RTCWEB Terminology

A Discussion of relation between RTCWEB Media Protocol Terminology and the PeerConnection API

RTP Related Terminology

- Multi-Media Session
  - A communication session between two or more entities
  - Can contain one or more RTP session
  - Can be represented by the signalling context
  - Example representations are in a SIP Dialog or RTSP Session
- RTP Session
  - One RTP SSRC space shared between 2 or more entities sending zero or more media streams
  - From a single end-points perspective, usually represented by:
    - A port(s) to receive RTP and RTCP
    - One or more destinations to send RTP and RTCP to.
  - Commonly single media type (RTCWEB multiplexing will have multiple types)
  - Media for a particular purpose and usage in an application context.

RTP Related Terminology

- SSRC
  - Sender Source identifier (also used by receiver only entities to ID its reports)
  - Identifies a single media source (e.g. camera, microphone, audio mix)
  - Intended to be unique within an RTP session
  - Multi-channel audio is commonly sent as a single SSRC using a media format capable either packetizing multiple channels or encoding multiple channels as one bit stream.
- CNAME
  - Canonical Name
  - Identifies a synchronization context
  - Unique within a multi-media session
  - CName is applied to the SSRCs in one or more RTP sessions that a receiver may synchronize
- Payload Type (PT)
  - An Identifier representing the encoding and packetization of the media present in the RTP packet body

WEBRTC API Terminology

- WebRTC 1.0: Real-time Communication Between Browsers
  - Based on W3C editors draft dated: 23 Aug 2011:
  - http://dev.w3.org/2011/webrtc/editor/webrtc.html
- MediaStream object
  - Contains zero or more tracks
  - A MediaStream object be forked to create a child that is equal or a subset of the parent object
  - MediaStream objects can share tracks so one object contains the same tracks, a sub or super set of tracks of another MediaStream object
  - Each MediaStream has a Label
- MediaStreamTrack
  - A single media source, multi-channel audio is a single track
  - Each track can originate from:
    - Media device, such as video camera or microphone
    - File which plays back in real-time
    - Received over network

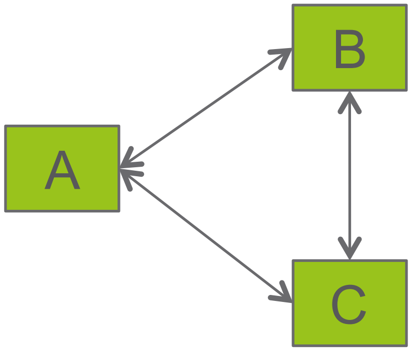
WEBRTC API Terminology

- Label
  - A Label used to identify a MediaStream when delivered to the other peer
  - Note: Child objects inherent their parent's Label
- PeerConnection
  - A communication association between two peers
  - Media negotiation signalling is done on PeerConnection level
  - Is configured with STUN and TURN server resources
  - Contains ICE, Media Transport, etc.

Discussion: MediaStream and Label

- A MediaStreamTrack can be mapped to a SSRC in an RTP session
- A MediaStreamTrack has a synchronization context, that could be represented by the CNAME
  - CNAME needs to be the same by all MediaStreamTracks that are part of the same context, e.g. captured in the same room
- A MediaStream sent by a PeerConnection can be represented by a list of RTP session:SSRC tuples
- The MediaStream **label** has no matching construct
  - The SDP a=label attribute labels RTP sessions, not a set of SSRCs in possibly several RTP sessions
  - Needs to be exchanged between peers
  - The MediaStream Label can't be CNAME:
    - The same track can be part of multiple MediaStreams
  - Needs to be specified!

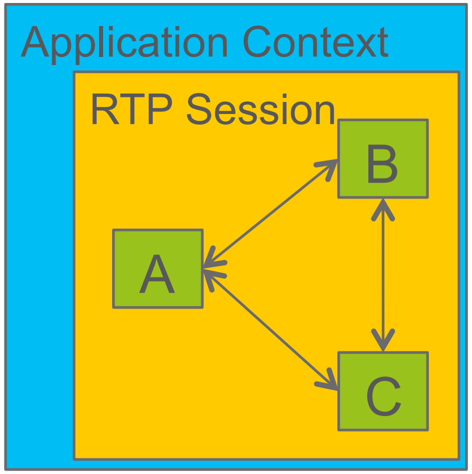
Discussion: RTP sessions

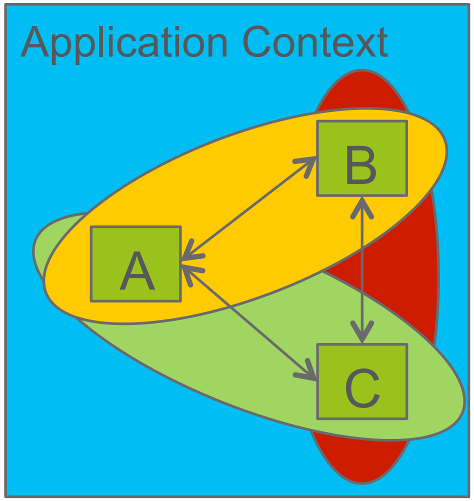


- Lets look at the small non-centralized conference:
  - Uses Mesh between A, B and C.
- Current PeerConnection definition results in one Peer Connection between each pair of peers, i.e AB, AC and BC
- The WebApp in A can bind the same MediaStream object to both PeerConnections (AB and AC)
  - Thus delivering the same media sources to both peers
  - The exact media configuration for each PeerConnection may be desirable to vary
    - Different amount of screen estate at peers
    - Application logic’s usage of streams
    - Capability negotiation between peers may result in different codecs etc.

Discussion: RTP sessions

- Two possible RTP Session structures:
  - The first is one RTP session over all PeerConnections
    - Implies same media streams to all participants
    - Allows for RTCP information for legs a peer is not directly involved in.
    - No Use case requiring this structure currently
  - The second alternative is to use individual RTP session(s) for each PeerConnection
    - Allows different rates and codecs in each PeerConnection
    - Adaptation modules needs to combine information across multiple PeerConnections and RTP sessions
- The application context may need common information across the peers
  - Identities of streams
  - Synchronization contexts





Discussion: Multi-Party RTP Sessions

- In the centralized conference usage a common RTP session needs to be supported
  - The RTP tools for centralized conferencing are built around a common session
  - The RTP session still exist in the context of only one PeerConnection from the end-point's (A, B, C or D) view
  - A Mixer has a different view, but is after all a more advanced RTP entity