Multiplexing RTP Sessions

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RTP Requirements for Multiplexing

• Different types of media (e.g., audio and video) are carried in different RTP sessions

• RTP requires an explicit lower-layer protocol to demultiplex different RTP sessions
  • This is typically done by running different RTP sessions on different transport-layer flows
  • Running several RTP sessions on a one transport-layer flow, without an explicit demultiplexing layer, causes problems (details are in the draft – not time to discuss here, but happy to talk in detail offline)
  • You may argue that the breakage is not important for the RTCWeb use cases, but some part of RTP do break, and some environments rely on those parts
Why is a Multiplexing Solution Desired?

• Perception that the standard RTP model is:
  • Slow – requires one additional RTT + ICE pacing interval in the typical case, to setup the additional flow (longer in the worst case)
  • Unreliable – the more rounds of ICE processing, the more likely that failures may occur (is there data on how often?)
  • Complex – needs additional ICE exchanges and more SDP (so what?)
  • Wasteful of port space – there are limits on the ability of NATs to support multiple port mappings (but, browsers open many concurrent TCP ports)

• Are these issues real problems? Why?
How to Multiplex?

• If we accept that multiplexing is needed, what is the best way to add it?

  • Define a new, explicit, multiplexing layer in RTCWeb
    • This could be as simple as a one-byte shim
  • Use an existing multiplexing layer
    • E.g., the UDP encapsulation of DCCP
  • Send requirements to AVTCORE for multiplexing RTP sessions
    • Require support for standard RTP semantics now, and adopt results of AVTCORE discussion as an extension when ready
    • [Reminder: the RTCWeb charter prohibits this group modifying RTP, and the AD has reiterated this on the list recently]
  • Give up on RTP, and use something else entirely, to avoid the problem…
    • But, what?
    • This might complicate legacy interop with RTP devices a little
Proposal

- RTCWeb needs to support the standard RTP model of one transport-layer flow per RTP session
  - For audio/visual conferences, this means two RTP sessions, and hence two UDP flows
  - The only solution for which the standards exist
  - Required for legacy interoperability

- We recognise that there are reasons to optimise this to reduce the number of UDP flows – request that AVTCORE consider developing multiplexing solutions