Audio/Video Transport Core Maintenance (AVTCore) Working Group meeting notes

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Monday, 3 March 2014

AVTCore Status Update - chairs

The slides are in https://tools.ietf.org/agenda/89/slides/slides-89-avtcore-0.pdf .

No issues.

Support of multiple RTP streams in the following two drafts: draft-ietf-avtcore-rtp-multi-stream-03, and draft-ietf-avtcore-rtp-multi-stream-optimisation-03 were presented by Magnus Westerlund

The slides are in https://tools.ietf.org/agenda/89/slides/slides-89-avtcore-1.pdf .

On multi streams draft.

Issue #1: The concern we have is that if an endpoint adds a lot of SSRCs in a short time-interval this creates a burst of initial RTCP compound packets.

Action: We should try and find a proposal to address this case that is a reasonable tradeoff. We need to sort out what are the rules to provide a non-zero delay.

Issue #2: In Section 6.2.2 it says that a future version of this memo will include examples of how to choose RTCP parameters for common scenarios.

Action: There is a need for examples (Jonathan's comment)

Issue #3: The Scheduling algorithm as describe above hasn't been tested with dynamic changes.

Colin plans to simulate this but be nice to see results of tests in an independent implementation.

Issue #4: Compatibility issues with AVG_RTCP_SIZE. The current proposal in the Scheduling includes an update to the AVG_RTCP_SIZE as total size / number of SSRCs where numbers of SSRCs are the ones that include SR or RR. This affects average transmission interval for non-updated RTCP senders.

Action: only likely to occur in circumstances with multiple streams and legacy receivers - just document issue and advise against aggregation rather than trying to signal.

Issue #5: Optimizations for Feedback Messages (AVPF).

Action: Implement the first proposal (in the slides) to simply cache feedback messages to allow piggybacking if other SSRC sends compound packet before the feedback is stale.

Next steps:

Varun volunteered to review, need more reviewers.

On multi stream optimization draft:

Bo Volunteered to reviews

RTP Congestion Control: Circuit Breakers for Unicast Sessions in draft-ietf-avtcore-rtp-circuit-breakers-05 presented by Colin Perkins.

The slides are in https://tools.ietf.org/agenda/89/slides/slides-89-avtcore-2.pdf .

Open issues based on Magnus review:

Media timeout circuit breaker triggers if RTP sent, but RTCP SR/RR show no packets received; likelihood of triggering higher when few RTP packets sent per RTCP interval.

Action: set threshold to don’t trigger if sending less than 3 packets per reporting interval. Colin will review and propose something in next rev of draft.

When using RTP/AVPF, do we need to give advice for triggering interval when using T_rr_interval.

Bernard: If you have a layered codec do you apply on a layer basis or to the whole thing.

Colin: treat congestion across the whole thing

Conclusion: Colin agrees with Magnus something is needed and will ask Magnus for text.

Bo volunteered to review the document.

SRTP EKT in http://tools.ietf.org/html/draft-ietf-avtcore-srtp-ekt-02 presented by Dan Wing

The slides are in https://tools.ietf.org/agenda/89/slides/slides-89-avtcore-3.pdf

The changes include sending EKT over SRTP and not over SRTCP.

Using Simulcast in RTP Sessions in https://tools.ietf.org/id/draft-westerlund-avtcore-rtp-simulcast-03.txt presented by Bo Burman.

The slides are in https://tools.ietf.org/agenda/89/slides/slides-89-avtcore-4.pdf

There are IPR disclosed for this document.

Presented RTP level simulcast features. SDP signaling was presented in MMUSIC.

In order to allow MANEs to process simulcast on the RTP layer need RTP level identifiers like "role" , participant, media source and quality level.

What is an appropriate RTP level Media Source identification in a Simulcast context?

CSRC , SRCNAME, appId, amended with semantics and end-to-end scope, msid, amended with RTP level information and end-to-end scope.

From the discussion it seems that none of the above is adequate as defined to provide description in the RTP layer.

There was also a comment about the need for such complexity of simulcast (have SVC).

Continue discussion offline.