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WebRTC: Media Transport and Use of RTP

draft-ietf-rtcweb-rtp-usage-05

Colin Perkins – University of Glasgow
Magnus Westerlund – Ericsson
Jörg Ott – Aalto University

Changes in -05

- Use RFC 2119 terminology by reference, don’t copy the definitions
- Note that the RTP/SAVPF profile with the updated list of recommended codecs is mandated, not the standard RTP/SAVPF profile.
- Clarify that the use of non-compound RTCP packets MUST be negotiated on the signalling channel before use, and that implementations are REQUIRED to support compound RTCP feedback packets if the remote endpoint does not agree to use non-compound RTCP packets
- Remove the reference to RFC 6222 and instead reference RFC 6222-bis
- Update references to RFC 5117 to point to the RTP Topologies update draft
- Clarify that a WebRTC sender is REQUIRED to understand and react to FIR messages it receives, but that sending FIR messages is OPTIONAL
- Rewrite Section 7 on rate control and media adaptation for clarity. Merge the previous Sections 7.1 and 7.2 into a single new section, and try to better explain the relationship between the RTP circuit breakers, the signalled SDP bandwidth limitations, and any RTP/AVPF TMMBR messages
- Add Section 13 on Open Issues
- Revise and expand Section 15 on Security Considerations

Open Issues: Congestion Control

- RTP congestion control algorithms will probably require some feedback information to be conveyed in RTCP. Are the tools that are mandated by this memo sufficient, or do we need additional information?
- RTP congestion control could be implemented using either a sender-based algorithm or a receiver-based algorithm. For interoperability, does this draft need to mandate which end is in charge of congestion control for a path?

Open Issues: RTCP XR

- Still open if any RTCP XR performance metrics are needed, as discussed in Section 8.

Open Issues: Number of SSRCs

- Are any requirements needed to agree and limit the number of simultaneously used SSRCs within an RTP session?
- Is an API needed for expressing any application level media mixing of an RTP media stream so that the correct CSRC list can be set?

Open Issues: Simulcast

- Anything needed in this draft to support simulcast?
  [Magnus]

Open Issues: QoS

- Possible documentation of what support for differentiated treatment that are needed on RTP level as the API and the network level specification matures?