Development of a Tool to Detect and Analyze Regular and Irregular Loudspeaker Distortions

Master Thesis

by

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STATUTORY DECLARATION

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Abstract

At the conversion from electrical energy to acoustic energy using an electro-dynamic loudspeaker, undesired distortions are generated. These can be divided into regular distortions caused by design limitations and irregular distortions caused by production faults. During the development phase of a loudspeaker, both types of distortions need to be minimized, so a clear detection of the root cause is necessary. Presently several measurement systems are focused on the detection of distortions and these have different methods for the analysis, but there is no system combining all methods. In this thesis, an analysis software is to be developed which uses data from common measurement systems and digital signal processes to provide root cause analysis for the acoustic development. This facilitates the improvement and elimination of distortions.

The main focus of this software program is the analysis of micro-speakers because common measurement systems are not developed for transducers of this size. Irregular distortions, like a hitting membrane, are very annoying because they are highly audible and can damage the loudspeaker during lifetime. For distortion root cause detection, the knowledge of the temporal occurrence is very important. Therefore the root of the distortion is combined with the actual membrane position to see critical areas of displacement. For applications where the membrane position is not measurable the software program calculates the excursion from the sound pressure signal. Using statistical parameters of the distortions can reduce the number of potential root causes. Analyzing the temporal trend of the distortion is necessary so a stimulus period-wise calculation has to be applied. Additionally, several kinds of frequency domain plots are available to analyze the spectral shape of the distortion and define optimum filter settings for the quality control at the production. The tool also provides a comparison of some filters settings to find the optimum parameters for each kind of distortion.

Zusammenfassung

Bei der Umwandlung elektrischer Energie in akustische Schalldruckwellen mittels elektro-dynamischen Lautsprecher entstehen Verzerrungen die grundsätzlich unerwünscht sind. Dabei handelt es sich um reguläre Verzerrungen, die durch die Grenzen bei einem Design des Lautsprechers entstehen, und irreguläre Verzerrungen, die durch Produktionsfehler verursacht werden. Bei der Entwicklung von Lautsprechern wird versucht beide Arten der Störung zu minimieren, deswegen ist es von großer Bedeutung die Ursache solcher Fehler zu kennen. Derzeit gibt es einige Messsysteme, die den Fokus auf der Detektion dieser Störungen mit teils unterschiedlichem Ansatz legen, aber es gibt noch kein System, das alle Methoden vereint. In dieser Arbeit entsteht eine Analysesoftware, die Daten herkömmlicher Messsysteme weiterverarbeitet und speziell für den Akustikentwickler darstellt. Dadurch soll die Ermittlung von Fehlerursachen deutlich erleichtert werden, um die Optimierung von Verzerrungen zu ermöglichen.

Hauptfokus dieses Programms liegt auf der Analyse von Kleinlautsprechern, da herkömmliche Messsysteme oft für solch kleine Abmessungen nicht optimal geeignet sind. Besonders störend sind irreguläre Verzerrung wie es etwa eine anstehende Membrane verursacht, denn diese Art der Verzerrung ist meist gut wahrnembar und kann im Betrieb auch stärker werden oder den Lautsprecher sogar zerstören. Um die Ursache der Störung genau detektieren zu können, ist es wichtig den zeitlichen

Ursprung zu wissen. Dabei wird auf die Membranposition referenziert, um mögliche Problemstellen zu erkennen. Falls die Auslenkung der Membran nicht gemessen werden kann, dann berechnet die Software die Membranposition anhand des gemssenen Schlladrucks. Statistische Parameter der Verzerrung sind oft hilfreich, um den Kreis der Fehlermöglichkeiten einzugrenzen. Hier ist die periodenweise Auswertung bezogen auf das Anregungssignals eine sehr gute Art um den zeitlichen Verlauf verfolgen zu können. Außerdem werden mehrere Darstellungen des Spektrums zur Verfügung gestellt um die Frequenzkomponenten der Verzerrungen zu analysieren und um optimale Filter für die Qualitätsüberprüfung bei der Produktion zu definieren. Dazu können mehrere Filter mit unterschiedlichen Einstellungen verglichen werden, um die besten Parameter für die jeweilige Art von Verzerrung zu finden.

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1 Introduction

1.1 Motivation

An electro-dynamic loudspeaker is not a simple linear transducer transforming electrical voltage and current directly proportional to sound pressure and particle velocity. The quality of a loudspeaker is mainly characterized by the occurrence, or rather the absence, of distortions, caused by nonlinearities in the speaker design, e.g. inconstant magnet field in the air gap.

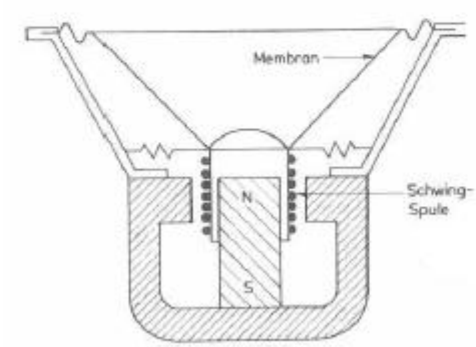

Figure 1: Electro-dynamic loudspeaker (Zollner und Zwicker 1998)

Additionally, defects during the production can generate irregular distortions which are often called “Rub & Buzz”. These kinds of distortions are likely generated by the two most common root causes, a rubbing voice coil or other loose parts leading to a buzzing noise. But there are lots of other root causes for irregular distortions which are also usually summed up in this parameter. It is generally easy to detect loudspeakers with this kind of failure because the bad masking conditions of the distortion making the defect prominently audible. This is usually caused by different frequency and statistic properties of the distortion. But tests with humans are not suitable for mass production, and a measurement system with lower detection level than the human ear is also desirable. Inaudible Rub & Buzz can also be a problem, because the effect can rise during use or it can damage the loudspeaker in the application.
This project is focused on the analysis of micro-loudspeakers and this kind of loudspeaker has some special properties due to its limited space. For a normal failure analysis the engineer has the following tools to discover the root cause:

- X-Ray
- Cross Section
- Disassembly
- Acoustic Measurement
- Stroboscope Analysis
- DC-Offset Measurements
- Rub & Buzz vs. Power

The acoustic measurement is the main test of the loudspeaker functionality because all samples are generally tested in this way before the shipment to the customer. So this test should guarantee the defect-less use of the product at the delivery and during lifetime. Lots of measurement systems offer this kind of quality test but each has its advantages and disadvantages.

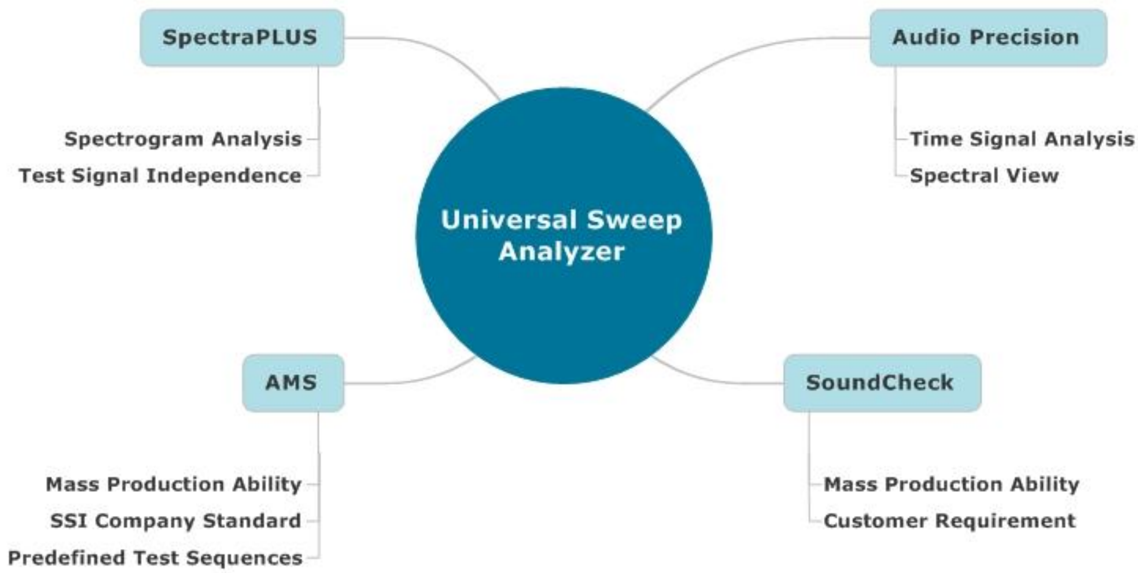

Figure 2: Overview of acoustic measurement systems

Figure 2 illustrates an overview of some common measurement systems and its advantages. SpectraPLUS and Audio Precision are systems for loudspeaker tests during the developments phase. They have many internal features like spectral view or time domain analysis. However, they are not useable for the test of big quantities. The software AMS (Acoustic Measurement System) was developed for the company Sound Solutions International and is optimized for short test durations. The purpose was to have a cheap solution for end-of-line testing in a fully automated production with a cycle time of about 1 second. Some customers require a unique test system for all their suppliers, so AMS cannot be used. SoundCheck is quite comparable with AMS and can be used in this case. It is also designed for the quality control in mass production.

The Universal Sweep Analyzer is a software program developed in this master thesis. It supports the acoustic engineer at the failure analysis and in deeper studies. To make the tool useable for all test systems it is necessary to define an interface for a flexible data import. The approach here is to measure all necessary signals with the customer-required test system and to only export the raw data. All further analyses are calculated in the Universal Sweep Analyzer to have all plots available independently of the used data acquisition system.

The main function of the Universal Sweep Analyzer is the root cause detection of failures. Additionally, it can be used for the definition of limits in mass production. Finding proper limits is very important: To avoid any kind of irregular distortion at loudspeakers shipped to customers, harsh limits are necessary. Too narrow limits lead to a waste of good samples. The Universal Sweep Analyzer supports this procedure independently of the used test system.

1.2 Assignment of Tasks

For the development and production of loudspeakers, every kind of distortion is important, and the acoustic engineer is interested in the occurrence, intensity and especially in the root cause of each. Lots of measurement systems are available which already have the focus on the detection of regular and irregular distortions. But micro-speakers have very extreme dimensions and are used in highly specialized applications, thus standard systems cannot be used without considering the area of validity.

This thesis does not aim to develop a complete measurement system including all data acquisition steps, but to develop software that should facilitate the post processing analysis of test signals. A standard measurement setup for a complete description of the loudspeaker including common acquisition software should be used, and the new tool should import this data for further analysis.

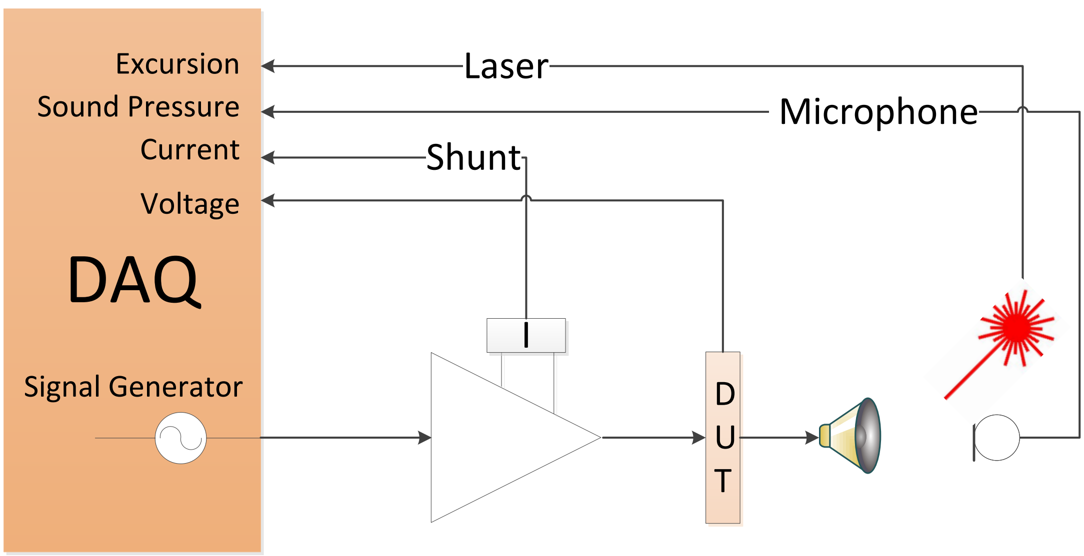

Figure 3: Standard Measurement Setup

The software program must be able to import all channels of the measurement and display it in different kinds of plots, e. g. the spectrum of the sound pressure signal. These graphs should support the engineer with the detection of any signal distortion and with discovering the root cause. The main focus is on the representation of the following properties that are very important for irregular distortions.

- Energy
- Impulsivity (Crest Factor)
- Stochastic and deterministic components
- Temporary occurrence (dependent on displacement)
- Stimulus frequency dependency
- Spectral shape

Optionally, the software provides a classification of the distortion in line with predefined thresholds for these parameters, providing the engineer with a list of possible root causes.

1.3 Overview

For the development of a software program which analyzes loudspeaker distortions it is important to summarize the properties and root causes of all kinds of distortions. A list of possible root causes including the split of regular and irregular distortions is shown in chapter 2. This separation is essential for finding critical parameters which can be helpful for the root cause detection. The properties of these parameters and the usual values for different kinds of defects are shown at the end of this section.

Chapter 3 illustrates the common methods for the analysis of loudspeaker distortions. A brief overview of mechanical methods is included but the main focus is on the acoustical analyses. The aim for the Universal Sweep Analyzer was to offer and explain all common parameters and diagrams.

For the development of the Universal Sweep Analyzer several signal analysis procedures are necessary. They calculate or transform the imported raw data to helpful signals and plots. These procedures are described in chapter 4. Generally, all signal processing steps are well-known and good descriptions are available. This chapter focusses on the basics of the algorithms and their use in this project.

Chapter 5 describes how the software program is structured and developed. First, all plots and visualization are explained to the operating person. It describes how to use the software for failure analysis. Afterwards, the details of the implementation including the used algorithms are illustrated.

Chapter 6 shows some generated defects of a loudspeaker and how to use the Universal Sweep Analyzer to find the root cause with a single sweep measurement. It describes the general approach of using the software, and is a manual for further failure analyses. Characteristic plots for each defect are shown to see usual values and shapes of the key parameters. At the end of this chapter a micro-speaker with an unknown defect is analyzed to prove the functionality for this kind of loudspeakers also.

2 Loudspeaker distortions

Several types of loudspeakers have been developed using different physical phenomena to generate acoustic waves, but the most common is the electro-dynamic loudspeaker. Especially for micro-speakers, this type is mainly used because of the possibility to generate sufficiently high sound pressure within very small dimensions.
The electro-dynamic loudspeaker transforms the electric energy to mechanical energy (force/velocity) and then to acoustic sound energy (pressure/velocity waves).

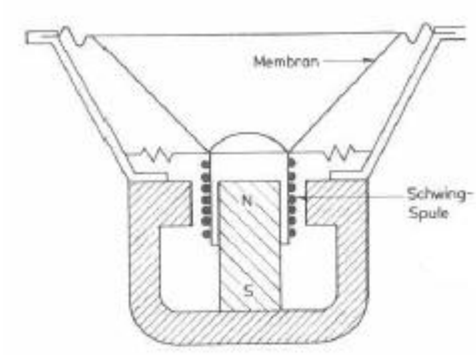

Figure 4: Electro-dynamic loudspeaker (Klippel 2013)

Due to some limitations in this loudspeaker design, the transition from the electric to the acoustic domain is always affected by different nonlinearities. This kind of nonlinearities lead to so-called “regular distortions” because the nonlinearities are inherent in the design and the resulting distortions occur in normal use, even without any defect.
Additionally, the loudspeaker can produce irregular distortions. These kinds of distortions do not appear in normal use without production related defects, but in case of a failure this nonlinearity can be very critical because it is highly audible and can damage the loudspeaker during its lifetime.

2.1 Causes of Regular Distortions

As mentioned before, regular distortions are caused by the design and occur in normal use. Lots of effects can cause nonlinearity, but this project has its focus on the irregular distortions so this is just a partial overview of some root causes (for in-depth explanation of these root causes see (Klippel 2005)).

2.1.1 Stiffness

To center the membrane in its rest position, different forms of suspensions are typically used. These methods have different operating modes, but all of them build a force to bring the membrane into the center position of the loudspeaker.

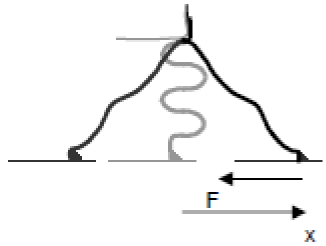

Figure 5: Suspension of a common loudspeaker (Klippel 2005)

In the simple model, the resulting force B is proportioned on the excursion, hence the stiffness is linear. This is valid for low displacements only because in the large signal domain the force rises exorbitantly more than the excursion.

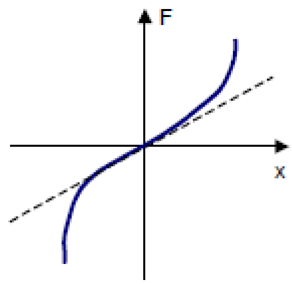

Figure 6: Force to Displacement relationship of standard membrane suspension (Klippel 2005)

Additionally, because of the visco-elastic behavior of the suspension material, the stiffness varies with the excitation frequency, even for low excursion and linear behavior. Both nonlinearities distort the resonance behavior of the oscillating circuit of mass and stiffness of the membrane relative to the excursion and its frequency, which has the effect of adding extra harmonics in the sound pressure.

2.1.2 Force Factor

The transformation from the electrical to the mechanical domain is caused by a current flowing through the voice coil in a constant magnetic field. An electrical current in a magnetic field creates a force acting on the conductor, the so-called Lorentz force. For a loudspeaker with a flexible mounted coil, this leads to a movement of the wire in the magnetic field.
Due to the limitation in the design with a permanent magnet and an air gap for the coil, the magnetic field is not constant over a wide area.

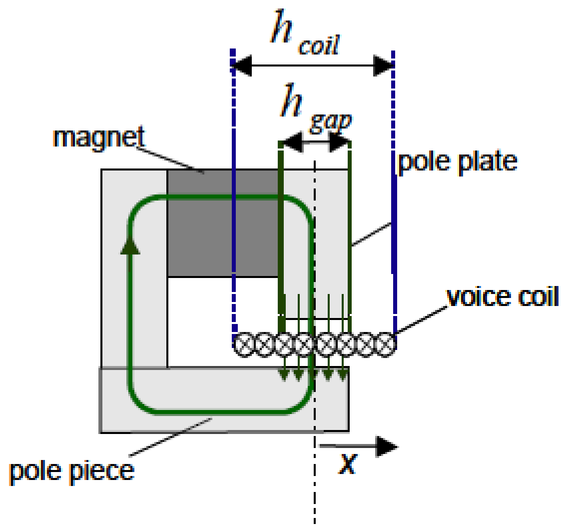

Figure 7: Structure of magnet and voice coil (Klippel 2005)

Assuming a constant magnet field in the area of h_{gap} and a decreasing field outside of the air gap, the force gets smaller with higher displacement of the voice coil.
The multiplication of the magnet field B and the length of the wire l represent the acting force on the conductor, so it is a good parameter for the analysis of the dependency between displacement and the electrical input power.

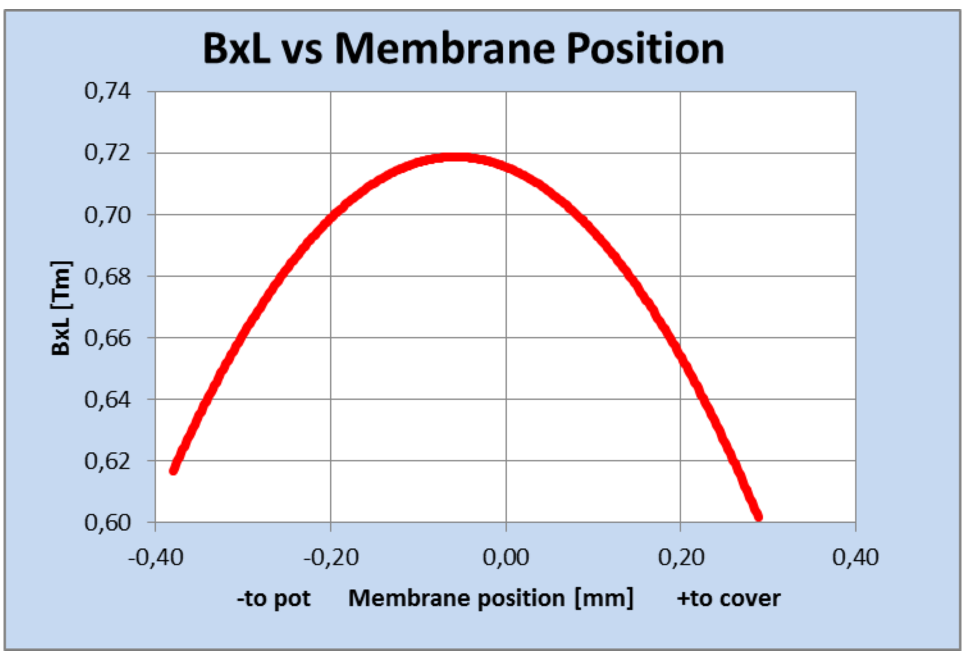

Figure 8: Bl vs membrane position of a common micro-speaker

A steeper curve indicates a smaller area with constant magnet field and leads to higher nonlinearities. A longer voice coil or thicker pole pieces normally make the Bl(x) curve flatter, thus decreasing the regular distortions because of the force factor.

2.1.3 Voice Coil Inductance

The inductance of the voice coil is dependent on their position in the air gap because the magnet flux is not constant over the whole working area of the voice coil. For a positive displacement, some windings of the coil are in the free field and the inductance is smaller than at the normal position. In the other direction, the inductance rises because of the higher magnet flux inside of the magnet.

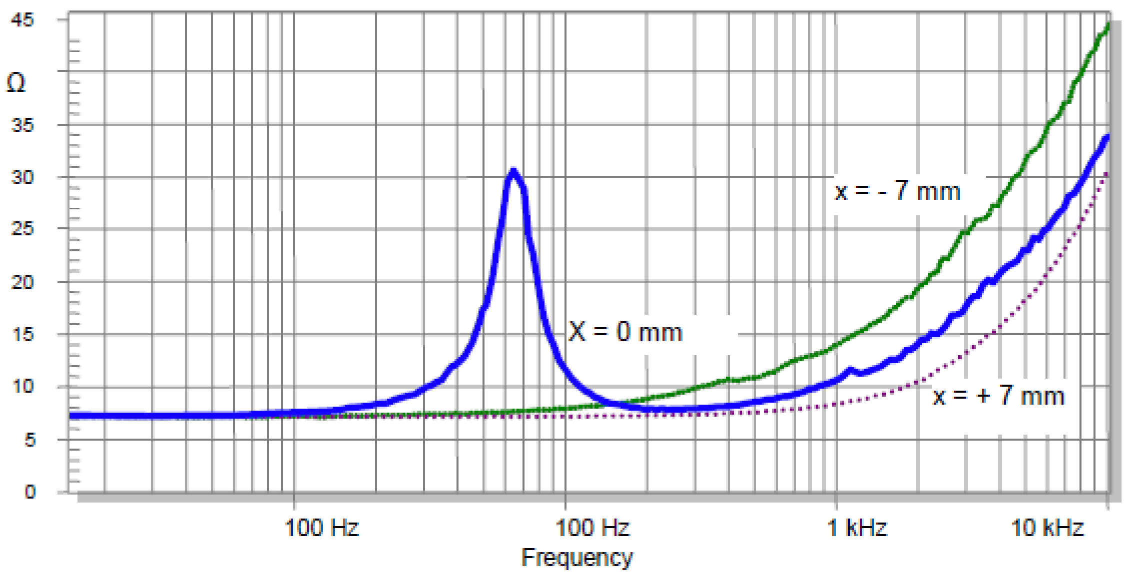

Figure 9: Dependency of displacement and voice coil inductance (Klippel 2005)

Additionally, the voice coil inductance depends on the magnitude of the input current i. Because of the nonlinear behavior of the relative permeability μ(i) the flux density B is not directly proportional on the magnetic field strength H.

$$B=\mu(i)*H$$

For high input currents the magnet is saturated and the μ is much lower, so the change of the flux density is smaller for a constant change of magnet field strength. This effect of the variability of the permeability is dependent on the magnet material and is called "Flux Modulation".

The sum of these effects causes harmonic distortions already at low amplitudes but increases with more input power.

Of course, lots of other root causes can produce nonlinearities, e.g. nonlinear material properties, geometric variations or the Doppler Effect of the moving membrane, but these effects are not so prominent for micro-speakers, and the main focus of my thesis is on irregular distortions and their detection.

2.2 Causes of Irregular Distortions

Regular Distortions of loudspeakers are caused by the design and the properties of the material and can be minimized through careful design and high quality materials, but they can never be completely eliminated. Although these effects are disturbing and need to be optimized, it is often not the most important issue because they have relatively low audibility and will not damage the loudspeaker.
Contrary, irregular distortions have completely different properties and root causes and can lead to much bigger problems. This kind of distortion normally appears if any defect in the production or assembly process occurs or happens during the speaker's lifetime. For inexperienced engineers, it is often difficult to detect the root cause of such a distortion because of the many possibilities. A small list of defects from (Klippel 2013) provides a handy overview. Their impact on the acoustics and how to detect them is explained later in 2.3.3 and 0.

2.2.1 Hitting Voice Coil

If the current in the voice coil produces a force which induces an excursion with too high displacement and the membrane hits the upper or lower mechanical limit, the membrane is stopped abruptly and the acoustic waveform is disturbed.

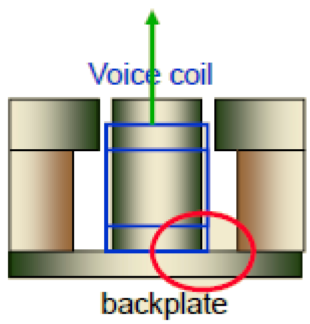

Figure 10: Hitting voice coil (Klippel 2010)

The distortion of the sound pressure signal has an impulsive character and occurs periodically with the stimulus. This property is helpful for the detection of the root cause.

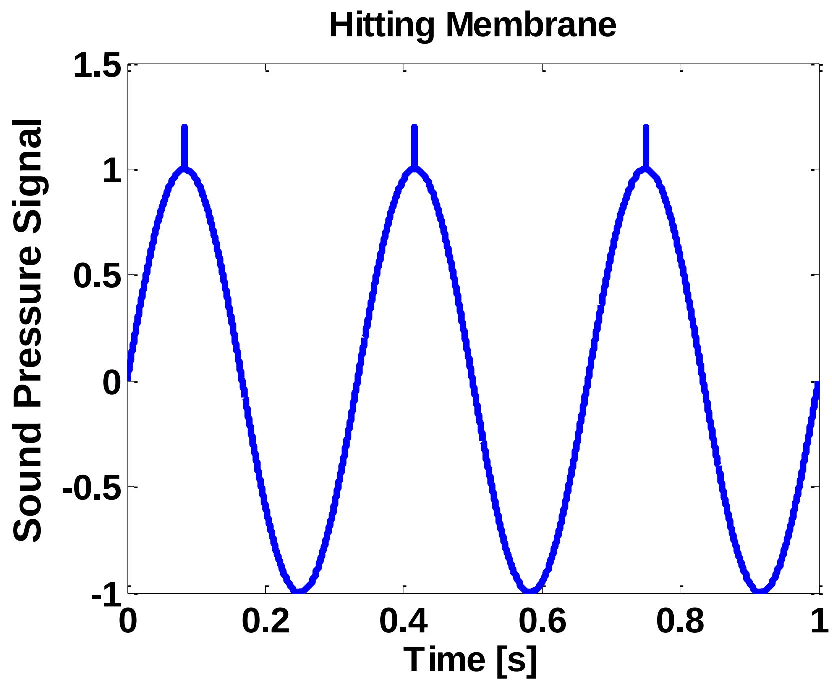

Figure 11: Representative distorted sound pressure signal of a loudspeaker with a hitting membrane

Normally this effect happens at the lower end of the coil movement where the voice coil or the voice coil former hits the back plate, but e.g. in the case of micro loudspeakers, for small place requirements, this effect exists at the upper end of the loudspeaker too. This can happen if, in front of the speaker, a basket is mounted to guarantee safe mounting of the application.

The limited excursion capabilities of the membrane cause a distortion at a known displacement so this effect appears at constant excursion amplitude and is called “excursion driven”. The distortion is deterministic because the error has a peak at each period with sufficient displacement and appears above a certain threshold only, thus the distortion is eminent or non-existent.

2.2.2 Rubbing Voice Coil

Another very common defect is a rubbing voice coil, and it can have several root-causes.

Either the basket is deformed because of improper mounting of the driver, because of transport damages, or because of a separated coil from the former due to too high input power. Another root cause can be displaced pole plates which cause an asymmetrical magnetic field, hence a space dependent force factor. Or in general: any asymmetry of stiffness, mass and force can be a root cause for a rubbing voice coil.

All these defects have nearly the same result in the output sound pressure level. Because of the rubbing of the voice coil the membrane doesn’t move unidimensionally along the main axis and a rocking mode arises.

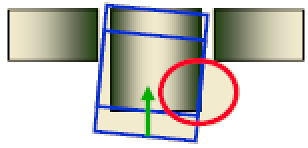

Figure 12: Rubbing voice coil (Klippel 2010)

The rubbing of the voice coil generates stochastic distortion but it appears synchronously with the stimulus. It is initiated by the tilting of the voice coil so it occurs at the highest acceleration which is corresponding with the positive and negative apex of the membrane displacement. Consequently, the distortion has a stochastic component because of the generated distortion itself, and a deterministic component because of the periodicity with the stimulus (Klippel 2013).
This assumption is valid for common Hi-Fi loudspeakers, but for micro-speakers it is slightly different. This kind of transducers normally have lower damping than Hi-Fi speakers, consequently the excursion slightly above the resonance frequency is still quite high. The rocking mode is normally located in this frequency range. It exists at the resonance frequency of the toroidal mode and is dominated by the design of the membrane and coil and their mass distribution. In standard-size loudspeakers, the excursion at the rocking mode is very small so the tilt due to the tumbling can be ignored at this frequency. But for micro-speakers the excursion is usually much bigger at the rocking mode so the most critical frequency range concerning a rubbing voice coil is at the rocking mode itself. Measurements have shown that the mode has the biggest tilt at the zero-position of the membrane, thus the distortion does not occur at the maximum displacement, but rather at the rest-position.

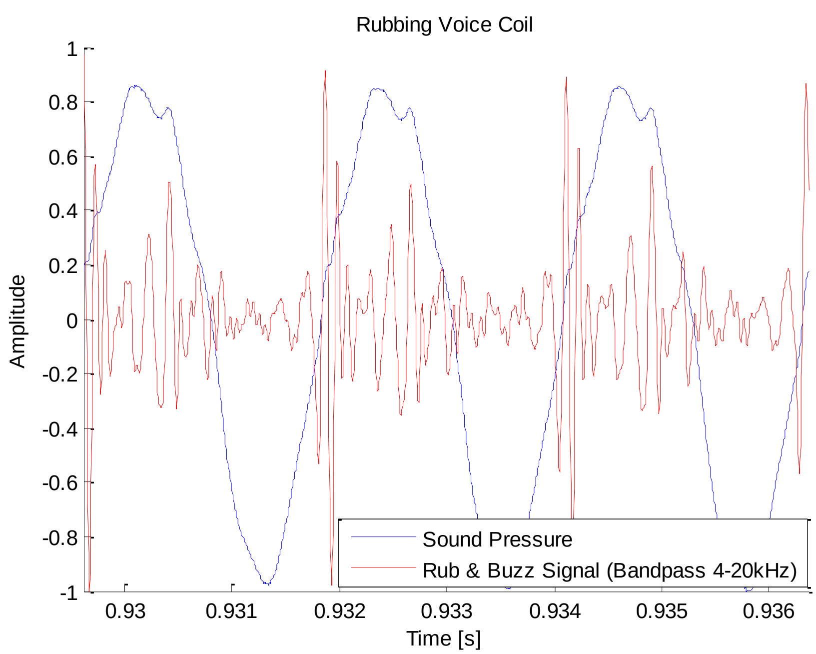

Figure 13: Sound pressure and Rub & Buzz signal of a loudspeaker with a rubbing voice coil

A real measurement of a rubbing voice coil with a micro-speaker is here shown where the distortion is at its maximum at the zero-position of the stimulus. The excitation frequency is at the rocking mode of the speaker. For the separation of the irregular distortion from the complete sound pressure signal, a band pass filter has been used. This is a simple method for the separation and is described in chapter 3.2.5.

2.2.3 Buzzing loose Joint

Most of such defects behave like nonlinear oscillators. This means that a new mode of vibration is excited above a critical amplitude of excursion. Then a separate vibration, triggered by the stimulus frequency, disturbs the output signal.

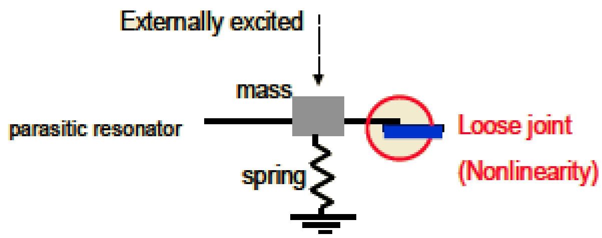

Figure 14: Buzzing loose joint (Klippel 2010)

This oscillation starts for each period at the same phase and the envelope of the signal has an exponential decay. The distortion is deterministic and the ratio of mechanical mass, spring and resistance of the loose joint determines its resonance frequency and the signal form. An interesting property of the buzzing loose joint is that above the critical amplitude, the power of the buzzing is independent of the power of the stimulus. This effect helps a lot in detecting the reason for a nonlinear distortion.

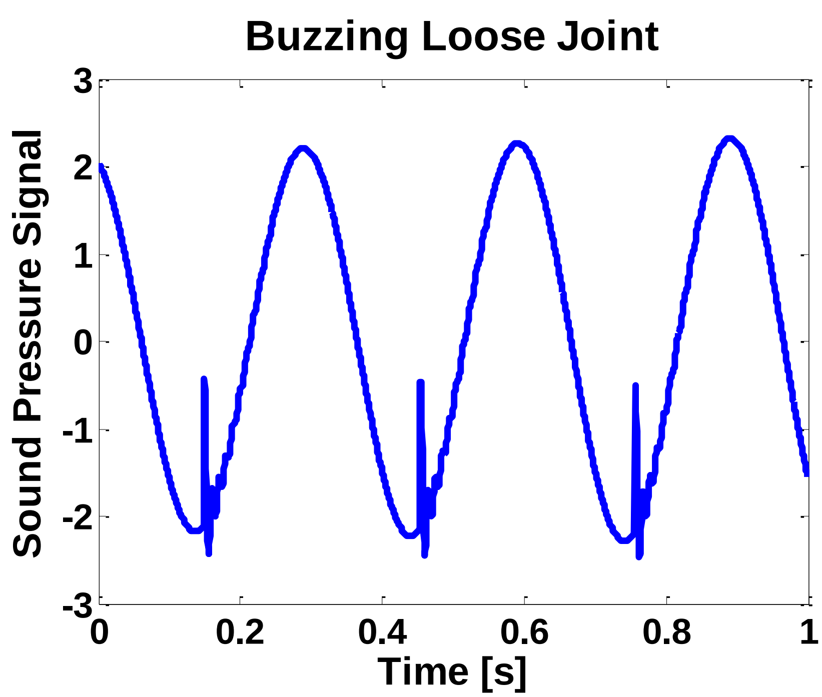

Figure 15: Typical distorted sound pressure signal of a loudspeaker with a buzzing loose joint

2.2.4 Flow Noise at Air Leak

Flow noise is very annoying because it has high audibility and might be inacceptable if customers get products with this defect. There are two different root causes of air leakage. The main part is tiny holes at the membrane because of improper gluing along the voice coil or the boundary of the membrane itself.

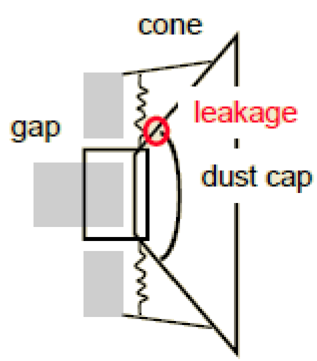

Figure 16: Leakage in the membrane (Klippel 2010)

Another effect occurs in the application with a closed box where the high pressure under the dust cap can cause very high velocities at the venting hole, which generates flow noise.

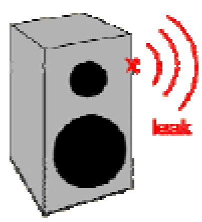

Figure 17: Leakage in the enclosure (Klippel 2010)

Air noise is a completely stochastic signal, but it appears in every period at the pressure maximum in the volume so it has a deterministic component too.
Due to the variability of leakage in front or back volumes or in membranes, there is no representative distortion. The temporal and spectral shape of the distortion depends on each application so there is no example shown here.

2.2.5 Loose Particle Hitting Membrane

Often during the production process, loose particles collect in the loudspeaker or in the loudspeaker box. When the loudspeaker is used, these particles hit the membrane randomly and generate distortions. Because of the randomness of this sound it might be very noticeable though the level is very low.
The distortion is a random process with constant output power but very high crest factor. The particles are accelerated by the cone displacement but they are not synchronized with the stimulus signal.

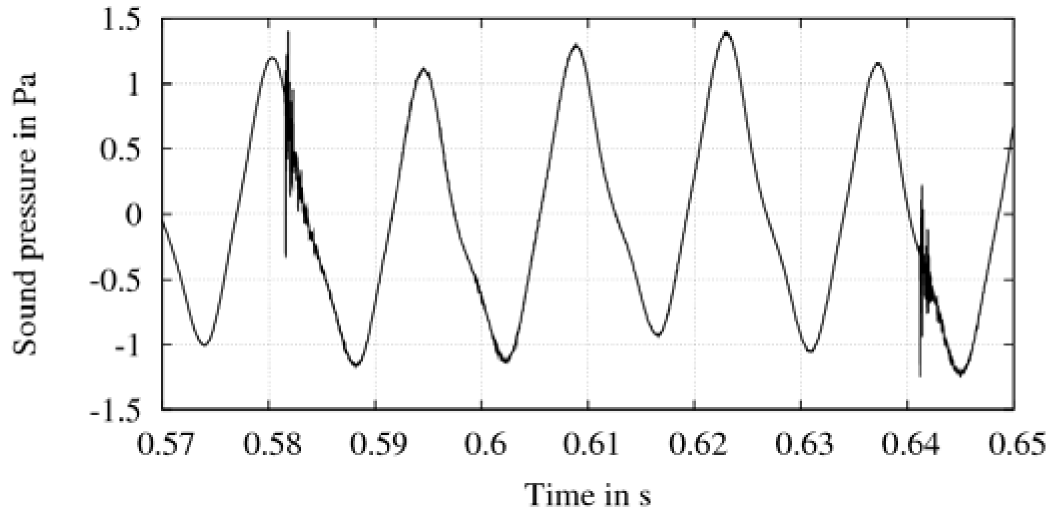

Figure 18: Sound pressure signal with loose particles (Klippel 2010)

2.3 Signal Properties of Distortions

The signal properties of each kind of nonlinearity must be clarified at the beginning of the distortion analysis. This is necessary to detect the distortion and its root cause, and to define parameters for a good indication and classification.
Regular and irregular distortions basically have different properties, consequently signal parameters and analysis methods should be separated to get a clear picture for each type.

2.3.1 Basic Signal Properties

A basic knowledge of statistical signal properties is assumed but some of them, which have high importance in this project, are chosen for detailed descriptions. The classification of the distortion, and especially for the root cause detection an analysis concerning signal properties, is necessary because such parameters can be compared or clustered easily. Otherwise the engineer has to analyze shapes within the time and frequency domain, something which requires a lot of experience and is not possible for absolute classification.

2.3.1.1 Impulsivity

Generally, the crest factor is a very good indicator for the impulsivity of a signal. It is the ratio of maximum and RMS (root mean square) value, consequently it is high for signals with high impulses and low energy elsewhere. Taking the crest factor of the whole sound pressure signal to check the impulsivity of a distortion is not a good approach. The reason is that due to the relatively small amplitude of impulsive distortions compared to the whole output of a loudspeaker, this parameter would be highly dependent on several parameters, such as the type of stimulus, the acquisition length and the input power. Thus three steps are required to get valid information when using the crest factor.

Firstly the crest factor is calculated on a band-pass filtered sound pressure signal, the so-called Rub & Buzz signal. Because of its simple realization, it is common to separate irregular distortions and normal output as described in 3.2.6. The crest factor is independent from the normal output of the loudspeaker and represents the impulsivity of the distortion.

Additionally, it is not very useful to only analyze one parameter for the whole signal because this value can be corrupted by ambient noise at undistorted regions or regular output with a stimulus frequency in the passband of the filter. Therefore this project uses the approach to calculate the crest factor of the distortion for each period of the stimulus separately. This software should work for any kind of input signal, so the challenge of this method is to detect a period of the stimulus without any information about it in advance. Using the common signal processing steps Short Time Fourier Transformation (4.1) and Phase Vocoder (4.4) it is possible to calculate the Crest, RMS and Peak values period-wise which is useful for classification of irregular distortions.

Additionally, to get an independent value of the used measurement system and its noise floor level, a relative crest factor with the average crest factor as reference is calculated.

2.3.1.2 Temporal Dependencies

As summarized in chapter 2.3.3, for the root cause detection it is essential to check the dependencies on the membrane position and its velocity and acceleration. A common form of analysis is to measure the sound pressure and displacement of the membrane simultaneously to make it possible to compare the occurrence of the distortion with the membrane position and movement. But often the membrane position is not measurable because of specialized applications or e.g. insufficient test equipment, so this project provides an estimation of the membrane position using the sound pressure signal and additional information. This approach uses the theory of the acoustic impedance of a circular Piston Diaphragm and is described in section 4.3.

For a better classification and a more user-friendly analysis method, the Stochastic and Deterministic Quadrant Detection are implemented. This analysis method maps either the root of a distortion for the deterministic one, or the energy content for the stochastic to the quadrant of the membrane. This way the engineer can see the dependency of the distortion and membrane position easily in one view. This method is described in chapter 4.4

2.3.1.3 Spectral Dependency / Q-Factor

We can assume a resonating distortion for some defects leaving the quality factor of the distortion visible in appearance over the frequency range. In this case it is assumed that the distortion is a separate system with a damped resonance. Only if it occurs in a very small frequency range it is a sign of a failure with high Q-Factor. The width of stimulus frequencies where the distortion occurs is a similar parameter for defects without a separate oscillation.
A common analysis method for this property is the envelope of the bandpass filtered sound pressure signal. Several methods for the envelope detection have been developed, and for this project I have chosen the method with the highest similarity to the human approach. The envelope is based on the maximum connection of each period and a detailed description can be found in section 4.2. This technique is easily

applied because the period-wise calculation is already provided for other methods such as the instantaneous crest factor.

2.3.2 Regular Distortions

Regular distortions are generally caused by nonlinear parameters that cause a nonlinear transition from electrical power to sound pressure. This means that the output of the system is not directly proportional to its input.

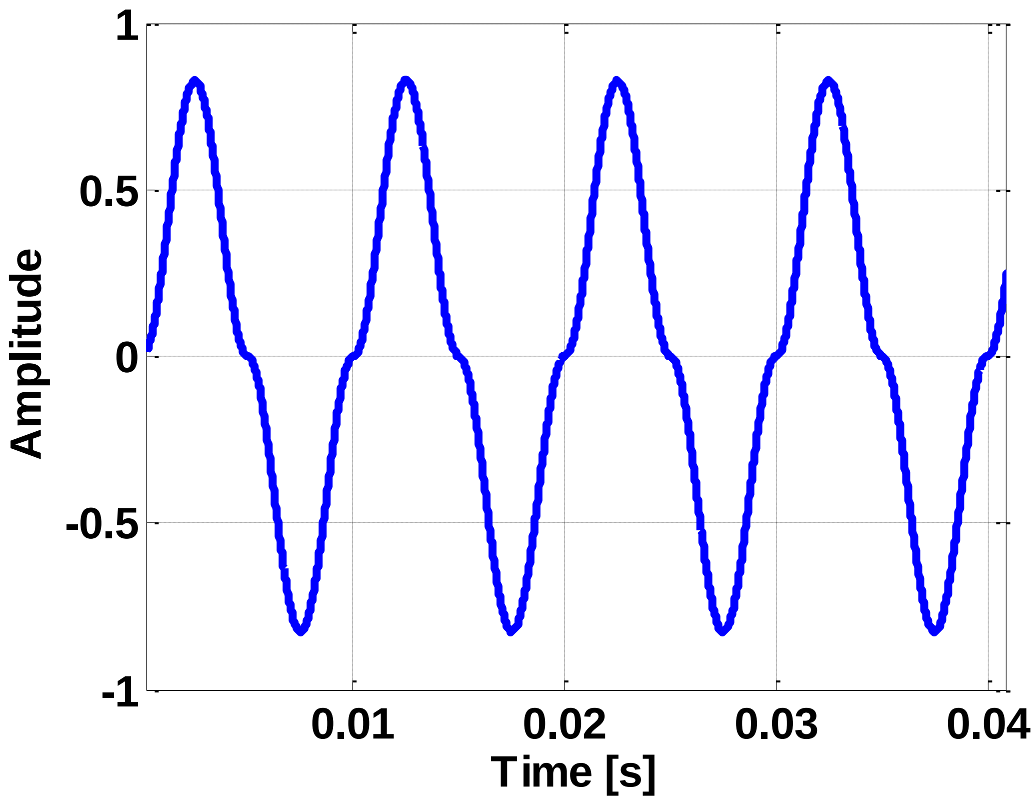

Figure 19: Time Signal of a corrupted sinusoidal

Such an output signal is not purely sinusoidal anymore. The stretching and compression of the signal shape cause harmonic distortions. It is visible in the spectrum with signal contents at multiples of the fundamental, so-called harmonics.

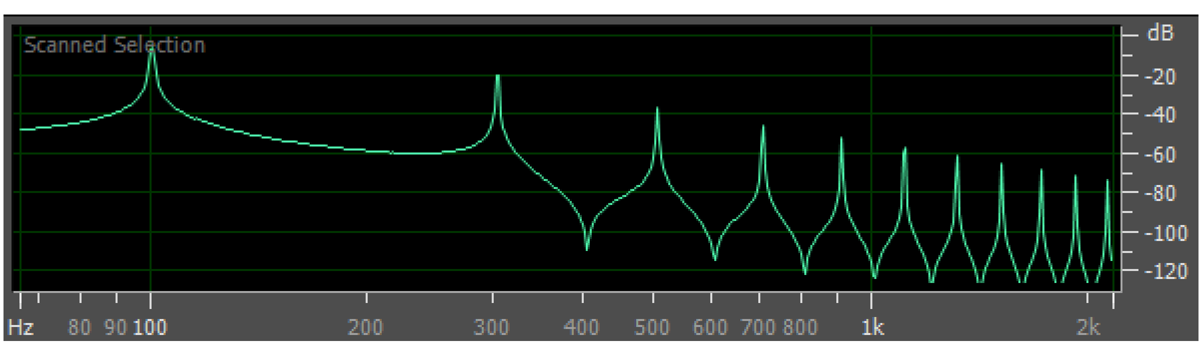

Figure 20: Spectrum of a corrupted sinusoidal

The analysis of the spectrum with consideration of all harmonics is a particularly detailed method and is only necessary for very specialized effects. However, for standard engineering work, the single parameter Total Harmonic Distortion (THD) is common. The THD is principally the ratio of fundamental and all harmonics, thereby two definitions are used that generally have the same approach with a small difference in definition.

The standardized definition in accordance with IEEE is the ratio of all harmonics and the fundamental:

$$THD_{IEEE} = \frac{\sqrt{U^2 - {U_1}^2}}{U_1}$$

U .... Total Signal
U_1 .... Fundamental

On the contrary, the IEC standard defines the ratio of all harmonics and the whole signal:

$$THD_{IEC} = \frac{\sqrt{U^2 - {U_1}^2}}{U}$$

Both methods are common and valid, but there is one important thing which needs to be kept in mind. The IEC THD method has a maximum value of 100%, whereas the IEEE can be infinite.
The THD is a proper method to judge regular distortions and select bad samples, but it is not perfectly suitable to detect the root cause of the distortion. The relationship between the shape of the transfer function and the spectral distribution is often useful for root detection.

A symmetrical transfer function causes high odd-order harmonics and low even-order harmonics, so in the spectrum every 2nd harmonic has high amplitude and the others low. A representative root cause for such a distortion is the variable force factor Bl(x).

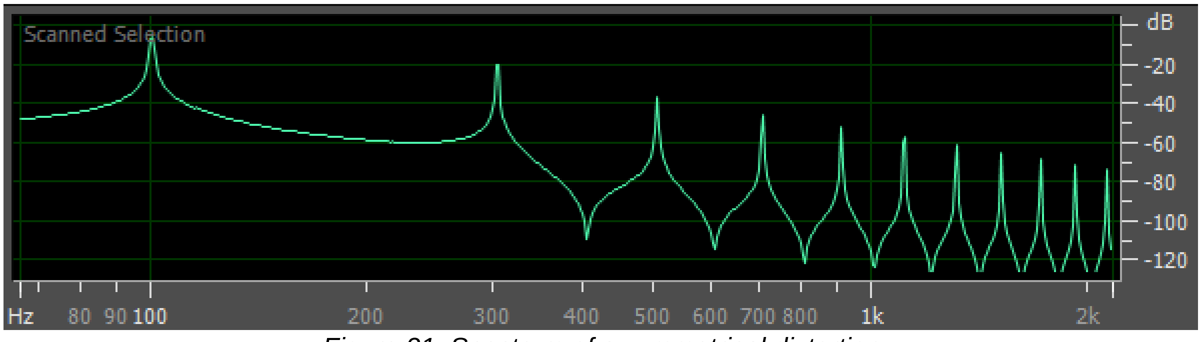

Figure 21: Spectrum of a symmetrical distortion

Asymmetrical systems generate all harmonics, so its spectrum includes even- and odd-order harmonics, normally with a smoothly decreasing shape. This kind of distortion appears if the voice coil is not perfectly centered in the air gap, or if the membrane torus has higher stiffness in one direction than in the other.

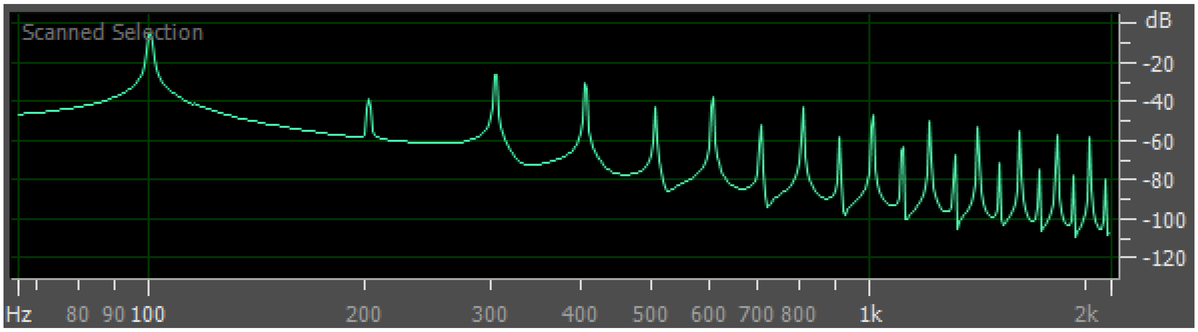

Figure 22: Spectrum of an asymmetrical distortion

Another possibility to detect the root cause of high THD is to check the appearance in the frequency range. Each parameter has a dominating frequency range, so its nonlinearity shows increased THD in this area only. Here is a brief overview of the most common nonlinearities and its dominating areas:

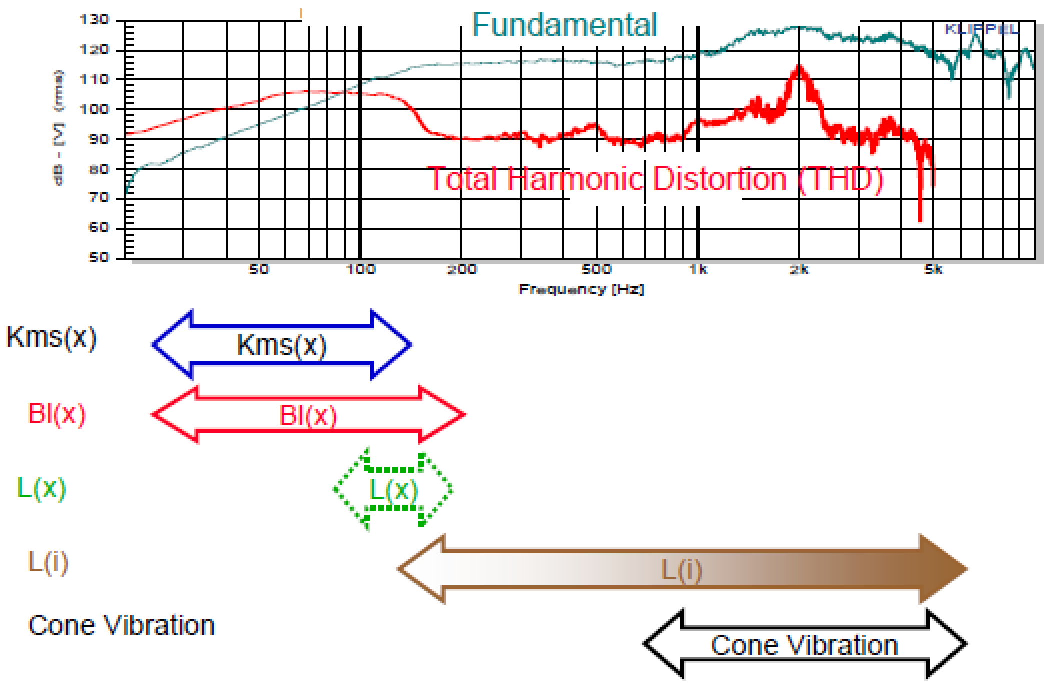

Figure 23: Relationship between dominating root causes and THD (Klippel 2013)

Kms........... Stiffness of the membrane
Bl............... Force factor
L................ Voice coil inductance

2.3.3 Irregular Distortions

Often the engineer analyzes the shape of the time signal and draws some conclusions by optical inspection, but this method needs a lot of experience and know-how and is not useful for automated detection. Therefore a detailed evaluation of the signal properties might be helpful. What follows is a brief introduction of the most important attributes for the acoustic distortion analysis:

2.3.3.1 Rub & Buzz Signal

First of all, for the description of the properties of irregular distortions an explanation of the so-called Rub & Buzz signal is necessary. During the production of big quantities of loudspeakers, quality control needs a fast and easy method for the detection of rubbing and buzzing failures. Most of these failures have prominent signal content in a certain frequency range, so a bandpass filter can separate the distortion from the rest of the signal. Such a bandpass filter is certainly not the best and most accurate way to detect these failures, but because of the low complexity of implementation it has become a proper method for the final quality check during production. This is the reason why this kind of signal is also often used in the development phase to analyze irregular distortions.
In the past Rub & Buzz was a synonym for irregular distortions because most of them are caused by rubbing or buzzing defects. Nowadays it is agreed that this term is not appropriate because irregular distortions can also have different root causes, such as air leakage, but it is still in use. Of course, the correct expression is the split into regular and irregular distortions as introduced by Wolfgang Klippel (Klippel 2013).

2.3.3.2 Stochastic

The time response of a stochastic signal is not describable by a mathematical function. Instead, statistical parameters are used to characterize it. Stochastic properties like power spectral density and probability distribution function are especially helpful for the analysis of such signals. Due to its randomness, trial averaging reduces the power of the signal, thus this method is an easy way to minimize the stochastic components of a signal. The best-known stochastic signal is Gaussian noise which for example is generated in a normal electric resistance caused by the Brownian movement.

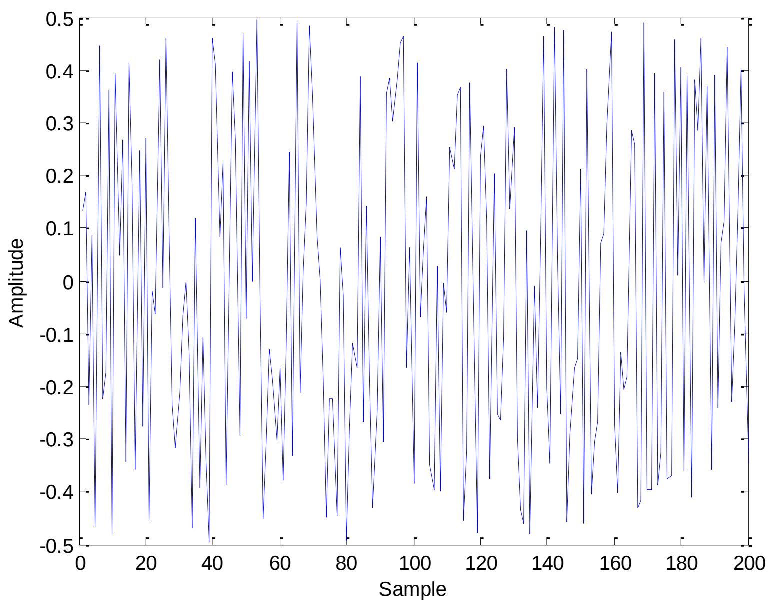

Figure 24: Random time signal

2.3.3.3 Deterministic

A deterministic signal is specified for each time point, consequently it is predictable and easy to classify. For each repetition the signal is exactly the same, so a trial averaging has no effect on it. Of course, measurement signals are never completely deterministic because each measurement device adds noise, but normally the deterministic components can be separated for further analyses. A common deterministic signal is a sine or cosine wave.

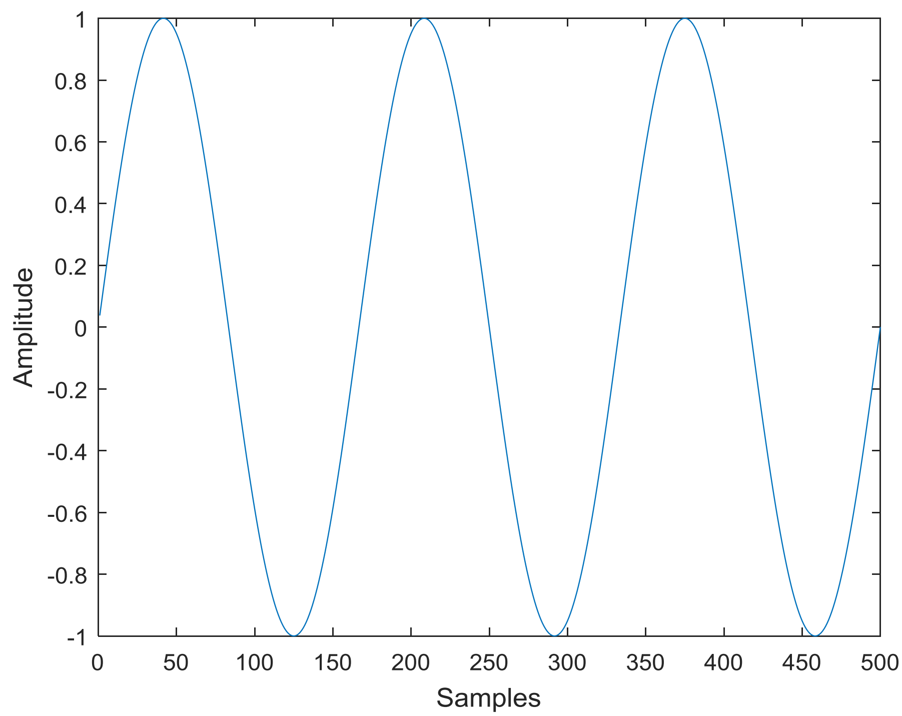

Figure 25: Deterministic time signal (Sinusoidal)

2.3.3.4 Impulsive

The impulsivity is a good indication of the power density in the time domain. If a signal has a very high impulse once and very low value the rest of the time, then it is called impulsive. Otherwise, if the value is almost constant, it is non-impulsive. This classification is helpful for the detection of irregular distortions because some root causes have this characteristic. Additionally, it is important to avoid such distortions because they have very high audibility. Through evolution, the human ear has adapted to detect such audible impulses because in nature, short term peaks are often signs of danger, e.g. an approaching tiger stepping on and breaking a twig. The human ear can detect such sounds although the signal has low energy and is hardly distinguishable in the spectrum.

Some statistical parameters, e.g. average, RMS or crest factor, are good indications for the detection of the impulsivity. The most impulsive signal is the Dirac impulsive which is infinite at a certain time point and zero otherwise.

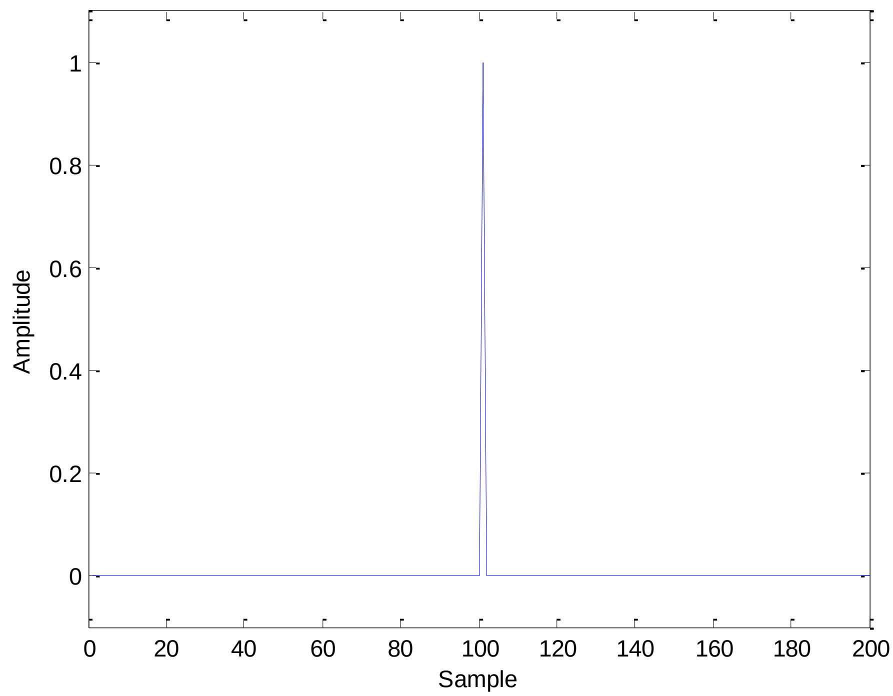

Figure 26: Impulsive time signal (Dirac)

2.3.4 Use for the Root Cause Detection of Irregular Distortions

The signal properties of all kinds of distortions are important for two reasons. On the one hand, some properties are the crucial factors for the audibility of the distortions so it mainly determines the risk for shipping such parts to the customer. On the other hand the properties help a lot in finding the root cause of the distortion. Consequently, the main focus of most of the analysis methods is on determining, qualifying and visualizing the signal properties of the distortion so that the acoustic engineer is able to find the root cause easily, and knows how to repair it or prevent it in future.

2.3.4.1 Filtering the Distortion Content

First of all, it is important to separate the distortion content from the complete sound pressure signal to be able to calculate the statistic properties of the distortion itself. As described in section 2.2, this process is difficult because of the variety of the distortion, but there is one small and simple method which works quite well. During mass production, a filtering method with high computational complexity is not possible because of the limited test time. A simple bandpass filter with constant cut-off frequencies is therefore used to separate the usually high frequency distortion content from the fundamental, including the low order harmonics. The important thing is to find suitable cut-off frequencies for the occurring distortion and application, so sometimes it is necessary to define several filters to identify all distortion types separately. A lower cut-off frequency of 5kHz and an upper one of 20kHz is usually a good choice to detect most of the distortions. But of course, a low cut-off frequency of 5kHz can lead to problems with high frequent stimuli. For example, regular distortions (low order harmonics) of a 1kHz stimulus are located at around 5kHz, so it is included in the bandpass filter, consequently the content is not irregular distortion only. The output of

this filter is generally called Rub & Buzz signal, and the procedure is described in chapter 3.2.5.
Another method to separate the distortion content is to filter the high order harmonics, for example 10th to 15th order. This is a good approach if the distortion is deterministic with strong dependency on the stimulus, but for detecting all defects it is not the best solution.
Some measurement system vendors are developing systems with adaptive filters to separate fundamental, regular and irregular distortions in addition. They build a model of the loudspeaker which estimates the regular distortions. This estimation is subtracted from the complete output signal to catch the irregular distortion including the noise floor. Common micro speakers have very high regular distortion content, so it is difficult to build such a model for them and therefore this thesis had no focus on this approach.

2.3.4.2 Dependency on Excursion

As seen in the description of each failure mode in chapter 2.2, the dependency on excursion is a determining factor. Commonly, power sequences or tests with additional DC offset are used to see the excursion dependency and the critical region (up- or downwards), but in this thesis the failure analysis is also doable in one sweep.
Therefore the position of the membrane in relation to the distortion appearance is important. To get this information without laser measurement, a simplification of the acoustic impedance of the circular piston diaphragm is used to calculate the membrane displacement from the acquired sound pressure signal. This algorithm is described in chapter 4.3.
In addition to this calculation, another issue in the acquisition of the data must be considered in order to achieve valuable dependencies between distortion and membrane position. Due to the propagation time of the acoustic wave, the sound pressure and all other acquired signals are not simultaneous. So if the membrane displacement gets measured by a laser, the delay of the sound pressure signal must be compensated to have all signals in phase. The minimum phase system of the loudspeaker is therefore extracted to compensate for the constant delay of the sound pressure signal. The details of this algorithm are described in section 4.5.
For further studies concerning the appearance of the distortion dependent on certain membrane positions, for example at the positive maximum, another algorithm is developed to split each period of the membrane movement into four quadrants. Root causes of distortions which are related to a specific membrane position can then be sorted into four groups: the top and bottom apex and the zero position for up- and downward movement. The energy content of the distortion and the root of it are thereby separated into these four quadrants to easily see the dependency on the critical membrane positions.

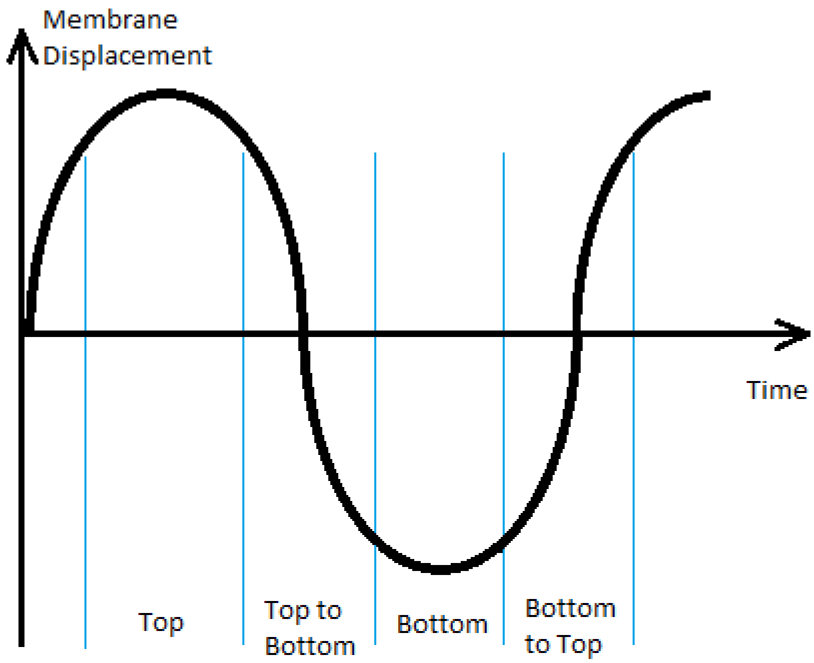

Figure 27: Separation into 4 quadrants of the membrane movement

In this thesis I develop two methods in support of the analysis of the excursion dependency, the so-called Stochastic and the Deterministic Quadrant Detection. The description of the algorithm can be found in chapter 4.4, and the application itself in chapter 5.1.6.

2.3.4.3 Frequency Dependency including Q-factor

The frequency dependency and the Q-factor of the distortion, respectively the width of the appearance in the frequency domain, can be good indications of the root cause. Using this information, it is possible to eliminate some possible root causes: For example, if the distortion occurs above the resonance frequency only consequently when the excursion has no maximum - then it cannot be a hitting membrane. Thus the software program that is to be developed must provide two common analysis methods: the spectrogram and the envelope of the time signal.
The Short Time Fourier Transformation algorithm is used for the spectrogram. Here, the critical parameters such as frame size, overlapping ratio or windowing method can be adjusted by the user to provide high flexibility. Details of the algorithm and the application are described in section 4.1.
The envelope of the time signal is a simplification of the normally huge data amount of a raw time signal and is useful for getting a first impression of the acquired system. For the Universal Sweep Analyzer, several algorithms were tried and the period-wise peak detection was chosen because it is comparable with the visual method by a human. The validation of the algorithms is in chapter 4.2.

2.3.4.4 Deterministic and Stochastic Content

To separate the deterministic and stochastic content of the distortion, time averaging is a useful operation. Re-measuring the loudspeaker several times and averaging the output in the time domain reduces the level of all stochastic components, so the

stochastic content of the distortion can be determined. Most of the data acquisition systems are able to perform this operation automatically so there is no need to implement it in the analysis tool of this master thesis, the Universal Sweep Analyzer.
Another option is the analysis of the crest factor of the so-called Rub & Buzz signal. A period-wise calculation of the crest-factor of this signal regarding the stimulus period is a good indication of the impulsivity and other statistical properties of the distortion. The period-wise calculation eases the inspection of the temporal trend of the impulsivity and additionally it simplifies the calculation because of the defined calculation frame length. Brief descriptions are in chapters 3.2.5 and 3.2.6.

2.3.4.5 Summary

Commonly, a loudspeaker is classified as “Rub & Buzz” but the root cause of the irregular distortion is unknown. For the failure analysis, the acoustic engineer can apply several different methods to reach a conclusion, so here is a short summary of them to support a structured method of checking all possible root causes:

Firstly, the dependency on excursion is checked because this is a main parameter of irregular distortion. Usually an acoustic measurement with a frequency sweep is used to test the loudspeaker during production. This sweep can be used to check the Rub & Buzz signal or the HOHD (high order harmonic distortion) and their occurrence in the frequency range to discover at which stimulus frequency the distortion appears. For example, if the output is distorted at low frequencies only then it is a sign that there is a relationship with the excursion. Using this knowledge, dependencies on velocity and acceleration can also be seen. For this analysis, the temporal envelope over a frequency axis is a useful view because the distorted frequencies can be seen easily. Additionally, the Q-factor (the bandwidth) of the distortion is shown clearly in this plot.
Rub & Buzz vs. power sequences or DC studies are other methods for checking the dependency on excursion. Using a power sequence shows if there is a critical amplitude where the distortion starts to exist, and the DC study can determine if the membrane hits at the positive or negative movement.
For the ability to see the critical position of the membrane with one sweep only, the Universal Sweep Analyzer has the Quadrant Detection implemented. This analysis separates the distortion appearance into four quadrants so the engineer can get the membrane displacement dependency with the standard acoustic test.

The spectral distribution of the distortion is a key parameter for the root cause detection. Using the spectrogram of the whole signal or the spectrum of a single time frame offers a visual inspection of the sound pressure signal in the frequency domain. This is also important for setting the Rub & Buzz filter’s cut-off frequencies to get the complete distortion content included.

Some defects generate extremely impulsive distortion so they have high audibility although the energy is low. It is therefore necessary to also check the impulsivity of the Rub & Buzz signal. The Universal Sweep Analyzer includes a stimulus-period-wise calculation of the Rub & Buzz signals’ Crest factor which is a good parameter for the impulsivity.

To split the stochastic and deterministic content of the distortion, time averaging of repeated measurements is a common method because the stochastic distortion is reduced with each repetition. Another method is the use of the deterministic and stochastic Quadrant Detection of the Universal Sweep Analyzer. This procedure uses the instantaneous crest factor to detect a deterministic distortion and estimates the origin of the distortion in each period of the stimulus.

The following table summarizes these key parameters in the root cause detection and gives a brief overview. The details of each analysis method are explained in chapter 3.

Root Cause	Dependency on Excursion	Dependency on Velocity	Dependency on Acceleration	Deterministic	Stochastic	Impulsive	Q-factor
Hitting Voice Coil	Directly	Indirectly	Indirectly	Yes	No	Yes	Low
Buzzing loose Joint	Conditionally	No	No	Yes	No	Mixture	High
Rubbing Voice Coil	Indirectly	Indirectly	Directly	Yes	No	Mixture	Medium
Flow Noise	No	Directly	Indirectly	Little	Yes	No	Low
Loose Particle	No	Little	No	No	Yes	Yes	Random

Table 1: Overview of all defects and its signal properties

Legend:

Directly............. Is caused by this domain
Indirectly .... Is not caused by this domain but there is a relationship due to the derivation
Conditionally .... There is no real dependency but it only occurs above a certain threshold
Deterministic.... Does it include a deterministic component which can be reproduced several times?
Stochastic .... Does it include stochastic components which can be minimized with time averaging of re-measurements?
Q-factor............ The Q-factor of the resonating distortion or for non-oscillating defects it represents the width of stimulus frequencies where the distortion occurs, the so-called bandwidth of the distortion

Table 1 is a good overview for the analysis of a loudspeaker with irregular distortions. It helps to eliminate possible root causes and leads to further failure analyses. With this knowledge it should be possible to distinguish the root cause of the distortion. To describe the loudspeaker behavior it is necessary to acquire signals like voice coil current and sound pressure. For this procedure several methods are developed – the optimum depends on the application. In the following chapter 3 there is a summary of the most common methods including some mechanical analyses which are also helpful for the root cause detection of a failure.

3 Analysis Method for finding Rub & Buzz root causes

In the previous chapter the most common kinds of effects are shown which can generate loudspeaker distortions. Knowing the characteristic signal properties of the distortion type helps for the root cause detection if all needed signals are available. Therefore, several methods have been developed to detect all kinds of defects and a summary is shown in this chapter.

Generally, these methods can be split into mechanical and acoustical procedures. The mechanical analysis focuses on the inspection of the critical dimensions in the rest position and the asymmetries during movement. The acoustic analysis requires a test where the loudspeaker is driven with a frequency sweep or any other proper signal where the distortion occurs, and a microphone which records the output.

3.1 Mechanical Analysis Methods

The first failure analysis method after detecting Rub & Buzz is often to have a look at the mechanics of the speaker so obvious defects like limited excursion space, holes or loose elements can be detected. Here is a brief summary of all common methods including advantages and limitations.

3.1.1 X-Ray

Often loudspeakers are used in closed boxes so it is difficult to see critical dimensions like excursion clearance, symmetry of the coil in the air gap or the tilt of the membrane. X-ray makes these parameters visible without the need for disassembly.

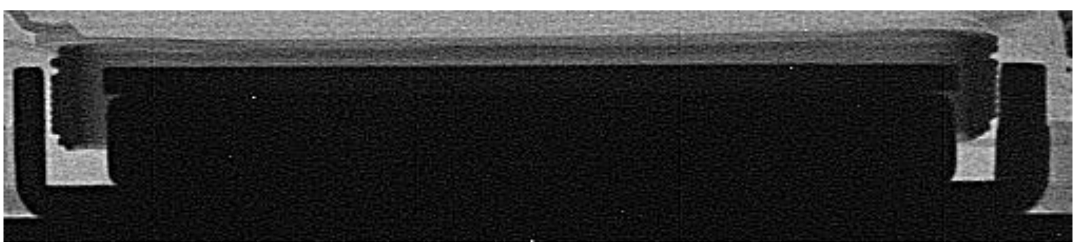

Figure 28: X-ray of a micro-speaker in xz-plane

This view shows if the membrane position is too high or too low and if it is tilted and gives a good indication of a hitting membrane resulting from limited excursion space or rocking mode.

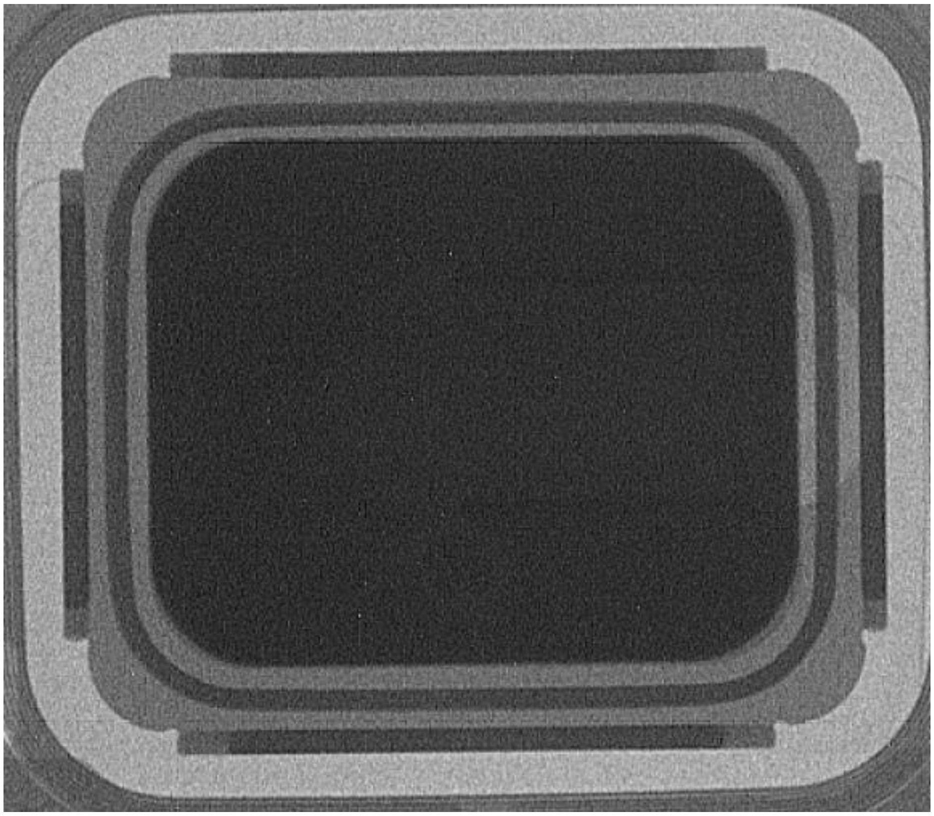

Figure 29: X-ray of a micro-speaker in xy-plane with centric buildup

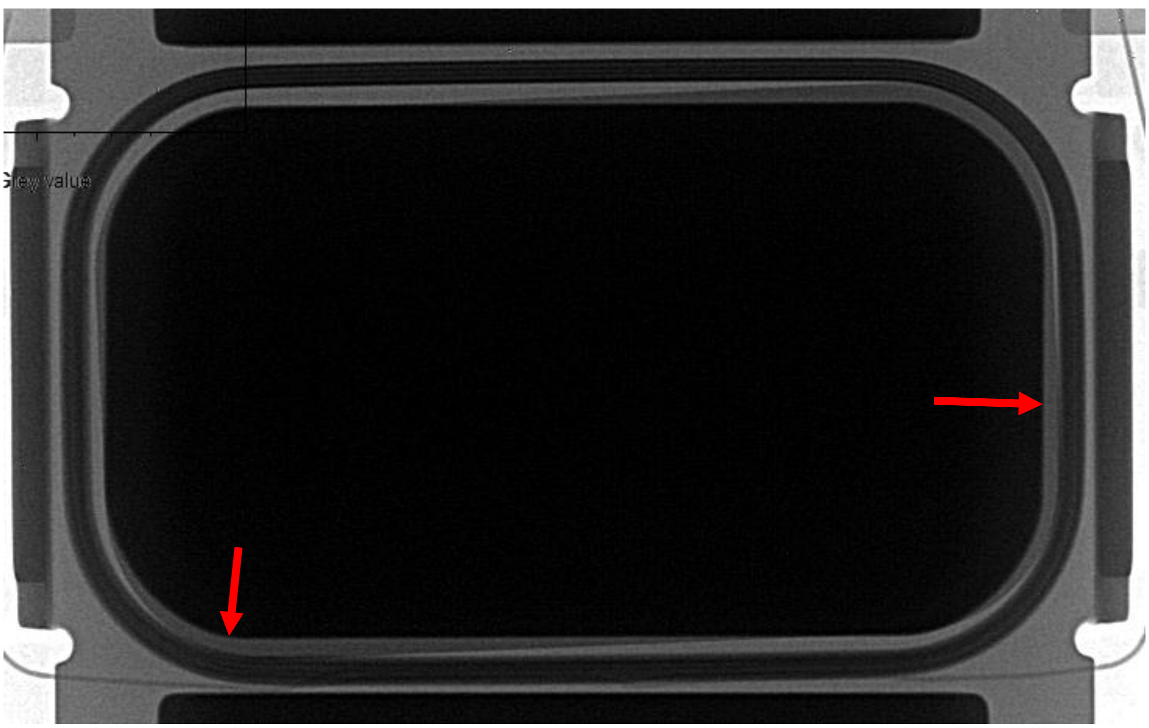

Figure 30: X-ray of a micro-speaker in xy-plane with eccentric top plate on the magnet

In Figure 29 a micro-speaker is shown with centric buildup, so the gap between coil and magnet or pot is quite constant and the risk of a rubbing voice coil is relatively small. In Figure 30 there is a micro-speaker with an eccentric mounted top plate on the magnet.

This leads to a touching voice coil at the top plate consequently to a rubbing voice coil in use.

Sometimes X-ray pictures can also be used to see loose elements such as poor positioning of the wire loop. But the parts must have sufficiently large dimensions to be visible in an X-ray.
This analysis method is helpful because it is a quick way of checking the mechanical dimensions, but generally it is hard to measure distances because of the small focus points. Some applications are also not suitable for X-ray inspection because of their metal housings.

3.1.2 Cross-Section

A particularly detailed way to check the mechanics is to do a cross-section of the loudspeaker and measure all important dimensions with a microscope. Through this method, the loudspeaker is put in a box and filled with fluid epoxy resin. After the glue has set, the complete loudspeaker can be ground down to the surface of the section in question.

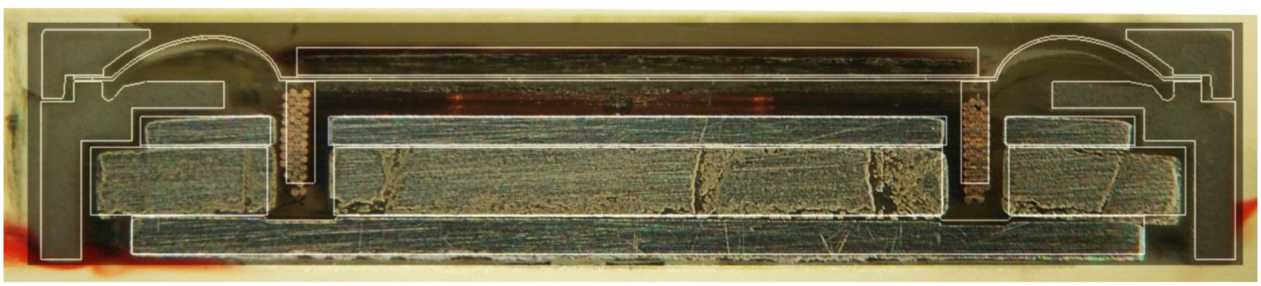

Figure 31:Cross-Section of a micro-speaker compared to the design

The advantage compared to X-ray is the possibility to analyze a certain section. The X-ray visualizes the reflection of the complete loudspeaker, while the area of the cross-section can be select according to the grinding depth.

3.1.3 Disassembly

Disassembling the loudspeaker or the application is a common way to detect some kinds of defects. Loose particles, holes and obvious wrong positions can be detected quickly, but normally it is quite difficult to measure certain distances or asymmetries. After finding the defect it is often possible to repair the loudspeaker, e.g. a loose particle can be removed or a bad wire loop position can be adjusted. Afterwards, the loudspeaker can be re-measured acoustically to see if the modification was able to eliminate the distortion. This method is useful to get proof of the root cause distortion.

3.1.4 Stroboscope Analysis

Because of the high frequency range in acoustics, it is not possible to visually detect the movement of the membrane. A stroboscope can be used to make the high frequencies visible to the human eye. This method benefits from the Nyquist theorem which describes an insufficient sampling of a high frequency wave. In other words, the stroboscope lights the membranes once in a period but the duration between two

pictures is slightly longer than the stimulus period, thus the pictures have developing phase. The human eye interpolates the pictures to a smooth motion which represents the fast movement on a slow time axis.
This method is useful in detecting rocking modes and tumbling because every movement against the main direction can easily be seen. A major advantage is that as a result of the decelerated visualization of the fast movement, the engineer gets a good impression of the asymmetry, and often this is helpful in finding the root cause of the rocking mode.

3.2 Acoustical Analysis Methods

Many defects are not visible by mechanical inspection and therefore acoustic methods must be applied. For the detection and classification of regular and irregular loudspeaker distortions, several acoustic analysis methods that require different signal processing steps are common. A standard test setup is described in chapter 3.2.1 to give an overview of the signals that are desirable, and their limitations. This kind of measurement is necessary to get the raw signals for all further analysis steps listed afterwards.

3.2.1 Measurement Setup

To be able to describe the electrical, mechanical and acoustical behavior of the loudspeaker completely, a detailed acoustic measurement is needed. For this test the following set of signals is required:

- Terminal voltage
- Current in the voice coil
- Excursion of the membrane
- Sound pressure at a known distance and position of the loudspeaker

Using the terminal voltage and the current in the voice coil, the parameters of the electrical model of the loudspeaker can be distinguished. The excursion signal can represent the mechanical behavior, and the sound pressure helps model the transformation from the mechanical to the acoustical domain. Thus the complete set of parameters of the loudspeaker model can be reconstructed and all defects can be determined with respect to all limitations of the model. Often loudspeakers are used in closed boxes so it is not possible to measure the excursion of the membrane, but for some failures it is very important to get information about the membrane position, and so an algorithm is developed to estimate the membrane displacement from the sound pressure. Consequently, the acquisition of the sound pressure in a known distance and position is sufficient. A detailed description of the algorithm is in chapter 4.3.

The following measurement setup was established based on a defined environment with low disturbances and few variable influences.

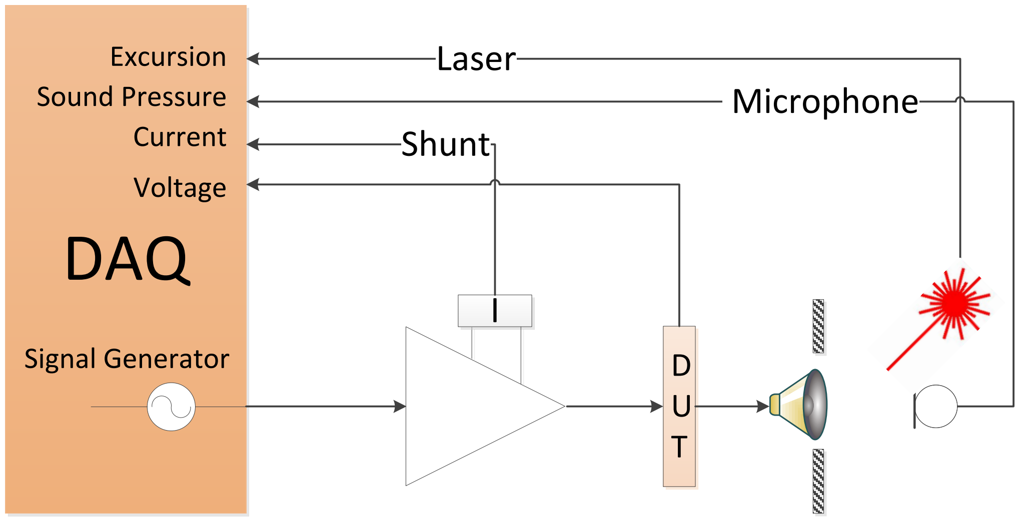

Figure 32: Measurement Setup with Baffle and Laser measurement

An infinite baffle guarantees that the loudspeaker only radiates in the half-plane. In reality, baffles with finite dimensions are used, but for ease of illustration one-sided radiation is assumed. This assumption simplifies further calculations a lot though the valid frequency range needs to be considered.
The microphone position should be in the near-field so that the acoustic impedance can be modeled in a simple way. But if the microphone is close to the loudspeaker, reflections from the microphone can disturb the half space condition and consequently the simple model would not be valid anymore.
Generally, all of the measurement equipment must have a linear behavior, so a correct measurement of the interesting parameters without any distortions can be guaranteed, otherwise nonlinear effects of the measurement chain can be misinterpreted as defects of the loudspeaker.

3.2.1.1 Test Signal

To ensure that all parameters of the loudspeaker model are accurate, high SNR (signal to noise ratio) in the whole considered frequency range is needed, so a signal must be chosen that guarantees sufficient signal components at these frequencies. Because of defects resulting in high Q-factor resonators the stimulus must have high frequency resolution. A continuous sweep with high energy fulfills all these requirements and is the optimal solution. For better SNR at low frequencies a sweep with logarithmic frequency distribution is chosen. Start and end frequency of the sweep should be outside of the frequency range of interest, and the transients at the beginning and ending must be considered.

3.2.2 Listening Test

The easiest way to analyze loudspeakers is the listening test. A trained person is able to detect irregular distortions already at low level, but the problem is that the test person’s ears get tired and so the measurement cycle time is limited. Additionally a human test needs several seconds for a decision and that is too much time for a test at an

automatized production. Another problem is the detection of regular nonlinear distortions, because the human's ears are not very sensible on this kind of distortions.
Summing up, the listening test is a quick proof for the rough detection of irregular distortions, but is not recommended for the detection of regular distortions and for the quality control in mass production.

3.2.3 Spectrogram

Another brief analysis method for the detection of all kinds of loudspeaker distortions is the spectrogram. It combines the time and frequency domain in one plot and uses the Short-Time-Fourier-Transformation (STFT) for the calculation. The STFT splits the signal into several time frames that can be overlapped to get a better time resolution. These frames are transformed in the frequency domain using the Fast-Fourier-Transformation (FFT) algorithm. A 3-D diagram visualizes the temporal trend of the spectral density of the amplitude.

Below is a spectrogram with regular nonlinear distortions visible in lines with increased amplitude that are parallel to the fundamental.

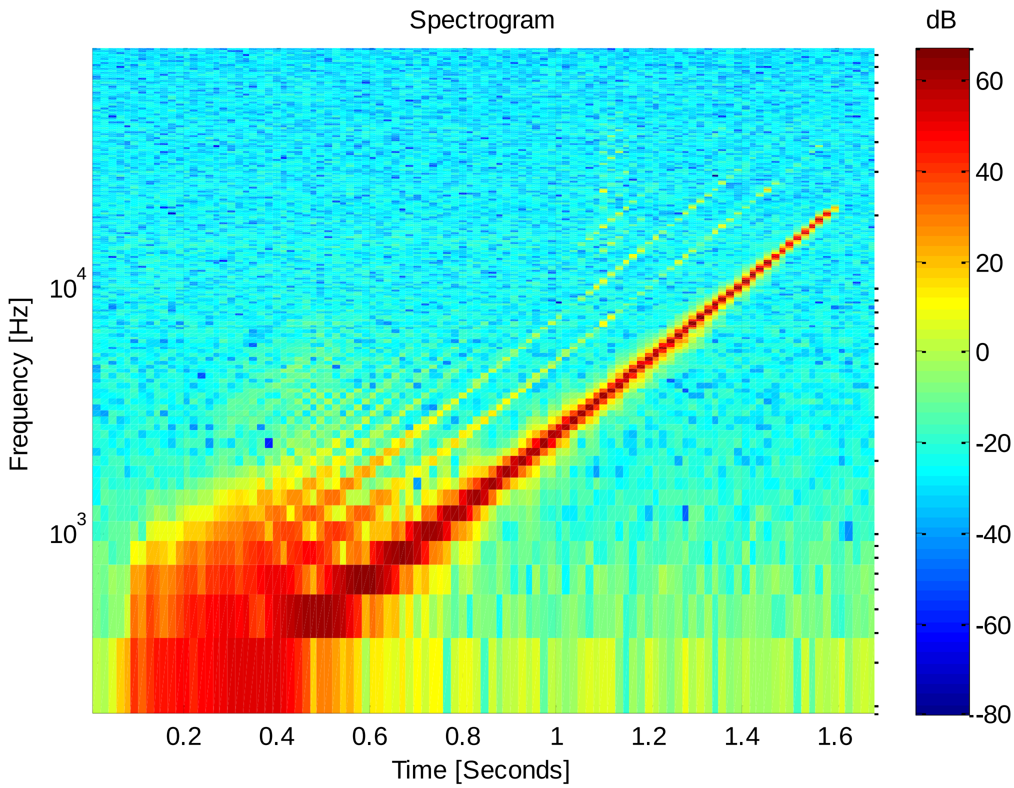

Figure 33: Spectrogram of a sample with high harmonic distortions with a sweep excitation

Irregular distortions have own oscillations, stochastic components or high order harmonics, so the shape in the spectrogram is different to regular distortions. In this plot, a hitting membrane is shown so the distortion has strong high order harmonics with impulsive characteristic. Other defects can look completely different in the spectrogram but the representative plots are shown in chapter 0.

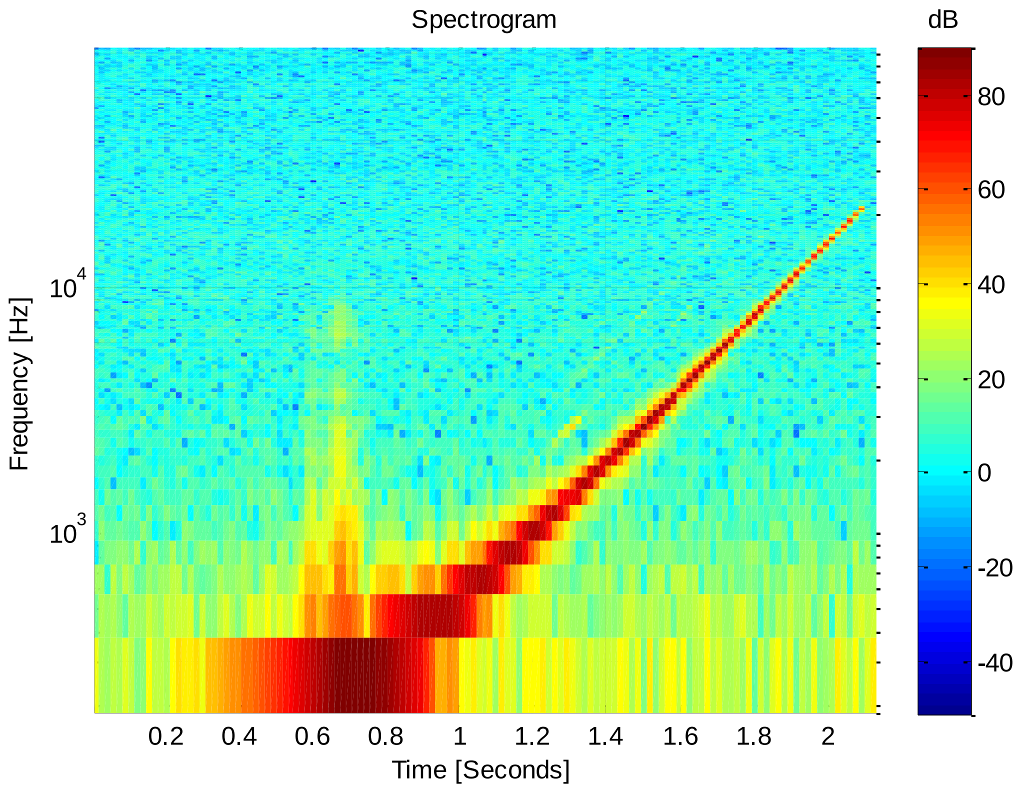

Figure 34: Spectrogram of a sample with R&B with sweep excitation

3.2.4 THD

The analysis of total harmonic distortion has become an accepted standard in loudspeaker testing. There are two common definitions which are described in 2.3.2.
For the calculation it is necessary to filter the fundamental and all harmonics with notch filters with small bandwidths to get the power of the stimulus and the harmonics. For a signal with varying frequency, this operation might be difficult to implement, but there are many measurement systems with good and fast implementations at a fair price available.
The problem with the analysis using THD is the poor usability for irregular defects, like Rub & Buzz. Because of the independence on the fundamental frequency, the distortion is normally not in the bandwidth of a harmonic filter, so the THD value is still small although Rub & Buzz exists.

3.2.5 Constant Band Pass Filter / Rub & Buzz Signal

As described above, the THD is a good parameter for regular distortions but it is less suitable to detect or identify irregular defects. Rub & Buzz normally has signal content at high frequencies independent from the stimulus signal, thus it is a non-harmonic distortion. A band pass filter with constant center frequency and bandwidth is a good solution. For an optimum result, the harmonic content should not be considered in the Rub & Buzz analysis and should be filtered out, but it is often neglected due to the low energy of the high order harmonics.
The challenge is to find good filter settings to keep the main part of the irregular distortion. High order filters have sharper separation in the frequency domain so the

separation of the irregular distortions is easier, but because of the long impulse response the temporal occurrence of the distortion is corrupted. A description of the filter setting can be found in chapter 2.3.4.1

Usually the envelope of the RMS value is analyzed and is called Rub & Buzz Signal because it represents the energy content of a distortion, but additional operations, such as calculating the instantaneous crest factor, is also helpful to detect the cause of the distortion.

3.2.6 Instantaneous Crest Factor

Another statistical parameter for a detailed analysis is the instantaneous crest factor. The crest factor is the ratio of peak value to RMS value, and this operation is performed on the filtered signal. Normally, a constant band pass filter is used, and the crest factor is calculated separately for each period of the fundamental. This is a particularly helpful indicator to separate deterministic and stochastic distortions. Deterministic errors, like a hitting voice coil, occur every period at the same phase with relative high peak value, but the signal energy of the distortions is generally small, so the crest factor at the distortion is much higher than at correct application. Additionally, because of the period-wise calculation, defects with high Q-factors are clearly visible and reasons identifiable. To get a crest factor which is independent from the measurement system in use, the value is referenced to the mean value and is interpreted in dB.

3.2.7 Trial Averaging

As explained in chapter 3.2.6, the instantaneous crest factor helps to separate errors between deterministic and stochastic ones, and trial averaging is another method. Modern measurement systems have the option to do several measurements consecutively and average them in the time domain. Exact synchronization of the signals is necessary because otherwise some frequency parts are eliminated due to phase errors and the result would be useless. Several measurement equipment vendors have focused on this issue and provide useful applications to automate this procedure.

Stochastic parts of the signal are, because of their random phase, reduced by time domain averaging. On the other hand, deterministic signals are still unmodified by this operation. Thus, trial averaging splits deterministic and stochastic distortions and helps classify the defect.

Averaging several measurements also improves the SNR, something which can be helpful for noisy environments or small measurement levels.

3.2.8 DC-Offset Measurement

In the case of a hitting part due to high excursion the question is whether the distortion occurs in upwards or downwards displacement. To find the answer, a very simple approach using DC-power is often used. The approach is to add positive or negative DC voltage to the original stimulus signal. If the distortion is caused by a too small clearance in one direction then the Rub & Buzz must disappear after adding a constant component to the original signal. This method normally shows the critical displacement direction clearly.

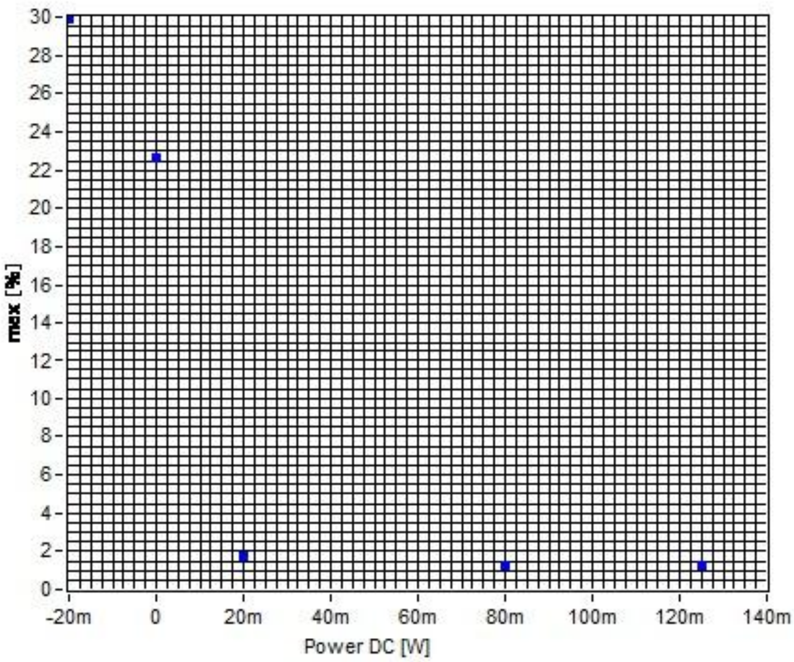

Figure 35: HOHD [%] vs. the DC power which is additionally applied on the frequency sweep

Without a permanent DC shift of the membrane there is high HOHD which represents Rub & Buzz, but after applying a small offset in the positive direction it is gone. This indicates that the membrane hits because of too little downwards clearance.
With this method the engineer has to keep in mind that because of the additional power, the loudspeaker gets more heated, and due to the temperature dependent properties, e.g. membrane stiffness or voice coil resistance, the excursion can be lower. Consequently, it is possible that the disappearance of the distortion is only because of higher temperature and lower excursion.

3.2.9 Rub & Buzz vs. Power

Another analysis method for a hitting membrane is to vary the AC power of the stimulus. The result shows the maximum power, consequently the maximum excursion, which can be applied on the loudspeaker with Rub & Buzz. This approach is useful to check the maximum excursion and input power for a given loudspeaker design. It can also be used to see how safe the membrane displacement with given application and stimulus is.

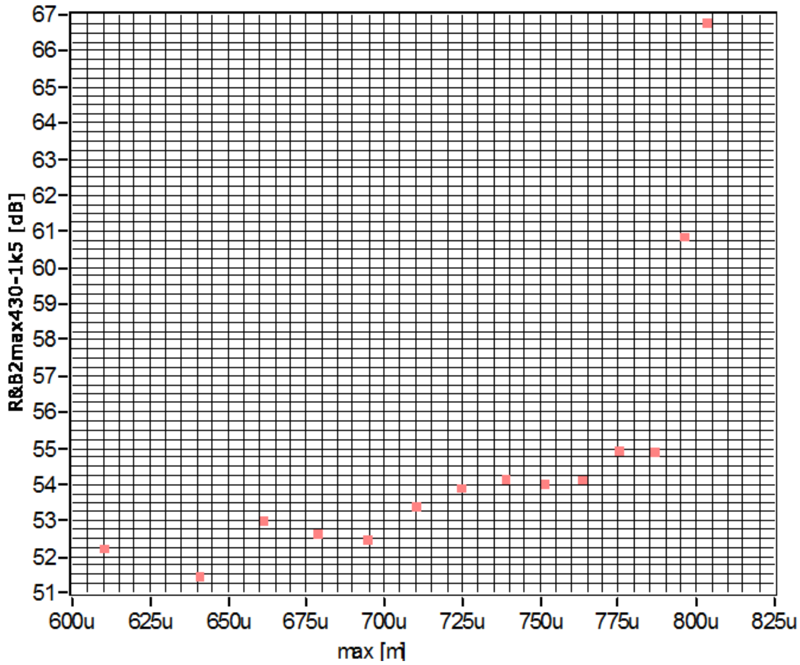

Figure 36: Rub & Buzz maximum [dBSPL] versus maximum peak-to-peak excursion [m]

Chapter 3 shows all common analysis methods which are useful to detect the root cause of irregular distortions. As mentioned in the introduction, there are several measurement systems available which provide these analyses methods. All have its advantages and disadvantages so it is not convenient to use only one system for all applications. In this master thesis a new software program called Universal Sweep Analyzer is developed to combine the advantages of all systems. The development of this tool requires several signal analysis procedures which are described in the following chapter 4.

4 Signal Analysis Procedures

As described in the previous chapter 3 there are many analysis methods available to detect distortions, but it is hard to combine all in one tool. Generally, one specific measurement device has a particular advantage but is not possible to add another function. So the engineer has to perform more than one measurement. However, all test systems start with the same step: the acquisition of the time signals. It must be possible to do all analysis steps with a single acquired time signal. For this procedure, some signal processing steps are necessary. Most of them are familiar, but may not have been used for this application. Short descriptions of the key functions are listed in this chapter. The details of the implementation into one analysis tool are explained in chapter 5.

4.1 Short Time Fourier Transformation

The spectral distribution of a signal is often important for a detailed analysis and the Discrete Fourier Transform (DFT) is a good choice to show it. The DFT is a transformation using the Fourier Analysis, which transforms a time discrete, finite and periodic signal in a discrete, periodic frequency spectrum. Therefore a number of samples of the signal are taken and the assumption is that this extract is periodic.

$$X_k = \sum_{n=0}^{N-1} x_n * e^{\frac{-j*2*\pi*k*n}{N}}$$

x_n .... sampled time signal
X_k .... Discrete Fourier Transform
n .... time index
k .... frequency bin
N .... period of the time signal

Because of the high computational complexity of this operation, a faster calculation algorithm is commonly used. The Fast Fourier Transform (FFT) is a specialized method of the DFT and is optimized for digital signal processors. The algorithm requires signals with a length equal to a power of 2.

$$N = 2^l$$

l .... natural number

One important property of the DFT is the periodicity criterion which assumes that all signals are periodic. Thus, only frequencies with periods of the frame length, or multiples of the fundamental frequency, are correctly represented in the DFT. But acquired signals are usually a mixture of signals with all frequencies. The values at the beginning and at the end are as a consequence not the same. This problem is called leakage-effect and can corrupt the spectrum appreciably. To reduce the impact of this issue, windowing functions are used. A scaling factor for each sample is multiplied with the origin in such a way that the signal takes course to zero at the borders. In this way, the discontinuities at the beginning and end are eliminated. But of course, the signal is changed as a result of this multiplication and a new signal is analyzed. The challenge is to find a good tradeoff between discontinuities and modification.

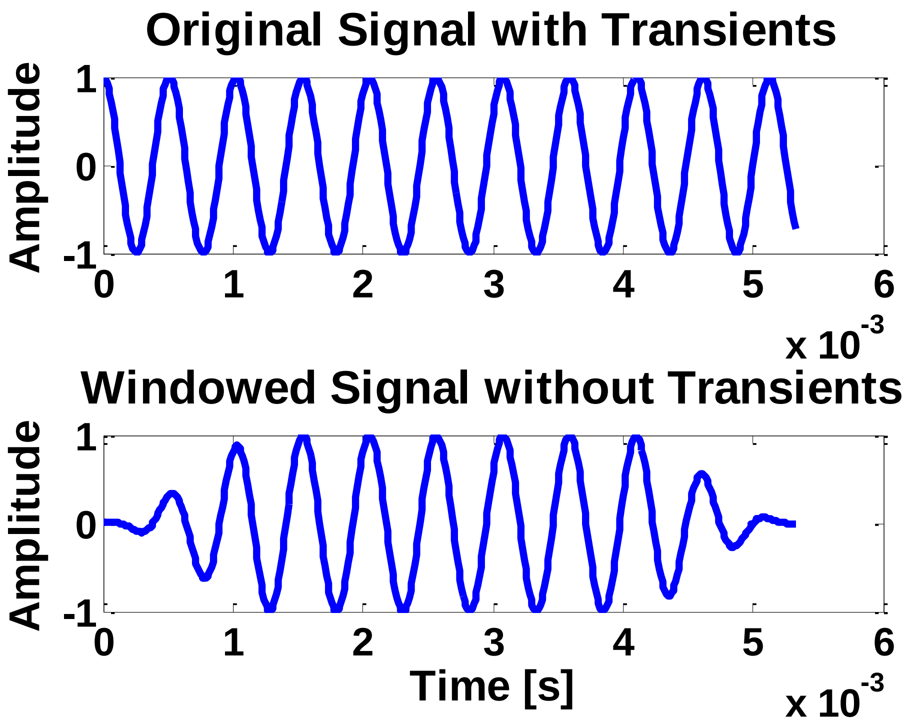

Figure 37: Signal with and without window function to avoid transients at the window borders

The critical point of the Fourier transform is the assumption of a stationary and periodic time signal. This generalization is problematic for common measurement signals due to its stochastic component and time variability. Especially for the analysis of loudspeaker distortions, the temporal trend of the spectral distribution has very high importance, and a Fourier Transform of the whole time signal does not display these phenomena. A visualization of the spectral components changing over time is necessary. The Short Time Fourier Transform (STFT) splits the complete time signal into several frames and transforms it to the frequency domain. Due to the separation in the time domain, the assumption of the stationarity must be valid only for a short time and the variability is still available.

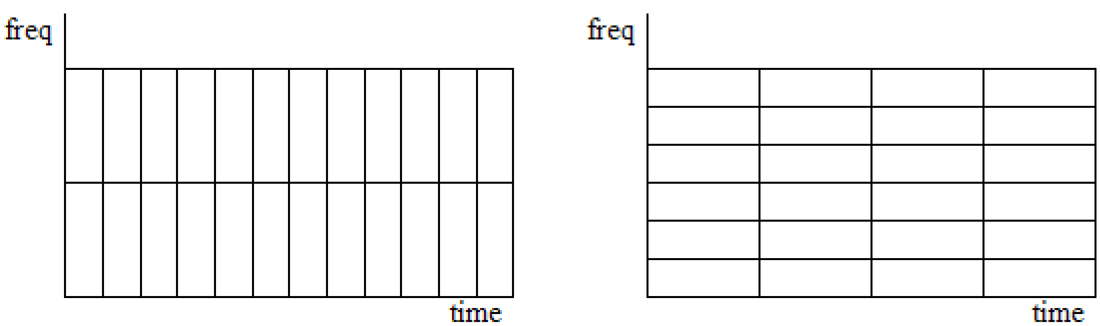

Figure 38: Time and Frequency Resolution: good time resolution (left) and good frequency resolution (right) (Short-time Fourier transform 2016)

The challenge of the STFT is to find the optimal window length. A long window improves the frequency resolution because it is also possible to transform low frequencies, but the time resolution is bad. On the other hand, a short window displays temporal development very well, but then the frequency resolution is bad. The uncertainty principle describes this behavior and defines the critical resolution.

$$\Delta t * \Delta f \geq 1$$

Δt..................... frame length (time resolution)
Δf..................... frequency resolution

To improve the time resolution, the overlapping method is often applied. The time segments do not start in succession, but the subsequent frame start within a time area which is also analyzed by the previous frame. The computational complexity and the memory resources are increased, but more details in the temporal development are visible.

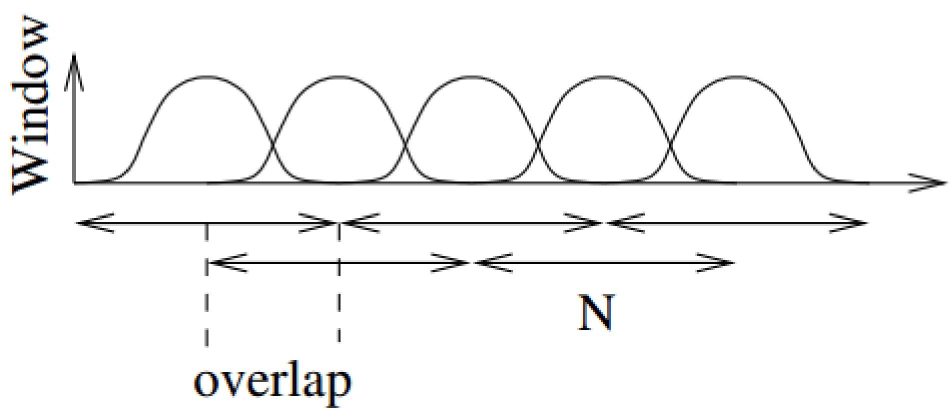

Figure 39: STFT with overlapping method to increase the time resolution (Heinzel, Rüdiger und Schilling 2002)

The use of STFT for the analysis of loudspeaker distortions generally needs good time and frequency resolution, but this is only possible simultaneously to a certain extent, as mentioned before. So it is a good approach to do several STFT analyses with different parameters. To find good filter parameters for the detection of irregular distortions, it is important to reconstruct the spectral envelope of the distortion exactly and a long window with high frequency resolution is a good method. But to detect defects with high Q-factors it is necessary to analyze using high time resolution because the distortions appears only briefly, and the exact detection of the critical stimulus frequency is essential. Because of this, an acoustic engineer needs a STFT with variable settings for the root cause analysis of distortions so he is able to separate the analysis in a detailed time and frequency view.

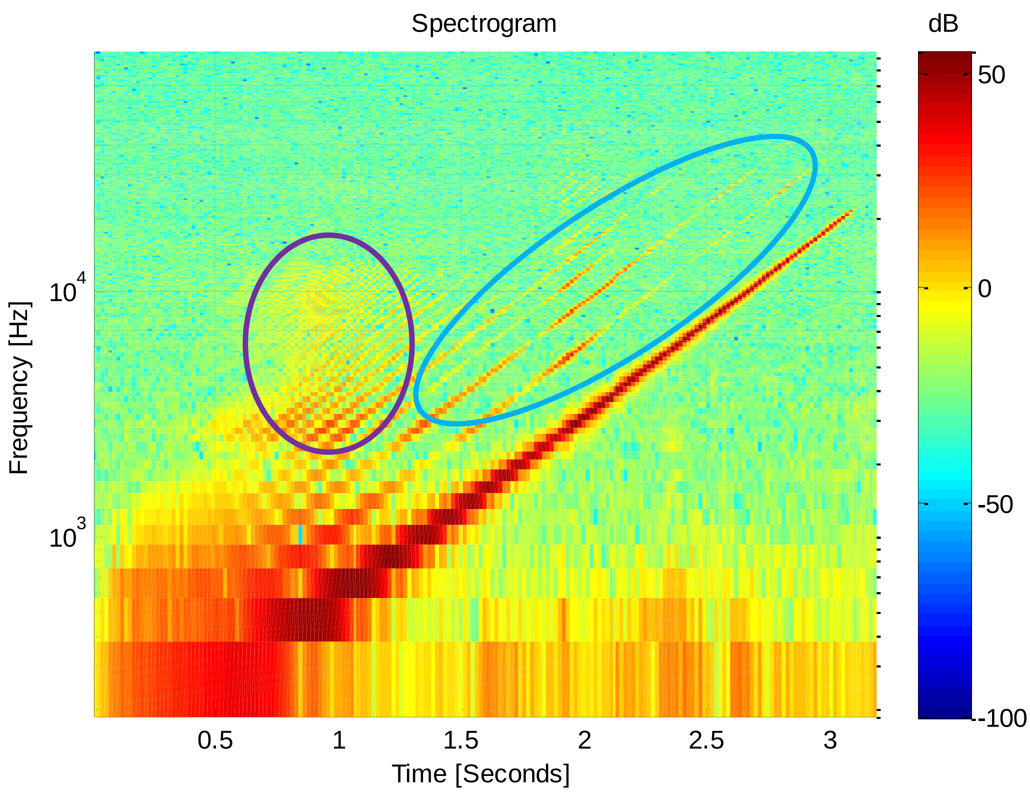

*Figure 40: STFT of a micro-speaker with **regular** and **irregular** distortions*

4.2 Envelope

The first step in checking the acoustic output of a loudspeaker is to have a look at a time signal of a microphone. This signal generally includes all information about the system, but due to the huge data amount it is rarely helpful for the acoustic engineer. The envelope of a time signal is a simplified representation where only maximum and/or minimum of each period is tracked. This approach gives a simpler overview because the redundant information is eliminated making it easier to focus on important issues, such as the temporal trend, frequency dependency or local peaks and dips.
Normally, the axis of abscissa for the envelope diagram is the time domain because the envelope is calculated with a time domain signal, but for the analysis of loudspeakers

another axis is common. For sweep measurements it is interesting at which stimulus frequency a certain type of distortion occurs, so it is useful to have the stimulus frequency on the x-axis. This enables easy detection of critical frequency ranges. To get a fine and accurate resolution on the axis of abscissa, the Phase Vocoder algorithm is used and is described in chapter 4.4.

For the calculation of the envelope, several algorithms are available making a comparison focusing on this special use necessary.

4.2.1 Squaring and Lowpass Filtering

An easy way to detect the envelope of a time signal is to square the signal and to lowpass filter the result. Firstly, due to the squaring in the time domain, only positive values exist. Secondly, the spectrum is modulated with itself, which means that it is split into parts: one at lower and one at higher frequencies. This effect is useful for lowpass filtering to eliminate the content at high frequency range. The end result only includes the low frequencies which correspond to the envelope of the signal. A simple multiplication with the square root of 2 corrects the magnitude of the result which is valid for sinusoidal signals.

$$x_{Env}(t) = \sqrt{2} * LP\{x(t)^2\}$$

$x_{Env}(t)$............... Envelope
LP.................... lowpass operation

This algorithm is suitable for signals with known frequency contents because a constant lowpass can be used to separate high and low frequencies. In this project the analysis tool must be useable for every kind of signal, therefore the lowpass filter must be adapted for each analysis. It is not possible to filter the whole signal with the same lowpass for sweep measurements over a wide frequency range, so an adaptive filter would be necessary. Of course this approach would be functional, but due to the complexity it is not necessary for this project.

4.2.2 Hilbert Transformation

The Hilbert Transformation is a method to remove redundancy from a signal, normally used in communication systems, but this operation is also useful for the envelope detection.
A real-valued signal has conjugate complex spectrum which means that the negative frequencies are the complex conjugate values of the positive ones. Thus, half of the spectrum has no additional information and is redundant. An analytical signal has only positive frequencies: the negative content is erased. This operation is called the Hilbert Transformation.

$$X_a(f) = X(f) * 2 * u(f) = X(f) + X(f) * \operatorname{sgn}(f)$$

$$x_a(t) = F^{-1}\{X(f) + X(f) * \operatorname{sgn}(f)\} = x(t) + j * (x(t) * \frac{1}{\pi * t}) = H\{x(t)\}$$

x(t)................... input time signal
x_a(t).................. analytic time signal
X(f) .... input spectrum
X_a(f)................. analytic spectrum
u(f)................... Heaviside function

As mentioned above, the main property and use case of the Hilbert Transformation is to reduce redundancy, but it also has another useful property. The magnitude of the Hilbert Transform represents the slow varying content of a signal, and the phase contains the fast variability.

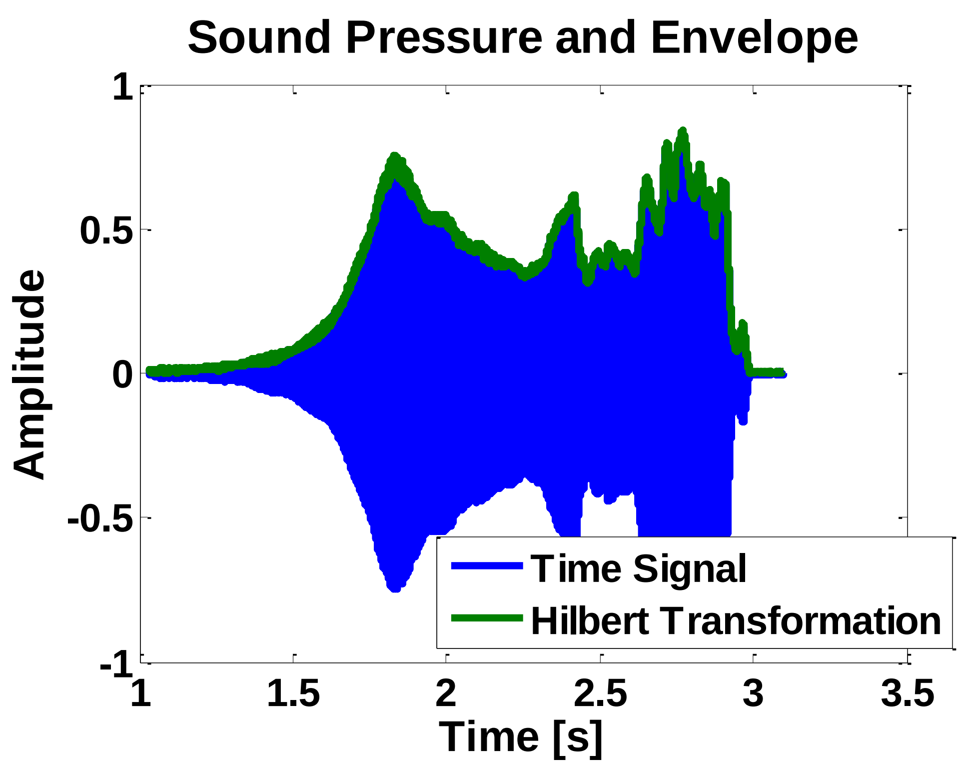

Figure 41: Envelope of a sweep sound pressure signal with Hilbert Transformation

In the first view the Hilbert Transformation can precisely reconstruct the envelope the way the human eye does it. The envelope follows the outer shape of the signal, but doesn't have the fast fluctuations of the time signal.

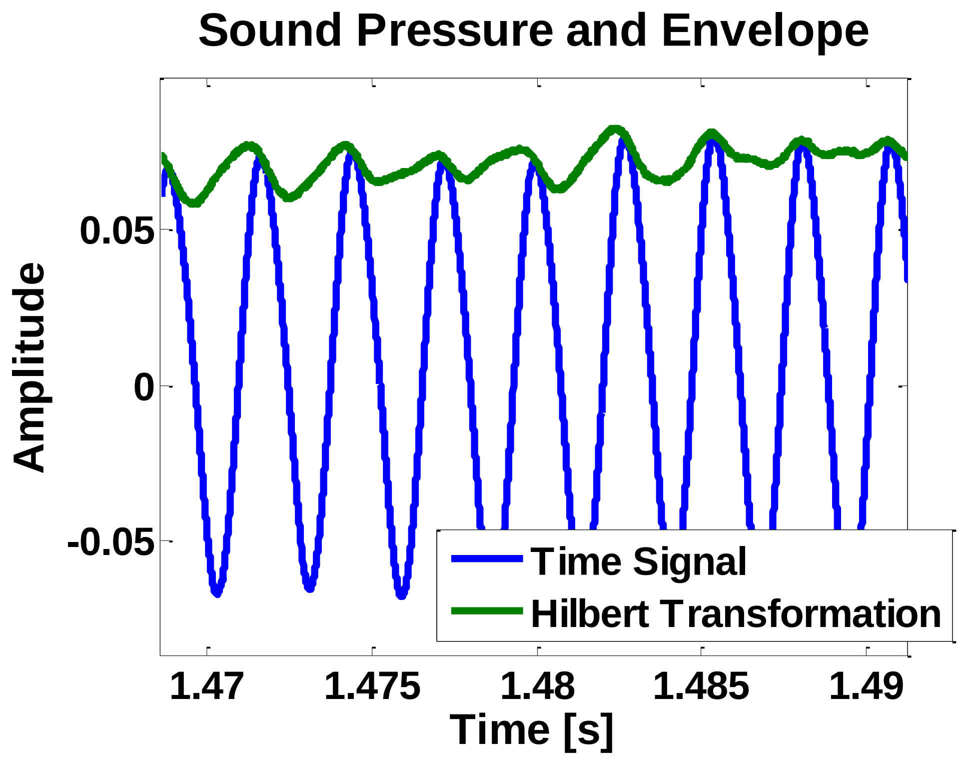

Figure 42: Zoom of the Envelope with Hilbert Transformation

But after having a closer look at the envelope, it is clear that the envelope still has some fluctuations. The Hilbert Transformation can in general connect the maxima of each period like a human eye but there is an additional “waviness” which might lead to misinterpretations.

4.2.3 Period-wise Peak

Another method to determine the envelope of a time signal is finding the peak in each period of the stimulus. The time signal is sub-divided into frames with different lengths where each frame contains one period of the stimulus. The algorithm tracks the maximum of each period and the position of it. This makes it possible to calculate other statistical parameters such as average, RMS or crest factor additionally without extra complexity. Afterwards an interpolation method connects all parameters to get a smooth curve. Therefore different interpolation methods are available, but because of the minor differences in result, this analysis is unimportant to this project.

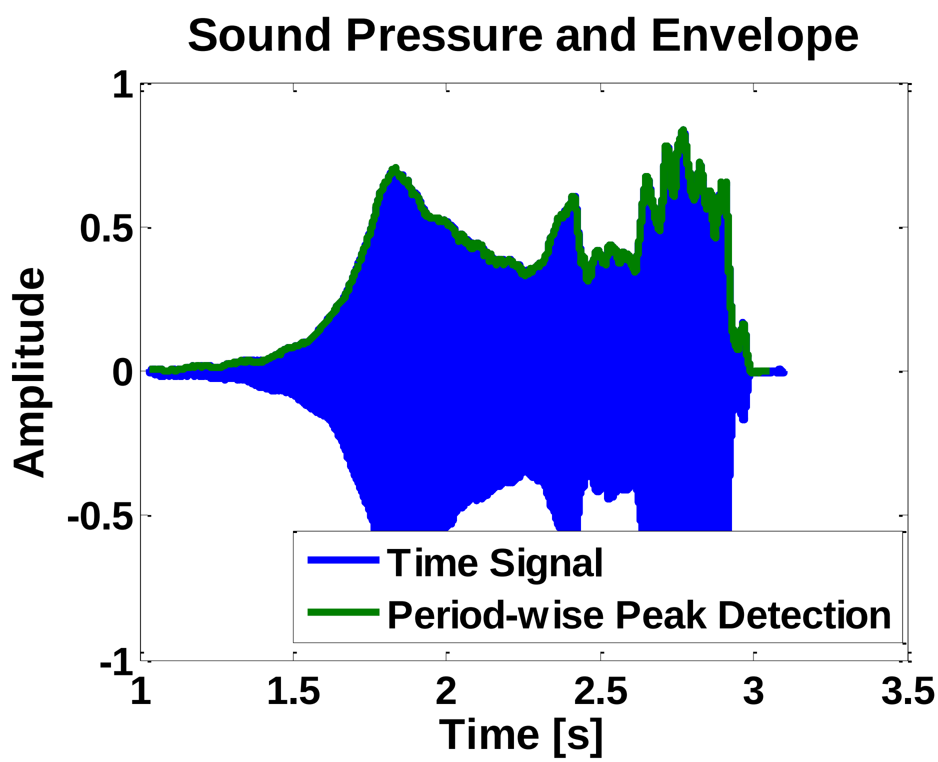

Figure 43: Envelope of a sweep sound pressure signal with peak detection

In the overview plot the result is comparable with the Hilbert Transformation, though it is already visible that the unevenness of the peak detection envelope is smaller.

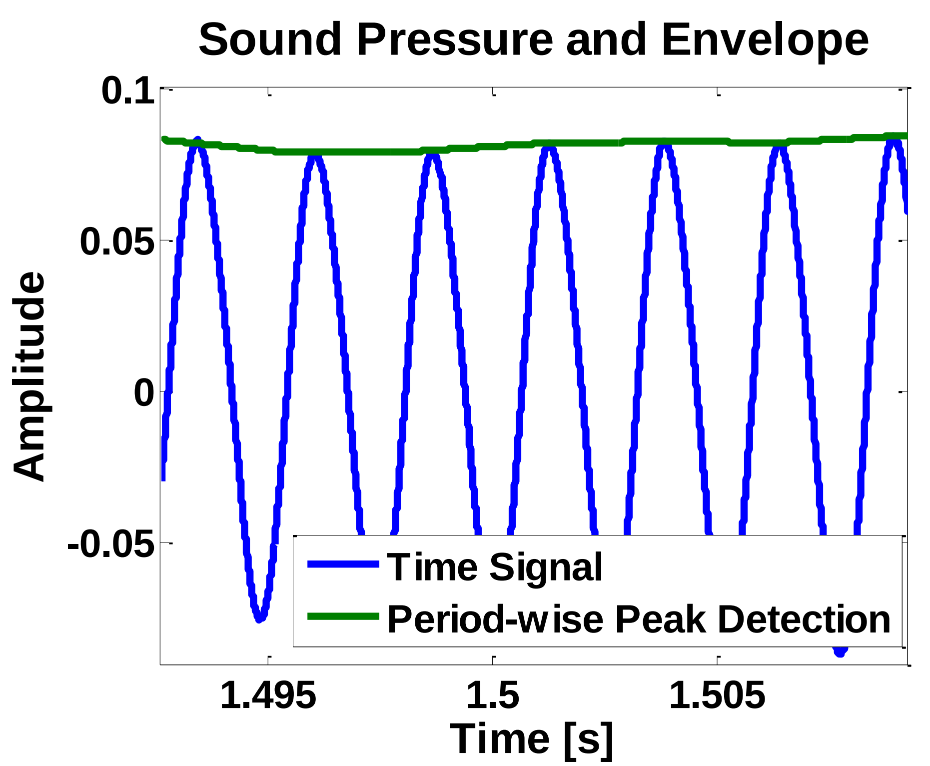

Figure 44: Zoom of the Envelope with period-wise Peak Detection

In the zoomed image, the connection from peak to peak in each period is clearly visible and represents the same method as the envelope construction by a human.

4.2.4 Choosing a suitable method

In this project the period-wise calculation of the maximum is chosen because this method corresponds best with the optical envelope detection by a human where the user observes a fast variable time signal and connects the maximum of each period to get an overview of the process. The period-wise maximum detection works in the same way. Additionally, the operation of the period-wise calculation is already done in this project for another analysis, so it only has to be adapted. The computational complexity is indeed higher than for other methods, but the calculation time is sufficiently fast to make up for that.

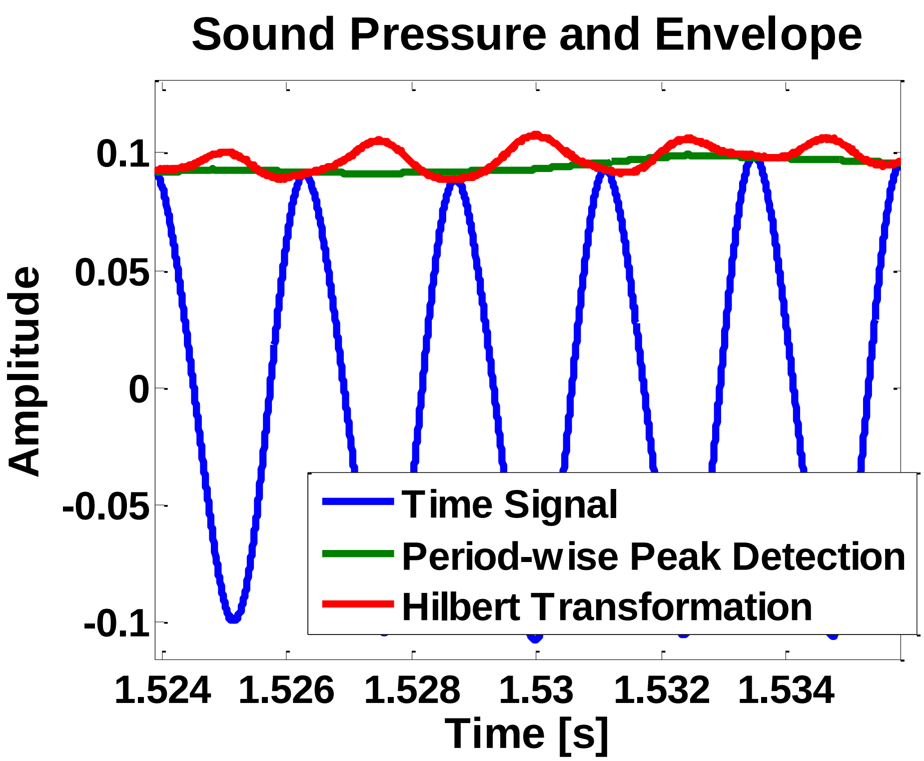

Figure 45: Envelope with Peak Detection and Hilbert Transformation

For the implementation in this project the RMS and the maximum value for each period are determined. Here the comparison of both values is shown.

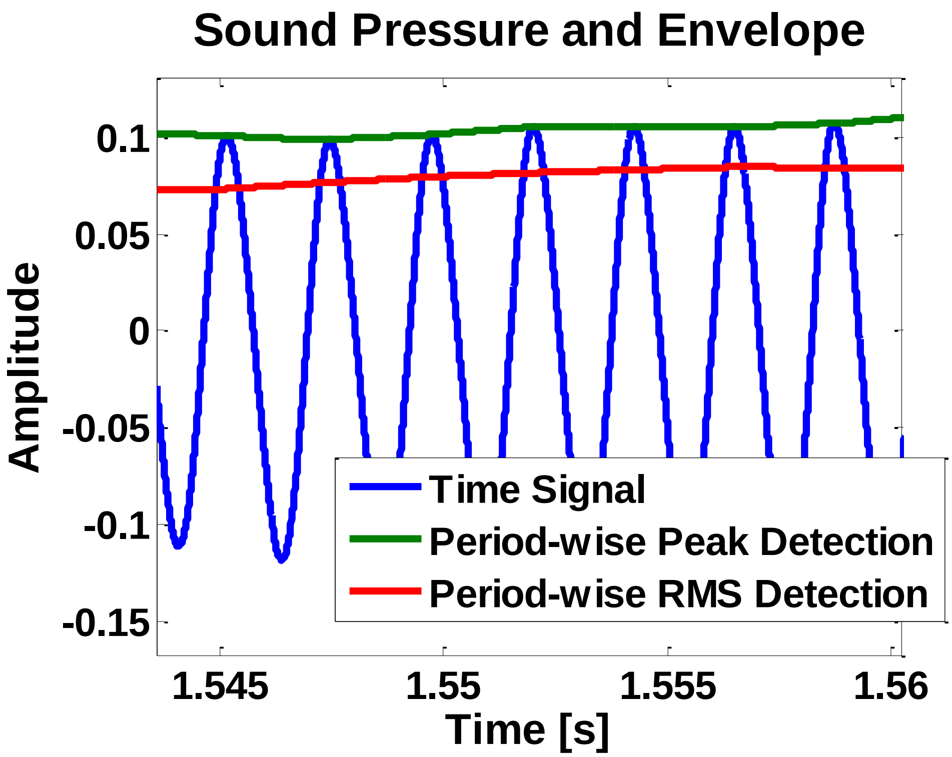

Figure 46: Peak vs. RMS detection

For sinusoidal signals there is a constant offset giving only minor changes. For the excursion signal the engineer is normally interested in the maximum value, so here the peak detection is applied. For sound pressure, voltage and current the energy related value RMS is more important, thus for these signals the RMS value is used.

4.3 Estimation of Membrane Excursion from Sound Pressure

For the root cause detection of an irregular distortion it is helpful to know the exact position of the membrane at any given time, therefore the excursion signal must be known. A measurement with a triangulating laser usually provides this information. But it is often hard to measure the displacement of the membrane in practice because either the laser cannot be placed in a standardized test set up, like a baffle or box, or the loudspeaker is used in a closed application. Because of this, the information about the displacement of the membrane must be estimated from other parameters of the loudspeaker. The sound pressure signal is easy to measure, and must be available for other distortion analyses too. The estimation of the mechanical domain from the acoustical is also much easier than from the electrical because of the unknown parameters of the mechanical oscillator circuit. So a measurement of the sound pressure must be done in a standardized environment in order to know the acoustic behavior of the loudspeaker, and the transformation from sound pressure to excursion can be done in a simple way. The calculation is explained in the following chapter 4.3.1.

4.3.1 Acoustic Impedance of the circular Piston Diaphragm

Acoustic sources with small dimensions compared to the propagating wavelengths can be modeled as isotropic point sources with equal propagation in all directions, but real

electro-dynamic transducers can normally not be simplified in that way when one considers the full frequency range of 20Hz to 20kHz or even more. For loudspeakers with a circular membrane, the acoustic impedance of a circular piston can be used as a good approximation for the description of the transformation from mechanical movement to acoustic sound propagation. The acoustic impedance of a rigid piston in an infinite baffle is typically used, although the membrane is not a perfectly rigid plate and the baffle is not infinite. But normally this assumption is sufficiently accurate within the important frequency range. Other geometries of loudspeaker membranes such as a rectangular micro-speaker can also be roughly approximated with this model. The detailed derivation of this approach including all assumptions and boundaries can be found in (Zollner und Zwicker 1998).
The simplification consists of a rigid piston radiating on an infinitely wide baffle into half-space. The magnitude and phase of the particle velocity at the membrane is constant, but at the baffle, due to physics, it is always zero. Because of this boundary condition, the sound pressure is dependent on the angle from the center axis. Assuming a circular piston, the sound pressure and particle velocity field has rotational symmetrical shape. Summing up: the sound pressure in front of a loudspeaker can be specified knowing the distance from it and the angle from the center axis. In this project, a standardized measurement on the center axis is always done, so the dependency of the angle can be ignored and the following relationship is used for the acoustic impedance:

$$\underline{Z} = \frac{\underline{p}}{\underline{v}} = Z_0 * (1 - 2 * \frac{J_1(2*k*a)}{2*k*a} + j * 2 * \frac{H_1(2*k*a)}{2*k*a})$$

$$Z_0 = c * \rho$$

$$k = \frac{\omega}{c} = \frac{2*\pi*f}{c} = \frac{2*\pi}{\lambda}$$

a Radius of the membrane
J_1 Bessel Function 1st order
H_1 Struve'sche Function 1st order

The first step to get excursion from sound pressure is the inversion of the acoustic impedance which is necessary in order to calculate the velocity. This model was calculated in MATLAB to get an impression of the magnitude and phase behavior.

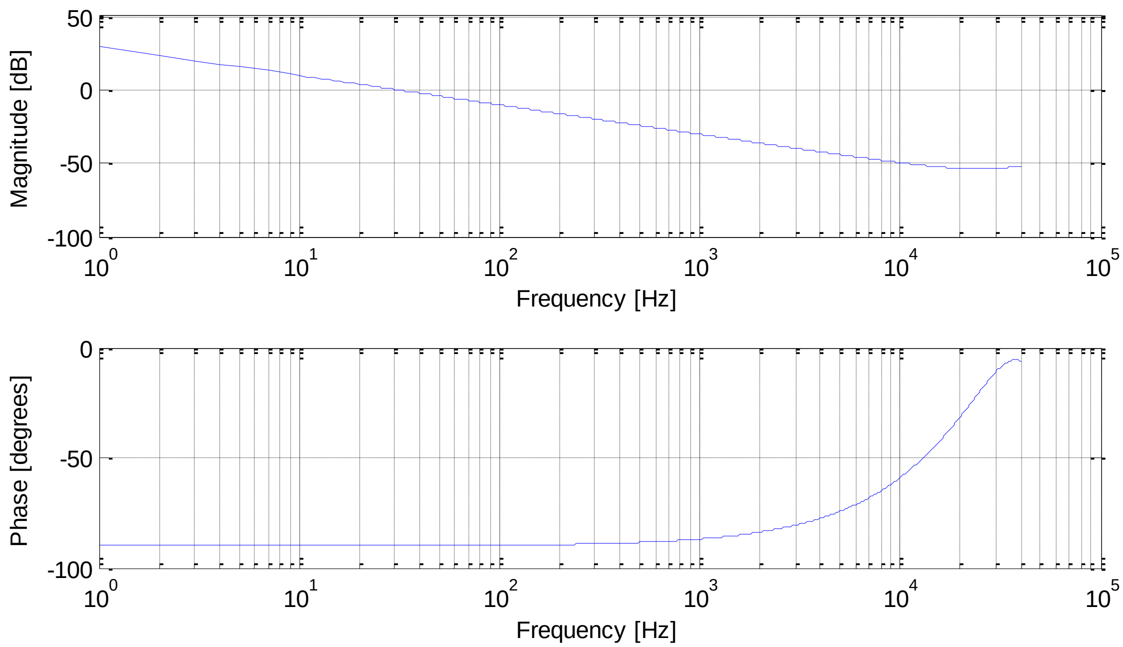

Figure 47: Inverse acoustic impedance of a piston with 1cm diameter

Now it is noteable, that the acoustic admittance has a nearly constant phase behavior and a 20 dB/Decade decay below 1 kHz. For such low frequencies and small microphone distances, the so-called near field area can be assumed where the acoustic admittance can be approximated with a simple integrator element. At higher frequencies, this simplification is not accurate any more, but normally the analysis of distortions of micro speaker with common dimensions is done in the near field area and there the error is negligible.

The next step for the transformation from sound pressure to excursion is the estimation from particle velocity to the excursion. Because of the identical conditions for membrane and air molecules next to it, a simple integration of the particle velocity leads to the displacement of the membrane.

Concerning normal membrane sizes of micro-speakers (~1cm) and measurement distances (1-3cm), the calculation of excursion from sound pressure can be approximated with two integrator elements. This simplification offers a very simple and fast calculation but causes also one problem which needs to be considered:
Real sound pressure signals recorded with a microphone are always without a constant component, but the acquired signal often has an offset from electrical noise, amplifier distortions or quantization noise. This DC component causes rising slopes after integration, which renders the result useless. The incoming sound pressure signal and the calculated particle velocity therefore need to be highpass filtered to eliminate the constant component. The challenge for this filtering procedure is to find the trade-off between computational complexity (length of the filter), cut-off frequency of the sound pressure signal, phase shift and DC attenuation. Several iterations have been necessary to find the best performance with the following filter settings:

Filter Type	FIR
Design Method	Window method
Matlab built-in function	FIR1
Order	4096
Cut-off frequency	100Hz
Filtering Method	Zero-Phase 4.6

Table 2: Filter Settings for reducing the DC content

A higher cut-off frequency would have unsatisfying phase error at the important frequency range, and lower cut-off frequency would not eliminate the DC component due to the integration sufficiently. So 100Hz is chosen for micro speakers when analyzing frequency range from 100 Hz to 1 kHz to get the most stable and phase real performance. Implementation with an IIR filter was also tried, but due to the risk of instability at high frequencies it is not an option for this use case.
Comment: Because of the computational complexity for long time signals with high sample rate, the sound pressure signal is down-sampled for this filtering. After the calculation of the excursion, the signal is up sampled again to have the same sample rate as all other signals.

4.4 Quadrant Detection using the Phase Vocoder

In this section the focus is on the key algorithm for the Quadrant Detection, the Phase Vocoder. The general description of the Quadrant Detection is in chapter 5.9. The Quadrant Detection separates time signals into 4 segments of the membrane movement period.

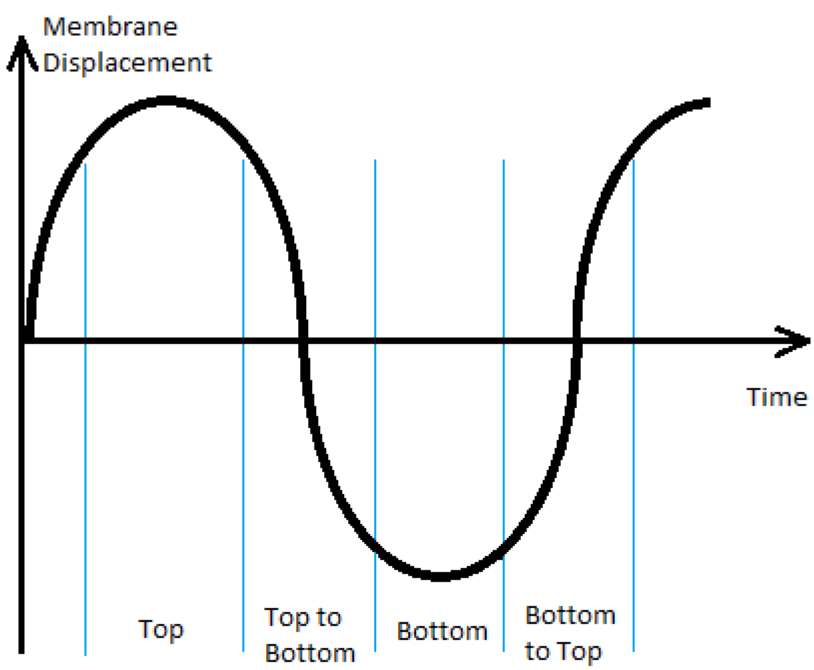

Figure 48: Quadrant separation in accordance with the membrane displacement

For this operation, the exact determination of the stimulus frequency and phase is necessary. For the normal use case, a STFT divides the whole time signal into frames and build the spectrum. Using this method, it is possible to detect the temporal trend of

the stimulus frequency, but due to the uncertainty principle it is only possible to have good frequency resolution or time resolution, but not both.
There are several algorithms available to get accurate frequency estimation, like the MUSIC or Eigenvector methods, and a short summary can be found in (Spectral density estimation 2016). To carry out the Universal Sweep Analyzer, the Phase Vocoder is chosen because this algorithm is based on the STFT, and this calculation is already done for other features. Normally the Phase Vocoder is used to analyze and synthesize speech or music, especially for data reduction, but it also includes an estimation of the instantaneous frequency and phase used here. The general approach is to use the additional phase information of following windows for a more detailed frequency resolution than the STFT provides itself. A detailed explanation of the algorithm can be found in (Puckette und Brown 1998), but the following short description should show the principles:

We use the STFT to obtain a rough analysis of the frequency and phase information of the stimulus.

$$X_k = \sum_{n=0}^{N-1} x_n * e^{\frac{-j*2*\pi*k*n}{N}}$$

Using the phase information of two consecutive windows it is possible to determine the instantaneous frequency more accurately.

$$X_{M,k} = \sum_{n=0}^{N-1} x_{n+M} * e^{\frac{-j*2*\pi*k*n}{N}}$$

$$\omega_{inst} = \frac{\arg(X_{M+H,k}) - \arg(X_{M,k})}{H}$$

H .... Hop size

For this method, a prominent tone causes the same instantaneous frequency value for several adjoining bins. If the SNR is bad or another tone has a comparable frequency, then the region with equal instantaneous frequency value is reduced. Thus, this algorithm works very well if the noise floor level is low and if the interesting tone is very prominent and without additional tones next to it. Here is an overview of the maximum error caused by an interfering tone next to the main stimulus. The details of this study can be found in (Puckette und Brown 1998, p171).

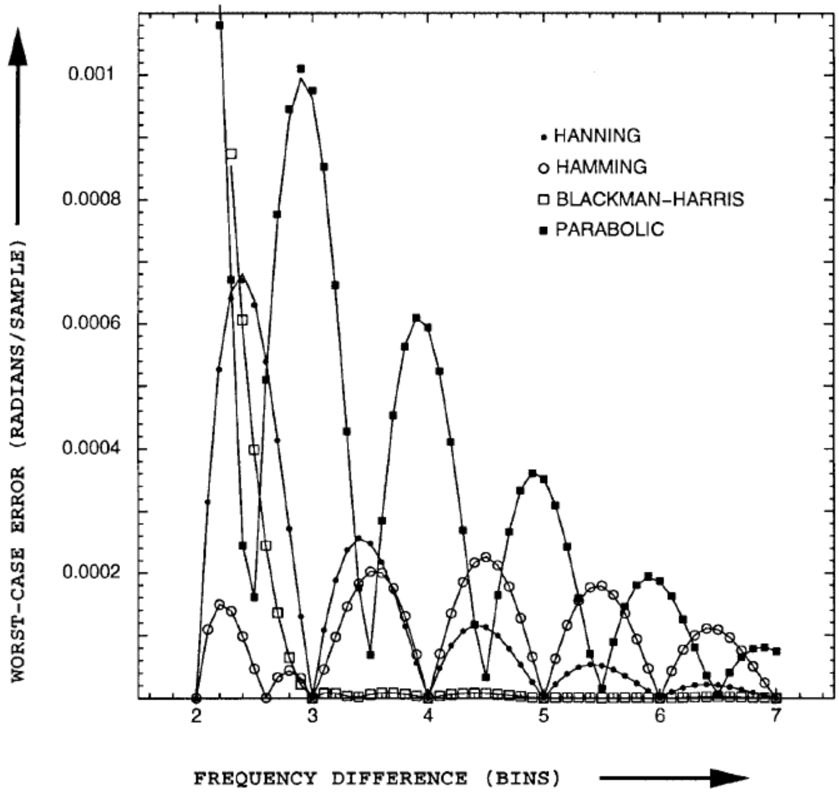

Figure 49: Predicted worst-case interference of one sinusoidal component estimate for a second component (shown as curves) and the measured interference (shown as points) (Puckette und Brown 1998, p171)

The instantaneous phase is the approach to get higher resolution for the phase of a tone. Using the additional information of the instantaneous frequency it is possible to get higher resolution of the instantaneous phase of a certain tone. In general, the information of the phase propagation of two consecutive frames is used to get a more accurate phase estimation of the instantaneous phase. The calculation is done by the subtraction of the instantaneous frequency value and the bin with the maximum magnitude in the spectrum.

$$\varphi_{inst} = \arg(X_k(k_{\max})) - \left(\frac{\omega_{est}}{\frac{f_s}{N}} - k_{\max}\right)$$

k_{max}.................. Bin with maximum magnitude
f_s...................... Sample rate

4.5 Extraction of a Minimum Phase System

Due to the propagation time of the acoustic wave from transducer to the microphone the sound pressure signal is not synchronous with the voltage and current signal of the loudspeaker. To be able to draw conclusions of the temporal appearance of impulsive distortions in accordance with the membrane position, sound pressure and membrane displacement, measured and calculated signals must be simultaneous. Therefore the delay in the sound pressure signal must be compensated, and the extraction of a minimum phase system is convinient for this operation.

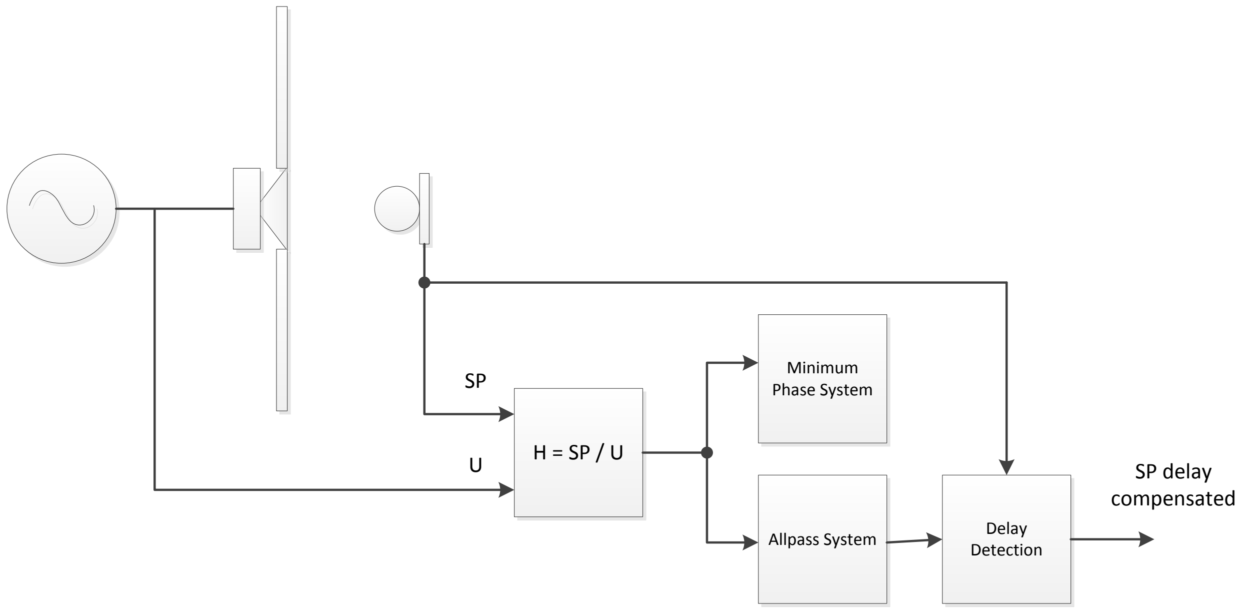

Figure 50: Block diagram of acoustic propagation time compensation

Each stable and causal system can be split into a minimum phase and an allpass system. The minimum phase system represents the magnitude behavior of the frequency response and has a phase response with the smallest possible group delay to reconstruct this magnitude response. The allpass system represents the additional constant delay and has no influence on the magnitude of the frequency response.

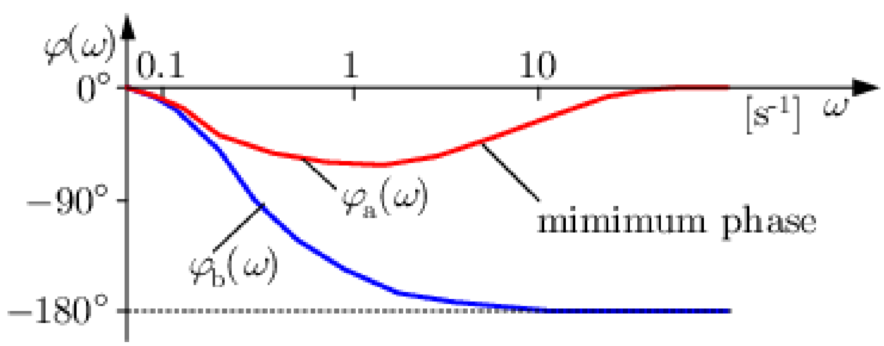

Figure 51: Phase response of a minimum phase and a non-minimum phase system with identical amplitude response (Schmid 2005)

This separation helps to find the linear phase shift caused by a constant delay in the system. This separation is of course only possible if the measurement system is a minimum phase system, therefore the delay of the laser must be compensated in advance.

Generally, a system can have poles and zeros inside, on and outside of the unit circle in the z-plane. If the system is stable, then all poles must lie inside the unit circle, but the zeros can be anywhere. In many cases we would like to find the inverse of a system,

and in this case the location of the zeros is important. The normal inversion transforms poles to zeros and vice versa. Then zeros outside of the unit circle become poles inside it which cause an instable inverse system. It is thus necessary to ensure that for the system to be inverted, all zeros lie inside the unit circle, otherwise the inverted system is instable. A common method to avoid this problem is to split the original system into a minimum phase and an allpass system and invert only the minimum phase system. Due to the constant amplitude of the allpass system this kind of inversion works well for the magnitude behavior. But a correct inversion of the phase response is not possible, because the phase response of the allpass system is not inverted; it stays the same throughout the inversion process.
Another application of this separation in the minimum phase and allpass system is the determination of a constant delay, like the propagation delay of the acquired sound pressure signal. The minimum phase system represents the complete variable phase response, so the allpass system only has the linear phase part which represents the constant delay of the system. Using this allpass system it is easy to determine the delay, e.g. with the use of correlation functions.

4.5.1 Theory

The relation of the minimum phase and the allpass system to its complete system is a simple multiplication of the transfer functions.

$$H(z) = H_{Min} * H_{AP}$$

The stable mixed-phase system has zeros in the complete z-plane and poles only inside the unit circle.

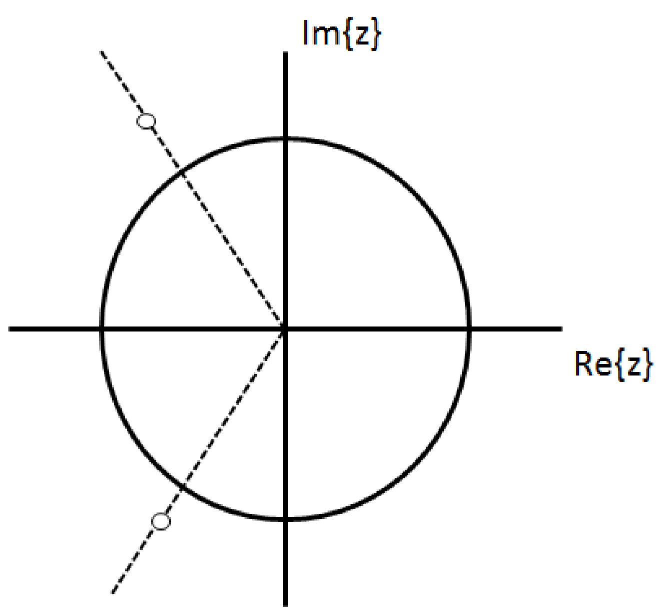

Figure 52:Pole-Zero-Diagram of a Mixed Phase System

The minimum phase system constrains all poles and all zeros to the inside of the unit circle of the original system. All zeros outside of and on the unit circle are eliminated in the minimum phase system.

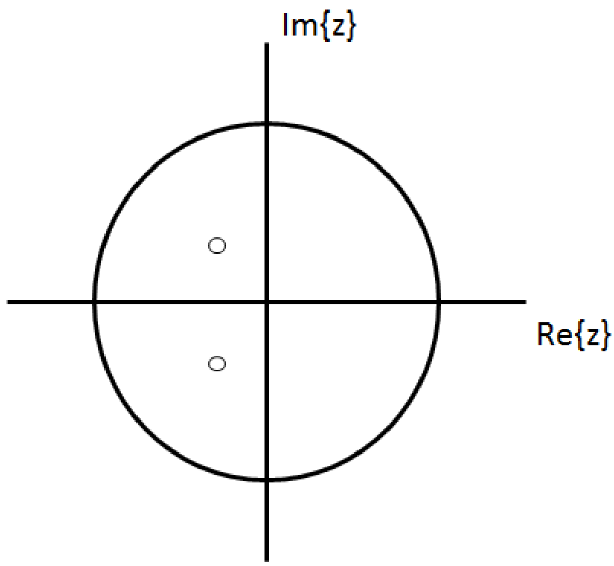

Figure 53: Pole-Zero-Diagram of the corresponding Minimum Phase System

These zeros (on and outside the unit circle) are extracted to the allpass system. Additionally, poles are added, mirrored at the unit circle, to get an allpass characteristic.

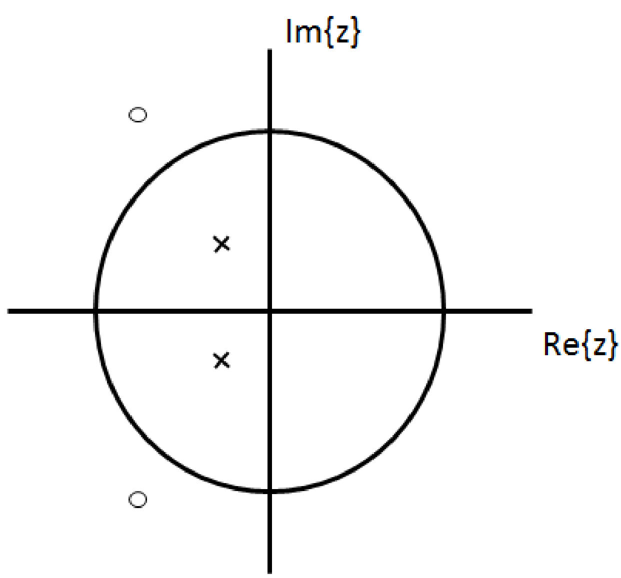

Figure 54: Pole-Zero-Diagram of the corresponding Allpass System

The modified inversion of the mixed phase system can be done by inversion of the minimum phase part and keeping the allpass part. Adding both inversions together, a kind of inversion of the original system is calculated. The magnitude representation of the real inversion is good, but the phase behavior is bad due to the instability of the real inversion. Of course, this approach is not the real inversion, however it is often more important to generate a stable system with bad phase reconstruction than having the real inverted system.

4.5.2 Calculation

The method described above is quite a simple algorithm, but for an effective calculation it is not the best solution. An alternate method to create a minimum-phase system with lower computational complexity is to transform the transfer function to the cepstral domain. To that end the following property of the complex Cepstrum is used:

Each minimum-phase zero in the frequency domain corresponds to a causal exponential decay in the Cepstrum, and each non-minimum-phase zero corresponds to an anti-causal exponential decay. Mirroring the negative cepstral area to the positive one corresponds to making all non-minimum-phase zeros to minimum-phase zeros. In doing that it is very important to ensure that the transformed spectrum has no hard notches and discontinuities as otherwise these edges would cause very slow decays in the Cepstrum and time-aliasing would be an issue. An interpolation of the spectrum is for this reason necessary to get a trustworthy result.

In this thesis, a readily finished function which implements the above algorithm in an efficient and safe way has been used. Further details are described in (Schmid 2005).

4.6 Zero-Phase Filtering

The Zero-Phase Filtering is a specific way of applying a digital filter on a signal with the advantage of avoiding any additional phase distortion to the signal. There are several signal processing steps in the implementation of the Universal Sweep Analyzer where the phase is very important so this method of digital filtering is valuable. For example, the filtering of the Rub & Buzz signal, where very sharp cut-off frequencies are wanted, or the DC reduction at the integration of the excursion estimation, where DC must be eliminated but frequencies above 100Hz have to be phase true, are use cases for the zero phase filtering. Thus, this method has very high importance in this thesis and a short description of the algorithm shows the features and limits of the used function (Van Veen 2012).

4.6.1 Theory

The key step of the Zero-Phase Filtering is the time-reversal property of the Time-Discrete Fourier Transform (DFT).

$$x[n] \overset{DTFT}{\longleftrightarrow} X\left(e^{jw}\right)$$
$$x[-n] \overset{DTFT}{\longleftrightarrow} X^{*}\left(e^{jw}\right)$$

Using this feature it is possible to filter an input signal, first in a forward direction and then in a second reverse step. In the following block diagram, all necessary operations and intermediate results have been made visible.

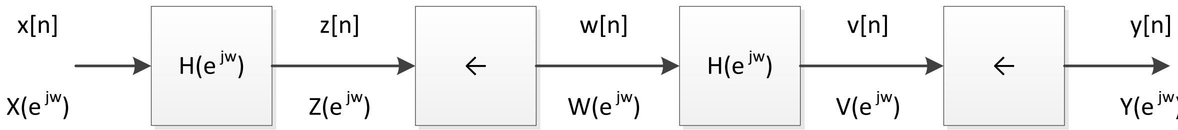

Figure 55: Schematics of Zero-Phase Filtering

Time-Domain	Frequency-Domain
$z[n] = x[n] * h[n]$	$Z(e^{jw}) = X(e^{jw}) \cdot H(e^{jw})$
$w[n] = z[-n]$	$W(e^{jw}) = Z^*(e^{jw})$
$v[n] = w[n] * h[n]$	$V(e^{jw}) = W(e^{jw}) \cdot H(e^{jw}) = \lvert H(e^{jw})\rvert^2 \cdot X^*(e^{jw})$
$y[n] = v[-n]$	$Y(e^{jw}) = Y^*(e^{jw}) = \lvert H(e^{jw})\rvert^2 \cdot X(e^{jw})$

Because of the forward and backward filtering, it is possible that only the magnitude response of the filter is applied to the input signal, and the phase response is unchanged.

4.6.2 Properties

The Zero-Phase Filtering has the big advantage that there is no phase distortion caused by the filtering procedure, so the phase difference between input and output spectrum is zero at each frequency point. There are, however, also some disadvantages that might be a problem if not taken into consideration.

- Acausal system
  Due to the reverse filtering, the Zero-Phase Filtering is an acausal operation. This has the effect that there is an output before the input is applied. Therefore, the complete input sequence must be available at the beginning of the filtering procedure, so the algorithm is not useable for a real time operation without block-wise calculation.
- Scaling error
  As seen before in the schematics (4.6.1), the effective magnitude response of the Zero-Phase Filtering is the squared magnitude, so for a correct implementation the square root of the target magnitude response must be used.
- Higher Computational Complexity
  Because of the forward and backward filtering, and two times a time-reversal operation, the computational complexity is much higher than the normal filtering procedure. This can cause a very long calculation time if long signals are used.

In this project the Matlab internal function filtfilt is used. This built-in function takes care of necessary computation steps which makes it user-friendly. But the high calculation time is still a problem if the input sequence is long. Hence this function should be used only if the zero-phase property is of high importance and if the input signal is not too long; otherwise the normal filtering operation should be preferred.

All signal analysis procedures described in this chapter are needed for the implementation of the Universal Sweep Analyzer. This software program loads acquired data and calculates useful signals for the failure analysis. The detailed functional specifications including all visualization and features are described in the following chapter 5. Additionally, the internal structure of the software and how the algorithms above are used in the program are shown.

5 Implementation of the Universal Sweep Analyzer

In this thesis a software program that includes all above described analysis methods in one tool is developed. It supports the acoustic engineer with failure analysis and distortion minimization and uses test data from several common measurement systems. The name Universal Sweep Analyzer (USA) is based on the flexibility of the analyzed signal type. The program can analyze all kinds of signals, so it is not specialized for sweeps, pure sine tones, or other common test signals. This makes it independent of the test system in use and use case. But it is of course developed for the analysis of sweeps because this stimulus type is very common for loudspeaker tests.
In the last chapter, the key algorithms described are used for data preparation. These signal processing steps are basic functions that are helpful for first verifications for programmers, however industry engineers need a powerful program with user interface, visualizations and import/export functions for their daily work. To satisfy these requirements, a Matlab program with Graphical User Interface (GUI) has been developed using the algorithms. This chapter contains the description of the requirements from the engineering side, including the implementation of the key algorithms for a user friendly interface.

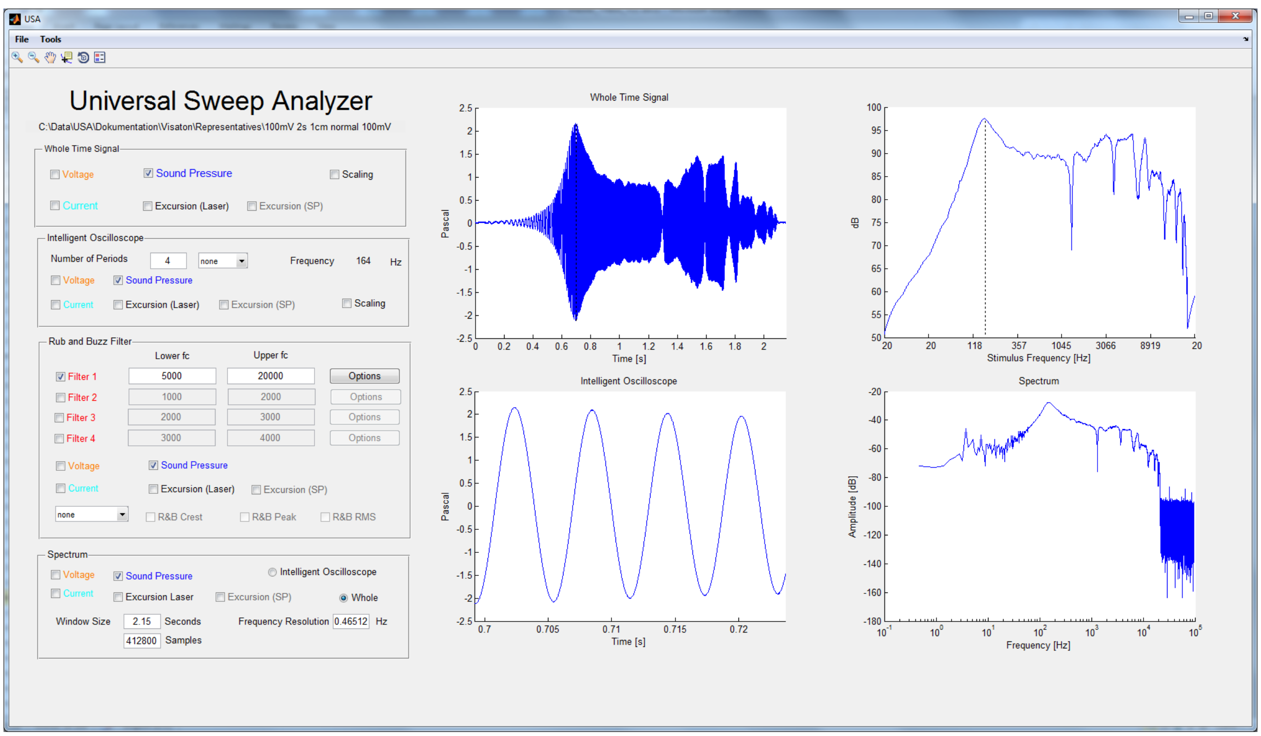

Figure 56: Main window of the Universal Sweep Analyzer

5.1 Functional Specifications

Many different measurement systems are available (e.g. Audio Precision, Soundcheck, Klippel) which have very good data acquisition and analysis methods. Some of them have good classifications of regular and irregular distortions too, but there is no existing system which is specialized for micro-speakers. As a result, the engineer has to do measurements with different systems, otherwise the analysis methods are limited.

Because of this, one post-processing tool is necessary to include all algorithms for detecting defects and find appropriate limits for the quality control in mass production. This system should import data from all typical measurement systems and should be useable without additional hardware. In order to be fast and flexible, an import function must be able to load predefined and user defined sets of data.

Many different diagrams must be provided so that the engineer can achieve high flexibility in the analysis. All key plots are in the main window to get an overview and further analyses with specialized algorithms are shown in separate windows.
All settings for the internal key algorithms and plot properties must be modifiable, but for inexperienced users the tool should as well be useable without long learning phase.

5.1.1 Display of Time Signals

To get a general impression of the measurement result, a plot of all available measured and calculated signals is recommended. For this purpose, a diagram with the time scale on the abscissa and two ordinates with the most important units are implemented. A scaling function to normalize all graphs simplifies the analysis. The scaling factors are displayed in a separate window if the user wants to see the true amplitude.

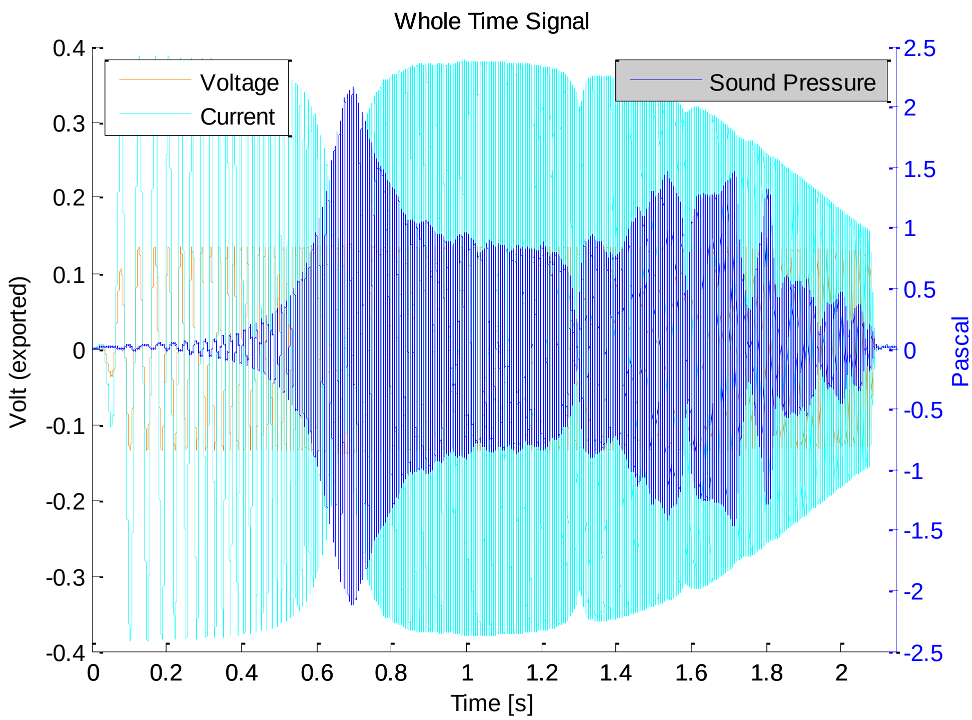

Figure 57: Time signal of a typical measurement including sound pressure, voltage and current

5.1.2 Intelligent Oscilloscope

Because of the large amount of information contained in a complete time signal of a sweep, a zoom function is necessary to be able to properly inspect the signal. A click in the time signal or envelope plot determines the middle time point of the detailed view.

Normally, the problem is that a cutout with a fixed length is very unsatisfying, because for typical sweeps the frequency changes very fast and so the relative size to the period depends on the position of the zoom. Therefore a display of a constant number of periods at each time point always shows a cutout with the right time scaling.

To get the right number of samples for a “period-constant” zoom, the result of the Phase Vocoder analysis is used. First, the time signal is sub-divided in equidistant frames with same lengths that are transformed in the frequency domain. The algorithm of the Phase Vocoder analyzes the phase behavior of the most dominant frequency. The phase difference between two consecutive frames corrects the DFT bin frequency to the instantaneous frequency. A detailed explanation of this procedure can be found in section 4.4.
Afterwards, the instantaneous frequency values for discrete time points are linearly interpolated to get a trend of the stimulus frequency for each sample.
This algorithm is normally applied to the terminal voltage because it has the best SNR and hence the instantaneous frequency is detectable in the widest frequency range. But if the terminal voltage is not available, then it is applied on the sound pressure signal.

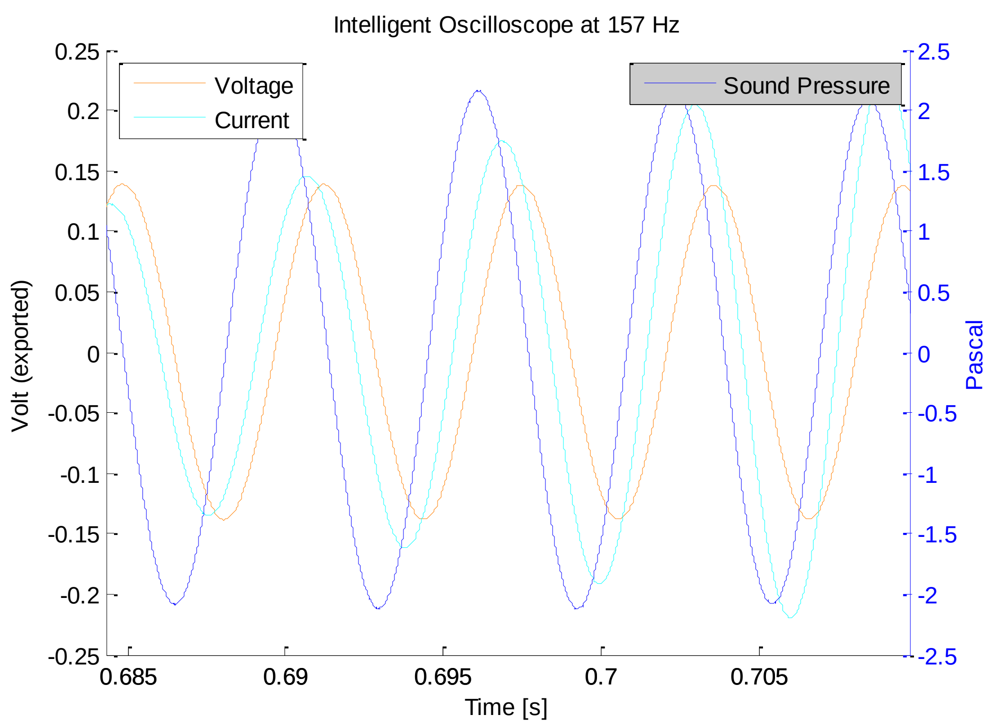

Figure 58: Intelligent Oscilloscope at the resonance frequency

5.1.3 Spectrum

Often the spectral distributions of all available signals are relevant to the acoustic engineer because it brings valid information of the distortion character that can support the root cause detection. The DFT of the complete time signal helps to get an overview of the general behavior, but it is not very useful for time-variable progression, especially for sweep measurements. Therefore a spectral view of a certain time point is of interest. Because of the uncertainty principle it is not possible to get a spectrum of a single

sample so a window with finite length must be chosen. For a better resolution in the frequency domain, a longer window is necessary, and for the analysis of a short event a bad frequency resolution is the inevitable compromise.
Normally the acoustic engineer does not want to think about frame lengths and frequency resolutions, and so the program has to support easy controlling. The user can input a required frequency resolution and get the length of the window to check the stationarity within the corresponding frame. If a known length of an impulsive distortion is known, then the other option is available and the engineer can input the window length in seconds or in samples to limit the window width. The frequency resolution is calculated automatically to see the limitation of the spectral analysis.

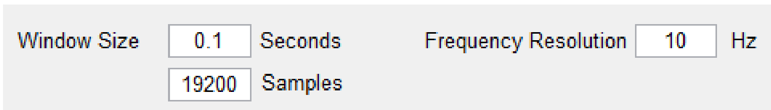

Figure 59: Settings panel for the spectrum plot

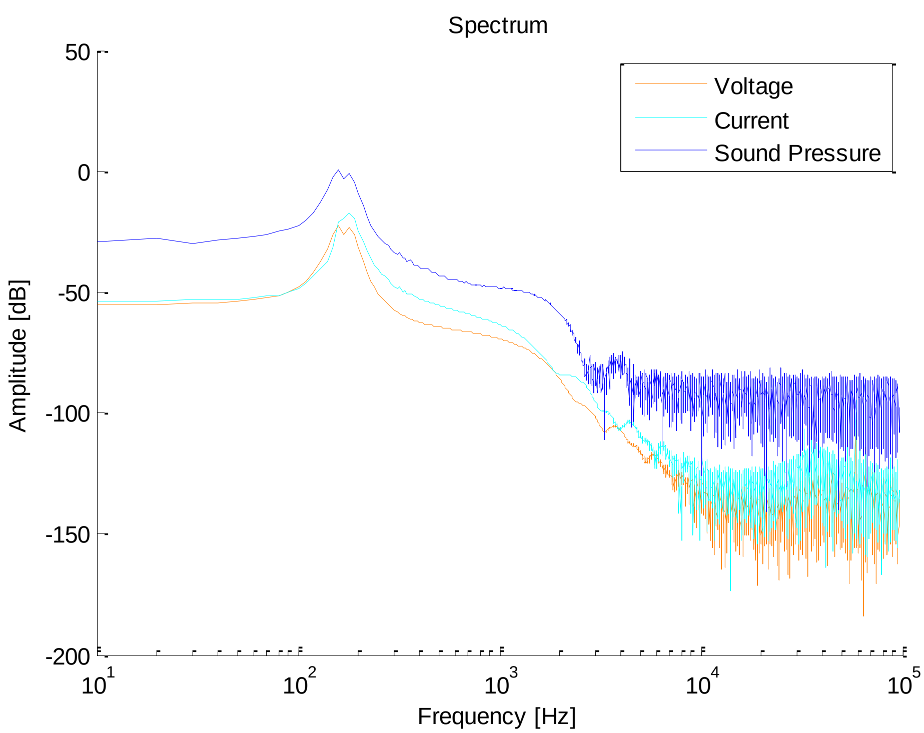

Figure 60: Spectrum of a frequency sweep at the resonance frequency. Due to the changing frequency and the finite small window length the fundamental is not a single frequency but a band.

5.1.4 Envelope over Frequency Axis

A useful view is the envelope of time signals, because for frequency sweeps it shows its behavior through the complete frequency range, but is still a plot in the time domain, so it is a "quasi frequency response". Normally an envelope connects all maxima of the time signal to get a smooth curve which corresponds to the outer limit, but for most types of analyses the RMS value is more important than the maximum. Thus, in this thesis for the envelope the RMS value of every stimulus period, the so-called

instantaneous RMS value, is connected to a smooth line. The detailed explanation can be found in section 4.2.
The envelope is displayed in time domain, but the time position is not interesting for the developer, rather he is interested in the stimulus frequency at that time. The abscissa is then "transformed" from the time domain to the corresponding stimulus frequency. Using the instantaneous frequency calculated before, the exact value for the stimulus frequency for every sample is available and so the envelope is plotted over the time samples with the corresponding frequency values displayed.

The instantaneous RMS, crest and peak values of the filtered signal, the Rub & Buzz signal, are additionally shown in this plot. These curves correlate strongly to irregular distortions like a hitting voice coil and helps the detection of production faults.

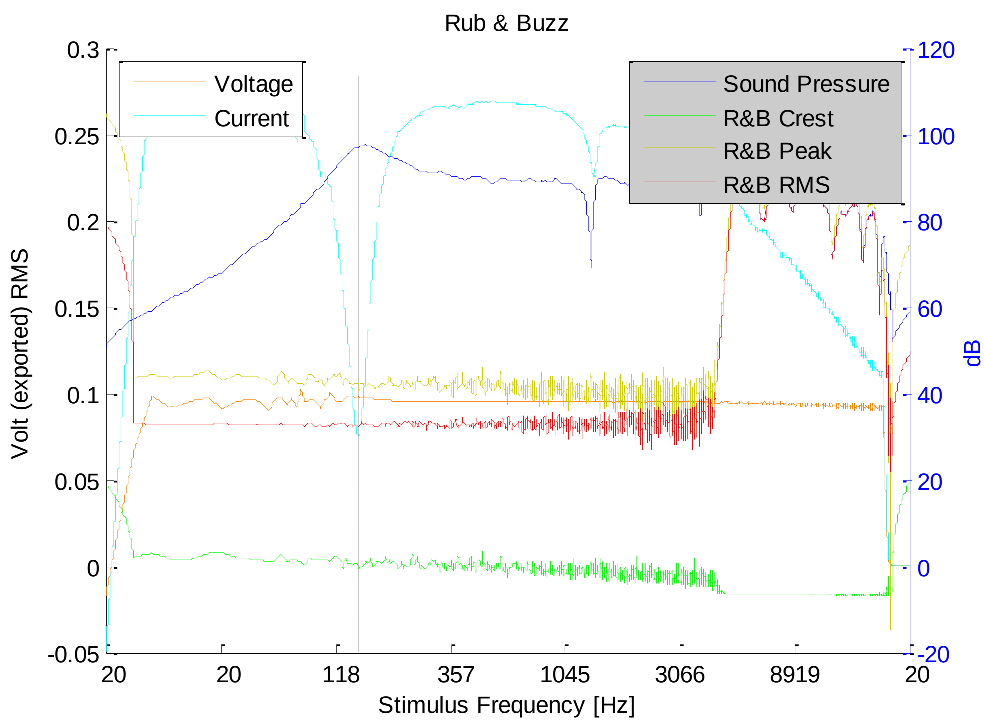

Figure 61: Envelopes of a normal measurement

Because of the flexibility of the input signal, it is not possible to fit the frequency trend and consequently the display of the frequency values cannot be done in equidistantly in the frequency domain. The time axis is therefore divided in regions with constant length and the instantaneous frequency of the stimulus at each point is displayed. Zooming in on the time domain reveals new frequency values.

5.1.5 Spectrogram

A standard analysis method for loudspeaker developers is the Spectrogram, which must be available in this program too. The spectrogram uses the STFT (see 4.1) and is a representation of the sound pressure signal in the frequency and time domain at a glance. The time signal is sub-divided into frames that are transformed in the frequency domain. This allows a spectral view dependent on time. To display all of this information in one plot, three axes are necessary. The abscissa is normally the time domain, the ordinate is the frequency domain and the amplitude is scaled with a color code. Additionally, the plot can be displayed at another angle so a three-dimensional view is available where the amplitude corresponds with the envelope.

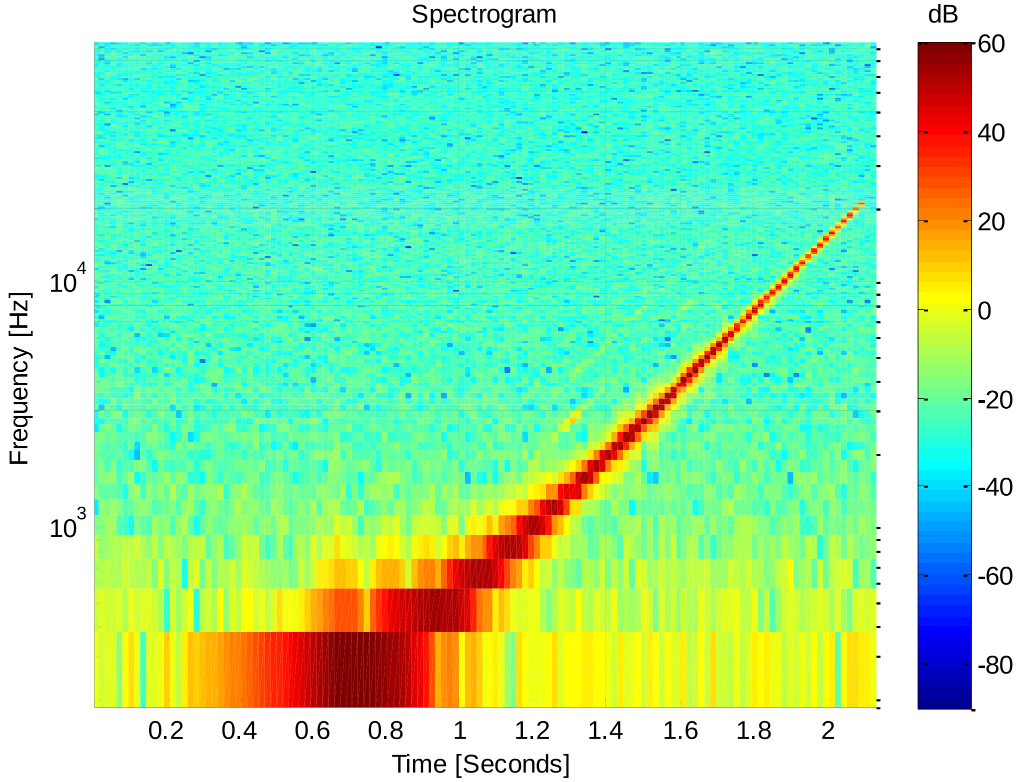

Figure 62: Spectrogram of a normal measurement

5.1.6 Quadrant Detection

Some defects are excursion driven, so their appearance correlates to a certain displacement or phase of the displacement. For a better understanding of this effect, an analysis method is developed that detects the temporal behavior of the distortion with respect to the excursion of the membrane. For this application, two different algorithms are used: one for deterministic distortions and one for stochastic ones.

The first method detects the origin of the distortion and displays the corresponding quadrant of the membrane. This calculation works only for deterministic defects; hence the crest factor of the bandpass filtered sound pressure signal is the threshold for this operation.

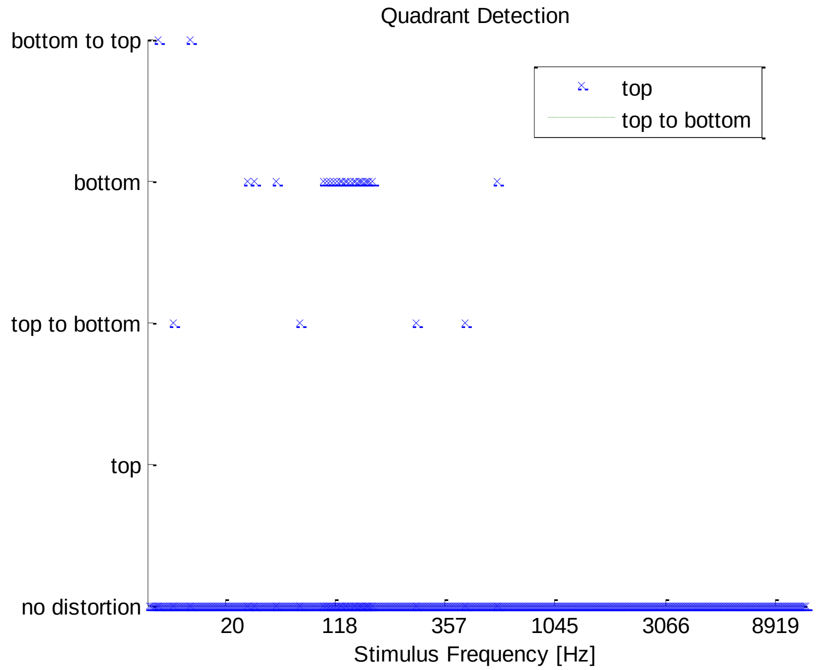

Figure 63: Deterministic Quadrant Detection

For all other distortions, the signal after the Rub & Buzz filter is split into sections corresponding to four quadrants of the excursion, thus the energy is subdivided and the most critical displacement area is easily seen.

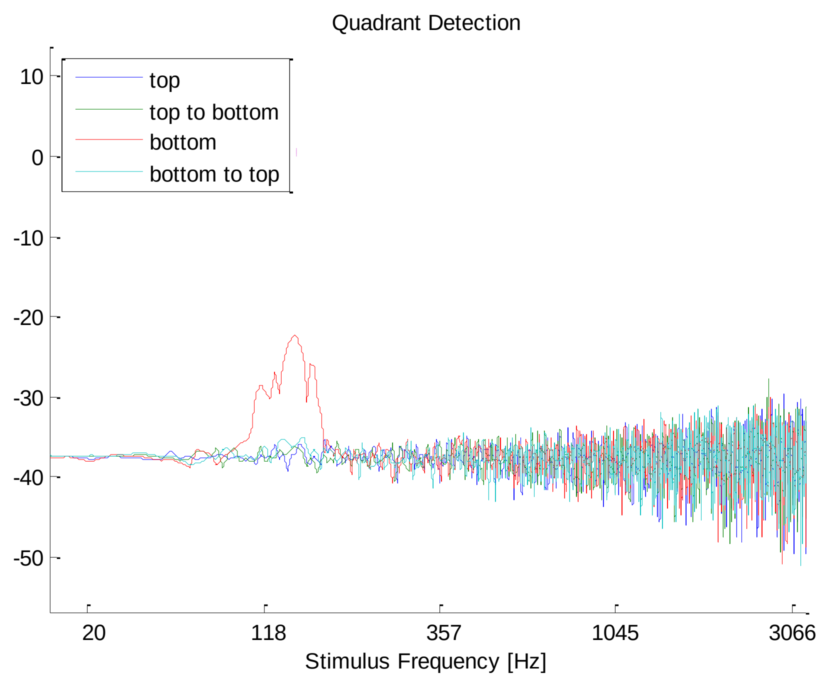

Figure 64: Stochastic Quadrant Detection

5.2 Import of Measurement Data

The Universal Sweep Analyzer is an analysis tool only and has not been developed to measure the loudspeaker itself. It is therefore necessary to import test data from other measurement systems, and a customize import function must also be available.
In general, all common physical domains, which describe a loudspeaker, are used in this program, and these signals must be importable. Often the engineer does not need a complete description of the loudspeaker, or he does not have the opportunity to measure all parameters. In this case the program can also read the available data and disable all functions that have insufficient information.
Furthermore, the calibration levels of each metering device are read from the import file, or they can be set manually.

5.2.1 APX

APX is the new measurement system from the company Audio Precision (AP) and is a universal acoustical measurement system with different predefined routines that make it flexible.

Figure 65: Hardware of the used APX

The Universal Sweep Analyzer is generally useable for all available signal forms in APX. The user simply needs to export the acquired waveform in the time domain. APX supports many routines with internal calculations and displays for detailed analysis, like the impulse response or the spectrum, without saving the underlying time signal. But for a useful export, a measurement template with a plot of the time signal is necessary.

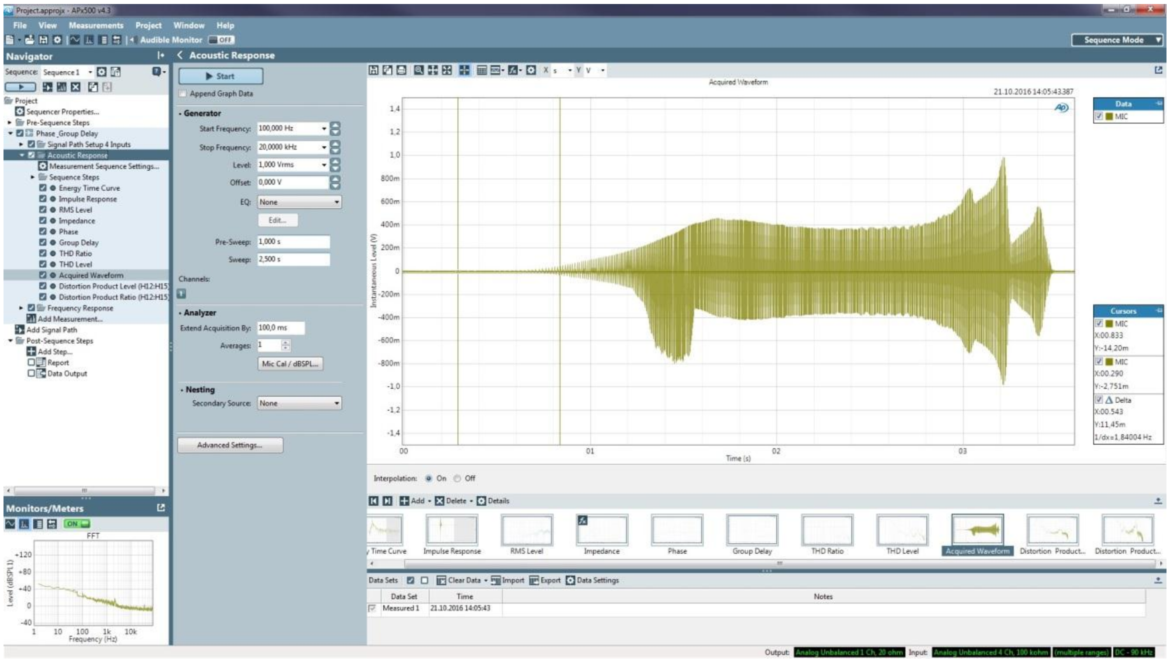

Figure 66: Screenshot of an acoustic measurement with APX

The user must then export the acquired waveform manually or with a predefined routine to get a file in the format ".mat", which is the standard Matlab file format. Because of the detailed internal structure of a .mat-file, the data set for long time signals can be quite big and that might cause problems with the export.
The exported file must have the following structure to be importable in the USA:
All data must be saved in a cell-array called "InstantaneousLevel". This array must have four rows where in the first row the name of the signal is saved and in the fourth row the data vector is placed. The time vector is placed in the first column, and the position of the others is searchable by name. The following names are defined and found automatically:

- "MIC" - sound pressure
- "UT" - terminal voltage
- "I Gras" - voice coil current (measured with a G.R.A.S[1] amplifier)
- "Laser" - membrane displacement

Normally, the export of APX has the same structure and the names are defined in the measurement itself.

5.2.2 Customize Import

Because of the necessary usability for all measurement systems and signals, it is important to import the common data formats. This program is available for the following data formats:

[1] G.R.A.S is an amplifier that was developed for the company Phillips. It amplifies the output voltage and the microphone signal for loudspeaker tests and additionally it offers a current measurement of the output channel.

- .mat - Matlab workspace
- .wav - Wave format (standard audio file)

All these file formats must have the same structure. All data sets must be saved in columns and the time axis must appear in the first column. The order of the other data sets can be chosen in the program.

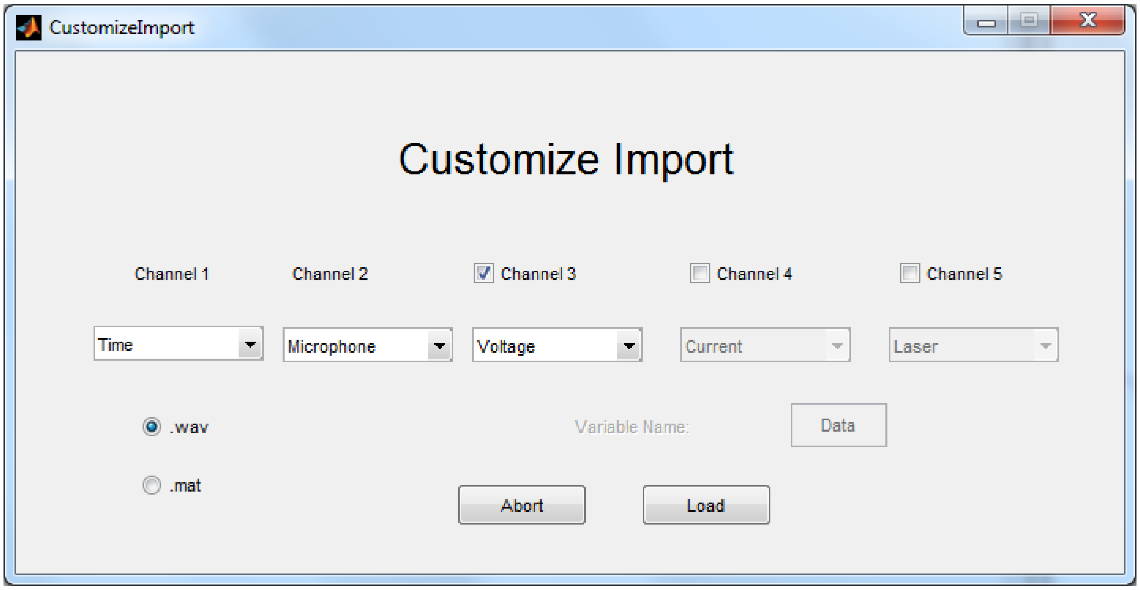

Figure 67: Settings to load a customized set of data

5.3 Synchronization of Microphone and Laser Signal

For a detailed analysis in the time domain it is important to get all data sets without any delay. Normally, the measured signals have delays because of systematic errors and calculation time of acquisition devices. It is therefore necessary to take a detailed look at the whole chain of the measurement setup.

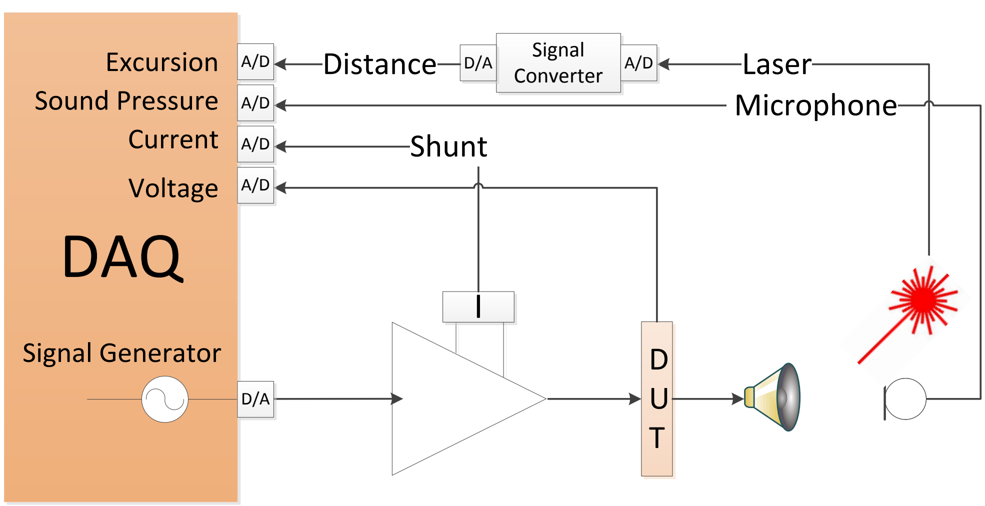

Figure 68: Measurement setup with A/D and D/A converters

Analog instruments are normally used to measure the target variable, but the analysis and further calculations are done in the digital domain, e.g. a computer. Thus, all signals have a delay from the analog to digital converter of the measurement system. But since the delay is the same for all channels, it does not result in synchronization issues.

5.3.1 Laser Delay

The excursion signal is normally measured with a laser inclusive the conversion from time to distance. This operation is done in the digital domain, so it is converted internally. To get the signal in the same way as the other inputs, the analog output of the laser system is used, which leads to an additional analog to digital and digital to analog conversion. Hence the excursion signal has a larger delay than the other signals and this needs to be compensated for.
In this project, a Keyence L-H052 has been used to measure the displacement of the membrane. Measurements with an impulse applied on the loudspeaker showed that the laser has a constant delay of 490µs so the Universal Sweep Analyzer is able to compensate for this delay. For measurement devices with different calculation times, this compensated delay can be changed under “Settings”.

5.3.2 Acoustic Propagation Time

Another signal with an additional delay is the sound pressure measured with a microphone. Because of the finite propagation velocity of the acoustic wave in air, the transmission from loudspeaker to microphone needs some time. This delay depends on the distance between source and detector and the physical environmental parameters, like temperature and humidity that cause a change in the propagation velocity. An additional problem is the definition of the acoustic center of loudspeaker and microphone, so the measurement of the distance between them is quite difficult. Consequently, the compensation for the acoustic delay due to an estimation of the distance is very rough and doesn’t deliver exact results. This is especially critical because a small error in the delay compensation causes a high phase deviation at high frequencies.
This means that an algorithm has to be developed, which compensates the delay between the microphone signal and all other signals (terminal voltage, current and already delay compensated excursion). Because of the phase behavior between voltage and sound pressure of an electro-dynamical transducer, a simple delay compensation using the auto- and cross-correlations of the time signals does not work. Calculating the delay from the phase response is also not possible because of flexibility in the system in use. If there is low signal content in a certain frequency range, then the phase response can be corrupted by the noise floor so the phase detection cannot be guaranteed to be accurate. Hence, the system of the electric to acoustic transition is split into the minimum phase and the allpass system, and the allpass system directly corresponds to the constant delay.

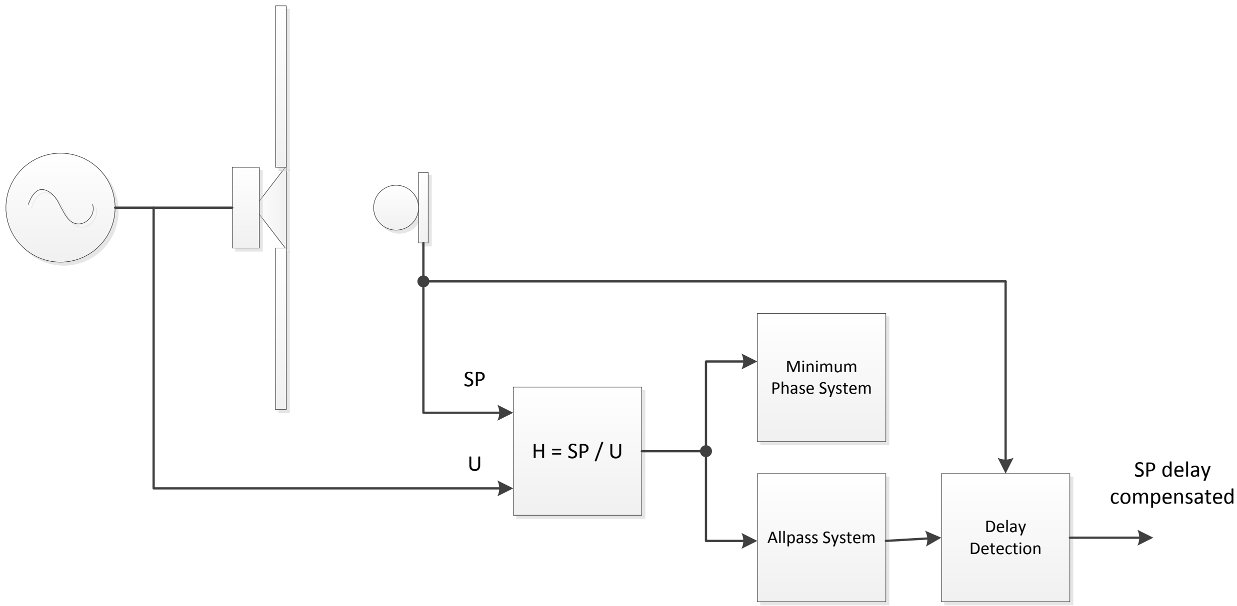

Figure 69: Block diagram of acoustic propagation time compensation

The impulse response of the extracted allpass system shows the acoustic delay and this is used for the compensation using a simple shift operation. The detailed explanation of the separation in minimum and allpass system is in 4.5.
This form of compensation for the delay of the microphone signal only works with a loudspeaker in a baffle and a microphone in front of it. For other set-ups, like the measurement of a loudspeaker in an artificial ear, the compensation does not work and the value must be changed manually in the program.

5.4 Estimation of Excursion from Sound Pressure Signal

Often the acoustic engineer wants to describe the loudspeaker in all physical domains. It is then necessary to know the input (electric), the output (acoustic) and also the domain in the middle: the mechanical. Normally the mechanical domain, respectively the displacement of the membrane, can be measured by a laser-vibrometer, but this measurement requires a special set up where the membrane is visible. Often this setup is not applicable or a front port generally does not permit a laser measurement. In this case an estimation of the excursion is necessary to get information about the mechanical behavior.
The detailed description of the transformation of a sound pressure signal to an excursion signal is shown in chapter 4.3.
This estimation works for the situation of a circular piston diaphragm in the near field, so the user must consider that this model works for low frequencies and small microphone distances only. Additionally, because of the bad SNR in the sound pressure signal below the resonance frequency, the estimation in this frequency area can result in incorrect results, so the engineer must always check the estimation for plausibility.

5.5 Intelligent Oscilloscope

The intelligent oscilloscope is a primitive but useful visualization of a time signal, which support the normal analysis of loudspeakers a lot. Its complete set of requirements is defined in 5.1.2.
Its approach is the instantaneous frequency, which is applied on the terminal voltage if available or otherwise on the sound pressure. The algorithm is described in chapter 4.4.
To choose the time area of the zoom, the user can click inside the whole time signal or envelope plot. The program takes the instantaneous frequency that corresponds to this time value and calculates the time range of the zoom, which depends on the desired number of periods. The plot shows the chosen time signals starting at half of the calculated length before the time point, to half of the subsequent window. Thus, the chosen time point is always in the middle of the zoom and a constant number of periods of all signals is displayed.

5.6 Spectrum

The calculation of the spectrum is very easily done, because Matlab provides a calculation time optimized function which calculates the Fast Fourier Transform of a time signal. The Fast Fourier Transform is a specialized Discrete Fourier Transform which needs a time signal with length of power of two samples. But the Matlab internal function also guarantees results for signals with different lengths by zero padding. The exact procedure is a secret of the software provider but several tests have shown good results, so this function can undoubtedly be used.
This project offers two different view options: The spectrum of the complete signal or the spectrum of a truncated time frame. For the plot of the whole signal, the Fourier transform is built without consideration for the length requirements of the FFT.
For the spectrum of an excerpt, the time point is chosen by a click on the whole time signal or envelope plot, like in the intelligent oscilloscope. The length of the window is defined by the frequency resolution or the length in samples or seconds.
For both cases, a window function is applied to the time signal to reduce leakage. By default, a Hann window is used, but the filter type can be changed under “Settings”.

5.7 Transformation from Time to Frequency Axis

As mentioned in 5.1.4, for the plots of the envelopes it is useful to have a frequency axis corresponding to the stimulus frequency instead of a time axis.
To get this axis, the pre calculated vector with instantaneous frequencies to each time sample is used and substituted at the x axis values. This replacement is done with the Matlab internal properties of the axes called “XTick” and XTickLabel”. This simple procedure causes a scale with non-equidistant frequency values, which is hard to read. Another option would be to fit the frequency behavior and set the trend of the abscissa to a linear or logarithmic scale depending on the sweep property. This procedure would work very well for sweeps with fixed settings, such as linear or logarithmic frequency increase. But the requirement for this software is to be independent from the test signal, consequently it must be useable for a completely unknown signal as well. So, the

approach with non-equidistant frequency values is chosen and an own routine for the x-axis labeling has been developed.

Figure 70: Envelope of a sound pressure signal with logarithmic sweep

The instantaneous frequency is marked at equidistant time points, but due to the undefined stimulus the frequency values are non-equidistant and uncommon to read. This is of course a little bit annoying because the visual interpolation is difficult. But the use of the zoom function with higher resolution on the frequency axis helps to see the frequency dependency.

5.8 Spectrogram

The spectrogram is a 3-D plot which shows the chronological process of the spectral distribution and is very useful for one-view-analysis. MATLAB provides an already finished function, called "spectrogram" which calculates the spectrogram and the power spectral density of a time signal. Required parameters are the frame length, the overlapping factor and the window function. These properties are adjustable in a separate GUI to customize the spectrogram.

The visualization of the spectrogram is a 2-D plot with abscissa and ordinate and the third dimension is scaled by a color code. Additionally, the amplitudes of the spectrogram are displayed in dB to be more robust with respect to high dynamics.

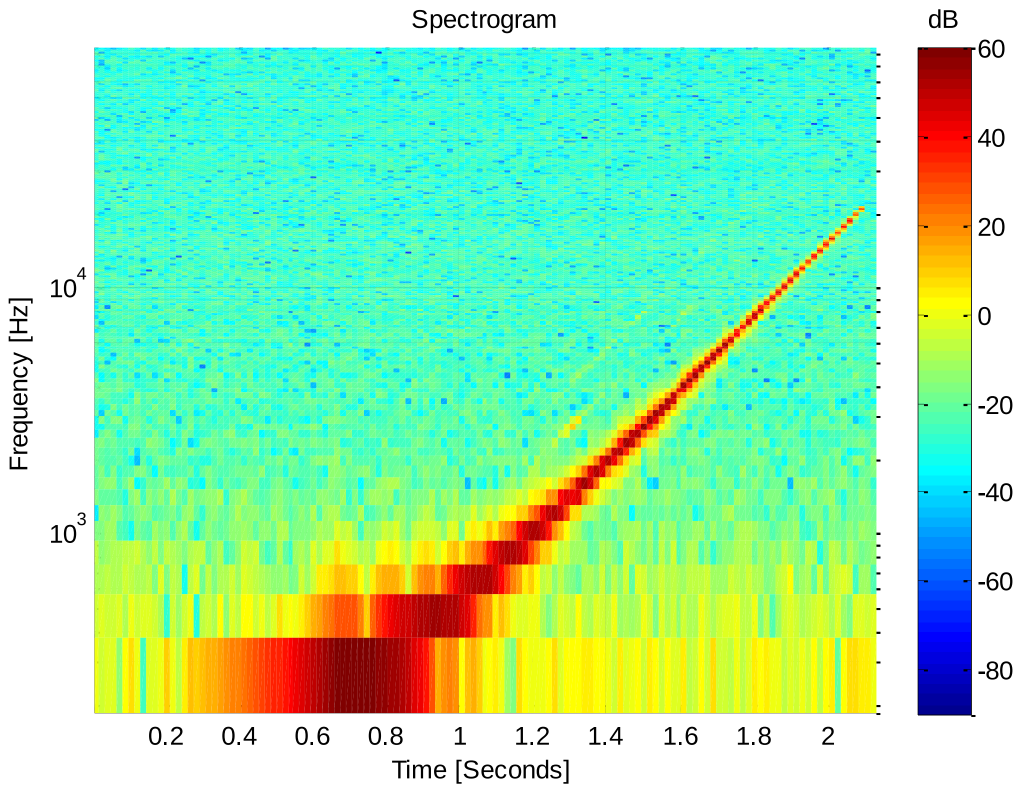

Figure 71: Spectrogram of a sweep measurement with less distortion

5.9 Quadrant Detection

For the root cause detection of irregular distortions, the temporal appearance is very important, as explained in 2.3.1.2. Therefore, the engineer wants to know at which position of the membrane the distortion occurs and if there is a dependency between the maximum excursion of the membrane and the amplitude of the distortion. For the root cause detection, the whole period of the membrane movement can be split into four quadrants that are prominent for certain failure modes. For example, the distortion content in the quadrant with highest negative displacement is a good indication for a hitting voice coil.

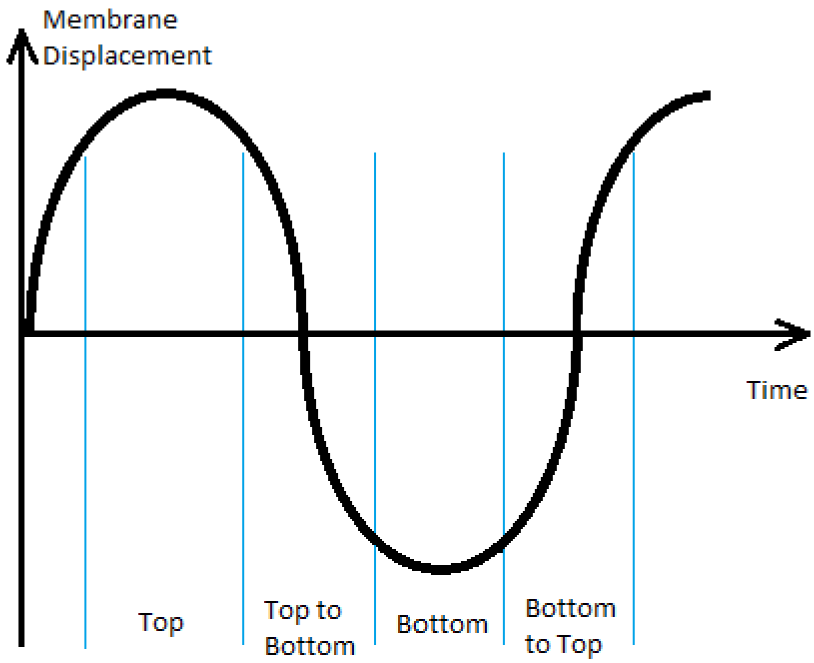

Figure 72: Quadrant Separation according the membrane displacement

The Quadrant Detection is an analysis method which works well for special kinds of defects, like for example a hitting at the top or bottom side or buzzing noise, but the user has to keep in mind that it is not useful for all kinds of defects. Additionally, the temporal separation into “quadrants” is also dependent on the statistical property of the distortion. Thus, the program has different algorithms for stochastic and deterministic defects. The classification of these two types is done in advance, using the following algorithm:

Separation in frames of one period

First, the algorithm from chapter 4.4 is used to split the excursion signal into frames with the length of one period of the excitation signal. If a measured excursion signal is available, then this one is used, otherwise the displacement of the membrane is estimated as described in 4.3. Then all operations for the classification of the distortion can be done period-wise with synchronization to the excursion.

Classification using the crest factor

Statistical properties like RMS and peak value are helpful in distinguishing stochastic and deterministic distortion. The crest factor is the ratio of peak to RMS value and is markedly different for these two types of signals making it a good decision base.
The most extreme kind of distortion is a spike for a deterministic signal and white noise for a stochastic signal.
Because of the theoretically unlimited amplitude range of Gaussian noise, its crest factor is infinite, but in practical applications the range is always limited, e.g. the input range of a measurement system or digital to analog converter. That is the reason why the electrical noise crest factor of the measurement system itself, the recorded ambient noise and pure stochastic distortion, cannot be estimated in advance.
A relative crest factor is therefore implemented, which is the ratio of the crest factor at the distortion divided by the average crest factor between the lowest valid frequency and

the lower cut-off frequency of the Rub & Buzz filter. So, only a relative crest factor is analyzed, and the program is independent from the measurement system in use.
With relative values it is possible to detect areas in the time signals with higher or lower crest factor. Impulsive distortions have generally higher crest factors and deterministic distortions generally have higher impulsive character, so relative crest factors above 0 dB indicates deterministic components, and the Quadrant Detection works with the algorithm described in 5.9.1. For further classification, the algorithm of 5.9.2 is used.

5.9.1 Deterministic Distortions

Some deterministic distortions, as explained above, have equal temporal shape in each period of the stimulus so the root can be determined at a specified phase of the excursion. For these kinds of distortions, the quadrant detection analyzes the bandpass filtered sound pressure signal and detects the maximum of the Hilbert transformation of the Rub & Buzz signal that represents the root of the defect. Theoretically the detection of the beginning of the distortions would be the optimal solution, but because of the zero-phase filtering procedure of the bandpass filter, the distortion is temporally smeared and the use of the Hilbert transformation improves the root detection.
The following steps ensure high robustness for different distortions and have low computational complexity.

5.9.1.1 Envelope

Peak detection of measurement signals is always difficult, because high frequent noise corrupts the result and there is no distinct maximum visible. The Hilbert Transform is therefore used to build the envelope of the signal. This has the effect that all high frequency components are eliminated, and additionally that the polarity of the distortion (positive or negative maximum) is independent.

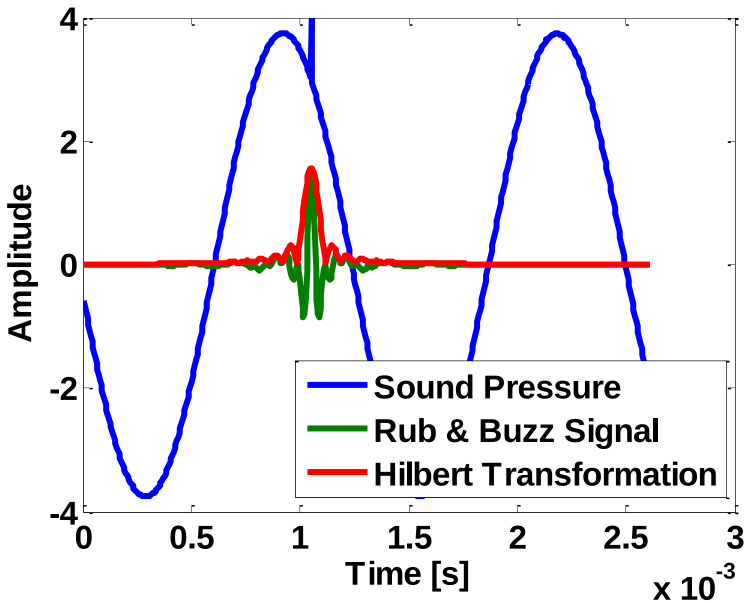

Figure 73: Root detection with Hilbert transformation of the Rub & Buzz Signal for the Deterministic Quadrant Detection

5.9.1.2 Position of maximum

The detection of a local maximum is an important topic in some areas of research, and there are many algorithms available that have different advantages. Here in this section the easiest method can be chosen. Normally the setting of the search area is the most critical parameter, but because of the period-wise calculation, this setting is done automatically. There is only one maximum in each frame and no separation between local and global maximum is necessary. This has the advantage that the simple Matlab function “max”, which is very robust and fast, can be used. Additionally, as explained above, the Hilbert transformation of the Rub & Buzz signal is used for the maximum detection so the sign of the peak doesn’t need to be considered because the Hilbert transformation is always positive.

5.9.1.3 Separation to quadrants

To get an overview of the root cause of a distortion, it is not necessary to know the exact phase of the origin and the allocation to a quadrant is sufficient. In addition, due to the possible long impulse response of the used bandpass filter, an exact phase determination is not trustworthy.
The separation into four “quadrants” for each time frame is done once at the loading procedure, so there is no additional calculation time for this operation. A detailed description of the used algorithm can be found in chapters 4.4 and 5.9.2.1.

5.9.1.4 Diagram

For the optimal visualization of the result, it is helpful to plot four steps, corresponding with the four quadrants, over the time. For time regions with too low distortion level or crest factor, no value is displayed. Additionally the time axis is replaced with the stimulus frequency, as used in 5.1.4, to be able to detect the critical frequency area.

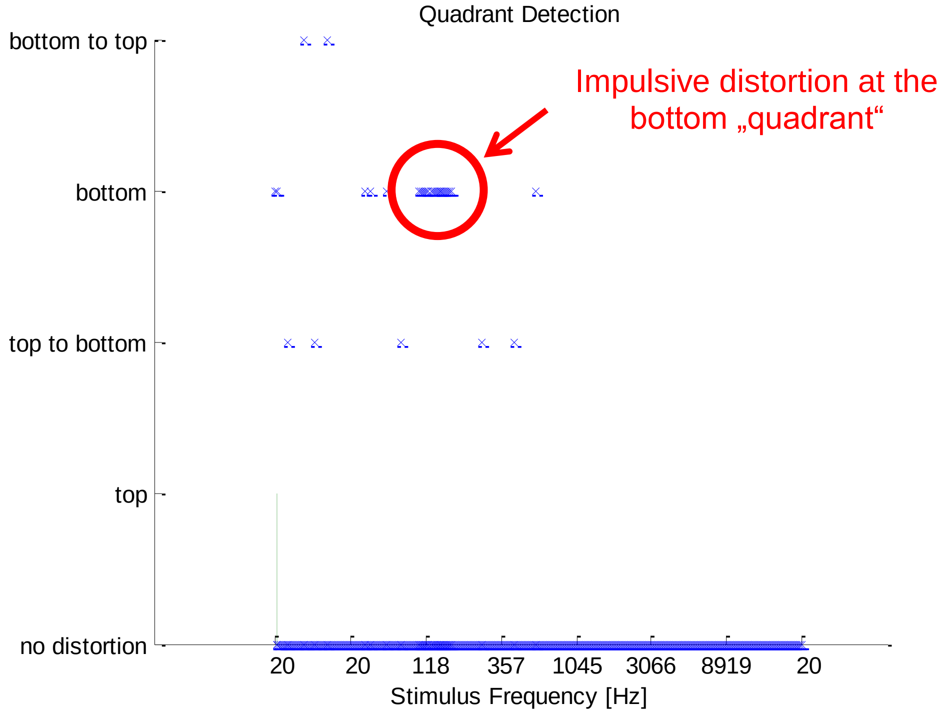

Figure 74: Deterministic Quadrant Detection with a hitting membrane at the bottom side

For most excitation frequencies, no impulsive distortion is detected, but around 180Hz there is a high concentration at the bottom end, which is a sign of a distortion. The other coarsely distributed single markers are errors caused by unstable noise floor and must be ignored.

5.9.2 Stochastic Distortions

Stochastic distortions normally have wide-stretched temporal appearance, so an exact separation of the origin into quadrants does not make sense. Nevertheless, there is a method to get an overview of the distribution over the four quadrants. In general, the algorithm separates the time signal into four pieces corresponding to each quadrant of the membrane excursion and calculates the envelope for each of it. The following steps are necessary:

5.9.2.1 Instantaneous Phase Detection

The bandpass filtered sound pressure signal is already truncated in frames with lengths of one period of the stimulus, but due to the unknown signal, respectively the temporal trend, the frame length is not always identical to the length of the stimulus period. Consequently, the phase of the stimulus within the frame is unknown, thus a subdivision into four sections and always starting at the same position doesn't work.

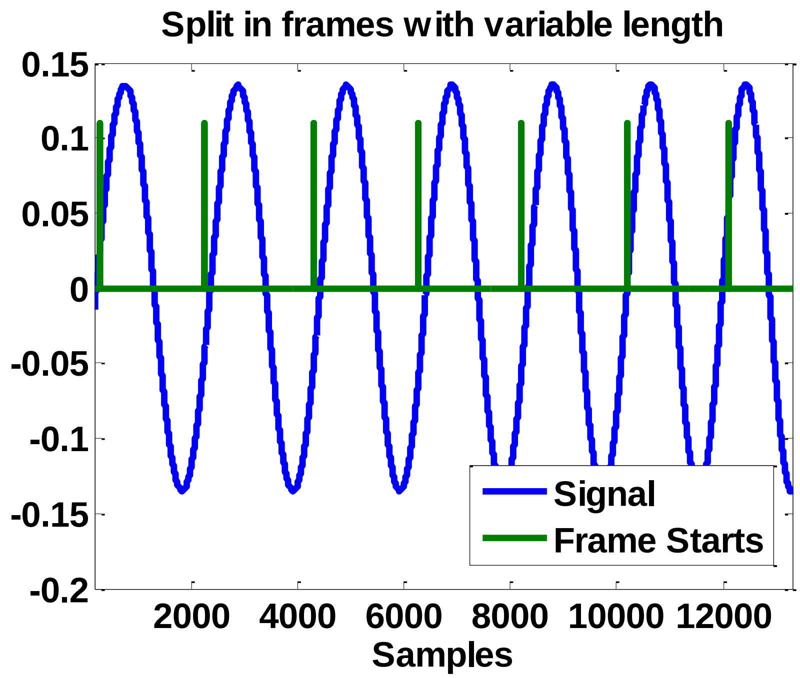

Figure 75: Frame starts with the variable frame calculation

The instantaneous phase of each frame needs to be calculated and the whole period can be separated starting with the correct phase information. The description of this algorithm can be found in chapter 4.4.

This knowledge allows the subdivision of the sound pressure signal into four quadrants of the excursion independent from the phasing of the frames.

5.9.2.2 Energy envelope

To get a set of curves that show the energy spreading of the distortion over the quadrants, the envelope of each time signal has to be built and plotted. The used algorithm is described in 4.2. For the analysis of the energy distribution, the RMS value is the better choice, so the envelope of the RMS value is determined.

5.9.2.3 Diagram

The diagram is principally the same as in chapter 5.1.4, but now the Rub & Buzz signal is split into four curves; one for each quadrant.

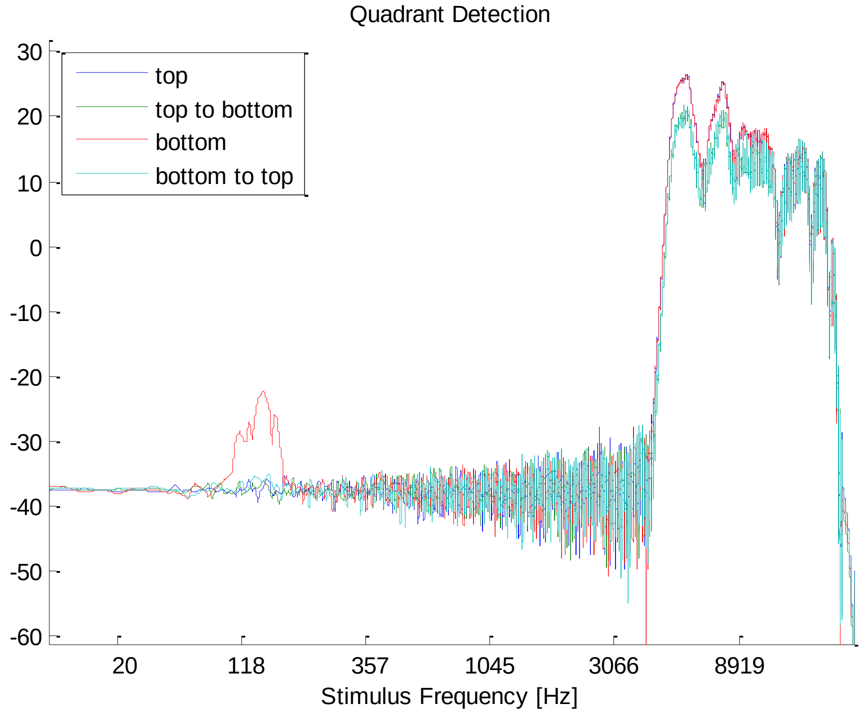

Figure 76: Stochastic Quadrant Detection with a hitting membrane at the bottom end

5.10 Settings

The target of this thesis is to develop a tool which supports the acoustic engineer with failure analysis during the loudspeaker development. It is therefore important that the engineer can use the software without long training, so most of the settings have to be predefined in a way that is applicable for most analyses. But to achieve high flexibility, it must be possible to adjust all important parameters so the tool can also be used for specialized analyses with unusual conditions.

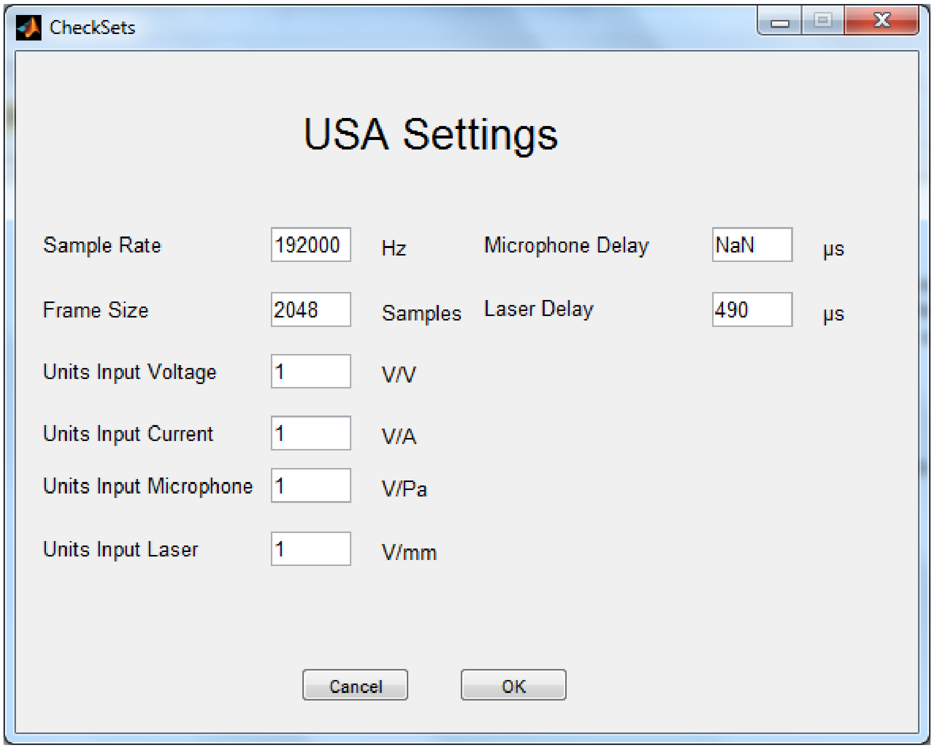

Figure 77: Settings panel for the Universal Sweep Analyzer

Sample Rate:

The chosen sample rate must be the same as in the acquired measurement signal, otherwise time stretching is applied. This value is normally updated at the signal import.

Frame Size:

This frame size is used for all internal block-wise calculations, e.g. STFT and Instantaneous Frequency detection. The default value is 2048, but it can be changed for detailed frequency or time resolution.

Units Input:

For correct axes scaling, the user can set a calibration value for each channel. For self-calibrating systems, these values can be exported: further details can be found in the manual.

Microphone Delay:

This analysis tool is very sensitive for delayed signals; therefore the acoustic elapsed time between loudspeaker and microphone must be compensated for. This procedure is done automatically when the measurement signal is imported and the value is shown in this field. For unusual applications with abnormal acoustic expansion, the automatic delay detection might be wrong and can be set manually.

Laser Delay:

For a correct delay compensation of all channels the additional delay of the excursion acquisition chain must be compensated too. The extra time of the laser to the other acquisition channels can therefore be measured or extracted from the datasheet. In this project a Keyence laser that has an additional delay by 490µs has been used, so this value is set as default.

6 Distortion Analysis using the Universal Sweep Analyzer

This chapter gives as instruction for each method of analysis including an explanation of how to reach a correct and distinct decision.
To get a complete description of the behavior of a loudspeaker, the electrical, the mechanical and the acoustic domains need to be determined. The measuring of all of these signals is often very complex, and of course sometimes impossible, just imagine measuring the membrane displacement in a situation with closed box with front port and mesh. Hence some assumptions are used to reduce the complexity of the acquisition procedure, and analogies of the loudspeaker are used to estimate the missing signals.

The detailed description of the measurement chain can be found in 3.2.1, but this chapter gives a short introduction of the procedure to get a set of measurement data.

First of all, the user must be clear about which measurement situation is acceptable. Either the loudspeaker is measured in a baffle with a microphone on axis to get a good representation of the half-space measurement, or a laser is used on axis to measure the membrane displacement while a free-field microphone only acquires the sound pressure off axis with free-field conditions. It is of course possible to combine both measurements in one, using a laser on axis in a baffle set up, but then a very small microphone must be used to minimize the reflections and guarantee a half-space measurement condition.

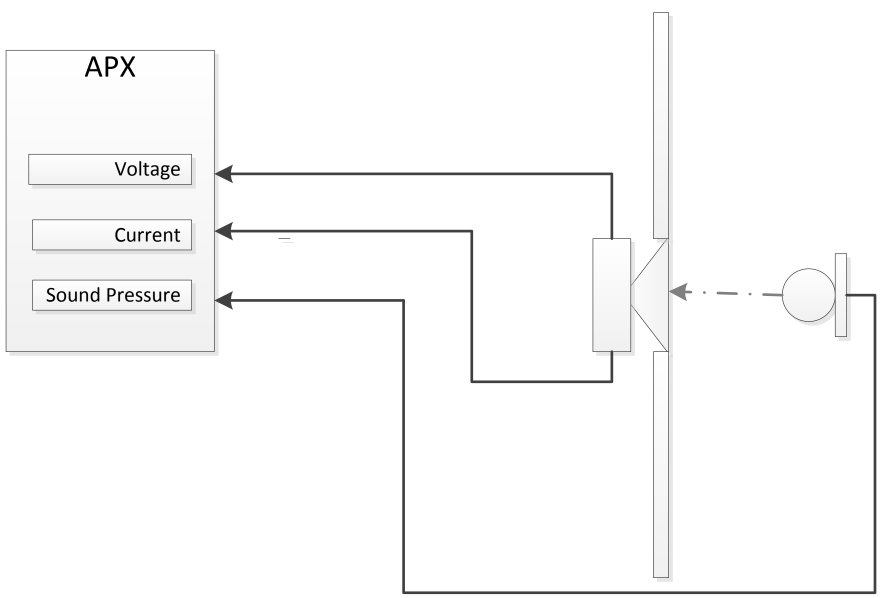

Figure 78: Setup for a baffle measurement

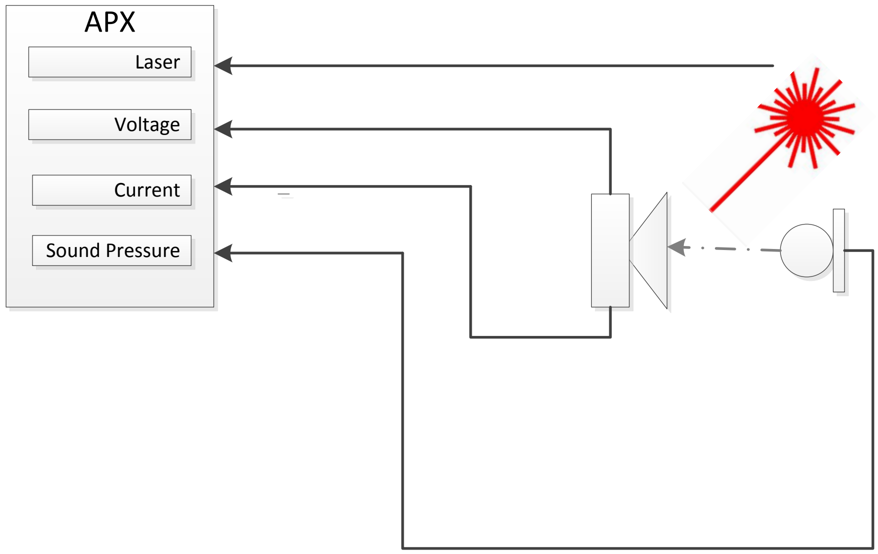

Figure 79: Setup of free-field measurement with laser

A proper signal for the measurement signal is either a stepped sine or a continuous sweep. The power has to be sufficiently high to excite the distortion of interest, but it shouldn't be too high to ensure a normal use without an absolutely distorted output signal.

The following list shows the procedural methods using the Universal Sweep Analyzer for all kinds of distortions.

6.1 Regular Distortions

Due to the limited choices in designing a loudspeaker, especially for micro-speakers, the regular acoustic output of a transducer already shows lots of distortions. This kind of distortion is often relatively high, and an inexperienced engineer can misinterpret it as a defect. Plots from a common loudspeaker without irregular defects are shown in the following figures to get an impression of regular distortions.

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	100mV RMS
Microphone Distance	1cm
Modification	Original setup

Table 3: Settings of a loudspeaker measurement in normal use

As described above, the loudspeaker was measured with APX and the data was imported into "USA" with a ".mat"-file in the format described in section 5.2.1. Here is the overview of all diagrams available in the software.

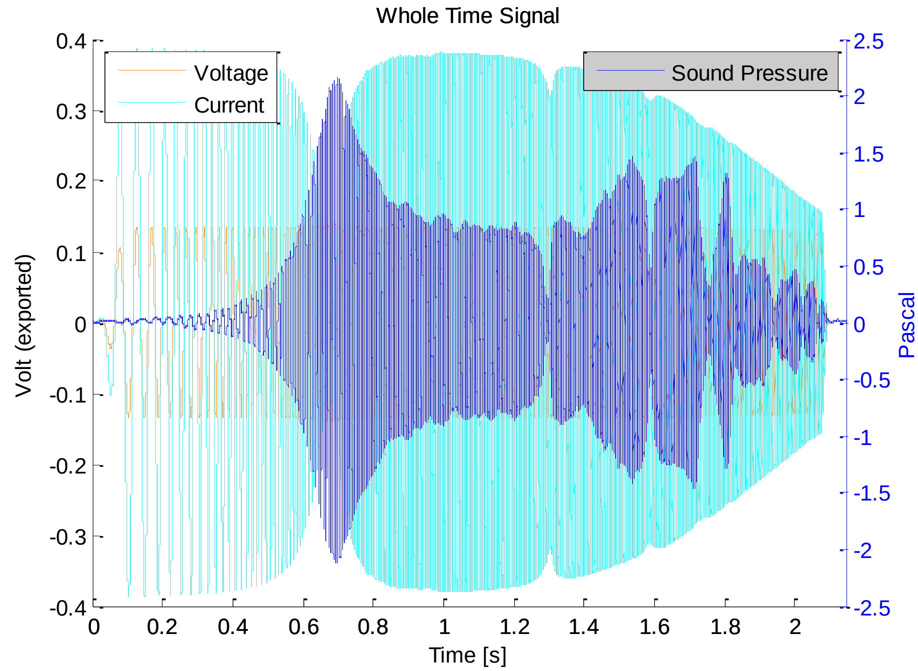

Figure 80: Time signal of a loudspeaker without defect

The time signal plot is handy to get a first impression of the measurement and to check the plausibility of all signals – especially if the signal of one channel was clipped during the measurement because of an improper input range selection.
Further analyses with the whole time domain signal are impracticable, so the Universal Sweep Analyzer provides a specialized zoom function.

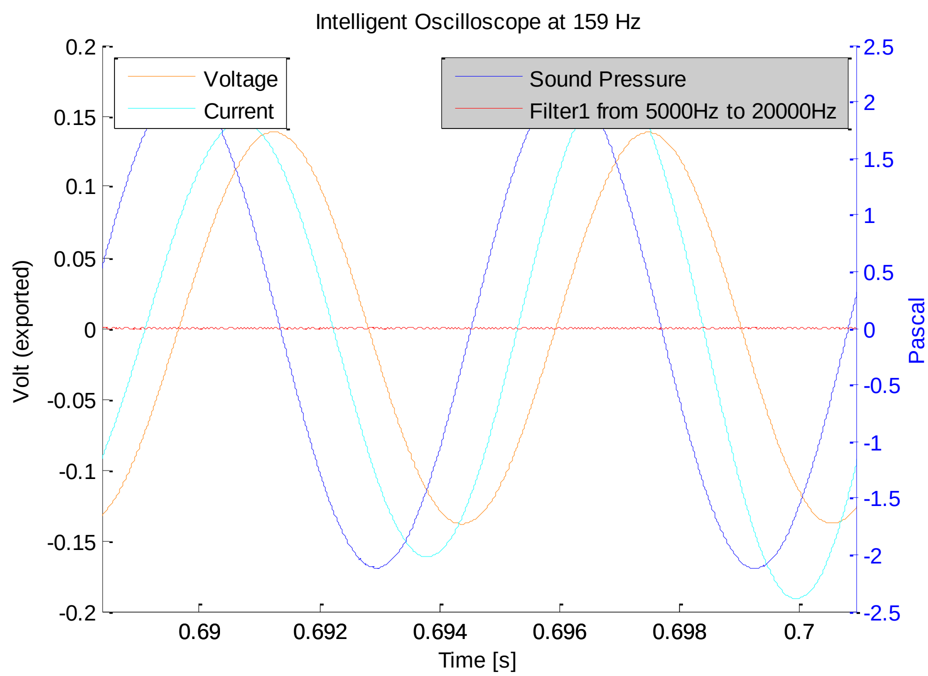

Figure 81: Intelligent Oscilloscope at the resonance frequency

For a detailed inspection, the intelligent oscilloscope assists in displaying a zoom of the time signal making the number of periods in the plot independent from the frequency of the stimulus.
This view is helpful for the analysis of phase relationships or the shape in the time domain in a certain frequency range, but to get an overview of the complete measurement other visualizations might be better.

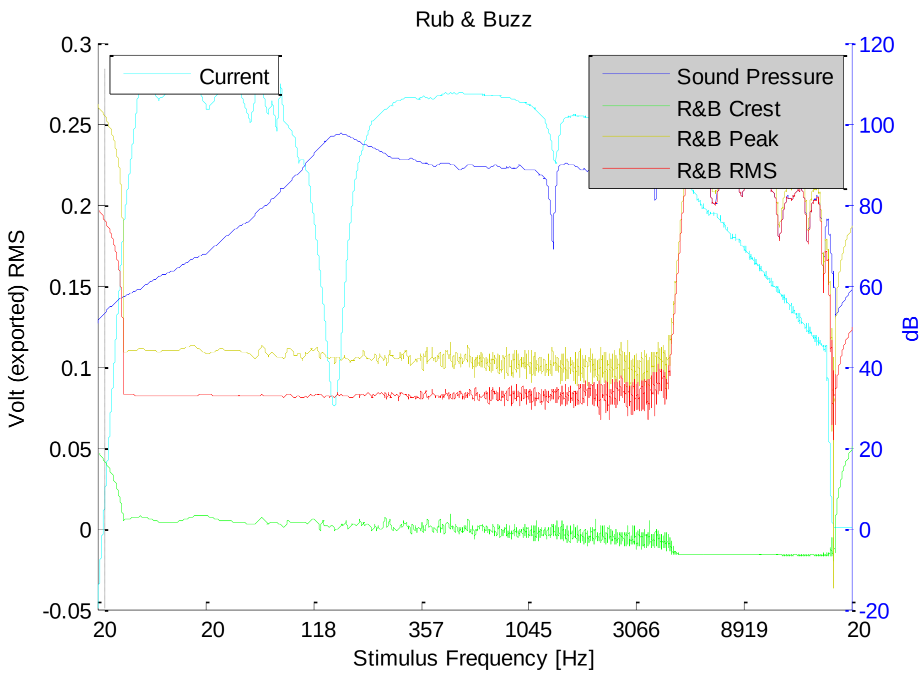

Figure 82: Envelope of time signal over instantaneous frequency axis

The envelope of the time signal is another visualization method of the frequency response and is a good indication for the behavior in several frequency ranges of a loudspeaker.
Additionally, statistic parameters of the so-called Rub & Buzz signal are displayed which are helpful for the detection of defects, as described in chapter 3.2.6.
Here you can see the data set of Crest, Peak and RMS in the pass- and stopband of the filter for a normal loudspeaker without any defect, tested with the measurement system of the company Sound Solutions International.

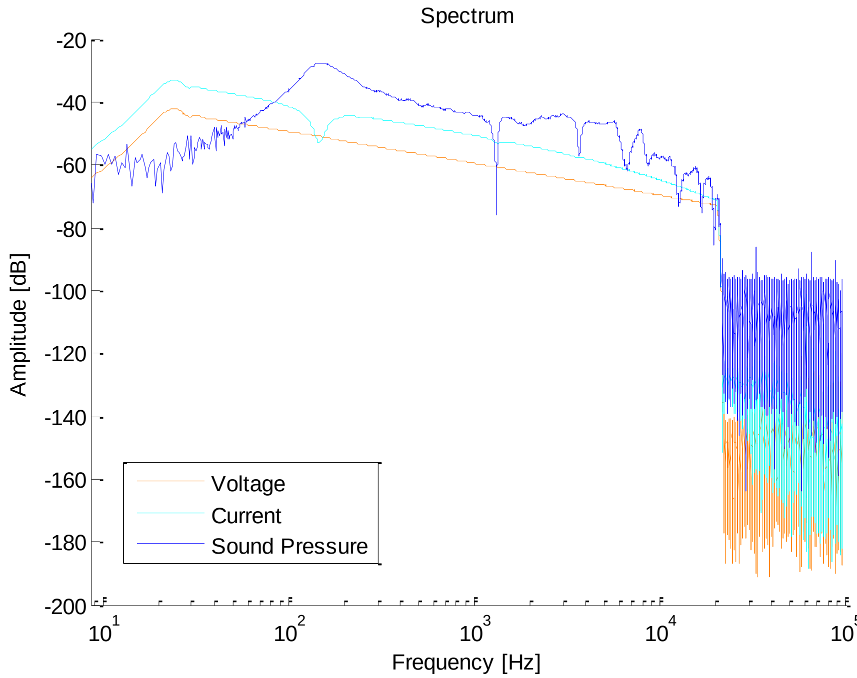

Figure 83: Spectrum of whole signal

In the spectrum, the frequency responses for the electrical and acoustical domain are visible. Additionally, the spectrum of the stimulus can be displayed to see the valid frequency range with sufficient SNR.

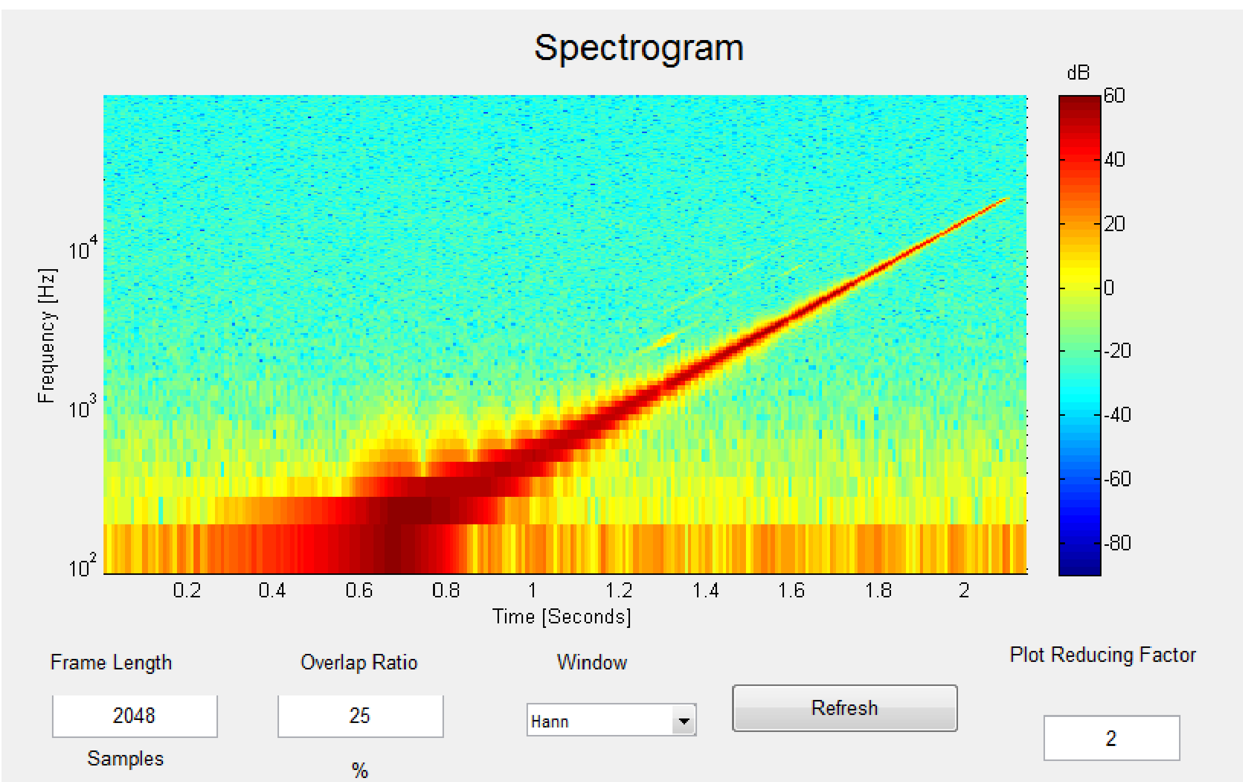

Figure 84: Spectrogram of sound pressure signal

In the spectrogram, the fundamental of the output signal is clearly visible. In addition, some harmonics in a certain frequency range are distinguishable, but the amplitude is very low. This is a good indication that the loudspeaker is driven in the small-signal behavior without irregular distortions.

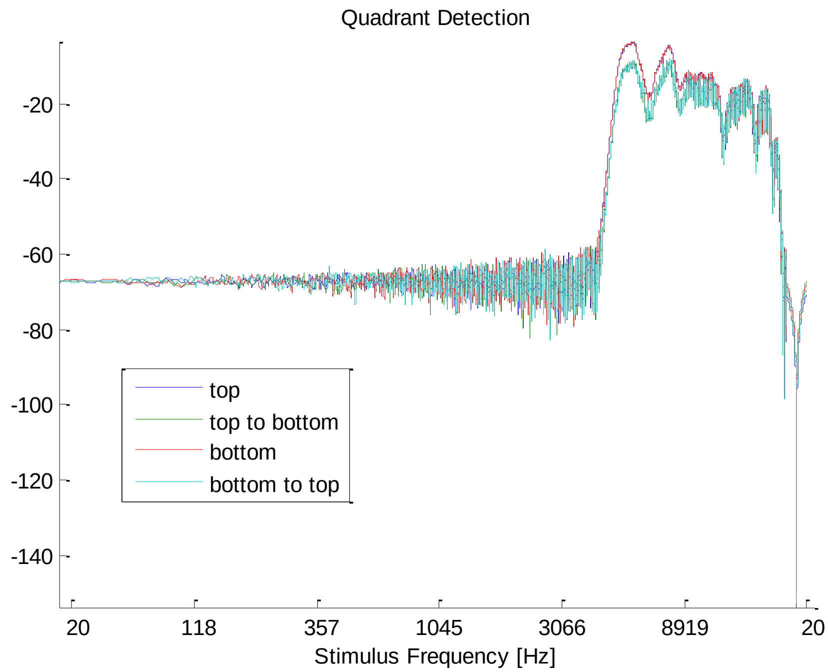

Figure 85: Stochastic Quadrant Detection

The Quadrant Detection splits the R&B RMS curves into four parts, one for each quadrant. If all four curves are comparable, it indicates that the filter signal has a constant energy over all four quadrants. This behavior implies that the speaker has no displacement region with abnormal sound pressure, consequently hitting at the top or bottom or a rubbing voice coil can be excluded as possible root causes.

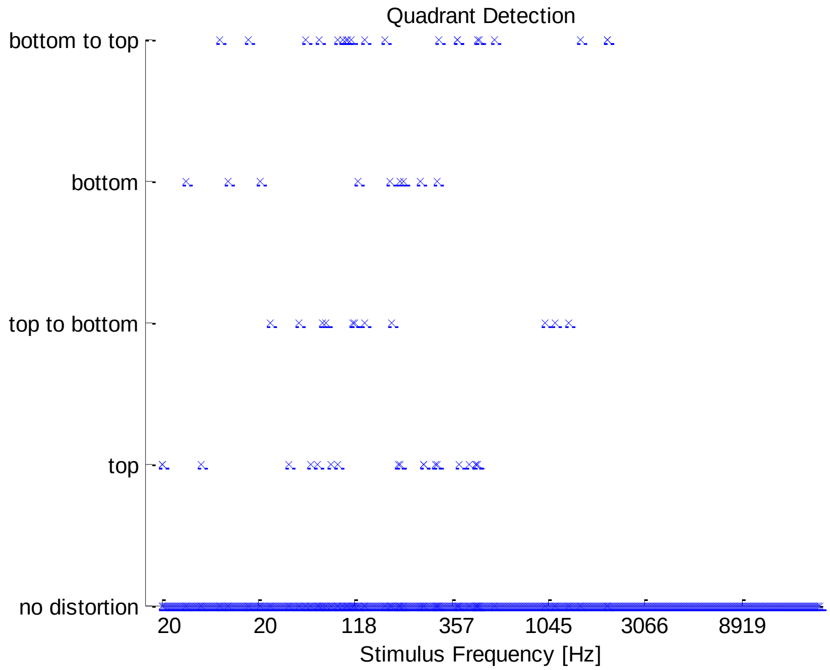

Figure 86: Deterministic Quadrant Detection

At the Deterministic Quadrant Detection, most of the periods show no distortion (relative instantaneous crest factor below 0dB) which already indicates that there is no impulsive defect, but there are some outliers which can be ignored because they seem to be randomly distributed.

6.2 Hitting Membrane at the Back

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	100mV RMS
Microphone Distance	1cm
Modification	Toothpick next to the membrane at the back

Table 4: Settings of a loudspeaker measurement with a hitting membrane at the back

In principle, it is easy to generate a hitting voice coil at the bottom side because normally the space at the back is the overall hard excursion limit, and if the loudspeaker is driven with too high power this limit will be reached. But at this driving point the loudspeaker already has lots of other distortions, and it is hard to separate regular and irregular distortions.

Figure 87: A toothpick next to the membrane at the backside to generate a hitting membrane

To get a limited membrane excursion without lots of other distortions, a toothpick was placed next to the membrane. This gave the membrane limited excursion space, similar to a hitting voice coil at the pot, but with very low displacement.

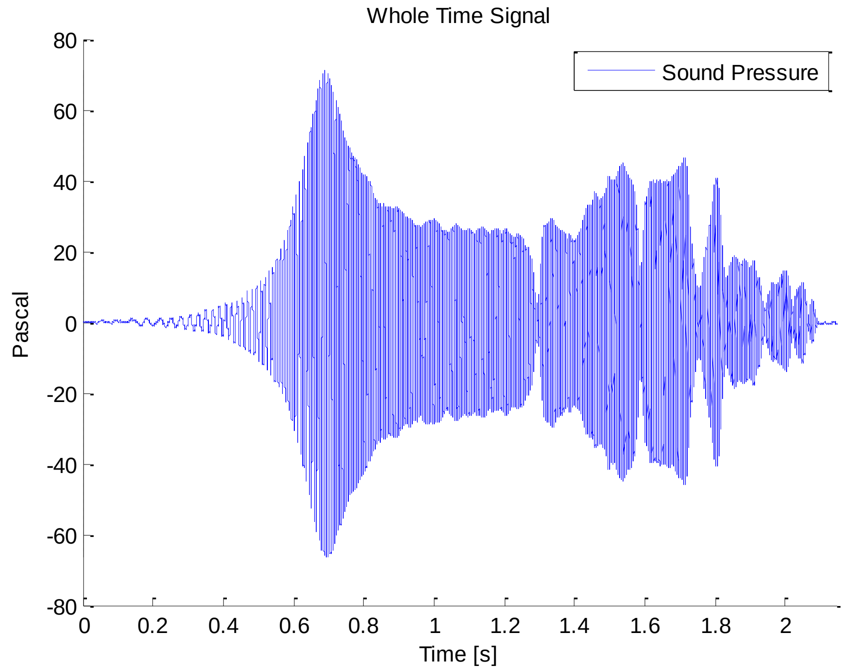

Figure 88: Time signal of loudspeaker with toothpick at the back

It is hard to distinguish any distortion when using the time domain signal only; the sound pressure looks comparable with the unmodified loudspeaker, but during the measurement the Rub & Buzz was clearly audible.

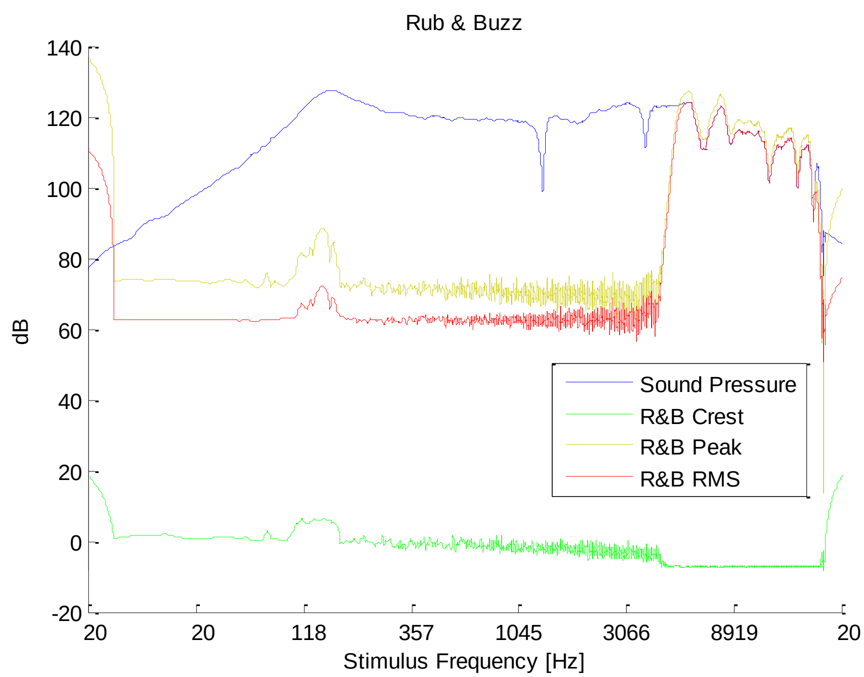

Figure 89: Envelopes of loudspeaker with toothpick at the back

In the sound pressure envelope there is again no visible change, but the Rub & Buzz curves show a region of about 118Hz with increased distortion. This is the frequency range with the highest excursion (at the resonance frequency) so it indicates an

excursion driven defect. The rise in the Rub & Buzz RMS value shows that there is additional sound pressure in the pass band of the Rub & Buzz filter. This is a good indication that within this frequency range, some irregular distortion occurs.
Additionally, at these frequencies the Rub & Buzz Crest and Rub & Buzz Peak values are increased, which means that the distortion is impulsive. This information helps for the detection of the root cause because the engineer can already eliminate some possible errors.

To get an impression of the shape of the distortion in the time domain the "Intelligent Oscilloscope" feature can be used for a detailed view in certain interesting stimulus frequencies. Here is a representative view at 140Hz.

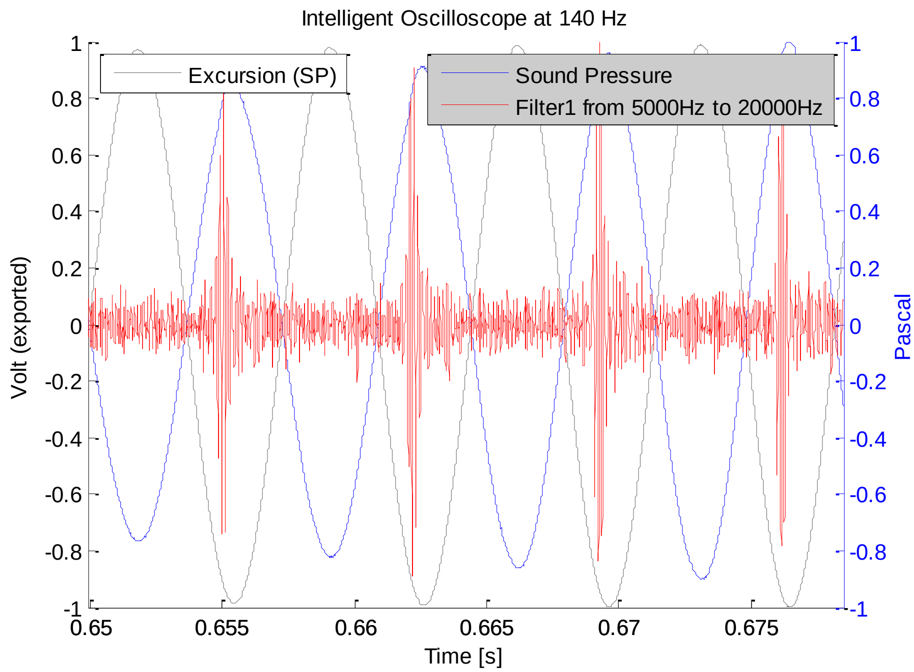

Figure 90: Intelligent Oscilloscope at 140Hz of a loudspeaker with toothpick at the back

For a better understanding of the critical position of the membrane, the calculated excursion signal can be used. In this example the Rub & Buzz signal was scaled for a better visualization of the band pass filtered signal compared to the original, so the origin of the failure is clearly visible although the level of the distortion is very low.

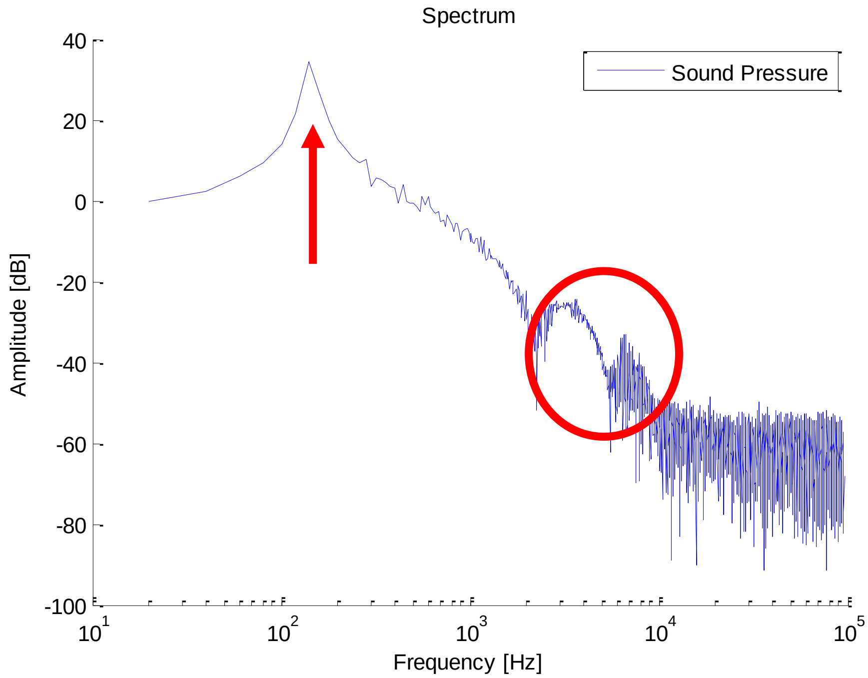

Figure 91: Spectrum at a time point where Rub & Buzz occurs

In Figure 91 the spectrum of the time range when the stimulus frequency is 150Hz. It is the moment when the membrane hits the toothpick. Non-harmonic distortions from 2 to 8 kHz are clearly visible and the spectrum indicates the critical frequency range of the distortion.

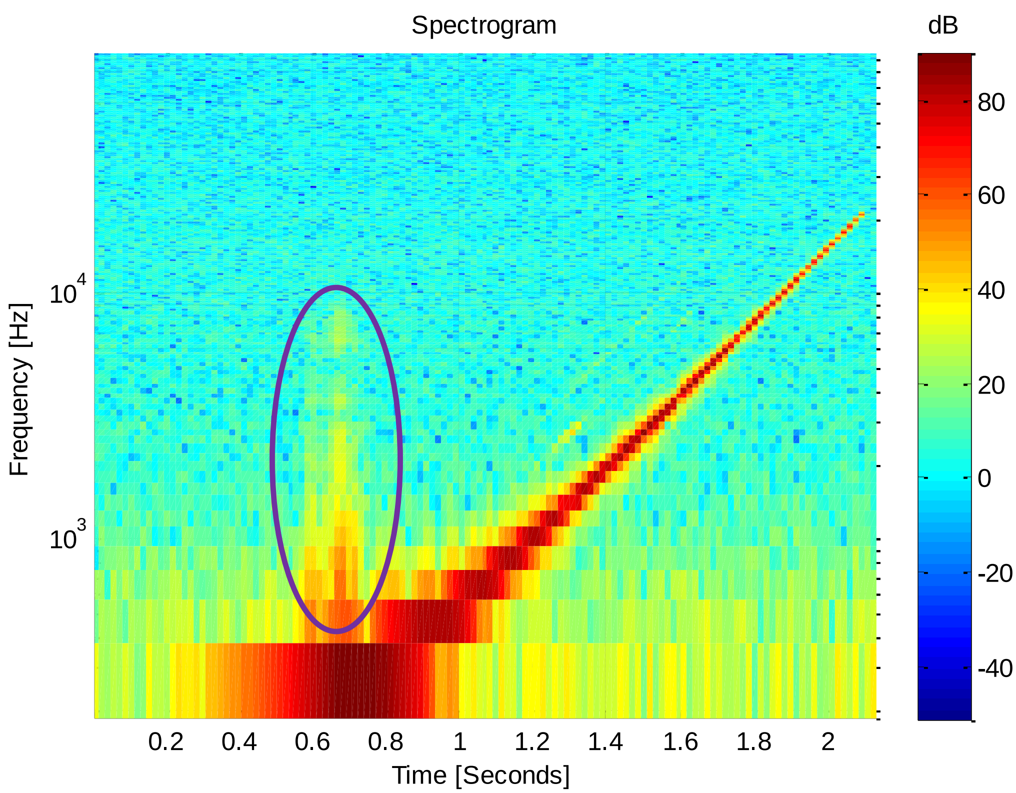

Figure 92: Spectrogram of a loudspeaker with toothpick at the back

The spectrogram shows the critical time region and is useful for a first impression, but for a detailed statement other tools are more helpful.

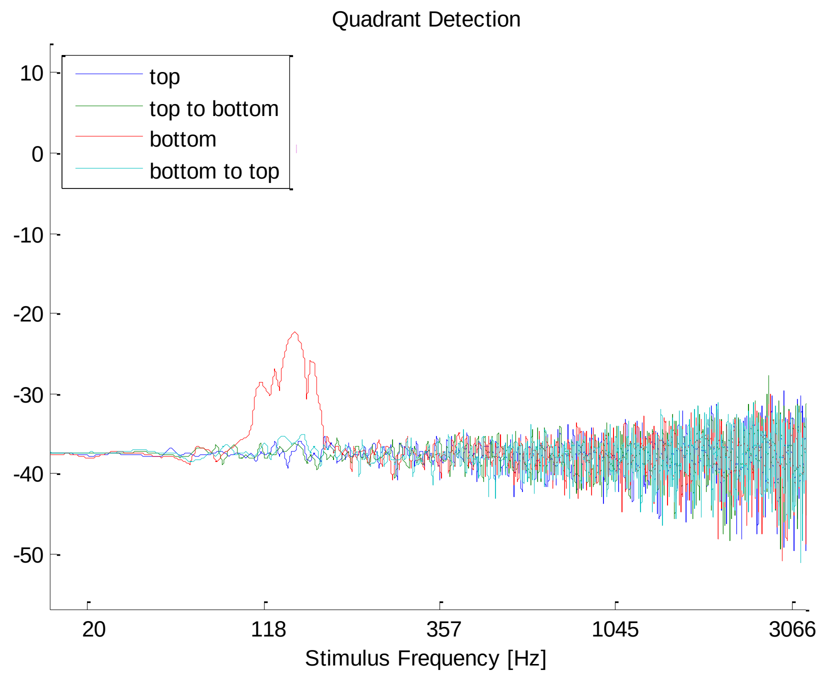

Figure 93: Stochastic Quadrant Detection of a loudspeaker with toothpick at the back

This analysis shows that the distortion appears only in one quadrant of the membrane displacement – when the membrane is at the bottom. This statement is very useful for the engineer because the appearance of the distortion limits the possible root causes.

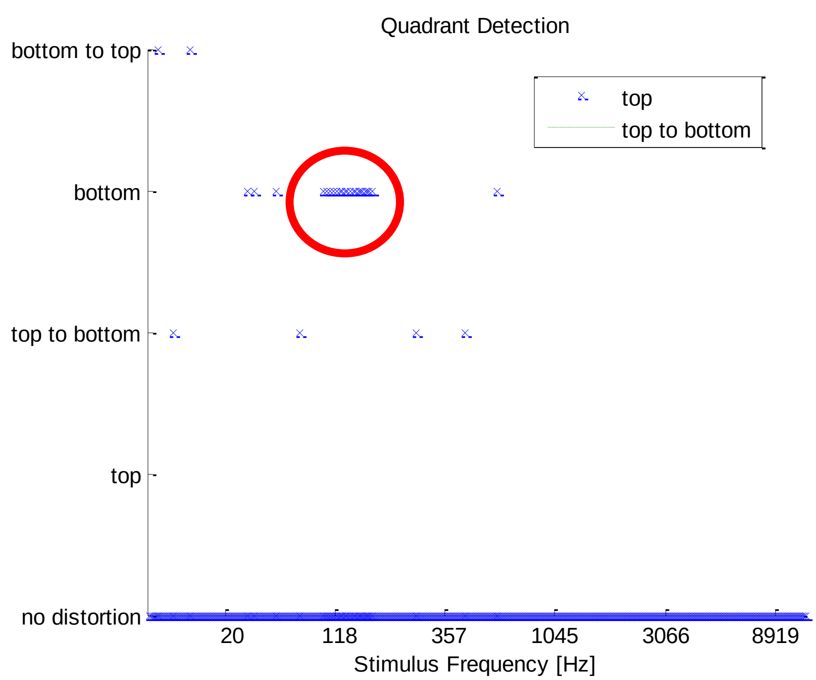

Figure 94: Deterministic Quadrant Detection of a loudspeaker with toothpick at the back

At the frequency range of around 120Hz, the Deterministic Quadrant Detection shows a distortion at the bottom quadrant where many markers are at the same quadrant. Due to bad SNR of sound pressure measurements in the baffle, especially at low frequencies, the Quadrant Detection displays false errors. This is caused by the high variability of the noise's Crest factor. If there is only noise in the Rub & Buzz signal then it is possible that some periods of analysis have much higher Crest factor than the average, consequently it is detected as a deterministic distortion. So the engineer has to take care when interpreting this result and should consider groups of markers only.

Taking all analyses into account, the engineer can detect the root cause very easily. For a first impression, the envelopes of the Rub & Buzz signals are helpful, and in this case the critical frequency range and the impulsivity of the distortion give a first hint of the root cause. The spectrogram and the spectrum support the analysis of the spectral distribution of the distortion. This is important for finding the best settings of the Rub & Buzz filter, especially if such a filter is used for the quality control in mass production. Using the Quadrant Detection, the engineer knows the appearance of the distortion according to the displacement of the membrane and so the root cause should be distinguishable. Summarizing all diagrams it is clear that the root cause in this example is hitting membrane or coil at the bottom.

6.3 Hitting Membrane at the Front

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	500mV RMS
Microphone Distance	1cm
Modification	Toothpick next to the membrane at the front

Table 5: Settings of a loudspeaker measurement with a hitting membrane at the front

Similar to the trial with a toothpick at the back, a hitting membrane at the front was generated. A microphone stand was placed in front of the membrane, and with too high displacements the membrane will hit and distortion will be generated.

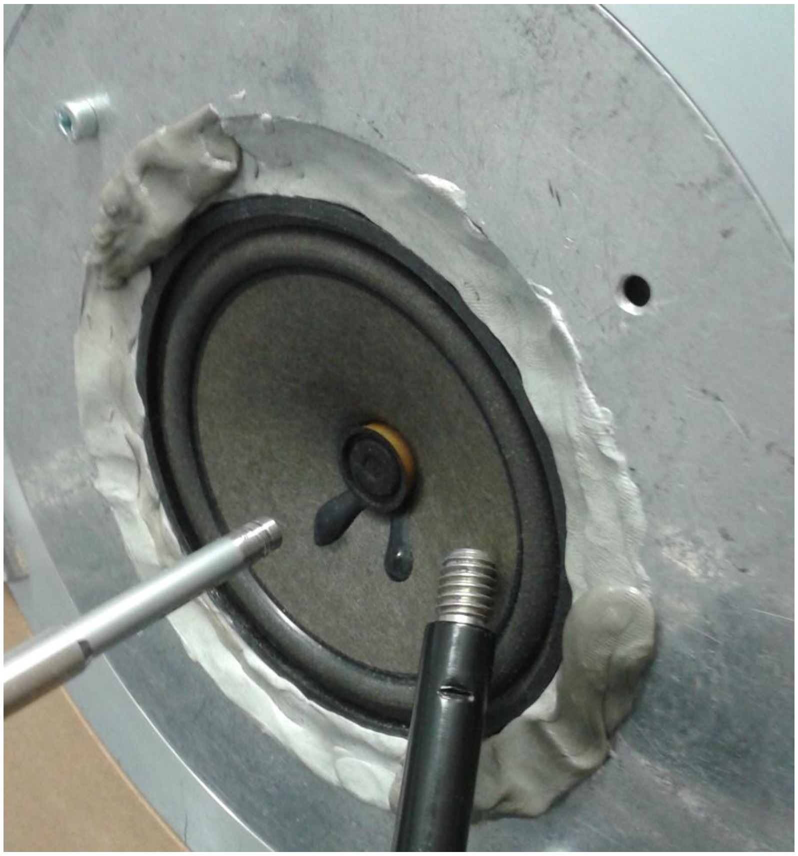

Figure 95: A microphone stand was placed in front of the loudspeaker to generate a hitting membrane

This setup facilitates a hitting membrane with low excursion to generate irregular distortions without big regular distortions, which might be a problem in getting a clear picture.

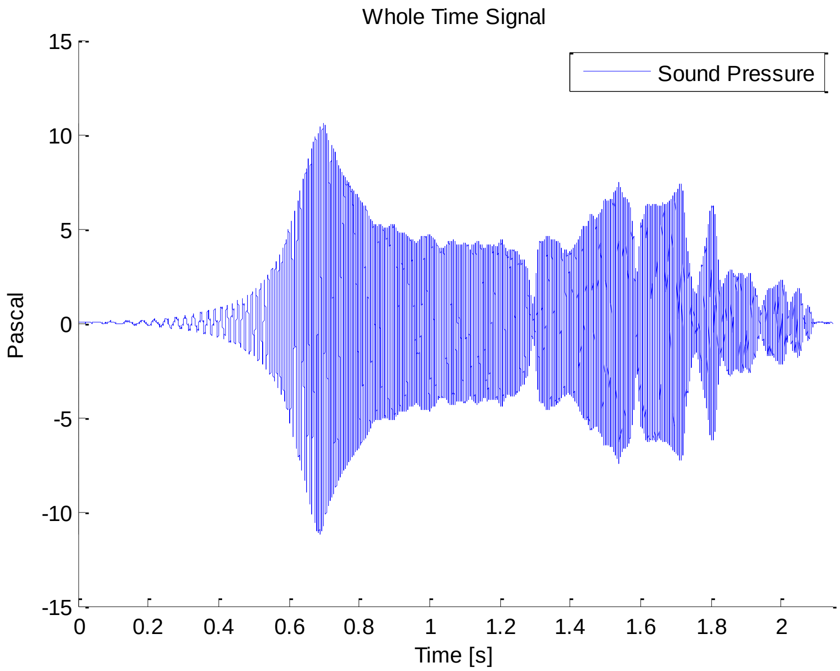

Figure 96: Sound pressure time signal of a loudspeaker with stand at the front

The time signal doesn't show any abnormalities and is comparable with regular use, so this view is not helpful for the Rub & Buzz detection.

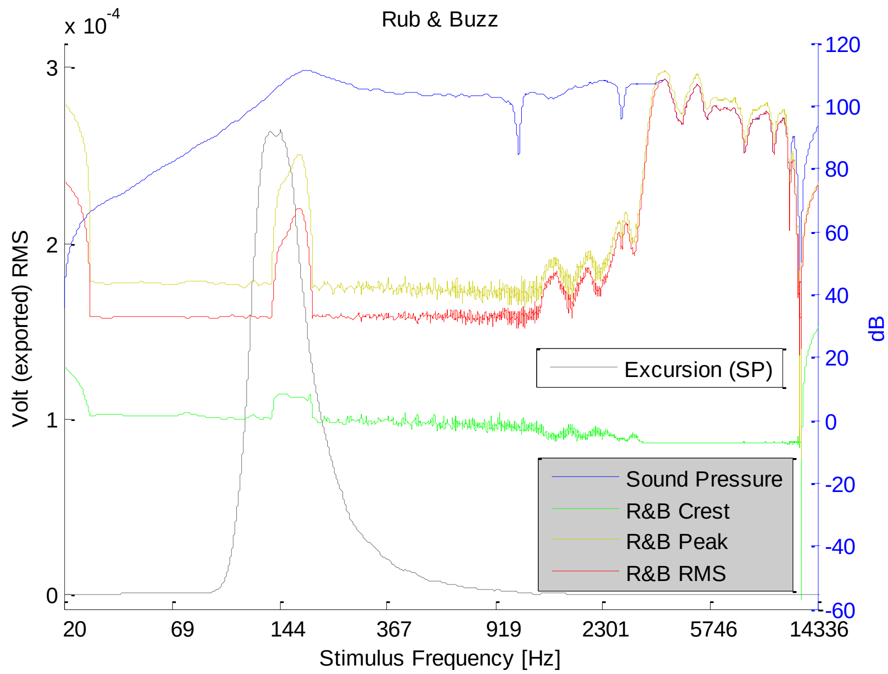

Figure 97: Envelopes of a loudspeaker with stand at the front

The rise in the Rub & Buzz signals at ~144Hz is a clear indication that there is an irregular distortion. Taking the calculated excursion signal into account it is evident that the loudspeaker has the highest displacement in this frequency range, so the distortion might be excursion driven.
Due to the high R&B crest factor at this region (maximum at 8dB), the distortion must be impulsive, something which already indicates that the membrane is hitting another part.

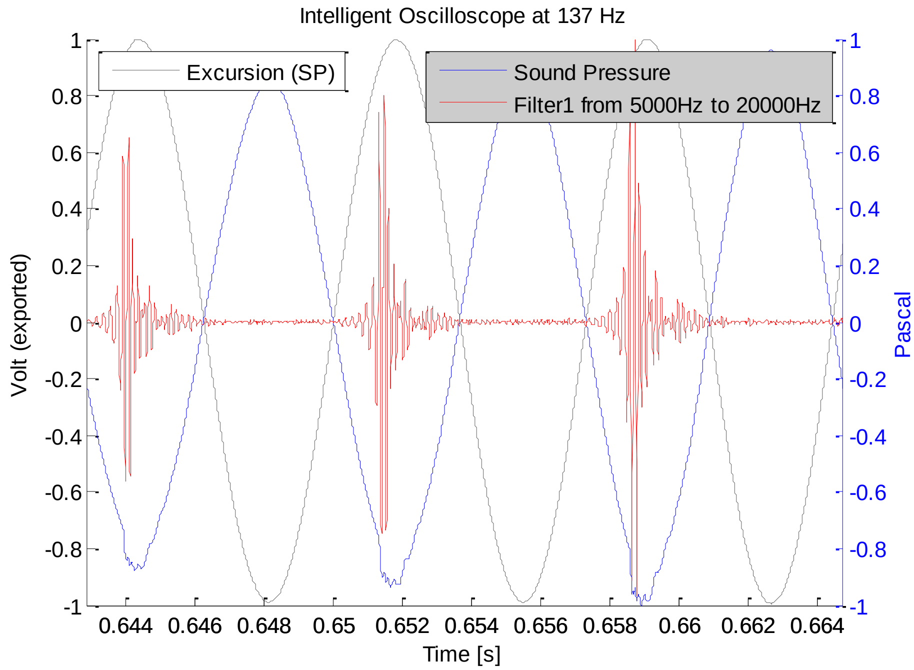

Figure 98: Intelligent Oscilloscope of a loudspeaker with a microphone stand at the front

This result is comparable with the toothpick at the back, but concerning the excursion signal it is clear that the distortion appears when the membrane is at the front. The exact shape of the filtered sound pressure signal might be confusing because of its transient effect, but this is caused by the bandpass filter only and can be improved with low order filters. One only needs to consider that low order filters can have problems separating regular and irregular distortion if the low order harmonics are at a comparable frequency range to the irregular distortion.

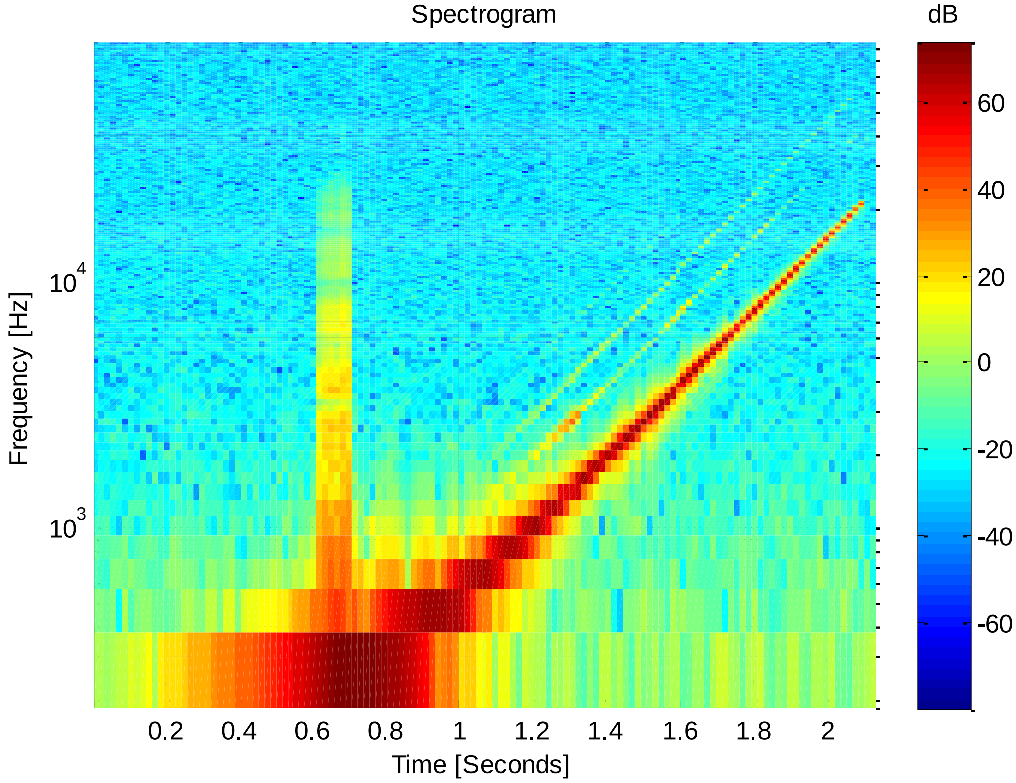

Figure 99: Spectrogram of a loudspeaker with stand at the front

The distortion is clearly visible, but for a detailed analysis this plot is not very useful. The spectrum of a short time frame when the noise occurs shows a better picture of the spectral shape.

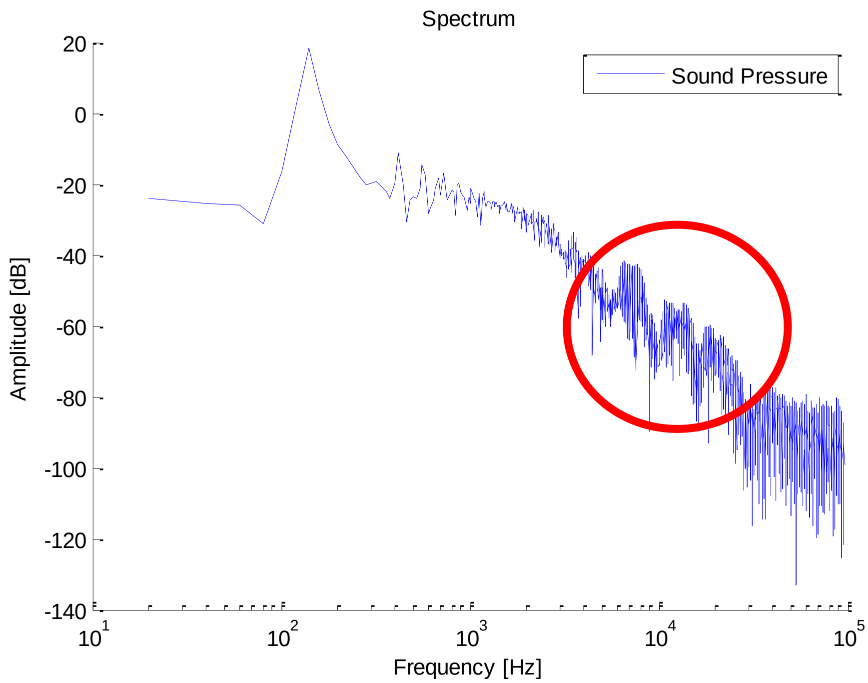

Figure 100: Spectrum of a loudspeaker with stand at the front when then R&B occurs

The increased sound pressure from 7 to 12 kHz indicates the spectral shape of the distortion. This frequency range should be used for a filter during production quality control.

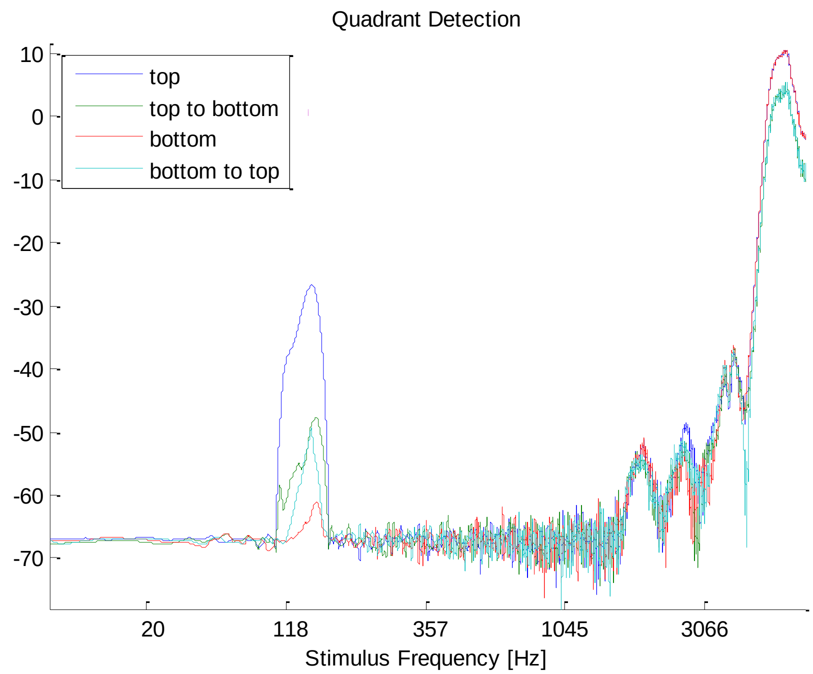

Figure 101: Stochastic Quadrant Detection of a loudspeaker with stand at the front

The distortion appears mainly in the top quadrant, but also in the adjacent quadrants. This is probably caused by the impulse response of the Rub & Buzz filter and must be ignored for the root cause detection. Lowering the filter order can help get better separation if the distortion has signal content at high frequencies only, but in this example the Rub & Buzz is only at slightly higher frequencies than the regular distortions.

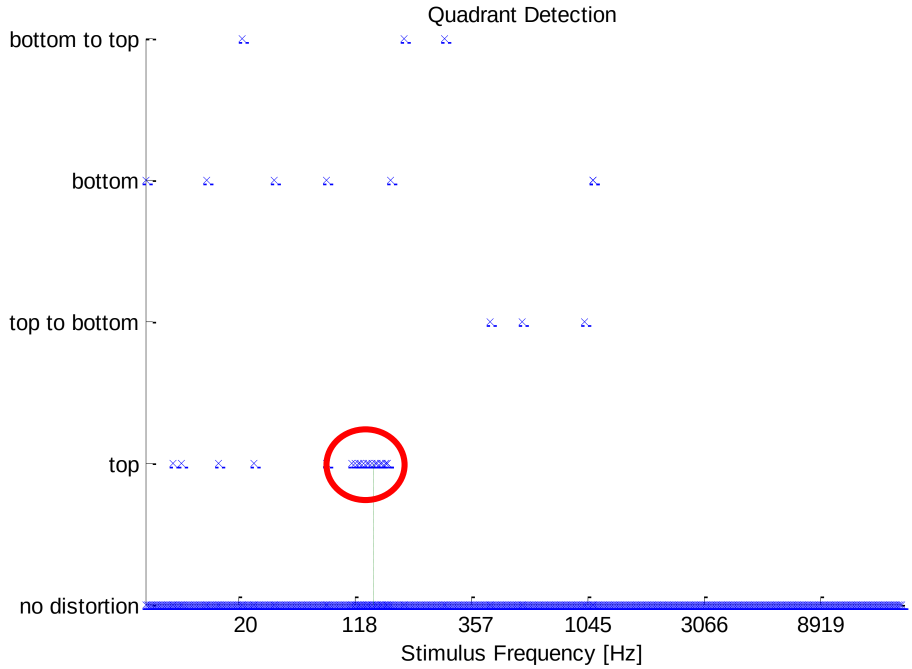

Figure 102: Deterministic Quadrant Detection of a loudspeaker with stand at the front

At about 120Hz there is a high concentration of markers at the top which indicates deterministic distortion in this region, and all other markers are false detections caused by bad SNR. This can be concluded because otherwise the failure mode would have extremely short transient effect, something which is not reasonable.

Summarizing all analyzes, it is clear that the distortion is impulsive, appears only in the top quadrant and is excursion driven, so the root cause must be a hitting part at the front.

6.4 Rubbing Voice Coil

Loudspeaker	Micro-speaker 18 x 13 x 2.5 mm
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	100Hz – 20kHz
Voltage	200mV RMS
Microphone Distance	1cm
Modification	Putty on one edge of the membrane

Table 6: Settings of a loudspeaker measurement with a rubbing voice coil

Loudspeakers for PA or industry applications are designed to be very robust against mechanical stress, so normally the air gap between voice coil and magnet or pot is quite large. This has the effect of reduced B*l, consequently low sensitivity, but for the customer the robustness is often more important.

Thus, for the generation of a representative rubbing voice coil, this type of loudspeaker is not very suitable since a high asymmetry for magnetic field, mass or stiffness needs to be applied.
Micro-speakers are designed with a tiny air gap to get as much sensitivity as possible, but for the centering of the coil only a suspension without a spider is used. For this type of loudspeakers, a rubbing voice coil is a common failure mode and is easily generated. Due to the limited space for micro-speakers, the design causes high nonlinearities that might lead to misinterpretation, so the acoustic engineer must be familiar with the typical distortions for this kind of speaker to be able to detect the irregular distortions. Therefore, a test result of a good speaker without irregular distortions is shown first:

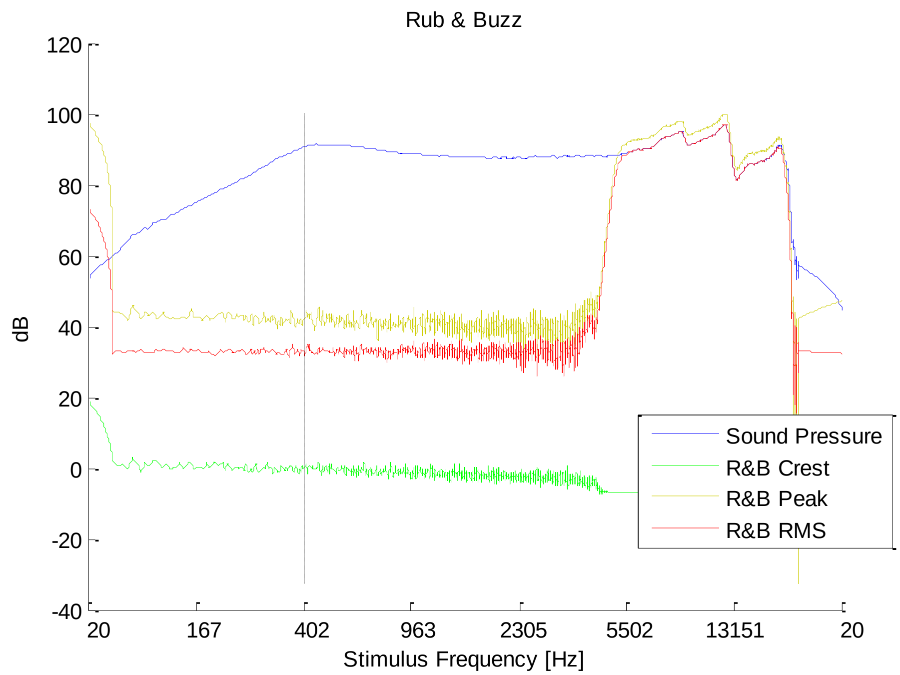

Figure 103: Envelopes of a micro-speaker without irregular distortions

This micro-speaker has a resonance frequency of about 400Hz and the regular distortions are quite high compared to a loudspeaker of normal size. In faultless use, the Rub & Buzz detection shows the absence of irregular distortions clearly because all curves are flat at low level in the whole stop-band of the Rub & Buzz filter.

To generate a rubbing voice coil, a piece of putty was added to the edge of the membrane so the mass and stiffness of the membrane is asymmetric and the movement of the voice coil is not purely vertical anymore. Using a sweep with sufficient amplitude, the acceleration at the resonance frequency is high enough to cause a rubbing voice coil.

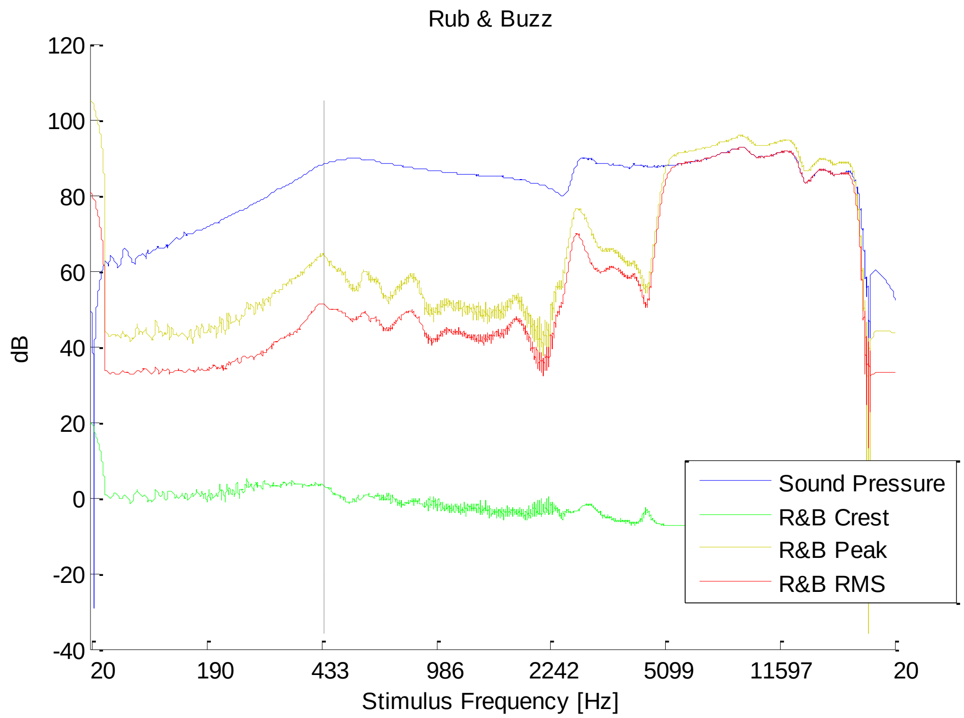

Figure 104: Envelopes of a micro-speaker with putty on the membrane

The area with increased R&B curves indicates irregular distortions in this frequency range. For a detailed check of the impulsivity, a zoom on the R&B Crest curve is needed.

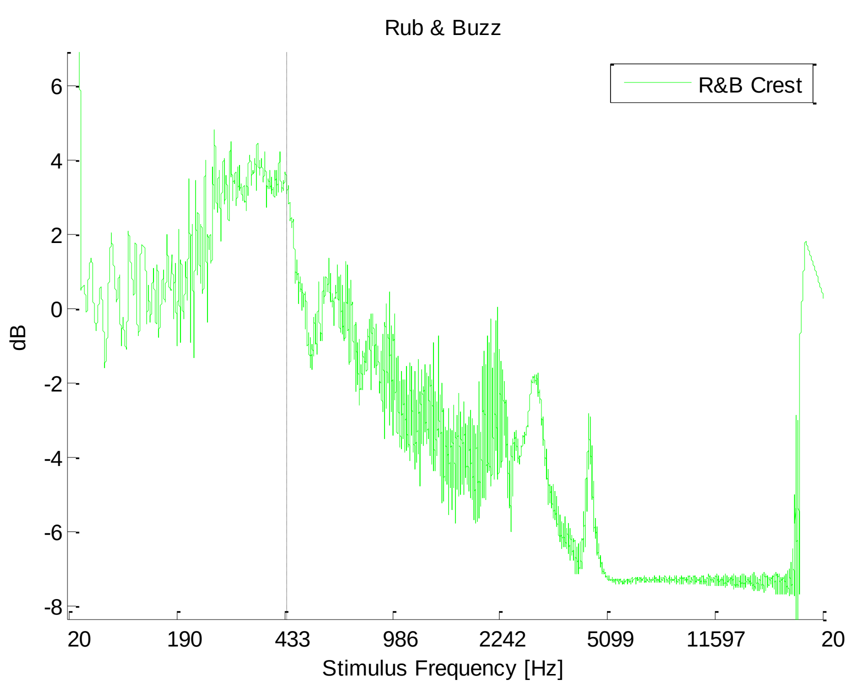

Figure 105: Envelope of the Instantaneous R&B Crest Factor

The maximum of the instantaneous crest factor is 4dB, so it is significantly higher than the noise floor but lower than a purely impulsive distortion.

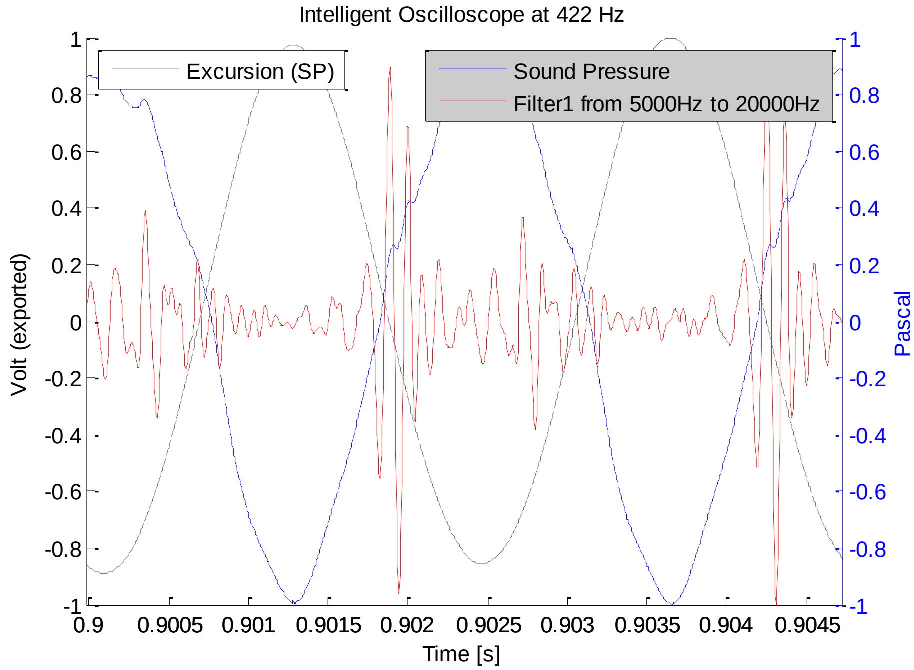

Figure 106: Time Signal at 422Hz of a micro-speaker with putty on the membrane

In Figure 106, several things can be seen:

- The occurrence of the distortion is at the 0 position of the membrane so it is not related to high membrane excursion.
- The distortion has stochastic character with impulsive components.
- The impulsive distortion occurs periodically – one appearance within each period of the stimulus.
- The long impulsive response of the Rub & Buzz filter spreads the energy of the short distortion to a wide area. This cannot be avoided without regular distortions in the Rub & Buzz signal because irregular distortions are next to the low order harmonics.

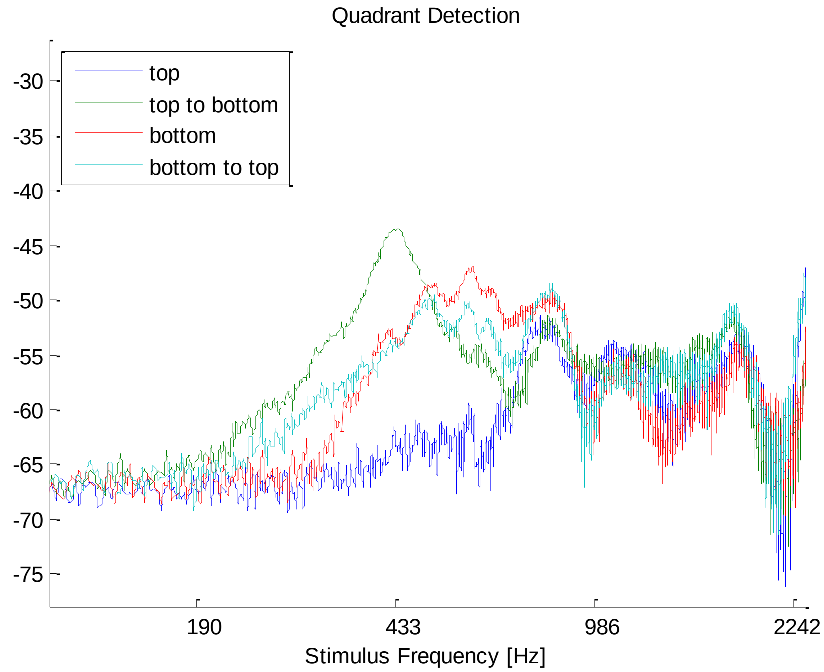

Figure 107: Stochastic Quadrant Detection of a micro-speaker with putty on the membrane

At the resonance frequency, the situation with highest acceleration, the distortion is most prominent at the transition from top to bottom, but at higher frequencies the distribution is changed a little bit. In the time domain, it is observable that the root causes are the long impulse response of the bandpass filter and the stochastic components, as seen in the Intelligent Oscilloscope plot. This leads to the shift of the energy maximum within one period. It is still clear that the root of the occurrence is not at the apex of the displacement, and the Deterministic Quadrant Detection reinforces this assumption as well.

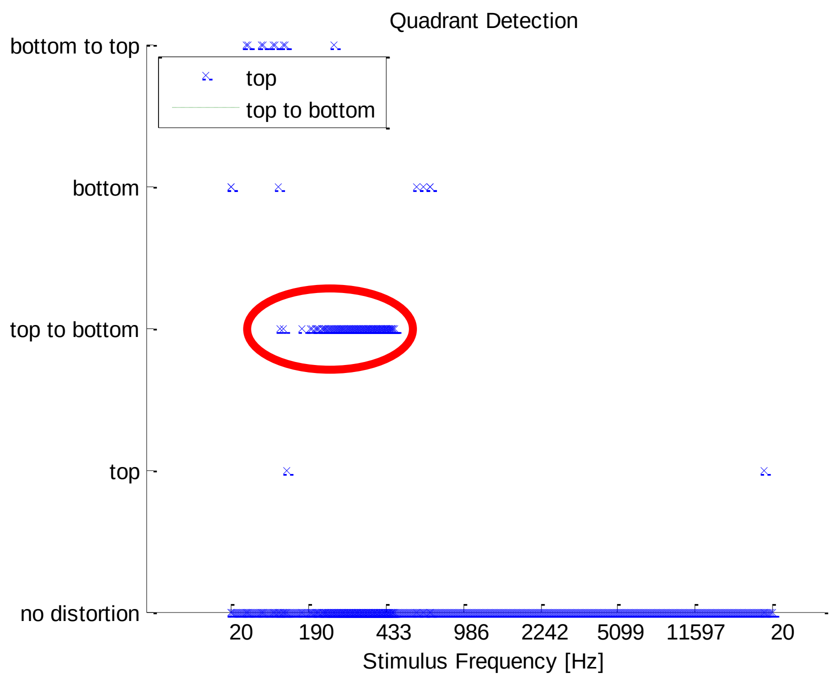

Figure 108: Deterministic Quadrant Detection of a micro-speaker with putty on the membrane

Most of the detections are at the transition from top to bottom and there are only a few numbers of outliers due to the noise floor.

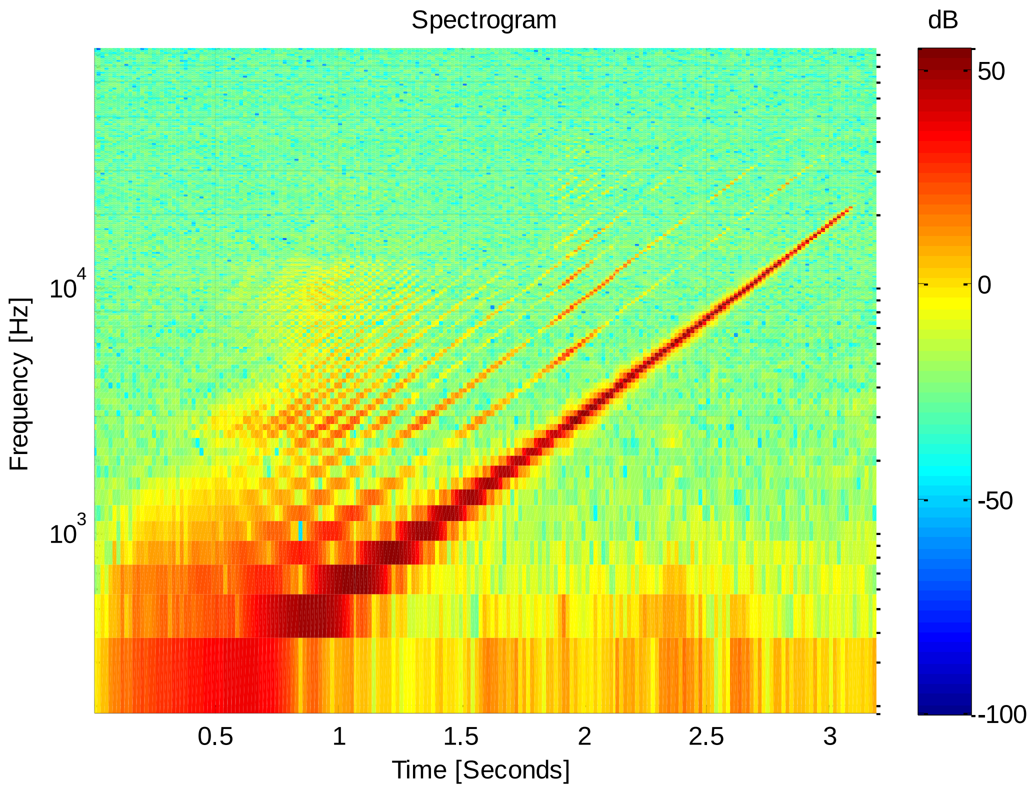

Figure 109: Spectrogram of a micro-speaker with putty on the membrane

Because of the high portion of harmonic distortions with micro-speakers, it is hard to distinguish irregular distortions in the spectrogram. However, it is possible to detect the rubbing voice coil in this plot too. In the time range at around 1 s, the spectral component at 8 kHz is a clear sign for non-harmonic distortion that fits with the detection of a rubbing voice coil. But of course, the spectrogram is not the best tool for this combination of driver and distortion.

This example showed that a rubbing voice coil shows no clear indication because the temporal occurrence is not always at the same quadrant, the instantaneous crest factor is not at an extreme range, and the spectral shape is a mixture of harmonic and non-harmonic distortion. But combining all analysis tools together and considering the strong dependency on the acceleration (maximum at resonance frequency), the root cause is clearly distinguishable without any additional measurement.

6.5 Buzzing Wire Loop

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	500mV RMS
Microphone Distance	1cm
Modification	Fixed wire next to the membrane

Table 7: Settings of a loudspeaker measurement with a buzzing wire loop

In electro-dynamic loudspeakers, a wire loop is commonly used to contact the moving voice coil. During normal production, it can happen that this wire loop is not accurately positioned, consequently the wire might touch the pot or coil during movement. To generate this failure mode, an ordinary wire is positioned in front of the loudspeaker. The moving membrane touches the loose wire which can vibrate as an own oscillating system, comparable with a buzzing wire loop.

Figure 110: Loudspeaker with a fixed wire at the front to simulate a buzzing wire loop

With sufficient excursion, we expect a vibrating wire which generates an own oscillation with harmonic distortion. This effect is evident in the envelope plot.

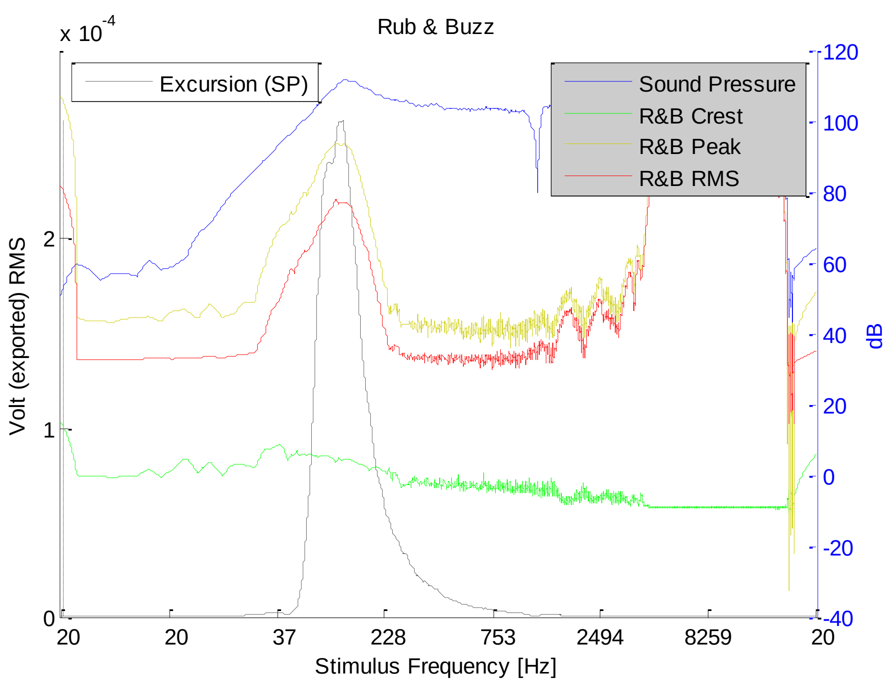

Figure 111: Envelopes of a loudspeaker with a buzzing wire

The peak and RMS values of the Rub & Buzz signal are increased in the frequency range with the highest excursion. The crest factor is also increased but is lower than with a hitting membrane. Additionally, the transition from distorted to undistorted is smoother than with a hitting membrane.

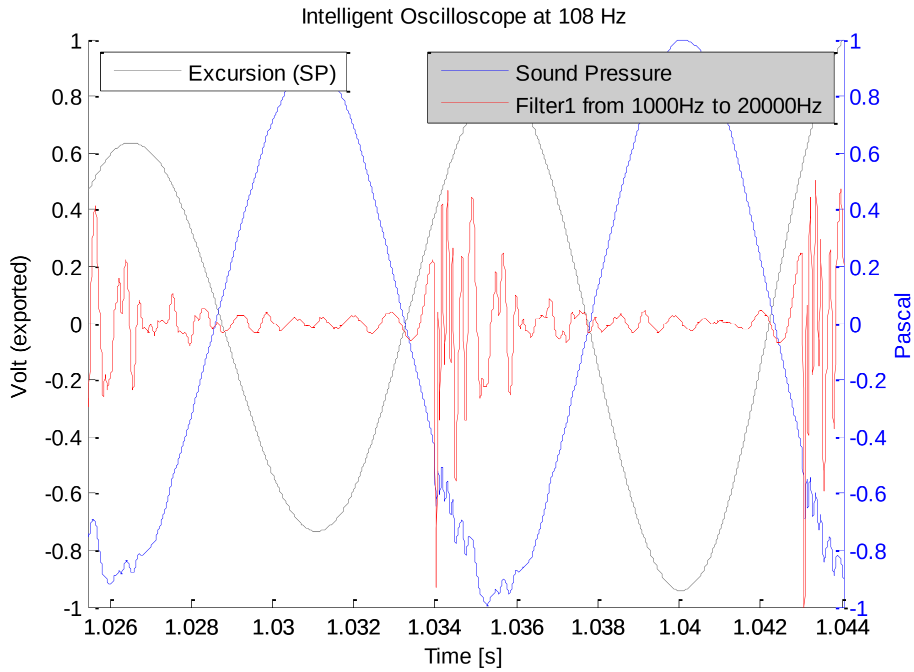

Figure 112: Intelligent Oscilloscope of a loudspeaker with a buzzing wire

Checking the time signal in the critical frequency range with sufficient excursion, the oscillating distortion is visible. The vibration is excited once in each period when the membrane moves upwards. At low stimulus frequency, as in the plot, the distortion can decay within one stimulus period, but at higher frequencies the next excitation occurs before the distortion has disappeared completely, so the own oscillation is not so clearly visible anymore.

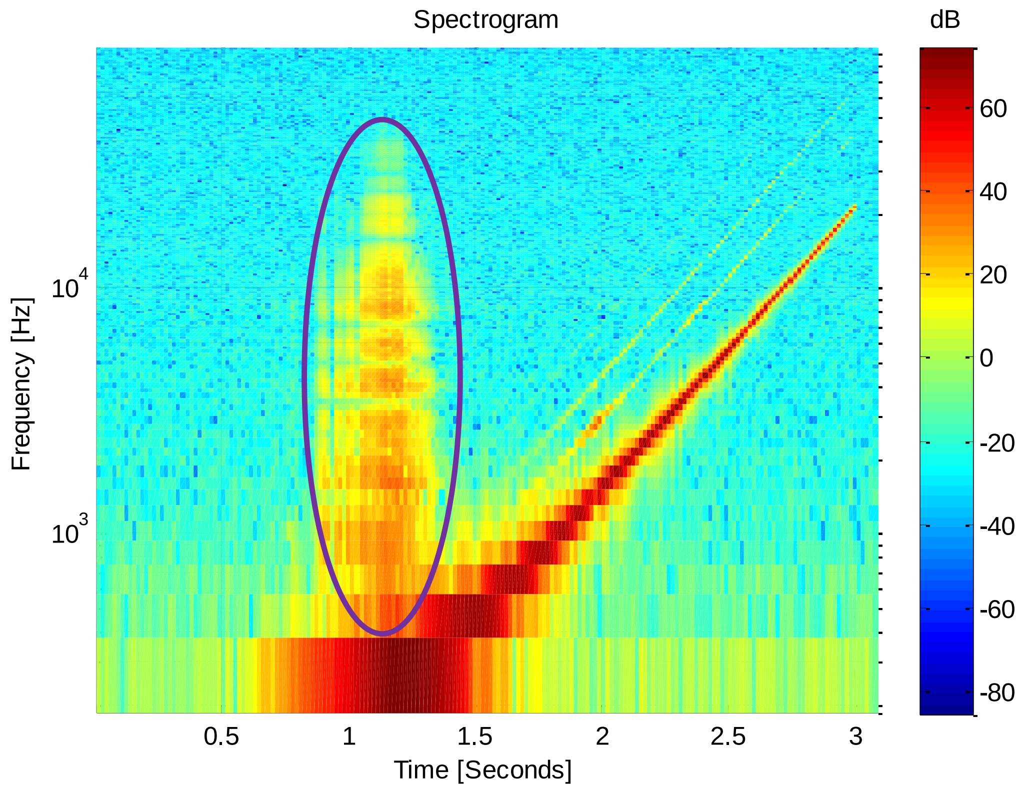

Figure 113: Spectrogram of a loudspeaker with buzzing wire

The distortion has a strong harmonic shape with a fundamental frequency of about 1kHz. This can be seen in the increased sound pressure level at 1, 2, 3… kHz. So a suitable Rub & Buzz filter needs a lower cut-off frequency of 1kHz to detect the distortion accurately.

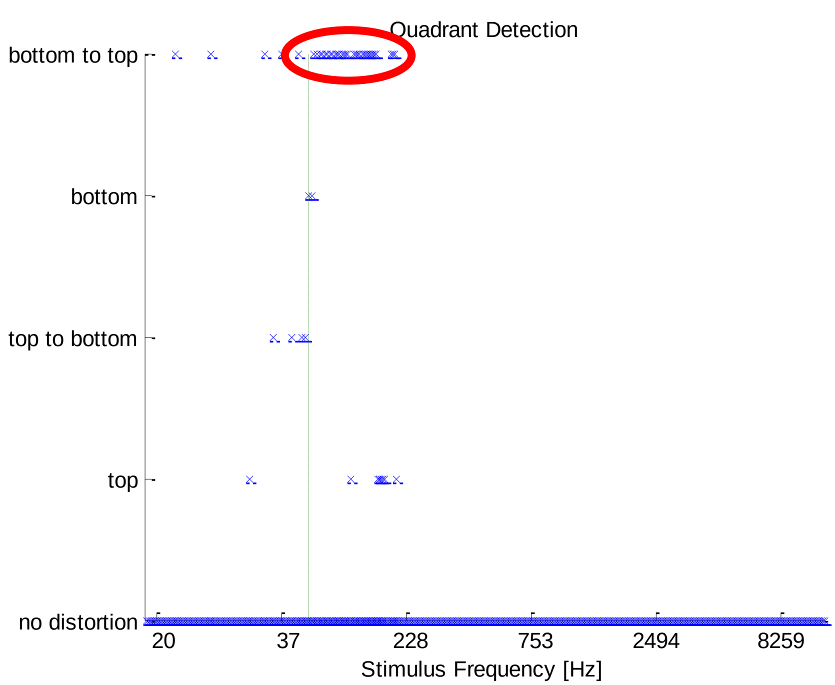

Figure 114: Deterministic Quadrant Detection of a loudspeaker with buzzing wire

A buzzing wire is a deterministic distortion and has quite high detection rate at the critical frequency range. Due to the high excursion, the root cause of the distortion is mainly at the bottom to top quadrant rather than at the top quadrant.

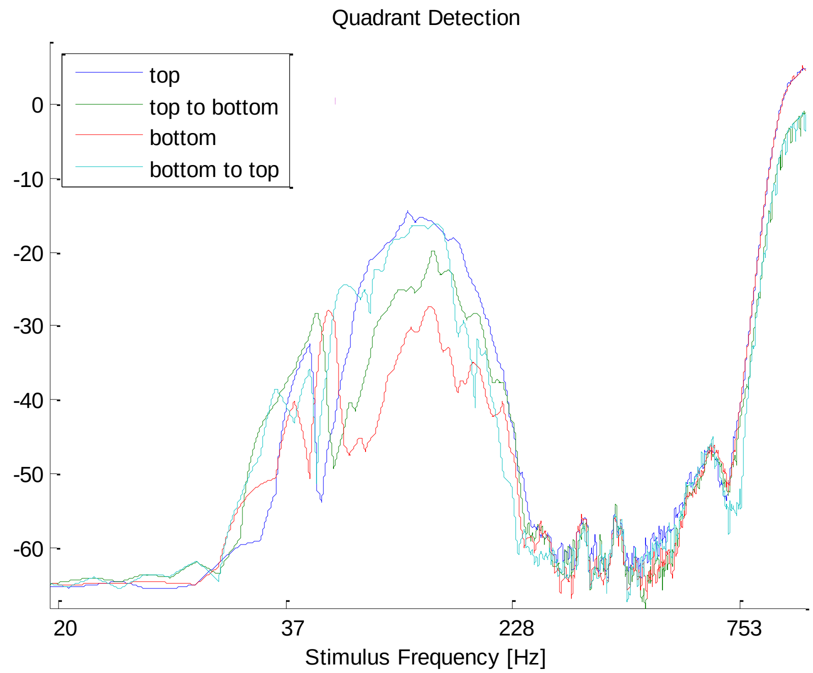

Figure 115: Stochastic Quadrant Detection of a loudspeaker with buzzing wire

Due to the long decay time of the resonator causing wide energy smearing, there is no clear separation between quadrants although the root only appears in one quadrant. Increased energy content in the top and bottom to top quadrants can be seen, something which gives a good indication of the displacement relation. At low frequencies with distortion content, the quadrant detection doesn't work correctly because the instantaneous frequency of the stimulus cannot be detected accurately because of the low SNR. Higher output voltage to increase the fundamental energy or a smaller microphone distance to decrease the noise floor could improve it.

6.6 Air Leakage in the Front Volume

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	2V RMS
Microphone Distance	1cm in front of the leakage
Modification	Plate mounted in front of the loudspeaker in a sealed condition to get a front volume. A hole in this plate generates a defined leakage.

Table 8: Settings of a loudspeaker measurement with air leakage at the front volume

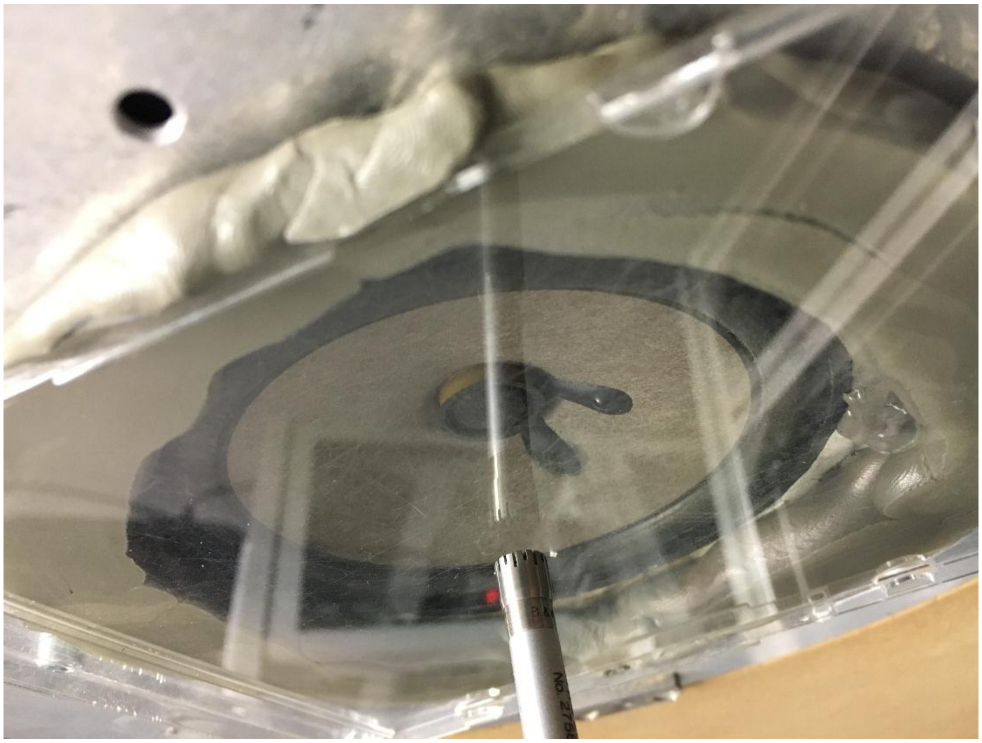

Figure 116: Loudspeaker with plate in front to get a sealed front volume. A hole generates a defined leakage

Micro speakers are often used in a closed back volume or with a front port, and in both applications leakage must be avoided. To generate this failure mode, a common loudspeaker with a defined tiny hole in a closed front volume is used. A microphone is placed in front of the leakage to record the distortion.

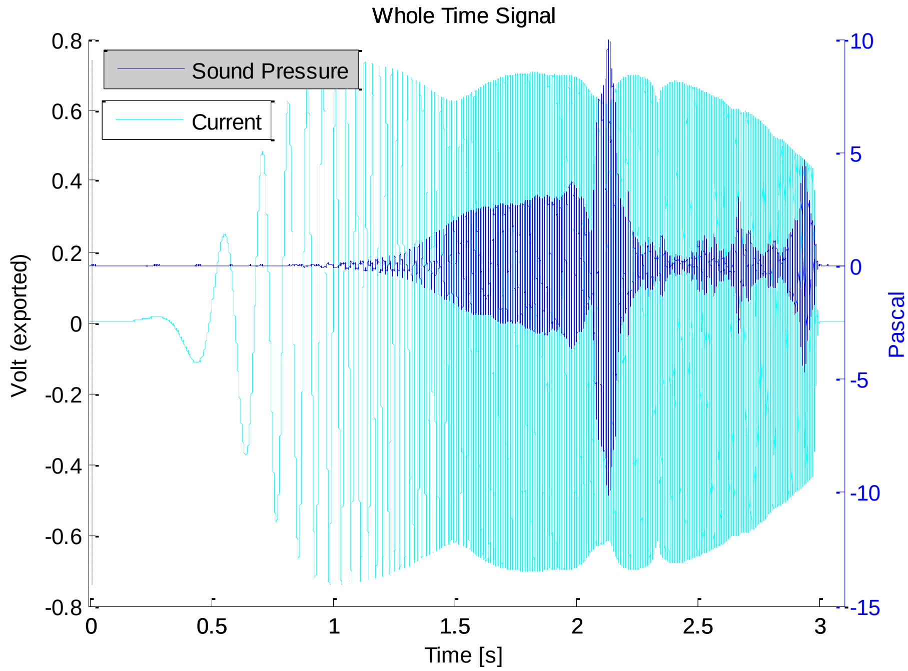

Figure 117: Whole Time Signal of a loudspeaker with front volume and leakage

Due to the closed front volume the additional stiffness increases the resonance frequency of the loudspeaker, so the overall performance is completely different to the other measurements.

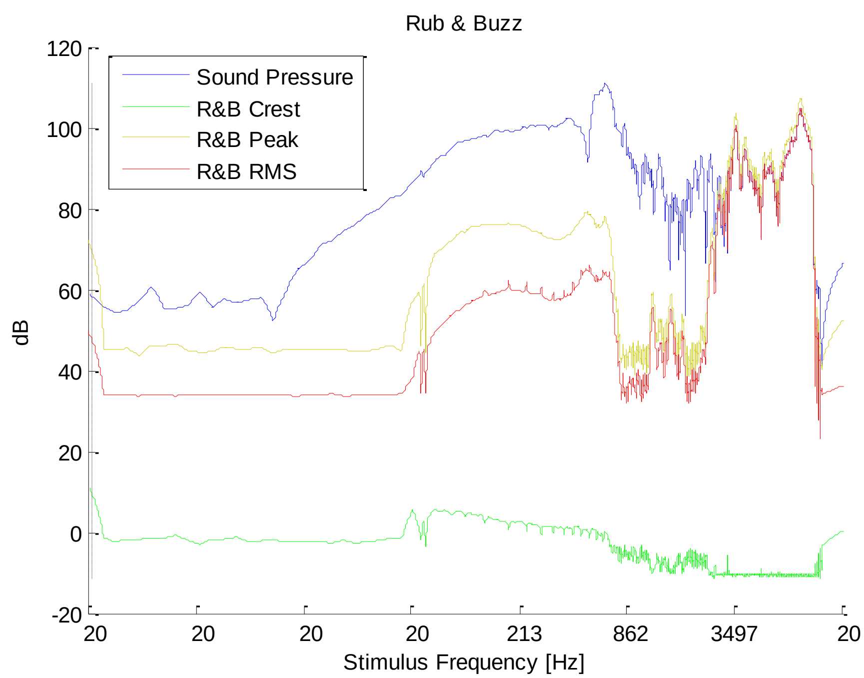

Figure 118: Envelopes of a loudspeaker with front volume and leakage

The R&B signal is increased in the frequency range next to the resonance frequency and is an indication that the distortion is velocity driven. The crest factor is surprisingly

high because - in theory - leakage generates pulsed noise and consequently the crest factor should not be very high.
Note: The lower cut-off frequency of this Rub & Buzz filter is at 3kHz to get the complete distortion, but this causes increased Rub & Buzz because of high THD at about 700Hz.

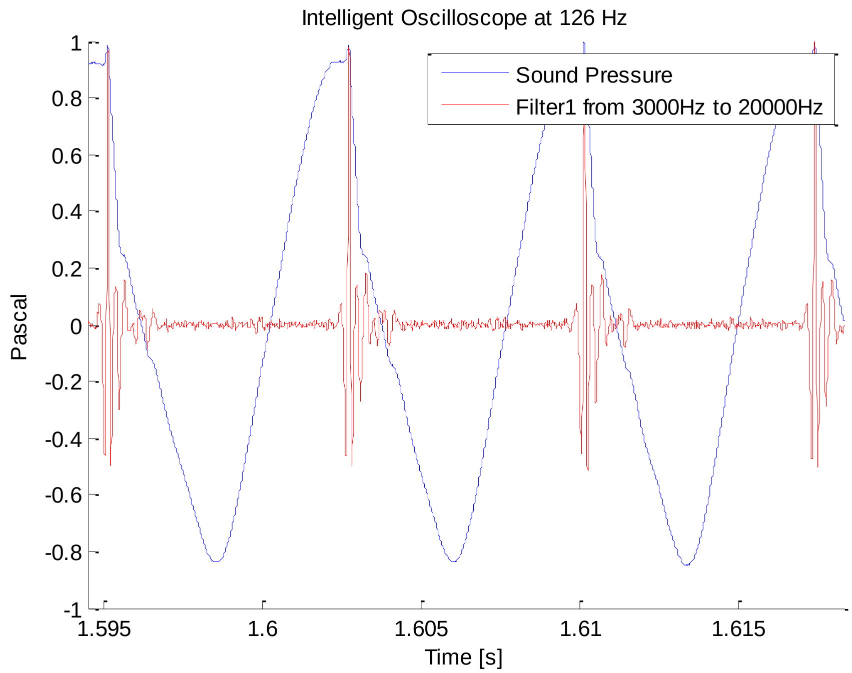

Figure 119: Intelligent Oscilloscope of a loudspeaker with front volume and leakage

Taking a closer look at the time signal in the distorted area reveals that the distortion is rather impulsive. Leakage theoretically causes a stochastic pulse that occurs at the velocity maximum. At this measurement, the fundamental is mainly from the backside of the loudspeaker, propagated over the baffle, so the fundamental and the distortion are not in phase. As a consequence, the appearance of the distortion cannot be analyzed. But the shape of the bandpass filtered sound pressure signal is correct and shows the distortion as a peak with small transient oscillation. This is possibly caused by the bandpass filter itself. Thus, the result is very surprising because the temporal shape of the distortion with leakage in a front volume is similar to a peak, rather than the pulsed noise one would theoretically assume. This behavior has been double checked with several applications of this loudspeaker and also with some micro-speaker applications and all had the same result. One explanation for this phenomenon is the transition from laminar into turbulent air flow in the hole, which can cause an impulsive noise, but this assumption has not yet been proven.

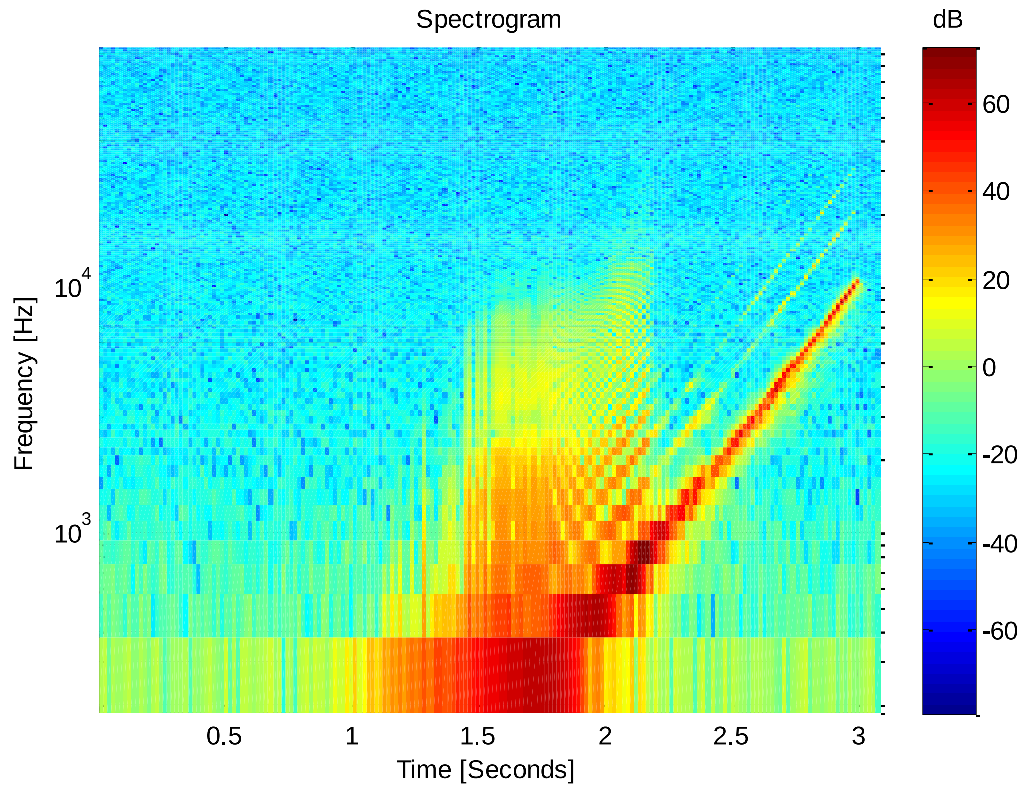

Figure 120: Spectrogram of a loudspeaker with front volume and leakage

In the distorted range around 1.5 seconds, the harmonics are increased due to the appearance of the stimulus in every period of the stimulus, and additionally there is a distortion with a white spectral shape in the range from 3 to 10 kHz because of the impulse.

6.7 Loose Particle

Loudspeaker	**Visaton FR 10 F**
Test Condition	Baffle - horizontal
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	20Hz – 20kHz
Voltage	500mV RMS
Microphone Distance	~1cm
Modification	Salt grains on the membrane

Table 9: Settings of a loudspeaker measurement with loose particles on the membrane

For a proper way to generate the effect of loose particles, salt grains can be put on the membrane. When the membrane moves, the grains jump and produce distortions that are impulsive and random. In reality, this failure mode can be very different, but this method generates and mimics the error quite well.

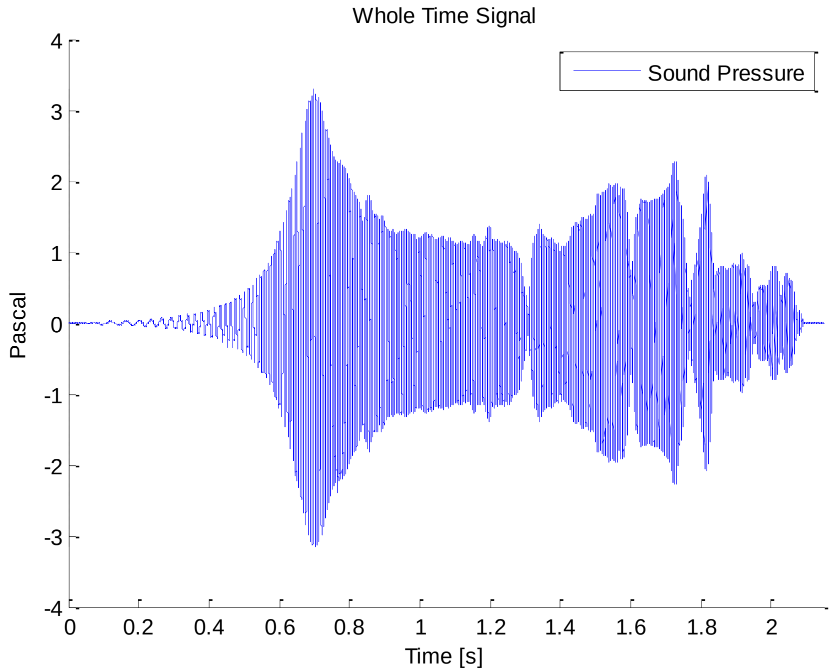

Figure 121: Sound pressure time signal of loudspeaker with salt grains

The sound pressure time signal shows no change in the normal loudspeaker, so it is not useful for failure analysis.

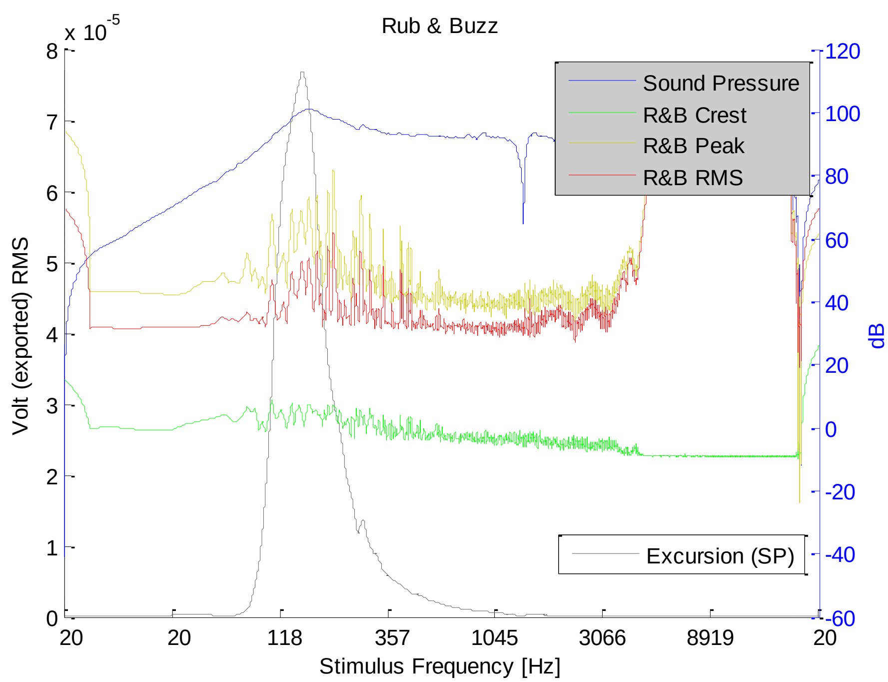

Figure 122: Envelopes of a loudspeaker with salt grains

The R&B curves show the distortion clearly, but the curves are not smooth: they are markedly rippled. This is an indication of a random appearance and an impulsive time shape. It is also discernible that the distortion appears only if the membrane excursion exceeds a threshold making it somehow displacement dependent.

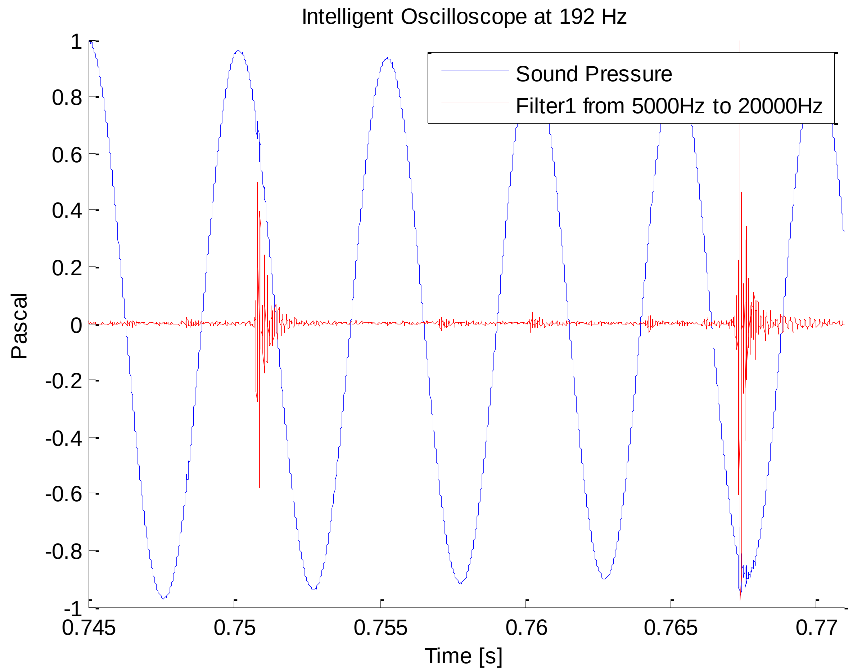

Figure 123: Intelligent Oscilloscope of loudspeaker with salt grains

This plot provides a good opportunity to check the temporal shape and appearance of the distortion. In this extract, there are five visible periods of stimulus, but only two arises from the defect. After inspecting several time slots, it is clear that this ratio is not constant because the appearance is random.
The temporal shape of the distortion is short for all occurrences, so the distortion must be impulsive.

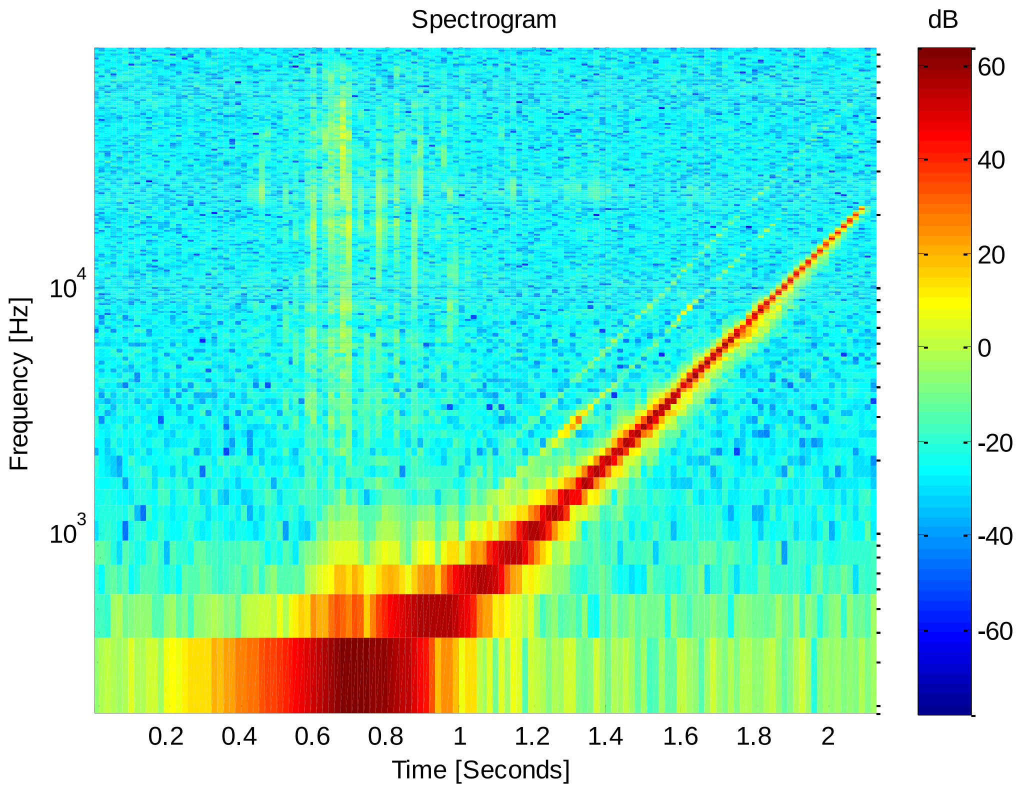

Figure 124: Spectrogram of a loudspeaker with salt grains

The spectrogram illustrates a distinct fundamental with small regular distortions and irregular distortions with white spectrum in a certain time range. This spectral property indicates an impulsive temporal shape. A close look at the time axis reinforces the random appearance of the defect.

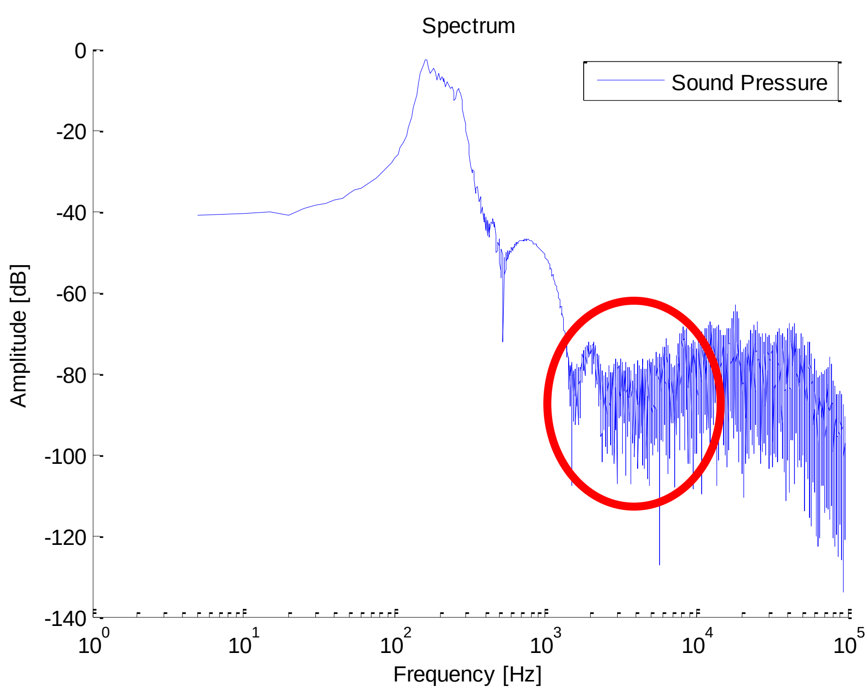

Figure 125: Spectrum of a loudspeaker with salt grains at a distorted region

This spectrum is from the time when irregular distortions occur, so the spectrum of the jumping salt grain is visible. The fundamental is at about 200Hz while the regular nonlinearities are quite low rendering the harmonics invisible. The spectrum from 5kHz to 15kHz is higher than normal and appear almost white, so this portion is related to the irregular distortion. Knowing this is very important for the setting of a Rub & Buzz filter to catch all the energy from this distortion and as little as possible from other noise or the fundamental and its regular harmonics.

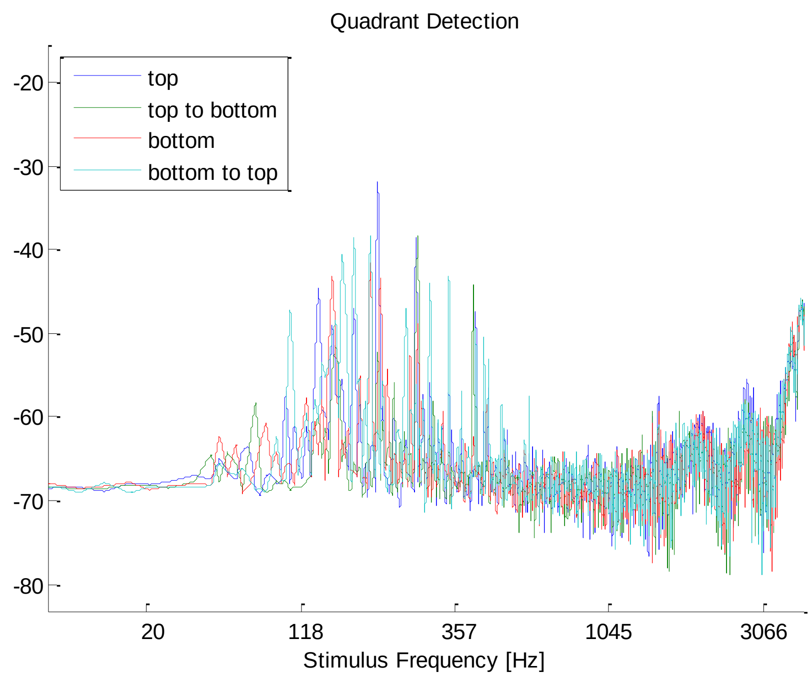

Figure 126: Stochastic Quadrant Detection of a loudspeaker with salt grains

There is no trend to one quadrant; all four curves show similar shapes with a rise from 100-400Hz and lots of peaks. This is a sign of the random appearance and impulsivity of distortion when the salt grains hit the membrane.

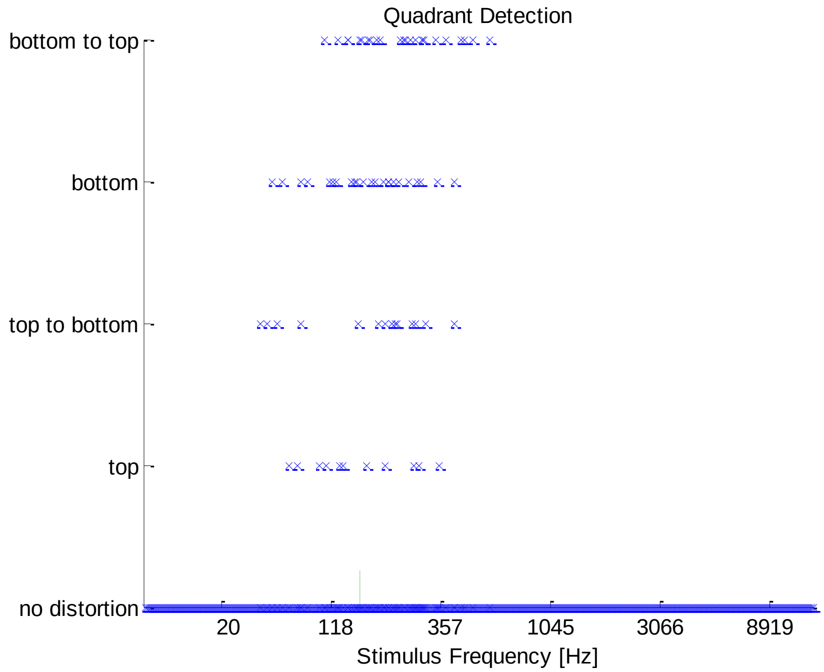

Figure 127: Deterministic Quadrant Detection of a loudspeaker with salt grains

There are distinct numbers of detection in the critical frequency range revealing impulsive distortions. But all four quadrants have almost the same number of markers, so there is no preferred quadrant.

Summarizing all views, it is clear that the distortion has random temporal appearance, is impulsive, has white spectrum, has no correlation to velocity or excursion but occurs only above a certain displacement, and is not quadrant specific. The possible root causes are thus restricted and an experienced engineer can distinguish loose particles with only one measurement.

6.8 Representative Micro-speaker failure

Loudspeaker	**Micro-speaker 18 x 13 x 2.5 mm**
Test Condition	Baffle
Test System	APX
Test Signal	Continuous Sweep 2s
Frequency Range	100Hz – 20kHz
Voltage	1000mV RMS
Microphone Distance	1cm
Modification	No modification, negative DC for failure analysis

Table 10: Settings of a test with a representative micro-speaker failure

To show the tool’s applicability for testing micro-speakers, a common sample with an unknown defect is used for the failure analysis. The speaker is driven with a frequency sweep where the distortion occurs at a certain frequency range.

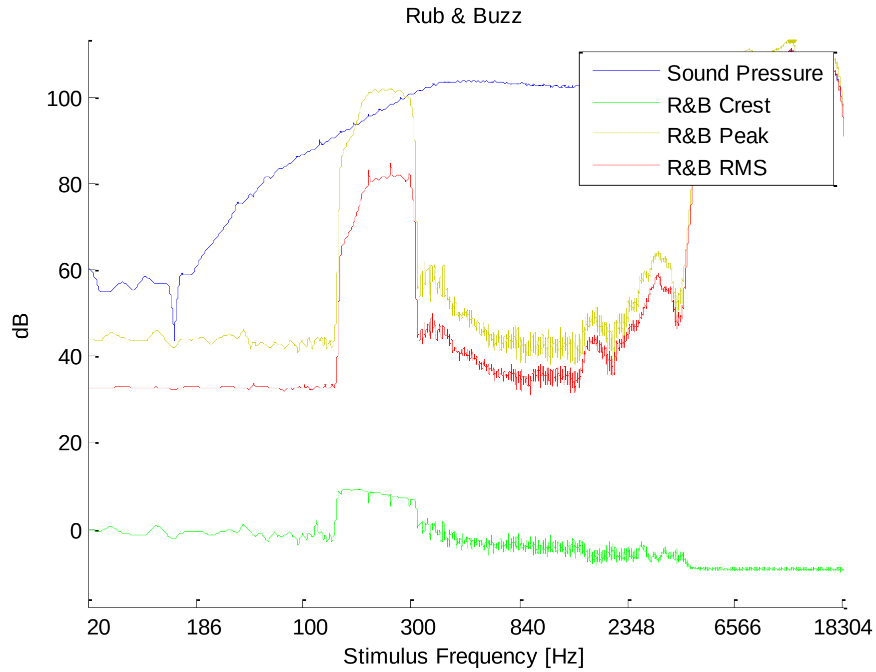

Figure 128: Envelopes of a micro-speaker with a representative failure

The resonance frequency of this speaker is 350Hz and the maximum excursion is caused by the high damping slightly below the resonance frequency, thus distortion occurs at frequencies with high excursion. The relative instantaneous crest factor is above 5dB making the distortion impulsive.

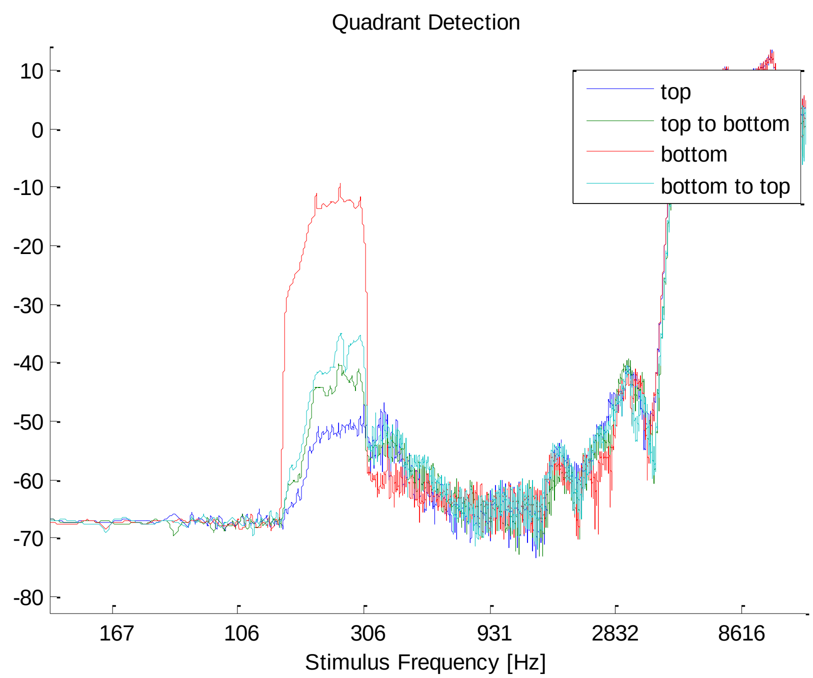

Figure 129: Stochastic Quadrant Detection of a micro-speaker with representative failure

Looking at the Stochastic Quadrant Detection one can see that the distortion energy is mainly in the bottom quadrant revealing that the defect is excursion dependent with a prominent quadrant.

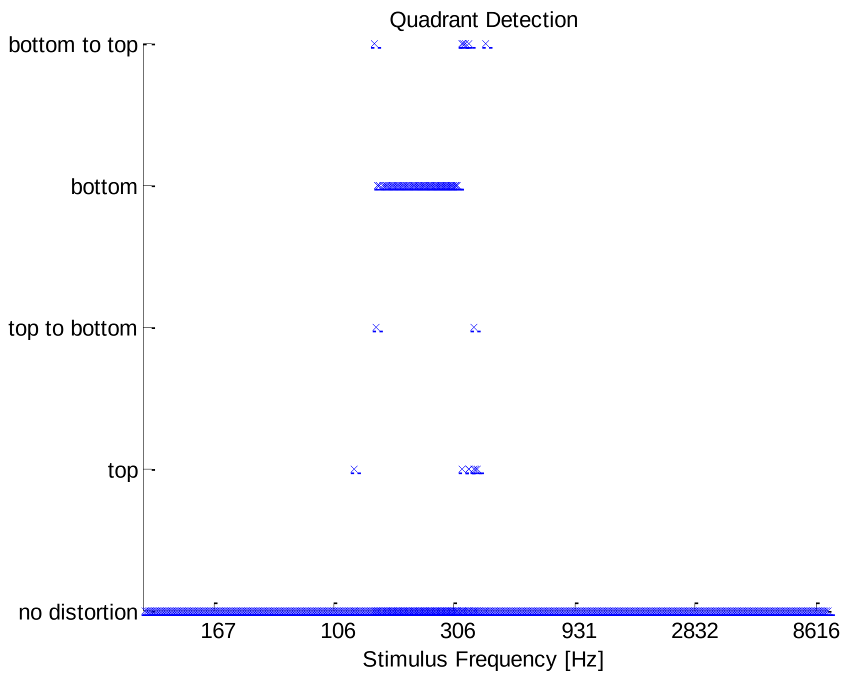

Figure 130: Deterministic Quadrant Detection

There is a significant group of markers at the bottom quadrant indicating a deterministic distortion with high excursion dependency.
Summing up these analyses it is clear that the root cause of the distortion is a hitting membrane or voice coil at the bottom side. To verify this failure analysis with the Universal Sweep Analyzer, two additional experiments are done.

Firstly, the same micro-speaker is driven with lower input power (630mV) where no distortion occurs.

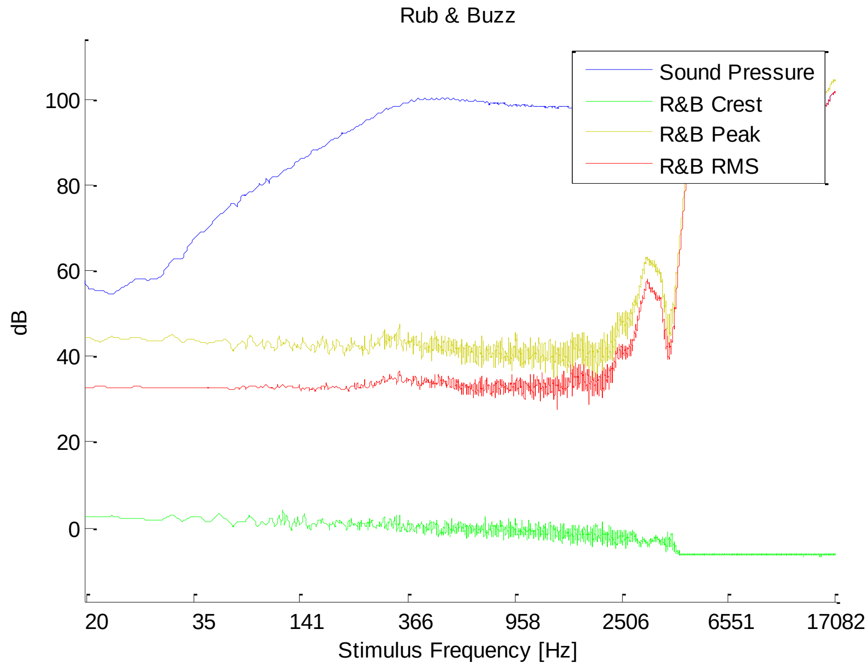

Figure 131: Envelopes of a micro-speaker without irregular distortions

The Rub & Buzz signals are not increased so the Universal Sweep Analyzer doesn't detect any irregular distortion.

To see what a hitting membrane at the bottom side looks like with this speaker, the same AC power with additional negative DC power is applied to shift the membrane permanently downwards. This approach guarantees that hitting at the bottom side is the root cause of the distortion.

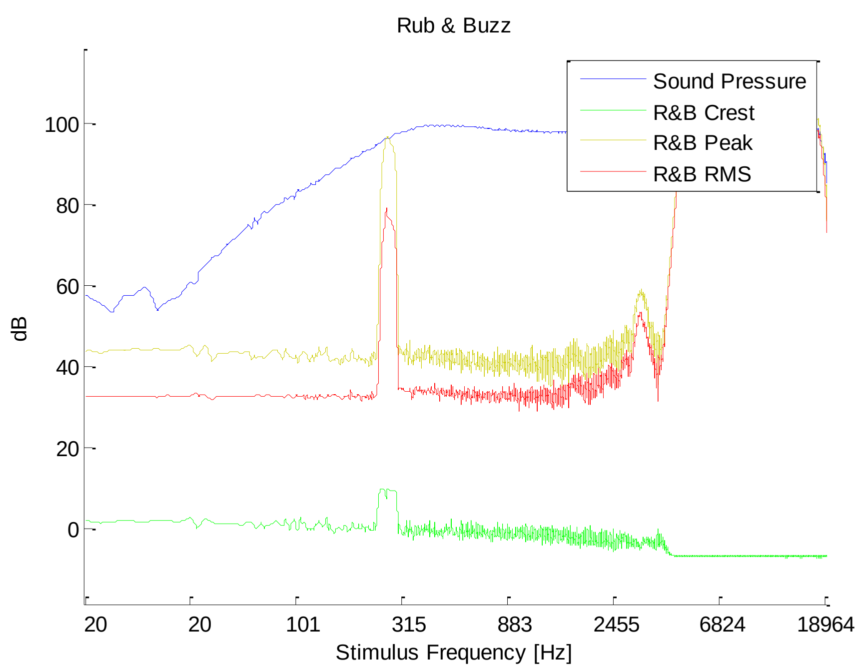

Figure 132: Envelopes of a micro-speaker with negative DC power to generate a hitting downwards membrane

The distortion occurs in a comparable frequency range as with the real failure shown above. The relative instantaneous crest factor is at the same level (~5dB).

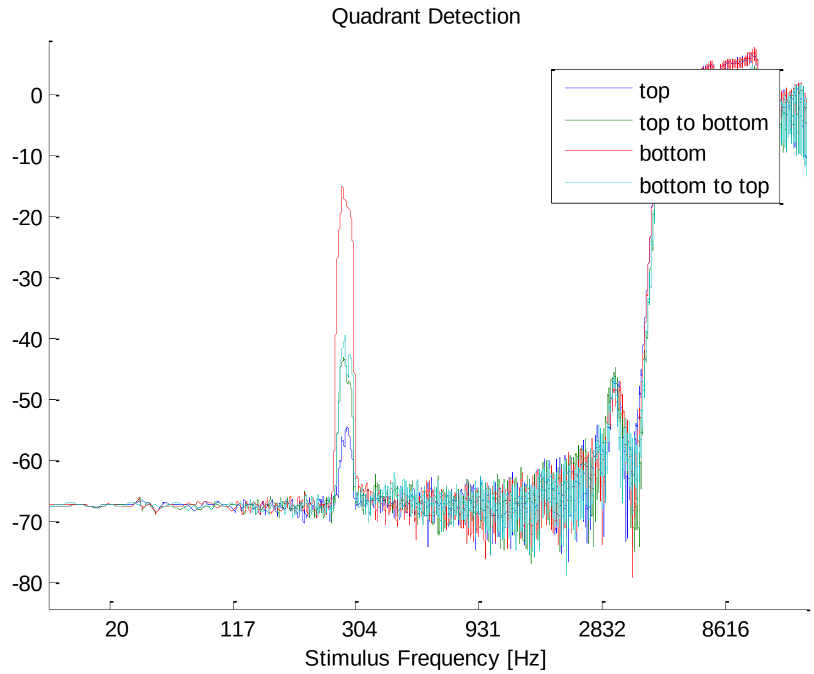

Figure 133: Stochastic Quadrant Detection of a micro-speaker with negative DC power to generate a hitting downwards membrane

The Quadrant Detection shows a similar picture as the real failure. The distortion in the bottom quadrant has much higher energy than in the other quadrants, so it clearly indicates a hitting membrane at the bottom. This proves that the analysis of the representative failure above is correct and the root cause detection with the Universal Sweep Analyzer works.

7 Conclusions

The aim of this master thesis is the development of a software program that supports the analysis of regular and irregular distortions and detects its root cause. To begin with, signal properties of the distortion caused by different defects are analyzed, something which is helpful for the error detection. With these properties in mind, the software program has been developed to include specific analyses functions and plots for the ability to detect the root cause of the distortion.

The first plot in this program is the time signal plot of all acquired signals. This view is helpful to check the feasibility of the measurement and to get an overview but a zoom is necessary to analyze details in the time domain. Therefore, the so-called Intelligent Oscilloscope was developed. It displays the time signal at a certain moment with a constant number of periods independent from the stimulus frequency.

Envelope plots were implemented to see the critical stimulus frequencies. This plot displays the maximum of each stimulus period over an axis with the instantaneous frequency of the stimulus. Additionally, statistical parameters of the so-called Rub & Buzz signal, a bandpass filtered sound pressure signal, are displayed in this plot. The Rub & Buzz signal is a simple method of filtering irregular distortions, and its statistics indicates the impulsivity of the defect.

The spectrum and spectrogram of the sound pressure signal are calculated for the analysis of the spectral distribution of the distortion. These views also help find the optimum Rub & Buzz filter.

To find the dependency of the membrane on the displacement, the so-called Quadrant Detection was developed. It splits the Rub & Buzz signal into four segments where each segment represents an area of the membrane displacement.

The implementation of these functions required several signal processing steps and simplifications, respectively assumptions, consequently the user has to consider several limitations.

For the analysis of the membrane excursion dependency without laser measurement, the membrane displacement is estimated from the acquired sound pressure signal. For this operation, a simplification of the acoustic impedance of the circular piston diaphragm is used, but this approach is only valid for micro-speakers measured in a baffle in the frequency range below 1 kHz.

The envelope plot and Quadrant Detection need an accurate estimation of the stimulus frequency. The Phase Vocoder approach is used to detect the instantaneous frequency of the sound pressure signal. This method works very well if the SNR is sufficiently high. But if a loudspeaker is tested below its resonance frequency, the output sound pressure level can be very low. Consequently, the noise floor is relatively high and the SNR can be too low for precise frequency detection.

The developed software program is able to synchronize all acquired signals, and this is important in order to detect temporal relationships between the distortion and all other signals. For this process, the separation of the allpass and minimum phase system is applied, something which is valid for a loudspeaker driven in a baffle or in free-field, but not in pressure chambers or artificial ears. If such applications are analyzed, then the acoustic engineer has to compensate for the delay manually or he needs to take the asynchronous signals into consideration.

One target of this thesis was to create an automatic root cause detection of irregular distortion, but unfortunately it did not work. Due to the variability of the different loudspeakers, especially micro-speakers, the distortions always have different absolute parameters, although the defect is the same. This makes it difficult to find threshold values for the separation of distortions into clusters. Additionally, some parameters, like the instantaneous crest factor of the Rub & Buzz signal, which represents impulsivity, are dependent on the presently used measurement system. An automatic failure detection which works safely for all kinds of loudspeakers was thus not possible to develop with the analysis methods, algorithms and requirements applied here, so for failure analysis, an experienced acoustic engineer is still required.
Maybe this automatic detection can be done with a more complex clustering algorithm that analyzes more statistical parameters and its dependencies for different failures, but this has not been tried in detail.
Another option would be to change the requirements and make a test sequence with different stimulus types and power levels to offer the possibility for automatic detection. This option has not been implemented because the software program has to be test system independent, but it would of course be an interesting task to develop a new measurement system or adapt a current one, with predefined sequences where the engineer only has to initiate the procedure, then gets the root cause detected automatically. Generally, this approach should be implementable though careful deliberations with regards to limitations would probably have to be done.

A major surprise occurred during the failure analysis of representative defects. Building air leakage in a closed volume always showed a failure mode which does not fit current theory. In standard literature, the distortion of an air leakage is described as a pulsed noise synchronized with the stimulus period, but the reality exposed an intensive impulse. Several applications with different loudspeakers were tried, but all gave similar results. An explanation is that the transition from laminar to turbulent air flow produces a drop in the acoustic impedance of the hole, which generates impulsive distortion. But this is just a hypothesis and needs to be proven with further trials.

Finally, a software program is available which works well for some measurement systems and test conditions, but an automatic load for most of the common measurement systems has not yet been found. Lots of measurements in a baffle have been analyzed and documented, and it works, but different use cases, as artificial ears or sound pressure chambers, have not been tested. This needs to be done to use such applications for the failure analysis also. And of course, the program is developed with Matlab which is a good programming language for the development phase, but if this software is used regularly then probably an implementation with a different programming language offers a better solution. LabVIEW would be a good choice for a new implementation because it has higher stability and is generally faster.

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