浙江大学计算机学院
数字媒体与网络技术

Digital Asset Management
数字媒体资源管理

4. Streaming multimedia

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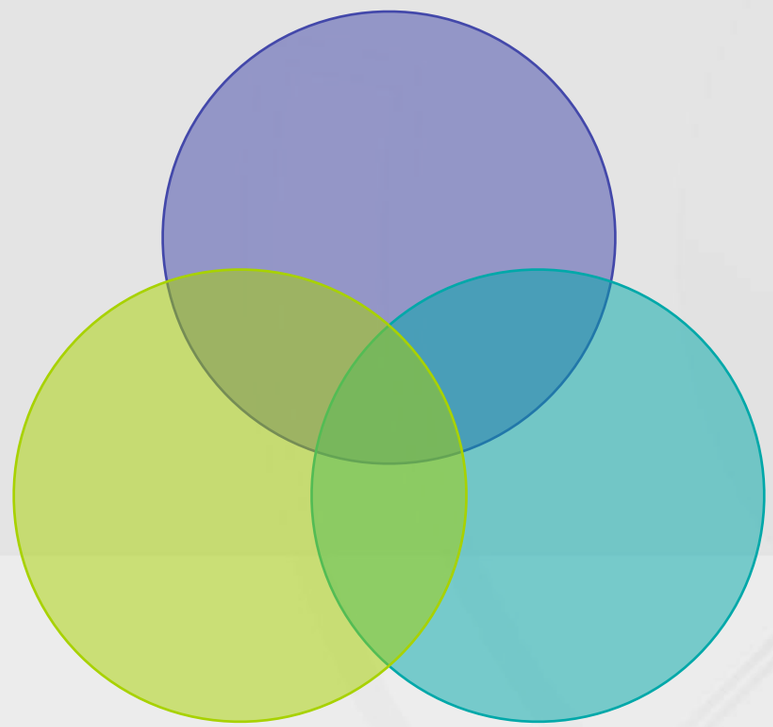
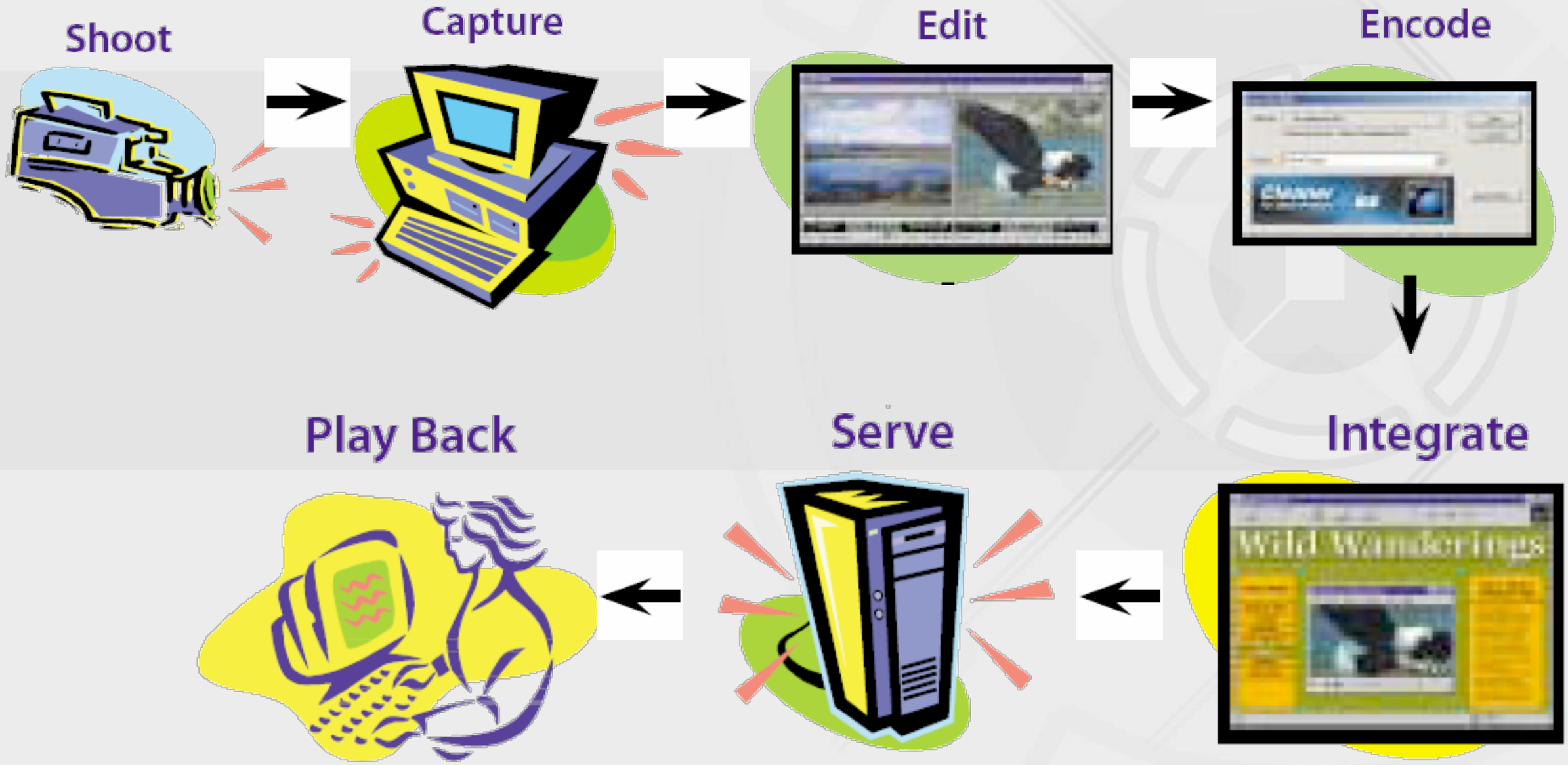
Keys of Streaming Media

- Algorithms (**)
- Standards (*****)
- Complete End-to-End systems (***)
- Research Frontiers(*)

Ubiquitous Multimedia



Workflow of Streaming Media



The primary characteristics of “streaming media”

- Three primary characteristics combine to define streaming media
  - Streaming media technology enables real-time or on-demand access to multimedia content via the Internet or an intranet.
  - Streaming media is transmitted by a media server application, and is processed and played back by a client player application, as it is received.
  - A streamed file is received, processed, and played simultaneously and immediately, leaving behind no residual copy of the content on the receiving device.

HOW DOES STREAMING WORK?

1 *The end-user selects a media file, causing a request to be sent to the Web server.*

Web Server

2 *The Web server relays the request to the media server.*

Media Server

Client

3 *The Media server streams the media file back to the end-user's computer.*

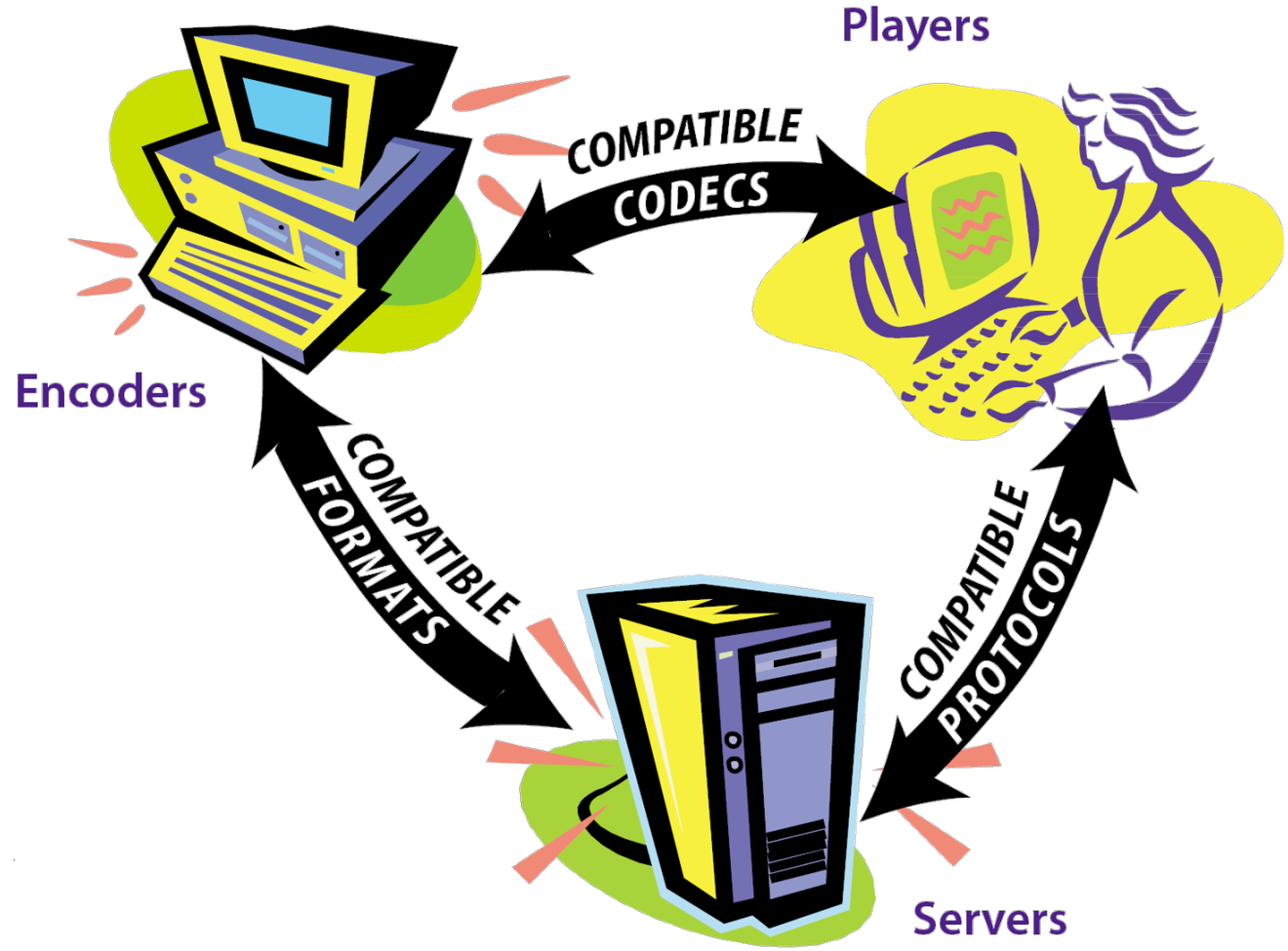
4 *The client, or "player," decodes and plays back the media file.*

WHERE DO STREAMS COME FROM?

- Streaming media architectures.
  - Streaming media architectures are comprised of
    - encoding and transmission methods,
    - server software, and
    - players (client software).
  - The three most popular streaming media architectures
    - RealMedia,
    - Windows Media, and
    - QuickTime.

It is all interrelated

- In a streaming architecture, everything must be compatible.



Streaming media formats

Architecture	Native Formats	Streaming Media File Extensions
QuickTime	QuickTime Format	.mov (sometimes .qt or .qti)
RealMedia	RealMedia Format	.rm
Windows Media	Advanced Streaming Format or Windows Media Video/Audio	.asf, .wmv, .wma

- **MPEG standard**
  - Windows Media Video v1 is a derivative of the MPEG-4 codec, which has been renamed to avoid confusion.
  - QuickTime 5 is the first full implementation of MPEG-4 for streaming media.

Streaming ...

- Progressive streaming transport (PST)
  - –use HTTP
  - –no jump

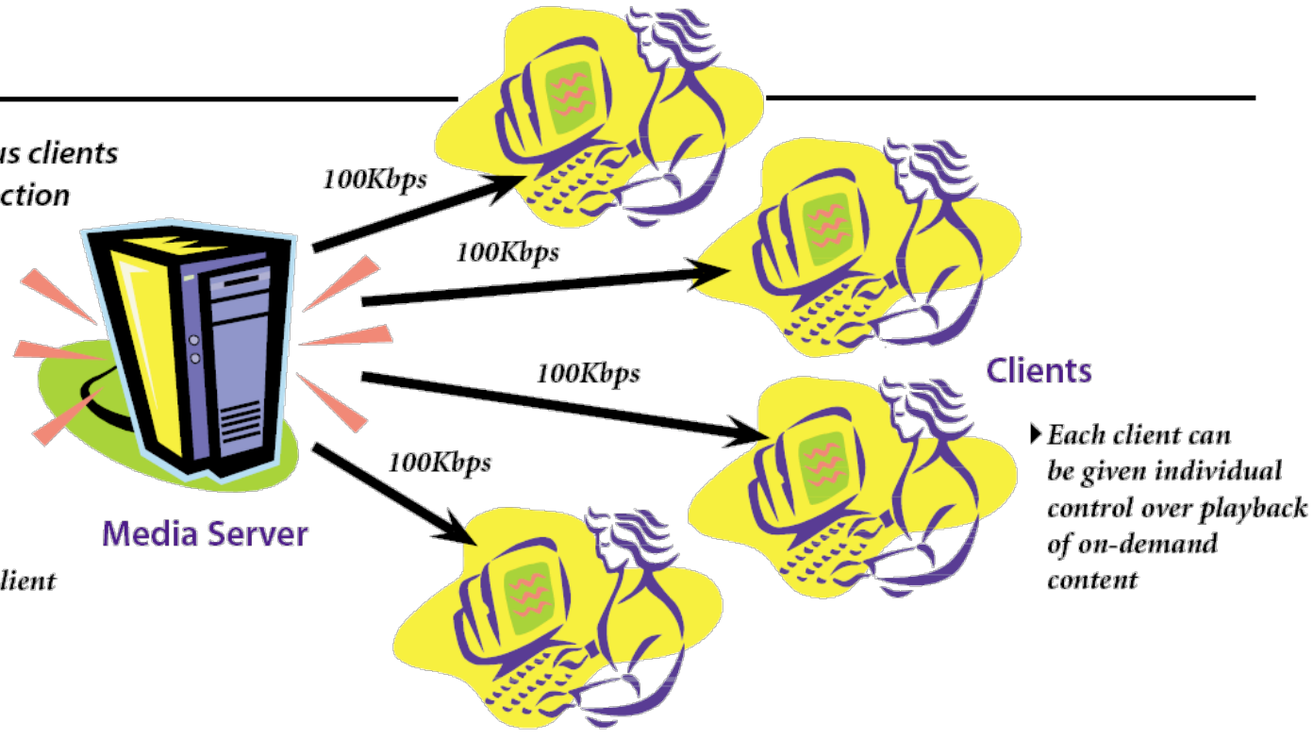
- Real-time streaming transport
  - –Real server (Real-time streaming protocol, RTSP)
  - –Windows Media server (M$ media server, MMS)
  - –Quicktime server

Unicast and Multicast

Unicasting

4 x 100Kbps simultaneous clients requires 400Kbps connection from server

- *Best choice for on-demand media*
- *Each client gets a different stream, even if they are watching the same movie*
- *Heavier load (CPU and bandwidth) on server per client*

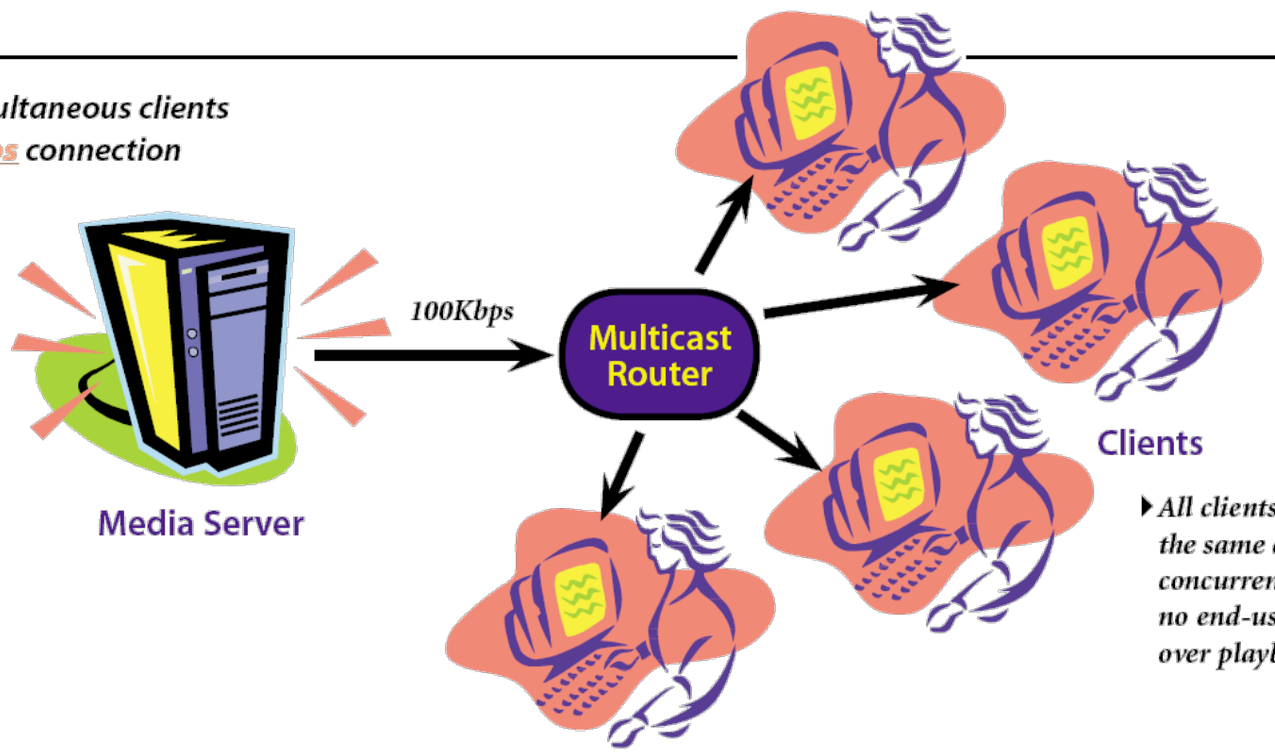


- *Each client can be given individual control over playback of on-demand content*

Multicasting

4 x 100Kbps simultaneous clients requires 100Kbps connection from server

- *Best for live or scheduled media*
- *Each client gets the same stream*
- *Conserves CPU processing power and bandwidth at server*



- *All clients playback the same content concurrently, with no end-user control over playback*

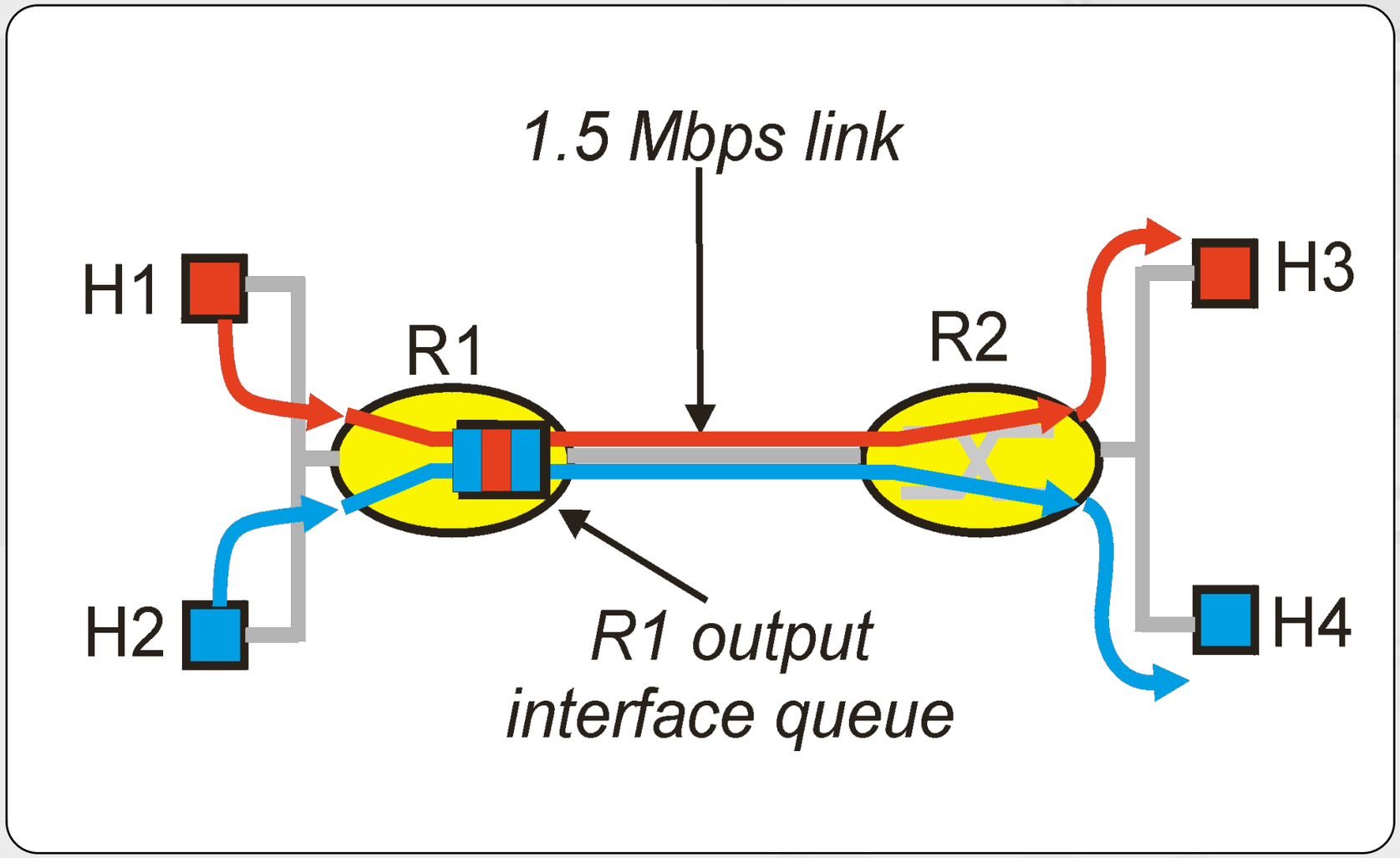
QoS: Quality of Service

- A defined measure of performance in a data communications system

- resource reservation control mechanisms
  - make the actual determination of which packets have priority

1. provide different priority to different users or data flows, or
2. guarantee a certain level of performance to a data flow in accordance with requests from the application program

QoS: a simple example

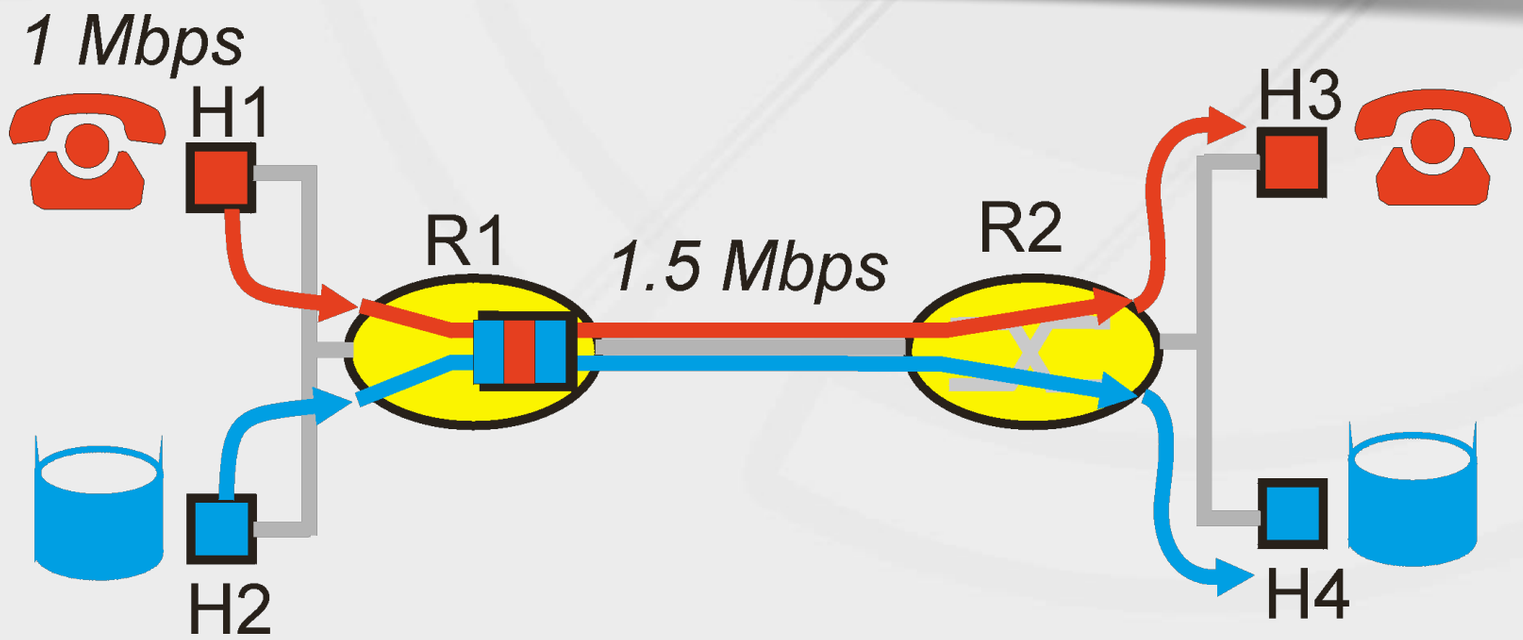


Improving QOS in IP Networks

Principles for QOS Guarantees

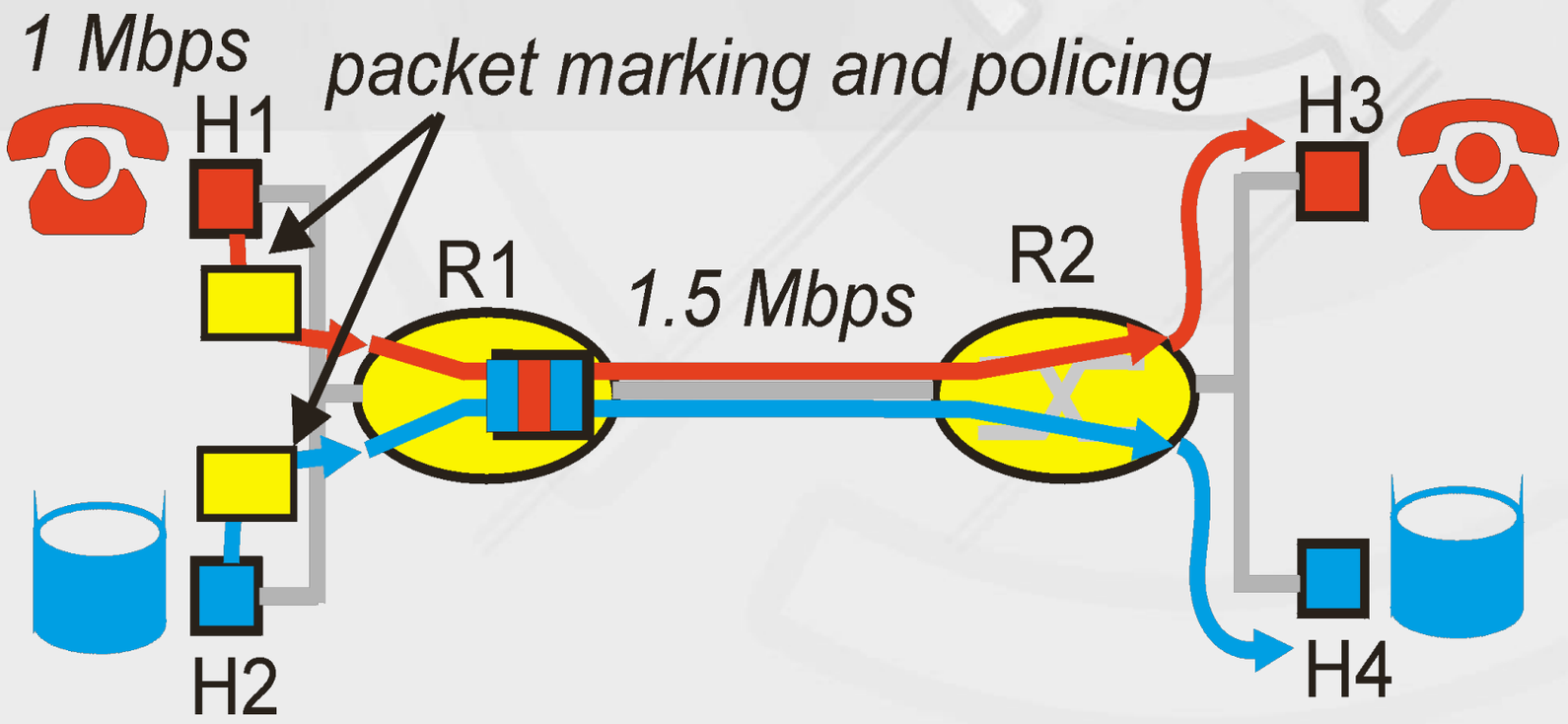
- **Consider:**
  - a phone application at 1Mbps and
  - an FTP application sharing a 1.5 Mbps link.
    - bursts of FTP can congest the router and cause audio packets to be dropped.
    - want to give priority to audio over FTP

- **PRINCIPLE 1: Marking of packets is needed for router to distinguish between different classes; and new router policy to treat packets accordingly**



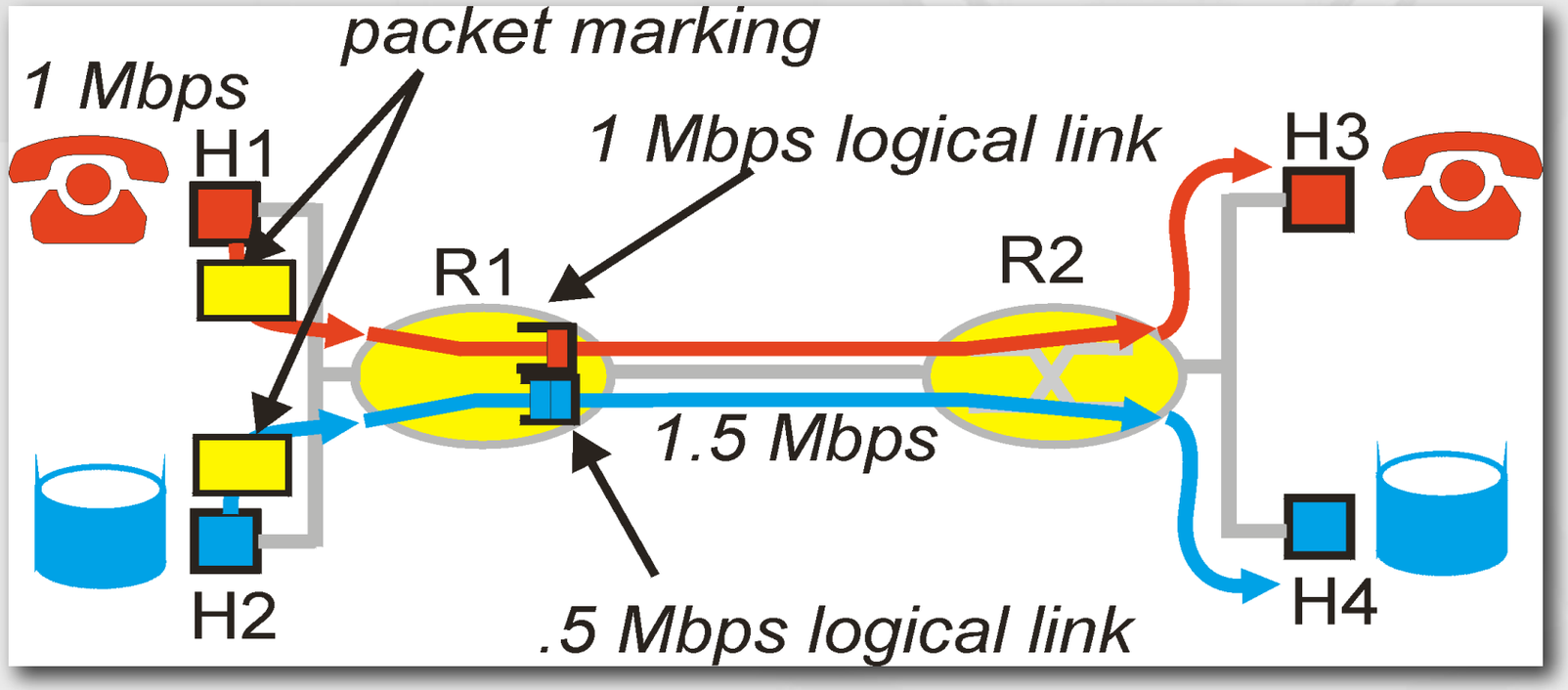
Principles for QOS Guarantees (more)

- Applications misbehave (audio sends packets at a rate higher than 1Mbps assumed above);
- **PRINCIPLE 2: provide protection (isolation) for one class from other classes**
- Require Policing Mechanisms to ensure sources adhere to bandwidth requirements; Marking and Policing need to be done at the edges:



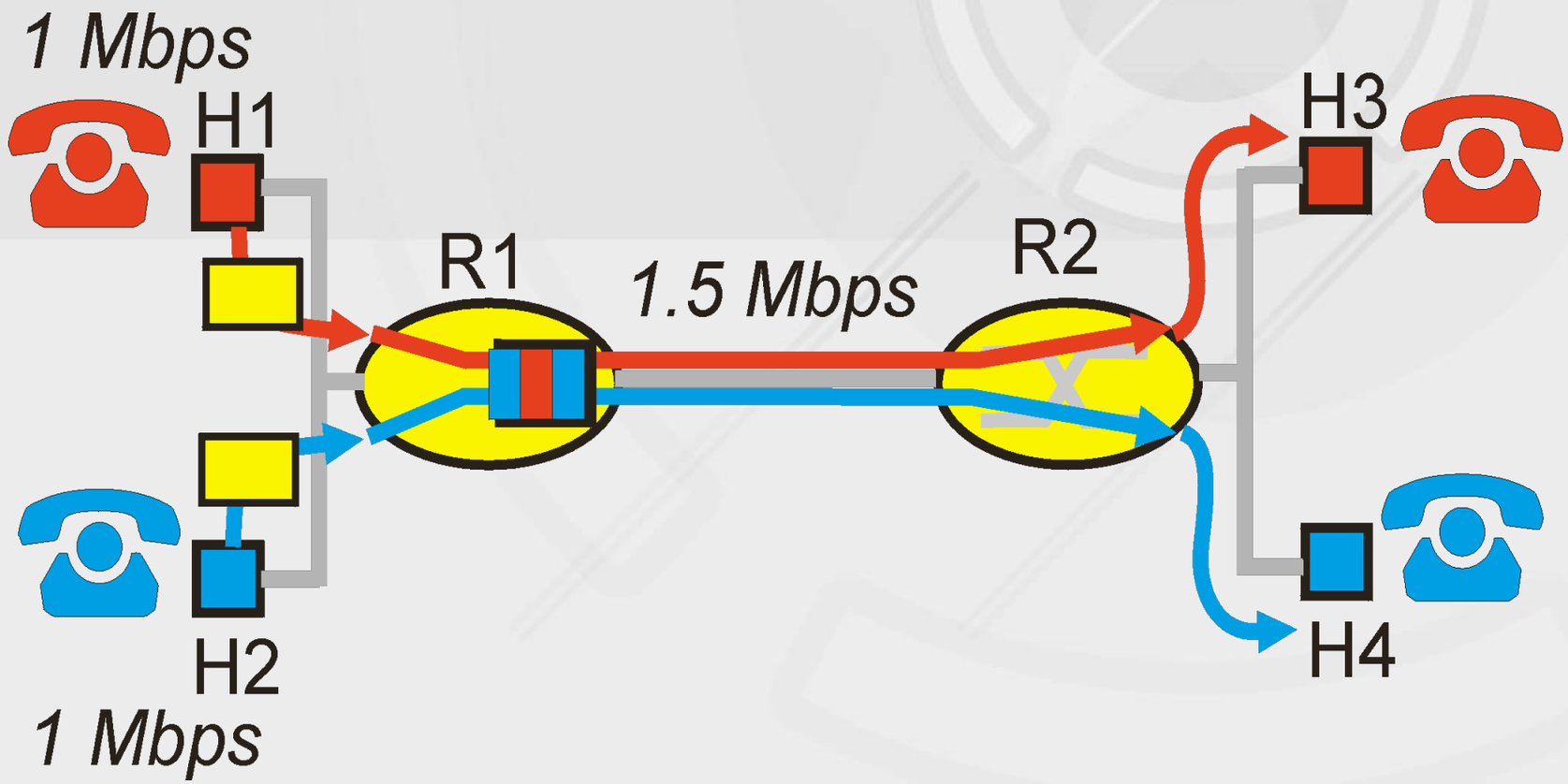
Principles for QOS Guarantees (more)

- Alternative to Marking and Policing: allocate a set portion of bandwidth to each application flow; can lead to inefficient use of bandwidth if one of the flows does not use its allocation
- **PRINCIPLE 3: While providing isolation, it is desirable to use resources as efficiently as possible**



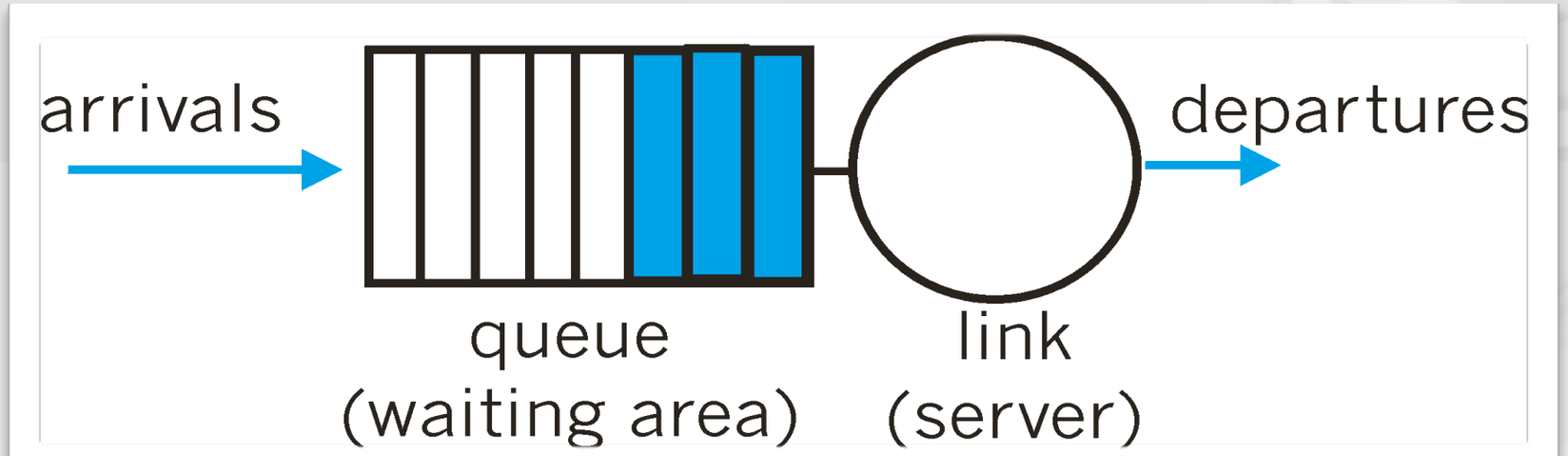
Principles for QOS Guarantees (more)

- Cannot support traffic beyond link capacity
- **PRINCIPLE 4: Need a Call Admission Process; application flow declares its needs, network may block call if it cannot satisfy the needs**



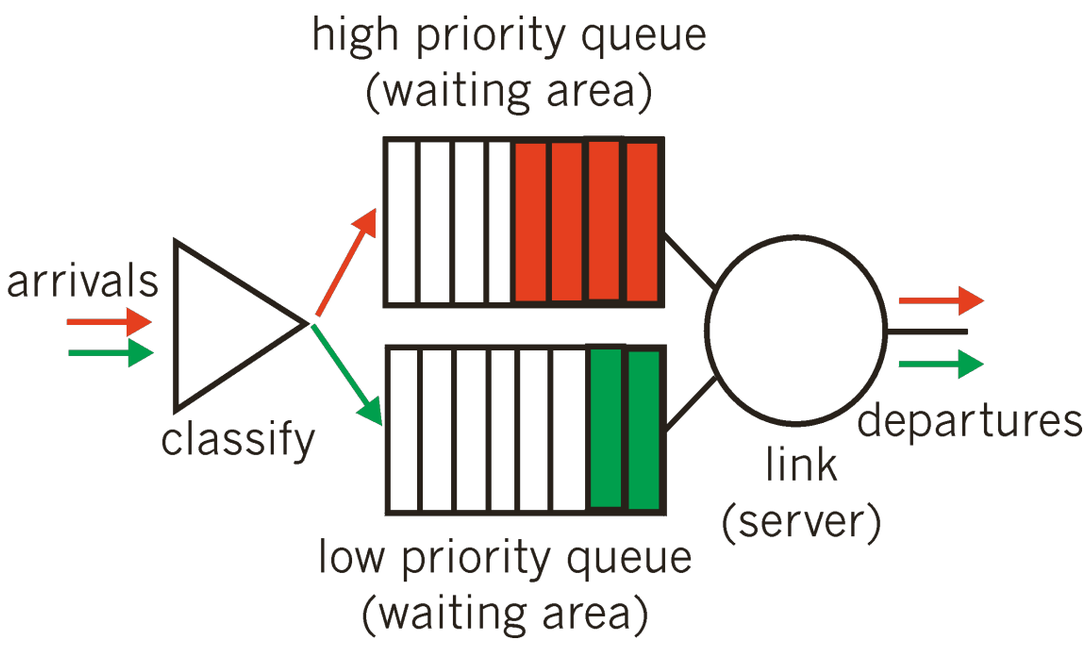
Scheduling And Policing Mechanisms

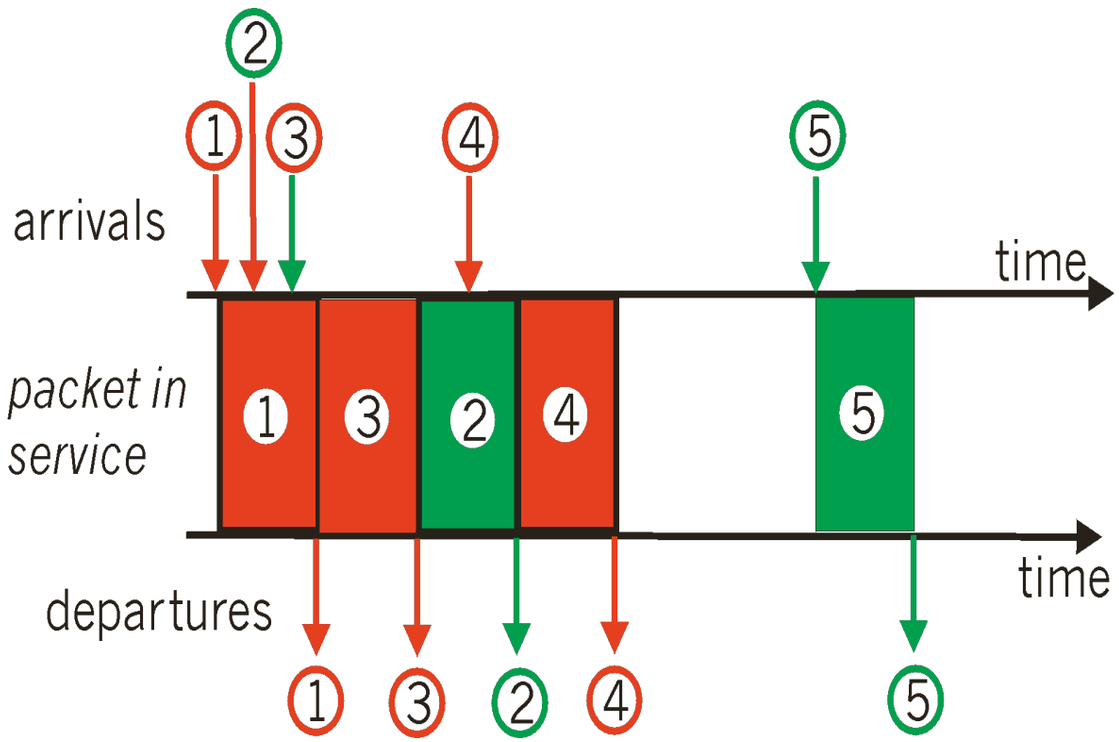
- Scheduling: choosing the next packet for transmission on a link can be done following a number of policies;
- FIFO: in order of arrival to the queue; packets that arrive to a full buffer are either discarded, or a discard policy is used to determine which packet to discard among the arrival and those already queued



Scheduling Policies

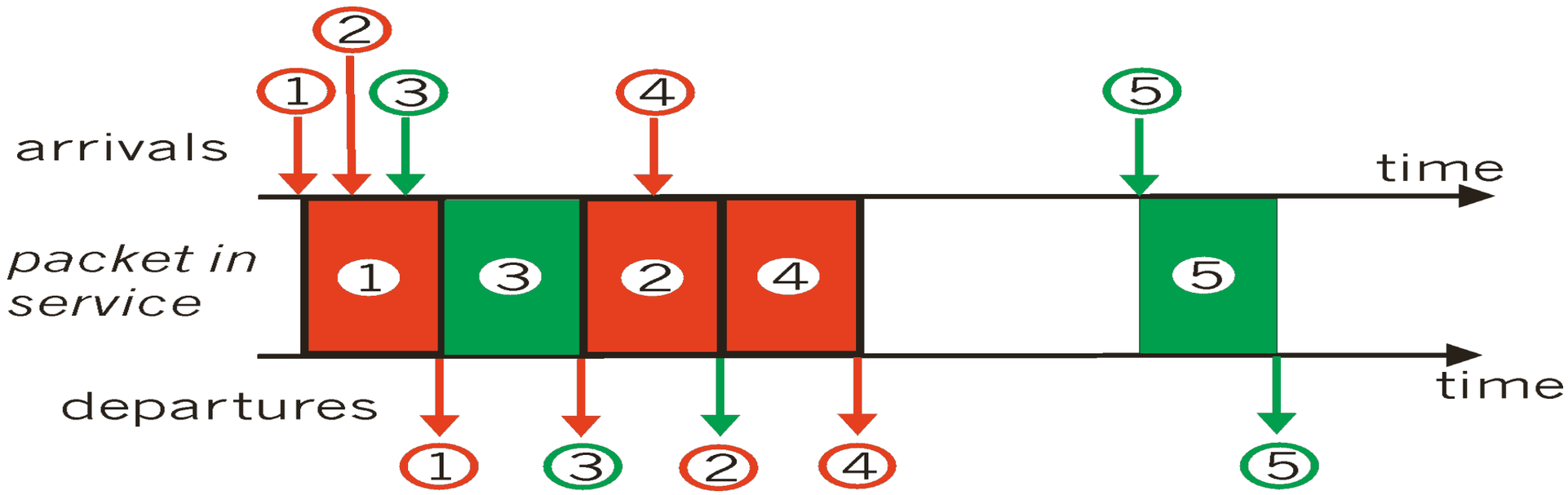
- Priority Queuing: classes have different priorities; class may depend on explicit marking or other header info, e.g. IP source or destination, TCP Port numbers, etc.
- Transmit a packet from the highest priority class with a non-empty queue
- Preemptive and non-preemptive versions





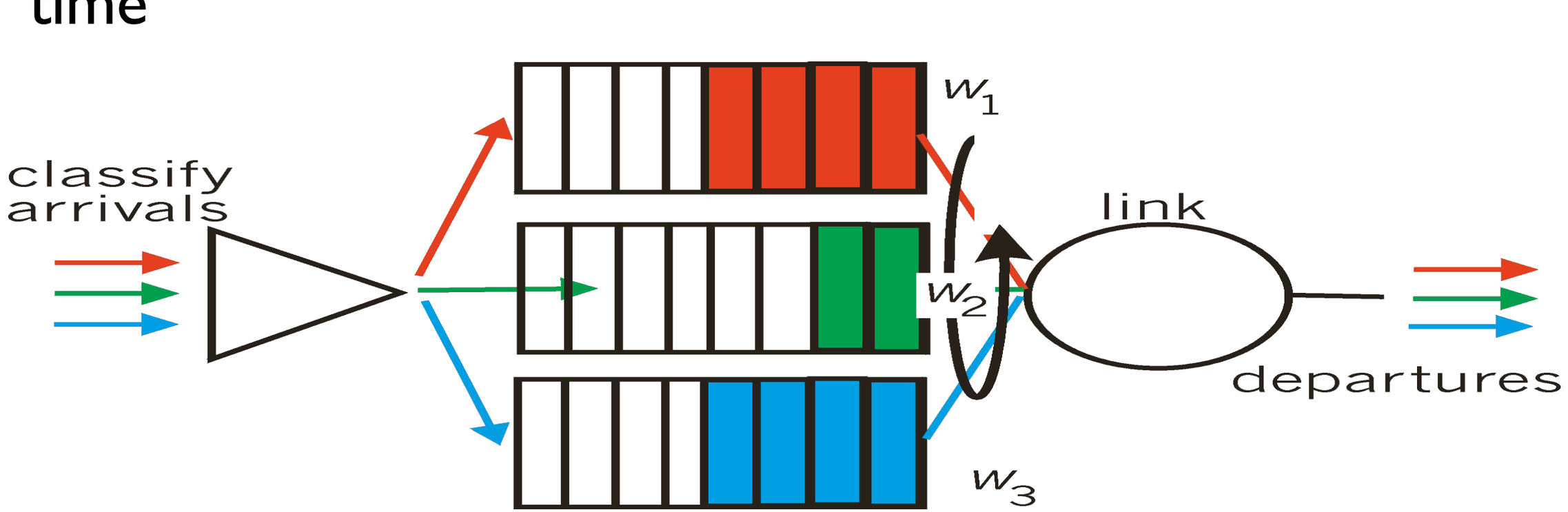
Scheduling Policies (more)

- Round Robin: scan class queues serving one from each class that has a non-empty queue



Scheduling Policies (more)

- Weighted Fair Queuing: is a generalized Round Robin in which an attempt is made to provide a class with a differentiated amount of service over a given period of time

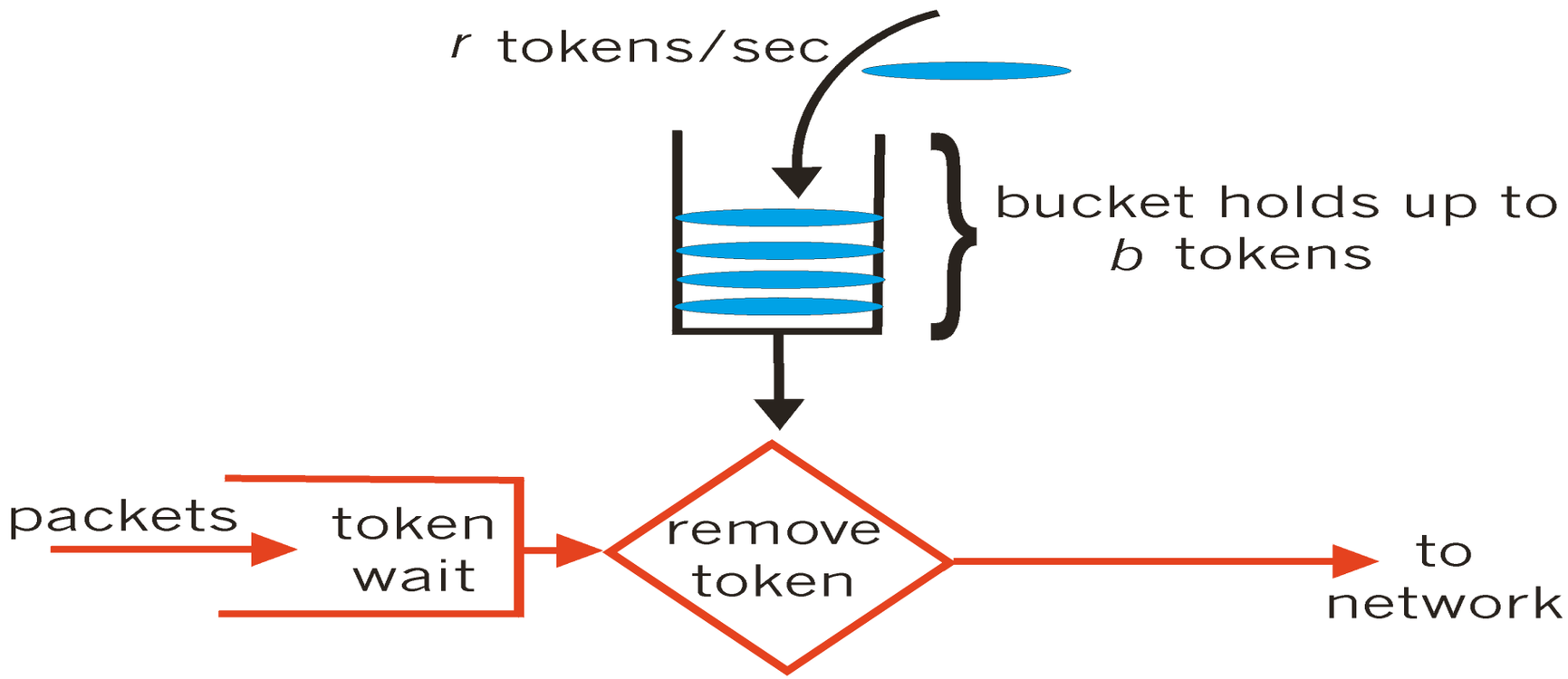


Policing Mechanisms

- Three criteria:
  - (Long term) **Average Rate** (100 packets/sec or 6000 packets/min), crucial aspect is the interval length
  - **Peak Rate**: e.g., 6000 p/min **Avg** and 1500 p/sec **Peak**
  - (Max.) **Burst Size**: Max. number of packets sent consecutively, i.e. over a short period of time

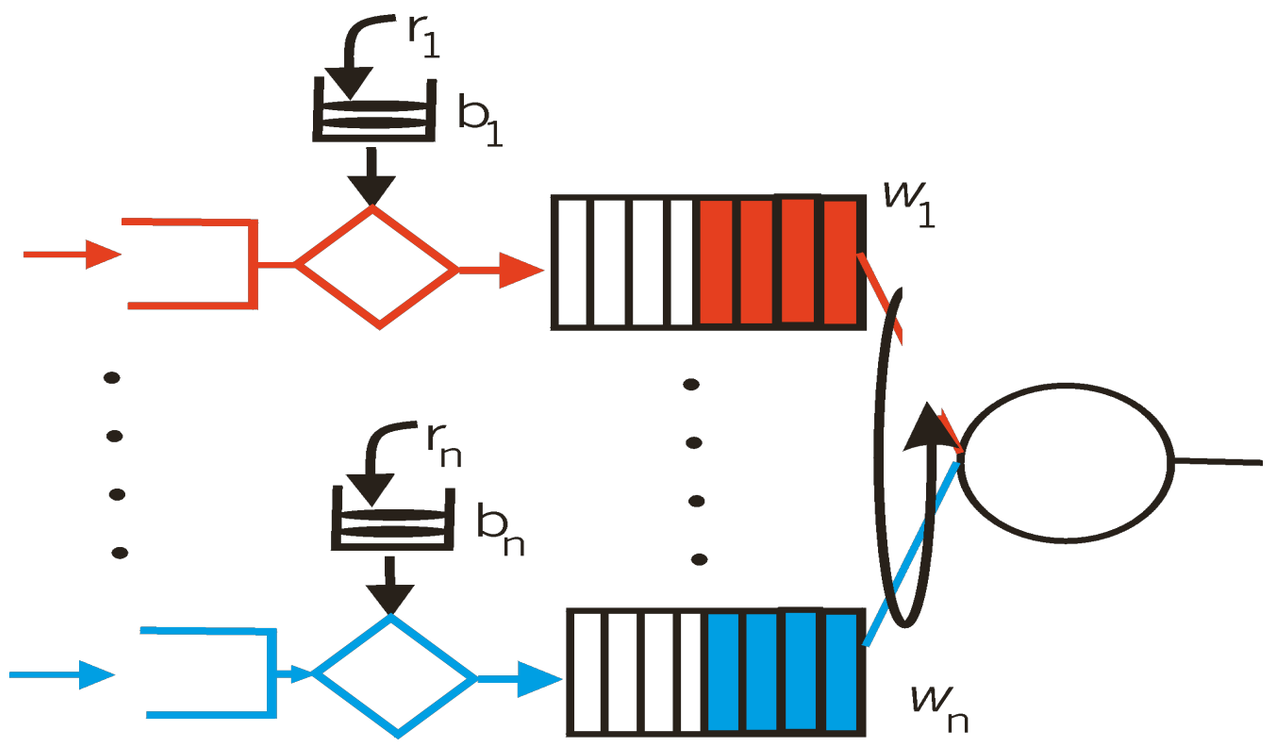
Policing Mechanisms

- **Token Bucket** mechanism, provides a means for limiting input to specified Burst Size and Average Rate.



Policing Mechanisms (more)

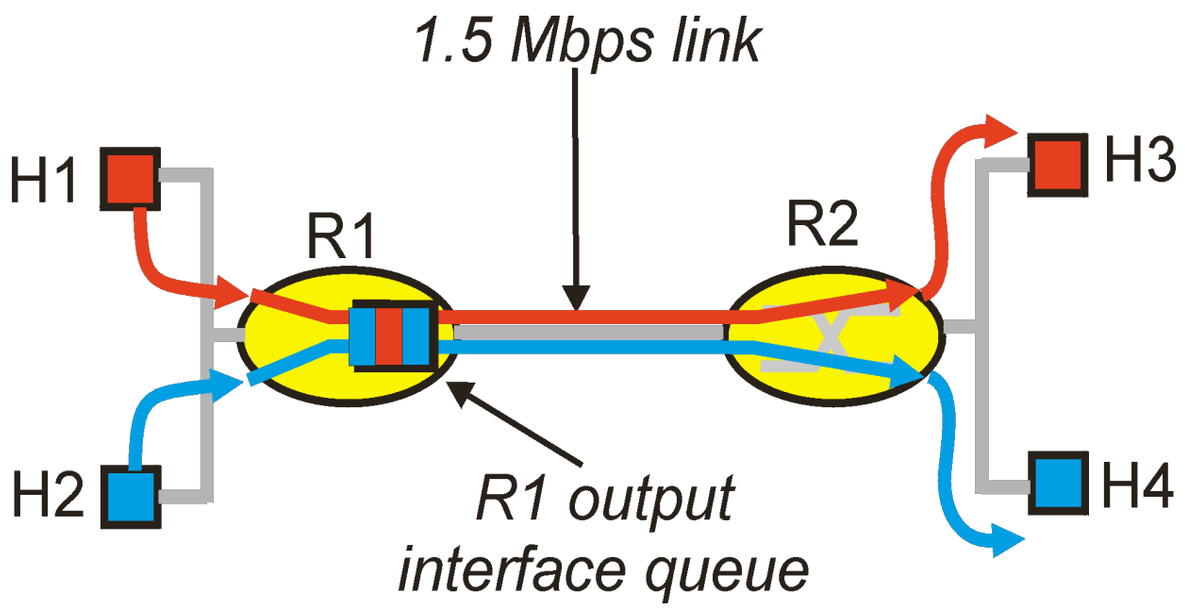
- Bucket can hold b tokens; token are generated at a rate of *r token/sec* unless bucket is full of tokens.
- Over an interval of length *t*, the number of packets that are admitted is less than or equal to *(r t + b)*.
- Token bucket and WFQ can be combined to provide upper bound on delay.



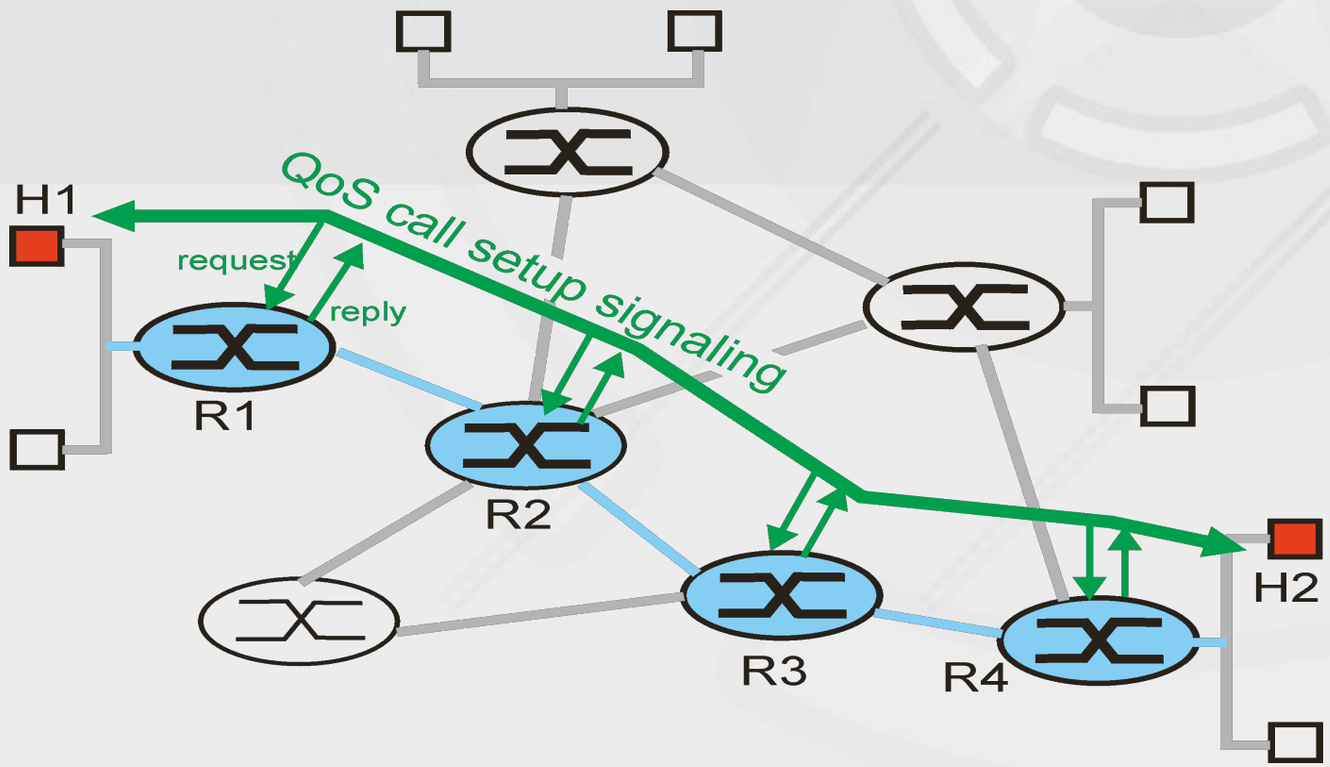
IETF IP QoS Efforts

- Policy based IP QoS Solutions
  - Integrated Services (RSVP protocol):
    - flow based
  - Differentiated Services (DiffServ byte settings):
    - packet based
  - Multi-Protocol Label Switching (MPLS):
    - flow+packet based

- IP Multicast and Anycast
- IPv6 QoS Support

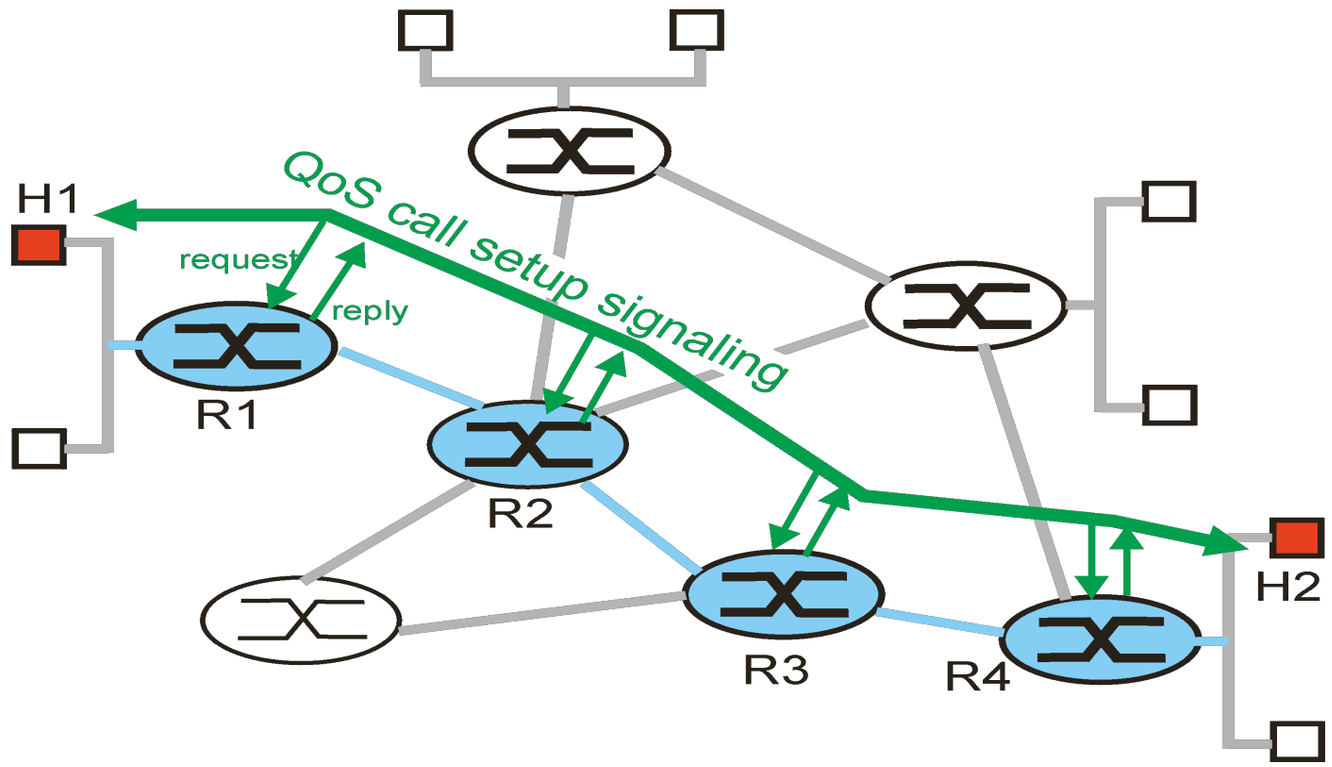


Connection Oriented QoS



Connection Oriented QoS

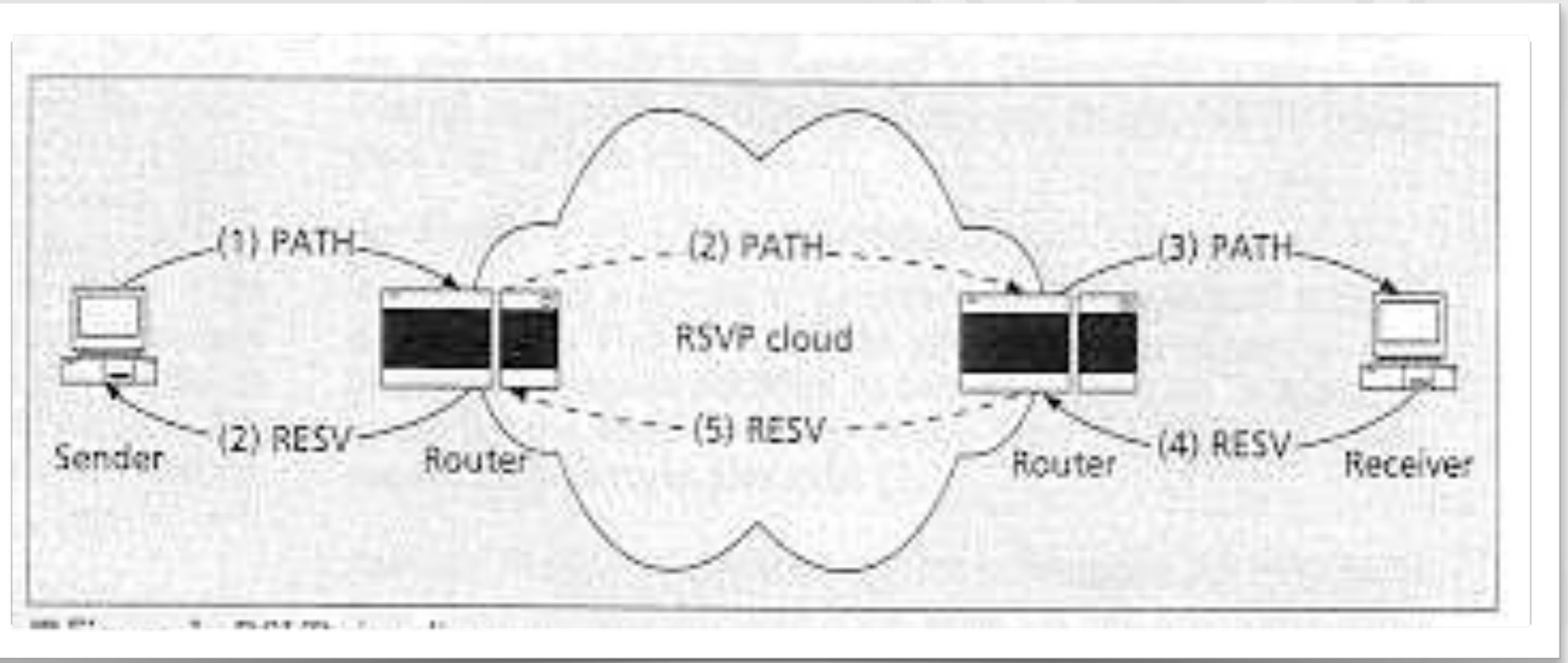
- **Integrated Services**: IETF RFC 1633
  - Defined by RSVP requires resource reservation at each hop end-to-end for each IP packet flow, and end-to-end signaling along nodes in the path
  - Reserve resources at the routers so as to provide QoS for specific user packet stream
  - This architecture does not scale well (large amount of states)
  - Many Internet flows are short lived, not worth setting up VC



Integrated Services / RSVP

- Sender sends a "**PATH**" message to the receiver specifying characteristics of traffic
  - every intermediate router along the path forwards the "PATH" message to the next hop determined by the routing protocol
- Receiver responds with "RESV" message after receiving "PATH". "RESV" requests resources for flow

RSVP
=
Resource reservation protocol



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- Diff Serv defines code points (DSCP) for the DS field, DE (default) = 000000 = best effort, and EF (Expedited Forwarding) = 101110 = low latency, etc.

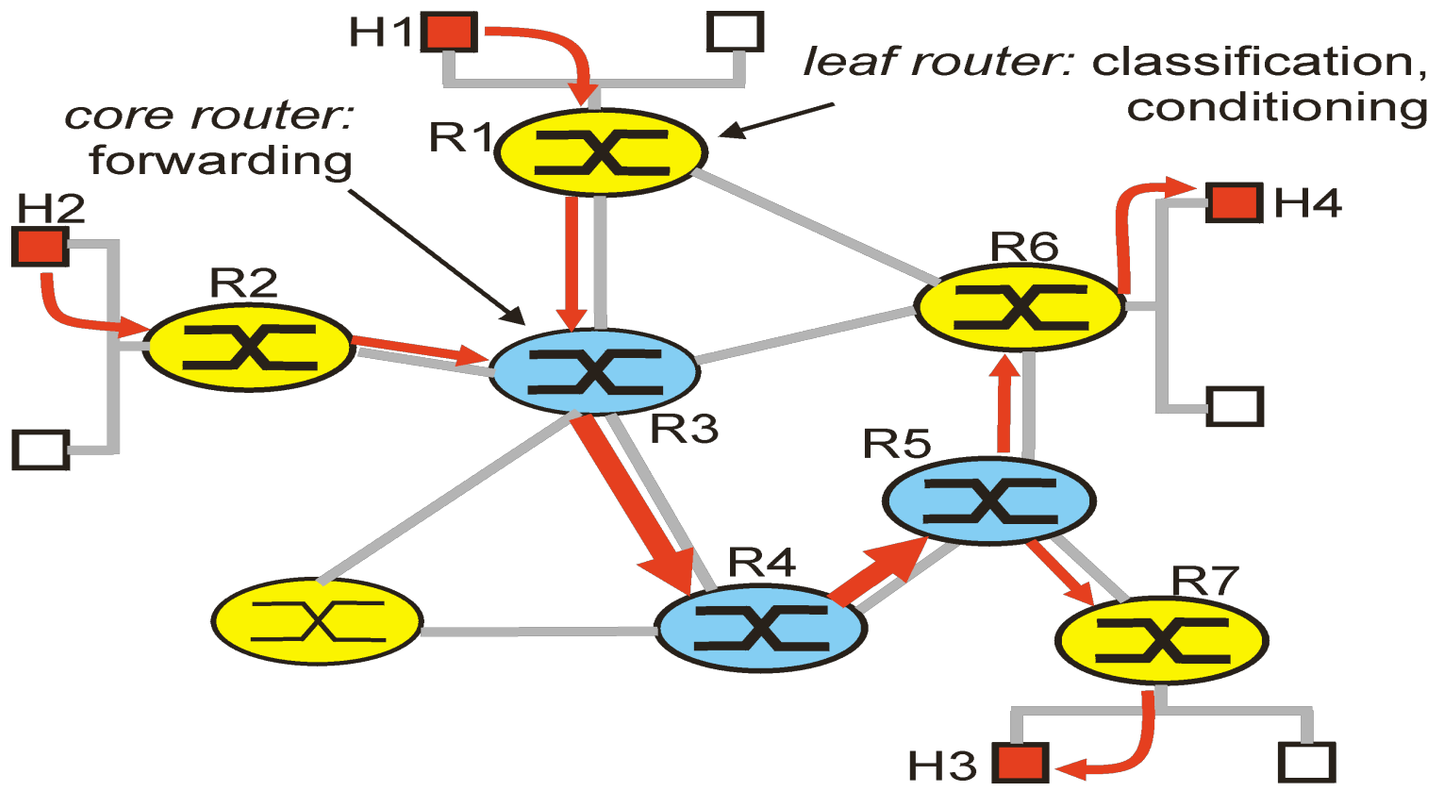
Differentiated Services

- Approach:
  - Only simple functions in the core, and relatively complex functions at edge routers (or hosts)
  - Do not define service classes, instead provides functional components with which service classes can be built

Edge Functions

- At DS-capable host or first DS-capable router

- **Classification**: edge node marks packets according to classification rules to be specified (manually by admin, or by some TBD protocol)
- **Traffic Conditioning**: edge node may delay and then forward or may discard



Core Functions

- **Forwarding**: according to "Per-Hop-Behavior" or PHB specified for the particular packet class; such PHB is strictly based on class marking (no other header fields can be used to influence PHB)

- BIG ADVANTAGE:

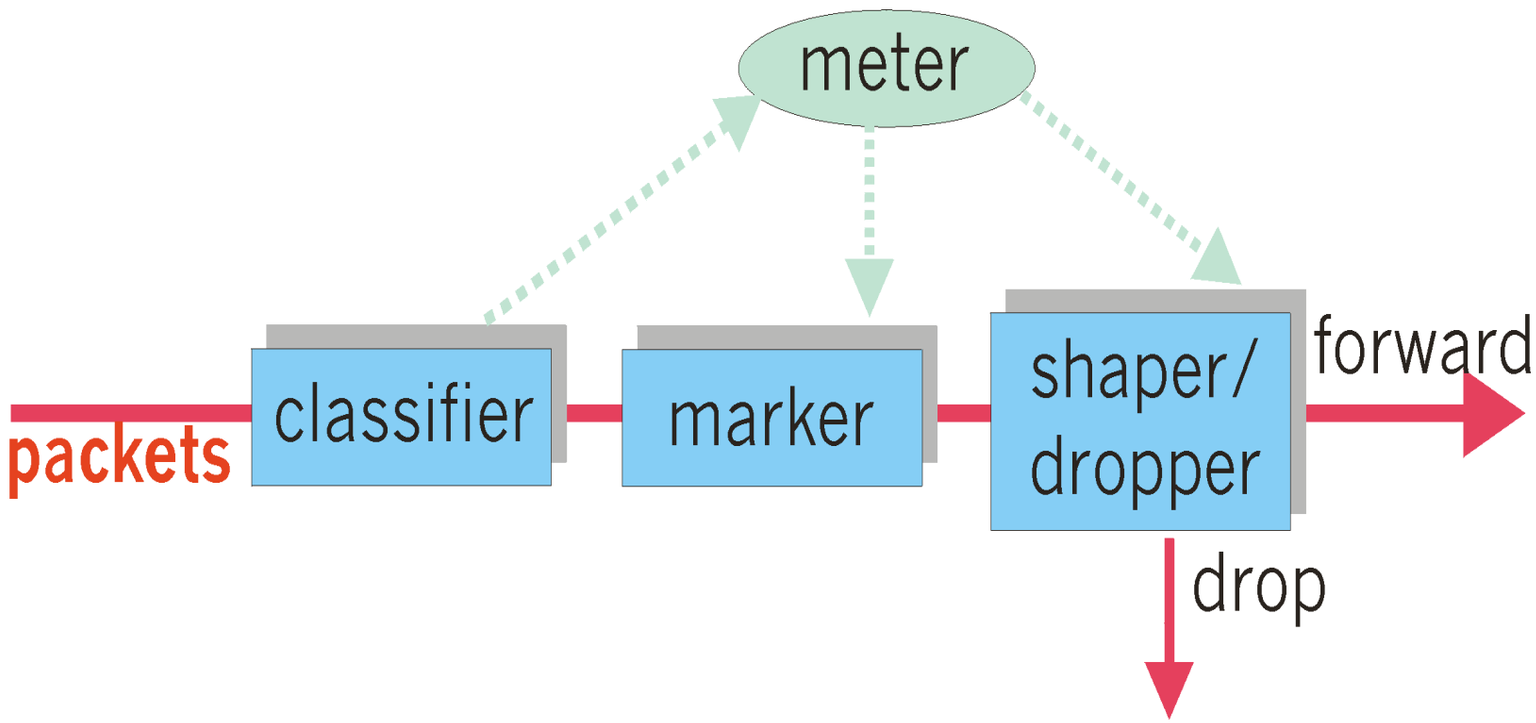
  No state info to be maintained by routers!

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

Classification and Conditioning

- It may be desirable to limit traffic injection rate of some class; user declares traffic profile (e.g., rate and burst size);
- traffic is metered and shaped if non-conforming



IPv6 Support of QoS

- IPv6 Flow Labels provide support for Data Flows
  - –Packet Prioritizing
    - sure that high priority traffic is not interrupted by less critical data

- IPv6 supports Multicast & Anycast
  - –Multicast delivers data simultaneously to all hosts that sign up to receive it
  - –Anycast allows one host initiate the efficient updating of routing tables for a group of hosts.

How should the Internet evolve to better support multimedia?

Integrated services philosophy:

- Change Internet protocols so that applications can reserve end-to-end bandwidth
  - Need to deploy protocol that reserves bandwidth
  - Must modify scheduling policies in routers to honor reservations
  - Application must provide the network with a description of its traffic, and must further abide to this description.
- Requires new, complex software in hosts & routers

Differentiated services philosophy:

- Fewer changes to Internet infrastructure, yet provide 1st and 2nd class service.
- Datagrams are marked.
- User pays more to send/receive 1st class packets.
- ISPs pay more to backbones to send/receive 1st class packets.

Streaming Stored Audio & Video

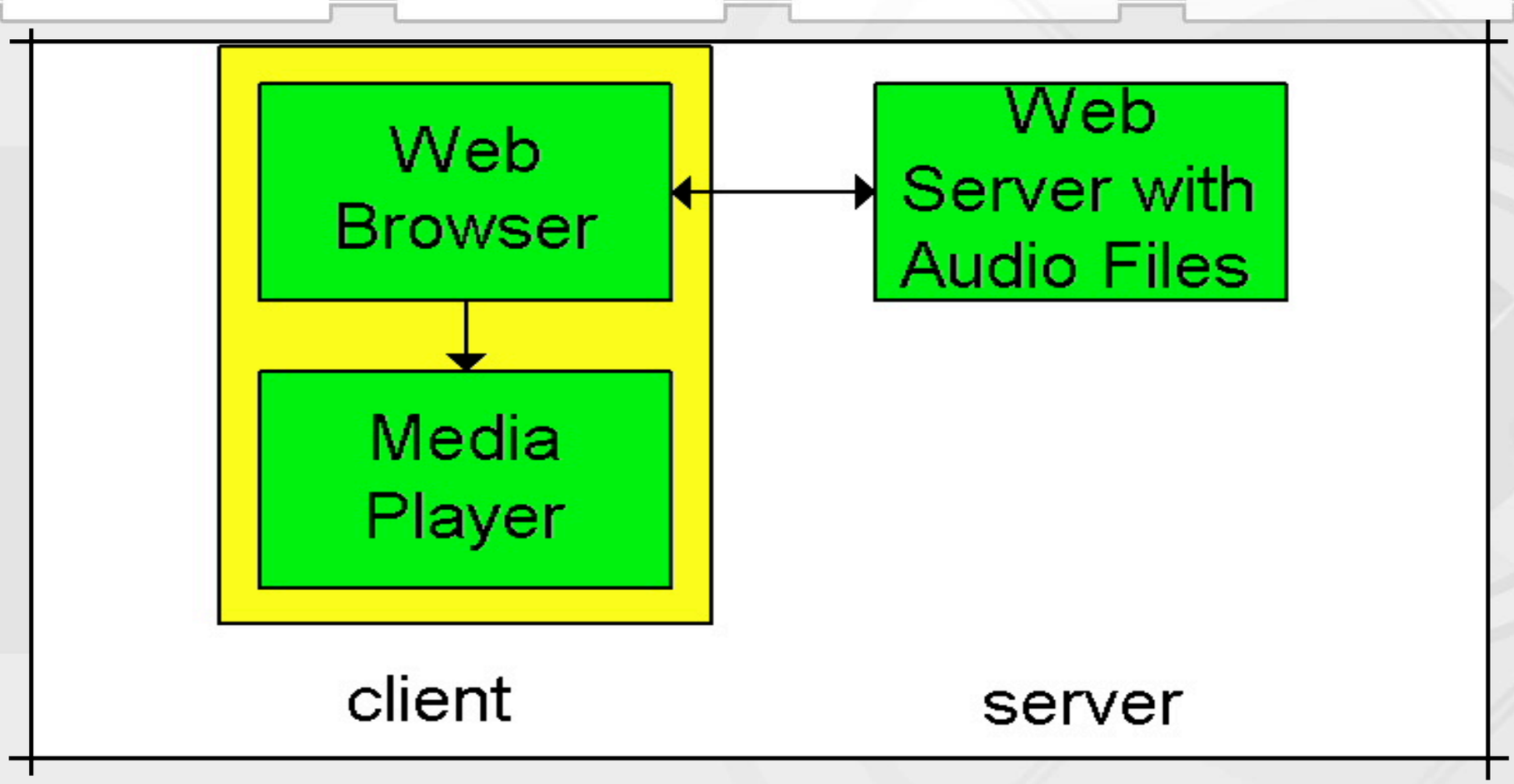
Streaming stored media:

- Audio/video file is stored in a server
- Users request audio/video file on demand.
- Audio/video is rendered within, say, 10 s after request.
- Interactivity (pause, re-positioning, etc.) is allowed.

Media player:

- removes jitter
- decompresses
- error correction
- graphical user interface with controls for interactivity
- Plug-ins may be used to imbed the media player into the browser window

Streaming from Web server (1)



Major drawback:

media player interacts with server through intermediary of a Web browser

Streaming from Web server (1)

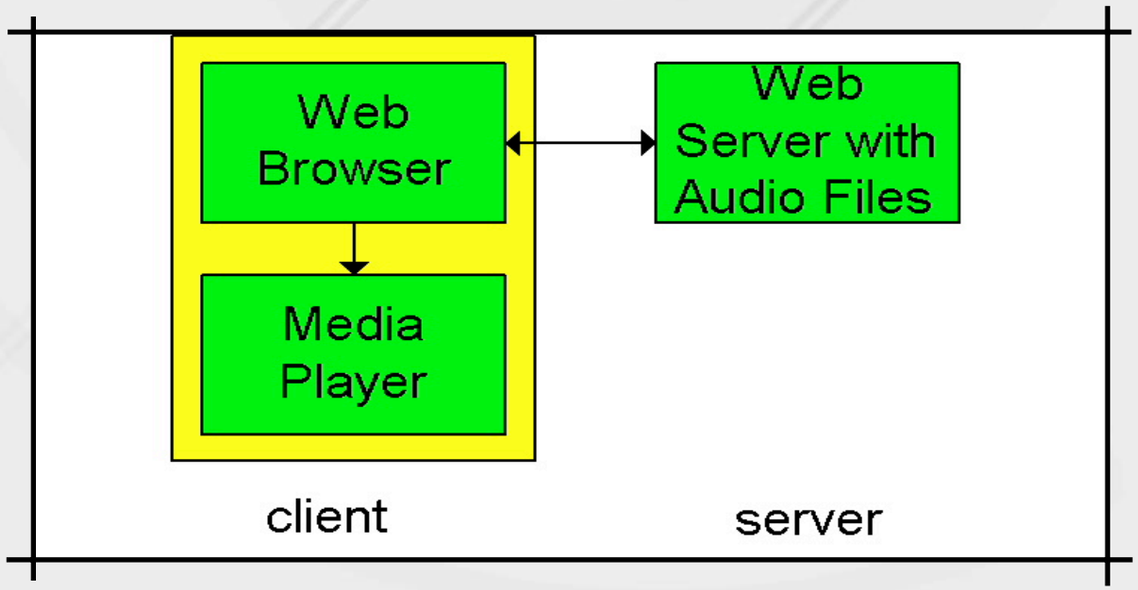
- Audio and video files stored in Web servers

naïve approach

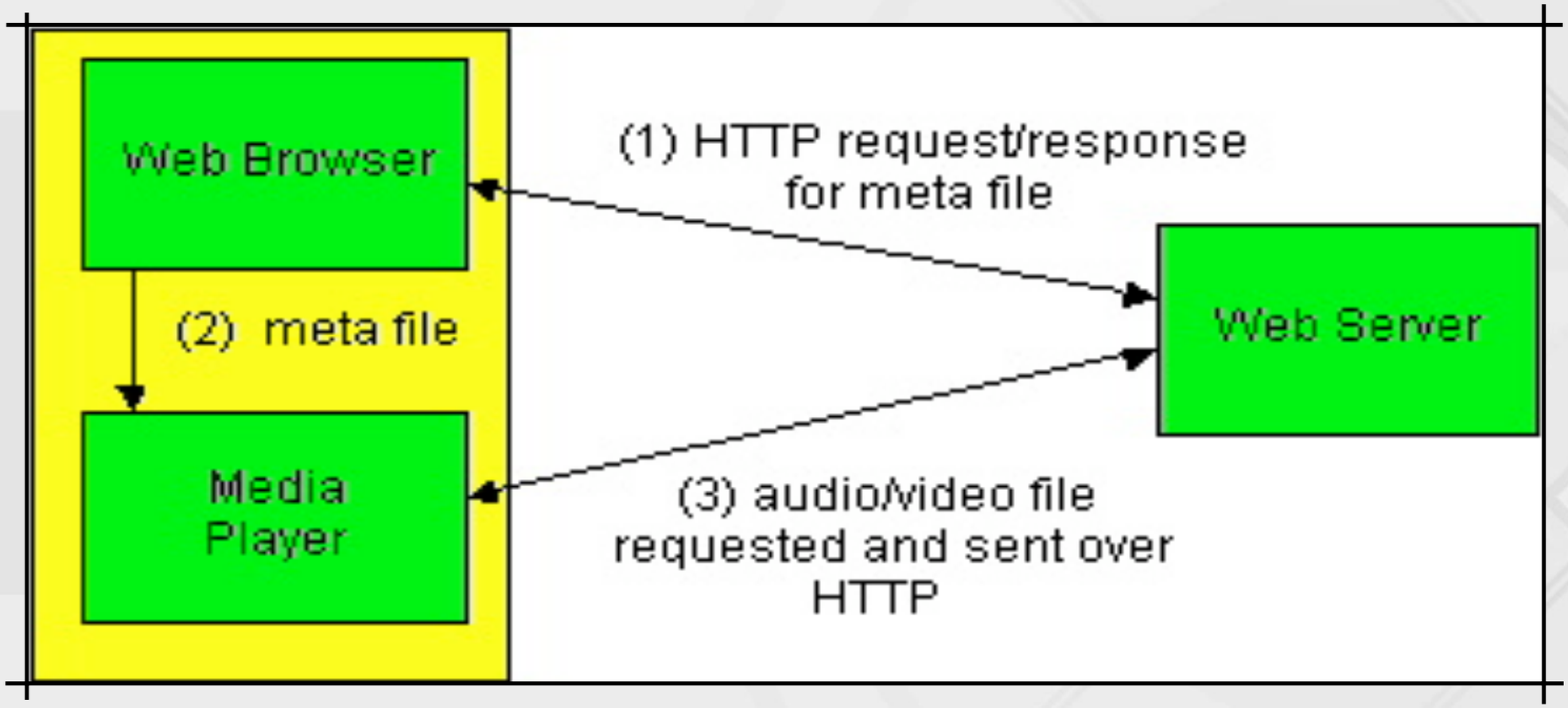
- browser requests file with HTTP request message
- Web server sends file in HTTP response message
- content-type header line indicates an audio/video encoding
- browser launches media player, and passes file to media player
- media player renders file

Major drawback:

media player interacts with server through intermediary of a Web browser



Streaming from Web server (2)



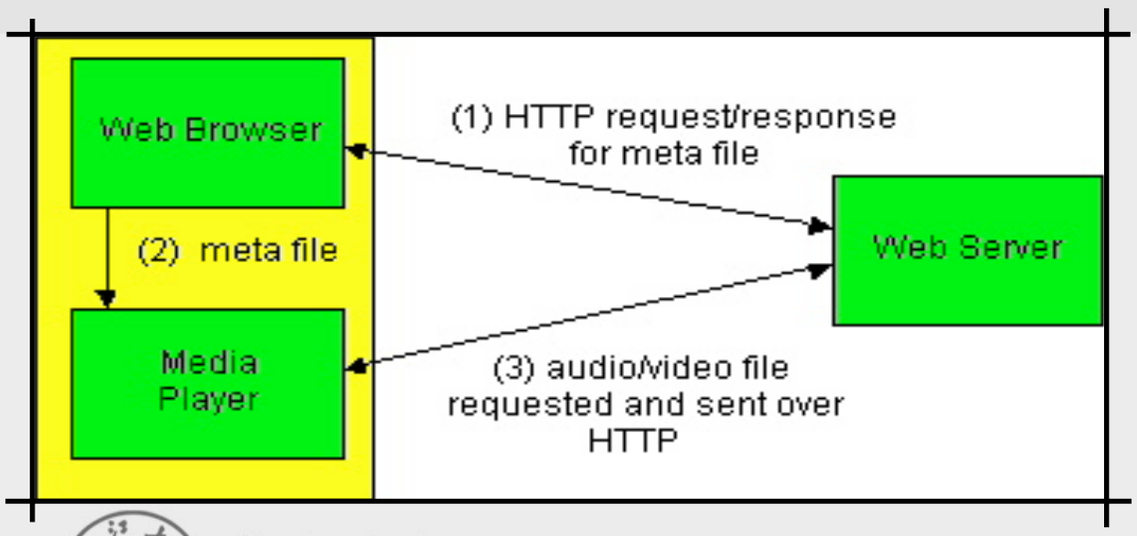
Some concerns:

- Media player communicates over HTTP, which is not designed with pause, ff, rwnd commands
- May want to stream over UDP

Streaming from Web server (2)

Alternative: set up connection between server and player

- Web browser requests and receives a **meta file** (a file describing the object) instead of receiving the file itself;
- Content-type header indicates specific audio/video application
- Browser launches media player and passes it the meta file
- Player sets up a TCP connection with server and sends HTTP request.



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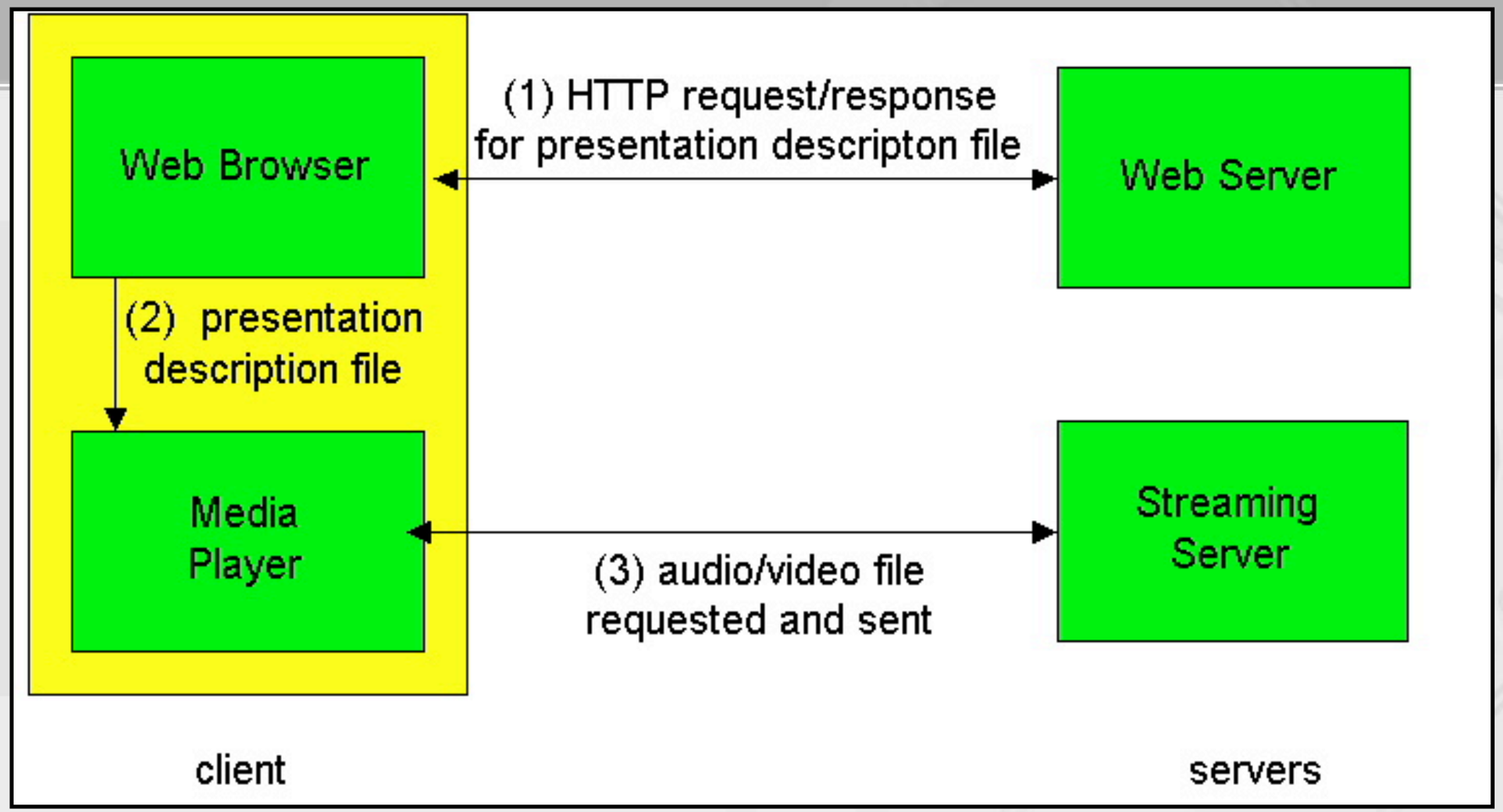
Streaming from a streaming server



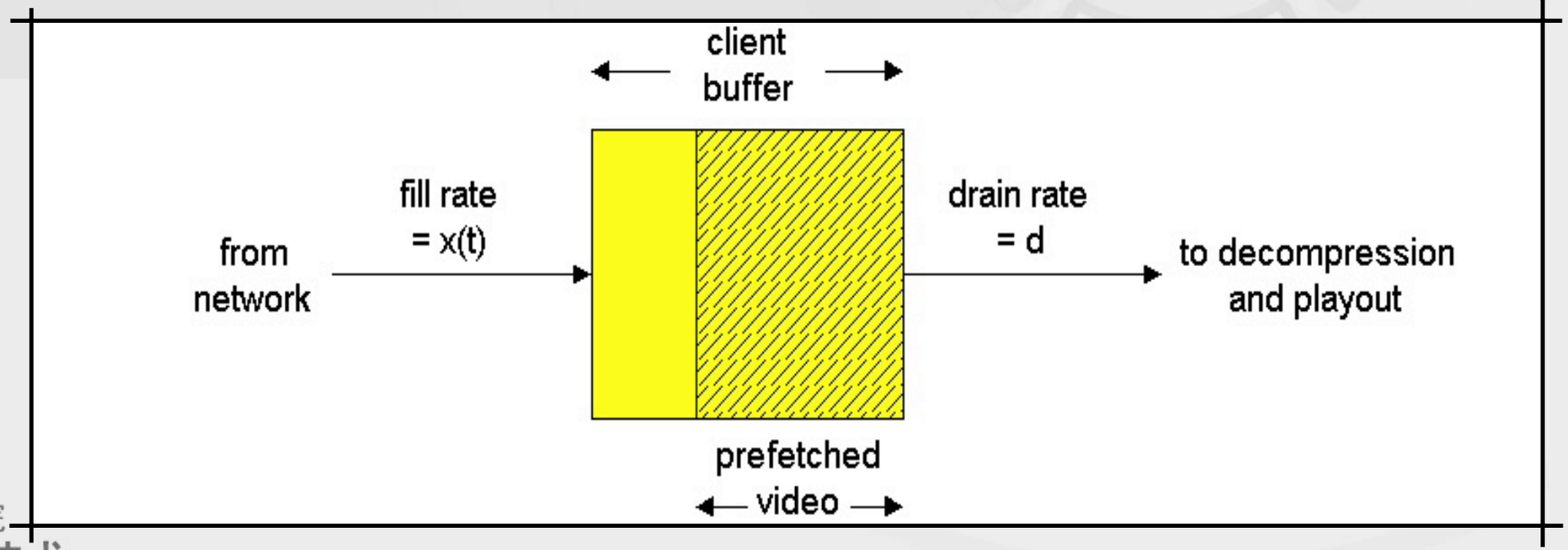
This architecture allows for non-HTTP protocol between server and media player

Can also use UDP instead of TCP.

Options when using a streaming server

- Send at constant rate over **UDP**. To mitigate the effects of jitter, buffer and delay playback for 1-10 s. Transmit rate = d, the encoded rate. Fill rate x(t) equals d except when there is loss.

- Use TCP, and send at maximum possible rate under TCP; TCP retransmits when error is encountered; x(t) now fluctuates, and can become much larger than d. Player can use a much large buffer to smooth delivery rate of TCP.



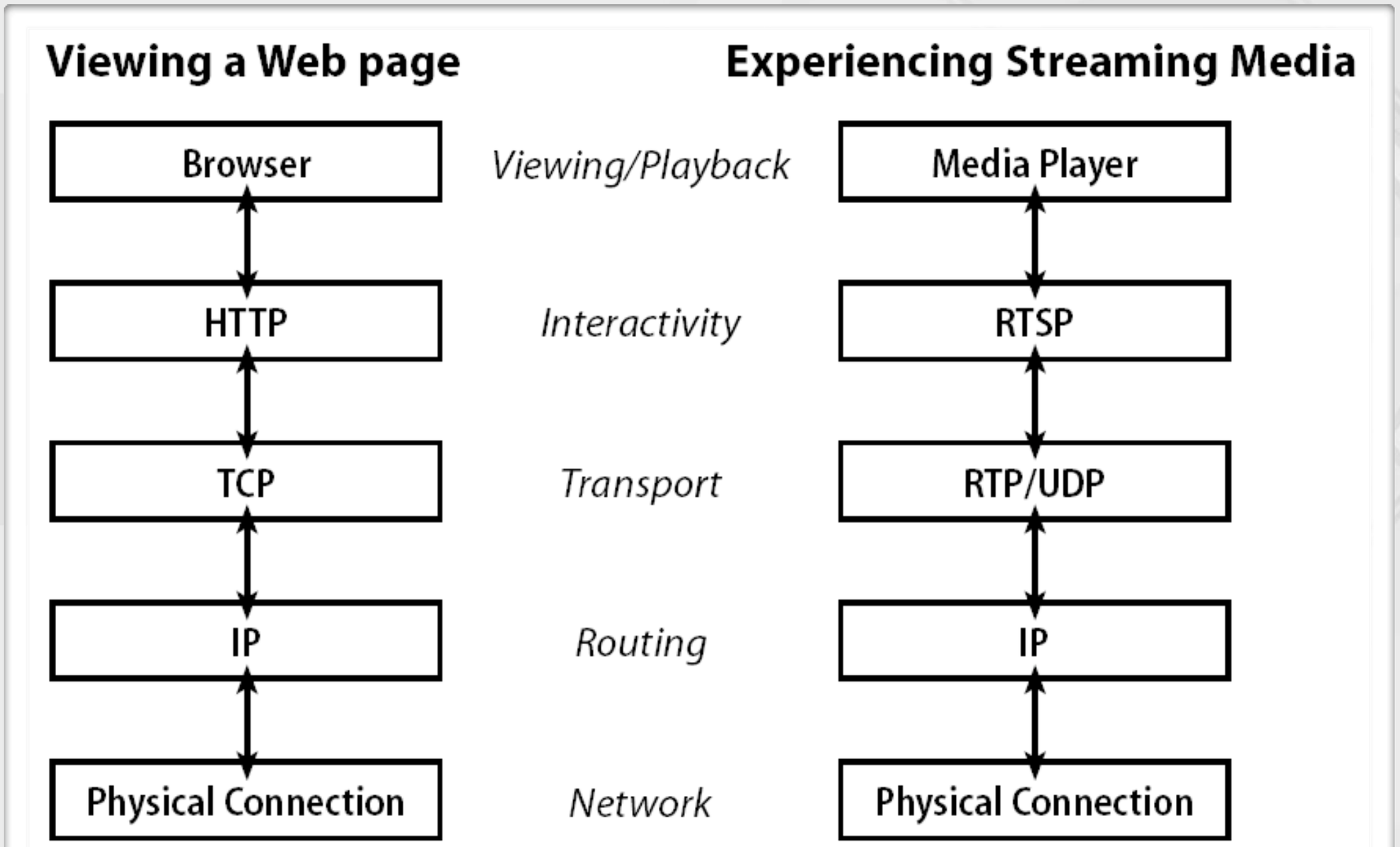
Streaming Protocols

- **RTP (Real-time Transport Protocol)**
  - similar to **HTTP** and to **FTP**—protocols used by Web servers
  - However, **HTTP** and **FTP** cannot be used for true streaming
    - both layered on top of **TCP**
  - layered on top of **UDP**
  - RTP enables a one-way stream, transmitting media from the server to the client

- **RTSP (Real-time Streaming Protocol)**
  - a two-way protocol which uses TCP to communicate, and
  - usually layered on top of RTP

- **RTCP (Real-time Transport Control Protocol)**

Streaming Protocols



Real Time Streaming Protocol: RTSP

HTTP

- Designers of HTTP had fixed media in mind: HTML, images, applets, etc.
- HTTP does not target stored continuous media (i.e., audio, video, SMIL presentations, etc.)

RTSP: RFC 2326

- Client-server application layer protocol.
- For user to control display: rewind, fast forward, pause, resume, repositioning, etc…

What it doesn't do:

- does not define how audio/video is encapsulated for streaming over network
- does not restrict how streamed media is transported; it can be transported over UDP or TCP
- does not specify how the media player buffers audio/video

RealNetworks

- Server and player use RTSP to send control info to each other

RTSP: out of band control

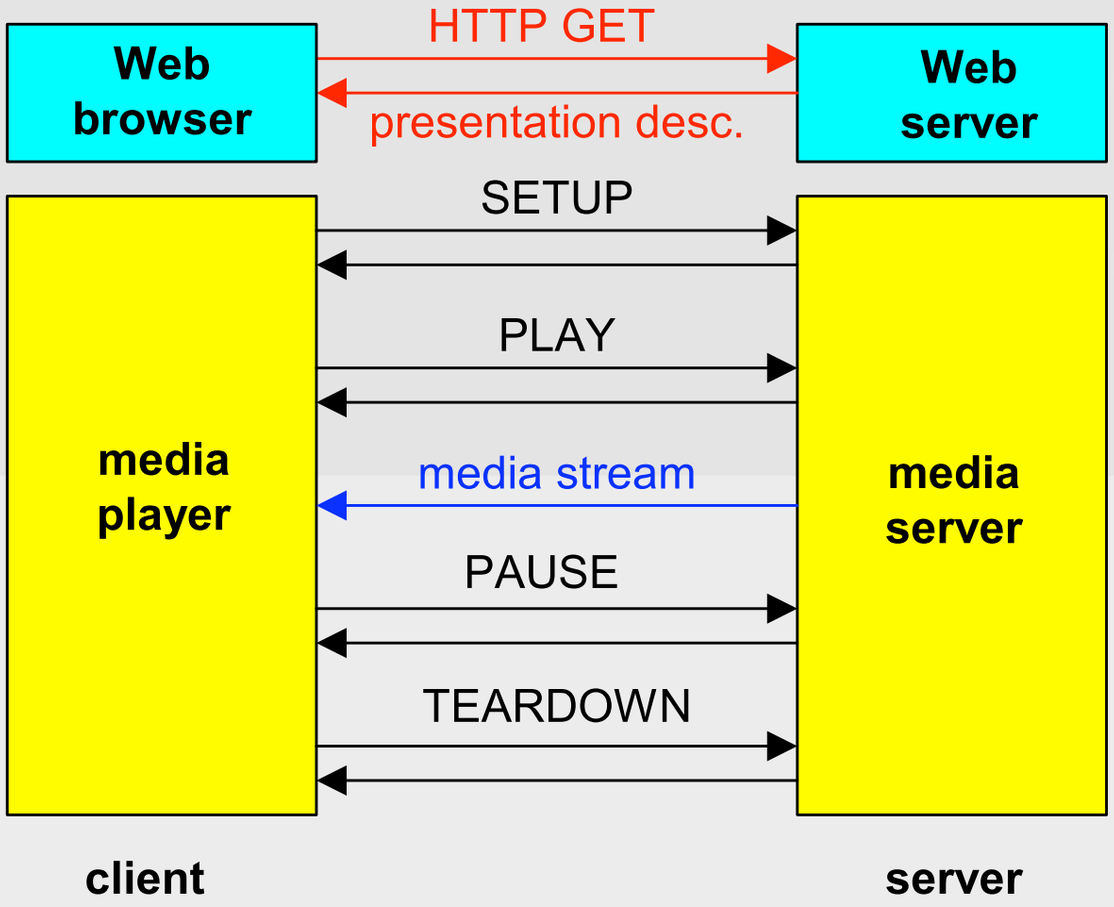
FTP uses an “out-of-band” control channel:

- A file is transferred over one channel.
- Control information (directory changes, file deletion, file renaming, etc.) is sent over a separate TCP connection.
- The “out-of-band” and “in-band” channels use different port numbers.

RTSP messages are also sent out-of-band:

- The RTSP control messages use different port numbers than the media stream, and are therefore sent out-of-band.
- The media stream, whose packet structure is not defined by RTSP, is considered “in-band”.
- If the RTSP messages were to use the same port numbers as the media stream, then RTSP messages would be said to be “interleaved” with the media stream.

RTSP initiates and controls delivery



- Client obtains a description of the multimedia presentation, which can consist of several media streams.
- The browser invokes media player (helper application) based on the content type of the presentation description.
- Presentation description includes references to media streams, using the URL method **rtsp://**
- Player sends **RTSP SETUP** request; server sends RTSP SETUP response.
- Player sends **RTSP PLAY** request; server sends RTSP PLAY response.
- Media server pumps media stream.
- Player sends **RTSP PAUSE** request; server sends RTSP PAUSE response.
- Player sends **RTSP TEARDOWN** request; server sends **RTSP TEARDOWN** response.

Meta file 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 session

- Each RTSP has a session identifier, which is chosen by the server.
- The client initiates the session with the SETUP request, and the server responds to the request with an identifier.
- The client repeats the session identifier for each request, until the client closes the session with the TEARDOWN request.
- RTSP port number is 554.
- RTSP can be sent over UDP or TCP. Each RTSP message can be sent over a separate TCP connection.

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

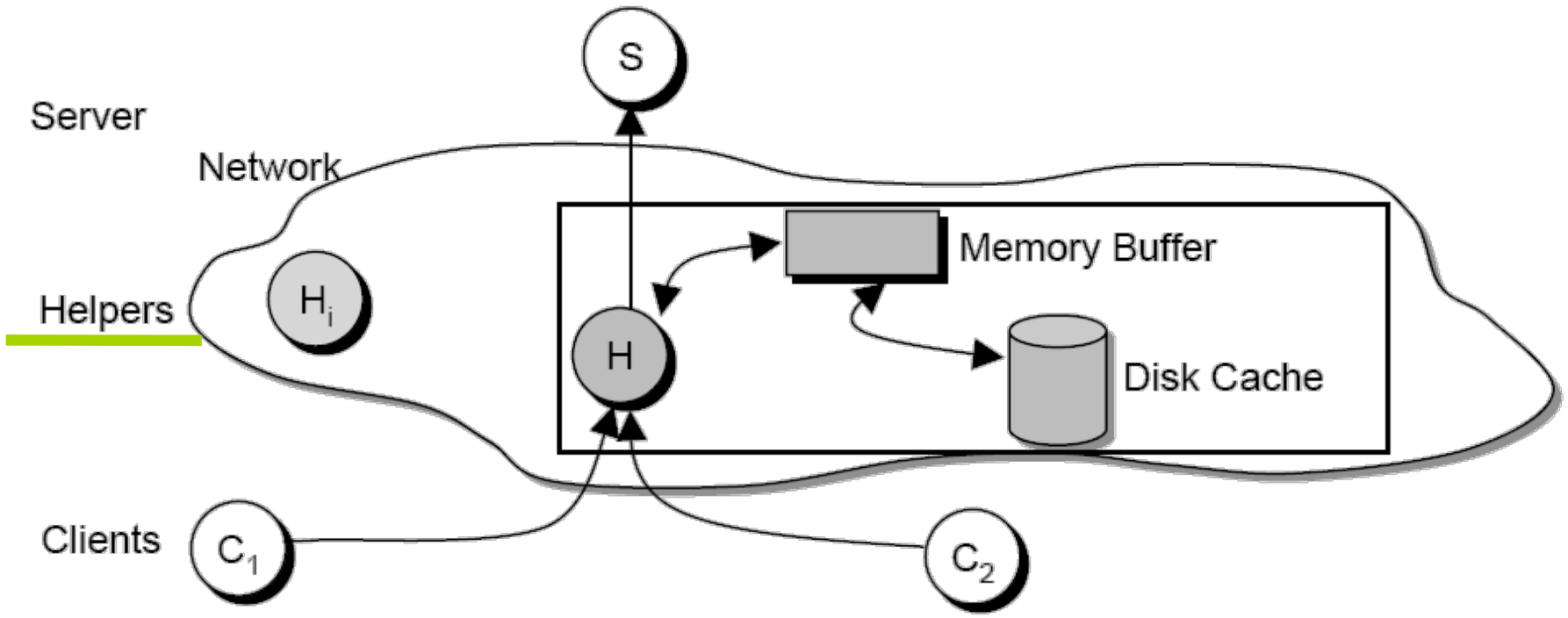
- C: TEARDOWN rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
- Session: 4231
- S: 200 3 OK

RTSP: streaming caching

- Caching of RTSP response messages makes little sense.
- But desirable to cache media streams closer to client.

- Much of HTTP/1.1 cache control has been adopted by RTSP.
  - Cache control headers can be put in RTSP SETUP requests and responses:
    - If-modified-since: , Expires: , Via: , Cache-Control:

Design and Implementation of a Caching System for Streaming Media



Application layer aware helper in the network

RTSP: streaming caching

- **Proxy cache** may
  - hold only segments of a given media stream.
  - start serving a client from its local cache, and then have to connect to origin server and fill missing material, hopefully without introducing gaps at client.

- When origin server is sending a stream through client, and stream passes through a proxy, proxy can use TCP to obtain the stream; but proxy still sends RTSP control messages to origin server.

Real-time interactive applications

- **PC-2-PC phone**
- PC-2-phone
  - Dialpad
  - Net2phone
- videoconference
- Webcams

Internet phone over best-effort (1)

Best effort

- packet delay, loss and jitter

Internet phone example

- now examine how packet delay, loss and jitter are often handled in the context of an IP phone example.
- Internet phone applications generate packets during talk spurts
- bit rate is 64 kbps during talk spurt

  - during talk spurt, every 20 msec app generates a chunk of 160 bytes = 8 kbytes/sec * 20 msec
  - header is added to chunk; then chunk+header is encapsulated into a UDP packet and sent out
  - some packets can be lost and packet delay will fluctuate.
  - receiver must determine when to playback a chunk, and determine what do with missing chunk

Internet phone (2)

packet loss

- UDP segment is encapsulated in IP datagram
- datagram may overflow a router queue
- TCP can eliminate loss, but
  - retransmissions add delay
  - TCP congestion control limits transmission rate
- Redundant packets can help

end-to-end delay

- accumulation of transmission, propagation, and queuing delays
- more than 400 msec of end-to-end delay seriously hinders interactivity; the smaller the better

Internet phone (2)

delay jitter

- consider two consecutive packets in talk spurt
- initial spacing is 20 msec, but spacing at receiver can be more or less than 20 msec

removing jitter

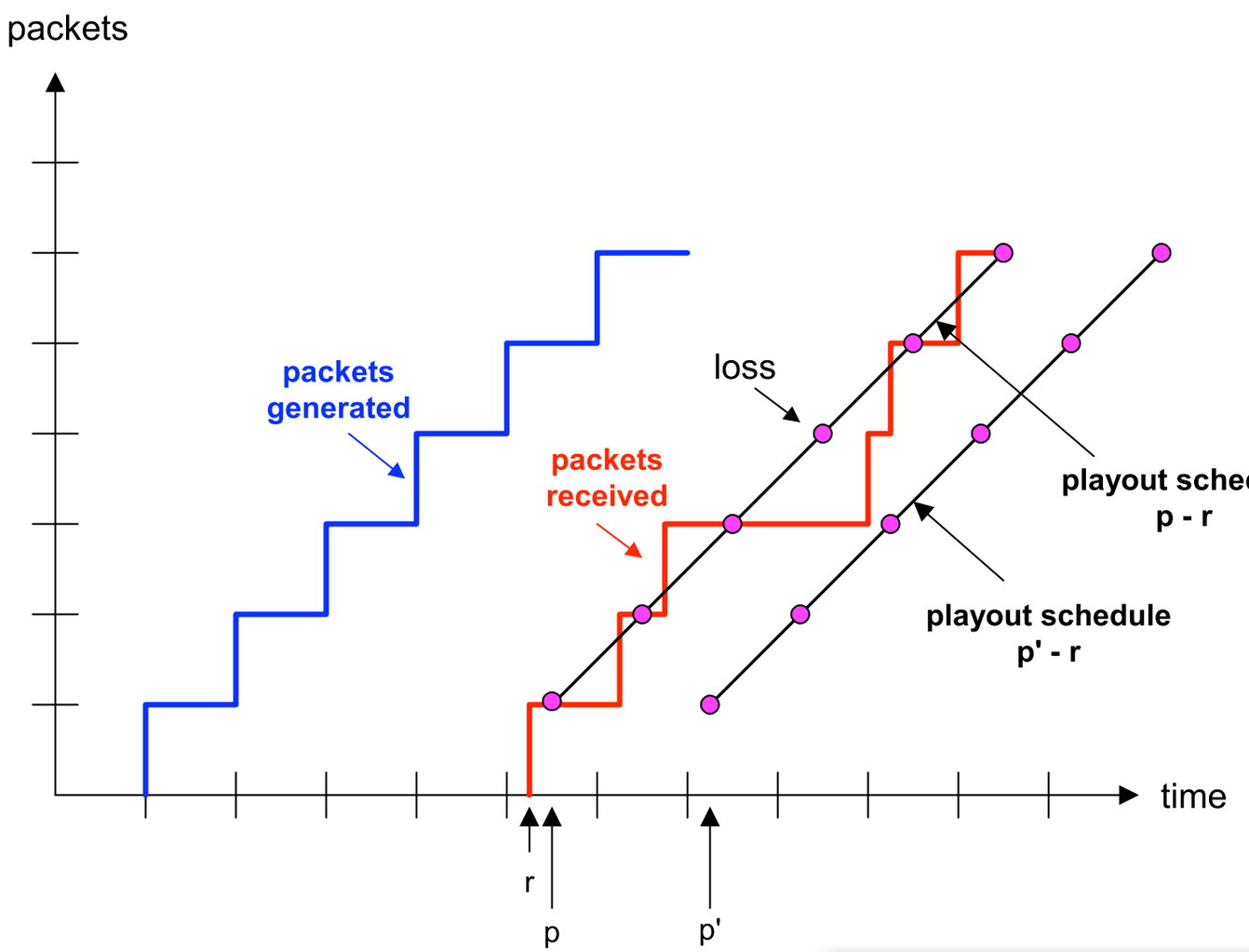
- sequence numbers
- timestamps
- delaying playout

Internet phone (3): fixed playout delay

- Receiver attempts to playout each chunk at exactly *q* msecs after the chunk is generated.
  - If chunk is time stamped *t*, receiver plays out chunk at *t*+*q*.
  - If chunk arrives after time *t*+*q*, receiver discards it.
- Sequence numbers are not necessary.
- Strategy allows for lost packets.

- Tradeoff for q:
  - larger q: less packet loss
  - smaller q: better interactive experience

Internet phone (4): 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)

- Estimate network delay and adjust playout delay at the beginning of each talk spurt.
- Silent periods are compressed and elongated.
- Chunks still played out every 20 msec during talk spurt.

t_i = timestamp of the ith 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 ith packet

d_i = estimate of average network delay after receiving ith 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).

Adaptive playout delay (2)

Also useful to estimate the average deviation of the delay, v_i :

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

The estimates d_i and v_i are calculated for every received packet, although they are only used at the beginning of a talk spurt.

For first packet in talk spurt, playout time is:

$$p_i = t_i + d_i + Kv_i$$

where K is a positive constant. For this same packet, the play out delay is:

$$q_i = p_i - t_i$$

For packet j in the same talk spurt, play packet out at

$$p_j = t_j + q_i$$

Adaptive playout (3)

How to determine whether a packet is the first in a talkspurt:

- If there were never loss, receiver could simply look at the successive time stamps.
  - Difference of successive stamps > 20 msec, talk spurt begins.

- But because loss is 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)

- Loss: packet never arrives or arrives later than its scheduled playout time

forward error correction (FEC): simple scheme

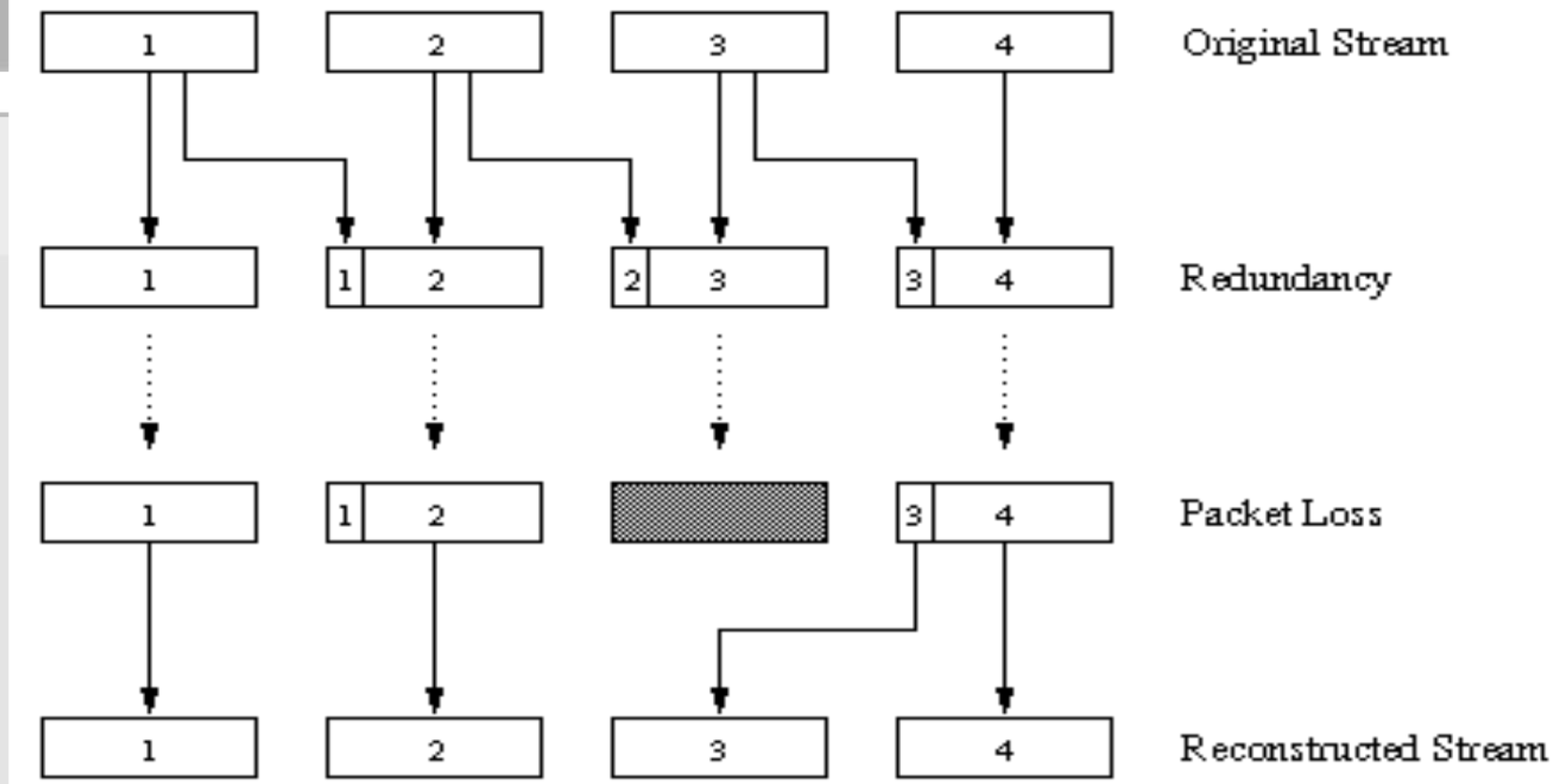
- for every group of n chunks create a redundant chunk by exclusive OR-ing the n original chunks
- send out n+1 chunks, increasing the bandwidth by factor 1/n.
- can reconstruct the original n chunks if there is at most one lost chunk from the n+1 chunks

- Playout delay needs to fixed to the 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

Recovery from packet loss (2)

2nd FEC scheme

- "piggyback lower quality stream"
- send lower resolution audio stream as the redundant information
- for example, nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps.
- Sender creates packet by taking the nth chunk from nominal stream and appending to it the (n-1)st chunk from redundant stream.

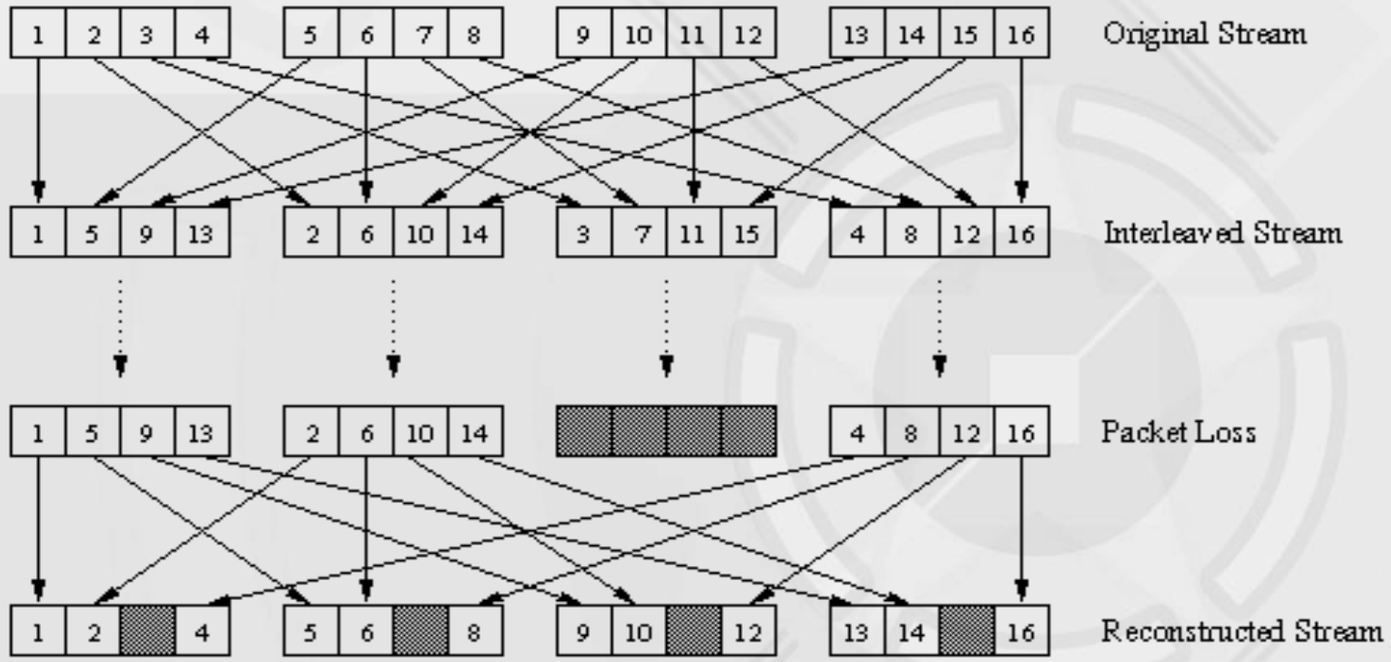


- Whenever there is non-consecutive loss, the receiver can conceal the loss.
- Only two packets need to be received before playback
- Can also append (n-1)st and (n-2)nd low-bit rate chunk

Recovery from packet loss (3)

Interleaving

- chunks are broken up into smaller units
- for example, 4 5 msec units per chunk
- interleave the chunks as shown in diagram
- packet now contains small units from different chunks



- Reassemble chunks at receiver
- if packet is lost, still have most of every chunk

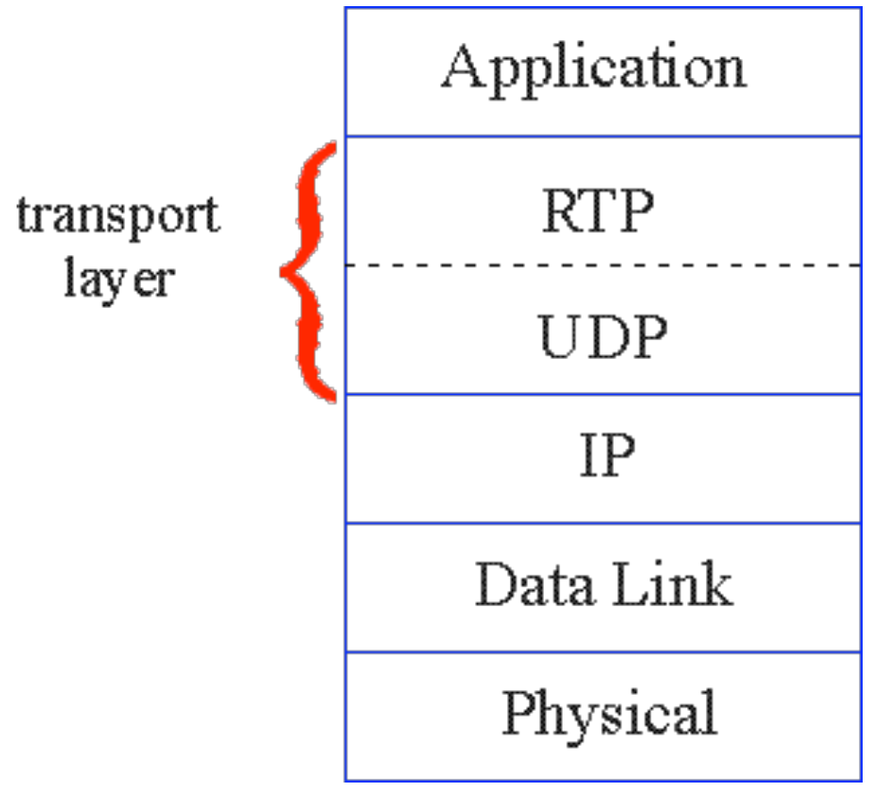
Recovery from packet loss (4)

Receiver-based repair of damaged audio streams

- produce a replacement for a lost packet that is similar to the original
- can give good performance for low loss rates and small packets (4-40 msec)
- simplest: repetition
- more complicated: interpolation

Real-Time Protocol (RTP)

- RTP specifies a packet structure for packets carrying audio and video data: RFC 1889.
- RTP packet provides
  - payload type identification
  - packet sequence numbering
  - timestamping



- RTP runs in the end systems.
- RTP packets are encapsulated in UDP segments
- Interoperability: If two Internet phone applications run RTP, then they may be able to work together

Real-Time Protocol (RTP)

- Provides standard packet format for real-time application
- Typically runs over UDP
- Specifies header fields below

RTP Header

- **Payload Type**: 7 bits, providing 128 possible different types of encoding; eg PCM, MPEG2 video, etc.
- **Sequence Number**: 16 bits; used to detect packet loss

RTP Header

Payload Type (7 bits): Used to indicate the type of encoding that is currently being used.

RTP Header

If a sender changes the encoding in the middle of a conference, the sender informs the receiver through this 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): The sequence number increments by one for each RTP packet sent; may be used to detect packet loss and to restore packet sequence.

Real-Time Protocol (RTP)

- **Timestamp**: 32 bytes; gives the sampling instant of the first audio/video byte in the packet; used to remove jitter introduced by the network

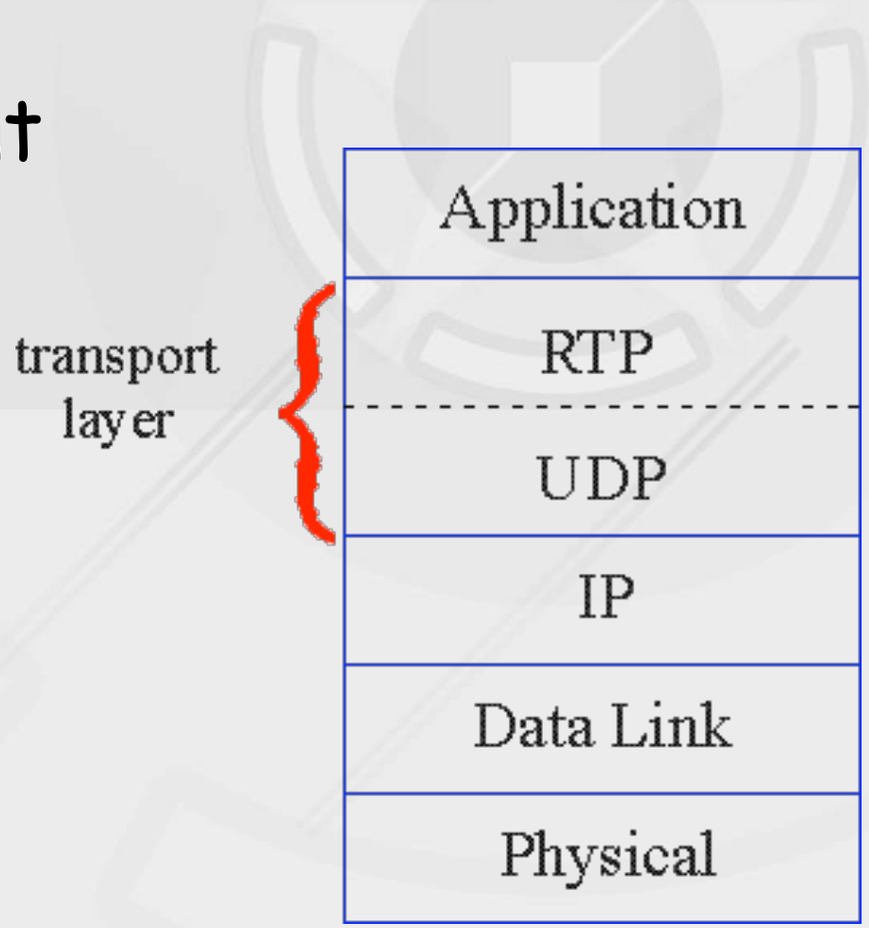
- **Synchronization Source identifier (SSRC)**: 32 bits; an id for the source of a stream; assigned randomly by the source

RTP Header

RTP runs on top of UDP

RTP libraries provide a transport-layer interface that extend UDP:

- port numbers, IP addresses
- error checking across segment
- payload type identification
- packet sequence numbering
- time-stamping



RTP Example

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

RTP and QoS

- RTP does **not** provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.
- RTP encapsulation is only seen at the end systems -- it is not seen by intermediate routers.
    - Routers providing the Internet's traditional best-effort service do not make any special effort to ensure that RTP packets arrive at the destination in a timely matter.
- In order to provide QoS to an application, the Internet most provide a mechanism, such as RSVP, for the application to reserve network resources.

RTP Streams

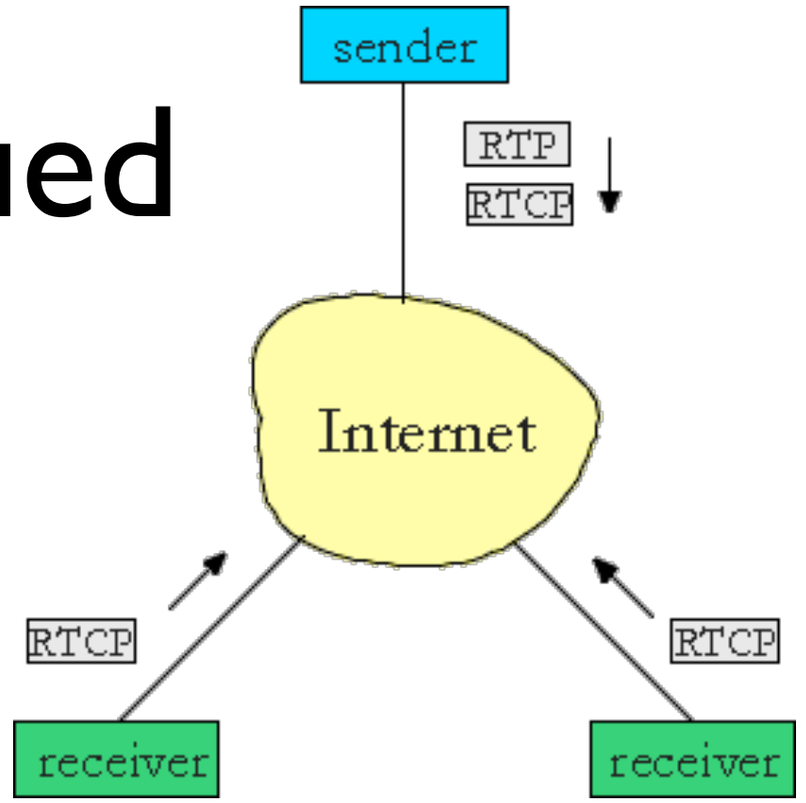
- RTP allows each source (for example, a camera or a microphone) to be assigned its own independent RTP stream of packets.
    - For example, for a videoconference between two participants, four RTP streams could be opened: two streams for transmitting the audio (one in each direction) and two streams for the video (again, one in each direction).

- However, some popular encoding techniques -- including MPEG1 and MPEG2 -- bundle the audio and video into a single stream during the encoding process. When the audio and video are bundled by the encoder, then only one RTP stream is generated in each direction.
- For a many-to-many multicast session, all of the senders and sources typically send their RTP streams into the same multicast tree with the same multicast address.

Real-Time Control Protocol (RTCP)

- Works in conjunction with RTP.
- Each participant in an RTP session periodically transmits RTCP control packets to all other participants. Each RTCP packet contains sender and/or receiver reports that report statistics useful to the application.
- Statistics include number of packets sent, number of packets lost, inter-arrival jitter, etc.
- This feedback of information to the application can be used to control performance and for diagnostic purposes.
- The sender may modify its transmissions based on the feedback.

RTCP - Continued



- For an RTP session there is typically a single multicast address: all RTP and RTCP packets belonging to the session use the multicast address.

- RTP and RTCP packets are distinguished from each other through the use of distinct port numbers.

- To limit traffic, each participant reduces his RTCP traffic as the 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 the RTP stream, the current time, the number of packets sent, and the number of bytes sent.

Source description packets:

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

Synchronization of Streams

- RTCP can be used to synchronize different media streams within a RTP session.

- Consider a videoconferencing application for which each sender generates one RTP stream for video and one for audio.

- The timestamps in these RTP packets are tied to the video and audio sampling clocks, and are not tied to the wall-clock time (i.e., to real time).

- Each RTCP sender-report packet contains, for the most recently generated packet in the associated RTP stream, the timestamp of the RTP packet and the wall-clock time for when the packet was created. Thus the RTCP sender-report packets associate the sampling clock to the real-time clock.

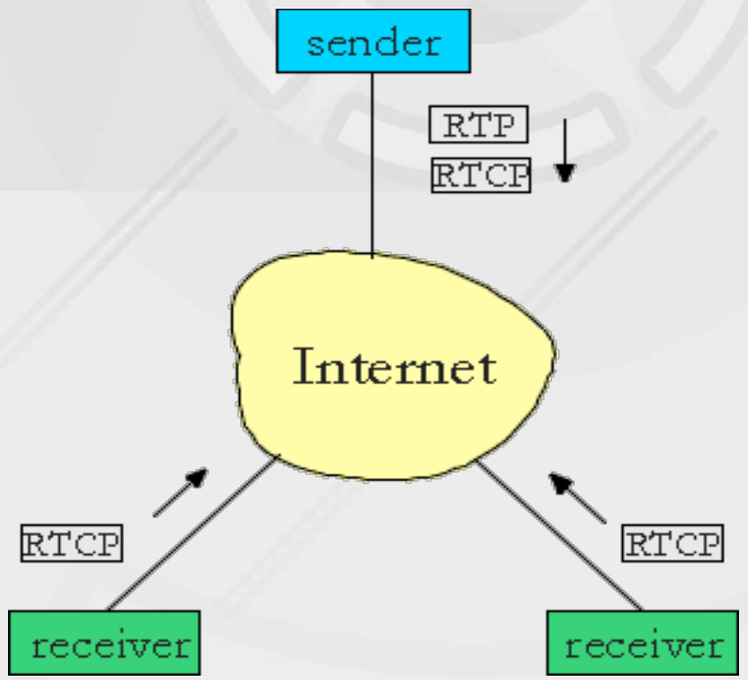
- Receivers can use this association to synchronize the playout of audio and video.

RTCP Bandwidth Scaling

- RTCP attempts to limit its traffic to 5% of the session bandwidth.
- For example, suppose there is one sender, sending video at a rate of 2 Mbps. Then RTCP attempts to limit its traffic to 100 Kbps.
- The protocol gives 75% of this rate, or 75 kbps, to the receivers; it gives the remaining 25% of the rate, or 25 kbps, to the sender.
- The 75 kbps devoted to the receivers is equally shared among the receivers. Thus, if there are R receivers, then each receiver gets to send RTCP traffic at a rate of 75/R kbps and the sender gets to send RTCP traffic at a rate of 25 kbps.
- A participant (a sender or receiver) determines the RTCP packet transmission period by dynamically calculating the the average RTCP packet size (across the entire session) and dividing the average RTCP packet size by its allocated rate.

RTP Control Protocol (RTCP)

- Protocol specifies report packets exchanged between sources and destinations of multimedia information
- Three reports are defined: Receiver reception, Sender, and Source description
- Reports contain statistics such as the number of packets sent, number of packets lost, inter-arrival jitter
- Used to modify sender transmission rates and for diagnostics purposes



RTCP Bandwidth Scaling

- If each receiver sends RTCP packets to all other receivers, the traffic load resulting can be large
- RTCP adjusts the interval between reports based on the number of participating receivers
- Typically, limit the RTCP bandwidth to 5% of the session bandwidth, divided between the sender reports (25%) and the receivers reports (75%)

Reference

- Fundamentals of Multimedia, Chapter 16
  - 机械工业出版社
-