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 (fmtcp)

- 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?