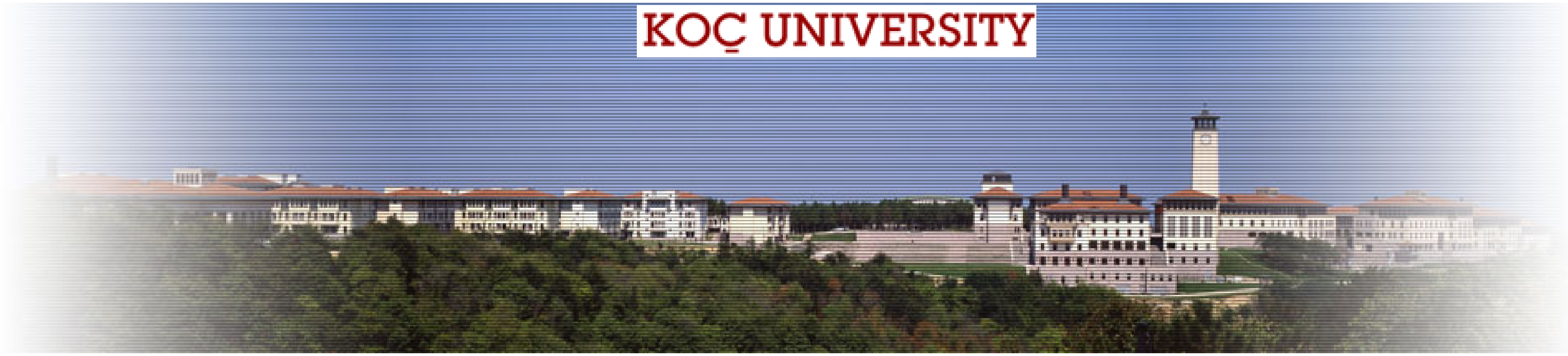

Peer-to-Peer Video

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PetaMedia Summer School
The Marmara Antalya
1 July 2011

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Overview

- **Introduction to Video Streaming**
  - Server-client vs. Peer-to-peer; Push vs. Pull
  - Application/Transport protocols
  - Video coding: MPEG-4 AVC/H264, MPEG HEVC, MPEG SVC
- **Server-Client Video Streaming**
  - Congestion Control and Packet loss remedies
  - Adaptive video streaming, DCCP, SVC
- **P2P Video Streaming**
  - Tree-based, Mesh-based
- **Quality of Experience**
  - Quality of service: Packet loss, delay
  - PSNR, freeze duration, start-up/channel switch delay
- **3D Video Coding and Streaming**

INTRODUCTION TO VIDEO STREAMING

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A Brief History of the Internet

- **Remote access to computing resources (1970's)**
  - *telnet*
- **Shared access to data and text communication (1980's)**
  - *FTP, NFS*
  - *USENET, e-mail*
- **A medium for information dissemination and interactive text and voice communication (1990's)**
  - WWW
  - *IM, VoIP*
- **A medium for streaming media and interactive video communication (2000's)**
  - *IP Radio*
  - *Video over IP, Youtube, IPTV*
  - *Skype, gmail-video, Telepresence*
  - *On-line video games*

Streaming Applications

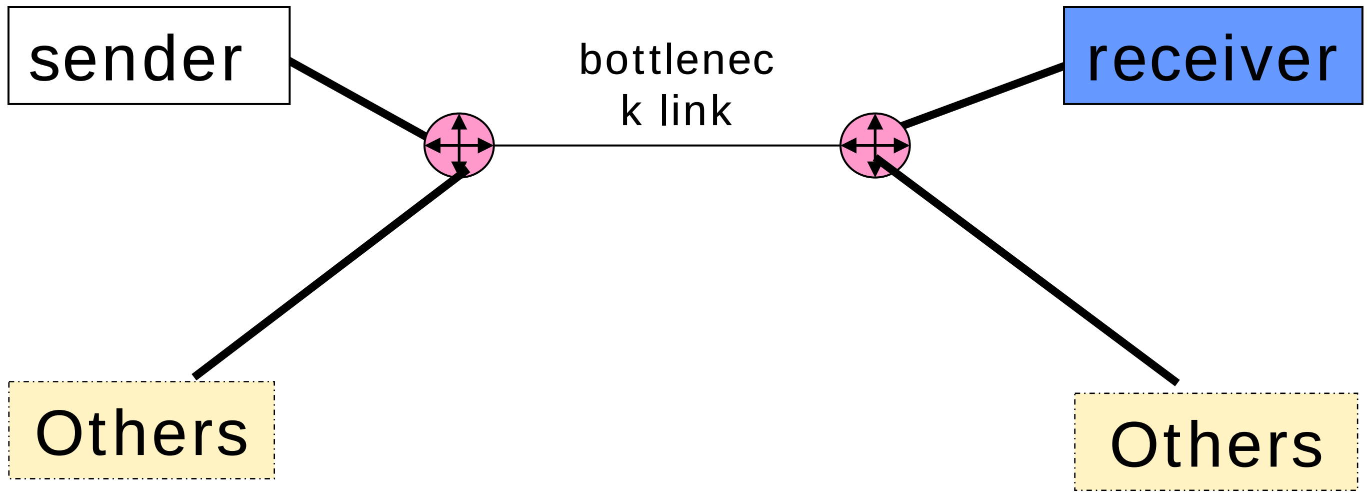
Streaming: Being able to start playback before downloading the whole media file

- **Video-on-Demand Streaming**
  - Commercial or user-generated stored video
- **Live/Real-time Program Streaming**
  - IPTV (live encoding or pre-encoded content)
- **Interactive Communications**
  - Telepresence
  - On-line gaming

Streaming Architectures

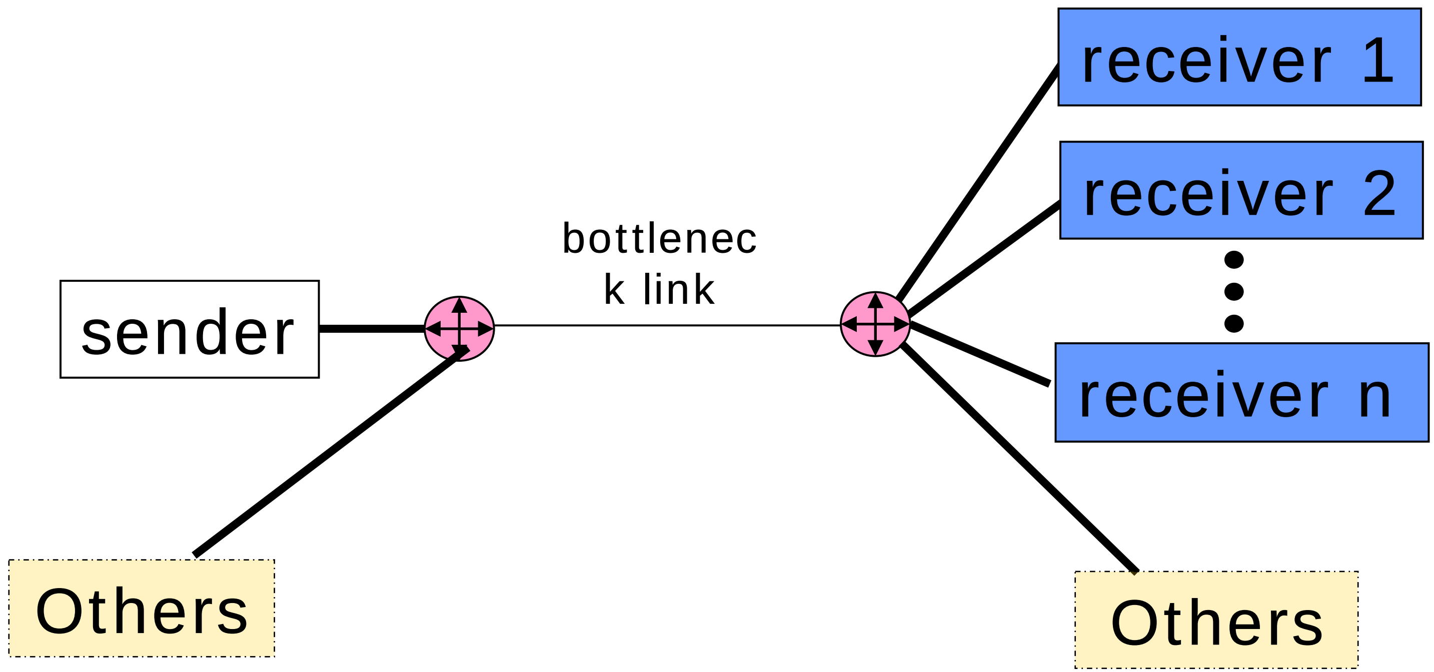
- **Server-Client**
  - Point-to-point unicast
  - Point-to-multipoint multicast
  - Content Distribution Network (CDN) node:
    A CDN node refers to a network entity that usually is deployed at the network edge to store content provided by the original servers, and serves content to the clients located nearby topologically.
- **Peer-to-peer**
  - Tree-based
  - Mesh-based

Unicast Streaming



point-to-point unicast streaming

Multicast Streaming

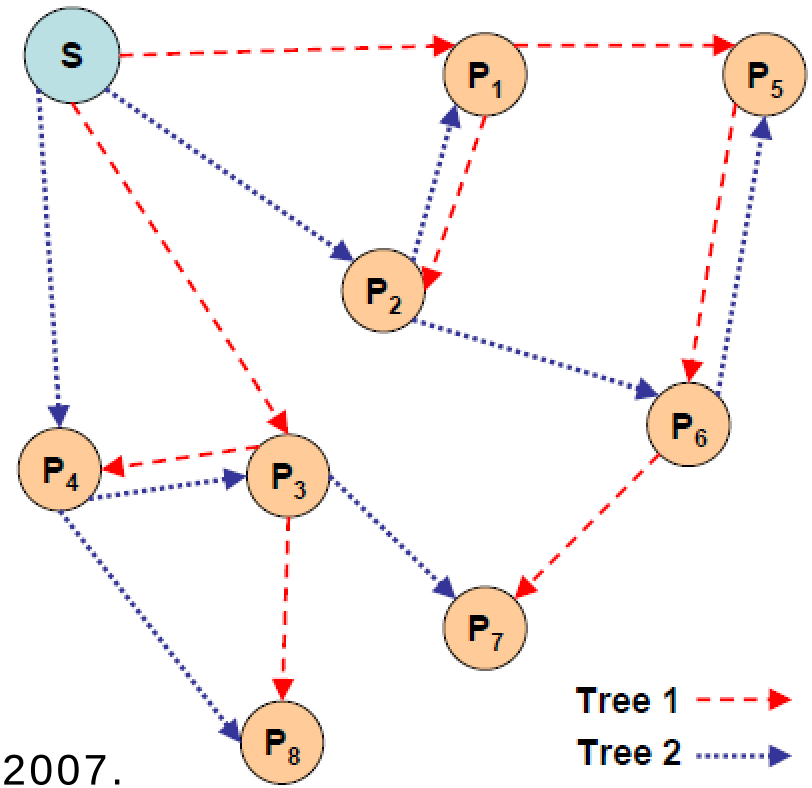


point-to-multipoint unicast/multicast streaming

Tree-Based P2P Streaming

- Broadcast/Multicast
  - Multiple multi-cast trees

Group members self-organize into a tree structure, based on which group management and data delivery is performed. Such structure and push-based content delivery have small maintenance cost and good scalability and low startup delay) and can be easily implemented. However, it may result in low bandwidth usage and less reliability.



Bacchichet, Schierl, Wiegand, Girod, “Low-delay P2P sreaming using scalable video coding,” Packet Video 2007.

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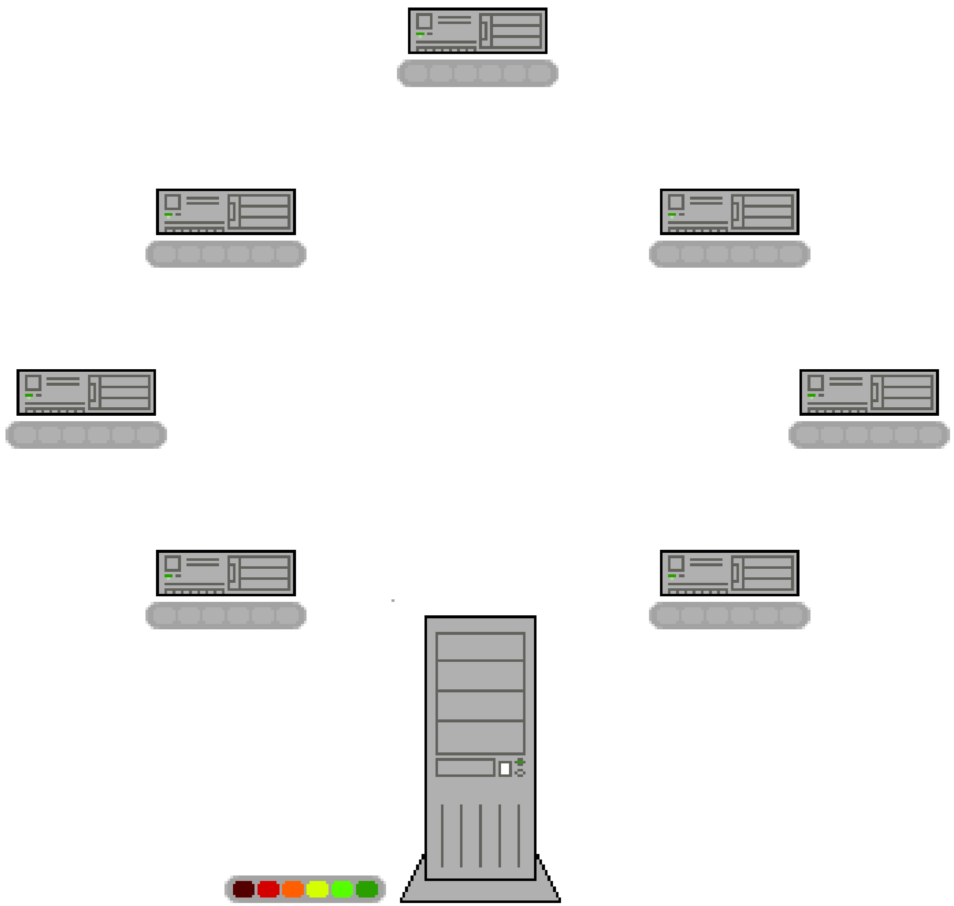
Mesh-Based P2P Streaming

- VoD
  - e.g., BitTorrent

A mesh uses multiple links between any two nodes. Thus, the reliability of data transmission and bandwidth usage are higher. However, the cost of maintaining mesh topology is much larger than that of a tree, and pull based content delivery results in high overhead, in particular in the start up delay.



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Push vs. Pull Models

- **Push Model Streaming Server**
  - Intelligence at the server/sender
  - Example: Apple Darwin Streaming Server

- **Pull Model Streaming Server**
  - Intelligence at the client/receiver
  - Example: HTTP Streaming

Media over IP

Application – IPTV, Telepresence, On-line games	
Presentation – Audio and Video Codecs	(media stream packets)
Session – SDP and RTSP /SIP and RTP	(RTP packets)
Transport – TCP/UDP/DCCP	(transport protocol packets)
Network – IP	(IP datagrams/packets)
Datalink – ADSL/DOCSIS/Ethernet/VLAN/Wifi/IEEE802.11/WiMax	
Physical – copper/coax/fiber/wireless	

e.g., RTP/UDP/IP or RTP/TCP/IP

A Protocol Suite

Application (RTP or HTTP)	
Reliable Stream (TCP)	User Datagram (UDP/DCCP)
Internet (IP)	
Network Interface	

- A *protocol port* is an abstract destination point
  - identified by a positive integer
  - OS provides the means to define or access protocol ports
  - protocol ports are buffered and access is usually *synchronous*
- A *transport address* is the pair:

  <IP Address; Port Number>

Transport Protocols

- *UDP is for unreliable, connectionless delivery adding “destination process selection” functionality to IP*
  - *no acknowledgement (ACK) mechanism*
  - *no message ordering or numbering*
  - *no flow control*
- TCP provides the following services to applications:
  - *procedures that make sure that data exchange is reliable*
  - *procedures for flow control*
  - *mechanisms to direct the traffic to specified processes on a machine*
  - *procedures to recover from transmission errors*
  - *procedures to initiate and end a connection*
  - *but, no guarantees on timely delivery*
- *DCCP*
  - *UDP + rate control*

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DCCP

- **Datagram Congestion Control Protocol**
- **Designed to replace UDP for media delivery.**
- **Provides congestion control without reliability.**
- **Selectable congestion control schemes:**
  - TCP-like, TCP-friendly, ...
- **NAT and firewall support: TCP-style explicit connection setup and teardown**
- **RFC 4340, 4341, 4342**
- **Comes with the recent Linux kernels.**

Video Coding

- **Single layer monocular video**
  - MPEG-4 AVC/H.264
  - MPEG HEVC
- **Scalable (Layered) Video**
  - MPEG SVC
- **3D Video**
  - Frame compatible coding
  - MPEG MVC/3DV

MPEG-4 AVC/H.264

- **Twice more efficient than MPEG-2**
  - Hierarchical B prediction
  - Intra prediction, better motion compensation
- **Rate-distortion performance**
  - HD video 5-10 Mbps
  - SD video 2-4 Mbps
- **Open source software**
  - Real-time encoder x264
  - Real-time decoder ffmpeg

MPEG HEVC

- **High Efficiency Video Coding - Current joint MPEG / VCEG project**
- **Aimed to be twice more efficient than MPEG-4 AVC/H.264**
  - Still block DCT based
  - Adaptive interpolation filtering
  - Variable block-size motion compensation
- **Reference software under development**

Scalable Video Coding (SVC)

- **Extension to MPEG-4 AVC/H.264**
- **Encode video in a base layer and one or more enhancement layers**
- **Scalability in the following domains:**
  - Spatial
  - Temporal
  - SNR
- **Open source software:**
  - Reference encoder -Joint Scalable Video Model (JSVM) C++, compiles on Windows and Linux
  - Real-time decoder - OpenSVC

SVC Example

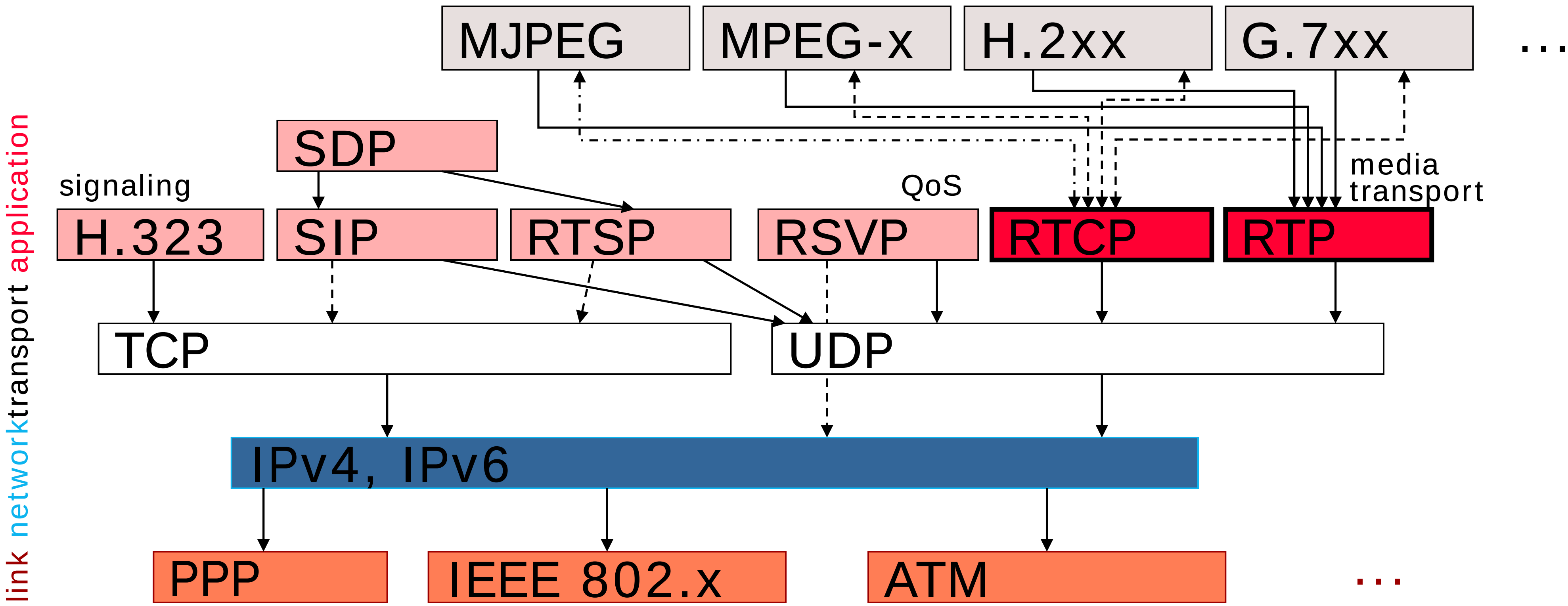
Type	Layer No	Bit Rate (Kbps)	PSNR (dB)
Base	0	70	30.61
MGS	1	110	31.43
	2	141	31.99
	3	174	32.86
	4	199	33.45
	5	224	34.06
	6	246	34.63
	7	259	34.78
	8	277	35.16
	9	304	36.00
	10	321	36.37
	11	335	36.61

Best Layer Configuration

- **How many enhancement layers?**
  - Number of possible extraction points – hence adaptability - increases with the number of enhancement layers
  - However, more the enhancement layers, less the Rate-Distortion (RD) performance.
  - Best configuration is found for videos with various spatial resolutions

SERVER-CLIENT VIDEO STREAMING

Push: Streaming Server Session



Real-time Transport Protocol - RTP

Defined by the Audio Video Transport (AVT) Workgroup of the IETF

- RFC 1889, “*RTP: A Transport Protocol for Real-Time Applications*”, January 1996
- obseleted by: RFC 3550, July 2003
- part of H.323 standard of ITU
- adopted by *Third Generation Partnership Program* (3GPP) and *Internet Streaming Media Alliance* (ISMA)

Basic services provided by RTP

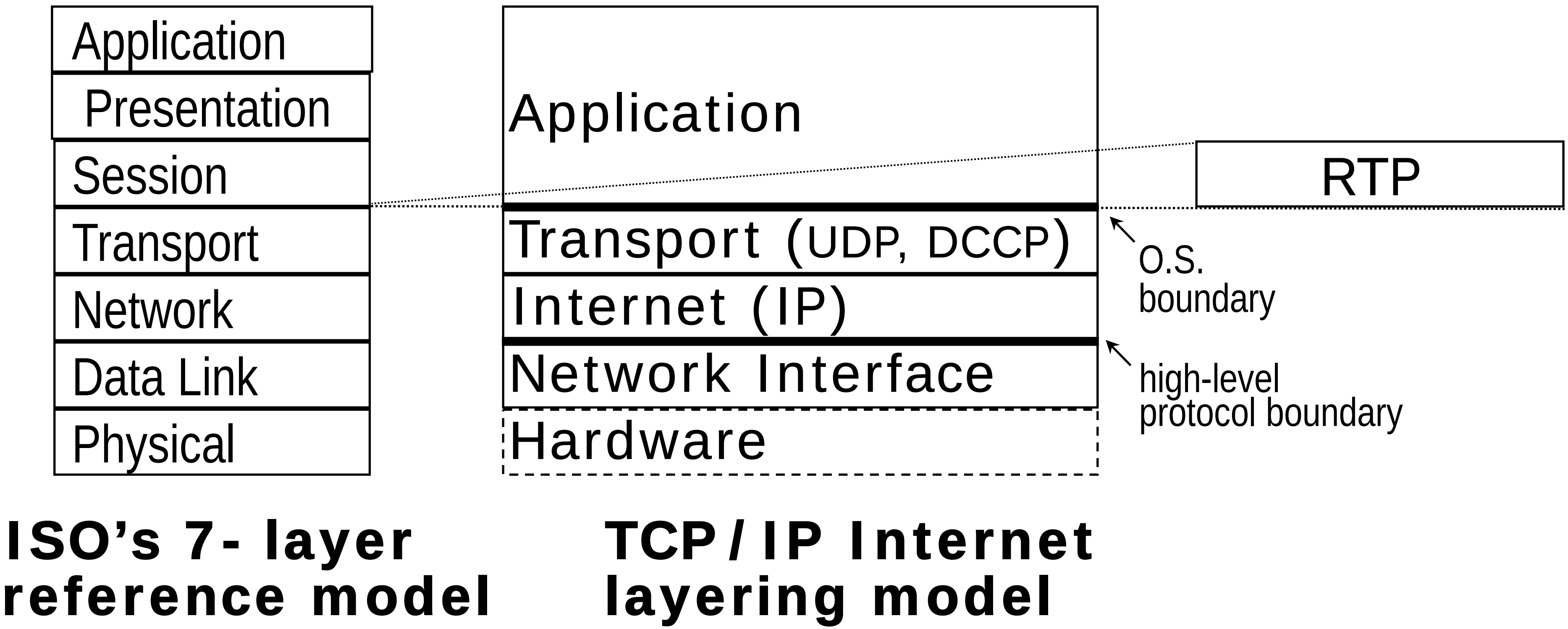
End-to-end delivery services for data with real time characteristics, such as interactive audio and video

1. Payload type identification
2. Sequence numbering
   - re-sequencing (if needed)
   - loss detection
3. Time stamping
   - media presentation timing and jitter removal
   - inter-media synchronization (lip-synch)
4. Delivery monitoring (through associated protocol: Real-time Control Protocol, **RTCP**)
   - quality of service feedback and rate adaptation

Real-time Transport Protocol - RTP

RTP typically runs on top of UDP, using its multiplexing and checksum services

- supports data transfer to multiple destinations using multicast distribution if provided by the underlying network



RTP: Audio and video sessions

- audio and video are transmitted as separate RTP sessions; i.e., separate RTP and RTCP packets are transmitted for each medium using two different UDP port pairs and/or multicast addresses
- each session generates RTP packets containing small chunks of audio/video data preceded by an RTP header
- RTP header specifies what kind of encoding is used which may change during the session, e.g., to accommodate for new bandwidth conditions
- RTP header contains sequence numbers and timing information for each transmitter that can be used for reordering packets and detecting losses
- RTP packets are then put into transport (TCP/UDP/DCCP) packets
- to associate the two sessions, both sessions should use the same *canonical* name in the RTCP packets of both sessions
- synchronized playback of audio and video can be achieved using timing information carried in the RTCP packets for both sessions

RTP Payload Formats

- Audio
  - G.722
  - 2.1
  - MPEG1/2
  - MP3
  - MPEG4
  - DAT, Linear
  - AMR, AMR-WB
  - iLBC
  - EVRC/SMV
  - ...

- Video
  - MPEG1/2
  - MJPEG
  - H.263
  - MPEG4
  - JVT
  - Uncompressed
  - DV
  - SMPTE 292
  - ...

- Other
  - Text
  - Pointers
  - MIDI
  - DTMF Digits, Telephony Tones & Signals
  - Comfort Noise
  - ...

RTP <u>does not</u>

- guarantee delivery
- prevent out-of-order delivery
- ensure timely delivery

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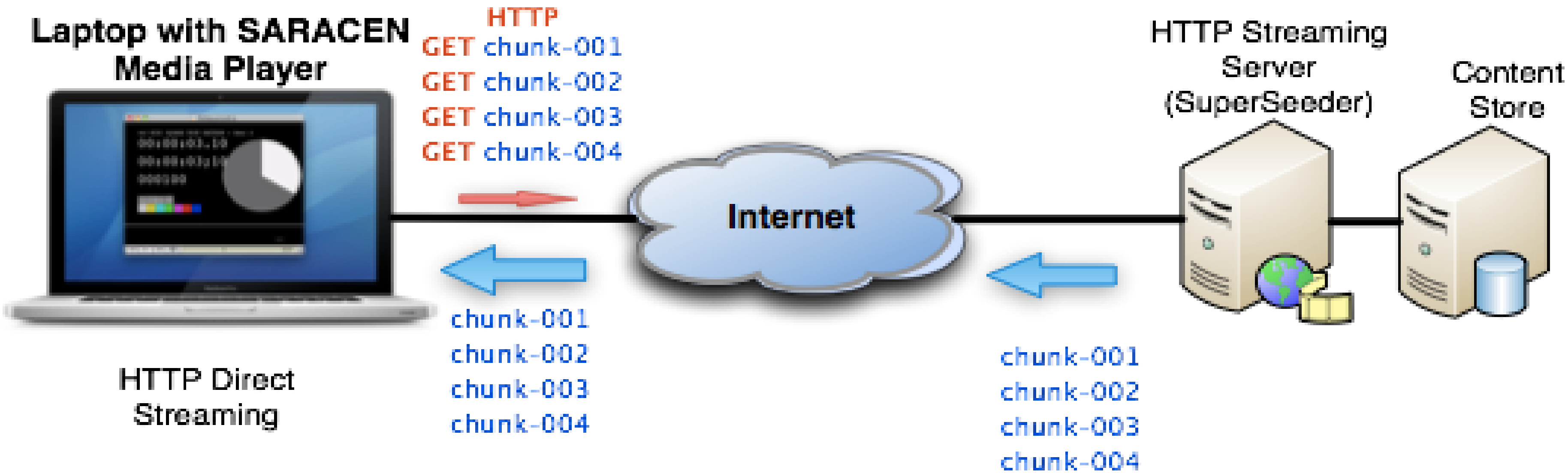
Pull: HTTP Streaming

- HTTP runs over TCP, the client sends “get” commands
- Progressive Download: video file is tagged, similar to download but playback starts before download is complete
- Adaptive Video Streaming: client requests chunks of video

HTTP Media Streaming Architecture



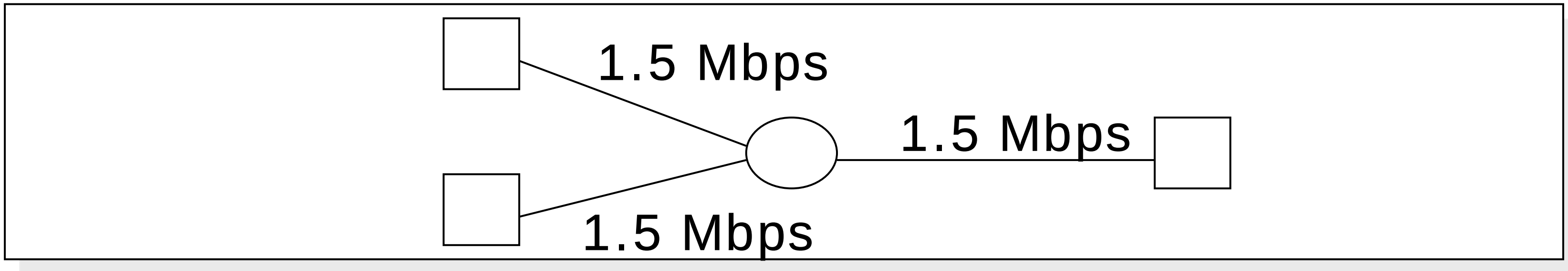
IP Delivery Impairments

What damage may happen to a packet delivered over the Internet?

- **Delayed**
- **Lost**
- Delivered out-of-order
- Replicated
- May have bit errors

Handled by transport protocols

Congestion



- **Different sources compete for resources (channel bandwidth, router queues) inside the network**
- **Why is it a difficult problem?**
  - *Sources are unaware of the current state of the resources*
  - *Sources are unaware of what each other is doing*
- **Congestion manifestations:**
  - Lost packets (buffer overflow at routers)
  - Long delays (queuing in router buffers)
    - ➢ **In many situations this will result in << 1.5 Mbps of throughput for the above topology (*congestion collapse*)**

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TCP Congestion Control

- **TCP interprets <u>packet drops</u> as signs of congestion and slows down**
  - *This is an assumption: packet drops are not a sign of congestion in all networks!*
    - *e.g., wireless networks*
- **TCP periodically probes the network to check whether more bandwidth has become available.**

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TFRC – TCP Friendly Rate Control

- A flow is "*reasonably fair*" if its sending rate is generally *within a factor of two* of the sending rate of a TCP flow under the same conditions
- TFRC has lower variation of throughput over time compared with TCP
  - more suitable for applications such as telephony or streaming media
  - TFRC responds slower than TCP to changes in available bandwidth
- TFRC is designed for applications that use a fixed packet size, and vary their sending rate in packets per second in response to congestion
- The receiver measures the <u>loss event rate</u> and feeds this information back to the sender.
- The sender also uses these feedback messages to measure the <u>round-trip time</u> (RTT).
- The <u>loss event rate</u> and <u>RTT</u> are then fed into TFRC's throughput equation, giving the acceptable transmit rate.
- The sender then adjusts its transmit rate to match the calculated rate.

Packet Loss Remedies (I)

- **Application Layer Framing - Packetization of bitstream consistent with the video structure.**
  - Handle fragmentation intelligently (*at the application layer*) No ***chop & ship*!**
  - Try to contain the effect of a packet loss to the content of the lost packet
  - Make each packet *self decodable*
    - Provide the necessary information for decoding the packet in the packet; headers, parameters, etc.
  - In H264/SVC, there are NAL units. A NAL unit is one or more (integer number of) slices.

Packet Loss Remedies (II)

- Application Layer <u>A</u>utomatic <u>R</u>epeat re<u>Q</u>uest (**ARQ**)
  - ¤ based on acknowledgements (**ACK / NACK**)
  - ¤ hard to use with multicast (*reliable multicast*)
  - ¤ delay problem (roundtrip & loss of ACK/NACK)
  - ¤ selective retransmissions may be more appropriate

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Packet Loss Remedies (III)

- Application Layer Forward Error Correction (**FEC**)
  - ¤ the simplest form is redundant transmission
  - ¤ bandwidth increase problem
  - ¤ delay may increase
  - ¤ FEC applied to one layer of a layered codec = Unequal Loss Protection (**ULP**)

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FEC Implementation in RTP (RFC 2733)

- A payload format to implement generic FEC for media encapsulated in RTP
- Engineered for FEC algorithms based on the exclusive-or (*parity*) operation
- Allows for the recovery of both the pay-load and critical RTP header fields
- FEC information is sent as a separate stream
  - *backwards compatible with non-FEC capable hosts*
  - *receivers that do not wish to implement FEC can just ignore (not receive) the FEC stream*

FEC Implementation in RTP (RFC 2733)

Scheme 1: a b c d ←

f(a,b) f(c,d) ← FEC stream

Scheme 2: a b c d ←

f(a,b) f(b,c) f(c,d) ← FEC stream

increased overhead and delay, recovery of some two consecutive losses

Scheme 3: f(a,b) f(a,c) f(a,b,c) f(c,d) f(c,e) f(c,d,e) ← FEC stream

reduced overhead, recovery of single losses

Scheme 4: a b c d ←

f(a,b,c) f(a,c,d) f(a,b,d) ← FEC stream

recovery of single, double and triple consecutive losses

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Packet Loss Remedies (IV)

- Guaranteed **QoS -** more expensive than *best effort*
  - *Differentiated services (DiffServ)*
  - *Integrated services*
  - *Resource reservation protocol (RSVP)*
  - *MPLS*

- Intranet applications are already available while large scale deployment is yet to be seen

- Backbone seems to be working well for now, but access may pose problems

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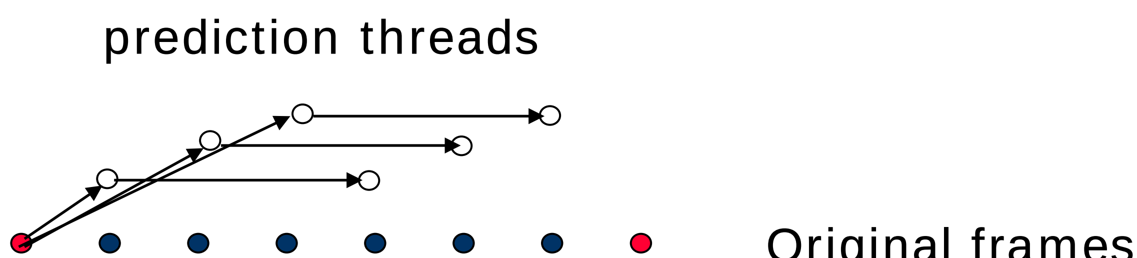
Error-Resilient Video Encoding (I)

- **Proper use of INTRA (*key*) updates**
  - Frequent *INTRA frames* to break long prediction chains
    - *Be careful about the increase in the instantaneous bit rate!*
  - Adaptive Intra Slice Refresh
  - A back channel to ask for *key updates*

Error-Resilient Video Encoding (II)

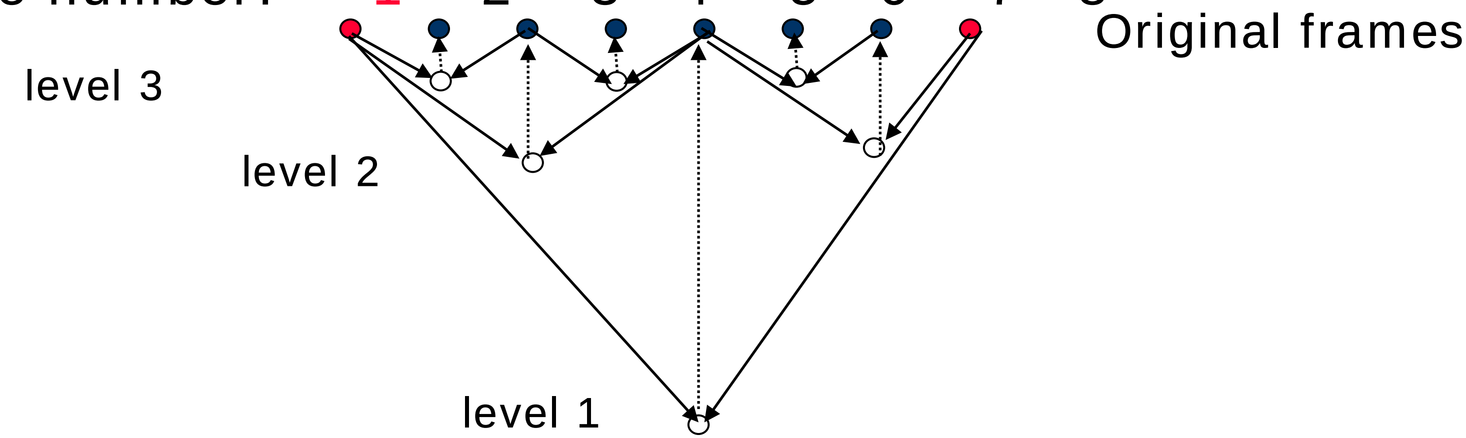
- **Robust Prediction Structures**
  - *Multiple reference frames – JVT*



Frame number: 1 2 3 4 5 6 7 8

  - *Hierarchical B (bidirectional) prediction – JVT*

Frame number: 1 2 3 4 5 6 7 8



Error-Resilient Video Encoding (III)

- **Scalable Video Coding**
- **Split the encoded bitstream into two or more layers based on:**
  - Frame rate — Temporal
  - Quality — SNR (signal to noise ratio)
  - Resolution — Spatial
  - Combinations of the above
- **Layered coding can be used to identify *high priority (HP)* parts (the *base layer)* of an encoded bitstream.**

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Error Concealment @ Decoder

- Frame/slice replication
- Spatial interpolation
- Motion-compensated interpolation

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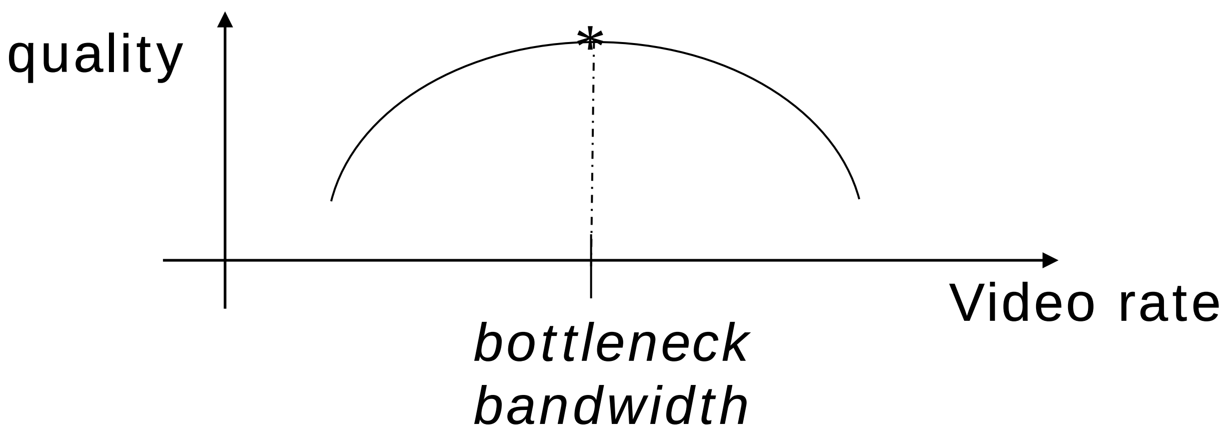
Avoiding Congestion

- **Getting on the backbone is <u>the</u> problem!**
- **Two protectors of the current Internet**
  - *Low-bandwidth access*
  - *Built-in TCP*
- **Future: Network-Aware Applications**
  - *Adaptive streaming → a wise choice!*

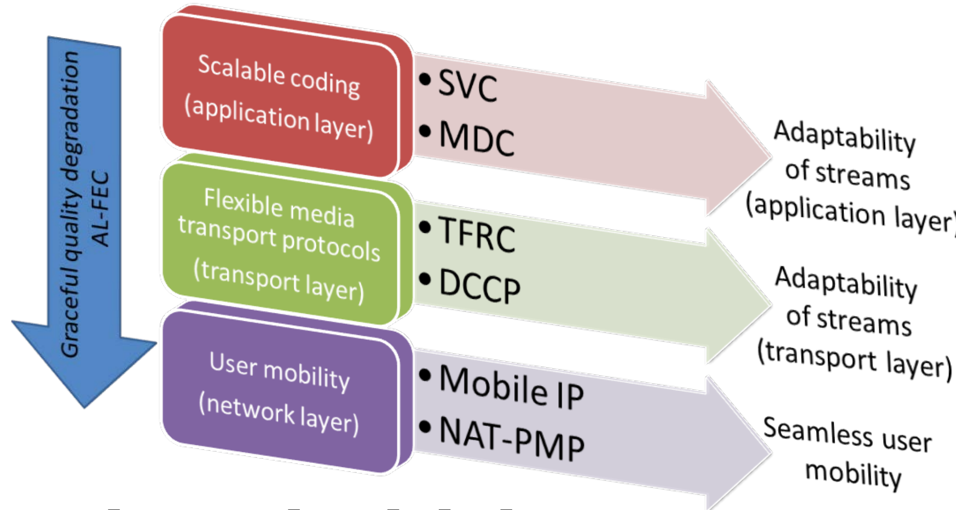


Network-Aware Applications

- **Single stream:**
  - No bandwidth adaptation:
    - Low quality
    - Long pre-buffering
    - Re-buffering
- **Multiple streams:**
  - Bandwidth adaptation thru stream-switching
    - Windows: Intelligent streaming
    - RealNetworks: SureStream
  - Adaptive streaming using SVC

Adaptive Streaming Platform

- **Congestion control at the transport layer**
  - TCP
  - UDP+TFRC
  - DCCP



- **Estimate instantaneous bandwidth**
  - Application layer: use signaling packets and formula
  - Protocol level: DCCP
- **Adaptation of the source rate**
- **Protection of the base layer (with SVC)**

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Video Rate Adaptation

- **Real-time encoder – rate control by QP selection**
- **Nonreference B-Frame skipping**
  - Jerkiness
- **Stream switching**
  - Coding inefficiency + delay
- **Enhancement Layer skipping (scalable coding)**
  - SNR (QP modification)
  - Spatial
  - Temporal (frame skipping)

Extracting the Layered Video

- **Given a target rate, how to extract a Group of Pictures (GoP)?**
  - **Flat extraction:** extract all frames in GoP at the same layer
  - **Hierarchic extraction:** give more bandwidth to the reference frames in hierarchic prediction
  - **Priority-based extraction:** insert priority information after encoding

Rate Adaptation in Multicast

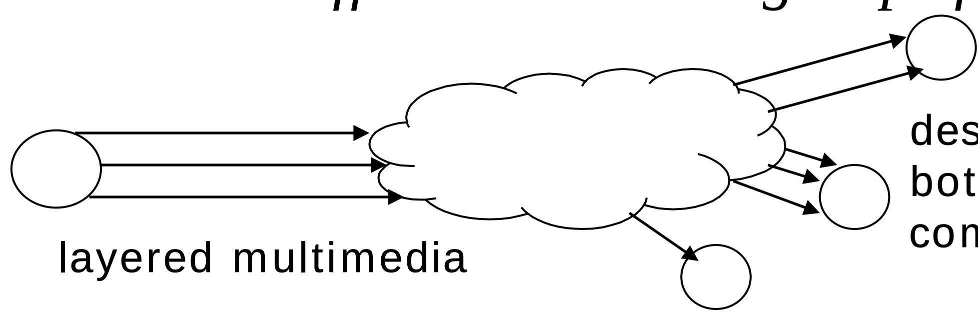
Bottleneck bandwidth:

- not the same for everyone in a multicast

Alternatives:

- use mean opinion
- receiver driven *layered* multicast for
  - *heterogeneous reception bandwidth*
  - *varying computing power*

—*Establish different multicast groups for different video rates*
—*Establish different multicast groups for base and enh streams*



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PEER TO PEER VIDEO STREAMING

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Definitions

- **Seeder:** Original provider of a media file.
- **Peer:** a program that implements a P2P streaming protocol. Peers not only receive streaming content, but also store and upload streaming content to other participants.
- **Swarm:** refers to a group of peers sharing the same content at a given time.
- **Chunk:** a basic unit of partitioned streaming media, which is used by a peer for the purpose of storage, advertisement and exchange among peers
- **Tracker:** maintains information about all clients to assist in efficient data sharing between clients. Specifically, it identifies the network location of each client uploading or downloading chunks associated with a swarm and which fragment(s) of a media file each client possesses.

P2P Streaming Protocols

- **PPSP:** **PPSP protocols refer to the key signaling protocols among various P2P streaming system components, including the tracker and peers**
- **Tracker protocol:** **provides communication between Service Trackers and Peers, by which Peers report streaming status to the Tracker and request candidate lists from the Tracker.**
- **Peer protocol:** **allows for information to be shared directly between Peers like Chunk Maps, Chunks, preferences, capabilities, etc.**

P2P Streaming Model

```
+------------------------------------------------------------+
|    +----------------------------------+                    |
|    |              Tracker             |                    |
|    +----------------------------------+                    |
|         ^  |                        ^                     |
|         |  |                        |                     |
|  query   |  |  peer list/            |streaming Status/    |
|          |  |  Parent nodes          |Content availability/|
|         |  |                        |node capability      |
|         |  |                        |                     |
|         |  V                        |                     |
|    +--------------+          +--------------+              |
|    |    Peer1     |<-------->|   Peer 2     |              |
|    +--------------+ content/ +--------------+              |
|                     join requests                         |
+------------------------------------------------------------+
```

Survey of P2P Streaming Applications
draft-ietf-ppsp-survey-01

Survey of P2P Applications

- **Mesh-based**
  - Joost; Octoshape; PPLive
  - Zattoo; PPStream
  - SopCast; TVants
- **Tree-based**
  - PeerCast
  - Conviva
- **Hybrid**
  - New Coolstreaming (original mesh-based)

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ALTO Services

- The IETF Working Group will design and specify an Application-Layer Traffic Optimization (ALTO) service that will provide applications with information to perform better-than-random initial peer selection.
- ALTO services may take different approaches at balancing factors such as maximum bandwidth, minimum cross-domain traffic, lowest cost to the user.
- The WG will consider the needs of BitTorrent, tracker-less P2P, and other applications, such as content delivery networks (CDN) and mirror selection.

- ALTO Problem Statement http://www.ietf.org/rfc/rfc5693.txt
- Xie, H., Krishnamurthy, A., Silberschatz, A., and R. Yang, "P4P: Explicit Communications for Cooperative Control Between P2P and Network Providers" <http://www.dcia.info/documents/P4P_Overview.pdf>.

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ALTO Protocol

- This WG will focus solely on the communication protocol between applications and ALTO servers. Note that ALTO services may be useful in client-server environments as well, although P2P is the first focus.
- If, in the future, IETF considers changes to other protocols for actually implementing ALTO services (e.g. application-layer protocols for Internet coordinate systems, routing protocol extensions for ISP-based solutions), such work will be done in strict coordination with the appropriate WGs.
- Mar 2011 Submit requirements document to IESG as Informational
- May 2011 WG Last Call of deployment considerations document
- Nov 2011 WG Last Call of discovery mechanism
- Feb 2012 Submit discovery mechanism to IESG as Proposed Standard

P2P-Next Project

- P2P-Next is an FP7 Integrated Project to develop an open source, efficient, trusted, personalized, user-centric and participatory television and media delivery mechanism with social and collaborative connotations using the emerging P2P paradigm, which takes into account the existing EU legal framework.
- New features include an UPnP Suite, Subtitles gossiping and the latest implementation of libswift, a new solution to replace TCP-IP as the transport protocol for content delivery. The open-source NextShare platform has been developed by the EC funded P2P-Next project
- The August 2010 release of the NextShare P2P content distribution platform is available at http://www.p2p-next.org
- The latest features of the NextShare release can be tested globally as a part of the "labs“ project in Wikimedia sites, for details see http://techblog.wikimedia.org/

SARACEN Project

- **FP7 STREP - Socially Aware, collaboRative, scAlable Coding mEdia distributioN**
- **SARACEN works on distribution of multimedia streams through p2p schemes, where the peers participate in an end to end media distribution system making use of social networking related information.**
- **An HTTP based prototype for P2P streaming of SVC video has been developed by INOV**
  - Media transport: video Chunks are requested by HTTP
  - Tracker and Peer Protocols: signaling via HTTP/XML encoded messages with appropriate XML body.

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Tracker Protocol Messages

CONNECT: used when a Peer connects to the system. The Tracker records the PeerID, Connect-time and several Peer properties and status.

DISCONNECT: used when the Peer intends to leave the system and no longer participate in any swarm. The Tracker deletes the corresponding activity records related to the Peer (including its status and all content status for all swarms).

JOIN: used for Peers to notify the Tracker that they wish to participate in a swarm.

LEAVE: used when Peers want to indicate they wish to leave a particular swarm.

FIND: allows Peers to request to the Tracker the Peer list for a swarm.

STAT_REPORT: allows the exchange of statistic and status data between Peers and Tracker to improve system performance.

Peer Protocol Messages

GET_PEERLIST: allows Peers to request the Peer List for a specific content or a particular swarm to other Peers.

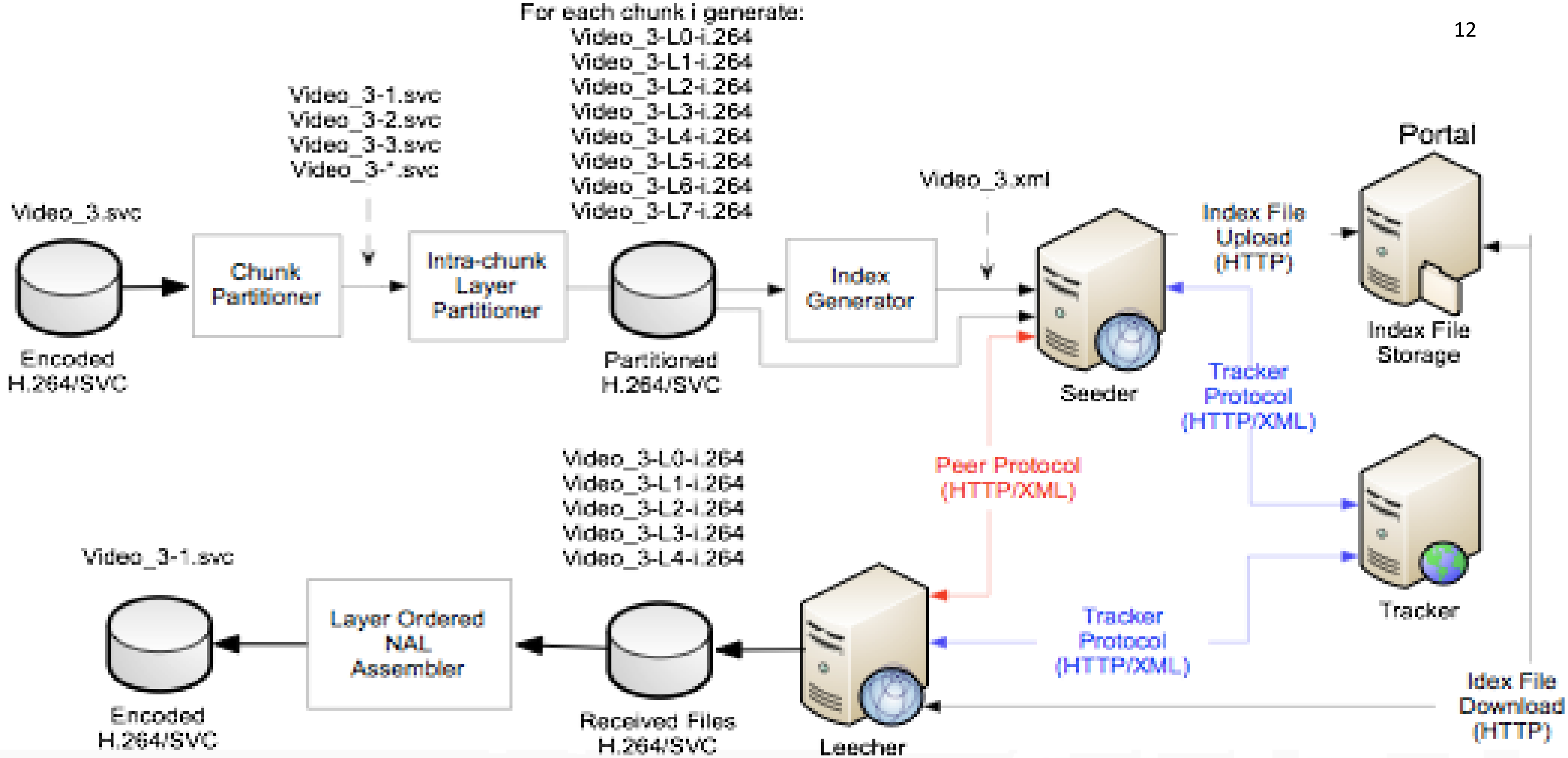
GET_CHUNKMAP: allows Peers to request chunks of a swarm the other Peer presently stores. The Chunk map returned by the other Peers lists ranges of Chunks (and Layers).

GET_CHUNKS: allows Peers to request Chunks (and Layers) to the other Peer. Note that Chunks requested are expressed as ranges.

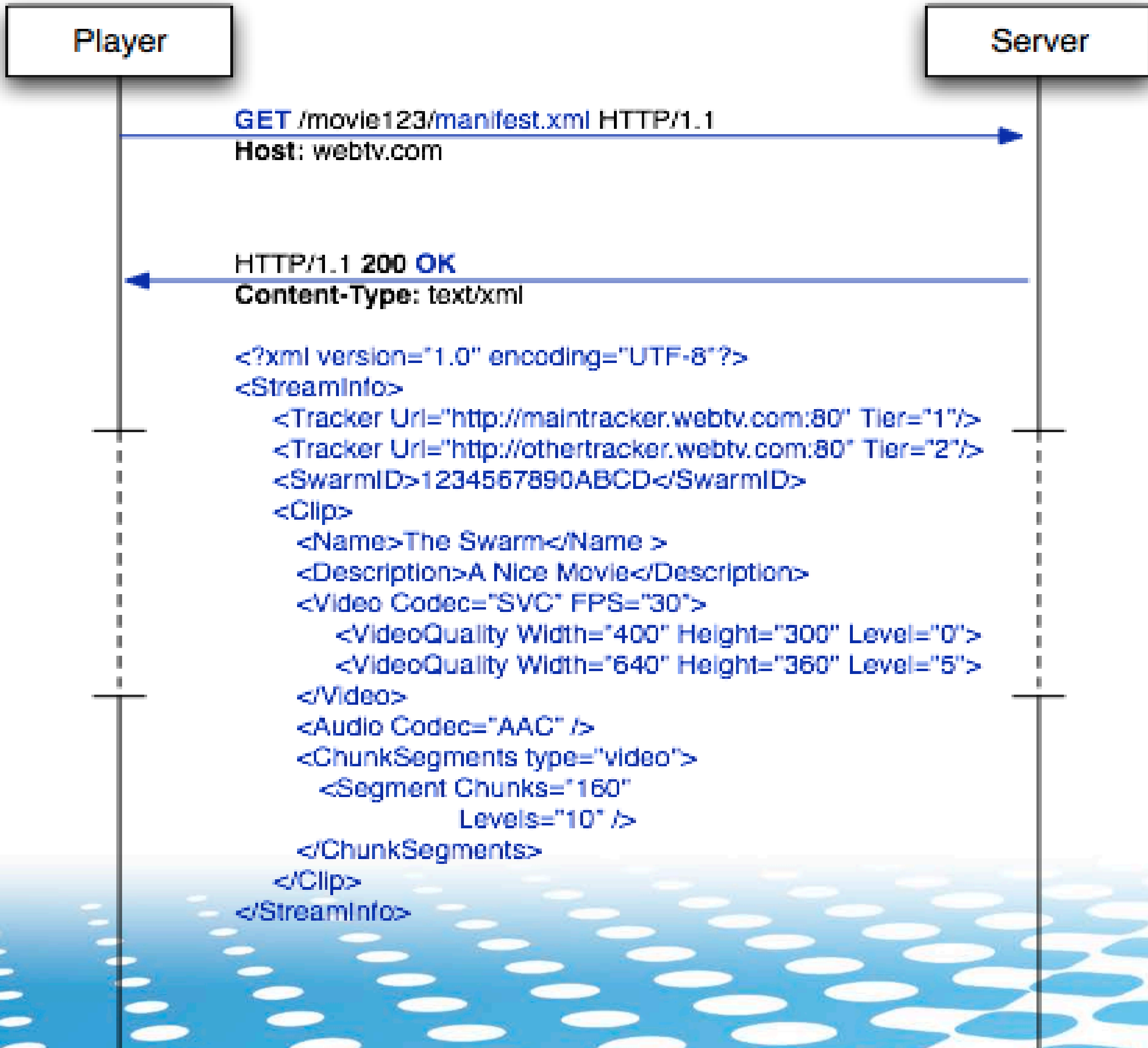
PEER_STATUS: allows a Peer to inform other Peers on its participation status (and statistic data like upload bandwidth). The information conveyed may include information related to Chunk trading like “Choke” (to inform the other Peer of the intention to stop sending data to it) and “Unchoke” (to inform the other Peer of the intention to start/re-start sending data to it).

HTTP Streaming Protocols (for SVC)

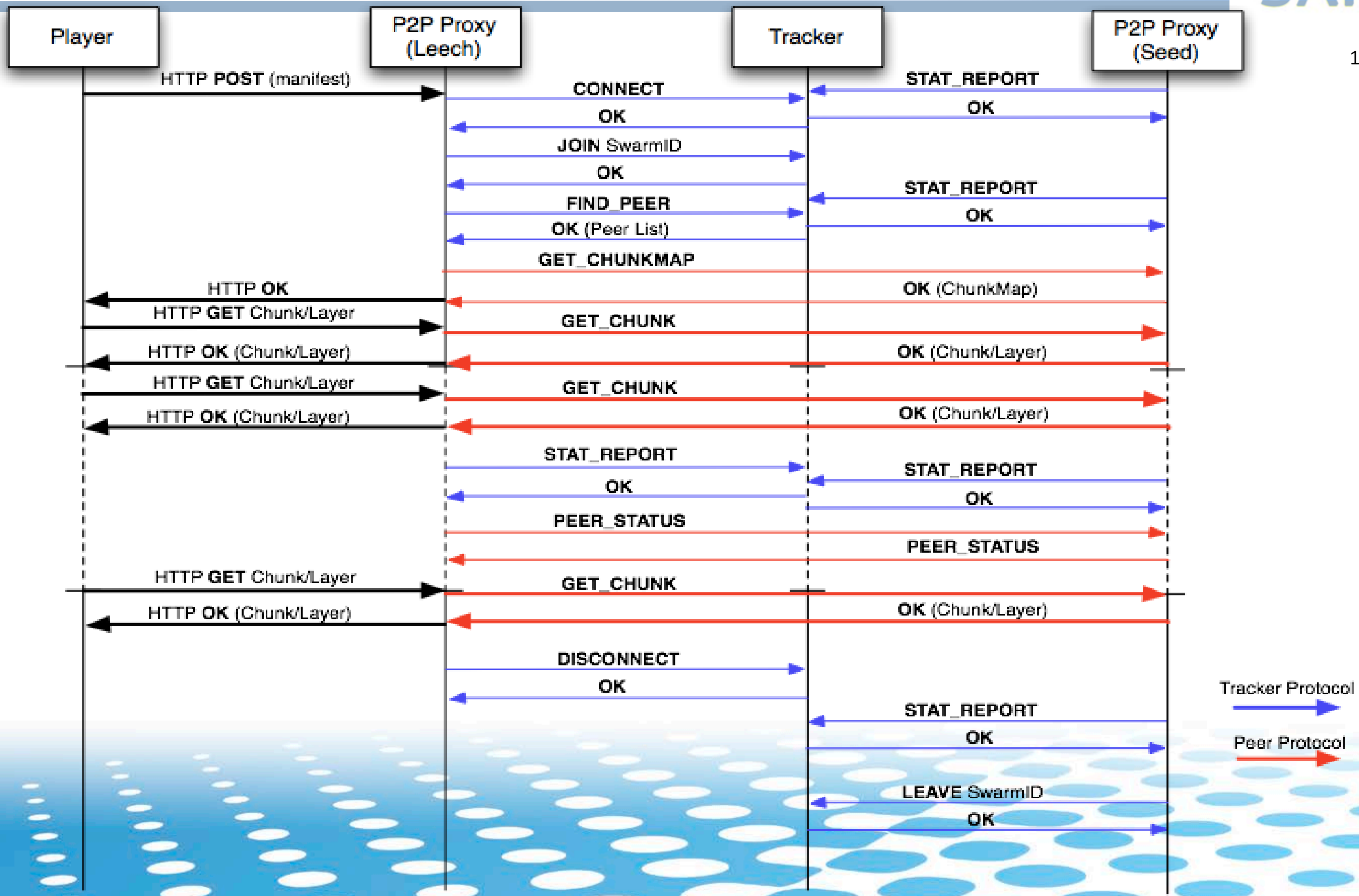
From Encoding to Distribution and Decoding



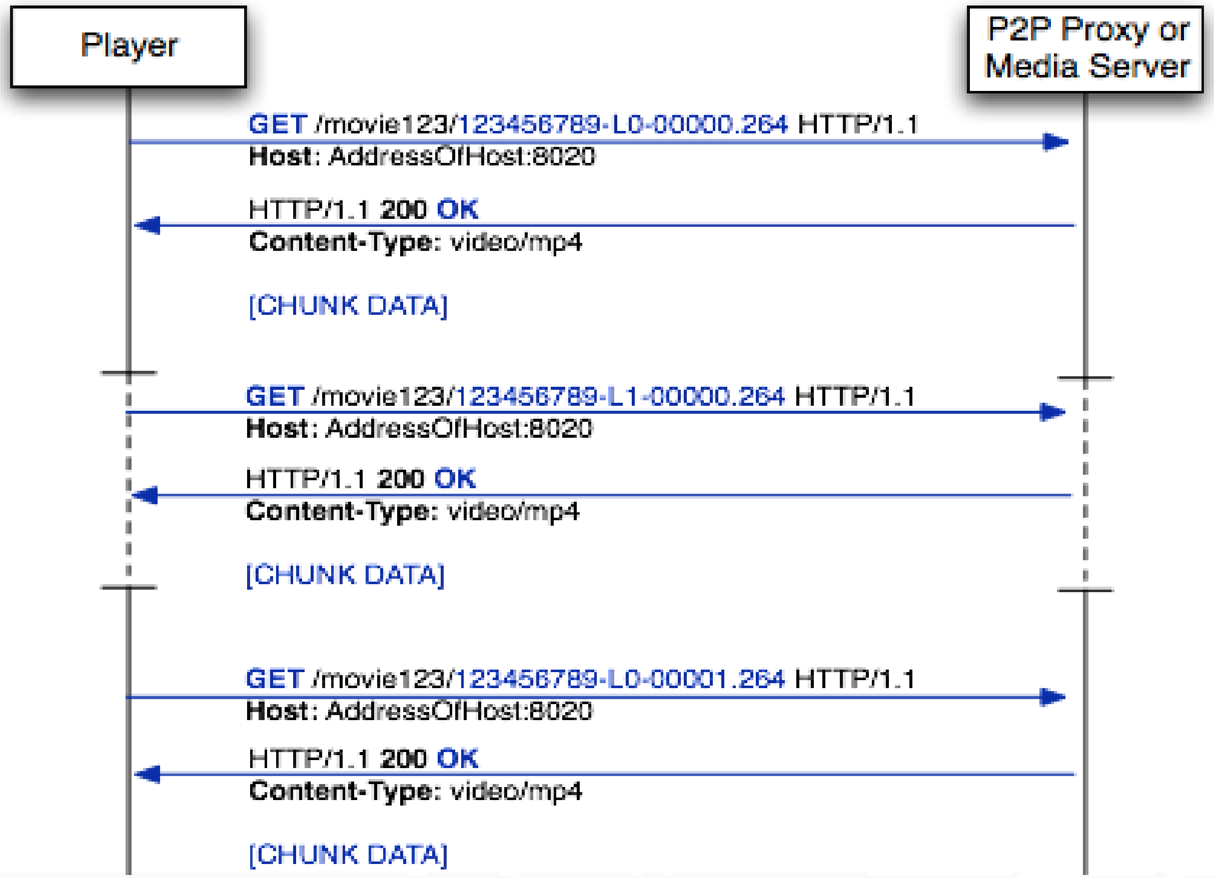
Player: GET Manifest from a Portal



Typical P2P Session

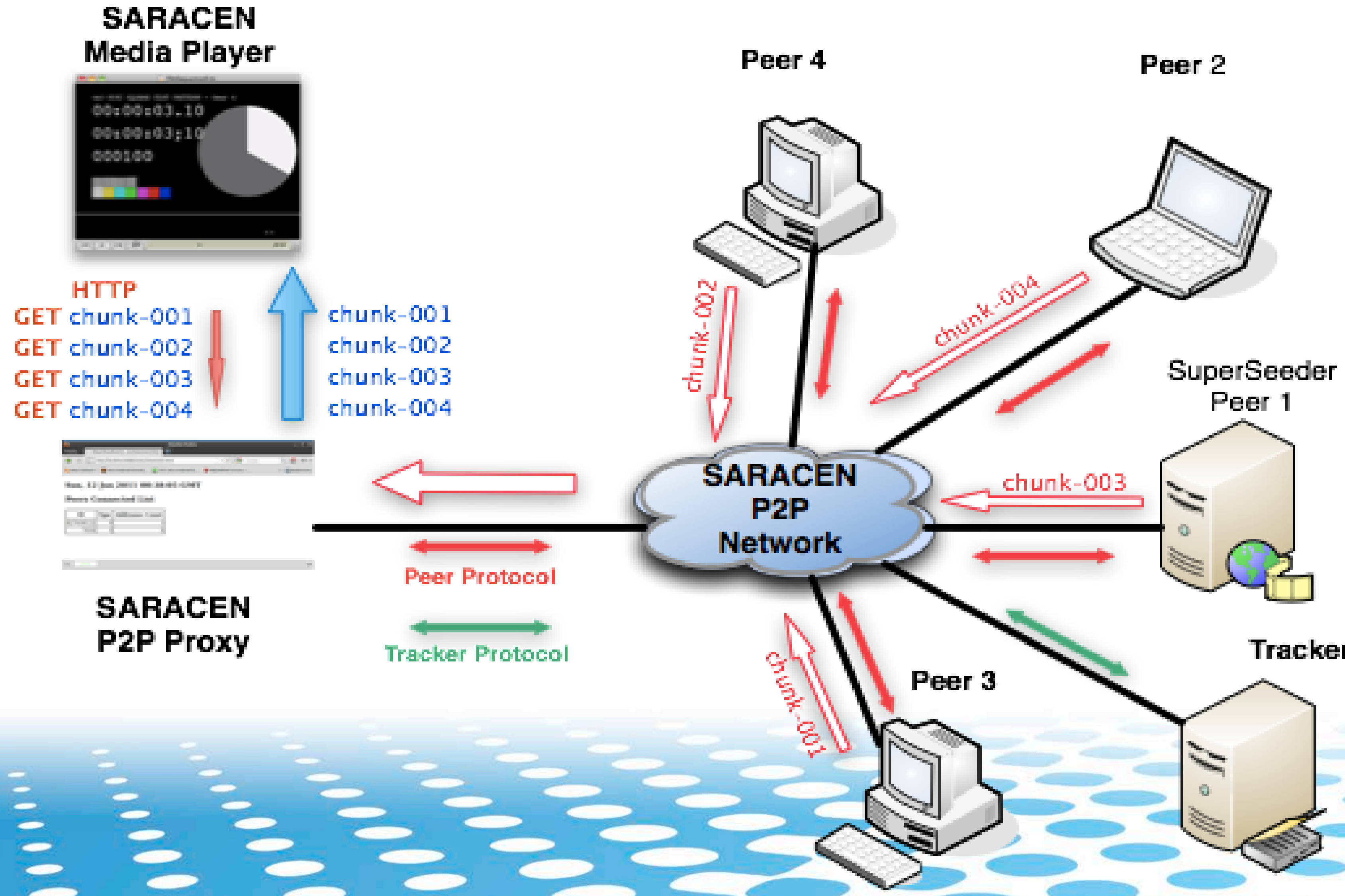


Player: GET chunks from P2P Proxy



SARACEN Streaming architecture

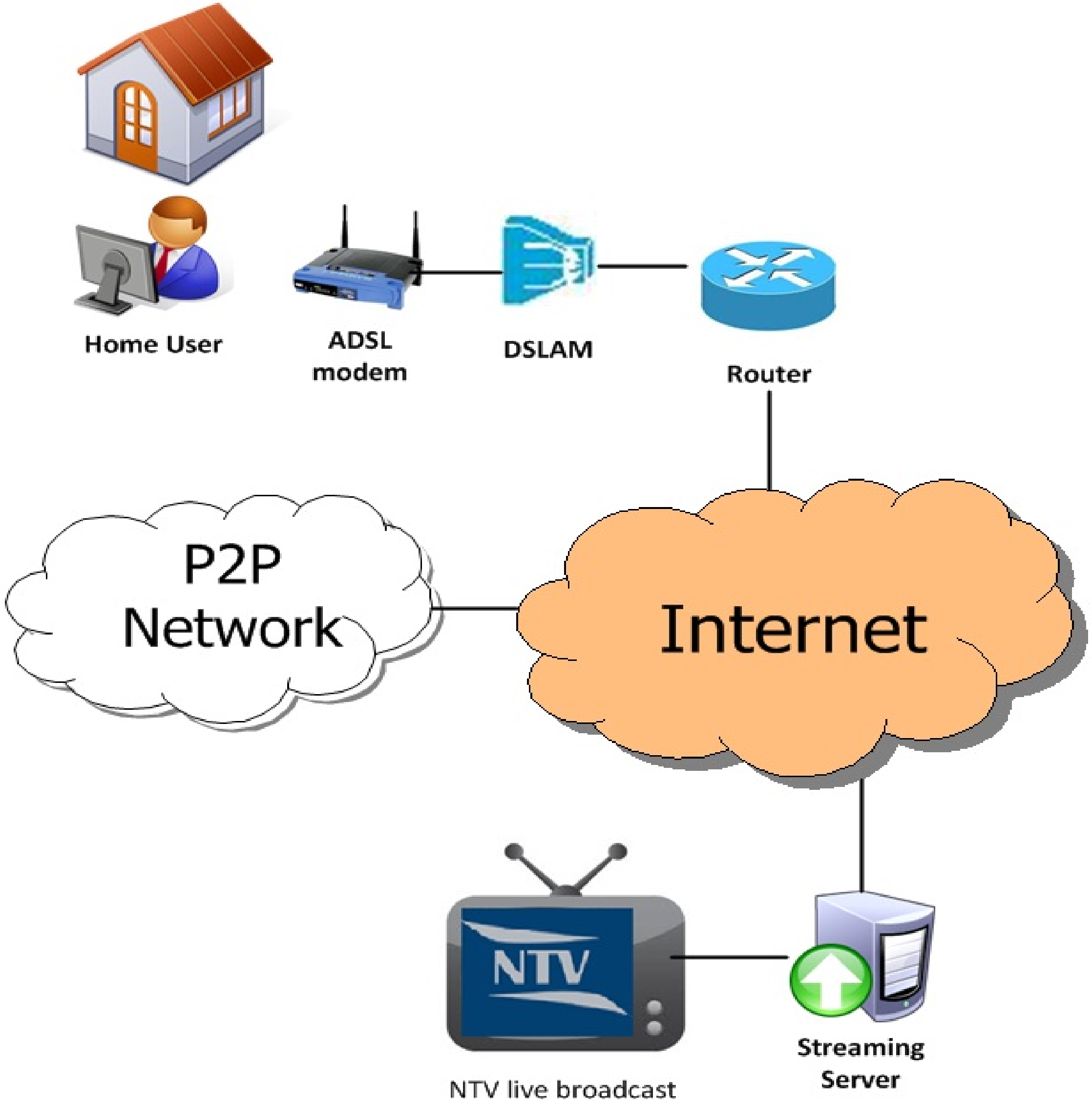
P2P HTTP Media Streaming



References

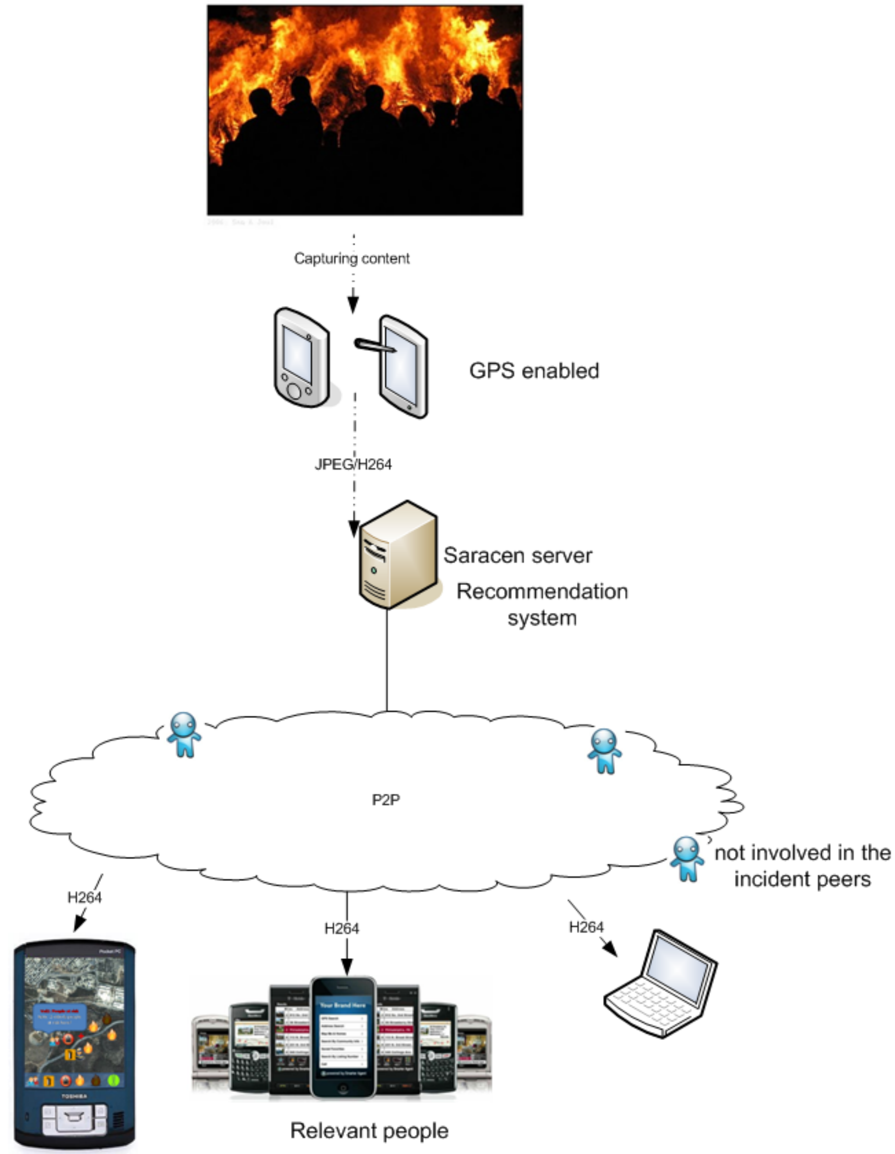
- **IETF draft published for the Peer Protocol (May 2011)**
  - “HTTP-based PPSP Peer Protocol”
    draft-cruz-ppsp-http-peer-protocol-00
- **IETF draft published for the Tracker Protocol (June 2011)**
  - “HTTP-based PPSP Tracker Protocol”
    draft-cruz-ppsp-http-tracker-protocol-00

Real-time streaming TV



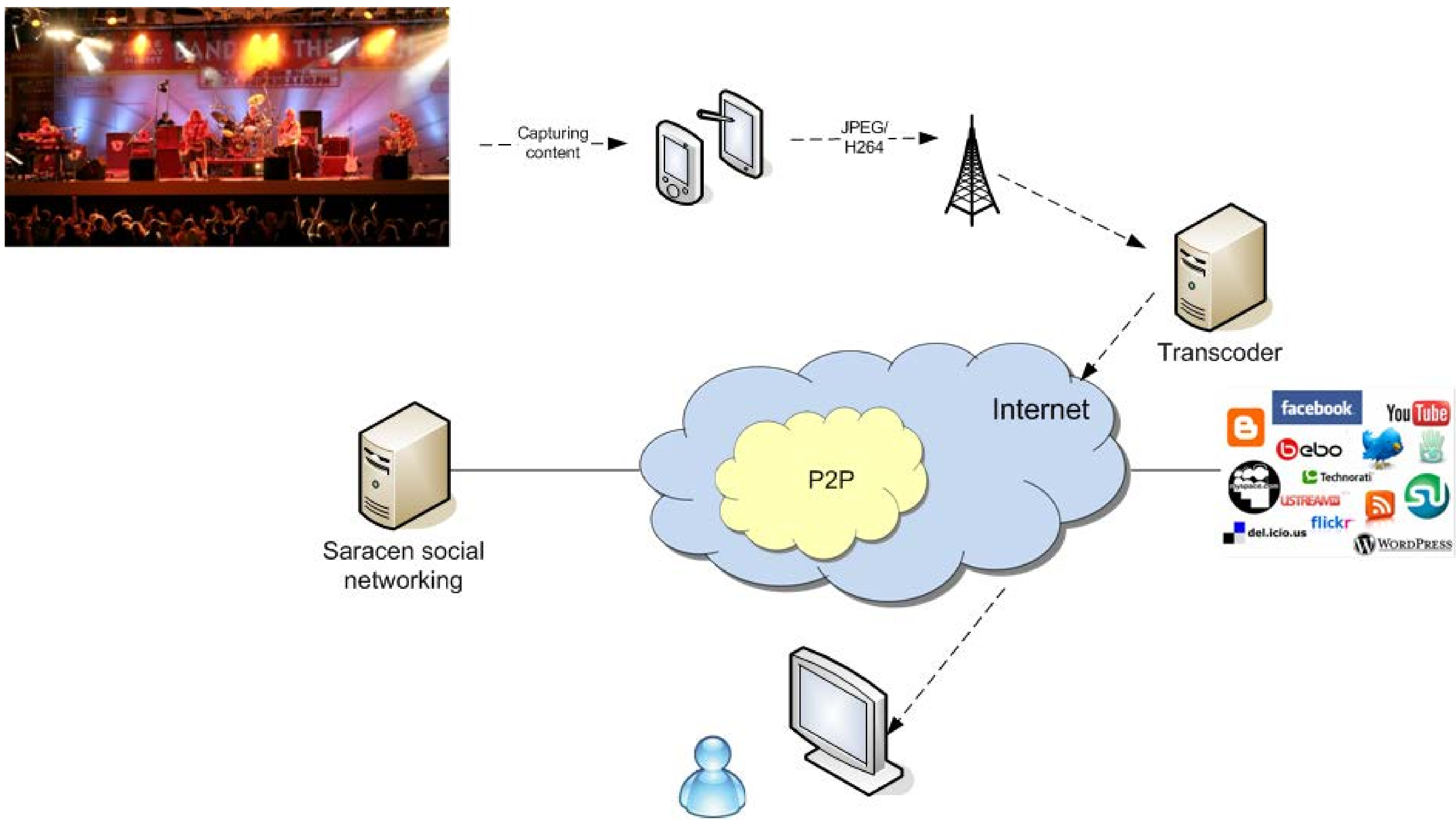
Social Media Applications

- **Rescue**
  - Forrest fire



Social Media Applications

- **Sharing media**
  - Concert



QUALITY OF EXPERIENCE

Information Society Technologies

Quality of Service

- **Packet loss**
- **Delay**
- **Delay jitter (variation)**

Information Society Technologies

Video Quality

- Start-up/switch delay
- Freeze duration
- Motion jumpiness (frame rate related)
- Picture quality (PSNR or other metrics)

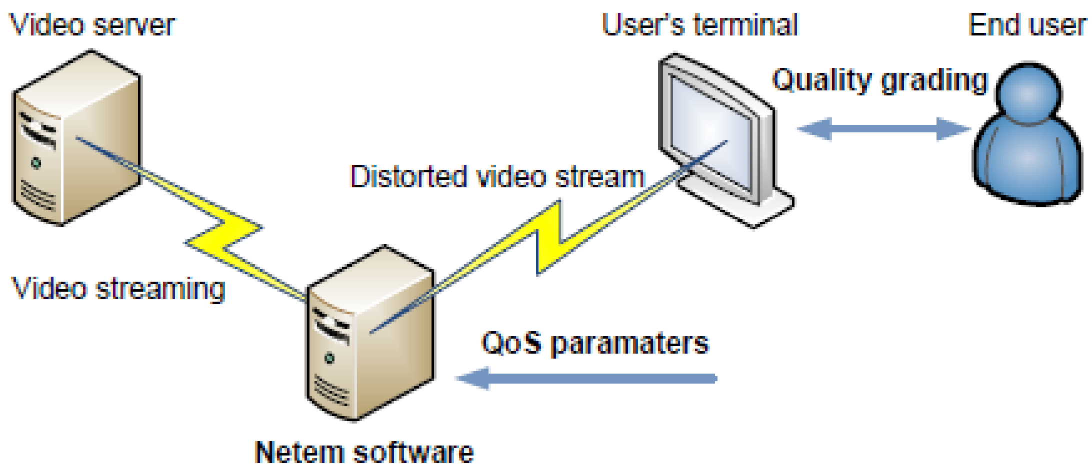
Objective Evaluation

- Full reference
- Reduced Reference
- No reference

Information Society Technologies

Subjective Evaluation

- Encoded video
- Transmitted video



- *ICT COST Action IC1003* European Network on Quality of Experience in Multimedia Systems and Services
- Its main objective is to develop and promote methodologies to subjectively and objectively measure the impact in terms of quality of future multimedia products and services.
- Action Chair: Touradj Ebrahimi

3D VIDEO CODING AND STREAMING

Information Society Technologies

Historical Perspective

- **Analog TV / video**
  - B&W
  - color
- **Digital TV / video**
- **HD TV / video**
- ***Next big step*: 3D HD TV / video**

and

- **3D video over IP, 3D IPTV, 3D webTV**

Information Society Technologies

- **3D Displays**
- **3D Content Generation**
- **3D Coding and Transport**
- **3D Quality of Experience**

- **Special Issue of Proceedings of the IEEE on 3D Media and Displays, April 2011**

Information Society Technologies

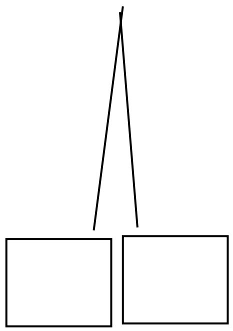
3DTV Project

- Network of excellence (2004-2008)

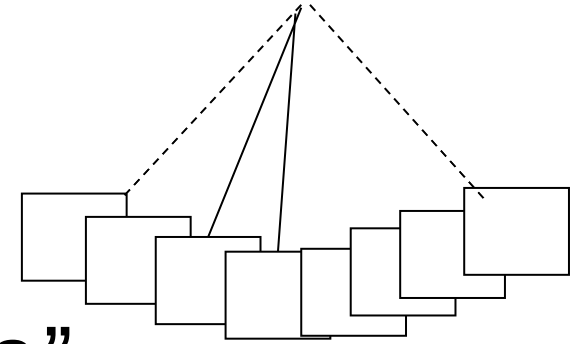
  http://www.3dtv-research.org/
- Follows ATTEST project led by Philips Research, Netherlands
- 21 partner institutions led by Bilkent University, Ankara, Turkey
- Capture, compression, transmission, and display of 3DTV content

3D Video Representations

- **Stereo** "fooling the eyes"
  - With glasses
  - Auto-stereoscopic (color, polarized, time-shutter)



- **Multi-view** (free-view within limited viewing angle)
  - N-views
  - View+Depth
- **Light-field**
- **Holographic** "optical replica"



3D Video Displays

Koç University 3D Video Lab
(with glasses)

HHI head-tracking auto stereo display

- Polarized stereo projection

Auto-stereoscopic

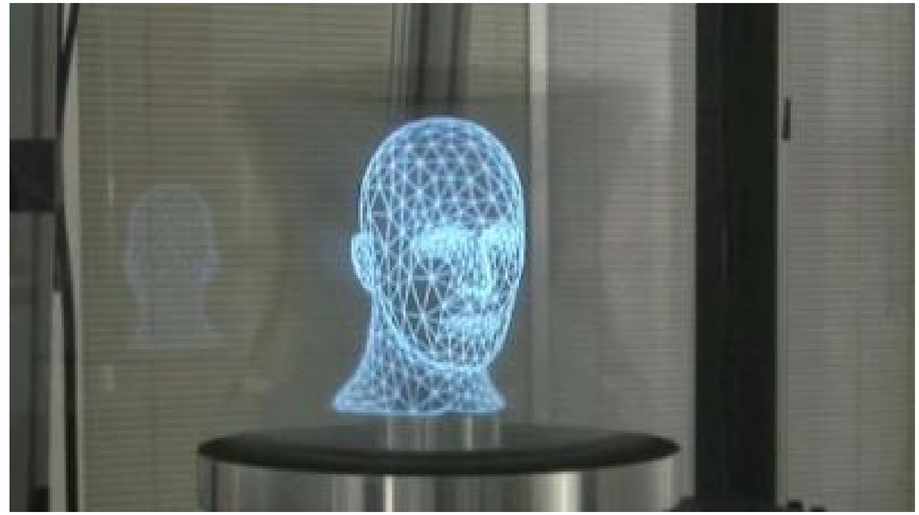
- Lenticular (~9 views)
- Parallax-barrier
- Retroreflective
- Head-tracking auto-stereoscopic

Nokia’s portable 3D display

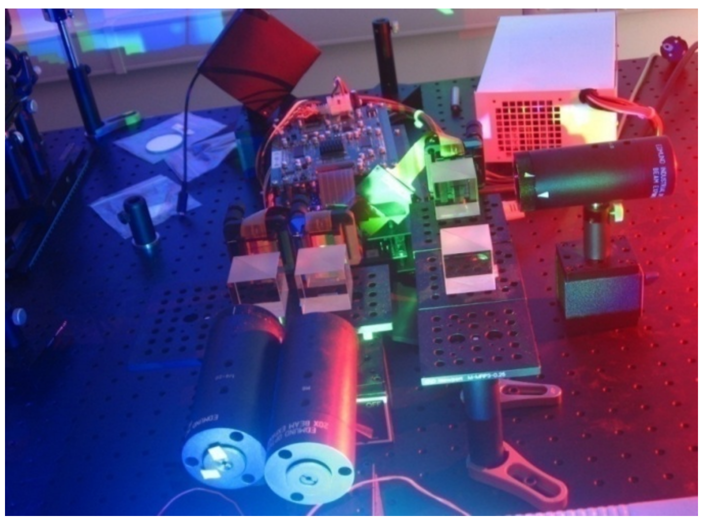
Information Society Technologies

3D Video Displays

Holografika Lightfield display
>45 views



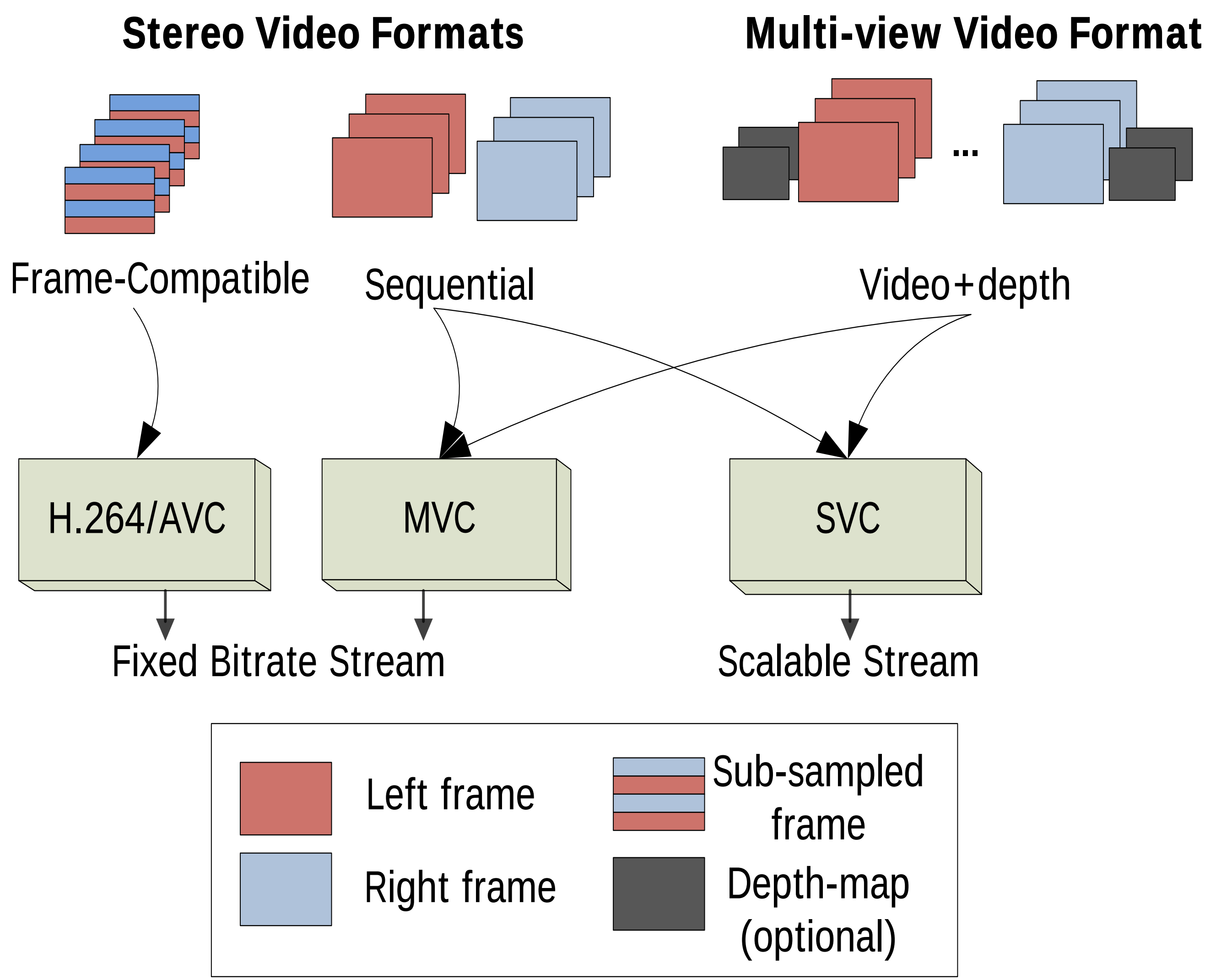
USC spinning mirrors holographic display



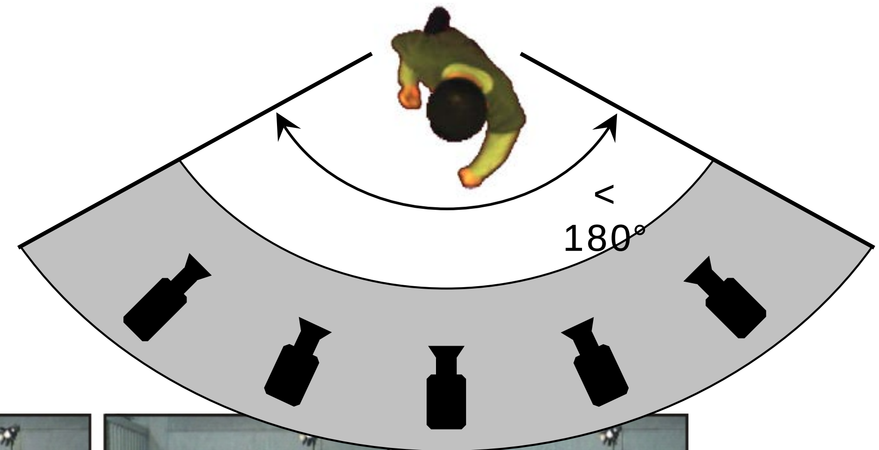
Future laser holographic 3D display

3D Video Compression

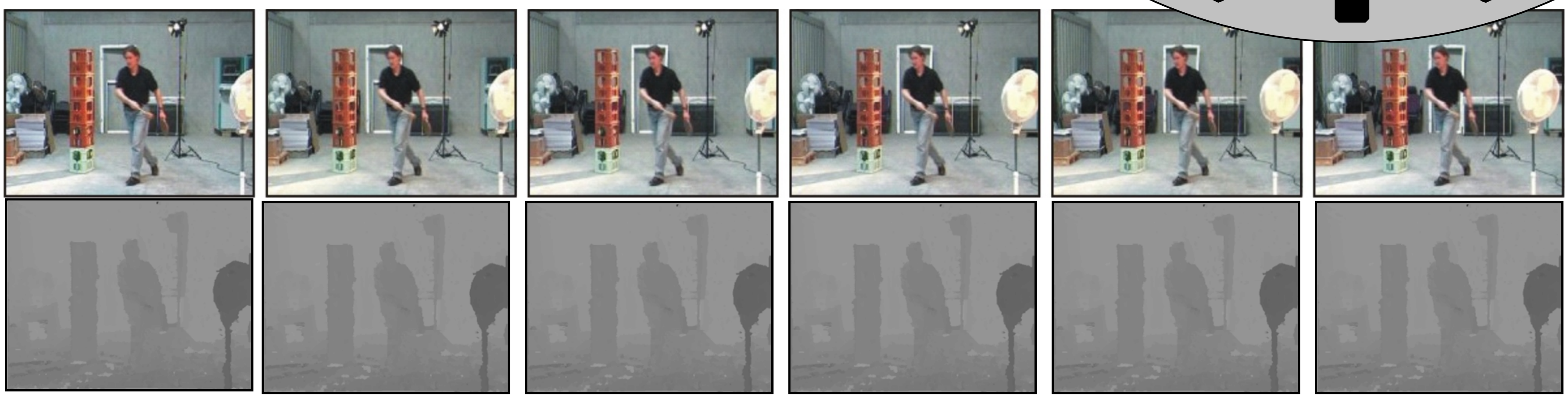
- **Monocular Video Coding - standards**
  - H264/AVC
  - Scalable Video Coding (SVC)
- **Multi-view Video Coding - standards**
  - Multi-view extension of H264/AVC
  - View+depth (MPEG-C)
- **Multi-view Video Coding – ongoing work**
  - Scalable Multi-view coding (SMVC)
  - Multiple description coding
- **3D video coding – future work**
  - Holographic video coding


Stereo Video Formats
Multi-view Video Format
...
Frame-Compatible
Sequential
Video+depth
H.264/AVC
MVC
SVC
Fixed Bitrate Stream
Scalable Stream
Left frame
Right frame
Sub-sampled frame
Depth-map (optional)

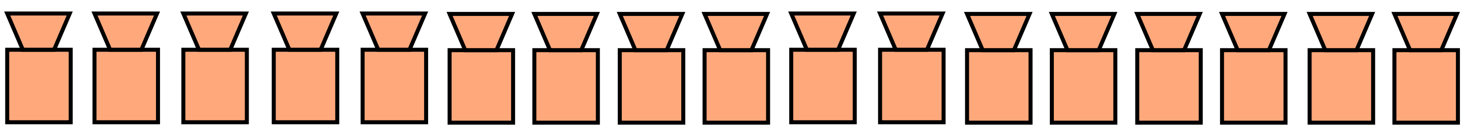
Multi-view Video + Depth (MVD)



Fraunhofer HHI, 3DTV project



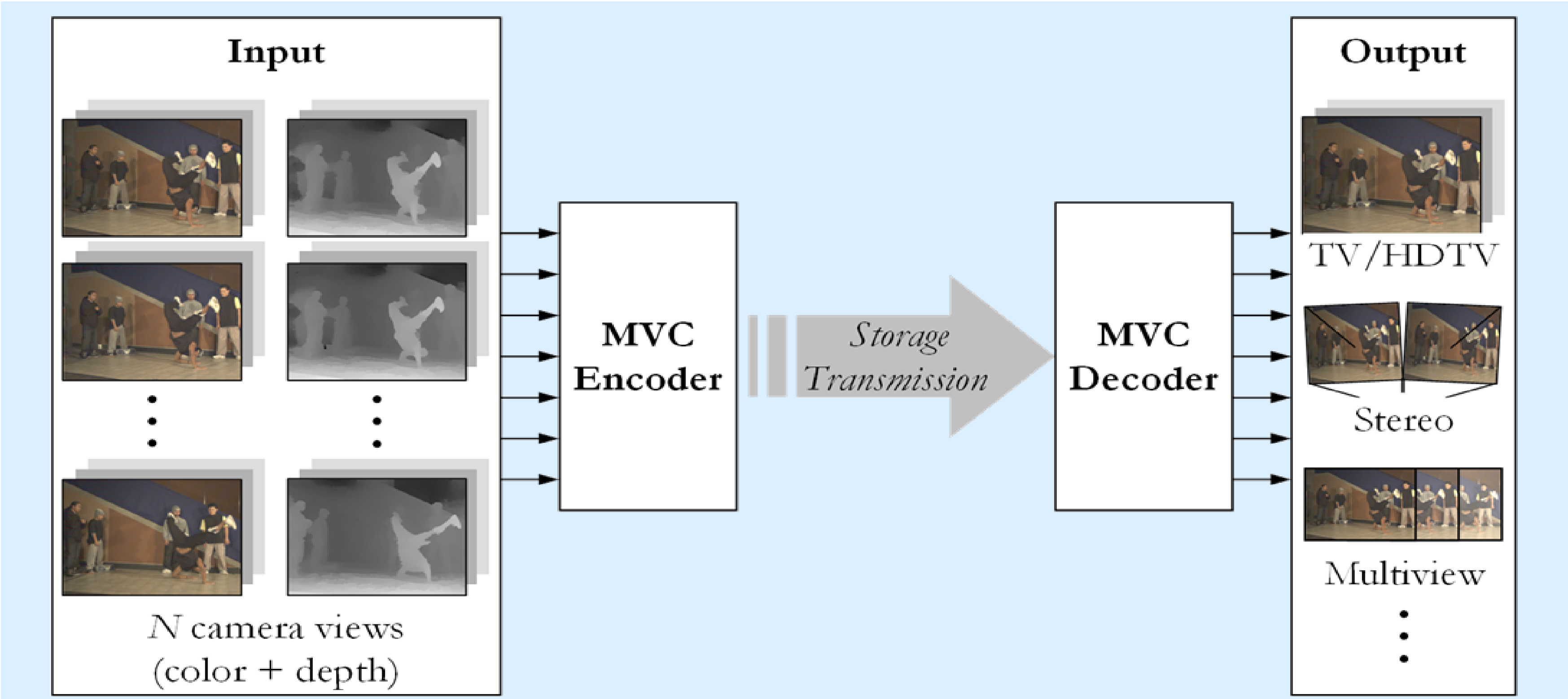
encoder



decoder

45 Virtual Intermediate Views

MVD Encoding



Open source real-time implementation (non-scalable codec) available at:
http://research.nokia.com/research/mobile3D

- The encoder may output one or more (scalable and/or multiple description) streams.
- Adaptation of the source rate to available network rate

Transport Issues with 3DTV

- *Stereoscopic*

  — standardized and broadcast by DVB

  — limited reconstruction of reality

- *Multi-view:*

  — N views of the scene

  — requires less than N times the bandwidth needed for one view if inter-view correlations are exploited

- *Light-fields*

  — very large amount of data need to be transmitted

- *Holographic:*

  — very realistic reconstruction of 3D objects

  — very large amount of data (100 G samples for a single 10 cm x 10 cm hologram)

  — effective compression techniques are not available

Stereo Video Streaming Platform

- **Supported by 3DTV project (Koç and METU)**
  - Linux and Windows, Demo at IBC 2007
- **Server side**
  - MVC and “view+depth” encoding support
  - RTP/UDP/IP (one slice per packet)
  - Variable rate FEC support
  - Requires ~1.6 Mbits/s for SD video incl. 10% FEC
- **Client side**
  - MVC decoding
  - Buffering
  - Multiple 3D display support
  - Frame-based concealment

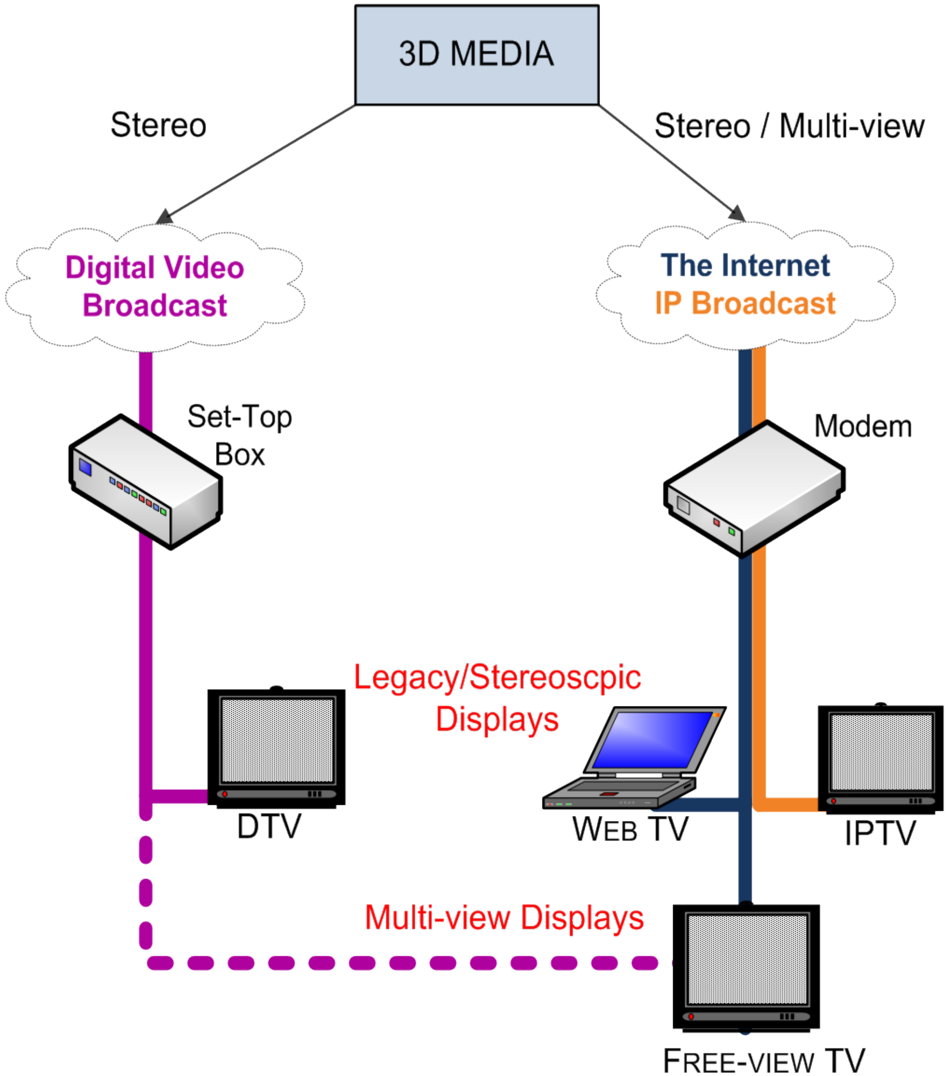
Information Society Technologies

DIOMEDES Project

- FP7 STREP
- DIOMEDES combines broadcast (DVB) and P2P streaming in one converged application. P2P technology will ensure that the network architecture is scalable to large numbers of users.
- 3D Quality of Experience and 3D visual attention models will improve compression efficiency, allowing optimal distribution in varying network conditions and adaptation to different display types.

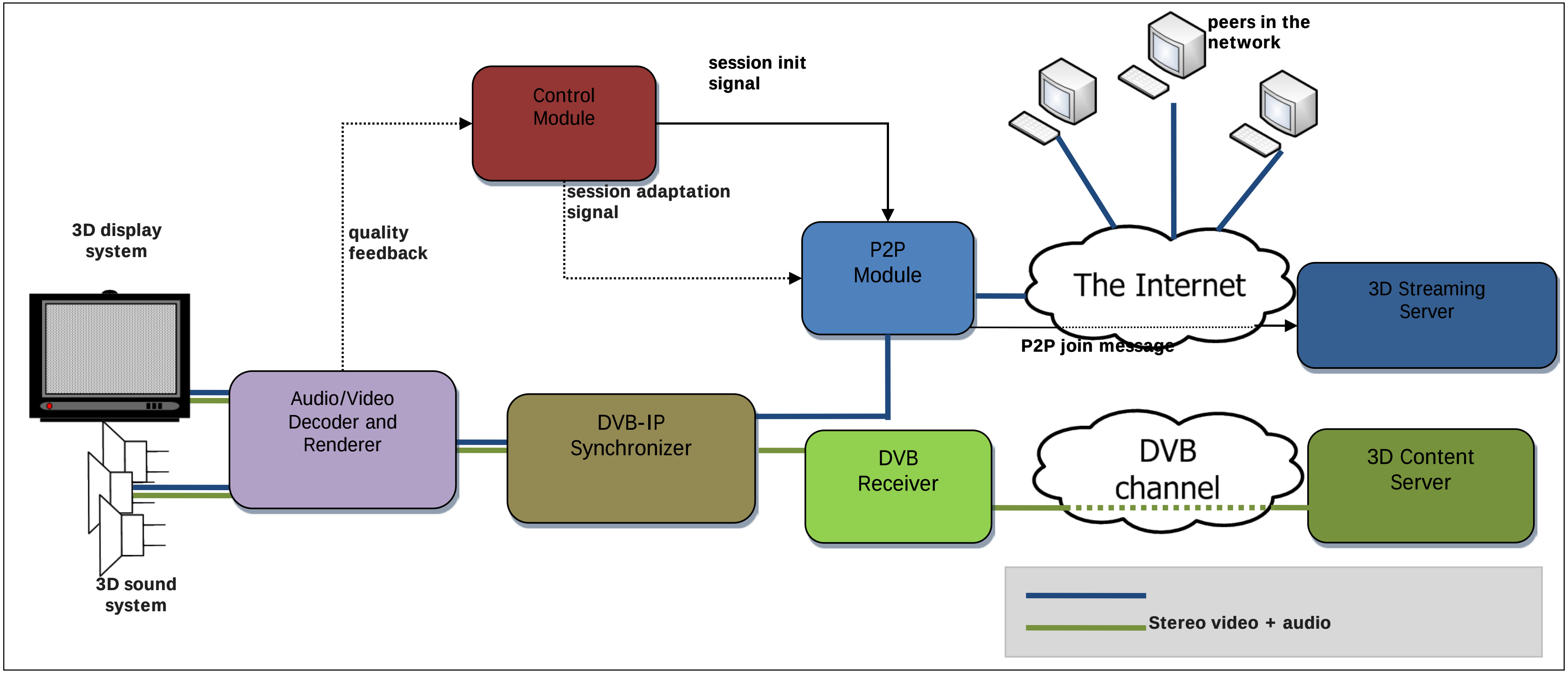
DIOMEDES Concept

- **DVB Stereo over broadcast (frame compatible format)**
- **Additional view over IP**



Information Society Technologies

DIOMEDES Client



3D Video QoE

- **Stereo sickness / headache**
- **Depth perception**
- **Display-dependent quality perception**
  - Full resolution displays
  - Reduced resolution zigging patterns

Information Society Technologies

HbbTV

- **Hybrid Broadcast Broadband TV is a new industry standard providing an open technology platform that seamlessly combines broadcast TV services with services delivered via broadband and also enables access to Internet only services for consumers using connected TVs and set-top boxes.**
- **NHK recently demonstrated a new content delivery system prototype, called Hybridcast Streaming. Currently, most 3DTV providers employ "side-by-side" format, which pumps two images together into a single 1080p frame. You lose resolution in the process. So how do you add that resolution back? By sending half of the signal over typical TV cabling, and the other half over the Internet.**