Technology Milestones

Voice Quality on Push-to-Talk Broadband Networks

Vocoder Strategy Overview

When connecting broadband to broadband, audio quality must be preserved as it passes between the distributed components of the Broadband Interoperability Platform (BIOP). Catalyst tested a prototype that leverages the Catalyst interworking product which, like other products in the critical communication industry, was originally designed to provide interoperability between radio channels. For bandwidth efficiency, the audio streamed between distributed Catalyst gateway components uses voice codecs (vocoders) that are bandwidth efficient and optimized for "radio quality" speech. For operations that involve LMR systems, LMR to LMR interoperability and even broadband to LMR interworking, these vocoders generally introduce no audible limitations since the fidelity of the audio on at least one leg of the audio path is already "radio quality".

As part of our development of a Broadband Interoperability Platform, Catalyst improved the fidelity of the audio that is sent between gateway components that make up the distributed BIOP by using **wideband vocoders**. We have chosen two vocoders as wideband options to our low bandwidth usage, narrow-band, "radio quality" codecs. We spent considerable time researching vocoders, examining tradeoffs between different vocoder technologies. The two vocoders that were prototyped were chosen based on different but complementary criteria.

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~~AMR-WB~~ (Adaptive Multi-rate Wideband)AMR-WB

We use the phrase “Full High Definition (HD) Vocoding” for sending audio between diverse systems. Unfortunately, this phrase blends “Full HD” with “HD Vocoding” and this blending does not yield a precise technical description. “Full HD” in the audio realm is usually associated with music, high bandwidth usage, and sampling frequences of at least 44.1 KHz. “HD Vocoding” is associated with speech and not music and has much more modest bandwidth specifications.

Our research calls out AMR (technically AMR-WB[i]) as the primary vocoder to be used to connect BIOP Push-to-Talk over Broadband (PTToB) systems. The reasoning behind this early selection was that AMR-WB is the codec required by 3GPP for Mission Critical Push-to-Talk (MCPTT). Being able to preserve AMR-WB between interoperable MCPTT compliant systems without transcoding would theoretically provide the best performance and audio quality for those configurations. Using AMR-WB when connecting proprietary systems or even for interworked systems that include LMR is a less clear-cut choice.

AMR-WB is a mature, (now considered legacy) vocoder whose patent expired in 2024[ii]. Prior to this, licensing fees and royalties were required to utilize this codec. The vocoder is generally described as a lossy wide-band vocoder for handling high-definition voice. It provides improved speech quality compared to narrowband voice coders (like G711 µ-law) which provide lower “toll quality” speech associated with traditional narrow-band telephony. AMR-WB provides a speech bandwidth of 50–7000 Hz while narrowband speech codecs are designed to support only 300–3400 Hz.

We have prototyped using the AMR-WB vocoder and used it to connect two prototype MCPTT gateway talkgroups together. All comparisons thus far have been subjective listening tests. Our listeners confirm that the AMR-WB-based connection provides audibly superior quality compared to our previous G711 µ-law-based connection.

Opus - Open-Source Codec

Opus is an audio coding format developed by the Xiph.Org Foundation and standardized by the Internet Engineering Task Force. It is designed to efficiently code speech as well as general audio in a single format and is low latency enough for real-time communication and low complexity enough for low-end processors. The open format is standardized through RFC 6716[iii]. It’s widely used as a voice over IP (VoIP) codec in applications such WhatsApp[iv] and the PlayStation 4[v].

Opus is a totally open, royalty-free, highly versatile audio codec[vi] that compares very favorably against similar codec technologies[vii]. AMR-WB is a voice codec, is limited to a 16KHz sampling rate and to a maximum data rate of 24 Kbps. So, while AMR-WB works very well at these maximums, it (like Opus) is lossy. While AMR-WB might be the ideal for connecting MCPTT-compliant systems to eliminate transcoding, there are many proprietary PTToB systems that do not use AMR-WB.

For non-MCPTT-compliant systems, however~~therefore~~, Opus is capable of providing a more versatile, higher quality and non-voice optimized codec. Opus can sample up to 48 kHz, supports bit-rates up to 510 Kbps, and is capable of supporting general audio and not just voice. For voice, it can also support not only wideband but also super wideband which provides a speech bandwidth of 50–14,000 Hz (compared to wideband's 50-7,000 Hz bandwidth).

We have prototyped using the Opus vocoder and used it to connect two prototype MCPTT gateway talkgroups together. Although AMR-WB cannot support a bandwidth greater than 24 Kbps, we can run Opus at the G711 µ-law 64Kbps benchmark bandwidth and evaluate its quality against AMR-WB in its highest quality mode. In this admittedly "unfair" configuration, in subjective audio comparisons, Opus is judged to provide superior audio quality to AMR-WB.

Background Noise Impact on Speech Quality

It is beyond the scope of our research into a ~~Broadband Interoperability~~BIOP Platform to evaluate non-speech audio sources, but Catalyst was a subcontractor to Northrop Grumman for a Mission Critical Voice codec evaluation project in 2007, funded by SAFECOM. In that project, we evaluated multiple narrow-band voice codecs including the highly optimized DVSI ones. Part of that project was to evaluate how these codecs handled speech that had background noise from sources such as helicopters, speed boats, and sirens. The inability of voice-optimized codecs such as DVSI AMBE full-rate to reproduce intelligible voice with these background noises present was surprising, stark, and concerning. Some of these noises even at relatively modest volume levels completely destroyed the codec's ability to reproduce intelligible voice.

For a future project, it would be informative to evaluate how these BIOP-candidate wideband voice codecs handle these types of background noises that are very often present in critical communication situations. Laboratory audio quality testing of codecs that use "clean" speech samples may give a very misleading evaluation of how that codec will perform "in the wild" during real-world critical communication events where the types of background noises cited above are often present.

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[i] VoiceAge, AMR-WB/G.722.2, https://voiceage.com/AMR-WB.G.722.2.html

[ii] VoiceAge Patent Portfolio by Standards, https://voiceage.com/Patent-Portfolio.html

[iii] Opus Interactive Audio Codec, https://opus-codec.org/

[iv] Hazra, Sudip; Mateti, Prabhaker (September 13–16, 2017). "Challenges in Android Forensics". In Thampi, Sabu M.; Pérez, Gregorio Martínez; Westphall, Carlos Becker; Hu, Jiankun; Fan, Chun I.; Mármol, Félix Gómez (eds.). Security in Computing and Communications: 5th International Symposium, SSCC 2017. Springer. pp. 286–299 (290).

[v] "Open Source Software used in PlayStation®4". Sony Interactive Entertainment Inc. https://doc.dl.playstation.net/doc/ps4-oss/

[vi] Opus Comparison – Codec Landscape https://opus-codec.org/comparison/

[vii] Results of the public multiformat listening test @ 64 kbps (March/April 2011) - https://listening-tests.hydrogenaud.io/igorc/results.html