RTP Multiple Stream Sessions and Simulcast

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IPR Disclosure

 › Telefonaktiebolaget LM Ericsson (publ)'s has made a Statement about IPR related to this draft in https://datatracker.ietf.org/ipr/1592/
Outline

› Problem Descriptions
  – A Target Scenario
  – Simulcast
  – Multiple Streams in Advanced RTP usages
  – Bandwidth Signaling
  – Codec Control / Optimization

› Problem Summary

› Extensions
  – Multiple Streams signaling
  – Bandwidth SDP attribute
  – Simulcast Grouping
  – SRCNAME SDES item
  – Codec Control

› Way Forward
  – Architecture document
  – Extensions
A Target Scenario

Active speaker

RTP Mixer

Listener

Listener

Listener
Simulcast

When using scalable coding with inter-layer prediction, the broad arrows will be slightly smaller, but all red arrows will also have to be sent.

Simulcast: sending multiple representations of the same source.
Simulcast and Scalable Encoding

» Simulcast is both an alternative and complementary to Scalable Encoding

» The trade-offs when it comes to efficiency are different
  – SVC encoding is more efficient in sender to mixer path
  – Simulcast is more efficient in mixer to receiver path
  – Combining scalable encoding with simulcast for best of both worlds

» Simulcast is codec agnostic

» Simulcast can be done for other purposes
  – Provide two different encodings for interoperability
  – Provide redundancy for robustness
Multiple Streams

A sample client, both sending and receiving multiple video streams
Multi-Stream Layering

- An RTP Session can contain 1..N SSRCs
- An RTP Session is identified by a lower layer identifier, such as a UDP port or five tuples
- A multimedia session contains one or more RTP sessions
Multi-Stream Simulcast Layering

A single source can be simulcasted as different versions
- Same actual source in several sessions;
  *don’t want to force new semantics into SSRC value*

Several sources can be rendered at the same device
Multi-Stream Issues

› More advanced use cases than point to point VoIP:
  – Video conferencing
  – Telepresence
  – IPTV
  – Etc.

› This can result in multiple media streams
  – Is the end-point capable of handling multiple simultaneous media streams of the same media type?
    › Legacy capabilities is likely one SSRC per direction
  – When should additional media streams be in the same RTP session, when in a new session?
  – When streams have relations, how to express that for:
    › Retransmission
    › Redundancy
    › Simulcast
Bandwidth Signaling

› Current SDP Bandwidth signaling insufficient in handling:
   – Asymmetric bandwidth capabilities in the path
   – Asymmetric bandwidth usage inherent from application
   – When different Payload Types have different bandwidth ranges
   – When multi-stream applications use multiple streams in each direction
   – The allowed burstiness of media sources is not explicit
Codec Control / Optimization

RTP Mixer

Active speaker

Listener

Simulcast

Red, sent streams are not consumed by anyone (e.g. based and resumed request)

NOTE! Just as valid for scalable coding!
Problem Summary

- We have a general RTP architecture clarity issue
  - We need to clarify multiple SSRCs in one RTP session
  - We need to discuss when appropriate to use multiple RTP sessions
  - We need to create common principles for streams that aren’t independent, but have common source.

- We need to do this for all reasonable topologies

- Multiple SSRCs in an RTP session appear to need signaling support to avoid legacy issues
- Simulcast is good tool, we need signaling and RTP association mechanisms to make it work
- Bandwidth configuration and capability declaration in asymmetric usages and encodings needs to be improved
- Scalable Codecs and Simulcast needs additional Codec Control tools to optimize sessions
Proposed Extensions (1/4)

› Multiple Streams Signaling

  – Separated directions
    › a=max-send-ssrc:96 2
    › a=max-recv-ssrc:96 5

  – Both payload specific and payload agnostic
    › a=max-recv-ssrc:98 6
    › a=max-recv-ssrc:99 4
    › a=max-recv-ssrc:* 8
Proposed Extensions (2/4)

› Bandwidth Signaling

– b= line not possible to extend with sufficient new semantics
– Per direction and payload type (also payload agnostic)

– Per source
  › a=bw:recv pt=96 SMT:tb=64000:320
  › a=bw:recv pt=97 SMT:tb=12200:128

– Entire media level aggregate
  › a=bw:send pt=* AMT:tb=384000:512

– Allow for future needed semantics to be defined
Proposed Extensions (3/4)

- Simulcast Grouping in SDP
  - Different semantics between directions
  - `a=group:SCS 1 2 3 …` (SimulCast Send intention)
  - `a=group:SCR 4 5 …` (SimulCast Receive capability / acknowledge)

- Simulcast Source Identification in RTP
  - SDES CNAME is defined as unique per endpoint, not per source
  - *New* SDES SRCNAME unique per actual media source
    - Indicate which streams are alternative encodings to each other
Proposed Extensions (4/4)

› Codec Control extensions is forthcoming
Anticipated Document Split

Current

- RTP Session and Multi-stream Architecture
- Multi-stream Signalling
- Simulcast Grouping
- Bandwidth
- Codec Control
Proposal for Going Forward

› That AVTCORE takes on the general Architecture questions:
  – Make it clear when appropriate to use multiple streams within an RTP session
  – How should one use RTP sessions and SSRCs when having alternative, complementary or redundant streams

› That the various extensions are submitted to the appropriate WG as individual pieces for progressing:
  – AVTCORE:
    › Architecture
    › Multi-stream Signaling
  – AVTEXT:
    › Simulcast Group Signaling
    › Codec Control
  – MMUSIC:
    › Bandwidth Signaling