Real-Time Protocol, RTP

- **TCP/IP**
  - Not suitable for real-time
  - Retransmissions can lead to high delay and cause delay jitter
  - Does not support multicast
  - Congestion control mechanism (slow start) not suitable for AV media

- **UDP/IP**
  - No defined technique for synchronizing
  - A feedback channel must be defined for quality control

- **RTP/RTCP**
  - A session consists of an RTP/RTCP pair of channels
  - Usually works over UDP/IP
  - End-to-end protocol

- **RTP Features**
  - Multicasting
  - Payload type identification
  - Time shaping
  - Sequencing
  - Delivery monitoring

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**Ideal Timing**

- **Send time**
- **Play time**

<table>
<thead>
<tr>
<th>Send time</th>
<th>Play time</th>
</tr>
</thead>
<tbody>
<tr>
<td>00.00.00</td>
<td>00.00.10</td>
</tr>
<tr>
<td>00.00.20</td>
<td>00.00.21</td>
</tr>
<tr>
<td>00.00.30</td>
<td>00.00.31</td>
</tr>
</tbody>
</table>

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**Reality**

<table>
<thead>
<tr>
<th>Send time</th>
<th>Play time</th>
</tr>
</thead>
<tbody>
<tr>
<td>00.00.00</td>
<td>00.00.11</td>
</tr>
<tr>
<td>00.00.10</td>
<td>00.00.21</td>
</tr>
<tr>
<td>00.00.20</td>
<td>00.00.31</td>
</tr>
<tr>
<td>00.00.30</td>
<td>00.00.41</td>
</tr>
<tr>
<td>00.00.40</td>
<td>00.00.51</td>
</tr>
</tbody>
</table>
Jitter

- Playback Buffer

At time 00:00:18

At time 00:00:28

At time 00:00:38

Sequence Numbers and Timestamps

- Consider audio data
  - What should the sender do during silence?
- Not send anything
  - Why might this cause problems?
- Receiver cannot distinguish between loss and silence

Solution:
- After receiving no PDUs for a while, next PDU received at the receiver will reflect a big jump in timestamp, but have the correct next seq. no. Thus, receiver knows what happened.
RTP/RTCP

- UDP doesn’t indicate ways to detect packet loss and restore packet sequence.
- RTP recover these problems (using sequence number, time stamping).
- RTP provides appropriate mechanisms used with QoS protocols.

RTCP

- What is RTCP?
  - Internet Protocol to monitor the quality of service and to convey information about the participants in an on-going session
- What RTCP does?
  - Works in conjunction with RTP
  - Each participant in RTP session periodically transmits RTCP control packets to all other participants.
  - Feedback can be used to control performance
    - Sender may modify its transmission based on feedback
      - Each RTCP packet contains sender and/or receiver reports
        - Statistics include number of packets sent, number of packets lost, interarrival jitter, etc.

RTCP Message types

- Sender report 200
- Receiver report 201
- Source description message 202
- Bye message 203
- Application specific message 204
Synchronization of Streams

- Timestamps in RTP PDUs are tied to the individual video and audio sampling clocks
- Timestamps are not tied to the wall-clock time, or each other!
- Each RTCP sender-report PDU contains (for most recently generated PDU in associated RTP stream):
  - The timestamp of RTP PDU
  - The wall-clock time for when PDU was created
- Receivers can use this association to synchronize the playout of audio and video

RTCP Bandwidth Scalability

- Uses 5% of session bandwidth.
  - Shared by all participants
  - 25% allocated to senders
  - 75% allocated to receivers
- Scaling characteristics
  - Reports sent periodically with random delay
  - Period based on estimated number of participants

Real-Time Streaming Protocol (RTSP)

- What RTSP does?
  - Allow a media player to control the transmission of a media stream.
    - (i.e. Pause/resume, repositioning of playback, fast forward and rewind).
  - Retrieve a media from a server.
  - Invite a server to add a media in an existing session.
  - Record a conference.
- What RTSP does NOT?
  - Does not define compression.
  - Define how audio and video are encapsulated.
  - Restrict how the media player buffers the audio/video.
Overview

RTSP Methods

OPTIONS  Get available methods
SETUP    Establish transport
ANNOUNCE Change description of media object
DESCRIBE Get (low-level) description of media object
PLAY     Start playback, reposition
RECORD   Start recording
REDIRECT Redirect client to new server
PAUSE    Halt delivery, but keep state
SET PARAMETER Device or encoding control
TEARDOWN Remove state

RTSP Operations
RTP, RTCP, RTSP?

- RTP is a transport protocol for the delivery of real-time data, including streaming audio and video.
- RTCP is a part of RTP and helps with QoS management.
- RTSP is a control protocol for initiating and directing delivery of streaming multimedia from media servers, the “Internet VCR remote control protocol”.
- RTSP does not deliver data, though the RTSP connection may be used to tunnel RTP traffic for ease of use with firewalls and other network devices.
- RTP and RTSP will likely be used together in many systems, but either protocol can be used without the other. The RTSP specification contains a section on the use of RTP with RTSP.