TDTS06: Computer Networks

Instructor: Niklas Carlsson
Email: niklas.carlsson@liu.se

Notes derived from "*Computer Networking: A Top Down Approach*", by Jim Kurose and Keith Ross, Addison-Wesley.

The slides are adapted and modified based on slides from the book's companion Web site, as well as modified slides by Anirban Mahanti and Carey Williamson.

Scalable Content Delivery
Motivation

- Use of Internet for content delivery is massive … and becoming more so (e.g., majority of all IP traffic is video content)
- Variety of approaches: HTTP-based Adaptive Streaming (HAS), broadcast/multicast, batching, replication/caching (e.g. CDNs), P2P, peer-assisted, …
- In these slides, we only provide a few high-level examples

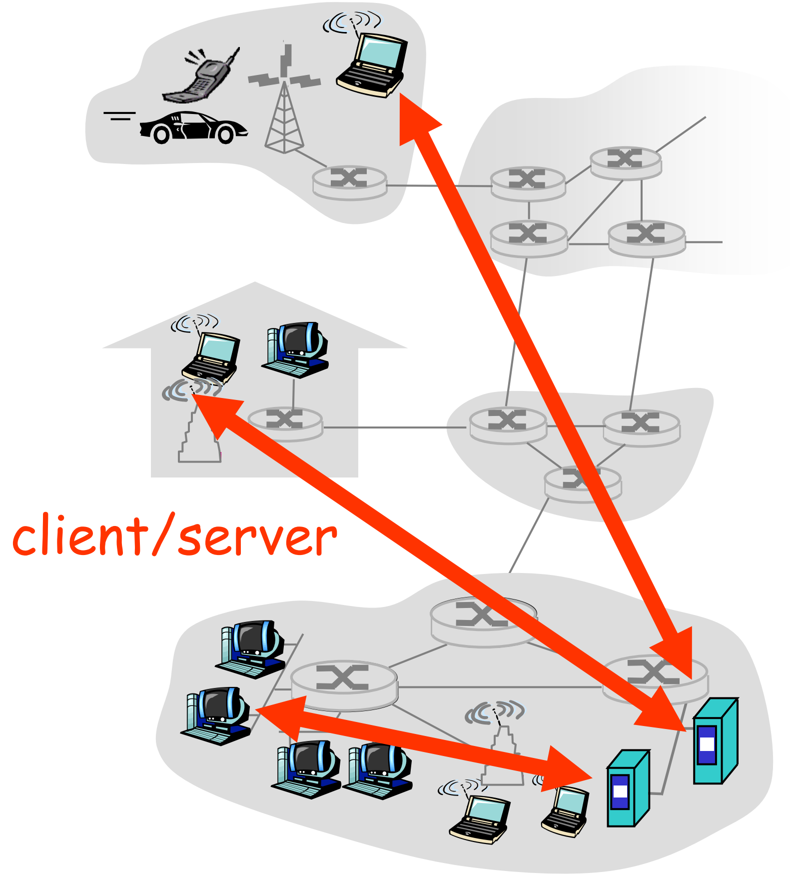
Why Study Multimedia Networking?

- Exciting and fun!
- Multimedia is everywhere
- Industry-relevant research topic
- Lots of open research problems

Service models

Client-server architecture

Client/server model has well-defined roles.



server:

- always-on host
- permanent IP address
- server farms for scaling

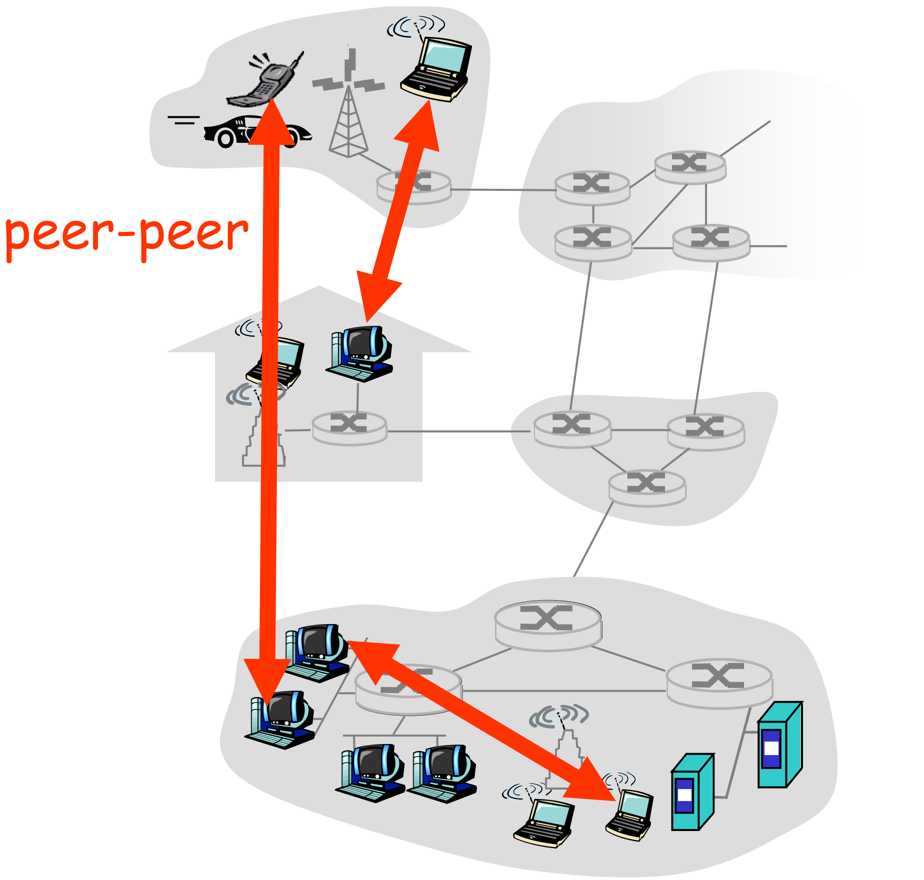
clients:

- communicate with server
- may be intermittently connected
- may have dynamic IP addresses
- do not communicate directly with each other

Pure P2P architecture

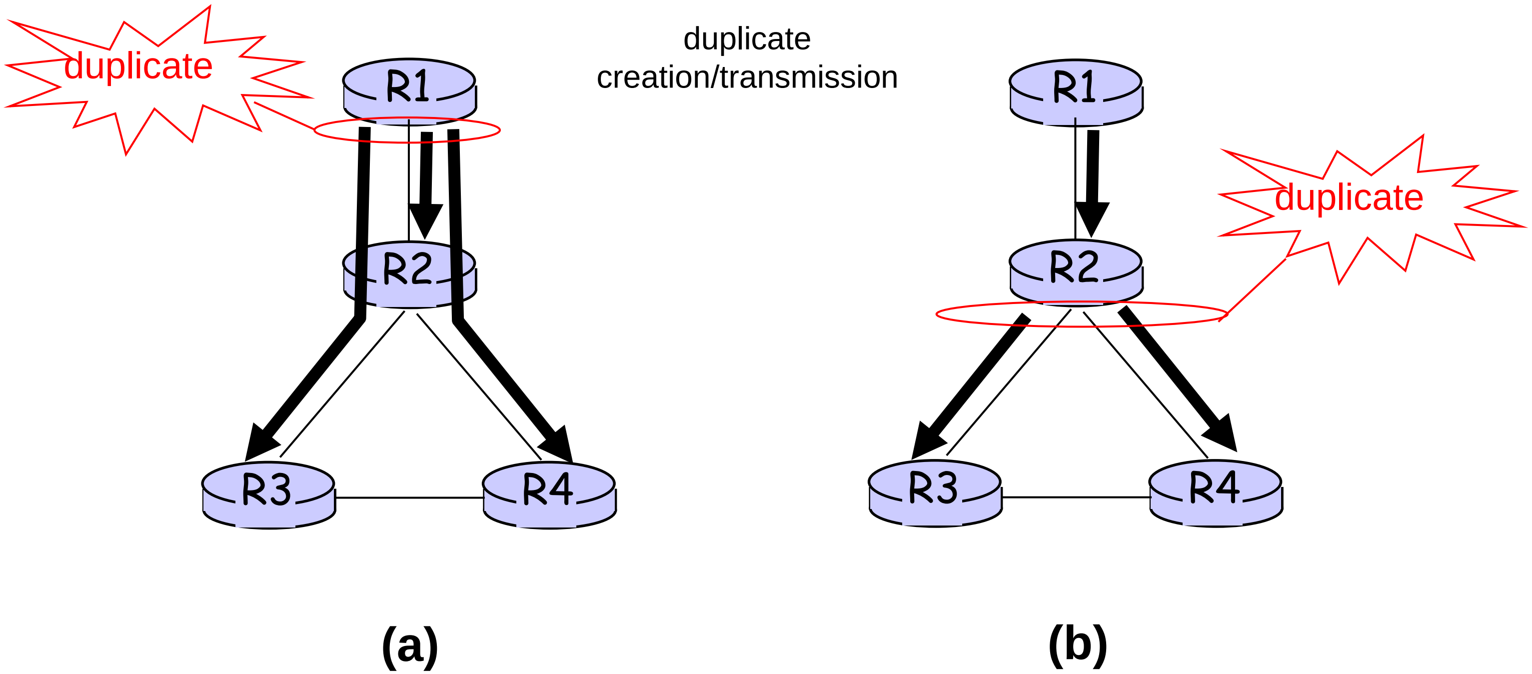
No fixed clients or servers: Each host can act as both client and server at any time

- *no* always-on server
- arbitrary end systems directly communicate
- peers are intermittently connected and change IP addresses



Additional Multimedia Support

Multicast/Broadcast



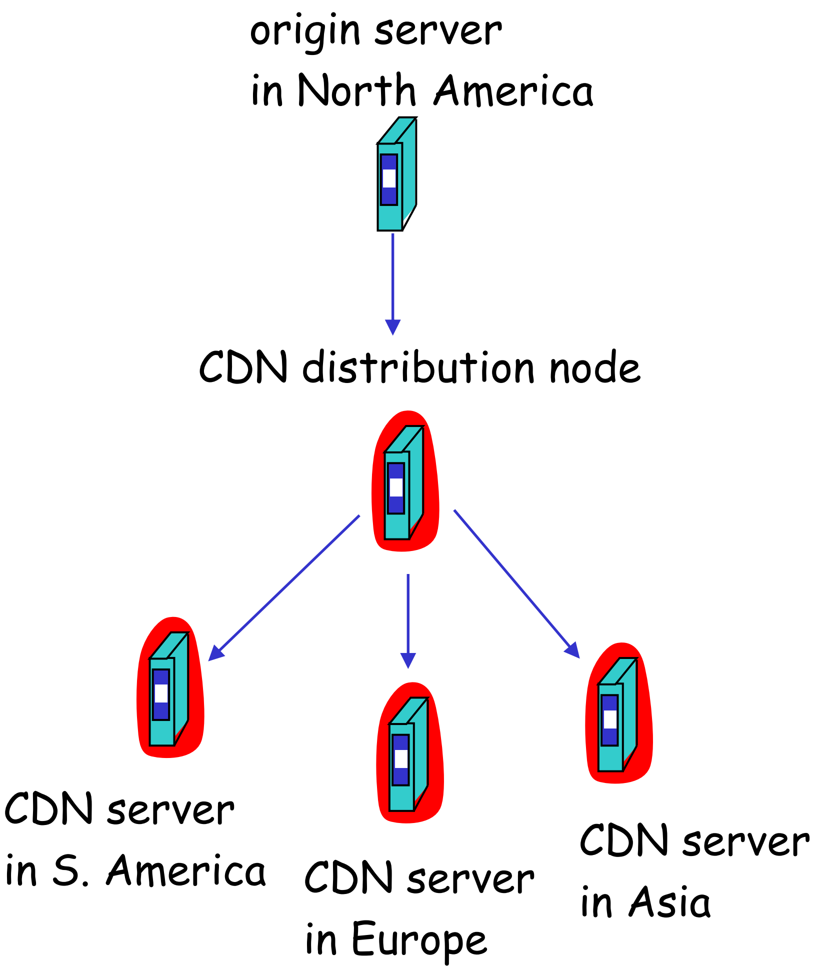
Source-duplication versus in-network duplication.
(a) source duplication, (b) in-network duplication

Also, application-layer multicast …

Content distribution networks (CDNs)

Content replication

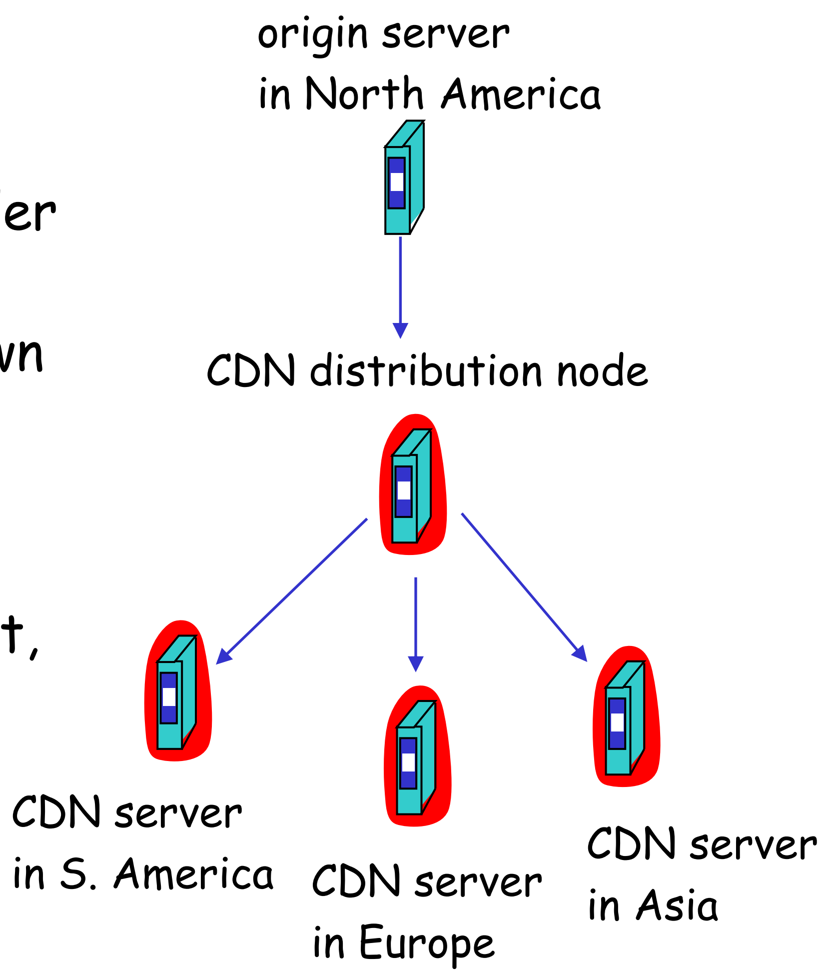
- replicate content at hundreds of servers throughout Internet (often in edge/access network)
- content "close" to user reduce impairments (loss, delay) of sending content over long paths



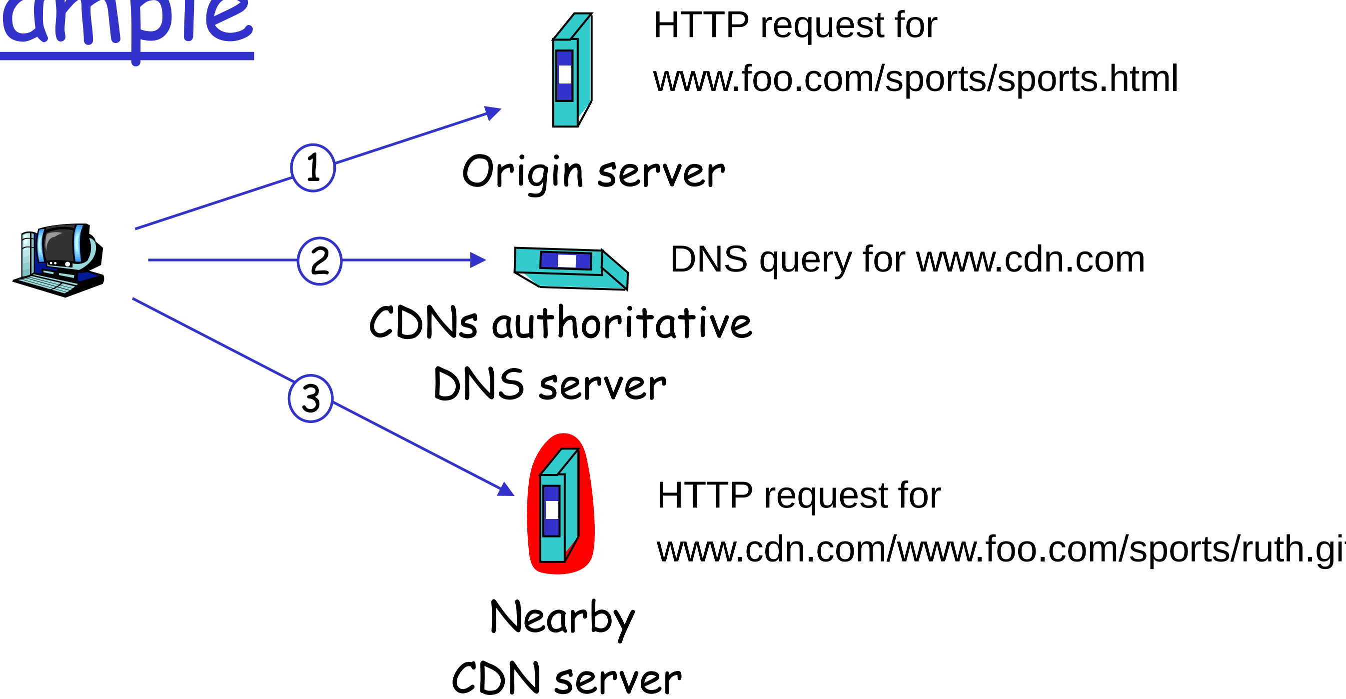
Content distribution networks (CDNs)

Content replication

- CDN (e.g., Akamai, Limewire) customer is the content provider (e.g., CNN)
- Other companies build their own CDN (e.g., Google)
- CDN replicates customers' content in CDN servers.
- When provider updates content, CDN updates servers



CDN example



origin server (www.foo.com)

- distributes HTML
- replaces:

  http://www.foo.com/sports.ruth.gif

  with

  http://www.cdn.com/www.foo.com/sports/ruth.gif

CDN company (cdn.com)

- distributes gif files
- uses its authoritative DNS server to route redirect requests

More about CDNs

routing requests

- CDN creates a "map", indicating distances from leaf ISPs and CDN nodes
- when query arrives at authoritative DNS server:
  - server determines ISP from which query originates
  - uses "map" to determine best CDN server
- CDN nodes create application-layer overlay network

Multimedia Networking

Principles

- Classify multimedia applications
- Identify the network services the apps need
- Making the best of "best effort" service
- Mechanisms for providing QoS

Protocols and Architectures

- Specific protocols for best effort delivery
- Architectures for QoS

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Multimedia Networking Applications

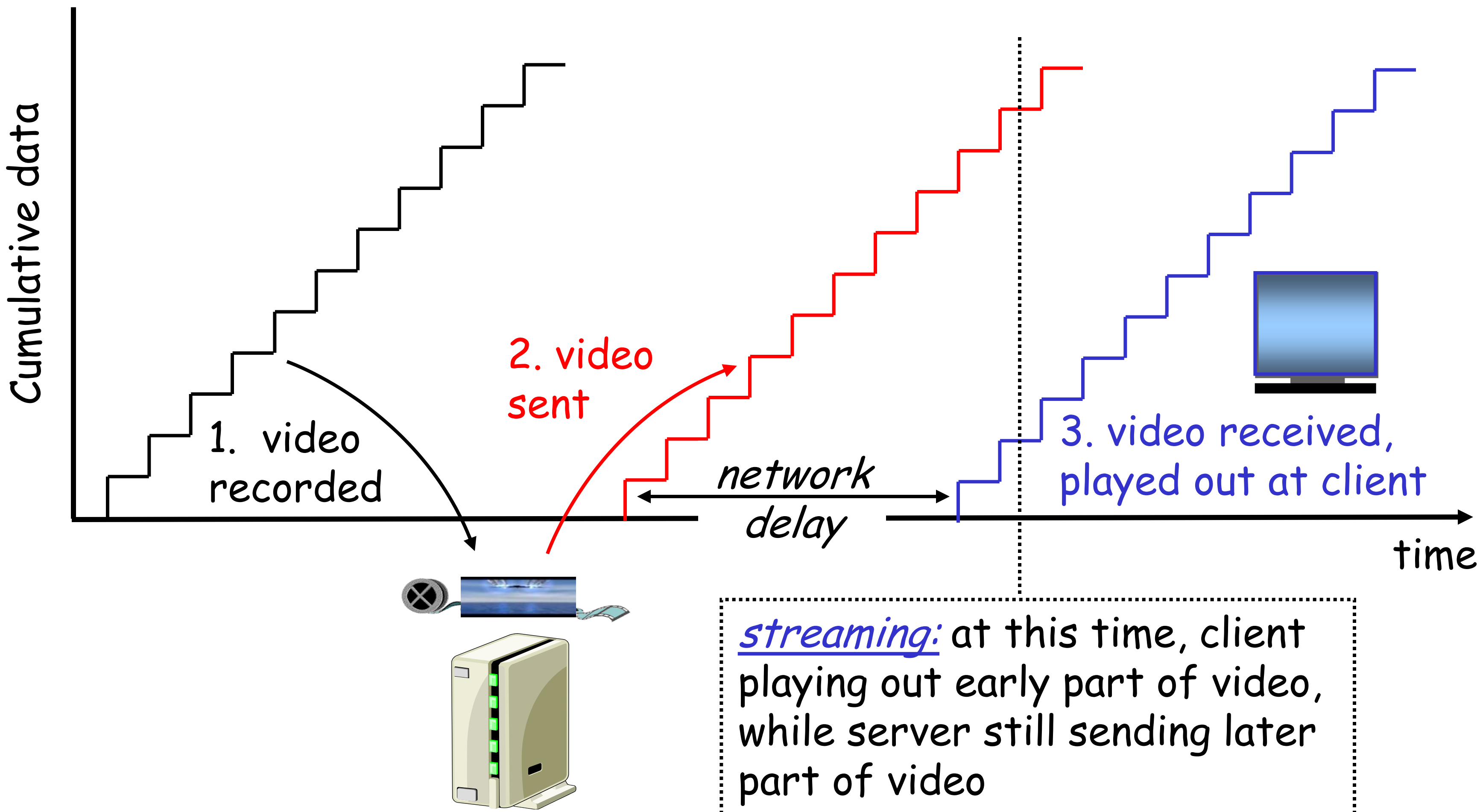
Classes of MM applications:

Multimedia Networking Applications

Classes of MM applications:

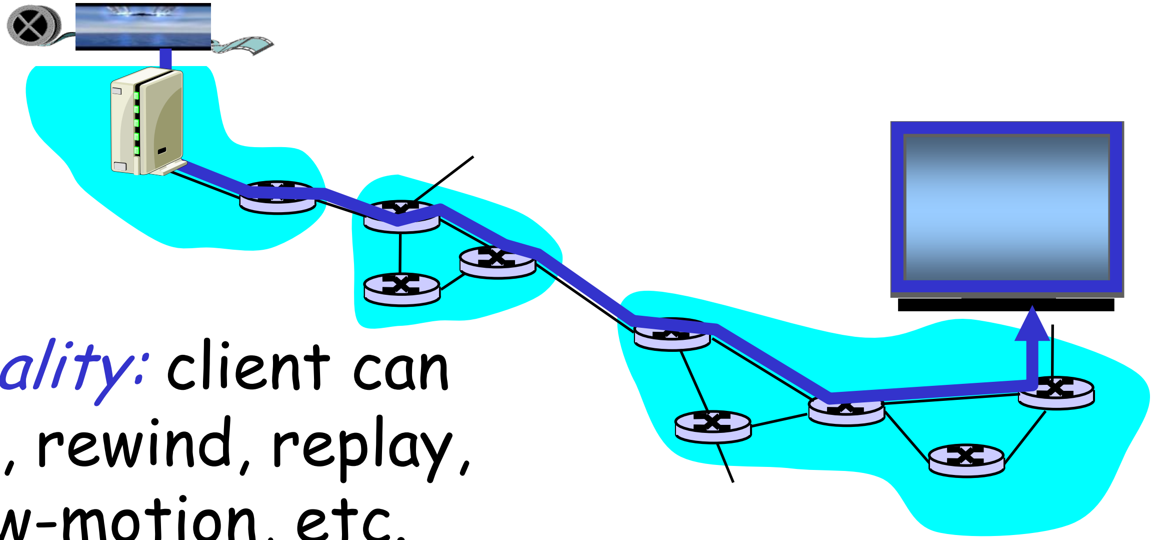
1) Streaming stored audio and video
2) Streaming live audio and video
3) Real-time interactive audio and video

Streaming Stored Multimedia (1/2)



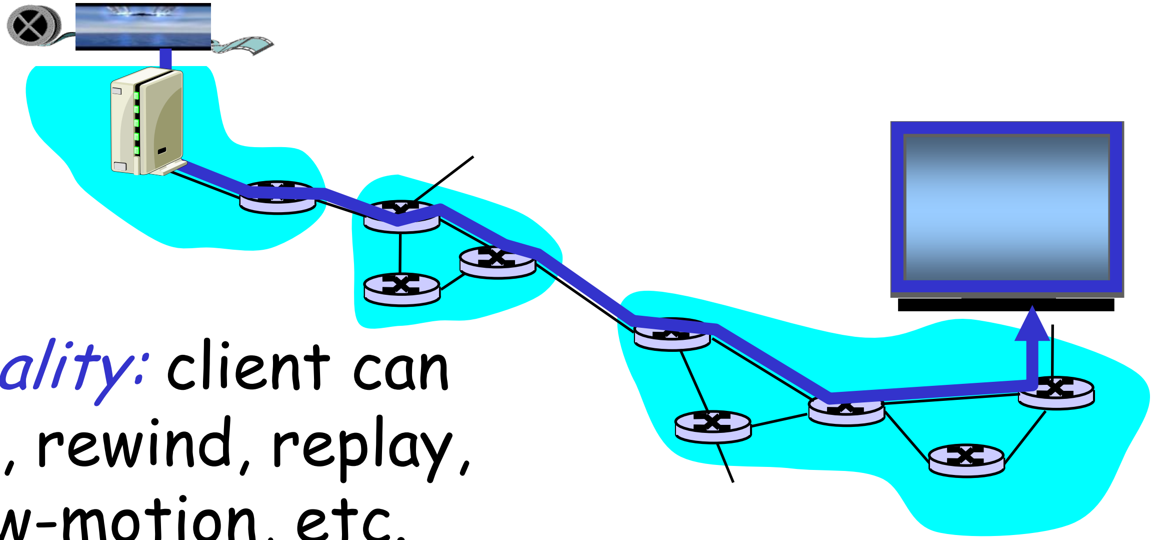
streaming: at this time, client playing out early part of video, while server still sending later part of video

Streaming Stored Multimedia (2/2)



- *VCR-like functionality:* client can start, stop, pause, rewind, replay, fast-forward, slow-motion, etc.

Streaming Stored Multimedia (2/2)



- *VCR-like functionality:* client can start, stop, pause, rewind, replay, fast-forward, slow-motion, etc.
  - 10 sec initial delay OK
  - 1-2 sec until command effect OK
  - need a separate control protocol?
- timing constraint for data that is yet to be transmitted: must arrive in time for playback

Streaming Live Multimedia

Examples:

- Internet radio talk show
- Live sporting event

Streaming Live Multimedia

Examples:

- Internet radio talk show
- Live sporting event

Streaming

- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity

- fast-forward is not possible
- rewind and pause possible!

Interactive, Real-time Multimedia

- applications: IP telephony, video conference, distributed interactive worlds

Interactive, Real-time Multimedia

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- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-layer (packetization) and network delays
    - higher delays noticeable, impair interactivity
- session initialization
  - callee must advertise its IP address, port number, frame rate, encoding algorithms

Multimedia Networking Applications

Fundamental characteristics:

Multimedia Networking Applications

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- Typically **delay-sensitive**
  - end-to-end delay
  - delay jitter

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- But **loss-tolerant**: infrequent losses cause minor transient glitches

- Unlike data apps, which are often delay-tolerant but loss-sensitive.

Multimedia Networking Applications

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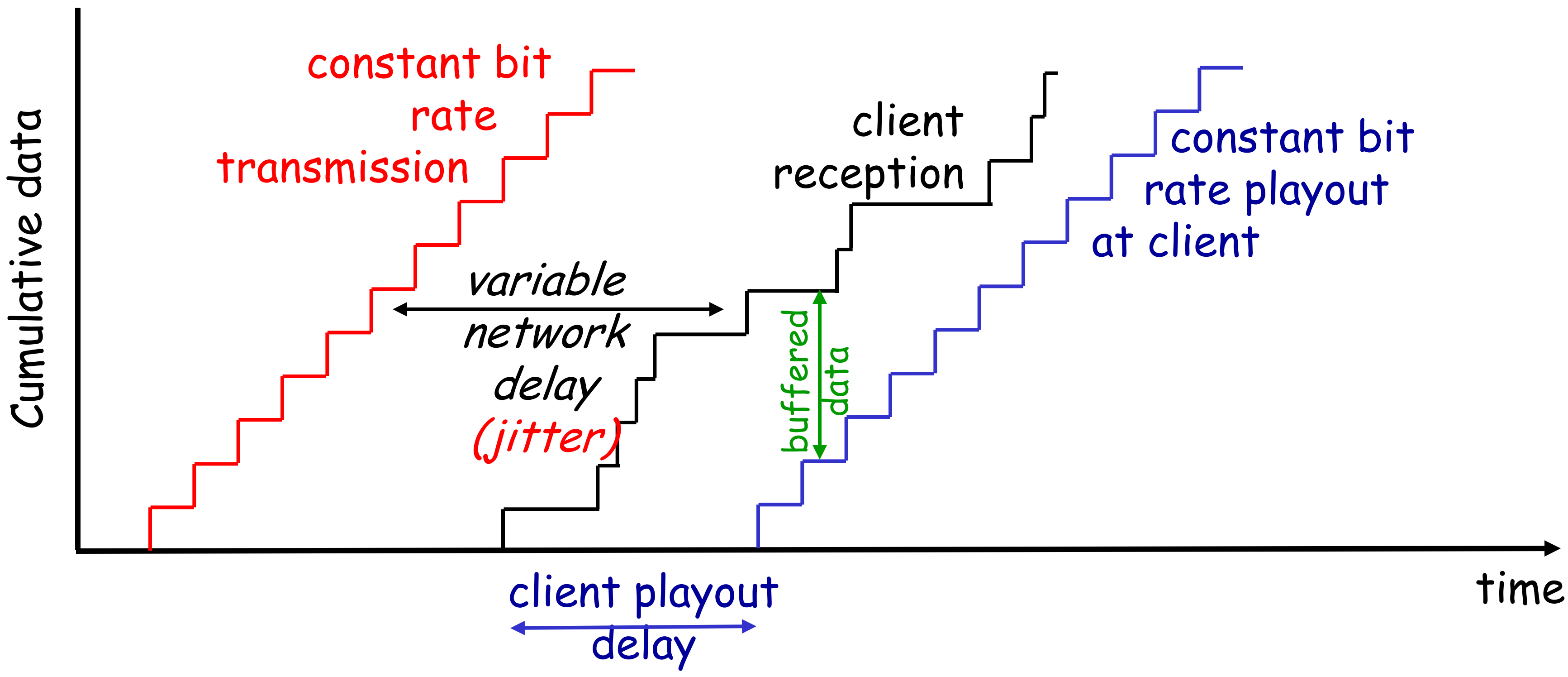
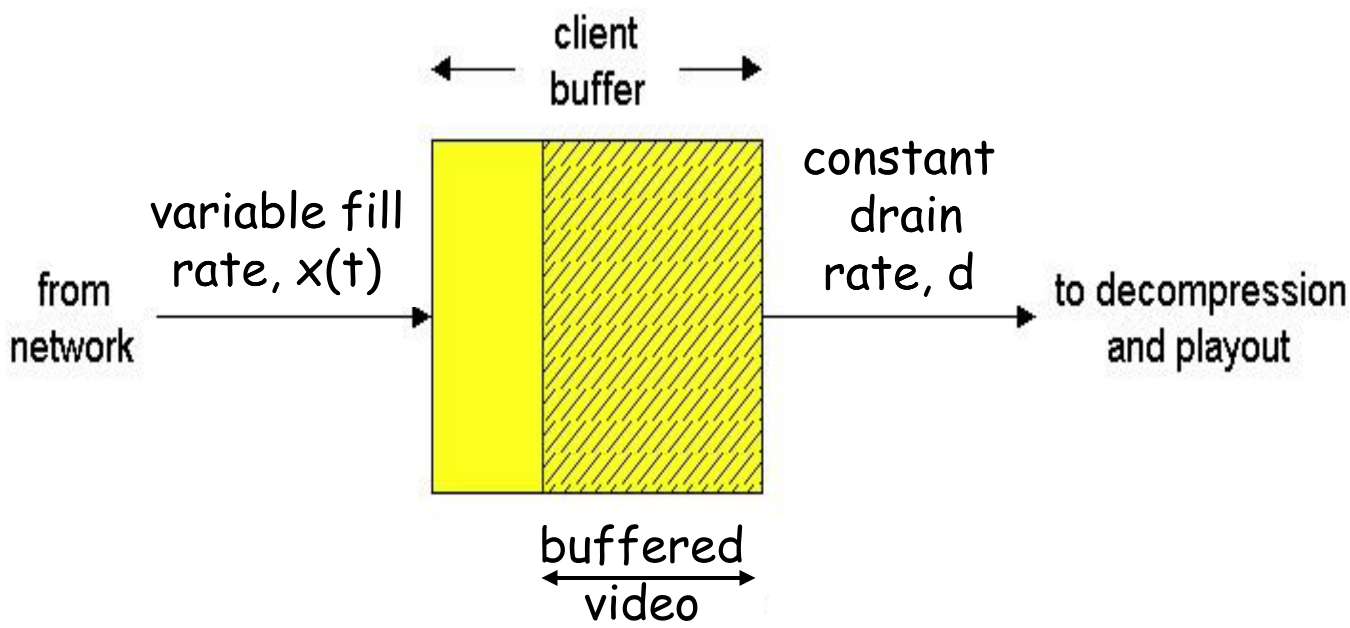
Jitter is the variability of packet delays within the same packet stream

Delay Jitter



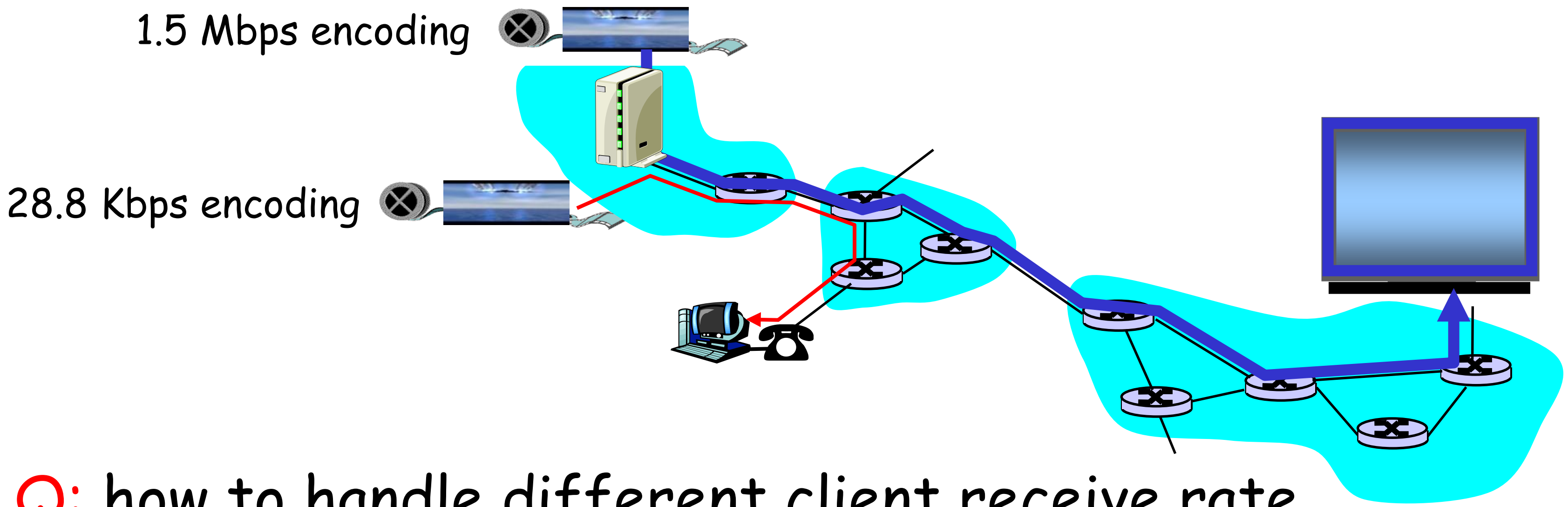
- consider end-to-end delays of two consecutive packets: difference can be more or less than 20 msec (transmission time difference)

Streaming Multimedia: Client Buffering



- Client-side buffering, playout delay compensate for network-added delay, delay jitter

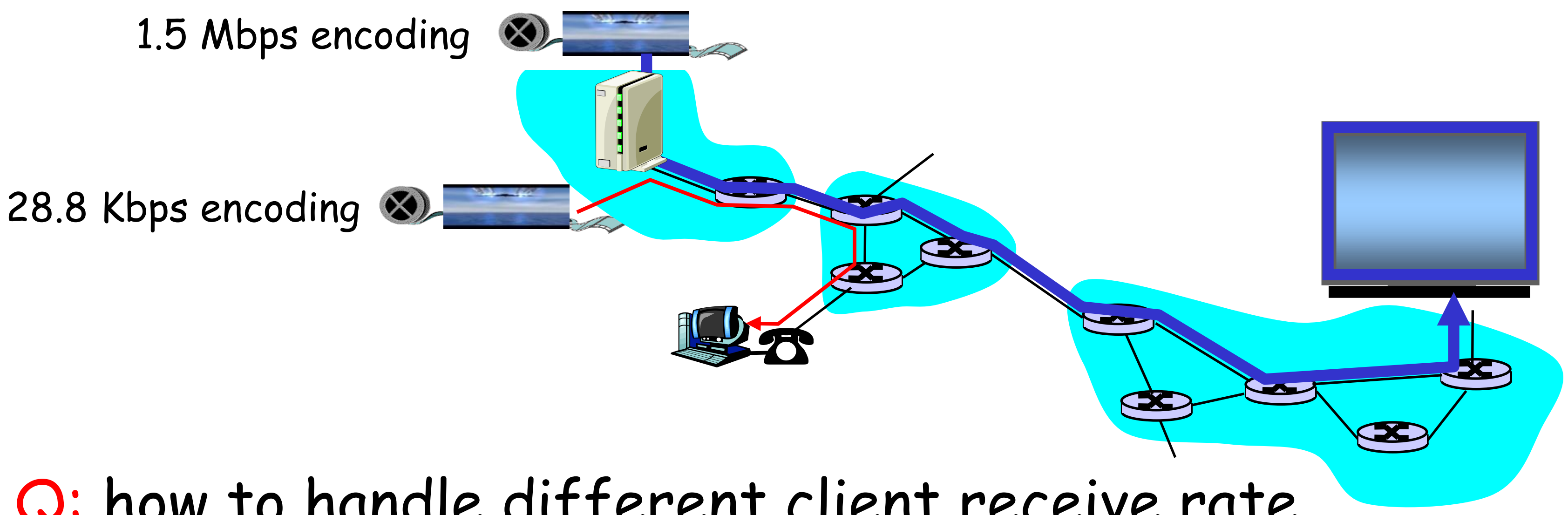
Streaming Multimedia: client rate(s)



Q: how to handle different client receive rate capabilities?

- 28.8 Kbps dialup
- 100 Mbps Ethernet

Streaming Multimedia: client rate(s)



Q: how to handle different client receive rate capabilities?
- 28.8 Kbps dialup
- 100 Mbps Ethernet

A1: server stores, transmits multiple copies of video, encoded at different rates

A2: layered and/or dynamically rate-based encoding

Consider first ...
Streaming Stored Multimedia

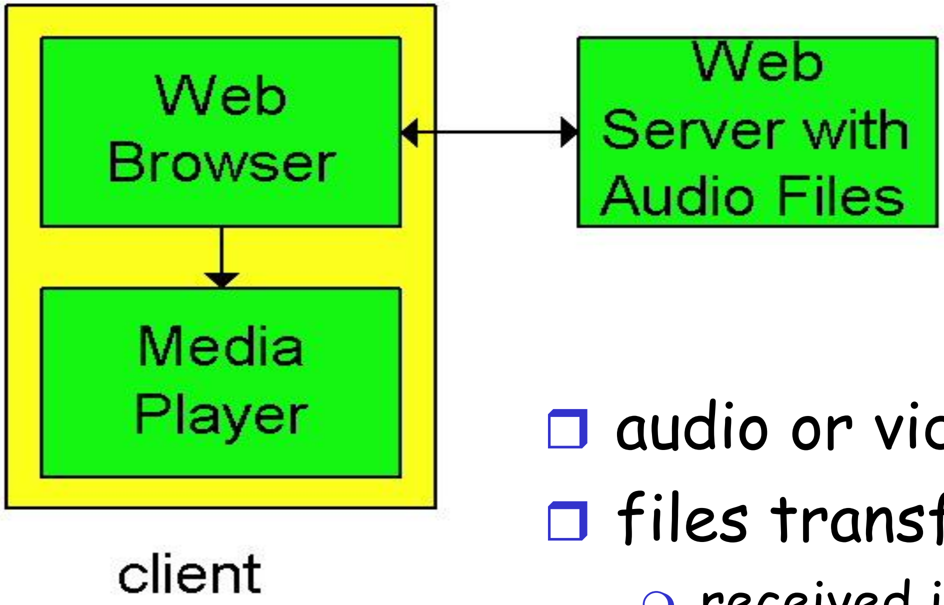
application-level streaming techniques for making the best out of best effort service:

- client-side buffering
- use of UDP versus TCP
- multiple encodings of multimedia

Media Player

- jitter removal
- decompression
- error concealment
- graphical user interface w/ controls for interactivity

Internet multimedia: simplest approach

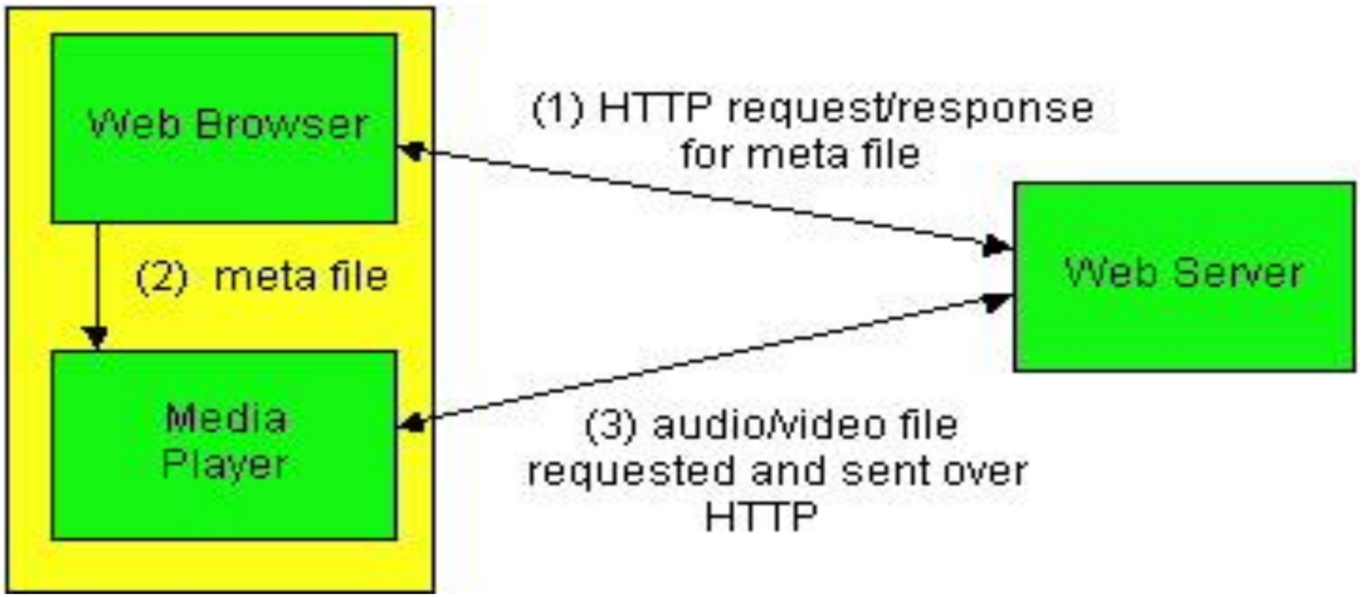


- audio or video stored in file
- files transferred as HTTP object
  - received in entirety at client
  - then passed to player

audio, video is downloaded, not streamed:

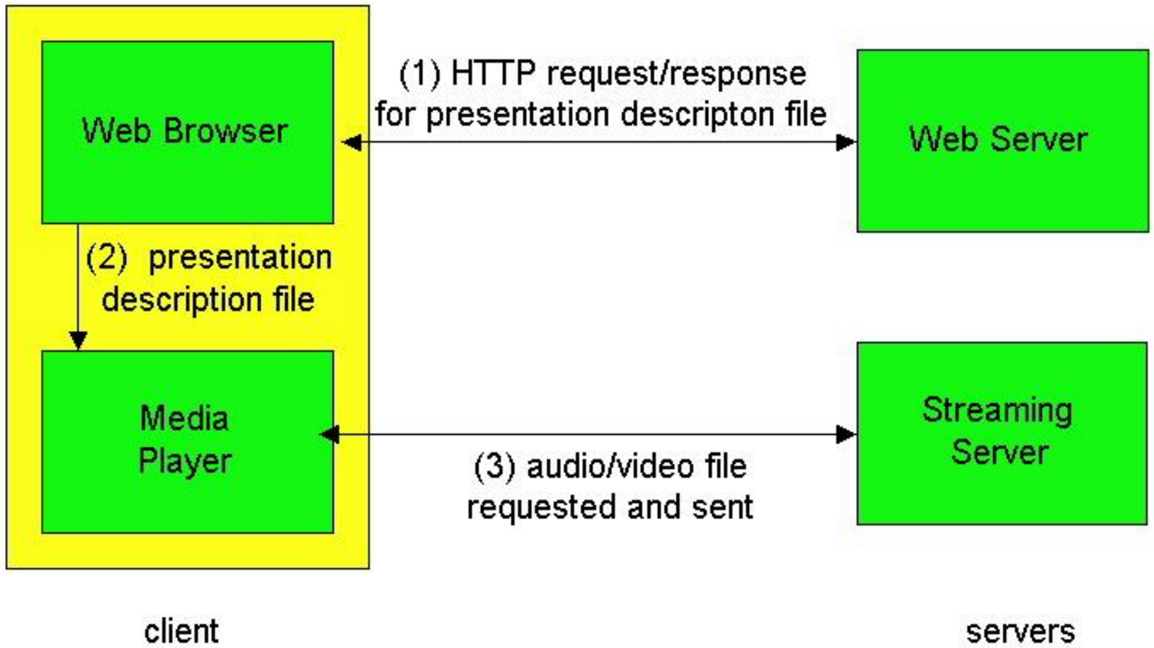
- long delays until playout, since no pipelining!

Progressive Download



- browser retrieves **metafile** using HTTP GET
- browser launches player, passing metafile to it
- media player contacts server directly
- server downloads audio/video to player

Streaming from a Streaming Server



- This architecture allows for non-HTTP protocol between server and media player
- Can also use UDP instead of TCP.

Streaming Multimedia: UDP or TCP?

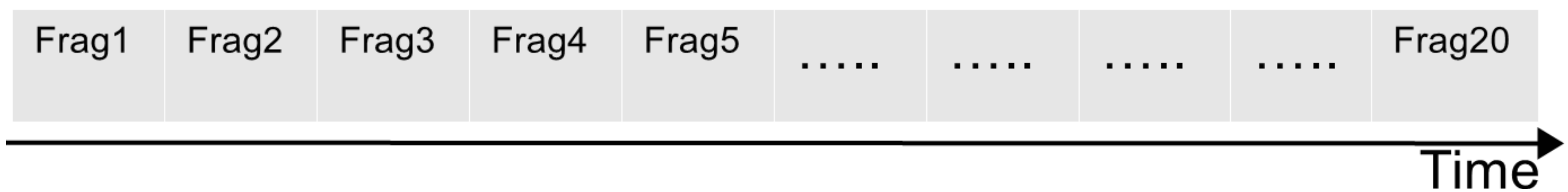
UDP

- server sends at rate appropriate for client (oblivious to network congestion !)
  - often send rate = encoding rate = constant rate
  - then, fill rate = constant rate - packet loss
- short playout delay (2-5 seconds) to compensate for network delay jitter
- error recover: time permitting

TCP

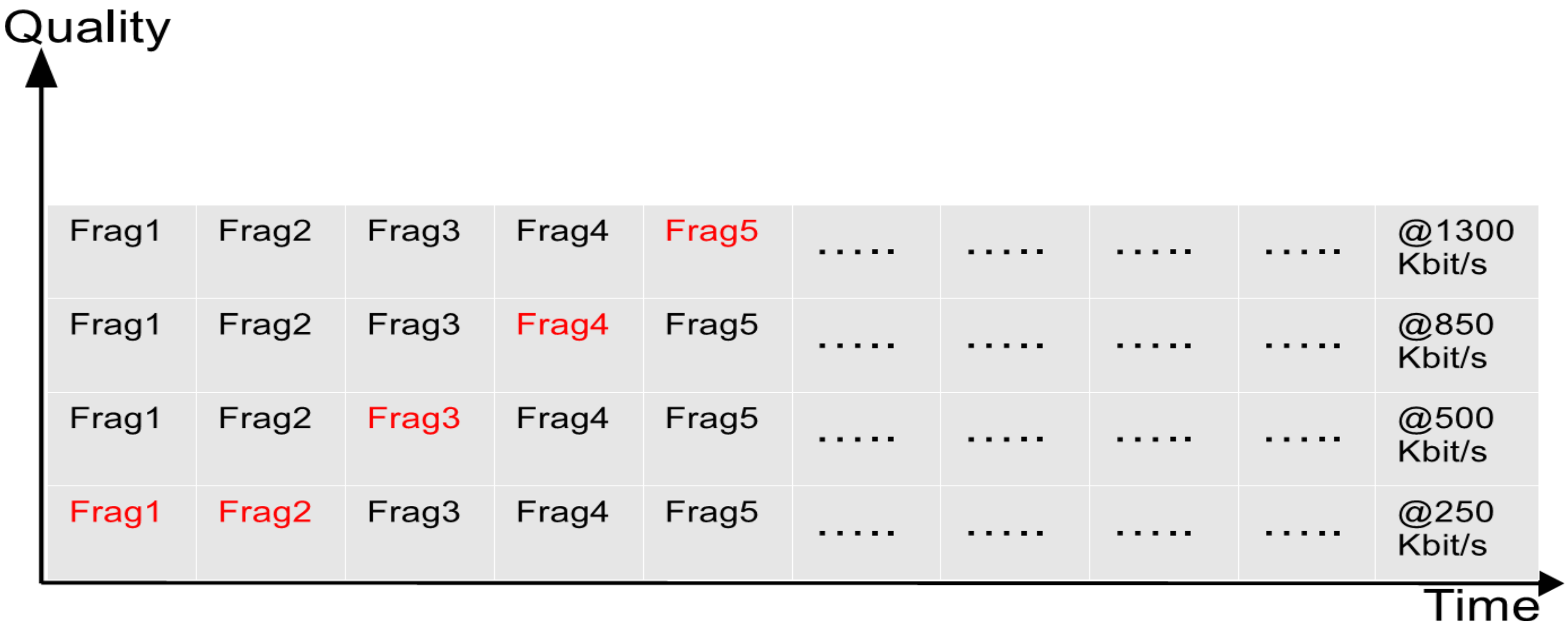
- send at maximum possible rate under TCP
- fill rate fluctuates due to TCP congestion control
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

HTTP-based streaming



- HTTP-based streaming
  - Allows easy caching, NAT/firewall traversal, etc.
  - Use of TCP provides natural bandwidth adaptation
  - Split into fragments, download sequentially
  - Some support for interactive VoD

HTTP-based adaptive streaming (HAS)

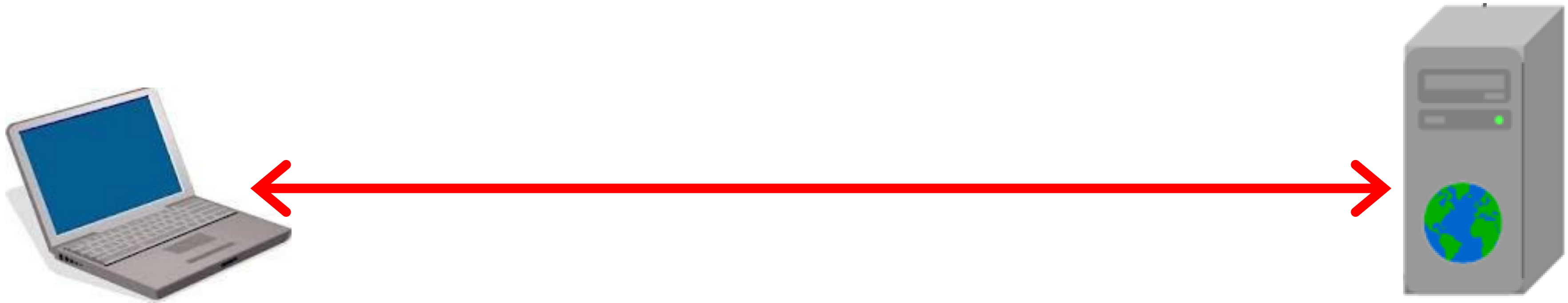


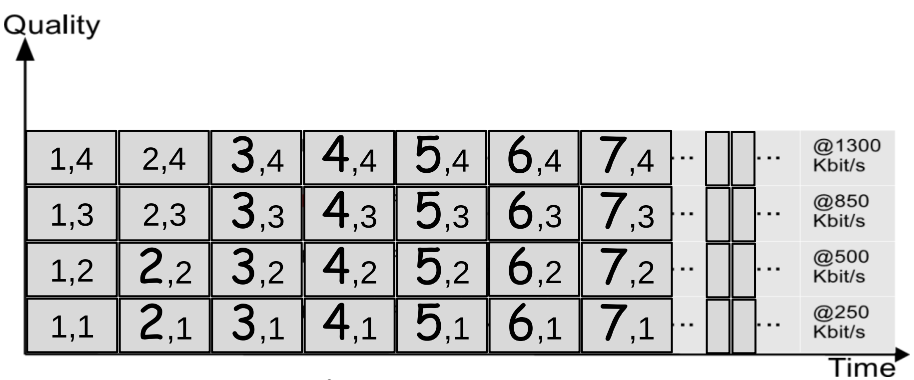
- HTTP-based adaptive streaming
  - Multiple encodings of each fragment (defined in manifest file)
  - Clients adapt quality encoding based on (buffer and network) conditions

Chunk-based streaming

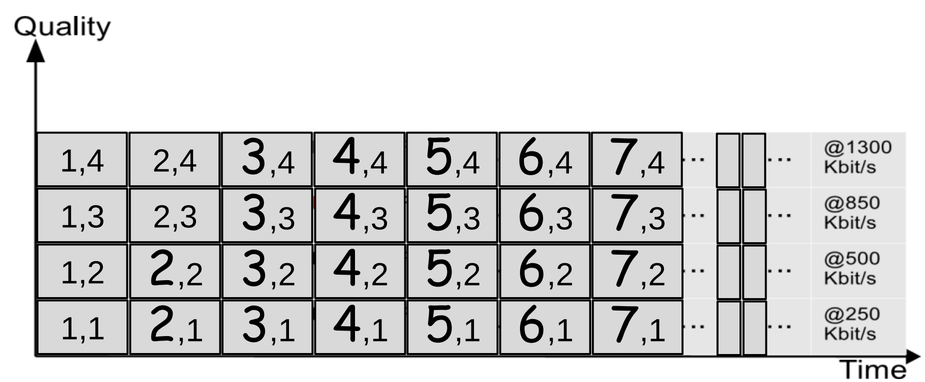
- Chunks begin with keyframe so independent of other chunks
- Playing chunks in sequence gives seamless video
- Hybrid of streaming and progressive download:
  - Stream-like: sequence of small chunks requested as needed
  - Progressive download-like: HTTP transfer mechanism, stateless servers

Example



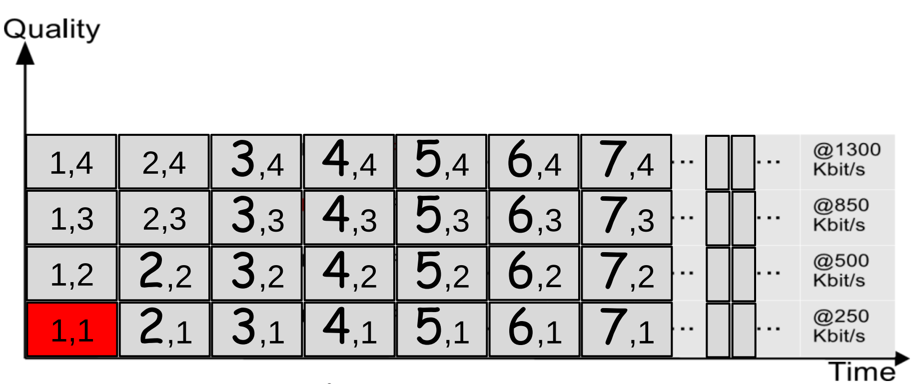


Client



Server

Example



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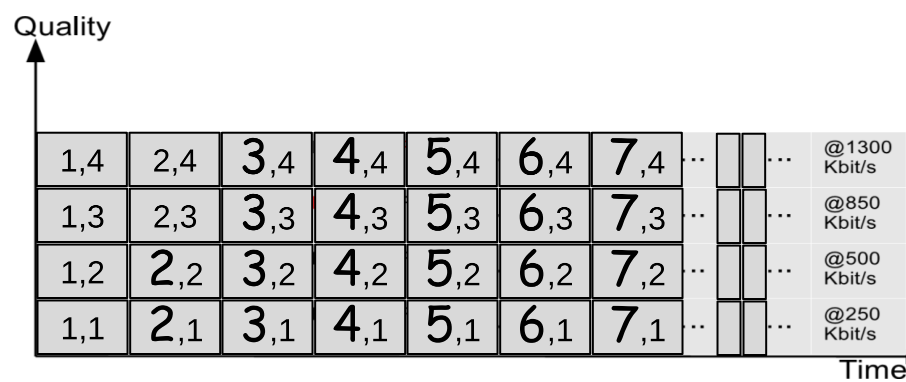

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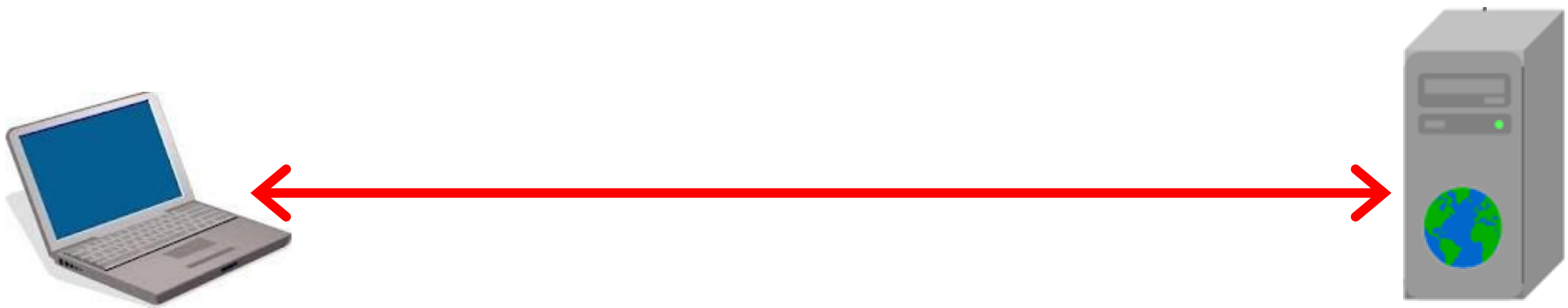

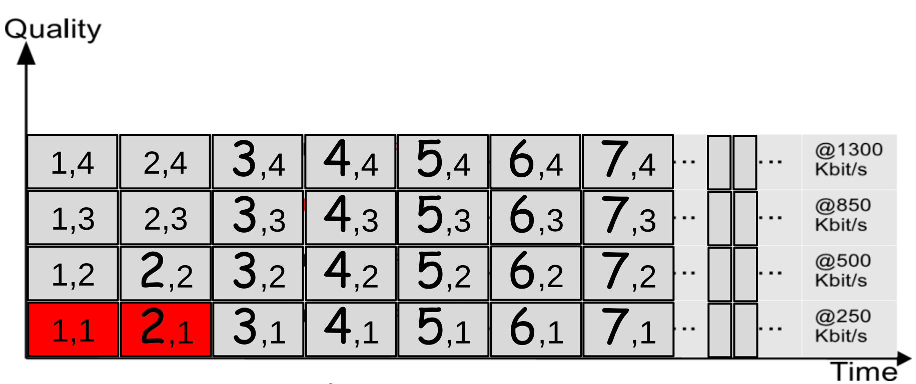

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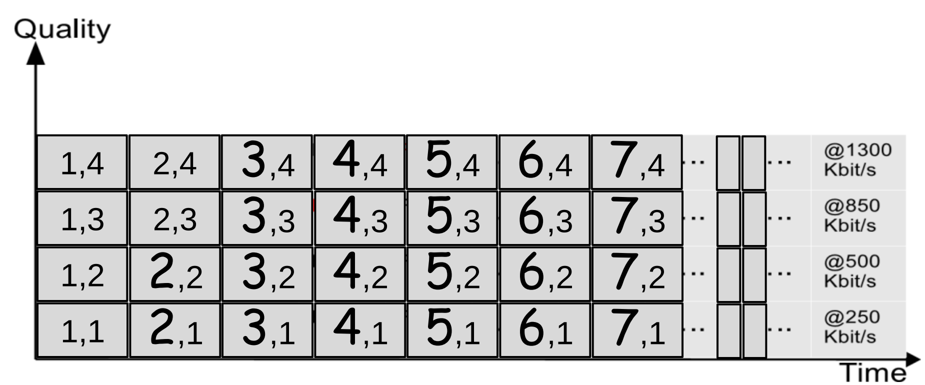

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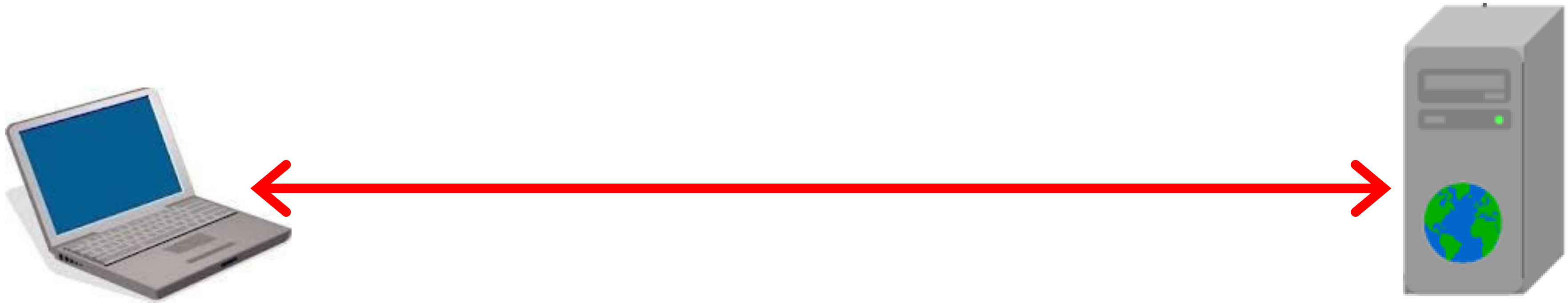

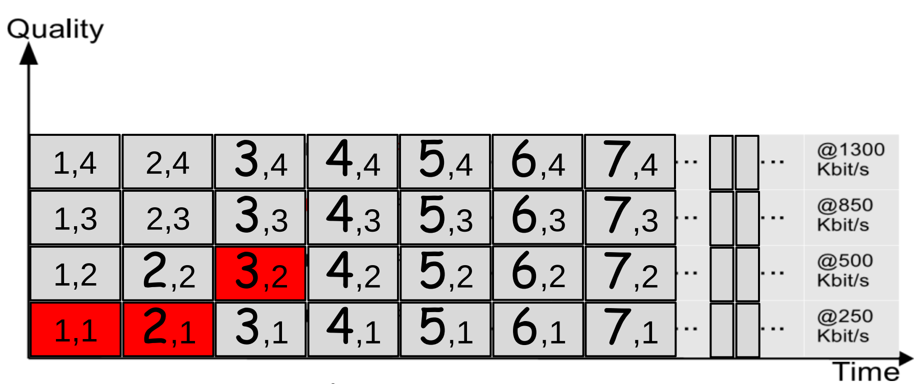

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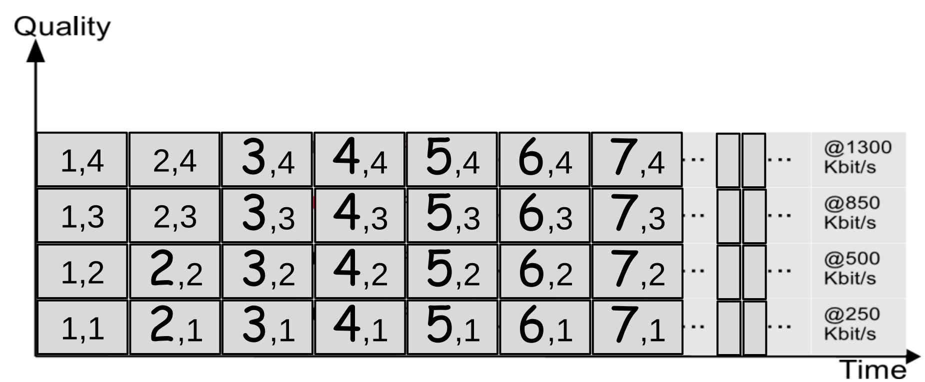

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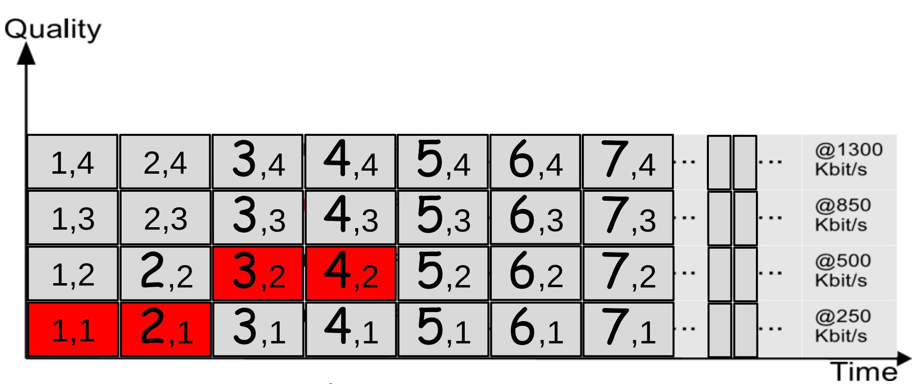

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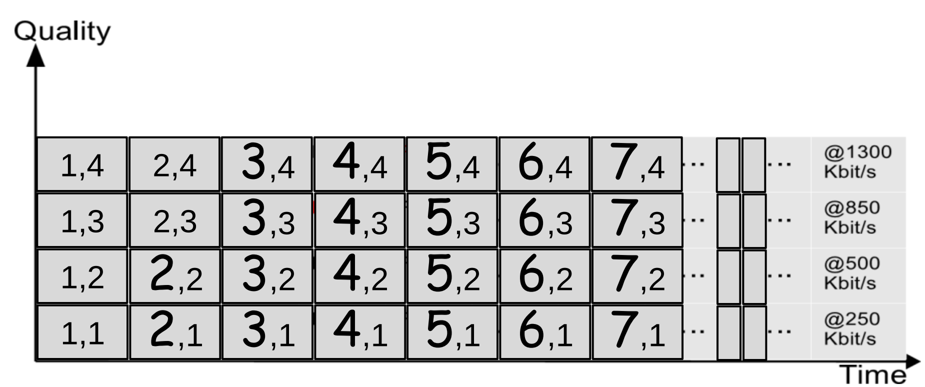

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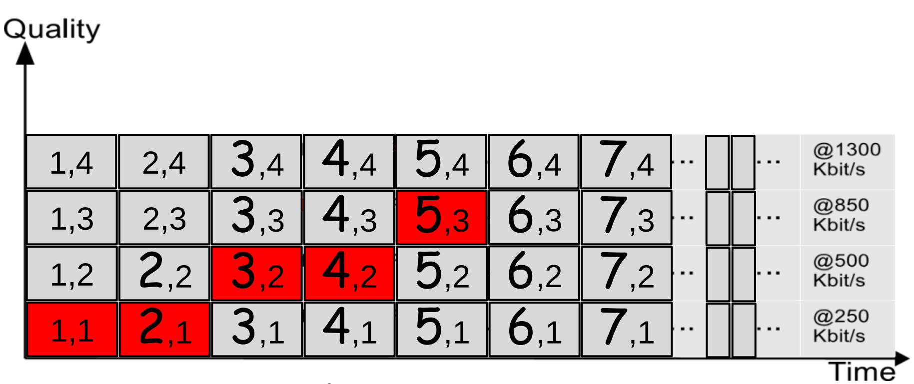

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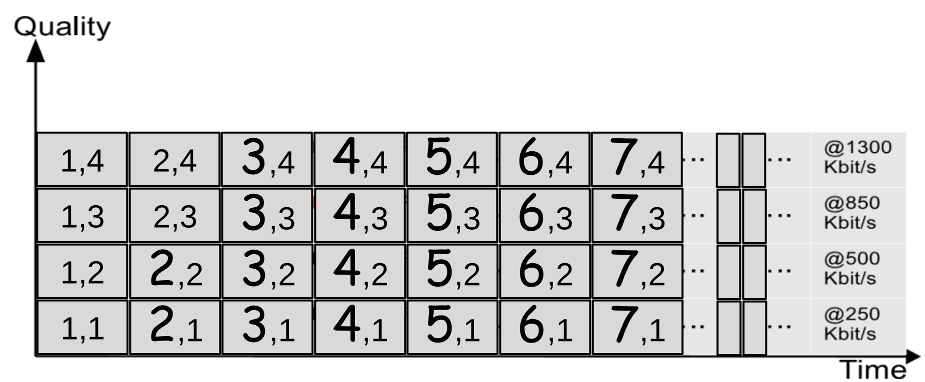

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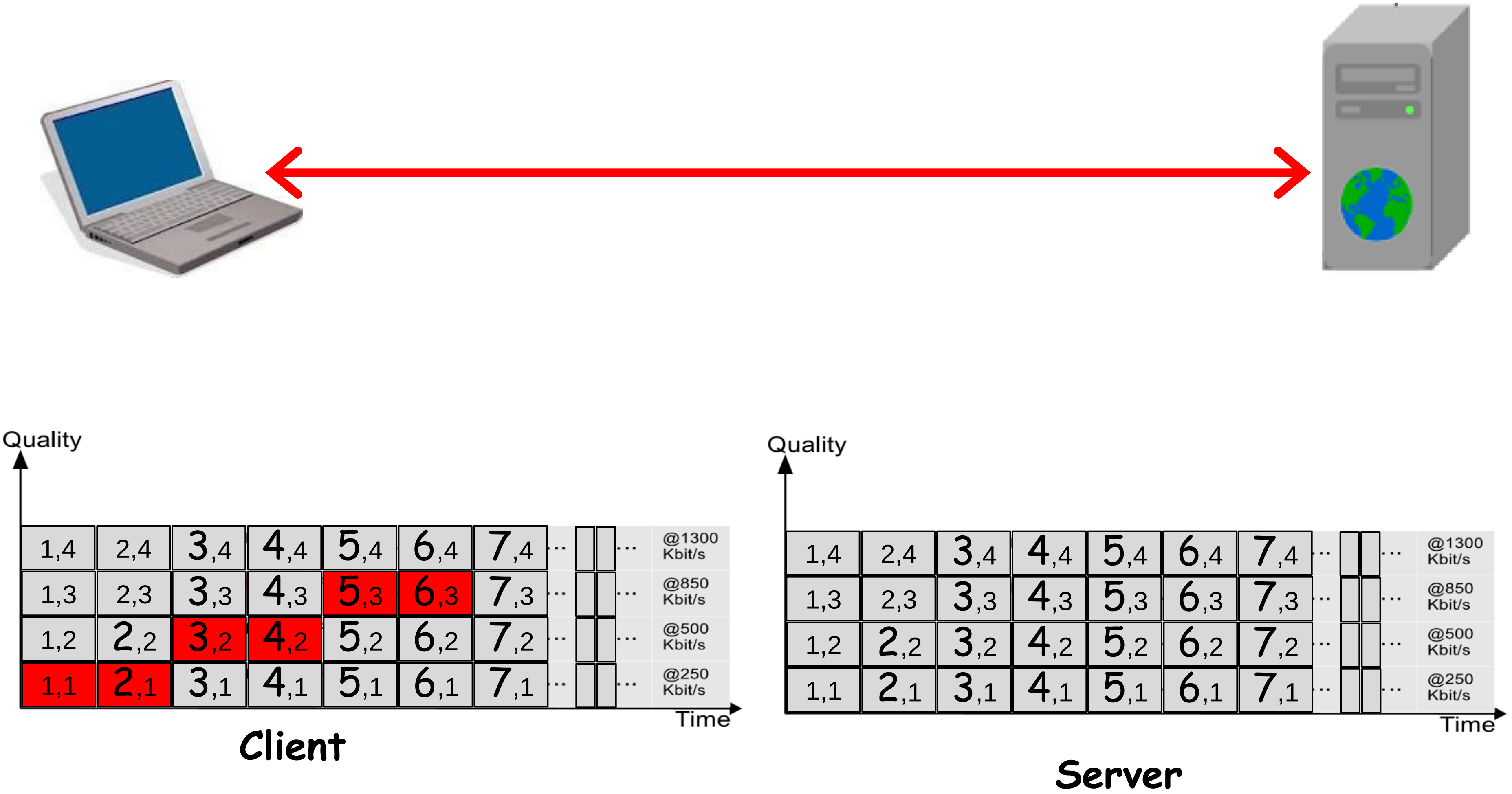
Quality
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@1300 Kbit/s
@850 Kbit/s
@500 Kbit/s
@250 Kbit/s
Time
Client
Quality
1,4
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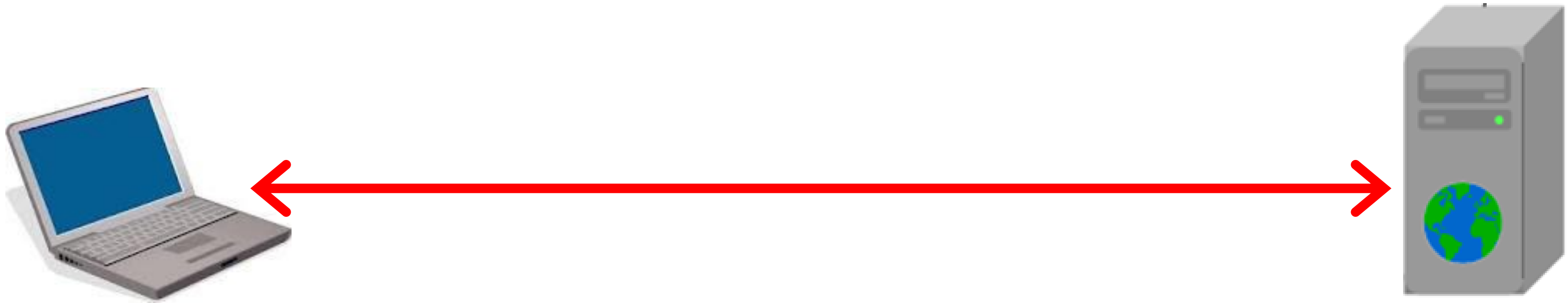

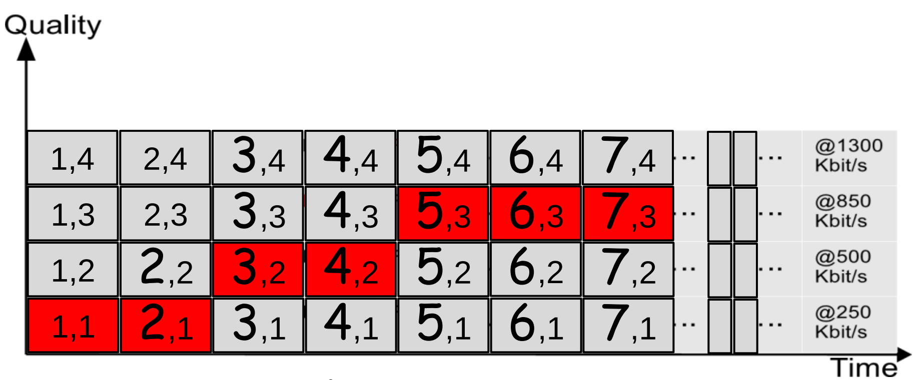

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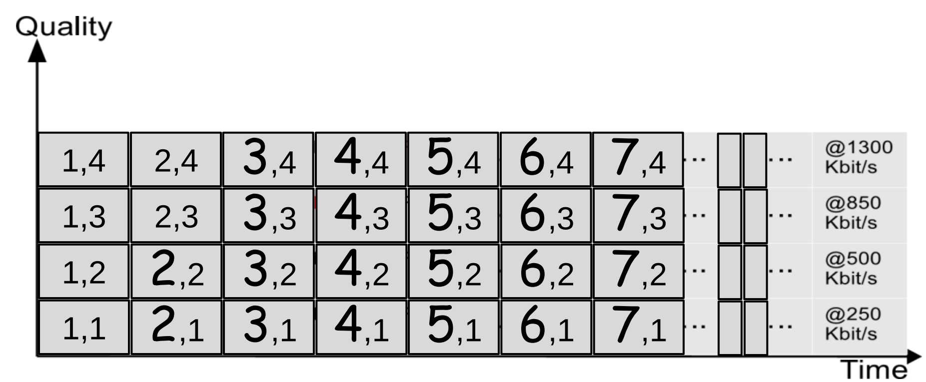

Server

HTTP-based Adaptive Streaming (HAS)

- Other terms for similar concepts: Adaptive Streaming, Smooth Streaming, HTTP Chunking
- Probably most important is *return to stateless server and TCP basis* of 1st generation
- Actually a series of small progressive downloads of chunks (or range requests)
- No standard protocol ...

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- No standard protocol ...
  - Apple HLS: HTTP Live Streaming
  - Microsoft IIS Smooth Streaming: part of Silverlight
  - Adobe: Flash Dynamic Streaming
  - DASH: Dynamic Adaptive Streaming over HTTP

Example players

	Player	Container	Type	Open Source
Microsoft Silverlight	Microsoft Smooth Streaming	Silverlight	Chunk	✗
NETFLIX	Netflix player	Silverlight	Range	✗
	Apple HLS	QuickTime	Chunk	✗
	Adobe HDS	Flash	Chunk	✓

Example: HAS and proxy

Clients' want

- High playback quality
- Small stall times
- Few buffer interruptions
- Few quality switches

HTTP Streaming (2)

- Adaptation:
  - Encode video at different levels of quality/bandwidth
  - Client can adapt by requesting different sized chunks
  - Chunks of different bit rates must be synchronized: All encodings have the same chunk boundaries and all chunks start with keyframes, so you can make smooth splices to chunks of higher or lower bit rates
- Evaluation:
  - Easy to deploy: it's just HTTP, caches/proxies/CDN all work
  - Fast startup by downloading lowest quality/smallest chunk
  - Bitrate switching is seamless
  - Many small files
- Chunks can be
  - Independent files -- many files to manage for one movie
  - Stored in single file container -- client or server must be able to access chunks, e.g. using range requests from client.

Examples: Netflix & Silverlight

- Netflix servers allow users to search & select movies
- Netflix manages accounts and login
- Movie represented as an XML encoded "manifest" file with URL for each copy of the movie:
  - Multiple bitrates
  - Multiple CDNs (preference given in manifest)
- Microsoft Silverlight DRM manages access to decryption key for movie data
- CDNs do no encryption or decryption, just deliver content via HTTP.
- Clients use "**`Range-bytes=`**" in HTTP header to stream the movie in chunks.

Slides from: V. Krishnamoorthi et al. "Helping Hand or Hidden Hurdle: Proxy-assisted HTTP-based Adaptive Streaming Performance", Proc. IEEE MASCOTS, 2013

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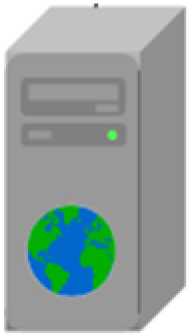

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HAS is increasingly responsible for larger traffic volumes
… proxies to reduce traffic??

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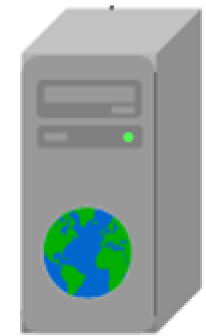

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Network providers' want

- High QoE of customers/clients
- Low bandwidth usage
- High hit rate

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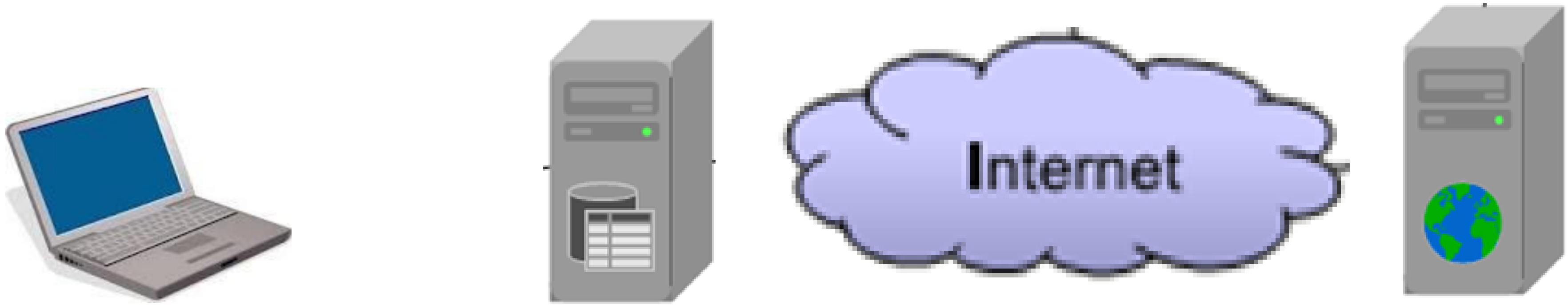

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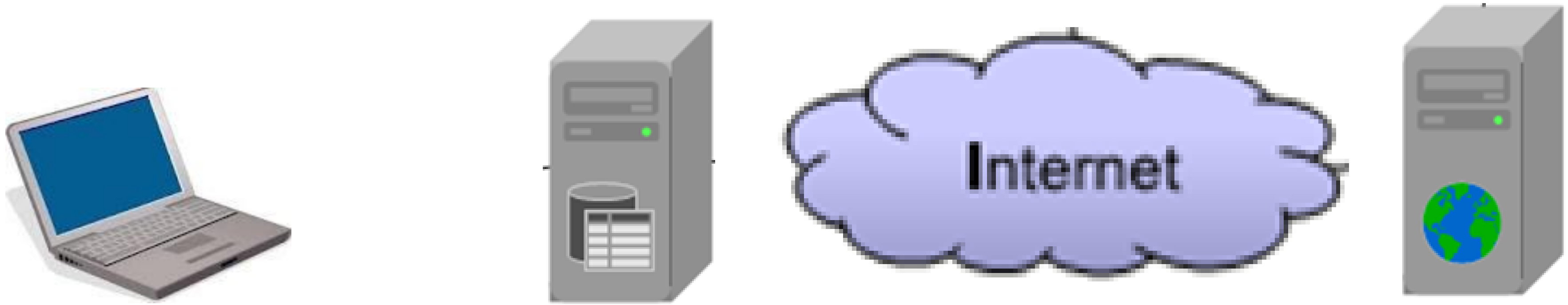
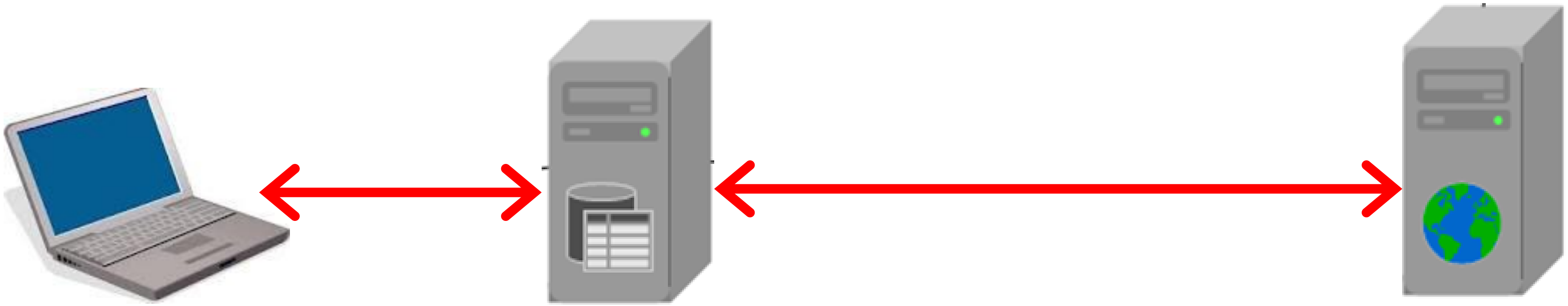


Proxy example …

Example: HAS and proxy



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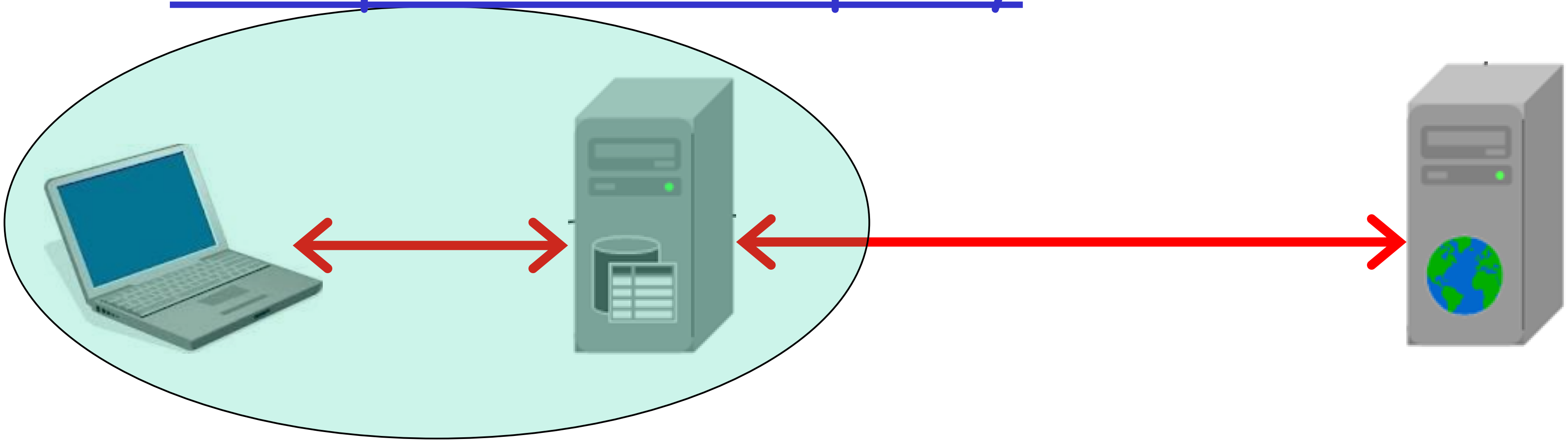

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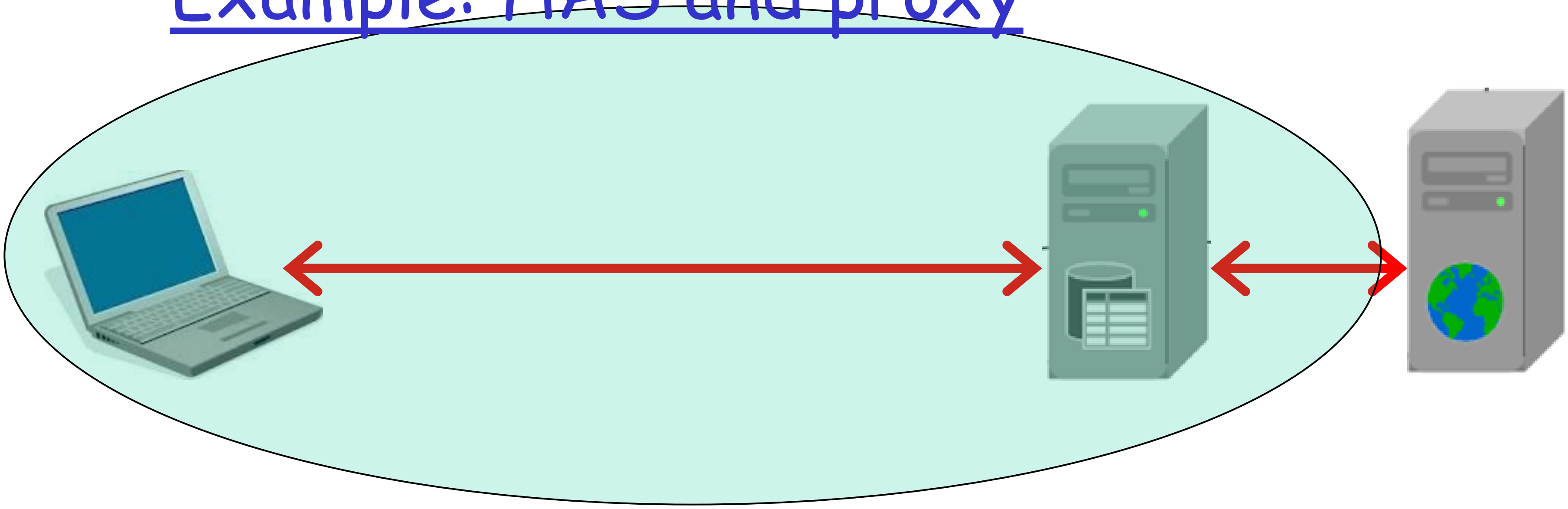
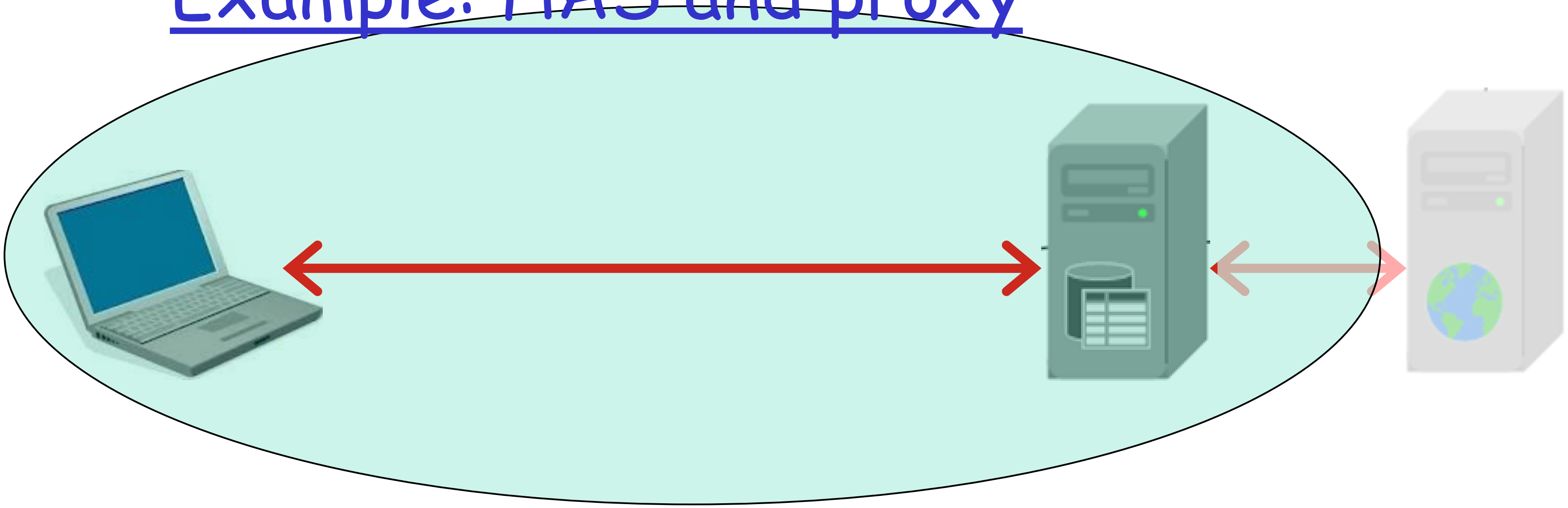
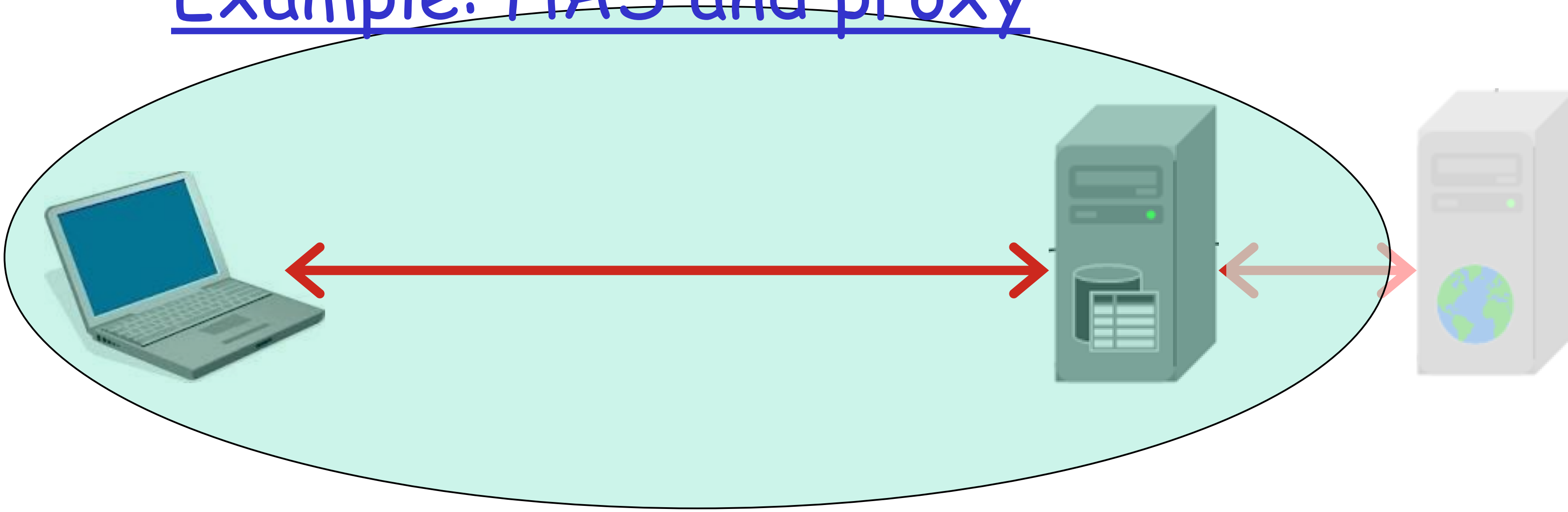
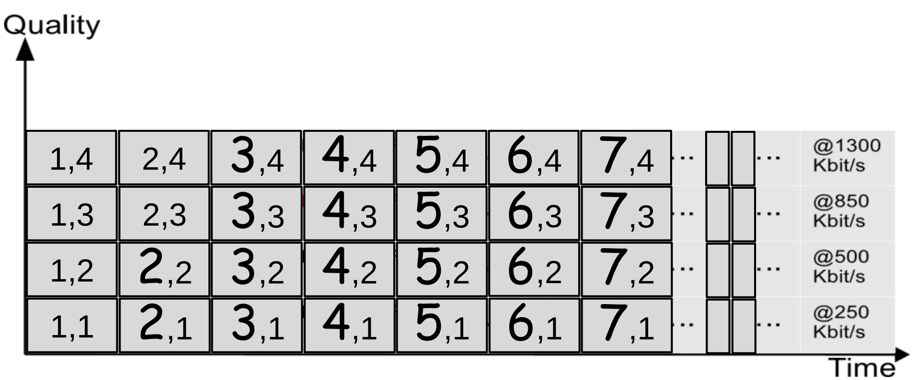


Example: HAS and proxy

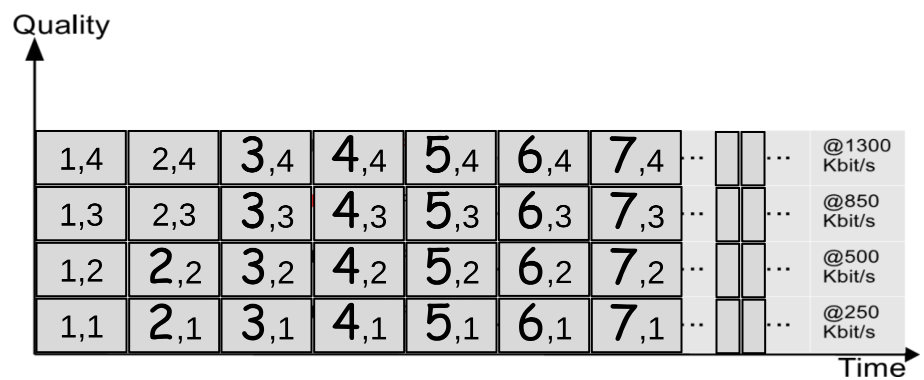


Example: HAS and proxy



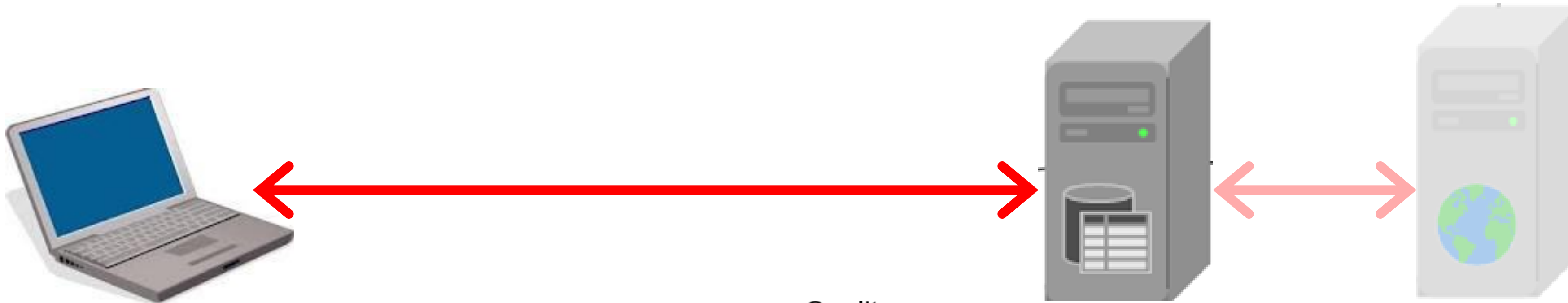


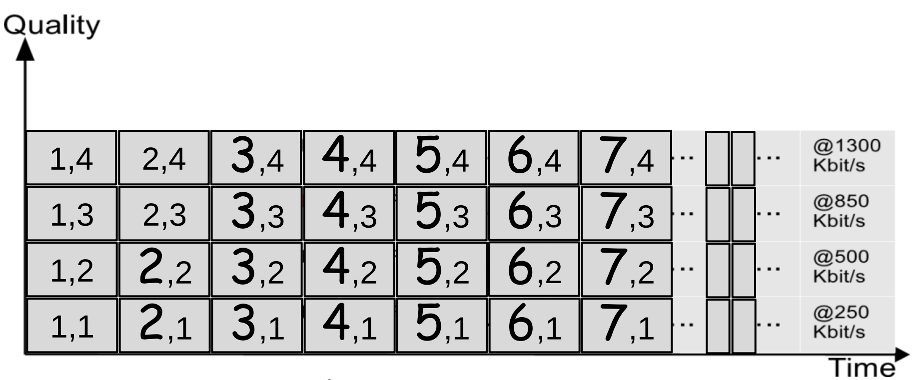
Client



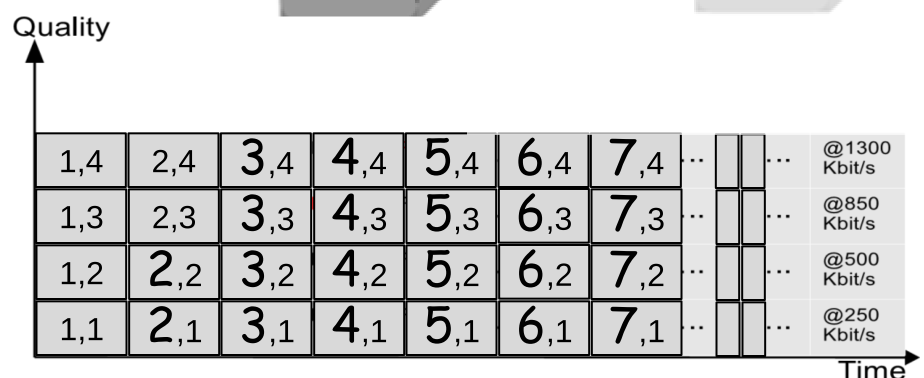
Proxy

Example: HAS and proxy

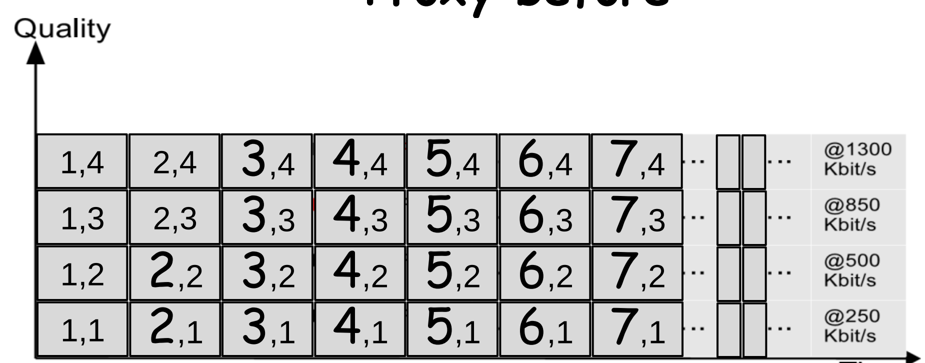




Client 1

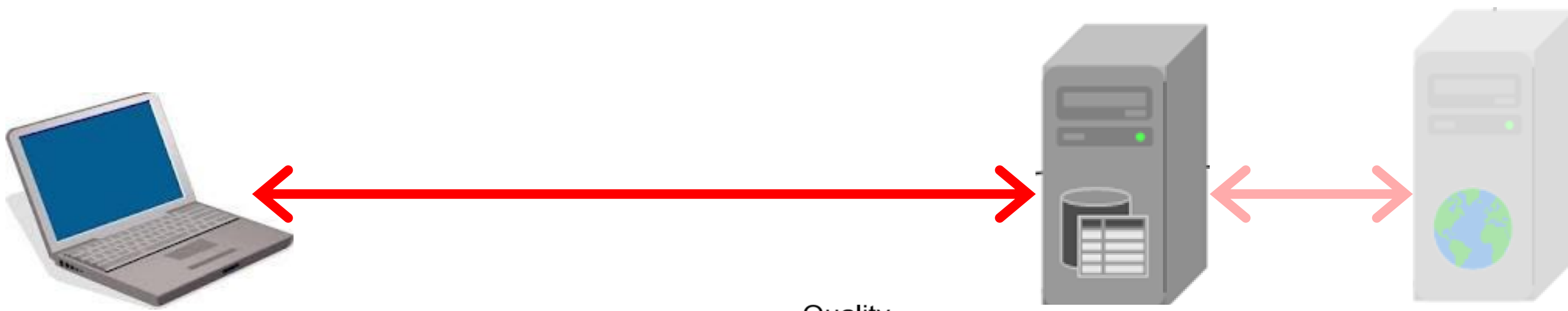


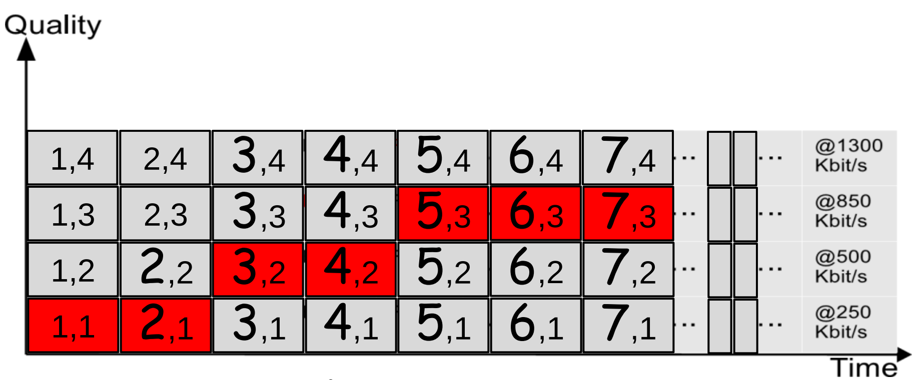
Proxy before



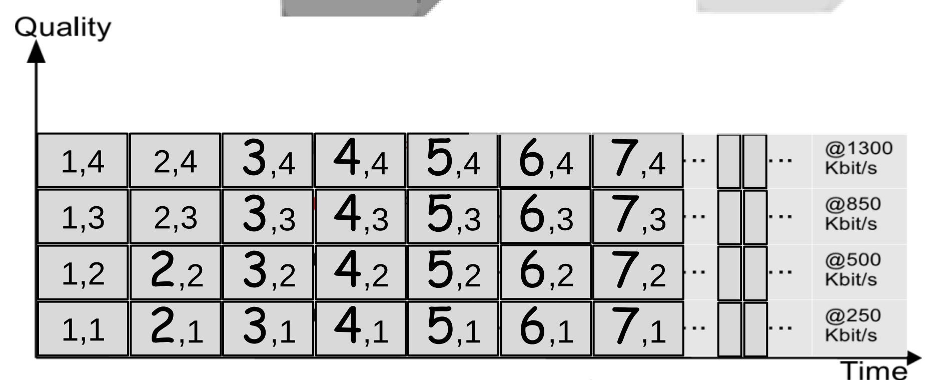
Proxy after

Example: HAS and proxy

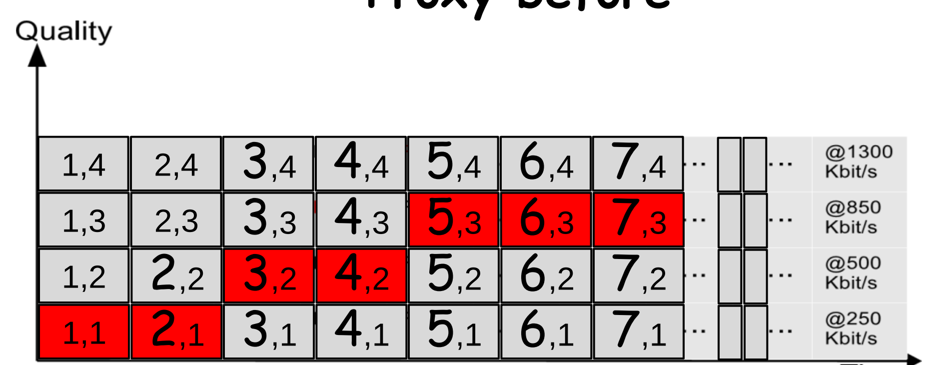




Client 1

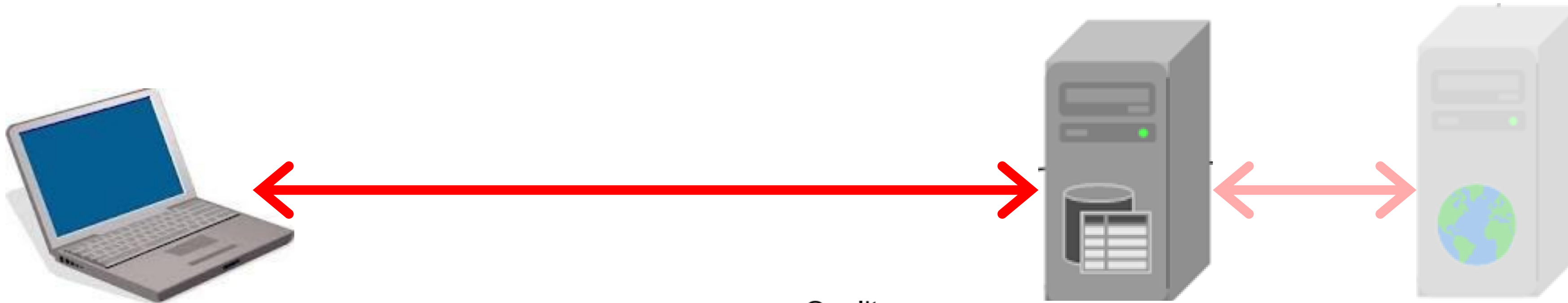


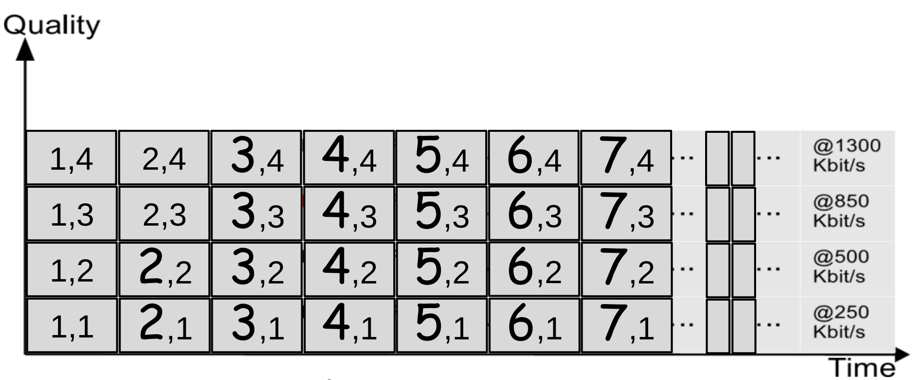
Proxy before



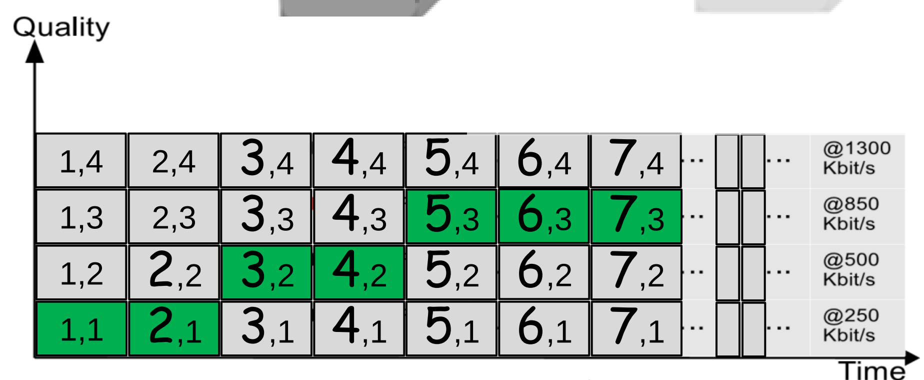
Proxy after

Example: HAS and proxy

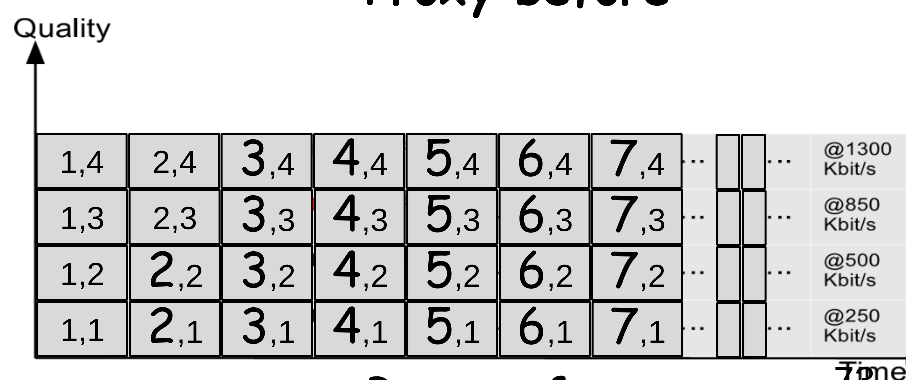




Client 2

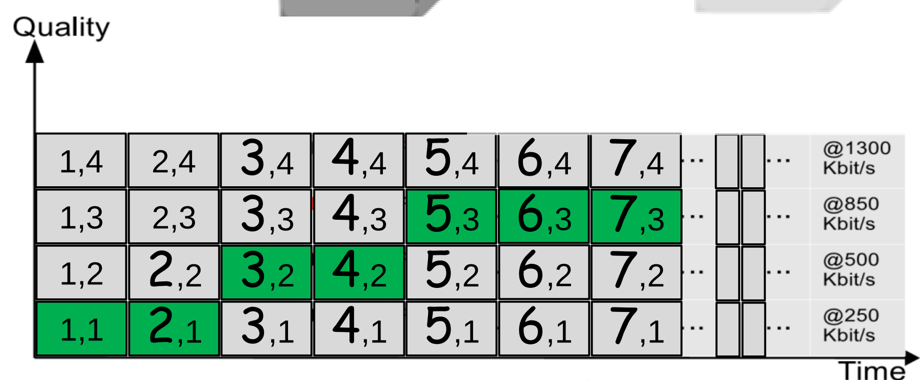


Proxy before

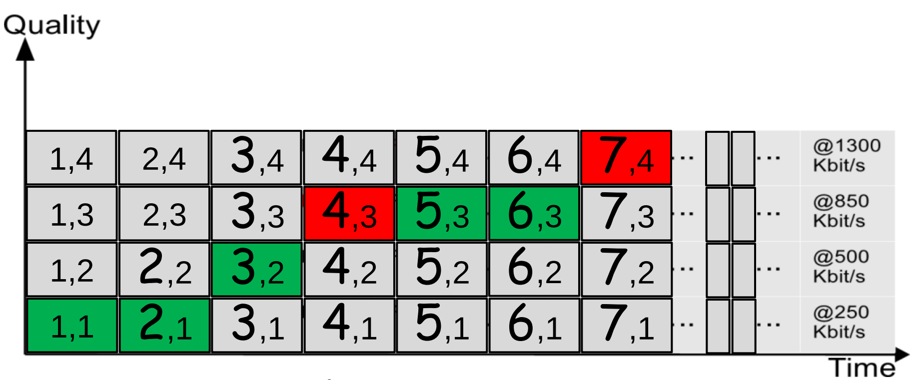


Proxy after

Example: HAS and proxy



Proxy before



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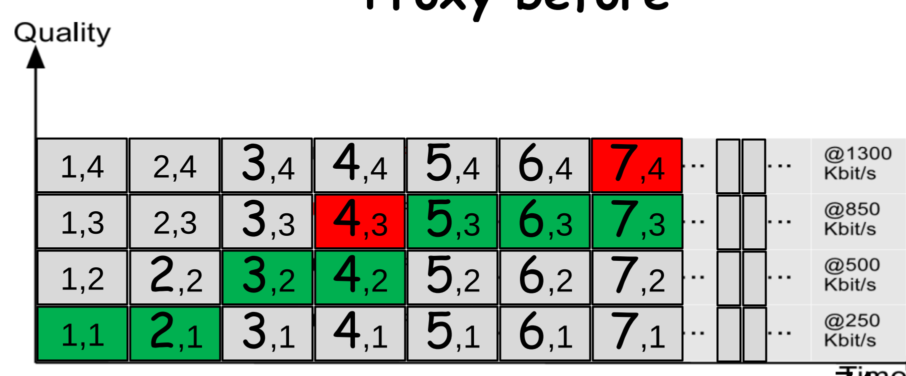
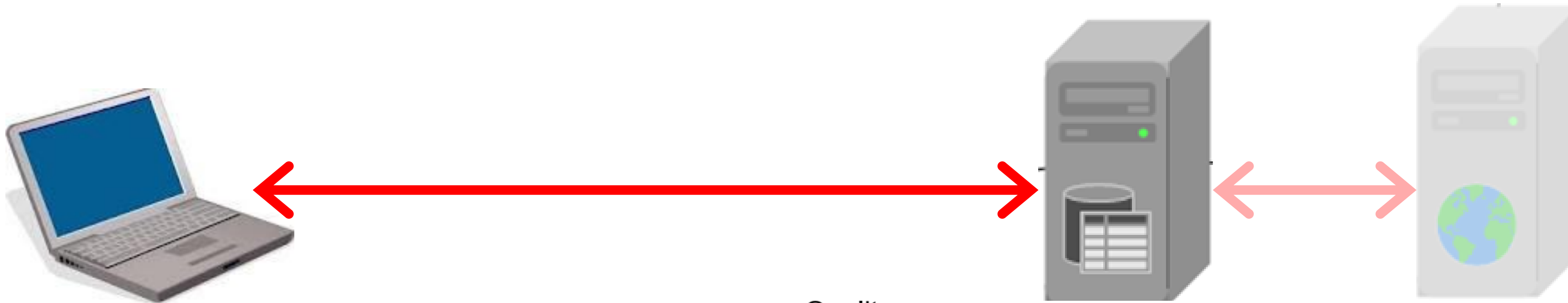
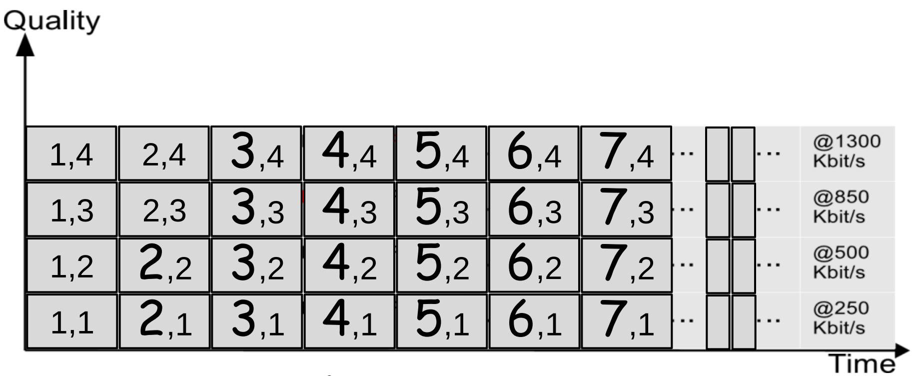


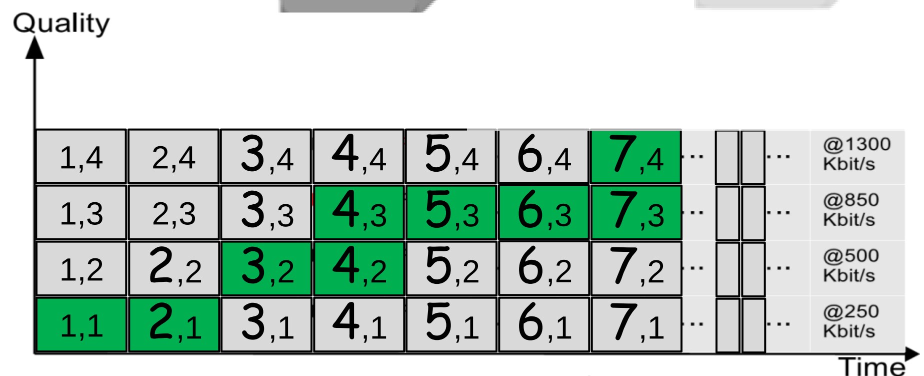
Proxy after

Example: HAS and proxy

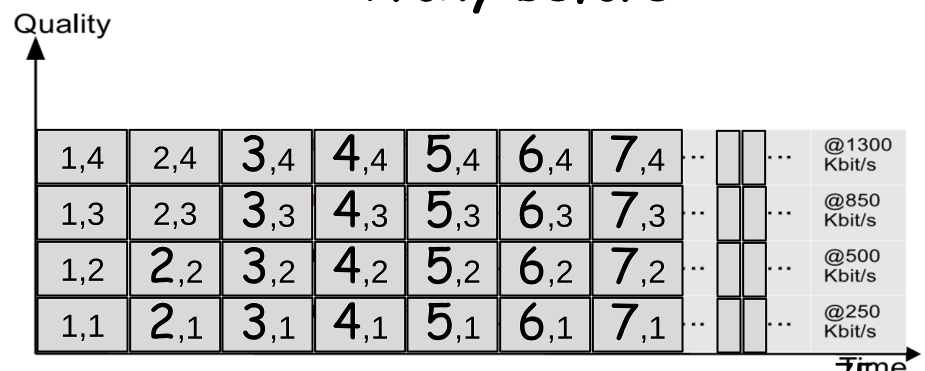




Client 3

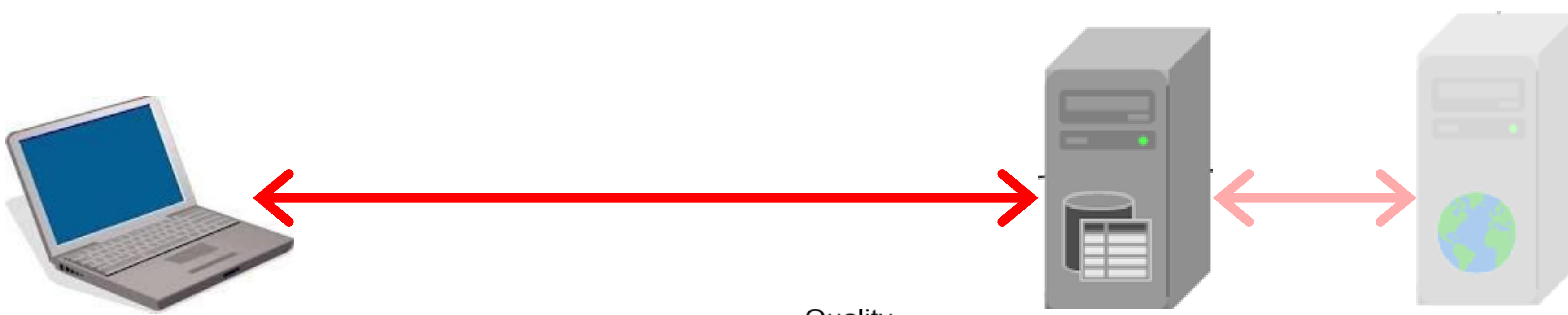


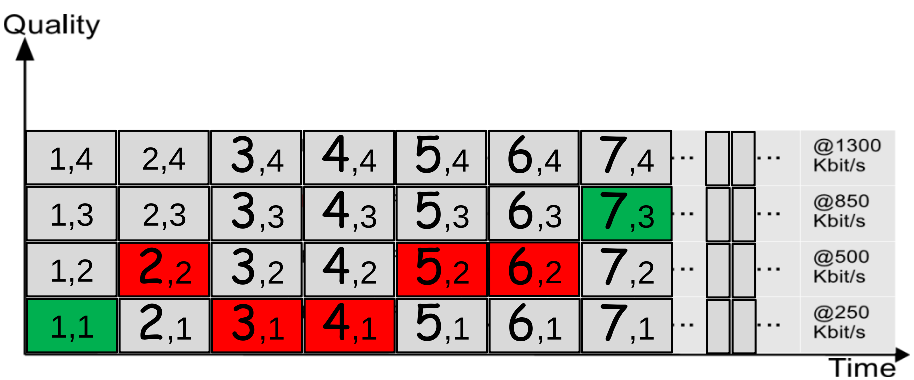
Proxy before



Proxy after

Example: HAS and proxy





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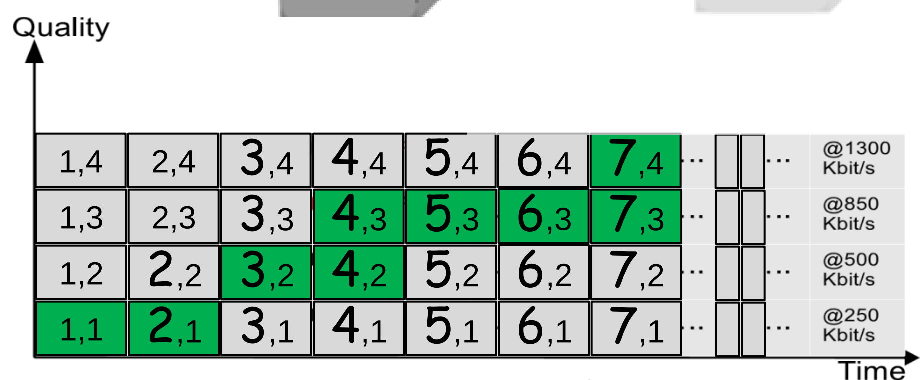
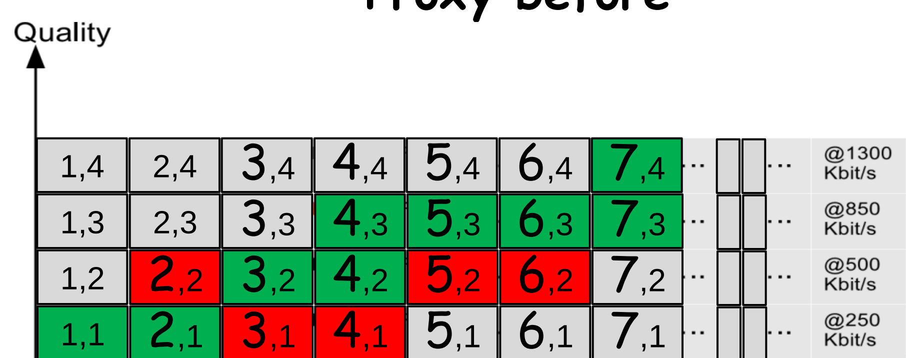


Proxy before



Proxy after

Streaming Multimedia: UDP or TCP?

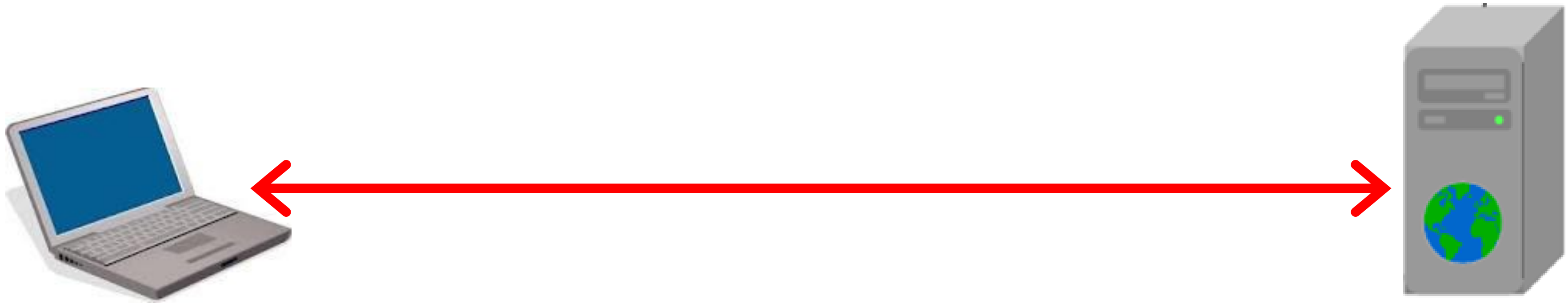
UDP

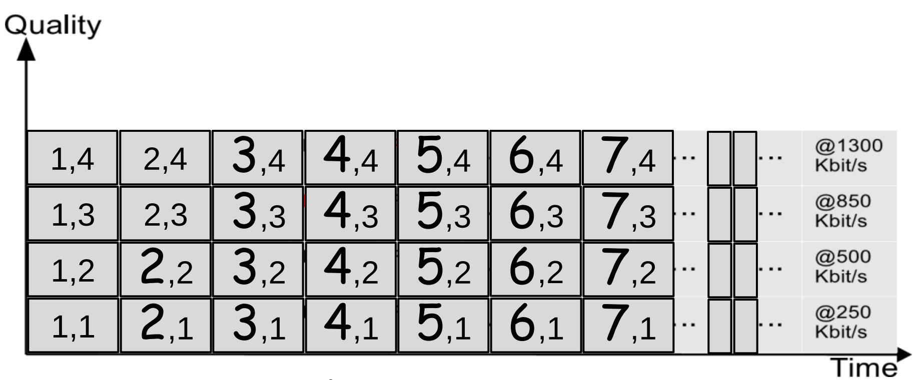
- server sends at rate appropriate for client (oblivious to network congestion!!)
  - often send rate = encoding rate = constant rate
  - then, fill rate = constant rate - packet loss
- short playout delay (2-5 seconds) to compensate for network delay jitter
- error recover: time permitting

TCP

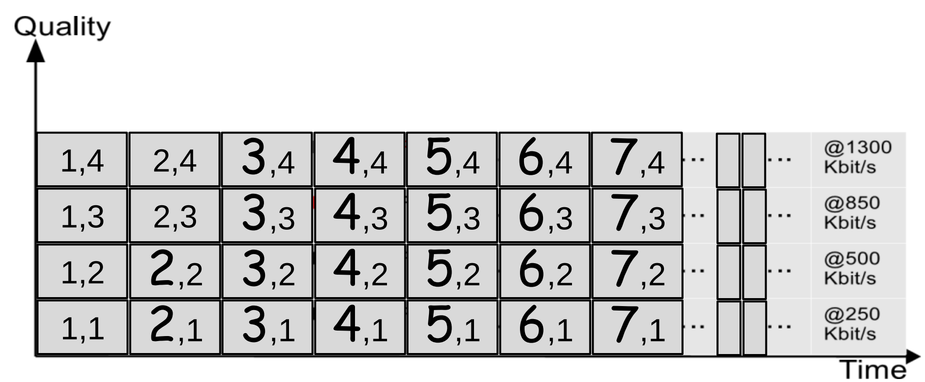
- send at maximum possible rate under TCP
- fill rate fluctuates due to TCP congestion control
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

Example



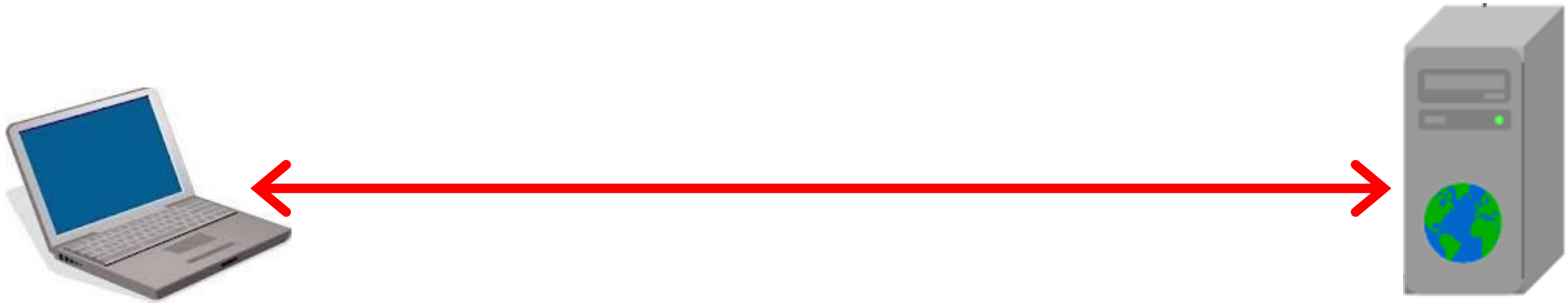


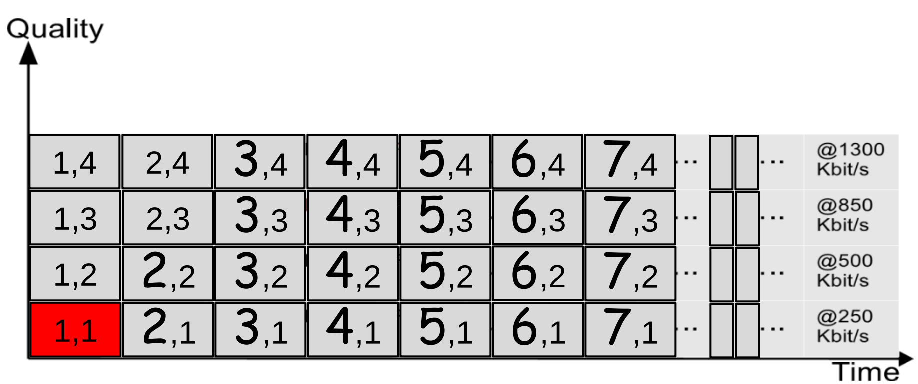
Client



Server

Example





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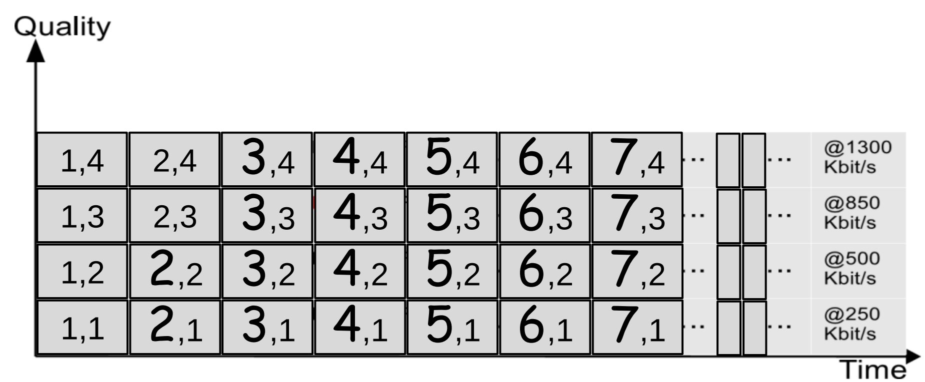
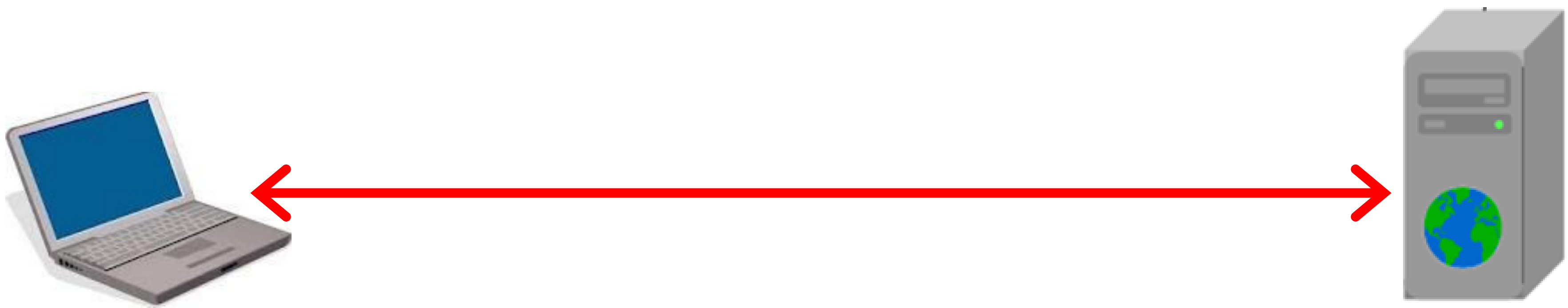
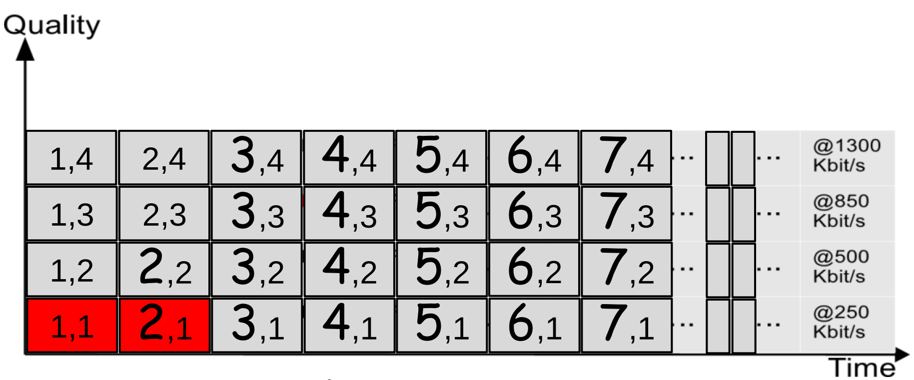


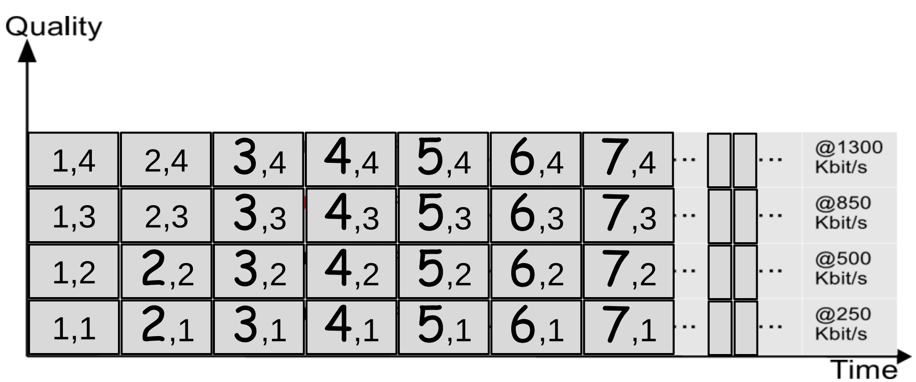
Server

Example



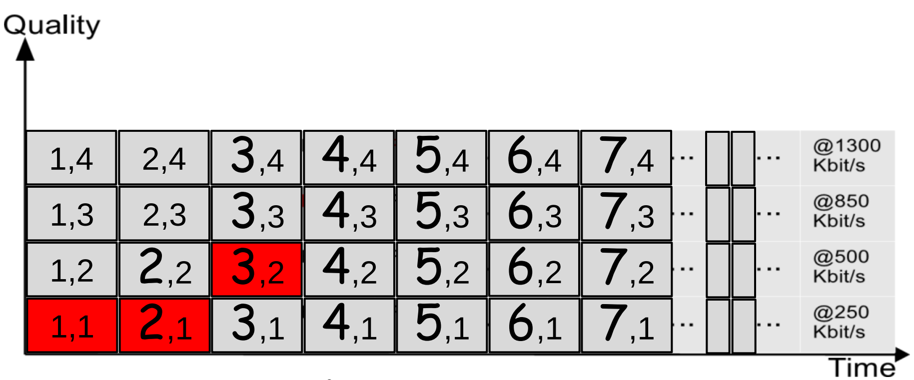


Client

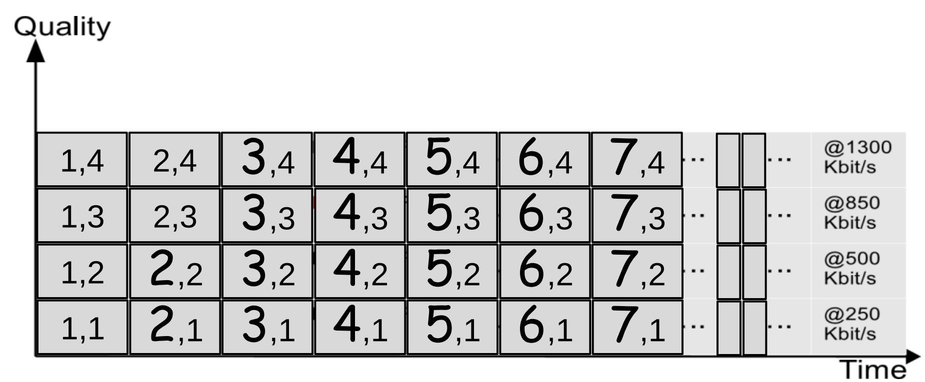


Server

Example

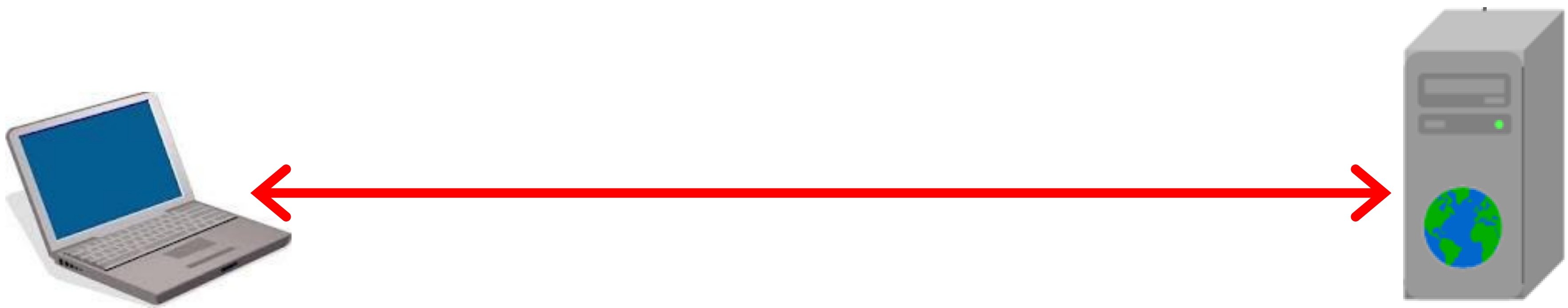


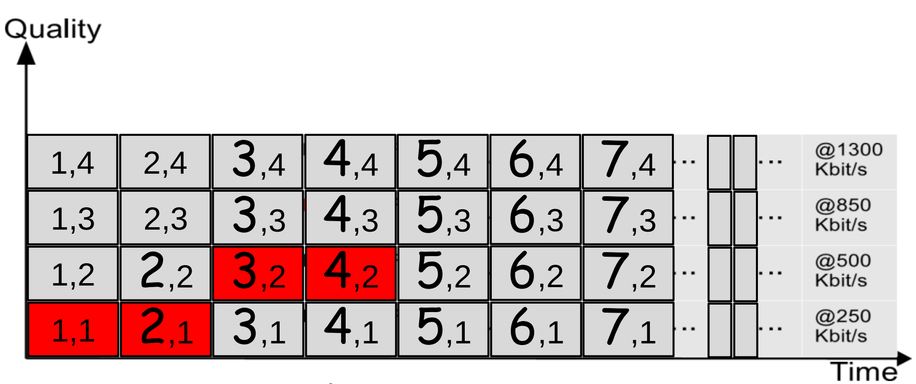
Client



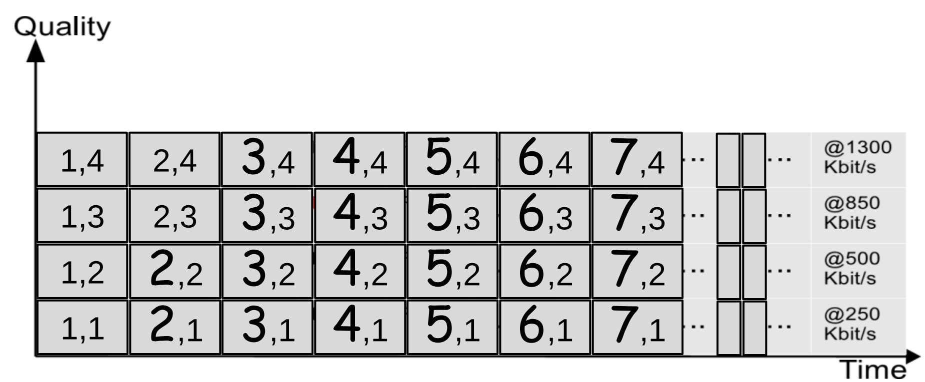
Server

Example



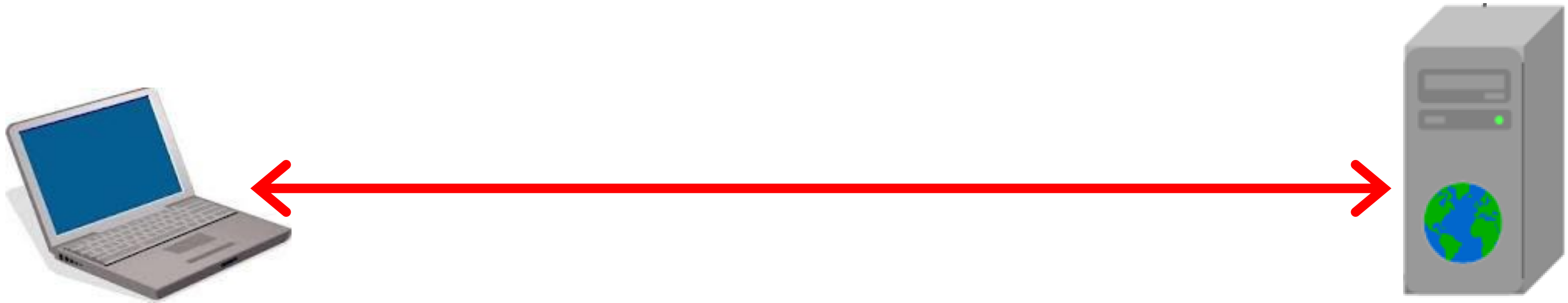


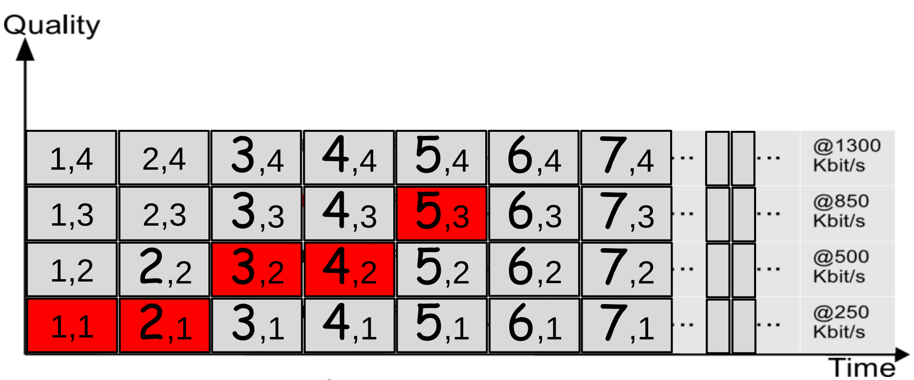
Client



Server

Example





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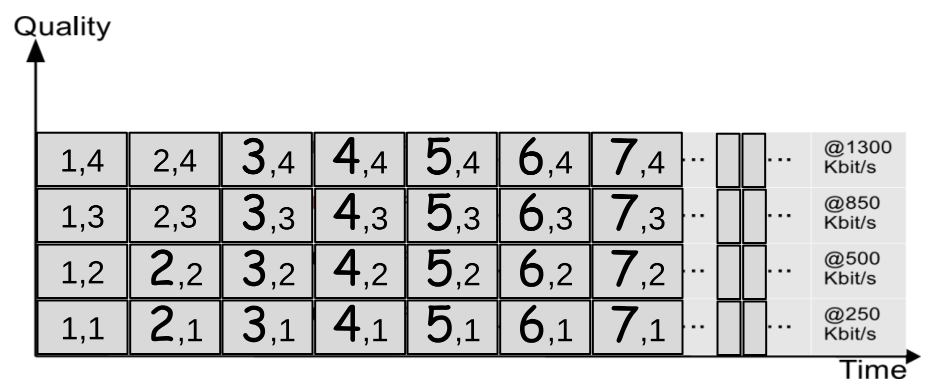
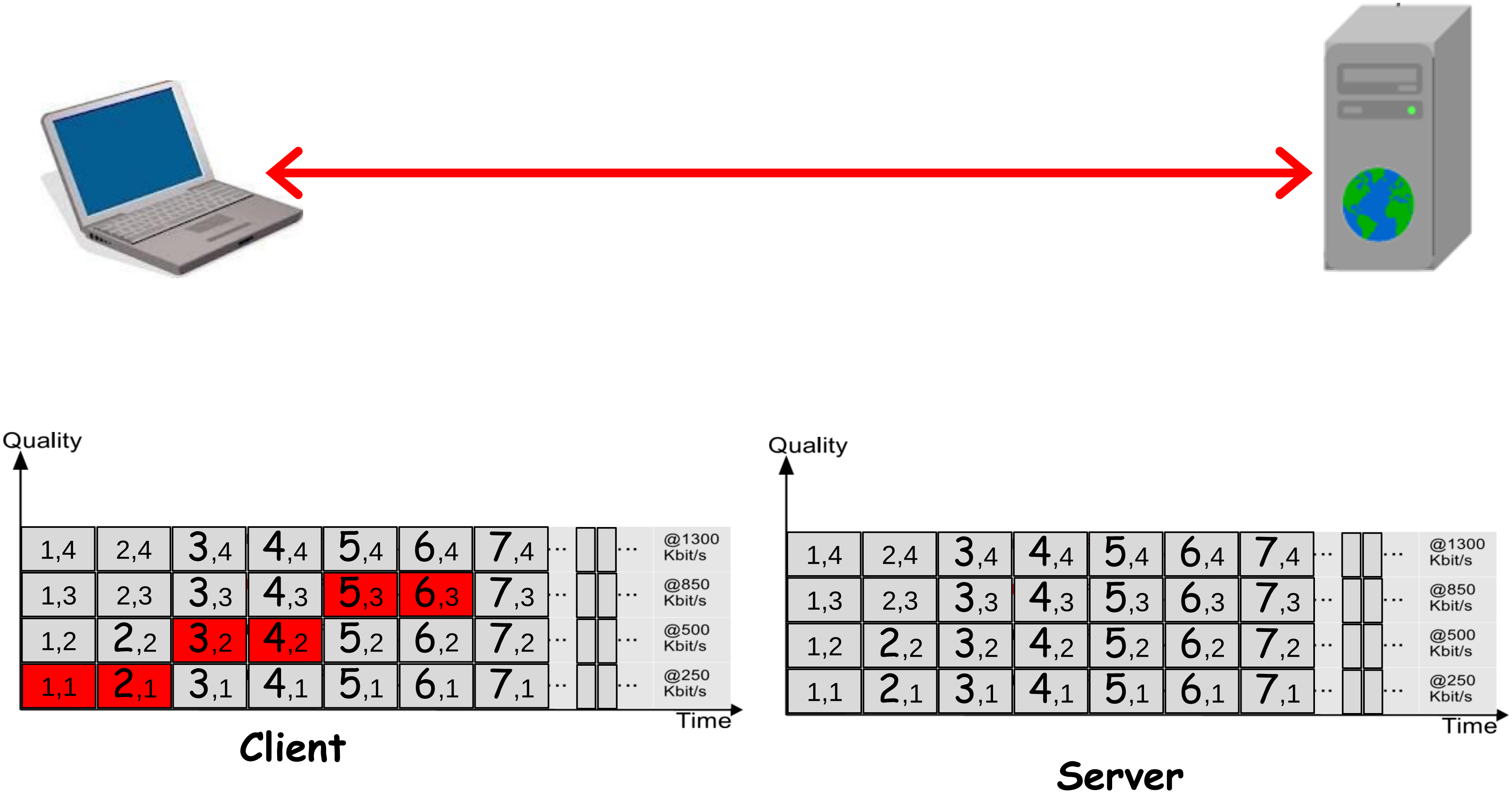


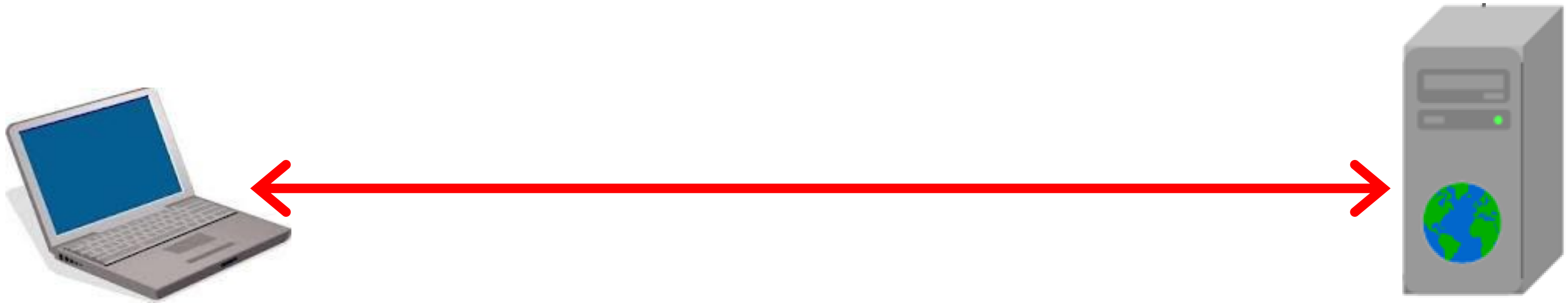
Server

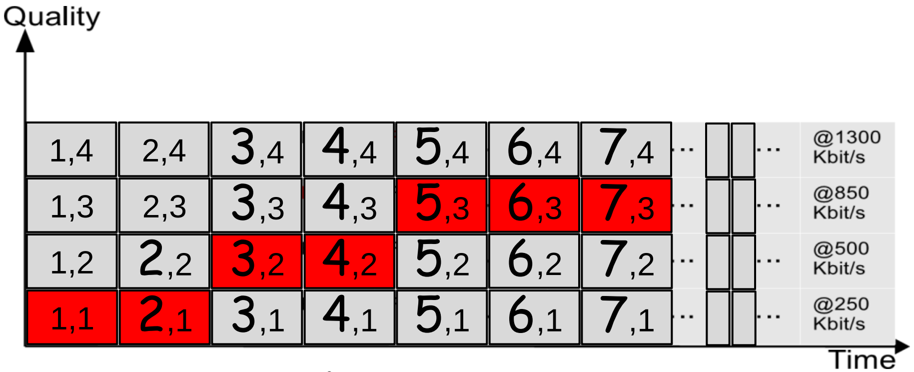
Example



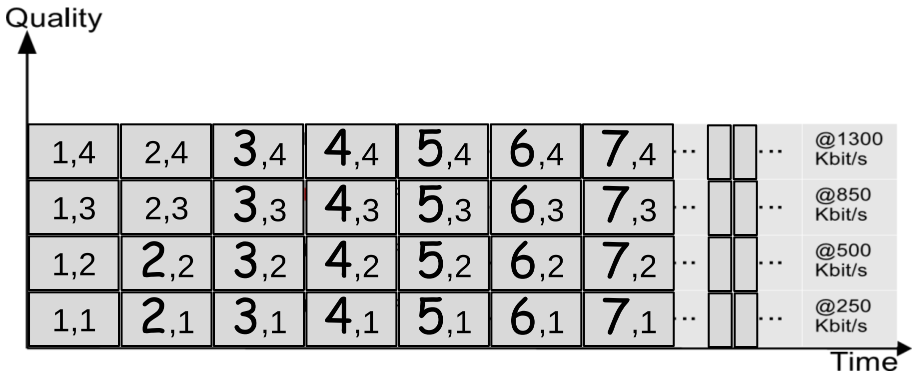
Quality
1,4 2,4 3,4 4,4 5,4 6,4 7,4 @1300 Kbit/s
1,3 2,3 3,3 4,3 5,3 6,3 7,3 @850 Kbit/s
1,2 2,2 3,2 4,2 5,2 6,2 7,2 @500 Kbit/s
1,1 2,1 3,1 4,1 5,1 6,1 7,1 @250 Kbit/s
Time
Client
Quality
1,4 2,4 3,4 4,4 5,4 6,4 7,4 @1300 Kbit/s
1,3 2,3 3,3 4,3 5,3 6,3 7,3 @850 Kbit/s
1,2 2,2 3,2 4,2 5,2 6,2 7,2 @500 Kbit/s
1,1 2,1 3,1 4,1 5,1 6,1 7,1 @250 Kbit/s
Time
Server

Example



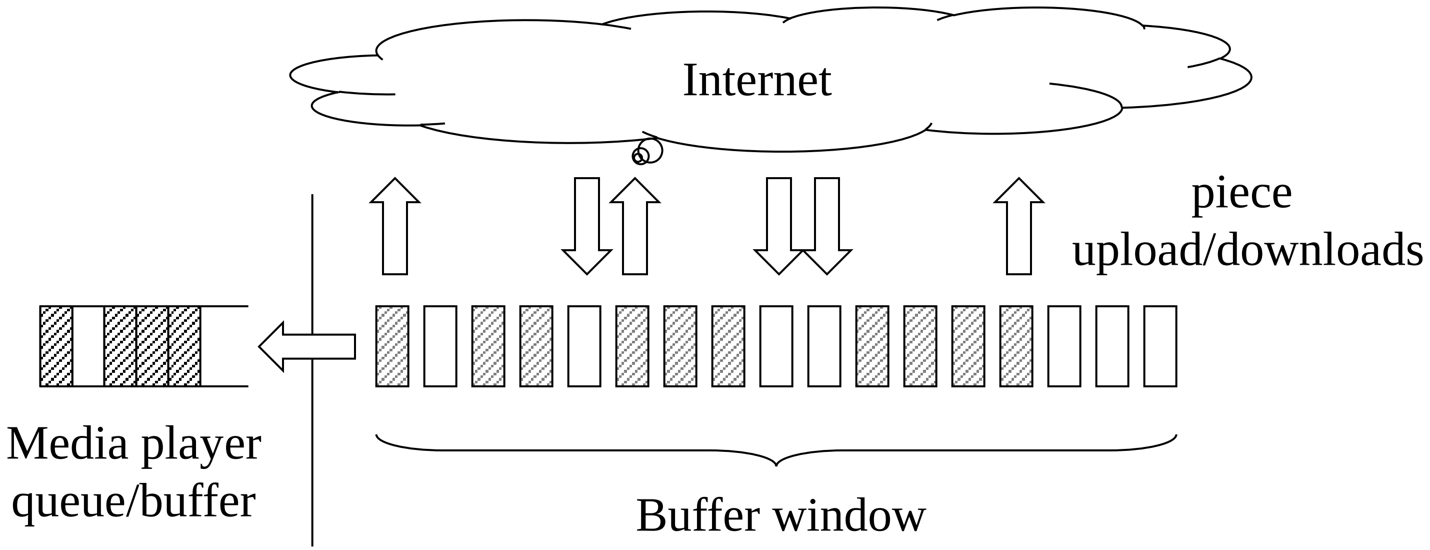


Client



Server

Live Streaming using Bittorrent-like systems



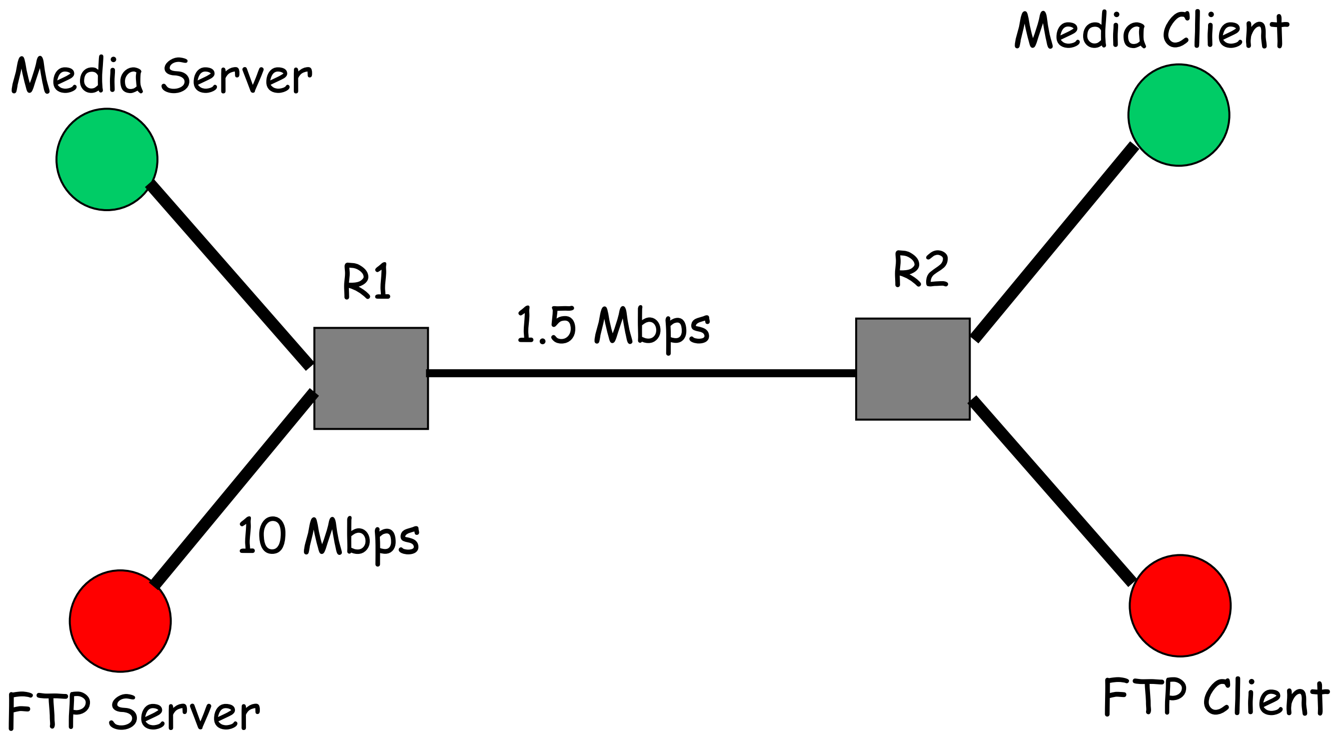
- Live streaming (e.g., CoolStreaming)
  - All peers at roughly the same play/download position
    - High bandwidth peers can easily contribute more …
  - (relatively) Small buffer window
    - Within which pieces are exchanged

Peer-assisted VoD streaming

- Can BitTorrent-like protocols provide scalable on-demand streaming?
- How sensitive is the performance to the application configuration parameters?
  - Piece selection policy (rarest vs. in-order tradeoff)
  - Peer selection policy
  - Upload/download bandwidth
- What is the user-perceived performance?
  - Start-up delay
  - Probability of disrupted playback

ACM SIGMETRICS 2008; IFIP Networking 2007/2009, IEEE/ACM ToN 2012

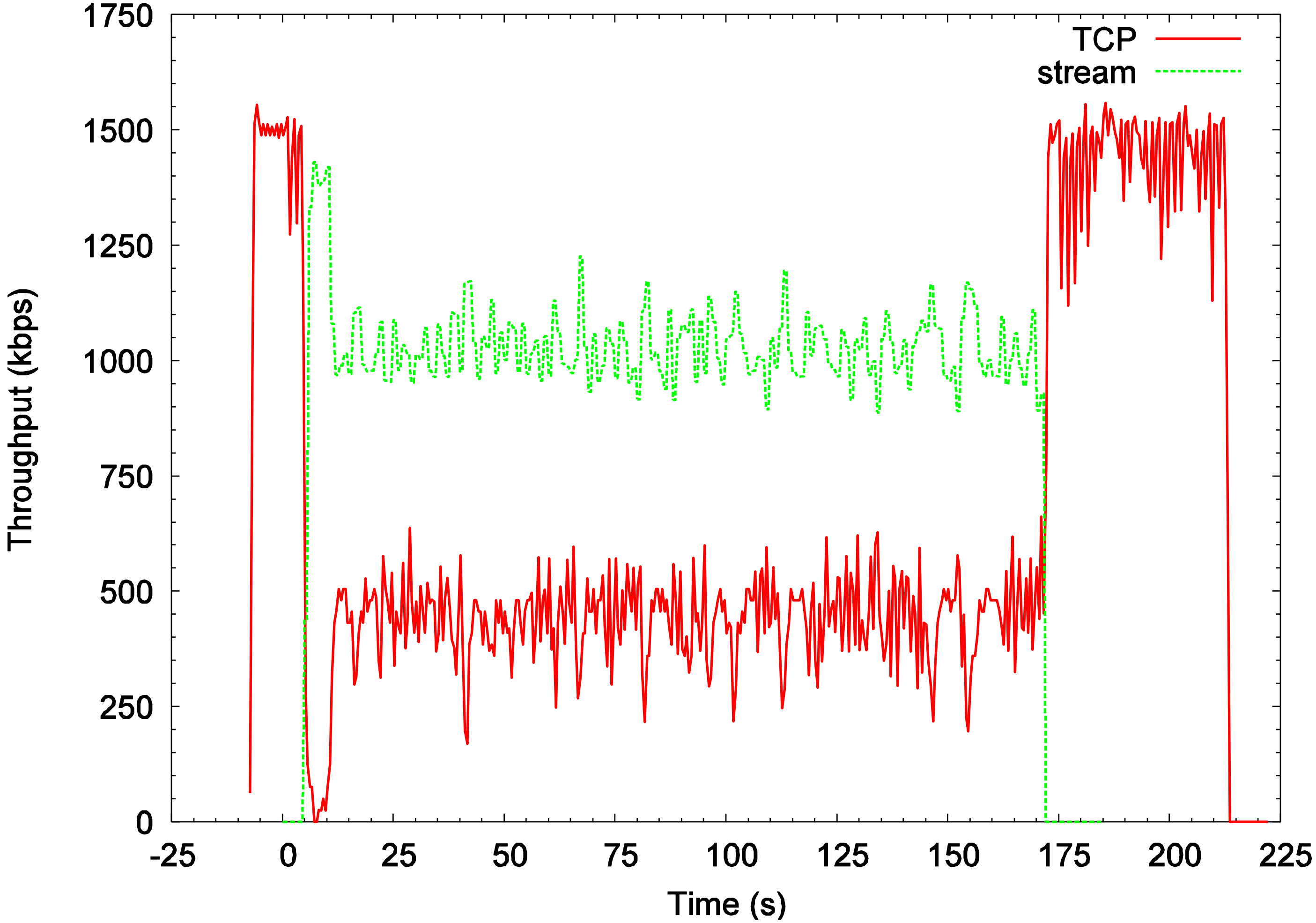
Fairness of UDP Streams (1/2)



- R1-R2 is the bottleneck link
- Streaming uses UDP at the transport layer; requested media encoded at 1 Mbps
- What fraction of the bottleneck is available to FTP?

Credit: MSc thesis work by Sean Boyden (2006)

Fairness of RealVideo Streams (2/2)



A protocol family for streaming

- RTSP
- RTP
- RTCP

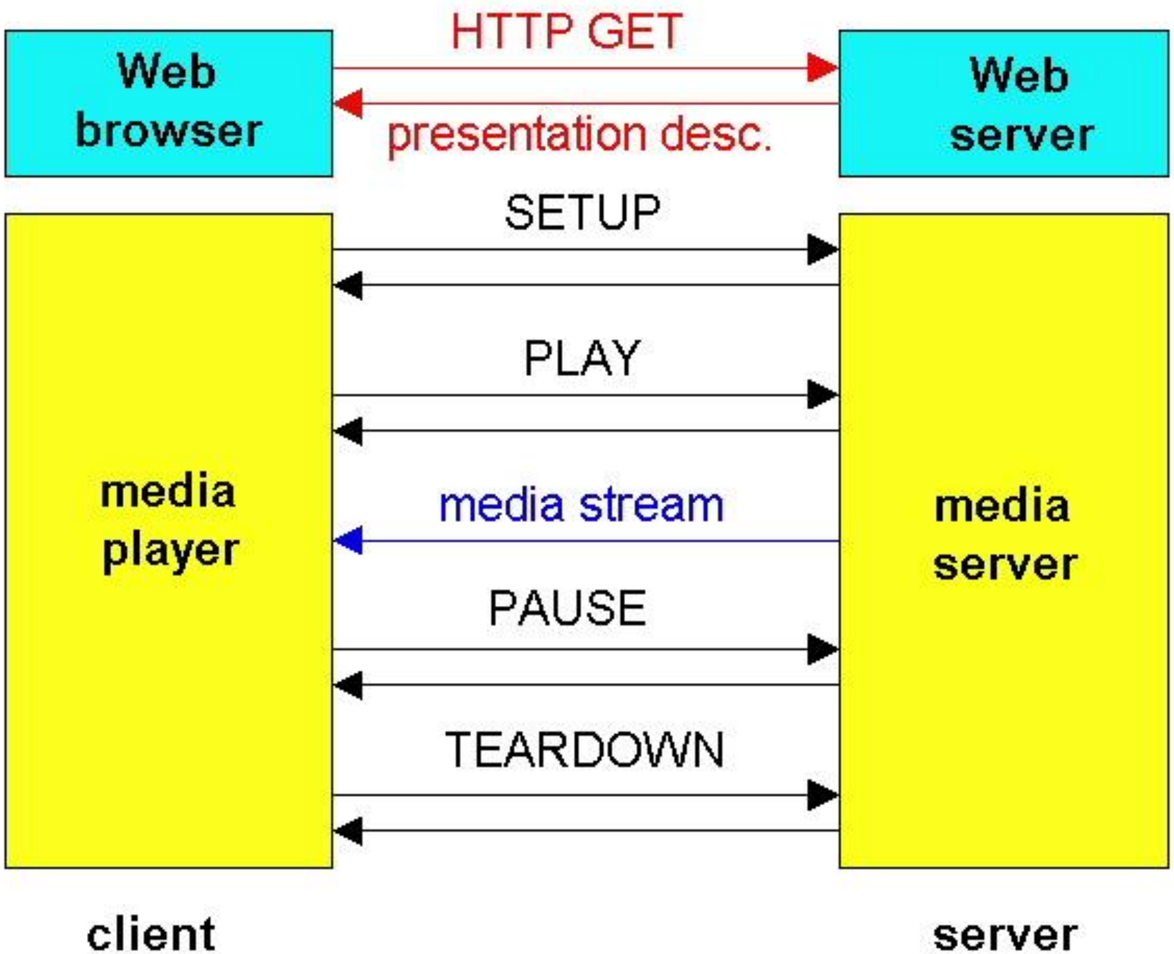
RTSP Example

Scenario:

- metafile communicated to web browser
- browser launches player
- player sets up an RTSP control connection, data connection to streaming server

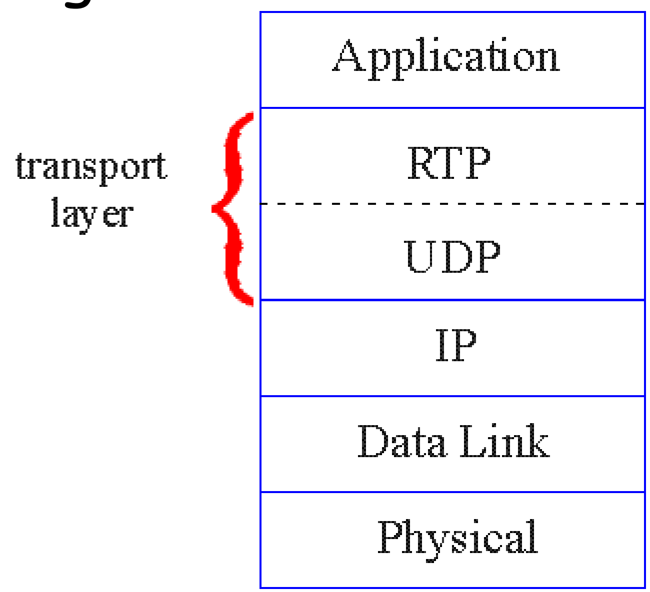
RTSP Operation

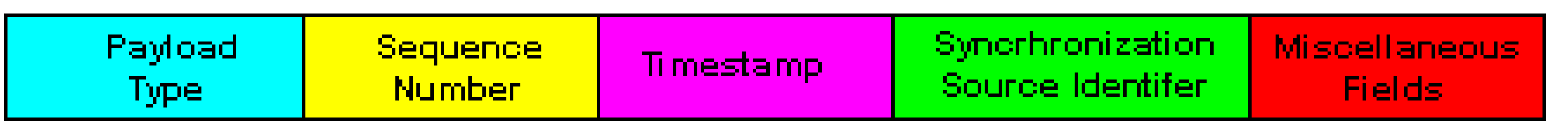
- RTSP out-of-band control messages
  - Port 554
- media stream is considered "in-band"



Real-Time Protocol (RTP)

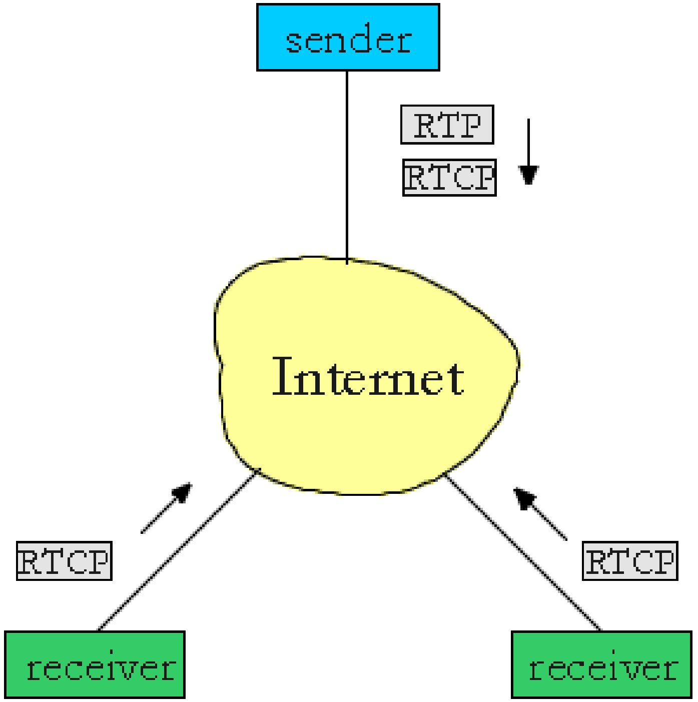
- RTP specifies packet structure for packets carrying audio, video data
- RFC 3550
- RTP runs in end systems
- RTP packets encapsulated in UDP segments





RTP Header

Real-time Control Protocol (RTCP)



Receiver report packets:

- fraction of packets lost, last sequence number, average interarrival jitter

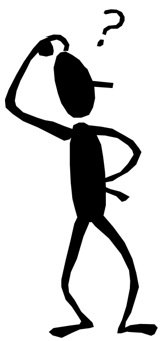
Sender report packets:

- SSRC of RTP stream, current time, number of packets sent, number of bytes sent

- RTCP attempts to limit its traffic to 5% of session bandwidth

Multimedia Over "Best Effort" Internet

- TCP/UDP/IP: *no* guarantees on delay, loss



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

Today's multimedia applications implement functionality at the app. layer to mitigate (as best possible) effects of delay, loss

Packet Loss

- network loss: IP datagram lost due to network congestion (router buffer overflow) or losses at wireless link(s)
- delay loss: IP datagram arrives too late for playout at receiver (effectively the same as if it was lost)
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - Tolerable delay depends on the application
- How can packet loss be handled?
  - We will discuss this next …

Receiver-based Packet Loss Recovery

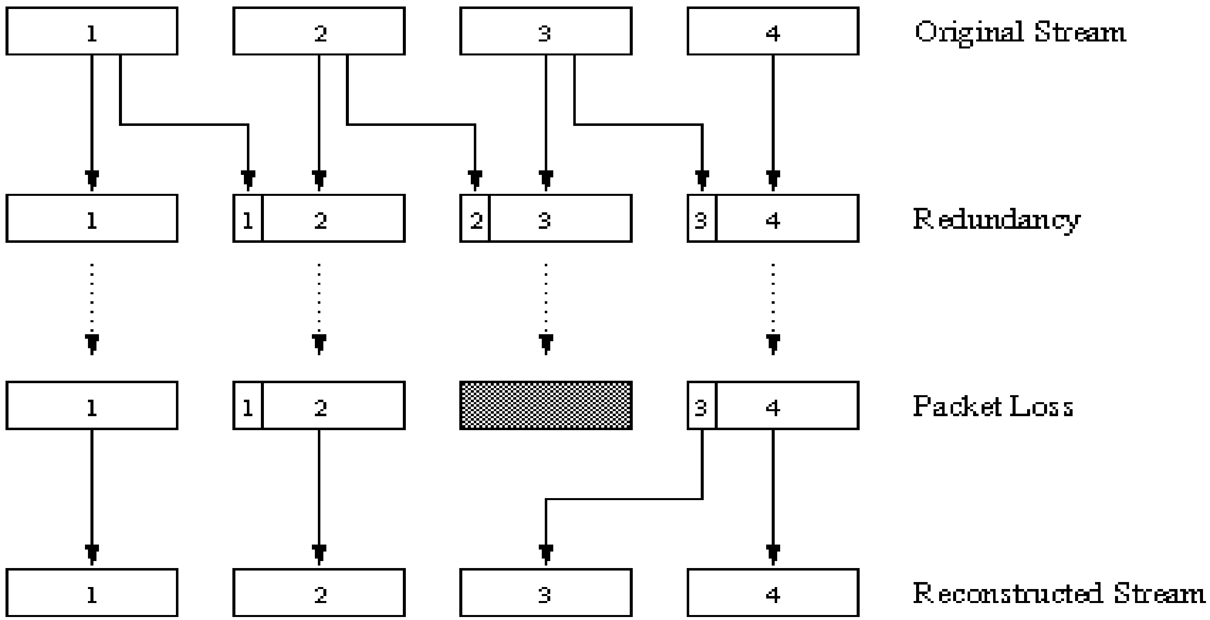
- Generate replacement packet
  - Packet repetition
  - Interpolation
  - Other sophisticated schemes
- Works when audio/video streams exhibit short-term correlations (e.g., self-similarity)
- Works for relatively low loss rates (e.g., < 5%)
- Typically, breaks down on "bursty" losses

Forward Error Correction (FEC)

- For every group of n actual media packets, generate k additional redundant packets
- Send out n+k packets, which increases the bandwidth consumption by factor k/n.
- Receiver can reconstruct the original n media packets provided at most k packets are lost from the group
- Works well at high loss rates (for a proper choice of k)
- Handles "bursty" packet losses
- Cost: increase in transmission cost (bandwidth)

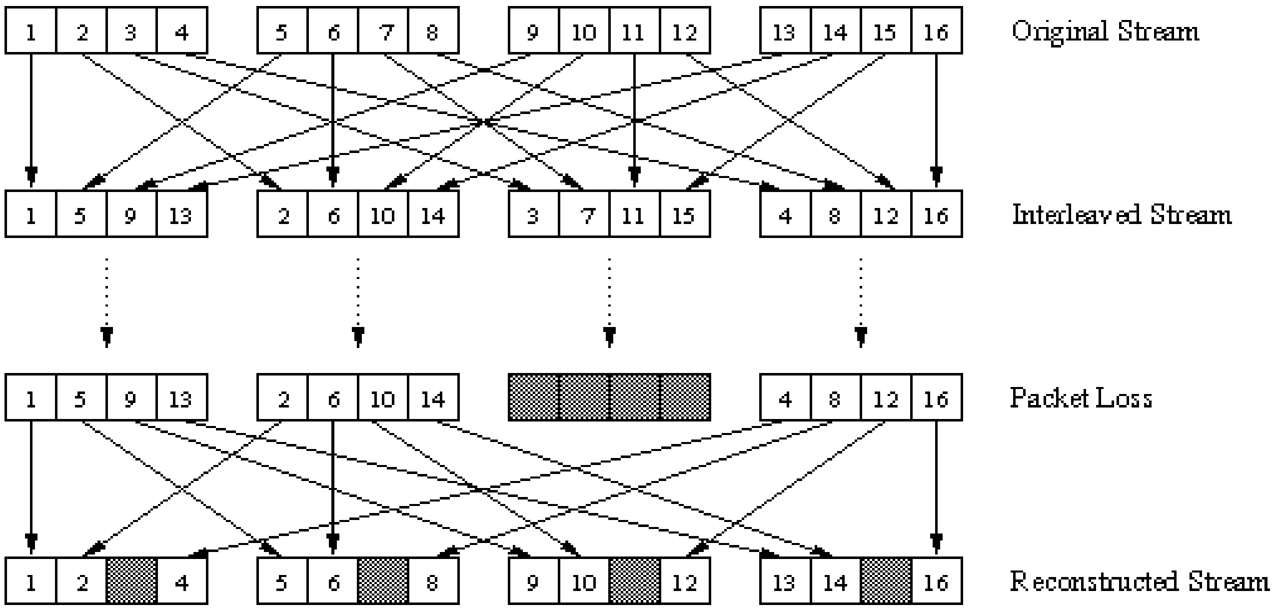
Another FEC Example

- "piggyback lower quality stream"



- Whenever there is non-consecutive loss, the receiver can conceal the loss.
- Can also append (n-1)st and (n-2)nd low-bit rate chunk

Interleaving: Recovery from packet loss



Interleaving

- Intentionally alter the sequence of packets before transmission
- Better robustness against “burst” losses of packets
- Results in increased playout delay from inter-leaving

More slides ….

Outline

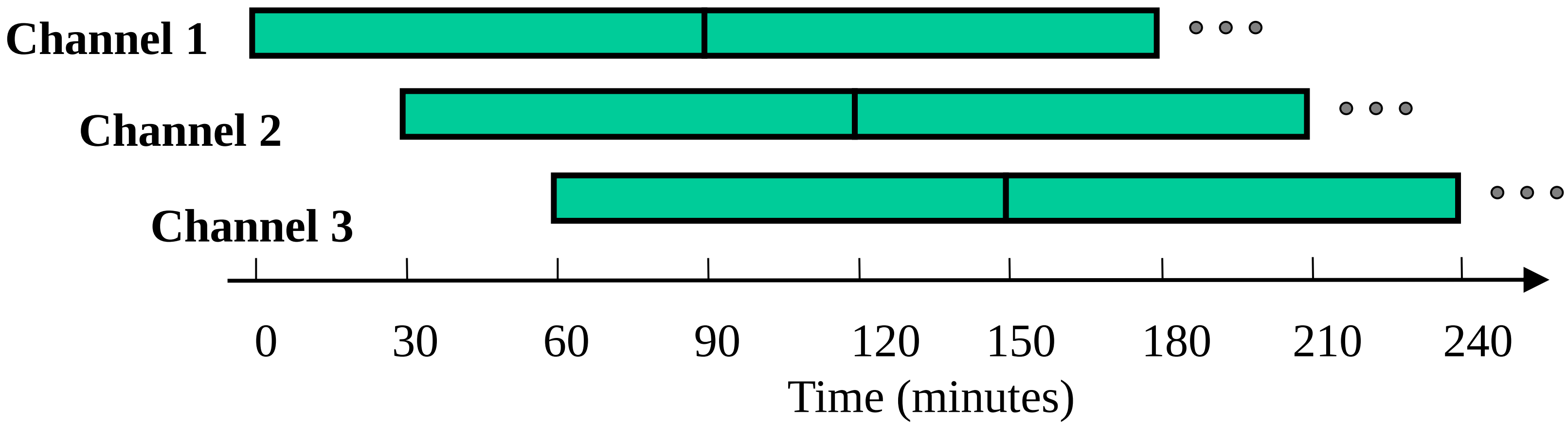
- Multimedia Networking Applications
- Streaming stored audio and video
- Scalable Streaming Techniques
- Content Distribution Networks
- Beyond Best Effort

Streaming Popular Content

- Consider a popular media file
  - Playback rate: 1 Mbps
  - Duration: 90 minutes
  - Request rate: once every minute
- How can a video server handle such high loads?
  - Approach 1: Start a new "stream" for each request
  - Allocate server and disk I/O bandwidth for each request
  - Bandwidth required at server= 1 Mbps x 90

Streaming Popular Content using Batching

- Approach 2: Leverage the multipoint delivery capability of modern networks
- Playback rate = 1 Mbps, duration = 90 minutes
- Group requests in non-overlapping intervals of 30 minutes:
  - Max. start-up delay = 30 minutes
  - Bandwidth required = 3 channels = 3 Mbps



Batching Issues

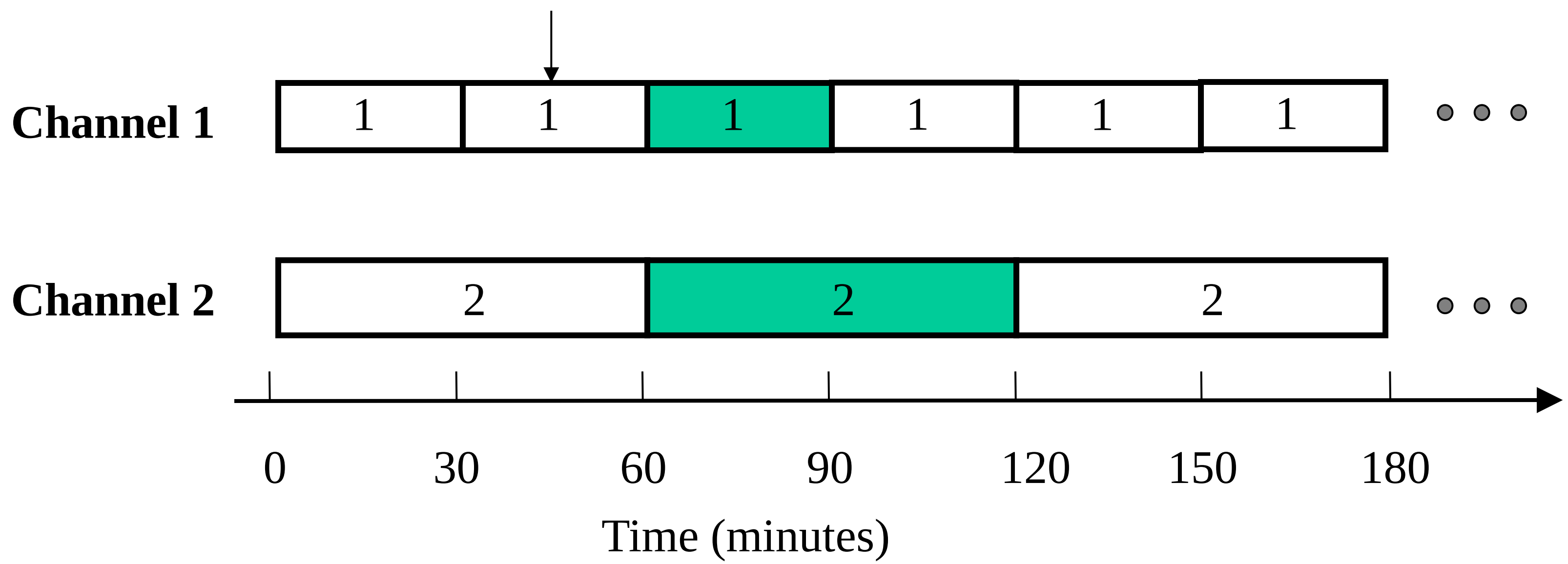
- Bandwidth increases linearly with decrease in start-up delays

- Can we reduce or eliminate "start-up" delays?
  - Periodic Broadcast Protocols

  - Stream Merging Protocols

Periodic Broadcast Example

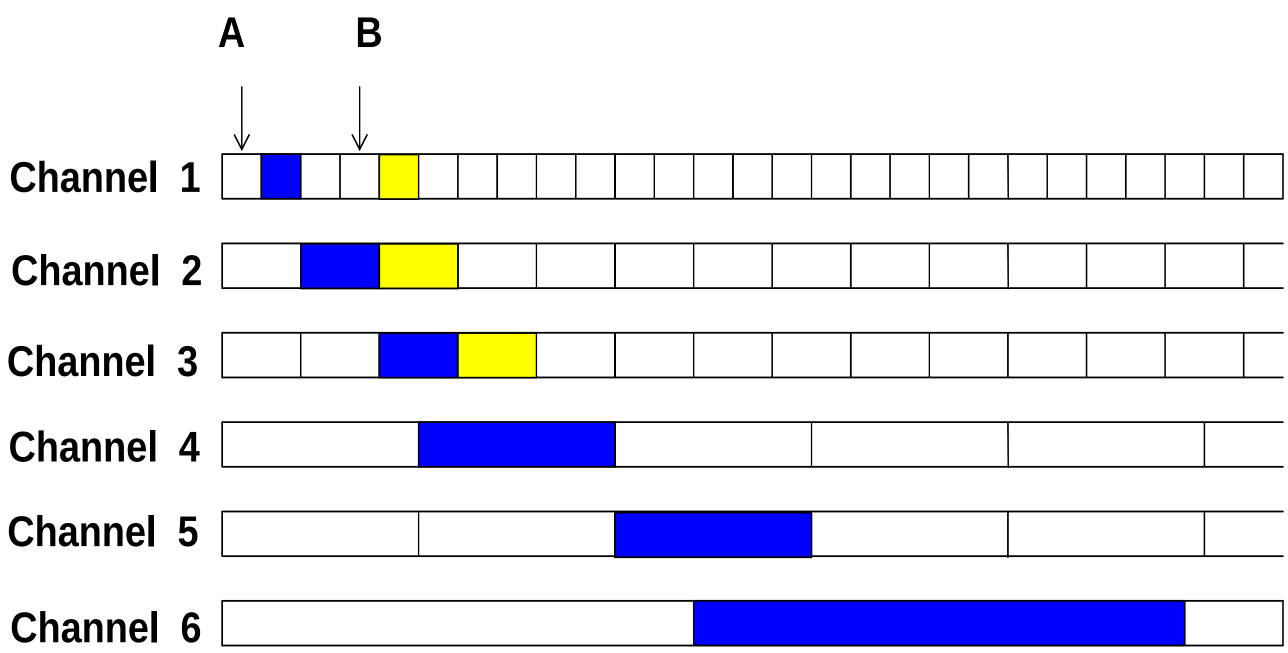
- Partition the media file into 2 segments with relative sizes {1, 2}. For a 90 min. movie:
  - Segment 1 = 30 minutes, Segment 2 = 60 minutes
- Advantage:
  - Max. start-up delay = 30 minutes
  - Bandwidth required = 2 channels = 2 Mbps
- Disadvantage: Requires increased client capabilities



Skyscraper Broadcasts (SB)

[Hua & Sheu 1997]

- Divide the file into *K* segments of increasing size
  - Segment size progression: 1, 2, 2, 5, 5, 12, 12, 25, …
- Multicast each segment on a separate channel at the playback rate
- Aggregate rate to clients: *2 x playback rate*



Comparing Batching and SB

Server Bandwidth	Start-up Delay	
	Batching	SB
1 Mbps	90 minutes	90 minutes
2 Mbps	45 minutes	30 minutes
6 Mbps	15 minutes	3 minutes
10 Mbps	9 minutes	30 seconds

- Playback rate = 1 Mbps, duration = 90 minutes

- Limitations of Skyscraper:
  - Ad hoc segment size progress
  - Does not work for low client data rates

Reliable Periodic Broadcasts (RPB)

[Mahanti *et al*. 2001, 2003, 2004]

- Optimized PB protocols (no packet loss recovery)
  - client fully downloads each segment before playing
  - required server bandwidth near minimal
  - Segment size progression is *not* ad hoc
  - Works for client data rates < 2 x playback rate
- extend for packet loss recovery
- extend for "bursty" packet loss
- extend for client heterogeneity

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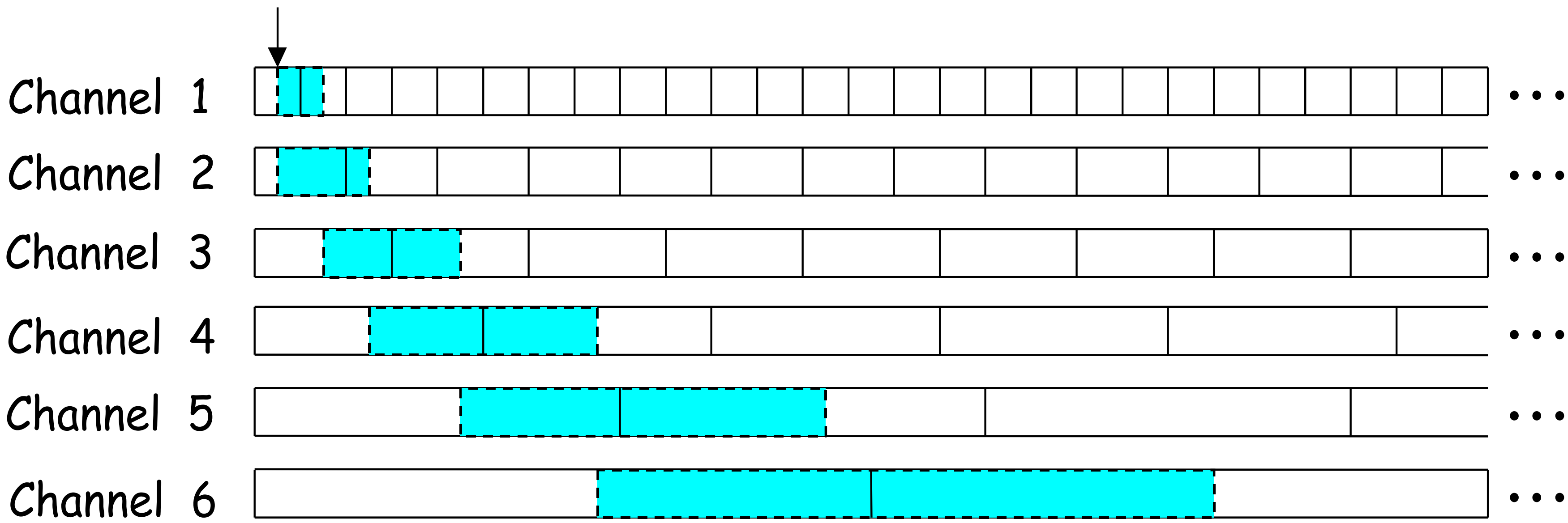
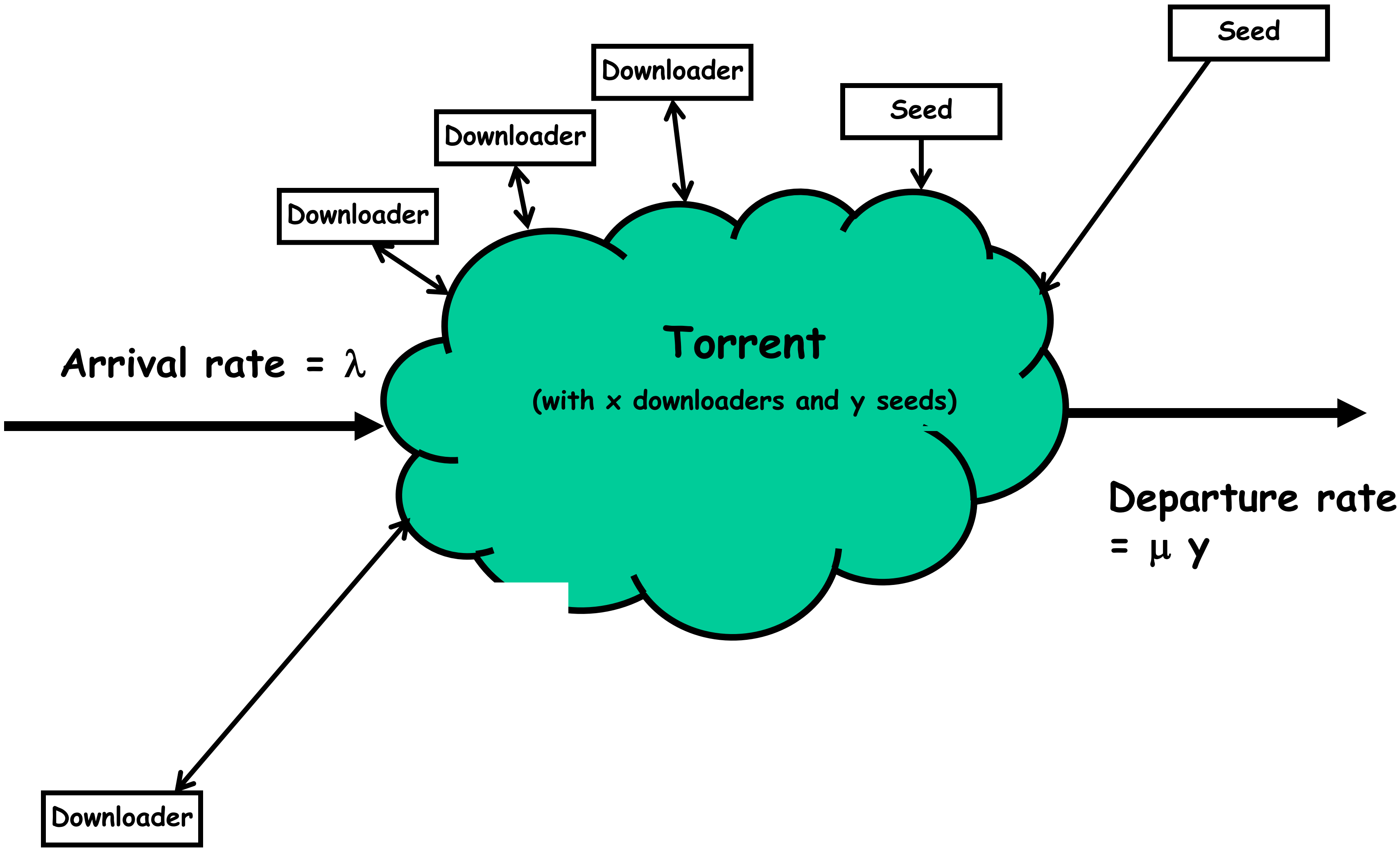
Optimized Periodic Broadcasts

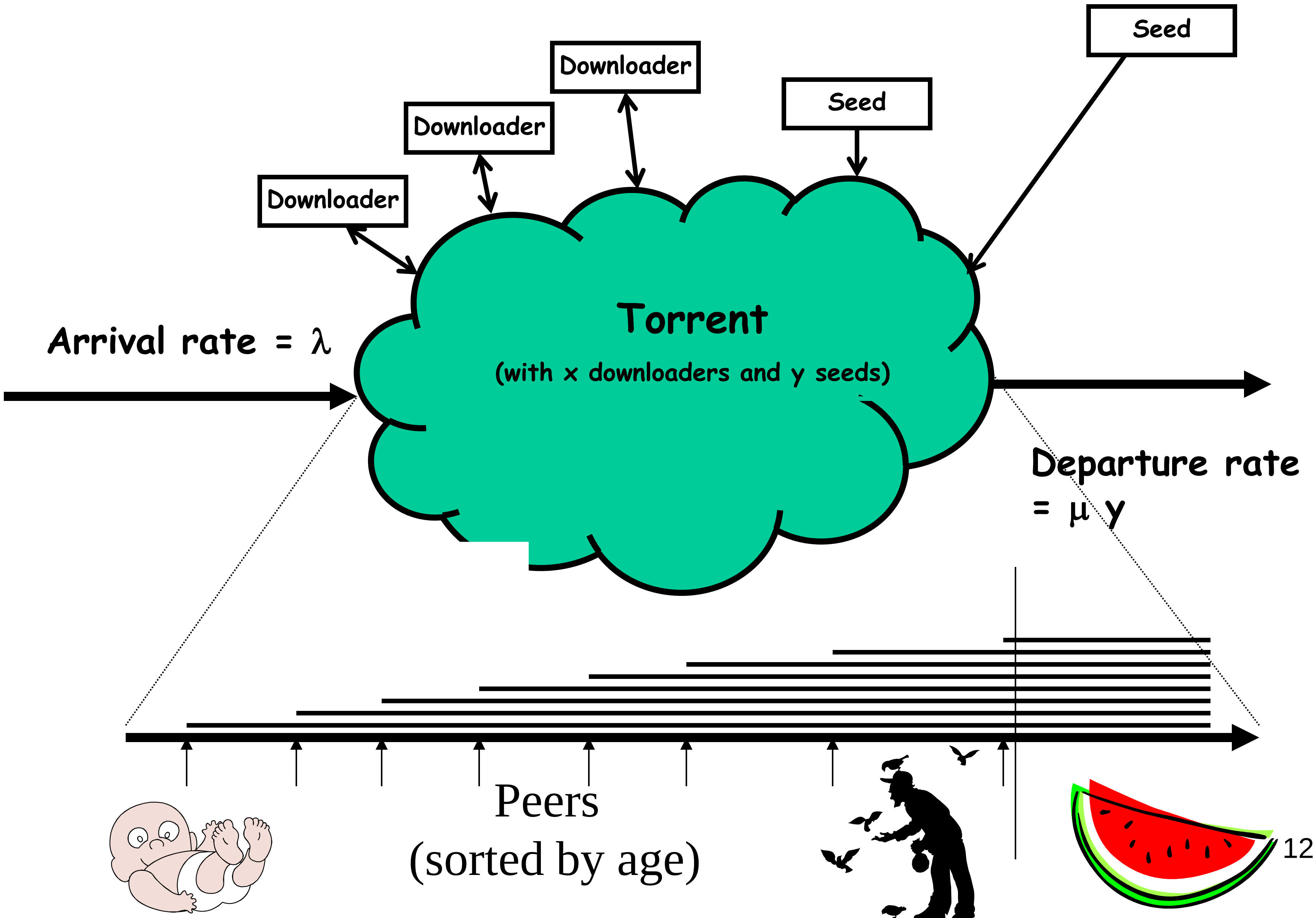


- r = segment streaming rate = 1
- s = maximum # streams client listens to concurrently = 2
- b = client data rate = s x r = 2
- length of first s segments: $\frac{1}{r} l_k = \frac{1}{r} l_1 + \sum_{j=1}^{k-1} l_j$
- length of segment k > s: $\frac{1}{r} l_k = \sum_{j=k-s}^{k-1} l_j$

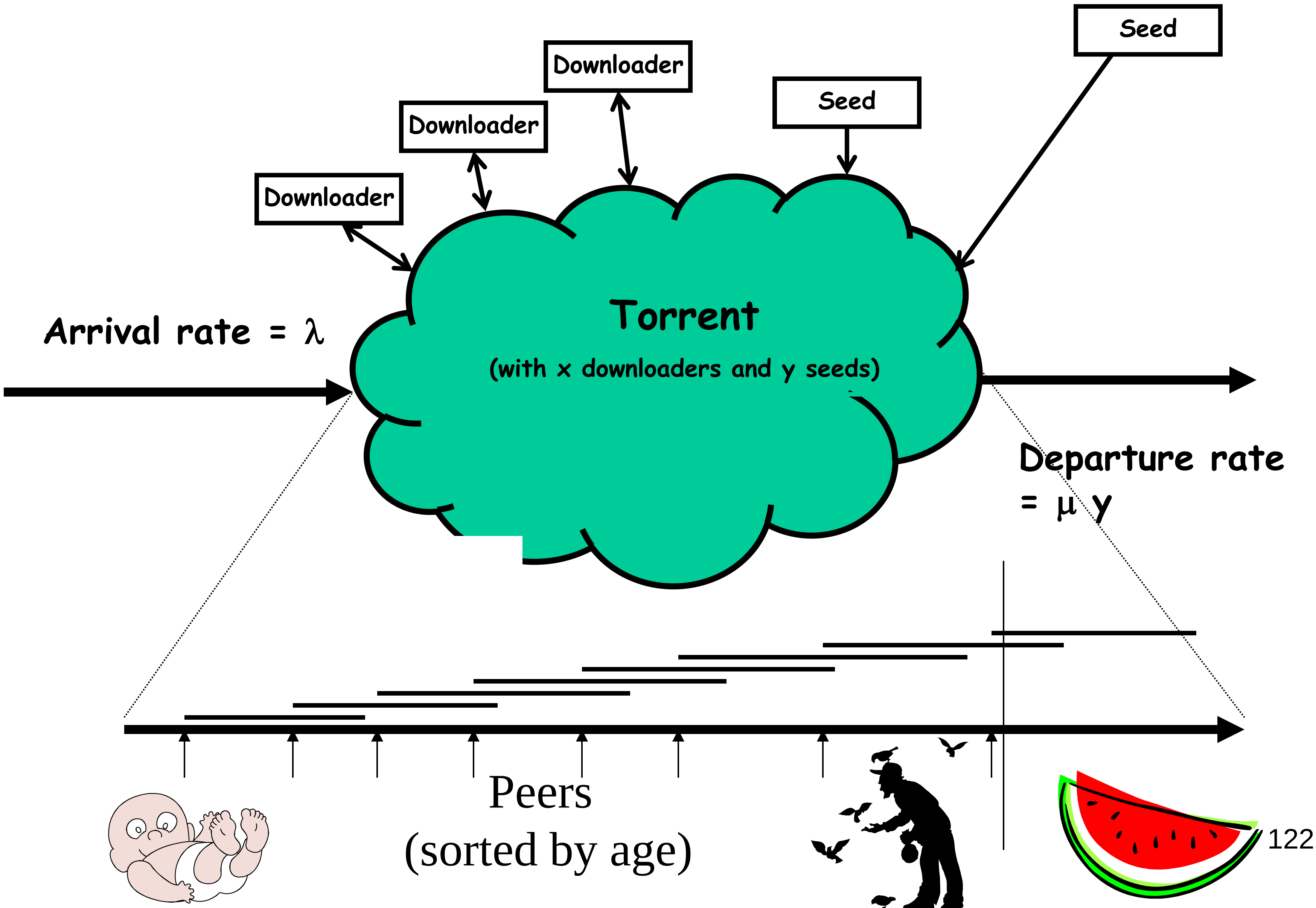
BitTorrent Model



BitTorrent Model (random)



BitTorrent Model (chaining)

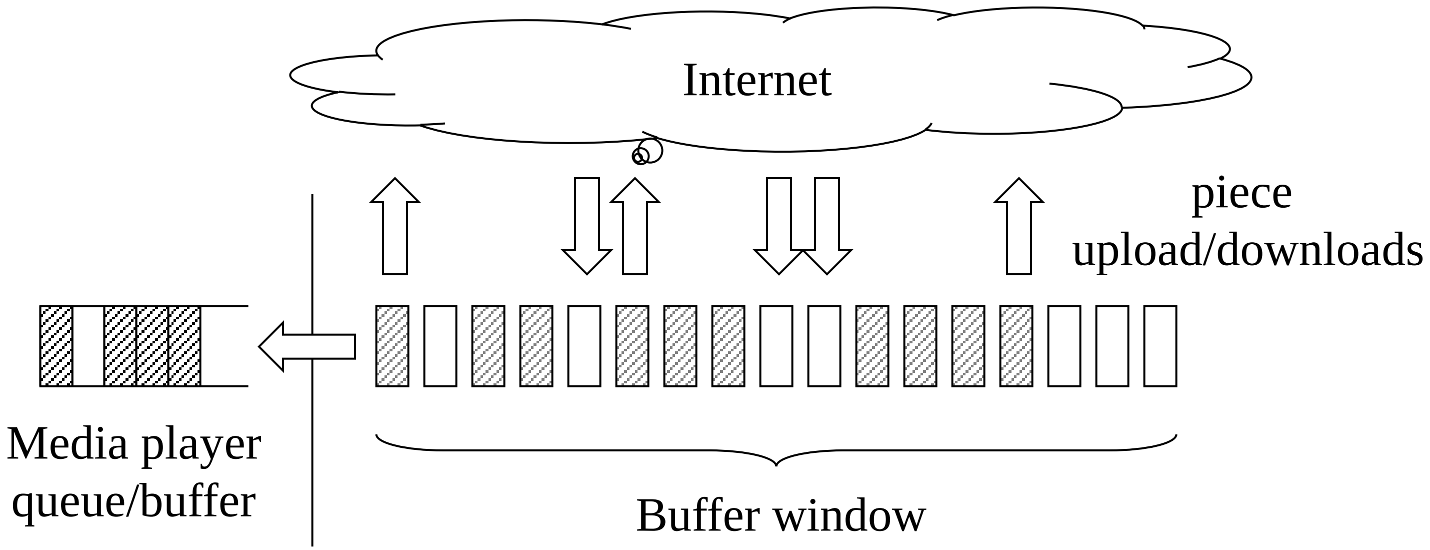


Peer-assisted VoD streaming
Some research questions ...

- Can BitTorrent-like protocols provide scalable on-demand streaming?
- How sensitive is the performance to the application configuration parameters?
  - Piece selection policy (rarest vs. in-order tradeoff)
  - Peer selection policy
  - Upload/download bandwidth
- What is the user-perceived performance?
  - Start-up delay
  - Probability of disrupted playback

ACM SIGMETRICS 2008; IFIP Networking 2007/2009, IEEE/ACM ToN 2012

Live Streaming using BT-like systems



- Live streaming (e.g., CoolStreaming)
  - All peers at roughly the same play/download position
    - High bandwidth peers can easily contribute more …
  - (relatively) Small buffer window
    - Within which pieces are exchanged

Outline

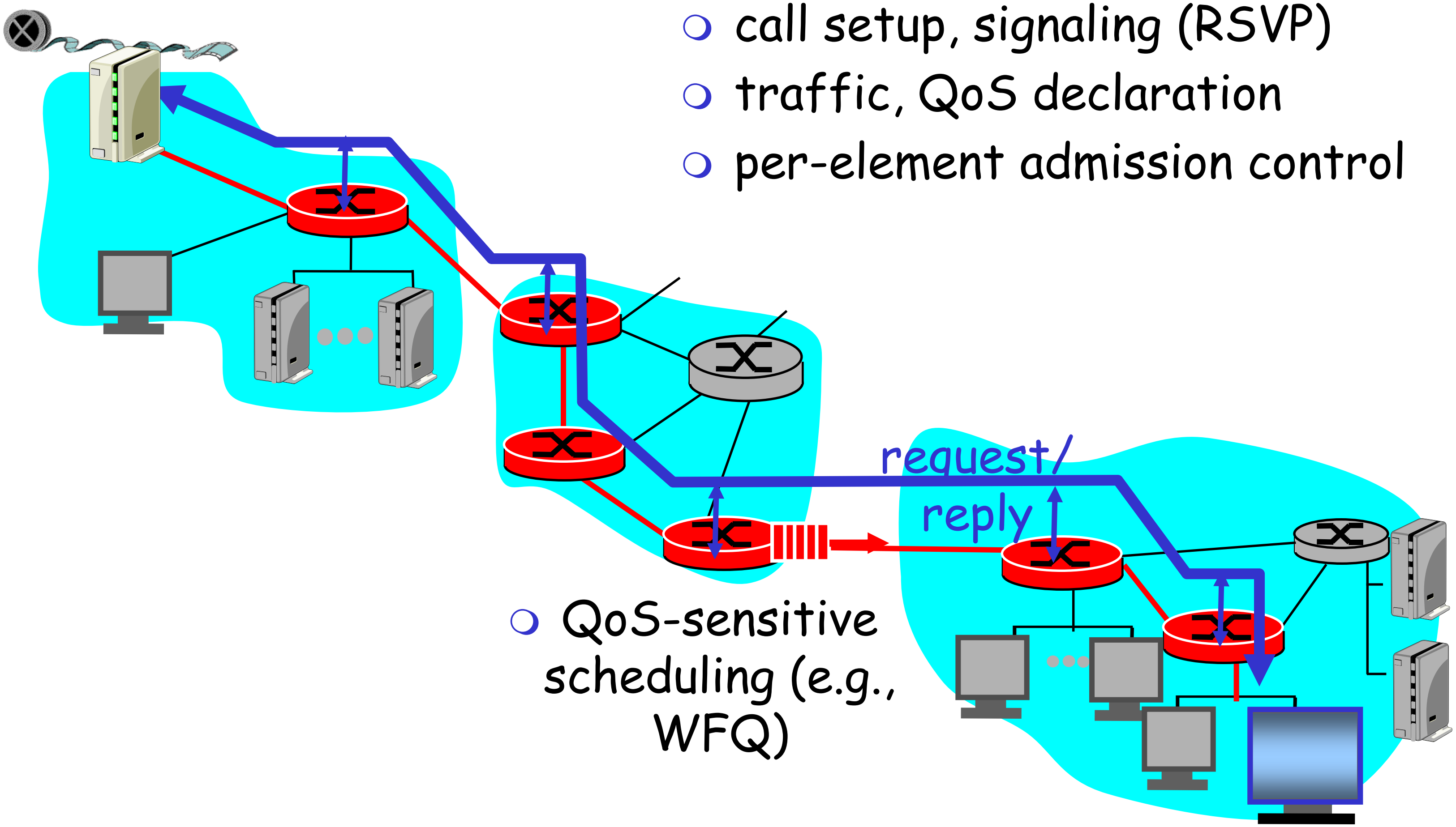
- Multimedia Networking Applications
- Streaming stored audio and video
- Scalable Streaming Techniques
- Content Distribution Networks
- Beyond Best Effort

Integrated Services (IntServ) Architecture

- architecture for providing QOS guarantees in IP networks for individual flows
- flow: a distinguishable stream of distinct IP datagrams
  - Unidirectional
  - Multiple recipient
- Components of this architecture:
  - Admission control
  - Reservation protocol
  - Routing protocol
  - Classifier and route selection
  - Packet scheduler

Intserv: QoS guarantee scenario

- Resource reservation
  - call setup, signaling (RSVP)
  - traffic, QoS declaration
  - per-element admission control



- QoS-sensitive scheduling (e.g., WFQ)

Call Admission

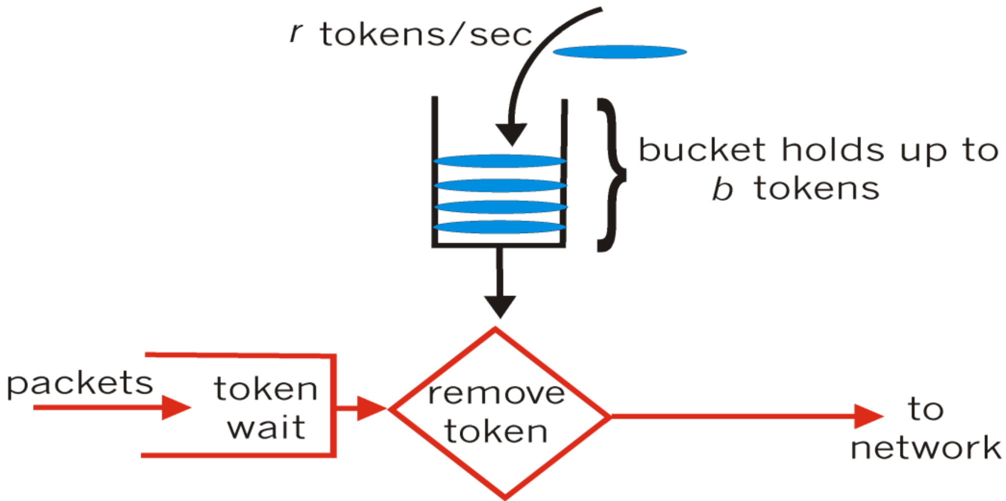
Arriving session must :

- declare its QoS requirement
  - R-spec: defines the QoS being requested
- characterize traffic it will send into network
  - T-spec: defines traffic characteristics
- signaling protocol: needed to carry R-spec and T-spec to routers (where reservation is required)
  - RSVP

Need Scheduling and Policing Policies to provide QoS

Policing: Token Bucket

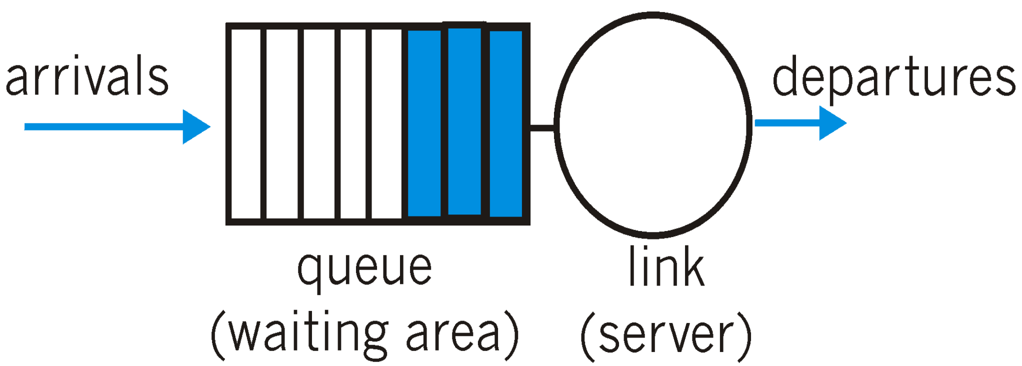
Token Bucket: limit input to specified Burst Size and Average Rate.



- bucket can hold b tokens
- tokens generated at rate *r token/sec* unless bucket full
- *over interval of length t: number of packets admitted less than or equal to (r t + b).*

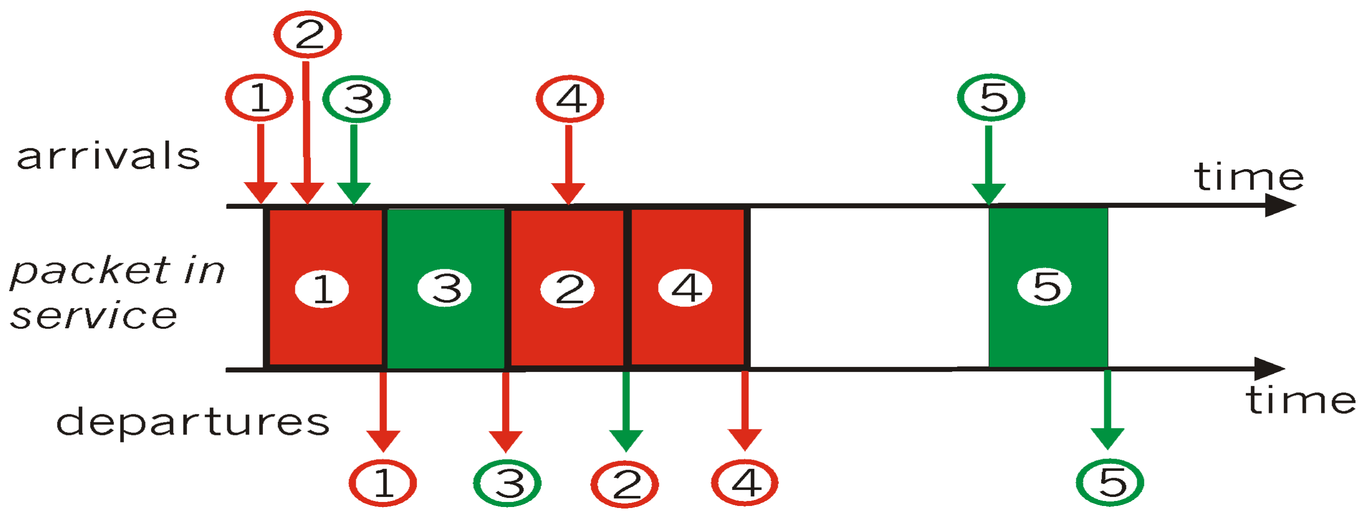
Link Scheduling

- scheduling: choose next packet to send on link
- FIFO (First In First Out) scheduling: send in order of arrival to queue
  - discard policy: if packet arrives to full queue: who to discard?
    - DropTail: drop arriving packet
    - Priority: drop/remove on priority basis
    - Random: drop/remove randomly (e.g., RED)



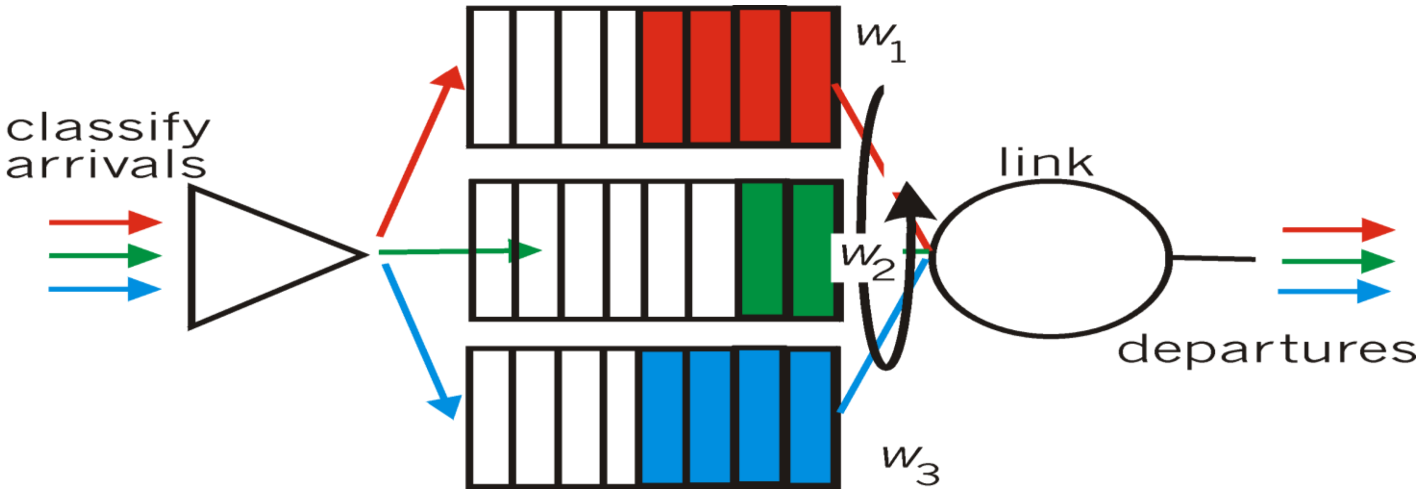
Round Robin

- multiple classes
- cyclically scan class queues, serving one from each class (if available)



Weighted Fair Queuing

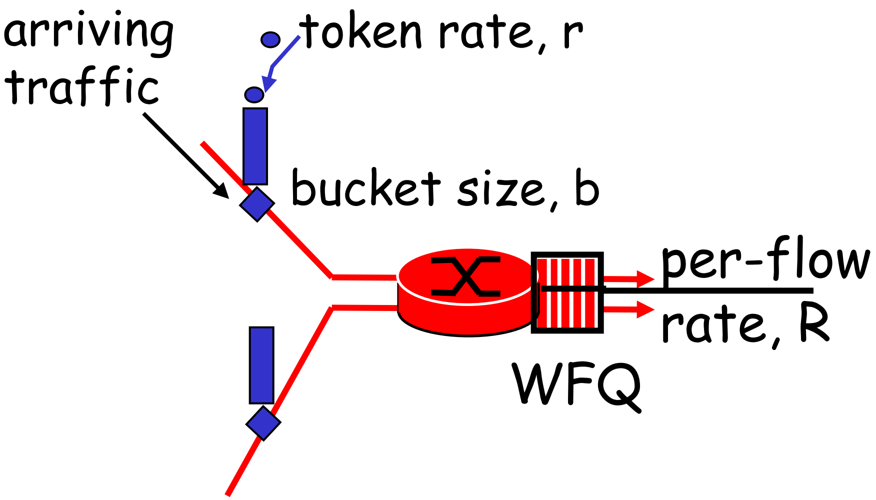
- generalized Round Robin
- each class gets weighted amount of service in each cycle



IntServ QoS: Service models [rfc2211, rfc 2212]

Guaranteed service:

- Assured data rate
- A specified upper bound on queuing delay



$$D_{max} = b/R$$

Controlled load service:

- "a quality of service closely approximating the QoS that same flow would receive from an unloaded network element."
- Similar to behavior best effort service in an unloaded network

Differentiated Services

Concerns with IntServ:

- Scalability: signaling, maintaining per-flow router state difficult with large number of flows
- Flexible Service Models: Intserv has only two classes. Desire "qualitative" service classes
  - E.g., Courier, xPress, and normal mail
  - E.g., First, business, and cattle class ☺

DiffServ approach:

- simple functions in network core, relatively complex functions at edge routers (or hosts)
- Don't define service classes, just provide functional components to build service classes

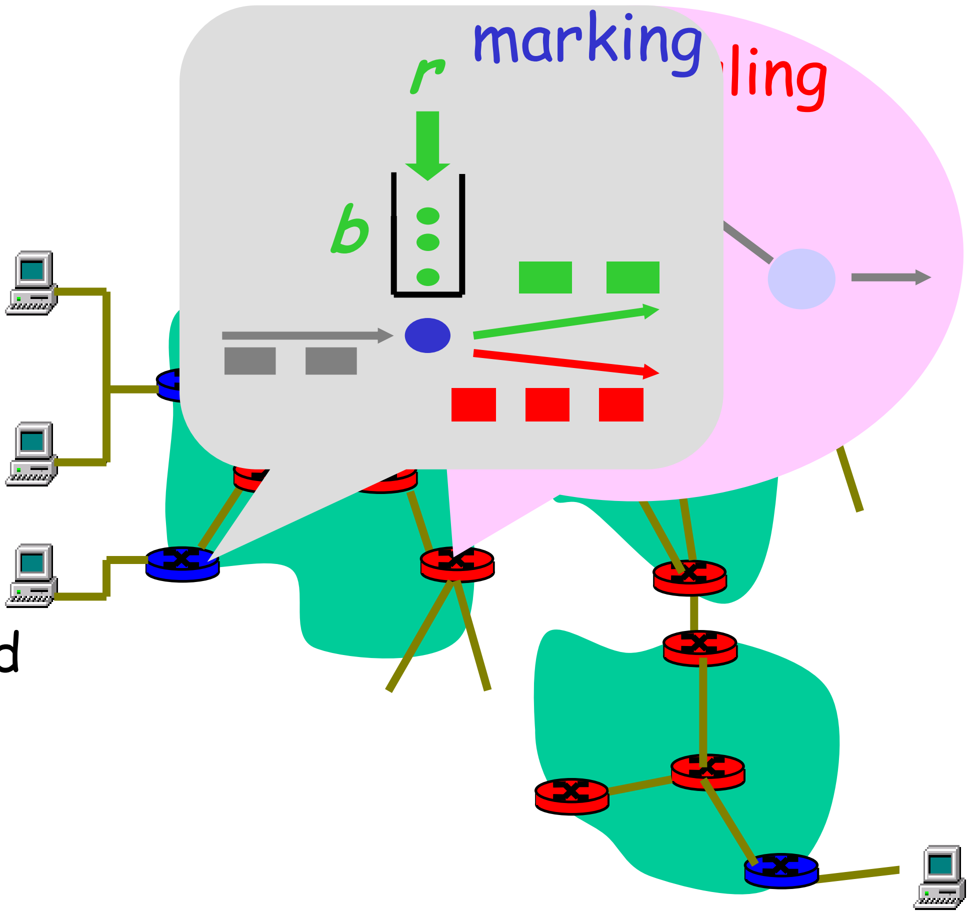
DiffServ Architecture

Edge router:

- per-flow traffic management
- Set the DS field; value determines type of service (PHB: Per-Hop Behavior)

Core router:

- buffering and scheduling based on marking at edge
- per-class traffic management



Traffic Classification/Conditioning

- How can packet markings be carried in IPv4 datagrams?
- Sender may agree to conform to a "traffic profile", thus a leaky bucket policer may be used at the network edge to enforce
  - Peak rate
  - Average rate
  - Burst size
- What happens when traffic profile is violated?
  - Employ traffic shaping?

Deployment Issues

- Single administrative domain
- Incremental deployment
- Traffic policing/shaping complexity
- Charging models

Signaling in the Internet

connectionless (stateless) forwarding by IP routers + best effort service = no network signaling protocols in initial IP design

- New requirement: reserve resources along end-to-end path (end system, routers) for QoS for multimedia applications
- RSVP: Resource reSerVation Protocol [RFC 2205]
  - " … allow users to communicate requirements to network in robust and efficient way." i.e., signaling !
- earlier Internet Signaling protocol: ST-II [RFC 1819]

RSVP Design Goals

1. accommodate heterogeneous receivers (different bandwidth along paths)
2. accommodate different applications with different resource requirements
3. make multicast a first class service, with adaptation to multicast group membership
4. leverage existing multicast/unicast routing, with adaptation to changes in underlying unicast, multicast routes
5. control protocol overhead to grow (at worst) linear in # receivers
6. modular design for heterogeneous underlying technologies

RSVP: does not…

- specify how resources are to be reserved
  - rather: a mechanism for communicating needs
- determine routes packets will take
  - that's the job of routing protocols
  - signaling decoupled from routing
- interact with forwarding of packets
  - separation of control (signaling) and data (forwarding) planes

Multimedia Networking: Summary

- multimedia applications and requirements
- making the best of today's "best effort" service
- scheduling and policing mechanisms
- next generation Internet: IntServ, RSVP, DiffServ, IPv6, IP-QoS