Telematics

Chapter 11: Multimedia Networking

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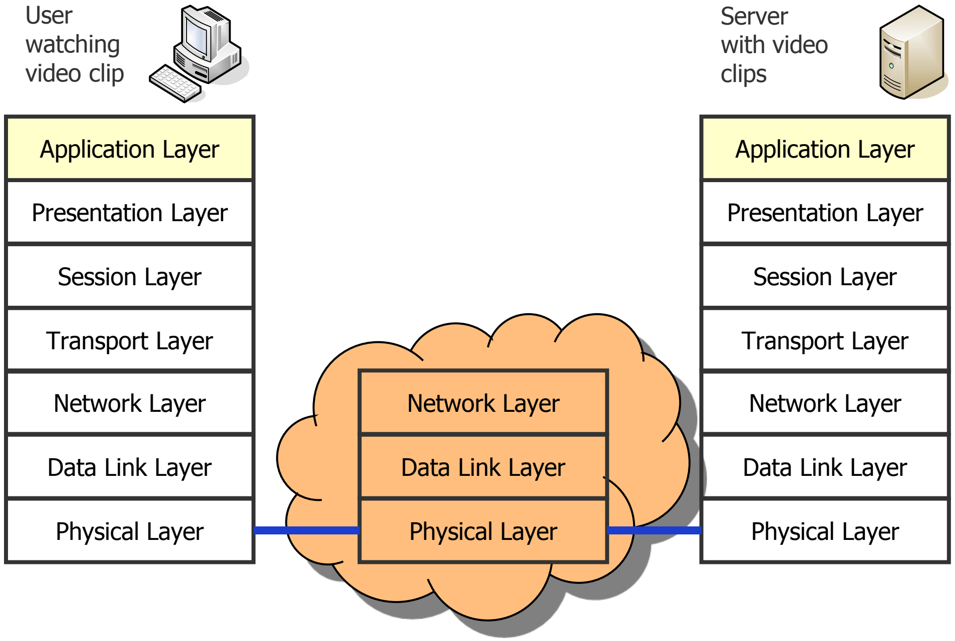


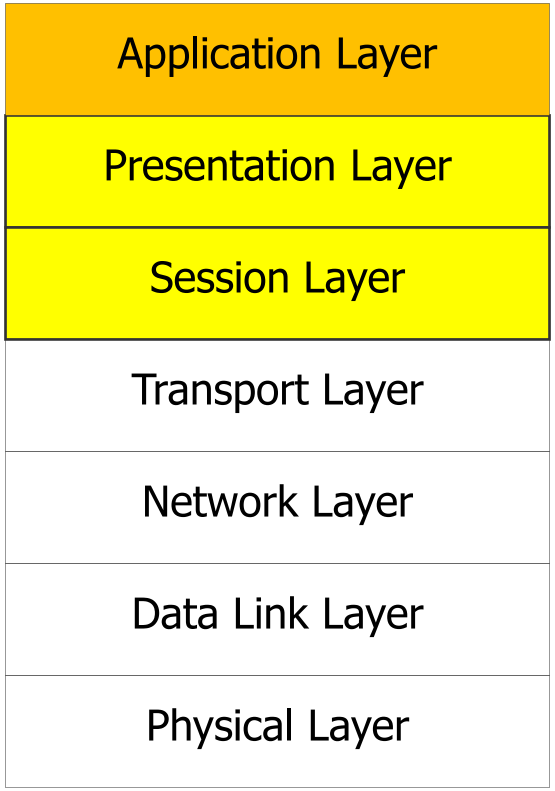
Contents

- Design issues
- Multimedia networking applications
- Streaming stored audio and video
- Making the best out of best effort service
- Protocols for real-time interactive applications
- Providing multiple classes of service
- Providing QoS guarantees

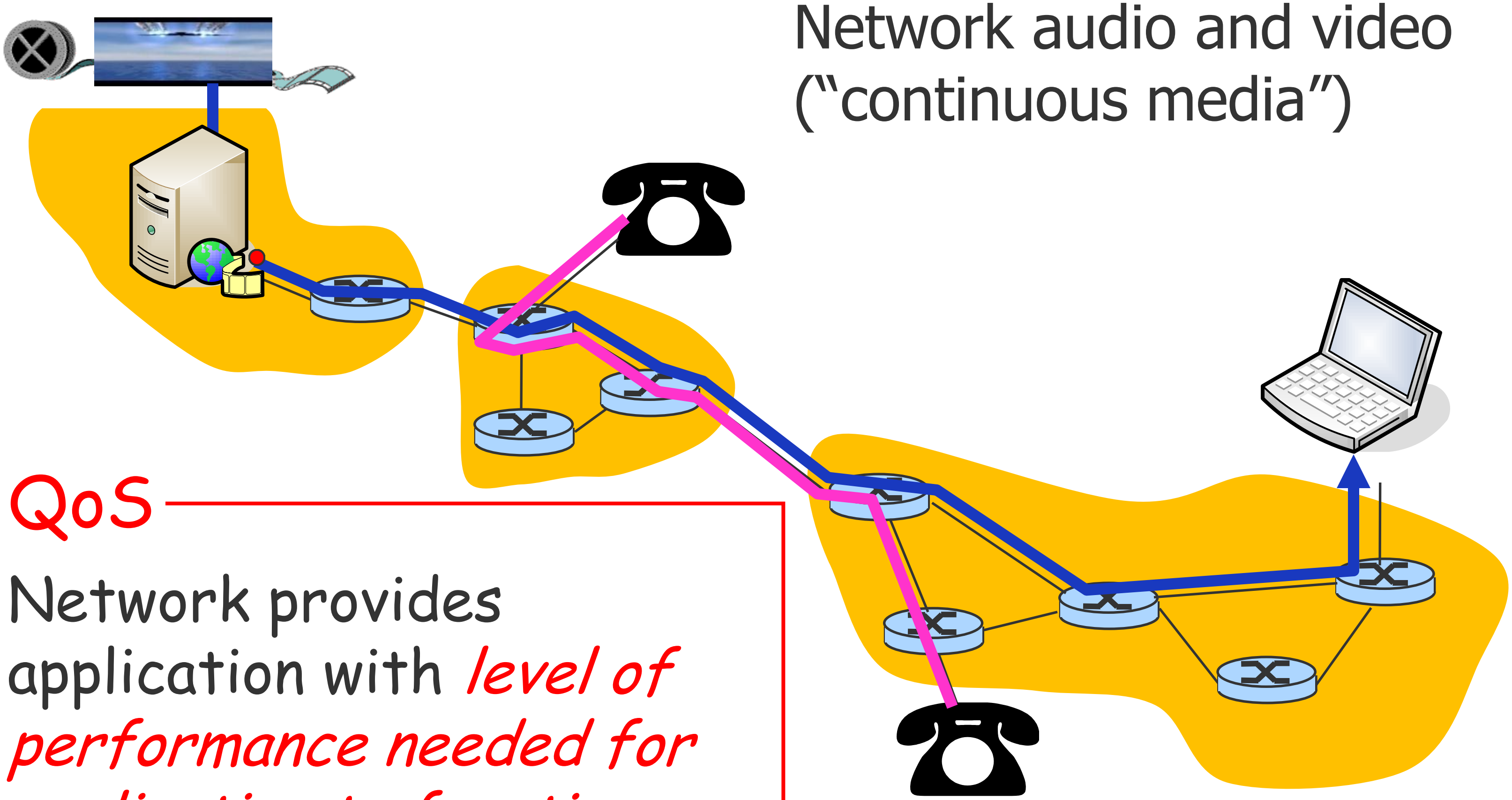
Design Issues

- Two kinds of applications
  - Non-real-time applications
    - Email, FTP,
  - Real-time applications
    - Audio, video
- Principles
  - Classify multimedia applications
  - Identify network services applications need
  - Making the best of the best effort service
- Protocols and Architectures
  - Specific protocols for best-effort
  - Mechanisms for providing QoS
  - Architectures for QoS

OSI Reference Model



Multimedia and Quality of Service: What is it?



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

QoS

Network provides application with *level of performance needed for application to function.*

Quality of Service in the Internet

- Problem today:
  - IP is packet switched, therefore no guarantees on transmission is given
    - Throughput, transmission delay, …
    - The Internet transmits data at **best effort**
  - But: many applications need a certain Quality of Service (QoS)

Application	Reliability	Delay	Jitter	Throughput
E-Mail	high	low	low	low
File Transfer	high	low	low	medium
Web Access	high	medium	low	medium
Remote Login	high	medium	medium	low
Audio on Demand	low	low	high	medium
Video on Demand	low	low	high	high
IP Telephony	low	high	high	low
Video Conference	low	high	high	high

QoS Parameters

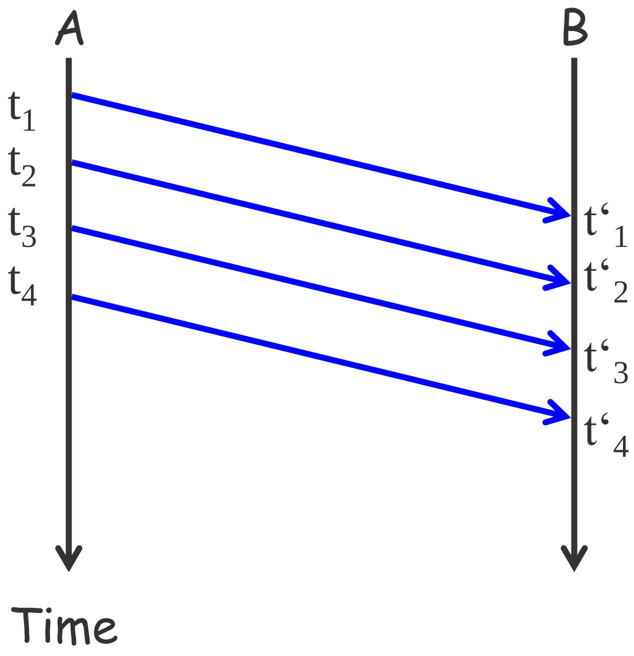
- Throughput [bit/s]
  - Which minimum/maximum/average data rate is necessary?
- Transmission delay [s]
  - Which maximum delay is tolerable?
- Jitter [s]
  - Which fluctuations in the transmission delay are tolerable?
- Availability [%]
  - With which probability the communication service is available?

Multimedia networking applications

MM Networking Applications

- Classes of MM applications:
    - Stored streaming
    - Live streaming
    - Interactive, real-time

- 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



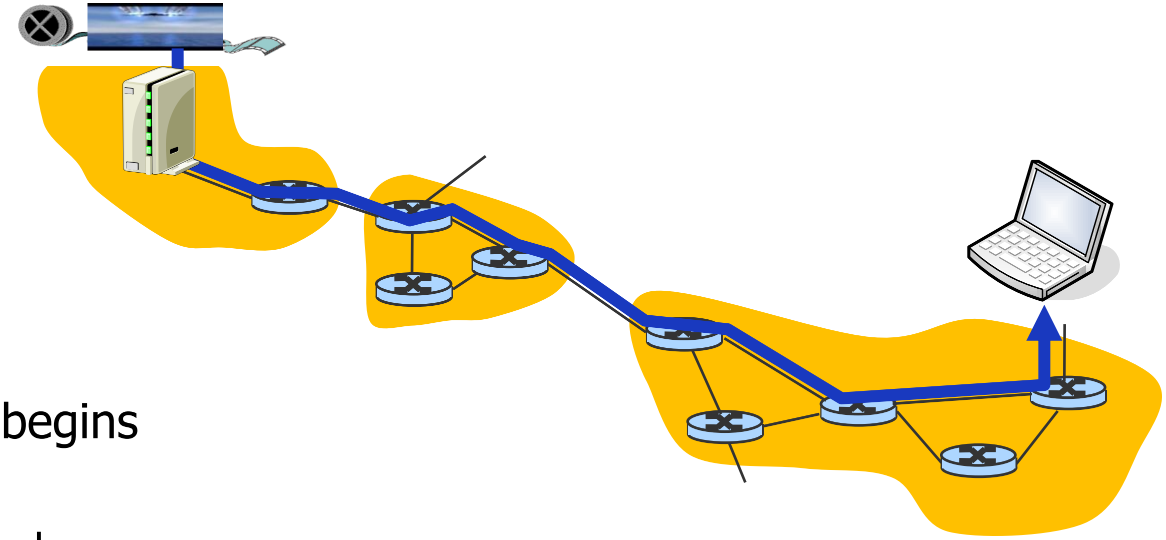
$d_i = t'_i - t_i$

$j_i = d_{i+1} - d_i$

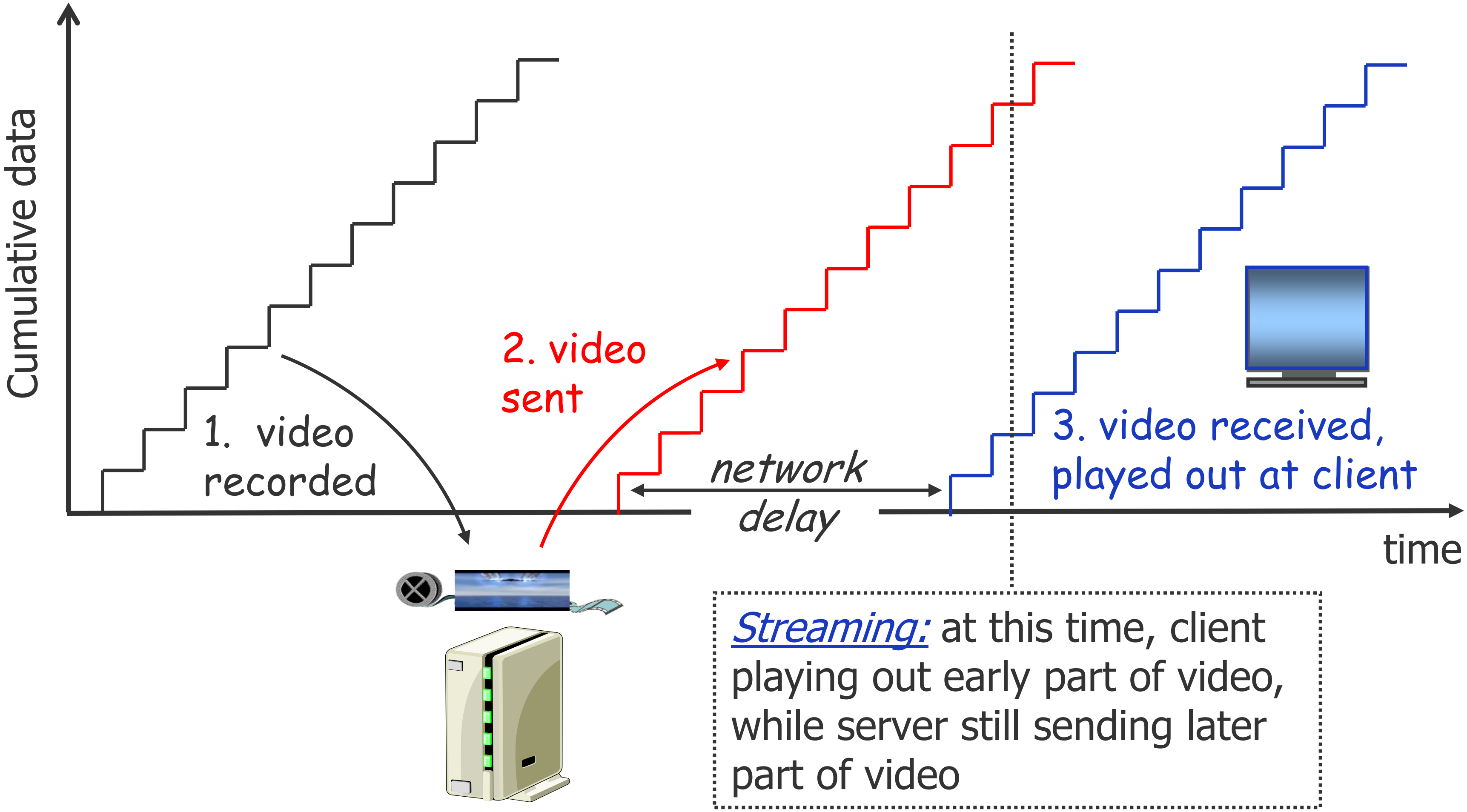
Jitter j_i is the variability of packet delays within the same packet stream

Streaming Stored Multimedia

- Stored streaming:
  - Media stored at source
  - Transmitted to client
  - Streaming: client play out begins before all data has arrived
    - Timing constraint for still-to-be transmitted data: in time for play out

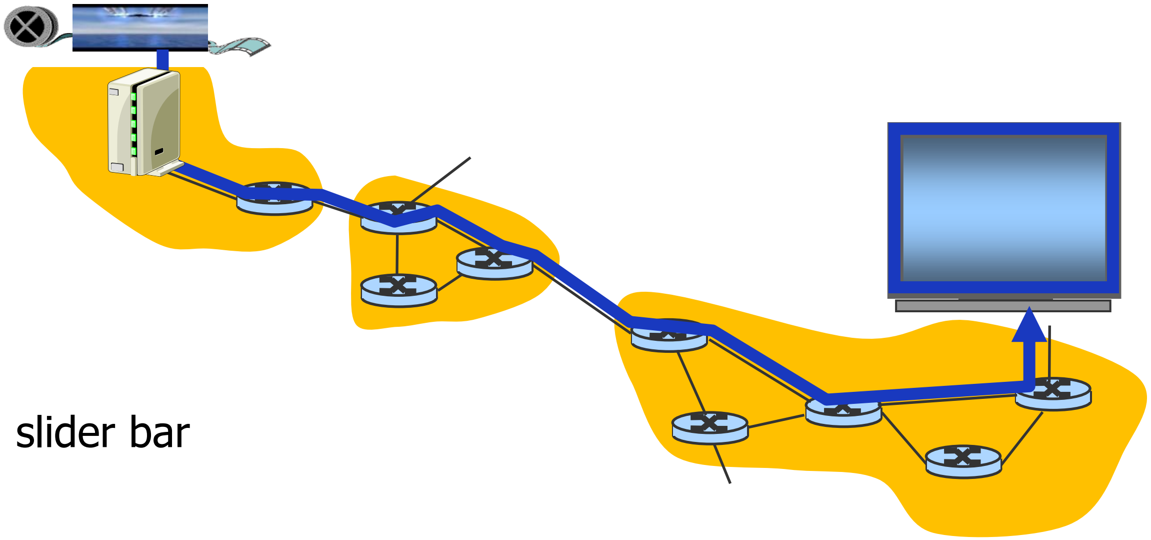


Streaming Stored Multimedia: What is it?



Streaming: at this time, client playing out early part of video, while server still sending later part of video

Streaming Stored Multimedia: Interactivity

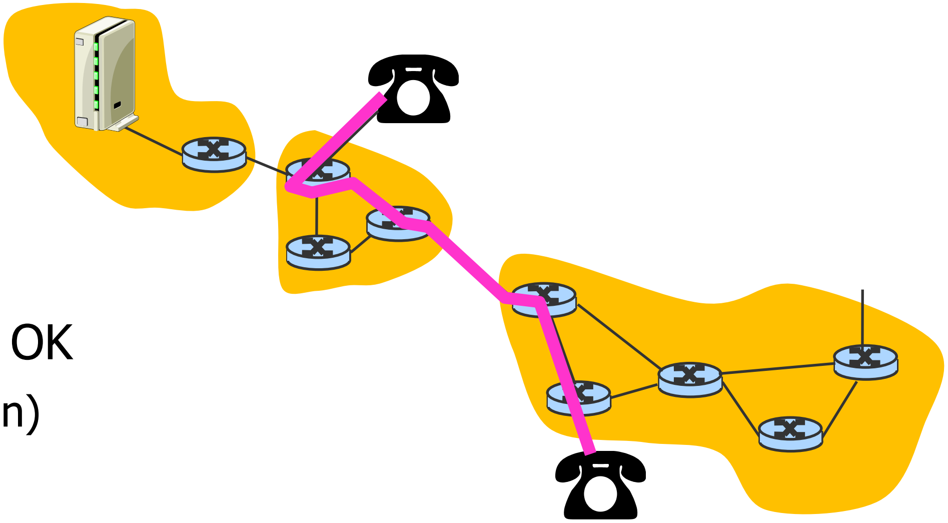


- VCR-like functionality:
  - Client can
    - pause, rewind, FF, push slider bar
  - Delay restrictions
    - 10 sec initial delay OK
    - 1-2 sec until command effect OK
- Timing constraint for still-to-be transmitted data: in time for play out

Streaming Live Multimedia

- Examples:
  - Internet radio talk show
  - Live sporting event
- 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!

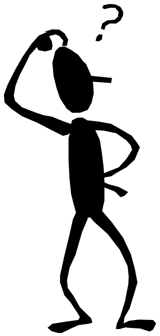
Real-Time Interactive Multimedia



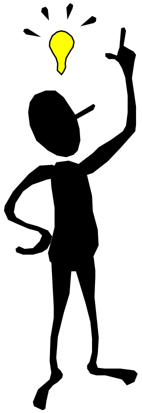
- Applications
  - IP telephony, video conference
  - Distributed interactive worlds
- End-to-end delay requirements
  - Audio: <150 msec good, <400 msec OK
    - Includes application-level (packetization) and network delays
    - Higher delays noticeable, impair interactivity
- Session initialization
  - How does callee advertise its IP address, port number, encoding algorithms?

Multimedia Over Today’s Internet

- TCP/UDP/IP: “best-effort service”
    - No guarantees on delay, loss



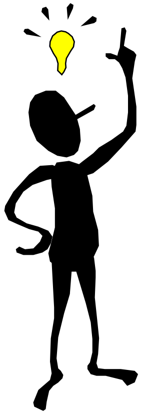
But you said multimedia apps require QoS and level of performance to be effective!



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

How should the Internet evolve to better support multimedia?

- Integrated services philosophy
  - Fundamental changes in Internet so that apps can reserve end-to-end bandwidth
  - Requires new, complex software in hosts & routers
- Laissez-faire
  - No major changes
  - More bandwidth when needed
  - Content distribution, application-layer multicast
    - Application layer
- Differentiated services philosophy
  - Fewer changes to Internet infrastructure
  - Provide small number of service classes (maybe two)



What's your opinion?

How should the Internet evolve to better support multimedia?

Approach	Unit of allocation	Guarantee	Deployment to date	Complexity	Mechanisms
Making the best of best-effort service	None	None or soft	Everywhere	Minimal	Application-layer support, CDN, over-provisioning
Differential QoS	Classes of flows	None or soft	Some	Medium	Policing, scheduling
Guaranteed QoS	Individual flows	Soft or hard, once a flow is admitted	Little	High	Policing, scheduling, call admission and signaling

Multimedia networking applications

Audio and Video Compression

A few words about audio compression

- Analog signal sampled at constant rate
  - Telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- Each sample quantized, i.e., rounded
  - $2^8=256$ possible quantized values
- Each quantized value represented by bits
  - 8 bits for 256 values

- Example:
  - 8,000 samples/sec, 256 quantized values ➡ 64,000 bps
  - Receiver converts bits back to analog signal
    - Some quality reduction

- Example rates
  - CD: 1.411 Mbps (stereo)
  - MP3: 96, 128, 160 kbps
  - GSM: 13 kbps
  - Internet telephony
    - G.729: 8 kbps
    - G.723.3: 5.3 kbps and 6.4 kbps

A few words about video compression

- Video: sequence of images displayed at constant rate
  - Frame rate 24 images/sec
- Digital image: array of pixels
  - Each pixel represented by bits
- Redundancy
  - Spatial (within image)
  - Temporal (from one image to next)

- Example
  - Single image of 1024 * 768 pixels
  - Each pixel encoded into 24 bits
  - ➡2.25 Mbyte without compression
  - ➡Compression ratio 10:1 results in < 250 kbyte

- Examples:
  - MPEG1 (CD-ROM) 1.5 Mbps
  - MPEG2 (DVD) 3-6 Mbps
  - MPEG4 <1 Mbps
    - Often used in Internet
- Research: layered (scalable) video
  - Adapt layers to available bandwidth

Streaming stored audio and video

Streaming Stored Multimedia

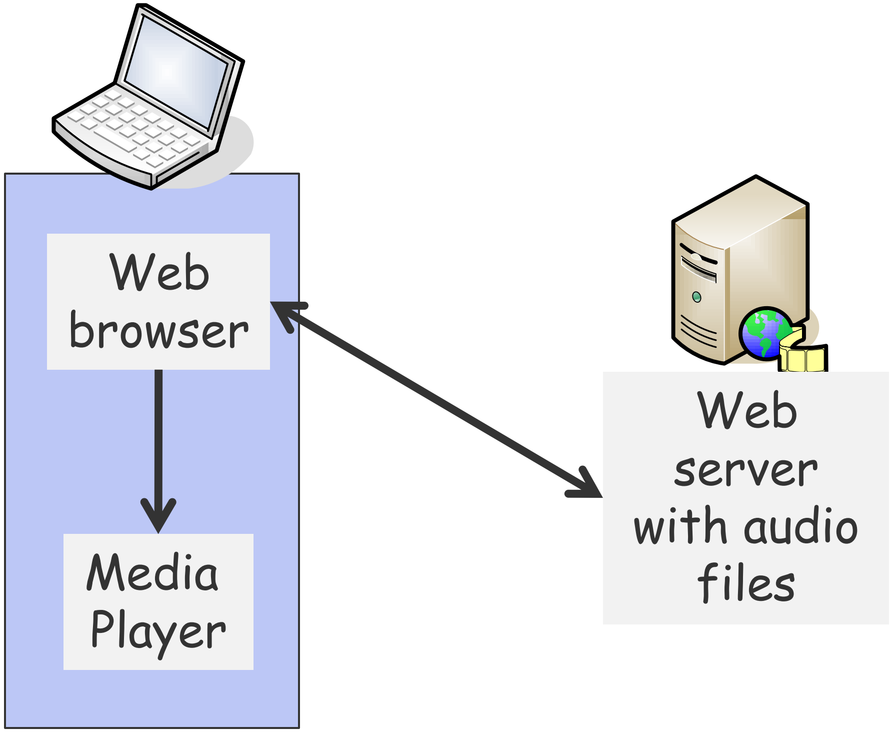
- Application-level streaming techniques for making the best out of best effort service:
  - Client-side buffering
  - Use of UDP versus TCP
  - Multiple encodings of multimedia

- Media Player
  - Jitter removal
  - Decompression
  - Error concealment
  - Graphical user interface with controls for interactivity

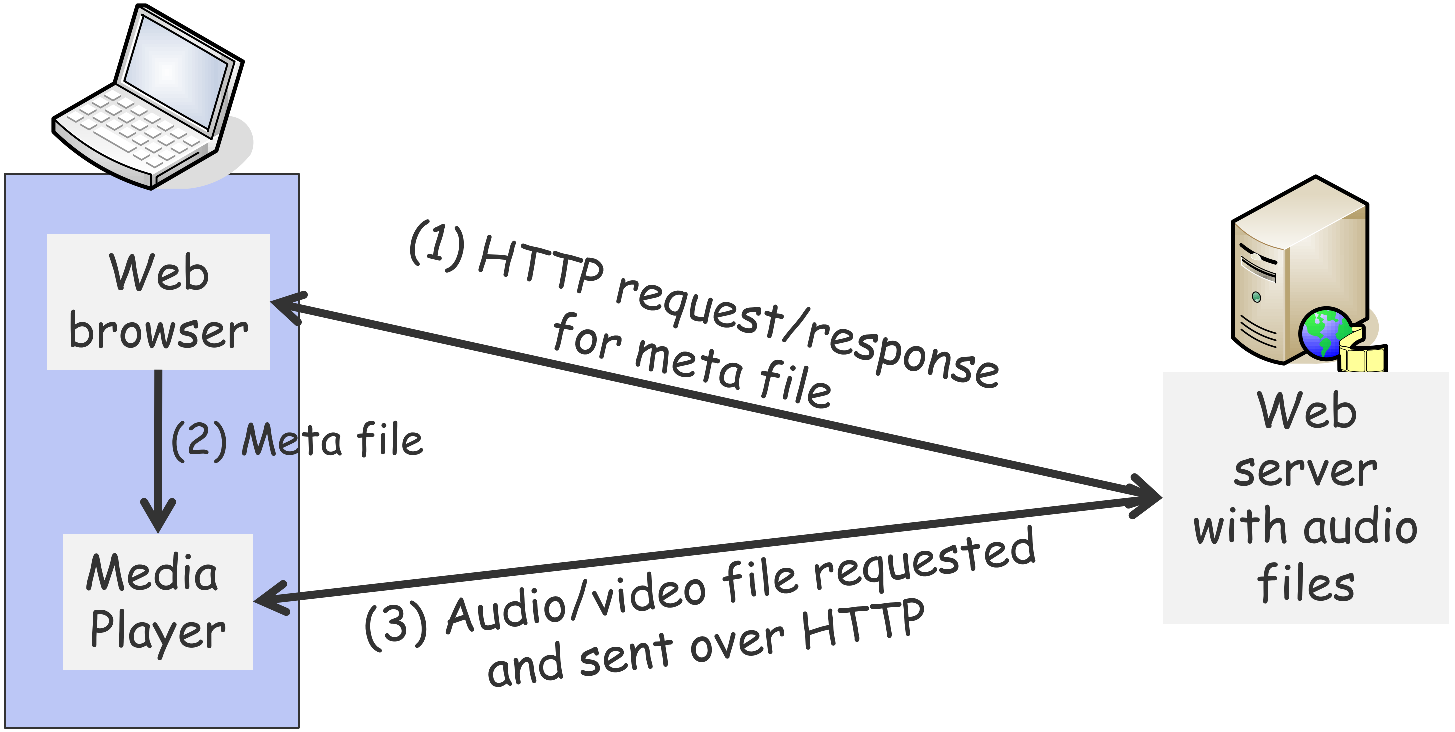
Internet multimedia: Simplest approach

- Audio or video stored in file
- Files transferred as HTTP object
  - Received in entirety at client
  - Then passed to player

- Audio, video not streamed:
  - No, “pipelining,” long delays until play out!

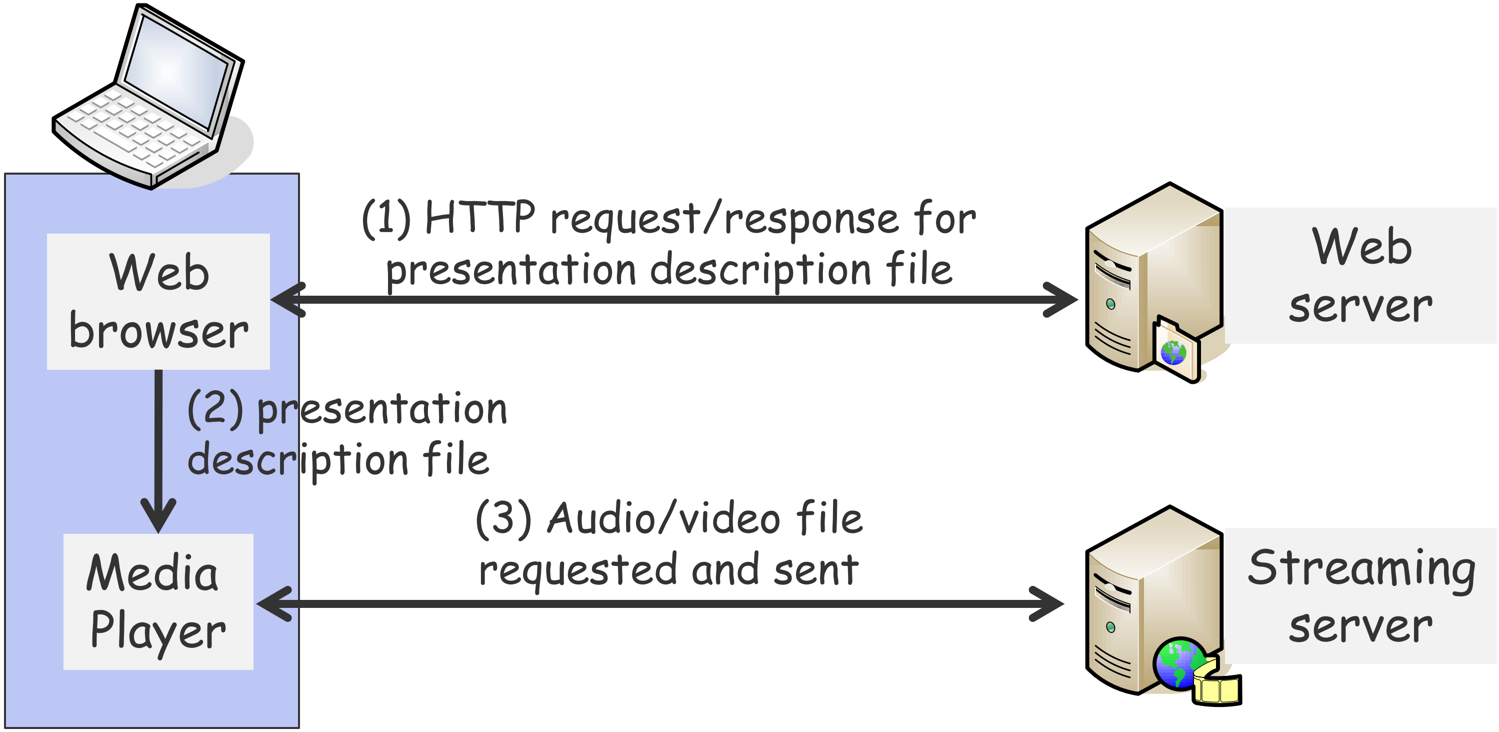


Internet multimedia: Streaming approach


Web browser
(2) Meta file
Media Player
(1) HTTP request/response for meta file
(3) Audio/video file requested and sent over HTTP
Web server with audio files

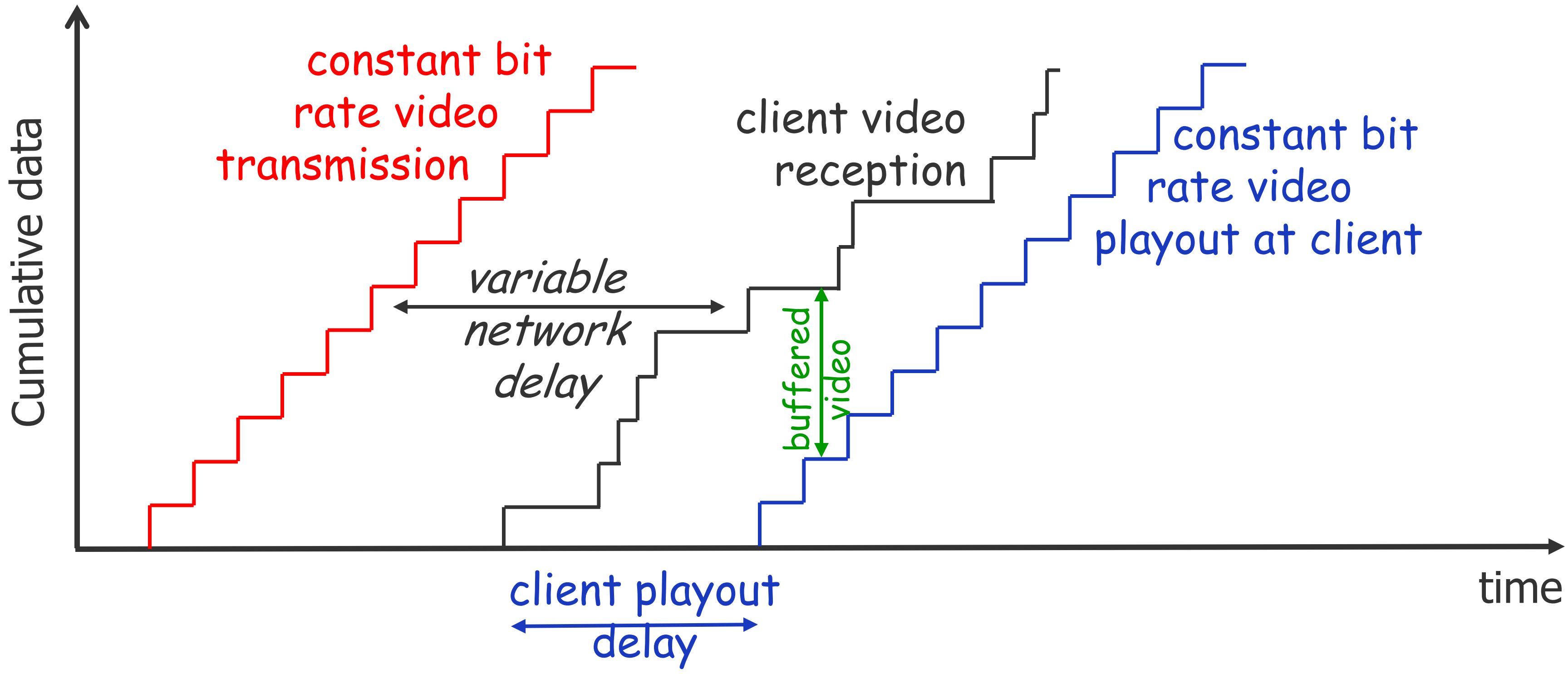
- Browser gets **metafile**
- Browser launches player, passing metafile
- Player contacts server
- Server streams audio/video to player

Streaming from a streaming server


Web browser
Media Player
(1) HTTP request/response for presentation description file
(2) presentation description file
(3) Audio/video file requested and sent
Web server
Streaming server

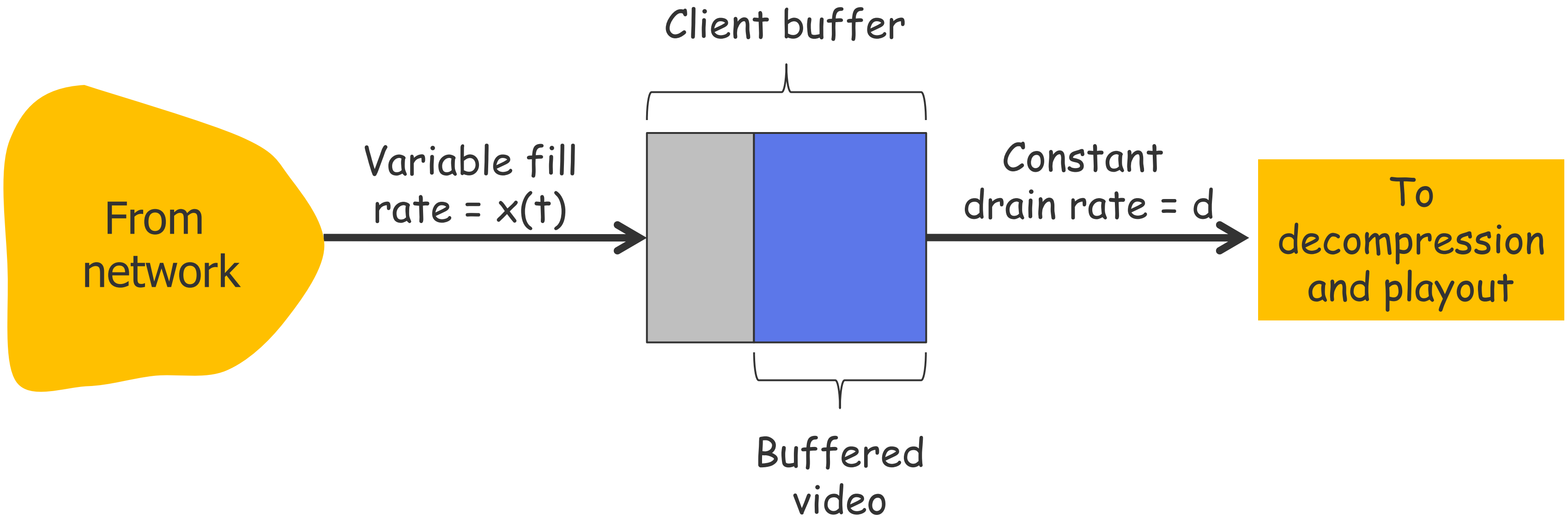
- Allows for non-HTTP protocol between server and media player
- UDP or TCP for step (3)

Streaming Multimedia: Client Buffering



- Client-side buffering, play out delay compensate for network-added delay, delay jitter

Streaming Multimedia: Client Buffering

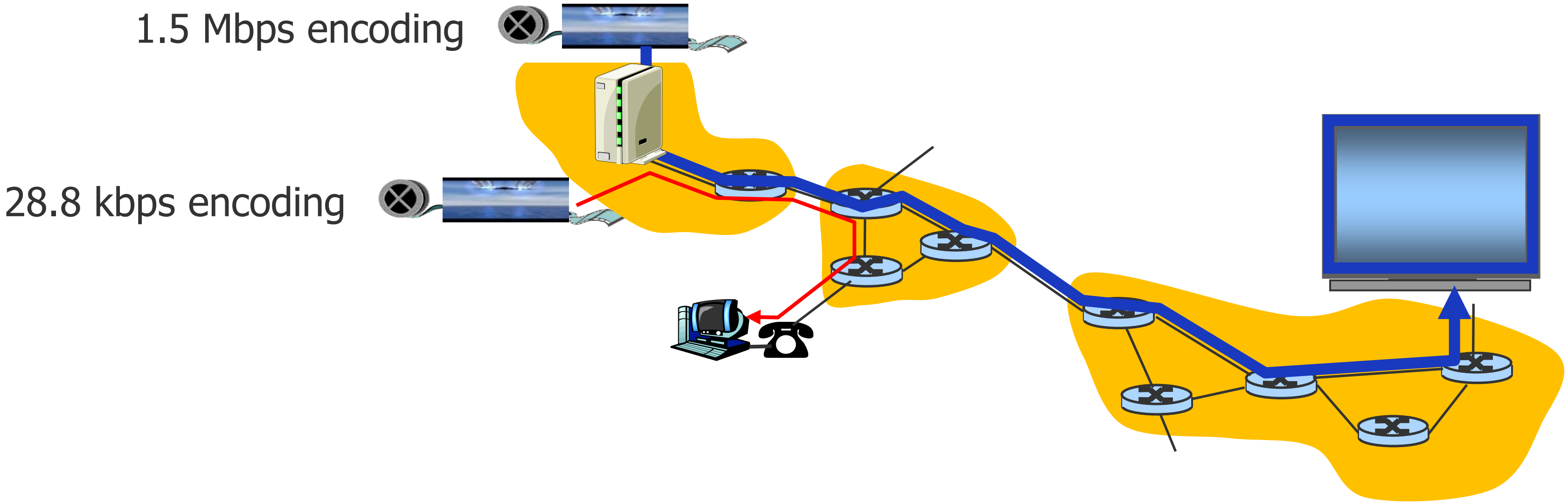


- Client-side buffering, play out delay compensate for network-added delay, delay jitter

Streaming Multimedia: UDP or TCP?

- UDP
  - Server sends at rate appropriate for client (oblivious to network congestion !)
    - Often send rate = encoding rate = constant rate
    - ➡ Fill rate = constant rate - packet loss
  - Short play out delay (2-5 seconds) to remove network jitter
  - Error recover: time permitting
- TCP
  - Send at maximum possible rate under TCP
  - Fill rate fluctuates due to TCP congestion control
  - Larger play out delay: smooth TCP delivery rate
  - HTTP/TCP passes more easily through firewalls

Streaming Multimedia: Client rate(s)



- Q: How to handle different client receive rate capabilities?
  - 28.8 kbps dialup
  - 100 Mbps Ethernet
- A: Server stores, transmits multiple copies of video, encoded at different rates

Streaming stored audio and video

RTSP

User Control of Streaming Media: RTSP

- HTTP
  - Does not target multimedia content
  - No commands for fast forward, etc.
- RTSP: RFC2326
  - Client-server application layer protocol
  - User control
    - rewind
    - fast forward
    - pause, resume
    - repositioning, etc.

- What it does not do:
  - Does not define compression schemes
  - Does not define how audio/video is encapsulated for streaming over network
  - Does not restrict how streamed media is transported (UDP or TCP possible)
  - Does not specify how media player buffers audio/video

RTSP: Out of band control

- FTP uses an "out-of-band" control channel:
  - File transferred over one TCP connection.
  - Control info (directory changes, file deletion, rename) sent over separate TCP connection
  - "Out-of-band", "in-band" channels use different port numbers

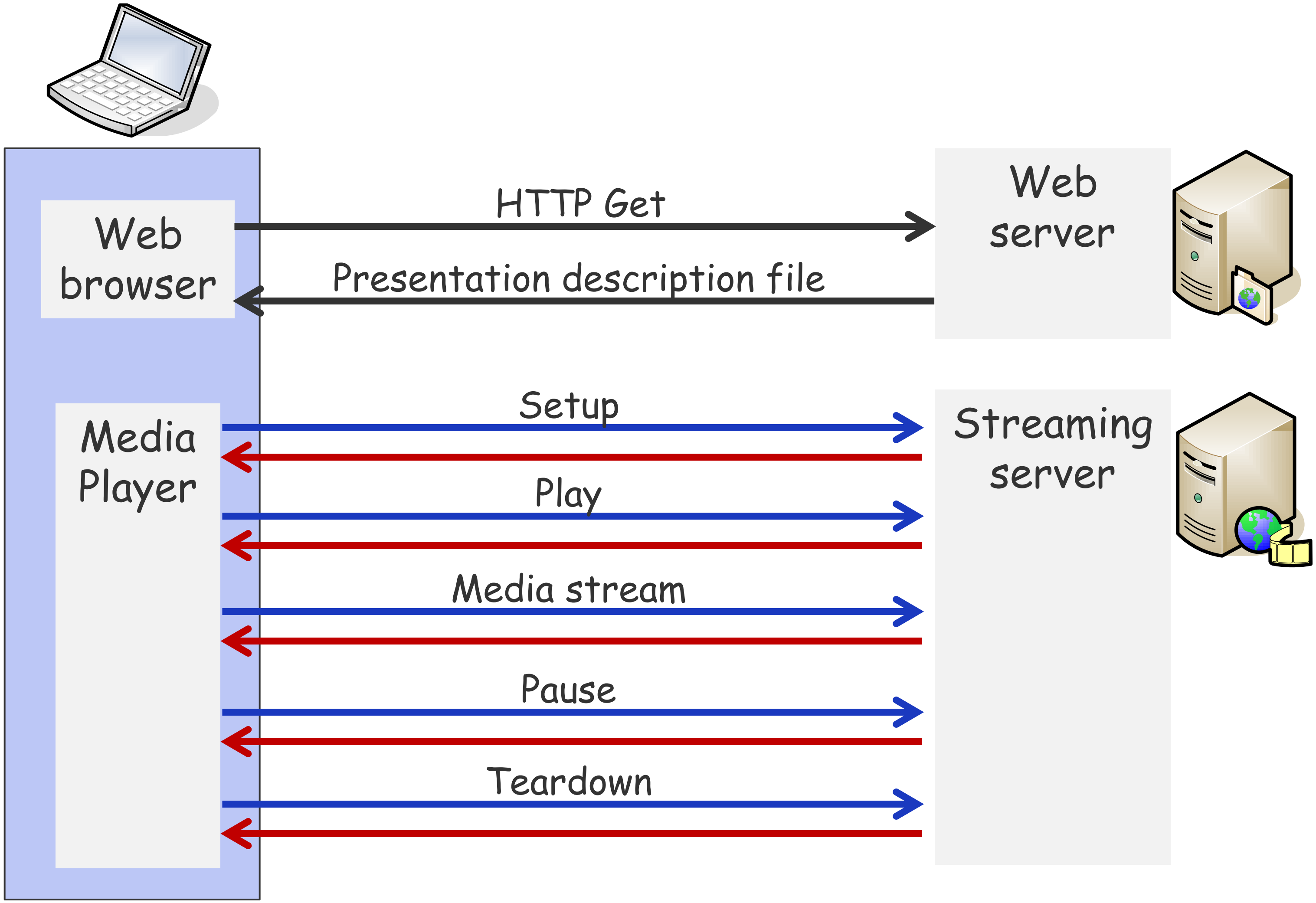
- RTSP messages also sent out-of-band:
  - RTSP control messages use different port numbers than media stream: out-of-band
    - Port 554
  - Media stream is considered "in-band"

RTSP Example

- Scenario:
  - Metafile communicated to web browser
  - Browser launches player
  - Player sets up an RTSP control connection, data connection to streaming server
- Metafile example

```
<title>Twister</title>
<session>
        <group language=en lipsync>
            <switch>
                 <track type=audio
                     e="PCMU/8000/1"
                     src = "rtsp://audio.example.com/twister/audio.en/lofi">
                 <track type=audio
                     e="DVI4/16000/2" pt="90 DVI4/8000/1"
                     src="rtsp://audio.example.com/twister/audio.en/hifi">
             </switch>
             <track type="video/jpeg‟ src="rtsp://video.example.com/twister/video">
         </group>
</session>
```

RTSP Operation



RTSP Exchange Example

```
C: SETUP rtsp://audio.example.com/twister/audio RTSP/1.0
   Transport: rtp/udp; compression; port=3056; mode=PLAY

S: RTSP/1.0 200 1 OK
     Session 4231

C: PLAY rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
     Session: 4231
     Range: npt=0-

C: PAUSE rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
     Session: 4231
     Range: npt=37

C: TEARDOWN rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
     Session: 4231

S: 200 3 OK
```

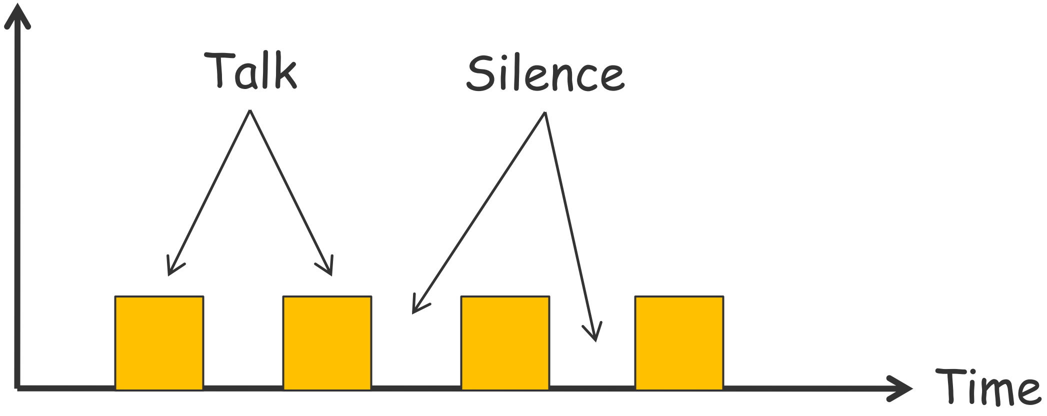
Difference to HTTP
RTSP tracks session and state of the client

Making the best out of best-effort service

Removing Jitter at the Receiver

Interactive Multimedia: Internet Phone

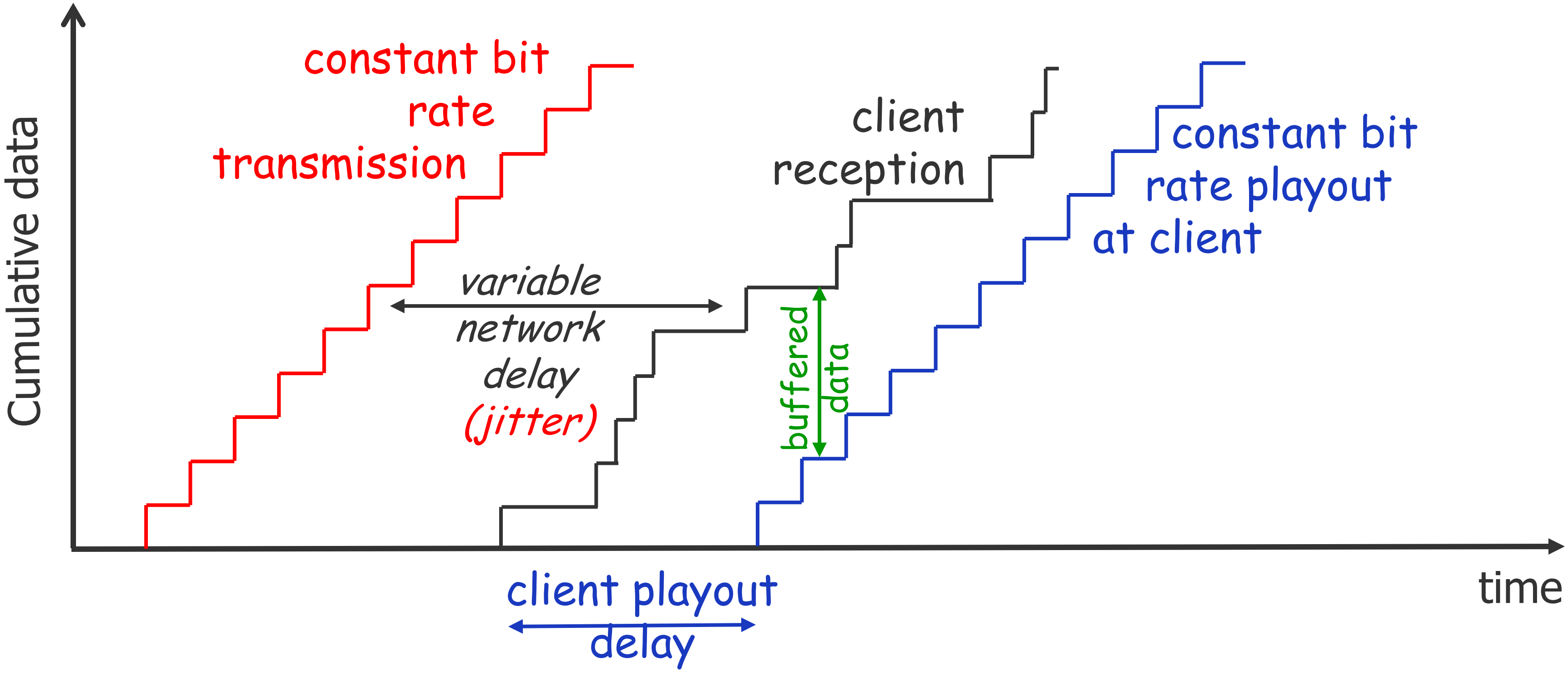
- Look at a PC-2-PC Internet phone example in detail
  - Speaker’s audio: alternating talk spurts and silent periods
    - 64 kbps during talk spurt
    - Packets generated only during talk spurts
    - 20 msec chunks at 8000 byte/sec ➡ 1 chunk = 160 byte data
  - Application-layer header added to each chunk
  - Chunk+header encapsulated into UDP segment
  - Application sends UDP segment into socket every 20 msec during talkspurt



Internet Phone: Packet Loss and Delay

- Network loss: IP datagram lost due to network congestion
  - Router buffer overflow
- Delay loss: IP datagram arrives too late for play out at receiver
  - Delays:
    - Processing
    - Queuing in network
    - End-system (sender, receiver) delays
  - Typical maximum tolerable delay: 400 ms
- Loss tolerance
  - Depending on voice encoding, losses concealed, packet loss rates between 1% and 10% can be tolerated

Delay Jitter



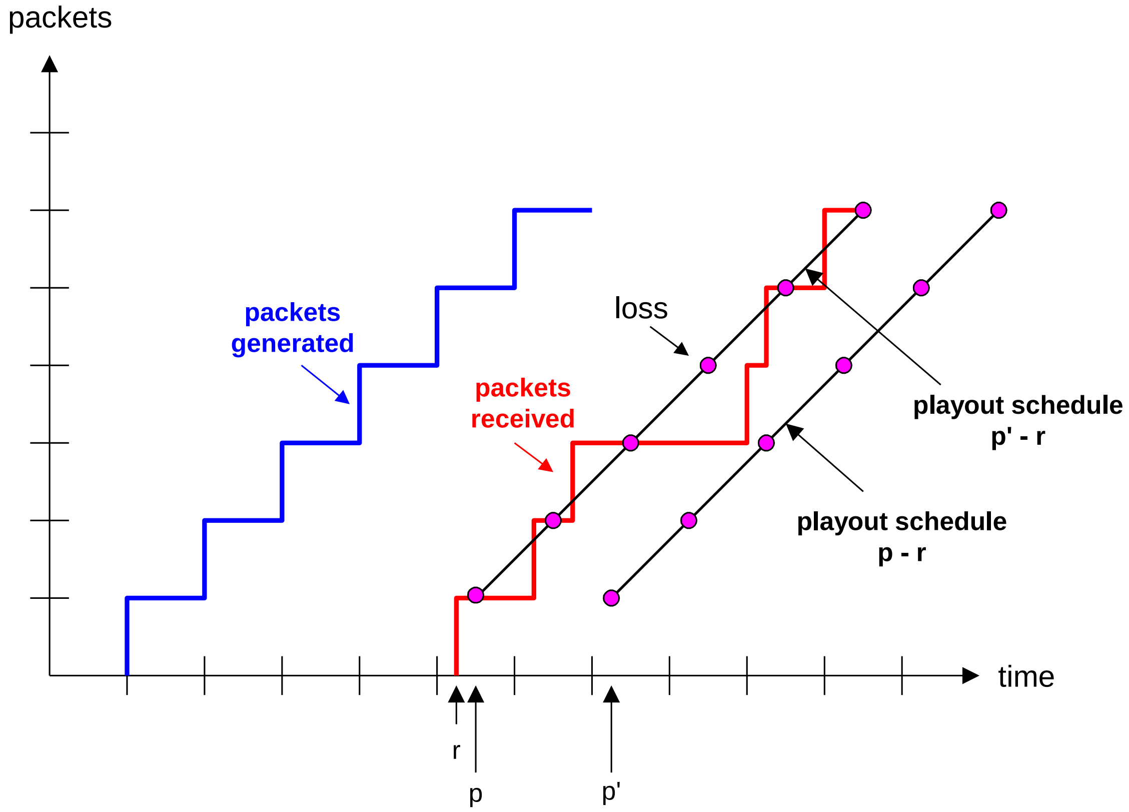
- Consider end-to-end delays of two consecutive packets: difference can be more or less than 20 msec (transmission time difference)

Internet Phone: Fixed Playout Delay

- Receiver attempts to play out 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 play out, data “lost”
- Tradeoff in choosing q:
  - Large q: less packet loss
  - Small q: better interactive experience

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'*



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

$$
\begin{aligned}
t_i &= \text{timestamp of the } i\text{-th packet} \\
r_i &= \text{the time packet } i \text{ is received by receiver} \\
p_i &= \text{the time packet } i \text{ is played at receiver} \\
r_i - t_i &= \text{network delay for } i\text{-th packet} \\
d_i &= \text{estimate of average network delay after receiving } i\text{-th packet}
\end{aligned}
$$

- 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., $u = 0.01$).

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., $K=4$

- Remaining packets in talkspurt are played out periodically

Adaptive Playout (3)

- Q: 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.

Making the best out of best effort service

Recovery from packet loss

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, increasing bandwidth 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

RTP Header (12 octets or more)
FEC Header (10 octets)
FEC Level 0 Header
FEC Level 0 Payload
FEC Level 1 Header
FEC Level 1 Payload
Cont.

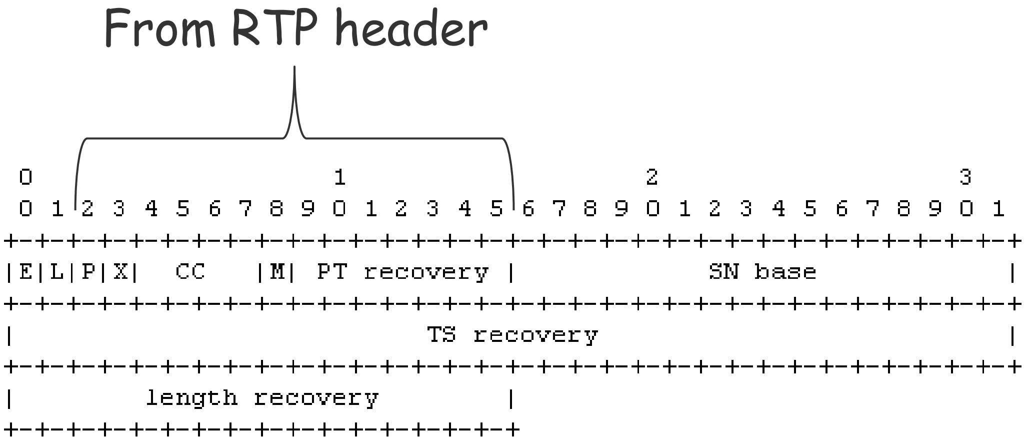
FEC Packet Structure RFC5109

Recovery from packet loss (1): FEC

- The FEC header is 10 octets
  - Extension flag (E bit)
  - Long-mask flag (L bit)
  - P recovery field
  - X recovery field
  - CC recovery field
  - M recovery field
  - PT recovery field
  - SN base field
  - TS recovery field
  - Length recovery field

RTP Header (12 octets or more)
FEC Header (10 octets)
FEC Level 0 Header
FEC Level 0 Payload
FEC Level 1 Header
FEC Level 1 Payload
Cont.

FEC Packet Structure RFC5109

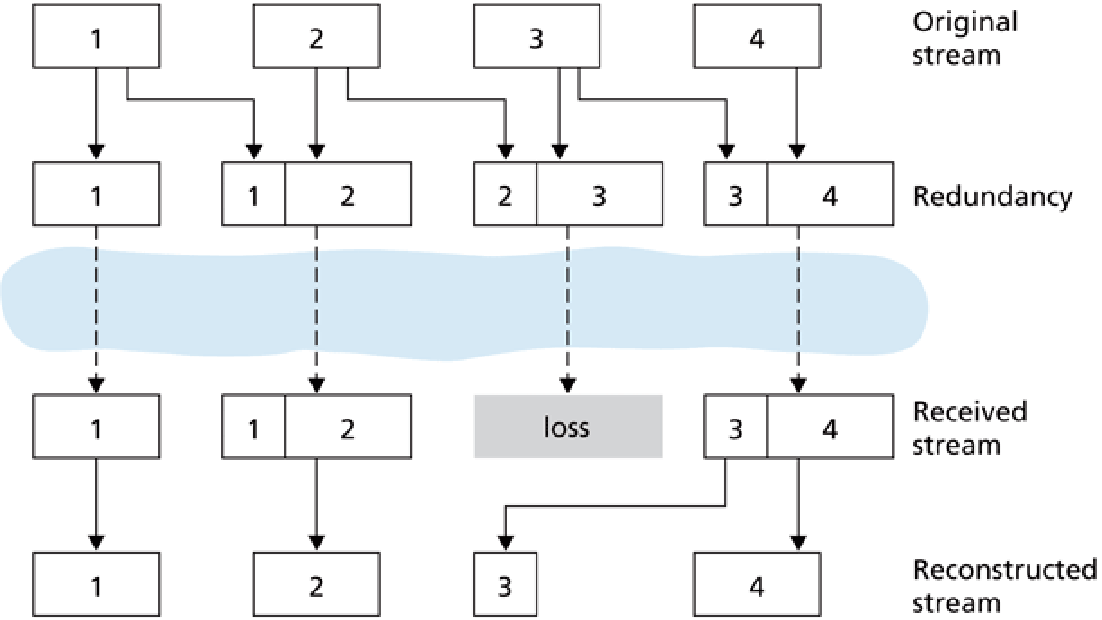


From RTP header

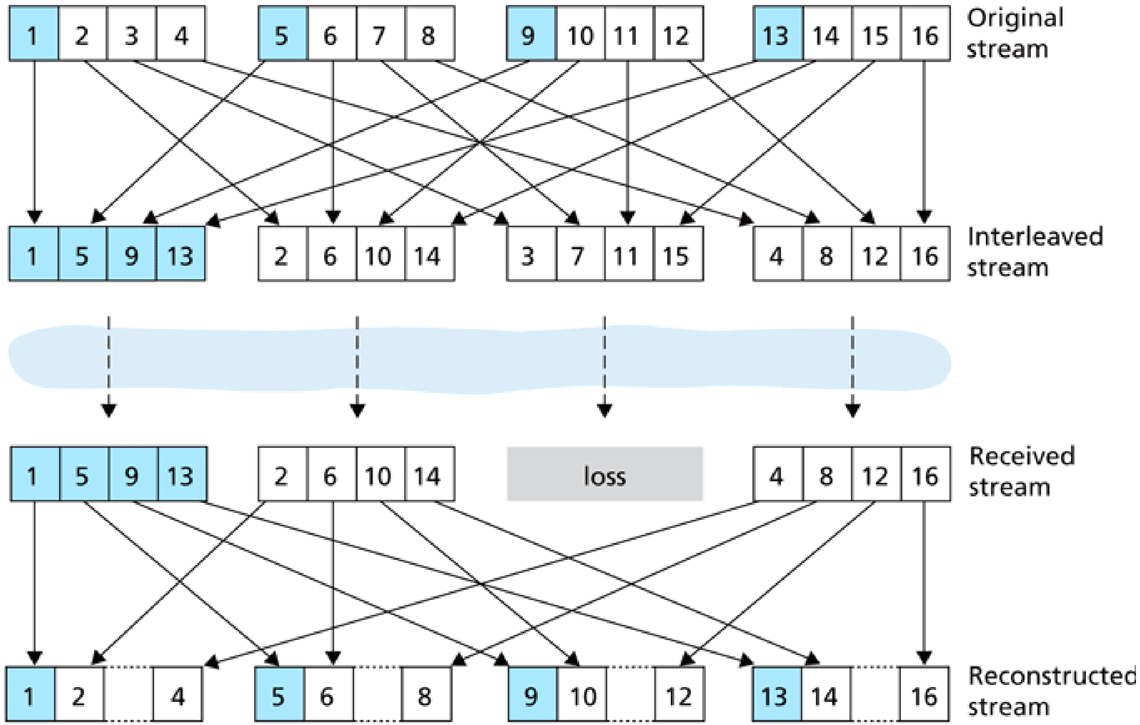
```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|E|L|P|X|  CC   |M| PT recovery |            SN base            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                          TS recovery                          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|        length recovery        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

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



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

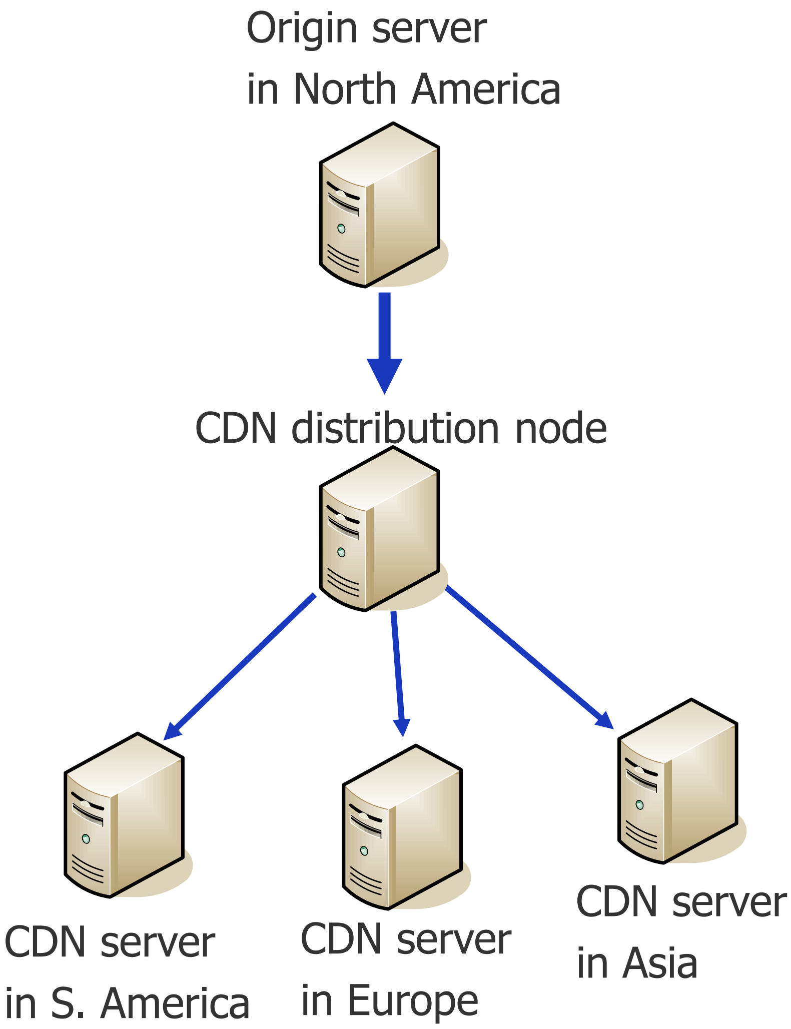
- If packet lost, still have most of every chunk
- No redundancy overhead, but increases playout delay

Making the best out of best-effort-service

Content Distribution Networks (CDN)

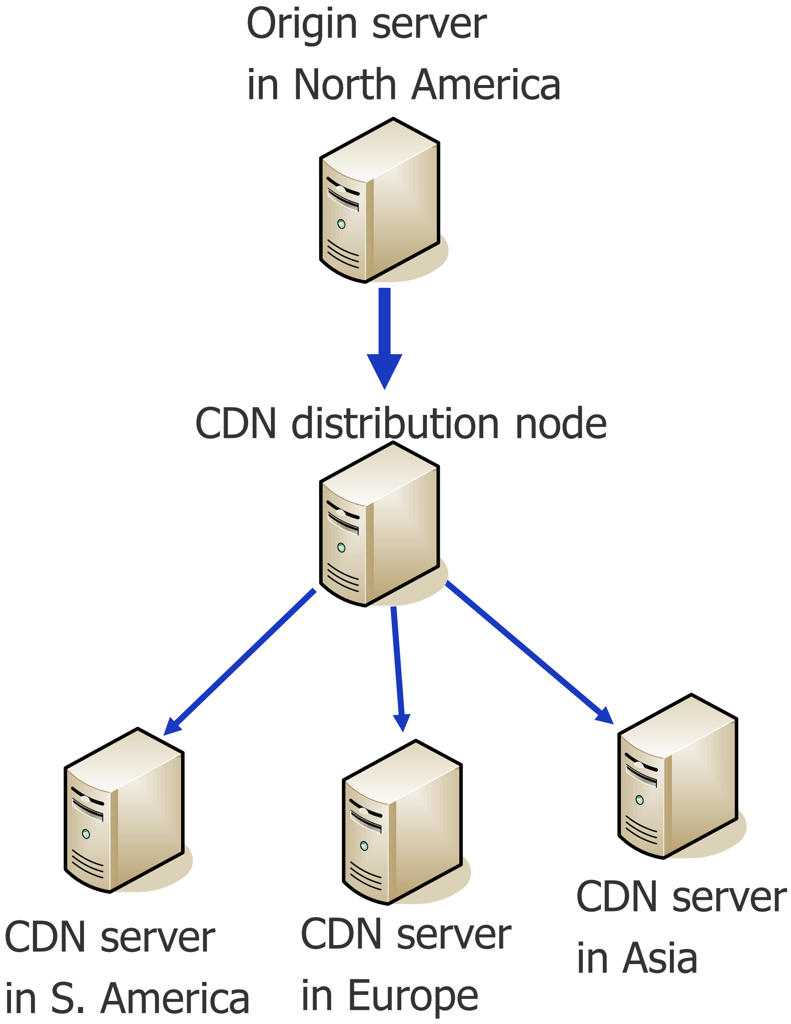
Content distribution networks (CDNs)

- 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



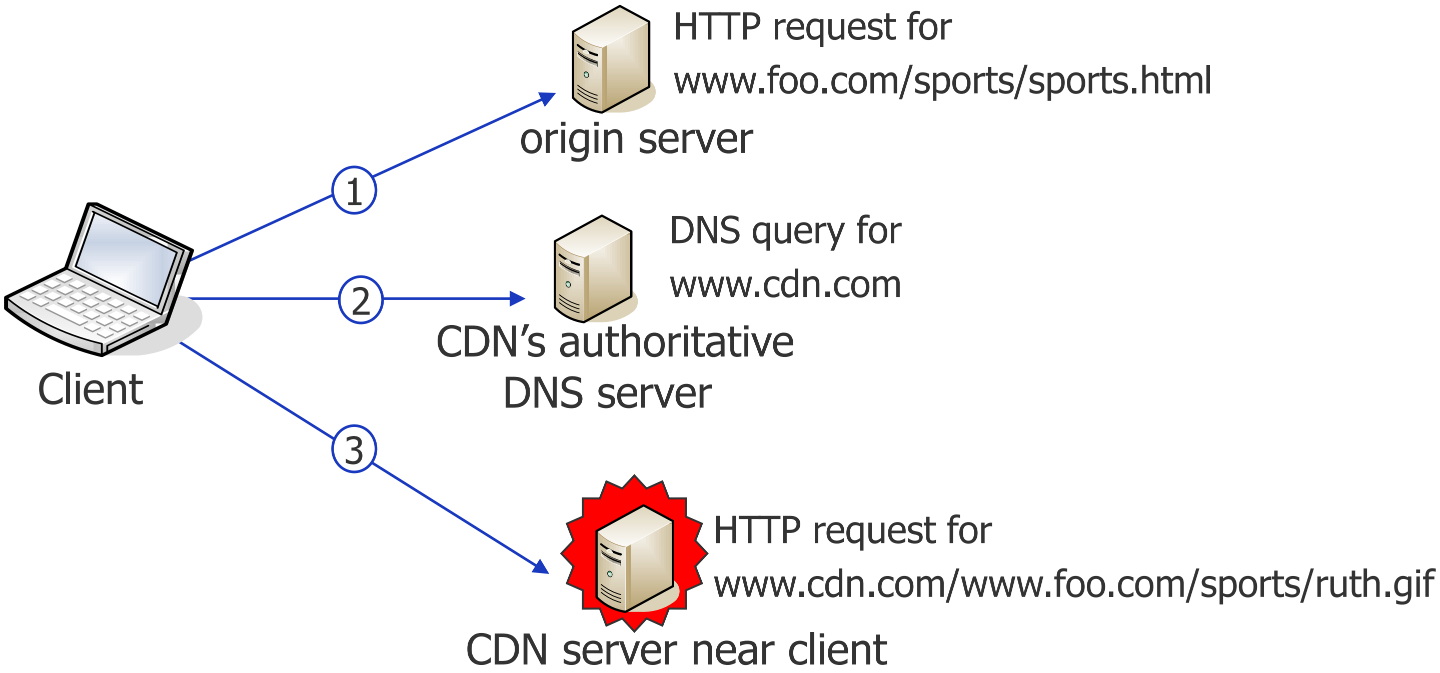
Content distribution networks (CDNs)

- Content replication
  - CDN (e.g., Akamai) customer is the content provider (e.g., CNN)
  - CDN replicates customers' content in CDN servers.
  - when provider updates content, CDN updates servers



CDN example

- Origin server (www.foo.com)
  - Distributes HTML
  - Replaces:
    http://www.foo.com/sports.ruth.gif
    with
    http://www.cdn.com/www.foo.com/sports/ruth.gif
- CDN company (cdn.com)
  - Distributes gif files
  - Uses its authoritative DNS server to route redirect requests



More about CDNs

- Routing requests
  - CDN creates a “map”, indicating distances from leaf ISPs and CDN nodes
  - When query arrives at authoritative DNS server:
    - Server determines ISP from which query originates
    - Uses “map” to determine **best** CDN server
  - CDN nodes create application-layer overlay network

Summary: Internet Multimedia / Bag of tricks

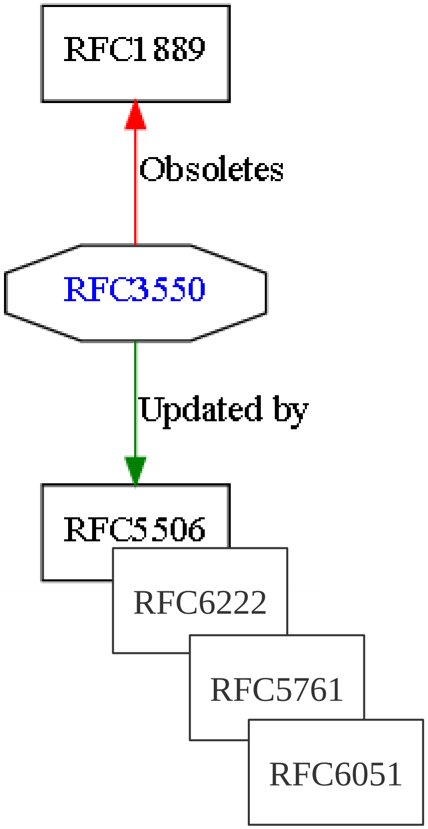
- Use UDP to avoid TCP congestion control (delays) for time-sensitive traffic
- Client-side adaptive playout delay: to compensate for delay
- Server side matches stream bandwidth to available client-to-server path bandwidth
  - Chose among pre-encoded stream rates
  - Dynamic server encoding rate
- Error recovery (on top of UDP)
  - FEC, interleaving, error concealment
  - Retransmissions, time permitting
- CDN: bring content closer to clients

Protocols for real-time interactive applications

RTP, RTCP, SIP

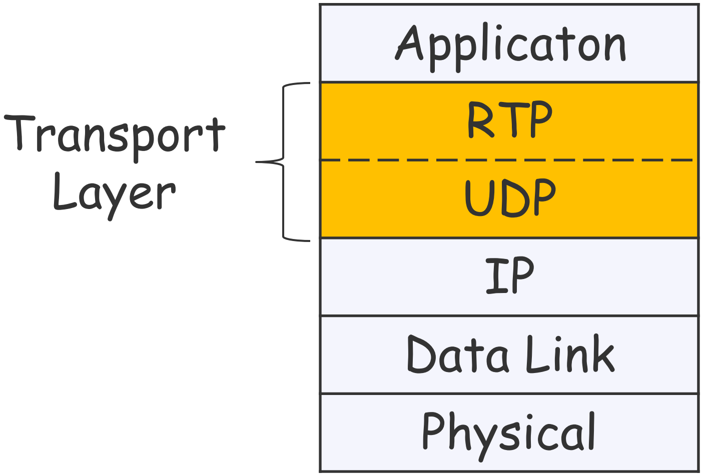
Real-Time Protocol (RTP)

- RTP specifies packet structure for packets carrying audio and 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 runs on top of UDP

- RTP libraries provide transport-layer interface that extends UDP
  - Port numbers, IP addresses
  - Payload type identification
  - Packet sequence numbering
  - Time-stamping



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.

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 providing best-effort service, making no special effort to ensure that RTP packets arrive at destination in timely matter.

RTP Header

- Version (V): This field identifies the version of RTP
- Padding (P): If P is set, the packet contains one or more additional padding octets
- Extension (X): If X is set, the fixed header MUST be followed by exactly one extension
- CSRC count (CC): number of CSRC identifiers that follow the fixed header
- Marker (M): interpretation of the marker is defined by a profile
- Payload Type (PT): identifies the format of the RTP payload
- Sequence Number: increments by one for each RTP **data packet** sent
- Timestamp: reflects the sampling instant of the **first octet** in the RTP data packet
- SSRC: field identifies the synchronization source
- CSRC list: 0 to 15 items. The CSRC list identifies the contributing sources for the payload contained in this packet.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

RTP Header

- Payload Type: 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 µ-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.

Payload Type	Sequence Number	Timestamp	Synchronization Source Identifier	Miscellaneous Fields

RTP Header

RTP Header (2)

- Timestamp field (32 bits 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 µs for 8 kHz sampling clock)
  - If application generates chunks of 160 byte 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 the RTP stream. Each stream in RTP session should have distinct SSRC.

Protocols for real-time interactive applications

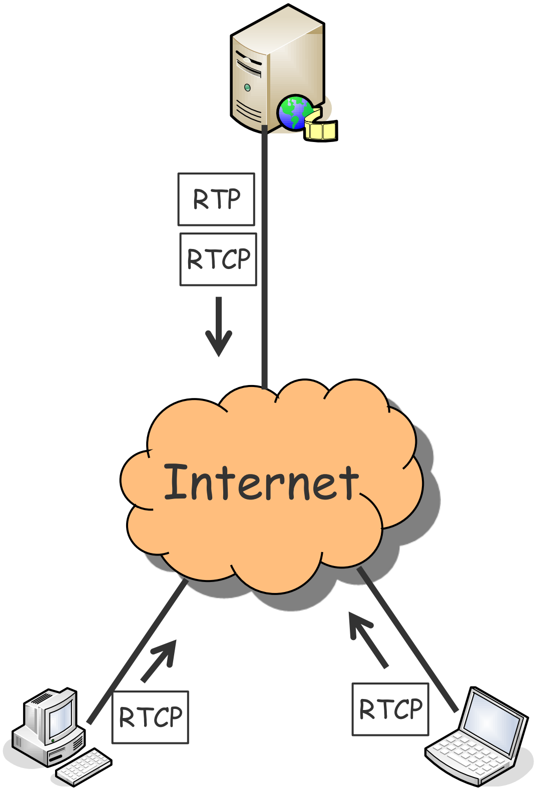
RTCP

Real-Time Control Protocol (RTCP)

- 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
  - Number of packets sent
  - Number of packets lost
  - Interarrival jitter, etc.
- Feedback can be used to control performance
  - Sender may modify its transmissions based on feedback

RTCP - Continued

- Each RTP session: typically a single multicast address
  - All RTP/RTCP packets belonging to session use multicast address.
- RTP and RTCP packets are distinguished from each other via distinct port numbers
  - RTCP Port = RTP Port + 1
- To limit traffic, each participant reduces RTCP traffic as number of conference participants increases



RTCP Packets

- Receiver report packets:
  - Fraction of packets lost,
  - Last sequence number
  - Average interarrival 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 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

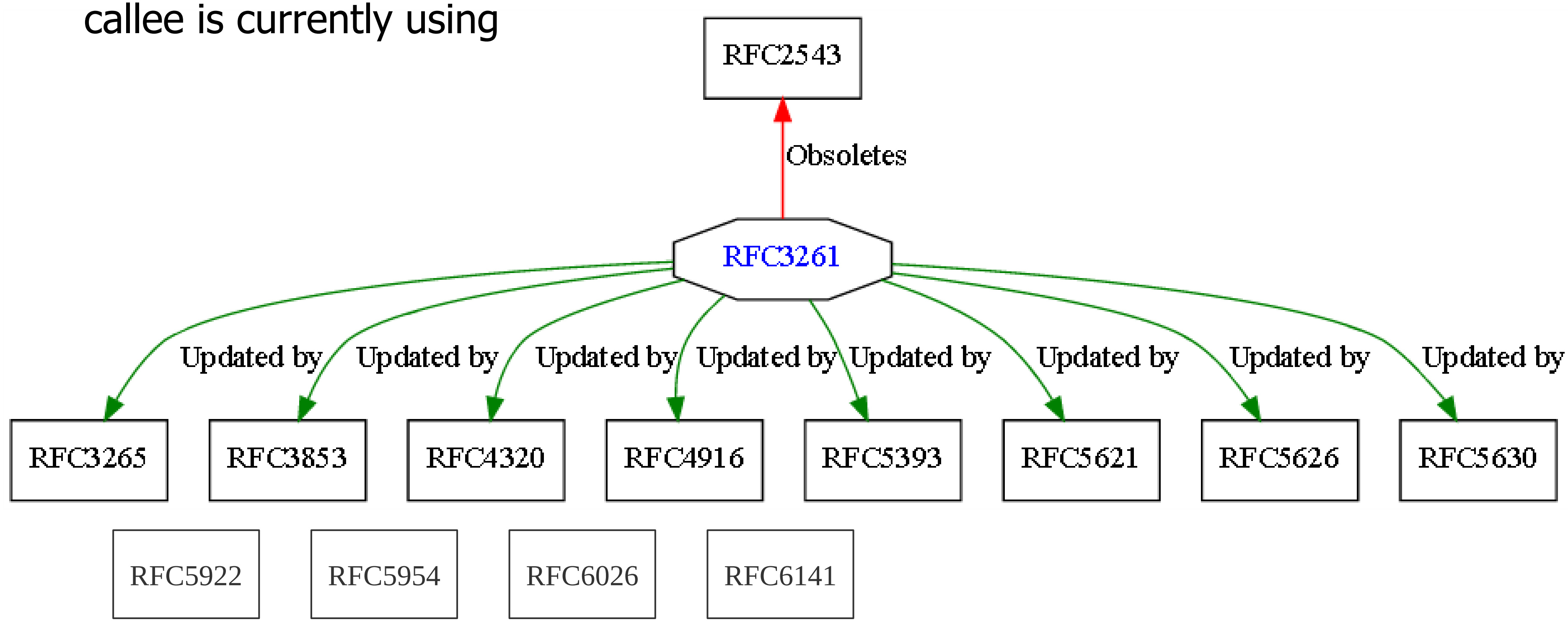
- RTCP attempts to limit its traffic to 5% of session bandwidth.
- Example
  - Suppose one sender, sending video at 2 Mbps. Then RTCP attempts to limit its traffic to 100 kbps.
  - RTCP gives 75% of rate to receivers; remaining 25% to sender
- 75 kbps is equally shared among receivers:
  - With R receivers, each receiver gets to send RTCP traffic at 75/R kbps.
- 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

Protocols for real-time interactive applications

Session Initiation Protocol (SIP)

SIP: Session Initiation Protocol [RFC 3261]

- 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
  - You can reach callee, no matter where callee roams, no matter what IP device callee is currently using

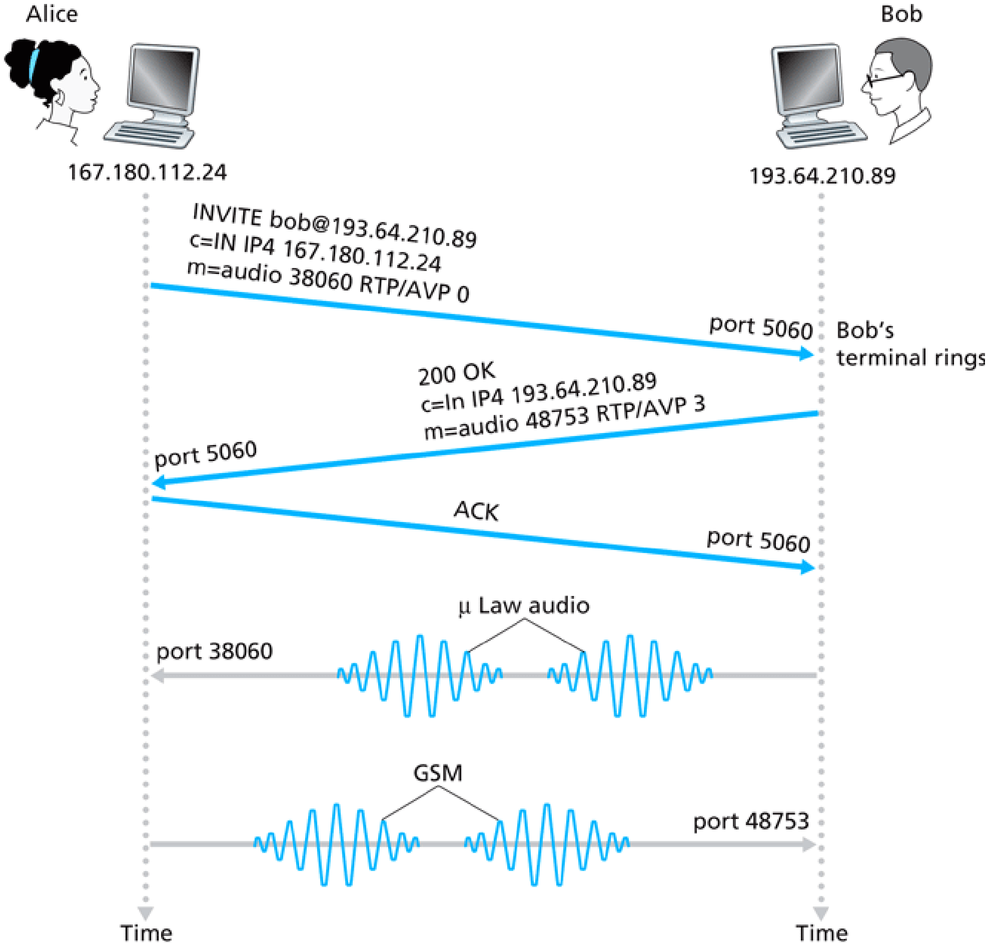


SIP Services

- Setting up a call, SIP provides mechanisms ..
  - For caller to let callee know she wants to establish a call
  - So caller and 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 others
  - Transfer, hold calls

Setting up a call to known IP address

- Alice’s SIP invite message indicates her port number, IP address, encoding she prefers to receive (PCM µlaw)
- Bob’s 200 OK message indicates his port number, IP address, preferred encoding (GSM)
- SIP messages can be sent over TCP or UDP; here sent over RTP/UDP.
- Default SIP port number is 5060.



Setting up a call (more)

- Codec negotiation:
  - Suppose bob doesn’t have PCM μlaw encoder.
  - Bob will instead reply with 606 not acceptable reply, listing his encoders Alice can then send new invite message, advertising different encoder
- Rejecting a call
  - Bob can reject with replies
    - “busy”
    - “gone”
    - “payment required”
    - “forbidden”
- Media can be sent over RTP or some other protocol

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 506
- Alice specifies in Via: header that SIP client sends, receives SIP messages over UDP

Name translation and user locataion

- Caller wants to call callee, but only has callee's name or e-mail address.
- Need 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)
- 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

Example

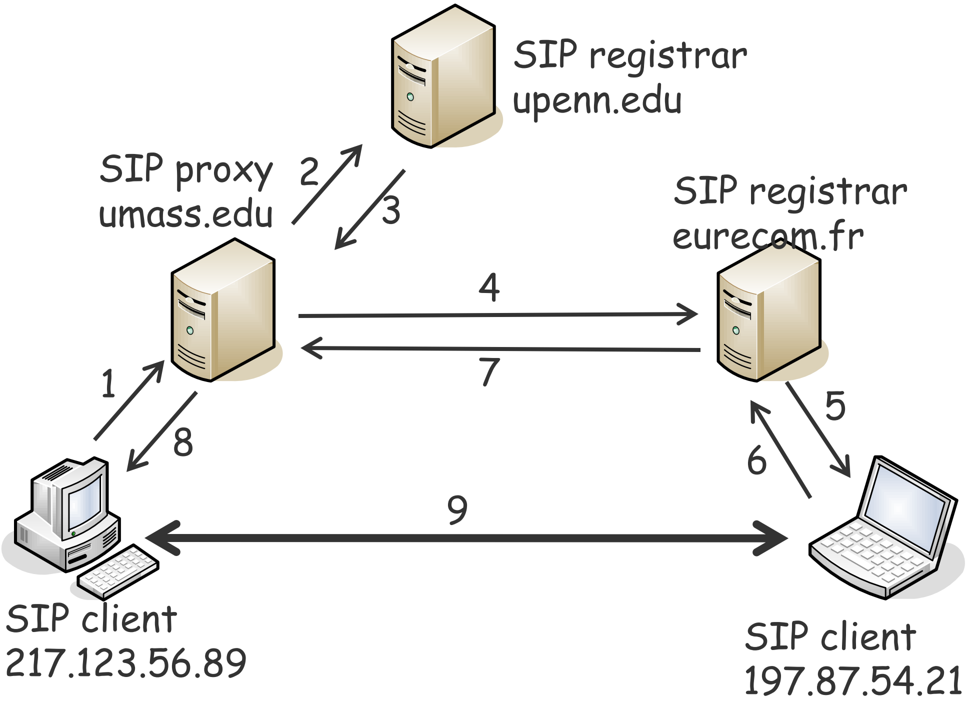
- Caller jim@umass.edu who places a call to keith@upenn.edu

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.

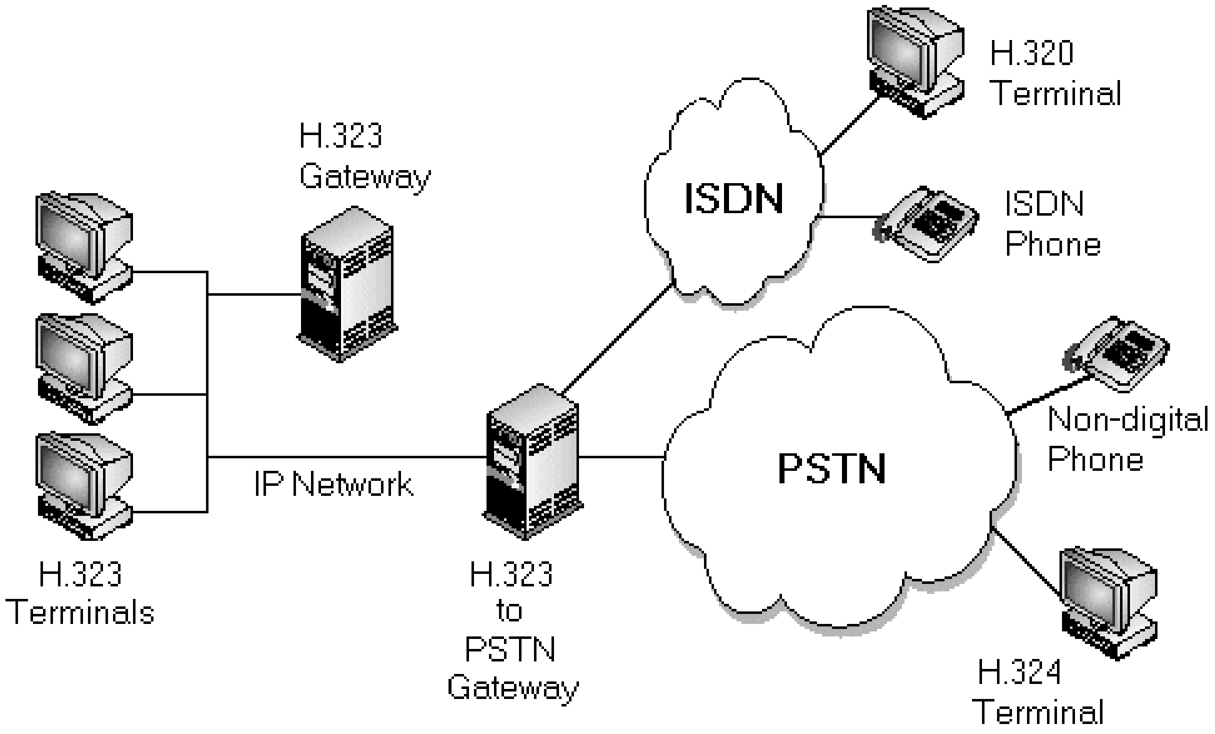
Note: there is also a SIP ack message, which is not shown.



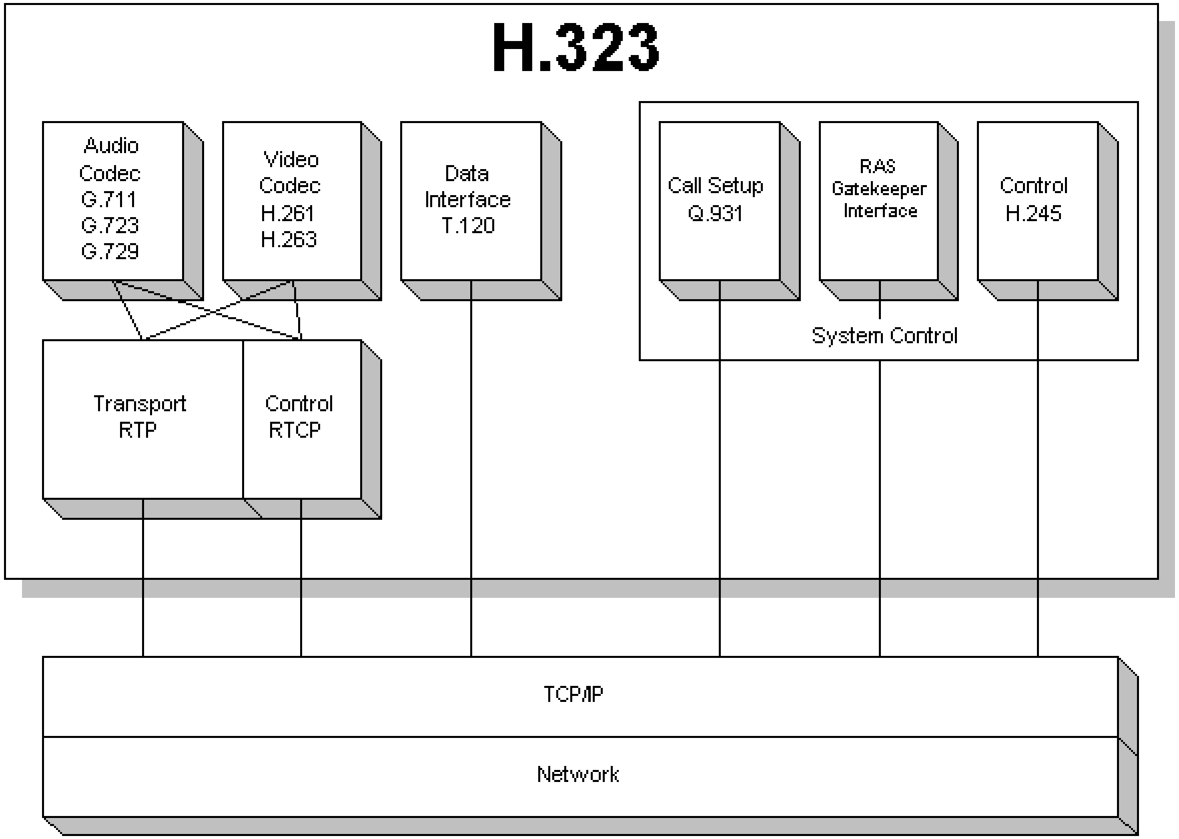
Protocols for real-time interactive applications

H.323

Alternative to SIP: H.323



Alternative to SIP: H.323



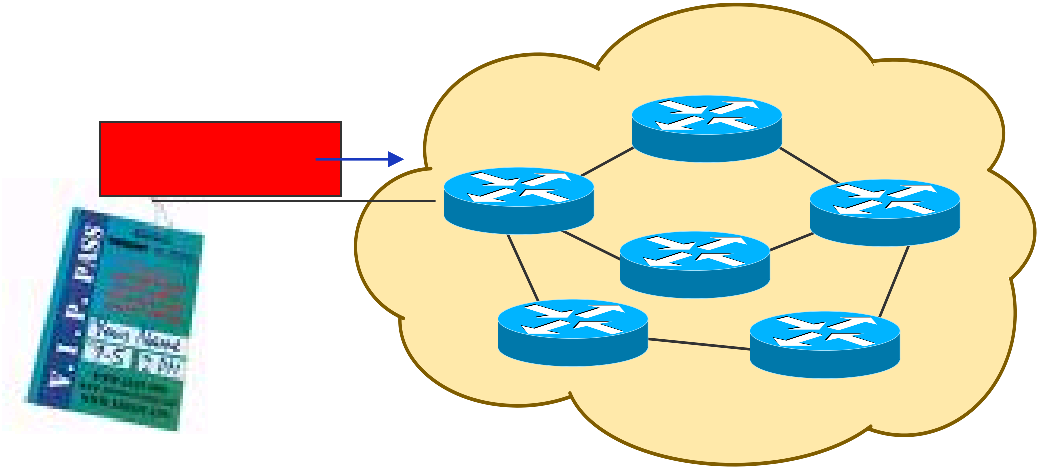
Comparison: SIP vs. H.323

- H.323 is another signaling protocol for real-time, interactive
- H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing:
  - Signaling
  - Registration
  - Admission control
  - Transport
  - Codecs
- SIP is a single component
  - Works with RTP, but does not mandate it
  - Can be combined with other protocols, services
- H.323 comes from the ITU (telephony)
- SIP comes from IETF:
  - Borrows much of its concepts from HTTP
  - SIP has Web flavor, whereas H.323 has telephony flavor
- SIP uses the KISS principle
  - Keep it simple stupid.

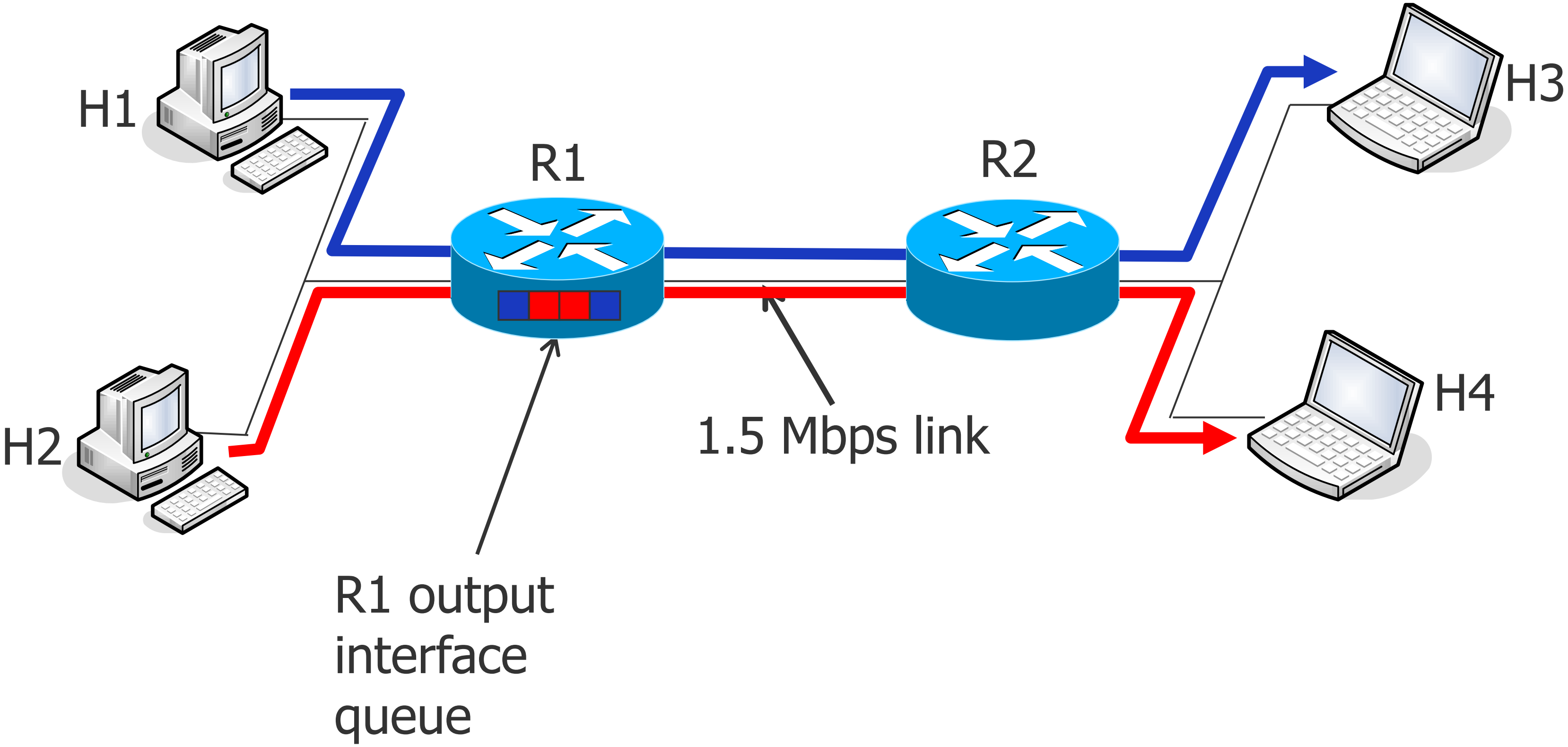
Providing multiple classes of service

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
  - Network treats different classes of traffic differently
    - Analogy: VIP service vs regular service
  - Granularity: differential service among multiple classes, not among individual connections
  - History:
    - Type of Service (TOS) bits in IPv4

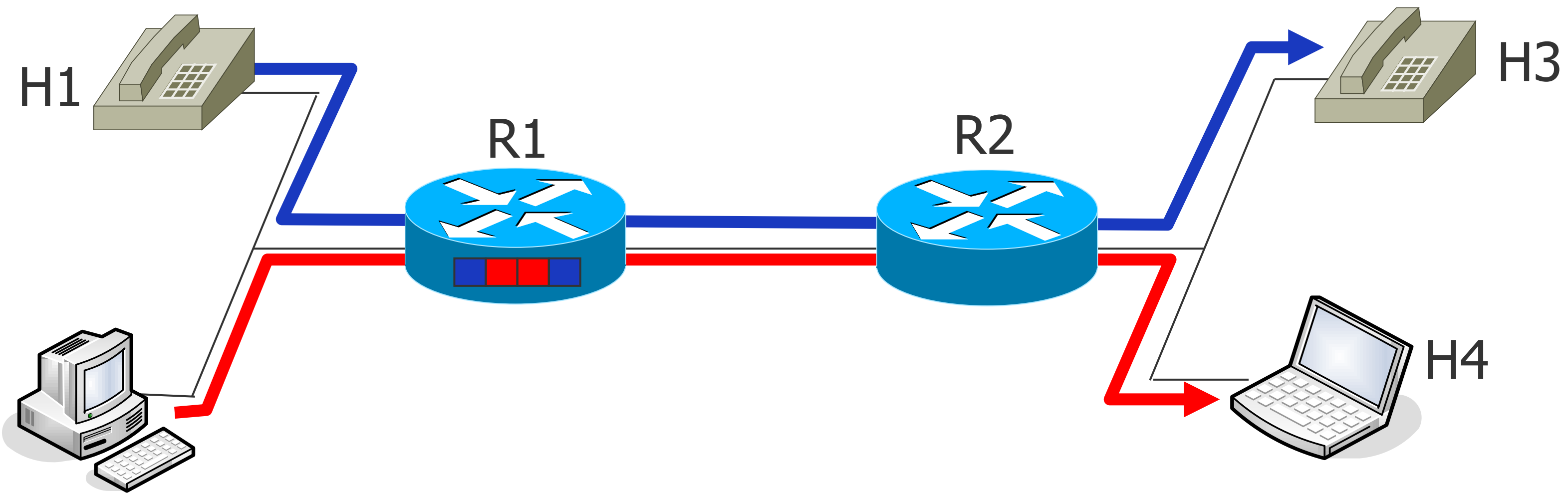


Multiple classes of service: Scenario



Scenario: Mixed FTP and Audio

- Example: 1Mbps IP phone and FTP share 1.5 Mbps link
  - Bursts of FTP can congest router ➡ cause audio loss
  - Want to give priority to audio over FTP

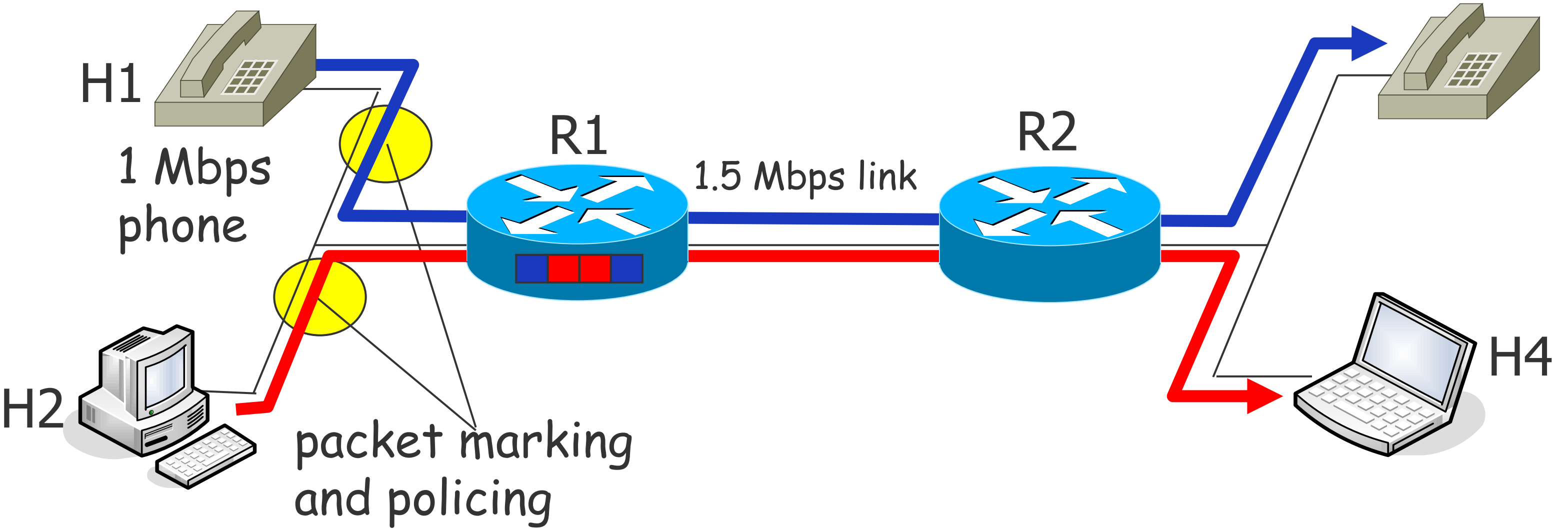


Principle 1

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

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)

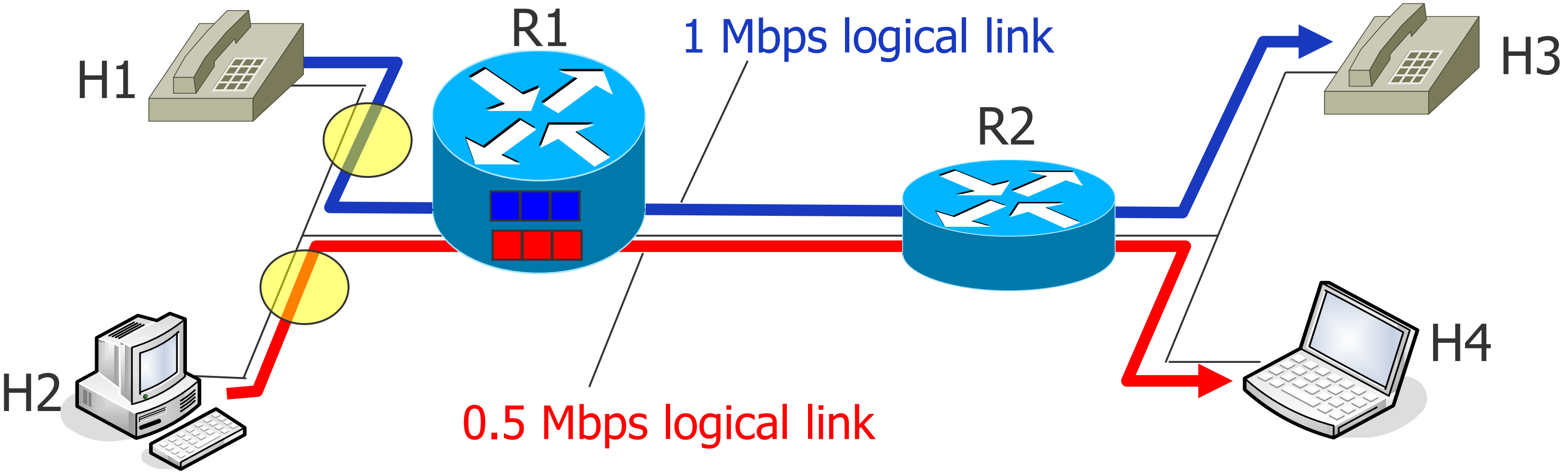


Principle 2

Provide protection (*isolation*) for one class from others

Principles for QOS Guarantees

- Allocating fixed (non-sharable) bandwidth to flow: inefficient use of bandwidth if a flow does not use its allocation



Principle 3

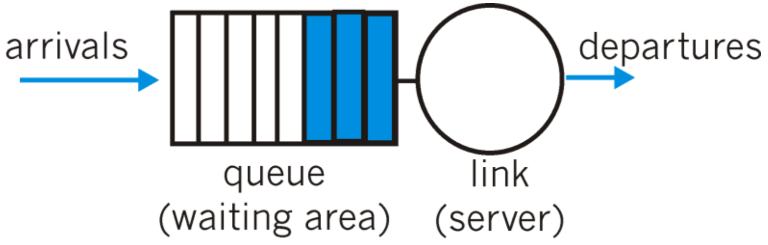
While providing isolation, it is desirable to use resources as efficiently as possible

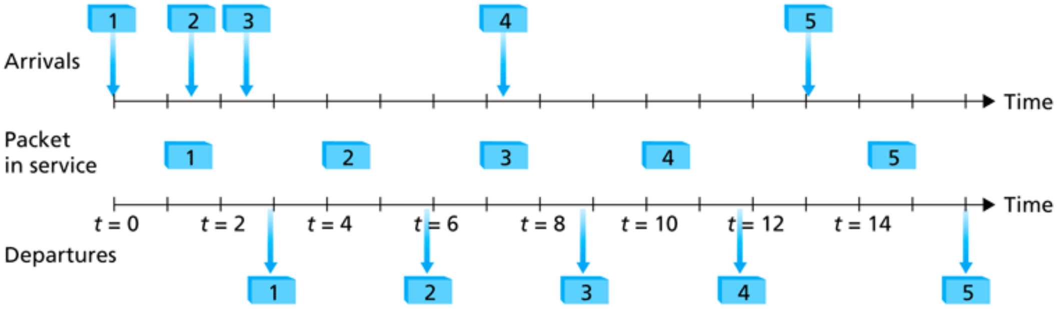
Providing multiple classes of service

Scheduling and Policing Mechanisms

Scheduling and Policing Mechanisms

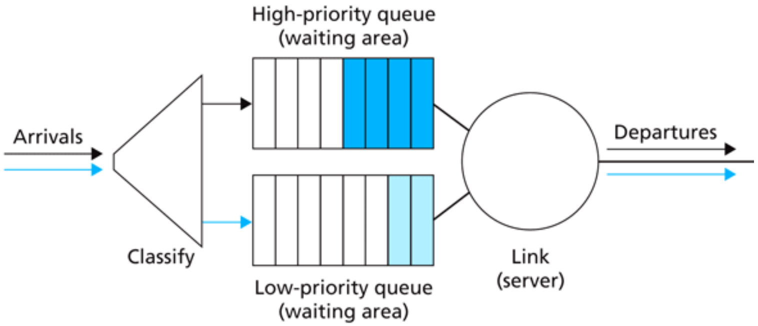
- Scheduling: choose next packet to send on link
- FIFO (first in first out) scheduling: send in order of arrival to queue
  - Real-world example?
  - Discard policy: if packet arrives to full queue: who to discard?
    - Tail drop: drop arriving packet
    - Priority: drop/remove on priority basis
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

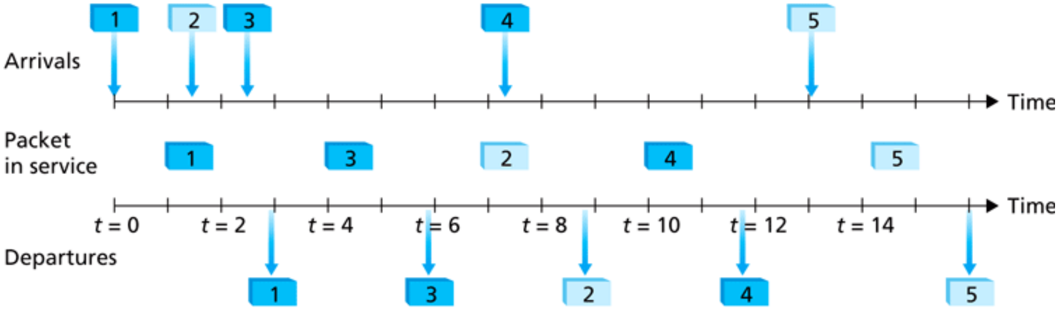




Scheduling Policies: Priority based

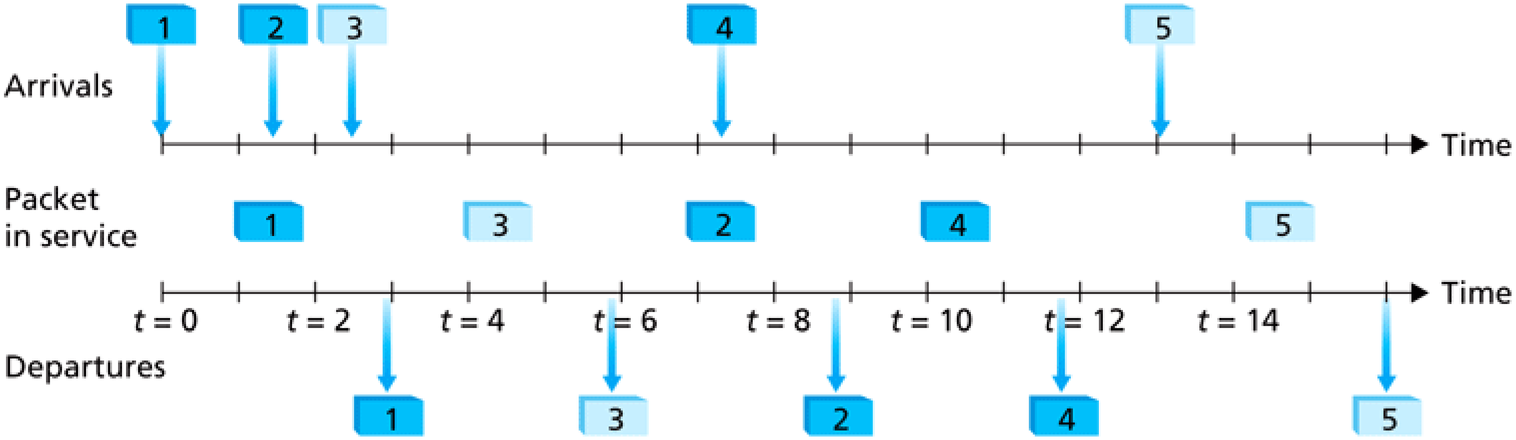
- Priority scheduling: transmit highest priority queued packet
- Multiple classes with different priorities
  - Class may depend on marking or other header info, e.g., IP source/dest, port numbers, etc.
  - Real world example?





Scheduling Policies: Round Robin

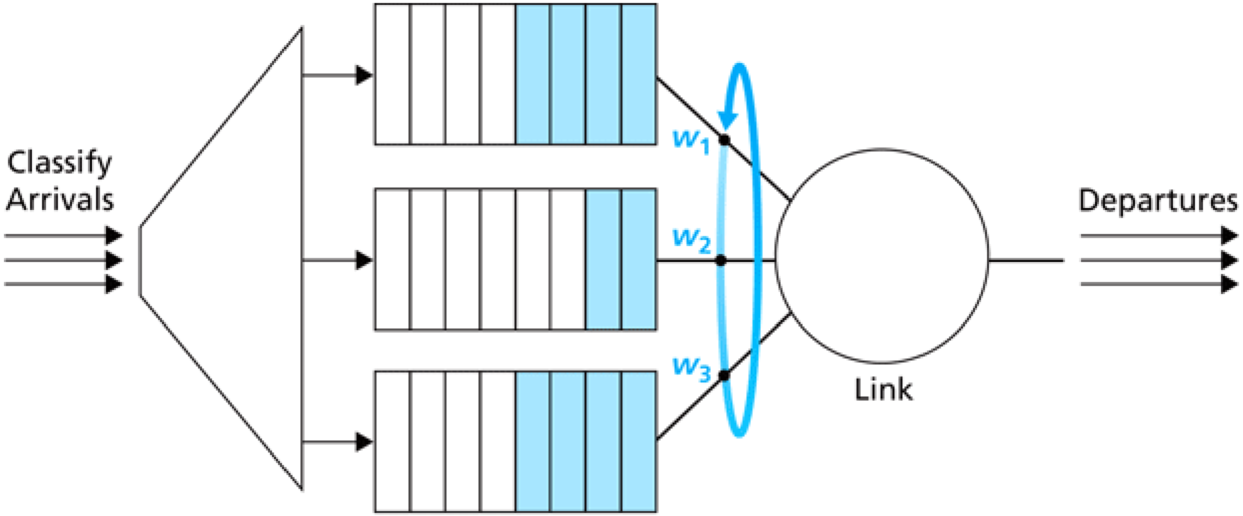
- Round robin scheduling:
  - Multiple classes
  - Cyclically scan class queues, serving one from each class (if available)
    - Class 1, Class 2, Class 1, ...
  - Real world example?



Scheduling Policies: Weighted Fair Queuing (WFQ)

- Weighted fair queuing
  - Generalized round robin
  - Each class gets weighted amount of service in each cycle
  - For a link with transmission rate R, class i will get a throughput of $R_i = R \times W_i$
  - Real-world example?

$$W_i = \frac{w_i}{\sum_{j=1}^{N} w_j}$$



Policing Mechanisms

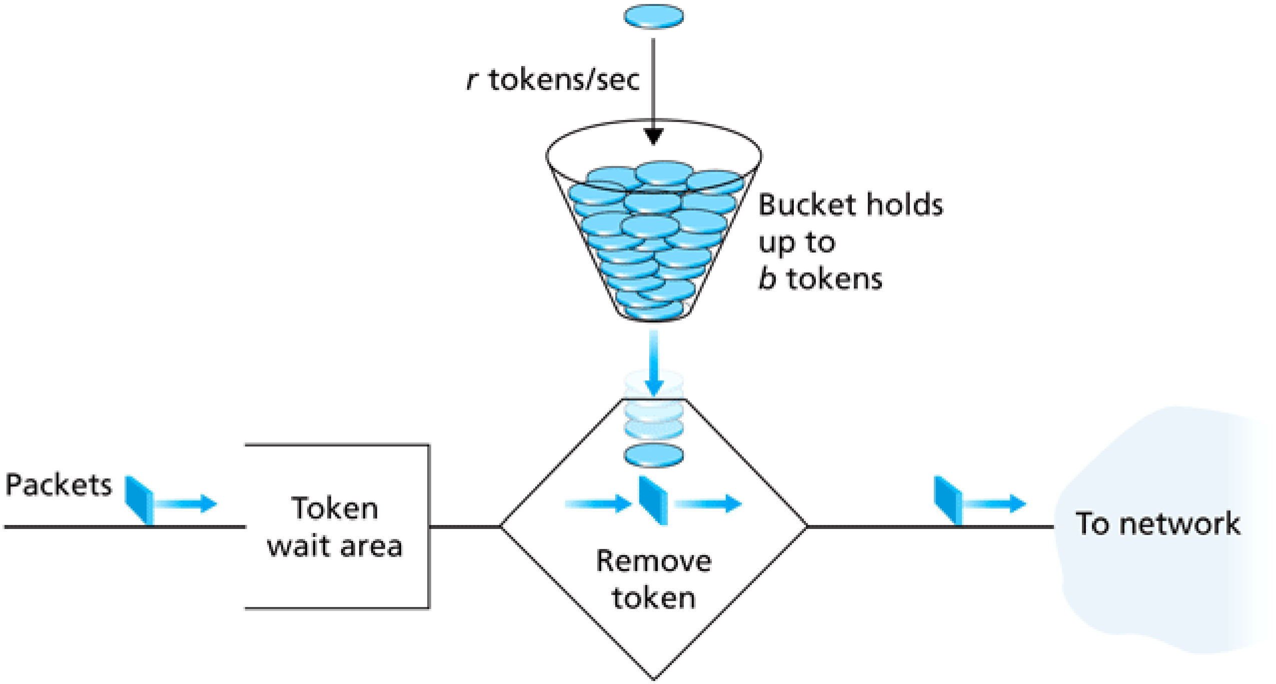
- Goal: Limit traffic to not exceed declared parameters

- Three common-used criteria:
  - Average Rate: how many packets can be sent per unit time (in the long run)
    - Long term criteria
    - Crucial question: What is the interval length?

      100 packets per sec or 6000 packets per min have same average!
  - Peak Rate: restriction of number of packets for a short time, e.g.,
    - Average rate: 6000 packets/min
    - Peak rate: 500 packets/sec
  - Burst Size: max. number of packets sent consecutively (with no intervening idle)

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 $\leq (r\ t + b)$

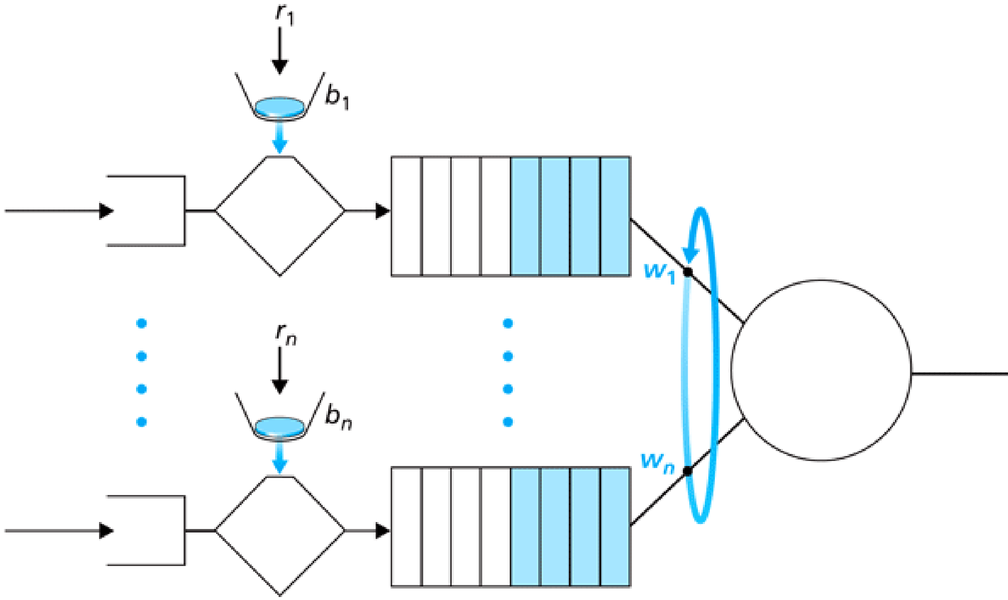
Policing Mechanisms

- Combine Token Bucket and WFQ to provide guaranteed upper bound on delay, i.e., QoS guarantee!
  - n flows with leaky buckets of parameters r_i and b_i
  - Each flow receives a share R_i of the bandwidth R

$$R_i = R \frac{w_i}{\sum_{j=1}^{N} w_j}$$

  - Maximum delay of flow i is d_{max}

$$d_{\max} = \frac{b_i}{R \frac{w_i}{\sum_{j=1}^{N} w_j}}$$



Providing multiple classes of service

Differentiated Services

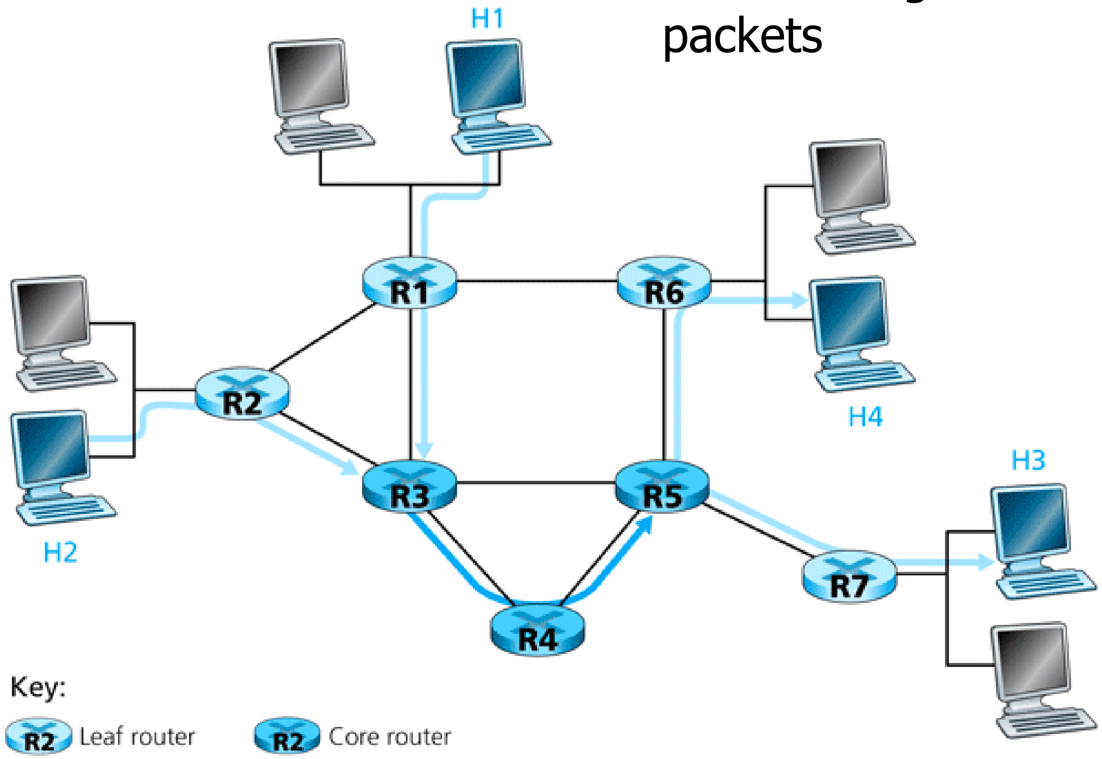
IETF Differentiated Services

- 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
- Do not define service classes, provide functional components to build service classes

Diffserv Architecture

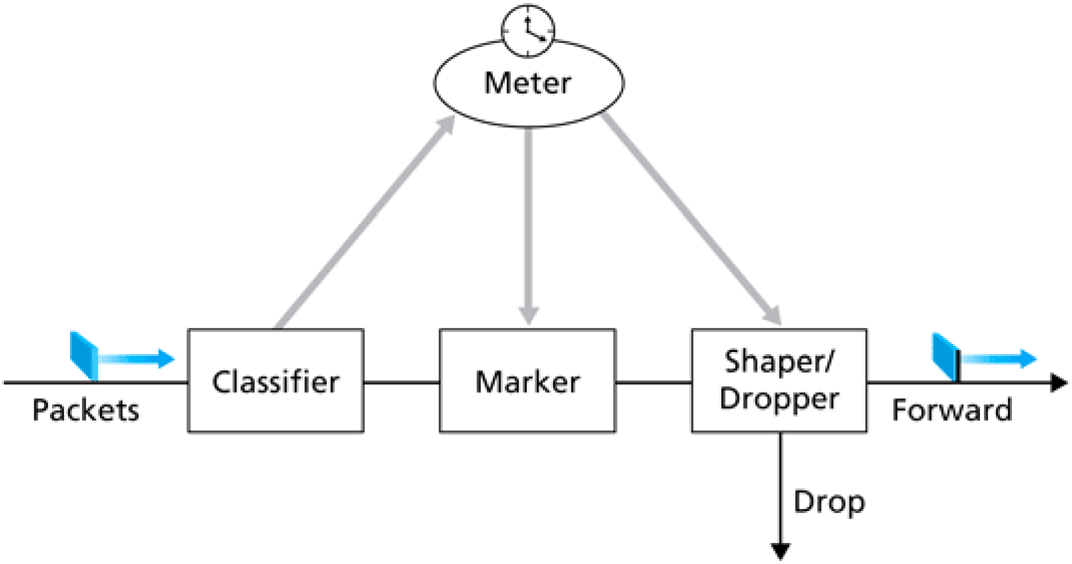
- Edge router:
  - Per-flow traffic management
  - Marks packets as in-profile and out-profile

- Core router:
  - Per class traffic management
  - Buffering and scheduling based on marking at edge
  - Preference given to in-profile packets



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



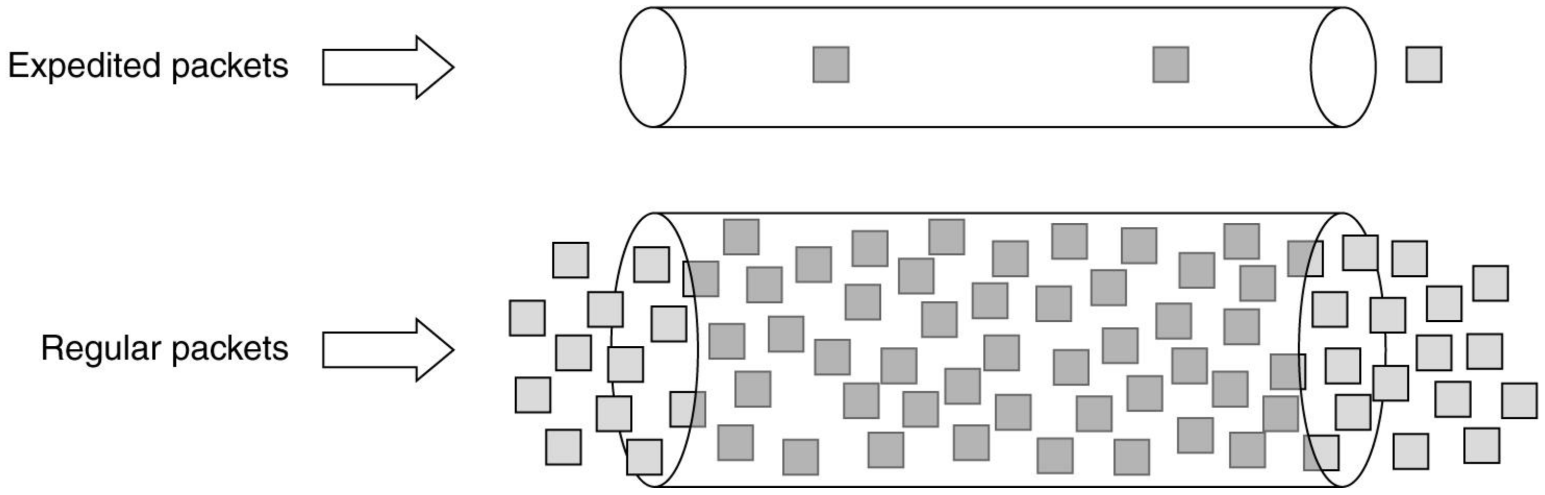
Per Hop Behavior (PHB)

- Per hop behavior (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

- Two PHBs are defined
  - Expedited forwarding behavior EF-PHB
  - Assured forwarding behavior AF-PHB

Expedited Forwarding

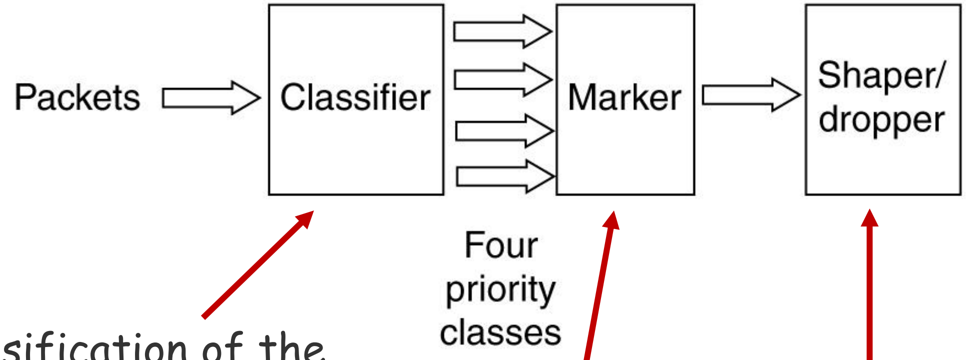
- Idea with Expedited Forwarding:
  - There are “regular” and “expedited” packets.
  - Expedited packets are forwarded to guarantee a minimum data transmission rate
  - Routers can manage two separated queues for these packet types (Weighted Fair Queuing).
  - Possible: low loss rate/delay/jitter as well as a guaranteed data transmission rate

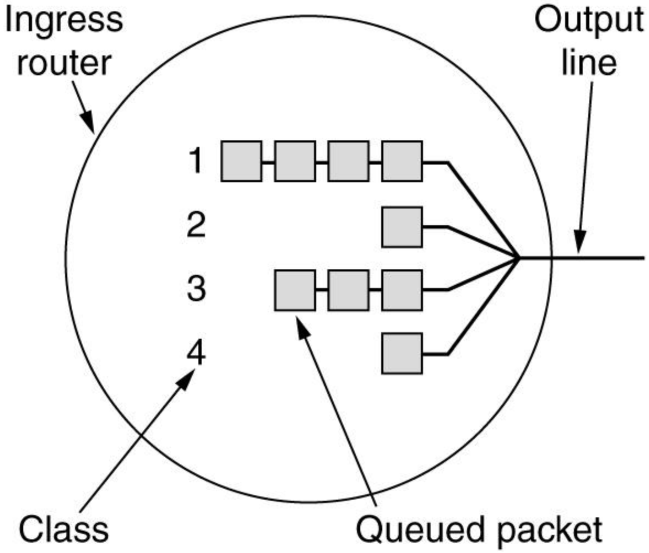


Assured Forwarding

- Improves differentiation: Definition of 4 priority classes with own resources (at the moment; there also is room for more classes)
- For each class, 3 drop probabilities are defined: low, medium, high
- Thus: altogether 12 different service classes
- Principle:
  - The priority class determines the portion of the transmission capacity of the routers
  - During high load, packets of lower priority would be discarded completely
  - Fairness: packets of each priority class should have chances to survive
  - Therefore definition of the probabilities for each class: by suitable selection of the probabilities, a small part of the lowest priority level still is still forwarded, while packets of higher priority classes are already discarded.

Assured Forwarding





1. Classification of the packets in service classes

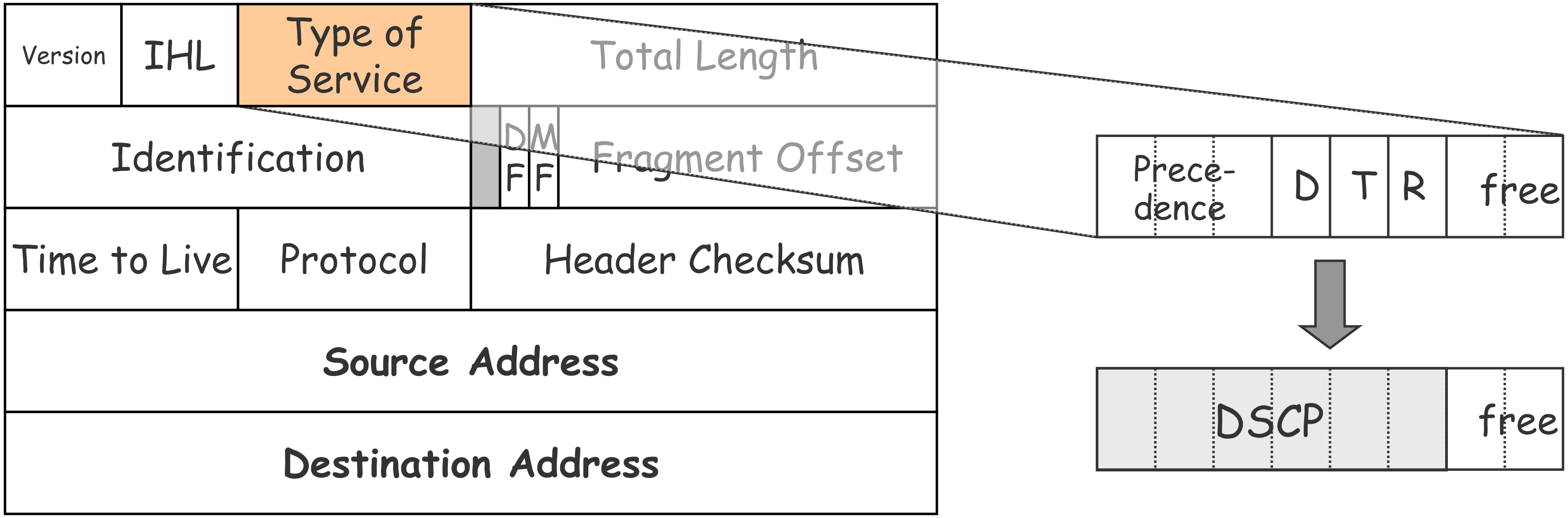
2. Appropriate choice of DSCP tags

3. Bring the data streams in a form according to their flow specification. Exceeding the specifications leads to discarding data. If thereafter still too much data are present, discarding of packets in accordance with their probability

4. Weighted Fair Queuing in accordance to priority classes

Application of DiffServ with IP

- Use of the Type of Service field in IPv4 for the classification (DSCP – Differentiated Service Code Point). The DSCP value defines the per-hop behavior of the packet from one router to the next one.



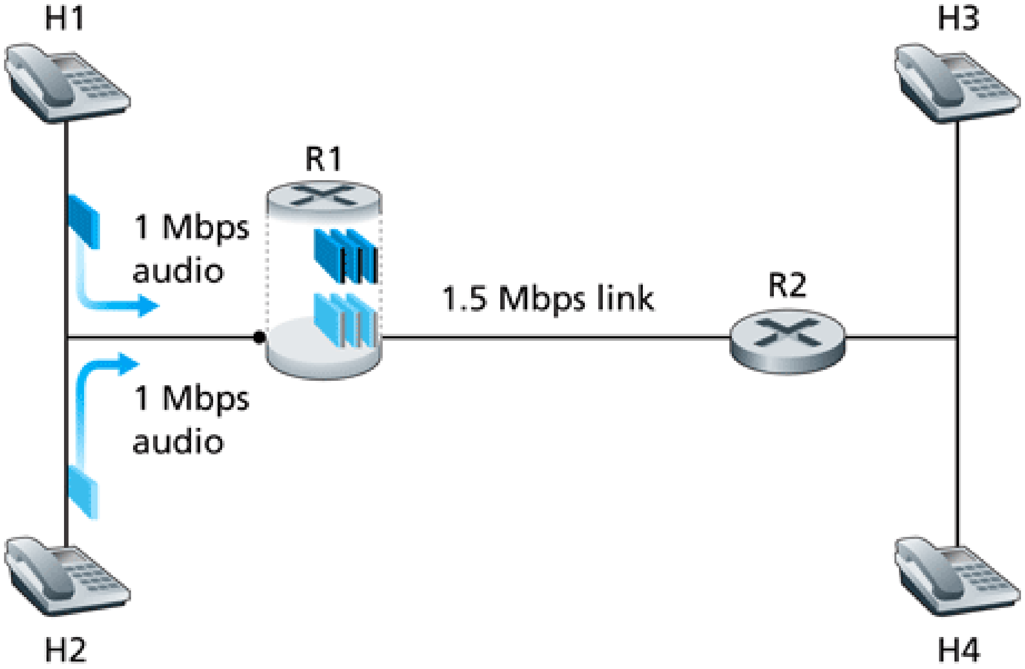
The code point defines the transmission class, which informs a router, how it has to treat the packet in forwarding.

Providing QoS guarantees

Principles for QoS Guarantees

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

- Required for QoS
  - Resource reservation
  - Call admission

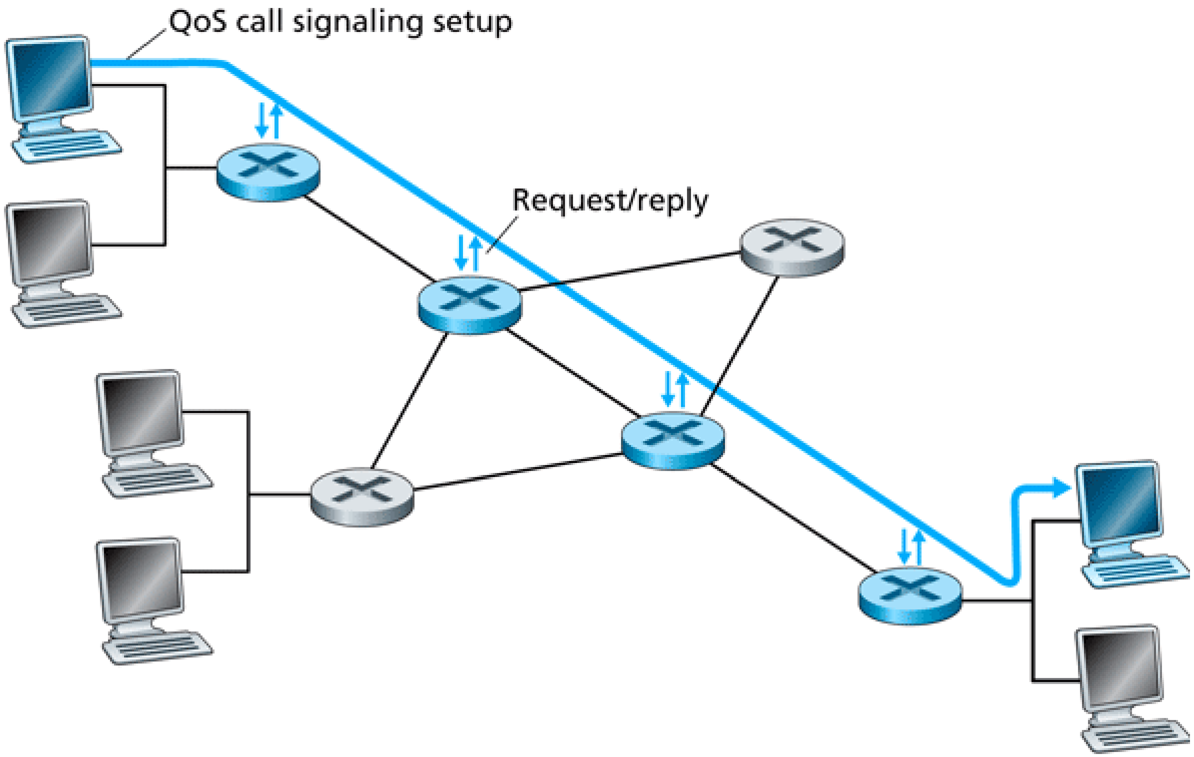


Principle 4

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

Principles for QoS Guarantees: Call Admission

- Call admission
  - Traffic characterization and specification of the desired QoS
  - Signaling for call setup
  - Per-element call admission



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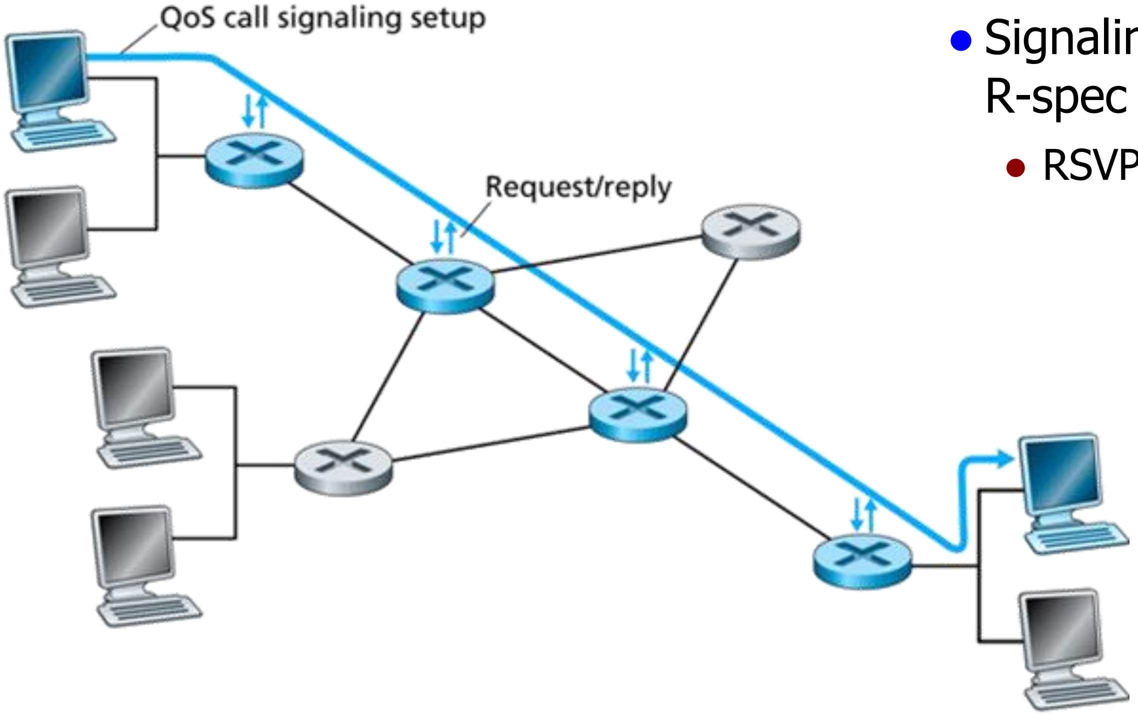


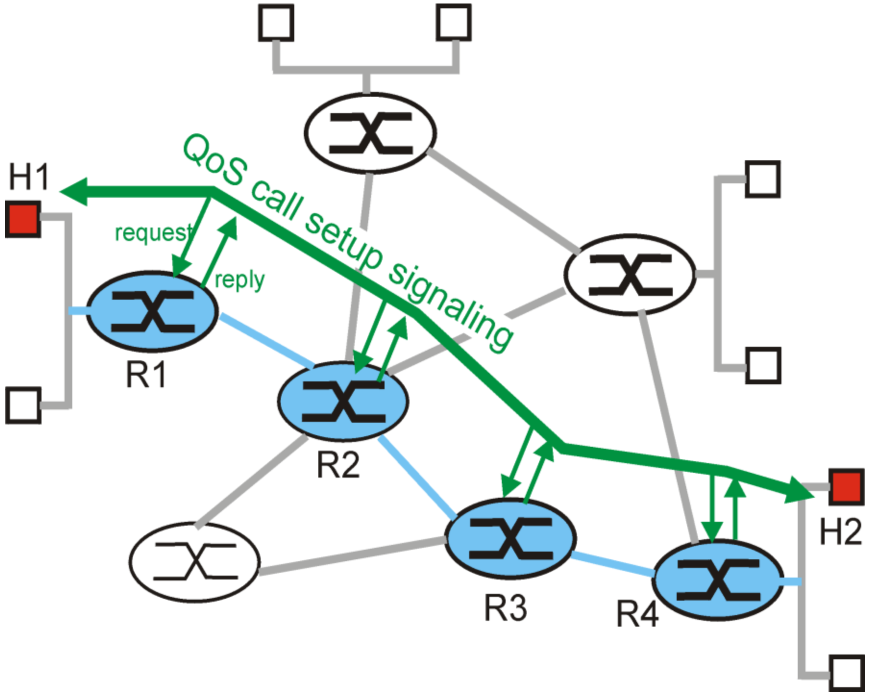
- Arriving session must
  - declare its QoS requirement
    - R-spec: defines the QoS being requested (RFC 2215)
  - Characterize traffic it will send
    - T-spec: defines traffic characteristics
  - Signaling protocol: needed to carry R-spec and T-spec to routers
    - RSVP

Providing multiple classes of service

Integrated Services

QoS in the Internet: Integrated Services

- 1995-1997 standardized by the IETF as an architecture for the transmission of multimedia streams
- Integrated Services (IntServ) uses RSVP for signaling of flow specifications as well as reservations and provides QoS for each data flow individually



- Thus the principle adds a connection-orientation to IP

Integrated Services

- Classes of QoS
  - Guaranteed: Data rate, delay and reliability are guaranteed. Deviations do not occur, packets are not discarded. Suitable for intolerant real-time applications.
  - Controlled Load: “weak” guarantees, small deviations are possible.
  - Principle: for data streams of this class the network behaves as best effort for an unloaded network. No guarantees are given, suitable for tolerant real-time applications.
  - Best Effort: as good as possible, normal Internet traffic.
- Problems
  - Scalability: for each data flow a router has to maintain own flow specifications
  - How can authorization and priority be treated for a reservation request?
  - The QoS classes are not sufficient to differentiate reasonably between different types of data streams.
- Possible: use IntServ only “at the edge” of large networks, where only few data flows are present

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]

RSVP Design Goals

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

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

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

Guaranteeing Delay: Resource Reservation

- Establish a path through the network and reserve fixed amounts of network capacity, buffer capacity and CPU time with the routers.
- Standardized by the IETF as Resource Reservation Protocol (RSVP)

- RSVP is **not** a routing protocol, but only an “addition” to such one.
  - A description of the requirements of the receiver in form of a flow specification (Level of the QoS) is needed.

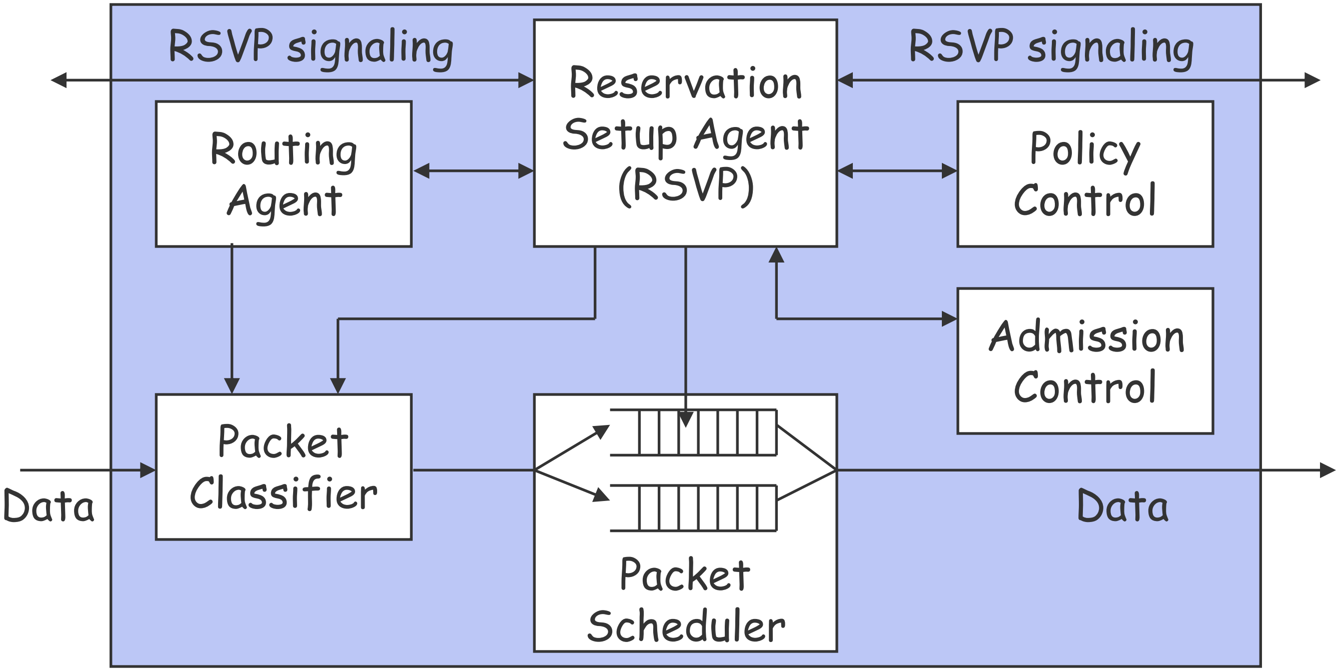
- Categories of flow specification are
  1. Best Effort: Take all free resources you can get
  2. Rate Sensitive: a guaranteed (minimal) data transmission rate is necessary
  3. Delay Sensitive: a guaranteed (maximal) delay is necessary

- RSVP can be used for Unicast and for Multicast

RSVP

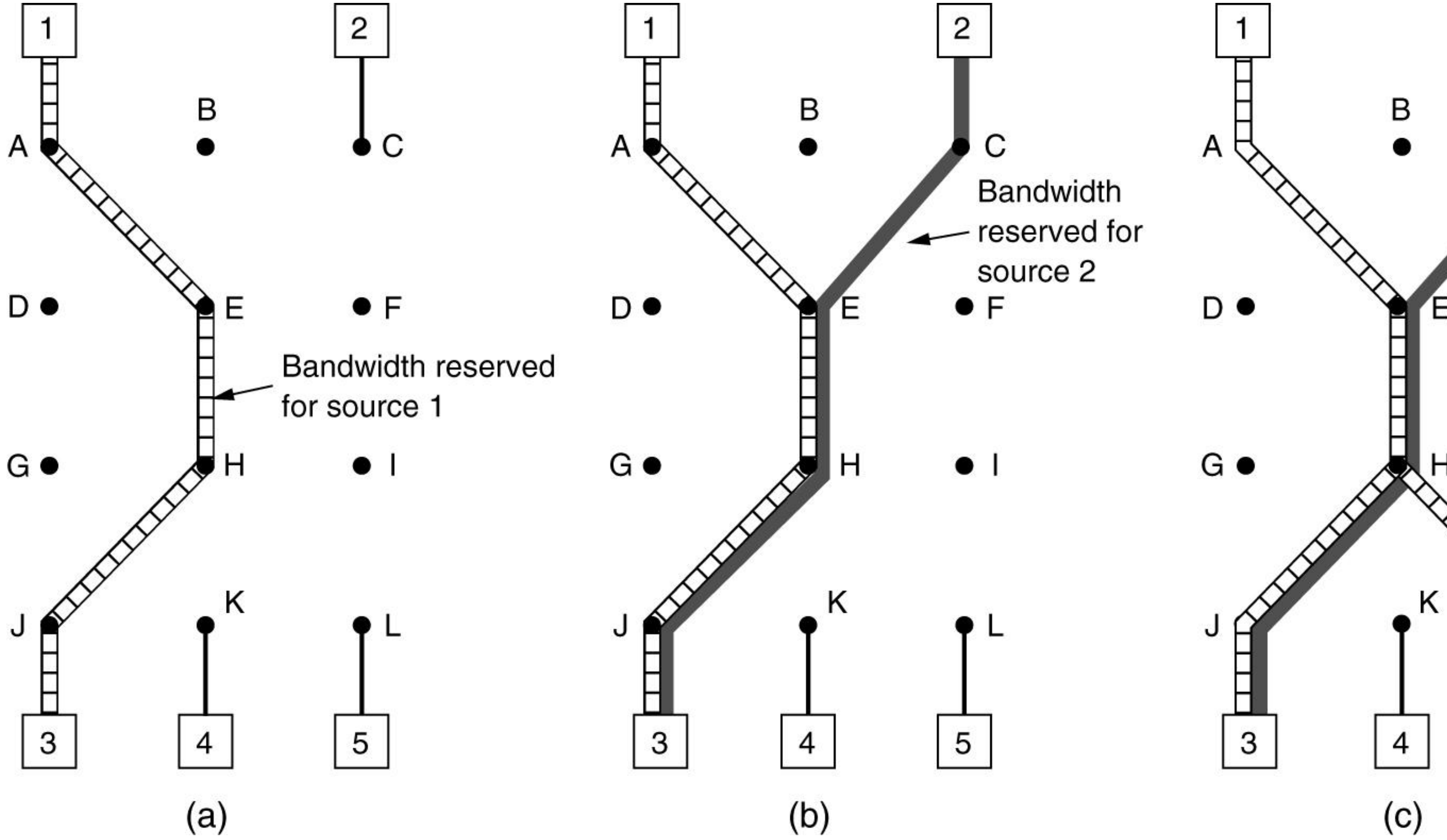
- Procedure of RSVP
  - The sender sends a RSVP Path Message to the receiver. Information about the routers on the network path are being collected and communicated to the receiver.
    - Additionally, also flow specifications for the receiver are provided.
  - The receiver sends back RSVP Reservation Messages along the collected path, which contain its flow specification. Every router on the path reserves resources accordingly.
  - If a reservation is not possible with a router, an error message is sent back.
  - As soon as the RSVP Reservation Message reaches the sender, the complete path is reserved, the sender begins to send its data.
  - To delete outdated reservations, timeouts are defined. If the receiver does not refresh its reservation before expiration of the timer, it is deleted.

IntServ Router



- *Packet Classifier*: classifies incoming packets regarding QoS class
- *Packet Scheduler*: sorts packets regarding basing on QoS class
- *Reservation Setup Agent*: configures Packet Classifier and Scheduler for new connections
- *Routing Agent*: instructs classifier about routes for data streams
- *Policy Control*: checks the permission for resource reservations
- *Admission Control*: decides if a new reservation can be accepted based on resources already used

RSVP with Multicast


1
2
A
B
C
D
E
F
G
H
I
J
K
L
3
4
5
Bandwidth reserved for source 1
Bandwidth reserved for source 2
(a)
(b)
(c)

(a) A path from receiver 3 to sender 1

(b) A second path for sender 2, because both data streams are independent

(c) For multicast, already existing reservations can be used

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