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Characterization Methodology and Requirement Specifications for the ETSI LC3plus codec

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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1 Scope

The present document specifies the subjective and objective methodologies developed in cooperation between TC STQ and TC DECT for the characterization of the Low Complexity Communication Codec Plus (LC3plus). It describes experimental tests and conditions used for subjective and objective testing. Based on these methodologies the performance requirements for this codec are specified.

The requirements in the present document are specified to characterize a high-quality codec for use in modern telecommunication networks, including but not limited to DECT and VoIP. A special focus is placed on the fact that end-to-end connections are often of hybrid nature concatenating different technologies and thus tandeming (i.e. transcoding) different codecs.

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.

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

[1] Recommendation ITU-T P.800 (08/1996): "Methods for subjective determination of transmission quality".

[2] Recommendation ITU-T P.863 (03/2018): "Perceptual objective listening quality prediction".

[3] Recommendation ITU-T G.722 (09/2012): "7 kHz audio-coding within 64 kbit/s".

[4] Recommendation ITU-T G.726 (12/1990): "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

[5] ETSI TS 103 634: "Digital Enhanced Cordless Telecommunications (DECT); Low Complexity Communication Codec plus (LC3plus)".

[6] Recommendation ITU-T G.191 (01/2019): "Software tools for speech and audio coding standardization".

[7] ETSI TS 126 442: "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point) (3GPP TS 26.442)".

[8] ETSI TS 126 173: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; ANSI-C code for the Adaptive Multi-Rate - Wideband (AMR-WB) speech codec (3GPP TS 26.173)".

[9] ETSI TS 126 073: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; ANSI-C code for the Adaptive Multi Rate (AMR) speech codec (3GPP TS 26.073)".

[10] Recommendation G.711 Appendix I (09/1999): "A high quality low-complexity algorithm for packet loss concealment with G.711".

[11] IETF RFC 8251: "Update to the Opus Audio Codec".

[12] Recommendation ITU-T G.711 (11/1988): "Pulse code modulation (PCM) of voice frequencies".

[13] Recommendation ITU-T G.722 (11/2006) "7 kHz audio-coding within 64 kbit/s; Appendix IV "A low-complexity algorithm for packet loss concealment with G.722".

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.

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

[i.1] ETSI TR 103 590: "Digital Enhanced Cordless Telecommunications (DECT); Study of Super Wideband Codec in DECT for narrowband, wideband and super-wideband audio communication including options of low delay audio connections (<= 10 ms framing)".

[i.2] IETF RFC 6716: "Definition of the Opus Audio Codec".

[i.3] 3GPP, S4-141392: "EVS-7c Processing functions for characterization phase", TSG S4#81.

[i.4] 3GPP, S4-141319: "EVS-8b EVS Permanent Document EVS-8b: Test plans for selection phase including lab task specification", TSG S4#81.

[i.5] 3GPP, S4-141372: "EVS-8c EVS Permanent Document EVS-8c: Test plans for characterization phase including lab task specification", TSG S4#81.

[i.6] IEEE: "A method for comparing the performance of EVS and other voice codecs under bursty packet loss", IPTcomm, 2018.

3 Definition of terms, symbols and abbreviations

3.1 Terms

Void.

3.2 Symbols

Void.

3.3 Abbreviations

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

ACR Absolute Category Rating
AMR-NB Adaptive Multirate speech codec Narrowband
AMR-WB Adaptive Multirate speech codec Wideband
BER Bit Error Rate
CBR Constant Bitrate
CELT Constrained Energy Lapped Transform
CuT Codec under Test
DCR Degradation Category Rating
DECT Digital Enhanced Cordless Telecommunications
DP DECT Profile
EVS codec for Enhanced Voice Services
EVS-WB EVS WideBand

FB	FullBand
FEC	Forward Error Correction
FER	Forward Error Correction RealTime Protocol
FP	Fixed Part
LC3plus	Low Complexity Communication Codec Plus
NB	NarrowBand
PLC	Packet Loss Concealment
PLP	Packet Loss Profile
PLR	Packet Loss Rate
PP	Portable Part
RF	Radio Frequency
RSSI	Received Signal Strength Indicator
RTP	RealTime Protocol
STL	Standard Template Library
SWB	Super WideBand
VoIP	Voice over IP
WB	WideBand

4 Introduction

The present document defines characterization methodologies as well as the performance requirements to be evaluated for the ETSI Low Complexity Communication Codec Plus (LC3plus) [5]. The performance of the codec was initially studied by the TC DECT group in ETSI TR 103 590 [i.1] which is considered as qualification of the codec.

The purpose of the characterization phase experiments is to demonstrate the performance of the codec over a set of conditions and the following use cases:

- Voice services in DECT and VoIP
- Interworking VoIP scenarios between different networks
- Music streaming as additional interesting use case

The characterization utilizes the set of characterization methodologies and configurations of subjective and objective experiments defined in clause 5. The experiments are designed in order to evaluate whether LC3Plus achieves the following codec objectives:

- Introduction of Super-Wideband (SWB) quality in voice services
- Increased capacity of DECT systems when compared to legacy DECT codecs
- Improved robustness for packet loss and bit errors
- Ensure suitable performance in case of transcoding or self-tandeming conditions

All details on the definition of codec objectives for DECT and VoIP and the derived performance requirements and performance objectives are specified in clause 6.

Clause 7 defines the statistical analysis to be conducted on the subjective results to verify that the performance of the Codec under Test (CuT) is sufficient in comparison to the specified performance requirement or performance objectives. In the present document, CuT always means ETSI LC3plus [5].

5 Characterization methodologies

5.1 Overview

The present clause describes the experiment design and the subjective and objective methodologies. The aim of the characterization test is to assess the clean channel performance, self-tandeming capabilities, cross-tandeming, as well as rate switching conditions and variation of the input speech level.

The characterization tests shall be conducted in the same way as the 3GPP EVS selection/characterization process [i.4] and [i.5].

5.2 Experiments

All test conditions shall be separated according to the category audio bandwidth and channel conditions. This results in six experiments, i.e. 3x audio bandwidth times 2x channel conditions. Additionally, one multi-bandwidth experiment shall be conducted in order to provide a quality overview.

Each experiment is evaluated using subjective and objective methodologies described in clauses 5.4 and 5.5.

Table 1 outlines the experiment setup:

Table 1: Experiment overview

Experiment number	Experiment label	Max. bandwidth of input	Channel conditions	Estimated number of conditions
1	NB clean	4 000 Hz	No error	40
2	NB error	4 000 Hz	Bit error & packet loss	32
3	WB clean	8 000 Hz	No error	60
4	WB error	8 000 Hz	Bit error & packet loss	32
5	SWB clean	16 000 Hz	No error	40
6	SWB error	16 000 Hz	Bit error & packet loss	35
7	M fullscale (see note)	20 000 Hz	No error & bit error & packet loss	44
NOTE: The M fullscale experiment contains all bandwidth conditions to span the complete P.800 quality range [1].				

A complete list of all experiments and conditions describing the exact configuration for each condition and the relevant comparison points are contained in archive ts_103624v010101p0.zip which accompanies the present document.

5.3 Item processing

The test items shall be processed according to the EVS processing plan [i.3]. For transcoding, no frame synchronization between the codecs shall be applied. The frequency masks used by 3GPP EVS characterization tests shall be applied to the input signals. The items shall be processed and prepared for the experiments using the STL 2009 [6] tools.

5.4 Subjective methodologies

All subjective experiments shall be conducted using the Recommendation ITU-T P.800 [1] procedure using speech material. Subjects shall be naïve listeners and native speakers. Experiments should be conducted in different languages and labs.

Table 2 shows the P.800 experiment configurations.

Table 2: P.800 experiment configuration

Parameter	Experiment						
	1	2	3	4	5	6	7
Rating scale	ACR	ACR	ACR	ACR	DCR	ACR	ACR
Min. number of listeners	24	24	24	24	24	24	24
Min. num. of talkers	4	4	4	4	4	4	4
Min. num. of samples per talker	6	6	6	6	6	6	6
Min. number of votes per sample	4	4	4	4	4	4	4
Min. number of votes per condition	96	96	96	96	96	96	96
Est. test duration in min. (see note)	47	41	63	41	68	38	45
NOTE: Estimation calculation contained in archive ts_103624v010101p0.zip.							

5.5 Objective methodologies

All experiments listed in Table 1 shall be assessed by the objective quality evaluation using the perceptual objective listening quality prediction tool standardized by ITU-T also known as Recommendation ITU-T P.863 [2].

Tests shall be run in the full band mode with full band reference files and appropriate degraded files.

6 Characterization test plan

6.1 Testing Conventions

6.1.1 Introduction

The following clauses specify performance requirements and conditions to be evaluated for the following use cases:

- DECT with clean channel conditions.
- DECT with error prone channel conditions.
- VoIP without packet loss conditions.
- VoIP including packet loss conditions.

Besides performance requirements, performance objectives are specified. The performance objectives are only foreseen as informative comparison conditions.

6.1.2 Software versions

The following software version for the different codecs shall be used:

- G.711 A-law: Recommendation ITU-T G.711 [12] and G.711 Appendix. I (PLC) [10].
- IETF RFC 8251 [11] OPUS: V1.1.3 (deployed) or V1.3.0 (latest), fix-point.

NOTE: OPUS is a codec in accordance with IETF RFC 6716 [i.2] and IETF RFC 8251 [11].

- EVS: EVS Codec ETSI TS 126 442 [7] V12.7.0 and V13.2.0 or latest one; EVS Codec ETSI TS 126 442 [7]. V12.12.0 and 13.7.0.
- LC3plus: Latest.
- G.722: Recommendation ITU-T G.722 [3] and G.722 Appendix IV [13] or Recommendation ITU-T G.722 [3] + Appendix IV.
- AMR-WB (G.722.2): ETSI TS 126 173 [8] V15.1.0 (latest).
- AMR-NB: ETSI TS 126 073 [9] V15.0.0 (latest).
- G.726: Recommendation ITU-T G.726 [4].

6.1.3 Test condition numbering

The test conditions are numbered according the scheme given in Table 3.

Table 3: Test condition numbering

NB	WB	SWB
DECT with clean channel conditions		
1xx	2xx	3xx
DECT with error prone channel conditions		
4xx	5xx	6xx
VoIP without packet loss conditions		
7xx	8xx	9xx
VoIP including packet loss conditions		
10xx	11xx	12xx

6.2 Characterization test plan for clean channels with application in DECT scenarios

6.2.1 Overview

CuT in DECT shall provide the same or better voice quality than the VoIP network provides and guarantees higher efficiency than DECT audio codecs used today, meaning same quality at lower bit rates to allow better DECT slot exploitation in conjunction with channel coding to provide better protection for bit errors and packet loss concealment.

As network interworking scenarios, the following cases shall be evaluated:

- Voice calls from legacy VoIP to DECT
- Voice calls from DECT to legacy VoIP
- Voice calls from DECT over legacy VoIP to DECT

DECT uses today G.726 (NB) and G.722 (WB). Today's VoIP terminals utilize G.711 (NB) and G.722 (WB).

6.2.2 NB conditions

The test shall verify the performance of the CuT in NB mode. Speech coding for narrowband speech connections using a normal 32 kbit/s payload DECT RF slot shall not be worse than what is achieved by Recommendation ITU G.726 [4]. The CuT shall enable the same range where communication is possible between DECT PP and FP as achieved at the date of publication of the present document by DECT-G.726 connections.

The voice quality by transcoding between VoIP G.711 to/from CuT shall not be worse than connections between VoIP-G711 and DECT-G.726.

Additional performance objectives should be defined in comparison to OPUS (CELT mode, constant bitrate mode (CBR), 32 kbit/s, complexity=0, FEC off, NB mode, 10 ms framing).

The following NB conditions shall be included into the test (Input speech levels to be applied are -16 dBov, -26 dBov, -36 dBov):

100. Direct reference conditions with limited audio bandwidth (cut off frequency of 4 kHz) but no speech coding.

CuT:

101. LC3plus 32 kbit/s, 10 ms framing.

Requirement:

102. G.726, 32kbit/s with G.711 Appendix I PLC.

Performance objective:

103. OPUS, CELT mode, CBR, 32 kbit/s, complexity = 0, FEC off, NB mode, 10 ms framing.