Audio/Video Transport Working Group

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Wednesday Agenda

● Introduction and status	10
● RTP spec and profile updates » MIME Registration, SDP bw modifiers	20
● RTP interop statement & testing	40
● Drafts to go to last call » RTP MIB; DTMF tones; FEC; (MP3)	20
● 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 **m=** (audio, video)
- Encoding (subtype) in **a=rtpmap**
- Fixed (possibly optional) parameters "rate" and "channels" also in **a=rtpmap**,
  "ptime" in **a=ptime**
- Encoding-specific parameters in **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?

SDP BW modifiers for RTCP

- 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
- L. Gannoun, "RTP Payload Format for X Protocol Media Streams," March, 1998
  - » 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?