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Automatic Calibration and Equalization of a Line Array System

School of Electrical Engineering

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ABSTRACT OF THE
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This final project presents an automated Public Address processing unit, using delay and magnitude frequency response adjustment. The aim is to achieve a flat frequency response and delay adjustment between different physically-placed speakers at the measuring point, which is nowadays usually made manually by the sound technician. The adjustment is obtained using four signal processing operations to the audio signal: time delay adjustment, crossover filtering, gain adjustment, and graphic equalization. The automation is in the calculation of different parameter sets: estimation of the time delay, the selection of a suitable crossover frequency, and calculation of the gains for a third-octave graphic equalizer. These automatic methods reduce time and effort in the calibration of line-array PA systems, since only three sine sweeps must be played though the sound system. For verifying the functioning of the system, both simulated signals and measurements have been conducted. A 1:10 scale model of a line array system has been designed and constructed in an anechoic chamber to test the automatic calibration and equalization methods and the results are analyzed.

Keywords: Live Sound, Line Array, Public Address, Automation, Loudspeaker measurement, Group delay, Graphic Equalization, Crossover Filtering

Preface

This final project has been realized in the Acoustics and Audio Signal Processing Department at Aalto University School of Electrical Engineering during the year 2015.

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Abbreviations

A/D	Analog to digital
D/A	Digital to analog
DSP	Digital Signal Processing
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
FOH	Front of House
HF	High frequency
HPF	Highpass filter
IIR	Infinite Impulse Response
LF	Low frequency
LPF	Lowpass filter
PA	Public Address
SPL	Sound Pressure Level

1 Introduction

1.1 Motivation

During the last decade, the music industry business model has shifted from record releasing to promote live performances, raising the number and quality of concerts. This situation has had also strong repercussion on the technical part of the live audio industry. Sound systems have become more complex, since the digitalization and computer control has invaded the mixing and processing part, with advanced digital mixing desks and system processors. Also acoustically many advances have made loudspeaker systems more efficient, with less distortion and with possibilities of controlling accurately the response in different target areas. In this sense, the control of those loudspeaker systems has also become more complex.

The vast majority of concert Public Address (PA) systems used nowadays consist of hanging line arrays. These speakers are used to reproduce the middle and high frequencies, usually above 100 Hz to 150 Hz. For the low frequency coverage different subwoofer configurations are used, which are usually placed on the floor in front of the stage.

The main problem with traditional loudspeaker systems was to achieve an equal coverage though the whole audience area with low distortion and in a equal way for the whole frequency range. Line arrays have improved tremendously the coverage for mid and high frequencies, by focusing the energy on a narrow vertical angle which is then pointed precisely with different aiming software tools.

However this higher complexity in the sound systems require also more complex signal processing techniques in order to achieve a flat and coherent response through the whole audio band. This different processing techniques are nowadays very differentiated when it comes to small and medium sized venues and in large venues. The small and medium sector of sound reinforcement is still making the adjustment operations usually manually.

In this project an automated PA processing unit, adapted for line arrays systems, is presented. The aim of this is to obtain the different parameter sets for adjusting the sound system and then apply those parameters to the audio signal processing chain. The calibration consists of four operations: apply a crossover filter to split the audio band between the subwoofers and the line array, adjust different time delays for the different loudspeakers, apply a certain gain to each of the loudspeakers, and equalize the complete system to achieve a flat magnitude frequency response.

Nowadays there are several automated systems, though they are usually bundled to a specific brand with pre-loaded speaker data, such as Meyer Sound's Galileo [1], or need additional tools to integrate it with the system processor. Moreover this systems are usually designed for large sized venues and are not affordable for integrate them with different manufacturers sound systems. The consequence of this is that most of the small and mid sized line array systems are still adjusted manually using PA processing units in addition to graphic equalizers. The manual procedure requires to play different excitation signals, usually pseudorandom pink-spectrum noise through the PA, which can take several minutes, as the sound technician observes the adjustment changes at the time he is manipulating the processing unit. This procedure is clearly disturbing and time consuming, for this reason it has to be adjusted and configured hours prior to the venue, as it couldn't be done with the audience in the venue area. The sound engineer adjusts the crossover frequency and filter type, the delay for the different loudspeakers and the graphic equalizer supported by a spectrum and phase analyzer. Those analyzers use two channels, one for the measured signal and the other for the reference loop-back, some commercial examples which are used nowadays are EASERA [2] and Smaart [3].

In this project an alternative integrated analyzer and parameter-calculation system is included in the system processor, allowing to automatically perform the adjustment of those four operations playing three sweep signals through the PA. This avoids all the manual procedure, and it can be done even with audience, being much more time and effort efficient.

In order to evaluate the system, a 1:10 factor scale model was tested in a anechoic chamber once the software has been tested with simulated signals and system responses.

1.2 Scope

This project outlines the design, testing and model scale implementation of the automated PA processor for line arrays. The different signal processing blocks and operations are described and implemented with Matlab. The testing is done first with emulated responses, created with Matlab functions and then with real measurements on the scale model. The benefits of this new system are highlighted and real applications are proposed.

The system has some limitations that were defined at the design phase due to time constraints. Compared to real application scenario, where at least stereo systems should be taken into account, in this project one channel system has been considered. Expansion to multichannel functioning should not be a problem, by adding the same measurement procedures independently to the other channel. Also when implementing the system, just two loudspeaker clusters have been taken into account, meaning that there is a low frequency driver acting as subwoofer and a mid and high frequency driver acting as one line array loudspeaker, which compared to real situations is a good approximation, as at a precise measuring point the influence of a cluster of subwoofers, if adjusted correctly, should have the same response, as well as the line array element.

Thus the work on this project has consisted in three main tasks:

1. Design of the signal processing operations and its order, defining the parameter extraction blocks, processing blocks and the audio chain.

2. Implementation of functions to simulate the responses of the loudspeaker system in order to test the designed processing blocks.

3. Definition, design and construction of the scale model in an anechoic chamber and perform measurements for testing the system and obtaining the results.

Despite at the moment of writing this project, the system has not been tested in a real concert PA system, the aim is to fulfill some tests during the next weeks using a *Das Audio Aero28* three-way line array.

1.3 Structure of the project

This project has been structured into the following chapters that are outlined below:

Chapter 2 gives a brief introduction to theoretical concepts that will be used for the explanation of the procedures used to perform the automated processor. Also some introduction to the live audio history and practical matters is given.

Chapter 3 presents the automated line array processor and its basic characteristics. It gives an overview to the different blocks that compose the signal processing path and the signal treatment to extract automatically the parameters.

Chapter 4 presents the different procedures that have been taken place for testing the functioning of the system, stressing on the description of the design and implementation of the scale model.

Chapter 5 explains in more detail how the different processing blocks have been implemented, using as reference signals the ones measured from the scale model.

Chapter 6 summarizes briefly the results and gives some future work proposals and applications.

2 Background

In this chapter a theoretical introduction to repeatedly used concepts and therms is given. However some basic knowledge about basic signal processing is supposed by the author. The first part deals with audio signal processing, stressing on theoretical concepts, filtering techniques used in the proposed system and audio signal characteristics. A brief introduction and state of art in live sound systems and its history will be made. Finally the chapter explains some theoretical concepts about line array systems, its application to real live sound loudspeakers and the configurations used in different venues.

2.1 Audio signal processing

2.1.1 Spectrum and dynamic range

The analog audio signal is an electric signal whose voltage or current waveform is proportional to the acoustic sound pressure. Sound waves have several properties that can affect the way humans perceive the sound, the ones that have more influence in our perception are: frequency, wavelength, amplitude, sound pressure, sound intensity and direction, and combination of these, as harmonic components

As the final destination of this signal processing is human hearing, the characteristics of the audio signals should then fulfill its requirements in frequency and amplitude range. It is considered that the human hearing goes from 20 Hz to 20 kHz in the frequency range, despite not the whole range is always audible. Very low frequencies are noticed as vibrations and with age the higher end of this range is lost, so for a average adult the limit is between 15 and 18 kHz. Regarding the amplitude, the range is measured in dB SPL, where 0 dB SPL equals a sound pressure of 20 μ pascals. The hearing range is between 20 dB and the pain limit, which is considered 120 dB SPL. However, sound systems usually reproduce higher SPL, as they are meant to cover long areas and distances.

The audio signal treated in this project is a digital waveform signal, meaning that it follows the characteristic of the analog audio signal, both in spectrum and in amplitude. Depending on the precision in which this signal is digitalized, the signal is a closer approach to the analog signal and the conversion loss is lower. In following chapters, the frequency and amplitude range is considered to be large enough in any point of the signal processing chain to not generate distortion due to exceeding the amplitude or frequency range.

The frequency band used in professional audio systems, and known as high fidelity systems, is considered the same as the audible band, from 20 Hz to 20 kHz. This band should be reproduced by the complete loudspeaker system. Thus, the sampling frequency has to be at least double of the maximum representable frequency according to Nyquist's law [4]. Professional standard sampling frequency is 48 kHz, but in the last years more and more audio systems are shifting to 96 kHz, as processors are capable of deal with large data streams even in real time. Higher sampling frequencies allow also to improve the response in the high end of the audible band, as aliasing filters can be smoother and in higher frequencies.

The number of bits used for each sample (sample resolution) limits the dynamic range of the audio signal. Large PA systems have huge dynamic ranges, as loudspeakers are able to reproduce easily up to 140 dB SPL. Usual acquisition is made at 24 bits per sample, however when operations are made with those signals a higher resolution is needed, usually up to 40 bits per sample. The signals used for this system are treated in Matlab, the data is represented by 64 bit double-precision floating-point numbers.

2.1.2 Impulse response measurements

The impulse response is the basic function which defines the reaction of a dynamic system to an external impulse. The reaction is function of time, thus it contains all frequencies in it.

Loudspeaker characteristics are usually obtained from its impulse response by different procedures. The most logical way to obtain the impulse response from a loudspeaker would be sending an impulse signal, but the need to limit input amplitude to maintain the linearity of the system makes it a non optimal system. Thus, different excitation signals have been used to derive from them the impulse response. The energy is spread in time-domain, avoiding the non desired energy peak. The two most used are pseudo-random maximum length sequences [5] and sinusoidal sweeps.

MLS measurements for audio systems are nowadays used for manual adjustment of loudspeaker systems. This method uses cross correlation between the input and output in order to recover a periodic impulse response of the system which increases its accuracy step-wise. In practice this means that to obtain the response of a loudspeaker or loudspeaker system, it is require to play a highly disturbing full band noise during several seconds.

Sine sweep measurements are based on time-domain de-convolution between input and output of the system. Furthermore, if the frequency distribution is logarithmic, several improvements can be achieved [6], the most relevant for our concerning is the ability to measure non-linear systems, as the linear response can be separated from the distortion.

After applying the de-convolution to the measured response, the signals produced by the nonlinear terms in are located at different places along the impulse response, thus the linear contribution to the response, which is proportional to the input signal can be separated from the other nonlinear terms. Due to the frequency response of the excitation signal increases exponentially, the lower frequencies expand on the measurement period. This has to be taken into account when designing the inverse filter, as it has to compensate the increased energy. As the average energy at lower frequencies is higher, the signal to noise ratio is also improved at low frequencies.

2.1.3 Frequency response

The frequency response is the measure of the spectrum in magnitude and phase at the output of a system given a known input. The frequency response can be measured directly with the same procedures as listed for the impulse response (impulse measurement, sweep signals or MLS sequences). The way to obtain the frequency response from a system is by calculating the Fourier transform of the impulse response.

The obtained signal has complex values, and the most direct way to represent it is with a polar diagram. However, as in signal processing many times phase and magnitude values want to be analyzed separately, the representation of the frequency response for audio systems is usually performed breaking down the phase (1) and the magnitude responses (2):

$$\phi(f) = \arg(X(f)), \tag{1}$$

$$|X(f)|, \tag{2}$$

where $X(f)$ is the complex value for the Fourier transform. The magnitude plot is usually measured in dB, and defined by the Equation 3 and the phase response in radiant or degrees, both versus the frequency in the horizontal axis. It is also typical to have a logarithmic frequency axis, as the human ear perceives has a logarithmic frequency and amplitude response.

$$G[dB] = 10\log_{10}(\frac{V}{V_0}), \tag{3}$$

where V is the voltage (or level if digital) V_0 is the reference voltage.

2.1.4 Phase and group delay

The phase delay and group delay are two alternative ways of extracting information of the phase response of a system. The phase delay shows the time shifting for each sinusoidal component at the output compared to the input:

$$\tau_\phi(f) = \frac{-\phi(f)}{f} \tag{4}$$

The group delay is the measure of the time delay of the amplitude envelope for a sinusoid, as it defines how much the envelope curve of a complex signal is delayed. The group delay is a very useful tool to measure distortion of a system. If the response of a system is non linear, the different frequency ranges have different delays. Group delay is measured from the phase response:

$$\phi(f) = \frac{-d\phi(f)}{df} \tag{5}$$

Distortion, and therefore large values of group delay, is a very noticeable effect on human hearing. According to psychoacoustic analysis [7, 8], group delay values should remain below the limits shown in Table 1 to be noticeable. This values have been measured with headphones or in low reverberation rooms, which makes the perception to be stronger, thus we could consider them as lower limits. This values are then a design constraint for the system, and have been measured at the end of the processing chain.

Table 1: *Audible group delay limits for octave frequency bands.*

Frequency (Hz)	Threshold (ms)
500	3.2
1k	2
2k	1
4k	1.5
8k	2

2.2 Crossover filters

Crossover filters are electric filters that divide the audio spectrum in different bands in order to use specific types of loudspeaker drivers and improve its performance. Ideally this filters split the band with a lowpass filter for the woofer and a symmetric highpass filter for the tweeter which reproduces the higher frequencies. A pass-band (or crossover band) has to be considered where both components share the reproduction of the a frequency range. Crossover filters can be active or passive depending on the need of external power and the point on the processing chain where it is placed. Also digital crossover filters are used.

Moreover, depending on the transfer function, three main types of designs have been used traditionally:

Butterworth: this filter type has a very flat frequency response and a reasonably linear phase in the passband. For this reason they are appreciated in audio applications. The magnitude transfer function for the lowpass filter follows the expression:

$$|H(f)|^2 = \frac{G_0^2}{1 + (\frac{f}{f_c})^{2n}}, \tag{6}$$

where f_c is the cutoff frequency and n is the order of the filter, thus for a first order filter an attenuation of 6 dB per octave is reached.

Linkwitz-Riley: this filters designed by engineers Siegfried Linkwitz and Russ Riley [9] take the advantage of the flatness of Butterworth filters and reduce the overlapping in the passband cascading two Butterworth second order filters. This filters present uniform magnitude response and in the crossover frequency the signals are attenuated 6 dB, which implies that their added magnitude is unitary. In addition, the phase difference in both ways is zero, which prevents distortions in the crossover band.

FIR: when working with digital crossover filters FIR filters can be used. The main advantage on using FIR filters is that they get a linear phase response when the impulse response is symmetric. As it is in digital domain, the Z-transform is used to represent the transfer function:

$$H(z) = \sum_{n=N1}^{N2} h(n)z^{-n}, \tag{7}$$

where N2 - N1 is the length of the impulse response and thus the order of the FIR filter.

Passive crossover filters were the first implemented crossovers, used in loudspeaker enclosures to divide the signal to the different drivers. Still nowadays are the most common solution in non professional audio equipment. The response of this filters is usually a first or second order type Butterworth. Passive filters are implemented with resistors, capacitors and inductors. This filters are easy to implement, but the main inconvenient is its waste of power, as a part of the energy is transformed into heat. Moreover the use of inductors and capacitors produce non linearity in the responses, provoking distortion and phase shifting and decreasing the quality of the reproduced sound.

Active crossover filters treat the signal before the power amplification process. This filters are based in operational amplifiers (OPAMPs) and allow to modify parameters dynamically. This units also allow usually to use different filter types, but the most common are Linkwitz-Riley responses from order 2 to 4 (cascading two second order Butterworth filters). Another advantage is the ability to apply gains and change the characteristic of the filter slope and cutoff frequency. This requires also the use of more power amplifiers as every way of the system has to be amplified independently.

Digital crossover filters are based on programmable digital signal processors. The input is A/D converted and filtered with DSP techniques to then convert it again to analog and driven to the power amplifiers. There are two types of digital crossover filters: digital simulation of analog filters or FIR filters.

Analog simulations are based on IIR filters and can affect the phase of the signal as the analog do. The analog designs as Butterworth or more common Linkwitz-Riley can be implemented with digital filters. In case of FIR filters, adapted and linear phase responses can be reached using high order filters. The main problem in this designs is the ringing that appears in the off-axis response [10].

Even so, many advantages make that digital crossover filters are becoming an industry standard:

- Very steep cutoff slopes can be easily reached. FIR filters require a higher order than IIR filters to achieve them, but nowadays as computational capabilities are increasing, and the latency issues in using long FIR filters in real time applications are becoming a problem from the past.

- A better control on the phase and frequency response is possible, adjusting the filters to different loudspeaker drivers.

- Analog components inaccuracies as inter-modulation problems and stability of the components (as temperature dependence) are not a problem any more with digital filtering.

- As the signal is treated in digital form, the possibilities for additional signal processing procedures is much easier. Commercial digital crossover units can have flexible number of inputs and outputs and routing options.

- Delay lines are usually implemented in these units, which allows to align different loudspeaker clusters or drivers in a loudspeaker.

2.3 Graphic equalization

Graphic equalization is one of the most used procedures in the audio industry, mainly in live audio applications, and the graphic equalizer has become a standard tool for tuning PA systems.

Equalization is a process that aims to alter the magnitude frequency response of an audio system, thus they could be considered as a frequency-specific volume control. Due to the condition of using filters that are not always linear, the equalization process affect magnitude and phase response, but those are designed in a way where the affection on the phase response is limited and controlled. Two basic types of equalizers can be discerned: graphic and parametric equalizers.

Parametric equalizers allow to modify the frequency response varying multiple parameters, commonly the bandwidth, central frequency and gain. This type of equalization is commonly used in mixing consoles, for shaping the frequency response of each channel independently, as it is able to cut or boost very precisely a specific frequency.

A graphic equalizer allows to control individually a certain fixed number of frequency bands. Moreover, this frequency bands are equally spaced on a logarithmic scale, thus following the human response. They are implemented using a bank of filters and selective amplifiers that are chained. Each of this filters has an individual slider-control that defines a boost or cut in its band. This sliders are placed side by side in ascending frequency band order, it is easy to identify graphically the overall frequency response curve.

The bandwidth of this filters is fixed, and it defines the number of needed bands to cover the audible spectrum. Octave band graphic equalizers are used for middle and high end stereophonic sound systems for consumer use. In professional applications the most common graphic equalizers present a bandwidth of 1/3 of octave. This means that the central frequencies of the filters are space 1/3 of octave between them. In order to cover the whole audible spectrum 31 bands are needed. This equalizers are used in live audio basically for three reasons: correct the response in hall acoustics, eliminate feedback frequencies (usually used in monitor tuning) and correct the response of loudspeaker system. This last application is the one which will be used in the system presented in this project.

Nowadays the digital implementation of graphic equalizer is widespread. Plenty of different architectures have been studied to implement the different filter banks.

2.4 Live sound systems

Live music industry has evolved a lot during the last decades, mostly from the 1960s on, when music bands started to perform in front of big audiences in theaters or open air venues. At the same time the musical tendencies, like Rock music, were heading towards the use of electric-based instruments, such as electric guitars, bass guitars, electronic keyboards or organs.

As the concerts were becoming bigger and the audience more noisy, the use of sound amplification to increase the volume of the band and reach the whole audience was needed, and the first sound systems appeared. At the beginning the solutions were very primitive, consisting in separate sound systems for each instrument. These independent systems consisted in low complexity combo amplifiers, usually with outputs between 50 and 100 watts. The microphones for the singers or acoustic instruments were connected to similar combo amplifiers similar to the guitar ones. Those instrument and microphone amplifiers were scattered around the stage, and oriented to the audience without any acoustical criteria. Bands like The Beatles toured with this type of scattered systems during the 1960s, and in many occasions the band was not heard at all by the audience. This situations drove to increase the necessities in power and quality in sound systems.

As no loudspeaker driver is able to reproduce the wide audible frequencies spectrum with the desired fidelity multi-way systems started it development for live audio applications. This requires the use of audio crossover filters. In early 1970s the first multi-way systems appear in the live sound scene derived from the movie pictures loudspeakers. The systems used passive crossover networks. The loudspeaker boxes used a two way configuration, based on a woofer and a horn-coupled high frequency driver. This loudspeakers were just stacked on each side of the stage without any specific acoustic criteria.

The use of more powerful loudspeaker led to a transition where the concept of PA appeared for the first time. The model changed from having every instrument connected to its own amplifier and loudspeaker to have both instruments and voice in one sound system. This also led to the appearance of the first mixing desks and the role of the sound engineer started to have some importance. However the result was not that great, as still the systems delivered a poor quality sound, lack of fidelity and realism, in part due to they were driven at full power and the lack of dynamic expression and the huge distortion were very present. At this point also stage monitoring for the musicians started to become a relevant point in the organization of a concert, as the increased sound level produced by the PA system did not allow the musicians to hear themselves or the other instruments.

By the end of the 1970s live performances occupied an important position in the music industry, and the live audio industry also developed with it. Sound engineers became more trained and acquired engineering criteria to improve the results of their job. Sound system manufacturers hired acoustic and electric engineers and more technical criteria started to be applied for specific live sound systems designs.

During the 1980s many multi-way loudspeakers appear, specifically designed for powerful live applications. Electronics also evolved during this period, and the first three and four-way loudspeakers appeared as active crossover networks started to be used. This drove also to a better quality in the amplification process, as specific amplifiers were used for each driver in the cabinet. The better structural properties of stages led to a new trend on the placement of the loudspeakers, hanging them in large clusters, as shown in Figure 1b. The divers used to be horn-loaded (meaning that the air mass is loading on the tip of a horn) and frontal radiating.

During mid 1990s also the digital crossover units started to appear for controlling loudspeaker systems. As additional features as delay adjustments were offered, this also led to a higher accuracy in the alignment of elements and phase coherence started to be taken seriously. This coincide with the appearance of the first line array systems, where alignment became a must.

One of the most well known systems used in the major tours during the end 1980s and 1990s was Clair Brothers S4, which represents very well this type of PAs. As seen in Figure 1a it consisted on a four way cabinet, with two 18-inch drivers for low frequencies, four 10-inch drivers for mid-low frequencies, two 2-inch drivers coupled to a horn for the mid-high frequencies and two 1-inch aluminum ribbon voice coil coupled to horns for the very high frequencies.

(a) Clair Brothers S4 cabinet.

(b) Hanged Clair Brothers S4 cabinets in a concert.

2.5 Line arrays

2.5.1 History

In the mid 1990s the French company L-Acoustics applied the well known technique of the line source effect to the first commercial concert sound line array, defining a new industry standard that goes on until these days and almost buried the traditional

The line source effect was studied by the acoustical engineer Harry F. Olson in the early 50s, in his book *Elements of Acoustical Engineering* [11]. These concepts were used to develop a column speaker in which vertically aligned drivers were used to reproduce mid range frequencies. However this systems were used in auditoriums or churches in order to achieve narrow vertical coverage and avoid to radiate sound to reflecting areas. The use was restricted to low fidelity voice signals.

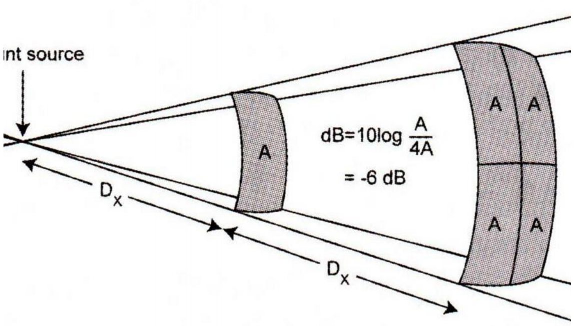
In 1992, French physicists Christian Heil and Marcel Urban presented a paper [12] in AES conference proposing to use the principles of line arrays studied by Olson for concert sound systems. Later the therm Wavefront Sculpture Technology (WST) [1] was introduced for referring to the criteria used for analyze and design continuous line sources. In 1994 the first commercial line array was released, the V-DOSC from L-Acoustics, Heil's founded company.

From this moment on, the majority of loudspeaker manufacturers have centered its efforts in the improvement of line array loudspeakers, and it has become an industry standard for any proper concert.

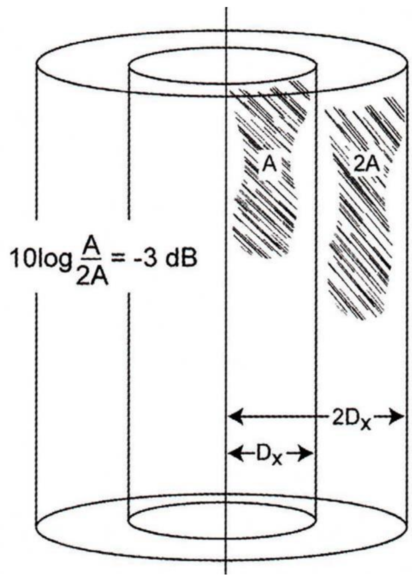
2.5.2 Line source effect

The line source effect is based on the inverse square law, which states that the sound intensity will decrease with distance in absence of reflection or reverberation following strictly geometrical considerations. Thus, with a point source in free space, spherical propagation is achieved, meaning that every time that the distance from the source is doubled the coverage area i multiplied by four as seen in Figure 2a. With a line source the propagation of waves is cylindrical, meaning that for every doubling in the distance the area covered is just twice as seen in Figure 2b.

If no propagation losses are considered, the relative intensity of the sound waves doubling the distance can be determined as -6 dB in point sources and -3 dB in line sources.



(a) Point source propagation.



(b) Line source propagation.

[1]Wavefront Sculpture Technology and WST are trademarks of L-ACOUSTICS

Point sources and one dimensional line sources can not exist in reality, but the calculations can be used as a model. The line source effect is limited with distance as the dimensions of the line source become less significant with distance, thus past a certain point it acts as a point source as its length becomes insignificant.

The line source effect has also frequency limitations, and looses its effectiveness at both high and low frequency end of the audible spectrum [13]:

- The length of the line array determines the longest wavelength (thus lowest frequency) where the directivity is controllable, and has the a line source behavior, following the Equation:

$$\mathrm{r} = \frac{\mathrm{L}^2\mathrm{f}}{700}, \tag{8}$$

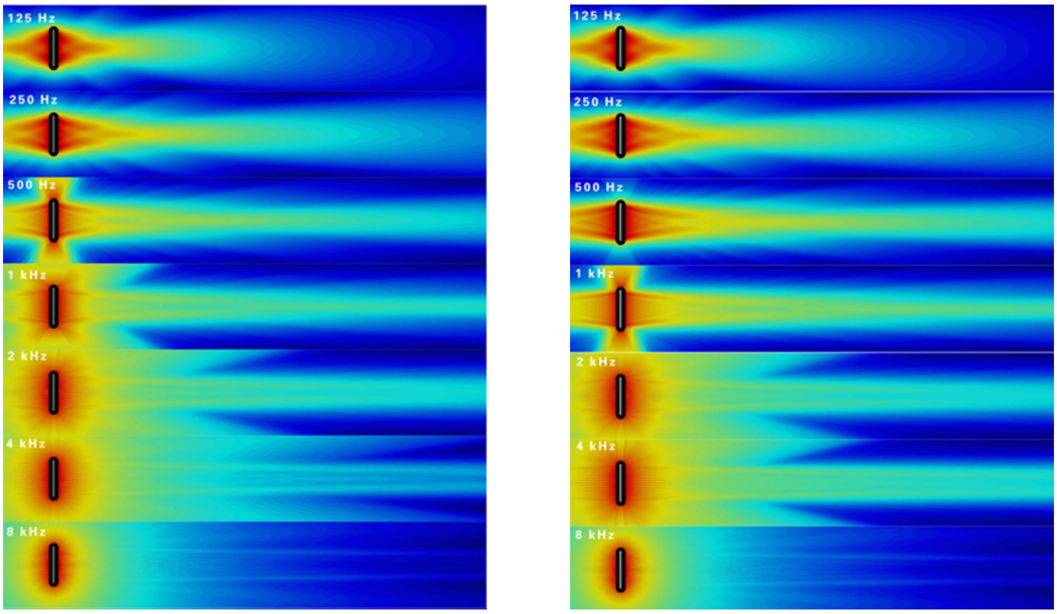
  where r is the distance where the line source effect is valid, L is length of the line source. Thus, the longer the array, the lower frequency that it will be able to control given a certain distance.

- In practice a line source is constructed with several aligned radiating elements. The wavefronts of those may no be plane, but have a certain curvature. The interference between those elements defines a minimum wavelength (thus highest frequency) for which different aligned radiating elements act as line source, and it is determined by the Equation:

$$\mathrm{d} < \frac{\lambda}{2}, \tag{9}$$

  where d is the distance between the radiating elements. Thus more and smaller sources are needed to space them closer in order to maintain the effect in the higher frequency range.

This limitations are shown in Figures 3a and 3b. The length of both arrays is 8 meters, thus at low frequencies no difference is appreciated. In Figure 3a the radiating elements present a separation of 0.5 meters, thus the source effects starts loosing its efficiency at about 400 Hz. In Figure 3b as the elements are twice as close, the sound field intensity is greater if observing the band from 500 Hz to 4 kHz.



(a) 16 omni-directional sources. (b) 32 omni-directional sources.

Figure 3: *Directional behavior of an 8 meter array per bands* [2].

Vertical interference between the different drivers is also a problematic aspect of line arrays, as a dispersion pattern appears due to the relative phase difference at the listening position, strongly noticed in the high frequency area.

However, lots of improvements using DSP techniques [14, 15], basically optimization of the responses using FIR filtering have been done the last years to control the response and directivity of line arrays. Also Wave Field Synthesis is used lastly to analyze and improve this systems [16].

2.5.3 Line arrays in live sound

The aim of using line arrays in live sound applications is that the whole audience has a equal coverage in the whole frequency band. The radiating elements in the array are designed to have a wide horizontal coverage (typically between 85 and 110 degrees) but very narrow vertical angle. However this is just possible for mid and high frequencies, as lower frequencies become omni-directional.

In practice, line arrays consist on several trapezoidal-shaped loudspeaker cabinets connected between them by specialized rigging tools as shown in Figure 4. The clusters are hanged at each side of the stage from the truss structures at a certain height, depending on the size of the venue, the coverage area and the size of the array it can vary between 5 and 15 meters. The rigging tools allow the adjustment of the relative angle between each pair of elements in order to be able to achieve a specific curvature depending on calculated parameters that change depending on the desired coverage.

[2]Picture copyright by Meyer sound.

Figure 4: *Meyer sound's Leopard line array* [3].

Line array elements can vary depending on the application and venue type. Large format line arrays are designed to cover long distances in festivals or open air concerts, and smaller versions are used for indoor auditoriums or concert halls, small sized venues or as reinforcement to cover smaller areas that the main line array does not. Typically manufacturers have different product families with large, medium and small format systems. These loudspeakers can have different driver configurations in order to cover its functioning band depending on the demanded SPL. Usually for large sized arrays three way design is implemented, with two 12 to 15 inches drivers are used for low frequencies in addition to 6 to 10 inches mid frequency drivers and multiple compression drivers. Smaller systems usually use a two way configuration mounting driver one driver up to 12 inches and two ore more compression drivers.

One of the most critical design points are the high frequency components. As explained in Chapter 2.5.2, for high frequencies it becomes difficult to maintain the line source effect, as the drivers have to be very close. For this reason the high frequency drivers are designed to work as point sources in a narrow vertical angle. The aim is to achieve a very narrow vertical dispersion (usually around 5 degrees) and wide horizontal dispersion, for this reason wave guides are used to couple the compression drivers. The narrow vertical dispersion also assures that the dispersion pattern is low due to point source interference. In three way array elements the mid

[3]Picture copyright by Meyer sound.

frequency areas is covered with multiple smaller drivers, to maintain the line source effect in the mid-range.

When considering the low end of the frequency band, as shown in Chapter 2.5.2 line arrays loose his line source and directivity properties for low frequencies. Moreover low frequency speakers use large drivers (usually between 18 and 21 inches), which makes them heavy. For this reason, the low end of the frequency band is covered with subwoofers placed on the ground in different configurations. Those cover a frequency range from the 20 Hz to between 100 or 200 Hz depending on the system.

The fact of using different loudspeakers which are physically at different positions, provoke that the arrival of the wavefronts to a certain listening position is not the same. Different arrival times of the wavefront from the subwoofers and the line array speakers, provoke phase shading, strongly noticed in the crossover band [17]. Figure 5 shows a two-dimensional diagram with approximate values of the relative delay problem in live sound applications. Thus, this requires that when line arrays are used, a delay adjustment procedure has to be made in order to virtually align the different speaker clusters.

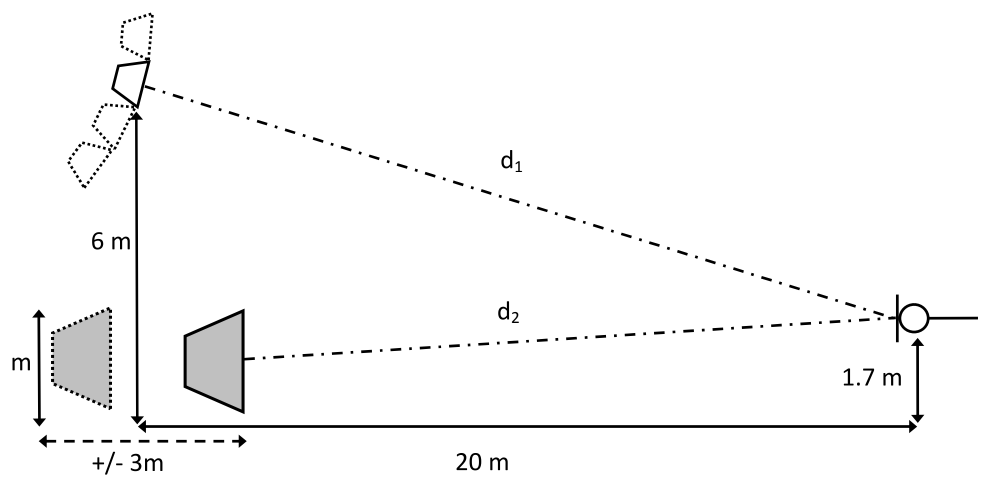


Figure 5: *Relative distances to measuring point.*

In practice, even specially designed loudspeakers are not perfectly continuous line sources. The gaps between the different driver elements and the cabinets themselves are non radiating portions of the total line source. Thus, the linear source effect is lost both in the low and the high end of the frequency band, and real line array designs cannot be described with the pure line source effect theory.

2.5.4 System configurations

Different configurations have been analyzed historically in the aim to optimize the use of line arrays, and reducing the acoustic shadows and increasing the distance of the line source effect [18, 19]. Four main configurations for line arrays can be determined:

Straight arrays: straight line sources match with the theoretical explanation of the line source effects. However, using those designs in practice produce polar response curves that vary substantially with length and frequency. At long throws and high frequencies they very narrow responses are obtained, which would drive to uneven coverage in the high frequency band.

Curved arrays: curved or arc sources produce wider directivity response than straight arrays. At high frequency the polar pattern is similar the the angle of the arc. This configuration is rarely useful, as even if the half bottom will be angled down to cover the first meters, the top half is angled upward pointing to the ceiling. However, when a wide vertical coverage is needed, as in theaters with several box seat floors this configuration is used.

J arrays: J arrays are a combination of arc and straight arrays. Those are a common approach to obtain an asymmetric pattern, curving the lower section of a line array. The a straight source is located above and adjacent to the arc source. The straight segment provides long throw and the arc segment provides coverage in the closes audience areas, determining a short throw. Together they provide an asymmetrical polar response in the vertical plane that is well suited for many venues.

Spiral arrays: a spiral or progressive array is based on a continuous, progressively curved profile. Those also provide an asymmetrical polar response in the vertical plane. Unlike a J array, they present a continuous curve. The top of the array is almost straight with angles of 1 degree between elements, and increases at the bottom to between 6 and 10 degrees. A well designed spiral array could have an almost constant directivity pattern with frequency. The array on Figure 4 is configured as a spiral array.

Spiral arrays are nowadays the most used configuration in the majority of the venues. This allow to make discriminations between different array elements that point to several target areas, allowing to apply different equalization or aligning parameters for short, mid or long throws.

When it comes to real applications, usually not only a stereo configuration is used. Several loudspeaker clusters cover specific areas that the main PA is not able to. So, typically up to six different clusters can be found on a concert PA:

1. Main PA: consists on the left and right stereo configuration that covers the majority of the audience area. It is the largest line array cabinet cluster, consisting usually from 6 to 20 speakers per side.

2. Outfills: these clusters are placed in a more outside lateral position of the stage, and oriented horizontally towards their respective sides. Those clusters are used to cover the side terraces that the horizontal dispersion of the main cluster is not able to. Usually they are formed by the same or smaller type of cabinets as the main PA, but with a lower number of them.

3. Frontfills: these are usually point source loudspeakers or individual line array cabinets that are placed along the front part of the stage, providing coverage for the first rows that the array curvature is not able to.

4. Delay towers: these clusters are used in very large venues, where one system is not enough to achieve the desired SPL to the most distant audience area. These are usually placed in a specific tower made of truss material, and placed 30 to 50 meters away from the stage. The loudspeakers are usually the same or a smaller model than the main PA.

5. Central channel array: if the stage is very wide and the main PA has a long horizontal separation, smaller line array cabinets are deployed at the center top of the stage, pointing with a strong tilt down to the first audience rows.

6. Subwoofer arrays: the subwoofers can be placed in many different configurations depending on the venue's physical possibilities and needs. In following parts more detailed options are discussed.

In order to define the proper parameters to achieve the coverage for a specific venue, acoustic modeling software tools are used. The design process starts by entering the dimensions of the area to cover and the required SPL. The program then suggests the number and arrangement (height of the upper element, inclination of the frame, and the different angles between the cabinets) of cabinets. The most known is EASE Focus [20], designed by German company AFMG, which can be used with different loudspeaker manufacturers data. Also custom made programs are offered by loudspeaker manufacturers, as Meyer Sound's MAPP [21], Electro-Voice LAPS, and JBL Vertec Line Array Calculator are available for their own systems.

Once the system is set up, aligning and equalization of the different loudspeaker clusters has to be done. The usual way to align and equalize a system is with help of acoustic measurement tools like Smaart Live [3]. These are based on dual-channel real-time FFT analysis, to calculate the magnitude and phase transfer function. The first channel is connected directly from one of the main outputs of the mixing console and the second channel is connected to a microphone placed in the audience listening area. This way, the phase of the different speaker clusters can be adjusted at the measuring position and the system can be equalized to obtain a flat response. However, this procedure requires playing pseudorandom pink-spectrum noise though the different loudspeakers while adjusting those parameters, which is usually disturbing and time consuming. For this reason it has to be adjusted and configured hours prior to the venue, as it couldn't be done with the audience in the venue area.

The physical division of the low frequencies cabinets allows also to configure different subwoofer arrays. As the wavelength increase at low frequencies, in order to arrange physical loudspeaker configurations more space is needed. In this sense, it is possible to take advantage of the fact that the subwoofers are separated from the array and placed on the ground, as they can be easily arrayed in different positions. This subwoofer arrays are based on a beamforming technique, by which the sound waves emitted by a group of driver can be aimed and shaped. In a beamformed array, the loudspeakers are driven separately and each signal has its own delay and level. Three kind of woofer arrays can be defined:

Broadside arrays: this is the most simple and usual form of deploying the subwoofers. It consists on several cabinets arranged in a horizontal row. Depending on its relative position, the radiation pattern can be modified to create light beamforming in order to point the most possible energy to the audience area and avoid an excess of SPL in the stage area.

Gradient arrays: here the subwoofers are arranged and driven in specific ways to provide directional patterns, usually cardioid and hyper-cardioid. Such arrays require the use of different signals for each cabinet, that may contain delays and polarity inversions to achieve their results. They can be bought as single cabinets or designed by positioning of subwoofers.

End-fire arrays: this consists in a number of cabinets are arranged in a spaced row pointing to the desired direction of radiation, and driven with progressive delay amount to create a very narrow pattern. End-fire arrays are rare, and are only useful in specific long-throw applications, outdoors or in huge venues.

3 System overview

In this chapter an overview of the design goals and the different signal processing procedures that take place for the realization of the automatic calibration and equalization of the line array system is performed.

In Chapter 3.1 the different requirements for ans signal processing operations are for adjusting the PA system is presented. The actual processing operations applied to the audio signal is explained. In Chapters 3.2 and 3.3 the different operations are explained in order to obtain automatically the parameters for the designed signal processing chain.

As explained in Chapter 2.5.2, the control of line arrays is being widely investigated and DSP processing is included in the line array cabinets to improve its performance and line source effect. However, the complete system, conformed by the different loudspeaker clusters tuning has not had such attention when it comes to its adjustment. The degrees of freedom in the placement of the subwoofers, caused by the adaptability of the systems to different venue types makes adjustment of the entire system a must every time it is set up in a different place.

The aim of this automated system is to make a complete adjustment of the PA by integrating parameter calculating mechanisms in the nowadays used digital crossover units, and integrate the measuring mechanisms with the processing chain. The presented system avoids all the manual procedures explained in Chapter 2.5.4, and it can be done even with audience, being much more time and effort efficient.

The calculation of the parameters does not need any real time intervention of the sound engineer, as once the system acquires the responses of the different loudspeakers the parameters are calculated in a few seconds. This makes possible that the measurement procedure is no more dependent on real-time changes, and pseudorandom noise can be avoided as measuring signal. Thus, much less annoying sweep measurements can be performed.

As explained in Chapter 1.2, the system that has been designed has some constraints. From now on, just two loudspeaker clusters will be taken into account, considering one mono loudspeaker array and one subwoofer cluster. So in this sense, a two way system can be considered from the system processor point of view. However, the array element could have multiple ways, but it is considered that the aligning and response of those is already treated by the loudspeaker's DSP procedures.

As line array elements can be divided in different vertical sections for short, mid or long throws, the system adjustment is also possible at several measuring positions, placed at different distances from the stage. In this project, just one measuring point has been taken in to account. This target area is considered to be at the Front of House (FOH) position, where the mixing desk is placed in concerts, as is the reference point for the sound technician [22]. Usually the FOH is placed at a mid throw area. The distance is not fixed, as it depends on the size of the venue, but usually it is between 15 and 30 meters from the stage. This could then be taken a a median measure between the front stage area (short throw) and the long throw.

Three sweeps will be used in total. As a two way system is considered, the first two sweeps are played separately through the low frequency (LF) speakers and through the high frequency (HF) speakers. These are recorded in order to then calculate the impulse responses of the subwoofers and the line array. With those responses the different sets of parameters for the processing chain are calculated. The third sweep runs through the designed processing chain and sounds through the whole PA, in order to verify that the expected response, that can be calculated beforehand, is accomplished, and if its not the case be able to re-adjust some parameters if necessary.

3.1 Audio signal processing chain

The automated processing unit is designed following the typical processing chain for PA systems shown in Figure 6. The usual PA processing rack includes the system crossover, two channels of analog graphic equalization and two channels of analog high quality compressors. The main output signal of the mixing desk is conducted through a graphic equalizer (and compressors if needed), in order to apply the equalization directly to the whole system. This is the usual way to equalize PA systems, as it is much more intuitive for the mixing engineer to have a graphic curve of the complete band instead of each component group separately. Even if nowadays almost all digital mixing desks include graphic equalization capabilities, it is still usual to see analog units as sound engineers like to have fast and direct access to it.

As explained in Chapter 2.2, the nowadays used digital crossover units have evolved to a more complex so called Loudspeaker Management Systems, which has also additional processing capabilities. Usual are:

- In/outputs definition and signal path definition. This allows to create sum signals from stereo mix for subwoofers, auxiliary buses for frontfills or outfills and define the number of crossover ways for the system.
- Filters selection: Linkwitz-Riley, Butterworth, Bessel, FIR filtering...
- Filters cutoff frequency and slope adjustment (typically from 6 to 48 dB/oct.).
- Parametric equalizers.
- Delay adjustment.
- Gain at inputs and outputs.

Following the usual chain, the output from the graphic equalizer feeds the processing unit, where the first step is applying the crossover filters to split the audio band into the different sub-bands for each speaker group or driver into a specific speaker. In this case two bands are used: mid and high frequencies to feed the line array and low frequencies to feed the subwoofers. The most widespread type for crossover filters are Linkwitz-Riley filters, and will be the ones used during for the automated processing unit.

Once the signal has been split into two bands, individual processing for the low frequency and high frequency ways can be applied. At this point, the delay is applied in order to acoustically align the speakers at a precise measurement point, and to achieve a coherent response in the crossover band. Also an overall gain is applied to each output. This gain compensates median level differences between the different system ways, achieving a relatively equal level for all the components in the system.

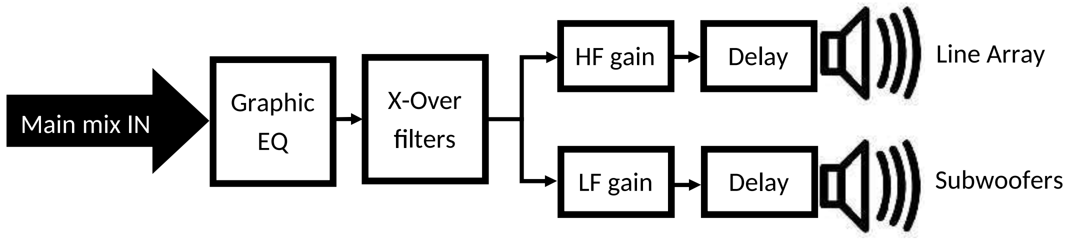


Figure 6: *One channel of the signal processing chain for a PA.*

In the design proposed in this project, the graphic equalizer is also integrated in the processing unit, offering additional capabilities compared to the state-of-art systems. Thus, with just one unit, complete control of the PA equalization and time delay adjustment can be achieved.

The designed automated processing unit includes the features shown in Figure 6 and the needed procedures to adjust the four sets of parameters automatically: graphic equalizer gains, crossover frequency, delay adjustment and output gain.

3.2 Parameter calculation

The calculation of the parameter sets for the automatic project. The calculation of these has to be understood as a inverse processing procedure, as in this case the desired response at the end of the processing chain is known.

This desired response is aimed to present the flattest possible magnitude frequency response and phase adjustment between the subwoofers and the line array. The phase adjustment is translated in minimum group delay difference in the passband. Thus, if observing the signal processing chain in Figure 6, the order of the parameter extraction is inverse, as following: time delay values extraction, cutoff frequency response, average output gain for each way and finally the graphic equalizer gain parameters.

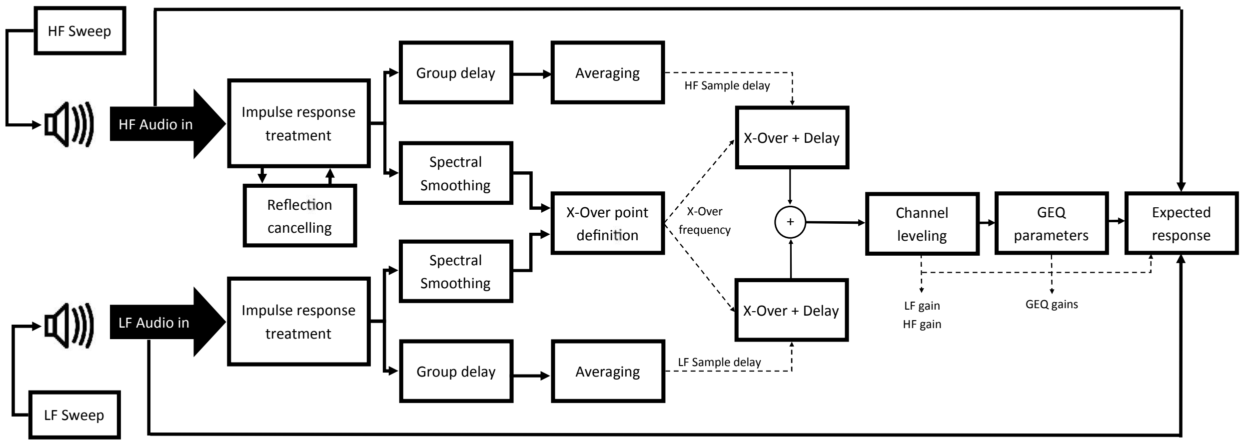


Figure 7: *Parameter calculation block diagram.*

In Figure 7 a flow graph for the parameter extraction is presented. In continuous line, the signal flow is represented. The dotted lines represent the use of the extracted parameters to perform a specific signal processing procedure. The main blocks of this system are:

- Sweep generation: to obtain a distortion-free response of the different loudspeaker ways [6], two logarithmic sweeps have been used to obtain separately the responses of the subwoofers and the line array. The responses are recorded in a synchronized way in order to extract the parameters just after the sweeps have been reproduced.

- Impulse response: the impulse responses are obtained using de-convolution mechanisms. Because of the properties of the logarithmic sweep, the linear response can be isolated. The responses are then treated to compensate the measuring system irregularities and truncated to obtain the desired characteristic.

- Reflection canceling: a ground reflection canceling mechanism is implemented. Its utility and implementation is explained in Chapter 5.6.

- Group delay: the group delay extraction from the impulse response is the basic procedure to define automatically the distance from the loudspeaker to the measuring point. By averaging its values in a certain band the estimation of the distance is made.

- Spectral softening: the magnitude frequency response is obtained from the impulse response. Softening of the response is used to avoid system or measurement irregularities to affect the parameter definition.

- Crossover frequency definition: this value is calculated from the magnitude frequency response, extracting the cutoff frequencies for each way of the system an determining the filters cutoff frequency from those.

- X-Over+Delay: this block processes the acquired signals with the already calculated time delay and crossover filters to obtain the summed signal. As the parameters for adjustment are the same that will be applied in the final signal chain, the summed response should correspond to the acoustic one at the measuring point for the complete system, thus from this response, full band parameters can be defined.

- Channel leveling: from this block the output gain is calculated, as an average full band gain can be extracted from the summed response and compared to the individual gains of the system.

- Graphic equalizer gains: those are also calculated from the full band magnitude frequency response, following the usual procedure in system tuning, where one graphic equalizer is used for the complete system tuning.

- Expected response: as the responses used for obtaining the parameters have been modified by different procedure like truncation, softening or the reflection cancellation mechanisms, the directly measured response is used to apply the processing chain with the calculated parameters and obtain a prediction of how the magnitude and phase responses from the system should be. This will allow to compare them to the measured response with the third sweep and check if they correspond.

Once the impulse responses have been obtained, the group delay and the frequency response are analyzed to extract the group delay values for each band and the crossover frequency. The signals are compensated to zero-time applying an inverse delay. Afterwards, the responses are filtered with the designed crossover filter.

An average gain is applied to each way before summing them to obtain the full band signal. The frequency response of the full band signal is used to calculate the gains of the graphic equalizer.

With these operations, all the needed parameters to adjust the PA are obtained. In order to simplify the system, the crossover filter type is not automatically adjusted, instead a fourth-order Linkwitz-Riley IIR filter has been designed and implemented with standard Matlab filter design functions. Thus, at the cutoff frequency, both ways are attenuated 6 dB and the filter presents a decay of 24 dB per octave.

3.3 Verification and re-adjusting

Despite that with parameters calculated in Chapter 3.2 would be enough to adjust the PA system, some additional procedures have been implemented to assure the proper functioning and adequacy of these parameters.

The presence of acoustic ground reflections caused mostly by the reflecting sound of the line array speakers ground can produce errors in the calculated parameters.

The well known comb filter effect appears on the frequency response, and can cause mostly errors in the automatic adjustment of the graphic equalization gains.

In order to verify that the response at the measuring point equals the expected response calculated by the last block in Figure 7, third full sweep though the whole PA (line array and subwoofers) is played. The parameters calculated in Chapter 3.2 are used to design a signal processing chain as the one explained in Chapter 3.1. The response of the full measurement is compared to the expected output calculated at the output of the graphic equalizer.

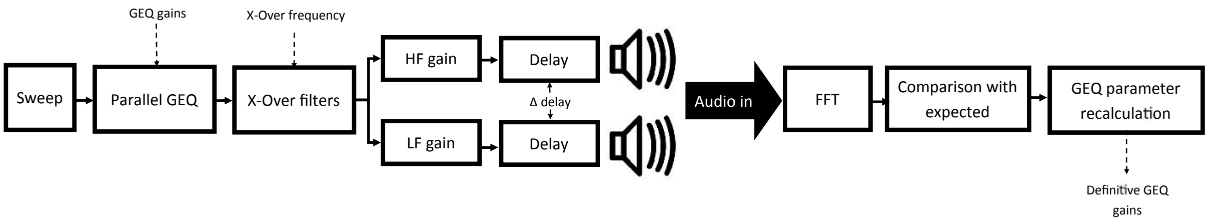


Figure 8: *Verification and re-adjusting block diagram.*

A block diagram of the verification and re-adjustment procedure is shown in Figure 8. In continuous line, the signal flow is represented. The dotted lines represent the use of the extracted parameters to perform a specific signal processing procedure. The main blocks of this system are:

- Sweep: the sweep signal fed as input signal has the same characteristics regarding frequency limits and duration as the the used for obtaining the subwoofer and line array responses.
- Parallel graphic equalizer: the sweep is filtered with the parallel graphic equalizer with the calculated gains in Chapter 3.2.
- Crossover filters: the Linkwitz-Riley IIR filter are implemented at this point. The sweep signal is divided by the fourth order filter into the two ways, for the subwoofer and the line array. The cutoff frequency is the calculated in Chapter 3.2.
- HF and LF gains: the obtained signals from the crossover are amplified by the determined gain calculated in 3.2.
- Delay: one of the signals is delayed in a certain amount of time. The exact procedure of determining which signal is delayed is explained in Chapter 5.2.
- Play and recording: once the sweep has been processed by the designed signal processing path, it is played though the whole PA. As the frequency increases exponentially with the sweep, the subwoofer starts sounding and progressively at the crossover frequency the signal is shifted to the line array element.

- FFT: the impulse responses are obtained and treated as the ones acquired in Chapter 3.2. Fourier transform is applied to obtain the magnitude and phase frequency responses.

- Comparison: the magnitude frequency response is compared via subtraction to the expected one calculated at the end of the Figure 7.

- Graphic equalizer parameter recalculation: the difference signal between these two responses is used to readjust the graphic equalizer if needed. If the graphic equalizer tries to equalize out the generated comb filter, by adding gain in the notch areas or subtracting it in the bumps, the difference signal presents a bump or notch, as the intensity of the reflected response increases in the same amount as the direct signal.

The difference between these signals is mostly caused by reflections. For this reason, a pre-processing in the responses is made to reduce the influence of the comb filter effect when ground reflections appear. However, as this solution does still not avoid completely the ground influence, this procedure is used as a security check in order not to force the tuning of the system unnecessary. If it is determined that a specific gain in a band is affected, this is set to nominal value, leaving this band un-equalized. This procedure is explained in more detail in Chapter 5.6.

4 Testing procedures

The functioning of the processing blocks explained in Chapter 3 has been tested in two stages. As the implementation of these blocks has been realized using Matlab environment, some additional functions have been made in order to simulate the responses of the measurement procedure.

Thus, the testing part is divided in two parts. First the simulation of the system responses using Matlab functions that aim to reproduce the conditions of a real measurement are shown in Chapter 4.1. Once the functioning of the processing blocks has been validated, a real implementation in a 1:10 scale model in an anechoic chamber has been designed and constructed.

For designing the scale model, different speakers have been measured to obtain similar scaled responses as a real PA system would have. In Chapter 4.2 the values for the distances and the target responses of the loudspeakers are defined. Also the specifications of the used equipment are presented. In Chapter 4.3 the setup of the system and the measurement procedures are explained.

4.1 Simulated responses

In order to model test the proper functioning of the implemented processing blocks, two Matlab functions have been made to simulate the effect of the speakers in a real situation. The first function models the response to the two first sweep measurements, thus, the effect of the subwoofer and the line array loudspeakers. The second function models the response of the full band verification sweep.

The part that has to be simulated comprises the signal path between the generated sweep and the recorded response. This includes the A/D and D/A conversion, the amplification, the response of the loudspeaker components, the time of flight of the wavefront from the loudspeaker to the microphone and the microphone response.

Thus, the following procedures have been applied to the reference signal (in this case the sweep) for modeling the before enumerated phenomenas:

- Filtering: different order Butterworth filters have been used to model the magnitude frequency response of the subwoofers and line array. The subwoofers present a high-cut at 25 Hz and low-cut at 250 Hz, the line array presents a high-cut at 200 Hz. Also additional frequency response modifications are made to test the graphic equalizer and the crossover frequency selection.

- Time-delaying: the responses are time delayed by adding zero padding at the begging of the signal. This models the time of flight of the sound waves to go from the loudspeaker to the measuring position. The number of samples is calculated with the following expression:

$$s = \frac{\Delta D f_s}{c}, \tag{10}$$

  where f_s stands for the sampling frequency and c for the speed of sound. ΔD is the relative distance between the subwoofers and the line array.

- Reflection generation: a ground reflection signal is simulated for the line array elements. The subwoofers are not taken in to account as they are placed on the ground and are not generating reflections. The path difference is calculated subtracting the reflected path from the direct path shown in Figure 9. The signal is duplicated, delayed and filtered to obtain a highpass response, which follows the ground reflection response. Finally it is phase inverted and multiplied by a reflection coefficient.

- Noise addition: average white Gaussian noise is added to the responses. This is an approximation to the sum of the different noise fonts, both acoustic background noise and the measurement system contributions (A/D amplifiers, cables, microphones...), such as inter-modulation, and thermal noise.

- Aliasing filtering: a lowpass filter at a cutoff frequency half of the sampling frequency.

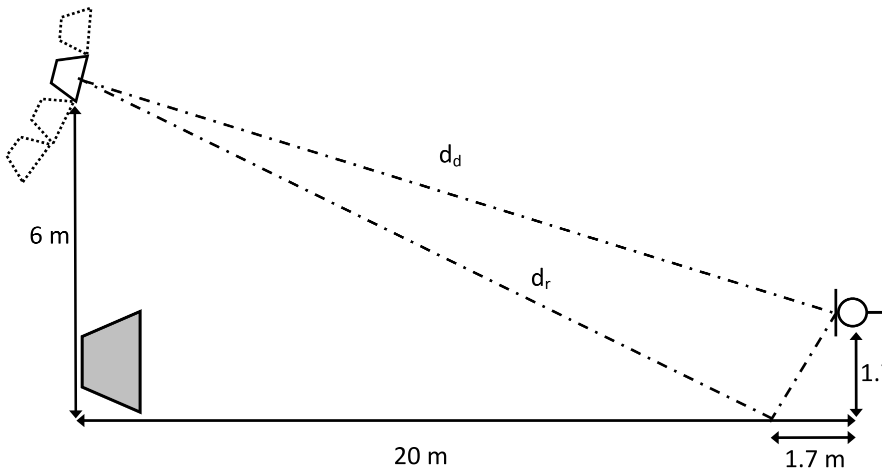


Figure 9: *Approximate path difference caused by ground reflection.*

In Figure 10 different impulse responses are shown for different relative distances between the subwoofers and the array and different ground reflection intensities. In Figure 11 the frequency responses are shown for different reflection intensities which create the comb filter effect, and different values of average Gaussian white noise.

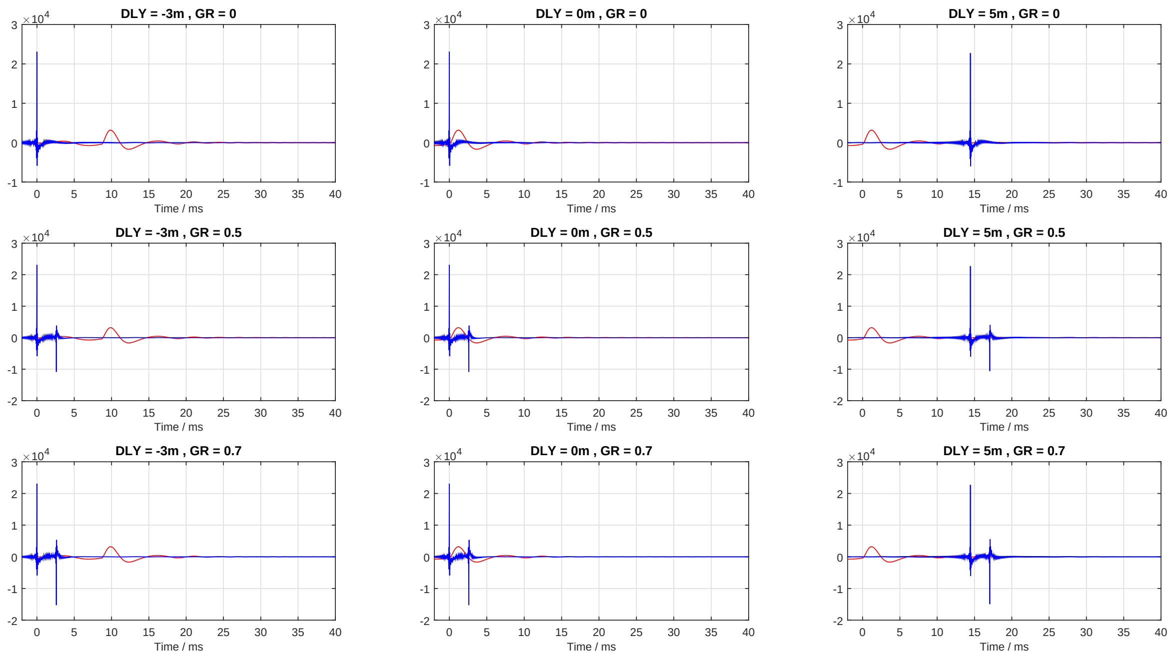


Figure 10: *Simulated subwoofer (red) and line array (blue) impulse responses for different ground reflection coefficients relative distance between sources (m<0 means the subwoofers are more distant than the line array).*

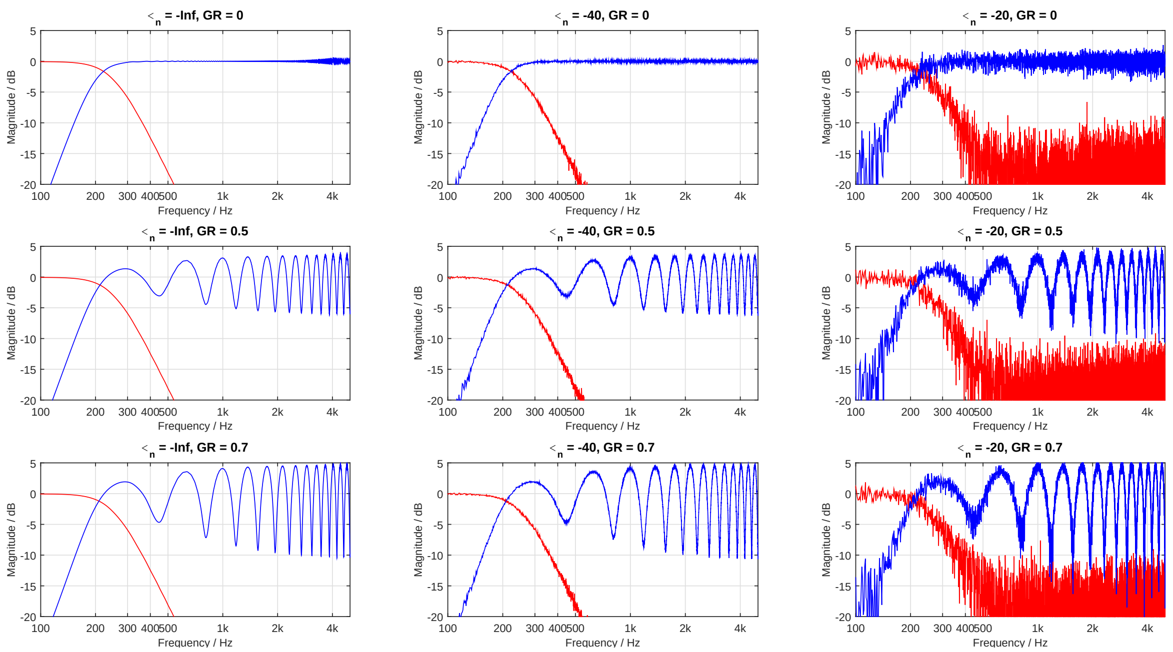


Figure 11: *Simulated subwoofer (red) and line array (blue) magnitude frequency responses for different ground reflection coefficients and average Gaussian noise powers.*

In Annex A.1 the Matlab function for the simulation of the line array and the subwoofers is presented. In Annex A.2 the Matlab function for the full sweep simulation is shown. The main difference between them is that the first one receives the same reference sweep signal and returns separately the two responses. The second function receives the processed sweep by the automatically adjusted processing chain, thus as the crossover is applied, two separate and different signals excite the subwoofers and the line array, but one signal is returned, as a single full band measurement is performed for the whole system.

4.2 Scale model definition

For now the possibility to test the system in real situation was not possible because of the physical and material constraints. It is difficult to have available a real line array system, because of its elevate rental price and the need of personnel to set it up. Thus, testing it with a real system in an external esplanade is not an option at this point. Therefore, a scale model has been implemented in an anechoic chamber.

The scaling factor is decided be the physical constraints of the available space to do the measurement. The chamber in which is has taken place is 3x3 meters between each end of the absorbing panels. The largest distance that has to be scaled is the one between the PA, which would be in real situation in the stage area, and the measuring position, which would be at the FOH position, which is usually around 20 meters. Thus, in order to have space to place the structure for holding the system, a maximum distance of around 2 meters is defined between the loudspeakers and the measurement microphone, from which a 1:10 factor scaling is obtained.

When designing the scale model, the approximate distances shown in Figure 5 have been divided by 10, but also the frequency range has to be scaled. Frequency is an inverse parameter to the wavelength. As wavelength is a distance related parameter, it is has to be down-scaled 10 times, thus frequency becomes an up-scaled variable. In Table 2 a comparison between the real case situation and the scale model main parameters can be observed.

Parameter	Real	Scale model
Distance to measuring point	20 m	2 m
Line array medium height	6 m	60 cm
Subwoofer frequency range	20 - 250 Hz	200 Hz - 2.5 kHz
Line array frequency range	150 Hz - 20 kHz	1.5 - 200 kHz
Sweep frequency range	20 Hz - 20 kHz	200 Hz - 200 kHz

Table 2: *Main parameters in real situation and in the scale model.*

The frequency range that has more interest in this application, is around the passband between the line array and the subwoofers, as the main phase adjustment, crossover frequency definition and equalization issues happen in the crossover and middle band. Therefore a compromise has been taken to limit the measuring range.

The problems of working with very high frequencies became an impediment when designing the scale model. For frequencies far above 20 kHz, ultrasound techniques have to be used. This also implies that the experiment cannot be taken place in normal conditions because of the increased air absorption. First an option with a fish tank filled with helium was considered, as helium has similar absorption and transmission factors as air for the ultrasound band.

Finally the scale model maximum frequency has been limited to 20 kHz, having a working band from 200 Hz to 20 kHz, which is equivalent to a limitation to 2 kHz in real application. This has made possible the use of standard audio material, and no special ultrasound loudspeakers and microphones had to be used to perform the measurements.

Once the frequency and size constraints are defined, the measurement material and parameters be defined to achieve the scaled responses. To perform the measurements, first the signal chain has to be defined. As the current implementation is on computer-based platform, a audio interface card has to be used to manage the input and output channels. In this case, as a two way system will be implemented, two output channels are used, and one input channel is used for the microphone audio acquisition at the single measuring position. The signal level at the output of the audio interface has to be amplified to reach a sufficient level to excite the loudspeaker membranes, thus a power amplifier is used. Figure 12 shown the measurement chain used for the scale model.

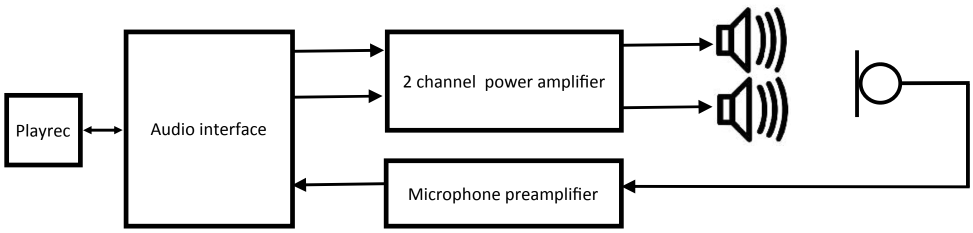


Figure 12: *Component chain involved in the measuring procedure.*

As the system is designed with Matlab functions, additional utilities have to be used to ensure that the playing and recording of audio signals is synchronized. *Playrec* is a package that provides versatile access to sound cards inputs and output streams in a multichannel configuration, that can be used on different platforms (Windows, Mac, Unix). In this case it is used to synchronize the sweep playing with the recording of the responses.

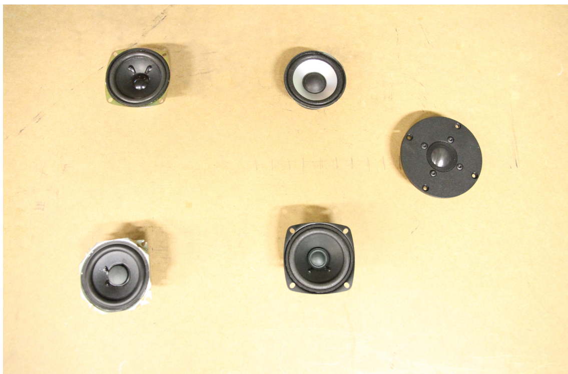
The chosen audio interface is a MOTU UltraLite MK3. This sound interface offers professional audio quality with samples rates up to 192 kHz. It has 8 line-level analog inputs and 10 analog outputs, upon other digital connection features. The most convenient feature that has been key to chose it for the purpose of the scale model measurements is the availability of on-board mixing and processing. A graphic interface allows to filter any input or output, which is very convenient to shape the responses of the subwoofer or the line array in order to test different configuration in a very dynamic and fast way.

The power amplifier used for driving the loudspeakers is a Quad electroacosutics 240 watt twin-channel amplifier. It uses current dumping technology, that provides extremely high linearity and low crossover and inter-modulation distortion. Current dumping is based on a combination of two amplifiers, a very high quality class A low power amplifier which amplifies precisely low amplitude ripples of the signal and a high power class B current dumping section that delivers the power part.

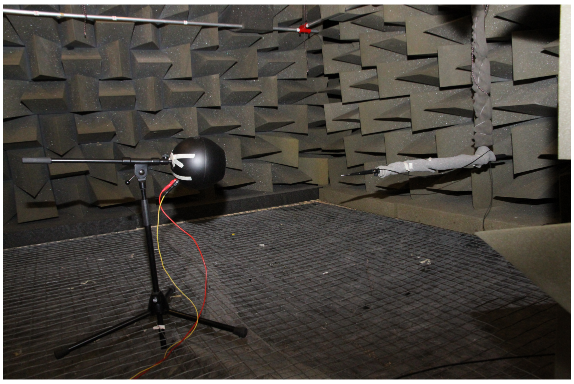
The microphone is a G.R.A.S. free-field omni-directional microphone. It uses a 1/4 inch membrane, and it is has an integrated pre-amplifier, thus it needs a specific module that supplies the polarization current and includes some basic gain and filtering controls. The output of this module is a line-level signal that feeds one of MOTU input channels. More detailed specifications are presented in Annex C.

Different loudspeakers were measured to determine the best option depending on its responses. As low frequency component, a sphere loudspeaker formed by a 5 inch cone component inserted in a spheroidal enclosure that can be seen under the measurement procedure in Figure 13b. Some of the measured loudspeaker components for the high frequency component can be seen on picture 13a. Four low quality speakers with 4 and 5 inch cones have been measured, and two SEAS high frequency tweeters with aluminum membrane (the SEAS 19TAF/G and the SEAS 25 TAF/G). The frequency range of the measurements is adapted to the component specifications in order not to damage them by applying too low frequencies:

- 5-inch drivers: 50 Hz to 40 kHz.
- SEAS tweeters: 500 Hz to 40 kHz.
- Sphere enclosure: 50 Hz to 30 kHz.



(a) Image of the different high frequency measured speaker components.



(b) Low frequency spherical loudspeaker under measuring test.

The responses of the loudspeaker components have been obtained using the logarithmic sweep measurements explained in Chapter 2.1.2. Finally the chosen driver has been the SEAS 19TAF/G because of its fairly linear magnitude frequency response up to 30 kHz. The datasheet of this component is availabe in Annex B. In Figure 14 the magnitude frequency responses of the chosen components is shown.

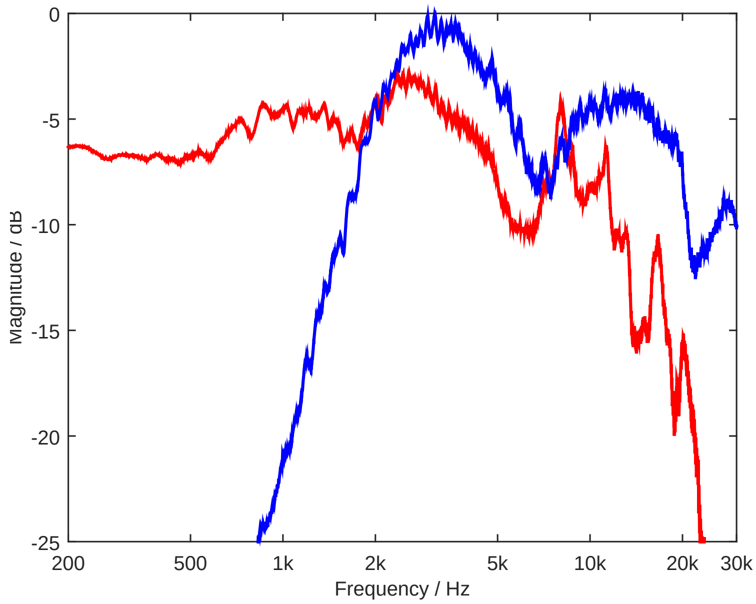


Figure 14: *Magnitude frequency responses of the choosen components. In red the low frequency sphere, in blue the SEAS 19TAF/G.*

4.3 Scale model measurement

With the chosen components and defined distances in Chapter 4.2, the construction of the scale model has been realized in the anechoic chamber.

As the anechoic chamber is a full anechoic chamber, the ground is covered by a metallic net, thus when walking on it the whole ground is vibrating and moving. The fact of using a 1:10 scale model implies that the precision in the distances has to be very accurate. For this reasons, and in order that the involved acoustic components do not move from their positions, those have been placed without touching the ground. A special structure has been build to hold the model PA system, reinforced with tubes. In Figure 15 the PA structure is shown. The structure allows to move freely the high and low-frequency speakers to arrange different setups and measuring distances easily and test different configurations.

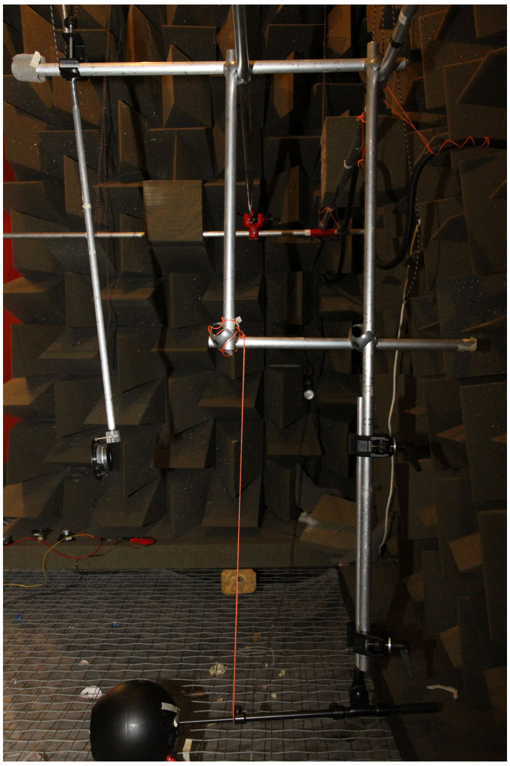


Figure 15: *Detail of the constructed structure to hold the model line array.*

In Figure 16 the measurement desk is shown. The computer runs Matlab and is connected to the MOTU interface. The two output channels of the MOTU are connected to the amplifier, which correspond to the high and low frequency loudspeakers, and one channel of the microphone module is used.

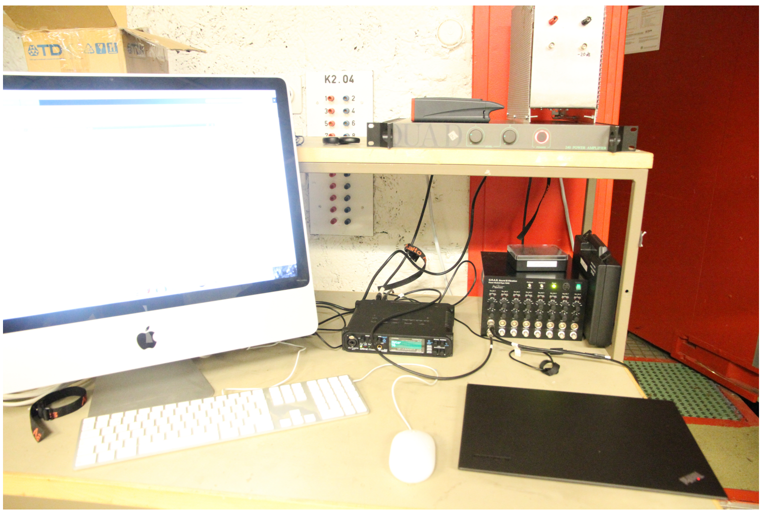


Figure 16: *Measurement desk.*

For choosing the sampling frequency a compromise has been taken between the size of the recorded files and its processing time and the required precision, deciding to use 96 kHz. The sampling frequency determines also the precision in the impulse response and group delay readings and thus the minimum discernible time and therefore distance difference:

$$\Delta D_{min} = \frac{c}{f_s} = \frac{346}{96000} = 3.6\text{mm} \tag{11}$$

This also determines the minimum discernible path difference between two signals. Thus, the distance precision in time-aligning the PA components in the scaled model is 3.6 mm. This is important to discern between a direct signal and a reflected, and it comes critical when dealing with the ground reflection, where a strong signal arrives just milliseconds after the direct sound. In Table 3 some values are given that illustrate this constraints, based on the scaled distances in Table 2.

Parameter	Real	Scale model
Array-sub path difference	-1.7 - 2.3 m	-17 - 23 cm
Array-sub time difference (ms)	-5 - 6.5	-0.5 - 0.65
Array-sub sample difference (@96 kHz)	-235 - 335	-23 - 33
Ground reflection path difference	0.9 m	9 cm
GR time difference (ms)	2.83	0.28
GR sample difference (@96 kHz)	136	14

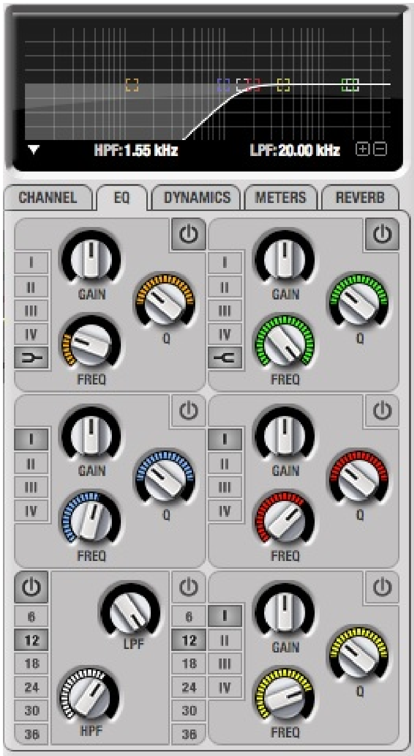
Table 3: *Reference values in real situation and in scale model for discerning different wavefront arrivals.*

The sweep is generated with a Matlab function, that returns the waveform signal to reproduce and the inverse reference spectrum. The inverse reference spectrum allow to obtain easily the impulse response convolving it with the recorded signal, as it already has the amplitude compensation caused by its logarithmic nature. The sweep time is a compromise between the generated data and the need of amplitude precision, as longer the sweep is, higher is the energy in the impulse response, and therefore the signal to noise ratio. Different sweep times have been tested, and finally 3 second sweeps have been used for measurements.

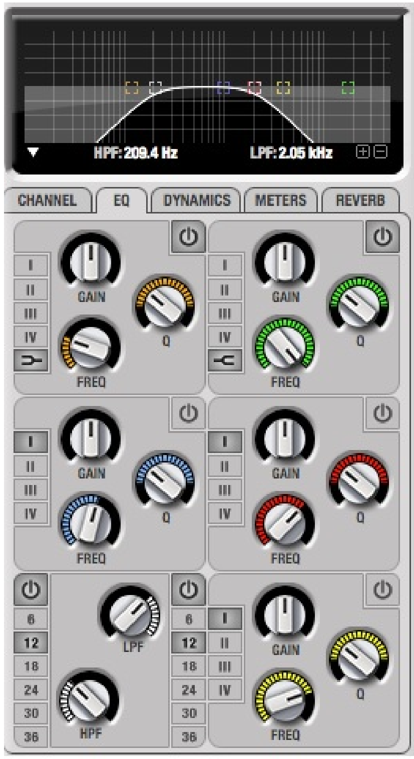
The sweep frequency range is from 200 Hz to 40 kHz for both ways of the system. This is defined so, as in some point of the system the responses are summed. If the responses are generated with different reference signals, re-sampling and shifting should be made. This way simple summing of the time-domain recordings can be made. Even if the evaluated system limits are between 200 Hz and 20 kHz, the extended range up to 40 kHz allows to extend the equalization of the system. Frequency shaping of the sweep signal is produced when its generated, the magnitude of frequency response is attenuated 3 dB at the sweep frequency limits, thus valid values should be taken with a certain margin to this limits.

As shown in Figure 14, the responses of the high and low frequency components are not an ideal approximation to the expected scaled responses. Therefore, the filters of the MOTU audio interface have been used to shape the responses. The definition of those filters can be easily done with the graphical interface of the MOTU's software, shown in Figure 17. Different filtering options have been tested, and the configuration which suits best with the requirement has been obtained with the following parameters:

- LF way:
  - HPF 170 Hz, 18 dB/oct.
  - LPF 1.66 kHz, 12 dB/oct.
- HF way:
  - HPF 1.03 kHz, 12 dB/oct.



(a) *High frequency channel filters.*



(b) *Low frequency channel filters.*

Figure 17: *MOTU filters graphical user interface.*

The filter configuration ensures that there is a certain overlap in the magnitude frequency responses, is a good approximation to a 10-scaled frequency range of a medium-sized line array and subwoofer configuration (20 - 150 Hz in subwoofers and from 110 Hz upwards in the line array).

A calibration process is performed before the measurements. This is done in order to verify and compensate if necessary the measurement system responses. The calibration process measures the response of the part of the measuring chain consisting in the MOTU audio interface input and output channels, the amplifier and the loudspeaker cables. Three parameters are critical to assure:

- Group delay: the application of non linear phase digital filters in the audio interface implies an alteration of the phase response, which affect to the overall group delay of the measuring system.
- Bandwidth: assure that none of the elements introduces hard bandwidth restrictions.
- Flatness of magnitude frequency response.

The response of the speakers is known from the measurements in Chapter 4.2, and the microphone specs present an almost flat and linear phase response for the functioning band, thus it is considered transparent.

The calibration is performed connecting the output of one channel of the amplifier to a 'dummy' resistor, creating a feedback loop. The resistor has a 20 dB attenuated output which is connected to the input channel of the audio interface. Using the same reference signal as the one which will be used for the measurement, the response is measured and stored for each channel independently. The use of this calibration responses is explained in Chapter 5.1.

The loudspeakers and the microphone are placed accurately on the constructed structure, and reflecting surfaces are covered with absorbing material to avoid its influence in the measurements. The tubes that are hole inside are covered on the edges to avoid its resonance. In order to determine accurately the relative position between the different components, a millimeter precision laser distance meter was used, and a millimeter ruler for short distances.

To test the ground reflection influence, wooden plates were placed below the measuring area. The lowest frequency that a squared reflecting plate is able to reflect, follows Equations 12 and 13.

$$f_A = \frac{c}{(\frac{1}{s} + \frac{1}{r})A^2 cos^2\phi}, \tag{12}$$

$$f_B = \frac{c}{(\frac{1}{s} + \frac{1}{r})B^2}, \tag{13}$$

where s stands for the distance from the sound source to the center of the reflector, r stands for the distance from the receiver to the center of the reflector, A is the length of the reflector, and B is the width of the reflector and ϕ is the reflection angle.

Considering the scaled sized showed in Figure 9 and the size of the reflector is 1.3 x 1.35 meters the minimum frequency that the reflector is capable of reflect is around 170 Hz, thus is below the minimum used frequency and valid for this scale model.

The procedure of placing the components together with the tube structure construction took several hours, and the measurements were conducted during 10 days. During this time different filtering options, relative distances, delay compensation and equalization methods explained in Chapter 5, and the influence of the ground reflections were tested. In Figure 18 the setup of the whole scale model is shown.

Once the measurement chain is calibrated, and the system is in position, very few physical changes had to be made. The measurements were conducted by running

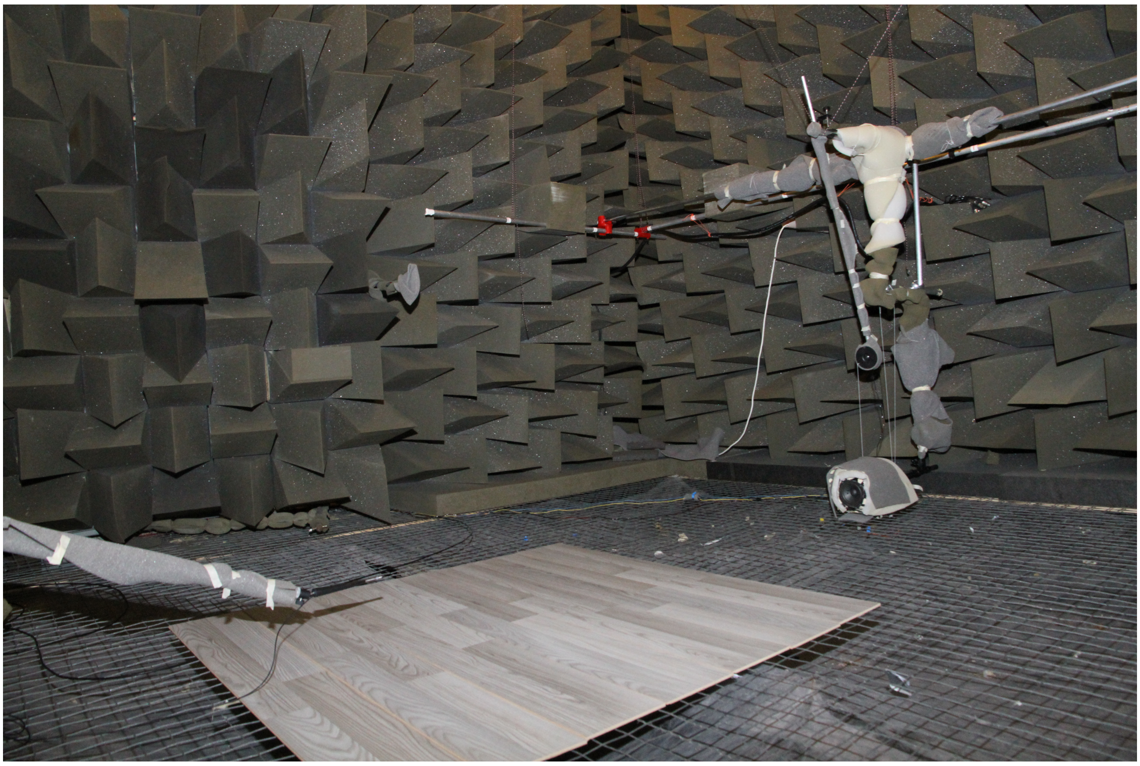


Figure 18: *Image of the scale model in the anechoic room. The bass and treble speakers are seen on the right, the microphone on the left, and the reflection plate below it.*

a Matlab script that includes the different functions. The sequence of functions was the following:

1. Sweep signal generation function: frequency limits and sweep time can be varied.

2. Sweep play and record: sequentially the LF and HF way sweep and record the signal.

3. Response acquisition: by deconvolution of the recorded signal with inverse reference spectrum.

4. Response treatment and automated parameter extraction.

5. Verification sweep processing: the third verification sweep is processed with the extracted parameters.

6. Verification sweep play and record.

7. Response acquisition and automated parameter recalculation if necessary.

Each measurement lasted about 15 seconds, between the sweeping (3 seconds each sweep), the computing time for the parameter extraction and the results plotting.

5 Calibration and equalization methods

In this chapter a more detailed insight to the parameter extraction procedures is given, using the measured responses from the scale model presented in Chapters 4.2 and 4.3.

In Chapter 5.1 the extraction and treatment of the impulse response obtained from the recorded signal is explained. In Chapter 5.2 the procedure to calculate automatically the delay to compensate the arrival time difference is presented. In Chapter 5.3 the procedures to define automatically the crossover frequency is explained. Once the expected signals for each way are calculated, the full-band operations can be done. In Chapter 5.4 the calculation of the output gain for each way is shown, and in Chapter 5.5 the gains of the graphic equalizer are set automatically. Chapter 5.6 explains the procedures used to verify the precision of the calculated parameter sets and how to avoid the affection of the ground reflection to the automatic calculation of those.

5.1 Impulse response acquisition

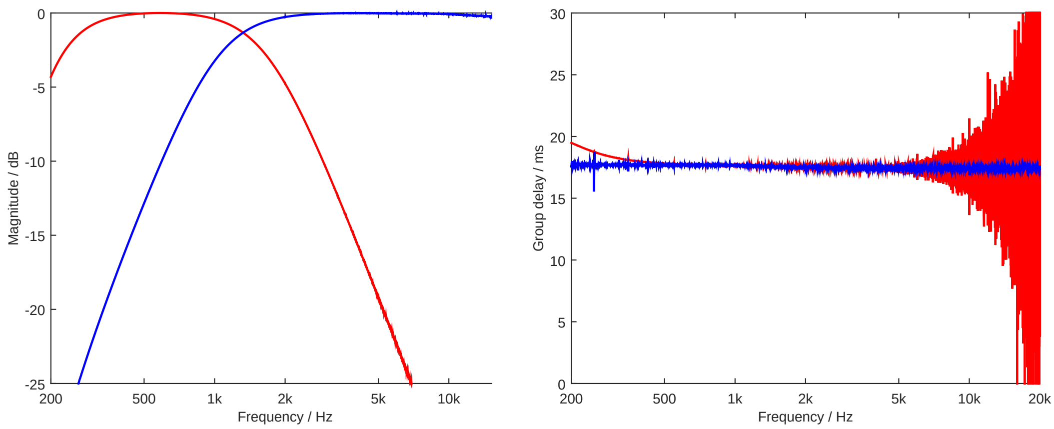
The impulse response of the low and high frequency ways is the basic function which will be used to extract the parameter sets. As the system is designed to work in rough environments, a pre-processing of the measured responses is made to avoid undesired reflections to affect the measure and to compensate the measurement system errors by using the calibration performed before the measurements.

The extracted functions from the impulse response that will be used for the automatic parameter sets definition are two:

- Magnitude frequency response: the magnitude frequency response of the LF and HF responses will be used to obtain three of the four sets of parameters:
  - Cutoff frequency point definition
  - Average output gain
  - Graphic equalization gains
- Phase response and group delay: the group delay derived from the phase response is used to define the distance of the loudspeaker to the measuring position, thus the time delay adjustment.

5.1.1 Calibration

The fact of using the audio interface filters to shape the responses of the HF and LF ways, present the magnitude response shown in Figure 19a. As this response is fairly flat and follows smoothly the response of the applied filters, no magnitude compensation is needed.



(a) *LF (red) and HF (blue) filters frequency response.* (b) *LF (red) and HF (blue) group delay added by the measurement system.*

However, not the same situation happens with the system latency, caused by use of buffers, inherent to the use of real time audio protocols, as the ones used to communicate the Matlab environment with the audio interface. Moreover, the use of non linear filters also modifies the phase responses affecting the group delay in a non-linear way. Thus, an offset delay is present at the measurement chain, mostly caused by the buffering and the non linear filters. Figure 19b shows the influence of those two phenomena over the group delay.

By averaging the group delay for each channel by its corresponding bandwidth, an average time latency is obtained. Multiplying this by the sampling frequency, the sample latency is obtained. For the filters defined in Chapter 4.3 this value is 1672 samples. This is presented as a delay in the impulse response, thus they are cropped by this number of samples.

The calibration procedure is also applied to the group delay, subtracting from the measured group delay curve the calibration curve:

$$\tau_g(f) = -\frac{d\phi_{meas}(f)}{df} + \frac{d\phi_{calib}(f)}{df}, \tag{14}$$

where $\phi_{meas}(f)$ is the phase response of the measured signal and $\phi_{calib}(f)$ stands for the phase response of the calibration.

5.1.2 Calibrated responses

In Figure 20 shows the beginning (first 10 millisecond out of 3 seconds) of the LF and HF measured impulse responses. The different shifting from the 0 time point in each response is caused by the time the wavefront travels from the speaker to the microphone. Thus, the different arrival times are caused show that the loudspeakers are at different physical positions in relation to the measurement point, and thus, are not time-aligned. The LF impulse response is scaled 10 times to be visible, as low frequencies have less impulse energy.

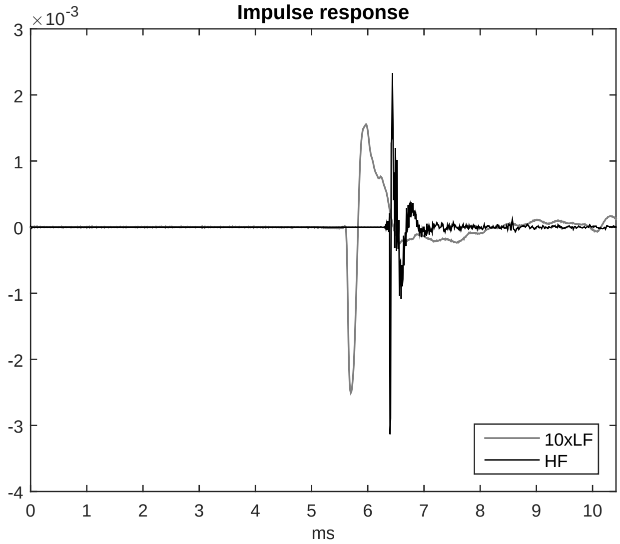


Figure 20: *Measured LF and HF impulse responses.*

Figure 21 shows the magnitude frequency response of the measured HF an LF ways at the measuring point. A strong ripple is appreciated, caused by reflections in the anechoic chamber and background noise.

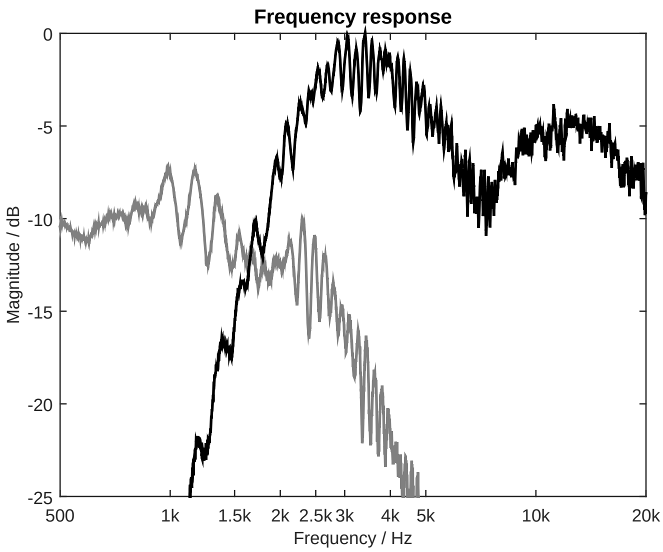


Figure 21: *Measured magnitude frequency responses.*

The different time of flight can also be evaluated form the group delay showed in Figure 22. It is visible that the frequency limits in which group delay becomes readable correspond to the part where energy is stronger (up to 2 kHz for the LF band and starting from 1.5 kHz for the HF band). In this band, the mean offset value is equivalent to the time of flight. The group delay is very sensible to noise and reflections, as observed, therefore some technique to mitigate it has to be used.

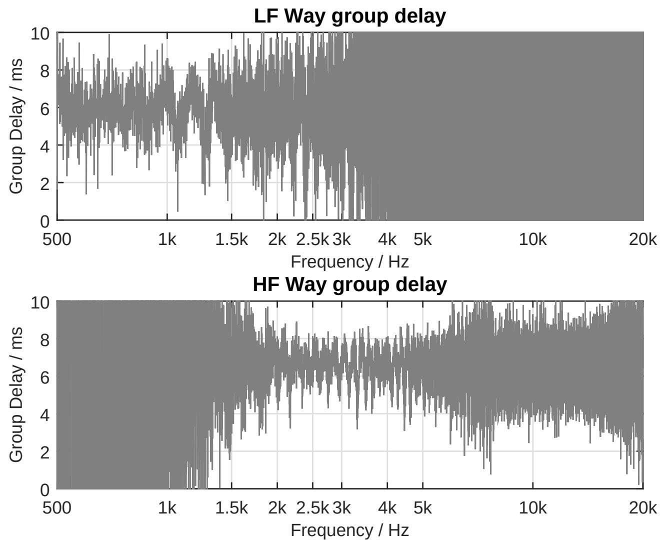


Figure 22: *Measured group delays without further processing.*

5.1.3 Impulse response truncating

The frequency response ripple and noisy group delay are caused mostly by late reflections in the impulse response. To avoid these effects, the impulse responses are truncated to the minimum number of samples possible. The compromise is between the minimum representable frequency and the aim to avoid reflections.

The functioning band of the system is from 200 Hz to 20 kHz, so the sweep initial frequency is 200 Hz (lower frequencies than 20 Hz in real case scenario would damage the drivers). As at this point the response is 3 dB below the nominal value (due to the sweep spectrum characteristics explained in Chapter 4.3), a margin is defined to the minimum operable frequency in terms of automated parameters extraction. Therefore, 300 Hz is chosen as the limiting band for parameter extraction. Also a lower margin is chosen to define the minimum representable frequency is chosen 250 Hz, and the impulse response length is defined by Equation 15.

$$N_{imp'} = \frac{f_s}{f_{min}}, \tag{15}$$

where f_s stands for the sampling frequency, f_{min} is the minimum representable frequency and $N_{imp'}$ is the number of samples of the impulse response.

When truncating the response, the shifting caused by the time of flight has to be taken into account, as the response starts with a certain delay depending on the distance of the speaker to the measuring point. A maximum distance has to be defined in order to add this to the calculated minimum impulse response length. In this case 5 meters have been chosen, which in real case scenario would allow to do measurements up to 50 meters distance. The final length in samples is calculated as:

$$N_{imp} = N_{imp'} + \frac{D_{max} f_s}{c}, \tag{16}$$

where c stands for the speed of sound, D_{max} the maximum measurable distance, and N_{imp} minimum length of the impulse response.

Once the impulse response is truncated, Fourier transform is applied again, and the magnitude frequency response and group delay are calculated. As shown in Figures 23 and 24 the responses are softened and readable. These responses will be used to compute the time delay for each way and to find the crossover point for the filters.

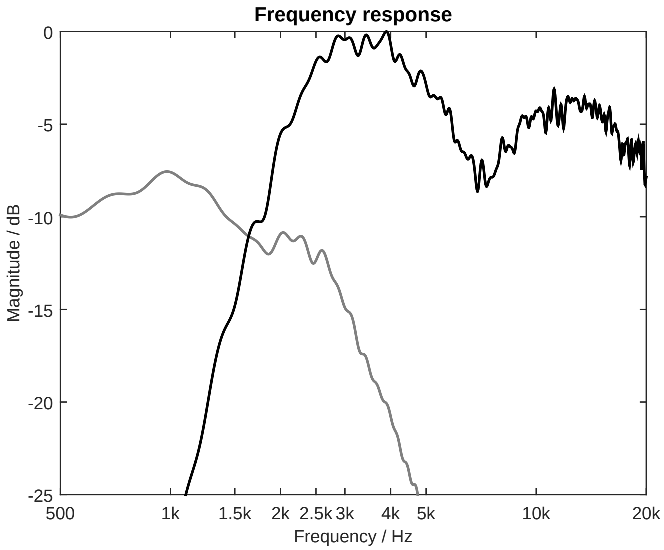


Figure 23: *Magnitude frequency response of the truncated LF and HF impulse response.*

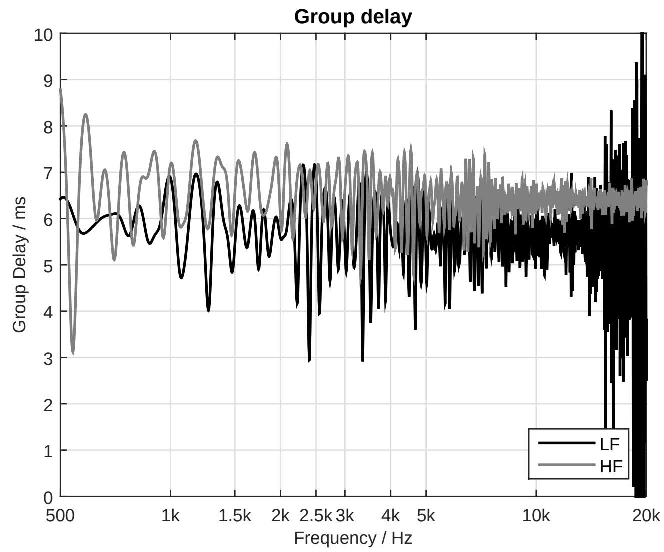


Figure 24: *Group delay curves of the truncated LF and HF impulse responses.*

5.2 Time alignment

The time alignment is the key operation to put in phase the different physically placed loudspeakers. The time delay for each way is calculated from the group delay. This gives the absolute latency caused by the time of flight of the wavefront between the loudspeaker and the microphone. This time of flight is converted to samples, and the recorded signals are truncated to align them to a zero-time point. Then both signals are time-aligned and with no latency.

To obtain the number of samples, the group delay is averaged in a certain band. This band is determined depending on the energy distribution of the signal, as the signal to noise ratio will be higher and the values for the group delay more accurate. For the LF band the averaging is between 1 and 5 kHz, and for the HF band between 2 and 10 kHz.

To achieve a fine adjustment in the signal truncation, fractional delay lines have been implemented using a linear interpolation filter [23]. The linear interpolation filter weights two consecutive samples according to the fractional part of the delay line, allowing to create new samples in fractional positions in between of those samples. This filter has a lowpass magnitude frequency response, thus it presents inconveniences when using full band signals. Therefore it has only been implemented in the LF band.

As the critical point on the delay adjustment is the relative delay between the signals more than the absolute value, when computing the delay for the LF band, the fractional part of the HF band has to be compensated. Thus, the fractional part of the HF band is subtracted from the LF band. Also some offset samples are subtracted from the calculated sample delays in both ways, as it was observed that the group delay is not giving exact values of the samples, but is giving sightly higher values. If the truncation values are higher, part of the signal is truncated, which is translated to mismatching in the relative time-position of the signals and if truncating the impulse response the peak is lost. The delay values are calculated as:

$$D_{HF'} = \frac{\sum_{f_1}^{f_2} \tau_g(f)}{f_2 - f_1} - N_{HF}, \tag{17}$$

$$D_{HF} = \lfloor D_{HF'} \rfloor, \tag{18}$$

$$D_{LF} = \frac{\sum_{f_3}^{f_4} \tau_g(f)}{f_4 - f_3} - N_{HF} - (D_{HF'} - D_{HF}), \tag{19}$$

where N stands for the offset in samples and D is the delay amount in samples. The values of f_3 and f_4 are the averaging band limits for the LF band, f_1 and f_2 are the averaging band limits for the HF band.

Way	Measured	Calculated	Δ
LF	1978	1957	-21
HF	2305	2310	5

Table 4: *Measured vs calculated distances in mm.*

In Table 4 a comparison between real measured distances with the laser distance meter, and calculated via group delay averaging is presented. The error is as low as 5 mm for high frequencies and 21 mm for low frequencies. The overall error is about 3 cm, which equals half wavelength at about 5.7 kHz (where cancellations could appear caused by phase mismatching), far beyond the interaction band of both ways, located between 1.5 and 2.5 kHz.

For obtaining the time-aligned responses, the recorded signals are truncated and de-convoluted again to obtain the impulse responses. Figure 25 shows how the compensated impulse responses have their peaks aligned and in zero-time position.

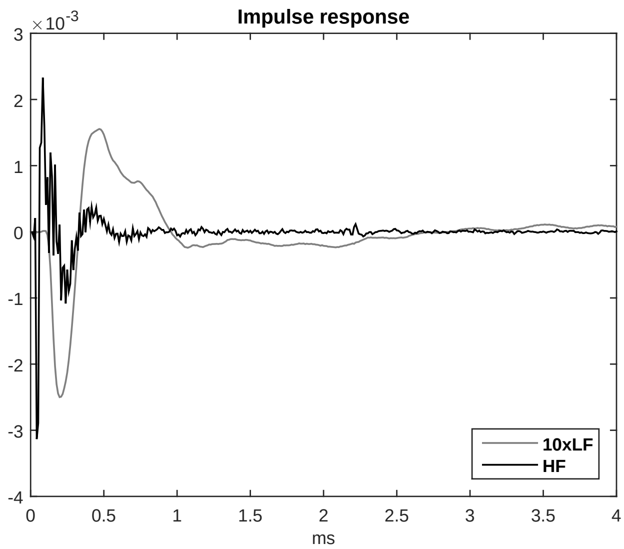


Figure 25: *Truncated and time aligned LF and HF impulses responses.*

Figure 26 shows the group delay of the time aligned signals. As it is observed, this is not exactly zero and flat. The slope is caused by the loudspeakers response, as they are not exactly linear or minimum phase. The added offset to the calculated time-delay is affecting the curves by not being exactly at zero mean value.

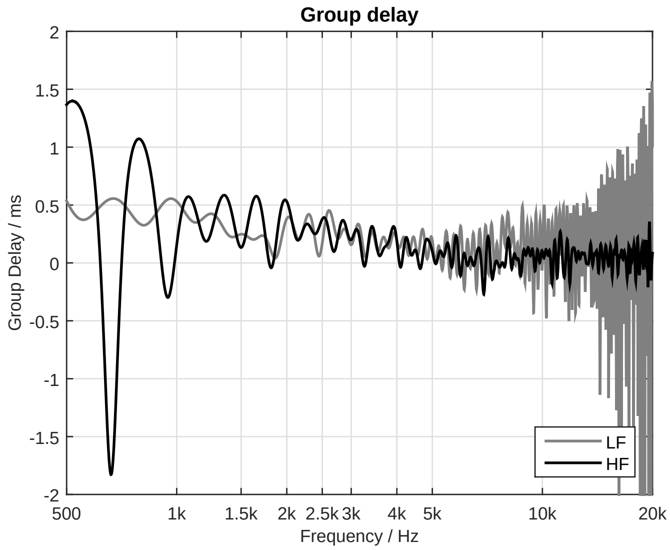


Figure 26: *Time aligned LF and HF group-delay curves.*

However, the absolute delay value is not the parameter that will be used in the signal processing chain, as it is supposed to work in real time. The adjustment of the phase is performed delaying only the closest source to the measuring position. Thus, the difference between the HF and LF calculated sample delays is used, and a signal agreement is used to determine which way has to be delayed. The LF delay is subtracted from HF delay. If the relative delay is positive means that the HF way (line array) is more distant, thus the LF way is delayed by its difference, which is the usual case (view Figure 5). If the relative delay is negative, the LF way is more distant, then the delayed element is the line array.

5.3 Choosing the crossover frequency

The automated selection of the crossover frequency is defined from the smoothed magnitude frequency responses obtained by the truncated impulse responses. For the LF and HF frequency responses the frequency point where the maximum magnitude value is presented is searched. As the responses can be very different depending on the component, and present its peaks in very distant bands, the crossover frequency band is limited to find the frequency band where the components have passed its cutoff band. In this case it is limited between 1 and 3 kHz, which would allow a crossover frequency definition in real situation from 100 to 300 Hz.

From the determined frequency peak value, a search for the -6 dB point is performed. For the LF band, the searching is performed from the peak toward higher frequencies, and for the HF band from the peak toward lower frequencies. These are estimated the cutoff frequencies for the subwoofers and the line array as shown in Figure 27. The crossover point can be determined by choosing a frequency point between the cutoff frequencies, allowing to exploit more the subwoofers using higher frequencies or the line array using lower frequencies in this band. In the model case, different options have been tested, and the middle point on the linear frequency scale has been used as the crossover frequency. In Annex A.3 the Maltab code for the procedure of finding the crossover frequency is shown.

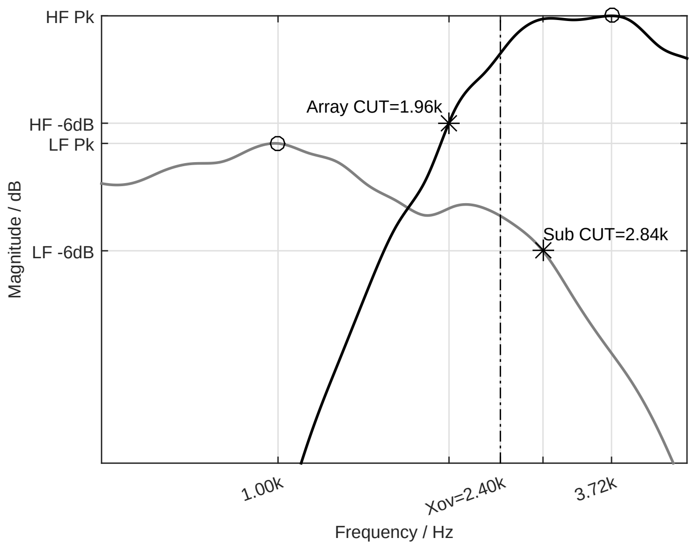


Figure 27: *Crossover frequency determination. The estimated peaks of LF and HF responses are marked with circles, and the –6-dB points are marked with asterisks. The selected crossover frequency is indicated with a vertical dash-dot line.*

At this point, the Linkwitz-Riley filters are designed and implemented. In Annex A.4 the Maltab function for implementing the filters is shown. Fourth order filters are used, chaining two second-order Butterworth filters with the chosen cutoff frequency. In Figure 28 the input frequency responses are shown as well as the filtered responses.

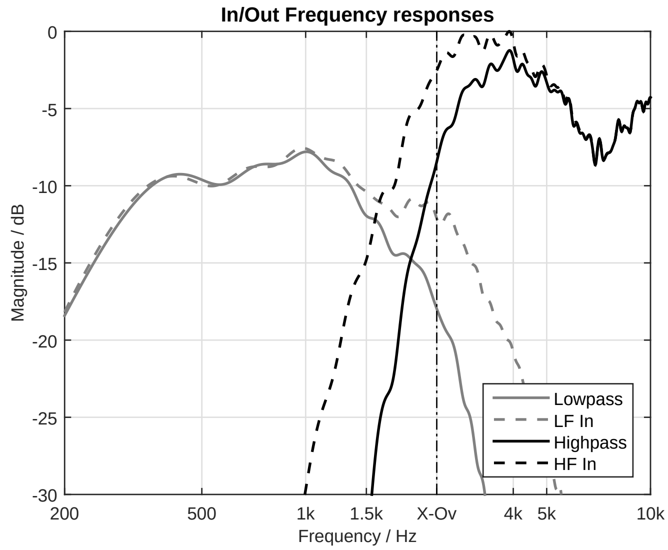


Figure 28: *Magnitude frequency responses of the crossover highpass and lowpass filtered outputs. The responses before crossover filtering are given for reference. The vertical dash-dot line indicates the crossover frequency.*

In Figure 29 it is observed that the group delay presents a very similar values in the crossover band and a low ripple in the passband. The strong ripples that are caused by the filters phase irregularities appear in the attenuated band of the HF and LF signals, thus its affection to the overall group delay is very low.

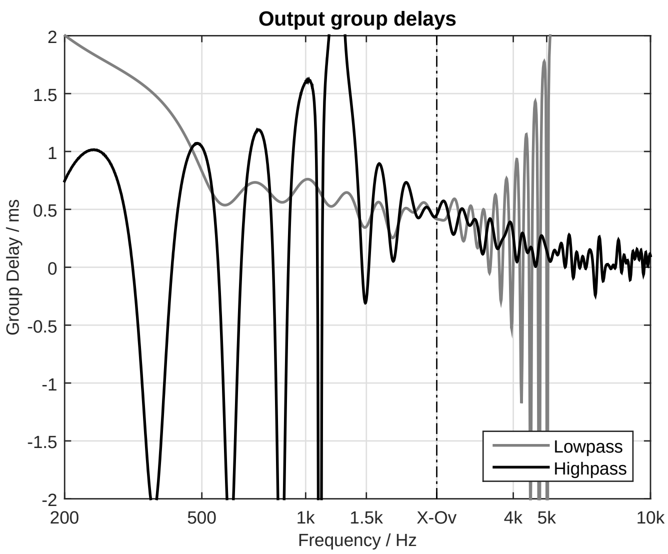


Figure 29: *Group-delay responses of the crossover highpass and lowpass filtered outputs. The vertical dash-dot line indicates the crossover frequency.*

In Figure 30 is shown that the group delay maintains an acceptable ripple in the whole band. If the responses were not correctly aligned, a strong peak in the group delay would appear around the crossover frequency.

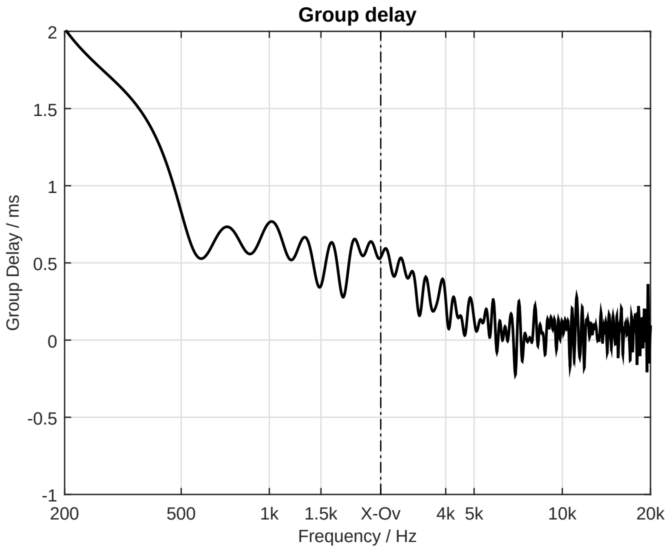


Figure 30: *Full-band group delay.*

5.4 Output gain correction

Before equalizing the whole system, an average gain is applied to the LF and HF way outputs. This is made to assure a closer-to-flat response and reach a similar mean SPL value for the full frequency range.

For this purpose, the average magnitude frequency response values are calculated for each way. The averaging is limited by the following defined frequency band limits:

- LF way: the lower frequency limit for the parameter extraction is 300 Hz, as explained in Chapter 5.1.3. As upper limit the crossover frequency is used.
- HF way: the lower limit is the crossover frequency. The upper limit is the maximum parameter extraction frequency, defined in 20 kHz.

A general target level is calculated by averaging those two values levels. The gain for each band is then obtained by dividing the target level by its band level. Once the signals are gained, they are summed to obtain a full-band signal. In this case, as observed in Figure 23, the subwoofer band has less level than the line array, thus the LF band is amplified by 3.6 dB and the HF band is attenuated by 2.5 dB. The full band signal before and after amplifying each band is shown in Figure 31.

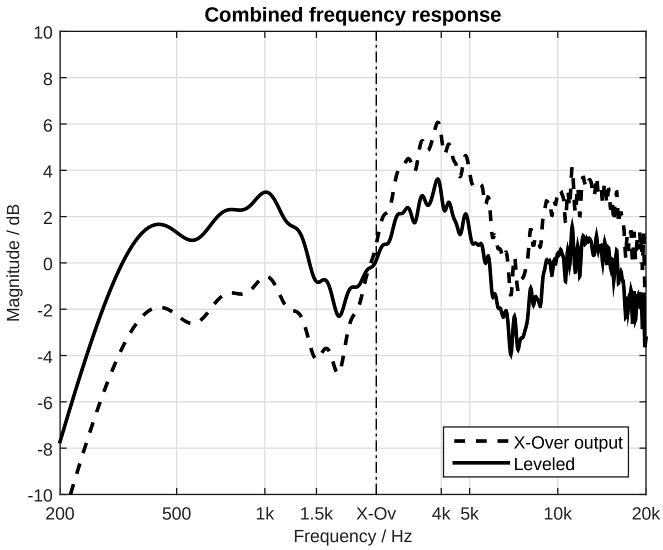


Figure 31: *Magnitude frequency responses before and after gain correction (leveling).*

5.5 Graphic equalization

The graphic equalizer used for the PA equalization is a third-octave parallel design [24]. Precise amplitude and minimum phase characteristics make it the most suitable option for the fine tuning of the system.

The full band frequency response is used to obtain the graphic equalizer gains. First the average level of the full band signal is calculated. As the −3 dB frequency limits of each third-octave band of the graphic equalizer are known, the average level of each band is calculated and the gains are obtained by dividing the average level by each band level. The 31-band frequency limits are shown in Table 5.

Band	1	2	3	4	5	6	7	8	9	10
f_L	17.5	22.4	28.2	35.5	44.7	56.2	70.8	89.1	112	141
f_H	22.4	28.2	35.5	44.7	56.2	70.8	89.1	112	141	178

Band	11	12	13	14	15	16	17	18	19	20
f_L	178	224	282	355	447	562	708	891	1120	1410
f_H	224	282	355	447	562	708	891	1120	1410	1780

Band	21	22	23	24	25	26	27	28	29	30	31
f_L	1780	2240	2820	3550	4470	5620	7080	8910	11200	14100	17800
f_H	2240	2820	3550	4470	5620	7080	8910	11200	14100	17800	22050

Table 5: *Lower and upper band limits for the 31-band parallel graphic equalizer.*

The Matlab code for the implementation of the automated gain definition is shown in Annex A.5.

As the parameter calculation is band limited between 300 Hz and 20 kHz, the frequency bands that are used in the scale model equalization are limited from the band 13 to the band 30. Figure 32 presents the obtained gains, which follow quite accurately the inverse of the magnitude frequency response observed after the output gain.

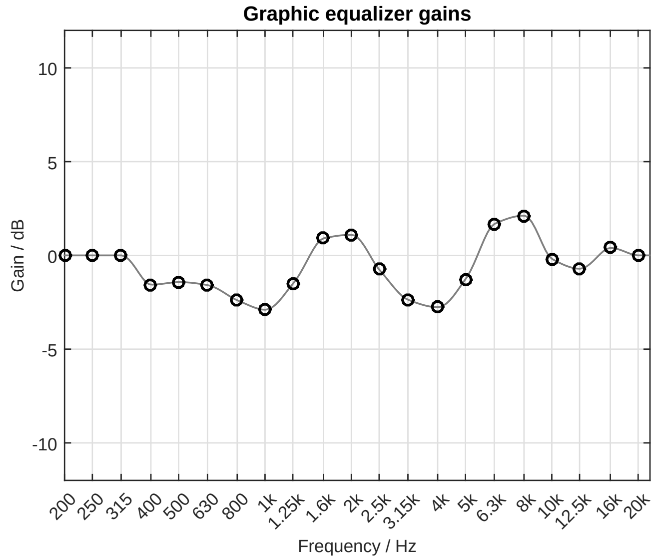


Figure 32: *Automatically calculated gains for the third-octave graphic equalizer.*

In Figure 33 the full-band magnitude frequency responses are shown before and after the last step of graphic equalization. The output has some ripple, mostly caused still by reflections, that has been affecting through all the steps of the process. However, the majority of the response is almost flat if evaluating the parameter calculation band (300 Hz to 20 kHz), and the ripple is lower than +/- 2 dB in any case.

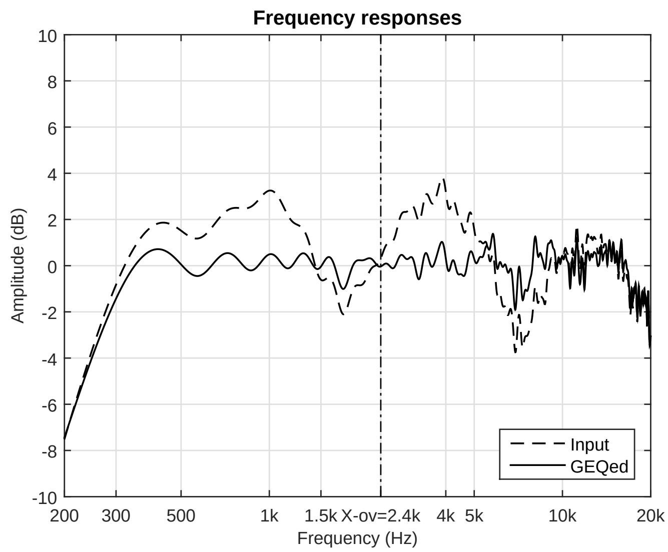


Figure 33: *Magnitude frequency response before and after graphic equalization.*

5.6 Verification sweep and ground reflection issues

In order to verify that the system has the expected time alignment and frequency response as the expected in Figure 33, a third sweep is performed. Adjusting the phase and magnitude of the sweep with the processing chain in Figure 6 using the calculated parameters, the response of the whole PA should correspond to the expected response in Figures 33. Usually unexpected responses can be obtained when reflections affect the measurements.

The four sets of calculated parameters in the previous chapters, are now used to define the signal processing chain for the system. Thus, the steps are the following:

1. Generation of the sweep signal with same characteristics as the previous ones.
2. Filtering of the time-domain signal with the graphic equalizer.
3. Applying the crossover filters to split the signal.
4. Apply the calculated gain for each way.
5. Apply the delay to one of the ways according to the signal standard explained in Chapter 5.2.
6. Play and record the full sweep and extract the responses.

Once the response is obtained, this is compared to the expected one at the output of the graphic equalizer by subtracting the magnitude frequency responses. In this case free-field responses have been measured without any reflection panels.

In Figure 34 is shown that the system follows almost exactly the the expected magnitude frequency response, and no cancellations are shown in the response.

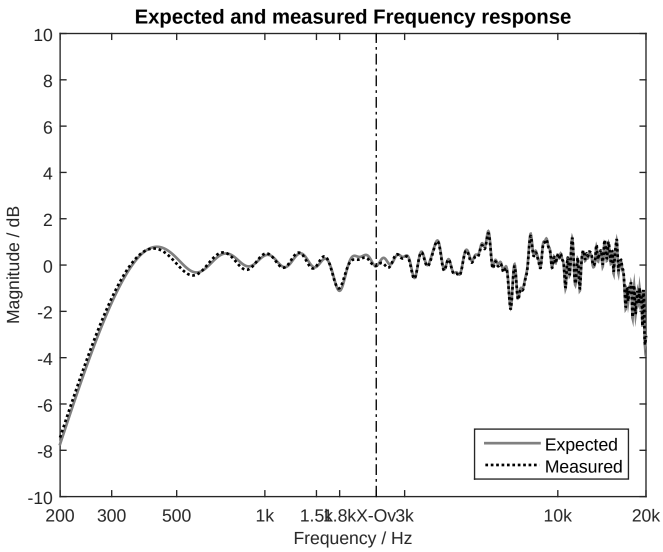


Figure 34: *Expected and measured overall magnitude response.*

Figure 35 the absence of strong peaks or irregularities at the crossover band show that the both interacting ways are time-aligned. Moreover, it is also shown that the average group delay values are below audible limits explained in Chapter 2.1.4.

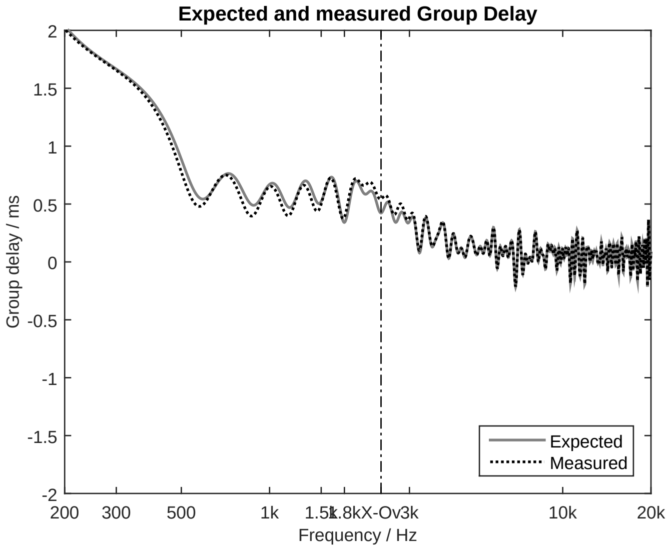


Figure 35: *(Top) Expected and measured overall group delay.*

This system is initially designed for use in outdoor venues or arenas, where the majority of reflective areas are far away from the measurement point, and the effects over the impulse response are shifted far from the direct sound impulse. However, ground reflections are a thing to have present, as the microphone is usually placed at ear height, 1 to 1.70 m depending if the audience is seated or standing. The ground reflections can affect the measurement creating a comb filter effect over the magnitude frequency response and a shifting in the group delay.

This can be seen in Figure 36, where the magnitude frequency response is shown obtained from measurement that has been performed with the wooden panels below the microphone position in the anechoic chamber. The distance of the microphone to the wooden plates is around 16 cm. As shown in Table 3 when designing the scale model, this causes an approximate path difference of 9 cm between the direct and the reflected wavefront, that is equivalent to 0.28 ms time difference. The path difference equals a half-wavelength distance for a frequency of about 1.8 kHz, where the first notch appears in the comb filter. As observed, the comb filter is only present in the HF band, as the LF band is parallel to the reflection plane, and the wavefront are guided. Therefore, all treatments for reducing the ground reflection influence will be performed only to the HF band.

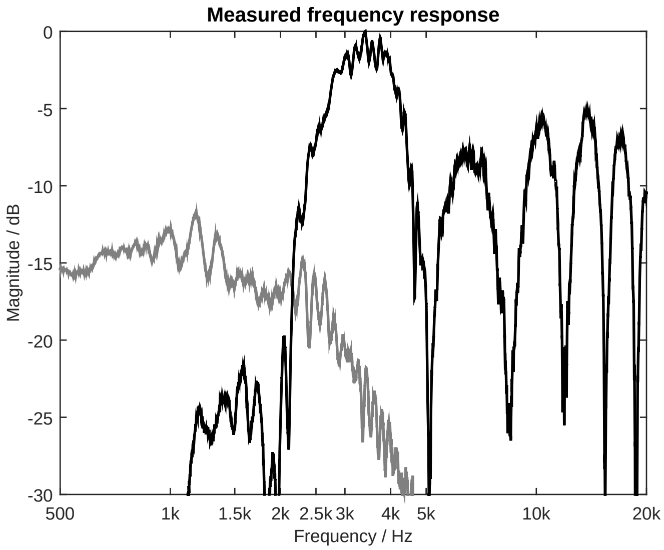


Figure 36: *Comb filter effect caused by ground reflection over the magnitude frequency response.*

The ground reflection in real case scenarios can present different intensities and responses (highpass, bandpass or lowpass) depending on the material of the ground. If focusing on live open-air or big venues, usually the ground is made of asphalt or concrete (arenas, big squares), grass or sand (sport stadiums), or even the people if the measurements are performed with audience. Concrete situation would be the most complicated, as it presents the lowest absorption coefficients (around 0.2) and has a highpass response. The wooden panels used in the scale model have a strong reflection coefficient and a highpass response, thus they are great to represent a worst case scenario.

By using the auto-correlation of the measured signal, the reflection peak can be identified. The time difference of 0.28 ms can also be seen in this case in the auto-correlation in Figure 37, where the second peak corresponding to the ground reflection is located. The relative height of the direct wavefront peak and the ground reflection gives information about the reflection intensity that the wooden panels produce. In this case, the peak reflection is around 0.5 the energy of the direct peak, which means that the reflected signal intensity is between around -3 dB below the direct sound intensity.

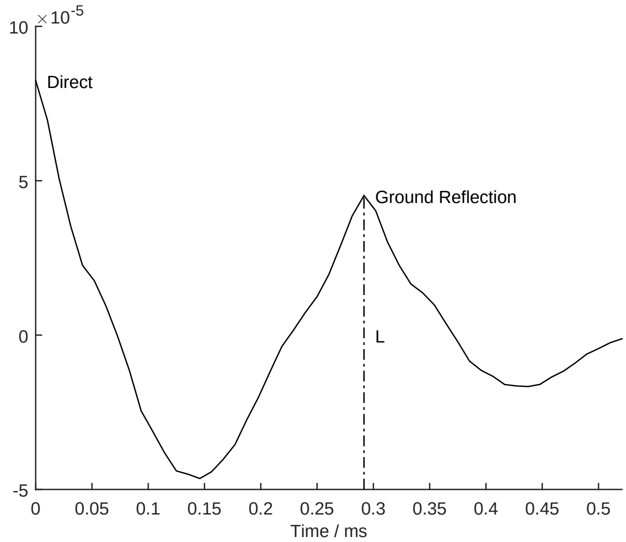


Figure 37: *Auto-correlation of input signal with ground reflection.*

To correct the comb filter effect of the reflection, a filter has been designed with an inverse response to the reflection. A first-order IIR filter, shown in Figure 38 creates a cancellation signal to the reflection. The parameters for this filter, delay L and coefficient r, are obtained from the auto-correlation of the measured signal. The distance between the zero-point peak and the second peak in the auto-correlation indicates the relative delay of the reflected signal, and therefore the delay L of the loop. The relative level between the the direct sound peak and the second peak determines the gain of the feedback loop r. This filter has been used as it is enough to obtain a proper frequency response to tune the system in Figure 39, where the magnitude frequency response of the filtered signal is shown.

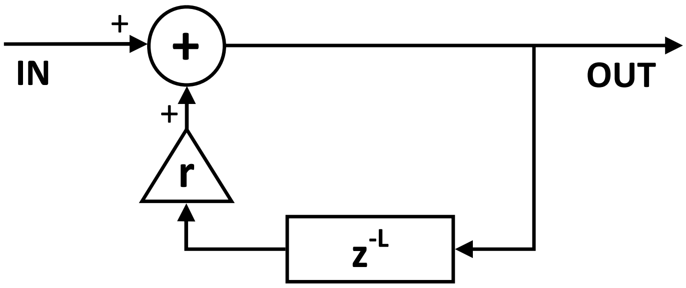


Figure 38: *Reflection cancellation filter.*

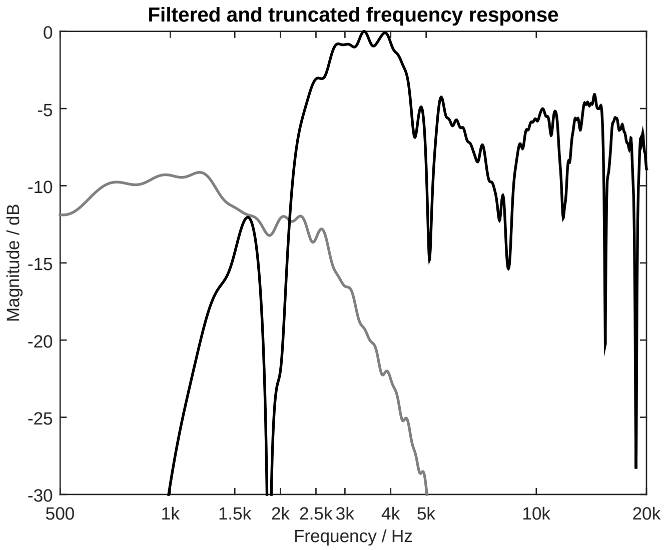


Figure 39: *Magnitude frequency response of the filtered signal with ground reflection.*

The main influence of the comb filter effect is noticed in the automated graphic equalizer. To avoid that a notch or a bump in the frequency band falsify the gains of the equalizer, the full band verification sweep is used to compare both responses. Figure 40 shows the expected and the measured magnitude frequency responses used for calculating the graphic equalizer gains. In Figure 41 the difference between the magnitude frequency responses is shown in gray. The average difference is averaged using as band limits the same as the graphic equalizer uses. If both, the difference and the equalizer gain in a certain band is higher than 2 dB, it is considered that the band gain has been falsified by the influence of reflection, thus the band remains un-equalized and the gain is set to 0 dB.

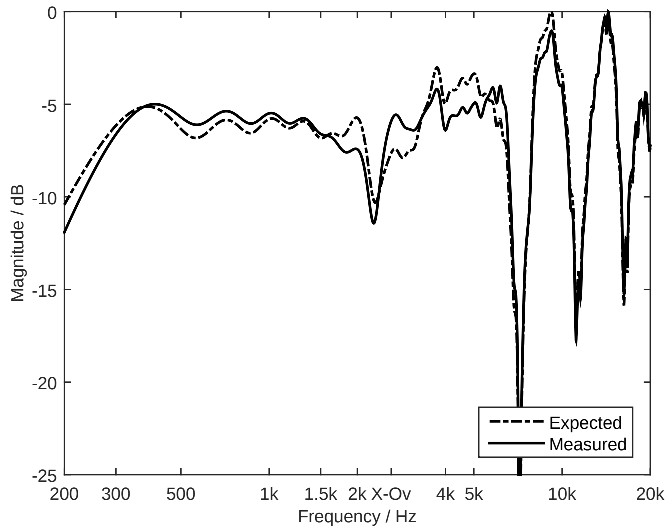


Figure 40: *Expected and measured magnitude frequency responses with ground reflection.*

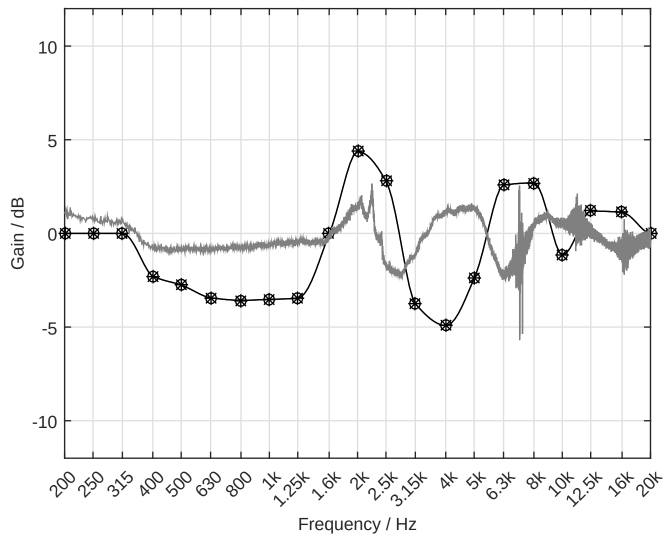


Figure 41: *In gray the difference between the expected and the measured magnitude frequency responses with ground reflection. Calculated gains for the graphic equalizer: in circles the first parameter calculation and with asterisks after the second verification sweep.*

The Matlab code for the verifying and re-calculation of the graphic equalizer gains is shown in Annex A.6.

As observed in Figure 41 the first parameter set for the equalizer (shown with circles) and the second set (presented with asterisks) are the same, meaning that the reflection filter has worked well. If it is compared to the equalization without ground reflection in Figure 32, very similar gain values are obtained. In this case no corrections to the graphic equalizer is done, as the difference signal between the expected signal and the measured signal is fairly low. In Figure 42 an example is shown when no correction filter is used or if it is not eliminating properly the ground reflection.

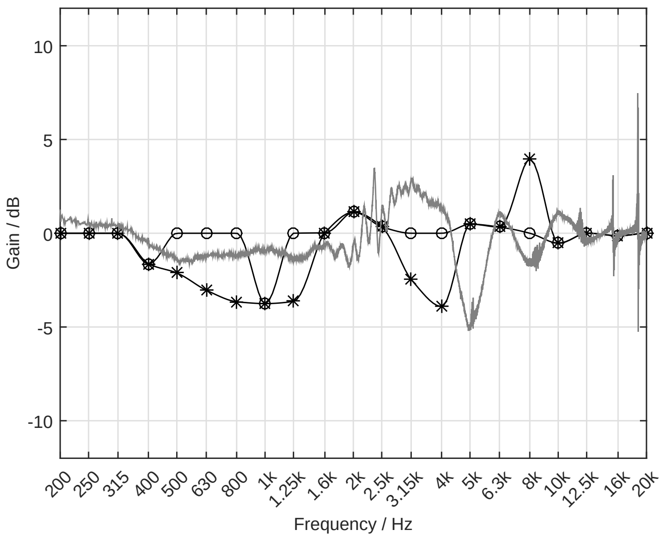


Figure 42: *In gray the difference between the expected and the measured magnitude frequency responses without reflection filter. Calculated gains for the graphic equalizer without reflection filter: in circles the first parameter calculation and with asterisks after the second verification sweep.*

6 Conclusions

An automatic calibration and equalization procedure for line-array-based PA systems has been presented in this project. The system has been designed using Matlab functions, and tested first with simulated signals and afterwards with a 1:10 scale model in an anechoic chamber.

The aim was to obtain four sets of parameters that are used in the audio signal processing chain for line array systems. Those have been successfully obtained from the subwoofer and line array responses to time align and equalize the whole system: delay time for the closest cabinets, crossover frequency, average gain for each way, and graphic equalization curve.

The results used to analyze the functioning of the designed procedures have been obtained from the scale model measurements, which have been taken place in the Aalto university School of Electrical Engineering anechoic chamber facilities.

Those measurements have been taken place in two rounds: during the first round the designed functions have been implemented for the extraction of the parameters, but errors in some parts of the parameter extraction and dis-adjustments caused by the measuring system came out. The Matlab functions were adapted to analyze the measured signals and some improvements were added, as the use of fractional delay lines and the reflection cancellation filter.

The obtained results were in the line with the expected ones: a close to flat frequency response and low group delay is obtained using phase (delay) adjustment in the crossover band. Magnitude and group delay responses were analyzed and compared to the predictions.

Also measurements with the ground reflection influence have been taken place. With those measurements the functioning of the verification mechanism and the reflection cancellation filter was tested. The calculated parameters with ground reflection were very similar to the ones without it, proving its resistance to them.

Such a system could be incorporated in PA processors making the line-array adjustment procedures a much faster and easier task than it is today. Therefore, one of the future developments of this project could be the implementation of the parameter extraction blocks in to a real time audio processing unit. This could also allow to implement a dynamic re-calculation of parameters depending on a desired response (for example, A weighing or pink-spectrum responses) or on external affections as changes in temperature or humidity.

The implemented solution is a simplified version of real case scenarios, where multiple loudspeaker clusters are involved and many times more than two ways are processed by the system crossover. Therefore, some improvements and expansions could be introduced to the system:

- Multi-channel support, for adjusting not just stereo systems, but whole PA systems involving frontfills, outfills or central channel configurations. This could be implemented adding more sweep measurements to locate the sources and create a decision mechanism that decides the optimal time delay for each cluster.

- Multiple measuring positions could be used to adjust different sections of a line array for mid, short or long throws.
- Multiple microphones could be used at a measuring position, and by using correlation of those signals improve the influence of ground reflections or even hall acoustics if used indoors.
- Subwoofer arrays could be adjusted with similar procedures and integrated in the adjusting system. Based on the acquisition of the responses and time-delay parameters. Different subwoofer arrays could be automatically configured, allowing to shape the array pattern and achieve a more efficient SPL coverage.

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A Matlab code

A.1 HF and LF response simulation

```
function [xmlf,xmhf] = get_measurement(xref,dly,nn,rm,fs)
path_difference = 0.9;
speed_of_sound = 346;
%Gaussian noise added to the responses
gauss = wgn(1,length(xref),nn);

% LF speakers response modelling
sublowcut = 25;
subhighcut = 250;

dl1 = fdesign.lowpass('N,F3dB',3,subhighcut,fs);
Hdl1 = design(dl1,'butter');

dl2 = fdesign.highpass('N,F3dB',2,sublowcut,fs);
Hdl2 = design(dl2,'butter');

xlow = filter(Hdl1,filter(Hdl2,xref));

% HP butterworth to modelate the High frequency speakers response
arraylowcut = 200;
dh1 = fdesign.highpass('N,F3dB',4,arraylowcut,fs);
Hdh1 = design(dh1,'butter');

xhi = filter(Hdh1,xref);

% 1st ground reflection signal:
r_dly = round((path_difference/speed_of_sound)*fs);
reflowcut = 600;
dr = fdesign.highpass('N,F3dB',1,reflowcut,fs);
Hdr = design(dr,'butter');
f_xhi = filter (Hdr,xhi);
xhi_r = -rm*f_xhi; % Inverted phase signal with relative amplitude rm

rref_hi = horzcat(zeros(1,r_dly),xhi_r(1:end-r_dly));

% Definition of HF and LF modeled signals:
xmlf = gauss+xlow;
xmhf = gauss+xhi+rref_hi;
if dly~=0
    dly_sig = wgn(1,abs(dly),nn); % Delay signal vector
    if dly<0
        xmlf = horzcat(dly_sig,xmlf(1:end+dly));
    else
        xmhf = horzcat(dly_sig,xmhf(1:end-dly));
    end
end
end
```

A.2 Full band response simulation

```
function xmfull = get_measurement2(low2,high2,acq_t,d,nn,rm,fs)
path_difference = 0.9;
speed_of_sound = 346;
%Gaussian noise added to the responses
gauss = wgn(1,length(low2),nn);

% LF speakers response modelling
sublowcut = 25;
subhighcut = 250;

dl1 = fdesign.lowpass('N,F3dB',3,subhighcut,fs);
Hdl1 = design(dl1,'butter');

dl2 = fdesign.highpass('N,F3dB',2,sublowcut,fs);
Hdl2 = design(dl2,'butter');

xlow = filter(Hdl1,filter(Hdl2,low2));

% HP butterworth to modelate the High frequency speakers response
arraylowcut = 200;
dh1 = fdesign.highpass('N,F3dB',4,arraylowcut,fs);
Hdh1 = design(dh1,'butter');

xhi = filter(Hdh1,xref);

% 1st ground reflection signal:
r_dly = round((path_difference/speed_of_sound)*fs);
reflowcut = 600;
dr = fdesign.highpass('N,F3dB',1,reflowcut,fs);
Hdr = design(dr,'butter');
f_xhi = filter (Hdr,xhi);
xhi_r = -rm*f_xhi; % Inverted phase signal with relative amplitude rm

% Definition of HF and LF modeled signals:

if d == 0
   xmlf2 = gauss+xlow;
   xmhf2 = gauss+xhi+rref_hi;
else
   dly_sig = wgn(1,abs(d),nn); % Delay signal vector
   if d<0
       aux = gauss+xlow;
       xmlf2 = horzcat(dly_sig,aux(1:end-abs(d)));
       xmhf2 = gauss+xhi+rref_hi;
   else
       aux = gauss+xhi;
       xmlf2 = gauss+xlow+rref_low;
       xmhf2 = horzcat(dly_sig,aux(1:end-d))+rref_hi;
   end
end
```

```
xmfull = xmlf2+xmhf2;
end
```

A.3 Crossover frequency definition

```
% 1/3 octave softening:
[LF_Freqresp_avg,freq] = octaveaveraging (LF_Fr,3,fs);
[HF_Freqresp_avg,~] = octaveaveraging (HF_Fr,3,fs);
% Search for Max points in the FFT's
    % Define frequency range 1-3 kHz
    xfmin = round(1000*(Nfft/fs));
    xfmax = round(3000*(Nfft/fs));
    % Max points for each way
    [LW,xl] = max(LF_Freqresp_avg(xfmin:xfmax));
    [LH,xh] = max(HF_Freqresp_avg(xfmin:xfmax));
% Search -6dB point
    % In LF -6dB is searched from the point forward in frequency:
    [~,f_low] = min(abs(LF_Freqresp_avg((xl+xfmin):xfmax)-(LW-6)));
    f_low = (f_low+xl+xfmin)*(fs/Nfft);
    % In HF -6dB is searched from the point backward in frequency:
    [~,f_high] = min(abs(HF_Freqresp_avg(xfmin:(xh+xfmin))-(LH-6)));
    f_high = (f_high+xfmin)*(fs/Nfft);
% Decision of cutoff point
    % Linear midpoint
    f_xover = round((f_low+f_high)/2);
```

A.4 Linkwitz-Riley filters implementation

```
function [yLP,yHP] = Xover_filtering2 (x,order,f_xover,fs)
% Coefficients of the lowpass and higpass Butterworth
wc = 2*f_xover/fs;
if mod(order,2)==0
    order = order/2;
end
% setup the Butterworth filters
[bL,aL]=butter(order,wc);
[bH,aH]=butter(order,wc,'high');
%Filtering
if mod(order,2) == 0     % Even order LR: Cascade of two butter filters
    yLP0 = filter(bL,aL,x);
    yHP0 = filter(bH,aH,x);
    yLP =  filter(bL,aL,yLP0);
    yHP =  filter(bH,aH,yHP0);
else                     % Odd order LR:  One butter filter
    yLP = filter(bL,aL,x);
    yHP = filter(bH,aH,x);
end
end
```

A.5 Graphic equalizer gains definition

```
function [gains] = GEQ_gains (spectrum,fmin,fmax,f_lim)
n_bands = length(f_lim)−1;  % f_lim contains the −3dB frequency limits
Pf = abs(spectrum);         % magnitude of the spectrum is used
m = mean (Pf(fmin:fmax));   % mean full−band value
gains = zeros (1,n_bands);  % gains initially set to 0
for i=1:n_bands
    if f_lim(i)<fmin        % if the band is out of frequency limits
        gains(i) = 1;       % it remains unequalized (unity gain)
    elseif f_lim(i+1)>fmax
        gains(i) = 1;
    else
        n1 = ceil(f_lim(i));
        n2 = ceil(f_lim(i+1));
        Lwin_i = n2−n1;
        s = sum(Pf(n1:n2))/Lwin_i;  % mean value of the band
        gains(i) = m/s;             % gain mean signal value / mean band
    end
end
end
```

A.6 Graphic equalizer gains verification and readjustment

```
function [eqgain2,m] = GEQ_gains2 (eqgain,dif,f_lim)
dif = abs(dif);
n_bands = length (eqgain);
eqgain2 = zeros (1,n_bands);   % gains initially set to 0dB
m=zeros(1,n_bands);
for i=1:n_bands
    % mean value of the difference signal in the band
    m(round(i)) = mean(dif(f_lim(i):f_lim(i+1)));
    if abs(eqgain(i))>1 && abs(m(i))>2   % if it is > 2dB
            eqgain2(i)=0;
    else
        eqgain2(round(i))=eqgain(round(i)); % otherwise the gain remains
    end
end
end
```

B SEAS 19TAF/G specifications

SEAS PRESTIGE

19TAF/G
H0414

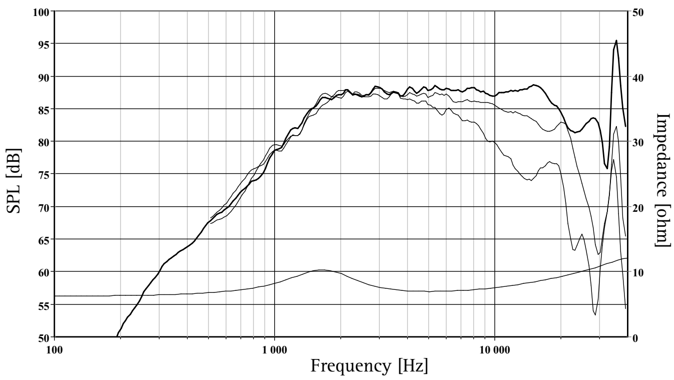
19mm High Fidelity dome tweeter with smooth, extended frequency response.

Aluminium diaphragm suspended by a specially designed soft Sonotex surround which allows a low fundamental frequency.

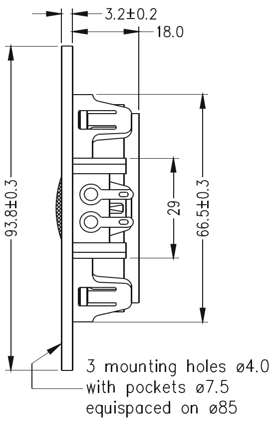
Injection moulded chassis in high grade glass fibre reinforced plastic keeps the moving parts in perfect alignment and provides good coupling to the cabinet.

Fine mesh grid protects the diaphragm and carries a phase plate which compensates for a slight axial roll off towards 20 kHz.

Voice coil windings immersed in magnetic fluid increase short term power handling capacity and reduce the compression at high power levels.



The frequency responses above show measured free field sound pressure in 0, 30, and 60 degrees, mounted in a 0.6m by 0.8m baffle. Input 2.83 Vrms, microphone distance 0.5m, normalized to SPL 1m. The impedance is measured without baffle using a 2V sine signal.



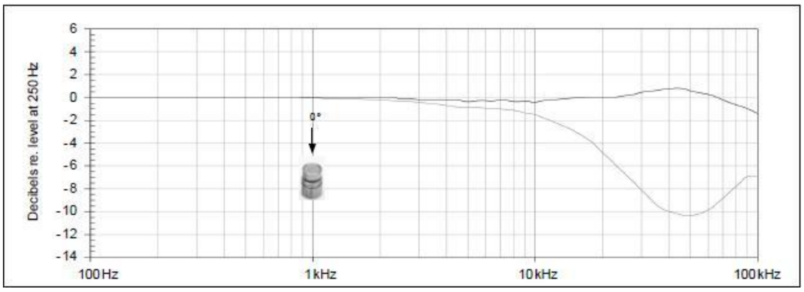
Nominal Impedance	8 Ohms	Voice Coil Resistance	6.2 Ohms
Recommended Frequency Range	4000 - 20000Hz	Voice Coil Inductance	0.05 mH
Short Term Power Handling *	220 W	Force Factor	2.6 N/A
Long Term Power Handling *	90 W	Free Air Resonance	1700 Hz
Characteristic Sensitivity (2.83V, 1m)	88 dB	Moving Mass	0.23 g
Voice Coil Diameter	19.5 mm	Effective Piston Area	4.0 cm²
Voice Coil Height	1.5 mm	Magnetic Gap Flux Density	1.4 T
Air Gap Height	2.0 mm	Magnet Weight	0.12 kg
Linear Coil Travel (p-p)	0.5 mm	Total Weight	0.30 kg

Jul 2007-1 *IEC 268-5, via High Pass Butterworth Filter 2500Hz 12 dB/oct. T19-102

SEAS reserves the right to change technical data

C Gras 46bf specifications

Parameter	Unit	Value
Frequency range (±1 dB)	Hz	10 to 40 k
Frequency range (±2 dB)	Hz	4 to 100 k
Dynamic range lower limit with G.R.A.S. preamplifier	dB(A)	35
Dynamic range upper limit with G.R.A.S. preamplifier @ +28 V / ±14 V power supply	dB	163
Dynamic range upper limit with G.R.A.S. preamplifier @ +120 V / ±60 V power supply	dB	172
Set sensitivity @ 250 Hz (±3 dB)	mV/Pa	3.6
Set sensitivity @ 250 Hz (±3 dB)	dB re 1V/Pa	-49
Output impedance	Ω	75
Power supply min. to max. (single/balanced)	V	28 to 120 / ± 14 to ± 60
DC-offset, min., single suppy	V	0.5 x Vs - 1
DC-offset, max., single suppy	V	0.5 x Vs + 4
DC-offset, balanced supply	V	-1 to 4
Microphone venting		Rear
IEC 61094-4 Compliance		WS3F
Temperature range, operation	°C / °F	-30 to 70 / -22 to 158
Temperature range, storage	°C / °F	-40 to 85 / -40 to 185
Temperature coefficient @250 Hz	dB/°C / dB/°F	-0.01 / -0.006
Static pressure coefficient @250 Hz	dB/kPa	-0.02
Humidity range non condensing	% RH	0 to 100
Humidity coefficient @250 Hz	dB/% RH	-0.0013
Influence of axial vibration @1 m/s²	dB re 20 µPa	60
TEDS UTID (IEEE 1451.4)		27 v. 1.0
Connector type		7-pin LEMO (FGG.1B.307)
CE/RoHS compliant/WEEE registered		Yes / Yes/Yes
Weight	g / oz	8 / 0.282



Typical frequency response (without protection grid). Upper curve shows free-field