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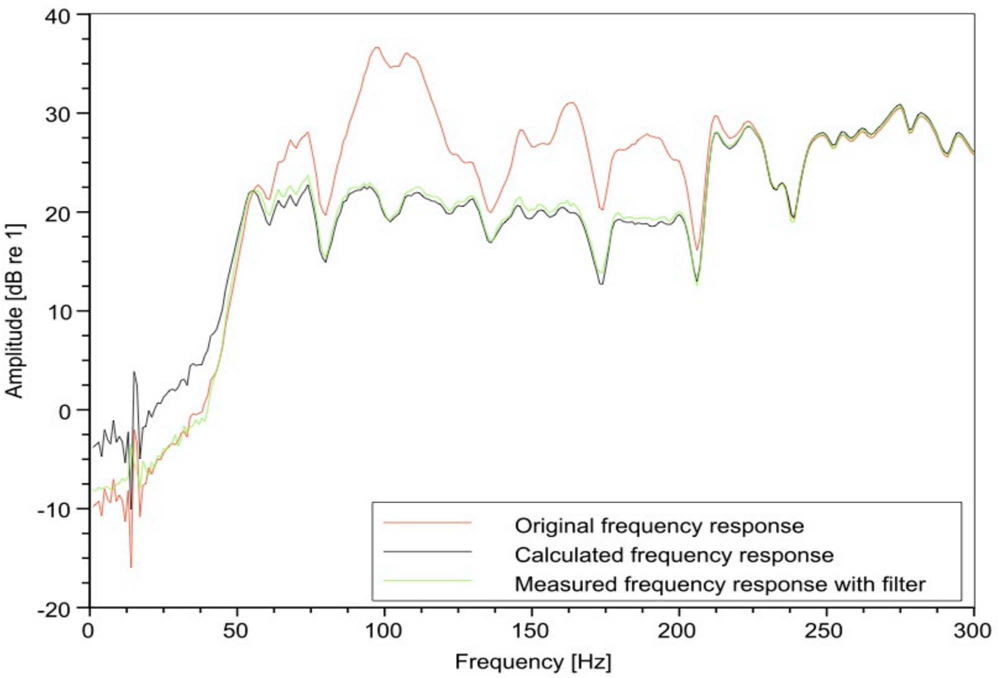


# Digital frequency compensation for loudspeakers used in rooms

## Using adaptive equalization

Master's Thesis in the Master's programme in Sound and Vibration


VIKTOR GRANBOM

Department of Civil and Environmental Engineering
*Division of Applied Acoustics*
*Room Acoustics Group*
CHALMERS UNIVERSITY OF TECHNOLOGY
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Division of Applied Acoustics
Room Acoustics Group
Chalmers University of Technology
SE-41296 Göteborg
Sweden

Tel. +46-(0)31 772 1000

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## Abstract

Equalization of loudspeakers and rooms has been performed since the late 1920's and is often a quite complicated task that requires a certain degree of knowledge about loudspeakers and room acoustics. This equalization is often done with built-in equalizers in stereo equipment or a separate equalizer. The two most common form of equalizers are the graphic equalizer or the parametric equalizer. Modern digital signal processing units (DSPs) have created new possibilities for equalization with total control and customization of all equalizer parameters, something that was restricted with analog equalizers due to their electrical components and their demand for space.

The purpose of this master thesis is to investigate if it is possible to implement digital room correction for loudspeakers using the Actiwave system of DSPs and digital amplifiers and if it can be used to noticeably improve the sound quality for everyday listening. This system should be fully automatic and only equalize the lower frequency region, e.g. under the Schroeder frequency where the modes of the room can be seen as resonant peaks and can be disturbing. Early in the process it was clear that this could be made so a software replica of the actual equalizer implementation was created in SciLab. This software uses exponential sine sweeps (ESS) to measure the room impulse response and a number of digital infinite impulse response filters (biquads) to equalize the room. Finite impulse response filters was also tested but due to their delay it was impossible to use these.

The results show that the software works as expected and that a noticeable difference can be heard when using the equalizer. The ESS measurement method is prone to background noise and can be used in a normal listening environment with good results.

The automatic equalizer removes most of the effect of modal resonances and also removes some distortion in the loudspeakers which is good. The downside of this is that the overall bass experience becomes somewhat flat. This was compensated for with a shelving filter with excellent results.

**Keywords:** Room Impulse Response, Inverse Filtering, Loudspeakers, Digital Signal Processing, Equalization

# Contents

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# 1 Introduction

### Background

Equalization of loudspeakers and rooms has been performed since the late 1920's and is often a quite complicated task that requires a certain degree of knowledge about loudspeakers and room acoustics. This equalization is often done with built-in equalizers in stereo equipment or a separate equalizer. The two most common form of equalizers are the graphic equalizer or the parametric equalizer. Modern digital signal processing units (DSPs) have created new possibilities for equalization with total control and customization of all equalizer parameters, something that was restricted with analog equalizers due to their electrical components and their demand for space.

### Purpose

The purpose of this master thesis is to investigate if it is possible to implement digital room correction for loudspeakers using the Actiwave system of DSPs and digital amplifiers and if it can be used to noticeably improve the sound quality for everyday listening.

### Task

Develope and if it is possible implement a fully functional measuring and filtering system in the Actiwave hardware with support for realtime filtering with low delays.

### Method

By measuring the impulse response of the room it is possible to create a filter for the loudspeaker that compensates for the frequency response of the loudspeakers and room. This was done in a number of steps. A semi-theoretical solution of the measuring system was created in Scilab. The measuring system was written in the C-language and should ported to Linux when the Linux core is up and running on the FPGA. When a measurement is performed, the generated filter coefficients are loaded on to another system of the FPGA that handles the sound reproduction and amplifying but for the test software these are only printed out as a variable.

# 2 Room acoustics theory

This chapter gives the reader some basic knowledge about two room acoustic concepts that is important when trying to equalize a room. These are room modes and room impulse response measurements. Two of the most used room impulse response measurement techniques , maximum length sequences and exponential sine sweeps, are also explained with their advantages and disadvantages.

## 2.1 Room Modes

In a normal listening environment, such as a livingroom, sound waves are reflected between the walls of the room and can coherently interfere with each other. Due to the coherence these reflections can amplify a resonance in the room. The resonances of a room are determined by the room geometry and can in the case of a perfect cuboid room be found by finding the eigenvalues of the wave equation [Eve 01]. These resonances are called room modes and can be troublesome when trying to achieve a good listening experience.

In a rectangular room there are three different types of modes that occurs, these can be seen in figure 2.1. The modes are called axial, tangential and oblique [Kin 00]. Axial modes are standing waves between two of the walls in a room, tangential modes are standing waves between four of the walls and oblique modes are standing waves between all of the walls in a room. The frequency of a mode can be calculated using equation 2.1, $n$ is the order of the mode and $l$ is the length of the room in that dimension. Of these three types the axial modes are most dominant and are also of the greatest interest when trying to equalize a room because they are the first modes to occur at lower frequencies.

$$f_{n_x n_y n_z} = \frac{c}{2}\sqrt{\left(\frac{n_x}{l_x}\right)^2 + \left(\frac{n_y}{l_y}\right)^2 + \left(\frac{n_z}{l_z}\right)^2} \tag{2.1}$$

An important acoustic measure when talking about room modes and the density of these is the schroeder frequency, which is defined in equation 2.2. $T_{60}$ is defined as the reverberation time of the room, i.e. the time it takes for the sound pressure level to drop by 60 dB and $V$ is the volume of the room. The schroeder frequency is the cross-over

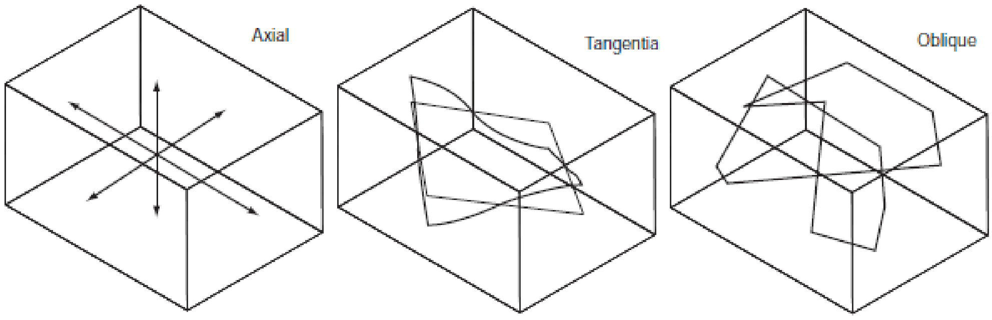



Figure 2.1: The three different types of room modes [Too 08].

frequency of where the individual modes no longer can be seen as resonant peaks but instead several modes add up to the total amplitude response of a certain frequency. However, this in not entirely true as the modal density is also a function of the room dimensions as seen in equation 2.1 and somewhat stochastic due to doors, windows and other properties of the room. Determining the schroeder frequency can be of importance when doing modal equalization because over this frequency it can be harder to get audible results when trying to equalize a single mode.

$$f_{schroeder} = 2000\sqrt{\frac{T_{60}}{V}} \tag{2.2}$$

It is also the spatial behavior of the lower order modes that makes them more interesting to equalize than higher order modes. From a spatial point of view the sound pressure in a room is not flat and depending on where the listener is located in the room, the sound pressure level will be perceived differently. If the listener is located at a point in the room where a mode has a maxima it will be perceived as higher than if the person were to be located at a point where the mode has a minima. Lower order modes have longer wavelengths which implies that if the person listening would make a smaller adjustment in listening position the difference in sound pressure level is smaller than for a higher order mode.

As example we consider two axial modes in the same axial direction, 100 Hz and 1000 Hz. The 100 Hz tone has a wavelength of 3.44 m and the 1000 Hz tone has a wavelength of 0.34 m. If the person listening is seated in the room where both these modes has a maxima and then relocates 0.34 m in the axial direction of the modes, the sound pressure for the 1000 Hz mode is at a minima whereas the 100 Hz mode is still close to maxima.

## 2.2 Room Impulse Response

Recording the sound from a loudspeaker in a room we perceive both the sound coming directly from the loudspeaker as well as reflected sound from the all the surfaces in the room. Without knowing how these reflections behave we can model the room as a black box that we insert an input to, in this case sound coming from the loudspeaker. At the other end we receive an output, in this case the recorded sound. Treating the microphone and loudspeaker as perfect and using an impulse as output from the loudspeaker and recording this we get the impulse response of the room. Because of the spatial distribution of sound pressure in a room this recorded room impulse response is only valid for this specific speaker and microphone location in the room and is different for all other locations [Bha 06].

In mathematical terms the simplest way to describe the room impulse response is to treat the room as a linear time-invariant system. The output of this system, $y(t)$, can be described as a convolution between the systems input signal, $x(t)$, and the room impulse response, $h(t)$.

$$y(t) = (x * h)(t) \tag{2.3}$$

Where a convolution is mathematically defined as:

$$(x * h)(t) = \int_{-\infty}^{\infty} x(\tau)h(t - \tau)d\tau \tag{2.4}$$

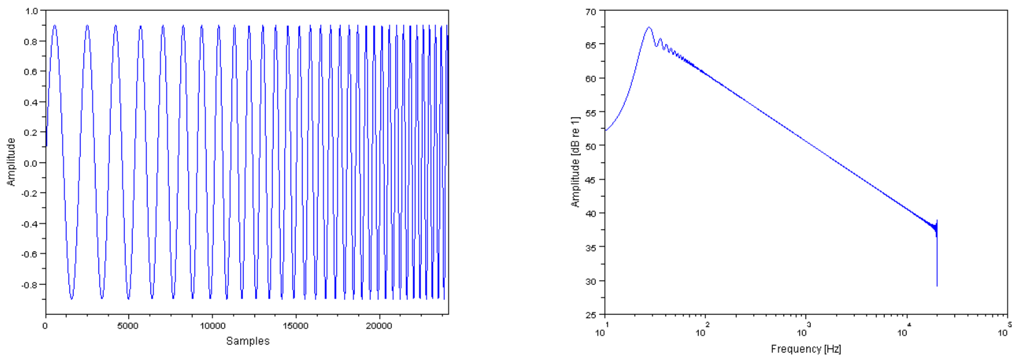
Using the fourier transformation of $h(t)$ we get the frequency response of the room. The frequency response can be divided into two parts, the magnitude response and the phase response. The magnitude response describes the amplitude of a certain frequency and the phase response the resulting delay of the system at this frequency. This is important in many types of signal and audio analysis.

There are many factors that could influence the recording of a room impulse response, for example the loudspeaker frequency response is not perfectly flat and this will alter the sound coming from the loudspeaker [Bha 06]. In some cases such as room equalization it is more suitable to treat the whole chain of sound source, amplifier, cables, loudspeakers and the room as a single system because all of these parts will alter the sound that the listener perceives in the end.

Recent years, a lot of studies have been made on how to create better room impulse response recordings that are unaffected from background noise and artifacts in the loudspeaker/recording chain and among them the exponential sine sweep and maximum length sequence techniques are the most prominent.

### 2.2.1 Exponential Sine Sweep Measurements

The exponential sine sweep (ESS) method was further developed by Angelo Farina in 2000 but has been used to some extent before that [Far 00]. It has become widely popular in recent years and as the name states it uses an exponential sine sweep to measure the room impulse response. With this method it is also possible to do measurements of harmonic distortion [Sta 02]. According to Holters [Hol 09] the goal was to construct a signal $x(n)$ that has an inverse function $x^{-1}(n)$ that can easily be determined in the sense that the convolution between the signal and its inverse results in a time-shifted dirac delta pulse. A simple explanation on how this can be done is made here but the full story on how to derive this can be found in [Far 00] and [Hol 09].




(a) Time domain (zoomed to better show the increasing frequency).

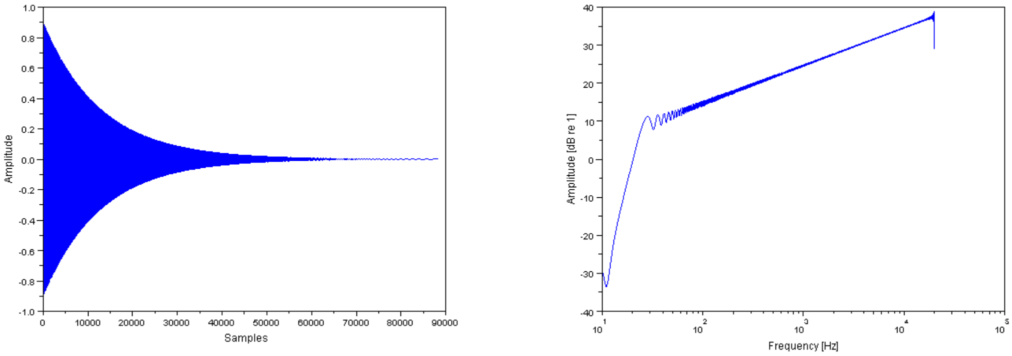
(b) Frequency domain.

Figure 2.2: Exponential sine sweep signal generated according to equation 2.5.

The sine sweep is generated according to equation 2.5 [Far 00] where $T$ is the length of the sweep in seconds, $w_1$ is the lowest frequency and $w_2$ the highest frequency of the sweep. An example of this can be found in figure 2.2.

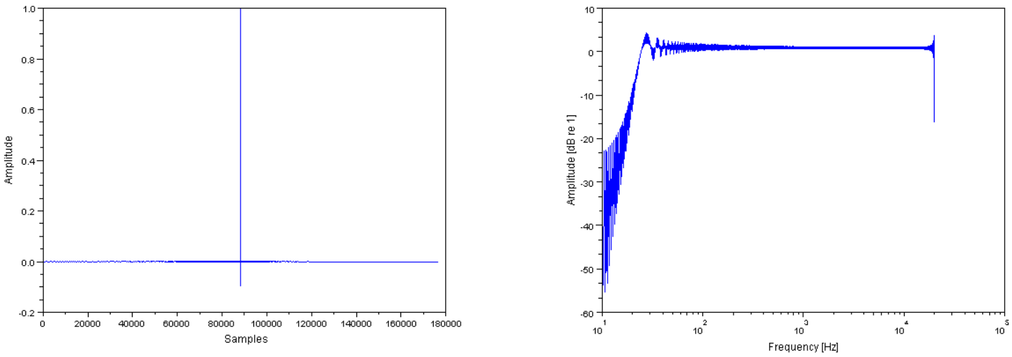
$$x(t) = \sin\left(\frac{w_1 T}{\ln\frac{w_1}{w_2}}\left(e^{\frac{t}{T}\ln\frac{w_1}{w_2}} - 1\right)\right) \tag{2.5}$$

Because of the $-6dB$ per octave slope of the sine sweep in figure 2.2(b) we must apply scaling to the inverse signal which is the time-reversal of the original signal. The scaling factor is $-6\ln\left(\frac{w_1}{w_2}\right)$. This can be seen in figure 2.3. The result of the convolution of the sine sweep and its inverse is found in figure 2.4. The passband ripple and overshoot




(a) Time domain. (b) Frequency domain.

Figure 2.3: Generated inverse signal for obtaining a dirac delta pulse.




(a) Time domain. (b) Frequency domain.

Figure 2.4: The result of convoluting a sine sweep with its inverse function.

found in figure 2.4(b) is a result of using a frequency band instead of all frequencies and can be limited using a time-domain window [Hol 09]. This can be realized with fade-in and fade-out of the sine sweep.

The main advantages of the exponential sine sweep method is that more energy is present in the lower frequency region of the signal and that non-linearities of the measured system results in anti-causal components in the impulse response [Hol 09]. In other words the harmonic distortion components in the impulse response are placed before the linear part of the impulse response and can easily be removed by truncating the impulse response [Sta 02]. The signal-to-noise ratio of this method is another advantage [Sta 02].

### 2.2.2 Maximum Length Sequence Measurements

A maximum length sequence (MLS) signal is a type of pseudo-noise sequence with known properties [Hol 09]. The auto-correlation of the MLS signal is a very good approximation of the Dirac delta function and therefore the room impulse response can be measured using a MLS sequence but instead of using the linear convolution as in 2.2.1 the circular cross-correlation of the input (the MLS signal) and the measured output is used to find the room impulse response.

The number of samples needed for creating a MLS signal of order $m$ is $L = 2^m - 1$ [Sta 02]. The individual samples take on values of -1 and 1 with phase between $-\pi$ and $+\pi$ [Ole 00]. Because of the periodicity of the MLS signal and the circular operations used, the room impulse response will also be periodic as can be seen in equation 2.6 [Sta 02].

$$h'[n] = \sum_{l=-\infty}^{+\infty} h[n + lL] \tag{2.6}$$

This equation shows one of the known weaknesses of MLS signals which is if the measured impulse response is longer than the MLS signal we get aliasing in the time domain. Because of this, one has to make sure that the MLS signal used is of sufficient length. For example a system using $f_s = 44100Hz$ and $m = 16$ results in $h'[n] \simeq 1.49s$. Another weakness is the sensitivity to non-linearities of the measured system. Using band-limited reconstruction of a discrete MLS signal results in a amplitude higher than 1 and if the system is not carefully calibrated this will results in non-linearities in the signal and will be added to the non-linear part of the room impulse response [Hol 09].

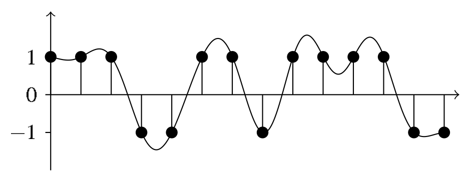



Figure 2.5: Discrete MLS signal and its band-limited reconstruction [Sta 02].

This calibration is in most cases a trial and error procedure that can be very time-consuming and results in a lower signal-to-noise ratio. Non-linearities of this type can be heard as crackling noises when auralizing the room impulse response [Sta 02]. A method of improving this has been proposed by Olesen [Ole 00]. This method uses several MLS signals of different length to be able to average out the non-linear distortion artifacts.

The advantage of using a MLS signal instead of an ESS signal is the insensitivity to background noise. The MLS technique randomizes the phase spectrum of a disturbing signal, such as white noise, which is not correlated with the MLS signal. The uniformly distributed noise can then be reduced by using averaging of several measurements [Sta 02].

# 3 Digital room correction

Digital room correction is a collection of methods used to give the listener a better listening experience in a room. The goal is to reduce the negative influence of the room by pre-filtering the audio source. There are mainly two approaches to digital room correction used today. The inverse room impulse response technique is based on FIR filters and the room equalization technique is based on IIR filters. Therefore it is important to get some introduction to these filters before a more thorough explanation of the two approaches are explained. A small introduction to the FPGA is also given as some basic knowledge on how these approaches can be implemented is needed for the understanding of the software.

## 3.1 Sampling

Sampling is the process of converting a continuous-time signal to a discrete-time signal [Wan 07].

$$x[n] = x(t) \mid_{t=nT} \quad n \in \mathbb{N} \tag{3.1}$$

To perform sampling the value of the continuous-time signal is read at a number of points $n$ with a certain time period between them, $T$. If the time interval is constant the sampling is called uniform and if the time interval changes the sampling is called non-uniform. The number of samples per second is called the sampling frequency and is denoted $f_s$. If the discrete-time signal is to be converted back to the analog signal the Nyquist criterion must be fulfilled. According to the Nyquist sampling theorem this is $f_s > 2f_{max}$ where $f_{max}$ is the highest frequency component of the analog signal [Wan 07]. If this condition is not met aliasing effects will occur.

## 3.2 Bandwidth

In general case of a passband filter with a center frequency $f_0$ the bandwidth $\Delta f$ is defined as the frequency range between the two points where the amplitude of the passband has dropped by -3 dB.

$$\Delta f = f_h \mid_{-3dB} - f_l \mid_{-3dB} \tag{3.2}$$

Where $f_h$ is the upper frequency limit and $f_l$ is the lower frequency limit. When talking about equalizer the bandwidth expression in not as consistently defined as one may think. One of the problems with using the -3 dB limits is that applying two filters with equal boost and cut to a signal will not produce a 0 dB gain [Bri 94]. Another problem is that designing a filter with a boost or cut under 3 dB is impossible [Mil 04]. Therefore another definition of bandwidth has been proposed to eliminate this problem. Gain is defined in decibels as the sound pressure level $p$ above a certain reference level $p_0$, see equation 3.3. Here the lower and upper frequencies are defined as the two points where the gain is half of the gain at the center frequency [Bri 94], see equation 3.4. This is the definition used in this thesis.

$$Gain = 20 \log \frac{p}{p_0} \tag{3.3}$$

$$\Delta f = f_h \mid_{Gain/2} - f_l \mid_{Gain/2} \tag{3.4}$$

## 3.3 The Quality Factor

In the world of equalizers the term bandwidth is rarely used. It is often replaced with quality factor, also known as $Q$.

$$Q = \frac{f_0}{\Delta f} \tag{3.5}$$

The quality factor is an inverse measure of the bandwidth with weighting of the center frequency. A high order $Q$ indicates a narrow frequency band and a low order $Q$ indicates a broad frequency band. This is a more useful expression for equalizers because it relates bandwidth to the human hearing and octave band scales.

## 3.4 Digital Filters

Digital filters are built up blocks of adders, multipliers and delay units [Wee 07]. They are generally used in some kind of digital signal processing unit (DSP) or personal computer and have a number of advantages over analogue filters. Some of these are:

- A digital filter can be programmed according to the users specification. This means that the filter can be changed on-the-fly without the need to change or adjust the hardware.
- Analog filters change their behavior depending on temperature and other external factors such as the quality of the components, such as capacitors. Digital filters are immune to this.
- Digital filters handle low frequencies better than analog filters.

### 3.4.1 Finite Impulse Response Filters

Finite impulse response filters or FIR as they are commonly called are characterized by their linear phase properties and as the name states their finite impulse response [Wan 07]. The general form of a FIR filter can be found in equation 3.6 and figure 3.1.

$$y[n] = \sum_{k=0}^{K} b_k x[n-k] \tag{3.6}$$

The finite part is derived from the fact that there is no internal feedback in the filter as

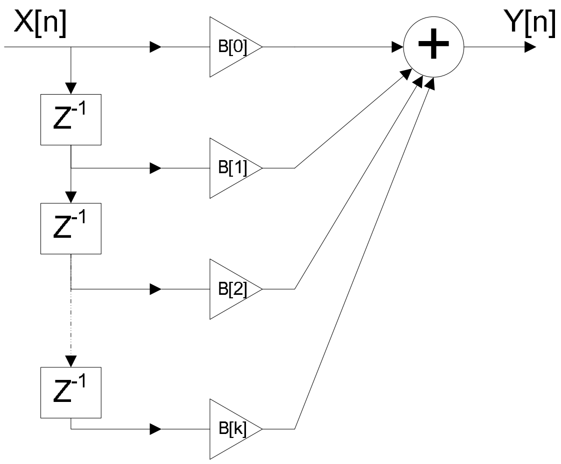



Figure 3.1: The general form of a FIR filter.

in the case of an IIR filter. Of course there are always exceptions such as the moving-average filter but the feedback of these filters is always a fixed number of samples so the theory is still valid. A FIR filter can not become unstable

The linear phase properties are very useful when dealing with audio. This means that all frequencies passing through the filter have the same delay. There is another type of FIR filters which are called minimum phase. They have a lower delay for the same impulse response as a linear phase but the disadvantage that they have a non-linear phase response (phase distortion). In the time-domain of impulse response this shows up as echoes of the original impulse response which makes them unsuitable to audio processing applications.

### 3.4.2 Infinite Impulse Response Filters

Infinite impulse response filters or IIR as they are commonly known, are the "classic" type of filters because they can always be derived form their analogue counterpart

[Wan 07]. This is a major advantage over FIR filters when designing the filter because the behavior of the analogue filter counterpart is for most cases well-known. Proper designed IIR filters are characterized by their impulse response that decay towards zero but never reaches it. An improper designed IIR filter can be unstable and in this case the impulse response will increase towards infinity. Higher order filters are more sensitive to this than lower order filters and because of this, higher order filters are mainly realized as cascaded second order filters [Ash 08].

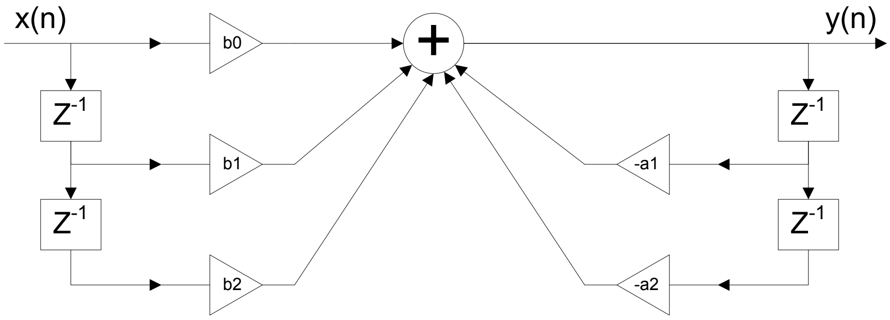



Figure 3.2: Biquad of direct form 1 implementation.

A digital IIR filter can be designed using the bilinear transform and one of the most common implementations of the IIR filter is the "biquad" that can be seen in figure 3.2 [Wan 07]. The biquad is a second order recursive filter that is named from its transfer function that in the Z-domain is the ratio of two quadratic functions, see equation 3.7.

$$H(z) = \frac{b_0 + b_1 * z^{-1} + b_2 * z^{-2}}{a_1 * z^{-1} + a_2 * z^{-2}} \tag{3.7}$$

## 3.5 Equalizers and Equalization

Equalization is often referred to as altering the magnitude response of a electronic system but it can also be altering the phase response. Equalizers are used for both recording and playback purposes and they have been used in audio industry since the 1930's [Ran 97]. All equalizers can be divided into a number of different groups depending on their architecture and usage areas. There are active and passive equalizers, graphic and parametric, analog and digital, sliding and rotary, constant-Q and proportional-Q amongst others [Ran 97].

The standard type of equalizer that was used from the 1960's and is still used for audio recording and playback is the graphic equalizer. There are thousands of different graphic equalizers but one most commonly used is the third octave band equalizer as seen in figure 3.3.It has in most cases fixed $f_c$ and $Q$-values for each band and the gain can be adjusted with sliders or knobs. A general misconception is that the curve seen on a graphic equalizer (the curve between the midpoint of all sliders) is the actual equalization filter curve. This is not the case, as Miller [Mil 04] points out. Figure 3.4 shows the filter curves for three different types of graphic equalizers which all have the same visual appearance as in figure 3.3 but different filter topology.

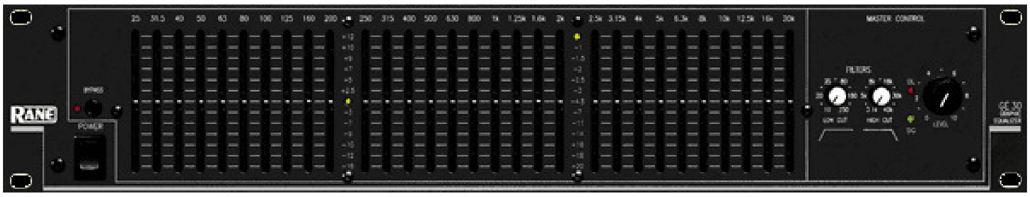



Figure 3.3: Graphic equalizer. Model: Rane GE 30 [Ran 85].

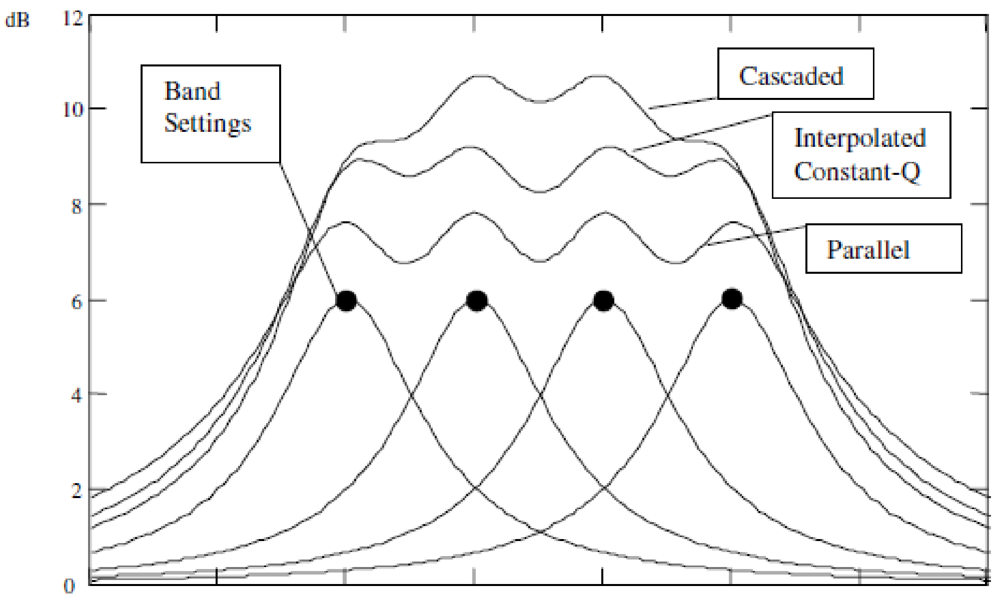



Figure 3.4: Combined responses for various types of graphic equalizers, all having the same $+6dB$ gain for the four bands [Mil 04].

Using an equalizer for recording purposes and for playback purposes are two different tasks. When recording audio, the purpose is to create a sound and therefore knowledge about the listening environment is not needed. When using a equalizer for playback the purpose is to recreate the original sound. Doing this in most cases requires some knowledge about room acoustics and the frequency response of the listening environment. This is hard to do by ear and often it is recommended to use some kind of real-time analyzer.

Nowadays the digital equalizer has become the most popular equalizer to use. In most cases there is no visible difference between a graphic analog and a graphic digital equalizer despite the fact that a digital equalizer has a wide range of configuration possibilities. When moving from analog to digital recording the digital equalizers used in programs as Logic Studio, Cubase or Acid Pro were made to resemble existing products that the user already was familiar with, such as a standard graphic third octave band equalizer, see figure 3.5. Home stereo and Hi-Fi equipment is just the same, new stereos and other products are made to resemble old graphic equalizers that the user is familiar with.


Figure 3.5: A digital implementation of a third octave band graphic equalizer [Ran 10].

One of the major benefit of digital audio products is the possibility to do an all-in-one solution for these kinds of problems. Instead of letting the user do all equalization by hand there are devices that combines a real-time analyzer and parametric equalizer to remove any negative influences of the room. This is however not common despite the fact that the use of measurement microphones has become a standard in mid-range and high-end digital receivers. Most of them only corrects the distance from the listening position to the speakers and do not perform any kind of automatic room equalization which easily can be done with the same microphone.

### 3.5.1 Minimum Phase vs. Mixed Phase Equalization

Minimum phase equalization is the most common equalization technique used. Using the term minimum phase equalizer indicates that there are another type of equalizer that is not minimum phase but that is rarely the case [Boh 86]. All IIR equalizers such as a graphic or parametric equalizer uses this technique though some of these equalizers has a built-in delay unit for controlling (delaying) the phase. A specific IIR filter can only be designed in one specific way with its specific magnitude and phase characteristics and therefore all IIR equalizers are minimum phase. Using a delay unit is not the same as a mixed phase equalization where the phase is actually controlled by filters and not just delayed.

Mixed phase equalization is often referred to as dereverberation and can be realized using a FIR filter with a minimum phase part and a all-pass part [Kar 05]. Deriving mixed phase filters are mathematically more complex than minimum phase filters and in most cases requires pre-filtering of the impulse response to work. Using an inverted impulse response without filtering will in theory result in a completely dereverberated listening position but due to practical reasons this does not work [Rub 00]. From the concept of room modes described in chapter 2.1 we know that altering listening position in the room will result in a new room impulse response for that specific location which is analog to moving just slightly from a complete dereverberated location in the room would require a new dereverberation for that specific location which is impossible to do in real-time.

### 3.5.2 FIR Equalization

FIR equalization is somewhat more complicated than IIR equalization but also have a number of advantages. Using a sufficient number of coefficients (filter taps) a mixed phase filter can be constructed. This can be done using pre-filtering techniques, different types of frequency warping, cascaded filters or pattern recognition [Bha 06]. The foundation of FIR equalization is the use of inverted impulse responses.

The impulse response can be inverted by using one of several techniques that are divided into non-parametric and parametric. Non-parametric inversion is direct inversion of the FFT or least mean square inversion. Parametric inversion is the use of AR/ARMA modeling. Of these methods the least mean square inversion has proved to be the most stable [Kar 05].

Creating a FIR filter covering the audible frequency range requires a lot of taps. High Q room resonances in the low frequency region requires a frequency resolution of about

1-2 Hz [Rub 00]. If the required frequency resolution is 2 Hz and using a sampling frequency of 44100 Hz this would result in a filter with 22050 taps for a 1 second long room impulse response. Without the possibility to pre-buffer audio this directly translates to a delay of 0.5 seconds which for many audio systems is unacceptable, especially when dealing with combined audio/video systems [ITU 98].

To overcome the problems with too many filter taps a number of methods has been developed. The use of auditory filter banks is one way of solving this problem [Smi 99]. Here the linear frequency axis is divided into a number of sections that are easier to handle with bilinear transformation. This can be done accordingly to any arbitrary scale but most common are the octave scales, ERB scale or the Bark scale. The Bark scale is the result of psychoacoustic research and divides the frequency axis into a number of critical bands that matches the human hearing and all bands have the width of one bark [Smi 99]. The FIR filter can then be designed with AR/ARMA modeling Another method is the use of multiple bands. Here the frequency axis is divided into a number of bands where the low frequency bands has a higher frequency resolution than the higher frequency bands [Rub 00]. Each band becomes a separate FIR filter that is in the end becomes part of a cascaded FIR filter with the other bands [Bha 06].

A free and well documented implementation of an equalizer using FIR filters is written by Denis Sbragion and a detailed guide on how he has done this can be found at his homepage [Sbr 10].

### 3.5.3 IIR Equalization

Digital IIR equalization is the equivalent to analog equalization. IIR equalization can be made possible with virtually any IIR filter but in most cases they are made as a digital implementation of an already existing analog equalizer. Just as in the analog case most digital IIR equalizers are built up by a number of filters in different order. As Miller states these filters can be of varying topology [Mil 04].

IIR equalization can be used with or without a room impulse response recording but it is often recommended to have a target function to equalize towards. Analog equalizer manuals often recommended using a real-time analyzer but this is often a built in feature of modern equalizers [Dbx 10]. In the case of equalization the major benefit of using digital IIR equalizers is full customization of all filter parameters such as $Q$, $f_0$ and filter curve.

## 3.6 FPGAs and ASICs

Designing and implementing digital circuits often results in a choice between using an integrated circuit (IC) or a application specific integrated circuit (ASIC) [Gro 08]. IC is a very wide definition covering all integrated circuits so the choice really stands between a field-programmable gate array (FPGA) (or equivalent) and an ASIC. An FPGA is as the name states field-programmable so the users can configure it for their own requirements and is built up by millions of logical gates that can be routed. Often this routing is made by some kind of intermediate language, for example VHDL. This makes FPGA an ideal platform to develop new products on. Modern FPGAs have become so fast and dense that it is also possible to implement a soft processor on the chip. Combining hardware design and software onto a single chip is makes it possible to have a complete product on chip and is referred to as a System-on-a-chip or SOC. Some of the disadvantages of an FPGA is that one can run out of logical space on the chip and in some cases the more logical space used the slower it runs, i.e. a lower clock cycle speed.

ASICs on the other hand are faster but has a fixed configuration. The downside of this is that they are extremely expensive to develop and is only affordable in larger volumes. The design can never be changed and therefore it is harder to develop products on this architecture.

# 4 Design of the Equalization Software

## 4.1 Introduction

The goal for the software was to create a program that measures the room impulse response of a given room where the loudspeakers are placed at an arbitrary position and calculate the appropriate filter coefficients to equalize lower frequency region where the modes can be audible.

The equalization software was created with two major programming goals. The first goal was to get a working prototype in SciLab for testing and verification. The second goal was if there was time left and the other hardware needed was available, it should be implemented in the FPGA.

Breaking this down into smaller goals, this means that the software created in SciLab should resemble the final implementation as much as possible. The final implementation should be created in the C programming language which makes SciLab (or MatLab) very suitable because both of these softwares are C compatible and the programming can be made according to the C programming standards.

## 4.2 Hardware Limitations

Before creating the software is it very important to be aware of the hardware limitations of the system. These set the general design guidelines of the software and the software must adapt to these limitations in order to perform well.

### 4.2.1 User Interaction

The hardware has no graphic interface and is solely depending on a remote control for user interaction. Therefore the equalization software must be designed to be used completely without user interaction. The only interaction should be a "Equalize" and "Equalizer On/Off" button on the remote control of the system. Because of this it is difficult to notice the user if the measurements made are valid or not and if the equalizer works as expected. The only way of doing it for now is to use some kind of auditory signal matched with a led light on the measurement unit.

### 4.2.2 Sampling Frequency

The sampling frequency is fixed at $44100Hz$ due to limitations in the hardware. As mentioned earlier, a too short recording of the impulse response in combination with a low sampling frequency can result in a poor frequency resolution. A good frequency resolution is very important here because of the peak-sensing algorithm used. If the resolution is too low peaks may not show up at the right frequency in the frequency plane or in worst case not at all which can be devastating for the function of the equalizer.

### 4.2.3 Filter Implementation

The filters used in the hardware are mainly biquads and for the purpose of a digital equalizer, used in this hardware, this is the only filter that can be used. However, this is a suitable choice of filter due to its versatility and ease of use. Lack of space on the FPGA chip have resulted in a maximum number of 10 biquads on each channel resulting in a delay of 20 samples of each channel. This is a really low delay, at the given sampling frequency this corresponds to 0.4535 ms. It is very important to keep this delay low because it adds up with all the other filters to a total delay of the whole system which should be at most 45 ms [ITU 98] to avoid lip sync error when using the system connected to video equipment.

### 4.2.4 Processor Speed and Memory

The recorded room impulse response must have a fixed length and be able to fit in the memory of the hardware. Memory space for this is not permanently allocated but is rather shared with the other processes running on the hardware and must be cleared once recorded and truncated.

The iterative process, described in section 4.5, used in the equalizer algorithm is fail-safe but time demanding. This is directly proportional to the length of the truncated impulse response of each channel. If the equalization takes too long to perform the user may suspect a malfunction and reset the system. This must be avoided att all times. More on this process in section 4.5.

## 4.3 Software Layout

The program consists of two major modules and a set of smaller modules that help the two major modules. As can be seen in figure 4.1 the FPGA consists of two parts, or at least two relevant parts for this thesis. These parts are the signal processing unit and the soft processor. The signal processing unit is responsible for handling all audio inputs and outputs. Should the audio input be analog as in the case of the measurement

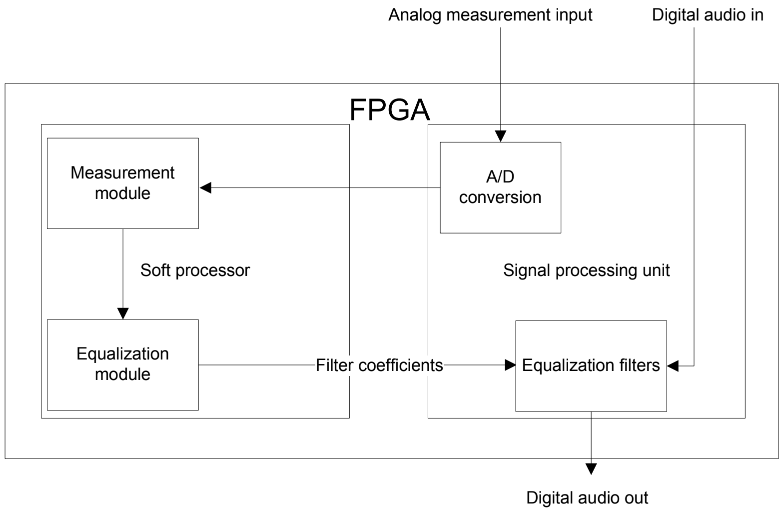



Figure 4.1: Simple schematics of the hardware layout of the FPGA showing the two parts of the FPGA and the two major modules of the software.

microphone input, the audio is sampled to digital form before it can be accessed from the program. It is also in the signal processing unit the actual biquads used for equalizing the sound are located. All biquads used in the equalizer are the same direct form 1 implementation and can be modified during runtime.

The soft processor part of the FPGA is where the software is located and running. The processor is running a compact version of a Linux distribution that is specialized for this FPGA. The software can access the audio inputs and outputs through the operating system which in turn can access and control parts of the signal processing unit. When the equalizer software is started, the software will first initialize a measurement of the room impulse response which is made in the measurement module. When the measurement module is finished it initializes the equalization module. This module calculates the filter coefficients which are sent to the signal processing unit via the operating system.

## 4.4 Measurement Module

The software created in SciLab uses an external audio input/output library created by Ronan Scaife [Ron 08] that is available as freeware on the internet. This library enables SciLab to work in the same way as if the code was running on the hardware with full duplex of all inputs and outputs. In other words an impulse response can be recorded directly from the software instead of being played from an external source. This greatly simplifies the procedure of recording an impulse response. As can be seen in figure 4.2 the right and left channel are recorded separately before the channel delay is calculated and the impulse responses are normalized and truncated to proper length.

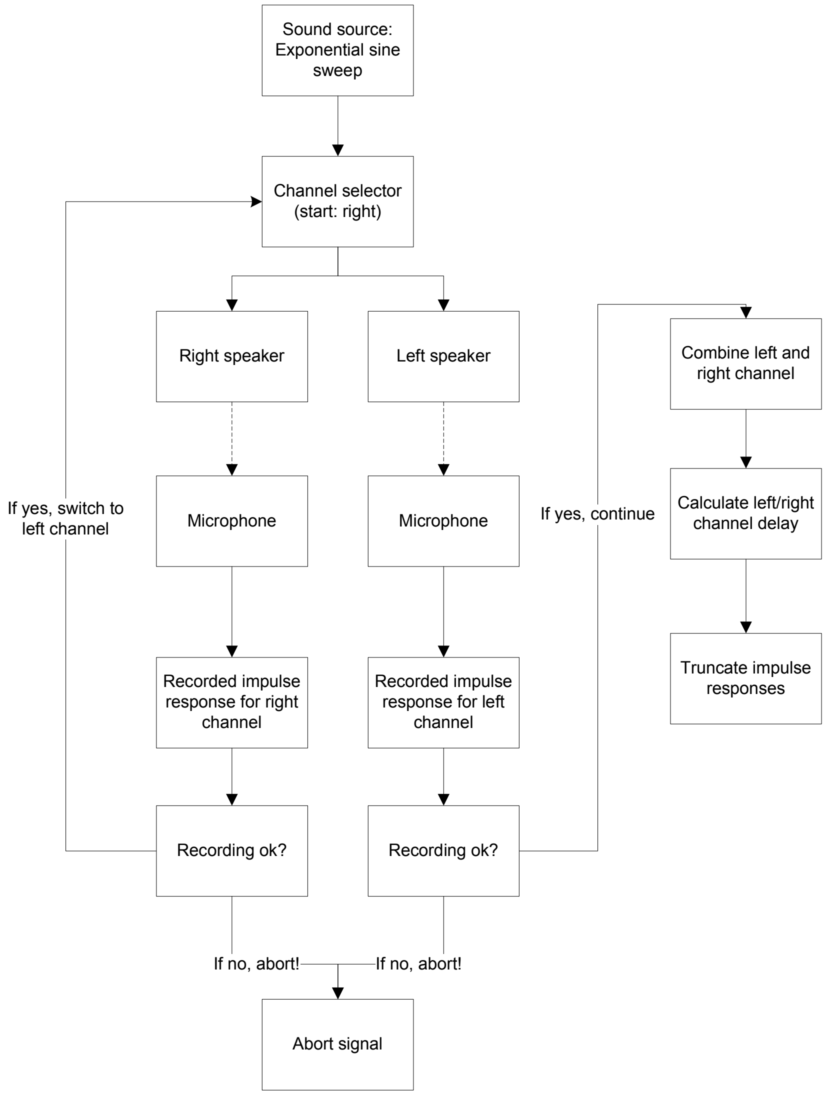



Figure 4.2: Schematics of room impulse response recording procedure.

The exponential sine sweep method was implemented and used as stimulus because of its noise rejecting properties as described in section 2.2.1. Though it is more sensitive to white background noise than the MLS method it rejects other types of background noise, such as TVs or people speaking, in a better way and it is more likely to get this kind of noise in a listening room than white noise.

### 4.4.1 Truncation and Normalization

When the impulse responses has been recorded and the speaker alignment correction has been calculated the impulse responses can be truncated. This is a fairly simple procedure which first locates the maximum of the impulse response, backs a number of samples for some margin and then truncates the impulse response to the last sample with an absolute normalized amplitude of over a pre-determined level. The software uses a minimum truncation so the frequency resolution is somewhere below 5 Hz. Experimental result shows that this regions result in some balance between runtime and correct equalization.

### 4.4.2 Average Gain Calculation

To equalize the room we need to know the average gain of the frequency response. This average gain sets the reference level for the gain used in the biquad filters and is calculated as the mean dB level for frequencies in the higher frequency region where the frequency response of the speaker is maximum flat.

### 4.4.3 Speaker alignment correction

The speaker alignment correction is based on the decorrelation of the impulse responses of the left and right channel. This is calculated as the difference in samples between the maxima of the right and left channel. An example of this can be seen in figure 4.3.

### 4.4.4 Frequency range

The fixed number of biquads used led to a fixed highest center frequency of 200 Hz. The results in 5.2 support this but the size of the room used is an important factor. A larger room can easily have a higher schroeder frequency and therefore a higher number of distinct resonance peaks that needs to be equalized. For that case the equalization may not be as good as expected but it is important to know that even if the equalization is not optimal it still works and helps to create a better listening experience. The lowest center frequency of correction was determined by the lowest frequency reproduced by the speakers. For the speakers provided by the company this was somewhere around 30 Hz but due to distortion this was later raised to 40 Hz so that the user does not think that there is something wrong with the speakers.

## 4.5 Equalization Module

The task of the equalization module is to feed data to the software biquad filter and return the filter coefficients used in the signal processing unit. This is an iterative process

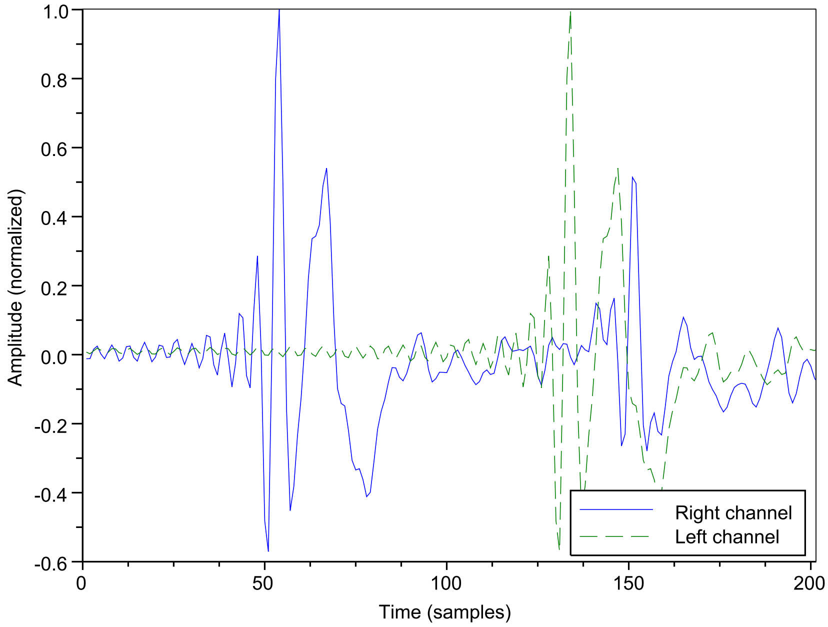



Figure 4.3: Typical impulse response zoomed in to show the delay of the arriving direct sound of the right channel. There is a delay of 79 samples between the two highest peaks corresponding to a difference of 0.61 meters.

that is performed up to 10 times for each channel. Given an impulse response with a lower and upper frequency limit, the equalizer module finds the highest peak in that region and the bandwidth of the peak and sends this to the biquad filter. The biquad returns the filter coefficients and the filtered impulse response to the equalizer module and the process is repeated up to 10 times.

### 4.5.1 Biquad Filter Design

The biquads used in the software generates filter coefficients needed for the signal processing unit. The software has no direct access to the signal processing units biquads and therefore we need to simulate the filters in the software before the coefficients are loaded in the signal processing unit. The biquad used in the SciLab software is a direct form 1 implementation visualized in figure 3.2. It uses a digitalized form of the parametric EQ transfer function in equation 4.1 that can be found in equation 3.7, with the coefficients in equation 4.2.

$$H(s) = \frac{s^2 + s * (A/Q) + 1}{s^2 + s/(A * Q) + 1} \tag{4.1}$$

$$\begin{aligned}
b_0 &= 1 + \alpha * A \\
b_1 &= -2 * \cos(w_0) \\
b_2 &= 1 - \alpha * A \\
a_0 &= 1 + \alpha/A \\
a_1 &= -2 * \cos(w_0) \\
a_2 &= 1 - \alpha/A
\end{aligned} \tag{4.2}$$

We also need to introduce some intermediate variables to make it easier to identify the coefficients, see equation 4.3. $k$ is a scaling coefficient and in the normal case equal to 1.

$$\begin{aligned}
\alpha &= \frac{\sin(\omega_0)}{(2 * Q * A)} \\
A &= 10^{\frac{gain}{40}}
\end{aligned} \tag{4.3}$$

The final expression for the implemented biquad is found in equation 4.4.

$$\begin{aligned}
y[n] = (b_0/a_0) * x[n] + (b_1/a_0) * x[n-1] + (b_2/a_0) * x[n-2] \\
-(a_1/a_0) * y[n-1] - (a_2/a_0) * y[n-2]
\end{aligned} \tag{4.4}$$

As can be seen from the equations we need to feed the biquad parametric EQ filter with some information from the measurements before we can use it. For this specific filter type we need the to know the $gain$, $f_s$, $f_0$ and $BW$ (or $Q$) before we can use it and these parameters can be derived from the measured room impulse response.

# 5 Results and Discussion

The previous chapters explained the theory behind equalization and room impulse measurements and how the software was designed. In this chapter the results of the measurements and equalization are presented and discussed. Measurement equipment and setup is also briefly explained here.

## 5.1 Measurement Equipment and Setup

To verify that the software works as expected a number of measurements in different listening environments were conducted. It was important that the software worked with different loudspeakers, in different rooms and under different listening conditions. The testing was performed on a laptop running the software in SciLab. A Behringer ECM8000 microphone and M-Audio FireWire 410 soundcard was used instead of the audio input of the Actiwave hardware, see figure 5.1.


Figure 5.1: Soundcard and microphone used during the measurements.

The testing could not be performed on the actual hardware because the Linux kernel that the software is supposed to run in was not working at the time of testing. Therefore it is important to see the measurement results as an indication of how the software works and not exact results. The final software must be tuned according to the software and microphone to be used.

## 5.2 $Q$ Scaling and number of Biquads Needed

Scaling $Q$ is equal to changing the bandwidth of the applied biquad filters. A higher $Q$-value creates more narrow filters. This in turn demands more biquads to achieve the same filtering effects. In figure 5.2 we see a comparison of how scaling $Q$ changes the frequency response of the room impulse response with applied equalization filter.

A lower $Q$-value tends to add too much damping to frequencies around the center frequency of the filter with the effect of over-damping the whole filtered part of the frequency response. Using a higher $Q$-value the filter under-damped instead. Using a higher number of biquads could compensate for this but that was not an option in this case.

## 5.3 Room Impulse Response Measurements using ESS

If the background noise has too much influence on the received frequency response the result in worst case is an equalization that make the loudspeakers sound worse than before the equalization. This must be avoided when dealing with products that is directed to non-professional users that is some cases can not determine this on their own.

The ESS and MLS measurement techniques had previously been tested by the company in an anechoic chamber with excellent results and therefore there was no need to evaluate the ESS measurement method regarding obtaining a correct frequency response of a loudspeaker. Instead it was decided it was much more important to test this measurement method in real life situations to see how it reacts to background noise of different types.

Figure 5.3 shows that the ESS method is very robust against typical background noise in a living room in the form of people speaking loudly, clapping their hands and the TV on at normal listening volume. In figure 5.4 the low frequency range of interest has been zoomed in. The frequency responses are very alike which confirms this.

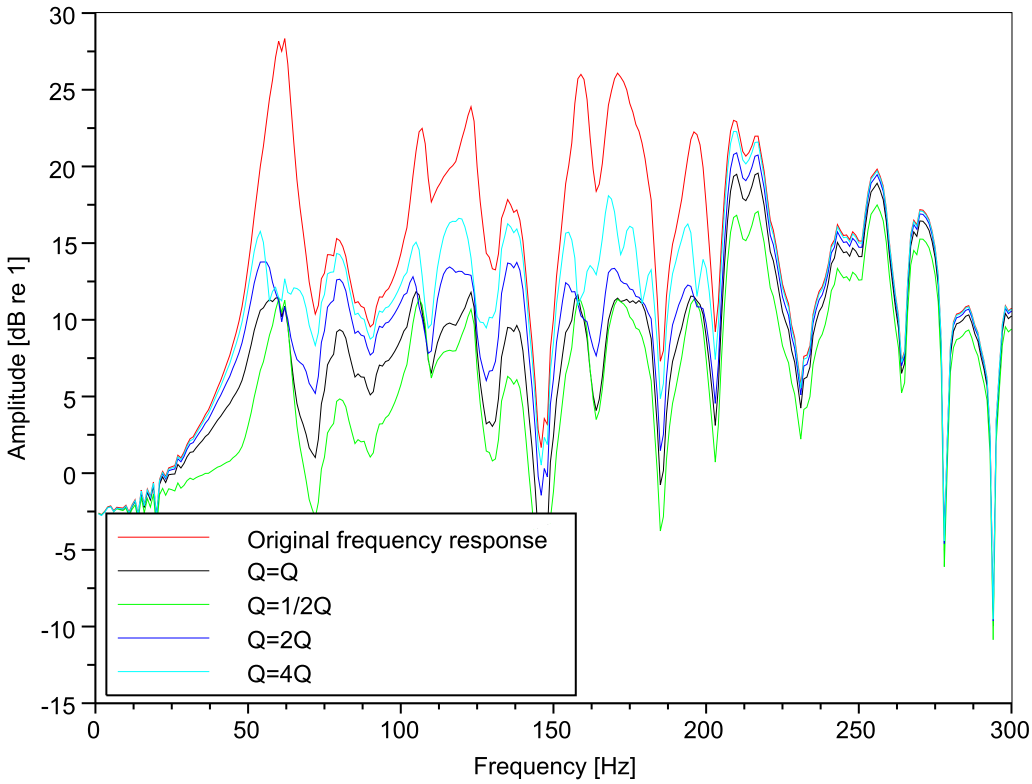



Figure 5.2: Comparison of different $Q$ scale factors. The biquad filters were applied for frequencies below $200Hz$. $Q = Q$ is the default value as stated in equation 4.3. $Q = 1/2Q$ is equivalent to half the original bandwidth and $Q = 2Q$ is equivalent to double the original bandwidth.

## 5.4 Speaker Alignment Correction

A number of test measurements were performed to see if the speaker alignment correction algorithm described in chapter 4.4.3 worked as expected. The microphone were placed at a number of positions in the room and the distance were measured with a measuring tape and then compared with the distance calculations of the algorithm. The results were conclusive and the difference in all cases were smaller than 10 cm which is well within the predetermined limits. It can be noted that it must be free sight between the speakers and the microphone for the algorithm to work.

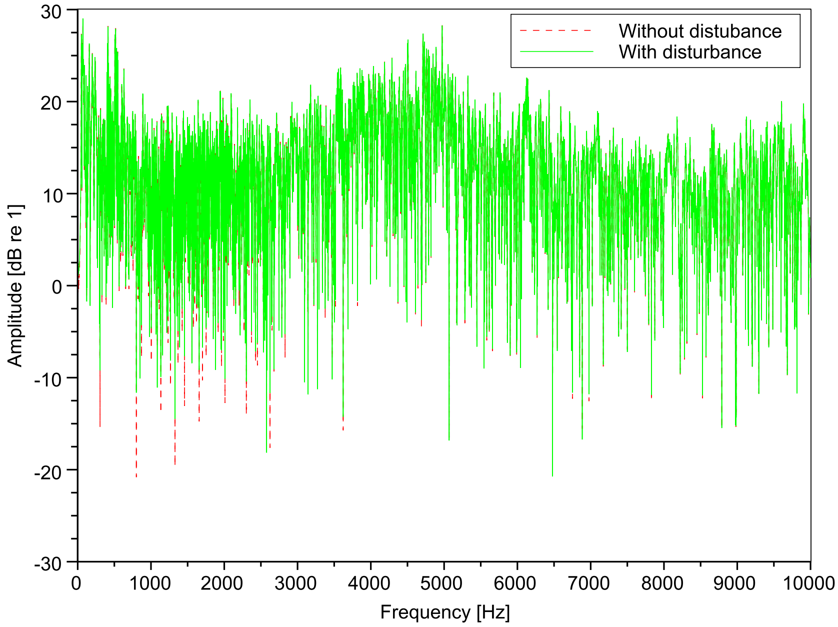



Figure 5.3: Frequency response of a room impulse response measured for 40 to 10000 Hz using the ESS method in a typical living room with minimal background noise conditions compared to a room impulse response of the same room with added background noise.

Informal listening tests were also performed at these listening positions. A song were played through the loudspeakers with and without delay and all in all cases the test persons preferred the song with corrected channel delay. Even though the channel delay was adjusted so that the sound from the two speakers arrived at exactly the same moment at the listening positing the test persons described the sound as coming from somewhere between the midpoint of the speakers and the non delayed speaker. This is hard to explain why but one theory is that when the visual position of the speakers and the arriving sound does not match the brain chooses the most logical solution. If one of the speakers is further away the brain perceives the sound as coming from the closest speaker even though it arrives at the same time. No further research was done on this phenomena because this was only a small part of the software and not a part of the thesis from the beginning.

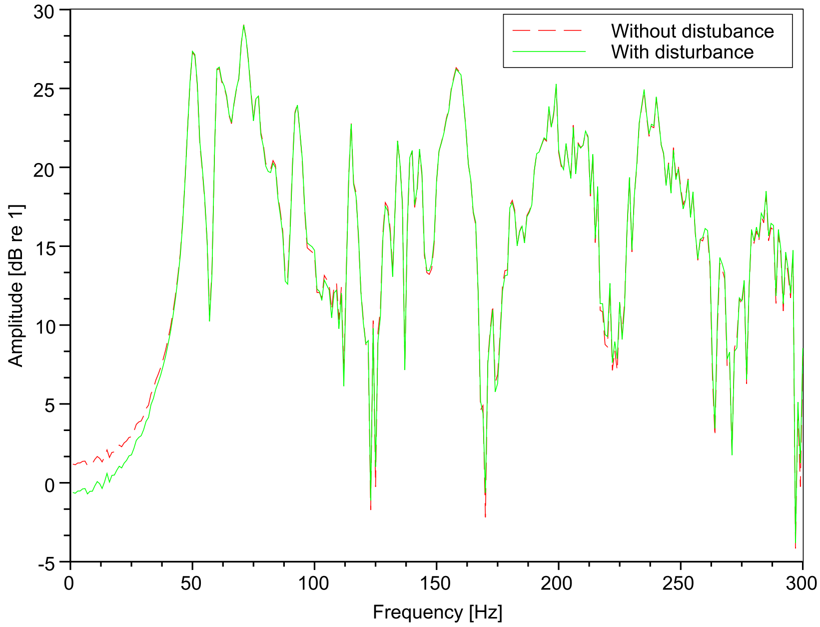


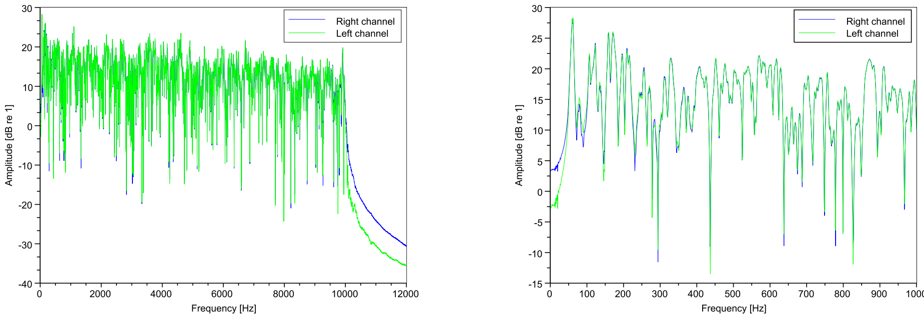

Figure 5.4: Same frequency response as in figure 5.3 zoomed in to show the lower frequencies.

## 5.5 Testing and verification of the equalization software

During the work with the software there were some key issues detected when verifying and testing the equalizer. The results of these measurements are presented and discussed here along with the results and measurements of the final and corrected software.

In figure 5.5 the measured frequency response of the office can be seen. The frequency response for frequencies under 10000 Hz is quite flat which would suggest that the room is well balanced and listening to music in the room confirms this. The room has no audible resonance peaks though some peaks can be seen in the zoomed in frequency response in 5.5(b).

The original equalizer filter algorithm applied on the impulse response of the office can be seen in figure 5.6. Looking at the zoomed in frequency responses we see that the biquad filters seems to be working as expected but the gain level of the filtered part is too low. Listening to music through the filter with same generated coefficients seems

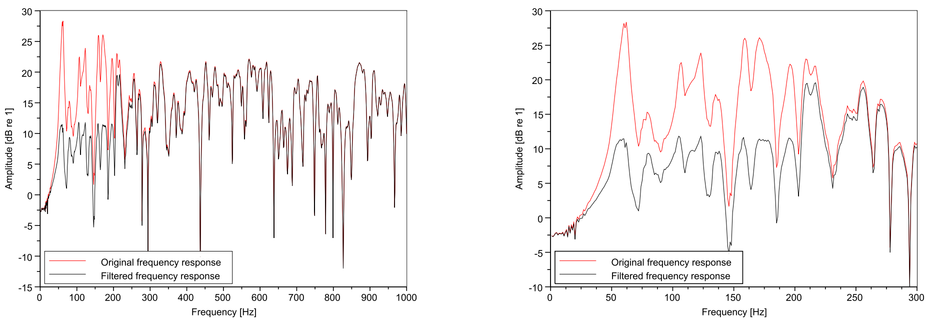



(a) Full frequency response.

(b) Frequency response zoomed in to show the low frequency resonance peaks.

Figure 5.5: Frequency response of the office measured with the ESS method for 40 to 10000 Hz. Length of the sweep was 5 seconds for each channel and the delay for the right channel was found to be 80 samples. Fourier transformation was applied to the impulse response after it was truncated to 1 second.

to support this perception. The music lacks bass and was perceived as lower in volume than the unfiltered music.




(a) Filtered and original frequency response.

(b) Zoomed in frequency responses to better show the low frequency filtering.

Figure 5.6: Original and filtered frequency response for the left channel. Filtering was made using 10 biquads with a highest filtering frequency of 200 Hz. Gain level was calculated to -11.3 dB.

From an ideal point of view the filtered part of the frequency response in 5.6(b) should be almost flat but here we see a frequency response that seems to be a compressed version of the original. There are several explanations to this. The most obvious explanation is how the $Q$-value is calculated. The software always finds the highest available peak and its center frequency but if there is two or more peaks closely together the amplitude drop may not be over $gain/2$ before the next peaks starts to build up and therefore the software calculates the $Q$-value for all peaks in the area where the gain level has not been under $gain/2$.

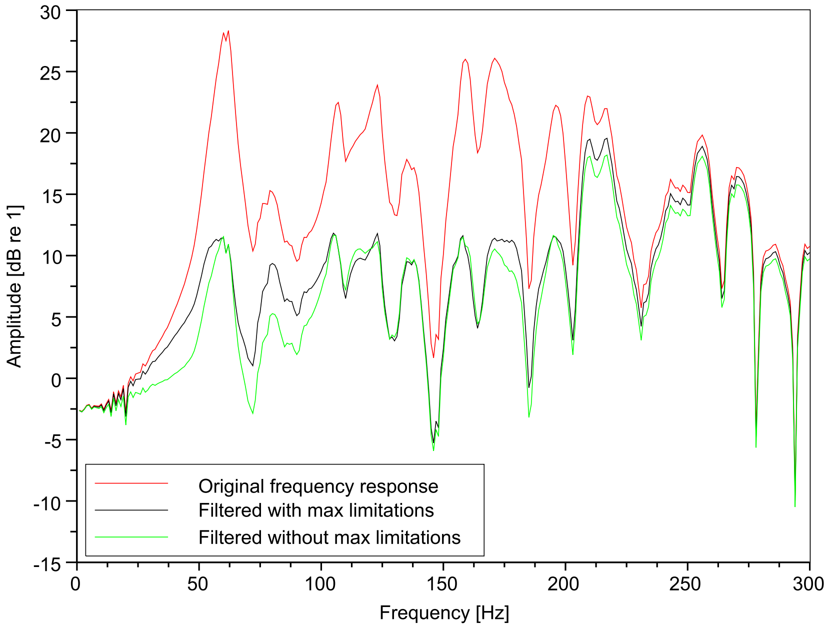



Figure 5.7: Frequency response of filter with and without restrains. The restrained filter has clearly a flatter frequency response than the filter without restrains.

In figure 5.7 a maximum bandwidth of 15 Hz restrain was applied to the filters along with a maximum gain of -15 dB restrain. This seems to have improved the filter algorithm and the frequency response of the restrained filter is noticeably flatter than the frequency response of the filter without restrains.

Another explanation is the filter shape of the biquads used. The peaks found in the frequency response are acuter that the anti-peaks of the equalization filter and therefore will frequencies outside the center frequency be over-damped. This can also result in amplification of the anti-resonances as can be seen in figure at around 150 Hz. However, this is not major problem though as narrow dips are nearly inaudible.

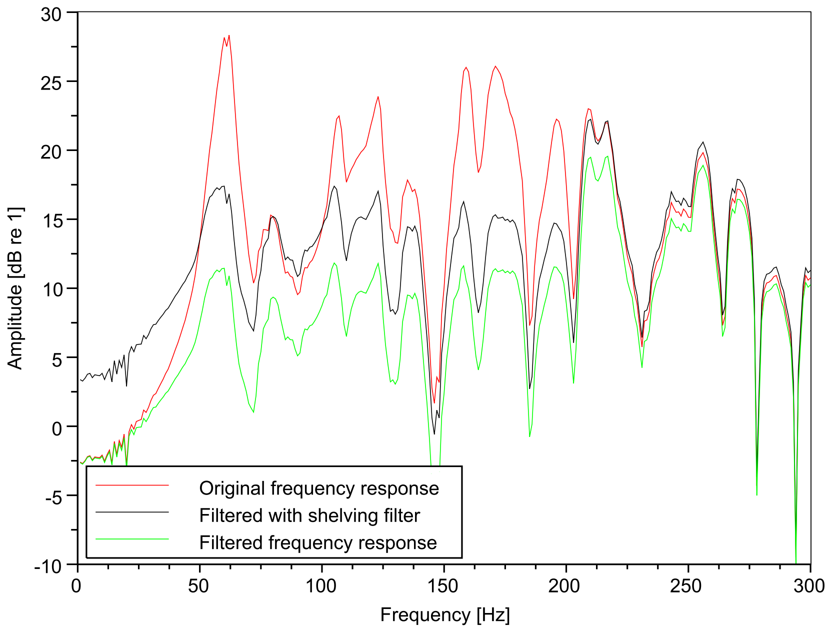



Figure 5.8: Frequency response of a filter with 10 parametric EQ biquads compared to a filter with 9 parametric EQ biquads and 1 low-shelving biquad. The shelving filter had a gain of +6dB.

The problem with the gain level is due to that the average gain algorithm does exactly what is says. It calculates the average gain and therefore the amplitudes of the filtered parts of the impulse response will be under that level. To correct for this the last biquad was used as a shelving filter instead of a parametric EQ filter. The result of this can be seen in figure 5.8.

Using the last biquad as a shelving filter does not appear to alter the flatness of the frequency response which is important. The result of using a higher gain level instead of a shelving filter here would have been a higher amplitude of the filtered part of the frequency response but also a greater difference in peaks and dips. Informal listening test using music filtered through the filter with the last biquad used as a shelving filter had good results. There was a clear difference in how the music sounded but not in volume. The music was described as having a more distinct bass and a more pleasant listening experience overall.

A interesting side effect of using these equalization filters was also found. The loudspeakers could be played at a much higher volume before audible distortion showed up which is a great advantage to the company.

## 5.6 In Situ Equalization

To verify that the equalization software worked as expected the software was tested in a standard size living room with quite poor listening conditions. The room had two opposite concrete walls with poor damping. The frequency response of the room and loudspeakers can be seen in figure 5.9.

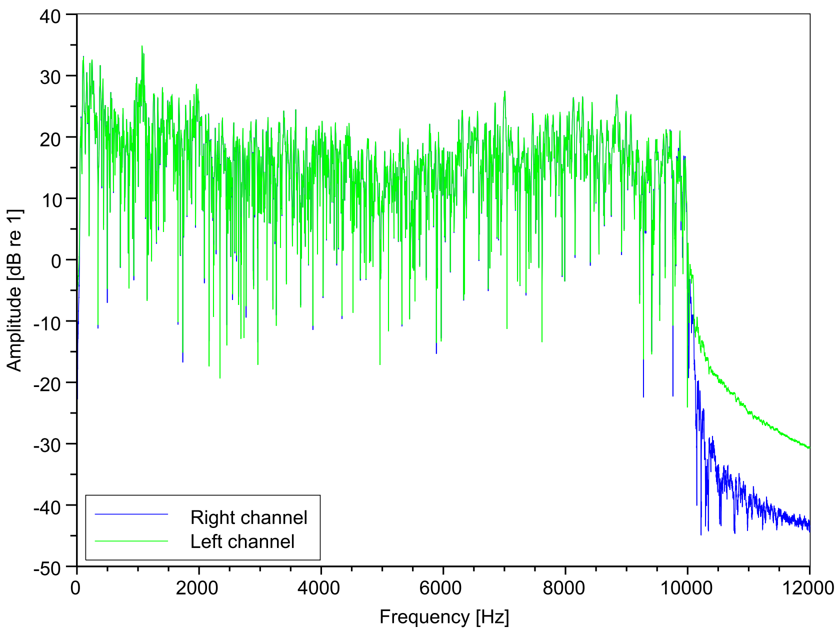
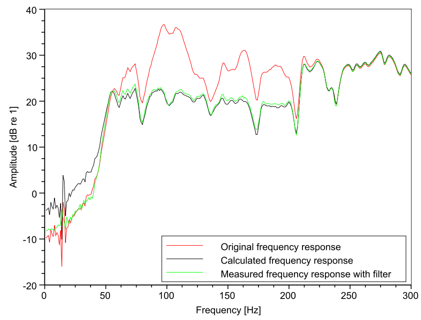
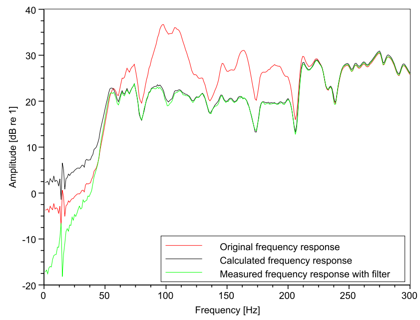



Figure 5.9: Frequency response at the first measurement position. Measured between 40 and 10000 Hz.

Instead of speakers provided by the company a small JBL stereo was used to even worsen the listening conditions. The software was used at the first listening location and the filters were adjusted for that position, see figure 5.10. For the other four listening position the room impulse response was first measured with the ESS method and then filtered with the coefficients that the software calculated for the first location. A second measurement was done but this time the ESS sweep was also filtered with the coefficients from the first position to make sure that the filter calculations matches the real measurement.



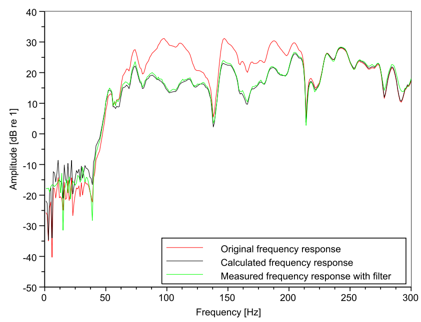

(a) Left channel.



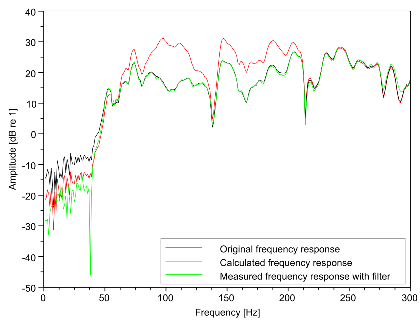

(b) Right channel.

Figure 5.10: First measurement position where the equalization software was used. The equalization filters had a highest center frequency of 200 Hz.

For all cases the calculated frequency response with filters added and the measurement with the modified ESS sweep had a very close match which is important when working with the software. If these two matches there is no need for always doing verification measurements if the filter is modified which saves a lot of time when working with the software.



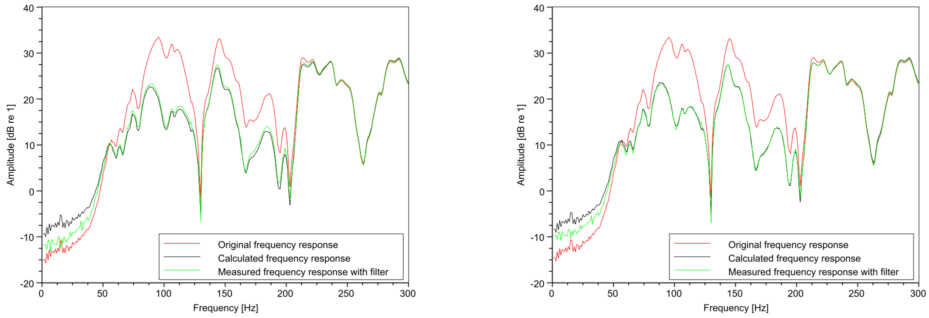

(a) Left channel.




(b) Right channel.

Figure 5.11: Second measurement position.

The first two measurements positions and especially the first one have a pretty flat frequency responses. These positions were also at the same distance from the loudspeakers and the normal listening positions in the room, i.e the sofa.

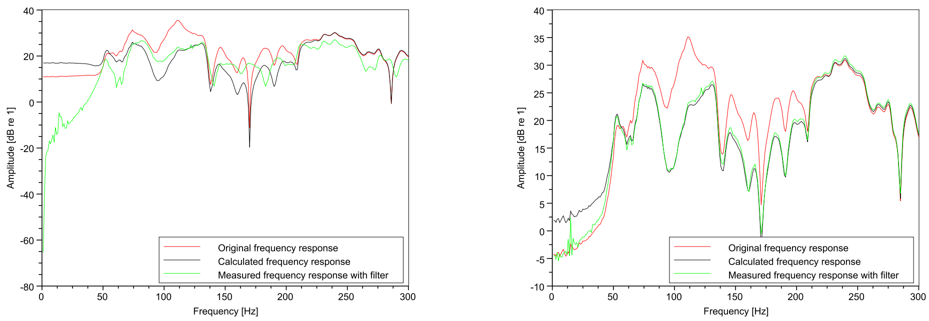



(a) Left channel.

(b) Right channel.

Figure 5.12: Third measurement position located at the centre of the room.

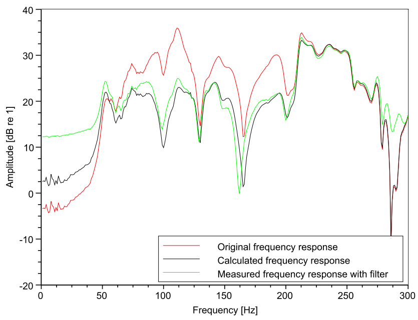
The frequency response of the last three measurements were not as flat as the first two. The fourth location measured at one of the corners in the room had the least agreement of measured calculated response though the scaling of figure 5.13(b) makes it look worse than it actually is.



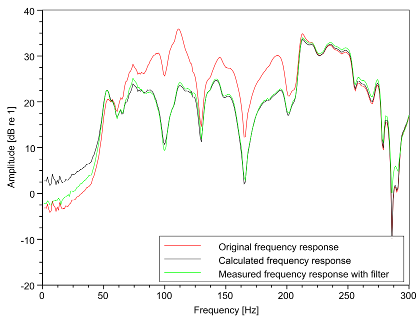

(a) Left channel.

(b) Right channel.

Figure 5.13: Fourth measurement position located at one of the corners.




(a) Left channel.




(b) Right channel.

Figure 5.14: Fifth measurement position located in the middle of the opposite wall of the stereo.

# 6 Conclusions

## 6.1 Summary

The task this thesis was to develope and if it is possible implement a fully functional measuring and filtering system in the Actiwave hardware with support for realtime filtering with low delays. This was to be done by measuring the room impulse response and then applying a equalization filter in the hardware.

A number of measurement methods was investigated in order to find the most appropriate method and both FIR and IIR filters were analyzed in order to create a good equalization filter. In the end, the test software created in SciLab could not be implemented in the hardware due to the fact that the Linux kernel on the FPGA it was supposed to run on was not functional. However, the test software shows good results and can successfully automatically equalize a pair of loudspeakers in a normal listening room. After the equalization filter was applied the low frequency region sounded clearer and the loudspeaker were less prone to distortion due to high volume.

## 6.2 Measurement Methods

Measurements with the ESS method are very prone to disturbing noise commonly occurring in a normal listening environment. Verification measurements done in this thesis support this theory which makes this measurement method s good choice. The ESS sweep is also very easy to filter and therefore control measurements are easily done.

## 6.3 Equalization and Filter Choice

Due to the already existing hardware used by the company, the IIR filter method was chosen as a FIR filter would add to much delay when added to the already existing FIR filter. Using a FIR filter could compensate both for phase and amplitude which is some cases can be a better choice when there is a need to remove early reflections.

The software implementation in SciLab is fully functional and is written in a way that it could be easily implemented in the hardware if needed. Some parts of it needs further work when the hardware is finished, microphone calibration and speaker calibration especially.

The average gain calculation should and can be replaced with a target curve instead. The shelving filter does a good job and should be kept in some form as removing peaks also to some degree lowers the amplitude of nearby frequencies. Therefore there is a need for some amount of general correction for all frequencies affected by the equalization filter.

The highest center frequency of the equalizer is for now not calculated but instead a fixed setting. Finding a modal criterion is hard and as the modal density to some degree is stochastic but the iterative equalization still works good over the Schroeder frequency and therefore this was not addressed further.

As a last conclusion informal listening tests using music played in a room with the applied equalization filter confirms that the equalizer noticeably improves the sound quality regarding a clearer bass and less distortion at higher volumes.

## 6.4 Suggestions for Further Work

The most obvious suggestion is of course the implementation of the equalization software in the hardware when the hardware is functional. Another suggestion is developing and testing of reference curves for the software to equalize against. Some kind of modified loudness curve is an example.

Another part that could be improved by further works is the optimal frequency range of equalization. By determining the schroeder frequency with the help of the room impulse response measurements, this frequency range can be adjusted for each room the software is used in which would improve the equalization further.

# References


[Gro 08] Grout, I.: *Digital Systems Design with FPGAs and CPLDs*, Newnes, Oxford, 2008

[Ron 08] Scaife, R.: *Simultaneous Audio I/O with Scilab*, Software, http://elm.eeng.dcu.ie/ scaifer/pa_wavplay/, October, 2008

[ITU 98] International Telecommunication Union/ITU Radiocommunication Sector: *ITU-R BT.1359-1 Relative Timing of Sound and Vision for Broadcasting*, http://www.itu.int/ITU-R/publications/rec/index.asp, 1998

[Ash 08] Ashby, D.: *Circuit Design: Know It All*, Newnes, Oxford, 2008

[Eve 01] Everest , F. A.: *The Master Handbook of Acoustics*, McGraw-Hill, USA, 2001

[Kin 00] Kinsler , E. et al.: *Fundamentals of Acoustics*, John Wiley & Sons, USA, 2000

[Far 00] Farina , A.: *Simultaneous measurement of impulse response and distortion with a swept-sine technique*, 108th AES Convention, Paris, February 2000

[Hol 09] Holters , M. et al.: *Impulse response measurements and their applicability in the real world*, Proc. of the 12th Int. Conference on Digital Audio Effects (DAFx-09), Como, Italy, September 1-4, 2009

[Sta 02] Stan, G. et al.: *Comparison of different impulse response measurement techniques*, JAES Volume 50 Issue 4 pp. 249-262, April 2002

[Ole 00] Olesen, S. et al.:*An improved MLS measurement system for acquiring room impulse responses*, Proceedings of NORSIG 2000, IEEE Nordic Signal Processing Symposium pp. 117-120, Kolmrden, Sweden, June 13-15 2000

[Too 08] Toole, F: *Sound Reproduction, Loudspeakers and Rooms*, Focal Press, Oxford, 2008

[Wee 07] Weeks, M: *Digital Signal Processing, Using MATLAB®and Wavelets*, Infinity Science Press LLC, Hingham, 2007.

[Ran 97] Rane Corporation: *Operator Adjustable Equalizers: An Overview*, http://www.rane.com/note122.html, Available 2010-04-08

[Ran 10] Rane Corporation: *Rane Series Equalizers*, http://serato.com/raneseries-equalizers, Available 2010-04-09

[Ran 85] Rane Corporation: *GE 30 Data Sheet*, http://www.rane.com/pdf/old/ge30dat.pdf, Available 2010-04-09

[Dbx 10] Dbx Professional Products: *DriveRack PA+ Cut Sheet*, ftp://ftp.dbxpro.com/pub/PDFs/Spec_Sheets/DRPAPlusCutsheet.pdf, Available 2010-04-09

[Kar 05] Karjalainen, M. et al.: *About Room Response Equalization and Dereverberation*, 2005 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, October 16-19, 2005

[Smi 99] Smith, J. and Abel, J.: *Bark and ERB Bilinear Transforms*, IEEE Transactions on Speech and Audio Processing Volume 7 pp. 697 - 708, 1999

[Ram 06] Ramos, G. and Lpez, J.: *Filter Design Method for Loudspeaker Equalization Based on IIR Parametric Filters*, J. Audio Eng. Soc., Vol. 54, No. 12, December, 2006

[Rub 00] Rubak, P. and Johansen, L.: *Design and Evaluation of Digital Filters Applied to Loudspeaker/Room Equalization*, AES Convention Paper 5172, Presented at the 108th Convention, Paris, France, Februari 19-22, 2000

[Mil 04] Miller, R.: *Equalization Methods with True Response using Discrete Filters*, AES Convention Paper 6088, Presented at the 116th Convention, Berlin, Germany, May 8-11, 2004

[Bri 94] ,Bristow-Johnson, R.: *The Equivalence of Various Methods of Computing Biquad Coefficients for Audio Parametric Equalizers*, AES Convention Paper 3906, Presented at the 97th Convention, San Fransisco, USA, November 10-13, 1994

[Bha 06] Bharitkar, S. and Kyriakakis, C.: *Cascaded FIR Filters for Multiple Listener Low Frequency Room Acoustic Equalization*, 2006 IEEE Intl. Conference of Acoustics, Speech & Signal Processing (ICASSP-2006), Toulouse, France, May, 2006

[Boh 86] Bohn, D.: *Exposing Equalizer Mythology*, RaneNote 115, 1986, http://www.rane.com/pdf/ranenotes/Exposing% 20Equalizer% 20Mythology.pdf, Available 2010-04-13

[Joh 07] Johansson, M.: *On Room Correction and Equalization of Sound Systems*, 2007, http://www.diracresearch.com/admin/download.asp?FileLink=upload% 2FPersonal% 2FOn_equalization_filters.pdf, Available 2010-02-18

[Sbr 10] Sbragion, D.: *DRC: Digital Room Correction*, http://drc-fir.sourceforge.net/doc/drc.html, Availiable 2010-04-13

[Kar 01] Karjalainen, M. and Paatero, T.: *Frequency Dependent Signal Windowing*, IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, 21-24 October, 2001

[Wan 07] Wanhammar, L. and Johansson, H.: *Digital Filters*, Department of Electrical Engineering, Linkping University, 2007

[Cor 09] Corbach , T. et al.: *Automated Equalization for Room Resonance Suppression*, Proc. of the 12th Int. Conference on Digital Audio Effects (DAFx-09), Como, Italy, September 1-4, 2009