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Contents

Foreword

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

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Introduction

Audio is a key component of an immersive multimedia experience and 3GPP systems are expected to deliver immersive audio with a high Quality of Experience. However, industry agreed methods to assess the Quality of Experience for immersive audio are relatively few and the present document seeks to address this gap by providing objective test methods for the assessment of immersive audio.

1 Scope

The present document specifies objective test methodologies for 3GPP immersive audio systems including channel based, object based, scene-based and hybrids of these formats. The subjective evaluation methods described in the present document are applicable to audio capture, coding, transmission and rendering as indicated in their corresponding clauses.

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] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] J. Fliege und U. Maier: "A two-stage approach for computing cubature formulae for the sphere," Dortmund University, 1999.

[3] ISO 3745 - Annex A: "Acoustics - Determination of sound power levels and sound energy levels of noise sources using sound pressure -- Precision methods for anechoic rooms and hemi-anechoic rooms - Annex A: General procedures for qualification of anechoic and hemi-anechoic rooms".

[4] ISO 1996 Acoustics: "Description, measurement and assessment of environmental noise".

[5] ANSI S1.4: "Specifications for Sound Level Meters".

[6] ISO 3: "Preferred numbers – Series of preferred numbers".

[7] B. Rafaely, “Analysis and design of spherical microphone arrays,” IEEE Transactions on Speech and Audio Processing, no. 13, 2005, pp. 135 – 143

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in 3GPP TR 21.905 [1].

spherical coordinates: The coordinate system used in this document is defined such that the x-axis points to the front, the y-axis to the left and the z-axis to the top (see Figure 0). Spherical coordinates are the distance r from the origin, the azimuth ϕ in mathematical positive orientation (counter-clockwise) and the elevation angle θ relative to the z-axis (with 0 degrees pointing to the equator and +90 degrees pointing to the North pole).

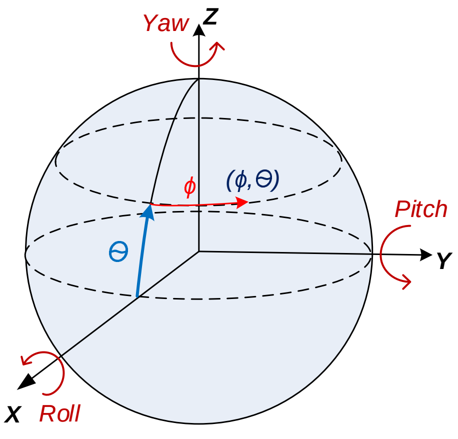

Figure 0: Spherical coordinate system

dBFS: dB full-scale, where 0 dBFS refers to the RMS level of a DC-free sinusoidal signal exercising the full scale of the digital interface/file.

3.2 Symbols

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

$\mathbf{LA_{eq}}$	the sound level in decibels equivalent to the total A-weighted sound energy measured over a stated period of time.
ϕ	azimuth
θ	elevation

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [1].

4 Objective Test Methodologies for Immersive Audio Systems

4.1 Objective Test Methodologies for Assessment of Immersive Audio Systems in the Sending Direction

4.1.1 Diffuse-field Send Frequency Response for Scene-based Audio

4.1.1.1 Introduction

This test is applicable to UEs capturing scene-based audio (e.g. First and Higher Order Ambisonics).

NOTE: Currently, the test method uses a periphonic loudspeaker array for generation of a diffuse-field. Additional loudspeaker setups for the derivation of the diffuse sound field are under consideration.

General test conditions

Free-field propagation conditions

- The test environment shall contain a free-field volume, wherein free-field sound propagation conditions shall be observed.
- The free-field sound propagation conditions shall be observed down to a frequency of 200 Hz or less.
- Qualification of the free-field volume shall be performed using the method and limits for deviation from ideal free-field conditions described in [3].

Test environment noise floor

Within the *free-field volume*, the equivalent continuous sound level of the test environment in each 1/3rd octave band, $L_{eq}(f)$, shall be less than the limits of the NR10 curve, following the noise rating determination procedures in [4].

4.1.1.2 Definition

The Diffuse-field Send Frequency Response for Scene-based Audio is defined as the transfer function, $G(f)$, between:

$\hat{P}(f)$, the estimated sound pressure magnitude spectrum obtained from a diffuse-field scene-based audio capture and reference synthesis at the geometric center of a *free-field volume*; and

$a)P(f)$, the sound pressure magnitude spectrum obtained from a diffuse-field microphone recording the same diffuse field at the origin of a spherical coordinate system.

Figure 1 describes a typical block diagram for the scene-based audio sending direction with measurement points when using a periphonic loudspeaker array.

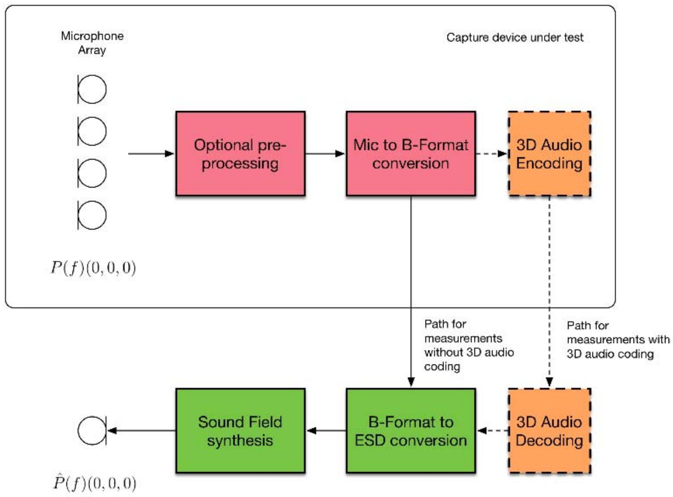

Figure 1: Scene-based audio capture block diagram for sending direction measurements

Definition of Equivalent Spatial Domain

The equivalent spatial domain representation, $\mathbf{w}(t)$, of a N^{th} order Ambisonics soundfield representation $\mathbf{c}(t)$ is obtained by rendering $\mathbf{c}(t)$ to K virtual loudspeaker signals $w_j(t)$, $1 \leq j \leq K$, with $K = (N+1)^2$. The respective virtual loudspeaker positions are expressed by means of a spherical coordinate system, where each position lies on the unit sphere, i.e., a radius of 1. Hence, the positions can be equivalently expressed by order-dependent directions $\boldsymbol{\Omega}_j^{(N)}=(\theta_j^{(N)}, \phi_j^{(N)})$, $1 \leq j \leq K$, where $\theta_j^{(N)}$ and $\phi_j^{(N)}$ denote the inclinations and azimuths, respectively. These directions are defined according to [2] and reproduced in Annex B for convenience.

The rendering of $\boldsymbol{c}(t)$ into the equivalent spatial domain can be formulated as a matrix multiplication:

$$\mathbf{w}(t) = (\boldsymbol{\Psi}^{(N,N)})^{-1} \cdot \mathbf{c}(t),$$

where $(\cdot)^{-1}(\cdot)^{-1}$ denotes the inversion.

The matrix $\mathbf{\Psi}^{(N,N)}$ of order N with respect to the order-dependent directions $\boldsymbol{\Omega}_j^{(N)}$ is defined by:

$$\mathbf{\Psi}^{(N,N)} := [\mathbf{S}_1^{(N)} \quad \mathbf{S}_2^{(N)} \quad \ldots \quad \mathbf{S}_K^{(N)}],$$

with:

$$\mathbf{S}_j^{(N)} := [S_0^0(\boldsymbol{\Omega}_j^{(N)}) \quad S_{-1}^{-1}(\boldsymbol{\Omega}_j^{(N)}) \quad S_{-1}^{0}(\boldsymbol{\Omega}_j^{(N)}) \quad S_{-1}^{1}(\boldsymbol{\Omega}_j^{(N)}) \quad S_{-1}^{1}(\boldsymbol{\Omega}_j^{(N)}) \quad \ldots \quad S_N^N(\boldsymbol{\Omega}_j^{(N)})]^T,$$

where $S_n^m(\cdot)$ represents the real valued spherical harmonics of the order n and degree m.

The matrix $\mathbf{\Psi}^{(N,N)}$ is invertible so that the HOA representation $\boldsymbol{c}(t)\boldsymbol{c}(t)$ can be converted back from the equivalent spatial domain by:

$$\mathbf{c}(t) = \mathbf{\Psi}^{(N,N)} \cdot \mathbf{w}(t)$$

4.1.1.3 Test method with periphonic array

4.1.1.3.1 Test Conditions

Periphonic loudspeaker array

a) A *periphonic loudspeaker array* shall be placed within the free-field volume with the geometric center of the *periphonic loudspeaker array* coinciding with the geometric center of the free-field volume.

b) The *periphonic loudspeaker array* shall have a radius greater or equal than 1 meter.

c) The *periphonic loudspeaker array* shall be composed of $(N+1)^2$ coaxial loudspeaker elements. Each of the $(N+1)^2$ coaxial loudspeaker elements shall be equalized (if necessary) and level compensated to conform with the operational room response curve limits given in [5] Section 8.3.4.1. N should be equal or greater than the maximum ambisonics order supported by the device under test (DUT), e.g. $N>=4$ for a DUT supporting 4th order Ambisonics capture.

d) The $(N+1)^2$ coaxial loudspeaker elements shall be positioned according to the azimuth and elevation coordinates given in Annex B.

e) All coaxial loudspeaker elements shall be oriented such that their acoustic axis intersects at the geometric center of the *free field volume*.

f) The radius of each coaxial loudspeaker element shall be such that, at the geometric center of the *free-field volume*, the far field approximation for the coaxial loudspeaker axial pressure amplitude decay holds true.

4.1.1.3.2 Measurement

Reference Spectrum measurement for periphonic loudspeaker array method

a) A diffuse-field / random incidence, or multi-field microphone is mounted in the *free-field volume* such that the tip of the microphone corresponds to the geometric center of the *free-field volume* and the geometric center of the *periphonic loudspeaker array*.

NOTE 1: Diffuse-field / random incidence microphones, are described in [5].

b) $(N+1)^2$ decorrelated pink noise signals are played simultaneously over each of the $(N+1)^2$ coaxial loudspeakers of the *periphonic loudspeaker array*.

c) The playback level is adjusted such that the *LAeq*, measured over a 30s time window at the geometric center of the *periphonic loudspeaker array,* is equal to 78dBSPL(A) ± 0.5dB.

d) The reference sound pressure at the geometric center of the *free-field volume*, *p(t)*, is captured with the diffuse-field or multi-field microphone.

e) The magnitude spectrum of the reference sound pressure, *P(f)*, is calculated for the 1/12th octave intervals as given by the R40 series of preferred numbers in [6].

NOTE 2: For ideal (calibrated) loudspeakers, the *P(f)* spectra should have equal energy in each 1/12th octave intervals.

Estimated Spectrum measurement

a) The scene-based audio capture device under test is mounted in the *free-field volume* such that its geometric center coincides with the geometric center of *free-field volume* and the geometric center of the *periphonic loudspeaker array*.

b) $(N+1)^2$ decorrelated pink noise signals are played simultaneously over each of the $(N+1)^2$ coaxial loudspeakers of the *periphonic loudspeaker array*. The pink noise signals shall be identical to the signals used for the reference spectrum measurement.

c) The B-format scene-based audio format representation (compressed or uncompressed, depending on the use case being tested) is stored for offline analysis.

d) The B-format scene-based audio format representation is uncompressed (if necessary) and converted to an *equivalent spatial domain representation* of order N_{DUT} (B-Format to ESD conversion in Figure 1), where N_{DUT} corresponds to the Ambisonics order of the device under test.

e) $\hat{p}(t)$, the estimate of the sound field at the geometric center of *the free-field volume* and *periphonic loudspeaker array*, is synthesized using the *equivalent spatial domain representation* of order N_{DUT}.

NOTE 3: $\hat{p}(t)$ can be taken from the W component of the B-Format signal, as an alternative to implementing the B-Format to ESD conversion in step d).

f) The magnitude spectrum of the estimated sound pressure, $\hat{P}(f)$, is calculated for the 1/12th octave intervals as given by the R40 series of preferred numbers in [6].

Calculation of send frequency response for scene-based audio

The send frequency response for scene-based audio, *G(f)*, is calculated as $G(f) = \frac{\hat{P}(f)}{P(f)}$.

4.1.1.4 Test method with loudspeaker array and turn table

4.1.1.4.1 Test Conditions

Loudspeaker array

a) A calibrated *loudspeaker array* shall be placed within the *free-field volume*.

b) The *loudspeaker array* shall comprise one or several semi-arcs having a radius greater or equal than 1 meter. The radius shall be reported.

c) The *loudspeaker array* shall be composed of $N+1$ loudspeaker elements. The ambisonic order N shall be reported.

d) Each loudspeaker in the array shall be calibrated with a frequency response of [at least 100 Hz-20,000 Hz] and minimum phase response.

e) The coordinates of the loudspeaker elements are defined according to a Gaussian spherical grid [7] of order N. Directions shall comply with Annex B.1 and the $N+1$ elevations of the spherical grid shall be reported.

Turn table

a) A turn table with a resolution of 0.5 degrees shall be used. The rotation axis of the turn table and the vertical axis of the semi-arcs shall be aligned The turn table shall be adjusted in height so that the device under test is positioned at the geometric center of the *loudspeaker array*.

b) For measurement, an azimuth step of $180/(N+1)$ degrees shall be used.

4.1.1.4.2 Measurement

Reference Spectrum measurement

a) A diffuse-field / random incidence, or multi-field microphone is mounted in the *free-field volume* such that the tip of the microphone corresponds to the geometric center of the *free-field volume* and the geometric center of the *loudspeaker array*.

NOTE 1: Diffuse-field / random incidence microphones, are described in [5].

Repeat steps b-c) with an azimuth angular resolution of 180/(*N*+1) degrees:

b) An exponential sweep sine signal is played over each of the *N*+1 loudspeakers of the *loudspeaker array*.

c) The impulse response at the geometric center of the *loudspeaker array* is measured for each loudspeaker position.

d) The magnitude spectrum of the reference sound pressure, *P(f)*, is calculated for the 1/12th octave intervals as given by the R40 series of preferred numbers in [6].

NOTE 2: For ideal (calibrated) loudspeakers, the *P(f)* spectra should have equal energy in each 1/12th octave intervals.

Estimated Spectrum measurement

a) The scene-based audio capture device under test is mounted in the *free-field volume* such that its geometric center coincides with the geometric center of *free-field volume* and the geometric center of the *loudspeaker array*.

b) Repeat steps b-c) with an azimuth angular resolution of 180/(*N*+1) degrees::

c) An exponential sweep sine signal is played over each of the *N*+1 loudspeakers of the *loudspeaker array*. The sweep signals shall be identical to the signals used for the reference spectrum measurement.

d) The impulse response at the geometric center of the *loudspeaker array* is measured for each loudspeaker position.

e) The magnitude spectrum of the estimated sound pressure, $\hat{P}(f)$, is calculated for the 1/12th octave intervals as given by the R40 series of preferred numbers in [6].

Calculation of send frequency response for scene-based audio

The send frequency response for scene-based audio, *G(f)*, is calculated as $G(f) = \frac{\hat{P}(f)}{P(f)}$.

Due to practical constraints (e.g. reflections on turn table), measurements for specific elevations (e.g. < - degrees) may be unreliable and discarded. In this case, the above measurement procedure may be conducted in two phases by measuring only directions for one hemisphere (e.g. top hemisphere, with elevations >0) in each phase. The device under test shall be flipped upside down between the two phases, and this two-phase approach shall be reported.

4.1.2 Directional response measurement for scene-based audio

4.1.2.1 Definition

The directional response for scene-based audio is defined as the transfer function, represented as an impulse response, $\mathbf{h}(\theta_i, \phi_i)$, between a device under test and a loudspeaker located at an equal distance *r* and L predefined directions, (θ_i, ϕ_i), *i*=1,...,L.

4.1.2.2 Test conditions

Free-field propagation conditions

- The test environment shall contain a free-field volume, wherein free-field sound propagation conditions shall be observed.

- The free-field sound propagation conditions shall be observed down to a frequency of 200Hz.

Test environment noise floor

The equivalent continuous sound level of the test environment in each 1/3rd octave band, $L_{eq}(f)$, shall be less than the limits of the NR10 curve, following the noise rating determination procedures in [4].

Loudspeaker array

A real or simulated loudspeaker array comprising L loudspeakers located be a set of predefined directions (θ_i, ϕ_i), i=1,...,L, from the geometric center of the *loudspeaker array* shall be used.

4.1.2.3 Measurement

For each loudspeaker position (θ_i, ϕ_i), i=1,...,L , the following procedure shall be used:

a) An exponential sweep sine test signal is played over the loudspeaker.

NOTE: The impact of codec on the exponential sweep sine test signal needs to be verified before performing the measurements. An activation signal may be needed.

b) The impulse response $\mathbf{h}(\theta_i, \phi_i)$ at the geometric center of the *loudspeaker array* is measured.

4.2 Objective Test Methodologies for Assessment of Immersive Audio Systems in the Receiving Direction

4.2.1 Headset Binaural Diffuse-field Receive frequency response for Scene-based audio

4.2.1.1 Introduction

This test is applicable to UEs rendering scene-based audio (e.g. First and Higher Order Ambisonics) over a binaural headset.

4.2.1.2 Definition

The Headset Binaural Diffuse-field Receive Frequency Response for Scene-based Audio (for left and right ears) is defined as the transfer function, $G_{L,R}(f)$, between:

a) $P_{L,R}(f)$, the binaurally recorded sound pressure magnitude spectra, obtained when a diffuse field signal in the equivalent spatial domain representation, w(t), is played on the DUT; and

b) $P_{ref\,L,R}$(f), the reference sound pressure magnitude spectra, obtained by direct convolution of the diffuse field signal in the equivalent spatial domain representation, w(t) with its corresponding set of HRTFs.

4.2.1.3 Test Conditions

Test environment noise floor

The equivalent continuous sound level of the test environment in each 1/3rd octave band, $L_{eq}(f)$, shall be less than the limits of the NR10 curve, following the noise rating determination procedures in [4].

The set of HRTFs used by the UE shall be documented and available to the test lab.

4.2.1.4 Measurement

Reference sound pressure magnitude spectra

The reference sound pressure magnitude spectra are derived offline. The reference sound pressure magnitude spectra for the left and right ears, $P_{ref\,L,R}(f)$ is the frequency domain representation of the convolution between the set of equivalent spatial domain signals, $\mathbf{w}$(t), with its corresponding set head related transfer functions $\boldsymbol{h}_{L,R}(t)$, for each direction j in an equivalent spatial domain of order N_{DUT}, i.e.:

$$P_{ref\ L,R}(f) = \mathcal{F}\left(\sum_{j=1}^{(N_{DUT}+1)^2} w_j(t) * h_{j\ L,R}(t)\right)$$

The signals $w_j(t)$, for $1 \le j \le (N_{DUT} + 1)^2$, are uncorrelated pink noise signals of 30s length.

Binaurally recorded sound pressure magnitude spectra

The binaurally recorded sound pressure magnitude spectra is obtained as follows:

a) The binaural headset is placed on a HATS.

b) The DUT shall be configured such that the set of HRTFs used for binaural rendering correspond to the HATS used for testing.

c) The DUT volume control (if any) is adjusted for its nominal setting.

d) The binaural time-domain signals are recorded with HATS.

e) The binaurally recorded sound pressure magnitude spectra, $P_{L,R}(f)$ is obtained by taking the Fourier transform of the binaurally recorded time-domain signals.

Calculation of headset binaural diffuse-field receive frequency response for scene-based audio

The headset binaural diffuse-field frequency response for scene-based audio, $G(f)$, is calculated for each supported Ambisonics order N_{DUT} as:

$$G(f) = \frac{P_{L,R}(f)}{P_{ref\ L,R}(f)}$$

4.2.2 Nominal System Sensitivity in Receive Direction for Channel-based audio

4.2.2.1 Introduction

This test is applicable to UEs rendering channel-based audio (e.g. 7.1.4) over a binaural headset.

4.2.2.2 Definition

The nominal system sensitivity in receive direction for channel-based audio is defined as the difference between the sound pressure level (in dBSPL(A)) produced by the DUT on HATS and the root mean square of the digital test signal (in dBFS).

4.2.2.3 Test Conditions

Test environment noise floor

The equivalent continuous sound level of the test environment in each 1/3rd octave band, $L_{eq}(f)$, shall be less than the limits of the NR10 curve, following the noise rating determination procedures in [4].

The specific HATS used for the recording shall be described in the test report. The set of HRTFs used by the UE shall be documented and available to the test lab.

4.2.2.4 Measurement

For each audio channel supported by the DUT, a pink noise signal with -18 dBFS RMS level is played, with the signals played only one channel at a time.

The *LAeq* (in dBSPL(A)) is measured continuously for a period of 30 s for each of the left and right ears.

The sensitivity $G_{i\ L,R}$ is expressed as the difference of the recorded sound pressure levels at the left and right ears and the root mean square digital level of the pink noise test signal, i.e. -18 dBFS.

$$G_{i\,L,R} = LA_{eq} - 18$$

4.2.3 Motion to Sound Latency in Dynamic Binaural Rendering Systems

4.2.3.1 Introduction

Motion to Sound latency is the time difference between the event of a change in head rotation and when the immersive audio signal is finally compensated for the head motion. The method in this specification is intended to verify that the overall motion-to-sound latency that a user experiences upon rotating their head is within acceptable limits.

The method allows full measurement of motion to sound, i.e. including both the latency of the head tracking sensor as well as the audio playback. This includes all components of a real setup and therefore contains all possible causes of additional latency that a user may experience.

The method also provides a latency value for the isolated audio processing of the binaural renderer without the aforementioned external hardware, assuming that the binaural renderer can process audio data as an audio processing plugin that can be evaluated in isolation.

NOTE: This method requires synchronized playback of two renderer instances and may not be suitable for the measurements of UEs where such synchronization is not possible.

4.2.3.2 Requirements

The following will be required:

Software:

- Audio processing software to run and record output of two renderers simultaneously
- Head tracker software

Hardware:

- Host machine for audio processing
- Head tracker hardware
- Stereo audio recording interface
- Stereo audio playback interface
- Mechanical setup to rotate the head tracking sensor in a precise and reproducible way

An exemplary hardware setup can be seen below in Figure 2, the method however can also be implemented using different systems under test and accompanying equipment:

Figure 2: Hardware Overview (Setup in Position 1 on the left, Position 2 on the right)

The audio processing environment uses two parallel signal chains, each containing its own instance of the same binaural renderer being tested. The test is concerned only with yaw angles, so values of pitch and roll should be set to zero at the beginning of the test and can be ignored thereafter.

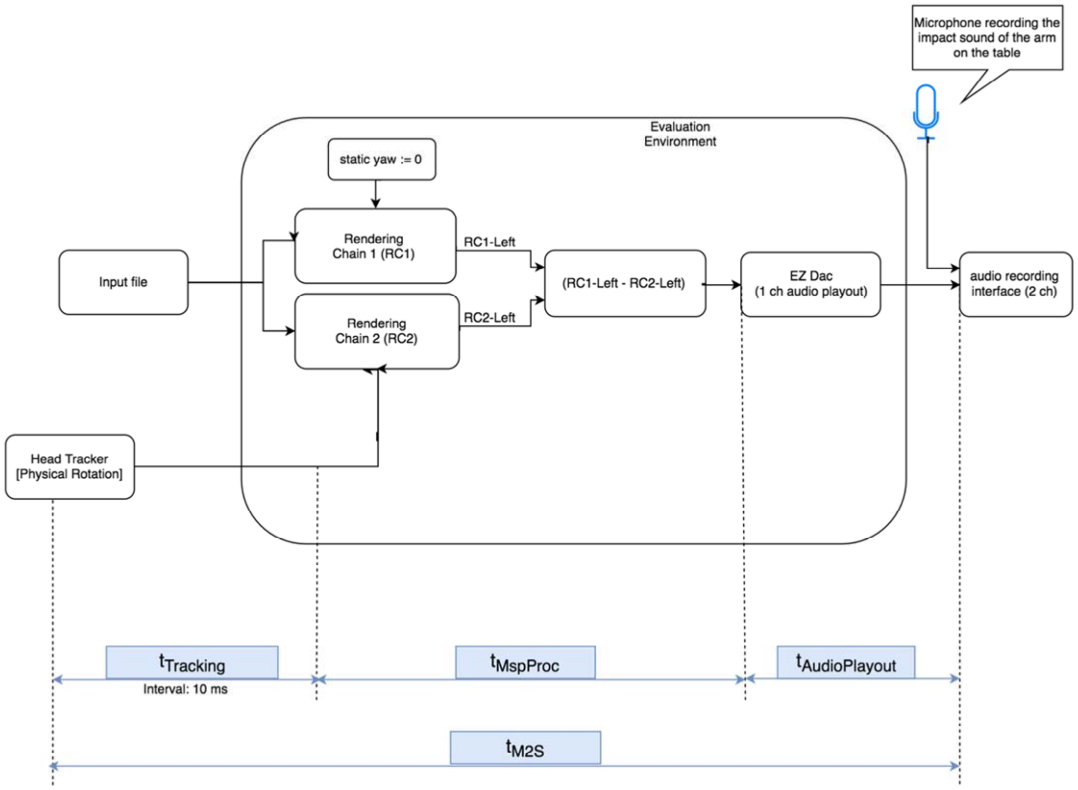

Figure 3: Generic Audio Processing Environment

The initial conditions are that Rendering Chain 1 (RC1) has a static yaw head rotation angle of 0 degrees and RC2 uses the physical rotation of the head tracker to get its yaw value. A white noise signal is virtually placed directly in front of the listener (0 degrees azimuth, elevation), meaning that rotation of the arm directly affects how the white noise source is rendered.

4.2.3.3 Calibration

The first step is to calibrate the final position of the rotating arm (Position 2 / P2). The rotating arm is moved manually and requires only a limited range of motion - from some small rotation away from the table (Position 1 / P1), 20 to 30 degrees will be ample, through until contact with the table (P2). The arm should be placed at P2 and set up so that this position also corresponds to 0 degrees yaw.

4.2.3.4 Evaluation Environment

An object within the evaluation environment, e.g. using Max/MSP, should be created to set the value of yaw to exactly 0 degrees once the real value of yaw (received from the head tracker) is <0.2 degrees.

NOTE: This tolerance value was chosen to be as small as possible while ensuring that it does not bounce (dependent on the accuracy of the tracking system)

This object should be designed to latch to zero once the actual value is under the tolerance threshold, so that any small accidental rebound of the rotating arm does not affect the yaw angle fed to the renderer - it artificially remains at exactly 0, which is important to ensure that both rendering chains have exactly the same head rotation when the arm is in its final position (P2). The output from the evaluation environment is captured by the recording audio interface, which therefore includes any latency introduced by playback.

4.2.3.5 Data acquisition

A test run begins by starting to record on the recording audio interface. The rotating arm is set to Position 1, then the audio processing set running and starts feeding the input source to both renderer instances. A microphone is positioned near the contact point at the table. This mono room microphone will be recorded synchronously with the output from the evaluation environment, with its purpose being to log the point of contact of the arm with the table, which should be done with a good amount speed and vigour so that the microphone picks up a loud knock at the table. Shortly after this (one or two seconds for example), with the test run now complete, playback and recording can be stopped.

Some milliseconds after the collision, the latest yaw value detected by the tracking system will have been passed into the evaluation environment ($t_{Tracking}$). With the target yaw value now reached (latched to zero in Max), both rendering chains will have the identical values of head rotation and therefore, after some further short delay, the output of both renderer instances will be identical.

4.2.3.6 Data Analysis

The overall motion-to-sound latency (tM2S) is taken as the time from the moment of collision until the point at which the two output signals are identical.

To easily visually inspect when this point occurs, one output channel of one signal chain (e.g. RC1-left) is subtracted from the same output channel of the complementary signal chain (RC2-left).

NOTE 1: This could be done manually in audio editor software after processing, but this would require recording at least three channels synchronously (one from each renderer chain, and one of the room microphone). Instead, the subtraction of signals can be done within Max/MSP, meaning only the output of this operation (one channel) and the room microphone can conveniently be recorded with a stereo audio interface. In addition to the stereo WAVE file recorded by a separate audio application, the Max/MSP application also writes to a separate mono WAVE file once it detects that it is in the final tolerated yaw position (latched on). This mono WAVE contains only the subtracted signals as described above, from which the tMspProc time can also be measured.

Evaluation is performed offline in audio editor software. The tMspProc time is measured from the start of the file until the point at which consecutive zero samples begin. This value encompasses any motion-to-sound latency caused by the tested renderer chain as well as any other latency caused by Virtual Studio Technology (VST) plugin framework buffering. The tMspProc time shall be measured from the audio frame boundary at which the latched-on yaw value is activated and applied within that audio frame.

NOTE 2: Since the yaw rotation update rate of the tracker is typically in the range of a few milliseconds, there is a framing mismatch when compared to the audio framing, but this mismatch will not be incorporated in the tMspProc value but rather only in the tM2S measurement.

An example measurement of the tMspProc is displayed in Figure 4. For tM2S this is measured by selecting the duration between the visible collision peak in the microphone channel and the point at which the other channel reduces to silence. Figure 5 shows an example measurement for the motion-to-sound latency.

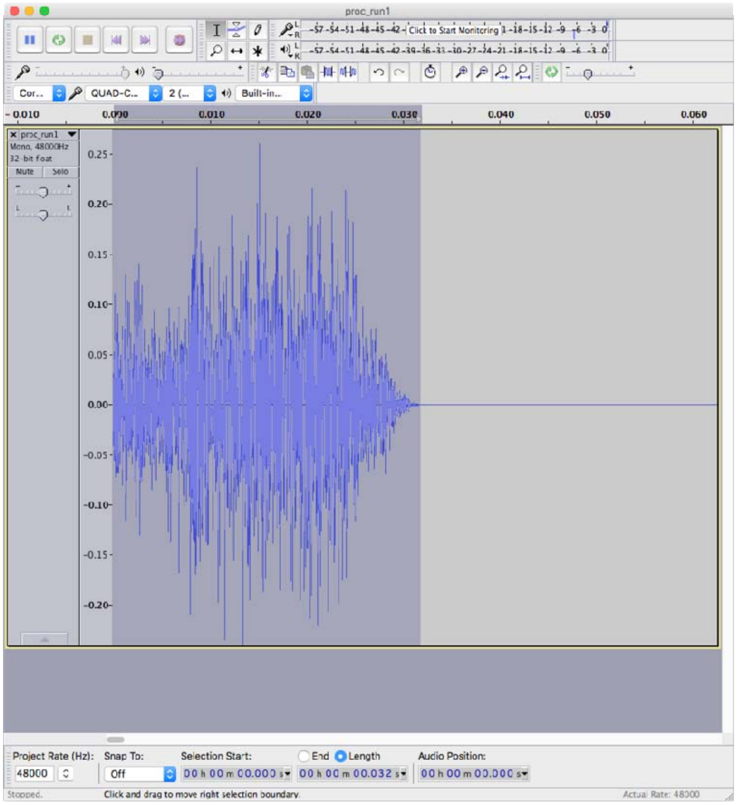

Figure 4: tMspProc latency measurement

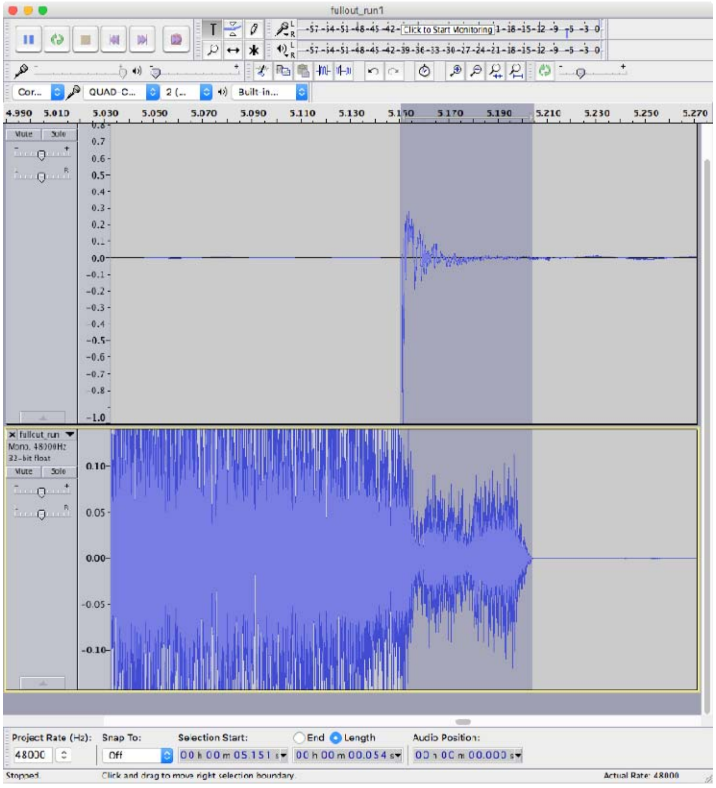

Figure 5: Motion-to-sound (tM2S) latency measurement

In Figure 5 the room recording is on the top, subtracted renderer output is on the bottom. Marked region is the time passed since the arm hits the table (recorded knock) and when the subtracted binaural renderer output reaches silence.

NOTE 3: Unlike the tMspProc measurement, the tM2S measurement is taken from signals recorded from hardware audio interfaces, hence it is not possible to look for continuous silence since the resulting file will always contain some noise added by the digital-to-analog and analog-to-digital converters. For this reason, it is important to ensure a high signal-to-noise ratio in the signal provided to audio interfaces, to make it easier to inspect where the cancellation occurs.

Annex A (normative): Order dependent directions

The following tables order-dependent directions $\boldsymbol{\Omega}_j^{(N)} = (\phi_j^{(N)}, \theta_j^{(N)}),\ 1 \leq j \leq K$, where $\theta_j^{(N)}$ and $\phi_j^{(N)}$ denote the elevations and azimuths in radians, respectively.

Index *j*	$\theta_j^{(N=1)}$	$\phi_j^{(N=1)}$
1	1.570796	0
2	-0.339837	0
3	-0.339837	2.094395
4	-0.339837	-2.0944

Index *j*	$\theta_j^{(N=2)}$	$\phi_j^{(N=2)}$
1	1.570796	0
2	-0.790277	0
3	0.363207	-1.95668
4	0.363207	1.956682
5	-0.844382	-1.95668
6	0.009757	-3.14159
7	-0.844382	1.956681
8	0.245128	0.687124
9	0.245129	-0.68712

Index *j*	$\theta_j^{(N=3)}$	$\phi_j^{(N=3)}$
1	1.570796	0
2	0.716698	0
3	-0.461173	1.119907
4	-1.034310	-0.25283
5	0.492174	1.155586
6	-0.165812	2.040481
7	-0.461172	-1.38118
8	-0.165813	0.270692
9	0.001916	-2.20417
10	0.653709	2.297267
11	0.653709	-2.80293
12	-0.192680	3.010956
13	-1.079056	2.154919
14	0.001915	-0.63529
15	0.616834	-1.41973
16	-0.887326	-2.46809

Index *j*	$\theta_j^{(N=4)}$	$\phi_j^{(N=4)}$
1	1.570796	0
2	0.747578	0
3	-0.168324	-2.00759
4	0.846499	1.927637
5	0.234515	-1.41208
6	0.699165	-2.10001
7	0.307091	2.512927
8	0.130649	1.667633
9	-0.677517	1.442383
10	0.136843	-0.60062
11	-1.317269	0.329968
12	-0.433118	-1.18621
13	-0.231864	2.983332
14	0.174242	-2.69222
15	-0.599985	0.507602
16	-0.382009	2.208977
17	-0.009394	0.952319
18	-1.013813	-1.71565
19	0.696199	0.934402
20	-0.602139	-0.38654
21	-1.041921	2.675958
22	-0.623111	-2.62842
23	0.054056	0.165012
24	0.855489	-1.02504
25	0.808243	-3.13121

Index *j*	$\theta_j^{(N=5)}$	$\phi_j^{(N=5)}$
1	1.570796	0
2	-0.454100	0
3	0.323739	-1.19666
4	-1.175381	0.184066
5	0.947221	0.124282
6	-0.193698	-2.84022
7	0.500281	-1.84701
8	-0.663529	0.698758
9	-0.613332	2.280239
10	-0.588043	-2.28482
11	0.946645	-2.37569
12	0.333311	2.883411
13	0.967374	-1.18504
14	0.436854	-2.76846
15	0.510141	0.763488
16	-0.063811	-0.46491
17	0.048266	-2.27504
18	-0.148392	1.762138
19	0.945735	2.804486
20	-0.125777	-1.69175
21	-0.241518	-1.0321
22	-0.063824	0.509415
23	-1.240392	-1.95737
24	0.542172	-0.567
25	0.043647	2.319619
26	-0.291045	2.853233
27	-0.841101	-3.07101
28	-1.213891	2.113132
29	-0.706626	-1.50877
30	-0.774625	-0.65404
31	-0.707445	1.464227
32	0.990842	1.373127
33	-0.122664	1.112751
34	0.598614	2.113949
35	0.306690	0.057137
36	0.381934	1.457925

Index *j*	$\theta_j^{(N=6)}$	$\phi_j^{(N=6)}$
1	1.570796	0
2	0.720144	0
3	-0.308365	3.024454
4	0.068431	2.080642
5	-0.495677	-2.21373
6	-0.018779	-2.03598
7	0.426043	1.678014
8	-0.259742	0.964363
9	0.179320	-3.03552
10	-0.249618	-2.70206
11	1.074183	0.581055
12	-0.781172	-2.80103
13	0.457849	0.550136
14	0.523951	-1.98436
15	-0.006246	-0.51212
16	-0.788507	-1.1411
17	0.228181	-2.48765
18	-0.418110	-1.62282
19	-0.512688	-0.57506
20	0.572140	2.286204
21	-0.867576	-0.08741
22	-0.624799	0.547028
23	-0.446687	1.878965
24	-0.789667	2.746717
25	1.047763	-0.76025
26	0.247192	-1.01978
27	0.720143	1.162107
28	-0.081819	1.507148
29	0.226040	1.062706
30	0.709088	-2.68135
31	-0.249096	-1.08377
32	0.573959	2.91352
33	1.069121	2.939099
34	0.135381	-1.53966
35	-0.057504	0.473238
36	-0.975369	-1.95522
37	-0.666036	1.294994
38	-1.146922	0.887936
39	-0.357070	2.427548
40	0.200642	-0.01608
41	-0.965084	1.97199
42	0.681666	-1.35341
43	0.112434	2.651183
44	0.528475	-0.57647
45	1.003627	1.857517
46	-1.275974	-0.77916
47	1.051102	-2.01121
48	-1.315079	3.087768
49	-0.326694	-0.00446

Annex B (normative): Directions in Gaussian spherical grid

B.1 Definition

A Gaussian grid of order N consists of $2(N+1)^2$ points associated to directions $\Omega_{i,j}^{(N)} = \left(\theta_i^{(N)}, \phi_j^{(N)}\right)$, $0 \leq i \leq N$ and $0 \leq j < 2(N+1)$, where $\theta_i^{(N)}$ and $\phi_j^{(N)}$ denote the elevation and azimuth, respectively. These directions are defined as follows [7]: The elevations $\theta_i^{(N)}$ are computed as the zeros of the (N+1)-th degree Legendre polynomial $P_{N+1}\left(\cos(\theta_i^{(N)})\right) = 0$, while the azimuths are given by $\phi_j^{(N)} = \frac{2\pi j}{2(N+1)}$ (in radians) or j.180/(*N*+1) (in degrees), $0 \leq j < 2(N+1)$.

Directions in test setup shall comply with the theoretical values $\Omega_{i,j}^{(N)}$ with an accuracy of +/-0.5 degree for all azimuths and +/-0.5 degree for elevations in the range [-80,+80] degrees. For elevation >80 degrees and <-80 degrees, the accuracy shall be respectively +4/-0.5 degrees and +0.5/-4 degrees.

B.2 Example loudspeaker array

An example implementation with an ambisonic order N = 29 is described below:

- A turn table with constant step size of 6 degrees and starting at 0 degree (to obtain 60 positions in azimuth).
- Two fixed semi-arcs of radius 2.5 meters separated in azimuth by 90 degrees with 15 loudspeakers on each semi-arc; the elevations of loudspeakers are given (in degrees) by -85, -80, -74, -68, -62, -56, -50, -44, -38, -32, -27, -21, -15, -9, -3, 3, 9, 15, 21, 27, 32, 38, 44, 50, 56, 62, 68, 74, 80, 85, where succesive values are alternatively allocated to each semi-arc.

NOTE: In practice, the elevation of -85 degrees may be replaced by a nearby value (e.g. -82 degrees) to leave room for the mounting structure at the bottom of the loudspeaker array.

Annex C (informative): Change history

Change history							
Date	**Meeting**	**TDoc**	**CR**	**Rev**	**Cat**	**Subject/Comment**	**New version**
2018-09	SA-81	SP-180644				Presented to TSG SA#81 for approval	1.0.0
2018-09	SA-81					Approved at TSG SA#81	15.0.0
2018-12	SA-82	SP-180969	0001	2		Corrections to test method with loudspeaker array and turn table	15.1.0
2020-03	SA-87-e	SP-200105	002	1	F	Correction of sensitivity calculation for immersive audio playback	15.2.0

History

Document history		
V15.0.0	October 2018	Publication
V15.1.0	April 2019	Publication
V15.2.0	March 2020	Publication