Master's Thesis

Application based adaptive sound enhancement for loudspeakers

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Abstract

This master thesis presents a system for adaptive enhancement of music to compensate for disturbances introduced by the environment and imperfections in the equipment, and additional sound effects to create an optimal sound. A method is presented to estimate the frequency response of the listening environment and the loudspeaker, and to compensate for these unwanted changes of the music from the source to the listener. Tests are made to demonstrate the effects of different loudspeakers and room environments including distortion measurements, and to show that the proposed method can eliminate these effects. The results show that the method is capable to eliminating the unwanted changes to within 1 dB of the original signal which increases both the perceived quality, and the consistency of the quality of the original music signal. This thesis shows that this kind of sound enhancement is possible to realize, using an adaptive system for calibration and modification of the signal to desired sound.

Till mor och far.

Acknowledgements

I would like to thank my advisor Dr. Mikael Swartling at LTH for support and feedback, being the best advisor I could ask for. I also wish to thank Dr. Nedelko Grbic at LTH and Fredrik Bågenholm and Markus Friberg at Orlo AB, for support and feedback on my work. Last but not least, I would like to thank Martin Ekstrand for the great support throughout the whole project.

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List of Abbreviations

AR	Auto Regressive
DSP	Digital Signal Processing
EQ	Equalizer
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
GUI	Graphical User Interface
IIR	Infinite Impulse Response
PSD	Power Spectral Densities
SPL	Sound Pressure Level
THD	Total Harmonic Distortion

Chapter 1

Introduction

1.1 Background

The company Orlo AB has designed an Audio Platform using proprietary software and high quality hardware, to bring outstanding sound quality with great clarity and rich bass from very compact loudspeakers. Orlo products are made to be used in different types of environment due to its portability, which brought out the need for an adaptive system that improves the sound quality in loudspeakers. The measurements for the project took place in the acoustic laboratory at LTH as well as in normal environments, to make sure the system works properly. The software used for the real-time system was Visual Studio and Matlab with an including real-time framework, to enable processing of the signal directly from the PC hardware.

Previous studies regarding room and loudspeaker equalization have been done. According to [1] a filter generation method have been developed to be able to equalize loudspeakers in small rooms using exponential sweeps targeting the response. This method is also based on a block-processing approach in real-time using a two-step equalization for loudspeaker and environment and is due to the fact that impact of room acoustics is hard to predict regarding the sound resproduction quality of loudspeakers. According to [2] warped digital filters have also been introduced instead of usual IIR and FIR filters for loudspeaker response equalization purposes, where the warped filters enables using smaller filter orders giving a stable and robust result. A similar study of a DSP-based correction of loudspeaker and room responses has been done according to [3], where several FIR and IIR filter techniques have been proposed for equalization purposes. These studies have mostly compared different equalization techniques to create a flat frequency response from the loudspeaker and the environment, whereas the solution for this thesis implements two parts. The first part generates loudspeaker and room equalization filters and the second part implements different sound effects in a block-processing real-time system for general purposes combined, unlike no other known developed system.

A fast approach to harmonic distortion measurements has been presented in [4]. The conslusion states that by sweeping the signal through all frequencies of the spectrum for a short period of time (approximately a few seconds), there is a possibility to measure the THD at chosen frequencies from the measured sweep. This enables estimation of the THD over the whole frequency spectrum in a more efficient way, compared to the solution used in this project.

1.2 Objectives

The primary goal of this thesis is to create a DSP-system that enhances the sound quality of loudspeakers, that would improve the sound quality of several loudspeakers on the market today and to decide appropriate methods to achieve autotuning. The system is in the end, to be implemented in a mobile phone application where an automated calibration process is carried out and the calibration of the loudspeakers made. Further algorithms for sound enhancement such as an equalizer, compression limiter and bass boost are presented and explained. The questions that are to be answered in this thesis are how well it is possible to control and compensate the frequency response of a signal from loudspeakers, how adaptive algorithms such as compression limiters and equal loudness curves are implemented, how well and efficient distortion measurements could be made, and the possibility of realizing this in a PC program.

1.3 Thesis outline

The thesis is divided into five chapters, where Chapter 1 includes a description of Orlo AB, an introduction to the project as well as the objectives of the thesis. Chapter 2 contains the problem setup of the project including theory behind all the algorithms that are used, such as transfer function and power calculation of a signal, total harmonic distortion, octave filter banks, the filtering process, and equal loudness curves. The chapter also describes the design of the system and how the different functions are implemented. Chapter 3 describes the measurement procedure, how the measurements were carried out during the process and also the corresponding results. The measurements and results include characterization of environment, speakers, microphones, distortion measurements, and sound interpretation. Chapter 4 describes the conclusion of the results and how the system can be used. Chapter 5 describes how the system could be improved and developed further.

Chapter 2

Problem setup and theory

This chapter describes the initial problems that were to be solved during the project and how these were tackled, including the different setups that were used for the corresponding sections. Theory that is relevant for implementing the solutions will also be presented in this chapter.

2.1 Initial problem formulation

When playing sound through a speaker in a room the sound is reproduced differently depending on the physical attributes and abilities of the hardware, such as the loudspeaker, and the environment. These external factors are added to the original signal that is played and therefore perceived differently. By measuring the frequency response in a certain point of the room while sound is played, the total transfer function of the system can be determined. This gives an idea of how different frequencies are amplified or attenuated in that specific location, depending on echoes of the room and hardware. The goal of this part of the project is to create a flat frequency response of the loudspeakers and the environment, that will be the foundation for further sound effects to be added.

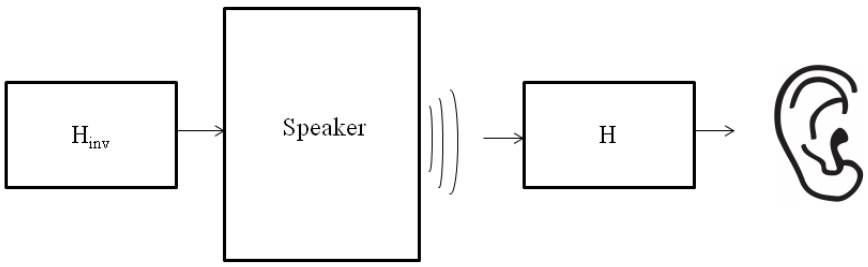

Figure 2.1: Setup of the system including transfer function H and the created compensation curve H_{inv}, that will create a flat frequency response where the listener is.

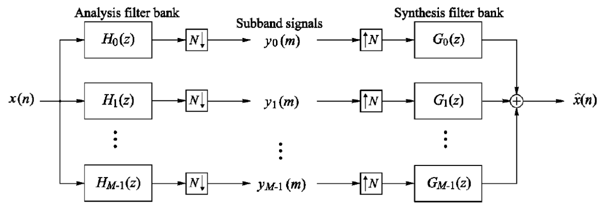

Figure 2.2: The analysis filter bank divides the signal into several narrow band-limited sub-band signals by filtering the original signal with corresponding bandpass filters.

The transfer function H is created due to different external factors, so the goal will be to create a system that calculates the filter H_{inv} that can compensate for these. The algorithms have to present a general solution, that gives the output signal a flat frequency response of the loudspeakers and the environment. As figured, the solution will be an inverse of the existing transfer function, and by filtering the output signal of the speaker with the H_{inv} filter the signal is flattened throughout the frequency spectrum.

The second task is to implement a octave-band filter bank to divide the signal into narrow band-limited sub-band signals across the frequency spectrum. When using the filter bank, the signal is processed in the frequency domain. The signal is easier to process with the filter bank, since the filtering becomes a gain or attenuation of the individual sub-band signals in the frequency domain. Additionally, multiple gain curves can easily be combined. [5]

Figure 2.2 shows how the signal is divided in sub-band signals as it is filtered through the filter bank. When this is realized, sound effects can be added by individually gaining or attenuating the sub-band signals, to create an optimized sound. Figure 2.3 shows the functions that should be implemented. Each of these functions creates a gain curve that weights the signal in amplitude, and is added to the compensation curve that is obtained from the calibration step. [6] [7]

When the compensation curve and gain curves have been applied to the signal, it goes through the synthesis filter bank to recreate the final signal which is the compensated signal with added sound effects. Figure 2.2 also shows how the sub-band divided signal is filtered into one output. The filtering of the signal is done blockwise with a block size of 2048 samples, and is processed continuously. A sample frequency of 48kHz is used due to the fact that high quality sound/music is played, and was specified from the beginning of the project. The software used for this project is mainly Matlab, with a built-in real-time framework that connects directly to the hardware of the PC which makes the solution more easily executable. The solution will also include compilation of C++ code from Matlab through Matlab Executable (MEX) external interface function, to speed up the calculations while still working with the Matlab environment.

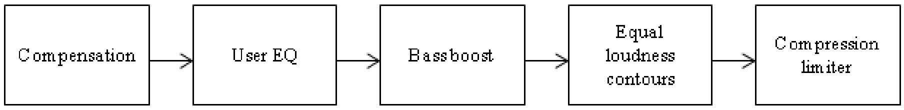

Figure 2.3: All of the gain curves that should be implemented in the system.These are sound effects that are to be added after the compensation is done.

2.2 Calibration

The function implementations of the system will consist of two parts - the calibration and the real-time process. The task of the calibration step is to estimate the total harmonic distortion of the center frequencies and estimate the transfer function of the whole system. The transfer function estimation is affected by the characteristics of the loudspeakers, the microphone, the environment, and the PC hardware components, and is made to be able to create the compensation curve H_{inv}. A measurement of the total harmonic distortion (THD) is also done in the calibration stage. It is done to set an output limit of the signal that the speakers manages to handle with respect to its hardware. The calibration part is a one-time event until the user for example moves the equipment or changes the premises in some way, creating a different environment that necessitates a new compensation curve.

2.2.1 Total harmonic distortion

The total harmonic distortion of a signal is a measurement of the distortion from harmonic frequencies. It is used to produce a more accurate reproduction of the signal in audio systems, by reducing harmonics added for example by electronics.

To be able to calculate the THD the power spectral density of the signal is calculated using Welch's method. Welch's method estimates the power of a signal at different frequencies by using periodogram spectrum estimate, obtained by converting the signal from time to frequency domain. The method is an improvement of the standard periodogram spectrum estimating method, and reduces the noise in the estimated signal depending of the frequency resolution. This is often desired due to noise caused by imperfect and finite data. Welch's method divides the data into a number of overlapping segments and allows windowing to each segment, which produces the modified periodograms that are to be averaged. This creates the power spectral density, which is used in the THD measurements to be able to more easily recognize the fundamental frequency and harmonics. [8]

Figure 2.4 shows the power spectrum of a recorded tone with fundamental frequency at 1kHz including its harmonics. The fundamental frequency has the highest amplitude, and the harmonics are its multiples. To calculate the THD of

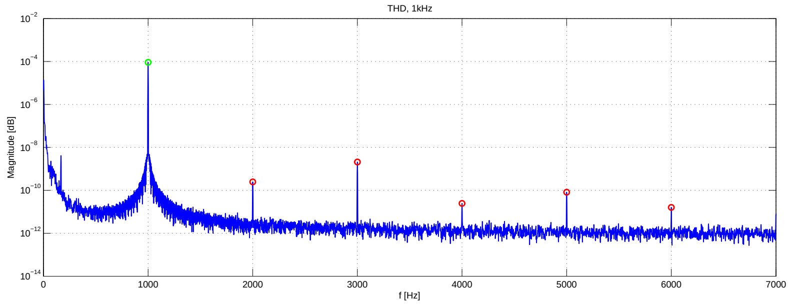

Figure 2.4: THD harmonics.

a certain frequency in decibel unit, the

$$THD_{dB} = 10 \cdot log_{10} \left(\frac{1}{P(k)} \sqrt{\sum_{n=2}^{N} P(k \cdot n)^2} \right) \tag{2.1}$$

where $P(k)$ is the calculated cross spectral density magnitude of the n^{th} harmonic in the i^{th} sub-band. Criterias of THD are often stated in percent, using

$$THD_{\%} = 100 \cdot 10^{\frac{THD_{dB}}{20}}. \tag{2.2}$$

For example

$$THD_{dB} = -33dB \rightarrow THD_{\%} \approx 2.2\%. \tag{2.3}$$

The THD measurement will be done in one center frequency per sub-band. The result of the THD will be used as a limit for the system, to avoid exceeding the actual capacity of the speakers. [9]

The THD measurement is carried out by playing and recording all the center frequencies with increasing amplitude, until a THD level larger than 10% is measured. Measurement of the THD starts with rounding the center frequencies into multiples of the used resolution, which is the sampling frequency divided by the FFT resolution. The rounded frequency will be the frequency that is played and recorded. The THD for the recorded signal is calculated according to formula 2.1, and the percentage compared to the limit of 10%. If the limit is reached, the measurement is continued with the next frequency and if not, the same measurement is done with increased volume. By the end of the measurement the vector G_{thd} has been created containing the dB-limits, when the THD is larger than 10%. [10]

2.2.2 Transfer function

The transfer function of a system is a mathematical representation of the relation between the input signal and the output signal of a linear system in terms of temporal frequency. There are several ways to estimate the transfer function of a system, such as using power spectrum or the variance of the signal when it is filtered through a filter bank. For this work the latter will be used. The transfer function of the real-time system will include the characteristics of the environment, the speaker, and the microphone as well as the characteristics of the internal system of the PC. [11]

To estimate the amplitude of the transfer function between X and Y, the ratio of the variances

$$var(X) = \frac{1}{N}\sum_{i=1}^{N} x_i^2 \tag{2.4}$$

is calculated of the input Y and the output X sub-band signals. In this project Y is the original signal, and X the recorded signal that is measured. The variance measures how the signal samples x_i are spread. This method makes the calculation of the transfer function

$$H = \frac{var(X)}{var(Y)}, \tag{2.5}$$

for each subband after the filtering of the signal through an octave-band filter bank.

2.2.3 Filter bank

An analysis filter bank consists of a set of filters arranged in a parallel bank as illustrated in figure 2.2. The filter bank is a group of band-pass filters called sub-band filters, which separates the signal into several components where each of them carries a single frequency sub-band of the signal. This is done to separate energy from a frequency region of a signal spectrum, as an ideal band-pass filter rejects all input signal energy that is outside the desired frequency range. Filter banks are used for performing spectral analysis and signal synthesis. The band-pass filters are implemented according to

$$H(z) = \prod_{k=1}^{L} H_k(z) = \prod_{k=1}^{L} \frac{b_{0k} + b_{1k}z^{-1} + b_{2k}z^{-2}}{1 + a_{1k}z^{-1} + a_{2k}z^{-2}}. \tag{2.6}$$

A filter bank with one-fifth octave band filters is used in this project and was specified from the start, which means the original signal is divided into approximately eight octaves between 60 Hz and 18 kHz, with five sub-bands for each octave, for a total of 42 sub-bands. In a one fifth-octave filter bank, the center frequencies

$$f_c[k] = 1000 \cdot 2^{\frac{k}{5}} \tag{2.7}$$

of the sub-bands, are defined relative a band-pass filter centered at $f_c[0]$=1kHz. Upper and lower band edges

$$f_{ch}[k] = \sqrt{f_c[k]f_c[k+1]} \tag{2.8}$$

and

$$f_{cl}[k] = \sqrt{f_c[k-1]f_c[k]} \tag{2.9}$$

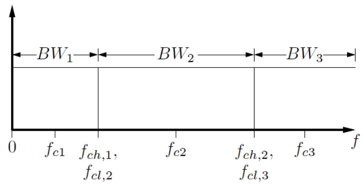

Figure 2.5: How the filter bank divides the signal in center frequencies in the frequency domain.

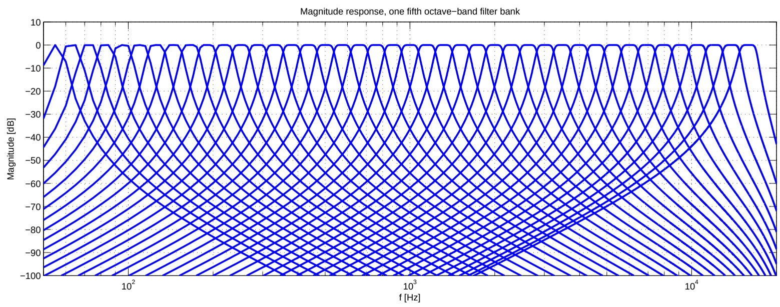

Figure 2.6: The band-pass filters of the used one-fifth octave filter bank, that filters the signal.

in the k^{th} band, are relative the centered frequency at 1kHz. Figure 2.5 shows an example of how a three-band filter bank is dividing the spectrum into sub-bands using previously mentioned formulas. From the formulas above, it can be shown that the bandwidth is

$$BW(k) = f_c[k]\frac{2^{\frac{1}{5}} - 1}{2^{\frac{1}{10}}}. \tag{2.10}$$

The quality factor Q of the filters, measures the center frequency divided by the bandwidth [12]. Since the bandwidth in formula 2.10 is proportional to center frequency, each sub-band filter is independent of k. This is the reason these kind of filter banks also are called constant-Q filter banks. Figure 2.6 shows how the band-pass filters are lined up over the spectrum. By filtering the signal with these band-pass filters, the sub-bands are created so that the signal is divided it into more managable components. Each filter passes a certain frequency interval that becomes a sub-band. The sub-bands will be weighted differently in magnitude, depending of the function that is to be implemented to create a desired output. [11]

2.2.4 IIR filter

There are two types of digital filter models: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters. The FIR filter is commonly realized with

Figure 2.7: The calibration setup consisting of a microphone, a PC, a speaker and an external sound card.

Moving Average (MA) models, described with a transfer function consisting only of a numerator polynomial. In comparison to the IIR filter that often is realized by an Autoregressive Moving Average (ARMA) model where the transfer function

$$H(z) = \frac{\sum_{k=0}^{M} b(k) z^{-}k}{\sum_{k=0}^{M} a(k) z^{-}k}, \tag{2.11}$$

where $a(0) = 1$. The IIR filter consists of both a numerator and a denominator polynomial. The impulse response of an IIR filter is infinite due to the feedback loop, but it also makes it possible for the filter to become unstable in comparison to the FIR filters which are always stable. IIR filters will be used in this project, as they are fast. [12] [13]

2.2.5 Setup

The calibration setup consists of a PC, a microphone, an external sound card, and speakers as seen in figure 2.7. The external sound card is plugged in via a USB port to the computer, and via RCA connectors to the microphone and the speaker. The reason for using an extern sound card is to eliminate additional interference and disturbances from inside of the PC, and to use a sound card with generally better flexibility and performance than the built in sound card. This is the calibration setup where the microphone is needed to make the transfer function estimation and the THD measurement. After the calibration the microphone is no longer needed and is replaced with for example a mobile phone, playing music through the framework containing the algorithms for adaptive enhancement.

The calibration function will return a compensation curve that is used when the real-time process starts. The compensation curve creates a foundation of the signal with a flat frequency response throughout the whole spectrum that will lay the foundation for forthcoming gain curves.

Figure 2.8: The setup of the real-time system.

2.3 The real-time system

The real-time system is the second part of this project. For this step the equipment consists of a PC, a microphone, an external sound card, and speakers as seen in figure 2.8. The external sound card is plugged in via a USB port to the computer, and via RCA connectors to the microphone and the speaker. As mentioned, the calibration stage is already done which means that the microphone is replaced with any device that generates a signal, for example a mobile phone playing music through the framework.

The real-time function has the main task of filtering every entering signal block with the final gain curve. The gain curve consists of weights with respect to the sub-band magnitude in frequency domain, and is a function output from the implemented algorithms. The final gain curve is the sum of the following calculated gain curves: the compensation curve, two equal loudness-curves, bass boost-curve, a limiter curve, equalizer, and microphone curve. The microphone characteristics has to be taken in consideration, since it affects the calibration. By subracting it from the total gain curve it will not affect the system.

2.3.1 Bass boost

To implement a bass boost to the system, a low shelf shelving filter is created. A shelving filter is designed to apply an equal gain change to all frequencies beyond the cut-off frequency. The filter allows low frequencies to be boosted as desirable. A first order shelving filter creates the transfer function, given certain parameters such as the gain, cut-off frequency, sampling frequency, and slope. The filter coefficients of the transfer function are computed as a new shelving filter is realized based on the shelving filter coefficient formulas according to Appendix A.

2.3.2 Subsonic filter

A subsonic filter is basically a steep slope, high pass filter in the lower frequencies that prevents undesired frequencies in the subsonic region to get to the output. The subsonic filter needs to have a steep slope in order to filter unwanted frequencies usually below 20Hz, and allow wanted frequencies through. These low frequencies are often not able to be reproduced and are not a desirable part of the audio spectrum. A subsonic filter was required from the beginning before using a filter bank, when frequencies throughout the whole spectrum had to be considered. Since this system uses a filter bank starting at approximately 60Hz, there is no need for an extra subsonic filter to be added. The filter bank only allows frequencies higher than the first band-pass filter at 60Hz and will therefore work as the wanted subsonic filter, to get rid of the frequencies in the subsonic region. This can also be seen in figure 2.6. [15]

2.3.3 Equal loudness contours

Originally, so called Fletcher-Munson curves were created and used as equal-loudness contours for the human ear. The curves present a measure of sound pressure level (SPL) over the frequency spectrum where a listener perceives the same loudness at different frequencies. The basic concept is that two tones with different frequency and the same amplitude have different perceived loudness. For example a tone of 1kHz and a level of 40dB has the same percieved loudness as a tone with frequency of 100 Hz and a level of 64 dB. Hence, there is a big difference of loudness over the frequency spectrum. Figure 2.9 shows the equal loudness contours according to Appendix A.

The unit of the curves is phons, which denotes loudness level for pure tones. The definition of the phon unit is the decibel sound pressure level of a sound at 1kHz that sounds just as loud. The fact that the difference between these curves is almost constant makes the implementation of the curves easier. This correlation will control the implementation of the curves when moving from one curve to another. Figure 2.10 shows the difference between adjacent curves, where the two diverging curves are from the two lowest equal loudness contours. These two curves are considered special case outliers and will not be taken in consideration, for the sake of approximating the equal loudness curves at higher sound pressure levels. Implementation of the curves was obtained from measured constants and sound pressure level equations. The Fletcher-Munson curve measurements were conducted in the 1930's, and a re-determination of the curves was carried out in the 1950's by Robinson and Dadson (known as Robinson-Dadson curves). The latter are now defined as the ISO226 standard, and are also the ones used in this project - therefore they will be referred to as equal loudness contours in this report. [16]

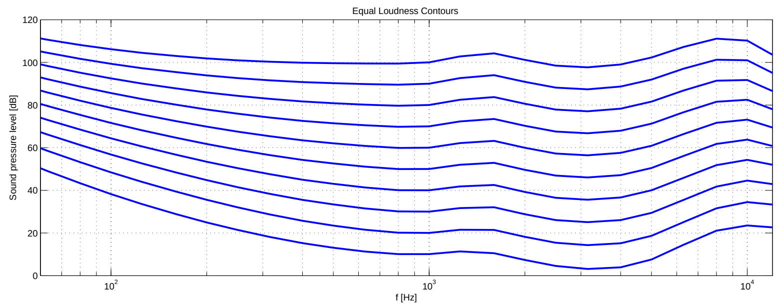

Figure 2.9: Equal loudness contours for ten different levels of phon.

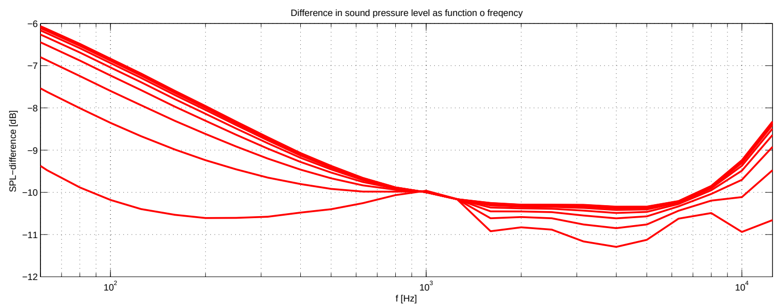

Figure 2.10: Difference of the equal loudness contours from upper to lower curve.

2.3.4 Compression limiter

The human ear is sensitive for energy changes in signals, which is the reason these variations should be handled and made subtle for the ear. Compression is used to compensate for higher levels to even out the signal, and works as a smooth attenuation combined with a limit of the output. To be able to set a limit for the output signal, an estimation of the energy in the signal is made using equation 2.12. This formula is used to estimate the momentum power of the signal, referred to as the short term power. The short term power

$$P_S(m) = \sum_{i=0}^{N} x_{b,i}(m)^2 \tag{2.12}$$

is a sum of all the samples i in a single block x_b squared. It is important to calculate, since some dynamic functions have to respond to sudden changes in the signal using the short term power to identify these variations. [18] The long term power

$$P_L(m) = P_L(m-1)\alpha - (1-\alpha)P_S(m) \tag{2.13}$$

is also calculated and used. Mainly, because of the fact that it is needed to handle changes in the signal and works as a memory of previous power values depending on the coefficient α. The signal goes through a first order autoregressive filter, using α for weighting the power depending on an increasing or decreasing signal. The constant

$$\alpha = e^{\frac{-n}{t \cdot f_s}} \tag{2.14}$$

is calculated from a given integration time t, block-length n and sampling frequency f_s. A long integration time gives a higher constant which results in a slow and stable change of the power, compared to a faster integration time. A fast integration time gives a fast change of the power and is in that sense more sensitive and reacts faster to variations in the signal. [8]

The compression ratio decides the amount of compression that should be applied, and is expressed in decibels. A threshold value is introduced to decide when the compression limiter should start compressing the signal. The compression is then made from the threshold point and above, which means that there will be no compression as long as the power is below the threshold. For example a ratio of 1.6:1, means that if the signal is exceeding the threshold by 1.6dB it will be compressed to 1dB above the threshold. It is therefore necessary to calculate the power when using downward compressors to be able to reduce the signal over a certain threshold. [19]

2.3.5 Gain curves

The microphone that is used to make the transfer function estimations has a certain frequency response that has to be accounted for, when using the compensation curve in the real-time system. The transfer function of the microphone will therefore be an input parameter to the real-time program as well as the compensation curve. The rest of the curves are either continuously calculated while running the program, or calculated only when a parameter changes as described in table 2.1.

The bass boost gain curve G_{bass}, will be the result of a created shelving filter with certain gain, cut-off frequency and Q-value that determines the slope of the curve. The volume gain curve is created using the difference in gain from one equal loudness contour to another, along with a gain parameter depending on the desired volume. The G_{vol} curve is therefore generated by recreating the volume based on amplitude scaling according to the equal loudness contours. Similarly to the volume gain curve, the power gain curve G_{power} is also calculated using the equal loudness contours. The gain curve changes according to the difference in the equal loudness curves when jumping from one curve to another. This is a dynamic curve as seen in the table, which means it is calculated continuously for every signal block, compared to the static gain curve G_{vol} that is only calculated when the user changes the volume. This is due to the fact that the system needs to know how the current signal block power behaves to be able to modify the signal before it is sent to the output.

Another dynamic curve is the power based gain curve G_{limit}, and is used to set

Curve	Description	Parameters	Behavior
G_{comp}	Created during the calibration stage.	-	Static
G_{mic}	Created from calibration.	-	Static
G_{bass}	Changes every time user changes gain or cut-off frequency.	Gain, cut-off frequency	Static
G_{vol}	Changes every time user changes volume from the GUI, according to the equal loudness curves.	Volume	Static
G_{eq}	Changes only when user changes the equalizer curve from the GUI.	Gain per sub-band	Static
G_{limit}	Changes with a given factor, every time the signal power goes beyond the threshold.	Ratio, threshold	Dynamic
G_{power}	Changes when there is a variation of the signal power, according to the equal loudness curves.	Power	Dynamic

Table 2.1: Gain curves of the real-time system.

a limit for the signal as the power increases and is also the implementation of the compression limiter. The input parameter to the function that creates this gain curve is simply a breakpoint or threshold, and a divisor that indicates how much the power should be down scaled. The compressor also uses attack- and release time to handle sudden changes in power. It is based on the integration time where the short term power is calculated. A too fast attack time can generate distortion whilst a longer time can destroy the equipment. The release time is usually longer than the attack time. A sudden increase in the signal should have a fast attack time to be able to respond with limiting the output, whilst the release time can be longer to smooth out the peak of the signal. Attack- and release times are decided by the user and recalculated into a coefficient for the power integrals that decides how much of the previous signal that should be accounted for, when creating the new one.

The user is finally given a forty-two band equalizer to edit freely, which creates the last gain curve G_{eq}, that is static and only changes depending on the users edit.

All of these gain curves are combined into one final gain curve and sent to the filtering function which gives a filtered signal as output to the speakers. These algorithms are calculated continuously as the real-time system is running, which makes it possible to play sounds from the computer input to the speaker where you get the modified filtered signal as output. The levels of the filtered output signal from the real-time program is compared to the THD measurements, and cut where the output exceeds the THD limit. This is done to prevent the actual limit of the hardware and distortion, from playing frequencies louder than the speakers allows.

2.4 Limitations

The characterization of different equipment is limited in the sense that it is difficult to characterize a single device thoroughly, such as a speaker. The reason for this is that the transfer function consists of several parts that it is affected by. This means that if not all the characteristics except for one is known, it is hard to draw any conslusion regarding accurate characteristics of a single part. The method used in this thesis will simply be a comparison between different devices, having the reference equipment plots as reference. Limitations were also made regarding the THD measurement, since this is carried out in a rather unefficient way. Usually the THD measurement is done in only one frequency (often set to 1kHz), which will not be enough in this case due to the fact that a wide frequency range of the signal is used. A THD estimation in only one point of the range would not conclude enough information about the THD for the whole span. Therefore, the goal of this part of the thesis will be to show a proof of concept by doing a thorough measurements of the THD at several frequencies to get proper results. There are several ways to estimate the THD in this case that are more efficient, but the latter method mentioned is considered enough for this kind of system. Limitations were also made in this part of the project regarding the fact that the accuracy of the result is difficult to determine.

Chapter 3

Measurements and results

This chapter contains all of the measurements that were carried out for this project, including corresponding results. All of the measurements were carried out in an echo free environment at the acoustic laboratory, if nothing else is stated. Several measurements were made to try out the implemented algorithms, focusing mainly on THD measurement, transfer function estimation and verification of the compensation as well as characterization of speakers and microphones.

3.1 THD estimation

The THD measure is done as a part of the calibration step to get information about the capacity of the speakers. This measurement is conducted by playing all the center frequencies one at a time with increasing volume. In each center frequency a THD value will be calculated. The setup for this measurement is seen in figure 2.1 using the reference microphone and speaker. The speaker volume is set close to maximum, to correctly estimate its maximum capacity. All of the center frequencies from the filter bank are played from the speaker with increasing amplitude in four steps, and recorded through the microphone. The reason for the increase in amplitude is to be able to find the THD limit, in this case when the THD exceeds 10%. The result is seen in the 'Reference speaker'-curve in figure 3.1, which shows the THD measurement of the center frequencies from one ampliude with the reference speaker.

Another THD measurement was done with a second cheaper and lower quality speaker, which corresponds to the second curve in figure 3.1. The results differs alot from the same measurement between the reference speaker and the test speaker, since the latter has higher THD at the same measured amplitude. This speaker has a poor element and its amplification is barely noticeable even though the test was carried out using its maximum volume.

The plots in figure 3.1 shows how the THD varies over the frequency spectrum. Observe that the data points above -20dB have exceeded THD limit. As this happens, there should be no further increase in amplitude of that frequency and the measurement should continue with the next center frequency instead. As seen

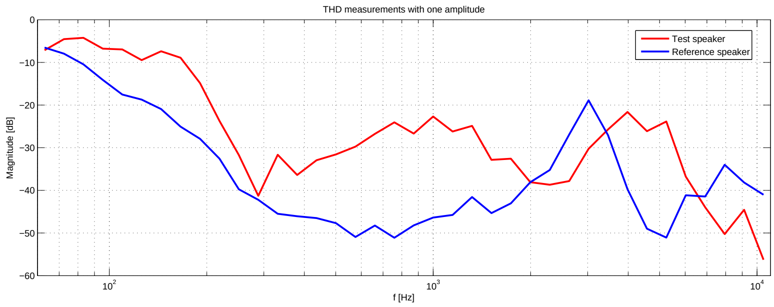

Figure 3.1: THD measurement of all center frequencies at one amplitude.

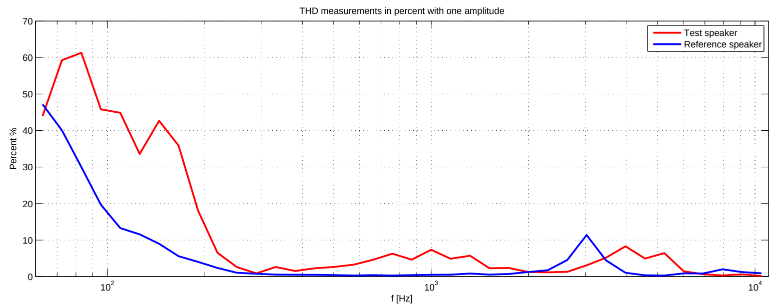

Figure 3.2: THD measurement in percent of all center frequencies.

in figure 3.1, the last center frequencies are not included, due to the fact that their harmonics are not in the current frequency range. The reason for this is the samplerate of 48kHz that is used - center frequencies above one fourth of the samplerate are therefore not measured. Since the THD is calculated in percent to check the THD-limit, figure 3.2 shows the same data with the THD in terms of percentage instead.

The result of the THD measurement shows that lower frequencies tend to generate a higher THD, especially frequencies below 150Hz. The mid-range span from 150Hz to 2kHz have relatively low THD values, which means this span tolerates a high gain without saturating the equipment or sounding bad. The frequencies in the treble region above 2kHz, generates a slight increase in THD that fades out above 4kHz to THD values below 2%. It is therefore especially important to not overboost low frequencies since they are very sensitive of an increase in amplitude with respect to the harmonics that are generated.

This becomes a way to characterize a speaker in the sense of how much they can handle without creating too much distortion. As seen in the 'Test Speaker'-curve in figure 3.1, its THD values rockets in the lower frequencies and is on average much higher compared to the reference speaker in the same figure. Only the lowest amplitudes stay below the THD limit of 10% which is the same as -20dB, but none of the lower frequencies are able to generate a THD within the stated criteria.

3.2 Transfer function estimation

By recording ten seconds of white noise played as output from the speakers, all frequencies are excited and enables estimation of the transfer function. The transfer function of the reference speaker measured in the acoustic laboratory is shown in the 'No microphone compensation'-curve in figure 3.3. This transfer function curve will be reoccurring throughout the measurement plots, and used as reference in result figures to compare other transfer function estimations.

As mentioned before, the compensation curve is obtained from the transfer function estimation in the calibration state. It has characteristics of the environment, speaker, microphone and PC. The frequency response of the microphone is known, which enables microphone compensation of the signal to make an accurate calibration. The microphone does not have a flat frequency response throughout the spectrum, and can therefore give error to the result with its characteristics. This is shown in the second curve in figure 3.3. Due to its frequency response, the microphone records a larger peak in higher frequencies compared to the reference equipment plot in figure 3.3. This makes an audible difference when compensating the signal, which makes it important to remove and compensate for. There is a difference of approximately 5dB at 9-10kHz when including the reference microphone in the transfer function, which shows that a compensation like this has to be done to not increase the error of the final output.

An example of how the calibration works in a normal environment such as a room in an apartment or an office, is presented. The plot in figure 3.4 is a transfer function estimation done in an office with an unknown speaker and the reference microphone, and has similar properties as the estimations in the acoustic laboratory. Evidently it will not look the same as the reference curve since there are many more external factors in an office that can change the characteristics of the curve. The algorithms are made to be general though, which means that this kind of calibration should give the same flat frequency response as the other measurements that were carried out in the acoustic laboratory.

3.2.1 Difference in characterization using white noise and music

All of the previous transfer functions have been obtained by using white noise as calibration signal. There is also a way to do the calibration using music, if the user would not want to calibrate the system with white noise. Figure 3.5 shows the transfer function estimation of the reference speaker and microphone, first obtained with white noise and the second curve obtained by calibrating with

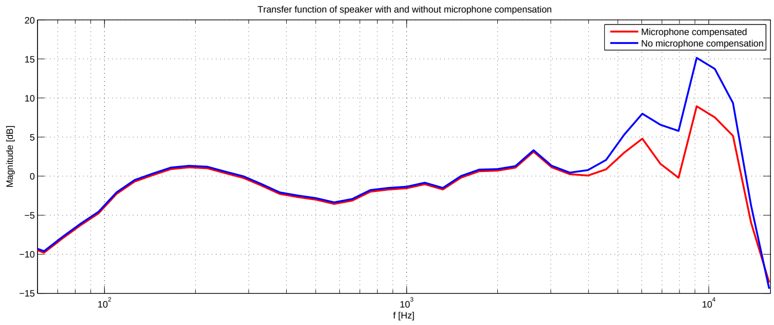

Figure 3.3: Transfer function estimation using the reference speaker and microphone.

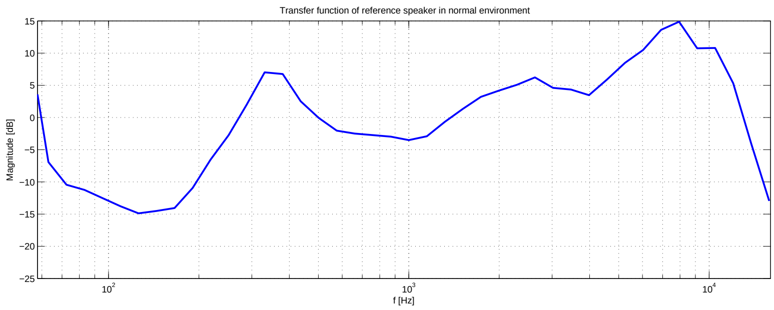

Figure 3.4: Transfer function in normal environment, normalized around zero.

music. The difference of the curves in the plot are not big, and means that it is perferctly fine to use music as calibration signal and still get good results of the transfer function estimation. This makes the calibration step easier and barely noticeable for the user, since it can be done while listening to music and then modify the signal based on the result. An important thing to mention is although the calibration process could be running in the background while playing music, the surroundings still have to be quiet to not give additional noise to the signal. If music is used as calibration signal, it is important that all frequencies are excited in that signal to be able to estimate the transfer function of the system.

3.3 Calibration verification

Now that the reference transfer functions are well known, measurements are made to try out the compensation algorithm. Starting with the reference speaker and microphone, the transfer function is measured and an inverse is created as seen in figure 3.6. The compensation curve is made using the inverse of the transfer function estimation curve, and is normalized around its mean to avoid intro-

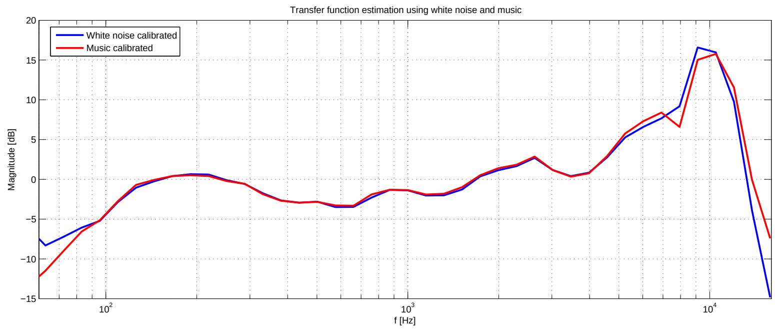

Figure 3.5: Transfer function estimations using white noise and music.

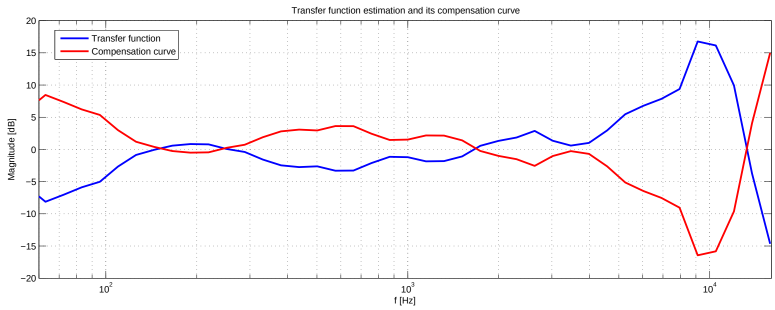

Figure 3.6: Transfer function estimation and its inverse, the compensation curve.

ducing a net gain or attenuation of the signal that is to be filtered. The compensation curve is filtered with a signal (consisting of white noise) through the filter bank and creates the compensated signal. To verify that the algorithms work, measurements are made of the filtered signal. The compensated signal is therefore played and recorded again, to measure the new transfer function after compensation. The result of the transfer function from the original signal and the calibrated signal is shown in figure 3.7.

The last verification of the compensation was done with a speaker with unknown characteristics. This compensation was done with white noise only, and the results can be seen in figure 3.8. Since the goal with these algorithms is to create a flat frequency response of the compensated output, the result is considered good. The plot shows a flat frequency response of the loudspeakers and the environment throughout the frequency spectrum, which was desired and basically the main goal of the calibration. Limits of the compensated curve were set to diverge not more than 3dB from the ideal flat frequency response, which is a fulfilled requirement with this measure. Observe that the plot of the signal is made for the whole system, which means that the characteristics of the microphone are not

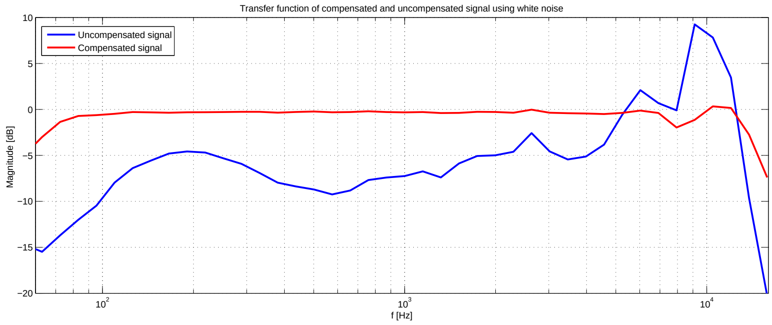

Figure 3.7: Compensated and uncompensated signal, done with white noise.

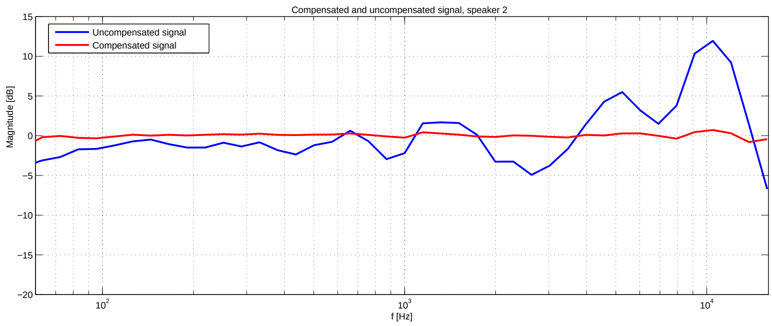

Figure 3.8: Compensated and uncompensated signal using white noise with test speaker 2.

compensated for. A compensation for the microphone would create a slight decrease of the transfer function in the treble. This is of course done in the final program, but is important to mention as the results are presented.

Consideration also has to be taken of the fact that the acoustic laboratory is a very ideal environment for conducting measurements such as these. The algorithms have proven to change the signal into what was desired from the beginning, which means it is nevertheless a proof of concept. This measurement was done by first calibrating using white noise, and then measured the transfer function using white noise. Further tests were also made (still using white noise as calibration signal) to estimate the transfer function and calibrating music instead. The result is shown in figure 3.9, and gives a similar results of the compensated signal compared to the plot in figure 3.7.

Continuously, similar measurements were done - this time by using music instead of white noise as calibration signal. Both white noise and music signal was therefore calibrated. Results from these measurements of the calibrated signal are

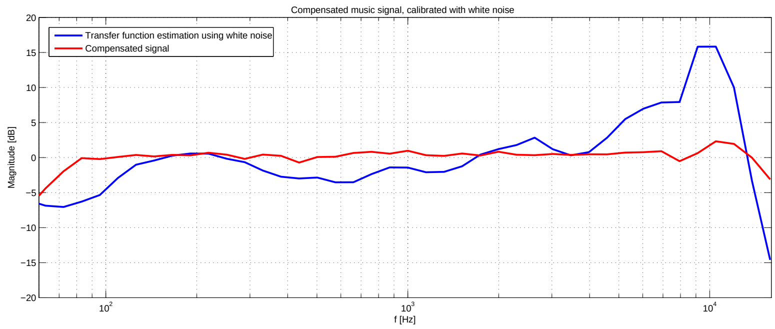

Figure 3.9: Compensated music signal, calibrated with white noise.

shown in figure 3.10 and figure 3.11.

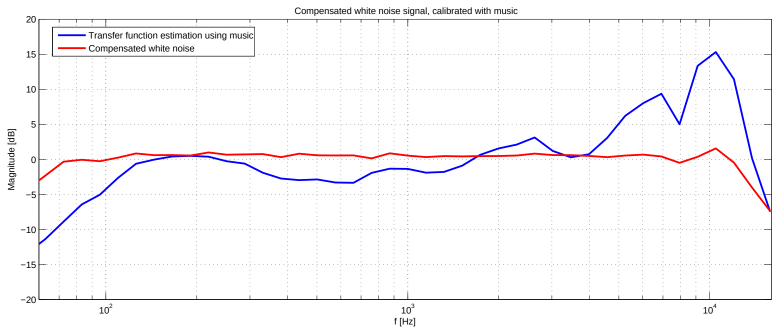

Figure 3.10: Transfer function for compensated noise using music as calibration signal.

When comparing the plot with the white noise calibrated signal in figure 3.7, the differences are neglectable since they are so small. The music calibrated signal also gives a similar flat frequency response within the wanted criterias, and a good result that emphasizes the fact that the calibration algorithms work properly. This was exactly what was desired from the algorithms - to obtain a flat frequency response after calibration using different calibration sounds. This creates the foundation of the signal to where other functions and sound effects can be added.

3.4 Acoustic laboratory characterization

To be able to continue with characterization of different speakers and microphones it is important to know about the characteristics of the acoustic laboratory. Several transfer function estimations of the acoustic laboratory were there-

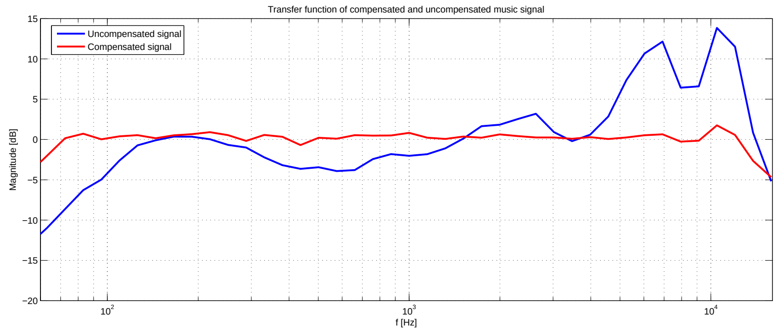

Figure 3.11: Transfer function for music calibrated music signal.

fore carried out, to be able to determine if the room has a flat frequency response throughout the whole frequency spectrum. Approximately twenty measurements were made with different positions of the speaker and microphone, with white noise used to estimate the transfer function. Figure 3.12 shows the transfer function from all of the measurements.

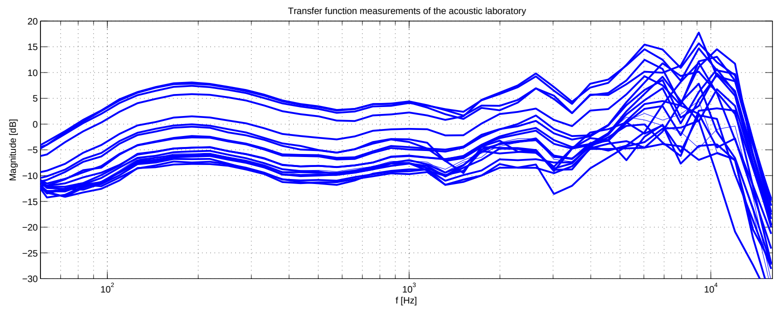

Figure 3.12: Transfer function estimation of the acoustic laboratory with different positioning of speaker and microphone.

The plots in figure 3.12 show that the laboratory environment has a relatively flat frequency response, since the transfer function is not dependent in positioning of the speakers or microphone. The curves are very similar to the reference curve, which shows that variations with different positioning are not big. Even though there are differencies in amplitude, this is due to the spacing between microphone and speaker, that changes the amplification of the signal. The important properties of this measurements is the characteristics of the transfer functions, which are very similar. It is therefore assumed that the acoustic laboratory is good for tests since it has a relatively flat frequency response and gives reliable measurement values.

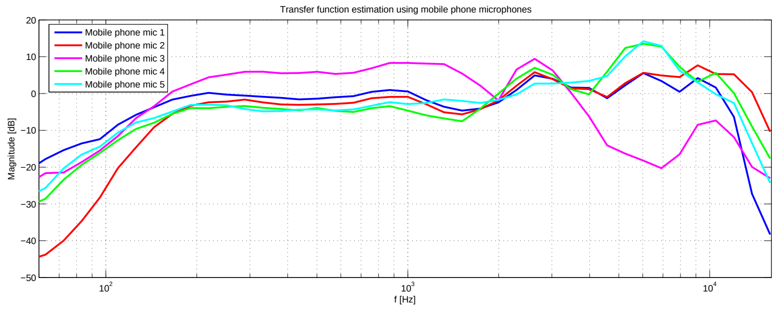

Figure 3.13: Transfer function estimations of five different mobile phone microphones.

3.5 Characterization of mobile phone microphones

As mentioned before, the characterization of the microphone has to be taken in consideration when making the calibration to approximate the compensation curve. Since the final implementation of the program is supposed to be realized in a mobile phone, measurements of different phone microphones were carried out. The reference speaker was used, along with the external sound card in a setup that is seen in figure 2.7, switching the reference microphone into the mobile phone microphone. Transfer function estimations of five microphones were done, using white noise. The tests were done using five common mobile phones from four different manufacturers. The plot in figure 3.13 shows the transfer function for each microphone tested.

The figure shows that there are some differences between the microphones in different mobile phones, which will have to be taken in consideration as the program is to be implemented in an application. Using a mean curve of the transfer functions would not give correct results even though the curves follow a certain pattern. This is due to the fact that there are variations of up to 10 dB at some frequencies that would ruin the initial calibration, and that cannot be neglected. Considering the fact that this is the result from only five measurements and mobile phones, there is probably a bigger difference if the test would include more and older mobile phones. The calibration part of the program needs the microphone characteristics of the specific device specified for it, to be able to compensate for this and make accurate measurements.

3.6 Characterization of test speakers

Several meaurements were carried out during the project regarding characterization of speakers, so two sets of test speakers with unknown characteristics were also measured. Since the previous plot in figure 3.3 gives information about the characteristics of the reference system, this can be compared to the measurements when switching speaker and/or microphone.

3.6.1 Characterization of test speaker 1

Characterization of a speaker with unknown frequency response was made to compare with the reference speaker. The same calibration setup was used as before - except for the different speaker. The plot of the estimated transfer function of the reference speaker and an unknown speaker is seen in figure 3.14.

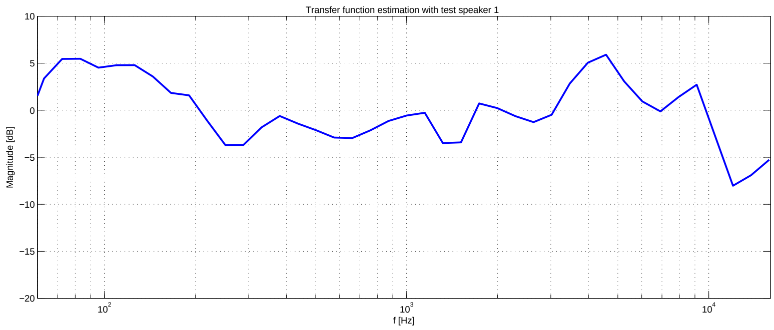

Figure 3.14: Transfer function of test speaker 1 and reference microphone.

Since the test environment is assumed to have a flat frequency response and the microphone characteristics are known, the only thing left are the characteristics of the speakers that can be compared. There is slight difference in the test speaker 1 curve comparing to the reference speaker. It does not have the same range of signal as the reference, but it has a relatively high magnitude in the lower frequencies compared to the reference speaker, and peaks at some of the middle frequencies between 1k-10kHz.

3.6.2 Characterization of test speaker 2

The characteristics of test speaker 2 were evidently unknown, knowing only that it was a studio monitor. This kind of equipment is assumed to have a flat frequency response, and is widely used in audio productions such as music recording and filmmaking. The measurement was carried out in a similar way as previously, with white noise as calibration signal. Using the reference microphone, figure 3.15 was obtained.

This plot shows that the test speaker 2 is relatively flat as one would assume, but a dip in magnitude in the mid-range frequencies characterizes it from the other speakers. The decrease in magnitude is of more than 5dB, and is an unexpected result considering the even form of the rest of the spectrum and due to the fact that it is a studio monitor.

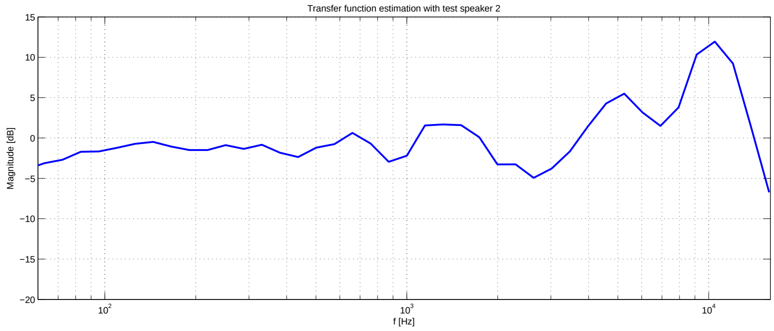

Figure 3.15: Transfer function of test speaker 2 and reference microphone.

3.7 Real-time system GUI

Results of the final GUI with the implemented functions in the real-time system are presented in figure 3.16. The first plot in the GUI shows the gain G_{total} that is the sum of all the gain curves. This user EQ can be seen in the same figure in the plot below, and makes it very easy for the user to modify the signal as wished. The different gain curves can be added and removed as wanted, by clicking the checkboxes of the corresponding function.

Some properties of the function can be modified as wanted. In the power box the user can decide and change release- and attack-time of the long term power integrator. The limit box enables the compressors to compress the signal when needed and the power box implements the equal loudness curve depending on how the power of the signal is changing. In the settings box the volume can be changed, which also changes the signal based on the equal loudness curves. The bass box allows the user to choose cut-off frequency and gain of the bass boost that can be added to the signal.

The part that is not shown in this figure is the calibration step, which is meant to be simply a startbutton that generates the process. The calibration function creates a flat frequency response throughout the spectrum of the signal and lays the foundation for forthcoming function curves. These will be added to the compensation curve, to modify the signal to what is desired of the user. In the GUI the user can decide the level of compensation in percent, and decide if the microphone should be compensated for by enabling the mic box.

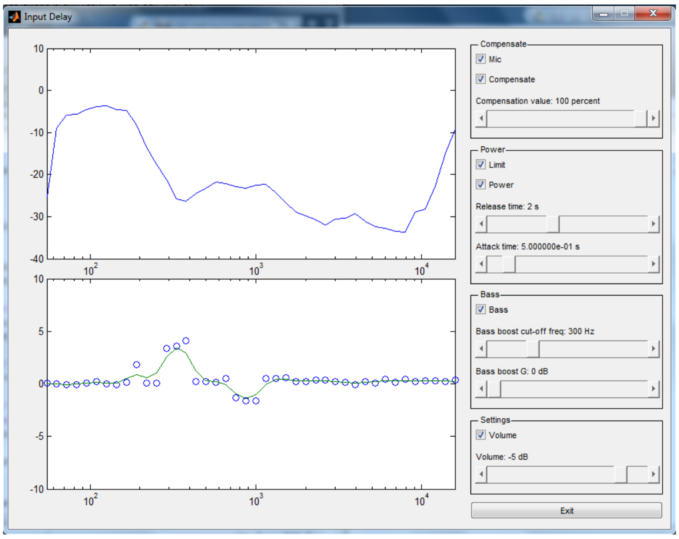

Figure 3.16: Real-time system GUI with all of the implemented functions.

Chapter 4

Conclusion

The goal of the calibration part was to be able to create a flat frequency response of the loudspeaker and the environment, within 3dB by compensating the signal being played. This was obtained as seen in the results, with an accuracy within 1dB on average. It concludes that the algorithms work as they are supposed to, in creating a flat frequency response of the loudspeakers and the environment. The compensation can be done with both white noise and music as calibration signal, which means that there are ways to simplify the calibration part. A more user friendly approach would be to run the calibration process in the background while listening to music and thereafter continue playing the compensated music. Observe that the compensation verification only was tested in the acoustic laboratory, since it is difficult to do this in a normal environment with lots of noise. The algorithms are confirmed to work in the acoustic laboratory, and are therefore assumed to work in the same way in any other environment. On the other hand there are several components to consider, that can make the measurements less accurate. The acoustic laboratory is for example assumed to have a flat frequency response based on the measurements with different positioning of speaker and microphone. This excludes the fact that the acoustic laboratory affects the transfer function estimations. When creating the compensation gain curve it is also important to have as little noise as possible, to create an accurate estimation of the environment. This is naturally very hard to achieve, but is also one of the things that are assumed in order for the algorithms to work properly. Similarly the characteristics of the microphone are important, to be able to estimate an accurate transfer function of the system. This leads to the conclusion that the given characteristics of the reference microphone, has to be accurate in order for the calibration to work - which is assumed when making the measurements.

The implementation of the different sound effects were realized as desired from the beginning regarding the equal loudness contours, the equalizer, the bass boost and the compression limiter. The level of accuracy when it comes to the equal loudness curves is not ideal, since the used curves have been estimated to forty-two points that are used in the real-time process. This leads to slight modifications in the equal loudness curves for them to fit the calculations, which are not necessarily ideal. These changes are nevertheless small and not something that the user notices, but it adds to the error and should be mentioned.

Looking at the THD measurement, the accuracy is hard to estimate since the estimation has been modified to fit this process. Problems that were faced during the measurement process was the fact that additional energy from other frequencies can coincide with the center frequencies and their harmonics, which gives an error in the THD calculation. These additional energies will most likely appear when making measurements without even knowing that they overlap, or being able to separate them from the fundamental frequency that they stem from. An example of this is the general power supply that generates the 50Hz disturbance, whose first harmonic coincides with the center frequency of 100Hz. Another limitation with the THD calculation is the fact that it only considers the five first harmonics of the fundamental frequency. This number is viewed as good enough for this project, but can give a somewhat difference in calculation. There are also several ways to improve the THD measurements and make it more user friendly and faster since the current process is not an efficient solution. Nevertheless it presents a good result and a proof of concept.

The final design of the PC program in which the real-time system is implemented, is fairly simple with possibilities to improve the design and make the calibration process automized. Certain properties that are editable of the implemented functions, are perhaps properties that should not be editable for commercial use, but for the manufacturer of the loudspeakers which in this case is Orlo. This means that the mobile phone application should have less function properties to edit, the sound effects should be implemented as default without the user being able to turn them off and the calibration should be an automized part of the system.

Chapter 5

Future work

The main idea is to have the real-time system implemented on a mobile phone application, to simplify the usage of the program. The main function is the automated calibration process, along with the built-in functions bass boost, compression limiter, volume, power, compensation curve and microphone compensation. The user will be able to decide equalizer, bass boost and/or a specific tuning made by Orlo. As mentioned before, it is important to be able to compensate for the microphone characteristics of the mobile phone microphone when calibrating. A solution to this would be to have the microphone characteristics of the most common mobile phone models in a library where the application simply can pick the corresponding curve to the current mobile phone that the system is on. This would of course require having access to the microphone characteristics of the most common mobile phone models. Further, the mobile phone is supposed to have the real-time system running as an application in the background whilst playing music, calibrated for the current environment that gives optimized sound from the loudspeakers.

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Appendix A

Formulas

A.1 Equal loudness contours

The equal loudness curves were obtained using the following equations. The sound pressure level L_p [dB] of a tone with frequency f, which has loudness level L_n is given from equation A.8.

$$L_p = \frac{10}{\alpha_f} \cdot lg(A_f) + 94 \tag{A.1}$$

where

$$A_f = 4.47 \cdot 10^{-3} \cdot (10^{0.025 L_N} - 1.14) + [0.4 \cdot 10^{(\frac{T_f + L_U}{10} - 9)}]^{\alpha} \tag{A.2}$$

where T_f is the threshold of hearing, α is the exponent for loudness perception and L_U is a magnitude of the linear transfer function normalized at 1kHz. These constants are all given from equal loudness level contour tables. The loudness level

$$L_N = 40 \cdot lg(B_f) + 94 \tag{A.3}$$

of a pure tone with frequency f which has a sound pressure level L_p, where

$$B_f = [0.4 \cdot 10^{(\frac{L_p + L_U}{10} - 9)}]^{\alpha} - [0.4 \cdot 10^{(\frac{T_f + L_U}{10} - 9)}]^{\alpha} + 0.005076 \tag{A.4}$$

according to [14].

A.2 Shelving filter

Transfer function of the shelving filter is obtained from the following formulas.

$$H(s) = \frac{A(s^2 + \frac{\sqrt{A}s}{Q} + A)}{As^2 + \frac{\sqrt{A}s}{Q} + 1)} \tag{A.5}$$

and

$$A = 10^{\frac{G}{20}}, \omega_0 = \frac{2\pi f_c}{f_s}, \alpha = \frac{sin(\omega_0)}{2Q}. \tag{A.6}$$

The filter coefficients for the low shelf shelving filter are

$$\begin{aligned}
b_0 &= A((A+1)+(A-1)cos(\omega_0)+2\sqrt{A}\cdot\alpha) \\
b_1 &= 2A((A-1)-(A-1)cos(\omega_0) \\
b_2 &= A((A+1)+(A-1)cos(\omega_0)-2\sqrt{A}\cdot\alpha) \\
a_0 &= (A+1)+(A-1)cos(\omega_0)+2\sqrt{A}\cdot\alpha) \\
a_1 &= -((2A-1)-(A-1)cos(\omega_0) \\
a_2 &= (A+1)+(A-1)cos(\omega_0)+2\sqrt{A}\cdot\alpha)
\end{aligned} \tag{A.7}$$

The final shelving filter transfer function

$$H_{Shelving}(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{a_0 + a_1 z^{-1} + a_2 z^{-2}}. \tag{A.8}$$

according to [17], [20] and [21].

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