NTNU
Innovation and Creativity

Compensation of Loudspeaker Nonlinearities

- DSP implementation

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Master of Science in Electronics

Submission date: August 2007

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Problem Description

A compensation system for nonlinear distortion in loudspeakers is simulated in Matlab and later implemented on DSP.

Assignment given: 05. February 2007
Supervisor: Peter Svensson, IET

Summary

Compensation of loudspeaker nonlinearities is investigated. A compensation system, based a loudspeaker model (a computer simulation of the real loudspeaker), is first simulated in matlab and later implemented on DSP for realtime testing. So far it is a pure feedforward system, meaning that no feedback measurement of the loudspeaker is used.

Loudspeaker parameters are drifting due to temperature and aging. This reduces the performance of the compensation. To fulfil the system, an online tracking of the loudspeaker linear parameters is needed (also known as parameter identification). Previous investigations (done by Andrew Bright and also Bo R. Pedersen) shows that the loudspeaker linear parameters can be found by calculations based on measurements of the loudspeakers current. This is a subject for further work.

Without the parameter identification, the compensation system is briefly tested, with the loudspeaker diaphragm excursion as output measure. The loudspeaker output and the output of the loudspeaker model are both monitored, and the loudspeaker model is manually adjusted to fit the real loudspeaker. This is done by realtime tuning on DSP. The system seems to work for some input frequencies and do not work for others.

Preface

This thesis is written to fulfil the requirements to obtain a master of science degree in electronics/acoustics, from NTNU - The Technical University of Norway.

During the last year I have been an exchange student at Ørsted DTU - The Technical University of Denmark - in the section for acoustic technology. The master thesis is carried out in corporation with Bang & Olufsen A/S, located in Struer – Denmark. In all eight weeks are spent at Bang & Olufsen, while the rest is spend at Ørsted DTU.

I would like to thank Finn Agerkvist - my supervisor at DTU, Peter Svensson - my supervisor at NTNU, Sylvain Choisel – my supervisor at B&O, and also Søren Beck, Gert Munch and the rest of the acoustical apartment in Struer, all for great helping and great inspiration.

Karsten Øyen, 1th. July 2007, Copenhagen

Contents

Expressions

A/D	Analog to Digital - converter
D/A	Digital to Analog - converter
IN1,IN2	input's on DPS-board.
Loudspeaker Model	Predicts vector X (Vector X consists of the loudspeaker diaphragm position(*x*), diaphragm velocity *(v)*, eddy-current(i_{L2}), voice-coil-current (i_{Le}).
Music source	Standard music signal from CD player or other source
Nonlinear Compensator	Removes the nonlinearities in the input signal: w.
OUT1-OUT4	output's on DSP-board
Parameter Identifier	Calculates the linear parameters (vector K) based on current measurement.
P. Amp	Power amplifier, standard voltage controlled HIFI-amplifier.

Notation

Symbol	Deskription	Unit
a	Loudspeaker diaphragm acceleration.	[m/s^2]
B:	Magnetic field.	
Bl	Force factor, the product of *B(magnet field)* and *l(effective length of voice-coil)*.	[N/A]
Bl_0	Force factor, at x=0.	[N/A]
C_str	Factor for tuning the compliance curve, stretches the x-axis.	
C_sca	Factor for tuning the compliance curve, scales the curve.	
C_t:	Mechanical compliance of driver suspension.	[m/N]
C_0:	Mechanical compliance of driver suspension, at x=0.	[m/N]
f_res:	Loudspeaker resonance frequency.	[Hz]
$F(n)$:	Matrix used in state space model.	
$G(n)$:	Matrix used in state space model.	
i_{Le}:	Voice-coil current.	[A]
i_{Lem}	Voice-coil current – measured value.	[A]
i_{L2}	Voice-coil eddy-current, induced in the loudspeaker's voice-coil.	[A]
G	Power Amplifier Gain.	
k:	Mechanical stiffness of driver suspension(1/C).	N/m]
k_0:	Mechanical stiffness of driver suspension, at x=0 (1/C_0).	N/m]
$K(n)$	Vector consisting of linear parameter.	
l:	Effective length of the voice-coil.	[m]
L_2:	Para-inductance of voice coil, due to eddy current losses.	[H]
L_e:	Voice coil inductance.	[H]
L_{e_0}:	Voice coil inductance, at x=0.	[H]
M_t:	Mechanical mass of driver diaphragm, air load and voice-coil.	[kg]
P_D	Pressure difference between the rear and the front of the loudspeaker diaphragm.	[Pa]
P_{AR}	Power Radiation	[Pa]
R_t	Mechanical resistance of total-driver losses	[kg/s]
R_2:	Electrical resistance, due to eddy current losses.	[Ω]
R_e:	Electrical voice coil resistance at DC.	[Ω]
Rs:	Shunt resistor for measuring voice-coil current: i_{Le}	[Ω]
Ra-Rd:	Feedback network for loudspeaker current -and voltage measurement.	[Ω]
$R_{in\ DSP}$	Input resistance for analogous inputs on DSP board.	[Ω]
S_D:	The area of the diaphragm.	[m^2]
Ts:	1/samplingfrequency	[s]
u (n):	Processed input signal for loudspeaker, digital, discrete.	[V]
$u_D(t)$	Driver voltage. Amplified, music signal applied to the loudspeaker.	[V]
$u_D(n)_m$	Driver voltage – measured value.	[V]
u (t):	Processed music signal, analogous signal.	[V]
U	Loudspeaker diaphragm volume velocity.	[m^3/s]
v:	Loudspeaker diaphragm velocity	[m/s]
w(n):	Signal from music source, digital, discrete.	[V]
w(t):	Signal from music source, analog signal.	[V]

x (n):	Loudspeaker diaphragm position	[m]
X(n):	Predicted state vector (one sample into the future). Consists of current, i_{Le}, eddy current, i_{L2}, voice-coil position x (n) - and velocity, $v(n)$.	
Z_{res}	Loudspeaker impedance at the resonance frequency	[Ω]
Z_{AR}	Acoustical impedance at the back side of the loudspeaker diaphragm	[Pa]
Z_{AF}	Acoustical impedance in the front of the loudspeaker diaphragm	[Pa]
$\frac{\partial L_e}{\partial x}$:	First derivate of L_e with respect to position ($x(n)$).	[H/m]
$\frac{\partial L_2}{\partial x}$:	First derivate of L_2 with respect to position - $x(n)$.	[H/m]

1. Introduction

Loudspeakers convert electrical signals into audible sound pressures. The conversion is however not perfect. Although this has been a subject for investigation for about one century - no one has yet made a perfect loudspeaker.

Loudspeakers generate distortion, meaning that the music signal is affected by the loudspeaker itself. We distinguish between linear distortion[1] and nonlinear distortion[2]. Linear compensation systems are found in nearly all loudspeakers[3]. Nonlinear compensation systems are very seldom.

Research on an electrical nonlinear compensation system has been done since the twenties. Nonlinear compensation (negative feedback systems[4]) is well used in amplifiers, but has not been a commercial successful in loudspeaker. Problems are related to the measurement of the feedback signal (the loudspeaker output, motion measurement[5]). Methods are so far not found to be appropriate. Though, new technology may have changed this.

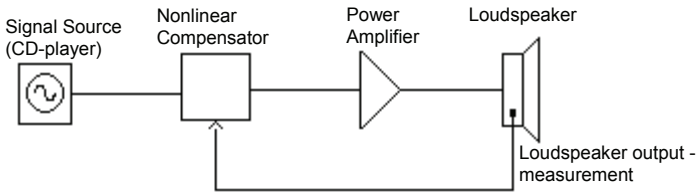


Figure 1.1. Feedback compensation system. The compensation is based on loudspeaker output measurement.

In the early nineties, when DSP-technology made new feature possible, the first attempts to implement nonlinear compensation without feedback, were made. This is known as feedforward processing. The compensation is based on a loudspeaker model – a computer simulation of the loudspeaker. The compensator has to be special customized to the loudspeaker, since detailed information about the loudspeaker is required to simulate its behaviour. Though, the performance of these compensators was low, due to inaccuracy in the loudspeaker model,

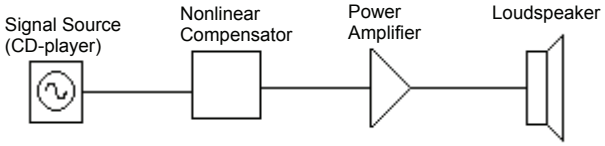


Figure 1.2. Feedforward compensation system. The compensation is based on computer-simulation of the loudspeaker.

[1] Linear distortion is when some of the frequency-bands are reproduced too loud, or not loud enough. If a loudspeaker has a “flat” frequency response, the linear distortion is low.
[2] Nonlinear distortion is when new frequency components are added by the loudspeaker. These are sums and multiplications of the frequency contents of the applied music signal.

[3] The traditional analogous filter in loudspeakers compensates for linear distortion – by flatting out the frequency response.
[4] Negative feedback - Parts of the output signal is feed to the input in reverse. Linear and nonlinear distortion is reduced traded for a lack of gain.
[5] Motion feedback means a measurement of the motion of the loudspeakers membrane. Position, velocity or acceleration is measured.

Several investigations are done to improve the performance of loudspeaker modelling. The problem is that loudspeakers change due to temperature and aging, while a computer-model don't. In 1998 one paper suggests to add a feedback measurement, to make the simulation able to follow the changes of the loudspeaker. This is known as the Adaptive feedforward controller. Later one also realised that a current measurement can be used as feedback signal, avoiding the traditionally problems due to motion measurement of the diaphragm. So far is this technology not available in any commercial products.

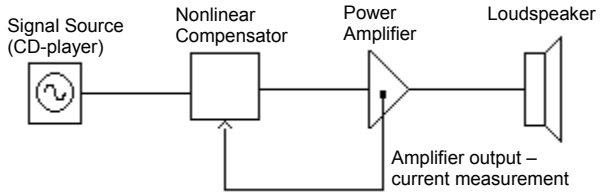


Figure 1.3. Adaptive Feedforward system. Compensation based on computer-simulation of the loudspeaker. The performance of the loudspeaker simulation is improved by use of a current measurement.

1.1 The Concept of the Adaptive Controller

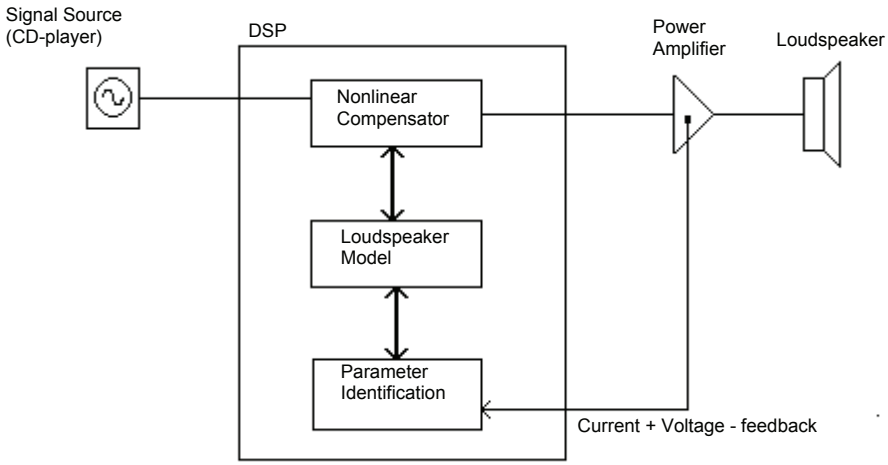


Figure 1.4. The concept: System for nonlinear compensation for loudspeakers.

Loudspeaker model
The loudspeaker model simulates the loudspeaker. The simulated output is send to the nonlinear compensator.

Nonlinear compensator
The nonlinear compensator removes nonlinear distortion, based on the simulation of the loudspeaker model. The loudspeaker model has predicted the loudspeaker nonlinearities, the inverse of the loudspeaker nonlinearities are added to the input, so the unwanted nonlinear distortion is cancelled out.

Parameter identification
This block detects changes of the loudspeaker, due to temperature drift and aging, and continuously updates the loudspeaker model.

1.2 Specifications for the project

During the first weeks at Bang & Olufsen in Struer, some goals for the project is settled. This page consists of some specified expressions, which are explained later in the report.

The Final Goal for the Project is:

- -to reduce intermodulation distortion (IMD) and total harmonic distortion (THD) for Loudspeakers, by implementing a DSP-compensation in front of the power amplification.

Specifications for the project

- -Nonlinearities in the force factor-Bl, the suspension-C and the inductance-Le, will be compensated for. The parameters Bl, C and Le will be treated as position dependent variables.
- -The compensated signal must be restricted within the range of the woofers specifications. Limiting or compressing will not be given priority to.
- -Actually compensating algorithms will first be simulated at in matlab, and later implemented on DSP for real time testing.
- -A simple model of the time varying suspension will be added.

Time Schedule at master project.

Date/Day	Week	Happening	Location
05.02(M)-09.02(F)	6	-Literature studding.	DTU
12.02(M)-09.03(F)	7-10	- Measure the nonlinear parameters at B&O woofers. - Implement the loudspeaker matlab simulation done in the course: "Advanced model modelling", on DSP.	B&O
12.03(M)-30.03(F)	11-13	-Simulate nonlinear compensation in matlab.	DTU
02.04(M)-06.04(F)	14	-Easter vacation.	
09.04(M)-03.05(Th)	15- 18	-Implement compensation code on DSP-board	DTU
08.05(Tu)-18.05(F)	19, 20	-Find a simple model of the time varying suspension.	B&0
21.05(M)-01.06(F)	21,22	-Test the compensation code on real speakers.	B&O
04.06(M)-30.06(F)	23-26	-Finish the project report.	DTU
30.06(F)	26	-Ending date.	

2. The Loudspeaker – a short introduction.

2.1 The History

The story of the electrodynamic loudspeaker begins in the late 20th century. Alexander Graham Bell patents the telephone in 1876, while Ernst Werner Siemens first describes the system in 1874. After further development it is finally patented by Rice and Kellogg in 1924. For the last century the absolute majority of loudspeakers are based on this principle.

2.2 The principle

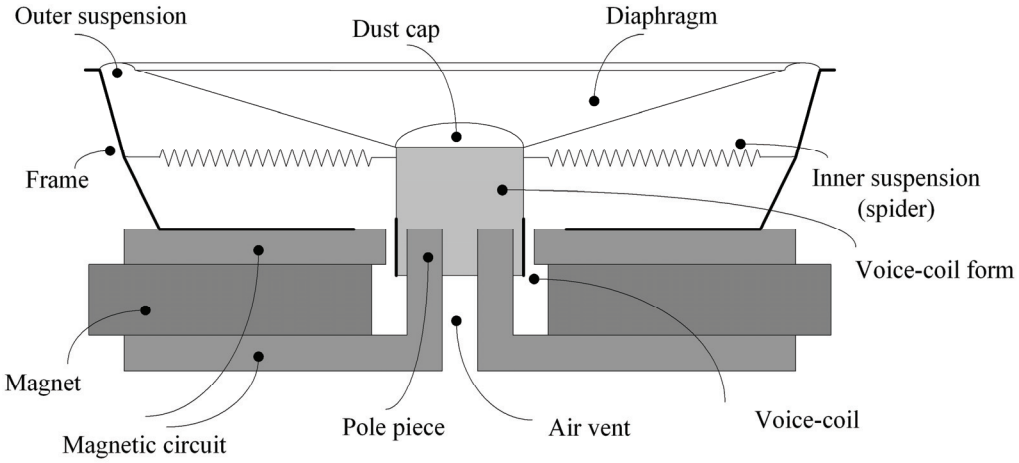


Figure 2.1 shows a cross section of the loudspeaker. (The drawing is taken from [Andersen, 2005])

The loudspeaker converts electrodynamic signals into audible sound pressures, created by the movement of the diaphragm. The diaphragm is attached to the voice-coil form, and these components moves vertically (figure 2.1). The lower part of the voice-coil form are surrounded by the voice-coil, and placed in the magnetic field. As electrodynamic signals are applied to the voice-coil, an electromagnetic force will appear between the voice-coil and the magnet, moving the diaphragm vertically.

2.3 The loudspeaker behaviour due to the resonance frequency.

Mechanically, the loudspeaker can be seen as a simple mass spring system. The loudspeaker resonance frequency is given by:

$$f_{res} = \sqrt{\frac{1}{C_{t\cdot} \cdot M_t}}$$

where *Ct* represent the softness/compliance of the outer suspension and the spider, and *Mt* are the mass of the moving parts, voice-coil, voice-coil-form, and the diaphragm (included the mass of the moving air)

The resonance frequency for typical six inch loudspeaker is normally between 50 and 150 Hz.

At resonance:

- The influence from the suspension and the mass equal.
- The current and the voltage have equal phase.

Below resonance:

- The suspension is the most important parameter.
- The phase of the current is delayed due to the voltage.

Above resonance:

- The mass is the most important parameter.
- The phase of the current in ahead of the voltage

2.4 Acoustical Power Radiation

The acoustical power radiation from a simple source is given by:

$$P_{AR} = \frac{1}{2} \cdot \frac{p_0 \cdot S^2}{2 \cdot \Pi \cdot c} \cdot (w^2 \cdot |x|)^2$$, from (Leach, 1999).

2.5 Generally About Loudspeaker Modelling

The purpose of the loudspeaker model is to predict the behaviour of the loudspeaker. The quality of the nonlinear compensation system in figure 1.4, totally depends on the accuracy of the loudspeaker model.

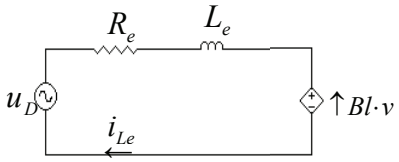
The final goal is to simulate the nonlinear loudspeaker. This is done in chapter 5. But first is the linear model is presented in chapter 3.

3. Linear Modelling

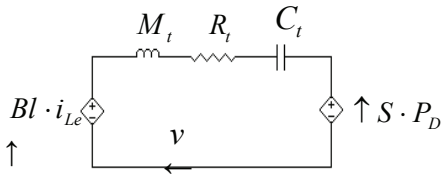
In this chapter the linear modelling is described. Electrical - mechanical and acoustical analogous circuits and equations first presented, and is later converted into the digital domain.

3.1 Electrical, Mechanical and Acoustical Analogous Circuits.

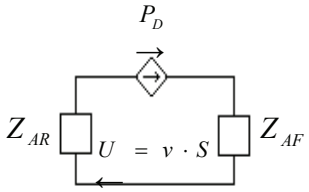
Electrical Circuit:



Mechanical Circuit:



Acoustical Circuit:



Electrical equation: (voltage)

$$u_D = R_e \cdot i_{Le} + L_e \cdot \frac{\partial i_{Le}}{\partial t} + i_{Le} \cdot \frac{\partial L_e}{\partial x} \cdot v + Bl \cdot v$$

Mechanical equation: (force)

$$Bl \cdot i_{Le} = M_t \cdot a + R_t \cdot v + \frac{1}{C_t} \cdot x + S \cdot P_D$$

Acoustical equation: (volume velocity)

$$P_D = U \cdot Z_{AB} + U \cdot Z_{AF}$$

Figure 3.1. The electrical, mechanical and acoustical - analogous circuits, with equations, describing the linear loudspeaker.

a	Loudspeaker diaphragm acceleration.	[m/ s^2]
Bl	Force factor, the product of *B(magnet field)* and *l (effective length of voice-coil)*. [N/A]	
C_t :	Mechanical compliance of driver suspension	[m/N]
i_{Le} :	Voice-coil current.	[A]
L_e :	Voice coil inductance.	[H]
M_t :	Mechanical mass of driver diaphragm, air load and voice-coil.	[kg]
P_D	Pressure difference between the rear and the front of the loudspeaker diaphragm.	[Pa]
R_e :	Electrical voice coil resistance at DC.	[Ω]
R_t	Mechanical resistance of total-driver losses	[kg/s]
S :	The area of the diaphragm.	[m^2]
$u_D(t)$	Driver voltage. Input voltage for the loudspeaker.	[V]
U	Loudspeaker diaphragm volume velocity	[m^2 /s]
v :	Diaphragm velocity	[m/s]
Z_{AR}	Acoustical impedance at the back side of the loudspeaker diaphragm	[Pa]
Z_{AF}	Acoustical impedance in the front of the loudspeaker diaphragm	[Pa]

3.1.1 Electrical System

Electrically the loudspeaker is a voice-coil moving in a magnet field. Simplified, the voice-coil is seen as a resistor (*Re*) and inductor (*Le*) in series. The system is driven by the electrical voltage: $u_D(t)$. The electrical equation in figure 3.1 describes the voltage drops.

$u_D(t)$ - the input voltage. (energy source)

$R_e \cdot i_{Le}$ - the voltage drop of *Re*.

$L_e \cdot \frac{\partial i_{Le}}{\partial t}$ - the voltage drop of *Le*.

$i_{Le} \cdot \frac{\partial L_e}{\partial x} \cdot v$ -If Le changes with the voice-coil position, the loudspeaker will also work as a small electric generator. In the linear model the inductance is considered to be a linear parameter, setting this term to zero, since the derivative of the Le with respect to position, is zero.

$Bl \cdot v$ - the voltage drop due to the mechanical system – the connection between the electrical and the mechanical system.

3.1.2 Mechanical System

Simplified the loudspeaker is – mechanically - seen as a simple mass spring system. The outer suspension and the spider are the spring (with compliance, *Ct*[6]). The weight of the moving parts, voice-coil, voice-coil-form, and the diaphragm (included the surrounding moving air) are the mass (*Mt*). Losses due to friction, is represented by *Rt*. The system is driven by the mechanical force: $Bl \cdot i_{Le}$

The mechanical equation in figure 3.1 describes how the force is distributed.

$Bl \cdot i_{Le}$ - the mechanical force acting on the voice-coil. (energy source)

$M_t \cdot a$ - the force acting on mass.

$R_t \cdot v$ - the force acting on mechanical resistance.

$\frac{1}{C_t} \cdot x$ - the force acting on spring.

$S \cdot P_d$ - the force acting on the acoustical system, the connection between the mechanical and the acoustical system)

3.1.3 Acoustical System

Acoustically the loudspeaker sees two impedances, the acoustical impedance of the air in front of the diaphragm (Zaf), and the acoustical impedance of the air in the back of the diaphragm (Zar). The system is driven by the pressure difference: P_D. The acoustical equation in figure 3.1 describes how the pressure is distributed.

P_D - the pressure acting on the air, (energy source)
(Pressure difference between the rear and the front of the loudspeaker diaphragm)

$U \cdot Z_{AF}$ - the pressure in front of the diaphragm.

$U \cdot Z_{AR}$ - the pressure at the rear of the diaphragm.

[6] Compliance – describes the softness of the spring. Compliance is the inverse of stiffness. C=1/K.

3.2 The State Space Model - linear

The state space model predicts the behaviour of the loudspeaker. The expected diaphragm position, diaphragm velocity and voice-coil current are calculated based on the electrical and mechanical circuits in figure 3.1. Vector X consists of these three parameters. The driver voltage is the only input signal for the model.

3.2.1 Forward Euler

For the discrete time implementation forward Euler is used. Bilinear transform and impulse invariance are alternative methods. These are not described here.

In the forward Euler method, is the next state value of a system, *X(n+1)*, predicted by looking at the present state value, and its derivatives.

$$X(n+1) = X(n) + T_S \cdot \frac{\partial x(n)}{\partial t} + \frac{{T_S}^2}{2!} \cdot \frac{\partial^2 x(n)}{\partial^2 t} + \frac{{T_S}^3}{3!} \cdot \frac{\partial^3 x(n)}{\partial^3 t} \ldots$$

Ts is the time difference between each sample. By just taking the first derivative in account, this simple equations are given;

$$X(n+1) \approx X(n) + T_S \cdot \frac{\partial x(n)}{\partial t} \qquad \& \qquad \frac{\partial x(n)}{\partial t} \approx \frac{X(n+1) - X(n)}{T_S}$$

This digital modelling requires a high sampling frequency, at last five times higher than the frequency of the signal, *X(n)*

3.2.2 Conversion of the analogous equations to digital, discrete domain.

The electrical analogous equation in figure 3.1 is transformed into discrete time.

The analogous equation:

$$u_D = R_e \cdot i_{Le} + L_e \cdot \frac{\partial i_{Le}}{\partial t} + i_{Le} \cdot \frac{\partial L_e}{\partial x} \cdot v + Bl \cdot v$$

The variables are set as functions of digital, discrete time, *n*

$$u_D(n) = R_e \cdot i_{Le}(n) + L_e \cdot \frac{\partial i_{Le}(n)}{\partial t} + i_{Le(n)} \cdot \frac{\partial L_e}{\partial x(n)} \cdot v(n) + Bl \cdot v(n)$$

3.2.3 Predicting the next state value of the Voice-coil Current, $i_{Le}(n+1)$

The derivative of the current is replaced with forward Euler:

$$u_D(n) = R_e \cdot i_{Le}(n) + L_e \cdot \frac{i_{Le}(n+1) - i_{Le}(n)}{T_S} + i_{Le} \cdot \frac{\partial L_e}{\partial x(n)} \cdot v(n) + Bl \cdot v(n)$$

The next state of the current is:

$$i_{Le}(n+1) = (1 - \frac{R_e}{L_e} \cdot T_S) \cdot i_{Le}(n) - \frac{T_S}{L_e} (\frac{i_{Le}(n) \cdot \partial L_e}{\partial x(n)} + Bl) \cdot v(n) + \frac{T_S}{L_e} \cdot u_D(n)$$

3.2.4 Predicting next state value of the diaphragm Velocity, $v(n+1)$

The calculation of the diaphragm velocity, *v(x).* is similar to the calculations of the current in 3.2.3. The result is:

$$v(n+1) = i_{Le}(n) \cdot \frac{Bl \cdot T_S}{M_t} - \frac{x(n) \cdot T_S}{M_t \cdot C_t} + v(n)(1 - \frac{R_t \cdot T_S}{M_t})$$

3.2.5 Predicting the diaphragm Position, $x(n+1)$

The diaphragm velocity is solved with simple forward Euler from 3.2.1.

$$x(n+1) = x(n) + T_S \cdot \frac{\partial x(n)}{\partial t}$$

Where the diaphragm velocity, *v*, is the first derivate of the diaphragm position, *x.*

$$x(n+1) = x(n) + T_S \cdot v(n)$$

3.2.6 The Final Matrix of The State Space Model - linear

The calculations in 3.2.3 – 3.2.4 are placed the matrix-system shown in figure 3.2. *X(n)* is the state vector, consisting of the voice-coil current(here notated as *i*), the diaphragm position and the velocity.

$$F(n) = \begin{bmatrix} 1 - \frac{R_e}{L_e} T_s & 0 & -\frac{T_s}{L_e(n)} \left(i(n) \frac{\partial L_e}{\partial x} + Bl \right) \\ 0 & 1 & T_s \\ \frac{Bl}{M_t} T_s & \frac{-T_s}{C_t M_t} & 1 - \frac{R_t T_s}{M_t} \end{bmatrix} \quad X(n) = \begin{bmatrix} i(n) \\ x(n) \\ v(n) \end{bmatrix} \quad G(n) = \begin{bmatrix} \frac{T_s}{L_e} \\ 0 \\ 0 \end{bmatrix} \quad u(n) = A \sin(2\pi f n T_s)$$

$$X(n+1) = F(n) X(n) + G(n) u(n)$$

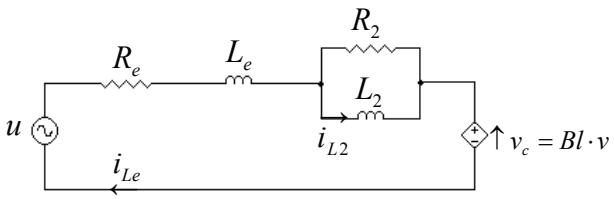
Figure 3.2. The equations of the state space model, simplified verson.

X(n+1) represents the simulated future output of the loudspeaker. The simple linear state model is now ready to be implemented in matlab.

3.3 Extension of the linear model. – including eddy currents.

As mention in 3.1.1, if the inductance, Le changes with the voice-coil position, the loudspeaker works as a small electric generator, and a current flow is generated in the voice coil. The current attempts to flow in small circles. Eddy currents was first described in [Thiele, 1961]. Due to eddy current losses the simple electric analogous circuit in figure 3.1, has been modified twice. Leach, [Leach,1999], added a resistor in parallel with Le, and in [Klippel, 2003] another modification is introduced. The voice-coil are modelled by Re, Le, R2 and L2, shown in figure 3.3.

Modified Electrical Circuit:



Equation for electrical circuit (voltage):

$$u = R_e \cdot i_{Le} + L_e \cdot \frac{\partial i}{\partial t} + L_2 \cdot \frac{\partial i_{L2}}{\partial t} + i_{Le} \cdot \frac{\partial L_e}{\partial x} \cdot v + i_{L_2} \cdot \frac{\partial L_2}{\partial x} \cdot v + Bl \cdot \frac{\partial x}{\partial t}$$

Figure 3.3. The extension of the simple analogous electrical circuit. This model was proposed by W. Klippel in 200.[Klippel, 2003], as a improvement of the simple model, just using Re and Le, shown in figure 3.1.

In figure 3.4 the loudspeaker output excursion is simulated, and the radiation is calculated based on the excursion (see 2.4). The green curve is simulated without eddy currents, and the blue with eddy currents. (The state space models described in 3.2.6 and 5.2.1 are used). The influence of the eddy currents are increasing proportionally with frequency.

Due to the masking phenomena in human hearing, is this small difference not audible.

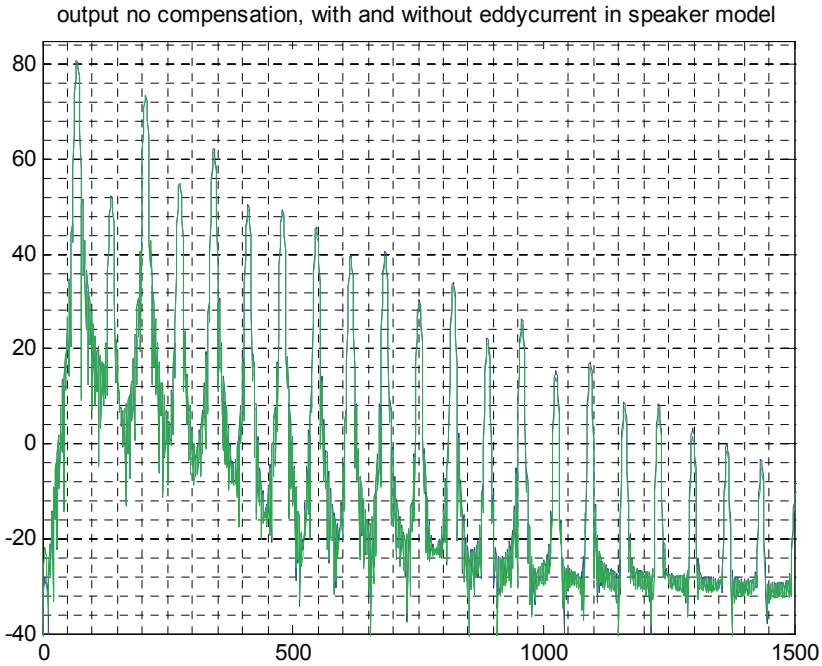


Figure 3.4. The eddy current influence. In green is a simulation without including eddy currents, and in blue the loudspeaker simulated with eddy currents.

4. Loudspeaker Nonlinearities

In chapter 3 linear modelling is described. Let look back on linearity. Loudspeakers, operating at small displacements, have a decent linear amplitude response. The shape of input voltage and the voice-coil excursion are in proportion. Linear amplitude response is illustrated in figure 4.1.

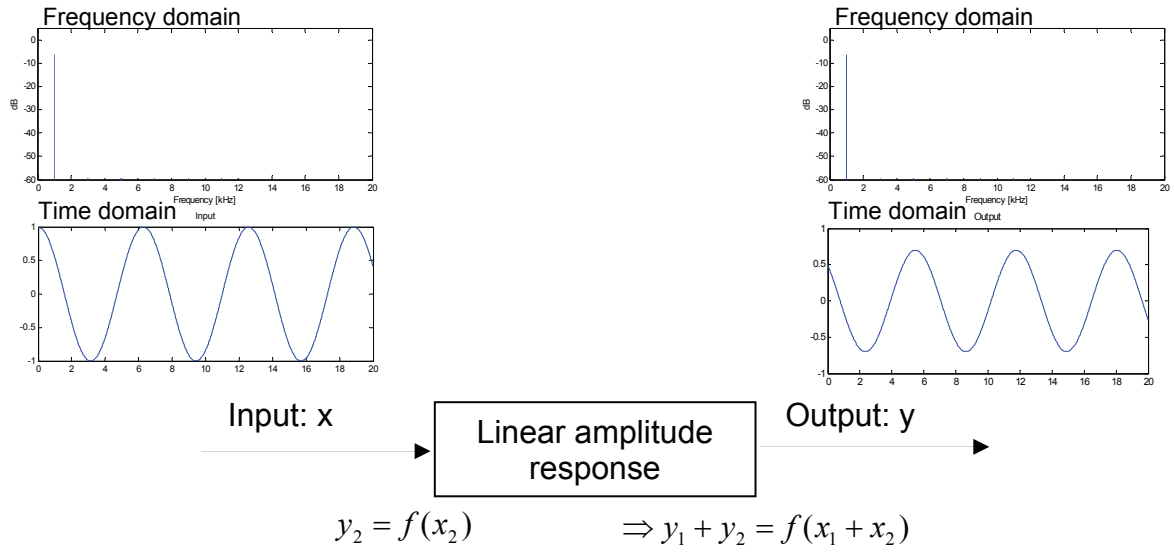


Figure 4.1. Linear amplitude response / superposition - the case for a loudspeaker operating within small displacement.

Operating at large displacements the loudspeaker amplitude response becomes nonlinear. The shape of input voltage and the output displacement are no longer in proportion. Nonlinear amplitude response is illustrated in figure 4.2.

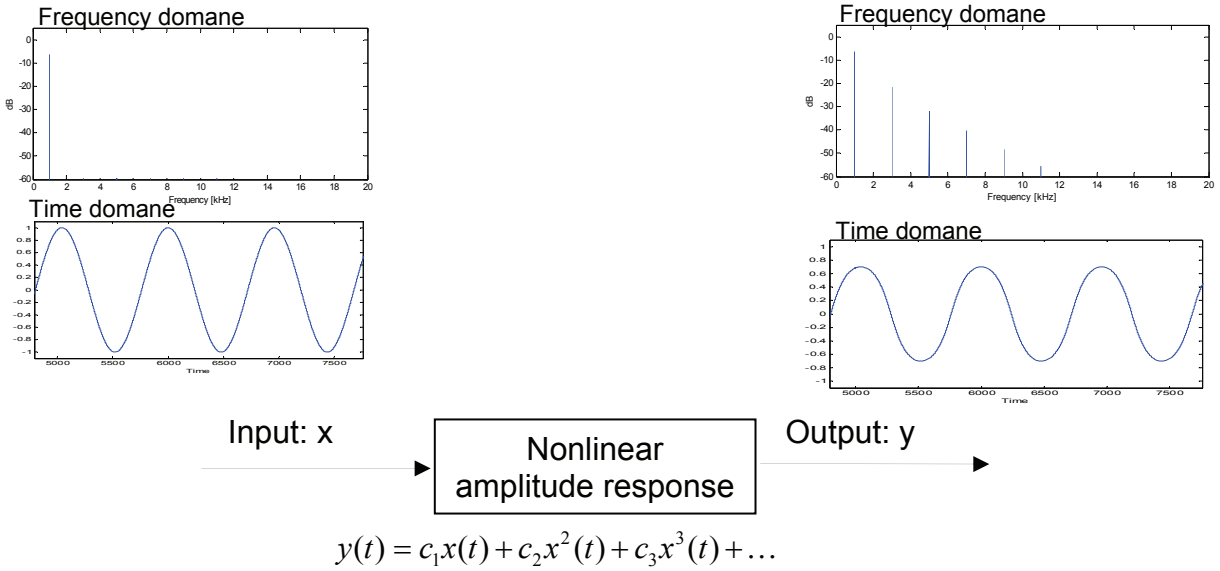


Figure 4.2. Nonlinear amplitude response - the case for a loudspeaker operating at large excursions.

The diaphragm excursion has a physical limit. Before the diaphragm excursion reaches its maximum, the loudspeaker will gradually start to limit the diaphragm excursion. This leads to a compression of the signal, as seen in figure 4.2. Both the force factor, Bl and the suspension are limiting the diaphragm excursion.

Nonlinear phenomena are mostly caused by the force-factor (Bl), the compliance of the suspension (C) and the inductance (Le). These parameters are dependent of the diaphragm position.

4.1 Position Dependent Parameters

4.1.1 The Position Dependent Force Factor

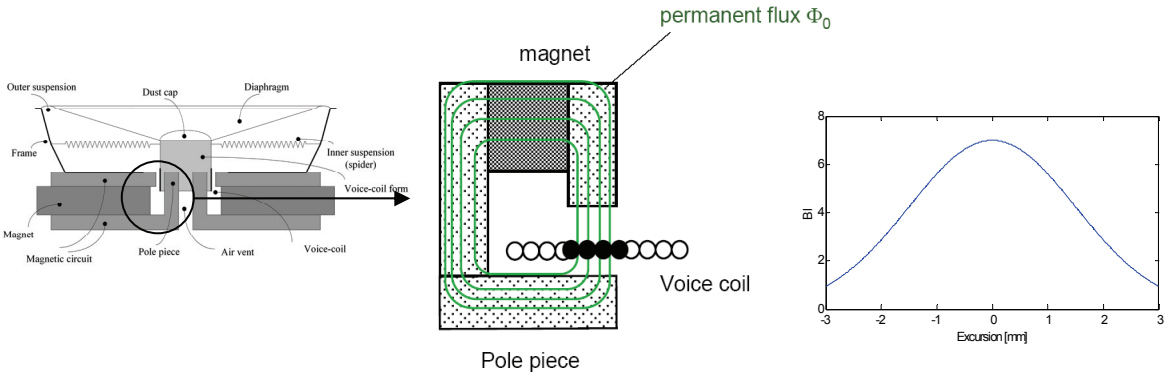


Figure 4.3. The position dependent force factor, Bl

Figure 4.3 shows a close up picture of the voice-coil inside the magnet field. The black circles represent the zero-position of the voice-coil, and the white circles represent the outer positions of the voice-coil. At large displacements parts of the coil will move out of the magnet field, and the magnetic force acting on the voice-coil is reduced. This influence on the force factor (Bl), is sketched in the right of figure 4.3.

4.1.2 The Position Dependent Suspension

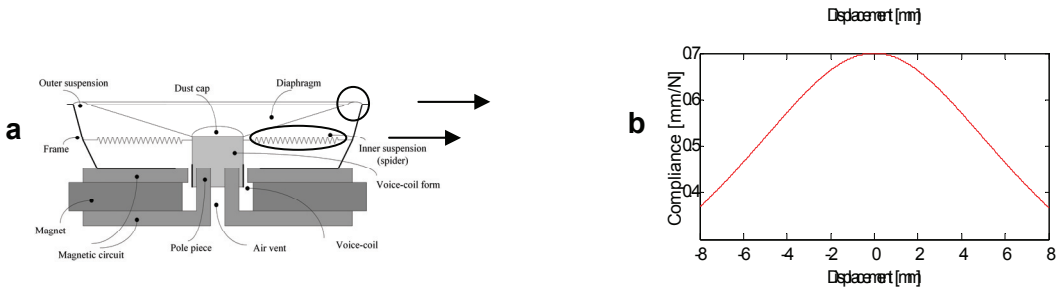


Figure 4.4. The position dependent compliance of the suspension, C

Figure 4.4, a shows the outer and inner suspension. In figure 4.4, b the position dependent compliance of the suspension is shown. The suspension is softest in the zero position of the voice-coil, and becomes less soft as the displacement increases.

4.1.3 The Position Dependent Inductance

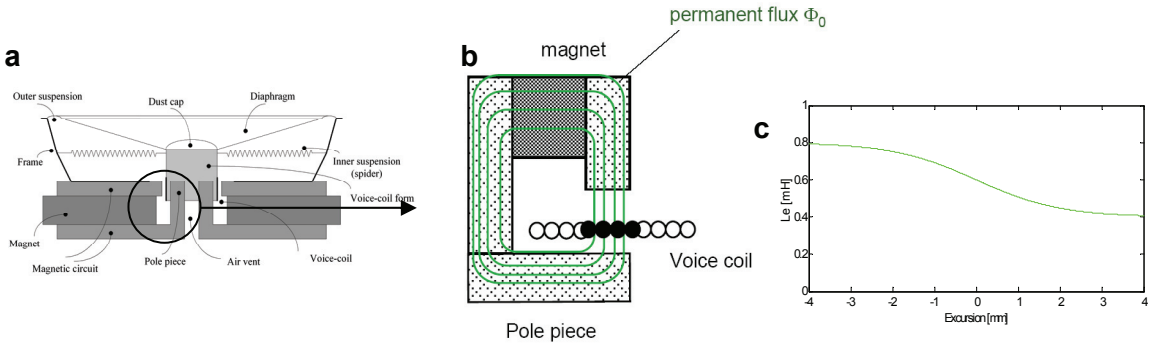


Figure 4.5. The position dependent inductance, Le

Figure 4.5, b show a close up picture of the voice-coil inside the magnet field. The black circles represent the zero-position of the voice-coil, and the white circles represent the outer positions of the voice-coil. The among of iron surrounding the voice-coil will decrease as it's moving out of the magnet, and the inductance decrease, as shown in figure 4.5, c.

4.2 Other Nonlinearities

Additionally there are a great number of other caused for nonlinear distortion, but the position dependent ones are known to be most important, and are the only one considered in this thesis. Some others are anyway mentioned below.

4.2.1 Compliance creep

Some loudspeaker suspension materials also exhibit significant creep. By creep means a "stretching" effect. The diaphragm motion continues after the voice-coil firm stops. In [Knudsen & Jensen, 1993] is this model suggested.

$$C_o(f) = C_o(f_{res}) \cdot \left[1 - \lambda \cdot \log 10(\frac{f}{f_S}) \right]$$, where λ is the creep factor.

4.2.2 The Current Dependent Inductance

The inductance is also current dependent (see figure 4.5). The current influence the permanent flux μ. As seen in the formula below will the inductance therefore also be slightly dependent of the current.

The formula for the inductance is: $L_e = \mu \cdot \frac{l \cdot N^2}{A}$

where *N is the number of turns of the voice-coil,* A is the distance from the voice-coil to the magnet, and μ is the permanent flux in the magnet.

4.3 Measuring Nonlinear Distortion (THD / IMD)

Depending on the non-linear system there are different measures of its non-linearity or in other words its distortion. Related to audio the usual measures are harmonic and intermodulation distortion. (denoted HD and IMD)

4.3.1 Harmonic distortion

Harmonic distortion means that the system will produce output at the integer multiples (harmonics) of the input frequency. The measure is defined as the ratio of the output at the n^{th} harmonics to the output at the fundamental frequency. If all the harmonics are taken into account then it is called the total harmonic distortion (THD) and is defined in the following ways:

$$THD = \frac{\sqrt{\sum_{n=2} Y(nf_0)}}{\sqrt{\sum_{n=1} Y(nf_0)}} \quad \text{or} \quad THD = \frac{\sqrt{\sum_{n=2} Y(nf_0)}}{\sqrt{Y(f_0)}}$$

In the first case the harmonics are compared to the whole output, therefore the value will always be less then 100%; in the second case they are compared only to the output at the fundamental so the value can reach infinity.

4.3.2 Intermodulation distortion

If the input of a nonlinear system contains more than one frequency then non only the harmonics but the differences and the sums of the frequencies appear. The intermodulation distortion is a simple measure in order to quantify this kind of property of a nonlinear system.
For the measurement two tones, a low and high (with respect to the bandwidth of the system) are used. In case of a loudspeaker the lower one is near the resonance frequency and the other one is at least 2.5 octave higher. The definition for IMD is the following:

$$IM_{n,1} = \frac{\left|Y(nf_1 + f_2)\right|}{Y_{RMS}} \leq \frac{\left|Y(nf_1 + f_2)\right|}{\left|Y(f_2)\right|}$$

5. Nonlinear Modelling

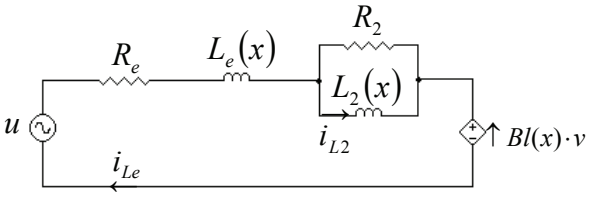
As seen in chapter 4, the loudspeaker is indeed a nonlinear component. The most important observation is that the inductance (Le), the compliance (Ct) and the force factor (Bl), all are position dependent. These parameters are in this chapter not treated as constants, like they are in the linear model in chapter 3.2.

5.1 Electrical, Mechanical and acoustical analogous Circuits.

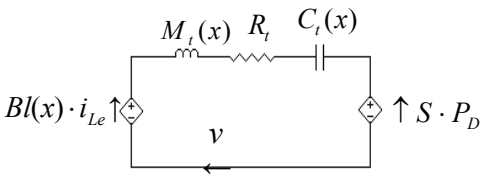
The analogous circuits are identical to the one used in the linear model in 3.1, except from a modification in the electrical circuit.(described in 3.3). The circuits are described in 3.1.

The main difference is that the inductance (Le), the compliance (Ct) and the force factor (Bl) here are dependent of the diaphragm position, *x*

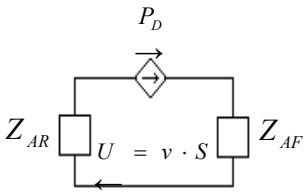
Electrical Circuit:



Mechanical Circuit:



Acoustical Circuit:



Electrical equation: (voltage)

$$u_D = R_e \cdot i_{Le} + L_e(x)\cdot\frac{\partial i_{Le}}{\partial t} + i_{Le}\cdot\frac{\partial L_e(x)}{\partial x}v + L_2(x)\cdot\frac{\partial i_{L2}}{\partial t} + i_{L_2}\cdot\frac{\partial L_2(x)}{\partial x}\cdot v + Bl(x)\cdot v$$

Mechanical equation: (force)

$$Bl(x)\cdot i_{Le} = M_t(x)\cdot a + R_t\cdot v + \frac{1}{C_t(x)}\cdot x + S\cdot P_D$$

Ac. equation: (volume velocity)

$$P_D = U\cdot Z_{AB} + U\cdot Z_{AF}$$

Figure 5.1. The electrical, mechanical and acoustical - analogous circuits, with equations, describing the nonlinear loudspeaker.

a	Loudspeaker diaphragm acceleration.	[m/ S^2]
Bl	Force factor, the product of *B(magnet field)* and *l(length of voice-coil)*.	[N/A]
C_t :	Mechanical compliance of driver suspension	[m/N]
L_2 :	Para-inductance of voice coil, due to eddy current losses.	[H]
i_{Le} :	Voice-coil current.	[A]
i_{L2}	Voice-coil eddy-current, induced in the loudspeaker's voice-coil.	[A]
L_e :	Voice coil inductance.	[H]
M_t :	Mechanical mass of driver diaphragm, air load and voice-coil.	[kg]
P_D	Pressure difference between the rear and the front of the loudspeaker diaphragm.	[Pa]
R_2 :	Electrical resistance, due to eddy current losses.	[Ω]
R_e :	Electrical voice coil resistance at DC.	[Ω]
R_t	Mechanical resistance of total-driver losses	[kg/s]
S :	The area of the diaphragm.	[m^2]
$u_D(t)$	Driver voltage. Input voltage for the loudspeaker.	[V]
U	Loudspeaker diaphragm volume velocity	[m^2 /s]

v:	Diaphragm velocity	[m/s]
Z_{AR}	Acoustical impedance at the back side of the loudspeaker diaphragm	[Pa]
Z_{AF}	Acoustical impedance in the front of the loudspeaker diaphragm	[Pa]

5.2 The State Space Model

The state space model predicts the behaviour of the loudspeaker. The expected diaphragm position, *x(n)*, diaphragm velocity, *v(n),* voice-coil current, *iLe(n),* and the eddy current, *iL2(n),* are calculated based on the electrical and mechanical circuits in figure 5.1. Vector X(n) consists of these four parameters. The driver voltage is the only input signal for the model.

The calculations of the diaphragm position, *x(n),* the diaphragm velocity, *v(n),* and the voice-coil current, *iLe(n)* are similar to the calculations done for the linear model in chapter 3.2. These are not further described for the nonlinear model.

5.2.1 Predicting the next state value of the Eddy current, $i_{L2}(n+1)$

The voltage drop of R2 and L2 in figure 5.1:

$$U_{R2} = R_2\left(i_{Le} - i_{L2}\right)$$

$$L_2 \cdot \frac{\partial i_{L2}}{\partial t} + \partial i_{L2} \cdot \frac{\partial L_2}{\partial t} = R_2\left(i_{Le} - i_{L2}\right)$$

$$L_2 \cdot \frac{\partial i_{L2}}{\partial t} = R_2\left(i_{Le} - i_{L2}\right) - i_{L2}\frac{\partial L_2}{\partial x}\frac{\partial x}{\partial t}$$

$$\frac{\partial i_{L2}}{\partial t} = \frac{R_2}{L_2} \cdot i_{Le} - \frac{R_2}{L_2} \cdot i_{L2} - \frac{i_{L2}}{L_2} \cdot \frac{\partial L_2}{\partial x} \cdot v$$

The variables are set as functions of digital discrete time, *n,* and the derivative of the current is replaced with forward Euler(described in 3.2.1):

$$\frac{i_{L2}(n+1) - i_{L2}(n)}{T_s} = \frac{R_2(n)}{L_2(n)} \cdot i_{Le}(n) - \frac{R_2(n)}{L_2(n)} i_{L2}(n) - \frac{i_{L2}(n)}{L_2(n)} \frac{\partial L_2(n)}{\partial x} \cdot v(n)$$

$$i_{L2}(n+1) = \frac{R_2(n) \cdot T_s}{L_2(n)} \cdot i_{Le}(n) - \frac{R_2(n) \cdot T_s}{L_2(n)} i_{L2}(n) - \frac{i_{L2}(n) \cdot T_s}{L_2(n)} \frac{\partial L_2(n)}{\partial x} \cdot v(n)$$

5.2.2 The Final Matrix of the State Space Model

The notation, $Bl(n) = Bl(x(n))$ is used.

$$F(n) = \begin{pmatrix} 1 - \frac{R_e + R_2(n)}{L_e(n)} T_s & \frac{R_2(n)}{L_e(n)} T_s & 0 & -\frac{T_s}{L_e(n)} \left(i_{Le}(n) \frac{\partial L_e(n)}{\partial x} + Bl(n) \right) \\ \frac{R_2(n) T_s}{L_2(n)} & 1 - \frac{R_2(n) T_s}{L_2(n)} & 0 & -\frac{T_s}{L_2(n)} i_{L2}(n) \frac{\partial L_2(n)}{\partial x} \\ 0 & 0 & 1 & T_s \\ \frac{Bl(n) T_s}{M_t} & 0 & \frac{-T_s}{C_t(n) M_t} & 1 - \frac{R_t T_s}{M_t} \end{pmatrix} \quad X(n) = \begin{bmatrix} i_{Le}(n) \\ i_{L2}(n) \\ x(n) \\ v(n) \end{bmatrix} \quad G(n) = \begin{bmatrix} \frac{T_s}{L_e(n)} \\ 0 \\ 0 \\ 0 \end{bmatrix} \quad u(n) = A \sin(2\pi f n T_s)$$

$$X(n+1) = F(n) X(n) + G(n) u(n)$$

Figure 5.2. The equations of the state space model - nonlinear modelling.

6. Loudspeaker Parameter Drifting

As mentioned in the introduction, the loudspeaker parameters drifts due to aging, temperature changes and production spread. Only drift due to temperature changes are considered in the thesis.

6.1 Temperature Drifting

Below the result of two investigations of parameter drift with respect to temperature are shown. Traditional low-frequency loudspeakers are investigated in [Krump, 1997], and a 6.5 inch loudspeaker unit is investigated in [Pedersen and Rubak, 2007].

Parameter	20 to 80oC
R_e , voice coil resistance,	20%
Bl, force factor.	-13%
M_t , moving mass	-10%
C , compliance suspension	-9%
R_t ,mechanical resistance	-42%

Parameter	20 to 50oC
R_e , voice coil resistance,	11%
Bl, force factor.	-6%
M_t , moving mass	-3%
C , compliance suspension	-21%
R_t ,mechanical resistance	-20%

Figure 6.1. Two investigations of loudspeakers parameter drift due to temperature changes. To the left:[Krump, 1997] and to the right: [Pedersen and Rubak, 2007]. Traditional low-frequency loudspeakers are investigated.

The strongest variations are found in the compliance of the suspension, *C,* and for the mechanical damping, *Rt*, while the moving mass is the most stabile parameter, as expected. In [Pedersen and Rubak, 2007], to the right, the variations in the passive mechanical system are more equally distributed between *C and Rt* . Only one loudspeaker is investigated in [Pedersen and Rubak, 2007],

The change of R_e is also significant, at 20% and 11%, about the same in both investigations due to that the temperature range is different. The result of the other parameters, are also pretty similar for the two investigations.

In Andrew Bright investigation [Bright, 2002] and [Klippel, 1998,c], it is found that the drift of the loudspeakers linear parameters are relative large, and that the drift of the nonlinear parameters are relative small.

6.2 The Drifting of the Compliance

The changes in the compliance of the suspension are known to be complex, but mainly the suspension gets softer as the temperature is increasing. The hole system is heating by the voice-coil, witch rapidly changes temperature.

In [Agerkvist, 2007] it is found that the largest changes are observed for the compliance, but also that the shape of the compliance-curve change. Bright's investigation is done for micro speakers, where the suspension is made of plastic, contra rubber materials used in traditional HIFI-speakers. Plastic materials are more stabile at these temperatures.

In [Agerkvist, 2007] is stretching and scaling of the compliance curve, used to fit the loudspeaker model.

6.3 Updating the Loudspeaker Model - Due to Parameter Drifting

Some kind of feedback from the loudspeaker is needed, to make the loudspeaker model able to follow the loudspeaker parameters drifting. Different papers suggest methods for updating the loudspeaker parameter, or often reefer to as “loudspeaker parameter identification”. In [Bright, 2002], a good overview is given. The most interesting methods use a current and voltage measurement as feedback, avoiding the traditionally problems due to motion measurements.(first described in [Klippel, 1998,c]) The impedance is analysed based on the voltage/current-measurement, and from the impedance can all the linear parameters be found. Further information can be found in [Bright, 2002].

7 Compensation of nonlinearities.

(The content of this chapter is briefly described in the introduction.)

There are three basic methods for compensation of nonlinearities, due to loudspeakers, or in general:

1. The negative feedback system.
2. The feedforward system.
3. The adaptive feedforward system.

7.1 The Negative Feedback System

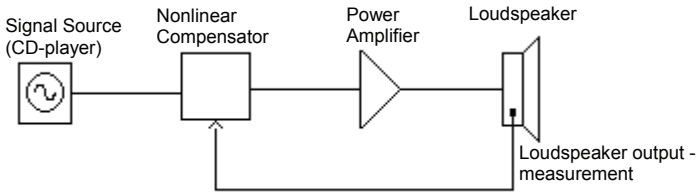


Figure 7.1. The negative feedback system

Parts of a system's output signal, is feed to the input in reverse. The system becomes more stabile due to temperature drifting, the linearity improves – all traded for a reduction of gain. Drawbacks are the need for a physical measurement of the feedback signal, and problems due to instability caused by oscillations.

Negative feedback control is widely used for audio amplifiers, as compensation of amplifier nonlinearities. The method was invented by Harold Stephen Black at Bell Laboratories in 1927. Nonlinear compensation systems for loudspeakers have not been a commercial success, though the idea of a negative feedback system was already proposed by Voigt in 1925. Problems are related to the output signal measurement. Methods have so far been considered to be too complicated or expensive for commercial use.

7.7.1 Different proposals for measuring the output of the loudspeaker:

1. Sound pressure, near-field or in listing position - using of a microphone.
2. Diaphragm acceleration - using an accelerometer.
3. Diaphragm position - using a laser.
4. Diaphragm velocity - using a secondary magnetic system / a second winding of the primary coil.

Sound pressure measurements:

The drawback of measuring the sound pressure is the influence of the room. Wall reflections and room-modes makes the sound pressure strongly dependent of position. It is possible to measure in the near-field, but the system would still need to be calibrated regularly.

Diaphragm measurement using an accelerometer

Disadvantages with the system are the extra weight due to the accelerometer, and its traditionally high cost. Today accelerometers have become smaller, lighter and cheaper, making the whole concept far more actually. Still, just a smaller minority of subwoofer producers are using them as part of a negative feedback system. A reason for it could be that the need for calibration results in high production costs.

Diaphragm position measurement using a laser

A laser measurement of the diaphragm position is the most accurate method available, but laser technology is too expensive for commercial use. It may be actual for real expensive active systems, with a laser placed inside the loudspeaker cabinet, measuring the rear side of the diaphragm.

Diaphragm velocity measurement using a secondary magnet system

In 1927 Hanna published a description of a motion feedback system, using a secondary magnet system for monitoring the diaphragm velocity. A cheaper solution is to add a secondary winding to the primary coil.

7.2 The Feedforward System.

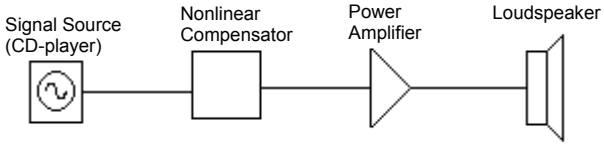


Figure 7.2. The feedforward system

For compensation of linear distortion the feedforward system are well used. The loudspeakers frequency response is corrected by use of tradition analogous filters. In active systems, digital filters are used, also giving the opportunity to lower the loudspeakers cut-off frequency.

The idea of Nonlinear compensation using the feedforward system:

The nonlinearities of the loudspeaker are predicted by the feedforward processor, either by real-time modelling of the loudspeaker (state space model), or by using pre-stored values (“look-up table”). Based on this information the inverse of the nonlinearities is added to input signal, so the unwanted nonlinearities of the loudspeaker will cancel out.

In 1992 Wolfgang Klippel published his famous paper “The mirror filter - a new basis for reducing nonlinear distortion and equalizing response in woofer systems.” ([Klippel, 1992]), presenting a feedforward system for compensating of linear and nonlinear distortion. Based on measurements of the loudspeaker nonlinearities and the state space model, the input signal is pre-distorted before entering the loudspeaker to cancel out loudspeaker nonlinearities.

In [Schurer, Slump and Herrmann, 1998] a “feedback linearization”-algorithm is presented. Klippel did later recognise that the mirror filter is based on the same principle as the feedback linearizator. The difference is that Klippel are pre-processing the input signal directly, while in [Schurer, Slump and Herrmann, 1998], the input signal is pre-processed based output of the compensator. Klippel has patented his mirror-compensator.

The compensator algorithm in [Schurer, Slump and Herrmann, 1998] is chosen in for this project, and further described in 7.4).

The quality of the compensation for a these pure feedforward systems totally depends on how accurate the loudspeaker model is. Drifting of loudspeaker parameters will reduce the performing, since no feedback is included.

7.3 The Adaptive Feedforward System.

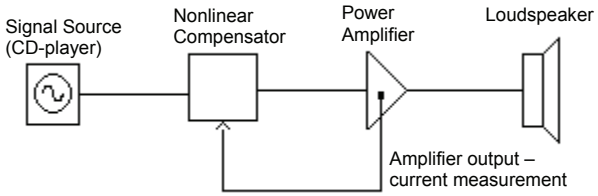


Figure 7.3. The adaptive feedforward system

Additionally to the pure feedforward system, a feedback signal from the loudspeaker is used for updating the parameters of the loudspeaker model, making the compensator able to handle loudspeaker parameters drifting. This is shortly described in 6.3.

This updating of the linear parameter is subject for further work.

7.4 The Chosen Compensation Algorithm.

The feedback linearization in [Schurer, Slump and Herrmann, 1998], is chosen as compensator algorithm. In retrospect it is seen like a feedforward controller.

7.4.1 Schematic diagram

The schematic diagram of the compensator is shown in figure 7.4

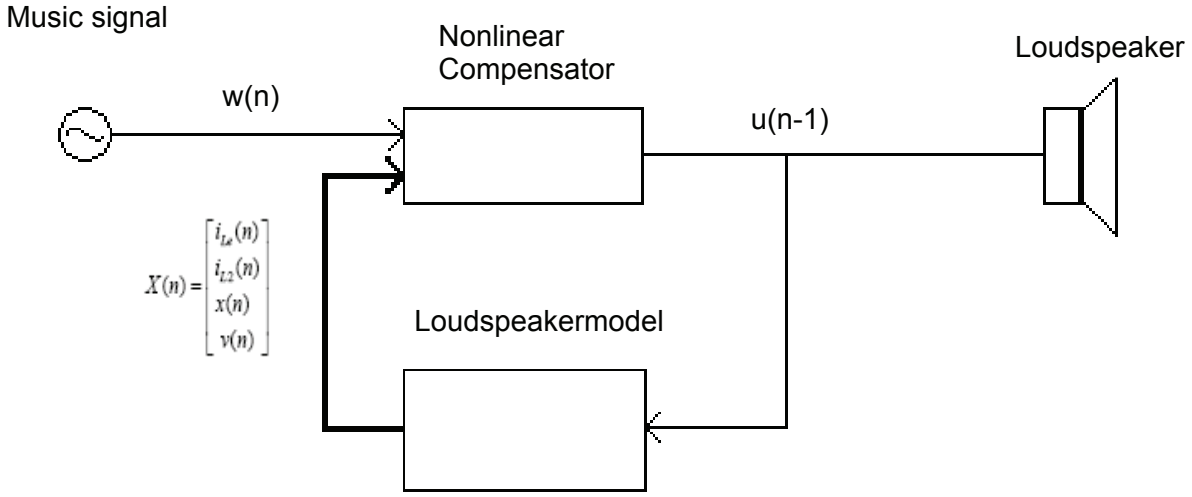


Figure 7.4. Schematic drawing of the compensator in [Schurer, Slump and Herrmann, 1998],

Signal (discrete,digital)		**Description**	**Unit**
w(n)		Input music signal	Voltage [V]
u(n-1)		Output music signal (one sample delayed)	Voltage [V]
X(n)		Predicted state vector (one sample into the future)	
	$i_{Le}(n)$	Predicted output current	Amper [A]
	$i_{L2}(n)$	Predicted eddycurrent	Amper [A]
	$x(n)$	Predicted coil-position, displacement	Meter [m]
	$v(n)$	Prediction coil-velocity	Meter/Second [m/s]

7.4.2 Compensator algorithm

The diaphragm position, x, the diaphragm velocity, v, and the voice-coil current, iLe, are functions of n.

$$u(n-1) = \frac{Le(x)}{Bl(x) + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le}} \left\{ v \cdot \left[\frac{\partial k(x)}{\partial x} \cdot x + \left[k(x) - k_0 \right] - \frac{\partial Bl(x)}{\partial x} \cdot i_{Le} - \frac{\partial^2 L_e(x)}{2 \cdot \partial^2 x} i_{Le}^{\ 2} - \frac{Bl^2}{\frac{\partial L_e(x)}{\partial x}} \right] \right.$$

$$\left. + \frac{R_e(k(x) - k_0)}{Le_0} x - \frac{R_e \left[2Bl(x) + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le} \right]}{2Le_0} i_{Le} + \frac{Bl_0}{Le_0} w(n) \right\} + R_e \cdot i_{Le} + Bl(x) \cdot v + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le} \cdot v$$

Parameter	**Description**	**Unit**
Bl	Force factor	[N/A]
K	Stiffness (1/Compliance)	[N/m]
Le	Inductance	[H]
Blx	First derivative of Bl with respect to x	[N/Am]
kx	First derivative of K with respect to x	[N/m2]
Lex	First derivative of Le with respect to x	[H/m]
Lexx	Second derivative of Le with respect to x	[H/m2]
Bl_0	Force factor at x=0, linear parameter	[N/A]
K_0	Stiffness of the compliance at x=0, linear parameter	[N/m]
Le_0	Inductance at x=0, linear parameter	[H]
Re	Coil resistance	[Ω]
Rt	Total mechanical resistance	[Kg/s]
L2	Eddycurrent inductance	[H]
R2	Eddycurrent resistor	[Ω]
Mt	Total mechanical mass	[Kg]
dLe/dx	First derivate of “Le” with respect to position	[H/m]
dL2/dx	First derivate of “L2” with respect to position	[H/m]
Ts	1/samplingfrequency	[s]

Signal (discrete,digital)	**Description**	**Unit**
w(n)	Input music signal	Voltage [V]
u(n-1)	Output music signal (one sample delayed)	Voltage [V]
iLe	Output current	Amper [A]
iL2	Eddycurrent	Amper [A]
x	Coil-position, displacement	Meter [m]
v	Voil-velocity	Meter/Second [m/s]

The three last terms in the equation are the voltage drop of the Re, the voltage drop due to the mechanical system, and the voltage generated by the loudspeaker itself. (see 3.1.1)

7.4.3 Modified Compensator algorithm - Including eddy-currents

The nonlinear compensation-algorithm article [Schurer, Slump and Herrmann, 1998] is designed for the simpler loudspeaker model described in 3.1. The eddy current is not included.

To fit the compensator to the extended model in figure 3.3, the expression $R_2\left[i_{Le} - i_{L2}\right]$ is simply added to include the voltage drop of the eddy currents. The result with and without this modification is shown in 9.4.

$$u(n-1) = \frac{Le(x)}{Bl(x) + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le}} \left\{ v \cdot \left[\frac{\partial k(x)}{\partial x} \cdot x + \left[k(x) - k_0\right] - \frac{\partial Bl(x)}{\partial x} \cdot i_{Le} - \frac{\partial^2 L_e(x)}{2 \cdot \partial^2 x} i_{Le}{}^2 - \frac{Bl^2}{\frac{\partial L_e(x)}{\partial x}} \right] \right.$$

$$\left. + \frac{R_e(k(x) - k_0)}{Le_0} x - \frac{R_e\left[2Bl(x) + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le}\right]}{2Le_0} i_{Le} + \frac{Bl_0}{Le_0} w(n) \right\} + R_e \cdot i_{Le} + Bl(x) \cdot v + \frac{\partial L_e(x)}{\partial x} \cdot i_{Le} \cdot v + R_2\left[i_{Le} - i_{L2}\right]$$

8. Loudspeaker - Measurements

8.1 Loudspeaker Parameters Measurement

8.1.1 The Klippel Analyser

The Klippel Analyser measures the parameters of a loudspeaker, both linear and nonlinear. The system is developed and patented by Wolfgang Klippel. The analyser can be linked to a standard windows computer, controlled by the software program db-Lab.

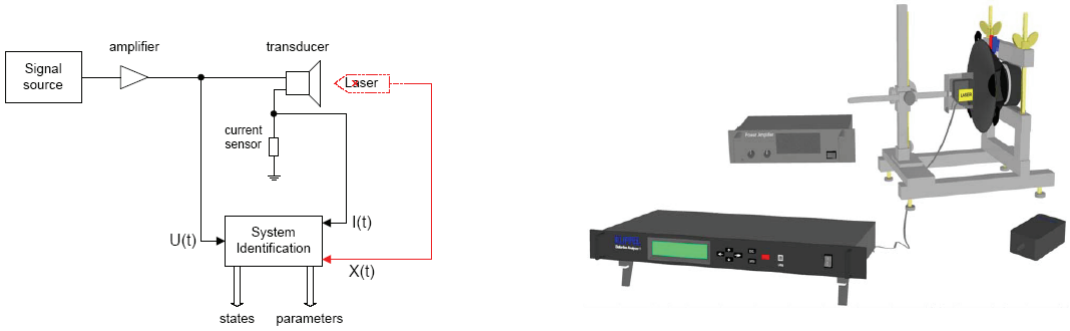


Figure 8.1. The Klippel Analyser. The system consists of a laser, a rack, and the analyser. To complete the setup a power amplifier and a windows computer running "dB-lab" software is required.

The loudspeaker is mounted in a rack, as seen in figure 8.1. The complete system contains of the analyser, a power amplifier and a windows computer running db-Lab software.

The operation of the Klippel analyser is based on system identification. Instead of directly measuring the nonlinear parameters, it uses a nonlinear loudspeaker model and extracts the parameters from that. In order to achieve this it measures the input and the output of the loudspeaker and adjusts the model so that it gives the same response as the real speaker for the same input. The nonlinear parameters are approximated with 8^{th} order polynomials. Parameters aren't valid outside of this range, so this simulation cannot be used to simulate driver behaviour at the physical limits of the excursion.

8.1.1.1 Measuring the Diaphragm Diameter.

The diameter of the diaphragm is required for the measurements. The diameter is measured with half of the surround included. Part of the surround is moving, while the other part is fixed, this is approximated by taking half of it in account in the measurement.

8.1.1.2 Laser Adjustment.

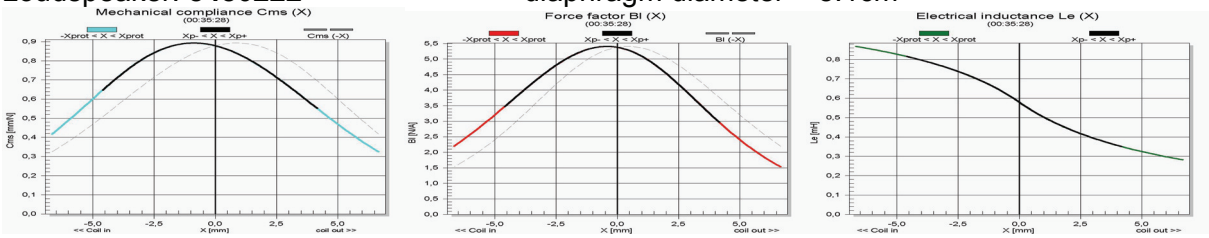
To avoid distortion in the laser measurement, the distance from the diaphragm to the laser-head has to be proper tuned. The range of the laser is narrow so it has to be set exactly to the mid position. A led-indicator is placed on the laser. A pulsing led indicates that the laser is in the right area, but still out of range. Just where the led changes from pulsing to permanently lightning, indicates the outer position. The optimal laser position is between the two outer positions.

8.1.2 The Result of Measurement 1.

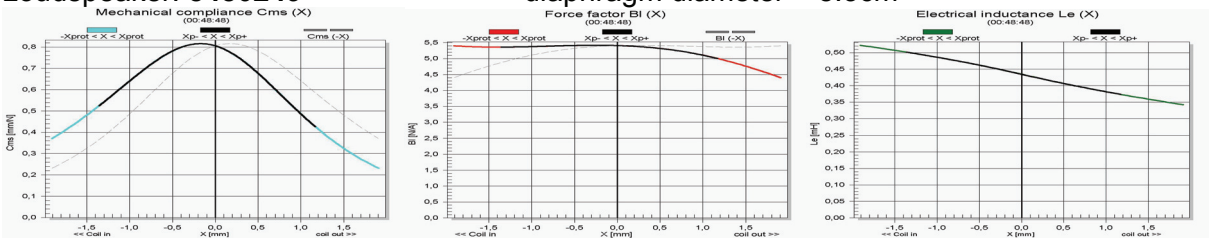
Seven loudspeakers were measured for this thesis. The result of the nonlinear measurement is shown in figure 8.2. Based on the result, three of the loudspeakers are chosen for further investigation. The measurement is found stored in:
\Measurements\KlippelAnalysator\measurements_28feb_B&O

Loudspeaker: **8480021** diaphragm diameter = 6.9cm

Loudspeaker: **8480222** diaphragm diameter = 8.1cm

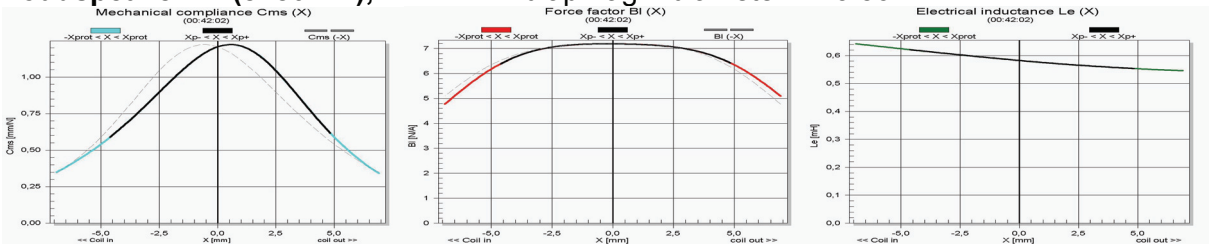


Loudspeaker: **8480249** diaphragm diameter = 8.3cm

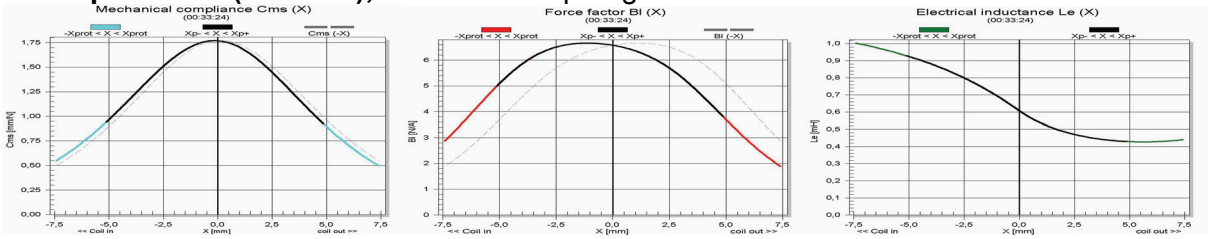


Loudspeaker: A (8480255), diaphragm diameter = 8.3cm

Loudspeaker: B (8480277), diaphragm diameter = 13.5cm



Loudspeaker: C (8480285), diaphragm diameter = 10.0cm



Loudspeaker: Tymphany 205-116 QC.131005 diaphragm diameter = 20.7cm

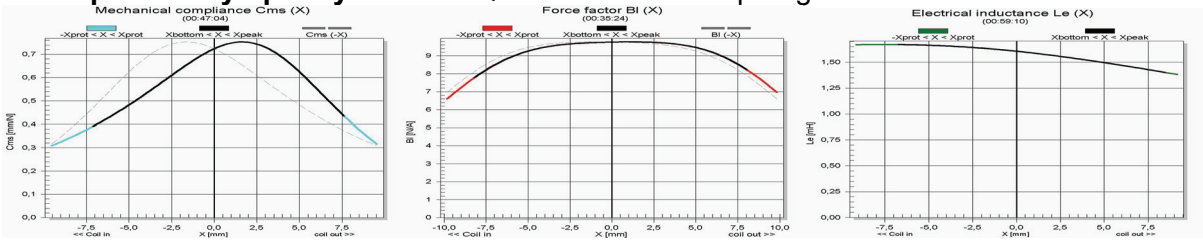


Figure 8.2 The result of measurement 1- nonlinear parameters. The black lines are the measured result and the colour lines are the polynom fitting done by the klippel analysator.

8.1.2.1 Measurement result.

The black lines are the measured result and the colour line is the polynom fitting done by the klippel analysator, to model the parameters. These polynomes are later used in the loudspeaker modelling.

The measured inductance for loudspeaker:8480021, not seems to be reliable, due to that the inductance increases with displacement. When the voice coil is moving away from the magnet, the inductance should decrease, due to the description in 4.1.3.

The measured force-factor for loudspeaker:84800249 is also a bit strange. The force factor should also decrease when the voice-coil is moving away from the magnet gap, due to the description in 4.1.1.

The loudspeakers who is chosen for futher investigation is

- “Loudspeaker A” - has a relatively symmetric Bl-factor and an unsymmetrical compliance.
- “Loudspeaker B” has both a relatively symmetric Bl-factor and compliance.
- “Loudspeaker C” has a relatively unsymmetrical Bl-factor and a symmetrical compliance.

8.1.3 The Results of Measurement 2 – updated software.

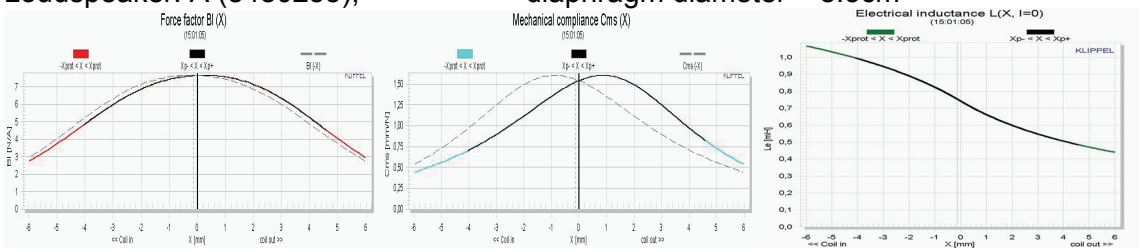
While I stayed at B&O in Struer, the dB-lab software was upgraded. Improved specifications for this “new generation LSI”-software are:

- -The inductance is not only measured with respect to position but also with respect to the current, described in 4.2.2.
- Contains of a “automatic heating mode”. - measures the thermal parameters. (runs for one hour).
- “Displacement function” - estimates coil displacement using current and voltage. Displays peak and bottom values, and compare them to the real displacement.

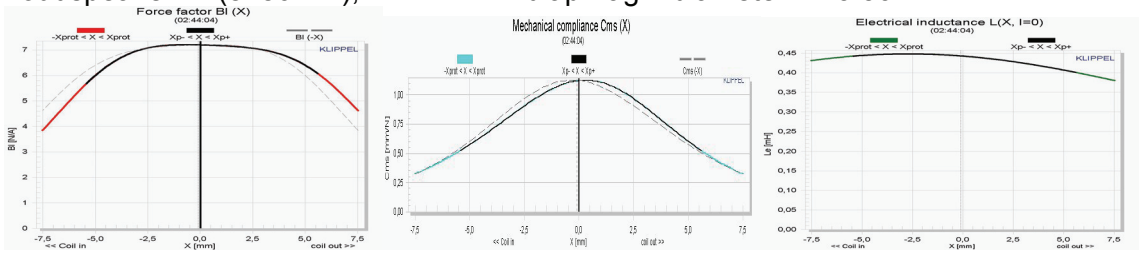
(Information is can be found on Klippels webside, www.klippel.de).

The result of measurements 2 are shown in figure 8.3. The measurement is stored in: \Measurements\KlippelAnalysator\measurements_06mars_B&O

Loudspeaker: A (8480255), diaphragm diameter = 8.3cm



Loudspeaker: B (8480277), diaphragm diameter = 13.5cm



Loudspeaker: C (8480285), diaphragm diameter = 10.0cm

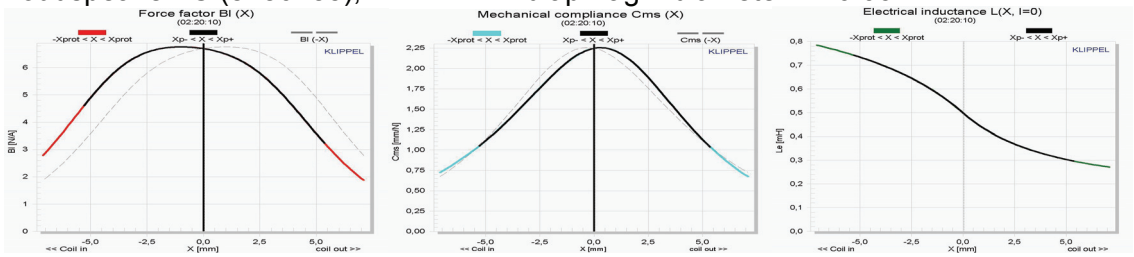


Figure 8.3 The result of measurement 2- nonlinear parameters. The black lines are the measured result and the colour lines are the polynom fitting done by the klippel analysator.

8.1.4 Comparing the Results of Measurement 1 and 2

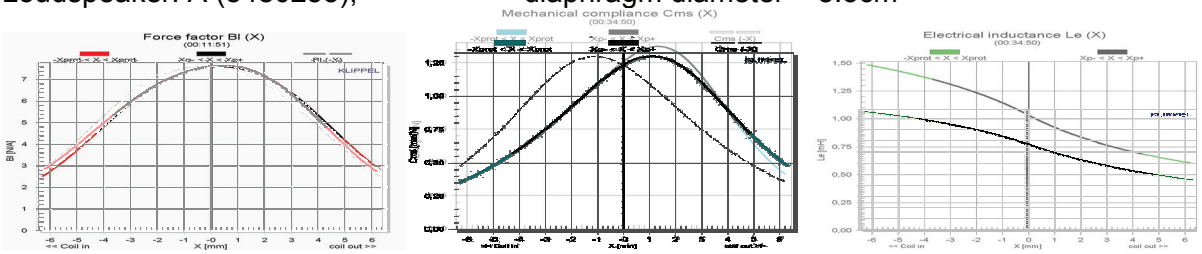
The result of measurement 1 and measurement 2, are compared in figure 8.4. The darkest curves are the results from measurement 2, and the grey curves are measurement 1. As seen are the inductance-curves changed. The new software is more accurate. (The lowest curves are from measurement 2, for the inductance)

The differences seen in the compliance of the suspension, is caused by changes in the suspension itself, and not by the software update. In 8.6 two measurements performed at the new software are compared, showing the same result.

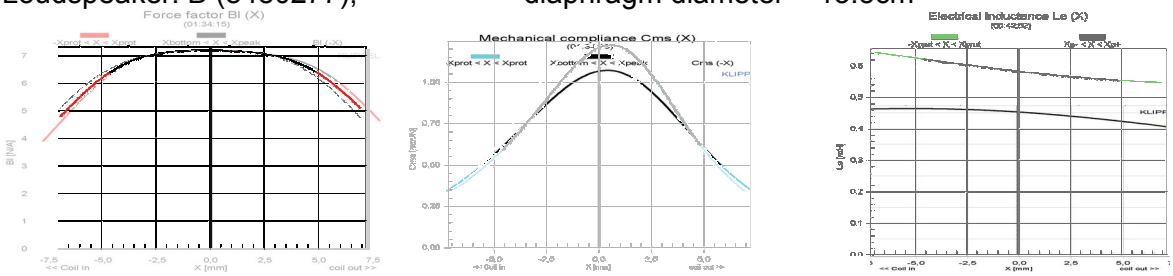
The force factor Bl has not changed.

The measurement is found in \Measurements\Klippel Analysator\measurements_06mars_B&O and \Measurements\Klippel Analysator\measurements_28feb_B&O

Loudspeaker: A (8480255), diaphragm diameter = 8.3cm



Loudspeaker: B (8480277), diaphragm diameter = 13.5cm



Loudspeaker: C (8480285), diaphragm diameter = 10.0cm

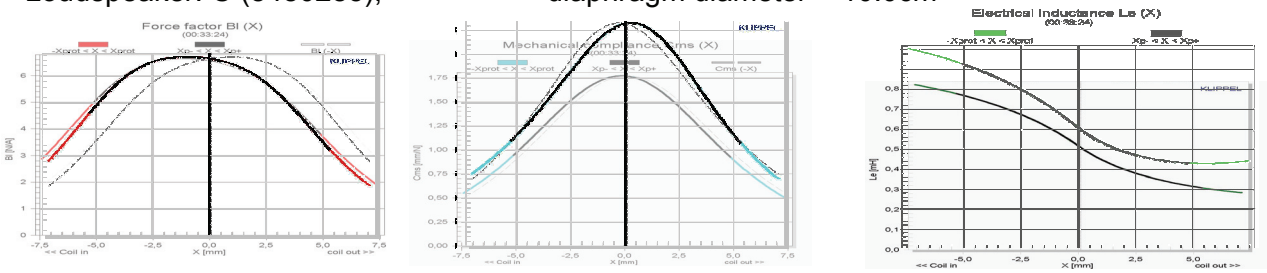


Figure 8.4 The result of measurement 1 and 2 are compared. Grey curves – measurement 1 (old version). Black curves – measurement 2 (new version) (The black lines are the measured result and the colour lines are the polynom fitting done by the klippel analysator).

8.1.5 Linear Parameters – Measurement.

Parameter	A(8480255)	B(8480277)	C(8480285)	Unit
Bl_0	7.65893	7.18583	6.70246	[N/A]
C_0	6.83108e-004	0.000538315	1.12449e-003	[m/N]
L_{e_0}	5.25436e-004	0.000389793	4.08976e-004	[H]
R_e	12.6263	5.02882	12.8636	[H]
R_2	5.65685	2.67697	3.36085	[Ω]
L_2	7.68891e-004	0.000565304	5.6789e-004	[H]
M_{mt}	6.15617e-003	0.0195471	8.82478e-003	[kg]
R_{mt}	1.16119	2.16858	0.589303	[Ω]
Fs	77.6104	49.0638	50.5232	[Hz]

Figure 8.5. The linear parameters measured with the Klippel Analyser.

Figure 8.5 contains the linear parameter of the three loudspeaker which where chosen for further investigation. The measurement is found in :
\Measurements\Klippel Analysator\measurements_06mars_B&O

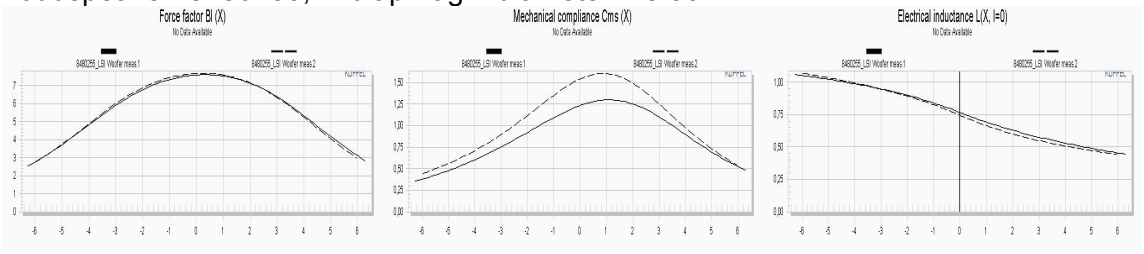
8.2 Nonlinear Parameters Drifting – Measurement.

In figure 8.6 two identical measurement of the nonlinear parameters are compared, for the three chosen loudspeakers. Loudspeaker “8480255” was in use for a while (heated) before the second measurement started. The compliance change a lot (becomes softer), while the Bl-factor and the inductance hardly drift at all. The shape of the compliance curve does not change dramatically, but changes are seen, according to Agerkvist’s investigation [Agerkvist, 2007]. (see 6.2) Loudspeaker “8480277” and “8480285” are not heated before the second measurement. Variations in the compliance are still seen.

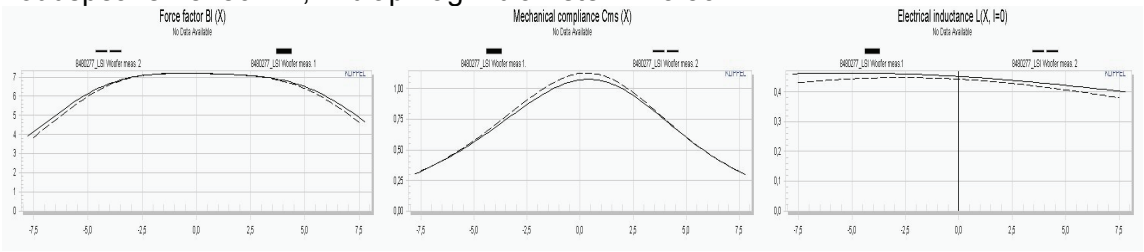
In figure 8.4 the measured result from the new and the old software of dB-Lab are compared. The changes seen in the compliance curves are caused by changes in the compliance itself, not by the software update.

The data is found in \Measurements\Klippel Analysator\sammenligning_LSI Woofer meas 1og2.

Loudspeaker: 8480255, diaphragm diameter = 8.3cm



Loudspeaker: 8480277, diaphragm diameter = 13.5cm



Loudspeaker: 8480285, diaphragm diameter = 10.0cm

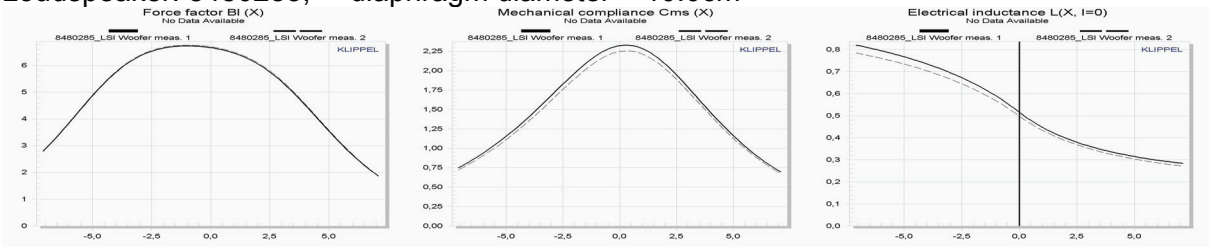


Figure 8.6. Two similar measurements are compared to detect parameter drifting

8.3 Loudspeaker Output Measurement – Displacement.

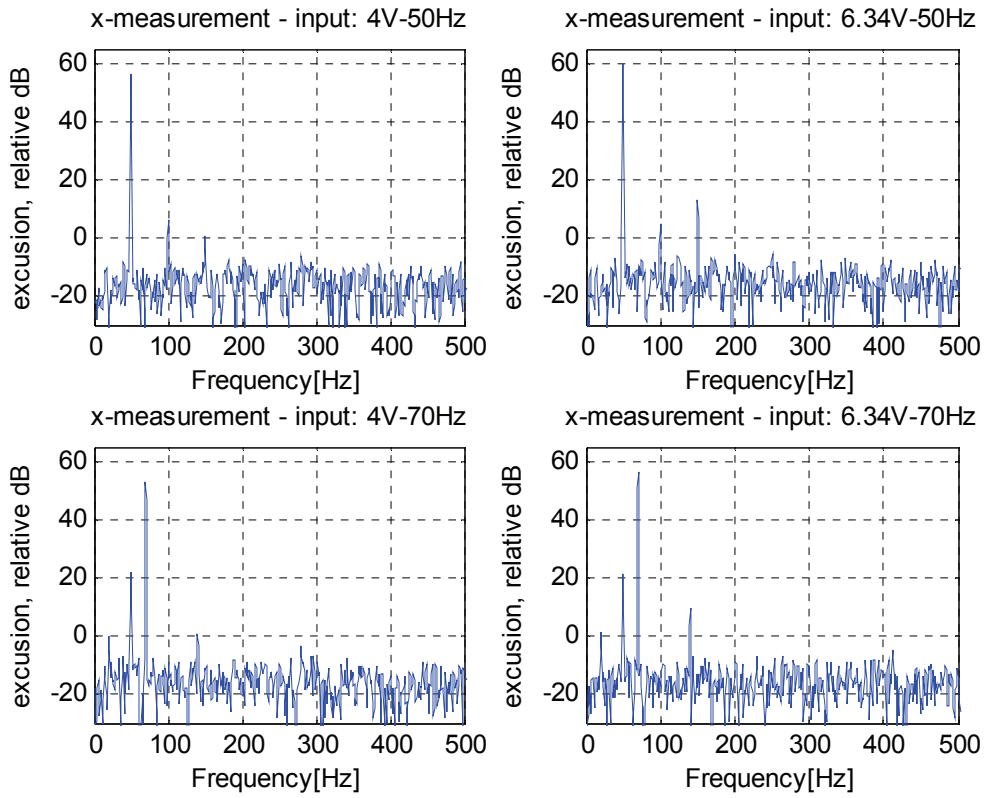


Figure 8.7 Excursion measurements. The Klippel Analyser's laser is used. The excursion signal is available at the X-output of the rear of the analyser. The voltage-amplitude are given in the figure. The level is to 1W and 2W corresponding to 8Ω.

The measurement is done at Bang and Olufsen, Struer. They're computer measurement system is used as frequency analyser, and the laser on the Klippel Analyser's is used as excursion measurement. The measured values are relative decibel values. The absolute excursion values are not identified.

More extensive excursion measurements were done at DTU. The measurement results are unfortunate useless. Some kind of unknown distortion/overdrive have influenced on the measurement.

net-power infection

The net-power infects the measurement, as seen in figure 8.7(for 70Hz). The 50Hz component is seen about 30 dB below the 70Hz signal. Due to intermodulation occurs also a 20Hz component. The third harmonic of the net power also detected. The magnitude of the net-power infection are small, anyway the problem would have been avoided by using another frequency than 50Hz.

9. Matlab simulation

9.1 Overview of Matlab functions

9.1.1 Linear_Loudspeaker_Model_Simplified **Appendix A1**
Simulation of the linear model in 3.2.6.

9.1.2 Nonlinear_Loudspeaker_Model_Simplified **Appendix A2**
Simulation of the nonlinear model without eddycurrent, The model in 3.2.6 is used with nonlinear parameters (Bl(x), C(x) and Le(x))

9.1.4 Nonlinear_Loudspeaker_Model **Appendix A3**
Simulation of the nonlinear model in 5.2.1..

9.1.5 Nonlinear_Compensator_Simplified **Appendix A4**
Simulation of the nonlinear compensation algorithm in 7.4.2

9.1.6 Nonlinear_Compensator **Appendix A5**
Simulation of the nonlinear compensation algorithm in 7.4.3

9.1.7 Compliance_Adjustment **Appendix A6**
Stretching and scaling the compliance curve

9.1.8 THD_Calculator **Appendix A7**
Function for the THD-calculation due to 4.3.1

9.1.9 IMD_Calculator **Appendix A8**
Function for the IMD-calculation in 4.3.2

9.1.10 Load_Nonlinear_Parameters **Appendix A9**
Loading polynomials &
Generating tables for the nonlinear parameters

9.1.11 Load_Linear_Parameters **Appendix A10**
Loading linear parameter

9.1.12 Plot_Nonlinear_Parameter **Appendix A11**
Plot the curves of the nonlinear parameters

9.1.13 Capture.m **Appendix A12**
VisualAudio to Matlab interface, previous developed at B&O

9.1.14 Main.m **Appendix A13**
Script to call the other functions call

9.2 The Modelling of the Nonlinear Parameter

The nonlinear parameters are approximated with 8th order polynomials in the Klippel analyser. Polynomials are also used in the loudspeaker model to model the nonlinear parameters.

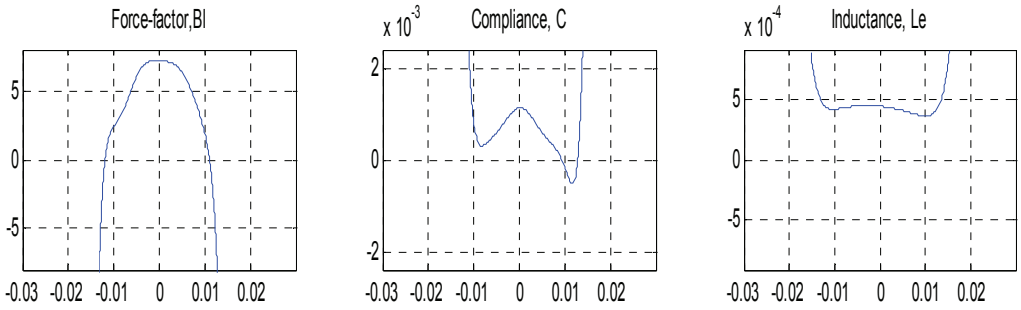


Figure 9.1 The nonlinear parameter plotted by polynomials

As seen in figure 9.1 are the polynomials not reliable outside the measured excursion. They diverge rapidly outside the range of the measurement. The parameters are measured within about ±4mm. This is found by looking at the black curve in figure 8.3, the nonlinear measurement result. (explaned in 8.1.2.1)

For better modelling other methods are recommended. [Agerkvist, 2007] and [Andersen, 2005] For Bl-factor and compliance modelling is inverse polynomial or sum of Gaussians suggested, and for the inductance is sigmoid functions suggested.

This is not given priority in this thesis.

9.2.1 Stretching / Scaling of the compliance

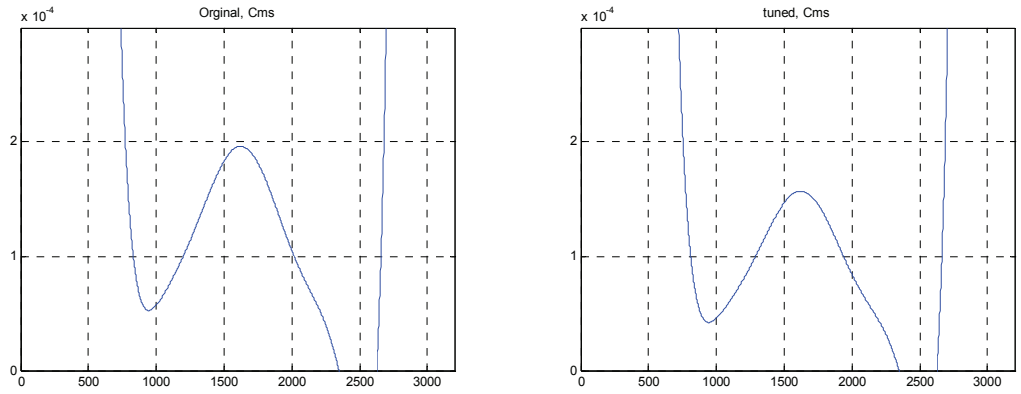


Figure 9.2 The nonlinear compliance of the suspension is scaled by a factor of 0.8

ddLe

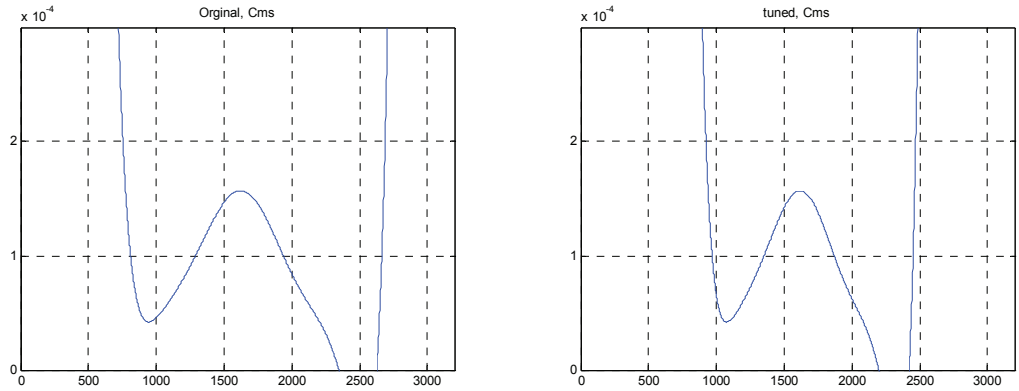


Figure 9.3 The nonlinear compliance of the suspension is stretched by a factor of 0.8

Since the compliance is the parameter that is mostly drifting, these simple adjustment technique is used to fit the loudspeaker model to the loudspeaker.

9.3 The Loudspeaker Model

The loudspeaker model based on the state space model in 5.2.1 is compared to the real loudspeaker model 50Hz/1W and 70 Hz/1W

9.3.1 Simulation in Matlab - Compared to Loudspeaker Measurement.

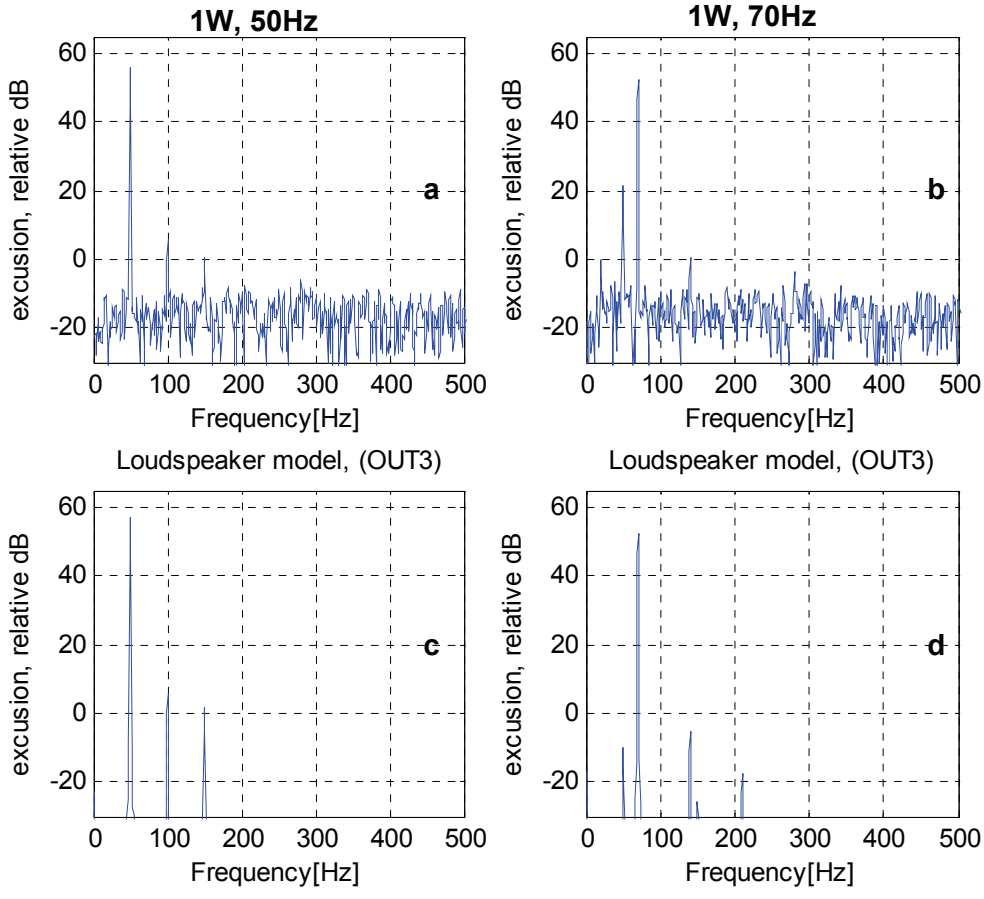


Figure 9.4. Measurement result. a,b: Measured loudspeaker excursion. c,d Loudspeaker model – excursion.

The measurement is done at Bang and Olufsen, Struer. They're computer measurement system is used as frequency analyser, and the laser on the Klippel Analyser's is used as excursion measurement. The measured values are relative decibel values. The absolute excursion values are not identified.

As seen in the figure below does the loudspeaker model fit the loudspeaker pretty well. The difference for the second harmonic is 0.8 dB is

"Absolute values"

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Loudspeaker	56,2	5,9	0,7
Loudspeaker model	57	5,9	1,5

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Loudspeaker	0	-50.3	-55.5
Loudspeaker model	0	-51.1	-55,5

1W -70Hz

The loudspeaker model does not fit the loudspeaker. The harmonics are underestimated in the loudspeaker model.

Absolute values

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Loudspeaker	52,7	0,8	-3.3
Loudspeaker model	52,6	-5,2	-

Relative values (the fundamental is scaled to 0 dB

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Loudspeaker	0	-51.9	-56
Loudspeaker model	0	-57.8	-

9.3.2 Model with Compliance Adjustment.- Compared to Real Loudspeaker.

The compliance curve is stretch and scaled with the factors shown below, to make the loudspeaker model fit the loudspeaker.

- C_{sca} =0.85, C_{str} = 0.92, *G*=4, $f_{w(n)}$ = 50
- C_{sca} =0.6, C_{str} = 1, G=4, $f_{w(n)}$ =70.

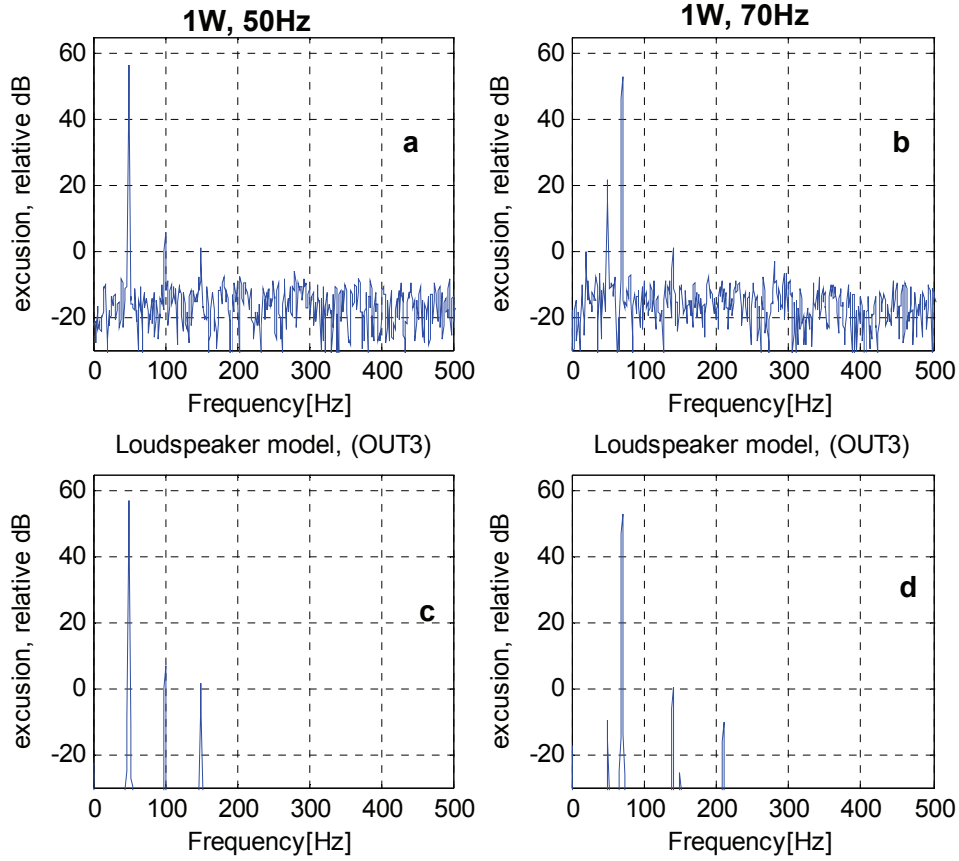


Figure 9.5. Measurement result. a,b: Measured loudspeaker excursion. c,d Loudspeaker model – excursion.

50Hz-measurement: the compliance are rescaled by a factor of 0.85 and stretched by a factor of 0.92.
70Hz-meausrement: the compliance are scaled by a factor of 0.6 and stretched by a factor of 1.

1W - 50Hz

The loudspeaker model is tuned by scaling and stretching the compliance curve. The stretching factor at 0.92 and the scaling factor at 0.85, gave the best result.

Absolute values

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Loudspeaker	56,2	5,9	0,7
Loudspeaker model	57	6.7	1,6

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Loudspeaker	0	-50.3	-55.5
Loudspeaker model	0	-50.3	-55,4

1W -70Hz

The loudspeaker model is tuned by scaling the compliance curve.

Absolute values

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Loudspeaker	52,7	0,8	-3.3
Loudspeaker model	53,1	-0,2	-

Relative values (the fundamental is scaled to 0 dB)

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Loudspeaker	0	-51.9	-56
Loudspeaker model	0	-53.3	-

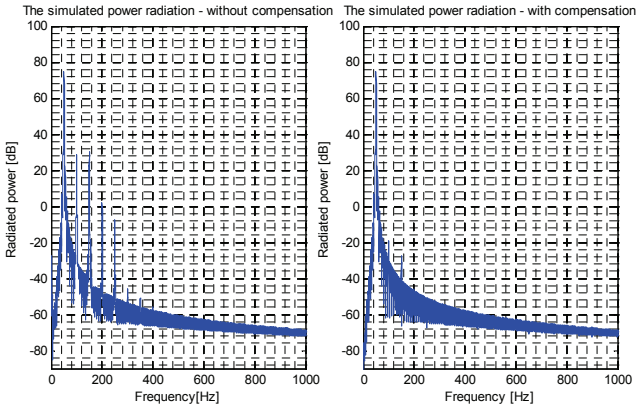
9.4 The Nonlinear Compensator

The formula given in 2.4 is used to calculate the simulated power radiation used for the THD and IMD calculations. Relative values are calculated.

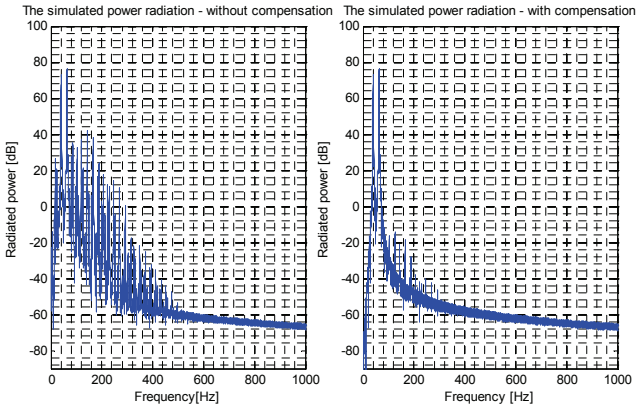
9.4.1 Simulation Result 1 – model and compensator without eddy current

The Loudspeker Model and the compensator are both simulated without eddy current. Loudspeaker model in 3.2.6, and compensator algorithm in 7.4.2.

50Hz/1W:



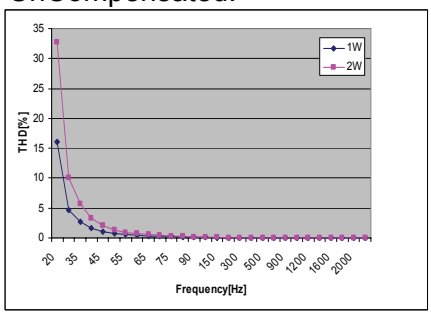
50Hz/1W + 70Hz/1W:



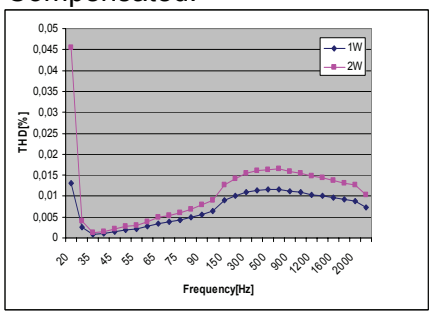
THD and IMD calculations - of the Simulated Power Radiation.

THD →

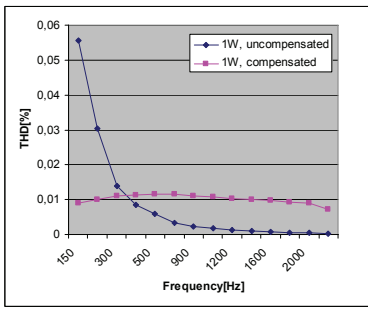
UnCompensated:



Compensated:

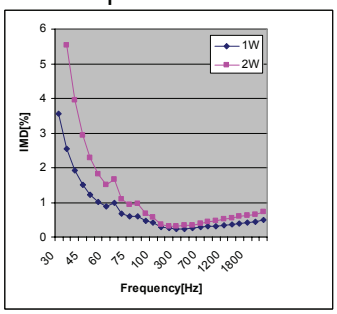


Together

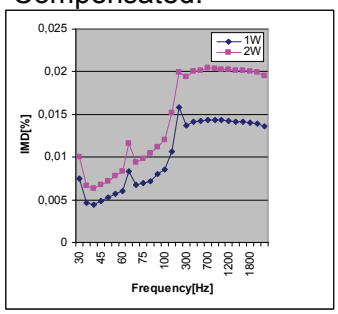


IMD →

UnCompensated:



Compensated:

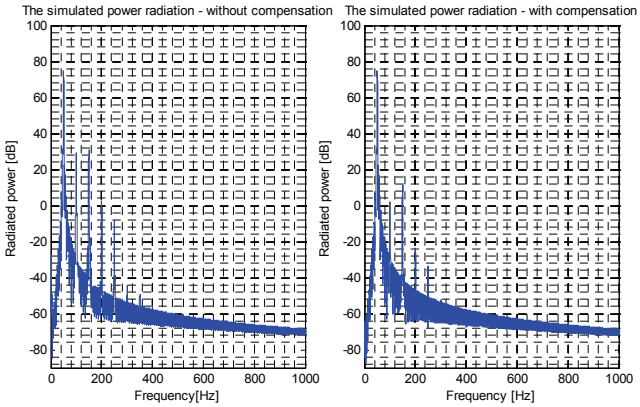


9.6 Loudspeaker model and comparator are simulated without eddy current.

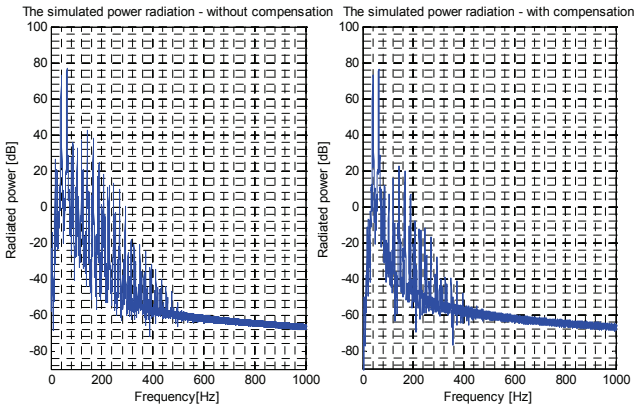
9.3.2 Simulation Result 2 – model with, compensator without eddy current

The Loudspeaker model is simulated with the model including eddy current (in 5.2.1) and the compensator algorithm without eddy current are used. (7.4.2)

50Hz/1W:



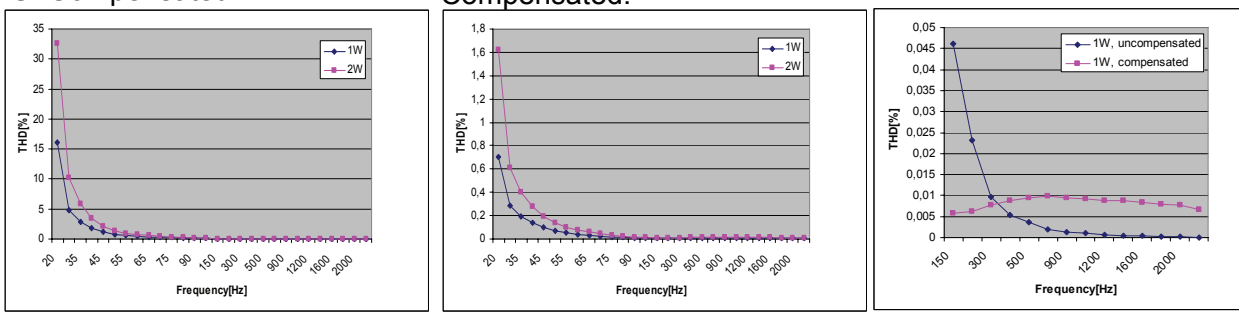
50Hz/1W + 70Hz/1W:



THD and IMD calculations - of the Simulated Power Radiation.

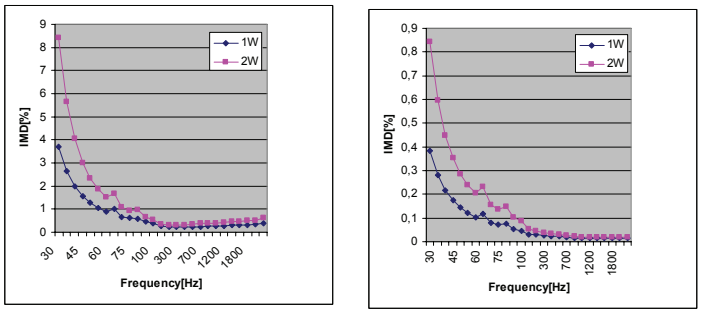
UnCompensated: Compensated:

THD



UnCompensated: Compensated:

IMD →

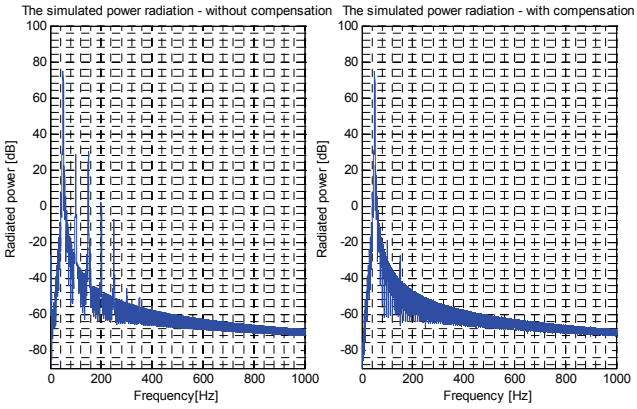


9.7 Loudspeaker model is simulated with eddy current and the comparator without.

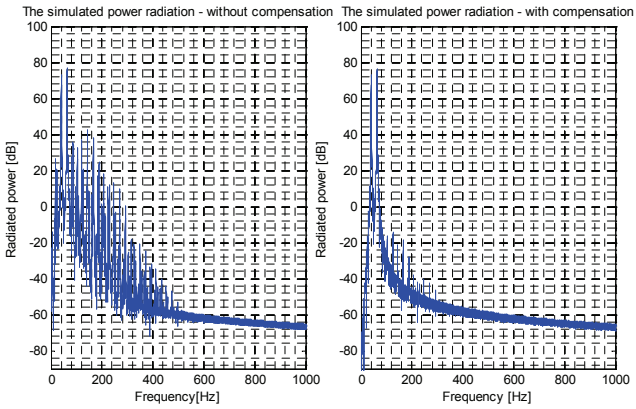
9.3.3 Simulation Result 3 – model and compensator with eddy current

The Loudspeker Model and the compensator are both simulated with eddy current. Loudspeaker model in 5.2.1, and compensator algorithm in 7.4.3.

50Hz/1W:



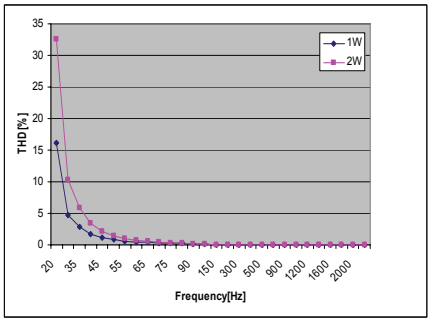
50Hz/1W + 70Hz/1W:



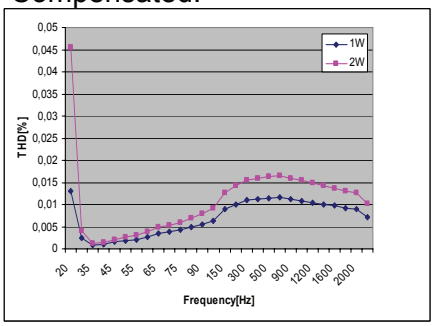
THD and IMD calculations - of the Simulated Power Radiation. (normal)

THD →

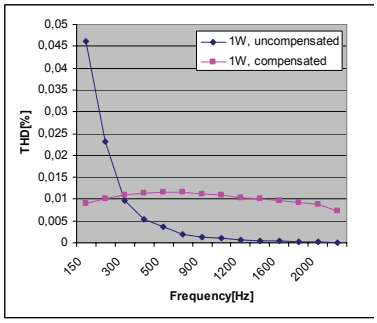
UnCompensated:



Compensated:

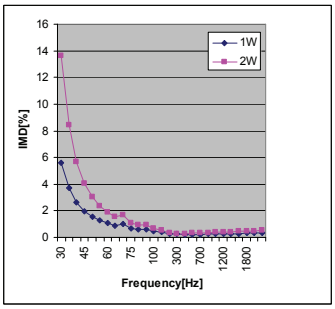


Together

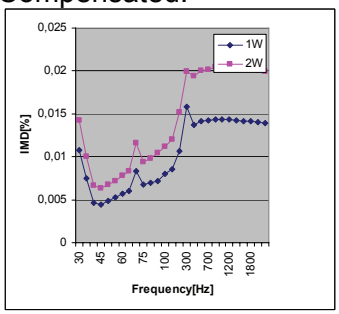


IMD →

UnCompensated:



Compensated:



9.8 Loudspeaker model and compensator are simulated with eddy current.

9.3.4 Comments of the simulation.

The originally compensator algorithm is working together with the simplified loudspeaker model, and the modified algorithm is working with the loudspeaker model with eddy currents. The

originally compensator does not work well at the loudspeaker model with eddy currents. In other words, the modification is working.

The IMD is measured with f1=63Hz and f2 starts at 20Hz, and increases first in steps on five dB. Later the step increases.

9.3.5 Instability problem with the compensation algorithm

If the amplitude of the excursion is excides about 8mm, the compensator gets unstable. This probably is caused by the polynomial modelling of the linear parameters. The polynomials diverge rapidly outside the measured range. (see figure 9.1)

10. DSP Programming

10.1 Overview of DSP functions

void linear_model **Appendix: B1**
The linear model in 3.2.6.

void nonlinear_model **Appendix: B2**
The nonlinear model in 5.2.1.

void nonlinear_model_comp **Appendix: B3**
The nonlinear compensation algorithm in 7.4.3 + The nonlinear model in 5.2.1.
Output: processed music signal u(n).

void get_linear_parameters **Appendix: B4**
The linear parameter nonlinear model in 3.2.6

void get_nonlinear_parameters **Appendix: B5**
Loudspeaker B (8480277)

void Gain **Appendix: B6**
Slow update of the Gain, or Gm in figure 12.2. (The compensator becomes instable if Gm changes to fast)

void Peak **Appendix: B7**
Peak detection

void Rms **Appendix: B8**
Rms calculation

void add_matrices_3 **Appendix: B9**
Adding two 3x1 matrixes

void mult_matrices_3 **Appendix: B10**
Multiplying two 3x1 matrixes

void add_matrices_4 **Appendix: B11**
Adding two 4x1 matrixes

void mult_matrices_4 **Appendix: B12**
Multiplying two 4x1 matrixes

10.2 Development kit/Software.

Sharc ADSP-21369 is used for the DSP-programming. The development kit is controlled by the software VisualDSP, through a USB connection to the computer. VisualDSP is used together with the software VisualAudio.

VisualAudio has a well-presented graphical interface. It is “relatively easy” to get started. A library of functions due to digital signal processing, are available as graphical blocks. These blocks are linked together graphically. The C-code of the blocks, are available and can also be modified in VisualDSP, or just be used as starting point for developing new blocks.

When operating in tuning-mode, VisualAudio offers real-time parameter-tuning while the DSP is running. A link to matlab can also be established in this modus.

Sharc ADSP-21369:

Key Features

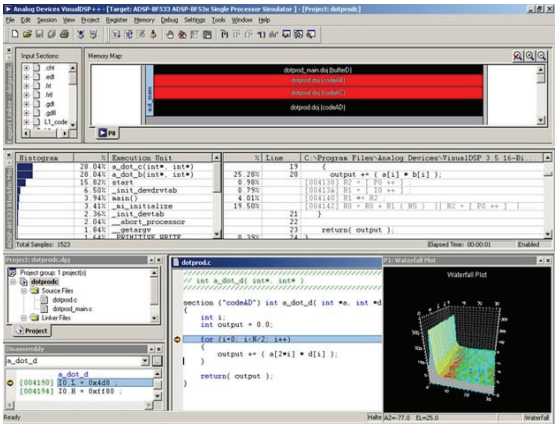
- ADSP-21369 SHARC Processor
- 1 M × 8-bit flash memory
- 1 M × 32-bit × 4 banks SDRAM
- 512 K × 8-bit SRAM
- 2 Mb SPI flash memory
- AD1835 stereo, 96 kHz, 24-bit codec
- 4 × 2 RCA jack for four channels of stereo audio output
- 1 × 2 RCA jack for one channel of stereo audio input
- Headphone jack (connected to one of stereo outputs)
- SPDIF In RCA jack
- SPDIF Out RCA jack
- ADM3202 RS-232 driver/receiver
- USB-based debugger interface
- JTAG ICE 14-pin header
- Evaluation suite of VisualDSP++ development tools
- Flash utility for downloading boot code to on-board flash memory
- Type A expansion interface with three connectors supporting external port, FLAG, DPI, and DAI interfaces
- 20-pin DPI header
- 26-pin DAI header
- Eight general-purpose LED
- Four push buttons: two general-purpose (connected to DAI), two IRQ (connected to flag pins)
- ELVIS interface
- USB cable
- 3.5 mm stereo headphones
- 6-foot RCA audio cable
- 6-foot 3.5 mm/RCA × 2 Y-cable
- CE certified

System Requirements

- Intel® Pentium® 166 MHz, or higher
- 32 MB RAM, minimum
- Microsoft® Windows® 98/2000/XP
- One available USB connector

Figure 10.1. ADSP-21369 SHARC EZ-KIT Lite Evaluation Kit.

Visual DSP



Key Features

Integrated Development and Debugging Environment

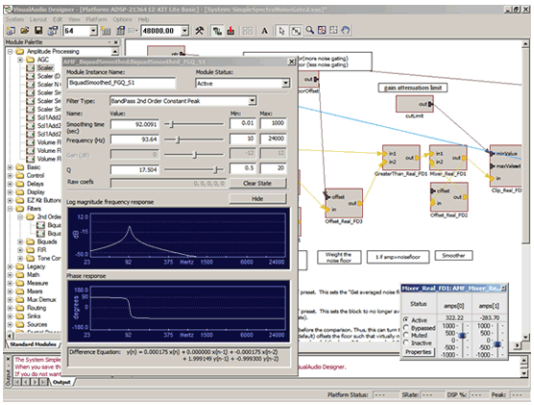
- Multiple project management
- Profiling and tracing of instruction execution
- Automation API and automation aware scripting engine
- Multiple processor (MP) support
- Background telemetry channel (BTC) support with data streaming capability
- Statistical profiling
- Graphical plotting capabilities
- Cache visualization
- Execution pipeline viewer
- Compiled simulation

Efficient Application Code Generation

- Native C/C++ compiler and enhanced assembler
- Profile-guided optimization (PGO)
- Expert linker with profiling capability
- Integrated source code control
- TCP/IP and USB support for Blackfin Processors
- Processor configuration/start-up code wizard for Blackfin Processors
- VisualDSP++ kernel (VDK) with multiprocessor messaging capability

Figure 10.2. VisualDSP – software for programming the

VisualAudio



Key features include:

- VisualAudio Designer™ graphical audio design tool
- Extensive library of optimized SHARC and Blackfin audio processing modules
- Real-time tuning interface
- Full-featured Automation API with an added MATLAB® layer
- Application-specific software platforms
- Real-time VU meters and FFT display
- Easy to integrate Layout Support Library
- Audio rate, control rate, and frequency domain signal types
- Standard 32-bit audio data types, plus high precision extensions
- Supports cross-processor SHARC and Blackfin development
- Extensible: add your own modules, decoders and platforms
- EZ-KIT Lite support: ADSP-21262, ADSP-21364, ADSP-21369, ADSP-BF533, ADSP-BF537
- Blackfin Audio EZ-Extender Board support

Figure 10.3. VisualAudio – software

For further specifications about Sharc ADSP-21369. VisulalDSP and Visual Audio can be found on: **http://www.analog.com**

10.3 The nonlinear compensation - Simulation.

10.3.1 Implementation of Tables.

To reduce the need for CPU-power on the DSP, tables for the nonlinear parameters is implemented. The values of the force-factor Bl, the inductance Le, the compliance C and the first derivative of Le (with respect to diaphragm position x), are stored in a tables during the initialisation routine. The length of the tables and the maximum excursion allowed, are made dynamical, so they later can be adjusted. The index of the tables is integers, from zero to the table length.

The implementation was first simulated in matlab.

10.3.1.1 Generation of tables:

The calculation of an 8-ordens polynomial is used to calculate the Bl, Le, C:

$$Parameter = a_0 + a_1 \cdot x + a_2 \cdot x^2 + a_3 x^3 + a_4 x^4 + a_5 x^5 + a_6 x^6 + a_7 x^7$$

And the first derivative used for calculating the derivative of Le is :

First derivative= $0 + a_1 + 2a_2 \cdot x + 3a_3 x^2 + 4a_4 x^3 + 5a_5 x^4 + 6a_6 x^5 + 7a_7 x^6$

The second derivate of Le and the first derivate of Bl and C are found with use of forward Euler, described in 3.2.1.

10.3.1.2 Linear Interpolation

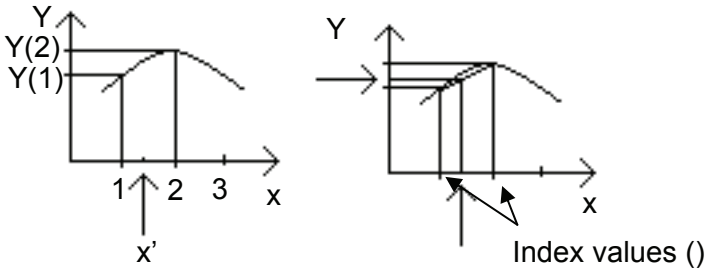


Figure 10.4 Linear interpolation

When the specific position: x' is between the to index value (integer) of the table, a strait tine is pulled between the neighbour values, Y(1) and Y(2). Y(x') is found one the line.

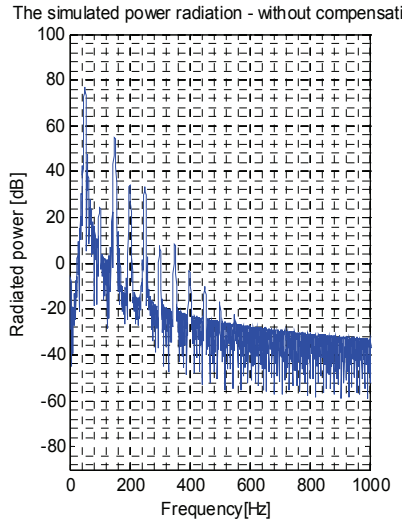
$Y(x') = Y(1) + (\left[x' - \text{int}(x')\right]) \cdot \left[Y(2) - Y(1)\right]$, if x is an integer.

In C-code "int" means the integer. This method improving the accuracy of the table implementation.

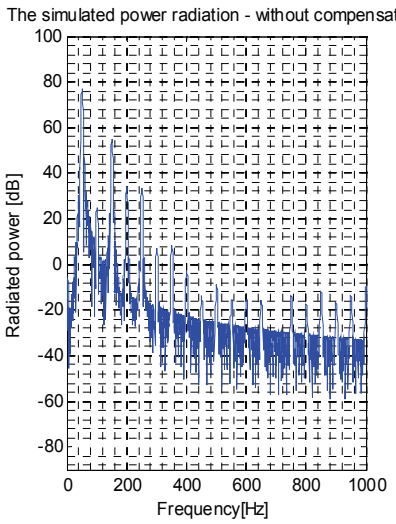
10.3.1.2 Simulating Result – Lookup table used in loudspeaker model

The maximum excursion is set to ±20mm, and different length of tables are simulated in matlab to find the required table length.

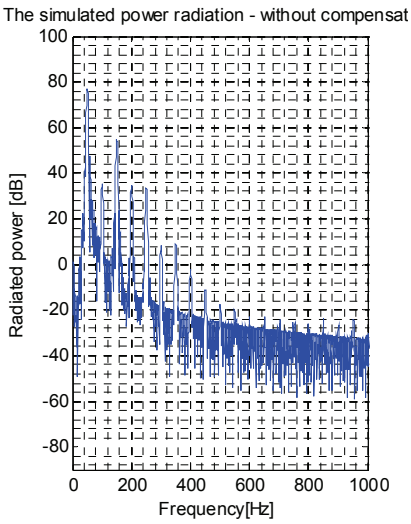
Matlab, Polynom



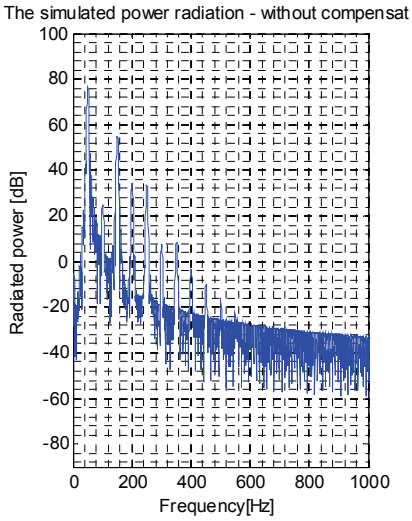
Matlab, table:3200



Matlab linear rounding, table 200



Matlab rounding, table 3200



Matlab rounding, table 32000

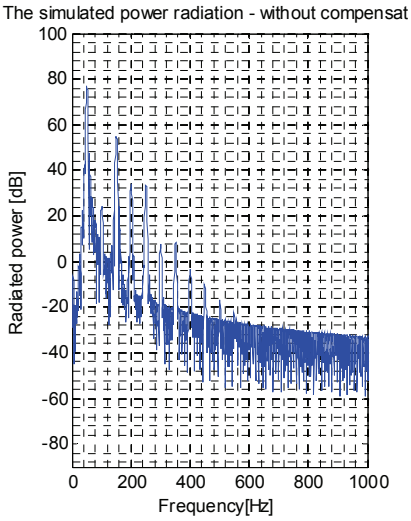


Figure 10.5 Comparing simulation of the loudspeaker model using different resolution in the tables for NonLinear Parameters

Linear Interpolation is really improving the result for in the simulation. A table-length of 3200 is chosen. No changes compared to the polynomial simulation is seen. The table length of 3200 means a solution on:

$\frac{\pm 20mm}{3200} = 12.5\mu m$ step size in the table.

10.3.1.2 Simulating Result – Lookup table used in Compensator

The nonlinear parameters of the loudspeaker model, is simulated with polynomials, while the nonlinear parameters of the compensator is simulated with use of lookup tables. This is done to imitate a real situation where the compensator is acting on a real loudspeaker.

Polynom

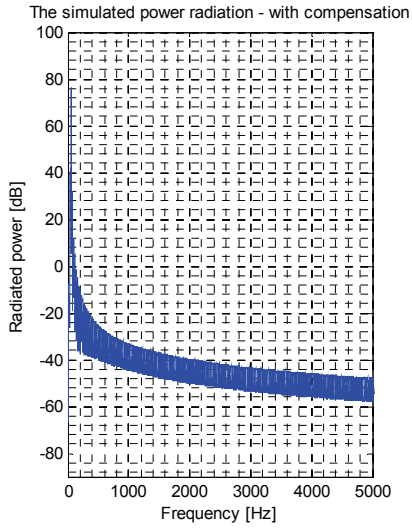


Table 3200

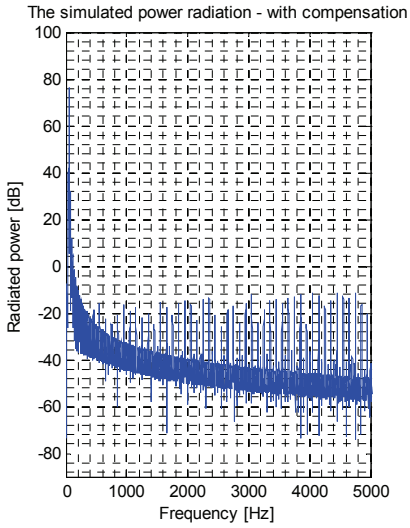


Table 12000

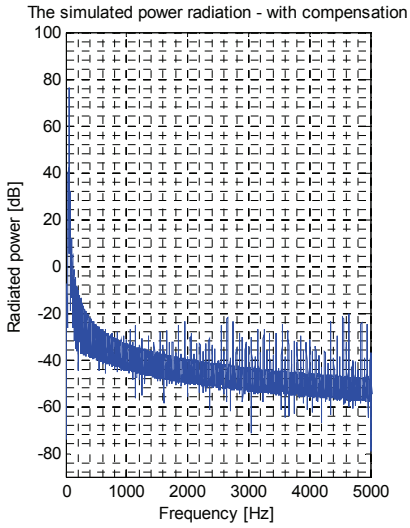


Table 3200, with rounding

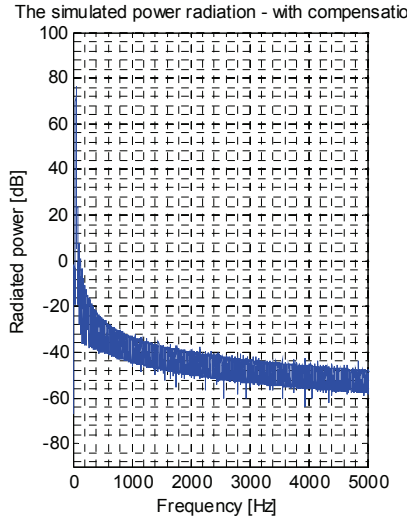


Figure 10.6 Compensation error due to table implantation. The nonlinear parameters of the loudspeaker model, is simulated with polynomials, while the nonlinear parameters of the compensator is simulated with use of lookup tables.

The compensation error due to table implantation is not critical due to the chosen table length and the use of linear interpolation.

(You may notice that the compensated output in figure 10.2 seems 100% cleared, compared to the small rest of distortion seen in the simulation of chapter 9. The reason is that the resolution and the length of frequency analysing method[7], is improved during the project. I then realized that some distortion still remained after the nonlinear compensation simulation.)

[7] Fft routine in matlab – Tool for frequency analyse in matlab.

10.4 Comparing DSP Simulation and Matlab Simulation

The C-code is running in realtime on the DSP board. Matlab are linked to Visual Audio using the function: “Capture” Visual Audio is operating set in tuning mode.

The received data from the DSP is frequency analysed in matlab.

10.4.1 Loudspeaker Model - excusion- Simulation on DSP - Compared to Matlab Simulation.

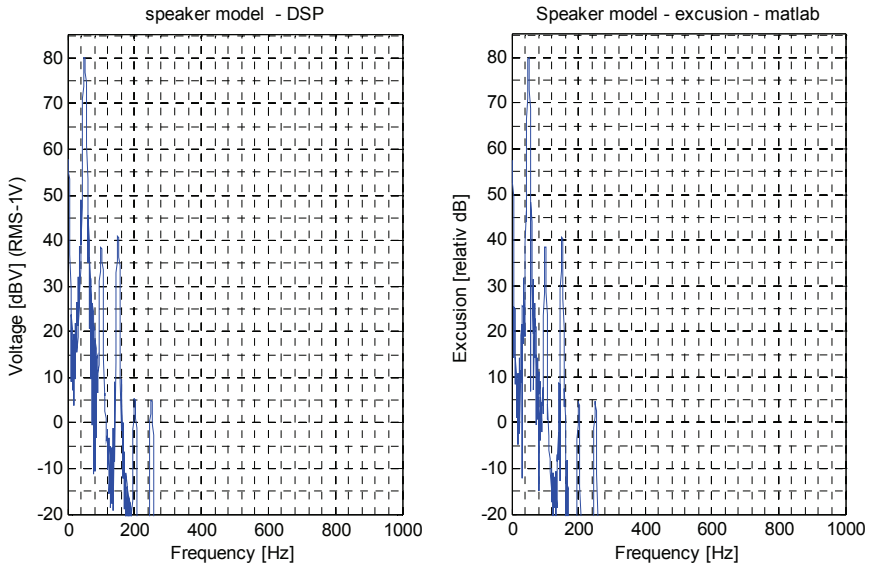
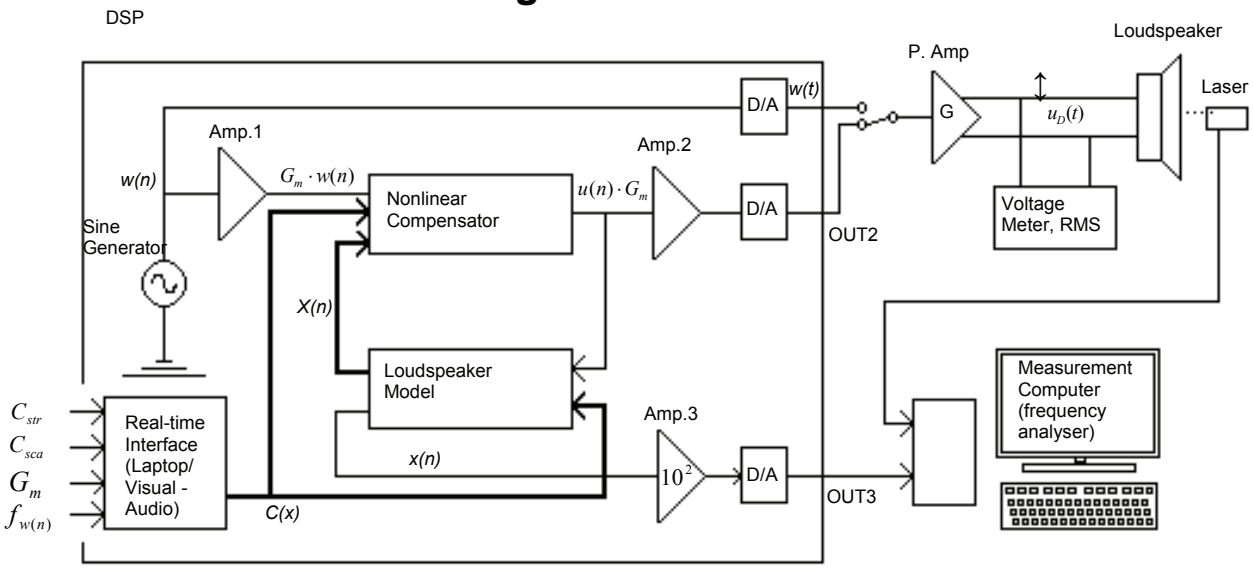


Figure 10.7. The loudspeaker model simulated on DSP, compared to matlab simulation.

The loudspeaker model simulated on DSP fits the matlab simulation. A similar test is done for the output of the compensator. The DSP and the matlab simulation is found to be equal. The document is unfortunate not to be found…

11 The final Measurement.

11.1 Schematic Drawing



11.2 Description of the Setup

Hardware

D/A	Digital to analog converter's on the DSP-board.
OUT1-OUT3	Analog Output's on the DSP-board.
P. Amp	Power amplifier. Type not specified. standard analog voltage controlled amplifier.
Laser	Laser for measuring the loudspeaker diaphragm position. Type: The Klippel Analyser's laser. The position signal is available at the X-output on the rear of the analyser.
Loudspeaker	Loudspeaker B. (mounted in the Klippel Analyser's rack)
Voltage Meter, RMS	Standard RMS voltage meter. Type not specified. Only sine waves are measured. For music signals are a "true RMS-voltage meter needed"
Measurement Computer	Measuring computer-system at B&O, Struer – in this setup only used as frequency spectrum analyser.

Software Blocks

Amp.1 – Amp.3	"Software" amplifiers.
Loudspeaker Model	Predicts vector *X(n)*. Described in 5.2.1
Music source	Sine-generator in Visual Audio.
Nonlinear Compensator	Removes the nonlinearities in the input signal *w(n)*. (Described in 7.4.3)
Real-time Interface	Real-time interface in VisualAudio *C_str*, *C_sca*, *f_w(n)*, and *G_m* are controlled on a laptop during the measurement (Described in 10.2)

Analog Signals

u(t)	Processed music signal	[V]
$u_D(t)$	Driver voltage.	[V]
$x(t)_m$	Measured excursion - laser measurement	[V]
w(t):	Music signal.(Unprocessed)	[V]

Discrete, digital signals

C(x)	Compliance – function of position, *x(n)*).	[m/N]
u (n)	Processed music signal.	[V]
w(n)	Music signal.	[V]
x(n)	Diaphragm position	[m]
X(n)	Predicted state vector (one sample into the future). Consists of current, i_{Le}, eddy current, i_{Le}, voice-coil position x (n) - and velocity, $v(n)$.	

Manually controlled Parameters (real time tuning with laptop/VisualAudio)

$f_{w(n)}$	The frequency of the sine-wave, w(n).
G_m	Internal gain. – measurement of the power amplifier gain, G.
C_{sca}	Scaling factor for the compliance curve. (Described in 9.2.1, figure 9.2)
C_{str}	Stretch factor for the x-axel for the compliance curve. (Described in 9.2.1, figure 9.3)

Others symbols

G	The gain of the power amplifier

Setup Description

An internal sine-generator in Visual Audio generates the music signal, *w(n)*. The frequency of *w(n), f_w(n)*, is available for real time tuning in VisualAudio. After the D/A converter *w(n) becomes w(t)* and is set to OUT.1 on the DSP board.

Amp.1 fits the amplitude of *w(n)* to the level where the loudspeaker is operating. *G_m* is manually set to equal the gain of the external power amplifier(*G), (the voltmeter is used)* . Amp.2 reintroduces the original level to *u(n)*. *u(n)* is D/A-converted and set to OUT.2. The level of the predicted diaphragm position, *x(n)* is multiplied by 100 (in Amp3), to get a decent signal to noise level for the measurement.

Update functionality of the linear parameters and current/voltage measurements are not implemented in the final setup. *C_str*, *C_sca*, *f_w(n)*, and *G_m* are manually controlled during the measurement. The DSP board are connected to a laptop, and the parameters are available for real time tuning, in VisualAudio's.

When the parameters are tuned in realtime, the DSP program adjusts the parameters little by little, until the wanted state is achieved. If the parameters changes too fast, the compensator algorithm gets instable.

11.3 Measurement result.

In figure 11.1 and 11.2 are the measurement results for 50Hz and 70Hz sine waves at 1W. On top are the loudspeaker outputs without compensation (excursion measurement, laser. OUT1 on the DSP board are connected to the power amp). In the middle are the predicted excursions from the loudspeaker model (measured at OUT3), and lowest are the loudspeaker outputs with compensation (excursion measurement, laser. OUT2 on the DSP board are connected to the power amp).

The measured values are relative decibel values. The absolute excursion values are not identified. The excursion signal from the loudspeaker model (OUT3, seen in figure 11.1, c&d) is simply scaled to fit the magnitude of the loudspeaker measurement, by tuning the gain of Amp.3. (the fundamentals are scaled to roughly equal).

net-power infection

The net-power infects the measurement, as seen in figure 11.1b. The 50Hz component is seen about 30 dB below the 70Hz signal. Due to intermodulation occurs also a 20Hz component. In figure 11.1d is the third harmonic of the net power detected. The magnitude of the net-power infection are small, anyway the problem would have been avoided by using another frequency than 50Hz.

11.3 Measurement result 1.

- C_{sca}=1, C_{str} = 1, *G*=4, $f_{w(n)}$ = 50&70.

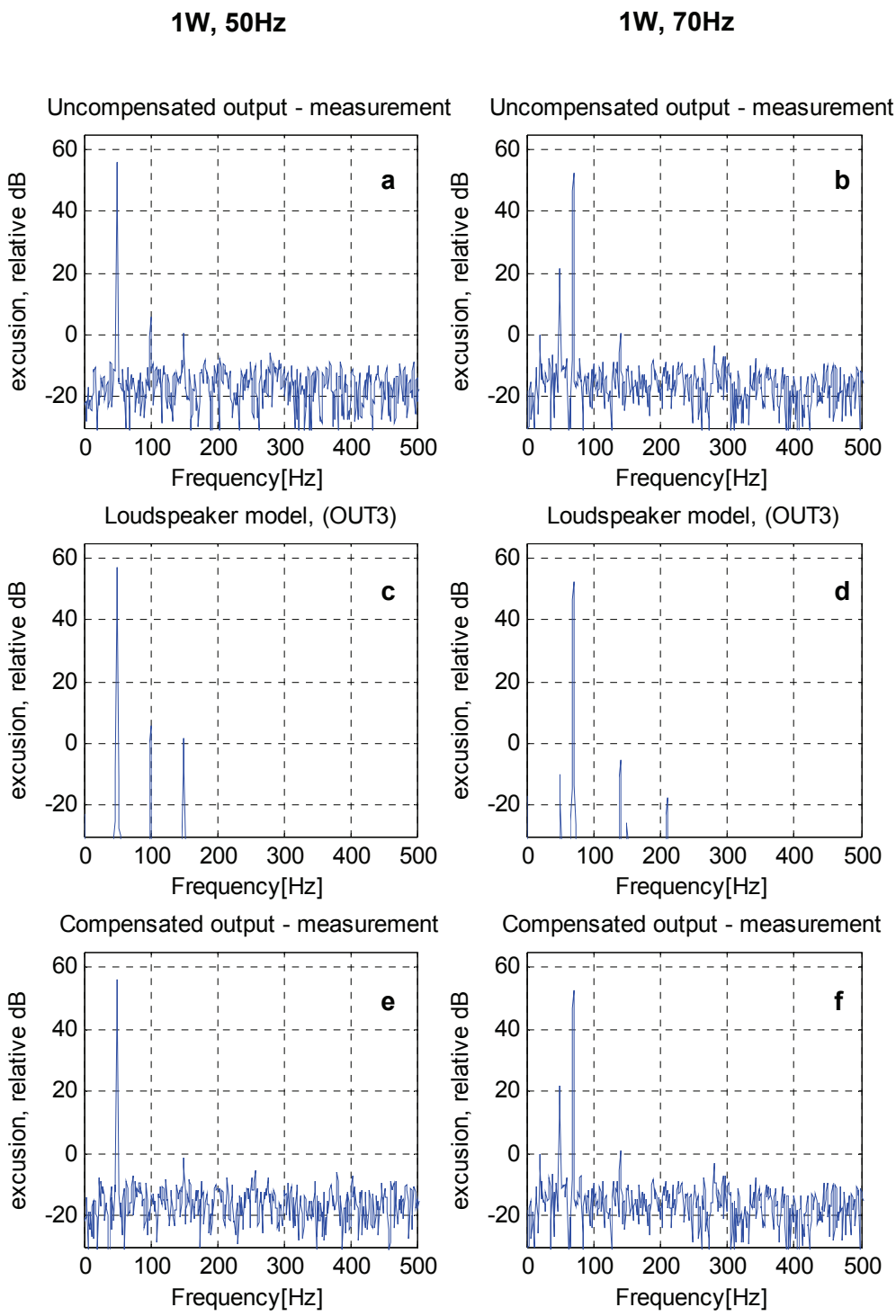


Figure 11.1. Measurement result. a,b: Loudspeaker excursion without compensation. c,d Loudspeaker model – excursion. e,f: Loudspeaker excursion with compensation.

(Data is collected from: \Measurement_result\FinalTest)

1W - 50Hz

The nonlinear compensation is somehow successful for the 50Hz input signal. In the compensated output the second harmonic (100Hz) is removed, while the third harmonic is decreased by 1.6 dB. As seen in the figure 11.1,c&d does the loudspeaker model fit the loudspeaker pretty well.

"Absolute values"

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Uncompensated output	56,2	5,9	0,7
Loudspeaker model	57	5,9	1,5
Compensated output	56,1	-	-1

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Uncompensated output	0	-50.3	-55.5
Loudspeaker model	0	-51.1	-55,5
Compensated output	0	-	-57.1

(Data is collected from: \Measurement_result\FinalTest)

1W -70Hz

For the 70Hz input signal the compensation does not work at all. The harmonics remains unchanged after the compensation. The loudspeaker model does not fit the loudspeaker. The harmonics are underestimated in the loudspeaker model.

Absolute values

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Uncompensated output	52,7	0,8	-3.3
Loudspeaker model	52,6	-5,2	-
Compensated output	52,7	0,8	-3.3

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Uncompensated output	0	-51.9	-56
Loudspeaker model	0	-57.8	-
Compensated output	0	-51.9	-56

Data is collected from: \Measurement_result\FinalTest)

11.3 Measurement result 2 – with adjustment of the compliance curve.

- C_{sca}=0.85, C_{str} = 0.92, *G*=4, $f_{w(n)}$ = 50
- C_{sca}=0.6, C_{str} = 1, G=4, $f_{w(n)}$=70.

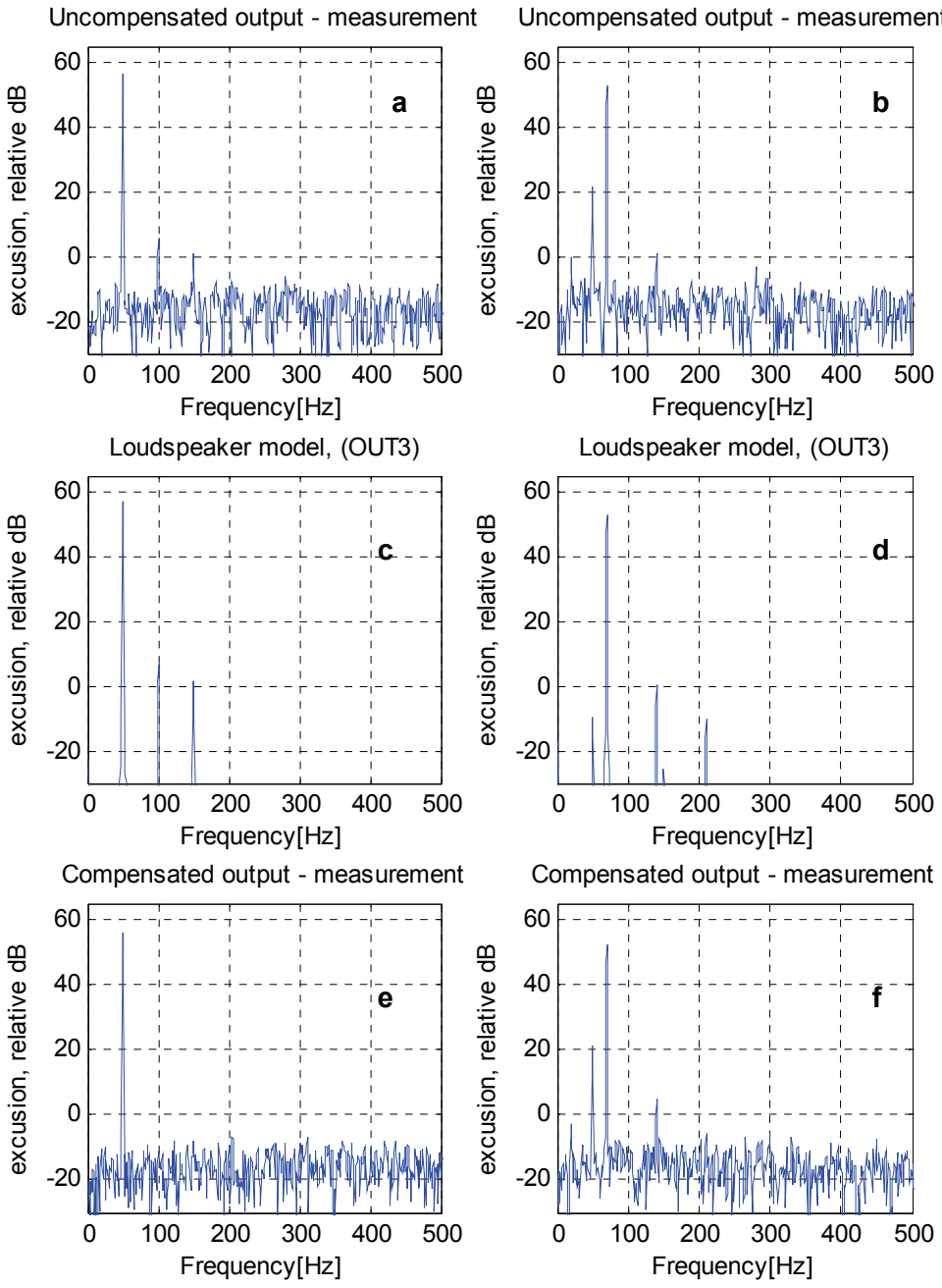


Figure 11.2. Measurement result. a,b: Loudspeaker excursion without compensation. c,d: Loudspeaker model – excursion. e,f: Loudspeaker excursion with compensation.
50Hz-measurement: the compliance are scaled by a factor of 0.85 and stretched by a factor of 0.92.
70Hz-meausrement: the compliance are scaled by a factor of 0.6 and stretched by a factor of 1.
(Data is collected from: \Measurement_result\FinalTest)

1W - 50Hz

The loudspeaker model is tuned by scaling and stretching the compliance curve. Both second and third harmonics (100Hz&150Hz), are removed for the compensated output signal.

Absolute values

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Uncompensated output	56,2	5,9	0,7
Loudspeaker model	57	6.7	1,6
Compensated output	56,1	-	-

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

50Hz, 1W	Fundamental	2. harmonic	3.harmonic
Uncompensated output	0	-50.3	-55.5
Loudspeaker model	0	-50.3	-55,4
Compensated output	0	-	-

(Data is collected from: \Measurement_result\FinalTest)

1W -70Hz

The loudspeaker model is tuned by scaling the compliance curve. The compensation is however not improved. The second harmonic actually increases.

Absolute values

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Uncompensated output	52,7	0,8	-3.3
Loudspeaker model	53,1	-0,2	-
Compensated output	52,7	4,9	-

(Data is collected from: \Measurement_result\FinalTest)

Relative values (the fundamental is scaled to 0 dB)

70Hz, 1W	Fundamental	2. harmonic	4.harmonic
Uncompensated output	0	-51.9	-56
Loudspeaker model	0	-53.3	-
Compensated output	0	-47.8	-

(Data is collected from: \Measurement_result\FinalTest)

11.4 Discussion

Result: The compensation is successful for 50Hz, 1W, not for 70Hz, 1W.

When the compensation failure is the loudspeaker model is likely to blame. The results of the simulation in 9.3.3, show that of the compensator itself is working.

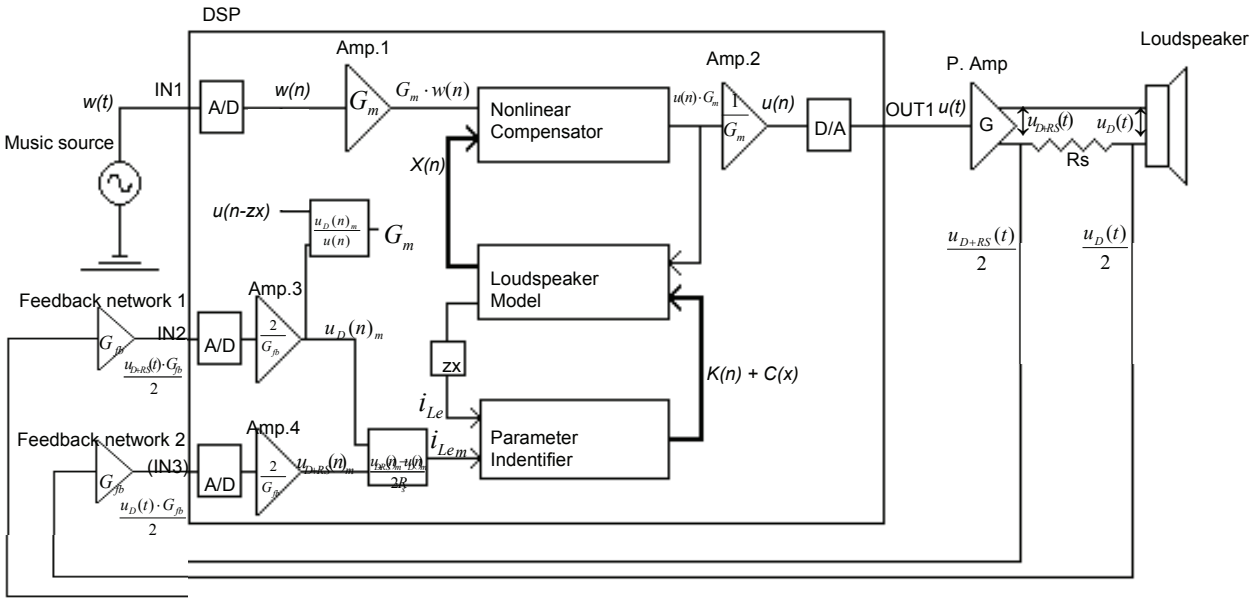
The loudspeaker resonance frequency is close to 50Hz. Above the resonance frequency will influence of the suspension gradually decrease.(2.3). Adjustment of the compliance may of that reason be more effective at 50Hz than 70Hz. The voltage and the current are also in same phase at the resonance frequency. This could make the loudspeaker simulation easier.

The loudspeaker model is tuned to fit the loudspeaker, by compliance adjustment, and the adjustment is based on frequency analyses. Apparently the tuning seems successful for both frequencies. The frequency contents of the loudspeaker model and loudspeaker are equalized. Signals with equal frequency spectrum do not necessary look the same in time domain. Tuning of the loudspeaker model by just looking into the frequency domain, may not be satisfactory.

The klippel analyser is way more complex than this simple manually suspension tuning procedure, but anyway the principle is the same. As described in 8.1.1, do the analyser use a nonlinear loudspeaker model and extracts the parameters by applying different test signals. But several setting of the nonlinear parameters can give the same output, for a specific test signal. As shown in figure 8.1.2.1, do the analyser in some cases failure. This result is measured with the old software.

12. The Future Setup

12.1 Schematic Drawing – The adaptive system



12.2 Description

Hardware

A/D	Analog to Digital converter's on the DSP-board.
D/A	Digital to analog converter's on the DSP-board.
Feedback network	Feedback network. Adapts the driver voltage from ±55V to ±1.5V.
IN1,IN2,IN3	Analog input's on DPS-board.(max voltage: ±1.5V)
Music source	CD player or similar music source.
OUT1	Analog Output on the DSP-board.
P. Amp	Power amplifier. Type not specified. standard analog voltage controlled amplifier.
Rs	Shunt resistor, due to the current measurement (high power resistor)

Software Blocks

Amp.1 – Amp.3	"Software" amplifiers.
Loudspeaker Model	Predicts vector *X(n)*. Described in 5.2.1
Nonlinear Compensator	Removes the nonlinearities in the input signal *w(n)*. (Described in 7.4.3)
Parameter Identifier	Updates linear parameters, in vector K. (Not described in detail, see 1.1)
zx	The time delay of the D/A converter and the power amplifier. (synchronizing the measured and simulated current)

Analog signals

u (t):	Processed music signal.	[V]
$u_D(t)$	Driver voltage.	[V]
$u_{D+RS}(t)$	Driver voltage + the voltage drop of Rs.	[V]
w(t):	Music signal.(Unprocessed)	[V]

Discrete, digital signals

C(x)	Compliance – function of position, *x(n)*.	[m/N]
i_{Le}	Voice-coil current - Predicted value by loudspeaker model	A]
i_{Lem}	Voice-coil current – measured value.	[A]
$K(n)$	Vector consisting the linear parameter.	
$u(n)$	Processed music signal.	[V]
$u_D(n)_m$	Driver voltage – measured value.	[V]
$u_{D+RS}(n)_m$	Driver voltage + the voltage drop of Rs - measured value.	[V]
w(n):	Music signal.	[V]
u (n):	Processed music-signal.	[V]
$X(n)$:	Predicted state vector (one sample into the future). Consists of current, i_{Le}, eddy current, i_{Le}, voice-coil position x (n) - and velocity, $v(n)$.	

Others symbols

G	The gain of the external power amplifier (P.Amp).
G_m	Internal gain. – Measurement of the power amplifier's gain, G
G_{fb}	Gain of the feedback network for driver voltage measurements. $(u_D(t))$

Description of the Setup

In the future setup is the update functionality of the linear parameters and current/voltage measurements implemented. An external source, for instance a CD-player, is generating the music-signal, *w(n)*.

Amp.1 fits the amplitude of *w(n)* to the level where the loudspeaker is operating. Amp.2 reintroduces the original level to *u(n)*. The internal gain, G_m, is automatically adjusted to equal the gain of the external power amplifier, G.

$$G_m = \frac{u_D(n)_m}{u(n)}$$

Rms values of u and u_D should be used. Gm must not be updated to fast. The compensator could get instable.

The block "Parameter Identification" updates the linear parameters and the compliance curve, based one the current measurement. The functionality is not described in this thesis.

The measured voice-coil current, i_{Lem} $i_{Le\ m} = \frac{u_{D+RS}(n)_m - u_D(n)_m}{2 \cdot R_s}$

12.3. Proposal for Voltage/current measurement.

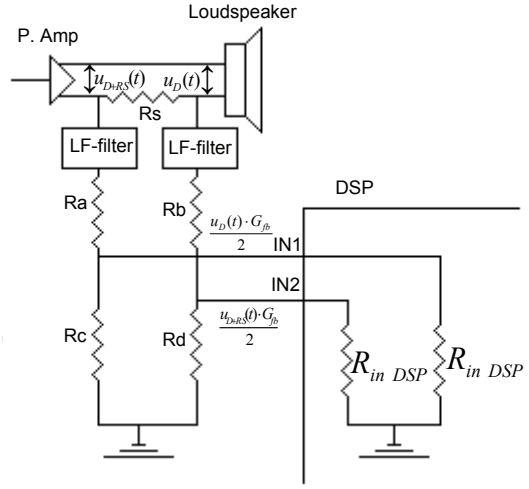


Figure 12.1. Proposal for the voltage/current measurement circuit. The symbols are described on the previous page.

B&O use digital amplifiers produced by ICE-Power, in their products. These amplifiers produce a 200kHz output voltage. The LF-filters in figure 12.1 avoids this noise to infect the circuits of the DSP board. The filter design is not considered in this thesis. (Filters for this purpose are available at B&O). LF-filtering is not necessary if a traditional analog power amplifier is used.

The maximal output voltage of the ICE-Power amplifier is 110V or ±55V. The maximal input voltage for the DSP-board is about ±1.5V. The gain of the feedback network is:

$$Gain_{feedback\ network} = \frac{U_{DSP\ input\ \max}}{U_{PowerAmp\ output\ \max}} \Rightarrow G_{fb} = \frac{1.5}{55} = \frac{1}{36.7}$$

The input impedance of the analog inputs on the DSP-board is 11kΩ. At the resonance frequency, loudspeakers impedance, Z_res, normally reaches about 50Ω. The impedance of the measuring circuits should be considerable larger than the main load, so the measuring circuit is not affects the voltage drop. Criterion for the feedback network:

1. $\frac{R_c}{R_a + R_c} = 36.7 \qquad \Rightarrow R_a \approx 36.7 \cdot R_c$

2. $R_a >> Z_{res}$ & $R_{in\ DSP} >> R_c \qquad \Rightarrow \frac{R_a}{Z_{res}} = \frac{R_{in\ DSP}}{R_c}$

Result:

$$R_c = \sqrt{\frac{R_{in\ DSP} \cdot Z_{res}}{36.7}} = 122.4 \approx \underline{\underline{120\Omega}} \quad \& \quad R_a \approx 36.7 \cdot R_c = 4492.8 \approx \underline{\underline{4.7k\Omega}}$$

$$R_a = 94 \cdot Z_{res} \quad \& \quad R_{in\ DSP} = 91.6 \cdot R_c$$

120Ω and 4.7kΩ are standard resistor values.
The circuit is not tested or simulated.

13. Conclusion.

Compensation of loudspeaker nonlinearities is investigated. A compensation system based on a loudspeaker model (a computer model/simulation of the real loudspeaker), is implemented on DSP. The system is briefly tested for pure tones, and the loudspeaker diaphragm excursion is used as output measure.

The loudspeaker parameters are measured with the Klippel Analyser, a tool for loudspeaker parameter detection. Based on these results are the loudspeaker modelled in matlab, and the result is briefly compared to the real loudspeaker. Adjustments of the compliance of the suspension, is used to improve the result.

The “feedback linearization” algorithm in [Schurer, Slump and Herrmann, 1998] is chosen as the nonlinear compensation-algorithm. It is slightly modified to fit the loudspeaker model used in this thesis. Originally it is designed for a simpler loudspeaker model, not including eddy currents. The result is satisfying. The nonlinear distortion is as good as eliminated in the simulation.

Due to temperature and aging the loudspeaker parameters are drifting. In previous investigations, like [Bright, 2002] it is found that the drifting mainly is located in the loudspeakers linear parameters. The loudspeaker model therefore needs an online tracking system (known as parameter identification), which continuously update the linear parameters. Previous investigations, and also measurements done in this thesis, show that nonlinear drifting in the compliance of the suspension also must be taken in account, at least for a traditionally hi-fi-loudspeaker. There was not time enough for looking into parameter identification.

Without the parameter identification, the compensation system is briefly tested in a simplified setup. The loudspeaker model is manually adjusted to fit the real loudspeaker. The compliance is tuned by realtime stretching and scaling functionality implemented on DSP. The system seems to work for some input frequencies and do not work for others. This is further described in 11.4.

To carry out an adaptive system, able to handle the loudspeaker drifting due to temperature and aging, is a subject for further investigation

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Appendix A Matlab Code

A1. linear_Loudspeaker_Model_Simplified

```
function [X ] = linear_Loudspeaker_Model_Simplified(u,a,b,c,M,Re,Rt,fs);

%linear modelling
%X(n)=state vector
%Bl=force factor vector
%C=total compliance vector
%M=total inductance vector
%R=Electrical resistance
%fs=sampling rate

Bl = b(1);
Le = a(1)*0.001;
C = c(1)*0.001;

Ts=1/fs;
%X=[i(n) ; x(n) ; 1/Ts*(x(];
F=[(1-((Re/Le)*Ts)) 0 (-Ts*Bl/Le);
    0 1 Ts;
    (Ts*Bl/M) (-Ts/(C*M)) (1-Rt*Ts/M)];
G=[Ts/Le ; 0 ; 0];

X=zeros(3,length(u));
X(:,1)= [ 0
          0
          0  ];

for k=1:(length(u))
    X(:,k+1) = F*X(:,k) + G*u(k);
end

end
```

A2.Loudspeaker_Model_Simplified

```
function [OUT] =
Loudspeaker_Model_Simplified(w,L2_0,R2_0,M,Re,Rt,fs,Le_tab,Bl_tab,C_tab,dLe_tab,table_size,max_x,ste
psize_table);

%linear modelling exercise
%X(n)=state vector
%Bl=force factor vector
%C=total compliance vector
%M=total inductance vector
%R=Electrical resistance
%fs=sampling rate

Le0 = Le_tab(table_size/2);
Ts=1/fs;
%X=[i(n) ; x(n) ; 1/Ts*(x(];
% Bl = b(1);
%     Le = a(1)*0.001;
%     C = c(1)*0.001;

%X=[i(n) ; x(n) ; 1/Ts*(x(];
X=zeros(3,length(w));
X(:,1)= [ 0
          0
          0  ];
str = 1;

for k=1:(length(w))
    index_realvalue = X(2,k)/stepsize_table + table_size/2;
       index = round(index_realvalue-0.5);
       index2 = round( (index-1600).*str+1600);
       dx  = index_realvalue - index;

       Bl_diff  = Bl_tab(index+1) - Bl_tab(index);
       Le_diff  = Le_tab(index+1) - Le_tab(index);
       C_diff   = C_tab(index2+1) - C_tab(index2);
       dLe_diff = dLe_tab(index+1) - dLe_tab(index);

       Bl  = Bl_tab(index)  + dx*Bl_diff;
       Le  = Le_tab(index)  + dx*Le_diff;
       C   = C_tab(index2)   + (dx*str)*C_diff ;
       dLe = dLe_tab(index) + dx*dLe_diff;

    F=[(1-((Re/Le)*Ts)) 0 -Ts/Le*(X(1,k)*dLe+Bl);
        0 1 Ts;
        (Ts*Bl/M) (-Ts/(C*M)) (1-Rt*Ts/M)];
    G=[Ts/Le ; 0 ; 0];

    X(:,k+1) = F*X(:,k) + G*w(k);
end
OUT(1,:) = X(1,:);
OUT(3,:) = X(2,:);
OUT(4,:) = X(3,:);
end
```

A3. Loudspeaker_Model

```
function [X ] =
Loudspeaker_Model(u,L2_0,R2_0,M,Re,Rt,fs,Le_tab,Bl_tab,C_tab,dLe_tab,table_size,max_x,stepsize_table
);
    %Nonlinear modelling exercise
    %X(n)=state vector
    %Bl=force factor vector
    %c=total compliance vector
    %b-total Bl vector
    %M=total mass
    %Re=Electrical resistance
    %Rt=Mechanical resistance
    %R2=electrical resistance due the eddy current losses
    %fs=sampling rate
    %unused output S1 S2 S3 S4 S5 S6 S7

    str = 1;
    Ts=1/fs;
    Le0 = Le_tab(table_size/2); % converting to Henry

    X=zeros(4,length(u));

    X(:,1)= [   0
                0
                0
                0   ];

    % simulation loop
    for k=1:64000%(length(u))
%       x2 = round(x1*round(tabel_size/20)+round(tabel_size/2))+1;%adding one since tabel starts at
1 (not zero)
%       index = round(X(3,k)/stepsize_table + table_size/2 + 0.5);
       index_realvalue = X(3,k)/stepsize_table + table_size/2;
       index = round(index_realvalue-0.5);
       index2 = round( (index-1600).*str+1600);
       dx  = index_realvalue - index;

       Bl_diff  = Bl_tab(index+1) - Bl_tab(index);
       Le_diff  = Le_tab(index+1) - Le_tab(index);
       C_diff   = C_tab(index2+1) - C_tab(index2);
       dLe_diff = dLe_tab(index+1) - dLe_tab(index);

       Bl  = Bl_tab(index)  + dx*Bl_diff;
       Le  = Le_tab(index)  + dx*Le_diff;
       C   = C_tab(index2)   + (dx*str)*C_diff ;
       dLe = dLe_tab(index) + dx*dLe_diff;

       L2 = Le*L2_0/Le0;
       R2 = Le*R2_0/Le0;
       dL2 = dLe*L2_0/Le0;

       %calculation of F and G matrix values
        F = [ 1-(Re+R2)/Le*Ts    R2/Le*Ts        0      -Ts/Le*(X(1,k)*dLe+Bl);
              R2*Ts/L2         1-(R2*Ts/L2)      0      -Ts/L2*X(2,k)*dL2;
                 0                 0             1                Ts ;
               (Ts*Bl/M)           0         (-Ts/(C*M))   (1-Rt*Ts/M)     ];

        G = [Ts/Le ; 0 ; 0; 0];

        %calculating the state variables
        X(:,k+1) = F*X(:,k) + G*u(k);

    end
end
```

A4. Nonlinear_Compensator_Simplified

```
function [OUT,u] = Nonlinear_Compensator_Simplified(w,Le_tab,Bl_tab,C_tab,
dLe_tab,L2_0,R2_0,M,Re,Rm,fs,table_size,max_x,stepsize_table);
    %Linear Feedback compensator (w,a,b,c,Le,Bl,C,L2_0,R2_0,M,Re,Rt,fs);
    %   u - compensated output voltage
    %   w - input voltage
    %   Bl_tab - force factor vector, tabel in mm, +-10mm
    %   Le_tab - inductance vector, tabel in mm, +-10mm
    %   C_tab - compliense vector, tabel in mm, +-10mm
    %   a - inductance polynomes, vector
    %   c - compliance polynomes, vector
    %   b - force factor polynomes, vector
    %   M - total mass
    %   Re - Electrical resistance
    %   Rm - Mechanical resistance
    %   R2 - electrical resistance due the eddy current losses
    %   fs - sampling rate

     Ts = 1/fs;                        % Stepsize
    Le_0 = Le_tab(table_size/2);     % inductance, converting to Henry
    Bl_0 = Bl_tab(table_size/2);     % force factor
    C_0 = C_tab(table_size/2);       % compliance, converting to m/N
    K_0 = 1/C_0;                     % stiffness N/m

    fase = 0;
    v = zeros(1,length(w));  %   LD(linear dynamics) output vector
    u = zeros(1,length(w));  %   compensator output vector
    X = zeros(3,length(w));  %   X-vector

    for k=1:fase+1

        X(:,k)= [  0           %   x1 - current
                   0           %   x3 - excution
                   0  ];       %   x4 - velosity
    end

    for k=1:64000
        %Loading nonlinear parameters from tables
        index_realvalue = X(2,k)/stepsize_table + table_size/2;
        index = round(index_realvalue-0.5);
        dx  = index_realvalue - index;

        Bl_diff  = Bl_tab(index+1) - (Bl_tab(index));
        Le_diff  = Le_tab(index+1) - (Le_tab(index));
        C_diff   = C_tab(index+1) - (C_tab(index));
        K_diff   = 1/C_tab(index+1) - 1/C_tab(index);
        dLe_diff = dLe_tab(index+1) - (dLe_tab(index));

        Bl  = Bl_tab(index)  + dx*Bl_diff;
        Le  = Le_tab(index)  + dx*Le_diff;
        C   = C_tab(index)   + dx*C_diff;
        K   = 1/C_tab(index) + dx*K_diff;
        dLe = dLe_tab(index) + dx*dLe_diff;

        dBl  = Bl_diff/stepsize_table;
        ddLe = dLe_diff/stepsize_table;
        dK   = K_diff/stepsize_table;

        %constants for speaker model
        L2 = Le*L2_0/Le_0;
        R2 = Le*R2_0/Le_0;
        dL2 = dLe*L2_0/Le_0;

        %compensator
        u(k) = Le/(Bl+dLe*X(1,k))*(X(3,k)*(X(2,k)*dK + (K-K_0) - dBl*X(1,k) ...
              - ddLe*X(1,k)^2/2 - Bl_0^2/Le_0) + Re*X(2,k)*(K-K_0)/Le_0 - ...
              Re*X(1,k)*(2*Bl+dLe*X(1,k))/(2*Le_0) + Bl_0*w(k)/Le_0) + Re*X(1,k) + ...
              Bl*X(3,k) + dLe*X(1,k)*X(3,k);% + R2*(X(1,k)-X(2,k));
%                   u(k) =w(k);
```

```
        %calculation of F and G matrix values
        F=[(1-((Re/Le)*Ts)) 0 -Ts/Le*(X(1,k)*dLe+Bl);
            0 1 Ts;
            (Ts*Bl/M) (-Ts/(C*M)) (1-Rm*Ts/M)];
            G=[Ts/Le ; 0 ; 0];
        %calculating the state variables
        X(:,k+1) = F*X(:,k) + G*u(k);


    end
    OUT(1,:) = X(1,:);
    OUT(3,:) = X(2,:);
    OUT(4,:) = X(3,:);
end
```

A5. Nonlinear_Compensator

```
function [X,u] = Nonlinear_Compensator(w,Le_tab,Bl_tab,C_tab,
dLe_tab,L2_0,R2_0,M,Re,Rm,fs,table_size,max_x,stepsize_table);
    %Linear Feedback compensator (w,a,b,c,Le,Bl,C,L2_0,R2_0,M,Re,Rt,fs);
    %   u - compensated output voltage
    %   w - input voltage
    %   Bl_tab - force factor vector, tabel in mm, +-10mm
    %   Le_tab - inductance vector, tabel in mm, +-10mm
    %   C_tab - compliense vector, tabel in mm, +-10mm
    %   a - inductance polynomes, vector
    %   c - compliance polynomes, vector
    %   b - force factor polynomes, vector
    %   M - total mass
    %   Re - Electrical resistance
    %   Rm - Mechanical resistance
    %   R2 - electrical resistance due the eddy current losses
    %   fs - sampling rate

    %Units for loudspeaker model
    %   Re[?]
    %   Rm[kg/s] / [N*s/m]
    %   mt[kg]!!
    %   Le0[H]!!
    %   Blo[N/A]
    %   Ko[N/m]

    %constants (first polynom is DC-value)
    Ts = 1/fs;                          % Stepsize
    Le_0 = Le_tab(table_size/2);        % inductance, converting to Henry
    Bl_0 = Bl_tab(table_size/2);        % force factor
    C_0 = C_tab(table_size/2);          % compliance, converting to m/N
    K_0 = 1/C_0;                        % stiffness N/m

    fase = 0;
    v = zeros(1,length(w));  %   LD(linear dynamics) output vector
    u = zeros(1,length(w));  %   compensator output vector
    X = zeros(4,length(w));  %   X-vector

    for k=1:fase+1

        X(:,k)= [  0           %   x1 - current
                   0           %   x4 - eddy current
                   0           %   x3 - excution
                   0  ];       %   x4 - velosity
    end
    %k=100
    %X(3,k)=0.0021
    %max_x = max_excursion;

    for k=1:64000
        %Loading nonlinear parameters from tables
        index_realvalue = X(3,k)/stepsize_table + table_size/2;
        index = round(index_realvalue-0.5);
        dx  = index_realvalue - index;

        Bl_diff  = Bl_tab(index+1) - (Bl_tab(index));
        Le_diff  = Le_tab(index+1) - (Le_tab(index));
        C_diff   = C_tab(index+1) - (C_tab(index));
        K_diff   = 1/C_tab(index+1) - 1/C_tab(index);
        dLe_diff = dLe_tab(index+1) - (dLe_tab(index));

        Bl  = Bl_tab(index)  + dx*Bl_diff;
        Le  = Le_tab(index)  + dx*Le_diff;
        C   = C_tab(index)   + dx*C_diff;
        K   = 1/C_tab(index) + dx*K_diff;
        dLe = dLe_tab(index) + dx*dLe_diff;

        dBl  = Bl_diff/stepsize_table;
        ddLe = dLe_diff/stepsize_table;
        dK   = K_diff/stepsize_table;
```

```
        %constants for speaker model
        L2 = Le*L2_0/Le_0;
        R2 = Le*R2_0/Le_0;
        dL2 = dLe*L2_0/Le_0;

        %compensator
        u(k) = Le/(Bl+dLe*X(1,k))*(X(4,k)*(X(3,k)*dK + (K-K_0) - dBl*X(1,k) ...
              - ddLe*X(1,k)^2/2 - Bl_0^2/Le_0) + Re*X(3,k)*(K-K_0)/Le_0 - ...
              Re*X(1,k)*(2*Bl+dLe*X(1,k))/(2*Le_0) + Bl_0*w(k)/Le_0) + Re*X(1,k) + ...
              Bl*X(4,k) + dLe*X(1,k)*X(4,k);% + R2*(X(1,k)-X(2,k));
        %calculation of F and G matrix values
        F = [ 1-(Re+R2)/Le*Ts    R2/Le*Ts        0      -Ts/Le*(X(1,k)*dLe+Bl);
              R2*Ts/L2         1-(R2*Ts/L2)     0      -Ts/L2*X(2,k)*dL2;
                 0                 0             1                Ts ;
               (Ts*Bl/M)           0         (-Ts/(C*M))   (1-Rm*Ts/M)     ];

        G = [Ts/Le ; 0 ; 0; 0];

        %calculating the state variables
        X(:,k+1+fase) = F*X(:,k) + G*u(k);
    end
end
```

A6 compliance_Adjustment

```
function [X,str]= compliance_Adjustment(C,stepsize_table,table_size);

    x  = -0.020:stepsize_table:0.020;
    X = 0.002;
    %scaling
    sca = 1; %Scaling-Gain
    C_tuned = C * sca;

    %stretching
    str = 0.8;  %Stretching-Gain

    %smuding, model becomes unstabil if
    smud_factor = 10;
    C_mid = (smud_factor*C + C_tuned)/(smud_factor+1);

    %index rutine
    index_realvalue = x/stepsize_table + table_size/2;
    index = round(index_realvalue-0.5);

    index2 = round((index-1600).*str+1600);

    figure(50)
        plot(index,C)
        axis([0 3200 0 0.0003])%max(C)*1.05])%0.0025])
        title('Orginal, Cms')
        grid
    figure(51)
        plot(index2,C_tuned)
        axis([0 3200 0 0.0003])%max(C)*1.05])%0.0025])
        title('tuned, Cms')
        grid
    figure(52)
        plot(index,C_mid)
        axis([0 3200 0 0.002])%max(C)*1.05])%0.0025])
        title('smud, Cms')
        grid
     X = C_tuned;
end
```

A7 THD

```
function [OUT, d, ind]=THD(Y,f0,fs,L,n,nf);
    % Y = spectrum(up to fs/2!), f0 = fundamental frequency, fs = sampling frequency.
    % L = size of spectrum vector,
    % n - the maximum order of distortion included
    % nf - % number of neighbour frequencies.

    %finding index for frequencies in spectrum
    for m=1:n+1;
        kind(m) = round(m*f0/(fs/(2*L)))+1;
    end
    ind = kind;
    Y1 = 0;
    Y0 = 0;
    %d = zeros(1,n+1);%[0 0 0 0 0 0]';
    for i=-nf : nf
        Y1 = Y1 + (sum((abs(Y(kind(2:end)+i))).^2));    %summing the energy in harmonics
        Y0 = Y0 + (sum((abs(Y(kind+i))).^2));           %summing the total energy
    end
    d = 10*log10(abs(Y(kind)));                         %freguency components energy
    OUT = 100*sqrt(Y1/Y0);
end
```

A8 IMD

```
function [OUT] = IMD(Y,f1,f2,fs,L,n,nf);
    % Y = spectrum, f1=the fundamental frequency, f2=the intermodulation frequency.
    % fs = sampling frequency, L= size of spectrum vector.
    % n = the maximum order of distortion included.
    % nf - number of neighbour frequencies.
    % k - index for all frequency components. intermodulation frequency +
    %     distortion freq. comp.
    % k0 - index for intermodulation frequency.

    % removing distortion frequency components less than zero.
    n_neg = -n;
    if n>round((f2-20)/f1-0.5)
        n_neg = -round((f2-20)/f1-0.5);
    end

    for m=n_neg:n                                          % finding index for distortion frequencies
        k(m-n_neg+1) = round((f2+m*f1)/(fs/(2*L)));        % + intermodulation frequency.
    end

    k0 = round((f2)/(fs/(2*L)));                    % finding index for intermodlation frequency.
    Y1 = 0;
    Y0 = 0;
    for i=-nf : nf
        Y1 = Y1 + (sum((abs(Y(k+i))).^2))-(abs(Y(k0+i))).^2;    % summing the energy in distortion.
        Y0 = Y0+ (sum((abs(Y(k+i))).^2));                      % summing the total energy.
    end
    OUT = 100*sqrt(Y1/Y0);
end
```

A9 Load_Nonlinear_Parameters

```
function [a,b,c,k,Le,Bl,C,dLe] = Load_Nonlinear_Parameters(speaker, a_ant, b_ant,
c_ant,tabel_size,max_x,table_stepsize);

    if speaker == 8480255
        a = [7.43101e-004 -7.82537e-002 0.205835 2030.62 54924.4 ...
            -6.57724e+007 -3.78117e+009 8.66467e+011 6.87626e+013];
        b = [7.37708 71.1914 -141077 -2.02181e+006 -1.53254e+009 ...
            1.21793e+010 9.21392e+013 7.81478e+013 -9.90756e+017];
        c = [1.65025e-003 0.138596 -64.0621 -11006.6 1.38078e+006 ...
            3.59878e+008 -8.77243e+009 -4.34949e+012 -1.14947e+014];

     elseif speaker == 8480277
        a = [4.42852e-004 -3.99246e-003 -0.730067 11.3856 2426.82 ...
            -8064.92 -3.80829e+007 1.0525e+008 3.24411e+011];
        b = [7.19845 -22.7689 -10185.5 3.70864e+006 -1.65104e+009  ...
            -5.0827e+010 2.04298e+013 1.40152e+014 -7.96303e+016];
        c = [1.12474e-003 1.91161e-002 -33.3252 -1199.15 636426 ...
            2.74833e+007 -7.21205e+009 -2.23333e+011 3.37872e+013];
        k = [1/c(1) -1000/(c(2)) 1/c(3) 1/c(4) 1/c(5) 1/c(6) 1/c(7) 1/c(8)];

    elseif speaker == 8480285
        a = [4.98714e-004 -6.53664e-002 1.10054 1739.08 10564 ...
            -4.51131e+007 -1.60088e+009 4.59639e+011 2.64098e+013];
        b = [6.69958 -131.573 -67754 -5.02744e+006 -2.97825e+009 ...
            2.80079e+011 8.79196e+013 -3.1649e+015 -7.36492e+017];
        c = [2.2468e-003 4.61464e-002 -77.8565 -3650.55 1.90038e+006 ...
            9.69742e+007 -2.72693e+010 -8.98781e+011 1.58282e+014];

    elseif speaker == 31221
        a = [0.58157 -0.031046 -0.0066093 0.0011830 0.00042152 -0.00015811 ...
            -6.4289e-5 1.1929e-5 4.5528e-6];%Le polynomial coefficients
        b = [7.03 -0.023848 -0.055244 0.0099365 -0.0042554 -0.00014267 ...
            9.478e-5 -7.2017e-6 2.3069e-7]; %Bl polynomial coefficients
        c = [0.241 0.0045478 -0.0050385 0.00022905 0.00012458 ...
            -0.10724e-4 -2.6118e-6 2.8368e-7 4.6371e-8];

    else
        message = ['Speaker "' num2str(speaker) '" is not presented in the database.']
        uiwait(msgbox(message));
    end
    %Removing unwanted coeffisients
    for i=0: 7-a_ant
       a(9-i) = 0;  %Le
    end

    for i=0: 7-b_ant
       b(9-i) = 0;      %Bl
    end

    for i=0: 7-c_ant
       c(9-i) = 0;      %C
    end
    for i=0: 7-c_ant
       k(9-i) = 0;      %C
    end

    x = -max_x:table_stepsize:max_x;
    Bl = polyval(fliplr(b),x);
    C = polyval(fliplr(c),x);
    Le = polyval(fliplr(a),x);
    dLe = polyval(fliplr(a(2:9).*[1 2 3 4 5 6 7 8]),x);
end
```

A10 Load_Linear_Parameter

```
function [Bl_lin,C_lin,Le_lin,M,Re,Rt,L2_0,R2_0] = Load_Linear_Parameter(speaker);

    if speaker == 8480255
        Bl_lin = 0;      Le_lin = 0;    C_lin = 0;
        Re = 12.6263;   Rt = 1.16119;   M = 6.15617e-003;
        R2_0 = 5.65685;     L2_0 = 7.68891e-004;

    elseif speaker == 8480277
        Bl_lin = 7.18583;      Le_lin = 0.000389793;    C_lin = 0.000538315;
        Re = 5.02882;   Rt = 2.16858;   M = 0.0195471;
        R2_0 = 2.67697;     L2_0 = 0.000565304;

    elseif speaker == 8480285
        Bl_lin = 0;      Le_lin = 0;    C_lin = 0;
        Re = 12.8636;   Rt = 0.589303;   M = 8.82478e-003;
        R2_0 = 3.36085;     L2_0 = 5.6789e-004;

    elseif speaker == 31221
        Bl_lin = 7.03;      Le_lin = 0.58157e-3;    C_lin = 0.241e-3;
        M = 9.4e-3;         Re = 5.85;              Rt = 6.812;
        L2_0 = 0.550e-3;    R2_0 = 6.4;%eddy current parameters%L2_0 = 0.1e-3;      R2_0 = 1.0;

    else
        message = ['Speaker "' num2str(speaker) '" is not presented in the database.']
        uiwait(msgbox(message));

    end
end
```

A11 plot_Nonlinearparameter

```
function plot_Nonlinearparameter(a,b,c,k,fs);
stepsize = 0.000001;
x = -0.020:stepsize:0.020;

Bl = polyval(fliplr(b),x);
C = polyval(fliplr(c),x);
Le = (polyval(fliplr(a),x));
K_p = polyval(fliplr(k),x);
K = zeros(1,length(C));
K = 1./C;

dLe = polyval(fliplr(a(2:9).*[1 2 3 4 5 6 7 8]),x);
dBl = polyval(fliplr(b(2:9).*[1 2 3 4 5 6 7 8]),x);
dC = polyval(fliplr(c(2:9).*[1 2 3 4 5 6 7 8]),x);
ddLe = polyval(fliplr(a(3:9).*[2 6 12 20 30 42 56]),x);

dBl2 = zeros(1,length(Bl));
dC2 = zeros(1,length(C));
dK = zeros(1,length(C));
dLe2 = zeros(1,length(Le));

for k=1:40000
    dBl2(k) = (Bl(k+1)-Bl(k))/stepsize;
    dC2(k) = (C(k+1)-C(k))/stepsize;
    dLe2(k) = (Le(k+1)-Le(k))/stepsize;
    dK(k) = (K(k+1)-K(k))/stepsize;
end

figure(40)
subplot(4,3,1)
    plot(x,Bl)
    axis([-0.030 0.03 -8 8])%max(Bl)*1.05])
    title('Force-factor,Bl')
    grid
subplot(4,3,2)
    plot(x,C)
    axis([-0.030 0.03 -2*0.0012 2*0.0012])%max(C)*1.05])%0.0025])
    title('Compliance, C')
    grid
subplot(4,3,3)
    plot(x,Le)
    axis([-0.030 0.03 -2*4.5558e-004 2*4.5558e-004])%max(Le(12000:20000))*1.05])
    title('Inductance, Le')
    grid
subplot(4,3,4)
    plot(x,dBl)
    axis([-0.020 0.02 -8e3 8e3])%max(Bl)*1.05])
    title('First derivate of force-factor dBl/dx')
    grid
subplot(4,3,5)
    plot(x,dC)
    axis([-0.020 0.02 -0.0009e3 0.0009e3])%max(C)*1.05])%0.0025])
    title('d_Cms')
    grid
subplot(4,3,6)
```

```
    plot(x,dLe)
    axis([-0.020 0.02 -6.1558e-002 6.1558e-002])%max(Le(12000:20000))*1.05])
    title('d_Le')
    grid
subplot(4,3,7)
    plot(x,dBl2)
    axis([-0.020 0.02 -8e3 8e3])%max(Bl)*1.05])
    title('d_Bl2')
    grid
subplot(4,3,8)
    plot(x,dC2)
    axis([-0.020 0.02 -0.0009e3 0.0009e3])%max(C)*1.05])%0.0025])
    title('d_C2')
    grid
subplot(4,3,9)
    plot(x,dLe2)
    axis([-0.020 0.02 -6.1558e-002 6.1558e-002])%max(Le(12000:20000))*1.05])
    title('d_Le2')
    grid
subplot(4,3,10)
    plot(x,K)
    axis([-0.020 0.02 0 6e3])%max(Bl)*1.05])
    title('K')
    grid
subplot(4,3,11)
    plot(x,K_p)
    axis([-0.020 0.02 0 6e3])%max(Bl)*1.05])
    title('K_p')
    grid
subplot(4,3,12)
    plot(x,ddLe)
    axis([-0.020 0.02 -7 7])%max(Bl)*1.05])
    title('ddLe')
    grid



end
```

A12 Capture

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%                                                                               %
%                        Bang & Olufsen A/S                                     %
%                                                                               %
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%
% Project name  : DIG-HT
%
% Module title  : Capture.c
%
% Module name   :
%
% Module type   :
%
% Description   :   Matlab interface for VA-Capture function
%
% Created       :   LYL 14-09-2006
%
% Responsible   :   LYL
%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%
% Changes history:
%
% Identifier    Date     Made by    Change
%-----------------------------------------------------------------
%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function [dataBuffer, numIn] = Capture(blockName);
path(path, 'C:\Program Files\Analog Devices\VisualAudio 2.5\Matlab');
path(path, 'C:\Program Files\Analog Devices\VisualAudio 2.5\MATLAB\mex');
path(path, 'C:\Program Files\Analog Devices\VisualAudio 2.5\MATLAB\ModuleDesigners');
path(path, 'C:\Program Files\Analog Devices\VisualAudio 2.5\Designer');


captureBlock = va_module(blockName,0);
captureBlock.matlabReading = 1;
SelectBuffer = captureBlock.bufReadReady;
if SelectBuffer == 0
    tmpBuffer = captureBlock.valueBuffer0;
else
    tmpBuffer = captureBlock.valueBuffer1;
end
captureBlock.matlabReading = 0;


numIn = length(captureBlock.valueBuffer1)/captureBlock.bufferSize;
bufferSize = captureBlock.bufferSize;
for i = 0:numIn-1
    dataBuffer(i+1,:) = tmpBuffer(1+i*bufferSize:bufferSize*(i+1));
end
```

A13 Main

```
%Model of nonlinear loudspeaker
%Using linear parameters from
%Using nonlinear parameters from
%X(n) - state vector
%Bl - force factor vector
%c - total compliance vector
%b - total Bl vector
%M - total mass
%Re - Electrical resistance
%Rt - Mechanical resistance
%R2 - electrical resistance due the eddy current losses
%fs - sampling rate

%Selecting Loudspeaker type
speaker = 8480277;

%sampling frequency
%   -for inputsignal w(n), loudspeakermodel and nonlinear compensator
fs = 48000;

%Table's for nonlinear parameters - properties.
table_size = 3200;     % tabel-size for nonlinear parameters
max_excursion = 0.02;   % [m]
stepsize_table = (2*max_excursion)/table_size;

%Loading Linear Parameters (Bl_lin,C_lin,Le_lin is not used later)
[Bl_lin,C_lin,Le_lin,M,Re,Rt,L2_0,R2_0] = Get_Linear_Parameters(speaker);

%Loading Nonlinear Parameters (Polynomial coefficients and tabels)
[a,b,c,K,Le,Bl,C,dLe] =
Get_Nonlinear_Parameters(speaker,8,8,8,table_size,max_excursion,stepsize_table);%Le,Bl,C
[C] = compliance_tuning(C,stepsize_table, table_size);

%Input signal, frequencies(Hz) and Input voltages, amplitude(V)(convert from rms to amplitude)
f = [50];% 25 30 35 40 45 50 55 60 65 70 75 80 90 100 150 200 300 400 500 700 900 1000 1200 1400
1600 1800 2000 3000];
fm = [63];
A = 4;  Am = 0*5.6569;  %5.6569
n = 1:96000;
result = zeros(6,length(fm));
k=1;
for j=1:length(fm)
    %Generates input signal (driver voltage - u_D)
    w = A*sin(2*pi*f(k)*n/fs) + Am*sin(2*pi*fm(j)*n/fs);

    %Loadspeaker Model
    %[X] =
Loudspeaker_Model(w,L2_0,R2_0,M,Re,Rt,fs,Le,Bl,C,dLe,table_size,max_excursion,stepsize_table);
    [X] =
Loudspeaker_Model_Simplified(w,L2_0,R2_0,M,Re,Rt,fs,Le,Bl,C,dLe,table_size,max_excursion,stepsize_ta
ble);
    %[X] =
Loudspeaker_Model_Simplified_Polynom(w,a,b,c,L2_0,R2_0,M,Re,Rt,fs,Le,Bl,C,dLe,table_size,max_excursi
on,stepsize_table);

    %Nonlinear Compensator
    %[X_comp,u] =
Nonlinear_Compensator(w,Le,Bl,C,dLe,L2_0,R2_0,M,Re,Rt,fs,table_size,max_excursion,stepsize_table);
    [X_comp,u] =
Nonlinear_Compensator_Simplified(w,Le,Bl,C,dLe,L2_0,R2_0,M,Re,Rt,fs,table_size,max_excursion,stepsiz
e_table);
    %[X_comp,u] =
Nonlinear_Compensator_Simplified_Polynom(w,a,b,c,Le,Bl,C,dLe,L2_0,R2_0,M,Re,Rt,fs,table_size,max_exc
ursion,stepsize_table);

    %Plot timedomain
%     figure(3);
%         plot([u(1:17001)' w(1:17001)'])%
%         axis([4000 10000 -10 10]);   grid;
%         %title('blue-compensated driver voltage, u_D(n), green-uncompensated driver voltage w(n)')
```

```
%         ylabel('Voltage[V]')
%         xlabel('Discrete time')
    %Frequency analyse
    Nf = 2^17;
    nx = 1:(Nf/2);
    y = fft(X(3,7001:64000).*hann(57000)',Nf);
    Pyy = (y.*conj(y))/Nf;
    Prad = (((Pyy(1:Nf/2))).*((nx*fs/Nf*2*pi).^4)); %*1/2*1.2*(0.0525^2*pi)^2/(2*pi*344)
    nx_comp = 1:15000/2;
    y_comp = fft(X_comp(3,7001:64000).*hann(57000)',Nf);
    Pyy_comp = (y_comp.*conj(y_comp))/Nf;
    Prad_comp = (((Pyy_comp(1:Nf/2))).*((nx*fs/Nf*2*pi).^4)); %*1/2*1.2*(0.0525^2*pi)^2/(2*pi*344)

    Y_prespeaker = fft(X(3,7001:64000).*hann(57000)',Nf);
    Y_prespeaker_voltage = abs ( Y_prespeaker(1:Nf/2));%5.287e3

    figure(5);
        plot(nx_comp*fs/Nf,20*log10(Y_prespeaker_voltage(1:15000/2))+31.3);
        axis([0 1000 -30 80]);    grid minor;
        title('The simulated power radiation - without compensation')
        ylabel('Radiated power [dB]')
        xlabel('Frequency[Hz]')
    %Frequency  plot
    figure(6);
        plot(nx_comp*fs/Nf,10*log10(Prad(1:15000/2)));
        axis([0 1000 -90 100]);    grid minor;
        title('The simulated power radiation - without compensation')
        ylabel('Radiated power [dB]')
        xlabel('Frequency[Hz]')
    figure(11);
        plot(nx_comp*fs/Nf,10*log10(Prad_comp(1:15000/2)));
        axis([0 1000 -90 100]);    grid minor;
        title('The simulated power radiation - with compensation')
        ylabel('Radiated power [dB]')
        xlabel('Frequency [Hz]')
```

Appendix B - C-Code

B1 linear_model

```
void linear_model(float *u, float *out, float Bl_lin, float Le_lin, float C_lin, float M, float Re, float Rt, unsigned short int fs, int tickSize, float
*X_overlap)
{
   float F[3][3];
   float G[3] = {0, 0, 0};
   float X[3][251];  //current, excursion, velosity
   float TEMP_X[3] = {0, 0, 0};//
   float TEMP_1[3] = {0, 0, 0};//
                  float TEMP_2[3] = {0, 0, 0};//
   float Ts = 1.0/fs;
                  int k, k2;

                  //fills zeros in X vektor - not necessary
                  for(k=0; k<3; k++)
                  {
                                 for(k2=0; k2<251; k2++)
                                 {
                                                X[k][k2] = 0;
                                 }
                  }

                  //calculates F and G matrices
                  F[0][0] = 1 -((Re/Le_lin)*Ts),          F[0][1] = 0,                                                                                F[0][2]
= -Ts*Bl_lin/Le_lin;
   F[1][0] = 0,                                                                                                  F[1][1] = 1,
                                 F[1][2] = Ts;
   F[2][0] = Ts*Bl_lin/M,                                                  F[2][1] = -Ts/(C_lin*M),          F[2][2] = 1-(Rt*Ts/M);

   G[0] = Ts/Le_lin,                                                                                             G[1] = 0,
                                                G[2] = 0;

   //Importing last predicted output from previous tick-block
   TEMP_X[0] = X_overlap[0];    TEMP_X[1] = X_overlap[1];          TEMP_X[2] = X_overlap[2];

   //calculates next value for output vector
   for(k=0; k<32; k++)//tickSize8
                  {

                  mult_matrices_3(F, TEMP_X, TEMP_1);
                  TEMP_2[0] = G[0]*u[k]; TEMP_2[1] = u[k]*G[1]; TEMP_2[2] = u[k]*G[2];
                  add_matrices_3(TEMP_1, TEMP_2, TEMP_X);
                  X[0][k] = TEMP_X[0];   X[1][k] = TEMP_X[1];  X[2][k] = TEMP_X[2];
                  out[k] = X[1][k];
    }

   //exporting last predicted output for next tick-block
   X_overlap[0] = TEMP_X[0];          X_overlap[1] = TEMP_X[1];          X_overlap[2] = TEMP_X[2];

}
```

B2 nonlinear_model

```
void nonlinear_model(float *u, float *out, float *Bl_tabel, float *C_tabel, float *Le_tabel, float *dLe_tabel, float R2_0, float L2_0, float M,
float Re, float Rt, unsigned short int fs, int tickSize, float *X_overlap_nonlinear, short int table_size, float max_excursion,float
stepsize_table)
{
   float F[4][4];                                                                                                    //matrice in state space
model
   float G[4] = {0, 0, 0, 0};                                                    //matrice in state space model
   float X[4][32];                                                                                                   //current, eddycurrent,
excursion, velosity
   float TEMP_X[4] = {0, 0, 0, 0};                          //current, eddycurrent, excursion, velosity
   float TEMP_1[4] = {0, 0, 0, 0};                          //current, eddycurrent, excursion, velosity
                  float TEMP_2[4] = {0, 0, 0, 0};                                //current, eddycurrent, excursion, velosity
   float Ts = 1.0/fs;
                  int k, k2, index, index2;
                  float Bl, C, Le, dLe, Le0, L2, R2, dL2, factor;
                  float dx, Bl_diff, Le_diff, C_diff, K_diff, dLe_diff, index_realvalue, K;

                  //Loading linear parameters
                  Le0 = Le_tabel[1600];                                                                  //  [H]

                  //fills zeros in X vektor - not necessary
                  for(k=0; k<4; k++)
                  {
                                    for(k2=0; k2<32; k2++)
                                    {
                                                      X[k][k2] = 0;
                                    }
                  }
                  factor = 0.88;

                  //Importing last predicted output from previous tick-block
   TEMP_X[0] = X_overlap_nonlinear[0];          TEMP_X[1] = X_overlap_nonlinear[1];
   TEMP_X[2] = X_overlap_nonlinear[2];                                TEMP_X[3] = X_overlap_nonlinear[3];

                  for(k=0; k<tickSize; k++)
                  {

                     index_realvalue = TEMP_X[2]/stepsize_table + table_size/2;
                     index = index_realvalue;
                     dx = index_realvalue - index;

                     index2 = ((index-1600)*factor +1600);

                     if(k==3)
                                    {
                                                      debugbuffer[0] = index_realvalue;
                                                      debugbuffer[1] = index;
                                                      debugbuffer[2] = dx;

                                    }
                     //Calculating diff
                     Bl_diff  = Bl_tabel[index+1] - Bl_tabel[index];
                     Le_diff  = Le_tabel[index+1] - Le_tabel[index];
                     C_diff               = C_tabel[index2+1]  - C_tabel[index2];
                     K_diff   = 1/C_tabel[index2+1] - 1/C_tabel[index2];
                     dLe_diff = dLe_tabel[index+1] - dLe_tabel[index];

                     //Loading nonlinear parameters from tables
                     Bl  = Bl_tabel[index] + dx*Bl_diff;                  // [N/A]
                     Le  = Le_tabel[index] + dx*Le_diff;                  // [H]
                     C   = C_tabel[index2] + (dx*factor)*C_diff;                                    // [m/N]
                     K   = 1.0/C_tabel[index2] + (dx*factor)*K_diff;      // [N/m]
                     dLe = dLe_tabel[index] + dx*dLe_diff;                // [H/m]


                     //Eddycurrent parameters
                     L2 = Le*L2_0/Le0;
                  // [H]
```

```
                R2 = Le*R2_0/Le0;
              // [ohm]
                dL2 = dLe*L2_0/Le0;
              // [H/m]

                //calculates F matrices
                F[0][0] = 1-(Re+R2)/Le*Ts;                F[0][1] = R2/Le*Ts;                F[0][2]
= 0;                F[0][3] = -Ts/Le*(TEMP_X[0]*dLe+Bl);//Ts/Le*;
              F[1][0] = R2*Ts/L2;                F[1][1] = 1-(R2*Ts/L2);                F[1][2]
= 0;                F[1][3] = -Ts/L2*TEMP_X[1]*dL2;//1-(Rt*Ts/M);
              F[2][0] = 0;                F[2][1] = 0;
                F[2][2] = 1;                F[2][3] = Ts;
              F[3][0] = (Ts*Bl/M);                F[3][1] = 0;
              F[3][2] = -Ts/(C*M);F[3][3] = 1-Rt*Ts/M;

              //calculates G matrice
              G[0] = Ts/Le,    G[1] = 0,                G[2] = 0;          G[3] = 0;

              //multiplies G and F matrice(TEMP_1 = TEMP_X x F)
              mult_matrices_4(F, TEMP_X, TEMP_1);

              TEMP_2[0] = G[0]*u[k]; TEMP_2[1] = u[k]*G[1];
              TEMP_2[2] = u[k]*G[2]; TEMP_2[3] = u[k]*G[3];

              //calculates X(n+1)
              add_matrices_4(TEMP_1, TEMP_2, TEMP_X);

              //filling data to state space vector
              X[0][k] = TEMP_X[0];              X[1][k] = TEMP_X[1];
              X[2][k] = TEMP_X[2];              X[3][k] = TEMP_X[3];

              //setting output
              out[k] = X[2][k];
  }

              //exporting last predicted output for next tick-block
  X_overlap_nonlinear[0] = TEMP_X[0];              X_overlap_nonlinear[1] = TEMP_X[1];
  X_overlap_nonlinear[2] = TEMP_X[2];              X_overlap_nonlinear[3] = TEMP_X[3];
}
```

A3. nonlinear_model_comp

```
void nonlinear_model_comp(float *w, float *out, float *Bl_tabel, float *C_tabel, float *Le_tabel, float *dLe_tabel, float R2_0, float L2_0,
float M, float Re, float Rt, unsigned short int fs, int tickSize, float *X_overlap_nonlinear, short int table_size, float max_excursion,float
stepsize_table)
{
   float F[4][4];                                                                                                    //matrice in state space
model
   float G[4] = {0, 0, 0, 0};                                                  //matrice in state space model
   float X[4][32];                                                                                                   //current, eddycurrent,
excursion, velosity
   float TEMP_X[4] = {0, 0, 0, 0};                          //current, eddycurrent, excursion, velosity
   float TEMP_1[4] = {0, 0, 0, 0};                          //current, eddycurrent, excursion, velosity
                  float TEMP_2[4] = {0, 0, 0, 0};                          //current, eddycurrent, excursion, velosity
   float Ts = 1.0/fs;
                  int k, k2, index, index2;
                  float Bl, Le, C, K;
                  float dBl, dK, dLe, ddLe, L2, R2, dL2;
   float Bl0,C0,K0,Le0,P0,P1,P2,P3,P4,u,v;
   float dx, Bl_diff, Le_diff, C_diff, K_diff, dLe_diff, index_realvalue, factor;

   //Loading linear parameters
   Bl0 = Bl_tabel[1600];                                                  // [N/A]
   Le0 = Le_tabel[1600];                                                  // [H]
   C0 = C_tabel[1600];                                                                    // [m/N]
   K0 = 1.0/C0;                                                                           // [N/m]

                  //fills zeros in X vektor - not necessary but...
                  for(k=0; k<4; k++)
                  {
                                 for(k2=0; k2<32; k2++)
                                 {
                                                X[k][k2] = 0;
                                 }
                  }

                  //Importing last predicted output from previous tick-block
   TEMP_X[0] = X_overlap_nonlinear[0];          TEMP_X[1] = X_overlap_nonlinear[1];
   TEMP_X[2] = X_overlap_nonlinear[2];                    TEMP_X[3] = X_overlap_nonlinear[3];

                  factor = 0.88;
   for(k=0; k<tickSize; k++)
                  {
                     //calculating index for tables, from excursion[m]
                     //index = TEMP_X[2]/stepsize_table + table_size/2 + 0.5;
                     index_realvalue = TEMP_X[2]/stepsize_table + table_size/2;
                     index = index_realvalue;
                     dx = index_realvalue - index;
                                    index2 = index;//((index-1600)*factor +1600);

                     //Calculating diff
                     Bl_diff  = Bl_tabel[index+1] - Bl_tabel[index];
                     Le_diff  = Le_tabel[index+1] - Le_tabel[index];
                     C_diff             = C_tabel[index2+1] - C_tabel[index2];
                     K_diff   = 1/C_tabel[index2+1] - 1/C_tabel[index2];
                     dLe_diff = dLe_tabel[index+1] - dLe_tabel[index];

                     //Loading nonlinear parameters from tables
                     Bl  = Bl_tabel[index] + dx*Bl_diff;                    // [N/A]
                     Le  = Le_tabel[index] + dx*Le_diff;                    // [H]
                     C   = C_tabel[index2]  + dx*C_diff*factor;                                    // [m/N]
                     K   = 1.0/C_tabel[index2] + dx*K_diff*factor;          // [N/m]
                     dLe = dLe_tabel[index] + dx*dLe_diff;                  // [H/m]

                     //Calculation derivatives, (dLe is loaded from table)
                     dBl  = Bl_diff/stepsize_table;                                                // [N/Am]
                     dK   = K_diff/stepsize_table;                                                 // [N/m2]
                     ddLe = dLe_diff/stepsize_table;                                               // [H/m2]

                     //Eddycurrent parameters
```

```
                    L2 = Le*L2_0/Le0;
                // [H]
                    R2 = Le*R2_0/Le0;
                // [ohm]
                    dL2 = dLe*L2_0/Le0;
                // [H/m]

                    //Compensator
                    u = Le/(Bl+dLe*TEMP_X[0])*(TEMP_X[3]*(TEMP_X[2]*dK + (K-K0) - dBl*TEMP_X[0] -
ddLe*(TEMP_X[0]*TEMP_X[0])/2 - (Bl0*Bl0)/Le0) + Re*TEMP_X[2]*(K-K0)/Le0 - Re*TEMP_X[0]*(2*Bl+dLe*TEMP_X[0])/(2*Le0) +
Bl0*w[k]/Le0) + Re*TEMP_X[0] + Bl*TEMP_X[3] + dLe*TEMP_X[0]*TEMP_X[3] + R2*(TEMP_X[0]-TEMP_X[1]);

                    //calculates F matrise
                    F[0][0] = 1-(Re+R2)/Le*Ts;          F[0][1] = R2/Le*Ts;          F[0][2]
= 0;          F[0][3] = -Ts/Le*(TEMP_X[0]*dLe+Bl);//Ts/Le*;
                F[1][0] = R2*Ts/L2;          F[1][1] = 1-(R2*Ts/L2);          F[1][2]
= 0;          F[1][3] = -Ts/L2*TEMP_X[1]*dL2;//1-(Rt*Ts/M);
                F[2][0] = 0;          F[2][1] = 0;
                    F[2][2] = 1;          F[2][3] = Ts;
                F[3][0] = (Ts*Bl/M);          F[3][1] = 0;
                F[3][2] = -Ts/(C*M);F[3][3] = 1-Rt*Ts/M;

                //calculates G matrise
                G[0] = Ts/Le,     G[1] = 0,          G[2] = 0;     G[3] = 0;

                //multiplies G and F matrice(TEMP_1 = TEMP_X x F)
                mult_matrices_4(F, TEMP_X, TEMP_1);

                TEMP_2[0] = u*G[0]; TEMP_2[1] = u*G[1];
                TEMP_2[2] = u*G[2]; TEMP_2[3] = u*G[3];

                //calculates X(n+1)
                add_matrices_4(TEMP_1, TEMP_2, TEMP_X);

                //filling data to state space vector
                X[0][k] = TEMP_X[0];          X[1][k] = TEMP_X[1];
                X[2][k] = TEMP_X[2];          X[3][k] = TEMP_X[3];

                //setting output
                out[k] = u;//X[2][k];
    }

                //exporting last predicted output for next tick-block
    X_overlap_nonlinear[0] = TEMP_X[0];          X_overlap_nonlinear[1] = TEMP_X[1];
    X_overlap_nonlinear[2] = TEMP_X[2];          X_overlap_nonlinear[3] = TEMP_X[3];
}
```

B4 get_linear_parameters

```
{
//loudspeaker B(8480277)
    *Re = 5.02882;
    *L2 = 0.000565304;
    *R2 = 2.67697;
    *Rt = 2.25391;
    *M = 0.0195471;
    *Le_lin = 0.000389793;
    *C_lin = 0.000538315;
    *Bl_lin = 7.18583;
}
```

B5 get_nonlinear_parameters

```
void get_nonlinear_parameters(float *Bl, float *C, float *Le, float *dLe, short int table_size, float max_excursion,float stepsize_table)
{
                short int i;
                float x=0;

                //polynomes for nonlinear parameters

   //31211
   //float Le_p[9] = {0.58157, -0.031046, -0.0066093, 0.0011830, 0.00042152, -0.00015811, -6.4289e-5, 1.1929e-5, 4.5528e-6};
   //float Bl_p[9] = {7.03, -0.023848, -0.055244, 0.0099365, -0.0042554, -0.00014267, 9.478e-5, -7.2017e-6, 2.3069e-7};
   //float C_p[9] = {0.241, 0.0045478,                -0.0050385,        0.00022905,        0.00012458, -0.10724e-4, -2.6118e-6,
2.8368e-7, 0.6371e-8};

   //8084277
   //float Le_p[9] = {4.42852e-004, -3.99246e-003, -0.730067, 11.3856, 2426.82, -8064.92, -3.80829e+007, 1.0525e+008,
3.24411e+011};
   //float Bl_p[9] = {7.19845, -22.7689, -10185.5, 3.70864e+006, -1.65104e+009, -5.0827e+010, 2.04298e+013, 1.40152e+014, -
7.96303e+016};
   //float C_p[9] = {1.12474e-003, 1.91161e-002, -33.3252, -1199.15, 636426, 2.74833e+007, -7.21205e+009, -2.23333e+011,
3.37872e+013};

   float Le_p[9] = {5.82481e-004, -7.58251e-003, 0.277347, 38.1499, -695.65, -1.7902e+006, -7.5998e+007, 4.13538e+010,
3.36663e+012};
   float Bl_p[9] = {7.18999, -5.79654, -13760.8, 1.21422e+006, -1.15292e+009, -5.35162e+009, 6.48184e+012, -2.18535e+014,
8.14653e+016};
   float C_p[9] = {1.20736e-003, 5.33297e-002, -43.9281, -4956.1, 1.02868e+006, 2.04476e+008, -1.12013e+010, -3.53739e+012, -
6.31026e+013};

   //creating tabells for nonlinear parameters
   for(i=0; i<table_size; i++)
   {
                x  = (i-table_size/2)*stepsize_table;  //(i-1600)/160000.0;

                Bl[i] = Bl_p[0] + Bl_p[1]*x + Bl_p[2]*pow(x,2) + Bl_p[3]*pow(x,3) + Bl_p[4]*pow(x,4) + Bl_p[5]*pow(x,5) +
Bl_p[6]*pow(x,6) + Bl_p[7]*pow(x,7) +Bl_p[8]*pow(x,8);
                C[i]  = C_p[0] + C_p[1]*x + C_p[2]*pow(x,2) + C_p[3]*pow(x,3) + C_p[4]*pow(x,4) + C_p[5]*pow(x,5) +
C_p[6]*pow(x,6) + C_p[7]*pow(x,7) +C_p[8]*pow(x,8);
                                Le[i] = Le_p[0] + Le_p[1]*x + Le_p[2]*pow(x,2) + Le_p[3]*pow(x,3) + Le_p[4]*pow(x,4) +
Le_p[5]*pow(x,5) + Le_p[6]*pow(x,6) + Le_p[7]*pow(x,7) +Le_p[8]*pow(x,8);

                                dLe[i]= Le_p[1] + 2*Le_p[2]*x + 3*Le_p[3]*pow(x,2) + 4*Le_p[4]*pow(x,3) + 5*Le_p[5]*pow(x,4) +
6*Le_p[6]*pow(x,5) + 7*Le_p[7]*pow(x,6) + 8*Le_p[8]*pow(x,7);
    }

}
```

B6 Gain

```
void Gain(float *Gain, float *table, int table_Size)
{
   int i;
   float sum = 0;
   for(i=0;i<table_Size;i++)
   {
                sum = sum + table[i];
   }
   *Gain = sum/table_Size;
```

B7 Peak

```
void Peak(float *Peak, float *table, int table_Size, unsigned short int tickSize)
{
                        int i;
                        float P=0;
   for(i=0; i<table_Size; i++)                          //summing one tick
                        {
                           if(table[i]>P)
                           {
                                          P = table[i];
                           }
                        }
                        *Peak = P/tickSize;
}
```

B8 Rms

```
void Rms(float *Rms, float *table, int table_Size, unsigned short int fs)
{

                        int i;
                        float sum=0;

   for(i=0; i<table_Size; i++)
   {
                        sum = sum + table[i];
   }

   *Rms = sqrt(sum/(fs));

}
```

B8 add_matrices_3

```
void add_matrices_3(float a[3], float b[3], float result[3])
{
  int i, j;
  for(i=0; i<3; i++)
  {
                                          result[i] = 0;
  }
  for(i=0; i<3; i++)

                        {
                           result[i] = a[i] + b[i];
                        }
}
```

B9 mult_matrices_3

```
void mult_matrices_3(float a[3][3], float b[3], float result[3])
{
  int i, k;
  for(i=0; i<3; i++)
  {
                                          result[i] = 0;
  }
  for(i=0; i<3; i++)
  {
                        for(k=0; k<3; k++)
                        {
```

```
                                        result[i] = result[i] + a[i][k] * b[k];
                        }
  }
}
```

B10 add_matrices_4

```
void add_matrices_4(float a[4], float b[4], float result[4])
{
                    int i, j;
  for(i=0; i<4; i++)
  {
                                        result[i] = 0;
  }
  for(i=0; i<4; i++)

                    {
                      result[i] = a[i] + b[i];
                    }
}
```

B11 mult_matrices_4

```
void mult_matrices_4(float a[4][4], float b[4], float result[4])
{
  int i, k;
  for(i=0; i<4; i++)
  {
                                        result[i] = 0;
  }
  for(i=0; i<4; i++)
  {
                      for(k=0; k<4; k++)
                      {
                                        result[i] = result[i] + a[i][k] * b[k];
                      }
  }
}
```

B12 main

```
//constant parameters
float X_overlap[3] = {0, 0, 0}, X_overlap_nonlinear[4] = {0, 0, 0, 0};
unsigned short int                      fs = 48000, f = 50;
unsigned short int table_size = 3200;
float max_excursion = 0.02, stepsize_table;

//Storage last value of tick.
float *X_overlappk = &X_overlap[0];
float *X_overlap_nonlinear_pk = &X_overlap_nonlinear[0];

float testin[1100], testout[110], model[110], pi = 3.1415927;
float *testinpk = &testin[0], *testoutpk = &testout[0], *modelpk = &model[0];
float Bl[3200], C[3200], Le[3200], dLe[3200];//, dC[3200], dBl[3200], ddLe[3200];
int i,init = 0, index=0, Rms_table_Size, counter1=0;
float x, P_Amp_Gain = 1, u_ticksum=0, u_Peak_tick=0, u_Peak=0, u_Rms=0, u_tab[1500];
float x, U_measured_ticksum=0, U_measured_Peak_tick=0, U_measured_Peak=0, U_measured_Rms=0, U_measured_tab[1500];
float Rms_vindowtime = 1, Gain_tab[1500], C_old[3200], C_change;
float W[32], U[32], Model[32], Gain_feedback=37, Limiter=1, peak=0;

SEG_MOD_FAST_CODE static void AMF_compensator_Render(AMF_compensator * restrict instance,float * restrict * buffers,int
tickSize)
{
   int i;
   float *w  = buffers[0]; // Don't use restrict because we say they can alias (see .h file)
   float *U_measured = buffers[1];
   float *in3 = buffers[2];
   float *in4 = buffers[3];
   float *u = buffers[4];
   float *w2 = buffers[5];
   float *Model = buffers[6];
   float *out4 = buffers[7];
   float compliance_scaler = instance->compensatoramp;
   //float compliance = instance->compliance;
//initialisering
   if(init==0)
   {
      //
      stepsize_table = (2*max_excursion)/table_size;

      //Loading linear parameters, 31211 is default
      get_linear_parameters(&Re,&L2,&R2,&M,&Rt,&Bl_lin,&Le_lin,&C_lin);

                  //Loading Nonlinear parameters tabels
                  get_nonlinear_parameters(Bl, C, Le, dLe, table_size,max_excursion,stepsize_table);

                  //changes in compliance
                  C_change = 0.001;

                  for(i=0; i<table_size; i++)
                  {
                              C_old[i] = C[i];
                  }

   }

//calculating Rms, Peak, P.Amp-Gain....
   u_ticksum=0, U_measured_ticksum=0;



//Setting Gain / (input signal)

   P_Amp_Gain = 8.944;


//controlling index
   init++;
   index++;
```

```
    if(index>Rms_table_Size)
    {
        init = 1;
        index = 0;
    }

//Simulating the Power Amplifier by scaling the input signal
    for(i=0; i<tickSize; i++)
                        {
                            w[i] = P_Amp_Gain*w[i];              //gain controll - input
                        }

//Scaling the complianse

                        //uptuning
                        if(C[1600] < (C_old[1600]*compliance_scaler)-(C_change*0.001) )
                        {
                                        for(i=0; i<table_size; i++)
                        {
                                        C[i] = C[i] + (C[i] * C_change);
                                        }
                        counter1++;
                        }

                        //downtuning
                        if(C[1600] > (C_old[1600]*compliance_scaler)+(C_change*0.001) )
                        {
                                        for(i=0; i<table_size; i++)
                        {
                                        C[i] = C[i] - (C[i] * C_change);
                                        }
                        counter1++;
                        }


//Comparator/Nonlinear linearization
    //linear_model(W, U, Bl_lin, Le_lin, C_lin, M, Re, Rt, fs, tickSize, X_overlap);
    nonlinear_model_comp(w, testoutpk, Bl, C, Le, dLe, R2, L2, M, Re, Rt, fs,
tickSize,X_overlap_nonlinear,table_size,max_excursion,stepsize_table);
    //nonlinear_model(w, modelpk, Bl, C, Le, dLe, R2, L2, M, Re, Rt, fs,
tickSize,X_overlap_nonlinear,table_size,max_excursion,stepsize_table);

//Reducing signal before the real Power Amplifier.
    peak = 0;
                        for (i=0; i<tickSize; i++)
    {
                        //u[i] = testoutpk[i];
                        u[i] = testoutpk[i]*1/P_Amp_Gain;
                        //u[i]= 1000*modelpk[i];//Model[i]
    }

    w2=w;


SEG_MOD_SLOW_CONST const AMF_ModuleClass AMFClasscompensator = {

    /* Flags */
    0,

    /* Render function */
    (AMF_RenderFunction)AMF_compensator_Render,  // render function

    /* Default bypass */
    (void *)0,

    /* Input descriptor - 1 input, and it is mono. */
    1, 0,

    /* Output descriptor - 1 output, and it is mono. */
    1, 0,
}
```