RTP Topology Considerations for Offer/Answer-Initiated Sessions

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Offer/Answer in one Slide

- SDP Offer/Answer has two modes
  - Multicast
    - Single view of session
    - If transport address is multicast
    - Not used very much
  - Unicast
    - Two separate views of session
    - Mostly, each side specifies what it wants to receive
      - (Oversimplification)
RTP Topologies (RFC 5117)

• RTP Sessions come in many topology types
• Topo-Multicast corresponds to Offer/Answer Multicast mode
• Topo-Point-to-Point corresponds most directly to Offer/Answer Unicast mode
• Other RFC 5117 topologies can also be assembled from Offer/Answer Unicast ... 
• ... except for Topo-Transport-Translator!
Topo-Multicast

• Network forwards packets among all participants.
• Single view of session.
• SDP Offer/Answer Multicast mode.
Middlebox forwards packets among all participants.

Single view of session.
Other RTP Topologies

- Topo-Media-Translator, Topo-Mixer, Topo-Video-Switch-Mixer, and Topo-RTCP-Terminating-MCU.
- Equivalent for this analysis.
- Middlebox modifies and forwards media and RTCP.
Why Offer/Answer doesn’t work for Topo-Transport-Translator

• Topo-Transport-Translator assumes a single view of the session.
  – A middlebox that just forwards all RTP and RTCP, without modifications.
  – At session layer, looks almost identical to Topo-Multicast.
  – Everyone shares the same session configuration.

• But as mentioned: Unicast Offer/Answer allows each endpoint to establish its own view of the session.
In particular (1): bandwidth

- An SDP offerer or answerer can specify a bandwidth (b= parameter) for any media stream (RTP session), indicating the bandwidth it wants the peer to use to send.
- This is independent of the peer’s specified bandwidth.
- RTCP timings (and thus RTP membership timeouts) are a function of SDP b= values (AS, RS, RR...).
In particular (2): media types

- Participants can remove media types from the answer or updated offer.
- Participants can change (some) fmtp parameters.
- Though NOT RECOMMENDED, they can change the mapping between media types and RTP payload type numbers.
  - And there are sometimes good reasons to do this.
And others...

- Other reasons too, such as ptime or many SDP extensions.
- Don’t need to get into them now.
But this is a good thing!

• If you know that your media and RTCP is always received by only one other party …
  – Whether it’s consuming it directly, or forwarding it after re-writing

• Some very useful optimizations are possible
Offer/Answer Asymmetric Bandwidth

- Asymmetric link, e.g. ADSL:
  - Bandwidths in each direction are very different, and non-rival at the bottlenecks.
  - What RTCP timeout should you use?
  - B sending RTCP less often because A has multiple senders makes no sense.
  - If you can assume a single receiver: each party uses 5% (or whatever) of its own bandwidth for RTCP, calculates remote timeouts accordingly.
RTCP receiver reports

• For the benefit of Topo-Multicast and Topo-Transport-Translator, in RTCP, every session member sends receiver reports about every sender.

• This means (in a many-to-many session) the number of reception reports is quadratic in the number of session members.

• If RTCP is going directly to only one other party, this is useless.
  – A middlebox is likely doing its own retransmission and repair, and thus directly consuming a large fraction of the RTCP.
  – Excess reports consume RTCP bandwidth which could be used for timely feedback of relevant data.
RTCP receiver reports (2)

• If you have multiple co-generated sources (e.g. CLUE, Bundle, RTX)
  – Having them send reports about each other is pointless: zero loss, jitter.
  – Having them send all send redundant reports about remote sources isn’t helpful.

• Reception reports from remote source A about another remote source B are rarely interesting or useful.
Normative recommendations

• For RTP sessions negotiated with unicast SDP offer/answer:
  – RTCP bandwidths, and timeouts, MUST be calculated independently in each direction.
  – Endpoints SHOULD NOT send reports from one of their own sources about another.
  – Endpoints SHOULD pick a single “reporting” source to send reception reports for each remote source.
Way forward

• What do people think of this?
  – I think this is important for BUNDLE, CLUE, some WebRTC models, and anything else doing source multiplexing...
  – The bandwidth issues apply even for single-source cases.

• Would this need to be a normative update to 3550? To 3264? Or just implementation guidance?

• Should this actually be more than one document, addressing different issues?