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Technical Report

**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
LTE;
Performance characterization of 3GPP audio codecs
(3GPP TR 26.936 version 8.0.0 Release 8)**

Reference

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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Foreword

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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where:

- x the first digit:
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- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
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1 Scope

The present document comprises the Technical Report for the Characterization of the 3GPP Audio Codecs, Enhanced aacPlus (*Eaac+*) and Extended AMR-WB (*AMR-WB+*), standardized by 3GPP in Release 6 for Packet-switched Streaming Service (PSS), Multimedia Messaging Service (MMS), Multimedia Broadcast and Multicast Service (MBMS), and IMS Messaging Service and Presence Service.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] ITU-R Recommendation BS.1534-1 (2003): "Method for the subjective assessment of intermediate quality level of coding systems".

[2] ISO/IEC JTC1/SC29/WG11/N5571 (2003): "Report on Informal MPEG-4 Extension 1 (Bandwidth Extension) Verification Tests".

[3] Tech 3296 (2003): "EBU Subjective listening tests on low bit-rate audio codecs".

[4] 3GPP TS 26.273: "ANSI-C code for the fixed-point Extended Adaptive Multi-Rate - Wideband (AMR-WB+) speech codec".

[5] 3GPP TS 26.290: "Audio codec processing functions; Extended Adaptive Multi-Rate - Wideband (AMR-WB+) codec; Transcoding functions".

[6] 3GPP TS 26.410: "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Floating-point ANSI-C code".

[7] 3GPP TS 26.411: "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Fixed-point ANSI-C code".

3 Abbreviations

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

ACR	Absolute Category Rating method
AMR-WB+	Extended AMR-WB
BLER	Block Error Rate
DCR	Degradation Category Rating method
DMOS	Degradation Mean Opinion Score
Eaac+	Enhanced aacPlus
EBU	European Broadcasting Union
EGPRS	Enhanced General Packet Radio Service
FEC	Forward Error Correction
FER	Frame Error Rate
HRTF	Head Related Transfer Function
ISF	Internal Sampling Frequency
ITU-T	International Telecommunications Union - Telecommunications

MBMS	Multi-media Broadcast and Multicast Services
MMS	Multi-media Messaging Service
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
MUSHRA	Multiple Stimulus with Hidden Reference and Anchors method
PLR	Packet Loss Rate
PSS	Packet-switched Streaming Service
ROHC	Robust Header Compression
RTP	Real-time transport protocol
RLC-PDU	Radio Link Control - Protocol Data Unit
SEAQ	System for the Evaluation of Audio Quality
UTRAN	UMTS Terrestrial Radio Access Network

4 Subjective test results

The main body of the present document summarizes the results from official tests conducted by 3GPP.

Further subjective test results were provided into the standardization process by various parties. Some of these results were part of the verification tasks. Two other contributions from codec proponents assessing the codecs across different bit rates in the low rate segment up to 24 kbps were provided in earlier phases of the process and are also attached to the present document (see S4-040439 and S4-040710 in attachment "TR 26.936 Annex B2 Additional Documents.zip").

In addition, there are further reports from other bodies. Clause A.1 comprises test results for the two standardized codecs carried out by ITU-T, where the 3GPP audio codecs served as reference codecs. The Annex also contains references to tests on Eaac+ carried out in MPEG and EBU, which provide information for configurations using the Eaac+ decoder in combination with a non-standardized encoder.

In the reporting of subjective test results, it is generally agreed that comparisons of results are valid only for conditions conducted within the same experiment. It is not valid, for example, to directly or statistically compare subjective test results for one codec across two bit-rates when those results have been obtained from different experiments. In general, this principle will be observed in the subjective test results presented in the following clauses.

4.1 Organization of the subjective test results

This report comprises data from subjective tests derived from standardization exercises organized in 3GPP. Each of the test exercises was conducted in accordance with a test plan and results were provided in a test report. Table 1 summarizes these test exercises including a series label, a description of the tests, and the specific Attachments containing the appropriate test plans and test reports.

Table 1: Subjective Test Series involved in the Technical Report

Standardization exercise	Label	Test plan	Test report
3GPP audio codec characterization test, phase 1	CT-P1	Attachment 1A	Attachment 1B
3GPP audio codec characterization test, phase 2	CT-P2	Attachment 1A	Attachment 1C
3GPP audio codec low rate selection test	ST-LR	Attachment 1D	Attachment 1E
3GPP audio codec high rate selection test	ST-HR	Attachment 1F	Attachment 1G

Table 2 lists the subjective tests that provided the results reported in the present document including details such as the test parameters, mode (mono vs. stereo), and number of listening labs. All of the subjective tests described in this report involved some common test parameters. These include the use of 15 subjects and 12 test items where the test items are sub-divided into three classes of Audio Content - four items each for Music-only, Speech-only, and Mixed Music+Speech audio content. All experiments in CT-P1 and CT-P2 used the same 12 items of material. The ST-LR tests used 24 items in each experiment, but the test items were split into two sets of 12 test items and allocated across to different labs running the same experiment. All experiments in ST-HR used the same 12 test items.

Table 2: Summary of subjective tests involved in the Technical Report

Series	Test parameter(s)	Mode	# Labs
CT-P1	3 bit-rates	mono	2
	3 bit-rates	stereo	2
CT-P2	4 PLR's, EGPRS	mono	1
	4 PLR's, EGPRS	stereo	1
	3 PLR's, UTRAN - Lower rates	stereo	1
	3 PLR's, UTRAN - Higher rates	stereo	1
ST-LR	14 kbps, PSS	mono	4
	18 kbps, PSS	stereo	4
	24 kbps, PSS	mono	4
	24 kbps, PSS	stereo	4
	14 kbps, MMS, 16 kHz sampling rate	mono	4
	18 kbps, MMS	stereo	4
	14 kbps, PSS, 3 % FER	mono	4
	24 kbps, MMS, 3 % FER	stereo	4
ST-HR	32 kbps, PSS	stereo	2
	48 kbps, PSS	stereo	2
	32 kbps, PSS, 1 % and 3 % FER	stereo	2

It should be noted that, for both audio codecs, the codecs used in the Characterization Test (i.e. CT-P1 and CT-P2 test series) were different (e.g., bug fixes, optimized configurations) from the candidate codecs used in the earlier Selection Test (i.e. ST-LR and ST-HR test series).

The Selection test results also provide information how the selected codecs perform in relation to Release 5 audio codecs.

The clauses of the present document are organized as follows:

- Clause 5: Performance Characterization for Audio Content:
  - This clause comprises a quality evaluation of the codecs across audio content. It contains results from codec characterization both without and with packet loss. It also contains a similar evaluation from the selection tests.
- Clause 6: Performance Characterization over Bit-rate:
  - This clause comprises the test results from codec characterization assessing the performance of the codecs across bit rate. It highlights quality vs. bit-rate in the lower bit-rate range. This clause contains relevant data for PSS, MBMS and MMS services.
- Clause 7: Performance based on Selection test results:
  - This clause comprises test results from the codec selection tests demonstrating the performance of the selected codecs relative to that of reference codecs (including audio and wideband speech codecs specified in Release 5). It covers an intrinsic quality comparison of the codecs and a quality comparison under stressed operating conditions in a lower bit-rate segment up to 24 kbps. The intrinsic quality comparison was designed to target PSS applications without channel impairments, and the stressed operating conditions contained terminal-generated MMS scenarios and PSS with channel impairments. Also included are high-rate codec selection results for a higher bit-rate segment assessing the performance of the *Eaac+* codec relative to reference codecs at 32 kbps and 48 kbps. These tests were targeting PSS and MMS applications and did not include *AMR-WB+* codec.
- Clause 8: Performance Characterization for Error Conditions:
  - This clause comprises the test results from codec characterization assessing the codec performance for various packet loss rates. The tests used the RTP packetization schemes specified for MBMS services as well as a packet loss simulator designed for such purposes. This clause is relevant for the design of application layer FEC and the definition of target BLER for MBMS services without FEC.

- Clause 9: Results of Verification Tests:
  - This clause comprises results from codec verification tasks which checked and assessed various codec aspects such as complexity, verification of the fixed point code, frequency response, delay, codec performance with 3D audio signals, rate switching performance, and content dependency.
- Annex A: Test results from other bodies:
  - Annex A includes test results carried out by ITU-T, where the two 3GPP audio codecs participated as reference codecs. These results consist of two parts, one part obtained by MUSHRA testing of music and mixed material, the other part obtained by MOS and DMOS testing (ACR and DCR) using speech material. The Annex also contains references to tests on *Eaac+* carried out in bodies outside of 3GPP, which provide information for configurations using the *Eaac+* decoder in combination with a non-standardized encoder.

All the results shown in the present document are generally valid for all Release 6 services and applications which are based on the Release 6 audio codecs. These are Packet switched Streaming Service (PSS), Multimedia Messaging Service (MMS), Multimedia Broadcast and Multicast Service (MBMS), IMS Messaging Service and Presence Service. During codec selection certain tests were designed to target a specific application, but are generally valid for all applications with similar constraints (e.g. bit-rate, mono/stereo). The only exception is terminal-generated MMS for which tests with reduced input signal bandwidth were carried out (restricted to 16 kHz sampling rate in some tests as reported in clauses 5.3 and 7.2) and for which a particularly low-complexity encoder for *AMR-WB +* was used (9.75k-lc in the results reported in clause 6).

4.2 Subjective test method

The subjective test results described in the present document were derived using the MUSHRA (**MU**ltiple **S**timulus with **H**idden **R**eference and **A**nchors) test methodology. The MUSHRA method is an ITU-R standardized test methodology for the subjective assessment of intermediate audio quality [1]. On each trial in a MUSHRA test the subject is presented with an unprocessed audio sample - the "Open Reference" (OR). By definition the quality of the OR is a score of 100 on the MUSHRA quality scale. The subjects' task is to then evaluate the quality of the same audio sample processed by each of the conditions involved in the test as well as the unprocessed condition, the Hidden Reference (HR), and two or more degraded Anchor conditions, typically low-pass filtered at 3.5 kHz and low-pass filtered at 7.0 kHz.

Figure 1 shows an example of a subject's response interface for a typical MUSHRA trial involving nine audio conditions. The subject is required to listen, first, to the OR (***Ref*** button in the figure) and then to each of the test conditions (buttons ***A*** through ***I*** in the figure). The assignment of test conditions is randomized among the buttons for each trial. Subjects register their ratings, 1 to 100, using the scale sliders above each button. The subject's task is to identify the HR condition and give it a rating of 100 and then to rate the remainder of the conditions relative to the HR condition. Subjects may listen to the samples as many times as they want and adjust their ratings accordingly. Subject's ratings are used in test analyses only if the subject can reliably identify the HR and correctly order the anchors and the HR.

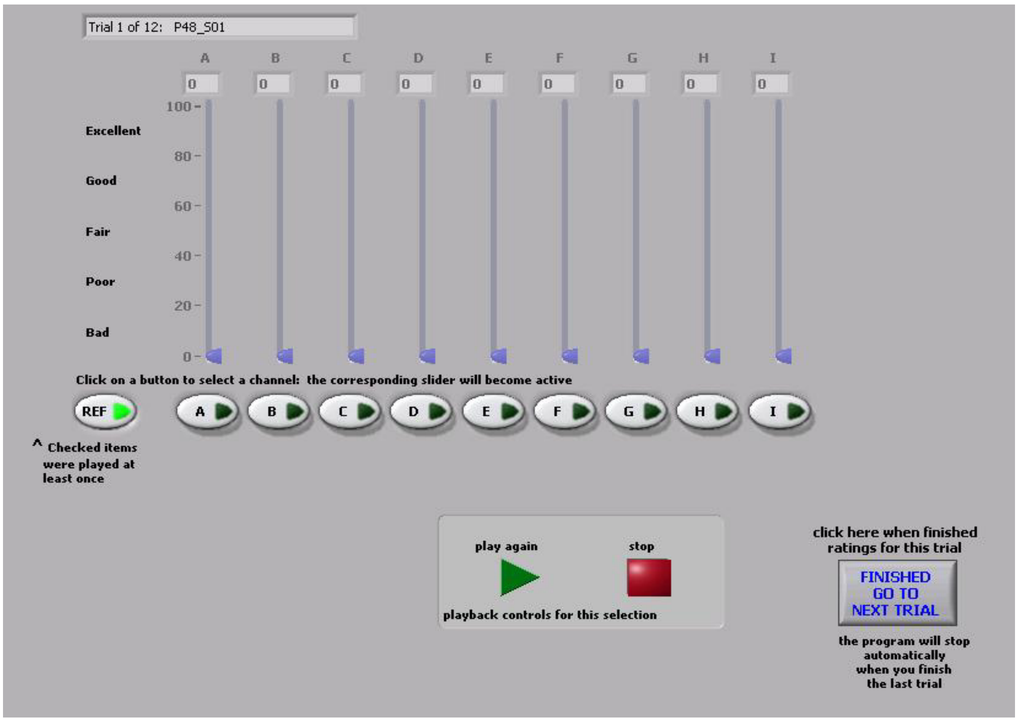


Figure 1: Example MUSHRA response interface

5 Performance Characterization for Audio Content

Results were extracted from three series of MUSHRA tests to characterize the 3GPP audio codecs for three different classes of audio content -- Music, Speech, and Mixed Music+Speech. Each of the MUSHRA tests described in the present document involved 12 test items, four items each for Music, Speech, and Mixed audio content.

5.1 Results from characterization phase 1

Figures 2 and 3 shows results from the experiments conducted in the CT-P1 test series. The Mean scores and 95 % Confidence Intervals shown in the figures are based on scores for two labs, 15 listeners, and four test-items per class of audio content (N = 2 × 15 × 4 = 120 votes).

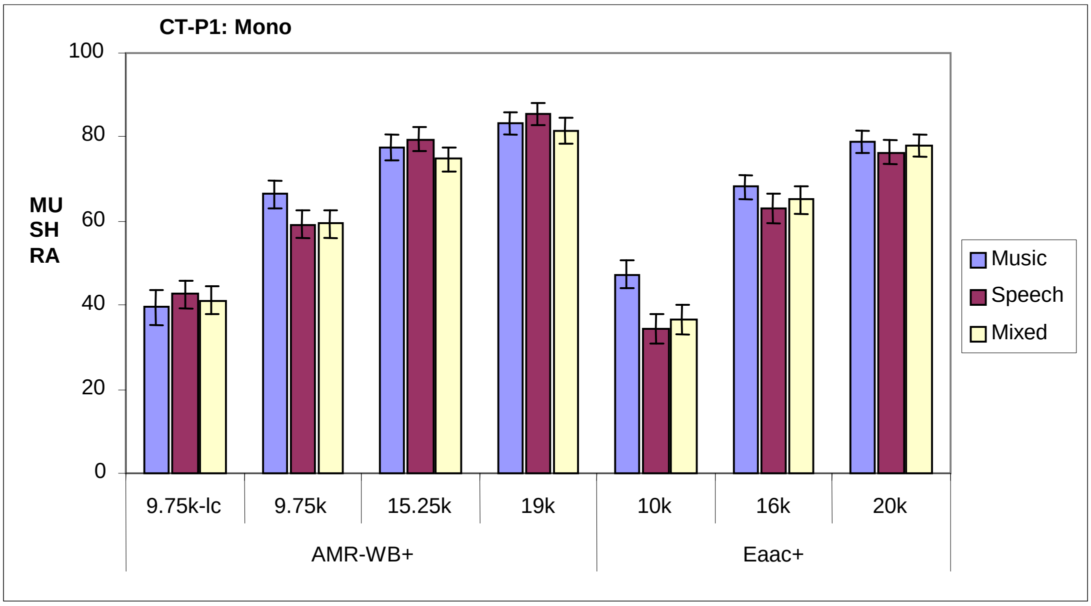


Figure 2: Audio Content for audio codecs across bit-rates (mono mode)

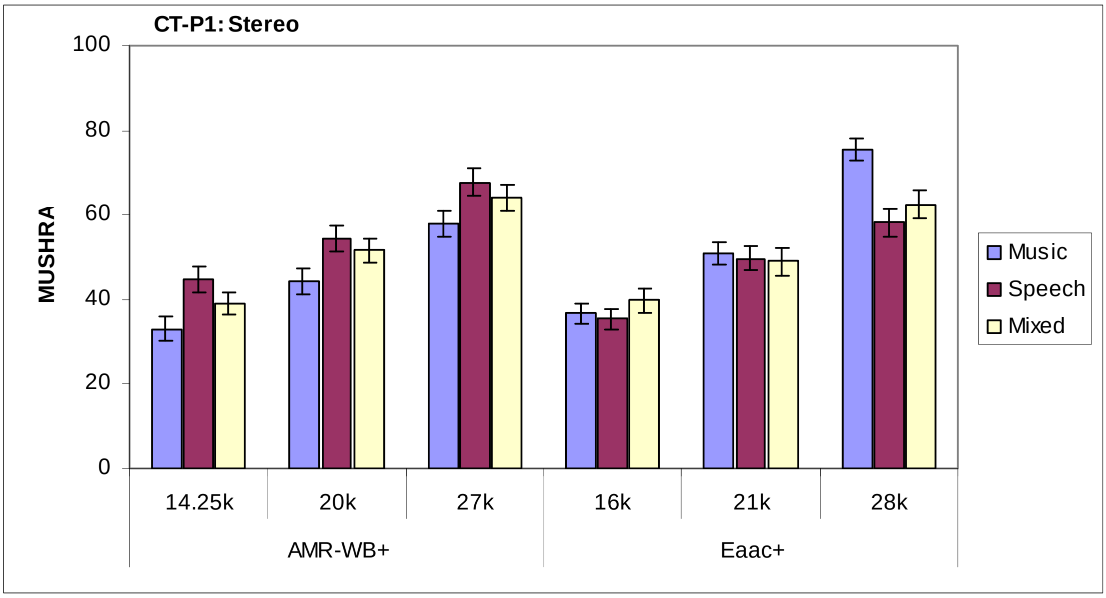


Figure 3: Audio content for audio codecs across bit-rates (stereo mode)

5.2 Results from Characterization phase 2

Results in this clause were derived in experiments containing packet losses. Bit-rates mentioned in figures 4 to 7 represent gross bit-rates including packetization overhead. Further information about how the packet loss conditions were defined, can be found in clause 8.

Figures 4 and 5 show results for audio content for the tests conducted in the CT-P2 test series for EGPRS under conditions of PLR. Figure 4 shows results for mono mode and figure 5 for stereo mode. The Mean scores and 95 % Confidence Intervals shown in the figure are based on scores for 15 listeners and four test-items per class of audio content ($N = 15 \times 4 = 60$ votes).

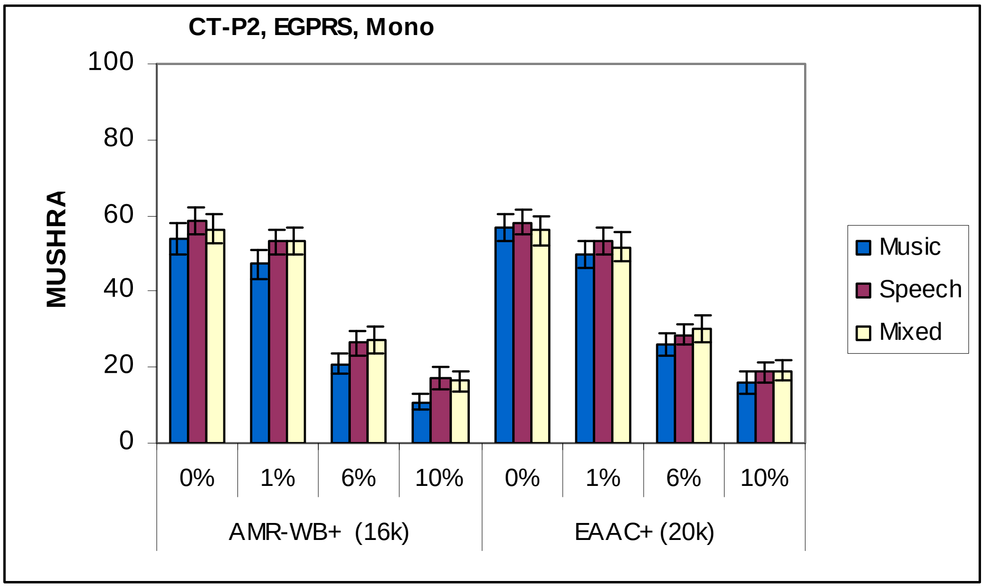


Figure 4: Audio content for audio codecs across PLR (EGPRS mono mode) (Bit-rates given are gross rates including packetization overhead)

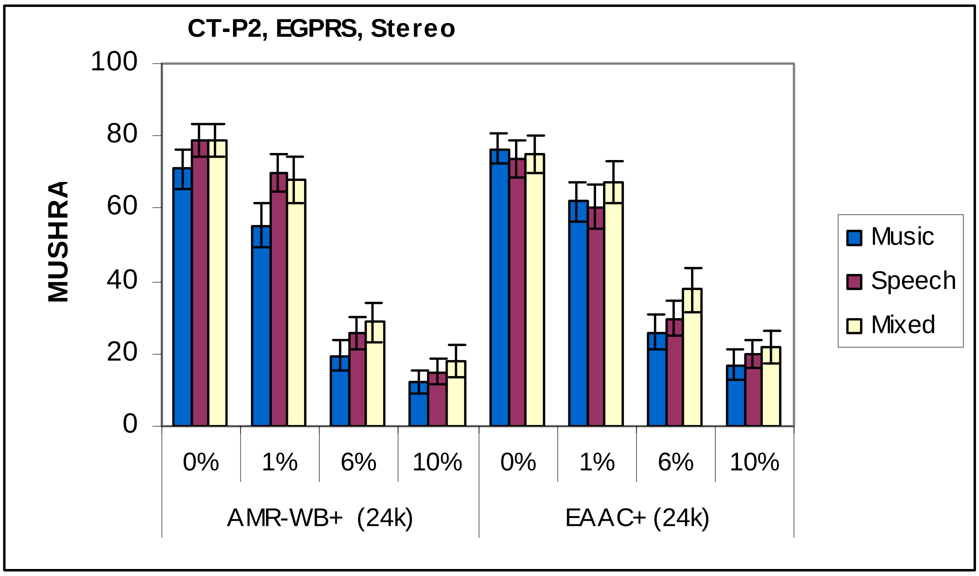


Figure 5: Audio content for audio codecs across PLR (EGPRS stereo mode) (Bit-rates given are gross rates including packetization overhead)

Figures 6 and 7 show results for audio content for the tests conducted in the CT-P2 test series for UTRAN, stereo mode under conditions of PLR. Figure 6 shows results for lower bit-rates (*AMR-WB+* at 20 kbps and *Eaac+* at 32 kbps) and figure 7 for higher bit-rate (both codecs at 40 kbps). The Mean scores and 95 % Confidence Intervals shown in the figure are based on scores for 15 listeners and four test-items per class of audio content (N = 15 × 4 = 60 votes).

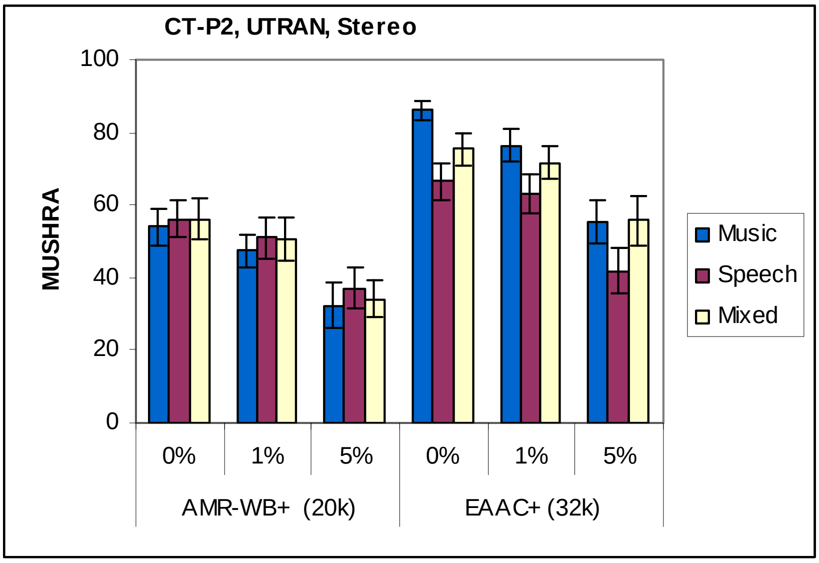


Figure 6: Audio content for audio codecs across PLR (UTRAN stereo mode) (Bit-rates given are gross rates including packetization overhead)

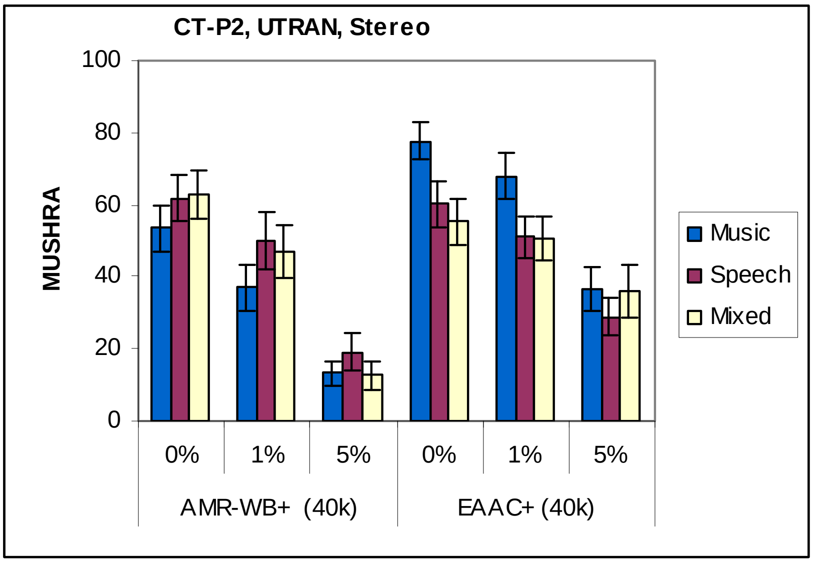


Figure 7: Audio content for audio codecs across PLR (UTRAN stereo mode) (Bit-rates given are gross rates including packetization overhead)

5.3 Results from Selection Test

Figure 8 summarizes results for a subset of four MUSHRA experiments conducted in the selection series of tests (ST-LR), each experiment involving the two audio codecs for the PSS application. Results are shown for each of the two audio codecs in each of four MUSHRA tests for the three classes of Audio Content. The results shown in figure 8 are based on votes from 15 subjects for four test-items per class of audio content in each of four listening labs ($N = 15 \times 4 \times 4 = 240$ votes). In general, these results show that *AMR-WB+* scores better for Speech content, relatively worse for Music content, with Mixed content between those values. On the other hand, *Eaac+* scores better for Music content, worse for Speech content, with Mixed content between those values.

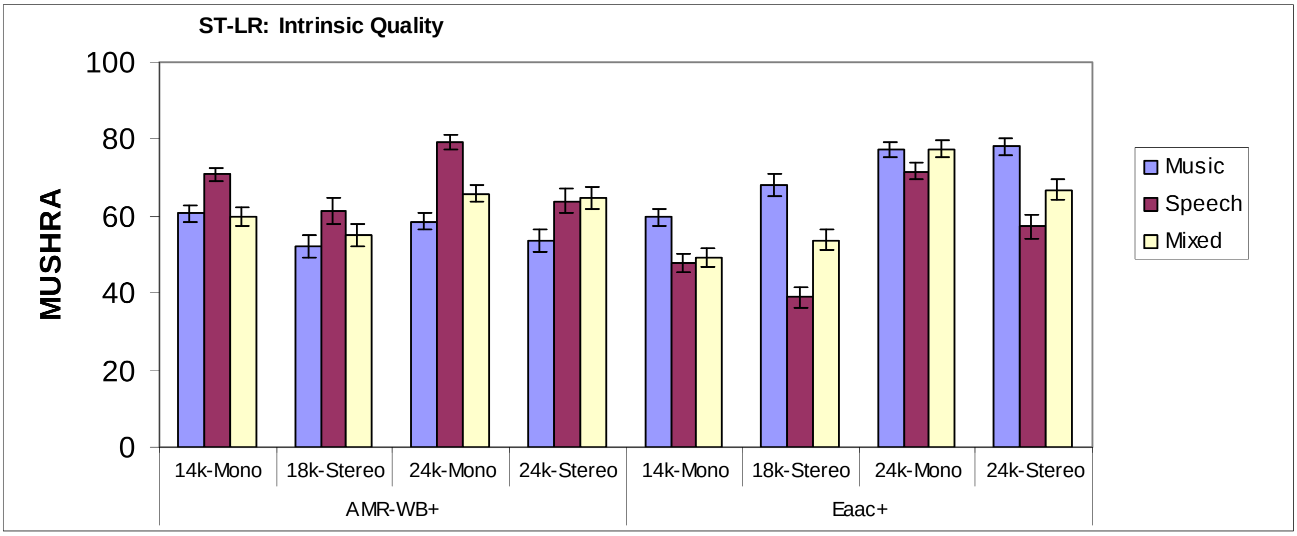


Figure 8: Results for four ST-LR tests (intrinsic quality) by class of audio content

Figure 9 summarizes results for a second subset of four ST-LR, two experiments for the MMS application without frame errors and two for the PSS application with 3 % frame errors. For the 14 kbps MMS case, the input and output sampling rate was restricted to 16 kHz. Results are shown for each of the two audio codecs in each of four MUSHRA tests for the three classes of Audio Content.

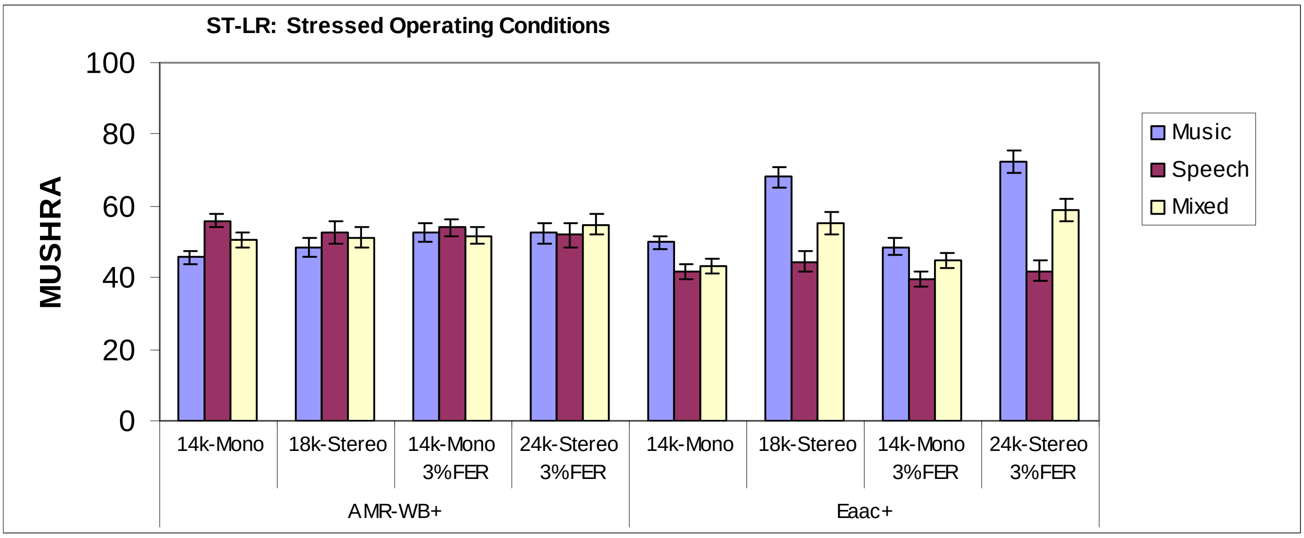


Figure 9: Results for four ST-LR tests (stressed operating conditions) by class of audio content

It can be observed from figures 4 to 9 that Extended AMR-WB tends to perform relatively better for speech than for music, while Enhanced aacPlus tends to perform relatively better for music than for speech.

The duration of a transmission frame in AMR-WB+ depends on the Internal Sampling Frequency. In the codec used in the selection phase, 20 ms transmission frames were used. Thus, frame errors correspond to 20 ms frames being lost. A transmission frame in Eaac+ corresponds to 2048 samples. Thus, frame errors at 48 kHz sampling rate correspond to 23.4 ms portions of signal being lost. Note that these frame error rates are different from packet loss rates since a packet may contain one or more frames, depending on the packetization scenario.

6 Performance characterization over bit-rate

Results from two experiments conducted in the CT-P1 series of tests contribute to the performance characterization of coding bit-rate for the two selected 3GPP audio codecs.

Figure 10 shows MUSHRA results from the CT-P1 test series for mono mode. These results are based on the MUSHRA ratings of 15 subjects, 12 test items, and two listening labs (N = 15 × 12 × 2 = 360). The experiment was designed to evaluate the performance of two audio codecs across bit-rates in a mono application. The figure shows average MUSHRA scores for *AMR-WB+* at coding rates of 9.75 kbps, 15.2 kbps and 19 kbps (which were suitable configurations of AMR-WB+ for the target bit-rates of 10 kbps, 16 kbps and 20kbps) plus a low-complexity mode at 9.75 kbps and for *Eaac+* at coding rates of 10 kbps, 16 kbps and 20 kbps. Also shown are the Mean MUSHRA scores for the three Anchor conditions - Low Pass 3.5 kHz, Low Pass 7 kHz, and the Hidden Reference. In addition to the Mean scores, the figure shows error brackets for each condition indicating the 95 % Confidence Intervals. The Mean scores and 95 % Confidence Intervals are based on 360 votes as indicated above. The results in figure 10 confirm that, for both audio codecs, MUSHRA performance increases with increases in bit-rate. Furthermore, at 9.75 kbps, the low complexity version of *AMR-WB+* (9.75-lc in the figure) scored significantly lower than the standard version. Except for the low complexity version of *AMR-WB+*, figure10 shows that *AMR-WB+* achieves a better performance that *Eaac+* at all tested bit-rates. Moreover, both codecs show a consistent increase in quality with growing bit-rate.

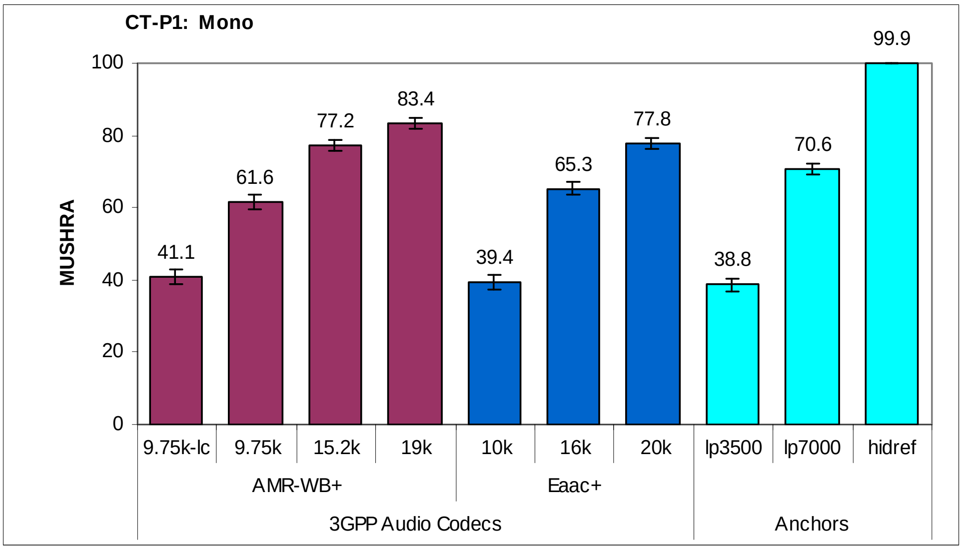


Figure 10: MUSHRA results for *AMR-WB+* and *Eaac+* across bit-rates (mono mode)

Figure 11 shows MUSHRA results from the CT-P1 test series for stereo applications. These results are similar to those in figure 10 except the audio codecs and anchor conditions were tested in stereo mode. The figure shows Mean MUSHRA scores with 95 % Confidence Intervals for *AMR-WB+* at coding rates of 14.25 kbps, 20 kbps and 27 kbps (which were suitable configurations of AMR-WB+ for the target bit-rates of 16 kbps, 21 kbps and 28 kbps) and for *Eaac+* at coding rates of 16 kbps, 21 kbps and 28 kbps. Also shown are the scores for three stereo Anchor conditions - Low Pass 3.5 kHz, Low Pass 7 kHz, and the Hidden Reference. As in figure 10, the Mean scores and 95 % confidence Intervals are based on 360 votes. Figure 11 shows that both codecs perform relatively similar at all tested bit-rates. Both codecs show a consistent quality increase with growing bit rate.

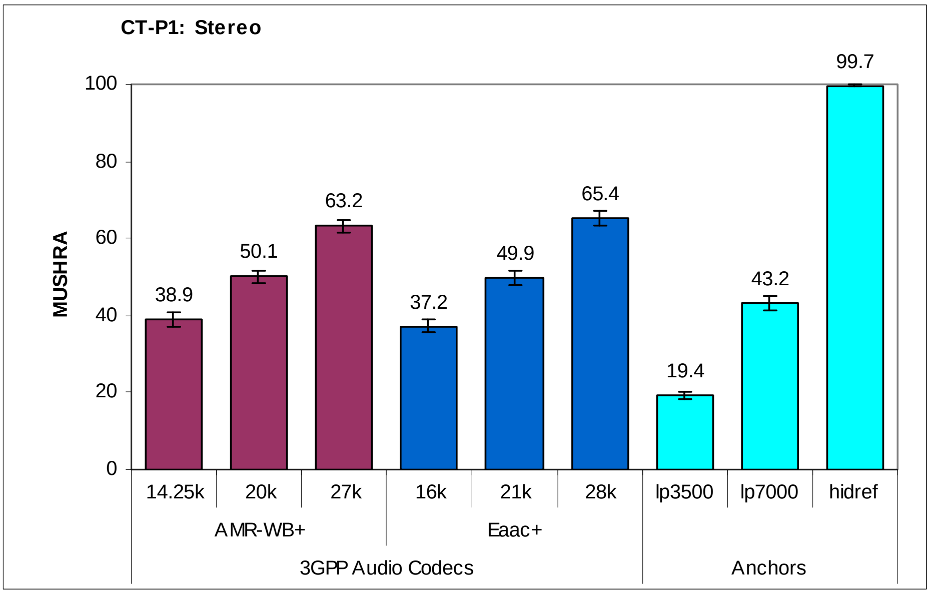


Figure 11: MUSHRA results for *AMR-WB+* and *Eaac+* across bit-rates (stereo mode)

7 Performance based on Selection Tests

The MUSHRA results presented in this clause for the application/service specific experiments were conducted in the Selection Test, ST-LR and ST-HR test series. Moreover, the Selection Test experiments involved earlier versions of the two audio codecs, *AMR-WB+* and *Eaac+*.

7.1 Intrinsic quality comparison in lower bit-rate range (up to 24 kbps)

Four MUSHRA experiments were conducted in the ST-LR series of tests which characterize the intrinsic performance of the two 3GPP Release 6 audio codecs. Figures 12 to 15 show the results of these experiments. Each experiment involved the two 3GPP Release 6 audio codecs plus two reference codecs, *AMR-WB* and *AAC*, operating at a common bit-rate. Each experiment was conducted in four listening labs. The results shown in figures 12 to 15 are based on the MUSHRA ratings of 15 subjects, 12 test items per experiment (2 different test sets used in 2 labs each), and four listening labs ($N = 15 \times 12 \times 4 = 720$). Two experiments were conducted to evaluate the performance of the audio codecs for PSS applications in mono mode, two experiments in stereo mode, and two experiments with 3 % FER.

Figures 12 and 13 show the results for the PSS application in mono mode at coding rates of 14 kbps and 24 kbps, respectively. For the PSS application in mono mode, *AMR-WB+* scores higher than *Eaac+* at 14 kbps (62.6 vs. 51.5) but lower at 24 kbps (67.4 vs. 75.8).

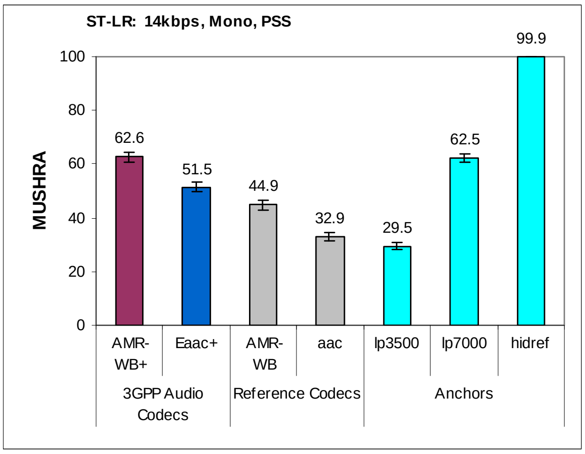


Figure 12: Results at 14 kpbs for PSS (mono)

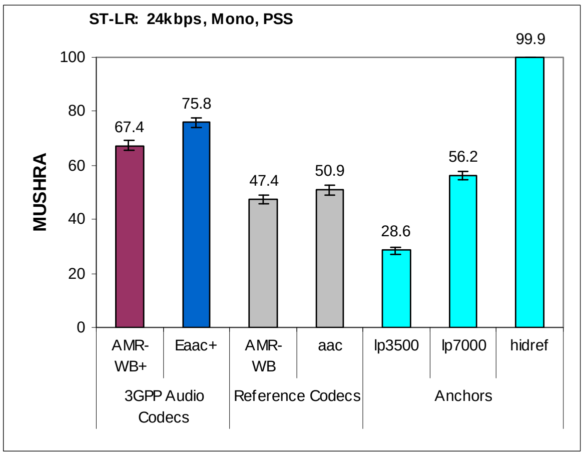


Figure 13: Results at 24 kbps for PSS (mono)

Figures 14 and 15 show the results for the PSS application in stereo mode at coding rates of 18 kbps and 24 kbps, respectively. For the PSS application in stereo mode, *AMR-WB+* scores higher than *Eaac+* at 18 kbps (55.6 vs. 53.3) but lower at 24 kbps (61.3 vs. 67.1).

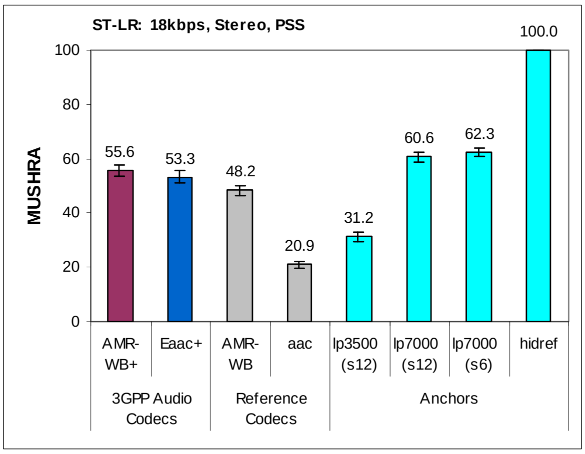
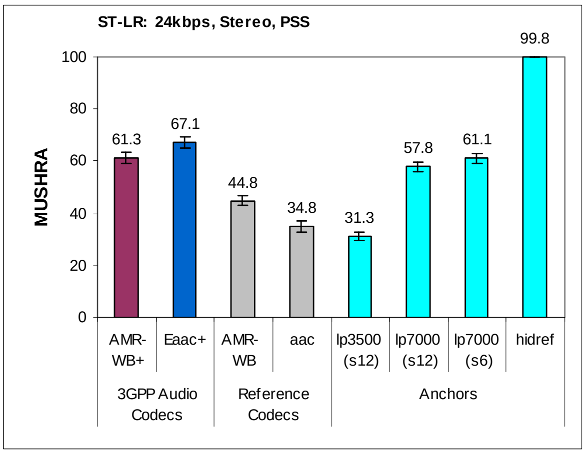


Figure 14: Results at 18 kpbs for PSS (stereo)



NOTE: For this stereo experiments there are three anchor conditions:
lp3500 (s12): original signal low-pass filtered at 3.5 kHz, stereo image reduced by 12 dB;
lp7000 (s12): original signal low-pass filtered at 7.0 kHz, stereo image reduced by 12 dB;
lp7000 (s6): original signal low-pass filtered at 7.0 kHz, stereo image reduced by 6 dB.

Figure 15: Results at 24 kbps for PSS (stereo)

7.2 Quality comparison under stressed operating conditions at lower bit-rates (up to 24 kbps)

Four MUSHRA experiments were conducted in the ST-LR series of tests that characterize the performance of the two 3GPP audio codecs under stressed operating conditions. Each experiment involved the two 3GPP Release 6 audio codecs plus two reference codecs, *AMR-WB* and *AAC*, operating at a common bit-rate for the MMS application and each experiment was conducted in four listening labs. The results shown are based on the MUSHRA ratings of 15 subjects, 12 test items per experiment (2 different test sets used in 2 labs each), and four listening labs ($N = 15 \times 12 \times 4 = 720$).

Figures 16 and 17 show the results for the PSS application for 3 % FER - mono mode at 14 kbps and stereo mode at 24 kbps, respectively. In the context of this test, the error conditions were simulated by applying a random frame erasure rate (FER) of 3 % to both codecs. Both codecs out-perform the reference codecs by a wide margin when operated under error conditions and show a pattern of behaviour, relative to each other, which is similar to that under unimpaired channel conditions.

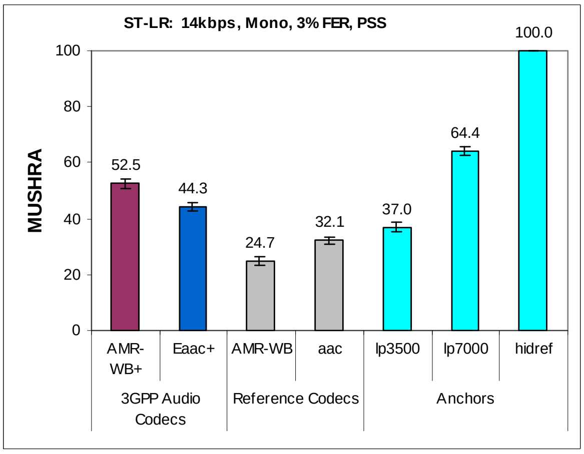


Figure 16: Results at 14 kbps mono, 3 % FER

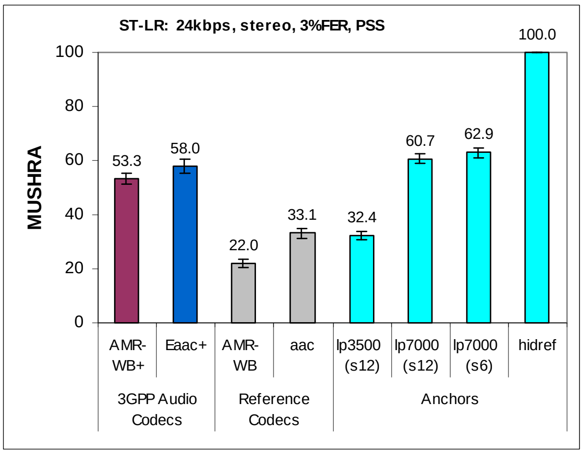


Figure 17: Results at 24 kbps stereo, 3 % FER

Figure 18 shows the results for the terminal-generated MMS application, mono mode at 14 kbps and figure 19 shows the results for stereo at 18kbps. Both experiments used the low complexity encoder.

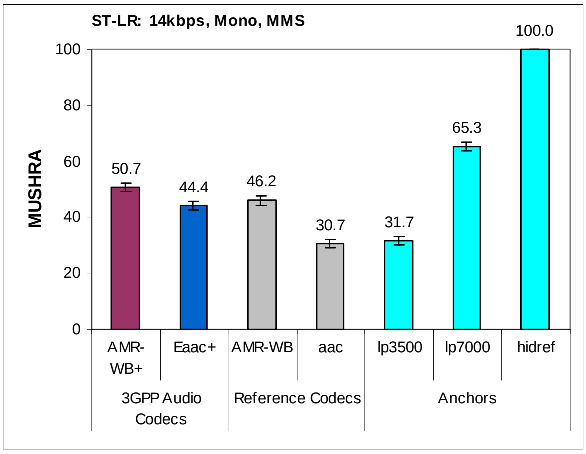


Figure 18: Results at 14 kbps for MMS (mono)

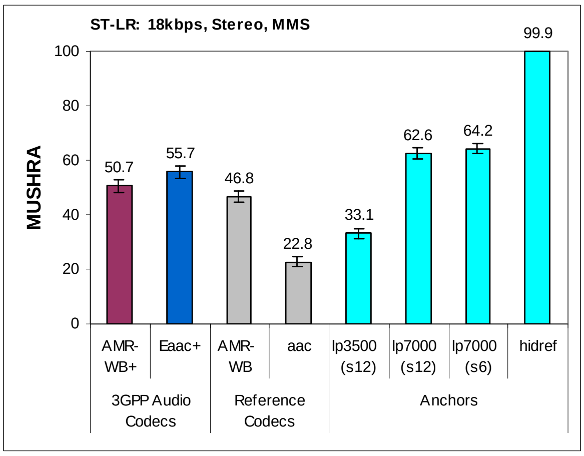


Figure 19: Results at 18 kbps for MMS (stereo)

For the MMS application in mono mode at 14 kbps, *AMR-WB+* scores higher than *Eaac+* (50.7 vs. 44.4). However, for the MMS application in stereo mode at 24 kbps, *Eaac+* scores higher than *AMR-WB+* (50.7 vs. 55.7). It should be noted that the MMS test at 14 kbps mono was conducted using samples that were sample-rate limited to 16 kHz.

7.3 Performance evaluation in higher bit-rate range (up to 48 kbps)

Three MUSHRA experiments were conducted in the ST-HR series of tests designed to characterize the intrinsic performance of one of the two 3GPP Release 6 audio codecs, Eaac+, at higher bit-rates. Figures 20 to 22 show the results of these experiments. Each experiment involved the *Eaac+* audio codec plus *AAC* as a reference codec both codecs operating at a common bit-rate and each experiment was conducted in two listening labs. The results shown are based on the MUSHRA ratings of 15 subjects, 12 test items, and two listening labs (N = 15 × 12 × 2 = 360). Two test evaluated the intrinsic performance of *Eaac+*, the third test evaluated error performance, again using random audio frame losses. It can be concluded that *Eaac+* out-performs the reference codec by a wide margin under both unimpaired and error conditions.

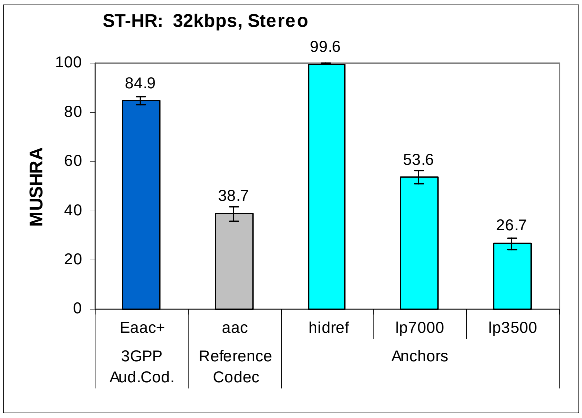


Figure 20: MUSHRA results at 32 kbps (stereo)

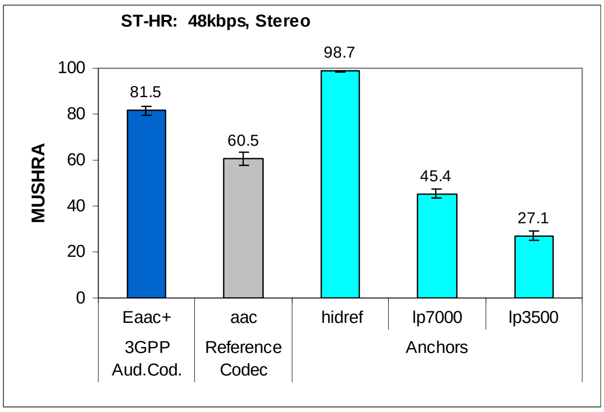


Figure 21: MUSHRA results at 48 kbps (stereo)

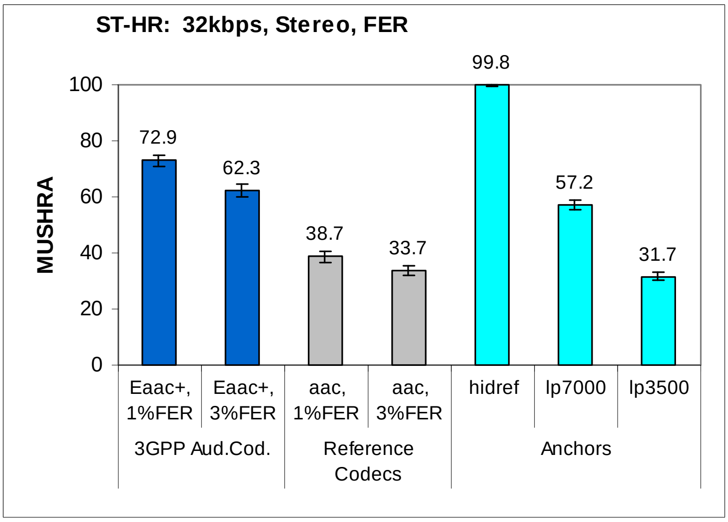


Figure 22: MUSHRA results at 32 kbps with 3 % errors (stereo)

8 Performance characterization for error conditions

This clause comprises the test results from codec characterization assessing the codec performance for various packet loss rates. This clause is relevant for the design of application layer FEC and the definition of target BLER for PSS and MBMS services. The chosen test cases do not imply that the codecs should be operated under all the tested packet loss rates.

The bit-rates mentioned in this clause are **gross** rates, which include packetization overhead. The performance relative to bit-rate and PLR depends on the chosen packetization and interleaving configuration. All packet loss rates mentioned in this context refer to RLC-PDU packet loss rates. Details on packetization, interleaving configurations and source bit-rates can be found in attachments S4-050453 for *AMR-WB+*, S4-050544 for *Eaac+* and S4-060099 for both codecs (in "TR 26.936 Annex B2 Additional Documents.zip").

Results from the CT-P2 series of subjective tests contribute to the performance characterization of the two 3GPP audio codecs under error conditions expressed in terms of percent Packet Loss Rate (PLR). In all, four MUSHRA experiments were conducted in the CT-P2 series -- two tests for Enhanced General Packet Radio Service (EGPRS), one in mono mode and one in stereo mode; two tests for under UMTS Terrestrial Radio Access Network (UTRAN), one in stereo mode at relatively lower bit-rates and one in stereo mode at relatively higher bit-rates.

Figure 23 shows MUSHRA results for the two 3GPP audio codecs across PLR under EGPRS, mono mode. Results are shown for *AMR-WB+* operating at 16 kbps and *Eaac+* operating at 20 kbps for PLR of 0 %, 1 %, 6 % and 10 %. Figure 24 shows results for EGPRS, stereo mode with both codecs operating at 24 k.

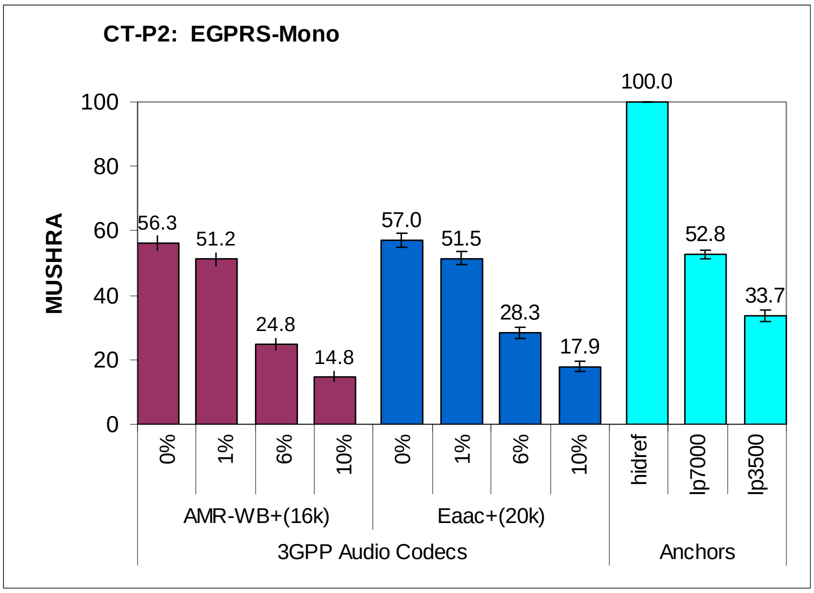


Figure 23: Results for EGPRS at four levels of PLR (mono mode) (Bit-rates given are gross rates including packetization overhead)

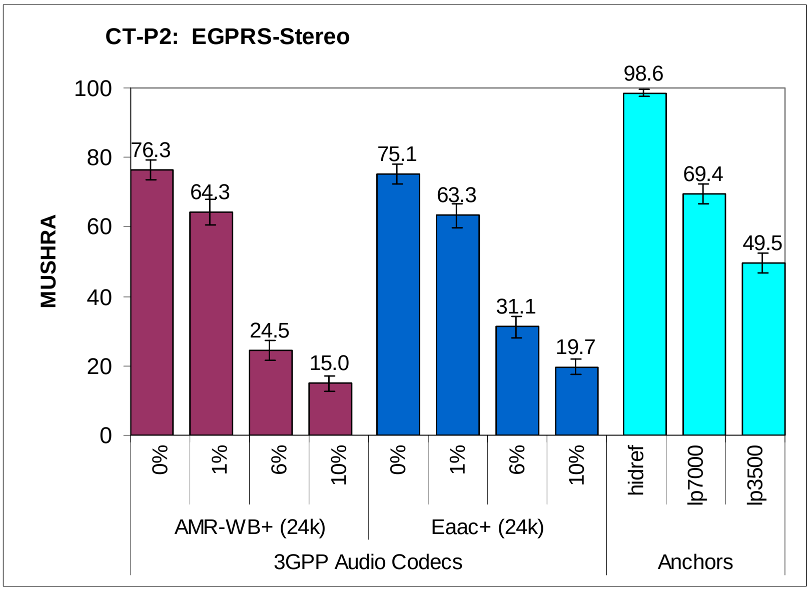


Figure 24: Results for audio codecs for EGPRS at four levels of PLR (stereo mode) (Bit-rates given are gross rates including packetization overhead)

Figure 25 shows MUSHRA results for the two 3GPP audio codecs across PLR under UMTS Terrestrial Radio Access Network (UTRAN), mono mode with the codecs operating at relatively lower bit-rates. Results are shown for

AMR-WB+ operating at 20 kbps and *Eaac+* operating at 32 kbps for PLR of 0 %, 1 % and 5%. Figure 26 shows results for UTRAN, stereo mode with both codecs operating at 40 k.

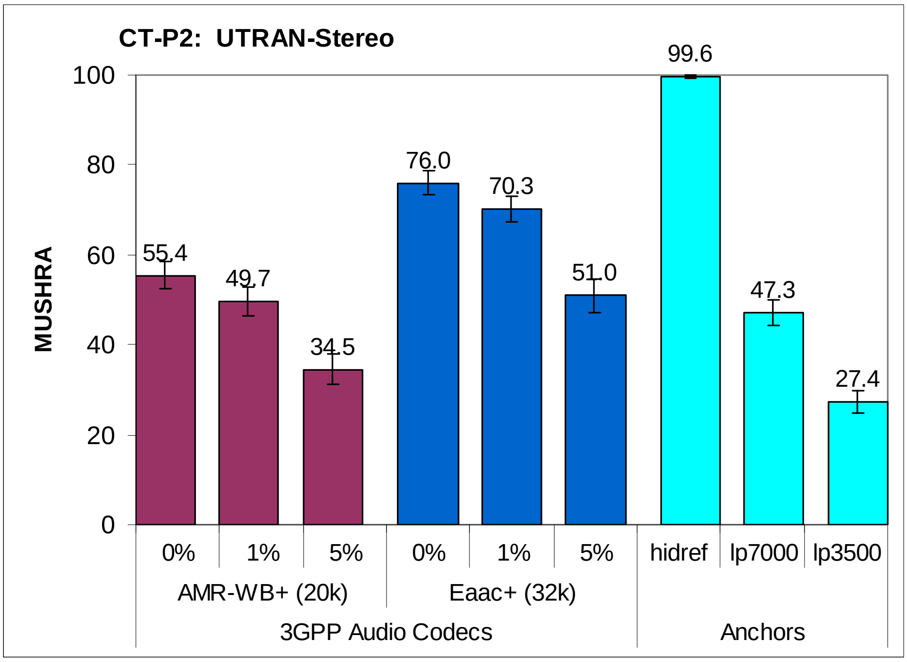


Figure 25: Results for UTRAN, lower-rate at three levels of PLR (stereo mode)
***AMR-WB+* was tested at 20 kbps, *Eaac+* at 32kbps**
(Bit-rates given are gross rates including packetization overhead)

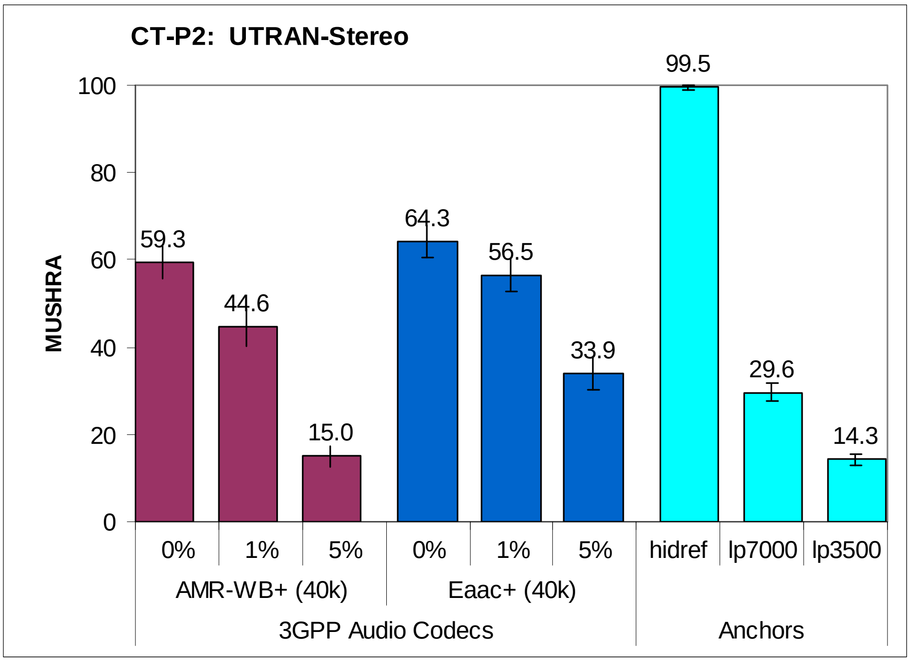


Figure 26: Results for UTRAN, Higher-rate at Three Levels of PLR (stereo mode)
(Bit-rates given are gross rates including packetization overhead)

In general, the results in figures 23 and 24 show that, for EGPRS, MUSHRA performance decreases with increase in PLR. Moreover, the performance profiles across PLR for the two codecs are similar for both the mono and stereo tests, and exhibit slightly higher quality losses for AMR-WB+ at higher PLRs. Figures 25 and 26 show a similar trend for UTRAN in stereo mode - performance decreases with increase in PLR. However, for the 40 kbps condition shown in figure 26 the quality loss with increasing error rate for AMR-WB+ is significantly higher than in the lower bit-rate conditions tested. This might be related to the use of a different RTP packetizer configuration without interleaving for AMR-WB+ in this test case as it can be seen in document S4-050453 (in attachment "TR 26.936 Annex B2 Additional Documents.zip").

A similar effect cannot be observed for Enhanced aacPlus. Again, as shown for the EGPRS experiments (figures 23 and 24), the performance profiles across PLR for the two codecs and for the two bit-rates are similar in the case of UTRAN.

9 Results of verification tests

This clause comprises results from codec verification tasks which checked and assessed various codec aspects such as complexity, verification of the fixed point code, frequency response, delay, codec performance with 3D audio signals, rate switching performance, and content dependency.

9.1 Complexity verification

The clause provides the results of the complexity verification of the fixed-point (FIP) code. The source code used for this verification task is that version which was obtained after completing the verification of the format of the C-code and correct implementation of complexity counters in agreement with verification entity STMicroelectronics.

9.1.1 AMR-WB+ complexity

The complexity has been evaluated using the "allcat.wav" stereo file, which covers the complete audio material used in the audio codec low-rate selection tests. For the mono test cases the "allcat.wav" file was externally down-mixed to mono.

Tables 3 and 4 show the evaluated conditions and corresponding complexity results.

Table 3: Encoder complexity figures

Condition	Command line options	Complexity (wMOPS) (Average/Worst Case)
14 kbps, mono	-rate 14 -mono	53.217 / 54.009
14 kbps, mono, low complexity	-rate 14 -mono -lc	29.004 / 33.331
24 kbps, mono	-rate 24 -mono	70.998 / 72.055
24 kbps, mono, low complexity	-rate 24 -mono -lc	38.074 / 45.815
18 kbps, stereo	-rate 18	69.279 / 70.479
18 kbps, stereo, low complexity	-rate 18 -lc	45.170 / 50.438
24 kbps, stereo	-rate 24	80.778 / 82.183
24 kbps, stereo, low complexity	-rate 24 -lc	51.799 / 59.311

Table 4: Decoder complexity figures

Condition	Command line options	Complexity (wMOPS) (Average/Worst Case)
14 kbps, mono	-mono	8.415 / 8.792
24 kbps, mono	-mono	10.382 / 10.981
36 kbps, mono	-mono	12.463 / 13.184
18 kbps, stereo	None	15.996 / 16.603
24 kbps, stereo	None	17.654 / 18.303
32 kbps, stereo	None	20.288 / 21.081
48 kbps, stereo	None	22.960 / 23.927

9.1.2 Eaac+ Complexity

The complexity numbers for the Enhanced aacPlus audio codec can be found in the following tables, the numbers have been derived using the "allcat.wav" item, which holds all the material from the selection test concatenated in one single item. For every test case the average and worst frame weighted MOPS figure has been derived. The worst case wMOPS figure over all test cases has been marked in **blue**.

The fixed-point C-code contains a pre-compiler directive named "ACCOUNT_ETSIOP_OVERHEAD_ SPLITWORD32". If this pre-compiler directive is set during compilation, the complexity figures will be as given in table 5. If this pre-compiler directive is not set during compilation, the complexity figures will be as given in table 6. Regarding the details of the corresponding difference in the fixed-point C-code, please see the comments in the SplitWord32() function in intrinsics.c.

Table 5: Weighted MOPS figures with ACCOUNT_ETSIOP_OVERHEAD_SPLITWORD32 set

	Test Case	Mono Encoder	Stereo Encoder	Decoder	Decoder, Mono only
wMOPS [average / worst frame]	14 m	26.51 / 28.87	26.50 / 31.61	19.15 / 21.20	14.73 / 16.80
	18 s	---	61.38 / 65.25	35.18 / 38.04	15.14 / 17.39
	24 m	29.51 / 34.28	29.51 / 34.26	20.98 / 23.84	15.93 / 18.74
	24 s	---	63.47 / 68.17	37.35 / 40.98	15.93 / 18.72
	32 s	---	64.61 / 71.02	**38.39 / 42.28**	16.47 / 19.60
	48 s	---	**64.17 / 77.63**	32.65 / 38.46	21.96 / 26.83

Table 6: Weighted MOPS figures with ACCOUNT_ETSIOP_OVERHEAD_SPLITWORD32 not set

	Test Case	Mono Encoder	Stereo Encoder	Decoder	Decoder, Mono only
wMOPS [average / worst frame]	14 m	23.80 / 25.41	23.79 / 29.04	15.86 / 17.73	12.38 / 14.38
	18 s	---	51.14 / 53.92	29.35 / 32.07	12.77 / 14.95
	24 m	26.52 / 29.60	26.52 / 29.66	17.53 / 20.13	13.51 / 16.21
	24 s	---	53.08 / 56.06	31.17 / 34.64	13.57 / 16.22
	32 s	---	54.11 / 58.39	**32.08 / 35.75**	14.05 / 17.09
	48 s	---	**57.46 / 65.89**	27.88 / 33.43	18.53 / 23.22

9.2 Frequency response verification

The input signal is the concatenation of the files that have been used for the evaluation of floating point and fixed point decoder. This file comprises 4 music items, 4 speech items and 2 mixed content items. The total length of the file is 84 s.

9.2.1 Frequency response of AMR-WB+

The frequency response of the configurations of AMR-WB+ used during characterization and in selection are different. The characterization phase configuration operated with a signal sampling frequency of 48 kHz while the selection test configuration operated at only 24 kHz. The resulting difference in frequency response in shown in the following.

9.2.1.1 Characterization phase configuration

This clause reports the results in computing the frequency response of the extended AMR-WB codec in the settings used for the characterization phase.

Frequency response computation

The AMR-WB+ codec was tested at following bit rates: 13.6 kbps mono, 18 kbps stereo, 23.85 kbps stereo and 48 kbps stereo.

The input file is stereo.The frequency response has been evaluated by computing the spectra for input signal and processed signal.

Results

- 14 kbps, mono output:
    - The output is band limited to 12 kHz as shown on the following image. Output signal is slightly attenuated in frequency range 7 kHz to 12 kHz.

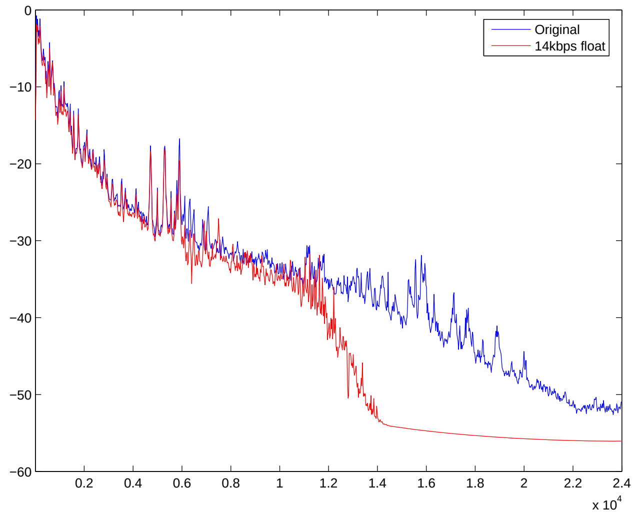


Figure 27

- 18 kbps, stereo output:
    - The output is band limited to 12 kHz as shown on the following images. Output signal is slightly attenuated in frequency range 7 kHz to 12 kHz.

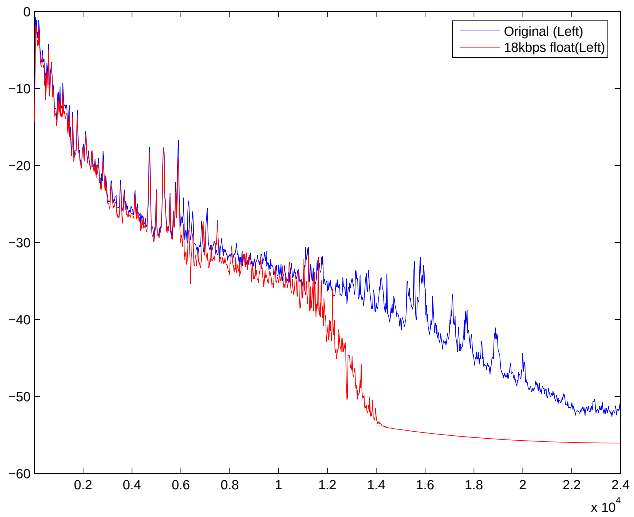


Figure 28

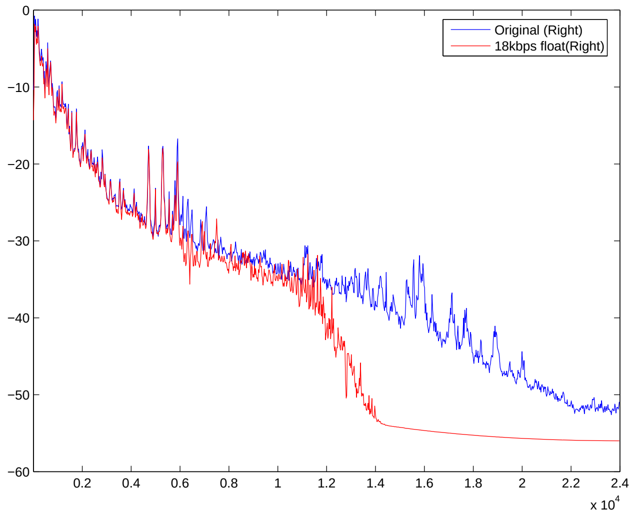


Figure 29

- 24 kbps, stereo output:
  - The output is band limited to 14 kHz as shown on the following images. Output signal is slightly attenuated in frequency range 6 kHz to 12 kHz.

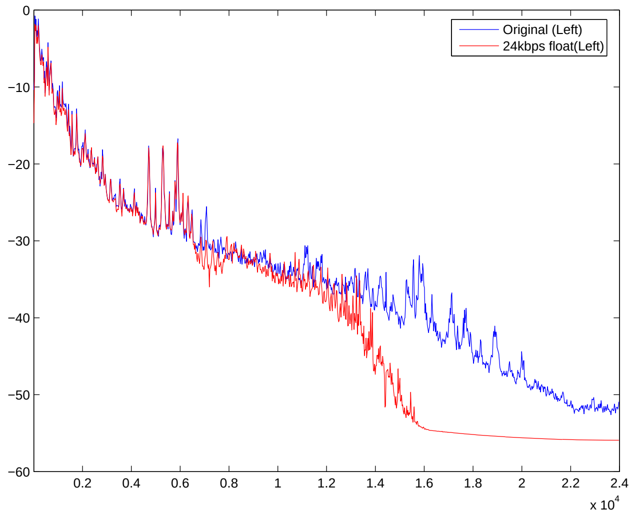


Figure 30

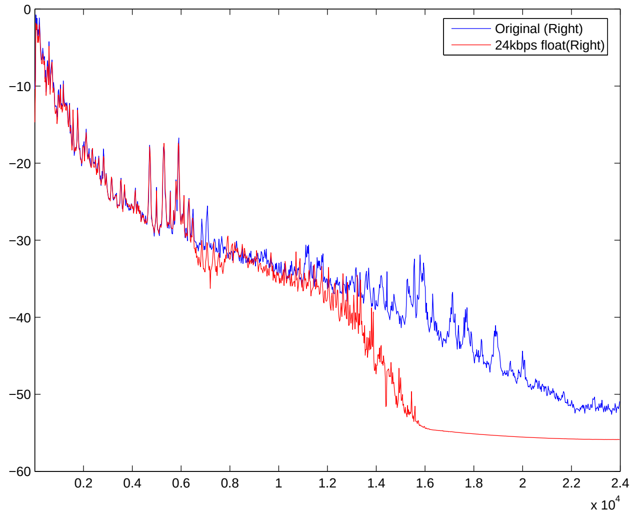


Figure 31

- 48 kbps, stereo output:
  - The output is band limited to 18 kHz as shown on the following images. Output signal is attenuated in frequency range 9 kHz to 11 kHz.

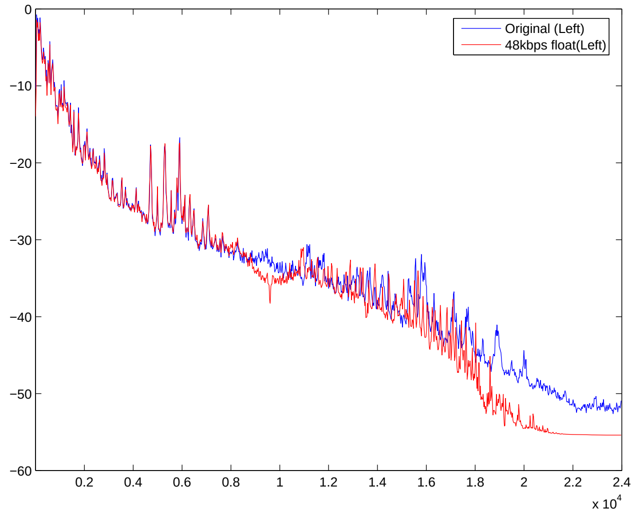


Figure 32

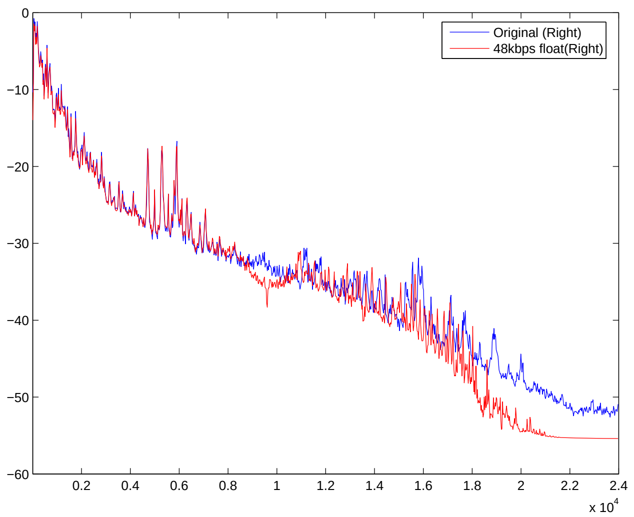


Figure 33

9.2.1.2 Selection phase configuration

This clause reports the results in computing the frequency response of the extended AMR-WB codec in the settings used for the selection phase for PSS application.

Frequency response computation

The AMR-WB+ codec was tested at following bit rates: 13.6 kbps mono, 18 kbps stereo, 24 kbps mono and 24 kbps stereo.

The input file is 24 kHz sampling frequency, mono or stereo depending on the mode index .The command line was the one used for selection using the mode index from 10 to 13.

The frequency response has been evaluated by computing the spectra for input signal and processed signal.

Results

- 14 kbps, mono output:
  - The output is band limited to 11 kHz as shown on the above image. Output signal is slightly attenuated in frequency range 6 kHz to 11 kHz.

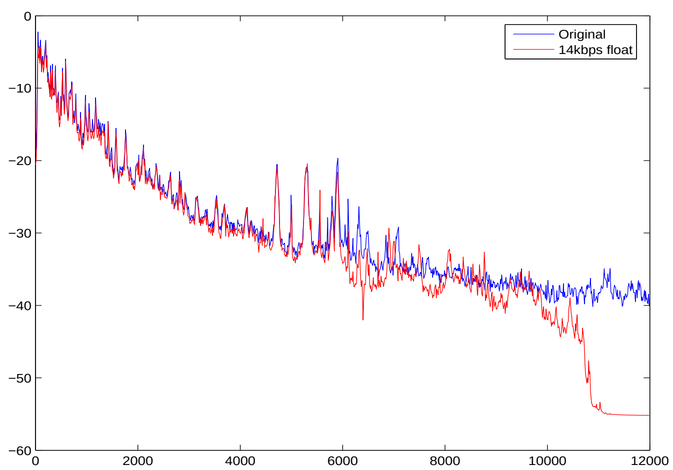


Figure 34

- 18 kbps, stereo output:

- The output is band limited to 11 kHz as shown on the images. Output signal is slightly attenuated in frequency range 4 kHz to 11 kHz.

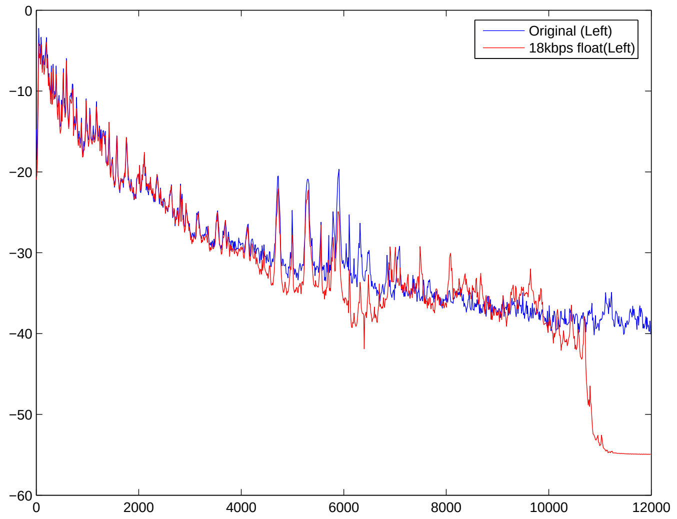


Figure 35

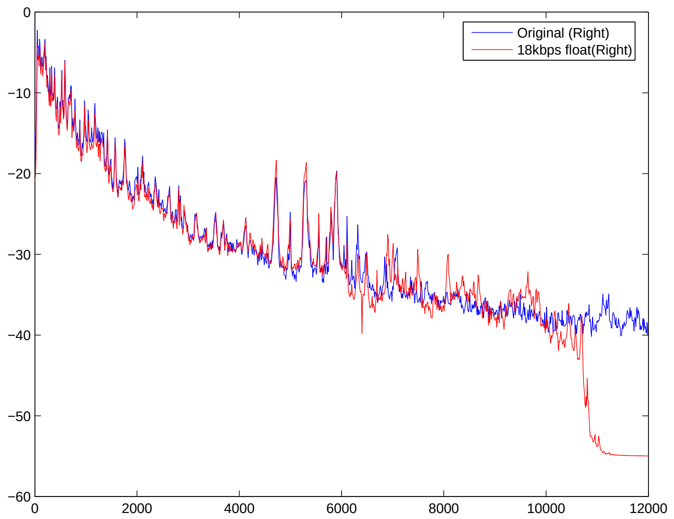


Figure 36

- 24 kbps, mono output:
    - The output is band limited to 14 kHz as shown on the following image. Output signal is slightly attenuated in frequency range 6 kHz to 12 kHz.

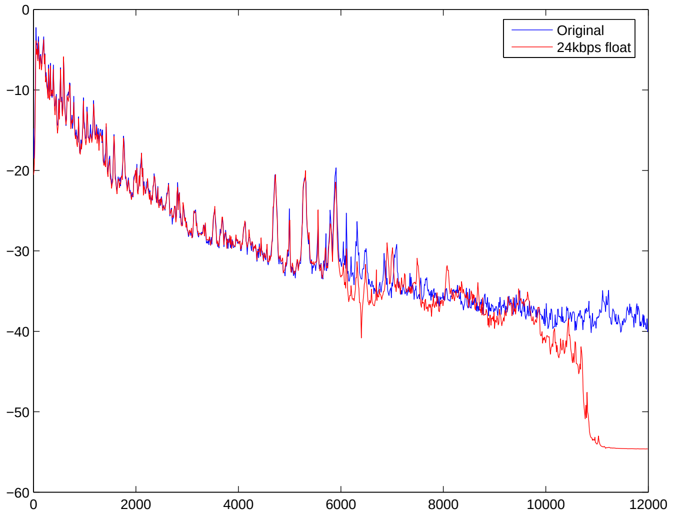


Figure 37

- 24 kbps, stereo output:
    - The output is band limited to 11 kHz as shown on the images. Output signal is attenuated in frequency range 6 kHz to 11 kHz.

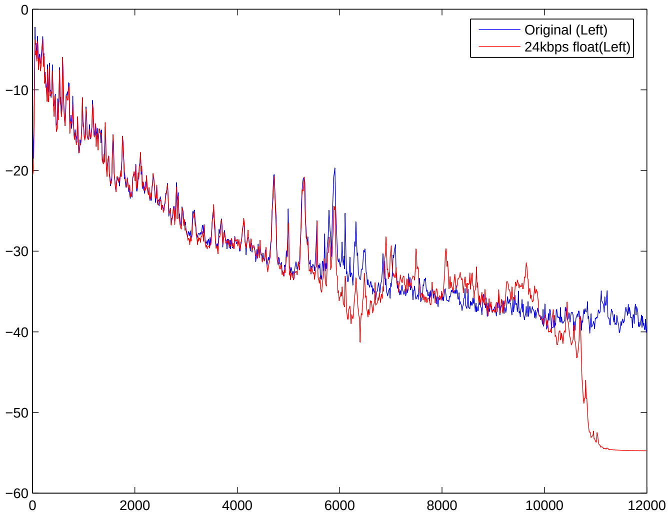


Figure 38

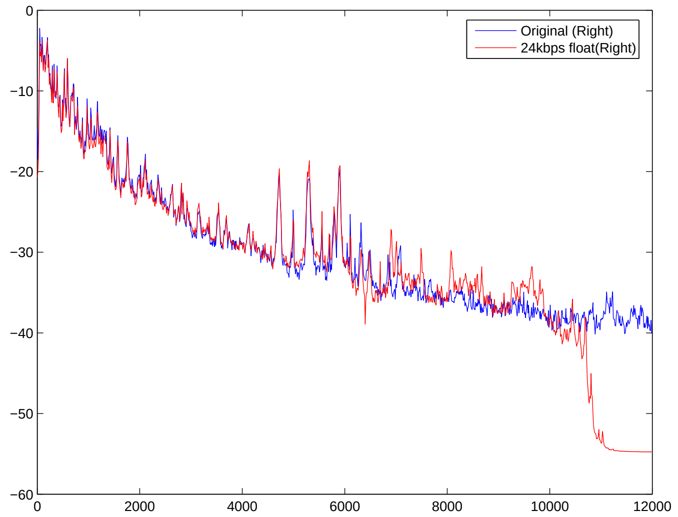


Figure 39

Conclusion

When used with the selection phase settings, the frequency response of AMR-WB+ is limited to 11 kHz.

9.2.2 Frequency response of Eaac+

This clause reports the results in computing the frequency response of the Eaac+ codec.

Frequency response computation

The Eaac+ codec was tested at following bit rates: 14 kbps mono, 18 kbps stereo, 24 kbps stereo and 48 kbps stereo.

The input file is stereo. The frequency response has been evaluated by computing the spectra for input signal and processed signal.

Results

- 14 kbps, mono output:
  - The output is band limited to 10 kHz as shown on the following images. Output signal is slightly attenuated in frequency range 7 kHz to 10 kHz.

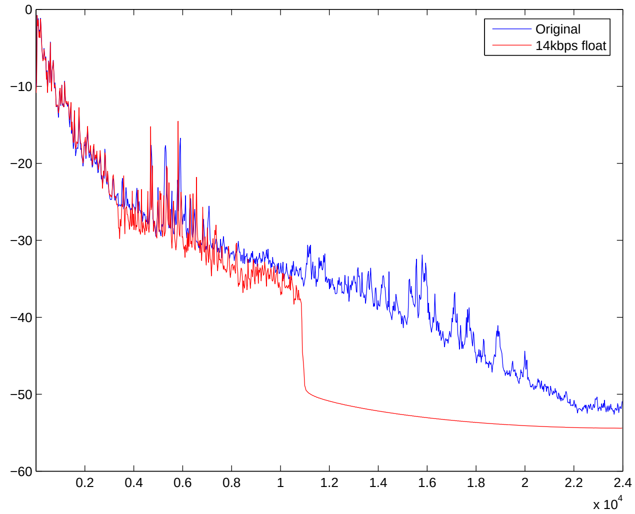


Figure 40

- 18kbps, stereo output:
  - The output is band limited to 12 kHz as shown on the following images. Output signal is attenuated in frequency range 6 kHz to 12 kHz.

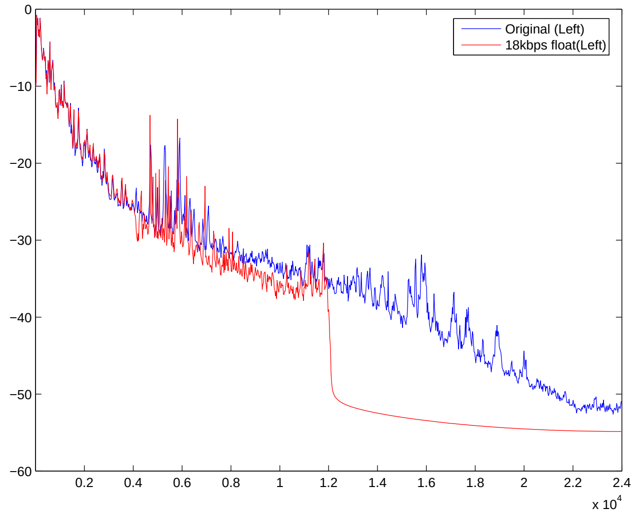


Figure 41

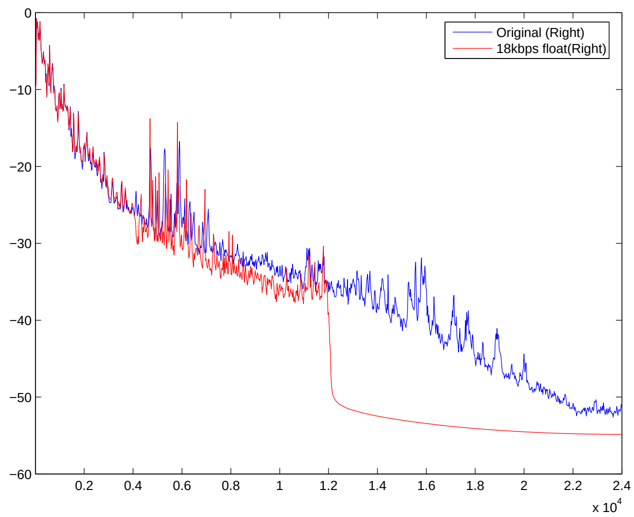


Figure 42

- 24 kbps, stereo output:
    - The output is band limited to 15 kHz as shown on the following images. Output signal is slightly attenuated in frequency range 6 kHz to 15 kHz.

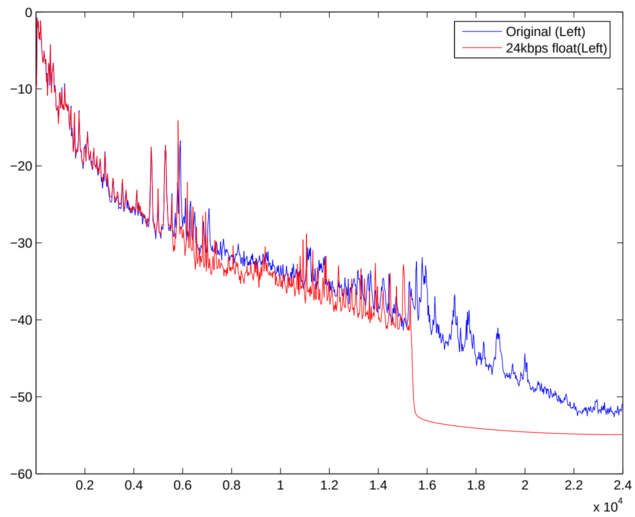


Figure 43

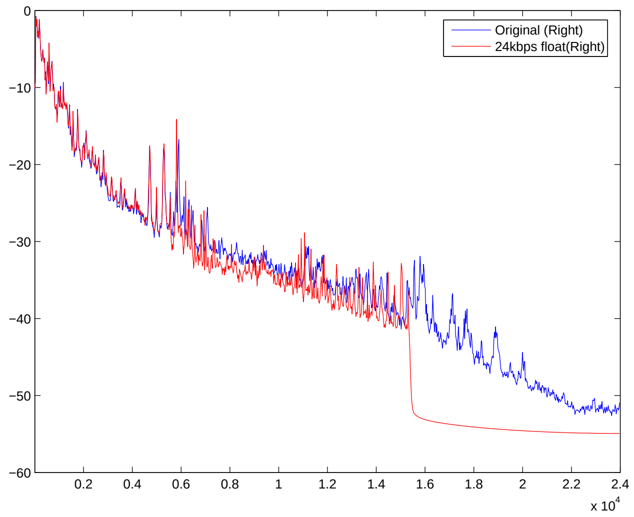


Figure 44

- 48 kbps, stereo output:
  - The output is band limited to 17 kHz as shown on the following images.

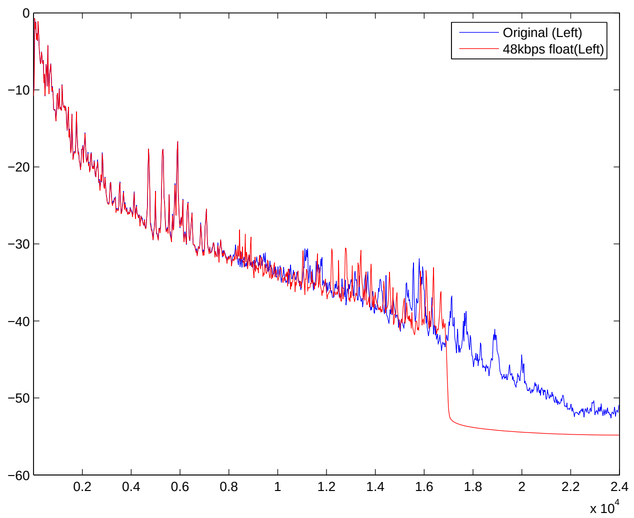


Figure 45

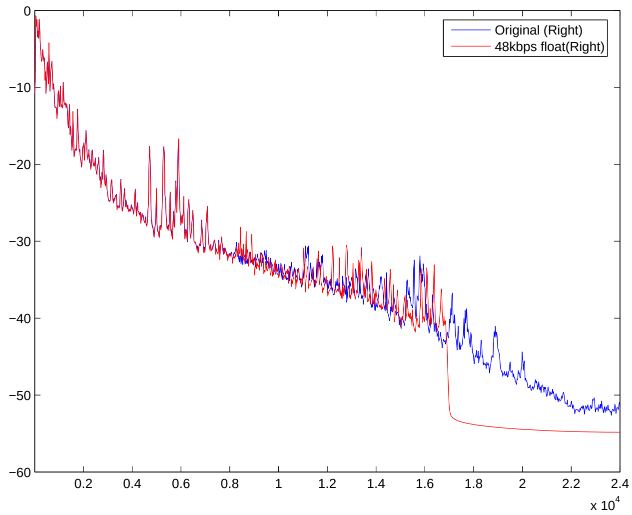


Figure 46

Conclusion

The frequency response of Enhanced aacPlus is also dependent on the bit rate. Attenuation can be observed for lowest bit rates; this attenuation is not constant over the frequency range.

9.3 Codec delay verification

9.3.1 AMR-WB+ codec delay

Introduction

This clause gives an analysis of the algorithmic delay of the extended AMR-WB codec. Delay figures are given for both mono and stereo operation, and at different Internal Sampling Frequencies (ISF).

Delay in mono operation

The delay analysis for mono operation is shown in table 7. The input signal sampling frequency of 48 kHz is used in the analysis. The table gives the encoder lookahead, decoder lookahead, and the superframe size. The algorithmic delay is measured as the superframe size plus the lookahead of both encoder and decoder. The breakdown of the delay is performed by listing the delay in samples for given operation, then the delay is converted from samples to ms based on the sampling frequency corresponding to that operation (in column Fs).

Table 7: Delay in mono operation for typical ISF of 25.6 kHz and for maximum ISF of 38.4 kHz, and minimum ISF of 12.8 kHz

	Typical ISF = 25.6 kHz			Min ISF = 12.8 kHz			Max ISF = 38.4 kHz		
	No of Samples	Fs (kHz)	Delay (ms)	No of Samples	Fs (kHz)	Delay (ms)	No of Samples	Fs (kHz)	Delay (ms)
Encoder lookahead									
Analysis lookahead	256	12.8	20.0000	256	6.4	40.0000	288	19.2	15.0000
Down-sampling	24	48.0	0.5000	48	48.0	1.0000	16	48.0	0.3333
Band-splitting	24	25.6	0.9375	24	12.8	1.8750	24	38.4	0.6250
Decoder lookahead									
Postfiltering	140	12.8	10.9375	140	6.4	21.8750	140	19.2	7.2917
Band-merging	12	12.8	0.9375	12	6.4	1.8750	12	19.2	0.6250
Upsampling	12	25.6	0.4688	12	12.8	0.9375	12	38.4	0.3125
Superframe size	1 024	12.8	80.0000	1024	6.4	160.0000	1024	19.2	53.3333
Total delay (ms)			**113.7813**			**227.5625**			**77.5208**

The algorithmic delay for the typical Internal Sampling Frequency (ISF) of 25.6 kHz is computed as 113.7813 ms. The algorithmic delay for the two extreme cases corresponding to maximum ISF of 38.4 kHz (32 kbps mono operation) and minimum ISF of 12.8 kHz (5.2 kbps mono operation) are also computed (77.5208 ms and 227.5625 ms, respectively).

The delay for other ISFs can be approximately estimated as 113.78 × (25.6/ISF), where ISF is the internal sampling frequency in kHz.

Delay in stereo operation

The delay analysis for stereo operation is shown in table 8. The input signal is assumed sampled at 48 kHz. The table gives the encoder lookahead, decoder lookahead, and the superframe size. The algorithmic delay is measured as the superframe size plus the lookahead of both encoder and decoder. The breakdown of the delay is performed by listing the delay in samples for given operation, then the delay is converted from samples to ms based on the sampling frequency corresponding to that operation (in column Fs).

The algorithmic delay for the typical internal sampling frequency (ISF) of 25.6 kHz is computed as 162.8438 ms. The algorithmic delay for the two extreme cases corresponding to maximum ISF of 38.4 kHz and minimum ISF of 12.8 kHz are also computed (77.5208 ms and 227.5625 ms, respectively).

The delay for other ISFs can be approximately estimated as 162.84 × (25.6/ISF), where ISF is the internal sampling frequency in kHz.

Table 8: Delay in stereo operation for typical ISF of 25.6 kHz and for maximum ISF of 38.4 kHz, and minimum ISF of 12.8 kHz

	Typical ISF = 25.6 kHz			Min ISF = 12.8 kHz			Max ISF = 38.4 kHz		
	No of Samples	Fs (kHz)	Delay (ms)	No of Samples	Fs (kHz)	Delay (ms)	No of Samples	Fs (kHz)	Delay (ms)
Encoder lookahead									
Analysis Lookahead	512	12.8	40.0000	512	6.4	80.0000	512	19.2	26.6667
Down-sampling	24	48	0.5000	48	48	1.0000	16	48	0.3333
Band-splitting	24	25.6	0.9375	24	12.8	1.8750	24	38.4	0.6250
Decoder lookahead									
Postfiltering	512	12.8	40.0000	512	6.4	80.0000	512	19.2	26.6667
Band-merging	12	12.8	0.9375	12	6.4	1.8750	12	19.2	0.6250
Upsampling	12	25.6	0.4688	12	12.8	0.9375	12	38.4	0.3125
Superframe size	1 024	12.8	80.0000	1024	6.4	160.0000	1024	19.2	53.3333
Total delay (ms)			**162.8438**			**325.6875**			**108.5625**

9.3.2 Eaac+ codec delay

Introduction

The clause holds the delay information for the Enhanced aacPlus codec.

Delay by component

The Enhanced aacPlus codec including pre-downsampling contains a number of delay sources, consisting of:

- 25 (fixed point encoder) resp. 6 (floating point encoder) samples for a 2 : 1 downsampling filter;
- 58 (fixed point encoder) resp. 5 (floating point encoder) samples for a 3 : 2 downsampling filter;
- 2 048 samples for an overlapped MDCT;
- 1 152 samples look-ahead buffer within the psychoacoustic module;
- 962 samples SBR encoding and decoding delay;
- 958 samples PS encoder downmix;
- 2 048 samples (1 Frame) decoder concealment;

- Bit-rate dependant bit-reservoir.

Delay overview of the constant delay contributors

When calculating the constant algorithmic codec delay contributors, the following additional considerations need to be taken into account:

- The PS downmix includes also the downsampling of the downmixed input signal, the 25 resp. 6 samples delay for the 2: 1 downsampling filter does not apply.
- For mono bit-rates below 12 kbps, Enhanced aacPlus operates with a sampling rate of 32 kHz. Here the 3: 2 downsampling filter is used for 48 kHz to 32 kHz input signal conversion.

Consequently this results in table 9.

Table 9

Encoding mode	2: 1 Down-sampler	3: 2 Down-sampler	MDCT	Psych look-ahead	SBR	Conceal ment	PS	Overall constant delay [samples, fixed point enc / floating point enc]
mono, br < 12 000	Yes	Yes	Yes	Yes	Yes	Yes	No	6293 / 6221
mono, br ≥ 12 000	Yes	No	Yes	Yes	Yes	Yes	No	6235 / 6216
PS-stereo, br < 36 000	No	No	Yes	Yes	Yes	Yes	Yes	7168
stereo, br ≥ 36 000	Yes	No	Yes	Yes	Yes	Yes	No	6235 / 6216

Delay contribution of the bit-reservoir

In cases where the codec is run in a streaming application, i.e. non-file based applications, the bitreservoir of AAC is also contributing to the overall codec delay. The bitreservoir has a size of 6 144 bits for mono and parametric stereo, and 2 × 6 144 bits for stereo. Depending on the choice of the bit-rate when running the encoder the additional delay in samples can be expressed by:

$$d[samples] = \frac{bitres\_size}{average\_frame\_size} \cdot frame\_len = \frac{bitres\_size}{(bitrate \cdot frame\_len)/f_s} \cdot frame\_len = \frac{bitres\_size \cdot f_s}{bitrate}$$

Table 10 holds some example calculations for the overall codec delay.

Table 10

Encoding mode	Constant delay contribution	Fs [Hz]	Bitreservoir Delay [samples]	Overall codec delay [ms]
mono, br =10 000	6293 / 6221	32 000	19 661	ca. 811 / 807
PS-stereo, br = 16 000	6235 / 6216	48 000	18 432	ca. 513
PS-stereo, br = 24 000	7168	48 000	12 288	ca. 405
stereo, br = 48 000	6235 / 6216	48 000	12 288	ca. 386

9.4 Performance verification with 3D audio signal

Introduction

This clause reports on subjective tests conducted by France Telecom for the Audio Codec verification phase.

France Telecom volunteered to verify the behaviour of recommended audio codec characterization on 3D-signals (binauralised virtual surround sound).

This clause reports on subjective tests to verify the codec behaviour on 3D-signals.

Test process

Void.

Test method

This method has been recommended at the ITU-R under the name BS.1534-1 [1]. It was developed in 1999 by the EBU Project Group B/AIM in collaboration with the ITU-R Working Party 6Q.

An important feature of this method is the inclusion of the hidden reference and bandwidth limited anchor signals. To adapt this method for testing 3D audio signals anchors with reduced spatial image (spatialisation closer to the head) were also introduced. For this test, anchor points were the band-limited (7 kHz and 10 kHz) reference signal with full spatial image and with reduced spatial image.

Test material

The test items can be classified in mixed content items where speech and music were combined and music items containing only music. 5 music items and 5 mixed content items were considered during this test. One of each type being used in the training session.

Some items were extracted from 5.1 recorded audio files and the others were created as synthetic 5.1 audio files. Then these items were binauralized through the binaural virtual surround technique.

One set of HRTF (Head Related Transfer Function) was used for all subjects.

Training phase

Each listener had a period of training, in order to get familiar with the test methodology, the use of the interface software and with the kind of quality they have to assess. This was as well an opportunity to adjust the restitution level that then remained constant during the test phase.

The training session contained 2 audio items.

User Interface

The MUSHRA method has the advantage of displaying all stimuli for one test item at a given bit-rate at the same time. The subjects were therefore able to carry out any comparison between them directly as well as to assess the quality comparing to the one of the explicit reference signal.

Implementation of MUSHRA user interface from CRC (SEAQ, System for the Evaluation of Audio Quality from Communications Research Center Canada)) was used in the tests. A screenshot of one implementation of the user interface is shown in figure 47. The buttons represent all the configurations/codecs under test including the hidden reference and both the anchor signals, and the reference, which is specially displayed on the left as "REF". Above each button, with the exception of the "REF" one, a slider is used to grade the quality of the test item according to the continuous quality scale.

For each of the test items, the signals under test were randomly assigned, with a different assignment for each subject. In addition, the test items were randomized for each subject within a session to avoid sequential effects. The session files were prepared by the host lab. There was one session file per listener.

The same randomization process was used for the training sessions : there was one training session per listener.

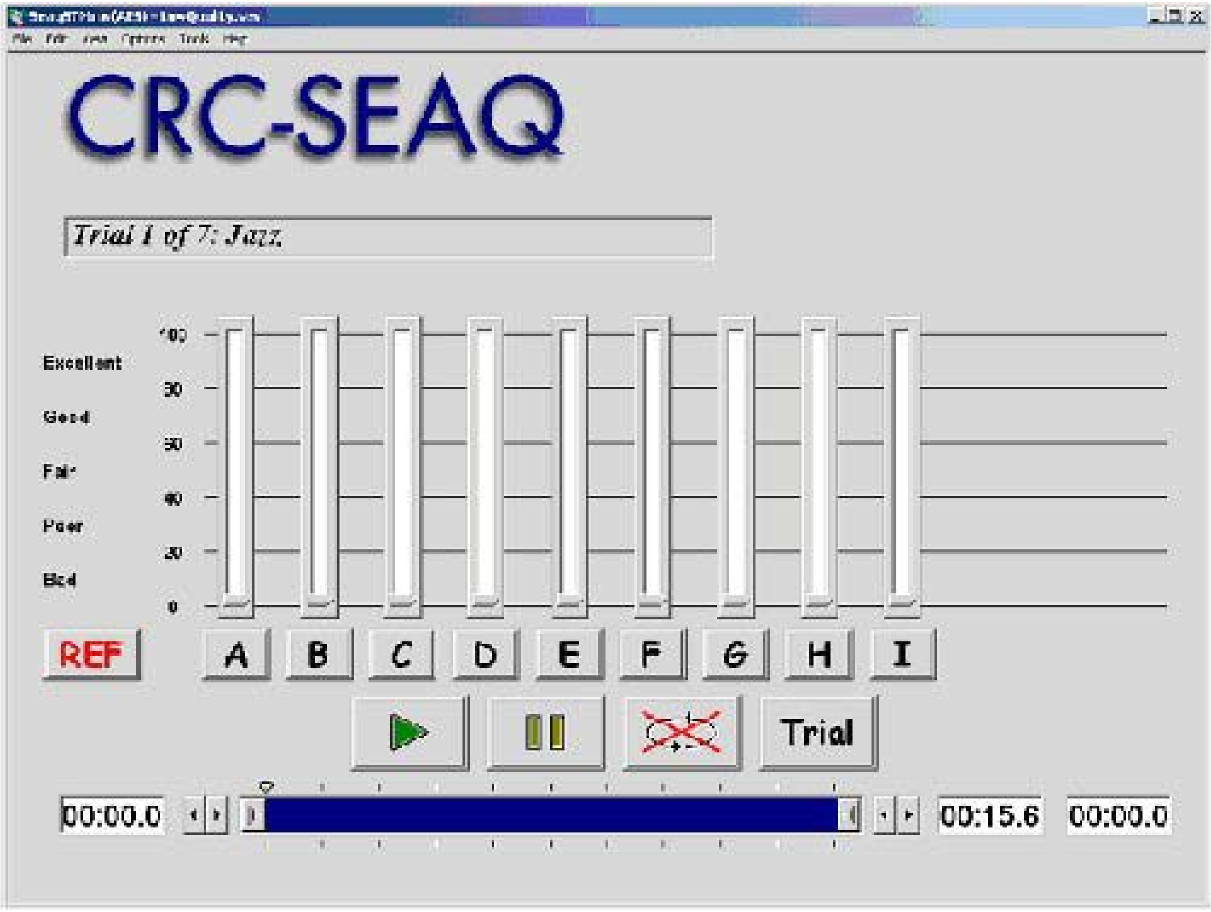


Figure 47: MUSHRA software

The Listening Panel

The listening panel consisted of 12 subjects, most of them experienced in audio but not only professionally involved. All the 12 remaining listeners were respectful regarding the listening instructions.

Tests duration

As mentioned above the test was preceded by a training period.

The training phase took about half an hour. This time was also used to describe the listening instructions and answer listeners' questions if any. If the listeners have faced difficulties in the assessment of the quality, this time was also used to explain them how to behave.

Then, one test took approximately 1 hour and 30 minutes (depending on listeners), including breaks. Every 20 minutes, the listener was asked to rest a bit by walking and breathing some fresh air.

Listening conditions

The tests were performed on the headphone STAX Signature SR-404 (open) and its amplifier SRM-006t. The subjects had the possibility to set the reproduction level individually before they started the actual test (during the training phase). The subjects were then restricted from changing the reproduction level during the test.

The test items were stored on a Windows 2k workstation. The digital sound was played through the PC board Digigram VX 222 and converted by 24 bits DAC (3Dlab DAC 2000).

The tests were run in an acoustically neutral room dedicated to such tests.

Statistical analysis

The statistical analysis method described in the MUSHRA specifications was used to process the test data. The results are presented as mean grades and standards deviation.

Experience has shown that the scores obtained for different test sequences are dependent on the criticality of the test material used. Therefore, this figure have been included in this report in order to provide a more complete understanding of codec performance by presenting results for different test sequences rather than only as aggregated averages across all the test sequences used in the assessment.

Test results

The test results are presented below.

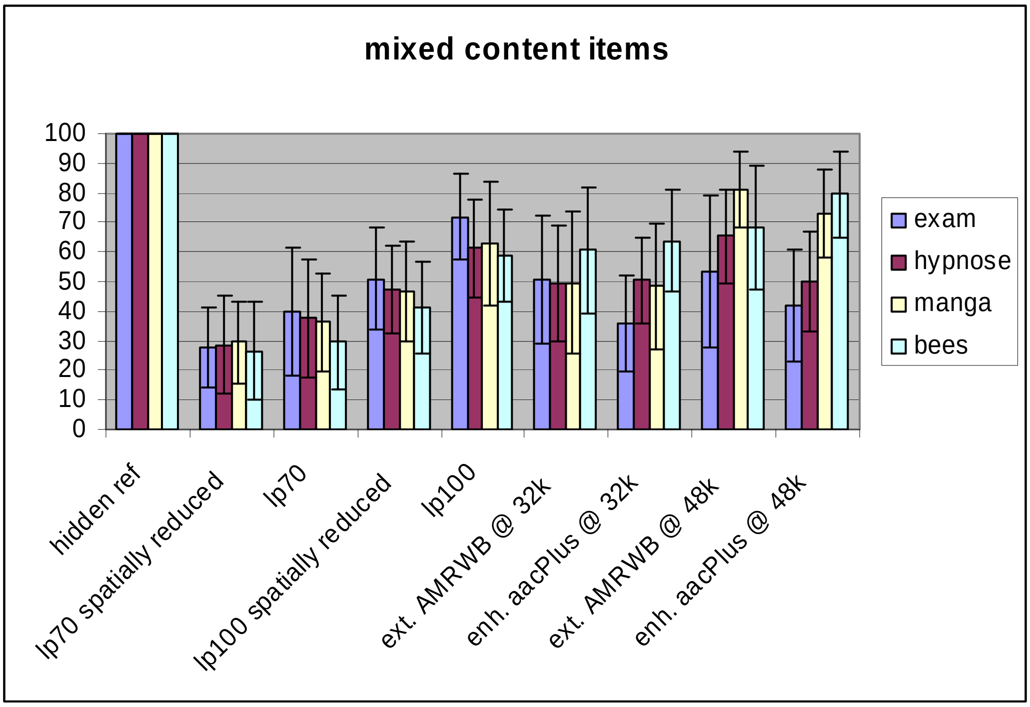


Figure 48

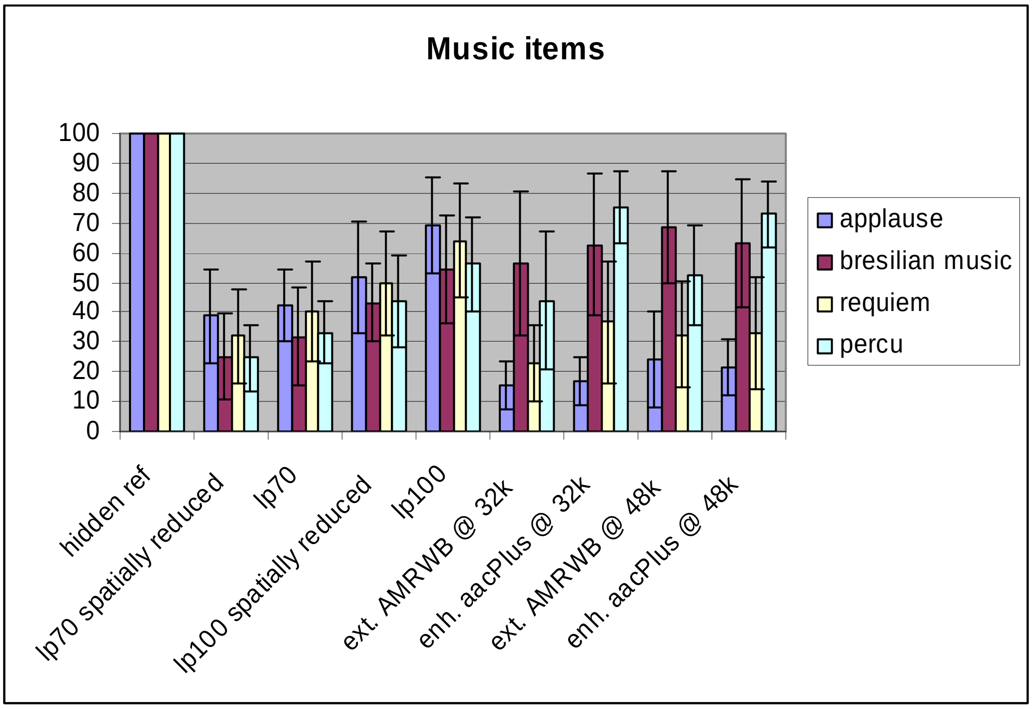


Figure 49

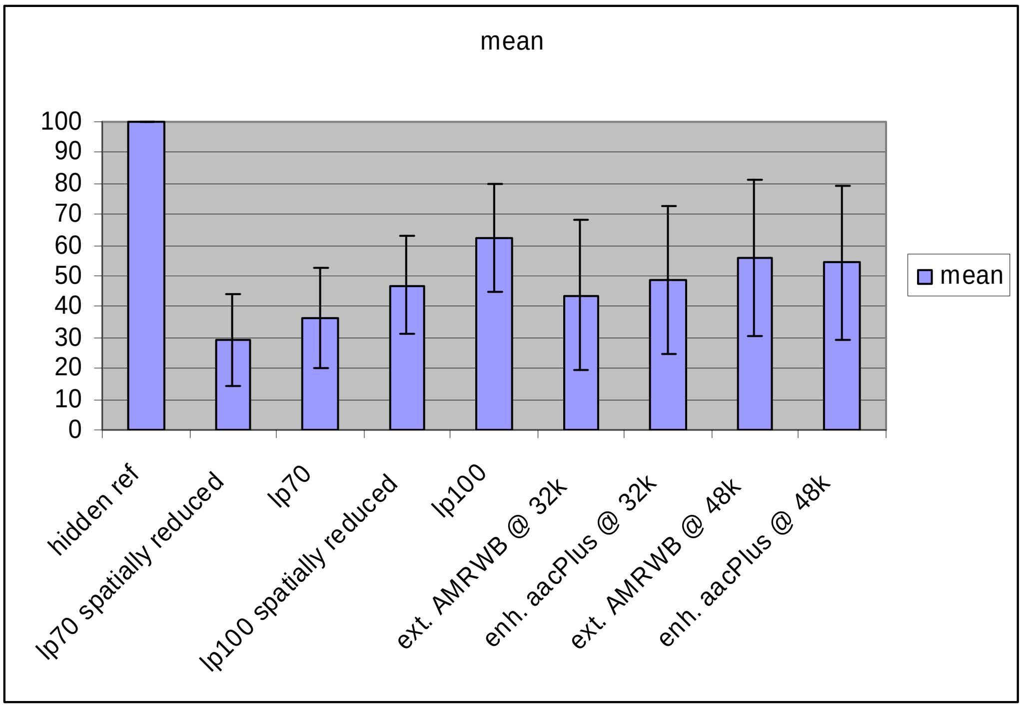


Figure 50

Conclusion

Even if the aggregated average scores over all test sequences tend to show that the quality is acceptable, for some items the perceived quality is rather poor. Moreover, the test was run with the highest bit rates in order to reach this quality.

9.5 Verification of rate switching performance

The ability of seamless bit rate switching of an audio decoder is a desirable property when, e.g. due to traffic load (congestion) the bit rate available for the service changes during the session. Seamless switching would allow changing the bit rate without audible artifacts.

9.5.1 Rate switching performance of AMR-WB+

Introduction

The motivation was to check whether seamless switching between various modes (i.e. no audible artifacts when switching the mode) would be possible by using the AMR-WB+ codec.

Test Methodology

The complete selection test material was used in the tests. It has a length of around 8 minutes and contains various kinds of speech, music, and mixed content in a randomized order.

The bit-rate switching was implemented through the configuration file feature of the AMR-WB+ fixed-point code in 3GPP TS 26.273 [4]. The bit-rate was changed every 1sec randomly between modes. So in total, around 480 switches were implemented by this approach.

mono and stereo processing were done in two separate experiments:

- For mono encoding, the sampling frequency was 16 kHz. The -lc option was applied. The internal sampling frequency was constant (25 600 Hz) in all cases. No DTX operation was applied. In the mono case, AMR-WB and AMR-WB+ modes were switched randomly in the mode index range of [0 to 15], see table 21 in 3GPP TS 26.290 [5]. In the decoder, the limiter option was turned on. The decoder was forced to provide mono output by using the -mono flag.
- For stereo encoding, the sampling frequency was 48 kHz. The internal sampling frequency was switched randomly in the range of [0.5 to 1.5] simultaneously with every mode switch. No DTX operation was applied. In the stereo case, AMR-WB+ extension modes were switched randomly in the mode index range of [24 to 47], see table 25 in 3GPP TS 26.290 [5]. In combination of mode index and internal sampling frequency switching, the bit-rate varied in the range of 7 kb/s to 45 kb/s. In the decoder, the -limiter option was turned on.

Listening

The decoded materials were examined by expert listeners for audible artifacts.

Results

The AMR-WB+ code was able to encode and decode fully the input material. No audible artifacts could be perceived in any case. Hence it can be stated that the AMR-WB+ codec is able to provide seamless bit-rate switching in the framework of the experiments we conducted.

9.5.2 Rate switching performance of Eaac+

Introduction

This clause presents the results of a study of the bit rate switching behaviour of the Eaac+ decoder.

Methodology

The following procedure was used for carrying out the task:

1) Generation of configuration files for the Eaac+ encoder steering the encoding bit rate to toggle between various rates of interest. The toggling was done with a rate of about one rate change per second.
2) Decoding of the generated audio parameter file with Eaac+ decoder (3GPP TS 26.410 [6]).

3) Informal listening by expert listeners and further analysis.

NOTE: The investigation was limited to bit rate toggling within a given Eaac+channel mode configuration, i.e. in between mono, in between PS stereo, and in between conventional stereo mode not using the PS tool. Switching across configurations, namely between mono and stereo or between PS stereo and conventional stereo (i.e. crossing the 36 kbps border) was not investigated since the used encoder did not provide such a functionality.

Audio data

The audio data used in the study was sound items from the selection test.

Experiments

The bit rates between which the toggling was performed were chosen with the preference to cause a change of the reconstructed audio bandwidth. In particular, the following cases have been investigated:

- mono: togglings between 12/17 kbps, 17/19 kbps, 21/23 kbps, 27/29 kbps, 17/29 kbps.
- PS stereo: togglings between 16/17 kbps, 17/19 kbps, 21/23 kbps, 27/29 kbps, 17/29 kbps.
- Conventional (non-PS) stereo: 43/45 kbps, 45/51 kbps.

Results

The standardized Eaac+ decoder was fully capable of decoding the bitstream files with toggled bit rate within the given Eaac+ channel mode configurations. No particular switching effects were encountered even though the rate change may lead to a slight change of the spectral nature of the reconstructed signal. In conclusion it can be stated that the Eaac+ decoder provides seamless bit rate switching capability as long as the Eaac+ channel mode configuration is not changed.

9.6 Verification of source code

The verification aims at the following objectives:

- to verify that the algorithms are based on a fixed-point 16/32-bit arithmetic;
- to verify that the reference source code is instrumented with the ETSI basic operators that are used for the AMR and AMR-WB speech codecs;
- to verify that the instrumentation follows the rules from document 'ETSI SMG-11 AMR#9: Complexity and delay assessment" (included in attachment "TR 26.936 Annex B3 Verification test documents.zip').

9.6.1 Source code verification of AMR-WB+

Introduction

According to the verification plan in Tdoc S4-050187 "Source code verification plan v1.0" (included in attachment "TR 26.936 Annex B3 Verification test documents.zip'), STMicroelectronics has conducted the verification of the reference source code from the extended AMR-WB (AMR-WB+) decoder.

Verification of the format of the C-code

The verification laboratory has verified the following items for both reference source codes:

- the source code of the AMRWB+ decoder compiles on MSVC (Microsoft Visual C);
- there is no obvious deviation from the ANSI-C standard that would prevent compliant ANSI-C compiler to compile the source code.

The verification laboratory has also verified that the only calls to external libraries that happened in the reference source codes were limited to:

- calls to libc: fwrite()/fread()/fseek() for file IO, printf()/fprintf() for command line text display, assert() for debug assertion, malloc()/free() for memory allocation at the initialization stage of the decoder;
- calls to libisomedia which is the library implementing the API for the 3GP file format.

Verification of the usage of the fixed-point arithmetic

The verification laboratory has verified that the audio decoding algorithm implemented through the reference source C code does not make usage of any floating-point instructions. It was checked that the algorithm relies exclusively on a 16/32-bit arithmetic. Other types of arithmetic (32/32-bit arithmetic and floating point arithmetic) are emulated through the 16/32-bit instructions.

The verification laboratory believes that the source code provides a full bit-exact reference representation from the behaviour of the audio decoding algorithm and therefore, that it is admissible for a bit-exact reference representation of the AMR-WB+ decoder.

Verification of the instrumentation

Objective

The verification laboratory has checked that the source code implements the basic operators and the instrumentation according to the rules described in the document 'ETSI SMG-11 AMR#9: Complexity and delay assessment". The primary objective is to ensure that the figures of complexity (wMOPs, stack depth) estimated at run time by the software decoder are comparable to the figures of complexity obtained from past exercises.

The secondary objective is to ensure that the source code can easily be ported on a DSP target by mapping basic operators or combination of basic operators onto the native intrinsic functions from the target platform.

General comments

The verification laboratory has mainly verified the functions that participate to the main loop of the decoding algorithm.

Therefore, the instrumentation of the functions corresponding to the initialization stages of the decoder were not carefully verified since those functions do not participate at all to the evaluation of the figure of complexity of the algorithm (but still, the bit-exactness and the fixed-point implementation were verified).

The verification laboratory reports also that the boundary between the file I/O operations that require to be instrumented and the file I/O operations that do not require to be instrumented is not very clear. In particular, the verification laboratory had no time to check whether the AMR-WB+ decoder has defined the same borders as limits of the source code instrumentation than AMR-WB or Eaac+.

The verification laboratory reports also that the source code for AMRWB+ does not fully conform to the proposed usage of the step increment. It exists cases where the increment is not a constant and the amount of iteration in the loop is not easy to obtain. Depending on the cases, the verification laboratory believes that the environment of such loop statements could be re-written in order to finally fulfill the step increment constraints; alternatively, those loop statements could be re-cast as software loops. In any case, the impact on the bit-exactness in null and the impact on the overall figure of complexity will not be significant.

List of files verified

Table 11: List of files that were verified

File name	Verified
common_fx/ALF_emph_fx.c	OK
Common_fx/bits_p_fx.c	OK
common_fx/Bitstream_fx.c	OK
common_fx/decim12k8_fx.c	OK
common_fx/bits_p_fx.c	OK
common_fx/fft3_fx.c	OK
common_fx/fft9_fx.c	OK
common_fx/gaintcx_fx.c	OK
common_fx/hf_func_fx.c	OK
common_fx/int_lpc_p_fx.c	OK
common_fx/join_split_fx.c	OK
common_fx/overs12k8_fx.c	OK
common_fx/q_gn_hf_fx.c	OK
common_fx/q_isf_hf_fx.c	OK
common_fx/r_fft_fx.c	OK
common_fx/re8_dec_fx.c	OK
common_fx/Re8_dic_fx.c	
common_fx/re8_ppv_fx.c	OK
common_fx/Re8_util_fx.c	OK
common_fx/read_dat_fx.c	OK
common_fx/rnd_ph16_fx.c	OK
common_fx/tables_plus_fx.c	
common_fx/tables_stereo_fx.c	OK
common_fx/util_plus_fx.c	OK
common_fx/util_stereo_x_fx.c	OK
common_fx/wavefiletools_fx.c	OK
common_fx/writ_dat_fx.c	OK
decoder_fx/avq_dec_fx.c	OK
decoder_fx/bass_pf_fx.c	OK
decoder_fx/d_gain2p_fx.c	OK
decoder_fx/d_isf_2s_fx.c	OK
decoder_fx/dec_ace_fx.c	OK
decoder_fx/dec_cp_state_fx.c	OK
Decoder_fx/dec_hf_fx.c	OK
Decoder_fx/dec_if_fx.c	OK
Decoder_fx/dec_lf_fx.c	OK
decoder_fx/dec_main_s_fx.c	OK
decoder_fx/dec_prm_fx.c	OK
decoder_fx/dec_tcx_fx.c	OK
decoder_fx/dec_wbplus_fx.c	OK
decoder_fx/Scale_dec_fx.c	OK
decoder_fx/tcx_ecu_fx.c	OK
stereo_fx_eks/d_stereo_x_fx.c	OK
stereo_fx_eks/dec_hi_stereo_fx.c	OK
stereo_fx_eks/dec_tcx_stereo_fx.c	OK

Comments

Based on the suggestions made by the verification laboratory, the instrumentation of the source code was slightly cleaned while keeping its bit-exact behaviour. The resulting source code complies with the rules from the document 'ETSI SMG-11 AMR#9: Complexity and delay assessment".

Conclusion

The verification laboratory feels confident that the source code of the AMR-WB+ decoder can serve as a bit-exact reference of the algorithm. The verification laboratory thinks that the instrumentation of the source code is conformed to the expectation of a 3GPP reference source code.

9.6.2 Source code verification of Eaac+

Introduction

According to the verification plan in Tdoc S4-050187 "Source code verification plan v1.0" (included in attachment "TR 26.936 Annex B3 Verification test documents.zip') , STMicroelectronics has conducted the verification of the reference source code from the enhanced aacPlus (Eaac+) decoder.

Verification of the format of the C-code

The verification laboratory has verified the following items for both reference source codes:

- the source code of the Eaac+ decoder compiles on a Linux platform, with GNU gcc version 3.2.3 and GNU make version 3.79.1;
- there is no obvious deviation from the ANSI-C standard that would prevent compliant ANSI-C compiler to compile the source code;
- the text of the specification document 3GPP TS 26.411 [7] matches the file tree structure delivered to the verification laboratory.

The verification laboratory has also verified that the only calls to external libraries that happened in the reference source codes were limited to:

- calls to libc: fwrite()/fread()/fseek() for file IO, printf()/fprintf() for command line text display, assert() for debug assertion;
- calls to libisomedia which is the library implementing the API for the 3GP file format.

Verification of the usage of the fixed-point arithmetic

The verification laboratory has verified that the audio decoding algorithm implemented through the reference source C code does not make usage of any floating-point instructions. It was checked that the algorithm relies exclusively on a 16/32-bit arithmetic. Other types of arithmetic (32/32-bit arithmetic and floating point arithmetic) are emulated through the 16/32-bit instructions.

The verification laboratory believes that the source code provides a full bit-exact reference representation from the behaviour of the audio decoding algorithm and therefore, that it is admissible for a bit-exact reference representation of the Eaac+ decoder.

Verification of the instrumentation

Objective

The verification laboratory has checked that the source code implements the basic operators and the instrumentation according to the rules described in the document 'ETSI SMG-11 AMR#9: Complexity and delay assessment" . The primary objective is to ensure that the figures of complexity (wMOPs, stack depth) estimated at run time by the software decoder are comparable to the figures of complexity obtained from past exercises.

The secondary objective is to ensure that the source code can easily be ported on a DSP target by mapping basic operators or combination of basic operators onto the native intrinsic functions from the target platform.

General comments

The verification laboratory has mainly verified the functions that participate to the main loop of the decoding algorithm.

Therefore, the instrumentation of the functions corresponding to the initialization stages of the decoder were not carefully verified since those functions do not participate at all to the evaluation of the figure of complexity of the algorithm (but still, the bit-exactness and the fixed-point implementation were verified).

The verification laboratory reports also that the boundary between the file I/O operations that require to be instrumented and the file I/O operations that do not require to be instrumented is not very clear. In particular, the verification laboratory had no time to check whether the Eaac+ decoder has defined the same borders as limits of the source code instrumentation than AMR-WB or AMR-WB+.

The verification laboratory reports also that the source code of Eaac+ does not fully conform to the proposed usage of the step increment. It exists cases where the increment is not a constant and the amount of iteration in the loop is not easy to obtain. Depending on the cases, the verification laboratory believes that the environment of such loop statements could be re-written in order to finally fulfill the step increment constraints; alternatively, those loop statements could be re-cast as software loops. In any case, the impact on the bit-exactness is null and the impact on the overall figure of complexity will not be significant.

List of files verified

Table 12: List of files that were verified

Directory	Module name	Verified
src/	main.c	OK
	fileifc.c	OK
	spline_resampler.c	OK
etsiop_aacdec/	aacdecoder.c	OK
	streaminfo.c	OK
	channelinfo.c	OK
	stereo.c	OK
	longblock.c	OK
	shortblock.c	OK
	pulsedata.c	OK
	block.c	OK
	pns.c	OK
	imdct.c	OK
	tns.c	OK
	bitstream.c	OK
	channel.c	OK
	conceal.c	OK
	datastream.c	OK
	aac_ram.c	OK
	aac_rom.c	OK
etsiop_sbrdeclib/	env_dec.c	OK
	aacpluscheck.c	OK
	env_calc.c	OK
	lpp_tran.c	OK
	sbrdecoder.c	OK
	sbr_dec.c	OK
	sbr_crc.c	OK
	hybrid.c	OK
	ps_bitdec.c	OK
	env_extr.c	OK
	freq_sca.c	OK
	ps_dec.c	OK
	qmf_dec.c	OK
	sbr_ram.c	OK
	sbr_rom.c	OK
etsiop_bitbuf/	bitbuffer.c	
etsiop_ffrlib/	fft_32x32.c	OK
	transcendent.c	OK
	transcendent_enc.c	OK
	intrinsics.c	OK
	vector.c	OK

Comments

Based on the suggestions made by the verification laboratory, the instrumentation of the source code was slightly cleaned while keeping its bit-exact behaviour. The resulting source code complies with the rules from the document 'ETSI SMG-11 AMR#9: Complexity and delay assessment' (apart from the double precision multiplication issue which is dealt in section "16x32 and 32x32 multiplication").

16x32 and 32x32 multiplication

The verification laboratory has noted that double precision multiplication is used. Such multiplication is difficult to simulate in full accuracy with the ETSI basic operators.

Apart for this workaround, the verification laboratory is not aware of any full precision 16x32 workaround used in 3GPP codecs and based on the ETSI basic operators.

The verification laboratory is aware of the workaround built in the AMR-WB source code and known as the DPF arithmetic. This workaround provides a 16x31 multiplication for 3 wops (weighted operations). It costs 5 wops in order to enter the DPF mode and costs 3 wops in order to exit the DPF mode.

The verification laboratory has noted that the Eaac+ has built another workaround for the 16x31 multiplication based on a split in two arrays, LSB / MSB, from the double precision argument. It is believed that it exists a mean to re-organize the processing and the data storage that would validates the proposed implementation. It is understood that if such a code and data re-organization was provided as a reference source code, it would cause a significant workload for a developer to revert back to the original organization which exhibit natively the 16x31 multiplications and is easily mapped onto DSP intrinsic functions.

Nevertheless, the current implementation does not exhibit such organization and therefore the verification laboratory can not state whether such workaround is valid.

Conclusion

The verification laboratory feels confident that the source code of the Eaac+ decoder can serve as a bit-exact reference of the algorithm.. The verification laboratory thinks that the instrumentation of the source code is conformed to the expectation of a 3GPP reference source code.

9.6.3 General discussion

During the source code verification phase, the instrumentation of the source code was slightly cleaned while keeping a bit-exact behaviour.

Regarding Eaac+, and more specifically the double precision multiplication, the verification laboratory recognizes that requesting from the source code to be clean enough so that the port on a DSP platform is straightforward (which is one of the objective targeted by the release of a reference source code) was in contradiction with the primary objective that requests an instrumentation which provides figures of complexity that scales smoothly between existing algorithms.

The verification laboratory understands that the nature of the audio codec technology (including the Eaac+ decoder, but not only) relies heavily on the 16x31-bit multiplication. The metric of the ETSI basic operators is designed in a way that penalizes heavily the usage of the double precision multiplication because such features were not common on baseband DSP. As a matter of fact, algorithms based on speech coding technology do not use heavily this kind of arithmetic.

STMicroelectronics thinks that if the metric of the ETSI basic operator was modified in order to lessen the penalty of the double precision multiplication, then, first of all, it would not impact significantly the wMOPs score of the algorithm, even if it was re-written in order to replace the DPF arithmetic by plain 16x31 and 31x31 arithmetic; second, we believe that the design choices, that were introduced at the very beginning of the design of the AMR-WB and AMR-WB+, partially based on the metric that was available at that time (penalizing the double precision multiplication), would not have been made definitely different, if the metric had been different and the 16x31 and 31x31 arithmetic was not penalized.

STMicroelectronics believes that the audio decoders under study will not be implemented on platforms that do not provide a support for the double precision multiplication. Therefore, we support the idea that the reference source code exhibits facilities for implementing the algorithm on platform supporting double precision multiplication.

Therefore, it is understood that the wMOPS values obtained from the current ETSI basic operators would not scale properly from the AMR and AMR-WB to the Eaac+.

The following way forward was agreed:

- The reference source code provides (for instance through compilation flags) two alternative (but bit-exact) implementations:
  - an implementation providing a source code fully based (but not necessarily optimized) for ETSI basic operators;
  - an implementation that allows a fast and efficient port for a DSP, where the 16x31 multiplication from the reference source code maps easily on the DSP intrinsic functions.
- It was agreed that in the characterization report, the wMOPs score obtained from the reference source codes is split in two values: the value that is directly due to the 16x31 and 31x31 arithmetic and the value that is due to

everything else. From those two values, any manufacturer will be able to derive an actual figure of complexity that scales properly on its own platform.

9.7 Content dependency

One aspect of this verification item was addressed by a contribution from a codec proponent in which a subjective test was conducted to assess the stereo performance of *AMR-WB+* and *Eaac+* with real-world critical music material having certain signal characteristics. The report on this experiment can be found in document S4-050710 (see Attachment "TR26.936 Annex B2 Additional Documents.zip" to the present document).

Annex A:
Test results from other bodies

This annex comprises test results for the two standardized codecs carried out by ITU-T, where the codecs participated as reference codecs. These results consist of two parts, one part obtained by MUSHRA testing of music and mixed material , the other part obtained by ACR (MOS) and DCR (DMOS) testing of speech material. Documents S4-050260-part1 and S4-050260-part2 (see Attachment "TR26.936 Annex B2 Additional Documents.zip" to the present document) contain the complete results for the MUSHRA and ACR and DCR results from the ITU-T tests.

Additional test results by MPEG [2] and EBU [3] provide further information for configurations using the *Eaac+* decoder in combination with a non-standardized encoder.

Since those tests were conducted outside of 3GPP, test conditions, sampling rates and/or bit-rates may not be directly applicable to use cases in a 3GPP environment.

A.1 Test results from ITU-T standardization of G.722-1, annex C

The results shown in figures A.1 to A.6 were derived from experiments involved in the ITU-T standardization of codec G.722.1, annex C. All six of these experiments were conducted under mono conditions and involved test items band-limited from 50Hz-14kHz. Two MUSHRA experiments were conducted in the ITU-Phase 2 series of tests that, taken together, characterize the performance of the two 3GPP audio codecs and one reference audio codec across two bit-rates. The MUSHRA tests were conducted with music and mixed content. Figures A1 and A2 show the results of G722-2 MUSHRA tests involving the three audio codecs at 24 kbps and 32 kbps, respectively. These results are based on the MUSHRA ratings of 20 subjects, 10 test items, and one listening lab (N = 20*10*1 = 200).

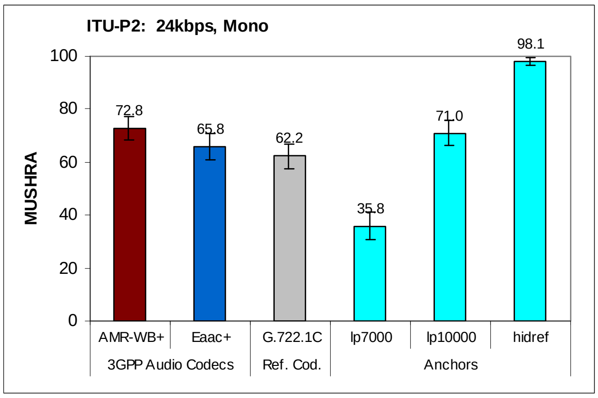


Figure A.1: MUSHRA results for audio codecs operating at 24 kbps

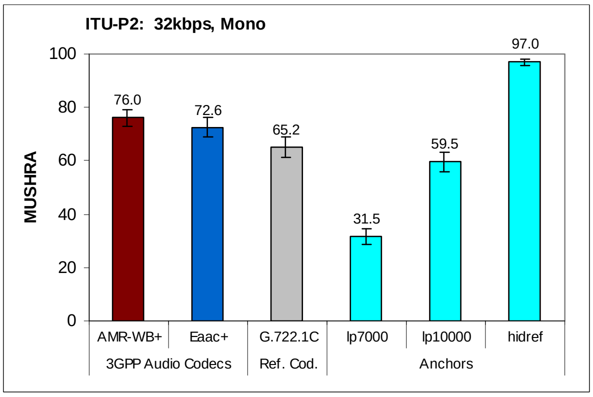


Figure A.2: MUSHRA results for audio codecs operating at 32 kbps

Figure A.3 shows MOS results from the ACR test in mono mode conducted in the ITU-Phase 1 test series (document 'ITU-T Standardization of G.722-1C, part1 (S4-050260).doc' included in Attachment 'TR26.936 Annex B2 Additional Documents.zip').. MOS results are shown for the two 3GPP audio codecs and for one ITU-T reference audio codec, *G.722.1-annex C*, across two bit-rates, 24 k and 32 kbps. The MOS tests were conducted with clean speech and speech with various types of background noises. For two of the audio codecs, MOS performance is better at the higher bit-rate (32 kbps). However, for *AMR-WB+*, MOS is higher for 24 kbps (4.11) than for 32 kbps (3.91). All of the test items involved in the ACR test involved Speech-only audio content.

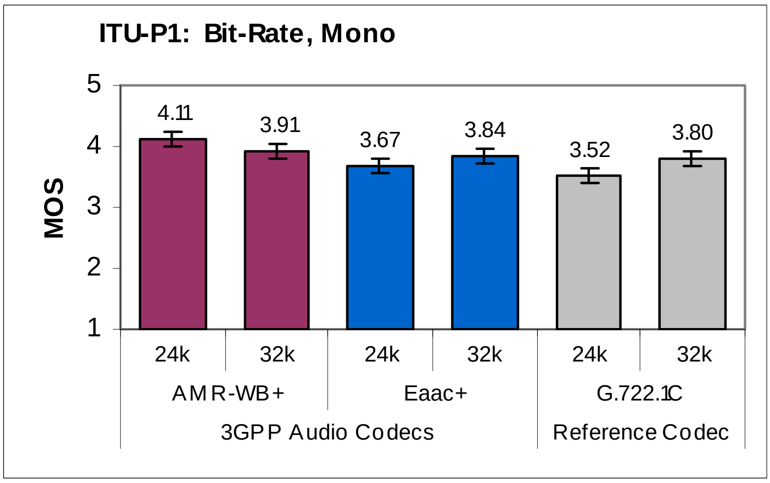


Figure A.3: MOS for audio codecs at bit-rates of 24 and 32 kbps (speech-only)

Figures A.4 to A.6 show results from the three ITU-P1 series of DCR tests, all for the mono mode. In each of these figures, DMOS results are shown for the three audio codecs across two bit-rates, 24 k and 32 kbps. The three DCR tests characterize the performance of the audio codecs in background noise conditions. Figure A.4 shows DMOS results for Office Noise, figure A.5 for Interfering Talker, and figure A.6 for Office Noise plus Interfering Talker. All of the test items involved in the DCR tests also involved Speech-only audio content.

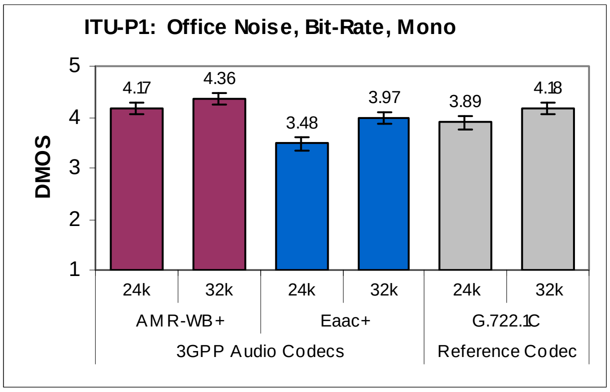


Figure A.4: DMOS for audio codecs at 24 and 32 kbps in office background noise

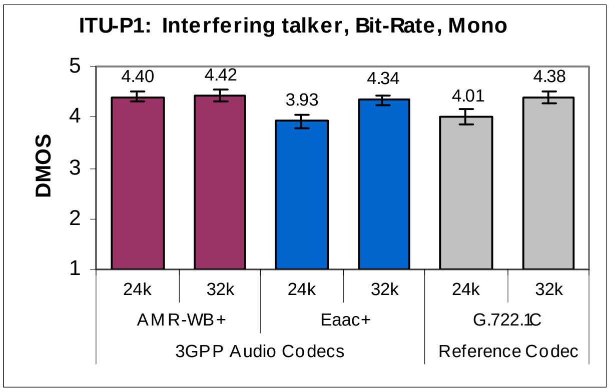


Figure A.5: DMOS for audio codecs at 24 and 32 kbps with interfering talker

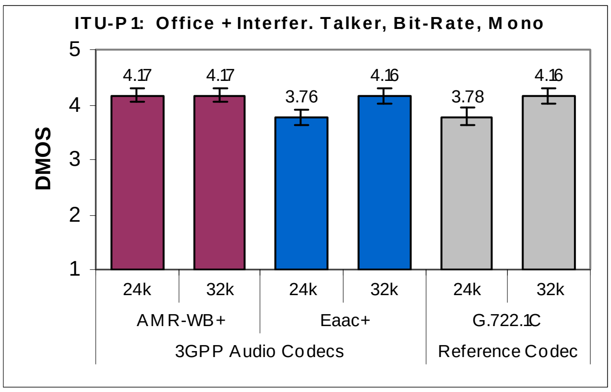


Figure A.6: DMOS for audio codecs at 24 and 32kbps in office noise and interfering talker

Annex B: Documents for information

The documents contained in the attached files :

- TR26.936 Annex B1 Official Test Documents.zip
- TR26.936 Annex B2 Additional Documents.zip
- TR26.936 Annex B3 Verification test documents.zip

provide useful additional information. The formatting of these documents does not follow the 3GPP drafting rules and some of them may contain further references to other temporary documents not essential for the purpose of this TR (and that may not be available for download, since they are not under permanent maintenance).

B.1 Official testing documents (file TR26.936 Annex B1 Official Test Documents.zip)

List of documents :

Attachment 1A : PSS/MMS[/MBMS] Audio Codec Characterization Test Plan

Attachment 1B : Global Analysis Laboratory Report for Phase-1 of the 3GPP Audio Codec Characterization Test for PSS-MMS-MBMS

Attachment 1C : Global Analysis Laboratory Report for Phase 2 of the 3GPP Audio Codec Characterization Test for PSS-MMS-MBMS Applications

Attachment 1D : AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan

Attachment 1E : Global Analysis Laboratory Report on 3GPP Low-Rate Audio Codec Exercises

Attachment 1F : PSS/MMS High-Rate Audio Selection Test and Processing Plan

Attachment 1G : Global Analysis Laboratory Report on 3GPP High-Rate Audio Codec Exercises

B.2 Additional information documents (file TR26.936 Annex B2 Additional Documents.zip)

List of documents :

S4-050260 - Characterisation test results of the 14khz low-complexity audio coding algorithm at 24, 32, and 48 kbps extension to ITU-T G.722.1: phase 1

S4-050260 - Characterisation test results of the 14khz low-complexity audio coding algorithm at 24, 32, and 48 kbps extension to ITU-T G.722.1: phase 2

S4-040439 - Additional information on AMR-WB+ performance

S4-040710 - Additional information: AMR-WB+ performance at very-low bit rates

S4-050453 - AMR-WB+ configurations in characterisation test phase 2

S4-050544 - Characterization test phase 2, settings used for Enhanced aacPlus

S4-050710 - Evaluation of codec behaviour with special input signals

S4-060157 - Background on Error Conditions for the Characterization in TR26.936

B.3 Verification test documents (file TR26.936 Annex B3 Verification test documents.zip)

List of documents :

AMR#9 Permanent document on complexity and delay assessment

S4-050187 - Source code verification plan

Annex C: Change history

Change history							
Date	**TSG # SA**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2005-12	30	SP-050807			Technical Report 26.936 (Release 6) approved at TSG SA#30	2.0.0	6.0.0
2005-12					Minor typographical corrections.	6.0.0	6.0.1
2006-03	31	SP-060015	0001	1	Clean-up of TR 26.936	6.0.1	6.1.0
2007-06	36				Version for Release 7	6.1.0	7.0.0
2008-12	42				Version for Release 8	7.0.0	8.0.0

History

Document history		
V8.0.0	January 2009	Publication