Real Time Protocol (RTP)

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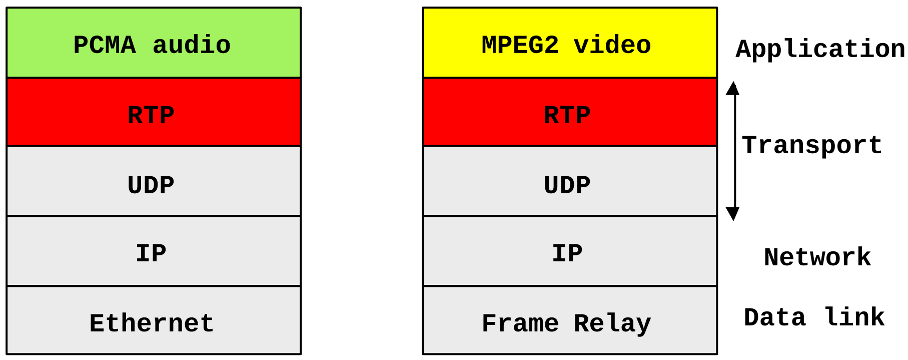
Multimedia applications

- Streaming multimedia applications need
  - hard real-time guarantees (do not tolerate losses or (excessive) delay jitter: need intserv, diffserv – next chapter)
  - soft real-time guarantees (do tolerate small losses and delay jitter: need RTP)
- Soft real time applications
  - should support mulicast
  - cannot wait for lost packets/segments/datagrams to be retransmitted
  - need to associate some timing information (timestamps) with packets/segments/datagrams
  - What about TCP ?
  - What about UDP ?

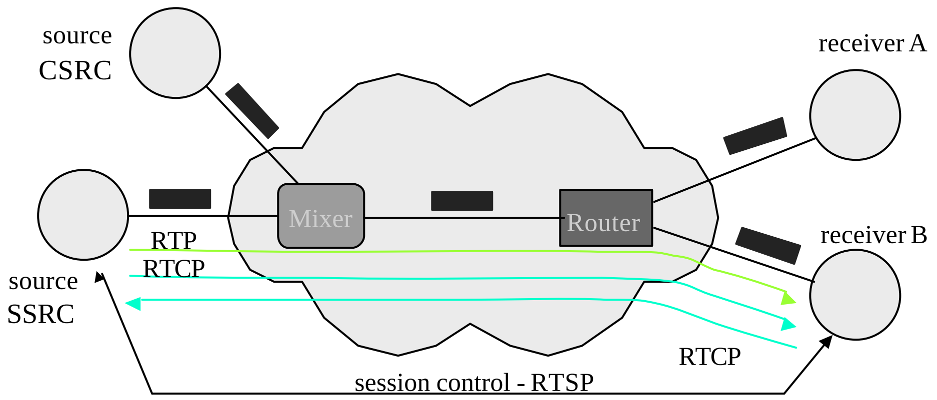
Real Time Transport Protocol (RTP)

- RTP
  - uses UDP
  - defines format of additional information required by the application (sequence number, time stamps)
  - uses a special set of messages (RTCP) to exchange periodic reports
  - one RTP session, one media flow



From a developer's perspective, RTP belongs to the application layer rather than the transport layer.

RTP



❑RTP session

- RTP port , RTCP port
- unicast or multicast IP adresses

Mixer is an intermediate system that combines RTP streams from different sources into a single stream. It can change the data format of the RTP packets.

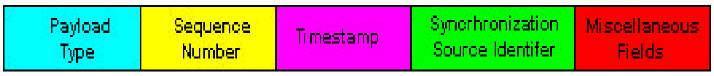
RTP

- Provides standard packet format for real-time application
- Specifies header fields below
- **Payload Type**: 7 bits, providing 128 possible different types of encoding; eg PCM, MPEG2 video, etc.
  - different media are not multiplexed
- **Sequence Number**: 16 bits; random number incremented by one for each RTP data packet sent; used to detect packet loss

RTP Header

RTP

- ❑ **Timestamp**: 32 bytes; gives the sampling instant of the first audio/video byte in the packet; used to remove jitter introduced by the network
  - clock frequency depends on applications
  - random initial value
  - several packets may have equal timestamps (eg. same video frame), or even in disorder (eg. interpolated frames in MPEG)
- ❑ **Synchronization Source identifier (SSRC)**: 32 bits; an id for the source of a stream; assigned randomly by the source
- ❑ Miscellaneous fields: Contributing Source identifier (CSRC)



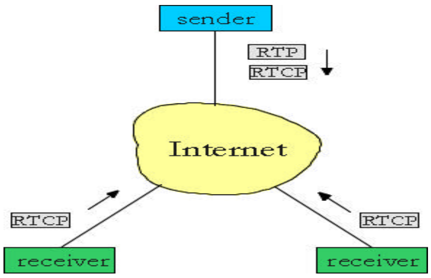
RTP Header

Type of the payload

- Audio
  - PCM A-law
  - PCM *m*-law
  - GSM
- Video
  - CelB
  - JPEG
  - H.261
  - MPEG

RTP Control Protocol (RTCP)

- Protocol specifies report packets exchanged between sources and destinations of multimedia information
- Three reports are defined: Receiver report (RR), Sender report (SR), and Source description (SDES)
- Reports contain statistics such as the number of packets sent, number of packets lost, inter-arrival jitter
- Used to modify sender transmission rates and for diagnostics purposes



RTCP Bandwidth Scaling

- If each receiver sends RTCP packets to all other receivers, the traffic load resulting can be large
- RTCP adjusts the interval between reports based on the number of participating receivers
- Typically, limit the RTCP bandwidth to 5% of the session bandwidth, divided between the sender reports (25%) and the receivers reports (75%)

RTCP

❑ Functions

- supervise the network QoS
  - flow control and congestion control
- identification of participants
  - persistent id (CNAME = Canonical Name)
- determine the number of participants
- session information
- traffic of RTCP < 5%

❑ Format of RTCP packets

- SR : *sender reports*
  - information on the source
  - source statistics
- RR : *reception reports*
  - receiver statistics
- SDES : *source description*
  - CNAME
- BYE : end of the participation
- APP : application specific functions

SR and RR : *sender and receiver reports*

- Information on the source (only in SR)
  - absolute timestamp (NTP)
  - timestamp (RTP)
  - number of packets sent RTP
  - number of bytes sent RTP
- Statistics report for source SSRC-1
- Statistics report for source SSRC-2
- ...
- Statistics report for source SSRC-n

Statistics report

- SSRC-n
- Fraction of lost packets
- Number of lost packets
- Last sequence number received
- Estimation of the jitter
- Timestamp of the last SR received
- Delay since the last SR received

Jitter estimation

- S_i - RTP timestamp RTP of packet *i*
- R_i - reception instant of packet *i*
- D_i - jitter estimation for packet *i*
  - $D_i = (R_i - R_{i-1}) - (S_i - S_{i-1})$
- J_i - temporal average of the jitter for packet *i*
  - $J_i = 15/16\, J_{i-1} + 1/16\, | D_i |$
- Used for adaptive playout

RTSP *(Real-Time Streaming Protocol)*

- Similar to HTTP
  - `rtsp://france-info.fr/actualites`
- Description of available media
  - SDP *(Session Description Protocol)*
- Allows to establish RTP sessions
- Session control
  - start, pause, resume, end