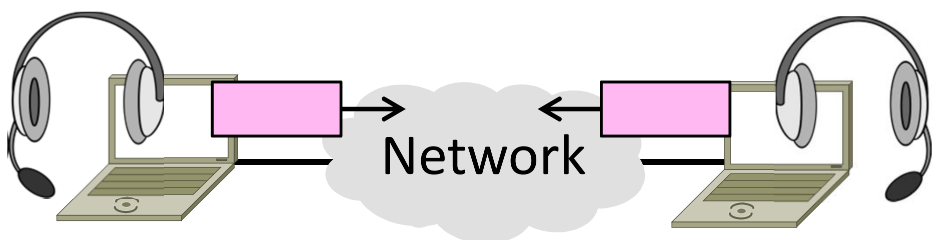
Computer Networks

Real-Time Transport (§6.4.3, §7.4.4-7.5.5)

David Wetherall (djw@uw.edu)
Professor of Computer Science & Engineering
UNIVERSITY *of* WASHINGTON

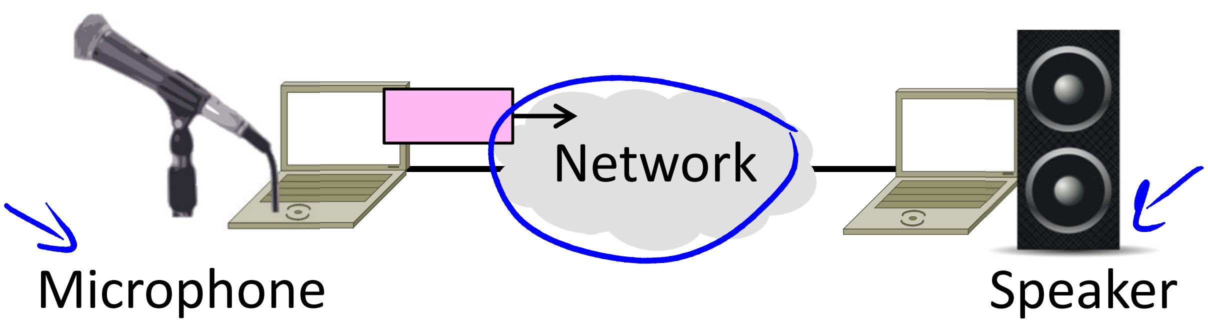
Topic

- Sending interactive real-time media over the network, e.g., VoIP
  - Using the best effort Internet
  - Playout buffer technique



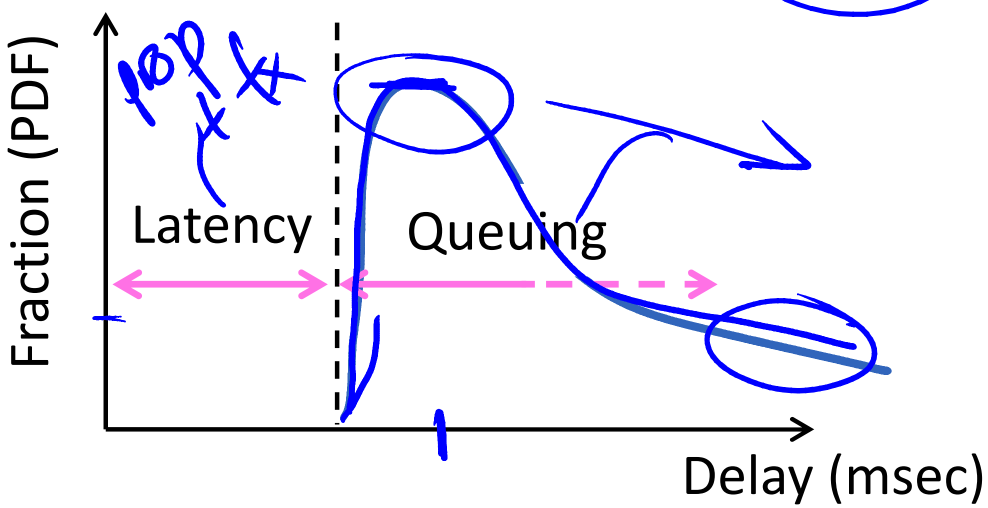
Challenge – Network Delay

- Consider one direction
  - Constant rate of media is generated at source, later consumed at receiver
  - Network must have enough bandwidth, and adds a delay



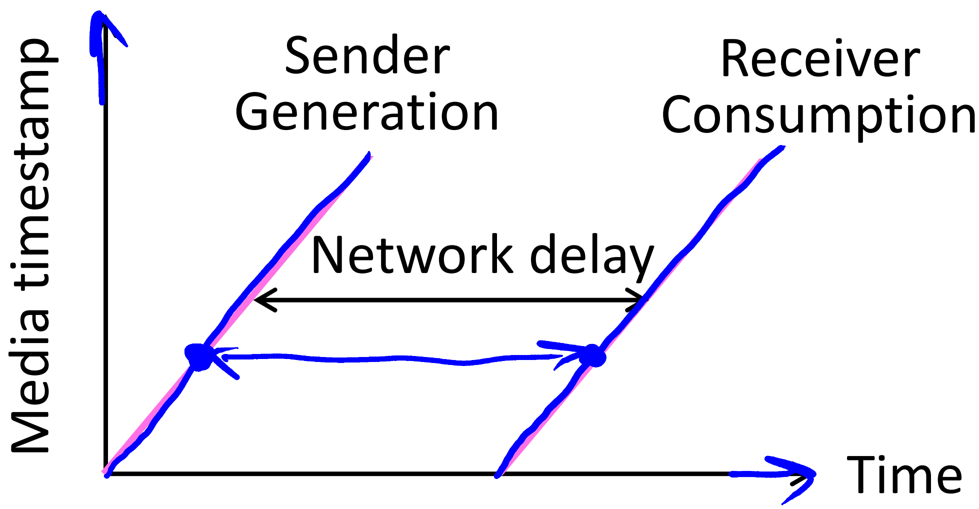
Network Delay (2)

- Network delay is variable
  - Message latency plus queuing delay
  - Variability in delay is called <u>jitter</u>



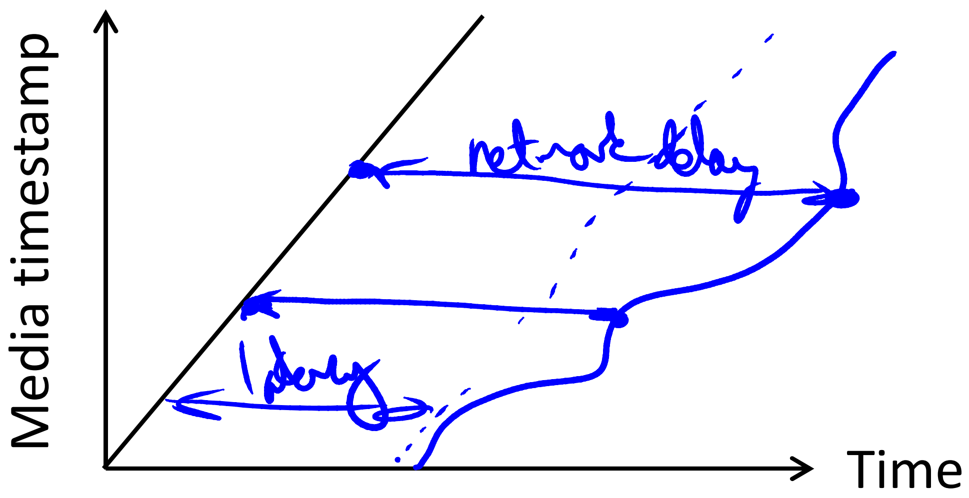
Playout

- Ideally want fixed, and small network delay for interactivity
  - Emulate the telephone network



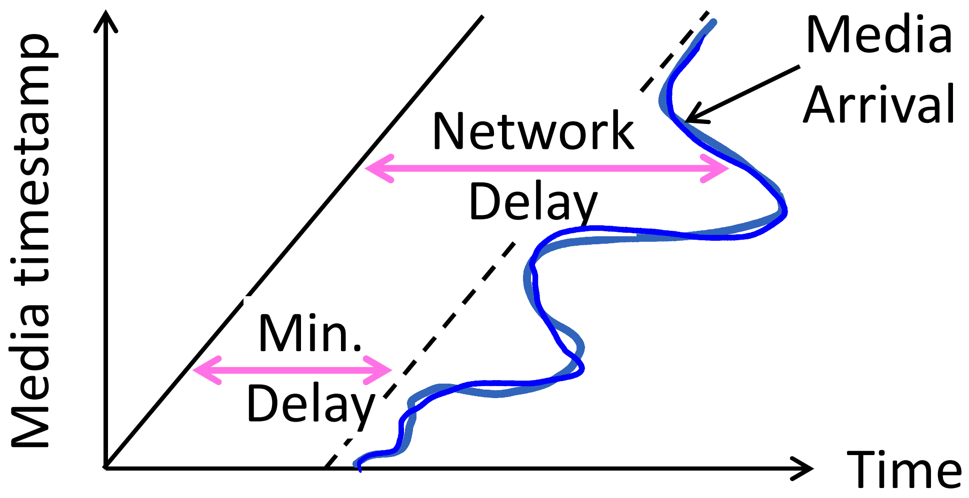
Playout (2)

- Media arrives at receiver after variable network delay



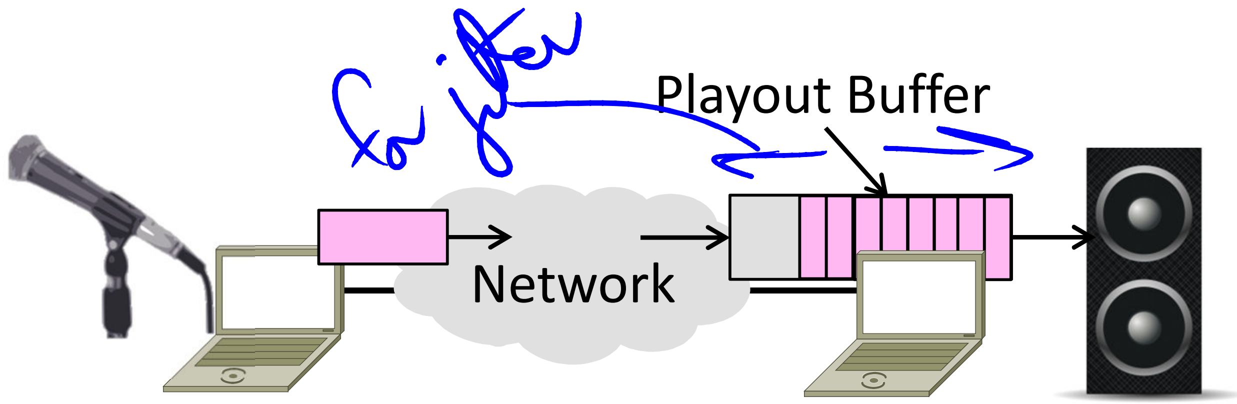
Playout (3)

- Media arrives at receiver after variable network delay



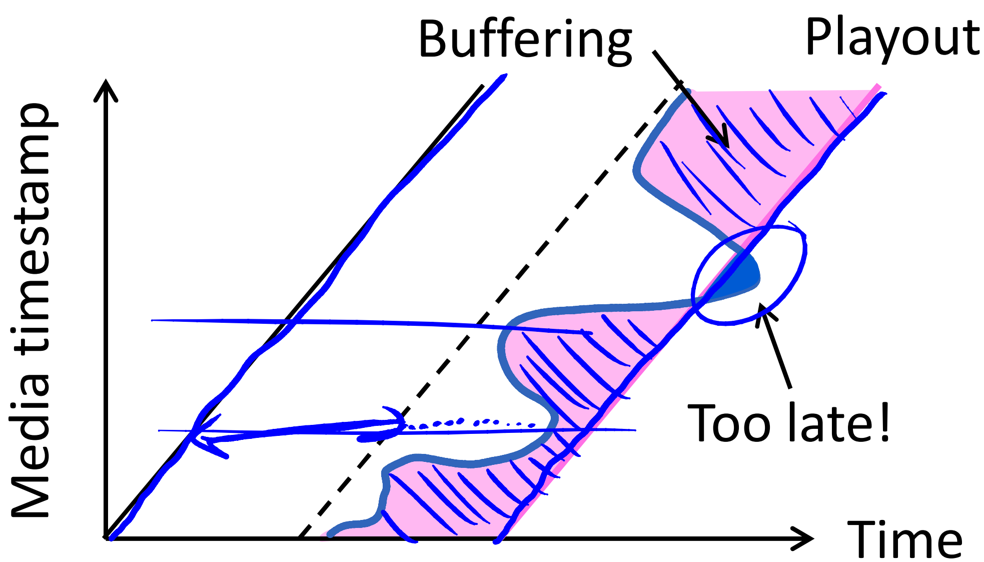
Playout Buffer

- Put media in <u>playout buffer</u> at receiver until consumption time
  - Smooth out variable network delay



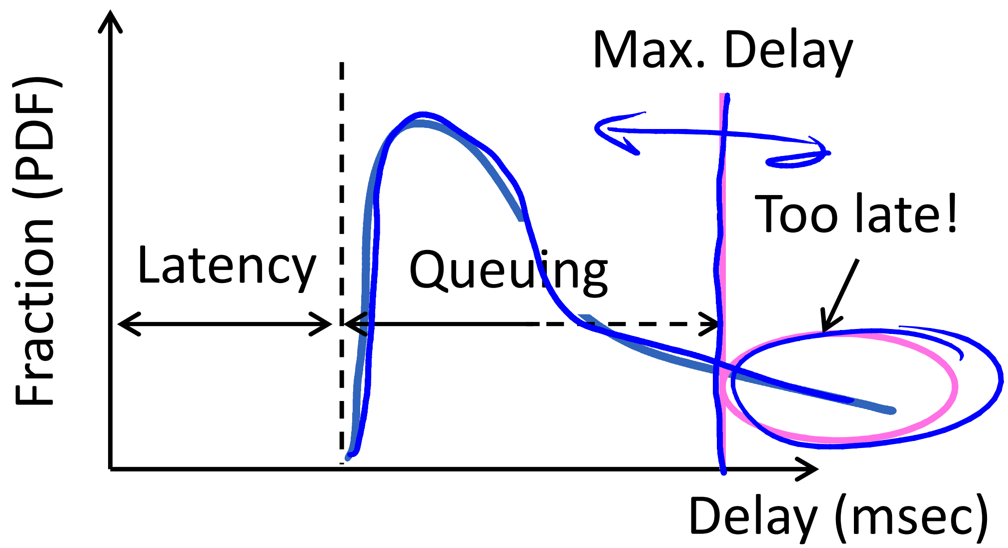
Playout Buffer (2)

- Media arrival curve determines time in playout buffer and deadline



Playout Buffer (3)

- Pick largest acceptable network delay to set the playout point



Playout Buffer (4)

- Tradeoff:
  - Larger acceptable network delay → larger buffer/delay, less loss
  - Smaller acceptable network delay → smaller buffer/delay, more loss
- Typically can't recover loss for interactive, real-time scenario
  - Instead, do without (glitch)

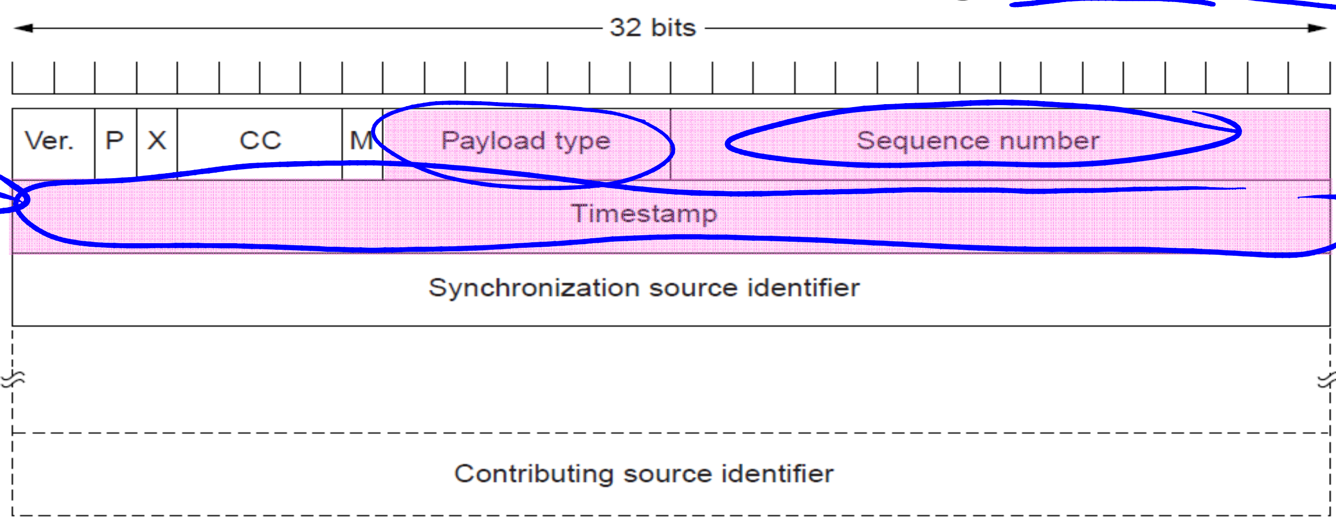
Components of a Real-Time Session

- A call consists of several parts:
  - Call setup, with SIP »
  - Session description, with SDP
  - Media transport, with RTP »
  - Media playout, with buffer

- May have audio/video, multiple parties, mobility, etc.

RTP (Real-time Transport Protocol)

- Used to carry media on top of best effort UDP (§6.4.3)
  - Header has media format, timestamp, sequence number, etc.
  - Media follows in standard formats, e.g., G.711, MP4



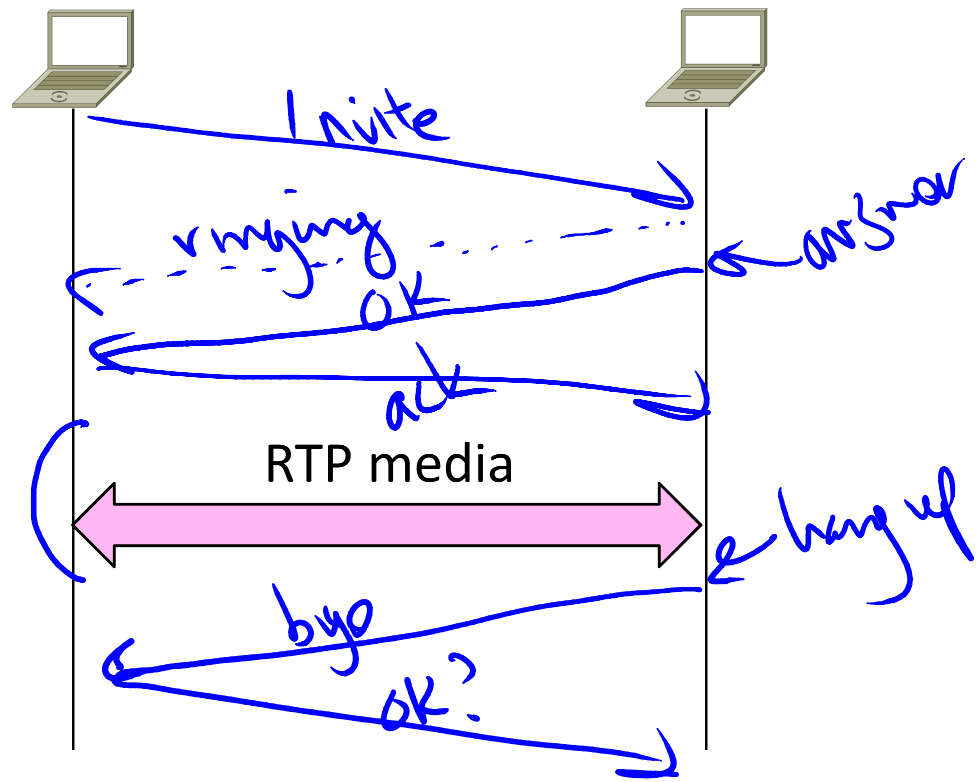
SIP (Session Initiation Protocol)

- Open protocol for establishing voice and video calls over IP
  - Provides the signaling; media is carried directly with RTP (or other)

- This is not Skype
  - It uses a proprietary protocol …

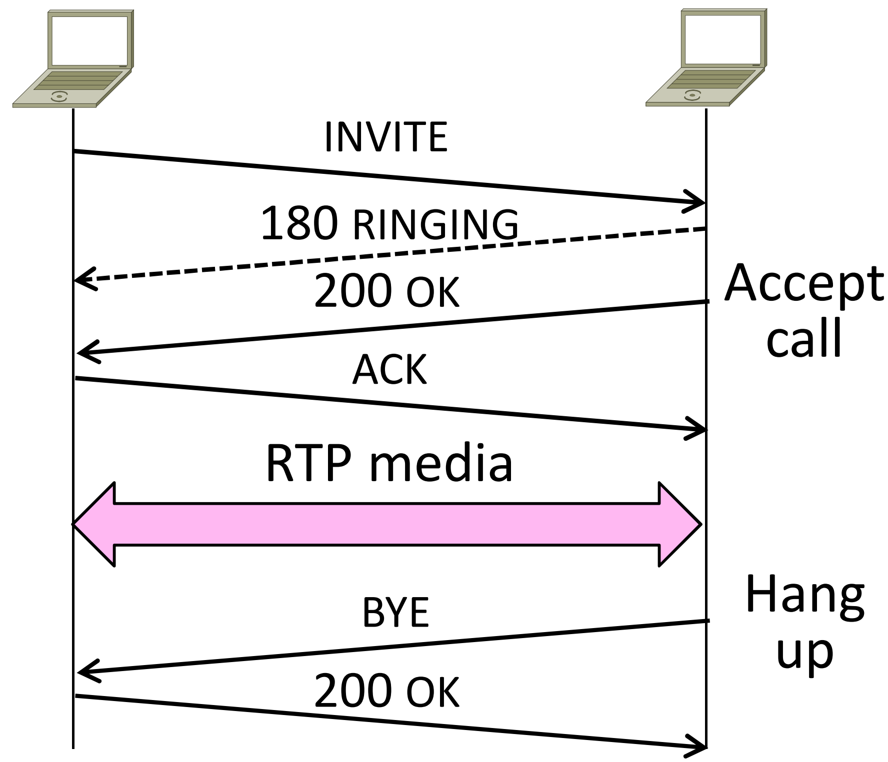
SIP Signaling

- Signaling for call control
  - Like HTTP, uses simple method/response codes
  - Runs on UDP or TCP
  - SIP proxy servers and registrars provide mobility (not shown)



SIP Signaling (2)

- Signaling for call control
  - Like HTTP, uses simple method/response codes
  - Runs on UDP or TCP
  - SIP proxy servers and registrars provide mobility (not shown)



END