WebRTC Media Transport and Use of RTP

draft-ietf-rtcweb-rtp-usage-12

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Changes in -11

- Incorporates results of discussion at IETF 88:
  - Receivers MUST support multiple synchronisation contexts (CNAMEs) per sender in a single RTP session; senders MAY use multiple CNAMEs
  - Update WebRTC API mapping, based on the current working draft API from the W3C
    - Proposals in W3C to remove the MediaStream concept from that API; if that occurs, this draft will have to be updated to match
  - Update text on forwarding MediaStreamTracks from one PeerConnection to another
    - Endpoint doing forwarding acts as back-to-back receiver and sender
    - Need to be able to handle outgoing stream as a MediaSource, i.e., be capable of transcoding the media to suit outgoing requirements, including bit-rate adaptation
  - Assorted editorial fixes and minor cleanups
Changes in -12

- Addresses remaining open issues:
  - Made use of SLI and RPSI feedback messages RECOMMENDED rather than OPTIONAL, based on list discussion
  - Defer to draft-ietf-rtcweb-security-arch for definitions of mandatory cipher suites, DTLS-SRTP protection profiles, key management, etc.
  - Clarify that the RTP packet stream association with MediaStreamTracks is the one that MSID provides
  - Add a WebRTC API requirement regarding CSRC list handling
  - Remove reference to the shim-based approach to running multiple RTP sessions on a single transport-layer flow
  - Remove discussion of simulcast
    - Leaves some comments in Section 12.1 regarding simulcast with existing functionality
  - Remove section 5.2.4 on Associating RTP Media Streams and Signalling Contexts, should be part of the WebRTC signalling specification
  - Assorted editorial fixes
Open Issues and Next Steps

• Open issue:
  • Metadata carried in RTP header extensions can be sensitive
  • Already RECOMMENDED that client-to-mixer and mixer-to-client audio level information sent in header extension is encrypted using RFC6094
  • Expand this recommendation to all RTP header extensions?

• Otherwise, authors have no open issues – review and/or working group last call solicited