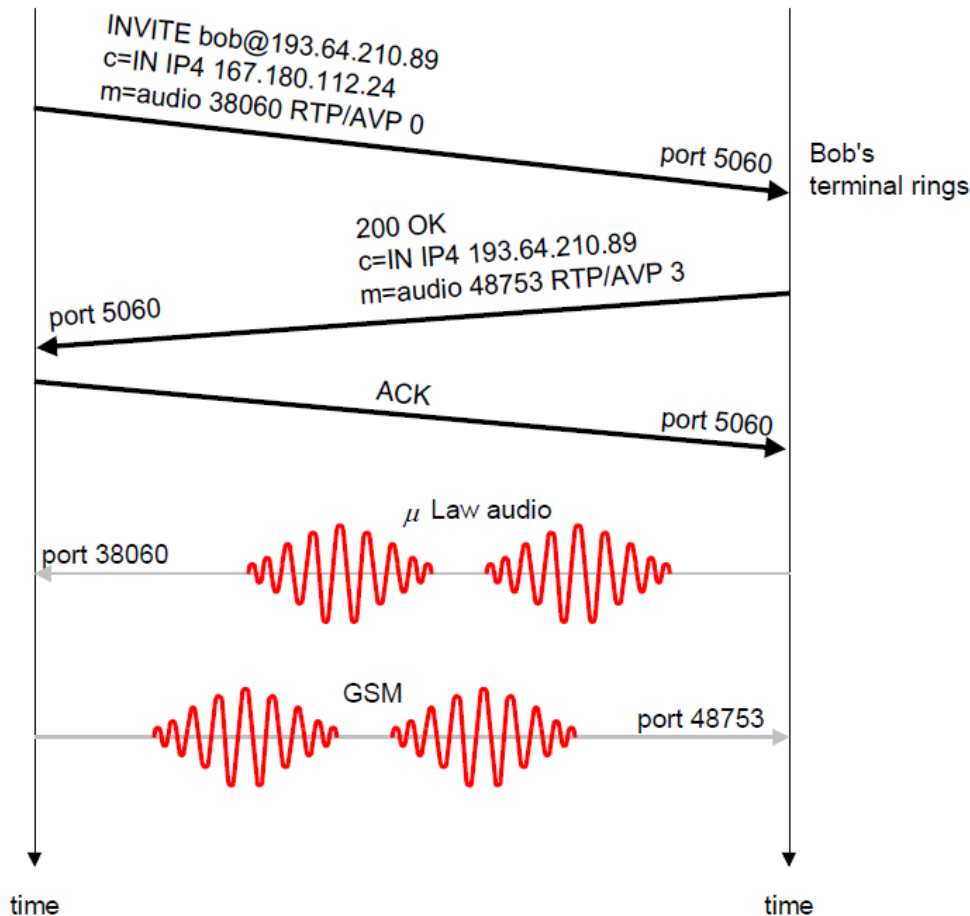


167.180.112.24

Bob



193.64.210.89



□ Alice's SIP invite message indicates her port number, IP address, encoding she prefers to receive (PCM ulaw)

□ Bob's 200 OK message indicates his port number, IP address, preferred encoding (GSM)

□ SIP messages can be sent over TCP or UDP; here sent over RTP/UDP.

□ default SIP port number is 5060.

Thank you

Any questions?