ETSI TS 103 634 V1.1.1 (2019-08)

Digital Enhanced Cordless Telecommunications (DECT); Low Complexity Communication Codec plus (LC3plus)

Reference

DTS/DECT-00338

Keywords

audio, codec, DECT, Full-Band, LC3plus, superwideband, voice, VoIP

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Digital Enhanced Cordless Telecommunications (DECT).

Clause 4 provides an overview of the LC3plus codec, whilst clause 5 provides detailed algorithmic descriptions of the encoder and the decoder.

Clause 6 introduces the bit exact, fixed point ANSI C source code, which is attached to the present document, that provides a reference implementation of the LC3plus audio codec. The conformance procedure for verifying optimized implementations is available in clause 7.

Annex A provides a description of the Application Layer Forward Error Correction function associated with the LC3plus codec.

Modal verbs terminology

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Executive summary

The present document is the specification of the Low Complexity Communication Codecs plus (LC3plus), a transformation-based audio codec operating at all common sampling rates and a wide range of bit rates. The present document includes beside the technical description of the core codec also packet loss concealment and forward error correction schemes such as a channel coder to be ready for use in applications like VoIP and DECT. Besides voice applications, the codec is also applicable for high quality music transmission up to transparency.

Introduction

With the introduction of the 3GPP Enhanced Voice Service (EVS) [i.1] in 2014, the mobile voice communication was enriched with the SWB audio quality. However, this technical development came along with a significant increase in computational complexity and memory demands which limits the deployment to relatively powerful mobile phones. LC3plus aims to provide the low complexity counterpart of EVS in order to make SWB also available on low-cost terminals such as VoIP or DECT. The codec allows perfect interoperability between mobile and other networks by means of transcoding and fits complexity wise very well to the requirements of DECT and VoIP terminal equipment. Due to the codec's flexible design the applications are not limited to voice services but can be extended to high quality music streaming as well.

1 Scope

The present document contains the specification of the Low Complexity Communication Codec plus (LC3plus). The specification includes a full algorithmic description of both the encoder and the decoder. It includes reference fixed-point and floating-point ANSI C source code and conformance test procedures.

The codec has been designed on the one hand for Digital Enhanced Cordless Telecommunications (DECT) and the New Generation DECT (NG-DECT) systems but also for VoIP and other applications such as music streaming.

The LC3plus codec provides the following basic features:

- Capability for speech and audio coding
- Several low delay modes
- Low computational complexity
- Multiple bitrates from 16 kbps up to 320 kbps and more
- Multiple audio bandwidth from narrow band to full-band and ultra-band
- High- resolution mode for high precision, high dynamic range and audio bandwidth up to the Nyquist frequency also for ultra-band
- Advanced error concealment
- Application Layer Forward Error Correction (AL-FEC) including channel coder functionality
- RTP payload format

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at https://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)", Rosenberg, J. and H. Schulzrinne, June 2002.

[2] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications", STD 64, Schulzrinne, H., Casner, S., Frederick, R. and V. Jacobson, July 2003.

[3] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, Schulzrinne, H. and S. Casner, July 2003.

[4] IETF RFC 4566: "SDP: Session Description Protocol", Handley, M., Jacobson, V., and C. Perkins, July 2006.

NOTE: Available at https://www.rfc-editor.org/info/rfc4566.

[5] IETF RFC 4855: "Media Type Registration of RTP Payload Formats", Casner, S., February 2007.

[6] IETF RFC 4867: "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs", Sjoberg, J., Westerlund, M., Lakaniemi, A., April 2007.

[7] European Broadcasting Union: "Sound Quality Assessment Material recordings for subjective tests".

NOTE: Available at https://tech.ebu.ch/docs/tech/tech3253.pdf and https://tech.ebu.ch/docs/testmaterial/SQAM_FLAC.zip.

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] 3GPP TS 26.445 (V16.0.0): "Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description".

[i.2] T. R. Fischer: "A pyramid vector quantizer", IEEE Trans. On Information Theory, July 1986, pp. 568-583.

[i.3] B. Fries and M. Fries: "Digital Audio: Essentials", February 2015, p. 159.

[i.4] Recommendation ITU-T G.191: "Software tools for speech and audio coding standardization", 2009.

[i.5] ETSI TR 103 633: "Low Complexity Communication Codec Plus (LC3plus); Characterization".

[i.6] C. Perkins, O. Hodson, V Hardman: "A Survey of Packet-Loss Recovery Techniques for Streaming Audio", IEEE Network, 1998.

[i.7] 3GPP TR 26.843 (V16.0.0): "Study on non-bit-exact conformance criteria and tools for floating-point EVS codec", August 2018, pp. 18-19.

[i.8] Recommendation ITU-T G.192: "A common parallel digital interface for speech standardization activities", March 1996, p. 14.

[i.9] "SoX", Sound eXchange.

NOTE: Available at https://sourceforge.net/projects/sox/files/sox/14.4.2/sox-14.4.2-win32.zip.

[i.10] "Wine", Wine-HQ.

NOTE: Available at https://www.winehq.org.

[i.11] Recommendation ITU-R BS.1387-1: "Method for objective measurements of perceived audio quality", November 2001.

[i.12] Recommendation ITU-T G.722: "7 kHz audio-coding within 64 kbit/s"', November 1988.

[i.13] Recommendation ITU-T G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)", December 1990.

[i.14] ISO/IEC 14496-3:2009: "Information Technology – Coding of Audio-Visual Objects – Part 3: Audio", March 2009.

3 Definition of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the following terms apply:

codec redundancy: redundancy created by the LC3plus codec

NOTE: The codec encodes two variants of a frame, one primary encoding and one secondary (or redundant) encoding. The primary encoding typically uses more bits than the secondary encoding.

frame: portion of the media for a single channel, e.g. speech or audio or a combination thereof, that is input to the encoder or output of the decoder for one channel

NOTE: A frame includes a frame duration of audio (see frame duration definition below).

frame aggregation: encapsulation of several non-redundant frames within the same packet

frame data: encoded media for a single audio frame and a single channel, either output from the encoder or input to the decoder

NOTE: Frame data may include any of the following: active audio, Silence Description (SID), NO_DATA errframe or Speech_bad frame. The frame data is not protected with the channel coding specified in Annex A.

frame data block: frame data for one or more channels for a single frame period

NOTE: For mono input audio signals, a frame data block includes the frame data for a single audio frame, see frame data. In this case, the frame data block is identical to the frame data. For stereo and multi-channel input audio signals, a frame data block contains the frame data from all channels. Thereby, a frame data block includes the same number of frames as there are channels.

frame duration: time duration for a frame

NOTE: For NB, WB, SSWB, SWB, FB, FBHR or UBHR, the frame duration is either 2,5 ms, 5 ms or 10 ms. For FBCD, the frame duration is either ca 2,72 ms, ca 5,44 ms or ca 10,88 ms.

frame period: time period for a frame, from time T until time T+frame_duration

full-band: speech or audio sampled at 48 kHz

full-band, compact disc: speech or audio sampled at 44,1 kHz

high-resolution mode: LC3plus operation mode for higher bit rates, higher precision and wider audio bandwidth

narrow-band: speech or audio sampled at 8 kHz

NO_DATA frame: type of frame data that spends no bits on encoding the audio

NOTE: A NO_DATA frame is sometimes included when creating the payload and a frame needs to be included but no active frame or SID frame is available, for example when sending redundancy with offset or multi-channel audio where some channels are idle or in DTX.

no request (NO_REQ): type of FDLR that includes no adaptation request

semi-super-wideband: speech or audio sampled at 24 kHz

speech_bad frame: type of frame data indicating that the frame data has been discarded because of errors

NOTE: For example, when a media gateway detects bit errors in the frame data it may discard the frame data and instead send a Speech_bad frame towards the receiver to explicitly indicate that the frame data was dropped.