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TECHNICAL SPECIFICATION

Characterization Methodology and Requirement Specifications for the ETSI LC3plus speech codec

Reference
RTS/STQ-301

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ETSI

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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) speech codec. 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.

In addition to its speech capabilities, the LC3plus codec has the option for high quality music streaming. This is out of scope of the present document.

The 2021 revision of the present document adds the test plan for the auditory test, see annex B, the results of the subjective characterization test for the ETSI LC3plus codec, see annex C as well as the related electronic attachment(s).

The 2024 revision of the present document adds the objective test results, see annex D and the Ie derivation as well as a new Ie derivation methodology for fullband conditions, see annex E together with the related electronic attachments.

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] 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 (03/2023): "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 ITU-T 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 Appendix IV (11/2006): "A low-complexity algorithm for packet loss concealment with G.722".

[14] ITU-T Handbook (2011): "Practical procedures for subjective testing". .

[15] Recommendation ITU-T G.192 (03/1996): "A common digital parallel interface for speech standardization activities".

[16] Recommendation ITU-T P.56 (12/2011): "Objective measurement of active speech level".

[17] Recommendation ITU-T P.50 (09/1999): "Artificial voices".

[18] ETSI TS 126 441: "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); General overview (3GPP TS 26.441)".

[19] Recommendation ITU-T G.722.2 (07/03): "Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)".

[20] Recommendation ITU-T P.833 (2001): "Methodology for derivation of equipment impairment factors from subjective listening-only tests".

[21] Recommendation ITU-T P.834 (2015): "Methodology for the derivation of equipment impairment factors from instrumental models".

[22] Recommendation ITU-T G.113 (2001): "Transmission impairments due to speech processing".

[23] Recommendation ITU-T G.107 (2000): "The E-Model, a computational model for use in transmission planning".

[24] Recommendation ITU-T P.833.1 (2009): "Methodology for the derivation of equipment impairment factors from subjective listening-only tests for wideband speech codecs".

[25] Recommendation ITU-T P.834.1 (2015): "Methodology for the derivation of equipment impairment factors from instrumental models for wideband speech codecs".

[26] Recommendation ITU-T G.113 Amendment 2 (2019): "New Appendix V - Provisional planning values for the fullband equipment impairment factor and the fullband packet loss robustness factor".

[27] Recommendation ITU-T G.107.1 (2019): "Wideband E-Model".

[28] Recommendation ITU-T G.107.2 (2023): "Fullband E-Model".

[29] Recommendation ITU-T G.113 Amendment 1 (2009): "Revised Appendix IV - Provisional planning values for the wideband equipment impairment factor and the wideband packet loss robustness factor".

[30] ETSI TS 126 445 (V17.0.0): "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); Detailed algorithmic description (3GPP TS 26.445 version 17.0.0 Release 17)". Codec for Enhanced Voice Services (EVS); Detailed algorithmic description.

[31] Recommendation ITU-T G.729.1 (2006): "G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[32] ETSI TS 146 053 (V16.0.0) (09-2020): "Digital cellular telecommunications system (Phase 2+) (GSM); ANSI-C code for the GSM Enhanced Full Rate (EFR) speech codec (3GPP TS 46.053 version 16.0.0 Release 16)".

[33] ETSI TS 146 006 (V17.0.0): "Digital cellular telecommunications system (Phase 2+) (GSM); Half rate speech; ANSI-C code for the GSM half rate speech codec (3GPP TS 46.006 version 17.0.0 Release 17)".

[34] Recommendation ITU-T G.729 (06/2012): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

[35] Recommendation ITU-T P.1401 (2020): "Methods, metrics and procedures for statistical evaluation, qualification and comparison of objective quality prediction models".

[36] Orange SA: "EID-AMR tool".

[37] McGill University, Telecommunications & Signal Processing Laboratory: "Audio File Programs and Routines".

[38] NTT: "AHEVS-165".

[39] Fraunhofer IIS: "Randomization Tool for EVS".

[40] Xiph.org: "OPUS 1.3".

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

3 Definition of terms, symbols and abbreviations

3.1 Terms

EPFsize size of each packet loss in milliseconds

3.2 Symbols

Void.

3.3 Abbreviations

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

ACR	Absolute Category Rating
AMR	Adaptive MultiRate speech codec
AMR-NB	Adaptive Multirate speech codec Narrow Band
AMR-WB	Adaptive Multirate speech codec Wide Band
ASL	Active Speech Level
BER	Bit Error Rate
BT	Better Than
CBR	Constant Bitrate
CELT	Constrained Energy Lapped Transform
CI95	Confidence Interval at 95 % probability level
CuT	Codec under Test
DCR	Degradation Category Rating
DECT	Digital Enhanced Cordless Telecommunications
DP	DECT Profile (e.g. DP0, DP1, etc.)
EID	Error Insertion Device
EN	English
EP	Error Protection
EPF	Error Pattern File
EPFsize(s)	Error Pattern Frame size(s)
EVS	codec for Enhanced Voice Services
EVS-WB	EVS WideBand
FB MB	Full Band Mixed Band
FB	FullBand
FEC	Forward Error Correction
FER	Frame Error Rate
FP	Fixed Part
LC3plus	Low Complexity Communication Codec Plus
MAN	Mandarin
MB	Mixed Band
MNRU	Modulated Noise Reference Unit
MOS	Mean Opinion Score
MOS-LQS	Mean Opinion Score - Listening Quality (Subjective)
MP3	MPEG Layer 3
MPEG	Motion Picture Experts Group
NB	Narrow Band
NWT	Not Worse Than
PCM	Pulse Code Modulation
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
SPL	Sound Pressure Level
SPL(A)	Sound pressure level, A-weighting
STD	Standard Deviation
STL	Software Tools Library
SWB	Super WideBand
URL	Uniform Resource Locator
VAD	Voice Activity Detection
VoIP	Voice over IP
WAV	WAVeform audio file

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.

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 a similar fashion as the 3GPP EVS selection/characterization process 3GPP S4-141319 [i.4] and 3GPP S4-141372 [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. Such experiment is subdivided in order to allow for a test in different languages.

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
7a	MB fullscale short American English (see note)	20 000 Hz	No error & bit error & packet loss	30
7b	MB fullscale short Mandarin (see note)	20 000 Hz	No error & bit error & packet loss	30
NOTE: The MB 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 subfolder "Conditions for the P.800 experiments" in archive ts_103624v010301p0.zip which accompanies the present document.

NOTE: The experiment labelled "FB M_fullscale" in the attachment is not actually used in the test, since it has been replaced by the experiment labelled "FB M_fullscale_short" which will be conducted in two languages.

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 tools provided in Recommendation ITU-T G.191 [6].

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 Recommendation ITU-T P.800 [1] experiment configurations.

Table 2: P.800 experiment configuration

Parameter	Experiment							
	1	2	3	4	5	6	7a	7b
Rating scale	ACR	ACR	ACR	ACR	DCR	ACR	ACR	ACR
Minimum number of listeners	24	24	24	24	24	24	24	24
Minimum number of talkers	4	4	4	4	4	4	4	4
Minimum number of samples per talker	6	6	6	6	6	6	6	6
Minimum number of votes per sample	4	4	4	4	4	4	4	4
Minimum number of votes per condition	96	96	96	96	96	96	96	96
Estimated test duration in minutes (see note)	47	41	63	41	47	38	29	29
NOTE: Estimation calculation contained in subfolder "Conditions for the P.800 experiments" in archive ts_103624v010301p0.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 versions for the different codecs shall be used:

- G.711 A-law: Recommendations ITU-T G.711 [12] and G.711 Appendix I (PLC) [10].
- IETF RFC 8251 [11] OPUS: 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].
- LC3plus: Latest.
- G.722: Recommendations ITU-T G.722 [3] and G.722 Appendix IV [13].
- AMR-WB (G.722.2): ETSI TS 126 173 [8].
- AMR-NB: ETSI TS 126 073 [9].
- G.726: Recommendation ITU-T G.726 [4].

6.1.3 Test condition numbering

The test conditions are numbered according to 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-T G.726 [4]. The CuT shall enable the same range where communication is possible between DECT PP and FP as achieved by DECT-G.726 connections of the previous technology.

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, 32 kbit/s with G.711 Appendix I PLC [10].

Performance objective:

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

The following transcoding scenarios shall be tested:

CuT:

104. G.711 -> LC3plus (32 kbit/s).
105. LC3plus (32 kbit/s) -> G.711.
106. LC3plus (32 kbit/s) -> G.711 -> LC3plus (32 kbit/s).

Requirement:

107. G.711 -> G.726 (32 kbit/s).
108. G.726 (32 kbit/s) -> G.711.
109. G.726 (32 kbit/s) -> G.711 ->G.726 (32 kbit/s).

Performance objective:

110. G.711 -> OPUS (32 kbit/s).
111. OPUS (32 kbit/s) -> G.711.
112. OPUS (32 kbit/s) -> G.711 -> OPUS (32 kbit/s).

The following codecs shall be tested for self-tandeming (double and triple):

113. LC3plus (32 kbit/s).
114. G.726 (32 kbit/s).
115. OPUS (32 kbit/s).
116. G.711 (64 kbit/s).

6.2.3 WB conditions

The test shall verify the performance of the candidate codec in WB mode for DECT scenarios. Speech coding for wideband speech connections using a 32 kbit/s payload for normal DECT RF slots shall not be worse than what is achieved recently by Recommendation ITU-T G.722 [3] using a 64 kbit/s payload for long DECT RF slots. The DECT evolution RF connection shall enable at least the same range where communication is possible between DECT PP and FP compared to previous G.722 DECT connections. It is envisioned that the range can be further extended.

The voice quality by transcoding between VoIP networks using G.722 to/from the DECT evolution speech codec shall not be worse than connections between VoIP-G.722 and DECT-G.722.

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

The following WB conditions shall be included into the test (input speech levels which shall be used are -16 dBov, -26 dBov and -36 dBov):

200. Direct reference condition with limited audio bandwidth with cut off frequency of 8 kHz, but no speech coding.

CuT:

201. LC3plus, 32 kbit/s, 16 kHz, 10 ms framing, 16 bits per audio sample.

Requirement:

202. G.722, 64 kbit/s.

Performance objective:

203. OPUS, CELT mode, Constant BitRate (CBR): 32 kbit/s, complexity=0, FEC off, WB mode, 10 ms framing.

To be characterized:

204. LC3plus for bitrates: 32 kbit/s, 48 kbit/s. Sampling rate of 16 kHz and nominal speech level. Short frame size (5 ms frame size) against regular frame size LC3plus 32 kbit/s codec (10 ms frame size).
205. LC3plus for bitrates: 64 kbit/s, 96 kbit/s. Sampling rate of 16 kHz and nominal speech level. Short frame size (2,5 ms frame size) against regular frame size LC3plus 32 kbit/s codec (10 ms frame size).

The following transcoding scenarios shall be tested:

CuT:

206. LC3plus (32 kbit/s) -> G.722 (64 kbit/s).
207. G.722 (64 kbit/s) -> LC3plus (32 kbit/s).
208. LC3plus (32 kbit/s) ->G.722 (64 kbit/s) -> LC3plus (32 kbit/s).

Requirement:

209. G.722 (64 kbit/s) -> G.722 (64 kbit/s).
210. G.722 (64 kbit/s) -> G.722 (64 kbit/s) -> G.722 (64 kbit/s).

Performance objective:

211. OPUS (32 kbit/s) -> G.722 (64 kbit/s).
212. G.722 (64 kbit/s) -> OPUS (32 kbit/s).
213. OPUS (32 kbit/s) -> G.722 (64 kbit/s) -> OPUS (32 kbit/s).

The following codecs shall be tested for self-tandeming (double, triple):

214. LC3plus (32 kbit/s).
215. OPUS (32 kbit/s).
216. G.722 (64 kbit/s).

6.2.4 SWB conditions

Speech coding for super-wideband speech connections using a long 64 kbit/s payload DECT RF slot shall not be worse than what is achieved by EVS-SWB at 13,2 kbit/s and better than what is achieved by Recommendation ITU-T G.722 [3] at 64 kbit/s. The DECT evolution RF connection shall enable the same range where communication is possible between DECT PP and FP as achieved today by G.722 DECT connections.

The voice quality degradation by transcoding between VoIP networks using OPUS (fullband mode) or EVS to/from DECT evolution speech codec shall be characterized.

Additional objectives should be defined in comparison to OPUS (CELT mode, CBR, 64 kbit/s, complexity=0, FEC off, FB mode, 10 ms framing).

The following conditions shall be tested (Input speech levels to be applied are -16 dBov, -26 dBov, -36 dBov):

300. Direct reference conditions with limited audio bandwidth but no speech coding. Lowpass cut-off frequency of 16 kHz shall be used.

CuT:

301. LC3plus, 64 kbit/s at sampling rate of 32 kHz. 10 ms framing, 16 bits per audio sample.

Requirement:

302. G.722 (64 kbit/s, WB, with Appendix IV PLC).
303. EVS (SWB at 13,2 kbit/s). No channel aware mode used.

Performance Objective:

304. OPUS, CELT mode, CBR, 64 kbit/s, complexity = 0, FEC off, fullband, mode, 10 ms framing.

To be characterized:

305. LC3plus for bitrates: 64 kbit/s, 96 kbit/s. Sampling rate of 32 kHz and nominal speech level. Short frame size (5 ms frame size) against regular frame size LC3plus 64 kbit/s codec (10 ms frame size).
306. LC3plus for bitrates: 96 kbit/s, 128 kbit/s. Sampling rate of 16 kHz and nominal speech level. Short frame size (2,5 ms frame size) against regular frame size LC3plus 64 kbit/s codec (10 ms frame size).

The following transcoding scenarios shall be tested:

307. EVS (SWB at 13,2 kbit/s) -> LC3plus (64 kbit/s).

308. LC3plus (64 kbit/s) -> EVS (SWB at 13,2 kbit/s).
309. LC3plus (64 kbit/s) -> EVS (SWB at 13,2 kbit/s) -> LC3plus (64 kbit/s).

310. EVS (SWB at 13,2 kbit/s) -> OPUS (64 kbit/s).

311. OPUS (64 kbit/s) -> EVS (SWB at 13,2 kbit/s).

312. OPUS (64 kbit/s) -> EVS (SWB at 13,2 kbit/s) -> OPUS (64 kbit/s).

313. LC3plus (64 kbit/s) -> OPUS (64 kbit/s).

314. OPUS (64 kbit/s) -> LC3plus (64 kbit/s).

315. LC3plus (64 kbit/s) -> OPUS (64 kbit/s) -> LC3plus (64 kbit/s).

The following codec shall be tested for self-tandeming (double, triple and quadruple):

316. LC3plus (64 kbit/s).
317. Opus (64 kbit/s).
318. EVS (SWB at 13,2 kbit/s).

6.3 Characterization plan for error prone channels with application in DECT scenarios

6.3.1 Overview

The packet loss concealment performance of CuT shall be evaluated compared to G.726 in NB and G.722 in WB and to SWB codec.

Table 4: DECT Packet loss and bit error rates

Packet Loss Profile	Normalized Averaged signal strength (RSSI)	PLR [%]	BER rounded [%]
DP0	1 (136 dB)	0	0
DP1	0,41 (56 dB)	0,99	0,01
DP2	0,35 (48 dB)	0,88	0,31
DP3	0,29 (40 dB)	7,39	2,92

The CuT shall be compared to the requirement condition under four typical signal strengths representing DECT packet loss and bit error profiles for a 10 ms framing. The configurations of the codecs include a specific setup to adapt to the DECT channel characteristic, e.g. configuration of the channel coder or addition of parity bit. The DECT error profiles are labelled DP0, DP1, DP2 and DP3 (see Table 4).

Switching of channel coder configurations (including rate switching) shall be tested as well. Switching within all possible channel configurations shall not be worse than only operating in the mode with highest protection.

For completeness, random patterns of 3 % and 6 % for a 10 ms frame size shall be tested. The random patterns are labelled as FER 3 % and 6 %.

6.3.2 NB conditions

The error profiles DP0, DP1, DP2, DP3 (see Table 5) and random FER 3 % and 6 % shall be applied for:

CuT:

400. LC3plus, 32 kbit/s, sampling rate 8 kHz, 10 ms framing.

Requirement:

401. G.726 32 kbit/s with G.711 Appendix I PLC [10].

6.3.3 WB conditions

The error profiles DP0, DP1, DP2, DP3 (see Table 5) and random FER 3 % and 6 % shall be applied to:

CuT:

500. LC3plus, 32 kbit/s, sampling rate 16 kHz, 10 ms framing.

Requirement:

501. G.722 64 kbit/s with G.722 Appendix IV PLC.

6.3.4 SWB conditions

The error profiles DP0, DP1, DP2, DP3 (see Table 5) and random FER 3 % and 6 % shall be applied for:

CuT:

600. LC3plus, bitrate 64 kbit/s, sampling rate 32 kHz, 10 ms framing.

Requirement:

601. G.722 64 kbit/s with G.722 Appendix IV PLC (G.722 conditions required to check that SWB connections achieve the same DECT distance of portable and fix part compared to legacy WB connections).

6.4 Characterization test plan for clean channels with application in VoIP scenarios

6.4.1 Overview

VoIP networks today use mainly G.711 for NB, G.722 for WB and OPUS for SWB. As SWB services are currently deployed in VoLTE, EVS-SWB may serve as the alternative reference point. However, OPUS is used in the following proposal.

A new ETSI VoIP codec shall provide the same or better speech quality than previous VoIP networks provide. It is envisioned that the new codec provides a Packet Loss Concealment (PLC) better than the PLC provided for G.711 and G.722 for narrowband and wideband calls.

For VoIP, the network interworking scenario with mobile phones shall be of main focus, leading to the transcoding conditions: VoIP to mobile terminals and vice versa.

As relevant mobile codecs AMR-NB, AMR-WB, EVS-WB and EVS-SWB shall be considered operating at the most commonly used configurations as outlined in the following clauses 6.4.2 to 6.4.4.

6.4.2 NB conditions

For narrowband VoIP network speech coding connections using up to 64 kbit/s payload, the CuT shall not be worse than G.711 coding.

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

CuT:

700. LC3plus, bitrate of 32 kbit/s, sampling rate 8 kHz, 10 ms framing, 16 bits per audio sample.

Requirement:

701. G.711 64 kbit/s, with Appendix I PLC.

Performance objective:

702. OPUS, CELT mode, CBR, 32 kbit/s, complexity = 0, FEC off, NB mode, 10 ms framing.
703. AMR-NB 12,2 kbit/s.

The following transcoding scenarios shall be tested:

CuT:

704. AMR-NB (12,2 kbit/s) -> LC3plus (32 kbit/s).
705. LC3plus (32 kbit/s) -> AMR-NB (12,2 kbit/s).

Requirement:

706. AMR-NB (12,2 kbit/s) -> G.711.
707. G.711 -> AMR-NB (12,2 kbit/s).

Performance objective:

708. AMR-NB (12,2 kbit/s) -> OPUS (32 kbit/s).
709. OPUS (32 kbit/s) -> AMR-NB (12,2 kbit/s).

6.4.3 WB conditions

VoIP network speech coding done by CuT for wideband speech connections using up to 64 kbit/s payload shall not be worse than G.722 coding.

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

CuT:

800. LC3plus, 32 kbit/s, sampling rate 16 kHz, 10 ms framing, 16 bits per audio sample.

Requirement:

801. G.722, 64 kbit/s with Appendix IV PLC will be used.

Performance objective:

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

The following transcoding scenarios shall be tested:

CuT:

803. AMR-WB (23,85 kbit/s) -> LC3plus (32 kbit/s).
804. LC3plus (32 kbit/s) -> AMR-WB (23,85 kbit/s).
805. AMR-WB (12,65 kbit/s) -> LC3plus (32 kbit/s).
806. LC3plus (32 kbit/s) -> AMR-WB (12,65 kbit/s).
807. EVS-WB (24,4 kbit/s) -> LC3plus (32 kbit/s).
808. LC3plus (32 kbit/s) -> EVS-WB (24,4 kbit/s).
809. EVS-WB (13,2 kbit/s) -> LC3plus (32 kbit/s).
810. LC3plus (32 kbit/s) -> EVS-WB (13,2 kbit/s).

Requirement:

811. AMR-WB (23,85 kbit/s) -> G.722 (64 kbit/s).
812. G.722 (64 kbit/s) -> AMR-WB (23,85 kbit/s).
813. AMR-WB (12,65 kbit/s) -> G.722 (64 kbit/s).
814. G.722 (64 kbit/s) -> AMR-WB (12,65 kbit/s).
815. EVS-WB (24,4 kbit/s) -> G.722 (64 kbit/s).

816. G.722 (64 kbit/s) -> EVS-WB (24,4 kbit/s).
817. EVS-WB (13,2 kbit/s) -> G.722 (64 kbit/s).
818. G.722 (64 kbit/s) -> EVS-WB (13,2 kbit/s).

Performance objective:

819. AMR-WB (23,85 kbit/s) -> OPUS (32 kbit/s).
820. OPUS (32 kbit/s) -> AMR-WB (23,85 kbit/s).
821. AMR-WB (12,65 kbit/s) -> OPUS (32 kbit/s).
822. OPUS (32 kbit/s) -> AMR-WB (12,65 kbit/s).
823. EVS-WB (24,4 kbit/s) -> OPUS (32 kbit/s).
824. OPUS (32 kbit/s) -> EVS-WB (24,4 kbit/s).
825. EVS-WB (13,2 kbit/s) -> OPUS (32 kbit/s).
826. OPUS (32 kbit/s) -> EVS-WB (13,2 kbit/s).

The following codecs shall be tested for self-tandeming (double, triple):

827. LC3plus (32 kbit/s).
828. OPUS (32 kbit/s).
829. EVS-WB (13,2 kbit/s).

6.4.4 SWB conditions

VoIP network speech coding for super wideband speech connection typically uses a payload of 64 kbit/s. The CuT shall not be worse than OPUS (CELT mode, Constant Bitrate Mode (CBR), 64 kbit/s, complexity = 0, FEC off, fullband mode, 10 ms framing) for coding.

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

CuT:

900. LC3plus, bitrate 64 kbit/s, sampling rate 32 kHz, 10 ms framing, 16 bits per audio sample.

Requirement:

901. OPUS, CELT mode, CBR, 64 kbit/s, complexity = 0, FEC off, fullband mode, 10 ms framing.

The following transcoding scenarios shall be tested:

902. EVS-SWB (24,4 kbit/s) -> LC3plus (64 kbit/s).
903. LC3plus (64 kbit/s) -> EVS-SWB (24,4 kbit/s).
904. EVS-SWB (13,2 kbit/s) -> LC3plus (64 kbit/s).
905. LC3plus (64 kbit/s) -> EVS-SWB (13,2 kbit/s).
906. EVS-SWB (24,4 kbit/s) -> OPUS (64 kbit/s).
907. OPUS (64 kbit/s) -> EVS-SWB (24,4 kbit/s).
908. EVS-SWB (13,2 kbit/s) -> OPUS (64 kbit/s).
909. OPUS (64 kbit/s) -> EVS-SWB (13,2 kbit/s).

The following codecs shall be tested for self-tandeming (double, triple):

910. LC3plus (64 kbit/s).
911. OPUS (64 kbit/s).
912. EVS-WB (13,2 kbit/s).

6.5 Characterization plan for Packet Loss Concealment (PLC) with application in VoIP scenarios

6.5.1 Overview

The following Packet Loss Profiles (PLPs) shall be applied.

Table 5: Packet Loss Profiles for VoIP application testing

N	Packet Loss Profile	Burst Length (No. Packets)	Average Loss Rate (%)	Comments
1	PLP0	0	0	No loss
2	PLP1	1	1,43	Uniform loss
3	PLP2	2	2,87	Synchronized with PLP1
4	PLP4	4	5,74	Synchronized with PLP1
5	PLP7	7	10,05	Synchronized with PLP1

The PLPs have been designed to test the impact of different bursts of losses on the quality of voice when different codecs and packet loss mitigation and resilience mechanisms are used (such as Packet Loss Concealment and Forward Error Correction). As shown in Table 5, each PLP has a distinct length of loss burst.

PLP1 was the first profile to be developed and it was designed by distributing individual losses (i.e. burst length of 1) randomly along the whole PLP. It was used to design the rest of the PLPs by increasing the size of the burst. For example, if an extract of packet statuses from PLP1 was 111110111111 (where 1 indicates a successfully received packet and 0 indicates a lost packet) PLP4 is then developed by switching the status of the 3 packets succeeding the lost one from 1 to 0 (i.e. 111110111111 becomes 111110000111). As a result, all PLP types have bursts that begin at the same location in the speech sample (see Figure 1 for an example applied to a 12 s speech sample consisting of 3 sentences).

NOTE: This only represents one approach to testing conditions with bursty packet loss and the applied loss profiles can be changed (e.g. to replicate the characteristics of particular networks and network conditions).

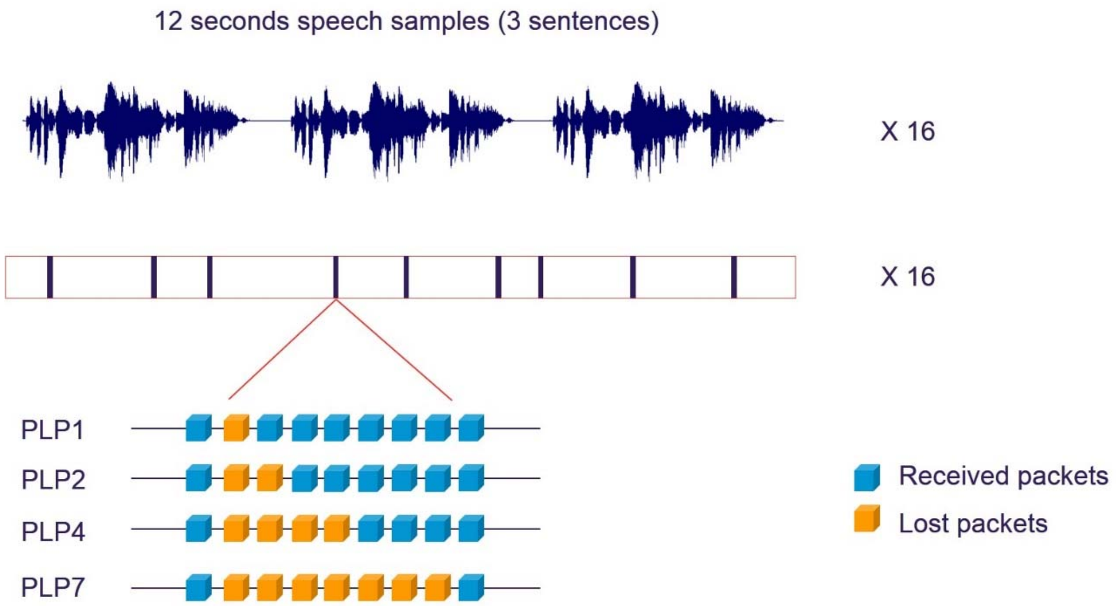

Figure 1: Testing conditions with bursty packet loss

PLC frames should be aligned to the same frames for all speech input frames; For VoIP, only 20 ms PLC frames will be triggered. This means for 10 ms frame codec always 2 frames of PLC indication are in a row.

For completeness, random pattern of 3 % and 6 % for a 20 ms frame size shall be tested. The random patterns are labelled as FER 3 % and 6 %.

As LC3plus may not use the full bit rate of 64 kbit/s of the VoIP transmission, RTP based redundancy modes should be tested as well. Here, besides the main LC3plus frame of the current frame, an additional redundant LC3plus frame with an offset of X packets is transmitted in the same RTP payload. In this scenario, the playout is delayed by X packet, but the redundant LC3plus frame can be used to handle packet losses. The assessment described in the IEEE paper [i.6] mentioned above uses an offset of 3 packets for the EVS codec. The same value should be used for LC3plus. The processed condition may be based on simulations on bit stream level.

It is envisioned that the CuT provides a Packet Loss Concealment (PLC) better than the currently provided PLC for G.711, G.722 and OPUS for NB, WB and SWB calls respectively.

6.5.2 NB conditions

The CuT PLC shall be as good as, or better than G.711 appendix I for random packet losses at 20 ms packet size.

The Packet Loss Profiles PLP0, PLP1, PLP2, PLP4 and PLP7 (see Table 5) and FER 3 % and 6 % shall be applied for:

CuT:

1000. LC3plus, bitrate of 32 kbit/s, sampling rate 8 kHz, 10 ms framing, 16 bits per audio sample.
1001. LC3plus, 64 kbit/s with RTP redundancy mode, sampling rate 8 kHz, 10 ms framing, 16 bits per audio sample (only relevant for PLP4 and PLP7; due to a redundancy configuration with offset=3, single and double losses in PLP1 and PLP2 can be completely compensated).

Requirement:

1002. G.711 64 kbit/s, with Appendix I PLC (Packet Loss Concealment).

6.5.3 WB conditions

The CuT PLC shall be as good as or better than G.722 Appendix IV for random packet losses at 20 ms packet size.

The Packet Loss Profiles PLP0, PLP1, PLP2, PLP4 and PLP7 (see Table 5) and FER 3 % and 6 % shall be applied for:

CuT:

1100. LC3plus, bitrate 32 kbit/s, sampling rate 16 kHz, 10 ms framing, 16 bits per audio sample.
1101. LC3plus, bitrate 64 kbit/s with RTP redundancy mode, sampling rate 16 kHz, 10 ms framing, 16 bits per audio sample (only relevant for PLP4 and PLP7, due to a redundancy configuration with offset=3, single and double losses in PLP1 and PLP2 can be completely compensated).

Requirement:

1102. G.722 64 kbit/s with Appendix IV PLC (Packet Loss Concealment).

6.5.4 SWB conditions

The Packet Loss Profiles PLP0, PLP1, PLP2, PLP4 and PLP7 (see Table 5) and FER 3 % and 6 % shall be applied for:

CuT:

1200. LC3plus, bitrate 64 kbit/s, sampling rate 32 kHz, 10 ms framing, 16 bits per audio sample.
1201. LC3plus, bitrate 64 kbit/s with RTP redundancy mode, sampling rate 32 kHz, 10 ms framing, 16 bits per audio sample (relevant for all PLPs).

Requirement:

1202. OPUS, CELT mode, CBR, 64 kbit/s, complexity=0, FEC off, FB mode, 10 ms framing.

7 Requirement verification

7.1 Requirement verification for subjective tests

Each condition in clauses 6.1 to 6.4 labelled as requirement shall be compared to the corresponding CuT condition using a Student's Dependent Groups t-test (single-sided at 95 % confidence level) on the subjective scores. This data will be the base for verifying that the CuT meets or exceeds the requirement. Additionally, the same rule shall be applied to the performance objectives, whereas the comparison between CuT and performance objective have only informative character.

A complete report for each requirement and objective can be found in clause C.2.7.

7.2 Requirement verification for objective tests

Each condition in clauses 6.2 to 6.5 labelled as requirement shall be compared to the corresponding CuT condition using a Student's Dependent Groups t-test (single-sided at 95 % confidence level) on the objective scores generated as described in clause 5.3. Additionally, the same rule shall be applied to the performance objectives.

The objective requirement verification is only for information. A complete report for each requirement and objective in comparison to the subjective results will be provided in a future revision of the present document.

8 Performance summary

The electronic attachment of the present document "Summary_TTF005_green_v2.xlsx" contained in subfolder "Raw voting data along with their statistical processing" in archive ts_103624v010301p0.zip which accompanies the present document, provides an overview of all statistical results of the LC3plus codec in comparison to requirement and objective conditions based on the subjective data.

The LC3plus passes all statistical tests (zero fails) where LC3plus is compared to the defined requirement and objective reference conditions. Altogether, 153 statistical tests were conducted. In 61 cases, the LC3plus passes the test by being "not worse than" (NWT) the reference condition. In 92 cases, the LC3plus passes the test by being "better than" (BT) the reference condition. This means in 60 % of all tests, LC3plus exceeds the performance reference point defined for DECT and VoIP applications.

9 Conclusion

LC3plus is recommended as the ETSI codec for global deployment in DECT and VoIP applications.

Annex A (normative): Conditions for the P.800 experiments

The conditions for the P.800 experiments are contained in subfolder "Conditions for the P.800 experiments" in archive ts_103624v010301p0.zip which accompanies the present document.

Compared to the conditions for the seven P.800 experiments as contained in archive ts_103624v010101p0.zip which was part of V1.1.1 of the present document, see https://www.etsi.org/deliver/etsi_ts/103600_103699/103624/01.01.01_60/ts_103624v010101p0.zip, the experiment FB MB_fullscale_short is added as smaller version of FB MB_fullscale in order to be realizable in two languages.

Annex B (normative): Subjective characterization test plan for the ETSI LC3plus codec

B.1 Introduction

This annex contains the subjective test plan for the ETSI LC3plus codec.

Test methodologies used for all planned subjective tests are based on Recommendation ITU-T P.800 [1], the ITU-T Handbook [14] and the guidelines provided in the main part of the present document.

The following steps are part of the test plan:

- Test samples are delivered in "ready-to-listen-to" format, means cropped and windowed, at a level corresponding to the final playout listening level, and grouped by listening panel.
- The randomizations are arranged different for each listening panel. The randomization tables are contained subfolder "Randomization tables" in archive ts_103624v010301p0.zip which accompanies the present document.
- The raw voting data along with their statistical processing (MOS calculation and conditional STD and CI95) are contained in subfolder "Raw voting data along with their statistical processing" in archive ts_103624v010301p0.zip which accompanies the present document.
- No sample name blinding or anonymization will be applied, as it is principally not needed.

B.2 Description of the subjective experiments

B.2.1 General considerations

- Four talkers (two males, two females) are used in each language.
- 24 subjects for each experiment, minimum 4 (2M+2F) talkers, minimum 6 samples per talker, minimum 4 votes per sample, minimum 96 votes per condition, each panel with an independent randomization.
- Preliminary or training conditions are selected from the material available by the listening lab.
- Randomizations are constructed under "partially-balanced/randomized blocks" experimental design described in the ITU-T Handbook [14].
- Test duration: maximum 2 hours per listening panel. Test duration comprises 50 % of actual listening time and 50 % test overhead including administration, initial briefing, preliminaries, and breaks.
- Listening level -21 dB Pa (73 dB SPL) equals to -26 dBov.
- Files are played back with diffuse-field equalized headphones and diotic presentation.
- Six (6) experiments are performed using ACR methodology of Recommendation ITU-T P.800 [1] and one experiment is performed using DCR methodology [1]. Overall, 7 experiments are performed.
- Listening environment: All tests are performed in an acoustically treated critical listening room that conforms to Recommendation ITU-T P.800 [1] requirements in full.

Details of the listening test instructions are provided in clause B.3.

B.2.2 Speech Material

Each source speech file contains a sentence pair and lasts exactly 8 s. Each sentence is centred inside a 4 s time window. Leading and trailing silence parts are longer than 0,5 s. The sentences are simple meaningful sentences, similar to those described in annex B.1.4 of Recommendation ITU-T P.800 [1].

The test languages will be American English (Experiment 1 to 6 and Experiment 7a) and Mandarin (Experiment 7b) as outlined in Table B.1.

B.2.3 Test samples

Each experiment sample database consists of a number of samples to be calculated by: nTalker(4) x nSamples(6) x nConditions (see Table B.1). The test sample are processed according to clause B.4.

B.2.4 Experiments

Table B.1 outlines the basic parameters of the experiments.

Table B.1: Experiment parameters

Experiment number	Experiment label (Excel sheet)	Experiment designator (file naming)	Max. audio bandwidth	Channel conditions	Number of conditions (nConditions)	Rating Scale	Language
1	NB clean	n1	4 000 Hz	No error	40	ACR	English
2	NB error	n2	4 000 Hz	Bit error & packet loss	32	ACR	English
3	WB clean	w1	8 000 Hz	No error	60	ACR	English
4	WB error	w2	8 000 Hz	Bit error & packet loss	32	ACR	English
5	SWB clean	s1	16 000 Hz	No error	40	DCR	English
6	SWB error	s2	16 000 Hz	Bit error & packet loss	35	ACR	English
7a	FB MB_fullscale short	f1	20 000 Hz	No error & packet loss & bit error	30	ACR	English
7b	FB MB_fullscale short	f2	20 000 Hz	No error & packet loss & bit error	30	ACR	Mandarin
NOTE: The FB MB_fullscale_short fullscale experiment contains all bandwidth conditions to span the complete quality range of Recommendation ITU-T P.800 [1].							

B.2.5 Test randomizations

For presentation to the subjects, the samples are grouped into 3 or 6 panels according to the "partially-balanced/randomized blocks" experimental design.

Table B.2 visualizes the items to be presented for each panel. The item selection follows the partially-balanced/randomized blocks design. The list are defined for 6 or 3 panels. For 3 panel presentation, panel 1 & 2, 3 & 4 and 5 & 6 need to be combined.

Table B.2: Experimental design of panels

	Block	Seq	Panel 1	Panel 2	Panel 3	Panel 4	Panel 5	Panel 6
0	1	1	an1f1s5.c34	an1f1s3.c13	an1f2s4.c16	an1f1s4.c28	an1f2s4.c39	an1f2s4.c26
1	1	2	an1m1s2.c13	an1m1s1.c33	an1m2s4.c26	an1m2s1.c31	an1m1s1.c01	an1m1s3.c37
2	1	3	an1f2s2.c05	an1f1s6.c29	an1f1s2.c23	an1f1s3.c33	an1f1s6.c07	an1f2s3.c38
…								
…								

The complete randomization tables are contained in subfolder "Randomization tables" in archive ts_103624v010301p0.zip which accompanies the present document.

B.3 Instructions for the subjective tests

B.3.1 P.800 ACR test instructions in English

In this experiment, you will be listening to short samples via headphones, and giving your opinion of the speech you hear.

Follow the instructions on the touchscreen in front of you, and listening to each sample, press the appropriate button to indicate your opinion on the following scale.

WHAT WAS THE QUALITY OF THE SAMPLE YOU HAVE JUST HEARD?

5 Excellent

4 Good

3 Fair

2 Poor

1 Bad.

For playing subsequent sample, follow the instructions on the screen. Please do not discuss your opinions with other subjects participating in the experiment. Thank you for your help in this experiment.

B.3.2 P.800 DCR test instructions in English

In this experiment, you will be listening to pairs of speech samples via headphones, and giving your opinion on the quality of the second sample compared to the quality of the first sample.

Follow the instructions on the touchscreen in front of you, and listening to each sample pairs, press the appropriate button to indicate your opinion on the following scale:

WHAT WAS THE QUALITY DEGRADATION OF THE SECOND SAMPLE COMPARED TO THE FIRST SAMPLE IN THE PAIR YOU HAVE JUST HEARD?

5 Degradation is inaudible

4 Degradation is audible but not annoying

3 Degradation is slightly annoying

2 Degradation is annoying

1 Degradation is very annoying.

For playing subsequent sample pair, follow the instructions on the screen. Please do not discuss your opinions with other subjects participating in the experiment. Thank you for your help in this experiment.

B.3.3 P.800 ACR test instructions in Mandarin Chinese

ACR主观测试示例说明

在本实验中，我们将对用于电信业务的语音频编解码系统进行评估。

实验包含若干个小节。在每一小节中，您将通过耳机听到一个音频。每个样本包含 同一个人说的两句话。请您认真听完整个样本，然后根据您对样本整体音质的感受，按下面的5分制进行打分：

5 非常好

4 好

3 一般

2 较差

1 很差

请注意：您的任务是对语音样本的整体音质进行评价。

在您听完一个样本后，请您对刚才听到的样本的整体音质按上述的评分方法进行打分。

您打完分之后，会有一个短暂的停顿，然后播放下一个样本。

请您在实验中不要和其他测听者讨论您打的分数。

B.3.4 P.800 DCR test instructions in Mandarin Chinese

DCR主观测试示例说明

在本实验中，我们将对用于电信业务中的语音频编解码系统进行评估。

实验包含若干个小节。每一小节中包含两个样本，每个样本为一个音频样本。每个小节中的第一个样本是参考样本，第二个样本的内容与第一个样本的内容完全相同，但它是经过了一个电信系统后得到的。请您认真听完每一小节中的两个样本，然后根据您察觉到的第二个样本相对于第一个样本的整体音质失真程度，按下面的5分制进行打分：

5 听不出音质失真

4 能听到音质失真，但不令人厌烦

3 能听到音质失真，且有一点令人厌烦

2 能听到音质失真，且令人厌烦

1 能听到音质失真，且非常令人厌烦

请注意：您的任务是对您所察觉到的第二个待测样本相对于第一个参考样本的失真程度给出相应的分数。请您在听完两个样本之后，按上述的评分方法进行打分。

请您在实验中不要和其他测听者讨论您打的分数。

B.4 Processing functions for the ETSI LC3plus codec characterization phase

B.4.1 Introduction

This clause and the following clauses define how audio material shall be prepared for the characterization tests of the ETSI LC3plus codec and provides details on the software modules and files required.

B.4.2 Definitions and formats

The following filename convention is used for source material:

Input<Fs> Fs (in kHz) is the input sampling frequency to a processing module (or sequence of modules). Fs is either 8, 16, 32 or 48.

Output<Fs> Fs (in kHz) is the output sampling frequency from a processing module (or sequence of modules). Fs is either 8, 16, 32 or 48.

The format of the above files is headerless PCM with samples stored in 16-bit 2's complements little endian format. Other command line filenames and variables are described in the relevant part of the present document.

The codecs use the Recommendation ITU-T G.192 [15] format as a common bit-stream interface, if available.

All executables are in the Win32 format, i.e. are native binaries for the 32-bit Microsoft Windows™ platform.

B.4.3 Processing Stages for the LC3plus codec characterization phase

B.4.3.1 Introduction

This clause and the following clauses define, in the form of diagrams, the processing stages required by the LC3plus codec under test. The latest version of the ITU-T Software Tool Library as provided in Recommendation ITU-T G.191 [6] is used for this processing.

The source material shall be 48 kHz sampled with 16 bit resolution. The format shall be headerless PCM, little endian. For full-band experiments the source material shall have frequency components above 16 kHz.

B.4.3.2 Source material pre-processing requirements

B.4.3.2.1 Input speech file naming

The filenames of the input speech samples are represented by:

aeegysz.48k

where:

- *ee* stands for the experiment number, e.g. n1 (see Table B.2)
- *g* is gender of talker (i.e. f for female and m for male) and *y* is the talker number: 1, 2, 3
- *s* stands for sample and z is the sample number; 1, 2, 3, 4, 5, ..., x

B.4.3.2.2 Processed speech file naming

The filenames of the processed speech samples are represented by:

aeegysz.cnn

where:

- *ee* stands for the experiment designator, e.g. n1 (see Table B.2)
- *g* is gender of talker (i.e. f for female and m for male) and *y* is the talker number: 1, 2, 3
- *s* stands for sample and z is the sample number; 1, 2, 3, 4, 5, ..., x
- *c* stands for the condition with the number nn = 01, 02, ... as specified in this annex.

B.4.3.3 File concatenation, separation, sequences and module initialization

B.4.3.3.1 Concatenated sequences processing

In all experiments, the pre-processed material will be processed in concatenated files comprising a preamble and a series of sentences for speech. The preamble will be 10 s long.

B.4.3.3.2 Preamble definition

The preamble in the speech path shall not be digital silence (i.e. 16 bit samples equal to zero), but a low-level random noise with amplitude between +4 and -4.

B.4.3.3.3 Frame error application

In frame erasure conditions, the erasures shall affect the same segments of speech signal for the all codecs in the test. This shall be done by compensating for all encoder-side delays (or for all encoder-side delays of the core layer in case of embedded codecs). Some codecs, e.g. EVS or LC3plus, have encoder and decoder delay compensation implemented in the executable. For all other codecs, delay shall be compensated prior to the processing by the reference encoder, as specified in Table B.4.

NOTE: Exact delay compensation might not be possible.

B.4.3.3.4 File naming for error patterns

Random frame erasure files will have the name "*patterns\eeEPFt1_rr_t2.g192*", where:

- *ee* stands for the experiment designator, e.g. n1 (see Table B.2)
- *t1* stands for original frame size of pattern

- *rr* stands for the frame loss rate in per cent, e.g. 06, or 10
- *t2* stands for adapted frame size of pattern, e.g. duplicated to align 20 ms to 10 ms frame size

Bursty frame erasure files will have the name "*patterns\eePLP_rr_t1.g192*", where:

- *ee* stands for the experiment designator, e.g. n1 (see Table B.2)
- *rr* stands for the frame loss rate frame in a row, i.e. 1, 2, 4 or 7
- *T1* stands for frame size of pattern

B.4.3.4 File naming for rate switching profiles

Rate switching profiles will have the name "*patterns\eersx*", where:

- *ee* stands for the experiment designator, e.g. n1 (see Table B.2);
- *rs* stands for rate switching file;
- *x* stands for the sequence in the experiment, i.e. 1, 2, etc.

B.4.3.5 Concatenation setup

Audio files are concatenated after level adjustment. For concatenation order, see Table B.3 below where all samples are concatenated from left to right and line by line. For speech files, the concatenation order is therefore *m1s1*, *f1s1*, *m2s1*, *f2s1*, … *mYs1*, *fYs1, m1s2, f1s2,* etc. The final concatenated file consists of the preamble and the audio file.

Table B.3: Concatenation order of speech files

Sample	Talker sequence in concatenated files						
1	m1	f1	m2	f2	..	mY	fY
2	m1	f1	m2	f2	..	mY	fY
…							
x							

For all experiments, the processed concatenated files should be divided into separated audio samples and named with the according file extension as specified in the test plan. This shall be performed before final up-sampling stage, and a cosine (Hanning) window of duration 100 ms shall be applied to the start and end of the separated files.

B.4.3.6 Processing for all conditions

B.4.3.6.1 Overview

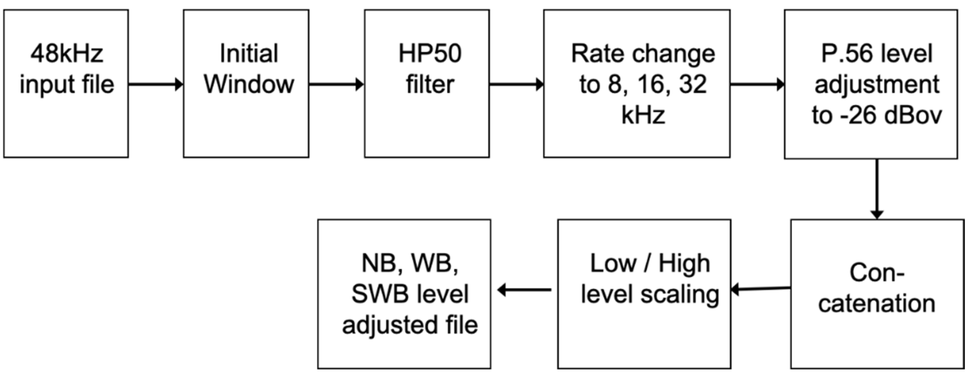

Figure B.1: Processing for all conditions

B.4.3.6.2 General processing stages

B.4.3.6.2.1 Direct conditions

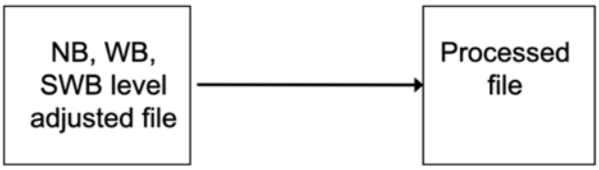

Figure B.2: Processing for direct conditions

B.4.3.6.2.2 Processing for MNRU reference conditions

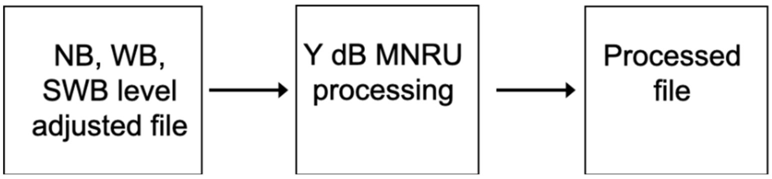

Figure B.3: Processing of MNRU anchors at Y dB

B.4.3.6.2.3 Codec conditions

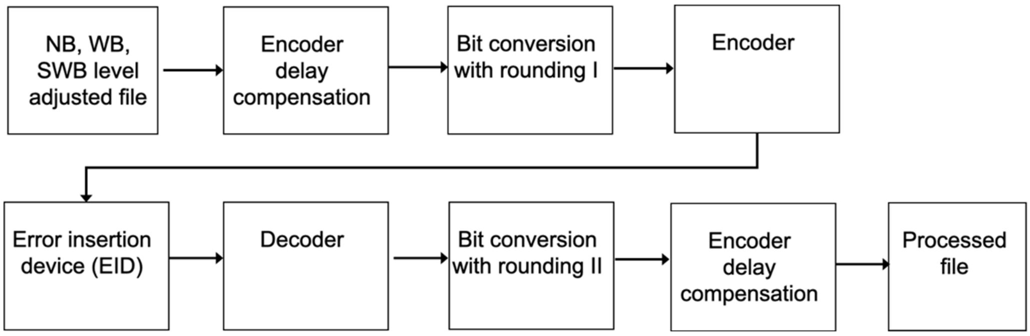

Figure B.4: Processing of codec conditions

Table B.4: Codec parameter table

Codec	Supported input	Encoder delay	Bit conversion	Decoder delay/ms
LC3plus	NB, WB, SWB	0	none	0
G.726	NB	0	13 bits	0
G.711	NB	0	13 bits	0
G.722	WB	11	14 bits	11
AMR-NB	NB	40	13 bits	0
AMR-WB	WB	80	14 bits	15
EVS	NB, WB, SWB	0	none	0
OPUS	NB, WB, SWB	0	none	0
NOTE: OPUS expects 48 kHz as SWB input, see clause B.4.4.3.3.7.				

B.4.3.6.2.4 Tandeming conditions

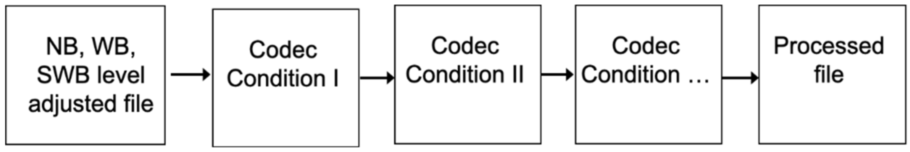

Figure B.5: Processing of tandemed codecs

The two or more codec stages as outlined in clause B.4.3 are processed.

NOTE: No encoder/decoder delay compensations are applied.

B.4.3.7 Post-processing

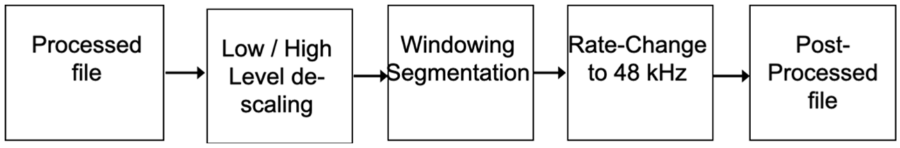

Figure B.6: Post-processing

B.4.3.8 Processing for multi-bandwidth conditions

For experiments containing conditions with multiple audio bandwidths, the relevant pre-processing steps for a dedicated maximum bandwidth are conducted. Some codecs need to be adapted to their native bandwidth by resampling steps.

In the native sample rate domain, the regular encoding, decoding and other processing steps for the codecs can be applied. Other processing steps might be delay compensation or error insertion.

B.4.4 Processing modules

B.4.4.1 Introduction

This clause and the following clauses describe the modules that shall be used in the processing of speech.

B.4.4.2 Pre- and post processing operations

B.4.4.2.1 General delay compensation for the STL filter tool

All filtering steps include a delay compensation step. For preparing the delay compensation, samples of the preamble are added to the end of the input file before applying the filter step. After completion of the filtering step, the samples are to be removed from the beginning of the filtered file.

B.4.4.2.2 Filtering operations

To produce a 50 Hz high pass filtered 48 kHz sampling file use:

```
filter.exe HP50_48KHZ Input48 Output48 960
```

B.4.4.2.3 Recommendation ITU-T P.56 active speech level adjustment

To normalize the P.56 ASL of an 8 kHz sampling file to -26 dBov, use:

```
sv56demo.exe -lev -26 -sf 8000 Input8 Output8 160
```

To normalize the P.56 ASL of a 16 kHz sampled file to -26 dBov, use:

```
sv56demo.exe -lev -26 -sf 16000 Input16 Output16 320
```

To normalize the P.56 ASL level of a 32 kHz sampled file to -26 dBov, use:

```
sv56demo.exe -lev -26 -sf 32000 Input32 Output32 640
```

To normalize the P.56 ASL level of a 48 kHz sampled file to -26 dBov, use:

```
sv56demo.exe -lev -26 -sf 48000 Input48 Output48 960
```

B.4.4.2.4 Low/high level de-/scaling

In case the speech level shall be -26 dBov, this processing step shall be bypassed. Otherwise, the following gain needs to be applied to the signal:

Table B.5: Scaling and descaling

Speech level	Scaling	Descaling
-16 dBov	G=10	G=-10
-36 dBov	G=-10	G=10

To apply the scaling operation with the gain G, use:

```
scaldemo.exe -dB -gain G -bits 16 -round –nopremask -blk BBB Input Output
```

where `BBB` is 160 for 8 kHz sampled files, 320 for 16 kHz sampled files and 640 for 32 kHz sampled files.

B.4.4.2.5 File concatenation

To concatenate files, the concat command is used:

```
concat.exe –undo undo_concat.txt file1 [file2 file3 …] catfile
```

Where `file1, file2,` … are the files to be concatenated and `catfile` is the concatenated file. The `undo_concat.txt` contains the parameters for segmentation.

In case additional silence sections need to be inserted before concatenation, use the following for all input files:

```
concat.exe silence fileX silence tmp_file
copy tmp_file fileX
```

where the silence file contains 0,2 s of digital silence and `fileX` is one of the input files.

B.4.4.2.6 Audio format conversion

B.4.4.2.6.1 PCM to WAV conversion

To convert a 32 kHz sampled PCM file to a WAV file for LC3plus, use:

```
CopyAudio.exe -F WAVE-NOEX -P integer16,0,32000,native,1,default -I "" input32
output32.wav
```

B.4.4.2.6.2 WAV to PCM conversion

To convert a WAV file to a 32 kHz sampled PCM file for LC3plus, use:

```
CopyAudio.exe -F noheader -D integer16 input32.wav output32
```

B.4.4.2.7 Sampling rate changes

B.4.4.2.7.1 Rate-change from 48 kHz to 8 kHz sampling

To produce an 8 kHz sampling file from a 48 kHz sampling file, use:

```
filter.exe –down SHQ3 input48 tmp16 960
filter.exe –down SHQ2 tmp16 output8 320
```

B.4.4.2.7.2 Rate-change from 48 kHz to 16 kHz sampling

To produce a 16 kHz sampling file from a 48 kHz sampling file, use:

```
filter.exe –down SHQ3 input48 output16 960
```

B.4.4.2.7.3 Rate-change from 48 kHz to 32 kHz sampling

To produce a 32 kHz sampling file from a 48 kHz sampling file, use:

```
filter.exe –up SHQ2 input48 tmp96 960
filter.exe –down SHQ3 tmp96 output32 1920
```

B.4.4.2.7.4 Rate-change from 8 kHz to 48 kHz sampling

To produce a 48 kHz sampling file from an 8 kHz sampling file, use:

```
filter.exe –up SHQ2 input8 tmp16 160
filter.exe –up SHQ3 tmp16 output48 320
```

B.4.4.2.7.5 Rate-change from 16 kHz to 48 kHz sampling

To produce a 48 kHz sampling file from a 16 kHz sampling file, use:

```
filter.exe –up SHQ3 input16 output48 320
```

B.4.4.2.7.6 Rate-change from 32 kHz to 48 kHz sampling

To produce a 48 kHz sampling file from a 32 kHz sampling file, use:

```
filter.exe –up SHQ3 input32 tmp96 640
filter.exe –down SHQ2 tmp96 output48 1920
```

B.4.4.2.7.7 Rate-change from 32 kHz to 16 kHz sampling

To produce a 16 kHz sampling file from a 32 kHz sampling file, use:

```
filter.exe –down SHQ2 input32 output16 640
```

B.4.4.2.7.8 Rate-change from 32 kHz to 8 kHz sampling

To produce a 8 kHz sampling file from a 32 kHz sampling file, use:

```
filter.exe –down SHQ2 input32 output16 640
filter.exe –down SHQ2 output16 output8 320
```

B.4.4.2.7.9 Rate-change from 16 kHz to 32 kHz sampling

To produce a 32 kHz sampling file from a 16 kHz sampling file, use:

```
filter.exe –up SHQ2 input16 output32 320
```

B.4.4.2.7.10 Rate-change from 8 kHz to 32 kHz sampling

To produce a 32 kHz sampling file from a 8 kHz sampling file, use:

```
filter.exe –up SHQ2 input8 output16 160
filter.exe –up SHQ2 output16 output32 320
```

B.4.4.2.7.11 Rate-change from 16 kHz to 8 kHz sampling

To produce a 8 kHz sampling file from a 16 kHz sampling file, use:

```
filter.exe –down SHQ2 input16 output8 320
```

B.4.4.2.8 Windowing and segmentation

B.4.4.2.8.1 Segmentation for NB conditions

To extract and window an `m` sample long file beginning at sample `s` from a 8 kHz single channel concatenated file, use:

```
astrip.exe –sample –smooth –wlen 800 –start s -n m input8 output8
```

B.4.4.2.8.2 Segmentation for WB conditions

To extract and window an `m` sample long file beginning at sample `s` from a 16 kHz single channel concatenated file, use:

```
astrip.exe –sample –smooth –wlen 1600 –start s -n m input16 output16
```

B.4.4.2.8.3 Segmentation for SWB conditions

To extract and window an `m` sample long file beginning at sample `s` from a 32 kHz single channel concatenated file, use:

```
astrip.exe –sample –smooth –wlen 3200 –start s -n m input32 output32
```

B.4.4.2.8.4 Initial windowing

To apply the initial windowing of a 48 kHz input speech file, use:

```
astrip.exe –sample –smooth –wlen 4800 –start s -n m input48 output48
```

B.4.4.2.9 Bit conversion

B.4.4.2.9.1 Conversion from 16 bit to 13 bit

To convert an 8 kHz narrowband signal from a 16 bit representation to a 13 bit representation including rounding, use:

```
scaldemo.exe -lin -gain 1 -bits 13 -round -nopremask -blk 160 input8 output8
```

B.4.4.2.9.2 Conversion from 16 bit to 14 bit

To convert a 16 kHz wideband signal from a 16 bit representation to a 14 bit representation including rounding, use:

```
scaldemo.exe -lin -gain 1 -bits 14 -round -nopremask -blk 320 input16 output16
```

B.4.4.2.10 Delay compensation for filter operations

The processing steps are delay-compensated in order to apply error insertion on the same parts of the audio signal and to be able to extract the original length and offset for each audio sample used in the tests.

The delay compensation is initialized by concatenating the file to be filtered and the first 960 samples of the preamble. After processing, the delay of the processing operation is compensated and the original file length is restored.

To compensate the delay for filter operations and reference conditions for encoder and decoder in the common scripts, use:

```
astrip –sample –start S+1 –n FILELENGTH input output
```

where `FILELENGTH` denotes the size in samples and the value for S for each filtering operation is given in Table B.6.

Table B.6: Delay compensation values for filter operations

Filter operation	Value for delay compensation after filtering operation
-up SHQ2	436
-up SHQ3	436
-down SHQ2	218
-down SHQ3	145
HP50_48KHZ	839
MSIN	92

B.4.4.3 Processing

B.4.4.3.1 Introduction

This clause and the following clauses describe the input file and channel operations processing that shall be used in the preparation of audio material.

B.4.4.3.2 MNRU reference conditions

To generate an MNRU reference at XXX dB for a file *Input8*, use:

```
mnrudemo.exe –Q XXX Input8 Output8 160
```

To generate an MNRU reference at XXX dB for a file *Input16*, use:

```
mnrudemo.exe –Q XXX Input16 Output16 320
```

To generate a P.50 MNRU reference at XXX dB for a file *Input48*, use:

```
p50mnru.exe Input48 Output48 XXX M
```

NOTE: Recommendation ITU-T P.50 [17] MNRU processing for SWB conditions requires rate-change steps between 32 kHz and 48 kHz.

B.4.4.3.3 Encoder and decoder

B.4.4.3.3.1 AMR-NB

To process a file *Input8* through the AMR-NB codec at XXX kbit/s, use:

```
amr_cod_vad2.exe [-dtx] BBB Input8 bitstream
amr_dec.exe bitstream Output8
```

where `BBB` is the bitrate mode corresponding to XXX as given in the following table.

XXX [kbit/s]	4.75	7.4	7.95	10.2	12.2
BBB	MR475	MR74	MR795	MR102	MR122

NOTE: AMR-NB operates at 20 ms frame length and has encoder and decoder delay of 5 ms and 0 ms, respectively.

B.4.4.3.3.2 G.711

To process a file *Input8* through the G.711 codec with 20 ms blocks, use:

```
g711demo.exe A lili Input8 Output8 160
```

where A stands for A-law.

NOTE: G.711 does not have algorithmic delay.

B.4.4.3.3.3 G.726

To process a file *Input8* through the G.726 codec with 10 ms blocks, use:

```
g711demo.exe A lilo Input8 out.g711 80
g726demo.exe A load 32 out.g711 bitstream 80

g726demo.exe A adlo 32 bitstream out.g726 80
g711demo.exe A loli out.g726 Output8 80
```

where A stands for A-law.

B.4.4.3.3.4 AMR-WB

To process a file *Input16* through the AMR-WB codec at XXX kbit/s, use Recommendation ITU-T G.722.2 [19] as:

```
amrwb_cod.exe [–dtx] –itu BBB Input16 bitstream
amrwb_dec.exe -itu bitstream Output16
```

where BBB is the bitrate mode corresponding to XXX as given in the following table.

XXX [kbit/s]	6.6	8.85	12.65	14.25	15.85	18.25	19.85	23.05	23.85
BBB	0	1	2	3	4	5	6	7	8

NOTE: AMR-WB operates at 20 ms frame length and has encoder and decoder delay of 5 ms and 0,9375 ms, respectively.

B.4.4.3.3.5 G.722

To process a file *Input16* through the G.722 codec at XXX kbit/s using 20 ms block size, use:

```
encg722.exe –fsize 320 –mode XXX Input16 bitstream
decg722.exe -fsize 320 –mode XXX bitstream Output16
```

where XXX is either 56 or 64.

NOTE: G.722 operates at 20 ms frame length and has an overall algorithmic delay of 1.625 ms.

B.4.4.3.3.6 EVS

To process a file *Input8* through the EVS codec at XXX bit/s, use:

```
EVS_cod.exe [-dtx] [-no_delay_cmp] XXX/SWF 8 Input8 bitstream
EVS_dec.exe [-no_delay_cmp] 8 bitstream Output8
```

where XXX is one of 5900, 7200, 8000, 9600 or 13200. For rate switching operation, XXX is replaced by a switching file (SWF).

To process a file *Input16* through the EVS codec at XXX bit/s, use:

```
EVS_cod.exe [-dtx] [-no_delay_cmp] XXX/SWF 16 Input16 bitstream
EVS_dec.exe [-no_delay_cmp] 16 bitstream Output16
```

where XXX is one of 5900, 7200, 8000, 9600, 13200, 16400, 24400, 32000, 48000, 64000, 96000 for testing non-IO modes or 6600, 8850, 12650, 14250, 15850, 18250, 19850, 23050, 23850 for testing AMR-WB IO modes. For rate switching operation, XXX is replaced by a switching file (SWF).

To process a file *Input32* through the EVS codec at XXX bit/s, use:

```
EVS_cod.exe [-dtx] [-no_delay_cmp] XXX/SWF 32 Input32 bitstream
EVS_dec.exe [-no_delay_cmp] 32 bitstream Output32
```

where XXX is one of 13200, 16400, 24400, 32000, 48000, 64000, 96000, 128000. For rate switching operation, XXX is replaced by a switching file (SWF).

To process a file *Input48* through the EVS codec at XXX bit/s, use:

```
EVS_cod.exe [-dtx] [-no_delay_cmp] –max_band FB XXX/SWF 48 Input48 bitstream
EVS_dec.exe [-no_delay_cmp] 48 bitstream Output48
```

where XXX is one of 16400, 24400, 32000, 48000, 64000, 96000, 128000. For rate switching operation, XXX is replaced by a switching file (SWF).

The switching file consists of XXX values indicating the bit rate for each frame in bit/s. These values are stored in binary format using 4 byte per value.

NOTE: EVS encoder and decoder provide delay compensated output files except if used for tandem conditions, where the option "-no_delay_cmp" is to be enabled.

B.4.4.3.3.7 OPUS

To process an *Input8* file through the Opus codec at XXX bit/s, use:

```
opus_demo.exe restricted-lowdelay 8000 1 XXX -cbr -bandwidth NB
[-no_delay_comp] [-epf patternFile] -framesize 10 -complexity 0 Input8 Output8
```

To process an *Input16* file through the Opus codec at XXX bit/s, use:

```
opus_demo.exe restricted-lowdelay 16000 1 XXX -cbr -bandwidth WB
[-no_delay_comp] [-epf patternFile] -framesize 10 -complexity 0 Input16 Output16
```

To process an *Input32* file through the Opus codec at XXX bit/s, apply rate-change from 32 kHz to 48 kHz to generate *Input48* and use:

```
opus_demo.exe restricted-lowdelay 48000 1 XXX -cbr -bandwidth FB
[-no_delay_comp] [-epf patternFile] -framesize 10 -complexity 0 Input48 Output48
```

And resample *Output48* to *Output32*

Where *patternFile* indicates an error pattern file in 16 bit format triggering concealment.

NOTE 1: patternFile is not in G.192 format.

NOTE 2: Opus encoder and decoder provide delay compensated output files except if used for tandem conditions, where the option "-no_delay_comp" is to be enabled. The delay compensation is separated into half delay portion for encoder and half for decoder to align the bit stream comparable to all other codecs for best PLC testing.

B.4.4.3.3.8 LC3plus

Convert the *Input8, Input16, Input24, Input32, Input48 to Input.wav*.

To process an *Input.wav* file through the LC3plus codec at XXX bit/s, use:

```
LC3plus.exe -v -E -formatG192 -cfgG192 CONFIG.txt -d D -frame_ms F [-epmode EP]
Input.wav bitstream
LC3plus.exe -v -E -formatG192 -cfgG192 CONFIG.txt -d D [-epmode EP] bitstream
Output.wav
```

Where *F* indicated the used frame duration in ms, i.e. 10, 5 or 2.5. *EP* indicates the used error protection class on encoder side, i.e. 1 to 4, and enabled epmode on decoder side.

Convert the *Output.wav* to *Output8, Output16, Output32, Output24*.

NOTE: LC3plus encoder and decoder provide delay compensated output files except if used for tandem conditions, where the option "-d 0" is enabled. The delay compensation is separated into half delay portion for encoder and half for decoder to align the bit stream comparable to all other codecs for best PLC testing, indicated by "-d 2".

B.4.4.4 Error Insertion Device (EID)

B.4.4.4.1 Introduction

For the conditions where random frame erasures are desired, frame erasure patterns are applied to the bitstream using tools from Recommendation ITU-T G.191 [6].

B.4.4.4.2 Frame error tool

For the G.711 A/µ-law, the following processing shall be used:

```
eid-int –ep g192 –factor 2 ep.g192 ep10.g192
g711iplc.exe ep10.g192 input.8k ouput.8k
```

where:

`ep.g192` is the error pattern file assuming 20 ms frames and `ep10.g192` is the error pattern file for the g711iplc tool where all entries are doubled to take the 10 ms frame grid of the g711iplc tool into account

`input.8k` is the G.711 decoded output file

`output.8k` is the G.711 decoded output file with packet loss concealment

For the G.726, the following processing shall be used:

```
G726_applyBE input epBER.g192 output
G726_applyFL input epPLC.g192 output
```

where:

`epBER/PLC.g192` is the error pattern file assuming 20 ms frames.

For all other reference codecs and LC3plus, the following processing shall be used:

`eid-xor.exe –vbr –ber g192bsin epBER.g192 g192bsout` (for bit errors)

`eid-xor.exe –vbr –fer g192bsin epPLC.g192 g192bsout` (for packet loss)

where:

`g192bsin` is the input bit stream

`epBER.g192` is the error pattern file indicating bit errors `epPLC.g192` is the error pattern file indicating frame losses

`g192bsout` is the output bit stream

B.4.4.4.3 Pattern generation

B.4.4.4.3.1 Random pattern

The error patterns used are generated using the gen-patt tool as follows:

```
gen-patt.exe -tailstat -fer -g192 -gamma 0 -rate XXX -tol 0.001      -reset -n
LENGTH -start 501 ep.g192
```

where `XXX` is the required erasure rate, e.g. 0.03 for 3 % and 0.06 for 6 % FER.

Different error patterns should be generated for each experiment. `LENGTH` is the number of 20 ms or 10 ms frames indicating the length of the whole input file (preamble and concatenated speech). Patterns are generated at highest erasure rate and lower erasures rates are generated by randomly changing bad frames to good frames.

B.4.4.4.3.2 Bursty pattern (PLP)

A random pattern is generated for the erasure rate 1,43 % and 20 ms frame size with the file name PLP_1_20_g192. To generate:

- PLP_2_20_g192: Each bad frame is followed by one additional bad frames
- PLP_4_20_g192: Each bad frame is followed by three additional bad frames
- PLP_7_20_g192: Each bad frame is followed by six additional bad frames

B.4.4.4.3.3 Converting pattern

To adapt to different frame sizes of codecs, e.g. from 20 ms to 10 ms, use:

```
eid-int.exe -ep g192 -factor 2 PLP_2_20_g192 PLP_2_10_g192
```

B.4.4.4.3.4 DECT RSSI pattern

The patterns are generating by randomly selecting frame out of measured DECT error profiles given a certain Receive Side Signal Indication (RSSI). The generated pattern for bit errors is denoted as "epSS_PPBER.192" and the pattern for frame loss is denoted as "epSS_PPPLC.192" where SS indicates the RSSI 80, 56, 48, 40 and PP is the payload length in bytes.

B.4.4.4.4 Switching profile generation

To generate EP mode switching profiles for LC3plus conditions, use:

```
gen-rate-profile.exe –layers B1,B2,…,Bx SWF 10 B1 LENGTH SEEDx
```

where `B1,B2,…,Bx` are all EP modes rates starting from the lowest one (`B1`) up to the highest one (`Bx`) multiplied by 100, i.e. 100, 200, 300, 400 for EP mode 1,2,3,4.

B.4.5 Binaries used in characterization phase

B.4.5.1 Introduction

All binaries are compiled and tested under Win32 platforms. The following clauses document the origin and the compilation of the binaries.

B.4.5.2 Tools

Table B.7: Recommendation ITU-T G.191

Source	Recommendation ITU-T G.191 [6] S4-120344 "Filter masks for EVS testing"
URL	http://ftp.3gpp.org/tsg_sa/WG4_CODEC/TSGS4_68/Docs/S4-120344.zip
Version/Release	G.191
Description	Software tools for speech and audio coding standardization
Comments	G.191 filter tool patched with S4-120344 to enable support for HP50 and SHQ filter
Executables	`oper, astrip, concat, sv56demo, filter, scaldemo, actlev, eid-xor, gen-patt, mnrudemo, gen-rate-profile, eid-int`
Status	Available

Table B.8: AMR-NB Error Insertion

Source	Orange: S4-120998 [36]
URL	http://ftp.3gpp.org/tsg_sa/WG4_CODEC/TSGS4_70/Docs/S4-120998.zip
Version/Release	-
Description	Error insertion device for AMR-NB bit streams
Comments	
Executables	`eid-amr`
Status	Available

Table B.9: WAV - PCM Converter

Source	McGill University, Telecommunications & Signal Processing Laboratory, Audio File Programs and Routines[37]
URL	http://www-mmsp.ece.mcgill.ca/Documents/Downloads/AFsp/AFsp-v9r0.tar.gz
Version/Release	Release v9r0; Software version of CopyAudio: v6r0 2003-05-08
Description	CopyAudio tool from the Afsp
Comments	
Executables	`CopyAudio.exe`
Status	Available

Table B.10: SWB MNRU

Source	NTT: AHEVS-165[38]
URL	http://ftp.3gpp.org/tsg_sa/WG4_CODEC/Ad-hoc_EVS/Docs/AHEVS-165.zip
Version/Release	-
Description	Recommendation ITU-T P.50 [17] MNRU
Comments	
Executables	`p50mnru.exe`
Status	Available

Table B.11: Randomization Tool

Source	Fraunhofer IIS: S4-121078 [3]
URL	http://ftp.3gpp.org/tsg_sa/WG4_CODEC/TSGS4_70/Docs/S4-121078.zip
Version/Release	-
Description	Tool for providing all randomizations depending on a master seed
Comments	
Executables	`random.exe`
Status	Available

B.4.5.3 Codecs

Table B.12: AMR-NB

Source	3GPP TS 26.073 [9]: ANSI-C code for the Adaptive Multi Rate (AMR) speech codec
URL	-
Version/Release	ETSI TS 126 073 [9]
Description	AMR narrow band fix point encoder and decoder software
Comments	Compiled with VAD version 1 and 2
Executables	`amr_cod_vad1.exe, amr_cod_vad2.exe, amr_dec.exe`
Status	Available

Table B.13: G.711

Source	Recommendation ITU-T G.191 [6]
URL	-
Version/Release	Version 3.3 of 02.Feb. 2010
Description	G.711 codec
Comments	
Executables	`g711demo.exe, g711iplc.exe`
Status	Available

Table B.14: G.726

Source	Recommendation ITU-T G.191 [6]
URL	-
Version/Release	Version 1.4 of 03. Feb.2010
Description	G.726 codec
Comments	
Executables	`g726demo.exe`
Status	Available

Table B.15: AMR-WB

Source	3GPP TS 26.173: ANSI-C code for the Adaptive Multi-Rate - Wideband (AMR-WB) speech codec
URL	-
Version/Release	ETSI TS 126 173 [8]
Description	AMR-WB fixed point encoder and decoder software
Comments	
Executables	`amrwb_cod.exe, amrwb_dec.exe`
Status	Available

Table B.16: G.722

Source	Recommendation ITU-T G.191 [6]
URL	-
Version/Release	COPYRIGHT CNET LANNION A TSS/CMC Date 24/Aug/90 COPYRIGHT Ericsson AB. Date 22/May/06 COPYRIGHT France Telecom R&D Date 23/Aug/06
Description	G.722 encoder and decoder software
Comments	
Executables	`encg722.exe, decg722.exe`
Status	Available

Table B.17: EVS

Source	3GPP 26.441
URL	-
Version/Release	ETSI TS 126 441 [18]
Description	EVS fix-point encoder and decoder software
Comments	
Executables	`EVS_cod.exe, EVS_dec.exe`
Status	Available

Table B.18: OPUS

Source	Xiph.org [40]
URL	https://ftp.osuosl.org/pub/xiph/releases/opus/opus-1.3.tar.gz
Version/Release	1.3
Description	Opus codec software compiled in fix-point arithmetic
Comments	Option to change delay alignment; added EPF support
Executables	`opus_demo_1_3_0`
Status	Available

Table B.19: LC3plus

Source	ETSI
URL	-
Version/Release	ETSI TS 103 634 [5]
Description	LC3plus codec software compiled in fix-point arithmetic
Comments	
Executables	`LC3plus`
Status	Available

Annex C (normative): Results of the subjective characterization test for the ETSI LC3plus codec

C.1 Introduction

This annex presents a report on subjective testing performed. MESAQIN.com performed the experiment according to the procedures and test plan specified in the main part of the present document. No deviations from or exceptions to the listening test procedures and specifications described in the test plan have been observed or experienced.

The raw voting data along with their statistical processing (MOS calculation and conditional STD and CI95) are contained in subfolder "Raw voting data along with their statistical processing" in archive ts_103624v010301p0.zip which accompanies the present document.

C.2 Summary of tests conducted and results obtained

C.2.1 Experiments

MESAQIN.com performed 8 different experiments (i.e. 7 experiments in English, one experiment in Mandarin) within the project. The experiment name, methodology (as defined in Recommendation ITU-T P.800 [1]) and language for each experiment are listed in Table C.1.

Table C.1: Allocation of experiments and languages

Exp.	Type	Methodology	Language
1	NB clean	ACR	EN
2	NB error	ACR	EN
3	WB clean	ACR	EN
4	WB error	ACR	EN
5	SWB clean	DCR	EN
6	SWB error	ACR	EN
7a	FB_MB_fullscale	ACR	EN
7b	FB_MB_fullscale	ACR	MAN

C.2.2 Speech material

MESAQIN.com used processed speech material delivered by Fraunhofer Institute in 48-kHz sampled with 16-bit resolution uncompressed wav files. The processed speech material for all tests conformed to restrictions indicated in "Practical procedures for subjective testing", ITU-T Handbook, 2011 [14]. The list of conditions is available in clause 6 of the present document. No sample name blinding or anonymization has been applied as it was principally not needed.

C.2.3 Listening Environment

The tests were performed in an acoustically treated critical listening room in MESAQIN.com Prague laboratories that conforms to the requirements of Recommendation ITU-T P.800 [1] in full. Its background noise during the tests was below than 30 dB SPL(A) with no peaks in audible acoustic frequency range and its reverberation time (60 dB) is 185 ms.

Listening equipment conformed to specified requirements (Recommendation ITU-T P.800 [1] and clause 5 of the present document) in full. Diffuse-field equalized Sennheiser™ headphones HD-650 have been used for all experiments. All headphones used have been calibrated and verified before and after performed experiments as required by Recommendation ITU-T P.800 [1] and [14]. A professional digital voting device has been used to collect the votes.

C.2.4 Test instructions and Presentation order

The instructions for subjects in each of the languages tested by the MESAQIN.com are available in clause B.3. The instructions to subjects were available in written during the entire test sessions. During the training session of each test, the instructions were verbally explained and if needed briefly discussed by a dedicated expert person that was able to answer questions from the subjects in their native language.

Test duration never exceeded 1,5 hours per listening panel. Test duration comprised of up to 50 % of actual listening time and test overhead including administration, initial briefing, preliminaries, and breaks.

The playout loudness calibration used: 73 dB SPL equals to -26 dBov.

For each speech sample, the MOS-LQS following ACR methodology for Experiments 1 to 4 and 6 to 7 have been assessed. In Experiment 5, the MOS-LQS following DCR methodology has been assessed.

C.2.5 Test Subjects

For each listening test, only native speakers of the tested language were used. Only normal-hearing subjects have been used for the experiments. Their hearing normality has been verified by subject self-assessment questionnaire during the subject hiring phase and verified prior the session by expert test. Each experiment required 24 listeners (6 panels of 4 listeners each). The age and gender information for the set of subjects used in each tested language tested by the MESAQIN.com are provided in clause C.2.6.

C.2.6 Age and gender information

Table C.2 provides age and gender information across all subjects within each language tested by MESAQIN.com.

Table C.2: Age and gender information

Language	#females/#males	Mean Age (years)	Age StdDev (years)
English	1,00	33,27	11,44
Chinese	1,00	30,67	10,45

C.2.7 Raw data delivery and statistical analysis

All raw voting data is available in "exp1.xlsx"-"exp7b.xlsx" in subfolder "Raw voting data along with their statistical processing" in archive ts_103624v010301p0.zip which accompanies the present document. They contain Means, Standard Deviations and CI95 of MOS-LQ for every condition within each test. Also, as required by in clause 7.1 of the present document, each condition in clauses 6.1 to 6.4 of the present document, labelled as requirement has been compared to the corresponding CuT condition using a Student's Dependent Groups t-test (single-sided at 95 % confidence level) on the subjective scores. This comparison has proven that the CuT meets or exceeds the requirements in all examined cases. Additionally, the same rule has been applied to the performance objectives, whereas the comparison between CuT and performance objective have only informative character. Also here the results shown that CuT meets or exceeds the performance objectives in all examined cases. The detailed results including all statistical evaluations are available in the digital attachment, separately for each experiment.

C.2.8 Discussion of any problems encountered during testing and the solution used to address the problem

During the tests no problems have been encountered.

Annex D (normative): Objective assessment of the subjective test results

D.1 Recommendation ITU-T P.863 software version

The software version of Recommendation ITU-T P.863 [2] used for the assessment was 3.16-2151, 64bit, Linux of the "POLQA OEM Library" by OPTICOM.

The default commandline was:

PolqaOemDemo64 -Ref *aeegysz.c01*.wav -Test *aeegysz.cnn*.wav -LC *LC* -Version 3 -DisableLevelAlignment -DisableSRConv

Where:

- *aeegysz.cnn* is defined in clause B.4.3.2.2 for experiment 1..7b.
- *LC* is the listening condition of P.863, i.e. NB, SWB or FB, see also Table D.1.

Table D.1: Listening conditions for P.863 according to experiment

Experiment number	Experiment label (Excel sheet)	Max. audio bandwidth	Channel conditions	Rating Scale	Listening Condition (*LC*)
1	NB clean	4 000 Hz	No error	ACR	NB
2	NB error	4 000 Hz	Bit error & packet loss	ACR	NB
3	WB clean	8 000 Hz	No error	ACR	FB
4	WB error	8 000 Hz	Bit error & packet loss	ACR	FB
5	SWB clean	16 000 Hz	No error	DCR	FB
6	SWB error	16 000 Hz	Bit error & packet loss	ACR	FB
7a	FB MB_fullscale short	20 000 Hz	No error & packet loss & bit error	ACR	FB
7b	FB MB_fullscale short	20 000 Hz	No error & packet loss & bit error	ACR	FB

The raw scores (MOS-LQO) of all items are available in "polqa_raw.xlsx" in subfolder "POLQA assessment" in archive ts_103624v010301p0.zip which accompanies the present document.

D.2 Statistical tests MOS-LQS vs MOS-LQO

D.2.1 Overview

All statistical tests which have been conducted for the MOS-LQS, have been repeated for MOS-LQO. The results are available in the file "T-test LQO vs LQS.xlsx" in subfolder "POLQA assessment" in archive ts_103624v010301p0.zip which accompanies the present document.

Using MOS-LQO as input for statistical tests, LC3plus still passes all statistical tests. For some statistical comparison, LC3plus pass with better than (BT) reference condition, for other with not worse than (NWT) the reference condition. Table D.2 provides a summary about BT and NWT passes for MOS-LQS and MOS-LQO data.

Table D.2: Overview of statistical results MOS-LQS vs. MOS-LQO

	MOS-LQS			MOS-LQO		
	NWT	BT	FAIL	NWT	BT	FAIL
Absolute	71	91	0	30	132	0
Relative	0,44	0,56	0,00	0,19	0,81	0,00

D.2.2 MOS-LQS vs MOS-LQO

D.2.2.1 Linear Regression

In order to remove linear offset in the subjective scores, for each experiment a linear regression model has been conducted and the MOS-LQO scores have been mapped. This aligns the MOS-LQS and MOS-LQO score to a similar scaling with respect to the P.800 scale and allow a comparison of the absolute scores. MNRUs and DIRECT conditions have been removed for this step. Figure D.1 shows the linear transformation of all experiments.

MOS-LQO$_E$ denotes the objective scores mapped to the common scale. Due to the low number of subjective votes per item (4 votes), all further analysis is per condition where the MOS-LQS is based on 96 subjective votes and the MOS-LQO is based on 24 objective scores.

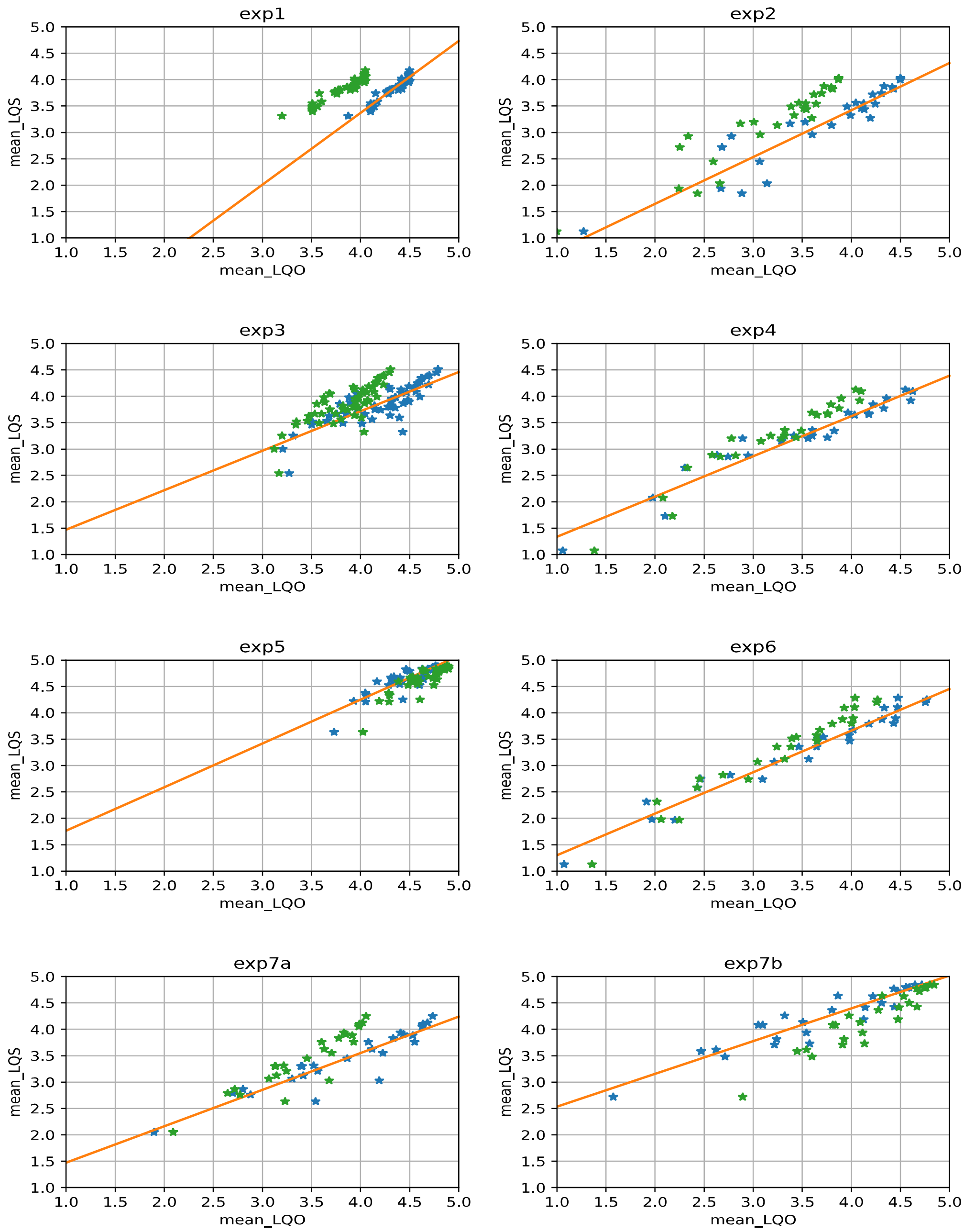

Figure D.1: Linear Regression MOS-LQS and MOS-LQE for all experiments. Blue: Original relation, Orange: linear regression mapping, Green: Mapped MOS-LQO$_E$ data

As can be observed, all experiments show a clear linear behaviour for the MOS-LQS vs. MOS-LQO$_E$ mapping and are therefore, used for further analysis. A further statistical analysis is given in Table D.3 where the metrices rmse*, maximum absolute error* (max_abs*) and Pearson correlation coefficient are provided for raw and mapped data according to [35]. Due to the linear mapping, the Pearson correlation coefficient is identical for raw and mapped data.

Table D.3: Statistical comparison of MOS-LQS and MOS-LQO

Exp	rmse*	rmse* (mapped)	max_abs*	max_abs* (mapped)	Pearson correlation coefficient
exp1	0.313553	0.000000	0.467932	0.000000	0.956596
exp2	0.394923	0.113191	0.881123	0.408811	0.931174
exp3	0.213693	0.071716	0.813683	0.439334	0.808003
exp4	0.165271	0.061156	0.480925	0.237171	0.965796
exp5	0.059737	0.030067	0.179028	0.134176	0.866480
exp6	0.155285	0.030458	0.399191	0.137404	0.979151
exp7a	0.303937	0.087513	0.806100	0.342792	0.928849
exp7b	0.360378	0.033494	0.890827	0.110372	0.940671

D.2.2.2 Outlier ratio

Conditions are considered as outliers if the 95 % confidence interval of the MOS-LQS does not intersect with the 95 % confidence interval of MOS-LQO$_E$. 29 out of 225 conditions are considered as outliers, leading to a ration of 12,8 %. All data is available in "ScoreMapping.xlsx" in subfolder "POLQA assessment" in archive ts_103624v010301p0.zip which accompanies the present document.

Table D.4: Overall outlier cases

Codec	Error prone channel	Number	Comments	LQO > LQS	Avg. Delta MOS
G.722	no	4	G.722 @ 64 kbps	4	0,430
G.722	yes	2	G.722 @ 64 kbps	2	0,270
LC3plus	no	4	LC3plus clean channel	0	-0,179
LC3plus	no	2	LC3plus WB @ 24 kbps	2	0,467
LC3plus	yes	10	8x DECT error pattern; 2x random/burst PLR	8	0,261
G.711	Yes	1	-	-	
G.726	No	0	-	-	
G.726	Yes	1	-	-	
Mixed tandeming	No	2	-	-	
Opus	No	2	-	-	
EVS-SWB	No	1	-	-	
SUM		29			

Table D.4 shows, that over all outlier cases:

- G.722 for clean channels was overestimated by POLQA with an average offset of 0.43 MOS
- G.722 for error prone channels was overestimated by POLQA with an average offset of 0.27 MOS
- LC3plus for clean channels was slightly underestimated by POLQA with an average offset of -0.18 MOS
  - As special case, LC3plus WB @ 24 kbit/s was overestimated by POLQA with an average offset of 0.47 MOS
- LC3plus for error prone channels was sometimes overestimated with an average offset of 0.26 MOS

Regarding LC3plus, 16 valid outliers out of 105 conditions have been found which leads to an outlier ratio of 15,2 % which is very close to the overall outlier ratio. In general, LC3plus assessment using Recommendation ITU-T P.863 [2] is possible.

Annex E (normative): Equipment Impairment Factors (Ie) for the different bandwidth modes of the codec

E.1 Testplan

E.1.1 Overview

This clause contains the subjective and objective test plan for the Equipment Impairment Factor estimation of the ETSI LC3plus codec. The list of conditions is identical for objective and subjective evaluation.

The list of experiments and conditions for all determinations of the Ie factors for the new ETSI codec is available in "P800_ETSI_TS_Ie_factor.xls" in subfolder "Ie_derivation" in archive ts_103624v010301p0.zip which accompanies the present document. The conditions are selected in a way to be conform with the corresponding Recommendation ITU-T and to cover a wide range of the MOS scale.

E.1.2 Experiments and conditions

The test plan consists of 9 experiments. For each bandwidth, three experiments are conducted in order to evaluated the *Ie* factors of LC3plus in the context of error-free transmission, transcoding (additivity check) and transmission errors. Table E.1 lists the required numbers of conditions for each experiment.

Table E.1: Overview of number of conditions per experiments

Bandwidth/Purpose	*Ie* in error-free (A)	Additivity check (B)	*Ie* for transmission errors I
NB	30	53	45
WB	28	61	53
FB	36	47	44

E.1.3 Speech material

Each source speech file contains a sentence pair and lasts exactly 8 s. Each sentence is centred inside a 4 s time window. Leading and trailing silence parts are longer than 0,5 s. The sentences are simple meaningful sentences, similar to those described in annex B.1.4 of Recommendation ITU-T P.800 [1].

The test language will be American English. Each experiment sample database consists of a number of samples to be calculated by: nTalkers(4) x nSamples(6) x nConditions (see Table E.1). The talkers consists of 2 male and 2 female talkers.

The test samples are processed according to clause E.1.6.

E.1.4 Considerations for subjective evaluation

- 24 subjects for each experiment, minimum 4 (2M+2F) talkers, minimum 6 samples per talker, minimum 4 votes per sample, minimum 96 votes per condition, each panel with an independent randomization.
- Preliminary or training conditions are selected from the material available by the listening lab.
- Randomizations are constructed under partially-balanced/randomized blocks experimental design described in the ITU-T Handbook [14].
- Test duration: maximum 2 hours per listening panel. Test duration comprises 50 % of actual listening time and 50 % test overhead including administration, initial briefing, preliminaries and breaks.

- Listening level -21 dB Pa (73 dB SPL) equals to -26 dBov.
- Files are played back with diffuse-field equalized headphones and diotic presentation.
- All nine (9) experiments are performed using ACR methodology of Recommendation ITU-T P.800 [1]
- Listening environment: All tests are performed in an acoustically treated listening environment that conforms to the requirements of [1].
- Listener instructions as listed below are used.

> In this experiment, you will be listening to short samples via headphones, and giving your opinion of the speech you hear.
>
> Follow the instructions on the touchscreen in front of you, and listening to each sample, press the appropriate button to indicate your opinion on the following scale.
>
> **WHAT WAS THE QUALITY OF THE SAMPLE YOU HAVE JUST HEARD?**
>
> 5 Excellent
>
> 4 Good
>
> 3 Fair
>
> 2 Poor
>
> 1 Bad

E.1.5 Considerations for objective evaluation

- The objective data is generated according to Recommendation ITU-T P.863 [2].
- The software used is version 3.16-2151, 64bit, Linux of the "Advanced OEM reference code by OPTICOM".
- For NB experiments, the listening condition is NB, for other experiments FB as listening condition is used.

E.1.6 Additional Processing functionalities

E.1.6.1 GSM Enhanced Full Rate

Source	3GPP TS 46.053 [32]: ANSI-C code for the GSM Enhanced Full Rate (EFR) speech codec
URL	https://www.3gpp.org/ftp/Specs/archive/46_series/46.053/46053-g00.zip
Version/Release	V16.0.0 (2020-07)
Description	GSM EFR encoder and decoder software
Comments	
Executables	coder, ed_iface, decoder
Status	Available

Command lines	`coder input bitstream.cod`
	`ed_iface bitstream.cod bitstream.dec`
	`decoder bitstream.dec output`

E.1.6.2 GSM Full Rate

Source	Recommendation ITU-T G.191 [6]
URL	https://www.itu.int/rec/T-REC-G.191-201901-S/en
Version/Release	1.2 of 02.Feb.2010
Description	GSM FR encoder and decoder software
Comments	Folder: T-REC-G.191-201901-S!!SOFT-ZST-E/Software/src/rpeltp
Executables	rpedemo
Status	Superseded (this is the latest published version)
Command lines	rpedemo -enc input bitstream.gsmfr rpedemo -dec bitstream.gsmfr output

E.1.6.3 GSM Half Rate

Source	3GPP TS 46.006 [33]: ANSI-C code for the GSM half rate speech codec
URL	https://www.3gpp.org/ftp/Specs/archive/46_series/46.006/46006-h00.zip
Version/Release	V17.0.0 (2022-04)
Description	GSM HR encoder and decoder software
Comments	change old typedef #define LW_SIGN (long)0x80000000 to #define LW_SIGN (unsigned int)0x80000000 change for loop in reid to while(1) to avoid limiting of frame number
Executables	gsm_hr, reid
Status	Available
Command lines	gsm_hr enc input bitstream.cod reid bitstream.cod bitstream.dec gsm_hr dec bitstream.dec output

E.1.6.4 G.726

Source	Recommendation ITU-T G.191 [6]
URL	G.191 : Software tools for speech and audio coding standardization (itu.int)
Version/Release	Version 1.4 of 03.Feb.2010
Description	G.726 encoder and decoder software
Comments	Folder: T-REC-G.191-201901-S!!SOFT-ZST-E/Software/src/g726
Executables	g726demo
Status	In force
Command lines	g726demo A load <kbps> input bstream.g192 <BlockSize> g726demo A adlo <kbps> bstream.g192 output <BlockSize>

E.1.6.5 G.729

Source	Recommendation ITU-T G.729 [34]: Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)
URL	https://www.itu.int/rec/T-REC-G.729-201206-I/en
Version/Release	Version 3.3 (Release 2, November 2006)
Description	G.729 encoder and decoder software
Comments	
Executables	coder, decoder
Status	In force
Command lines	coder input bitstream decoder bitstream output

E.1.6.6 G.729 Annex E

Source	Recommendation ITU-T G.729 [34], Annex E
URL	https://www.itu.int/rec/T-REC-G.729-201206-I/en
Version/Release	Version 1.3 (Release 2, November 2006)
Description	G.729 encoder and decoder software
Comments	Folder: T-REC-G.729-201206-I!!SOFT-ZST-E/Software/G729_Release3/g729/c_code
Executables	codere, decodere
Status	In force
Command lines	`codere input bitstream <0|1>` `decodere bitstream output`

E.1.6.7 G.729.1

Source	Recommendation ITU-T G.729.1 [31]
URL	https://www.itu.int/rec/T-REC-G.729.1-200908-I!Cor1/en
Version/Release	Version 1-5 - August 29, 2009
Description	G.726 encoder and decoder software
Comments	Folder: T-REC-G.729-201206-I!!SOFT-ZST-E/Software/G729_Release3/g729/c_code
Executables	/Software/G.729.1_V1.5_Part1_fixed_point_MB/Release/encoder.exe /Software/G.729.1_V1.5_Part1_fixed_point_MB/Release/decoder.exe
Status	In force
Command lines	encoder.exe -r <bitrate> input bitstream decoder bitstream output

E.2 Methodology for the determination of Ie factors for the LC3plus codec

E.2.1 Narrowband (NB) Ie factors

E.2.1.1 Introduction

The conditions for each specific part of the experiment for objective and subjective determination of Ie,NB are listed in the following tabs of the excel sheet "P800_ETSI_TS_Ie_factor.xls" in the subfolder "Ie_derivation" in archive ts_103624v010301p0.zip which accompanies the present document.

- Part A - Clean speech: tab "NB_clean_A"
- Part B - Tandeming: tab "NB_clean_B"
- Part C - Transmission errors: tab "NB_error_C"

E.2.1.2 Objective determination

This methodology follows Recommendation ITU-T P.834 [21].

E.2.1.3 Subjective determination

This methodology follows Recommendation ITU-T P.833 [20].

E.2.2 Wideband (WB) Ie factors

E.2.2.1 Introduction

The conditions for each specific part of the experiment for objective and subjective determination of Ie,WB are listed in the following tabs of the excel sheet "P800_ETSI_TS_Ie_factor.xls" in the subfolder "Ie_derivation" in archive ts_103624v010301p0.zip which accompanies the present document.

- Part A - Clean speech: tab "´WB_clean_A"
- Part B - Tandeming: tab "WB_clean_B"
- Part C - Transmission errors: tab "WB_error_C"

E.2.2.2 Objective determination

This methodology follows Recommendation ITU-T P.834.1 [25].

E.2.2.3 Subjective determination

This methodology follows Recommendation ITU-T P.833.1 [24].

E.2.3 New Methodology for Fullband (FB) Ie factors

E.2.3.1 Introduction

The conditions for each specific part of the experiment for objective and subjective determination of Ie,FB are listed in the following tabs of the excel sheet "P800_ETSI_TS_Ie_factor.xls" in the subfolder "Ie_derivation" in archive ts_103624v010301p0.zip which accompanies the present document.

- Part A - Clean speech: tab "FB_clean_A"
- Part B - Tandeming: tab "FB_clean_B"
- Part C - Transmission errors: tab "FB_error_C"

Recommendation ITU-T G.107.2 [28] describes the FB version of a computational model, known as the E-model. This model makes use of impairment factors which can be derived for the NB and WB use case according to Recommendations ITU-T P.833 [20] and P.834 [21].

This clause specifies a new methodology for how fullband impairment factors (Ie,FB) and packet loss robustness factors (Bpl,FB) can be derived for a speech codec from the results of listening-only tests and using instrumental models. Objective and subjective determination differ only in the method of obtaining MOS values, the rest of the procedure is the same for both types. For the objective determination of FB Ie values the MOS-LQO values are derived by objective assessment with Recommendation ITU-T P.863 [2] in FB mode. For the subjective determination of FB Ie values the MOS-LQS values are derived by subjective tests in compliance with Recommendation ITU-T P.800 [1]. Hereafter the score will be referred to as MOS to be applicable for both methods.

As experimental evidence suggests that SWB and FB conditions are perceptually equivalent when being evaluated on a 5-point ACR scale according to [1], both bandwidths are subsumed under *Ie,FB* and the reference conditions below are presented in 32 kHz sampling rate.

E.2.3.2 Addition for subjective determination

The listening-only test from which *Ie,FB* values are to be derived should fulfil the general requirements for listening-only tests given in [1] and [31] as well as the requirements set in [20]. However, in contrast to [20], the test should be carried out in a mixed-band mode (SWB and WB) and with diotic headphone presentation. Furthermore, it is advisable to add MNRU reference conditions as anchors for WB and SWB only in the listening test.

E.2.3.3 Reference conditions for SWB speech codecs without transmission errors

When fullband equipment impairment factors for codecs disregarding transmission errors are determined the set of 19 reference codec conditions given in Table E.2 should be included in the test conditions. This list has been chosen from the codecs for which values are already defined in Appendix IV and V to Recommendation ITU-T G.113 [26], [29] and they have been selected to cover the whole range of *Ie,FB* values.

Table E.2: Reference conditions for super wideband speech codecs without transmission errors

No.	Abbreviation	Codec Type	Band-width	Reference	Operating rate (kbit/s)	Ie,FB value
1	SWB Clean	Linear PCM, 16 bit	SWB	-	-	0
2	EVS-SWB@48	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	48	10.2
3	EVS-SWB@32	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	32	8.7
4	EVS-SWB@24.4	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	24.4	7.2
5	EVS-SWB@16.4	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	16.4	10.8
6	EVS-SWB@13.2	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	13.2	17.1
7	EVS-SWB@9.6	ACELP/MDCT	SWB	ETSI TS 126 445 [30]	9.6	22.7
8	WB Clean	Linear PCM, 16 bit	WB	-	-	19
9	G.722@64	ADPCM	WB	ITU-T Rec. G.722 [3]	64	24
10	G.729.1@32	CELP with TDBWE and TDAC	WB	ITU-T Rec. G.729.1 [31]	32	24
11	G.722.2@23.05	CELP	WB	ITU-T Rec. G.722.2 [19]	23.05	26
12	G.722.2@23.85	CELP	WB	ITU-T Rec. G.722.2 [19]	23.85	29
13	G.722@56	ADPCM	WB	ITU-T Rec. G.722 [3]	56	29
14	G.729.1@24	CELP with TDBWE and TDAC	WB	ITU-T Rec. G.729.1 [31]	24	35
15	G.722.2@15.85	CELP	WB	ITU-T Rec. G.722.2 [19]	15.85	38
16	G.722.2@12.65	CELP	WB	ITU-T Rec. G.722.2 [19]	12.65	39
17	G.722.2@8.85	CELP	WB	ITU-T Rec. G.722.2 [19]	8.85	45
18	G.722@48	ADPCM	WB	ITU-T Rec. G.722 [3]	48	60
19	G.722.2@6.6	CELP	WB	ITU-T Rec. G.722.2 [19]	6.6	60

It is important to check the additivity of the newly derived super wideband equipment impairment factor in the framework of other equipment impairment factor values defined so far. If such an additivity check is not performed, the property of a simple summation of equipment impairment factors in order to cater for codec tandems should not be regarded as valid. Table E.3 gives a minimum number of additional reference conditions which should, in any case, be included in the test set to allow for a rough additivity check. The chosen number of codecs is limited by the available Ie,FB values for SWB Codecs in [29].

Table E.3: Reference conditions for the additivity check in tandem operation

No.	Tandem operation	Operating rate (kbit/s)	Ie,FB value
20	EVS-SWB@48*(new codec)	48	10,2 + *Ie,FB*(new codec)
21	EVS-SWB@32*(new codec)	32	8,7 + *Ie,FB* (new codec)
22	EVS-SWB@24.4*(new codec)	24.4	7,2 + *Ie,FB* (new codec)
23	EVS-SWB@16.4*(new codec)	16.4	10,8 + *Ie,FB* (new codec)
24	EVS-SWB@13.2*(new codec)	13.2	17,1 + *Ie,FB* (new codec)
25	EVS-SWB@9.6*(new codec)	9.6	22,7 + *Ie,FB* (new codec)
26	(new codec) *EVS-SWB@48	48	*Ie,FB* (new codec) + 10,2
27	(new codec) *EVS-SWB@32	32	*Ie,FB* (new codec) + 8,7
28	(new codec) *EVS-SWB@24.4	24.4	*Ie,FB* (new codec) + 7,2
29	(new codec) *EVS-SWB@16.4	16.4	*Ie,FB* (new codec) + 10,8
30	(new codec) *EVS-SWB@13.2	13.2	*Ie,FB* (new codec) + 17,1
31	(new codec) *EVS-SWB@9.6	9.6	*Ie,FB* (new codec) + 22,7
NOTE: A*B designates asynchronous tandeming of codecs A and B, B followed by A.			

E.2.3.4 Reference conditions for SWB codecs with transmission errors

When fullband equipment impairment factors for codecs under the effects of transmission errors are determined, the same reference conditions as given in Table E.2 should be applied. In addition to these conditions, supplementary conditions may be included in the test from the codecs listed in Appendix V of G.113 Amendment 2 [26].

For codecs with transmission errors, reference conditions allowing for an additivity check should be included in the test set as well. Unfortunately, when different error rates are to be tested, this additivity check can lead to an experimental size that is barely manageable. For this reason, no mandatory list of reference conditions is given here.

E.2.3.5 Test method

The test method generally follows the recommendations given in Recommendations ITU-T P.833 [20] and P.834 [21]. Table E.4 summarizes the test conditions to be included in the different parts of the experiment.

The test conditions do not need to be included into one single test or test session; in fact, the number of test conditions in one session should be limited to avoid listener fatigue, following the recommendations given in Recommendation ITU-T P.800 [1].

Table E.4: Overview of test conditions for the different parts of the experiment

Part	Purpose	Test conditions	Mandatory/ Optional	Min. overall Σ test cond.
A	Determination of *Ie,FB* for the new codec in error-free conditions	Reference conditions 1-19 from Table E.2	Mandatory	19
		New codec in single operation, at 3 speech input levels	Mandatory	
		Additional super wideband codec references	Optional	
B	Additivity check	Reference conditions 20-31 from Table E.3	Mandatory	14
		New codec alone in double and triple self-tandem operation	Mandatory	
		New codec in double and triple tandem operation with other codecs	Optional	
C	Determination of *Ie,FB* for the new codec in transmission error conditions	Min. n = 10 references according to [26] (e.g. 4 codecs at 4 different error rates)	Mandatory	$n + m$
		New codec in single operation in different transmission error conditions (m conditions)	Mandatory	
		Additional references according to [26]	Optional	

E.2.3.6 Derivation of FB equipment impairment factors

E.2.3.6.1 Introduction

The methodology for deriving FB equipment impairment factors follows mostly P.833 [20] and P.834 [21], however with some modifications of the calculations. It consists of three to five steps, depending on whether transmission errors are considered or not:

- Step 1: Scale transformation of the test data.
- Step 2: Derivation of a stable *Ie,FB* value for the codec under test, in single codec operation without transmission errors, via a linear interpolation of the test results.
- Step 3: Additivity check.

If transmission errors are under investigation, the following additional steps are to be taken:

- Step 4: Derivation of stable *Ie,FB* values for different transmission error conditions in single codec operation.
- Step 5: Additivity check (optional).

E.2.3.6.2 Detailed process

E.2.3.6.2.1 Scale transformation (step 1)

Mean Opinion Scores (MOSs) are determined for all test conditions, including the references listed in Table E.4. These MOS results shall be linearly transformed to the MOS(NB) scale in case that the maximum MOS_{max} value of the test is higher than 4.5. This normalization is applied for all values according to:

$$MOS_n = \frac{MOS-1}{MOS_{max}} \cdot 3.5 + 1 \qquad (1)$$

The normed MOS values shall be transformed to the extended R-scale. For this purpose, they are first transformed to the non-extended, narrow-band R_{NB}-scale (range [0;100], subscript *NB*) by solving the following equations numerically for R_{NB}. This corresponds to an inversion of the relationship between MOS and R_{NB} given in the E-model:

$for\ MOS_n = 1.0$: $R_{NB} = 0$

$for\ 1.0 < MOS_n < 4.5$: $MOS_n = 1 + 0.035 \cdot R_{NB} + R_{NB} \cdot (R_{NB} - 60) \cdot (100 - R_{NB}) \cdot 7 \cdot 10^{-6}$

$for\ MOS_n \geq 4.5$: $R_{NB} = 100$

These values have to be transformed to the extended FB *R*-scale (range [0;148]). For this purpose, a linear expansion of the obtained R_{NB} values to the super wideband R_{FB} values (subscript *FB*) is carried out:

$$R_{FB} = R_{NB} \cdot 1.48 \qquad (2)$$

From the resulting values for R_{FB}, the corresponding raw *Ie,FB,obs* values (subscript *obs* for "observed") can be calculated by defining the R_{FB}-value for reference condition No. 1 (see Table E.2) as an anchor, thus:

Ie,FB,obs = R_{FB} (condition No.1) – R_{FB} (test condition) (3)

This equation results in *Ie,FB,obs* for the reference condition No. 1 always set to 0. The outcome of step 1 is an *Ie,FB,obs* value for each test condition. It reflects the specific test condition, and it is not necessarily consistent with full band equipment impairment factors defined in Appendix IV to [29].

E.2.3.6.2.2 Linear interpolation of the test results (step 2)

For all 19 reference conditions of Table E.2, as well as possibly for all supplementary reference conditions involving only codecs for which *Ie,FB* values have already been defined, pairs of defined equipment impairment factors *Ie,FB,def* and observed values *Ie,FB,obs* are now available. These pairs can be represented as a scatter plot. A linear interpolation using a straight line:

$$\mathrm{Ie,FB,obs} = \mathrm{a}\cdot\mathrm{Ie,FB,def} + \mathrm{b} \qquad (4)$$

can now be made. The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}\left(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j\right)*\left(Ie,FB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j\right)}{\sum_{i=1}^{n}\left(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j\right)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j - a\cdot\frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j \qquad (5)$$

From this approximation, a stable equipment impairment factor value for the codec under test (*Ie,FB* = *Ie,FB,def*) can be derived.

In rare cases, the linear transformation may result in a negative *Ie,FB* value for the codec under investigation. This might occur if the related subjective ratings are close to or better than the one for reference condition No. 1 of Table E.2. In this case, *Ie,FB* should be set to zero instead.

E.2.3.6.2.3 Additivity check (step 3)

The equipment impairment factor derived in step 2 does not necessarily satisfy the additivity property of *Ie,FB*. This has to be checked for double and triple self-tandeming of the new codec alone and mixed tandems with codecs for which *Ie,FB* values have already been defined. The procedure is the same as described in Recommendation ITU-T P.833 [20]. If more than 4 out of 14 tandem conditions (2 pure tandems of the codec under investigation and 12 reference tandem conditions Nos 20-31, see Table E.3) show major deviations from the interpolation line, the additivity property should not be regarded as having been satisfied. In this case, the fullband equipment impairment factor derived from the experiment will not properly represent the degradations occurring in tandem operations of the new codec.

E.2.3.6.2.4 Derivation of *Ie,FB* values for transmission error conditions (step 4)

For conditions with transmission errors the effective equipment impairment factor Ie,eff,FB shall be determined. Ie,eff,FB is derived using the codec-specific value for the equipment impairment factor at zero packet-loss Ie,FB and the packet-loss robustness factor Bpl. With the packet-loss probability Ppl, Ie,eff,FB is calculated using the equation from Recommendation ITU-T G.107.2 [28]:

$$Ie,eff,FB = Ie,FB + (132 - Ie,FB)\cdot\frac{Ppl}{Ppl+Bpl}$$

Corresponding values for Ie,FB and Bpl can be found in Appendix V of Recommendation ITU-T G.113 [26].

For all *n* reference conditions with *Ie,eff,FB* values under transmission error conditions pairs of observed equipment impairment factors *Ie,FB,obs* and defined ones *Ie,eff,FB*, exist. These pairs can be added to the scatter plot derived in step 2.

Any large deviations may indicate that the rating behaviour changed for the error conditions in comparison to the error-free case; the resulting *Ie,FB* values for codecs under transmission errors should then be considered with care.

Graphs or tables of *Ie,FB* values for transmission error conditions can be used for a plausibility check. Goodness of fit measure like coefficient of determination or maximum square error are also helpful to investigate if the results are reasonable. The minimum consistency to be reached is to have non-decreasing fullband equipment impairment factor values for increasing transmission error rates. If major inconsistencies are detected, the experiment should be rerun, this time including more speech input levels and/or speakers to base a decision on.

E.2.3.6.2.5 Additivity check (step 5)

A similar additivity check as in step 3 can be carried out on the fullband equipment impairment factor values derived in step 4, using all the available tandem conditions of the codec under investigation with transmission errors introduced and other codecs for which impairment factors have already been derived. As the number of test conditions may become very high in case of transmission errors, this step is optional.

E.2.3.6.2.6 Derivation of Packet loss robustness factor (step 6)

The packet loss robustness factor *Bpl* for the codec under investigation can be calculated by identifying the *Bpl* that fits the *Ie,eff,FB* curve the best. The optimal value can be determined by calculating the least square error for *Ie,FB,obs(Ppl)* and *Ie,eff,FB(Ppl).*

E.3 Results of the Ie factor assessment

E.3.1 Narrowband (NB) Ie factors

E.3.1.1 Objective assessment

E.3.1.1.1 Introduction

This clause describes the calculation procedure of the equipment impairment factor *Ie,NB* and the packet loss robustness factor *Bpl* for LC3plus. It is determined according to Recommendation ITU-T P.834 [21]. MOS values are obtained by evaluating coded 8 kHz speech signals with Recommendation ITU-T P.863 [2].

E.3.1.1.2 Determination of Ie for LC3plus in error-free conditions

E.3.1.1.2.1 Measuring MOS

Figure E.1 shows the mean MOS of the 14 reference conditions (see Recommendation ITU-T P.834 [21], table 1) and four LC3plus configurations under test with -16, -26 and -36 dBov and bitrates of 16, 20, 24 and 32 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

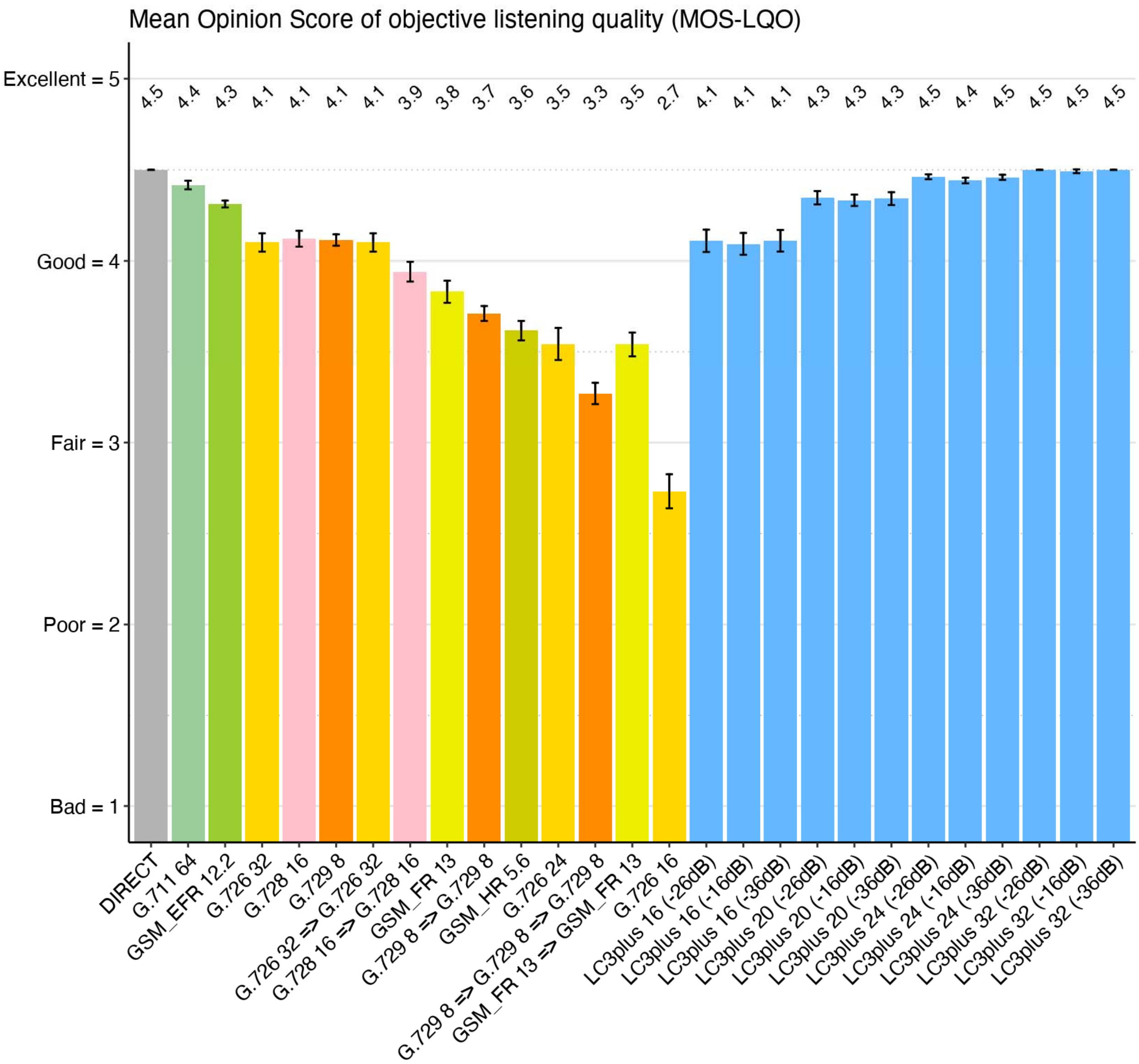

Figure E.1: Recommendation ITU-T P.863 results for NB clean speech signals. Mean scores and 95 % confidence intervals

E.3.1.1.2.2 Scale transformation

All Recommendation ITU-T P.863 test results are transformed from the MOS scale to the scale of the equipment impairment factor *Ie*,*NB*. In a first step R is determined by solving the following equations:

$$\begin{array}{ll} \text{for } MOS = 1.0: & R = 0 \\ \text{for } 1.0 < MOS < 4.5: & MOS = 1 + 0.035 \cdot R + R \cdot (R - 60) \cdot (100 - R) \cdot 7 \cdot 10^{-6} \\ \text{for } MOS \geq 4.5: & R = 100 \end{array}$$

The relation between MOS and R-value is presented graphically in Figure E.2 and shows that differences on the outer end of the MOS scale are weighted stronger on the R scale.

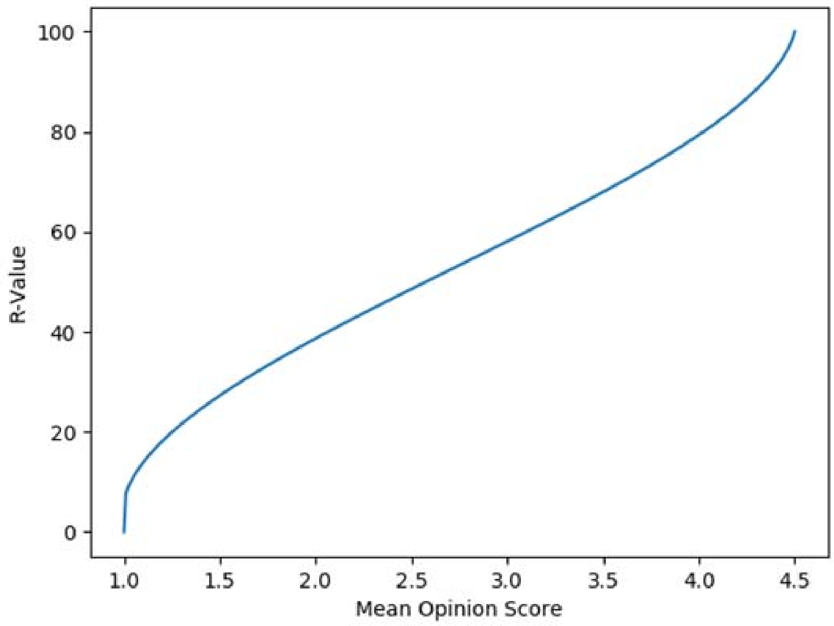

Figure E.2: Relation between MOS and R-value

With G.711 taken as reference anchor, the observed impairment factor *Ie,NB,obs* is calculated as:

Ie,NB,obs = R(G.711) - R(test condition)

for all other reference and test conditions depicted in Figure E.1. The values for MOS, R, *Ie,NB,obs* and *Ie,NB,def* are listed in Table E.5.

Table E.5: Measured MOS, R, *Ie,NB,obs* and *Ie,NB,def* for all reference conditions

Codec References	MOS	R	Ie,NB,obs	Ie,NB,def
G.711@64	4.42	93.58	0.0	0.0
GSM_EFR@12.2	4.31	88.95	4.63	5.0
G.726@32	4.1	82.11	11.47	7.0
G.728@16	4.12	82.69	10.89	7.0
G.729@8	4.11	82.5	11.09	10.0
G.726@32 => G.726@32	4.1	82.11	11.47	14.0
G.728@16 => G.728@16	3.94	77.87	15.71	14.0
GSM_FR@13	3.83	75.21	18.37	20.0
G.729@8 => G.729@8	3.71	72.49	21.09	20.0
GSM_HR@5.6	3.62	70.43	23.15	23.0
G.726@24	3.54	68.88	24.7	25.0
G.729@8 => G.729@8 => G.729@8	3.27	63.31	30.27	30.0
GSM_FR@13 => GSM_FR@13	3.54	68.83	24.75	40.0
G.726@16	2.73	52.99	40.59	50.0

E.3.1.1.2.3 Linear interpolation of the test results

According to [21] pairs of defined equipment impairment factors *Ie,NB,def* and observed values *Ie,NB,obs* are plotted in Figure E.3. A linear regression between the measured and the already known variables can be modeled as:

$$Ie,NB,obs = a \cdot Ie,NB,def + b$$

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,NB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j) * (Ie,NB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,obs_j)}{\sum_{i=1}^{n}(Ie,NB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,NB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j$$

with n = 14 reference conditions and Ie,NB,obs and Ie,NB,def taken from Table E.5.

The results of the coefficients a and b are presented in Figure E.3 together with the regression line, approximating all the reference pairs in a least-squares sense. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination $R^2$of 0.9 indicates a strong fit of the model. The only condition outside of the confidence interval is "GSM_FR@13 => GSM_FR@13". The estimation of Ie,NB by the objective method according to P.863 is probably too optimistic for this tandeming condition because the subjective evaluation shows a higher value here.

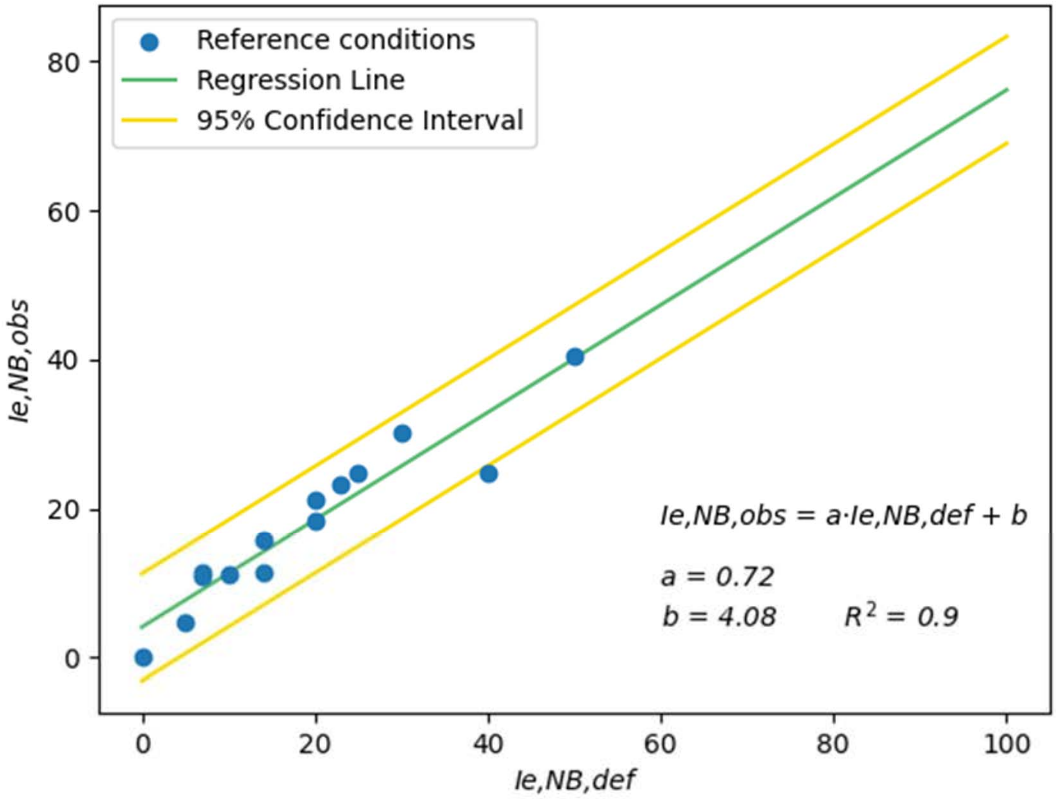

Figure E.3: Linear Regression between observed and defined *Ie,NB*

E.3.1.1.2.4 Determination of a stable *Ie,NB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,NB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,NB,def* for LC3plus based on the parameters a and b. Negative *Ie,NB* values 0 are set to zero. The stable impairment factors are show in Table E.6 and calculate as follows:

$$Ie, NB, \mathrm{def}(LC3plus) = \frac{Ie,NB,obs - b}{a}$$

Table E.6: Measured mean MOS, R, *Ie,NB,obs* and stable *Ie,NB;def* for all conditions under test

Codec under Test	MOS	R	Ie,NB,obs	Ie,NB,def
LC3plus @16	4.11	82.21	11.37	10.11
LC3plus @20	4.35	90.06	3.52	0.0
LC3plus @24	4.46	95.81	-2.23	0.0
LC3plus @32	4.5	99.64	-6.06	0.0

E.3.1.1.2.5 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.7 are encoded and MOS values are measured as shown in Figure E.6. R and *Ie,NB,obs* calculation are performed according to clause E.3.1.1.2.2.

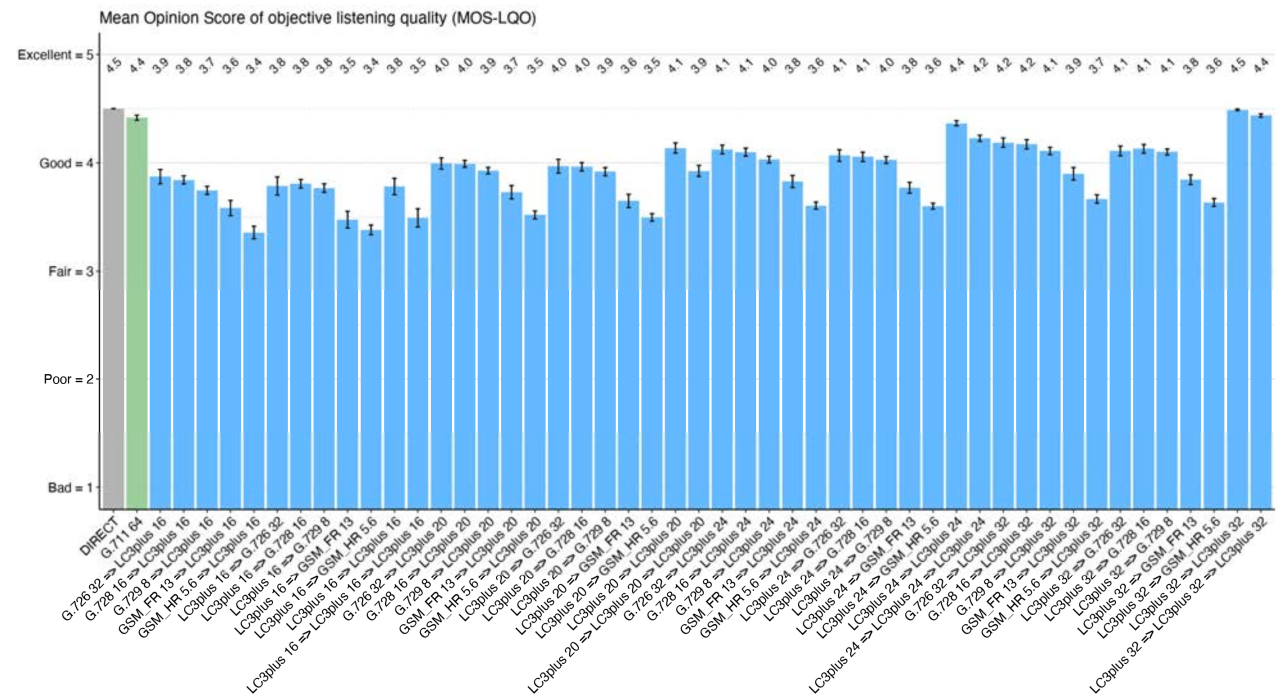

Figure E.4: Recommendation ITU-T P.863 results for NB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

For the additivity check the *Ie,NB,def* values in Table E.7 are derived from *Ie,NB,def* in Table E.5 (reference conditions) and *Ie,NB,def* in Table E.6 (test conditions). For example the tandeming of G.726@32kbit/s and LC3plus@16kbit/s corresponds to a summation of the respective *Ie,NB,def* values: *Ie,NB,def* = 7.0 + 10.11.

Table E.7: Measured mean MOS, R, *Ie,NB,obs* and calculated *Ie,NB,def* for all conditions tandeming conditions for the additivity check

Codec	MOS	R	Ie,NB,obs	Ie,NB,def
G.726@32 => LC3plus@16	3.87	76.2	17.38	17.11
G.728@16 => LC3plus@16	3.84	75.49	18.09	17.11
G.729@8 => LC3plus@16	3.75	73.27	20.32	20.11
GSM-FR@13 => LC3plus@16	3.58	69.7	23.88	30.11
GSM-HR@5.6 => LC3plus@16	3.36	65.05	28.53	33.11
LC3plus@16 => G.726@32	3.79	74.19	19.39	17.11

Codec	MOS	R	Ie,NB,obs	Ie,NB,def
LC3plus@16 => G.728@16	3.81	74.66	18.92	17.11
LC3plus@16 => G.729@8	3.77	73.75	19.83	20.11
LC3plus@16 => GSM-FR@13	3.48	67.47	26.11	30.11
LC3plus@16 => GSM-HR@5.6	3.38	65.53	28.05	33.11
LC3plus@16 => LC3plus@16	3.78	74.07	19.51	20.22
LC3plus@16 => LC3plus@16 => LC3plus@16	3.49	67.8	25.79	30.33
G.726@32 => LC3plus@20	3.99	79.21	14.37	7.0
G.728@16 => LC3plus@20	3.99	79.13	14.45	7.0
G.729@8 => LC3plus@20	3.93	77.55	16.04	10.0
GSM-FR@13 => LC3plus@20	3.73	72.88	20.7	20.0
GSM-HR@5.6 => LC3plus@20	3.52	68.37	25.21	23.0
LC3plus@20 => G.726@32	3.97	78.58	15.0	7.0
LC3plus@20 => G.728@16	3.96	78.43	15.15	7.0
LC3plus@20 => G.729@8	3.92	77.31	16.28	10.0
LC3plus@20 => GSM-FR@13	3.65	71.11	22.47	20.0
LC3plus@20 => GSM-HR@5.6	3.5	67.88	25.7	23.0
LC3plus@20 => LC3plus@20	4.14	83.14	10.44	0.0
LC3plus@20 => LC3plus@20 => LC3plus@20	3.93	77.48	16.1	0.0
G.726@32 => LC3plus@24	4.12	82.74	10.84	7.0
G.728@16 => LC3plus@24	4.1	82.04	11.54	7.0
G.729@8 => LC3plus@24	4.03	80.16	13.43	10.0
GSM-FR@13 => LC3plus@24	3.83	75.15	18.43	20.0
GSM-HR@5.6 => LC3plus@24	3.61	70.18	23.4	23.0
LC3plus@24 => G.726@32	4.07	81.18	12.4	7.0
LC3plus@24 => G.728@16	4.05	80.83	12.76	7.0
LC3plus@24 => G.729@8	4.03	80.07	13.51	10.0
LC3plus@24 => GSM-FR@13	3.77	73.81	19.77	20.0
LC3plus@24 => GSM-HR@5.6	3.6	70.08	23.51	23.0
LC3plus@24 => LC3plus@24	4.37	91.11	2.48	0.0
LC3plus@24 => LC3plus@24 => LC3plus@24	4.23	85.9	7.68	0.0
G.726@32 => LC3plus@32	4.19	84.64	8.95	7.0
G.728@16 => LC3plus@32	4.17	84.16	9.42	7.0
G.729@8 => LC3plus@32	4.11	82.35	11.23	10.0
GSM-FR@13 => LC3plus@32	3.9	76.84	16.74	20.0
GSM-HR@5.6 => LC3plus@32	3.67	71.5	22.08	23.0
LC3plus@32 => G.726@32	4.11	82.37	11.22	7.0
LC3plus@32 => G.728@16	4.13	82.91	10.67	7.0
LC3plus@32 => G.729@8	4.1	82.08	11.51	10.0
LC3plus@32 => GSM-FR@13	3.84	75.51	18.07	20.0
LC3plus@32 => GSM-HR@5.6	3.63	70.8	22.78	23.0

Codec	MOS	R	Ie,NB,obs	Ie,NB,def
LC3plus@32 => LC3plus@32	4.49	98.74	-5.16	0.0
LC3plus@32 => LC3plus@32 => LC3plus@32	4.44	94.71	-1.13	0.0

As done before the pairs of defined and observed *Ie,NB* are plotted against each other in Figure E.5. The majority of points lie within the confidence interval and hence proof the additivity of the newly calculated *Ie,NB,def* for LC3plus according to Recommendation ITU-T P.834 [21]. In other words the added defined impairment values correspond to the observed values.

The conditions outside of the confidence intervals are:

"LC3plus@20 => LC3plus@20 => LC3plus@20" with an error of 12.02
"LC3plus@32 => LC3plus@32" with an error of -9.24

The additive model does not consider an offset for tandeming of very good quality codecs. Of course 0 + 0 + 0 is still 0 and the decrease in quality after even the 10th tandeming is not taken into account. This behaviour can be observed at condition "LC3plus@20 => LC3plus@20 => LC3plus@20". The outlier at "LC3plus@32 => LC3plus@32" can be neglected. At 32kbit the quality is so good that Ie becomes negative. If the same rule was applied as in clause E.3.1.1.2.4 this value would be set to 0 and therefore move inside the confidence interval.

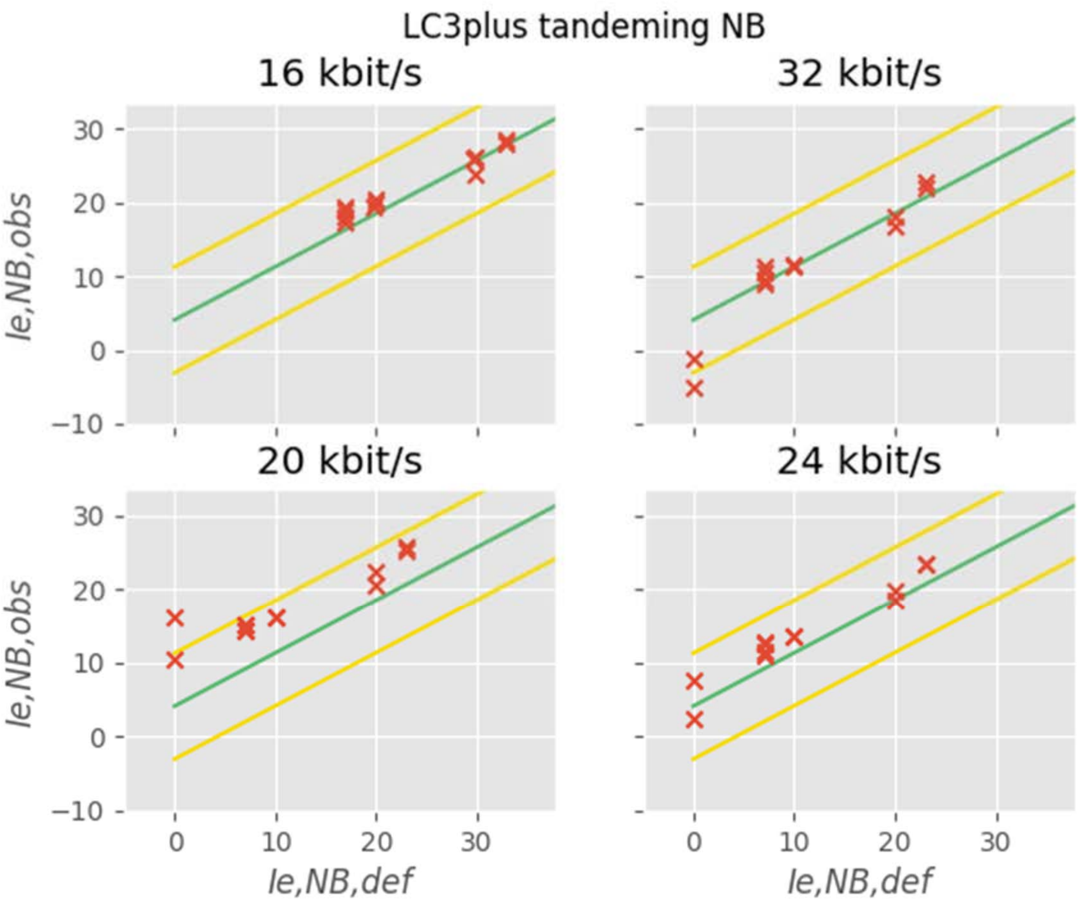

Figure E.5: Observed and defined *Ie,NB* for tandeming conditions with original regression line

E.3.1.1.2.6 Determination of *Ie,NB* for LC3plus under transmission error conditions

To determine *Ie,NB* values for LC3plus under transmission error conditions the same reference conditions as in Table E.5 (14 clean codecs) plus 12 supplementary reference conditions from Appendix I/G 113 [22] are needed.

Figure E.6 shows the Recommendation ITU-T P.863 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes (EPFsizes).

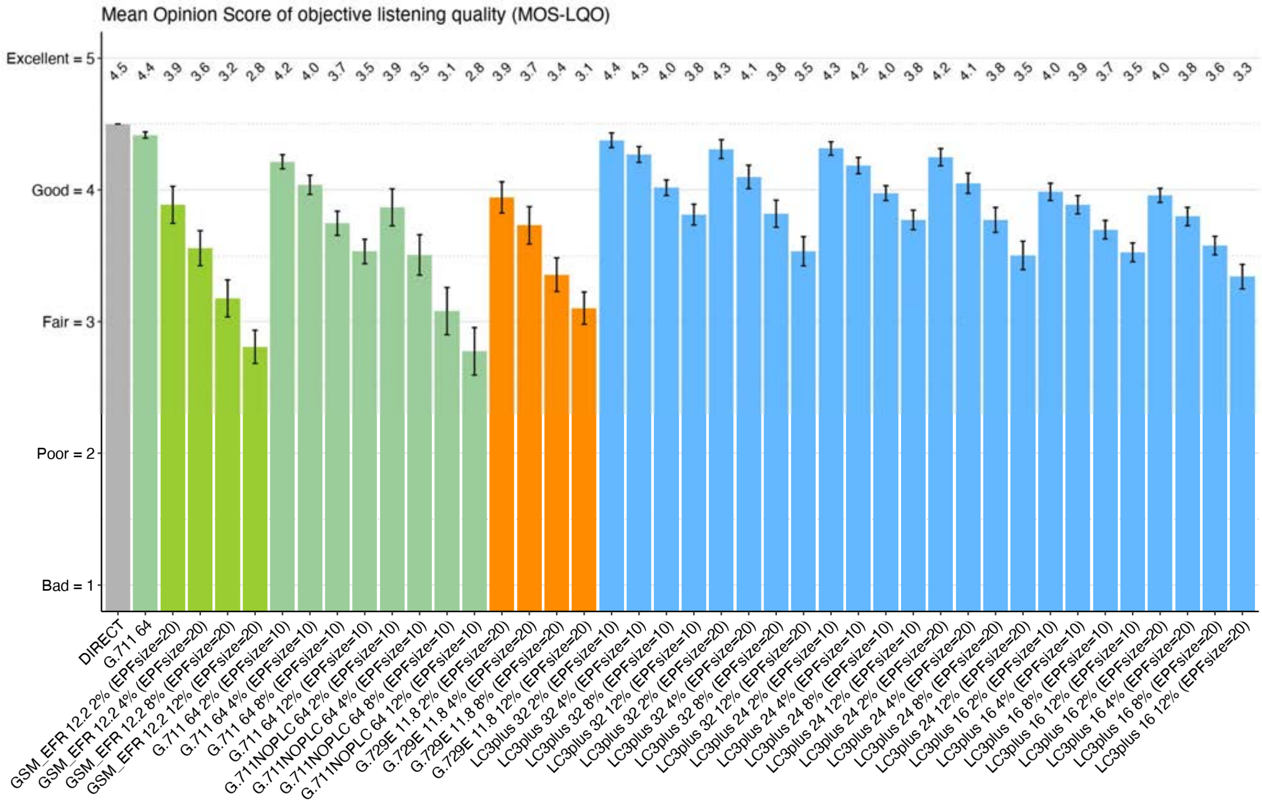

Figure E.6: Recommendation ITU-T P.863 results for NB transmission error speech signals. Mean scores and 95 % confidence intervals

For 12 errorprone reference conditions an effective equipment impairment factor *Ie,eff,NB* is calculated according to the following equation taken from Recommendation ITU-T G.107 [23] (equations 7-29 and 7-30):

$$BurstR = \frac{1 - \frac{Ppl}{100}}{\mathrm{q}}$$

$$Ie, eff, NB = Ie, NB + (95 - \ Ie, NB) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl}$$

The actual values for Ppl and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor Bpl is taken from Recommendation ITU-T G.113 [22].

Table E.8: Parameters used to calculate *Ie,eff,NB*

Condition	Ie,NB	*Ppl* (assumed)	Bpl	*Ppl* (measured)	q (measured)	BurstR	Ie,effNB	Ie,NB,obs
GSM_EFR	5	2	10.0	1.98	0.98	1.0	19.87	17.05
GSM_EFR	5	4	10.0	3.96	0.97	0.99	30.44	24.42
GSM_EFR	5	8	10.0	7.93	0.93	0.99	44.59	32.1
GSM_EFR	5	12	10.0	11.89	0.88	1.01	54.02	39.15
G.711	0	2	25.1	1.91	0.98	1.01	6.73	8.08
G.711	0	4	25.1	3.82	0.96	1.0	12.56	13.19
G.711	0	8	25.1	7.64	0.92	1.0	22.17	20.3
G.711	0	12	25.1	11.46	0.89	1.0	29.76	24.94
G.711NOPLC	0	2	4.3	1.91	0.98	1.01	29.29	17.5
G.711NOPLC	0	4	4.3	3.82	0.96	1.0	44.71	25.49
G.711NOPLC	0	8	4.3	7.64	0.92	1.0	60.76	33.97
G.711NOPLC	0	12	4.3	11.46	0.89	1.0	68.97	39.8
G.729E	4	2	8.1	1.98	0.98	1.0	21.88	15.67
G.729E	4	4	8.1	3.96	0.97	0.99	33.77	20.66
G.729E	4	8	8.1	7.93	0.93	0.99	48.75	28.55
G.729E	4	12	8.1	11.89	0.88	1.01	58.29	33.54

Next the observed *Ie,NB* values are calculated according to clause E.3.1.1.2.2. The results listed in Table E.8 show increasing impairment factors for increasing transmission error rates. The new pairs of *Ie,NB,obs* and *Ie,eff,NB* are added in the scatterplot Figure E.7. Seven of the transmission error conditions lie outside of the confidence interval. Therefor a new interpolation line (purple) is defined by the formula and coefficients depicted in Figure E.7. It is calculated from all 14 clean and 16 error-prone reference conditions and shows a strong fit with R^2 of 0.86.

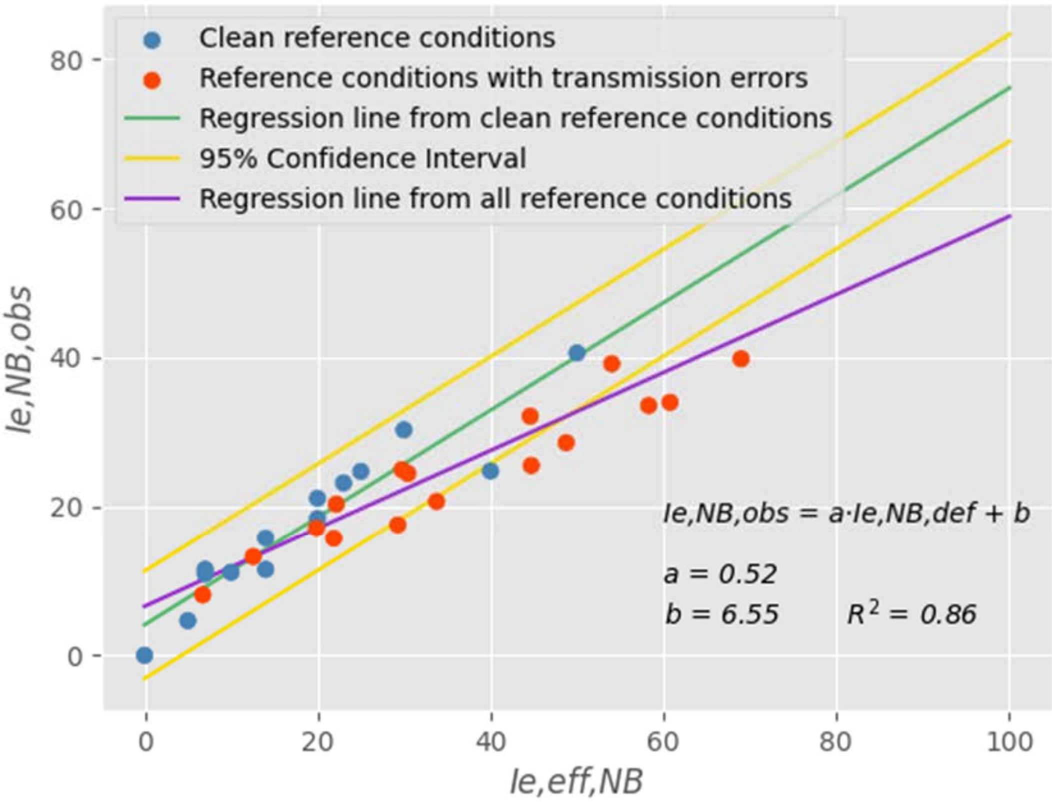

Figure E.7: Linear Regression between observed *Ie,NB,obs* and defined *Ie,eff,NB* for clean and error-prone reference conditions

With these new coefficients a and b stable *IeNB,def* values for LC3plus with transmission errors can be derived (Table E.9) without existing *Bpl* values.

Table E.9: Stable *Ie,NB,def* values for LC3plus under transmission error conditions

Codec under Test	Ie,NB,obs	Ie,NB,def
LC3plus@32 2% (EPFsize=10)	1.97	0
LC3plus@32 4% (EPFsize=10)	6.24	0
LC3plus@32 8% (EPFsize=10)	13.77	13.78
LC3plus@32 12% (EPFsize=10)	18.81	23.4
LC3plus@32 2% (EPFsize=20)	4.74	0
LC3plus@32 4% (EPFsize=20)	11.55	9.54
LC3plus@32 8% (EPFsize=20)	18.64	23.08
LC3plus@32 12% (EPFsize=20)	24.91	35.05
LC3plus@24 2% (EPFsize=10)	4.57	0
LC3plus@24 4% (EPFsize=10)	9.0	4.67
LC3plus@24 8% (EPFsize=10)	14.85	15.84
LC3plus@24 12% (EPFsize=10)	19.73	25.16
LC3plus@24 2% (EPFsize=20)	6.93	0.72
LC3plus@24 4% (EPFsize=20)	12.87	12.06
LC3plus@24 8% (EPFsize=20)	19.72	25.14
LC3plus@24 12% (EPFsize=20)	25.57	36.31
LC3plus@16 2% (EPFsize=10)	14.6	15.36
LC3plus@16 4% (EPFsize=10)	17.03	20.0
LC3plus@16 8% (EPFsize=10)	21.39	28.33
LC3plus@16 12% (EPFsize=10)	25.09	35.39
LC3plus@16 2% (EPFsize=20)	15.28	16.66
LC3plus@16 4% (EPFsize=20)	19.13	24.01
LC3plus@16 8% (EPFsize=20)	24.0	33.31
LC3plus@16 12% (EPFsize=20)	28.85	42.57

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern Frame sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,NB* curve the best. The optimal value can be determined by calculating the least square error for the every four pairs of *Ie,NB,obs(Ppl)* and *Ie,eff,NB(Ppl)*.

Table E.10: Packet loss robustness factors for LC3plus

Bitrate (kbit/s)	EPFsize (ms)	Ie,NB,def	Bpl
32	10	0.0	49.1
32	20	0.0	25.1
24	10	0.0	39.1
24	20	0.0	22.2
16	10	10.11	27.6
16	20	10.11	20.0

Figure E.8 and Figure E.9 show *Ie,eff,NB(Ppl)* with fixed *Bpl* from Table E.10 and an assumed BurtsR value of 1. For 16 kbit/s the curve fits very good for both error pattern frame sizes. For 24 and 32 kbit/s the curve shows a weak fit, because it is forced to go through (0|0) by the *Ie,eff,NB* equation. The model has difficulties fitting higher quality codecs.

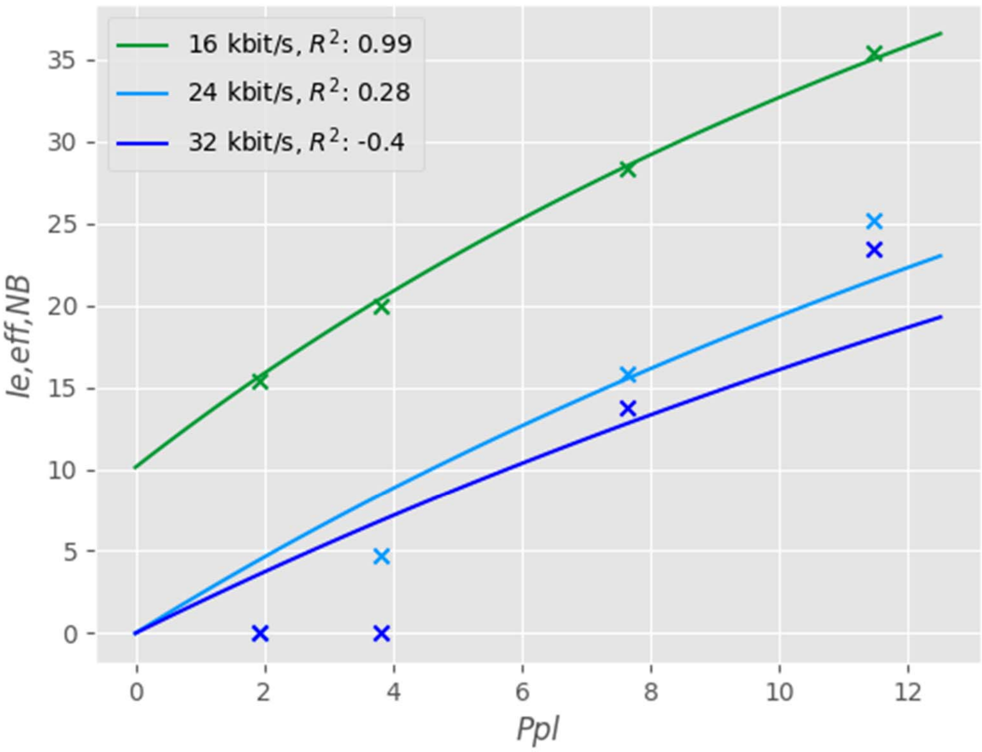

Figure E.8: *Ie,eff,NB* approximation with best fitting *Bpl* and observed *Ie,NB* values vs. packet loss rate for 10 ms EPFsize

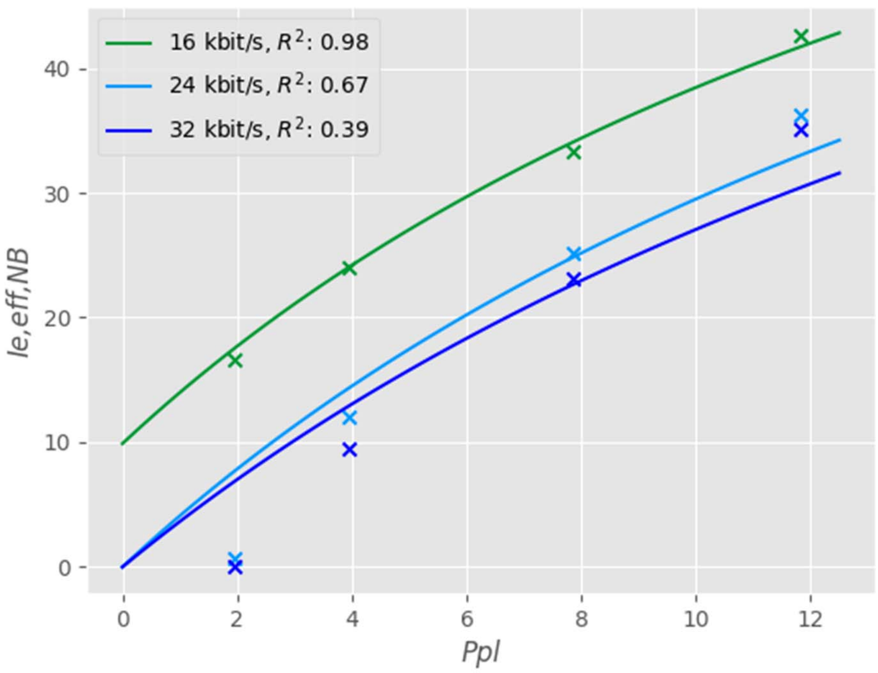

Figure E.9: *Ie,eff,NB* approximation with best fitting *Bpl* and observed *Ie,NB* values vs. packet loss rate for 20 ms EPFsize

E.3.1.2 Subjective assessment

E.3.1.2.1 Introduction

This clause describes the calculation procedure of the equipment impairment factor *Ie,NB* and the packet loss robustness factor *Bpl* for LC3plus. It is determined according to Recommendation ITU-T P.833 [20]. MOS values are obtained by performing P.800 [1] listening tests according to considerations in clauses E.1.3 and E.1.4.

E.3.1.2.2 Determination of Ie for LC3plus in error-free conditions

E.3.1.2.2.1 Measuring MOS

Figure E.10 shows the mean MOS of the 14 reference conditions (see Recommendation ITU-T P.834 [21], table 1) and four LC3plus configurations under test with -16, -26 and -36 dBoV and bitrates of 16, 20, 24 and 32 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

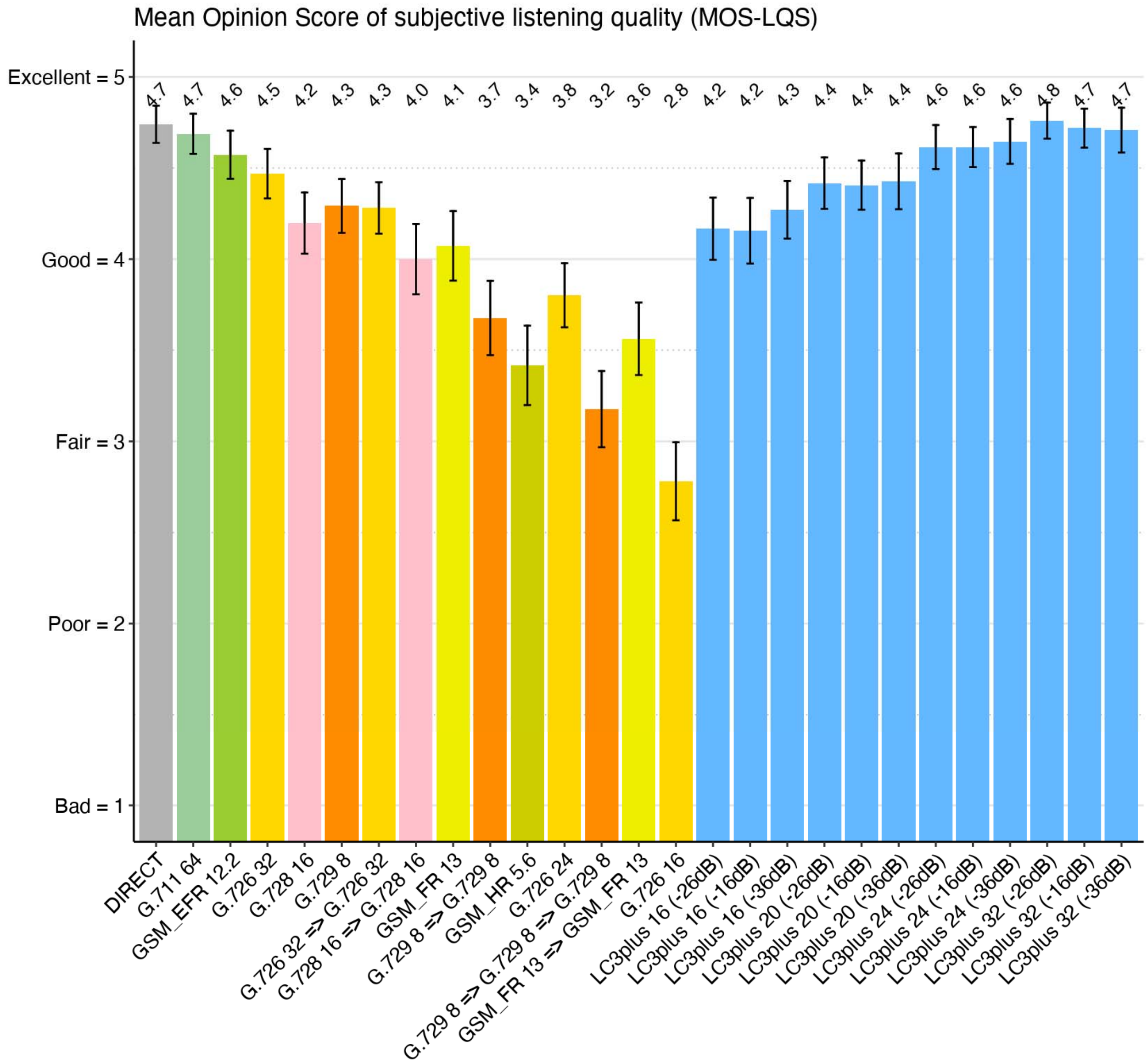

Figure E.10: Recommendation ITU-T P.800 results for NB clean speech signals. Mean scores and 95 % confidence intervals

E.3.1.2.2.2 Scale transformation

All Recommendation ITU-T P.800 test results are transformed from the MOS scale to the scale of the equipment impairment factor *Ie,NB*. In a first step R is determined by solving the following equations:

$$\begin{aligned} &\text{for } MOS = 1.0: && R = 0 \\ &\text{for } 1.0 < MOS < 4.5: && MOS = 1 + 0.035 \cdot R + R \cdot (R-60) \cdot (100-R) \cdot 7 \cdot 10^{-6} \\ &\text{for } MOS \geq 4.5: && R = 100 \end{aligned}$$

The relation between MOS and R-value is presented graphically in Figure E.11 and shows that differences on the outer end of the MOS scale are weighted stronger on the R scale.

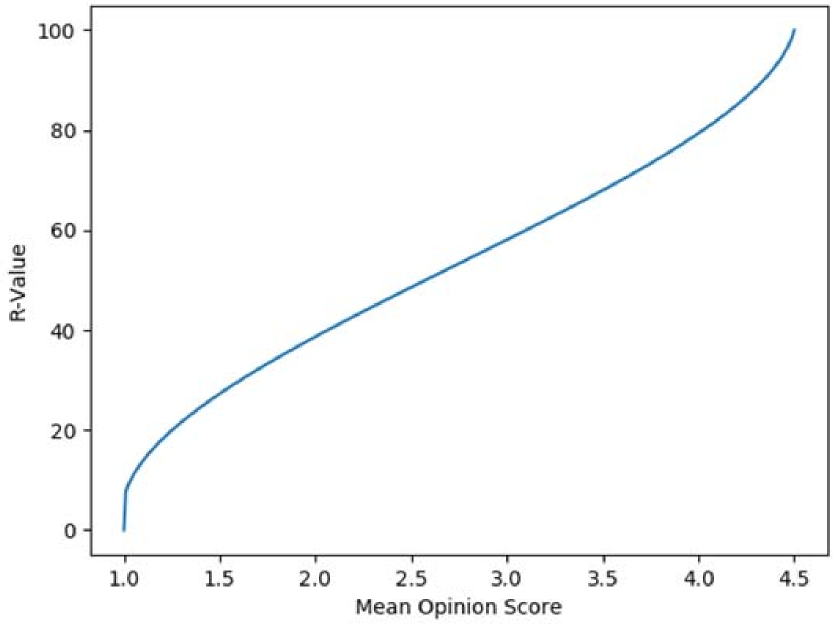

Figure E.11: Relation between MOS and R-value

With G.711 taken as reference anchor, the observed impairment factor *Ie,NB,obs* is calculated as:

Ie,NB,obs = R(G.711) - R(test condition)

for all other reference and test conditions depicted in Figure E.10. The values for MOS, R, *Ie,NB,obs* and *Ie,NB,def* are listed in Table E.11:

Table E.11: MOS, R, *Ie,NB,obs* and *Ie,NB,def* for all reference conditions

Codec References	MOS	R	Ie,NB,obs	Ie,NB,def
G.711@64	4.69	100.0	0.0	0
GSM_EFR@12.2	4.57	100.0	0.0	5
G.726@32	4.47	96.89	3.11	7
G.728@16	4.2	85.0	15.0	7
G.729@8	4.29	88.18	11.82	1
G.726@32 => G.726@32	4.28	87.78	12.22	14
G.728@16 => G.728@16	4.0	79.37	20.63	14
GSM_FR@13	4.07	81.32	18.68	20

Codec References	MOS	R	Ie,NB,obs	Ie,NB,def
G.729@8 => G.729@8	3.68	71.73	28.27	20
GSM_HR@5.6	3.42	66.26	33.74	23
G.726@24	3.8	74.54	25.46	25
G.729@8 => G.729@8 => G.729@8	3.18	61.49	38.51	30
GSM_FR@13 => GSM_FR@13	3.56	69.28	30.72	40
G.726@16	2.78	53.91	46.09	50

E.3.1.2.2.3 Linear interpolation of the test results

According to Recommendation ITU-T P.833 [20] pairs of defined equipment impairment factors *Ie,NB,def* and observed values *Ie,NB,obs* are plotted in Figure E.12. A linear regression between the measured and the already known variables can be modeled as:

$$Ie,NB,obs = a \cdot Ie,NB,def + b$$

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,NB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j) * (Ie,NB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,obs_j)}{\sum_{i=1}^{n}(Ie,NB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,NB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,NB,def_j$$

with n = 14 reference conditions and Ie,NB,obs and Ie,NB,def taken from Table E.11.

The results of the coefficients a and b are presented in Figure E.12 together with the regression line, approximating all the reference pairs in a least-squares sense. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination $R^2$of 0.82 indicates a strong fit of the model.

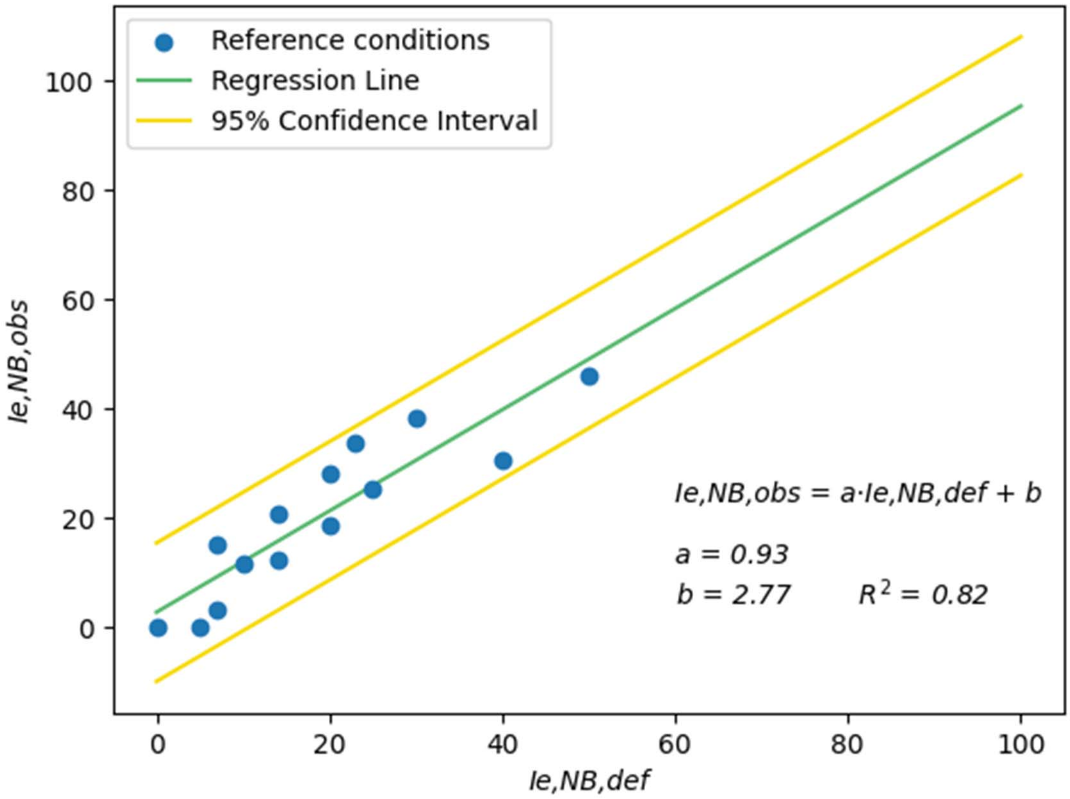

Figure E.12: Linear Regression between observed and defined *Ie,NB*

E.3.1.2.2.4 Determination of a stable *Ie,NB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,NB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,NB,def* for LC3plus based on the parameters a and b. Negative *Ie,NB* values 0 are set to zero. The stable impairment factors are show in Table E.12 and calculate as follows:

$$Ie,NB,\mathrm{def}(LC3plus) = \frac{Ie,NB,obs-b}{a}$$

Table E.12: MOS, R, *Ie,NB,obs* and stable *Ie,NB,def* for all conditions under test

Codec under Test	MOS	R	*Ie,NB,obs*	*Ie,NB,def*
LC3plus@16	4.17	85.0	15.0	13.2
LC3plus@20	4.42	93.59	6.41	3.93
LC3plus@24	4.62	100.0	0.0	0.0
LC3plus@32	4.76	100.0	0.0	0.0

E.3.1.2.2.5 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.13 are encoded and MOS values are shown in Figure E.13. R and *Ie,NB,obs* calculation are performed according to clause E.3.1.2.2.2.

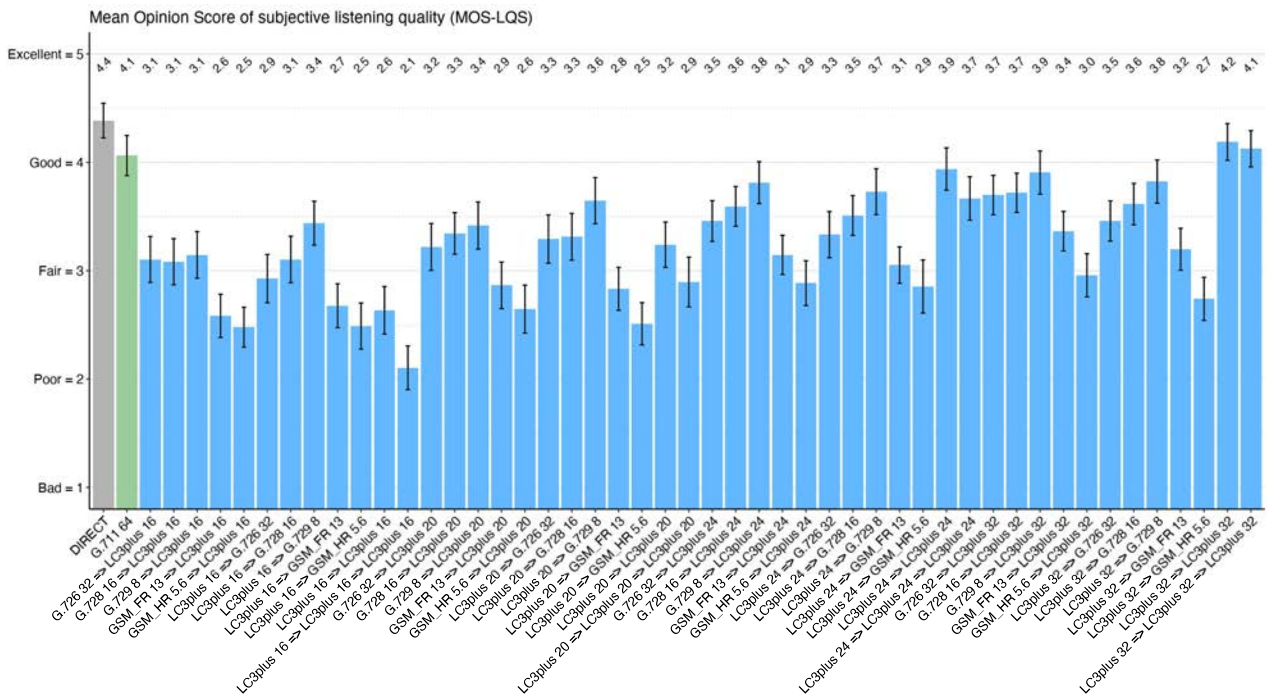

Figure E.13: Recommendation ITU-T P.800 results for NB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

For the additivity check the *Ie,NB,def* values in Table E.13 are derived from *Ie,NB,def* in Table E.11 (reference conditions) and *Ie,NB,def* in Table E.12 (test conditions). For example the tandeming of G.726@32kbit/s and LC3plus@16kbit/s corresponds to a summation of the respective *Ie,NB,def* values: *Ie,NB,def* = 7.0 + 14.24.

Table E.13: MOS, R, *Ie,NB,obs* and calculated *Ie,NB,def* for all conditions tandeming conditions for the additivity check

Codec	MOS	R	*Ie,NB,obs*	*Ie,NB,def*
G.726@32 => LC3plus@16	3.1	60.08	20.97	20.2
G.728@16 => LC3plus@16	3.08	59.67	21.38	20.2
G.729@8 => LC3plus@16	3.15	60.89	20.16	23.2
GSM-FR@13 => LC3plus@16	2.58	50.15	30.9	33.2
GSM-HR@5.6 => LC3plus@16	2.48	48.17	32.88	36.2
LC3plus@16 => G.726@32	2.93	56.69	24.36	20.2
LC3plus@16 => G.728@16	3.1	60.08	20.97	20.2
LC3plus@16 => G.729@8	3.44	66.69	14.36	23.2
LC3plus@16 => GSM-FR@13	2.68	51.94	29.11	33.2
LC3plus@16 => GSM-HR@5.6	2.49	48.38	32.67	36.2
LC3plus@16 => LC3plus@16	2.64	51.14	29.91	26.4
LC3plus@16 => LC3plus@16 => LC3plus@16	2.1	40.81	40.24	39.6
G.726@32 => LC3plus@20	3.22	62.31	18.73	10.93
G.728@16 => LC3plus@20	3.34	64.79	16.26	10.93
G.729@8 => LC3plus@20	3.42	66.26	14.79	13.93
GSM-FR@13 => LC3plus@20	2.86	55.51	25.54	23.93
GSM-HR@5.6 => LC3plus@20	2.65	51.35	29.7	26.93

Codec	MOS	R	*Ie,NB,obs*	*Ie,NB,def*
LC3plus@20 => G.726@32	3.29	63.75	17.3	10.93
LC3plus@20 => G.728@16	3.31	64.17	16.88	10.93
LC3plus@20 => G.729@8	3.65	71.05	10.0	13.93
LC3plus@20 => GSM-FR@13	2.83	54.9	26.15	23.93
LC3plus@20 => GSM-HR@5.6	2.51	48.76	32.29	26.93
LC3plus@20 => LC3plus@20	3.24	62.73	18.32	7.86
LC3plus@20 => LC3plus@20 => LC3plus@20	2.9	56.09	24.95	11.79
G.726@32 => LC3plus@24	3.46	67.1	13.95	7.0
G.728@16 => LC3plus@24	3.59	69.94	11.11	7.0
G.729@8 => LC3plus@24	3.81	74.79	6.25	10.0
GSM-FR@13 => LC3plus@24	3.15	60.89	20.16	20.0
GSM-HR@5.6 => LC3plus@24	2.88	55.89	25.16	23.0
LC3plus@24 => G.726@32	3.33	64.57	16.48	7.0
LC3plus@24 => G.728@16	3.51	68.17	12.88	7.0
LC3plus@24 => G.729@8	3.73	72.88	8.17	10.0
LC3plus@24 => GSM-FR@13	3.05	59.08	21.97	20.0
LC3plus@24 => GSM-HR@5.6	2.85	55.3	25.75	23.0
LC3plus@24 => LC3plus@24	3.94	77.79	3.25	0.0
LC3plus@24 => LC3plus@24 => LC3plus@24	3.67	71.51	9.54	0.0
G.726@32 => LC3plus@32	3.7	72.19	8.86	7.0
G.728@16 => LC3plus@32	3.72	72.66	8.39	7.0
G.729@8 => LC3plus@32	3.91	77.01	4.04	10.0
GSM-FR@13 => LC3plus@32	3.36	65.21	15.84	20.0
GSM-HR@5.6 => LC3plus@32	2.96	57.28	23.77	23.0
LC3plus@32 => G.726@32	3.46	67.1	13.95	7.0
LC3plus@32 => G.728@16	3.62	70.38	10.66	7.0
LC3plus@32 => G.729@8	3.82	75.03	6.02	10.0
LC3plus@32 => GSM-FR@13	3.2	61.9	19.15	20.0
LC3plus@32 => GSM-HR@5.6	2.74	53.13	27.91	23.0
LC3plus@32 => LC3plus@32	4.19	84.68	-3.63	0.0
LC3plus@32 => LC3plus@32 => LC3plus@32	4.12	82.79	-1.74	0.0

As done before the pairs of defined and observed *Ie,NB* are plotted against each other in Figure E.14. All points lie within the confidence interval and hence proof the additivity of the newly calculated *Ie,NB,def* for LC3plus according to Recommendation ITU-T P.833 [20]. In other words the added defined impairment values correspond to the observed values.

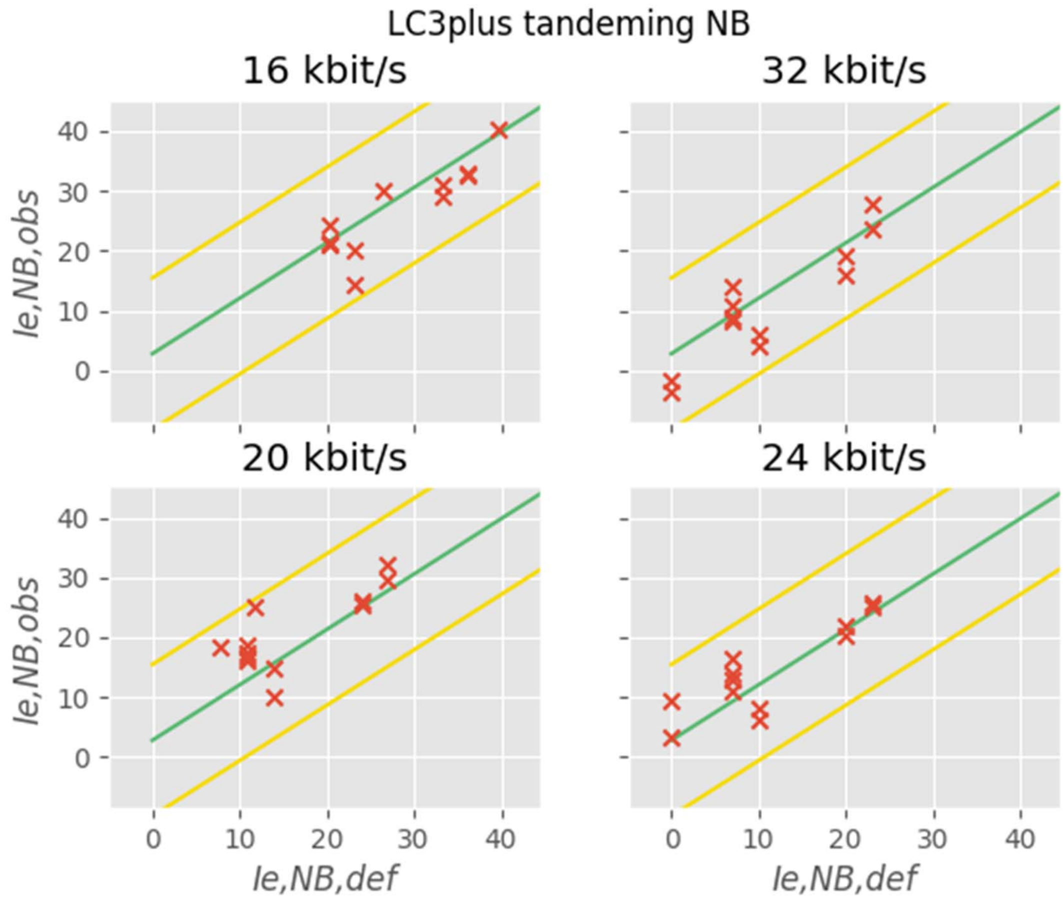

Figure E.14: Observed and defined *Ie,NB* for tandeming conditions with original regression line

E.3.1.2.2.6 Determination of *Ie,NB* for LC3plus under transmission error conditions

To determine *Ie,NB* values for LC3plus under transmission error conditions the same reference conditions as in Table E.11 (14 clean codecs) plus 12 supplementary reference conditions from Appendix I/G 113 [22] are needed.

Figure E.15 shows the Recommendation ITU-T P.800 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes(EPFsizes).

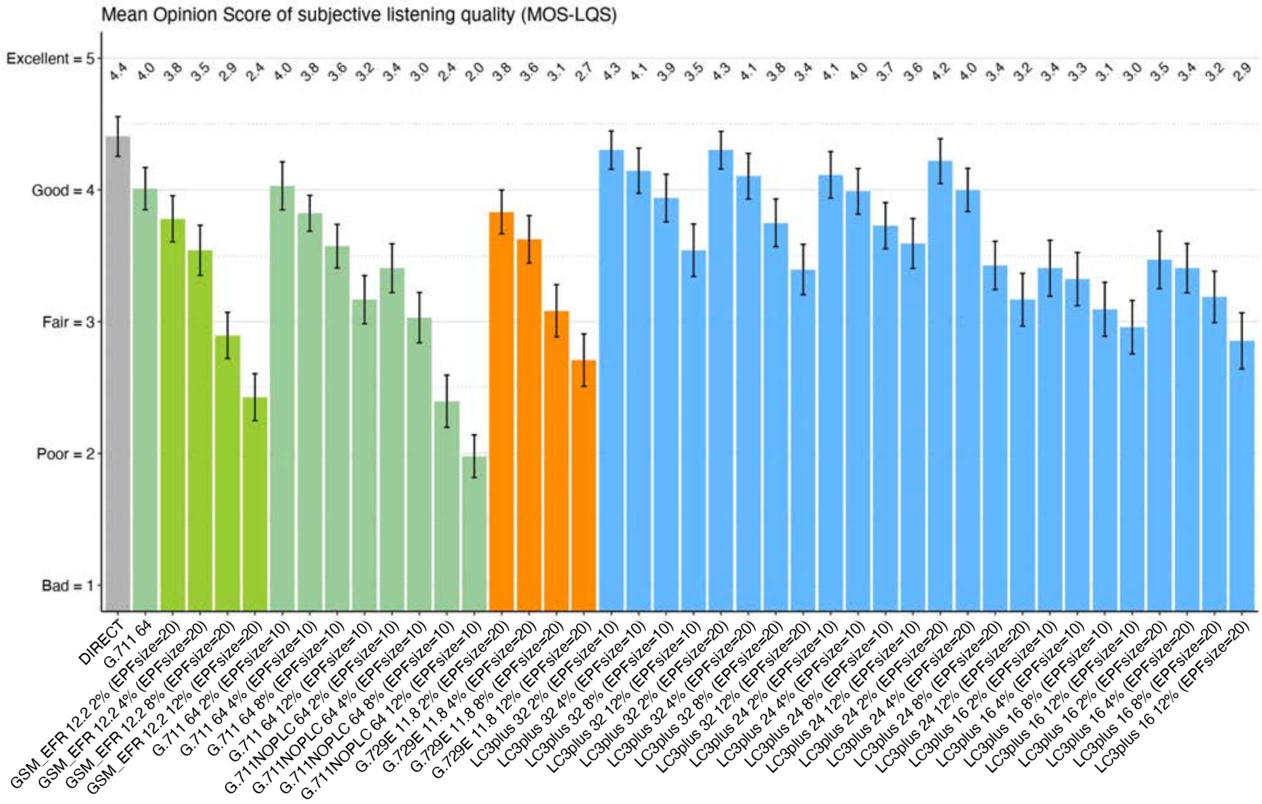

Figure E.15: Recommendation ITU-T P.800 results for NB transmission error speech signals. Mean scores and 95 % confidence intervals

For 12 errorprone reference conditions an effective equipment impairment factor *Ie,eff,NB* is calculated according to the following equation taken from Recommendation ITU-T G.107 [23] (equations 7-29 and 7-30):

$$BurstR = \frac{1-\frac{Ppl}{100}}{q}$$

$$Ie,eff,NB = Ie,NB + (95 - Ie,NB) \cdot \frac{Ppl}{\frac{Ppl}{BurstR}+Bpl}$$

The actual values for Ppl and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor Bpl is taken from Recommendation ITU-T G.113 [22].

Table E.14: Parameters used to calculate *Ie,eff,NB*

Condition	*Ie,NB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,effNB*	*Ie,NB,obs*
GSM_EFR	5.0	2	10.0	1.98	0.98	1.0	19.87	5.58
GSM_EFR	5.0	4	10.0	3.96	0.97	0.99	30.44	10.79
GSM_EFR	5.0	8	10.0	7.93	0.93	0.99	44.59	23.54
GSM_EFR	5.0	12	10.0	11.89	0.88	1.01	54.02	32.46
G.711	0.0	2	25.1	1.91	0.98	1.01	6.73	-0.55
G.711	0.0	4	25.1	3.82	0.96	1.0	12.56	4.61
G.711	0.0	8	25.1	7.64	0.92	1.0	22.17	10.14
G.711	0.0	12	25.1	11.46	0.89	1.0	29.76	18.33

Condition	*Ie,NB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,effNB*	*Ie,NB,obs*
G.711NOPLC	0.0	2	4.3	1.91	0.98	1.01	29.29	13.6
G.711NOPLC	0.0	4	4.3	3.82	0.96	1.0	44.71	20.96
G.711NOPLC	0.0	8	4.3	7.64	0.92	1.0	60.76	33.06
G.711NOPLC	0.0	12	4.3	11.46	0.89	1.0	68.97	41.38
G.729E	4.0	2	8.1	1.98	0.98	1.0	21.88	4.37
G.729E	4.0	4	8.1	3.96	0.97	0.99	33.77	9.03
G.729E	4.0	8	8.1	7.93	0.93	0.99	48.75	19.96
G.729E	4.0	12	8.1	11.89	0.88	1.01	58.29	27.1

Next the observed *Ie,NB* values are calculated according to clause E.3.1.2.2.2. The results listed in Table E.14 show increasing impairment factors for increasing transmission error rates. The new pairs of *Ie,NB,obs* and *Ie,eff,NB* are added in the scatterplot Figure E.16. The new interpolation line (purple) is defined by the formula and coefficients depicted in Figure E.16. Since the new data points do not gather around the original line and only 3 points fall inside the confidence interval, a new regression line is calculated from the 16 error-prone reference conditions as stated in Recommendation ITU-T P.833 [20]. This new line shows a strong fit with R^2 of 0.92.

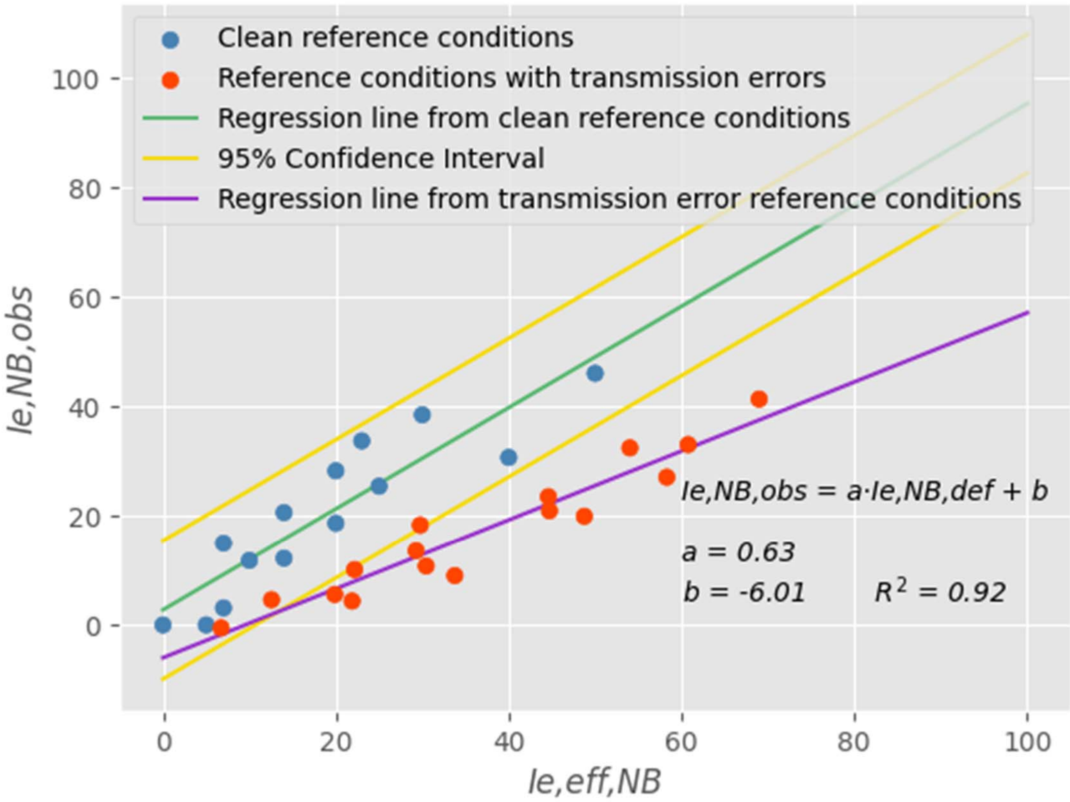

Figure E.16: Linear Regression between observed *Ie,NB,obs* and defined *Ie,eff,NB* for clean and error-prone reference conditions

With these new coefficients a and b stable *IeNB,def* values for LC3plus with transmission errors can be derived (Table E.15) without existing *Bpl* values.

Table E.15: Stable *Ie,NB,def* values for LC3plus under transmission error conditions

Codec under Test	*Ie,NB,obs*	*Ie,NB,def*
LC3plus@32 2% (EPFsize=10)	-8.92	0
LC3plus@32 4% (EPFsize=10)	-3.78	3.52
LC3plus@32 8% (EPFsize=10)	1.84	12.42
LC3plus@32 12% (EPFsize=10)	10.79	26.59
LC3plus@32 2% (EPFsize=20)	-8.92	0
LC3plus@32 4% (EPFsize=20)	-2.56	5.46
LC3plus@32 8% (EPFsize=20)	6.28	19.45
LC3plus@32 12% (EPFsize=20)	13.8	31.36
LC3plus@24 2% (EPFsize=10)	-2.87	4.96
LC3plus@24 4% (EPFsize=10)	0.52	10.33
LC3plus@24 8% (EPFsize=10)	6.75	20.2
LC3plus@24 12% (EPFsize=10)	9.7	24.87
LC3plus@24 2% (EPFsize=20)	-6.04	0
LC3plus@24 4% (EPFsize=20)	0.26	9.92
LC3plus@24 8% (EPFsize=20)	13.17	30.36
LC3plus@24 12% (EPFsize=20)	18.33	38.53
LC3plus@16 2% (EPFsize=10)	13.6	31.04
LC3plus@16 4% (EPFsize=10)	15.26	33.67
LC3plus@16 8% (EPFsize=10)	19.75	40.78
LC3plus@16 12% (EPFsize=10)	22.36	44.91
LC3plus@16 2% (EPFsize=20)	12.31	29.0
LC3plus@16 4% (EPFsize=20)	13.6	31.04
LC3plus@16 8% (EPFsize=20)	17.92	37.88
LC3plus@16 12% (EPFsize=20)	24.33	48.03

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,NB* curve the best. The optimal value can be determined by calculating the least square error for the every four pairs of *Ie,NB,obs(Ppl)* and *Ie,eff,NB(Ppl)*.

Table E.16: NB Packet loss robustness factors for LC3plus

Bitrate (kbit/s)	EPFsize (ms)	*Ie,NB,def*	*Bpl*
32	10	0.0	41.5
32	20	0.0	31.1
24	10	0.0	30.9
24	20	0.0	20
16	10	13.2	14.3
16	20	13.2	15.4

Figure E.17 and Figure E.18 show *Ie,eff,NB(Ppl)* with fixed *Bpl* from Table E.16 and an assumed BurtsR value of 1. For 24 kbit/s the curve fits very good for 10 ms EPFsize. For the other bitrates and 20 ms EPFsize the depicted curve is a bad approximation of the subjective data as shown by R^2. A negative R^2 indicates that a simple mean over all *Ppl* would have a better fit than the depicted curve. For 10 ms 16kbit/s a flatter curve would be needed, but is not possible with the given *Ie,eff,NB* equation. For 32 kbit/s the curve shows a weak fit, because it is forced to go through (0|0) by the *Ie,eff,NB* equation. The model has difficulties fitting higher quality codecs.

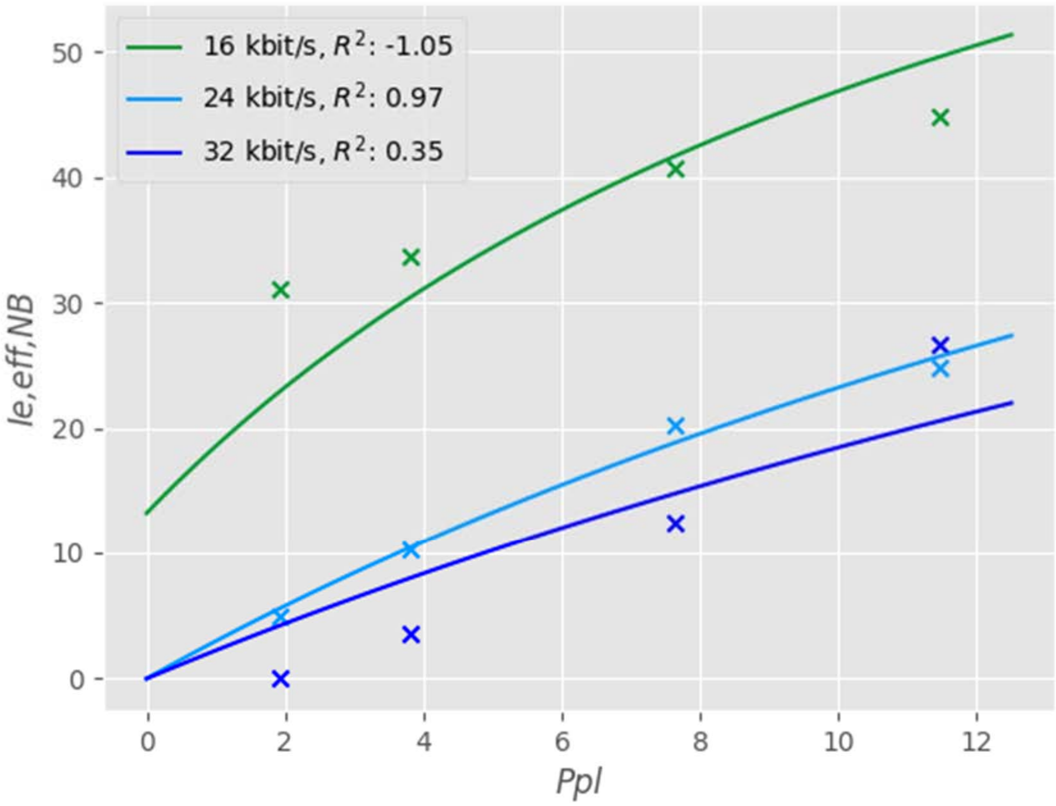

Figure E.17: *Ie,eff,NB* approximation with best fitting *Bpl* and observed *Ie,NB* values vs. packet loss rate for 10 ms EPFsize

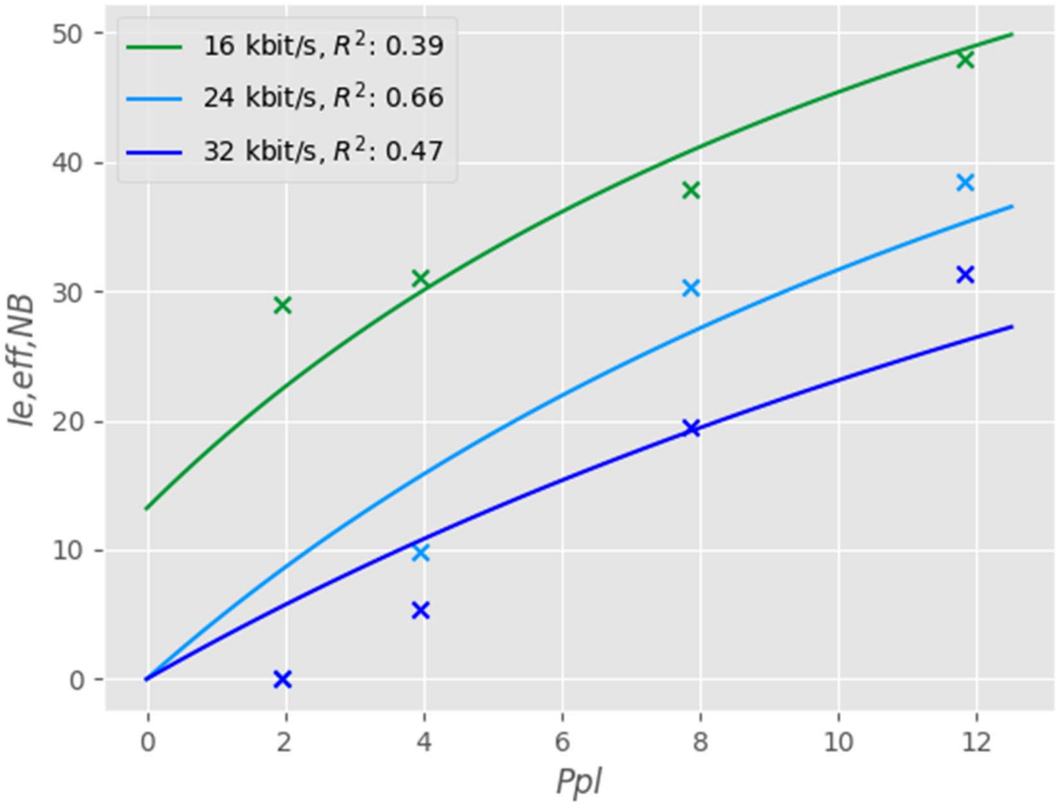

Figure E.18: *Ie,eff,NB* approximation with best fitting *Bpl* and observed *Ie,NB* values vs. packet loss rate for 20 ms EPFsize

E.3.2 Wideband (WB) Ie factors

E.3.2.1 Objective assessment

E.3.2.1.1 Introduction

This clause describes the derivation of the equipment impairment factors *Ie,WB* and the packet loss robustness factor Bpl for LC3plus in wideband mode. It is determined according to Recommendation ITU-T P.834.1 [25]. MOS values are obtained by evaluating coded 16 kHz English speech signals with Recommendation ITU-T P.863 [2].

E.3.2.1.2 Determination of *Ie,WB* for LC3plus in error-free conditions

E.3.2.1.2.1 Measuring MOS

Figure E.19 shows the mean MOS of the 12 reference conditions and LC3plus configurations under test with -16, -26 and -36 dBOV bitrates of 16, 24, 32 and 48 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

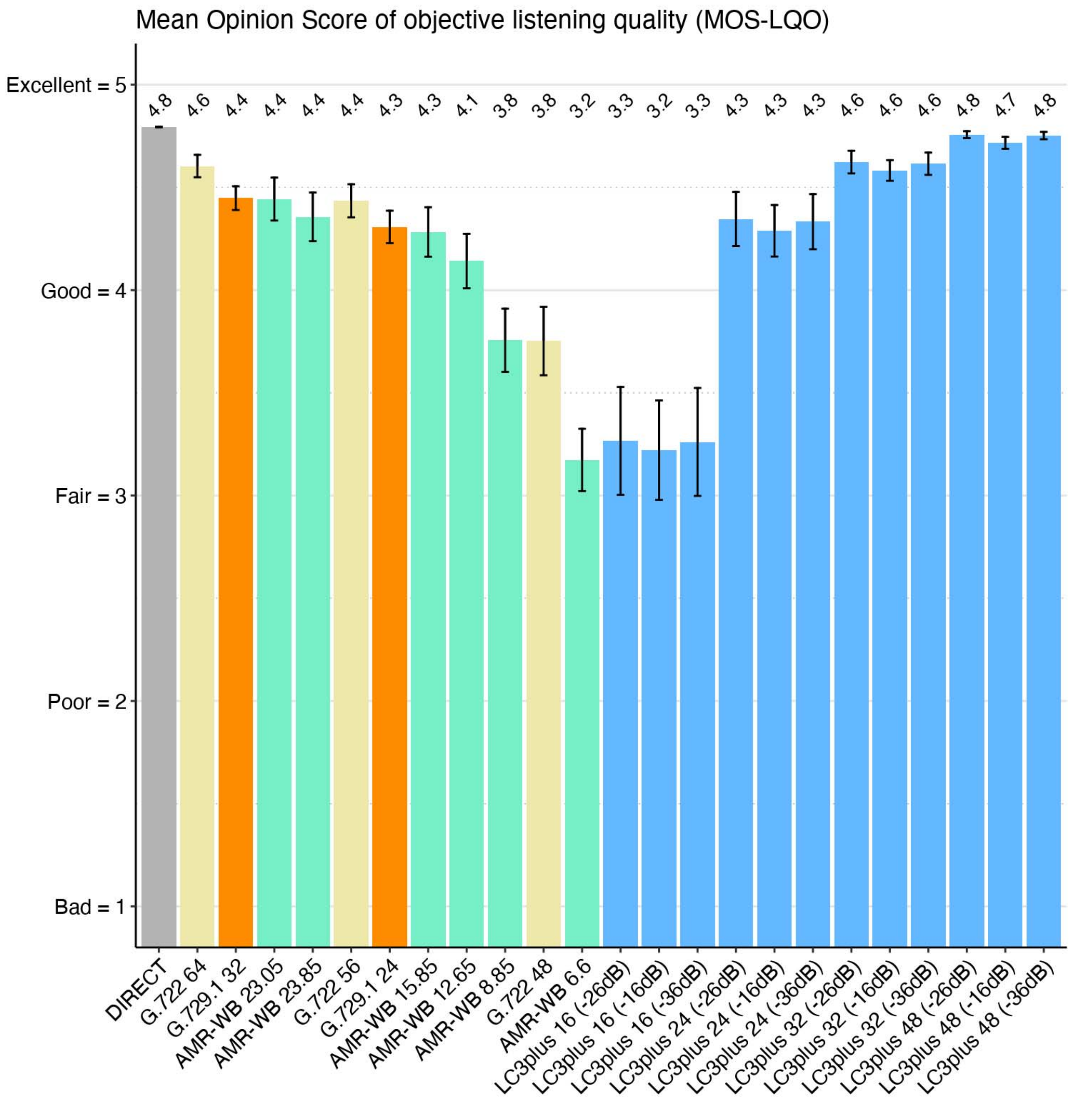

Figure E.19: Recommendation ITU-T P.863 results for WB clean speech signals. Mean scores and 95 % confidence intervals

E.3.2.1.2.2 MOS normalization

To be able to use the same MOS - R transformation as in Recommendation ITU-T P.834 [21] the MOS values from the experiment need to be normalized to fit the narrow band scale in case the maximum value is greater than 4.5 according to:

$$MOS_n = \frac{MOS-1}{MOS_{max}} \cdot 3.5 + 1$$

E.3.2.1.2.3 Scale transformation

All Recommendation ITU-T P.863 test results are transformed from the MOS(NB) scale to the scale of the equipment impairment factor *Ie,WB*. In a first step R is determined by solving the following equations:

$for\ MOS_n = 1.0$: $\quad R_{NB} = 0$

$for\ 1.0 < MOS_n < 4.5{:}\quad MOS_n = 1 + 0.035 \cdot R_{NB} + R_{NB} \cdot (R_{NB} - 60) \cdot (100 - R_{NB}) \cdot 7 \cdot 10^{-6}$

$for\ MOS_n \geq 4.5{:}\qquad R_{NB} = 100$

E.3.2.1.2.4 R scale extension

For WB the R scale is extended by:

$$R_{WB} = R_{NB} \cdot 1.29$$

E.3.2.1.2.5 Calculation of observed impairment factor

The observed impairment factor *Ie,WB,obs* is calculated with the *DIRECT* signal as reference anchor:

$$\text{Ie,WB,obs} = R_{WB}\ \text{(DIRECT)} - R_{WB}\ \text{(test condition)}$$

for all other reference and test conditions depicted in Figure E.19. The values for MOS, MOS_n, R_{WB} and *Ie,WB,obs* and *Ie,WB,def* are listed in Table E.17:

Table E.17: Observed and defined *Ie,WB* reference values and intermediate values

Codec References	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
DIRECT	4.79	4.5	100.0	129.0	0.0	0.0
G.722@64	4.6	4.32	89.42	115.35	13.65	5.0
G.729.1@32	4.45	4.18	84.45	108.94	20.06	7.0
AMR-WB@23.05	4.44	4.18	84.32	108.77	20.23	8.0
AMR-WB@23.85	4.36	4.1	81.98	105.75	23.25	10.0
G.722@56	4.43	4.17	84.08	108.47	20.53	10.0
G.729.1@24	4.31	4.05	80.73	104.14	24.86	16.0
AMR-WB@15.85	4.28	4.03	80.12	103.35	25.65	17.0
AMR-WB@12.65	4.14	3.9	76.81	99.08	29.92	20.0
AMR-WB@8.85	3.76	3.54	68.84	88.8	40.2	41.0
G.722@48	3.75	3.54	68.77	88.71	40.29	41.0
AMR-WB@6.6	3.17	3.0	58.17	75.04	53.96	56.0

E.3.2.1.2.6 Linear interpolation

According to Recommendation ITU-T P.833.1 [24] pairs of defined equipment impairment factors *Ie,WB,def* and observed values *Ie,WB,obs* are plotted in Figure E.20. A linear regression between the measured and the already known variables can be modeled as:

$$\text{Ie,WB,obs} = \text{a}\cdot\text{Ie,WB,def} + \text{b}$$

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,\mathrm{WB},def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j) * (Ie,WB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,obs_j)}{\sum_{i=1}^{n}(Ie,WB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,WB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j$$

with n = 12 reference conditions and Ie,WB,obs and Ie,WB,def taken from Table E.16.

The results of the coefficients a and b are presented in Figure E.20 together with the regression line, approximating all the reference pairs in a least-squares sense. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination R^2 of 0.91 indicates a strong fit of the model. The only condition outside of the confidence interval is the reference.

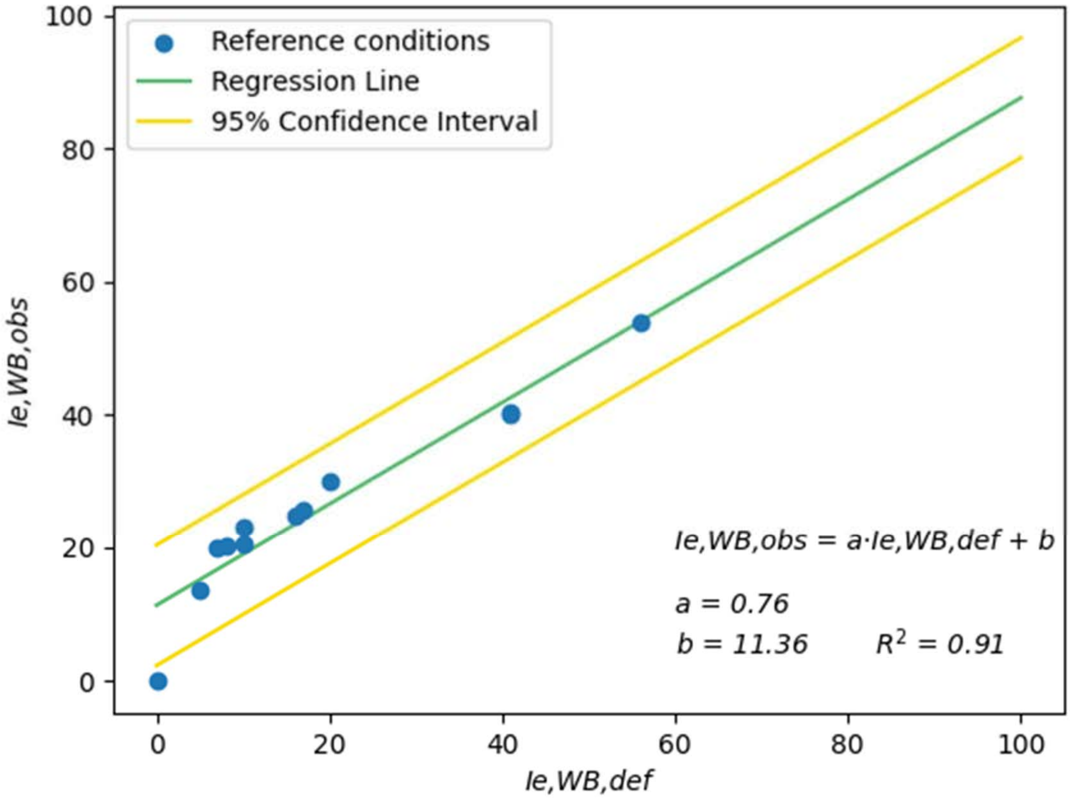

Figure E.20: Linear Regression between observed and defined *Ie,WB*

E.3.2.1.2.7 Determination of a stable *Ie,WB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,WB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,WB,def* for LC3plus based on the parameters a And b. Negative *Ie,WB* values are set to zero. The stable impairment factors are shown in Table E.18 and calculate as follows:

$$Ie,WB,def\,(LC3plus) = \frac{Ie,WB,obs-b}{a}$$

Table E.18: Stable *Ie,WB,def* for LC3plus and intermediate values

Codec under Test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
LC3plus@16	3.27	3.07	59.51	76.77	52.23	53.55
LC3plus@24	4.35	4.07	81.11	104.63	24.37	17.05
LC3plus@32	4.62	4.33	89.53	115.49	13.51	2.82
LC3plus@48	4.76	4.45	95.67	123.41	5.59	0.0

E.3.2.1.2.8 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.19 are encoded and MOS values are shown in Figure E.21. Normalization, scale transformation and extension are performed according to the previous clause to derive *Ie,WB,obs* values for the tandeming conditions.

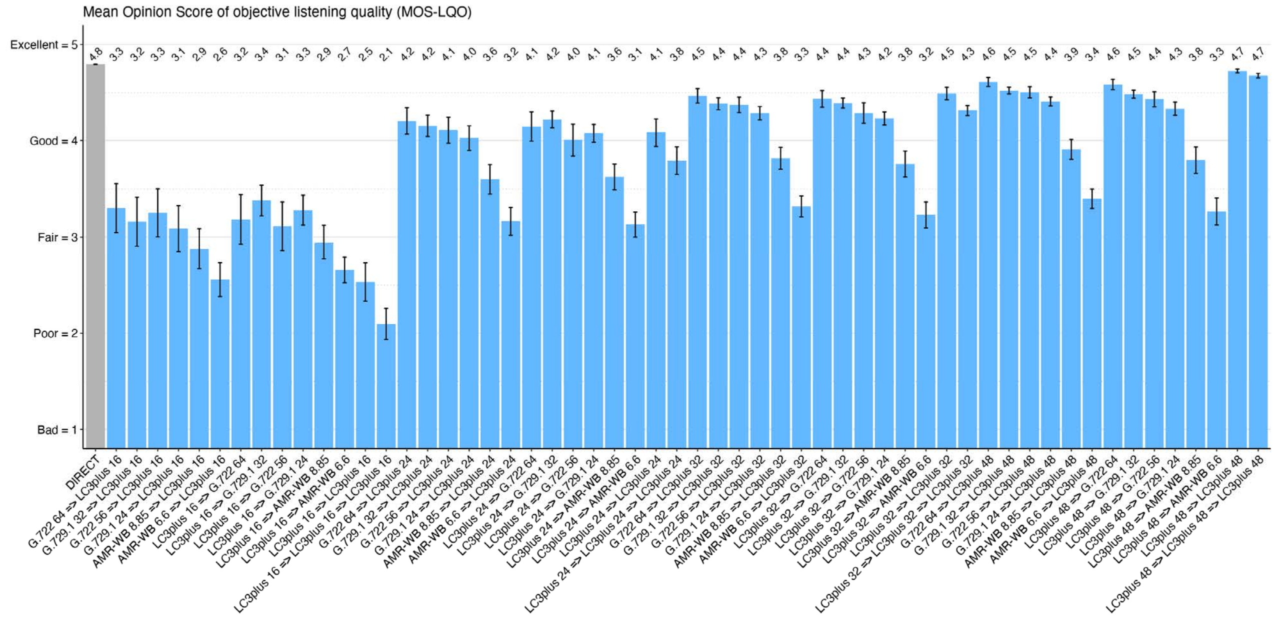

Figure E.21: Recommendation ITU-T P.863 results for WB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

Ie,WB,def values of tandeming of codec1 and codec2 are calculated as the sum of *Ie,WB,def(codec1)* and *Ie,WB,def(codec2)*.

Table E.19: Observed and defined *Ie,WB* values for tandeming and intermediate values

Codec under test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
G.722@64 => LC3plus@16	3.3	3.12	60.43	77.95	51.05	58.55
G.729.1@32 => LC3plus@16	3.16	2.99	57.93	74.73	54.27	60.55
G.722@56 => LC3plus@16	3.25	3.08	59.55	76.82	52.18	63.55
G.729.1@24 => LC3plus@16	3.09	2.93	56.68	73.12	55.88	69.55
AMR-WB@8.85 => LC3plus@16	2.88	2.73	53.0	68.36	60.64	94.55
AMR-WB@6.6 => LC3plus@16	2.56	2.43	47.31	61.03	67.97	109.55
LC3plus@16 => G.722@64	3.18	3.01	58.35	75.27	53.73	58.55
LC3plus@16 => G.729.1@32	3.38	3.19	61.83	79.76	49.24	60.55
LC3plus@16 => G.722@56	3.11	2.95	57.09	73.65	55.35	63.55
LC3plus@16 => G.729.1@24	3.28	3.1	60.04	77.45	51.55	69.55
LC3plus@16 => AMR-WB@8.85	2.95	2.8	54.19	69.9	59.1	94.55
LC3plus@16 => AMR-WB@6.6	2.66	2.53	49.09	63.33	65.67	109.55
LC3plus@16 => LC3plus@16	2.53	2.41	46.88	60.47	68.53	107.1
LC3plus@16 => LC3plus@16 => LC3plus@16	2.1	2.01	38.93	50.22	78.78	160.65
G.722@64 => LC3plus@24	4.21	3.96	78.26	100.96	28.04	22.05
G.729.1@32 => LC3plus@24	4.16	3.91	77.13	99.5	29.5	24.05

Codec under test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
G.722@56 => LC3plus@24	4.11	3.87	76.09	98.15	30.85	27.05
G.729.1@24 => LC3plus@24	4.03	3.79	74.32	95.88	33.12	33.05
AMR-WB@8.85 => LC3plus@24	3.6	3.4	65.85	84.95	44.05	58.05
AMR-WB@6.6 => LC3plus@24	3.16	3.0	58.0	74.82	54.18	73.05
LC3plus@24 => G.722@64	4.15	3.9	76.97	99.29	29.71	22.05
LC3plus@24 => G.729.1@32	4.22	3.97	78.62	101.42	27.58	24.05
LC3plus@24 => G.722@56	4.01	3.77	73.9	95.34	33.66	27.05
LC3plus@24 => G.729.1@24	4.08	3.84	75.4	97.26	31.74	33.05
LC3plus@24 => AMR-WB@8.85	3.62	3.42	66.32	85.55	43.45	58.05
LC3plus@24 => AMR-WB@6.6	3.13	2.97	57.41	74.06	54.94	73.05
LC3plus@24 => LC3plus@24	4.08	3.84	75.54	97.44	31.56	34.1
LC3plus@24 => LC3plus@24 => LC3plus@24	3.79	3.58	69.54	89.7	39.3	51.15
G.722@64 => LC3plus@32	4.47	4.2	85.02	109.68	19.32	7.82
G.729.1@32 => LC3plus@32	4.39	4.12	82.76	106.76	22.24	9.82
G.722@56 => LC3plus@32	4.37	4.11	82.43	106.33	22.67	12.82
G.729.1@24 => LC3plus@32	4.29	4.03	80.26	103.53	25.47	18.82
AMR-WB@8.85 => LC3plus@32	3.82	3.6	70.01	90.31	38.69	43.82
AMR-WB@6.6 => LC3plus@32	3.32	3.14	60.76	78.38	50.62	58.82
LC3plus@32 => G.722@64	4.44	4.17	84.11	108.51	20.49	7.82
LC3plus@32 => G.729.1@32	4.39	4.13	82.93	106.98	22.02	9.82
LC3plus@32 => G.722@56	4.29	4.03	80.29	103.58	25.42	12.82
LC3plus@32 => G.729.1@24	4.23	3.98	78.94	101.83	27.17	18.82
LC3plus@32 => AMR-WB@8.85	3.76	3.54	68.86	88.82	40.18	43.82
LC3plus@32 => AMR-WB@6.6	3.23	3.06	59.18	76.34	52.66	58.82
LC3plus@32 => LC3plus@32	4.49	4.22	85.71	110.57	18.43	5.64
LC3plus@32 => LC3plus@32 => LC3plus@32	4.32	4.06	80.93	104.4	24.6	8.46
G.722@64 => LC3plus@48	4.61	4.33	89.67	115.68	13.32	5.0
G.729.1@32 => LC3plus@48	4.52	4.25	86.65	111.78	17.22	7.0
G.722@56 => LC3plus@48	4.5	4.23	86.13	111.11	17.89	10.0
G.729.1@24 => LC3plus@48	4.41	4.14	83.37	107.54	21.46	16.0
AMR-WB@8.85 => LC3plus@48	3.91	3.68	71.85	92.69	36.31	41.0
AMR-WB@6.6 => LC3plus@48	3.4	3.21	62.17	80.2	48.8	56.0
LC3plus@48 => G.722@64	4.58	4.31	88.74	114.47	14.53	5.0
LC3plus@48 => G.729.1@32	4.48	4.21	85.52	110.32	18.68	7.0
LC3plus@48 => G.722@56	4.43	4.17	83.99	108.35	20.65	10.0
LC3plus@48 => G.729.1@24	4.34	4.08	81.44	105.06	23.94	16.0
LC3plus@48 => AMR-WB@8.85	3.8	3.58	69.65	89.84	39.16	41.0
LC3plus@48 => AMR-WB@6.6	3.26	3.09	59.79	77.13	51.87	56.0
LC3plus@48 => LC3plus@48	4.73	4.44	94.72	122.19	6.81	0.0
LC3plus@48 => LC3plus@48 => LC3plus@48	4.68	4.39	92.31	119.08	9.92	0.0

As done before the pairs of defined and observed *Ie,WB* are plotted against each other in Figure E.22. If more than four out of 14 tandem conditions show major deviations from the interpolation line, the additivity property should not be regarded as being satisfied [25]. This condition is fulfilled for LC3plus at 32 and 48 kbps. For 16 kbps the triple self tandeming of LC3plus appears as biggest outlier but also other data points show major deviations from the regression line. Clearly the observed Ie values are better than expected by addition of *Ie,WB,def.* Therefore *Ie,WB* values for LC3plus at 16 kbps do not proof to be additive within the objective method. Also LC3plus at 24 kbit/s fails to fulfil the condition of additivity for objective evaluation by a narrow margin. For the other LC3plus bitrates the added defined impairment values correspond to the observed/measured values.

The conditions outside of the confidence intervals are:

"AMR-WB 8.85 => LC3plus 16" with an error of -22.88
"AMR-WB 6.6 => LC3plus 16" with an error of -27.0
"LC3plus 16 => G.729.1 24" with an error of -12.89
"LC3plus 16 => AMR-WB 8.85" with an error of -24.42
"LC3plus 16 => AMR-WB 6.6" with an error of -29.3
"LC3plus 16 => LC3plus 16" with an error of -24.57
"LC3plus 16 => LC3plus 16 => LC3plus 16" with an error of -55.19

"AMR-WB 8.85 => LC3plus 24" with an error of -11.61
"AMR-WB 6.6 => LC3plus 24" with an error of -12.93
"LC3plus 24 => AMR-WB 8.85" with an error of -12.21
"LC3plus 24 => AMR-WB 6.6" with an error of -12.17
"LC3plus 24 => LC3plus 24 => LC3plus 24" with an error of -11.1

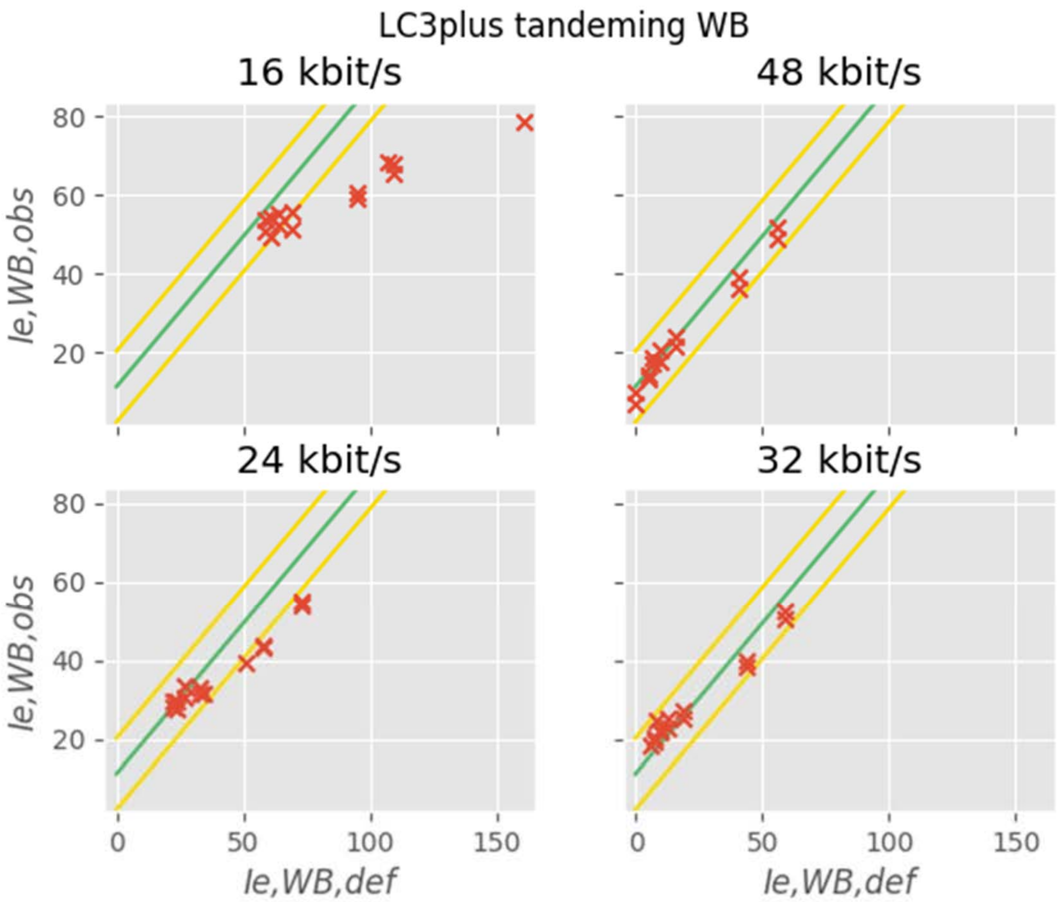

Figure E.22: Observed and defined *Ie,WB* for tandeming conditions with original regression line

E.3.2.1.2.9 Determination of Ie,WB for LC3plus under transmission error conditions

To determine *Ie,WB* values for LC3plus under transmission error conditions the same reference conditions as in Table E.17 plus 16 supplementary reference conditions from Appendix I/G 113 [22] are needed.

Figure E.23 shows the Recommendation ITU-T P.863 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes (EPFsizes).

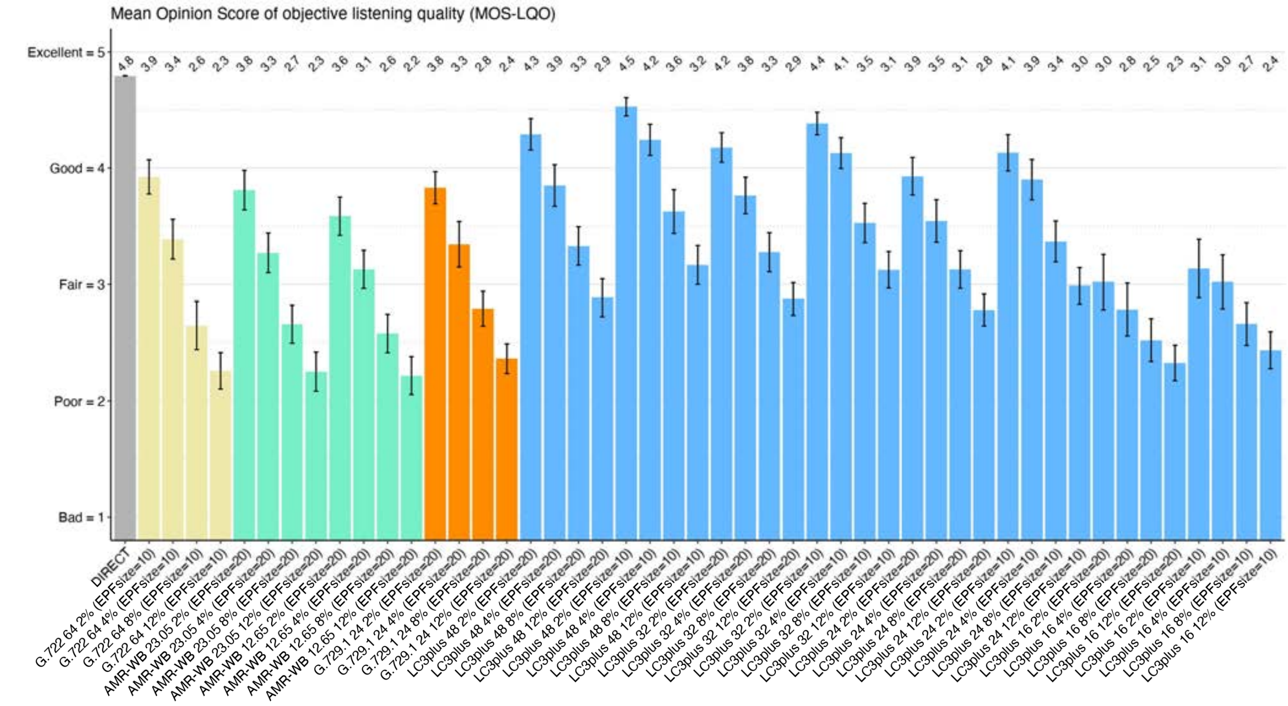

Figure E.23: Recommendation ITU-T P.863 results for WB transmission error speech signals. Mean scores and 95 % confidence intervals

For the error-prone reference conditions an effective equipment impairment factor *Ie,eff,WB* is calculated according to the following equation taken from Recommendation ITU-T G.107.1 [27] (equation 7-15):

$$Ie,eff,WB = Ie,WB + (95 - Ie,WB) \cdot \frac{Ppl}{Ppl+Bpl}$$

The actual values for *Ppl* and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor *Bpl* is taken from [32]. G.722.2 conditions were selected in a way that the given *Ie,WB* values from Recommendation ITU-T P.833.1 [24] are not contradicting the values listed in [32] (Recommendation ITU-T P.833.1 [24] refers to [32]). This way the *Bpl* values from [32] should be consistent with the *Ie,WB* values of these conditions.

Table E.20: Parameters used to calculate *Ie,eff,WB*

Condition	*Ie,WB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,eff,WB*	*Ie,WB,obs*
G.722@64	5.0	2	5.1	2.03	0.98	1.0	30.59	35.91
G.722@64	5.0	4	5.1	4.05	0.96	1.0	44.85	49.03
G.722@64	5.0	8	5.1	8.1	0.92	1.0	60.24	65.8
G.722@64	5.0	12	5.1	12.17	0.88	1.0	68.42	74.84
AMR-WB@23.05	8.0	2	4.6	1.96	0.97	1.01	33.98	38.83
AMR-WB@23.05	8.0	4	4.6	3.92	0.94	1.02	48.01	51.74
AMR-WB@23.05	8.0	8	4.6	7.82	0.92	1.0	62.79	65.55
AMR-WB@23.05	8.0	12	4.6	11.73	0.87	1.01	70.49	75.0
AMR-WB@12.65	20.0	2	4.3	1.96	0.97	1.01	43.47	44.36
AMR-WB@12.65	20.0	4	4.3	3.92	0.94	1.02	55.75	54.98
AMR-WB@12.65	20.0	8	4.3	7.82	0.92	1.0	68.4	67.39

Condition	*Ie,WB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,eff,WB*	*Ie,WB,obs*
AMR-WB@12.65	20.0	12	4.3	11.73	0.87	1.01	74.88	75.83
G.729.1@24	16.0	2	7.3	1.96	0.97	1.01	32.71	38.32
G.729.1@24	16.0	4	7.3	3.92	0.94	1.02	43.59	50.04
G.729.1@24	16.0	8	7.3	7.82	0.92	1.0	56.87	62.59
G.729.1@24	16.0	12	7.3	11.73	0.87	1.01	64.69	72.39

Next the observed *Ie,WB* values are calculated according to the previous clause. The results listed in Table E.20 show increasing impairment factors for increasing transmission error rates. The new pairs of *Ie,WB,obs* and *Ie,eff,WB* are added in the scatterplot Figure E.24. It can be observed that the original interpolation line still describes the new error-prone reference conditions adequately with a strong fit of $R^2 = 0.93$. At higher error rates a light deviation outside the confidence interval can be observed.

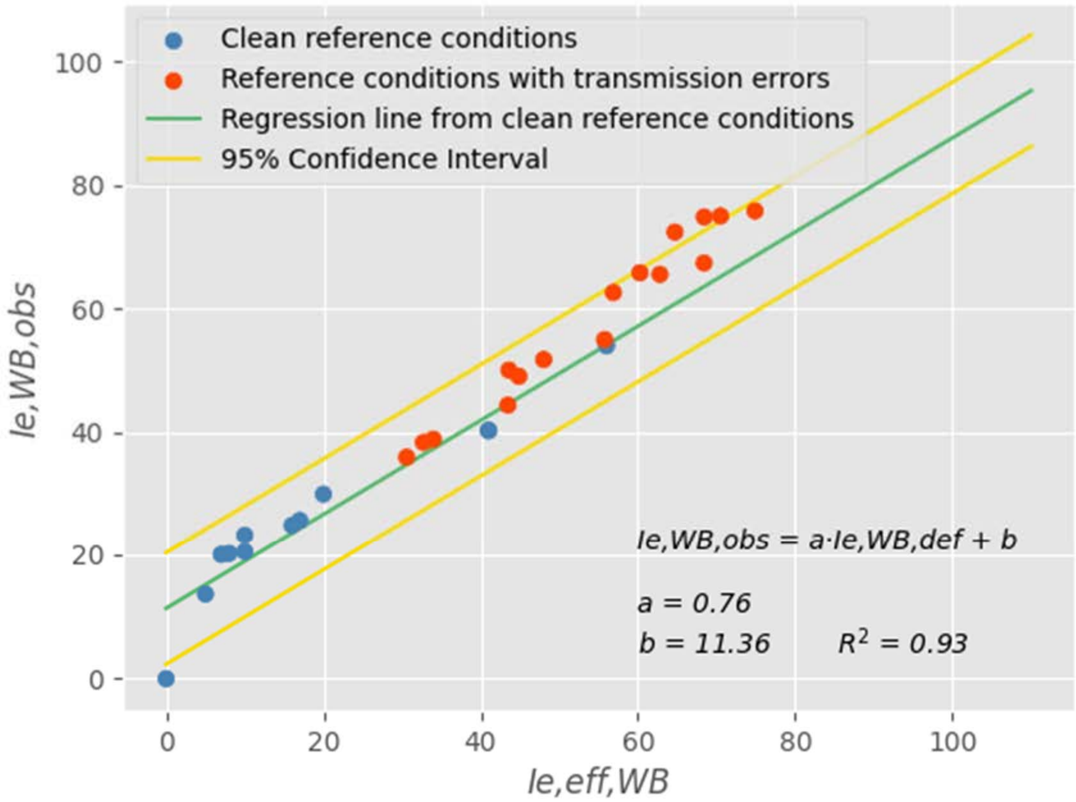

Figure E.24: Original regression line, observed *Ie,WB,obs* and defined *Ie,eff,WB* for clean and error-prone reference conditions

With the coefficients a and b stable *Ie,WB* values for LC3plus with transmission errors can be derived (Table E.21) without existing *Bpl* values.

Table E.21: Stable *Ie,WB,def* values for LC3plus under transmission error conditions

Codec under Test	*Ie,WB,obs*	*Ie,WB,def*
LC3plus@48 2% (EPFsize=20)	25.43	18.44
LC3plus@48 4% (EPFsize=20)	37.81	34.66
LC3plus@48 8% (EPFsize=20)	50.38	51.13
LC3plus@48 12% (EPFsize=20)	60.46	64.33
LC3plus@48 2% (EPFsize=10)	16.97	7.35
LC3plus@48 4% (EPFsize=10)	26.9	20.36
LC3plus@48 8% (EPFsize=10)	43.38	41.96
LC3plus@48 12% (EPFsize=10)	54.11	56.01
LC3plus@32 2% (EPFsize=20)	28.87	22.94
LC3plus@32 4% (EPFsize=20)	39.99	37.51
LC3plus@32 8% (EPFsize=20)	51.62	52.75
LC3plus@32 12% (EPFsize=20)	60.7	64.65
LC3plus@32 2% (EPFsize=10)	22.35	14.4
LC3plus@32 4% (EPFsize=10)	30.27	24.78
LC3plus@32 8% (EPFsize=10)	45.76	45.07
LC3plus@32 12% (EPFsize=10)	55.07	57.27
LC3plus@24 2% (EPFsize=20)	35.74	31.95
LC3plus@24 4% (EPFsize=20)	45.33	44.51
LC3plus@24 8% (EPFsize=20)	55.01	57.19
LC3plus@24 12% (EPFsize=20)	62.85	67.47
LC3plus@24 2% (EPFsize=10)	30.21	24.7
LC3plus@24 4% (EPFsize=10)	36.53	32.98
LC3plus@24 8% (EPFsize=10)	49.48	49.95
LC3plus@24 12% (EPFsize=10)	58.16	61.32
LC3plus@16 2% (EPFsize=20)	57.44	60.38
LC3plus@16 4% (EPFsize=20)	62.73	67.31
LC3plus@16 8% (EPFsize=20)	68.68	75.1
LC3plus@16 12% (EPFsize=20)	73.25	81.09
LC3plus@16 2% (EPFsize=10)	54.81	56.93
LC3plus@16 4% (EPFsize=10)	57.38	60.3
LC3plus@16 8% (EPFsize=10)	65.52	70.96
LC3plus@16 12% (EPFsize=10)	70.68	77.72

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern Frame sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,WB* curve the best. The optimal value can be determined by calculating the least square error for every four pairs of *Ie,WB,obs(Ppl)* and *Ie,WB,eff(Ppl)*.

Table E.22: Packet loss robustness factors for LC3plus

Bitrate (kbps)	EPFsize (ms)	*Ie,WB,def*	*Bpl*
48	20	0.0	7.0
48	10	0.0	11.0
32	20	2.82	7.0
32	10	2.82	10.2
24	20	17.05	7.3
24	10	17.05	11.6
16	20	53.55	7.4
16	10	53.55	12.0

Figure E.25 and Figure E.26 show *Ie,eff,WB(Ppl)* with fixed *Bpl* from Table E.22. The approximations using the *Ie, eff,WB* formula vary in accuracy, depending on bitrate and EPFsize but all show a strong fit according to R^2. It can be observed that the fitting is better for 20 ms EPFsize.

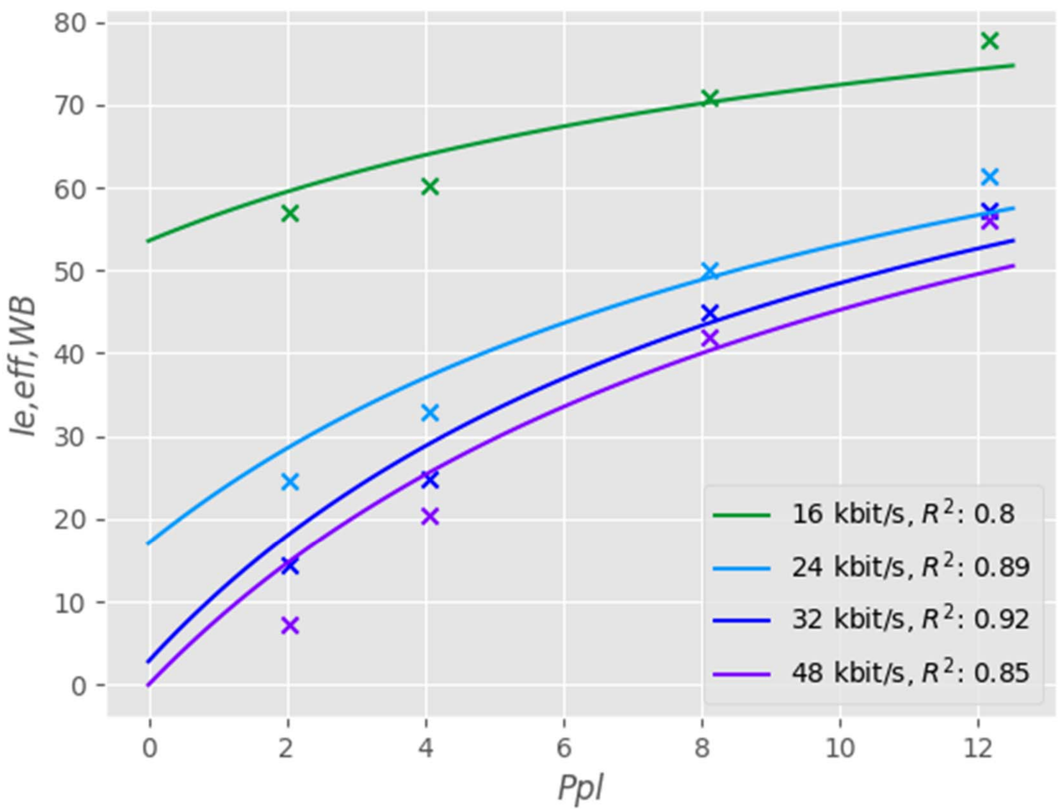

Figure E.25: *Ie,eff,WB* approximation with best fitting *Bpl* and observed *Ie* values vs. packet loss rate for 10 ms EPFsize

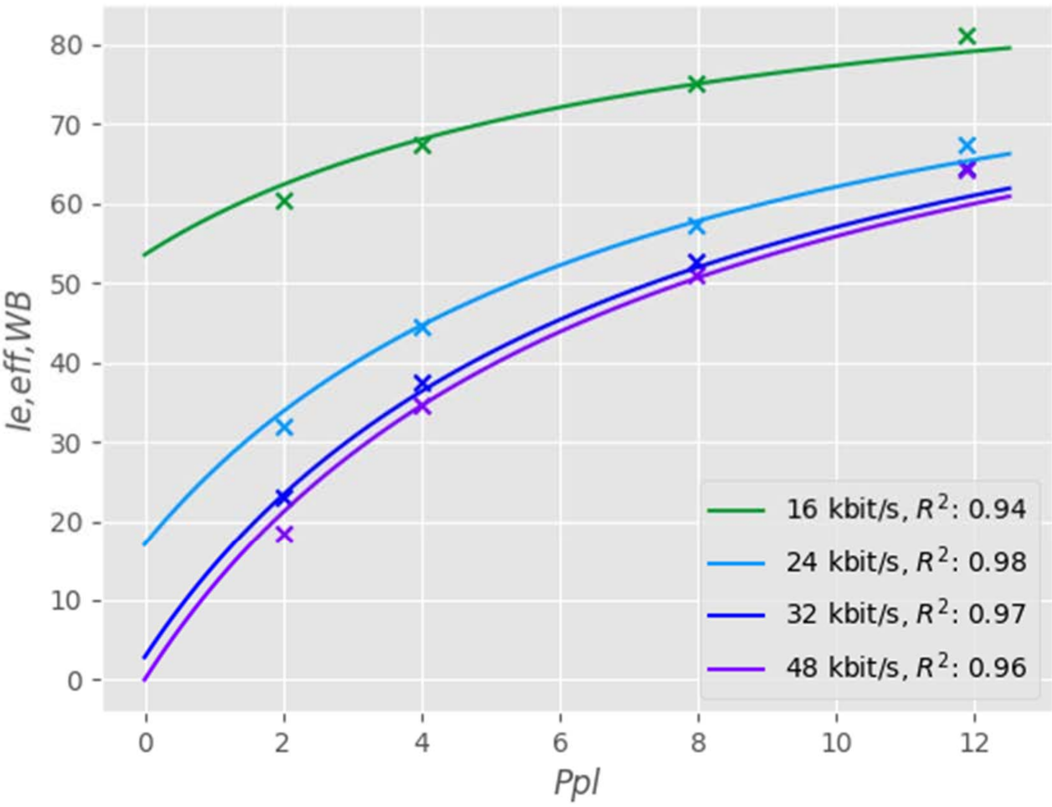

Figure E.26: *Ie,eff,WB* approximation with best fitting *Bpl* and observed *Ie* values vs. packet loss rate for 20 ms EPFsize

E.3.2.2 Subjective assessment

E.3.2.2.1 Introduction

This clause describes the derivation of the equipment impairment factors *Ie,WB* and the packet loss robustness factor Bpl for LC3plus in wideband mode. It is determined according to Recommendation ITU-T P.833.1 [24]. MOS values are obtained by performing P.800 [1] listening tests according to considerations in clauses E.1.3 and E.1.4.

E.3.2.2.2 Determination of *Ie,WB* for LC3plus in error-free conditions

E.3.2.2.2.1 Measuring MOS

Figure E.27 shows the MOS of the 12 reference conditions and LC3plus configurations under test with -16, -26 and -36 dBOV bitrates of 16, 24, 32 and 48 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

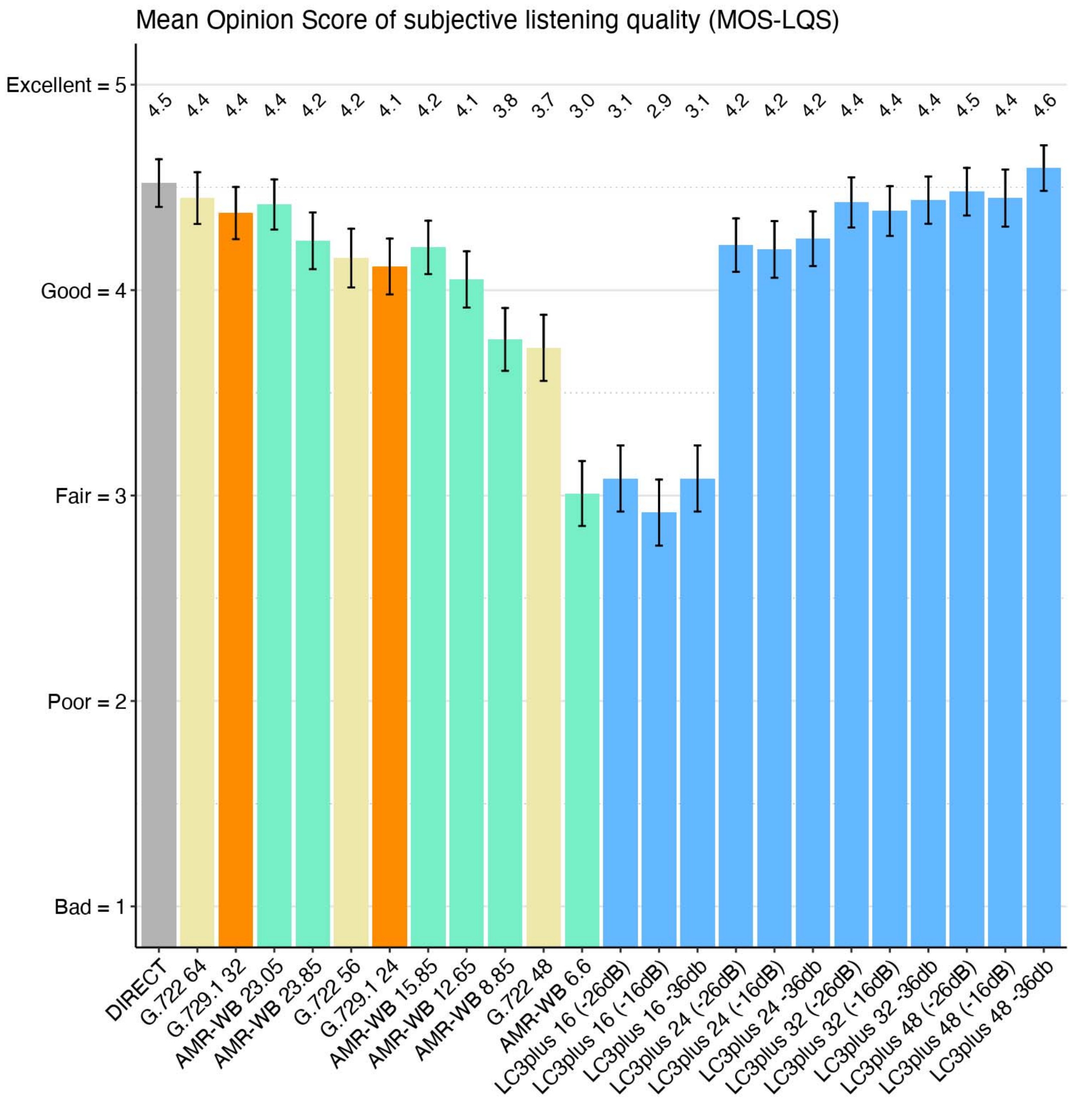

Figure E.27: Recommendation ITU-T P.800 results for WB clean speech signals. Mean scores and 95 % confidence intervals

E.3.2.2.2.2 MOS normalization

To be able to use the same MOS - R transformation as in Recommendation ITU-T P.833.1 [24] the MOS values from the experiment need to be normalized to fit the narrow band scale in case the maximum value is greater than 4.5 according to:

$$MOS_n = \frac{MOS-1}{MOS_{max}} \cdot 3.5 + 1$$

E.3.2.2.2.3 Scale transformation

All Recommendation ITU-T P.800 test results are transformed from the MOS(NB) scale to the scale of the equipment impairment factor *Ie,WB*. In a first step R is determined by solving the following equations:

$for\ MOS_n = 1.0$: $\quad R_{NB} = 0$

$$for\ 1.0 < MOS_n < 4.5: \quad MOS_n = 1 + 0.035 \cdot R_{NB} + R_{NB} \cdot (R_{NB} - 60) \cdot (100 - R_{NB}) \cdot 7 \cdot 10^{-6}$$

$$for\ MOS_n \geq 4.5: \qquad R_{NB} = 100$$

E.3.2.2.2.4 R scale extension

For WB the R scale is extended by:

$$R_{WB} = R_{NB} \cdot 1.29$$

E.3.2.2.2.5 Calculation of observed impairment factor

The observed impairment factor *Ie,WB,obs* is calculated with the *DIRECT* signal as reference anchor:

Ie,WB,obs = R_{WB} (DIRECT) - R_{WB} (test condition)

for all other reference and test conditions depicted in Figure E.27. The values for MOS, MOS_n, R_{WB} and *Ie,WB,obs* and *Ie,WB,def* are listed in Table E.23:

Table E.23: Observed and defined *Ie,WB* reference values and intermediate values

Codec References	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
DIRECT	4.52	4.5	100.0	129.0	0.0	0.0
G.722@64	4.45	4.43	94.18	121.49	7.51	5.0
G.729.1@32	4.38	4.35	90.66	116.95	12.05	7.0
AMR-WB@23.05	4.42	4.4	92.56	119.41	9.59	8.0
AMR-WB@23.85	4.24	4.22	85.72	110.58	18.42	10.0
G.722@56	4.16	4.14	83.15	107.26	21.74	10.0
G.729.1@24	4.12	4.1	81.98	105.75	23.25	16.0
AMR-WB@15.85	4.21	4.19	84.71	109.28	19.72	17.0
AMR-WB@12.65	4.05	4.03	80.26	103.54	25.46	20.0
AMR-WB@8.85	3.76	3.74	73.21	94.44	34.56	41.0
G.722@48	3.72	3.7	72.3	93.26	35.74	41.0
AMR-WB@6.6	3.01	3.0	58.04	74.87	54.13	56.0

E.3.2.2.2.6 Linear interpolation

According to Recommendation ITU-T P.834.1 [25] pairs of defined equipment impairment factors *Ie,WB,def* and observed values *Ie,WB,obs* are plotted in Figure E.28. A linear regression between the measured and the already known variables can be modeled as:

Ie,WB,obs = a·Ie,WB,def + b

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,WB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j) * (Ie,WB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,obs_j)}{\sum_{i=1}^{n}(Ie,WB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,WB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,WB,def_j$$

with n = 12 reference conditions and Ie,WB,obs and Ie,WB,def taken from Table E.23.

The results of the coefficients a and b are presented in Figure E.28 together with the regression line, approximating all the reference pairs in a least-squares sense. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination $R^2$of 0.92 indicates a strong fit of the model.

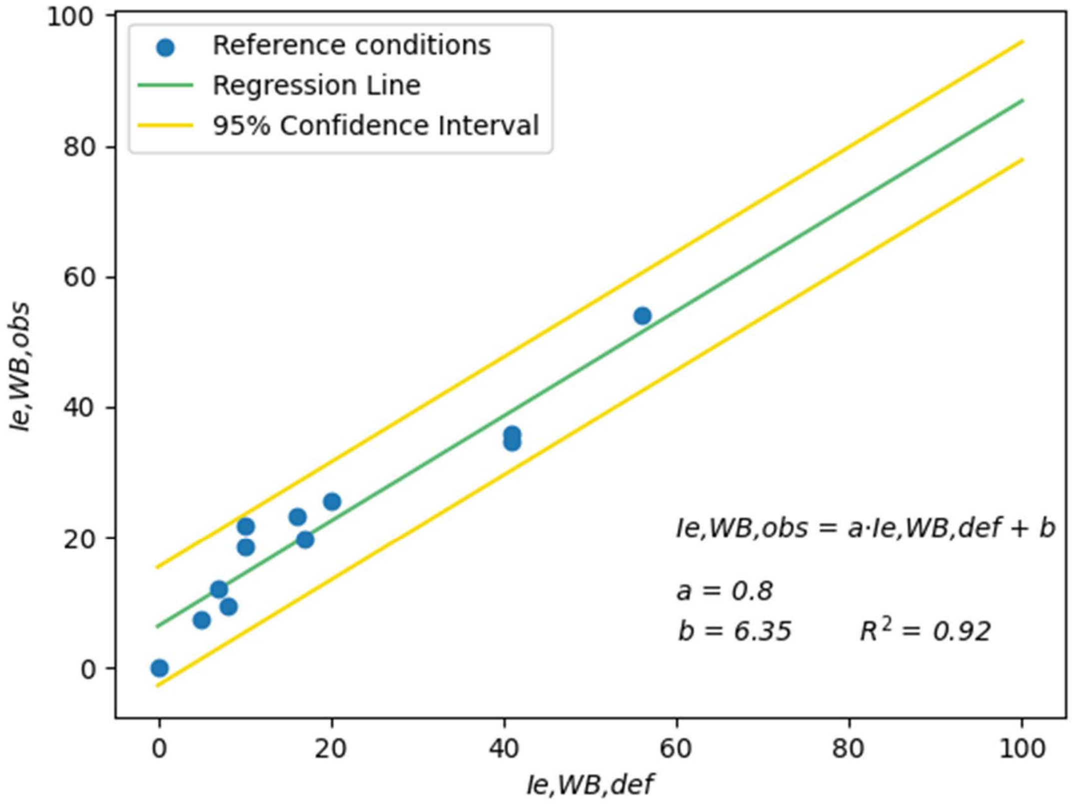

Figure E.28: Linear Regression between observed and defined *Ie,WB*

E.3.2.2.2.7 Determination of a stable *Ie,WB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,WB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,WB,def* for LC3plus based on the parameters a And b. Negative *Ie,WB* values are set to zero. The stable impairment factors are shown in Table E.24 and calculate as follows:

$$Ie, WB, def\,(LC3plus) = \frac{Ie{,}WB{,}obs - b}{a}$$

Table E.24: Stable *Ie,WB,def* for LC3plus and intermediate values

Codec under Test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
LC3plus@16	3.08	3.02	58.38	75.31	53.69	58.81
LC3plus@24	4.22	4.2	85.16	109.85	19.15	15.9
LC3plus@32	4.43	4.4	92.55	119.39	9.61	4.04
LC3plus@48	4.48	4.49	98.38	126.91	2.09	0.0

E.3.2.2.2.8 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.25 are encoded and MOS values are measured as shown in Figure E.29. Normalization, scale transformation and extension are performed according to the previous clause to derive *Ie,WB,obs* values for the tandeming conditions.

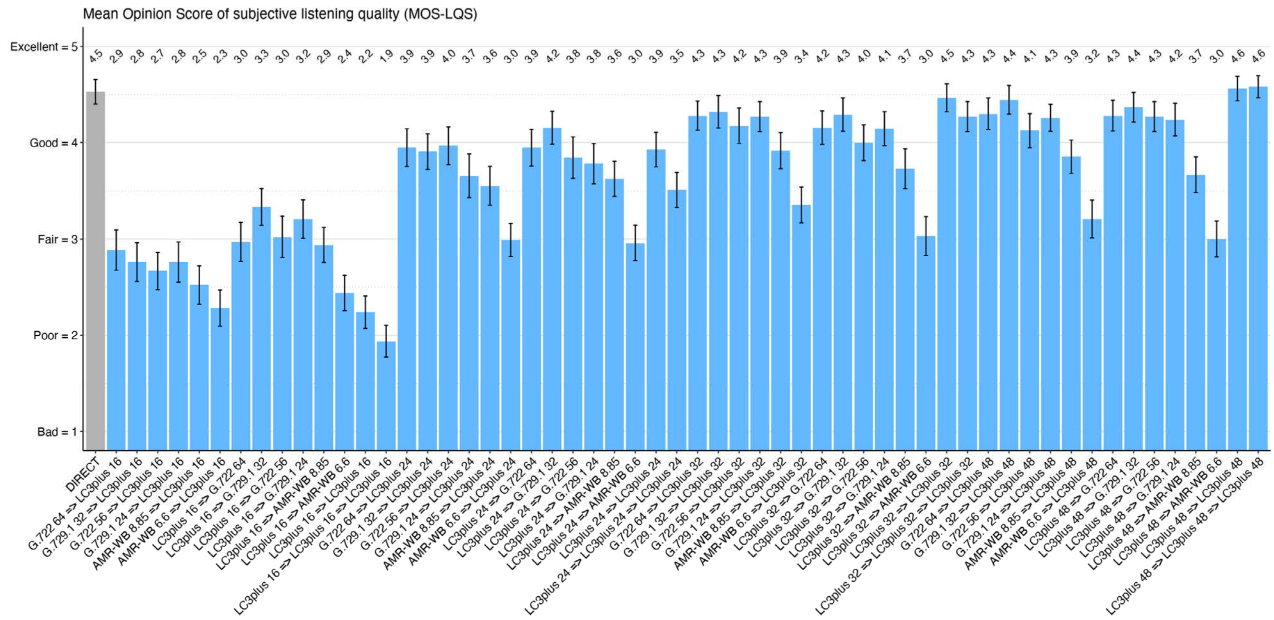

Figure E.29: Recommendation ITU-T P.800 results for WB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

Ie,WB,def values of tandeming of codec1 and codec2 are calculated as the sum of *Ie,WB,def(codec1)* and *Ie,WB,def(codec2)*.

Table E.25: Observed and defined *Ie,WB* values for tandeming and intermediate values

Codec under test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
G.722@64 => LC3plus@16	2.88	2.84	55.06	71.02	52.17	63.81
G.729.1@32 => LC3plus@16	2.76	2.72	52.74	68.03	55.15	65.81
G.722@56 => LC3plus@16	2.67	2.63	51.02	65.81	57.38	68.81
G.729.1@24 => LC3plus@16	2.76	2.72	52.74	68.03	55.15	74.81
AMR-WB@8.85 => LC3plus@16	2.52	2.49	48.3	62.3	60.89	99.81
AMR-WB@6.6 => LC3plus@16	2.28	2.25	43.75	56.44	66.75	114.81
LC3plus@16 => G.722@64	2.97	2.92	56.62	73.03	50.15	63.81
LC3plus@16 => G.729.1@32	3.33	3.28	63.49	81.91	41.28	65.81
LC3plus@16 => G.722@56	3.02	2.97	57.58	74.28	48.9	68.81
LC3plus@16 => G.729.1@24	3.21	3.16	61.1	78.82	44.37	74.81
LC3plus@16 => AMR-WB@8.85	2.94	2.89	56.04	72.29	50.9	99.81
LC3plus@16 => AMR-WB@6.6	2.44	2.4	46.74	60.29	62.9	114.81
LC3plus@16 => LC3plus@16	2.24	2.21	42.96	55.42	67.77	117.62

Codec under test	MOS	MOS_n	$\mathrm{R_{NB}}$	$\mathrm{R_{WB}}$	***Ie,WB,obs***	***Ie,WB,def***
LC3plus@16 => LC3plus@16 => LC3plus@16	1.94	1.92	36.93	47.64	75.55	176.43
G.722@64 => LC3plus@24	3.95	3.88	76.37	98.52	24.67	20.9
G.729.1@32 => LC3plus@24	3.91	3.84	75.39	97.26	25.93	22.9
G.722@56 => LC3plus@24	3.97	3.9	76.87	99.16	24.03	25.9
G.729.1@24 => LC3plus@24	3.66	3.59	69.95	90.23	32.96	31.9
AMR-WB@8.85 => LC3plus@24	3.55	3.49	67.81	87.48	35.71	56.9
AMR-WB@6.6 => LC3plus@24	2.99	2.94	57.01	73.54	49.65	71.9
LC3plus@24 => G.722@64	3.95	3.88	76.37	98.52	24.67	20.9
LC3plus@24 => G.729.1@32	4.16	4.08	81.6	105.26	17.93	22.9
LC3plus@24 => G.722@56	3.84	3.78	73.99	95.45	27.74	25.9
LC3plus@24 => G.729.1@24	3.78	3.72	72.6	93.66	29.53	31.9
LC3plus@24 => AMR-WB@8.85	3.62	3.56	69.3	89.4	33.79	56.9
LC3plus@24 => AMR-WB@6.6	2.96	2.91	56.41	72.77	50.42	71.9
LC3plus@24 => LC3plus@24	3.93	3.86	75.88	97.88	25.3	31.8
LC3plus@24 => LC3plus@24 => LC3plus@24	3.51	3.45	66.97	86.39	36.8	47.7
G.722@64 => LC3plus@32	4.28	4.2	85.22	109.93	13.26	9.04
G.729.1@32 => LC3plus@32	4.32	4.25	86.56	111.67	11.52	11.04
G.722@56 => LC3plus@32	4.18	4.1	82.17	106.0	17.19	14.04
G.729.1@24 => LC3plus@32	4.27	4.2	84.91	109.53	13.66	20.04
AMR-WB@8.85 => LC3plus@32	3.92	3.85	75.65	97.58	25.6	45.04
AMR-WB@6.6 => LC3plus@32	3.35	3.3	63.9	82.43	40.76	60.04
LC3plus@32 => G.722@64	4.16	4.08	81.6	105.26	17.93	9.04
LC3plus@32 => G.729.1@32	4.29	4.22	85.56	110.38	12.81	11.04
LC3plus@32 => G.722@56	4.0	3.93	77.61	100.12	23.07	14.04
LC3plus@32 => G.729.1@24	4.15	4.07	81.33	104.91	18.28	20.04
LC3plus@32 => AMR-WB@8.85	3.73	3.67	71.48	92.21	30.97	45.04
LC3plus@32 => AMR-WB@6.6	3.03	2.98	57.77	74.53	48.66	60.04
LC3plus@32 => LC3plus@32	4.47	4.39	92.18	118.91	4.28	8.08
LC3plus@32 => LC3plus@32 => LC3plus@32	4.27	4.2	84.91	109.53	13.66	12.12
G.722@64 => LC3plus@48	4.3	4.23	85.88	110.79	12.4	5.0
G.729.1@32 => LC3plus@48	4.45	4.37	91.24	117.7	5.49	7.0
G.722@56 => LC3plus@48	4.12	4.05	80.77	104.19	19.0	10.0
G.729.1@24 => LC3plus@48	4.26	4.18	84.57	109.1	14.09	16.0
AMR-WB@8.85 => LC3plus@48	3.85	3.79	74.21	95.74	27.45	41.0
AMR-WB@6.6 => LC3plus@48	3.21	3.16	61.1	78.82	44.37	56.0
LC3plus@48 => G.722@64	4.28	4.2	85.22	109.93	13.26	5.0
LC3plus@48 => G.729.1@32	4.37	4.29	88.18	113.75	9.43	7.0
LC3plus@48 => G.722@56	4.27	4.2	84.91	109.53	13.66	10.0
LC3plus@48 => G.729.1@24	4.24	4.16	83.97	108.33	14.86	16.0
LC3plus@48 => AMR-WB@8.85	3.67	3.61	70.18	90.53	32.66	41.0
LC3plus@48 => AMR-WB@6.6	3.0	2.95	57.19	73.78	49.41	56.0

Codec under test	MOS	MOS_n	R_{NB}	R_{WB}	*Ie,WB,obs*	*Ie,WB,def*
LC3plus@48 => LC3plus@48	4.56	4.48	97.85	126.22	-3.04	0.0
LC3plus@48 => LC3plus@48 => LC3plus@48	4.58	4.5	100.0	129.0	-5.81	0.0

As done before the pairs of defined and observed *Ie,WB* are plotted against each other in Figure E.30. If more than four out of 14 tandem conditions show major deviations from the interpolation line, the additivity property should not be regarded as being satisfied [25]. This condition is fulfilled for LC3plus at 24, 32 and 48 kbp/s. For 16 kbps the triple self tandeming of LC3plus appears as biggest outlier but also other data points show major deviations from the regression line. Clearly the observed Ie values are better than expected by addition of *Ie,WB,def*. Therefore *Ie,WB* values for LC3plus at 16 kbps do not proof to be additive within the subjective method. For the other LC3plus bitrates the added defined impairment values correspond to the observed/measured values.

The conditions outside of the confidence intervals are:

"G.729.1 24 => LC3plus 16" with an error of -11.41
"AMR-WB 8.85 => LC3plus 16" with an error of -25.79
"AMR-WB 6.6 => LC3plus 16" with an error of -32.01
"LC3plus 16 => G.729.1 32" with an error of -18.04
"LC3plus 16 => G.722 56" with an error of -12.83
"LC3plus 16 => G.729.1 24" with an error of -22.19
"LC3plus 16 => AMR-WB 8.85" with an error of -35.78
"LC3plus 16 => AMR-WB 6.6" with an error of -35.86
"LC3plus 16 => LC3plus 16" with an error of -33.25
"LC3plus 16 => LC3plus 16 => LC3plus 16" with an error of -72.8

"AMR-WB 8.85 => LC3plus 24" with an error of -16.44
"AMR-WB 6.6 => LC3plus 24" with an error of -14.57
"LC3plus 24 => AMR-WB 8.85" with an error of -18.36
"LC3plus 24 => AMR-WB 6.6" with an error of -13.8

"AMR-WB 8.85 => LC3plus 32" with an error of -17.0
"AMR-WB 6.6 => LC3plus 32" with an error of -13.92
"LC3plus 32 => AMR-WB 8.85" with an error of -11.63

"AMR-WB 8.85 => LC3plus 48" with an error of -11.9
"LC3plus 48 => LC3plus 48" with an error of -9.39
"LC3plus 48 => LC3plus 48 => LC3plus 48" with an error of -12.16

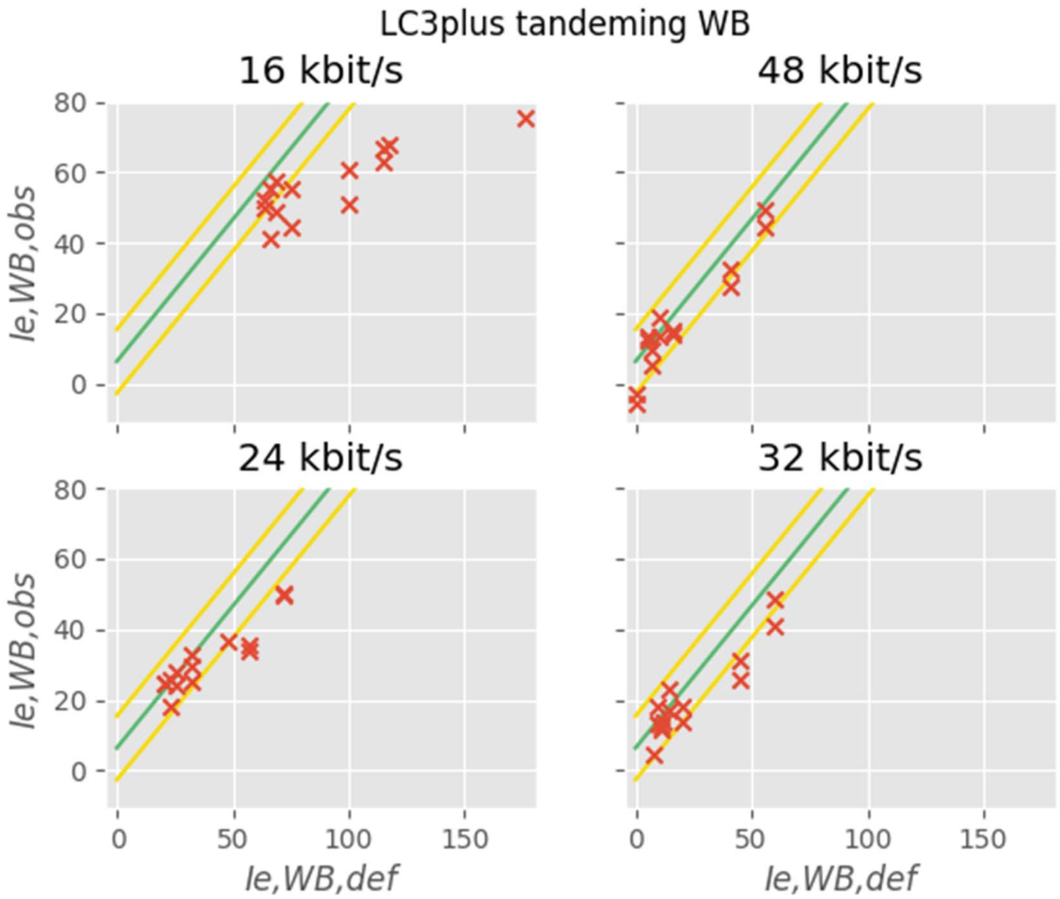

Figure E.30: Observed and defined *Ie,WB* for tandeming conditions with original regression line

E.3.2.2.2.9 Determination of *Ie,WB* for LC3plus under transmission error conditions

To determine *Ie,WB* values for LC3plus under transmission error conditions the same reference conditions as in Table E.23 plus 16 supplementary reference conditions from Appendix I/G 113 [22] are needed.

Figure E.31 shows the Recommendation ITU-T P.800 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes (EPFsizes).

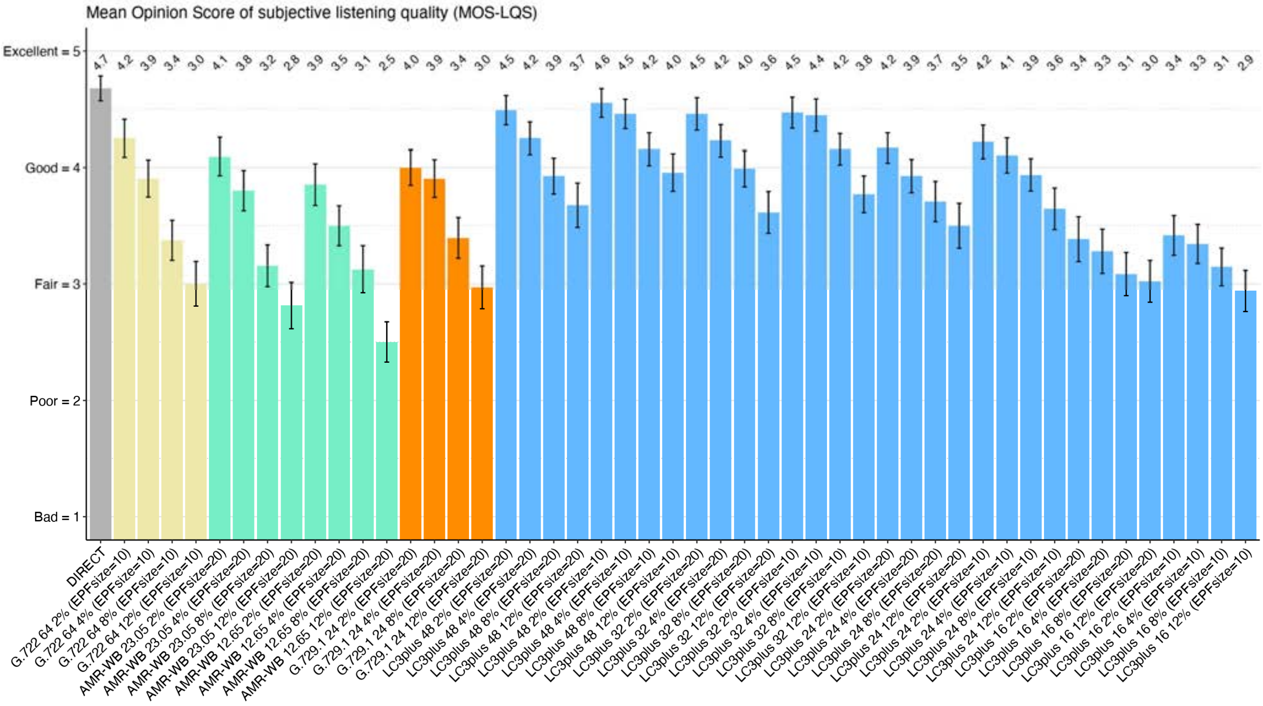

Figure E.31: Recommendation ITU-T P.800 results for WB transmission error speech signals. Mean scores and 95 % confidence intervals

For the error-prone reference conditions an effective equipment impairment factor *Ie,eff,WB* is calculated according to the following equation taken from Recommendation ITU-T G.107.1 [27] (equation 7-15):

$$Ie,eff,WB = Ie,WB + (95 - Ie,WB) \cdot \frac{Ppl}{Ppl+Bpl}$$

The actual values for *Ppl* and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor *Bpl* is taken from [32]. G.722.2 conditions were selected in a way that the given *Ie,WB* values from Recommendation ITU-T P.833.1 [24] are not contradicting the values listed in [32] (Recommendation ITU-T P.833.1 [24] refers to [32]). This way the *Bpl* values from [32] should be consistent with the *Ie,WB* values of these conditions.

Table E.26: Parameters used to calculate *Ie,eff,WB*

Condition	*Ie,WB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,eff,WB*	*Ie,WB,obs*
G.722@64	5.0	2	5.1	2.03	0.98	1.0	30.59	23.36
G.722@64	5.0	4	5.1	4.05	0.96	1.0	44.85	33.91
G.722@64	5.0	8	5.1	8.1	0.92	1.0	60.24	47.56
G.722@64	5.0	12	5.1	12.17	0.88	1.0	68.42	56.45
AMR-WB@23.05	8.0	2	4.6	1.96	0.97	1.01	33.98	28.42
AMR-WB@23.05	8.0	4	4.6	3.92	0.94	1.02	48.01	36.75
AMR-WB@23.05	8.0	8	4.6	7.82	0.92	1.0	62.79	52.79

Condition	*Ie,WB*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	q (measured)	BurstR	*Ie,eff,WB*	*Ie,WB,obs*
AMR-WB@23.05	8.0	12	4.6	11.73	0.87	1.01	70.49	60.81
AMR-WB@23.65	20.0	2	4.3	1.96	0.97	1.01	43.47	35.34
AMR-WB@12.65	20.0	4	4.3	3.92	0.94	1.02	55.75	44.5
AMR-WB@12.65	20.0	8	4.3	7.82	0.92	1.0	68.4	53.52
AMR-WB@12.65	20.0	12	4.3	11.73	0.87	1.01	74.88	68.13
G.729.1@24	16.0	2	7.3	1.96	0.97	1.01	32.71	31.23
G.729.1@24	16.0	4	7.3	3.92	0.94	1.02	43.59	33.91
G.729.1@24	16.0	8	7.3	7.82	0.92	1.0	56.87	47.05
G.729.1@24	16.0	12	7.3	11.73	0.87	1.01	64.69	57.17

Next the observed *Ie,WB* values are calculated according to the previous clause. The results listed in Table E.26 show increasing impairment factors for increasing transmission error rates. The new pairs of *Ie,WB,obs* and *Ie,eff,WB* are added in the scatterplot Figure E.32. It can be observed that the original interpolation line still describes the new error-prone reference conditions adequately with a strong fit of $R^2 = 0.91$.

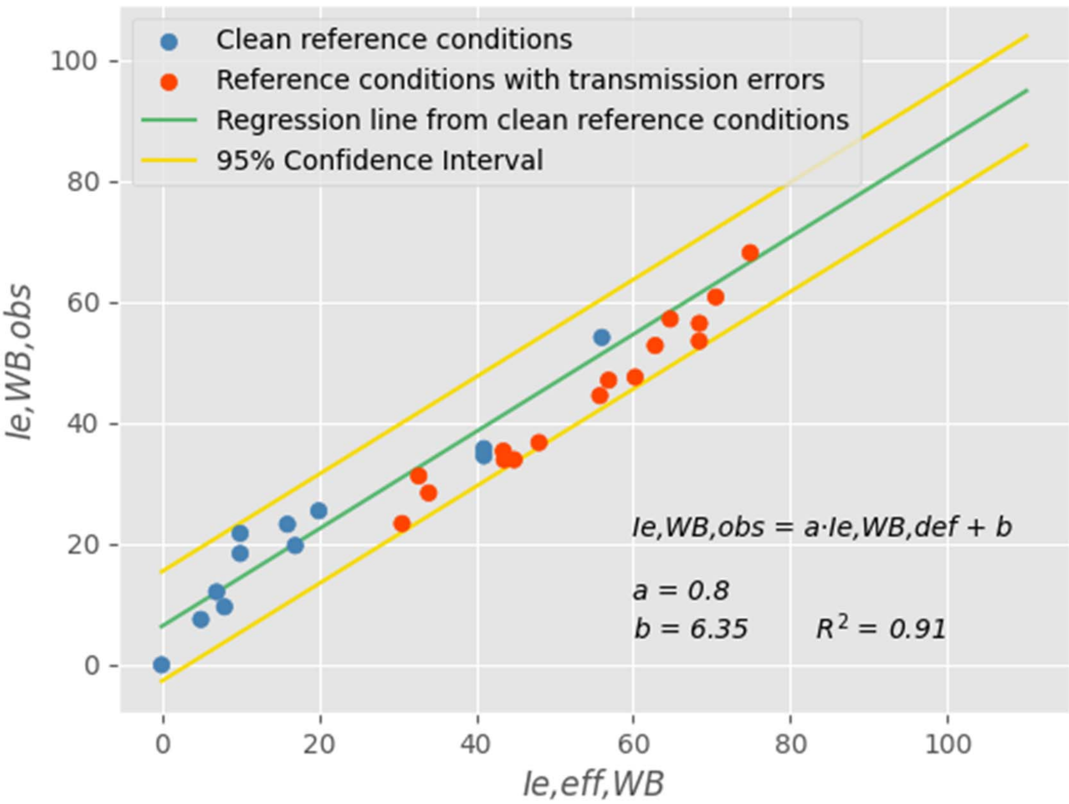

Figure E.32: Original regression line, observed *Ie,WB,obs* and defined *Ie,eff,WB* for clean and error-prone reference conditions

With the coefficients a and b stable *Ie,WB* values for LC3plus with transmission errors can be derived (Table E.26) without existing *Bpl* values.

Table E.27: Stable *Ie,WB,def* values for LC3plus under transmission error conditions

Codec under Test	*Ie,WB,obs*	*Ie,WB,def*
LC3plus@48 2% (EPFsize=20)	13.78	9.23
LC3plus@48 4% (EPFsize=20)	23.36	21.13
LC3plus@48 8% (EPFsize=20)	33.32	33.5
LC3plus@48 12% (EPFsize=20)	40.03	41.84
LC3plus@48 2% (EPFsize=10)	10.55	5.21
LC3plus@48 4% (EPFsize=10)	15.26	11.06
LC3plus@48 8% (EPFsize=10)	26.47	24.99
LC3plus@48 12% (EPFsize=10)	32.44	32.41
LC3plus@32 2% (EPFsize=20)	15.26	11.06
LC3plus@32 4% (EPFsize=20)	24.07	22.01
LC3plus@32 8% (EPFsize=20)	31.52	31.27
LC3plus@32 12% (EPFsize=20)	41.62	43.82
LC3plus@32 2% (EPFsize=10)	14.76	10.44
LC3plus@32 4% (EPFsize=10)	15.71	11.62
LC3plus@32 8% (EPFsize=10)	26.47	24.99
LC3plus@32 12% (EPFsize=10)	37.58	38.8
LC3plus@24 2% (EPFsize=20)	26.12	24.56
LC3plus@24 4% (EPFsize=20)	33.32	33.5
LC3plus@24 8% (EPFsize=20)	39.2	40.81
LC3plus@24 12% (EPFsize=20)	44.5	47.4
LC3plus@24 2% (EPFsize=10)	24.41	22.43
LC3plus@24 4% (EPFsize=10)	28.11	27.03
LC3plus@24 8% (EPFsize=10)	33.01	33.12
LC3plus@24 12% (EPFsize=10)	40.83	42.84
LC3plus@16 2% (EPFsize=20)	47.32	50.9
LC3plus@16 4% (EPFsize=20)	49.82	54.01
LC3plus@16 8% (EPFsize=20)	54.5	59.82
LC3plus@16 12% (EPFsize=20)	55.96	61.64
LC3plus@16 2% (EPFsize=10)	46.54	49.93
LC3plus@16 4% (EPFsize=10)	48.31	52.13
LC3plus@16 8% (EPFsize=10)	53.02	57.98
LC3plus@16 12% (EPFsize=10)	57.89	64.03

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern Frame sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,WB* curve the best. The optimal value can be determined by calculating the least square error for every four pairs of *Ie,WB,obs(Ppl)* and *Ie,WB,eff(Ppl)*.

Table E.28: Packet loss robustness factors for LC3plus

Bitrate (kbps)	EPFsize (ms)	*Ie,WB,def*	*Bpl*
48	20	0.0	14.9
48	10	0.0	24.5
32	20	4.04	17.0
32	10	4.04	24.5
24	20	15.9	16.9
24	10	15.9	25.4
16	20	58.81	(300)
16	10	58.81	(300)

Figure E.33 and Figure E.34 show *Ie,eff,WB(Ppl)* with fixed *Bpl* from Table E.28. The approximations using the *Ie, eff,WB* formula does not work for LC3plus@16kbit/s since this condition is already discarded by the additivity check. For the other bitrates it shows a very good fit.

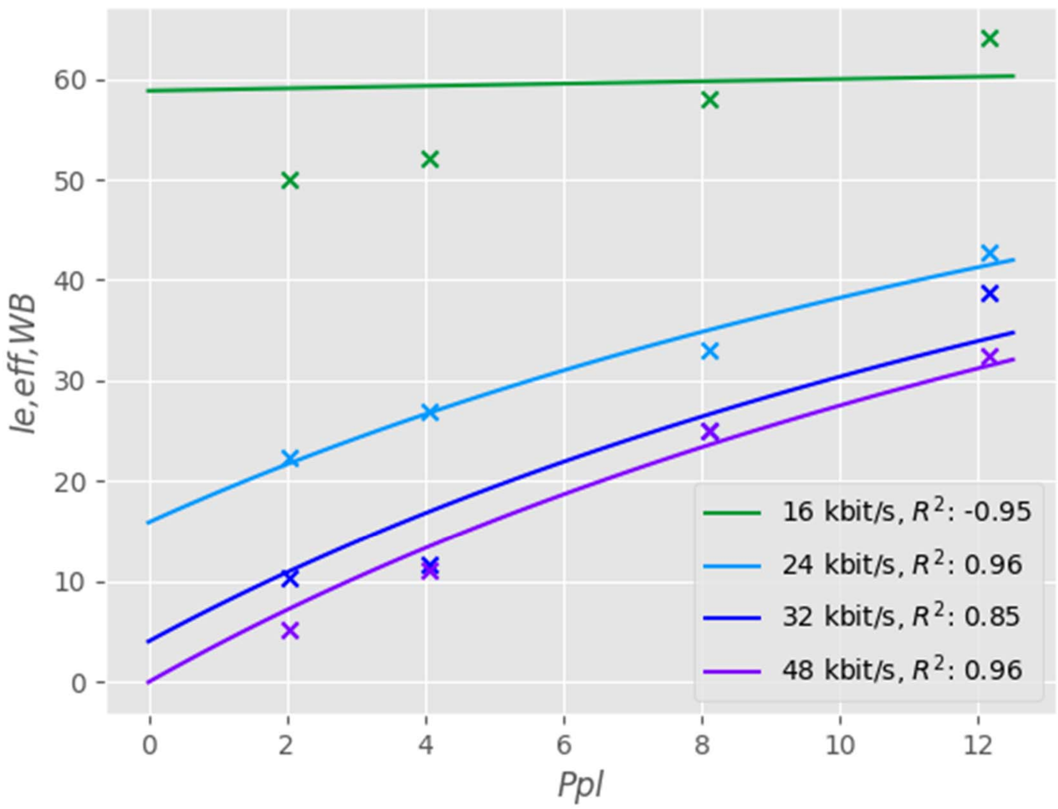

Figure E.33: *Ie,eff,WB* approximation with best fitting *Bpl* and observed *Ie* values vs. packet loss rate for 10 ms EPFsize

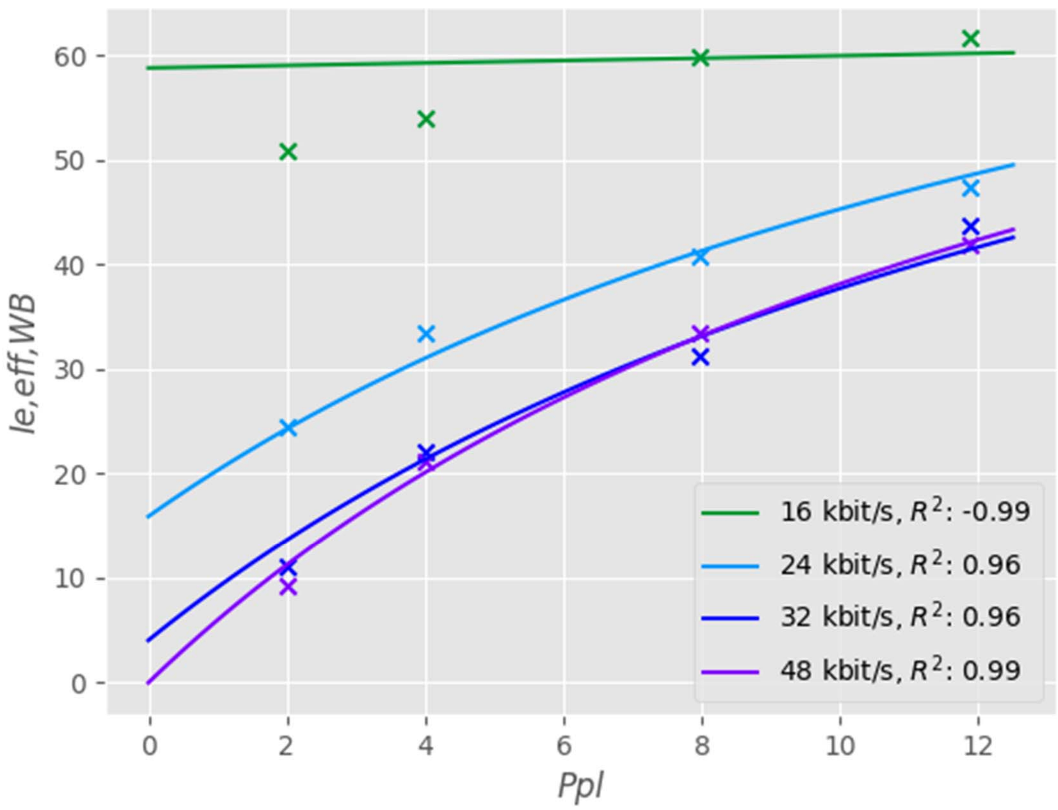

Figure E.34: *Ie,eff,WB* approximation with best fitting *Bpl* and observed *Ie* values vs. packet loss rate for 20 ms EPFsize

E.3.3 Fullband (FB) Ie factors

E.3.3.1 Objective assessment

E.3.3.1.1 Introduction

This clause describes the derivation of the equipment impairment factors *Ie,FB* and the packet loss robustness factor *Bpl* for LC3plus in super wideband mode. It is determined according to clause E.2.3. MOS values are obtained by evaluating coded 32 kHz English speech signals with Recommendation ITU-T P.863 [2].

E.3.3.1.2 Determination of *Ie,FB* for LC3plus in error-free conditions

E.3.3.1.2.1 Measuring MOS

Figure E.35 shows the MOS of the reference conditions and LC3plus configurations under test with -16, -26 and -36 dBOV bitrates of 32, 48 and 64 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

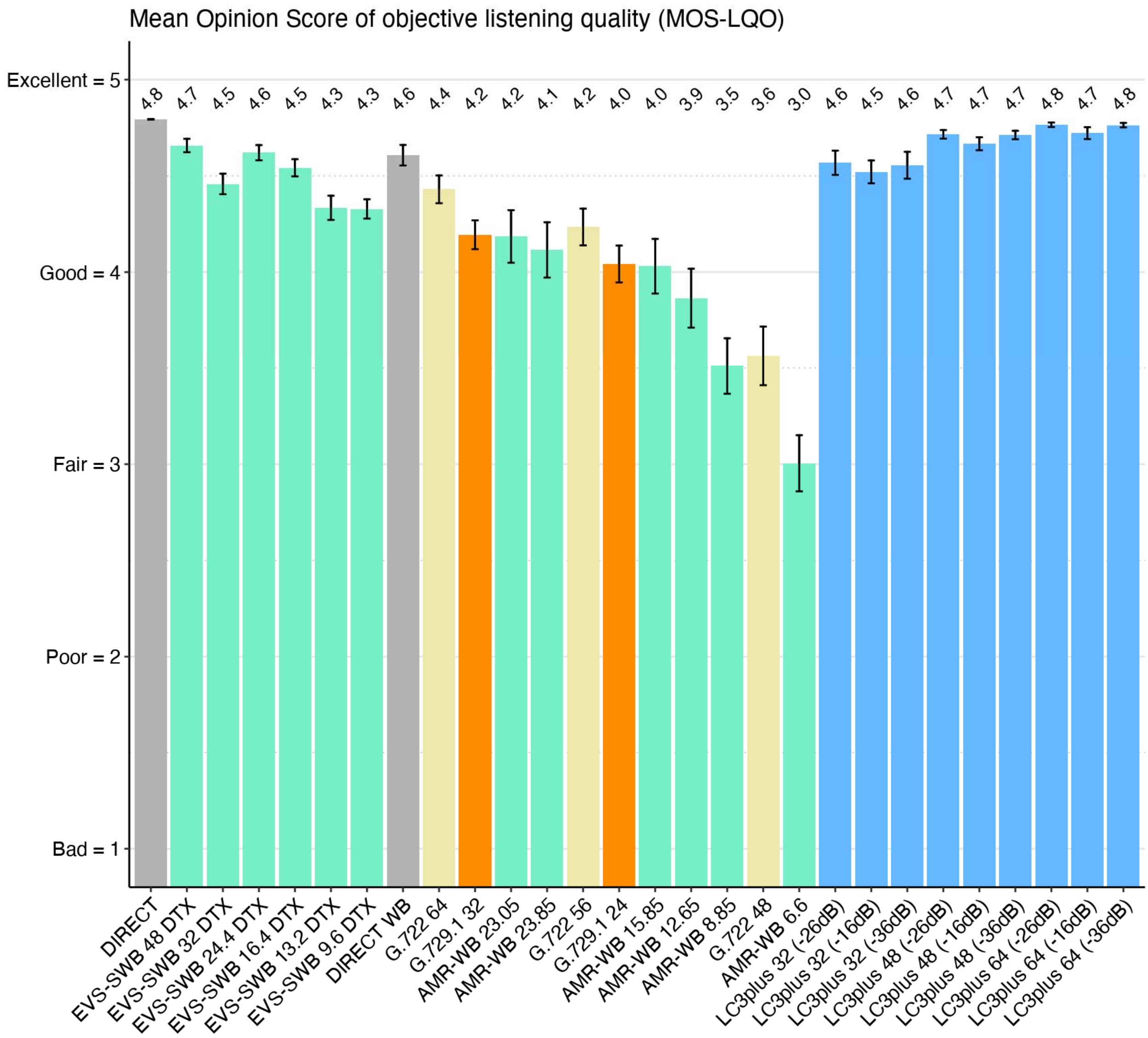

Figure E.35: Recommendation ITU-T P.863 results for SWB and WB clean speech signals. Mean scores and 95 % confidence intervals

E.3.3.1.2.2 MOS normalization

To be able to use the same MOS - R transformation as in Recommendation ITU-T P.834 [21] the MOS values from the experiment need to be normalized to fit the narrow band scale in case the maximum value is greater than 4.5 according to:

$$MOS_n = \frac{MOS-1}{MOS_{max}} \cdot 3.5 + 1$$

E.3.3.1.2.3 Scale transformation

All Recommendation ITU-T P.863 test results are transformed from the MOS(NB) scale to the scale of the equipment impairment factor *Ie,FB*. In a first step R is determined by solving the following equations for R_{NB}:

$for\ MOS_n = 1.0$: $R_{NB} = 0$

$for\ 1.0 < MOS_n < 4.5$: $MOS_n = 1 + 0.035 \cdot R_{NB} + R_{NB} \cdot (R_{NB} - 60) \cdot (100 - R_{NB}) \cdot 7 \cdot 10^{-6}$

$for\ MOS_n \geq 4.5$: $R_{NB} = 100$

E.3.3.1.2.4 R scale extension

For FB the R scale is extended by:

$$R_{FB} = R_{NB} \cdot 1.48$$

E.3.3.1.2.5 Calculation of observed impairment factor

The observed impairment factor *Ie,FB,obs* is calculated with the DIRECT signal as reference anchor:

$$Ie,FB,obs = R_{FB}\ (DIRECT) - R_{FB}\ (test\ condition)$$

for all other reference and test conditions depicted in Figure E.35. The values for MOS, MOS_n, R_{FB}, *Ie,FB,obs* and *Ie,FB,def* are listed in Table E.29:

Table E.29: Observed and defined *Ie,FB* reference values and intermediate values

Codec References	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
DIRECT	4.79	4.5	100.0	148.0	0.0	0.0
EVS-SWB@48 DTX	4.66	4.37	91.52	135.45	12.55	10.2
EVS-SWB@32 DTX	4.46	4.19	84.76	125.44	22.56	8.7
EVS-SWB@24.4 DTX	4.62	4.34	90.04	133.26	14.74	7.2
EVS-SWB@16.4 DTX	4.54	4.27	87.31	129.22	18.78	10.8
EVS-SWB@13.2 DTX	4.33	4.08	81.4	120.48	27.52	17.1
EVS-SWB@9.6 DTX	4.33	4.07	81.26	120.26	27.74	22.7
DIRECT WB@NA	4.61	4.33	89.56	132.55	15.45	19.0
G.722@64	4.43	4.16	83.96	124.27	23.73	24.0
G.729.1@32	4.19	3.95	78.02	115.46	32.54	26.0
AMR-WB@23.05	4.19	3.94	77.81	115.16	32.84	27.0
AMR-WB@23.85	4.12	3.87	76.24	112.83	35.17	29.0
G.722@56	4.23	3.98	78.96	116.86	31.14	29.0
G.729.1@24	4.04	3.81	74.65	110.48	37.52	35.0
AMR-WB@15.85	4.03	3.8	74.4	110.11	37.89	36.0
AMR-WB@12.65	3.86	3.64	70.99	105.06	42.94	39.0
AMR-WB@8.85	3.51	3.32	64.24	95.07	52.93	60.0
G.722@48	3.56	3.37	65.21	96.51	51.49	60.0
AMR-WB@6.6	3.0	2.85	55.21	81.71	66.29	75.0

E.3.3.1.2.6 Linear interpolation

According to Recommendation ITU-T P.834.1 [25] pairs of defined equipment impairment factors *Ie,FB,def* and observed values *Ie,FB,obs* are plotted in Figure E.36. A linear regression between the measured and the already known variables can be modeled as:

$$\text{Ie,FB,obs} = \text{a} \cdot \text{Ie,FB,def} + \text{b}$$

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j) * (Ie,FB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j)}{\sum_{i=1}^{n}(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j$$

with *n = 19* reference conditions and *Ie,FB,obs* and *Ie,FB,def* taken from Table E.29. The resulting regression line and data points are presented in Figure E.36. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination R^2 of 0.93 indicates a strong fit of the model.

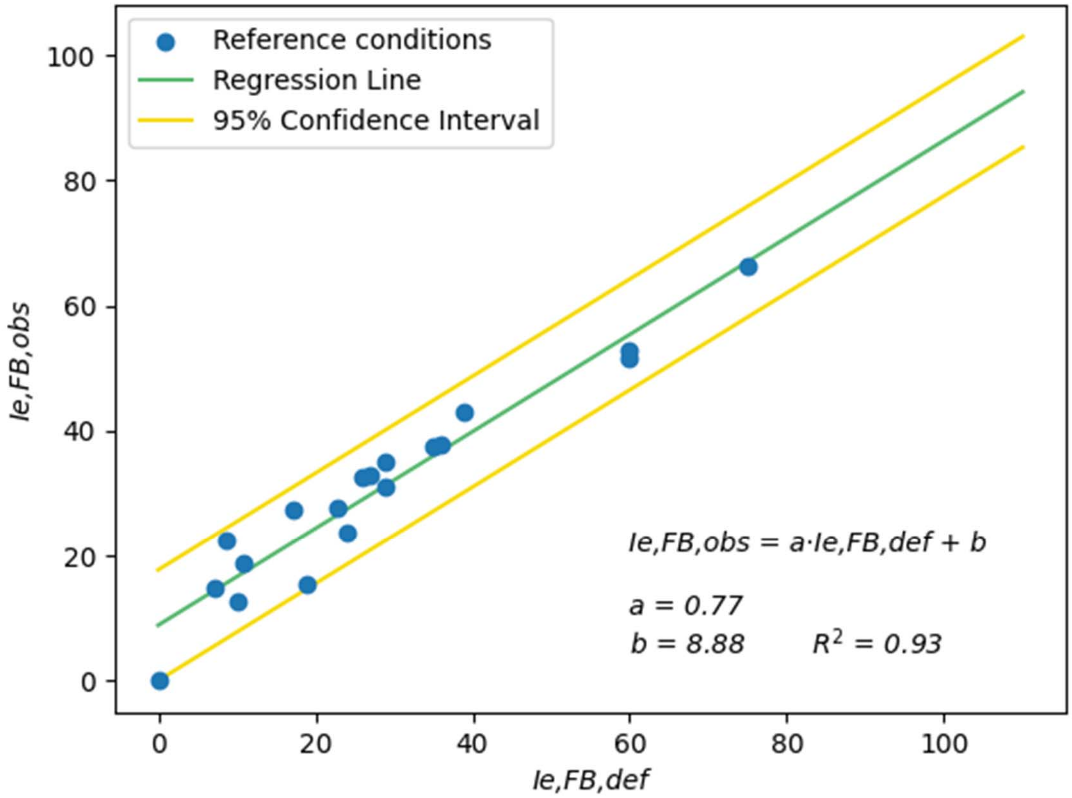

Figure E.36: Linear Regression between observed and defined *Ie,FB*

E.3.3.1.2.7 Determination of a stable *Ie,FB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,FB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,FB,def* for LC3plus based on the parameters a and b. Negative *Ie,FB* values are set to zero. The stable impairment factors are shown in Table E.30 and calculate as follows:

$$Ie,FB,def(LC3plus) = \frac{Ie,FB,obs - b}{a}$$

Table E.30: Stable *Ie,FB,def* for LC3plus and intermediate values

Codec under Test	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32	4.57	4.27	87.51	129.51	18.49	12.4
LC3plus@48	4.72	4.41	93.34	138.14	9.86	1.27
LC3plus@64	4.77	4.46	96.22	142.4	5.6	0.0

E.3.3.1.2.8 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.31 are encoded and MOS values are measured as shown in Figure E.37. Normalization, scale transformation and extension are performed according to the previous clause to derive *Ie,FB,obs* values for the tandeming conditions.

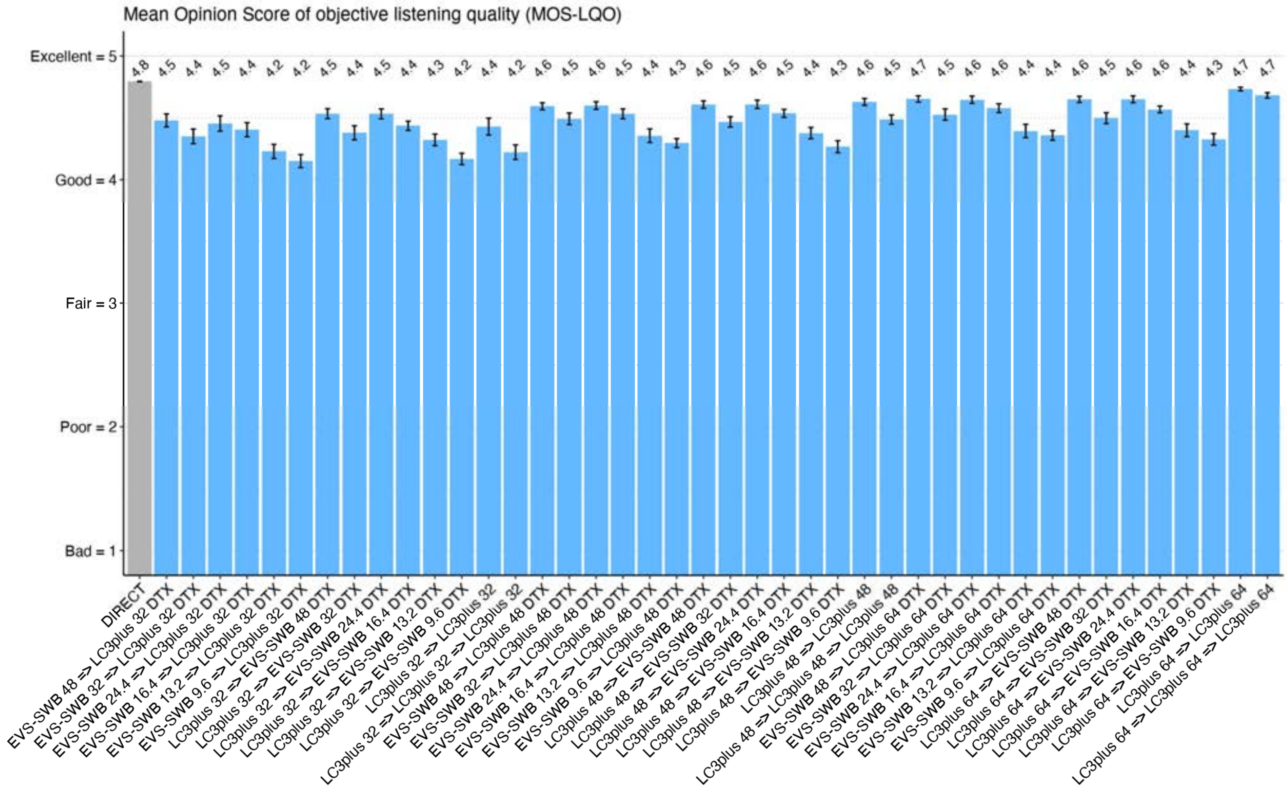

Figure E.37: Recommendation ITU-T P.863 results for SWB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

Ie,FB,def values of tandeming of codec1 and codec2 are calculated as the sum of *Ie,FB,def(codec1)* and *Ie,FB,def(codec2).*

Table E.31: Observed and defined *Ie,FB* values for tandeming and intermediate values

Codec under test	MOS	MOS_n	R_{NB}	R_{FB}	Ie,FB,obs	Ie,FB,def
EVS-SWB 48 DTX => LC3plus 32	4.48	4.21	85.38	126.36	21.64	21.32
EVS-SWB 32 DTX => LC3plus 32	4.35	4.09	81.82	121.1	26.9	19.82
EVS-SWB 24.4 DTX => LC3plus 32	4.45	4.19	84.66	125.3	22.7	18.32
EVS-SWB 16.4 DTX => LC3plus 32	4.4	4.14	83.25	123.22	24.78	21.92
EVS-SWB 13.2 DTX => LC3plus 32	4.48	4.21	85.38	126.36	21.64	22.6
EVS-SWB 9.6 DTX => LC3plus 32	4.35	4.09	81.82	121.1	26.9	21.1
LC3plus 32 => EVS-SWB 48 DTX	4.45	4.19	84.66	125.3	22.7	19.6
LC3plus 32 => EVS-SWB 32 DTX	4.4	4.14	83.25	123.22	24.78	23.2
LC3plus 32 => EVS-SWB 24.4 DTX	4.23	3.98	78.83	116.66	31.34	29.5
LC3plus 32 => EVS-SWB 16.4 DTX	4.15	3.91	77.02	113.99	34.01	35.1
LC3plus 32 => EVS-SWB 13.2 DTX	4.53	4.26	87.06	128.85	19.15	22.6
LC3plus 32 => EVS-SWB 9.6 DTX	4.38	4.12	82.58	122.21	25.79	21.1
LC3plus 32 => LC3plus 32	4.53	4.26	87.01	128.77	19.23	19.6
LC3plus 32 => LC3plus 32 => LC3plus 32	4.44	4.17	84.13	124.51	23.49	23.2
EVS-SWB 48 DTX => LC3plus 48	4.32	4.07	81.11	120.04	27.96	29.5
EVS-SWB 32 DTX => LC3plus 48	4.17	3.92	77.42	114.58	33.42	35.1
EVS-SWB 24.4 DTX => LC3plus 48	4.43	4.16	83.95	124.25	23.75	24.8

Codec under test	MOS	MOS_n	R_{NB}	R_{FB}	Ie,FB,obs	Ie,FB,def
EVS-SWB 16.4 DTX => LC3plus 48	4.22	3.97	78.69	116.45	31.55	37.2
EVS-SWB 13.2 DTX => LC3plus 48	4.59	4.32	89.06	131.8	16.2	11.47
EVS-SWB 9.6 DTX => LC3plus 48	4.49	4.22	85.77	126.94	21.06	9.97
LC3plus 48 => EVS-SWB 48 DTX	4.6	4.32	89.27	132.12	15.88	8.47
LC3plus 48 => EVS-SWB 32 DTX	4.53	4.26	87.01	128.77	19.23	12.07
LC3plus 48 => EVS-SWB 24.4 DTX	4.36	4.1	81.97	121.32	26.68	18.37
LC3plus 48 => EVS-SWB 16.4 DTX	4.3	4.04	80.44	119.06	28.94	23.97
LC3plus 48 => EVS-SWB 13.2 DTX	4.61	4.33	89.56	132.55	15.45	11.47
LC3plus 48 => EVS-SWB 9.6 DTX	4.47	4.2	85.02	125.83	22.17	9.97
LC3plus 48 => LC3plus 48	4.61	4.33	89.64	132.67	15.33	8.47
LC3plus 48 => LC3plus 48 => LC3plus 48	4.54	4.26	87.13	128.96	19.04	12.07
EVS-SWB 48 DTX => LC3plus 64	4.65	4.37	91.3	135.13	12.87	10.2
EVS-SWB 32 DTX => LC3plus 64	4.53	4.25	86.83	128.51	19.49	8.7
EVS-SWB 24.4 DTX => LC3plus 64	4.65	4.36	91.06	134.76	13.24	7.2
EVS-SWB 16.4 DTX => LC3plus 64	4.58	4.3	88.54	131.03	16.97	10.8
EVS-SWB 13.2 DTX => LC3plus 64	4.39	4.13	82.98	122.8	25.2	17.1
EVS-SWB 9.6 DTX => LC3plus 64	4.36	4.1	82.03	121.41	26.59	22.7
LC3plus 64 => EVS-SWB 48 DTX	4.65	4.37	91.17	134.93	13.07	10.2
LC3plus 64 => EVS-SWB 32 DTX	4.5	4.23	85.94	127.19	20.81	8.7
LC3plus 64 => EVS-SWB 24.4 DTX	4.65	4.37	91.2	134.98	13.02	7.2
LC3plus 64 => EVS-SWB 16.4 DTX	4.57	4.29	88.17	130.5	17.5	10.8
LC3plus 64 => EVS-SWB 13.2 DTX	4.4	4.14	83.12	123.02	24.98	17.1
LC3plus 64 => EVS-SWB 9.6 DTX	4.33	4.07	81.2	120.18	27.82	22.7
LC3plus 64 => LC3plus 64	4.73	4.44	95.17	140.84	7.16	0.0
LC3plus 64 => LC3plus 64 => LC3plus 64	4.68	4.4	92.56	136.99	11.01	0.0

As done before the pairs of defined and observed *Ie,FB* are plotted against each other in Figure E.38. If more than four out of 14 tandem conditions show major deviations from the interpolation line, the additivity property should not be regarded as being satisfied. This condition is fulfilled for LC3plus at 32, 48 and 64 kbit/s.

The only condition outside of the confidence interval is:

"LC3plus 48 => LC3plus 48 => LC3plus 48" with an error of 9.49

The possible decrease in quality after the third tandeming of a codec with Ie of 0 is not taken into account by the model.

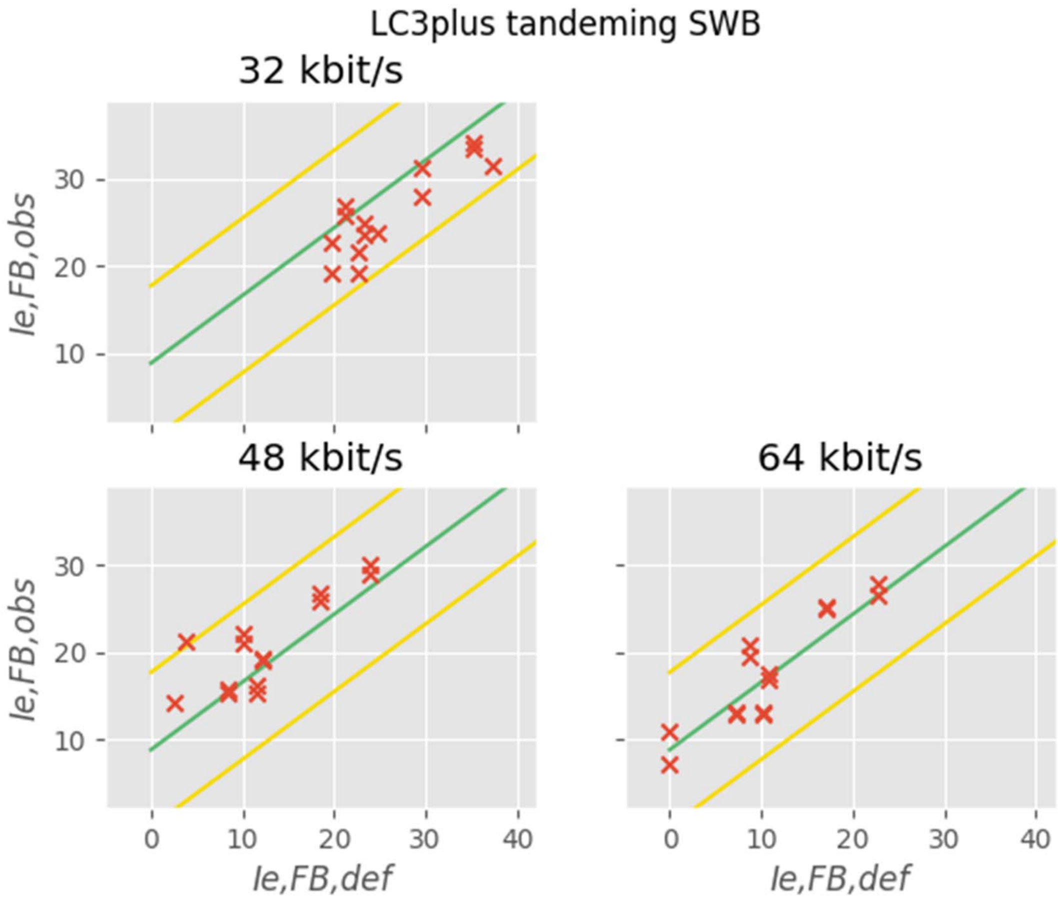

Figure E.38: Observed and defined *Ie,FB* for tandeming conditions with original regression line

E.3.3.1.2.9 Determination of *Ie,FB* for LC3plus under transmission error conditions

To determine Ie values for LC3plus under transmission error conditions the same reference conditions as in Table E.29 plus 16 supplementary reference conditions from Appendix V/G 113 Amd. 2 [26] are needed.

Figure E.39 shows the Recommendation ITU-T P.863 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes (EPFsizes).

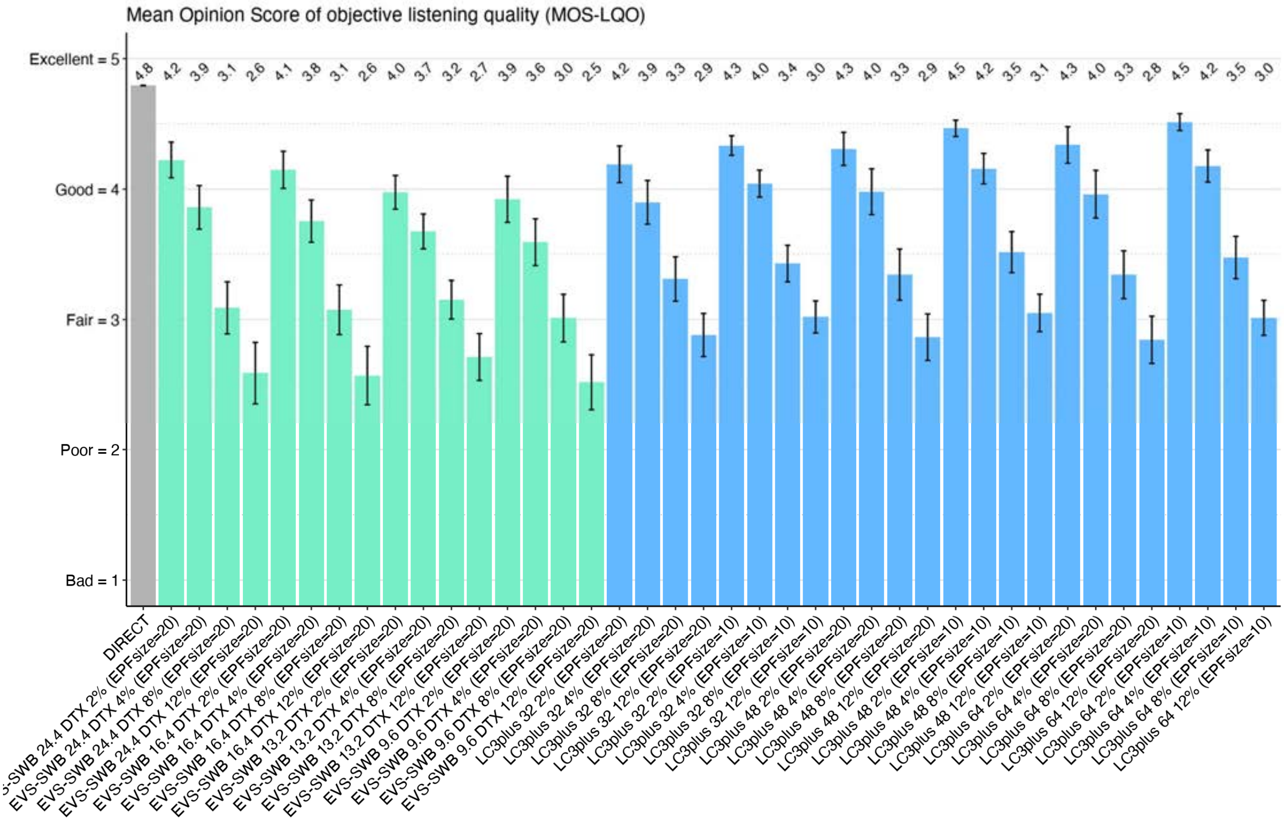

Figure E.39: Recommendation ITU-T P.863 results for SWB transmission error speech signals. Mean scores and 95 % confidence intervals

For the error-prone reference conditions an effective equipment impairment factor *Ie,eff,FB* is calculated according to the following equation taken from Recommendation ITU-T G.107.2 [28].

$$Ie,eff,FB = Ie,FB + (132 - Ie,FB) \cdot \frac{Ppl}{Ppl+Bpl}$$

The actual values for *Ppl* and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor *Bpl* is taken from G.113 Amendment 2 [26].

Table E.32: Parameters used to calculate *Ie,eff,FB*

Condition	*Ie*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	*q* (measured)	*BurstR*	*Ie,FB,eff*	*Ie,FB,obs*
EVS-SWB@24.4	7.2	2	11.4	2.09	0.99	0.99	26.57	31.52
EVS-SWB@24.4	7.2	4	11.4	4.18	0.97	0.99	40.67	43.08
EVS-SWB@24.4	7.2	8	11.4	8.37	0.92	1.0	60.02	64.09
EVS-SWB@24.4	7.2	12	11.4	12.54	0.87	1.01	72.58	77.08
EVS-SWB@16.4	10.8	2	10.3	2.09	0.99	0.99	31.28	34.1
EVS-SWB@16.4	10.8	4	10.3	4.18	0.97	0.99	45.77	46.14
EVS-SWB@16.4	10.8	8	10.3	8.37	0.92	1.0	65.12	64.46
EVS-SWB@16.4	10.8	12	10.3	12.54	0.87	1.01	77.35	77.56

Condition	*Ie*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	*q* (measured)	*BurstR*	*Ie,FB,eff*	*Ie,FB,obs*
EVS-SWB@13.2	17.1	2	11.7	2.09	0.99	0.99	34.54	39.61
EVS-SWB@13.2	17.1	4	11.7	4.18	0.97	0.99	47.33	48.38
EVS-SWB@13.2	17.1	8	11.7	8.37	0.92	1.0	65.0	62.45
EVS-SWB@13.2	17.1	12	11.7	12.54	0.87	1.01	76.54	73.85
EVS-SWB@9.6	22.7	2	13.0	2.09	0.99	0.99	37.86	41.23
EVS-SWB@9.6	22.7	4	13.0	4.18	0.97	0.99	49.28	50.69
EVS-SWB@9.6	22.7	8	13.0	8.37	0.92	1.0	65.5	66.12
EVS-SWB@9.6	22.7	12	13.0	12.54	0.87	1.01	76.37	78.91

Next the observed Ie,FB values are calculated according to the previous clause. The results listed in Table E.32 show increasing impairment factors for increasing transmission error rates. The new pairs of Ie,FB,obs and Ie,eff,FB are added in the scatterplot Figure E.40. It can be observed that the original interpolation line still describes the new error-prone reference conditions adequately with a strong fit of $\boldsymbol{R^2}$ = 0.93. At higher error rates a light deviation outside the confidence interval can be observed.

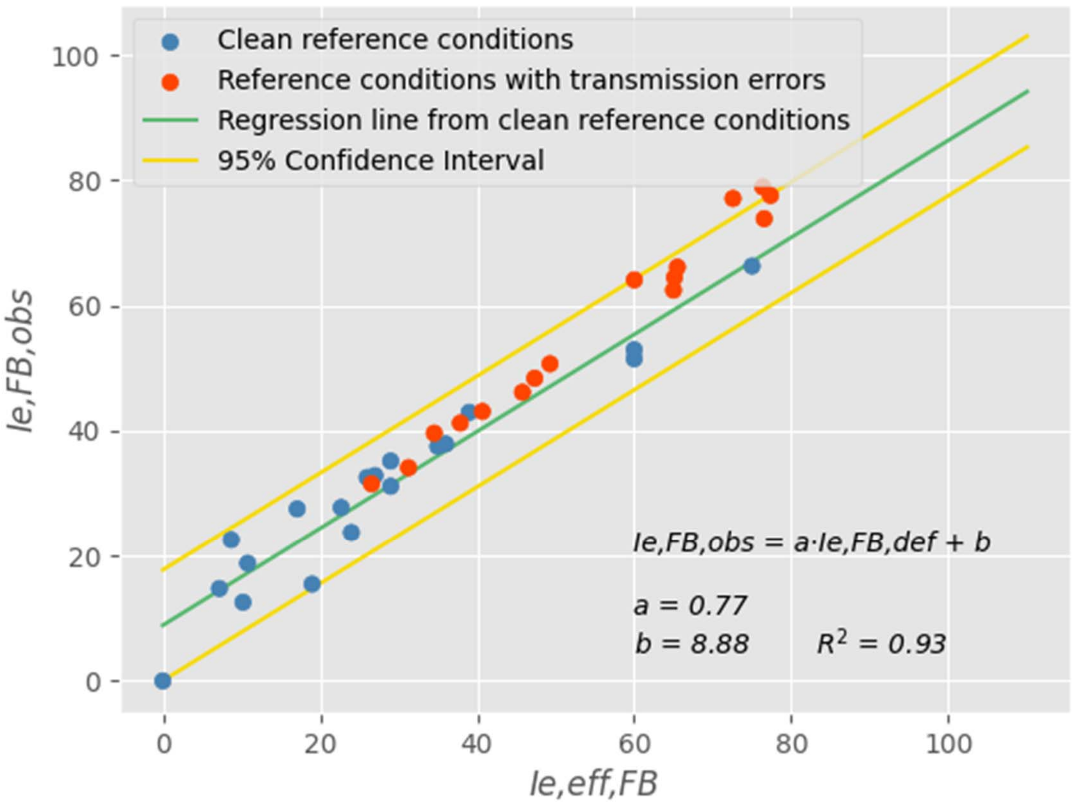

Figure E.40: Original regression line, observed *Ie,FB,obs* and defined *Ie,eff,FB* for clean and error-prone reference conditions

With the coefficients a and b stable *Ie,FB* values for LC3plus with transmission errors can be derived (Table E.33) without existing *Bpl* values.

Table E.33: Stable *Ie,FB* value for LC3plus under transmission error conditions

Codec under Test	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32 2% (EPFsize=20)	32.67	30.7
LC3plus@32 4% (EPFsize=20)	41.93	42.65

Codec under Test	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32 8% (EPFsize=20)	58.29	63.76
LC3plus@32 12% (EPFsize=20)	69.48	78.2
LC3plus@32 2% (EPFsize=10)	27.52	24.06
LC3plus@32 4% (EPFsize=10)	37.52	36.96
LC3plus@32 8% (EPFsize=10)	55.13	59.68
LC3plus@32 12% (EPFsize=10)	65.88	73.56
LC3plus@48 2% (EPFsize=20)	28.48	25.29
LC3plus@48 4% (EPFsize=20)	39.46	39.46
LC3plus@48 8% (EPFsize=20)	57.39	62.6
LC3plus@48 12% (EPFsize=20)	69.91	78.76
LC3plus@48 2% (EPFsize=10)	22.23	17.23
LC3plus@48 4% (EPFsize=10)	33.79	32.15
LC3plus@48 8% (EPFsize=10)	52.77	56.64
LC3plus@48 12% (EPFsize=10)	65.09	72.54
LC3plus@64 2% (EPFsize=20)	27.34	23.82
LC3plus@64 4% (EPFsize=20)	40.04	40.21
LC3plus@64 8% (EPFsize=20)	57.42	62.64
LC3plus@64 12% (EPFsize=20)	70.44	79.44
LC3plus@64 2% (EPFsize=10)	20.1	14.48
LC3plus@64 4% (EPFsize=10)	33.12	31.28
LC3plus@64 8% (EPFsize=10)	53.89	58.08
LC3plus@64 12% (EPFsize=10)	66.06	73.79

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern Frame sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,FB* curve the best. The optimal value can be determined by calculating the least square error for the pairs of *Ie,FB,obs(Ppl)* and *Ie,eff,FB(Ppl).* Table E.34 shows that a shorter EPFsize is expectedly associated with a higher *Bpl* for the same bitrate.

Table E.34: Packet loss robustness factors for LC3plus

Bitrate (kbit/s)	EPFsize (ms)	*Ie,FB,def*	Bpl
32	20	12.4	11.0
32	10	12.4	13.1
48	20	1.27	9.2
48	10	1.27	11.5
64	20	0.0	9.0
64	10	0.0	11.1

Figure E.41 and Figure E.42 show Ie,eff,FB(Ppl) with fixed Bpl from Table E.34. All approximations using the Ie,eff,FB formula show a strong fit according to $\boldsymbol{R^2}$ for both EPFsizes.

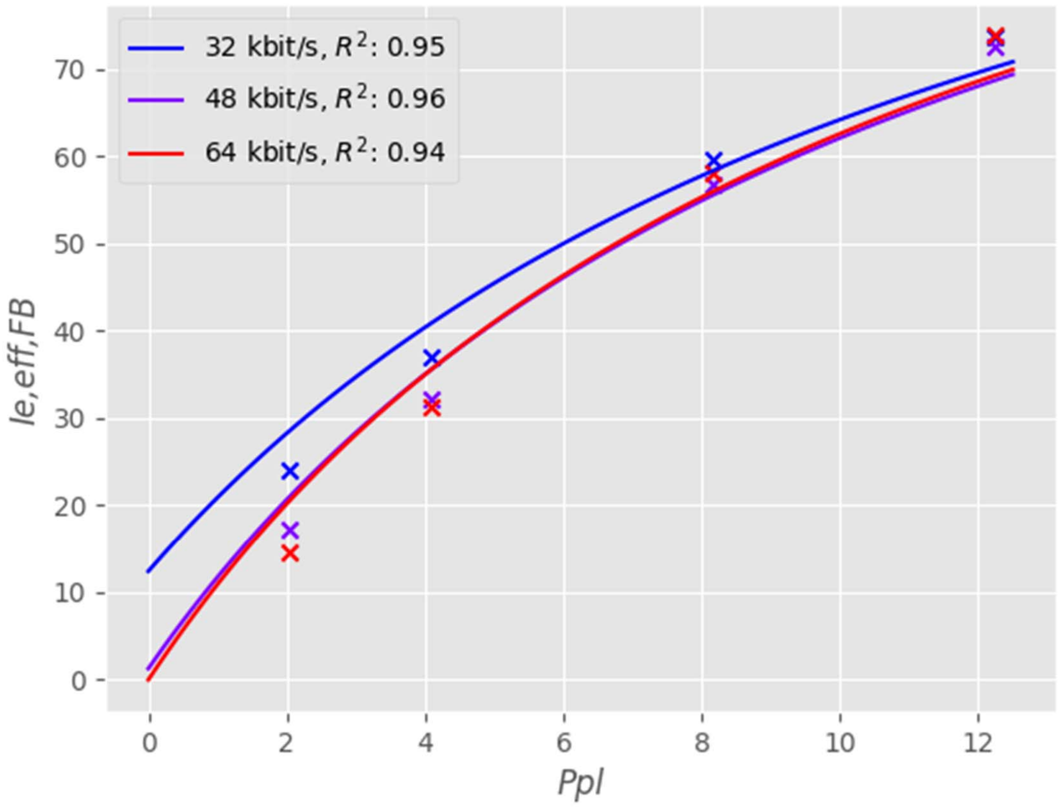

Figure E.41: ***Ie,eff,FB*** **approximation with best fitting** ***Bpl*** **and observed** ***Ie*** **values vs. packet loss rate for 10 ms EPFsize**

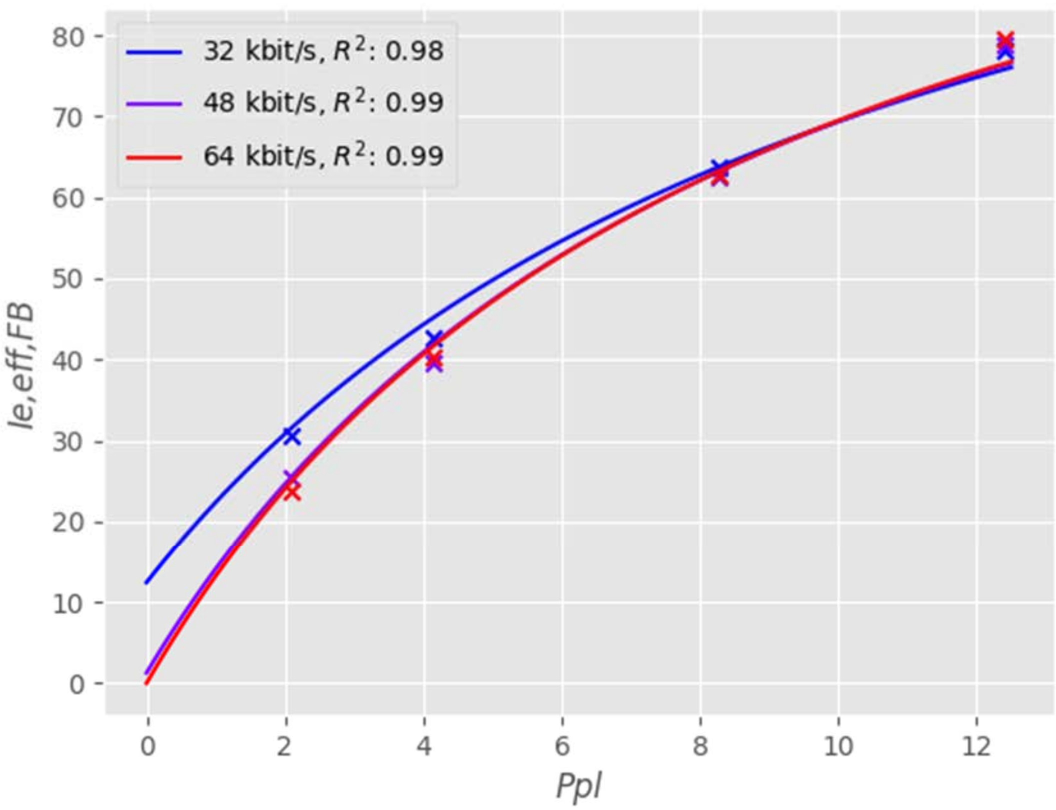

Figure E.42: ***Ie,eff,FB*** **approximation with best fitting** ***Bpl*** **and observed** ***Ie*** **values vs. packet loss rate for 20 ms EPFsize**

E.3.3.2 Subjective assessment

E.3.3.2.1 Introduction

This clause describes the derivation of the equipment impairment factors *Ie,FB* and the packet loss robustness factor *Bpl* for LC3plus in super wideband mode. It is determined according to clause E.2.3. MOS values are obtained by performing Recommendation ITU-T P.800 [1] listening tests according to considerations in clauses E.1.3 and E.1.4.

E.3.3.2.2 Determination of *Ie,FB* for LC3plus in error-free conditions

E.3.3.2.2.1 Measuring MOS

Figure E.43 shows the MOS of the reference conditions and LC3plus configurations under test with -16, -26 and -36 dBOV bitrates of 32, 48 and 64 kbit/s. In the following calculations, the mean across all levels at the same bitrate is used for LC3plus.

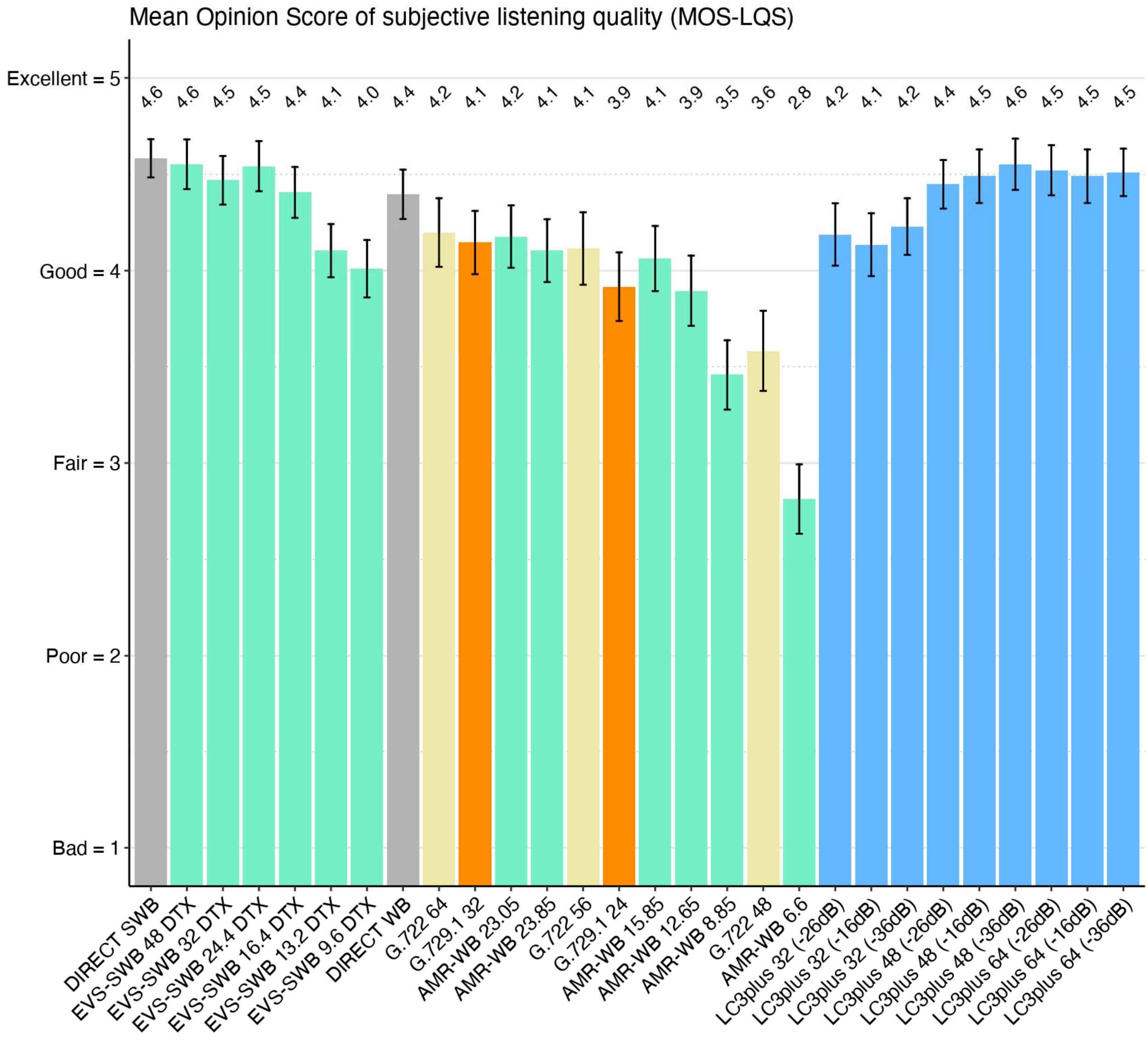

Figure E.43: Recommendation ITU-T P.800 results for SWB and WB clean speech signals. Mean scores and 95 % confidence intervals

E.3.3.2.2.2 MOS normalization

To be able to use the same MOS - R transformation as in Recommendation ITU-T P.833 [20] the MOS values from the experiment need to be normalized to fit the narrow band scale in case the maximum value is greater than 4.5 according to:

$$MOS_n = \frac{MOS-1}{MOS_{max}} \cdot 3.5 + 1$$

E.3.3.2.2.3 Scale transformation

All Recommendation ITU-T P.800 test results are transformed from the MOS(NB) scale to the scale of the equipment impairment factor *Ie,FB*. In a first step R is determined by solving the following equations for R_{NB}:

$for\ MOS_n = 1.0$: $R_{NB} = 0$

$for\ 1.0 < MOS_n < 4.5$: $MOS_n = 1 + 0.035 \cdot R_{NB} + R_{NB} \cdot (R_{NB} - 60) \cdot (100 - R_{NB}) \cdot 7 \cdot 10^{-6}$

$for\ MOS_n \geq 4.5$: $R_{NB} = 100$

E.3.3.2.2.4 R scale extension

For FB the R scale is extended by:

$$R_{FB} = R_{NB} \cdot 1.48$$

E.3.3.2.2.5 Calculation of observed impairment factor

The observed impairment factor *Ie,FB,obs* is calculated with the DIRECT signal as reference anchor:

Ie,FB,obs = R_{FB} *(DIRECT)* - R_{FB} *(test condition)*

for all other reference and test conditions depicted in Figure E.44. The values for MOS, MOS_n, R_{FB}, *Ie,FB,obs* and *Ie,FB,def* are listed in Table E.35:

Table E.35: Observed and defined *Ie,FB* reference values and intermediate values

Codec References	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
DIRECT	4.58	4.5	100.0	148.0	0.0	0.0
EVS-SWB@48 DTX	4.55	4.47	96.95	143.48	4.52	10.2
EVS-SWB@32 DTX	4.47	4.39	92.18	136.42	11.58	8.7
EVS-SWB@24.4 DTX	4.54	4.46	96.22	142.41	5.59	7.2
EVS-SWB@16.4 DTX	4.41	4.33	89.52	132.49	15.51	10.8
EVS-SWB@13.2 DTX	4.1	4.03	80.21	118.72	29.28	17.1
EVS-SWB@9.6 DTX	4.01	3.94	77.85	115.22	32.78	22.7
DIRECT WB@NA	4.4	4.32	89.14	131.93	16.07	19.0
G.722@64	4.2	4.12	82.76	122.48	25.52	24.0
G.729.1@32	4.15	4.07	81.33	120.36	27.64	26.0
AMR-WB@23.05	4.18	4.1	82.17	121.62	26.38	27.0
AMR-WB@23.85	4.1	4.03	80.21	118.72	29.28	29.0
G.722@56	4.12	4.04	80.5	119.14	28.86	29.0
G.729.1@24	3.92	3.85	75.65	111.96	36.04	35.0

Codec References	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
AMR-WB@15.85	4.06	3.99	79.16	117.16	30.84	36.0
AMR-WB@12.65	3.9	3.83	75.16	111.24	36.76	39.0
AMR-WB@8.85	3.46	3.4	65.94	97.58	50.42	60.0
G.722@48	3.58	3.52	68.44	101.3	46.7	60.0
AMR-WB@6.6	2.81	2.77	53.72	79.51	68.49	75.0

E.3.3.2.2.6 Linear interpolation

According to Recommendation ITU-T P.833.1 [24] pairs of defined equipment impairment factors *Ie,FB,def* and observed values *Ie,FB,obs* are plotted in Figure E.45. A linear regression between the measured and the already known variables can be modeled as:

$$Ie,FB,obs = a \cdot Ie,FB,def + b$$

The interpolation line parameters a and b are calculated in the least square sense as follows:

$$a = \frac{\sum_{i=1}^{n}(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j) * (Ie,FB,obs_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j)}{\sum_{i=1}^{n}(Ie,FB,def_i - \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j)^2}$$

$$b = \frac{1}{n}\sum_{j=1}^{n} Ie,FB,obs_j - a \cdot \frac{1}{n}\sum_{j=1}^{n} Ie,FB,def_j$$

with $n = 19$ reference conditions and *Ie,FB,obs* and *Ie,FB,def* taken from Table E.35. The resulting regression line and data points are presented in Figure E.44. The margin of error for the confidence interval is calculated by multiplying the 95 % t-value with the standard deviation of the residuals. The coefficient of determination R^2 of 0.91 indicates a strong fit of the model.

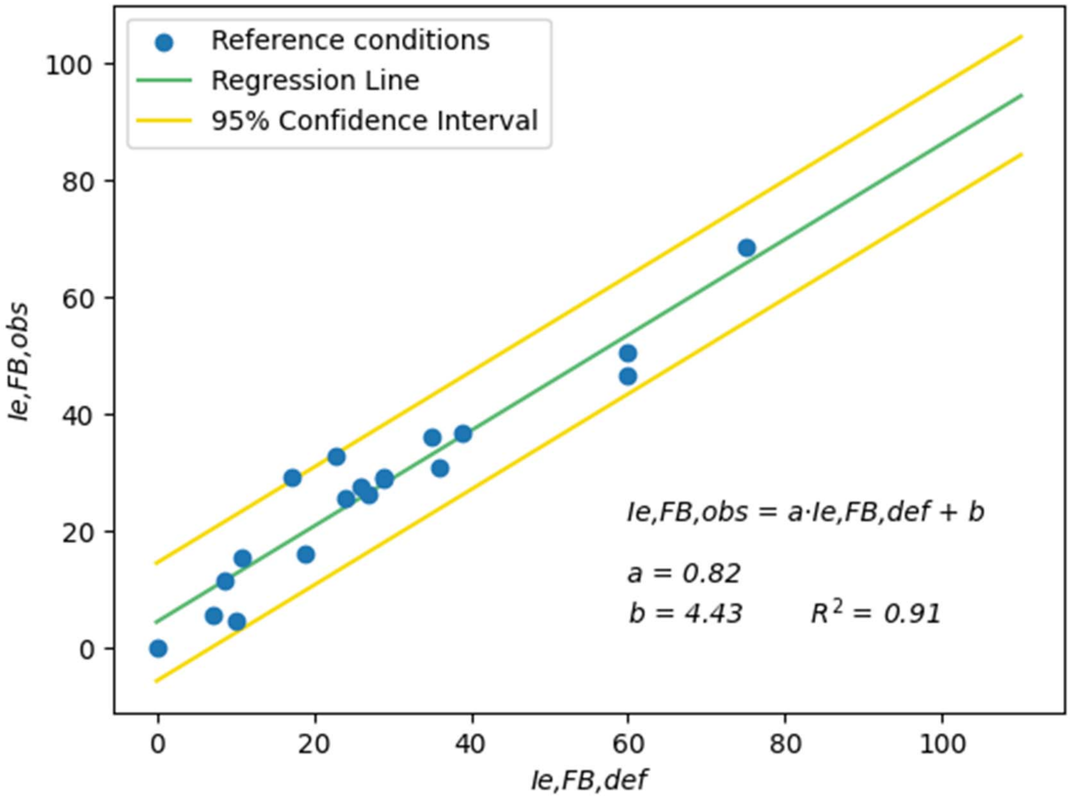

Figure E.44: Linear Regression between observed and defined *Ie,FB*

E.3.3.2.2.7 Determination of a stable *Ie,FB* value for LC3plus

In the next step the calculated regression line is used to map the observed values *Ie,FB,obs* of the LC3plus conditions to new equipment impairment factors *Ie,FB,def* for LC3plus based on the parameters a and b. Negative *Ie,FB* values are set to zero. The stable impairment factors are shown in Table E.36 and calculate as follows:

$$Ie, FB, def(LC3plus) = \frac{Ie{,}FB{,}obs - b}{a}$$

Table E.36: Stable *Ie,FB,def* for LC3plus and intermediate values

Codec under Test	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32	4.19	4.11	82.37	121.9	26.1	26.5
LC3plus@48	4.45	4.42	93.53	138.43	9.57	6.29
LC3plus@64	4.52	4.43	94.09	139.25	8.75	5.28

E.3.3.2.2.8 Additivity check

To proof that the calculated impairment values meet the condition of additivity, a check with multiple tandeming conditions is performed. The conditions listed in Table E.37 are encoded and MOS values are measured as shown in Figure E.45. Normalization, scale transformation and extension are performed according to the previous clause to derive *Ie,FB,obs* values for the tandeming conditions.

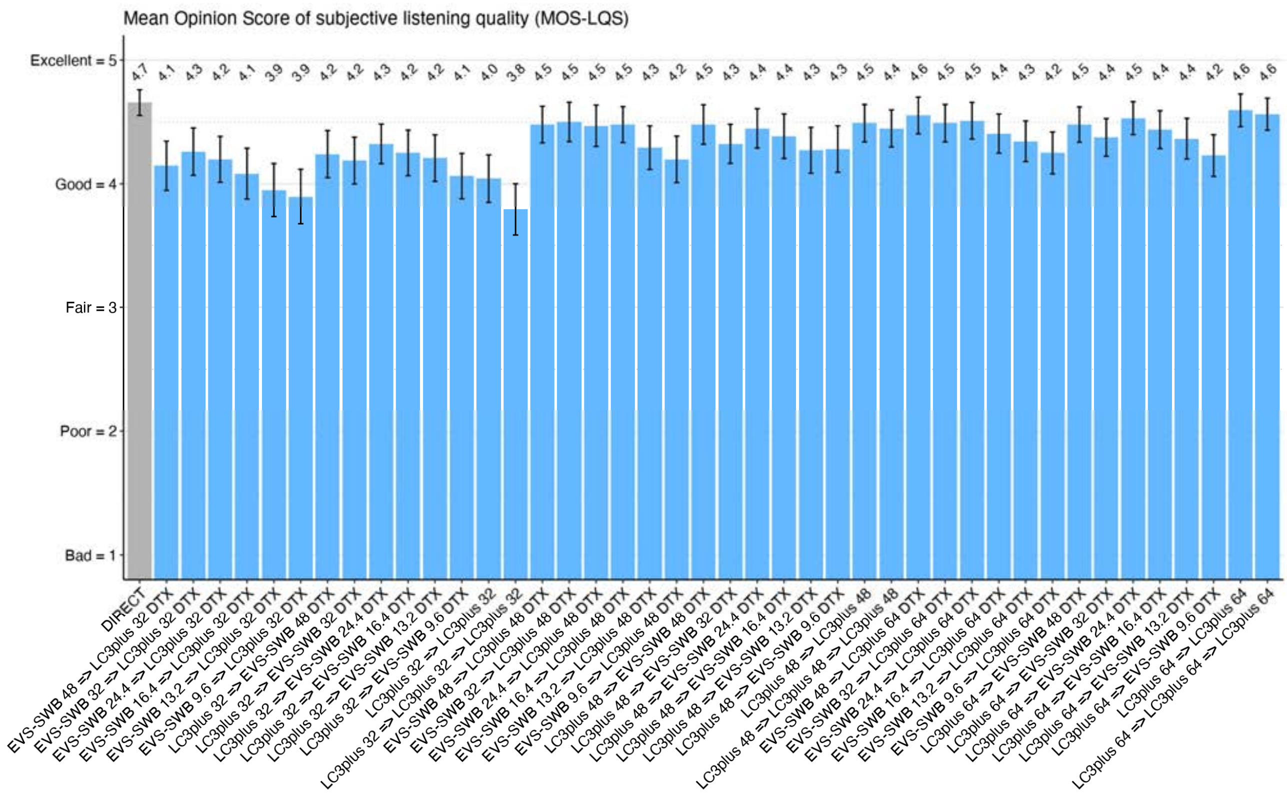

Figure E.45: Recommendation ITU-T P.800 results for SWB clean speech signals with codec tandemings for additivity check. Mean scores and 95 % confidence intervals

Ie,FB,def values of tandeming of codec1 and codec2 are calculated as the sum of *Ie,FB,def(codec1)* and *Ie,FB,def(codec2).*

Table E.37: Observed and defined *Ie,FB* values for tandeming and intermediate values

Codec under test	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
EVS-SWB 48 DTX => LC3plus 32	4.15	4.01	79.68	117.92	30.08	36.7
EVS-SWB 32 DTX => LC3plus 32	4.26	4.12	82.67	122.36	25.64	35.2
EVS-SWB 24.4 DTX => LC3plus 32	4.2	4.06	81.01	119.89	28.11	33.7
EVS-SWB 16.4 DTX => LC3plus 32	4.08	3.95	78.13	115.63	32.37	37.3
EVS-SWB 13.2 DTX => LC3plus 32	3.95	3.82	75.01	111.01	36.99	43.6
EVS-SWB 9.6 DTX => LC3plus 32	3.9	3.77	73.86	109.31	38.69	49.2
LC3plus 32 => EVS-SWB 48 DTX	4.24	4.1	82.13	121.55	26.45	36.7
LC3plus 32 => EVS-SWB 32 DTX	4.19	4.05	80.75	119.51	28.49	35.2
LC3plus 32 => EVS-SWB 24.4 DTX	4.32	4.18	84.47	125.02	22.98	33.7
LC3plus 32 => EVS-SWB 16.4 DTX	4.25	4.11	82.4	121.95	26.05	37.3
LC3plus 32 => EVS-SWB 13.2 DTX	4.21	4.07	81.27	120.28	27.72	43.6
LC3plus 32 => EVS-SWB 9.6 DTX	4.06	3.93	77.65	114.93	33.07	49.2
LC3plus 32 => LC3plus 32	4.04	3.91	77.16	114.19	33.81	53.0
LC3plus 32 => LC3plus 32 => LC3plus 32	3.79	3.67	71.64	106.02	41.98	79.5
EVS-SWB 48 DTX => LC3plus 48	4.48	4.33	89.66	132.7	15.3	16.49
EVS-SWB 32 DTX => LC3plus 48	4.5	4.35	90.48	133.92	14.08	14.99
EVS-SWB 24.4 DTX => LC3plus 48	4.47	4.32	89.28	132.14	15.86	13.49
EVS-SWB 16.4 DTX => LC3plus 48	4.48	4.33	89.66	132.7	15.3	17.09
EVS-SWB 13.2 DTX => LC3plus 48	4.29	4.15	83.57	123.69	24.31	23.39
EVS-SWB 9.6 DTX => LC3plus 48	4.2	4.06	81.01	119.89	28.11	28.99
LC3plus 48 => EVS-SWB 48 DTX	4.48	4.33	89.66	132.7	15.3	16.49
LC3plus 48 => EVS-SWB 32 DTX	4.32	4.18	84.47	125.02	22.98	14.99
LC3plus 48 => EVS-SWB 24.4 DTX	4.45	4.3	88.51	131.0	17.0	13.49
LC3plus 48 => EVS-SWB 16.4 DTX	4.38	4.24	86.38	127.84	20.16	17.09
LC3plus 48 => EVS-SWB 13.2 DTX	4.27	4.13	82.98	122.81	25.19	23.39
LC3plus 48 => EVS-SWB 9.6 DTX	4.28	4.14	83.26	123.22	24.78	28.99
LC3plus 48 => LC3plus 48	4.49	4.34	90.09	133.33	14.67	12.58
LC3plus 48 => LC3plus 48 => LC3plus 48	4.45	4.3	88.51	131.0	17.0	18.87
EVS-SWB 48 DTX => LC3plus 64	4.55	4.4	92.75	137.27	10.73	15.48
EVS-SWB 32 DTX => LC3plus 64	4.49	4.34	90.09	133.33	14.67	13.98
EVS-SWB 24.4 DTX => LC3plus 64	4.51	4.36	90.89	134.52	13.48	12.48
EVS-SWB 16.4 DTX => LC3plus 64	4.41	4.26	87.07	128.86	19.14	16.08
EVS-SWB 13.2 DTX => LC3plus 64	4.34	4.2	85.1	125.95	22.05	22.38
EVS-SWB 9.6 DTX => LC3plus 64	4.25	4.11	82.4	121.95	26.05	27.98
LC3plus 64 => EVS-SWB 48 DTX	4.48	4.33	89.66	132.7	15.3	15.48
LC3plus 64 => EVS-SWB 32 DTX	4.38	4.23	86.06	127.37	20.63	13.98
LC3plus 64 => EVS-SWB 24.4 DTX	4.53	4.38	91.79	135.85	12.15	12.48
LC3plus 64 => EVS-SWB 16.4 DTX	4.44	4.29	88.16	130.47	17.53	16.08

Codec under test	MOS	MOS_n	R_{NB}	R_{FB}	*Ie,FB,obs*	*Ie,FB,def*
LC3plus 64 => EVS-SWB 13.2 DTX	4.36	4.22	85.75	126.91	21.09	22.38
LC3plus 64 => EVS-SWB 9.6 DTX	4.23	4.09	81.83	121.11	26.89	27.98
LC3plus 64 => LC3plus 64	4.59	4.44	94.96	140.53	7.47	10.56
LC3plus 64 => LC3plus 64 => LC3plus 64	4.56	4.41	93.29	138.07	9.93	15.84

As done before the pairs of defined and observed *Ie,FB* are plotted against each other in Figure E.46. If more than four out of 14 tandem conditions show major deviations from the interpolation line, the additivity property should not be regarded as being satisfied. This condition is fulfilled for LC3plus at 48 and 64 kbit/s. For 32 bit/s the data points are all below the regression line. That means that the observed values *Ie,FB,obs* are better than expected by the *Ie,FB,def* calculation. Still the regression only works poorly for this bitrate.

The conditions outside of the confidence interval are:

"LC3plus 32 => EVS-SWB 13.2 DTX" with an error of -12.37
"LC3plus 32 => EVS-SWB 9.6 DTX" with an error of -11.6
"LC3plus 32 => LC3plus 32" with an error of -13.96
"LC3plus 32 => LC3plus 32 => LC3plus 32" with an error of -27.47

The biggest deviations show at triple self tandeming where LC3plus performs better than predicted by the model.

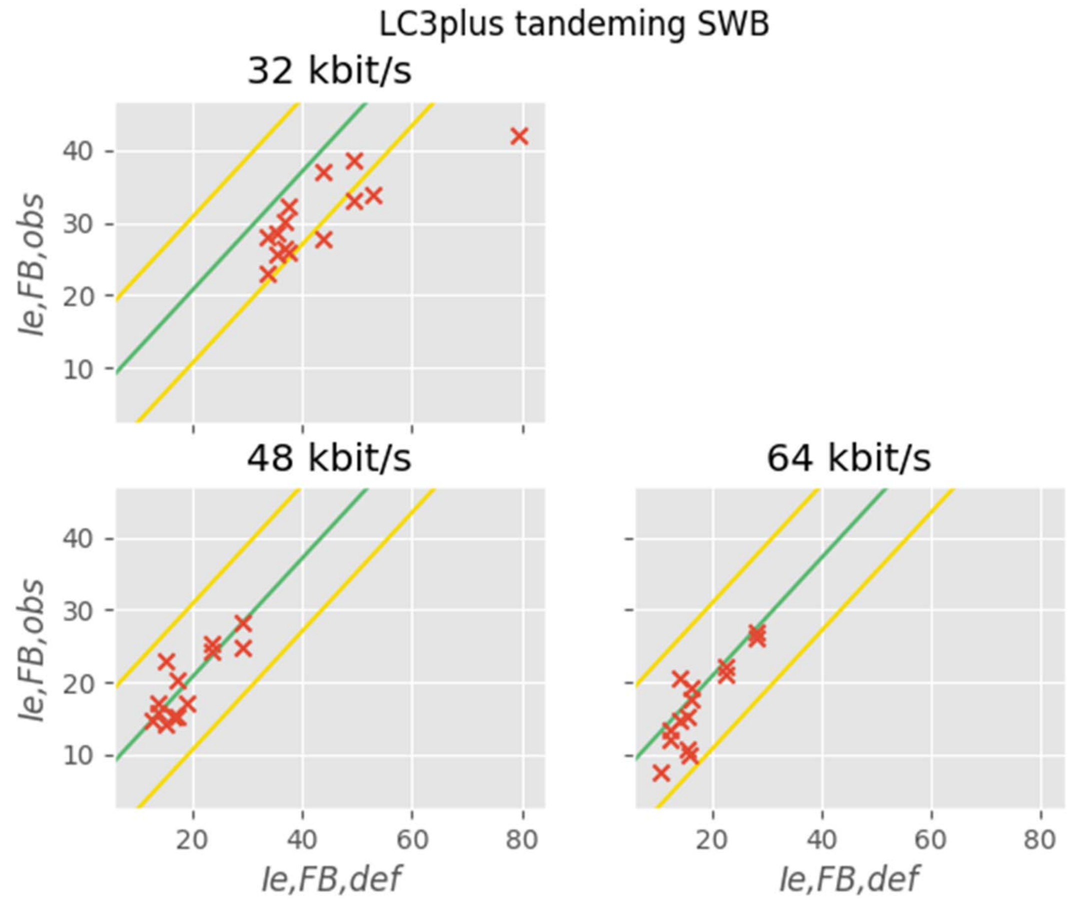

Figure E.46: Observed and defined *Ie,FB* for tandeming conditions with original regression line

E.3.3.2.2.9 Determination of *Ie,FB* for LC3plus under transmission error conditions

To determine Ie values for LC3plus under transmission error conditions the same reference conditions as in Table E.35 plus 16 supplementary reference conditions from Appendix V/G 113 Amd. 2 [26] are needed.

Figure E.47 shows the Recommendation ITU-T P.800 results of these 12 new reference codecs with the additional LC3plus conditions to be evaluated at different bitrates, error rates and Error Pattern Frame sizes (EPFsizes).

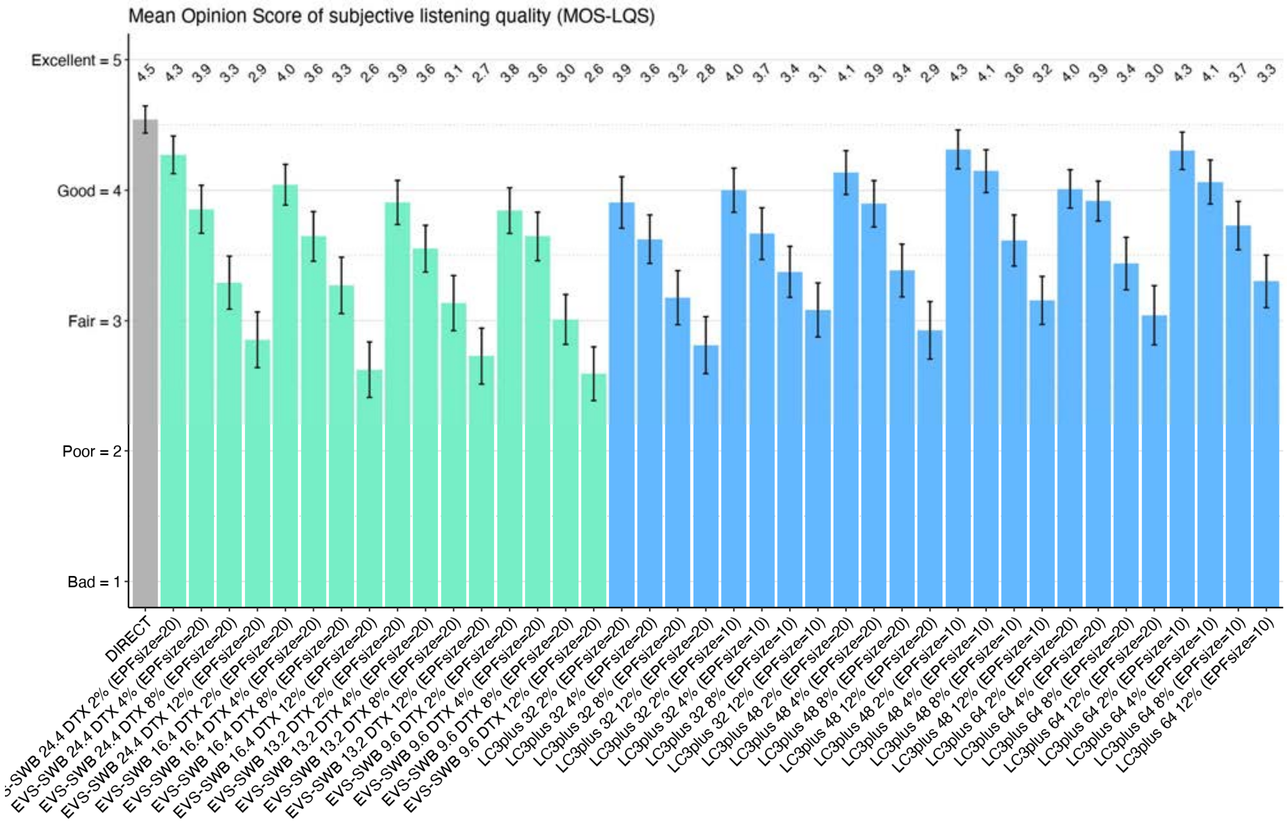

Figure E.47: Recommendation ITU-T P.800 results for SWB transmission error speech signals. Mean scores and 95 % confidence intervals

For the error-prone reference conditions an effective equipment impairment factor *Ie,eff,FB* is calculated according to the following equation taken from Recommendation ITU-T G.107.2 [28].

$$Ie,eff,FB = Ie,FB + (132 - Ie,FB) \cdot \frac{Ppl}{Ppl+Bpl}$$

The actual values for *Ppl* and q are determined numerically from the randomly generated error patterns. The packet loss robustness factor *Bpl* is taken from G.113 Amendment 2 [26].

Table E.38: Parameters used to calculate *Ie,eff,FB*

Condition	*Ie*	*Ppl* (assumed)	*Bpl*	*Ppl* (measured)	*q* (measured)	*BurstR*	*Ie,FB,eff*	*Ie,FB,obs*
EVS-SWB@24.4	7.2	2	11.4	2.09	0.99	0.99	26.57	20.57
EVS-SWB@24.4	7.2	4	11.4	4.18	0.97	0.99	40.67	37.06
EVS-SWB@24.4	7.2	8	11.4	8.37	0.92	1.0	60.02	54.44
EVS-SWB@24.4	7.2	12	11.4	12.54	0.87	1.01	72.58	66.78
EVS-SWB@16.4	10.8	2	10.3	2.09	0.99	0.99	31.28	30.3
EVS-SWB@16.4	10.8	4	10.3	4.18	0.97	0.99	45.77	43.84
EVS-SWB@16.4	10.8	8	10.3	8.37	0.92	1.0	65.12	55.05
EVS-SWB@16.4	10.8	12	10.3	12.54	0.87	1.01	77.35	73.13
EVS-SWB@13.2	17.1	2	11.7	2.09	0.99	0.99	34.54	35.26
EVS-SWB@13.2	17.1	4	11.7	4.18	0.97	0.99	47.33	46.75

EVS-SWB@13.2	17.1	8	11.7	8.37	0.92	1.0	65.0	58.92
EVS-SWB@13.2	17.1	12	11.7	12.54	0.87	1.01	76.54	70.25
EVS-SWB@9.6	22.7	2	13.0	2.09	0.99	0.99	37.86	37.4
EVS-SWB@9.6	22.7	4	13.0	4.18	0.97	0.99	49.28	43.84
EVS-SWB@9.6	22.7	8	13.0	8.37	0.92	1.0	65.5	62.44
EVS-SWB@9.6	22.7	12	13.0	12.54	0.87	1.01	76.37	74.0

Next the observed *Ie,FB* values are calculated according to the previous clause. The results listed in Table E.38 show increasing impairment factors for increasing transmission error rates. The new pairs of *Ie,FB,obs* and *Ie,eff,FB* are added in the scatterplot Figure E.48. It can be observed that the original interpolation line still describes the new error-prone reference conditions adequately with a strong fit of R^2 = 0.95.

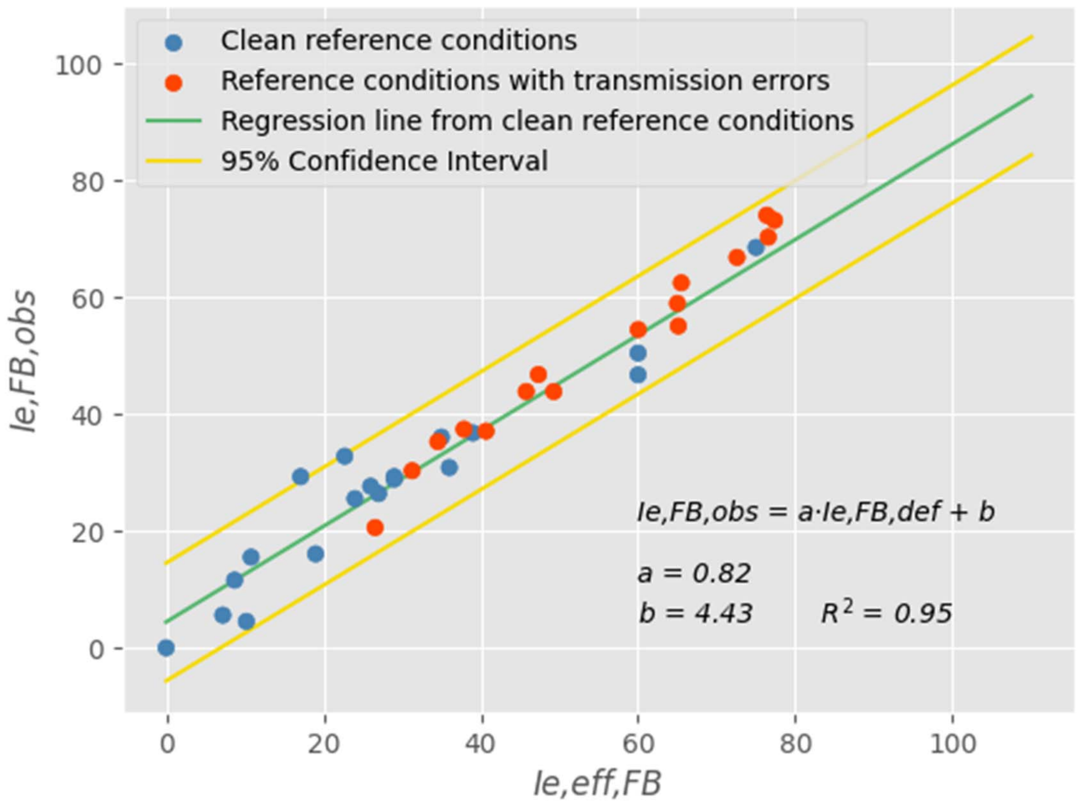

Figure E.48: Original regression line, observed *Ie,FB,obs* and defined *Ie,eff,FB* for clean and error-prone reference conditions

With the coefficients a and b stable *Ie,FB* values for LC3plus with transmission errors can be derived (Table E.39) without existing *Bpl* values.

Table E.39: Stable *Ie,FB* value for LC3plus under transmission error conditions

Codec under Test	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32 2% (EPFsize=20)	35.26	37.7
LC3plus@32 4% (EPFsize=20)	44.5	49.0
LC3plus@32 8% (EPFsize=20)	57.73	65.17
LC3plus@32 12% (EPFsize=20)	67.92	77.63
LC3plus@32 2% (EPFsize=10)	31.88	33.57
LC3plus@32 4% (EPFsize=10)	43.18	47.38

Codec under Test	*Ie,FB,obs*	*Ie,FB,def*
LC3plus@32 8% (EPFsize=10)	52.03	58.2
LC3plus@32 12% (EPFsize=10)	60.39	68.42
LC3plus@48 2% (EPFsize=20)	26.62	27.13
LC3plus@48 4% (EPFsize=20)	35.61	38.13
LC3plus@48 8% (EPFsize=20)	51.74	57.85
LC3plus@48 12% (EPFsize=20)	64.75	73.76
LC3plus@48 2% (EPFsize=10)	18.46	17.16
LC3plus@48 4% (EPFsize=10)	26.16	26.57
LC3plus@48 8% (EPFsize=10)	44.81	49.38
LC3plus@48 12% (EPFsize=10)	58.33	65.91
LC3plus@64 2% (EPFsize=20)	31.51	33.11
LC3plus@64 4% (EPFsize=20)	34.88	37.23
LC3plus@64 8% (EPFsize=20)	50.17	55.93
LC3plus@64 12% (EPFsize=20)	61.54	69.83
LC3plus@64 2% (EPFsize=10)	19.03	17.85
LC3plus@64 4% (EPFsize=10)	29.49	30.64
LC3plus@64 8% (EPFsize=10)	41.2	44.96
LC3plus@64 12% (EPFsize=10)	54.15	60.8

The packet loss robustness factor *Bpl* for LC3plus at different bitrates and Error Pattern Frame sizes (EPFsizes) can be calculated by identifying the *Bpl* that fits the *Ie,eff,FB* curve the best. The optimal value can be determined by calculating the least square error for the pairs of *Ie,FB,obs(Ppl)* and *Ie,eff,FB(Ppl).* Table E.40 shows that a shorter EPFsize is expectedly associated with a higher *Bpl* for the same bitrate.

Table E.40: Packet loss robustness factors for LC3plus

Bitrate (kbit/s)	EPFsize (ms)	*Ie,FB,def*	Bpl
32	20	26.5	14.2
32	10	26.5	18.7
48	20	6.29	11.4
48	10	6.29	15.8
64	20	5.28	11.6
64	10	5.28	16.7

Figure E.49 and Figure E.50 show *Ie,eff,FB(Ppl)* with fixed *Bpl* from Table E.40. All approximations using the *Ie,eff,FB* formula show a strong fit according to R^2 for both EPFsizes.

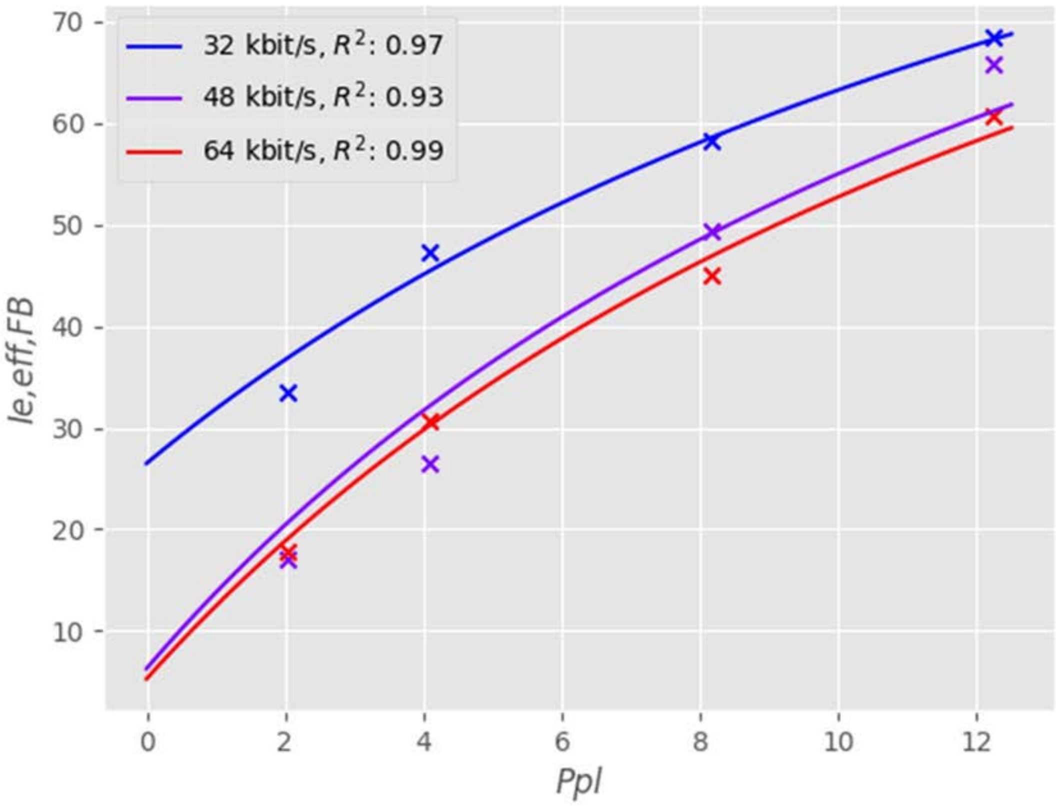

Figure E.49: *Ie,eff,FB* approximation with best fitting *Bpl* and observed *Ie* values vs. packet loss rate for 10 ms EPFsize

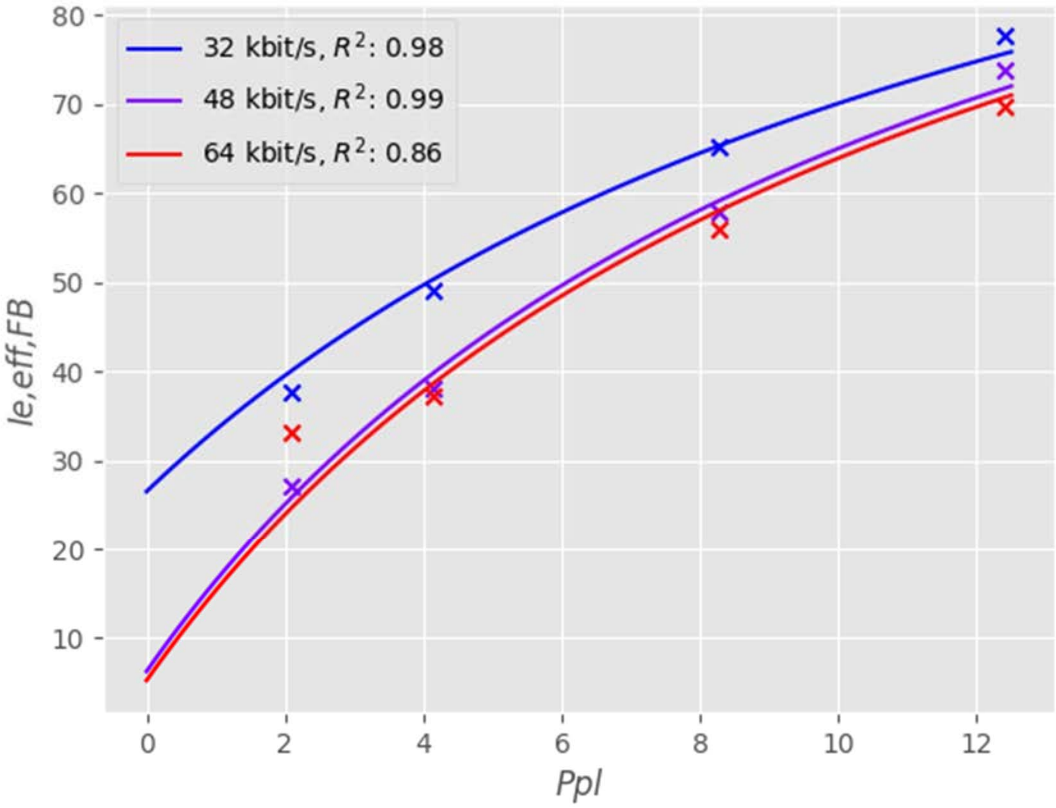

Figure E.50: Ie,eff,FB approximation with best fitting Bpl and observed Ie values vs. packet loss rate for 20 ms EPFsize

E.4 Overview and proposed provisional Ie values for the LC3plus codec

E.4.1 Proposed provisional Ie,NB and Bpl factors for LC3plus

This clause provides a comparison of objective and subjective *Ie,NB* values derived according to Recommendations ITU-T P.833 [20] and P.834 [21]. Generally, the NB-model shows some weaknesses when calculating Ie for codecs with quality as good or better than the reference condition G.711 and for more advanced PLC technologies. Table E.41 shows that both objective and subjective assessment result in similar *Ie,NB* factors with a maximum difference of 3.9.

Table E.41: Ie,NB for LC3plus derived by objective and subjective assessment

Codec under Test	***Ie,NB,def*** **objective assessment**	***Ie,NB,def*** **subjective assessment**
LC3plus@16	**10.1**	13.2
LC3plus@20	0	**3.9**
LC3plus@24	0	**0**
LC3plus@32	0	**0**

Table E.42 displays the *Bpl* NB values for transmission error conditions. Considering the partially bad fitting behaviour of the *Ie,eff* approximation (see Figures E.8, E.9, E.17, E.18) it is recommended to use the objectively derived *Ie,NB* and *Bpl* values for LC3plus at 16kbp/s and the subjectively derived values for LC3plus at 20, 24 and 32 kbp/s.

Table E.42: Packet loss robustness factors for LC3plus NB derived by objective and subjective assessment

Bitrate (kbit/s)	***Bpl*** **objective assessment**	***Bpl*** **subjective assessment**
LC3plus@32 (EPFsize=10)	49.1	**41.5**
LC3plus@32 (EPFsize=20)	25.1	**31.1**
LC3plus@24 (EPFsize=10)	39.1	**30.9**
LC3plus@24 (EPFsize=20)	22.2	**20**
LC3plus@16 (EPFsize=10)	**27.6**	14.3
LC3plus@16 (EPFsize=20)	**20**	15.4

To compare impairment and packet loss robustness to other codecs, Figures E.51 and E.52 rank the Ie,NB and Bpl values for LC3plus alongside the known standardized parameters for NB. The lowest impairment factors and highest robustness factors are ranked on top.

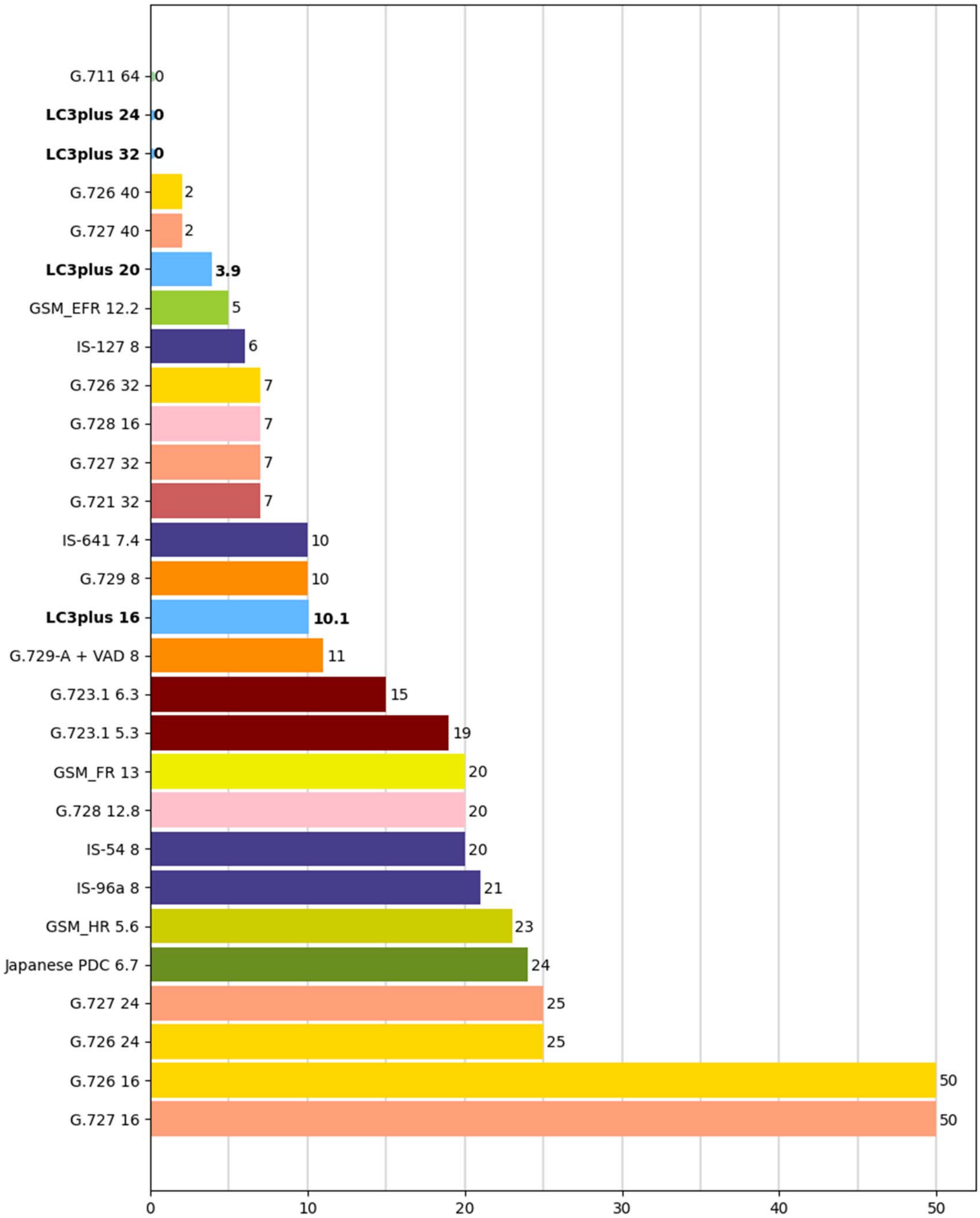

Figure E.51: Ranked *Ie,NB* of official reference NB codecs and LC3plus at bitrates in kbit/s

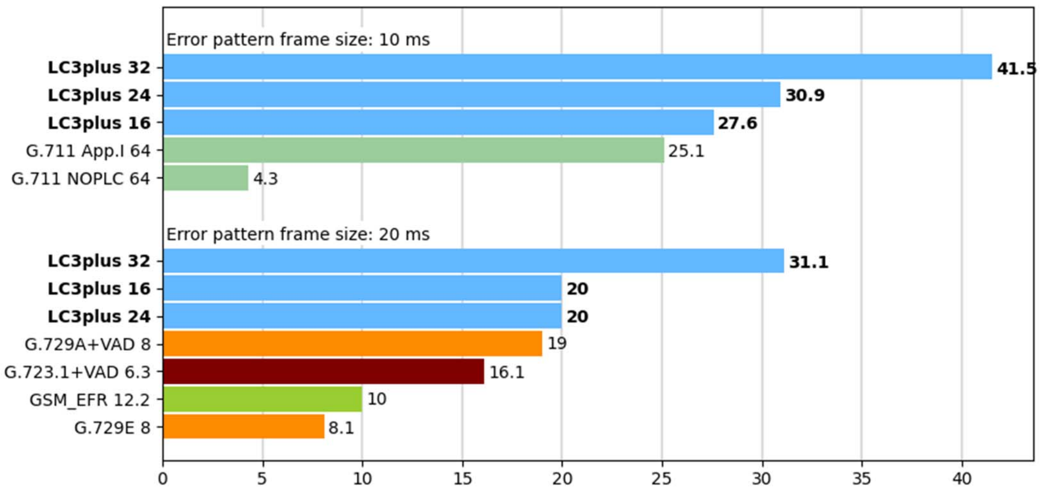

Figure E.52: Ranked *Bpl* factors of official reference NB codecs and LC3plus at bitrates in kbit/s

E.4.2 Proposed provisional Ie,WB and Bpl factors for LC3plus

This clause provides a comparison of objective and subjective *Ie,WB* values derived according to Recommendations ITU-T P.833.1 [24] and P.834.1 [25]. LC3plus at 16 and 24 kbit/s in objective evaluation and LC3plus at 16 kbit/s in subjective evaluation did not fulfil the condition of additivity and is therefore not listed int the following tables. The negative effect of LC3plus@16 in tandeming is much weaker than expected by the additive model. Table E.43 shows that both objective and subjective assessment result in very similar *Ie,NB* factors.

Table E.43: Ie,WB for LC3plus derived by objective and subjective assessment

Codec under Test	*Ie,WB,def* objective assessment	*Ie,WB,def* subjective assessment	*Mean from subj. and obj. Ie,WB,def*
LC3plus@24	-	15.9	**15.9**
LC3plus@32	2.8	4	**3.4**
LC3plus@48	0	0	**0**

Table E.44 displays the *Bpl* WB values for transmission error conditions. It is recommended to use the mean of subjectively and objectively derived, valid *Ie,WB* and *Bpl* values.

Table E.44: Packet loss robustness factors for LC3plus WB derived by objective and subjective assessment

Bitrate (kbit/s)	*Bpl* objective assessment	*Bpl* subjective assessment	Mean from subj. and obj. *Bpl*
LC3plus@48 (EPFsize=20)	7	14.9	**11**
LC3plus@48 (EPFsize=10)	11	24.5	**17.8**
LC3plus@32 (EPFsize=20)	7	17	**12**
LC3plus@32 (EPFsize=10)	10.2	24.5	**17.4**
LC3plus@24 (EPFsize=20)	-	16.9	**16.9**
LC3plus@24 (EPFsize=10)	-	25.4	**25.4**

To compare impairment and packet loss robustness to other codecs, Figures E.53 and E.54 rank the Ie,WB and Bpl values for LC3plus alongside the known standardized parameters for WB. The lowest impairment factors and highest robustness factors are ranked on top.

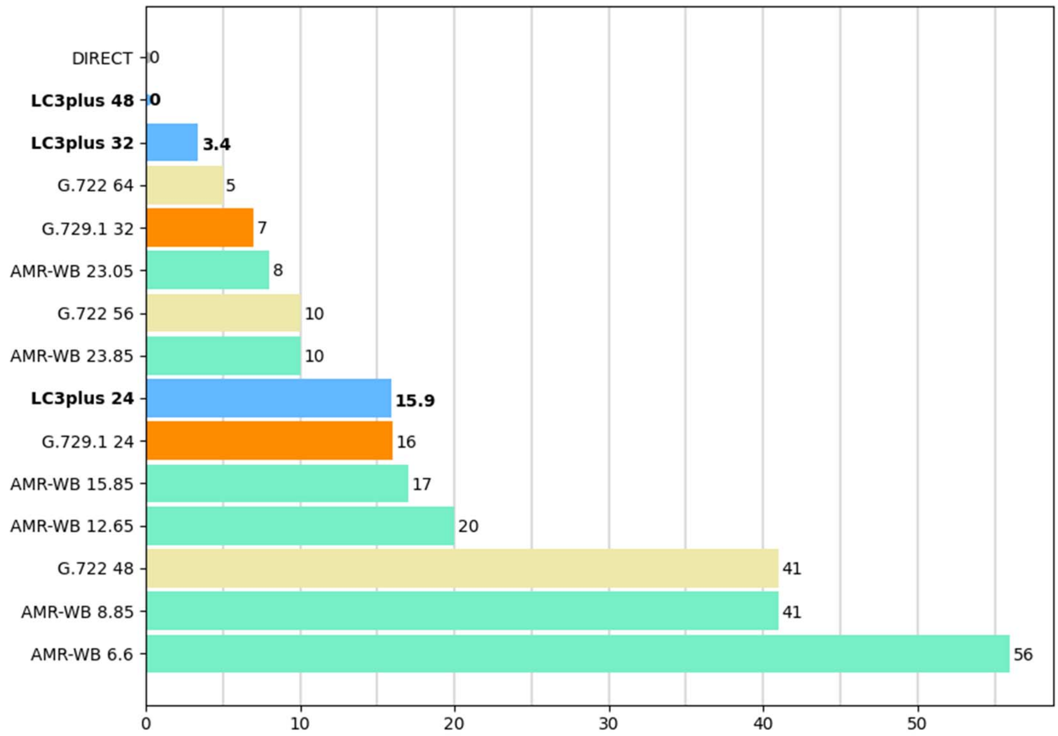

Figure E.53: Ranked *Ie,WB* of official reference WB codecs and LC3plus at bitrates in kbit/s

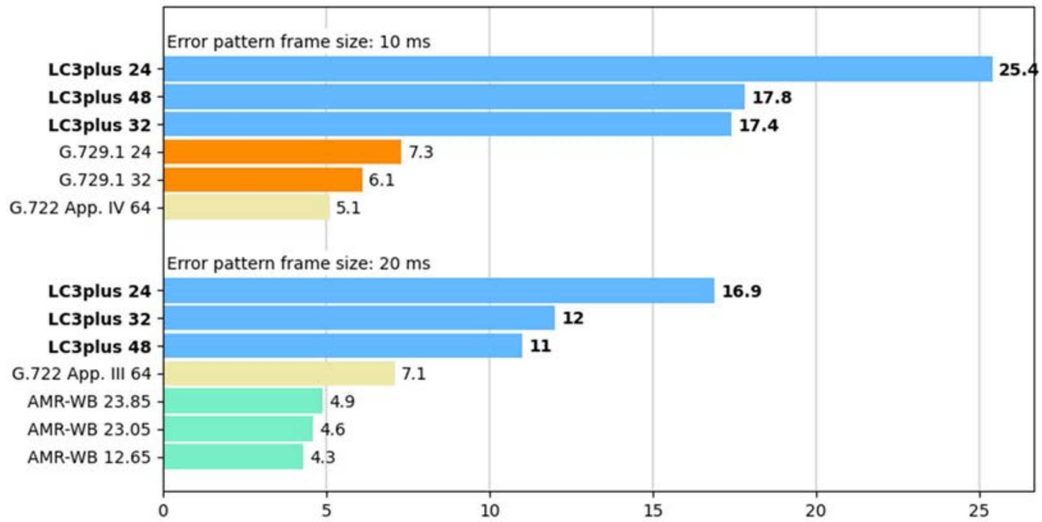

Figure E.54: Ranked *Bpl* factors of official reference WB codecs and LC3plus at bitrates in kbit/s

E.4.3 Proposed provisional Ie,FB and Bpl factors for LC3plus

This clause provides a comparison of objective and subjective *Ie,FB* and *Bpl* values derived according to clause E.2.3. Table E.45 shows that the objectively and subjectively assessed *Ie,FB* values differ significantly.

Table E.45: Ie,FB for LC3plus derived by objective and subjective assessment

Codec under Test	*Ie,FB,def* objective assessment	*Ie,FB,def* subjective assessment	*Mean from subj. and obj. Ie,WB,def*
LC3plus@32	12.4	26.5	**19.5**
LC3plus@48	1.3	6.3	**3.8**
LC3plus@64	0	5.3	**2.6**

Table E.46 displays the *Bpl* FB values for transmission error conditions. It is recommended to use the mean of subjectively and objectively derived *Ie,FB* and *Bpl* values.

Table E.46: Packet loss robustness factors for LC3plus FB derived by objective and subjective assessment

Bitrate (kbit/s)	*Bpl* objective assessment	*Bpl* subjective assessment	Mean from subj. and obj. *Bpl*
LC3plus@32 (EPFsize=20)	11	14.2	**12.6**
LC3plus@32 (EPFsize=10)	13.1	18.7	**15.9**
LC3plus@48 (EPFsize=20)	9.2	11.4	**10.3**
LC3plus@48 (EPFsize=10)	11.5	15.8	**13.7**
LC3plus@64 (EPFsize=20)	9	11.6	**10.3**
LC3plus@64 (EPFsize=10)	11.1	16.7	**13.9**

To compare impairment and packet loss robustness to other codecs, Figures E.55 and E.56 rank the Ie,FB and Bpl values for LC3plus alongside the known standardized parameters for FB. The lowest impairment factors and highest robustness factors are ranked on top.

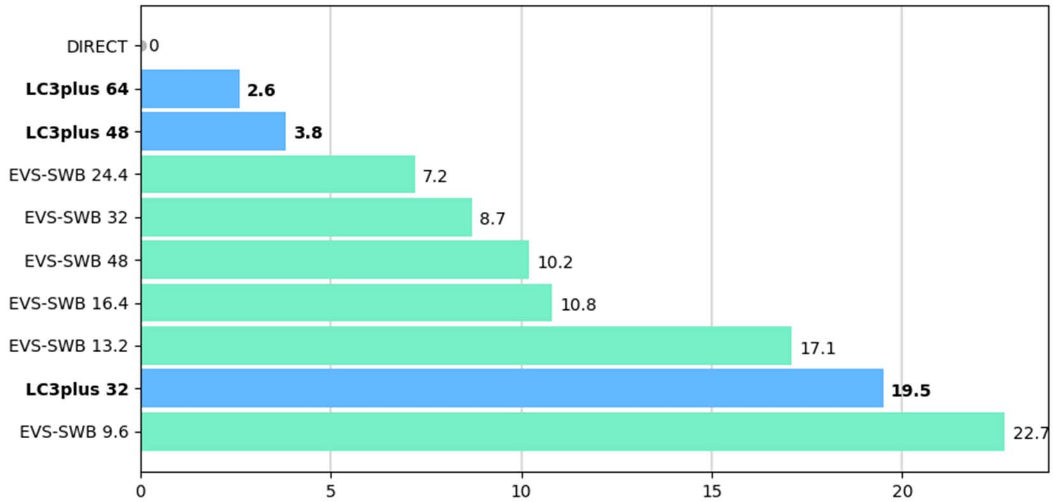

Figure E.55: Ranked *Ie,FB* of official reference FB codecs and LC3plus at bitrates in kbit/s

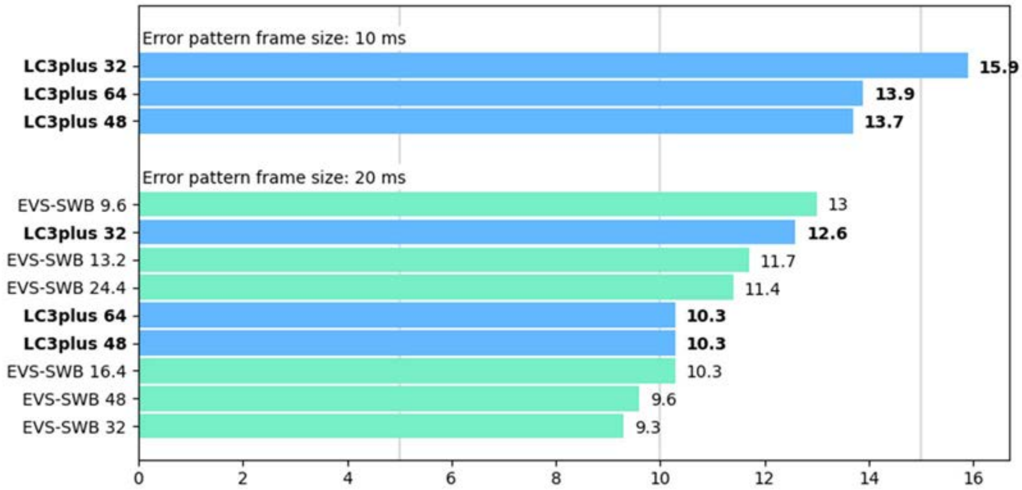

Figure E.56: Ranked *Bpl* factors of official reference FB codecs and LC3plus at bitrates in kbit/s

Annex (informative): Bibliography

- ETSI TS 126 071: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Mandatory speech CODEC speech processing functions; AMR speech Codec; General description (3GPP TS 26.071)".

History

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