Major action points:

Start WGLC on frame markings and cc-feedback.

The notes from the etherpad – thanks James Gruessing

Note Well, Note Takers, Agenda Bashing
No comments

Status of working group drafts:

With IESG:

draft-ietf-avtcore-multiplex-guidelines-09

Roni: Speaking as one of the authors of the document, we missed a comment from Bernard and will probably add some clarification next week

draft-ietf-payload-tsvcis-04
========
* There is still an open discuss
Barry: The status is that the authors are waiting for the AD to respond, correct?
Roni: I will have to look
Barry: I will check the status

draft-ietf-payload-rtp-ttml-06
========
* Has been approved by the IESG and has been sent to the RFC Editor

WG documents:

draft-ietf-payload-rtp-jpegxs-02
* Comments on the list and requires more review

draft-ietf-payload-tetra-03

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Roni: This is ready for WGLC

draft-ietf-payload-vp9-07

Jonathan: I did a minor refresh, but awaiting on some informative text

Jonathan: I'll just go ahead and publish it and stop it being a draft forever

ACTION: Jonathan to report if ready for WGLC

draft-ietf-avtext-framemarking-10
========
Mo: Quite a few comments, one of the majors items was the bottom four bits were all zeros in the short format, making them not possible to reserve for future use
Jonathan: If you do need to change some bits, you'll need some signalling anyway
Mo: A solution would be to have different URN
Mo: I change the document so that the B bit is 0 when the temporal layer is 0
Mo: Clarified that TID = 0 and LID = 0 where TL0PICIDX
Mo: Layer ID mappings further clarified
Mo: Editorial changes to the extension mappings
Mo: I think we should go into WGLC
Bernard: The draft is simple to implement, which modes of AV1 is it applicable to?
Mo: Promoters of AV1 aren't buying into this specification, and where you have nested temporal hierarchies you could use this
Mo: There is no AV1 spec in IETF presently
Bernard: Frame marking is in WebRTC and when it has AV1 this should be addressed
Mo: This mode question comes up for VP9 as well
Bernard: It would be helpful to give examples of frame marking
Mo: I'd be adverse to enumerating things that are newish, especially if they don't see deployment
Roni: the frame marking header extensions for new RTP payload should be in a section in those RTP payload specifications see https://tools.ietf.org/html/draft-ietf-avtext-framemarking-10#section-3.2.1.5

ACTION: Start WGLC beginning of December

RTCP Feedback:
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Colin: Three changes: Mention REMB, update references and author contact details, and relation to Holmer RMCAT draft
Colin: I believe this draft is ready for WGLC announce also in RMCAT
Mo: I think this document will be helpful for this document to be read for new protocol designers, particularly for the people working on the QUIC recovery draft
Colin: As a FYI
Jonathan: QUIC ACK packets don't include timestamps (they were removed)
Zahed: Is this WG willing to take in changes from other working groups?

Colin: This draft says nothing about congestion control
Colin: I suggest we WGLC here and RMCAT

ACTION: Start WGLC beginning of December, copy RMACT about the WGLC

draft-zhao-avtcore-rtp-vvc-00
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Bernard: MSRT was never added to WebRTC, so the jury is not out, it's back

Jonathan: Does anyone have any need for the multi-stream

Mo: I think we can safely omit MRMT, but unclear on the complexities of MRST

Bernard: We implemented MRST for H.264, and unaware of HEVC

Stefan: If they are doing it with HEVC they aren't compliant with RFC 7798, and this was underspecified so we could ship it out the door

Jonathan: VP9 made a decision not to include any of the multi-stream modes

Are we leaning towards a frame marking solution

Bernardo: I think the answer to having a generic payload header extension mechanism is "yes"

Stefan: need an RTP header extension that will be in the RTP payload specification

draft-hellstrom-mmusic-multi-party-rtt-00

Roni: I will consult with the Area Director on which working group will progress with this work

Jonathan: Where to do it depends on what solution we do, which makes it sort of a chicken and egg problem

Gunnar: It has topics for a number of different areas

Barry: I think the avtcore and mmusic chairs should discuss, if neither think it's the right place we should take it to dispatch

Magnus: Gunnar, do you think the multi-stream multi-part solution is clear enough in the specifications for this?

Magnus: If I take the RTP model for text, each user has their own SSRC, the stream is forwarded to all receivers and the question is how do you display this? And from that, how do you deal with legacy?

Gunnar: That would be the ideal solution, with one SSRC per sending user in a stream, so you can detect when text was loss, but I don't want to complicate implementations as multiple SSRCs appear not supported in the same session in SDP

Barry: Magnus do you have a thought on where this work belongs?

Magnus: This is tough, some of the solutions to these legacy problems are where you are codifying the internal of the text formats, and the rest is signalling, I think a large part of it is in mmusic

Gunnar: For mmusic it's capability negotiation, do we go SDP attribute or SIP media attribute

Harald: Just a word of warning, this is generic multi-party chat service. We have a long and glorious history of global failures, starting from IRC, through to XMPP and vary other efforts. You will have all the problems of multi-party chats, and you might want to take a step back or shut down the scope of use to pursue a point solution.

Roni: You should figure out what would be simple to implement and to start with that.

???: In all of these deployments, are they only just text or are they sometimes mixing with other media? If it's purely text, RTP and SDP are a giant mistake.

Gunnar: Most common deployments involve audio and real time text

Roni: The document mentions video and closed captions

Barry: I don't want you to forget mls

Bernard: This is part of next-gen 911 architecture, right?

Jonathan: We should ensure rum is in the loop for this

Jonathan: The RUE is possibly being used, and one of the use cases is from a RUE to a PSAP

Gunnar: In that situation, it's because the PSAP wants to conference the next party, which is a simple three-party call

Jonathan: I guess my point is what solution spaces will be easy for that community depends on what we do

Jonathan: It does seem like people feel like it's in scope for mmusic and not here

Roni: Try to focus on what solution will progress it faster

Jonathan: Both from standardisation and from implementation without too much pain

Summary:
The author should select the mode that will be simpler to implement and propose it as a solution.
The document should be discussed in MMUSIC? AVTCore chairs will update the MMUSIC chairs

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