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 <= encoding rate
  - Passes more easily through firewalls/NATs

UDP vs. TCP (cont'd)

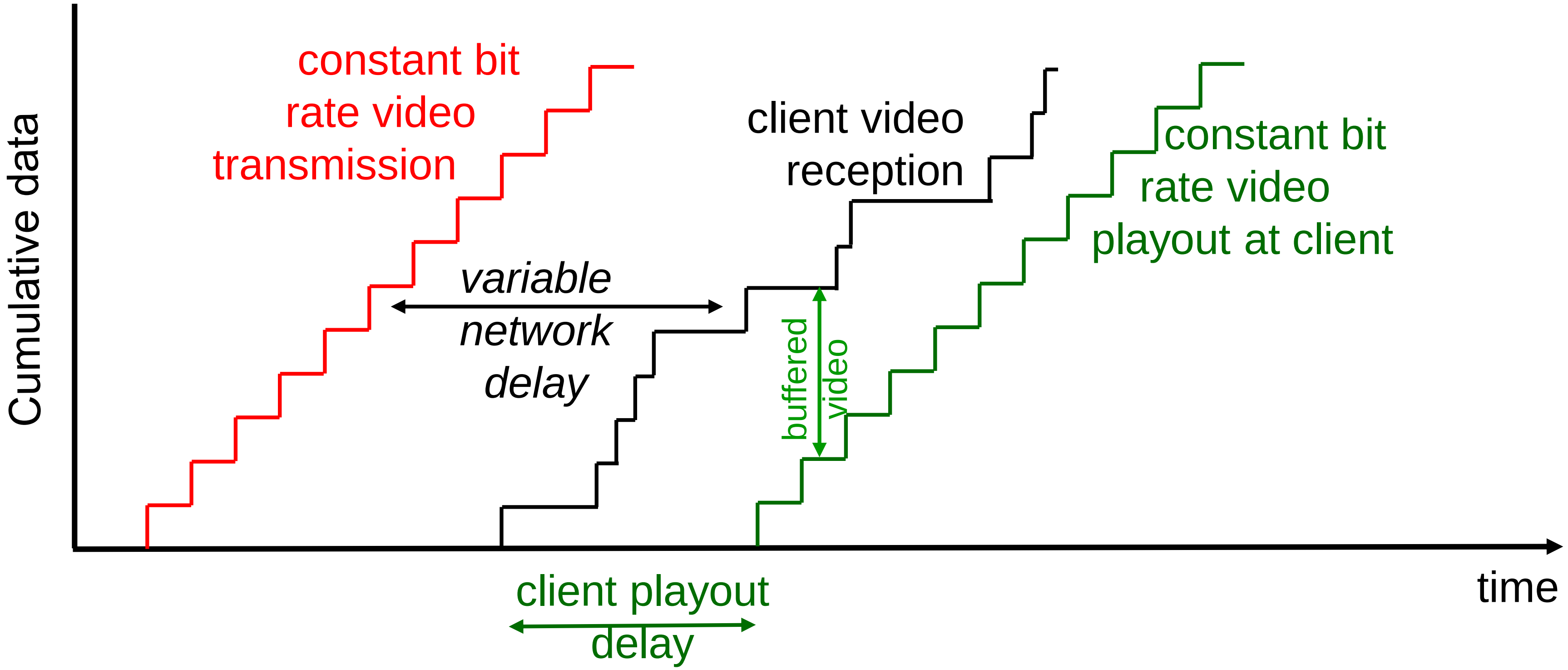
- UDP
  - Well suited for media with short playout delay (no time for retransmissions anyway)
  - Not well suited if reliable data transfer is important
- TCP
  - Suited for media with long playout delay
  - Easier to pass firewalls/NATs with

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 Mbs. 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:

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

$$\text{with}$$

$$t_i = \text{time packet } i \text{ is sent}$$

$$r_i = \text{time packet } i \text{ received}$$

- 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

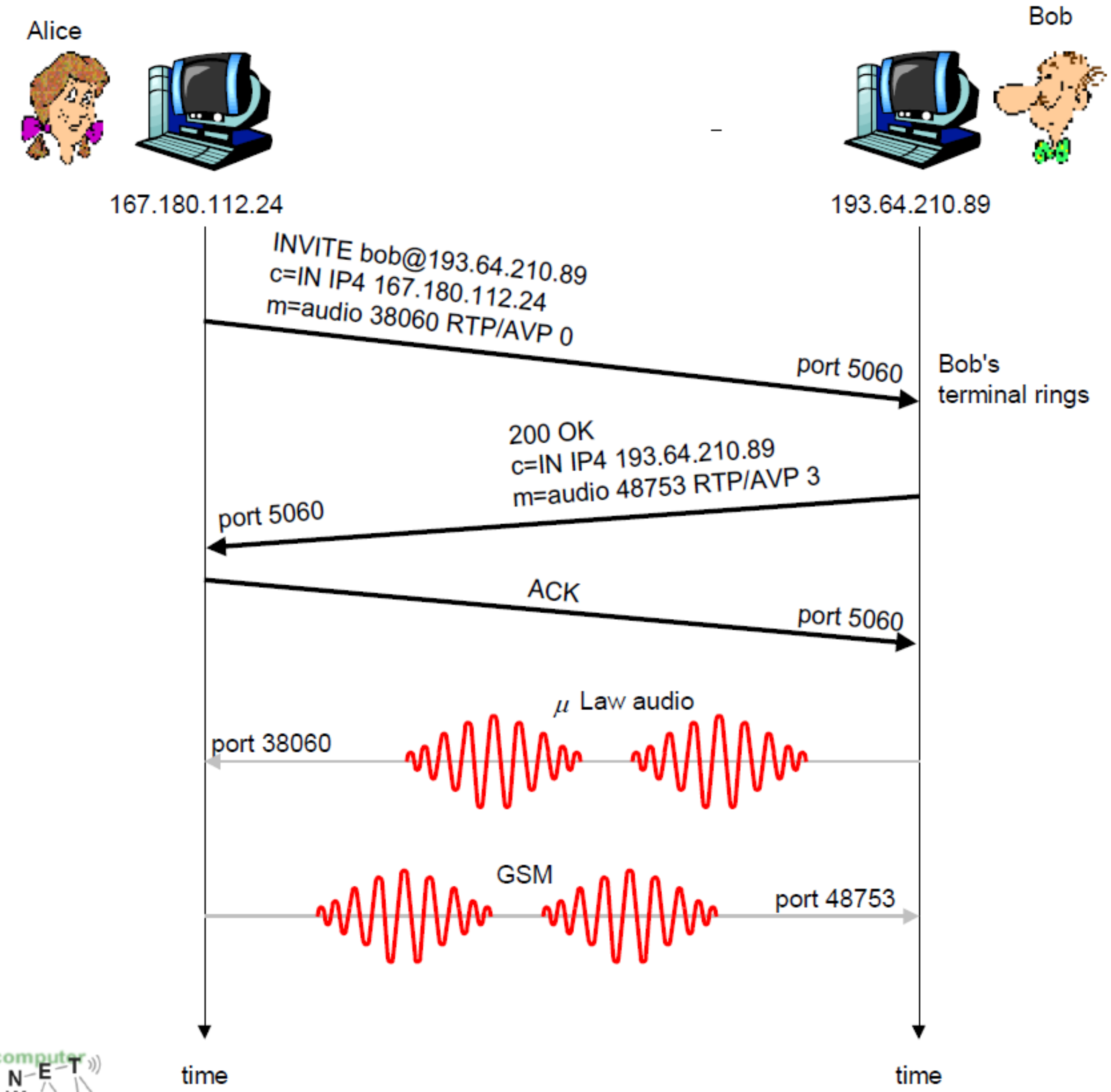
RTCP - Synchronization of Streams

- Scenario: Videoconference application with different audio/video streams for each participant
- Audio/video streams are time stamped but not tied to common wall clock time (sampling clock is independent of other clocks)
- RTCP Sender Report contains RTP time stamp and NTP time for last RTP packet
- Combination of RTP timestamp and NTP time allows for synchronization of streams

SIP

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

Setting up a call to known IP address



- 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?