RTP

The Real-Time Transport Protocol

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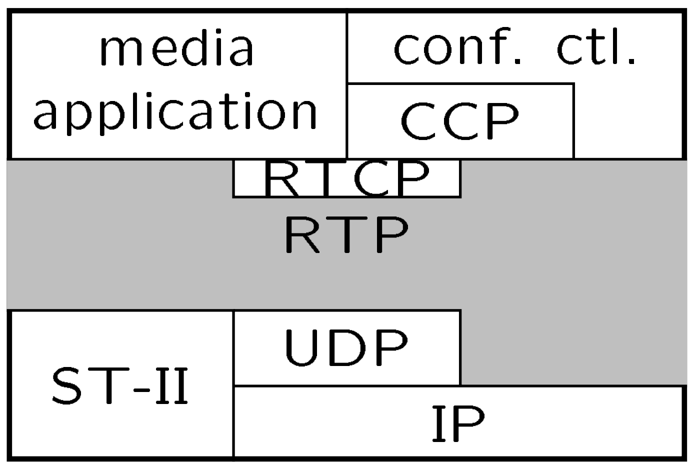
Overview

RTP = Transport services for *real-time* applications (audio, video, simulation, remote control, ...)

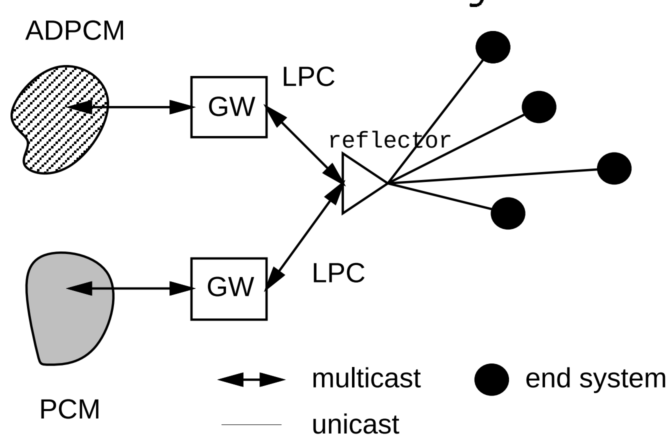
- architecture and concepts
- goals and services
- protocol components:
  - RTP
  - RTCP
- status, future plans

Architecture and Concepts

Embedding into Internet protocols:



Gateways, reflectors and end systems:



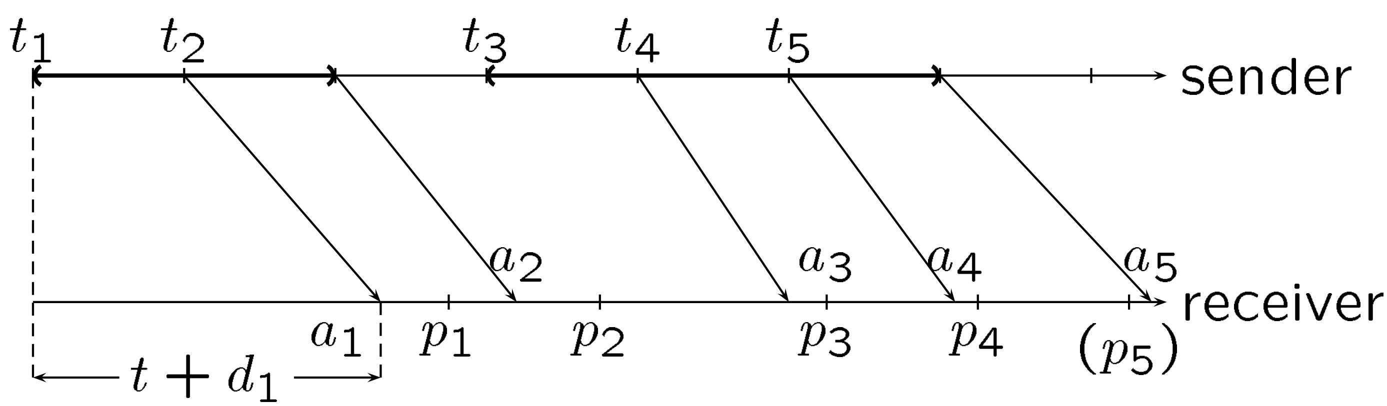
Synchronization

playout synchronization

intra-media synchronization (simulation)

inter-media synchronization (lip sync)

$\Rightarrow$use *synchronization units* (voice: talkspurts) to adjust delays to changing network conditions



Goals

- content flexibility (not just audio and video)
- extensible ⇒options
- independent of lower-layer protocols (IP, ST-II, UDP, ...)
- unicast and multicast
- gateway-compatible
- bandwidth efficient ⇒keep header to 8 bytes
- international (text, encodings)
- processing efficient ⇒alignment, avoid arithmetic
- implementable with current hardware and operating systems

Protocol Services

RTP: RT transport protocol (unidirectional):

- framing (if needed)
- identify synchronization source
- transfer media data (unreliably)
- demultiplexing (flows, e.g., combined audio and video)
- synchronization and sequencing support
- next layer (e.g., media) identification

RTCP: control protocol (bidirectional):

- identify participants
- describe content
- quality-of-service information
- request for retransmission

Optional:

- packet length (for stream oriented transport)
- synchronization source (for reflectors)

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| packet length (optional)                                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| address of synchronization source (optional)                  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|Ver| flow      |O|S|  content   | sequence number              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| timestamp (seconds)           | timestamp (fraction)          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| options ...                                                   |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Ver: version (value: one)

flow: flow index (multiplexing)

content: index into encoding table

O: options present

S: synchronization bit (*end* of sync unit)

option length: in 32-bit words

sequence number: count packets

timestamp: fixed point, NTP-format, wallclock time

RTP options

RTP, RTCP options share ...

- same format
- same numbering space

content source identification:

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| CSRC          | length = 2    | id unique within host         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| IP address of content source                                  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

beginning of playout unit:

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| BOP           | length = 1    | sequence number               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

RTCP

- forward direction:

  CDESC: content description:

  - map 'content' field into content (media) description
  - port number for reverse control
  - timestamp clock quality
  - content-dependent data

  SDESC: source naming (e.g., login name)

  FDESC: text describing flow content

  BYE: source bids farewell

- reverse direction (from receiver back to source):

  QOS: quality of service (packets received, delay range)

  RTR: request for retransmission

Status and Future Plans

- product of Internet Engineering Task Force (IETF)
- audio-video transport (AVT) working group
- process: drafts $\surd \longrightarrow$ informational RFC, experimental standard $\stackrel{?}{\longrightarrow}$ implementations $\stackrel{?}{\longrightarrow}$ draft standard
- rough consensus at IETF meeting in Washington, D.C. (November 1992)
- four documents:
  1. rationale, design alternatives $\Rightarrow$ informational RFC
  2. protocol specification $\Rightarrow$ experimental standard RFC
  3. profile for audio-visual conferences: names and numbering for encodings, meaning of certain header fields, ... $\Rightarrow$ IANA (Internet Assigned Numbers Authority)
  4. implementors agreement about media encodings (H.261 in packets, multichannel audio, codec API, ...)
- re-use parts (naming, format) in conference control, conference server