Multimedia protocols

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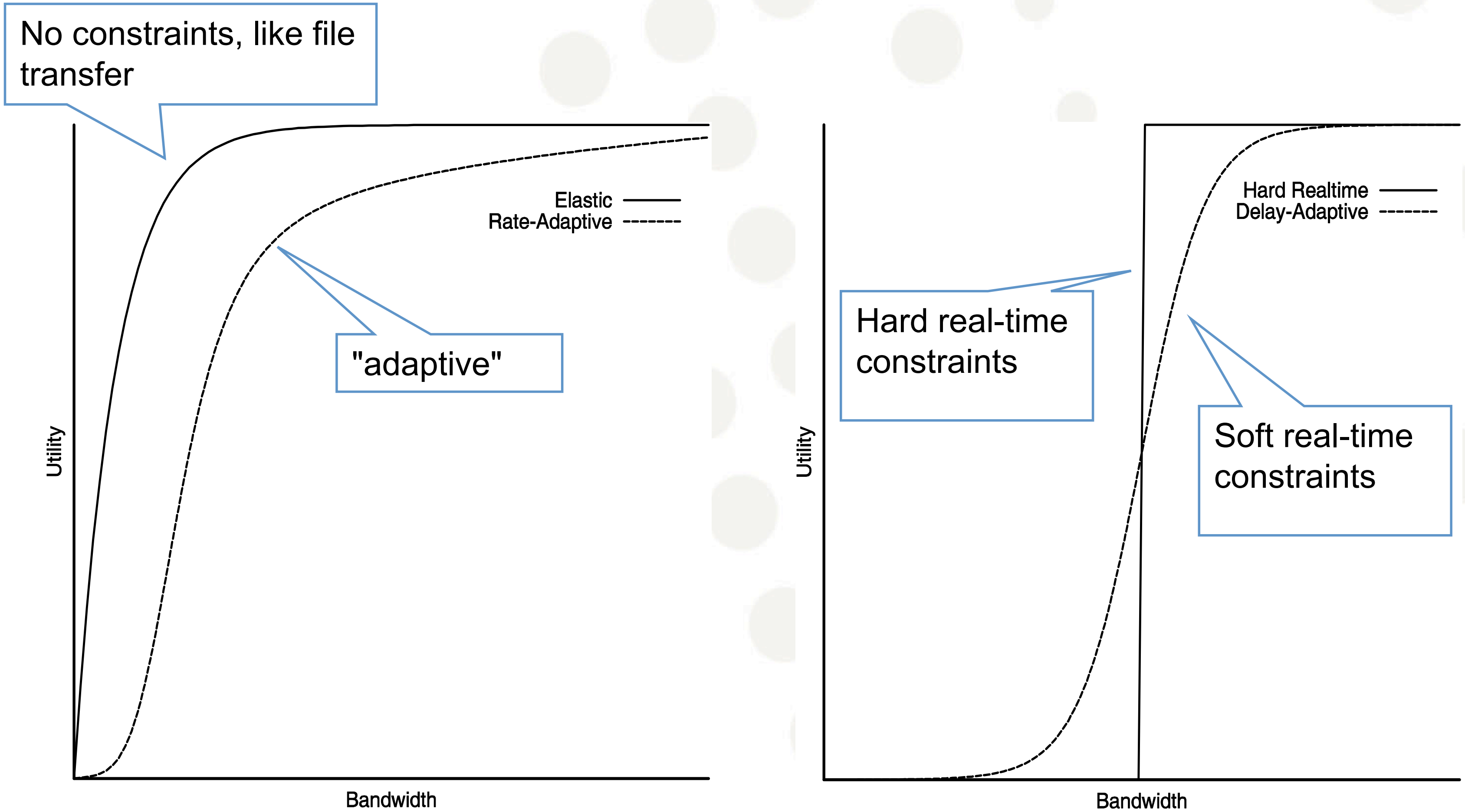
Multimedia

- Combined use of more content forms: text, graphics, audio, video, ...
  – Networks context: multimedia usually means that audio and/or video are used
- Only real-time multimedia of interest
  – Downloading a movie is not much different from downloading a large piece of software (but, note: it's large)
  – Here, "Real-time" means soft real-time
- Requirements differ:
  – one-way streaming media: compensate network fluctuations by buffering; buffer size → initial delay + time lag (can be bad for live TV broadcasts...)
  – interactive application: buffer size → delay during usage
  – Often, timely is more important than reliable delivery → avoid retransmissions

Characterizing multimedia streams

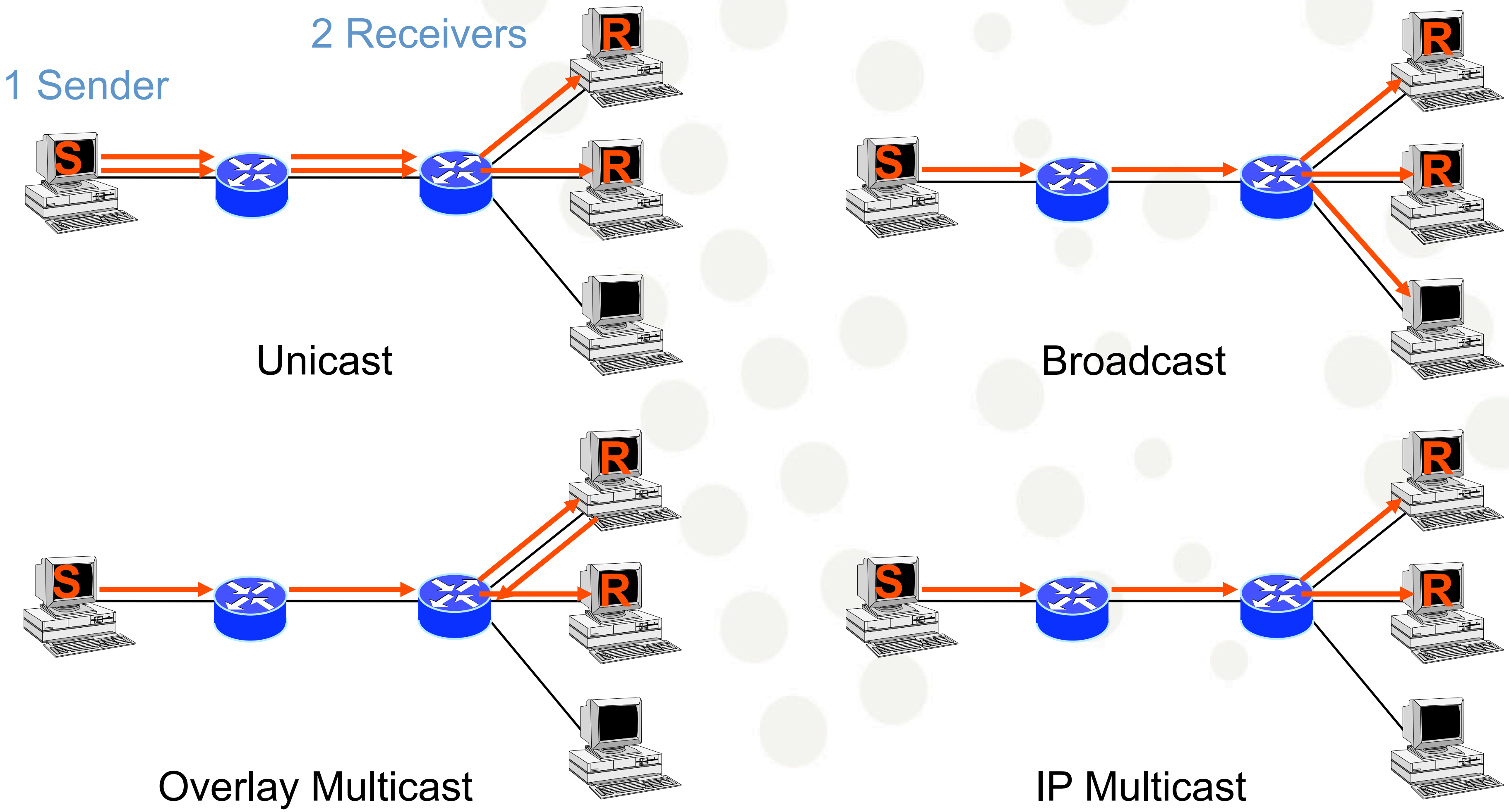


Quality of Service (QoS)

- How to support multimedia bandwidth / delay requirements:
  - use special network mechanisms that can do it (QoS)
  - or dimension the network accordingly

- Both approaches cost money
  - Dimensioning: usually less. It's also less risky...
  - Internet QoS was once a big thing (because of notion: "value-added services" = more money), but is now a history lesson
  - So we end it here ☺ and assume a non-QoS-Internet from now on
  - Note: perfectly dimensioned networks are also not assumed: not very interesting (and not always possible – e.g. WiFi)
    - Remember, multimedia content is large; there is never a "good enough"

Transmission modes



Multicast issues

- Required for applications with multiple receivers only
  - video conferences, real-time stream transmission (e.g. radio, TV), ..

- Issues:
  - group management
    - protocol required to dynamically join / leave group:
      Internet Group Management Protocol (IGMP)
    - state in routers: hard / soft (lost unless refreshed)?
    - who initiates / controls group membership?
  - congestion control
    - scalability (ACK implosion), dealing with receiver heterogeneity, fairness

- Multicast congestion control mechanism classification:
  - sender- vs. receiver-based, single-rate vs. multi-rate (layered),
  - reliable vs. unreliable, end-to-end vs. network-supported

depends on content!

Multimedia content fluctuates

- This is natural: sometimes we talk, sometimes we don't, sometimes we move, sometimes we don't.
  - exploited by compression schemes
  - Necessary to cope with size of multimedia content

- Typical values:

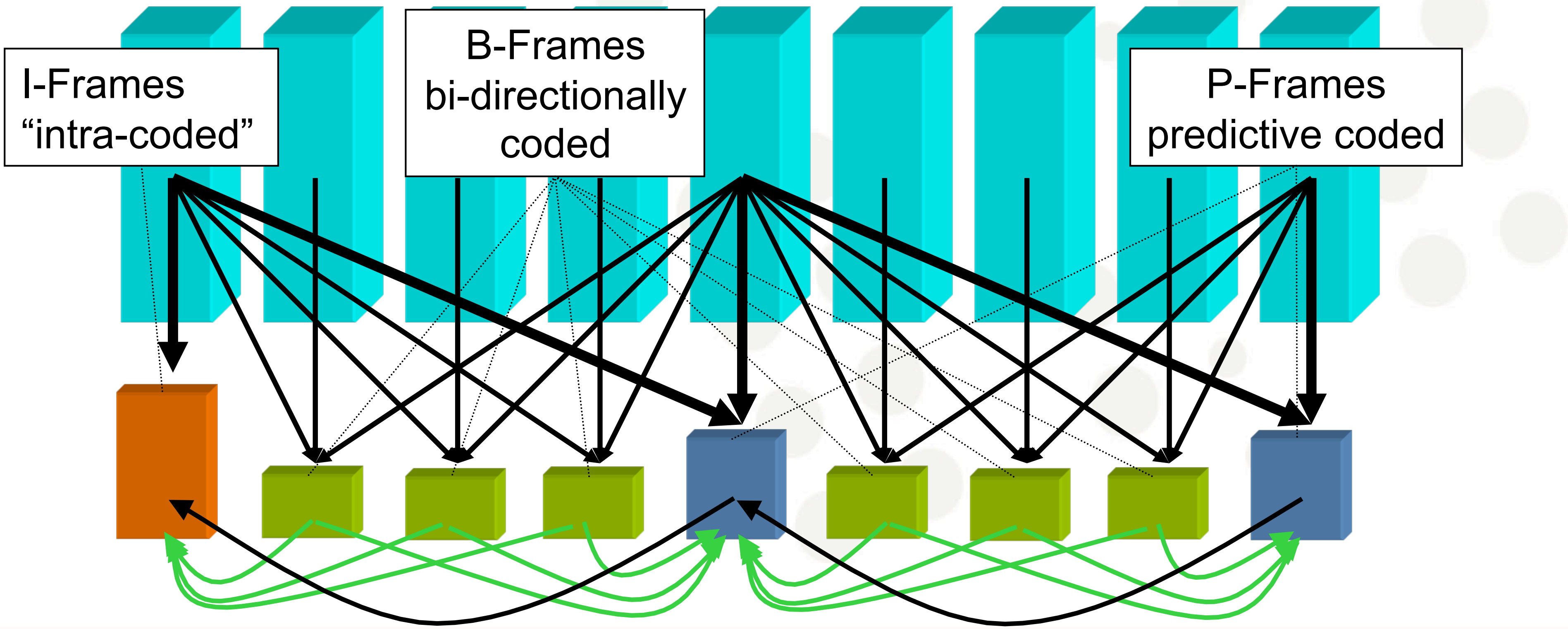
  Uncompressed
    - video: 140 – 216 Mbit/s; audio (CD): 1.4 Mbit/s; speech: 64 Kbit/s

  Compressed audio & video:
    - VOD: down to 1.2 – 4 Mbit/s; Conf.: down to 128 Kbit/s

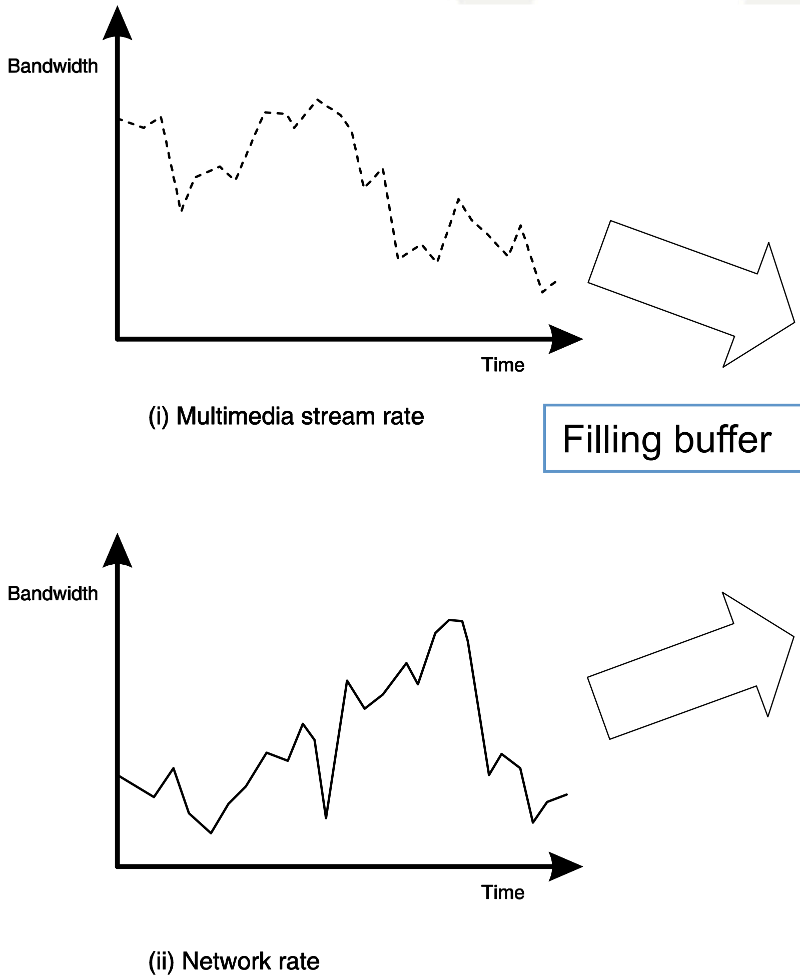
  - Compressed speech: down to 6.2 Kbit/s

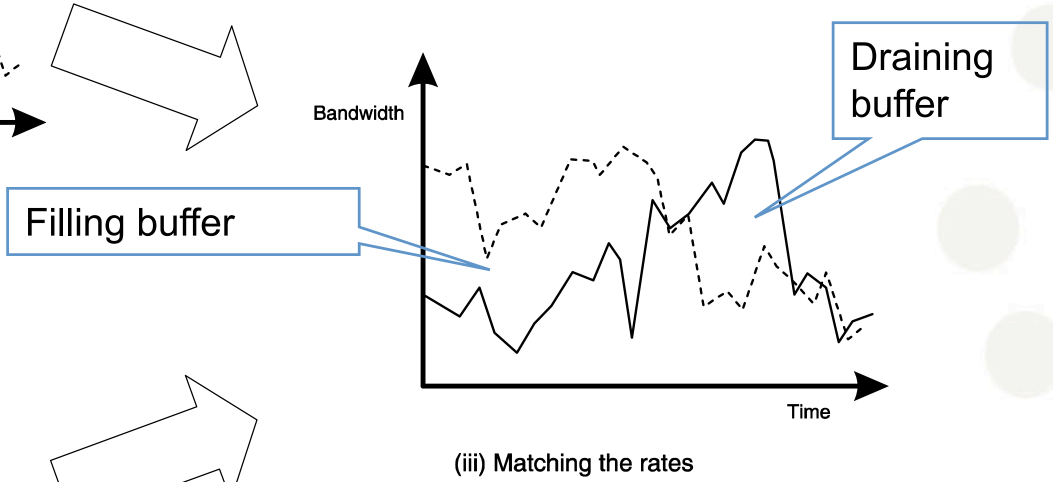
Example: MPEG-1

- International Standard: Moving Pictures Expert Group
  - Compression of audio and video for playback (1.5 Mbit/s), real-time decoding
- Sequence of I-, P-, and B-Frames



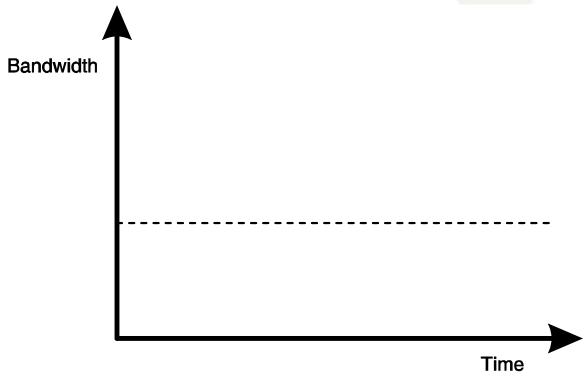
Matching stream and network rates



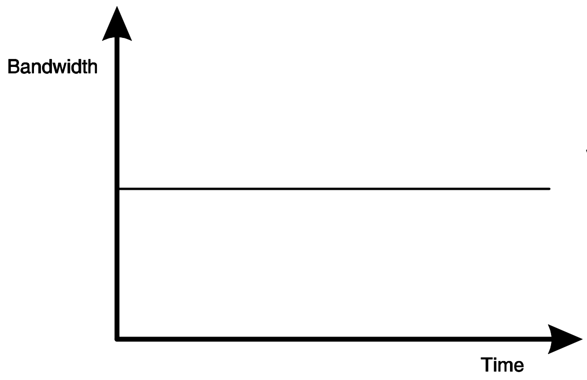


- Works if lucky, and buffer large enough
- Large buffer ⇔ interactivity

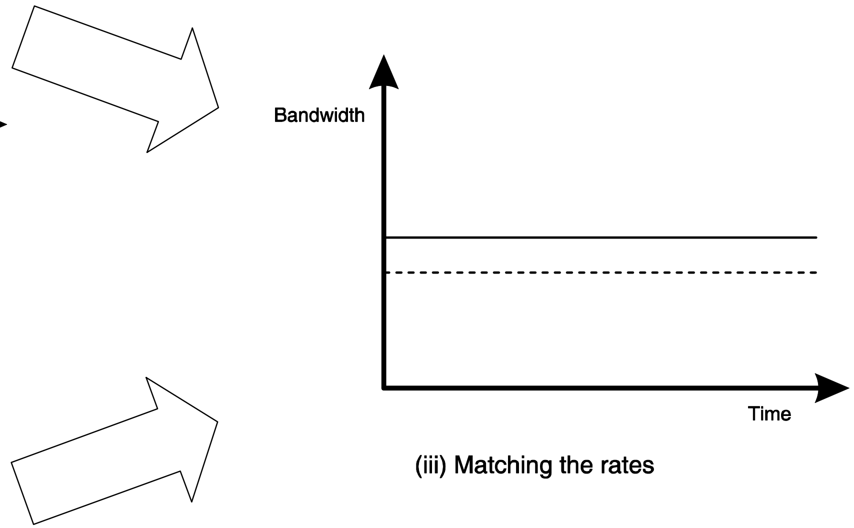
Matching stream and network rates /2



(i) Multimedia stream rate



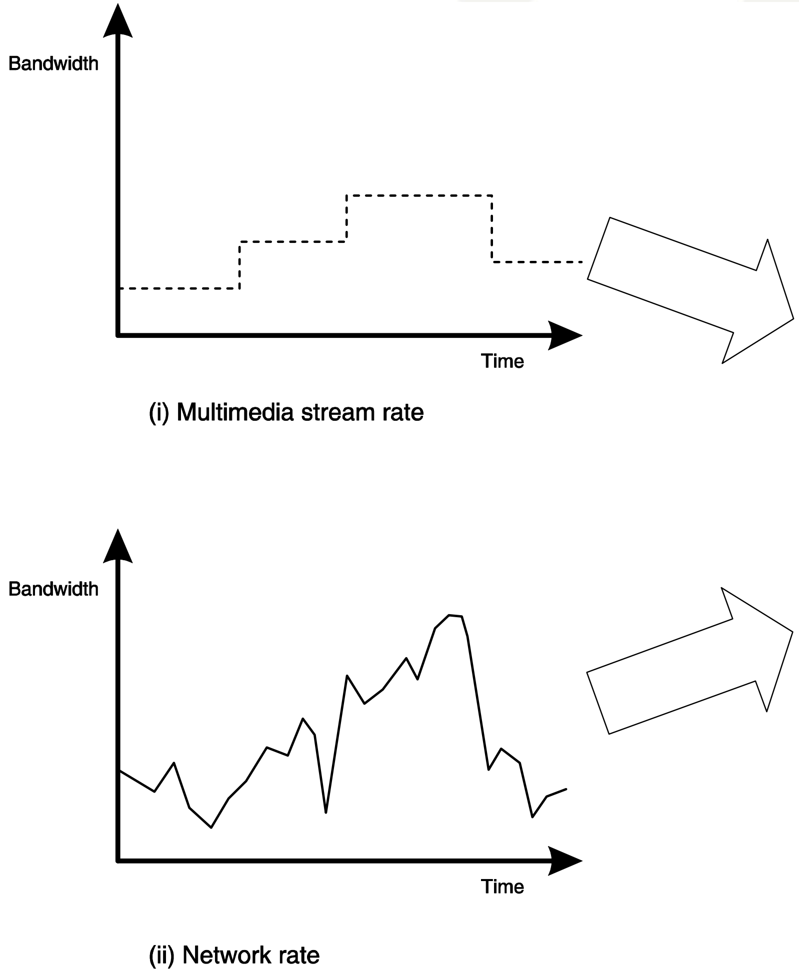
(ii) Network rate

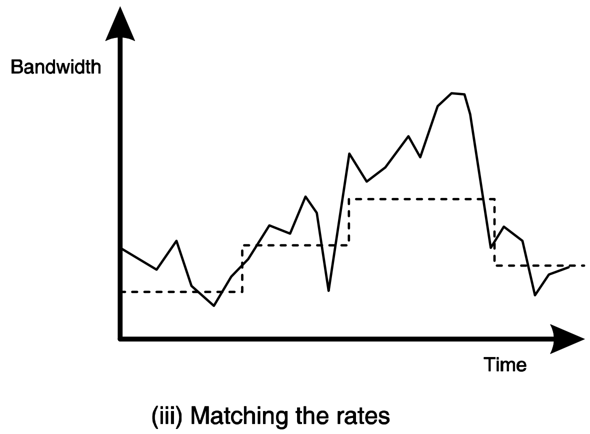


(iii) Matching the rates

- Ideal case
- Realistic?

Matching stream and network rates /3



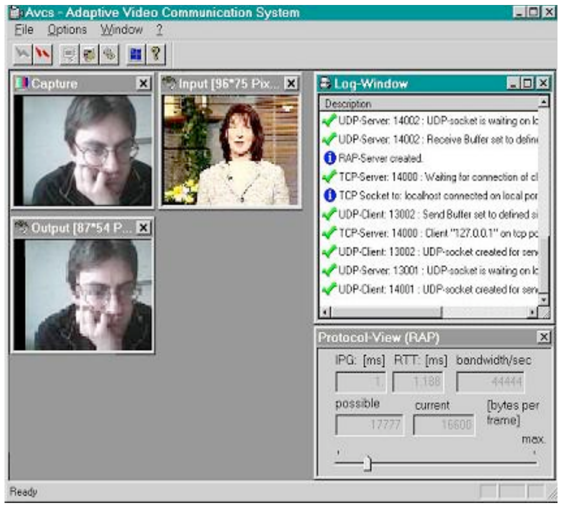


- "Adaptive Multimedia Application"
- Smoother network bandwidth would facilitate matching

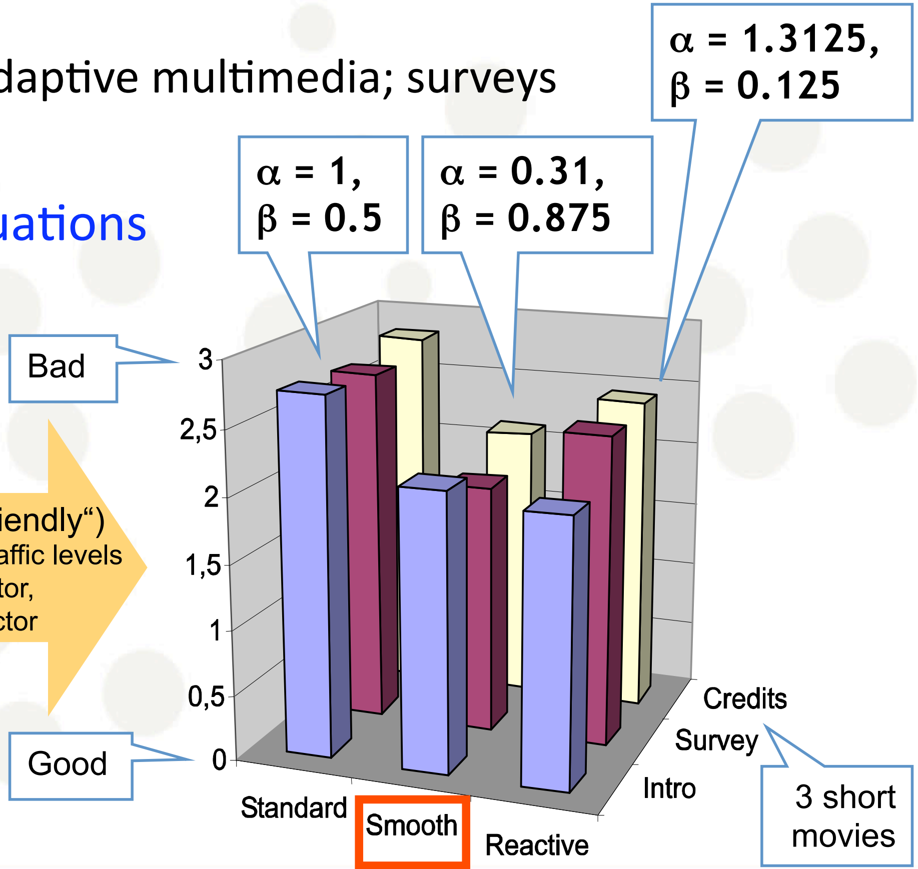
Adaptive multimedia: the user experience

- Studied by several research groups
  - Automatically evaluate "user experience" by judging received content based on knowledge about users
  - Study heartbeat etc. of users who test adaptive multimedia; surveys

- Consistent result: users do not like fluctuations



RAP (TCP-“friendly“)
5 different BG traffic levels
α = increase factor,
β = decrease factor



Resulting transport layer problem

- How to be fair towards TCP ("TCP-friendly") and have a relatively stable ("smooth") rate
  - Several ways to do this
  - Well known example: TCP-Friendly Rate Control (TFRC)
  - Determines sending rate by calculating how much TCP would send under similar conditions
  - Note: TFRC is not a protocol (only a congestion control mechanism)

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + t_{RTO}\left(3\sqrt{\frac{3p}{8}}\right)p(1+32p^2)}$$

s: packet size
R: rtt
t_{RTO}: TCP retransmit timeout
p: steady-state loss event rate

Datagram Congestion Control Protocol (DCCP)

- Motivation: provide unreliable, timely delivery
  - e.g. VoIP: significant delay = ☹, but some noise = ☺
  - UDP: no congestion control
    - unresponsive applications endanger others (congestion collapse) and may hinder themselves (queuing delay, loss, ..)

- DCCP realizes congestion control in the OS, where it belongs

DCCP /2

- Roughly:
  - DCCP = TCP – (bytestream semantics, reliability)
    = UDP + (congestion control w/ ECN, handshakes, ACKs)

- Main specification does not contain congestion control mechanisms
  - CCID definitions (e.g. TCP-like, TFRC, TFRC for VoIP)

- IETF standard – but not used much (up to now ?)

One-way streaming over TCP

- Assumption: buffering (delay) doesn‘t matter
  ⇒ no need for a smooth rate!
- Little loss case: TCP retransmissions won‘t hurt
- Heavy loss case:
- DCCP: 1, ~~2~~, ~~3~~, 4, ~~5~~, ~~6~~, 7, ~~8~~, ~~9~~, 10…
- TCP: (assume window = 3): 1, ~~2~~, ~~3~~, 2, ~~3~~, ~~4~~, 3, ~~4~~, ~~5~~, 4…
  - Application would detect: 4 out of 10 expected packets arrived
    ⇒ should reduce rate
  - Is receiving 1, 4, 7, 10 instead of 1, 2, 3, 4 really such a big benefit?
    Or is it just a matter of properly reacting?
    In RealPlayer and MediaPlayer, TCP can be used for streaming…
    seems to work well (also in YouTube!)

Real-time Transport Protocol (RTP)

- Designed for requirements of (soft!) real-time data transport
  - **NOT** a transport protocol
  - Two Components: RTP and RTP Control Protocol (RTCP)

- Provides several important functions
  - sequencing and loss detection (sequence numbers)
  - synchronization (timestamps)
  - payload identification (RTP profiles)
  - (via RTCP) QoS feedback and session information
  - scalable multicast support (...)
  - mixers and translators to adapt to bandwidth limitations
  - support for changing codecs on the fly, encryption

RTP Packet Format

- Relatively long header (>40 bytes)
  - overhead carrying possibly small payload
  - header compression
  - other means to reduce bandwidth (e.g. silence suppression)
- Header extensions for payload specific fields possible
  - Specific codecs
  - Error recovery mechanisms
- RTP can be used over any transport protocol – usually UDP

Profiles and Payload Types

- Profiles define codecs used to encode the payload data and their mapping to payload format codes ("Payload Type" header field)

- Each profile is accompanied by several payload format specifications
  - e.g. audio: G.711, G.723, G.726, G.729, GSM, QCELP, MP3, DTMF etc., and video: H.261, H.263, H.264, MPEG

- A complete specification of RTP for a particular application usage requires a profile and/or payload format specification(s)

Example profiles

- Profile for Audio and video conferences with minimal control defines
  - a set of static payload type assignments
  - mechanism for mapping between payload formats
  - and a payload type identifier (in header) using the Session Description Protocol (SDP)
    - mapping can be dynamic, i.e. per-session

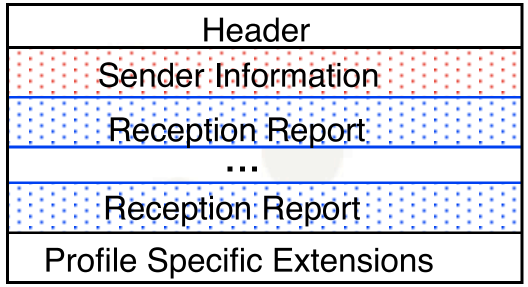
- Secure Real-time Transport Protocol (SRTP) = profile that provides cryptographic services for the transfer of payload

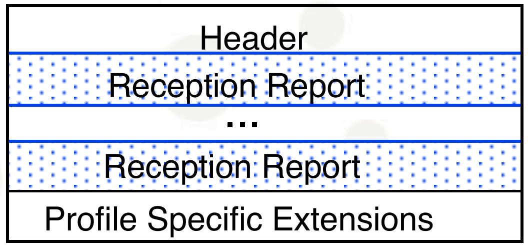
RTP Control Protocol (RTCP)

- Monitoring
  - of QoS / application performance
- Feedback to members of a group about delivery quality, loss, etc.
  - Sources may adjust data rate
  - Receivers can determine if QoS problems are local or network-wide
- Loose session control
  - Convey information about participants and session relationships
- Automatic adjustment to overhead
  - report frequency based on RTP sending rate and participant count

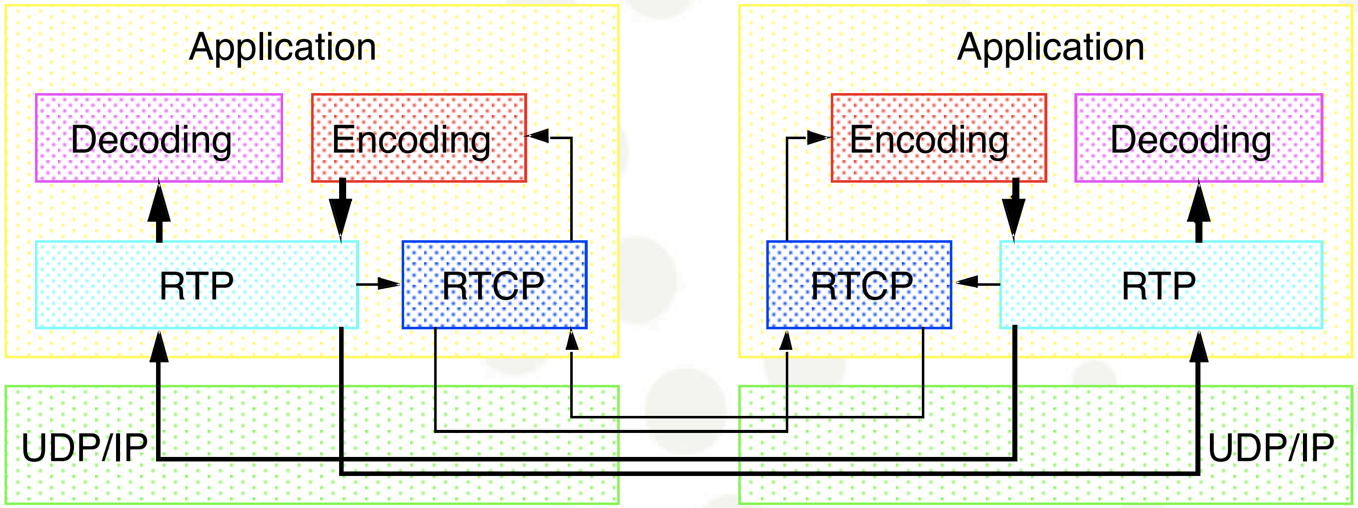
RTCP Sender / Receiver Reports

- Sender report
  - Sender Information
    - Timestamps, Packet Count, Byte Count
  - List of statistics per source
- Receiver report
  - For each source
    - Loss statistics
    - Inter-arrival jitter
    - Timestamp of last SR
    - Delay between reception of last SR and sending of RR
- Analysis of reports
  - Cumulative counts for short and long time measurements
  - NTP timestamp for encoding- and profile independent monitoring





RTP Quality Adaptation

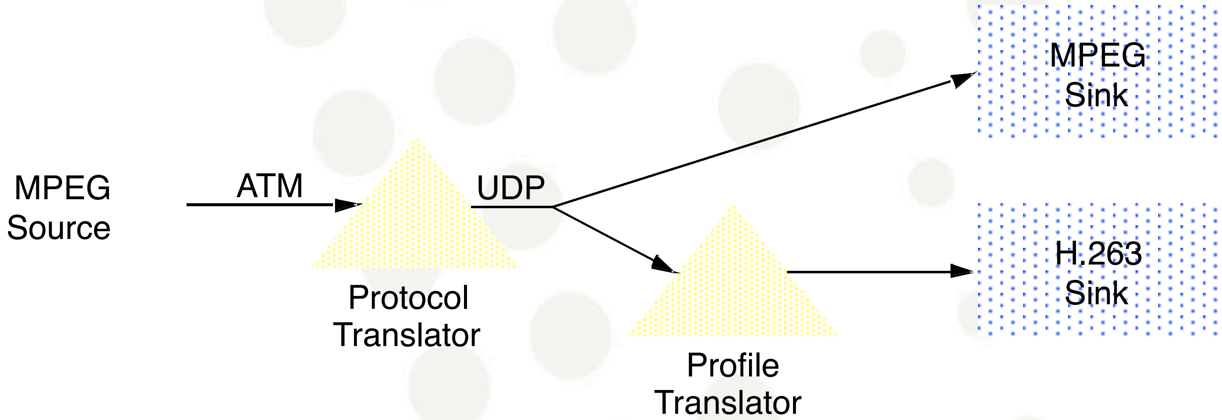


- Component interoperations for control of quality
- Evaluation of sender and receiver reports
- Modification of encoding schemes and parameters
- Adaptation of transmission rates
- Hook for possible retransmissions (outside RTP)

RTP Mixer

- Reconstructs constant spacing generated by sender
- Translates audio encoding to a lower-bandwidth
- Mixes reconstructed audio streams into a single stream
- Resynchronizes incoming audio packets
  - New synchronization source value (SSRC) stored in packet
  - Incoming SSRCs are copied into the contributing synchronization source list (CSRC)
- Forwards the mixed packet stream
- Useful in conference bridges

RTP Translator



- Translation between protocols
  - e.g., between IP and ST-2
  - Two types of translators are installed
- Translation between encoding of data
  - e.g. for reduction of bandwidth without adapting sources
- No resynchronization in translators
  - SSRC and CSRC remain unchanged

Signaling Protocols

- Control of media delivery by sender or receiver
  - Sender and receiver "meet" before media delivery

- Signaling should reflect different needs
  - Media-on-demand
    - Receiver controlled delivery of content; explicit session setup
  - Internet telephony and conferences:
    - Bi-directional data flow, live sources; (mostly) explicit session setup, mostly persons at both ends
  - Internet broadcast
    - Sender announces multicast stream; no explicit session setup

Real-Time Streaming Protocol (RTSP)

- Internet media-on-demand
  - Select and playback streaming media from server
  - Similar to VCR (start, stop, pause, ..), but
    - Potentially new functionality
    - Integration with Web
    - Security
    - Varying quality
- RTSP is also usable for
  - Near video-on-demand (multicast)
  - Live broadcasts (multicast, restricted control functionality)
  - ...

RTSP Approach

- In line with established Internet protocols
  - Similar to HTTP 1.1 in style
  - Uses URLs for addressing: rtsp://video.server.com:8765/videos/themovie.mpg
  - Range definitions
  - Proxy usage
  - Expiration dates for RTSP DESCRIBE responses
  - Other referenced protocols from Internet (RTP, SDP)

- Functional differences from HTTP
  - Data transfer is separate from RTSP connection; typically via RTP
  - Server maintains state – setup and teardown messages
  - Server as well as clients can send requests

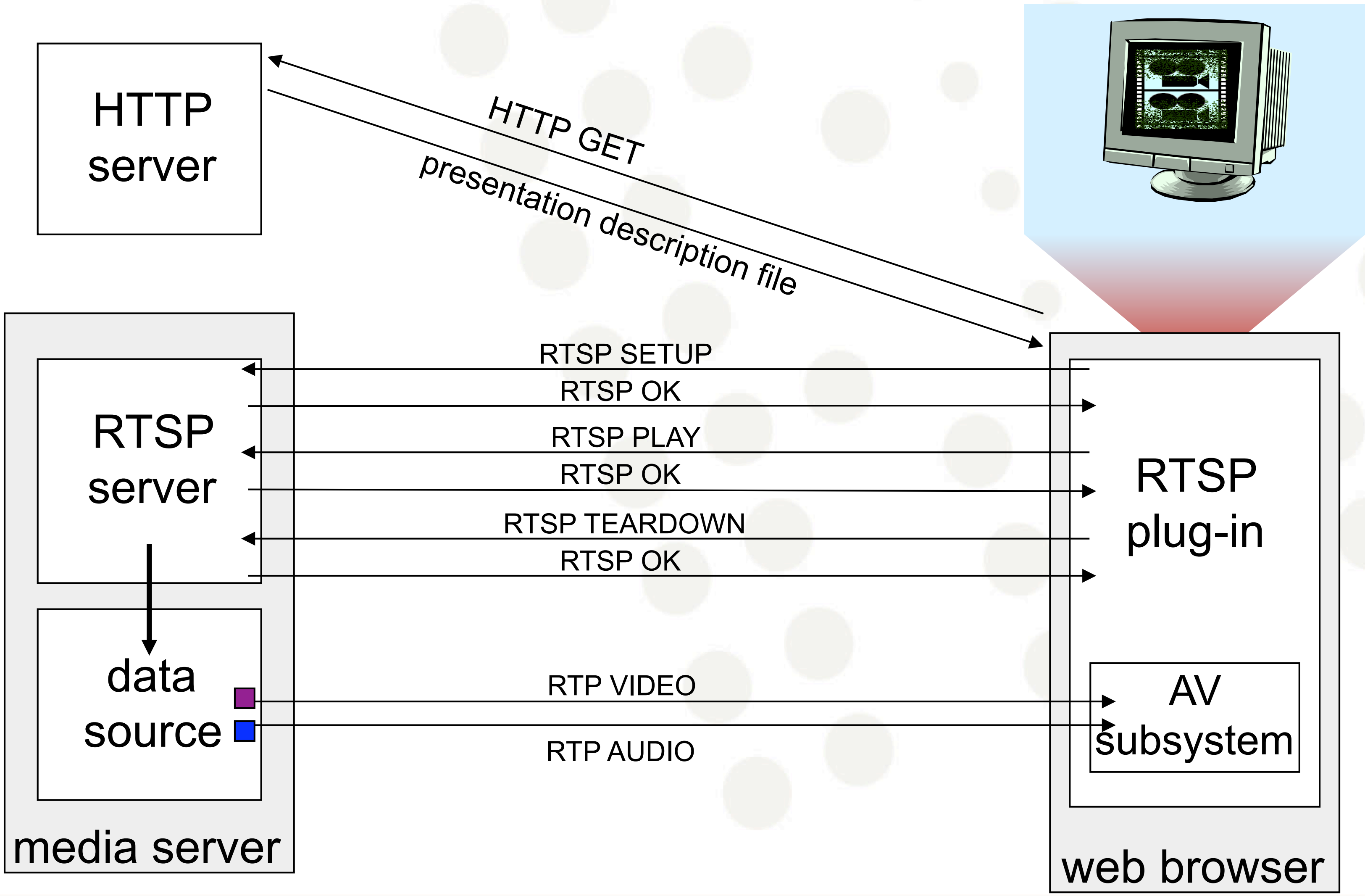
RTSP Features

- Rough synchronization
  - Media description in DESCRIBE response
  - Timing description in SETUP response
  - Fine-grained through RTP sender reports
- Aggregate and separate control of streams possible
- Virtual presentations: synchronized streams from multiple servers
  - Server controls timing for aggregate sessions
  - RTSP Server may control several data (RTP) servers
- Load balancing through redirect at connect time
  - Use REDIRECT at connect time
- Caching
  - Only RTSP caching so far

RTSP Methods

OPTIONS	C → S C ← S	determine capabilities of server/client
DESCRIBE	C → S	get description of media stream
ANNOUNCE	C ↔ S	announce new session description
SETUP	C → S	create media session
RECORD	C → S	start media recording
PLAY	C → S	start media delivery
PAUSE	C → S	pause media delivery
REDIRECT	C ← S	redirection to another server
TEARDOWN	C → S	immediate teardown
SET_PARAMETER	C ↔ S	change server/client parameter
GET_PARAMETER	C ↔ S	read server/client parameter

RTSP Integration



Session Initiation Protocol (SIP)

- Lightweight generic signaling protocol
- Internet telephony and conferencing
  - call: association between number of participants
  - signaling association as signaling state at endpoints (no network resources)
- Several "services" needed
  - Name translation, user location, feature negotiation, call control

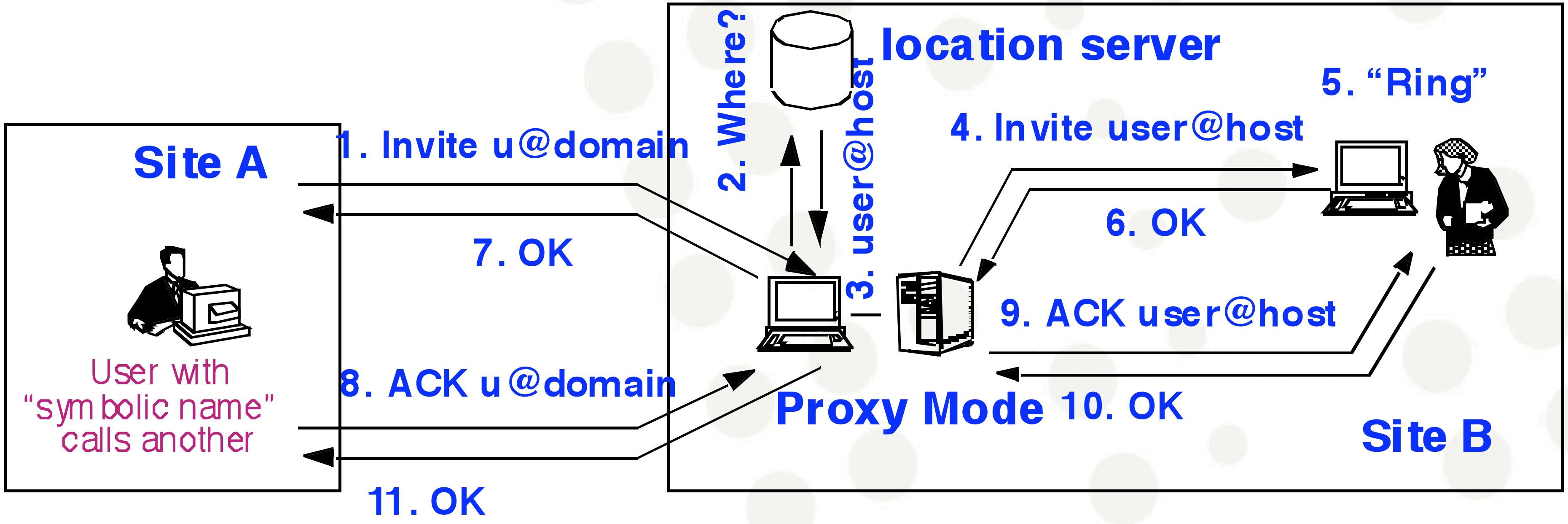
SIP Basics

- Establish calls between users
  - directly or forwarding (manual and automatic)
  - re-negotiate call parameters
  - terminate and transfer calls
- Supports personal mobility (change of terminal)
  - through proxies or redirection
- Control, location and media description (via SDP)
- Extensible
  - IMS – Internet Multimedia Subsystem – the next generation of telecoms' service gateways

SIP – Methods

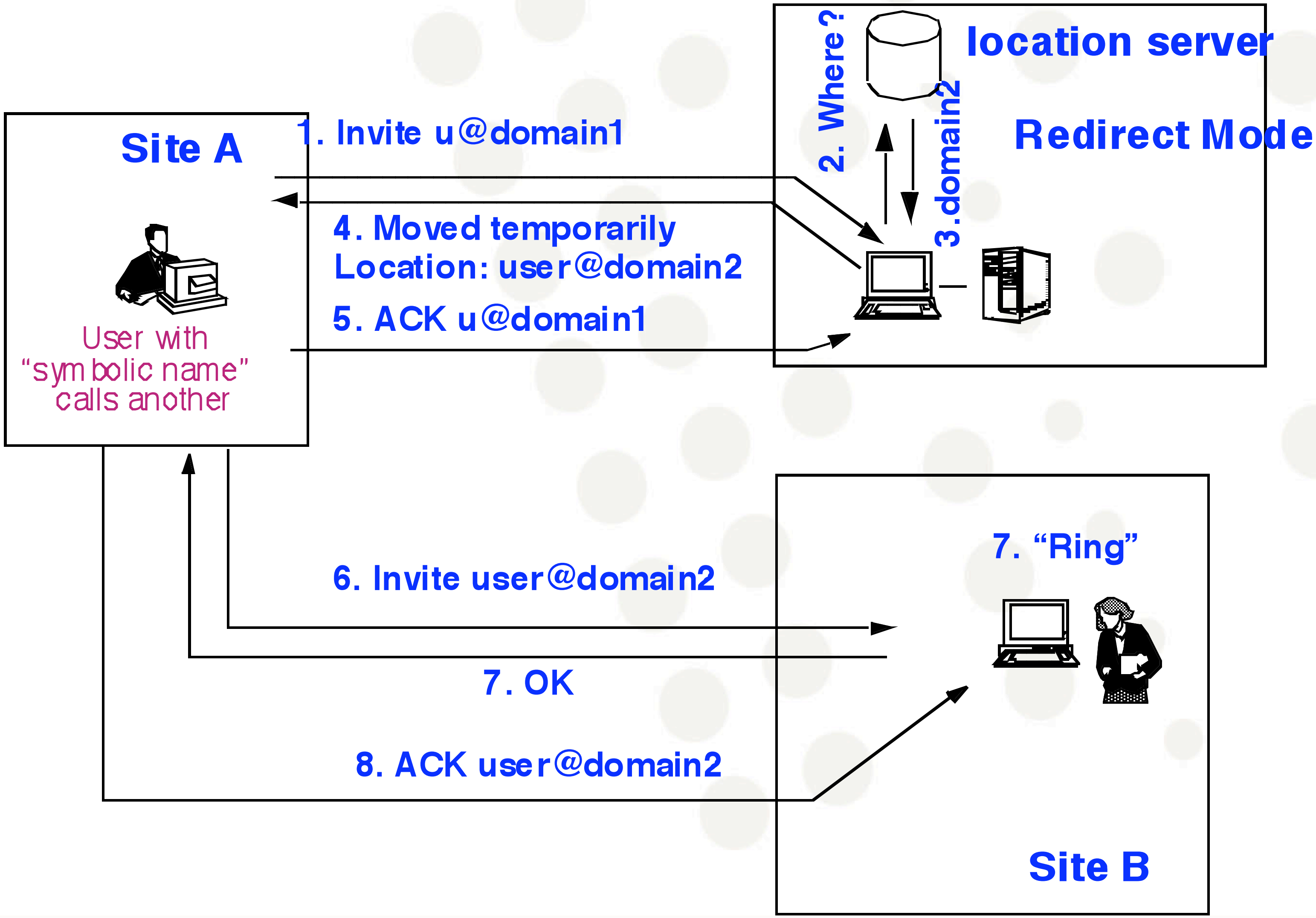
- Basic Methods:
  - INVITE: session setup – like RTSP SETUP and DESCRIBE in one
  - ACK: like RTSP ACK
  - OPTIONS: like RTSP OPTIONS
  - BYE: end a session
  - CANCEL: terminate an ongoing session setup operation
  - REGISTER: register a user in a location server, update location, …
- Additional Methods (partially standardized):
  - INFO: carry information between User Agents
  - REFER: ask someone to send an INVITE to another participant
  - SUBSCRIBE: request to be notified of specific event
  - NOTIFY: notification of specific event

SIP Operation – Proxy Mode



- Proxy forwards requests
  - possibly in parallel to several hosts
  - cannot accept or reject call
  - useful to hide location of callee

SIP Operation – Redirect Mode



PSTN: SS7 / SIGTRAN

(PSTN = Public Switched Telephone Network)

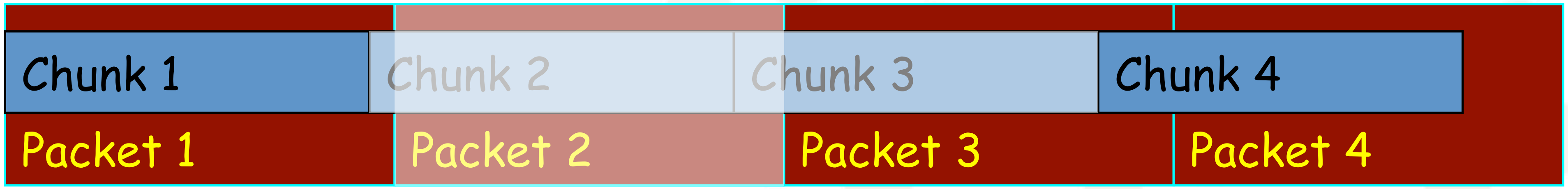
- SS7: telephony signaling protocols
  - mainly call setup and teardown
  - international standard + national variants
  - services such as call forwarding (busy and no answer), voice mail, call waiting, conference calling, calling name and number display, ...
- SIGTRAN: IETF standards, most importantly SCTP
  - *efficiently* transferring such data over the Internet

SCTP services: SoA TCP + extras

Services/Features	SCTP	TCP	UDP
Full-duplex data transmission	yes	yes	yes
Connection-oriented	yes	yes	no
Reliable data transfer	yes	yes	no
Unreliable data transfer	yes	no	yes
Partially reliable data transfer	yes	no	no
Ordered data delivery	yes	yes	no
Unordered data delivery	yes	no	yes
Flow and Congestion Control	yes	yes	no
ECN support	yes	yes	no
Selective acks	yes	yes	no
Preservation of message boundaries (ALF)	yes	no	yes
PMTUD	yes	yes	no
Application data fragmentation	yes	yes	no
Multistreaming	yes	no	no
Multihoming	yes	no	no
Protection agains SYN flooding attack	yes	no	n/a

Application Level Framing (ALF)

- Concept applied in RTP and SCTP
  - Byte stream (TCP) inefficient when packets are lost
  - Application may want logical data units (“chunks“)



- ALF: app chooses packet size = chunk size

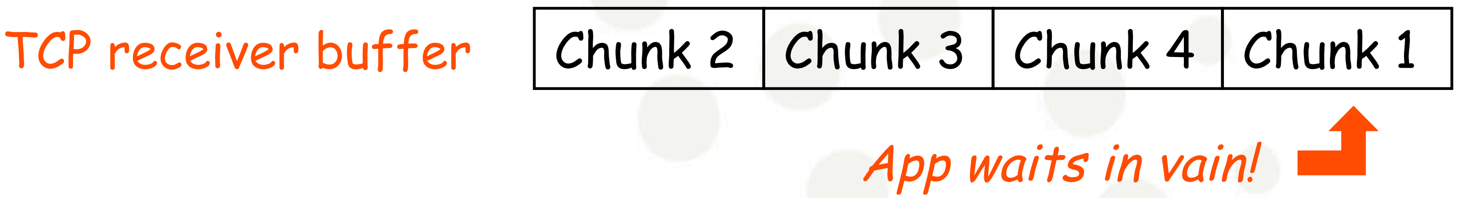
  packet 2 lost: no unnecessary data in packet 1,

  use chunks 3 and 4 before retrans. 2 arrives

- 1 ADU (Application Data Unit) = multiple chunks \ ALF still more efficient!

Unordered delivery & multistreaming

- Decoupling of reliable and ordered delivery
  - Unordered delivery: eliminate Head-Of-Line blocking delay



- Support for multiple data streams (per-stream ordered delivery)
  - Stream sequence number (SSN) preserves order *within* streams
  - no order preserved *between* streams

Multihoming

- ...at transport layer! (i.e. transparent for apps, such as FTP)
- TCP connection ⇔ SCTP association
  - 2 IP addresses, 2 port numbers ⇔ 2 sets of IP addresses, 2 port numbers
- Goal: robustness *(not load balancing – yet?)*
  - automatically switch hosts upon failure
  - eliminates effect of long routing reconvergence time
- TCP: no “keepalive“ messages when connection idle
- SCTP monitors reachability via ACKs of data chunks and heartbeat chunks

References

- Michael Welzl, "Network Congestion Control: Managing Internet Traffic", John Wiley & Sons, Ltd., August 2005, ISBN: 047002528X

- INF3190 2009 slides by Carsten Griwodz