Outline

1.	Introduction and Motivation	
2.	Interactive Web Applications	Part I: Web Technologies for Interactive MM
3.	Web Programming with Java	
4.	Communities, the Web, and Multimedia	
5.	Digital Rights Management	Part II: Content-Oriented Base Technologies
6.	Cryptographic Techniques	
7.	Multimedia Content Description	
8.	Electronic Books and Magazines	Part III: Multimedia Distribution Services
9.	Multimedia Content Production and Management	
10.	Streaming Architectures	
11.	Web Radio, Web TV and IPTV	
12.	Multimedia Conferencing	Part IV: Conversational Multimedia Services
13.	Signaling Protocols for Multimedia Communication	
14.	Visions and Outlook	

10 Streaming Architectures

Literature:

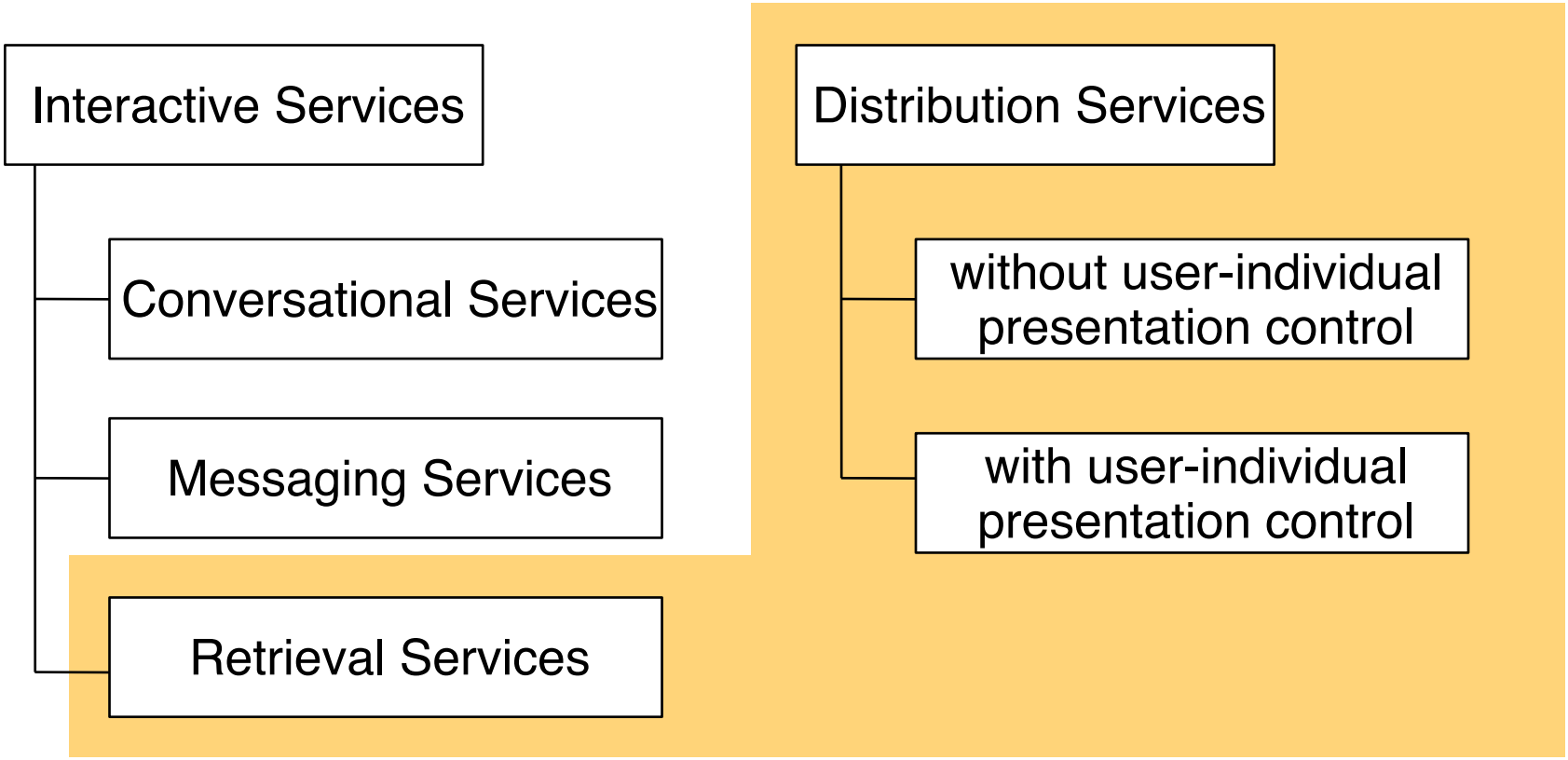
David Austerberry: The Technology of Video & Audio Streaming, Focal Press 2002

Gregory C. Demetriades: Streaming Media, Wiley 2003

Tobias Künkel: Streaming Media – Technologien, Standards, Anwendungen, Addison-Wesley 2001

A Classification of Multimedia Services

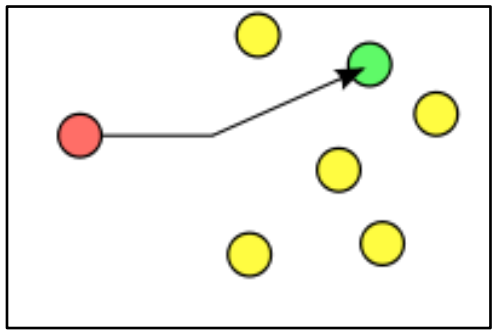
- According to ITU-T recommendation I.211 “B-ISDN Service Aspects”



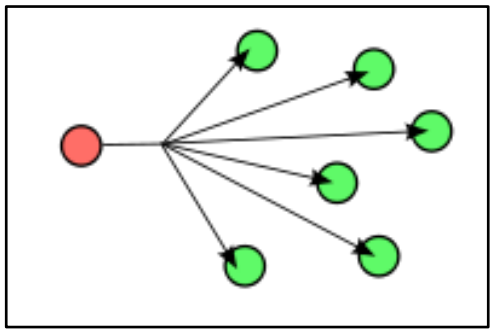
Presentation Control in Distribution Services

- **Without** user-individual presentation control:
  - Continuous “live” stream of information from sender to receiver(s)
  - Replacement of other distribution media (e.g. radio) by digital networks
    - » Real-time service (e.g. fixed start time for transmission)
- **With** user-individual presentation control:
  - Pause, resume, skip backward, skip forward
  - **Server** control:
    - » Individual stream (or group of closely related streams)
  - **Client-local** control:
    - » Outdated solution for interactivity: “Near Video-on-Demand” (NVOD) = Staggered broadcast of multiple transmissions of the same content
    - » Time-shifted recording enhances interactivity (pause/resume)
  - Full transmission of video content on local storage before playback enables full interactivity
    - » Modern solutions enabled by high-capacity local disks (e.g. AppleTV, Sky Anytime)

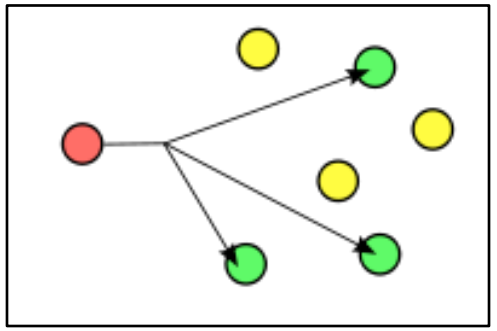
Unicast, Broadcast, Multicast, Anycast



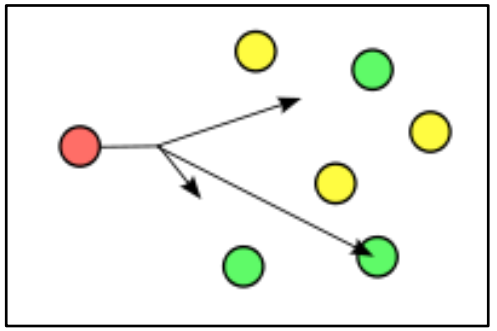
Unicast:
One specific receiver



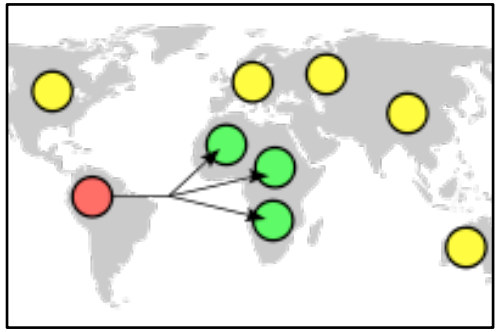
Broadcast:
Many receivers, all on the network



Multicast:
Many receivers, all of a specific group



Anycast:
One receiver, "nearest" of a specific group



Geocast:
Many receivers, all of a geographic region

Pictures: Wikipedia

Bandwidth Economy

- Fully heterogeneous individual requests:
  - Required bandwidth = stream bandwidth x number requests
- Homogeneity of request helps saving bandwidth:
  - Same content for many clients, but different playback times:
    - » Broadcast with local buffering
  - Same content at same time for many clients
    - » Multicast (splitting streams)
- Pre-planning saves bandwidth
  - (Individual) transmission of pre-booked content in non-real time ("download and play")

Streaming, Streaming Media

- *Streaming media* means real-time delivery of moving images, moving text and sound, from a server to client (over the Internet).
- Delivery types for audio and video content:
  - *Download and Play:* Content must be downloaded completely to the client before it can be played
  - *Progressive Download:* Playback is started while download is still in progress. Download rate independent of program bit rate.
  - *True Streaming:* Delivered media is viewed/listened in “real-time”.
    - » Playback takes place with roughly the same rate as delivery of data
    - » Delay between send and receive event of data packet kept small
- Subtypes of True Streaming:
  - *Static File Streaming:* Delivery of pre-recorded media files.
    Often also called *on-demand* delivery (e.g. *Video on Demand)*
  - *Live Streaming*

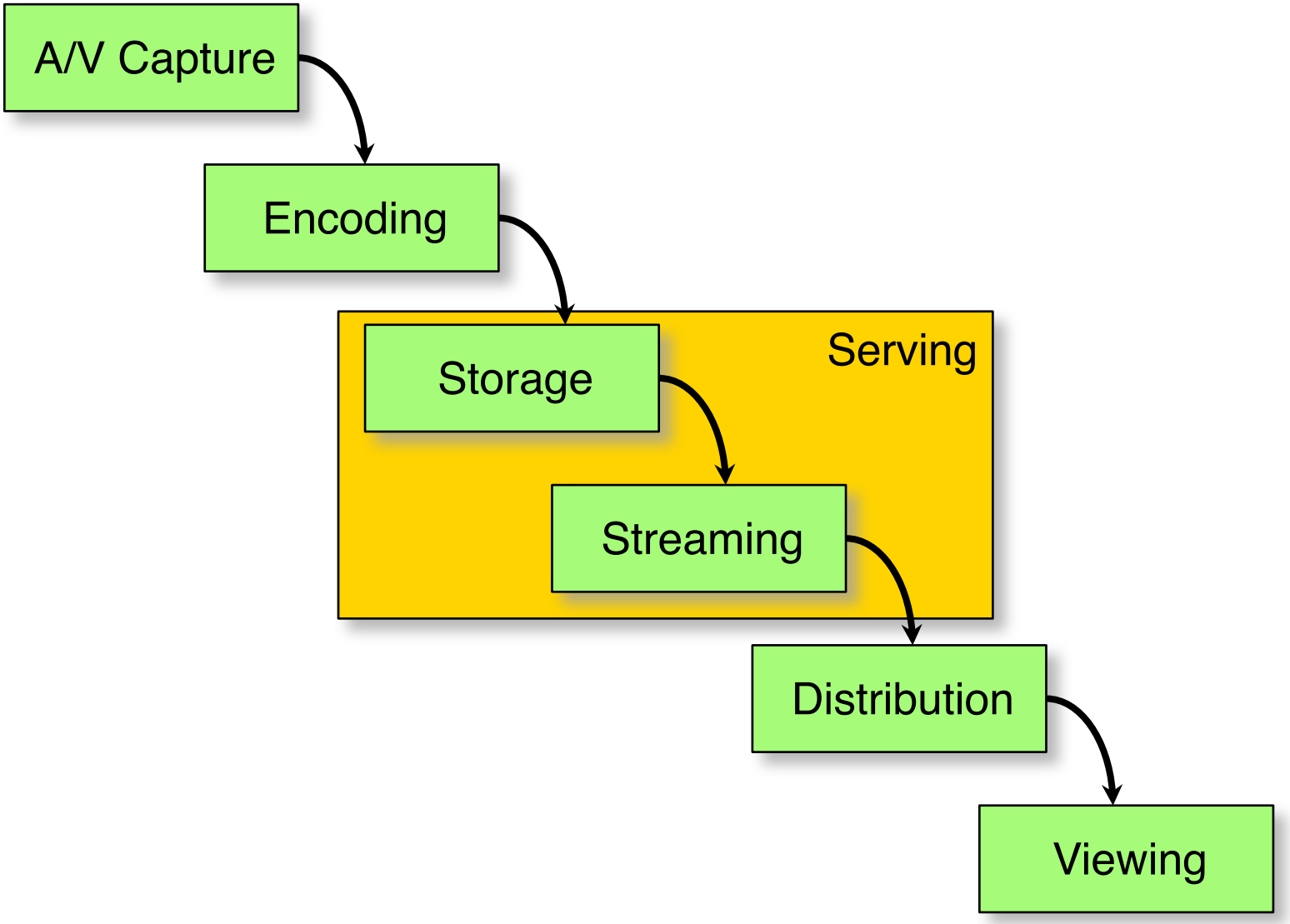
Based on material from www.streamingalliance.org

Session

- A *session* is an association between communicating parties, which
  - Persists over a limited time span
  - Incorporates at least two parties
  - May comprise a large number of communication connections of different characteristics
- Examples of sessions:
  - Movie streamed to consumer, consisting of audio and video parts
  - Multimedia conference among five participants, consisting of audio and video source from each of of the participants (plus possibly some global information)

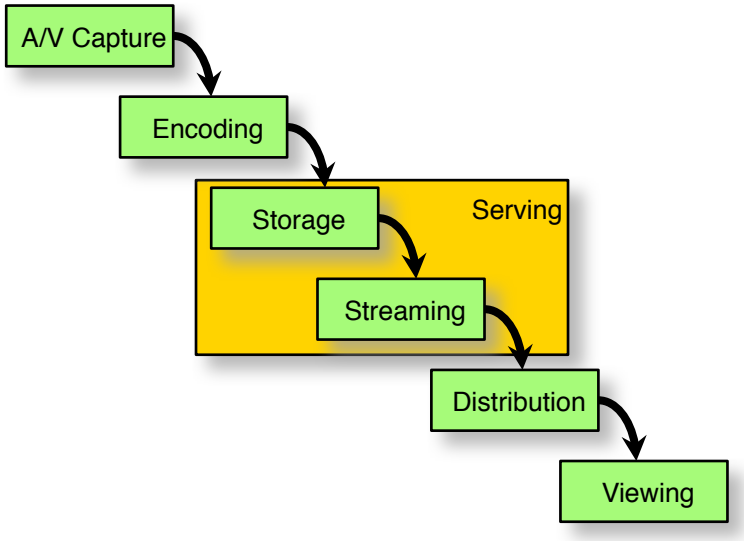
- Session awareness at which levels?
  - At application level: unavoidable
  - At network level: possible
    - » Requires specific protocols

Streaming Delivery Chain

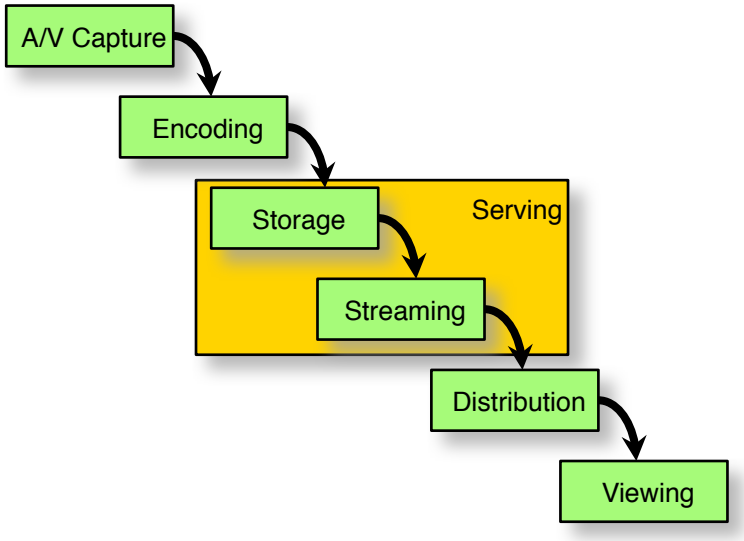


Encoding

- Format conversions
  - E.g. analog/digital conversion
  - E.g. downscaling of picture size
- Compression
  - Adequate for player capabilities and typical transmission bandwidth
- Indexing
  - Analyzing internal structure
- Metadata creation
  - Possibly including digital rights specification

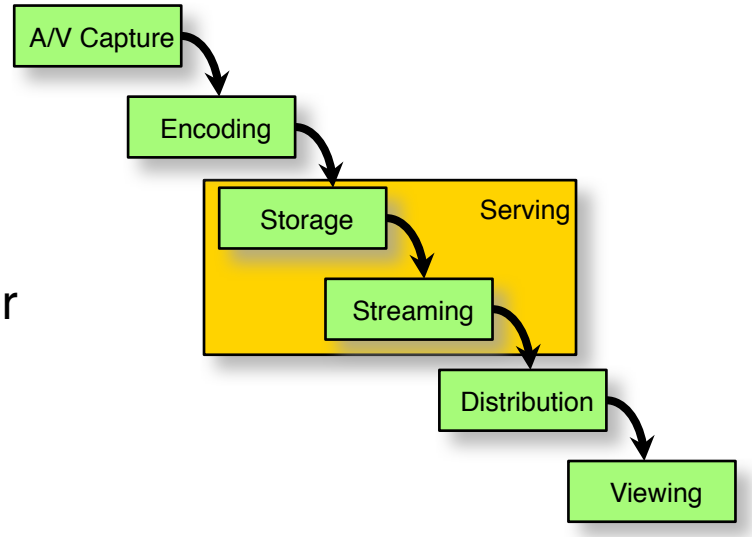


Serving



- Storage
  - Live: only buffering and archiving
  - Static files: archive management, retrieval
- Streaming
  - Pull model:
    - » Request-response driven similar to Web server
  - Push model:
    - » Server sends packets at regular intervals
- Interactivity
  - VCR-like control (PLAY, STOP, PAUSE, FFWD, REW)
  - Random access based on various criteria
  - Hyperlinks in A/V material (“hypervideo”)
- Transcoding
  - Encoding adapted to client capabilities

Distribution



- Key topic: Quality of Service (QoS)
  - Determining realizable bandwidth, delay, jitter

- Key concepts:
  - Overprovisioning
  - Detailed reservations (“Integrated Services”, reservation protocol RSVP)
    - » Difficult to scale to large numbers of users
  - Traffic classes (“Differentiated Services”)
    - » Difficult to control access to privileges
  - Resource management layer
  - Technology-specific solutions
    - » E.g. ATM (Asynchronous Transfer Mode)

10 Streaming Architectures

Literature:

David Austerberry: The Technology of Video & Audio Streaming, Focal Press 2002

Stephan Rupp, Gerd Siegmund, Wolfgang Lautenschlager: SIP – Multimediale Dienste im Internet. dpunkt 2002

A. Begen, T. Akgul, M. Baugher: Watching Video over the Web, Part I: Streaming Protocols, *IEEE Internet Computing*, March/April 2011

IP and TCP

- Internet Protocol
  - Network communication protocol (ISO layer 3)
  - Packets transferred from address to address (through routers)
  - Main problems:
    - » Variable network latency
    - » Packet order on arrival may be different than on sending
    - » Packets may be lost
- Transport Control Protocol (TCP)
  - Connection establishment (by “three-way handshake”)
    - » Connections are sequences of associated IP packets
  - Sequencing of bytes with forwarding acknowledgement number
  - Non-acknowledged bytes are re-transmitted after a defined time period
  - Flow control
- For audio/video streaming:
  - Retransmissions (and associated delays) are harmful
  - Lost packets can be tolerated to some extent

UDP

- User Datagram Protocol (UDP)
- Extremely simple transport protocol over IP
  - Connectionless (TCP: connection-oriented)
  - Unreliable (TCP: reliable)
  - No flow control (TCP: has flow control)
- Contents of a UDP datagram:
  - Ports used by application program
  - Checksum
- Basically adequate for media data transport
  - In particular for **pull**-model true streaming
  - Very efficient, protocol overhead of TCP avoided
  - Flow control and handling of packet loss have to be handled by higher protocol layer

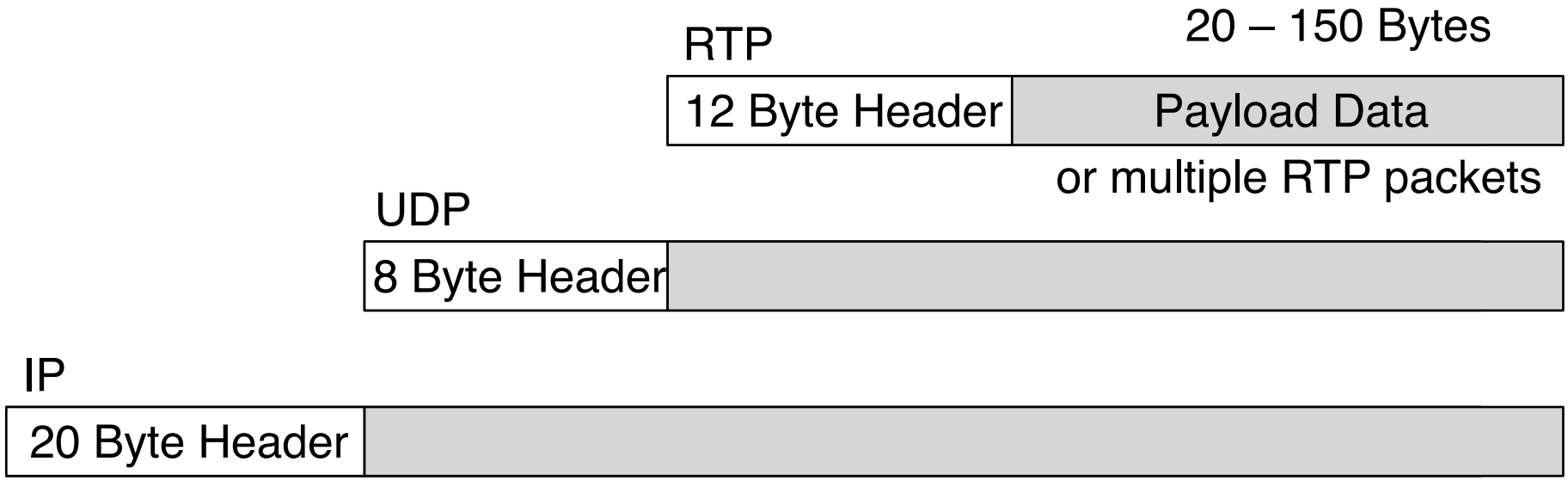
Streaming with HTTP?

- **Pull** model:
  - Client requests next slice of audio/video information from server
  - Server responds with audio/video data
- Very popular in Web-based multimedia services
  - e.g. YouTube video
  - e.g. MP3-Streaming with ShoutCast, Live365
- Advantage:
  - Ubiquitous protocol, no problems with firewalls etc.
- Main problems:
  - HTTP usually run over TCP, creates overhead
  - Progressive download rather than true streaming
  - Sometimes user has to select requested quality/bitrate
- Consequence:
  - Specialized streaming protocols: “Adaptive HTTP Streaming” (see later)

Real-Time Transport Protocol RTP

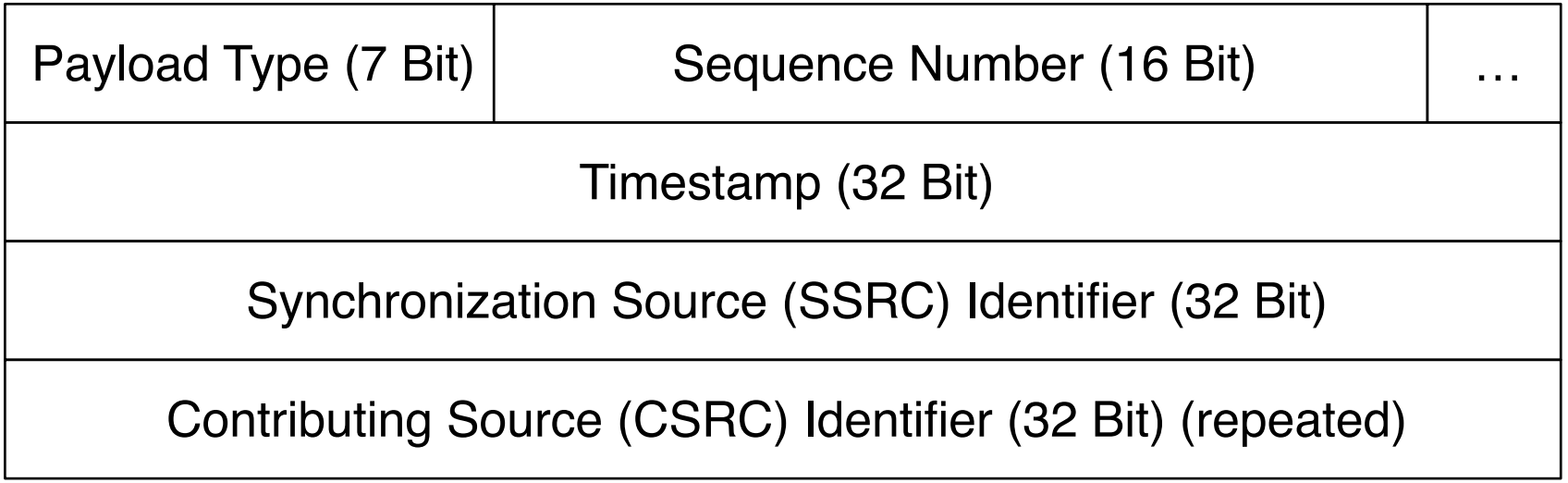
- Transport protocol specifically developed for streaming data
  - IETF (Internet Engineering Task Force) RFC (Request for Comments) 1889
- RTP packets contain
  - Sequence number
  - Time stamp
  - Identification of sender and destination
- RTP usually carried over UDP
- Very important:
  - **RTP does not at all change the way how IP packets are transferred in the network!**
  - To achieve “Quality of Service”, additional network technologies are required (see above)
- RTP used (for instance) by:
  - Apple QuickTime architecture
  - RealSystems streaming architecture

RTP Packets and Other Protocols



- IP Header:
  - Source address, destination address, length, time to live, ...
- UDP Header:
  - Port numbers (source and target processes), length, checksum

RTP Header Format

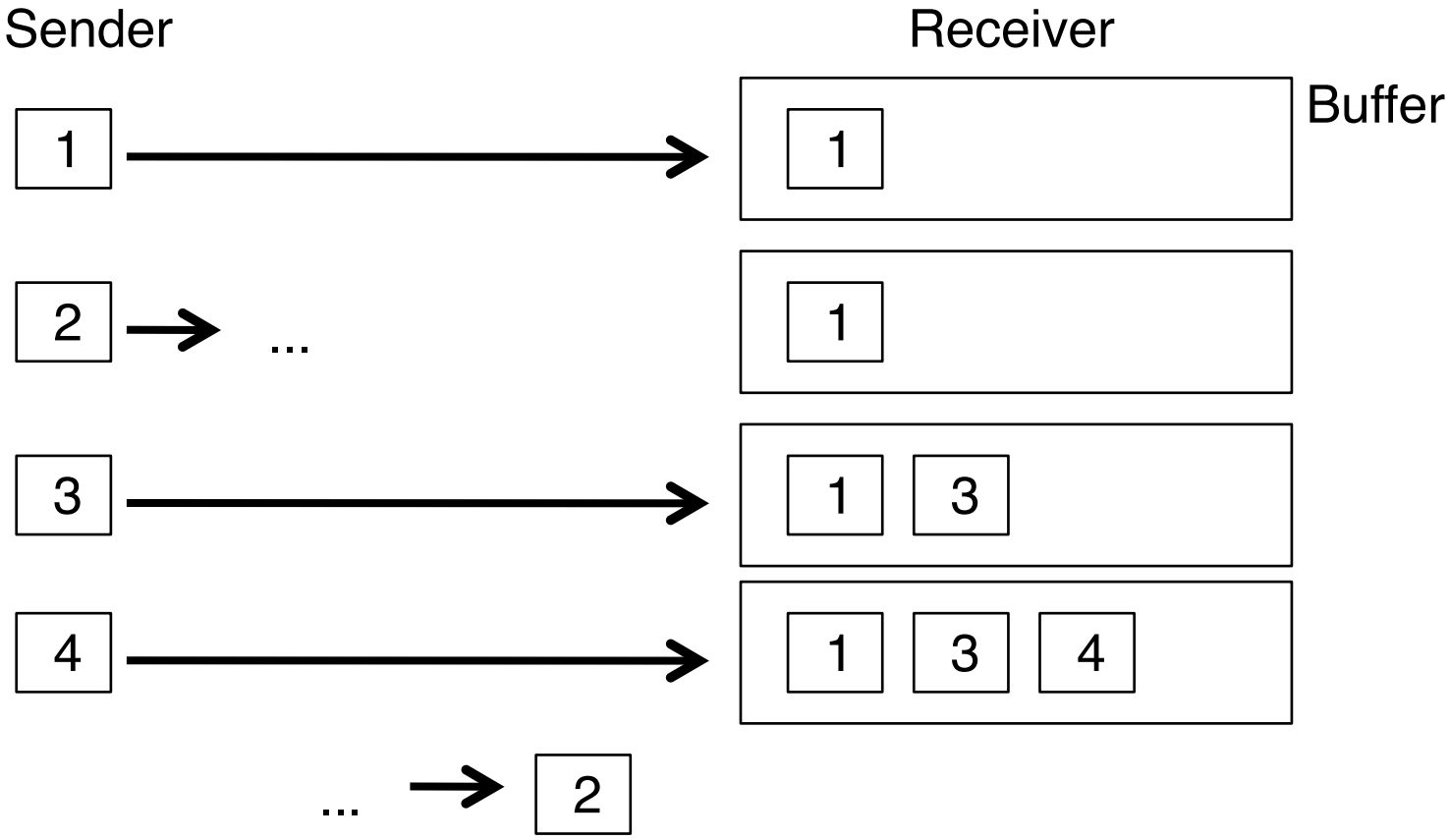


- Payload Type: Source coding (codec used)
- SSRC: Identification of sending data source, defined basis for sequence numbers and time stamps
- CSRC (optional): Identifications for original data sources which have been *mixed* together to form the synchronization source

Buffer Size Allocation

- Streaming quality relies on
  - Features of the network, and
  - *Adaptive* codecs on client/server side
  - E.g. intelligent use of buffering
- *Constant* delay does not require buffering
- Buffers are necessary to deal with *jitter (packet delay variation)*
  - Varying network delays
- Estimation of buffer sizes
  - Based on preliminary test transmissions
  - Adaptively during content transmission
- Problem:
  - Keep buffers small to achieve proper “real-time”
  - Avoid “buffer underrun” in case of exceptionally long delays

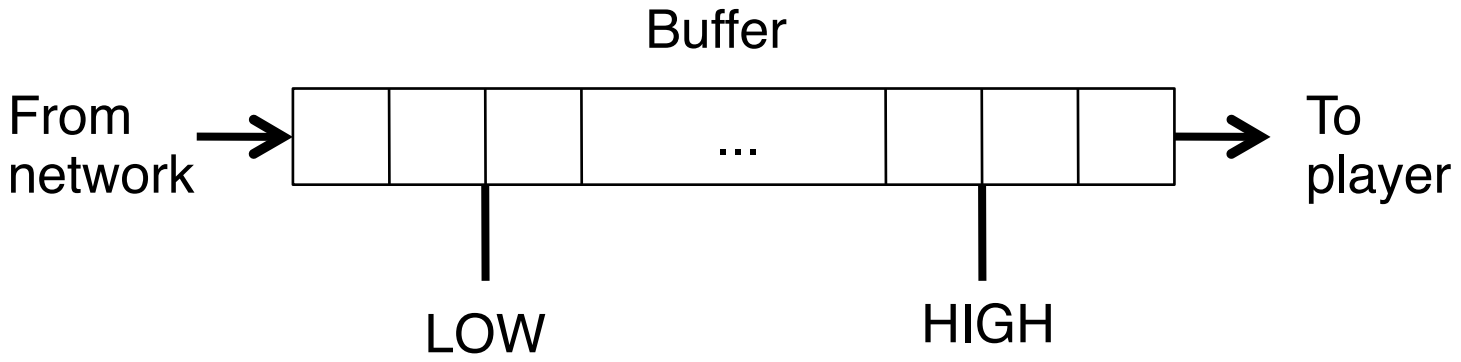
Jitter and Loss Compensation



- Options for application on receiver side:
  - Wait *(not* adequate), repeat last packet (1), interpolate (between 1 and 3)
  - Missing audio information is difficult, missing video can be compensated

Adaptive Transmission Rate Control

- Application-level mechanism
- Define “low” and “high” thresholds on buffer
- Communication between client and server
  - Decrease effective transmission rate when high threshold is reached
  - Increase effective transmission rate when low threshold is reached
- Changing the effective transmission rate
  - Change sending rate
  - Change content quality (frame rate, resolution etc.)



Real-Time Control Protocol RTCP

- RTCP controls the transmission (not the setup of connection)
- RTCP periodically sends monitoring information to all participants in a streaming session
- Main functions of RTCP:
  - Feedback on QoS of transmission
    - » Information for adaptive codecs, e.g. whether problem is local or global
  - Identification of sender by “canonical name”
    - » Helpful when synchronization source changes
    - » Supports lip synchronization between audio and video
  - Number of participants in a session
    - » Adaptation of sending rate of RTCP control information to number of participants, to avoid network overload
  - Transmission of additional information, e.g. names of session participants

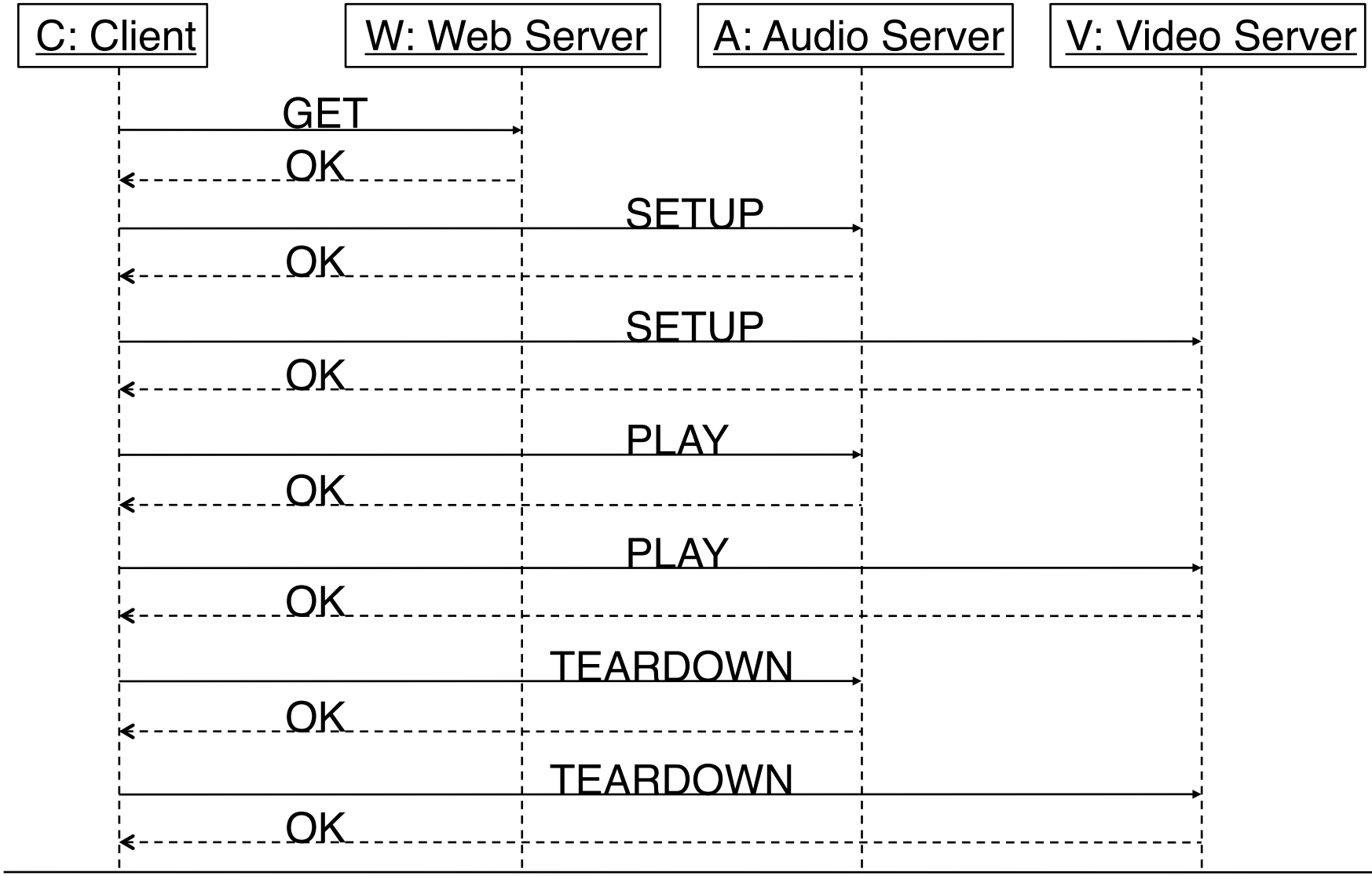
Real Time Streaming Protocol RTSP

- Client-server multimedia presentation protocol, designed specifically for streamed media
  - IETF (Internet Engineering Task Force) RFC (Request for Comments) 2326 (“MMUSIC” work group)
    - » February 1998, draft revision February 2004
  - “The Internet VCR remote control protocol” (www.rtsp.org)
  - Independent of the use of RTP for transport
  - Syntactically similar to HTTP 1.1 (carried over TCP or UDP)
- Main operations supported by RTSP:
  - Retrieval of media from media server
  - Invitation of a media server to a conference
- Key terminology
  - Aggregate control (e.g. for audio & video)
  - Server control (clients should be able to stop streaming from a server)
  - Transport & capability negotiation (e.g. disallowing a “seek” function)

Main Methods of RTSP

- SETUP:
  - Causes the server to allocate resources for a stream and create a RTSP session.
- PLAY:
  - Starts data transmission on a stream allocated via SETUP
  - Fast forward (scale ratio parameter)
- PAUSE:
  - Temporarily halts a stream without freeing server resources.
- REDIRECT:
  - Indicates that the session should be moved to new server / location
- PING:
  - Prevents the identified session from being timed out.
- TEARDOWN:
  - Frees resources associated with the stream. The RTSP session ceases to exist on the server.

Example Session with RTSP



10 Streaming Architectures

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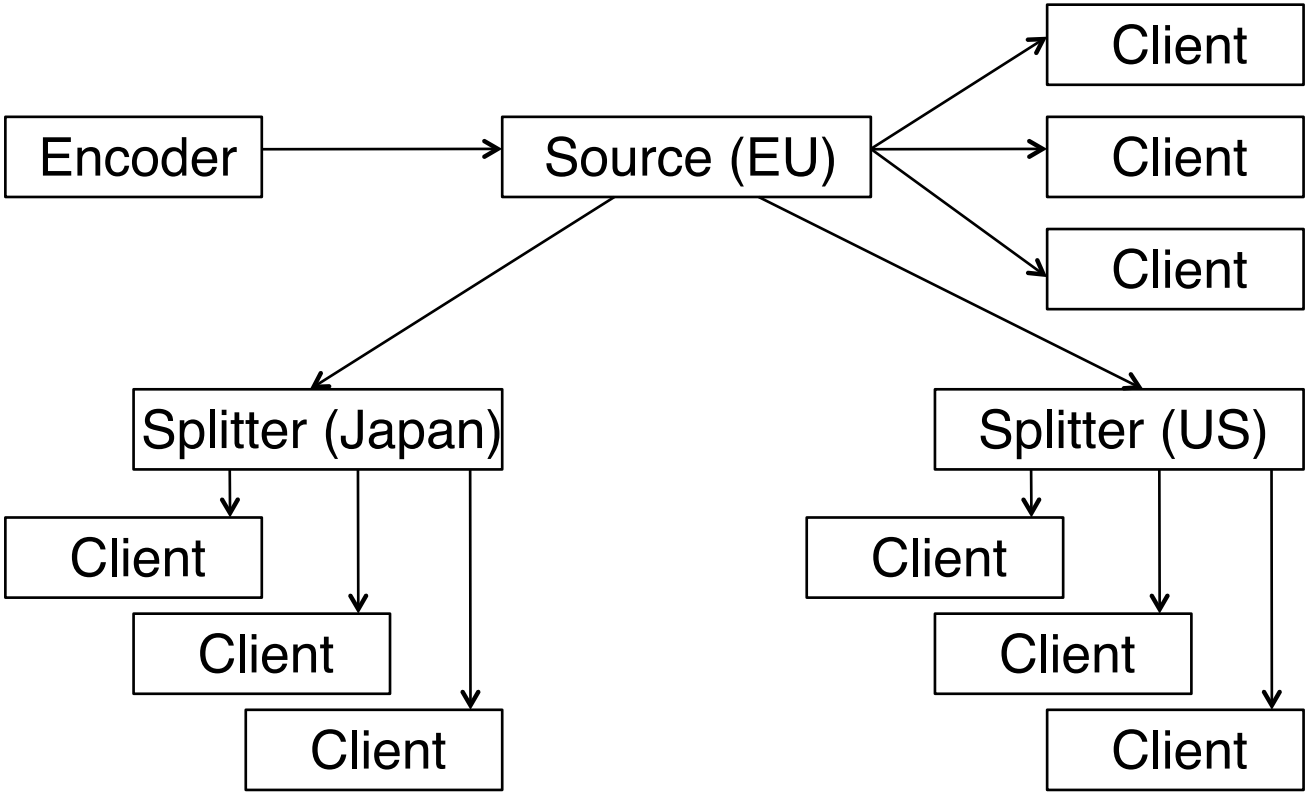
Gregory C. Demetriades: Streaming Media, Wiley 2003
Xueyan Tang et al.: Web Content Delivery, Springer 2005

Splitting

- Video servers are limited in capacity
- Assuming clients at spatially distant locations
  - Intermediate, forwarding server is useful: “splitter”



Content Delivery Networks (CDN)

- Serve content closer to to the user
  - "edge serving"
- Main components of CDN:
  - Smart routing
  - Edge caching of content
  - Proxy serving
  - Splitting of live webcasts

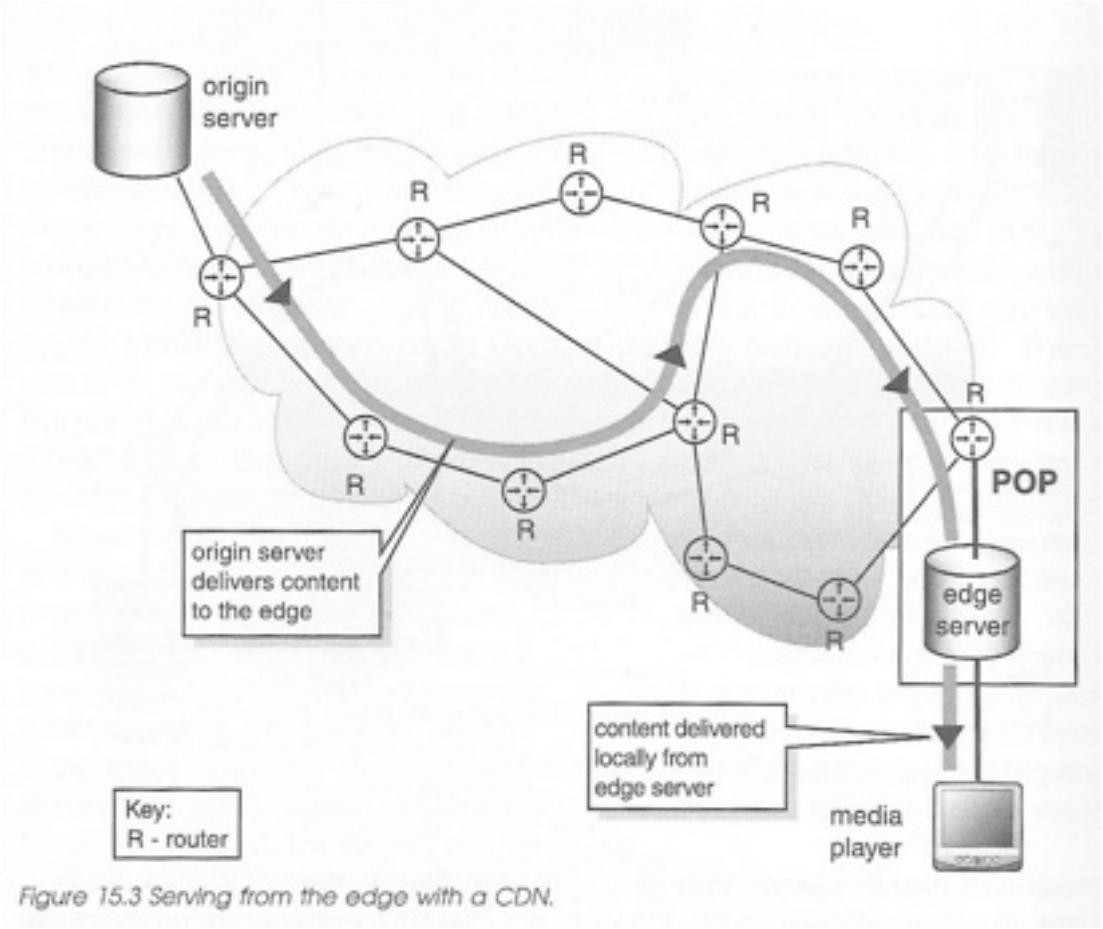


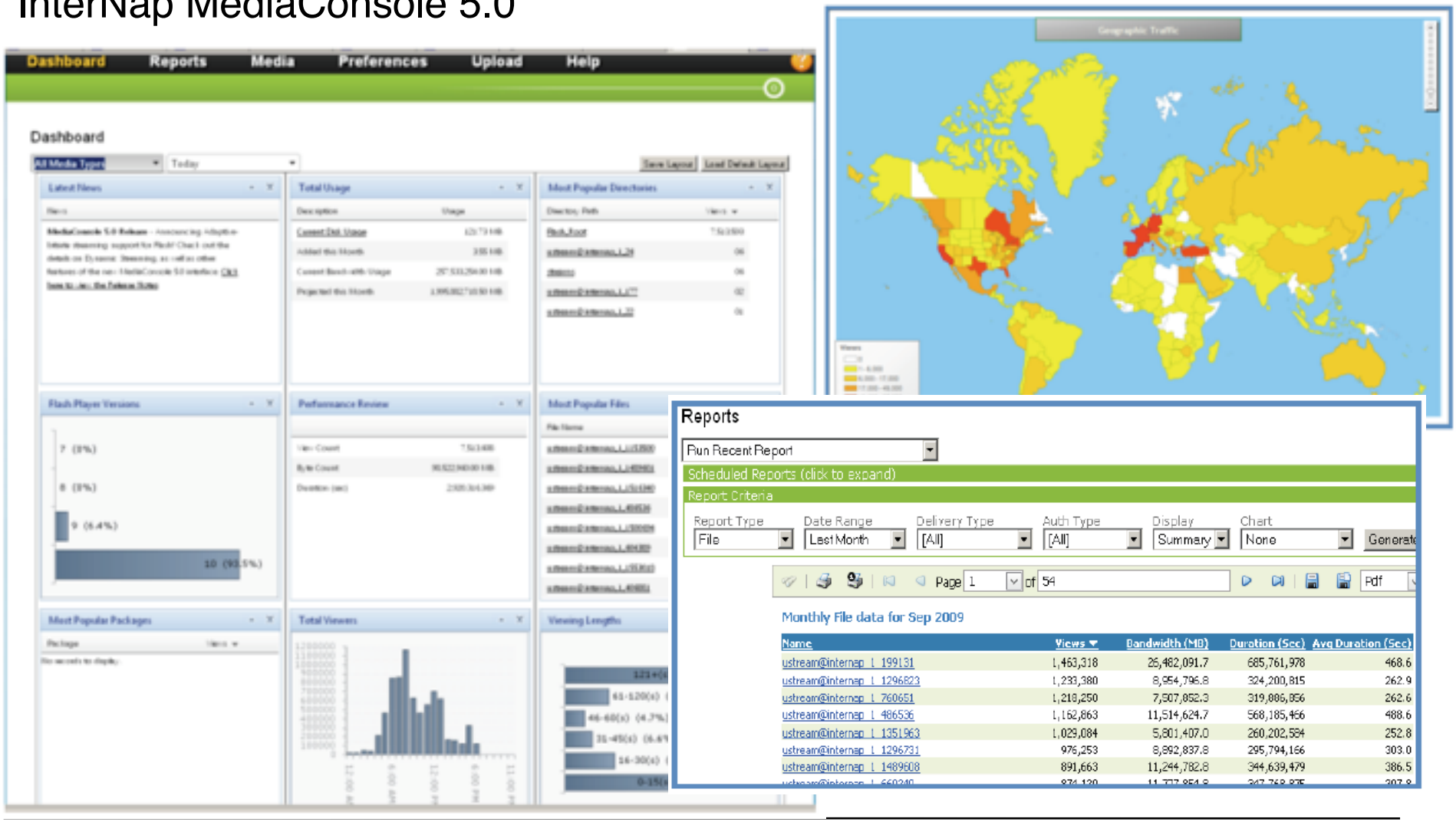
Figure from Austerberry 2002

Content Delivery Networks

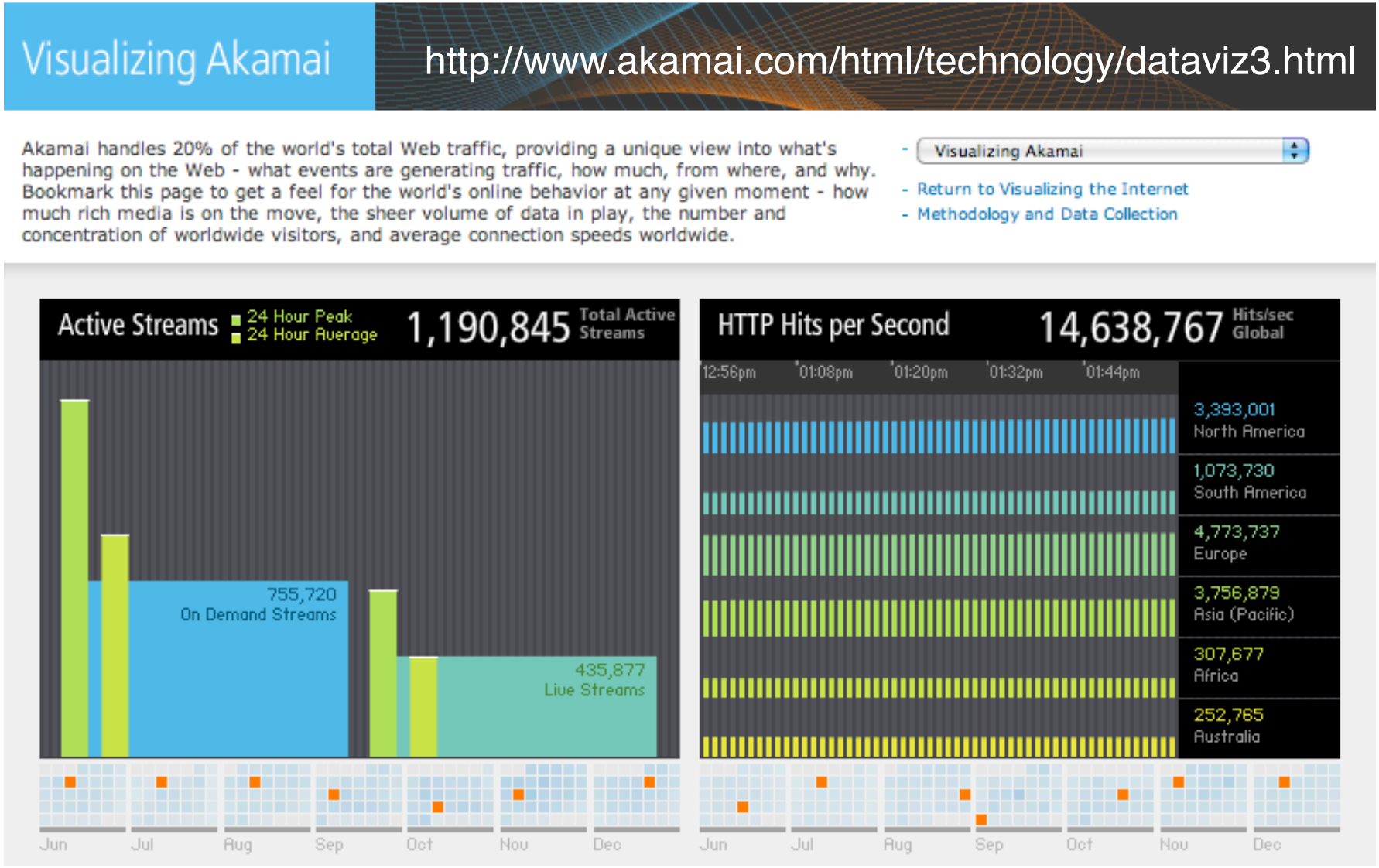
- “Overlay server infrastructure”
  - Network of centrally operated Web and streaming servers
  - Geographically distributed, present in main ISP networks
  - Flexibly used for content from various sources
- Content delivery as a service
  - Content delivery service provider owns server infrastructure
  - Content owner pays for having content delivered to customers
- Examples:
  - Akamai (delivers 20% of all Web traffic according to akamai.com, 2012)
    - » Runs 95,000+ servers in 71 countries
  - InterNap CDN Services
    - » Traditionally specialized in media streaming (acquisition of VitalStream)
  - Amazon CloudFront
- Streaming of a/v media (e.g. movies) gains strategic importance

Example of Monitoring/Administration Interface

InterNap MediaConsole 5.0



Example: Visualizing Akamai



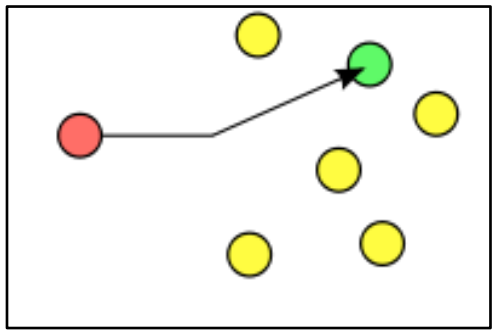
Key Problems in CDNs

- Replica placement:
  - Where to place copies of web sites or other content
  - Problem is in general NP-hard (Karlsson, Karamolis, 2004)
  - Replica placement algorithms (RPA) achieve a suboptimal solution within reasonable time frame
  - Global information is difficult or costly to get - RPA uses local information mostly
  - CDN providers typically try to observe global network performance to some extent
- Request routing:
  - Mechanism and policy of redirecting client requests to a suitable server containing the requested content
  - Redirection algorithm: Decides what node to direct a client request to
  - Redirection mechanism: Way of redirecting the request (client, network)

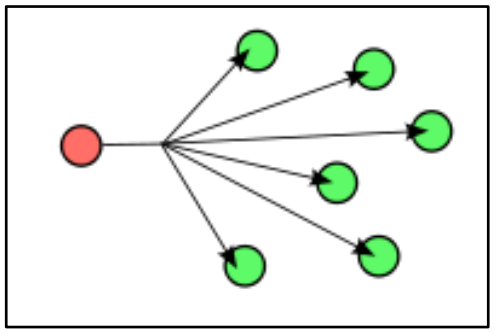
Streaming Media in CDN

- General idea: Local proxy caching. But: ...
  - Huge size (1 KB vs. 100 MB)
    - » To cache only portions of the original?
  - Intensive bandwidth usage
    - » Minimizing bandwidth consumption as primary consideration
  - High interactivity
    - » E.g. premature termination is frequent (Chen et al. 04: approx. 90 %)
  - However: Media content is rather static (compared to Web pages)
- Caching algorithms
  - Different for homogeneous and heterogeneous clients (in bandwidth/format)
  - Sliding interval caching: sequential access, mainly effective for similar requests in short time period
  - Prefix caching: Saves time to load remaining parts
  - Segment caching: Generalization of prefix caching to support fast forward
  - Rate-split caching: Lower layer from original server, higher layer from proxy
  - Co-operative proxy caching (e.g. Acharya/Smith: MiddleMan)

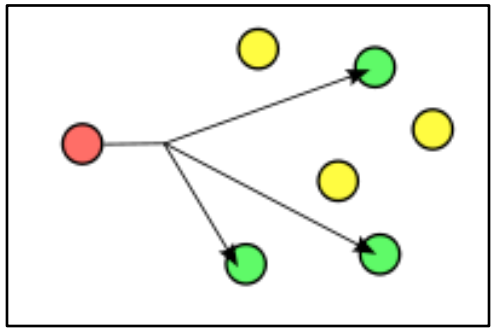
Unicast, Broadcast, Multicast, Anycast



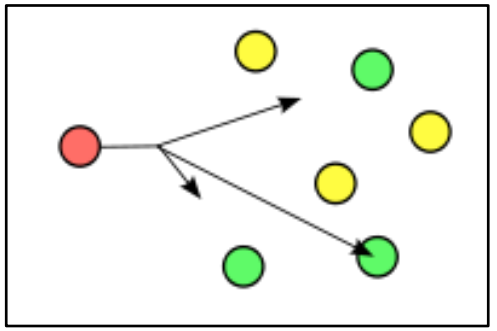
Unicast:
One specific receiver



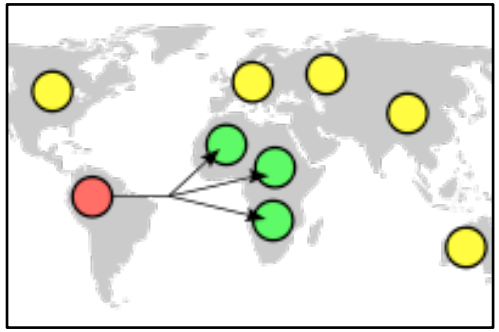
Broadcast:
Many receivers, all on the network



Multicast:
Many receivers, all of a specific group



Anycast:
One receiver, "nearest" of a specific group

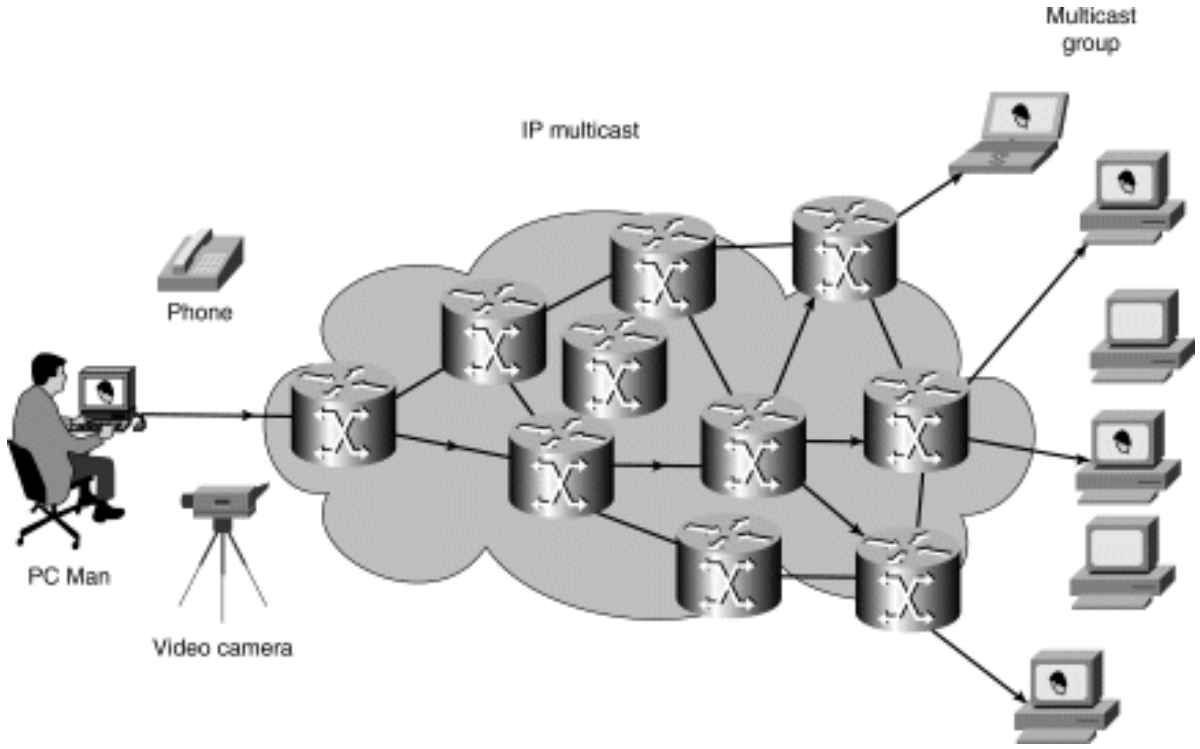


Geocast:
Many receivers, all of a geographic region

Pictures: Wikipedia

IP Multicast

- Multicast relatively easy to integrate in routers
- IP address class D (224.0.0.0 through to 239.255.255.255) reserved for multicast (multicast groups)
- Registration/deregistration with IGMP (Internet Group Management Protocol)



- Reliable multicast: e.g. “Mbone” overlay network
- Multicast still rarely used in today’s Internet

IP Version 6

- Next generation of the IP protocol
- 128 Bit address space
  - Intended to relieve shortage of IP v4 addresses
- Built-in support for multicast
  - Specific multicast addresses
- Uptake of IP version 6 is (strangely) slow

10 Streaming Architectures

10.1 High-Level Streaming Architecture

10.2 Real-Time Data Transport

10.3 Scalability and Multicast

10.4 Selected Commercial Streaming Architectures

RealNetworks, QuickTime, Windows Media

Literature:

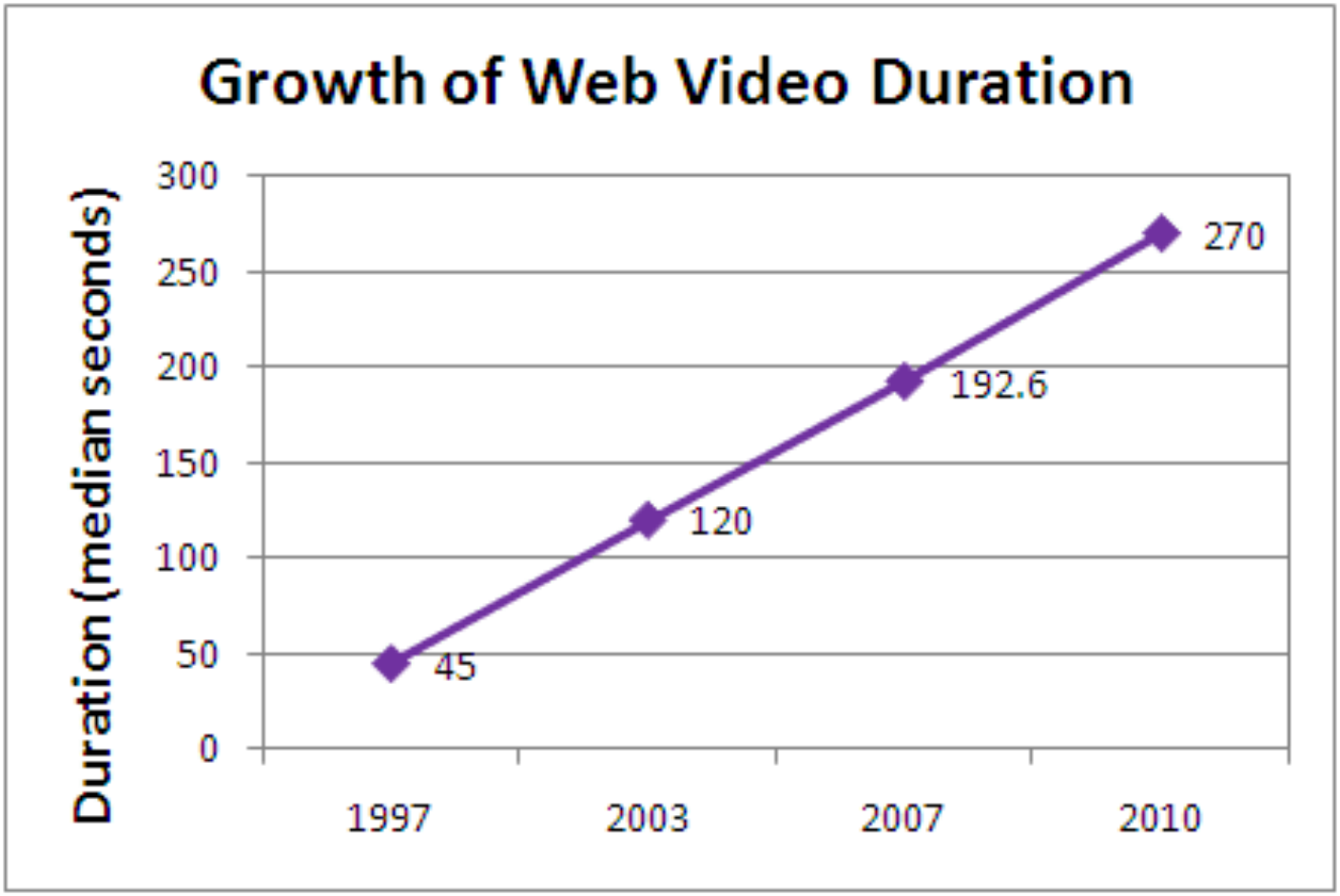
Tobias Künkel: Streaming Media – Technologien, Standards, Anwendungen, Addison-Wesley 2001

David Austerberry: The Technology of Video & Audio Streaming, Focal Press 2002

Usage of Video Streaming (1)

Marc Gosschalk, comscore.com, October 2011

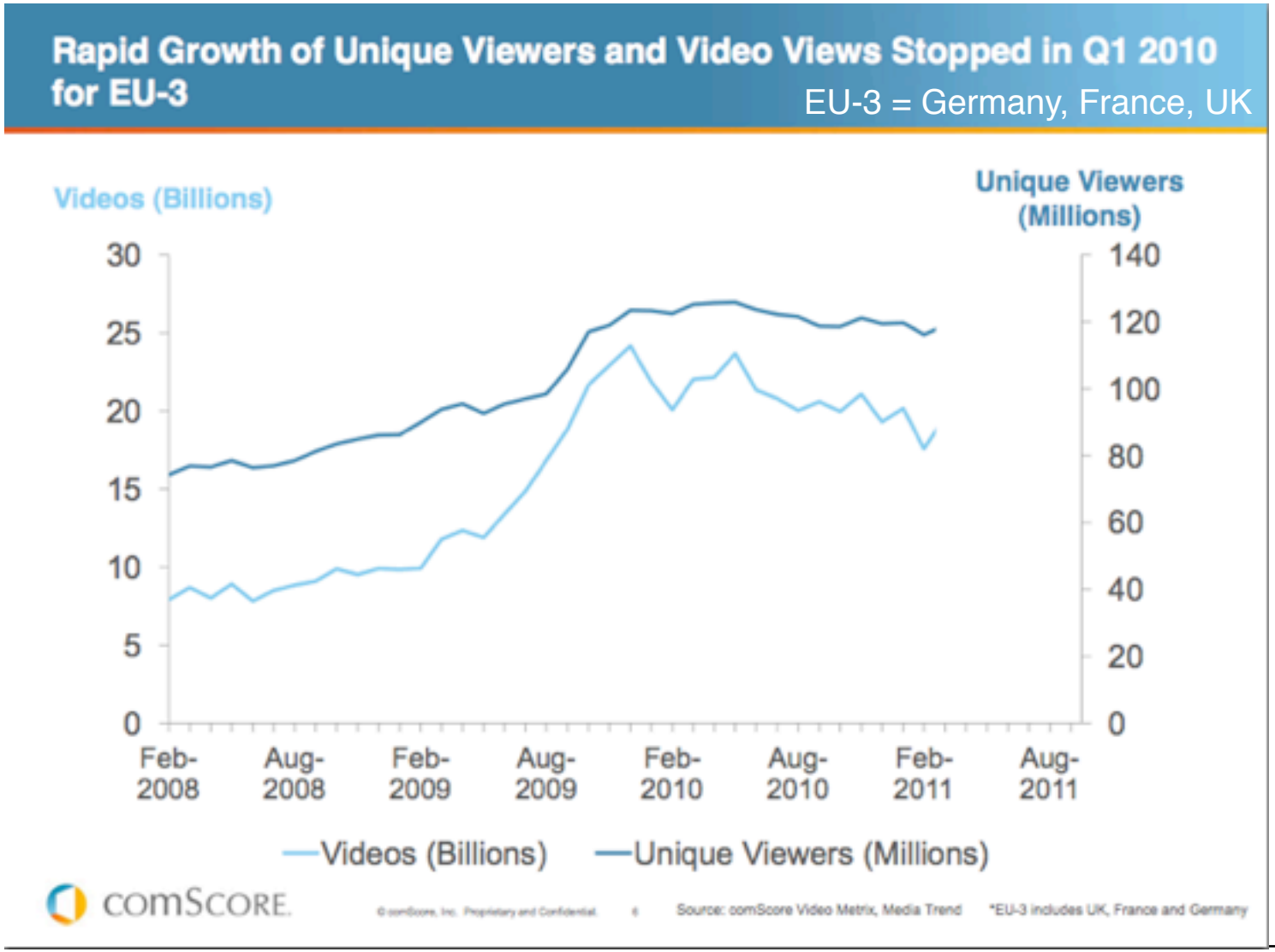
Increasing Size of Video Files



Growth
in length and
in quality!

Source: WebSiteOptimization.com

Usage of Video Streaming (2)



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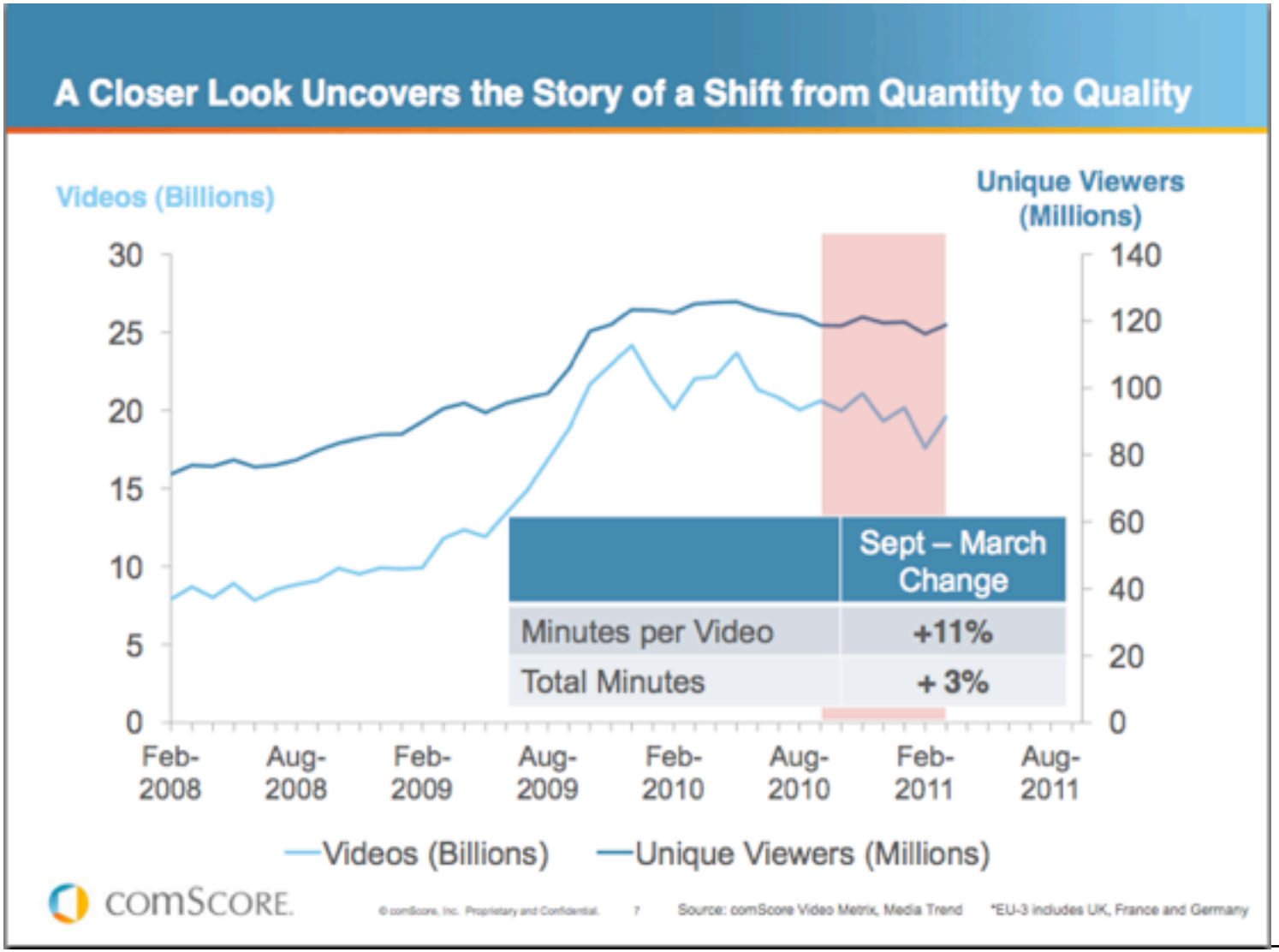
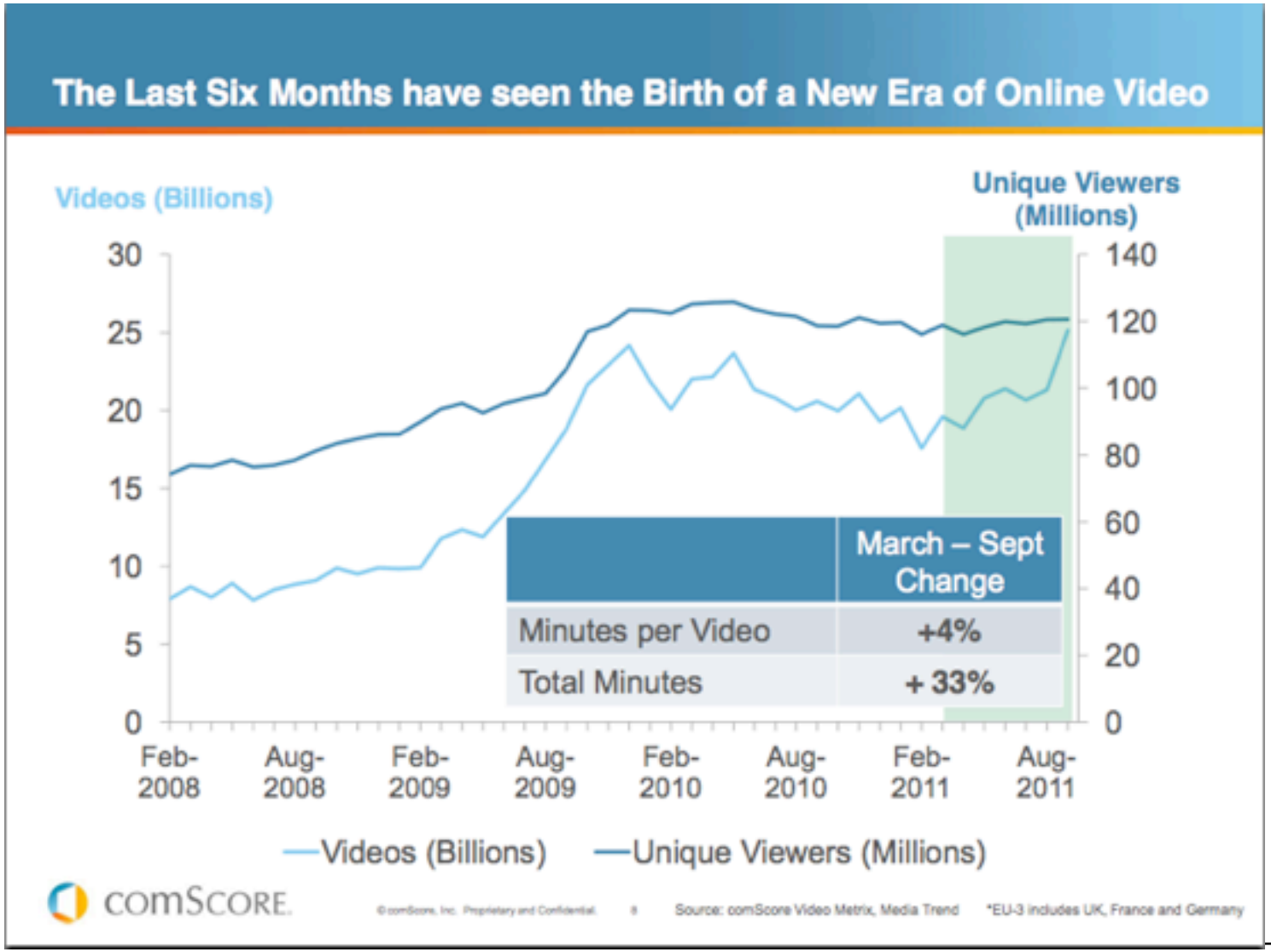


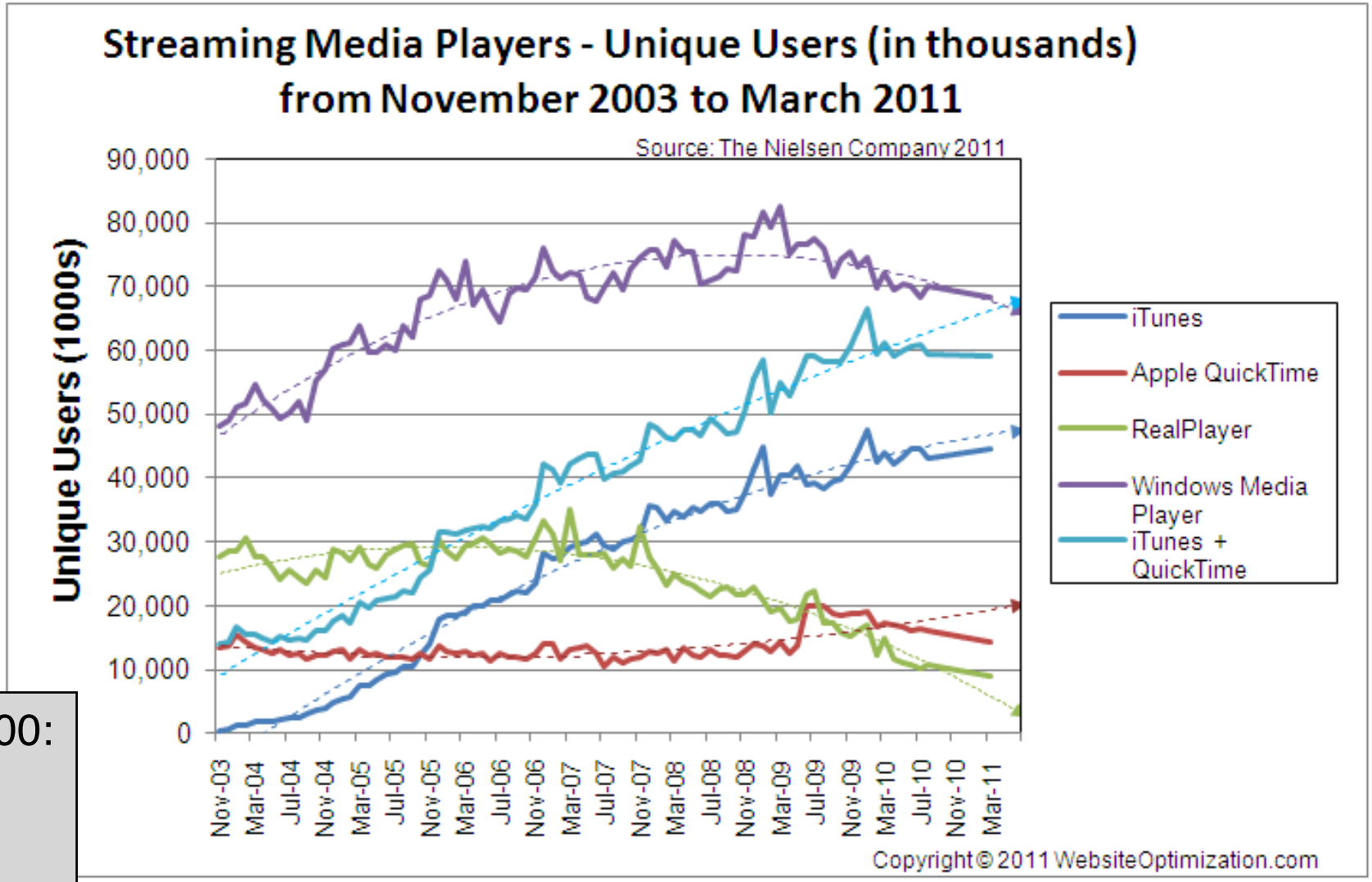
Marc Gosschalk, comscore.com, October 2011

Usage of Video Streaming (4)



Marc Gosschalk, comscore.com, October 2011

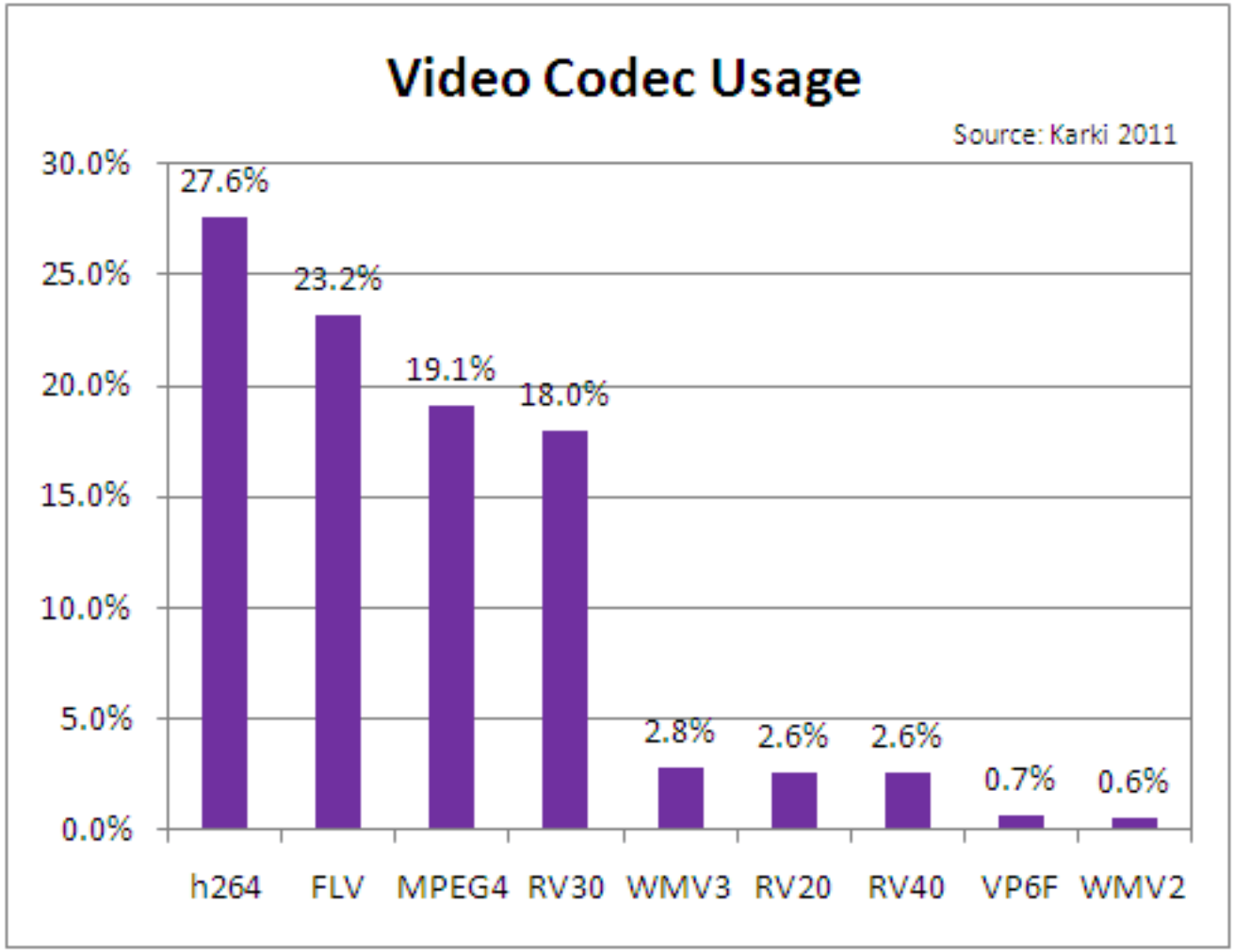
Market Shares of Streaming Players



Back in 2000:
Real 28%
Win 22%
QT 4%
(streaming media.com)

Prediction: iTunes+QT will pass Windows Media by 2012

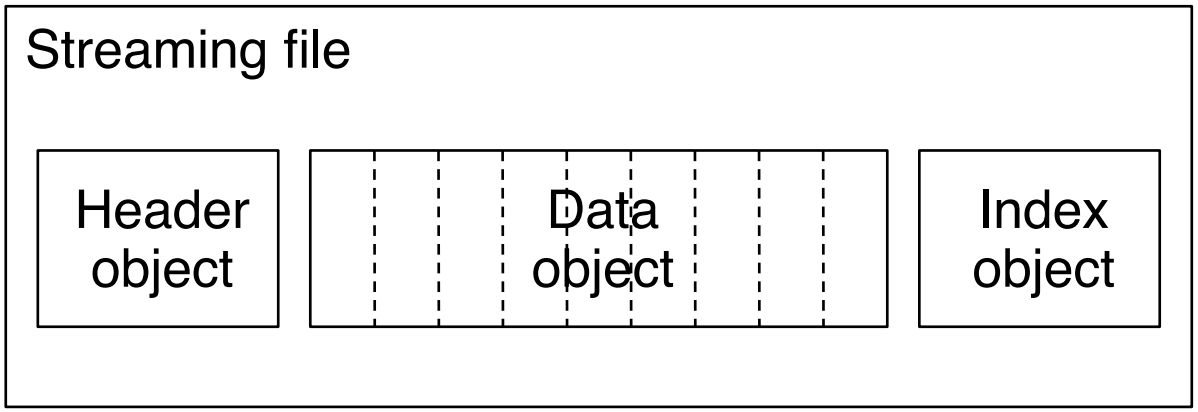
Video Codec Usage



from WebSiteOptimization.com

RV = Real Video
WMV = Windows Media Video

Streaming File Formats



(from ASF)

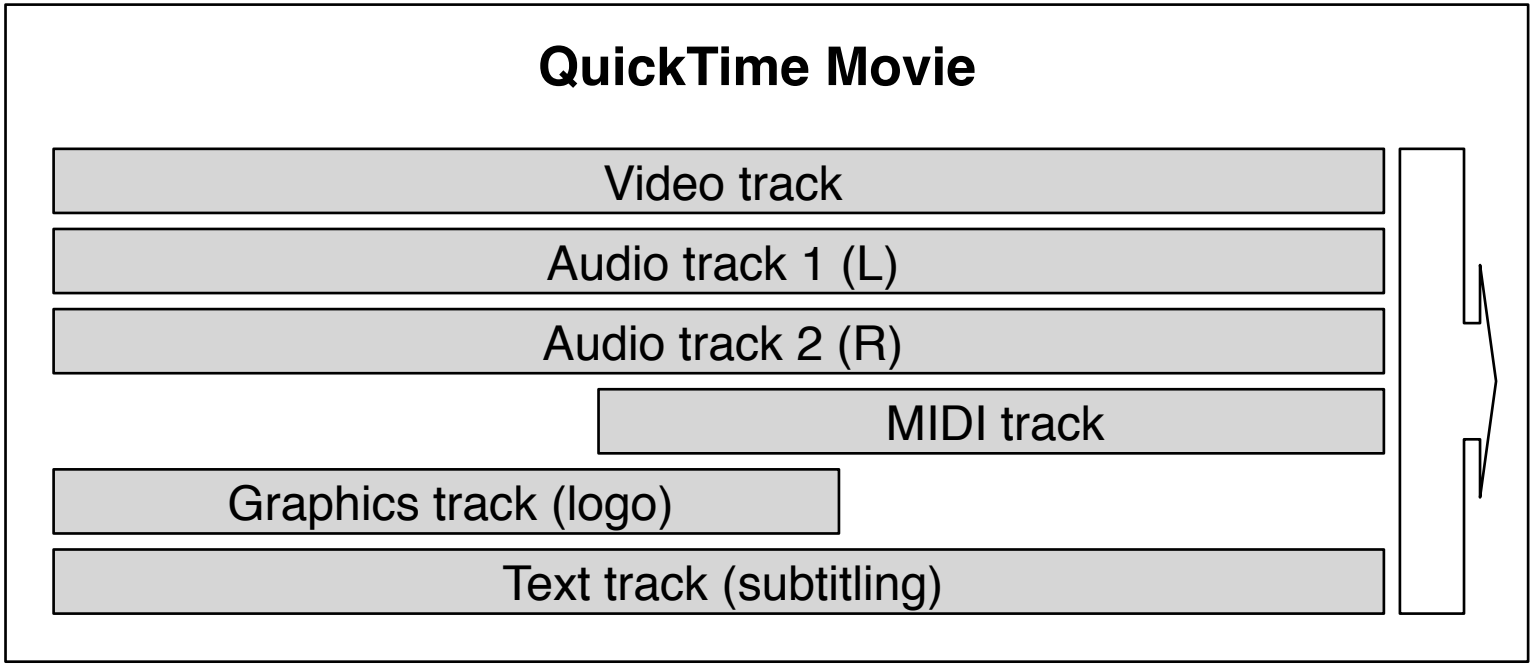
- Header, Data: As in other audio/video formats
  - Additional timing control information used to manage flow rate
- Index Object: Aid for client navigation
- Main streaming file formats:
  - Microsoft: Advanced Streaming Format (ASF), Windows Media Video (WMV), Windows Media Audio (WMA)
  - RealNetworks: RealMedia (RM), RealAudio (RA)
  - Apple: QuickTime Hinted Movie (MOV)

Apple QuickTime

- Platform-independent multimedia architecture
  - Supports MacOS and Windows
- History:
  - First version (1.0) for MacOS 6 in 1991
  - QuickTime 2.0 also for Windows 1994
  - Streaming support since version 4.0 (1999)
  - Current version 10.1 for MacOS 10.7 (and 7.7.1 for Windows)
- QuickTime consists of:
  - Framework, API, file format
- Applications using QuickTime:
  - QuickTime Player, iTunes, Logic, Final Cut, Premiere, Avid, ...
- Codec support:
  - Open plugin architecture (*QuickTime components*)
  - Huge selection of pre-installed components
- Digital Rights Management is intrinsic part of QuickTime

QuickTime Movie Files

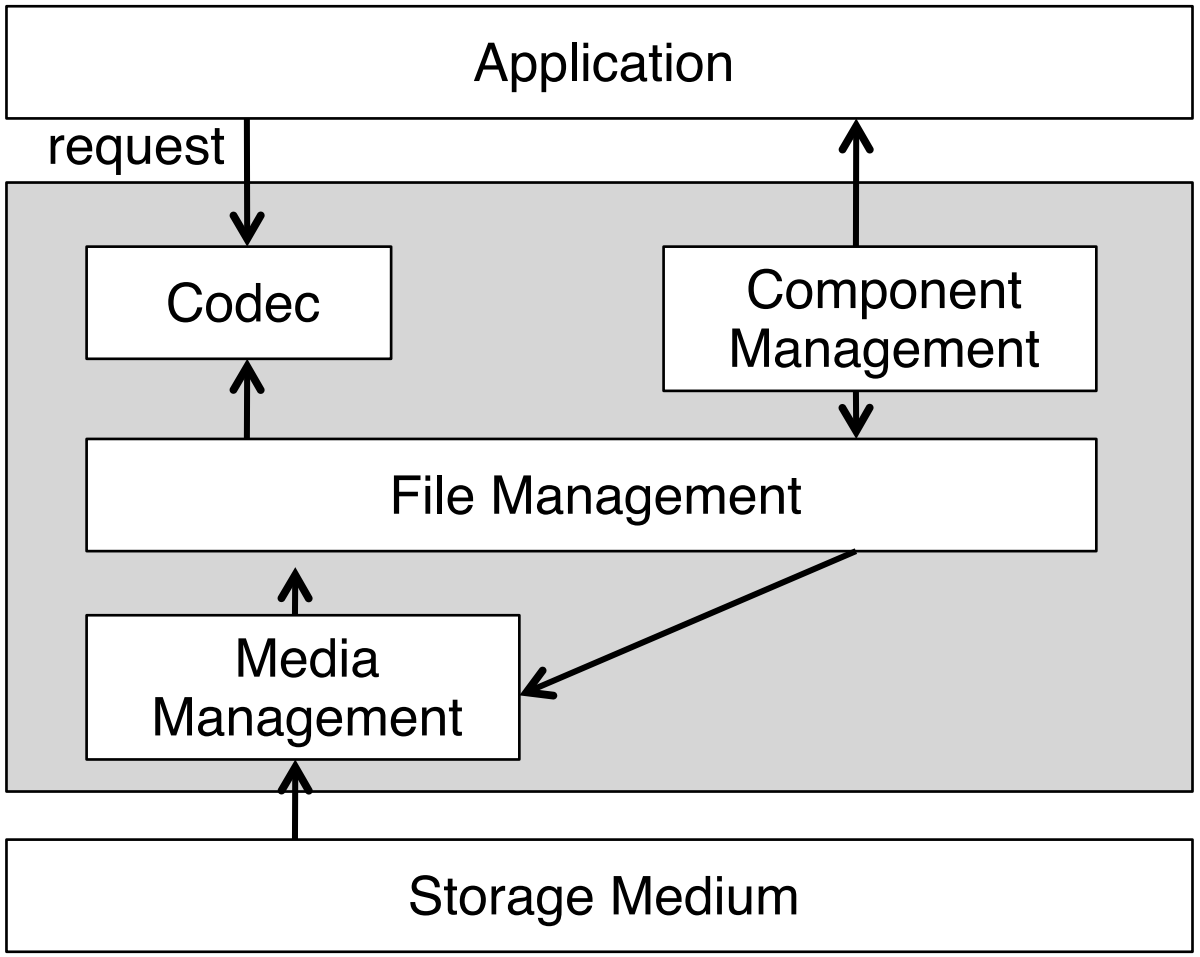
- Modular and flexible architecture
  - Multimedia files organized in tracks
  - Example:



Types of QuickTime Tracks

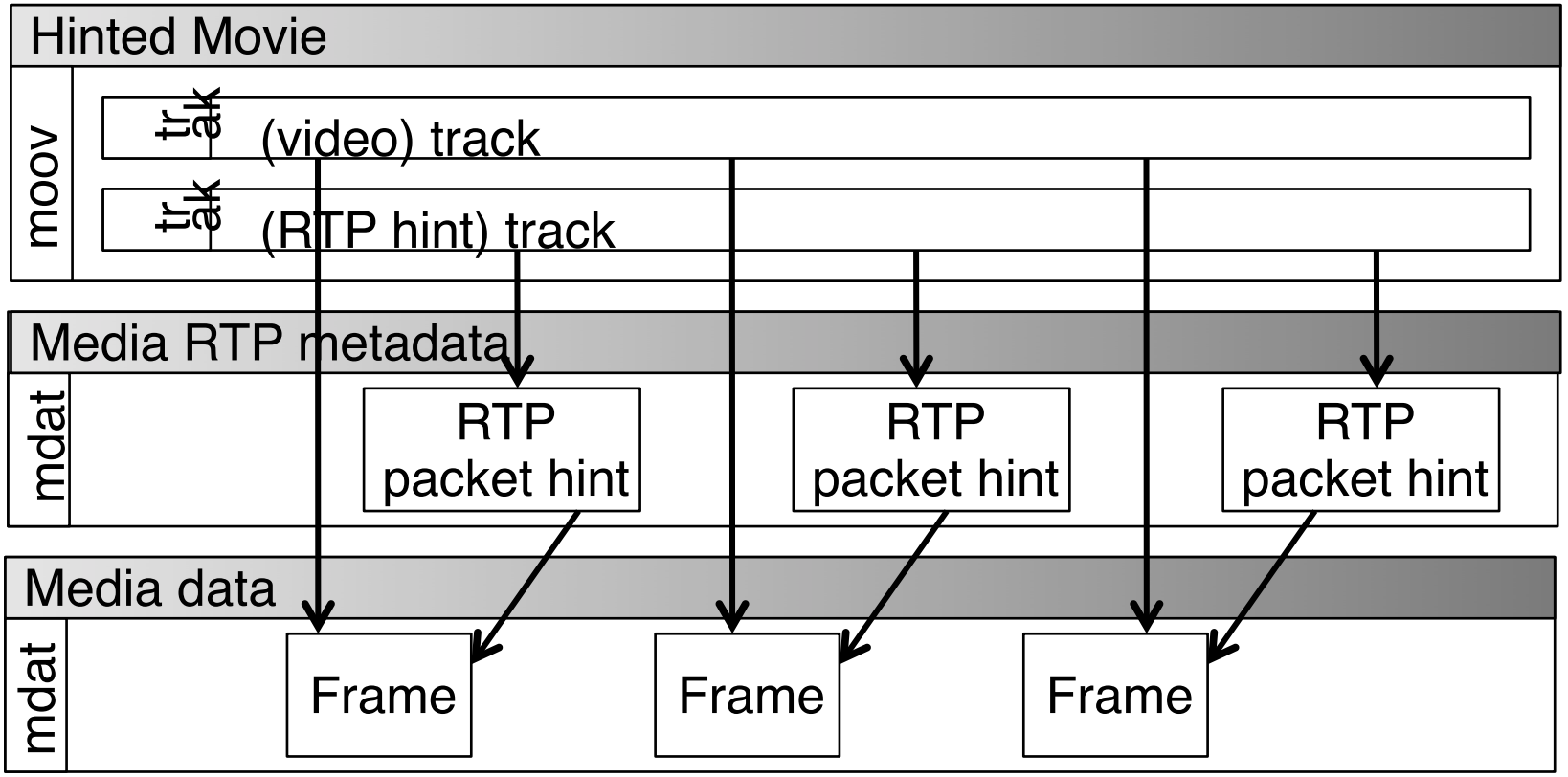
- Movie track: Copyright info, annotations, ...
- Audio track(s)
- Text track: Titles, subtitles, credits, notes, ...
- Sprite track: Images with animatable, programmable behaviours
- Flash track: SWF animation
- QuickTime VR track: VR objects, panorama movies
- Video track: Digital video, 3D animation, ...
- Music track: MIDI
- Chapter track: Inserts addressable entry points
- 3D track: Contains QuickDraw 3D metafile objects
- Streaming track: References to streams from a server source
- Hint track: Additional information for streaming (see below)

QuickTime Media Abstraction Layer



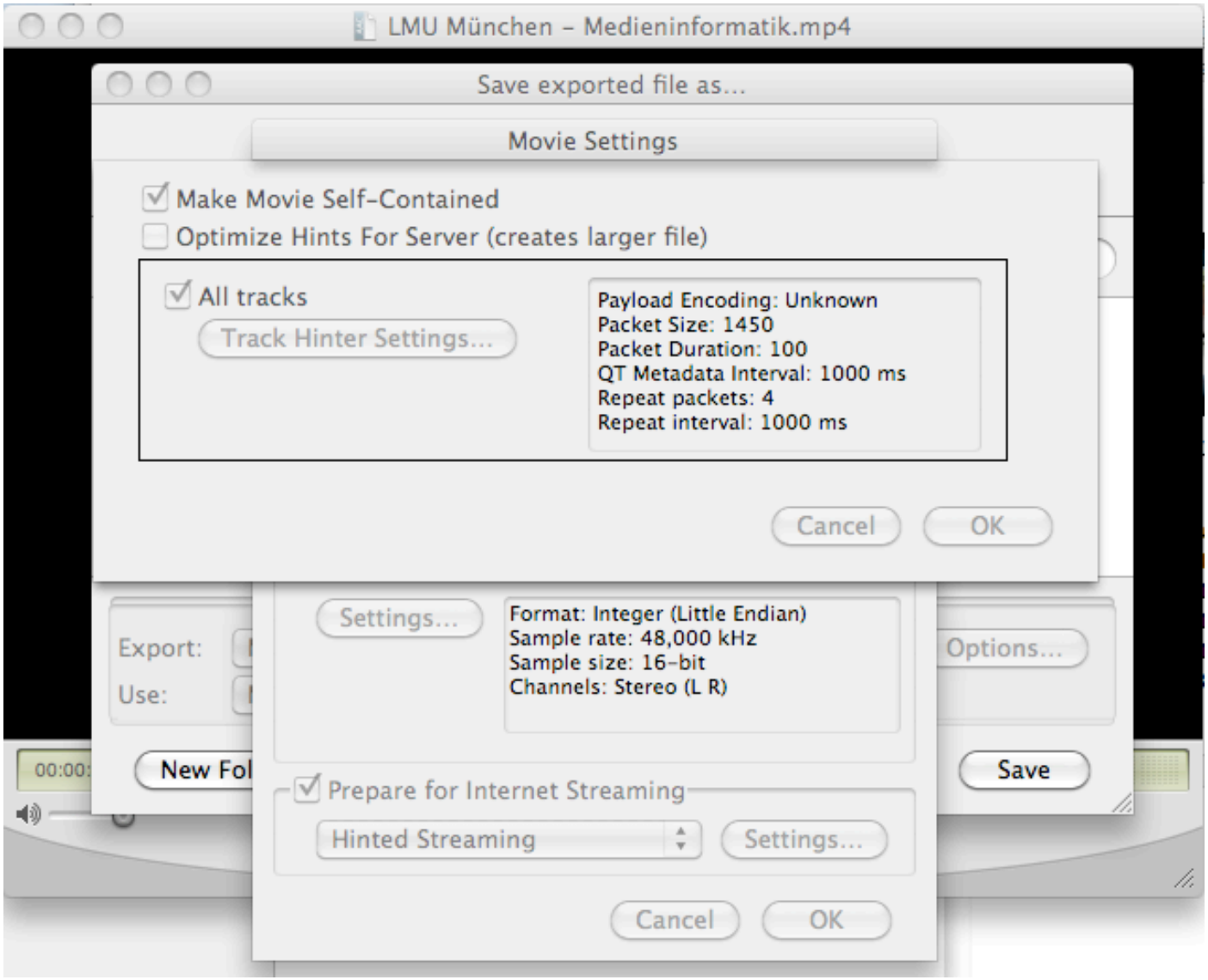
QuickTime Toolbox (Media Abstraction Layer)

Hint Tracks in QuickTime and MPEG-4

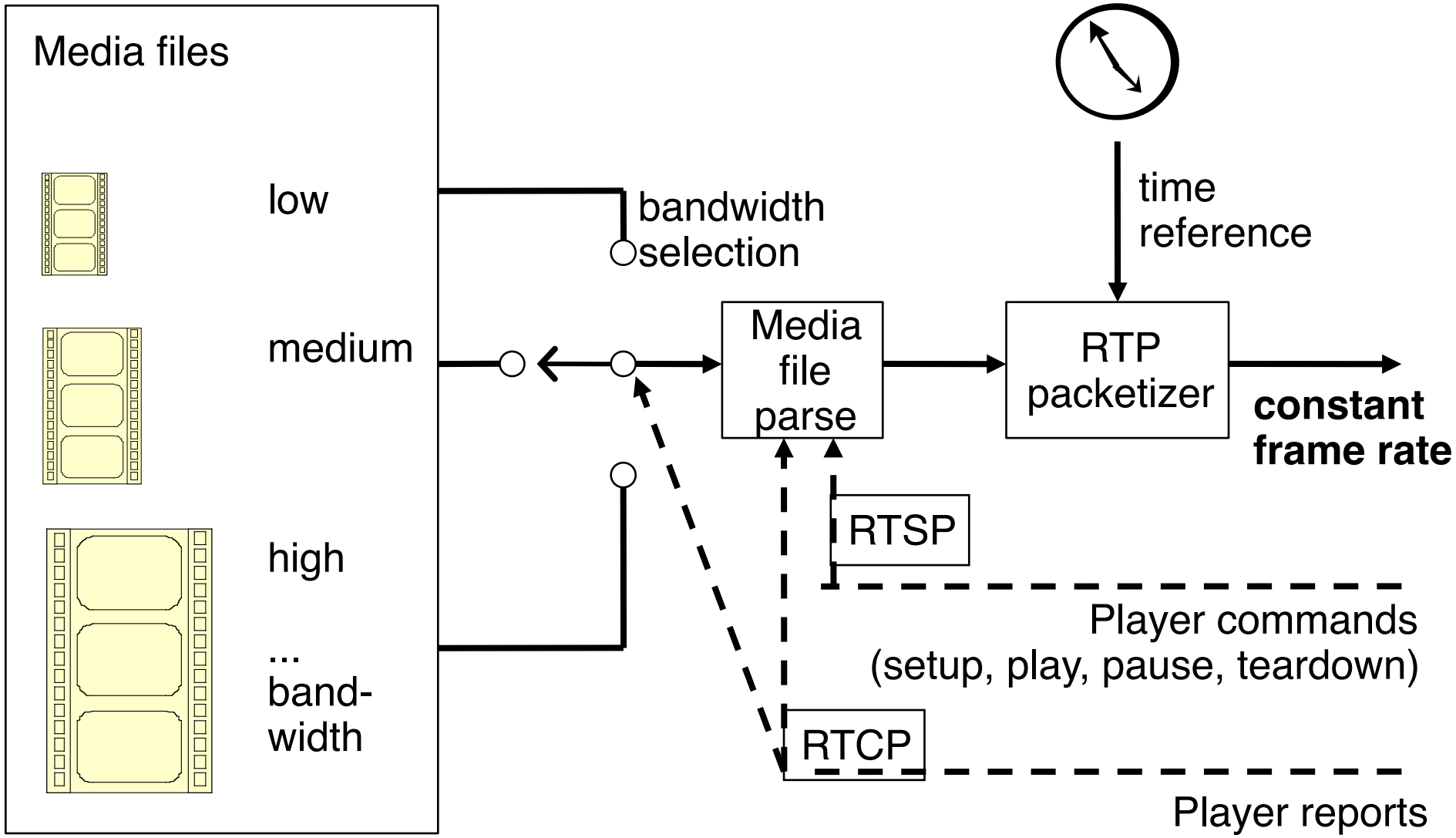


- Hint track gives server software pointers to the RTP information to serve the relevant media chunks
- Concept from QuickTime, integrated in MPEG-4 (streaming)

Adding a Hint Track *(Steuerspur)*



Adapting to Network Congestion



Realisations for Rate Adaptation

- Multiple bit rate files
  - RealNetworks “SureStream”, Windows Media “multiple bit rate”
  - Several bit rates in one file
  - Compatible only with streaming servers, not with Web servers
  - Adaptation by change of picture size not supported
- Alternate movies (QuickTime)
  - Player receives pointers to assemble the actual program
  - Usable for adapting bit rate and other parameters
  - Usable also for different language versions and other applications
- MPEG-4 Scalable Streams
  - Similar to “progressive” technique in picture compression
  - Basic low-resolution stream transmitted
  - Additional “helper” streams can add more detail and improve quality

Trend: Adaptive HTTP Streaming

- Conventional streaming technologies (push-model, based on RTP, RTSP etc.):
  - Apple QuickTime Streaming
  - Microsoft Windows Media
  - Adobe Flash
- Emerging adaptive streaming technologies (pull-model, based on HTTP):
  - Apple HTTP Live Streaming
  - Microsoft Smooth Streaming
  - Adobe HTTP Dynamic Streaming
- Basic idea: Small video file fragments
  - Fragments exist at different bit rates
  - Index file for all available fragments
  - Client requests appropriate next fragment by GET request

3GP-DASH: Open Standard for HTTP Streaming

- Dynamic Adaptive Streaming over HTTP

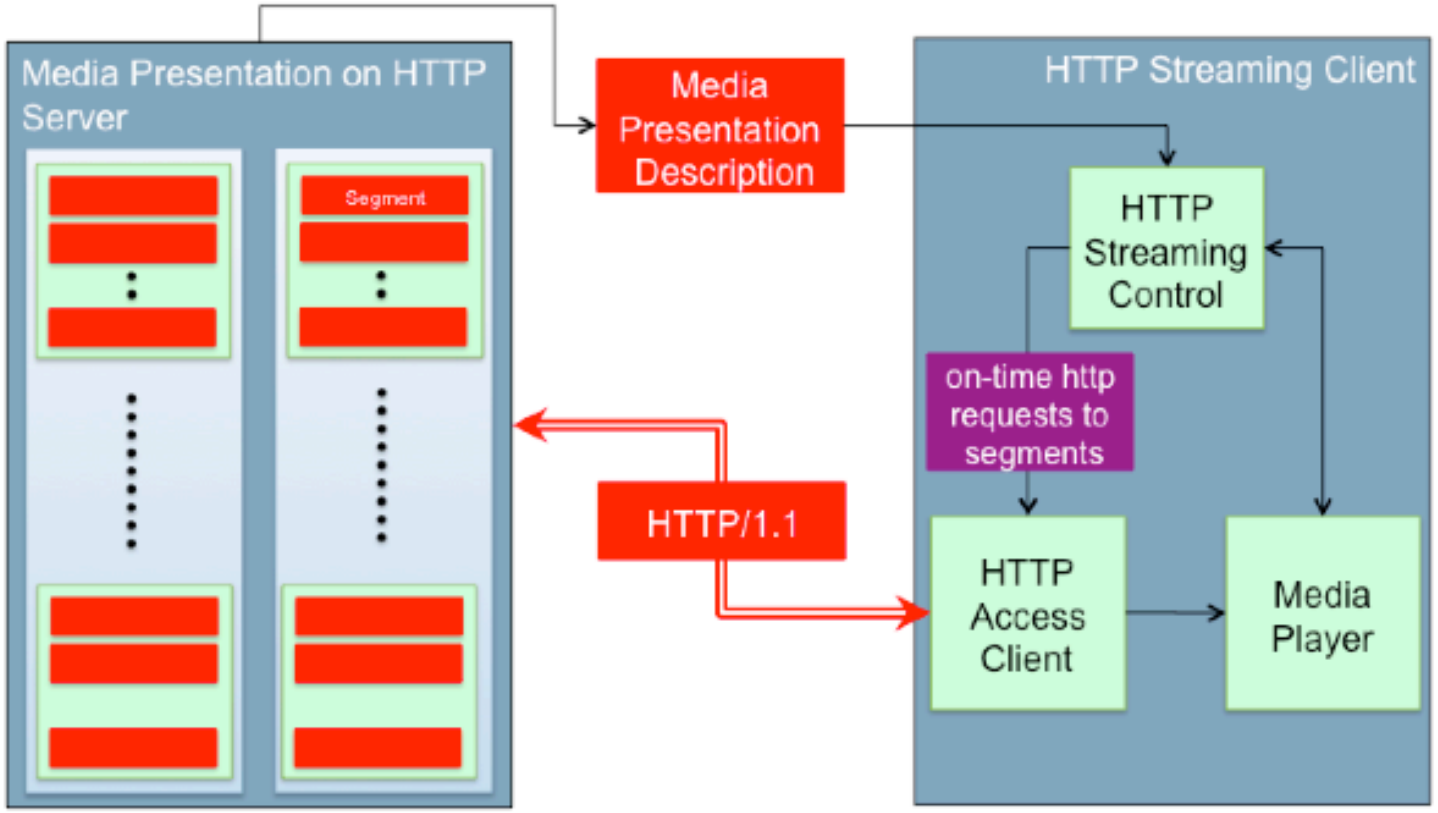


Figure 3 Solution overview – 3GP-DASH

3rd Generation Partnership Project (3GPP)

Source: Thomas Stockhammer, Qualcomm