



Multimedia Networking

Acknowledgements

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Goals

Principles

- classify multimedia applications
- identify application requirements
- making the best of best effort service

Protocols and Architectures

- specific protocols for best-effort
- mechanisms for providing QoS
- architectures for QoS

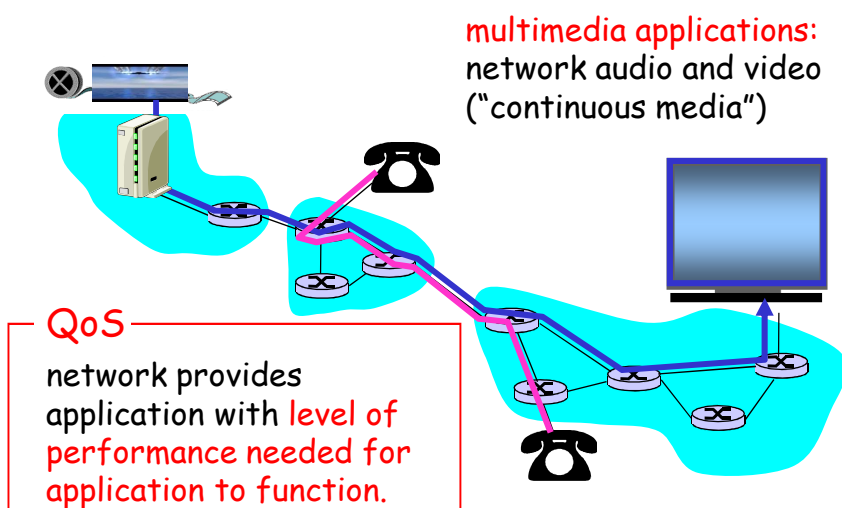
Roadmap



- ❑ **Multimedia networking applications**
- ❑ Streaming stored audio and video
- ❑ Protocols and architectures for audio/video streaming
 - Streaming Protocols
 - Content Distribution Networks (CDNs)
- ❑ Real-time interactive applications
- ❑ Protocols for real-time interactive applications
 - RTP
 - RTCP
 - SIP
- ❑ Providing multiple classes of service
- ❑ Providing QoS guarantees

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Multimedia and Quality of Service (QoS)



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Audio digitalization/compression



- ❑ Analog signal sampled at constant rate
 - telephone: 8,000 samples/sec
 - CD music: 44,100 samples/sec
- ❑ Each sample quantized, i.e., rounded
 - e.g., $2^8=256$ possible quantized values
- ❑ Each quantized value represented by bits
 - 8 bits for 256 values
 - example: 8,000 samples/sec, 256 quantized values --> 64 Kbps
- ❑ Receiver converts bits back to analog signal:
 - some quality reduction

Audio digitalization/compression



Example rates

- ❑ Internet telephony: 5.3 kbps and up
 - GSM: 13 Kbps
 - PCM: 64 Kbps
- ❑ CD: 705.6 Mbps (mono) 1.411 Mbps (stereo)
- ❑ MPEG-1 Layer 3 (MP3): 96, 128, 160 kbps

Video digitalization/compression



- Video: sequence of images displayed at constant rate
 - e.g. 24 images/sec
- Digital image: array of pixels
 - each pixel represented by bits (luminance + colour)
- Redundancy
 - spatial (within image)
 - temporal (from one image to next)

Video digitalization/compression



Examples:

- MPEG 1 (CD-ROM) 1.5 Mbps
- MPEG2 (DVD) 3-6 Mbps
- MPEG4 (often used in Internet, < 1 Mbps)

Research:

- layered (scalable) video
 - adapt layers to available bandwidth

MM Networking Applications



Classification

- ❑ Stored streaming applications
 - YouTube
 - Video on demand
- ❑ Live streaming applications
 - Internet radio
 - IPTV (e.g., live sporting event)
- ❑ Interactive, real-time applications
 - Internet Telephony
 - Audio/video conferencing

MM Networking Applications



Fundamental characteristics

- ❑ typically **delay sensitive**
 - end-to-end delay
 - delay jitter
- ❑ **loss tolerant**: infrequent losses cause minor glitches
- ❑ antithesis of data, which are loss *intolerant* but delay *tolerant*.

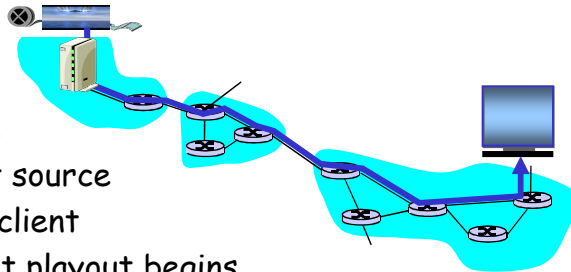
Jitter is the variability of packet delays within the same packet stream

Streaming Stored Multimedia



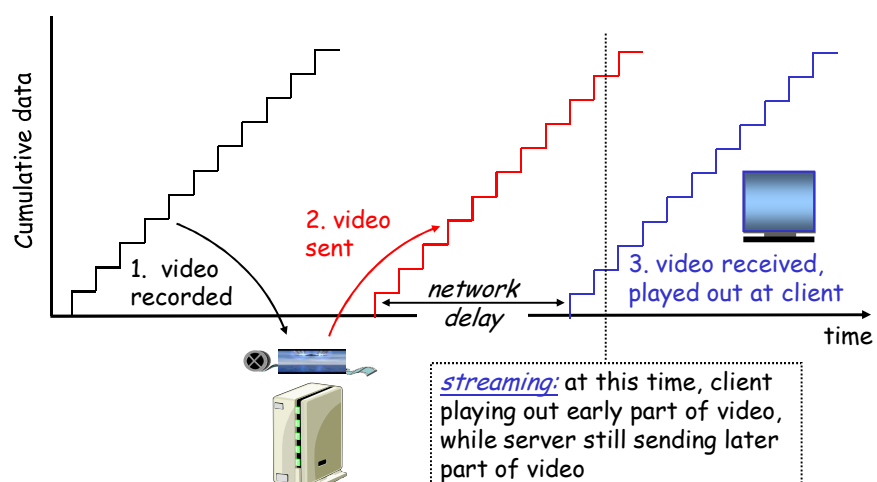
Stored streaming:

- media stored at source
- transmitted to client
- **streaming**: client playout begins *before* all data has arrived
- timing constraint for still-to-be transmitted data: in time for playout



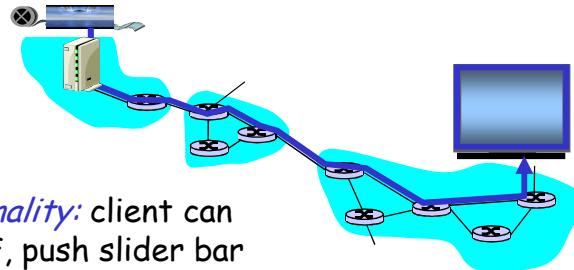
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Streaming Stored Multimedia



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Streaming **Stored** Multimedia: Interactivity



- *VCR-like functionality*: client can pause, rewind, FF, push slider bar
 - 10 sec initial delay OK
 - 1-2 sec until command effect OK
- timing constraint for still-to-be transmitted data: in time for playout

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Streaming **Live** Multimedia



Streaming (as with streaming *stored* multimedia)

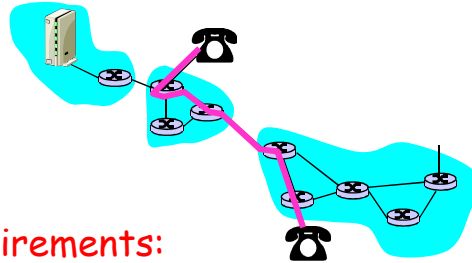
- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity

- fast forward **impossible**
- rewind, pause possible!

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Real-Time Interactive Multimedia



□ end-end delay requirements:

- audio: < 150 msec good, < 400 msec OK
 - includes application-level (packetization) and network delays
 - higher delays noticeable, impair interactivity

□ session initialization

- how does callee advertise its IP address, port number, encoding algorithms?

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Multimedia Over Today's Internet



TCP/UDP over IP: "best-effort service"

- No guarantees on delay, jitter, loss



But we just said multimedia apps require QoS and level of performance to be effective!



Today's Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss

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How should the Internet evolve to better support multimedia?



Laissez-faire

- ❑ No major changes
- ❑ More bandwidth when needed (**overprovisioning**)
- ❑ Application layer solutions
 - content distribution networks
 - application-layer multicast

Differentiated services

- ❑ fewer changes to Internet infrastructure, yet provide 1st and 2nd class service



Integrated services

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

What's your opinion?

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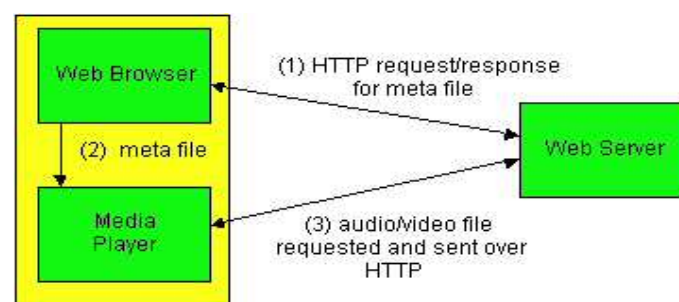
Streaming Stored Multimedia



- Application-level streaming techniques
 - client-side buffering
 - use of UDP versus TCP
 - UDP streaming
 - HTTP, Adaptive HTTP streaming (use TCP)
 - multiple encodings of multimedia
- Media Player
 - Jitter removal
 - Decompression
 - Error concealment
 - Graphical user interface with controls for interactivity

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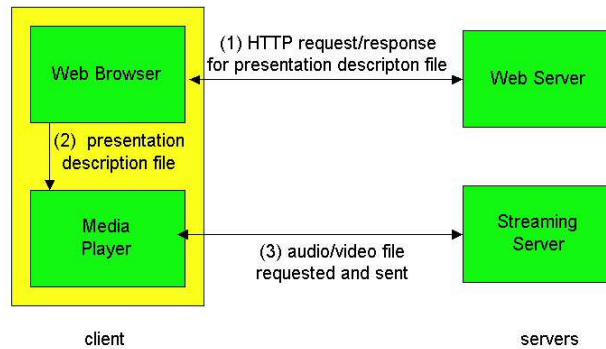
Streaming Protocols (1)



- browser GETs **metafile**
- browser launches player, passing metafile
- player contacts server
- server **streams** audio/video to player

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Streaming Protocols (2)



- Allows for non-HTTP protocol between streaming server and media player
- UDP typically used, with RTP and RTSP, for step (3)

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Streaming Approaches



- UDP streaming
- HTTP streaming
- Dynamic Adaptive Streaming over HTTP (DASH)

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UDP Streaming



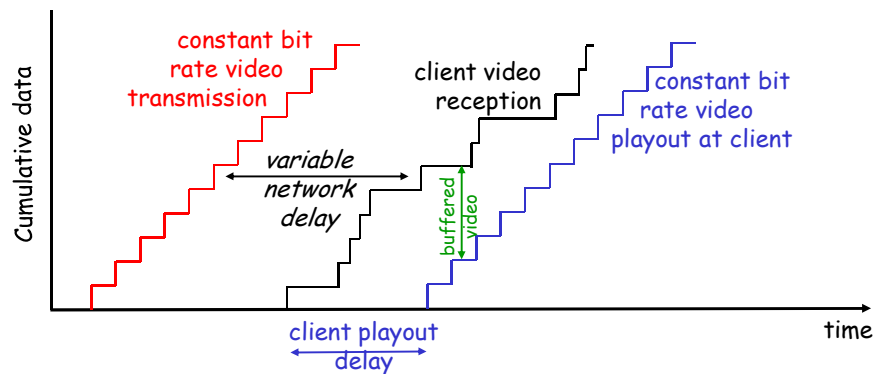
- ❑ Uses UDP as transport protocol
 - With RTP as encapsulation protocol, and RTSP as control protocol
 - This requires two additional protocols, thus increasing installation costs
- ❑ UDP can fail to provide continuous playout
 - Due to variation in the network operating conditions
- ❑ Many gateways are configured to block UDP traffic

HTTP Streaming



- ❑ Uses TCP as underlying transport protocol
 - Despite of congestion, reliability and flow control
- ❑ No problems in traversing gateways
- ❑ No needs for additional control protocols (RTP , RTSP)
 - This reduces installation costs
 - HTTP byte-range header in GET messages used for repositioning
- ❑ Allows pre-fetching of data
- ❑ Youtube and Netflix use HTTP (over TCP) as their streaming protocol
- ❑ All users receive the same encoding of video

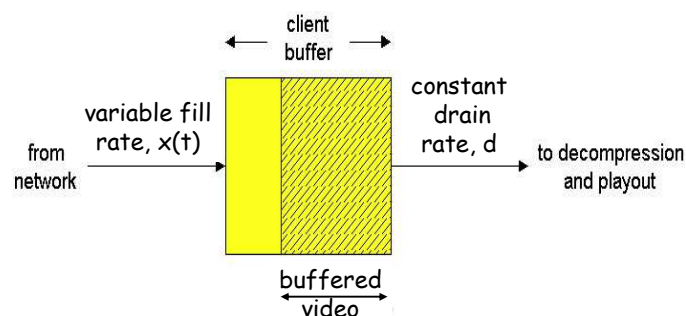
Streaming: Client Buffering



- client-side buffering, playout delay compensate for network-added delay, delay jitter

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Streaming: Client Buffering



- client-side buffering, playout delay compensate for network-added delay, delay jitter

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Adaptive Streaming



Dynamic Adaptive Streaming over HTTP (DASH)

- ❑ Different encoded version of streaming data
 - Different bit rate and quality
 - Clients select the encoded version on the basis of the available bandwidth
 - Important for mobile users
- ❑ Different versions have different URL
 - provided in the manifest (presentation description) file
- ❑ Rate determination algorithm for measuring the available bandwidth
 - Smooth decrease of the bit rate to avoid noticeable changes in the streamed sequence
 - Better utilization of the de-jittering buffer

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Roadmap



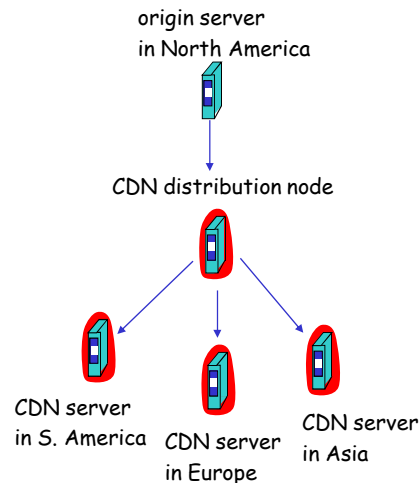
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Content Distribution Networks (CDNs)

Content replication

- ❑ challenging to stream large files (e.g., video) from single origin server in real time
- ❑ *solution: replicate* content at hundreds of servers throughout Internet
 - content downloaded to CDN servers ahead of time
 - placing content "close" to user avoids impairments (loss, delay) of sending content over long paths
 - CDN server typically in edge/access network

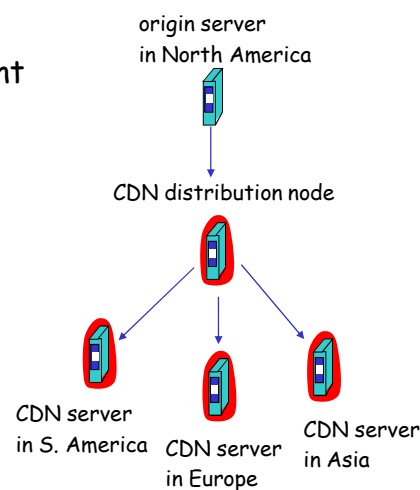


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Content Distribution Networks (CDNs)

Content replication

- ❑ CDN customer is the content provider (e.g., CNN)
- ❑ CDN replicates customers' content in CDN servers.
- ❑ when provider updates content, CDN updates servers
- ❑ CDN may either private or third-party company
 - Google's CDN for YouTube
 - Akamai for Netflix, Hulu, ...



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CDNs: Server Placement



Enter Deep

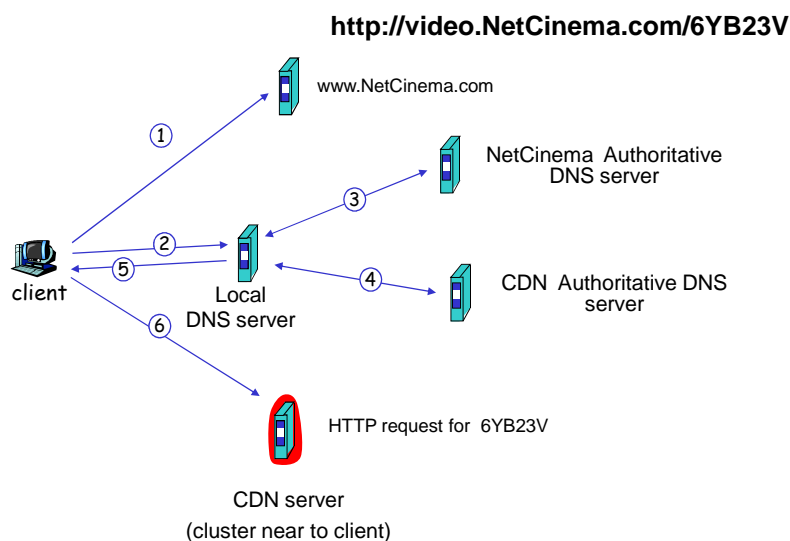
- Server clusters are located in access ISP all over the world
 - typically a dozen of servers per cluster
 - Akamai has cluster in 1700 locations
- Increased performance
- Cluster management is a big challenge

Bring Home

- Large clusters in a small number of locations
- Easier management of clusters
- To improve performance site locations are chosen strategically

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CDN Operations



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Cluster Selection Strategy



□ Geographically Closest

- The cluster closest to the LDNS is assigned (map of LDNS locations)
- May not be the closest on the Internet path (some clients use a remote LDNS)
- Does not consider traffic variations

□ Real-time measurements

- Clusters send periodic probe messages to LDNSs
- Measures the delay between SYNACK and ACK
- ...

□ IP anycast

- Same IP address used for all clusters
- BGP selects the best path (in terms of # of Ass)

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Case Studies



□ NetFlix

- Amazon cloud for content storing
- Third-party CDNs for distribution
- DASH for client-server interaction

□ YouTube

- Private Google CDN
- HTTP streaming

□ KanKan (PPTV, ...)

- P2P video streaming
- Similar to BitTorrent
 - DHT for content tracking
 - priority is give to chunks to play in the near future
- Uses UDP

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Real-time interactive applications



- ❑ PC-2-PC phone
 - Skype
- ❑ PC-2-phone
 - Skype
 - Dialpad
 - Net2phone
- ❑ Videoconference with webcams
 - Skype
 - Polycom

We will look at a PC-2-PC Internet phone example in detail

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Internet Phone



- Speaker's audio: alternating talk spurts, silent periods.
 - 64 kbps during talk spurt
 - pkts 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 talkspurt

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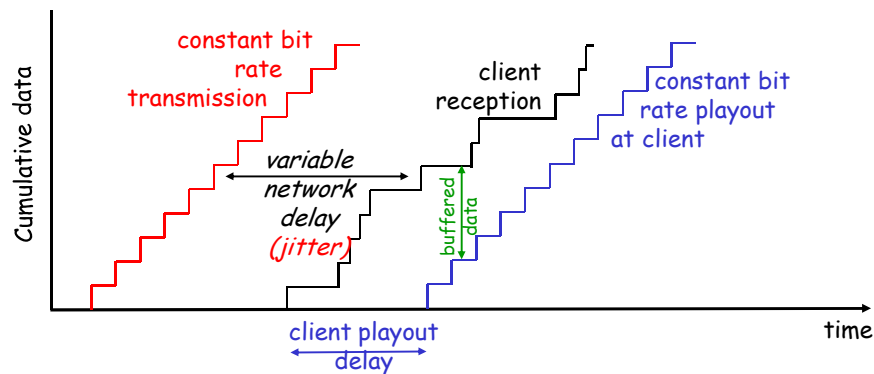
Packet Loss and Delay



- **Network loss:** IP datagram lost due to network congestion
 - buffer overflow at some intermediate router
- **Delay loss:** IP datagram arrives too late for playout at receiver
 - delays: processing, queueing in network; end-system (sender, receiver) delays
 - typical maximum tolerable delay: 400 ms
- **loss tolerance:** depending on voice encoding, loss concealment technique, packet loss rates between 1% and 10% can be tolerated.

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Delay Jitter



- consider end-to-end delays of two consecutive packets: difference can be more or less than 20 msec (transmission time difference)

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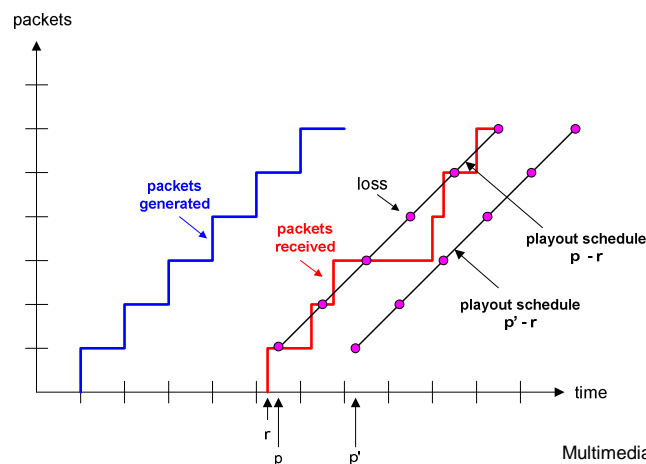
Fixed Playout Delay

- 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, **data "lost"**
- Tradeoff in choosing q :
 - *large q* : less packet loss
 - *small q* : better interactive experience

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Fixed Playout Delay

- sender generates packets every 20 msec during talk spurt.
- first packet received at time r
- first playout schedule: begins at p
- second playout schedule: begins at p'



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Adaptive Playout Delay (1)

- **Goal:** minimize playout delay, keeping late loss rate low
- **Approach:** adaptive playout delay adjustment:
 - estimate network delay, adjust playout delay at beginning of each talk spurt.
 - silent periods compressed and elongated.
 - chunks still played out every 20 msec during talk spurt.

t_i = timestamp of the i th packet

r_i = the time packet i is received by receiver

p_i = the time packet i is played at receiver

$r_i - t_i$ = network delay for i th packet

d_i = estimate of average network delay after receiving i th packet

dynamic estimate of average delay at receiver:

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

where u is a fixed constant (e.g., $u = .01$).

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Adaptive Playout Delay (2)



- also useful to estimate average deviation of delay, v_i :

$$v_i = (1-u)v_{i-1} + u |(r_i - t_i) - d_i|$$

- estimates d_i , v_i calculated for every received packet (but used only at start of talk spurt)

- for first packet in talk spurt, playout time is:

$$p_i = t_i + d_i + Kv_i$$

where K is positive constant (e.g., 4)

- remaining packets in talkspurt are played out periodically

Adaptive Playout Delay (3)



How does receiver determine whether packet is first in a talkspurt?

- if no loss, receiver looks at successive timestamps.
 - difference of successive stamps > 20 msec --> talk spurt begins.
- with loss possible, receiver must look at both time stamps and sequence numbers.
 - difference of successive stamps > 20 msec **and** sequence numbers without gaps --> talk spurt begins.

Recovery from Packet Loss (1)



Forward Error Correction (FEC): simple scheme

- ❑ For every group of n chunks create redundant chunk by exclusive OR-ing n original chunks
- ❑ Send out $n+1$ chunks
 - bandwidth increases by factor $1/n$.
- ❑ Can reconstruct original n chunks if at most one lost chunk from $n+1$ chunks
- ❑ Playout delay: enough time to receive all $n+1$ packets
- ❑ Tradeoff:
 - increase n , less bandwidth waste
 - increase n , longer playout delay
 - increase n , higher probability that 2 or more chunks will be lost

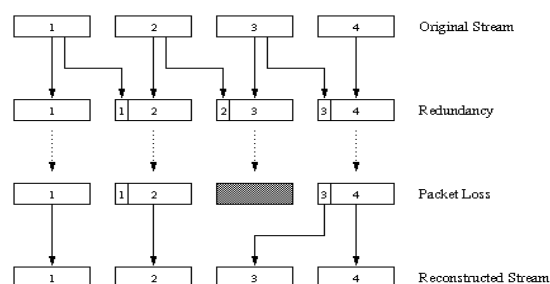
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Recovery from Packet Loss (2)



2nd FEC scheme

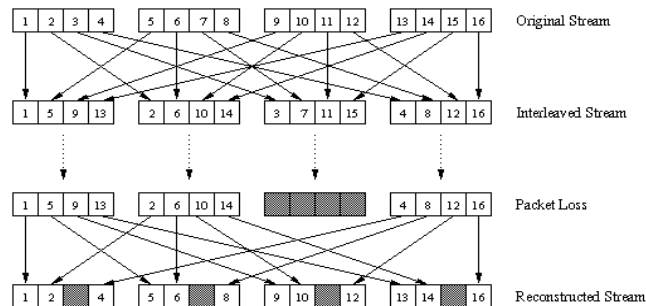
- ❑ "piggyback lower quality stream"
- ❑ send lower resolution audio stream as redundant information
- ❑ e.g., nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps.



- ❑ whenever there is non-consecutive loss, receiver can conceal the loss.
- ❑ can also append $(n-1)$ st and $(n-2)$ nd low-bit rate chunk

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Recovery from Packet Loss (3)



Interleaving

- chunks divided into smaller units
- for example, four 5 msec units per chunk
- packet contains small units from different chunks
- if packet lost, still have most of every chunk
- no redundancy overhead, but increases playout delay
 - Mainly used in streaming of stored audio

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Real-time Multimedia: Summary

- use **UDP** to avoid TCP congestion control (delays) for time-sensitive traffic
- client-side **adaptive playout delay**: to compensate for delay
- server side **matches stream bandwidth** to available client-to-server path bandwidth
 - dynamic server encoding rate
- error recovery (on top of UDP)
 - FEC, interleaving, error concealment
 - retransmissions, time permitting

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Roadmap

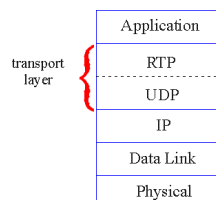


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Real-Time Protocol (RTP) [RFC 3550]



- ❑ RTP specifies packet structure for packets carrying audio, video data
- ❑ RTP packet provides
 - payload type identification
 - packet sequence numbering
 - time stamping
- ❑ RTP runs in end systems
- ❑ RTP packets encapsulated in UDP segments



- ❑ interoperability:
 - if two Internet phone applications run RTP, then they may be able to work together

RTP Example



- ❑ Consider sending 64 kbps PCM-encoded voice over RTP.
- ❑ Application collects encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk.
- ❑ Audio chunk + RTP header form RTP packet, which is encapsulated in UDP segment
- ❑ RTP header indicates type of audio encoding in each packet
 - sender can change encoding during conference.
- ❑ RTP header also contains sequence numbers, timestamps.

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RTP and QoS



- ❑ RTP does **not** provide any mechanism to ensure timely data delivery or other QoS guarantees.
- ❑ RTP encapsulation is only seen at end systems (not) by intermediate routers.
 - routers provide best-effort service, making no special effort to ensure that RTP packets arrive at destination in timely matter.

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RTP Header



RTP Header

Payload Type (7 bits): Indicates type of encoding currently being used. If sender changes encoding in middle of conference, sender informs receiver via payload type field.

- Payload type 0: PCM mu-law, 64 kbps
- Payload type 3: GSM, 13 kbps
- Payload type 7: LPC, 2.4 kbps
- Payload type 26: Motion JPEG
- Payload type 31: H.261
- Payload type 33: MPEG2 video

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.

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RTP Header (2)



- **Timestamp field (32 bytes long):** sampling instant of first byte in this RTP data packet
 - for audio, timestamp clock typically increments by one for each sampling period (for example, each 125 usecs for 8 KHz sampling clock)
 - if application generates chunks of 160 encoded samples, then timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.
- **SSRC field (32 bits long):** identifies source of + RTP stream.
 - Random number.
 - Each stream in RTP session should have distinct SSRC.

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Developing Applications using RTP

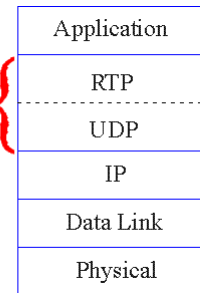


RTP libraries provide transport-layer interface that extends UDP:

Information to be provided

- port numbers, IP addresses
- payload type identification
- packet sequence numbering
- time-stamping
- SSRC

transport layer



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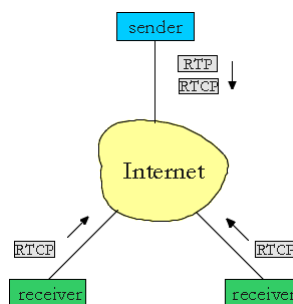
Real-Time Control Protocol (RTCP) [RFC 3550]



- ❑ Works in conjunction with RTP.
- ❑ Each participant in RTP session periodically transmits RTCP control packets to all other participants.
- ❑ Each RTCP packet contains sender and/or receiver reports
 - report statistics useful to application:
 - # packets sent
 - # packets lost
 - inter-arrival jitter, etc.
- ❑ Feedback can be used to control performance
 - sender may modify its transmissions based on feedback

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RTCP: Example



- ❑ RTP session: typically a single multicast address
 - ❑ Both sender and receivers send RTCP packets
- ❑ RTP, RTCP packets distinguished from each other via distinct port numbers
 - RTCP port number = RTP port number +1
- ❑ To limit traffic, each participant reduces RTCP traffic as number of conference participants increases

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RTCP Packets



Receiver report packets:

- SSRC of RTP stream, fraction of packets lost, last sequence number, average inter-arrival jitter

Sender report packets:

- SSRC of RTP stream, current time, number of packets sent, number of bytes sent

Source description packets:

- e-mail address of sender, sender's name, SSRC of associated RTP stream
- provide a mapping between the SSRC and the user/host name

Synchronization of Streams



- RTCP can synchronize different media streams within a RTP session
- Consider videoconferencing app for which each sender generates one RTP stream for video, one for audio.
- Timestamps in RTP packets tied to the video, audio sampling clocks
 - **not** tied to wall-clock time
- Each RTCP sender-report packet contains (for most recently generated packet in associated RTP stream):
 - timestamp of RTP packet
 - wall-clock time for when packet was created.
- receivers uses association to synchronize playout of audio, video



RTCP Bandwidth Scaling

- ❑ RTCP attempts to limit its traffic to 5% of session bandwidth.
 - Example: one sender, sending video at 2 Mbps. Then RTCP attempts to limit its traffic to 100 Kbps.
- ❑ RTCP gives 75% of rate to receivers; remaining 25% to sender
- ❑ 75 kbps is equally shared among receivers:
 - with R receivers, each receiver gets to send RTCP traffic at $75/R$ kbps.
 - Each receiver send reports to any other → R is known
- ❑ Sender gets to send RTCP traffic at 25 kbps.
- ❑ Participant determines RTCP packet transmission period
 - by calculating avg RTCP packet size (across entire session) and dividing by allocated rate

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SIP: Session Initiation Protocol [RFC 3261, 5411]



SIP long-term vision:

- ❑ All telephone calls, video conference calls take place over Internet
- ❑ People are identified by names or e-mail addresses, rather than by phone numbers
 - E.g., Skype name
- ❑ You can reach callee, no matter
 - where callee roams,
 - what IP device callee is currently using

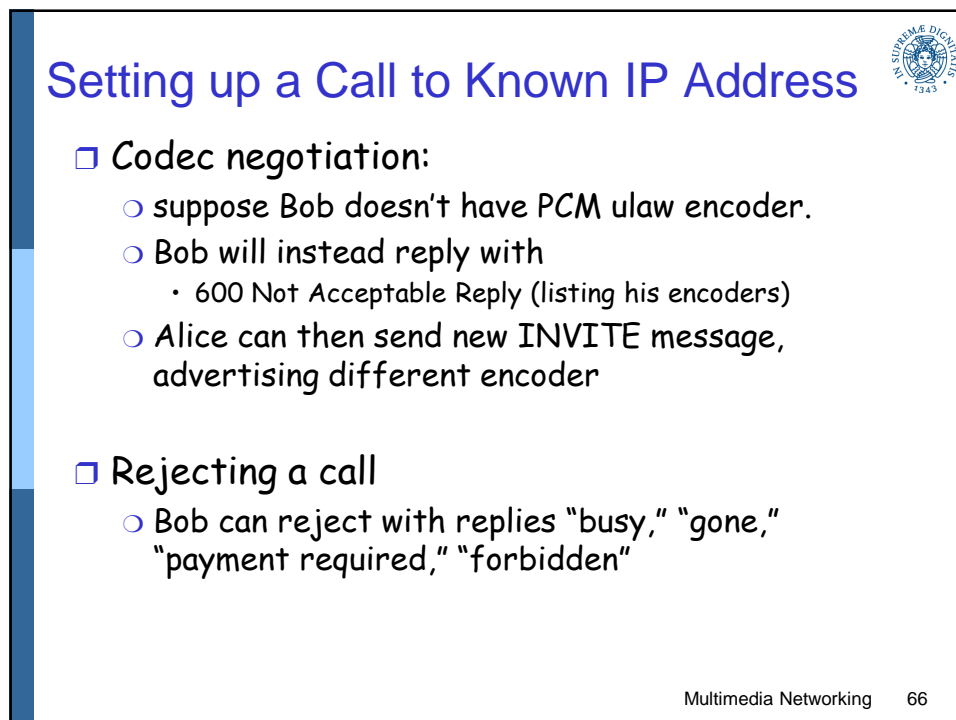
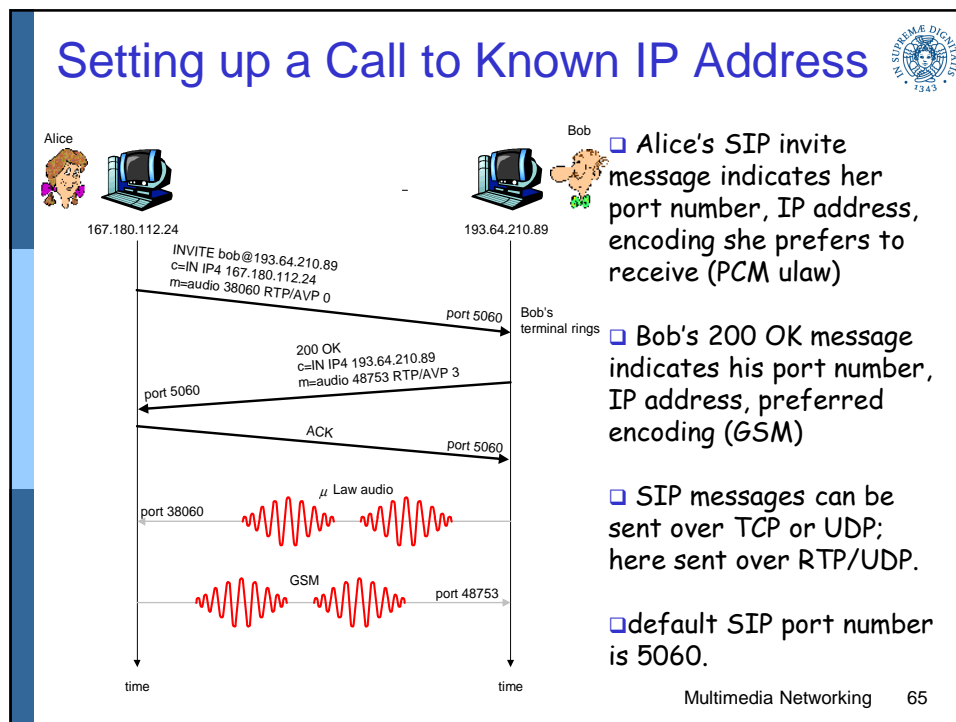
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SIP Services



- ❑ Setting up a call, SIP provides mechanisms ..
 - for caller to let callee know he/she wants to establish a call
 - so caller, callee can agree on media type, encoding
 - to end call
- ❑ Determine current IP address of callee:
 - maps mnemonic identifier to current IP address
- ❑ Call management:
 - add new media streams during call
 - change encoding during call
 - invite other participants
 - transfer, hold calls

Multimedia Networking 64



Example of SIP Message



```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885

c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

Notes:

- HTTP message syntax
- sdp = session description protocol
- Call-ID is unique for every call.

□ Here we don't know Bob's IP address. Intermediate SIP servers needed.

□ Alice sends, receives SIP messages using SIP default port 5060

□ Alice specifies in Via: header that SIP client sends, receives SIP messages over UDP

Multimedia Networking 67

Name Translation and User Location



- Caller wants to call callee
 - but only has callee's name or e-mail address.
- Needs to get IP address of callee's current host:
 - User moves around
 - DHCP protocol
 - User has different IP devices (PC, PDA, car device)
- Result can be based on:
 - time of day (work, home)
 - caller (don't want boss to call you at home)
 - status of callee (calls sent to voicemail when callee is already talking to someone)

Multimedia Networking 68

Service Provided by SIP Servers



- ❑ SIP registrar server
- ❑ SIP proxy server

SIP Registrar



- ❑ When Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server (similar function needed by Instant Messaging)

Register Message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

SIP Proxy



- ❑ Alice sends invite message to her proxy server
 - contains address sip:bob@domain.com
- ❑ Proxy responsible for routing SIP messages to callee
 - possibly through multiple proxies.
- ❑ Callee sends response back through the same set of proxies.
- ❑ Proxy returns SIP response message to Alice
 - contains Bob's IP address
- ❑ Proxy analogous to local DNS server

Multimedia Networking 71

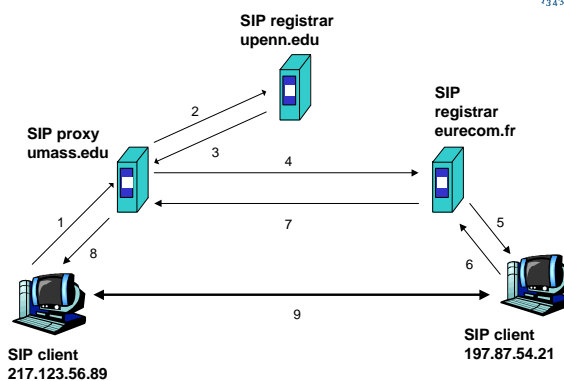
Example

Caller jim@umass.edu
with places a call to
keith@upenn.edu

- (1) Jim sends INVITE message to umass SIP proxy.
- (2) Proxy forwards request to upenn registrar server.
- (3) upenn server returns redirect response, indicating that it should try keith@eurecom.fr

- (4) umass proxy sends INVITE to eurecom registrar. (5) eurecom registrar forwards INVITE to 197.87.54.21, which is running keith's SIP client. (6-8) SIP response sent back (9) media sent directly between clients.

Note: also a SIP ack message, which is not shown.



Multimedia Networking 72

Roadmap



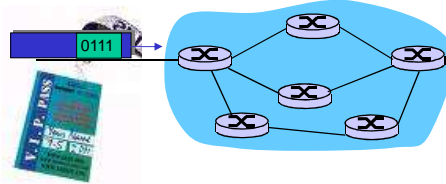
- ❑ Multimedia networking applications
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- ❑ Protocols for real-time interactive applications
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 - RTCP
 - SIP
- ❑ Providing multiple classes of service
- ❑ Providing QoS guarantees

Multimedia Networking 73

Providing Multiple Classes of Service

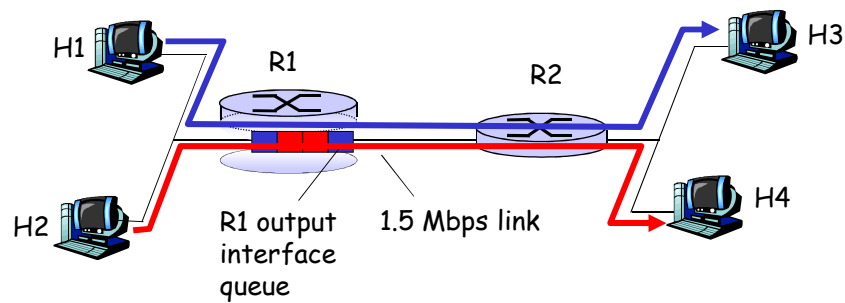


- ❑ Thus far: making the best of best effort service
 - One-size fits all service model
- ❑ Alternative: multiple classes of service
 - Partition traffic into classes
 - Manage different classes of traffic differently
- ❑ Granularity:
 - Differential service among multiple classes, not among individual connections
- ❑ History: ToS bits



Multimedia Networking 74

Multiple Classes of Service: Scenario

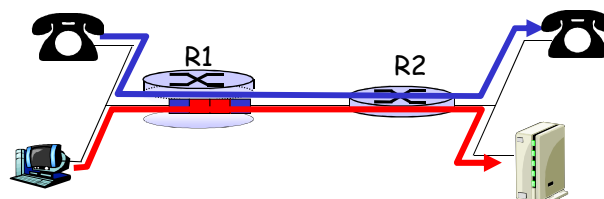


Multimedia Networking 75

Principles for QOS Guarantees



- Example: 1Mbps IP phone, FTP share 1.5 Mbps link.
 - bursts of FTP can congest router, cause audio loss
 - want to give priority to audio over FTP



Principle 1

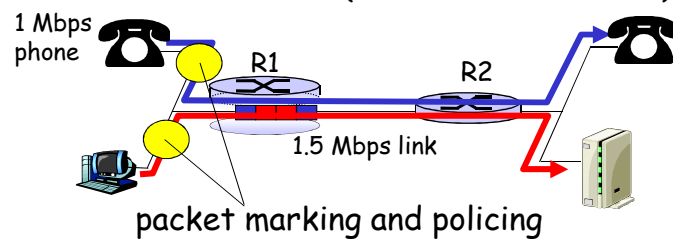
Packet marking needed for router to distinguish between different classes;
and new router policy to treat packets accordingly

Multimedia Networking 76

Principles for QOS Guarantees



- ❑ what if applications misbehave (audio sends higher than declared rate)
 - policing: force source adherence to bandwidth allocations
- ❑ marking and policing at network edge:
 - similar to ATM UNI (User Network Interface)



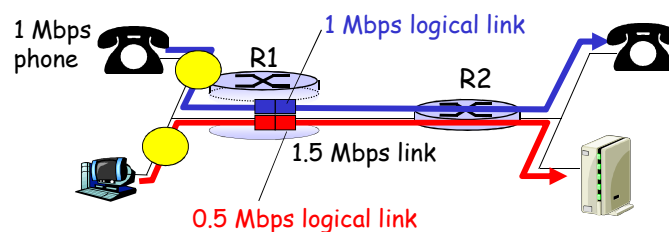
Principle 2 —
Provide protection (*isolation*) for one class from others

Multimedia Networking 77

Principles for QOS Guarantees



- ❑ Allocating *fixed* (non-sharable) bandwidth to flow: *inefficient* use of bandwidth if flows doesn't use its allocation



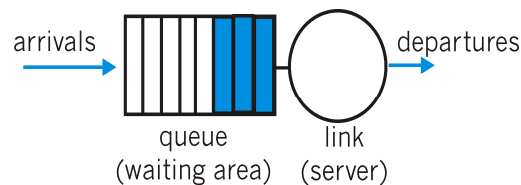
Principle 3 —
While providing isolation, it is desirable to use resources as efficiently as possible

Multimedia Networking 78

Scheduling And Policing Mechanisms



- **Scheduling:** choose next packet to send on link
- **FIFO (first in first out) scheduling:**
 - send in order of arrival to queue
 - **discard policy:** if packet arrives to full queue: who to discard?
 - Tail drop: drop arriving packet
 - priority: drop/remove on priority basis
 - random: drop/remove randomly



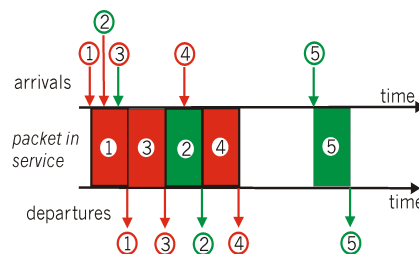
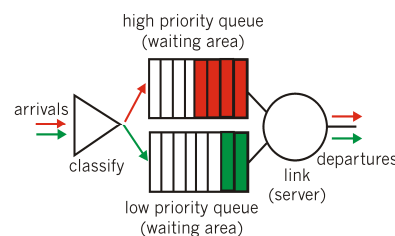
Multimedia Networking 79

Priority-based Scheduling



Transmit highest priority queued packet

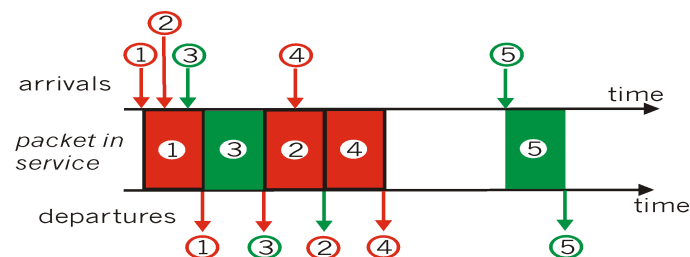
- Multiple *classes*, with different priorities
 - class may depend on marking or other header info, e.g. IP source/dest, port numbers, etc..
 - Real world example?



Multimedia Networking 80

Round Robin Scheduling

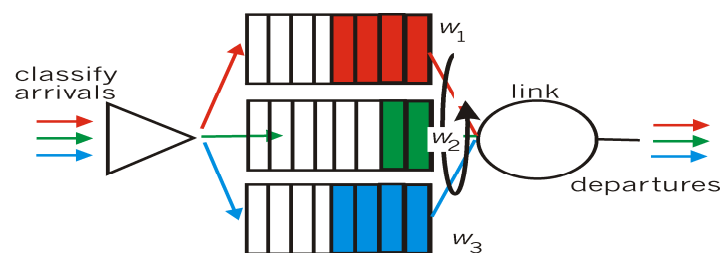
- Multiple classes
- Cyclically scan class queues, serving one from each class (if available)
- real world example?



Multimedia Networking 81

Weighted Fair Queuing

- Generalized Round Robin
- Each class gets weighted amount of service in each cycle
- real-world example?



Multimedia Networking 82

Policing Mechanisms



Goal: limit traffic to not exceed declared parameters

Three common-used criteria:

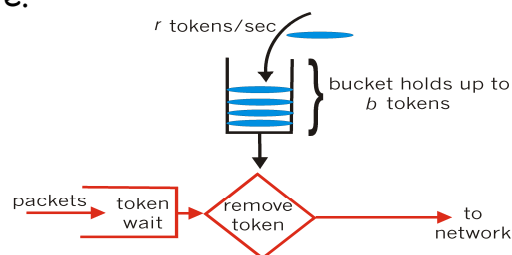
- ❑ **(Long term) Average Rate:** how many pkts can be sent per unit time (in the long run)
 - crucial question: what is the interval length: 100 packets per sec or 6000 packets per min have same average!
- ❑ **Peak Rate:** e.g., 6000 pkts per min. (ppm) avg.; 1500 ppm peak rate
- ❑ **(Max.) Burst Size:** max. number of pkts sent consecutively (with no intervening idle)

Multimedia Networking 83

Policing Mechanisms



Token Bucket: limit input to specified Burst Size and Average Rate.

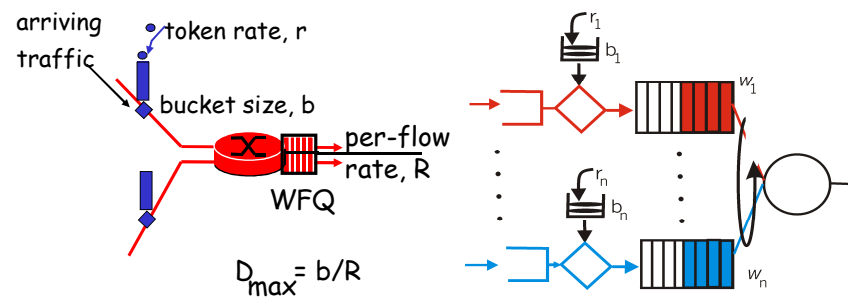


- ❑ bucket can hold b tokens
- ❑ tokens generated at rate r token/sec unless bucket full
- ❑ over interval of length t : number of packets admitted less than or equal to $(r \cdot t + b)$.

Multimedia Networking 84

Policing Mechanisms (more)

- Token bucket, WFQ combine to provide guaranteed upper bound on delay, i.e., **QoS guarantee!**



Multimedia Networking 85

IETF Differentiated Services (DiffServ)

- Want "qualitative" service classes
 - "behaves like a wire"
 - relative service distinction: Platinum, Gold, Silver
- Scalability:**
 - simple functions in network core, relatively complex functions at edge routers (or hosts)
 - signaling, maintaining per-flow router state difficult with large number of flows
- Don't define service classes
 - Just provide functional components to build service classes

Multimedia Networking 86

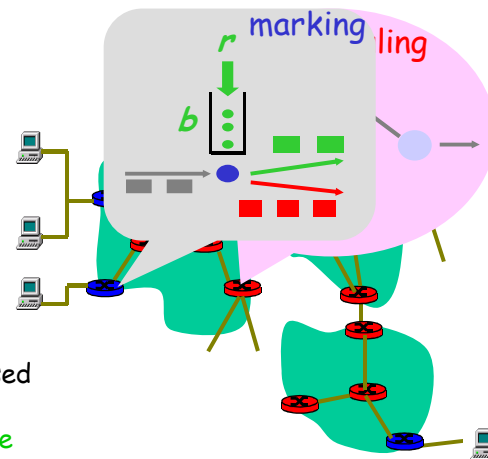
Diffserv Architecture

Edge router: 

- per-flow traffic management
- marks packets as **in-profile** and **out-profile**

Core router: 

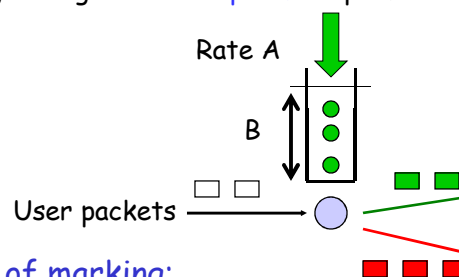
- per class traffic management
- buffering and scheduling based on **marking** at edge
- preference given to **in-profile** packets



Multimedia Networking 87

Edge-router Packet Marking

- **profile**: pre-negotiated rate A , bucket size B
- packet marking at edge based on **per-flow** profile



Possible usage of marking:

- class-based marking: packets of different classes marked differently
- intra-class marking: conforming portion of flow marked differently than non-conforming one

Multimedia Networking 88

Classification and Conditioning



- Packet is marked in the Type of Service (TOS) in IPv4, and Traffic Class in IPv6
 - 6 bits used for Differentiated Service Code Point (DSCP) and determine PHB that the packet will receive
 - 2 bits are currently unused



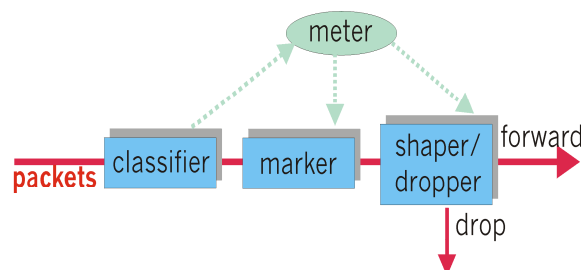
Multimedia Networking 89

Classification and Conditioning



May be desirable to limit traffic injection rate of some class:

- user declares traffic profile (e.g., rate, burst size)
- traffic metered, shaped if non-conforming



Multimedia Networking 90

Per Hop Behaviour (PHB)



- PHB result in a different observable (measurable) forwarding performance behavior
- PHB does not specify what mechanisms to use to ensure required PHB performance behavior
- Examples:
 - Class A gets x% of outgoing link bandwidth over time intervals of a specified length
 - Class A packets leave first before packets from class B

Forwarding (PHB)



PHBs being developed:

- **Expedited Forwarding:** pkt departure rate of a class equals or exceeds specified rate
 - logical link with a minimum guaranteed rate
- **Assured Forwarding:** 4 classes of traffic
 - each guaranteed minimum amount of bandwidth
 - each with three drop preference partitions

Roadmap



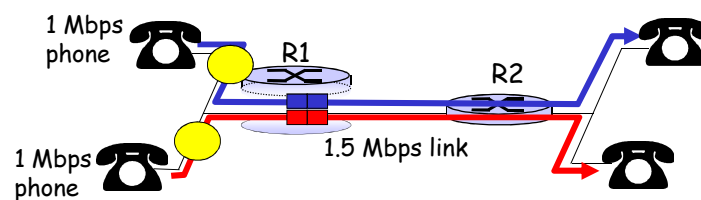
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Multimedia Networking 93

Principles for QOS Guarantees



- ❑ Basic fact of life: can not support traffic demands beyond link capacity

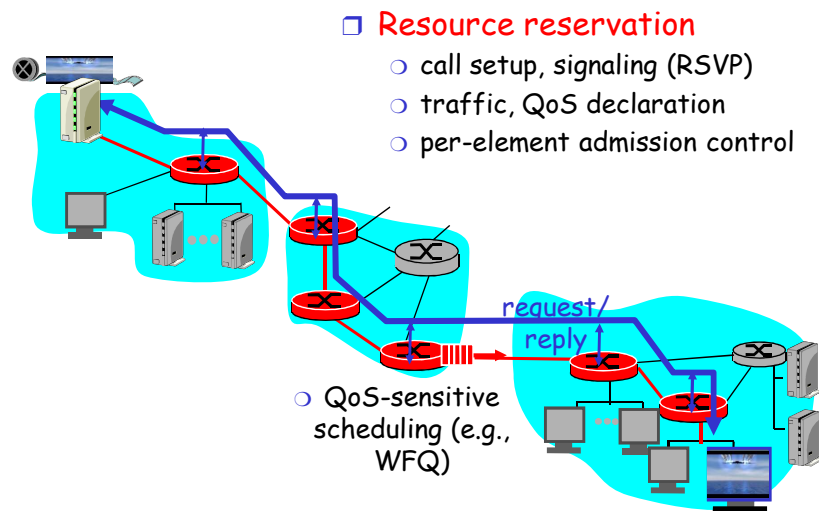


Principle 4

Call Admission: flow declares its needs, network may block call (e.g., busy signal) if it cannot meet needs

Multimedia Networking 94

QoS Guarantee Scenario



Multimedia Networking 95

IETF Integrated Services (IntServ)

- Architecture for providing QOS guarantees in IP networks for individual application sessions
- Resource reservation: routers maintain state info (a la VC) of allocated resources, QoS req's
- Admit/Deny new call setup requests:

Question: can newly arriving flow be admitted with performance guarantees while not violated QoS guarantees made to already admitted flows?

Multimedia Networking 96

Call Admission



Arriving session must:

- Declare its QOS requirement
 - **R-spec**: defines the QOS being requested
- Characterize traffic it will send into network
 - **T-spec**: defines traffic characteristics
- Signaling protocol: needed to carry R-spec and T-spec to routers
 - **RSVP**

Multimedia Networking 97

Intserv Service Models [RFC 2211, 2212]

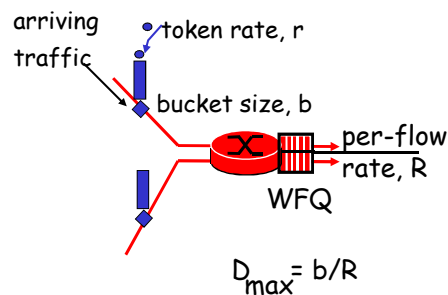


Guaranteed service:

- worst case traffic arrival: leaky-bucket-policed source
- simple (mathematically provable) **bound** on delay [Parekh 1992, Cruz 1988]

Controlled load service:

- "a quality of service closely approximating the QoS that same flow would receive from an unloaded network element."



Multimedia Networking 98

Signaling in the Internet



connectionless
(stateless)
forwarding by IP
routers + best effort
service = no network
signaling protocols
in initial IP
design

- ❑ **New requirement:** reserve resources along end-to-end path (end system, routers) for QoS for multimedia applications
- ❑ **RSVP:** Resource Reservation Protocol [RFC 2205]
 - " ... allow users to communicate requirements to network in robust and efficient way." i.e., signaling !
- ❑ earlier Internet Signaling protocol: ST-II [RFC 1819]

Multimedia Networking 99

RSVP Design Goals



1. Accommodate **heterogeneous receivers** (different bandwidth along paths)
2. Accommodate different applications **with different resource requirements**
3. Make **multicast a first class service**, with adaptation to multicast group membership
4. **Leverage existing multicast/unicast routing**, with adaptation to changes in underlying unicast, multicast routes
5. **Control protocol overhead** to grow (at worst) linear in # receivers
6. **Modular design** for heterogeneous underlying technologies

Multimedia Networking 100

RSVP: does not...



- ❑ Specify how resources are to be reserved
 - rather: a mechanism for communicating needs
- ❑ Determine routes packets will take
 - that's the job of routing protocols
 - signaling decoupled from routing
- ❑ Interact with forwarding of packets
 - separation of control (signaling) and data (forwarding) planes

Multimedia Networking 101

RSVP: Overview of Operation



- ❑ **Senders, receiver join a multicast group**
 - done outside of RSVP
 - senders need not join group
- ❑ **Sender-to-network signaling**
 - **path message**: make sender presence known to routers
 - path teardown: delete sender's path state from routers
- ❑ **Receiver-to-network signaling**
 - **reservation message**: reserve resources from sender(s) to receiver
 - reservation teardown: remove receiver reservations
- ❑ **Network-to-end-system signaling**
 - path error
 - reservation error

Multimedia Networking 102

Summary



Principles

- ❑ classify multimedia applications
- ❑ identify network services applications need
- ❑ making the best of best effort service

Protocols and Architectures

- ❑ specific protocols for best-effort
- ❑ mechanisms for providing QoS
- ❑ architectures for QoS
 - multiple classes of service
 - QoS guarantees, admission control