AUTOMATIC CALIBRATION AND EQUALIZATION OF A LINE ARRAY SYSTEM

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ABSTRACT

This paper presents an automated Public Address processing unit, using delay and magnitude 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 three signal processing operations to the audio signal: time delay adjustment, crossover filtering, 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 through the sound system. Measurements have been conducted in an anechoic chamber using a 1:10 scale model of a line array system to verify the functioning of the automatic calibration and equalization methods.

1. INTRODUCTION

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. Live audio systems have become more complex due to the computer-controlled digital signal processing part and the acoustical improvements using innovative loudspeaker systems.

The vast majority of concert Public Address (PA) systems used nowadays consist of hanging line arrays, which help reduce acoustic shadows and increase the distance of the line source effect [1, 2]. 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.

These loudspeaker configurations requires various signal processing techniques in order to achieve a flat and coherent response. Basically the calibration consists of three operations: apply a crossover to split the audio band between the subwoofers and the line array, adjust a time delay [3] and equalize the complete system to achieve a flat magnitude response.

Figure 1 shows a two-dimensional diagram with approximate values of the relative delay problem. Different arrival times of the wavefront from the subwoofers and the line array speakers, which are physically at different positions provoke phase shading, strongly noticed in the crossover band. 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 our work, just one measuring point has been taken in to account. The target area is usually placed at the Front of House position [4].

A similar phenomenon occurs between the loudspeaker components in the line array, which can be treated with different DSP procedures to compensate those problems [5], but the relative position between the array elements is limited. Lots of improvements using DSP and Wave Field Synthesis have been done the last years to control the response and directivity of line arrays [6, 7] but the complete system tuning has not had such attention. In the case of the subwoofers and the line array, those can be placed at very different positions and configurations depending on the venue, the characteristics of the stage, and the space where it takes place.

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 [8], or need additional tools to integrate it with the system processor. 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 [9] pink-spectrum noise through the PA, which is disturbing for the audience and time consuming. For this reason it has to be adjusted and configured hours prior to the venue. The sound engineer adjusts then the crossover frequency and filter type, the delay of the different loudspeakers and the graphic equalizer supported by a spectrum and phase analyzer. Some examples are EASERA [10] and Smaart [11].

In this paper we explain a method to automatically perform the adjustment of those three operations playing three sweep signals through the PA. This avoids all the manual procedure, and it can be done even with audience, as spreading the spectral energy along time and makes it less noticeable and disturbing than pseudorandom noise, being therefore much more time and effort

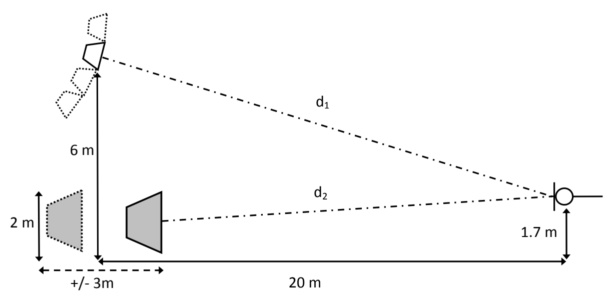

Figure 1: *Relative distances to measuring point.*

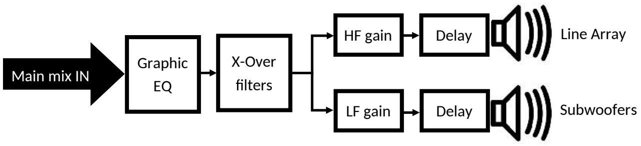

Figure 2: *One channel of the signal processing chain for a PA.*

efficient.

In order to evaluate the system, a 1:10 factor scale model was tested in an anechoic chamber. Thus, the distances were divided by 10, and the frequencies were multiplied by 10. To be able to work with standard audio material and avoid problems due to increased air absorption and distortions in ultrasound band, the system has been tested with frequencies up to 20 kHz in the scale model. This limitation is affordable as the main phase, crossover and equalization issues happen in the crossover and middle band. Therefore the results shown in the paper have to be analyzed as 10 times scaled to a real model and band limited to 2 kHz in a real application.

This paper is organized as follows. In Section 2, the calibration system and the test setup are explained. Section 3 describes the details of signal processing operations. Section 4 discusses the ground reflection problem and how it can be treated. Conclusions and future applications are explained in Section 5.

2. SYSTEM OVERVIEW

In this section a brief explanation of the different steps in the signal processing and parameter calculation is given. In Section 2.1 an introduction to the actual processing operations applied to the audio signal is made. In Sections 2.2 and 2.3 the different operations are explained in order to obtain automatically the parameters.

Three sweeps will be used in total. The first two sweeps collect separately the responses of the subwoofers and the line array to calculate the different audio processing chain parameters. The third sweep runs trough the designed processing chain and sounds through the whole PA for verifying and re-adjusting some parameters if necessary.

2.1. Audio processing chain

The automated processing unit is designed following the typical processing chain for PA systems shown in Figure 2. The main output signal of the mixing desk is conducted through a graphic equalizer, 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.

The output of the graphic equalizer feeds the PA processing unit. The first step is applying the crossover filters to divide the audio band into the different sub-bands (or ways) for each speaker group or driver into a specific speaker. In this case two bands are used: mid and high frequencies (HF) to feed the line array and low frequencies (LF) to feed the subwoofers. Different filter types are used for this purpose, but the most widespread type for digital crossover filters are Linkwitz-Riley filters, as they obtain a flat pass-band response and zero phase difference in the crossover frequency [12].

Once the signal has been split into two bands, individual processing for the low frequency and high frequency cabinets can be applied. At this point, the delay is applied in order to acoustically align the speakers at a precise point, and to achieve a coherent response in the crossover band. Also an overall gain is applied to each output.

2.2. Target responses

As in this case the desired output (flat response and phase coherent) is known, in order to be able to generate an adequate target response for each of the processing steps, the reverse procedure has been followed. A block diagram is shown in Figure 3.

To obtain a distortion-free response of the different loudspeaker ways [13], two logarithmic sweeps have been used to obtain separately the responses of the subwoofers and the line array. The impulse responses are obtained using de-convolution and using the second half part, which corresponds to the linear part [14].

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.

2.3. Verification and re-adjusting

In order to verify the suitability of the calculated parameters and correct errors, caused mostly by ground reflections, a third full sweep though the whole PA (line array and subwoofers) is played. A block diagram is shown in Figure 4.

The parameters calculated in Section 2.2 are used to design a signal processing chain as the one explained in Section 2.1. The same sweep signal as the used in the first step is fed as input signal.

The response of the full measurement is compared to the expected output calculated at the output of the graphic equalizer. The difference signal between these two sweeps is used to readjust the graphic equalizer if needed.

The difference between these signals is mostly caused by reflections in the measurement which could not been canceled with pre-processing of the responses. In this case, if the graphic equalizer tries to equalize out the generated comb filter, the affected band is set to nominal value, leaving this band un-equalized.

2.4. Scale model

A 1:10 scale model of the PA system has been implemented in an anechoic chamber. Different speaker configurations have been tested in addition to filters implemented in the sound interface to obtain a similar scaled response of a real PA system.

In Figure 5 the setup in the anechoic chamber is shown. The microphone is a quarter inch free field pressure microphone. A structure holds a tweeter emulating an array element and a sphere loudspeaker with a 5 inch cone emulating the subwoofer. The

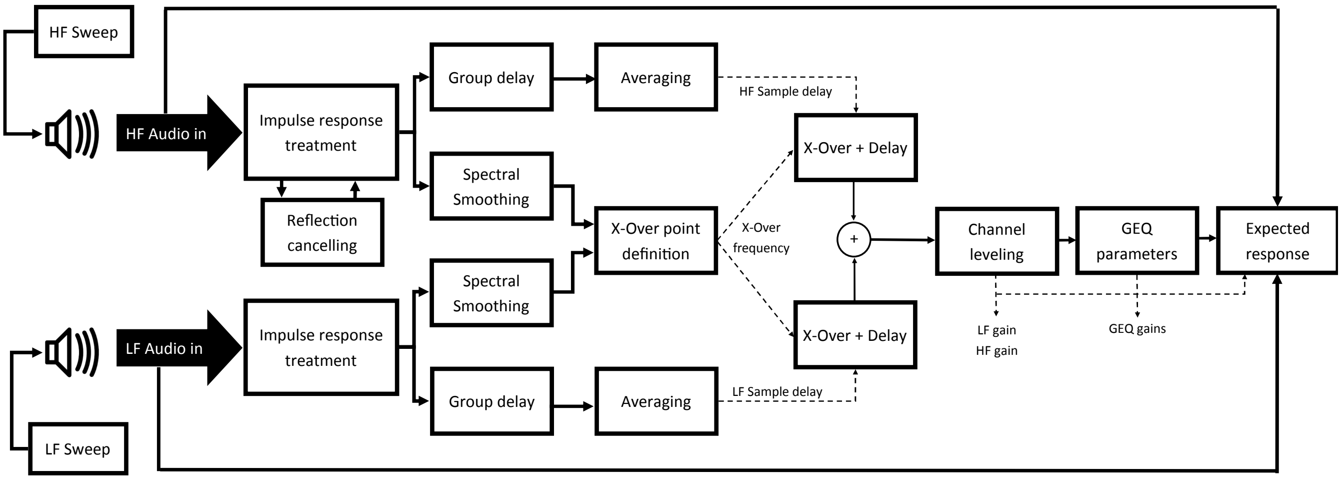

Figure 3: *Parameter calculation block diagram.*

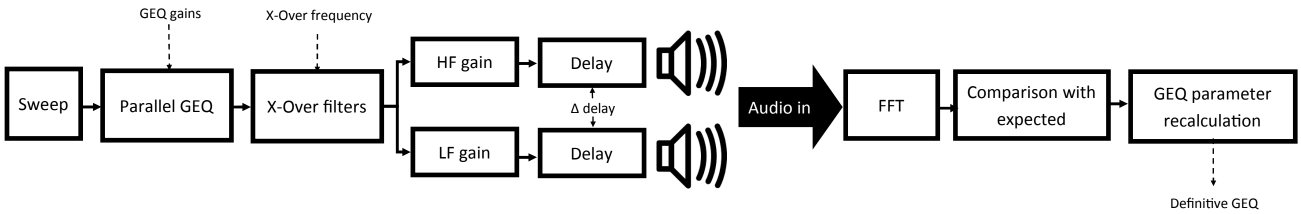

Figure 4: *Verification and re-adjusting block diagram.*

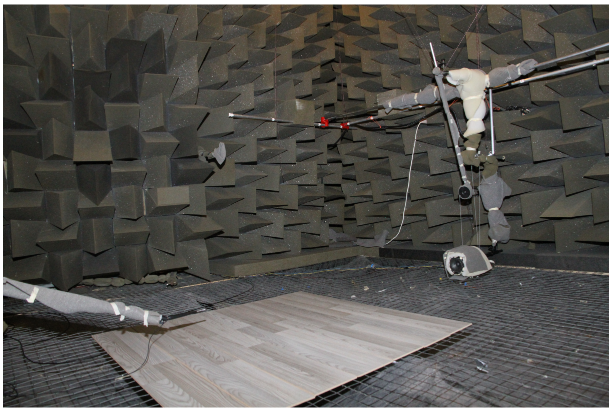

Figure 5: *Image of the scale model in the anechoic room. The bass and treble speakers are seen on the right, the microphone on the left, and a reflection plate below it.*

structure allows to move freely the high and low-frequency speakers to arrange different setups. In order to test the influence of the ground reflection, laminated wooden plates have been used.

The prediction is band limited from 300 Hz to 20 kHz, which equals a real case scenario from 30 Hz to 2 kHz. The sweeps have been performed from 200 Hz to 40 kHz during 3 seconds, as the used HF speaker has a reasnoable flat response up to this frequency, and could be equalized up to this band. A 96-kHz sampling frequency has been used for the anechoic measurements.

The fact of testing the system with a 1:10 scale model implies that the precision of the time related measures has to be also 10 times higher. Both measurements, with and without ground reflection, have been taken place.

The wooden panels used in the scale model exaggerate the intensity of a ground reflection in real case scenarios, with an average absorption coefficient around 0.4, which means that the reflected signal intensity is between -2 and -3 dB below the direct sound intensity.

3. CALIBRATION AND EQUALIZATION METHODS

In this section a detailed explanation of the procedures for obtaining the parameters explained in Section 2.2 is given.

3.1. Impulse response acquisition

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.

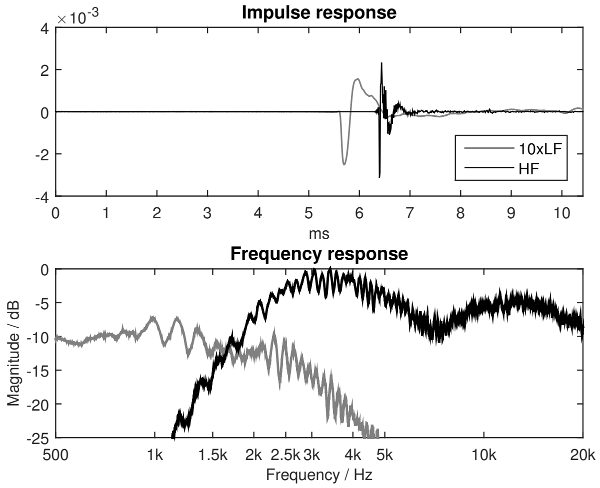

Figure 6: *(Top) Measured LF and HF impulse responses and (bottom) the corresponding magnitude frequency responses.*

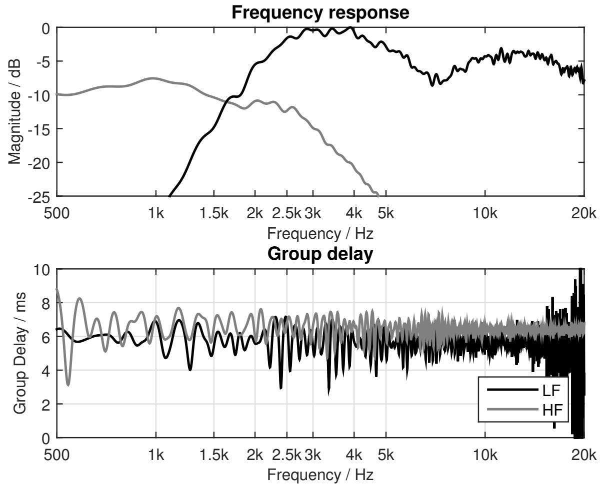

Figure 8: *(Top) Magnitude and (bottom) group-delay curves of the truncated LF and HF impulse responses.*

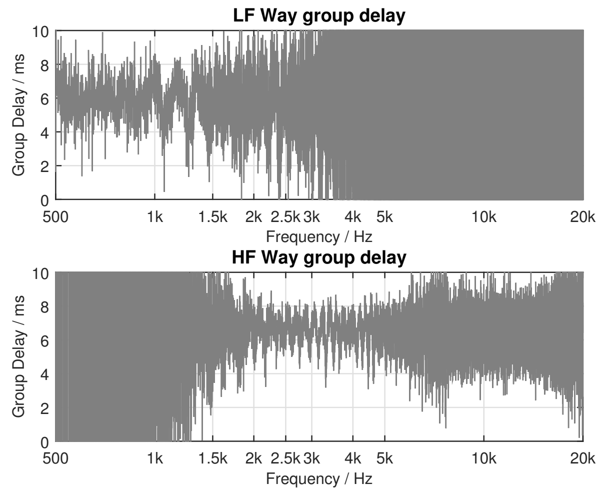

Figure 7: *Measured group delays without further processing.*

An initial calibration of the system is made. The group delay added by the sound cards and signal flow from the measurement system is measured using a feedback loop. Afterwards it will be subtracted from the measured group delay.

The different shifting from the 0 time point in each response shown in Figure 6 is caused by the time the wavefront travels from the speaker to the microphone. Thus, the different arrival times are caused by the different distances to the measuring point, as shown in Figure 1.

The different distances have also influence in the group delay. The measured group delay is very noisy as shown in Figure 7. 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 minimum frequency in this case has been chosen 250 Hz, in order to have a margin below the 300 Hz where the prediction mechanism starts.

When truncating the response, the delay 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:

$$N_{imp} = \frac{f_s}{f_{min}} + \frac{D_{max} f_s}{c}, \tag{1}$$

where f_s stands for the sampling frequency, c for the speed of sound, f_{min} is the minimum representable frequency, D_{max} is the maximum measurable distance, and N_{imp} is the number of samples of the impulse response.

In Figure 8 the improvement on the group delay and the frequency response is shown. These responses will be used to compute the time delay for each way and to find the crossover point for the filters.

3.2. Time alignment

The time delay is calculated from the group delay. To compensate the measuring system delay, caused by filters and A/D converters, it is subtracted from the measured response in a calibration process;

$$\tau_g(f) = -\frac{d\phi_{meas}(f)}{df} + \frac{d\phi_{calib}(f)}{df}, \tag{2}$$

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

In order to get accurate values of the time delay, the group delay of each way is averaged in the frequency range where the energy is located.

To achieve fine tuning in the calculated delay, fractional delay lines have been implemented using a linear interpolation filter [15]. As the linear interpolator, which is a two-tap FIR filter, has a lowpass magnitude response, it has only been implemented in the LF band. The fractional part of the HF band is subtracted from the LF band to compensate the relative delay. Also some offset samples are subtracted from the calculated time delay in order not to truncate the impulse response peak. The delay values are calculated as:

$$D_{HF\prime} = \frac{\sum_{f_1}^{f_2} \tau_g(f)}{f_2 - f_1} - N_{HF}, \tag{3}$$

$$D_{HF} = \lfloor D_{HF\prime} \rfloor, \tag{4}$$

$$D_{LF} = \frac{\sum_{f_3}^{f_4} \tau_g(f)}{f_4 - f_3} - N_{HF} - (D_{HF\prime} - D_{HF}), \tag{5}$$

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 1 and 5 kHz for LF and f_1 and f_2 are 2 and 10 kHz in the HF band. These limits are chosen as the signals contain its energy this range, thus the group delay has valid values. In Table 1 a comparison between real measured distances 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, far beyond the interaction band of both ways.

Once the delay values have been determined, the signals are truncated according to the group delay adjusting them to the zero point. In Figure 9 the compensated impulse responses and the group delay are shown. The group delay is not exactly zero and flat. The slope is caused by the loudspeakers, as they are not exactly linear or minimum phase.

3.3. Choosing the crossover frequency

The crossover frequency is defined from the smoothed frequency responses obtained by the truncated impulse responses. For the LF and HF frequency responses the peak frequency and its magnitude value is searched. From the peak frequency a search for the -6 dB point is performed. For the LF band, the search is made 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 10. A limited crossover frequency band is determined, to avoid that peaks in higher areas of the HF frequency response cause shifting in the calulated crossover frequency (in the scale model between 1 and 4 kHz).

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.

Table 1: *Measured vs calculated distances in mm.*

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

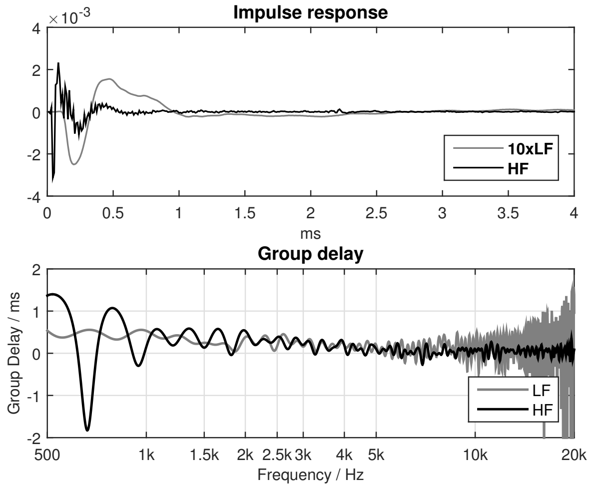

Figure 9: *(Top) Truncated and time aligned LF and HF impulses responses and (bottom) their group-delay curves.*

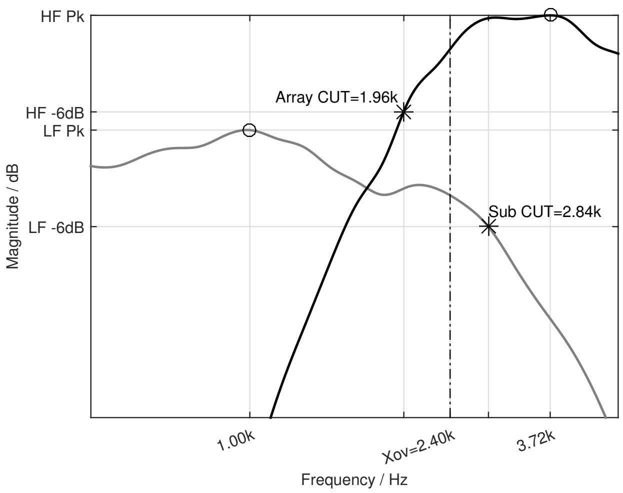

Figure 10: *Crossover frequency determination. The estimated peaks of LF and HF responses are marked with circles, and the –6-dB points are marked with asterisks. The selected crossover frequency is indicated with a vertical dash-dot line.*

At this point, the Linkwitz-Riley filters are designed and implemented. Two second-order Butterworth filters with the chosen cutoff frequency are chained. In Figure 11 the input frequency responses are shown as well as the filtered responses. Also it is observed that the group delays of the filtered signals are very close to each other in the crossover band.

Next an average gain is applied to each band before obtaining the full band signal. The average level values are calculated for each way. Then a general target level is calculated by averaging

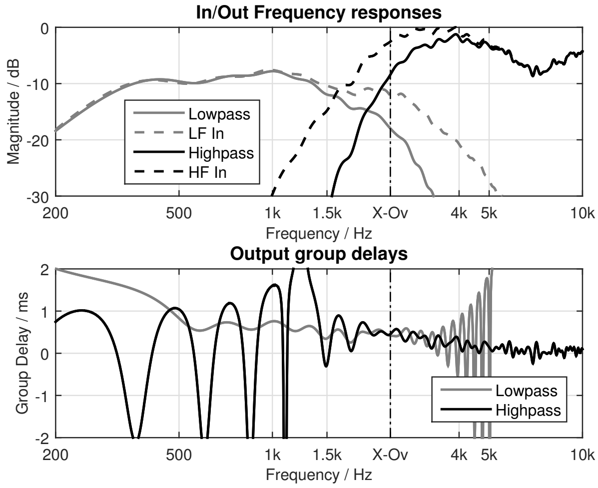

Figure 11: *(Top) Magnitude and (bottom) group-delay responses of the crossover highpass and lowpass filtered outputs. In the top plot, the responses before crossover filtering are given for reference. The vertical dash-dot line indicates the crossover frequency in this and in the following three figures.*

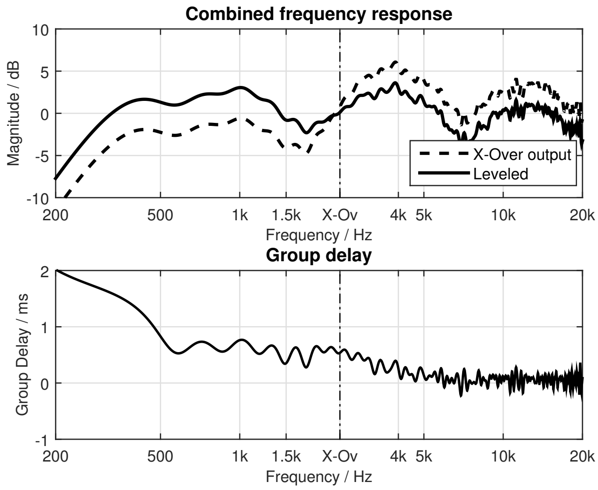

Figure 12: *(Top) Magnitude responses before and after gain correction (leveling) and (bottom) the full-band group delay.*

those two values levels. The gain is easily obtained by dividing the target level by its calculated level. Once the signals are gained, they are summed to obtain a full band signal.

As observed in Figure 6, 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 12 It is also observed 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.

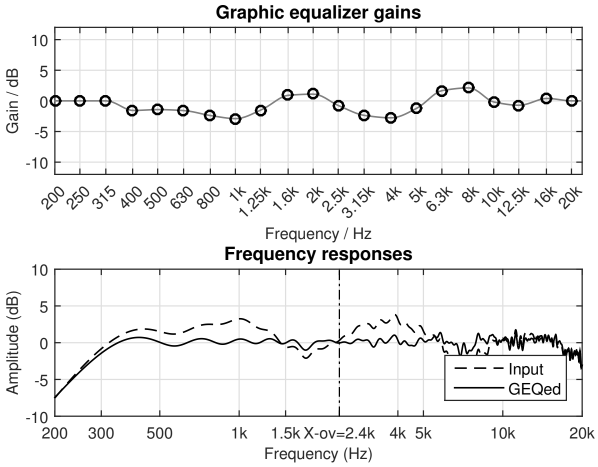

Figure 13: *(Top) Automatically calculated gains of the third-octave graphic equalizer and (bottom) the overall magnitude response before and after equalization.*

3.4. Graphic equalization

The graphic equalizer used for the PA equalization is a third-octave parallel design [16]. 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 frequency bands are limited from 315 Hz to 20 kHz. Figure 13 presents the obtained gains are presented and the comparison of the system frequency response before and after filtering with the parallel graphic equalizer.

3.5. Verification sweep

In order to verify that the system has the expected time alignment and frequency response, a third sweep is performed and processed with the processing chain in Figure 2. For determining the delay for each way, the difference between both time delays calculated in Section 3.2 is calculated. The way with the shortest time delay (i.e. the speaker closest to the measurement position - usually the subwoofer, as in Figure 1) is delayed by the number of samples of the calculated time difference. The fractional delay line is only implemented for the LF way.

In Figure 14 is shown that the system is time aligned as no peaks appear in the group delay and no cancellation occurs in the frequency domain. Additionally, the obtained frequency response is fairly linear, with ripple of less than +/- 2 dB in almost all the frequency band. Regarding the group delay, the values are below audible limits [17].

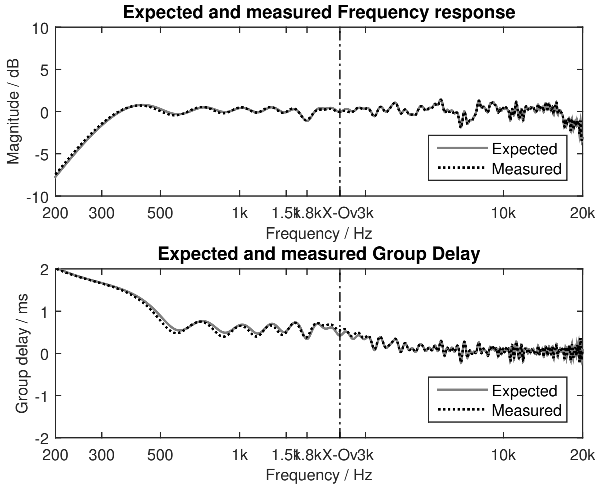

Figure 14: *(Top) Expected and measured overall magnitude response and (bottom) group delay.*

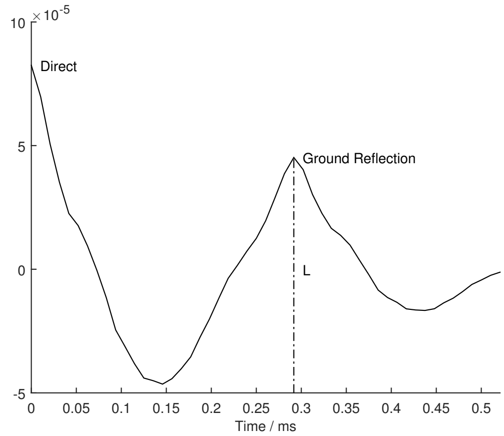

Figure 15: *Autocorrelation of input signal with ground reflection.*

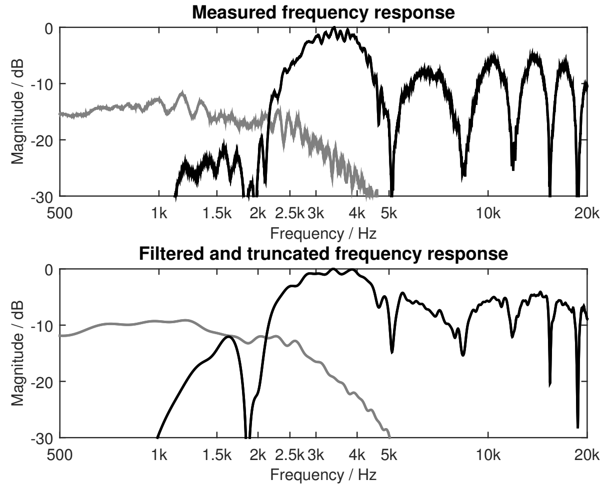

Figure 16: *(Top) Measured magnitude response with the reflection plate and (bottom) the corrected magnitude response.*

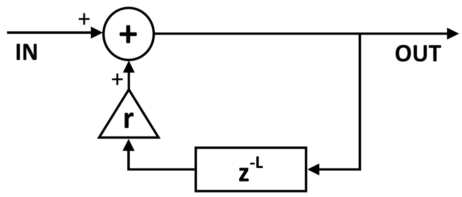

Figure 17: *Reflection cancellation filter.*

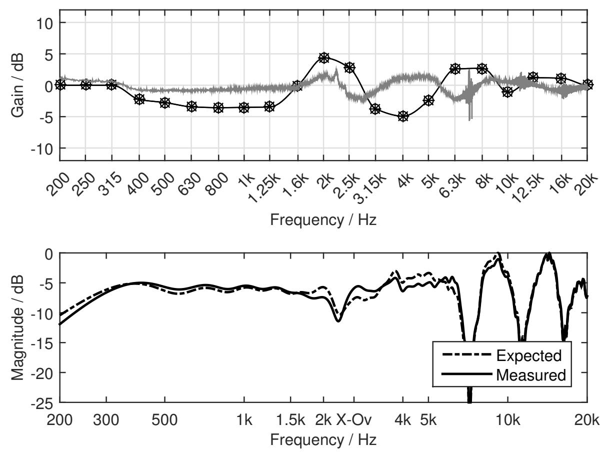

Figure 18: *(Top) Difference between the expected and measured magnitude responses and the calculated gains for the graphic equalizer: asterisks correspond to the first calculation and circles second calculation. (Bottom) Expected and measured magnitude responses.*

4. GROUND REFLECTION ISSUES

As this system is initially designed for use in outdoor venues or arenas, 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. This is not the case with the ground reflection, 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 frequency response and a shifting in the group delay.

The upper frequency response in Figure 16 shows the influence of the ground plates used in the measurement. The extra distance of the ground reflection is approximately 17 cm, which is equivalent to a half wavelength approximately at 2 kHz, where the first notch appears.

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 17 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, shown in Figure 15. 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 as shown in Figure 16.

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. The average difference between the expected and the measured band is calculated. If both, the difference and the equalizer gain in a certain band is higher than 2 dB, it is considered that the band has been falsified and the gain is set to 0 dB.

In Figure 18 the calculated equalization is made with the corrected response in Figure 16. If it is compared to the equalization without ground reflection in Figure 13, very similar responses 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.

5. CONCLUSION

An automatic calibration and equalization system for line-array PAs has been presented in this paper. The procedure for calculating the different parameters has been tested using a 1:10 scale model, and the obtained results have been described and analyzer. Four sets of parameters 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 parameters.

The obtained results are in the line with what we expected: a close to flat frequency response and low group delay can be obtained using phase (delay) adjustment in the crossover band. Such a system could be incorporated in PA processors making the line-array adjustment a much faster and easier task than it is today.

Even so, some improvements could be introduced to the system. Future work could be oriented in adapting the system for multi-channel operation, being able to equalize stereo or more channels (as central channel or outfills). Also subwoofer arrays could be adjusted with similar procedures, based on obtaining the response of each cabinet separately. This would allow to use phase adjustments to generate different subwoofer configurations and SPL patterns.

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