Virtual Room Characterization

Parrinelli Sofia

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Abstract

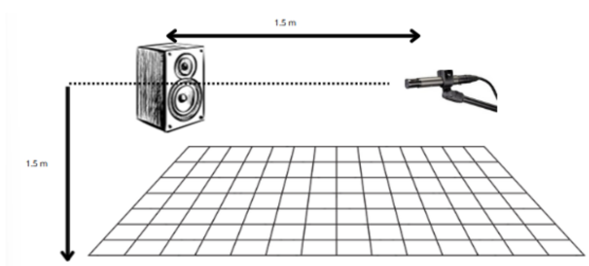
This report provides a comprehensive overview of the procedures and analyses conducted up until November regarding the primary focus of the thesis: the acoustic characterization of the virtual room.

1 Loudspeakers compensation

The aim is to reach the optimal starting point in order to analyze the response of the virtual room. In order to do that, a flat response of the loudspeakers is necessary.

If subjected to an exponential sweep signal, the loudspeaker exhibits non completely flat behaviour in the spectrum, the goal, thus, is to stimulate it by using an suitably predistorted signal that is able to compensate the irregularities.
The recording are carried out in a dead room, sufficiently far away and isolated from unwanted reflection. The following distances are reported:

- 1.9 meters from the first obstacle
- 1.5 meters from the floor
- 1.5 meters: distance microphone to the loudspeaker



An omnidirectional microphone was used.
The loudspeaker's crossover points were configured at 400-2500 Hz for the woofer and 2500- Hz for the tweeter, utilizing a Butterworth filter with a slope of 12 dB per octave.
The sweep used to stimulate the loudspeaker lasts for 1 seconds plus 3 seconds of silence, spanning the frequency range of 22-22000 Hz and includes both a fade-in and a fade-out.

As an initial step, the inverse filter has been created using the Kirkeby method [**1**].
Thus, the inverse of the sine sweep, obtained through Kirkeby, has been convolved with the recorded sound, yielding the impulse response of the loudspeaker. The Kirkeby inverse filter has been then employed once more to derive the inverse of this impulse response.
By convolving the inverse of the impulse response with the initial sweep, a predistorted sweep signal has been . This signal guarantees a flat response when used to stimulate the loudspeaker. Although the loudspeaker already exhibits a relatively flat response, the predistorted sweep ensures a balanced stimulation that is sometimes necessary.

The analysis done in one loudspeaker is considered applicable for each one of the other 24 present in the virtual room.

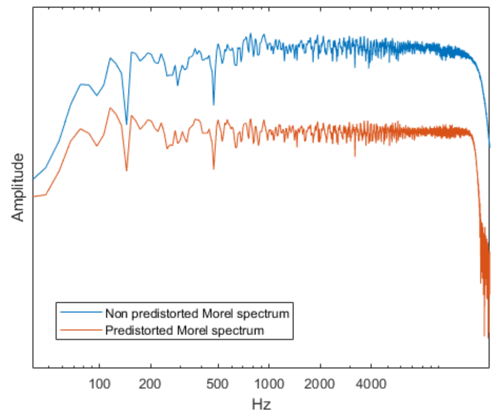


Figure 1: Comparison between the spectrum of the original impulse response with the predistorted one.

2 Virtual Room measurements

2.1 Preliminary considerations

An omnidirectional microphone was used for the measurements in the virtual room. For each set up the microphone was positioned at the driver's location at ear height.

The objective is to analyze the spectrum of the impulse response achieved by linearly summing all the individual impulse responses generated through the sequential excitation of each loudspeaker, therefore find the acoustical parameters: reverberation time, clarity and early decay time.
The ultimate objective is to propose potential solutions for optimizing the acoustic arrangement, establish an acoustical parameter-based description of the environment.

As done for the compensation of the speakers, the crossover points were configured at 40-2500 Hz for the woofer and 2500 - Hz for the tweeter, utilizing a Butterworth filter with a slope of 12 dB per octave.
The pre-distorted sweep was used for all the following measurements.
The responses obtained from the individual excitation of each loudspeaker are initially aligned to account for any potential irregularities in their placement and are then summed together.

For improved spectrum visualization, the *smoothSpectrum function* provided by the Matlab toolbox IoSR was employed. This function mitigates the spectral fluctuations by applying a 1/5-octave smoothing using a Gaussian window with a standard deviation of $f/5$, where f represent the central frequency.

The headrest has been removed in order to avoid sound absorption or reflections and any potential obstruction of the microphone's reception.

All the responses have been accurately calibrated, and the magnitude of the spectrum has been adjusted to ensure the correct values.

2.2 First set up

In this first set up the TV has been kept inside the room.
The numbers above each spectrum in the following picture corresponds to their respective loudspeaker identifiers.

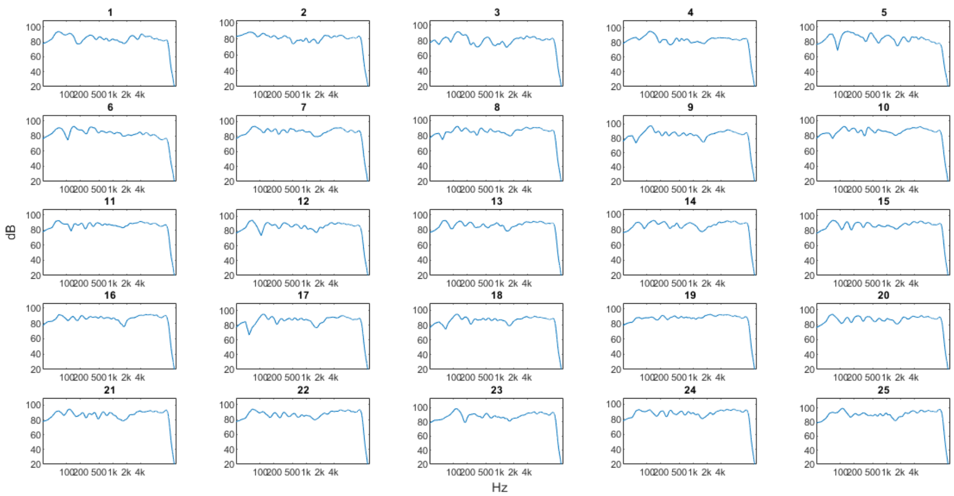


Figure 2: Spectrum of each single loudspeaker. Set up 1

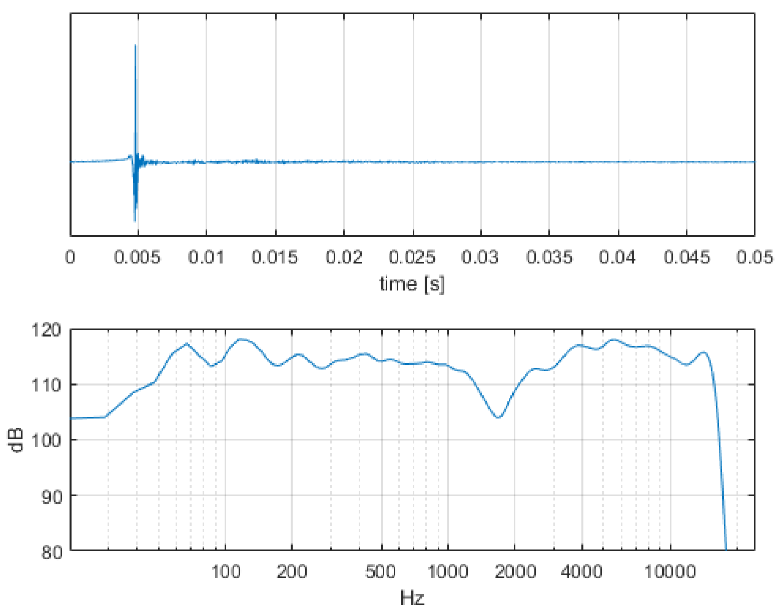


Figure 3: Impulse Response in time and spectrum of the sum of all aligned loudspeakers' responses. Set up 1

At first sight a relevant decay can be noted in correspondence of 1650 Hz.

2.3 Second set up

As a second set up the TV was covered with absorbing polyester fiber panel 10 cm thick.

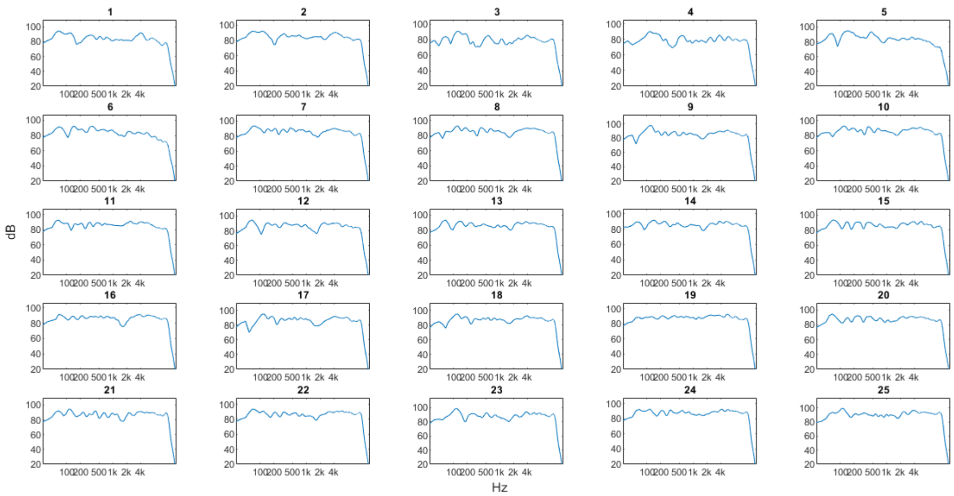


Figure 4: Spectrum of each single loudspeaker. Set up 2

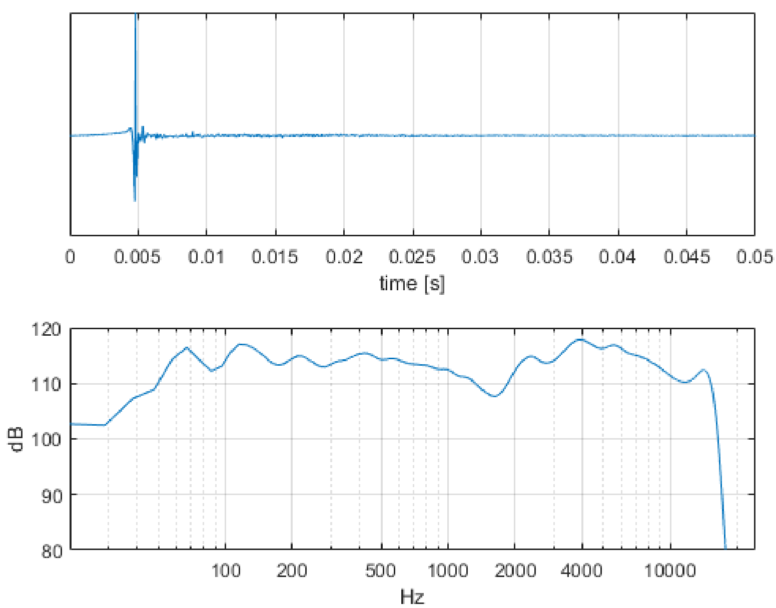


Figure 5: Impulse Response in time and spectrum of the sum of all aligned loudspeakers' responses. Set up 2

2.4 Third set up

For this configuration the TV has been removed.

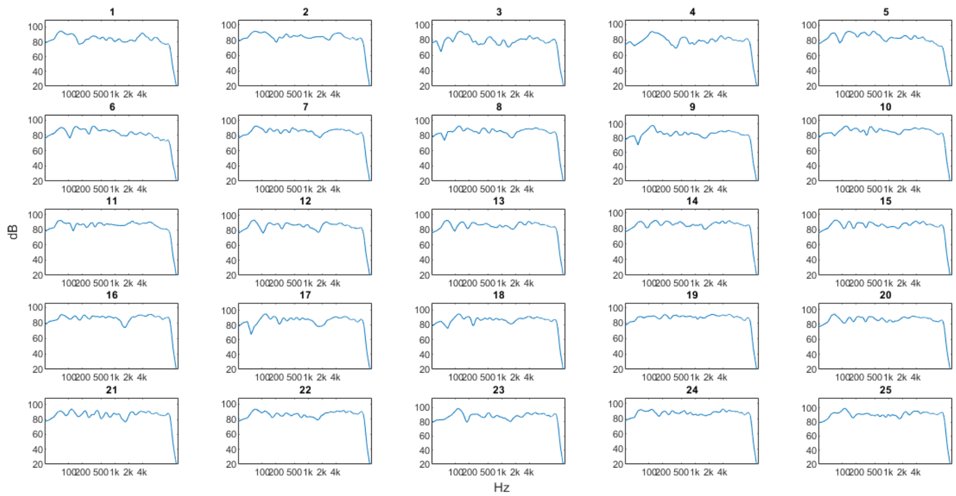


Figure 6: Spectrum of each single loudspeaker. Set up 3

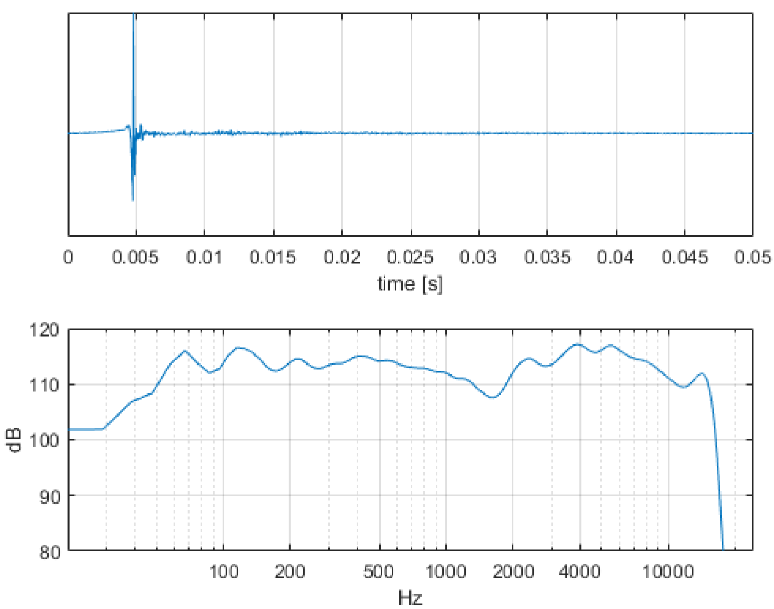


Figure 7: Impulse Response in time and spectrum of the sum of all aligned loudspeakers' responses. Set up 4

2.5 Fourth setup

The TV is now still not present and the door is covered by an absorbing panel 5 cm thick.

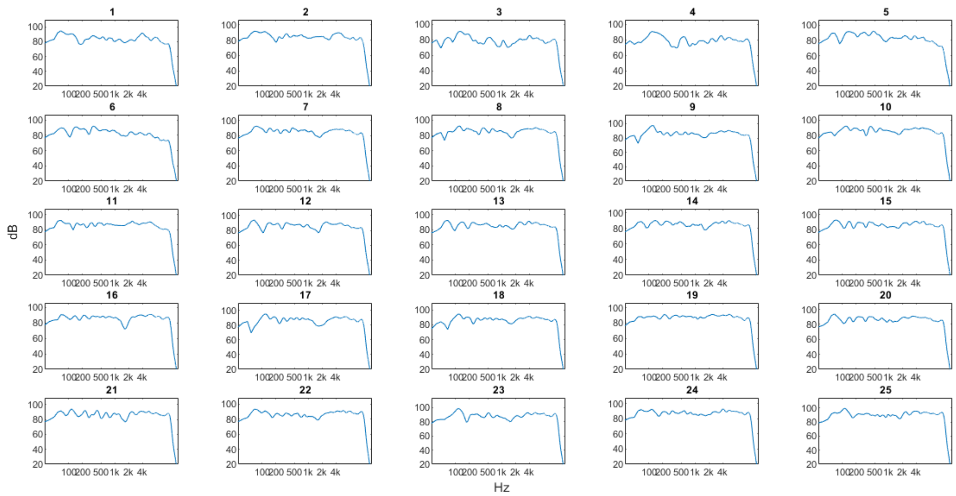


Figure 8: Spectrum of each single loudspeaker. Set up 4

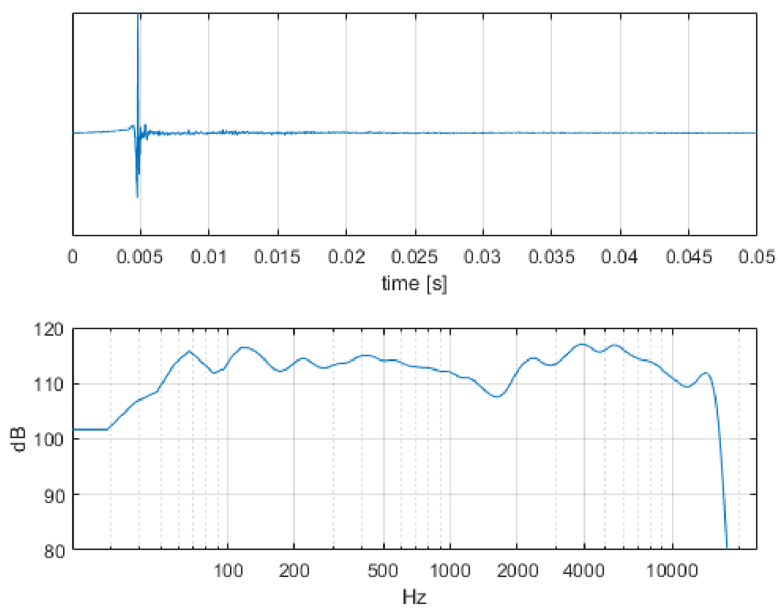


Figure 9: Impulse Response in time and spectrum of the sum of all aligned loudspeakers' responses. Set up 4

2.6 Fifth and last set up

In this case, the TV is again present, the door not covered anymore but the seat is removed as well as the pedal and its support are removed too.

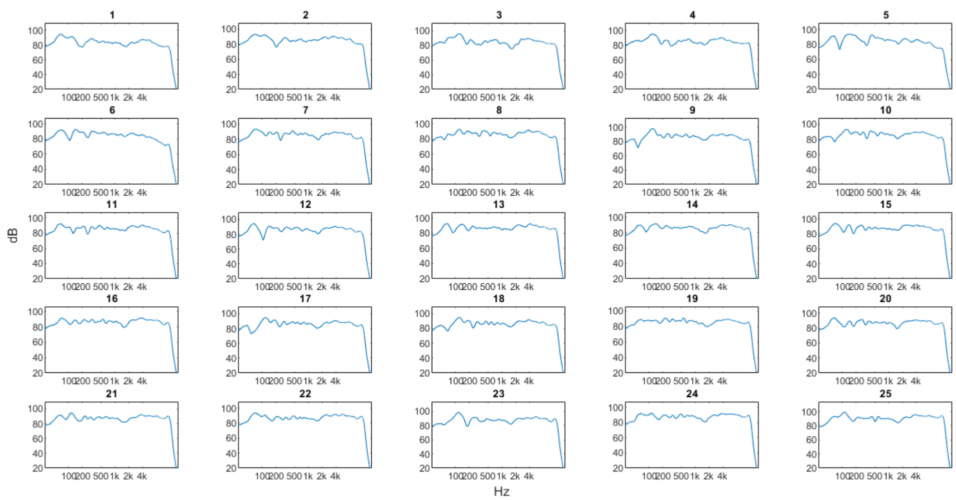


Figure 10: Spectrum of each single loudspeaker, the pedal and its support has been removed

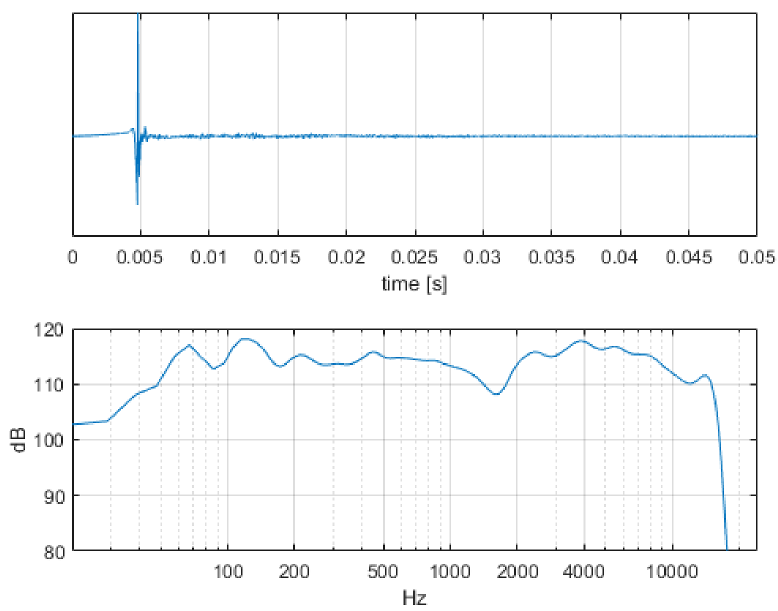


Figure 11: Impulse Response in time and spectrum, the pedal and its support has been removed

2.7 Considerations

The decay around 1650 Hz is always present, in the total spectrum for all the configurations.
Therefore, measurements by utilizing the non predistorted sweep were conducted to see if the problem could be a wrong compensation of the loudspeakers due to a possible wrong ambient of measurement. However, it has been proven that the results do no change even with the non predistorted sweep, therefore this aspect is not the cause of the problem.

The following graphs display the spectra of various configurations together, providing a clearer visualization of all the configurations and their differences.

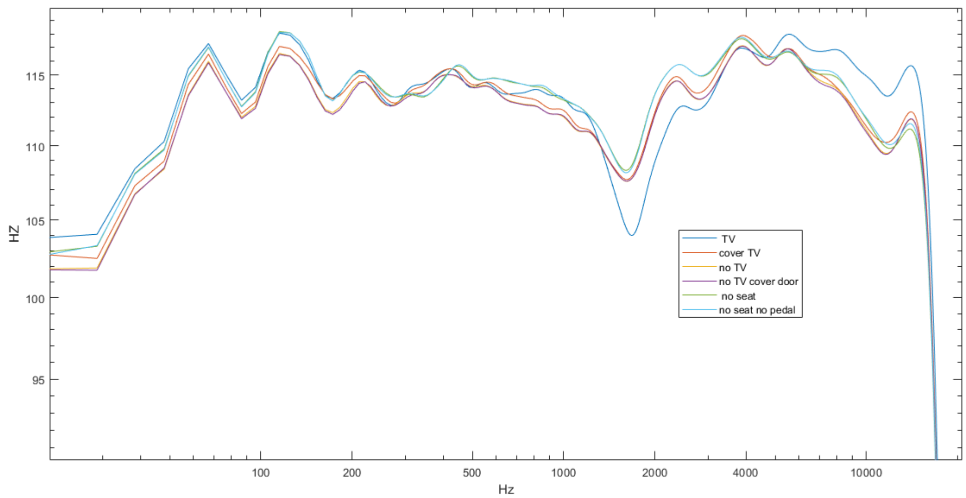


Figure 12: Spectrums of the different setups

2.8 Analyzing the problem

Further analyses were conducted with the aim of examining the issue and identifying its root cause.
Consequently, the direct responses from two speakers, identified as problematic in all configurations (speakers 7 and 9), were analyzed.
The microphone was placed at a distance of 80 cm, centered between the tweeter and woofer, testing both with pre-distortion and without, with and without a grille.
The analyses were further deepened using Smaart, capable of displaying impulse response and its real-time spectrum.
It was deduced that the issue persisted even in the case of direct sound, ruling out a specific room reflection, or a mistake in the pre-distortion, as the cause.
Since the response of speaker No. 19 was correct in the 1650 Hz zone, it was observed that the difference between this speaker and the others lay in the fact that the tweeter of speaker 11 was the only one with a non-inverted phase. The phases of the tweeters of the other speakers had been inverted in a previous use of the room.

2.9 Phase correction applied

Once the phase of all speakers was corrected, the following final results were obtained, by testing the pre-distorted sweep with the TV and door presence and then their covering through an absorbing polyester fiber panel 10 cm thick for the TV and 2,5 cm thick for the door.

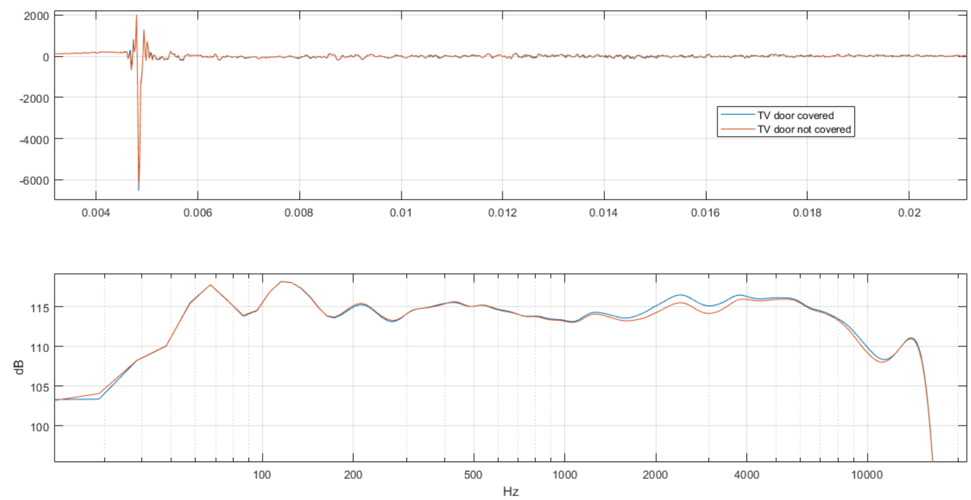


Figure 13: Sum of the single Morels after reversing the tweeter

The response is now much more linear in frequency.
A gain was applied to loudspeaker n. 13 of +4 dB and to loudspeaker n. 25 of -4 dB to both tweeter and woofer to compensate loss of balance in the volume.

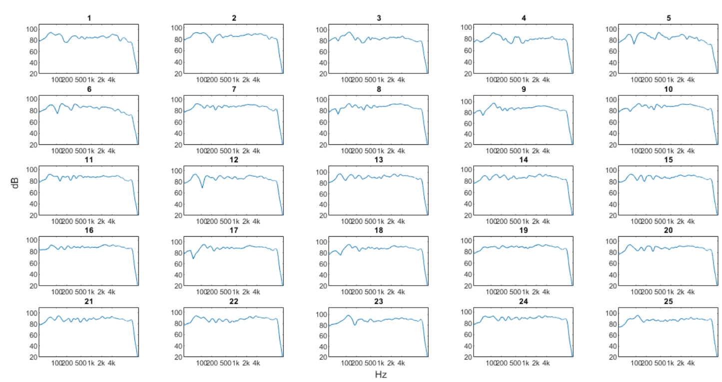


Figure 14: Door and TV not covered, tweeter fixed

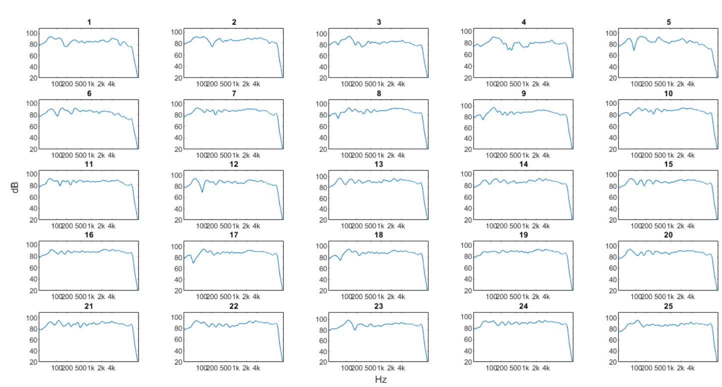


Figure 15: Door and TV covered, tweeter fixed

The previous problem is completely fixed and it can be observed that the two configurations do not lead to great differences in the time and frequency responses of the room.

3 Analysis with balloons, obtained acoustic parameters

To study the room itself without influences from the speakers, balloons were popped in 3 different positions using the two configurations mentioned in the previous section: with the TV and door present, and then covered.
Three positions of explosion were tested:

- Centered: 70 cm upon the microphone in driver position
- Back: 1.20 m distant from the microphone in the door direction
- Front: 1.40 m distant from the microphone, in the TV direction

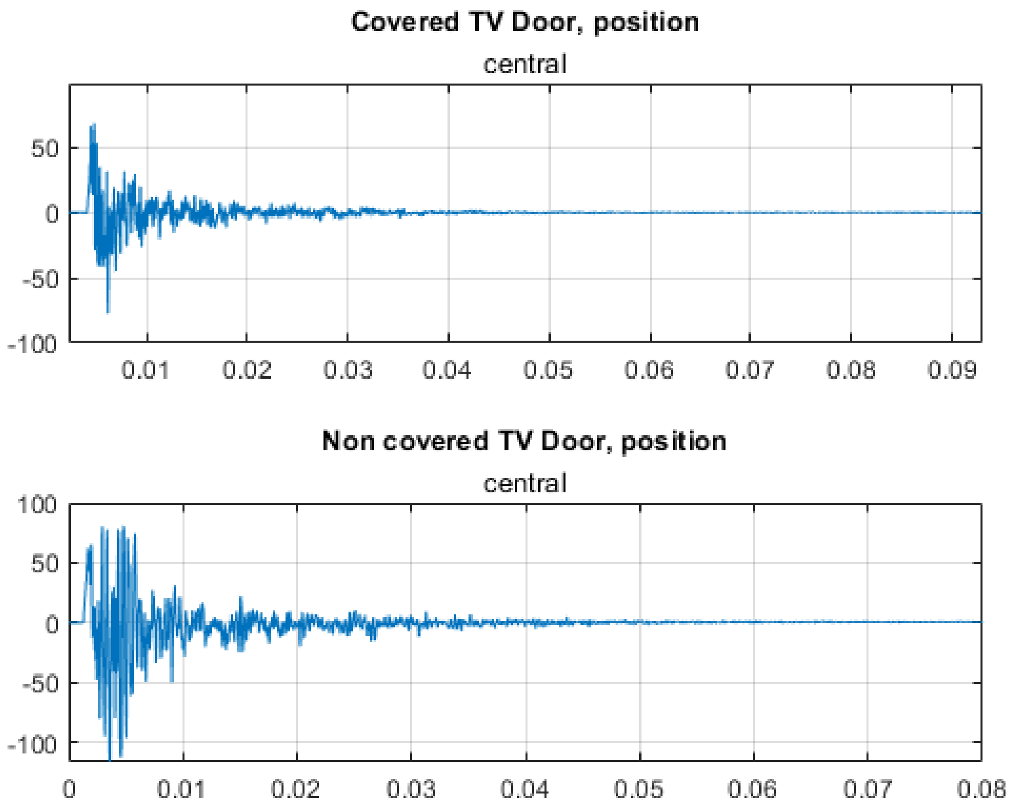


Figure 16: Impulse time trend in the balloon popped in central position, covering TV and door and not covering them

In the presented spectrum, the emphasis should be directed towards comparing the two configurations rather than delving into the details of the frequency trends. This is because the spectrum of the balloon explosion itself may not exhibit a flat profile.

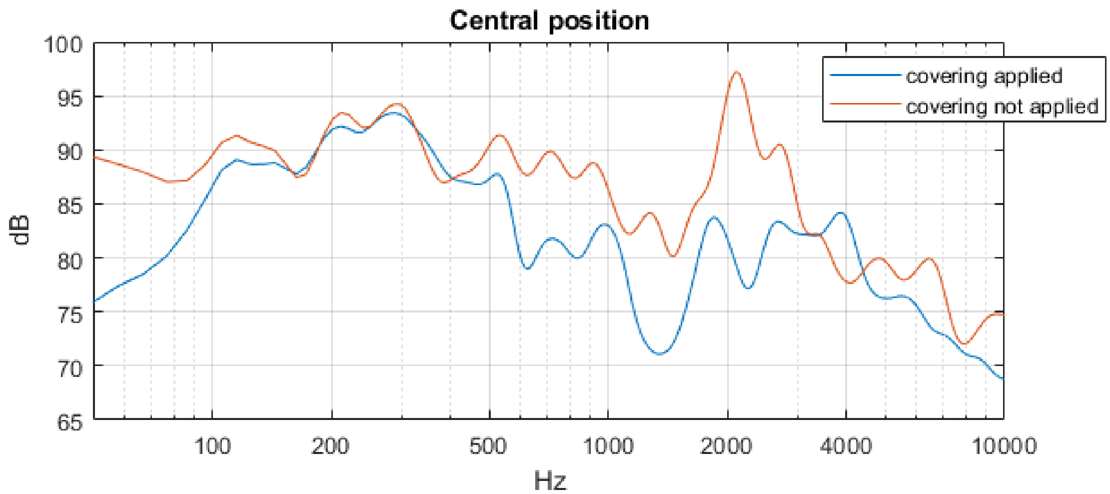


Figure 17: Impulse spectrum in the balloon popped in central position, covering TV and door and not covering them

Upon bursting the balloon, the people in the room distinctly noticed a sharp and bright sound emanating from the ceiling. This can be attributed to the absence of any covering, allowing for additional reflections over time.
These reflections can be traced back to the metal structure supporting the upper loudspeaker, potentially causing the peak just above 2000 Hz in the absence of any covering. Therefore, it is plausible that the panel above the TV is responsible for absorbing these reflections.

In the following bar graphs the detected acoustic parameters are displayed. The signal under study was properly cut and analysed by using the toolbox provided by IoSR: iosr.acoustic.irStats.
The utilized function returns the necessary parameters by employing a method based on ISO 3382-1:2009. This involves using reverse cumulative trapezoidal integration to estimate the decay curve and a linear least-square fit to determine the slope between 0 dB and -20 dB for the T20 calculation. Therefore, the signal is appropriately truncated prior to analysis—specifically, just before the onset of the direct sound and immediately after surpassing the noise threshold. Consequently, there is no need for a noise compensation algorithm, and the decay curve is accurately detected.
The parameters extracted are:

- T20 (Reverberation Time) calculated by determining the decay time from the obtained decay curve. This is achieved through cumulative trapezoidal integration of the band-filtered impulse response between the direct sound and -20 dB.
- Speech clarity index: $C_{50} = 10\log_{10}\frac{\int_a^{50ms}(x(t))^2\,dt}{\int_{50ms}^{b}(x(t))^2\,dt}$ where a represents the detected instant of the direct sound and b the threshold for the noise level.
- Early Decay Time calculated by determining the decay time from the obtained decay curve. This is achieved through cumulative trapezoidal integration of the band-filtered impulse response between the direct sound and -10 dB.

The values are displayed in octave bands: the octave-band filters are calculated according to ANSI S1.1-1986 and IEC standards.

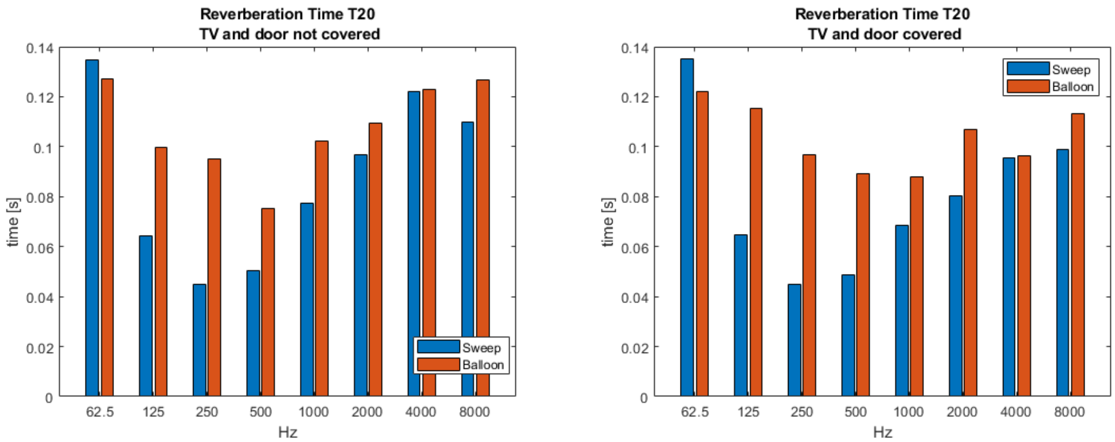


Figure 18: Center position acoustic results: T20

It is possible to observe that the T20 values are optimal even without the absorbing cover.

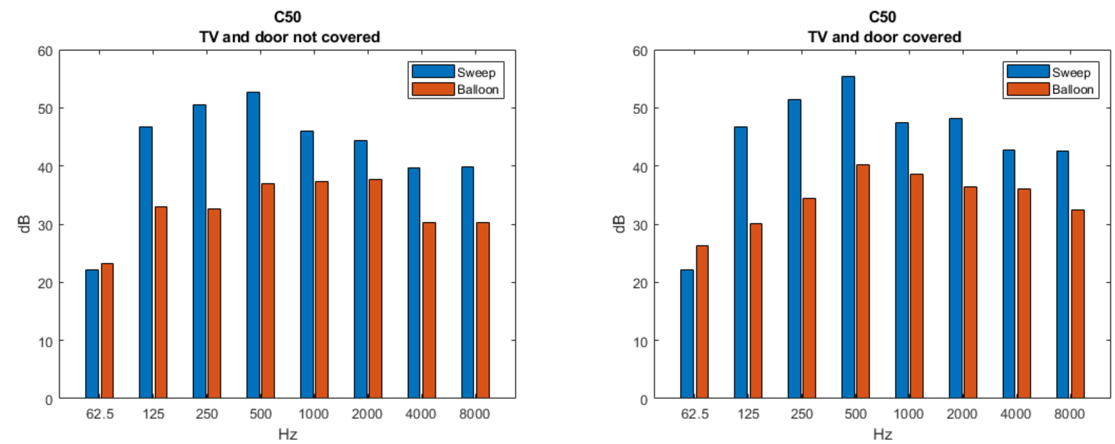


Figure 19: Center position acoustic results: C50

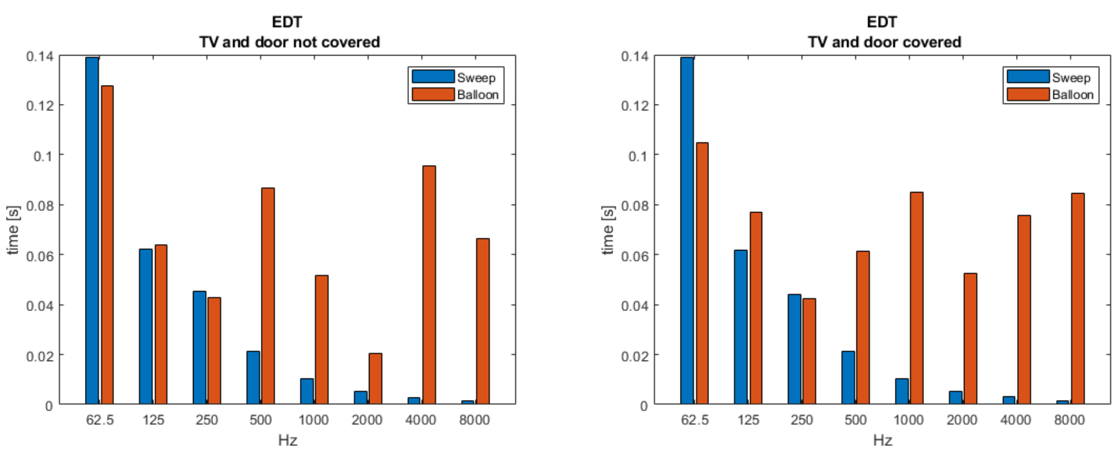


Figure 20: Center position acoustic results: Early Decay Time

For the various tested positions of the balloon popping, the following comparison could be shown:

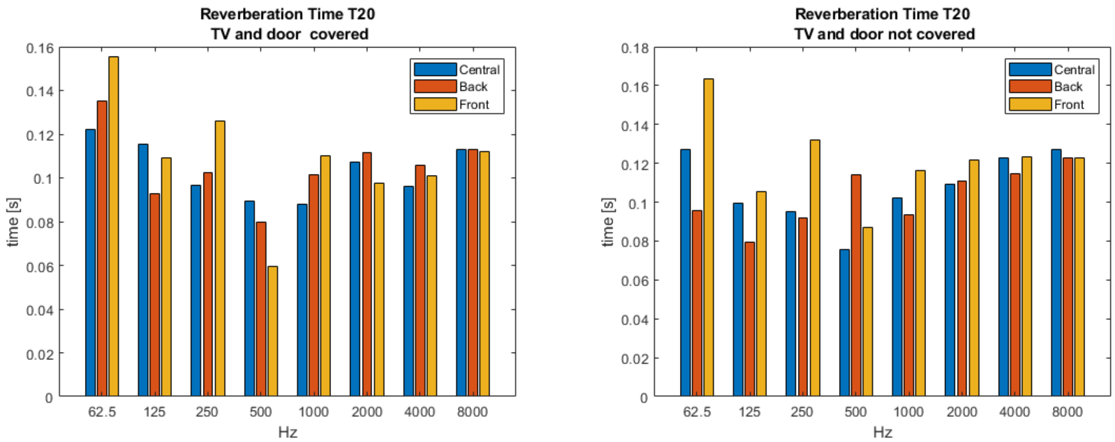


Figure 21: Balloon in different positions:T20

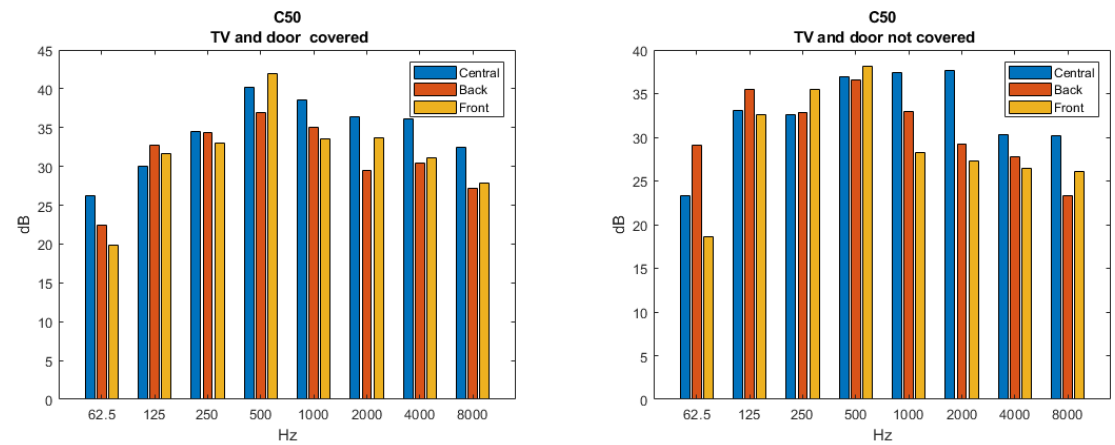
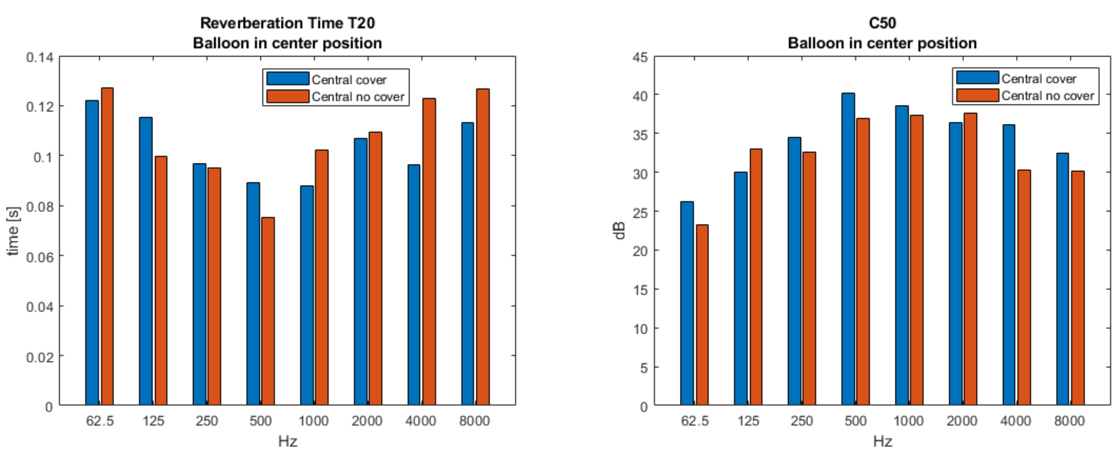


Figure 22: Balloon in different positions:C50

Focusing on the comparison between cover set up and not cover one in the central position:



It can be inferred that the optimal acoustic parameters are effectively achieved in both situations. Therefore, the decision is not to alter any room predisposition and leave it as is.
The room has been confirmed to possess heightened semi-anechoic characteristics, distinguished by exceptional clarity and minimal reverberation. This is guaranteed by efficient sound absorption achieved through a well-designed ambient projection.

4 Subwoofers analysis

During this analysis, the two subwoofers are placed beneath the seat.
A non-predistorted 15 second sine sweep, 10-10000 Hz, 0.2 fade in, was sent to the subwoofers for the examination of them. The microphone was positioned, as always, at the driver's seat.
The obtained impulse response of the two simultaneously playing, inside the virtual room, is depicted below:

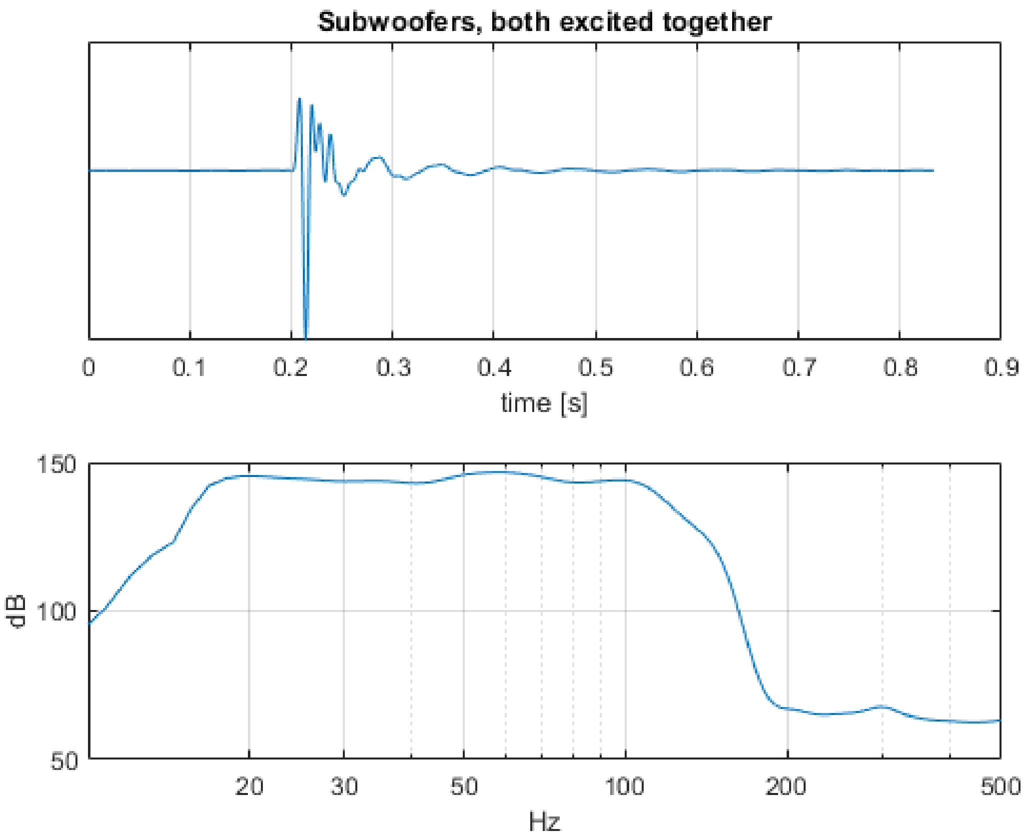


Figure 24: subwoofer analysis

Additionally, the impulse response of the sum of the single responses aligned is depicted below:

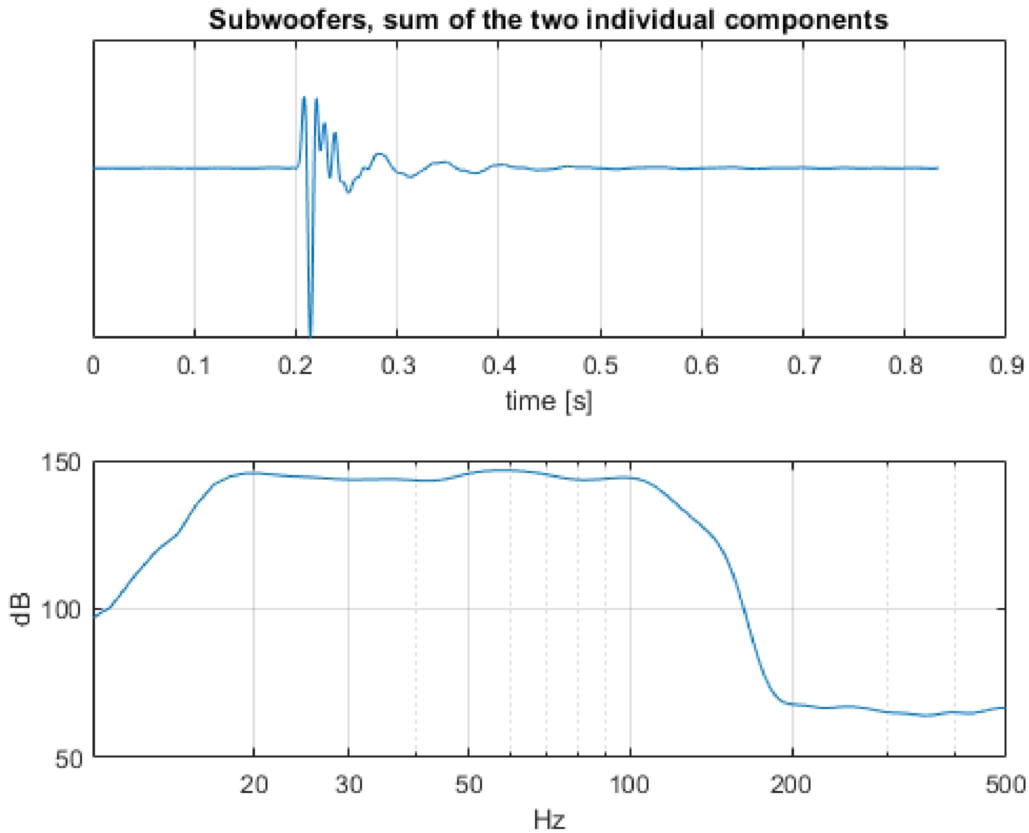
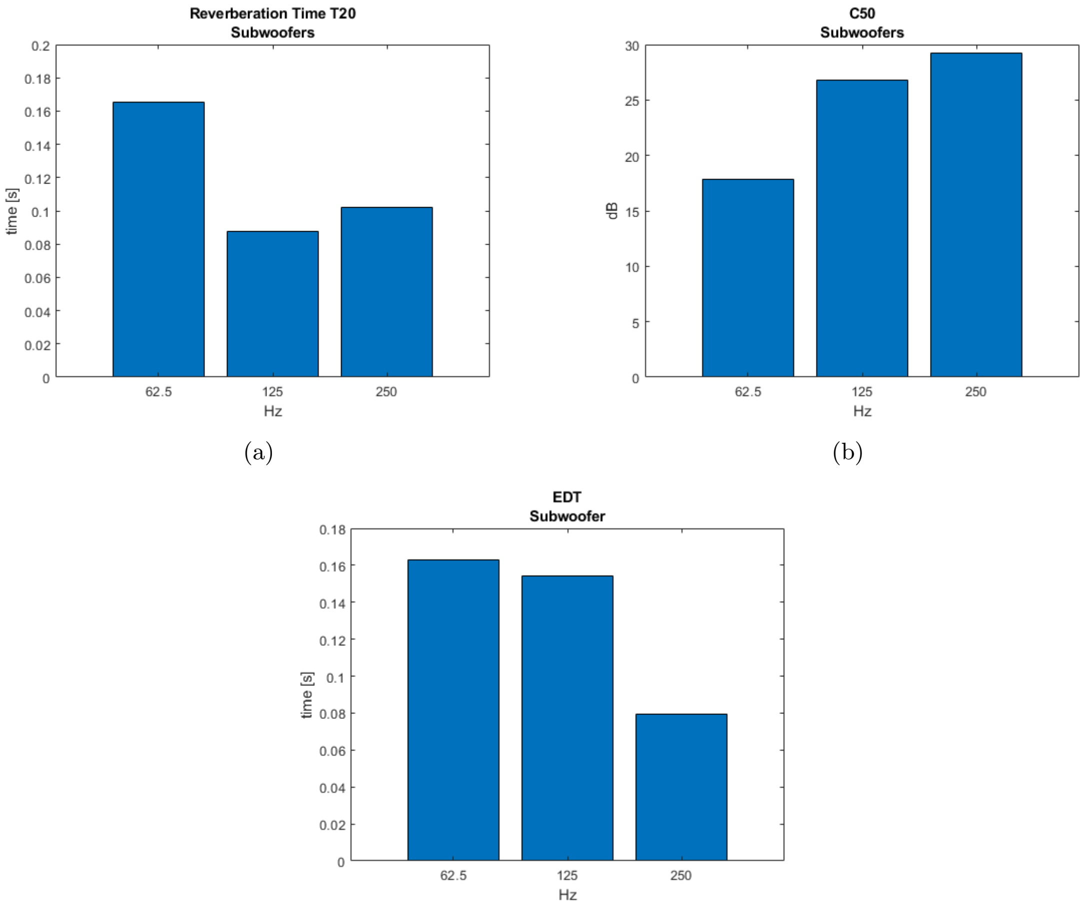


Figure 25: subwoofer analysis

The spectrum exhibits adequately flat frequency behaviour within the desired range. Consequently, there is no need for predistortion to address any irregularities.
Since the responses obtained with the two methods, 24 and 25, are nearly identical, the parameter extraction is carried out using the data from the first signal, 24.



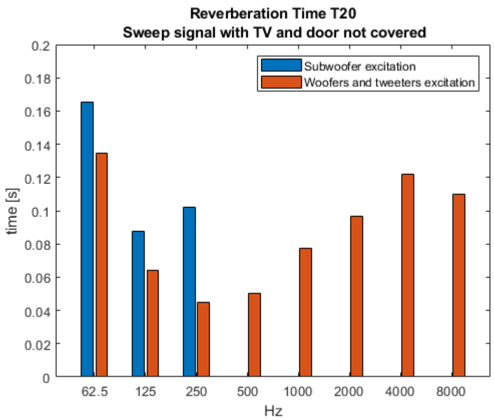
(a)

(b)

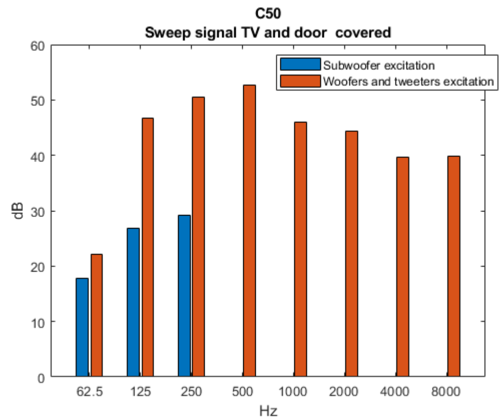
(c)

Figure 26: Subwoofer excitation, acoustical parameters

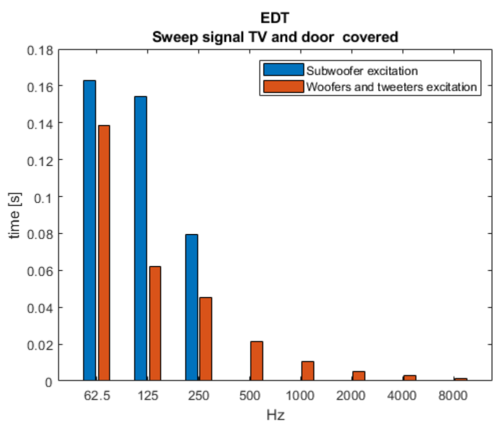
In this final plot, for a clearer visualization, the resulting values are presented when the room is excited by the sweep from the subwoofer and from the speakers at their respective frequencies:



(a)



(b)



(c)

Figure 27: Subwoofer excitation, acoustical parameters

5 Next steps

Once the structure and characteristics of the room, as well as the subwoofer placement, have been defined, the tuning phase of the speakers within the virtual room is initiated.
The objectives to achieve are as follows:

- Gain match between the speakers
- Ensures phase alignment between each woofer and tweeter.
- Adjust the delays between each speaker ensuring phase alignment.
- Generate a Finite Impulse Response (FIR) filter for each speaker to achieve equalization.

As a final step, the Ambisonics system will be evaluated both with and without the customized tuning for the room, utilizing specified parameters such as Spatial Correlation and Level Difference to assess the audio system.

References

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[3] Claus Lynge Christensen, George Koutsouris, Jens Holger Rindel, 'The ISO 3382 Parameters: Can we simulate them? Can we measure them?', Toronto, Canada International Symposium on Room Acoustics, June 9-11, 2013.

[4] Mladen Russa, Luka Kraljević, Maja Stella, Marjan Sikora, 'Acoustic Performance Analysis of Anechoic Chambers Based on ISO 3745 and ISO 26101: Standards Comparison and Performance Analysis of the Anechoic Chamber at the University of Split', University of Split, FESB – Laboratory for Smart Environment Technologies, R. Boškovića 32, Split, Croatia.

[5] Andrzej Miskiewicz, Tomira Rogala, Teresa Rosciszewska, Tomasz Rudzki, Tadeusz Fidecki:'Concert Hall Sound Clarity: A Comparison of Auditory Judgments and Objective Measures', Laboratory of Musical Acoustics, Department of Sound Engineering, The Fryderyk Chopin University of Music, Okolnik 2, 00-368 Warszawa, Poland.

[6] Andrea Venturi, Angelo Farina, Lamberto Tronchin, 'On the Effects of Pre-processing of Impulse Responses in the Evaluation of Acoustic Parameters on Room Acoustics', ICA 2013 Montreal, Montreal, Canada, June 2013.

[7] Densil Cabrera, 'Acoustic Clarity and Auditory Room Size Perception', Faculty of Architecture, Design and Planning, University of Sydney, Sydney, NSW 2006, Australia.

[8] Doheon Lee, Jasper van Dorp Schuitman, Xiaojun Qiu, Ian Burnett, 'Development of a Clarity Parameter Using a Time-Varying Loudness Model', June 8, 2018.

[9] Lamberto Tronchin, Giulio Ernesto Curà, Valerio Tarabusi, 'The Enhancement of the Arlecchino Listening Room: Adding Stereo Dipole to Ambisonics', DIENCA – CIARM, Bologna.