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 draft-ietf-avt-rtp-new-03.ps.txt
  - Profile is draft-ietf-avt-profile-new-05.ps.txt
- 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: \( b=\text{AS:<} \text{kb/s}> \)
- Sender RTCP bw: \( b=\text{RS:<} \text{kb/s}> \)
- Receiver RTCP bw: \( b=\text{RR:<} \text{kb/s}> \)
- Example:
  - \( b=\text{AS:100} \)
  - \( b=\text{RS:1.25} \)  \( \text{<<< Can we use fractions?} \)
  - \( b=\text{RR:3.75} \)  \( \text{(SDP spec says no!)} \)
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