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Chair of Future Communication
Faculty of Computer Science
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050069

VO Netzwerktechnologie für Multimedia Anwendungen

Lecture 4: Multimedia Networking

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Endowed by

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Bachelor Informatik (Medieninformatik)
WS 2011/12

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Chapter 3: Multimedia Networking

Overview:

- 2.1 Multimedia Networking Applications
- 2.2 Streaming stored audio and video
- 2.3 Real-time Multimedia: Internet Phone study
- 2.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP
- 2.5 IP Telefony, SIP, and H.323
- 2.6 Distributing Multimedia: content distribution networks

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Streaming Stored Multimedia

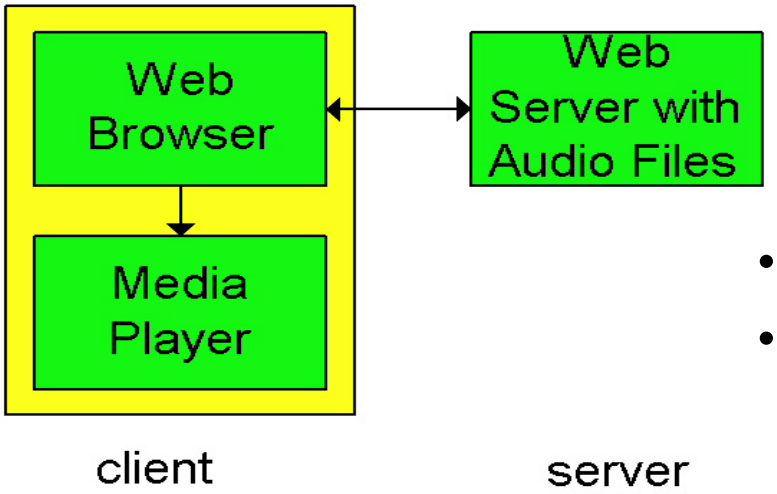
Application-level streaming techniques for making the best out of best effort service:

- 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 not streamed:
no, “pipelining,” long delays until playout!

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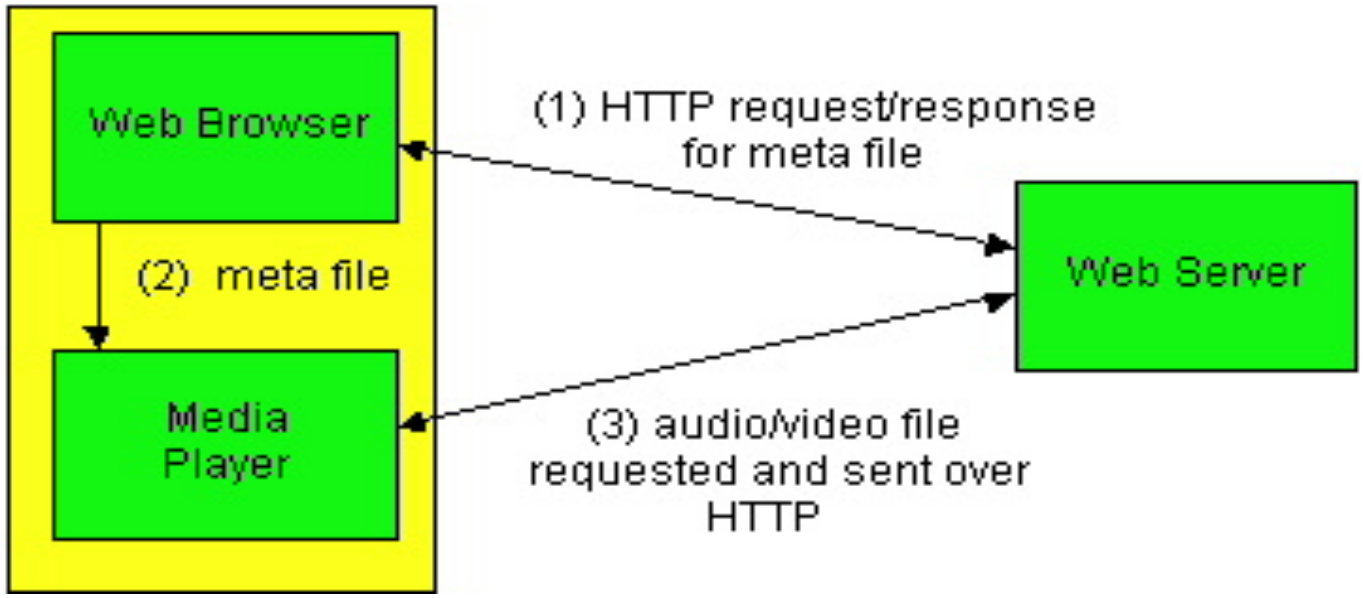
Internet multimedia: streaming approach



- browser GETs metafile
- browser launches player, passing metafile
- player contacts server
- server streams audio/video to player

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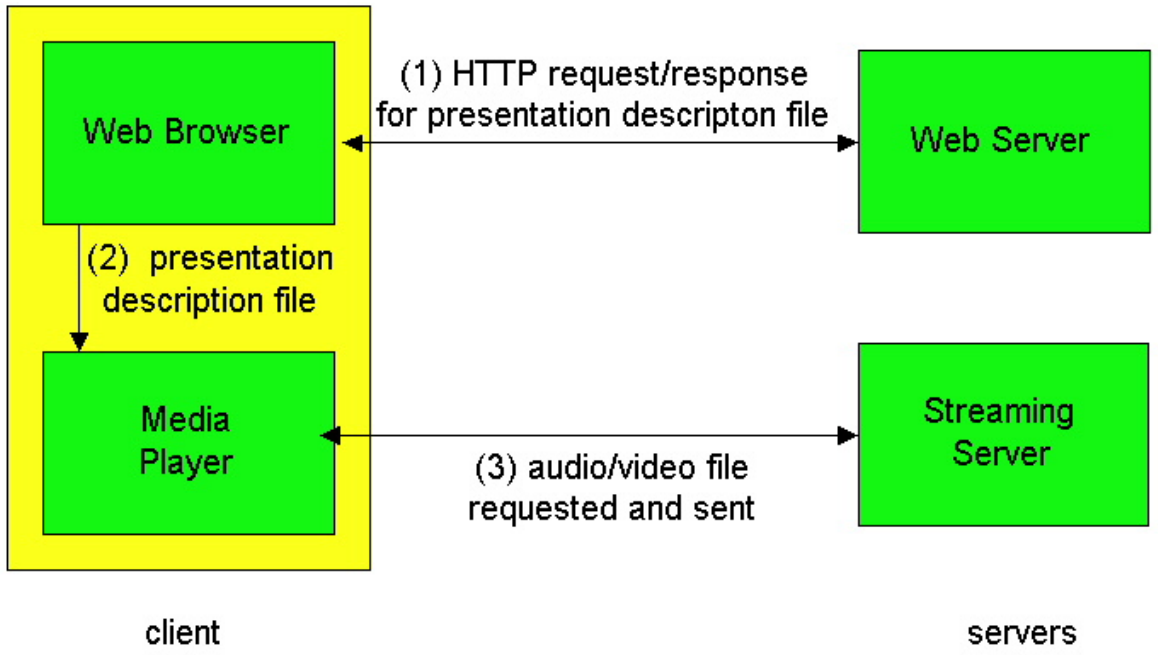
Streaming from a streaming server



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

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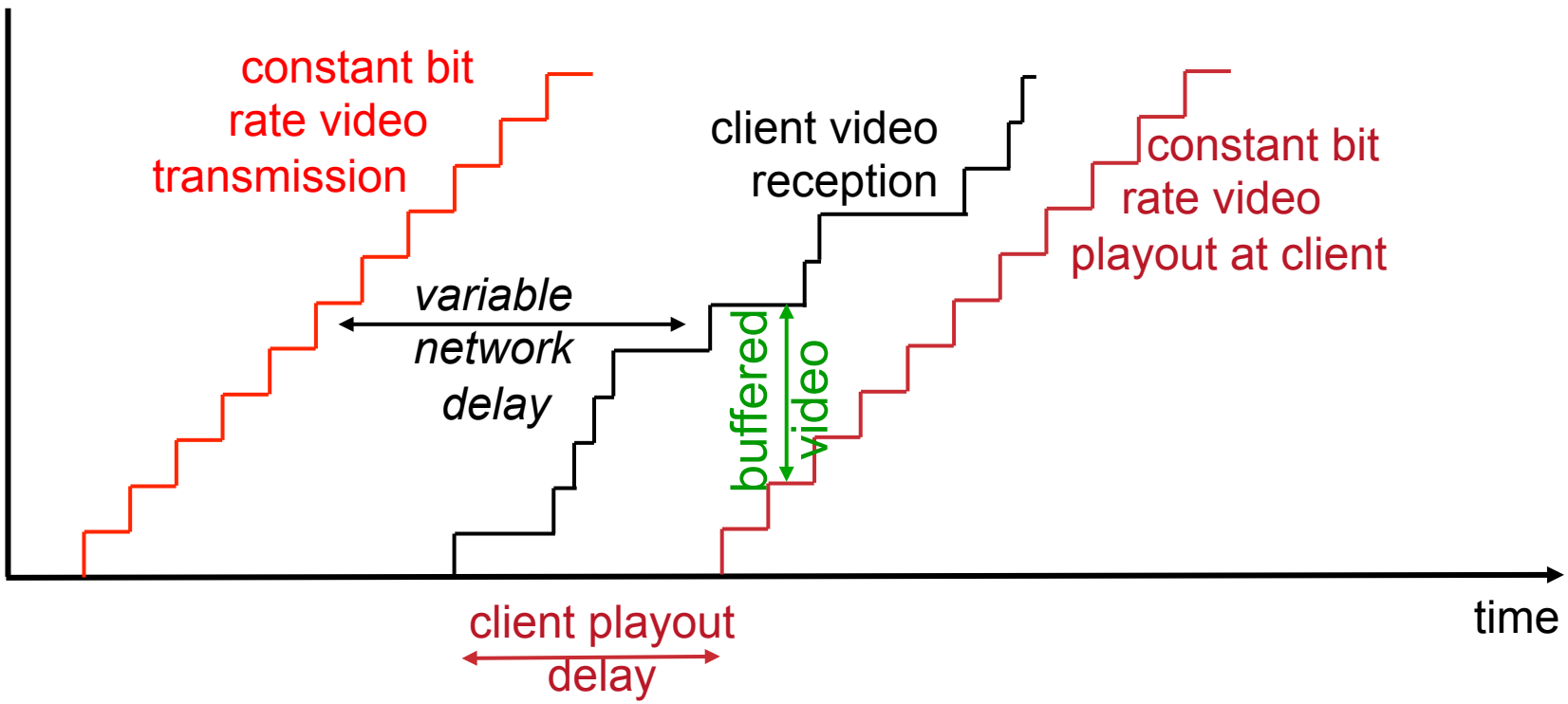
Streaming Multimedia: Client Buffering



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

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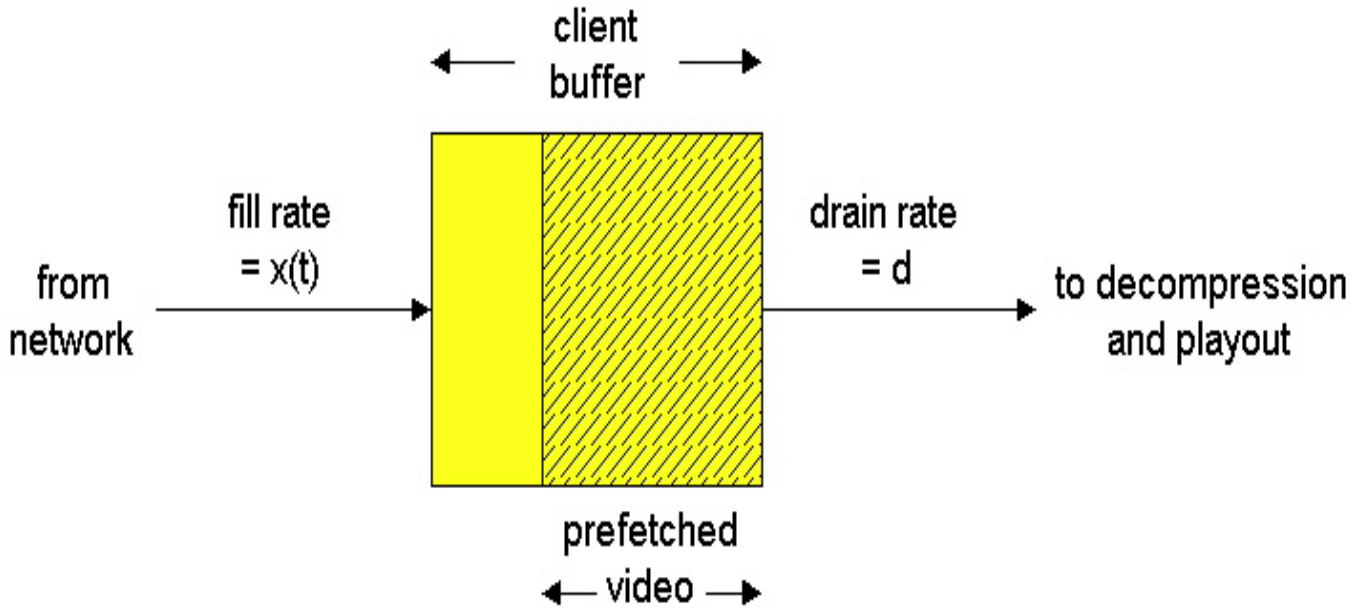
Streaming Multimedia: Client Buffering



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

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Streaming Multimedia: UDP or TCP?

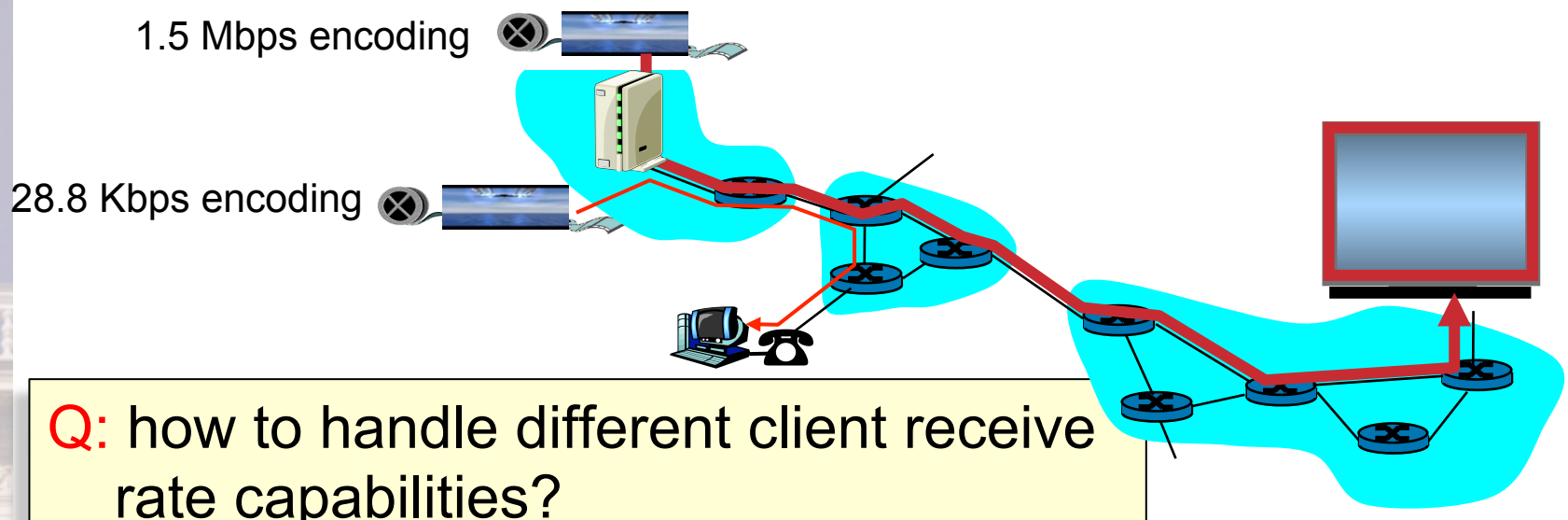
UDP

- 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 recovery if time is 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

Streaming Multimedia: client rate(s)



Q: how to handle different client receive rate capabilities?

- 28.8 Kbps dialup
- 100Mbps Ethernet

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

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User Control of Streaming Media: RTSP

HTTP

- Does not target multimedia content
- No commands for fast forward, etc.

RTSP: RFC 2326

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

What it doesn't do:

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

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RTSP: out of band control

FTP uses an "out-of-band" control channel:

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

RTSP messages are also sent out-of-band:

- RTSP control messages use different port numbers than the media stream: out-of-band.
  - Port 554
- The media stream is considered "in-band".

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RTSP Example

Scenario:

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

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Metafile Example


```
<title>Twister</title>
<session>
        <group language=en lipsync>
                <switch>
                    <track type=audio
                          e="PCMU/8000/1"
                          src = "rtsp://audio.example.com/twister/audio.en/lofi">
                    <track type=audio
                          e="DVI4/16000/2" pt="90 DVI4/8000/1"
                          src="rtsp://audio.example.com/twister/audio.en/hifi">
                 </switch>
               <track type="video/jpeg"
                          src="rtsp://video.example.com/twister/video">
         </group>
</session>
```

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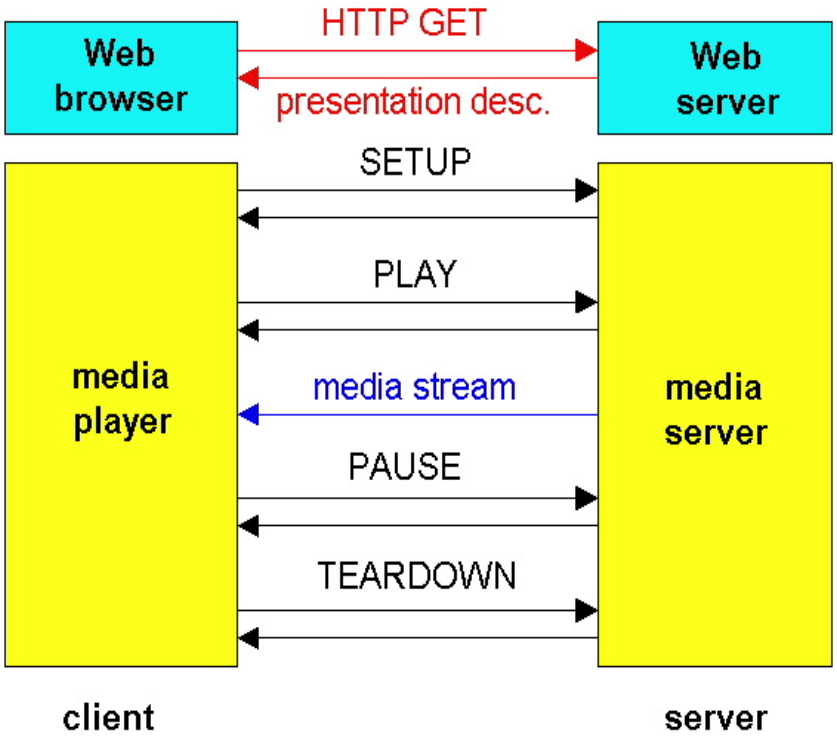
RTSP Operation



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RTSP Exchange Example

C: SETUP rtsp://audio.example.com/twister/audio RTSP/1.0
Transport: rtp/udp; compression; port=3056; mode=PLAY

S: RTSP/1.0 200 1 OK
Session 4231

C: PLAY rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
Session: 4231
Range: npt=0-

C: PAUSE rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
Session: 4231
Range: npt=37

C: TEARDOWN rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
Session: 4231

S: 200 3 OK

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Problems with Interactive Streaming Control

- VCR-control-like interactions desired
  - Play forward
  - Play backward
  - Fast forward
  - Fast backward
  - Set bookmark
  - Jump to bookmark
- Frames required for these actions might be missing
  - ⇒ request and transmission latency
  - ⇒ Buffer and prefetch strategy required
- Assumption
  - Client-pull architecture: clients request frames explicitly

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L/MRP Buffer Management Algorithm

- Problems
  - Buffer space limited
  - MPEG frames have different importance
  - Importance of streamed frames also depends on current playback time
  - Bitstream order vs. display order
- Buffer management algorithm
  - Idea: relevance function depends on
    - Frame type
    - Time distance to playback point / bookmark
  - Least/most relevant for presentation:
    - Request most important frames for presentation
    - Toss least important frames for presentation from buffer

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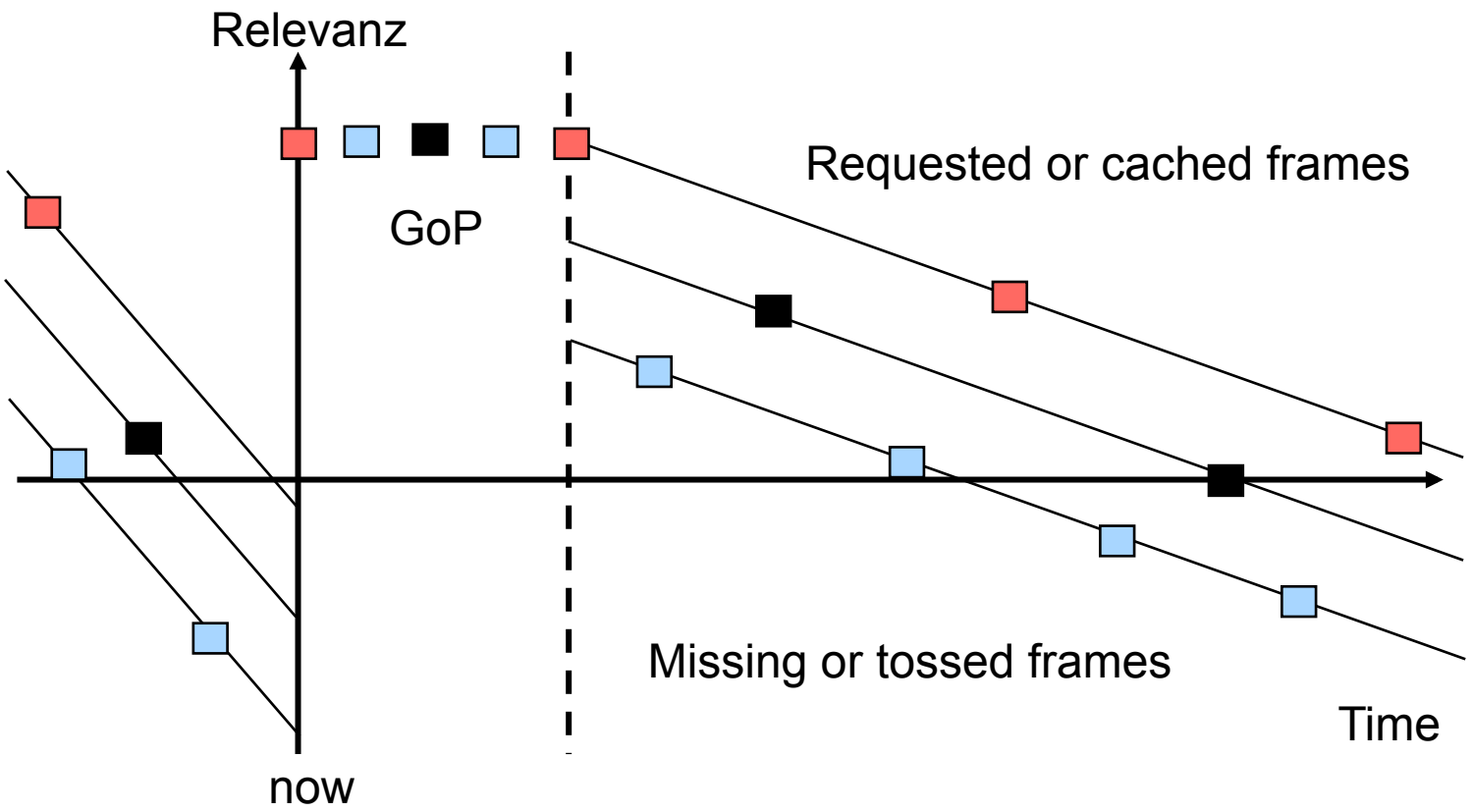
Typical Relevance Function and Memory Occupation



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Implementation of an Interactive Streaming Application

- Problems
  - Requesting too many frames in advance increases reaction time, e.g., to bookmark jumps etc.
    - Already requested frames will be delivered by the server before sending „bookmark“ frames
  - Request only most important frames if bandwidth is insufficient
    - Transmit only I-frames and save bandwidth by retaining P- and B-frames
- Solution: controlled prefetching
  - Requested number of frames (bytes) must not exceed a dynamic threshold
  - Consequences
    - Short response times from server
    - Automatic rate reduction of the stream

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Real-time interactive applications

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- PC-2-PC phone
  - instant messaging services are providing this
- PC-2-phone
  - Dialpad
  - Net2phone
- videoconference with Webcams

Going to now look at a PC-2-PC Internet phone example in detail

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Interactive Multimedia: Internet Phone

Introduce Internet Phone by way of an example

- speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
- pkts generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes data
- application-layer header added to each chunk.
- Chunk+header encapsulated into UDP segment.
- application sends UDP segment into socket every 20 msec during talkspurt.

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Internet Phone: Packet Loss and Delay

- network loss: IP datagram lost due to network congestion (router buffer overflow)
- delay loss: IP datagram arrives too late for playout at receiver
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - typical maximum tolerable delay: 400 ms
- loss tolerance: depending on voice encoding, losses concealed, packet loss rates between 1% and 10% can be tolerated.

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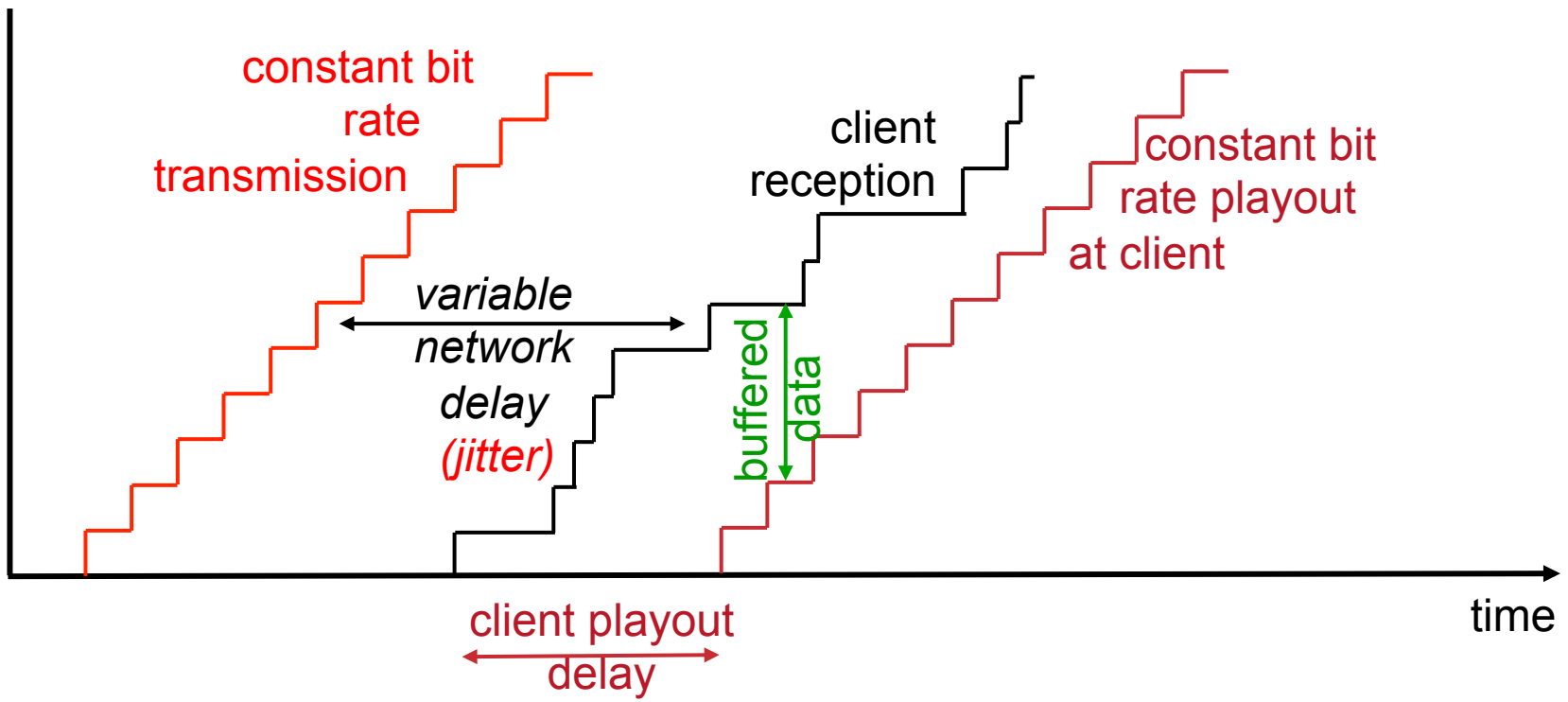
Delay Jitter



- Consider the end-to-end delays of two consecutive packets: difference can be more or less than 20 msec

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Internet Phone: Fixed Playout Delay

- Receiver attempts to playout each chunk exactly q msecs after chunk was generated.
  - chunk has time stamp t: play out chunk at t+q .
  - chunk arrives after t+q: data arrives too late for playout, data “lost”
- Tradeoff for q:
  - large q: less packet loss
  - small q: better interactive experience

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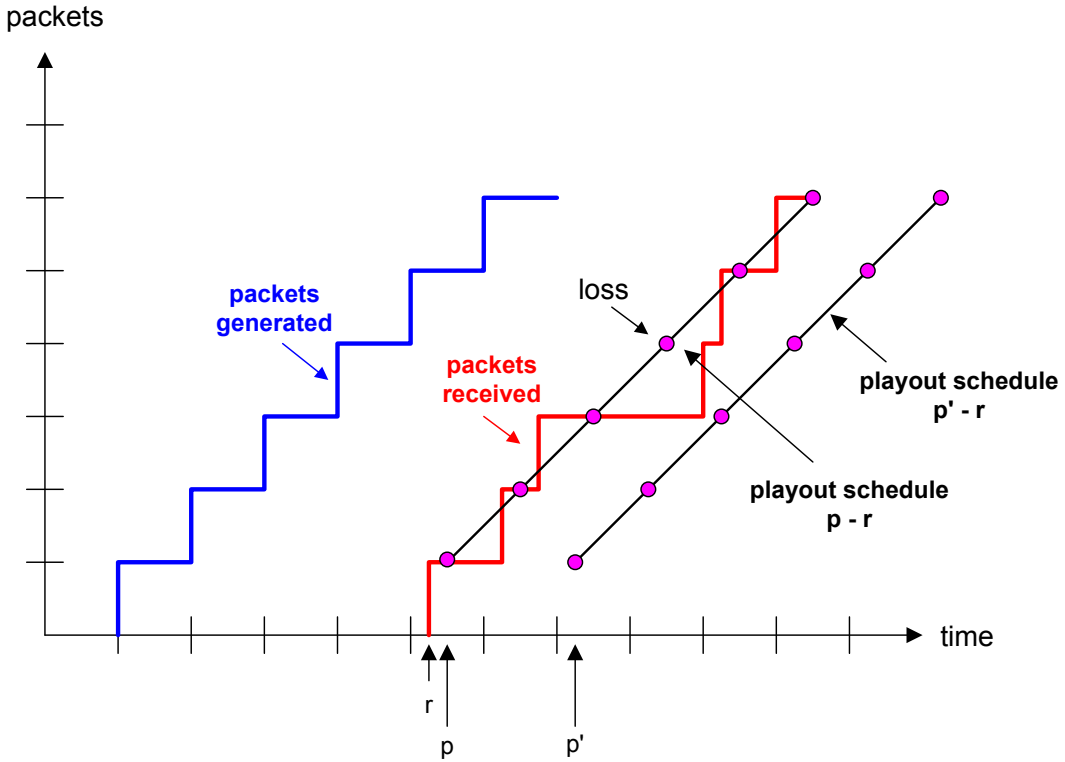
Fixed Playout Delay

- Sender generates packets every 20 msec during talk spurt.
- First packet received at time r
- First playout schedule: begins at p
- Second playout schedule: begins at p’



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Adaptive Playout Delay, I

- Goal: minimize playout delay, keeping late loss rate low
- Approach: adaptive playout delay adjustment:
  - Estimate network delay, adjust playout delay at beginning of each talk spurt.
  - Silent periods compressed and elongated.
  - Chunks still played out every 20 msec during talk spurt.

t_i=timestamp of the ith packet

r_i=the time packet i is received by receiver

p_i=the time packet i is played at receiver

$r_i - t_i$=network delay for ith packet

d_i=estimate of average network delay after receiving ith packet

Dynamic estimate of average delay at receiver:

$$d_i = (1-u)d_{i-1} + u(r_i - t_i)$$

where u is a fixed constant (e.g., $u = .01$).

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Adaptive playout delay II

Also useful to estimate the average deviation of the delay, *vi* :

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

The estimates *di* and *vi* are calculated for every received packet, altho they are only used at the beginning of a talk spurt.

For first packet in talk spurt, playout time is:

$$p_i = t_i + d_i + Kv_i$$

where K is a positive constant.

Remaining packets in talkspurt are played out periodically

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Adaptive Playout, III

Q: How does receiver determine whether packet is first in a talkspurt?

- If no loss, receiver looks at successive timestamps.
  - difference of successive stamps > 20 msec -->talk spurt begins.
- With loss possible, receiver must look at both time stamps and sequence numbers.
  - difference of successive stamps > 20 msec and sequence numbers without gaps --> talk spurt begins.

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Recovery from packet loss (1)

forward error correction (FEC): simple scheme

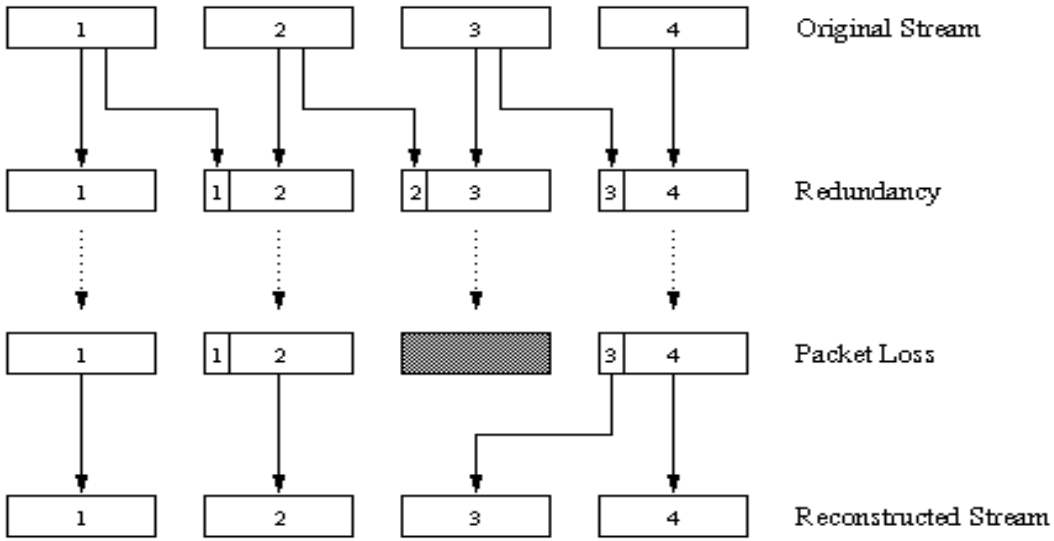
- for every group of n chunks create a redundant chunk by exclusive OR-ing the n original chunks
- send out n+1 chunks, increasing the bandwidth by factor 1/n.
- can reconstruct the original n chunks if there is at most one lost chunk from the n+1 chunks

- Playout delay needs to be fixed to the time to receive all n+1 packets
- Tradeoff:
  - increase n, less bandwidth waste
  - increase n, longer playout delay
  - increase n, higher probability that 2 or more chunks will be lost

Recovery from packet loss (2)

2nd FEC scheme

- "piggyback lower quality stream"
- send lower resolution audio stream as the redundant information
- for example, nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps.



- 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

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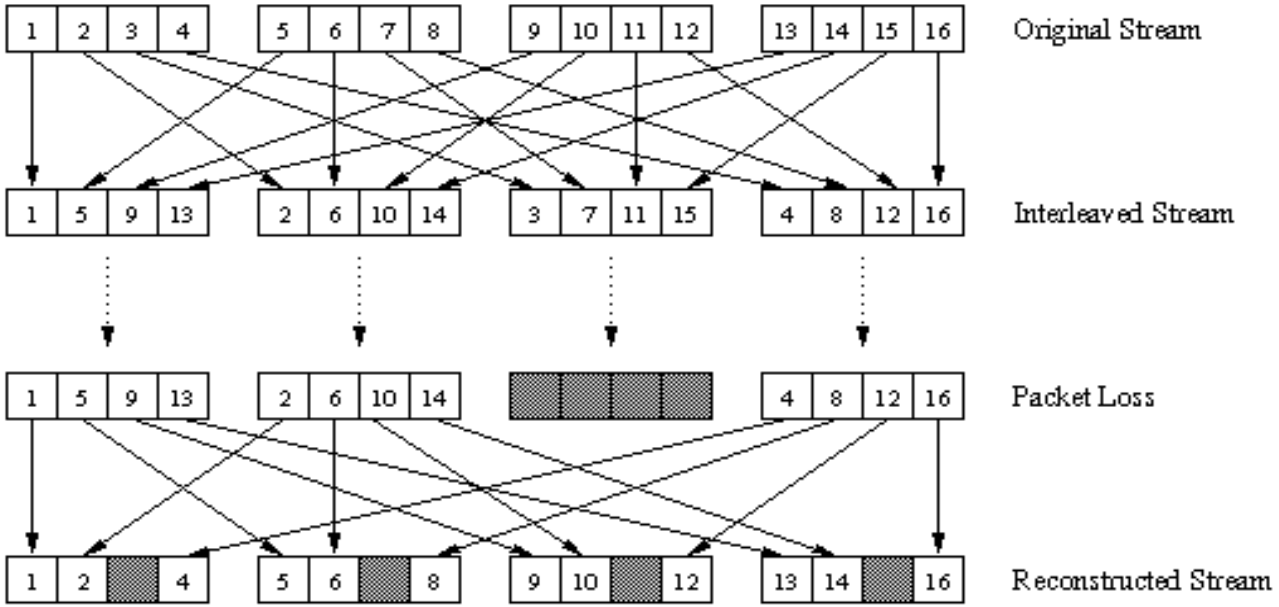
Recovery from packet loss (3)



Interleaving

- chunks are broken up into smaller units
- for example, 4 5 msec units per chunk
- Packet contains small units from different chunks
- if packet is lost, still have most of every chunk
- has no redundancy overhead
- but adds to playout delay

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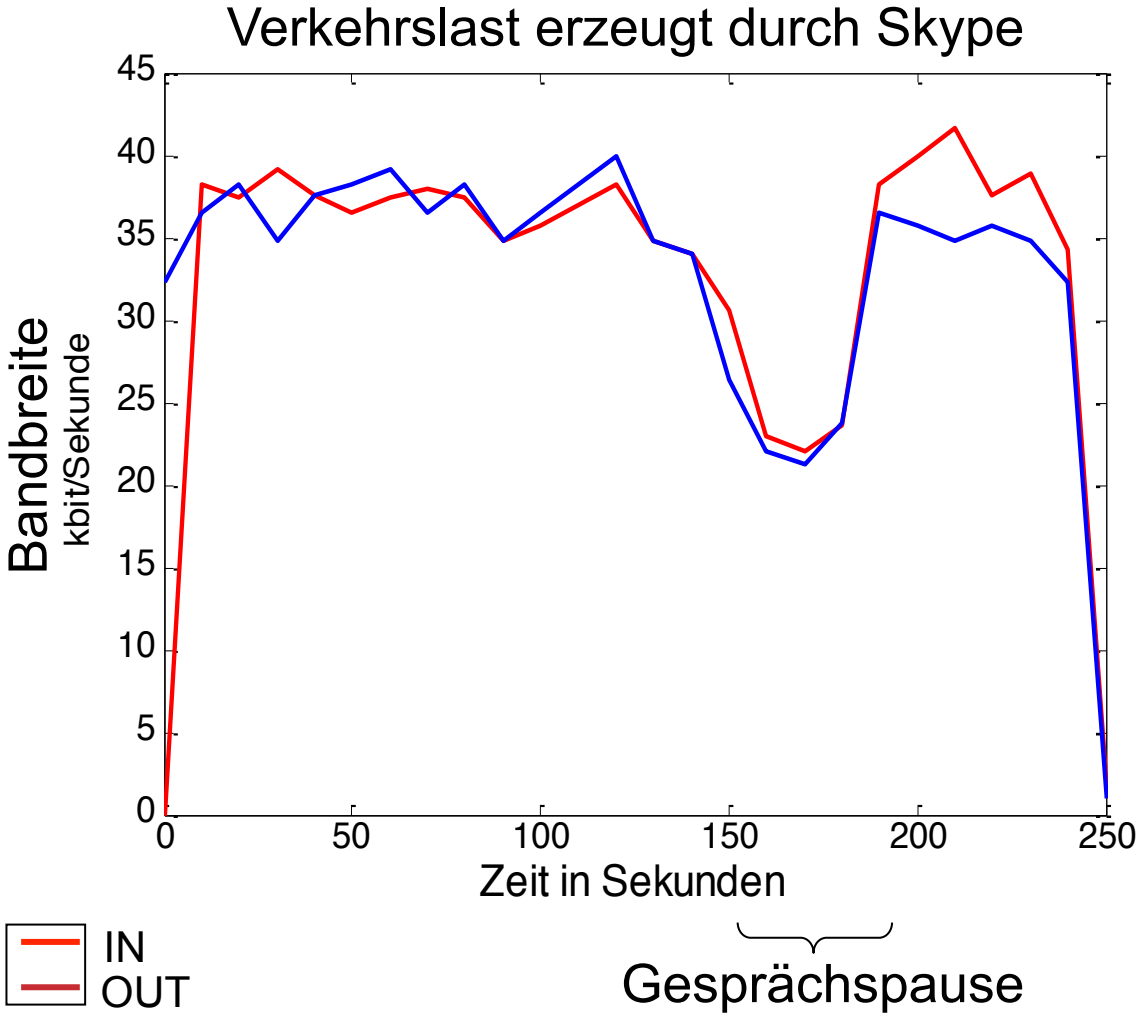
Bandbreitenmessung VoIP (Skype)



Mittelwert:
IN: 32,4 kbit/s
OUT: 32,6 kbit/s

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Bandbreitenmessung VoIP (SIP)

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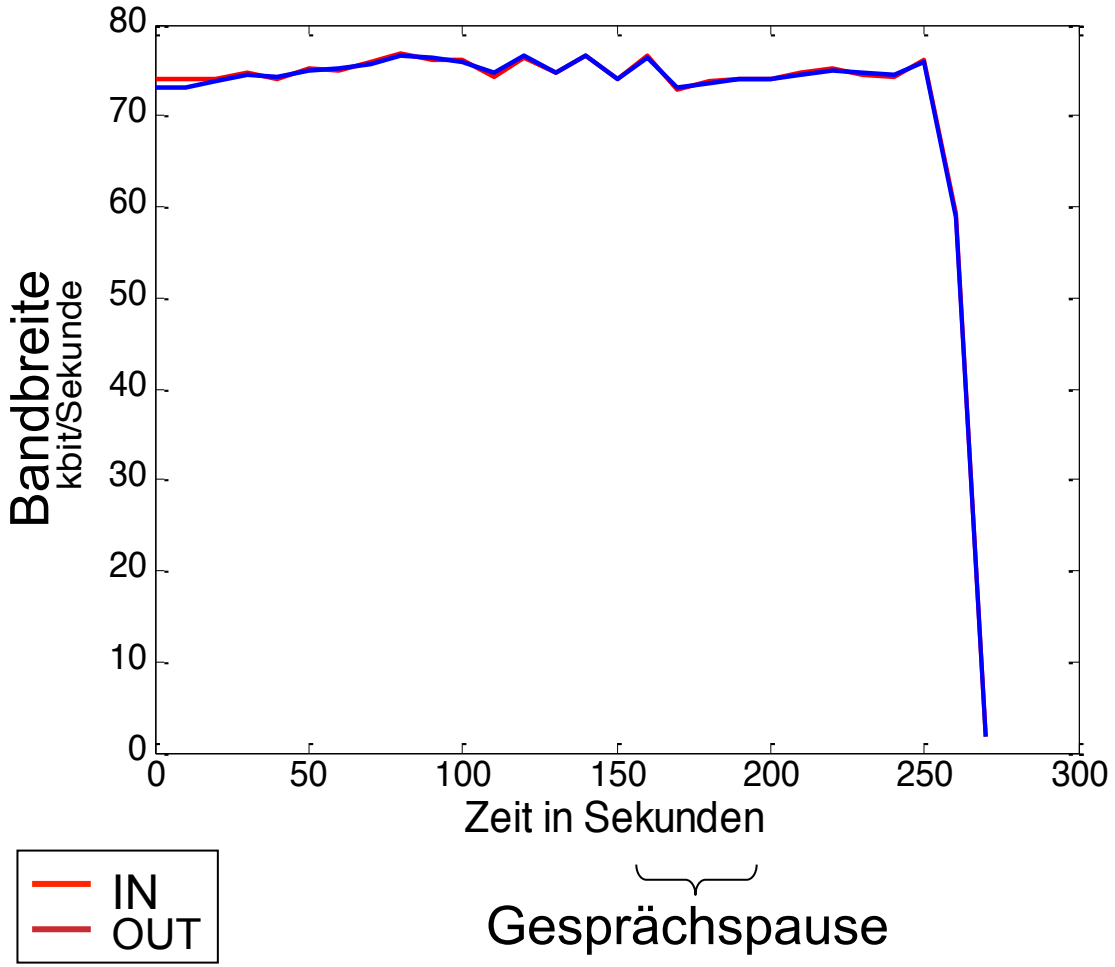
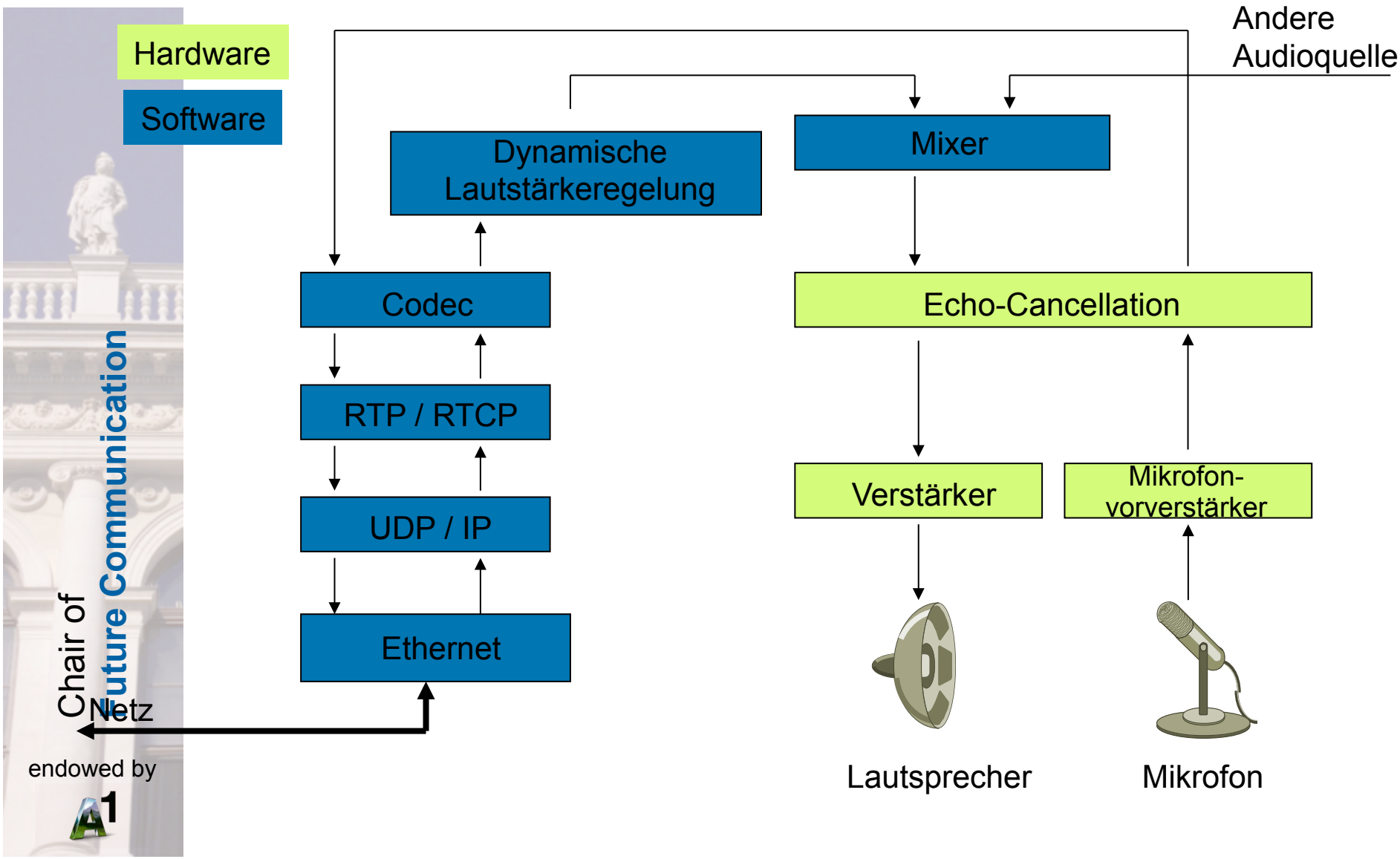
Verkehrslast erzeugt durch SIP (Codec: G711u)



Mittelwert:
IN: 71,8 kbit/s
OUT: 71,6 kbit/s

Lokale Enflussfaktoren - Beispiel PC



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Anforderungen Sprachqualität

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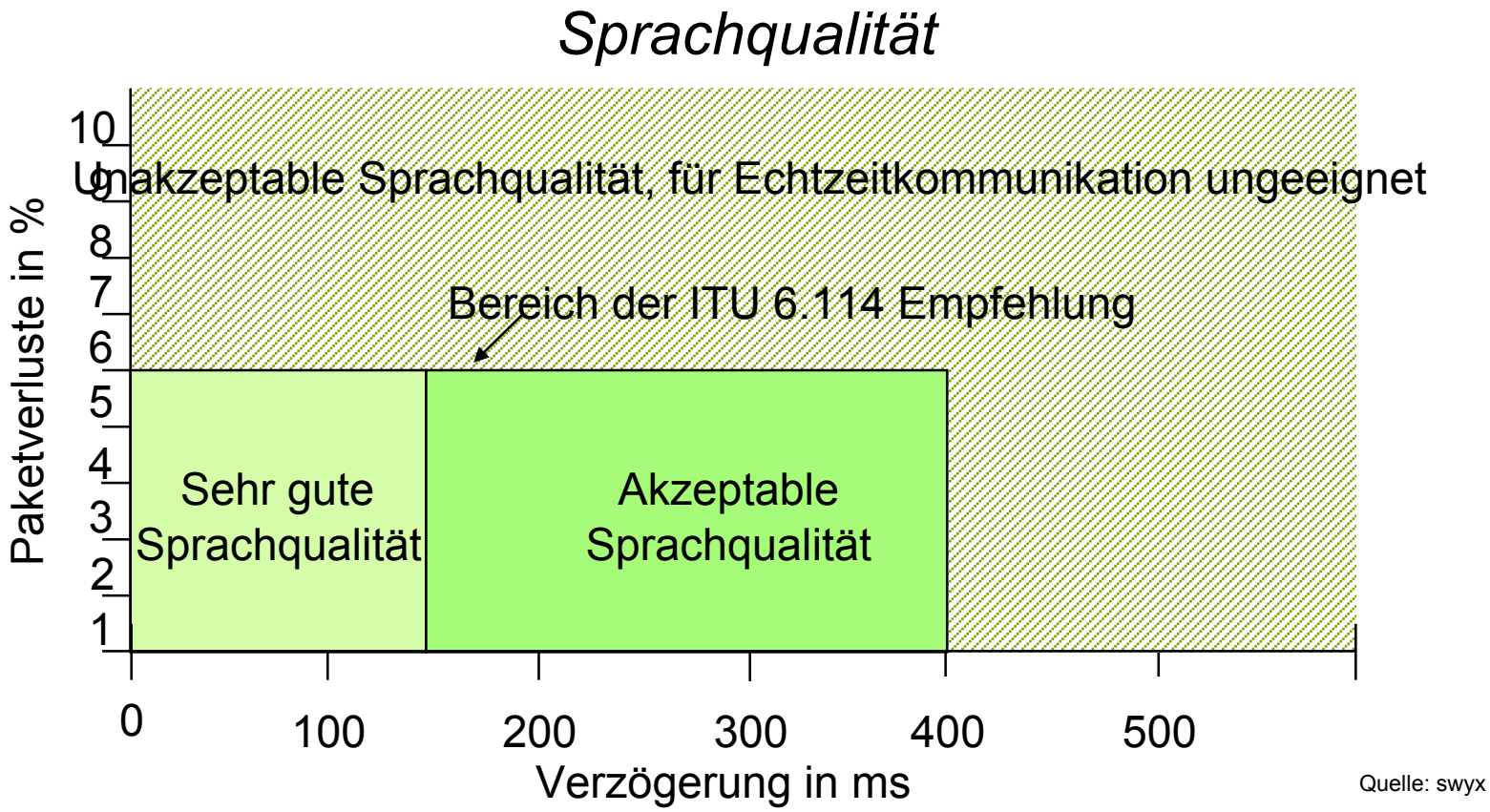
- Die Qualität der Sprache in IP-Netzen hängt wesentlich von den Paketverlusten und von der Verzögerung ab.



Quelle: swyx

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Bewertung der Übertragungsgüte

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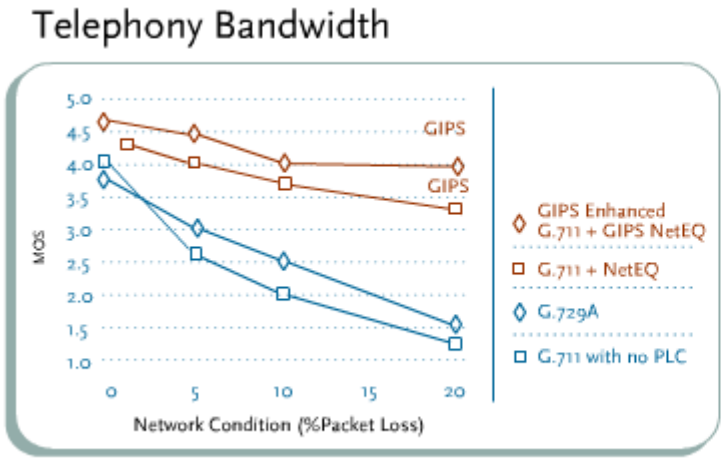
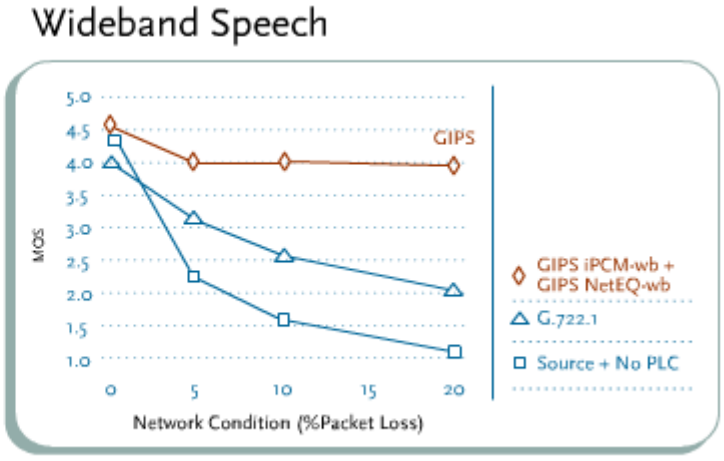
- Bewertung der subjektiven Übertragungsqualität
  - Übertragung einer vorgegebenen Audiodatei
  - Aufzeichnung am Empfänger
  - Vergleich mittels eines standardisierten Algorithmus ergibt:
    - Mean Opinion Score (MOS)

Sehr gute Sprachqualität in leiser Umgebung	Excellent	5.0
Natürliche Sprachqualität wie digitales Telefon	Good	4.0
Akzeptabel, erfordert aber teilweise Konzentration	Fair	3.0
Schwer zu verstehende Sprache	Poor	2.0
Kaum zu verstehen, Unterbrechungen	Bad	1.0

Skype - Sprachqualität

- Einsatz eines speziellen Voicecodecs
  - Global IP Sound





  - Reagiert adaptiv auf Bandbreitenveränderungen
  - Kann das Verhältnis Prozessor-/Bandbreitenlast optimieren

Quelle: www.globalipsound.co

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Summary: Internet Multimedia: bag of tricks

- use UDP to avoid TCP congestion control (delays) for time-sensitive traffic
- client-side adaptive playout delay: to compensate for delay
- server side matches stream bandwidth to available client-to-server path bandwidth
  - chose among pre-encoded stream rates
  - dynamic server encoding rate
- error recovery (on top of UDP)
  - FEC, interleaving
  - retransmissions, time permitting
  - conceal errors: repeat nearby data