RTCP Sender Report

- **SR**
  - Header Info
  - Sender Info
  - Receiver report blocks
  - Option
    - Profile-specific extension

<table>
<thead>
<tr>
<th>Header Info</th>
<th>Sender Info</th>
<th>Receiver report blocks</th>
<th>Option</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>V=2</strong></td>
<td><strong>P</strong></td>
<td><strong>PT=SR=200</strong></td>
<td><strong>Length</strong></td>
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<tr>
<td><strong>header</strong></td>
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<tr>
<td><strong>SSRC of sender</strong></td>
<td><strong>NTP Timestamp (most significant word)</strong></td>
<td><strong>RTP Timestamp</strong></td>
<td><strong>sender's packet count</strong></td>
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<tr>
<td><strong>SSRC_1 (SSRC of first source)</strong></td>
<td><strong>fraction lost</strong></td>
<td><strong>cumulative number of packets lost</strong></td>
<td><strong>interarrival jitter</strong></td>
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<tr>
<td><strong>SSRC_2 (SSRC of second source)</strong></td>
<td><strong>delay since last SR (DLSR)</strong></td>
<td><strong>profile-specific extensions</strong></td>
<td></td>
</tr>
</tbody>
</table>

VoIP 2-39
Resemble to a RTP packet

- Version
  - 2
- Padding bit
  - Padding octets?
- RC, report count
  - The number of reception report blocks
  - 5-bit
    - If more than 31 reports, an RR is added
- PT, payload type
- SSRC of sender
- **NTP Timestamp**
  - Network Time Protocol Timestamp
    - The time elapsed in seconds since 00:00, 1/1/1900 (GMT)
    - 64-bit
      - 32 MSB: the number of seconds
      - 32 LSB: the fraction of a seconds (200 ps)
  - A primary time server
    - NTP, RFC 1305
    - Station WWV, Fort Collins, Colorado, by NIST
    - Station WWVB, Boulder, Colorado, by NIST
- RTP Timestamp
  - Corresponding to the NTP timestamp
  - Use the same units and has the same offset as used for RTP timestamps
  - For better synchronization
- Sender’s packet count
  -Cumulative within a session
- Sender’s octet count
  -Cumulative
RR blocks

- SSRC_n
  - The source identifier

- Fraction lost
  - Fraction of packets lost since the last report issued by this participant
  - By examining the sequence numbers in the RTP header

- Cumulative number of packets lost
  - Since the beginning of the RTP session

- Extended highest sequence number received
  - The sequence number of the last RTP packet received
  - 16 lsb, the last sequence number
  - 16 msb, the number of sequence number cycles
- Interarrival jitter
  - An estimate of the variance in RTP packet arrival

- Last SR Timestamp (LSR)
  - The middle 32 bits of the NTP timestamp used in the last SR received from the source in question
  - Used to check if the last SR has been received

- Delay Since Last SR (DLSR)
  - The duration in units of 1/65,536 seconds
RTCP Receiver Report

- RR
  - Issued by a participant who receives RTP packets but does not send, or has not yet sent
  - Is almost identical to a SR
    - PT = 201
    - No sender information
RTCP Source Description Packet

- Provides identification and information regarding session participants
  - Must exist in every RTCP compound packet

- Header
  - V, P, SC, PT=202, Length

- Zero or more chunks of information
  - An SSRC or CSRC value
  - One or more identifiers and pieces of information
    - Email address, phone number, name
    - Defined in RFC 1889
    - A unique CNAME
      - E.g., user@host
- **RTCP BYE Packet**
  - Indicate one or more media sources are no longer active

- **Application-Defined RTCP Packet**
  - For application-specific data
  - For non-standardized application
Calculating Round-Trip Time

- Use SRs and RRs
- E.g.
  - Report A: A, T1 → B, T2
  - Report B: B, T3 → A, T4
  - RTT = T4-T3+T2-T1
  - RTT = T4-(T3-T2)-T1
  - Report B
    - LSR = T1
    - DLSR = T3-T2
Calculation Jitter

- The mean deviation of the difference in packet spacing at the receiver
  - $S_i =$ the RTP timestamp for packet $i$
  - $R_i =$ the time of arrival
  - $D(i,j) = (R_j - S_j) - (R_i - S_i)$
- The Jitter is calculated continuously
  - $J(i) = J(i-1) + (|D(i-1,i)| - J(i-1))/16$
Timing of RTCP Packets

- RTCP provides useful feedback
  - Regarding the quality of an RTP session
  - Delay, jitter, packet loss
  - Be sent as often as possible
    - Consume the bandwidth
    - Should be fixed at 5%

- An algorithm, RFC 1889
  - Senders are collectively allowed at least 25% of the control traffic bandwidth
  - The interval > 5 seconds
  - 0.5 – 1.5 times the calculated interval
  - A dynamic estimate the avg RTCP packet size