UNIVERSITAT POLITÈCNICA DE VALÈNCIA

School of Design Engineering

Design and Implementation of a 15W, 2.1 portable speaker with bluetooth connectivity and digital signal processing using the ESP32 microcontrollers

End of Degree Project

Bachelor's Degree in Industrial Electronics and Automation Engineering

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Abstract

This project encompasses the design and implementation of a portable speaker with a power draw of 15W, bluetooth connectivity, DSP technology and 5h battery life. This is a multidisciplinary project covering different fields of study such as acoustics, baffle design, system control, microcontroller programming, hardware implementation and 3D modelling among others.

Resumen

Este proyecto engloba el diseño e implementación de un altavoz portátil con una potencia de 15W, conectividad bluetooth, tecnología de DSP y duración de batería de 5h. Este es un proyecto multidisciplinar que abarca diferentes campos de estudio como la acústica, diseño de bafles, control de sistemas, programación de microcontroladores, implementación de hardware y modelado 3D entre otros.

Resum

Este projecte engloba el disseny i implementació d'un altaveu portàtil amb una potència de 15W, connectivitat bluetooth, tecnologia de DSP i duració de bateria de 5h. Este és un projecte multidisciplinari que comprén diferents camps d'estudi com l'acústica, disseny de bafles, control de sistemes, programació de microcontroladors, implementació de maquinari i modelatge 3D entre altres.

Acronyms

I^2S Inter-IC Sound. 17, 33–35

BMS Battery management system. 19, 20, 67

BW Bandwidth. 4

DAC Digital-analog converter. 17, 33, 34, 38, 39, 41, 45, 46, 68

DSP Digital signal processing. 2, 4, 6, 11, 53

FR Frequency response. 2, 4, 6–8, 10, 11, 14, 15, 41, 42, 45, 47, 49, 51–53, 67, 68

HPF High-pass filter. 10, 43

IC Integrated circuit. 17, 19

LPF Low-pass filter. 10, 39

SoC System on a chip. 18

TF Transfer function. 4, 6, 41, 43, 49, 50

µC Microcontroller. 2, 14, 16, 17, 19, 22, 33, 34, 41, 53

Contents

Part I

Technical Report

Chapter 1

Object

The basis of this project consists of designing and building the best portable speaker possible in terms of sound quality, battery life and versatility. The system will incorporate a wide variety of technologies such as bluetooth, Digital signal processing (DSP), advanced baffle design methods and Microcontroller (µC) control.

There are various measurable parameters that provide information about the sound quality of a speaker. The most relevant of them is the Frequency response (FR), which, if its measured in the correct conditions, defines the transfer function of a speaker.

When it comes to the design process of a speaker, the driver and the enclosure are the basis of the final FR, and at the same time, they are the principal cause of irregularities in its linearity.

DSP algorithms can manipulate the FR of a sound system in a very precise way and can thus improve it.

This document summarises the application of these technologies to a portable loudspeaker, including the design and physical implementation.

Chapter 2

Background

2.0.1 Evolution of speakers and sound technology

Since 1857 Édouard-Léon Scott de Martinville invented the Phonautograph in Paris, a lot of different devices were invented to record and play audio signals. Although these devices were capable of transmitting sound, it wasn't possible to play audio signals in real time until the invention of what is known as a loudspeaker driver.

A loudspeaker is essentially an electroacoustic transducer. In other words, converts electric signals into sound. It works by the same principle than a dynamic microphone, but is connected in reverse.

At first, the development of loudspeakers was driven by the need of a real-time oral communication at long distances, causing that the first loudspeakers were installed in telephones by Johann Philipp Reis back in 1861.

In 1915 Peter L. Jensen and Edwin Pridham made the first moving coil (dynamic) loudspeaker as a part of their product "Magnavox" and quickly became the standard.

The improvement of loudspeaker technology continued, primarily driven by the motion picture industry.

During the 1950's the invention of the electric guitar and the growing popularity of rock 'n' roll music increased the need for better and louder sound systems.

Since then and until now, the music industry has continued to grow massively, and increasingly, the general public has been demanding a better sound quality.

2.0.2 Definition of sound quality

Acoustic enclosures and speakers in general are extremely complex systems when it comes to modelling and predicting their FR. This causes that, even two speakers from the same model and manufacturer can have differences in their FR.

There is another factor that has a huge impact in FR, which is the room acoustics. If a speaker is placed and measured in two different rooms the two FR graphs obtained will be completely different.

That means that the sound system is not playing each frequency with the same energy, thus changing the nature of the original recording.

It is because of these reasons that worldwide music producers and sound engineers decided to create a standard for their recordings, which consists on a flat response and a BW as wide as possible. If the frequency response of soundsystems used in professional recording studios is measured, the FR plot will be very similar in every case. The reason of that relies on the fact that sound engineers want to make decisions based purely on the actual sound that was recorded and not on a sound that is influenced by the speakers and/or room acoustics TF.

This standard created a reference for what is known as sound quality. If a sound system does not manipulate the nature of the sound as it was recorded originally, the listener will hear the best recreation of what the sound engineer created in the studio. In other words, the quality of a speaker can be quantified by the linearity and bandwidth of its FR.

2.0.3 The arrival of DSP active speakers

Before DSP started to become relevant in the audio industry, speakers were composed basically by a driver, an enclosure and, in some cases, passive analog filters to separate the signal if there were various drivers per baffle.

As the resources were limited, manufacturing flat FR speakers implied a complex mathematical design, precise building and expensive materials. Moreover, when it was intended to design a speaker that could handle low frequency signals, it usually resulted in large and heavy enclosures. If there existed any mistake in the design or manufacturing process that could affect the FR of the final product, it would not be possible to correct it once it was built.

The arrival of DSP technology in audio applications changed drastically the way that personal audio products were being designed and manufactured back then. At first, DSP was used primarily for communications and speech processing for telephones. It wasn't until the era of smartphones and IoT in the decade of the late 2000's that DSP microchips started to get much more powerful and interesting, bearing in mind that, at this time, the vast majority of audio records were distributed in digital formats as MP3.

Nowadays, most of the personal audio playback devices in the market incorporate DSP technologies for various applications, one of them being the correction and improvement of the FR of speakers and headphones.

By the use of DSP in a speaker, it is possible to manipulate the final FR of it after it was designed and manufactured. To add on, it also brought the possibility of designing speakers with a reduced size that could handle considerably low frequencies. This permitted brands to create speakers with better sound quality at a lower price than before.

The integration of these products into the market resulted in an scenario where the average consumer of multimedia content is enjoying better sound quality. At the same time, this is causing consumers to increasingly demand better sound quality, what is resulting in an exponential market growth of the music industry and the portable wireless speaker brands.

As can be seen in the next figure, these conditions are making the bluetooth speaker market grow year after year.

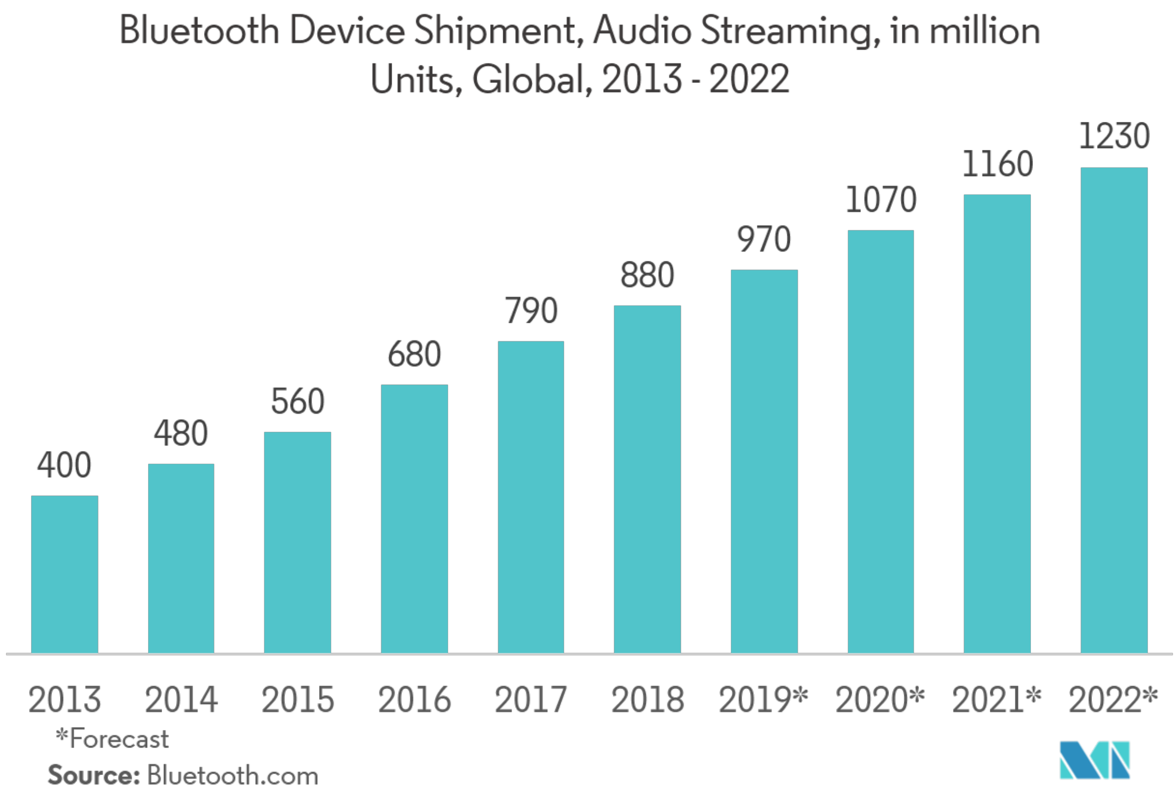


Figure 2.1: Bluetooth speaker market share analysis provided by bluetooth.com

Chapter 3

Needs, limitations and considerations

3.1 Needs

The main objective of this project is building a portable speaker with bluetooth connectivity, sufficient acoustic power for an open-air environment and a battery life long enough for an extended playback session.

At the same time, the FR has to be the flattest possible, and the bandwidth should cover as many frequencies as it can be.

The device has to allow the user to program the TF of the DSP stage once it is fully assembled.

As the product will be used outdoors, building materials have to be able to resist high levels of humidity and ultra-violet radiation.

3.2 Limitations

The size and weight of the speaker is the biggest limitation. It should be small and light enough to be carried in a regular backpack. To quantify this criteria, the size limit will be 350x200x200mm, meaning that the final product has to fit in a rectangular prism of the mentioned dimensions.

Manufacturing costs per unit should not exceed 400€.

3.3 Considerations

Some of technical features that supply the needs exposed before can conflict with the limitations at some point. One example of that is the fact that the speaker bandwidth is dependant of the enclosure size, meaning that the bass response will be limited by how big and heavy the final product is desired to be.

Internal enclosure volume is also affected by the volume of internal components as the battery cells, which limits the maximum amount of components to the point that it creates a deviation between the real and predicted internal volume of the enclosure.

At the same time, the most expensive part of all the components is the battery. The price of it will grow linearly with the chosen capacity.

All the stages of the system will have an accumulative impact in the final sound quality. That means that, to achieve the flattest FR possible, it is needed to make decisions carefully and make sure that each stage by itself is as transparent as possible with the signal.

The objective is building a 2.1. system, which consist of a pair of mid-high frequency drivers, that provide stereo sound, assisted by a mono channel woofer that extends the bass response.

The enclosure design must be as precise as possible to reduce the need of doing corrections in the future. Therefore, the electro-mechanical modelling coefficients will have to be measured experimentally in order to design the enclosure.

Chapter 4

Alternatives and justification of the chosen solution

4.1 Driver layout

When it comes to choosing a driver layout for a speaker a lot of different configurations are possible, each one with their benefits and inconveniences.

4.1.1 Mono

Mono audio is the simplest configuration which can be used to design sound system. In essence, it means that there is going to be only one audio channel where sound is going to be played.

The most significant benefit is that it doesn't create nulls and peaks in higher regions of the FR caused by the interference of multiple sound sources.

Additionally, it commonly results in smaller speakers because of its simplicity and the fact that it only needs one enclosure to work. This explains why this configuration is the most chosen one in the smallest commercial bluetooth speakers.

Although mono audio only uses one channel to work, the output signal can be sent to more than one driver. This is what is called as a multiple way speaker.

The inconvenient of mono speakers is that the spatial sound image is null. The standard format in which music is distributed nowadays is stereo, and sound engineers use both channels to create a virtual image of the different instruments in the song. This is called as binaural sound, and it makes the listening experience much more complex and rich.

4.1.1.1 Single way

A single way mono speaker is composed by only one driver and its enclosure.

4.1.1.2 Multiple way

A multiple way mono speaker is composed by two or more drivers and their corresponding enclosures. The signal is filtered to feed each driver with a certain range of frequencies in which it works most efficiently.

A 2-way baffle, for instance, has a woofer that handles lower frequencies and a tweeter that is driven with the higher frequency components of the signal.

Figure 4.1: 2-way baffle

To make a 2-way baffle work as intended, it is necessary to filter the signal with a crossover (either digital or analog) that separates the raw signal into two filtered ones by using a Low-pass filter (LPF) for the woofer and a High-pass filter (HPF) for the tweeter.

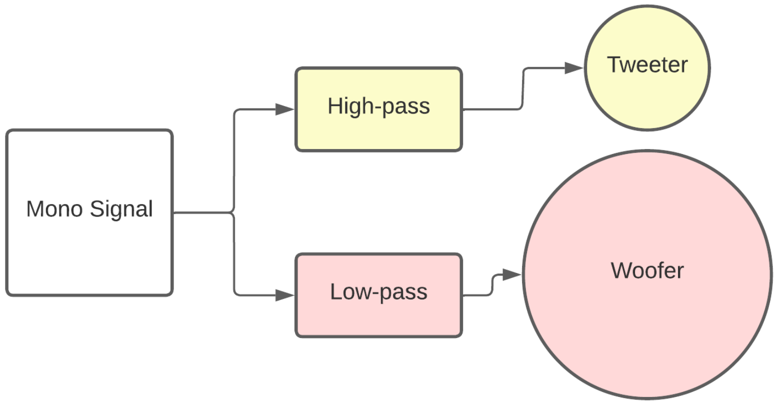


Figure 4.2: Block diagram of a crossover

4.1.2 Stereo

As mentioned before, stereo sound is always a priority if it is intended to achieve a pleasing listening experience. This factor is the only benefit of it and apart from that, all that comes next to the implementation of stereo in a sound system are inconveniences. Nevertheless, the relevance of it in the listening experience makes it a priority for high quality audio applications.

As stereo uses two output channels, each one containing different signals thus creating two separated sound sources, the complexity of the system increases significantly.

The most problematic phenomenon is the interference. As there exist two separated sound sources, the sound-field created is more inconsistent in its FR, which results in destructive and constructive interference events in those frequencies in the spectrum that have a smaller wavelength (i.e. high frequencies).

To demonstrate that, two FR measurements were made in an acoustically-treated room with a pair of reference studio speakers from the same brand and model.

The first measurement is made in mono conditions, by only using the right speaker. The second measurement was made with the two speakers playing the frequency sweep at the same time. In both cases the measurement microphone was placed in the listening position of the sound system, which is a place in the middle of both speakers where the listener sits down to achieve the best stereo image.

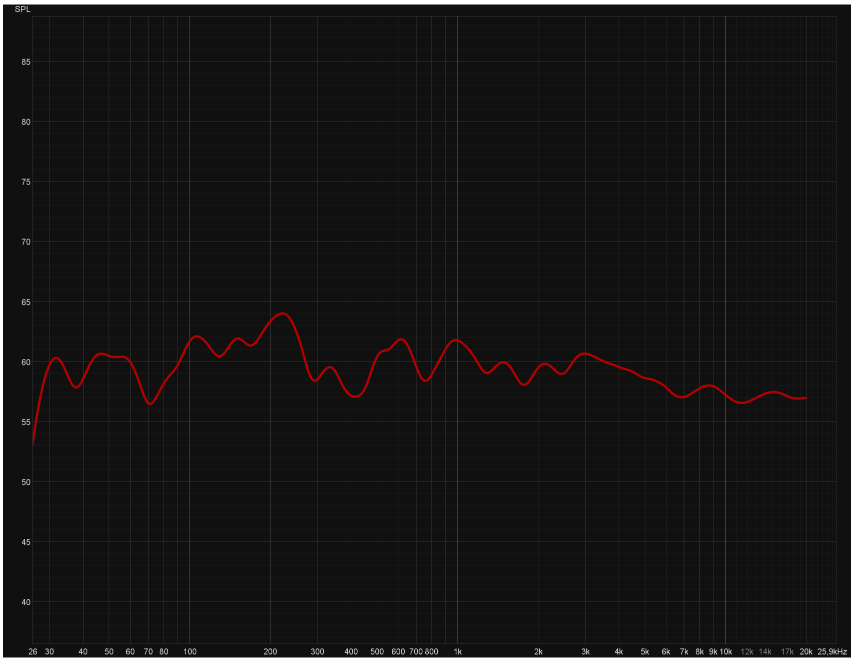


Figure 4.3: Measurement in mono

Figure 4.4: Measurement in stereo

As can be seen in the second graph, the high frequency region presents two noticeable nulls where sound waves are cancelled, reducing the consistency of the FR. This deviation cannot be corrected with the use of systematic DSP filtering, because the interference is not homogeneous and varies in the space.

It is worth to mention that, if the measurements were made in a perfect anechoic chamber which is free of standing waves, the lower region of the FR should look similar in both measurements.

This phenomenon is also influenced by how separated the two drivers are. As the separation increases, the frequencies affected are lower and lower. This has to be taken into consideration because the separation perceived by the listener increases when the drivers are separated as well. The best stereo image is achieved by creating an equilateral triangle between the speakers and the listening position.

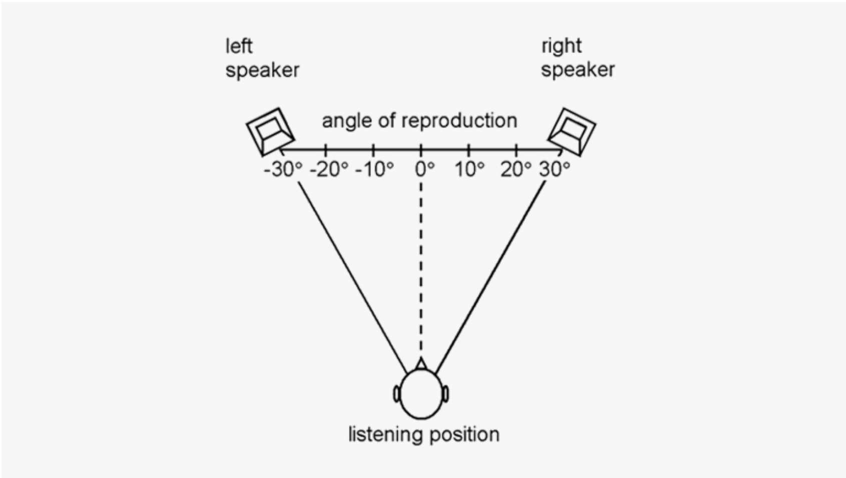


Figure 4.5: Ideal stereo sound system configuration

4.1.3 Mixed layouts

The earlier configurations can be combined with each other, leaving place for mixed layouts that provide different characteristics. The most common is 2.1 stereo.

The amount of combinations possible when it comes to speaker layouts is much larger. Keeping in mind that more complex layouts exceed the scope of this project, it is worth mentioning 3-way baffles, line arrays or surround systems like the 5.1 or 7.1 configuration.

4.1.3.1 2.1 configuration

A 2.1 sound system is composed by a woofer in mono configuration and a pair of tweeters in stereo. As the input signal is stereo, it needs to be separated with a crossovers. The woofer channel also needs to process the signal to convert it into mono.

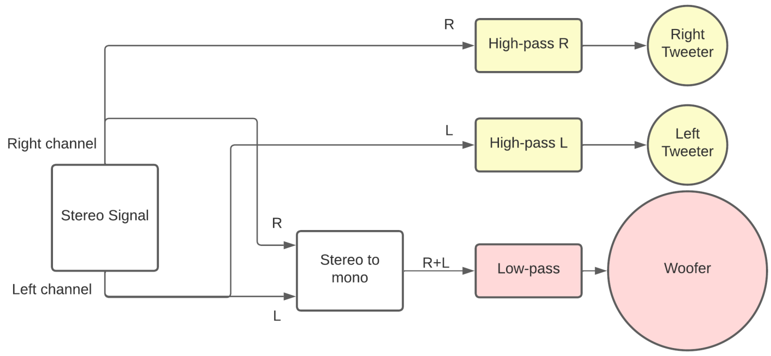


Figure 4.6: 2.1 sound system block diagram

This system is interesting because it combines some of the benefits of mono systems without losing the compatibility with stereo sound. The largest and heaviest parts of a speaker are those that manage lower frequencies. For example, a woofer driver usually has a diameter from 4 to 18 inches, while tweeters don't tend to have a diameter of more than 3 inches. The same happens with enclosures, as a woofer enclosure will always be far bigger than a tweeter one.

By reducing the amount of woofer drivers the size of the speaker gets considerably smaller, and the same thing occurs to the electronic components.

It can also increase efficiency in case that the track that is being played has low frequency phase issues between the two channels, because the signal cancellation occurs at the processing stage and not when the drivers are being fed with power.

Based on all the information exposed above, it was decided that this driver layout was the one that suits the project needs the best.

Chapter 5

Detailed description of the chosen solution

The speaker that is going to be designed and implemented is a 2.1 baffle that uses a µC to do all the signal processing required in order to separate the frequency components and correct the final FR of the system.

5.1 Block diagram

To get a basic technical idea of the project, there is a block diagram that shows the processing stages that the signal goes through.

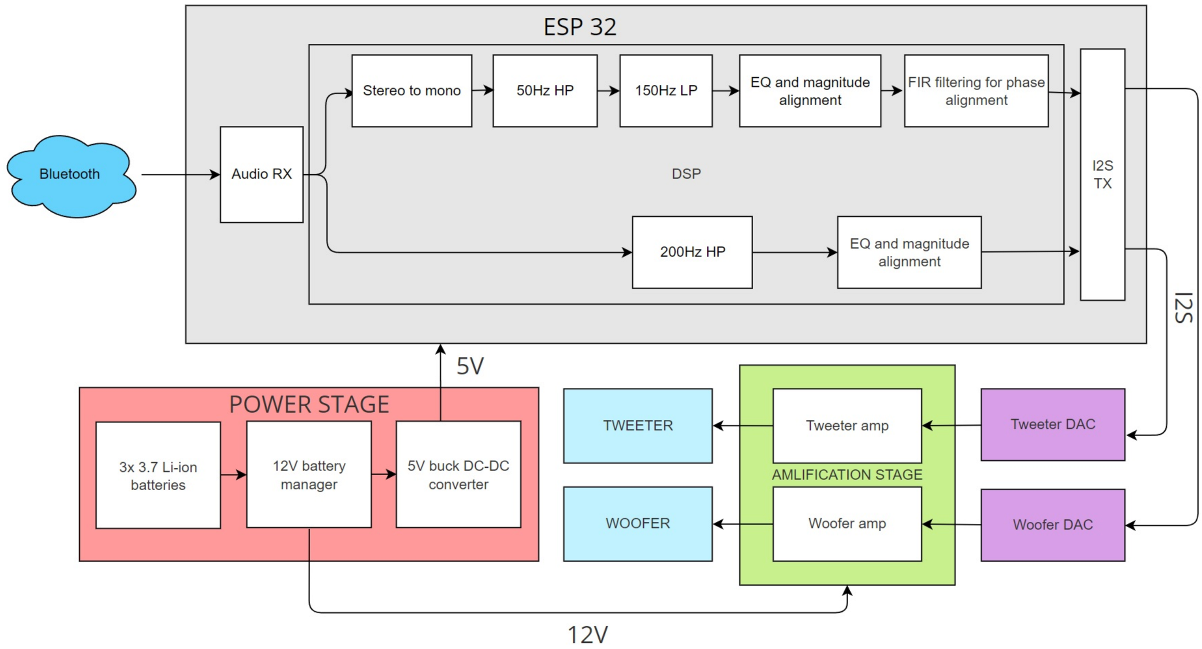


Figure 5.1: Block diagram of the system

Some of the values shown on the block diagram are for guidance. It will be necessary to measure the FR once the product is assembled to determine the required signal process dedicated to correct the FR.

The design of the system shall be approached in reverse. That means that the first design and implementation decisions concerning the processing of the signal will cover the last steps of the chain exposed in the block diagram, while the last decisions will determine the behaviour of the stages where the signal goes through at first. This is due to the very nature of the project, where the signal is intended to be continuously adjusted to correct real acoustic deviations in the frequency response. For example, DSP algorithms algorithms are to be configured after the final system's response is measured acoustically.

That said, the first step of the design process is the material selection.

5.2 Materials

5.2.1 Hardware

5.2.1.1 Mid-high tweeter driver

The drivers used for high and mid frequency stereo signal reproduction will be a pair of 2" paper cone drivers with a rated power of 5W each.

Figure 5.2: Tweeter driver

5.2.1.2 Woofer driver

The driver used for low frequencies mono reproduction will be a 4" paper cone driver with a rated power of 25W and very high efficency.

Figure 5.3: Woofer driver

5.2.1.3 Microcontroller

In order to archieve the best frequency response the audio signal will be processed using DSP algorithms that will be coded into the Microcontroller (µC).

It has been decided to use the SoC ESP32 from the company Espressif for this project, because of its small size, low power consumption, low price and the fact that it has bluetooth, Wi-Fi and radio integrated.

Figure 5.4: Microcotroller

Taking advantage of the fact that we have a Microcontroller (µC) unit, the final product could be upgraded in the future to include useful functions as power saving mode, extra bass mode, battery indicator or volume control.

5.2.1.4 DAC module

It will be needed to have 3 independent output channels. Therefore, It won't be enough with a single stereo Digital-analog converter (DAC) unit.

By using 2 stereo DAC the system will be provided with 4 independent audio outputs.

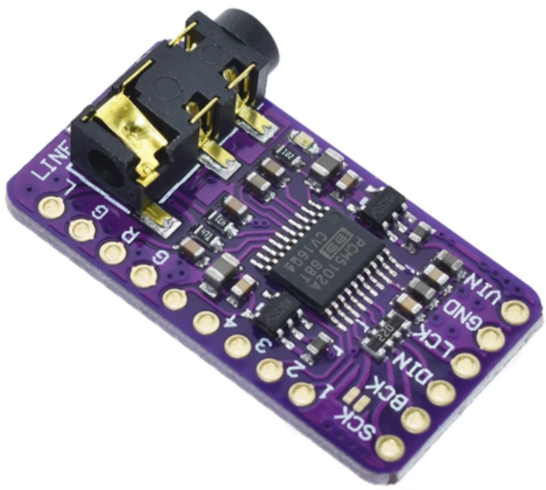


Figure 5.5: DAC module

The Digital-analog converter (DAC) module is based on the IC PCM5102A from Texas Instruments, which is designed to work with PCM data and Inter-IC Sound (I^2S) communication.

These are the fetures given by the manufacturer:

- DAC channels: 2

- DAC SRN: 112 dB
- Sampling frequency: 384 kHz
- Interface: I2S
- Data format: PCM
- Resolution: 32 bits
- Very low noise level
- Integrated high-performance audio PLL
- Intelligent mute system
- Power supply: 3.3 V or 5 V

5.2.1.5 Power amplifiers

The two tweeter drivers will be amplified by a stereo amp. On the other hand, the woofer driver will be powered with a mono amp.

The amplification stage will be done with linear response amplifiers, because the audio will be received by the SoC via bluetooth in order to process it digitally, so it's not needed to get an amplifier which includes analog filtering, for example.

The operating voltage will be 12V and, in this case, efficiency is a priority, which leds to the decision of using class-D amplifiers to improve battery life and reduce the size of the speaker.

Tweeters

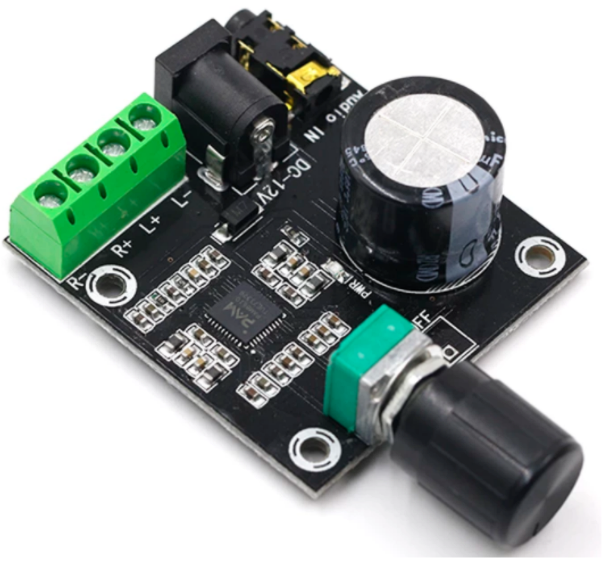


Figure 5.6: Tweeter amplifier

This is a class-D 15+15W amplifier that integrates the PAM8610 circuit, a very high quality and high efficiency amplifier. In some cases, a class-D amplifier can reduce the sound quality of high frequency speakers, because it digitalizes the signal in the amplification process. However, this model has a fast sampling frequency and provides a very good output signal.

Woofer

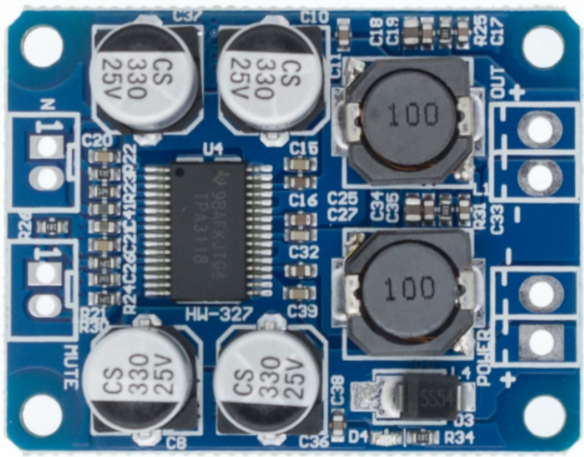


Figure 5.7: Woofer amplifier

The woofer amplifier circuit uses the IC TPA3118 class-D 60W mono amplifier, which will give enough power for the woofer unit.

Both amplifiers have different sensibilities, so it will be necessary to adjust the gain individually in the µC in order to get the flattest response possible before filtering the final frequency response.

5.2.1.6 Battery pack

A battery pack will be built to suit the project's power needs. It will be made out of a Battery management system (BMS) and 3.7 Li-Ion cells that provide 12V with enough current to power all the electronics at full volume.

The BMS unit is based on the IC bm3451, which controls the charge and discharge cycles of a set of 3 to 5 3.7V batteries in series. By using the 3S configuration, the battery pack will deliver a nominal and maximum voltage of 11.1 V and 12.6 V, respectively.

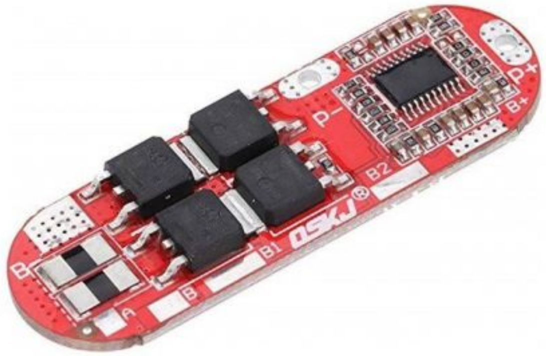


Figure 5.8: BMS module

To achieve the best battery life possible, it is critical to use high capacity cells. Another important aspect for better power efficiency lies in internal resistance. The lower it is, the more efficient the battery pack is when delivering power. For those reasons, the INR18650-M6 cells from LG were chosen for this project. They are rated with a minimum capacity of 3450 mAh per cell and 3.6V nominal voltage, which, assuming that the battery pack will be made of 9 cells, ultimately translates into a capacity of 111.78 Wh and a maximum output current of 25A (limited by the BMS).

5.2.2 Test equipment

5.2.2.1 Measurement microphone

For acoustic measurements, the chosen microphone was the Behringer emc8000, which provides an omnidirectional polar pattern and a flat frequency response.

Figure 5.9: Measurement microphone

5.2.2.2 Data acquisition interface

The acoustic and electronic frequency response measurements were done using the sound interface Komplete Audio 2 from the company Native Instruments. This device is essentialy an ADC with its corresponding signal conditioning hardware for high precision and low noise measurements.

Figure 5.10: Audio interface

5.2.2.3 Oscilloscope

In this project it was used the oscilloscope Rigol DS1054Z, mainly to measure signal quality and channel delay and synchronization.

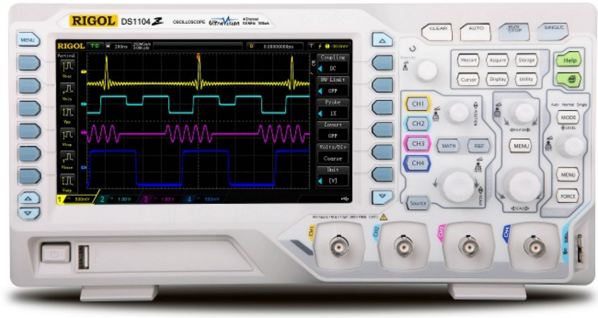


Figure 5.11: Oscilloscope

5.2.3 Software

5.2.3.1 ESP-IDF

For the programming of the Espressif µC, the actual company offers a C programming environment based on the real-time operative system FREERTOS, Python scripts and CMake compiler. It also contains all the libraries and example projects needed for most of the cases.

5.2.3.2 Visual Studio Code

The IDE chosen was VS Code, an open-source code editor and programming environment released by Microsoft. From this software, it is possible to open a terminal and call commands to build and flash the code into the µC.

5.2.3.3 Matlab

Matlab will be used for filter design purposes and transfer function simulation, which includes the discretization process of continuous filters.

5.2.3.4 WinISD

WinISD is a simulation software which eases the task of designing speaker enclosures given the desired design parameters and electro-mechanical coefficients of a physical driver.

5.2.3.5 Room Equalizer Wizard

Room Equalizer Wizard is an open-source software mainly used for room acoustics measurement. Which is interesant about this software is that it can be used to measure frequency responses of electric signals, plotting a very accurate bode diagram of the system. Then, it will be used for acoustic measurement of the drivers, as well as the DAC output frequency response. Another useful feature of this software is the frequency-dependant impedance measurements and Thielle-Small parameter calculation, which will be used to measure the modelling variables required by WinISD in order to design the speaker enclosure.

5.2.3.6 Fusion360

This software will be used to design the enclosure of the speaker, as well as preparing the files required for the 3D printer and the laser CNC cutter.

5.3 Design process

5.3.1 Enclosure design

The earlier project stages have the biggest impact on the final result. Therefore, to achieve a flat frequency response, it is necessary to start with a good-sounding enclosure.

5.3.1.1 Driver's free-air frequency response measurement

To make a decision about the enclosure dimensions and frequency tuning, it is common to do an open-baffle measurement of the driver[1].

The testing equipment includes a mount for the drivers, a measurement microphone and a sound interface with +48V phantom power.

Figure 5.12: Tweeter mounted on an open-baffle configuration

The resulting frequency response can give a brief idea of what can be expected of the final frequency response. The results are the following.

Woofer

As shown in the following bode diagram, the woofer driver will reach 90Hz with ease. By using a bass reflex enclosure and DSP bass enhancements we can expect it to reach the 75Hz mark.

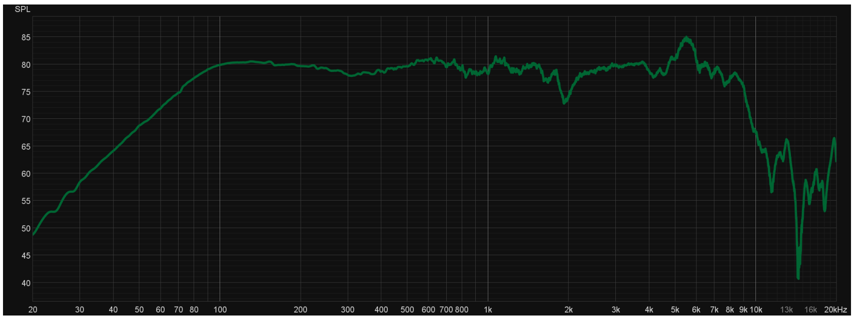


Figure 5.13: Woofer open-baffle frequency response

Tweeter

On the other hand, the tweeter driver has a better response in the high-frequency section, whereas the sound pressure in the bass section is reduced. By looking at the bode diagram, the tweeter enclosure and the crossover frequency should be tuned to match a cutoff frequency of 200Hz

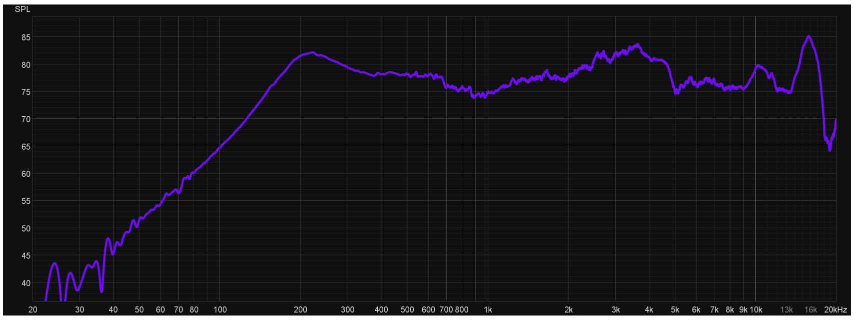


Figure 5.14: Woofer open-baffle frequency response

5.3.1.2 Driver's electro-mechanical coefficients measurement

To begin with the enclosure design it is needed to measure the Thielle-Small parameters [2] of the drivers.

These parameters are a set of electro mechanical modelling parameters that measure the driver's behaviour at low frequencies. They can be used to simulate the frequency response of speaker baffles for a certain driver. The software REW will be used for that purpose.

REW uses a model for the electrical impedance component of the driver based on "Electrodynamic Transducer Model Incorporating Semi-Inductance and Means for Shorting AC Magnetization"[3].

The mechanical impedance model incorporates elements that cater for the frequency-dependence of compliance. It uses the LOG model of viscoelasticity, from "Low-Frequency Loudspeaker Models That Include Suspension Creep" [4]

This model calculates the electro-mechanical coefficients of a driver from its impedance data. By using REW, the impedance can be measured and then use the obtained data to calculate the modelling parameters.

The measuring equipment needed includes an audio interface with two or more input channels and a measuring probe that can be built by following the next electric diagram.

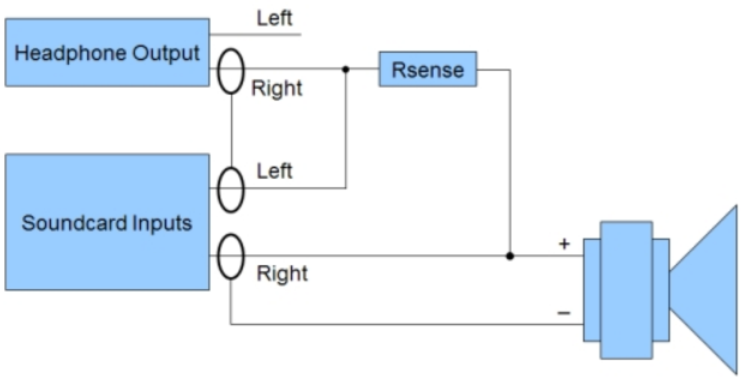


Figure 5.15: Driver impedance measuring probe electric diagram

It is important to calibrate the probe previously, which includes measuring the impedance of the terminals and the audio output of the interface.

There are several methods that we can use for measuring the thielle-small parameters of a loudspeaker. In this case, we are going to use the added mass method, due to its simplicity and accuracy.

This methods requires to measure the impedance of the driver in 2 different conditions.

Free-air impedance measurement The first one is the free-air measurement, which consists of measuring the driver horizontally, letting the cone vibrate freely in the air.

Figure 5.16: Driver free-air impedance measurement

Added-mass impedance measurement The second measurement is done in the same conditions than the first one, but this time, a known value of mass is added to the cone.

Figure 5.17: Driver added-mass impedance measurement

For this case, it was used 8.4g of adhesive putty, weighed on a precision scale. This product works great for this purpose and it's commonly used for high accuracy measurements.

Notice that the driver is standing on a bottle to let the air resonate freely under the membrane. This is done to reduce the effect of standing waves between the driver and the desk that can affect the experiment negatively.

Impedance measurement results

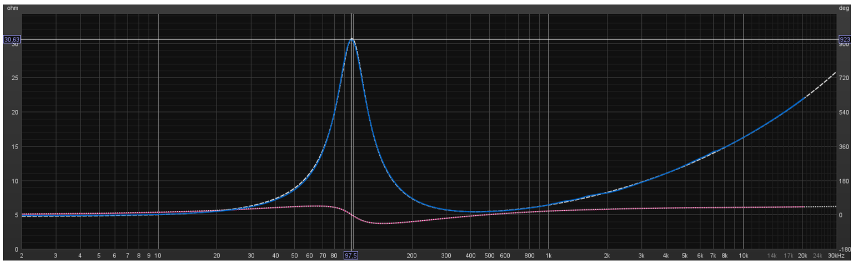


Figure 5.18: Woofer driver impedance bode diagram on free-air conditions

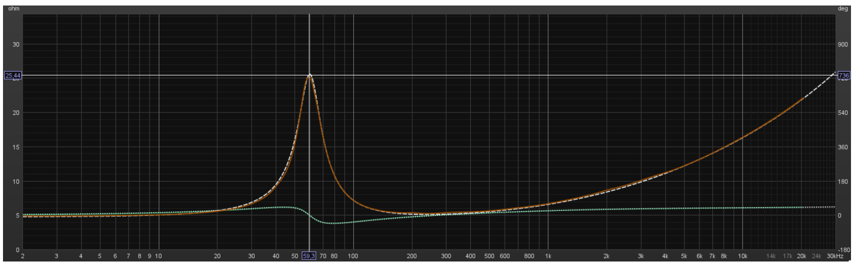


Figure 5.19: Woofer driver impedance bode diagram with an added mass of 8.4g

Thielle-Small parameter calculation

With the measurements that were done, REW can calculate the Thielle-Small parameters.

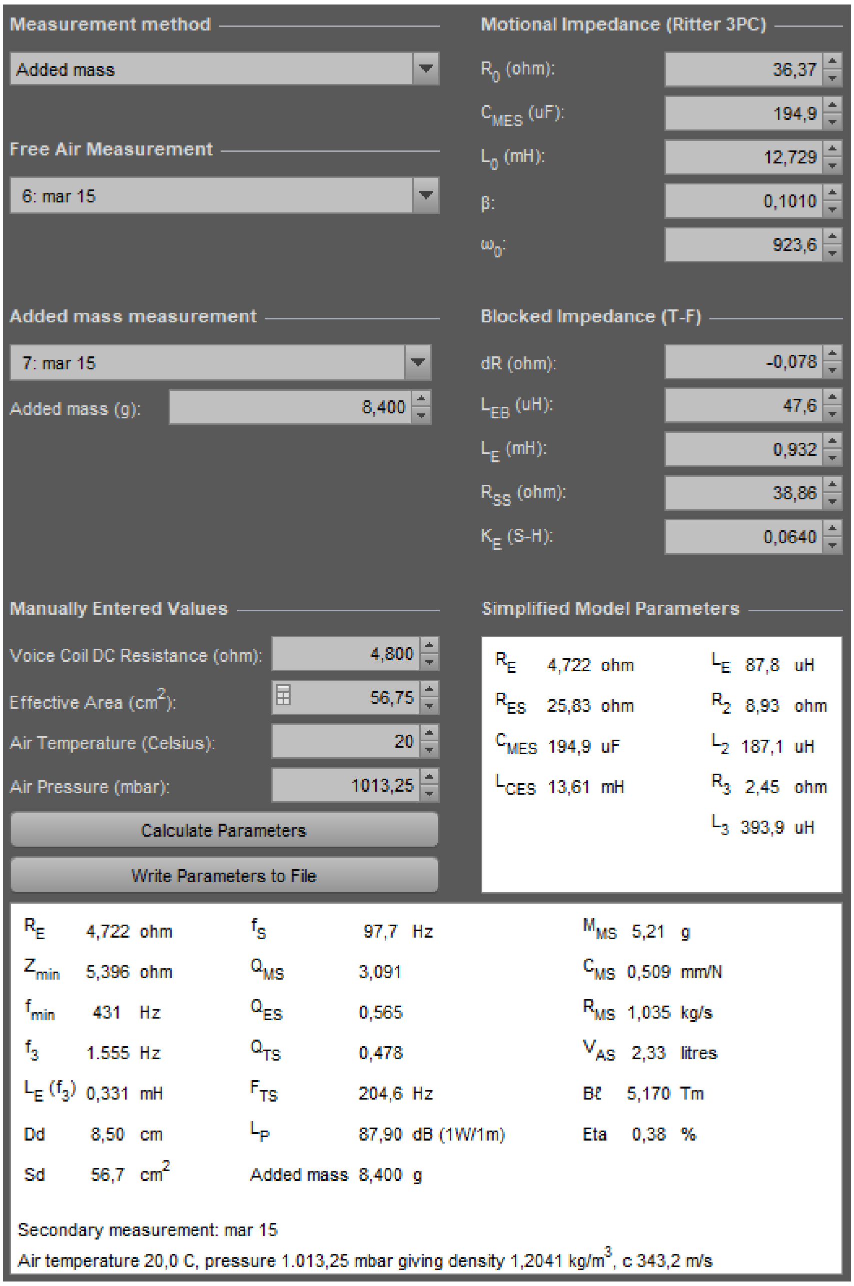


Figure 5.20: Woofer driver's Thielle-Small parameters

5.3.1.3 Acoustic simulation of the enclosure

After measuring both the tweeter and the woofer parameters, they can be introduced into the simulation software. This will be done by using WinISD.

The woofer baffle will be a bass-reflex enclosure[1]. This type of design is based on a second order system which includes a resonance at the tuning frequency, and gives a steeper rollof on the lower frequencies.

In other words, it gives a higher magnitude response at the tuning frequency by reducing the energy output at frequencies below.

As one of the objectives is achieving a high-efficency speaker with a long battery life, increasing the bandwidth to cover those frequencies that can be reproduced easily while attenuating the ones that are more power-hungry is a wise decision.

After introducing the parameters into the driver properties, the tuning frequency and volume of the enclosure can be adjusted. It is intended to extend the bass response as possible, but as a counterpart, it may create a dip in the magnitude of the frequencies above the tuning frequency.

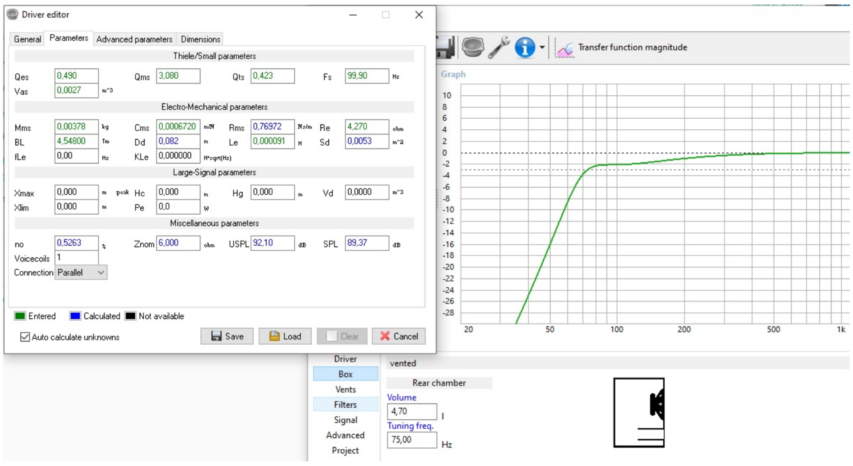


Figure 5.21: Woofer enclosure simulation on WinISD

Based on the two conditions mentioned above, the tuning frequency was adjusted to 75Hz, which is the sweet spot where the bandwidth is extended the most without compromising the higher bass frequencies excessively.

About the volume, as it increases, the bass port resonance and the frequencies above increase as well in terms of magnitude. It was finally adjusted to 4,7 liters, which is a reasonable size that won't result on an excessively big enclosure and doesn't make the port resonance too pronounced.

Once these two parameters were defined in the software, the bass port area has to be defined so the software can calculate the tube length.

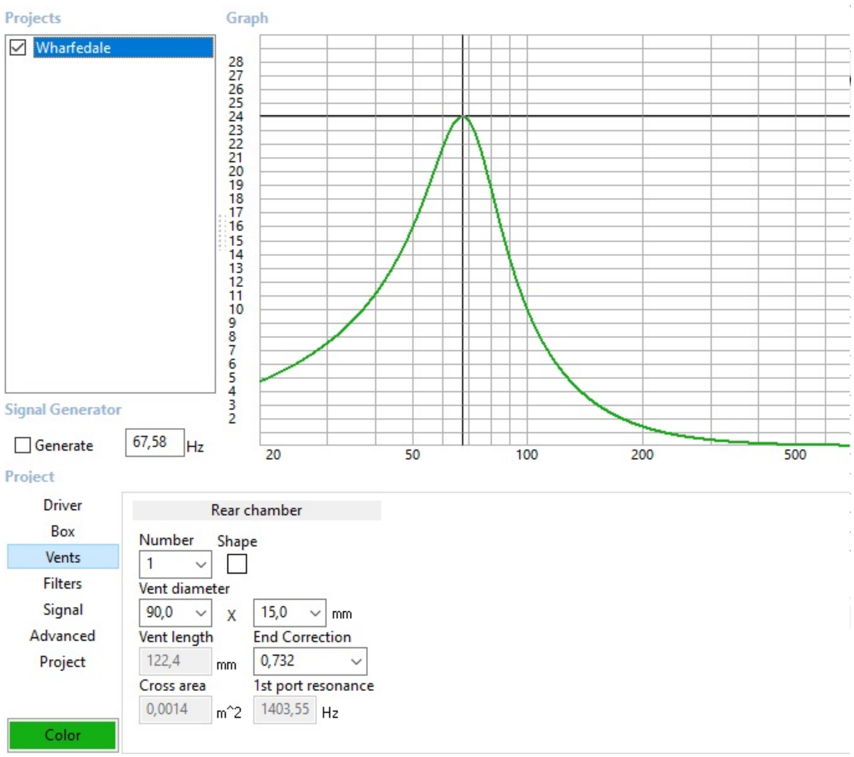


Figure 5.22: Woofer enclosure simulation on WinISD

As the surface area is increased, the port air velocity is reduced. A high air velocity introduces resonances in the final frequency response, causing harmonic distortion. This is something to take into consideration, as distortion is desired to be as low as possible. The limitation is that, as the surface is increased, the tube has to be larger in order to resonate at the desired frequency.

To decide the port area, taking into account the conditions above, the input power was set to 20W, which is slightly above the maximum power that is going to be applied to the woofer. Then, the air velocity inside the port can be plotted for that output power.

As documented in many speaker enclosure design guides, it should not exceed the 25 m/s mark at the resonant frequency, because at this point, the port distortion starts to be audible and it increases exponentially.

5.3.1.4 Enclosure 3D modelling

After the enclosure volume and tube shape has been determined, the continuing process lies in the enclosure modelling.

Once a design was decided to be the best one among some other sketches in terms of appearence, strucure consistency, practicity and simplicity, the enclosure was modeled to match the calculated parameters

The first step was to measure the driver's dimensions. With those dimensions, some sketches were drown on Fusion 360, and then, they were extruded to create the parts of the enclosure.

The image below shows an example of the modelling process.

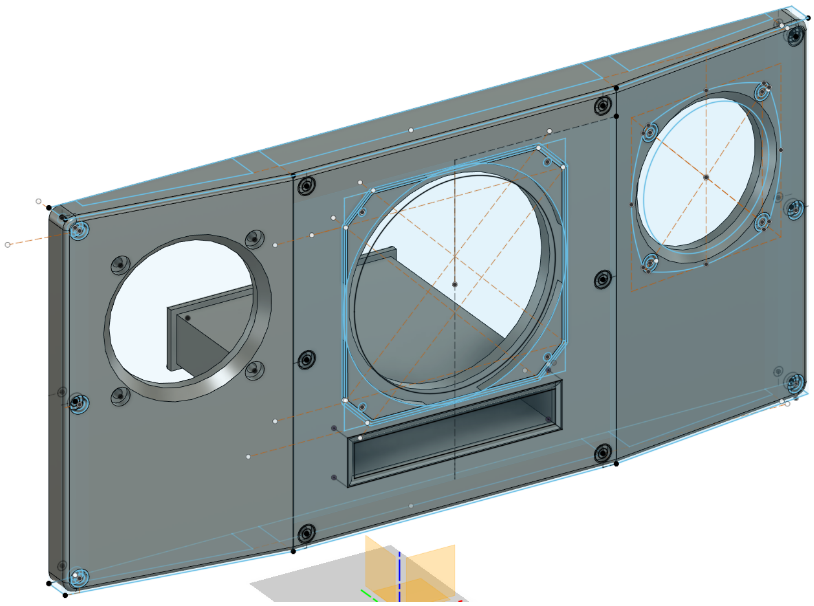


Figure 5.23: Enclosure's front panel modelling process

Considering that the front of the enclosure is going to be 3D-printed, it is convenient to separate some parts like the tube, because if something bad happens during the printing process, or the calculations were mistaken, it won't be necessary to discard the entire part.

Figure 5.24: Enclosure's final model for the front panel (front)

The tube is specially susceptible of miscalculations. Even a length offset of a few millimeters can change the resonant frequency of the tube, thus changing the bass response. Therefore, even considering that it would have been easy to print the front panel and the tube together, it was better to separate it.

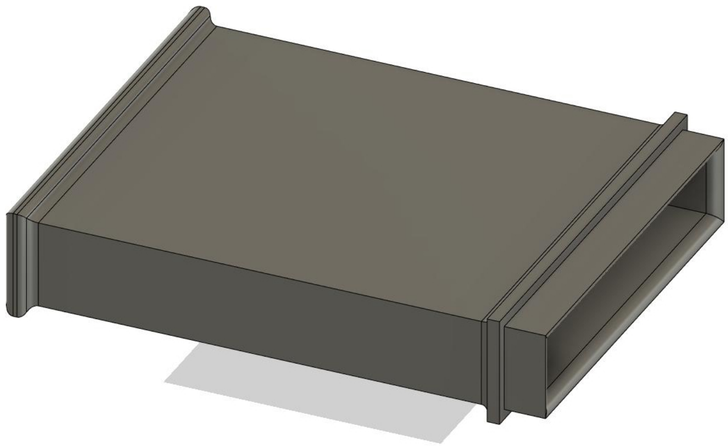


Figure 5.25: Enclosure's final model for the bass-reflex tube

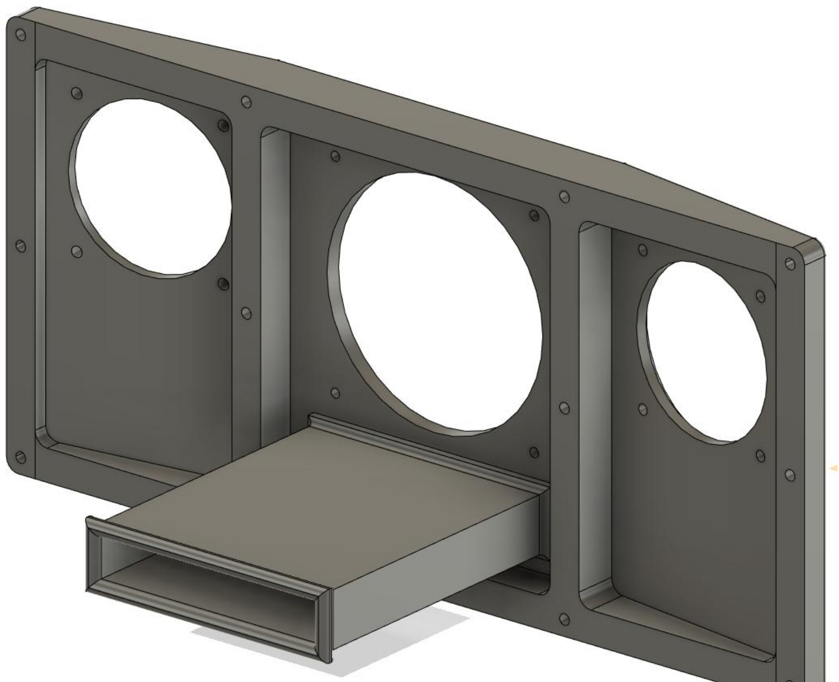


Figure 5.26: Enclosure's front panel and tube assembled in it (behind)

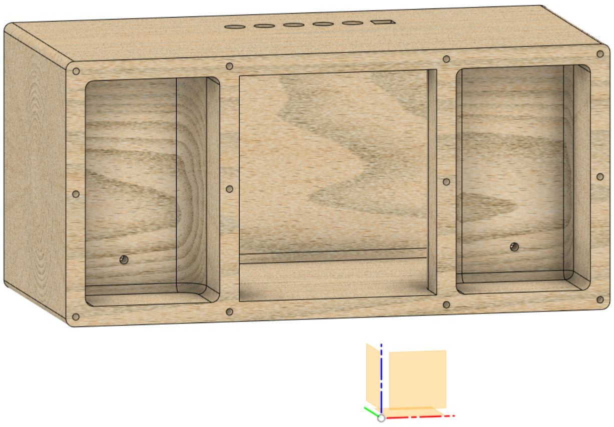


Figure 5.27: Back of the enclosure

Metallic treads were added for the screws, which makes the product much more durable if it's going to be assembled and disassembled regularly, which was expected to be a future task of the project. For that purpose, insertions were added in order to put nuts inside the screw holes.

Finally, an exploded view of the full enclosure.

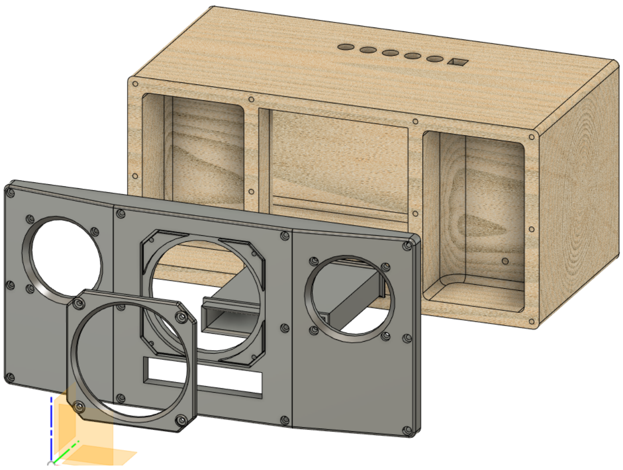


Figure 5.28: Exploded view of the model parts

Considering the frequency range that the speaker is going to reproduce, some research was made to decide the width of the enclosure walls. It was chosen to be 12 mm wide, printed with a 50% infill. The sound will be good enough and the speaker won't be too heavy.

5.4 Implementation

These are the steps taken in the implementation process. The earlier steps need to be completed in order to continue with the later ones.

1. Digital stage assembly
2. Digital data receiver programming (bluetooth)
3. DSP algorithm developing
4. Full system implementation
5. Filter adjustement and frequency response correction

5.4.1 Digital stage assembly

The digital stage is composed by the two DAC circuits and the µC. Audio will be received by the µC as bluetooth data, convert it into PCM format and transfer it to the DAC by using the I^2S peripheral.

The two DAC modules were assembled in a prototype PCB connected to the µC.

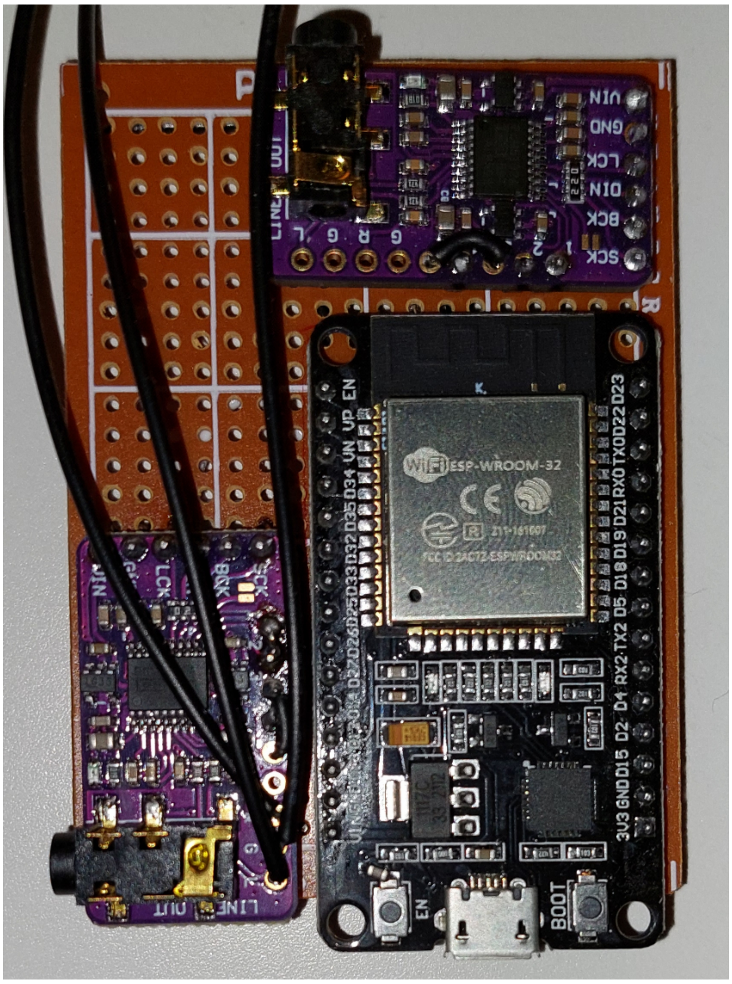


Figure 5.29: DAC and ESP32 soldered on a perfboard (top)

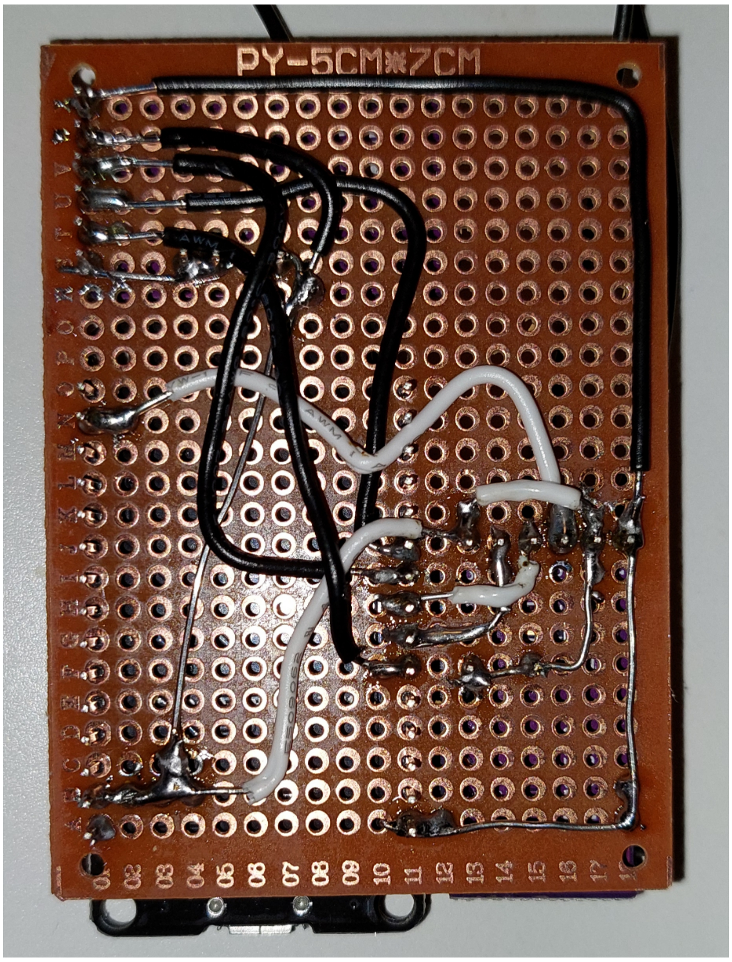


Figure 5.30: DAC and ESP32 soldered on a perfboard (bottom)

5.4.2 Digital data receiver programming

At this point, the next thing to do is programming and testing the ESP32 to ensure that the data is being received and transmitted correctly.

To start with it, it is needed to download the ESP-IDF tools for Espressif devices. It can be found in the official website of the manufacturer [5].

This pack includes example projects for various applications. Each one already incorporates the FREERTOS libraries and the C structures needed to configure operation modes and peripherals. One of these projects is a2dp_sink, which is basically a bluetooth audio receiver that incorporates I^2S communication with the PCM-5102 DAC.

By building and flashing the project onto the µC, it was noted that some things needed to be changed:

- The output volume of the signal couldn't be changed.
- Every 10 seconds the output volume raised by a 10% automatically
- The program is configured for 2.0 stereo and it needs to be 2.1

The first problem mentioned was caused by the fact that the code did not include a function to multiply each audio sample with the gain desired, even though there is a 7-bit variable that stores the volume value numerically.

The variable data allocates the sample value in the memory of the ESP32 using PCM format, those samples being written by the ring-buffer that receives them via bluetooth. The function assigned for this task is called "I2S_Task_Handler", and it is also responsible of sending data blocks through the I^2S interface.

Knowing that, it's easy to implement the desired function.

```
static void bt_i2s_task_handler(void *arg)
{
    uint8_t *data = NULL;
    size_t item_size = 0;
    size_t bytes_written = 0;


/* RECEIVE DATA FROM RINGBUFFER AND WRITE IT TO I2S DMA TRANSMIT BUFFER */
    for (;;) {
     data = (uint8_t *)xRingbufferReceive(s_ringbuf_i2s, &item_size, (TickType_t)portMAX_DELAY);
        if (item_size != 0)
        {
         int16_t * pcmdata = (int16_t *)data;
         for (int i=0; i<item_size/2; i++) {
            int16_t temp = *pcmdata;
            temp = temp * logarithm[volume_step];
            *pcmdata = (int16_t)temp;
            pcmdata++;
      }

         // NEW POINTER FOR PARALLEL PROCESSING
            uint8_t* backup = (uint8_t*) malloc(item_size);

            // COPY SAMPLES FROM DATA TO BACKUP
            for (int i=0; i<item_size; i++)
            {
               *backup = *data;
               data++;
               backup++;
            }

            // SET POINTERS BACK TO THEIR ORIGINAL POSITION
            for (int c=0; c<item_size; c++)
            {
               data--;
               backup--;
            }

            // SIGNNAL PROCESSING
         process_woofer(data, item_size);
         process_tweeter(backup, item_size);
```

```
          // DAC0 AND DAC1 WRITE
         i2s_write(0, data, item_size, &bytes_written, portMAX_DELAY);
         i2s_write(1, backup, item_size, &bytes_written, portMAX_DELAY);


            vRingbufferReturnItem(s_ringbuf_i2s,(void *)data);

            // RELEASE DYNAMIC MEMORY USED BY BACKUP
            free(backup);
        }
    }
}
```

Note that, in the I2S_Task_Handler function, there are two i2s_write functions, one for each of the two i2c channels that are available in the ESP32.

This is done because, as mentioned earlier, 3 audio channels are needed, one of them receiving a different signal treatment than the other two, being that the woofer channel.

The logarithm variable is a lookup table that represents a discretization of a log10 function in 16 steps in total, and is a more efficient alternative than using the <math.h> library functions. This is neccesary to control the volume consistently, because the human psychoacoustic perception of sound pressure works in a logarithmic scale [6].

It was decided to use 16 steps because the majority of volume control scrolls in today's android phones have 16 positions. In case that the connected device has different volume positions, the program will set the volume step which is nearest to the desired volume.

Then, a criteria is established for dB increments in volume. 100% means 0dB, and from there, a constant amount of decibels is subtracted each step, in this case, being -2 dB. Therefore, the first step is -inf dB, the second one -30 dB, the third one -28 dB and this series continues until the last step, 0dB, which is the same as a 1 in the linear scale.

After this, we have to convert our dB steps into linear scale gain, which is the one that will be multiplied by the samples in order to achieve the desired output.

This conversion can be made with this equation:

$$Gain = 10^{\frac{dB}{20}}$$

By using this formula, the next lookup table is created.

dB	Linear gain factor
0	1
-2	0.79
-4	0.63
-6	0.50
-8	0.40
-10	0.32
-12	0.25
-14	0.20
-16	0.16
-18	0.13
-20	0.10
-22	0.008
-24	0.006
-26	0.005
-28	0.003
-inf	0

The implementation in the code is as follows.

```
extern uint8_t s_volume;
static uint8_t logarithm[16] = {0, 0.03, 0.05, 0.06, 0.08, 0.1, 0.13, 0.16, 0.2, 0.25, 0.32,
    0.4, 0.5, 0.63, 0.79, 1};
```

The last thing to do to stream audio in 2.1 format is configuring the second i2s peripheral with the same parameters as the first one.

```
   /* I2S configuration parameters for channel 1*/
   i2s_config_t i2s_config = {
       .mode = I2S_MODE_MASTER | I2S_MODE_TX,          /* only TX */
       .sample_rate = 44100,
       .bits_per_sample = 16,
       .channel_format = I2S_CHANNEL_FMT_RIGHT_LEFT,   /* 2-channels */
       .communication_format = I2S_COMM_FORMAT_STAND_MSB,
       .dma_buf_count = 16,
       .dma_buf_len = 128,
       .intr_alloc_flags = 0,                          /* default interrupt priority */
       .tx_desc_auto_clear = true                      /* auto clear tx descriptor on underflow
           */
   };

   /* enable I2S channel 1*/
   i2s_driver_install(0, &i2s_config, 0, NULL);

   i2s_pin_config_t pin_config = {
       .bck_io_num = 27,
       .ws_io_num = 33,
       .data_out_num = 32,
       .data_in_num = -1                               /* not used */
```

```
    };
    i2s_set_pin(0, &pin_config);

    /* I2S configuration parameters for channel 2*/
    i2s_config_t i2s_config_2 = {
        .mode = I2S_MODE_MASTER | I2S_MODE_TX,
        .sample_rate = 44100,
        .bits_per_sample = 16,
        .channel_format = I2S_CHANNEL_FMT_RIGHT_LEFT,
        .communication_format = I2S_COMM_FORMAT_I2S_MSB,
        .dma_buf_count = 16,
        .dma_buf_len = 128,
        .intr_alloc_flags = 0,                        /* default interrupt priority */
        .tx_desc_auto_clear = true                    /* auto clear tx descriptor on underflow
            */
};

    /* enable I2S channel 2*/
    i2s_driver_install(1, &i2s_config_2, 0, NULL);

    i2s_pin_config_t pin_config_2 = {
        .bck_io_num = CONFIG_EXAMPLE_I2S_BCK_PIN,
        .ws_io_num = CONFIG_EXAMPLE_I2S_LRCK_PIN,
        .data_out_num = CONFIG_EXAMPLE_I2S_DATA_PIN,
        .data_in_num = -1
    };

    i2s_set_pin(1, &pin_config_2);
```

The value assigned to the variable .dma_buf_len was changed from 64 to 128, because as it is needed to double the amount of data transmitted than when it was configured for using only one DAC, the ringbuffer also needs to be two times bigger. Omitting to do so will result in playback problems as missing samples and noises.

5.4.3 DSP algorithm implementation

5.4.3.1 Stereo to mono conversion

The first thing to do is obtaining a 2.1 signal, which means that one of the two stereo signals needs to be converted into mono. This is done by using a monosum algorithm, which sums the right and left sample into the output sample.

```
static void process_woofer (uint8_t * data, size_t item_size) {

   int16_t *samples = (int16_t *) data;
   int16_t *outsamples = (int16_t *) data;


   for (int i=0; i<item_size; i=i+4)

   {
```

```
      //COPY SAMPLES AND MAKE MONOSUM
      int32_t preinsample = *samples;
       samples++;
      preinsample += *samples;
       samples++;
       float insample = (float) preinsample;

       // LOW PASS FILTER
       insample = lpf_200Hz_W_ZOH(insample)*0.25; //GAIN ADJUSEMENT

       // 78Hz BOOST
       insample = Wf_78Hz_Boost(insample);

        // OVERFLOW PROTECTION
       if(insample >= 16383){insample = 16383;}
       else if(insample <= -16383){insample = -16383;}

       //SEND MONO SAMPLE INTO ONE OUTPUT CHANNEL
      *outsamples = 0; //MUTE THE LEFT CHANNEL
      outsamples++;
       *outsamples = ((int16_t) insample); //SEND TO RIGHT CHANNEL
      outsamples++;

   }
}
```

5.4.3.2 Crossover implementation and discretization method comparison

To prevent phase issues, It is mandatory to have the same phase response in the cutoff frequency [7].

As the tweeter enclosure is sealed, There is already a first order system which has a +90 degree phase shift in its rollof. It is intended to create a second order crossover, which means that the tweeter channel will implement a first order high pass, while the woofer low pass filter will be a second order one in order to match the +180 phase. Both cutoff frequencies must be equal.

IIR design can be done in the continuous domain, and then, discretize it. The most common discretization method for this purpose is Tustin.

Different discretization methods will be evaluated by measuring the signal quality out of the DAC.

Woofer low pass IIR filter

The next task will be adding the LPF seen in the code, which is the one that sets the crossover at 200Hz.

The most efficient way to create a second order transfer function with a cutoff frequency and damping factor that we desire is to use a normalized polynomial for butterworth filters.

This is the basic structure of a second order system:

$$\frac{kw_n^2}{s^2 + 2\zeta w_n s + w_n^2}$$

The denominator of a second order butterworth filter with a Wn of 1 rad/s is defined by this expression:

$$(s^2 + 1.414214s + 1)$$

Now it is known that the damping factor (ζ) must be 1.414214, which results in a quality factor of 0.707.

By knowing that the cutoff frequency is going to be 200 Hz -> 1250 rad/s, the filter's transfer function can be designed as follows.

The filter T.F. must be:

$$\frac{1250^2}{s^2 + 1.414 \cdot 1250s + 1250^2}$$

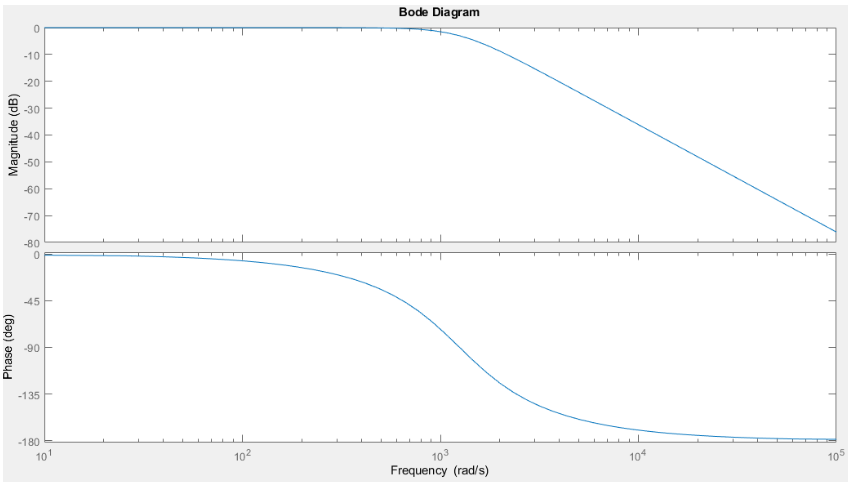


Figure 5.31: Bode diagram of the low pass filter

After discretizing the filter using the ZOH method, the next discrete T.F. is obtained:

$$\frac{0.0003964z + 0.0003911}{z^2 - 1.96z + 0.9607}$$

The resulting difference equation is:

$$Y_k = U + 0.0003964U_{k-1} + 0.0003911U_{k-2} + 1.96Y_{k-1} - 0.9607Y_{k-2}$$

The difference equation relates the output to the input and past values of input and output. In other words, the algorithm is an iteration that changes the processing made to the output every time that a new sample is introduced into the filter. This equation can be implemented in the code as a function.

```
static float wy = 0;
static float wy_1 = 0;
static float wy_2 = 0;
static float wu_1 = 0;
static float wu_2 = 0;

static float lpf_200Hz_W(float wu)
{
   wy = wu_1 * 0.0003964 + wu_2 * 0.0003911 + wy_1 * 1.96 - wy_2 * 0.9607;

   wu_2 = wu_1;
   wy_2 = wy_1;
   wy_1 = wy;
   wu_1 = wu;

   return wy;
}
```

Once the code is compiled and flashed into the µC, it is possible to measure the TF of the DAC output for a frequency sweep to obtain the FR at this point.

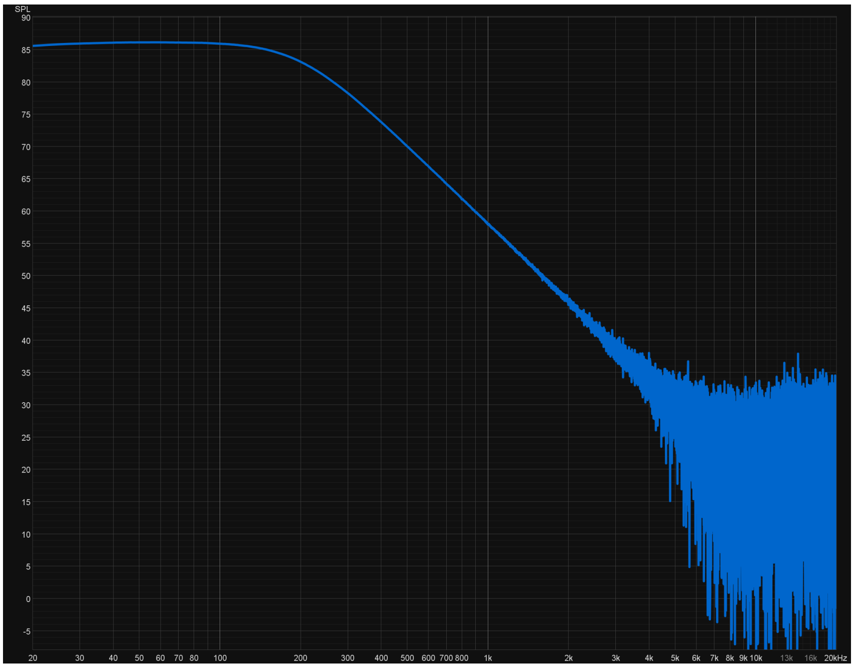


Figure 5.32: Measurement for low pass filter - ZOH

Tustin method test

On the other hand, if the TF is discretized with the Tustin method, the result is as follows:

$$\frac{0.0001969z^2 + 0.0003937z + 0.0001969}{z^2 - 1.96z + 0.9607}$$

The resulting difference equation is:

$$Y = 0.0001969U_k + 0.0003937Uk - 1 + 0.0001969Uk - 2 + 1.960Yk - 1 - 0.9607Yk - 2$$

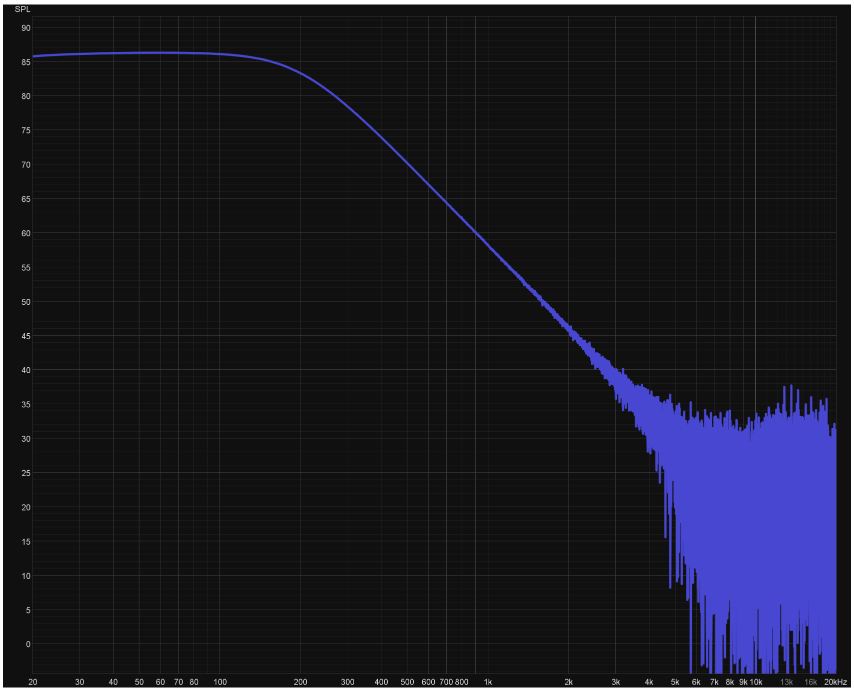


Figure 5.33: Measurement for low pass filter - Tustin

It is clear that there is no difference in terms of FR. It is also important to look at other parameters to compare the signal quality of these discretization methods.

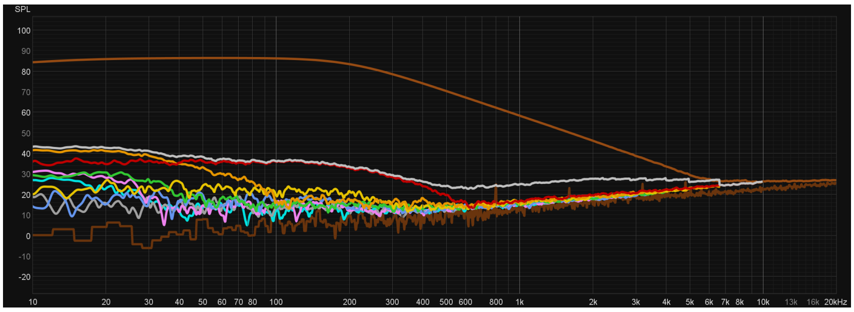


Figure 5.34: Distortion and background noise in ZOH filter

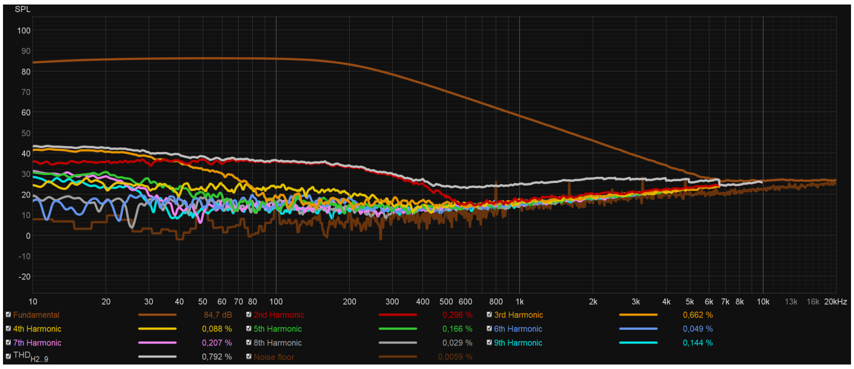


Figure 5.35: Distortion and background noise in Tustin filter

By looking at the distortion and background noise, it can be concluded that both filters behave similarly.

Note that in both cases the frequency response at high frequencies is very inconsistent. This is because, as the magnitude reduces, the signal to noise ratio becomes smaller too. As can be seen in the distortion measurement there is a trace that represents the background noise, and at high frequencies the noise overcomes the fundamental frequency.

Tweeter high pass IIR filter

The design process for the HPF algorithm starts by creating a transfer function with the desired characteristics.

It is needed to design a 1st order high-pass filter with a cutoff frequency of 200Hz, so let's begin by constructing the TF and simulate it in matlab.

- The filter must have a +20dB/dec slope from 0Hz to 200Hz. -> Derivator
- The slope from 200Hz onwards must be 0dB/dec. -> Pole with natural frequency of 200Hz

This is the resulting TF.

$$\frac{s}{s+1250}$$

This is the the system's response represented in a bode diagram.

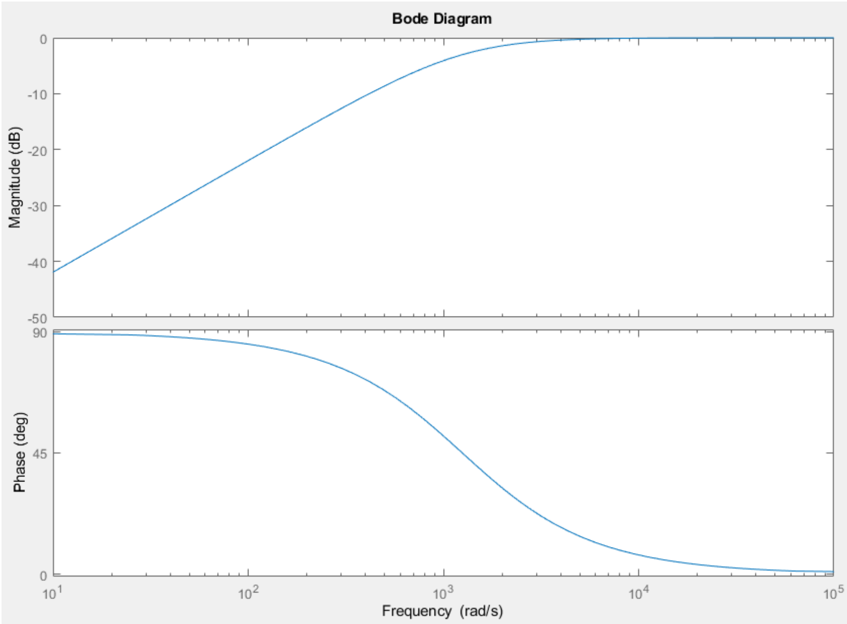


Figure 5.36: Bode diagram of a 200Hz High-Pass filter

Now the filter will be discretized using the ZOH method with a sampling frequency of 44.100 Hz, which is the sampling frequency configured in the ESP32 code.

This is the transfer function of the filter in the discreet domain.

$$\frac{z-1}{z-0.9721}$$

The resulting difference equation is as follows.

$$Y_k = U_k - U_{k-1} + 0.9721 Y_{k-1}$$

```
static float ly = 0;
static float ly_1 = 0;
static float lu_1 = 0;

static float ry = 0;
static float ry_1 = 0;
static float ru_1 = 0;

// LEFT CHANNEL FILTER
static float hpf_200Hz_L(float u)
{
  ly = u - lu_1 + 0.972053270286608 * ly_1;
  ly_1 = ly;
  lu_1 = u;
  return ly;
}

// RIGHT CHANNEL FILTER
```

```
static float hpf_200Hz_R(float u)
{
   ry = u - ru_1 + 0.972053270286608 * ry_1;
   ry_1 = ry;
   ru_1 = u;
   return ry;
}
```

It is mandatory to use two different functions for stereo filtering, one for each channel. It's not possible to use the same function for both channels, because the old iterations change the past input and output values.

After compiling the code, the DAC output FR was measured.

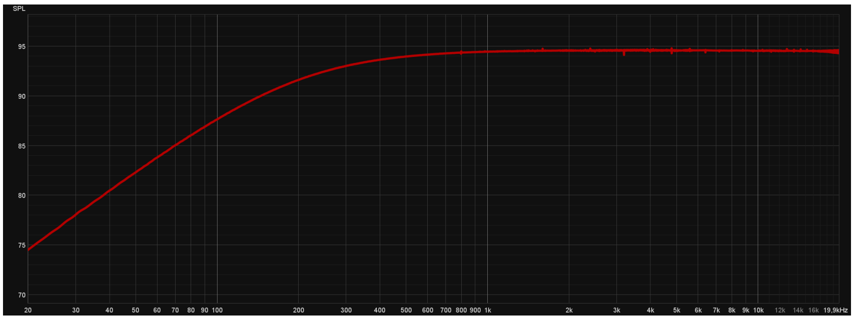


Figure 5.37: Frequency response measurement after adding the 200Hz High-Pass filter

5.4.4 Full system implementation

5.4.4.1 Hardware assembly

All the hardware components were connected and powered by the battery pack. To make it more robust, a 3D printed plastic plate was used to allocate each piece in its place.

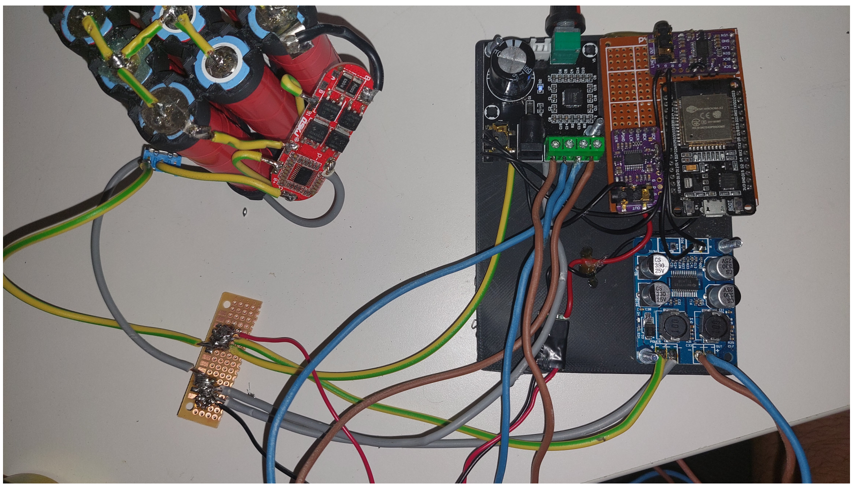


Figure 5.38: Hardware components mounted and powered by the battery pack

5.4.4.2 Enclosure building

The wooden part of the enclosure is made from two layers of 6cm thick plywood that were cut by a laser cutter. After all the pieces were glued together, screw nuts were added to make it easier to open the enclosure in the future.

The wood was treated with hydrophobic varnish to prevent water and environmental damage. As a last step, a rubber seal was glued over the nuts.

The front cover was printed using ABS plastic. After that, some sanding was needed to prepare the surface for painting. Finally, the piece was painted with acrylic spray paint.

After the two parts of the enclosure were finished, the electronics were introduced. Then, before assembling them together, some textile damping material was placed inside the tweeter cavities to reduce standing waves and enhance the frequency response of these drivers.

The holes in the enclosure were used to add a USB programming port, a charging port, a power switch, a button and two jack outputs connected to the DAC circuits for development and measuring purposes.

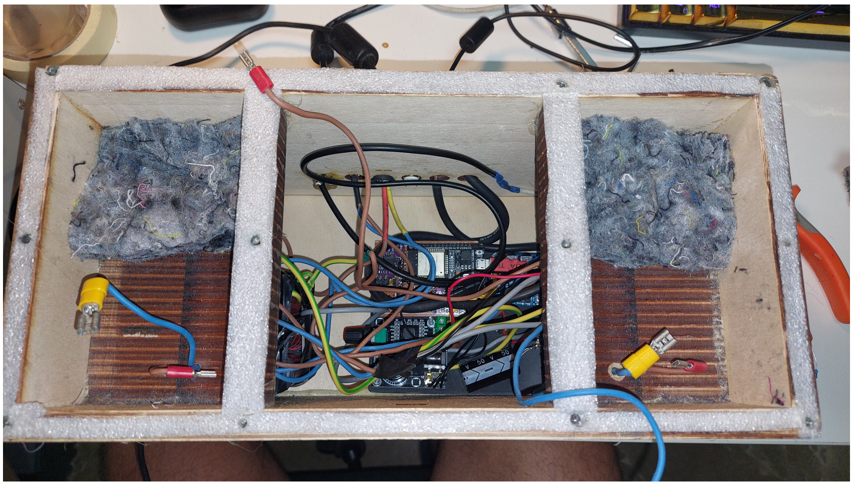


Figure 5.39: Enclosure finished before closing it with the front panel

Figure 5.40: Final product assembled

5.4.5 Filter adjustment and frequency response correction

After the system is built, the final task consists on correcting the FR.

The first adjustment consists in the input gain adjustment of the amplifiers. For that purpose, various measurements were taken in a process of trial and error. The final adjustment included the introduction of a voltage divider with resistors to reduce the amplitude before the woofer amplifier. The tweeter amplifier was regulated by using the in-built variable resistor.

5.4.5.1 Raw frequency response measurement

The decisions on the bass frequencies must be taken by looking at the full system response, because the woofer channel sums the stereo signal into mono.

On the other hand, for high frequency correction, measurements need to be taken in mono to avoid interference between the two tweeters.

The measurements were made in an acoustically treated room that follows the LEDE (live-end dead-end) scheme.

Figure 5.41: Measurement process with the microphone placed 1m away from the speaker

Full system

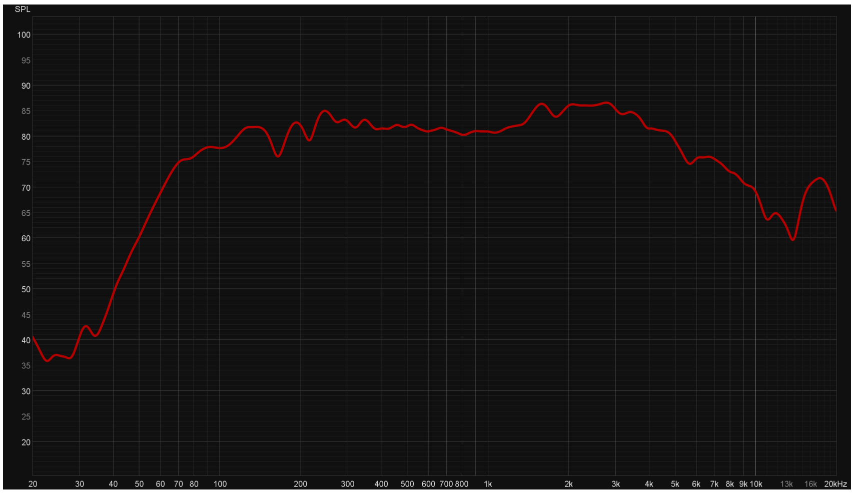


Figure 5.42: Acoustic frequency response of the speaker in full operation

By looking at the graph, it is clear that the bass region needs to be boosted to match the higher frequency magnitude.

Apart from that, the crossover frequency is right, as there is not a significant dip or peak in it.

Left tweeter muted

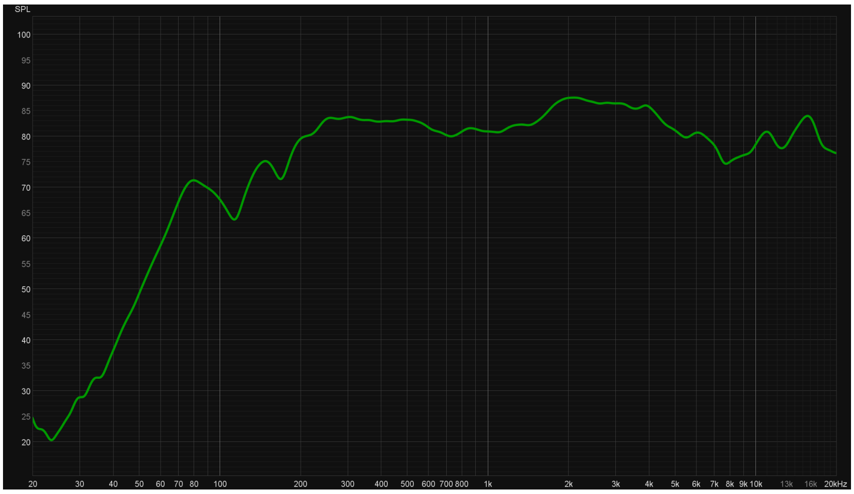


Figure 5.43: Acoustic frequency response of the right tweeter and the woofer together

Right tweeter muted

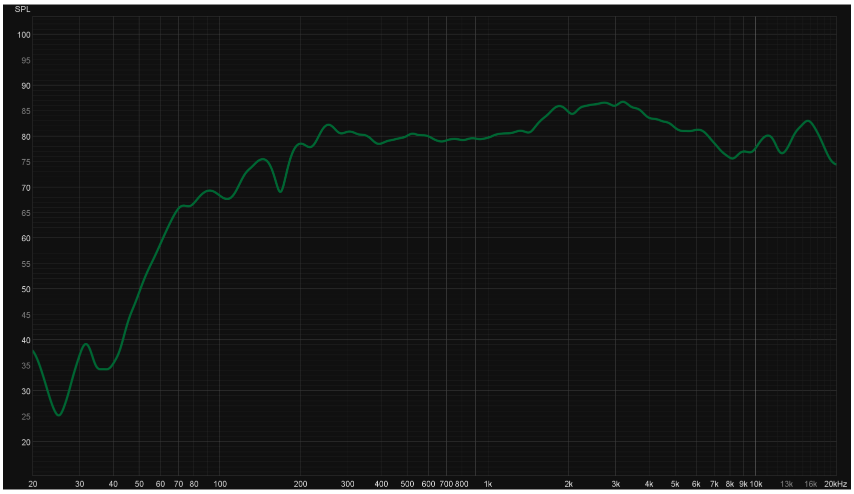


Figure 5.44: Acoustic frequency response of the left tweeter and the woofer together

On the other hand, the FR of the two tweeters has a notorious build up around 2.5 KHz. Therefore, it will be needed to reduce the magnitude on this region.

5.4.5.2 DSP Magnitude correction

Bass boost

A bell filter was applied at 78Hz to increase the magnitude in the bass region.

$$\frac{s^2 + 2.5 * 50 * s + 490^2}{s^2 + 50 * s + 490^2}$$

The TF was discretized using the tustin method, giving the next discrete system.

$$\frac{1.001z^2 - 1.999z + 0.998}{z^2 - 1.999z + 0.9989}$$

The code implementation is as follows.

```
static float wyb = 0;
static float wyb_1 = 0;
static float wyb_2 = 0;
static float wub_1 = 0;
static float wub_2 = 0;

// 78Hz Bass Boost
static float Wf_78Hz_Boost(float wub)
{
   wyb = 1.998743507439808* (wyb_1 - wub_1) + 0.998017058334367 * wub_2 - 0.998866890476781 *
       wyb_2 + 1.000849832142414 * wub;

   wub_2 = wub_1;
   wub_1= wub;
   wyb_2 = wyb_1;
   wyb_1 = wyb;
```

```
  return wyb;
}
```

Figure 5.45: Woofer's DAC frequency response measurement

2KHz treble cut

To reduce the magnitude at 2KHz, the tweeter channel was processed with a bell filter that has a gain of 0,4.

$$\frac{s^2 + 0.4 * 20000 * s + 14250^2}{s^2 + 20000 * s + 14250^2}$$

The resulting TF is as follows.

$$\frac{0.8914z^2 - 1.555z + 0.7466}{z^2 - 1.555z + 0.638}$$

And finally, the implementation in the code.

```
//Variables for 2KHz cut left
static float lyc = 0;
static float lyc_1 = 0;
static float lyc_2 = 0;
static float luc_1 = 0;
static float luc_2 = 0;

//Variables for 2KHz cut right
static float ryc = 0;
static float ryc_1 = 0;
static float ryc_2 = 0;
static float ruc_1 = 0;
static float ruc_2 = 0;

static float Wf_2kHz_Cut_L(float luc)
{
```

```
  lyc = 1.554677273766128 * (lyc_1 - luc_1) + 0.746611599629706 * luc_2 - 0.638016570899580 *
      lyc_2 + 0.891404971269874 * luc;

  luc_2 = luc_1;
  luc_1= luc;
  lyc_2 = lyc_1;
  lyc_1 = lyc;
  return lyc;
}

static float Wf_2kHz_Cut_R(float ruc)
{
  ryc = 1.554677273766128 * (ryc_1 - ruc_1) + 0.746611599629706 * ruc_2 - 0.638016570899580 *
      ryc_2 + 0.891404971269874 * ruc;

  ruc_2 = ruc_1;
  ruc_1= ruc;
  ryc_2 = ryc_1;
  ryc_1 = ryc;
  return ryc;
}
```

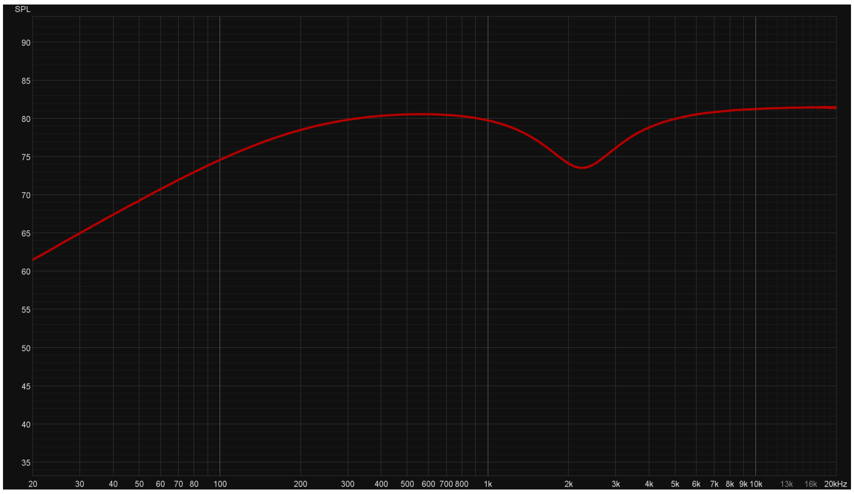


Figure 5.46: Tweeter's DAC frequency response measurement

5.4.6 Final frequency response

The next graph represent the FR of the full system in standard conditions.

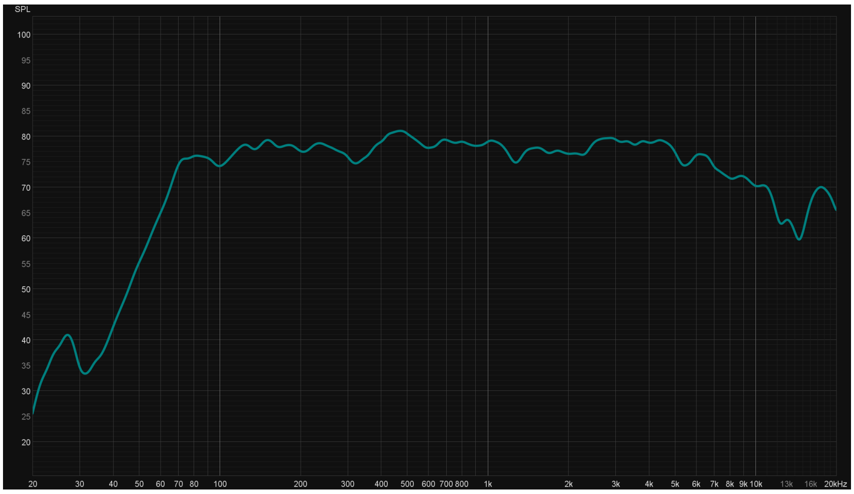


Figure 5.47: Acoustic frequency response of the system after the correction

As the graph shows, the FR improved enormously, and this is something that can be appreciated in the final sound without a doubt.

Note that the higher frequencies fall over 5KHz, due to the fact that is a stereo measurement in the center of the speaker where there exists a destructive interference. To evaluate the treble region, it is needed to do another measurement using only one channel and setting the microphone next to the tweeter.

The next graph represents the FR of the right channel of the speaker. The measurement was done in the same conditions than the previous one.

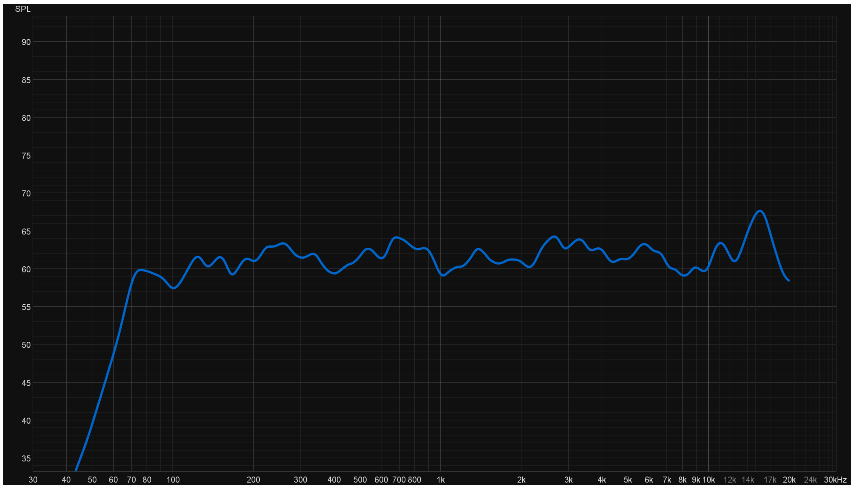


Figure 5.48: Acoustic frequency response of the right channel after the correction

5.5 Results and conclusions

5.5.1 Power specifications

The power handling of the speaker was measured as the maximum consumption power playing pink noise at maximum volume without clipping. As the power draw has short transients in time, a set of capacitors was used to maintain it as constant as possible.

The measured power was 18.7 W.

5.5.2 Sound quality

The combination of stereo sound and a flat response over all the frequency spectrum provides a remarkable sound that fully satisfies the initial expectations. The data exposed in the measurements reflects the results of the project in terms of sound quality as it was explained at the beginning of the document.

5.5.3 Battery life

The battery life was tested in the conditions that are expected to be the regular use of the final customer. That means, constant music playback at around 60% max volume.

The implementation of DSP in a µC that lacks of dedicated hardware for such purpose resulted in a high power consumption when the speaker was in idle state. To be exact, the µC and the buck converter alone drain 3.3W constantly.

Even with that, the battery lasted for more than 5h in all the tests. The best case scenario reported more than 8h, and the worst was around 5h, when the music was played at full volume the entire test.

5.5.4 Conclusions

Among all the difficulties related to the development process, the final product meets the initial objectives in all aspects.

Apart from that, it's worth to mention that the means to carry out acoustic quality measurements were good but not optimal. Even though the room was acoustically treated, some of the irregularities seen in the acoustic FR are caused by the room modes itself.

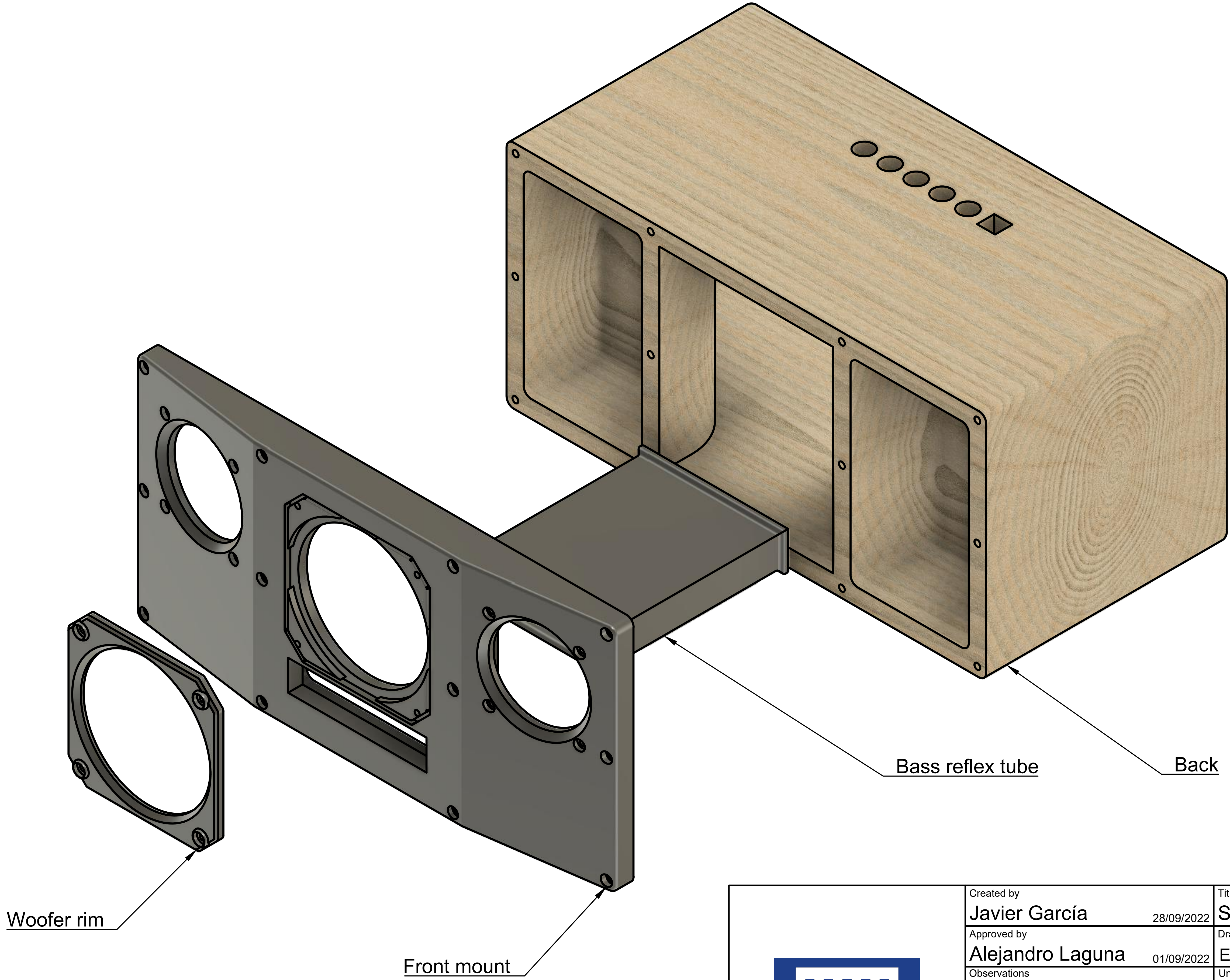
To prevent that, the measurements should have been done in an anechoic chamber capable of absorbing most of the reflections that cause nulls and peaks in the measured FR.

Part II

Plans

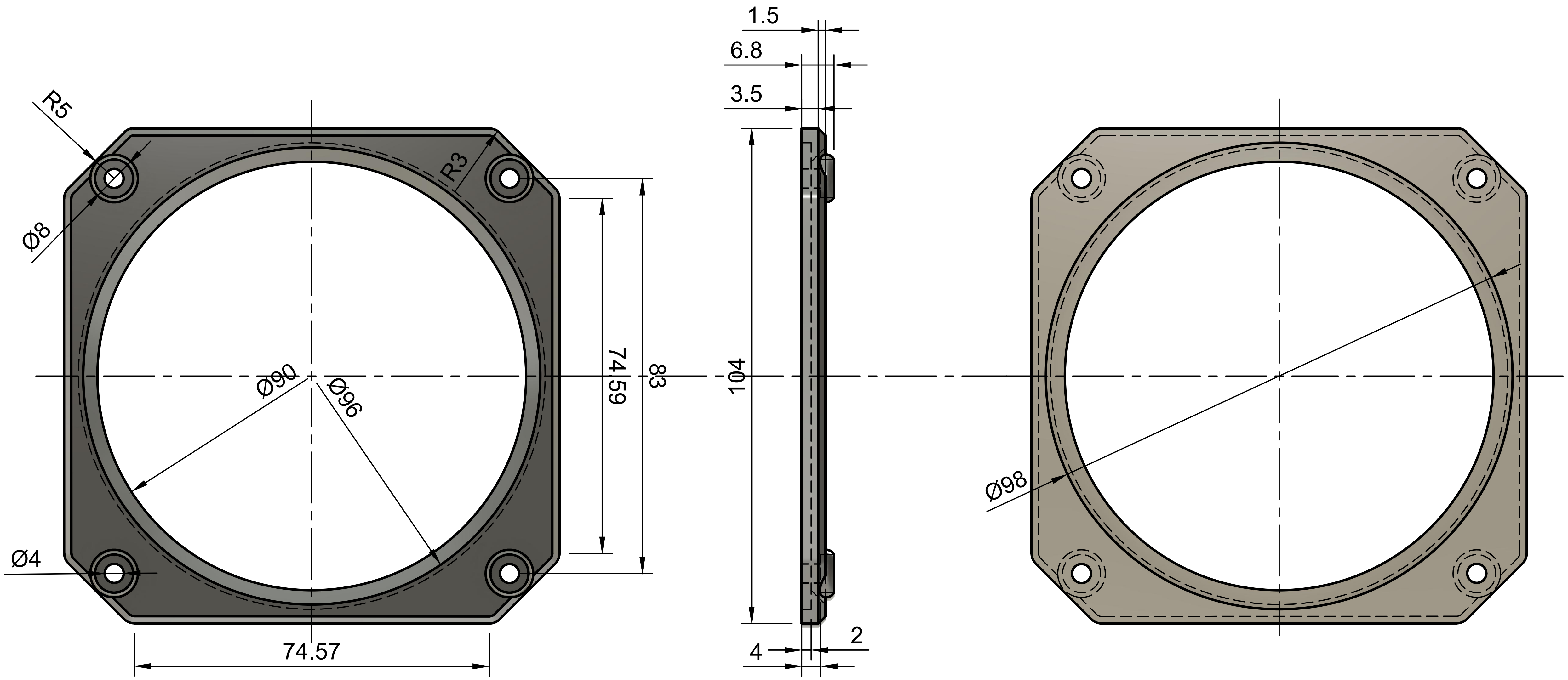
Chapter 6

Plans

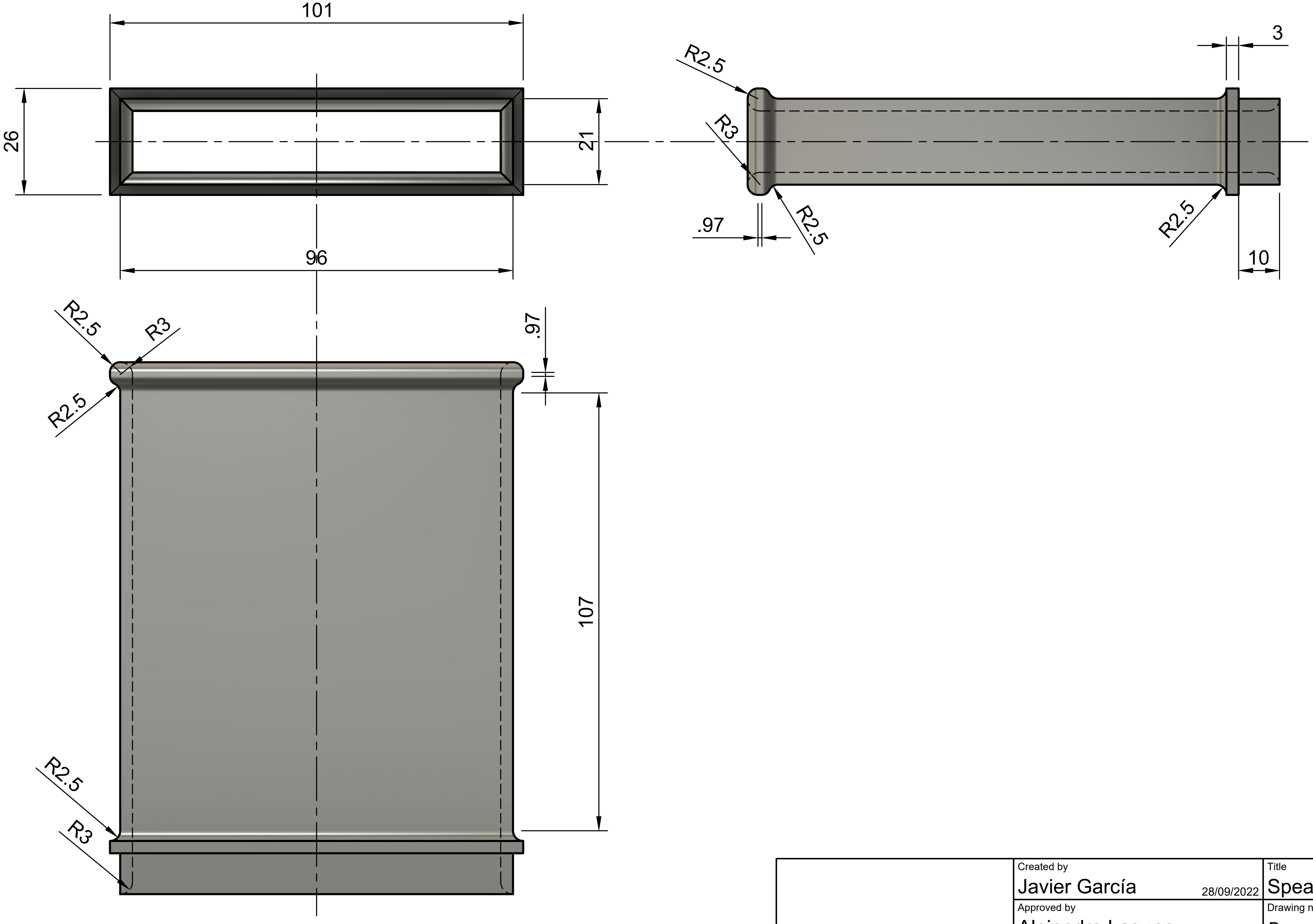


Escuela Técnica Superior de Ingeniería del Diseño

Created by: Javier García 28/09/2022	Title: Speaker enclosure	
Approved by: Alejandro Laguna 01/09/2022	Drawing name: Exploded view	
Observations	Units: mm	Reference number: P01
	Scale: 1:2	Sheet Nº: 1



Escuela Técnica Superior de Ingeniería del Diseño	Created by Javier García 28/09/2022	Title Speaker front
	Approved by Alejandro Laguna 01/09/2022	Drawing name Woofer rim
	Observations	Units: mm — Reference number: P02
		Scale: 1:1 — Sheet Nº: 2



Escuela Técnica Superior de Ingeniería del Diseño

Created by
Javier García 28/09/2022

Approved by
Alejandro Laguna 01/09/2022

Observations

Title
Speaker front

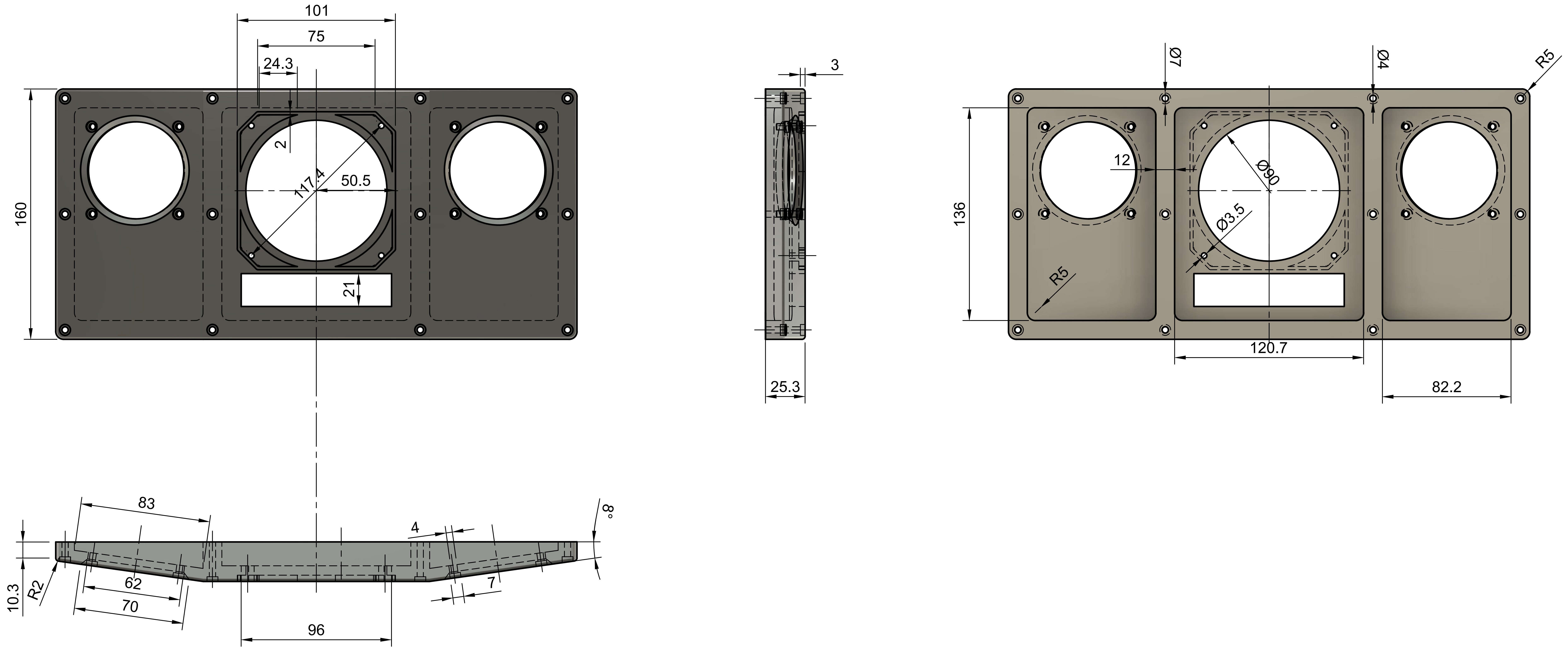
Drawing name
Bass reflex tube

Units
mm

Reference number
P03

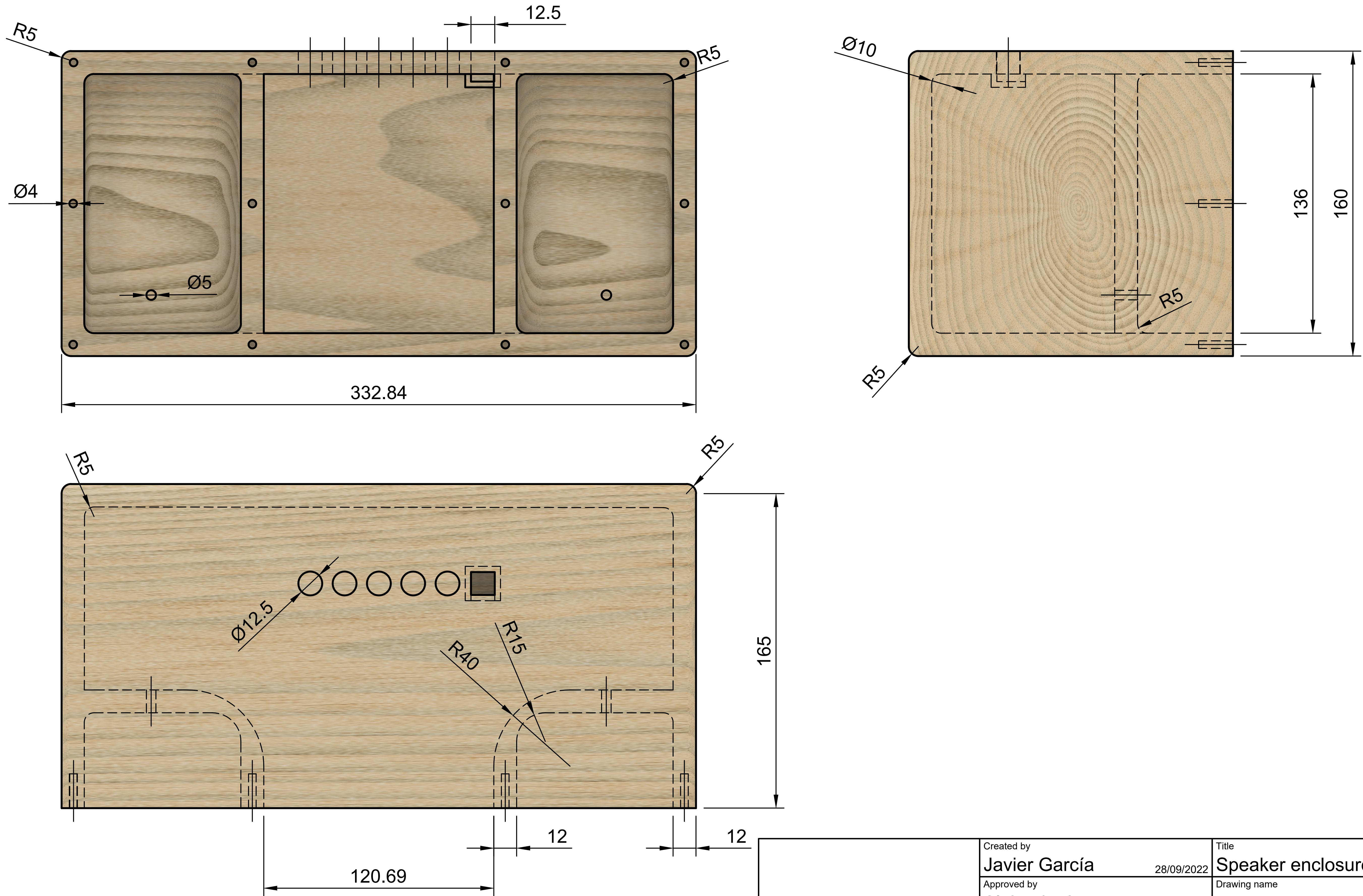
Scale
1:1

Sheet Nº
3



Created by	Javier García 28/08/2022	Title: Speaker enclosure
Approved by	Alejandro Laguna 01/09/2022	Drawing name: Front mount
Observations		Units: mm / Reference Nº: P04
		Scale: 1:2 / Sheet Nº: 4

Escuela Técnica Superior de Ingeniería del Diseño



Escuela Técnica Superior de Ingeniería del Diseño

Created by Javier García 28/09/2022	Title Speaker enclosure	
Approved by Alejandro Laguna 01/09/2022	Drawing name Back	
Observations	Units mm	Reference number P05
	Scale 1:2	Sheet Nº 5

Part III

Budget

Chapter 7

Budget

this chapter summarizes the budget necessary to carry out the development and manufacture of the product.

7.1 Partial Budget

7.1.1 Hardware cost

For the development and manufacturing of this product, a wide amount of items were needed. These pieces of hardware and their costs are shown in the table below.

7.1.1.1 Service costs

The design process required a consumer-grade Windows PC to operate with the software exposed at the beginning of the document. The cost of the computer used was 880€and the estimated lifespan is about 5000h. Power consumption was included in the cost of it. On the other hand, those physical components that make up the speaker's enclosure were manufactured with the assistance of a 3D printer and a CNC laser wood cutter. Therefore, external manufacturing services need to be considered. The Dremel 4250 is a tool that was acquired for this project, and the estimated lifespan is 2000h.

Service	Uds.	Quantity	Unit Price (€)	Total (€)
Consumer-grade computer	h	500	0,20	100,00
3D printing services	h	20	5,00	100,00
Laser CNC wood cutting services	h	1	50,00	50,00
Dremel 4250 complete kit	h	30	0,067	2,00
			Total	152,00

Table 7.1: Machinery and service cost.

7.1.1.2 Hardware components

The list below compiles the physical components that make up the product, as well as their cost and the sum of all.

Component	Uds.	Quantity	Unit Price (€)	Total (€)
Wharfedale 103.78 4" woofer	u	1	40,00	40,00
Cambridge Soundworks tweeter	u	2	20,00	40,00
ESP32 microcontroller	u	1	5,00	5,00
Texas Instruments I2S DAC module	u	2	5,00	10,00
50W mono audio amplifier	u	1	1,69	1,69
15W stereo audio amplifier	u	1	3,40	3,40
Li-Ion 18650 cell	u	9	2,10	18,90
BMS	u	1	5,00	5,00
USB-A female port	u	1	0,86	0,86
16mm copper cable	m	2	2,39	4,78
DIN912 M4 screw	u	1	0,07	1,68
M4 nut	u	12	0,06	0,72
M4 insertion nut	u	12	0,08	0,96
ABS 3D printer filament	kg	1,2	28,00	33,60
			Total	178,41

Table 7.2: Hardware components cost.

7.1.2 Software cost

Most of the software used for designing the product was open-source and/or free to use. Even with that, both Fusion360 from Autodesk and Matlab from Mathworks required a paid license to work. The cost of all the software tools is listed below. It's worth to mention that some of these software licenses were acquired at student price.

Software	cost (€)
MatLab + SysControl	70,00
VS Code	0,00
REW	0,00
WinISD	0,00
ESP-IDF	0,00
Fusion360	0,00
Total	70,00

Table 7.3: Software cost.

7.1.3 Labour cost

The labour cost is formed by the price and amount of time that was invested by the the people involved in the project in order to carry it out. The development was done by Javier García Serrano as student and author of the thesis.

Worker	**Uds.**	**Quantity**	**Unit Price (€)**	**Total (€)**
Mr. Javier	h	700	10,00	7000,00
			Total	7000,00

Table 7.4: Labour cost.

7.2 Total budget

The total cost implies the addition of the partial budgets exposed above. This amount will be increased by various coefficients. A 10% overhead cost shall be considered and applied to the brute cost. Finally, the corresponding taxes (VAT) will be added.

Partial Budget	***Total (€)***
1. Machinery and service cost	152,00
2. Hardware components cost	178,41
3. Software cost	70,00
4. Labour cost	7000,00
Manufacturing cost per unit	**330,41**
Complete development cost	**7400,41**
10% Overhead cost	814,05
Cost before taxes	**8140,45**
21% VAT	1709,5
Total cost	**9.849,94**

Table 7.5: Total cost.

Part IV

Solicitation document

Chapter 8

Solicitation Document

8.1 Scope of the document

This document summarises the technical conditions that the product has to meet once is finished. It also contains information about the manufacturing process, including the technical, facultative and legal conditions of it.

In case of documentary inconsistency the first document overrides this one.

8.2 General regulations

The following laws are applied directly to audio reproduction systems.

- **Directive 2001/95/EC** of the European Parliament and of the Council of 3 December 2001 on general product safety. "DOCE" No. 11 of 15 January 2002, pages 4 to 17 (DOUE-L-2002-80044).
- **Law 37/2003** of 17 November 2003 on noise. State leadership, Official State Gazette 276 of 18/11/2003 (BOE-A-2003-20976).
- **Directive 2003/10/EC** on the minimum health and safety requirements regarding the exposure of workers to the risks arising from physical agents (noise) (Seventeenth individual Directive within the meaning of Article 16(1) of Directive 89/391/EEC).
- **Royal Decree 286/2006**, Of March 10, On The Protection Of The Health And Safety Of Workers From The Risks Related To Exposure To Noise (DOUE-L-2003-80227).
- **Decision 2009/490/EC**: Commission Decision of 23 June 2009 on the safety requirements to be met by European standards for personal music players pursuant to Directive 2001/95/EC of the European Parliament and of the Council

The product shall include the following elements.

- **Business name of the supplier.** May be substituted by the trademark registered in the name of the supplier.
- **Model reference,** that shall meet the information exposed in the commercial invoice.
- **Assembly and/or use instructions if the applied regulations require it.** Shall be written in the official language of the customer's country and provide the required information to ensure the product is used correctly.
- **CE marking.**

8.3 Technical specifications and conditions

The following conditions must be accomplished during the manufacturing and FR correction process.

In addition, the final product must meet a set of specifications that ensure the correct functioning and performance of the speaker.

8.3.1 Manufacturing process

Enclosure Every component that make up the enclosure, regardless of the material or the production process through which it has been built, must meet the plans dimensions with a maximum error of $\pm 2mm$.

Wood parts must be treated with hydrophobic varnish.

All paints used must be adequate for outdoor use.

The fit of the front to the rear must be sufficiently precise so that all screws can be assembled and tightened correctly.

The sealing point between the front part and the back of the enclosure must be isolated with rubber material as butyl or high density foam.

In order to properly insulate the enclosure in terms of acoustics, the compression exerted by the screw towards the parts must be balanced, thus not allowing screws that do not exert sufficient force to compress the layer of insulating material underneath that particular screw.

Electronics

The connections must be safe in terms of electric conductivity and isolation. Once the electronics assembly is done, the continuity must be tested to ensure that there is not any short-circuit or false contact.

The output cables that are connected to the speaker drivers must include the adequate terminals to ensure a safe and easy assembly or disassembly in the future.

The idle power drain must not exceed $200mA$ out from the BMS and semiconductors shall not overheat.

8.3.2 Frequency response correction

All the decisions that concern the filter design process must be based on the actual FR of the speaker. For that purpose, the conditions under which the acoustic measurements are taken must allow the FR not to be significantly altered by external factors such as the transfer function of the room itself.

The signal measured in the DAC terminals needs to be clean and free of clipping, overflow, distortion or noise. For this purpose, the floating point processing power must be taken into consideration, so that the number of filters that can be applied is limited by the number of decimal multiplications that the processor is capable of performing in real time.

8.3.3 product specifications

Frequency response and sound quality

The final FR of the full system in normal operation must meet the following requirements.

- Bandwidth of 75Hz - 16KHz or greater with a criteria of $\pm 3dB$.
- A FR linearity of $\pm 4dB$ within the margins of the specified bandwidth.
- Total Harmonic Distortion of 5% or less over all the operating frequencies.

power output and battery performance

The speaker must be able to drain at least $15W$ of power and must be limited by software to protect the integrity of the drivers and their coils.

The real capacity of the battery pack must exceed $60Wh$ with a voltage range of $[3.2, 4.2]\,V$ and the maximum current has to be greater than $3A$.

Bibliography

[1] Vance Dickason. *The Loudspeaker Design Cookbook*. Google-Books-ID: _K3SAAAACAAJ. Audio Amateur Press, Dec. 2007. 275 pp. ISBN: 978-1-882580-47-7 (cit. on pp. 23, 27).

[2] *Thiele/Small parameters*. In: *Wikipedia*. Page Version ID: 1100463633. July 26, 2022 (cit. on p. 24).

[3] Knud Thorborg and Claus Futtrup. "Electrodynamic Transducer Model Incorporating Semi-Inductance and Means for Shorting AC Magnetization". In: *J. Audio Eng. Soc* 59.9 (2011), pp. 612–627 (cit. on p. 24).

[4] Morten H. Knudsen and J. Grue Jensen. "Low-Frequency Loudspeaker Models That Include Suspension Creep". In: *Journal of the Audio Engineering Society* 41.1 (Feb. 1, 1993). Publisher: Audio Engineering Society, pp. 3–18 (cit. on p. 24).

[5] *Bluetooth A2DP API - ESP32 - — ESP-IDF Programming Guide latest documentation*. URL: `https://docs.espressif.com/projects/esp-idf/en/latest/esp32/api-reference/bluetooth/esp_a2dp.html` (visited on 09/10/2022) (cit. on p. 34).

[6] *Acoustics/Fundamentals of Psychoacoustics - Wikibooks, open books for an open world*. URL: `https://en.wikibooks.org/wiki/Acoustics/Fundamentals_of_Psychoacoustics` (visited on 09/11/2022) (cit. on p. 36).

[7] Ethan Winer. *The Audio Expert: Everything You Need to Know About Audio*. Google-Books-ID: 6FoPEAAAQBAJ. Routledge, Dec. 15, 2017. 809 pp. ISBN: 978-1-351-84007-1 (cit. on p. 39).