Opus Audio Codec

Research Group: Telecommunications Software and Systems Group (TSSG)

Address: Waterford Institute of Technology, West Campus, Carriganore, Waterford, Ireland

Authors: Brendan O'Farrell

Email: bofarrell@tssg.org

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Description:

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1. Opus interactive audio Codec

The Opus codec is designed for interactive speech and audio transmission over the Internet. It is designed by the IETF Codec Working Group and incorporates technology from Skype's SILK codec and Xiph.Org's CELT codec.

1.1. Technology

The Opus codec is designed to handle a wide range of interactive audio applications, including Voice over IP, videoconferencing, in-game chat, and even remote live music performances. It can scale from low bit-rate narrowband speech to very high quality stereo music.Opus-codec.org [http://www.opus-codec.org/]

Features:

- Bit-rates from 6 kb/s to 510 kb/s
- Sampling rates from 8 to 48 kHz
- Frame sizes from 2.5 ms to 60 ms
- Support for both constant bit-rate (CBR) and variable bit-rate (VBR)
- Audio bandwidth from narrowband to full-band
- Support for speech and music
- Support for mono and stereo
- Support for up to 255 channels (multistream frames)
- Dynamically adjustable bitrate, audio bandwidth, and frame size
- Good loss robustness and packet loss concealment (PLC)
- Floating point and fixed-point implementation

For the full specification see latest Internet Draft [http://tools.ietf.org/html/draft-ietf-codec-opus-16].