Communication Networks II
Multimedia Communications / QoS
Specific Topics:
RTP, RTCP

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Scope

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<tr><td rowspan="2">L5</td><td>Applications</td><td rowspan="2">Terminal access</td><td rowspan="2">File access</td><td rowspan="2">E-mail</td><td rowspan="2">Web</td><td colspan="2" rowspan="2">Peer-to-Peer</td><td rowspan="2">Inst.-Msg.</td><td>IP-Tel.</td></tr>
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UDP, TCP, SCTP</td><td rowspan="3">Netw. Transitions</td><td rowspan="3">Security</td><td rowspan="3">Addressing</td><td colspan="2">Transport QoS - RTP</td></tr>
<tr><td>L3</td><td>Network Layer
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Legend: **KN I** **KN II**

Overview

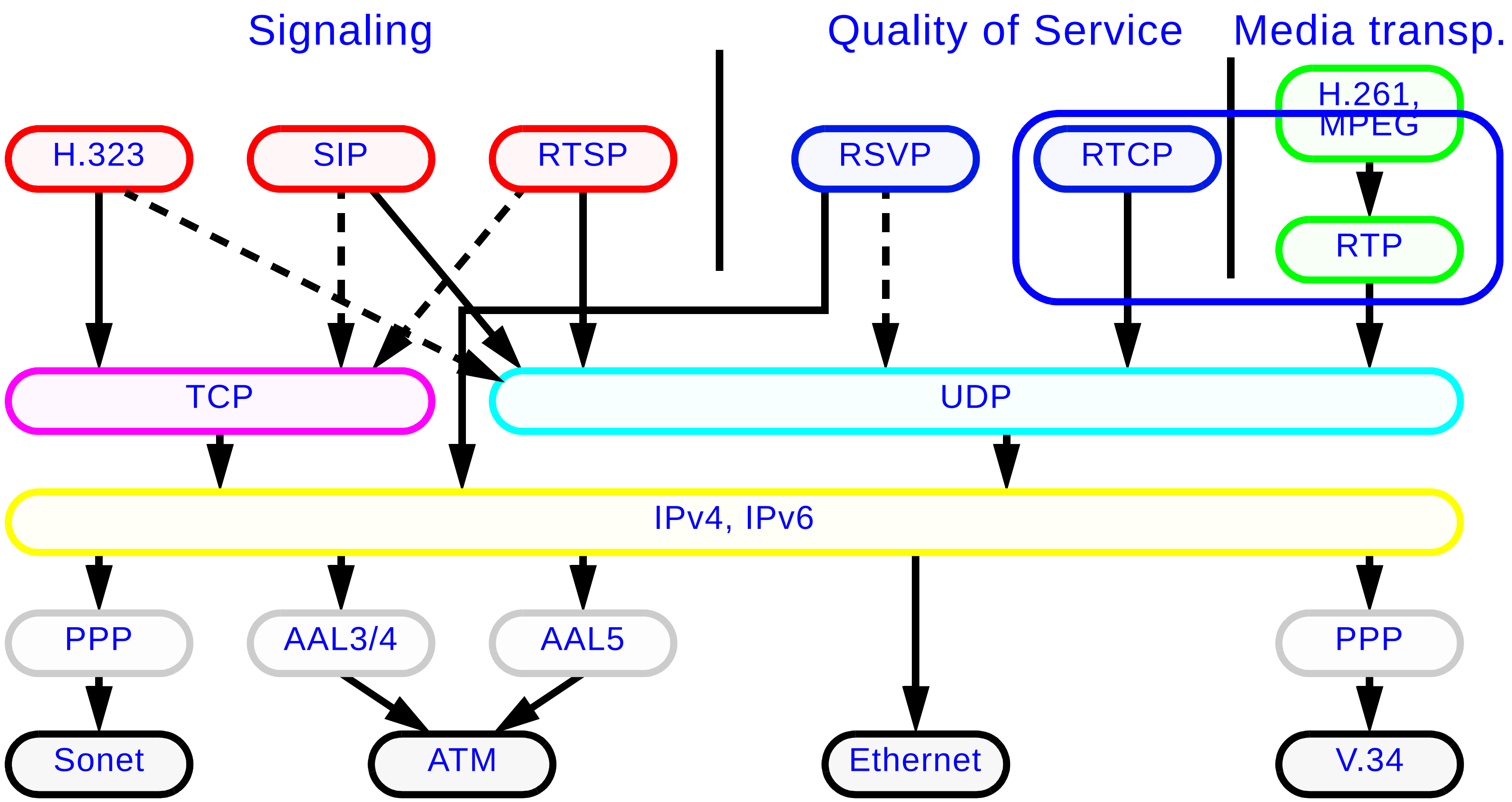
1. **Motivation: Real-time and Multimedia Protocols of the Internet**

2. **RTP + RTCP Duties**

3. **Packet Format: RTP + RTCP**

4. **Mixer & Translator**

1. Motivation:
Real-time and Multimedia Protocols of the Internet



Multimedia Transport Protocols

Motivation

- **separate flows for each media stream**
  - simplifies applications
  - allows for different QoS
- **receiver adaptation**
  - buffering to smooth out jitter
    - which usually always exists to some extend
  - timestamp necessary
- **synchronization**
  - of various media streams
  - adapting play-out buffers
- **framing service**
  - splitting media stream into adequate PDUs

⇒ **Real-Time Transport Protocol RTP**

- **see**
  - RFC 1889, RTP: A Transport Protocol for Real-Time Applications
  - RFC 1890, RTP Profile for Audio and Video Conferences with Minimal Control

`and around e.g. http://www.cs.columbia.edu/~hgs/rtp/`

2. RTP + RTCP Duties

Transport Layer

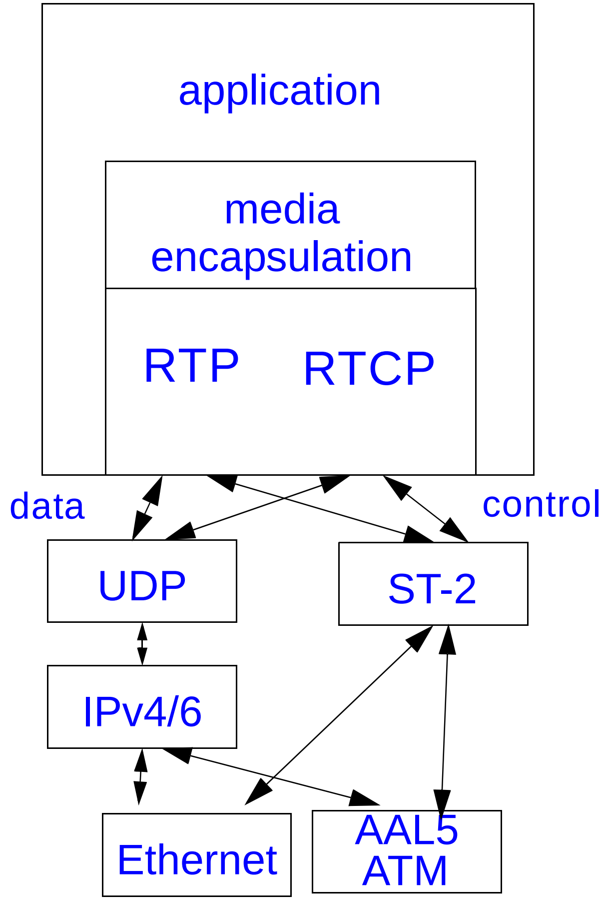
- **Real-Time Transport Protocol (RTP)**
- **Real-Time Transport Control Protocol**

End-to-end transport functions, but is NOT

- a real transport protocol
  - no checksums
  - no multiplexing
- a real-time protocol
  - no reservations
  - no guarantees

Adds functionality to existing transport protocols

- designed to work with UDP
  - works also with TCP
  - also to work with ATM (AAL5)
- **functions like**
  - session layer (in OSI terminology)
  - integrated with applications



Real-Time Transport Control Protocol (RTCP)

Companion protocol to RTP

- **function:**
  - monitoring QoS
  - to convey information about
    - participants
    - session relationships
- **monitor application performance**
  - feedback to sender about delivery quality, loss, etc.
- **automatic adjustment to overhead**
  - report frequency based on participant count

Typically,

- **"RTP does ..." means "RTP with RTCP does ..."**

RTP with RTCP Functions

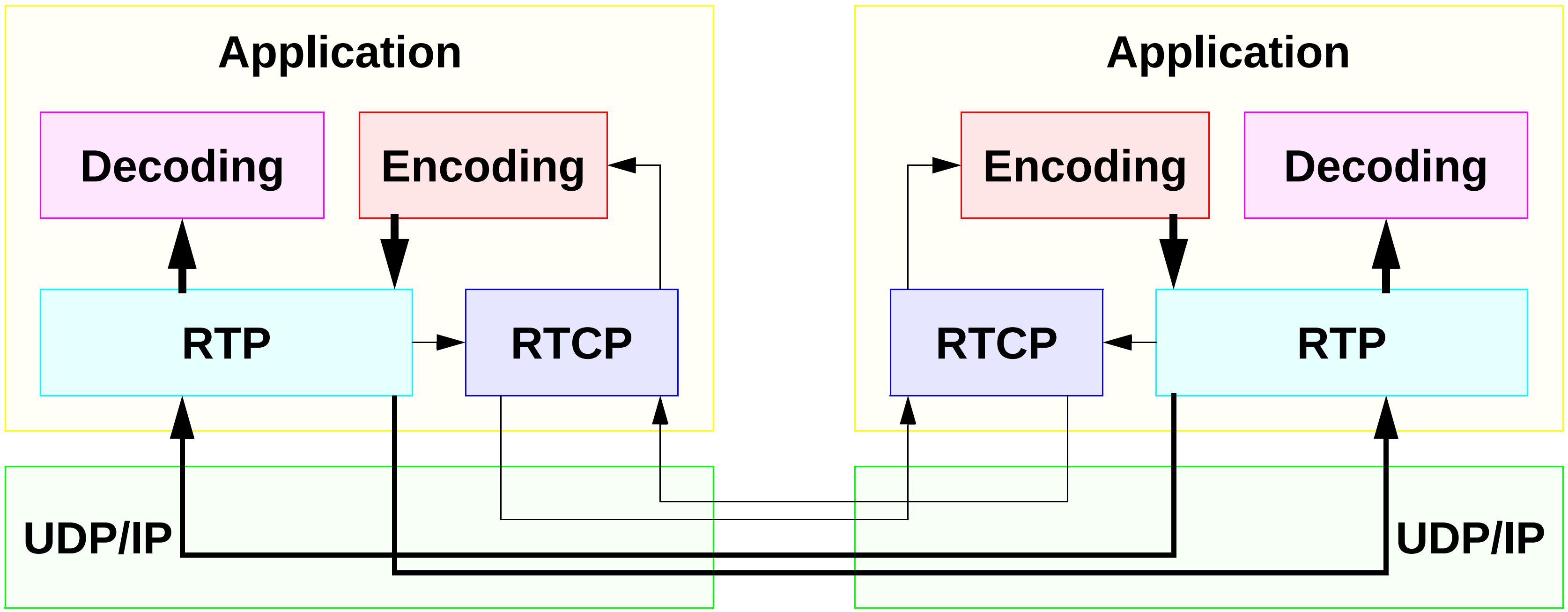
RTP with RTCP provide:

- **support for transmission of real-time data**
- **over multicast or unicast network services**

functional basis for this:

- **sequence numbering**
- **determination of media encoding, i.e. payload type identification**
- **source identification (process), i.e. payload type identification**
- **synchronization**
- **framing i.e. follows principle of**
  - application level framing and
  - integrated layer processing
- **error detection, i.e. delivery monitoring**
- **encryption**
- **timing, i.e. time stamping**
- **unicast and multicast support**
- **support for stream “translation” and “mixing”**

RTP+RTCP: Quality Control



Component interoperations for control of quality

- evaluation of sender and receiver reports
- modification or encoding schemes and parameters
- adaptation of transmission rates

3. Packet Format: RTP + RTCP

Header structure

bytes	20	8	12	
	IP hdr	UDP	RTP HDR	Payload

Uses (most commonly) UDP

- simple
- unreliable
- connectionless
- multicast

- **RTPC to control the stream**
  - media encoding with profiles

RTP Profile	Media Enc.	Bits/Sample	Samp.Rate	Packet Rate
0	PCM μLaw	8	variable	
6	DVI14	4	16 kHz	20 ms
9	G.722	8	16 kHz	20 ms

RTP Header

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

V 2 bit – Version number (2)
P 1 bit – Padding (last byte contains information how many padding bytes are inserted)
X 1 bit – Extension (extension header inserted)
CC 4 bit – CSRC Count (-Number of mixed Sources- max. 15)
M 1 bit – Marker (payload profile dependent; e.g., frame-boundaries)
PT 7 bit – Payload Type
Sequence number 16 bit
Timestamp 32 Bit - timestamp of SSRC
SSRC 32 Bit – synchronization source (random number), identification of sender, whose timestamp is master time stamp
CSRC 32 Bit – list of identifiers of those contributing to (mixed) packet

RTCP Packets

Characteristics

- periodic control packets
- quality control
- participants list
- occupies only a fraction of the used bandwidth

Header

```
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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P| subtype |   PT= SR =200 |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           SSRC/CSRC                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                          name (ASCII)                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                   application-dependent data                ...
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- non application specific, e.g.

SR Sender Report (statistics from active senders: bytes sent -> estimate rate)
RR Receiver Report (statistics from receivers)
SDES Source Descriptions (Canonical Name = user@host; name, email, location, ...)
BYE explicit leave

- application/media specific

APP extensions, application specific

RTP Profiles - Payload Type

RTPC to control the stream

- **media encoding with profiles**

Payload type identification

RTP Profile	Media Enc.	Bits/Sample	Samp.Rate	Packet Rate
0	PCM μLaw	8	variable	
6	DVI14	4	16 kHz	20 ms
9	G.722	8	16 kHz	20 ms

- **each content encoding needs its coding specification**
- **defines a set of payload type codes and their mapping to payload formats**
  - H.261, H.263, H.263+ (ITU-T), Real, ..
  - Motion JPEG, MPEG1 & MPEG2, Bundled MPEG, CellB video encoding
  - BT.656-3 encoding
  - HTTP encoding
  - ASF (Advanced Streaming Format)
  - DTMF (dial tone multiple frequence) Digits
  - Layered Multimedia Streams
  - Redudant Encodings Audio Data

RTP Profiles - Payload Type (2)

Some details

- **CellB video encoding**
  - variable bit-rate video coding scheme
    - "high" quality
    - low bit-rate image
    - at low computational cost.
  - bytestream consists of
    - instructional codes and
    - information about the compressed image
  - Request for Comment 2029

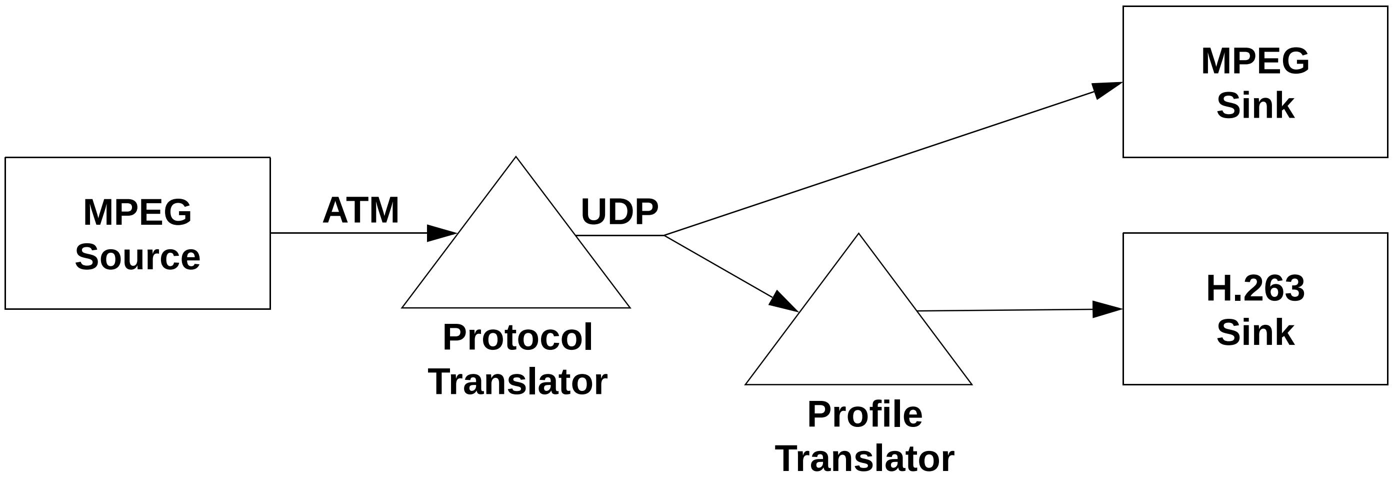
- **ASF Advanced Streaming Format**
  - the payload format for encapsulating Advanced Streaming Format
  - extensible file format for synchronized multimedia streams
  - not tied to any particular media type or compression scheme
  - allows a user to create new media type
    - define a general encapsulation scheme that allows any ASF stream, regardless of its media type, to be sent as an RTP payload
  - ASF that do not require reliable transmission.

4. Mixer & Translator

Mixer functions

- **reconstructs**
  - constant spacing generated by sender
- **translates**
  - e.g. audio encoding to a lower-bandwidth
- **mixes**
  - reconstructed audio streams into a single stream
- **resynchronizes**
  - incoming audio packets
    - new synchronization source value (SSRC) stored in packet
    - incoming SSRCs are copied
      into the contributing sync. source list (CSRC)
- **forwards**
  - mixed packet stream

RTP Translator



- **translation between**
  - IP and other protocols or protocol families
  - e.g., between IP and e.g. Stream Type Protocol ST-2
- **two translators are installed**
  - may change the encoding of data
  - no resynchronization in translators
- **SSRC and CSRC remain unchanged**

SSRC synchronization source (random number), identification of sender, whose timestamp is master time stamp

CSRC list of identifiers of those contributing to (mixed) packet

RTP Identifiers