RTP Payload Format for Opus Speech and Audio Codec
`draft-ietf-payload-rtp-opus`

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Opus Overview

- Audio codec RFC 6716

- Covers most audio applications

    - Bitrates from 6 to 510 kb/s

    - Narrowband (8 kHz) to fullband (48 kHz)

    - Frame size from 2.5 ms to 60 ms (packets up to 120 ms)

    - Mono and stereo support

- Configuration changeable dynamically

- Selected as MTI for WebRTC

Payload Design Goals

- No negotiation failure
- Keep all the flexibility
- Keep ability to switch dynamically
- Not targeting surround

RTP Payload

- Simple: 1 RTP payload = 1 Opus packet
  - May contain multiple frames, up to 120 ms
- Timestamp based on 48 kHz clock
- Decodable with no OOB signalling
  - Configuration signaled in-band
- RTP padding NOT RECOMMENDED
  - Opus-level padding is better

SDP

- Mapping is always opus/48000/2
  - Even when the audio is narrowband mono
- All fmtp parameters are informative
  - Designed to optimize quality
  - May be ignored without affecting interoperability

SDP Parameters (fmtp)

- maxplaybackrate: "Running speakers at this rate"

- stereo: "I only have one speaker"

- sprop-maxcapturerate: "Running mic at this rate"

- sprop-stereo: "I only have one mic"

- cbr: "Please send CBR"

- useinbandfec: "I can make use of Opus FEC"

- usedtx: "Please don't transmit silence"

- Sender can choose to ignore any of these (e.g. to avoid transcoding)

Outstanding Issues

- Constraining DTX duration?
- Setting the marker bit

Questions?