Digital feedback control of the frequency response of a conventional loudspeaker

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Abstract

Automatic control design and Hi-Fi loudspeakers are two areas that not very often are combined. In 1976 Karl Erik Ståhl performed a master thesis project at KTH where he, with analog circuits, made a positive feedback loop to manipulate the mechanical parameters of a loudspeaker. That project introduced the idea to use control design when constructing loudspeakers. In this project this idea is pursued.

For a subwoofer, the interesting thing from a control perspective is that it is the low frequency range that has to be controlled as opposed to the high frequency range which is normally the case in disturbance and servo problems. This master thesis project will present a solution to this problem where a digital signal processor is used to handle the feed back information. The IMC controller implemented in the processor is based on models derived from data, measured in the tailor made laboratory set-up that was built for the project. In order to satisfy the sampling rate requirements, the complexity of the control algorithm had to be restricted. Despite this limitation in the equipment, the frequency response of the loudspeaker was improved significantly at low frequencies.

Contents

Chapter 1

Introduction

1.1 Background

This Master's Thesis Project started as an idea of improving the sound experience from a regular loudspeaker with the help of control design. A loudspeaker has always a characteristic that is similar to a highpass filter. This makes it very hard to reproduce frequencies that is close to the lower limit of what the human is capable to hear. In general a larger speaker cabinet is able to play lower frequencies than a smaller cabinet, if the same radiator is used. The idea of this project is similar to a master thesis project preformed in 1976 by Karl Erik Ståhl [Stå76]. Ståhl used analog circuits and positive feedback to manipulate the mechanical properties of a loudspeaker. Ståhl finished his invention and got an international patent for it and the technique is today called ACE-bass[1].

One of the main differences with this project compared to Ståhl's is the fact that this project uses a digital signal processor to make the control design. By making it digital, the equipment can be made smaller, cheaper and contain a more complex control algorithm than can be made with analog circuits.

The win of making a control system for a loudspeaker is that the speaker cabinet can be made smaller but still have the same sound characteristics as a larger one.

[1] Read more about it on http:\\www.audiopro.com\products\subwoofers\tekn_acebass.asp

1.2 Aim of thesis

The first aim of this thesis is to design and build a loudspeaker that is suitable for this project. The speaker cabinet is chosen to be the smallest possible to fit the radiators. This gives the loudspeaker the desired characteristics, i.e a loudspeaker that has a obvious highpass frequency response equipped with radiators that has a long stoke length. The stroke length is needed to compensate for the small cabinet.

A final aim was to build a controller for the woofer radiator and thereby change the frequency characteristics of the loudspeaker.

1.3 Thesis Outline

Chapter 1 describes the background and the aims of this thesis. Then chapter 2 goes through the theoretical details of the different thesis parts, such as control theory and speaker theory. To get insight in the system that is discussed, the system is described in chapter 3. The reader just gets a rough overview of the system from reading this chapter. To get a better description, reading [JA03] is recommended.

Chapter 4 and 5 is the core of this report. These chapters covers modelling the system respectively the results. Chapter 5 includes, besides results, the implementation part that appeared to affect the final result in a major way. In chapter 6 future work is discussed. Chapter 7 includes conclusion and acknowledgements.

Chapter 2

Theory

The ultimate loudspeaker has completely flat frequency response that stretches over the whole frequency spectra that the human ear can hear, i.e 20 Hz to 20 kHz. It is very hard to achieve a relatively small loudspeaker that can reproduce the lowest frequencies of the human hearing spectra.

2.1 Speaker theory

In this master thesis a loudspeaker is looked upon as a system that has an input signal and a output signal, affected by nothing more than a transfer function like $Y(s) = G(s)X(s)$. The transfer function describes the frequency response for a specific loudspeaker.

The appearance of the describing transfer function $G(s)$ depends on many different physical factors such as its volume, radiator, bass reflex port, humidity and so on. Every loudspeaker has its individual characteristics that not necessarily has to be linear. The unlinear part is not discussed in this project.

A common loudspeaker consists of two main contributors that shapes the frequency response. These are the cabinet and the radiator. Depending of the size, design and volume the characteristics of a certain loudspeaker can differ a lot.

2.1.1 The cone speaker

To be able to get a mathematical theoretical model of the speaker it is important to identify each and every physical component of every single part of the system. The function of a speaker is represented in three different domains; the electrical, the mechanical and the acoustic domain. A simple way to describe the domains is that the source from the amplifier is electrical which makes the speaker move mechanically and then make an acoustic sound. In these three domains, every physical part has to be identified and then transformed into one of the domains to get a complete system to derive a theoretical mathematical model from. When drawing a scheme of the systems, electrical components are always used to describe the components in the different domains.

The loudspeaker radiator that is used in this project is a cone speaker. For such a speaker the mechanical impedance consists of mainly three components: Mass, suspension and resistance. The mass originates from the mass of the cone and coil. There is also a contribution to the mass from the air that is moving as the cone is moving. The suspension originates from the mounting of the cone, and is concentrated to the periphery and the center of the cone. The resistive losses of the cone is also a product of the mounting. The attenuation that the air provides in the magnetic gap is also a source of resistive losses. An electric analogy scheme representing the mechanical system is presented in Figure 2.2. The impedance in the electric domain consists of two components, resistance and inductance. These originate from the coil. The electrical scheme is represented in Figure 2.1. The acoustical domain is shown in Figure 2.3.

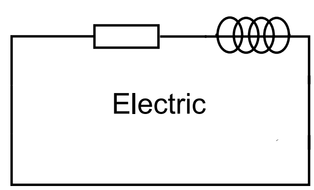

Figure 2.1: Electrical scheme.

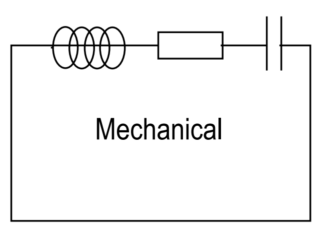

Figure 2.2: Mechanical analogy scheme of a cone speaker

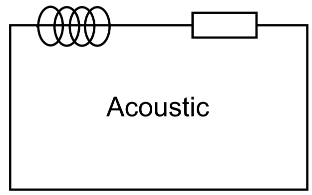

Figure 2.3: Acoustical analogy scheme.

2.1.2 Analogy for the transformed domains

The electrical schemes 2.1, 2.2 and 2.3 have to be transformed into the same domain to be able to be put together the whole system.

There are two ways of modelling the transition between two domains: by flow or by intensity. The best way to think of this is to look at it as energy floating from one part of the system into other parts of the system. In the electric domain this corresponds to the relation flow and electric current, intensity and voltage. In the mechanical domain the flow is velocity and the intensity is force, and in the acoustic domain we have that the flow is acoustic flow and the intensity is pressure. When the connection between the elements are of flow type the transition of energy is made by a gyrator, and when the connection between the elements are of intensity type the transition is made by a transformer[1]. In the case of the cone speaker, the transition between the mechanical and the acoustic domain is of intensity type and therefore a transformer mediate the energy. Between the electrical and the mechanical domain, a flow is supplying the connection between the domains and therefore a gyrator is the component here.

The resulting scheme is describing the transfer function from electrical signals from the amplifier to the sound pressure created from the cone. The transitions are quite complex and are further described in [Gus04], page 15 to 22. The model described with the schemes above just take the cone speaker in consideration. As seen below many more components and have to be considered if the speaker box is included in the system.

2.1.3 Ported Cabinet

The ported loudspeaker cabinet has besides the opening where the radiator is mounted, a second opening, the port. The port creates together with the volume a Helmholtz resonator that is driven by the backside of the radiator.

[1]Moore about this in [TG91]

At the specific frequency where the port and cabinet has a resonance, the impedance has a maximum. This frequency is called the Helmholtz frequency. The maximum of the impedance gives that the velocity of the radiator has a minimum. This is valid no matter what the resonance frequency of the radiator is. In this stage the main sound pressure comes from the port and just a fraction from the radiator. Compared to a closed cabinet, the ported cabinet achieves the same sound pressure with smaller radiator motion and lower distortion. This is the main advantage for the ported cabinet. Figure 2.4 shows a ported cabinet with a single radiator mounted in it.

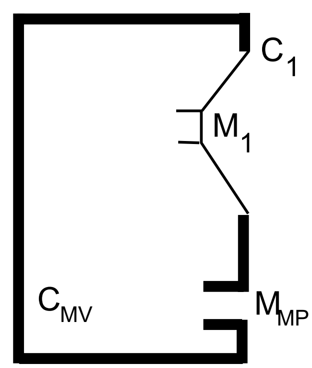

Figure 2.4: A ported cabinet with the cone speaker mounted into it.

To be able to understand the Helmholtz resonator it can be a good idea to study the mechanical analogies. A common ported box like in Figure 2.4, has an electrical analogy as can be seen in Figure 2.5. The port has an acoustic mass M_{AP}, which corresponds to the mechanical mass $M_{MP} = M_{AP}{S_P}^2$, where S_P is the opening area. The acoustic mass is affected by the pressure inside the cabinet which is equal to the force on the mechanical mass and the cabinets mechanical capacitance. The port does also have an resistive component R_{MP} that is affected by the same velocity as M_{MP} and therefore placed in series with it [Gus04].

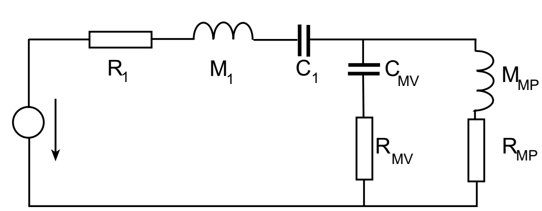

Figure 2.5: Mechanical analogies for the speaker in Figure 2.4.

If trying to reproduce lower frequencies than the Helmholtz frequency, there will occur an acoustic short circuiting between the radiator and the port. This because the radiator and port is in phase with each other and the net result is then practically zero. More about speaker theory can be found in [Lil93].

2.2 Control theory

Controlling a loudspeaker radiator is quite different from common control design. The thing that differ this system from usual systems is that it is the lower frequencies that are to be controlled and not the high frequencies that is the common case.

To be able to control the speaker with a closed loop, a feedback signal has to be fed back from the output. The obvious solution is to have a microphone in front of the radiator, in order to feedback the sound pressure from the loudspeaker. There are two specific problems with that solution. The microphone would be very sensitive to reflections of the sound waves and the output signal would therefore not only describe the frequency response of the loudspeaker but also the room characteristics. The controller would then get a really hard time if the sound pressure from the speaker was low compared to the rest of the noise in the room. A more serious problem with the microphone solution would be the delay that would effect the bandwidth of the system negatively. If the delay is introduce into a system with feedback, it will get a severe negative phase shift. This will give a reduced phase margin. If the delay is large compared to the desired cut frequency, the stability of the system can be hard to achieve. The delay would of course depend of how close the microphone is to the radiator but if a single microphone is meant to cover both the active and passive radiator, the distance to each of them has to be equal and of consideration.

The optimal solution is of course a signal that represents the output from the speaker, that is not delayed and is not changed by the room characteristics or any other non-static source. As mentioned in Section 2.1 the impedance of the radiator is depending of the frequency, therefore changes the current through the speaker in relation to the sound pressure. The natural way to get a feedback signal was therefore to measure the current that flows through the radiator and then use that signal.

The natural way to describe the system was therefore as made in Figure

2.6. The input signal to the system is a voltage, coming from a source such as a CD-player, and transformed into a current that is measured and represented as voltage that is used as the feedback signal.

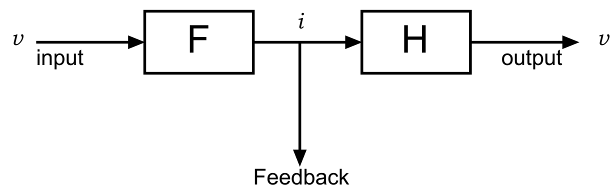

Figure 2.6: Schematics of the open loop and the feedback signal.

Because of the system configuration H can not be manipulated by a feedback. To be able to compensate for H, feed forward control, which in fact means a static filter, is used for the desired output. An obvious solution way for the total system to was therefore to create a control loop based on the system in Figure 2.6. Figure 2.7 shows how the final control design was made and what the system looked like. When the current in the closed loop is measured, a voltage is retrieved that is proportional to the current.

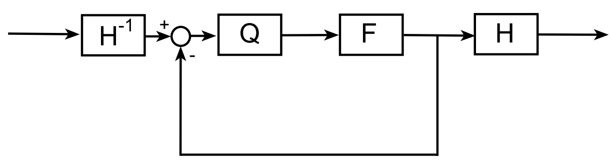

Figure 2.7: Theoretical system

2.2.1 Internal Model Control

The control design used for this project was Internal Model Control (IMC). The IMC method demands a model G of a stable system G_0, see Figure 2.8. The structure of this control design is made so that only the difference between the modelled output and the actual output is fed back.
The error signal that is fed back is according to figure above:

$$w = y - Gu \tag{2.2.1}$$

This signal is then fed back via a transfer function Q and added to the reference signal:

$$u = -Q(y - Gu) + Q\tilde{r}, \quad \tilde{r} = \widetilde{F}_r r \tag{2.2.2}$$

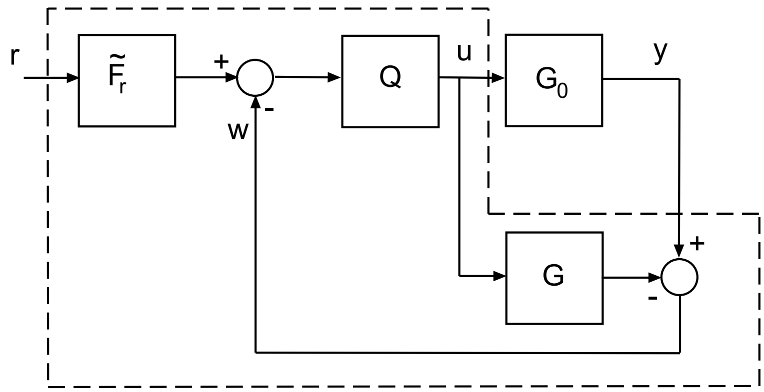

Figure 2.8: Feedback with the model error

The name Internal Model Control comes from the way of describing and calculating the controller in terms of feedback via Q of the new information $y - Gu$, where the model is included in the signal. This demands the model to be a good approximation of the real system.

If disregarding the reference signal, the transfer function from u to y will be:

$$u = \frac{-Q}{1 - QG} y \qquad \Rightarrow \qquad F_y = \frac{Q}{(1 - GQ)} \tag{2.2.3}$$

If the model is identical to the real system, i.e. has no disturbances or model errors, the closed loop will be:

$$y = [\tilde{F}_r r - u(G_0 - G)]QG_0 \qquad \Rightarrow \qquad G_c = \tilde{F}_r QG \tag{2.2.4}$$

This gives the sensitive and complimentary sensitive function:

$$S = 1 - GQ \tag{2.2.5}$$

$$T = \frac{GF_y}{1 + F_y G} = GQ \tag{2.2.6}$$

The transfer function from the output error signal w to u then becomes:

$$G_{wu} = -Q = \frac{-F_y}{(1 - F_y G)} \tag{2.2.7}$$

The ideal choice of Q is $Q = G^{-1}$ which would give $S \equiv 0$ and $T \equiv 1$, but this is impossible since that would make $F_y \equiv \infty$. $Q = G^{-1}$ is still to strive for. More about IMC and how to design the controller can be studied in

[TG97].

When making a controller that shall be implemented in a digital signal processor, it is necessary to build a control structure without algebraic loops. The loop that contains the modelled system is such a loop. The implementation technique is to rearrange the control loop and get a new controller that has the exact same function as the original. The rearrangement for the IMC controller can be seen in Figure 2.9.

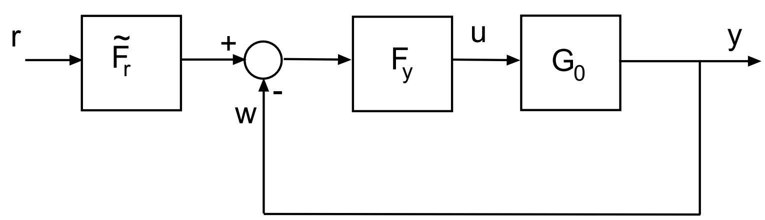

Figure 2.9: Rearranged IMC controller

The new controller F_y has the transfer function:

$$F_y = \frac{Q}{(1 - QG)} \tag{2.2.8}$$

2.2.2 Transfer function minimization

If having a model that has a higher order than by any reason is to large to be used. This could for example be the case if a controller is going to be implemented in a digital signal processor.

The model of the true system is here referred to as F_2 and the controller that is wanted to reduce is called F_y. The desired reduced controller is referred to as C_y. All parts of the system is discrete counterparts of continuous systems. The minimized controller C_y is the one that minimized the expression:

$$\min_{C_y} \int_{-\pi}^{\pi} |\frac{F_2(e^{i\omega})C_y(e^{i\omega})}{1 + F_2(e^{i\omega})C_y(e^{i\omega})} - \frac{F_2(e^{i\omega})F_y(e^{i\omega})}{1 + F_2(e^{i\omega})F_y(e^{i\omega})}|^2 dw$$

This is made by creating both a new input signal and an output and then use an identification toolbox to try to get a model that has a lower order that the previous one. The white noise that was used to derive the previous model is used as input to get the signal u:

$$u = \frac{F_2}{(1 + F_2F_y)^2} e$$

$$y = C_y u$$

The signal y is the same output signal that was used when calculating the model that the controller was based upon. The signals u y can now be used with e.g Ident, the Matlab identification toolbox.

Chapter 3

System Description

This master thesis project is breaking new ground in such areas as combining control design and loudspeaker building. A tailor made laboratory set-up was built for this purpose. This makes it inevitable to configure a system, custom made for this project. The system is mainly constructed by three parts; the loudspeaker cabinet with its radiators, a current measurement device, the amplifier and a Digital Signal Processor. Because of the special needs for this project the loudspeaker cabinet was made from scratch with suitable radiators, mounted in it. Also the current measurement device had to be exclusively developed.

The components are put together in a quite complex structure where the DSP is the hub of the creation. A schematic of the whole system can be seen in Figure 3.1, where the part inside the dashed line is implemented into the DSP.

More about the system can be read in [JA03], where all parts are explained in detail.

3.1 The Loudspeaker

One of the aims of this thesis is to build a loudspeaker that has the suitable properties for the ways and means of this project. The desired loudspeaker frequency response is highpass characteristic with low sound pressure reproduction in frequency range from 20 Hz to approximately 80 Hz. This was achieved by mounting quite large radiators with long stroke length in a relatively small cabinet.

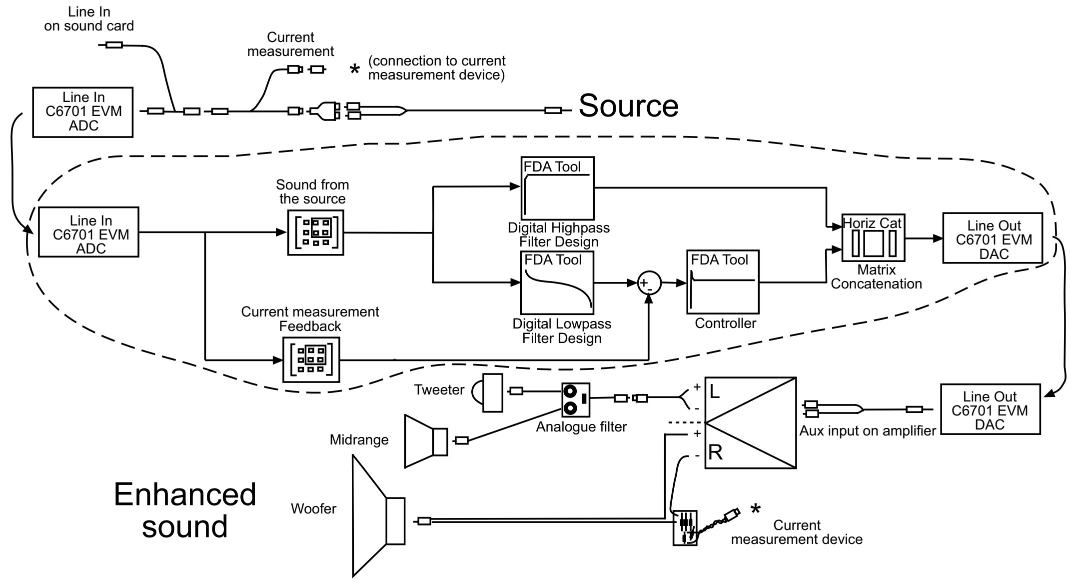

Figure 3.1: Schematics of the total system.

When listening to a subwoofer only makes it hard to hear if the sound is good. In order to make the sound impression complete, a tweeter and midrange was also included in the loudspeaker solution.

3.1.1 Cabinet

When making a quite small cabinet with subwoofer properties it is common to use a ported cabinet, i.e a box with a tube mounted in to it. The diameter and length of the tube is depending on the radiators, size of the cabinet and wanted resonance frequency for the loudspeaker. In the case when it is desired to have the cabinet as small as possible and with relatively large elements, the tube would get several meters long which obviously would take up a lot of volume in the cabinet. A ported box with a tube is therefore a not so good choice in this speaker design.

An alternative for using the tube is to have a passive radiator, which means a radiator without the coil or magnet. The passive radiator consist of merely the cone that can be loaded with extra weight to correspond to an extra long tube. This passive element is sometimes referred to as slave radiator.

The speaker configuration became a box with two cavities, one that holds the midrange radiator and the tweeter, the other holds the active and the passive woofers. The woofer cavity has an inner volume of 16 liter. See Figure 3.2 where the cavities are shown without the lid. The two cavities are hermetically separated from each other.

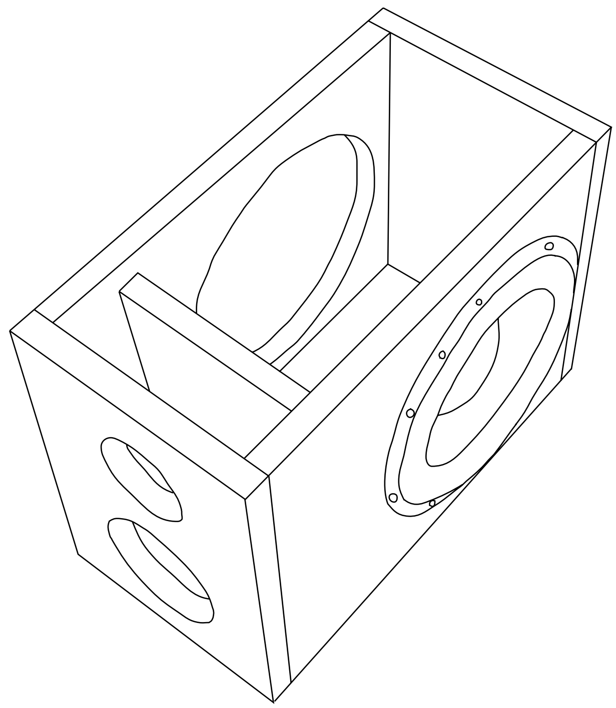

Figure 3.2: Blueprint of the speaker cabinet

The tweeter and midrange are mounted in the front of the box and the active and passive radiator on opposite sides of the main cavity. The front cavity is designed suitable for the midrange demands so that a flat frequency response as possible can be obtained.

3.1.2 Radiators

The choice of the tweeter and midrange was based on the suitable frequency range and the flatness of the frequency response in that range. For the tweeter, the choice fell upon a SEAS H535 25TAC/D and the recommended frequency range is 2000 Hz to 25000 Hz. The midrange radiator has a recommended frequency range between 100 Hz to 4000 Hz, and is a 5" SEAS H422 MP14RCY. There is an external analogue filter that separates the frequencies feeding the tweeter and the midrange loudspeaker. The crossover frequency is at 2500Hz.

For woofer radiator, the long stroking 10” Peerless SWR 269 with corresponding passive radiator, was chosen. This constellation together with the cabinet, a Helmholtz resonance at 26,3 Hz was achieved.

The data sheets concerning these radiators are found at the back of the loudspeaker manual [JA03].

3.2 Current Measurement Device

The signal that is used for feedback is the current that flows through the woofer radiator. In order to get a measurement of the current to the digital signal processor the level of the voltage has to be below 3.3 volts because of the maximum input to the DSP. The current that flows through the radiator is relatively high, so therefore the resistors that are used must be able to handle large amounts of power. The analogue design of the current measurement

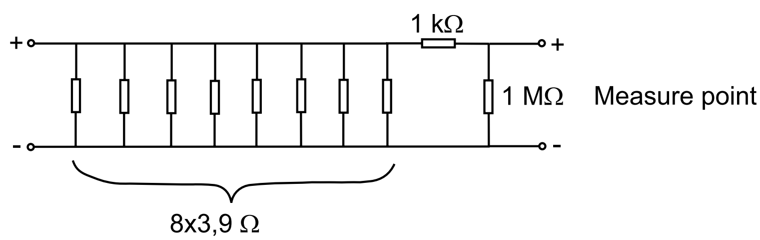

Figure 3.3: Analogue design

device is shown i Figure 3.3.

3.3 Harman/Kardon Amplifier

The Amplifiers consists a of regular HIFI amplifier, Harman/Kardon Citation Seventeen and a preamp, Harman/Kardon Citation Nineteen. Data sheets for these amplifiers are at the back of the manual. The frequency response for the amplifier is flat, which have been verified by several laboratory experiments.

3.4 Digital Signal Processor

The Digital Signal Processor (DSP) is one of the most important parts of this system. To be able to create the feedback loop and manipulate the signal in the desired way the choice of DSP is crucial. The requirements to fulfill is a DSP with fast AD/DA-converters and be able to perform a lot of calculations per second. One of the decisive reasons depended on that the signal to the midrange and tweeter also was going through the DSP. This was made in order to get the same delay on the signals. To be able to preserve a good sound quality of the frequencies in the upper range, it is necessary to sample the source with at the least 44,1 kHz.

The DSP that is used is a Texas Instrument TMS320C6701 EVM kit with Code Composer Studio 2.10 bundled with it. The DSP has a ADC and a DAC that are able to use a sampling frequency of 48000 Hz. A data sheet for the DSP is found at the back of the manual [JA03].

Chapter 4

Modelling

In order to make the control design as good as possible, it is important to model the system well. A good model can increase the knowledge of the system and make important robustness and stability analysis. In several control algorithms the model is included in the controller, like in IMC that is used in this project.

In the book [Lil93] and in Section 2.1.1, it is shown how a loudspeaker can be modelled as an electrical circuit that has a specific transfer function. The theoretical transfer function gives an indication of the order of the function that describes the system. This proves not to be as simple as it sounds like when applied in a practical case like this project.

From the system description in Figure 2.7 we see that models of the blocks F and H are required. The system from the input voltage of the amplifier to the sound pressure originating from the radiators will be referred to as $G(s)$. The block desired has the same input as $G(s)$ and the output signal is the current that flows through the speaker. This transfer function is called $F(s)$. We have the relation:

$$G(s) = F(s)H(s) \tag{4.0.1}$$

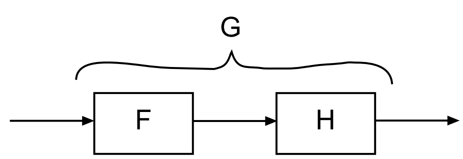

Figure 4.1: Flowchart of Equation 4.0.1

The transfer function $H(s)$ describes how the sound pressure is related to the current. $H(s)$ has not been modelled because of practical reasons and is therefore based on models G(s) and F(s) through:

$$H(s) = G(s)F^{-1}(s) \tag{4.0.2}$$

As seen in Figure 2.7, the transfer function $H(s)$ is not included in the closed loop and therefore not to be taken in consideration when creating the controller. This makes $H(s)$ just a compensation link as shown in Figure 2.7.

There are basically two ways of making a model of a system. The first one is to build a model using physical relations where the parameters represents real values of the system components. This can be quite easy if it is a small and non complex systems where all parameter values are known. If the system is large and have disturbances, errors or unknown components, such *tailor made* models can be really hard to construct. The other way of describing system is to use predefined model structures. The parameters in the different structures have no physical interpretations and their only purposes is to explain the properties of the system. These models are very flexible and are called *black box* models and are appropriate to use in the most common cases.

4.1 Identification toolbox and Validation

For the modelling part of this project, the Matlab identification toolbox Ident was used. Ident is a Graphical User Interface (GUI) made to be a convenient solution to calculate models for input and output data from a process. The toolbox contains all necessary tools for making a well adjusted model to a certain system. Functions like down sampling, mean removal, trend removal and filtering are all included in the GUI.

Validation is also made in Ident by using a piece of the data sequence that is not used when calculating the models. The output from the calculated model is then compared to the real output. The model then gets a indication of how well the model agrees to the real system. If the value is 100 the match is perfect.

4.2 Open Loop System

The theoretical transfer function $G(s)$ is put into practice with the arrangement in Figure 4.2. The reason to add up the acoustic sound pressure with 2 microphones is so the output signal will represent the *whole hearing impression.*

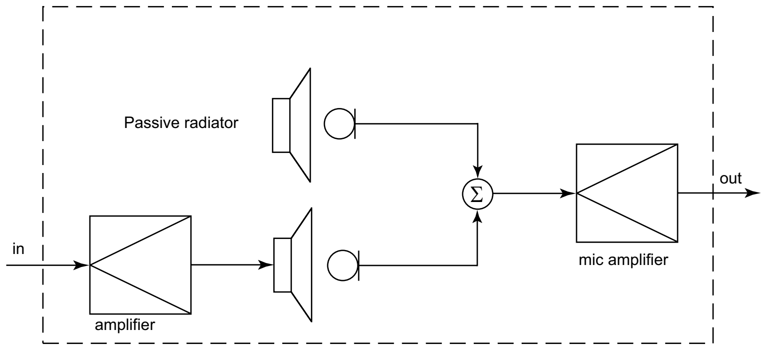

Figure 4.2: Schematics of the open loop system.

As input signal to the open loop system bandlimited white noise was used. The frequency range was bound from 0 to 250 Hz. This was made by filtering white noise with a $20th$ order butterworth filter. The power spectrum of the input signal is shown in Figure 4.3 and is clearly of white noise type.

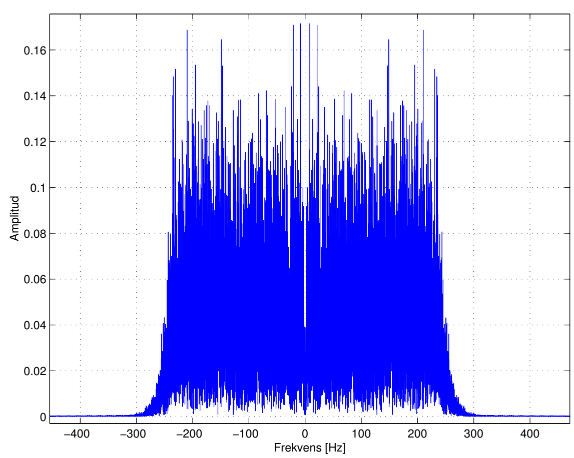

Figure 4.3: Power spectrum for the input signal that reaches the system.

When the signal has passed the open loop system, an output signal with highpass characteristics can be observed in Figure 4.4.

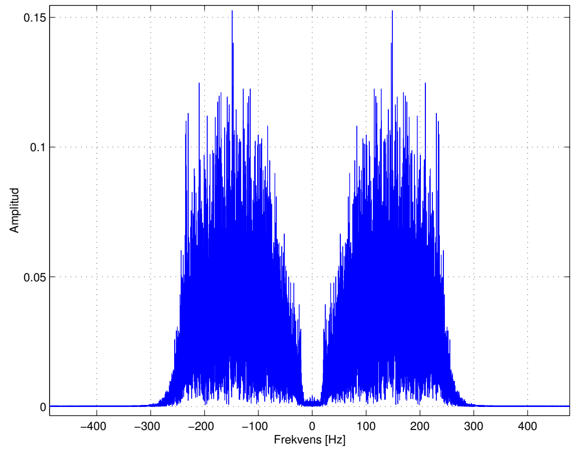

Figure 4.4: Power spectrum for the signal that is the system output.

These input and output data is then used to get a *transfer function estimate* (TFE) to have when validating the eventual models. TFE is a non-parametric estimate of the data sequence. The TFE is a calculated by the Matlab function with the same name. This is how the function works according to the Matlab help section:

> *TFE estimates the transfer function of the system with input X and output Y using Welch's averaged periodogram method. X and Y are divided into overlapping sections, each of which is detrended, then windowed by the WINDOW parameter, then zero-padded to length NFFT. The magnitude squared of the length NFFT DFTs of the sections of X are averaged to form Pxx, the Power Spectral Density of X. The products of the length NFFT DFTs of the sections of X and Y are averaged to form Pxy, the Cross Spectral Density of X and Y. Txy is the quotient of Pxy and Pxx; it has length NFFT/2+1 for NFFT even, (NFFT+1)/2 for NFFT odd, or NFFT if X or Y is complex. If you specify a scalar for WINDOW, a Hanning window of that length is used. Fs is the sampling frequency which does not effect the transfer function estimate but is used for scaling of plots.*

TFE from this data is shown i Figure 4.5 as well as a parametric model estimated from the received data. The parametric model is of output error type [TG91] and has an order in the nominator and denominator of 10. When making the models in Ident it became very clear that the it was the output error model that was suitable for this system. Other models that are common are ARX, ARMAX and Box-Jenkins, these are explained in [TG91].

Because of the limited frequency range that the model has to be valid for, the order can be significantly reduced. Model reduction was performed by deleting zeros an poles that was not in the frequency range of interest. Consideration has to be taken so that there is an equilibrium between poles and zeros in the frequency interval outside the one of interest. If there are a plethora of poles there will be heavy positive slope in the frequency response that will correspond to high gains where it is undesired. The plot in Figure 4.5 shows the TFE and the unreduced model.

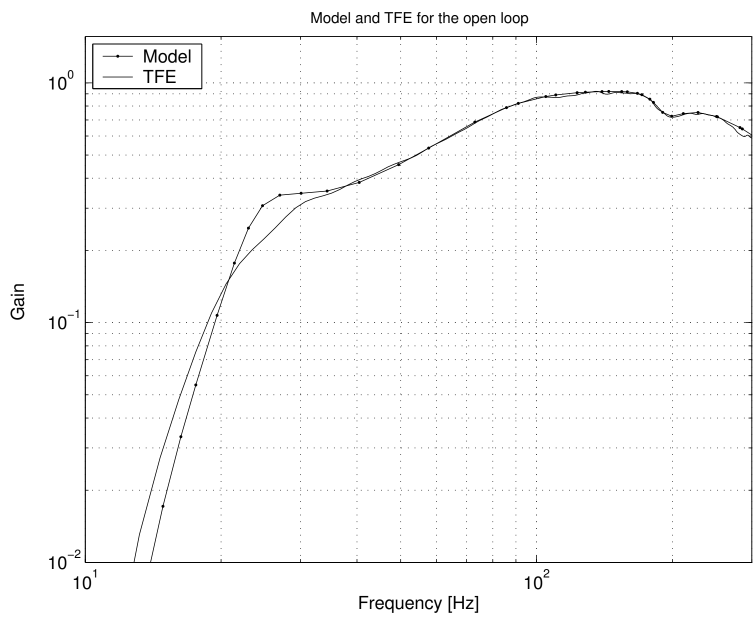

Figure 4.5: Transfer function estimate and Model for the open loop system.

4.3 Current measurement model

The system that previously been referred to as $F(s)$ is going to be modelled in this section. The input signal is the voltage that comes from a source, e.g a CD-player, and output signal is a voltage that is a measurement of the current that flows through the speaker. See Figure 4.6.

This model is the most important because the controller is going to be based upon it. The input signals to the current measurement was the same as for the open loop system. In fact the whole scenario for the modelling is

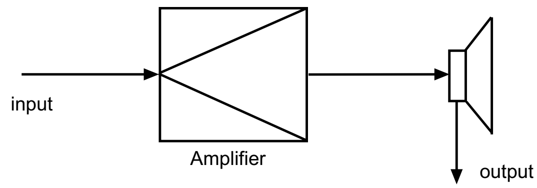

Figure 4.6: Schematics of the open loop system.

identical to the closed loop. Even the output error model showed to be the most accurate model type. The output plot shows clearly that the transfer function $F(s)$ is related to the radiator impedance. This is of course both obvious and expected, as can be seen on the data sheets for the radiator in [JA03].

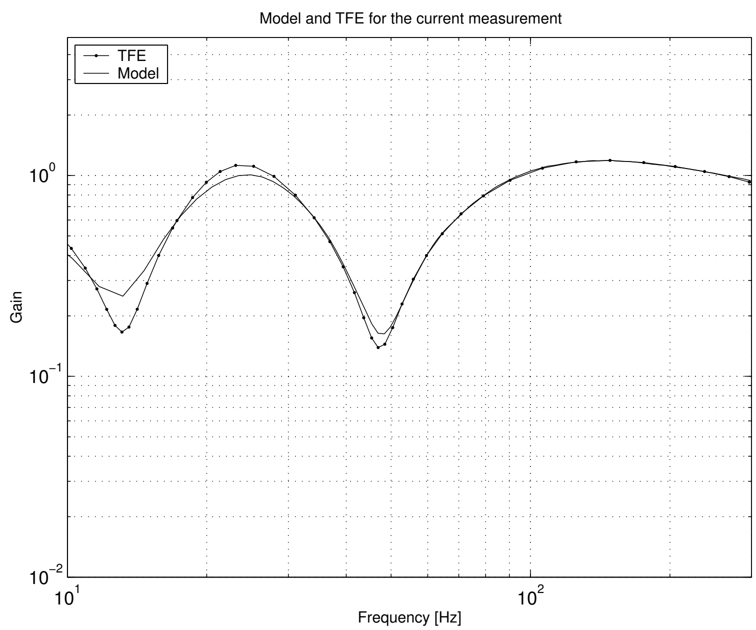

Figure 4.7: Schematics of the current measurement system.

The model in Figure 4.7 is an oe10101-model (nominator and denominator is of 10th order) and has a fit-value in Ident that is just above 96. This model is following the TFE quite well but the order is higher that it could be. A model of this high order is also difficult to work with and it takes a lot of calculations from the DSP if this was to be used.

Chapter 5

Results and Discussion

5.1 Control Design

The physical limitations of the system sets the conditions for the control design. As mentioned in section 2.1.3, there is no idea to try to play frequencies that are below the Helmholtz resonance frequency because of acoustic shortage that appears. This makes it ideal to have the closed loop frequency response shaped as a highpass filter with cut off frequency at 26 Hz. The transfer function for the desired high pass characteristics will be referred to as $L(s)$, which gives the closed loop:

$$L(s) = \frac{Q(s)G_0(s)}{1 + Q(s)(G_0(s) - G(s))} \tag{5.1.1}$$

This will give the IMC-controller Q(s):

$$Q(s) = \frac{L(s)}{G_0(s) - L(s)(G_0(s) - G(s))} \tag{5.1.2}$$

From Equation 5.1.2 and 2.2.8 a controller to the system configuration i Figure 2.9 can be derived. The highpass filter $L(s)$ was of first order, if the order was higher the system became unstable. The controller was calculated from the oe10101-model of F presented in Section 4.3.

A bode plot of the magnitude for the closed loop for the ideal controller is shown in Figure 5.1.

In the plot it is obvious that the closed loop shows the first order higpass filter with a cutoff frequency at 26 Hz that is desired. The magnitude has decreased 3 dB at 26 Hz.

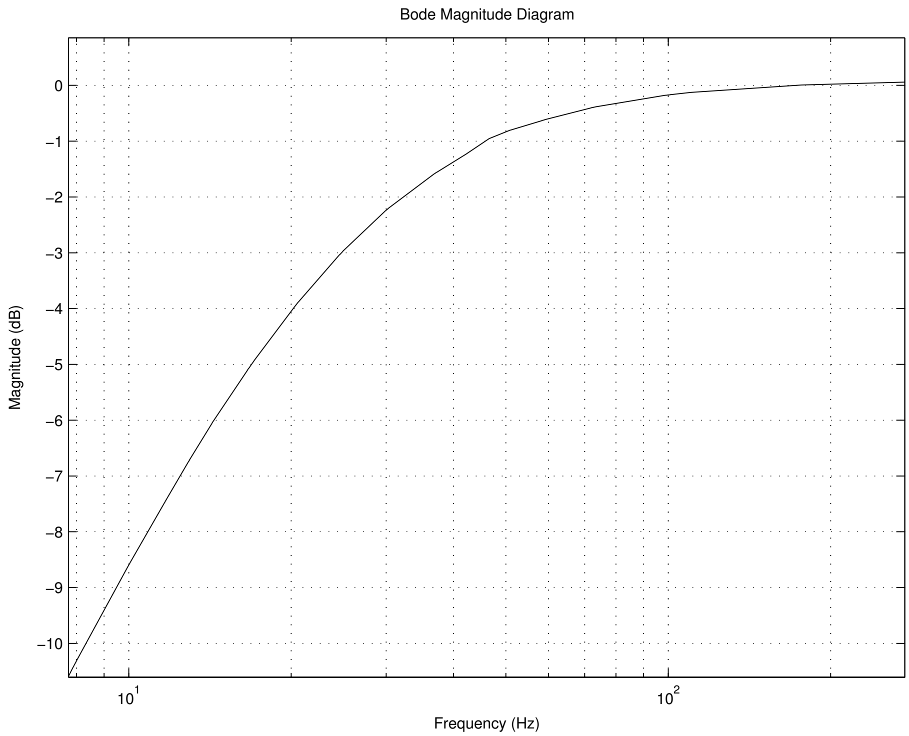

Figure 5.1: Bode magnitude plot of the closed loop with ideal controller.

5.2 Implementation

Implementation of the calculated controller was the bottleneck of this project. The way to create a program for implementation, a model was made in the Matlab toolbox Simulink. From that model auto generated C-code was made in Simulink. The reason for not making the C-code by hand was to save time. The Simulink model that was used can be seen in Figure 5.2. As the figure

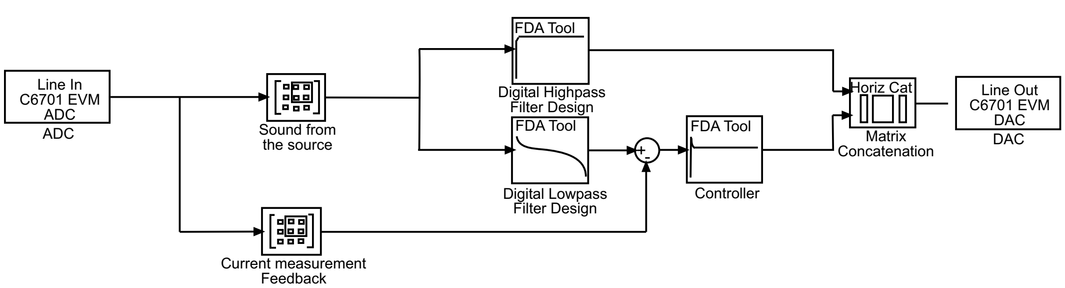

Figure 5.2: The Simulink model.

shows the structure of the model that the code was to be built on was quite

Model	Best fit (Max 100)
oe220	37,03
oe230	38,09
oe240	38,09
oe320	37,52
oe330	38,06
oe340	45,90
oe420	38,59
oe430	60,31
oe440	61,33

Table 5.1: The tabular shows how well the models are agreeing with the an unparametric model.

simple and contains only three filters, one addition and three matrix manipulations. In an early stage it was obvious that the filter taps have to be heavily reduced if this structure was to be implemented. The highpass filter was reduced to first order and the lowpass to second, both of IIR[1] type. The feed forward link $H^{-}1$ could not be implemented because of the limitations of the code generator from Simulink to the DSP. This makes it hard to, in reality, show what the result had been with a the calculated controller and feed forward link.

The controller was also to be heavily reduced because of the limitations of the DSP. The controller that was derived from Equation 5.1.2 was reduced using the method in Section 2.2.2. When experiments were made with the DSP it became clear that the highest order of the controller that could be implemented were second order. Even trials building the controller in cascades failed.

When calculating models in Ident, the second order models where not agreeing with the original model as much as desired. A list of the different model proposal can be seen in Tabular 5.1. As the tabular shows there is a big gap from oe420 and oe430 which made the choice fall on oe430 and then introduced another solution to get a controller of second order.

The controller oe430 was chosen to be the controller to continue to work with because of theoretical good appearance when simulating the closed loop.

[1]Infinite impulse response. More in [Wil97]

This controller is shown i Figure 5.3. As can be seen in the figure, there is a lot of controlling in frequencies above 250 Hz and also below the Helmholtz resonance. Possibilities of reducing the order of the controller were therefore pleasant.

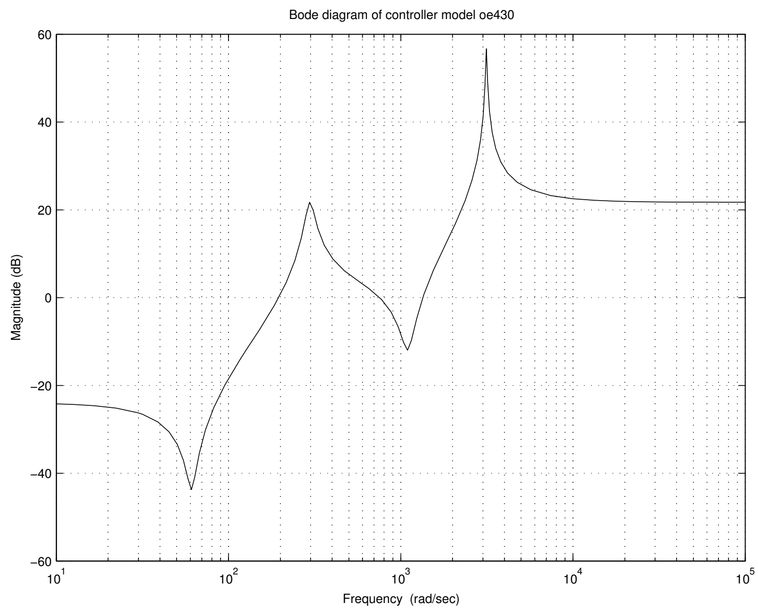

Figure 5.3: The reduced oe430 model of the controller.

5.2.1 Fine tuning the controller

The oe430 model of the controller does still have a higher order than accepted. As mentioned above, the second order is maximum. The controller in Figure 5.3 has poles and zeros that are not of interest for the purpose of the controller. The range to work within is up to 250 Hz. The fine tuning for this controller consisted in deleting poles and zeros by hand to get the closed loop as close as possible to the desired system properties. The deleting resulted in the second order controller in Figure 5.4.

5.3 Results

When fine tuning the controller, the step response and impulse response became a bit oscillating. But since the DSP set the limitations the oscillations had to be disregarded. The step and impulse response of the closed system

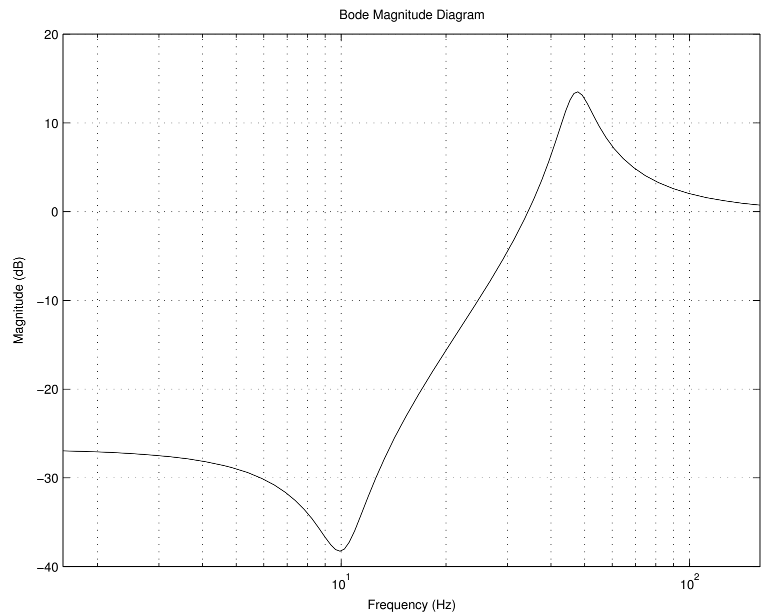

Figure 5.4: Bode magnitude diagram of the fine tuned controller.

can be seen in Figure 5.5. These plots show that the system is stable though a small oscillation is introduced.

The frequency response for the closed loop system is shown in Figure 5.6. It is not hard to see the difference between the theoretical closed loop in Figure 5.1 and this one. Although the response clearly has the characteristics of a highpass filter that it is supposed to have.

After successful theoretical simulations the real system had to be measured. The second order controller could be implemented in the DSP and the closed loop could be measured with the current measurement device. A transfer function estimation can be seen in Figure 5.7 and the result was pleasing. Since the feed forward link not was implemented the frequency response was quite similar to the theoretical one in Figure 5.1. The true system has an slight amplitude dip in the high range that the theoretical system do not have.

The plot of the measured closed loop shows that it is possible to control the current through a speaker radiator. If the whole theoretical system inclusive the feed forward link could have been implemented, the system output could have been measured and thereby evaluated against the original frequency response in Figure 4.5.

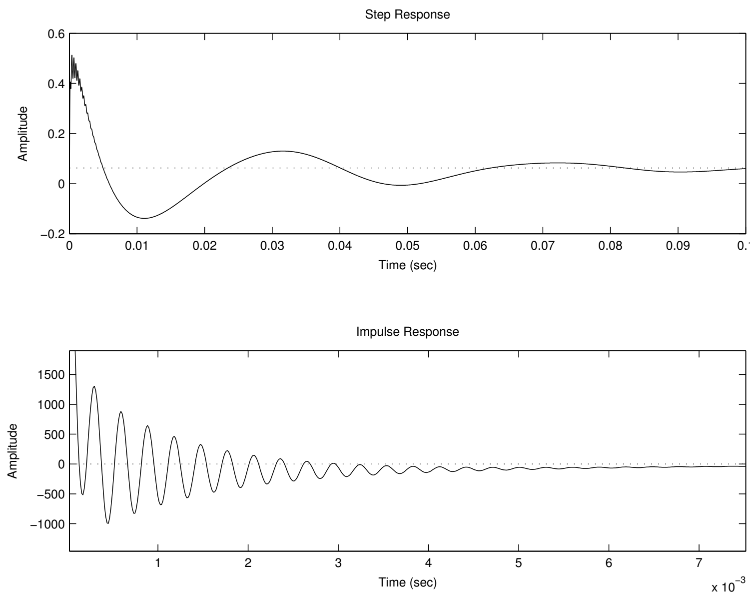

Figure 5.5: Step and impulse response for the closed system with the fine tuned controller.

One of the biggest worries with the control was that it may be possible to perform, but would not sound well. It was not that sure that the ear would like the controlled sound. In the long run, it is the pleasant sound that is desired to achieve. After controlling the loudspeaker it is most subjectively certain to state that the sound impression have improved although the feed back link is not yet implemented.

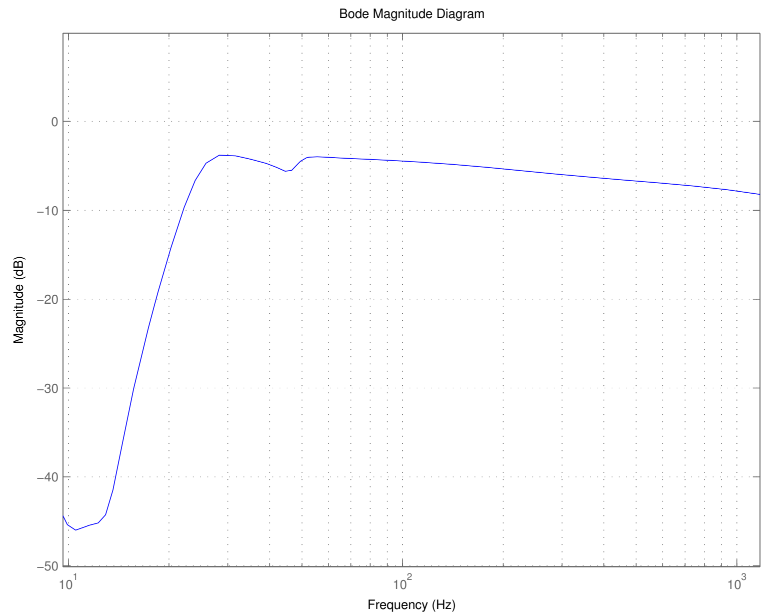

Figure 5.6: Closed loop for the system with fine tuned controller.

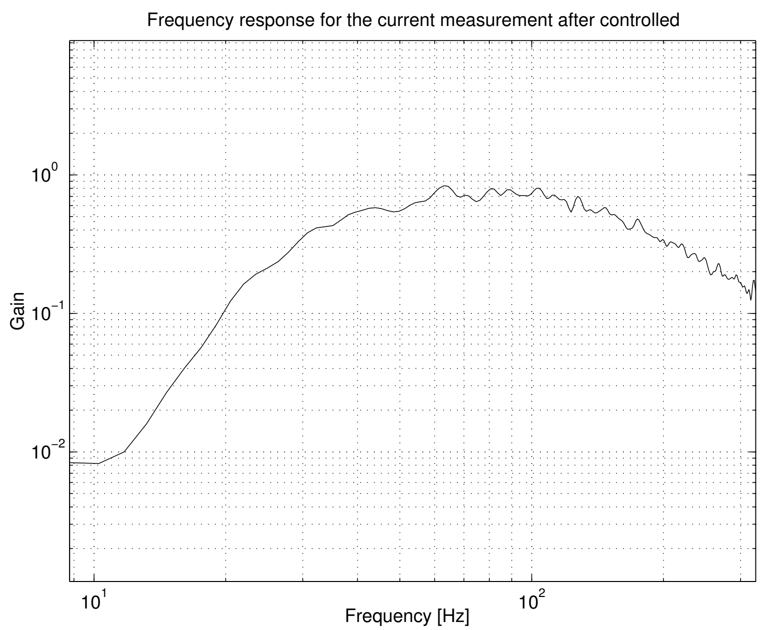

Figure 5.7: Plot of the real current controlled system

Chapter 6

Future work

6.1 Feed forward Link

The aim in this thesis was to completely control the speaker radiator and thereby change the frequency response for the loudspeaker. As shown above the control of the current flowing through the radiator was successful but because of the lack of knowledge in programming C it was not possible to implement the feed forward link H^{-1}, seen in Figure 2.7, in the DSP. If the filter was implemented it would have been possible to study how much the frequency response had changed after the control design was introduced.[TG89]

6.2 Optimized C-code

Because of the extent of this Master Thesis Project there was no time to learn how to program C. The knowledge how to use Simulink and thereby autogenerate C-code from a model was to be used. The created C-code that was not optimized for audio or control design applications.

With the controller implemented in the DSP that is used in this project there is a notable delay from input to output in the DSP. Perhaps a better C-code could minimize this delay and thereby make the product applicable to existing loudspeaker configurations. If this is possible, the product could be used as a complimentary subwoofer to consumers home cinema packages.

6.3 More advanced modelling and controller

If the control algorithm was able to be optimized for this cause, it would be possible to implement a controller of higher order than now is used. These possibilities would demand high accuracy in the modelling part and could also open up for other controllers than the IMC-controller used in this master thesis.

6.4 Stand alone product

A natural extension to the work preformed in this Master Thesis project is to build a stand alone device that is integrated with an active[1] loudspeaker. This would show that the actual control device could be made very small and probably cost effective, if manufactured in great quantities.

[1] An active loudspeaker is when an amplifier is integrated with the speaker cabinet. This is common for subwoofers.

Chapter 7

Conclusions and Acknowledgments

7.1 Conclusion

The combination between the two areas Audio technology and Automatic control is not very common. With the results that was achieved in this project in a relatively short time and just basic knowledge in Audio technology, more complex creations with even better results should not be impossible to explore. This hybrid between these areas shows that automatic control could be introduced into a lot of different areas and probably improve processes that never before were associated with Automatic control.

7.2 Acknowledgments

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This master thesis report was written in LaTeX.

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