Overview

- IETF Audio/Video Transport WG
  - RTPv1 RFC 1889 (January 1996)
  - RTPv2 draft-ietf-avt-rtp-new-09.txt (March 2001)
- Real-Time Protocol (RTP)
  - understand: « a framing protocol for real-time applications »
  - does not define any QoS mechanism for real-time delivery!
- Real-Time Control Protocol (RTCP)
  - its companion control protocol
  - does not guaranty anything either!

Overview (cont’d)

- Protocols and their application field…
  - stream description: SDP, SMIL
    describe the session and content
  - stream control: RTSP
    remote control the session
  - media transport: RTP
    send data and metadata
  - resource reservation (if any!): RSVP, DiffServ
    make sure the communication path offers appropriate guarantees…
    … otherwise Best-Effort transmissions!

Topics

- RTP
- RTCP
- RTSP
RTP Features

- Provide support for the end-to-end transport of real-time data over datagram network
- A framing protocol that does not define any QoS mechanism for real-time delivery
- RTP does not assume anything about the underlying network
- RTP does not offer any reliability or flow/congestion control
- Designed primarily for multicast of real-time data, can also be used in unicast
- Designed for both one-way transport and interactive services

RTP generalities

- Data functions (RTP)
  - content labeling
  - source identification
  - loss detection
  - resequencing
  - timing
    - intra-media synchronization: remove jitter with playout buffers
    - inter-media synchronization: hp-synchro between audio-video

Typical use

- Usually...
  - uses UDP (TCP is not for real-time!!!)
  - no fixed UDP port negotiated out of band
  - UDP port for RTCP = UDP port for RTP + 1
  - usually one media per RTP session (i.e. port pair)

RTP Packet Encapsulation

- IP header
- UDP header
- RTP header
- RTP payload

RTP Packet Format

- Timestamping
  - Sender sets the timestamp. Timestamps increase by the time covered by a packet
  - Used for the receiver to reconstruct the original timing
  - Also used for synchronization of substreams

- Sequence numbering
  - Used to put the packets in order
  - Also used for packet loss detection
RTP Services (cont’d)

• Payload type identifier
  — Specify the payload format and the encoding/compression schemes
  — Help the receivers to know how to interpret and play out the payload data
• Source identification
  — Allows the receiver to know where the data is coming from

RTCP Introduction

• Designed to work in conjunction with RTP
• In RTP sessions, participants periodically send RTCP packets to convey feedback on quality of data delivery and information of membership
• 5 packet types are defined

RTCP generalities

• periodic transmission of control packets
• several functions
  — feedback on the quality of data distribution
  — let everybody evaluate the number of participants
  — persistent transport-level canonical name for a source, CNAME
    • usually user@host
    • will not change, even if SSRC does!
    • provides binding across multiple media tools for a single user

RTCP generalities (cont’d)

• Five RTCP packets
  — SR sender reports
    tx and rx statistics from active senders
  — RR receiver reports
    rx statistics from other participants, or from active senders if more than 31 sources
  — SDES source description, including CNAME
  — BYE explicit leave
  — APP application specific extensions

RTCP generalities (cont’d)

• distribution
  — use same distribution mechanisms as data packets (n-way multicast)
  — multiple RTCP packets can be concatenated by translators/mixers
  ⇒ compound RTCP packet
• scalability with session size
  — RTCP traffic should not exceed 5% of total session bandwidth
  — requires an evaluation of number of participants RTCP tx interval = f(number of participants)
  — at least 25% of RTCP bandwidth is for source reports
  — let new receivers quickly know CNAME of sources!

SR RTCP packets

• includes
  — SSRC of sender identify source of data
  — NTP timestamp when report was sent
  — RTP timestamp corresponding RTP time
  — packet count total number sent
  — octet count total number sent
  — followed by zero or more receiver report…
  — example:
    source 1 reports, there are 2 other sources


RR RTCP packets

- includes
  - SSRC of source: identify the source to which this RR block pertains
  - fraction lost: since previous RR (SR) sent (\(= \text{int}(256 \times \text{lost}/\text{expected})\))
  - cumul # of packets lost: long term loss
  - highest seq # received: compare losses
  - interarrival jitter: smoothed interpacket distortion
  - LSR: time when last SR heard
  - DLSR: delay since last SR

RTCP Services

- QoS monitoring and congestion control
- Source Identification
- Inter-media synchronization
- Control information scaling

RTP/RTCP

- RTP session
  - Application specifies a pair of destination transport addresses (one network address + a pair of ports for RTP and RTCP)
  - In a multimedia session, each medium is carried in a separate RTP session with its own RTCP packets
- RTP/RTCP provides functionality and control mechanisms necessary for carrying real-time content. But they are not responsible for the higher-level tasks such as assembly and synchronization

RTSP

- RTSP introduction
  - A client-server multimedia presentation protocol to enable controlled delivery of streamed multimedia data over IP network
  - An application-level protocol designed to work with lower-level protocols
- RTSP operations
  - Retrieval of media from media server
  - Invitation of a media server to a conference
  - Adding media to an existing presentation

RTSP generalities

- IETF standard
  - RFC 2326
- Real-Time Streaming Protocol
  - acts as a « network remote control »
- supports the following operations:
  - retrieval of a media from a server
  - invitation of a media server to a conference
  - recording of a conference

RTSP generalities (cont’d)

- Protocol design
  - text-based protocol
  - transport protocol independent
  - supports any session description (sdp, xml, etc.)
  - similar design as HTTP with differences yet!
    - client \(\rightarrow\) server and server \(\rightarrow\) client requests
    - server maintains a « session state »
    - data carried out-of-band
    - works either with unicast or multicast
## RTSP URL

- Whole presentation
  - rtsp://media.example.com:554/twister
- A track within the presentation
  - rtsp://media.example.com:554/twister/audio

## RTSP Message

- a request (client → server or server → client)
  
  ```
  PLAY rtsp://video.example.com/twister/video RTSP/1.0
  CSeq: 2
  Session: 23456789
  Range: start=0; end=10
  }
  RTSP/1.0 200 OK
  CSeq: 2
  Session: 23456789
  Range: start=0; end=20
  }
  RTP-Info:
  url=rtsp://video.example.com/twister/video;
  seq=1231232
  ```

## RTSP methods

- Major methods
  - SETUP: server allocates resources for a stream and starts an RTSP session
  - PLAY: starts data rx on a stream
  - PAUSE: temporarily halts a stream
  - TEARDOWN: free resources of the stream, no RTSP session on server any more

- Additional methods
  - OPTIONS: get available methods
  - ANNOUNCE: change description of media object
  - DESCRIBE: get low level descr. of media object
  - RECORD: server starts recording a stream
  - REDIRECT: redirect client to new server
  - SET_PARAMETER: device or encoding control

## 3.2- Example: media on demand, unicast

### Step 1: get description (in SDP format)

```
C 1 0 8 7 6 5 4 3 2 1 0
media descr
```

### Step 2: open streams with RTSP

```
W 1 0 8 7 6 5 4 3 2 1 0
audio descr
```

### Step 3: play

```
V 1 0 8 7 6 5 4 3 2 1 0
video descr
```

### Step 4: teardown

```
C 1 0 8 7 6 5 4 3 2 1 0
```

## RTSP Implementations

- Apple Darwin QuickTime Streaming Server
  - server M: Mac OS, Unix SDP, Full RTP/RTP, server
- Apple QuickTime 4
  - client M: Mac OS, Windows SDP, QuickTime also supports SDP files that describe multiscr.
  - with standard devices.
- Cern Wrtpp
  - server N: Linux, SDP
- Columbia University rtspd
  - server N: Solaris SDP, container files
- Entera Light Weight Streaming Application
  - server W: Windows, Unix SDP
- IBM RTSP Toolkit
  - server W: Solaris, SDP
- Kom TU-Darmstadt KOM-Player
  - player L: Linux, AIX SDP, MPEG-1 System, alternative RTSP server for IBM Video Charger
- Kompella C
  - server W: Windows, SDP
- Kwetschbaumer and Koenig srs
  - server W: SDP
- Mac OS, Unix
  - Server W: Unix, SDP
- Mbone, Neutral Media
  - RTCP
- NCSA
  - Open source: record and play
- NCSA
  - Open source, record and play
- Oracle Corporation Oracle Video Server
  - client, server W: Windows, Unix SDP, MPEG-1 System, ATM, QAM, IP
- Real Networks Real Proxy
  - server W: Windows, Unix SDP, ATM, QAM, IP
- Real Networks RealServer G2
  - server W: Windows, Unix SDP, SMIL, supports RTP for RTSP-based clients
- Real Networks RealServer G2
  - server W: Windows, Unix SDP, based on ATM
- Real Systems, RealNetworks
  - server W: Solaris, SDP
- Silicon Valley Internet Media
  - server W: Solaris, SDP
- Sun JMF 2.1
  - client S: Solaris, Windows, Mac OS
- Sun Media Central Server
  - server S: Solaris
- Vovida
  - server L: Open source, record and play
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  - server L: Open source, record and play
Data Encoding

- Data is sampled and quantized
- Pulse Code Modulation (PCM)
  - Audio CD
    - 44100 samples/second, 16 bits/sample: 705.6 Kbps (mono) and 1.411 Mbps (stereo)
  - Differential PCM
    - Encode subsequent samples as diffs from first
- GSM: 13 Kbps,
- G.729: 8 Kbps
- G.723.3: 6.4 & 5.3 Kbps

Data Encoding

- JPEG
  - For still images: uses spatial correlation within image
- MPEG
  - For moving images: uses spatial and temporal correlations
  - Uses a sequence of frames
    - I frame: Reference frames (doesn’t depend on other frames)
    - P frames: Predictor frames (depends on past I or P frames)
    - B frames: Bidirectional frames (interpolation between I and P frames in past and future)
  - In size I > P > B
  - Usually 12 frames from I to I
  - Frequency of P and B based on experience / application and network characteristics
    - MP1 (CD-ROM): 1.5 Mbps
    - MP2 (Hi-quality DVD): 3.6 Mbps
    - MP3: 96 Kbps, 128 Kbps, 160 Kbps