Multimedia Networking and Quality-of-Service Support

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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 (possibly produced a long time ago)
- Transmitted to client when requested

Stored Multimedia over the Internet: The basic approach

- Web Server with Audio Files
- Web Browser
- Media Player

- Audio or video stored in files
- Files transferred as HTTP objects
  - received in entirety at client
  - then passed to player
- Media player
  - decompression depending on media format
  - user interface allows 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
    - client playout may begin before all data has arrived

Streaming from a Streaming Server

- This architecture allows for special-purpose (non-HTTP) protocol between server and media player

Streaming Multimedia Performance

- Timing constraint for still-to-be transmitted data: in time for playout
- Sensitive to delay jitter: variability of packet delays within the stream
- Solution: playout delay and client-side buffering

Streaming Multimedia: TCP or UDP?

TCP
- send at maximum possible rate under TCP
  - rate fluctuates due to TCP congestion control
  - hard to control

UDP
- server sends at rate appropriate for client (oblivious to network congestion)
- retransmission of lost packets (at the application-layer) when timing constraints permit
- network unfriendly, unfair to other TCP streams employing congestion control
- incompatible with HTTP ⇒ need non-HTTP streaming client/server
**Real-Time Interactive Multimedia**

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

**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 \( \Rightarrow \) delay loss
- Tradeoff for q:
  - large q: less packet delay loss
  - small q: better interactive experience
  - typical maximum tolerable delay: 400 msecs

**Internet Phone: Packet Loss**

- Network loss:
  - packet dropped by router (due to network congestion)
  - delay loss
  - Loss tolerance: depending on voice encoding, packet loss rates below 10% can be tolerated
- Beyond that?
  - small tolerable playout delay prohibits retransmission
  - 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 need by 1/n
- can reconstruct the original n chunks if there is at most one lost chunk from the n+1 chunks

Impact of a larger n:
- less bandwidth waste
- longer playout delay (playout delay needs to be long enough for the client receiving all n+1 packets)
- 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

Real-time Transport Protocol (RTP)

- Real-time transport protocol (RTP)
  - specifies a packet structure for carrying audio/video data
  - interoperability: If two Internet phone applications run RTP, then they can understand each other’s data

<table>
<thead>
<tr>
<th>Payload type</th>
<th>Sequence number</th>
<th>Timestamp</th>
</tr>
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<tbody>
<tr>
<td>...</td>
<td>...</td>
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- payload type: indicates type of encoding.
- sequence#: helps detect loss and restore packet sequence.
- timestamp: reflects the data sampling/playout time.
- RTP normally runs on UDP, but can run on TCP.

Multimedia Networking Summary

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

Application techniques for multimedia networking:
- Use UDP for time-sensitive traffic
- Compensate delay jitters:
  - playout delay and client-side buffering
- Error recovery (on top of UDP)
  - retransmissions, time permitting (for stored multimedia)
  - conceal errors (for real-time interactive multimedia): FEC, piggybacking, interleaving
Improving QoS inside the Internet

Thus far (practically doable now): applying application-level end-to-end techniques to enhance the “best effort” service of the Internet.

Future ideal: incorporating quality-of-service techniques into Internet routers.

Specific goals and mechanisms
Overall Internet architectures

QoS Goals and Mechanisms

- Packet classification – packet marking at network entrance
- Traffic regulation – policing
- Efficiency and fairness – router scheduling
- Admit only when there is enough resource – reservation

Let’s next look at policing and scheduling mechanisms ...

Policing Mechanism

Goal: limit traffic to not exceed specified traffic pattern (at the entrance of the network).

Characteristics of traffic pattern (in terms of bandwidth usage):
- Average rate: how many packets can be sent per unit of time
- Peak rate: the maximum or 95 percentile rate
- Burst size: maximum number of burst packets sent beyond the average rate over any time period
  - Over any interval of length t: number of packets ≤ (r_t + b)
  - For a conforming traffic, if available bandwidth \( R \geq r \), then the queue never exceeds \( b \) packets, therefore each packet’s delay ≤ \( b/R \).

How to enforce a traffic pattern with an average rate and a burst size?

Policing Mechanism: Leaky Bucket

- Bucket can hold \( b \) tokens
- Tokens generated at rate \( r \) tokens/sec unless bucket full
- A token must accompany each admitted packet
  - \( r \) tokens/sec
  - Bucket holds up to \( b \) tokens

- Over any interval of length \( t \): number of packets admitted less than or equal to \( (r \cdot t + b) \)
- Limit input to specified average rate and burst size
**Packet Scheduling at Routers**

- **FIFO (first in first out) scheduling**: send in order of arrival
- Multiple classes
  - **Round robin scheduling**: cyclically scan class queues, serving one from each class (if available)
  - **Priority scheduling**: transmit packet in the highest priority queue
  - **Weighted fair queueing**: each class gets weighted amount of service to reach a certain target

**Internet QoS Architectures**

- **Integrated services (Intserv) approach**:
  - fundamental changes in Internet for guaranteed services
  - Resource reservation
    - request with QoS declaration
    - per-router resource reservation
  - QoS-sensitive scheduling (e.g., WFQ)

**Differentiated Services**

- **Concerns with Intserv**:
  - scalability & overhead: signaling, maintaining per-flow router state difficult with large number of flows
  - deployment challenge: requires new complexity in hosts & routers

- **Differentiated services (Diffserv) approach**:
  - provide differentiated services to different classes, but may not give firm guarantee on delay and bandwidth
  - lower overhead and better scalability
  - simpler functions in network routers

**Multimedia Networking and Quality of Service Support: Summary**

- Multimedia applications and their QoS requirements
- Applying application-level end-to-end techniques to enhance the “best effort” service of the Internet
- QoS support within the Internet: policing and scheduling, reservation mechanisms
- Proposed Internet QoS architectures: Intserv, Diffserv
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