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WebRTC: Media Transport and Use of RTP

draft-ietf-rtcweb-rtp-usage-15

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Changes in -15

- Reflects comments from interim meeting:
  - Change draft-ietf-avtcore-rtp-multi-stream-optimisation to “MAY support, MUST signal before use”
  - Reference draft-ietf-avtcore-rtp-multi-stream to justify recommendation of T_rr_interval = 4 seconds when using RTP/SAVPF profile
  - Clarify implementations MUST signal extensions before use (but need to be robust to non-signalled extensions)
  - Add discussion of how to mitigate potential denial of service attacks due to malicious configuration of RTCP parameters

Open Issue: FEC (1/3)

- The draft currently states:
  - "There are several block-based FEC schemes that are designed for use with RTP independent of the chosen RTP payload format. At the time of this writing there is no consensus on which, if any, of these FEC schemes is appropriate for use in the WebRTC context. Accordingly, this memo makes no recommendation on the choice of block-based FEC for WebRTC use."
- Some have expressed desire to add FEC support, but no suggestions for what scheme to use

Open Issue: FEC (2/3)

- Possible FEC schemes:
  - RFC 2733 – parity FEC; requires FEC use same SSRC as original media, so doesn’t work with bundle; abuses RTP header fields for FEC, so can’t be used with mixers
  - RFC 5109 – parity FEC with uneven level protection (ULP); unnecessary complexity due to ULP; requires FEC use same SSRC as original media, so doesn’t work with bundle
  - SMPTE 2022-1 – extends RFC 2733 for 2d interleaved parity FEC; sets CC, CSRC, and timestamp to zero, so doesn’t work with mixers; requires SSRC=0 and PT=96, so can’t support >1 FEC stream per RTP session
  - RFC 6015 – interleaved parity FEC; abuses RTP header fields for FEC, so can’t be used with mixers; works with bundle
  - RFC 6682 – Raptor FEC; works with bundle
  - RFC 6865 – Reed-Solomon FEC; not well integrated with RTP
- IPR declarations exist

Open Issue: FEC (3/3)

- Options going forward:
  1. Leave the draft unchanged, with no FEC recommended in this version
  2. MAY use RFC 6015 interleaved parity FEC, with guidance explaining why it's problematic with mixers
  3. MAY use RFC 6682

- Proposal: no FEC in this version; revisit in a later version of the spec

Open Issue: DTX

- There is no mention of DTX for audio in the draft; suggested to mandate support for RFC 3389
  - The rtp-usage draft doesn't mandate any codecs or RTP payload formats
  - Proposal: if wanted, this should be done in draft-ietf-rtcweb-audio

Open Issue: multiple CNAMEs

- Potential conflict between Sections 4.9 and 11:

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Section 4.9 says:

  Taking the discussion in Section 11 into account, a WebRTC end-point
  MUST NOT use more than one RTCP CNAME in the RTP sessions belonging
  to single RTCPeerConnection (that is, an RTCPeerConnection forms a
  synchronisation context).

and then Section 11 says:

  ... This is motivating the
  strong recommendation in Section 4.9 to only use a single CNAME.

     The requirement on using the same CNAME for all SSRCs that
     originate from the same end-point, does not require a middlebox
     that forwards traffic from multiple end-points to only use a
     single CNAME.
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  - Had assumed “end-point” excludes middleboxes – might need to clarify in draft-ietf-avtext-rtp-grouping-taxonomy
  - Middleboxes can use multiple CNAME values
  - End-points use a single CNAME – no strong preference on SHOULD vs. MUST here; what is the sense of the room?

Next Steps

- Are there any other open issues?

- Working group last call, and send to IESG