

herisSon – Spatial Room Impulse Response (SRIR) measurements for the evaluation of electro-acoustic systems in auditoriums

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Summary

Measured spatial room impulse responses are a very useful tool to understand the behaviour of acoustic spaces by analysing them in both temporal and spatial domains. **herisSon** is a tool designed at LINK ACOUSTIQUE that implements different techniques for analysing and presenting SRIR measurements results in an updated version compared to the one presented by the author at the Auditorium Acoustics Conference in 2023 [1]. The tool expands the previous research by providing an updated 3D visualization, the possibility of analysing the 3D response in 1/1 octave bands, ISO 3382 parameters, new acoustical parameters and visualization proposed by A. Bassuet [2] and a proposed analysis of directional reverberation in 3D. All these new features have been also used to test auditoriums, amplified music halls, immersive sound systems and the installation of an electroacoustic system in a concert hall originally conceived for classical music. The paper presents the results of these measurements and discuss what are the next steps to improve our understanding and presentation of the results.

1 INTRODUCTION

This paper describes the implementation and new developments of a series of MATLAB routines for the measurement and analysis of Spatial Room Impulse Responses (SRIRs) that employ the Spatial Decomposition Method (SDM), 3rd order Ambisonics and the exponential sweep sine test signal in the version presented by the author in 2011 [3].

The first implementation of herisSon (the name of the tool) has been presented at the Auditorium Acoustics conference in Athens in 2023 [1].

The paper is divided into two main sections:

- Firstly, a description of the new developments and the corrections of previous problems, notably a better evaluation of the direct sound, a newly and more regular distribution of the receiver sphere, the visualization proposed by A. Bassuet [2], a direction-dependent reverberation analysis [3], 1/1 octave bands analysis of the directional of arrivals and the calibration of the measurement system in anechoic conditions.
- Secondly, we present a series of measurements showing how herisSon has helped us to evaluate the performance of electro-acoustic systems in auditoriums.

2 HERISSON

2.1 Overview

herisSon is composed of two main sections:

1. **Measurements:** this is the tool's section that is used for the actual SRIR measurements, implementing the ESS method as described in [3]. This section is hard coded to use a Zylia ZM-1 microphone as input and a single source or multichannel source as output.
2. **Analysis:** this is the section where the SRIRs are analysed using the SDM method and 3rd order ambisonics, described in [1], for microphone arrays and for B-format signals [4]. This is also the section where the main visualization tool helps to investigate the SRIR in details. This is the section where most of the recent development occurs.

Figure 1 below shows the orientation of the front of the microphone that is applicable to all results shown in this document.

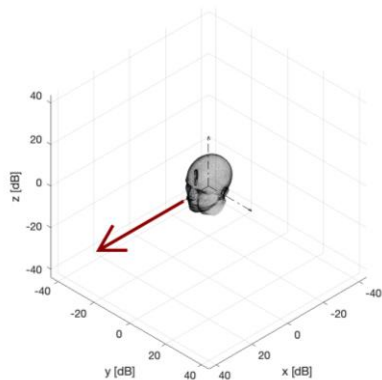


Figure 1: Orientation of the front of the microphone

2.2 New Developments

2.2.1 Direct Sound evaluation

After having performed several measurements campaigns with different sound source configurations (either dodecahedron or PA systems), we have found that the proposed evaluation of the direct sound in our previous iteration of the tool [1] was prone to errors especially in the case of a PA system where the sound comes from more than one loudspeaker.

According to [5], it is important to correctly evaluate the onset of the Room Impulse Response (RIR) to have a correct evaluation of the energetic and statistical acoustical indices (C80, Tc, etc).

Our approach to evaluate the IR's onset is based on two different algorithms derived from the ITA toolbox [6]:

- **Case of one source (usually dodecahedron):** The onset detection proposed in the ISO 3382-1, paragraph A.3.4, where the onset is calculated as the time when the signal emerges from the background noise but it is still 20dB lower than the maximum peak
- **Case of multiple sources (usually PA system):** The onset detection is based on cross correlation method (CCM) to measure time delay estimation (TDE) between different sources in order to better evaluate which source represents the direct sound [7]

Once the onset is detected, we developed an algorithm based on the idea that the direct sound cannot be considered as just one single acoustic event but as a series of very close timing events, based on the SDM method analysis characteristic of one acoustic event per sample [8].

The algorithm steps are as such:

1. Detection of the IR's onset
2. Creation of a 4ms analysis window to detect the highest peak in this temporal window frame
3. Calculation of the time difference between the onset and the highest peak
4. Creation of the direct sound window, starting at the IR's onset, with a length equal 2 times the time difference between the onset and the highest peak

Figure 2 below shows the different steps of the algorithm.

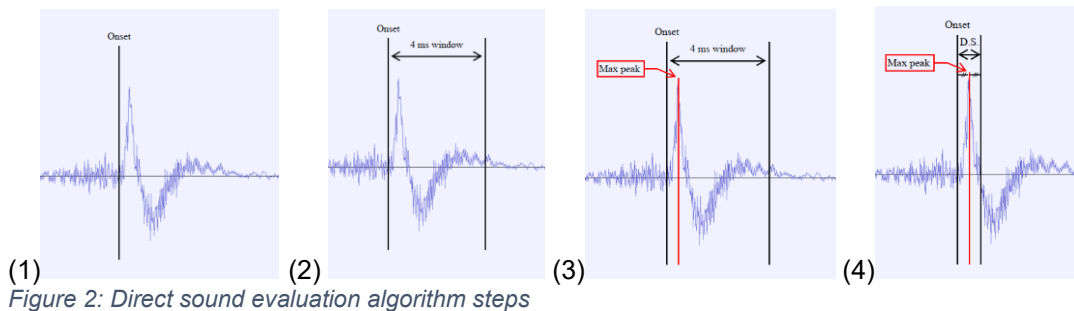
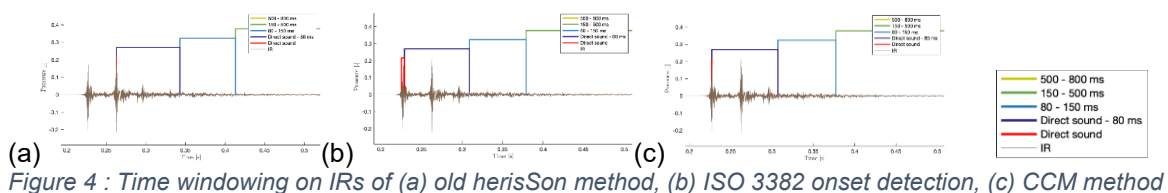
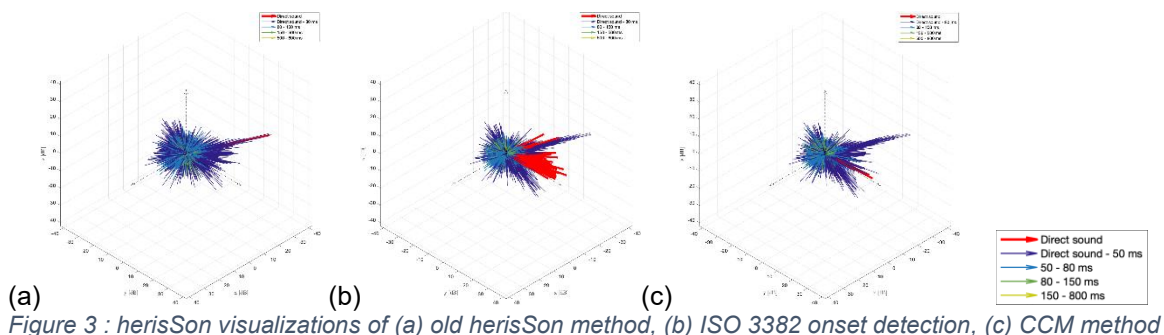


Figure 3 and Figure 4 below show the results with different direct sound evaluation techniques, on the left our old model based on the highest peak in the IR = direct sound, in the center the new algorithm following the ISO 3382 onset detection, on the right the new algorithm following the CCM (more suited for this case) for a measurement in a concert hall with a multichannel sound system for a source on the left of the receiver.



The results in the figures above also show that the CCM method is more efficient than the ISO 3382 onset method in the case of multi-channel sound systems.

It is useful to know that the size of the sliding window in the SDM analysis to evaluate the DOAs of acoustic events has been set at 48 samples for our microphone with a diameter of 9.8cm, which is in accordance with the results shown in [8] (paragraph 2.2.2) where the optimal window size for an array's diameter of 10cm @48000Hz should be between 36 and 64 samples.

2.2.2 Discretization of the receiver sphere

The original SDM code uses an angular resolution for the discretization of the receiver sphere that can go down to 1 degree, having the same angle on the azimuth and the elevation. This distribution consist of longitude-latitude points that divide the azimuth and elevation equally, resulting in a concentration of points at the top and bottom of the sphere (Bivariate trapezoidal rule in [9], Figure 5 (a)).

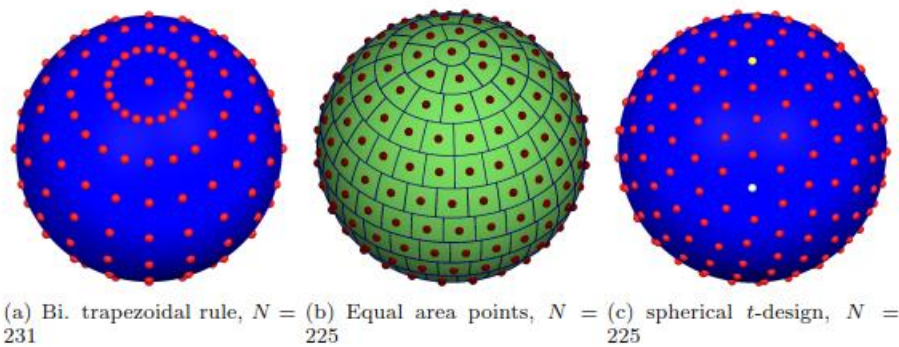


Figure 5 : Different points distribution on a sphere: (a) Bivariate trapezoidal rule – original SDM, (b) Equal area points – herisSon SDM, (c) Spherical t -design – herisSon 3rd order Ambisonics

As the SDM calculates the pressure value of the peaks at each angle as a log-sum of the detected DoAs that fits into that angle region, we have found that a more appropriate and equal distribution of the points on the receiver sphere was the Equal area points case (Figure 5 (b)).

This distribution also provides an equal area distribution of the sphere resulting in a more uniform visualizations of the DoAs and the acoustic events.

In the case of herisSon, a resolution of more than 200 points is used.

Equally, we use the 3rd order ambisonics signal to calculate a series of acoustic parameters (only the RT is fully developed and shown in this paper) and their directional characteristics; in this case a spherical t -design (Lebedev grid) of 146 points is used (Figure 5 (c) shows a denser example of 225 points) due to the fact that it is easier to derive spherical harmonics equations for these types of grids [10].

2.2.3 Visualization and new acoustical parameters proposed by A. Bassuet [2]

During our latest presentation at the Auditorium Acoustics conference 2023 in Athens, different colleagues have proposed that we look at the work of A. Bassuet [2] and the visualizations of DoAs proposed.

Given that the amount of information that we derive from the DoAs estimation using the SDM method is more detailed than the angular sectors demanded by the new visualization, the only thing we had to do is to collect all the data from our analysis and sum all the values corresponding to each sector.

Additionally, we have performed a normalization of the values of each sector based on the maximum of any time windows.

Figure 6 and Figure 7 below show the new visualizations and the way we have evaluated the new acoustical parameters LH (Low to High) and FR (Front to Rear).

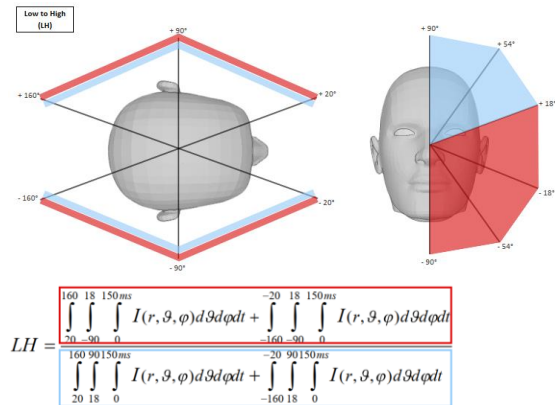


Figure 6 : LH parameter angle sectors of interest vs equation

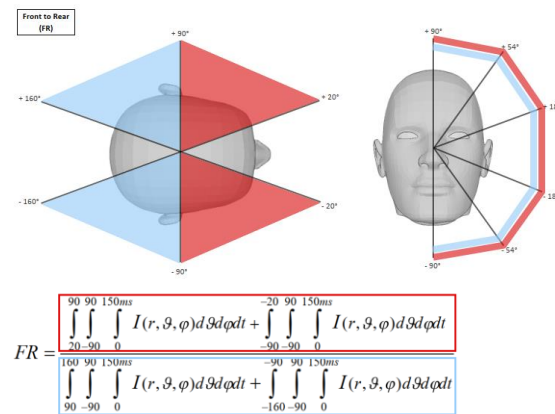


Figure 7 : FR LH parameter angle sectors of interest vs equation

It is important to mention that, in our view, the visualization proposed by A. Bassuet is easier to read when you want to understand the general tendency of the acoustic reflections without going in too much detail on the DoAs and the evaluation of problematic reflections.

Figure 8 below shows a measurement example implementing the new visualizations, the new acoustical parameters and some other ISO 3382 parameters.

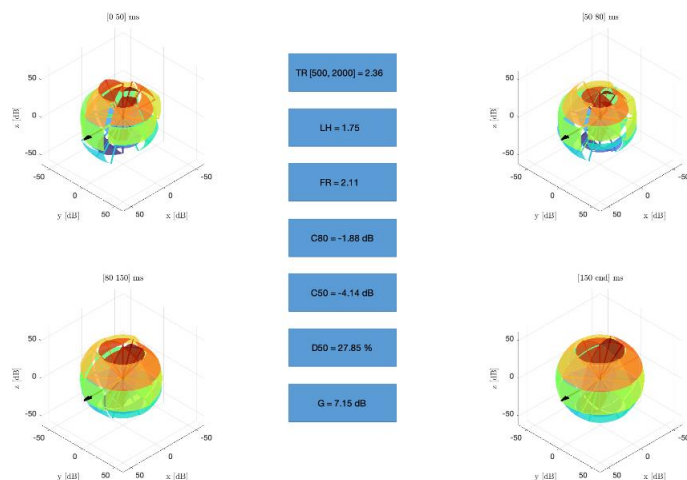


Figure 8 : A. Bassuet visualization, new acoustical parameter LH et FR, ISO 3382 parameters (RT, C80, C50, D50 and G)

2.2.4 Direction-dependent reverberation analysis [11]

According to the work by B. Alary et al. : *“Recent work has shown that reverberant sound fields have perceptually salient position and direction dependent characteristics which should be considered in six degrees of freedom (6DoF) sound reproduction.”*

This work has been pivotal in the development of the analysis of direction-dependent reverberation that also helped us building the foundation for future analysis of direction dependent ISO 3382 acoustic parameters.

Our approach is based on the processing of the 3rd order ambisonics signals (N3D ACN) of our microphone, calculated in MATLAB thanks to a script donated by Prof. Farina, and then processed to derive 146 IRs for each direction of the Lebedev grid described in paragraph 2.2.2.

In order to derive the IRs, we calculate the spherical harmonics coefficients for each direction of the 146 points Lebedev grid using equations in [10], we multiply each ambisonics channel with the corresponding coefficients and finally we sum the resulting IRs to create one single IR per direction. All the above is made through a matrix product as shown in the equation below:

$$\begin{bmatrix} Y_{0,0\ 1} & \cdots & Y_{3,3\ 1} \\ \vdots & \ddots & \vdots \\ Y_{0,0\ n} & \cdots & Y_{3,3\ n} \end{bmatrix} \times \begin{bmatrix} C_{0,0} \\ \vdots \\ C_{3,3} \end{bmatrix}$$

Where :

- $Y_{0,0}$ to $Y_{3,3}$ are the ambisonics channels corresponding to the 16 channels of the 3rd ambisonics file
- The subscript 1 to n indicates the samples
- $C_{0,0}$ to $C_{3,3}$ are the spherical harmonics coefficient derived from equations in [10]

Once the 146 IRs are derived, we calculate the T30 using the ITA toolbox [6] and then we create the visualization as shown below for each octave band between 125Hz and 4kHz.

Figure 9 below shows a measurement example, made at the auditorium M. Ravel in Lyon, where the reverberation is clearly dependent on the direction of arrival of the late energy.

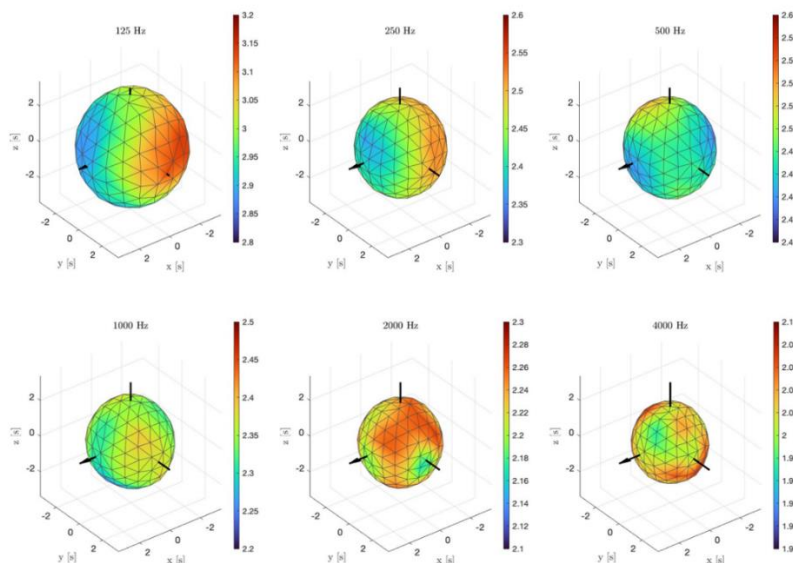


Figure 9 : Direction-dependent reverberation analysis – Auditorium M. Ravel (Lyon) 1st floor balcony

2.2.5 Octave band analysis of the SRIR

At the time of writing this paper and according to the authors' knowledge, there is no available tool that shows the DoAs estimation on a 3D hedgehog in 1/1 octave band analysis. Our approach is dependent on the SDM method and its algorithm.

The SDM method needs an omnidirectional microphone placed at the center of the array to calculate the pressure and the reference of each reflections, all other microphones are only used to evaluate the DoAs (using the TDOA evaluation method [8]).

It is very complex to process the microphones used to evaluate the DoAs through an 1/1 octave band filter, as those filters can create phase shift and other artifacts that will compromise the DoAs estimation.

On the other hand, the central omnidirectional microphone is only used to calculate pressure and timing of acoustic events, so it is considered resilient to all the artifacts introduced by those filters. The 1/1 octave band filters are in accordance with IEC 61260-1:2014.

By applying the 1/1 octave band to the central omnidirectional channel, we derive 6 reference IRs for each octave band that are then used as reference for the DoAs estimation at each frequency using the same microphones signals (in our case the 19 signals corresponding to each capsule of the ZYLIA microphone).

Figure 10 below shows the result of a measurement (Source-receiver distance 1m) in the anechoic chamber at the Department of Acoustics at the Ecole Centrale de Lyon; dB values normalized to the maximum peak value in the global results.

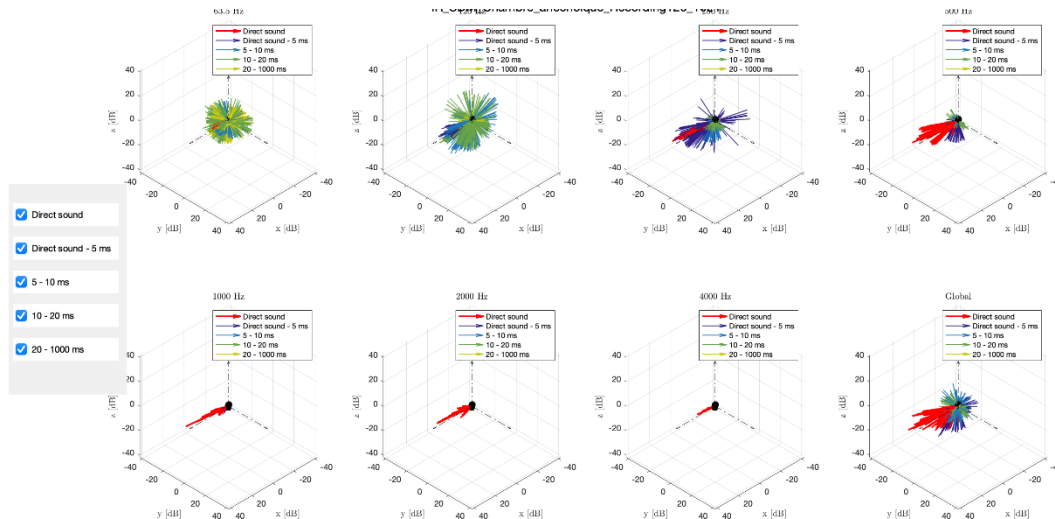


Figure 10 : Octave band analysis of the SRIR – Source – receiver distance 1m

Analysis and discussion of the measurements' result is described in the next paragraph.

2.2.6 Measurement system calibration in anechoic chamber

At the end of August 2024, we have performed a series of measurements to characterize and verify the quality of the herisSon measurement chain and to calibrate it in order to be able to evaluate the acoustic parameter G (Strength) for all our measurements.

Figure 11 below shows the set-up in the anechoic chamber of the department of Acoustics at the Ecole Centrale de Lyon.

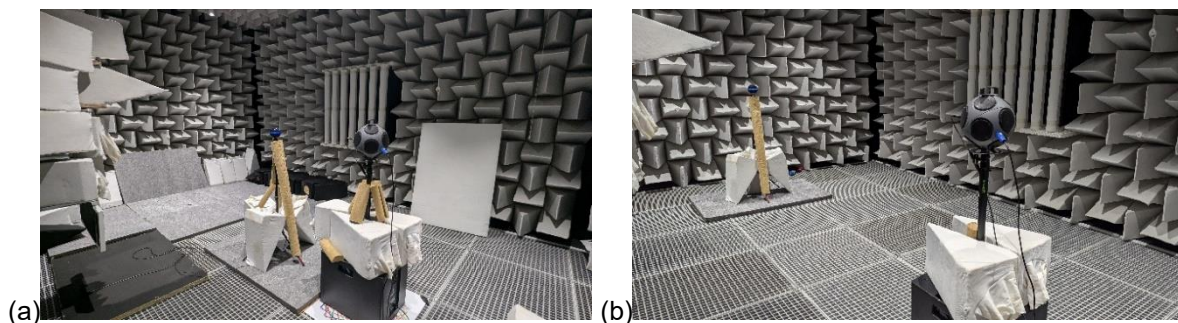


Figure 11 : Measurements in anechoic chamber – (a) Source-Mic 1m, (b) Source – Mic 3.73m

The source is composed of a LOOKLINE dodecahedron and a subwoofer (15") with a cutoff frequency of 100Hz.

Measurements were performed in different set-ups as listed below :

- Microphone – Source distance 1m (Figure 11 (a)) (white panel in the picture is a melamine foam panel)
- Microphone – Source distance 3.73m (Figure 11 (b))

Additionally we have also performed a series of measurements for the calibration of the measurement chain for Strength (G) measurements in accordance with [12], rotating the source every 15° with a source-receiver distance of 3.73m.

Figure 12 below shows the rotational plan used as a reference for the source rotation.

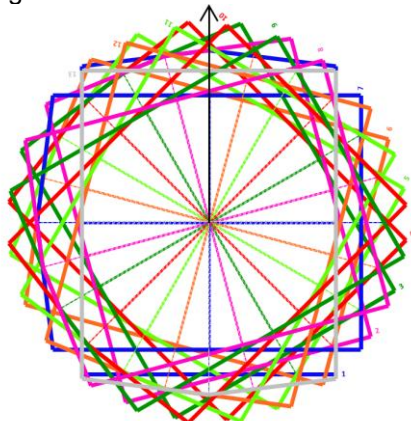


Figure 12 : Rotational plan for source rotation

Figure 13 to Figure 14 below show the results of the measurements in the two different scenarios.

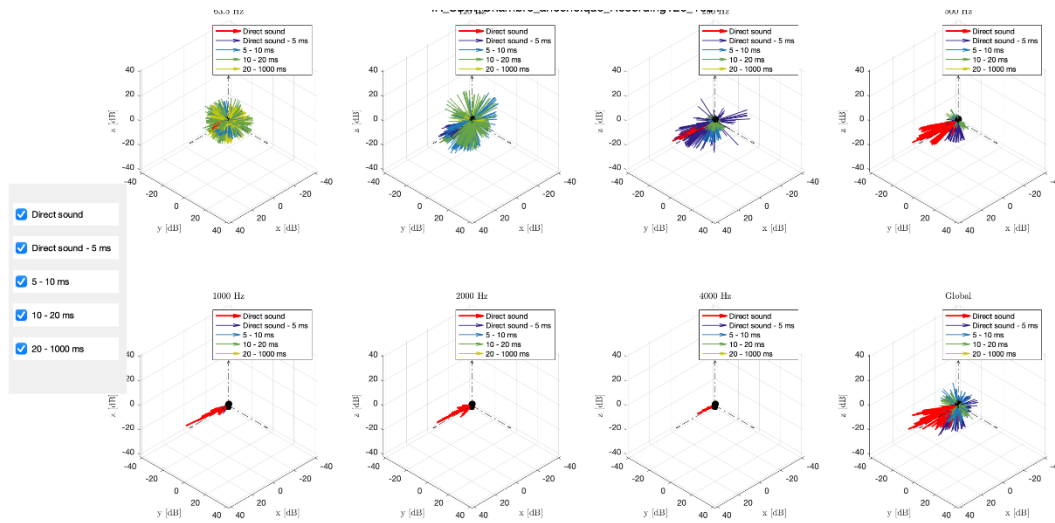


Figure 13 : Measurements in anechoic chamber – Source – receiver distance 1m

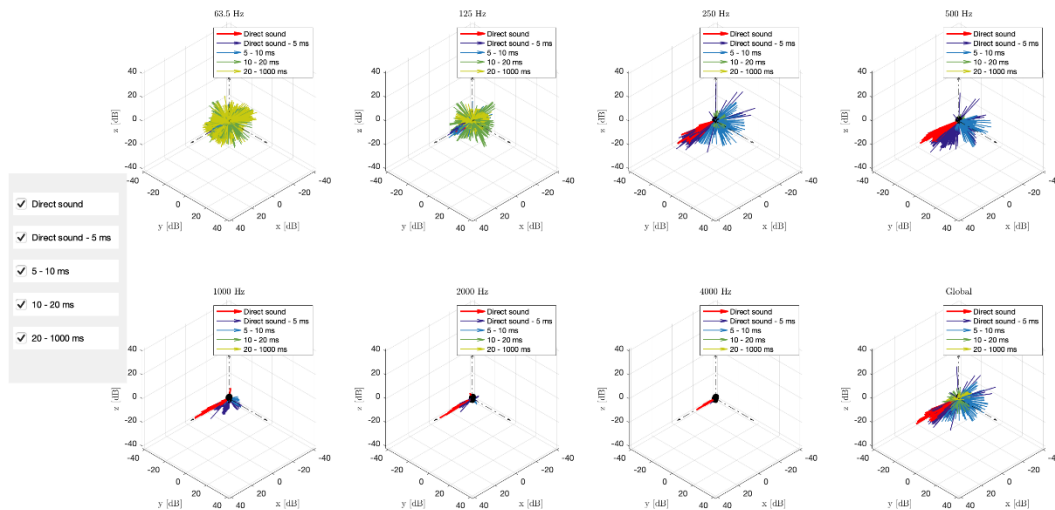


Figure 14 : Measurements in anechoic chamber – Source – receiver distance 3.73m

From the results in Figure 13 to Figure 14 we can derive the following:

- The cut-off frequency of the anechoic chamber is around 200Hz, which is clearly visible on the 63Hz, 125Hz and 250Hz results by the amount of energy in the late part of the IR (after 10ms from the direct sound)
- In the 250Hz and 500Hz results, in the case of SRC-REC distance of 3.73m, we can see a series of reflections coming from the back and from the left of the microphone and from the floor; equally in the case of SRC-REC distance of 1m, we can still see some reflections from the back at 250Hz. We are still investigating the reasons of this but in our view those “parasite” reflections are a byproduct of different factors:
 - On the left of the microphone there is a big silencer (the anechoic chamber is used for aeroacoustics tests) and it could provide some reflections

- Our source has a resonance frequency of about 350Hz that can be heard in the raw recordings, so we suppose that this resonance creates some false reflections by spatial spread and spectral distortion of the acoustic events (to be investigated further) in particular at the back of the microphone
 - The floor is not perfectly anechoic due to the presence of a metal grid of a certain thickness (not known)
- In the case of SRC-REC distance of 1m and at 500Hz, we can clearly see the effect of different drivers of the source influencing the detection of the direct sound
- Results above 1kHz are as expected

The results in the anechoic chamber present some reflections that are not expected for this type of environment and more investigation is needed to understand the reasons; we also believe that the room was not perfect for the job as too many elements are present that diverts its acoustic response from an anechoic environment.

3 HOW HERISSON HAS HELPED US TO EVALUATE THE PERFORMANCE OF ELECTRO-ACOUSTIC SYSTEMS IN AUDITORIUMS – MEASUREMENTS RESULTS

This section shows a series of measurements results performed in auditoriums designed for classical and general acoustic music that have been equipped with electro-acoustic systems.

We present two cases:

1. The case of a well-known concert hall in the north-west of France where we have worked to help the management to make a choice between two loudspeakers' manufacturers for the new electro-acoustic system to be installed in the venue;
2. The case of an immersive ambisonics system installed in a well-known venue in Belgium where we demonstrated the capability of our measurement system to evaluate an immersive sound system.

3.1 Concert Hall in France

The hall under study is a concert hall in the north west of France that is well known for its orchestra but that is also used for stand-up comedy and amplified music performances.

Our brief was to provide a series of acoustic measurements that can help decide which electro-acoustic system manufacturer would be chosen by the management to provide the new PA system in the hall.

We worked to assess how the electro-acoustic systems, in a stereo configuration, interact with the acoustics of the hall and provide the best performance possible in the challenging acoustics.

Figure 15 below shows the venue stage with its originally installed sound system on the left and a view of the audience on the right.

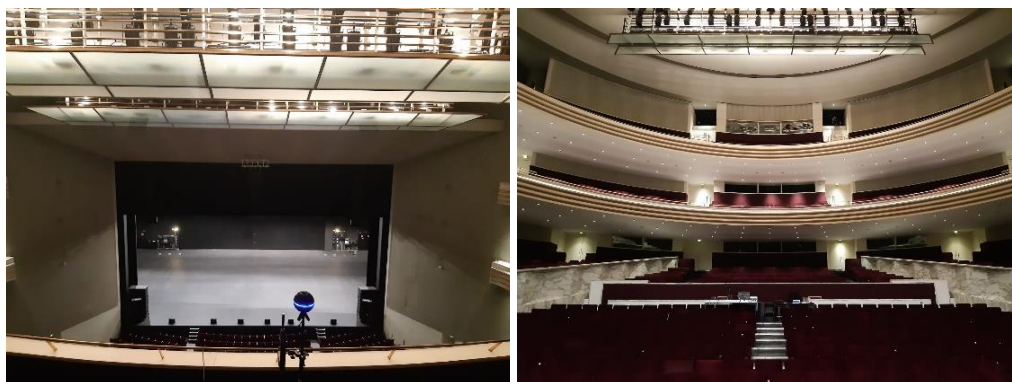


Figure 15 : Venue stage with its originally installed sound system (L) and audience (R)

The sound system was composed of 2 line arrays at the top left and top right of the stage, two lines of subwoofer on the left and right of the stage and a series of fills around the venue.

We performed a series of benchmark measurements to characterize the performance of the existing sound system and the acoustics of the concert hall (in this case using our dodecahedron + subwoofer).

Figure 16 below shows the T_{30} measured at the ground floor ("Zone RDC") and inside the stage tower ("Scène").

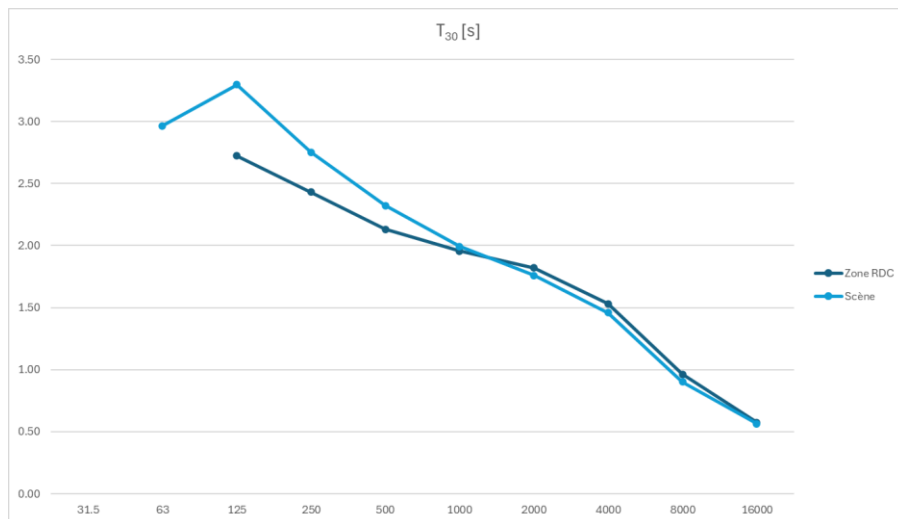


Figure 16 : T_{30} measurements at the ground floor and inside the stage tower (average over 10 measurements)

In this study we present the results for two receiver positions (Figure 17) :

- Seat M7 at ground floor
- Seat C43 at second floor balcony

Source configurations are :

- Dodecahedron on stage Figure 17
- Existing PA system
- Manufacturer 1
- Manufacturer 2

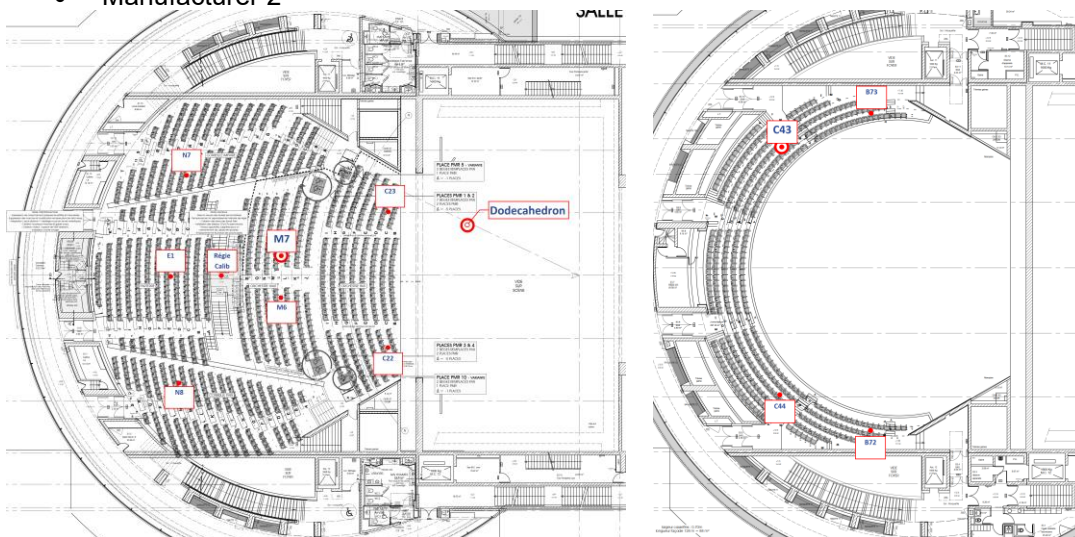


Figure 17 : Receivers' positions with indicated seat number

The aim of the study is to inform the client on how each electro-acoustic system performs in the challenging acoustic environment.

Figure 18 shows the results of the measurement in seat M7 using the dodecahedron on stage, while Figure 19 shows the same results in 1/1 octave band analysis.

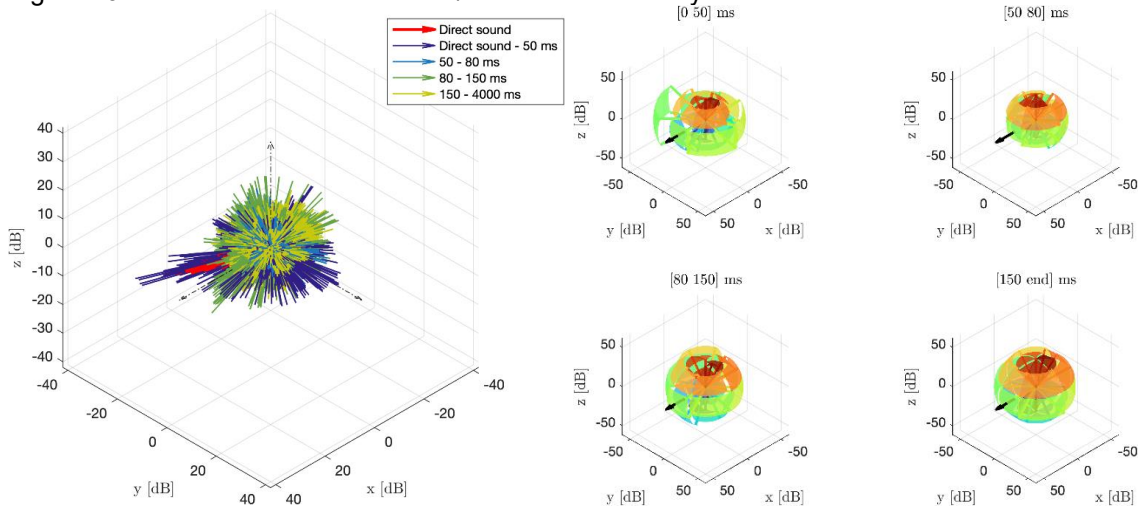


Figure 18 : Source Dodecahedron, Receiver M7 – overall 3D response

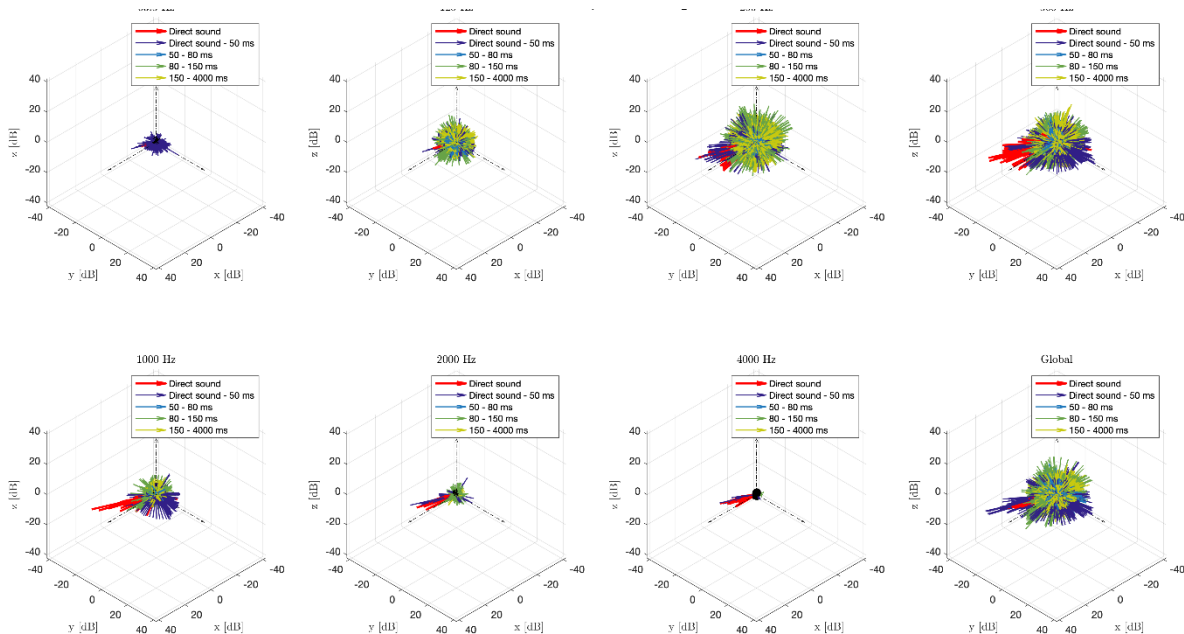


Figure 19 : Source Dodecahedron, Receiver M7 – 1/1 octave band 3D response

The results in Figure 18 and Figure 19 show a good amount of early (<50ms) lateral reflections, especially from the left side of the microphone where the closest wall is located, mostly in the 500Hz et 1kHz frequency range.

Equally, we can spot some late lateral energy (>80ms) coming from the top right but at about 20dB less than the direct sound and 10dB less than the early lateral energy that should not create any issues but it should be taken into account when analysing the results with the electro-acoustic system.

Figure 20 shows the results of the measurement in seat M7 using the existing electro-acoustic system, while Figure 21 shows the same results in 1/1 octave band analysis.

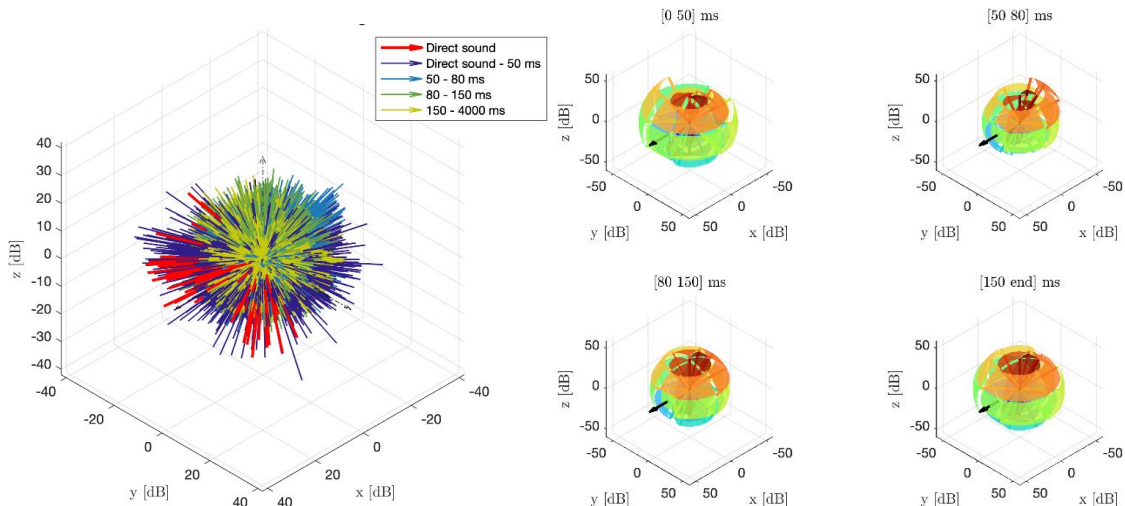


Figure 20 : Existing electro-acoustic system, Receiver M7 – overall 3D response

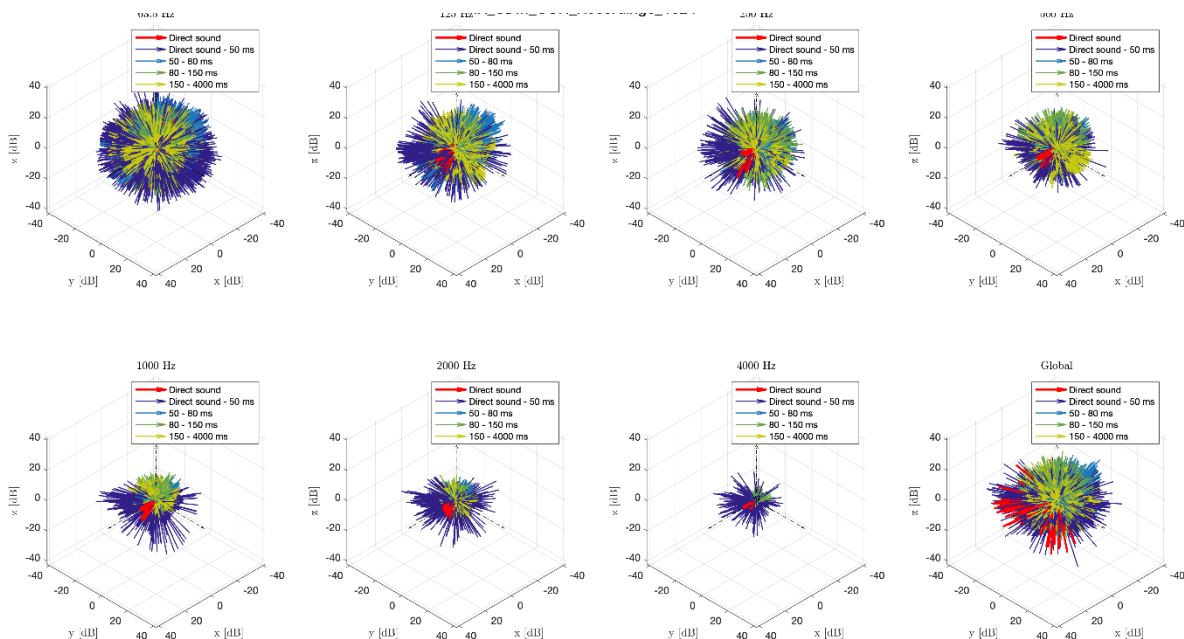


Figure 21 : Existing electro-acoustic system, Receiver M7 – 1/1 octave band 3D response

The results in the case of the existing electro-acoustic systems show a clear difference in the response of the hall in the same location as the previous measurements:

- Direct sound is perceived as coming from the front with a stereo spread typical of the sound system in study
- Low frequencies (63Hz and 125Hz) are louder and show a good amount of early reflections (<80ms) as well as some late reflections (between 80 ms and 150ms) mostly from the sides but with some presence from the back of the microphone
- Lateral reflections remain quite predominant in the higher frequencies (>250Hz) with the early ones (<80ms) louder than late ones (between 80 ms and 150ms)

- Overall the existing electro-acoustic systems seems to be well designed for the acoustics of the hall
- Figure 22 below shows the direction-dependent T30 with small directional characteristics (we can see a slight longer T30 along the front and back of the microphone) mostly due to the stage tower and the curved back wall.
- Additionally, we can see that at 16kHz the direction dependent T30 shows a strange behaviour; during the measurement we can hear a delayed test signal in the very high frequencies coming from the system (like a second signal superposing with a delay of about 600ms) that is most probably due to heavy DSP processing.

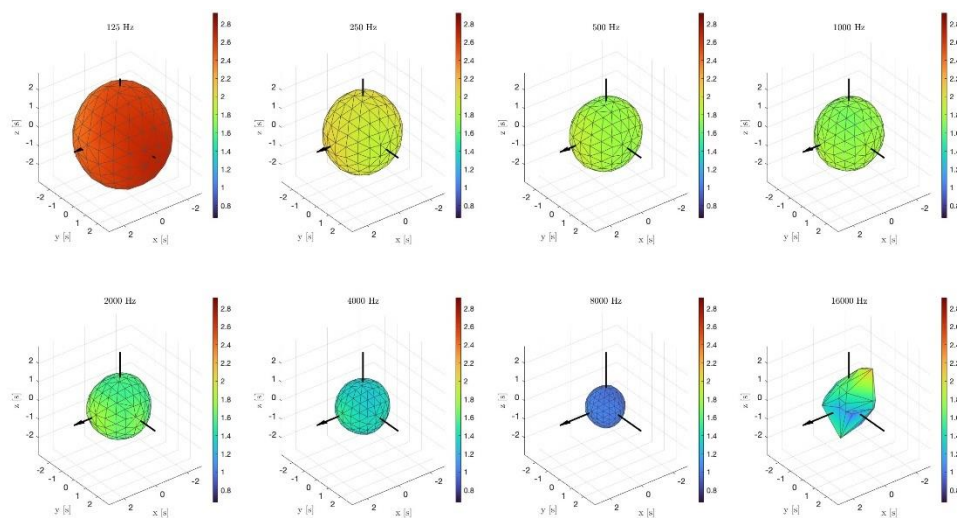


Figure 22 : Existing electro-acoustic system, Receiver M7 - Directional T30

Next step is analysing the results for the electro-acoustic systems proposed by the two manufacturers as a substitute to the old existing system.

Figure 23 below shows the overall 3D response for Manufacturer 1 (on the left) and Manufacturer 2 (on the right).

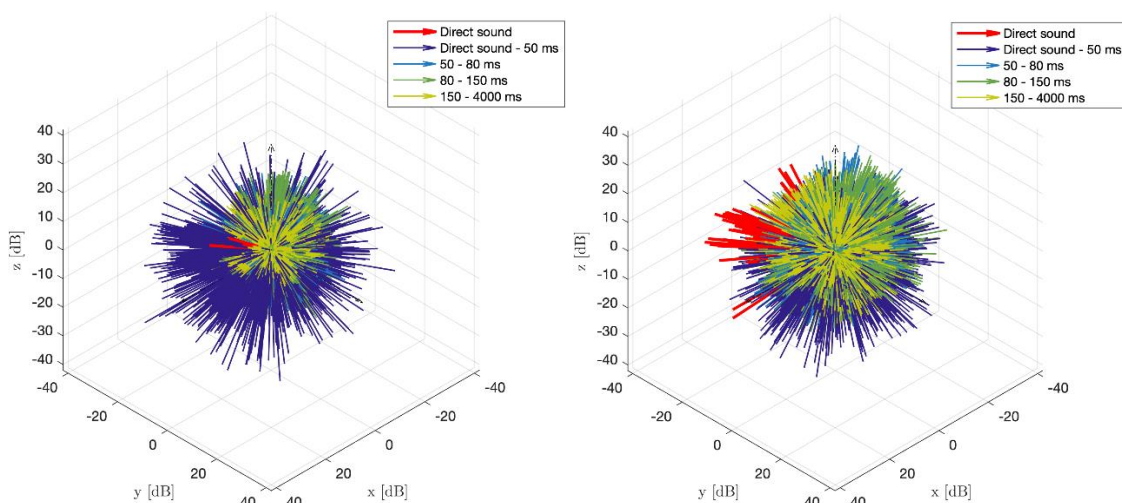


Figure 23 : Manufacturer 1 (L) and Manufacturer 2 (R) electro-acoustic system, Receiver M7 – overall 3D response

Figure 24 et Figure 25 show the 1/1 octave analysis for Manufacturer 1 and Manufacturer 2 respectively.

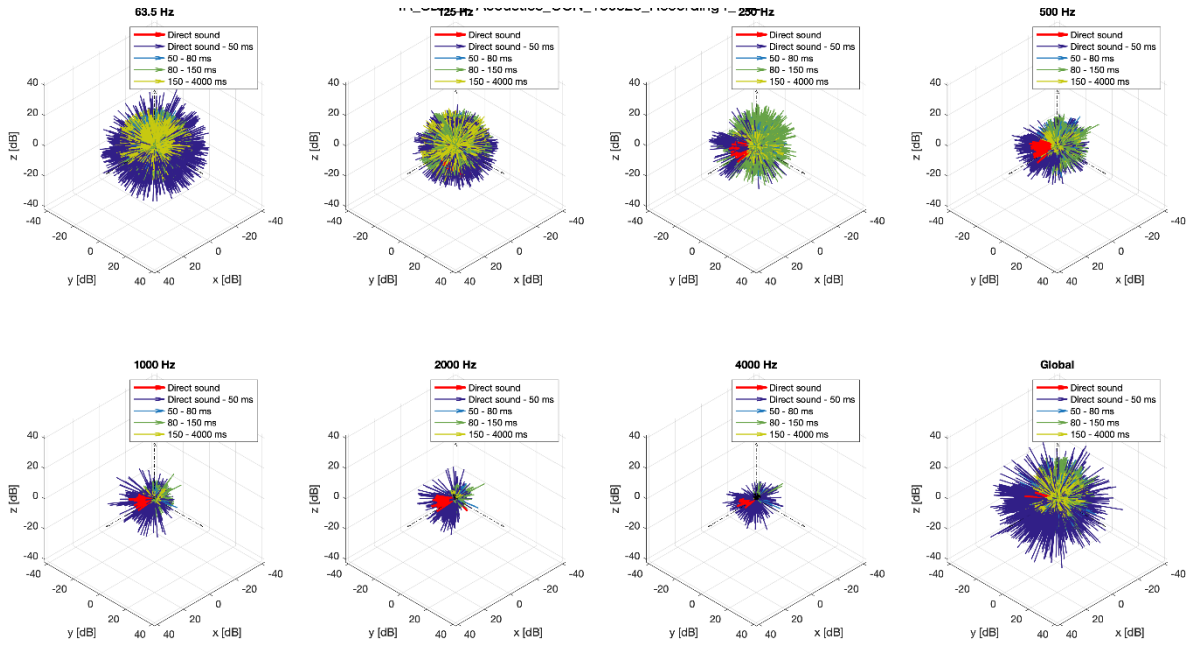


Figure 24 : Manufacturer 1 electro-acoustic system, Receiver M7 – 1/1 octave band 3D response

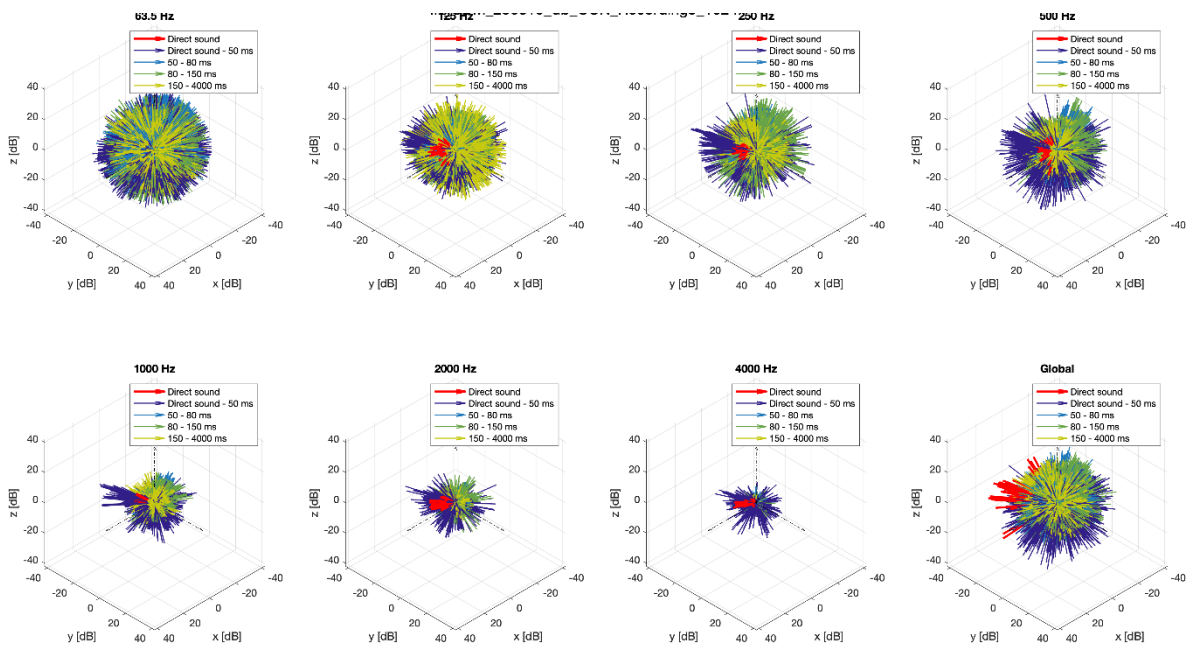


Figure 25 : Manufacturer 2 electro-acoustic system, Receiver M7 – 1/1 octave band 3D response

Results at the receiver M7 location show that:

- Alignment and coherence between the stereo pair is better for Manufacturer 1 as our tool detect a correct virtual source in the middle of the loudspeakers, while for Manufacturer 2 it detects a more prominent right loudspeaker
- Manufacturer 1 presents more early reflections (<80ms) than Manufacturer 2
- Manufacturer 1 excites less late energy (>150ms) than Manufacturer 2, especially in the low frequencies (63Hz et 125Hz) which allows better directivity control for the subwoofers

Our first analysis shows already that the design proposed by Manufacturer 1 is more adapted to the hall, but we would like to look at the results on the balconies, in particular the second floor balcony, where coverage with a stereo system could be trickier.

Figure 26 below shows the overall 3D response for Manufacturer 1 (on the left) and Manufacturer 2 (on the right) for the second floor balcony seat position C43.

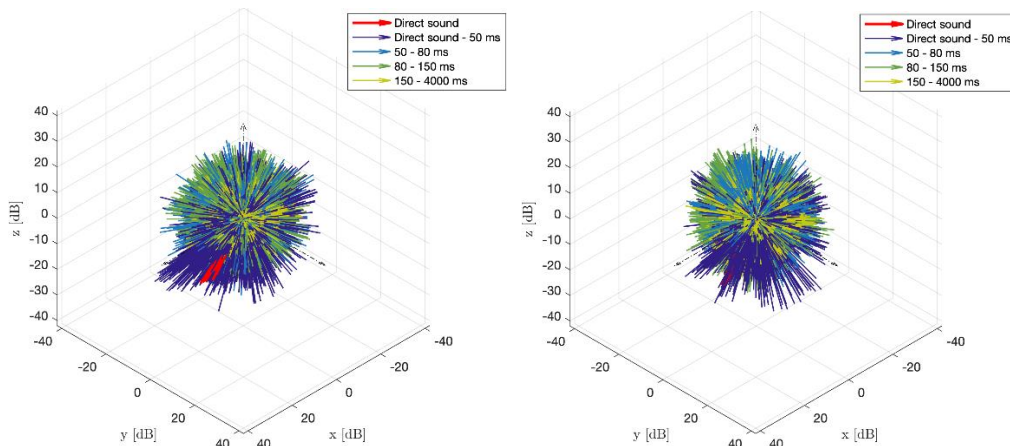


Figure 26 : Manufacturer 1 (L) and Manufacturer 2 (R) electro-acoustic system, Receiver C43 – overall 3D response

Figure 27 and Figure 28 show the 1/1 octave analysis for Manufacturer 1 and Manufacturer 2 respectively for the second floor balcony seat position C43.

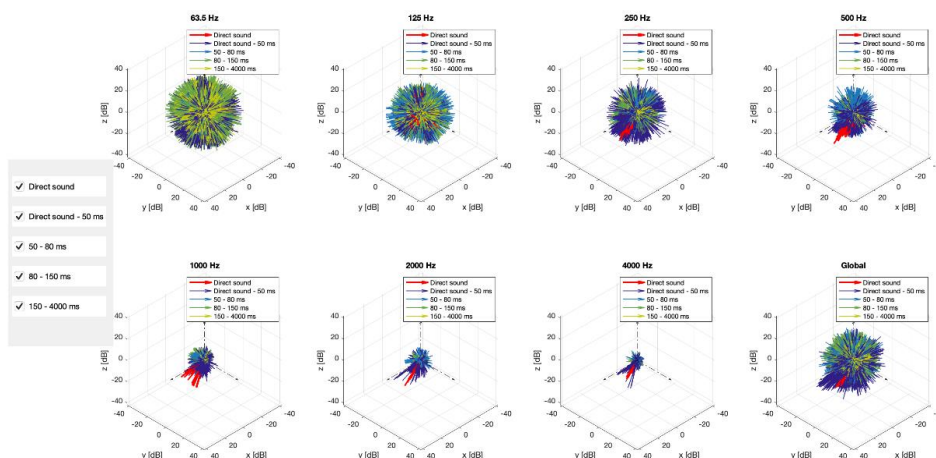


Figure 27 : Manufacturer 1 electro-acoustic system, Receiver C43 – 1/1 octave band 3D response

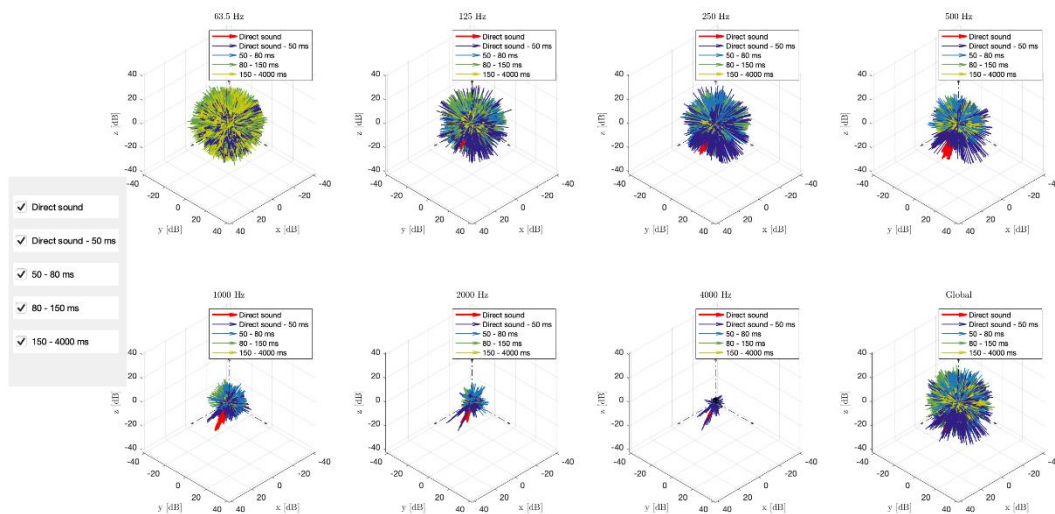


Figure 28 : Manufacturer 2 electro-acoustic system, Receiver C43 – 1/1 octave band 3D response

Results at the receiver C43 location show that:

- Alignment and coherence between the stereo pair is correct for both manufacturers as our tool detect a correct virtual source towards the left of the microphone
- Early reflections (<80ms) are equally distributed between the manufacturers but Manufacturer 1 seems to have a better directivity control (less side reflections on the left)
- Manufacturer 1 excites less late energy (>150ms) than Manufacturer 2, especially in the low frequencies (63Hz et 125Hz), which allows better directivity control for the subwoofers

All results show how herisSon can determine how an electro-acoustic system performs in a challenging acoustic environment by providing informative visualizations to determine crucial acoustic response.

The interpretation of the 3D visualizations is still tricky to people who are not familiar with the technique and can sometime lead to misinterpretations.

We aim to improve this in the next versions of the tool but the initial results are very promising and our conclusions have been also supported by a subjective listening test of the two systems by a jury of 30 people (professionals et general public) that have chosen Manufacturer 1 as the preferred system for the hall.

3.2 Concert Hall in Belgium

The hall under study is a concert hall in the north west of Belgium that is well known for its acoustics, who has been subjected to a renovation by a respected acoustics firm in the 90s.

One of the authors has worked with a British artist to help her deliver her immersive sound performance by designing the electro-acoustic system and by providing the software for the ambisonics encoding and decoding.

The systems is comprised of 8 loudspeakers and 4 subwoofer distributed around the audience as shown in Figure 29 below.

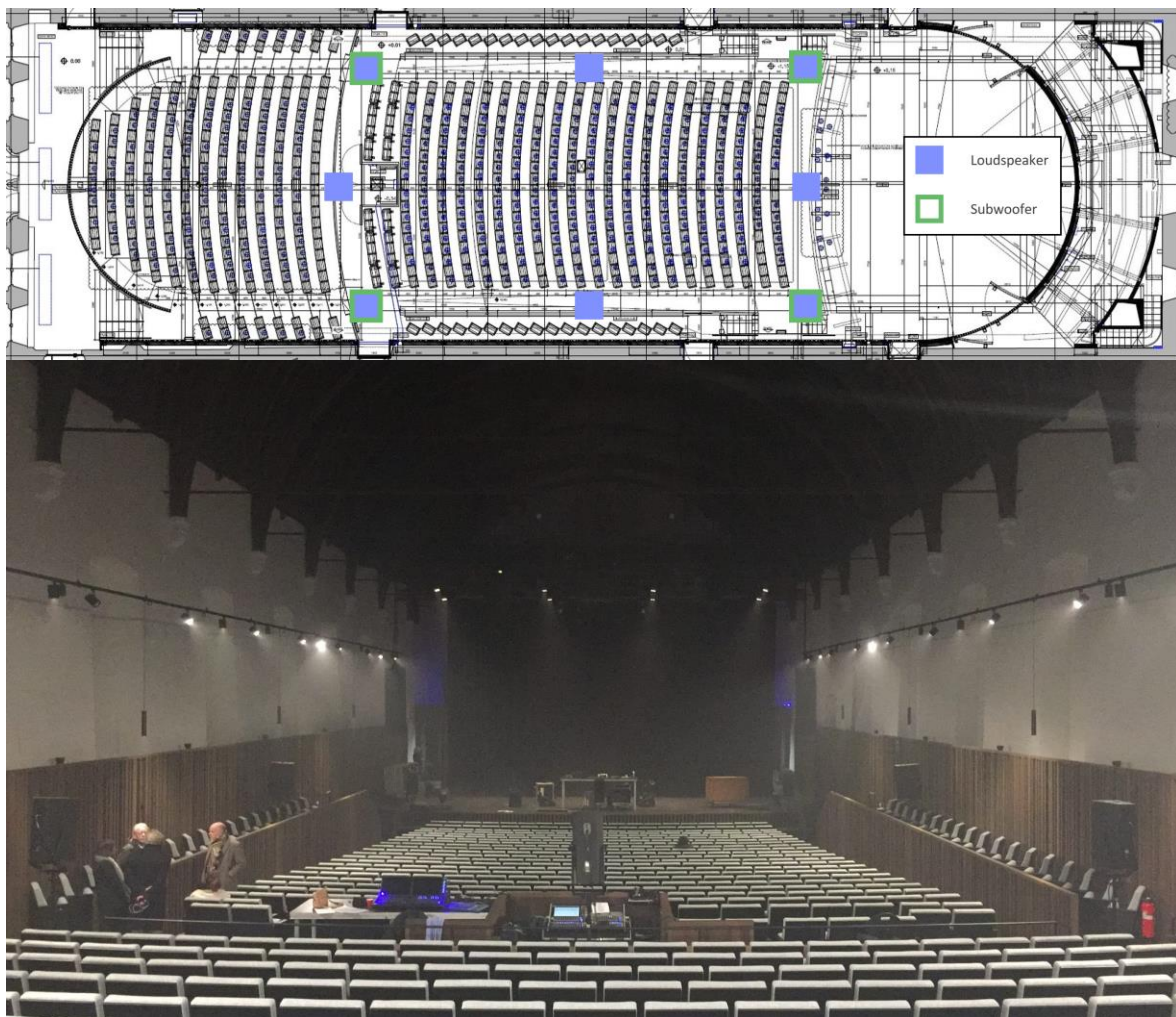


Figure 29 : Loudspeakers and subwoofer locations

The aim of the test is to check if herisSon is able to detect the direction of the direct sound according to a virtual source's placement on the ambisonics encoder and how the acoustics of the hall influences the performance of the immersive system, considering that, at the time of writing this paper and to our knowledge, no acoustic standard exist that defines any acoustic parameters for the task.

No other acoustic measurements have been performed to evaluate the acoustics of the hall.

The ambisonics decoder is AllRAD (3rd order) with correction for the non-isotropic distribution of the loudspeakers, using the IEM plug-in suite [13].

Figure 30 below shows virtual sources and the receiver location, including the direction of the front axis of the receiver.

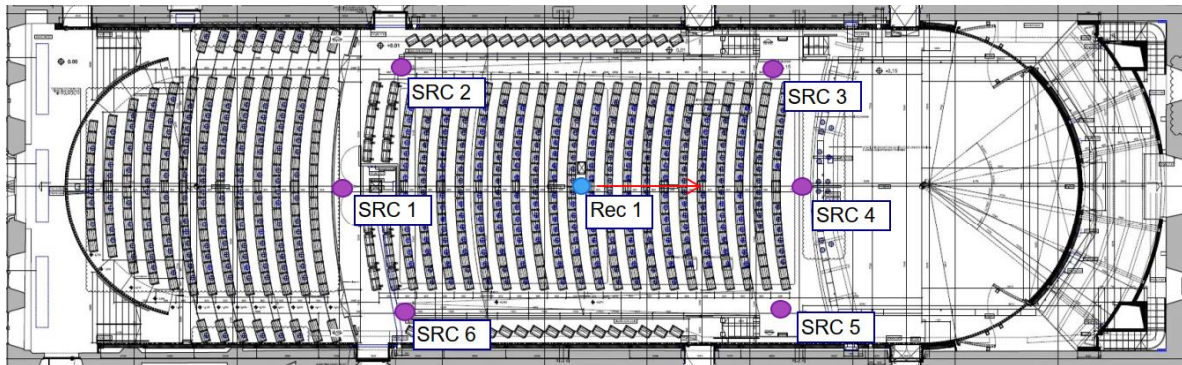


Figure 30 : Virtual sources and receiver locations.

The first measurement shown is the one for the position of the source SRC 4 for validation as it is the easier position to evaluate (straight in front of the microphone).

Figure 31 below shows the overall 3D response while Figure 32 shows the 1/1 octave analysis.

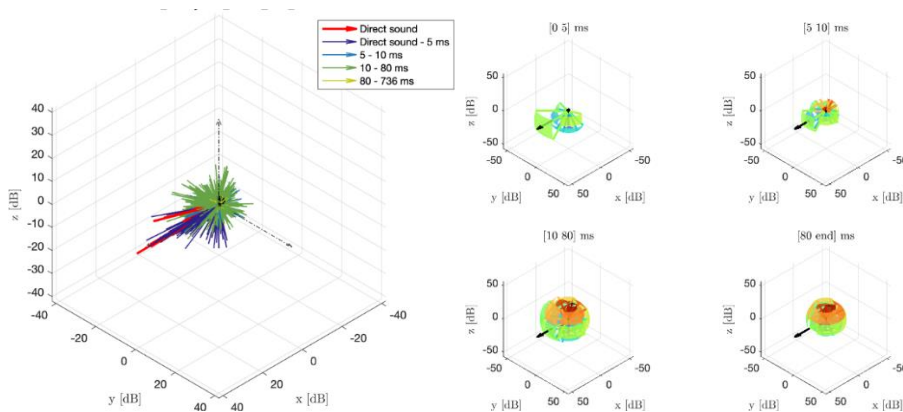


Figure 31 : SRC 4 – overall 3D response

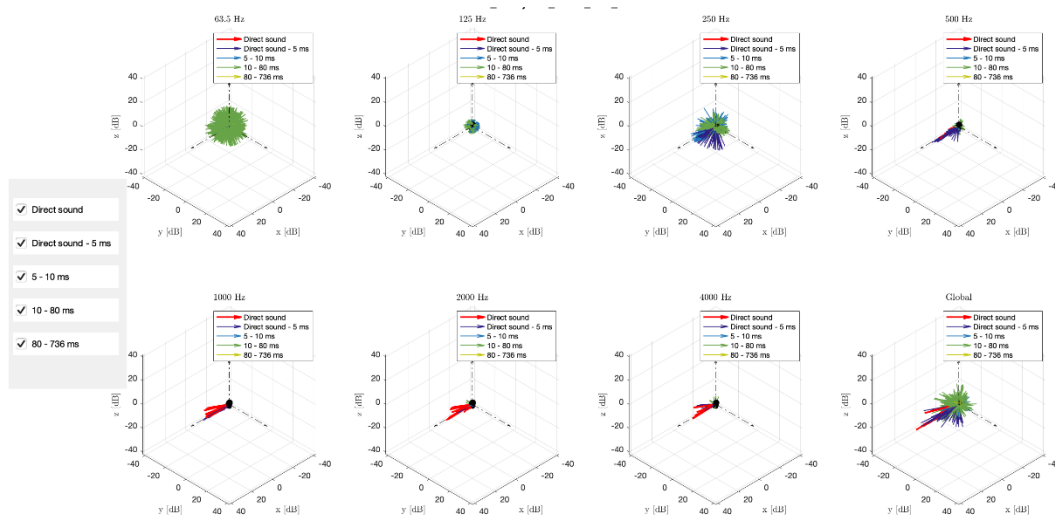


Figure 32 : SRC 4 – 1/1 octave band analysis

The results for SRC 4 show that herisSon is capable of correctly detecting the direction of the direct sound; the IR has been cut quite short (736ms to reduce the effect of the reverberation on the analysis).

The next results is for the position SRC 6 (back right of the microphone) with the full IR of about 3s; this position is chosen as it is a challenging location due to the non-isotropic distribution of the loudspeakers (the loudspeakers on the left and the right of the microphone are closer than the ones to the front and back).

Figure 33 below shows the overall 3D response while Figure 34 shows the 1/1 octave analysis.

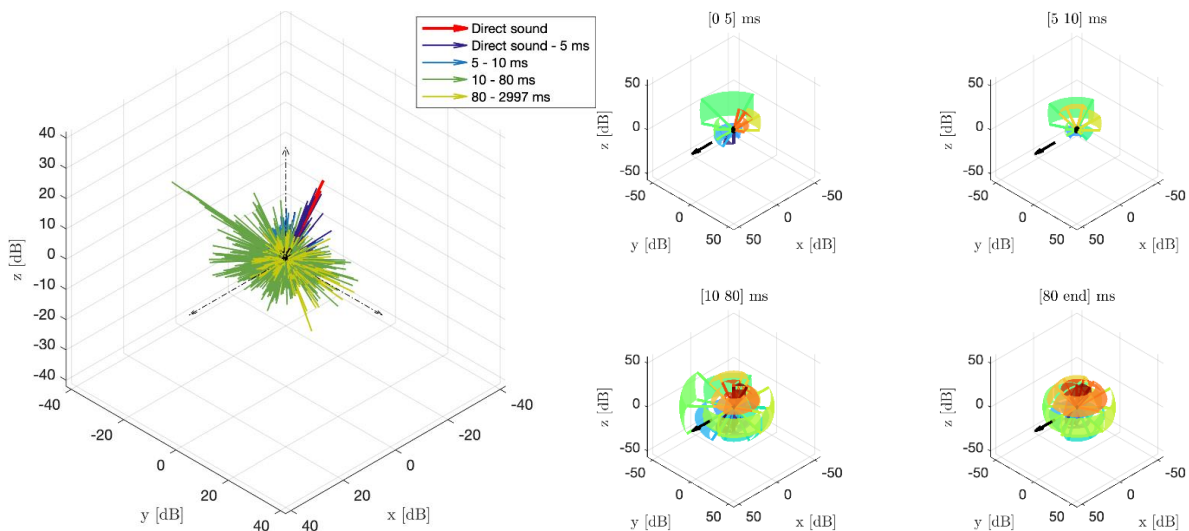


Figure 33 : SRC 6 – overall 3D response

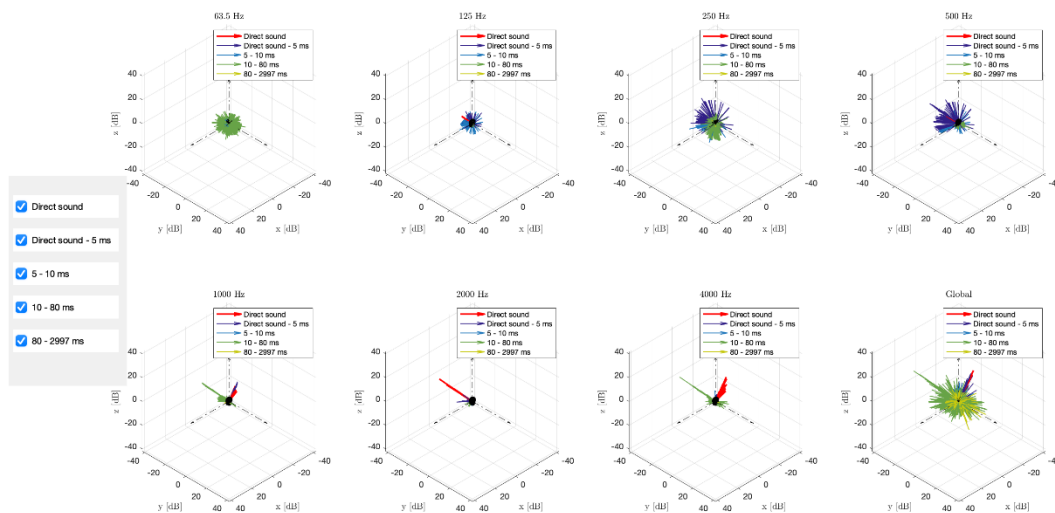


Figure 34 : SRC 6 – 1/1 octave band analysis

The results for SRC 6 show again that herisSon is capable of correctly detecting the direction of the direct sound.

Additionally we can clearly see the strong signal coming from the loudspeaker on the right (closer to the microphone) and the effect of the compensation for the non-isotropic distribution of the loudspeakers (delay of about 19ms) that allows the measuring system to correctly evaluate the location of the virtual source.

Only the result at 2kHz in Figure 34 shows an erroneous evaluation of the virtual source position; this is due to the fact that the filter reduces drastically the levels of the IR at this frequency so the onset algorithm is not able to determine the correct start of the IR but it evaluate that it starts at the loudest signal (closer loudspeaker).

The measurement also shows that the reverberant field is relatively anisotropic and that its level is not detrimental to the correct localization of the virtual source as the location of the loudspeaker system at about head height helps with reducing the excitation of the reverberant field of the hall.

In order to confirm the findings for SRC 6, we have performed another measurement in position SRC 2 turning off the compensation for the non-isotropic distribution so, effectively, having no delay applied to any loudspeaker.

Figure 35 below shows the overall 3D response while Figure 36 shows the 1/1 octave analysis.

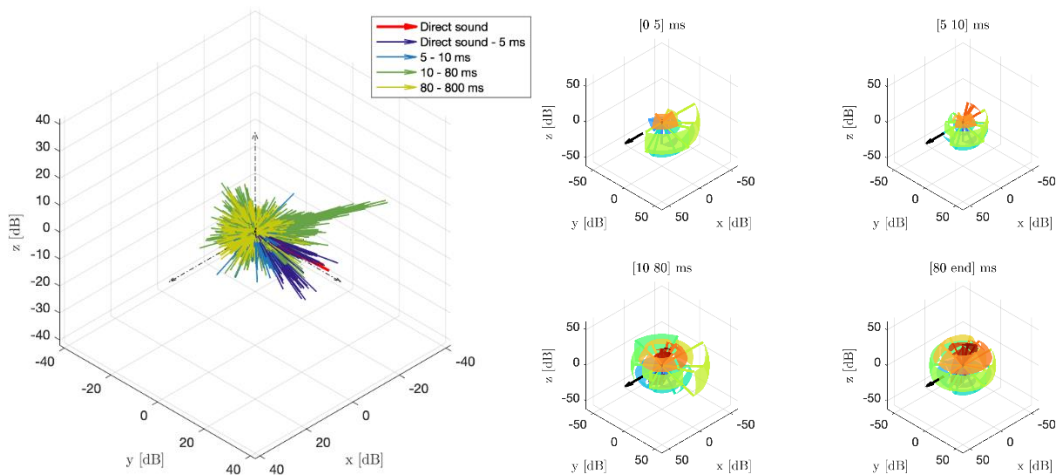


Figure 35 : SRC 2 – overall 3D response

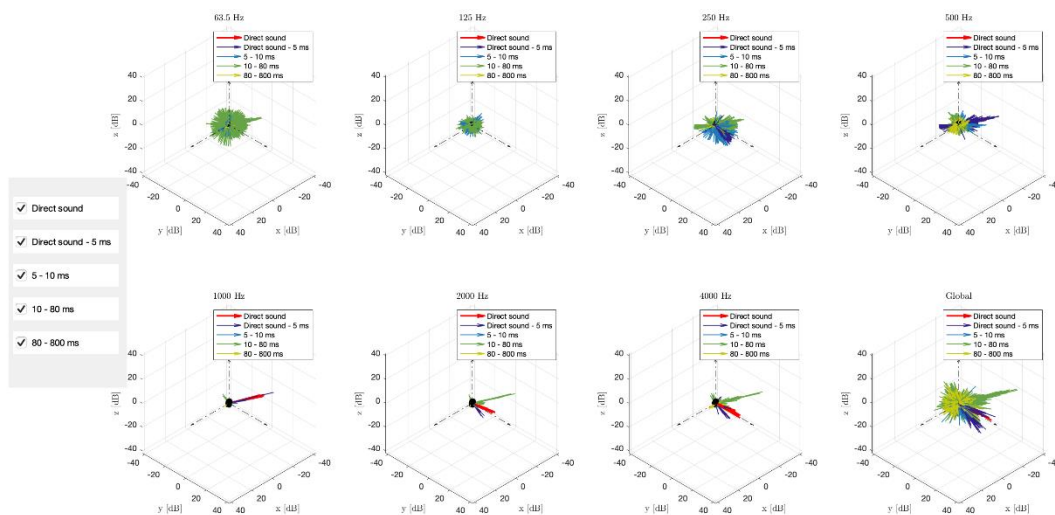


Figure 36 : SRC 2 – 1/1 octave band analysis

The results for SRC 2 show that herisSon is capable of correctly detecting the direction of the direct sound coming from the closer loudspeaker and so effectively detecting a fault in the decoding process of the loudspeaker systems where the compensation correction has been turned off.

4 CONCLUSIONS

In this paper we presented the new developments of our SRIRs measurement and analysis tool notably a better evaluation of the direct sound, a newly and more regular distribution of the receiver sphere, the visualization proposed by A. Bassuet [2], a direction-dependent reverberation analysis [3], octave bands analysis of the directional of arrivals and the calibration of the measurement system in anechoic conditions.

All developments have been very useful in the analysis process and to inform our understanding of particular problems we had in the previous iteration of our tool [1], notably the presence of “parasite” acoustic events in the measurements in anechoic conditions (Figure 13 and Figure 14).

The 1/1 octave band analysis have helped us to isolate the events coming from unexpected directions to the 250Hz – 500Hz range that corresponds with the resonance frequency of our dodecahedron source (around 300Hz) and so they have given us a possible path to follow to resolve this issue.

Equally the new visualization developed by A. Bassuet [2] have been very useful to understand the general behaviour of the acoustic field in 3D, compared to the more complex 3D hedgehog visualization, and the direction-dependent reverberation analysis have opened new paths to the analysis of acoustic ISO 3382 parameters in 3D.

In the second part of the paper, we have shown its application to evaluate the performance of electro-acoustic systems in auditoriums by the presentation of measurements in two different halls equipped with a stereo and an immersive electro-acoustic system respectively.

The measurements have shown that our approach is able to determine the performance of a stereo PA system in a hall by analysing its 3D response in different locations by evaluating the coherence and better alignment of the stereo pair of loudspeakers, the ratio of beneficial early lateral reflections to late reflections, the reduced excitation of the reverberant field and a better low frequency response.

Additionally, herisSon has been used to analyse an immersive sound system (ambisonics), inside a hall designed for classical and acoustic music in general, where the results have been promising in the research of new parameters for the evaluation of the reproduction quality of these systems.

Further development will focus on fine tuning the DoAs estimation, by reducing the parasite acoustic events and by improving the precision of our measurement chain.

Also, we would like to start analysing ISO 3382 direction-dependent acoustic parameters, in particular the energetic and statistical acoustical indices (C_{50} , C_{80} , G , G_{early} , G_{late} , etc) and to develop further the analysis of immersive sound systems.

5 ACKNOWLEDGEMENT

The authors would like to thank all our colleagues who have shown interest in our work at the Auditorium Acoustics conference in 2023, in France and Belgium for all their comments, suggestions, corrections, and general enthusiasm that have pushed us to go further and further in our research.

Special thanks to Prof. Angelo Farina who has very kindly and patiently helped us every time we have been asking questions or looking for help to put together missing pieces in the code. His availability and knowledge of acoustics is a gift to our profession and we will be missing him dearly.

We would like to dedicate this paper and all our work on SRIR to his memory.

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