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Real Time Transport Protocol (RTP)

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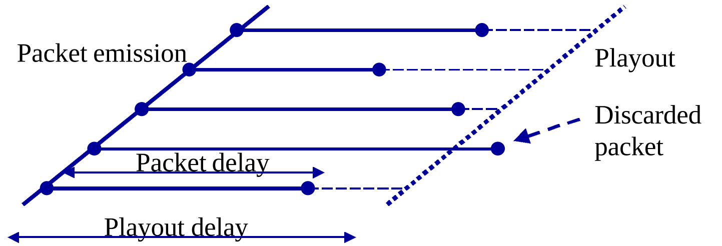
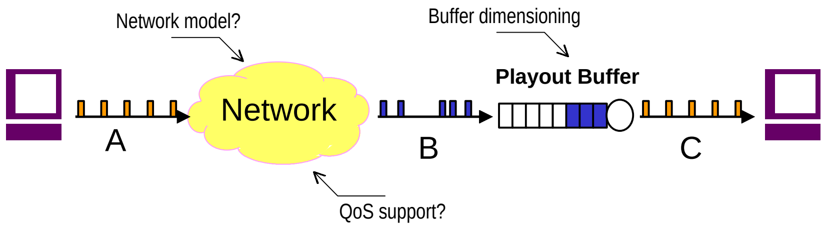
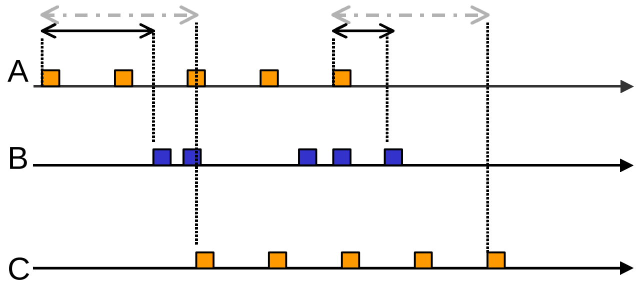
Requirements for a Real-time Protocol

- Packets generated at a steady rate (e.g. audio/video streams) must be played back at the same constant rate at the destination
  - **sufficient playout buffers at destination to account and equalize network delay**
  - **packets must contain sequence number and timing information to play back in right order and at same steady rate**
- Playout buffer dimensioning may require assumptions on
  - **source characterization**
  - **network model (that in turn requires background traffic characterization)**



Jitter - Delay Variation





Requirements for a Real-time Protocol

- TCP is not well-suited for real time applications, because:
  - **TCP congestion control does not apply to non adaptive real time applications**
    - it implies that applications are adaptive - some are not!
  - **TCP retransmission mechanism leads to intolerable delays**
    - real-time data cannot be re-transmitted
  - **does not preserve time relationship between source and destination**
  - **TCP does not support multicast**
- UDP is better adapted to real time applications
  - **UDP supports multicast**
  - **it has no congestion control and no retransmission mechanisms**
- but it lacks some functionality:
  - **it does not convey timing or sequencing information from source to destination**

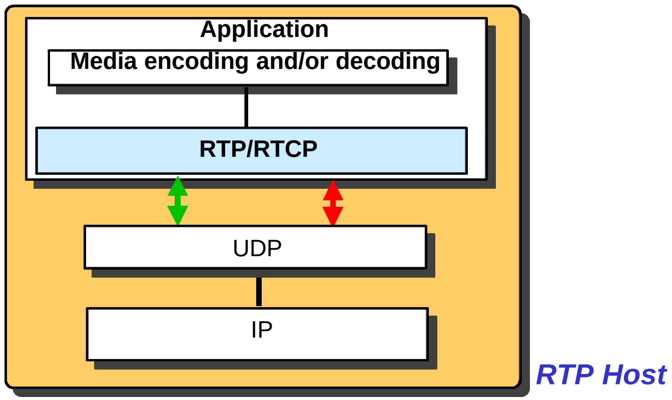
Real-time Transport Protocol (RTP)

- RFC 3550 and 3551 (2003)
  - **previously defined by RFC 1889 and 1890**
  - **normally on top of UDP**
- End-to-End transport for real-time applications:
  - **sequence numbering**
  - **payload type identification**
  - **timestamping**
  - **delivery monitoring**
- RTP characteristics:
  - **may make use of other transport protocols such as UDP**
  - **only end-to-end: does not require resources inside the network**
  - **any "media" (voice/video/...) any codec**
  - **both point to point and multicast communications**
- RTP does not:
  - **establish connections**
  - **guarantee delivery (no error control)**
  - **perform resource reservation**

Real-time Transport Protocol (RTP)

- RTP is typically implemented as a part of the application and not of the operating system
- RTP does not depend on any particular address format
- RTP requires that fragmentation and reassembly are taken care of by lower layers
- RTP is NOT a complete transport layer protocol; it integrates real time applications by providing them specific functionality
  - **RTP can rely on various lower layer technologies**

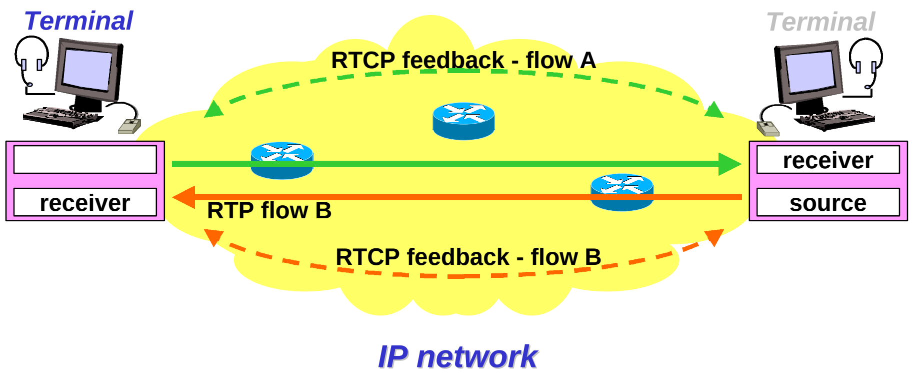
Protocol Architecture



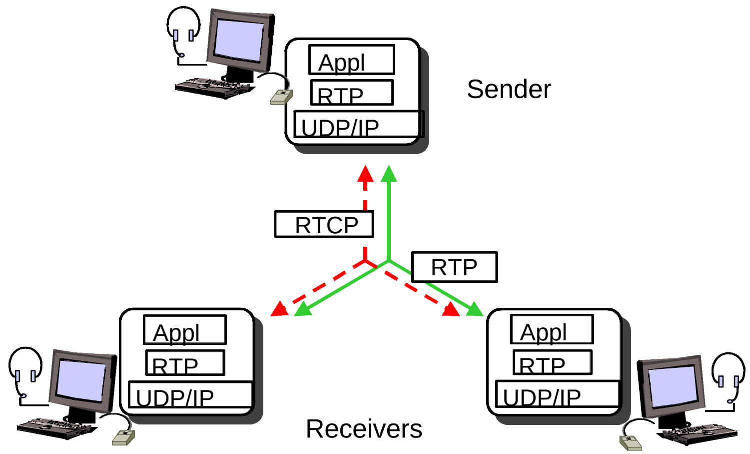
RTP Session

- Is the association among a set of participants communicating through RTP
- For each of them the session is defined by:
  - **destination IP address (unicast or multicast)**
  - **a pair of consecutive destination UDP ports**
    - the lowest for carrying data (RTP), and
    - the highest for RTCP control
- In a multimedia session each medium is carried by a separate RTP session

Point-to-point RTP



Point-multipoint (multicast) RTP



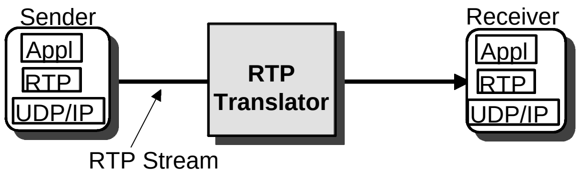
RTP Relay

- It is an intermediate system (IS) that acts as an RTP-level relay agent within the network
- It accepts packets coming from an input source, performs necessary transformations on them and then relays them towards destination(s)
- It is needed when source and destination cannot exchange packets directly (e.g. different formats)
- Two types of RTP Relays are defined:
  - **RTP Translator**
  - **RTP Mixer**
- Why
  - **to save bandwidth (mixer and translator)**
  - **to traverse firewalls**
  - **transcoding**

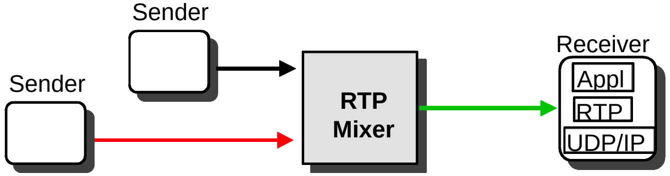
Mixers and Translators

- Mixer
  - **An RTP intermediate Relay system that mixes several RTP streams into a single one**
  - **it receives one or more RTP streams**
  - **possibly changes media encoding**
  - **mixes into a RTP single output stream**
  - **makes timing adjustments among the contributing streams (synchronization)**
  - **maintains contributing source IDs (CSRCs), and generates a new source ID (SSCR)**
- Translator
  - **An RTP intermediate relay system that operates on individual RTP streams**
  - **It forwards RTP packets maintaining the same Synchronization SouRCe identifier (SSRC)**
  - **It is often used for media encoding conversion (without mixing), replication from multicast to unicast and application-level filters in firewalls**

Mixers and Translators

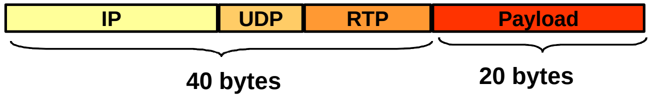


✓ Intermediate RTP system that operates on individual RTP streams
✓ May translate one media encoding to another but will maintain same SSRC



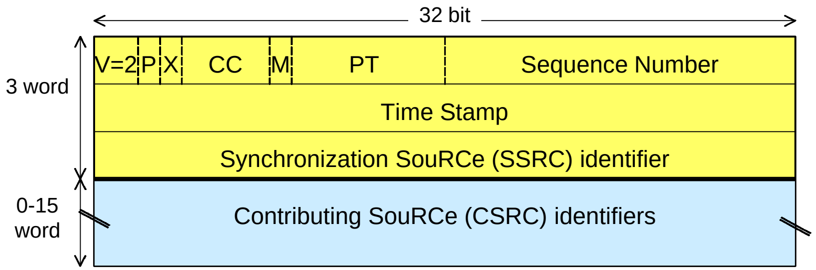
✓ Mixes several RTP streams into one new stream (may change the encoding)
✓ Generates new SSRC but originating sources are carried forward to receiver (CSRC)

RTP packets



- IP header : 20 octets **(IP addresses, ...)**
- UDP header : 8 octets **(port identifier, ...)**
- RTP header : 12 octets
  - **source id**
  - **payload type**
  - **sequence number**
  - **time stamp**
- Payload: encoded voice or multi-media stream (i.e. 20 bytes each 20 ms)

RTP Header Format



V = Version [2bit]
P = Padding [1bit]
X = Extension [1bit]
CC = CSRC Count [4bit]
M = Marker [1bit]

RTP Header Format (1)

- Version (2bit)
  - **current version = 2**
- Padding (1 bit)
  - **If set, the packet contains one or more additional padding octets at the end; the last octet contains the number of padding octets**
- Extension (1bit)
  - **If set, the fixed header is followed by exactly one header extension**
- Marker Bit (1 bit)
  - **marks significant events (depends on the payload type), e.g. frame boundaries ("in-band signaling")**
- Payload Type (7 bit)
  - **identifies the format of the RTP payload and determines its interpretation by the application (e.g. PCM A-law, G.729 codecs...)**
  - **additional payload type codes may be defined dynamically through non-RTP means**

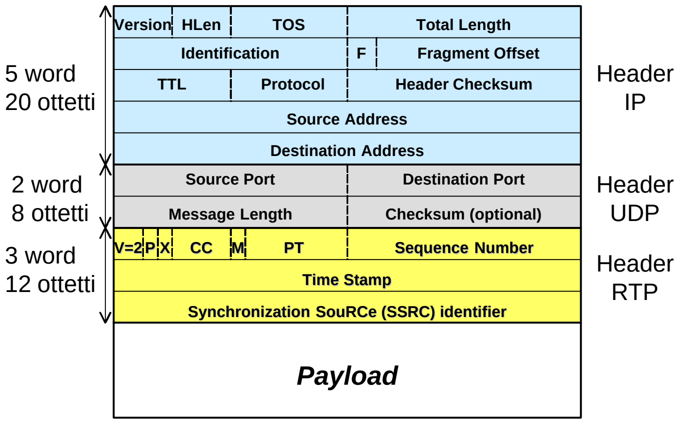
RTP Header Format (2)

- Sequence Number (16 bit)
  - increments by one for each RTP data packet sent,
  - may be used by the receiver to detect packet loss and to restore packet sequence
  - the initial value of the sequence number is random (unpredictable)
- Time Stamp (32 bit)
  - the timestamp reflects the sampling instant of the first octet in the RTP data packet
  - the clock frequency is dependent on the format of the data (for PCM audio codec f= 8kHz, T= 125μs)
  - if RTP packets are generated periodically, the nominal sampling instant as determined from the sampling clock is to be used
  - the initial value of the timestamp is random

RTP Header Format (3)

- Sinchronization SouRCe - SSRC (32 bit)
  - identifies the synchronization source
  - this identifier is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier
- Contributing SouRCe - CSRC (32 bit)
  - The CSRC list identifies the contributing sources for the payload contained in this packet
  - CSRC identifiers are inserted by mixers
  - the CSRC can be both a terminal identifier or a mixer identifier
- CSRC Count - CC (4 bit)
  - counts the number of CSRC (0-15)

RTP over IP: RTP/UDP/IP



RTP Payload Types

Payload Type	Encoding Name
0	PCMU
1	1016
2	G721
3	GSM
4	unassigned audio
5	DV14 (8KHz)
6	DV14 (16KHz)
7	LPC
8	PCMA
9	G722
10	L16 Stereo
11	L16 Mono
12	TPSO
13	VSC
14	MPA
15	G728
16-23	unassigned

Payload Type	Encoding Name
23	RGB8
24	HDCC
25	CelB
26	JPEG
27	CUSM
28	NV
29	PicW
30	CPV
31	H261
32	MPV
33	MP2T
34-71	unassigned
72-76	reserved
77-95	unassigned
96-127	dynamic

RTP Control Protocol (RTCP)

- RTCP (RTP Control Protocol) is based on the periodic transmission of control packets to all participants in the session (multicast)
  - **The transmission rate is determined independently by everyone by observing the number of participants**
- RTCP Primary Functions:
  - **QoS monitoring and feedback**
    - Riporta verso tutti i partecipanti della sessione (nel caso pt-pt solo alla sorgente) stime della QoS percepita: perdite e jitter
    - In caso di sorgenti adattative (ad es. coder voce) RTCP può fornire il necessario “feedback”
  - **Inter-media synchronization (e.g. lip synching audio and video from single RTP sender)**
  - **identification of population of multicast group**
  - **Minimum session control e.g. (BYE message)**
- RTCP does NOT do retransmission

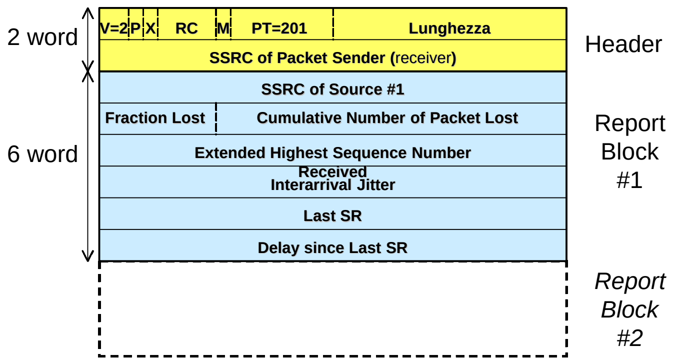
RTP Control Protocol (RTCP)

- RTCP packets are multicast to all participants
  - **rate of RTCP sent packets is dependent on network size (5% of RTP traffic)**
- RTP uses EVEN port numbers, RTCP uses next ODD port numbers
- Note: RTCP is defined to work together with RTP, however not all RTP applications do use it
  - **example, when another signaling protocol is present**

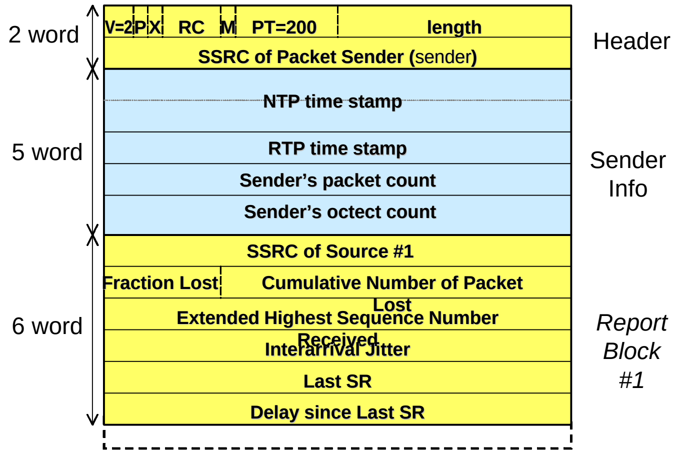
RTCP Packet Types

- Sender Report (SR)
  - **used by participants that are active senders to send data statistics/information (bytes sent, timestamps, etc.)**
- Receiver Report (RR)
  - **used by receivers to send back some statistics (estimated packet loss, inter-arrival jitter, round trip delay, etc.)**
- Source Description (SD)
  - **carries additional information on a source (e.g. CNAME, e-mail, phone no., etc.) which can be used to associate multiple RTP streams from a single sender**
- Bye
  - **used by a source to leave the session**
- Application Specific
  - **application specific functions**

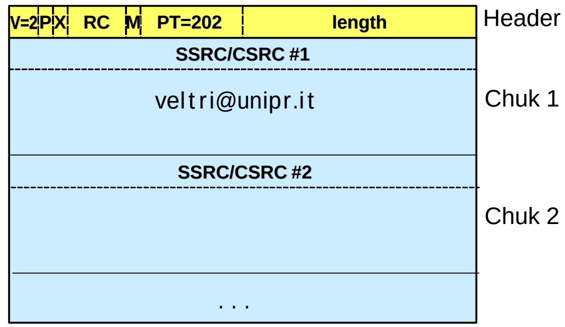
RTCP Packet Receiver Report (RR)



RTCP Packet
Sender Report (SR)



Pacchetto RTCP
Source Description (SDES)



RTCP Transmission Rate

- The control traffic should be limited to a small and known fraction of the session bandwidth
- It is suggested that the fraction of the session bandwidth allocated to RTCP be fixed at 5%

TX/RX