Internet Media-on-Demand: The Real-Time Streaming Protocol

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May 28, 1999
Overview

• Internet media-on-demand
  – why bother – I already have a TV and VCR
  – Internet integrated-services architecture
  – problems

• real-time stream protocol (RTSP) ➜ “Internet VCR”

• session description
Internet multimedia (on demand)

VOD trials not exactly successful... Internet MM different:

- just one service among many ➔ reverse economics from VOD
- re-use existing infrastructure
- flexible media: modem, wireless, cable, LAN, ...
- quality scales from stamp-size flipbook to HDTV – adaptive
- side information easy (closed captioning)
- easy integration with WWW
- easy integration with recording – click-on-page-to-record
- security through encryption
- cheap authoring, service ➔ lots of content

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Internet multimedia

Same infrastructure, different delivery modes:

**on demand:** unicast

**near on-demand:** staggered transmission on multicast

VCR control

**multicast:** niche markets to audience of millions

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Applications

- lectures, seminars
- on-demand instruction
- entertainment: specialty content
- remote digital editing
- voice mail
Internet radio

- 12,140 U.S. AM and FM radio stations, only 100 in Germany
- FM quality (56 kb/s) ➤ backbone capacity of 680 Mb/s
- New York City: 45 FM stations ➤ 2.5 Mb/s
- DirecTV: 31 audio channels ➤ 1.7 Mb/s
- easy time-shifting, content-labeling ➤ near media-on-demand

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Problems

**bandwidth:** 64–128 kb/s for talking heads, 1.5 Mb/s for movies

**quality:** packet loss, predictability

**reliability:** makes CATV look good...

**billing infrastructure:** pay-per-view?

**cheap receivers:** shouldn’t cost more than set-top box
Internet streaming media requirements

- retrieval of media from media server
  - video-on-demand ➜ unicast
  - near video-on-demand ➜ time-staggered multicast
- live events (Mbone-style) ➜ multicast
- remote digital editing ➜ queued play lists, recording
- remote device control
- integration with conferences
- transport, content, description-neutral

Have some proprietary protocols, need interoperability
Streaming multimedia

RTSP

HTTP

SDP

IP multicast

RTP

RSVP

streaming audio/video

RTSP

SIP

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Internet real-time & multimedia protocols

**resource reservation:** RSVP, YESSIR, ...

**media transport:** RTP

**stream control:** RTSP

**stream description:** SDP, SMIL (W3C), RTSL, ...

Related work: DSM-CC, but much simpler
RTSP features

- “rough” synchronization (fine-grained RTP sender reports)
- virtual presentations = synchronized playback from several servers command timing
- load balancing using redirection at connect, during stream
- supports any session description
- device control camera pan, zoom, tilt
- caching: similar to HTTP, except “cut-through”
RTSP protocol design

- similar design as HTTP (TCP + UDP, HTTP, ...)
- HTTP = “the Internet RPC protocol”
- supports any session description
- control “tracks” (audio, video) and “presentation” (movie)
- remote digital editing
RTSP sessions

TCP connection ≠ RTSP session ➞ session maintained by identifier

- one TCP connection per session ➞ firewalls, bidirectional

- one TCP connection per ≥ 1 command ➞ no server state

- UDP
  - multicast, low latency
  - ➞ “passing around the remote”
  - ➞ limit server connection state (live events!)
RTSP and HTTP: similarities

- protocol format: text, MIME-headers
- request/response = request line + headers + body
- status codes
- security mechanisms
- URL format
- content negotiation
RTSP protocol design

RTSP is not HTTP

- server state needed
- different methods
- server → client
- data carried out-of-band
- avoid HTTP mistakes:
  - relative request paths
  - no extension mechanism
  - 8859.1 coding

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RTSP: HTTP inheritance

• simple servers are easy, Apache for industrial-strength

• re-use HTTP extensions:
  – authentication (basic, digest, …)
  – PICS = content labeling
  – JEPI = electronic payments
  – PEP = protocol extensions

• SSL for security
RTSP operation

client

HTTP GET

session description

setup

play

RTP audio

RTP video

RTCP

pause

teardown

media server

web server

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RTSP functionality

**retrieval**: media-on-demand for continuous media
- first, get presentation description
- unicast
- multicast, client chooses address
- multicast, server chooses address (NVOD)
- independent of stream file format subsets or combinations of files

**conference participant**: “invite” to conference, controlled by several people

**live streaming**: ability to add media

one session = single time axis
Control

**Aggregate control:** one command ➠ control several streams

- content may be in *container file* (QuickTime, .wav, ASF, MPEG systems stream, rtpdump, …)
- on single server

**Per-stream control:** each stream has own command

- across container files
- several servers
RTSP URLs

whole presentation:
rtsp://media.example.com:554/twister

track within presentation:
rtsp://media.example.com:554/twister/audiotrack

but: name hierarchy ≠ media hierarchy ≠ file system
RTSP: Web integration

1. web page with “program guide”

2. contains pointer to presentation description (say, SMIL):

   <session>
   <group>
     <track src="rtsp://audio.mtv.com/movie">
     <track src="rtsp://video.mtv.com/movie">
   </group>
   </session>

3. RTSP sets up and controls delivery

4. RSVP reserves resources

5. RTP delivers data

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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

commands may be pipelined
RTSP time

- normal play time (NPT): seconds, microseconds
- SMPTE timestamps (seconds, frames)
- absolute time (for live events)

allow absolute timing of events: ➤ “start playing movie at 10:05.34, at NPT = 10 s” ➤ synchronize distributed servers

- DSM-CC: single pending command
- RTSP: edit list (play 10-12, play 15-20, . . . ) ➤ editing
Request headers

Accept: media description formats
Accept-Encoding: encoding of media format
Accept-Language: human language
Authorization: basic and digest authentication
Bandwidth: client bandwidth available
Conference: conference identifier
From: name of requestor
If-Modified-Since: conditional retrieval
Range: time range to play
Referer: how did we get here?
Scale: (play time)/(real time)
Speed: speed-up delivery
User-Agent: software
## Response headers

<table>
<thead>
<tr>
<th>Header</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Location</td>
<td>redirection</td>
</tr>
<tr>
<td>Proxy-Authenticate</td>
<td>authenticate to proxy</td>
</tr>
<tr>
<td>Public</td>
<td>methods supported</td>
</tr>
<tr>
<td>Retry-After</td>
<td>busy; come back later</td>
</tr>
<tr>
<td>Server</td>
<td>server software</td>
</tr>
<tr>
<td>Vary</td>
<td>cache tag</td>
</tr>
<tr>
<td>WWW-Authenticate</td>
<td>request authorization</td>
</tr>
</tbody>
</table>
**RTSP reliability**

- if TCP, send request once
- if UDP, retransmit with RTT (estimate: 500 ms)
- CSeq for request sequence
- Timestamp for RTT estimation
- atomicity: may pack requests into PDU
- kludge: data interleaving for TCP
RTSP descriptions

contains streams + initialization information [+ network info]:

- RTSP DESCRIBE
- http, email, ...
- command line
- updated via ANNOUNCE; both C-to-S and S-to-C

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Unicast session: get description

C→W: GET /twister.sdp HTTP/1.1
    Host: www.example.com
    Accept: application/sdp

W→C: HTTP/1.0 200 OK
    Content-Type: application/sdp

v=0
o=-- 2890844526 2890842807 IN IP4 192.16.24.202
s=RTSP Session
m=audio 0 RTP/AVP 0
a=control:rtsp://audio.com/twister/audio.en
m=video 0 RTP/AVP 31
a=control:rtsp://video.com/twister/video
Unicast session: open streams

C→A: SETUP rtsp://audio.com/twister/audio.en RTSP/1.0
   CSeq: 1
   Transport: RTP/AVP/UDP;unicast
            ;client_port=3056-3057

A→C: RTSP/1.0 200 OK
   CSeq: 1
   Session: 12345678
   Transport: RTP/AVP/UDP;unicast
             ;client_port=3056-3057;
             ;server_port=5000-5001

C→V: SETUP rtsp://video.com/twister/video RTSP/1.0
   CSeq: 1
   Transport: RTP/AVP/UDP;unicast
            ;client_port=3058-3059
V->C: RTSP/1.0 200 OK
CSeq: 1
Session: 23456789
Transport: RTP/AVP/UDP;unicast
;client_port=3058-3059
;server_port=5002-5003
Unicast session: play

C->V: PLAY rtsp://video.com/twister/video RTSP/1.0
    CSeq: 2
    Session: 23456789
    Range: smpte=0:10:00-

V->C: RTSP/1.0 200 OK
    CSeq: 2
    Session: 23456789
    Range: smpte=0:10:00-0:20:00
    RTP-Info: url=rtsp://video.com/twister/video
              ;seq=12312232;rtptime=78712811

C->A: PLAY rtsp://audio.com/twister/audio.en RTSP/1.0
    CSeq: 2
    Session: 12345678
    Range: smpte=0:10:00-
RTSP session teardown

C->A: TEARDOWN rtsp://audio.com/twister/audio.en RTSP/1.0
   CSeq: 3
   Session: 12345678

A->C: RTSP/1.0 200 OK
   CSeq: 3

C->V: TEARDOWN rtsp://video.com/twister/video RTSP/1.0
   CSeq: 3
   Session: 23456789

V->C: RTSP/1.0 200 OK
   CSeq: 3
PLAY and PAUSE

- several ranges (≥ 1 PLAY) are queued
- PAUSE intercepts first matching time point
- PLAY parameters:
  - Scale: NPT speed ↑
  - Speed: delivery bandwidth ↓
  - Transport: for near-video-on-demand
- mute vs. pause
- implementation: calendar queue
REDIRECT

- server tells client: go elsewhere
- Location header contains URL
- load balancing
- needs to do TEARDOWN and SETUP

S->C: REDIRECT rtsp://example.com/fizzle/foo RTSP/1.0
    CSeq: 732
    Location: rtsp://bigserver.com:8001
    Range: clock=19960213T143205Z-
RECORD

- may use URL or create own ⬤ return new URL in Location

C->S: RECORD rtsp://example.com/meeting/audio.en RTSP/1.0
    CSeq: 954
    Session: 12345678
    Conference: 128.16.64.19/32492374

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Interaction with RTP

- **PLAY** response announces RTP timestamp and sequence number
- allow discarding of packets before break

RTP-Info: url=rtsp://foo.com/bar.avi/streamid=0;seq=45102, url=rtsp://foo.com/bar.avi/streamid=1;seq=30211
Near video-on-demand

- in wide area, video-on-demand not scalable
- near on-demand, with positioning, pause
- popular content delivered every 5 minutes
- RTSP PLAY $t \rightarrow$ join appropriate multicast group for $t$
- easy in Internet: IP multicast groups $\Rightarrow$ no network signaling
- may be able to “catch up” with group
RTSP caching

- proxy caching of *content*, not RTSP responses
- except: DESCRIBE

- parameters similar to HTTP:
  - no-cache: don’t cache
  - public: anybody may cache
  - private: only end-user may cache
  - no-transform: conversion disallowed
  - only-if-cached: only if proxy has content
  - max-stale: except beyond expiration date
  - min-fresh: shelf life left
  - must-revalidate: ask first, proxy later
RTSP extensions

- add headers, methods

- **Require** header for must-understand extensions:

  ```
  Require: org.ietf.rtsp.foobar
  501 Not implemented
  ```
SIP and RTSP integration

- provide transport parameters to RTSP explicitly
- H.323 needs introductions conference identifier
RTSP status

- IETF MMUSIC working group ➔ RFC 2326
- active contributors: Columbia University, Netscape, RealNetworks; IBM, INRIA, Microsoft, …
- implementations in progress:
  - Columbia University (NT, Unix)
  - IBM
  - Lucent
  - Netscape
  - RealNetworks (G2)
- may use existing Mbone tools
RTSP implementation

Example: Columbia rtspd

- share parser with SIP (Internet telephony) server
- basic UDP and TCP (per connection) threads: listen for RTSP requests, assign to session
- thread that picks up timed PLAY and PAUSE request
- thread that cycles through multimedia file
- RTP packetizer

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Summary

- Internet multimedia-on-demand ➔ integrated services Internet
- building block for virtual reality systems
- conferencing ↔ telephony ➔ same tools, formats, network
- WebTV as VOD, Internet telephony terminal?
- digital TV: specialized protocols ➔ IP over the air
- Columbia MarconiNet for TV/radio network architecture
- 18 GB disk ➔ download movie at night?