

# A Practical Guide to Streaming Media

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*[www.thrupoint.net](http://www.thrupoint.net)*



# What this tutorial is about!

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- Streaming Media – Using IP to distribute audio and video
- Discussion of streaming media technologies and best practices, both current and emerging
  - Streaming Servers
  - Caching
  - Content Routing
  - Network Infrastructure

# What this tutorial is not about!

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- How to edit and make good content
- “Stationary” multimedia, i.e. it must be delivered over IP to be discussed here
- Analogue audio and video technology
- Videoconferencing
- Pure theory, our discussion is in practical terms
- Recommendations on commercial implementations

# Who is Thrupoint?

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- Thrupoint – <http://www.thrupoint.net>
  - The largest global pure network consultancy
  - Over 500 engineers
  - Formed in 1996
  - Headquartered in New York City

# What Does ThruPoint Do?

- Internetworking Solutions for:



**Communications**



**Financial Services**



**Life Sciences**



**Media**

- Component Solutions Include:

- IP Telephony • IP Streaming • Wireless LAN • Address Management
- Integrated Network Management • Network Audit • Network Infrastructure
- Optical Infrastructure • PKI • Security Audit • Penetration Testing
- VPN • Firewall Architecture • Intrusion Detection • Internet Offload
- Carrier Voice over Packet • DSL Infrastructure • IP & ATM
- Sonet/SDH Infrastructure • Multi Service Infrastructure

- Practice Areas Include:

- Network Transport • Network Management • Network Enabling
- Security • Voice/Video • Content Networking

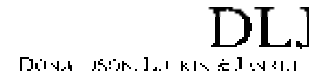
# Where does ThruPoint Operate

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- 17 Markets in North America & 7 International in 2001



# Who Are ThruPoint's Clients?



# Agenda

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- Introduction
- Digitisation of Media Content
  - Audio
  - Video
- IP infrastructure for streaming media
  - QoS
  - Multicast



# Agenda

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- Delivery Technologies
  - Downloading & Streaming
  - Rich Media
  - Interoperability
  - Content Distribution
  - Content Indexing
  - Caching
  - Content Routing
  - Redundancy
  - Security
- Practical Tips
- Future

# Introduction

## Streaming Media Applications

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- Download-and-play
  - Content that will be replayed multiple times on client system, e.g. Music or Movies
- Streamed live
  - Content that is live, e.g. Television and Radio
- Streamed on-demand
  - Content that is played whilst receiving in buffer, e.g. audio/video clips, pay-per-view

# Introduction

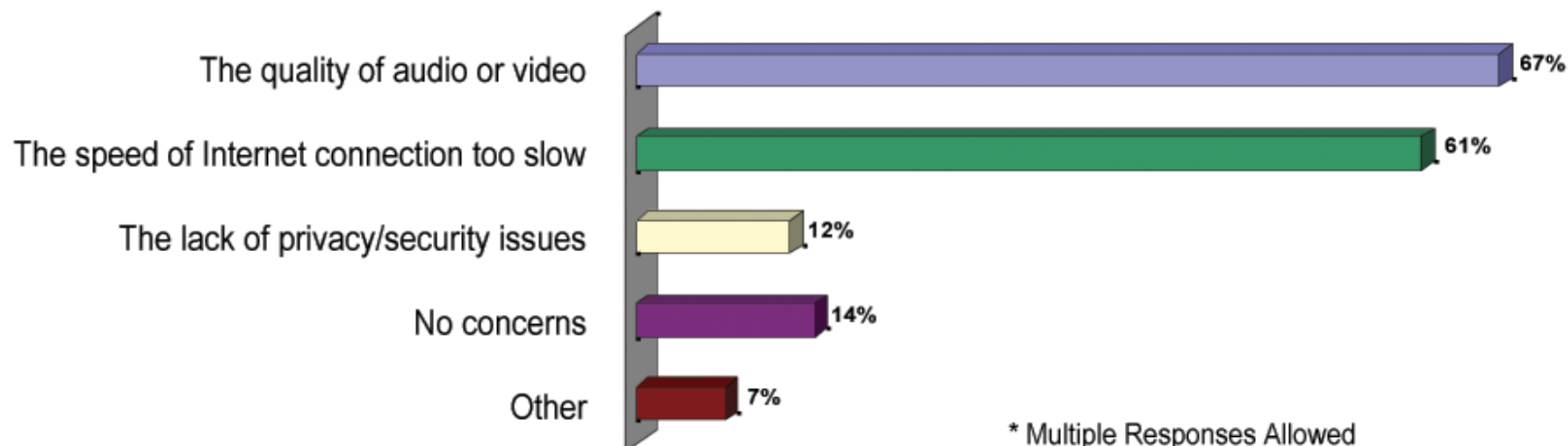
## Key Numbers

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- Streaming Media will grow to over US\$ 2.5 Billion by 2004 (Internet Research Group)
- Over 10% of corporate websites use streaming media for corporate communications
- However



## Quality and Speed are Major Concerns



### DIFFERENCES ACROSS MAJOR DEMOGRAPHICS

AGE: older users are more concerned with privacy and security than younger users; younger users are more concerned with speed than older users

# Agenda

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- Introduction
- Digitisation of Media Content

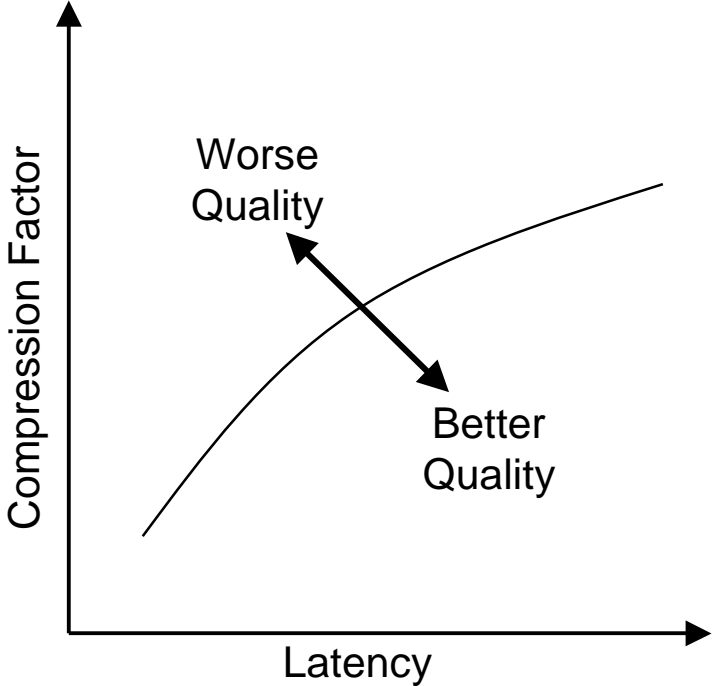
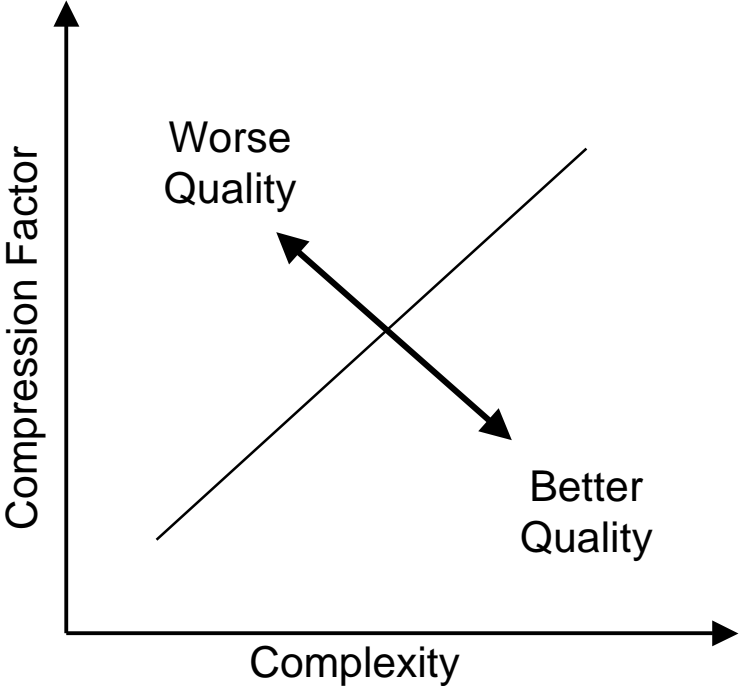
# Digitisation of Media Codecs

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- Codec - Compression and Decompression method to compress audio / video whilst preserving quality
- All codec schemes used for streaming are “lossy”, parts of signal are discarded to achieve compression
- Codecs compress based upon mathematical algorithms and / or perceptual coding algorithms
- Codecs consist of four parameters:
  - Compressed rate – The bit rate for the payload
  - Complexity – The better the compression and resulting quality, the more CPU processing the Codec needs on each end.
  - Quality – The higher the quality, the lower the compression
  - Delay – More complex Codecs often take longer to compress and decompress

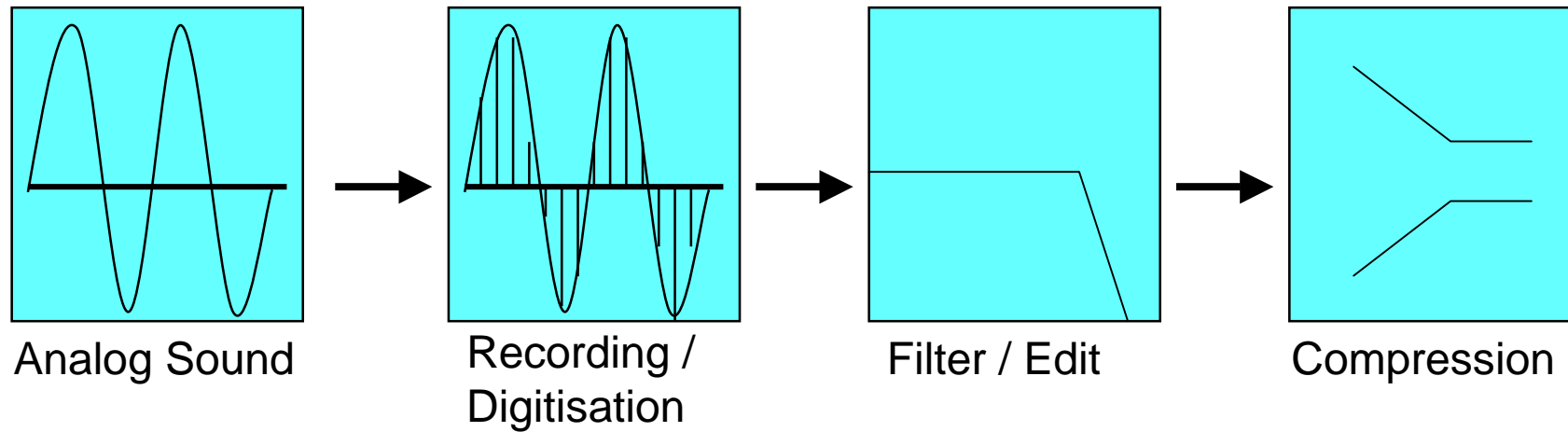
# Digitisation of Media Codecs

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# Digitisation of Media - Audio

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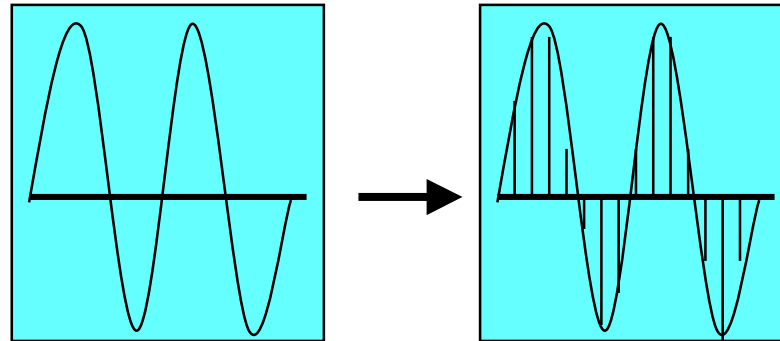




# Digitisation of Media

## Audio Recording / Digitisation

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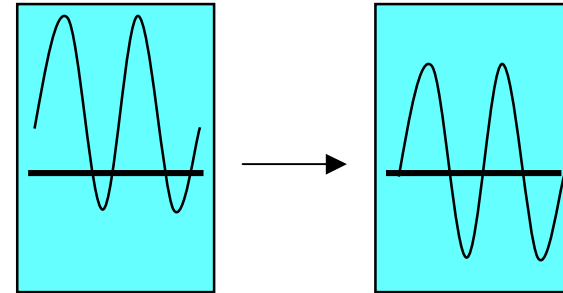
- Our ears only hear frequencies reliably up to 16 kHz, up to 20 kHz for children
- Sampling must be done at least twice the frequency of the original signal to recreate - Nyquist Theorem

# Digitisation of Media

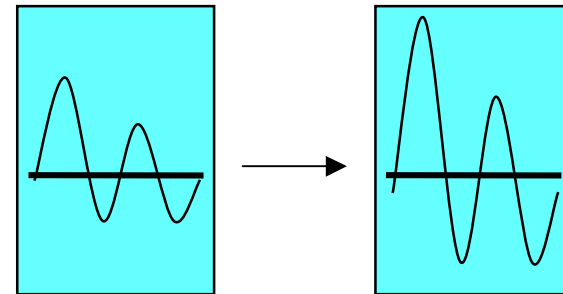
## Audio Filtering / Editing

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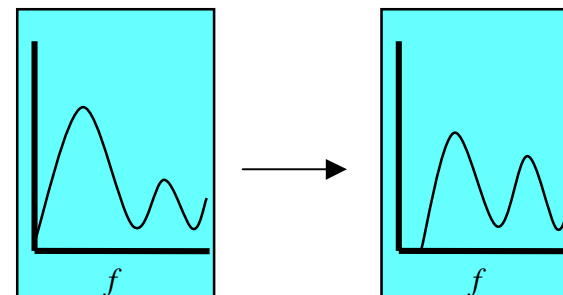
- Remove DC Offset



- Normalize



- Equalize



# Digitisation of Media

## Audio Compression

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- Different codecs are optimized towards different content and different bitrate:

Codec	Content Type	Bitrate
MP3	Hi-Fi music	56 kbit/s or more
G.723.1	Voice	5.3 / 6.3 kbit/s
Qmusic	Hi-Fi Music	56 kbit/s or less
Ogg Vorbis	Hi-Fi Music	56 kbit/s or more
RealNetworks Codecs	Various	Various
Microsoft Codecs	Various	Various

# Digitisation of Media

## Audio Compression - MP3

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- MPEG-1 Layer III
- ISO Standards based codec
- Royalty must be paid to patent-holders for codec implementations, and for sale of MP3 encoded media
- At a compressed bitrate of 128 kbit/s, achieves close to CD-like quality (Compression 11:1)
- Uses perceptual coding techniques addressing the perception of sound waves by the human ear
- Can encode using CBR (Constant Bit Rate) or VBR (Variable Bit Rate)

# Digitisation of Media

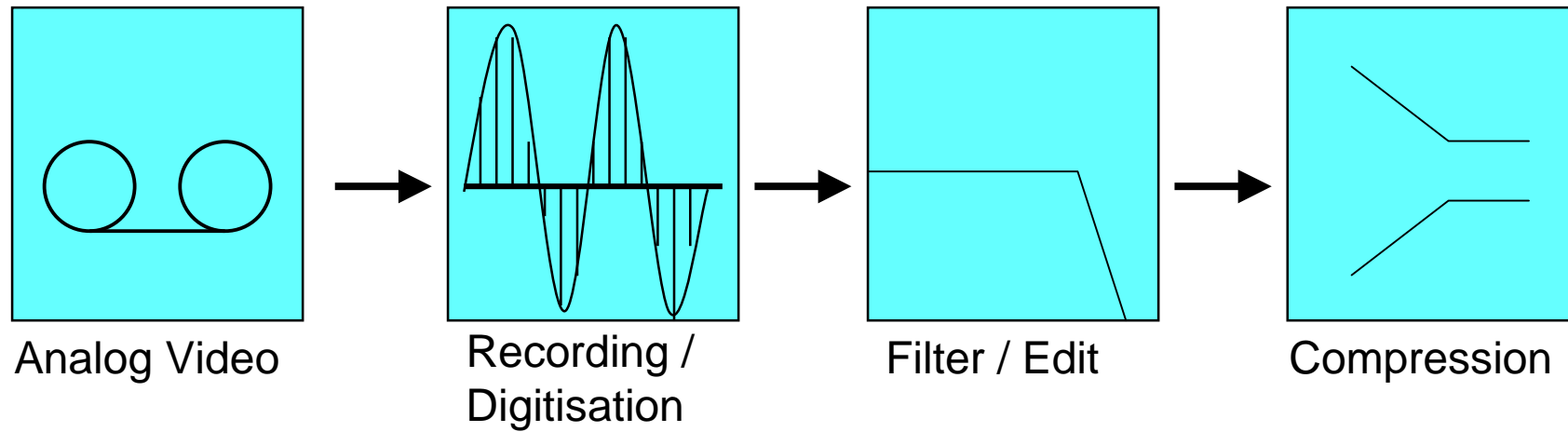
## Audio Compression - Demo

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- Voice [Microsoft @ 6 kbps](#)
- Voice [Microsoft @ 32 kbps](#)
- Music [RealNetworks @ 8 kbps](#)
- Music [MP3 @ 128 kbps](#)

# Digitisation of Media - Video

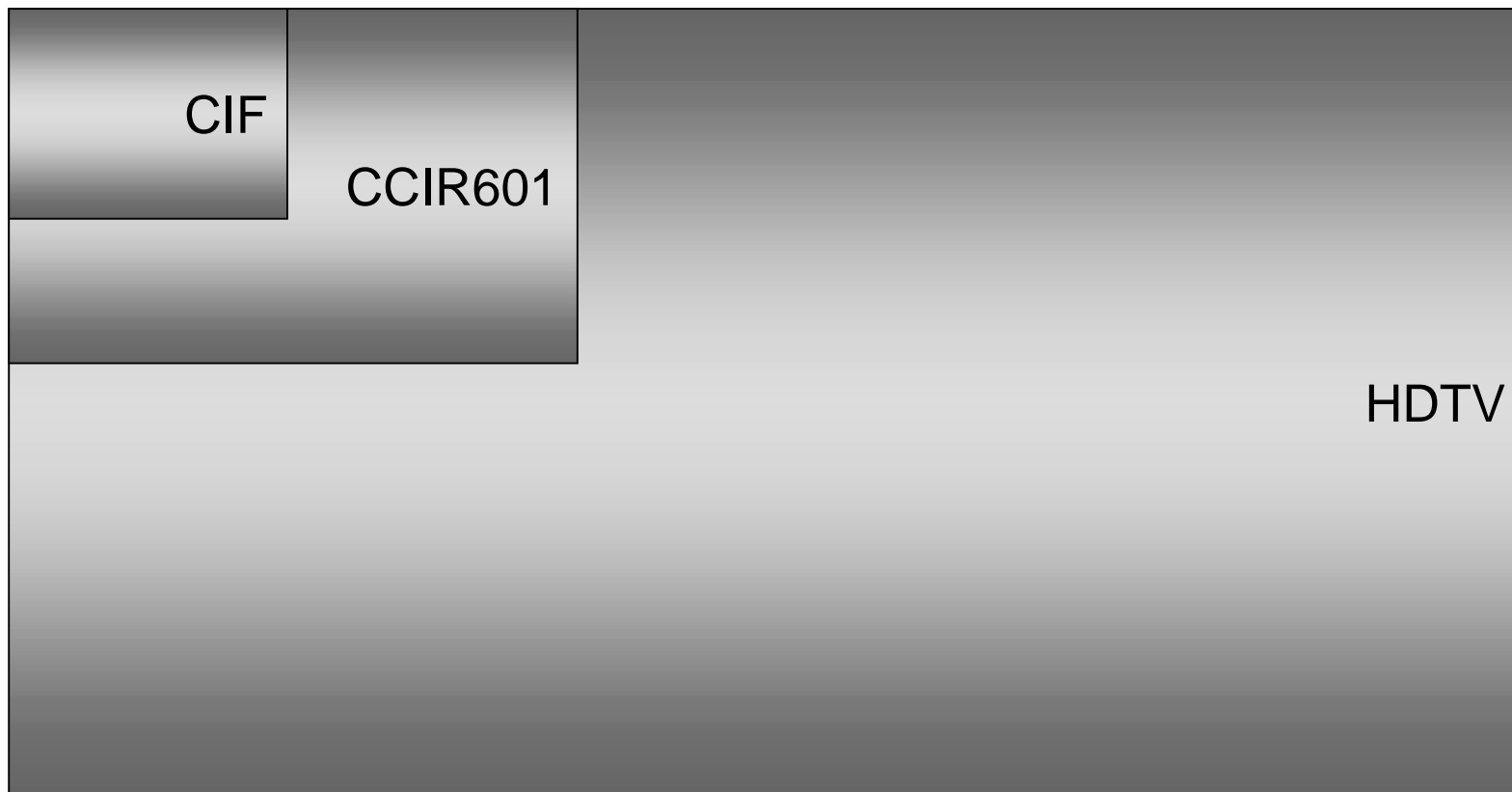
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# Digitisation of Media

## Video Picture Size

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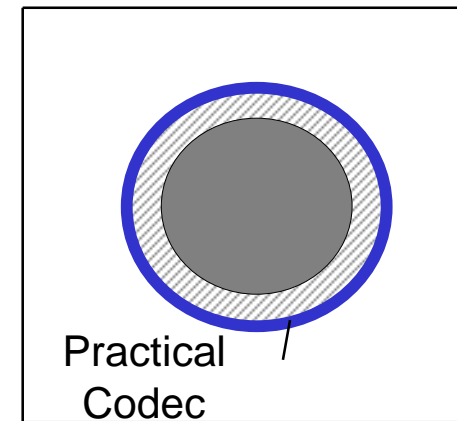
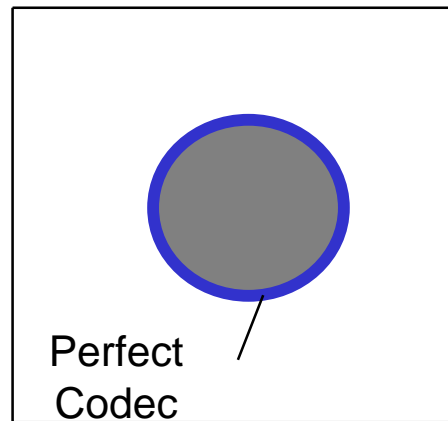
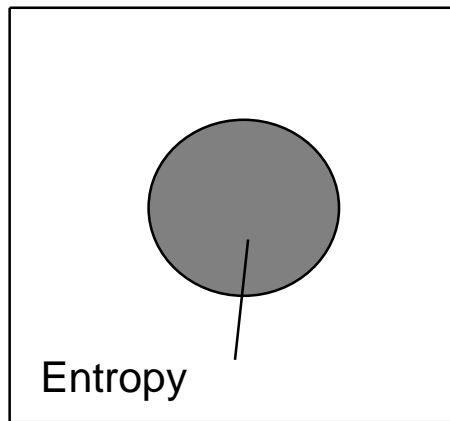


# Digitisation of Media

## Video Codecs

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- Perfect codec would only transmit difference between frames – Entropy
- In practice, codecs aren't able to process video perfectly, and will have to transmit more than entropy





# Digitisation of Media

## Video Compression

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- Different codecs are optimized towards different content types and bitrates:

Codec	Optimized for
H.261	Videoconferencing, n* 64 kbit/s
H.263	Videoconferencing, low bitrates
MJPEG	Real-time compression, editing
MPEG-4	Streaming media, low bitrates
MPEG-1	TV Quality, up to 1.5 Mbit/s
MPEG-2	Broadcast quality, up to 6 Mbit/s
Microsoft Codecs	Varies
RealNetworks Codecs	Varies

# Digitisation of Media

## Video Compression – H.26X

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- ITU Codecs developed for videoconferencing. Algorithms optimized for pictures with low motion
- H.261 developed for ISDN conferencing, thus  $n * 64$  kbit/s
- H.263 is an improved version of H.261, approximately 30% less bits per pixel
- H.263 not limited to  $n * 64$  kbit/s
- Use of certain algorithms for compression
  - Motion prediction and estimation
  - Discrete Cosine Transformation
  - Both spatial and temporal algorithms

# Digitisation of Media

## Video Compression – MJPEG

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- MJPEG is basically JPEG compression frame-by-frame
- Spatial compression only
- Can run in software on a regular PC in real-time
- Format used when temporal compression (over time) is a concern. This might be for situations where compression and decompression occurs multiple times (e.g. editing), artifacts due to staggered compression are far less with MJPEG than MPEG-1/2

# Digitisation of Media

## Video Compression – MPEG-1/2

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- Defined by MPEG (Moving Picture Expert Group) within ISO
- Algorithms made for general video compression, handle motion well
- Spatial and temporal compression
- **MPEG-1** - Optimized for CIF resolution. Bitrate at CIF is aprox. 1.2 Mbit/s. Used on Video-CD
- **MPEG-2** - Improvement on MPEG-1. Optimized for CIF or CCIR601 resolution. Bitrate at CCIR601 is aprox. 6 Mbit/s. Used on DVD. Licensing fees apply

# Digitisation of Media

## Video Compression – MPEG-4

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- MPEG-4 is similar to MPEG1 / 2 in that it is a compression standard, but in addition encompasses coding of audio-visual objects:
  - Allows interaction with the content
  - Improves reusability and coding of the content
  - Allows content-based scalability
- MPEG-4 is currently optimized at QCIF and CIF resolution at:
  - Below 64 kbit/sec
  - 64 - 384 kbit/sec
  - 384- 4 Mbit/sec
- Also optimized for CCIR601 and HDTV:
  - 2, 15 and 38.4 Mbit/sec
- Finally, work being done for MPEG-4 for Studio use:
  - 2000 x 4000 pixels
  - Beyond 300 Mbit/sec

# Digitisation of Media

## Video Compression – MPEG

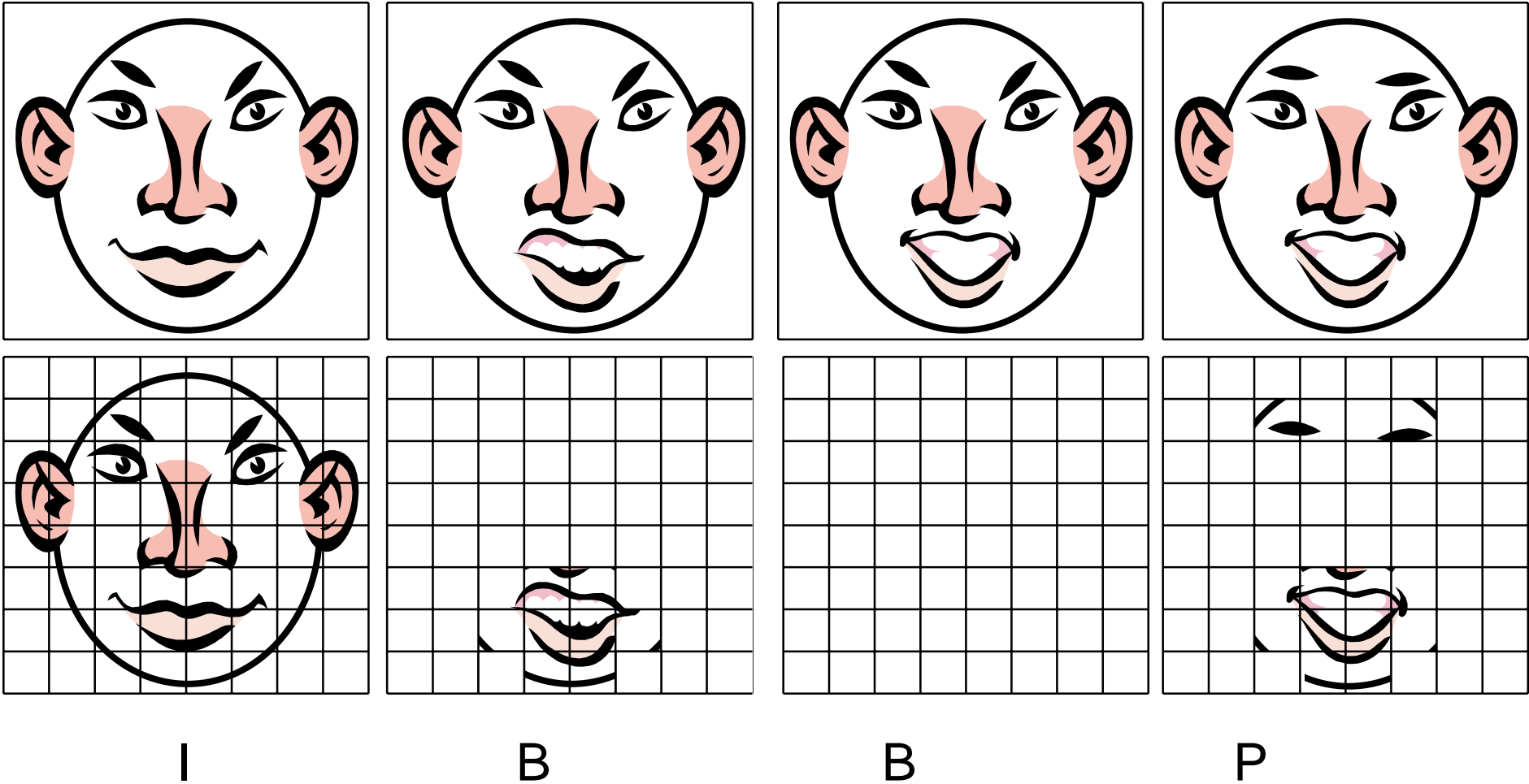
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- Has concept of Intra / reference frames and predicted frames
- Intra frames (I-frames) contain entire picture. I-frames are JPEG compressed
- Picture is broken up into quadrants called macroblocks, typically 8x8 or 16x16 pixels
- Predicted frames (P-frames) only transmit the difference between itself and the previous P or I frame for each respective macroblock
- Bi-directional frames (B-frames) use information from previous and next I or P frame and only transmits the macroblocks that aren't matched in either

# Digitisation of Media

## Video Compression - MPEG

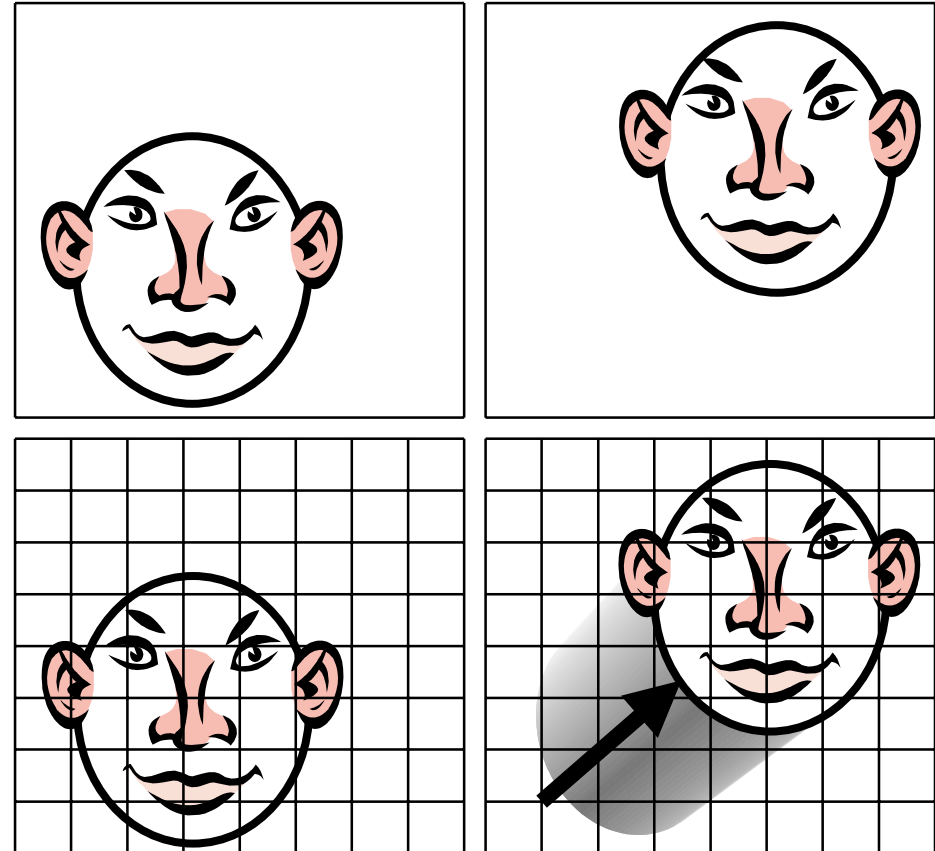
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# Digitisation of Media Motion Compensation

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- Can compress further by predicting movement and shifting picture using vector of movement
- Vectors are on a macro-block basis





# Digitisation of Media

## Video Compression - MPEG-4

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- Can create unique objects
- Objects can change independently of each other
- Demo: <http://www.objectvideo.com>



# Digitisation of Media

## Video Compression - MPEG-4

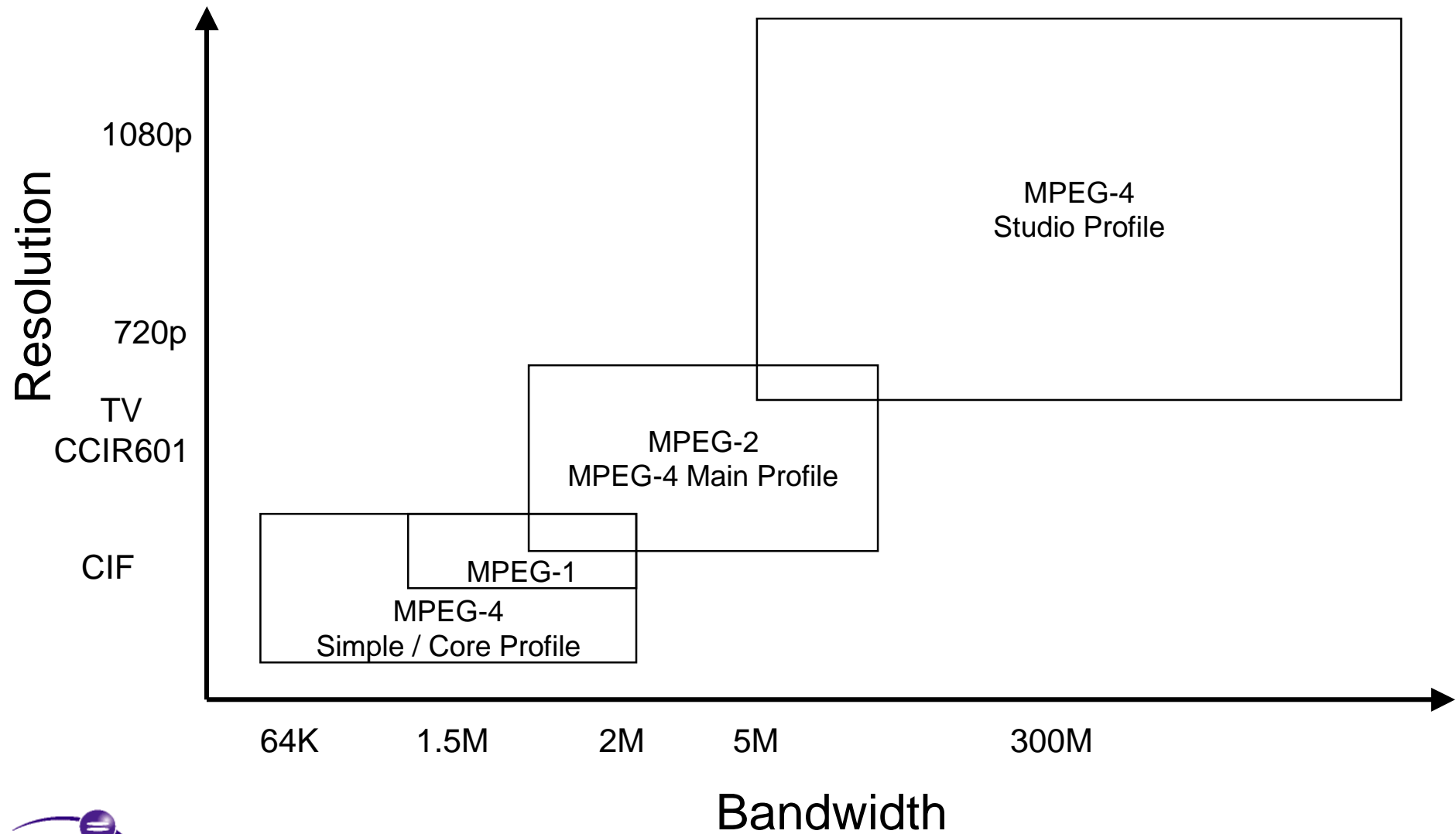
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- Can pan and zoom in a scene without having to re-transmit objects



# Digitisation of Media

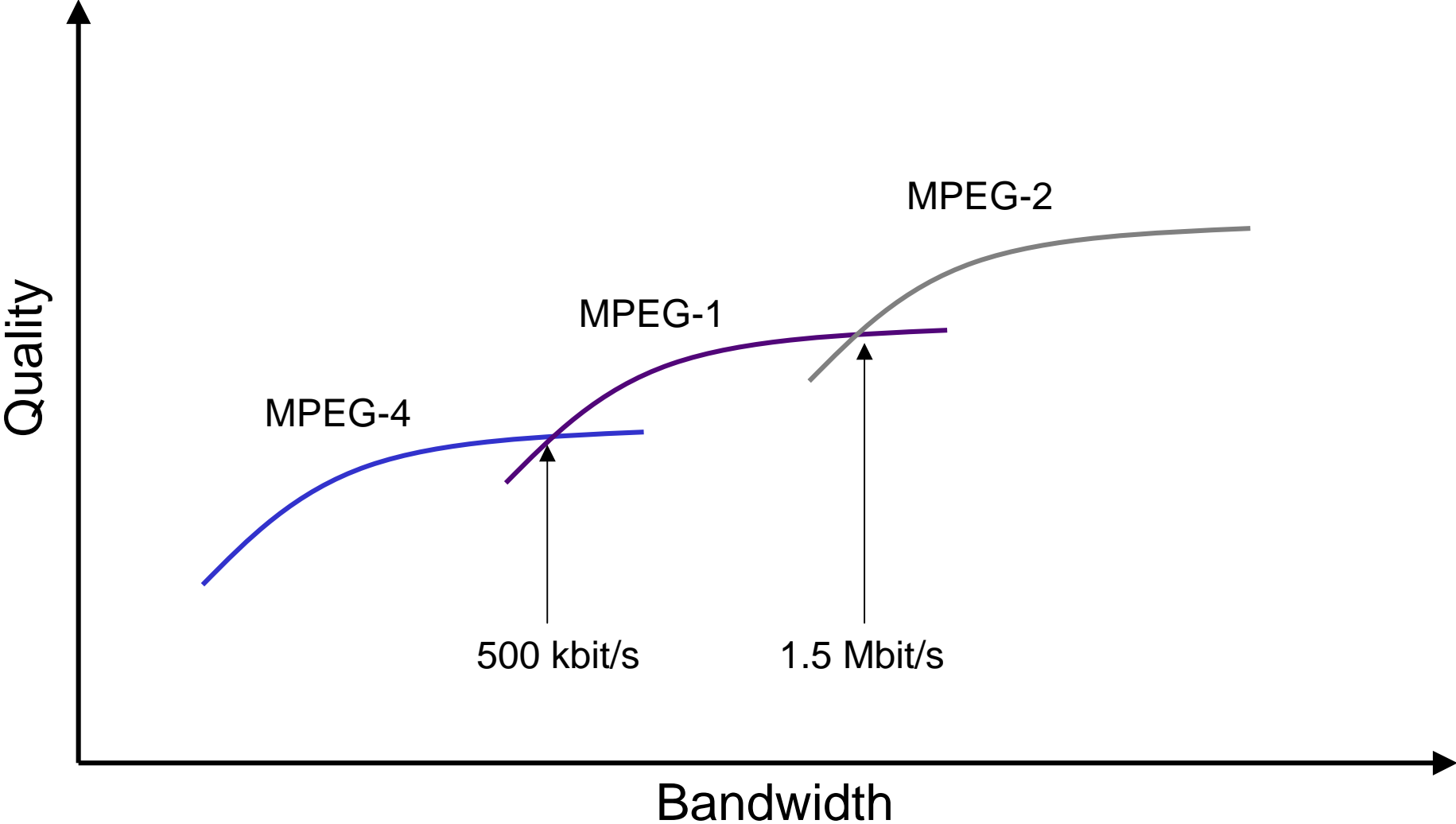
## Resolution vs. Bandwidth



# Digitisation of Media

## Typical usage of MPEG-1/2/4

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# Digitisation of Media

## Video Description – MPEG-7

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- MPEG-7 – “Multimedia Content Description Interface”
- MPEG-7 focuses on representing information about the content, not the content itself
- Will contain information to facilitate indexing, searching and retrieval of video and audio clips
- Scheduled to be finished July 2001

# Digitisation of Media

## Video compression - Demo

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- QCIF @ 8 fps – 20 kbit/s Apple QuickTime
- CIF @ 12 fps – 100 kbit/s Microsoft
- CIF @ 24 fps – 500 kbit/s RealNetworks
- CIF @ 24 fps – 2 Mbit/s MPEG-1

# Digitisation of Media

## Video compression - OpenDivX

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- Open-source codec based upon MPEG-4
- Widespread use for encoding of movies, similar to usage of MP3 for music
- Initial codec was called DivX, 'ripped' from Windows Media MPEG-4

### Demonstration

- CIF @ 24 fps – 500 kbit/s
- 544x432 @ 25 fps – 875 kbit/s video, 128 kbit/s audio

# Agenda

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- Introduction
- Digitisation of media content
- IP infrastructure for streaming media



# IP infrastructure for streaming media Access – Speed Comparison

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<b>3:00 MP3 Song sampled at 128 kbit/s</b>	
<b>Dialup</b>	7 minutes
<b>DSL Cable Modem</b>	36 seconds

<b>Audio Quality</b>	
<b>Dialup</b>	22 kHz stereo, or 44.1 kHz mono
<b>DSL Cable Modem</b>	44.1 kHz stereo (At least)

<b>Audio / Video Quality</b>	
<b>Dialup</b>	8 kHz mono / 160 x 120 @ 7.5 fps
<b>DSL Cable Modem</b>	44.1 kHz stereo / 320 x 240 @ 30 fps

# IP infrastructure for streaming media

## QoS

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- RTP flows are inherently intolerant to packet-loss
- Some streaming media packages retransmit lost packets in the application layer, but not all
- Various Quality-of-Service mechanisms used for ensuring end-to-end delivery:
  - **Diffserv** - Marking of IP packets to classify the treatment received by the IP network
  - **Queuing** - The algorithms employed by routers and switches to prioritize packets
  - **Shaping / Policing** - Employed on routers together with queuing

# IP infrastructure for streaming media

## QoS – Definition of Terms

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- **Flow:** A stream or set of packets on a media that can be considered as an aggregate
- Classification of packets as belonging to a flow based on certain parameters:
  - Same source or destination IP address / network
  - Same user generating them
  - Same protocol
  - Same source or destination port
  - Same URL

# IP infrastructure for streaming media

## QoS – Definition of Terms

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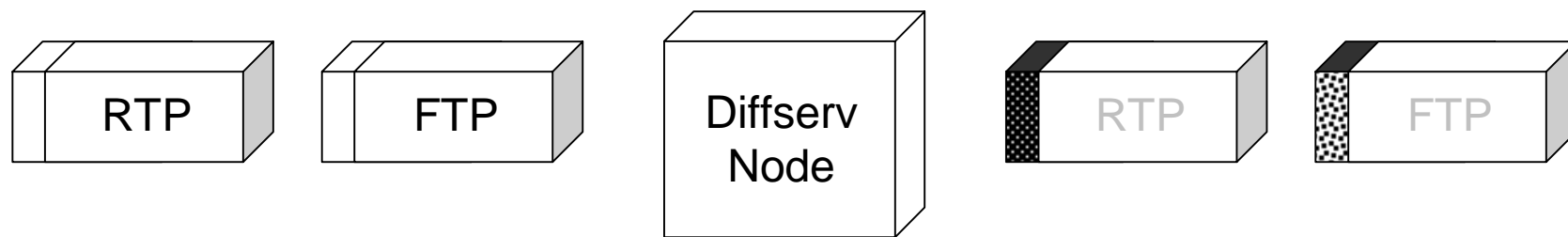
- **QoS:** A network service that specifies the performance of traffic flows through one or more networks
  - Includes technologies such as Queuing and DiffServ
  - Can be measured by availability, delay, jitter, throughput, packet loss rates etc.
  - The technology focus is through router or switch behavior, but the customer focus is through services delivered (Black-box)
- **Policy:** A network service that specifies where QoS should be applied, and if flows should be allowed
  - Binds network strategies to QoS mechanisms

# IP infrastructure for streaming media

## Diffserv – Classification

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- IETF standard - Multiple RFCs
- Packets get classified and marked in the Diffserv Codepoint (DSCP) field in the IP header
- Diffserv-aware Routers and Switches will give packets certain treatment based upon marking
  - All packets marked the same way - Treated the same way



Unmarked packets  
arrive at node

Node classifies packets  
based upon policies

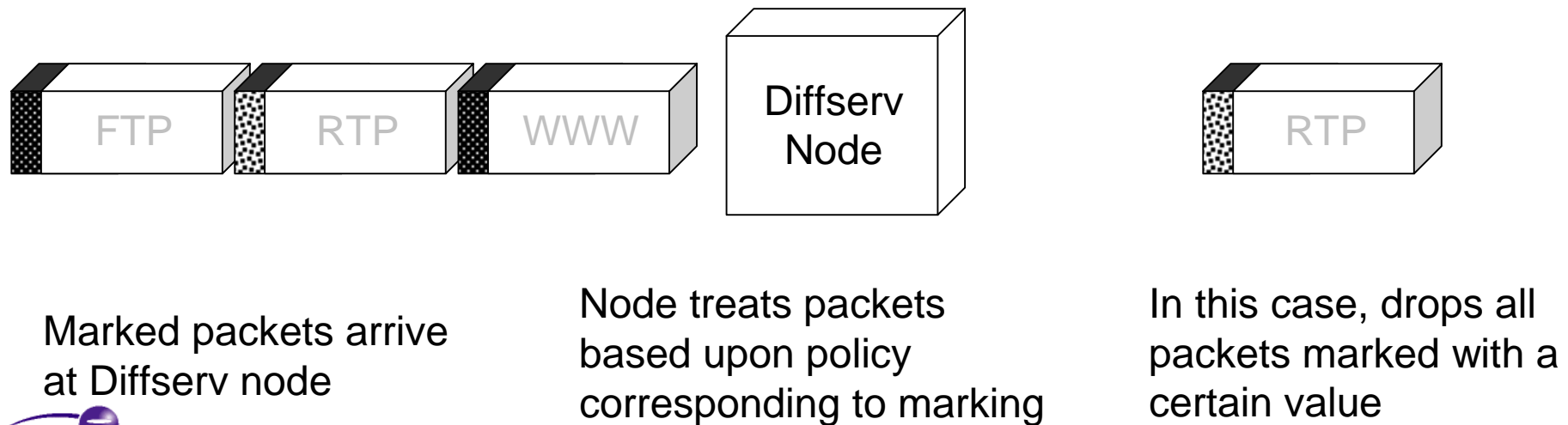
Packets marked in DSCP  
Field, no further classification  
is necessary

# IP infrastructure for streaming media

## Diffserv – Packet Treatment

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- Packet treated same way as others marked the same
  - Core devices know only markings, not flows
    - No per-flow knowledge required in the core
    - Per-flow knowledge kept at the network edge



# IP infrastructure for streaming media

## Diffserv – Policy Manager & Domains

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- **Policy Manager**

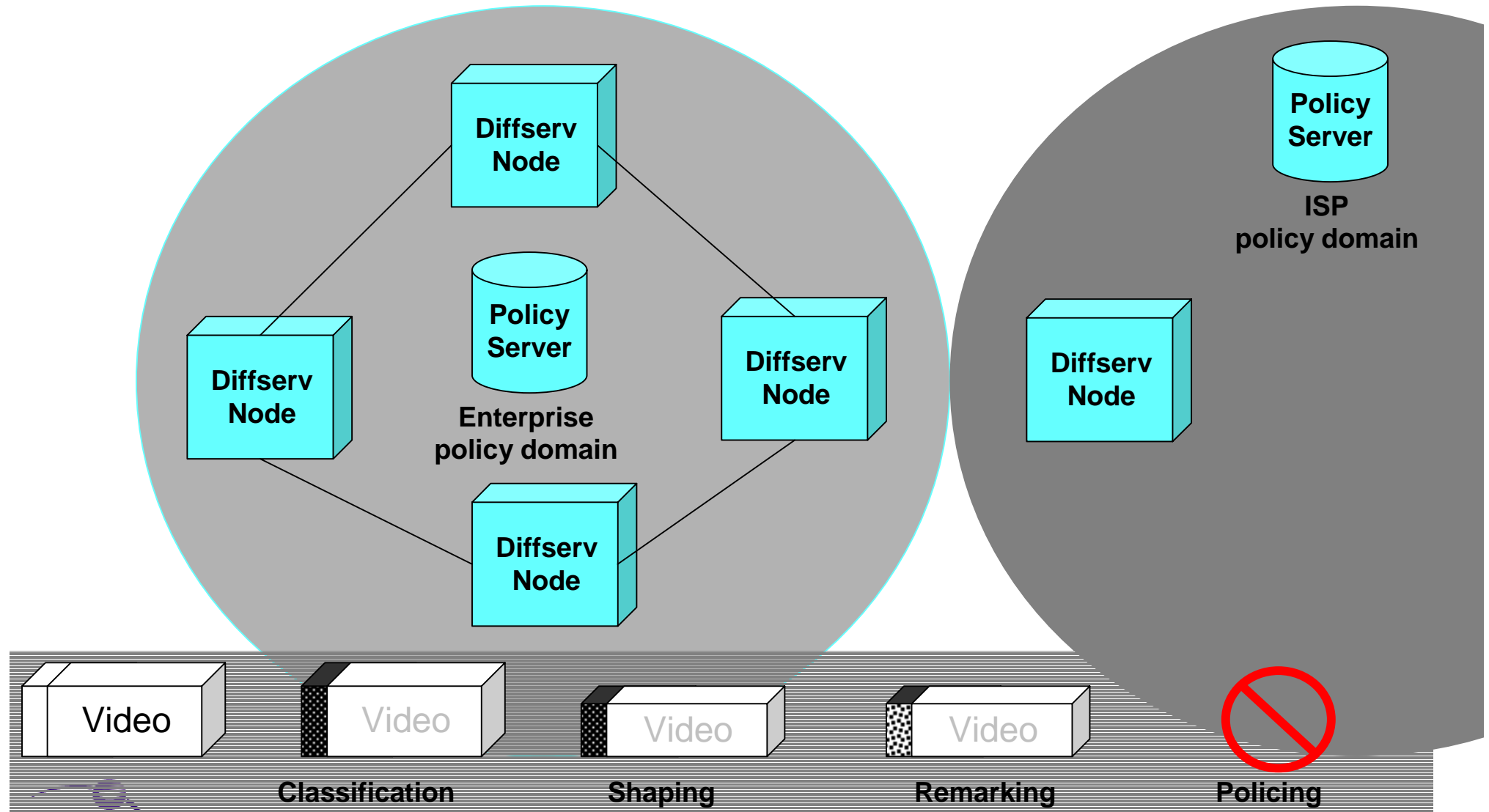
- Who/what gets access to a service (authenticate and authorize)
- May be static; e.g., set up by a network administrator
- May be dynamic; e.g., receive requests and configure Diffserv nodes accordingly

- **Domain**

- A contiguous network over which a consistent set of Diffserv policies could be administered - one Policy Manager per domain
- Domain examples would be an enterprise's network or an ISP's network

# IP infrastructure for streaming media

## Diffserv – Policy Manager & Domains





# IP infrastructure for streaming media

## Queuing and Dropping

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- **Queuing** – Mechanisms that queue packets for outbound transmission based upon their priority
- **Dropping** – Mechanisms that drop certain packets based upon their priority before congestion occurs
- These mechanisms run within a switch or a router, and can use DSCP marking to determine a packet's priority

# IP infrastructure for streaming media

## Queuing and Dropping

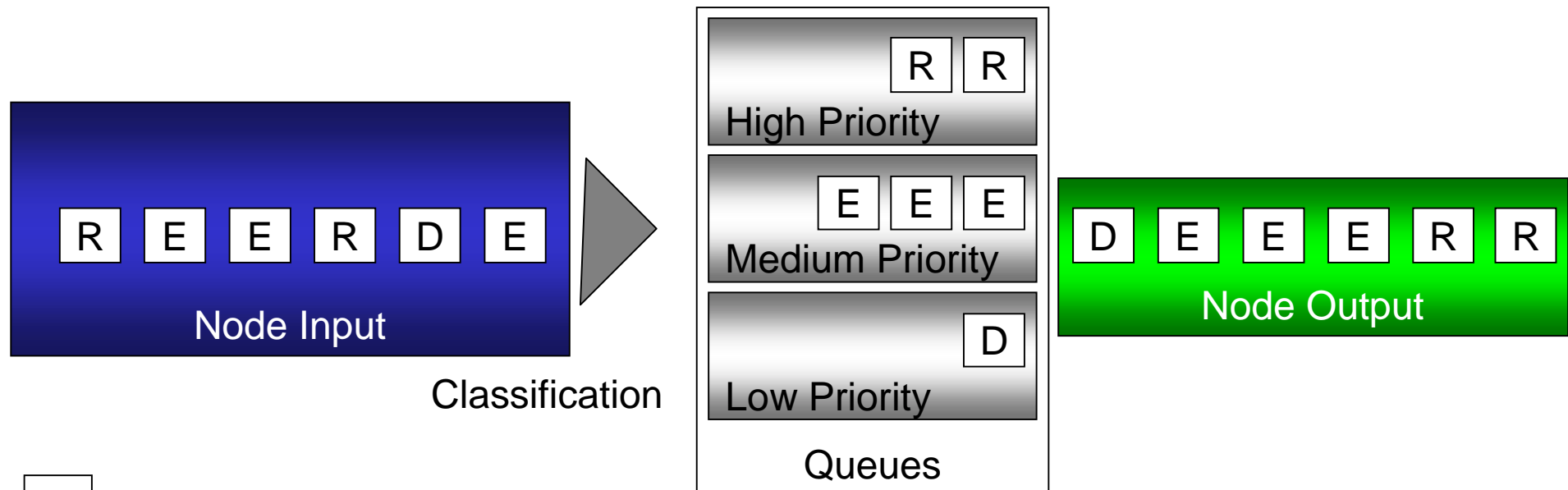
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- **Priority Queuing** – A prioritization scheme where a packet with a higher priority gets strict precedence in front of a lower prioritized packet
- **Class Based Queuing** – A prioritization scheme where a packet's chance of getting dequeued depends on its classification

# IP infrastructure for streaming media

## Priority Queuing

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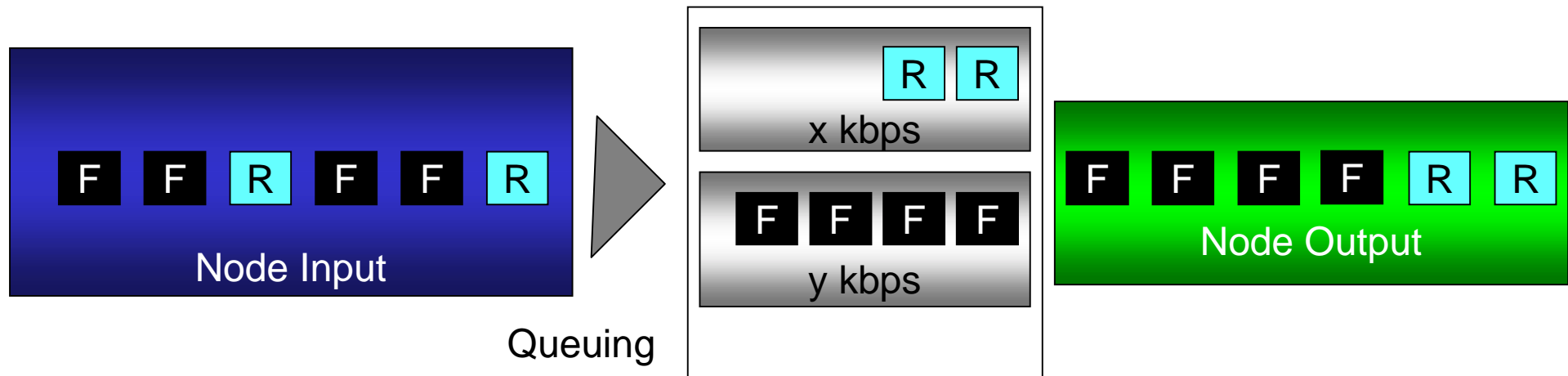
- E Email
- R RTP – Video
- D Other Data

- Higher priority traffic gets absolute priority. This can “freeze” out other traffic

# IP infrastructure for streaming media

## Class Based Queuing

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**F** FTP

**R** RTP – Video

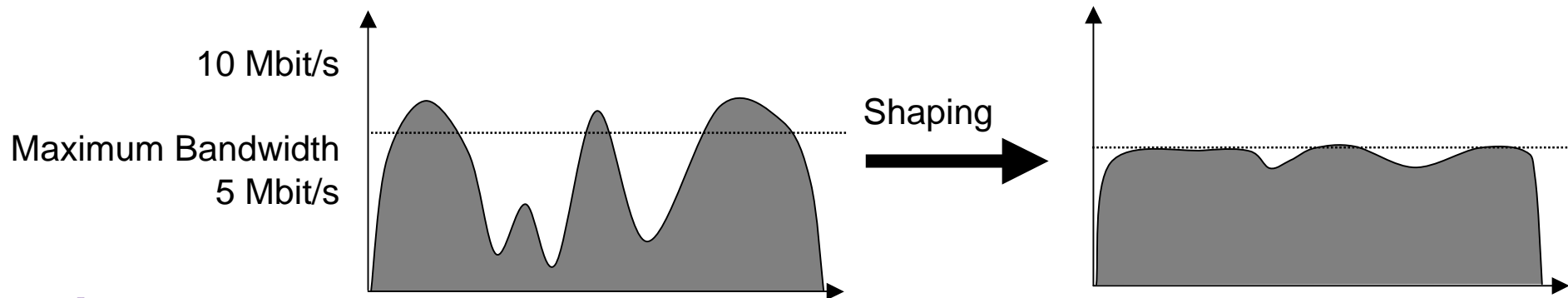
- Each class is guaranteed a minimum BW during congestion, irregardless of other flows

# IP infrastructure for streaming media

## Shaping

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- Shaping forces traffic to conform to a certain traffic pattern
- Essentially smoothes out bursts by buffering until egress capacity available
- Shaping is typically used where traffic is transmitted to another network that enforces a maximum bitrate (e.g. Frame Relay CIR)

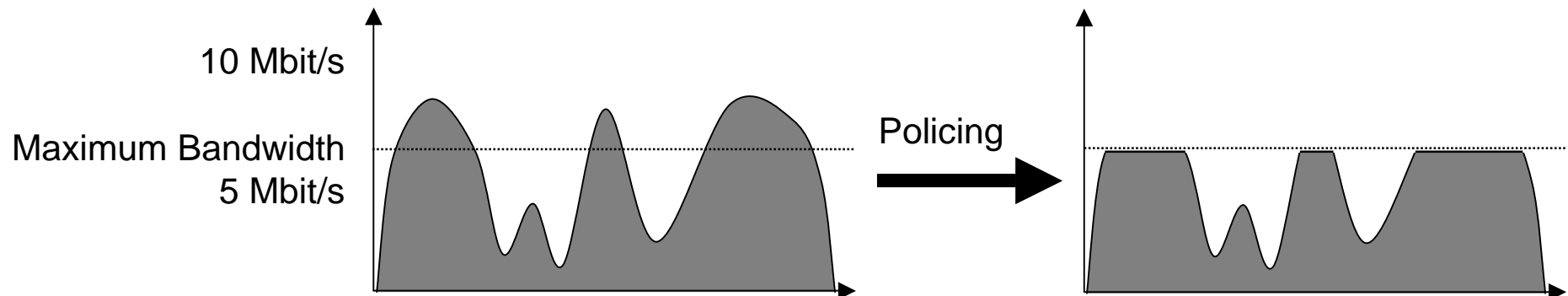


# IP infrastructure for streaming media

## Policing

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- Policing forces traffic to conform to an absolute bitrate
- Any traffic above bitrate gets dropped
- Policing is typically used where traffic is coming into a network above an agreed upon maximum bitrate



# IP infrastructure for streaming media

## QoS Conclusion

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- RTP streams can be intolerant to packet-loss
- To ensure consistent delivery various QoS mechanisms should be used:
  - Marking to ensure that packets get prioritized treatment
  - CBQ to deliver priority-tagged packets during congestion
  - Understand shaping & policing that happens with flows to conform to SLAs

# IP infrastructure for streaming media

## Multicast

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- In Multicast, a sender transmits one copy of a stream to an IP Multicast group
- The network ensures that the stream is distributed to all the end-stations that wish to receive it
  - Via replication in Multicast-enabled nodes
  - Natively via transmission media (satellite etc.)
- Built for applications where multiple users need to receive same data at same time
  - Broadcasting
  - Conferences
  - Real-time data feed, e.g. market data

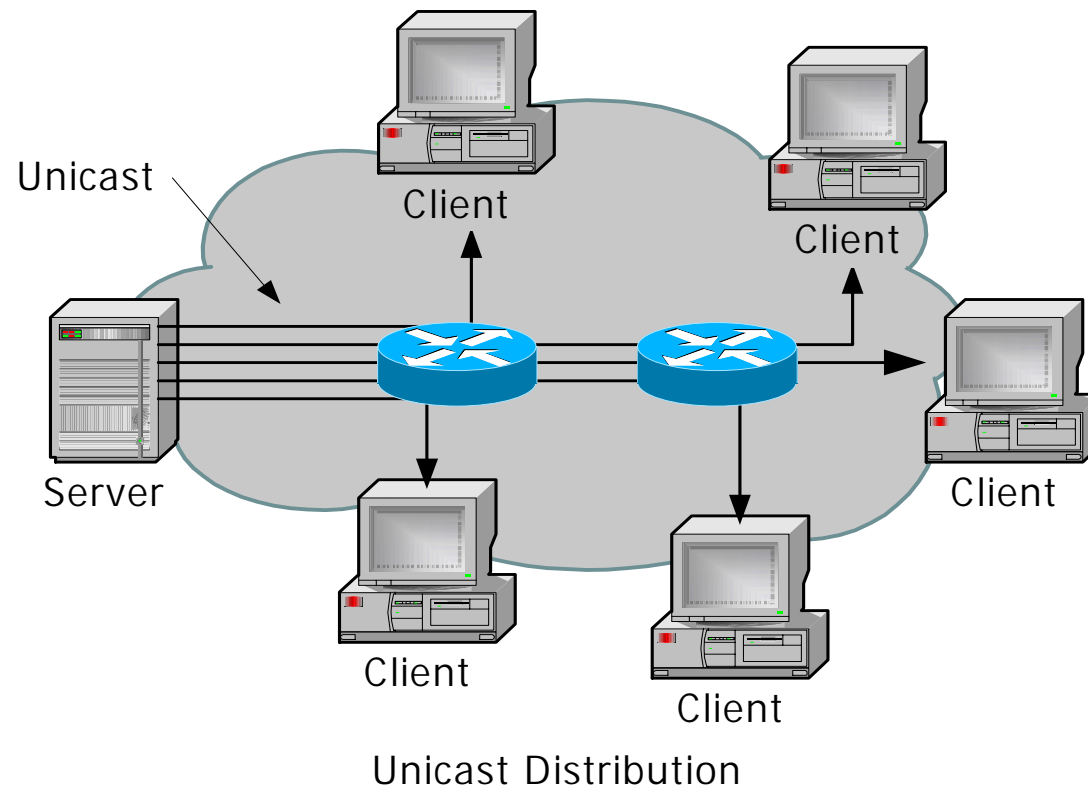


# IP infrastructure for streaming media

## Unicast streaming

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- Media Server must transmit one unique stream to each Client

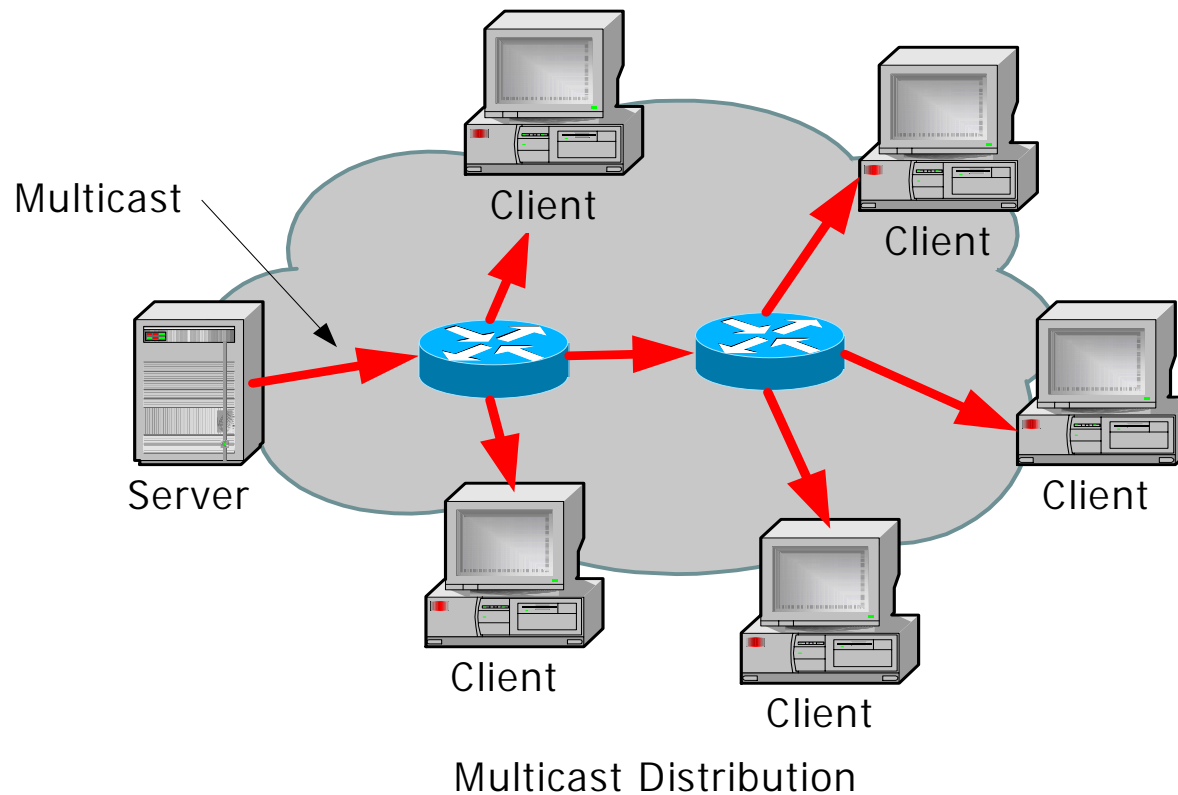


# IP infrastructure for streaming media

## Multicast streaming

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- Media server transmits one stream to Multicast-enabled network

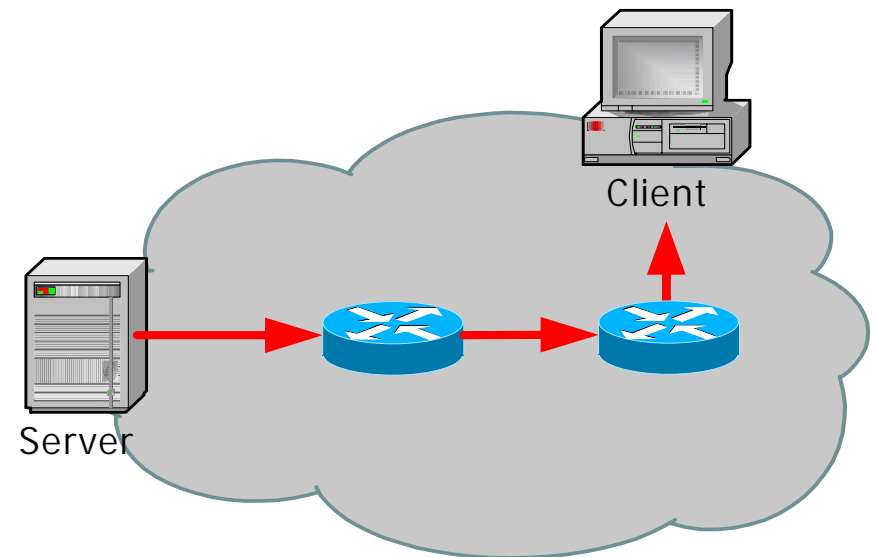
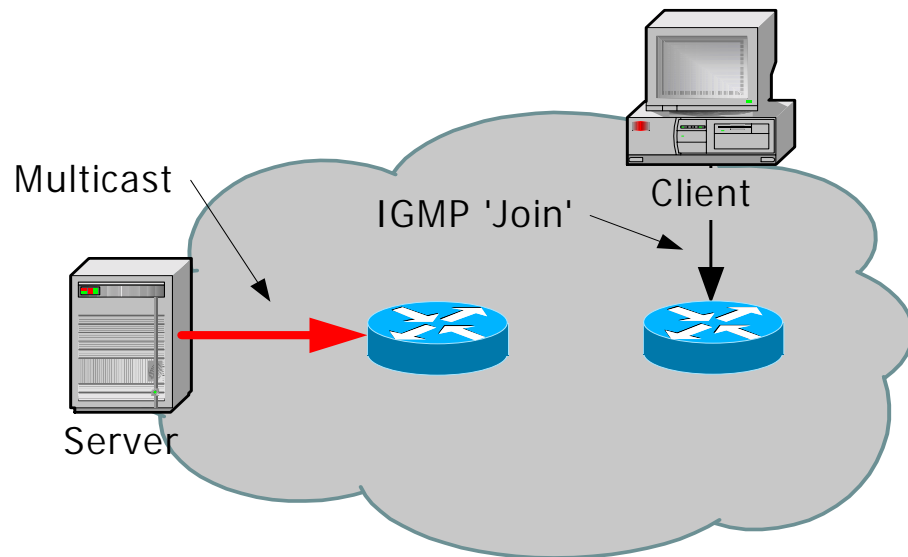


# IP infrastructure for streaming media

## Multicast Client signaling

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- **IGMP** - Internet Group Management Protocol
  - Protocol for joining and leaving a Multicast group
  - Can perform 'Join', 'Leave'. Routers send periodic keep-alives
  - Must be supported in IP stacks on client, switches and routers



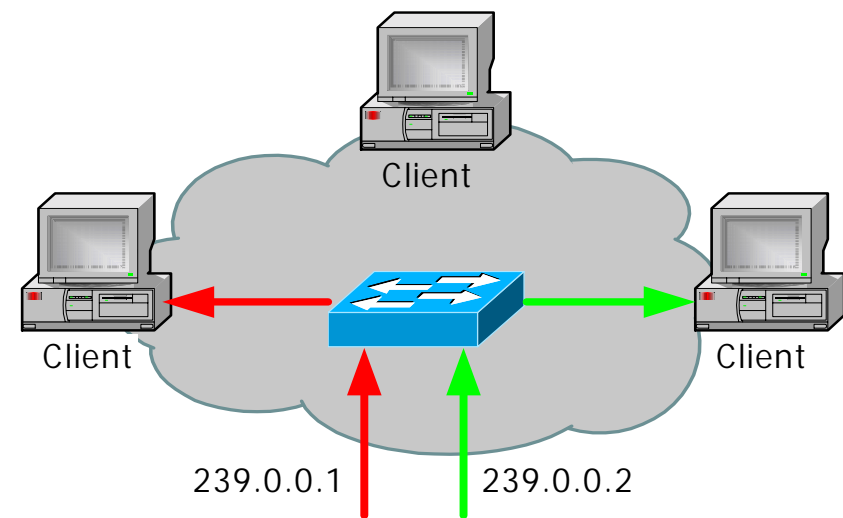
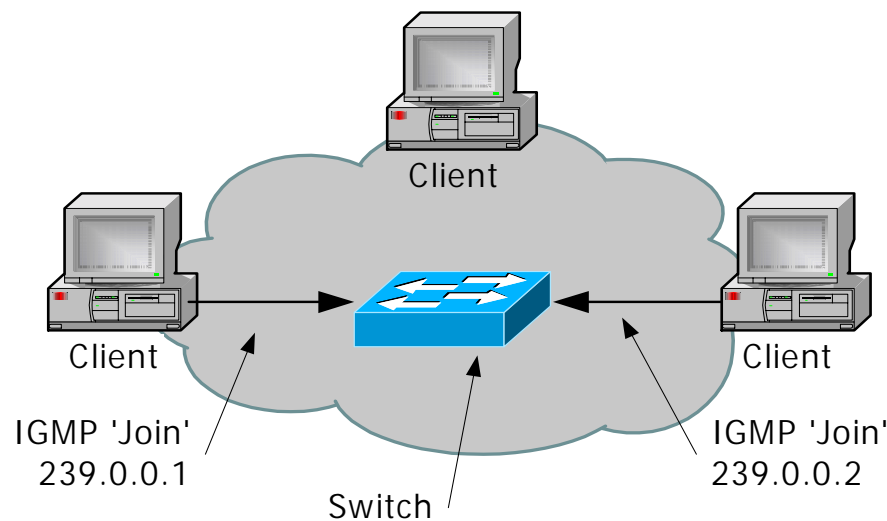
# IP infrastructure for streaming media

## Multicast Client signaling

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- **IGMP Snooping**

- Allows switches to 'snoop' IGMP queries from clients to routers to keep state
- Ensures that switch only replicates Multicast packets to interfaces with clients that joined that Multicast group
- If switch doesn't support IGMP snooping, will then have to flood each interface with packets



# IP infrastructure for streaming media

## Multicast Tree

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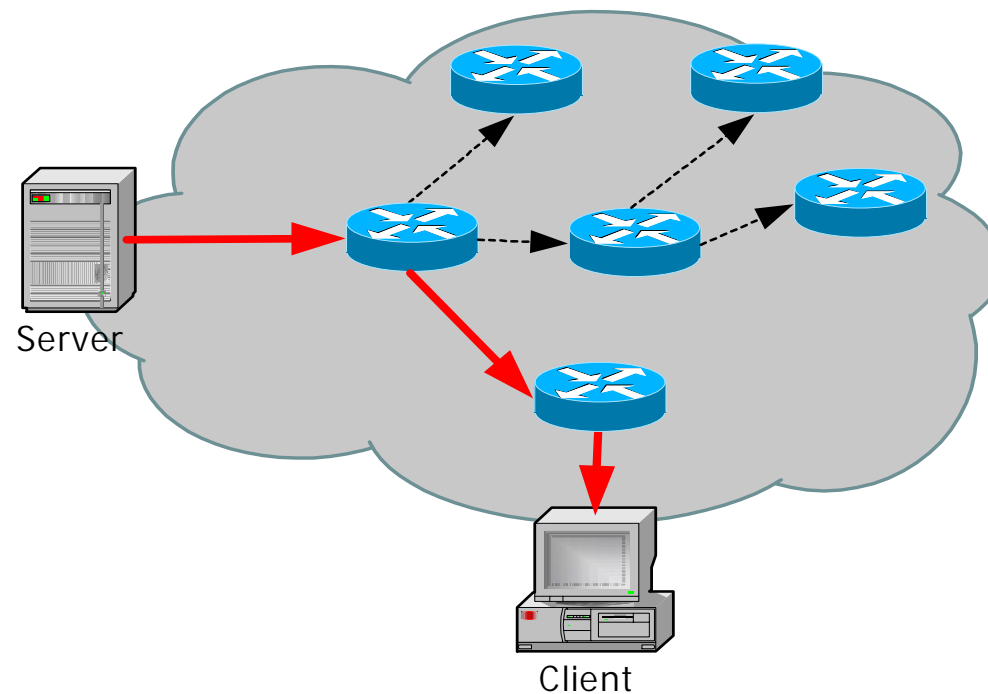
- Within a Multicast network, it isn't necessary to send the multicast distribution to all routers
- Only routers with end-stations that are joined to a group need be part of the distribution tree
- Various Multicast routing protocols exist that facilitate this to different degrees
  - PIM Dense Mode
  - PIM Sparse Mode
  - DVMRP
  - MOSPF

# IP infrastructure for streaming media

## Multicast PIM Dense Mode

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- Multicast stream is flooded to all routers
- If no listeners on router, router will prune
- Process repeats itself every three minutes

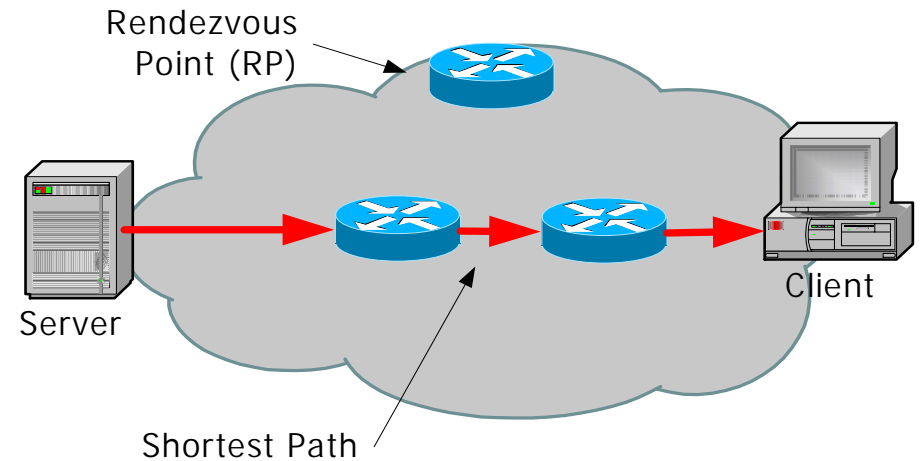
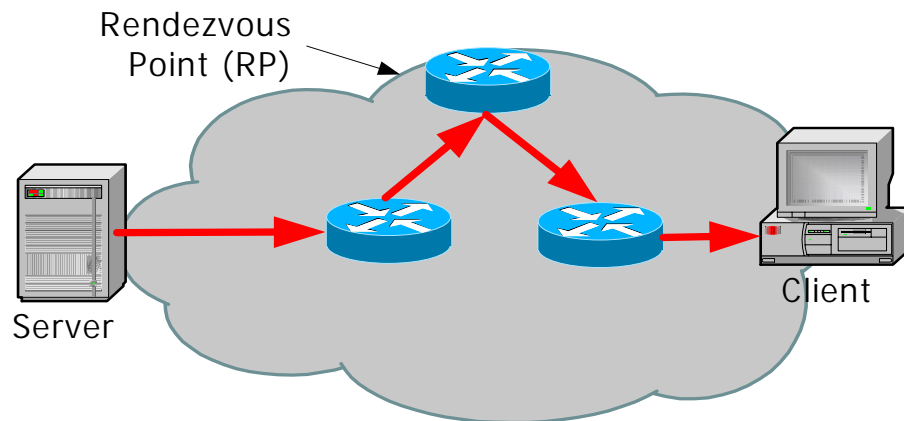


# IP infrastructure for streaming media

## Multicast PIM Sparse Mode

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- Multicast stream is only replicated to “listening” routers
- Has core device called Rendezvous Point (RP) that listeners and senders join initially
- Routers can then opt to create shortest path tree



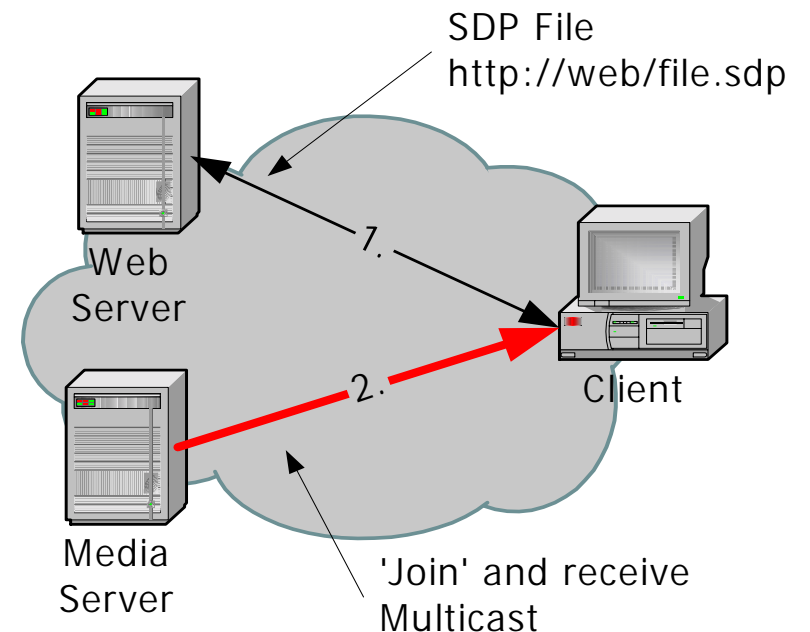
# IP infrastructure for streaming media Session Description Protocol (SDP)

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- SDP is a protocol that describes multimedia sessions.
- Used with streaming and Multicast as a mechanism to describe a media-stream
- Text file that includes parameters such as:
  - Multicast group and port
  - Codec information
  - Payload information (Audio / Video etc.)

1 – Client fetches and reads SDP file from web-server

2 – Client uses information in SDP file to attach to appropriate Multicast group and use correct Codec etc.





# IP infrastructure for streaming media

## Source Specific Multicast (SSM)

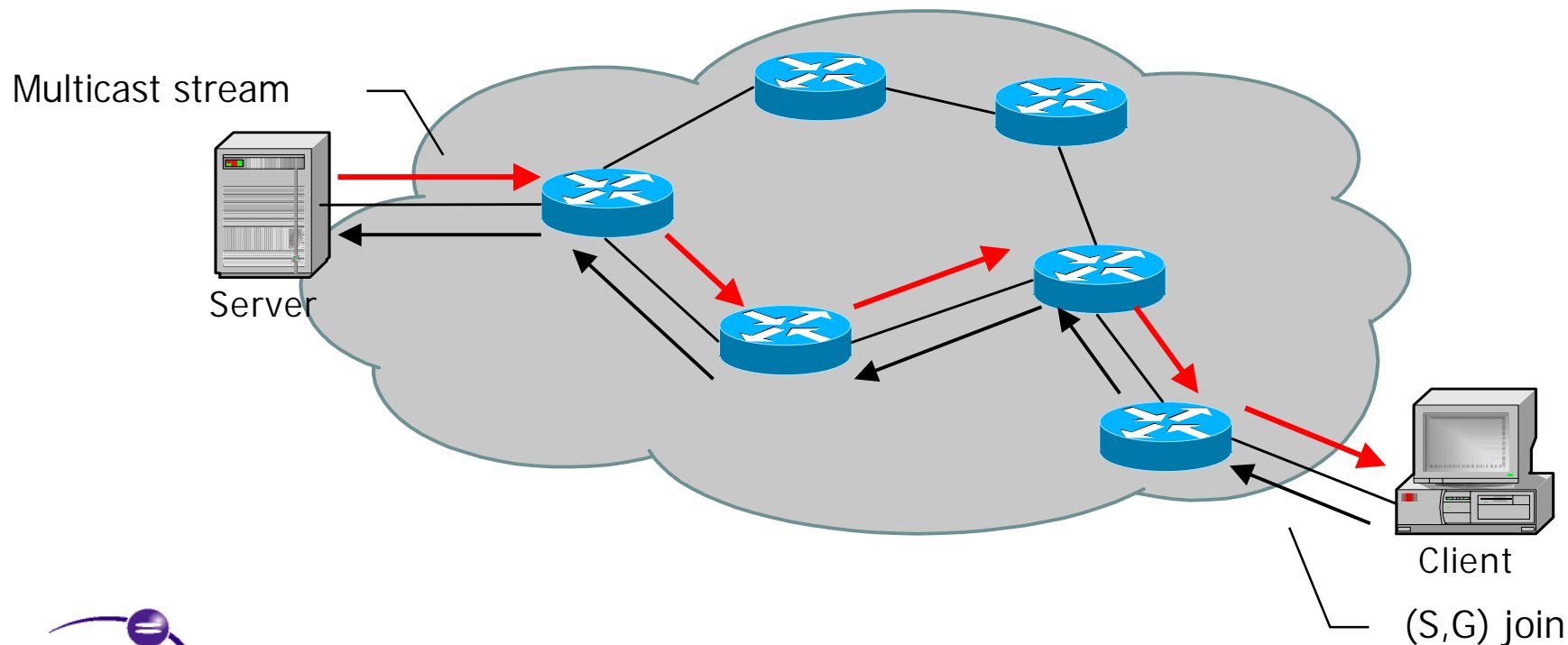
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- SSM is attempting to simplify Multicast deployment by taking into consideration practical realities:
  - Streaming (and many other applications) only have one source
  - Source is usually well known, and can be distributed via out-of-band mechanism (SDP & HTTP etc.)
  - Problems with RP:
    - Complicated
    - Security issues, “anyone” can become sender
    - No guarantee of shortest path

# IP infrastructure for streaming media Source Specific Multicast (SSM)

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- SSM is built upon PIM-SM
- IGMP v3 must be deployed for SSM
- Always uses SPT



# IP infrastructure for streaming media

## Pragmatic General Multicast (PGM)

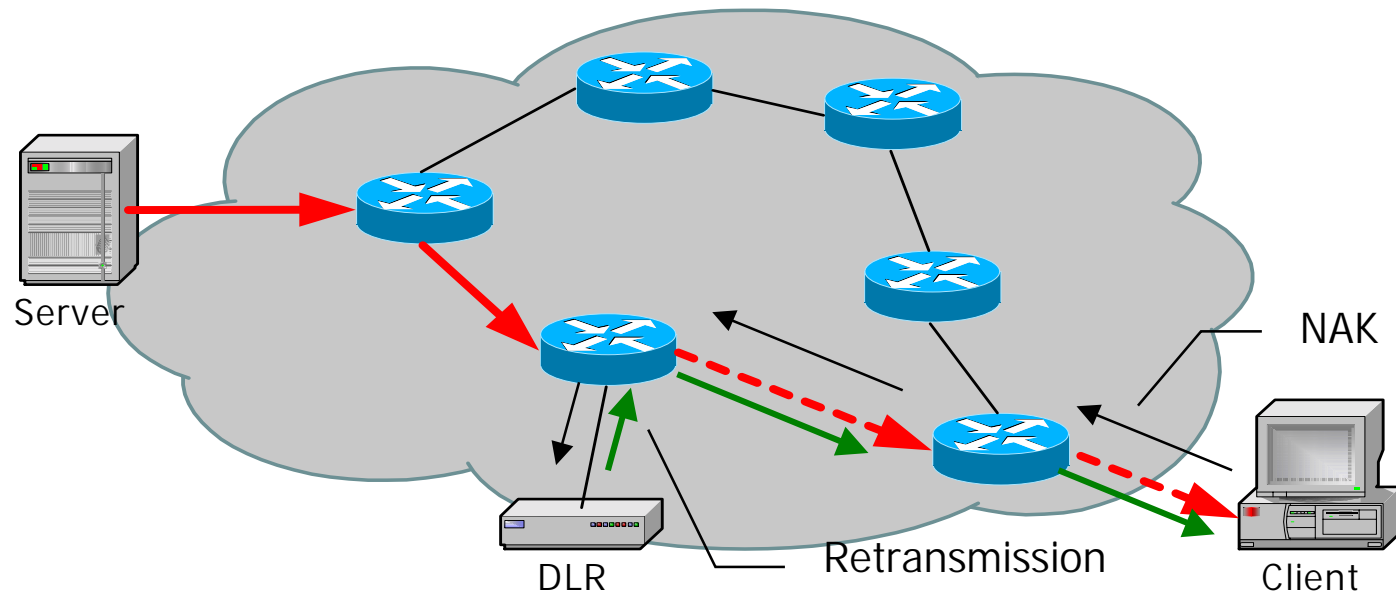
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- PGM is a reliable Multicast protocol
- Is designed for scenarios where absolute reliability, acknowledgment or packet order is not required
- Thus pragmatic...
- Uses NAKs (Negative Acknowledgments) to request retransmission
- Network elements will assist in sending NAKs up the source-tree

# IP infrastructure for streaming media Pragmatic General Multicast (PGM)

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- Retransmission of missing packets happens either by source, or by Designated Local Repairers (DLRs) that will resend missing packet to requestor
- Can also send FEC information pro-actively or on-demand



# IP infrastructure for streaming media

## Multicast Conclusion

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- Use Multicast when doing live streaming
- Sparse mode protocols such as PIM-SM are preferable, much less impact on infrastructure
- Satellite distribution is inherently multicasting
- Difficult to set up outside own network, ISP or end-user might not be multicast enabled
- SDP or similar mechanism should be used to distribute stream information in a Multicast environment
- SSM should ease deployment
- PGM should increase reliability

# Agenda

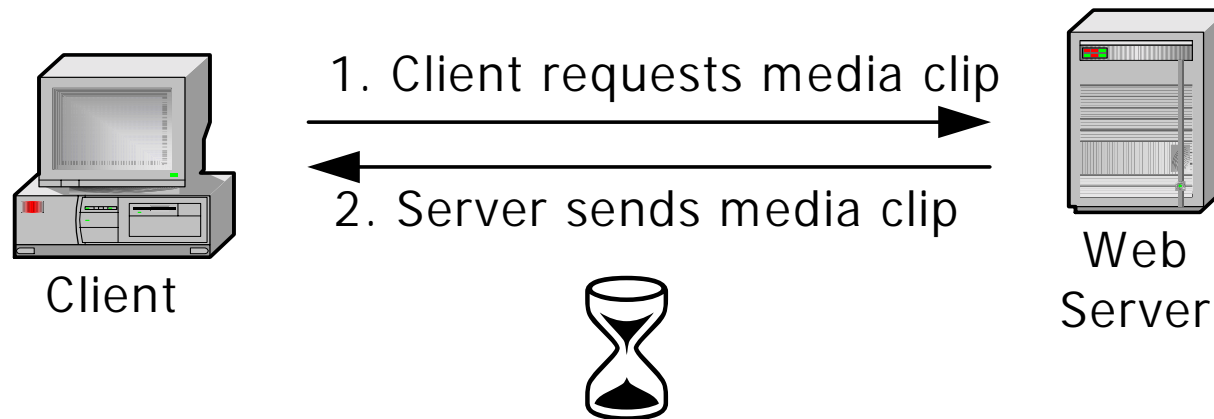
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- Introduction
- Digitisation of media content
- IP infrastructure for streaming media
- Delivery Technologies

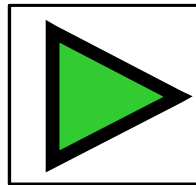
# Media Delivery Downloading

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- Media is downloaded from webserver to client and stored locally



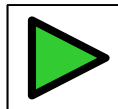
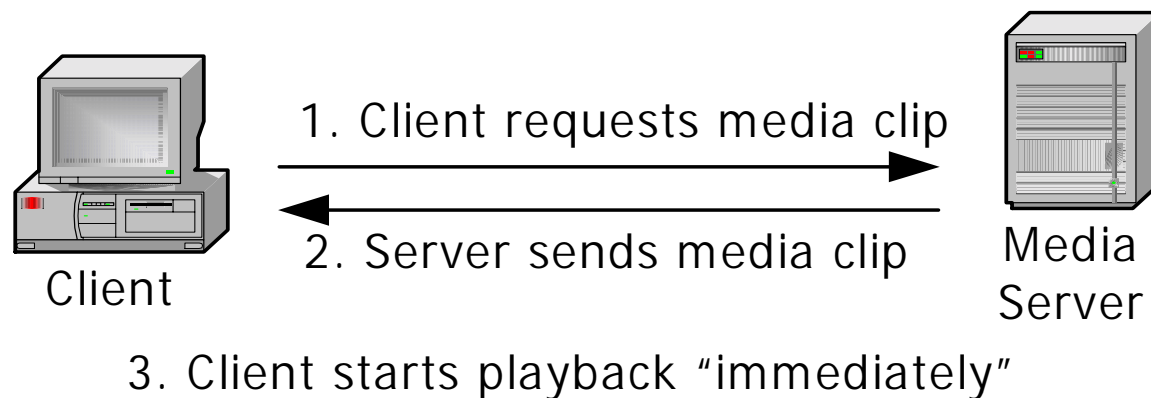
3. Once downloaded, Client can play clip



# Media Delivery Streaming

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- Once media clip is requested, server starts delivering clip to client using “streaming”
- Streaming is when the client starts playback of received data immediately, and doesn't store more than small buffer locally (Typically ~5 seconds worth)
- Uses specialized server, usually not webserver





# Media Delivery

## Downloading vs. Streaming

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- Advantages of each method:

<b>Streaming</b>	<b>Downloading</b>
Faster play time	Low infrastructure impact
No local storage	Low server cost
Copyright protection	Higher quality clips
Supports live mode	Better for replaying

- Streaming is far more popular as it delivers immediate playback for end-user, despite the extra costs and complexity

# Media Delivery

## Streaming protocols

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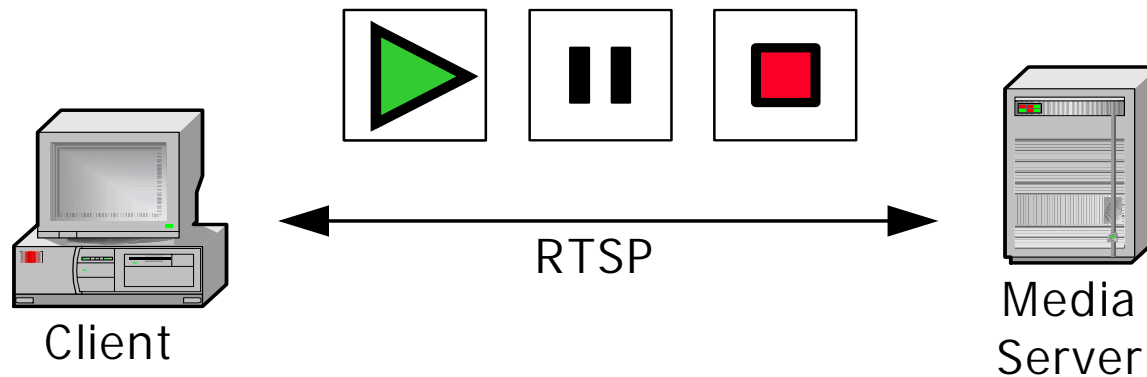
- So many protocols, so little interoperability
  - RTSP Real Time Streaming Protocol (IETF)
  - RTP Real Time Protocol (IETF)
  - RTCP Real Time Control Protocol (IETF)
  - RDT RealNetworks proprietary streaming
  - PNA RealNetworks proprietary streaming control (old)
  - MMS Microsoft proprietary streaming control
  - HTTP Used for streaming some types of content

# Media Delivery

## RTSP

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- Defined by the IETF (RFC2326)
- Communication between streaming client and server, i.e. application layer protocol
- Starts and controls streams. Negotiates Codecs etc.
- Uses TCP socket 554

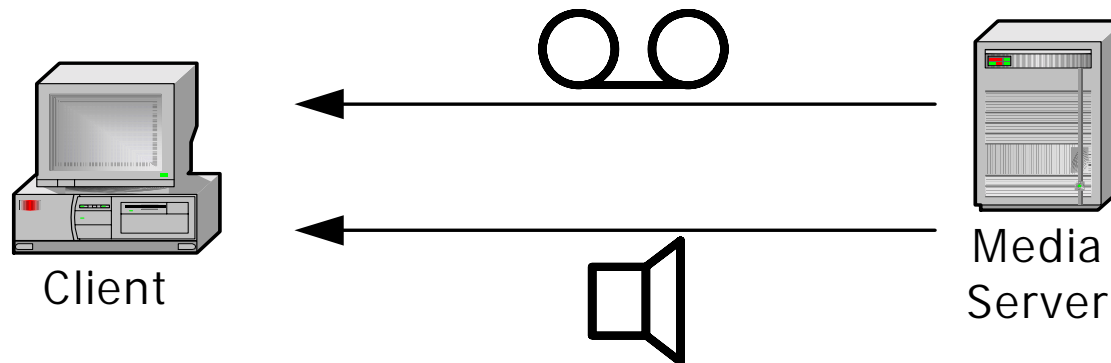


# Media Delivery

## RTP

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- Defined by the IETF (RFC1889)
- RFC1890 describes payload types
- Used for streaming audio or video over UDP
- Uses dynamic port range
- Provides timing reconstruction, loss detection and media identification
- Separate RTP "channel" for each media

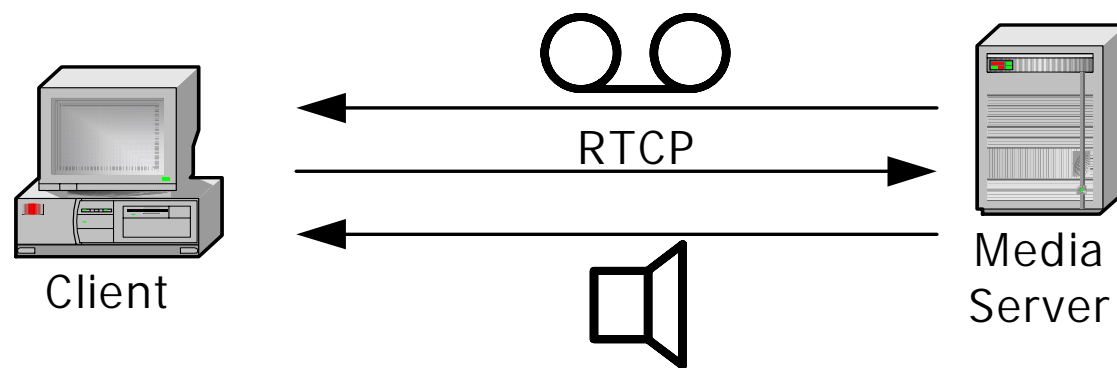


# Media Delivery

## RTCP

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- Also by the IETF, same RFC as RTP
- Control protocol for RTP, uses UDP
- Provides monitoring of quality
- Feedback from clients listening to streams, especially useful for Multicast

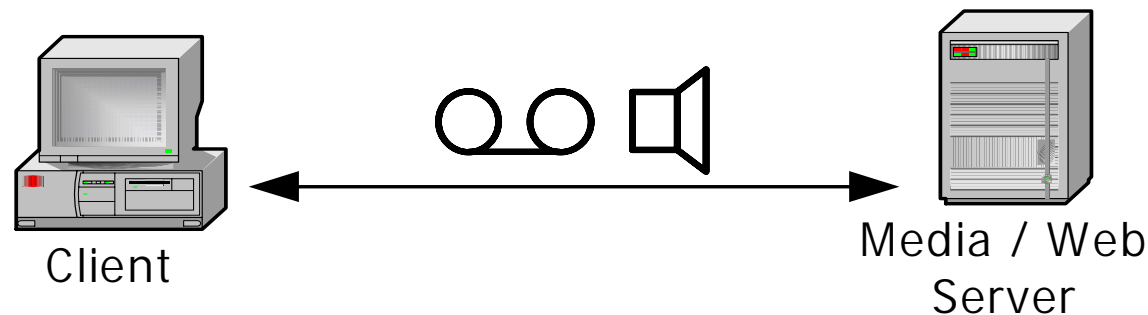


# Media Delivery

## HTTP

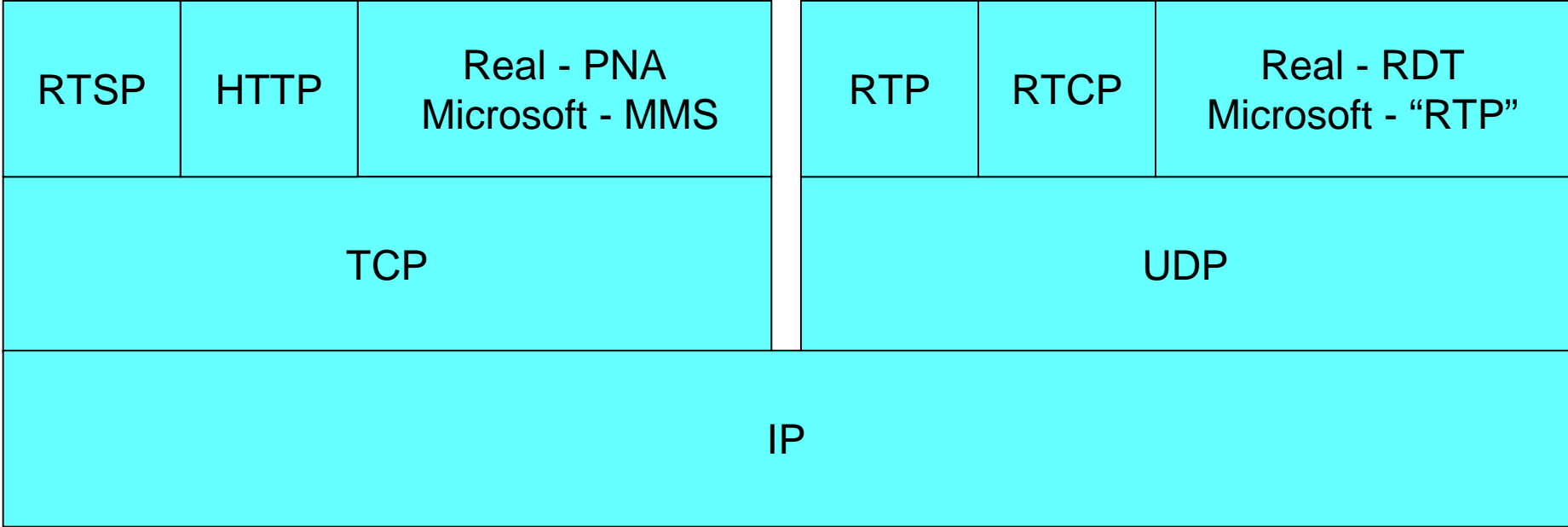
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- HTTP used for streaming in some cases
  - Where RTP doesn't describe how to stream media-type yet
  - To get through firewalls
- Not optimized for streaming, runs over TCP. Cannot support bitrate negotiation



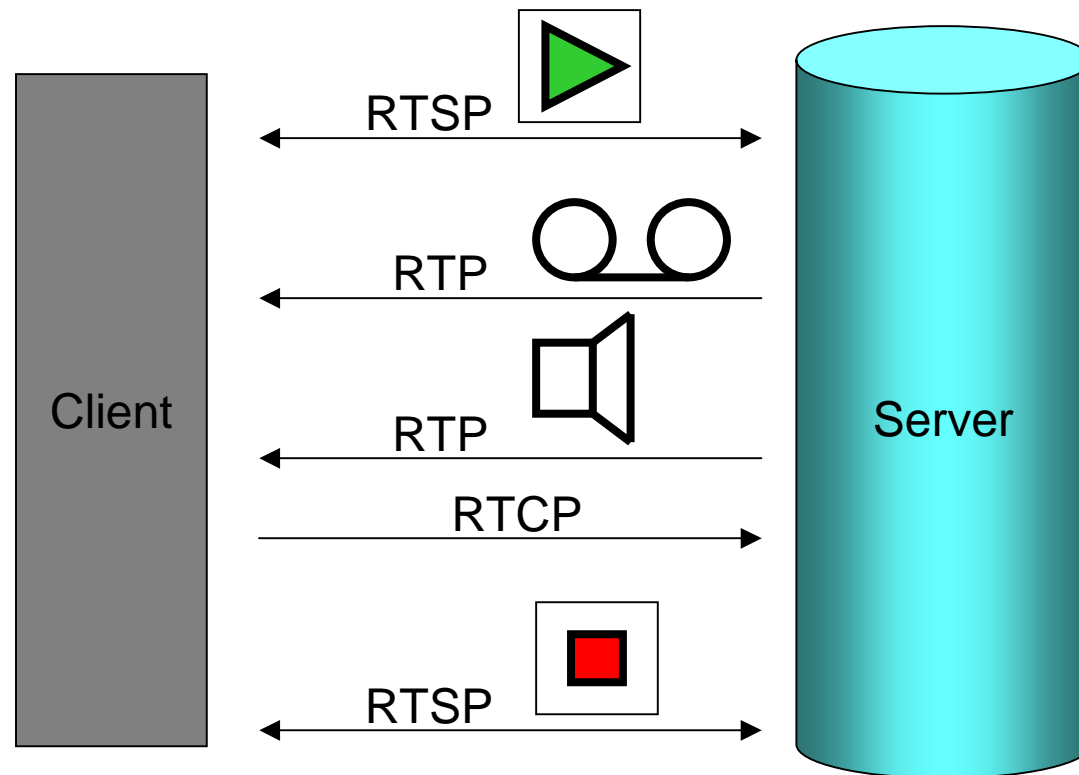
# Media Delivery Protocols – How do they play together

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# Media Delivery Protocols – How do they play together

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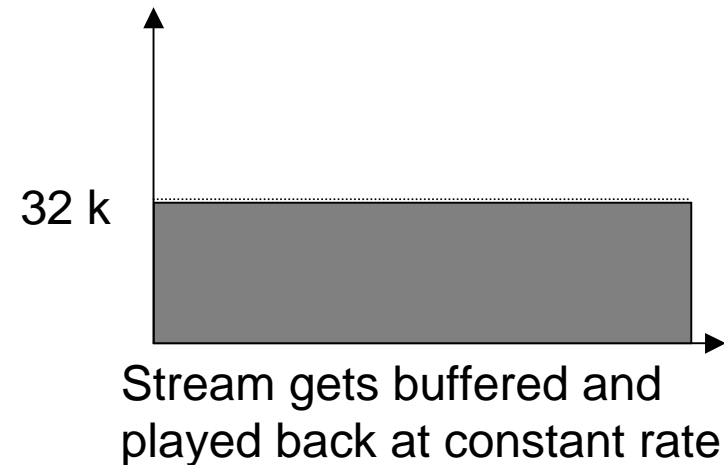
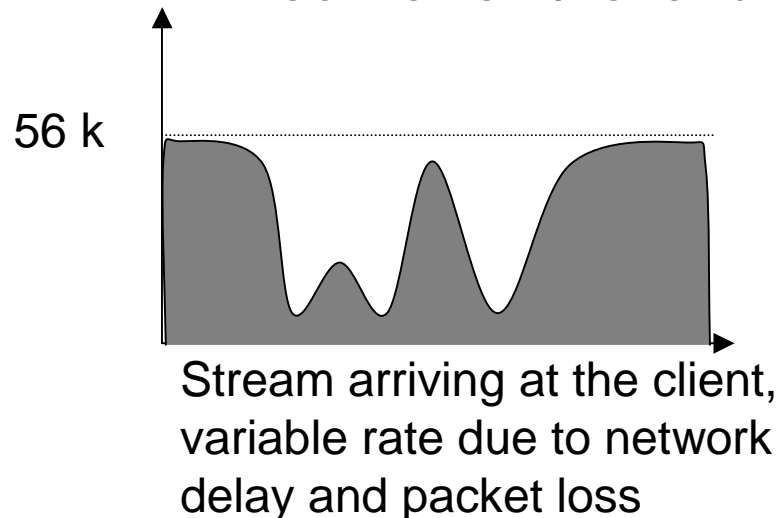


# Media Delivery

## Streaming – Buffering

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- Media client buffers certain amount:
  - To avoid jitter and to play back content in time sequence, using RTP timestamps
  - To compensate for packet loss in network between server and client

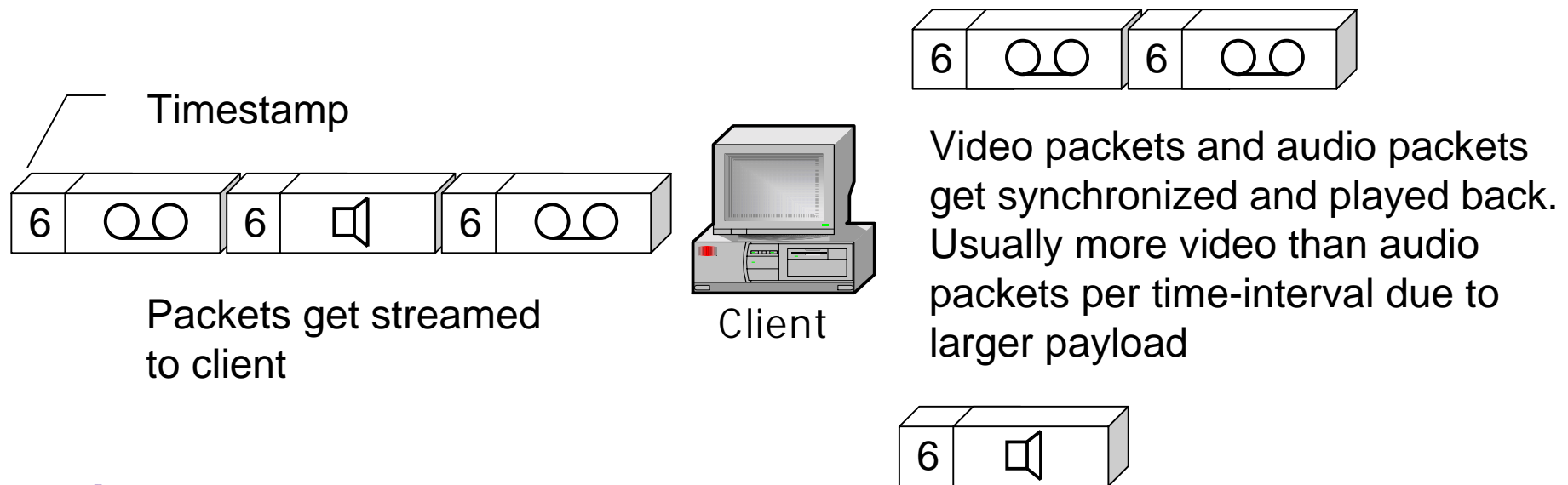


# Media Delivery

## Streaming – Synchronization

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- Audio and Video get transmitted over separate RTP streams
- Timestamp in RTP header used for inter-media synchronization



# Media Delivery

## TCP / UDP

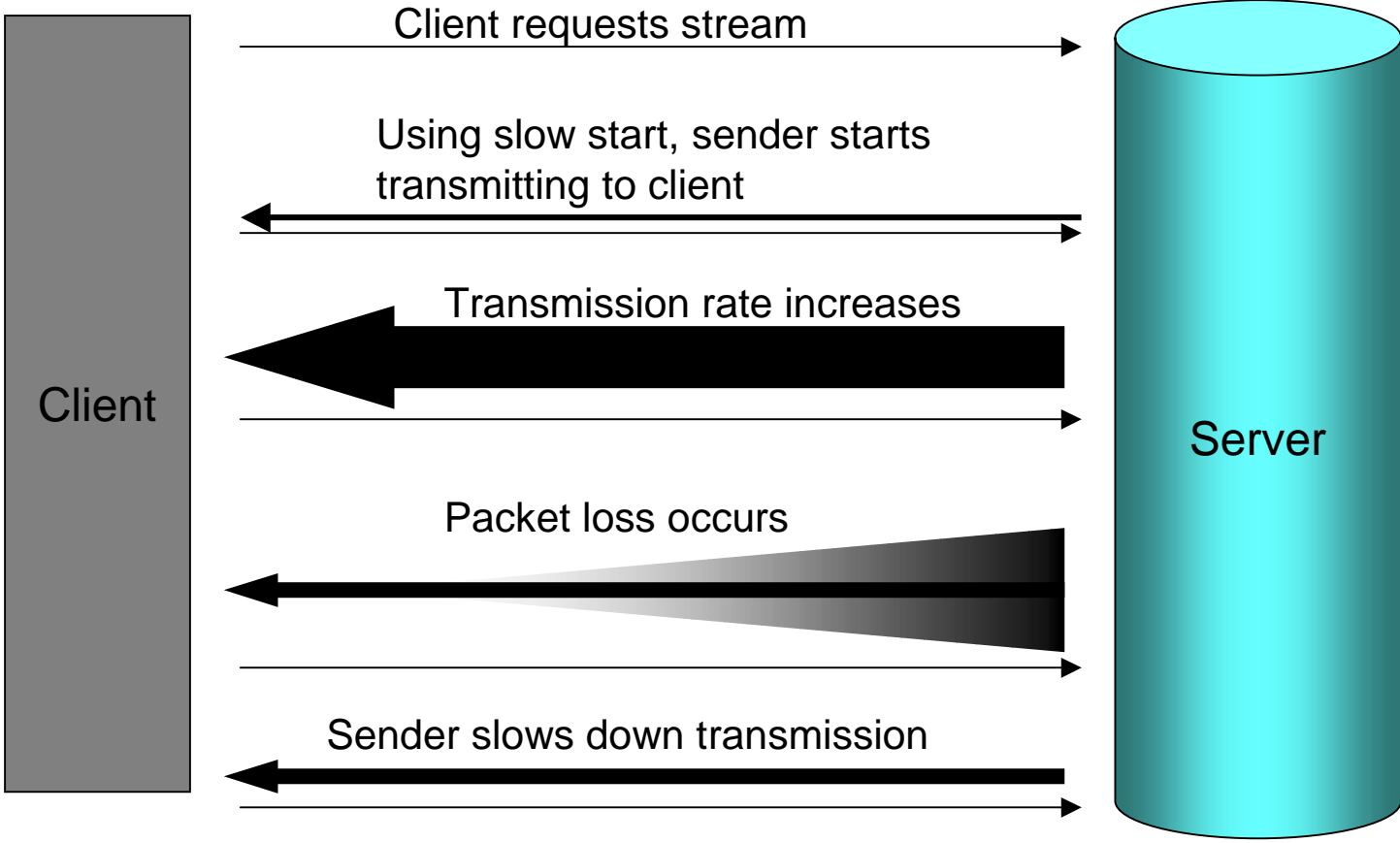
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- **TCP** - Maintains two-way connection between sender and receiver. Sender re-transmits packets if no acknowledgement is received from receiver
- **UDP** - Transmits packets at a set bitrate with assumption that they are received at receiver, no re-transmission

# Media Delivery

## TCP transmission

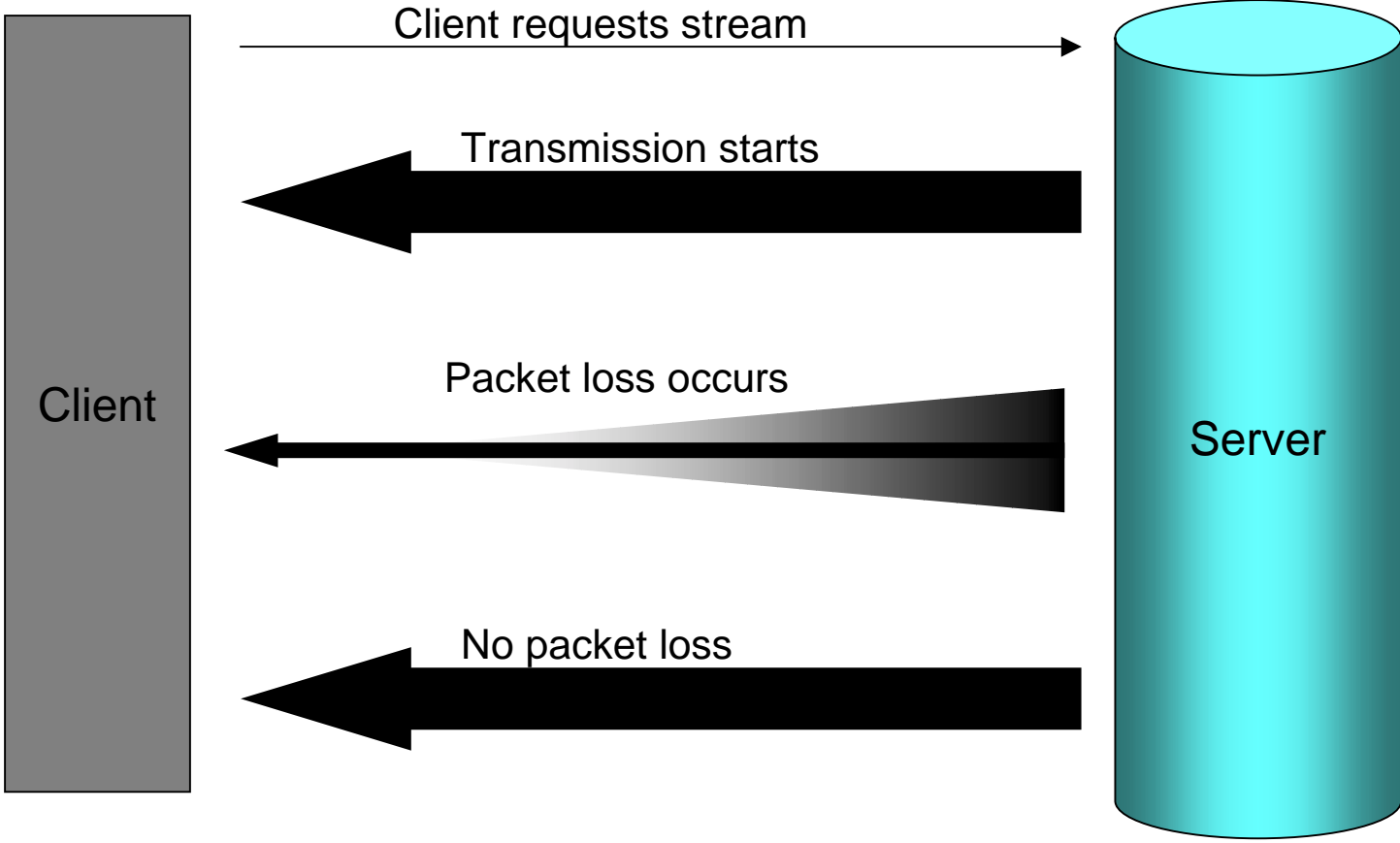
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# Media Delivery

## UDP transmission

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# Media Delivery

## UDP and RTP

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- TCP not suitable for streaming due to certain functions
  - Slow start
  - Retransmission
  - Sliding window
- Thus UDP is used, has lower overhead
  - Retransmission unnecessary, no time to wait with real-time stream
- RTP necessary for additional functionality not in UDP
  - Timing information for media synchronization
  - Feedback from client to server

# Media Delivery

## Stream overhead

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- Total stream size is
  - Audio payload + Video payload + Headers
  - Headers = IP 20 bytes + UDP 8 bytes + RTP 12 bytes = 40 bytes
  - Header overhead in bits/sec depends on how many packets-per-second are streaming

Streaming Type	Packets Per Second
Real Audio @ 20 kbit/s	5
Windows Media Video @ 300 kbit/s	30
Real Video @ 300 kbit/s	80

- Packets-per-second is dependent on the application and media payload

# Media Delivery Targeting Bitrate

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- Efficient bitrate available to media streams is lower than link-rate due to various factors:
  - Header overhead
  - Packet loss
  - Other applications
- As a rule-of-thumb, calculate with only 75% available bitrate

Link Type	Effective Media Bitrate
Dialup @ 28.8 kbit/s	21.6 kbit/s
Dialup @ 56 kbit/s	42 kbit/s
ISDN @ 128 kbit/s	96 kbit/s



# Media Delivery

## Multi-bitrate streams

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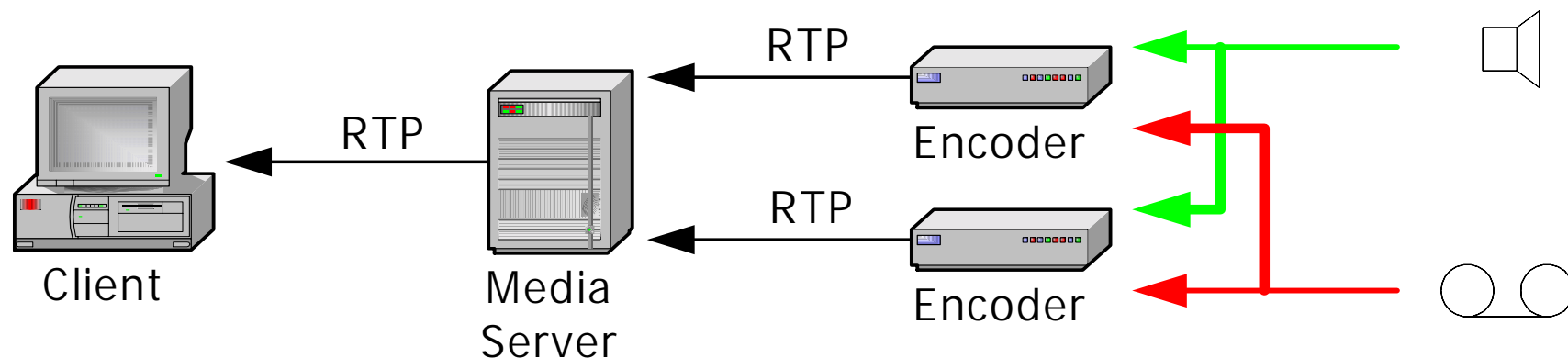
- Streaming packages have the ability to serve streams with multiple simultaneous bitrates
- The server and client negotiate which bitrate to play stream at. They can also scale up and down bitrate depending on congestion

# Media Delivery

## Live encoding

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- For live streams, the content must be encoded on-the-fly and streamed
- Typical architecture employs a dedicated encoder that streams to a server
  - For various bitrates either use multiple encoders or encoder with multi-bitrate capability

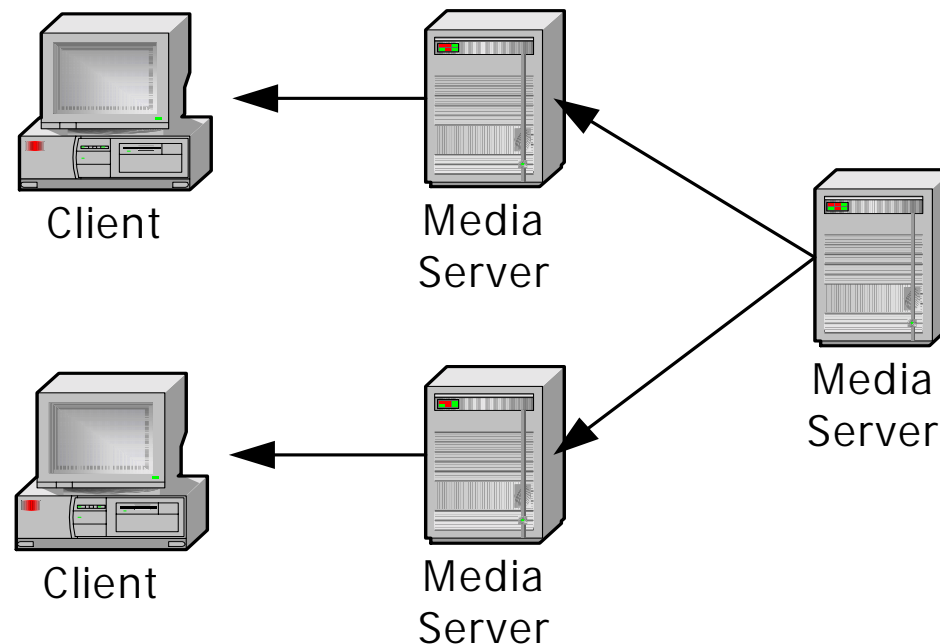


# Delivery Technologies

## Distributed Media Servers

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- Media servers can be set up in a distributed structure
  - Used for live where Multicast isn't an option, server replicates incoming stream to downstream servers
  - Used for on-demand to move content closer to end-user
- Concept is often called "splitting"

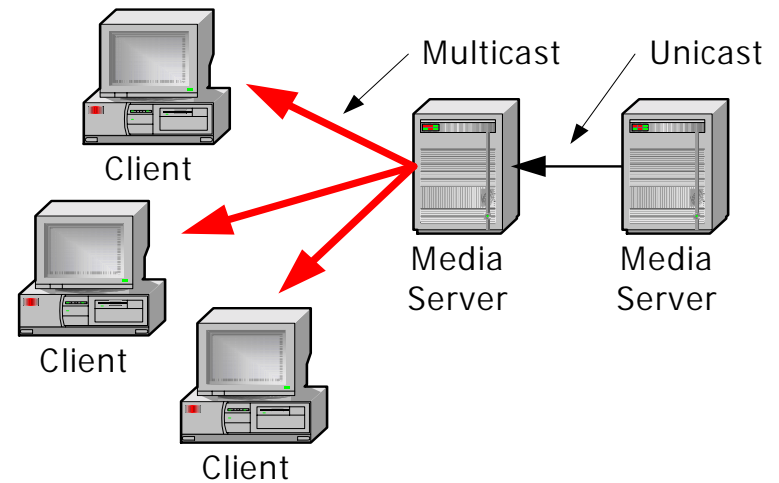


# Delivery Technologies

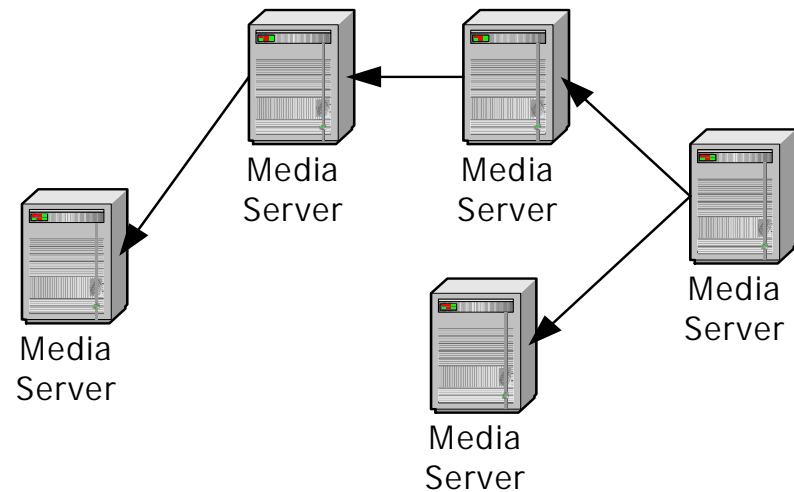
## Distributed Media Servers

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Can usually mix Unicast and Multicast in any combination



Can use multi-tiered architecture

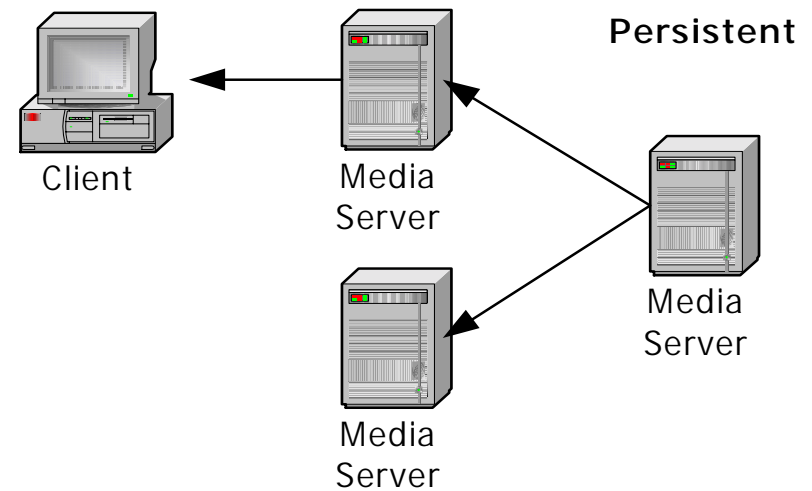


# Delivery Technologies

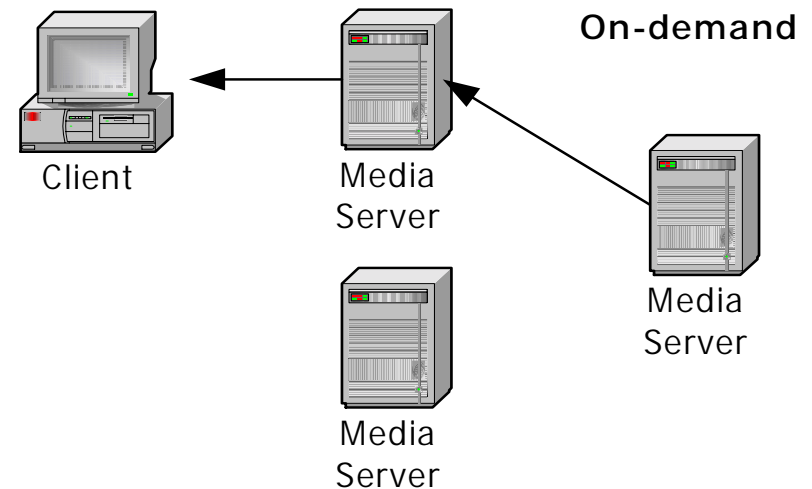
## Distributed Media Servers

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- “Push” architecture - streams flow persistently downstream
- Low delay for initial client request



- “Pull” architecture - streams only flow downstream if requested by client
- Higher delay for initial request



# Delivery Technologies

## Streaming Conclusion

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- Systems using RTSP / RTP are becoming industry standard
- HTTP / TCP not very suitable for streaming, although occasionally used
- Calculations must be done to ensure that total bitrate of stream, including headers, fits within target bitrate
- Where Multicast isn't available a distributed media server architecture can be used

# Delivery Technologies

## Error Correction

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- Streaming systems can utilize various forms of error correction to compensate for packet-loss:
  - Application retransmission of lost packets
  - Simple duplication of information, each packet contains replicate information
  - Transmission of checksums to allow correction of minor loss
  - Codecs that recover gracefully from packet-loss, where some loss is indiscernible to the viewer / listener
  - Systems that create parity packets from the original content. Packet order is irrelevant, but enough packets must be received to recreate original

# Delivery Technologies

## Rich Media

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- Platforms integrate additional features to pure audio / video streaming
  - Synchronization of presentations with audio / video
  - Slideshow formats, e.g. JPEGs with audio
  - VR formats, e.g. where the user controls viewpoint
  - Control of other applications with streaming players
  - SMIL (Synchronized Multimedia Integration Language) is a standards-based language that describes the presentation of rich media. Based upon XML



# Delivery Technologies

## Rich Media

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- There are many different forms of Rich Media. Some examples demonstrated are:
  - [Streaming Presentations](#)
  - [Slide-shows](#)
  - [Virtual Reality](#)
  - [Web-embedded streams](#)

# Delivery Technologies

## SMIL

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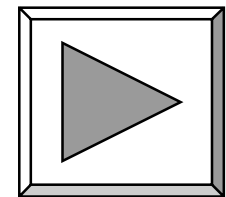
- SMIL is an XML-based media description format, and describes media in terms of space and time
- Defines the layout of content to a player, and the sequence that clips should be played back
- Can specify playback in parallel or sequence
- Has the ability to switch files based upon user Locale or bitrate

# Delivery Technologies

## SMIL Demo

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```
<smil>
  <head>
    <meta name="title" content="SMIL demonstration" />
    <meta name="author" content="Matthew Liste" />
    <meta name="copyright" content="© Matthew Liste" />
    <layout type="text/smil-basic-layout">
      <root-layout width="352" height="280" background-color="black"/>
      <region id="videotext" left="0" top="250" width="352" height="40" z-index="2"
      />
      <region id="audiotext" left="0" top="120" width="352" height="40" z-index="3"
      />
      <region id="video" left="0" top="0" width="352" height="240" z-index="1" />
    </layout>
  </head>
  <body>
    <seq>
      <par>
        <text src="videotext.rt" region="videotext"/>
        <video src="video.rm" region="video"/>
      </par>
      <par>
        <audio src="audio.rm"/>
        <text src="audiotext.rt" region="audiotext"/>
      </par>
    </seq>
  </body>
</smil>
```



SMIL Demo

# Delivery Technologies

## Streaming Interoperability

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- Although the various streaming platforms more or less use standards-based platforms, they are still not interoperable
- Industry is moving towards interoperability
- Interoperability must exist on four levels
  - Codecs
  - File format
  - Streaming transport protocol
  - Streaming initiation protocol

# Delivery Technologies

## Content Distribution

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- On-demand content must be distributed to remote servers
- Timing
  - Distribution before requests
  - “Just-in-time” delivery, e.g. when first request comes
- Distribution Mechanisms
  - Satellite
  - Multicast
  - Content management packages
- Functionality
  - Publishing at same time “everywhere”
  - Expiring / Deleting content

# Delivery Technologies

## Content Distribution – Timing

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- Pre-usage distribution
  - Use with media servers which need entire clip to be present to serve
  - For distribution of content during low-usage hours
  - Uses more total bandwidth as content must get distributed to all servers regardless of usage or not
- Just-in-time distribution
  - Used with caches that can start replaying content whilst receiving
  - Need to have bandwidth available at the particular moment that the content is requested

# Delivery Technologies

## Content Distribution – How

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- Unicast point-to-point distribution from content management to each server, or propagated through server tree
- Satellite distribution is inherently powerful for distributing content
- A Multicast enabled backbone is well suited for content distribution

# Delivery Technologies

## Content Distribution Functionality

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- Publishing Content
  - Often want availability at same time everywhere
  - Can wait until content is at all sites prior to publishing URL
  - Can use mechanisms on remote site to make content available at certain time
- Deleting Content
  - Equally as important as distribution is expiry of the content
  - Can be done through OS-specific tools such as 'cron' jobs or other home-grown scripts
  - Commercial packages that distribute content usually have mechanisms to expire and delete content



# Delivery Technologies

## Content Indexing

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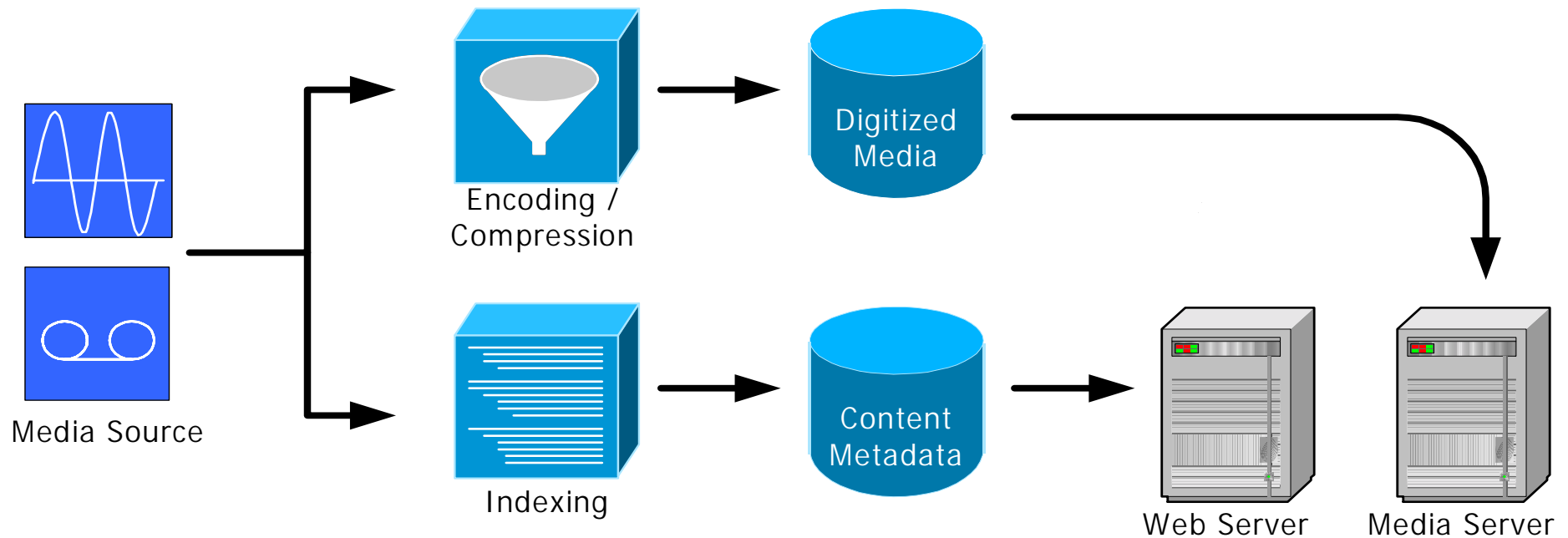
- Streaming Media cannot be automatically indexed like text-based media
- Need to generate index as reference to media
- Software available to do this with numerous features:
  - Face recognition
  - Audio-to-text, used as index
  - In-picture text recognition
  - Use of closed captioning or subtitles

# Delivery Technologies

## Content Indexing

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- The content is indexed in parallel with encoding
- Metadata is generated that can be used as an index to refer to the media itself



# Delivery Technologies

## Content Indexing

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- Demo - <http://www.virage.com/showcase/index.html>

# Delivery Technologies

## Caching

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- A cache is a device that temporarily stores web-objects for later retrieval
- First time a client requests content, the cache retrieves content from server. Subsequent retrievals of content come from cache, not from server
- Typically used to store content “closer” to end-users, thus limiting bandwidth used towards upstream content provider
- Three different cache mechanisms
  - **Proxy** - Each client is configured to use a specific cache. This cache can in turn be configured to use other caches
  - **Transparent** - A router or switch in the path of the client-server request intercepts query and redirects to a cache
  - **Network-layer** - Cache is situated physically in path and intercepts queries traversing it

# Delivery Technologies

## Streaming Media Caches

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- Caches are implementing support for streaming media
- This enables the caches with two functions:
- **Splitting of live streams**
  - Can often “transpose” from Multicast to Unicast and vice-versa
- **Caching of on-demand streaming content**
  - Does “instant replay”, where the cache starts streaming to the client whilst receiving the content
- Caveat: Due to the non-interoperability of streaming platforms, caches must support each vendor’s platform

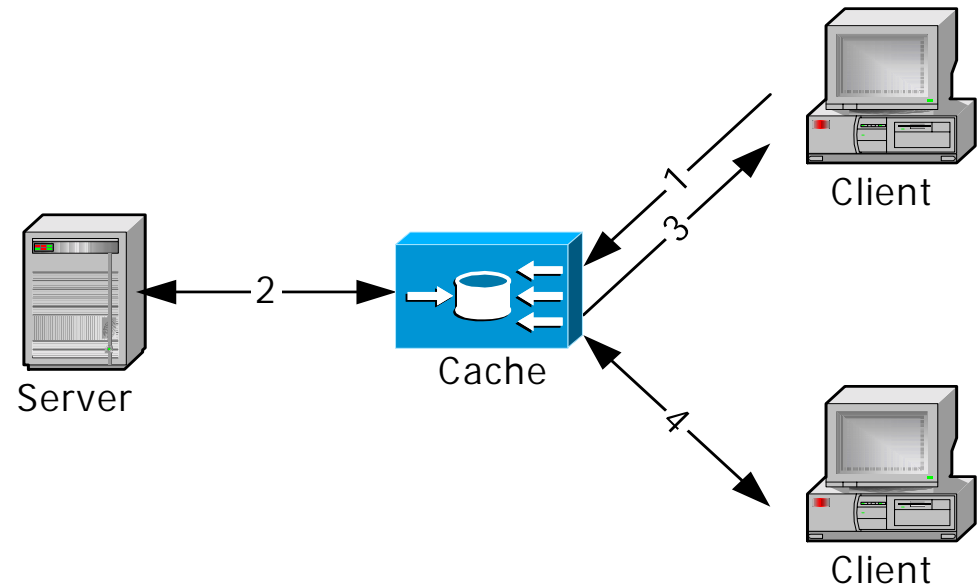
# Delivery Technologies

## Proxy Caching

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- Each client (Web-browser or media-client) must be statically configured with address of proxy cache, uses proxy for certain protocols (e.g. HTTP / RTSP etc.)
- Cache auto-discovery protocols such as WPAD could be used
- Also used for security purposes to better control access

- 1 - Client requests content from cache
- 2 - Content not in cache, so cache pulls content from source
- 3 - Cache delivers content to requestor
- 4 - Next client requesting same content gets served instantly from cache



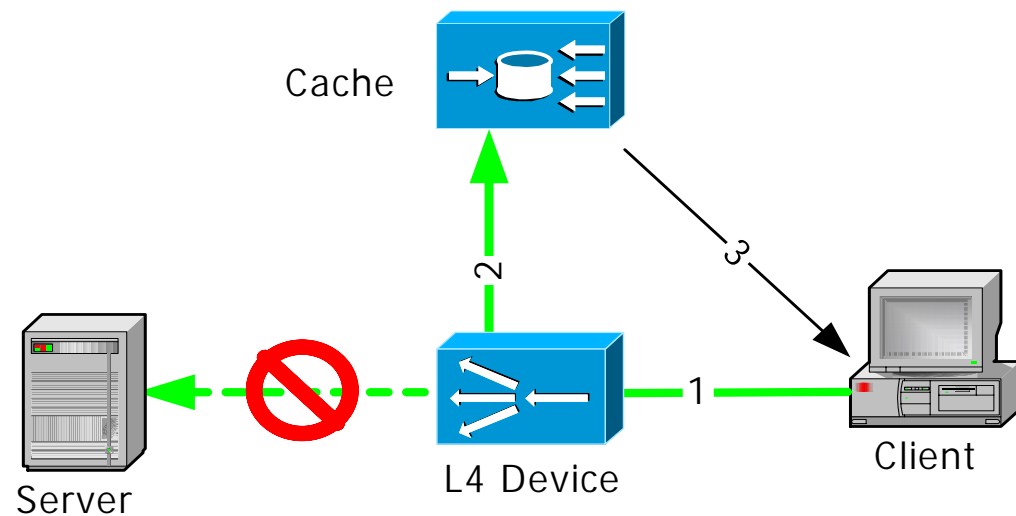
# Delivery Technologies

## Transparent Caching

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- No configuration necessary on client side
- Content request is redirected by layer-4 aware device
  - L4 switch
  - Router using Cisco's Web Cache Control Protocol (WCCP)
  - Router implementing policy-based routing

- 1 - Client requests content from server, L4 device intercepts query
- 2 - Query is redirected to local cache
- 3 - Cache delivers content to requestor

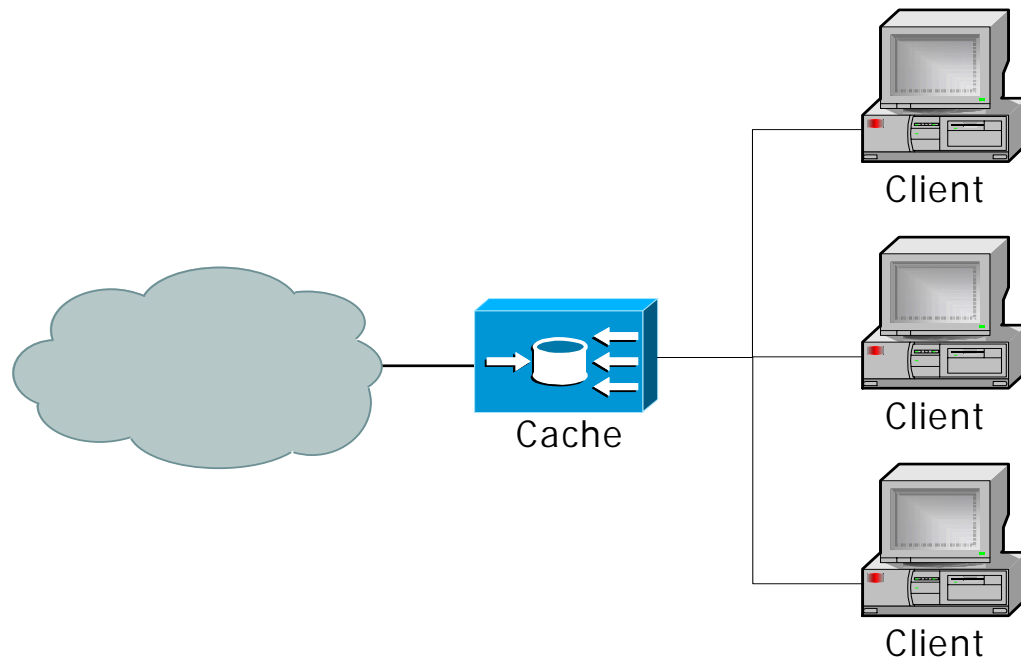


# Delivery Technologies

## Network-Layer Caching

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- Cache sits between router and LAN
- All requests must intrinsically be fed through cache
- Built-in Layer-4 function “routes” relevant traffic to internal cache
- Usually has two Ethernet interfaces





# Delivery Technologies

## Caching Methods – Pros and Cons

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- Transparent caching usually easiest to deploy and has built-in stability
- Regardless of method chosen, must have support for streaming media platform / protocol used
- HTTP support not sufficient!

Cache Type	Pros	Cons
Proxy	Cheap Ubiquitous support	Static client configuration
Transparent	No client configuration Provides support for redundancy	Must have L4 device Adds extra “hop” to retrieval
Network-Layer	No client or L4-device configuration Fast	Extra point-of-failure

# Delivery Technologies

## Cache Structures

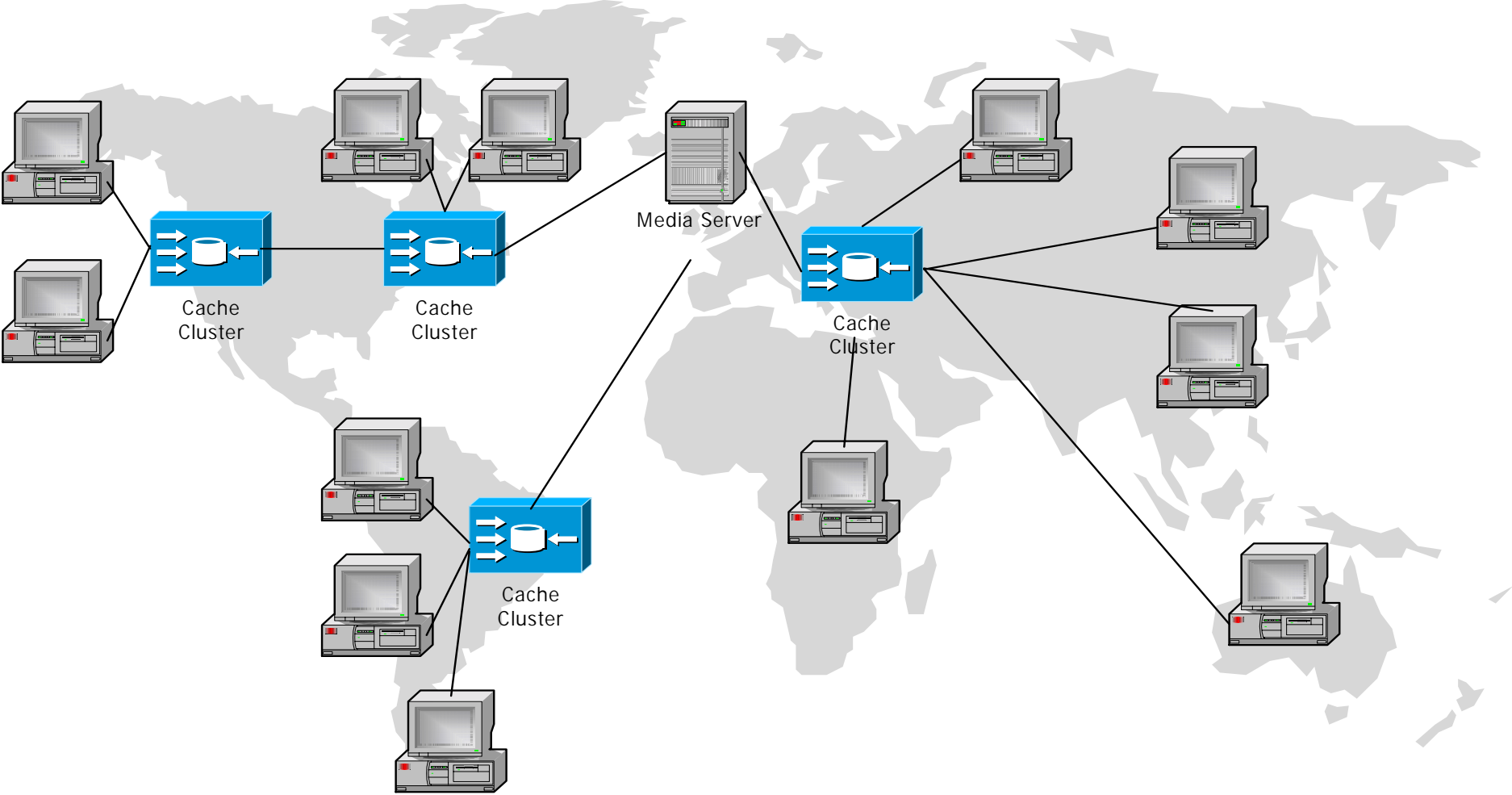
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- Caches can be set up in hierarchies, where a cache will contact an upstream cache to get content
- In a cluster, caches request content from peer caches
- Internet Cache Protocol (ICP) can be used between caches to request content (Or proprietary protocols used by caching vendors)

# Delivery Technologies

## Cache Structures

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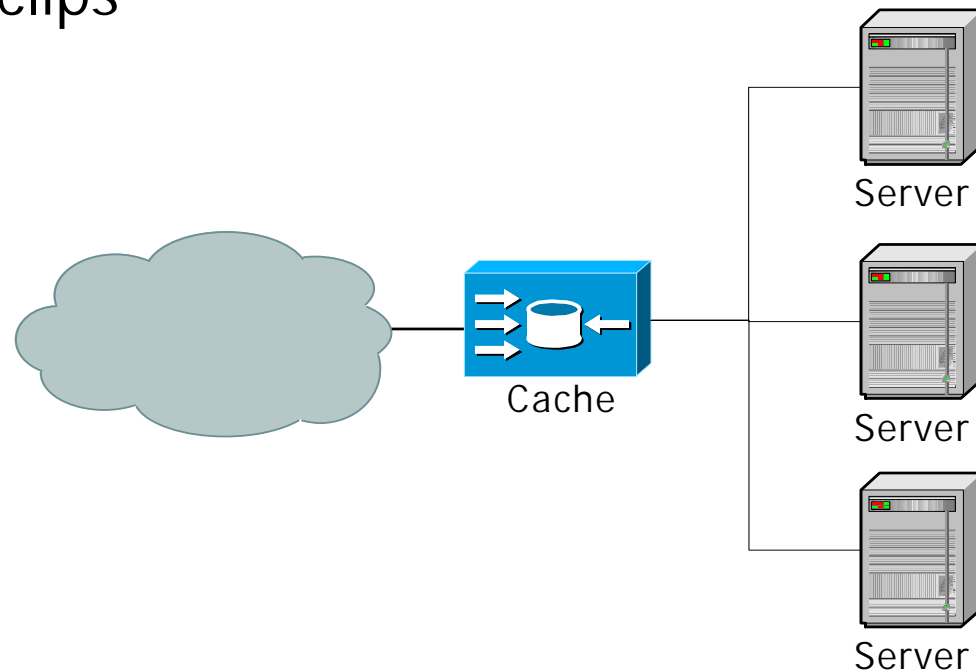


# Delivery Technologies

## Front-end Caches

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- Can also use cache to sit in “front” of server-farm to offload requests
- Typically used for content that has a frequent request rate, for example an audio advertisement that gets played before all clips



# Delivery Technologies

## Caching Conclusion

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- Caching is a mature technology today for static content
- Support for streaming content is becoming available
- A cache-based infrastructure is very attractive for streaming media because it is platform-agnostic
- Intelligent usage of dynamic caching can greatly reduce bandwidth usage
- Need a lot of disk-space in cache:
  - 60 minute video clip @ 300 kbit/s = 135 Mbytes

# Delivery Technologies

## Content Routing

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- Content Routing - How to match a client with the right content from the right place
- How:
  - DNS redirection
  - HTTP redirection
  - URL rewrite
- Where:
  - Globally
  - Locally
- Redirection systems:
  - DNS servers
  - Layer 4-7 switches
  - Custom software

# Content Routing

## Centralized / De-centralized

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- Two different models to redirect client requests
- **Centralized:** All requests are initially handled by a global content router which in turn redirects to distributed systems
  - DNS Redirection
  - HTTP Redirection
  - URL Rewrite
- **De-centralized:** The requests are handled directly by the distributed systems
  - WCCP
  - Policy-based Routing
  - Anycast

# Delivery Technologies

## Content Routing - Where?

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- The content routing service needs to decide where to send the client request
- Many different parameters can comprise this decision policy:
  - Geographical proximity
  - Network proximity (i.e. by router-hops, bandwidth etc.)
  - Server load
  - Content availability
  - Fairness (i.e. round-robin)
  - End-user type (Paying customer gets access to better server than non-paying)

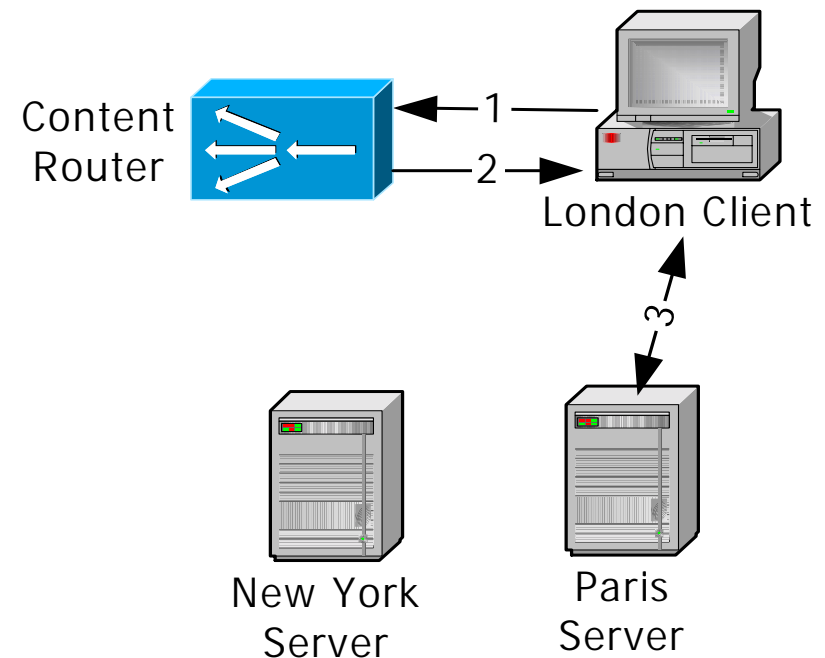


# Delivery Technologies

## Global Content Routing

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- Redirection between sites
- Usually used to serve content from “closest” possible site
  - Not necessarily geographical proximity, but could be hop-count, ping time etc.



1 - Client requests content from Content Router  
2 – Content Router picks closest server based upon network proximity, returns IP address for that server

3 - “Closest” Server serves client

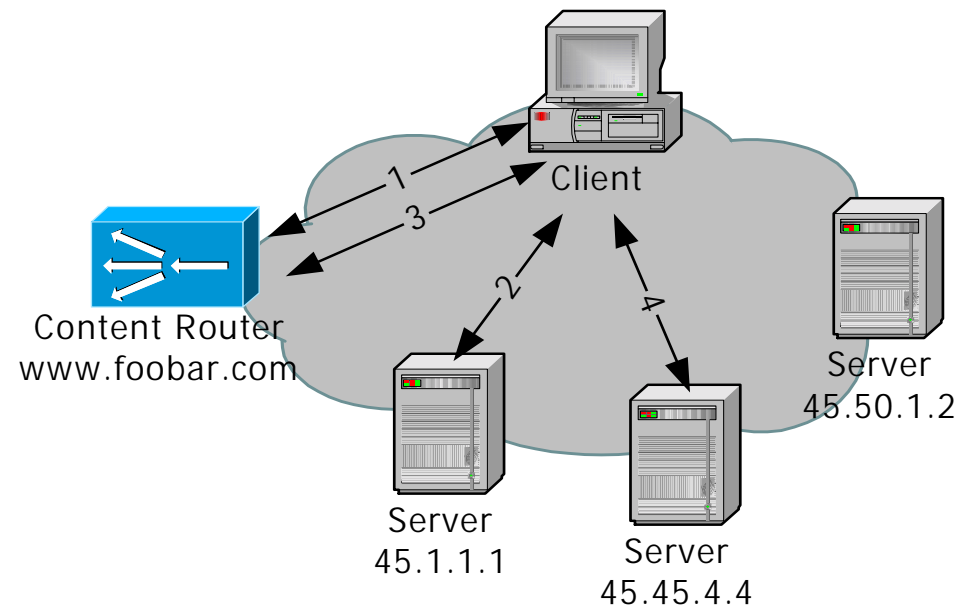
# Delivery Technologies

## DNS Content Routing

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- Offers content routing appearing as a DNS server
- The content router will choose an IP address for a DNS query according to its policies

- 1 - Client requests IP address for www.foobar.com
- 2 - DNS server returns 45.1.1.1
- 3 - Client requests IP address for same hostname later
- 4 - DNS server returns 45.45.4.4



# Delivery Technologies

## HTTP Redirect

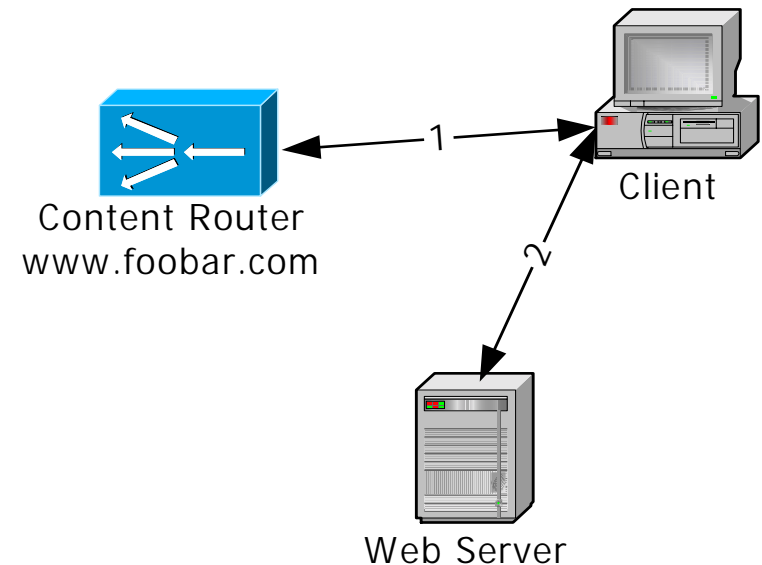
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- Uses the "302 Redirect" directive in HTTP/1.1 to redirect the browser to a new site

1 - Client requests URL from content router

<http://www.foobar.com/content>

2 - Content Router send HTTP redirect to client, thus making client load page from new server



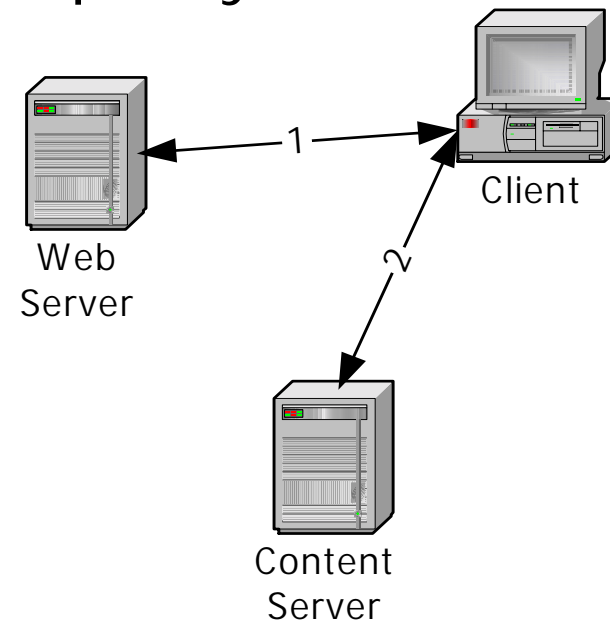
# Delivery Technologies

## URL Rewrite

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- Technique used to point client to server through URLs
- Web server will run script that dynamically changes URLs based upon policy

- 1 - Client requests URL from Web server
- 2 - Using script, Web server gives client URLs pointing to content server

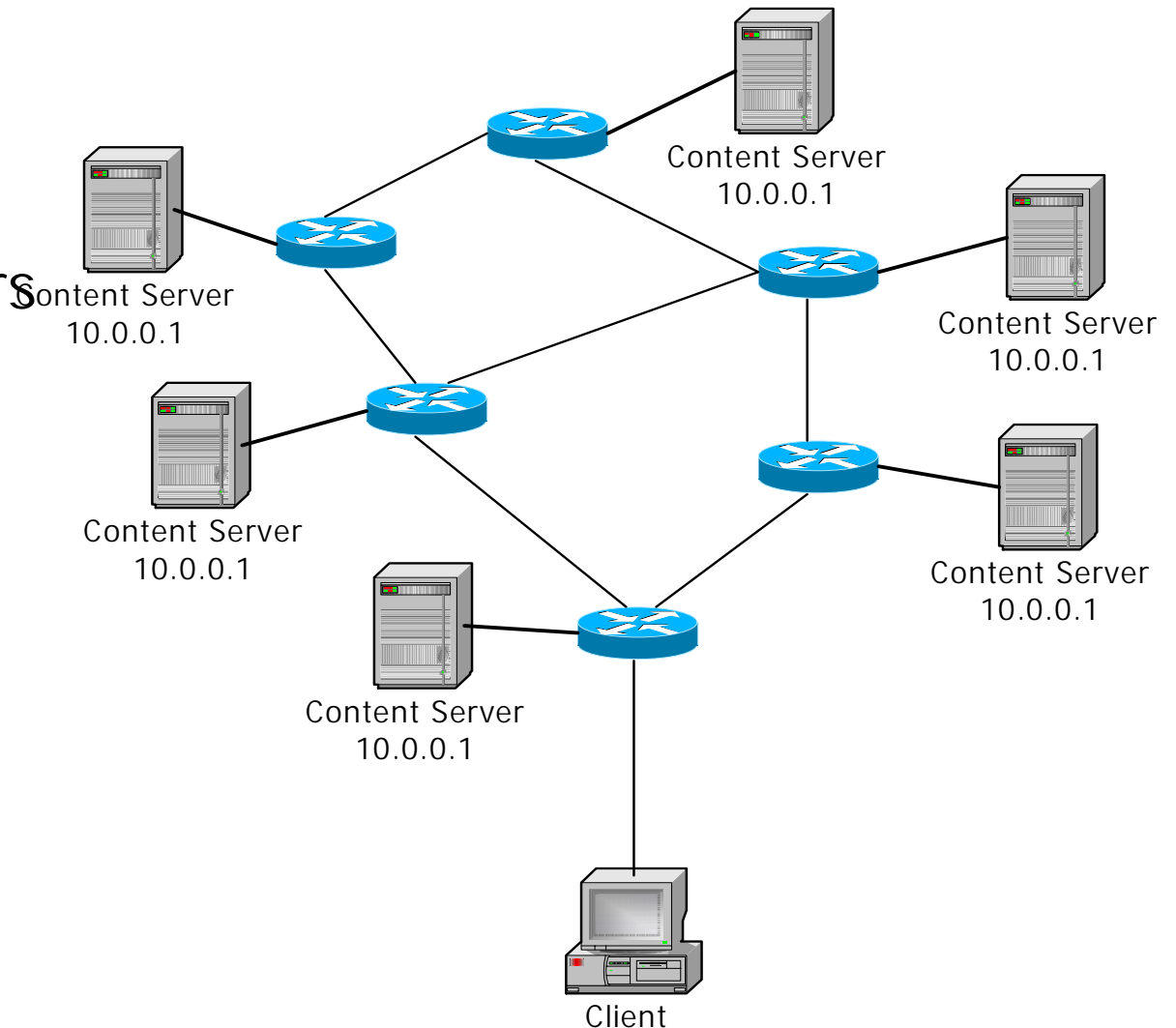


# Delivery Technologies

## Content Routing - Anycast

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- All servers have same IP address
- Servers either directly attached or update routers via routing protocol
- Routers use regular routing table to direct requests to closest server



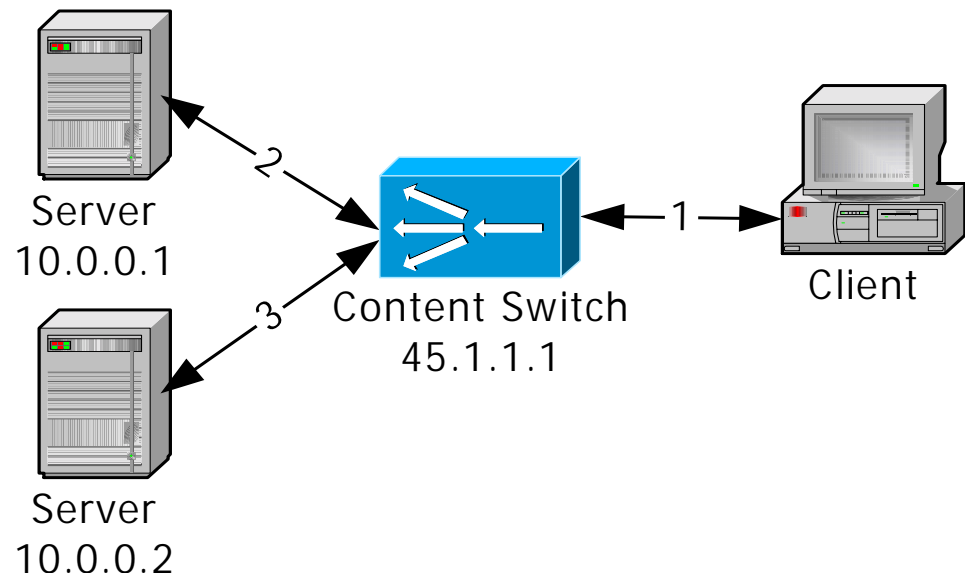
# Delivery Technologies

## Local Content Routing

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- Often known as Content Switching
- Multiple servers can have the same virtual IP address (VIP) assigned
- From the client-side, the request looks like it is to one IP address, although it could go to a different server each time

- 1 – Client requests content from 45.1.1.1
- 2 – Content Switch sends request to 10.0.0.1
- 3 – Next request goes to 10.0.0.2



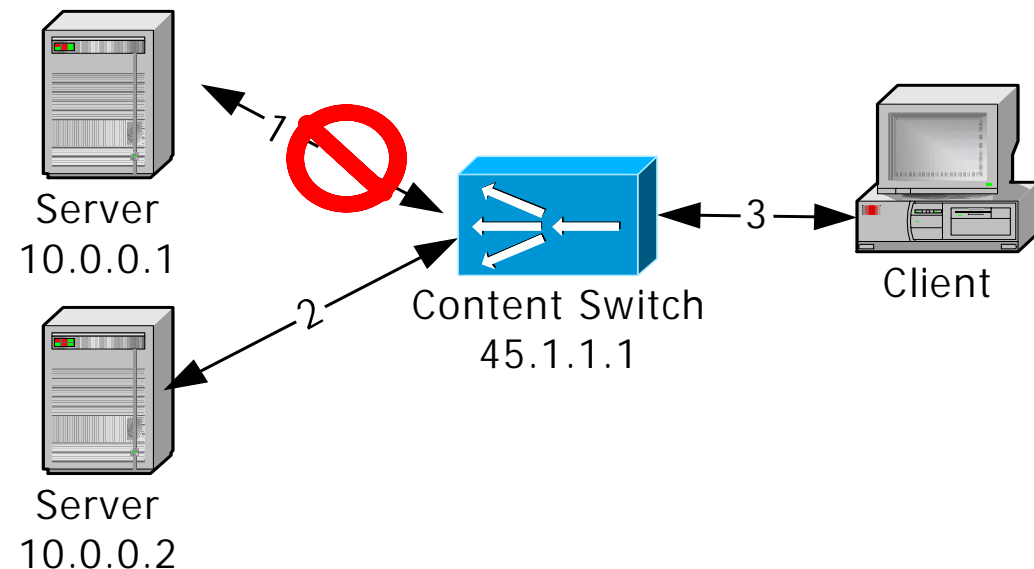
# Delivery Technologies

## Intelligent Content Switching

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- Has awareness of server status through agent running on server
- Generally tests availability of content, not just server

- 1 – Content Switch constantly tests the various servers, and finds that service on 10.0.0.1 is down
- 2 – Content Switch verifies that 10.0.0.2 is ok
- 3 – When request comes from client, it is redirected to 10.0.0.2



# Delivery Technologies

## Persistence

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- Since back-end servers often do not maintain same session state with clients, content switches need to maintain persistence between server and client
- Four methods to maintain session persistence:
  - Source IP address
  - Cookies
  - SSL Hash
  - URL ID

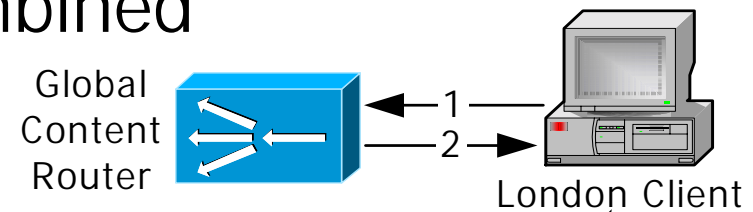


# Delivery Technologies

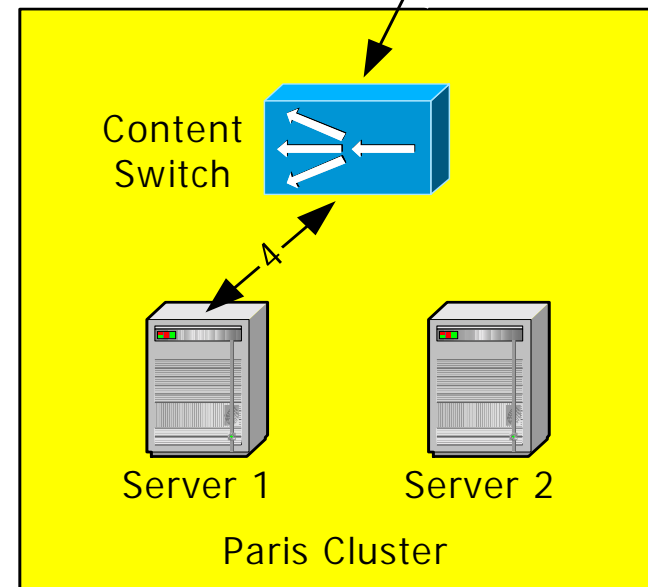
## Tiered Content Routing

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- Obviously, both global and local Content Routing are often combined



- 1 - Client makes request to global Content Router
- 2 - Content Router picks closest cluster based upon network proximity, and returns IP address for Content Switch for that cluster
- 3 - Local Content Switch at cluster chooses server
- 4 - Server serves client



# Delivery Technologies

## Content Routing Conclusion

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- Content Routing ultimately gives clients better performance in a transparent manner
- All the content is seemingly available from a root site, then various “smoke-and-mirror” methods serve the client from the relevant server
- Powerful way to bypass network bottlenecks

# Delivery Technologies

## Redundancy

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- Redundant infrastructure needs to be enabled for reliable delivery
- Use parallel encoders and root servers for live streaming
- For streaming servers, basically just a combination of previous techniques:
  - Content Routing with content availability awareness
  - Caching with WCCP or similar mechanism for redirection to multiple caches

# Delivery Technologies

## Security

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- Several (very different) aspects of security need consideration when delivering streaming media
- Copyright protection
  - For pay-per-view or downloadable content the content provider wishes to ensure that the content is protected from unauthorized copies
- User authentication
  - To ensure that only authorized users get access to the content user authentication schemes must be implemented
- Firewalls
  - How content is moved through firewalls and proxies

# Delivery technologies

## Copyright protection

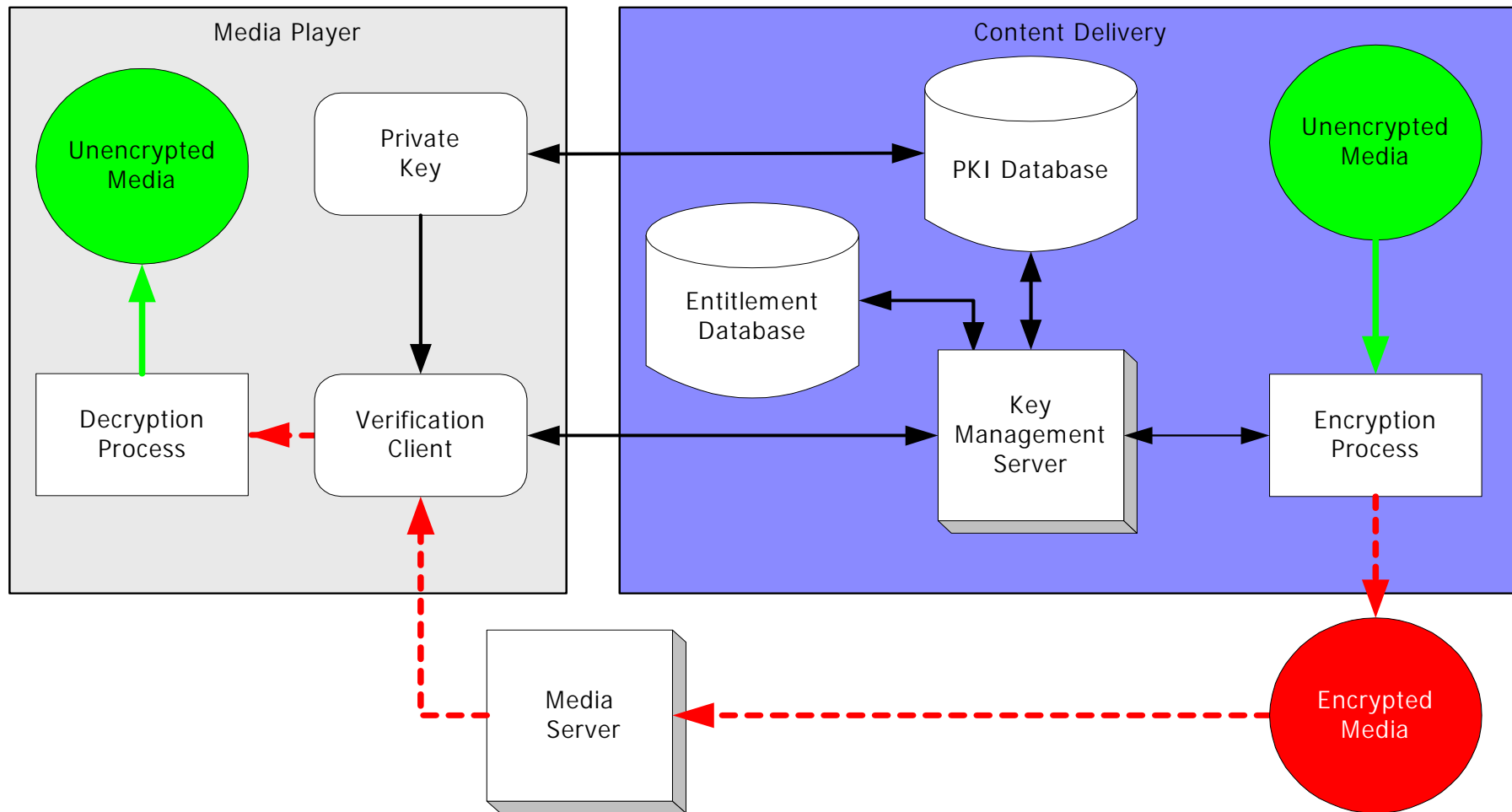
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- Ease of copying makes digital distribution difficult for content owners
- MP3 and DivX technology has made illegal copying extremely easy, industry is playing catch-up
- Proprietary formats exist to address these issues and implement the following technologies:
  - Watermarking
  - Encryption
  - Copyright protection
  - Licensing
- Secure distribution can enable new features:
  - Pay-per-view/listen
  - Song / Movie rental
  - Right to copy (or not)

# Delivery technologies

## Securing Media

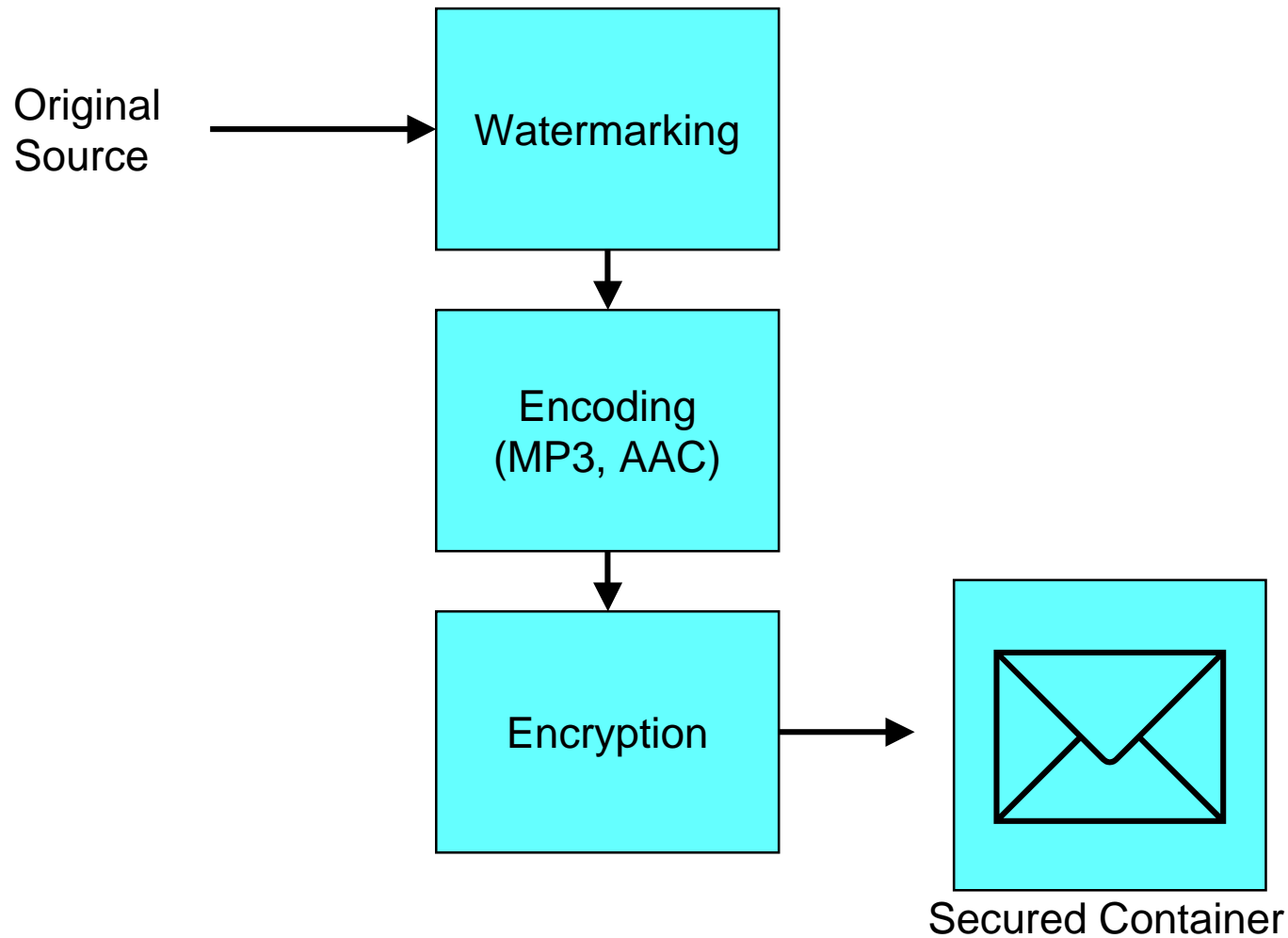
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# Delivery technologies

## Copyrighting techniques

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# Delivery technologies

## Securing Streamed Media

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- Streaming media is inherently copy-protected, the players do not store the content
- Of course, someone could design a player that would store streaming content
  - Streambox have made a player that can record content being streamed from a RealServer, and are getting sued by RealNetworks
  - Logically identical to a VCR

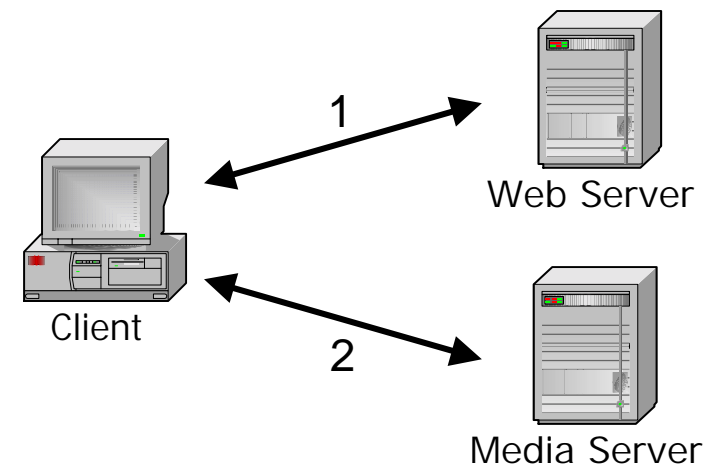


# Delivery Technologies Authentication

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- Media servers can authenticate users just as web servers can
- Why not just use web-server authentication ?
  - Because players can get content directly from media server if the URL is known
- “Protection” - The player doesn't reveal the URL

1 - Client requests URL from web server, which feeds back URL to media server  
2 - Client then requests stream from media server



# Delivery Technologies

## Firewalls

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- Firewalls block streaming content by default
- RTSP uses fixed TCP port 554, MMS uses fixed TCP port 1755
  - But RTP uses dynamic UDP ports negotiated in session setup
- Several options to get streaming content through firewalls:
  - Use firewalls that support streaming protocol(s), through stateful awareness
  - Open up ports used for RTP, these vary by vendor
  - Use streaming proxies, such as caches with that functionality
  - Last resort, use HTTP streaming

# Delivery Technologies

## Stateful Firewalls

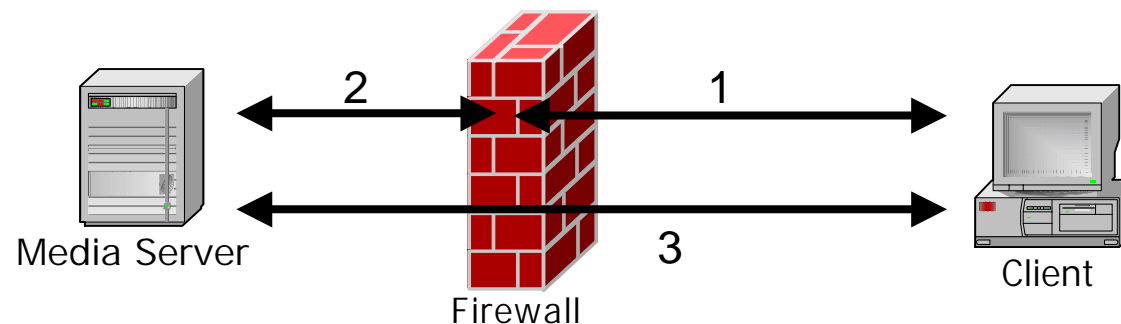
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- This is the preferred manner for streaming through a firewall

1 - Client requests stream from server using RTSP on port 554

2 - 554 is passed through firewall, and session negotiations between server and client are inspected by firewall

3 - Firewall opens up negotiated ports for session

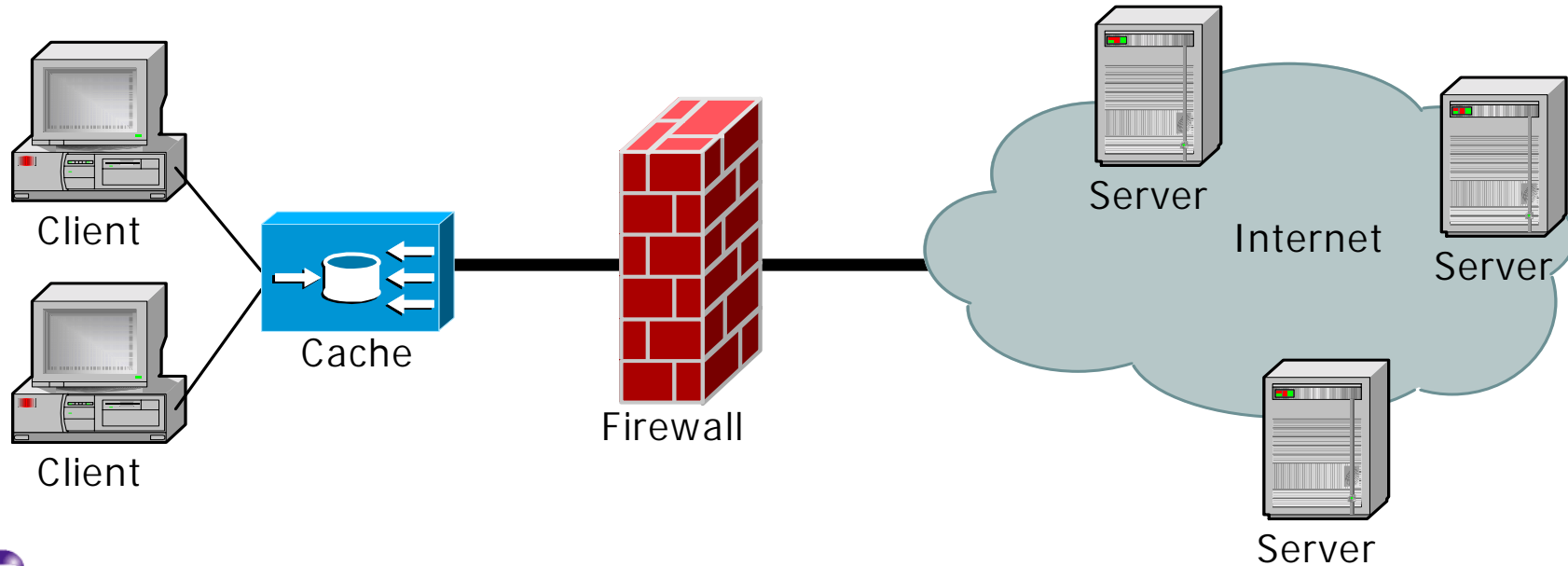


# Delivery Technologies

## Streaming in through Firewalls

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- Can also front-end firewall with proxy cache to bring in external content

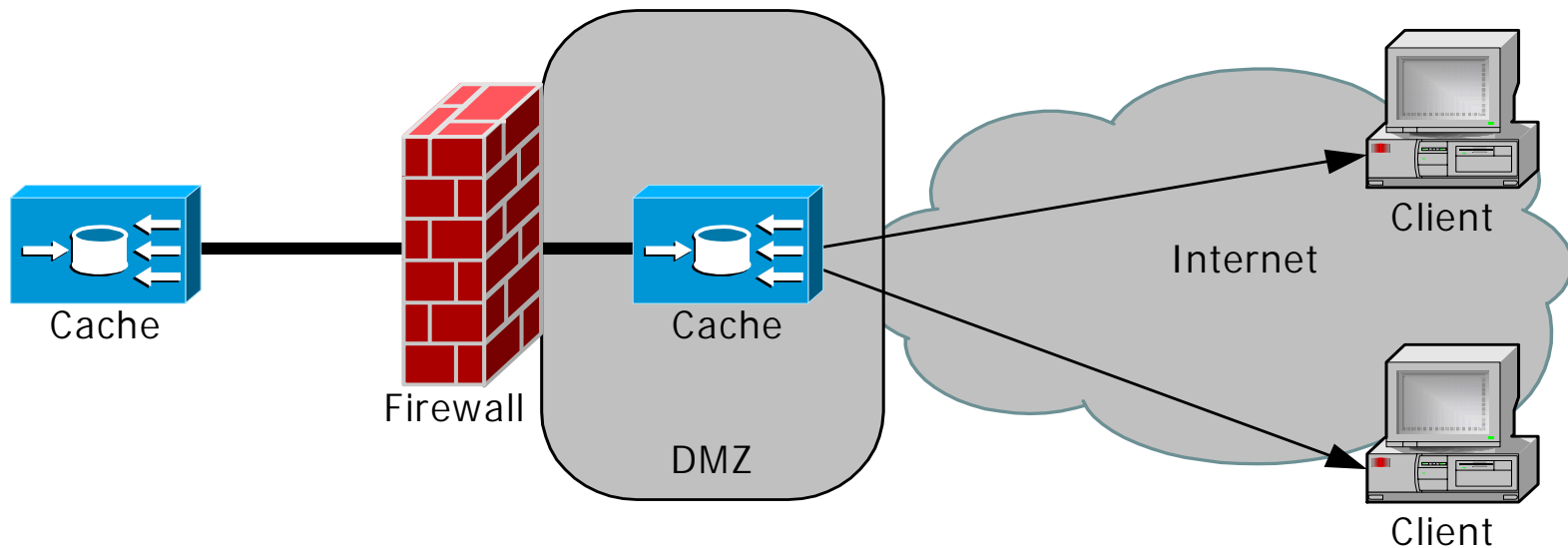


# Delivery Technologies

## Streaming out through Firewalls

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- If streaming content to external users, can deliver through firewall by using proxy cache



# Delivery Technologies

## Security - Conclusion

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- If content needs to be protected, any of the encryption formats need to be used
- If content is only to be made available to certain users, authenticate on media server, not on web server
- When streaming in through firewall, use either stateful firewall, or tunnel through using proxy

# Agenda

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- Introduction
- Digitisation of media content
- IP infrastructure for streaming media
- Delivery Technologies
- **Practical Tips**

# Practical Tips

## Delay

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- The low bitrate streaming packages are not suitable for real-time
- The aggregate delay, from encoder to server to player can be over 20 seconds
- If the encoding and decoding is done in hardware the delay will usually decrease
- Some of the high bitrate packages (MPEG-1/2) encode and decode in hardware and achieve far lower delays



# Practical Tips

## Stream degradation

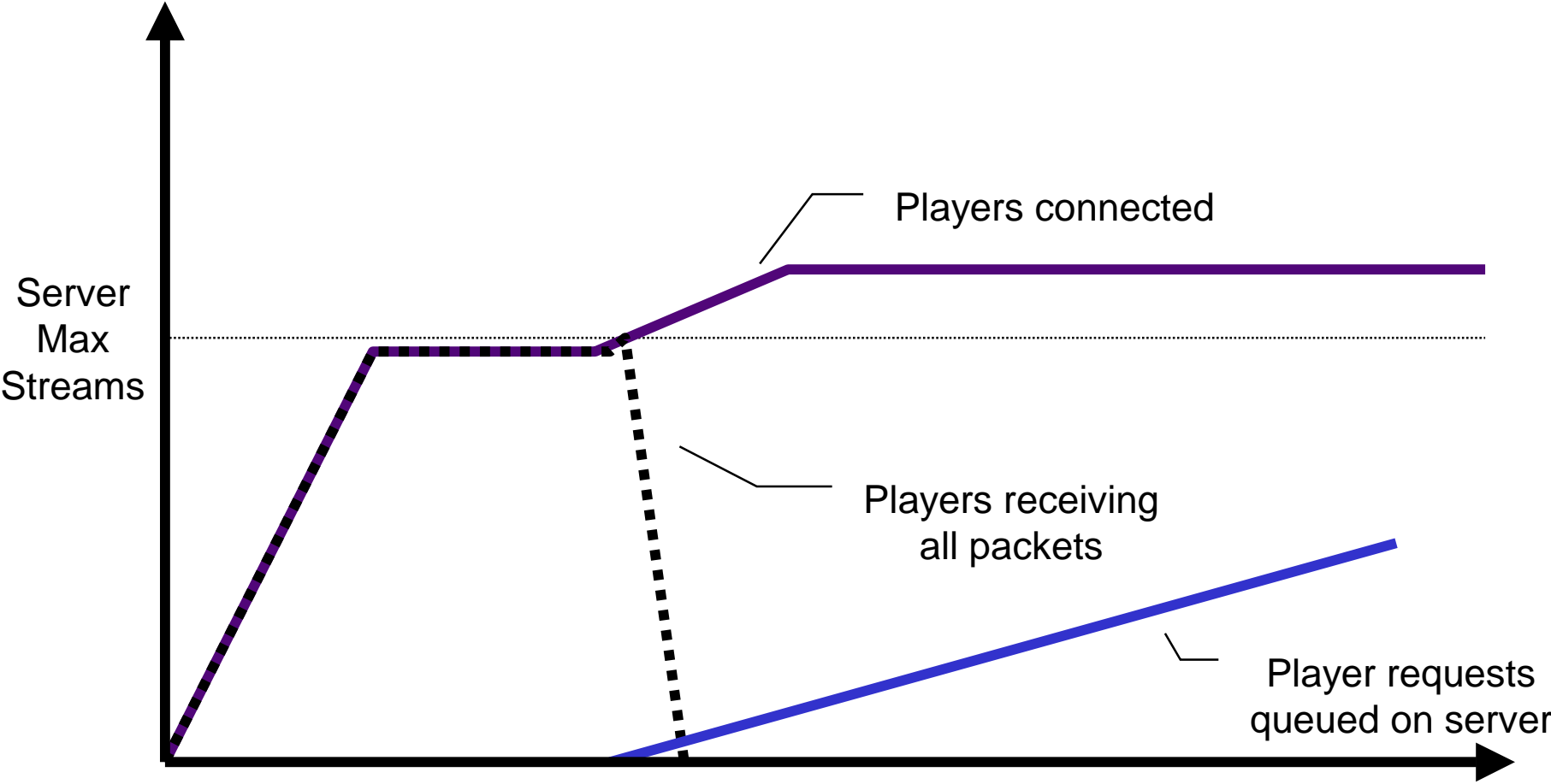
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- Stream delivery can get degraded due to network congestion or server overload
- Network generally will drop randomly from all streams, thus affecting delivery to all clients
- When server becomes overloaded it will drop random packets to all streams
- Clients will re-request content, thus loading server even further
- How to avoid this:
  - Set up media server to only serve N streams, when N+1 player tries to connect they get connection refused.
  - Use L4 functionality to restrict access to server. When maximum is achieved stop serving
  - This way one ensures that already connected players still get served efficiently

# Practical Tips

## Stream degradation

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# Practical Tips

## Compression parameters

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- Always leave overhead in regards to the target audience, i.e. use 20 kbit/s for 28.8 modem users
- It can often be better to encode video at half the framerate with high clarity than full framerate with low clarity
- Always use good codec for audio, a clip with good audio and poor video is more palatable than the opposite
- If squeezed on bandwidth it's better to encode one good mono channel than two bad stereo channels

# Practical Tips

## Splitting up content

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- To increase usefulness of caching, content should be split up:
- If playing repetitive content from media server, for example advertisements, split this out as separate clips instead of encoding them with all other clips
- By using play-lists, player will load and play advertisement, then requested content
- By splitting out these clips, the caches will serve them instead of the media server. This is useful both with front-end caches, and also for “downstream” ISP networks that cache

# Agenda

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- Introduction
- Digitisation of media content
- IP infrastructure for streaming media
- Delivery Technologies
- Practical Tips
- Future

# Future

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- Low bitrate
  - Audio and video content to wireless devices over IP
- Residential streaming
  - Higher bitrates are making video-on-demand at VCR quality commercially viable
  - As codecs improve, computers get faster and bandwidth increases, DVD quality streaming with MPEG-2 will be available
  - When will we have HDTV over IP?
- Upcoming features
  - Searchable content
  - Subtitling
  - Rewind and control of live streams
  - Interactive content
  - VR content where viewpoint can be controlled

# Future ISMA

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- ISMA – Internet Streaming Media Alliance
  - <http://www.ism-alliance.org/>
  - “ISMA, a group of industry leaders in content management, distribution infrastructure and media streaming, have joined together to promote open standards for developing end-to-end media streaming solutions”
- Current goal is to develop standard for streaming MPEG-4 over IP
- Founded by Apple, Cisco, IBM, Kasenna, Phillips and Sun.
- Conspicuously, both RealNetworks and Microsoft are missing from the alliance

# Future Transcoding

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- Media Servers are being developed that can transcode media on the fly
- Media is stored in best potential quality, for example MPEG-2, and server 'downgrades' stream in real time depending on player's requirement
- Advantages:
  - Only need one media server for all content types
  - Lower storage requirements
  - Higher flexibility, can serve any format requested
  - Removes need for end-users to support multiple different media players
- However:
  - CPU Intensive, expensive to do this in real time



# Future

## MPEG-4 Fine Granular Scalability (FGS)

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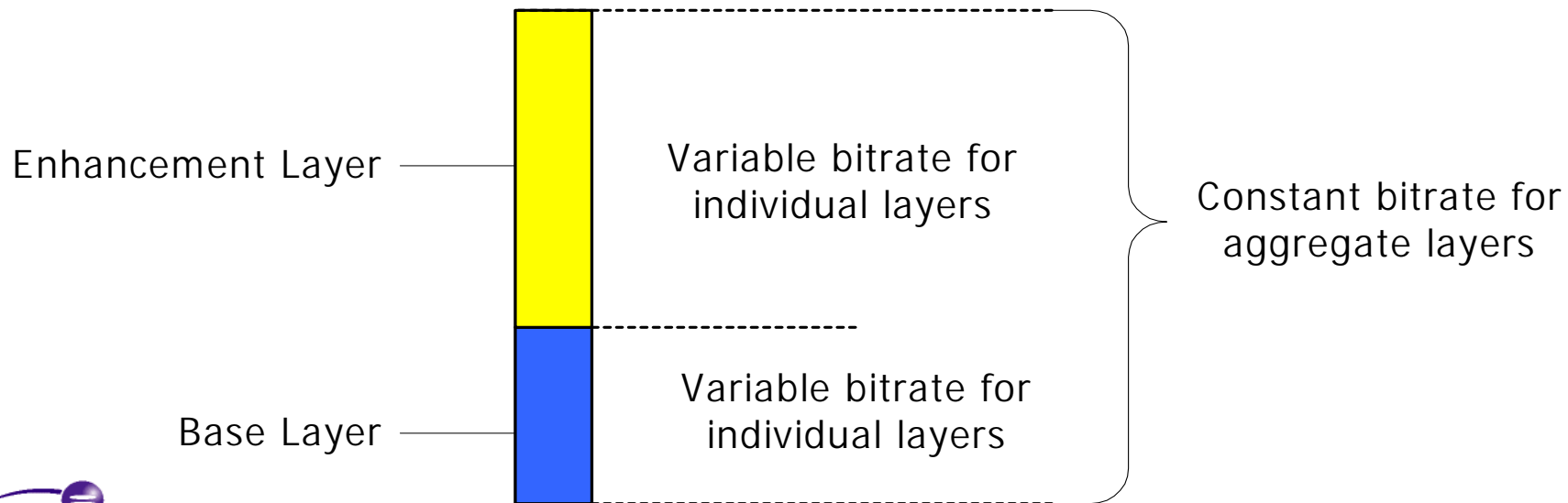
- Recently adopted as part of MPEG-4 for streaming applications – MPEG-4 v4
- Optimized for streaming through:
  - Resiliency to packet-loss
  - Handles variability in bandwidth between client and server
  - Server scalability through intrinsic support of Multicast
- Key technology is video scalability:
  - Encodes the video sequence as multiple layers
  - Base plus enhancement layer
  - Each layer can be received by player separately

# Future

## MPEG-4 Fine Granular Scalability

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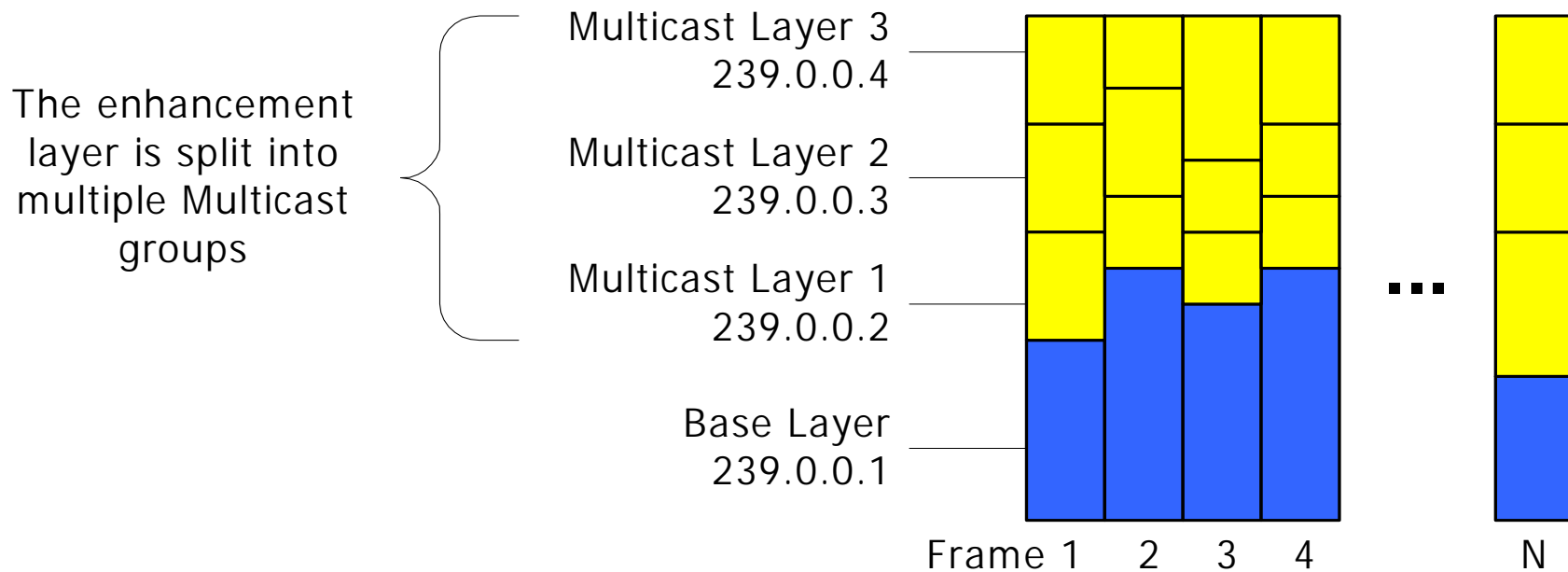
- The base layer holds all the critical information
- The enhancement layer is 'optional'
- Content is encoded using constant bitrate.
- Server can transmit however much needed of the enhancement layer, eliminating the need to encode at multiple bitrates depending on client connectivity



# Future MPEG-4 FGS and Multicast

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- Could split up enhancement layer into multiple Multicast groups
- If an enhancement layer is degraded, doesn't severely impact stream rendering



## References – Standards

### IETF – <http://www.ietf.org>

- RFC1889 RTP: A Transport Protocol for Real-Time Applications
- RFC2032 RTP Payload Format for H.261 Video Streams
- RFC2190 RTP Payload Format for H.263 Video Streams
- RFC2250 RTP Payload Format for MPEG1/MPEG2 Video
- RFC2326 Real Time Streaming Protocol (RTSP)
- RFC2327 SDP: Session Description Protocol
- RFC2435 RTP Payload Format for JPEG-compressed Video
- RFC2474 Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Header
- RFC2475 An Architecture for Differentiated Services

### ITU – <http://www.itu.int>

- H.261 Video codec for audiovisual services at p x 64 kbit/s
- H.263 Video coding for low bit rate communication

### ISO – <http://www.iso.ch>

- ISO/IEC 11172 MPEG-1 - Coding of moving pictures and associated audio for digital storage media at up to about 1.5 Mbit/s
- ISO/IEC 13818 MPEG-2 - Generic coding of moving pictures and associated audio information
- ISO/IEC 14496 MPEG-4 - Coding of audio-visual objects

### ISO MPEG Working Group – <http://drogo.cselt.it/mpeg/>

- MPEG-4 Overview Version 3 and Version 4
- MPEG-7 Overview Multimedia Content Description Interface

## References – Websites

### Useful

- Caching <http://www.caching.com/>
- Codecs <http://www.codeccentral.com>
- MPEG <http://www.mpeg.org>  
<http://bmrc.berkeley.edu/frame/research/mpeg/mpeg2faq.html>
- MP3 <http://www.askmp3.com>
- Multicast <http://www.ipmulticast.com/>
- Streaming <http://www.streamingmedia.com>  
<http://www.streamingmediaworld.com>

## Dictionary of Terms

<b>AAC</b>	MPEG-2 Advanced Audio Coding
<b>ASF</b>	Microsoft Windows Media Advanced Streaming Format
<b>ASX</b>	Microsoft Windows Media Metafile
<b>Audio Equalization</b>	Technique to amplify or suppress certain frequencies
<b>Audio Normalization</b>	Technique to maximize audio levels after recording

<b>AVI</b>	Microsoft Windows Audio Video Interleave Format
<b>Cable Modem</b>	Modem that allows Internet access via the cable television connection
<b>Cache</b>	System within a network that monitors object requests and stores objects it retrieves from a server. On subsequent requests for the same object, the cache delivers the from its storage rather than passing the request on to the origin server
<b>CBR</b>	Constant Bit-Rate
<b>CBQ</b>	Class-Based Queuing
<b>CCIR601</b>	ITU standard for digital "broadcast quality" video, e.g. at PAL / NTSC / SECAM resolution
<b>CIF</b>	Common Intermediate Format – a 352x288-pixel video format specified at 29.97 fps
<b>Codec</b>	Compressor / decompressor, a codec is an algorithm for compressing and decompressing data
<b>Content Delivery Networks</b>	CDN – Networks that have multiple servers thus pushing content “closer” to the end-user
<b>DC Offset</b>	Low-frequency, inaudible noise that results from equipment grounding problems
<b>Diffserv</b>	Differentiated Services, a IETF standard defining an architecture for implementing scalable service differentiation in the Internet
<b>Digital Media</b>	Any media, e.g. Audio, Video or Pictures in digital format
<b>DSL</b>	Digital Subscriber Lines – technology that allows Internet access via telephone connection
<b>DVD</b>	Digital Versatile Disc – CD-sized disk that can hold from 4.7 to 17 GB of data, used with MPEG-2 for movies
<b>DVMRP</b>	Distance Vector Multicast Routing Protocol – The first Multicast routing protocol
<b>DWDM</b>	Dense Wavelength Division Multiplexing – Optical technology used to increase bandwidth over existing fiber optic backbones
<b>Firewall</b>	A system to prevent unauthorized access to or from a private network
<b>Flow</b>	A stream or set of packets on a media that can be considered as an aggregate
<b>FPS</b>	Frames per second
<b>G.711</b>	ITU standard for digitization of voice, uses 64 kbit/s
<b>G.723.1</b>	ITU standard for digitization of voice, uses 5.3 or 6.3 kbit/s
<b>H.261</b>	ITU standard for digitization of video for videoconferencing, uses N*64 kbit/s
<b>H.263</b>	Improved version of H.261, not fixed at N*64 kbit/s
<b>HDLC</b>	High Level Data Link protocol, a layer 2 protocol used over serial links
<b>HDTV</b>	High Definition Television, up to 1920 x 1080 pixels
<b>HTTP</b>	Hypertext Transfer Protocol
<b>ICP</b>	Internet Cache Protocol, used for inter-cache communication
<b>IETF</b>	Internet Engineering Task Force
<b>IGMP</b>	Internet Group Management Protocol, used between hosts and routers to announce that a host wishes to join a Multicast group
<b>IP</b>	Internet Protocol
<b>ISO</b>	International Organization for Standardization
<b>ITU</b>	International Telecommunication Union

<b>JPEG</b>	Joint Photographic Experts Group, also the name of a picture compression standard that the group created
<b>Layer-4 Switch</b>	Switch that switches based upon information in L4
<b>MBONE</b>	Multicast Backbone
<b>Microflow</b>	A stream or set of packets with same IP source, destination and port, thus considered as an aggregate
<b>MJPEG</b>	Motion JPEG, video compression codec utilizing frame-by-frame JPEG compression, e.g. spatial compression only
<b>MMS</b>	Microsoft Media Server protocol
<b>MOSPF</b>	Multicast OSPF, Multicast routing protocol
<b>MOV</b>	Windows proprietary movie format
<b>MP3</b>	MPEG-1 Layer III audio codec, suitable for music
<b>MPEG</b>	Moving Picture Experts Group, also the name of a group of video compression standards that the group created
<b>MPLS</b>	Multiprotocol Label Switching, a protocol used to combine layer-3 routing information with layer-2 switching to improve delivery
<b>MSBD</b>	Microsoft Media Stream Broadcast Distribution protocol
<b>Multicast</b>	Where the network replicates a stream of packets to multiple recipients that choose to receive this stream, the "opposite" of Unicast
<b>Multimedia</b>	The integrated presentation of text, graphics, sound and video
<b>NSC</b>	Microsoft Netshow Channel File, a descriptor of a Multicast channel used with Microsoft Media Services
<b>Nyquist Theorem</b>	A theorem that states the highest frequency that can be accurately represented by a digital sample is one-half of the sampling rate
<b>OC-12</b>	A SONET rate at 622 Mbit/s
<b>OC-48</b>	A SONET rate at 2.488 Gbit/s
<b>OC-192</b>	A SONET rate at 9.952 Gbit/s
<b>PIM</b>	Protocol Independent Multicast routing protocol
<b>PNA</b>	Progressive Networks Audio (Old RealNetworks streaming protocol)
<b>Policing</b>	The process that forces network traffic to conform to an absolute bitrate
<b>POS</b>	Packet Over Sonet
<b>PPP</b>	Point-to-Point Protocol, a layer 2 protocol used over serial links
<b>PQ</b>	Priority Queuing, the queuing mechanism where higher priority traffic gets absolute priority through a router or switch
<b>Proxy Cache</b>	A cache which a web client or media client points to directly
<b>QoS</b>	Quality-of-Service, specifies a guaranteed throughput level
<b>Queuing</b>	The algorithms employed by routers and switches to prioritize packets
<b>Ram</b>	RealNetworks metafile, lists clips to be played in sequence
<b>Rdist</b>	UNIX utility to maintain identical copies of files over multiple hosts
<b>RDT</b>	RealNetworks proprietary packet protocol
<b>Redirection</b>	The method of pointing a client to the most "relevant" server of a cluster of servers with identical content
<b>Robocopy</b>	NT utility to distribute copies of files over multiple hosts
<b>Rsync</b>	Newer version of Rdist

<b>RTCP</b>	RTP Control Protocol
<b>RTP</b>	Real-time Transport Protocol
<b>RTSP</b>	Real-Time Streaming Protocol
<b>Sampling</b>	The method where a digital sample of an analogue signal is taken at a certain interval
<b>Shaping</b>	The method where network devices force traffic to conform to a maximum bitrate by delaying packets
<b>SMIL</b>	Synchronized Multimedia Integration Language, an XML-like language used to describe complex multimedia presentations
<b>SONET</b>	Synchronous Optical Networks, a layer 2 protocol used over optical serial links
<b>Splitting</b>	The method where a media server can replicate a live stream originating at another media server
<b>Streaming</b>	The method where media data is fed to the client whilst it is being viewed. A small amount of data is buffered at the client to ensure smooth playback
<b>SureStream</b>	RealNetworks proprietary technology that stores multiple bitrates in the same file, and serves the most appropriate bandwidth to the client depending on requested and available bandwidth
<b>S-Video</b>	Super-Video, the technology for transmitting video in two separate components, color and brightness
<b>TCP</b>	Transmission Control Protocol, used with IP with guarantee of packet delivery
<b>ToS</b>	Type-of-Service, field in the IP header describing requested service for packet
<b>UDP</b>	User Datagram Protocol, used with IP but has no error recovery services. Usually used for streaming media
<b>Unicast</b>	A one-to-one session between a client and server, the "opposite" of Multicast
<b>VBR</b>	Variable Bit-Rate
<b>VIP</b>	Virtual IP address, the method where multiple hosts have the same IP address
<b>VIV</b>	Vivo video format
<b>WAV</b>	Windows Audio format
<b>WCCP</b>	Web Cache Control Protocol, Cisco proprietary protocol used to transparently redirect content requests from clients to cache
<b>WFQ</b>	Weighted Fair Queuing
<b>WPAD</b>	Web Proxy Auto Discovery
<b>WRED</b>	Weighted Random Early Detection, the method where a router or switch discards packets based upon policies before congestion occurs
<b>XML</b>	Extensible Markup Language, possible successor to HTML
<b>Y/C</b>	Video cable standard to carry Luminance (Y) and Chrominance (C) as separate signals