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WebRTC: Media Transport and use of RTP

draft-ietf-rtcweb-rtp-usage-07

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

Changes in -07

- Attempt to address comments received:
  - Section 4.1: expand discussion of multiple SSRCs in single RTP session
  - Section 4.3: expand discussion of payload type assignment; explain how to associate media streams with SDP m= lines using payload types
  - Section 4.3: note mandatory to implement codecs are specified elsewhere; any codec that can be negotiated in SDP can be used
  - Section 4.4: clarify single RTP session is used for all RTP media streams
  - Section 4.8: clarify MUST accept RTP/RTCP using non-signalled SSRCs; explain how a=ssrc: lines can associate media streams with SDP m= lines
  - Section 5.2.1: clarify rapid sync extensions in addition to RTCP SR sync
  - Section 7.3: discuss interoperability between sender- and receiver-driven congestion control; not solvable until algorithms from RMCAT known
  - Section 8: reflect the discussion about RTCP XR use at IETF 86
  - Assorted editorial fixes throughout

Next Steps

- Sections 11, 12, and Appendix A not yet updated
  - Comments welcome, but these sections will be re-written in next version
- Other sections are believed essentially complete
  - **Please review and send comments to the list – the draft needs detailed review from implementors**