Audio/Video Transport
Working Group

45th IETF, Oslo
14-15 July 1999

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Wednesday Agenda

● Introduction and status  10
● RTP spec and profile updates  20
  » MIME Registration, SDP bw modifiers
● RTP interop statement & testing  40
● Drafts to go to last call  20
  » RTP MIB; DTMF tones; FEC; (MP3)
● MPEG-4 payload format  30
● DV video; 12-bit audio for DV  20
Thursday Agenda

- Multicast feedback / new Profiles 25
  » Rosenberg, Casner

- RTP header compression / muxing 50
  » Degermark, Pazhynannur, Koren, El-Khatib

- New payload formats
  » AAC audio 10
  » Pointers 10
  » Text for H.323 10
  » Virtual worlds 15
RTP Drafts in Process

- Drafts awaiting publication:
  - QCELP payload format

- Drafts in WG last call:
  - RTP MIB – Comments & revision
  - SSRC Sampling
  - Payload format guidelines – new Security

- A dozen+ new drafts this meeting!
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-04.ps.txt
  » Profile is draft-ietf-avt-profile-new-06.ps.txt

● Companion drafts now ready

● Ready for WG Last Call
  » Pending a couple of questions...
Companion Drafts for RTP Spec

- RTP interoperability statement
  » draft-ietf-avt-rtp-interop-00.txt
- RTP testing strategies – Informational
  » draft-ietf-avt-rtptest-00.txt
- RTCP scalability conformance – Inform.
  » draft-ietf-avt-rtcptest-01.txt
- SSRC sampling – Experimental
  » draft-ietf-avt-rtpsample-04.txt
Recent Changes to RTP Spec

- Example protocols given in definition of “non-RTP means”
- Clarified that Network Time Protocol is not required
- Clarified that implementation MAY choose a different policy than the example algorithm in keeping packets when a collision occurs
- Explained that third-party monitor may receive data and not send RTCP
- A few minor tweaks to code in A.7 based on feedback from Ross Finlayson
Questions on RTP Spec

- SSRC collision algorithm and mobile phones: SHOULD follow new address?
  » Replace pseudo-code with pseudo-C?
- RTP says ports MUST be distinct for layered codings; SDP says this is illegal if addresses differ
- Should avg_rtcp_size be changed to exclude received RTCP packets?
Recent Changes to RTP Profile

- Reference to MIME registration draft reworded to make it non-normative
- Fixed numbers in wrong columns of Table 1
- Removed G726-16/24/40, G727, SX* because no packetization has been defined
- Clarifications for packetization of G722, VDVI
- Didn’t add tables for GSM-HR, GSM-EFR

> Louise Spergel, TIPHON Vice-Chair, says TS 101 318 is a referenceable standard
Questions on RTP Profile

- Should G722 clock rate equal sampling frequency of 16000, or sample-pair rate of 8000 as it is RFC 1890 and H.225.0?
- Should CN payload type be changed from 19 to 13 (as it is in IMTC VoIP Forum spec)? [13 used to be VSC and is reserved]

Need interoperability checklist for profile, too
MIME Registration

- New draft-ietf-avt-rtp-mime-00.txt
- 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 to payload format spec, parameter definitions as needed
- Registers all the payload names from RTP A/V Profile
MIME and SDP

Draft specifies MIME to SDP mapping:

- MIME major type in \texttt{m=} (audio, video)
- Encoding (subtype) in \texttt{a=rtpmap}
- Fixed (possibly optional) parameters “rate” and “channels” also in \texttt{a=rtpmap}, “ptime” in \texttt{a=ptime}
- Encoding-specific parameters in \texttt{a=fmtp} with default format “param=value”
MIME Registration Issues

- Merged in template from RFC 2586 for audio/L16 encoding
- Defines audio+video types as video
- Declare no conflict: audio/basic is 8kHz, PCMU is variable
- What to do with audio/mpeg vs MPA?
- What to do with vnd.qcelp vs QCELP?
- What to do with vnd.wave and vnd.avi?
New draft-ietf-avt-rtcp-bw-00.txt

Specifies two new SDP bw modifiers:
  » Sender RTCP bw: b=RS:<b/s>
  » Receiver RTCP bw: b=RR:<b/s>

→ Deviates from SDP spec to use b/s

Specifies precedence:
  » RS or RR at media level
  » RS or RR at session level
  » Default per session bw at media level
  » Default per session bw at session level
Other Drafts – Not On Our Radar

- L. McCarthy, “RTP Profile for Source Authentication and Non-Repudiation of Audio and Video Conferences,” May, 1999
  » draft-mccarthy-smug-rtp-profile-src-auth-00.txt

  » draft-ietf-avt-X11-new-00.txt
Motivations for New Profile

- Reduce the amount of (S,G) multicast routing state induced by many receivers sending RTCP
- Avoid distributing RTCP identity and feedback information for privacy or competitive reasons
- Proponents of EXPRESS single-source multicast model need this
Unicast RTCP to Source/Monitor

- Must avoid implosion
  - Tell receivers how often to send RTCP
- Don’t enable packet bombing
  - Don’t allow receivers to be told to send RTCP to an innocent bystander host

All-multicast model uses distributed control to preclude sabotage by single entity: bad guy can only slow the rate
Initial Idea

- Unicast to source which forwards to multicast group; receivers run the current distributed interval calculation
  - Reduced multicast routing state
  - Full RTCP functionality retained
  - Could aggregate/summarize RTCP info
    - RTCP transit time to receivers is increased
    - Slightly more load on source & its local link
Authentication is Mandatory

- If RTCP is sent to unauthenticated address, packet bombing is possible

- Can be simple for reflected RTCP:
  - Authenticate SDP announcement
  - Include source’s address in SDP
  - Send RTCP to source address only when RTP packets are received from that source
    - Need to remove this constraint if you want to allow a third-party monitor or multiple monitors
Variations for Privacy

- Filter SDES and reception report blocks, just send empty RR to provide SSRC
- Receivers ignore BYE which could be spoofed
  » Then BYE reconsideration doesn’t work
Second Variation

- Source doesn’t reflect RTCP, just sends a new RTCP packet type with number of receivers
  - Receivers limit rate at which this number is allowed to decrease so spoofer can’t send small number and cause implosion
  - Second limit for BYE rate to replace BYE reconsideration for simultaneous leaves
Issues

- What if more than one simultaneous source or alternative sources?
- How does this interact with adjustable RTCP bw, esp. for low-speed links?
- This idea is quite young - there are bound to be other issues when we think harder
Should we do this?

● Deployment of enhanced multicast routing will take a year or more
● So will development and deployment of this profile?
● Would this profile add value in the long run?
● Is this too easy to get wrong?