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Bitrate Requirements for Opus with First, Second and Third Order Ambisonics reproduced in 5.1 and 7.1.4

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ABSTRACT

In this paper, we present a study on the Basic Audio Quality of first, second and third order native Ambisonics recordings compressed with the Opus audio codec at 24, 32 and 48 kbps bitrates per channel. Specifically, we present subjective test results for Ambisonics in Opus decoded to ITU-R BS.2051-2 [1] speaker layouts (viz., 5.1 and 7.1.4) using IEM AllRAD decoder [2]. Results revealed that a bitrate of 48 kbps/channel is transparent for Basic Audio Quality for second and third order Ambisonics, while larger bitrates are required for first order Ambisonics.

1 Introduction

Ambisonics is a full-sphere surround sound technique for representation of sound fields in terms of orthogonal basis functions, known as spherical harmonics [3]. The maximum order of the harmonic functions determines the channels number, the spatial reproduction granularity, and the required reproduction loudspeakers density[4].[1] MPEG-H 3D Audio has been offering support for Ambisonics as scene-based audio as part of ATSC 3.0 for television broadcasting since 2015 [5, 6]. Ambisonics also has an important role in gaming and Metaverse applications of Virtual Reality [7, 8].

Opus is a royalty-free open-source perceptual codec [9, 10, 11] supporting first and higher-order Ambisonics. RFC 6716 defines the Opus codec as a multi-purpose audio codec, capable of automatic alternation between SILK and CELT modes for speech and music, respectively [12]. Opus can encode and decode audio streams either as mono or as joint stereo to achieve higher quality at a lower bitrate. The performance of the Opus codec has been established for mono and stereo audio [9, 10, 11, 13, 14] and multi-channel surround [15, 16, 17].

RFC 7845 defines the Opus encapsulation in the Ogg container for the mono, stereo and multi-channel surround cases in mapping family 1 [18]. RFC 8486 extends said encapsulation to first and higher order Ambisonics [19] introducing two additional mapping families:

- Mapping family 2 (multi-mono coding, each audio channel being independently encoded as a mono

[1] xOA represents x-th order Ambisonics

stream), recommended for first order Ambisonics [19].

- Mapping family 3 (fixed channel matrixing before and after multi-stereo coding), recommended for higher order Ambisonics [19, 20, 21].

Previous work on the performance of the Opus codec with Ambisonic audio has evaluated the Listening Quality (LQ) and Localization Accuracy (LA) of Opus-encoded first and higher Ambisonics both in mapping family 2 [22, 23] and 3 [24, 25].

In this study, the Basic Audio Quality (BAQ) of first, second and third order Ambisonics mixes:

- compressed with the Opus audio codec at 24kbps, 32 kbps, and 48 kbps per channel, and
- reproduced over 5.1 and 7.1.4 speakers layouts,

is investigated, with the purpose of determining a transparency bitrate for the reproduction of Opus-encoded Ambisonics rendered to commercial speakers layouts.

Key differences with prior work [22, 23], [24, 25] include: (a) decoding to ITU-R BS.2051-2 layouts 5.1 and 7.1.4 as opposed to binaural or spherical array decoding, (b) including second order Ambisonics, and (c) using native Ambisonic mixes and recordings.

2 Preliminary observations

Informal preliminary listening sessions involving a small panel of trained listeners were conducted to specify the scope of the research. The following conditions were investigated:

- Ambisonics encoders and decoders
  - Higher order Ambisonics toolbox in Matlab from Archontis Politis [26].
  - IEM All-Round Ambisonics Decoder (AllRAD) [2].
  - O3A plugin from Blue Ripple [27, 28].
- Opus mapping families 2 and 3
- Compression bitrates - 16, 24, 32, 48 and 64 kbps per channel

From these preliminary pilot sessions, we

- fixed the choice of the decoder towards IEM AllRAD,
- fixed the preference of mapping family 2 for 1OA and 2OA, and mapping family 3 for 3OA,
- estimated a bitrate for perceptual transparency between 24 and 48 kbps per channel for first, second and third order Ambisonics.

In addition, discussions between the experimenter and the pilot assessor pool suggested that the compression introduced audible artefacts in the reverberation part of the test samples, which lead to the inclusion of recordings of highly reverberant spaces in the test sample.

3 Methods

3.1 Setup

Listening tests took place in a 7 m (L) x 5.33 m (W) x 3.05 m (H) listening room (Figure 1) equipped with 11 loudspeakers and 4 subwoofers supporting 5.1 and 7.1.4 channel-based playback. The loudspeakers' and listener's positions were based on ITU-R BS.2051-2 [1] standard. Loudspeakers were level-matched at the listeners's position and met the ITU-R BS. 1116-3 [29] specification in terms of room response curve within the 50 Hz -16 kHz frequency range.

Fig. 1: Listening room

A tablet interface using Max/MSP (Figure 2) provided the assessors with the control of playback, selection, rating and comments of the test signals. A segment looping function enabled the assessors to focus on artefacts in restricted sections of the audio clips. The listening test software was implemented using a customised Max/MSP program to achieve a double-blind system, allowing for randomization of audio samples playback order and testing items mapping order on each trial.

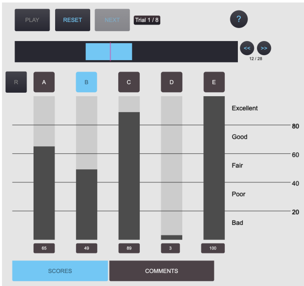


Fig. 2: Test user interface

3.2 Design

A listening experiment was conducted to establish a transparency bitrate for compressed Ambisonics with the following parameters:

- the first three Ambisonics orders, labeled 1OA, 2OA and 3OA,
- 5.1 and 7.1.4 speaker layout,
- 3 treatments, including the total bitrates displayed in Table 3.

The experiment was divided into 6 independent sessions - one for each Ambisonics order and speaker layout combination, as shown in Table 1. The order of the sessions was manually distributed across the participants.

A familiarization session preceding the grading process enabled the participants to navigate through labeled test content. The familiarization included, for each listening test sample: the reference signal, the low anchor and the 24 kbps/channel compressed signal.

Table 1: Test sessions breakdown

	5.1	7.1.4
1OA	Session 1	Session 2
2OA	Session 2	Session 3
3OA	Session 5	Session 6

Each session consisted of 8 MUSHRA-inspired trials [30], where assessors were

- oriented towards focusing on both localization cues and timbre impairments,
- asked to rate the BAQ on a 0-100 scale for each of the test signals (3 compressed signals (Table 3), low anchor and hidden reference),
- and invited to leave comments directly in the test interface.

The reference signal was the original uncompressed Ambisonics file, decoded for the corresponding speaker layout reproduction. The low-anchor was a 3.5 kHz low-pass filtered version of this reference signal [30]. The other signals were the treatments under test as shown in Table 3.

3.3 Listening test content

Each test session was run with 4 audio samples with 1 replicate, distributed in 8 trials, including:

- a reverberant audio clip - later labeled as "Ambient",
- a music audio clip - labeled as "Music",
- a male speech audio clip - labeled as "Speech",
- a sport game audience recording - labeled as "Sport".

Apart from the speech track, which content was spatially panned, the listening test samples were obtained using native Ambisonics recordings:

- first order Ambisonics recordings for 1OA test sessions,
- third order Ambisonics recordings for 3OA test sessions,
- the same third order Ambisonics recordings truncated to second order for 2OA test sessions.

3.4 Audio processing

An Ambisonics-enabled command-line interface opus-tools 0.2 (Opus 1.3) was used to encode 1OA and 2OA to mapping family 2 and 3OA to mapping family 3. After compression, the test tracks were decoded back to WAV for loudspeaker rendering. The IEM plug-in implementation of the All-Round Ambisonics Decoder (AllRAD) [2] was selected for 5.1 and 7.1.4 rendering. Finally, EBU R 128 Loudness normalization [34] was applied. Audio clips were also truncated to sections shorter than 20 seconds.

Table 2: Listening material sources

Label	Source	Content
Ambient	1OA, AmbiXes [31]	Foot steps, bells and distant voices in a reverberant church
Ambient	3OA, Munk Productions [32]	Foot steps and water dripping in a reverberant cave
Music	1OA, Soundfield by Rode [33]	String quartet
Music	3OA, Google provided	Voice, guitar and percussion music track
Speech	4OA, Google provided	English male speech
Sport	1OA, Soundfield by Rode [33]	American football game audience
Sport	3OA, mh acoustics	American football game audience

4 Results

4.1 Assessors

A total of 17 listeners participated to the experiment, including 15 audio professionals and two untrained listeners, with an average of 13 participants per session.

The assessor panel was subjected to a post-screening process in accordance with the ITU-R BS.1534 [30] MUSHRA standard, where assessors' ability to spot the hidden reference was verified. The set of results was also scanned for outliers.

The eGauge Method for Assessor Screening ([35], [36]) for evaluation of agreement, reliability and discrimination was then applied to the dataset. This method gives a measure for assessors' discrimination and reliability skills, along with an evaluation of the agreement with the panel. Assessors whose reliability or discrimination is estimated above the permutation level (the horizontal line in the plots in Tables 8 and 7) can be considered as experienced for the purposes of the test. Results in the appendix show that the panels of assessors were well-suited for each test (Cf Tables 9, 6).

Table 3: Treatments under test, in kbps (total bitrate)

Bitrate per channel	1OA	2OA	3OA
24	96	216	384
32	128	288	512
48	192	432	768

4.2 Analysis method

The test sessions were analysed separately as independent tests. As the data sets did not meet the requirements for ANOVA analysis[2], data were analysed with a robust two-way ANOVA for trimmed means [37] to assess the effects of the treatment and sample factors on the BAQ ratings. It should be noted that as this method relies on an adjusted critical value, no degrees of freedom are reported.

Post-hoc paired comparisons were then applied, using a similar method, focusing on the following pairs of conditions:

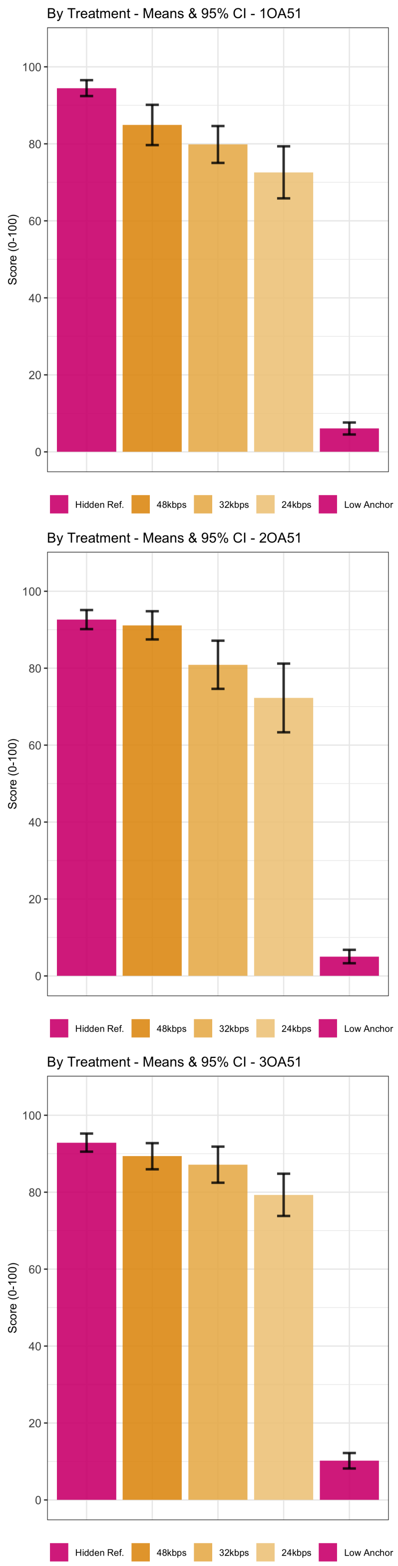
- Hidden reference vs 24 kbps/channel,
- Hidden reference vs 32 kbps/channel,
- Hidden reference vs 48 kbps/channel.

In the case of a significant effect of the sample or interaction between treatment and sample, similar one-way tests were applied to subsets of the data, grouped by audio sample.

In the following sections, $p_{bitrate}$ refers to the p-value associated with the post-hoc comparison between the ratings given to the hidden reference and to the bitrate in question.

[2] For each test, Levene's and Shapiro-Wilk's tests were significant, indicating a violation of the assumptions of homogeneity of variance and of normal distribution of the data.

Table 4: Overall 5.1 tests results, by test session



4.3 Session 1 - 1OA 5.1

ANOVA results revealed that the effect of the treatment was significant, with $F = 5426.6$, $p < .001$.

The post-hoc test for bitrates comparisons showed that all the compression rates under test were rated significantly lower than the reference (respectively $p_{48kbps} < 0.05$; $p_{32kbps} < 0.0001$; $p_{24kbps} < 0.0001$).

4.4 Session 2 - 1OA 7.1.4

For this session, the effect of the treatment was also found significant, with $F = 6571$, $p < 0.001$, along with a significant effect of the interaction between the treatment and the audio sample ($F = 36.27$, $p < 0.05$). Each tested bitrate scored statistically significantly lower than the reference (respectively $p_{48kbps} < 0.001$; $p_{32kbps} < 0.05$; $p_{24kbps} < 0.0001$).

Listeners were not able to discriminate the test bitrates and hidden references for the speech and sport tracks ($p_{48kbps,speech} = 0.99$, $p_{48kbps,sport} = 0.16$; $p_{32kbps,speech} = 0.053$, $p_{32kbps,sport} = 0.23$; $p_{24kbps,speech} = 0.11$, $p_{24kbps,sport} = 0.14$) compared with the other test samples. No transparency bitrate was found for the ambient sample ($p_{48kbps,ambient} < 0.0001$), while only 48 kbps/channel showed no significant difference to the reference for the music sample ($p_{48kbps,music} = 0.32$).

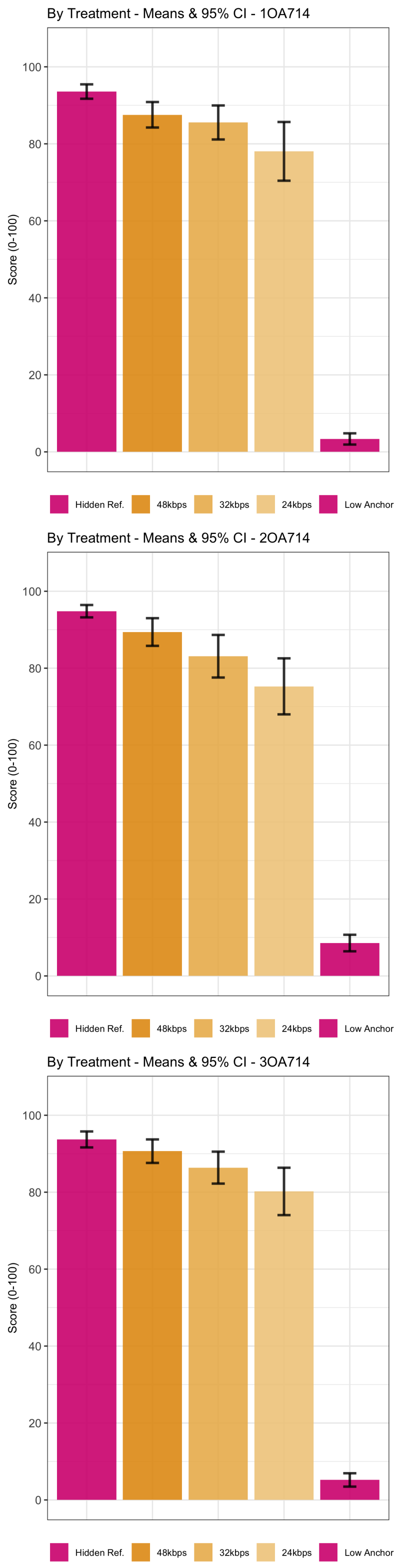
4.5 Session 3 - 2OA 5.1

The treatment factor was found significant with $F = 3552$, $p < 0.001$, as well as the sample factor, with $F = 16.2$, $p < 0.01$.

The post-hoc comparisons revealed that bitrates under 32 kbps/channel were rated significantly lower than the reference (respectively $p_{32kbps} < 0.01$; $p_{24kbps} < 0.001$). However, no significance was found between the 48 kbps/channel condition and the reference, with $p_{48kbps} = 0.909$.

The music sample generally lead to more discrimination between treatments and lower scores. The 24 kbps/channel and 32 kbps/channel conditions were rated significantly lower than the reference ($p_{32kbps,music} < 0.05$; $p_{24kbps,music} < 0.0001$). Comparatively, only the 24 kbps/channel condition was rated significantly different for all other samples, with $p_{24kbps,ambient-speech-sport} < 0.05$.

Table 5: Overall 7.1.4 tests results, by test session



4.6 Session 4 - 2OA 7.1.4

The treatment factor was found significant with F = 3796.281725, p < 0.001.

Bitrates under 32 kbps/channel were both rated significantly lower than the reference ($p_{32kbps} < 0.001$; $p_{24kbps} < 0.0001$), whereas the difference in ratings between the reference and the 48 kbps/channel condition showed no statistical significance, with $p_{48kbps} = 0.094$.

4.7 Session 5 - 3OA 5.1

Similarly to the previous test, the treatment factor was found significant with F = 3551, p < 0.001, with no other significant effect.

Bitrates as low as 32 kbps/channel were not rated significantly different than the reference ($p_{24kbps} < 0.0001$; $p_{32kbps} = 0.091$; $p_{48kbps} = 0.24$)

4.8 Session 6 - 3OA 7.1.4

Anova showed a significant effect with F = 4594 , p < 0.001. The effect of the interaction between treatment and sample was also found as significant with F = 31.1, p < 0.05. The post-hoc comparisons revealed a significant difference for bitrates of 32 kbps/channel and lower ($p_{32kbps} = 0.019$; $p_{24kbps} < 0.001$), while the 48 kbps/channel condition was not significantly rated differently than the reference ($p_{48kbps} = 0.27$).

More discrimination between conditions was achieved with the music track, with no compression level being rated significantly close to the reference ($p_{48kbps,music} < 0.05$; $p_{32kbps,music} < 0.0001$; $p_{24kbps,music} < 0.0001$). By contrast, only the 24 kbps/channel condition differed significantly from the reference for the other 3 test samples ($p_{24kbps,ambient-speech-sport} < 0.05$).

4.9 Comments

Collected comments related to both timbral and spatial aspects of the sound signals. The strong agreement between assessors on the content of the comments lead to the following generak remarks on the perceived alterations induced by compression:

- Timbral impairments were generally reported for all kind of sample and in all test sessions, including a loss of details, a loss of high frequencies and a presence of compression artefacts.
- Background noises that were previously perceptible in the original signals (typically the ambient and sport samples) were enhanced by the compression.

- Reverberance was audibly altered by the compression in the ambient track, which originally contained an important amount of reverberation.
- In the speech sample, which contained a smoothly moving sound source, a loss in spatial resolution was reported.
- For the other samples (ambient, music, and sport), the sound stage was found narrower when compressed with Opus.
- Inaccuracy in the spatial reproduction was also mentioned, including elevation effects.

5 Discussion

This study did not allow for a conclusion on a transparency bitrate for the reproduction of 1OA through 5.1 and 7.1.4 speakers layouts, although the speech and sport samples were not rated significantly differently for compression rates as low as 24 kbps/channel.

It however appears that 48 kbps/channel compression does not significantly affect the perceived audio quality of the reproduction of 2OA and 3OA reproduced over 5.1 and 7.1.4 speaker layouts, and that a bitrate of 32 kbps/channel is transparent for the reproduction of 3OA through 5.1 speakers.

It should be noted that music samples were generally given lower scores, which could be due to localisa=tion errors being easier to perceive for stationary sound sources. When removing these samples from the scope of the analysis, the 32 kbps/channel condition is found transparent for 2OA and 3OA.

However, assessors were much less able to discriminate between the test signals for the ambient and sport samples, possibly indicating that Opus processing is suited for the reproduction of complex sound scenes, where spatial inaccuracies are harder to spot.

6 Conclusion

Six listening experiments were conducted to estimate how much Ambisonics recordings could be compressed using Opus without losing audio quality transparency. Experiments were conducted for the first three Ambisonics order, in each test reproduced to either 5.1 or 7.1.4 speakers layout. Although the study could not conclude on this question for first order Ambisonics, results indicate that for second and third order Ambisonics, files compressed to 48 kbps/channel through Opus are perceptually transparent regarding Basic Audio Quality.

Further investigation could include more specific experimental designs to assess separately the Listening Quality and the Localisation Accuracy.

7 Acknowledgements

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8 Appendix

Table 6: eGauge Agreement plots of the assessors panel, by test session

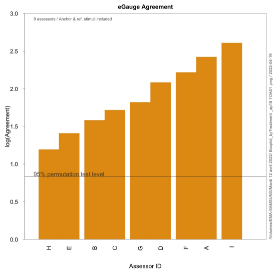
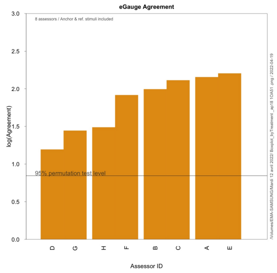
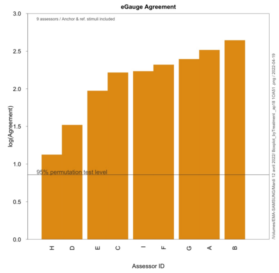
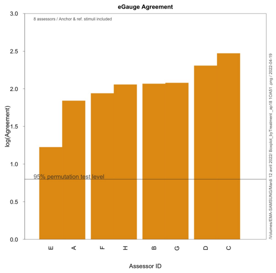
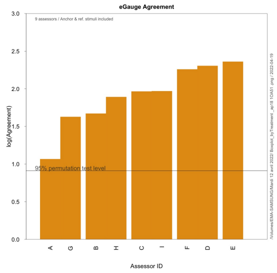
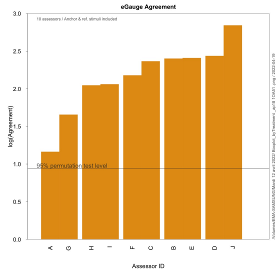
	1OA	2OA	3OA
5.1			
7.1.4			

Table 7: Discrimination plots of the assessors panel, by test session

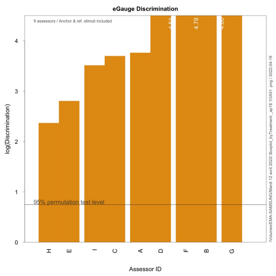
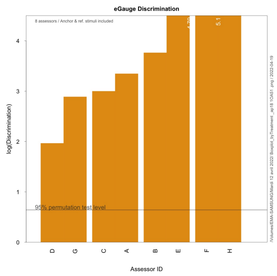
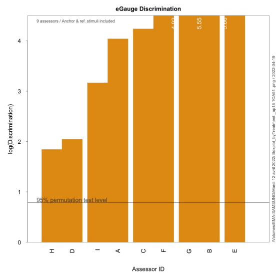
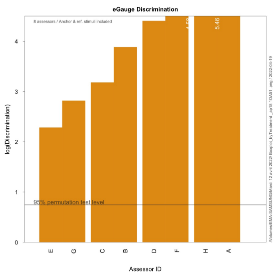
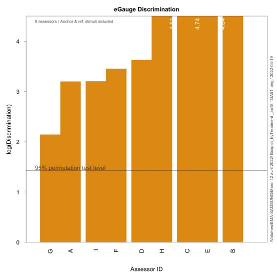
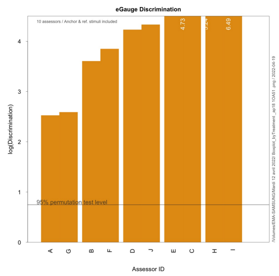
	1OA	2OA	3OA
5.1			
7.1.4			

Table 8: eGauge Reliability plots of the assessors panel, by test session

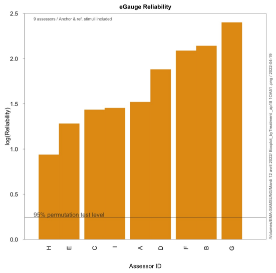
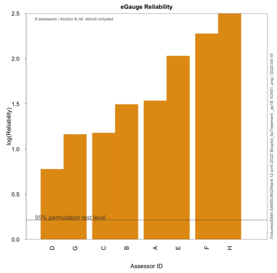
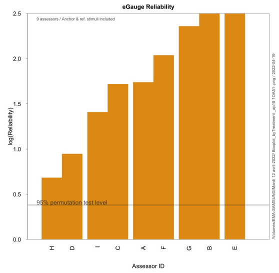
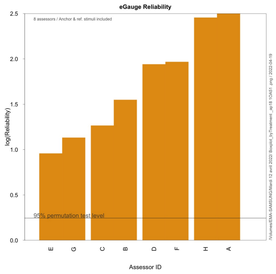
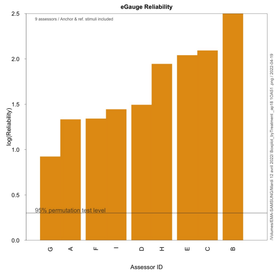
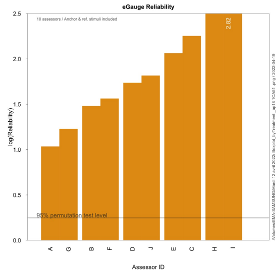
	1OA	2OA	3OA
5.1			
7.1.4			

Table 9: eGauge Discrimination vs Reliability plots of the assessors panel, by test session - values contained in the top right quadrant of the graph, delimited by the two lines on the graph, can be considered as experiences for the scope of the study.

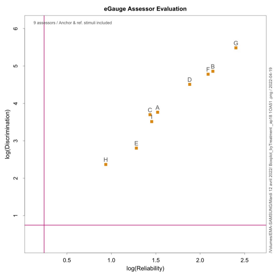
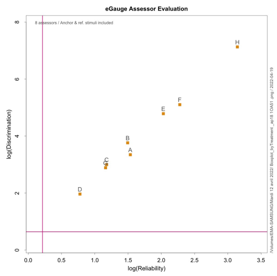
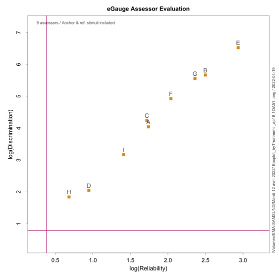
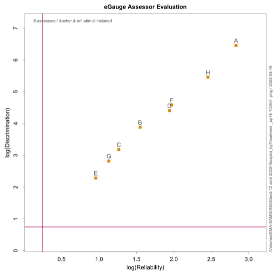
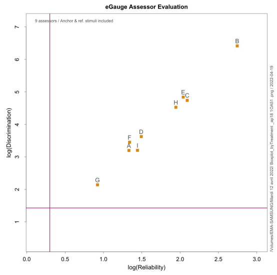
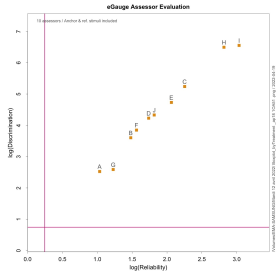
	1OA	2OA	3OA
5.1			
7.1.4			

Table 10: Tests results by audio sample, by test session

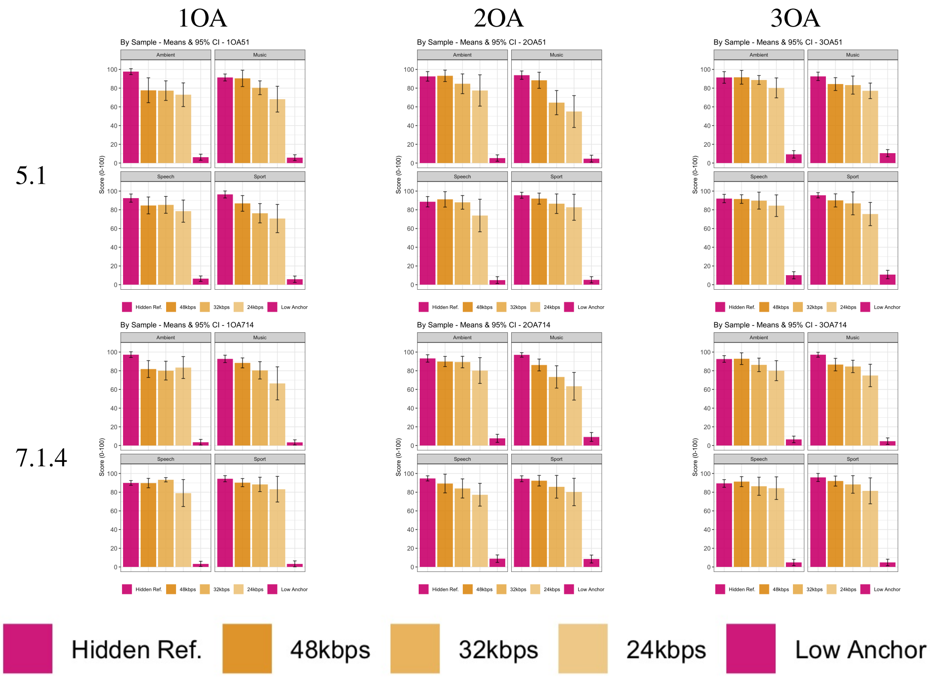


Table 11: Overall scores box plots, by test session