Popular protocols for serving media

- Network transmission control
  - RTP – Realtime Transmission Protocol
  - RTCP – Realtime Transmission Control Protocol

- Session control
  - Real-Time Streaming Protocol (RTSP)
  - Session Description Protocol (SDP) – textual representation of session

- VOIP – SIP – Session Initiation Protocol
  - Signaling for IP Telephony

- SAP – Session announcement protocol for multicast sessions
RTP and RTSP

- RTP usage – in several application audio and video tools (vat, vic)
- RTP follows the principle of application level framing and integrated layer processing
- RTP/UDP/IP is being used by the current streaming session protocols such as RTSP
- Session protocols are actually negotiation/session establishment protocols that assist multimedia applications
- Multimedia applications such as QuickTime, Real Player and others use them
Real-time Transmission Protocol (RTP)

- RTP provides end-to-end transport functions suitable for real-time audio/video applications over multicast and unicast network services.

<table>
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<tr>
<th>Layer 4</th>
<th>RTP</th>
<th>RTCP</th>
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<td>User Datagram Protocol</td>
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<tr>
<td>Internet Protocol</td>
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<tr>
<td>Ethernet 802.13 or Wi-Fi 802.11</td>
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<tr>
<td>PHY (Wired or Wireless)</td>
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</table>
Relation between RTP and RTCP

```
Application
Decoding → Coding
RTP  →  RTCP

UDP/IP
```

```
Application
Coding → Decoding
RTCP  →  RTP

UDP/IP
```
RTCP: Control and Management

- Out-of-band control information for RTP flow.
  - Monitors QoS for RTP in the delivery and packaging of multimedia data
  - Used periodically to transmit control packets to participants in a streaming multimedia session.
  - Provides feedback on the quality of service being provided by RTP
  - Gathers statistics on media connection
    - Bytes sent, packets sent, lost packets, jitter, feedback and round trip delay
    - Application may use this information to increase the quality of service, perhaps by limiting flow or using a different codec
RTCP Functions

- There are several types of RTCP packets:
  - Sender report packet,
  - Receiver report packet,
  - Source Description RTCP Packet,
  - Goodbye RTCP Packet and
  - Application Specific RTCP packets.

- RTCP itself does not provide any flow encryption or authentication means. [SRTCP](#) protocol can be used for that purpose.
RTP Services

- Payload Type Identification
  - Determination of media coding
  - Source identification
  - RTP works with Profiles
    - Profile defines a set of payload type codes and their mappings to payload formats

- Sequence numbering
  - Error detection

- Time-stamping
  - Time monitoring, synchronization, jitter calculation

- Delivery monitoring
RTP Services – Support of heterogeneity

- **Mixer service**
  - Allows for resynchronization of incoming audio packets
  - Reconstructs constant 20 ms spacing generated by sender
  - Mixes reconstructed audio streams into single stream
  - Translated audio encoding to lower bandwidth
  - Forwards lower bandwidth packet streams

- **Translator service**
  - Allows for translation between IP and other high speed protocols
  - May change encoding data
Payload Formats

- **Static Payload formats**
  - Established in RTP Profile
  - Payload type 0 := μ-law audio codec
- **Dynamic Payload formats**
  - Applications agree per session on payload format
  - H.263, JPEG, MPEG
Session Management (Layer 5)

- Important part of multimedia communication
- Separates control aspects from transport aspects

**SESSION MANAGER**

- Conference control
- Participant Management
- Configuration control
- Media control

**Session Control Protocol**

**Presentation data communication**
- Continuous data communication
- Continuous data communication

**video** → **whiteboard**

**audio** → **video**
Session Manager

- Tasks:
  - Membership control
  - Monitoring of shared workspace
  - Coordination of Media control management
  - Exchange of QoS parameters
  - Conference control management – establishment, modification, termination
Session Control

- Session Described by
  - Session state
    - Name of session, start, valid policies
- Session management – two steps for state processing
  - Establishment of session
  - Modification of session
Session Control

- Conference Control
  - Centralized or distributed approach

- Media Control
  - Synchronization

- Configuration Control
  - Negotiation of QoS parameters, admission control and reservation/allocation of resources

- Membership Control
  - Invitation of users; registration of users, change of membership
RTSP

- Enables controlled, on-demand delivery of real-time data such as audio and video
- Intends to control multiple data delivery sessions
- Provides means for choosing delivery channels
  - UDP
  - Multicast UDP,
  - TCP
Real-Time Streaming Protocol (RTSP)

- Application Protocol for control of multimedia streams
- This is not an application data transmission protocol, just remote control protocol between client and server

Diagram:
- Client: Audio Video Decoder, RTP, RTSP
- Server: RTP, RTSP, Audio Video Decoder
- Session Control
## RTSP Methods

<table>
<thead>
<tr>
<th>Request</th>
<th>Direction</th>
<th>Description</th>
</tr>
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<tbody>
<tr>
<td>OPTIONS</td>
<td>S &lt;-&gt; C</td>
<td>Determine capabilities of server (S) or client (C)</td>
</tr>
<tr>
<td>DESCRIBE</td>
<td>C -&gt; S</td>
<td>Get description of media stream</td>
</tr>
<tr>
<td>ANNOUNCE</td>
<td>S &lt;-&gt; C</td>
<td>Announce new session description</td>
</tr>
<tr>
<td>SETUP</td>
<td>C -&gt; S</td>
<td>Create media session</td>
</tr>
<tr>
<td>RECORD</td>
<td>C -&gt; S</td>
<td>Start media recording</td>
</tr>
<tr>
<td>PLAY</td>
<td>C -&gt; S</td>
<td>Start media delivery</td>
</tr>
<tr>
<td>PAUSE</td>
<td>C -&gt; S</td>
<td>Pause media delivery</td>
</tr>
<tr>
<td>REDIRECT</td>
<td>S -&gt; C</td>
<td>Use other server</td>
</tr>
<tr>
<td>TEARDOWN</td>
<td>C -&gt; S</td>
<td>Destroy media session</td>
</tr>
<tr>
<td>SET_PARAMETER</td>
<td>S &lt;-&gt; C</td>
<td>Set server or client parameter</td>
</tr>
<tr>
<td>GET_PARAMETER</td>
<td>S &lt;-&gt; C</td>
<td>Read server or client parameter</td>
</tr>
</tbody>
</table>
RTSP Extensions

- **Timing**
  - RTSP needs to hide latency variations
  - PLAY request may contain information about when request is to be executed

- **Three types of timestamps**
  - SMPTE (the same as in TV production)
    - Format: hours:minutes:seconds:frames
  - Normal play time
    - Measured relative to beginning of stream and expressed in ours, minutes, seconds and fractions of second
  - Absolute time
    - Wall clock
Session Description Protocol (SDP)

- Text format for describing multimedia sessions
- Not really a protocol (similar to markup language like HTML)
- Can be carried in any protocol, e.g., RTSP or SIP
- Describes unicast and multicast sessions
There are five terms related to multimedia session description:

- Conference: It is a set of two or more communicating users along with the software they are using.
- Session: Session is the multimedia sender and receiver and the flowing stream of data.
- Session Announcement: A session announcement is a mechanism by which a session description is conveyed to users in a proactive fashion, i.e., the session description was not explicitly requested by the user.
- Session Advertisement: same as session announcement
- Session Description: A well defined format for conveying sufficient information to discover and participate in a multimedia session.
Sample SDP file

v=0
o=- 19 1077294547 IN IP4 127.0.0.0
s=QuickTime
t=0 0
a=range:npt=now-
a=control:rtsp://127.0.0.1/mystream.sdp
a=isma-compliance:2,2.0,2
m=audio 0 RTP/AVP 96
c=IN IP4 0.0.0.0
b=AS:8
a=rtpmap:96 mpeg4-generic/8000/1
a=fmtp:96 profile-level-id=15;mode=AAC-
hbr;sizeLength=13;indexLength=3;indexDeltaLength=3;config=1588
a=mpeg4-esid:101
m=video 0 RTP/AVP 97
c=IN IP4 0.0.0.0
b=AS:30
a=rtpmap:97 H264/90000
a=fmtp:97 packetization-mode=1;profile-level-id=4D400A;sprop-parameter-sets=J01ACqkYUI/LgDUGAQa2wrXvfAQ=,KN4JF6A=
a=mpeg4-esid:201
a=cliprect:0,0,120,160
a=framesize:97 160-120
VOIP
Signaling for IP Telephony

- Internet Telephone — needs ability of one party to signal to other party to initiate a new call

- Call — association between a number of participants

  - Note: there is no physical channel or network resources associated with the session layer connection, the connection exists only as signaling state at two end points
IP Telephony Signaling Protocol
(Requirements)

- Name translations and user location
  - Mapping between names of different levels of abstraction
    - Email address to IP address of host
- Feature negotiation
  - Group of end systems must agree on what media to exchange and their respective parameters
    - Different encodings, rates
- Call Participant Management
  - Invite participants to existing call, transfer call and hold other users
IP Telephony Signaling
(Requirements)

- Feature change
  - Adjust composition of media sessions during the course of call
    - Add or reduce functionality
    - Impose or remove constraints due to addition or removal of participants

- Two signaling protocols:
  - SIP (IETF Standard)
  - H.323 (ITU Standard)
SIP (Session Initiation Protocol)

- SIP Goal: invite new participants to call
- Client-Server protocol at the application level
- Protocol:
  - User/Client creates requests and sends to server;
  - User agent server responds;
- SIP requests can traverse many proxy servers
- Server may act as redirect server
- Proxies or redirect servers cannot accept/reject requests, only user agent server can
- Requests/Responses are textual
### SIP - Message

- Calls in SIP – have unique call ID (carried in Call-ID header field of SIP message)
- Call identifier is created by the caller and used by all participants
- SIP messages have information
  - Logical connection source
  - Logical connection destination
  - Media destination
  - Media capabilities (use SDP)
SIP – Addressing and Naming

To be invited and identified, called party must be named

SIP chooses email-like identifier
- user@domain
- user@host
- user@IPaddress
- phone-number@gateway

SIP’s address: part of SIP URL
- sip:j.doe@example.com
- URL can be placed on web page

Interactive audio/video requests translation
- name@domain to host@host
Call Setup Process using SIP

1. INVITE sip:johnsmith@test.com
2. INVITE sip:johnsmith@test.com
3. Where is johnsmith?
4. At Jsmith
5. INVITE sip:johnsmith@jsmith.test.com
6. 200 OK
7. 200 OK
8. 200 OK
9. ACK
10. RTP Audio/Video data
SIP Redirect Server Operation

1. INVITE sip:johnsmith@test.com

2. Where is johnsmith?

3. At play

4. 302 Moved temporarily
   Contact: sip:johnsmith@play.test.com

5. INVITE sip:johnsmith@play.test.com

6. 200 OK

7. RTP Audio/Video data

SIP user agent

Location Service

play
SIP Requests/Methods

- **INVITE**—Indicates a client is being invited to participate in a call session.
- **ACK**—Confirms that the client has received a final response to an INVITE request.
- **BYE**—Terminates a call and can be sent by either the caller or the callee.
- **CANCEL**— Cancels any pending searches but does not terminate a call that has already been accepted.
- **OPTIONS**—Queries the capabilities of servers.
- **REGISTER**—Registers the address listed in the To header field with a SIP server.
SAP – Session Announcement Protocol

- RTSP and SIP are designed for one-on-one session
- SAP is multicast announcement protocol
- Protocol
  - Distributed servers periodically send multicast packets (advertisements) containing descriptions of sessions generated by local sources
  - Advertisements are received by multicast receivers on well-known, static multicast address/port
- Advertisement contains SDP information to start media tools needed in the session