DEVELOPMENT OF SOFTWARE TO GET A THREE DIMENSION IMPULSE RESONPONSE GRAPHIC INTERPRETATION

Joaquin Guiñazú
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email: joaquin_guinazu@hotmail.com
Alan Nicolás Godoy
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email: godoyanicolas@gmail.com
Nicolás Pelella
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email: nicolaspelella@gmail.com

The purpose of this article is to detail the procedure that was carried out to develop a software that allows the processing of impulse responses in three dimensions. As main functions, this software supports audio files that are recorded both in format A and in format B, since it has a conversion algorithm from A to B format. Then, as the main features, the software allows the visualization of a hedgehog graph, the impulse response in two dimensions, and a table showing the intensity values as a function of time. The hedgehog is a three-dimensional graph that represents the intensity vectors according to where the reflection in space comes from. Finally, the software allows the user to configure some parameters, such as filtering intensities based on a threshold, percentage of overlap of the window to calculate the intensity, configure the window size, and specify the arrival time of the direct sound. The software code was written in Python and its purpose is to facilitate the user understanding of the acoustic characteristics of an enclosure based on the processing of impulse responses recorded with soundfield or ambisonic microphones and that it's features allows to make a deep analysis of sound field evolution.

Keywords: Hedgehog, Impulse Response, Python, Soundfield

1. Introduction

One of the most used techniques to characterize a room is to obtain its impulse response in a monophonic or binaural way. It allows obtaining acoustic parameters of the room such as clarity or reverberation time. However, this type of technique does not allow obtaining information regarding the origin of the reflections in space. In the present work, the characteristics and development of a software processing impulse responses obtained in ambisonic format are specified. It is developed in Python and seeks to allow the user to have a representation of the temporal and spatial content of the reflections that make up the impulse response.

As a starting point and guide, the thesis work carried out by the Sound engineer Federico Cacavelos "IR360" was taken, as well as commercial software that have the same purpose: IRIS.

It seeks to be a more robust work in relation to the aforementioned, but without losing important components in quality in the results. It will seek to continue with a constant development of this software, so that, in the future, it can become a reliable tool and free for the community of scientists and researchers in the field.

2. Theorical FrameWork

2.1 Impulpse Responses

The characteristics of a linear, time-invariant component of any system are fully described by its impulse response $h(t)$. It is thus desirable to acquire the impulse response as accurately as possible. In traditional impulse response measurement, periodic pulse and Maximal-Length Sequence (MLS) are often used as excitation signals[1]. The impulse response for a room will typically be an oscillatory signal with a large number of periods. The envelope of the signal will be irregular but typically have a fast attack-time and an exponential decay. The impulse response may be measured as the response of the room to a very short acoustic pulse. However, it will in most cases where sources other than a loudspeaker are used, be difficult to have sufficient control of the spectral content and the directional characteristics of the excitation. To obtain the required control of the excitation signal, the impulse response is in most practical cases obtained by digital signal processing. The room is excited by a known signal for a certain time and the impulse response is calculated from the response to the excitation. The excitation signal is distributed over a longer period of time to increase the total radiated energy. This procedure will enhance the achievable dynamic range and reduce the influence of extraneous noise.

2.2 Ambisonic

Ambisonics is a suite of surround sound capture and storage techniques developed in 1970 that takes into account the mechanisms of localization of the sounds of the auditory system and is based on these principles to reproduce the directional characteristics of a sound field beyond those that are possible to obtain through the use of differences in amplitude or phase of a conventional stereophonic recording.[2]

The main advantage of Ambisonics is its ability to reproduce an immersive sound field, independent of the number and position of the listener's speakers; providing the largest listening area of any surround sound system today. Even walking outside the speaker array the listener can appreciate the sound field with good localization.

2.2.1 Format B

It is the standard signal in Ambisonics and each channel is identified by the abbreviations W, X, Y and Z. There are two conventions within the B-format standard of Ambisonics: AmbiX and FuMa. They are quite similar, but not interchangeable: they are differentiated by the sequence in which the four channels, with AmbiX, for example, arranged W Y Z X instead of W X Y Z.

2.2.2 Format A

When recording with Ambisonics microphones,four signals are obtained in the so-called A-format. Because the format in which Ambisonics works is the B-format, it is necessary to perform a A-B format conversion to be able to work with those signals. This type of microphones uses four capsules placed in the shape of tetrahedron as can be seen in the figure 1.

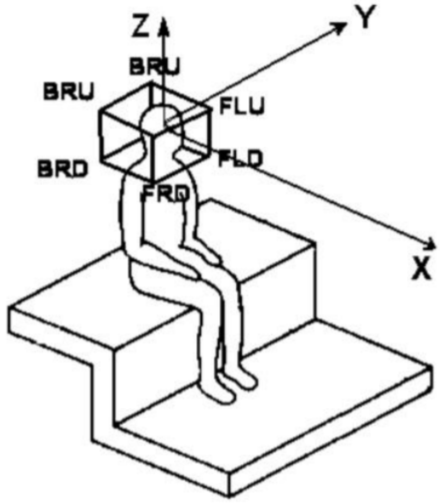

Figure 1: Format A Capsules orientation

2.2.3 A to B Conversion

The fundamentals of converting from A-Format to B-Format are done to through matrix operations of the signals collected from the capsules.

$$\begin{aligned} W &= FLU + FRD + BLD + BRU \\ X &= FLU + FRD - BLD - BRU \\ Y &= FLU - FRD + BLD - BRU \\ Z &= FLU - FRD - BLD + BRU \end{aligned} \tag{1}$$

Where W, X, Y, Z are the B format signal channels and FLU, FRD, BLD, BRU are the A format signal channels.

Considering that the signals were captured by microphones that do not match at exactly the same point, you must use two specific filters to position them virtually in the center of the tetrahedron. The first must be applied to W ', while the second must be applied to the remaining signals. These filters perform substantial changes in the gain of the signals, but also slightly affect the phase of the same[2][3]

$$\begin{aligned} F_W &= \frac{1 + \frac{j\omega}{c} - \frac{1}{3}(\frac{\omega r}{c})^2}{1 + \frac{1}{3}\frac{j\omega r}{c}} \\ F_{XYZ} &= \sqrt{6}\frac{1 + \frac{1}{3}j\omega r - \frac{1}{3}(\frac{\omega r}{c})^2}{1 + \frac{1}{3}\frac{j\omega r}{c}} \end{aligned} \tag{2}$$

Where r is the distance of each capsule towards the center of the tetrahedron (in meters), ω is the angular frequency rad/s and c is the speed of sound m/s (assumed to be 343 m / s).

2.3 Intensity vectors calculation

The techniques employed to analyse 3D impulse response is based on the work of Farina and Tronchin [5]. It exploits the capabilities of the B-Format signal of detecting the direction-of-arrival of each impinging wavefront by computing the “instantaneous” sound intensity vector I and the instantaneous value of the energy ratio rE and is based on the same vector decomposition scheme initially proposed in [16], also related to the later SIRR method [12]. The three components of the sound intensity vector can be simply obtained from the B-Format Impulse Response by means of the following equations:

$$\begin{aligned} I_x &= W.X \\ I_y &= W.Y \\ I_z &= W.Z \end{aligned} \tag{3}$$

Where w, x, y and z represent the four signals of the B-Format IR. Keeping in mind these signals are proportional to sound pressure and particle velocity, the total energy density can be computed by means of the following equation

$$E_D = \frac{W^2 + X^2 + Y^2 + Z^2}{c} \tag{4}$$

The modulus of the sound intensity vector is:

$$|I| = \sqrt{I_x^2 + I_y^2 + I_z^2} \tag{5}$$

The ratio between the active intensity and energy density is computed as:

$$r_E = \frac{|I|}{E_D c} \tag{6}$$

Finally, the azimuth (horizontal) “a” and elevation (vertical) “e” angles are simply obtained from trigonometric equations, i.e.:

$$\begin{aligned} a &= arctan \frac{I_y}{I_x} \\ e &= arctan \frac{I_z}{|I|} \end{aligned} \tag{7}$$

It is important to note that each vector symbolizes the intensity in a window of integration of a certain time. Although it is suggestive to interpret each line as a single reflection, in many cases, a time window encapsulates more than one reflection, and the resulting vector is determined at all times by the combination of the reflections included in that window.

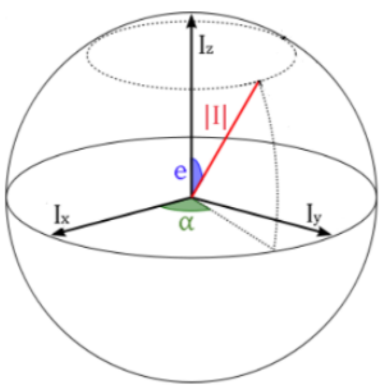

Figure 2: Resultant intensity vector

2.4 Direct Sound and Reflections

To begin with, it's necessary to separate the impulse response into what would be the direct sound from what the reflections are. This is done since the impulse response is needed under anechoic conditions to be able to perform the cross correlation. For this to be done successfully, there needs to be a considerable time gap between the direct sound and the reflections.

2.5 State of Art

Currently there are several software that calculate and display the 3D response of an impulse response. On the one hand, we have the EASE software[4], which, based on geometric simulations on a designed enclosure, in addition to allowing you to obtain synthetic impulse responses, allows you to see the resulting hedgehog. Another software, more focused on 3D signal processing, is IRIS[5]. The results delivered by this software can be used to relate sound rays to physical features of the room, observe the directional distribution of early and late sound energy, as well as identify surfaces causing problematic reflections. Finally, there is the IR360 software developed by the sound engineer Federico Nahuel Cacavelos as thesis work[2] and which was taken as a guide for the development of the software that was carried out for this paper. IR360, like IRIS, and our software, allows a 2D projection of the resulting hedgehog to be placed on a plant image of the place where the measurement was made. This allows a clearer view of the distribution of the signal in the space where it was reproduced. In addition, it has several options for editing the audio to be processed, as well as for displaying the results: number of windows to be displayed, their overlap, choice of direct sound, etc. Many of these options are also available in our software.

3. Python Code Development

For the processing of Room Impulse Responses and the computation of the 3D Rir, an open-source piece of Python software was developed. For easy user interaction, a GraphicalUser Interface (GUI) was created with the help of the cross-platform GUI toolkit for the Python language WxPython.

3.1 Processing

In order to describe how the results were obtained, a description of each process is made.

- The analysis of each impulse response is made from the B format, so the conversion is carried out if necessary.
- The signals are cut considering the start of the signal by detecting the maximum peak of the 4 channels.
- Each channel is filtered with the virtual position correction filter.
- Signal windowing. By default, direct sound is detected considering from the first maximum to the first minimum, and the rest of the signal is sectioned with the same window length. In other case the user can determine the value of the windows in milliseconds considering the direct sound and the integration of the rest of the signal in different time sections. There are three possibilities of overlap 25 %, 50 %, or no overlap. In this way, a temporarily sectioned vector is done.
- The wavelength when increasing frequency, begins to be comparable with the distances between capsules. In order to avoid distortions effects, signals are processed with a low pass filter with cutoff frequency of about 5 Khz. Even using filters with linear phase, its application can introduce phase modifications in frequency. For this reason, time window segmentation must be done before using filters.
- The intensity vectors are generated with the (X, Y, Z) coordinates.

- The coordinates vectors are normalized with direct sound, and then the module of each position is calculated in dBs.
- With the direction vectors and the calculated module, the interactive graphic presentation is carried out.

In Figure 3 a flow chart is presented with the basic structure of the data processing, in order to describe the operation of the code performed.

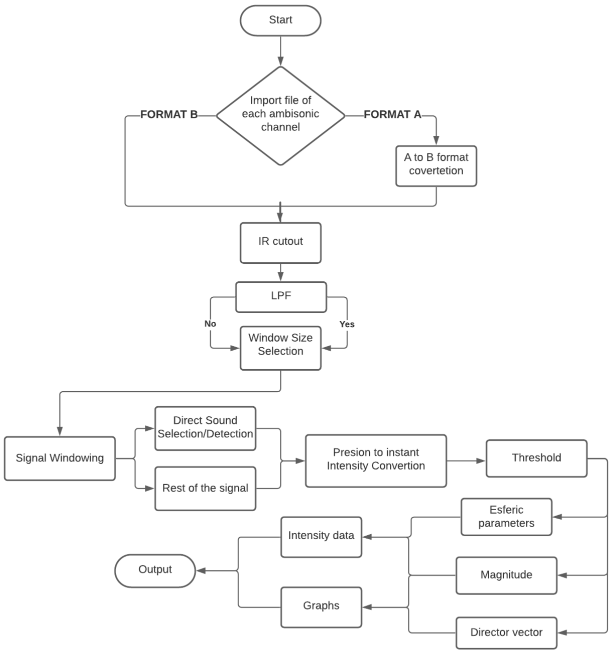

Figure 3: Signal processing flow chart

3.2 GUI

Figure 4 shows the initial GUI window in tab *Parámetros*. The user can import the files in format A or B. In both cases, the program requires the 4 files in .wav format. Also it can be imported an image of the plant view. When the files are selected, the user can modify some characteristics of the processing. These are described below.

- *Tamaño de ventana [ms]:* The user determine the integrating window length of the signal. This specification does not include the direct sound. If this element is left blank the windows length is the default value.
- *Sonido directo [ms]:* The user determine the direct sound window length. If this element is left blank the windows length is the default.
- *Solapamiento:* The user determines the overlap percentage, between 25%, 50 % or none overlapping.
- *Filtro pasa bajos:* The user choose between applying or not the low pass filter.
- *Threshold:* This value refers to the levels that are taken into account in the graph. Values below the threshold are not displayed.

Once the process is done the W channel signal is shown together with the direct sound portion obtained automatically.

In this view there are three more tabs with some other information obtained after the signals processing.

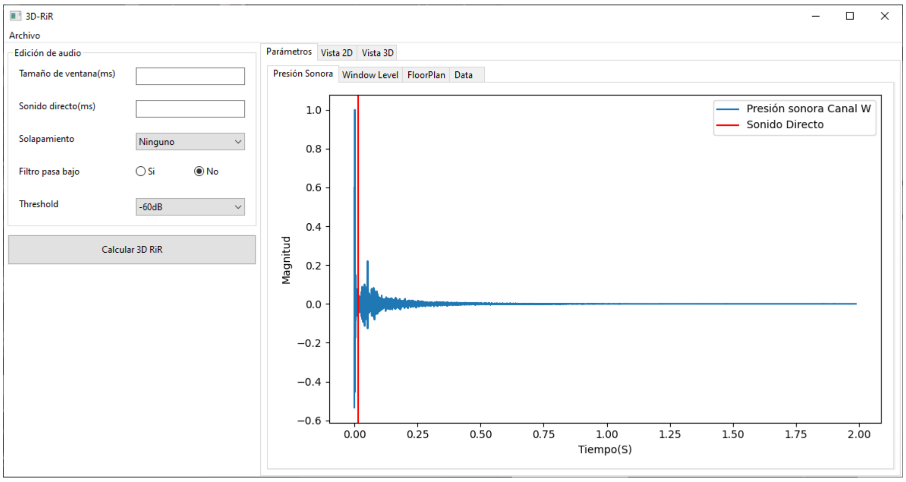

Figure 4: Inital GUI

- *Ventanas:* In this section it can be seen the signal sectioned in windows, represented as red dots. This representation is sisplayed above the impulse response for channel W.
- *Vista de planta:* From "Archivo-> Importar -> Imagen de planta", the user can import a floor plan view of the venue where the measurement were taken off. In this tab the uploaded image is displayed.
- *Data:* The results obtained are shown in table format, with the values of each integrated window, magnitude, elevation and azimuth.

The figure 5 shows the 3D response obtained with the software. As can be seen, the software allows the image to be rotated in 3 dimensions as well as showing the intensity of each of the spines obtained when it is clicked. It is represented on a color scale that symbolizes the arrival time of each integrated window. The software also allows you to export this image in 3D, the intensity data, as well as the Azimuth, Elevation and time values of each window in CSV files or Excel files.

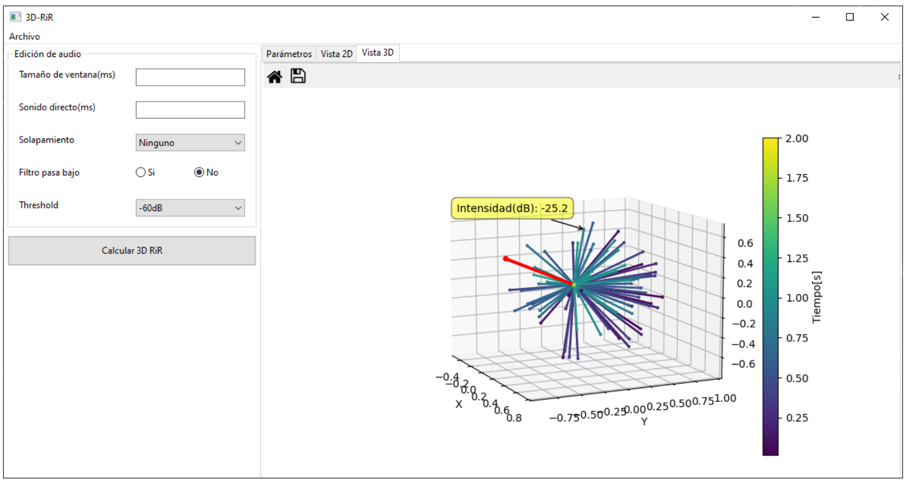

Figure 5: 3D acoustic impulse response

Finally, the figure 6 shows the "Vista 2D" tab with the picture of the place where the measurement was done, in order to show the 2D projection of the hedgehog obtained on top of it. The user can place the 2d view of the hedgehog at the receiving position. It is also allowed to rotate and scale the result in order to perform a proper analysis. This image together with the 2D projection of the hedgehog, can be exported in a PNG image.

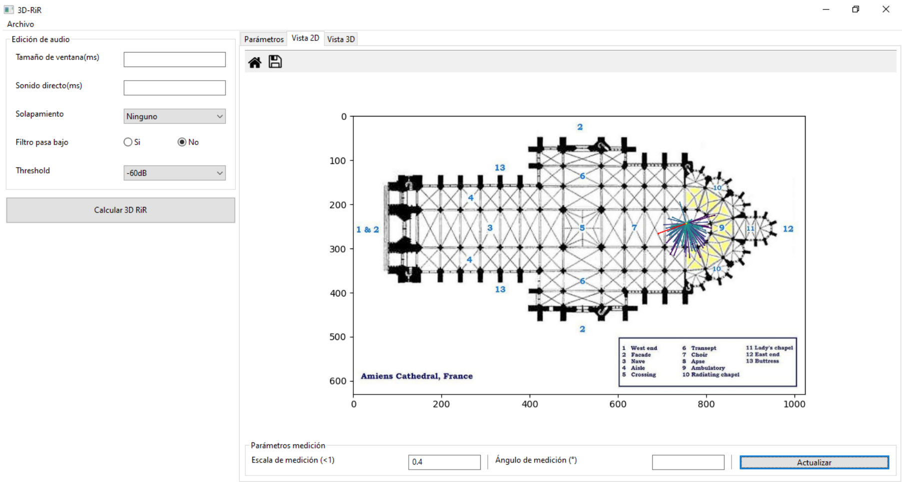

Figure 6: 2D View of measurement on floor-plan

4. Results

In order to obtain and analyze some results, the processing of Format B impulse responses is carried out. The signal were obtained from the Open Air [6] library in this case measured in The Arthur Sykes Rymer Auditorium. The Arthur Sykes Rymer Auditorium is one of the three large performance spaces on campus of the University of New York. In this case the measurements were made with two sources and seven reception positions distributed in the room. The floor plan and the source and receiver positions are shown in Fig. 7.

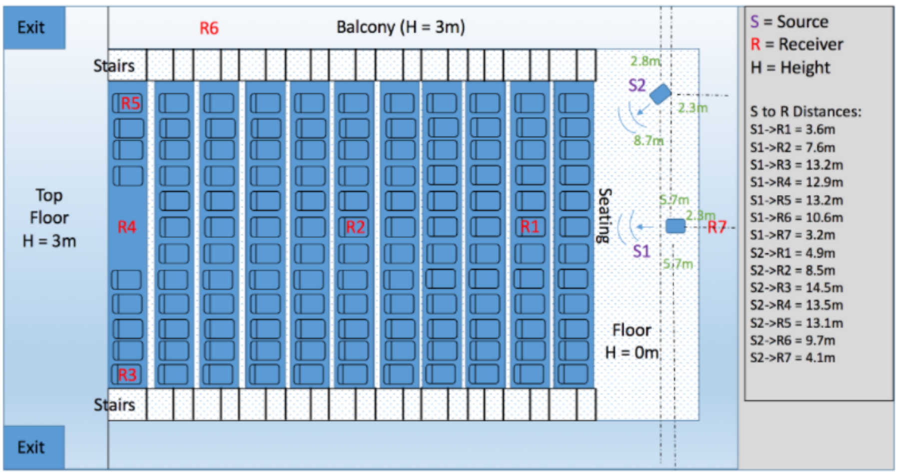

Figure 7: 2D View of measurement on floor-plan

The characteristics of the measurement are presented below:

- Source Sound: Genelec 8030
- Signal: Exponential Sine sweep (25 seconds, 20 Hz to 20 kHz)
- Microphone: Soundfield ST350 Kit
- Sample Rate: 96 kHz
- Number of Channels: 4
- Bits per Sample: 24

4.1 Description and analysis of data obtained

To carry out the analysis of the data obtained, the most representative results are chosen and conclusions are taken. The signal corresponds to the measurement made with source 1 in position 7 (S1R7). The source is located under the balcony on the stage, and the microphone is positioned on the balcony above and behind the source.

Details of measurement:

- Source Receiver Distance (m): 3.3 m
- Source Height: 1.5 m
- Receiver Height: 4.5 m

The following GUI settings were used for the results shown in this section:

- The direct sound was automatically detected.
- Window size equal to that of direct sound.
- 50% overlap.
- Low-pass filter deactivated.
- -60dB threshold.

Figure 8 shows the direct sound selected automatically by the software. It has a length of approximately 10ms and, by means of the resulting graph, it can be stated that the selected portion is consistent with what one would believe to be the direct sound of the signal. This is why it was decided to leave this value for the direct sound length obtained by default.

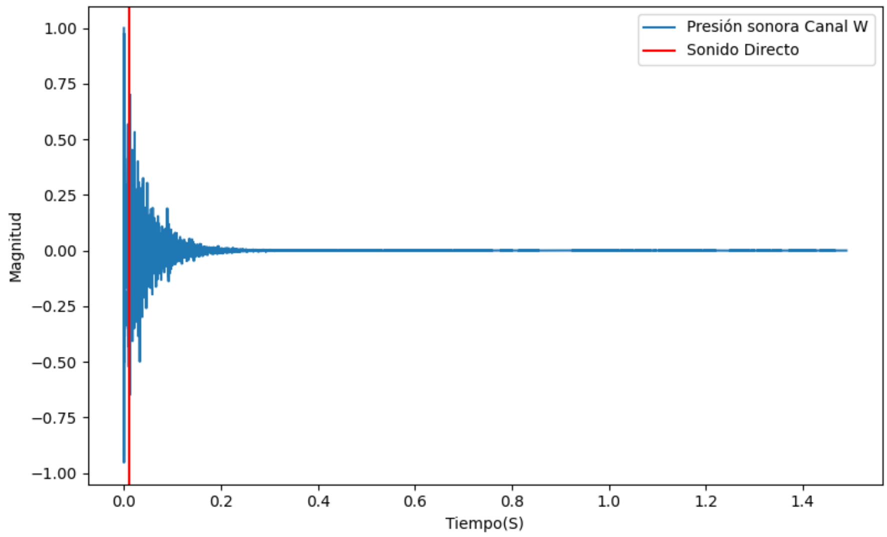

Figure 8: Channel W pressure levels and direct sound selection

The figure shows the windows in which the signal will be divided for processing. Its length, as stated above, will be equal to that of the direct sound (10ms). It can be seen that from about 200ms the signal level is very small compared to the beginning. Since a threshold eb -60dB has been set for direct sound, the windows corresponding to this section of the signal will not have enough energy to exceed it and therefore will not appear in the final graphics.

In figure 17 we can see that the direct sound points downwards. This corresponds to the position of the measurement point relative to the source, detailed above. Furthermore, it is observed that the first reflections also come from the front, but in the entire ZY plane. They can correspond to the reflections generated by the stands and the ceiling of the venue or even the back wall. Finally, we see that the

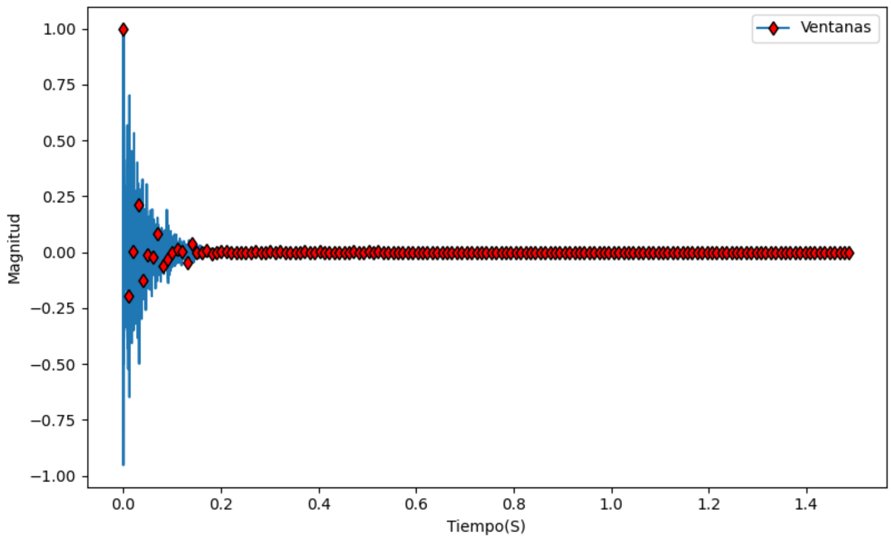

Figure 9: Niveles de presión del canal W y ventanas a graficar

latest reflections, and that exceed the configured threshold, come from behind the measurement point and from below. This makes sense since they come from the wall behind the microphone and the balcony floor, therefore, they are the last to be generated, logically.

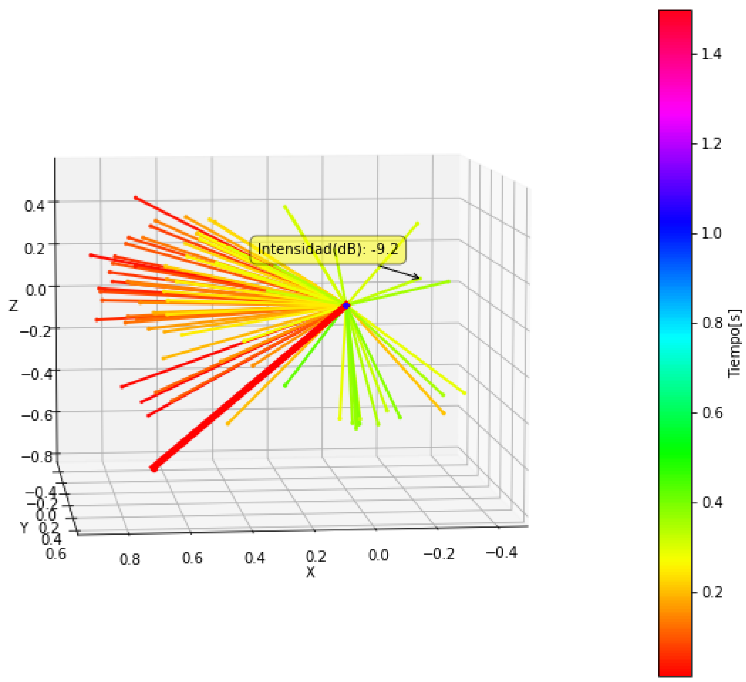

Figure 10: 3D view of impulse response

This last point becomes more evident when the measurement is shown in 2D on the floor plan image of the room where it was made (figure 17). It can be seen that the rear reflections are much less than the front ones, in addition to being generated later.

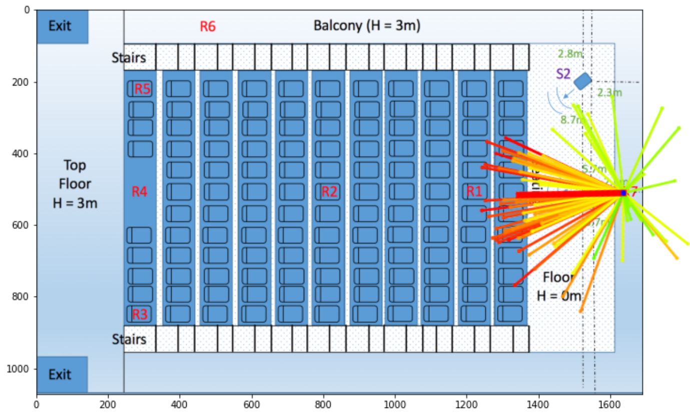

Figure 11: 2D view of impulse response on floor plan

5. Conclusions

The purpose of this work is to be able to provide software that enables the user to characterize an enclosure based on soundfield measurements of impulse responses in three dimensions. In addition, the software development took into account the possibility that the user can make customized configurations in order to give greater versatility to signal processing.

Some conclusions can be drawn about the results obtained. The developed software could not be validated and compared with some commercial software, in spite of this, it can be concluded that the study of the signal is carried out for the most part, successfully. From the visualization of the graphs obtained, it is observed that there is consistency with the phenomenon studied. In all cases the temporal distribution of the reflections is congruent to the positions of the source and the receiver.

Finally, although the results are expected values and have different functionalities, there are possible improvements to be able to further deepen the validation of the results. Among these possible improvements, the FFT method could be added as an alternative processing to obtain intensity from pressure levels. Also, on the other hand, the method of Du et al [7] could be added for the identification of peaks in the response. This method looks for a way that only the vectors that are represented are graphed due to peaks in the signal of window integrations over time and the rest are disregarded. Being able to add these functionalities would add comparative results, and would further nurture the analysis of the room characterization.

References

1. Antoni Torras Rosell. *Methods of Measuring Impulse Responses in Architectural Acoustics*. Technical University of Denmark, 2009.

2. Federico Nahuel Cacavelos. *Interpretación gráfica de respuestas al impulso en tres dimensiones*. Universidad Nacional de Tres de Febrero, 2018.

3. Angelo Farina. *A-format to B-format conversion. [Online].* http://pcfarina.eng.unipr.it/Public/B-format/A2B-conversion/A2B.html, 2015.

4. AFMG. *EASE- Enhanced Acoustic Simulator for Engineers*. https://ease.afmg.eu, 1990.

5. Marshall Day Acoustics. *IRIS*. http://www.iris.co.nz, 2021.

6. The Open Air Library. *Arthur Sykes Rymer Auditorium, University of York.* https://www.openair.hosted.york.ac.uk, 2018.

7. P. Du W. A. Kibbe and S. M. Lin. *Improved peak detection in mass spectrum by incorporating continuous wavelet transform based pattern matching.* Bioinformatics, 2006.

6. Appendix

Below are the magnitude, azimuth and elevation values obtained for each of the windows processed in the impulse response that was analyzed in the results section. The data was exported directly from the GUI.

The resulting 3D and 2D graphics on plant image are also attached for the impulse responses obtained with different combinations of sources and measurement points with the same GUI parameter settings that were used for the measurement analyzed in the results section. This measurements are listed below:

- Source one and measurement point two (S1R2).
- Source one and measurement point four (S1R4).
- Source two and measurement point six (S2R6).

N°	Tiempo(ms)	Magnitud(dB)	Azimuth(°)	Elevación(°)
Sonido Directo	0	0	-7,9	138,5
1	10,07	-5,3	-31,4	127,4
2	20,2	-9,2	-24,1	117,6
3	30,34	-11,4	11,7	54,8
4	40,47	-15,4	-18,6	87,8
5	50,61	-17,4	-18	88
6	60,75	-17,5	9,7	71,8
7	70,88	-21,6	13,8	60,1
8	81,02	-18,5	0,5	84,8
9	91,15	-19,8	4,8	75
10	101,29	-25	27,2	116,1
11	111,42	-26,6	-26,3	98,3
12	121,56	-26,5	8,3	94,7
13	131,7	-28,9	-8,8	51,9
14	141,83	-31,1	-4,5	93,5
15	151,97	-33,7	55,6	100,6
16	162,1	-35	10,5	97,6
17	172,24	-37,1	4,1	53,6
18	182,37	-36,7	-27,5	103,5
19	192,51	-35,7	-19,7	113,5
20	202,65	-43,8	-57,9	124,3
21	212,78	-41	19,5	51,6
22	222,92	-42,2	-28,5	67,9
23	233,05	-44	0,5	84
24	243,19	-46,4	-19,7	105,4
25	253,32	-45,6	10,6	75,9
26	263,46	-47,5	-11,9	60,5
27	273,6	-54,5	38,8	96,8
28	283,73	-51,5	81,5	139,5
29	293,87	-51,4	-70,1	50,1
30	304	-55,7	-44,1	45,5
31	314,14	-53	-59,4	99,4
32	324,27	-56,3	0,4	139,1
33	334,41	-55,4	87,4	71,7
34	344,55	-54,8	-50,8	158,9
35	354,68	-60,1	43,1	76,2
36	364,82	-56,7	57,5	162,7
37	374,95	-55,4	-45	167,3
38	385,09	-56,4	-77,6	161,3
39	395,22	-65,5	29,3	79,6
40	405,36	-60,3	-76,6	86
41	415,5	-61,6	33,3	98,9
42	425,63	-61,8	69,6	125,4
43	435,77	-68,1	18,3	36,7
44	445,9	-62,2	-39,2	45,4
45	456,04	-62,9	36,1	55,4
46	466,17	-64,2	-28,3	14
47	476,31	-63,1	-71,8	84,7
48	486,45	-68,9	32,7	105,3
49	496,58	-61,3	-72,1	63,5
50	506,72	-66,6	-70,7	73,6
51	516,85	-62,5	-20,4	118,7
52	526,99	-61,6	-24,9	101,9
53	537,12	-62,3	-33,2	115,4
54	547,26	-64,8	64,3	121,3
55	557,4	-63,7	71,9	87
56	567,53	-66,4	-71,1	84,2
57	577,67	-66,1	26,5	115,1
58	587,8	-64,8	88,3	26,8
59	597,94	-69,4	-33,8	51,9
60	608,07	-71,9	-7	103,5
61	618,21	-67,9	-1,5	106,5
62	628,34	-68,4	84,7	33,1
63	638,48	-68,6	30	131,2
64	648,62	-64,8	54,4	160,3
65	658,75	-66,7	-85,2	135,1
66	668,89	-71,2	-81,8	58,5
67	679,02	-68,9	37,1	118,7
68	689,16	-69,1	54,1	109
69	699,29	-69	-62,3	73,8
70	709,43	-73,8	-49,1	62,9
71	719,57	-69,5	89,9	118
72	729,7	-67,8	-77,9	153,3
73	739,84	-67,8	30,3	148,1
74	749,97	-67,8	34,9	173,8
75	760,11	-66,4	61,6	157,9
76	770,24	-73,6	65,8	66,9
77	780,38	-70,3	-52,9	34,4
78	790,52	-69,8	63,6	26
79	800,65	-72,4	-71,2	79,3
80	810,79	-67,6	53,7	156,8
81	820,92	-66	86,1	144,1
82	831,06	-67,6	88,2	154,5
83	841,19	-69,1	-28,2	153,4
84	851,33	-68	88,6	146
85	861,47	-69	53,7	101,6
86	871,6	-71,6	74,2	126,4
87	881,74	-71,8	-27,9	124,2
88	891,87	-74,2	2,8	95,4
89	902,01	-74	-34,1	118
90	912,14	-68,3	-74,9	126,5
91	922,28	-71,4	87,8	109,9
92	932,42	-72,8	73,3	93,5
93	942,55	-69,2	-86,4	122,9
94	952,69	-72,6	8,3	161
95	962,82	-69,8	-11,4	145,7
96	972,96	-68,9	-67,6	138,8
97	983,09	-70,6	73,1	129,5
98	993,23	-70,8	86,1	91,9
99	1003,37	-68,5	-79,6	123,8
100	1013,5	-71,2	89,2	135,1
101	1023,64	-75,4	65,2	102,5
102	1033,77	-68,5	-77	126,1
103	1043,91	-65,2	89,3	124,2
104	1054,04	-68,6	76,8	138
105	1064,18	-70,2	76,5	128,7
106	1074,32	-69,2	-47,3	120
107	1084,45	-70,3	-14,3	142,8
108	1094,59	-76,1	-11,6	155
109	1104,72	-74,7	-0,3	105
110	1114,86	-72,7	19,6	147,7
111	1124,99	-74,9	65,5	57,1
112	1135,13	-74	63,2	131,9
113	1145,27	-73,4	26,5	73,4
114	1155,4	-71,6	3,9	167,2
115	1165,54	-69	62,9	167,2
116	1175,67	-67,8	78,6	139
117	1185,81	-71,3	55,2	161,1
118	1195,94	-72,6	-4,1	118,5
119	1206,08	-73	7,7	119,5
120	1216,22	-70	88,9	122,8
121	1226,35	-72,6	76,1	94,8
122	1236,49	-71,6	48,4	105,9
123	1246,62	-70,4	84,3	143,9
124	1256,76	-69,6	49,2	135,1
125	1266,89	-72,6	6,7	142,3
126	1277,03	-70,8	78,5	139,7
127	1287,17	-71,9	-58,4	117,3
128	1297,3	-70,7	55,8	135,6
129	1307,44	-70,9	12,1	119
130	1317,57	-72	25,8	153,3
131	1327,71	-70,7	-76,2	102,1
132	1337,84	-71,1	68,8	114,1
133	1347,98	-70,1	55,7	123,1
134	1358,12	-71,7	-14,7	125,4
135	1368,25	-72,7	62	148,7
136	1378,39	-71	66,6	131,8
137	1388,52	-69,1	48,4	144
138	1398,66	-74,2	-62,9	153
139	1408,79	-73,2	84,5	116,3
140	1418,93	-71,3	56,6	76,2
141	1429,07	-70,7	85,9	156,5
142	1439,2	-73,1	31,3	97,8
143	1449,34	-73,1	-80,5	130,6
144	1459,47	-68,6	-60,6	109,5
145	1469,61	-70,5	-86,7	89,4
146	1479,74	-71,6	36,2	136,9
147	1489,88	-73,3	47,9	104,6
148	1500,02	-91,5	79,8	129,9

Figure 12: Obtained data of measurement S1R7

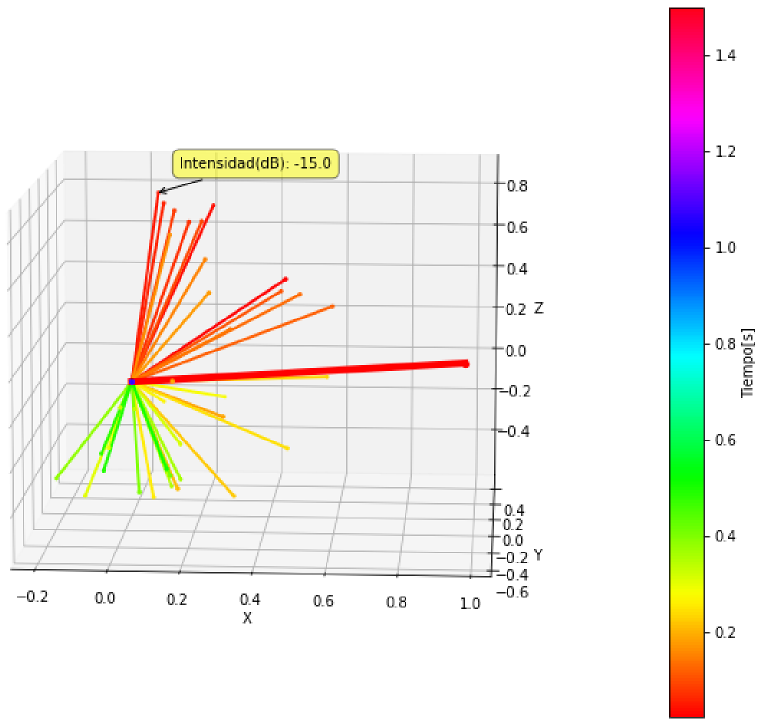

Figure 13: 3D view of measurement S1R2

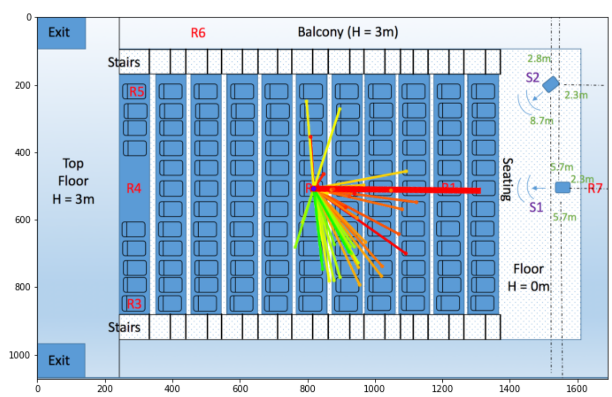

Figure 14: 2D view of measurement S1R2 on floor plan

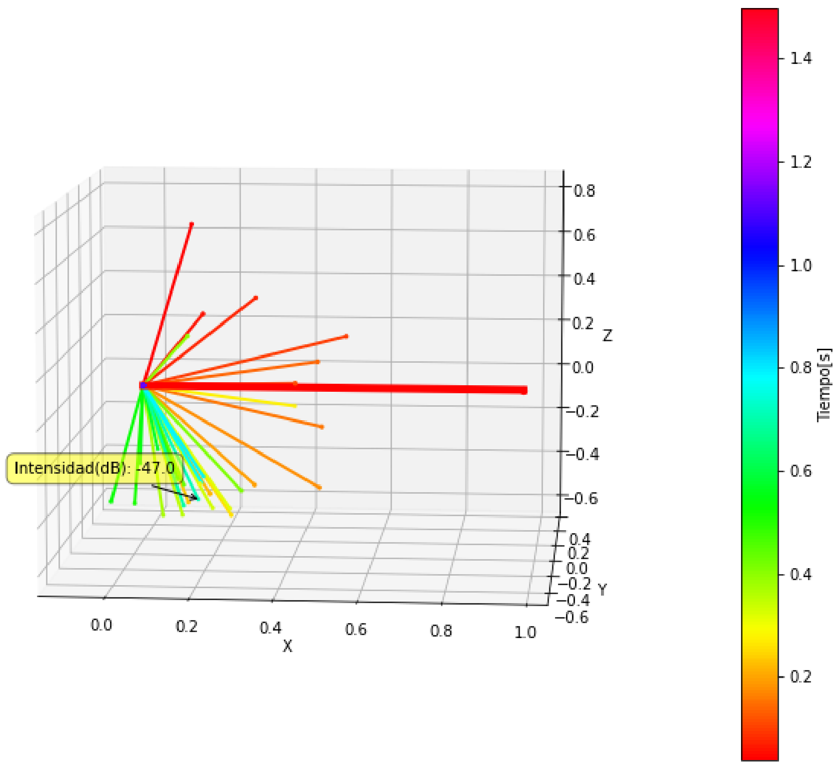

Figure 15: 3D view of measurement S1R4

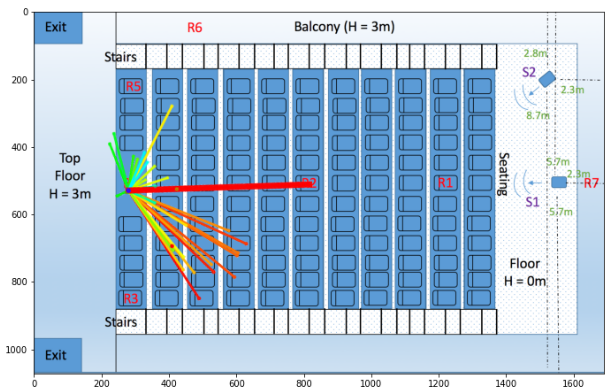

Figure 16: 2D view of measurement S1R4 on floor plan

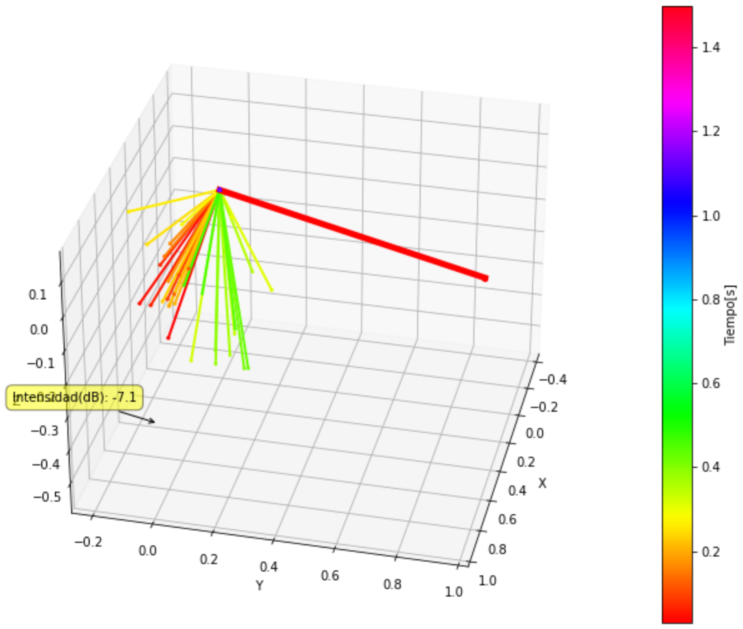

Figure 17: 3D view of measurement S2R6

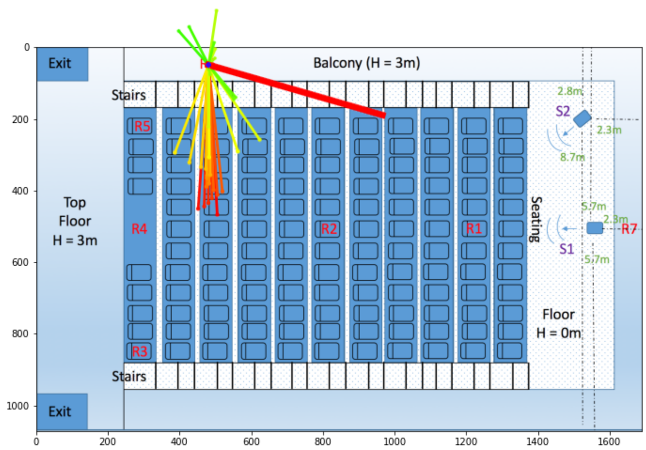

Figure 18: 2D view of measurement S2R6 on floor plan