



UNIVERSITA' DEGLI STUDI DI PARMA
Dipartimento di Ingegneria dell'Informazione

Real Time Transport Protocol (RTP)

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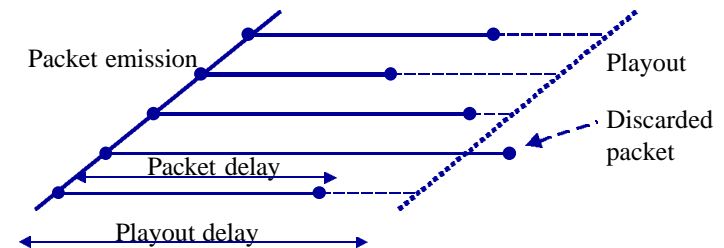


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RTP

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)



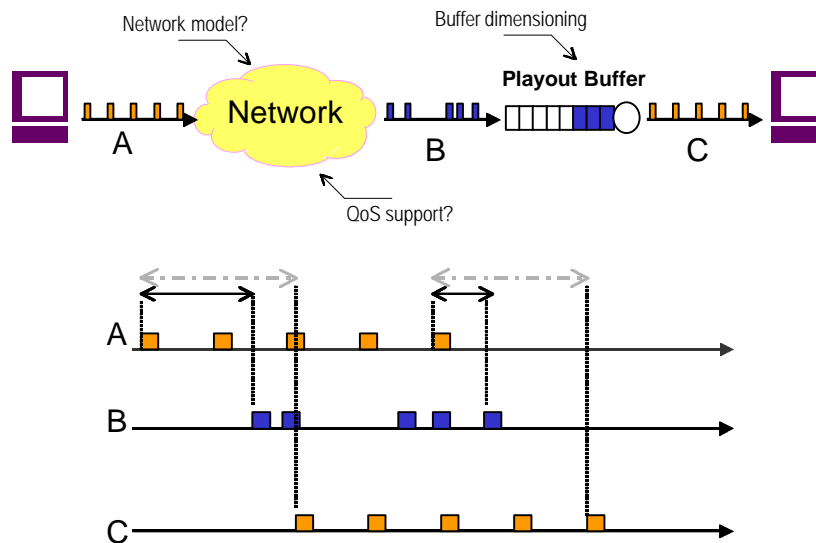
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RTP

Jitter - Delay Variation



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RTP

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

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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

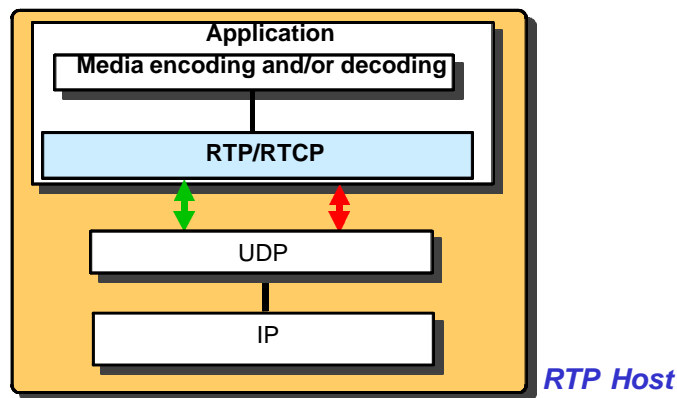
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Real-time Transport Protocol (RTP)

- 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

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Protocol Architecture



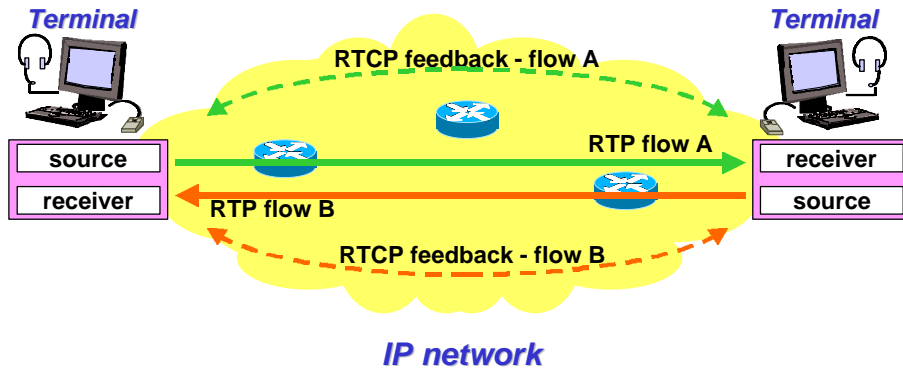
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RTP Session

- 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

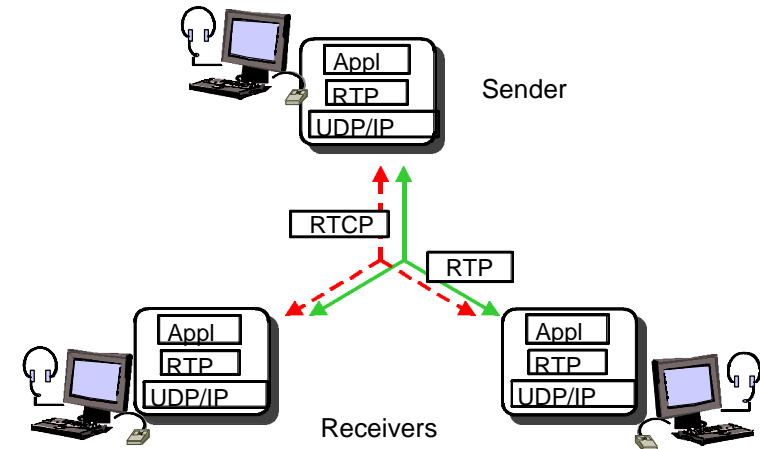
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Point-to-point RTP



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Point-multipoint (multicast) RTP



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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

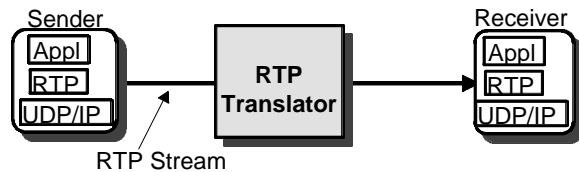
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Mixers and Translators

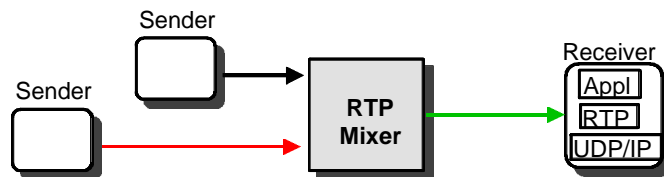
- Mixer
 - An RTP intermediate Relay system that mixes several RTP streams into a single one
 - it receives one or more RTP streams
 - possibly changes media encoding
 - mixes into a RTP single output stream
 - makes timing adjustments among the contributing streams (synchronization)
 - maintains contributing source IDs (CSRCs), and generates a new source ID (SSRC)
- 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

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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)

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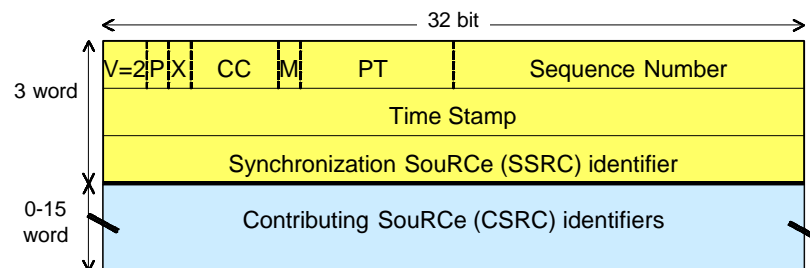
RTP packets



- 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)

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RTP Header Format



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

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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

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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 $f = 8\text{kHz}$, $T = 125\mu\text{s}$)
 - 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

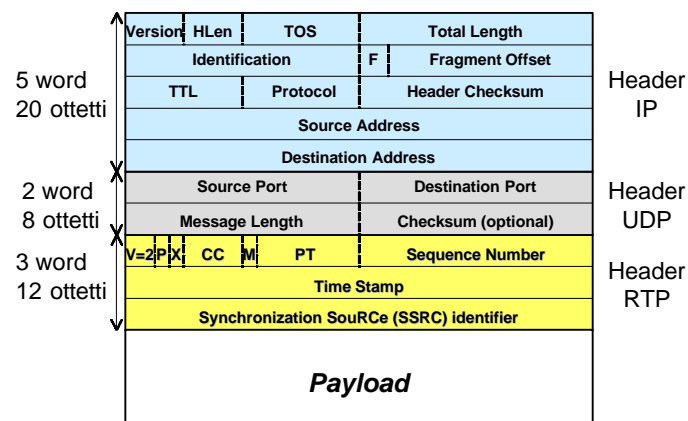
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RTP Header Format (3)

- Synchronization 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)

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RTP over IP: RTP/UDP/IP



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RTP Payload Types

Payload Type	Encoding Name	Payload Type	Encoding Name
0	PCMU	23	RGB8
1	1016	24	HDCC
2	G721	25	CeIB
3	GSM	26	JPEG
4	unassigned audio	27	CUSM
5	DV14 (8KHz)	28	NV
6	DV14 (16KHz)	29	PicW
7	LPC	30	CPV
8	PCMA	31	H261
9	G722	32	MPV
10	L16 Stereo	33	MP2T
11	L16 Mono	34-71	unassigned
12	TPSO	72-76	reserved
13	VSC	77-95	unassigned
14	MPA	96-127	dynamic
15	G728		
16-23	unassigned		

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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

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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

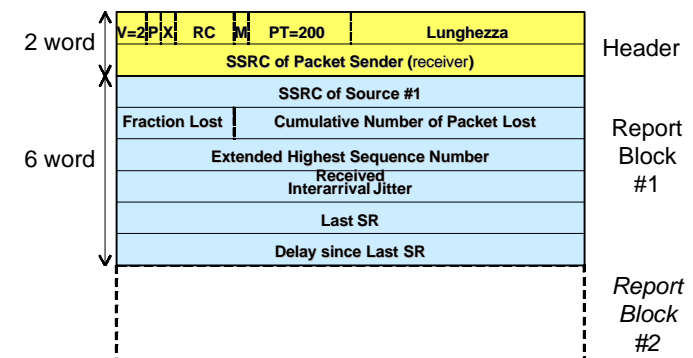
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RTCP Packet Types

- 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

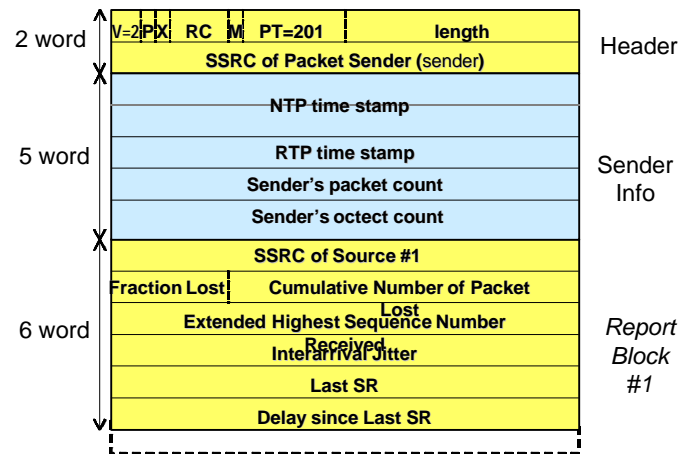
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RTCP Packet Receiver Report (RR)



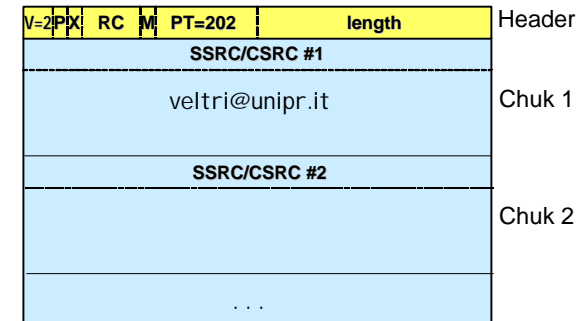
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RTCP Packet Sender Report (SR)



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Pacchetto RTCP Source Description (SDS)



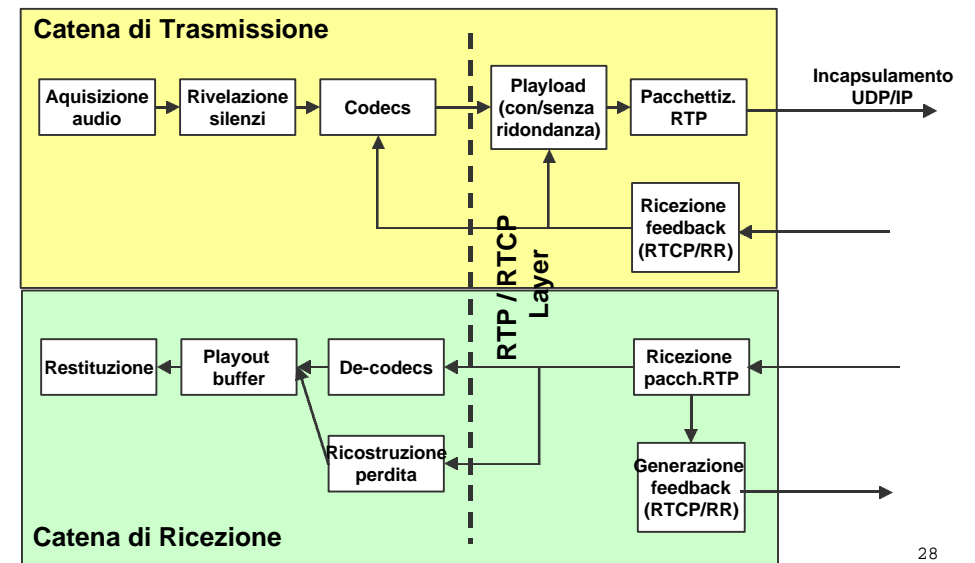
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RTCP Transmission Rate

- 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%

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TX/RX



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