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 multicast
  - 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

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From a developer’s perspective, RTP belongs to the application layer rather than the transport layer.
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; e.g., 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

- **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 (e.g. same video frame), or even in disorder (e.g. 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)
Type of the payload

- Audio
  - PCM A-law
  - PCM μ-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