

Real-Time Transport Protocol

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RTP RFCs

- RTP (and RTCP)
 - RFC 1998 (1996), RFC 3550 (2003)
- Basic audio and video profiles
 - RFC 1980 (1996), RFC 3551 (2003)
- High quality audio
 - RFC 3190 (2002)
- Session description protocol (SDP)
 - RFC 2327 (1998), RFC 4566 (2006)

Sessions

- Multimedia session
 - Set of concurrent RTP sessions among a common group of participants
- RTP session
 - A media stream. Separate RTP sessions may be used for associated audio and video.
 - Point-to-point
 - Point-to-multipoint
 - Multipoint-to-multipoint

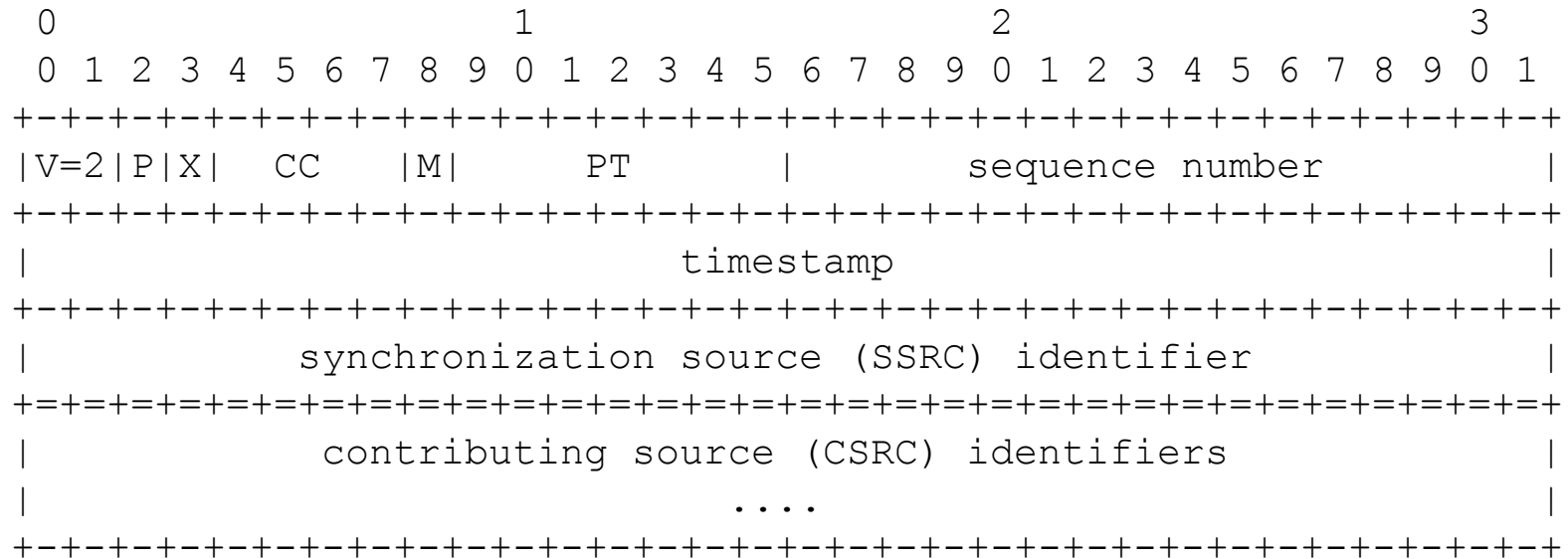
RTP packet

Ethernet header
IP header
UDP header
RTP header
RTP header extension (optional, profile specific)
Payload
Ethernet frame check sequence

RTP IP parameters

- RTP datagrams uses even port numbers
- Corresponding RTCP messages uses next-higher odd port number
- Unicast or IP multicast destination address is used

RTP header



RTP header fields

- **V**ersion (2 bits) – 2 is current version
- **P**adding (1 bit) – 1 indicates presence of one or more padding octets are appended. First pad octet indicates amount of padding.
- **EX**tension (1 bit) – Header extension present
- **C**ontributing source **C**ount (4 bits) – Number of CSRCs following header
- **M**arker (1 bit) – Application specific marker bit
- ***P**ayload **T**ype (7 bits) – Determines payload interpretation. Static and dynamic payload types are supported.*
- Sequence number (16 bits) – Increments for each RTP packet sent
- *Timestamp (32-bits) – Synchronization information for payload.*
- *Synchronization source (32 bits) – Uniquely identifies a stream source in an RTP domain*
- Contributing sources (0 to 15 items, 32 bits each) – Identifies sources for mixed content.

Synchronization source (SSRC)

- Unique 32-bit identifier within an RTP session
- Applicable to sessions with multiple sources (i.e. multipoint-to-multipoint routing)
- Source IP address is not adequate due to NAT and possibility of multiple sources per device
- Collision resolution and loop detection procedures defined

Payload type

- 128 available payload types (PTs) in 7-bit PT field
- Common static PTs are defined in RFC 3551
 - 0 – 8 bit 64 kb telephone audio
 - 10 – Stereo 44.1 kHz, 16-bit audio
 - 11 – Mono 44.1 kHz, 16-bit audio
- PTs 96 through 127 are reserved for dynamic allocation within an RTP session
 - SIP and SDP are often used to communicate these assignments
- Source may change PT during a session
 - Telephone audio vs. comfort noise
 - HDMI mode change
 - Marker bit used to flag a change

Timestamp

- Reflects sampling instant for beginning of payload data
 - Origination time
 - Receivers determine presentation time
- Units and offset are PT and SSRC specific
- Sample period is used in audio applications
- Clock must advance monotonically and linearly (with 32-bit wrap)
- Resolution must meet greater of application synchronization accuracy requirements and requirements for producing useful jitter measurement at receivers
- Timestamp mapped to real time clock in RTCP sender reports

RTP Control Protocol (RTCP)

```

0          1          2          3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
header |V=2|P|      RC      |      PT=SR=200      |      length      |
+-----+-----+-----+-----+-----+-----+-----+-----+
|
+=====+
sender |      NTP timestamp, most significant word      |
info   +-----+-----+-----+-----+-----+-----+-----+
|      NTP timestamp, least significant word      |
+-----+-----+-----+-----+-----+-----+-----+
|      RTP timestamp      |
+-----+-----+-----+-----+-----+-----+-----+
|      sender's packet count      |
+-----+-----+-----+-----+-----+-----+-----+
|      sender's octet count      |
+=====+
report |      SSRC_1 (SSRC of first source)      |
block  +-----+-----+-----+-----+-----+-----+-----+
1      | fraction lost |      cumulative number of packets lost      |
+-----+-----+-----+-----+-----+-----+-----+
|      extended highest sequence number received      |
+-----+-----+-----+-----+-----+-----+-----+
|      interarrival jitter      |
+-----+-----+-----+-----+-----+-----+-----+
|      last SR (LSR)      |
+-----+-----+-----+-----+-----+-----+-----+
|      delay since last SR (DLSR)      |
+=====+
report |      SSRC_2 (SSRC of second source)      |
block  +-----+-----+-----+-----+-----+-----+-----+
2      |      :      ...      :      |
+=====+
|      profile-specific extensions      |
+-----+-----+-----+-----+-----+-----+-----+

```

RTCP message types

- Sender report
 - RTP to NTP time mapping
 - Packet and octet counts
- Receiver report
 - Packet loss, data and sender reports
 - Jitter on data packet arrivals
 - Round-trip delay
- Source description
 - Machine name, user name and contact info, etc.
- Goodbye
- Application specific

RTCP transmission

- Multiple reports may be packed into a single transmission
- Cooperative periodic scheduling desires to use a fixed percentage of bandwidth for RTCP vs. RTP
- Information refreshed at different rates based on priority and timeliness

IEEE 1733

- Defines an application-specific RTCP message type (208) transmitted periodically by each source (talker)
 - StreamID (64 bits)
 - GMIdentity (64 bits)
 - 802.1AS to RTP time mapping