Audio/Video Transport Working Group

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Wednesday Agenda

- Introduction and status 5
- RTP spec and profile updates 10
  - » Conformance Tests for RTP Scalability 15
  - » MIME Registration of Payload Types 10
- Drafts to go to last call 30
  - » QCELP; Guidelines; RTP MIB; FEC
- New payload formats 25
  - » MP3 audio; DV video; Interleaving
- RTCP SDES location reports 10

Thursday Agenda

• MPEG4 payload format	15
• RTP multiplexing discussion	30
• Generic payload format	5
• Reprise: RTCP for large groups	?

RTP Drafts in Process

- RFCs recently published:
  - » 2508 - IP/UDP/RTP header compression
- No drafts awaiting publication
- Some ready for WG last call
- Several new drafts this meeting
- Some that didn’t make the deadline:
  - » Transport of DTMF & MF tones
  - » RTP implementation checklist

Status of RTP

- RFC1889, 1890 published as Proposed Standards in January 1996
- Internet-Draft revisions for Draft Std.
  - » Spec is <u>draft-ietf-avt-rtp-new-03.ps,txt</u>
  - » Profile is <u>draft-ietf-avt-profile-new-05.ps,txt</u>
- Spec and Profile drafts now complete!
- Ready for WG Last Call for Draft Std?

Recent Changes to RTP Spec

- Clarified that payload type may change during session
- Jonathan Rosenberg carefully reviewed Sections 6.2 and 6.3 containing major changes from RFC 1889
  - Several minor corrections
  - Removed requirement to retain state for inactive participants for 30 minutes -- can cause overestimate, isn't needed with reconsideration
- Also tested & fixed code in Appendix A.7

Companion Drafts for RTP Spec

- SSRC Sampling to be Experimental
  - » draft-ietf-avt-rtpsample-02.txt
- New scalability conformance test draft to be Informational
  - » draft-ietf-avt-rtcptest-00.txt

Recent Changes to RTP Profile

- Completed use of MUST, SHOULD, MAY
  - » Rules for marker bit are SHOULD (video is new)
- Allow override of default 5% RTCP bandwidth
  - » Need separate draft to specify SDP BW modifiers for explicit RTCP sender and receiver BW
- New "Changes from RFC 1890" section and security considerations section
- Added GSM-HR, GSM-EFR, QCELP, BT656, H263-1998 and BMPEG

More Changes to RTP Profile

- No explicit changes for generic formats; specify in payload formats & SDP extensions
- Refers to separate draft for MIME registration
  - » draft-hoschka-rtp-mime-00.txt
    by Philipp Hoschka
  - » Need to publish both drafts together

SDP BW modifiers for RTCP

- Session bandwidth: b=AS:<kb/s>
- Sender RTCP bw: b=RS:<kb/s>
- Receiver RTCP bw: b=RR:<kb/s>
- Example:

  b=AS:100

  b=RS:1.25 <=== Can we use fractions?

  b=RR:3.75 (SDP spec says no!)

MIME Registration

- Defines procedure for registration:
  - » Gives template for new type names
  - » For any existing types that match, just add RTP-specific “encoding considerations”
  - » Info required: reference payload format spec, define parameters as needed
- Registers all the payload names from RTP A/V Profile using a table

MIME Registration Issues

- It’s a “rough draft” needing completion
- Merge draft-alvestrand-audio-l16-01.txt
  - » Pick up “channels” parameter
  - » Conflict for “sample-rate” vs “rate” param.
- Will define audio+video types as video
- Declare no conflict: audio/basic is 8kHz, PCMU is variable
- What to do with vnd.wave and vnd.avi?

MIME and SDP

- MIME major type in **m=** (audio, video)
- Encoding (subtype) in **a=rtpmap**
- Fixed (possibly optional) parameters "rate" and "channels" also in **a=rtpmap**
- Encoding-specific parameters in **a=fmtp** as "type=value"

Drafts ready for Last Call

- PureVoice (QCELP) payload format

  draft-mckay-qcelp-02.txt

  » Issues raised during WG Last Call have been addressed: encryption removed
- Guidelines for RTP payload formats

  draft-ietf-avt-rtp-format-guidelines-01.txt,.ps
- RTP MIB - Ready for last call?

Multiplexing Questions

- Should AVT standardize RTP muxing?
- If yes, more than one scheme?
- Which one(s)?

Strawman Proposal

- AVT standardizes one scheme: GeRM
  - » attractive for MPEG-4
- Use Tmux (RFC1692) for reduced processing (and less compression)
- Applications for which GeRM is not satisfactory may specify their own multiplexing schemes, but these are not standardized by AVT

RTCP for large groups

- Some methods we've discussed:
  - » Sampling of receivers to respond
  - » Summarization/aggregation (router/agent)
  - » Unicast to source which forwards mcast
- Define these methods as new profiles
- Profile specified in SDP as:

  m=audio 1234 RTP/XXX 121 0 5