Multiple RTP Flows in a Single Media Line

draft-ivov-rtcweb-noplan

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How Many Streams Can We Fit Here:

```
v=0
o=carol 24872 24872 IN IP4 100.3.6.6
s=-
t=0 0
c=IN IP4 192.0.2.4
a=group:BUNDLE audio video

m=audio 5000 RTP/SAVPF 96 0 8
a=mid:audio
a=rtpmap:96 opus/48000/2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000

m=video 5002 RTP/SAVPF 97 98
a=mid:video
a=rtpmap:97 VP8/90000
a=rtpmap:98 H264/90000
```

Multiple Streams. Single M Line

- Advantages:
  - It just works
  - No Offer/Answer when adding/removing streams
  - No added glare risk
  - No need to pre-announce SSRCs
    (more possible topologies)
  - Allows apps to choose fine-tuned signalling:
    Custom, XCON, RFC4575, WebRTC JS, CLUE channels
  - Does not prohibit Plan A or Plan B

- Need to work out:
  - Discovery of multi-stream capabilities
  - Optionally:
    - define new mechanisms for layering, FEC, SIM, RTX

How do I know if adding a 2nd stream is OK?

- max-send-ssrc / max-recv-ssrc
  - not sure how this relates to layering
- leave it to upper-layer signalling

How do we demux when using BUNDLE?

- payload-types
  (we have 96 but even 32 should be enough for the conservative)

- What if we run out of payload types?
  use two bundles (or no bundle at all)

- How do I know that SSRC “1111” needs to go to the left screen (and codec chain) and “2222” to the right one?
  use upper-layer signalling