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1 Summary

This document reports on the verification test of the MPEG-4 Parametric Audio coding for high quality audio (also abbreviated in the remainder of this document as SSC). The SSC coder has its optimum performance at bitrates around 24kbps stereo and inherently provides for independent pitch and tempo scaling at the decoder side. The parametric stereo component that is part of this technology can be combined with MPEG-4 HE-AAC, thereby typically increasing the coding efficiency by 33% for bit-rates up to 40kbps stereo.

The verification test compares the performance of the MPEG-4 SSC coder to that of the MPEG-4 AAC Profile coder (i.e. AAC LC codec). The verification test shows that for mono and stereo, the SSC codec, in mean performance, offers a coding gain of more than 25% as compared to the performance of AAC, when both coders operate at or near 24kbps/channel. Furthermore, when operated in stereo mode at 24kbps stereo, the tests also show that for no item the SSC codec performs worse than the AAC codec when both operate at the same bitrate.

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3 Introduction

In mid-1999 the International Standard ISO/IEC 14496-3, MPEG-4 Audio Version 1 was issued and in early 2000 the ISO/IEC 14496-3 / AMD1, MPEG-4 Audio Version 2 was issued. Numerous tests have been conducted by MPEG (see references) to verify that the MPEG-4 standard contains state of the art technology. However, WG11 is always interested in new developments which may provide improvements over the existing MPEG-4 standard and which may lead to extensions of MPEG-4 or to new work items. For this reason, at the 53rd MPEG meeting, in Beijing, MPEG issued a Call for Evidence Justifying the Testing of Audio Coding Technology (N3641). Evidence submitted in response to the Call was examined at the 55th MPEG meeting, in Pisa, and it was determined that there was technology that might improve upon the MPEG-4 standard. Based on the results of the Call for Evidence, work was begun in WG11 to standardize technology parametric coding that could be applied to high quality audio signals.

This work on parametric coding has led to the standardisation of SSC, a parametric coder for high quality audio. SSC partitions a full bandwidth audio signal into transient components, individual sinusoids and noise components. Each of these components is efficiently parameterised and represented in the SSC encoded bitstream. An additional feature of SSC is its inherent property to support independent tempo and pitch scaling at the decoder side. Furthermore, the SSC coder comprises parametric stereo (PS) that can be combined with other audio coders, thereby enhancing the coding efficiency. Specifically, when PS is combined with MPEG-4 HE-AAC, additional reduction in complexity is obtained due to overlapping modules in PS and HE-AAC.

3.1 Test Methodology

For the verification of the new technology, a MUSHRA test (see Annex C) was conducted comparing the performance of SSC with that of MPEG-4 AAC when coding mono and stereo signals at bitrates in the neighborhood of 20 kb/s.

4 Codecs under test

There were two codecs under test for each of the stereo and mono tests. The first codec is the MPEG-4 AAC Profile codec, which is used as a reference of the current state of the art MPEG-4 compression technology, and which comprises the MPEG-4 AAC Low Complexity coding tool. The other codec is the parametric audio coder (SSC).

The codecs under test for the mono and stereo test are shown in Table 1 and Table 2 respectively, which also shows the labels used for each codec in the tables and plots throughout the remainder of this report.

Table 1 - Coders under test in mono test

Coding scheme	Label	Bitrate [kbps]	Sampling rate [kHz]	Typical audio bandwidth [kHz]
MPEG - 4 AAC Profile	MP4-AAC-24	24	24	7
	MP4-AAC-32	32	32	10.5
SSC	SSC-16	16	44.1	18
	SSC-20	20	44.1	18
	SSC-24	24	44.1	18
Anchors and references	H-Ref-Orig	16-bit PCM	44.1	24
	H-Ref-3.5	16-bit PCM	44.1	3.5
	H-Ref-7	16-bit PCM	44.1	7.0

Table 2 - Coders under test in stereo test

Coding scheme	Label	Bitrate [kbps]	Sampling rate [kHz]	Typical audio bandwidth [kHz]
MPEG - 4 AAC Profile	MP4-AAC-24	24	24	7
	MP4-AAC-32	32	32	10.5
	MP4-AAC-48	48	44.1	11
SSC	SSC-16	16	44.1	18
	SSC-20	20	44.1	18
	SSC-24	24	44.1	18
Anchors and references	H-Ref-Orig	16-bit PCM	48	24
	H-Ref-3.5	16-bit PCM	48	3.5
	H-Ref-7	16-bit PCM	48	7.0

5 Test material

For both the mono and stereo MUSHRA tests the items listed in Table 3 were used. The items were selected from 53 potential candidate items by a selection panel at IRT. The applied pre-selection procedure is outlined in Annex B. The nine items listed in Table 3 were considered critical or have typical bahaviour for all of the systems under test.

Table 3 – The 9 selected items for the mono and stereo MUSHRA test

Item No.	filename	signal
1	sc02	Meistersinger
2	te02	Dulcimer
3	te07	Male Speaker
4	te10	Fireworks
5	te15	Carmen
6	te26	Orchestra
7	te40	Orchestra & Trumpet
8	te45	Chanson
9	te48	Eric Clapton

For the test the MP4-AAC reference bitstreams were provided by Fhg. The SSC encoded bitstreams, decoder and the decoded wav-file outputs, were provided by Philips.

The hidden references and anchors were provided by IRT. The bitstreams and decoded output were verified by IRT.

6 Test Center

Both the mono and the stereo tests were performed at Institut für Rundfunktechnik (IRT) Germany. The test equipment was a PC with SEAQ-MUSHRA-software of Canadian Research Company (CRC) Version 2.03, which fed a Stax-Lambda-Pro Headphone system through a high-quality D/A-converter.

7 Test Results

7.1 Introduction

The test was performed in two passes, beginning with the stereo selection. Each set was estimated to be done in 15 through 30 minutes, but few test subjects needed up to nearly 1 hour. Besides two anchors and one hidden reference there were 9 sliders (mono: 8) to assess each of the 9 different sound signals.

A statistical analysis was done on the listening test data. The following plots display the mean values (asterisk) and 95% confidence intervals (vertical tick) over all items for every coding scheme. Detailed plots of comparisons between codecs on a per-item basis, and comparisons between items on a per-codec-basis are given in Annex E.

7.2 Mono results

The plot in Figure 1 shows the mean result per codec of the subjective test of mono signals.

It shows that the SSC coder operating at 16 kb/s and 20 kb/s, indicated by labels SSC-16 and SSC-20, each have a mean subjective quality that is comparable to the mean quality of MPEG-4 AAC operating at 24 kb/s, indicated by label MP4-AAC-24, where “comparable” is in a statistical sense at the 95% significance level.

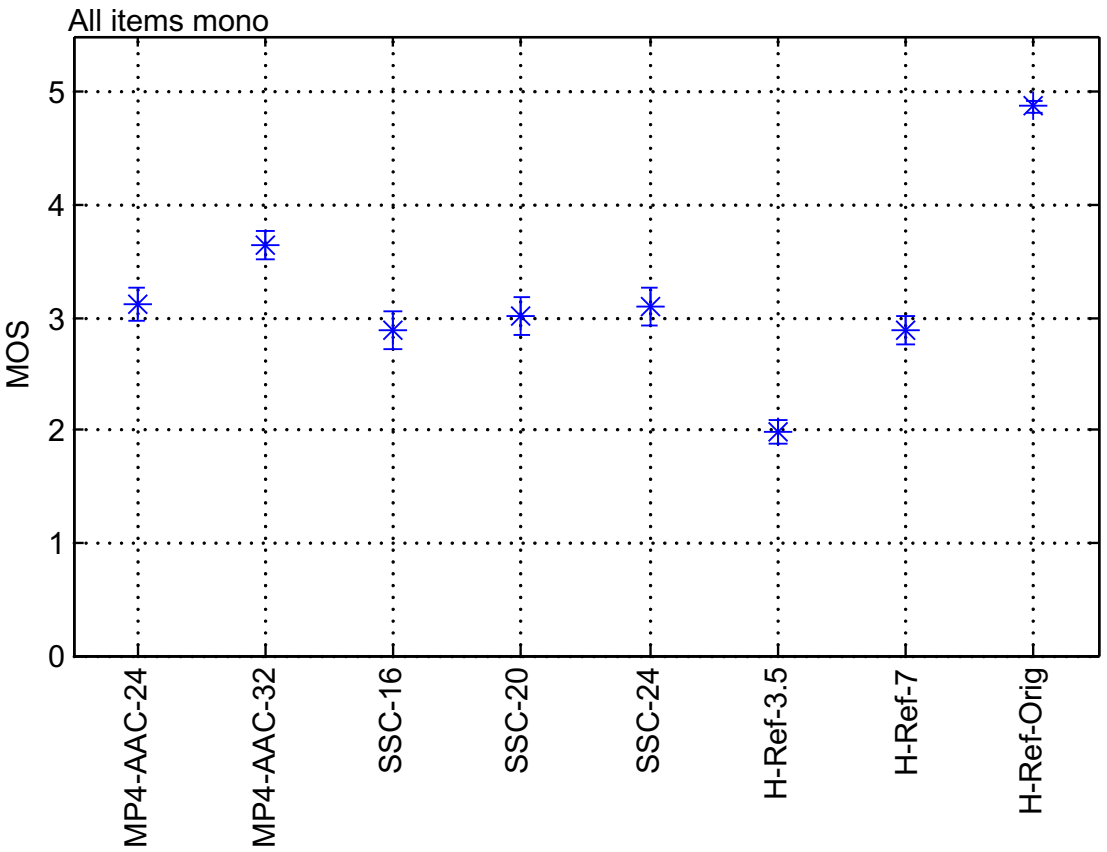


Figure 1 - MUSHRA Mono test results

Comparing the SSC technology operating at 24kbps with MPEG-4 AAC at 24kbps, it is evident from the plots given in Annex E, that for two items the new technology performs worse in a statistical sense than MPEG-4 AAC (i.e. the 95% confidence interval upper limit of these SSC items is less than the lower limit of MPEG-4 AAC). For the remaining items, the confidence intervals are at least overlapping.

7.3 Stereo results

The plot in Figure 2 shows the mean result per codec of the subjective test of stereo signals.

It shows that the SSC coder operating at 16 kb/s, indicated by label SSC-16, has a grand mean score that is higher than the grand mean score of MPEG-4 AAC operating at 24 kb/s, indicated by label MP4-AAC-24, where “better” is in a statistical sense at the 95% significance level.

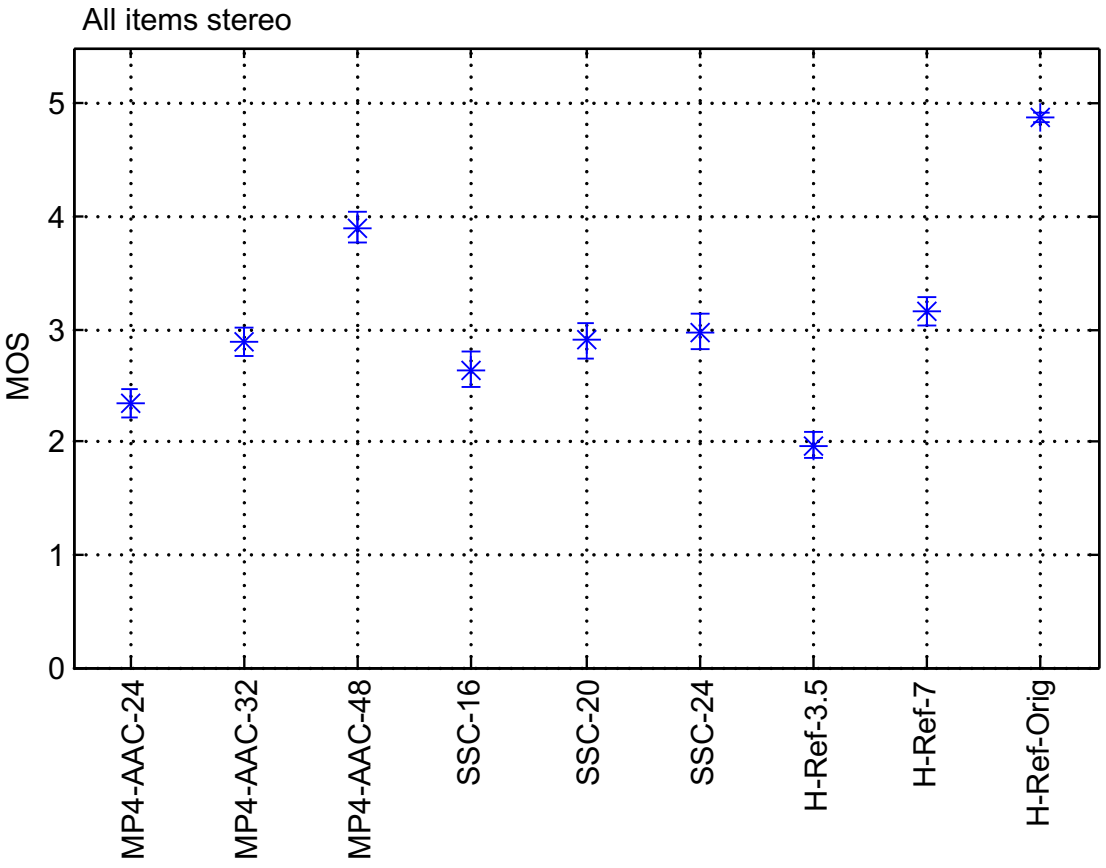


Figure 2 - MUSHRA stereo test results

When comparing the SSC operating at 24kbps, with MPEG-4 AAC at 24kbps, it is evident from the plots given in Annex E that for no item the new technology is worse in a statistical sense than MPEG-4 AAC (i.e. the 95% confidence interval upper limit of the SSC items is never less than the lower limit of MPEG-4 AAC).

7.4 Discussion

A noteworthy observation that can be made from the subjective test results of the parametric coder, is that for the same bit-rate, the subjective performance for mono and stereo is almost identical, see Figure 3. This illustrates the performance of the parametric stereo tool. For the same bitrate, the parametric stereo tool enhances the SSC coder operating in mono to stereo for almost the same quality. On several other occasions, this has also been demonstrated in combination with the MPEG4 HE-AAC core coder. As can also be inferred from Figure 3, for traditional waveform based audio coders such as AAC, when going from mono to stereo for the same bitrate a clear drop in quality is observed.

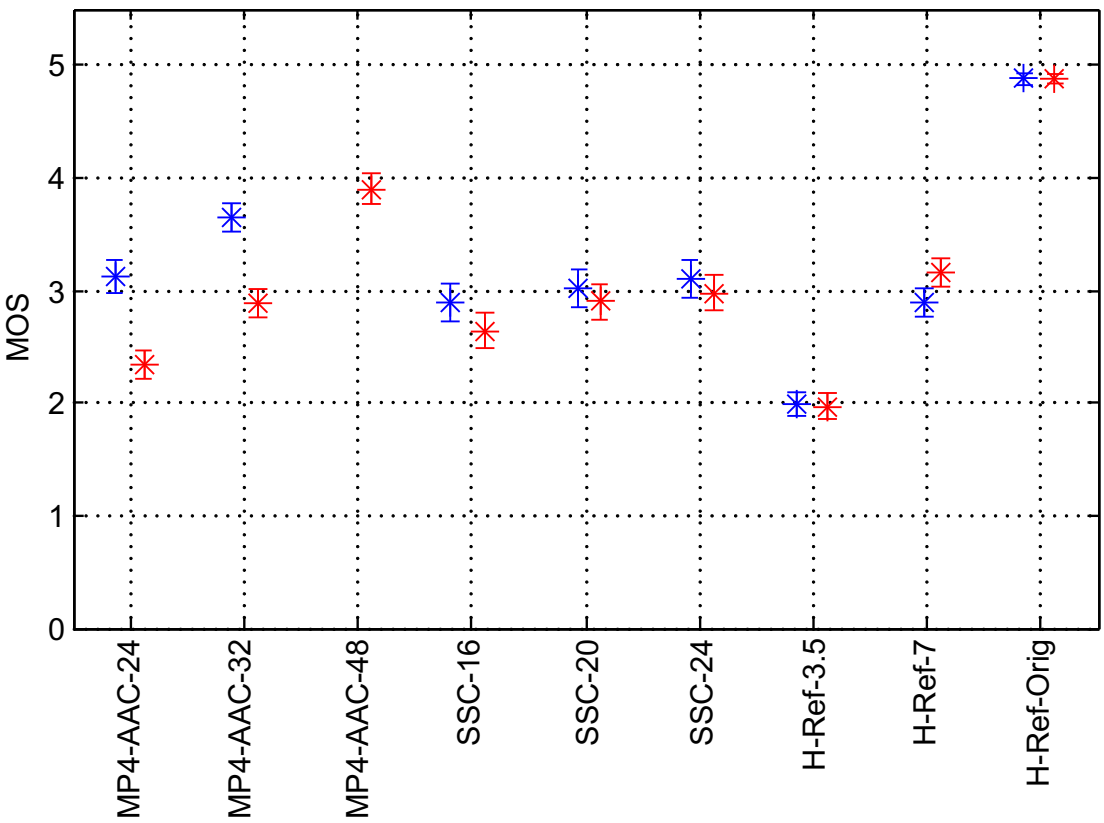


Figure 3 - Comparison mono and stereo results. Left hand scores represent mono (Blue) and right hand scores (Red) represent stereo results

For the Extension 2 parametric audio coder, when compared to MPEG4 AAC, the most distinctive quality discriminator is its intrinsic property to maintain full bandwidth (44.1kHz sampling frequency) at low bit-rates. As can be seen from Table 1 and Table 2, for these low bitrates, a traditional waveform coder operates at reduced sampling frequency in order to limit distortions. This results in band limitation at the decoded signal output. Consequently, for input material that is already severely band limited, the parametric coder is not able to display its full bandwidth feature for band-limited items.

The formal verification test results for mono show that for two items, te10 and te26, the parametric coder performs with overlapping confidence intervals, or worse, with respect to waveform coding at the same bit-rate. It is noted that the te26 item is band limited.

For illustration, the power spectral density functions of the excerpts in the formal verification test have been measured and an estimate of the bandwidth is listed in Table 4. This estimate for the bandwidth is defined as the lowest frequency beyond which the PSDF is 40dB less than its maximum. Note that the orchestra excerpt (te26) has by far the largest fall-off. For the orchestra excerpt (te26), only 3kHz bandwidth is measured.

Table 4 – The items under test with their respective bandwidth

Item index	Description	Bandwidth [kHz]
sc02	Meistersinger	7
te02	Dulcimer	12
te07	Male speaker	11
te10	Fireworks	11
te15	Carmen	15
te26	Orchestra	3
te40	Orchestra & trumpet	7
te45	Chanson	13
te48	Eric Clapton	18

Observing the absolute scores for te26 for AAC and SSC, this item shows not to be critical for both coders for all configurations tested. Particularly to the limited bandwidth of this item, the AAC score is higher than its average performance. As a result, a statistically significant, but not very relevant difference with respect to SSC is obtained for this item.

To illustrate typical behaviour of the MPEG4 SSC coder, in addition to the formal verification test, an informal subjective test has been conducted in order to get an impression on the subjective quality for all the items that were input to the pre-selection procedure, see Table 7 in Annex B. In a Differential Category Rating (DCR) test with 8 subjects, the performance of the parametric coder at 24kbps stereo has been compared with respect to AAC operating at 32kbps stereo. The results are provided in Figure 4.

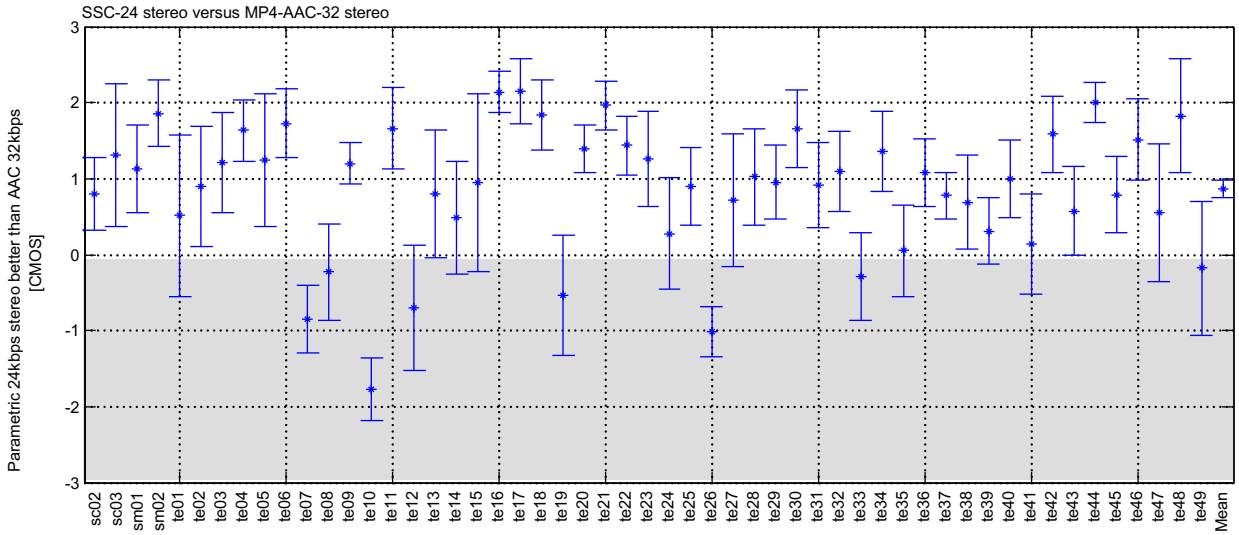


Figure 4 – DCR test comparing SSC at 24kbps stereo versus AAC at 32kbps stereo

Figure 4 shows that the parametric coder at 24 kbps performs statistically significantly better than AAC at a 33% higher bitrate of 32 kbps for 35 out of the 53 excerpts and with statistically overlapping intervals for 15 excerpts. Thus, the parametric coder operating at 24kbps stereo performs comparable to, or significantly better than AAC at 32kbps stereo for all except 3 excerpts. These 3 excerpts are all part of the set in the formal verification test, i.e. te07 (German male), te10 (fireworks) and te26 (orchestra), see Table 3. Combining the formal test results from Figure 2 and Figure 4, it is observed that at the same bit rate of 24kbps stereo, for SSC none of the 53 items input to the pre-selection test performs worse than AAC in a statistically significant sense.

8 Performance goals of standardization effort

At the start of the work item, the following performance goals have been set.

Using the MUSHRA test methodology, and with the target bit-rate for the SSC coder set to 24 kbit/s per channel for general audio signals, the Parametric technology shall satisfy the following two criteria.

1 With the SSC coder operating at the target bit-rate and MPEG-4 operating at 25% higher bit-rate, the SSC coder shall have a mean score that is comparable to or better than the mean score of MPEG-4, where comparisons are in a statistical sense at the 95% significance level.

2 With both coders operating at the target bit-rate, for no item shall the developed technology be worse in a statistical sense at the 95% significance level.

9 Conclusions

The verification tests all clearly show that the parametric technology (SSC) on average performs as well as MPEG-4 AAC Profile when the latter is operating at a 25% higher bitrate. The tests also show that for stereo

no item is the new technology worse than MPEG-4 AAC when both coders operate at the same bitrate. Therefore for stereo, the acceptance criteria outlined above have clearly been satisfied.

The verification test also demonstrates the strength of the standardized parametric stereo module, in that virtually the same quality level is obtained for stereo as for mono, at the same total bit rate.

10 Acknowledgements

The following persons are acknowledged for their contribution to the verification tests:

Table 5 - Acknowledgements

Name	Company
Schuyler Quackenbush	Audio Research Labs
Martin Link	IRT
Johannes Hilpert	FhG IIS-A
Bernhard Grill	FhG IIS-A
Andy Gerrits	Philips Digital Systems Laboratories

11 References

[1] Multi stimulus test with hidden reference and anchor (MUSHRA) – "EBU method for subjective listening tests of intermediate audio quality, Recommendation ITU-R BS". 1534 available at http://ecs.itu.ch

[2] Audio Subgroup - "Call for Proposals for New Tools for Audio Coding" ISO/IEC JTC1/SC29/WG11/N3793. January 2001

[3] David Meares et al. - "Report on the MPEG-2 AAC Stereo Verification Tests" ISO/IEC JTC1/SC29/WG11/N2006. February 1998.

[4] Frank Feige et al. – "Revised specifications of the MPEG-2 AAC Stereo Verification Tests", ISO/IEC JTC1/SC29/WG11/N1845. October1997

[7] Audio Subgroup - "Call for Evidence for New Tools for Audio Coding" ISO/IEC JTC1/SC29/WG11/N3641

[8] Audio Subgroup – "Workplan for Verification Tests of MPEG-4 Parametric Audio Coding" ISO/IEC JTC1/SC29/WG11/N6446

Annex A - Testing schedule for the verification test

Table 6 - Testing schedule for the verification test

Activity	Completion Date	Responsibility	Contact
Original items on ftp site	Fri Apr 2, 2004	Philips	W.Oomen
Coded test items on ftp site (6 weeks)	Mon May 24, 2004	Philips for SSC FhG for AAC	W.Oomen B.Grill
Pre-selection Test material (1 week)	Mon May 31, 2004	IRT	M.Link
Verification of decoding (1 week)	Mon May 31, 2004	IRT	M.Link
Listening test complete (4 weeks)	Mon Jun 28, 2004	IRT	M Link
Data analysis and Test Report complete (2 week)	Mon Jul 12, 2004	IRT	M.Link

Annex B – Preselection process

First a pre-selection of around 53 audio samples was considered for feasible material. It resulted from the items that were delivered together with appropriate decoder-software as encoded data streams as well as reference material. Then samples with significant "behaviour" were selected out of these. The choice relates to critical material to the codecs at their given bitrates and it should cover the variety of typical programme material. According to the agreed pre-selection, these items appeared to be a good representation for partially critical material to the codecs, but also representing usual programme material, instead of just being so-called "killer-signals".

For the pre-selection in the presence of original and encoded material, all items were mounted in a multi-track project of the audio production software "samplitude", with each track carrying one condition (i.e. codecs at designated bitrates, original). Based on the different (or similar) behaviours of the codecs, and the type of sound, it was attempted to find a complete coverage of all possible material. The item fireworks (te10) represents the genre of noise of radio-plays or movie-soundtracks, and outdoor atmosphere. The item orchestra (te26) and orchestra&trumpet (te40) possibly are two intervals in the same recording, but the different appearances of the musical presentation as well as the phrases in "pp" and "forte" have resulted in selecting both. As (primarily) solo instruments dulcimer (te02) and Carmen (te15) (second half) have been selected. Note that te26 and te40 also fall partly in this category.

Table 7 - Items input to the preselection

filename	signal
te01	Dorita
te02	We shall be happy
te03	Castanets
te04	Harpsichord
te05	Pitch Pipe
te06	Glockenspiel
te07	Male German Speech
te08	Suzanne Vega
te09	Tracy Chapman
te10	Fireworks
te11	Ornette Coleman
te12	Bass Synth
te13	Bass guitar
te14	Hyden Trumpet Concert
te15	Carmen
te16	Accordion/Triangle
te17	Tambourine
te18	Percussion
te19	Male speech
te20	George Duke
te21	Asa Jinder

te22	Dire Straits
te23	Dalarnas Spelmansforbund
te24	Stefan Nilsson
te25	Stravinsky
te26	Ravel
te27	Triangles
te28	Clay
te29	spiral wave
te30	"aimai"
te31	ether
te32	Palmtop boogie
te33	<CROISEMENT I> pour hautbois, violon et contrebasse
te34	drifting
te35	dramatics
te36	O1
te37	Fourth
te38	Interlude by Halves for violin, flute and piano
te39	accellation
te40	atmosphere
te41	fanfare
te42	Kids Drive Dance(KDD)
te43	Bass clarinet
te44	Bransle
te45	Brel
te46	Guitar + Castanets
te47	Fools
te48	Layla
te49	Music Rain
sc02	Orchestra
sc03	Contemporary Pop
sm01	Bagpipe
sm02	Glockenspiel

Annex C - Test methodology

A.1.1 MUSHRA

Several of the tests in this report used the MUSHRA method described in [1]. EBU Project Group B/AIM developed this in 1999, in collaboration with ITU-R Working Party 6Q. An important feature of this method is the inclusion of the hidden reference and two bandwidth limited anchor signals (7 kHz and 3.5 kHz).

MUSHRA prescribes a quality scale where the intervals are labeled "bad", "poor", "fair", good" and "excellent" as opposed to BS.1116. The value on the lower end of the scale is zero, the value on the upper end is 100. No decimals are given. This scale has the advantage to be harmonized with video quality.

In the preparation of the test, the test-engineer preferred to use the BS.1116 scale from Table 8 instead of the 100 point MUSHRA scale. The rationale for this choice is that voting is easier when you are used to a scaling range. In this case scores from 1 to 5 seemed to be easier as this corresponds to grades that are given at German (high) schools. The number of descriptors for the BS.1116 scale is the same as used on the 100 point MUSHRA scale.

Table 8 – BS.1116 grading scale

Impairment	Grade
Imperceptible	5.0
Perceptible, but not annoying	4.0
Slightly annoying	3.0
Annoying	2.0
Very annoying	1.0

The length of the sequences did not exceed 20 seconds to avoid fatiguing listeners and to reduce the total duration of the listening test.

A.1.1.1.1 Training phase

In order to get reliable results, it was mandatory to train the subjects in special training sessions in advance of the test. Usually the subjects are not familiar with the given test procedure as well as with the kind and nature of quality impairments. Thus a training phase must precede the actual test. The behavior of acoustical presentation, just by clicking the buttons and moving the sliders, always takes several minutes of acquisition time to people who never before were in touch with such kind of listening tests.

In the beginning a short explanation was given to each subject, concerning

- Switching through signals of different audio qualities
- Presence of a 'hidden reference' (At least one vote at the highest grade is mandatory)
- Same set of codecs on each item by random allocation
- Meaning of the quality scale tags
- Varying quality impressions by different items

Specially the meaning of the lowest grade was emphasized, i.e. the range for voting must stay where it was heard, and it must not be expanded to the lower end by force.

In most cases the subject just had to switch through all qualities of the first item in the set, as all subjects were from IRT and they were familiar with that kind of listening tests. On request a more detailed explanation together with trials was given to those who are beginners.

After the training phase the subject was left alone, but he could call in case of any further problems. Actually no questions on the test arose after the initial explanations.

A.1.1.1.2 User - interface

Compared to ITU-R BS.1116, the MUSHRA method has the advantage of displaying all stimuli (conditions) for one test item. The subjects were therefore able to carry out any comparison between them directly.

The whole test was divided in two sessions, each containing only one type of conditions (mono or stereo). Figure 5 below shows the user-interface presenting one item under test. The buttons represent the reference, which is specially displayed on bottom left, and all the codecs under test, including the hidden reference and both anchor points (band-limited processed reference), called test items. Above each button, with the exception of the button for the reference, a slider was used to grade the quality of the test item according to the continuous quality scale used. For each of the items, the signals under test were randomly assigned. In addition, the test items were randomized for each subject within a session. To avoid sequential effects, each subject was running the two sessions in randomized order.

None of the subjects had the same items order and the same order in the conditions presentation.

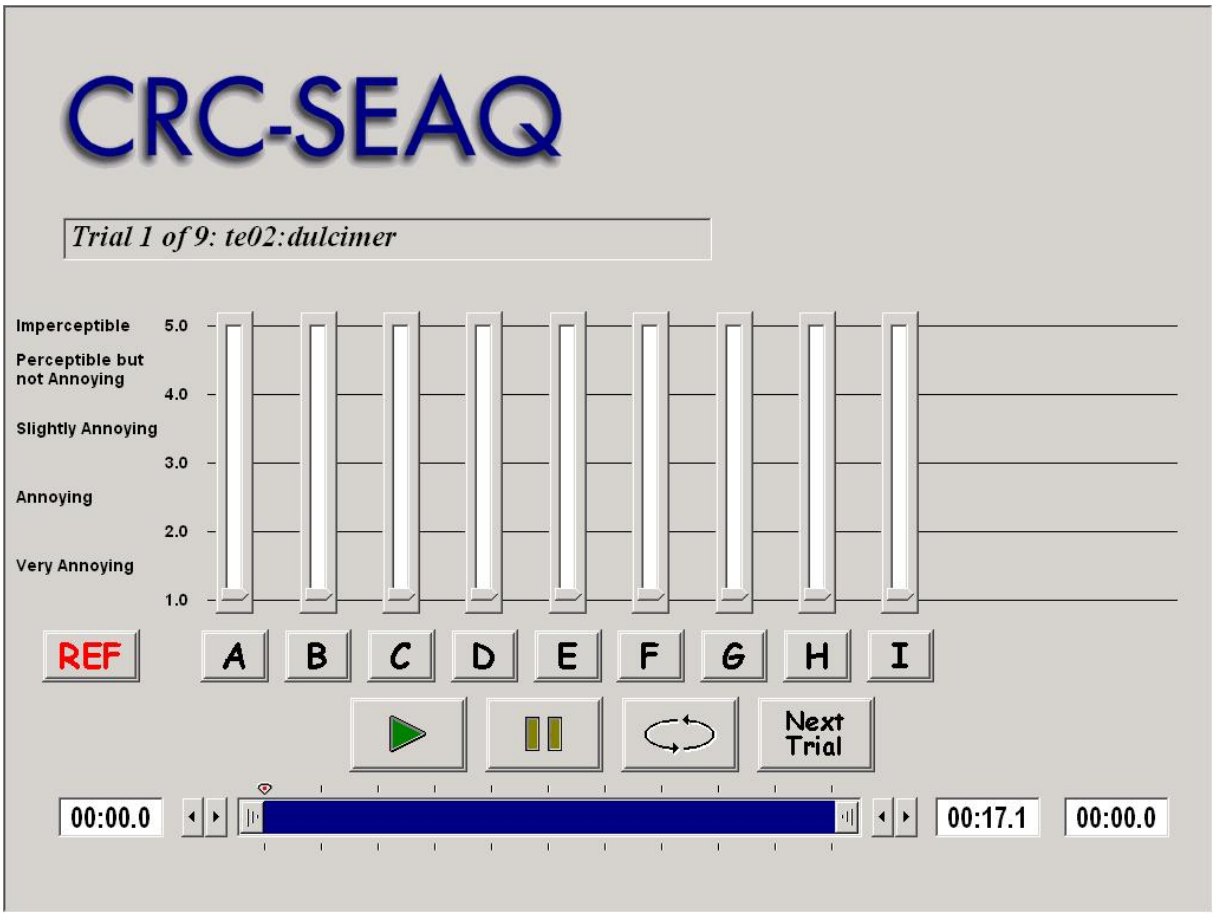


Figure 5: User interface for MUSHRA tests

Annex D - Statistical Analysis

A.1.2 General

The statistical analysis followed standard MUSHRA procedure. The calculation of the averages of the scores of all listeners remaining after post-screening will result in the Mean Subjective Scores (MSS). The first step of the analysis of the results is the calculation of the mean score $\bar{u}_{jk}$, for each of the presentations:

$$\bar{u}_{jk} = \frac{1}{N}\sum_{i=1}^{N} u_{ijk} ,$$

where u_i is the score of observer i for a given test condition j and sequence k 1… N is the number of observers. Confidence intervals were also calculated which was derived from the standard deviation and the size of each sample. The 95% confidence interval is given by:

$$\left[\bar{u}_{jk} - \delta_{jk}, \bar{u}_{jk} + \delta_{jk}\right] ,$$

where:

$$\delta_{jk} = 1.96\frac{S_{jkl}}{\sqrt{N}},$$

and the standard deviation S_{jk} is given by: $S_{jk} = \sqrt{\sum_{i=1}^{N}\frac{\left(\bar{u}_{jk} - u_{ijk}\right)^2}{\left(N-1\right)}}$.

With a probability of 95%, the absolute value of the difference between the experimental mean score and the “true” mean score (for a very high number of observers) is smaller than the 95% confidence interval, on condition that the distribution of the individual scores meets certain requirements.

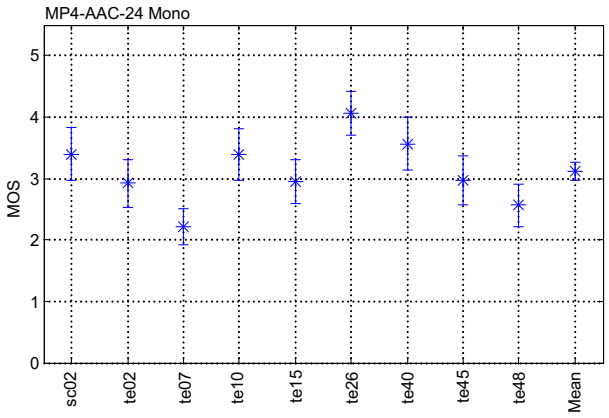
Similarly, a standard deviation could be calculated for each test condition. It is noted however that this standard deviation will, in cases where a small number of test sequences are used, be influenced more by differences between the test sequences used than by variations between the assessors participating in the assessment.

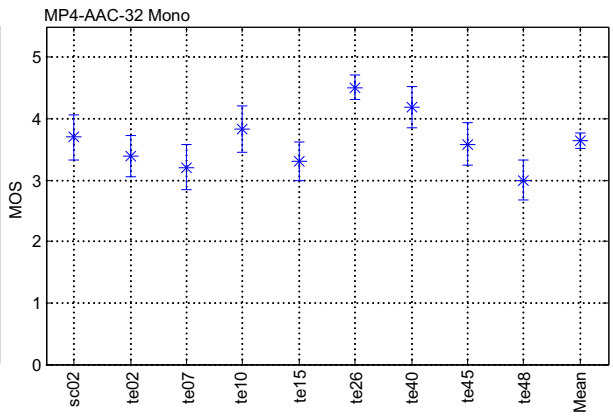
Annex E - Detailed results

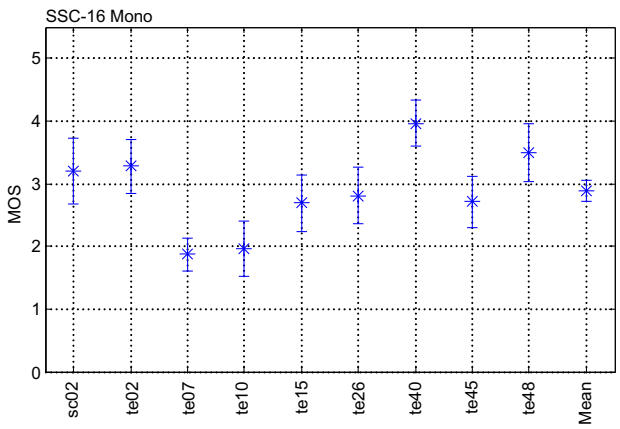
The trailing number at the MP4-AAC or SSC labels designates the bitrate in units of kbit/s. At 48 kBit/s only the AAC - stereo version was available. The label at the left top corner indicates the coder, item or configuration (mono/stereo) for which the results are shown. The voting of the anchor signals with band limitations of 3.5 khz or 7 khz influences the relationship of results of former listening tests. These results are also shown together with the polling of the hidden reference signal, which proves the reliability of the results.

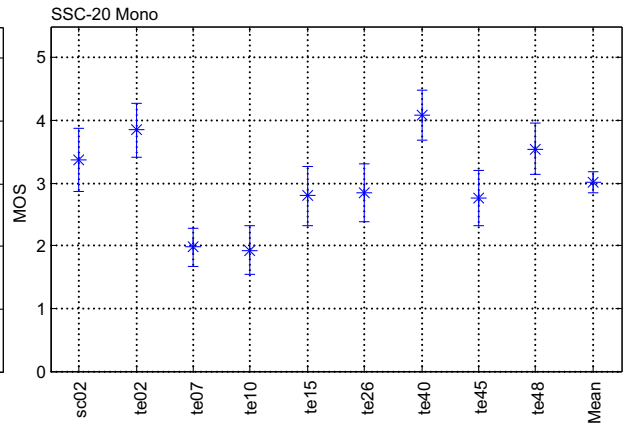
E.1 Mono Items

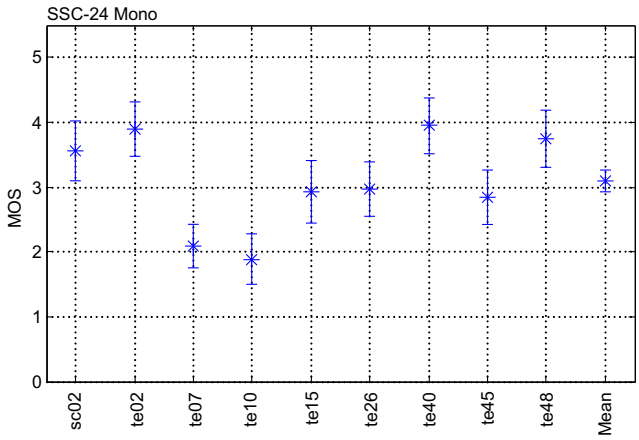
E.1.1 Results per coder

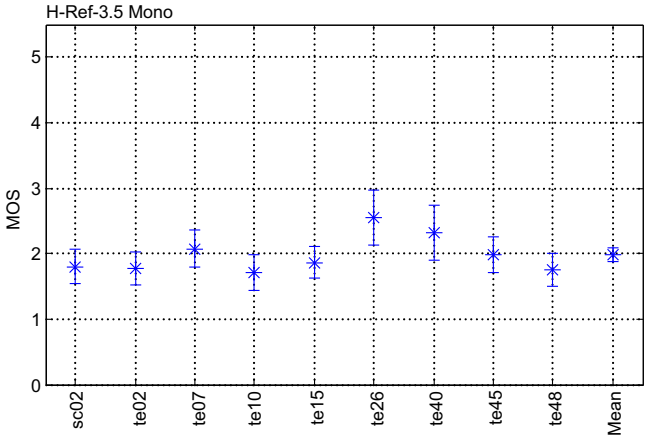


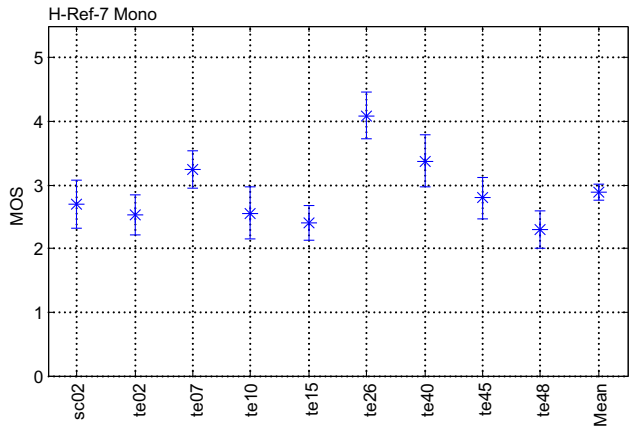


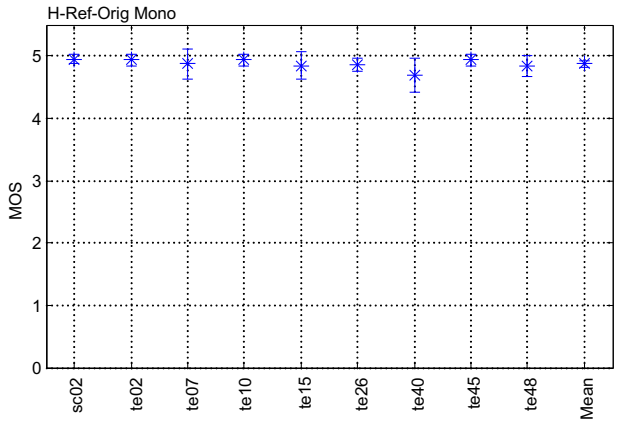




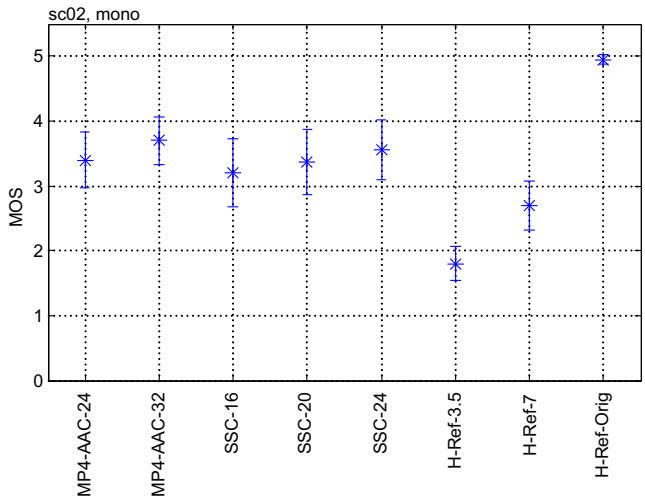


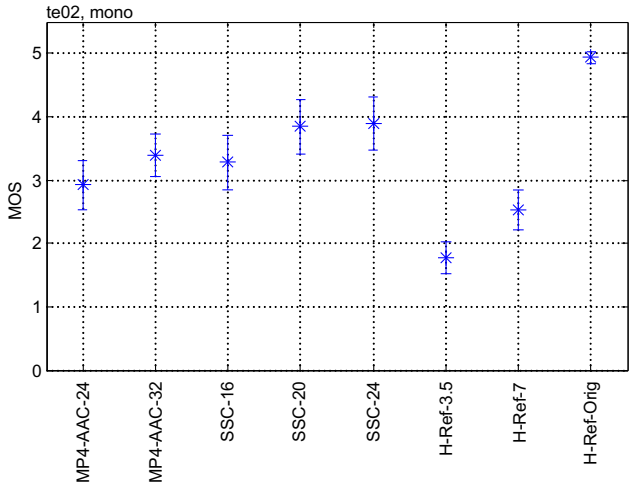


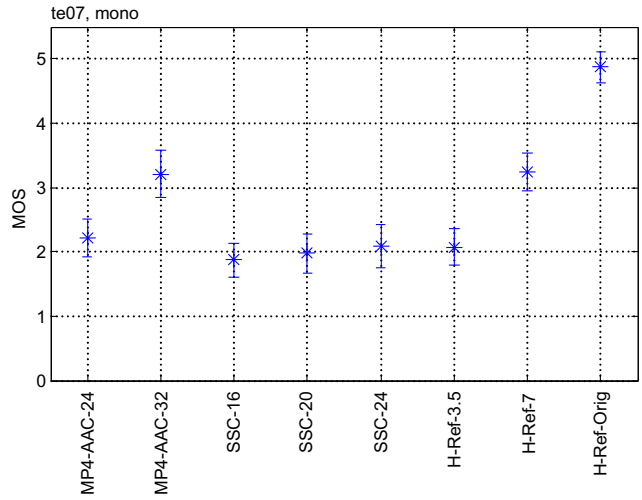


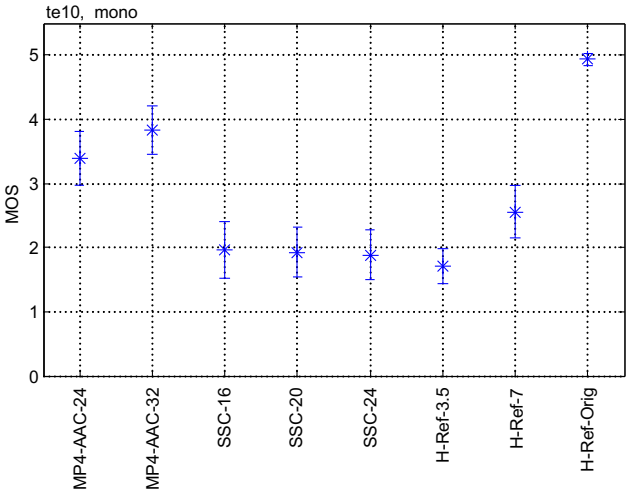


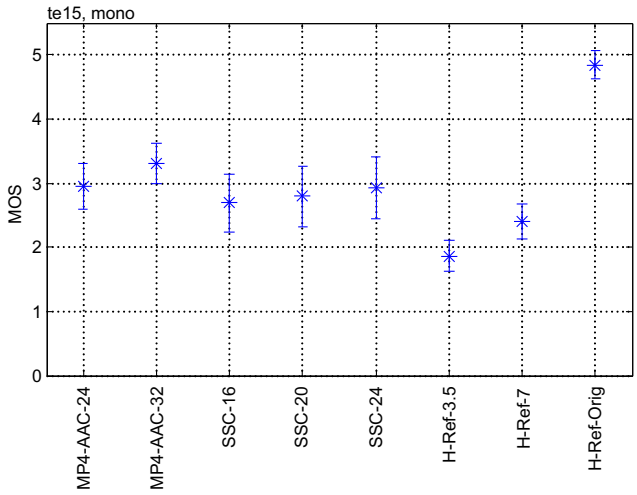
E.1.2 Results per item







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SSC-20
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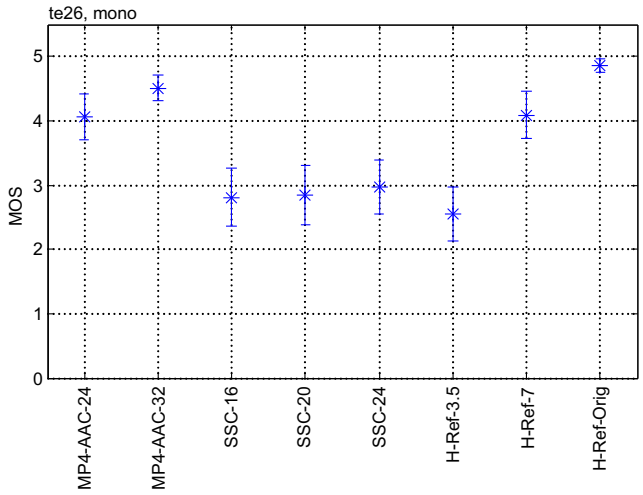
H-Ref-7
H-Ref-Orig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MOS
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SSC-20
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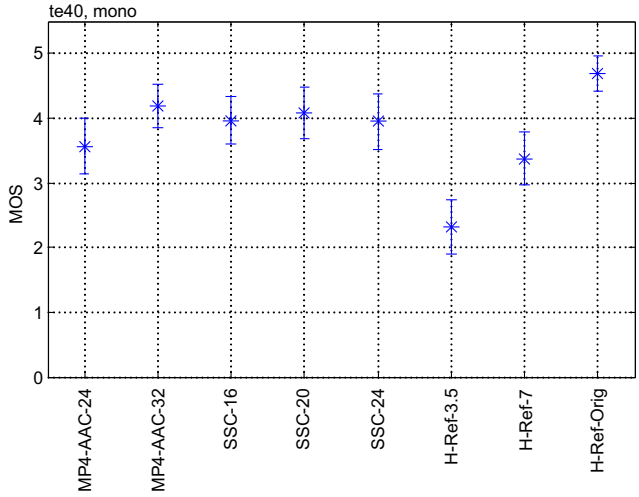
H-Ref-3.5
H-Ref-7
H-Ref-Orig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MP4-AAC-24
MP4-AAC-32
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SSC-20
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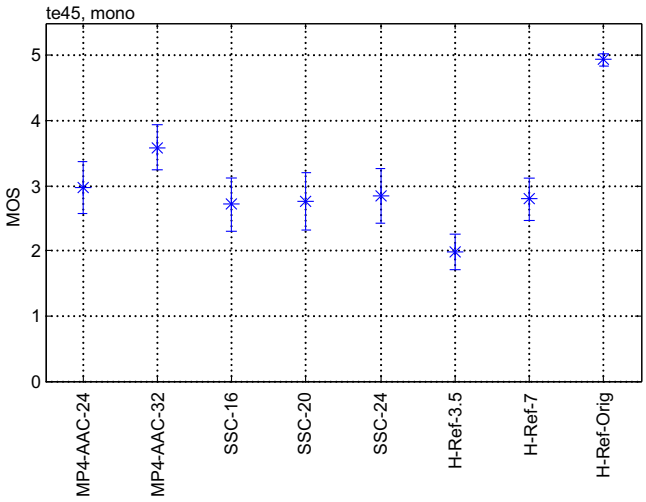
H-Ref-7
H-Ref-Orig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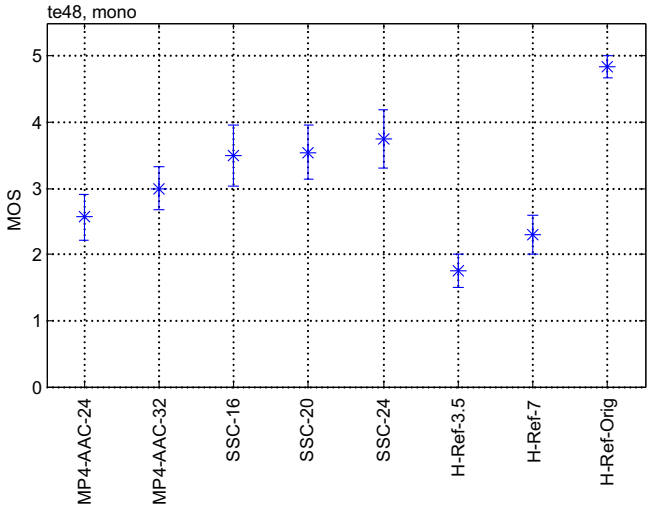
SSC-20
SSC-24
H-Ref-3.5
H-Ref-7
H-Ref-Orig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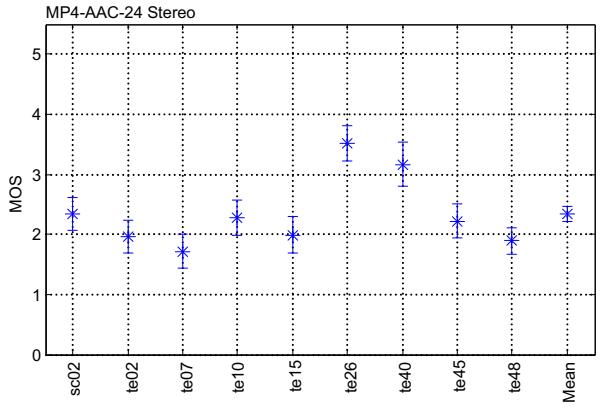
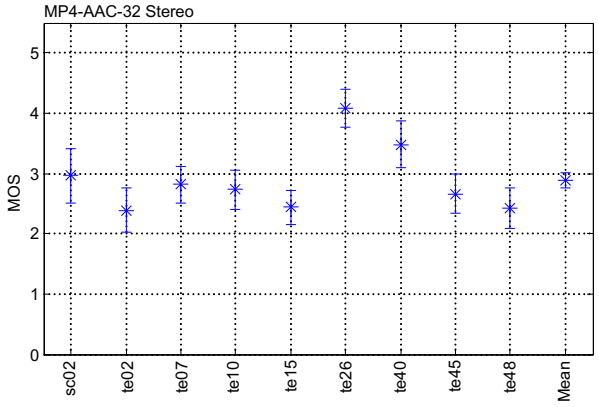
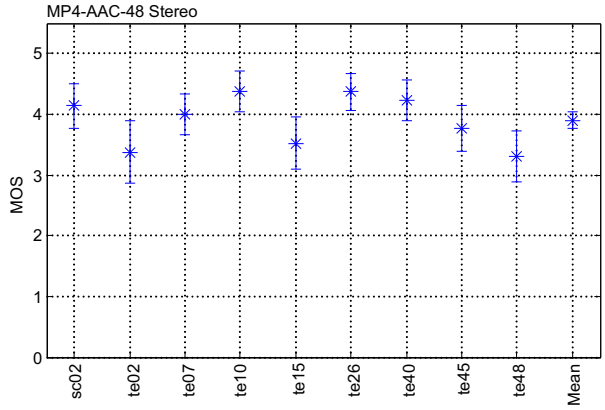
MP4-AAC-24
MP4-AAC-32
SSC-16
SSC-20
SSC-24
H-Ref-3.5
H-Ref-7
H-Ref-Orig

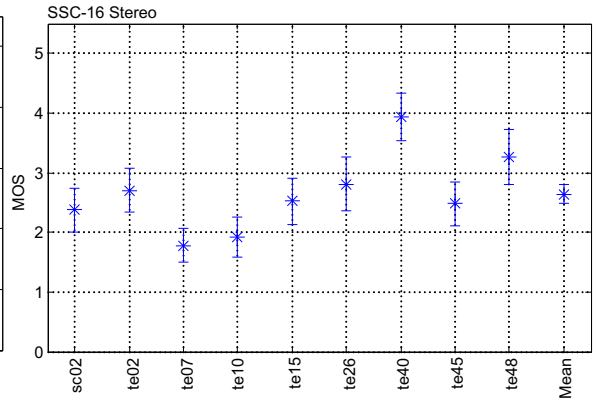
E.2 Stereo Items

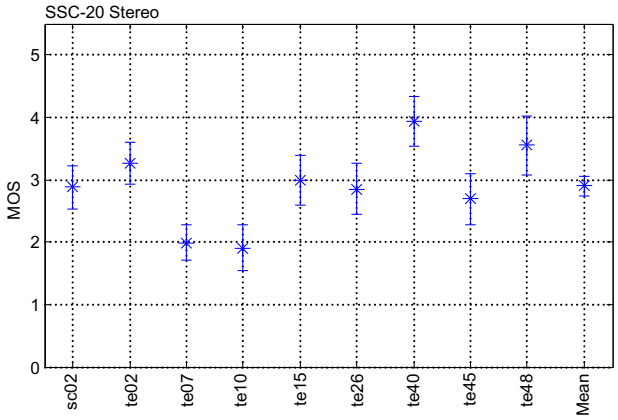
E.2.1 Results per coder

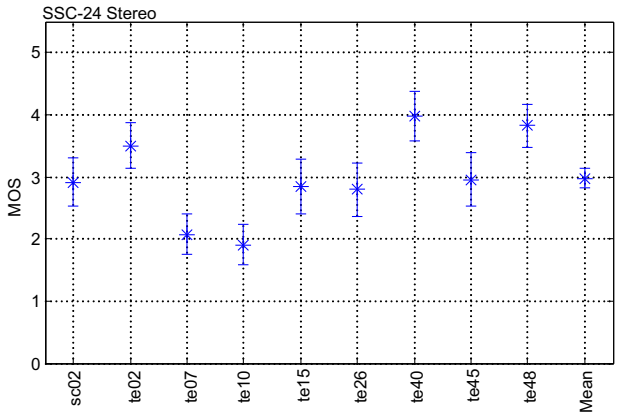


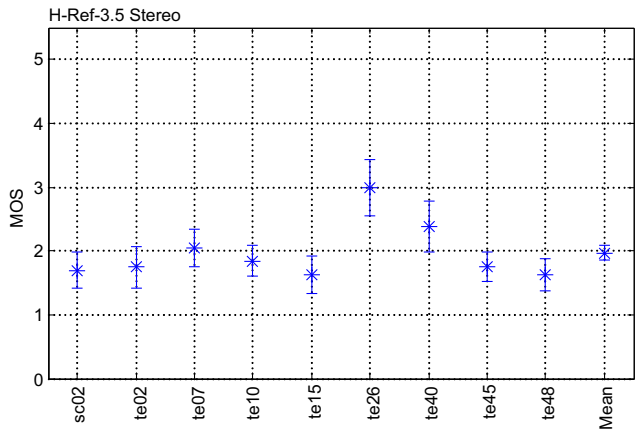


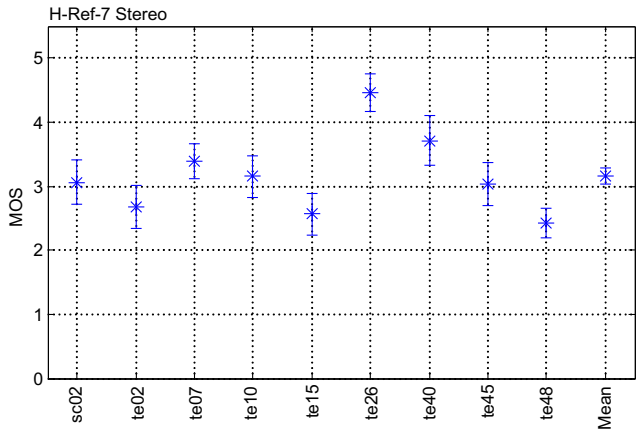


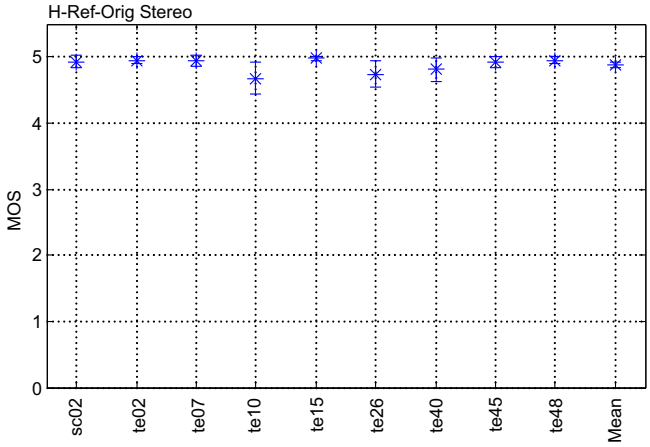




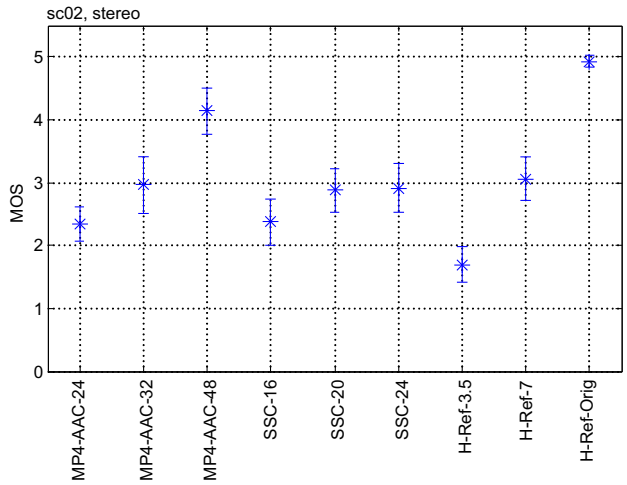


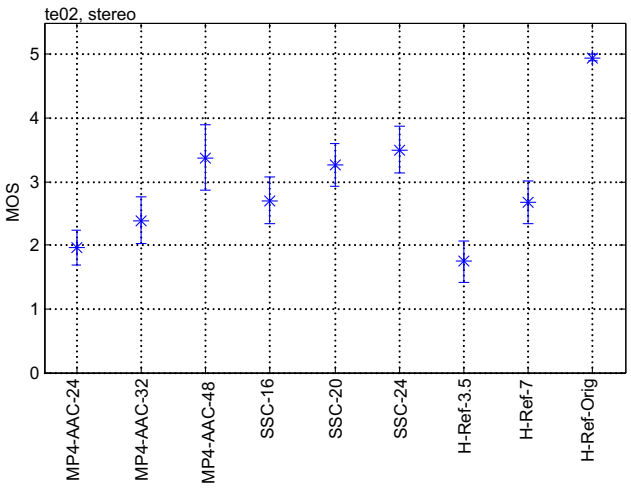


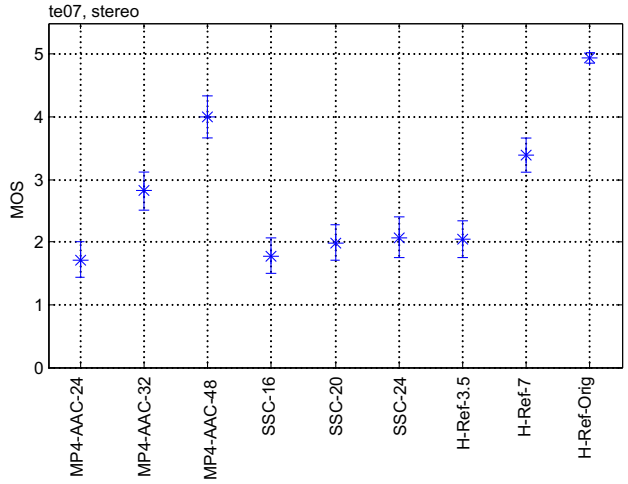




E.2.2 Results per item



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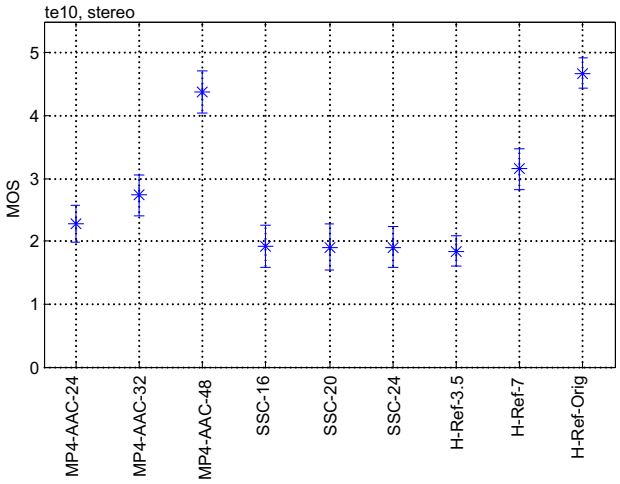
MP4-AAC-24
MP4-AAC-32
MP4-AAC-48
SSC-16
SSC-20
SSC-24
H-Ref-3.5
H-Ref-7
H-Ref-Orig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MP4-AAC-24
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SSC-24
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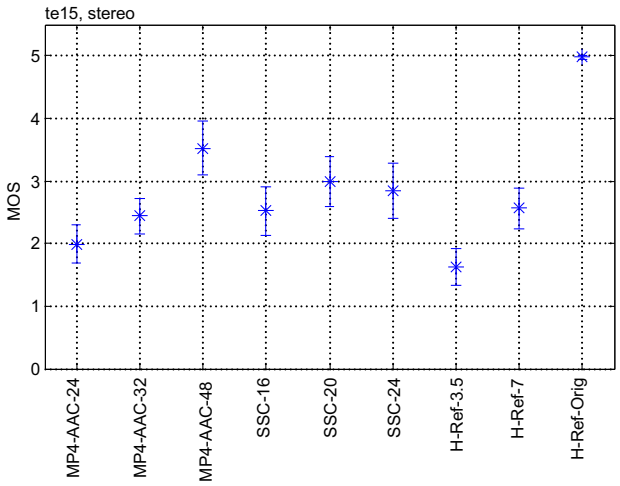
H-Ref-Orig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SSC-20
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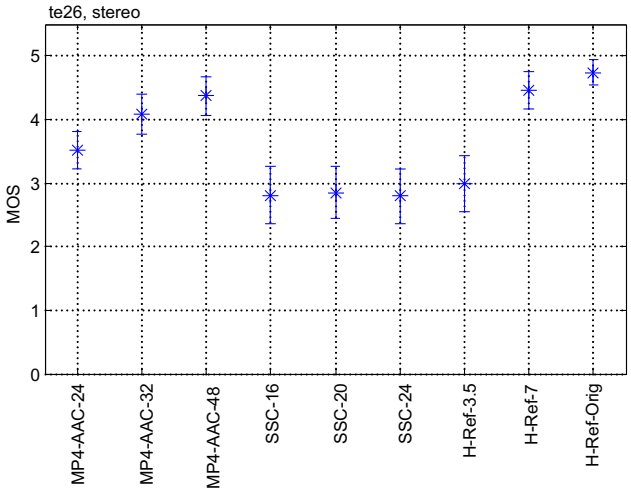
H-Ref-Orig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H-Ref-3.5
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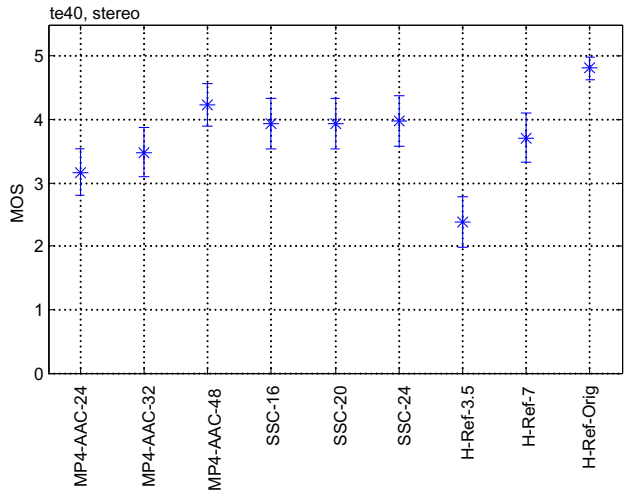
H-Ref-Orig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MP4-AAC-24
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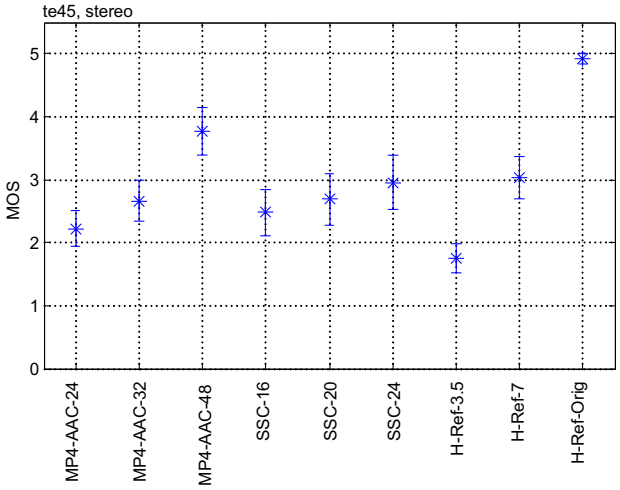
H-Ref-Orig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SSC-20
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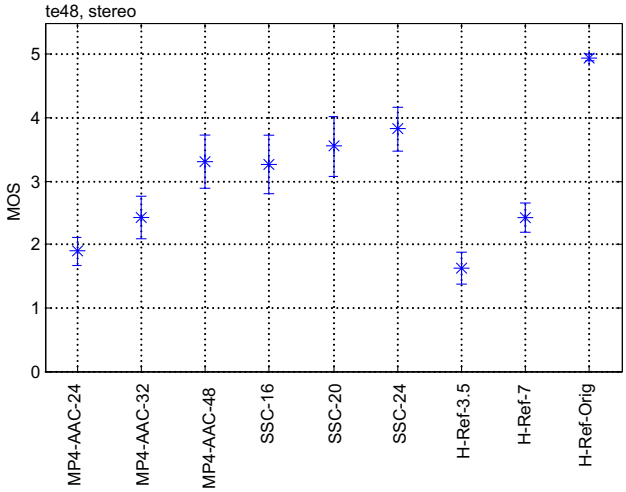
H-Ref-Orig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MP4-AAC-48
SSC-16
SSC-20
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H-Ref-7
H-Ref-Orig