

# Multimedia Networking and Quality-of-Service Support

Kai Shen

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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.

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## Stored Multimedia

- Media stored at source (possibly produced a long time ago)
- Transmitted to client when requested

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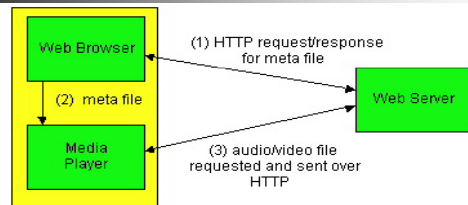
## Stored Multimedia over the Internet: the basic approach

**client**

- 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 playback!

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## Stored Multimedia over the Internet: streaming approach



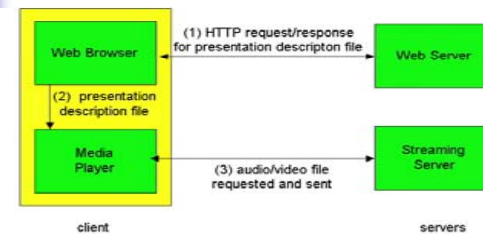
- browser GETs **metafile**
  - location of the media file, type of encoding, ...
- browser launches the media player, passing metafile
- the media player contacts server and the server **streams** audio/video to player
  - client playout may begin before all data has arrived

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## Streaming from a Streaming Server



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

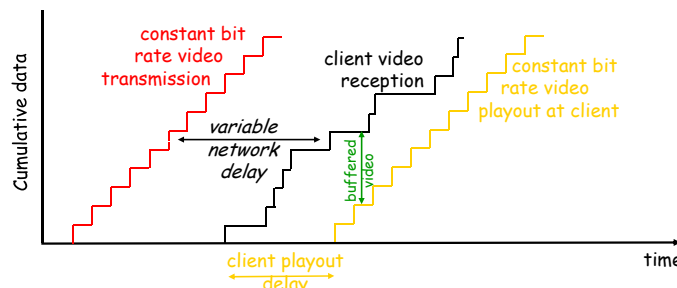
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## Streaming Multimedia Performance

- 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



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## Streaming Multimedia: TCP or UDP?

### TCP

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

### UDP

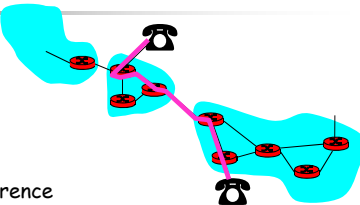
- server sends at rate appropriate for client (oblivious to network congestion!)
- retransmission of lost packets (at the application-layer) when timing constraints permit
- network unfriendly, unfair to other TCP streams employing congestion control
- incompatible with HTTP  $\Rightarrow$  need non-HTTP streaming client/server

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## 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

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## Internet Phone: Playout Delay

- Receiver attempts to playout each chunk exactly  $q$  msec 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 ms

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## 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 between 1% and 10% can be tolerated
- beyond that?
  - small tolerable playout delay prohibits retransmission
  - we need loss concealments - recover from packet loss without retransmission

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## 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

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### Recovery from Packet Loss (2)

Original Stream

Redundancy

Packet Loss

Reconstructed Stream

**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?

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### Recovery from Packet Loss (3)

Original Stream

Interleaved Stream

Packet Loss

Reconstructed Stream

**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

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### 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

...	Payload type	Sequence number	Timestamp	...
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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.
- What does RTP run on?

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### Multimedia Networking

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

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## Improving QoS inside the Internet

**Thus far:** applying application-level end-to-end techniques to enhance the "best effort" service of the Internet.

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

[Goals and Mechanisms](#)  
[Proposed architectures](#)

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## QoS Goals and Mechanisms

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

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

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## 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  $\leq (r \cdot t + b)$

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

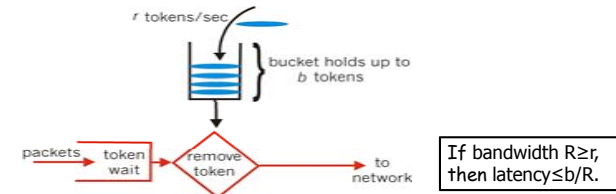
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## Policing Mechanism: Leaky Bucket

- bucket can hold  $b$  tokens
- tokens generated at rate  $r$  tokens/sec unless bucket full
- a token must accompany each admitted packet



- 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

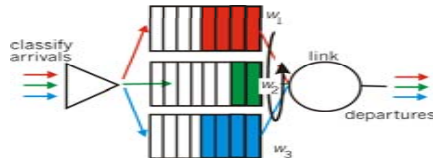
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## Packet Scheduling at Routers

- **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



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## How should the Internet evolve to better support QoS for multimedia apps?

### No major changes:

- apply only application-level end-to-end techniques

### Integrated services (Intserv) philosophy:

- fundamental changes in Internet so that apps can reserve end-to-end bandwidth
- requires new, complex software in hosts & routers

### Differentiated services philosophy:

- fewer changes to Internet infrastructure, yet provide differentiated services to different classes
- may not give firm guarantee on delay and bandwidth

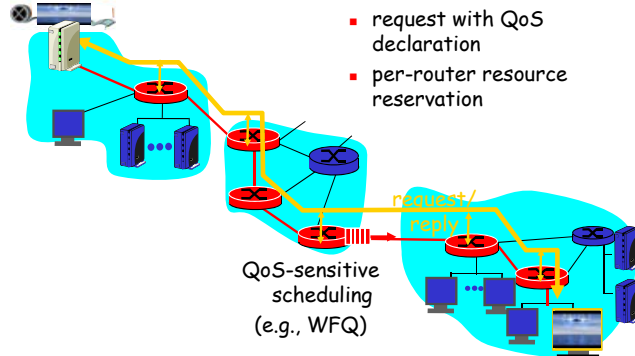
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## Intserv: An Illustration

- **Resource reservation**
  - request with QoS declaration
  - per-router resource reservation



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## Differentiated Services

### Concerns with Intserv:

- **Scalability & overhead:** signaling, maintaining per-flow router state difficult with large number of flows

### Diffserv approach:

- simple functions in network core, relatively complex functions at edge routers (or hosts)
- provide differentiated services to different classes, may not give firm guarantee on delay and bandwidth

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## 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
- Next generation Internet QoS architecture: Intserv, Diffserv

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