Network Issues for Multimedia Presentations

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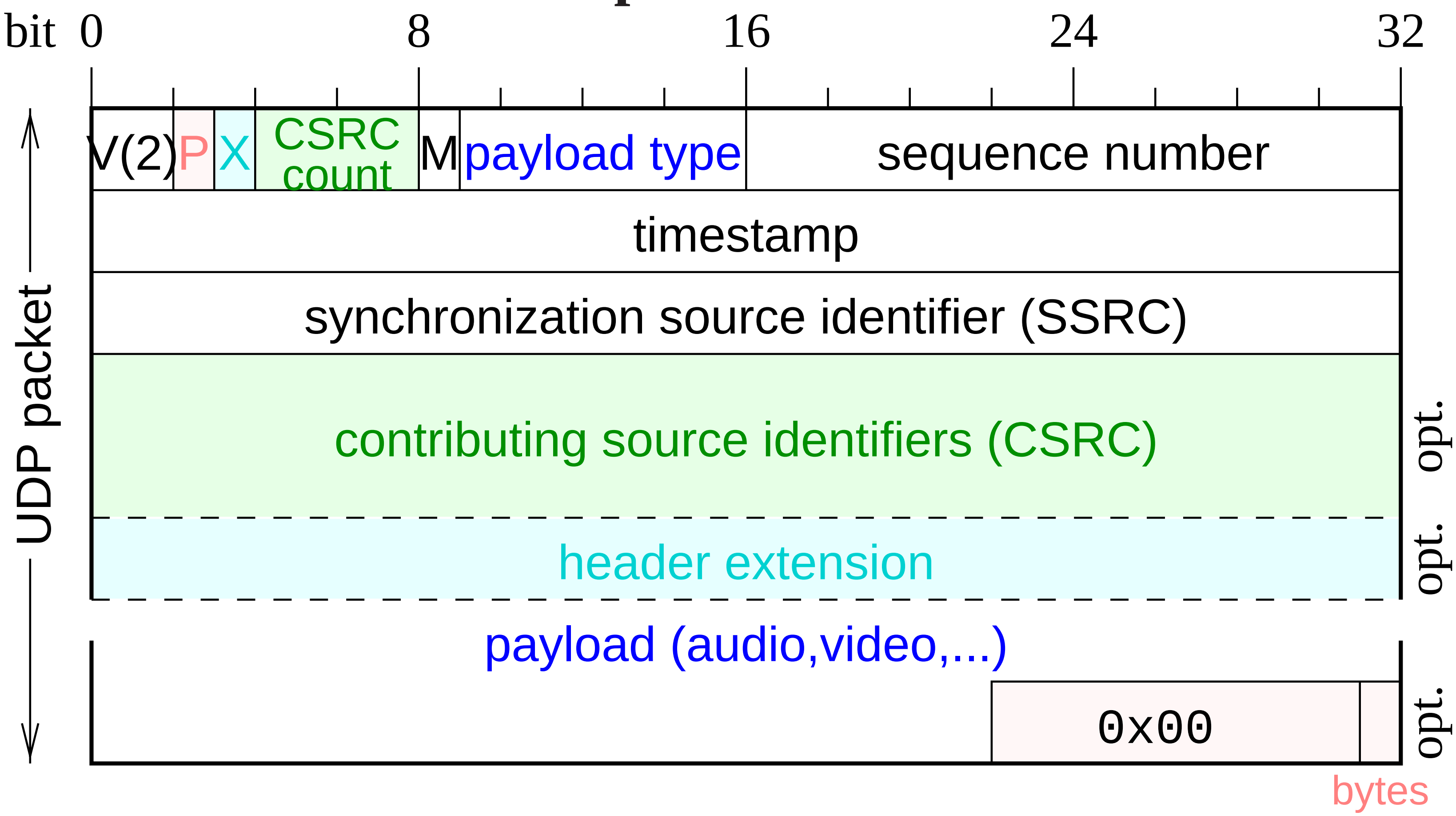
Overview

- synchronizing different sources using RTP
- Internet hardships: jitter, packet loss, reordering, ...
- smoothing
- bandwidth reservation
- variable bandwidth, adaptation

RTP packet header



RTP packet header

Payload type: audio, video encoding method; may change during session

SSRC: sychronization source ➠ each source picks at random ➠ may change after *collision*!

sequence number: incremented by 1 for each packet ➠ gaps ≡ loss

P: padding (for encryption) ➠ last byte contains padding count

M: marker bit; indicates frame, beginning of talkspurt ➠ allow delay adjustment

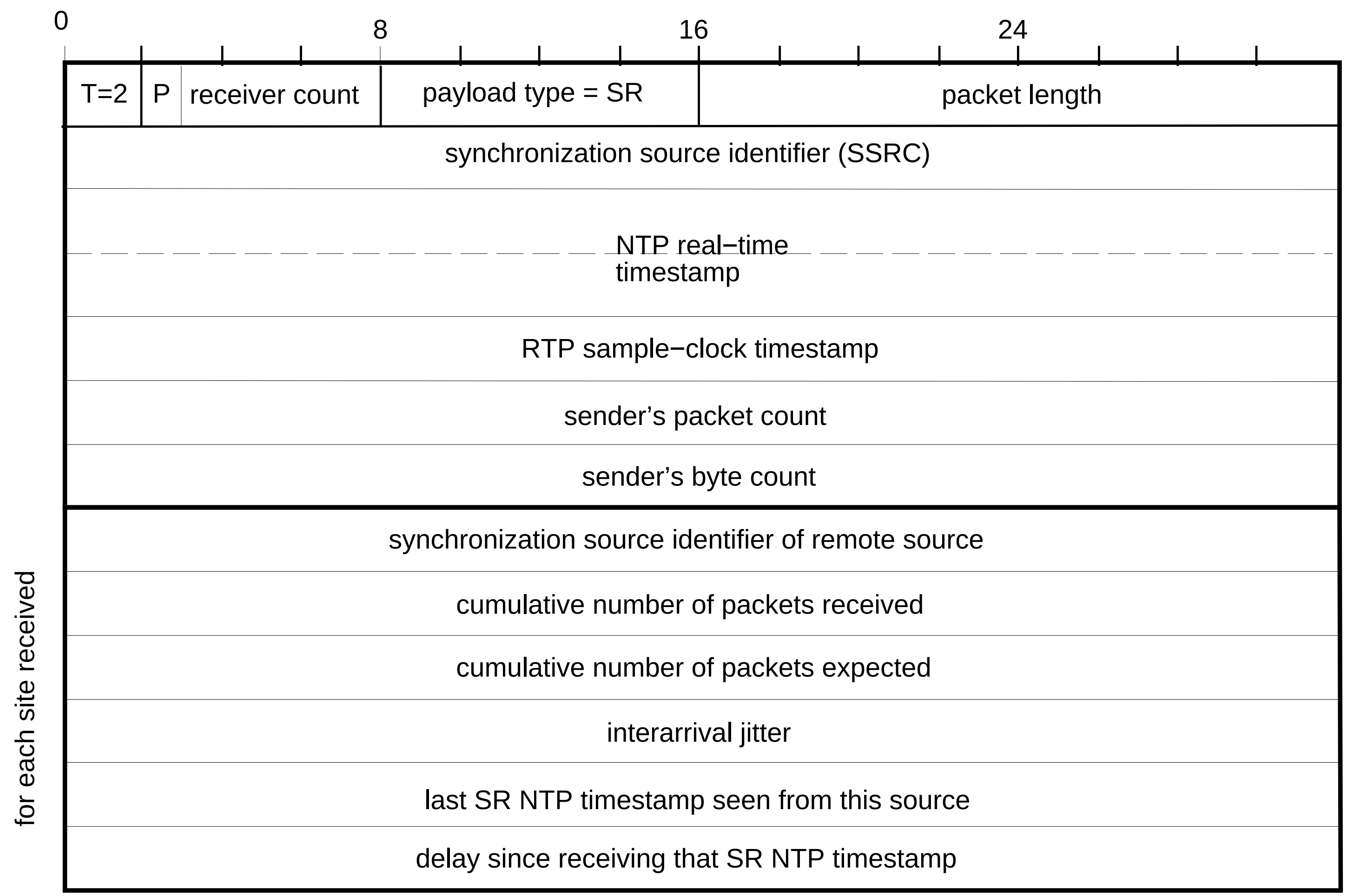
CC: content source count (for mixers)

CSRC: list of identifiers of those contributing to (mixed into) packet

RTP timestamp

- incremented by 1 for each sample (e.g., 160 for 20 ms packets @ 8000 Hz)
- random starting value
- constant rate for each audio payload type (e.g., 8000 Hz for PCM μ-law, 44100 Hz for linear, 16-bit)
- 90 kHz for video
- several video frames may have same timestamp
- ➠ gaps $\equiv$ silence
- time per packet may vary
- video frame maybe split (carefully…) over several packets
- typical: 20 to 100 ms of audio

RTCP sender reports (SR)



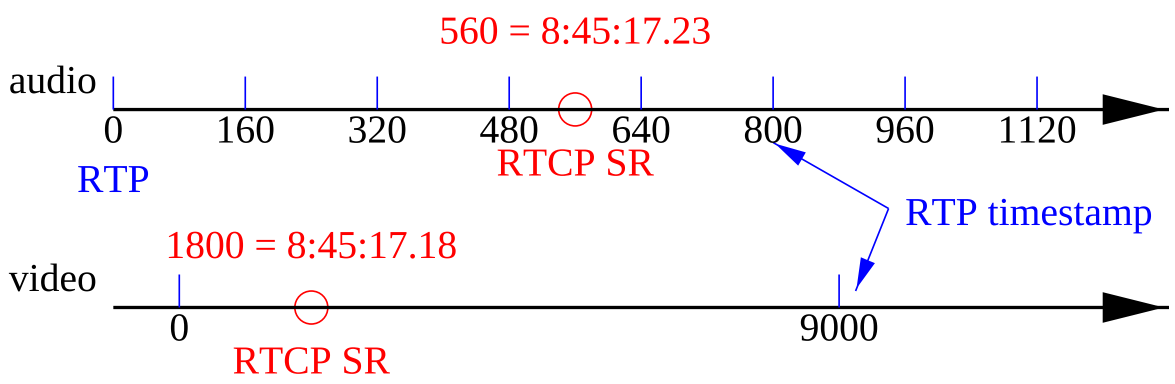
Stream-Level Synchronization

- no difference: single source, multiple sources
- assume: synchronized clocks

Intermedia synchronization

= synchronization between different streams (audio, video, slides, . . .)

- timestamps are offset with random intervals
- may not tick at nominal rate
- every RTCP sender report correlates "real" time (wallclock time) with RTP timestamp
- ➠ compute when sample was generated
- delay playout of all but one stream to achieve synchronization



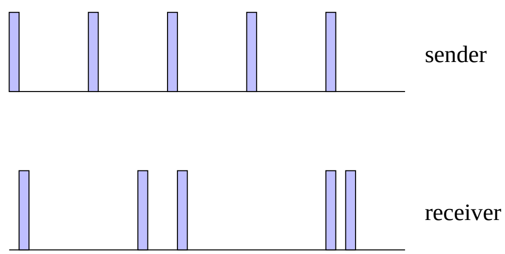
Network impairments

duplicates: not uncommon for multicast; generally harmless

packet loss: up to several percent ➠ audible clicks, loss of encoding state; not as bad for conferencing video

delay: due to transmission on slow links, propagation (5 μs/km), switching

delay jitter: arrival distortion:



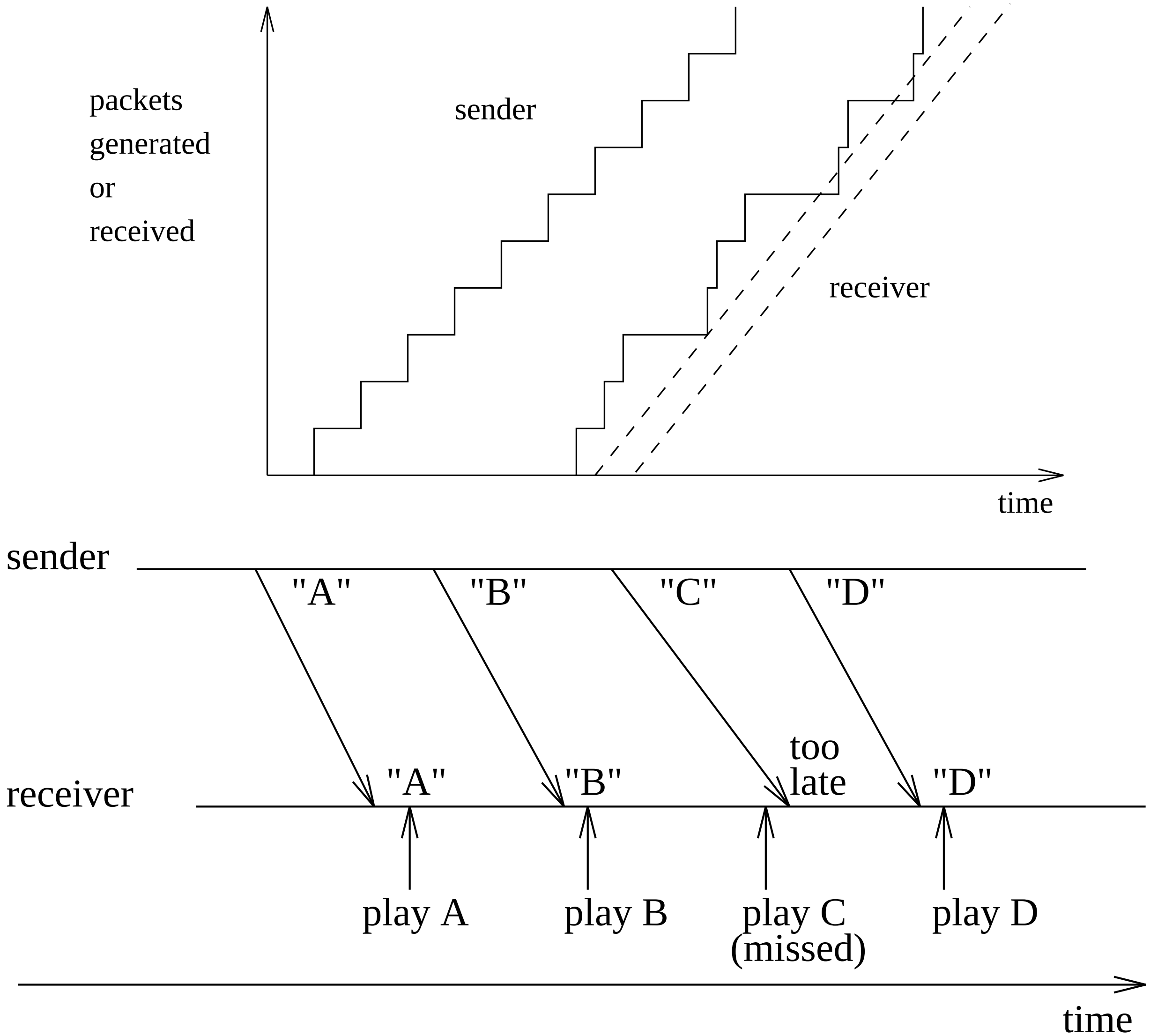
caused by queueing (resource contention) in nodes

Playout delay

interactive: adaptive ➠ trade late loss for delay

stored media: pre-buffer a few seconds

Playout delay



Smoothing

- video data: bursty (I, B, P; motion; scene changes)
- ➠ smoothing = delay or pre-play
- buffer at receiver

Bandwidth Reservation

- out-of-band ➠ RSVP
- media server sends PATH messages to all receivers with Tspec
- Tspec: r, b, p, packet size range
- receivers that need reservation return RESV with Rspec

Bandwidth Adaptation

Sender: RTCP feedback ➠ sender changes rate

Receiver: receivers subscribe to different layers or codecs