Multimedia Networking

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Assignment #5

- Choose one of the given choices or propose something new
- Demo in 12/7-12/9 (also turnin stuff as usual)
- For your own topic:
  - must substantially related to computer networks.
  - must contact the TA and me to have approval before next Monday.
  - depending on the scope of the project, you may work in a group of two.
- For some assignments, you might need to access PlanetLab
  - go to my office after next Monday's class and I will set them up for you

Multimedia and Quality of Service

Multimedia applications: network audio and video ("continuous media")

QoS support
network provides application with level of performance needed for application to function.

Stored Multimedia

- media stored at source
- transmitted to client
- client playout may begin before all data has arrived (streaming)
  - timing constraint for still-to-be transmitted data: in time for playout
  - sensitive to delay jitter: variability of packet delays within the same packet stream
Real-Time Interactive Multimedia

- applications:
  - IP telephony, video conference
  - sensitive to both end-to-end delay and jitters
- end-end delay requirements:
  - audio: < 150msec good, < 400msec OK
  - higher delays noticeable, impair interactivity
  - includes application-level (compression/decompression) and network delays

Multimedia Networking

Classes of MM applications:
- Stored audio and video
- Real-time interactive audio and video

Fundamental characteristics:
- Typically delay sensitive
  - delay jitter
  - end-to-end delay
- But loss tolerant: infrequent losses cause minor glitches

QoS support for multimedia networking:
- No direct support from the Internet (best effort, no delay guarantee)
- Employ application techniques to mitigate problems

Outline

- Multimedia applications
- Stored multimedia
- Real-time interactive multimedia
- RTP (Real-time Transport Protocol)

Stored Multimedia over the Internet: the basic approach

- audio or video stored in files
- files transferred as HTTP object
  - received in entirety at client
  - then passed to player
- media player
  - decompression depending on media format
  - user interface allow interactive control
  - long delays until playout!
Stored Multimedia over the Internet: streaming approach

- browser GETs metafile
  - location of the media file, type of encoding, ...
- browser launches the media player, passing metafile
- the media player contacts server and the server streams audio/video to player
  - pipelining: client playout may begin before all data has arrived

Streaming from a Streaming Server

- This architecture allows for non-HTTP protocol between server and media player
- Can also use UDP instead of TCP.

Streaming Multimedia: Client Buffering

- Client-side buffering, playout delay compensates for network-added delay, delay jitter
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Real-time Interactive Multimedia: Internet Phone
- speaker’s audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurts
- packets generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes data
- application-layer header added to each chunk.
- chunk+header encapsulated into UDP segment.
- application sends UDP segment into socket every 20 msec during talk spurt.

Internet Phone: Playout Delay
- Receiver attempts to playout each chunk exactly q msecs after chunk was generated.
  - chunk has time stamp t: play out chunk at t+q.
  - chunk arrives after t+q: data arrives too late for playout ⇒ delay loss
- Tradeoff for q:
  - large q: less packet delay loss
  - small q: better interactive experience
  - typical maximum tolerable delay: 400 ms

Internet Phone: Packet Loss
- network loss:
  - IP packet lost due to network congestion (router buffer overflow)
- loss tolerance: depending on voice encoding, packet loss rates between 1% and 10% can be tolerated
- beyond that?
  - small tolerable playout delay prohibits retransmission
  - beyond, we need loss concealments - recover from packet loss without retransmission
Recovery from Packet Loss (1)

Forward Error Correction (FEC):
- for every group of n chunks create a redundant chunk by exclusive OR-ing the n original chunks
- send out n+1 chunks, increasing the bandwidth by factor 1/n
- can reconstruct the original n chunks if there is at most one lost chunk from the n+1 chunks
- playout delay needs to be long enough for the client receiving all n+1 packets

Impact of a larger n:
- less bandwidth waste
- longer playout delay
- higher probability that 2 or more chunks will be lost

Recovery from Packet Loss (2)

piggyback lower quality stream:
- send lower resolution audio stream as the redundant information
- the receiver can conceal non-consecutive loss
- What is the playout delay?

Recovery from Packet Loss (3)

Interleaving
- chunks are broken up into smaller units (e.g. four 5 msec units per chunk)
- Packet contains small units from different chunks
- if packet is lost, still have most of every chunk
- has no redundancy overhead, but adds to playout delay

Streaming Multimedia on the Internet: Summary

- use UDP to avoid TCP congestion control (delays) for time-sensitive traffic
- compensate delays and jitters:
  - client-side buffering
- error recovery (on top of UDP)
  - retransmissions, time permitting (for stored multimedia)
  - conceal errors: FEC, interleaving (for real-time interactive multimedia)
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Real-time Transport Protocol (RTP)

- RTP specifies a packet structure for packets carrying audio and video data
- RTP packet provides
  - payload type identification
  - packet sequence numbering
  - timestamping
- RTP runs in the end systems
- RTP packets are encapsulated in UDP segments
- Interoperability: If two Internet phone applications run RTP, then they may be able to work together

RTP Header

<table>
<thead>
<tr>
<th>Payload type</th>
<th>Sequence number</th>
<th>Timestamp</th>
<th>...</th>
</tr>
</thead>
</table>

- Payload Type (7 bits): Indicates type of encoding currently being used.
- Sequence Number (16 bits): Increments by one for each RTP packet sent, and may be used to detect packet loss and to restore packet sequence.
- Timestamp field (32 bits): Reflects the sampling instant of the first byte in the RTP data packet.

What does RTP do and not do?

- RTP specifies packet structure for multimedia data.
- RTP does not specify actions on receiving such packets.
- RTP does not provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.
- RTP encapsulation is only seen at the end systems: it is transparent to intermediate routers.
Disclaimer

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