# Development of a Low-Cost Characterization System for Feed-Forward ANC Headphones

Diploma Thesis

Completed by


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Graz, June 2011

# STATUTORY DECLARATION

I declare that I have authored this thesis independently, that I have not used other than the declared sources / resources and that I have expressly marked all material which has been quoted, either literally or by content from the sources used.

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# Acknowledgment

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# Abstract

The performance of a properly-working feed forward ANC system mainly depends on the accurateness of the implemented filter curves. To achieve good attenuation levels, these filter curves have to be evaluated individually for every headphone type. The measurement process for the evaluation of the required filter gain and phase response has, until now, been considered a complex process. Assuming an acoustically optimized measurement chamber makes the process inflexible. High-performance hardware, like measurement microphones, pre-amplifier and the IEC 711 standard coupler are expensive and, additionally, require in-depth knowledge of acoustics and signal processing routines to diligently perform measurements. Last, but not least, the pace of the measurements marks an important point towards supporting cost-optimized product development.

The aim of this thesis is to develop an all-in-one low-cost solution which includes the adaptation of the measurement environment, resulting in downsizing from room size to the required manageable size, the development of the necessary hardware components, like the acoustic coupler based on the IEC 711 standard coupler, and the development of a measurement software, including an easy-to-handle GUI.

The finalized measurement system should allow fast and easy characterization of ANC headphones in an office environment. An additionally-developed ANC performance evaluation tool should allow determination of the active headphone performance.

# Kurzfassung

Die Leistung eines gut funktionierenden Feed Forward ANC Systems hängt entscheidend von der Genauigkeit der implementierten Filterkurven ab. Um eine gute Dämpfung der Außengeräusche zu erreichen, müssen die Filterkurven individuell für jeden Kopfhörertyp bestimmt werden. Der Messvorgang zur Bestimmung der nötigen Filterkurven und Phasenlagen ist bis dato sehr aufwendig. Die Voraussetzung eines akustisch optimierten Messraumes trägt zusätzlich zur Unflexibilität des Messvorganges bei. Die benötigten Hardwarekomponenten wie Messmikrofone, Vorverstärker und der IEC 711 Standard Akustikkoppler sind kostenintensiv und setzen gute Kenntnisse in Akustik und Signalverarbeitung voraus um ein zufriedenstellendes Messergebnis zu erzielen. Nicht zuletzt spielt auch die Dauer der Messung eine entscheidende Rolle für eine kostenoptimierte Produktentwicklung.

Das Ziel dieser Diplomarbeit ist die Entwicklung einer kostengünstigen All-In-One Lösung. Das inkludiert die Anpassung der Messumgebung durch die Reduktion auf handliche Dimensionen, die Entwicklung der erforderlichen Hardware Komponenten, wie unter anderem einen Akustikkoppler basierend auf den Vorlagen des IEC 711 Standard Kopplers und der Entwicklung einer entsprechenden Messsoftware inklusive GUI, welches die einfache Bedienung des Messsystems ermöglichen soll.

Das fertige Messsystem soll die einfache und schnelle Charakterisierung von ANC Kopfhörern in einer typischen Büroumgebung ermöglichen. Ein zusätzlich entwickeltes Programm soll die Bestimmung der Leistung von bereits charakterisierten ANC Kopfhörern ermöglichen.

# Index

# List of Illustrations

# List of Used Abbreviations

| | |
|---|---|
| **DUT** | **D**evice **U**nder **T**est |
| **AH** | **A**rtificial **H**ead |
| **ANC** | **A**ctive **N**oise **C**anceling |
| **IEM** | **I**n **E**ar **M**onitor |
| **PCB** | **P**rinted **C**ircuit **B**oard |
| **GUI** | **G**raphical **U**ser **I**nterface |
| **APT** | **A**ctive **P**erformance **T**ool |
| **FCT** | **F**ilter **C**alculation **T**ool |
| **CAT** | **C**h**a**racterization **T**ool |

# 1 Introduction

Drawn against the background of improved technical feasibility, active noise cancelling (ANC) systems are increasingly gaining in significance in various acoustic situations. This spans the areas of aircraft engine insulation to resonance reduction of wind turbines to sound insulation design in car engines and improvement of communication lines for helicopter pilots.

This master's thesis focuses on ANC in the context of headphones. Specifically, the emphasis is on measuring equipment to be developed for so-called 'ear buds' (also called 'in-ear headphones or in ear monitors (IEMs)'[1]).

The principle behind the ANC procedure is generally known as destructive interference. Here, interfering noise is picked up by microphones externally attached to the headphones and transmitted to the ear phased inversely via the loudspeaker, taking into account a transfer function modified by the earphones. By means of this procedure, environment noise - particularly low frequencies - may be reduced very well.

Imperative necessity of highly accurate measuring of the transmission line requires an extremely cost-intensive and complex measuring system for calibration and recording the correct filter curves and phasing. High cost for the measuring device and related high development cost are the main reasons for a relatively high sales price for headphones.

This master's thesis aims at developing a low-cost solution for the measuring system, so headphones can be offered at a competitive price.

The unique challenge with this master's thesis is about developing measuring equipment in manageable size and sizing all components, factoring in the aspect of minimizing cost but still not suffering any setbacks in accuracy of measurement or quality. Further, the completed measuring system has to be easy-to-use. Tentative sales price for the completed measuring system is EUR 1,000.--.

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[1] Headphones inserted into the ear canal

## 1.1 Overview

A short summary of fundamental investigations can be found in chapter 2. It includes the most important headphone basics and the principles of active noise cancelation, focusing on feed-forward systems. The second part deals with signal processing fundamentals which are necessary to understand the development process of the measurement system.

Chapter 3 deals with the development of the measurement box, including speaker, hardware and absorbers, under the aspect of cost minimization and local-independency. It includes a detailed description of the ANC – measurement process as well as the measured performances of the developed parts. Additionally, a big part of this chapter is the development of the IEC 711-based acoustic coupler.

The development of the measurement software can be found in chapter 4. The software based on MATLAB, which uses exponential sine sweeps to measure frequency and phase response curves of the device under test (DUT) and calculates ideal filter transfer function. It contains a description of the software features and the most important signal processing steps implemented. The second section of chapter 4 deals with the graphical user interface (GUI) and gives a detailed description of the GUIs functions.

A recapitulation of the finalized measurement system can be found in Chapter 5. Finally, chapter 6 deals with a summary of the developed system and added a short outlook.

For additionally and detailed information, Appendix A, B and C includes a user manual, all relevant datasheets and design drawings of all developed parts. Furthermore the enclosed CD-ROM includes inter alia, the measurement software and a 3D-model of the measurement system.

# 2 Theory

The functionality of this measurement system puts the emphasis on the development of active noise cancelling (ANC) headphones. The following chapter introduces the fundamentals of different headphone types, ANC filter calculations, signal processing basics and measurement techniques, which are required for understanding the process of achieving the ideal filter curves and determining the ANC performance of the device under test (DUT). Further, it should be made possible to comprehend the development process of different hardware parts included in the measurement system.

## 2.1 Headphones

First of all, it should be noted that there is a huge amount of different headphones types. Within the limitations of this thesis, the measurement system was developed for just one of them. However, the design of the system should provide an opportunity for further developing the essential parts - like the acoustic coupler - and enhancing the system for other headphone types[2].

The huge amount of different headphone types can be classified along the lines of the following 2 major groups. [CON11] [WIK11]

### 2.1.1 Headphone types

#### 2.1.1.1 Over-ear and on-ear headphones

**Over-ear headphones** (fig. 2.1 a), also called circumaural headphones or full-sized headphones, are the biggest types of headphones. Their ear pads enclose the whole ear. This headphones group can be further separated into closed and open types. The attribute of the closed type is the encompassing sealing against noise from the outside. The disadvantage can be found in the same property, because the sealing also extends to noise that maybe should be heard, like sirens or other important acoustic messages. On the other hand, there is the open type. In this case, the sound from the outside could particularly reach the ear, but, otherwise, part of the sound from the headphone could also be heard outside. Over-ear headphones are often used in combination with feed-backward ANC systems.

[2] Of course, this enhancement is naturally limited by the dimensions of the box. This means, for example, the development of circumaural headphones is not possible without changing the hardware dimensions.

**On-ear headphones** - also called supra-aural - are also available as open and closed versions. They weigh less then over-ear headphones because they are much smaller. As fig. 2.1 b) shows, their ear pads are placed directly on the pinna. They can be used single-sided in combination with feed-forward or feed-backward ANC systems, for example, as intercom headsets. Furthermore, their usage in combination with mobile devices is common, which also leads to a combination with ANC systems. There are scores of portable on-ear headphones, like neck band over-ear models, sports-style on-ear or wireless models of all mentioned headphones.

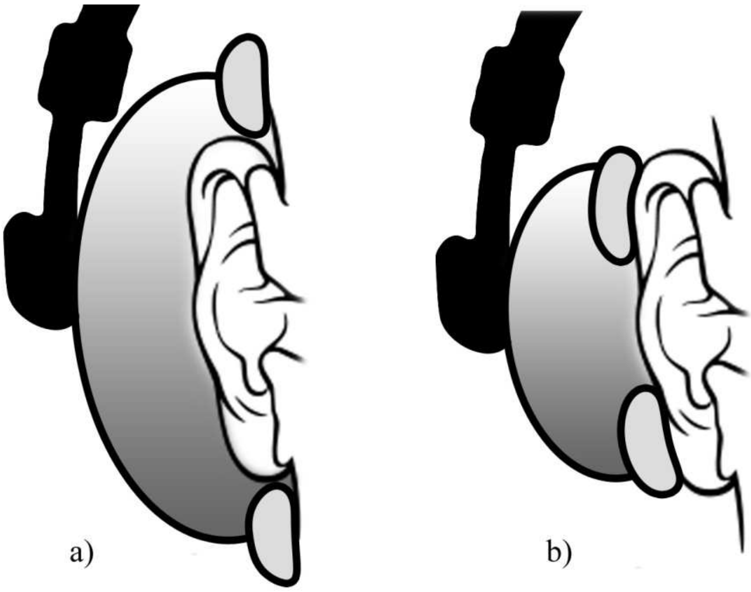



Fig. 2.1 a) Over ear headphone, b) On ear headphone

#### 2.1.1.2 Intra-concha and intra-canal

Ear buds or **intra-concha** headphones (fig. 2.2 a) are the most common types in conjunction with portable usage. The ear pad lies directly at the end of the ear canal, but, because of the specific design, it is not possible to seal the ear canal completely. Noise from outside may pass through. This often induces listening with high volumes to being truncated from the environment. The big advantage is low price, resulting from effectiveness during production because of big lots. Because of variability in wearing positions, they are not appropriate for ANC technique.

The other types are called **intra-canal** headphones (fig. 2.2 b). They are also called insert-style, in-ear monitors (IEMs), or canal phones. This type of headphones is fully inserted into the ear canal. This principle allows for good fit in the ear canal, which leads to favorable sealing from the environment. In order to achieve best performance in wearing comfort, it is possible to fit the front tip of the headphone (canal tip) to the customer's ear canal. This also leads to better sound performance and better sealing from the environment, as well. Additionally, most manufacturers provide the properties of 2 or 3 changeable canal tips because of the variability of ear canals with different users.

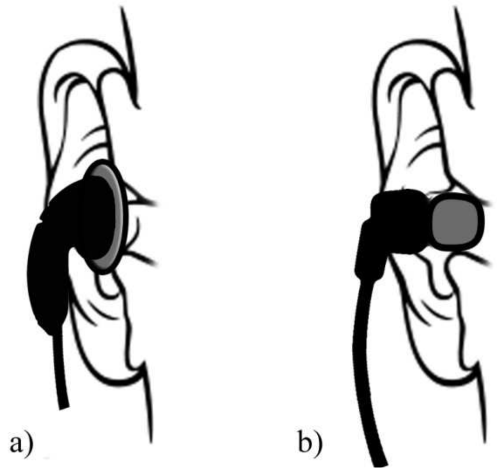



Fig. 2.2 a) IEM's intra concha, b) IEM's intra canal

## 2.1.2 In Ear Monitors (IEMs)

As mentioned before, in the first instance the measurement system was developed for only one of the headphone types introduced. This refers to the type of in-ear monitors or IEM's headphones (fig. 2.2 b). Thousands of different IEM's have been developed over the past decades. The application area opened plenty of different fields but the principles how IEM's work are always the same. The following chapter gives an introduction to the main techniques of common IEM's, especially in relation to ANC performance, so we can become more familiar with this type of headphones.

### 2.1.2.1 Global features

As shown in fig. 2.3, the feed-forward ANC-headphone barely differs from the standard IEM. The two main differences are the microphone on the outside of the headphone for recording ambience noise, and the leakage hole between the speaker front volume and the ear canal (described in 2.1.2.2). Depending on the headphone model, the design and direction of the microphone changes, but the functionality is always the same. To get the best results in ANC performance, the usage of a high quality, low noise omni-directional microphone is recommended. Through the enhancement of MEMS[3] microphones over the past few years, those could also be used, though they have to be carefully chosen to achieve good low frequency response. To provide good low frequency response, the speaker should be as big as possible with at least 8 mm diameter. Additionally, it is important to separate the 3 chambers - speaker front volume, speaker back volume and microphone housing - acoustically from each other. This is important to minimize crosstalk and to avoid acoustic short-circuits. [SCH10]

[3] MEMS (Micro electrical-mechanical System) microphones have a pressure-sensitive diaphragm which is etched directly into a silicon chip, mostly implemented with an integrated preamplifier. [Lee09]

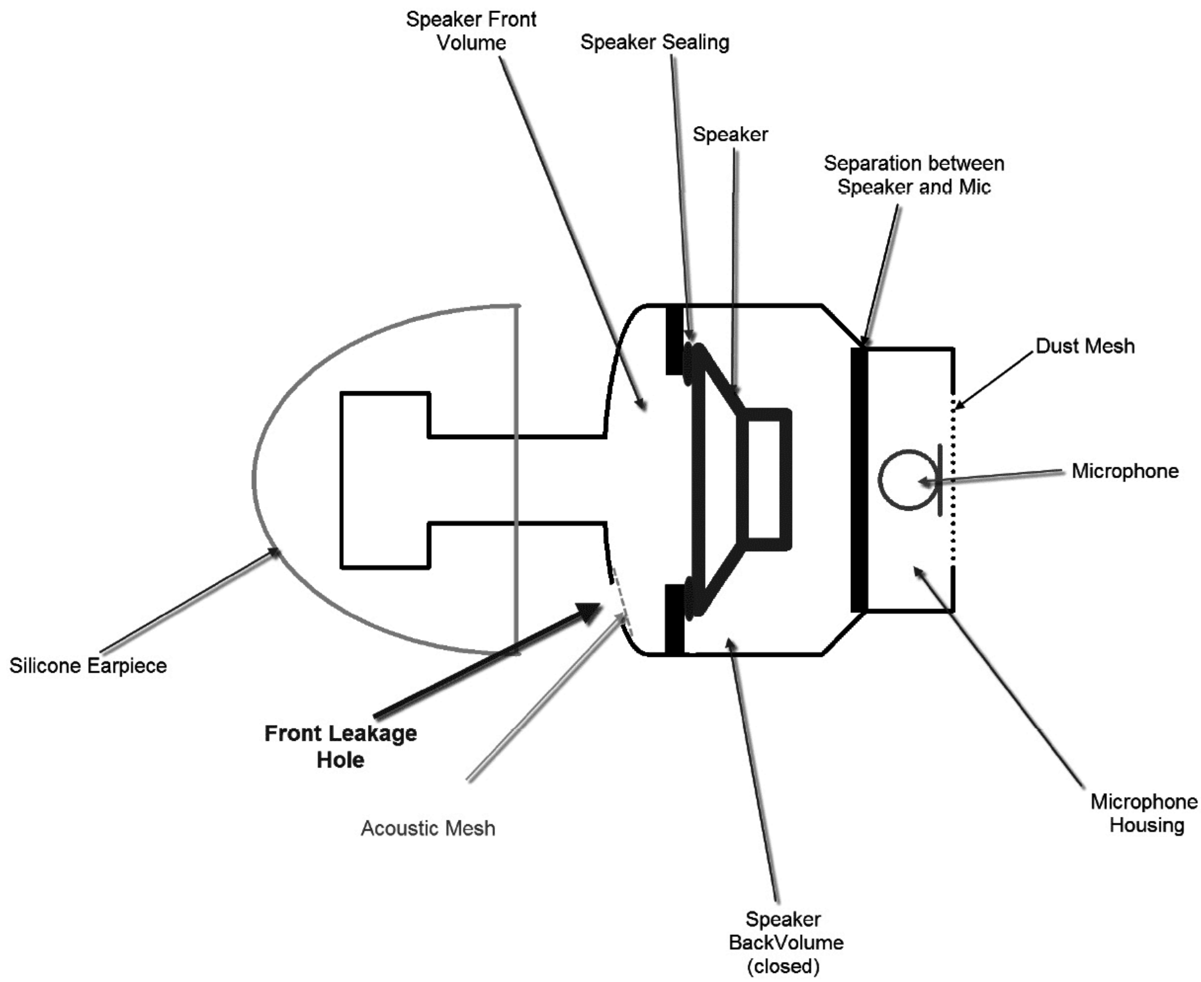



Fig. 2.3 Reference design of an ANC – IEM

### 2.1.2.2 The leakage hole

If IEMs are used in combination with feed-forward noise cancelation, it is necessary to have a leakage hole with defined size between the speaker front volume and the ear canal. This supports constant working conditions for any ear canal volume, which means good ANC performance for many different people. It can be interpreted as a kind of pressure equalization between the IEM and the ear canal because of hermetic sealing of the IEM caused by the canal tip. To find a good leakage hole size, two aspects should be noticed. A big leakage hole size leads to bad passive attenuation at low frequencies and poor low frequency music playback response. Small leakage size leads to bad adaptation to different ear canals. Therefore, a trade-off between good low frequency response and good support for ANC performance has to be found. Thus, the determination of the passive noise cancelation could be a useful tool to find a good leakage hole size. [SCH10]

After finding the right leakage size, a possibility for making further small frequency-dependent changes is the deployment of different acoustic meshes to cover the leakage hole. fig. 2.4 shows the example of an IEM's frequency response with a good leakage size.

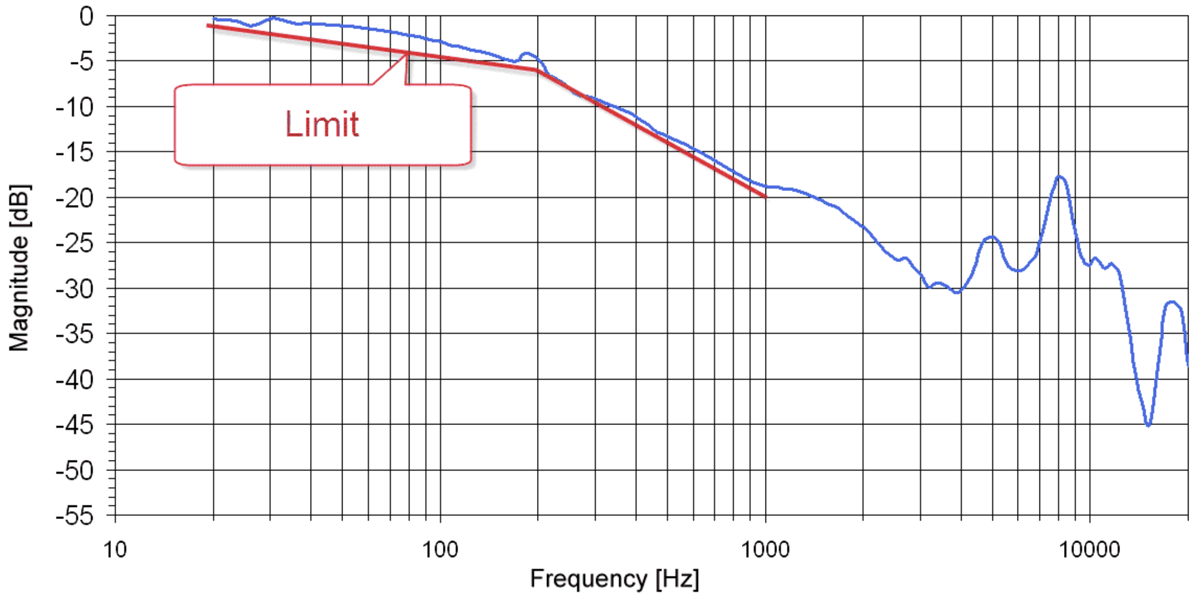



Fig. 2.4 Example for a good leakage hole size [SCH10]

## 2.2 The principals of active noise cancelation

Active noise cancelation is a method for cancelling or – rather – minimizing undesirable noise. This noise can be classified into two types of noise, broadband and narrowband noise. Broadband noise has its energy distributed over a wide frequency range, for example, ambience sound in airplanes. However, narrowband noise shows its energy only at a small frequency range. It is often produced by the oscillation or vibration of mechanical parts, for example, the turbo charger of an engine. Generally, ANC systems work best at low frequencies. This fact has two main reasons. First, the number of modes in a cavity grows rapidly with an increase of frequency [BRO05] and, second, the phase response of the DUT at higher frequencies has a complexity which could not be adapted using standard filter technology.

There are several methods of implementing noise cancelation to work against such disturbing sounds. Below, the feed-forward method, which was implemented over the course of this thesis, will be discussed in detail.

In fact, all noise canceling methods are mainly based on the principle of a phase shift at 180°. As a consequence, this leads to complete signal canceling in case of a pure sine signal, caused by the phenomenon of destructive wave interference (fig. 2.5).

Signal $\mathrm{y}_1(t)$ represents a sine with the amplitude A and the phase $\varphi_1$

$$\mathrm{y}_1(t) = A \cdot \sin(\omega\mathrm{t} + \varphi_1) \qquad [2.1]$$

added with the sinusoidal signal $\mathrm{y}_2(t)$

$$y_2(t) = B \cdot \sin(\omega t + \varphi_2 + 180°) \qquad [2.2]$$

leads to

$$y_1(t) + y_2(t) = 0, \qquad if\ A = B\ and\ \varphi_1 = \varphi_2 \qquad [2.3]$$

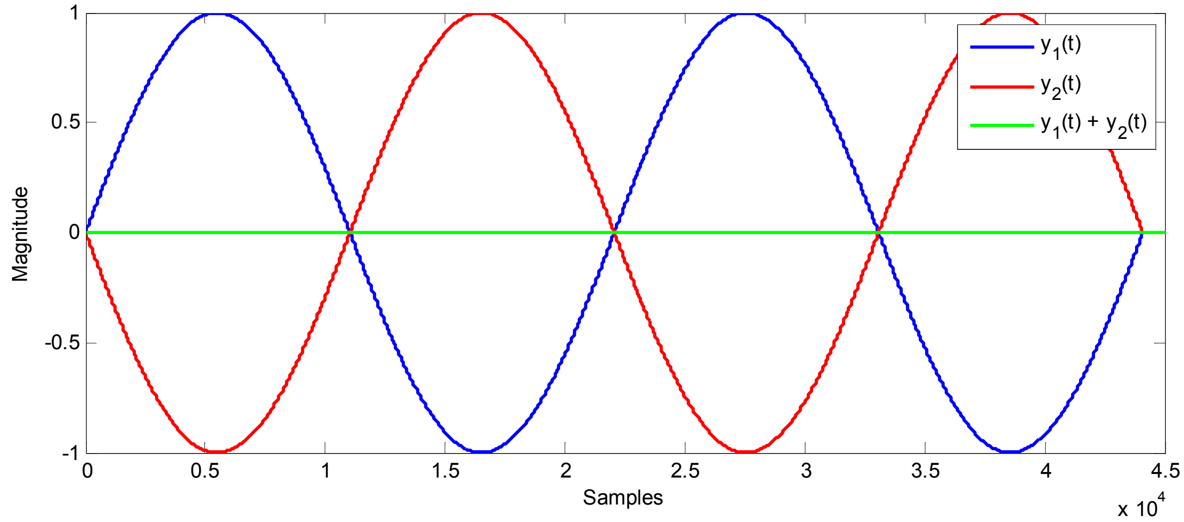



Fig. 2.5 Sine cancelation

Unfortunately, most signals we are surrounded by are not only single tones. The signals the system has to deal with have a complex wave form (fig. 2.6). These complex signals consist of many different frequencies and amplitudes which interfere with each other. Therefore, it is not possible to completely blank undesirable noise. But the advancement in development over the last decades allows realistic attenuation rates in properly working ANC systems at approximately 25 dB up to 35 dB.

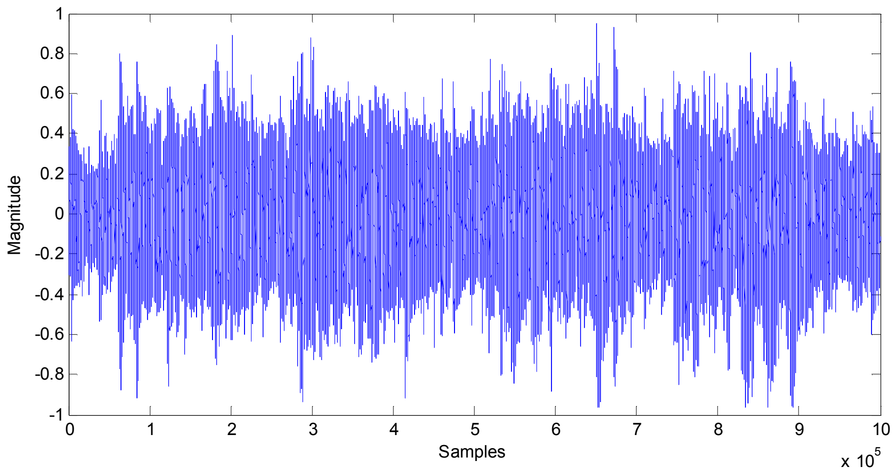



Fig. 2.6 Complex waveform

## 2.3 Feed-forward systems

Feed-forward noise cancelation is a straight forward ANC system at a very high performance level. This method of noise cancelation is especially suitable for small applications. In contrast to feed-backward systems, a loop back, which takes the error signal from in front of the speaker, is not part of the system. Further advantages are:

- Low current consumption
- No latency (cf. adaptive systems)
- Less production costs because of simple topology
- Increased stability, because of no feedback loop

The feed-forward ANC system consists of two parts, a passive and an active part. It is a combination of electric and acoustic properties. Last, but not least, the design of the headphone itself exerts influence on the system because of the relation with passive attenuation.

fig. 2.7 shows the principle process of a feed-forward ANC system as a combination of the passive and the active attenuation parts.

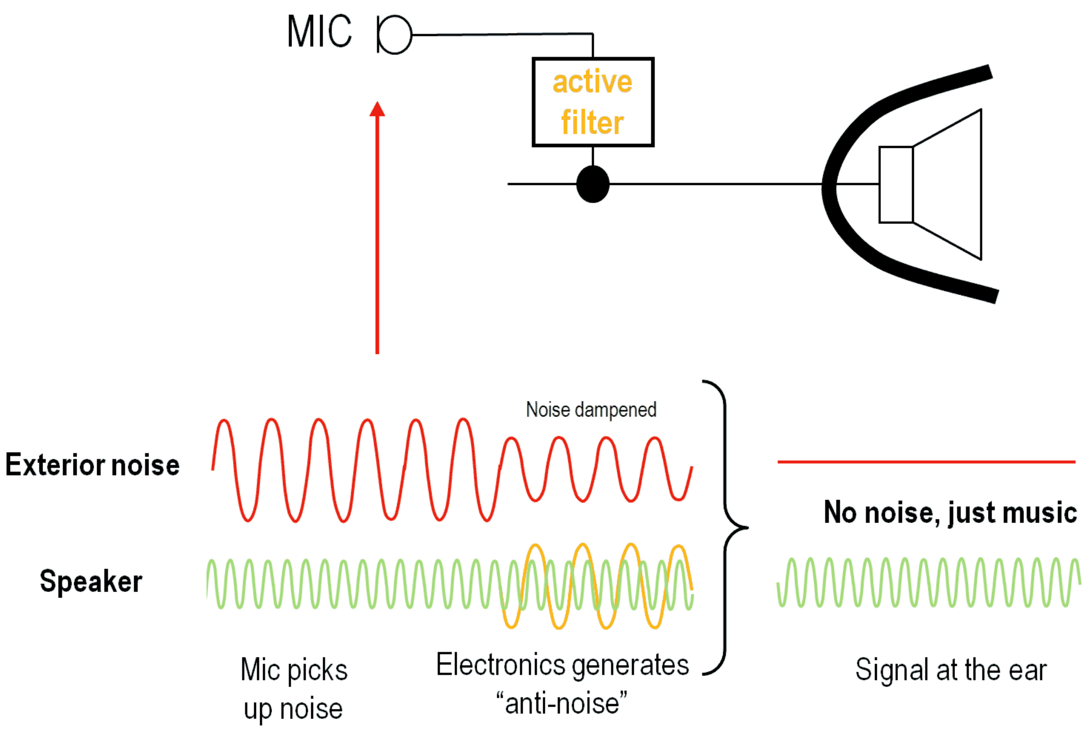



Fig. 2.7 Principles of feed-forward ANC

### 2.3.1 Passive attenuation

The passive attenuation part is caused by the mechanical construction of the headphone. It is similar to using ear muffs to protect the ears against damage caused by extremely loud music or noise. The sealing of the ear canal (fig. 2.2 b) by the headphone tip attenuates the high frequency band very well. It works like a low pass filter with a cut-off frequency at around 300 Hz. The passive attenuation in the upper frequency range highly depends on the structure of the ear muff and is dependent on combining with different ear canals. This means it is instrumental to have a well-designed ear tip for balancing the differences between different ear canals. The best working solution would be about fitting the front tip of the headphone (canal tip) especially to each customer's ear canal, suggested in 2.1.1.2. But this may be too expensive for most users. The attenuation of the low frequencies depends on the size of the leakage hole and the resistance of the acoustic mesh used to cover the hole. Further, the attenuation is limited by size and mass of the IEM. Generally, good passive cancelation of low frequencies is much more difficult than for high frequencies. fig. 2.9 shows the typical passive attenuation curve of an IEM.

### 2.3.2 Active attenuation

The attenuation of the lower frequency range has to be implemented with active components. This electro-acoustic compensation path includes the red marked paths 2, 3 and 4, as illustrated in fig. 2.8. The filter, included in the signal path, equates the filter curve calculated in 2.3.3. The filter gain and the filter phase are the decisive points in the performance of the ANC system.

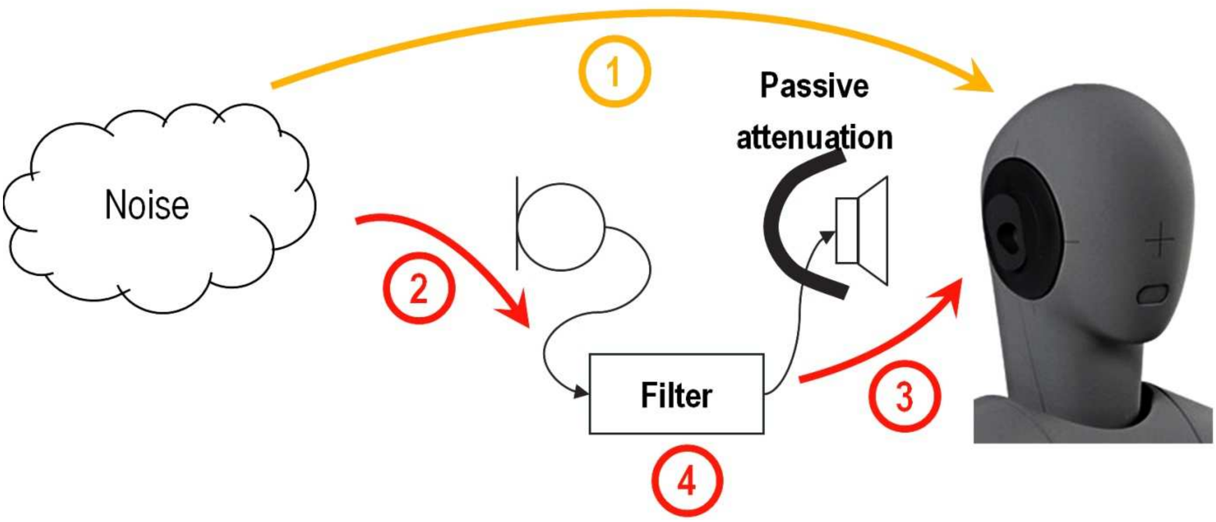



Fig. 2.8 Active and passive noise path

| Path | Function |
|---|---|
| 1 | Passive Attenuation |
| 2 | Noise detected by the ANC microphone |
| 3 | Sound from speaker to the ear |
| 4 | Filter, implemented in the system (depends on path 1, 2 and 3) |

Tab. 2.1 Feed-forward ANC path description

### 2.3.3 Filter determination

The key to a properly-working ANC system is in determining the accurate filter gain and phase curve, corresponding with the existing system. To achieve these values, the measurement of the acoustic and electric paths shown in fig. 2.8 is necessary. These 3 measurements are fundamental for determining the filter coefficients and are described below. To identify the frequency and phase response of the DUT, several measurement techniques can be used. During this thesis, all developments and measurements were made with an exponential sweep as excitation signal. A detailed description of the measurement technique applied can be found in 2.4. Also, a detailed description of the hardware used as part of this thesis can be found further down in chapter 3.

#### 2.3.3.1 Measurement: Passive attenuation at measurement microphone

The passive attenuation curve represents the attenuation caused by the IEM itself. It describes the complete acoustic attenuation influence of the whole IEM at the point of the measurement microphone inside the acoustic coupler (cf. 3.4) which represents the human ear canal. To perform this measurement, it is necessary to mount the IEM inside the provided acoustic coupler (fig. 3.10) and measure the attenuated signal in the simulated ear canal[4]. Both gain and phase responses of the measurement were needed. fig. 2.9 shows the typical frequency and phase response of an IEM measured with the acoustic coupler prototype developed during this thesis.

[4] A real simulation of the human ear canal is not necessary. As a result of the calculations for the filter curve, it can be shown that the transfer function of the acoustic coupler will be canceled during the calculation.

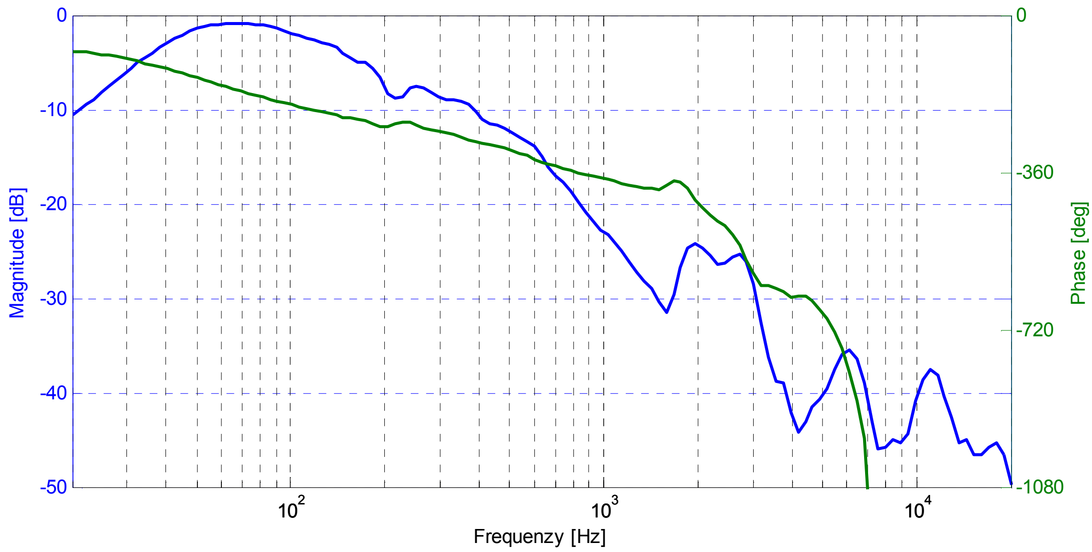



Fig. 2.9 Typical filter gain and phase response, passive attenuation

#### 2.3.3.2 Measurement: Noise at ANC microphone

The measurement at the ANC microphone represents the detection of the pure noise signal. As mentioned in 2.1.2.1, it is recommended to use a microphone with omni-directional characteristics. This works against the fact that the noise detection of the ANC microphone depends on the position of the listener's head. There are systems with more than one microphone in use, but for applications like IEMs or similar small applications, this is not practicable. The property of omni-directional microphones, to behave more like cardioidals the higher the frequency, does not have that much of an impact because the higher frequency range is already covered by the passive attenuation effect and is not captured by the filter anymore.

Fortunately, the effect of directional impact on the ANC microphone is getting smaller the lower the frequencies actually are. Because of the bigger wavelength at lower frequencies, the diffraction effect increases and has a positive impact on noise detection. The combination with the passive attenuation at the higher, more directional frequencies results in a properly-working system covering the whole audible frequency range between 20 Hz and 20 kHz. Fig. 2.10 shows the typical frequency and phase response of the noise measurement at the ANC microphone.

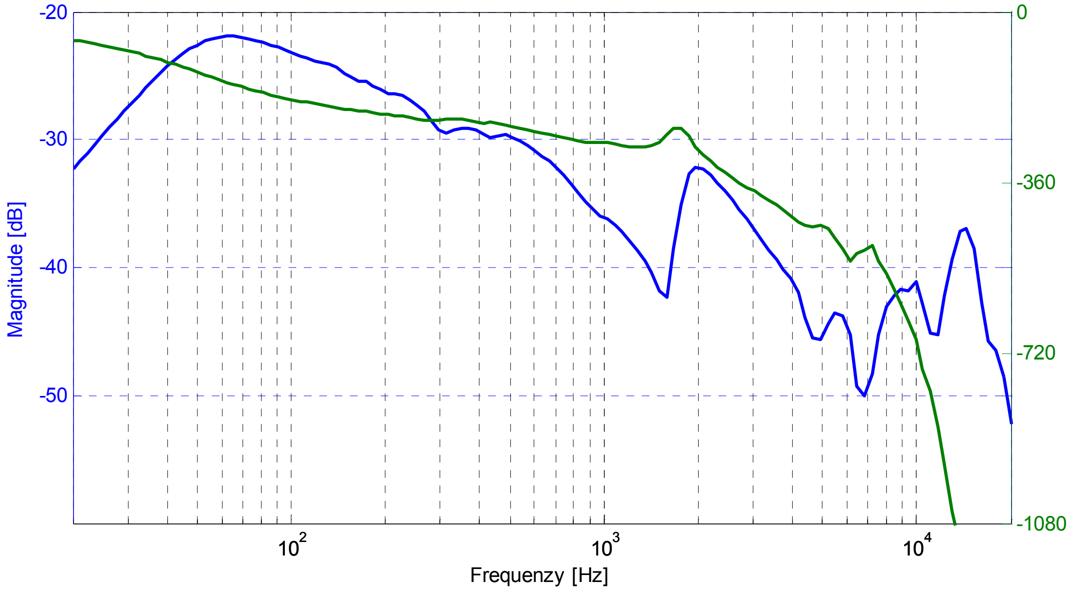



Fig. 2.10 Typical filter gain and phase response, Noise 2 ANC microphone

#### 2.3.3.3 Measurement: Sound at measurement microphone

Measurement 3, as illustrated in fig. 2.8, records the excitation signal as reproduced by the headphone speaker. The resulting frequency and phase response, as shown in fig. 2.11, represents the speaker characteristics of the headphone. The influence of the simulated ear canal will be canceled during the filter calculation (cf. 2.3.3.4). Therefore, this is the last measurement we need to describe all electro-acoustic paths of the IEM. The next step is the calculation of the filter curve for the active filter.

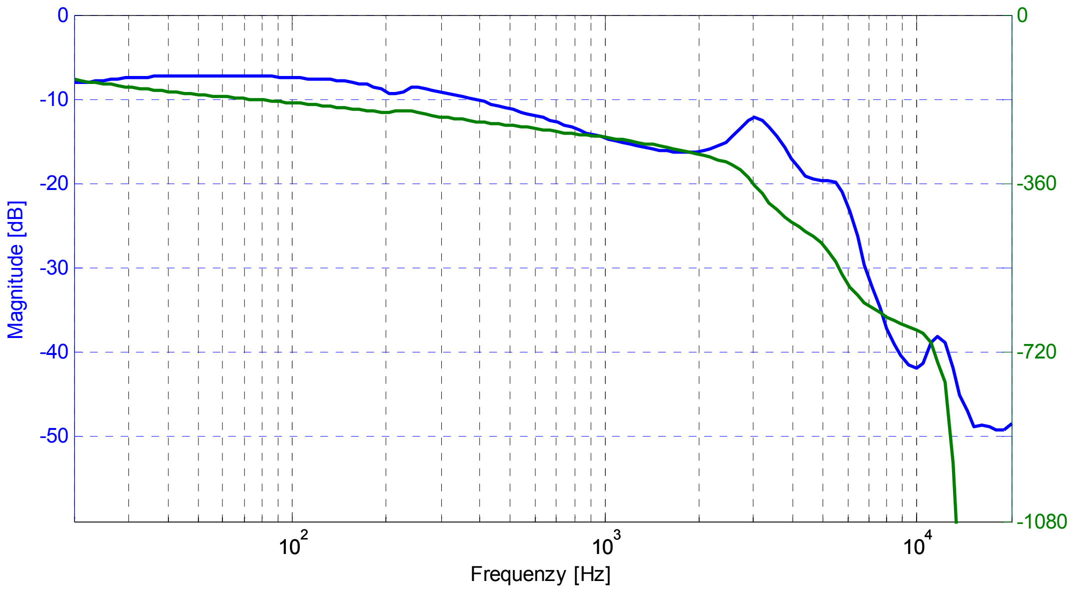



Fig. 2.11 Typical filter gain, Speaker 2 microphone

#### 2.3.3.4 Filter calculation

To achieve the correct frequency and phase values for the required filter curve, some easy calculations are necessary.

$$A_{Filter} = A_1 - (A_2 + A_3) \tag{2.4}$$

leads to the needed filter amplitude $A_{Filter}$, with $A_n$ as amplitudes of the single measurements and

$$\varphi_{Filt} = 180° + \varphi_1 - (\varphi_2 + \varphi_3) \tag{2.5}$$

leads to the needed phase $\varphi_{Filt}$, with $\varphi_n$ as phase values of the single measurements.

This resulting filter now includes all 3 paths of the system. The filter gain and the phase shift with 180° correct the incoming noise, caused by equation [2.4] and **Fehler! Verweisquelle konnte nicht gefunden werden.**], in a way that noise from the outside will be canceled.

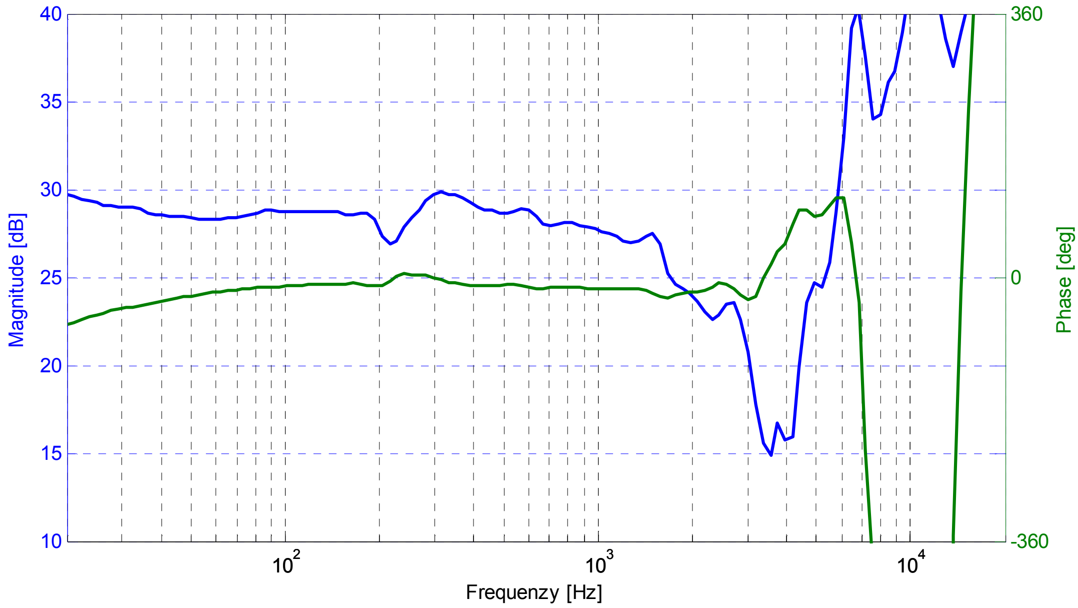



Fig. 2.12 Typical frequency and phase response for a calculated ANC filter

#### 2.3.3.5 Filter topology and implementation

The last step after the filter calculation is the implementation of the resulting curves. Because of the limitations in analog filter circuits, the implemented filter is a smoothed version of the calculated curves. Another reason for the smoothing process is the inaccuracy of the calculated filter at high frequencies. Especially the one-by-one implementation of the phase response at high frequencies makes no sense because of massive phase shifting at high frequencies (cf. fig. 2.12). Fig. 2.13 shows a calculated filter and the implemented filter gain and phase compared to each other.

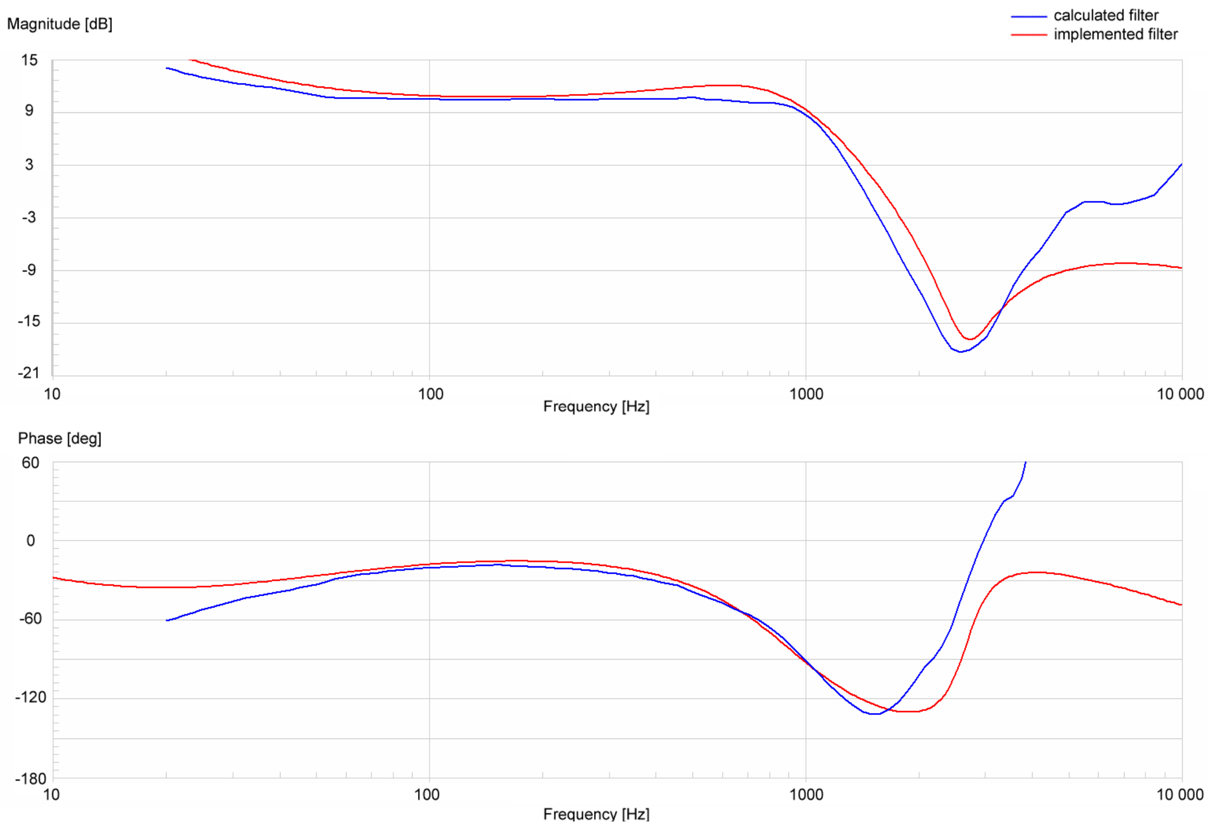



Fig. 2.13 Calculated filter vs. implemented filter

Fig. 2.14 shows the filter topology of the implementation and fig. 2.15 shows a sample schematic of the implementation.

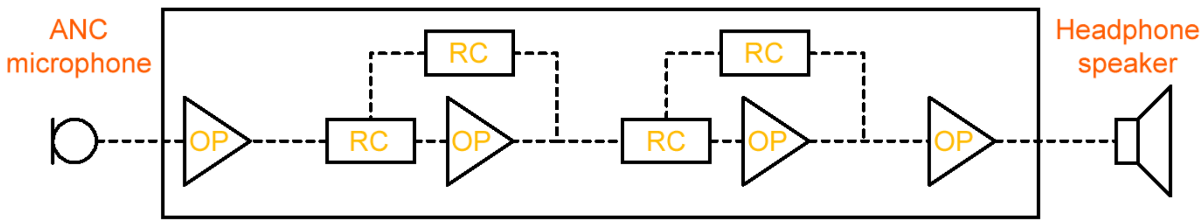



Fig. 2.14 Filter topology

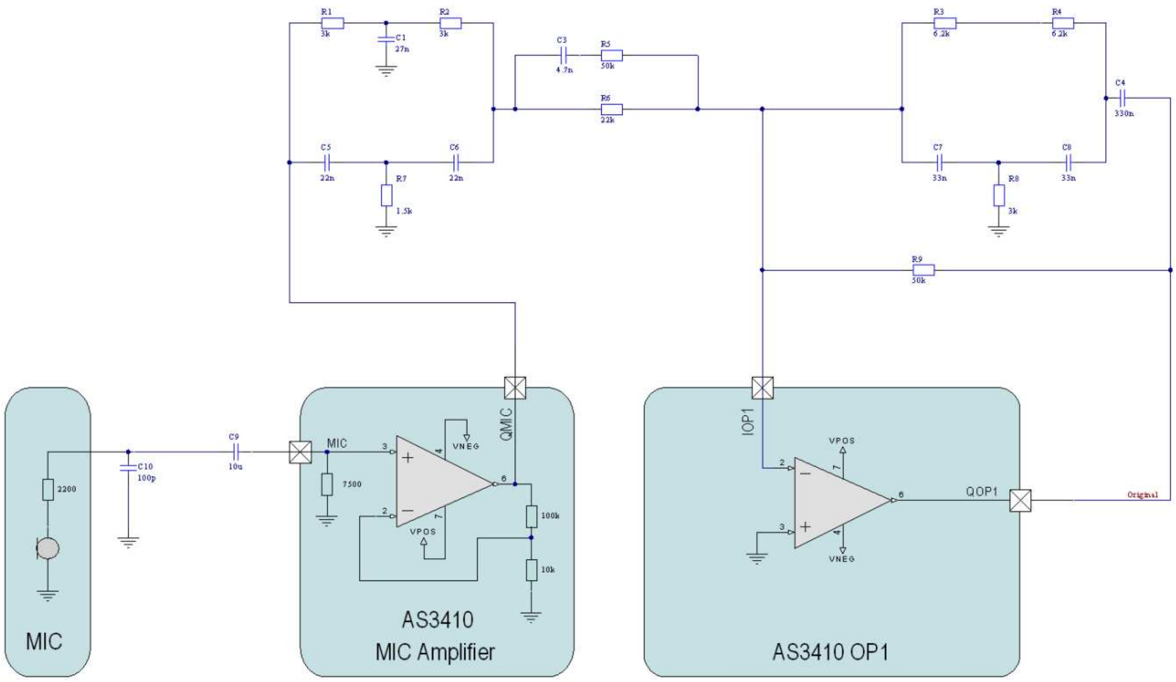



Fig. 2.15 Sample of a filter schematics

## 2.4 Sweep measurements

There are several ways towards achieving the required frequency and phase responses discussed in 2.3. The decision to perform the measurements with an fft-based exponential sine sweep measurement is based on a number of reasons. A big point here is, of course, about accuracy of measurement. The measured values should be as accurate as possible within the range of possibilities of this measurement setup. This refers, in the first instance, to the electro-acoustic components, which show the property of distorting the ideal signals, like the measurement microphone and the speaker used. Another point is the pace of measurements. The customers for whom this system was developed want to keep measurement time short.

### 2.4.1 Excitation signal

There are some advantages using a sine sweep as excitation signal. The small crest factor of 3 dB (theoretically) means a high Signal to Noise Ratio (SNR). Alternatively, the SNR level could be raised by sweep duration. Additionally, sweeps have a higher immunity against time-invariance and non-linearities caused by the electro-acoustic signal chain that can be cut off in a windowing process (fig. 2.19) and/or could be analyzed separately. Further advantages are the possibility of implementation in time domain and a relatively short excitation signal – down to 1 second or less - depends on the application. [MÜL01]

Another method of increasing the SNR and dealing with undesired noise is the Synchronized averaging in time domain method (SATD) [WES10]. The SATD method makes it possible to raise the SNR by increasing the number of measurements. This is a good method if the sound pressure level of the undesirable noise has a much higher level than the excitation signal. The improvement of the SNR is caused by synchronous averaging of the achieved system responses. With this method, the SNR could be improved almost arbitrarily, but at the expense of the duration of measurement. The improvement of the SNR is approximately

$$\log_2 n \cdot 3 = SNR_+, \qquad [2.6]$$

where $n$ is the number of measurements and $SNR_+$ is the increase in dB.

There are several kinds of sweep types which can be used for system identification measurements. The most common types are linear sweeps and exponential (also called logarithmic) sweeps. It is possible to generate these sweeps in the time or in the frequency domain. The implementation in the frequency domain produces the advantage of the amplitude spectrum to be shaped according to the desired spectral energy distribution. This allows creating arbitrary non-linear types of sweeps with varying energy distribution [WES10]. Sweeps generated in the time domain have the advantage of a perfect envelope. This results in an ideal crest factor of 3 dB. Therefore, the spectrum has some irregularities, especially at the beginning of the sweep - at the low frequencies (fig. 2.16 & fig. 2.17). This is caused by the switch-on process. But this effect should not have any impact on the analyses of the frequency response of the DUT. The subtraction of the reference signal during the signal processing stages described in 2.5 yields to a balancing of these effects.

Below the implementation in time domain of the linear and the exponential sweep will be discussed.

## 2.4.2 Linear sweeps

A linear sine sweep is a sine signal with a time-dependent frequency response. The frequency increase follows a linear function. The implementation of the linear sine sweep could be described as follows [WES10]:

The general form of a sine sweep $x(t)$ with amplitude $A$ and phase $\varphi(t)$, depending on the pulsatance $\omega(t)$ is

$$x(t) = A \cdot \sin \varphi(t), \qquad \varphi(t) = \int \omega(t) dt \qquad [2.7]$$

To generate a linear sine sweep in the time domain, the instantaneous frequency $\omega(t)$ is following a linear function:

$$\omega(t) = k \cdot t + d. \qquad [2.8]$$

With the assumption of the start frequency $\omega_{start}$, the end frequency $\omega_{stop}$ and the phase shift $\varphi(0)$ during the start of the sweep

$$\omega(0) = \omega_{start}, \qquad \omega(T) = \omega_{stop}, \qquad \varphi(0) = 0, \tag{2.9}$$

and the following boundary conditions

$$d = \omega_{start}, \quad k = \frac{\omega_{stop} - \omega_{start}}{T}, \tag{2.10}$$

the linear sweep in time domain has the following form

$$x(t) = A \cdot \sin\left[2 \cdot \pi \left(\frac{f_{stop} - f_{start}}{2 \cdot T} \cdot t + f_{start}\right) \cdot t\right]. \tag{2.11}$$

The amplitude spectrum of the linear sweep is uniformly distributed over the whole defined frequency range. Fig. 2.16 shows the magnitude of a linear sweep in time and frequency domain[5]. As also shown in fig. 2.16, the magnitude stability of the linear sine sweep at low frequencies is worse according to the exponential sine sweep shown in fig. 2.17. Especially in the illustration with a logarithmically-scaled frequency axis, the variability of the magnitude of the linear sweep becomes apparent. This behavior is a result of the linear increase of the frequency over time. The sweep "spends" the same time on each frequency over the whole frequency range (fig. 2.18).

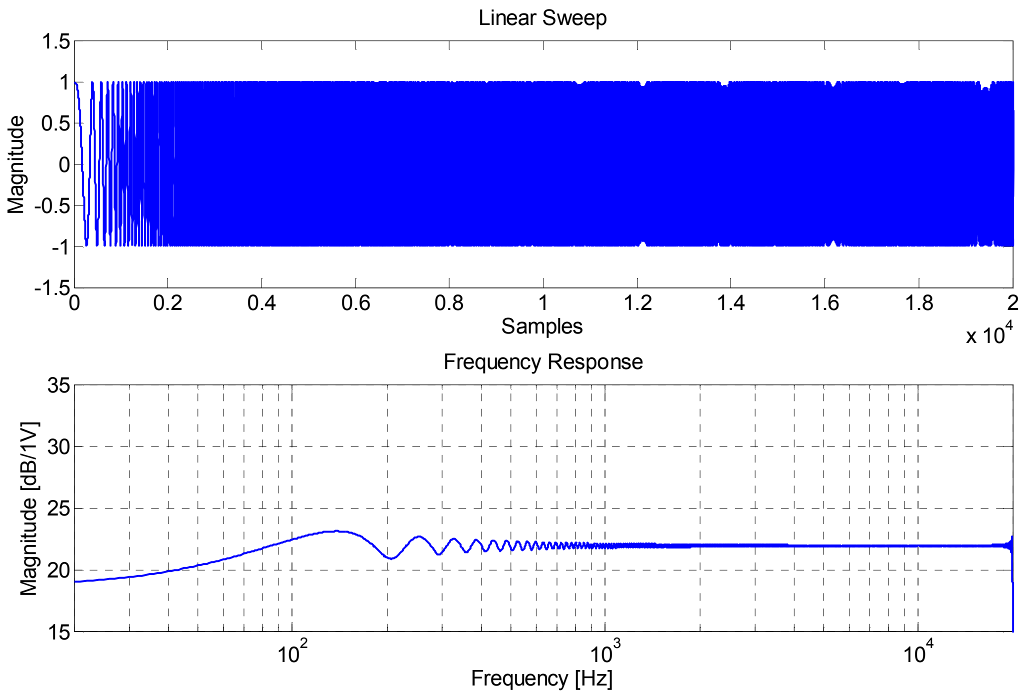



Fig. 2.16 Linear sweep - Time domain (top), Magnitude spectrum (bottom)

[5] The sweep duration was 2 seconds and the frequency range was 20 Hz up to 20 kHz with a sample rate of 44.1 kHz.

### 2.4.3 Exponential sweeps

In contrast to the linear sweep, the instantaneous frequency $\omega(t)$ of the exponential sweep follows an exponential function. In the time domain, this can be described by the following equation [WES10]:

$$\omega(t) = k \cdot a^{\frac{t}{\tau}} \qquad [2.12]$$

The first two boundary conditions lead to the determination of the 2 unknown constants $k$ and $\tau$.

$$k = \omega_{start}, \quad k \cdot a^{\frac{T}{\tau}} = \omega_{stop} \Rightarrow \tau = \frac{T}{\log_a\left(\frac{\omega_{start}}{\omega_{stop}}\right)} \qquad [2.13]$$

The phase $\varphi(t)$ could be described as

$$\varphi(t) = k \cdot \tau \frac{a^{\frac{t}{\tau}}}{\ln a} + C \qquad [2.14]$$

With the third boundary condition

$$C = -\frac{k \cdot \tau}{\ln a}, \qquad [2.15]$$

the phase φ(t) is represented in the time domain as

$$\varphi(t) = \frac{k \cdot \tau}{\ln a} \cdot \left[a^{\frac{t}{\tau} \cdot \log_a\left(\frac{f_{stop}}{f_{start}}\right)} - 1\right]. \qquad [2.16]$$

The exponential sweep $x(t)$ in time domain has the following form

$$x(t) = A \cdot \sin\left\{2\pi \cdot \frac{f_{start} \cdot t}{\ln\left(\frac{f_{stop}}{f_{start}}\right)} \cdot \left[\left(\frac{f_{stop}}{f_{start}}\right)^{\frac{t}{T}} - 1\right]\right\} \qquad [2.17]$$

The amplitude spectrum of an exponential sweep is distributed like pink noise. This is caused by the exponential increase of the frequency over time. As a result of this, the sweep "spends" more time at lower frequencies (fig. 2.18) during generation in contrast to the linear sweep. As shown in fig. 2.17 the weight of the energy distribution is more concentrated at the lower frequencies[6]. Additionally, the cut-off frequency for the instabilities at the low end is shifted down to approximately 50 Hz in contrast to the linear sweep. This is the reason why an

[6] According to the linear sweep, the sweep duration of the logarithmic sweep was 2 seconds and the frequency range was 20 Hz up to 20 kHz with a sample rate of 44.1 kHz.

exponential sweep is used for ANC measurements. Good ANC performance is especially important at lower frequencies. Thus, more energy at the lower frequencies leads to more accuracy at lower frequencies.

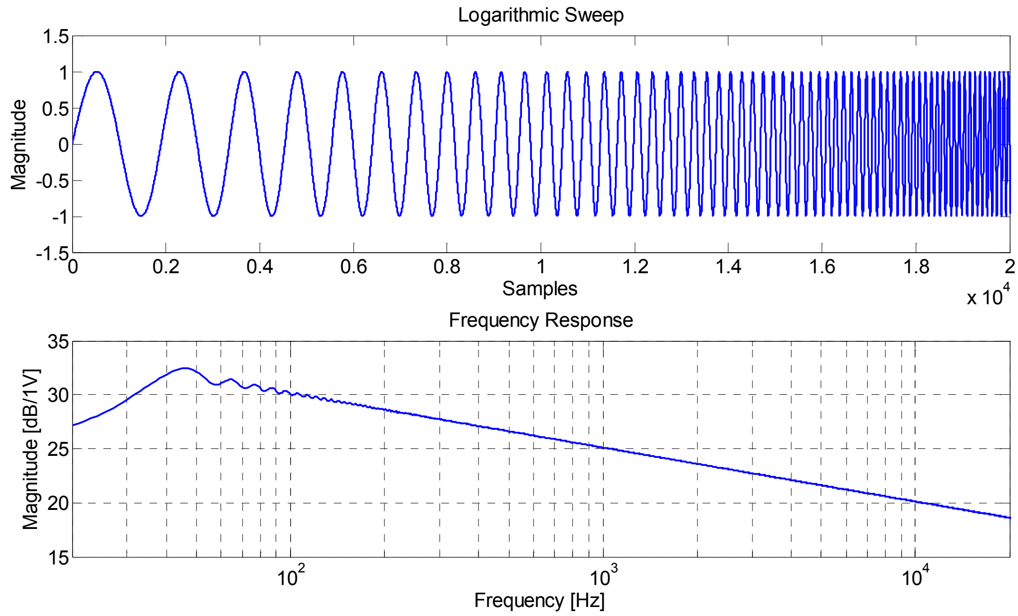



Fig. 2.17 Exponential sweep - Time domain (top), Magnitude spectrum (bottom)

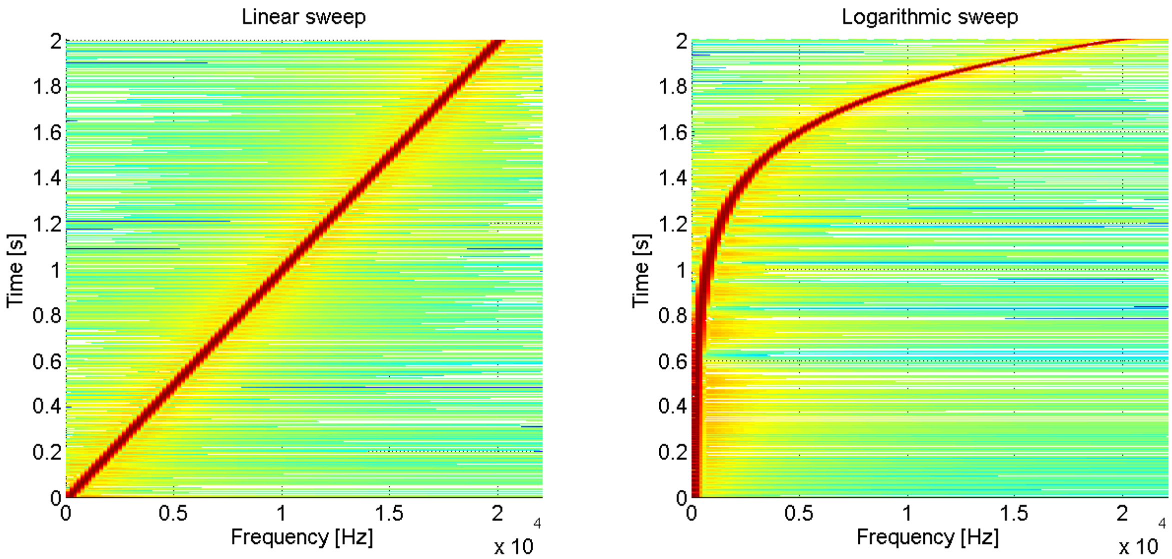



Fig. 2.18 Frequency response over time, linear and exponential sweep

## 2.5 Signal processing

To achieve the final frequency and phase response of any DUT by using the exponential sweep, the signal processing stages illustrated in fig. 2.19 have to be performed [MÜL01]. Below, the main theories for understanding these various signal processing stages will be discussed. We assume that the underlying system is linear and time-invariant.

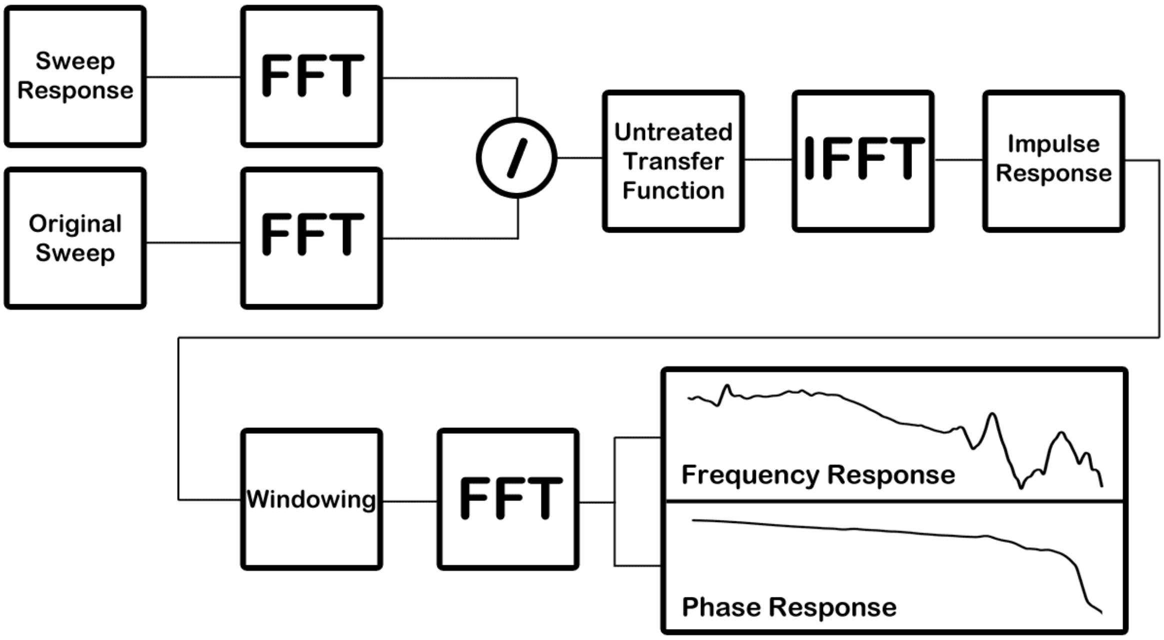



Fig. 2.19 Signal processing flowchart for fft based sweep measurements

## 2.5.1 Sweeps

The generation of the exponential sine sweep was discussed in 2.4.3. Now, this sweep was used as an excitation signal for the system identification. As illustrated in fig. 2.19, the input of the signal processing stages is the sweep response of the DUT and the original excitation signal. The runtime of both signals is exactly the same, with the exception of the sweep response having been delayed over the additional runtime caused by the DUT[7].

A big point about doing measurements in bigger rooms is padding the sweep signal with zeros. This is necessary to make sure the sweep response was not cut off during measurement. The length of the zeros should be at least as long as the combined delayed direct sound and reverberation time. Especially through using short sweeps in big rooms, the reverberation time - particularly at low frequencies - could be considerably longer than the sweep itself [MÜL08]. But due to the fact of good attenuation inside the measurement box, reverberation has no relevant influence on the transfer function. As a precaution, the sweep for this measurement system was padded with 22000 Samples. This is approximately equal to 0.5 seconds at a sample rate of 44.1 kHz.

[7] Because of the very low microphone-speaker distance, about 1 cm, the delay could be ignored. In fact, at a sample rate of 44.1 kHz, the time resolution is approximately equal to the time the sound needs to pass this distance.

## 2.5.2 FFT and transfer functions

The goal of the measurement system is about achieving the frequency and phase response of the DUT through applying the exponential sweep. The solution for this endeavor can be brought about via fourier transformation. The fourier transformation is an essential tool in all matters of acoustics. The basic property of the fourier transformation is the transformation of a signal from the time domain in the frequency domain. This is founded on the principle that each periodic signal could be split into single sine and cosine signals. The sum of these signals represents the complete signal. Usually, the fourier transform is calculated as fast fourier transform, or FFT. The FFT is an algorithm that leads to effective calculation of the fourier transform. This results in less computing time.

In case of this project, the appearance of all signals is time-discrete. Therefore, the discrete fourier transformation (DFT) can be used. The definition of the DFT has the following form: [OPP99]

$$X[k] = \sum_{k=0}^{N-1} x[n] W_N^{kn}, \qquad k = 0, 1, \ldots, N-1 \tag{2.18}$$

with $W_N = e^{j(\frac{2\pi}{N})kn}$.

The inverse discrete fourier transformation (IDFT) is defined by

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] W_N^{-kn}, \qquad n = 0, 1, \ldots, N-1 \tag{2.19}$$

The next step on the way to the frequency and phase response of the DUT is the determination of the transfer function. Based on system theory, the transfer function *H(f)* of a system is defined in the frequency domain as

$$H(f) = \frac{Y(f)}{X(f)}, \tag{2.20}$$

with *X(f)*as input signal and *Y(f)* as output signal.

In order to achieve the transfer function of the DUT, we only have to divide the spectrum of the sweep response by the spectrum of the reference sweep. The result is the untreated transfer function. This transfer function includes all harmonic distortions and other possible noise effects which could interrupt the result. But, as discussed in 2.4, one advantage of the sweep measurement is less sensibility towards uncorrelated annoying noise. Therefore, it is necessary to compute the IDFT, so to achieve the impulse response of the system.

### 2.5.3 Impulse response

The computation of the impulse response allows cutting out a big part of undesired background noise and non-linearities caused by the signal chain. This is possible because the impulse response contains the same information as the transfer function. Via the FFT, it is possible to transform the impulse response into the transfer function, and vice-versa, and use the advantages of both domains [MÜL08].

The cut-off process of the undesired signal parts can be performed by windowing the impulse response. Other than the mentioned advantages of windowing, the window size has to be chosen carefully. In fact, a small window size eliminates all reflections, non-linearities and other undesirable effects but it could also have a negative impact on the frequency characteristics of the transfer function. Depending on window size, the frequency response will be more flattened the shorter the window is. This effect especially appears at low frequencies [MÜL08]. By windowing the frequency response of the DUT in this measurement system, there is no risk of flattening caused by a short window. The reason for this is the well-attenuated measurement box with practically no reflections and very little non-linearities. Thus, the window length was chosen at 44000 samples to achieve very good resolution at low frequencies. After windowing the impulse response, another DFT computation leads to the desired frequency and phase response of the DUT.

### 2.5.4 Interpolation and smoothing

As discussed in 4.2.3 the measured data was treated by means of an interpolating and a smoothing process. These processes will ensure that the amount of measurement data does not get out of bounds. Below, the algorithms used for both processes are being described.

#### 2.5.4.1 Interpolation

The algorithm for the interpolation is a straight-forward linear interpolation. There was the possibility of using more accurate algorithms but these algorithms are much more complex and, therefore, require more time for computing. Furthermore, the resolution of the measurement is high enough, so linear interpolation is a sufficient tool. Additionally, in this case, interpolation was not used for generating missing data because of low resolution. It is rather used as a tool for achieving data point between the measured values - to achieve the data points for logarithmic scaling (cf. 4.2.3).

#### 2.5.4.2 Smoothing

The smoothing is necessary for approximating the measured data on the later-implemented filter curve (cf. 4.2.3). Therefore, the implemented smoothing algorithm is using a moving average filter to compute the smoothed filter curve. This process is equivalent to a low pass filtering with the response of smoothing. The moving average filter can be described by the following difference equation:

$$y_s(i) = \frac{1}{2N+1}\big(y(i+N) + y(i+N-1) + \cdots + y(i-N)\big), \qquad [2.21]$$

with $y_s(i)$ as the smoothed value for the $i$ th data point and $N$ numbers of neighboring data points on either side of $y_s(i)$ and 2N+1 as span [MAT10].

# 3 The Measurement System

The following measurements are depending on the main principles of feed-forward noise cancelation as described in 2.2.

Note: In the following chapter, to simplify matters the following abbreviations will be used:

AH = Artificial Head

ANC = Active Noise Cancelling

IEM = In Ear Monitor

A detailed list of all abbreviations used can be found at the beginning of this thesis.

Specifications and Datasheets depending AH, Panasonic WM-61A microphone, G.R.A.S. measurement microphone with preamplifier, evaluation board, all used speaker types, IEC 711 coupler and other relevant technical information can be found in Appendix B. Furthermore Appendix C includes some of the construction drawings of developed parts like the coupler prototypes and some parts of the measurement box. A 3D model and detailed construction drawings of all parts developed during this thesis can be found at the enclosed data CD.

The goal of this thesis is not only about optimizing the cost factor of this measurement system. The other important point is about simplification in handling the system. This comprised a proper software graphical user interface (GUI), as well as simple-to-handle hardware ambience. Development of the required parts to attain such a system will be described below.

## 3.1 The principle measurement process

Up until now, the measurements were performed in an acoustically well-damped measurement room. For performing the measurements of steps 1 to 4, as described later in this chapter, an AH was taken. The IEM was coupled to an occluded-ear simulator, as recommended in [IEC06] with the AH. The calibrated measurement microphone is located at the base of the ear simulator [IEC10] (inside the AH).

The generated test signal was a sine sweep with a frequency range from 40 Hz to 10 kHz. An 8” high-end studio speaker with a very linear frequency response represents the signal source via an audio amplifier and was placed ~ 500 mm symmetrically in front of the IEM. The evaluation board AS 3501 was used as preamplifier for the ANC microphone and as amplifier for the IEM speaker. Audio Precision was used for all amplitude and phase measurements and signal generation.

Fig. 3.1 shows the principle measurement process applied up until now. This represents a good and precise measurement set-up with the very best equipment money can buy. But there are still a few things to improve with this system. This setup consists of a couple of components which have all been connected in the right way. In most the cases it is not possible to leave the system in the measurement setup and you lose plenty of time setting up your system and making it work correctly (including all calibration gadgets). Further, this is a very high–priced solution making it uneconomical/impossible for smaller companies to reproduce. Actually, it is not necessary to be that precise because of the limited usage range of the measurement application. This also means that some of the performance parameters can be reduced to a level adequate for the ANC measurement, so to try to achieve the best possible price-performance ratio for this characterization system.

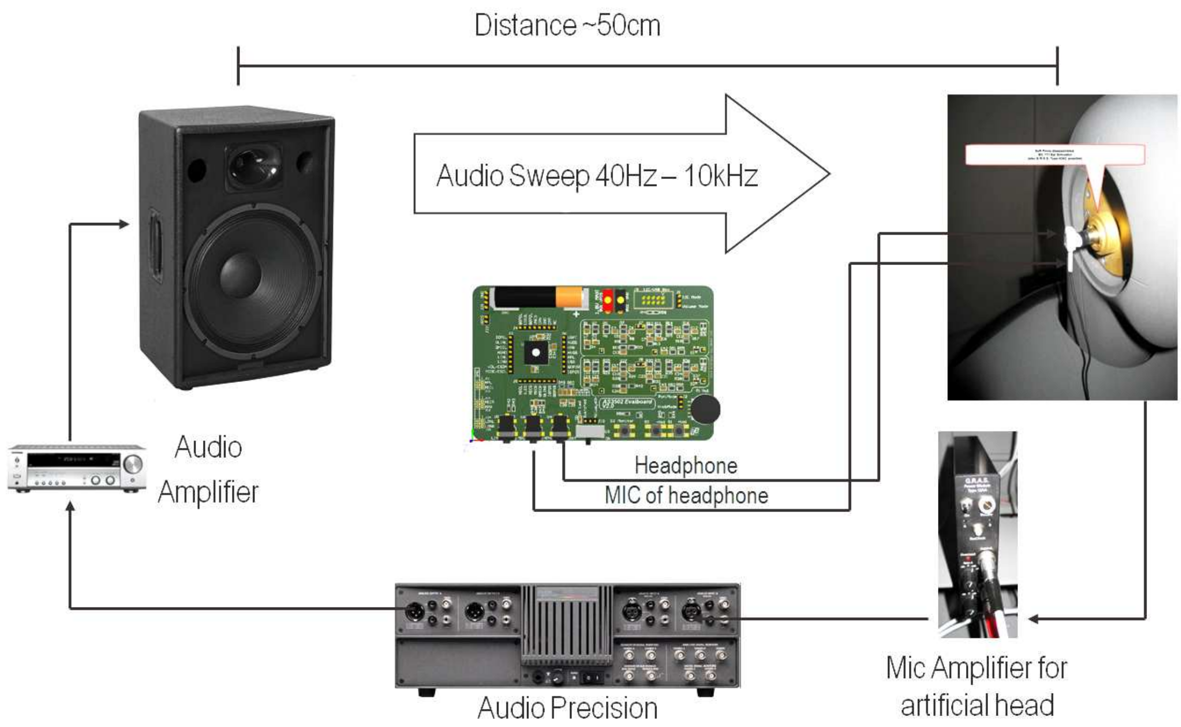



Fig. 3.1 Application flow

## 3.2 System peripherals

### 3.2.1 The speaker types

During the evaluation process, three different types of speakers were used. Type 1 is a high-end speaker and was used as reference source for all measurements in the acoustic chamber. Type 2 and type 3 were especially selected taking into account the aspect of lower and middle-priced segments and are intended to be used in the measurement box.

| | Type 1 | Type 2 | Type 3 |
|---|---|---|---|
| **Speaker** | 8” KSdigital CS8 Studiomonitor, coaxial | 4” Visaton FR 10, with tweeter dome | 5” Ciare HX135 |
| **Cost** | High price segment | Lower price segment | Middle-priced segment |
| **Frequency range** | 38-28.000 Hz[8] | 80-20.000 Hz[8] | 50-20.000 Hz[8] |
| **Nominal power** | 80 Watts | 30 Watts | 40 Watts |
| **Impedance** | 8 Ohms | 8 Ohms | 8 Ohms |
| **Resonance frequency** | unknown | 88 Hz | 54 Hz |
| **Used in box** | | X | X |
| **Used in chamber** | X | | |

Tab. 3.1 Different Speakers used for measurements

## 3.2.2 The microphones

Two types of microphones were used during the evaluation period. The permanent parallel usage of both microphone types ensured constant verification of the measurement results.

### 3.2.2.1 High-end measurement microphone

A high-end microphone was used for basic evaluation measurements of the acoustic coupler and the measurement box. Also, the frequency & phase response of the two speakers used in the box was measured using this microphone. This was handy to have as reference to the cost-optimized measurement solution.

Microphone type: G.R.A.S. 40 AF, ½” free-field microphone, omni-directional.

### 3.2.2.2 Measurement microphone used in the acoustic coupler

The microphone finally used in the characterization system is a 6 mm electret condenser type (fig. 3.2). This makes it possible to mount the microphone directly into the acoustic coupler (fig. 3.10) to achieve optimal space-saving technology. Additionally, the signal path is as short as possible and the attenuation on the backside of the coupler will increase. The diameter of the microphone was adapted with adhesive tape to perfectly match the diameter of

[8] Frequency response drop-down at -10 dB

the cavity. To avoid external noise and achieve best possible signal-to-noise ratio, a coaxial cable was used for signal transfer.

Microphone type: Panasonic WM-61A omni-directional back electret condenser microphone cartridge.

Fig. 3.2 Adapted measurement microphone Panasonic WM-61A

## 3.2.3 The microphone – pre-amplifiers

### 3.2.3.1 High end pre-amplifier

The G.R.A.S. microphone pre-amplifier Type 26HG was used as associated facility for the G.R.A.S. high-end measurement microphone.

### 3.2.3.2 Pre-amplifier for the acoustic coupler microphone

In order to guarantee best performance of the measurement system in association with design and cost, a custom made pre-amplifier, to fit the particular needs of the measurement microphone, was built. The pre-amplifier was assembled on a circuit board the coupler was also mounted on directly. This results in space-saving architecture, low power consumption and, additionally, attenuation on the backside of the coupler (see fig. 3.3). The printed circuit board (PCB) was especially designed to minimize reflections. Thus, the width of the 3 bulks the PCB was fixed with was kept minimal in dimensions. With a thickness of 3 mm the PCB has, 20 mm width was enough to be sure that the PCB becomes not instable or vibrates too much. To support inter-changeability of the acoustic couplers and also guarantee safe signal transmission, the connector on the backside of the PCB was designed as SMA Jack.

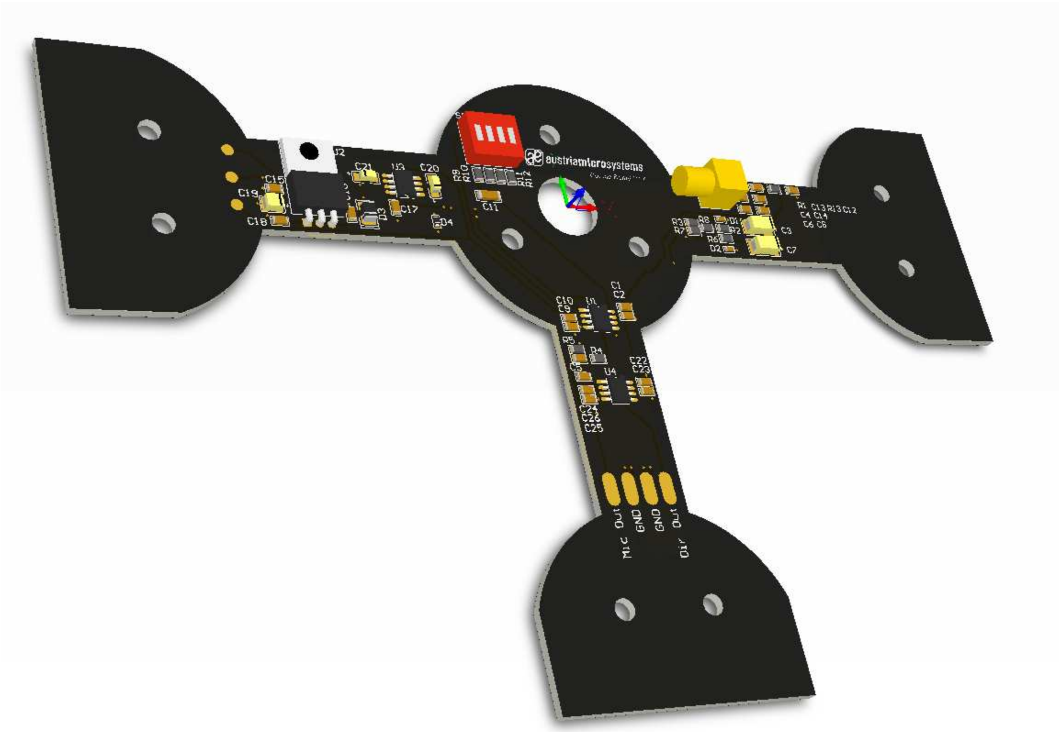



Fig. 3.3 3D model – PCB backside with pre-amplifier circuit

Fig. 3.5 shows the schematic of the pre-amplifier circuit for the measurement microphone. The circuit works with 5V power supply. A built-in switch provides for adjusting microphone gain. A second op-amplifier (fig. 3.5, right bottom) guarantees stable working conditions for the additionally-supported direct output signal of the measurement microphone. A power supply circuit reduces the incoming voltage of 9V down to the required 5V (fig. 3.4).

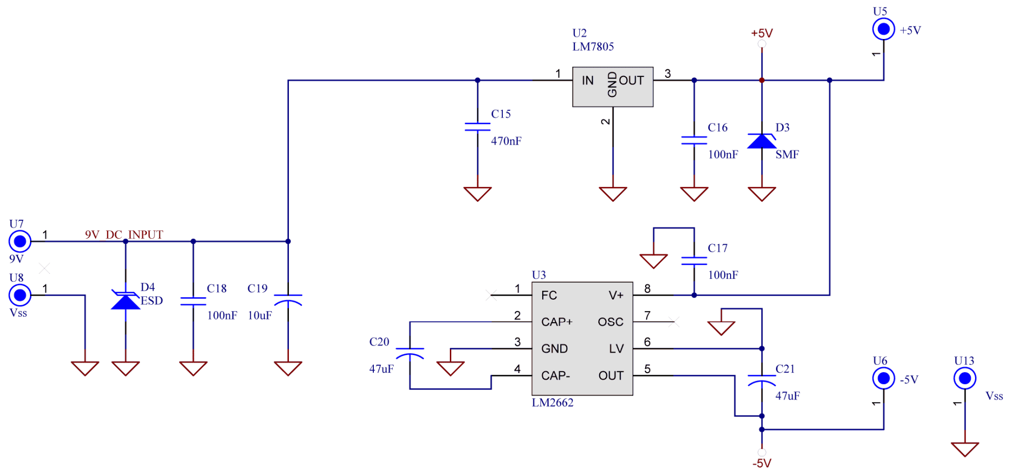



Fig. 3.4 Schematic power supply microphone pre - amplifier

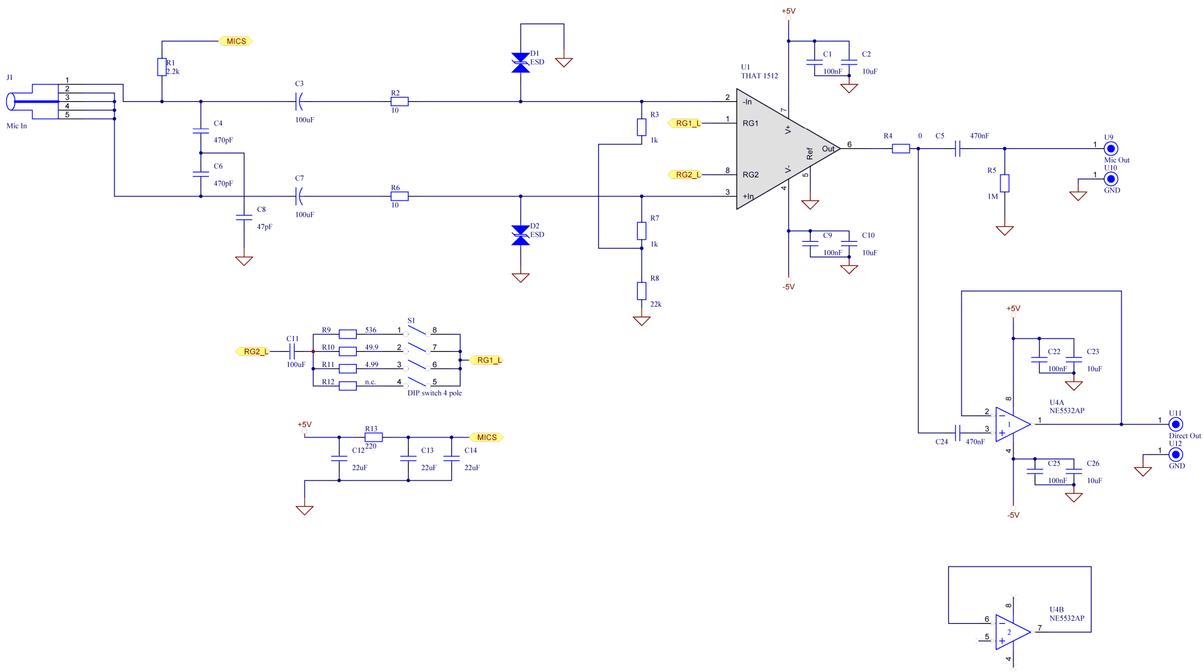



Fig. 3.5 Schematic measurement microphone pre-amplifier

# 3.3 Measurements

All measurements for the development of the acoustic coupler were made under the same conditions. At this point you will find a detailed description of the ambience and the equipment used.

## 3.3.1 Conditions

### 3.3.1.1 Measurement room

| | |
|---|---|
| Dimensions (L/W/H): | 230 x 230 x 290 m |
| Reverberation Time @ 1 kHz: | 0.084 sec |
| Temperature: | 26 C° |

## 3.3.2 Equipment

| | |
|---|---|
| Speaker: | KSdigital CS8 Studiomonitor |
| Measurement software / hardware: | Audio Precision (SYS - 2722) |
| General measurement microphone: | G.R.A.S. 40 AF |
| Measurement microphone coupler: | Panasonic WM-61A |

### 3.3.3 Configuration details

| | |
|---|---|
| Distance speaker - IEM: | 235 mm (symmetrically in front of the speaker) |
| Distance coupler - ground: | 1250 mm |
| Signal type: | Stepped Sine sweep 20 (40) Hz - 20 kHz |

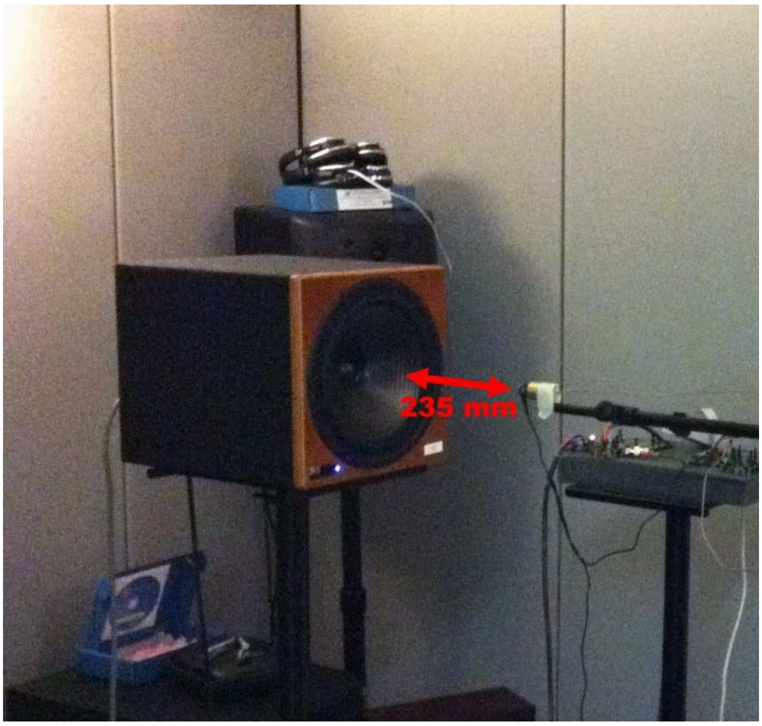



Fig. 3.6 Measurement setup

### 3.3.4 The artificial head

The AH developed and distributed by *Head acoustics* was equipped with an IEC 711 acoustic coupler. In order to perform the measurements, it was necessary to modify the AH in 2 steps[9]:

1) To avoid undesirable reflections produced at the uncovered surface of the AH's - "earless" side, the surface was covered with broadband absorber material (fig. 3.11, left)

2) To minimize reflections from the AH's shoulder (hard-reflecting material), the shoulder was covered with a commercial sweater. (fig. 3.11, left)

[9] It was not possible to use the IEC 711 Coupler and the AH - human ear simulation (physically) simultaneously because the dimensions of the acoustic coupler did not match the human ear adapter. Therefore, the measurements were made without the ear adapter. This takes us to the next item.

### 3.3.5 The enhancement of the measured frequency domain

As can be seen from fig. 3.12 and the following figures, measurements do not start before 40 Hz. The reason for this is the limited frequency range between 40 Hz and 10 kHz during all measurements in the past. Based on these results, we started working drawn against the background of the same conditions, but later decided to increase the frequency range within the course of development. Thus, all further measurements from acoustic coupler prototype 2 onwards were made at the frequency range between 20 Hz and 10 kHz.

## 3.4 Prototype acoustic coupler

### 3.4.1 Development

The basic idea for the development of the adaptive acoustic coupler is about simplification of the IEC 711 standard coupler. The IEC 711 occluded-ear simulator simulates the acoustic transfer impedance for the occluded normal adult human ear. Within the specified frequency range of 100 Hz to 10 kHz, it fully works as ear simulator [IEC10]. Above and below (20 Hz – 20 kHz), it can be used as acoustic coupler.

In the case of ANC measurements, it is not necessary to have an acoustic coupler with the right transfer function of the ear. It is more important to pay attention to other features, like the resonance frequency and size of the cavity and the possibility of making it easy to reproduce. Mentioned below at 3.4.1.2 you will find a list of the most important requested properties.

#### 3.4.1.1 Technical requirements

Technical requirements were based on the proposals in [IEC10] and [IEC06]. Fig. 3.7 shows a design drawing of the acoustic coupler prototype.

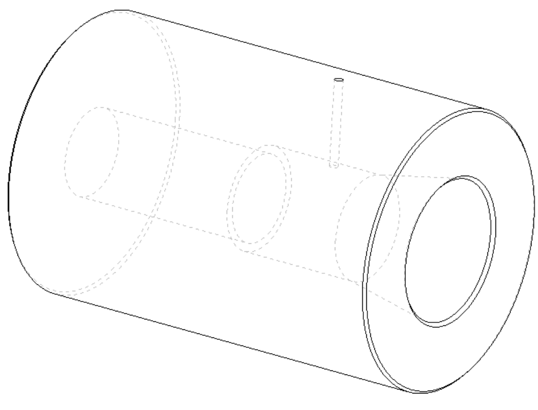


Fig. 3.7 Design drawing, acoustic coupler

#### 3.4.1.2 Material

In order to meet the requirements of the IEC and achieve best possible performance, brass was selected as basic material for the coupler. The important properties in relation to this use are:

- hard / dimensionally stable
    - guarantees minimal response of vibrations
    - minimizes solid-borne sound
    - damping extraneous sound outside the cavity
- non-magnetic
    - avoids magnetic coupling between microphone and speaker
- non-porous
    - reliability of a long endurance

### 3.4.2 Construction

#### 3.4.2.1 The cavity

The most important property is the size of the cavity. The recommended diameter is 7.5 ± 0.04 mm. The length of the principal cavity shall be such as to produce a half - wavelength resonance of the sound pressure above the main measurement area. In case of this acoustic coupler, this means above 10 kHz. As the following measurements show, 2 different lengths of the cavity were tested.

#### 3.4.2.2 The pressure equalization

The combination of the IEMs ear tip (with its rubber-like surface) and the coupler produces a hermetically sealed-cavity. This leads to pressure alteration by inserting the IEM into the coupler and will result in a negative effect on measurements. This is why a vent has to be installed to equalize static pressure. The time constant for the pressure equalization is not supposed to be longer than 1.5 seconds.

To minimize effects on pressure conditions in the cavity and secure sound damping from outside to comply with the standard (damping from outside 16dB @ 100Hz, 6dB/Oct with increasing frequency), the volume of the vent should be as small as possible [IEC06].The pressure equalization was implemented by means of a 0.6 mm drilled hole filled with a 0.5 mm conductor.

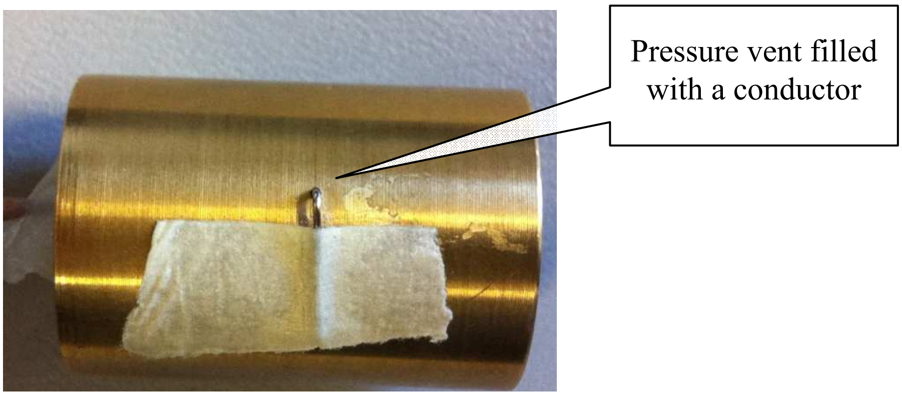



Fig. 3.8 Acoustic coupler with pressure vent

#### 3.4.2.3 The IEM pick-up

The dimensions and the angle of the cone were duly selected. An angle of 11° guarantees the purchase of the specific IEM and ensures its constant position during measurement.

#### 3.4.2.4 The measurement microphone pick-up

It is important to have the measurement microphone mounted in a hermetically-sealed fashion in the coupler. No sound or vibrations should reach the microphone from the backside. To prevent such effects, the pickup hole for the microphone has about 12 mm in length (the microphone has 6 mm). The remaining cavity on the backside of the microphone is filled with hot glue with an age-hardening effect. Fig. 3.9 shows an acoustic coupler prototype with hermetically-sealed backside.

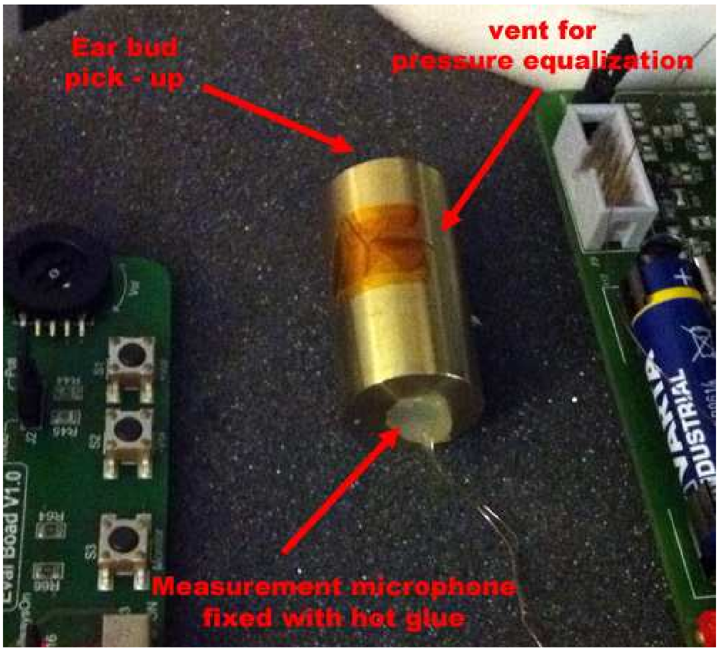



Fig. 3.9 Acoustic coupler, prototype 1 with sealed backside

## 3.4.3 Acoustic coupler prototype 1

Fig. 3.10 illustrates the principle of the acoustic coupler. Prototype 1 was developed with following dimensions:

Effective cavity length: 20.5 mm
Effective cavity volume: 905.6 mm²
Cavity pick-up angle: 11°

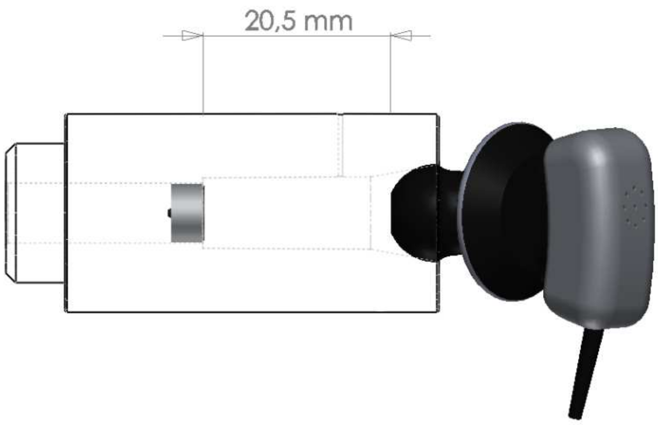



Fig. 3.10 Design drawing, acoustic coupler prototype 1

Note:

At this stage of the thesis analysis, the custom-made pre-amplifier for the measurement microphone was still in development. Meanwhile, the evaluation board AS 3501 was used as pre-amplifier. The performance of the evaluation board will suffice for the measurement process.

Fig. 3.11 Comparing measurements with prototype 1 and AH

In a first step, the properties of the pure performance of the acoustic coupler prototype were measured (fig. 3.11, right). To minimize influence from ambience, measurements were performed outside the measurement box. In order to achieve best possible performance, the measurements were made in a semi-anechoic room under conditions mentioned at 3.3.1.

The main target of the following measurements in chapter 3.4 was to approximate the results achieved with the artificial head in the measurement chamber over the past few years as closely as possible. However, as we will learn later on, there was some influence of the AH on the measurements having corrupted some of the results.

#### 3.4.3.1 Measurements coupler type 1 vs. AH

There are 2 big differences in this first comparison of measurements between acoustic coupler prototype 1 and the AH.

1) There is a frequency minimum at 2 kHz on the AH measurement, probably depending on reflections from the AH's shoulder and/or Head (fig. 3.12, fig. 3.13). This frequency minimum also appears at the filter calculation (fig. 3.15) because of the dependence of the single measurements.

2) The measurement of the acoustic coupler in fig. 3.14 shows a compression of the frequency response at the range of 400 - 1500 Hz. The reason is an included AGC (automatic gain control) - feature coupled with the pre-amplifier circuit of the evaluation board AS 3501. This feature has been removed during all further measurements.

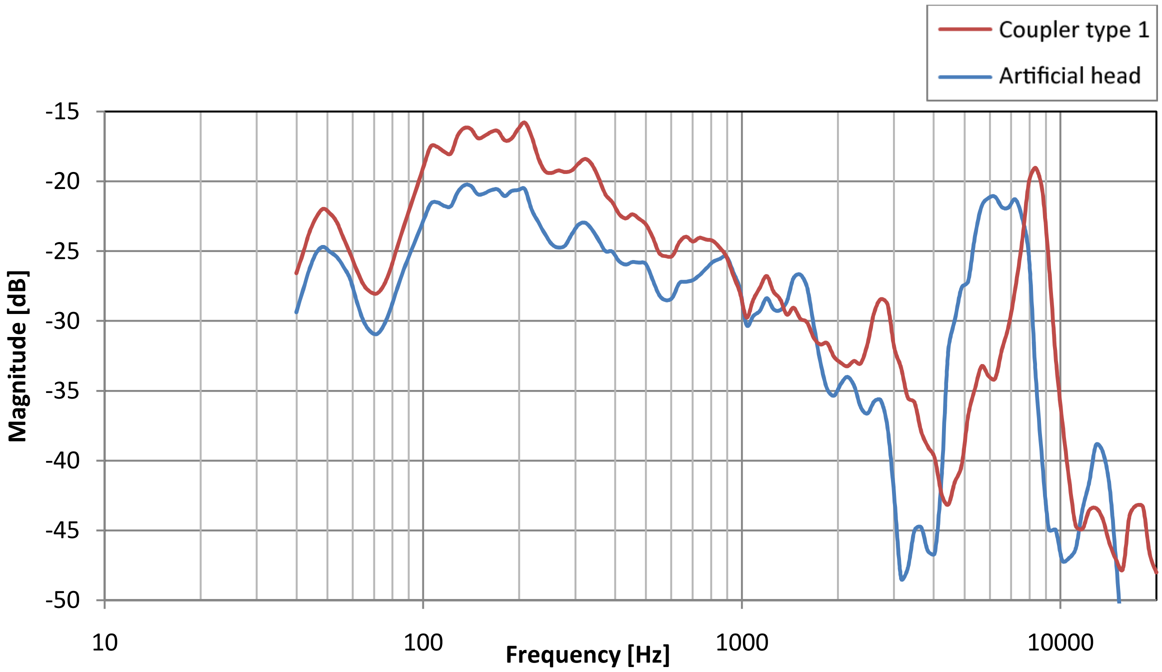



Fig. 3.12 Passive attenuation measurement

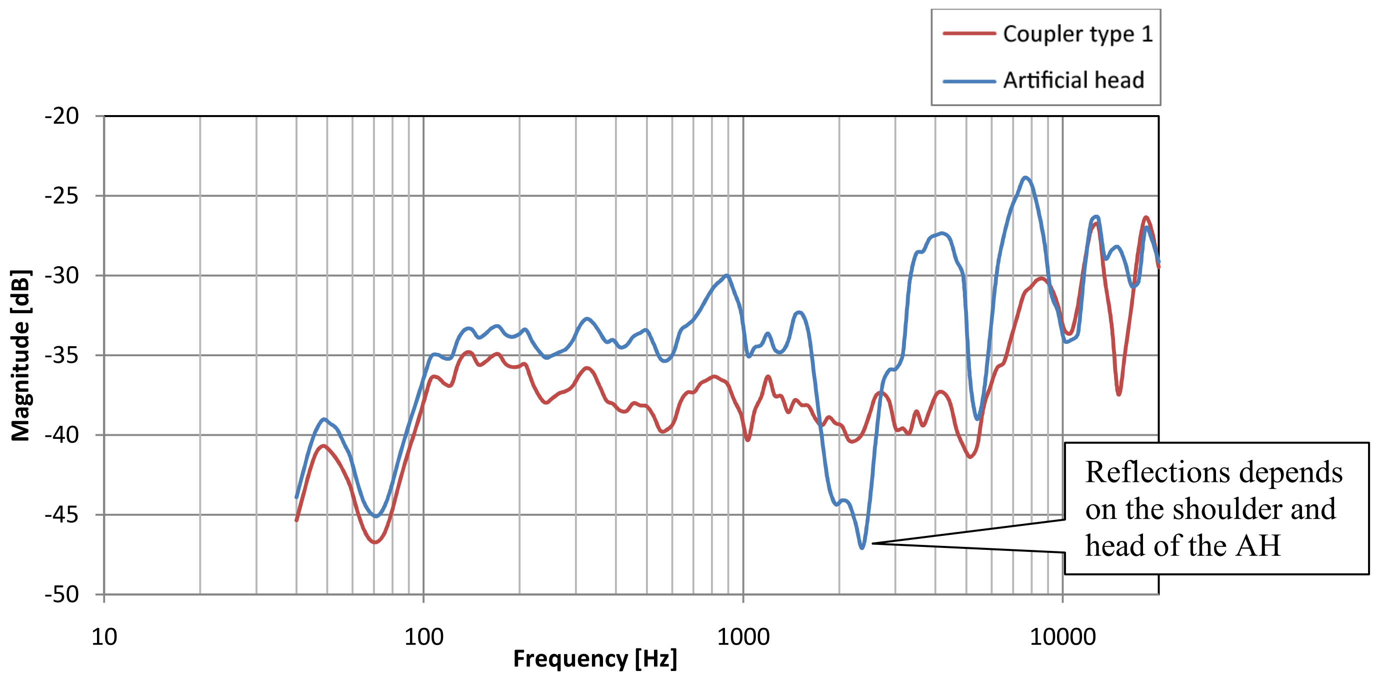



Fig. 3.13 Noise to ANC microphone measurement

Note that the reference is mounted in the AH. The acoustic coupler prototype was simply mounted on a microphone stand (Fig. 3.11).

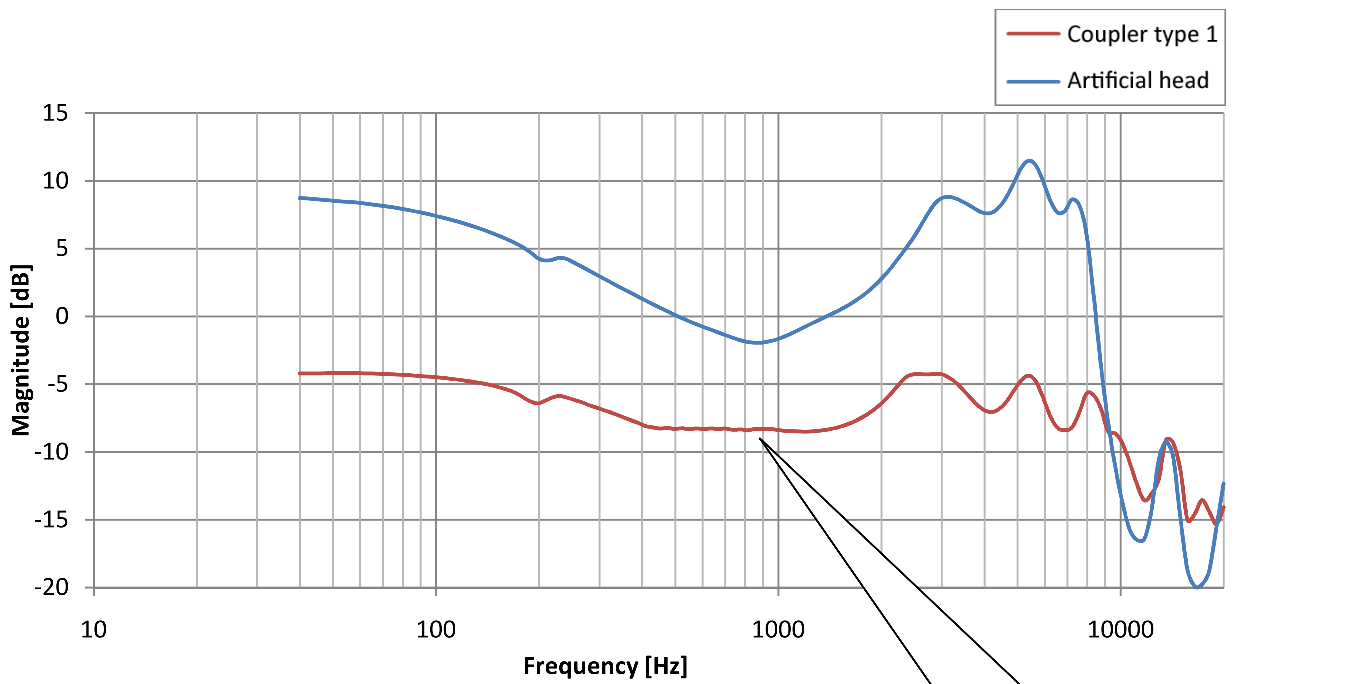



Fig. 3.14 IEM to measurement microphone measurement

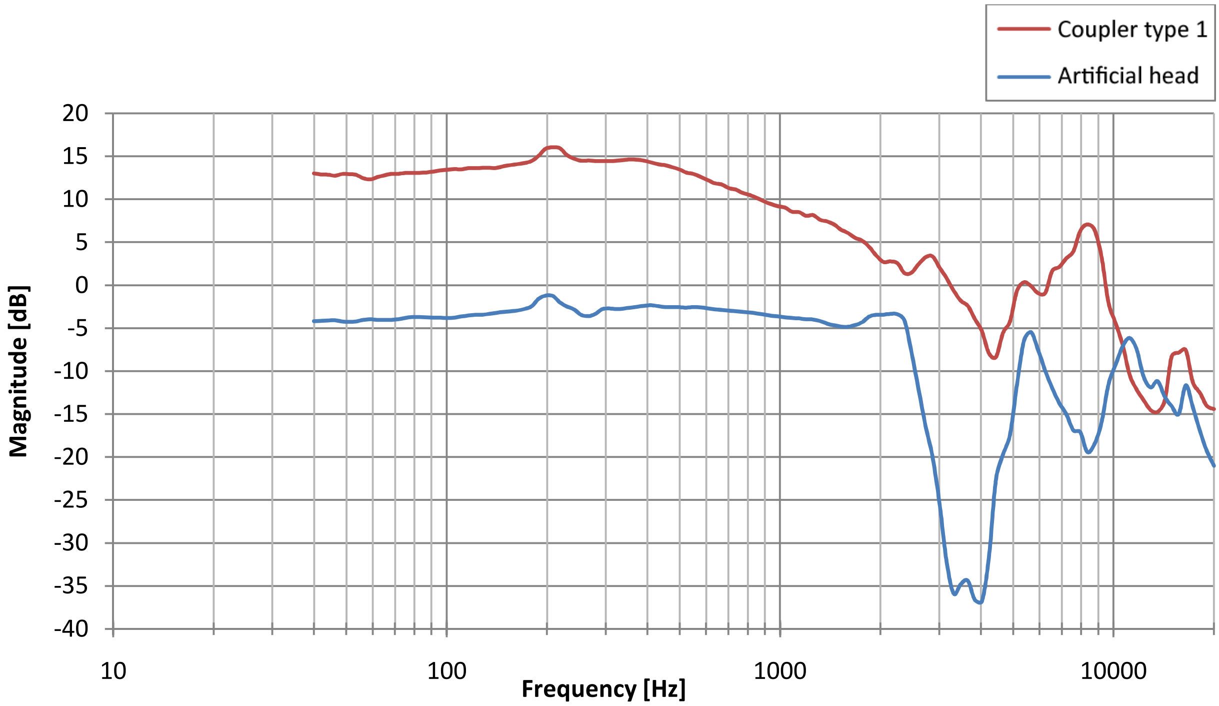



Fig. 3.15 Filter calculation

Furthermore, the cavity length at 20.5 mm is not ideal. As mentioned in 3.4.2.1, the length of the cavity should be such as to produce a half-wavelength resonance of the sound pressure above the main measurement area. With the current cavity length, the resonance frequency @ half-wavelength amounts to 8366 Hz. To achieve the resonance frequency above 10 kHz, the cavity length should be at most at 17.15 mm.

Additionally, it just turned out that the angle of the cavity pick-up (where to put in the IEM) at 11° was not ideal. Sometimes the IEM was lacking in grip and dropped out. I tried to fix the problems mentioned with prototype 2.

### 3.4.4 Acoustic coupler prototype 2

The following properties where modified in relation to prototype 1 to achieve better performance of the acoustic coupler. Fig. 3.16 illustrates the functionality of acoustic coupler type 2. Prototype 2 was developed with the following dimensions:

| | |
|---|---|
| Effective cavity length: | 12 mm |
| Complete cavity length: | 16 mm |
| Cavity pick-up angle: | 9.5° |

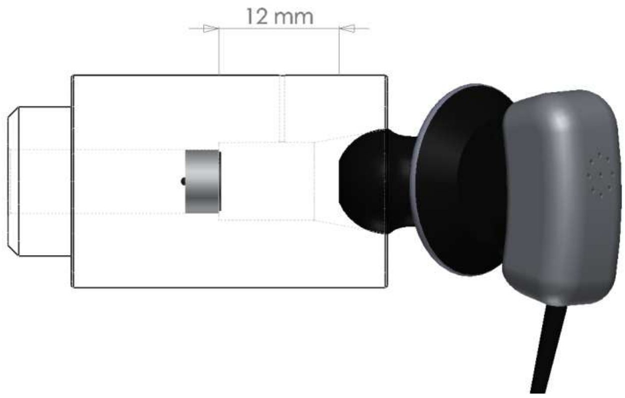



Fig. 3.16 Design drawing, acoustic coupler prototype 2

Fig. 3.17 shows some photographs of the acoustic coupler prototype 2 from different sides. On the left side, the built in measurement microphone can be seen.

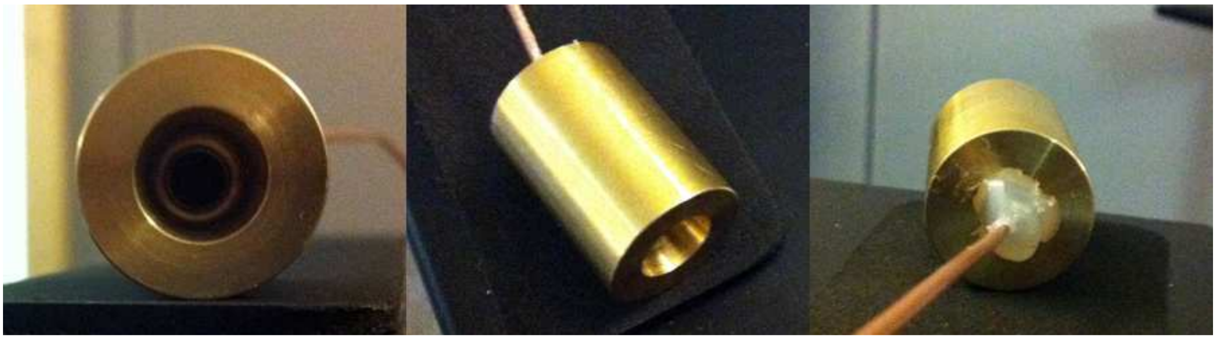


Fig. 3.17 Acoustic coupler prototype 2

### 3.4.4.1 Measurements coupler prototype 1 vs. prototype 2

Fig. 3.18 shows the results of the improved acoustic coupler type 2. The implementation of the suggested modifications results in the following advantages:

- The resonance frequency of the cavity is now located above 10 kHz; the length of 12 mm results in a $f_r$ = 14291 Hz
- The smaller size of the cavity makes for easier handling
- A cavity pick-up angle of 9.5° provides safe mounting of the IEM

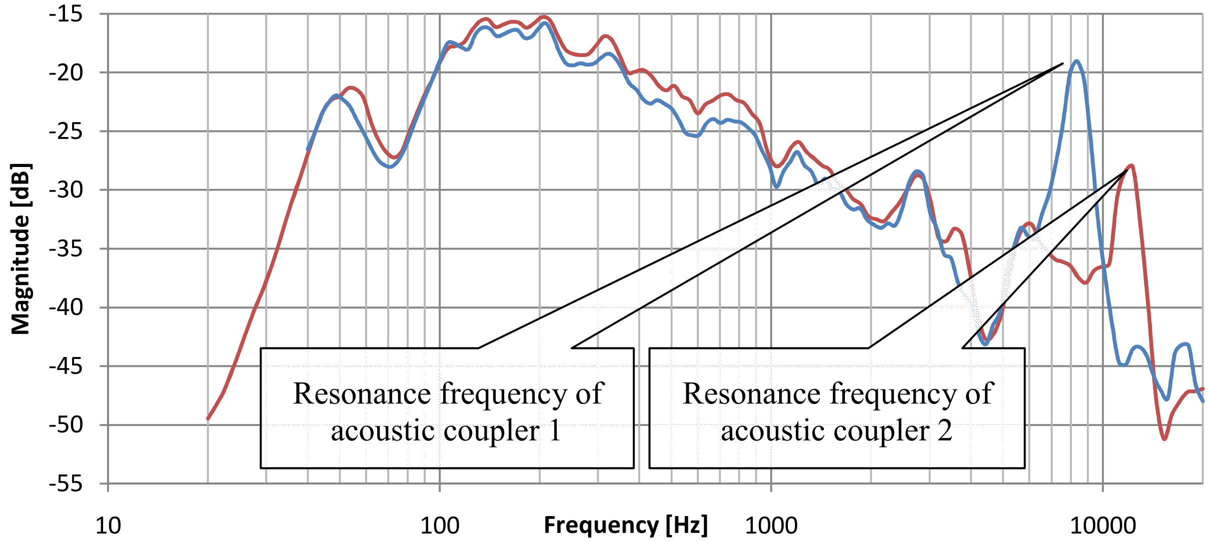



Fig. 3.18 Passive attenuation comparison prototype 1 and 2

## 3.4.5 Measurements coupler type 2 vs. IEC 711 standard coupler

In order to verify development of the acoustic coupler and slowly change to fit a realistic scenario, comparing the coupler prototype and the IEC 711 standard coupler (fig. 3.19) became necessary. The following measurements will have us evaluate the performance of the acoustic coupler compared to the "stand-alone" version of the IEC 711 standard coupler. This brings with it the advantage of comparing the performances without influences by the AH.

The following measurements were made inside the measurement box prototype described below in chapter 3.5. The setup of the box is equivalent to the final setup as described below in chapter 3.6.1. The DUT was a Sony MDR NC 22 IEM (fig. 3.19).

Fig. 3.19 Stand alone IEC 711 standard coupler with IEM

The comparison between the acoustic coupler prototype 2 and the IEC 711 coupler shows a magnitude gap in the frequency range from 4 kHz up to 10 kHz (fig. 3.20, fig. 3.22). The reason for this can be found in the built-in resonators in the IEC 711 coupler. These so-called "annular groves" cause an increase of magnitude at the specific frequencies analog to the human ear canal. "*The IEC 711 based occluded-ear simulator simulates the acoustic transfer impedance for the occluded normal adult human ear. However, it does not simulate the leakage between an ear mould and a human ear canal.*" [IEC10]

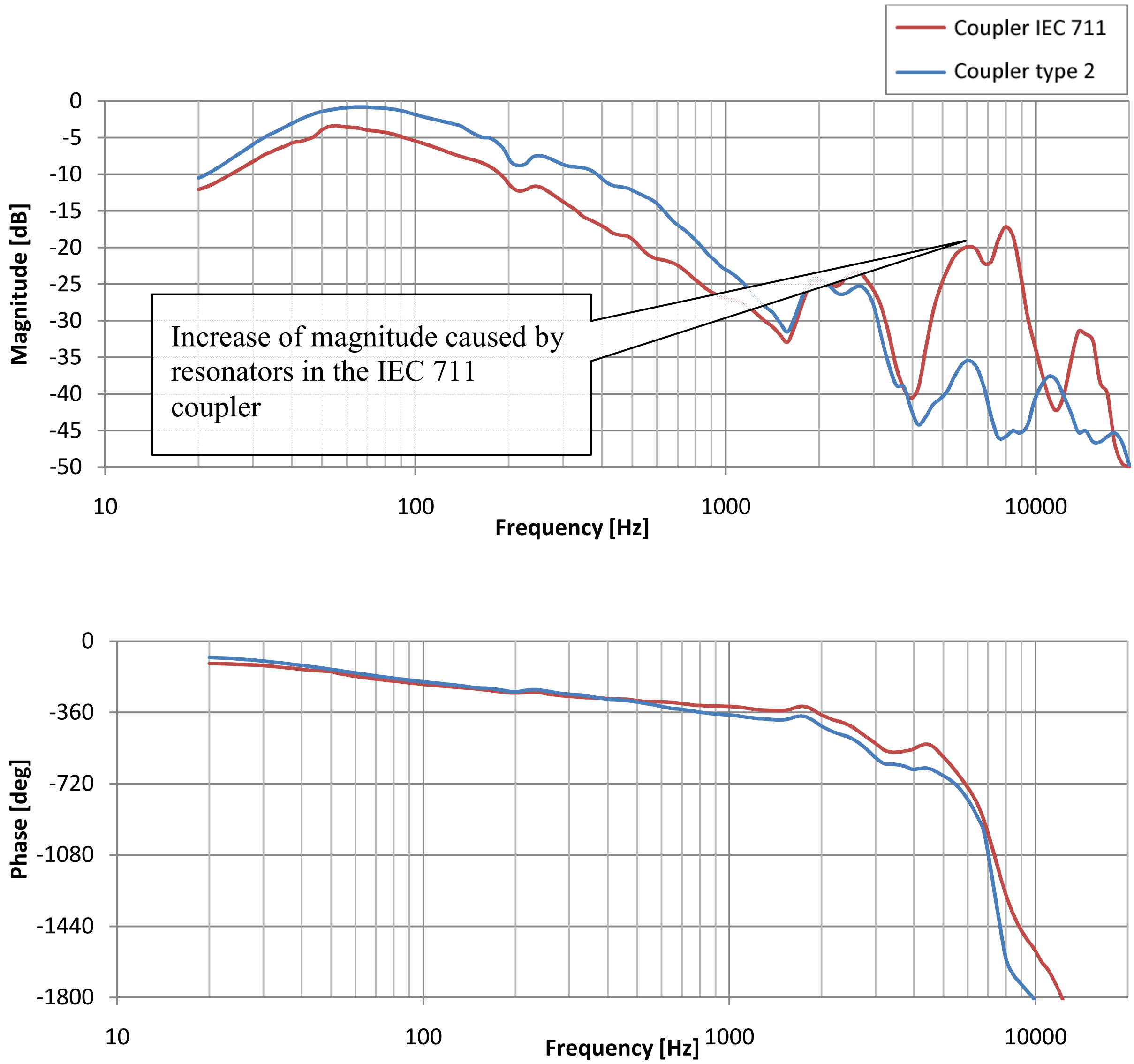
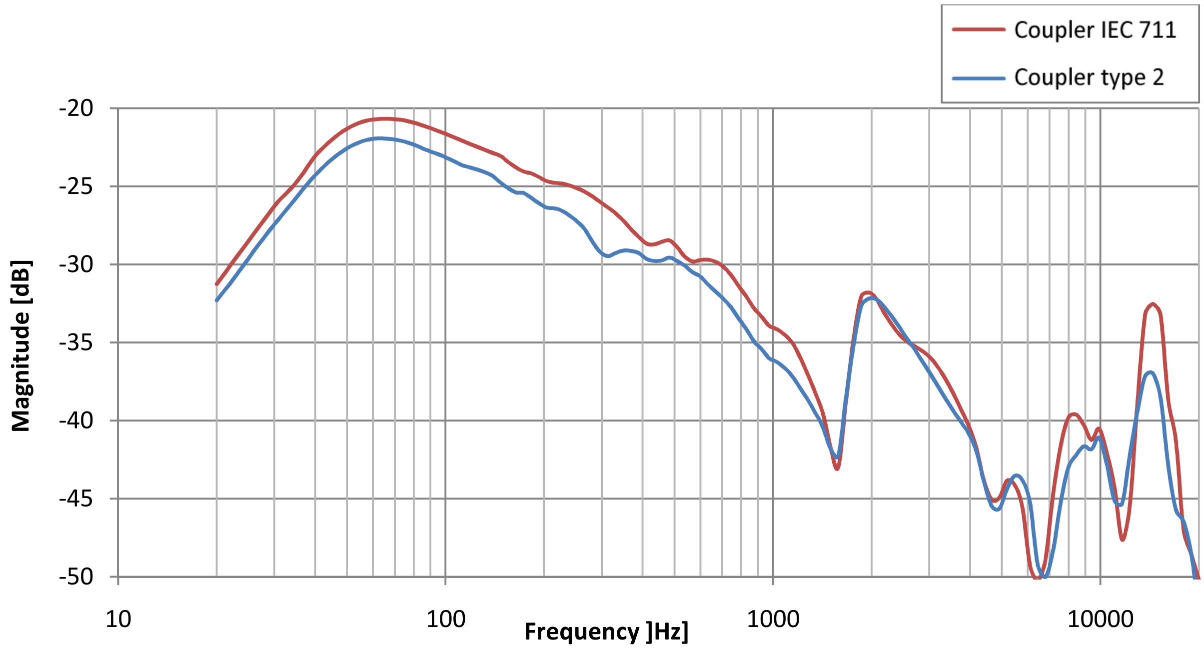



Fig. 3.20 Passive attenuation measurement



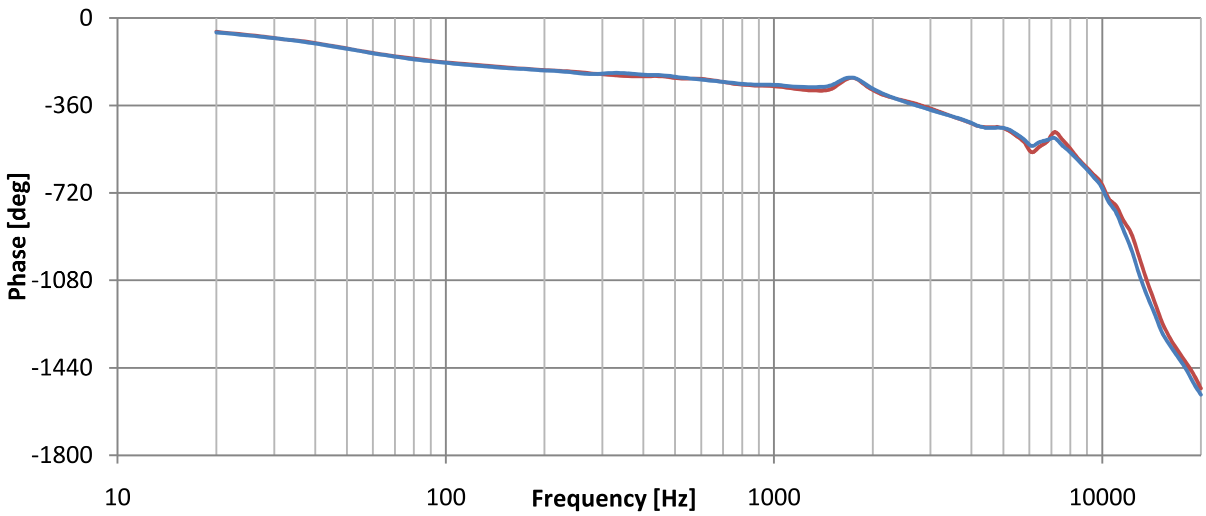




Fig. 3.21 Noise to ANC microphone measurement

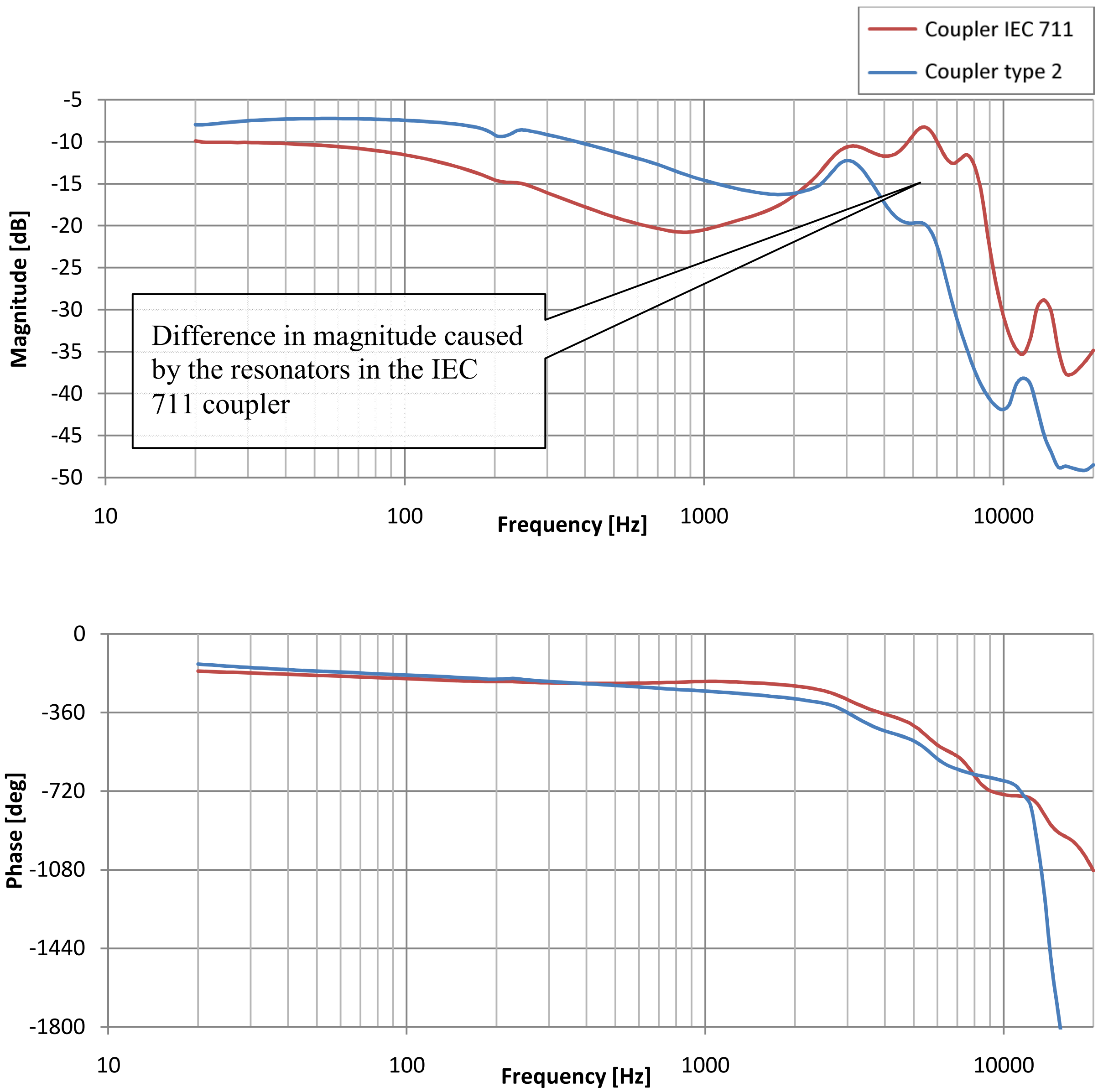



Fig. 3.22 IEM to measurement microphone

Fortunately, these circumstances are not considered a problem because of the calculation rules leading to the final filter curve (Fig. 3.23).

$$\text{Filter curve} = \text{Passive att.} - (\text{Noise to ANC} + \text{IEM to measurement mic.}) \qquad [3.1]$$

This algorithm compensates for most of the differences in frequency response by way of simple subtraction. As shown in fig. 3.23, the averaged filter curves (Magnitude and Phase) of the acoustic coupler prototype 2 and the IEC 711 coupler are even similar in the important frequency range between 20 Hz and 10 kHz.

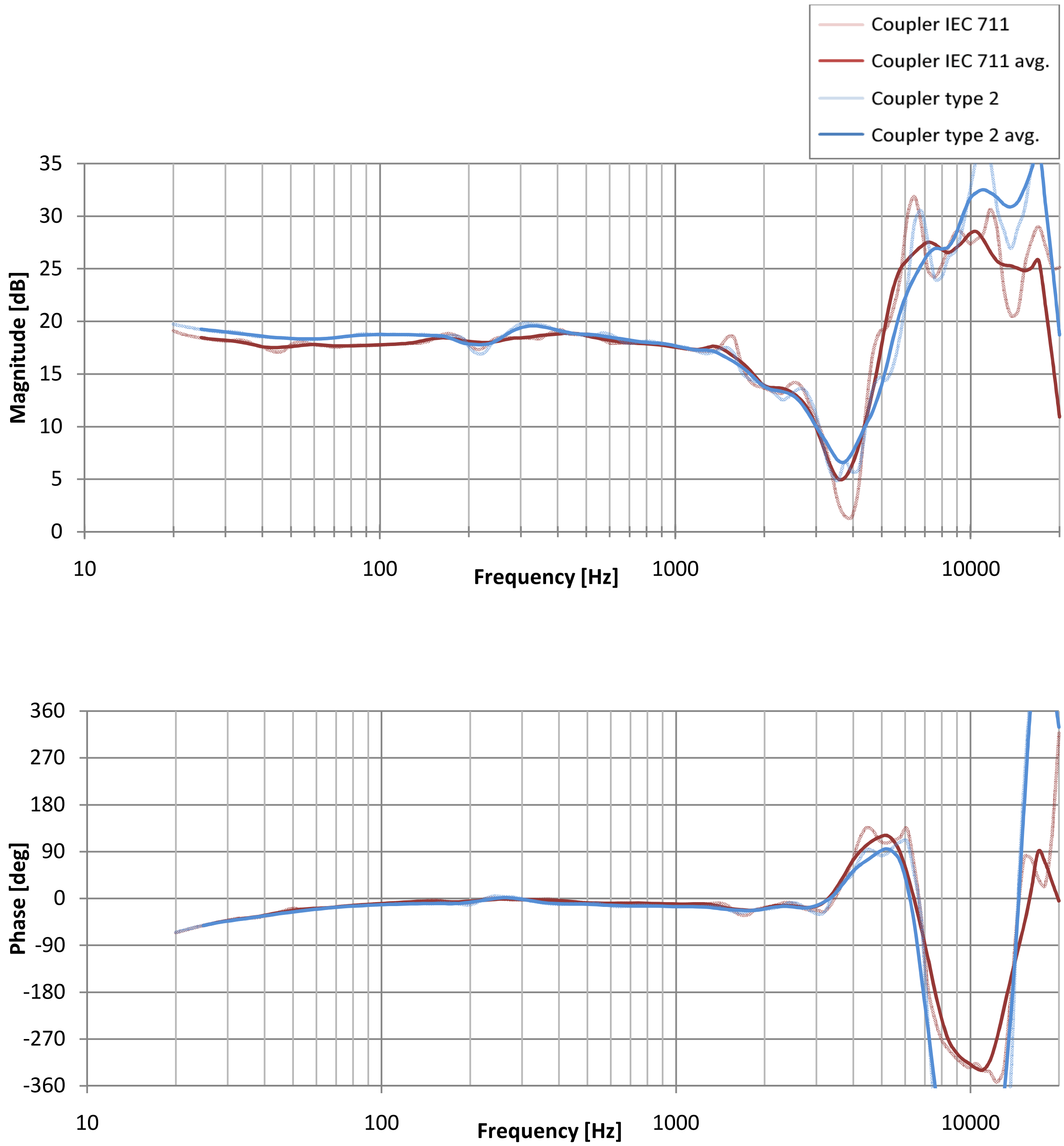



Fig. 3.23 Filter calculation

# 3.5 Prototype measurement box

## 3.5.1 Development

The basic idea for the development of the measurement box was about resizing the previous measurement ambience, so far a measurement room, to manageable and portable size. The performance data for the measurement box should, if possible, be nearly similar to data for the measurement chamber. The box should also have acceptable noise attenuation to the outside and vice-versa to the inside, so it could be used in an office environment, without having any problems with disturbing noise. Additionally, all these parameters had to be implemented bearing in mind the aspect of minimizing cost, but without suffering any setback in accuracy of measurement or quality. Fig. 3.24 shows the first prototype of the measurement box as mentioned above.

Fig. 3.24 Measurement box prototype

These are the big points taken into account during the design phase:

- Manageable size/weight
    - The selected material
    - Keep basic dimensions as small as possible
    - Design of the first prototype: flexible in size

- Include the following components
  - Speaker
  - Coupler
  - Preamp (Microphone)
  - Preamp (Speaker)
  - Necessary connectors
  - USB Soundcard
- Minimize cost
  - The selected material
  - Well-considered design
- Make it easy to reproduce
  - The selected material
  - Well-considered design
  - Documentation of construction plans

### 3.5.2 Design

The measurement box mainly consists of two chambers (fig. 3.25). The smaller chamber (#1) on the backside of the speaker serves as speaker cabinet. The bigger chamber (#2) represents the measurement chamber. The decision to use MDF (medium density fiberboard) as basic material for the body was based on the reason of low cost and that it is easy to handle. The fact that it is easy to handle is particularly important for the prototype development. This provides the option of making changes during the development process. Moreover, MDF is a well-used material in speaker cabinet design and has been used over the last decades producing excellent performance values.

As shown in fig. 3.25, the front and back ends of the box were designed in a flexible way. This offers two more options during the development/measurement process, if needed. If necessary, some of the speaker resonance can be controlled to especially have an impact on the resonance frequency of the measurement chamber itself and the speaker cabinet.

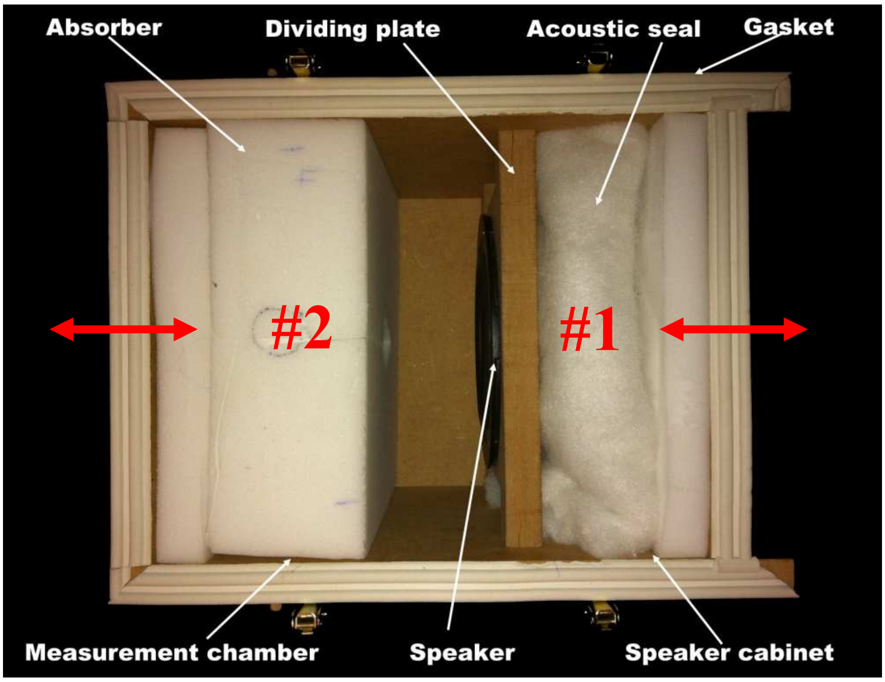



Fig. 3.25 Measurement box inside

In addition to above-mentioned options, the dimensions of the box depend on the following parameters:

- Speaker size (diameter/depth)
- Acoustic coupler size (+ IEM size)
- Distance coupler - speaker
- Thickness of absorber material (depends on frequency to attenuate)
- Preamplifier microphone (very small) / preamplifier speaker (mounted on the outside of the box)

### 3.5.2.1 Acoustic feedback

As can be seen from fig. 3.31, the dividing plate between the two chambers was prepared to cover four different possibilities of acoustic feedback related to the speaker. This was supposed to create more space for producing an impact on frequency response and handle resonance effects that may occur in relation to the box.

### 3.5.2.2 The absorber material

In order to reduce resonance frequencies in the measurement box, absorber material has to be used. Due to the small dimensions of the box, the inserted absorber does not impact on the entire frequency range. Especially in the lower frequency range, effectiveness decreases. The sound absorption value of the used absorber material (BASF Basotect®) can be seen in fig. 3.26.

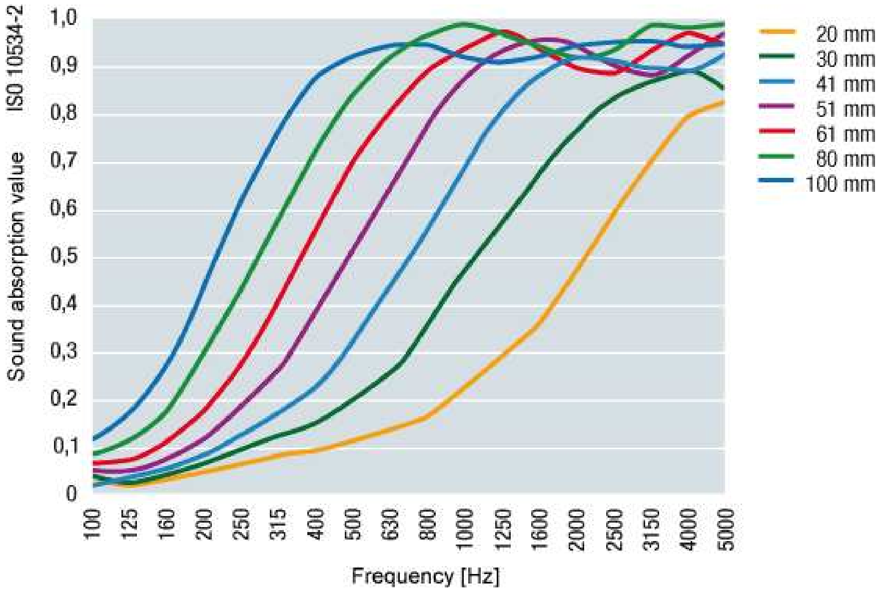



Fig. 3.26 Sound absorption value, (www.basotect.com, Jun. 2011)

## 3.5.3 Measurement setup

Speaker types: 4” Visaton FR 10, full range speaker with tweeter dome, 30 Watts
5” Ciare hx135, full range speaker, 40 Watts

Measurement Software/Hardware: Audio Precision
General measurement microphone: G.R.A.S. 40 AF
Measurement signal: Exponential stepped sine sweep

## 3.5.4 Frequency and phase response of the empty box

The first measurement relating the measurement box was the frequency response of the empty box. This has to reasons, first, of course, to get an idea of what the frequency response looks like and where resonances appear. The second reason is to see the impact of the absorber material in relation to the empty box.

The main Specifications for the empty measurement box are:

- Measurement box empty, no absorber material
- Speaker cabinet empty, no absorber material
- No acoustic feedback between measurement chamber and speaker cabinet
- Chamber size: (small) 110 x 215 x 215 *(l x b x h)*
- Cap closed

Fig. 3.27 Empty measurement box

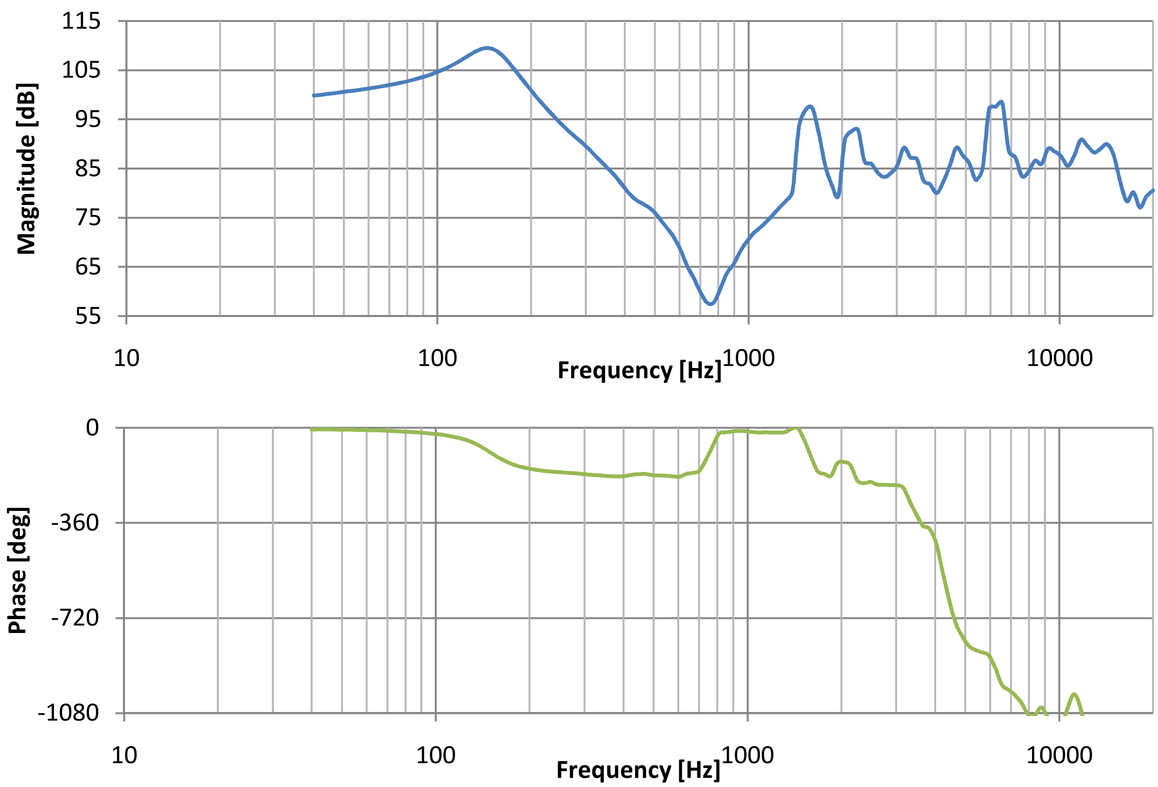



Fig. 3.28 Frequency and phase response of empty measurement box

### 3.5.5 Variation in chamber size

During the development process, the dimensions of the measurement box had to be achieved. To define these dimensions under the aspect of a good handling but also a good performance, the prototype box was designed flexible in size. Two variations of the chamber size where selected and the frequency response of the box was measured. The measurement was performed under the following conditions:

- Cap closed
- Absorber material thickness: 45 mm
- Speaker cabinet filled with loose sound sponge
- 2 different sizes of the measurement chamber:

| | **Small** | **Big** |
|---|---|---|
| **Measurement chamber** | 110 x 215 x 215mm | 170 x 215 x 215mm |
| **Speaker cabinet** | 80 x 215 x 215mm | 100 x 215 x 215mm |

Fig. 3.29 Small chamber size with absorber

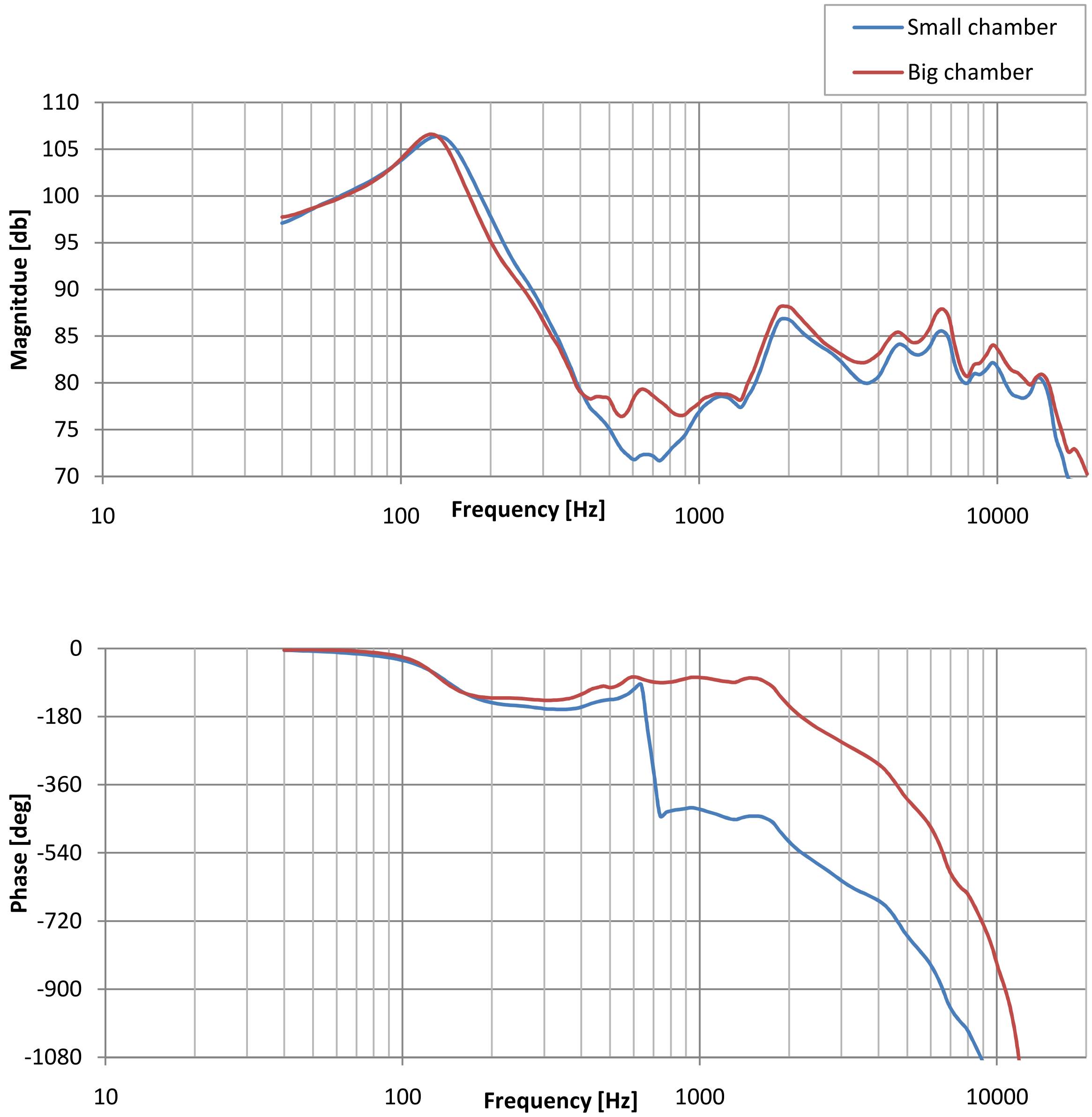



Fig. 3.30 Comparison of small and big chamber

The variation in speaker cabinet size has no influence on the frequency response of the speaker or on any resonances produced by the speaker. The reason for this is that only minimal volume changes are possible and, furthermore, the combination with an acoustic feedback effect (as described below in 3.5.2.1) makes the influence drop towards zero. Therefore, the decision was made in favour of the small speaker cabinet size.

The only apparent difference between the big and the small measurement chamber size can be found at the more flat frequency response between 400 Hz and 1000 Hz using the bigger chamber size (fig. 3.30). This was the reason for using this size in further development.

### 3.5.6 Variation in acoustic speaker feedback

The reason for supporting 4 different speaker front plates (fig. 3.31) during development was about creating more impact on the frequency response of the speaker, especially in the lower frequency range. This effect - which enables the property of having impact - is the acoustic speaker feedback. The speaker feedback describes the possibility of the sound waves to enter the speaker plate from the backside of the speaker to the front side and vice-versa. This results in a more or less intense (depending on the permeability of the speaker plate) acoustic short-circuit. The following measurements will prove the influence of theses plates. Tab. 3.2 shows the properties of the 4 different speaker front plates. Fig. 3.32 shows that the effect mainly impacts at lower frequencies.

| Type | Acoustic feedback | Plate area [%] | Area [mm²] |
|---|---|---|---|
| **Plate 0** | Non | 100 | 46225 |
| **Plate 1** | Minimum | 98.7 | 45625 |
| **Plate 2** | Moderate | 86.0 | 39775 |
| **Plate 3** | Maximum | 65.4 | 30225 |

Tab. 3.2 Properties of the various acoustic feedback plates

Fig. 3.31 Four different acoustic feedback plates

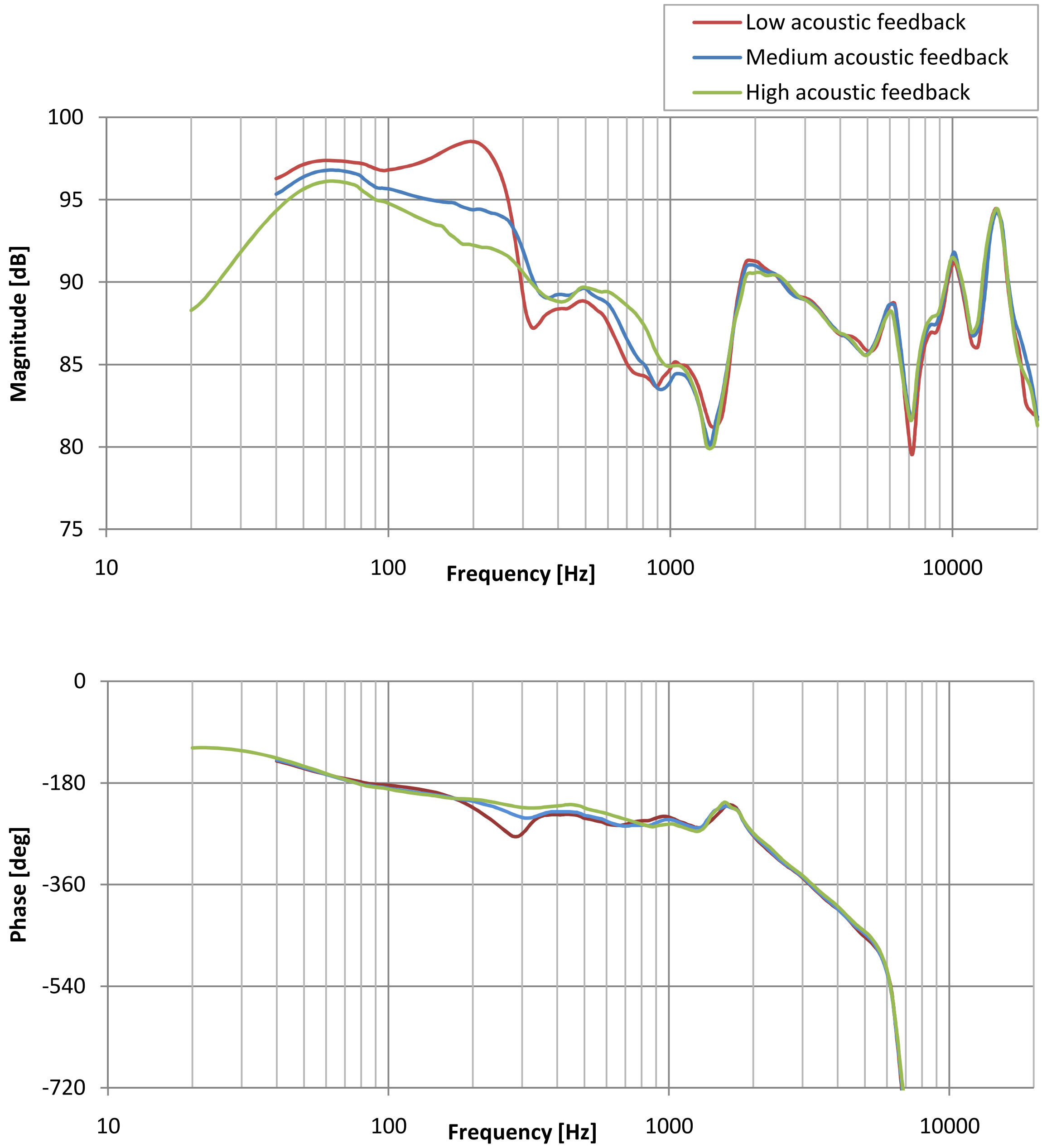



Fig. 3.32 Frequency and phase response of different acoustic feedback plates

As shown in fig. 3.32, the plate with high acoustic feedback produces a flatter frequency response between 100 Hz and 300 Hz. Due to this, the phase response is more flat, too, and does not show any significant warping. The intention of building this box in such a way as to achieve an as-flat-as-possible frequency response calls for selecting plate 3, with the property of highest acoustic feedback. Based on this decision, all further measurements where performed using plate 3.

## 3.5.7 Variation in microphone position

During measurement of different microphone positions, the shape of the absorber material inside the box was developed further into the final shape needed for combining it with the acoustic coupler. The goal was to achieve highest possible absorption in due consideration of required distance, later on including the acoustic coupler and the IEM. Furthermore, the amplitude of the diaphragm of the speaker had to be considered. These requirements resulted in a cone-like shape. Fig. 3.33 shows a prototype version of this absorber part.

Fig. 3.33 Shape of the absorber material closely to the speaker

### 3.5.7.1 Variation in axis position and distance

Fig. 3.34 shows 3 different measurements of various microphone positions. The goal was to find the best position, with a frequency response that is flat as possible. The measurements in fig. 3.35, follows the same principle, with the difference, that these measurements deals with the distance of the microphone.

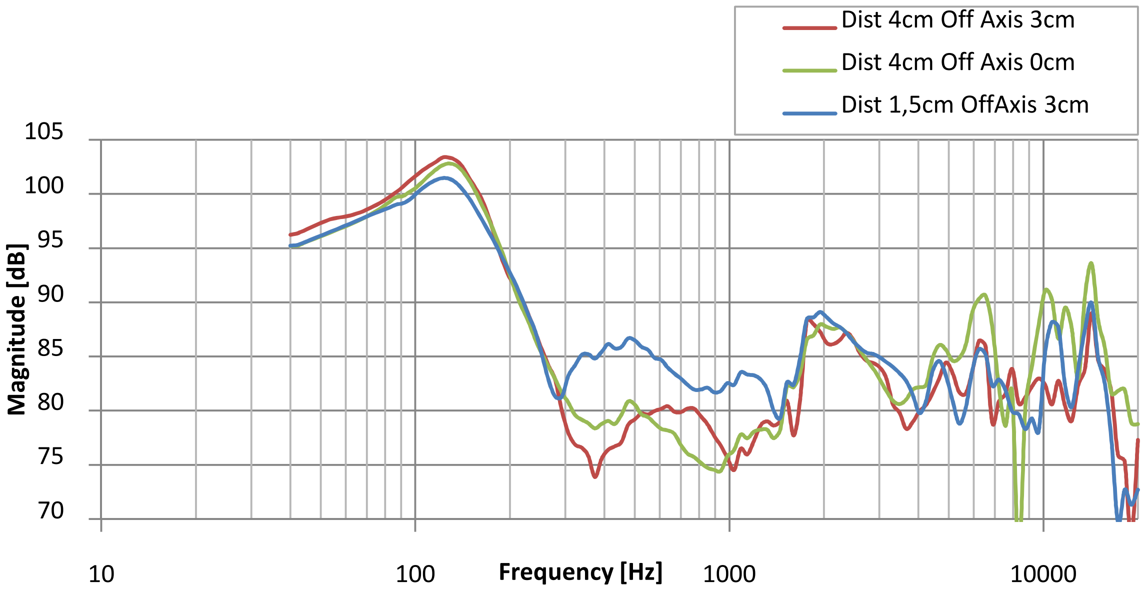



Fig. 3.34 Frequency response in different microphone positions

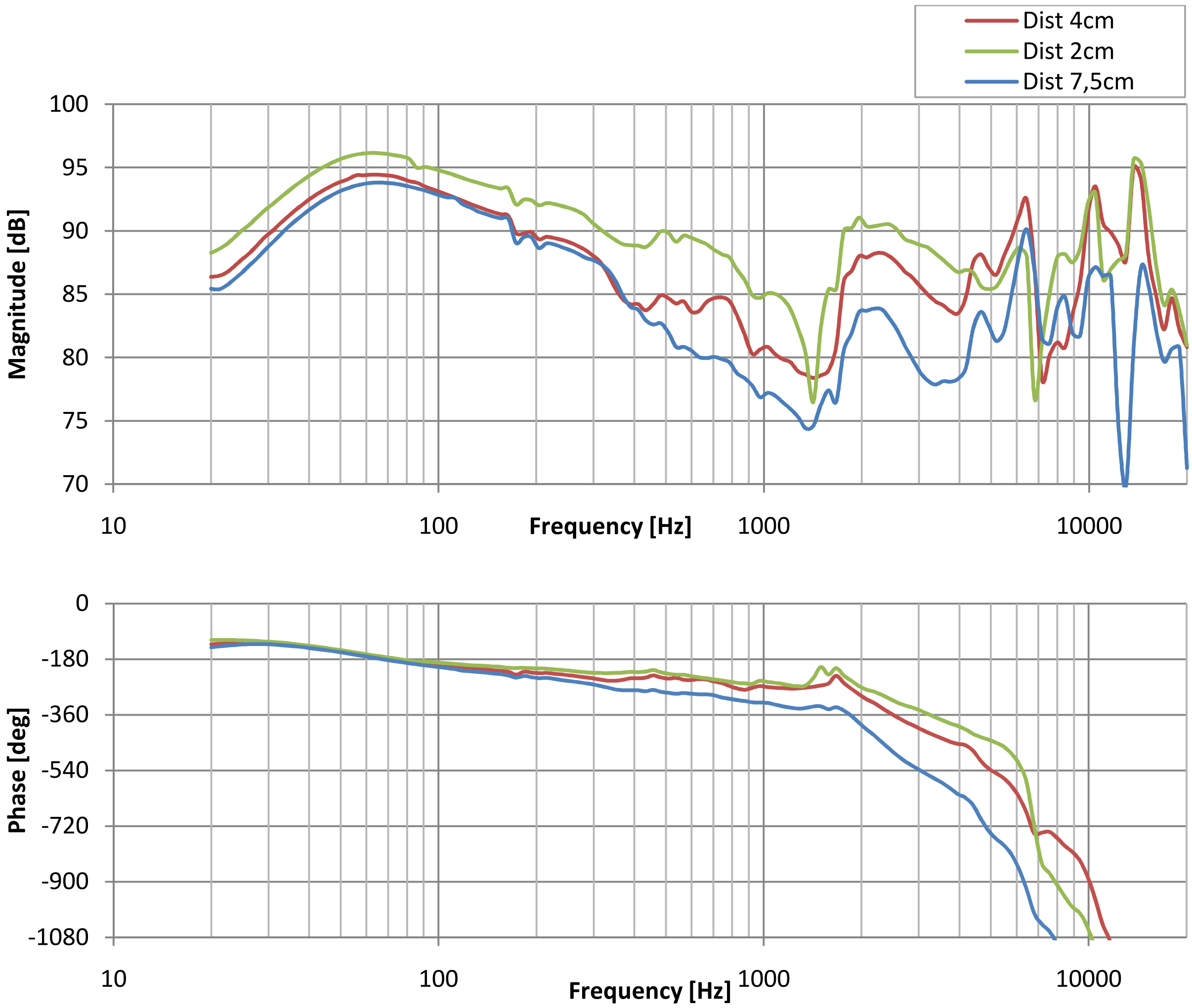



Fig. 3.35 Variation in microphone distance

The important goal here is also about finding a position for the microphone where frequency response is as flat as possible. However, there are a few other problems surfacing in this case.

Because of the low distance between microphone and speaker, the microphone is in the near-field of the membrane and alters the frequency response with partial oscillations. These partial oscillations are frequency-dependent and increase with increasing frequency. A rule of thumb says that partial oscillations start at the wavelength of the perimeter of the speaker diaphragm. In this case, we are dealing with a 5” speaker with a ⌀ 122 mm diaphragm. This leads to a start frequency of approximately 900 Hz. It also depends on the hardness of the diaphragm. If it is soft, the frequency is a little bit lower and vice-versa.

The second problem, also based on close distance between microphone and speaker, are interferences in the sound field, which leads to cancellation and peak effects in frequency response. This effect shows extensively in case of sideways movement of the microphone (fig. 3.34). These two circumstances lead to the decision of leaving the microphone on axis and choosing the microphone-to-speaker distance at 4 cm. This was the flattest possible solution in terms of measured frequency response.

### 3.5.8 Variation of the speaker

One big goal of this thesis is about cost-optimization of the measurement system. This aspect was considered in every development step of this project. Even though we are still dealing with a speaker yielding good results, we should not pass on determining a very low-priced speaker alternative, too. This speaker has a smaller diameter and 10 Watts less nominal power. Additionally, the speaker is equipped with a tweeter dome which probably produces other effects, especially in the near-field of the speaker. The following measurements compare the 5” and the 4” speaker. The microphone-to-speaker distance amounts to 40 mm.

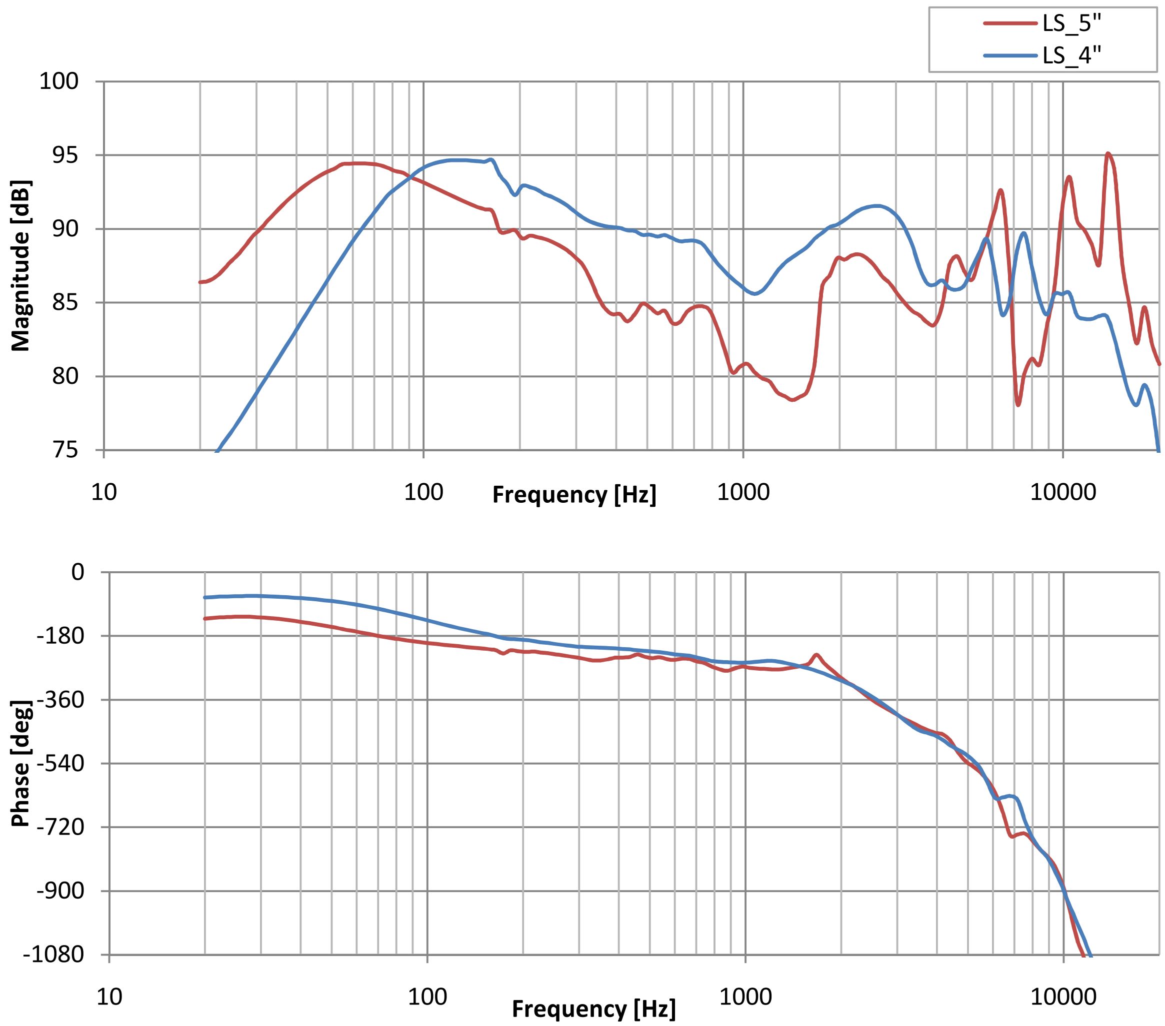



Fig. 3.36 Comparison of 5" and 4" speaker

Fig. 3.36 shows the prospective performance of the speaker at low frequencies. Depending on the smaller diameter of the 4” speaker, low frequency support is not satisfying. In fact, proper bass performance is important for the final measurement system to arrive at good ANC performance. Although frequency response in the higher range looks very promising, the 5” solution had been preferred because of better bass performance.

# 3.6 Combination of acoustic coupler and measurement box

After having completed all measurements and determinations of parameters of each single part of the measurement box, in this section all parts were assembled and the performance of the complete box was measured.

## 3.6.1 Final setup

The list below specifies all parts finally used:

**Speaker**

| | |
|---|---|
| Speaker type: | 5” Ciare hx135, full range |
| Cost: | Middle-priced segment |
| Impedance: | 8 Ohms |
| Resonance frequency: | 54 Hz |
| Nominal power: | 40 Watts |

**Acoustic coupler**

Acoustic coupler prototype 2

| | |
|---|---|
| Effective cavity length: | 12 mm |
| Complete cavity length: | 16 mm |
| Cavity pick-up angle: | 9.5° |

**Measurement box specifications**

| | |
|---|---|
| Measurement chamber: | 170 x 215 x 215 mm |
| Speaker cabinet: | 80 x 215 x 215 mm |
| Acoustic feedback: | Plate 3 (maximum feedback) |
| Absorber material: | Maximum absorption |

Cap closed for all measurements.

**DUT**

Sony MDR - NC22

Fig. 3.37 shows a front and the back view of the acoustic coupler including the mounted IEM. The acoustic coupler was embedded in the absorber material. A top view of the prototype measurement box can be seen in Fig. 3.38. Both chambers were filled with absorber material and a speaker seal was additionally placed at the speaker cabinet.

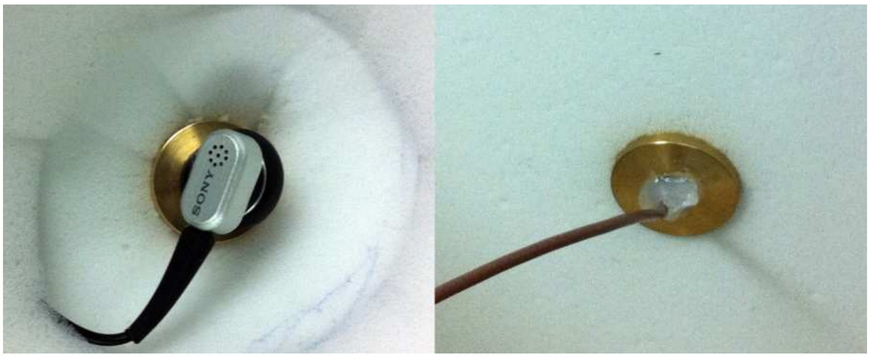



Fig. 3.37 Acoustic Coupler front and back view with IEM

Fig. 3.38 Measurement box top view

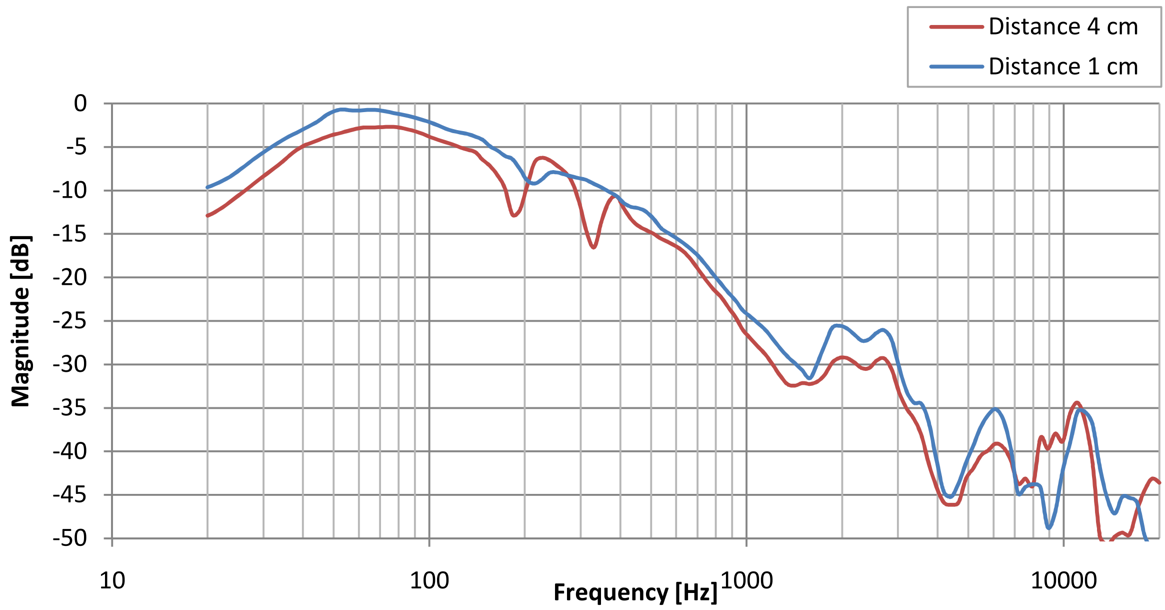
## 3.6.2 First measurements

The first ANC measurements with the acoustic coupler in the box show that the frequency response slightly drops at 180 Hz and 320 Hz (fig. 3.39). Such drops only occur in the passive attenuation measurement. This gives way to the assumption that this effect could only be produced by resonances of the box or the speaker itself. The λ/4 - wavelength of 320 Hz equates 270 mm, which is the exact length of the whole measurement box. The drop at 180 Hz may depend on the directional characteristics in the near-field of the speaker. The fact that such frequency drops only occur in the passive attenuation measurement (cf. fig. 3.39 vs. fig. 3.40) gives room to the assumption that the position of the measurement microphone corresponds with the minimum amplitude of the sound wave.

To get rid of this problem, some more measurements are required to specify a more favorable distance between coupler and speaker. The following measurement shows the comparison between 40 mm distance (until now) and 10 mm.

### 3.6.3 Variation in distance

As can be clearly seen from fig. 3.39, the frequency response of the measurement box has a much better performance. Mounting the microphone at 10 mm distance decreases the frequency drops at 180 Hz and 320 Hz. Based on this measurement, the final position for the coupler was chosen such as to create an IEM – speaker distance of 10 mm. Of course, the results of these measurements could not be generalized. The relatively flat frequency response at 1 cm is a result of especially combining the parts used in this available configuration.



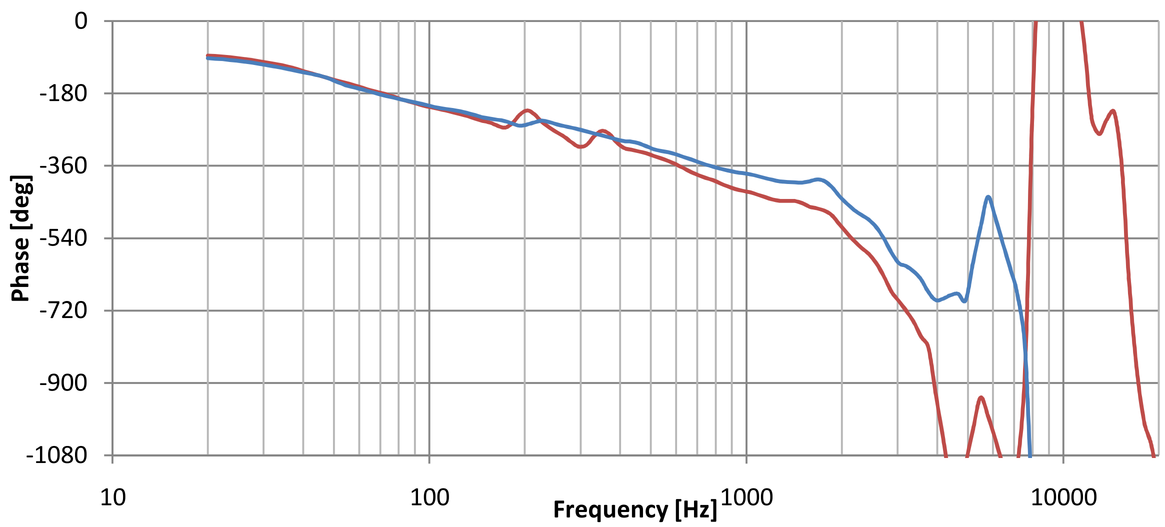




Fig. 3.39 Passive attenuation

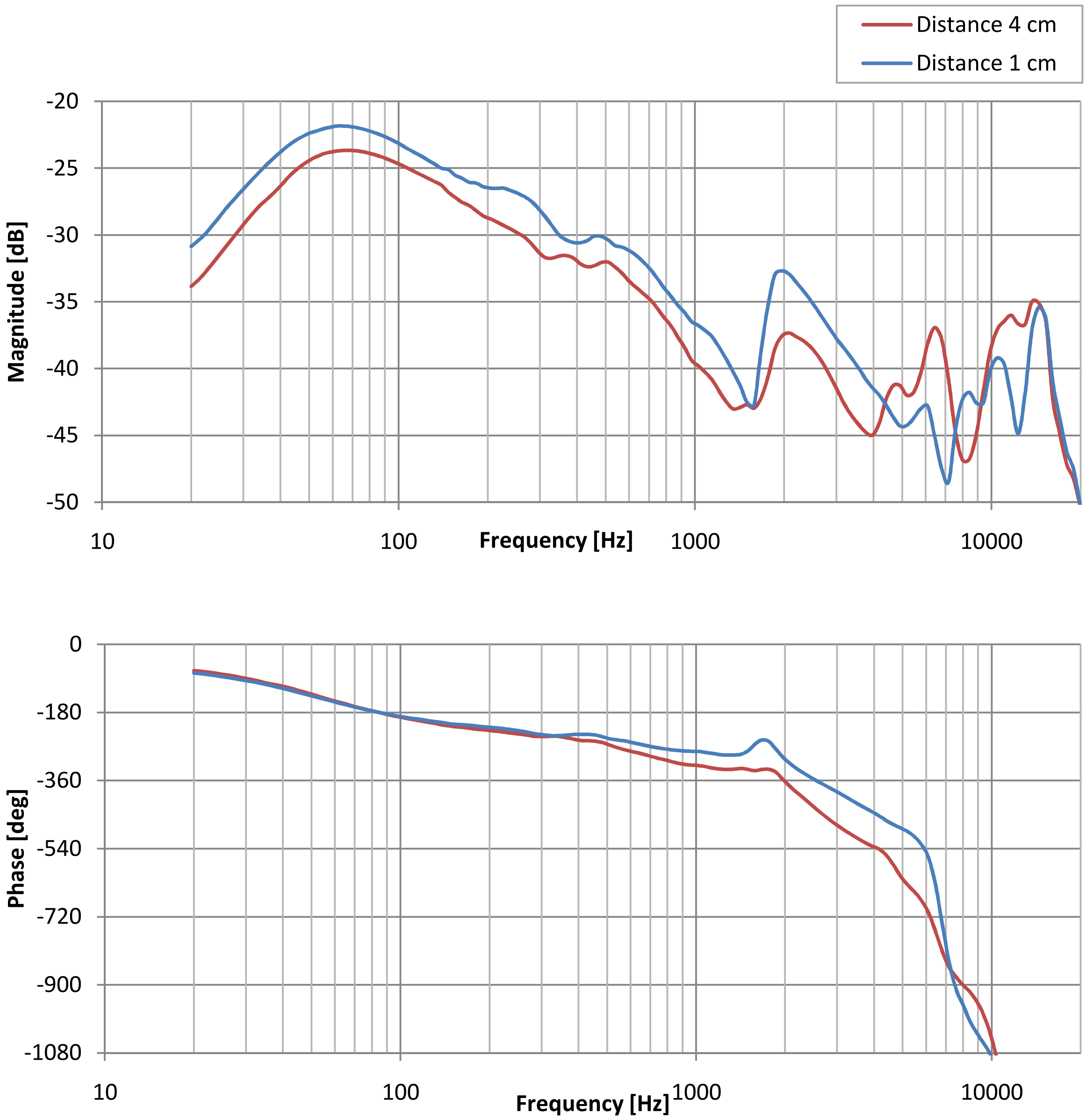



Fig. 3.40 Noise - ANC microphone

# 3.7 Performance of the prototype measurement system

## 3.7.1 Attenuation behavior

This measurement system will most likely be applied in a usual office ambience. In this case, there are no special provisions related to noise attenuation around the measurement system. Therefore, it was necessary to provide for proper attenuation performance of the box in both directions, outside→in and inside→out. The selected material and the decision to keep the absorption value inside the box as high as possible paved the way towards proper attenuation. Additionally, we attached great importance to sealing all possible leakages. This includes a special gasket strip between cap and body and hermetic sealing of all holes.

Fig. 3.41 Attenuation measurements

### 3.7.1.1 Inside→Out

Fig. 3.42 shows the attenuation from the inside of the box to the ambience around. To arrive at the resulting attenuation curve, 6 measurements were performed. 4 + 1 measurements were made from each side around and from the top side of the box. The resulting curve is the arithmetic average of those 5 curves. 1 measurement was made inside the box to get the relative attenuation value by subtracting the averaged outside curve from the inside curve.

Pink noise was used as source signal and the distance was 0.5 m. The reason for this distance is the application area as described at the top of this section. It is based on the assumption that the standard user will be positioned at nearly 0.5 m distance during usage of the measurement system on its desktop.

Basically, the attenuation is, of course, frequency-dependent. But, as shown in fig. 3.42, it is not that linear in magnitude over the measured frequency, as we may have expected. Especially at lower frequencies around 100 Hz, attenuation seems to be working better than in the higher range. This was founded on the reasons that the nominal power of the speaker inside the box is limited. This, of course, becomes noticeable at lower frequencies when compared to fig. 3.43 - where attenuation from outside→in seems to be relatively linear. The reason for this is in the performance of the speaker used outside the box. This system produces a well-balanced and linear frequency response, also down to nearly 20 Hz, due to deployment of a subwoofer. This results in correct reproduction of the source signal at all measured frequencies.

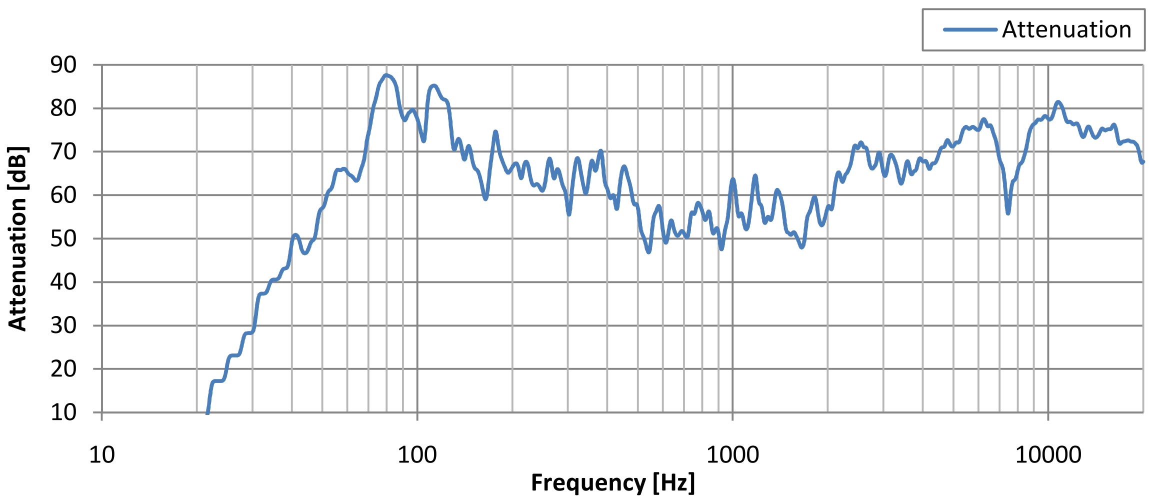



Fig. 3.42 Attenuation to the outside

### 3.7.1.2 Outside→In

The attenuation to the inside of the measurements box is important, too. The crucial point is about the noise outside the box that may produce disturbance during the measurement process. To avoid such disturbances, particular care was taken regarding all gasket seals around the box. To achieve the attenuation curve, 2 measurements were necessary; one measurement inside the box and one on the outside. For both measurements, pink noise was represented via the 4 loudspeakers and 1 subwoofer around the box. The difference of the 2 measurements results in attenuation of the box shown in fig. 3.43.

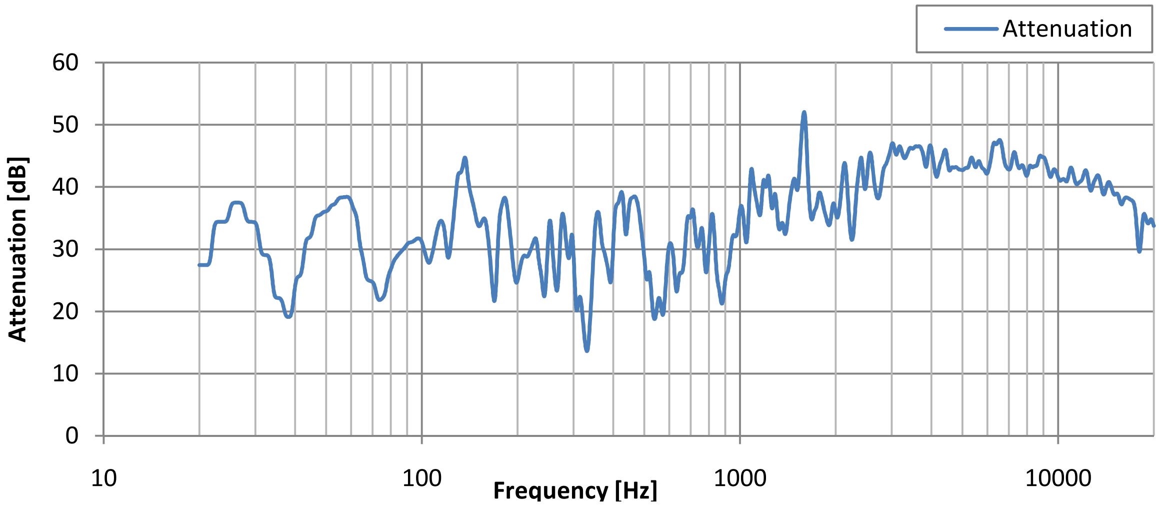



Fig. 3.43 Attenuation from outside to inside the box

## 3.7.2 Reverberation time

For the sake of completeness, the reverberation time inside the box was measured, too. The best possible performance we need in our case is very low reverberation at all frequencies because we want to measure the source signal without any disturbances. The applied measurement system is called “CLIO” and the source signal used was a sine sweep. The measurement was averaged over periods and the results show in fig. 3.44. As expected, we got very low values, especially at frequencies above 500 Hz. But the values at 0.1 s at 200 Hz and 0.3 s at 63 Hz are also quite satisfying. A list with detailed reverberation values can be found in

Appendix B - Datasheets.

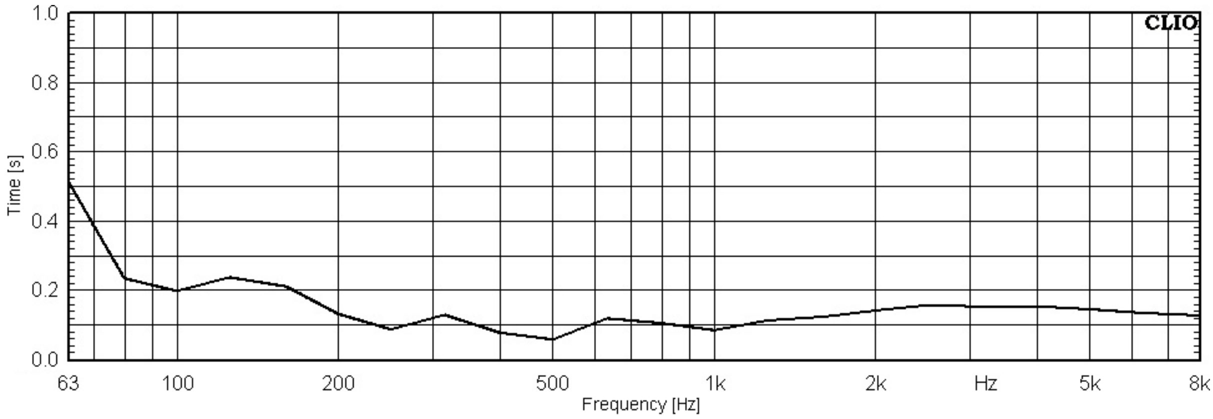



Fig. 3.44 Reverberation time inside the measurement chamber

## 3.8 Comparing 3 different IEMs

In order to evaluate correct functioning of the measurement system, the ANC filter curve of 3 different IEMs (fig. 3.45) was measured and compared. These are the DUTs:

1) Sony MDR - NC22 (Reference headphone)
2) Philips SHN4600
3) Audio Technica (Prototype)

Fig. 3.45 Sony MDR-NC 22, Philips SHN 4600, Audio Technica prototype

Figures 3.46 – 3.49 show all performed measurements needed for an ANC - filter determination. The figures compare each of the above mentioned DUTs in every single measurement. Fig. 3.49 shows the calculated ANC – filter curve. The figures demonstrate the different characteristics and performances of the various IEMs. Furthermore the measurements with different IEMs should verify the performance of the measurement system dealing with various DUTs.

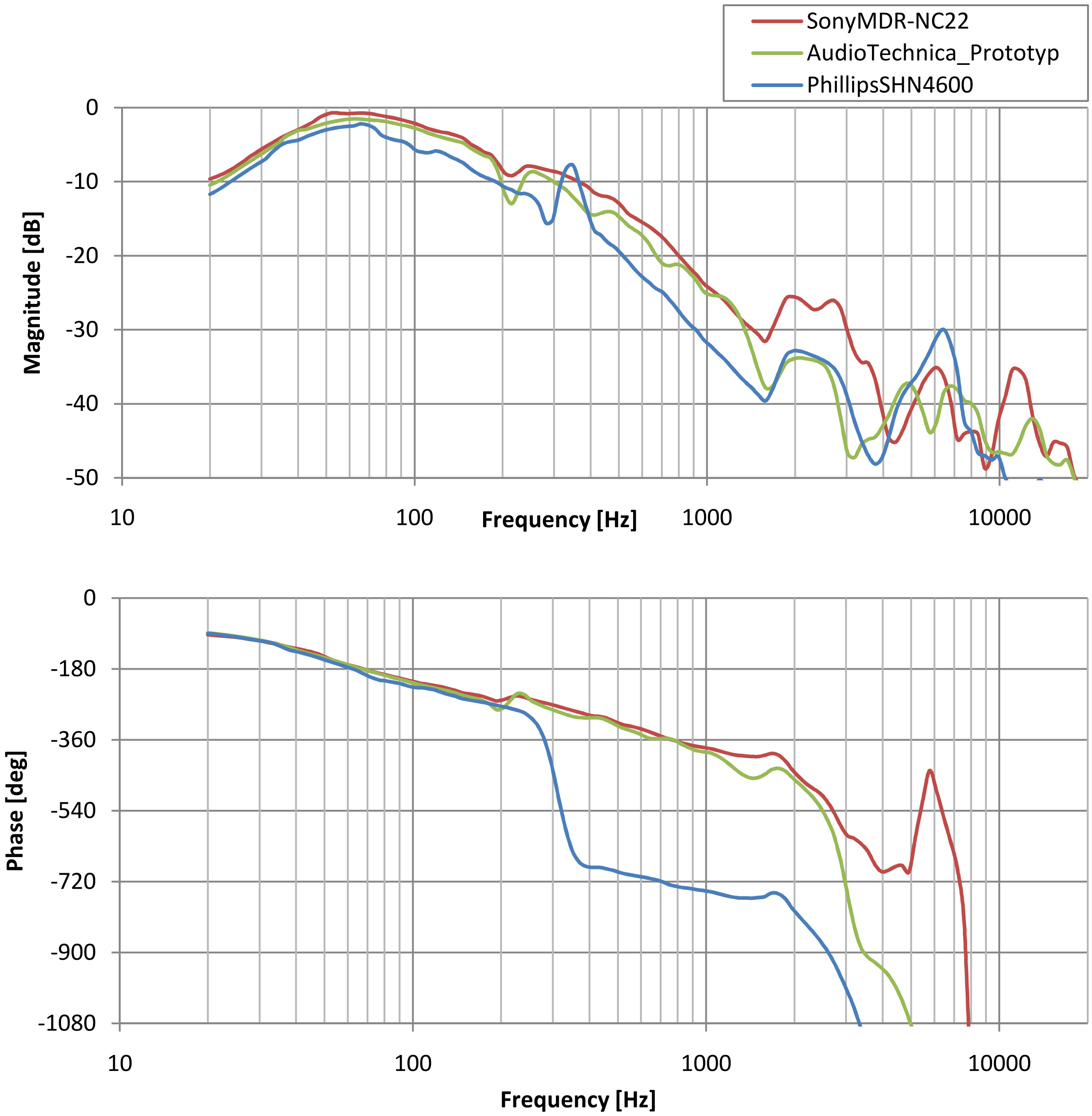



Fig. 3.46 Passive attenuation

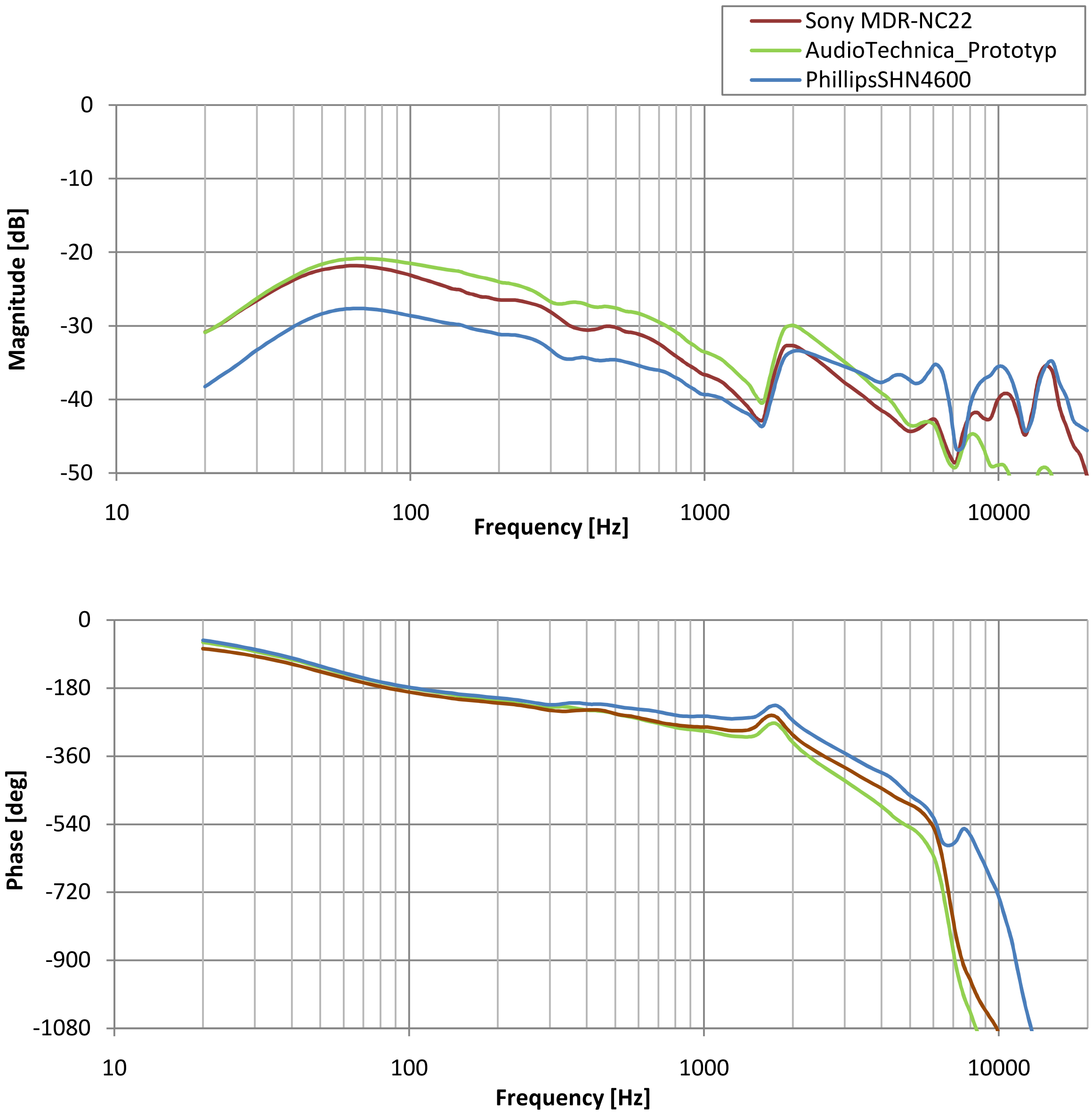



Fig. 3.47 Noise to ANC - microphone

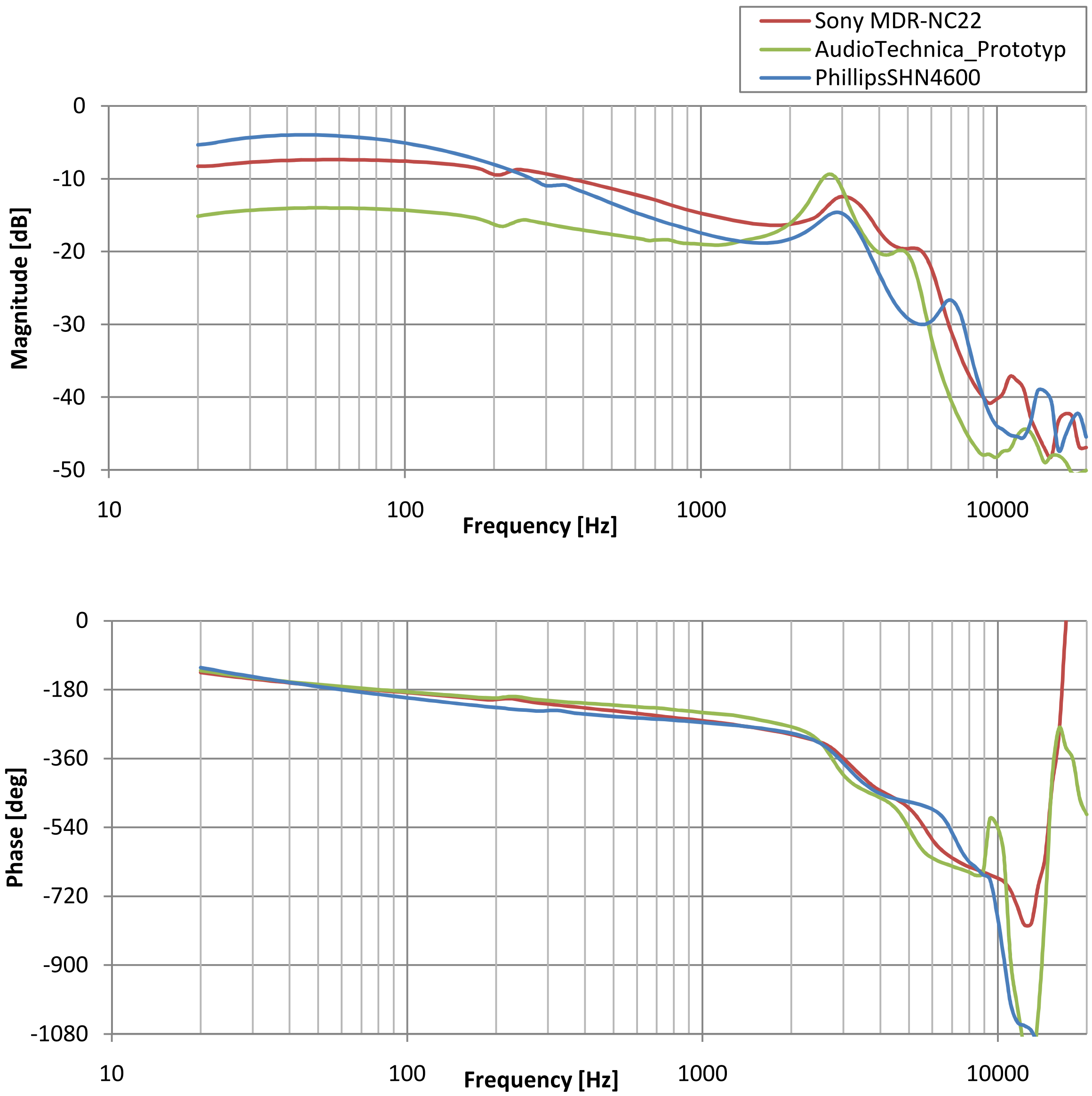



Fig. 3.48 IEM to measurement microphone

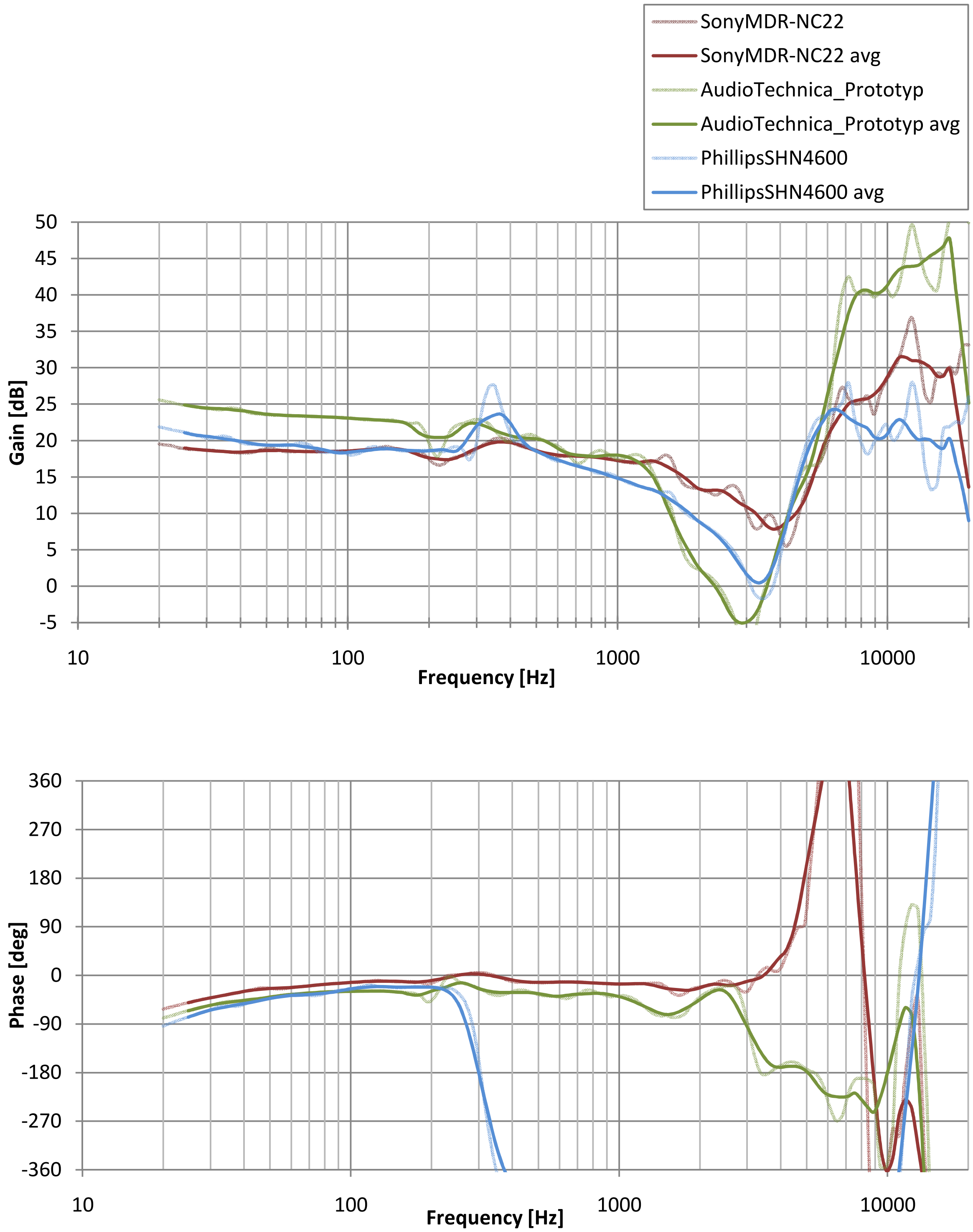



Fig. 3.49 Filter Gain and Phase of 3 different IEM’s

# 4 The Software solution

Another big part of this diploma thesis is about the design of a measurement software solution, based on the measurement hardware system developed and described in chapter 3. The principal task of the software is to provide the option of controlling the measurement system and guaranteeing simple operating during the measurement process. The software should accompany the employee/user during every single step of the measurement process and should help in trying to reduce the potential of human errors during the process. A further requirement is the simple-to-handle installation process via .exe format and a solidly-working process on Windows XP, Windows Vista and WIN 7 platforms. The software will be developed for one special type of audio interface which will be included in the hardware package. The programming environment was MATLAB R2010a.

## 4.1 Software features

The developed software parts should include the following features:

- Support multichannel I/O[10]
- Exponential sweeps as excitation signal
- Output: Frequency and phase response of DUT
- Calculation of ideal filter curve
    - Display Curves
    - Export data and/or graphic files
- Calculation of ANC performance
- GUI (Graphical User Interface)
- Installation guide and user manual

[10] Maximum 4 channels at the same time (2 Out, 2 In)

# 4.2 Signal processing

There are different methods to get the frequency and phase response of the DUT. Each method is asking for a different source signal to achieve the required impulse response and, finally, the frequency response. In chapter 2 some of the most common methods were introduced, and also the reason why an exponential sine sweep was used for the characterization of the ANC filter curves.

## 4.2.1 Verification of the environment

The first step, before developing the software, was to evaluate the necessary hardware components. This step was essential because the way of programming depends on the possibilities of the hardware used. Especially the possibilities of the I/O – Device determinate the performance of the software and the approach of the basic concept.

A big point in doing "real time" measurements with sweeps is the latency of the system. It is of fundamental importance to know about the latency behavior of the system you work with. The reason for this is the crucial information of delay between the two sweeps used during the calculation of impulse response (see fig. 4.1).

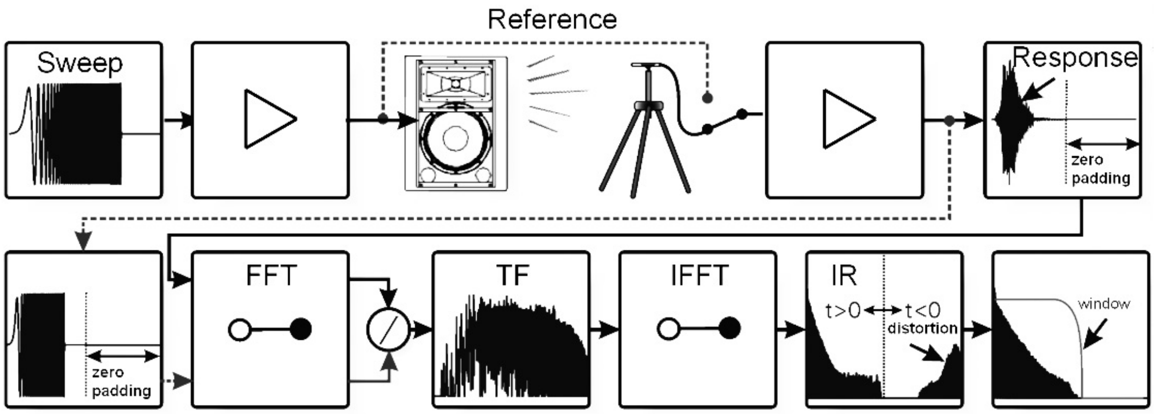



Fig. 4.1 FFT-based impulse response measurements, (Weinzierl, et al., 2009)

There are several ways of adapting the response sweep to the reference sweep, but, in any case, the delay caused by the system has to be considered. To avoid calculation of the system depending on delay and arrive at an operative system working independently from any delay it is useful to send each required signal over the same paths (fig. 4.2) - especially the reference signal. In this case there is no need to worry about the delay at all because each signal has the same path to go. The only important thing is to make sure every channel has the same delay time. Fig. 4.3 shows a channel delay measurement with the soundcard used to improve equal latency of each channel.

A further advantage of this method lies with the autonomy versus industrially-manufactured soundcard driver protocol ASIO, which provides low and constant latency for multichannel I/O, because MATLAB did not support ASIO at the time of developing this software. Furthermore, the measurements did not require "real" multichannel I/O because it was not necessary to run more than 2 I/O channel pairs at once (2 OUT, 2 IN). This kind of stereo in- and output is provided in MATLAB.

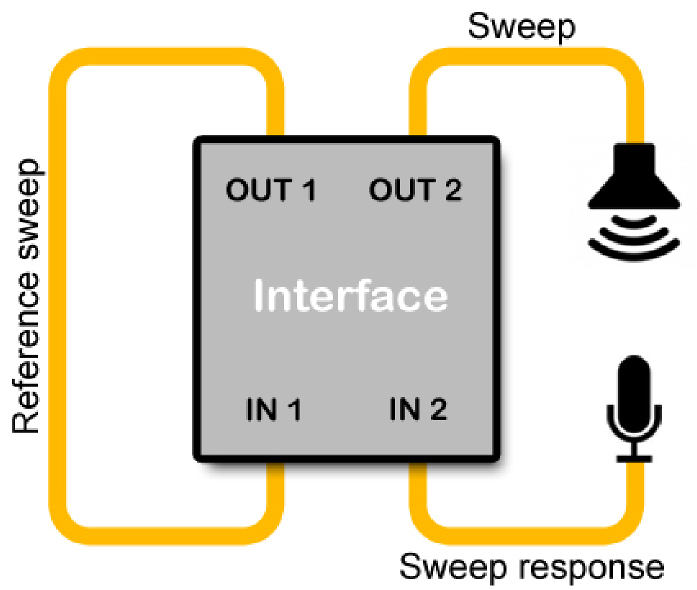



Fig. 4.2 I/O delay compensation

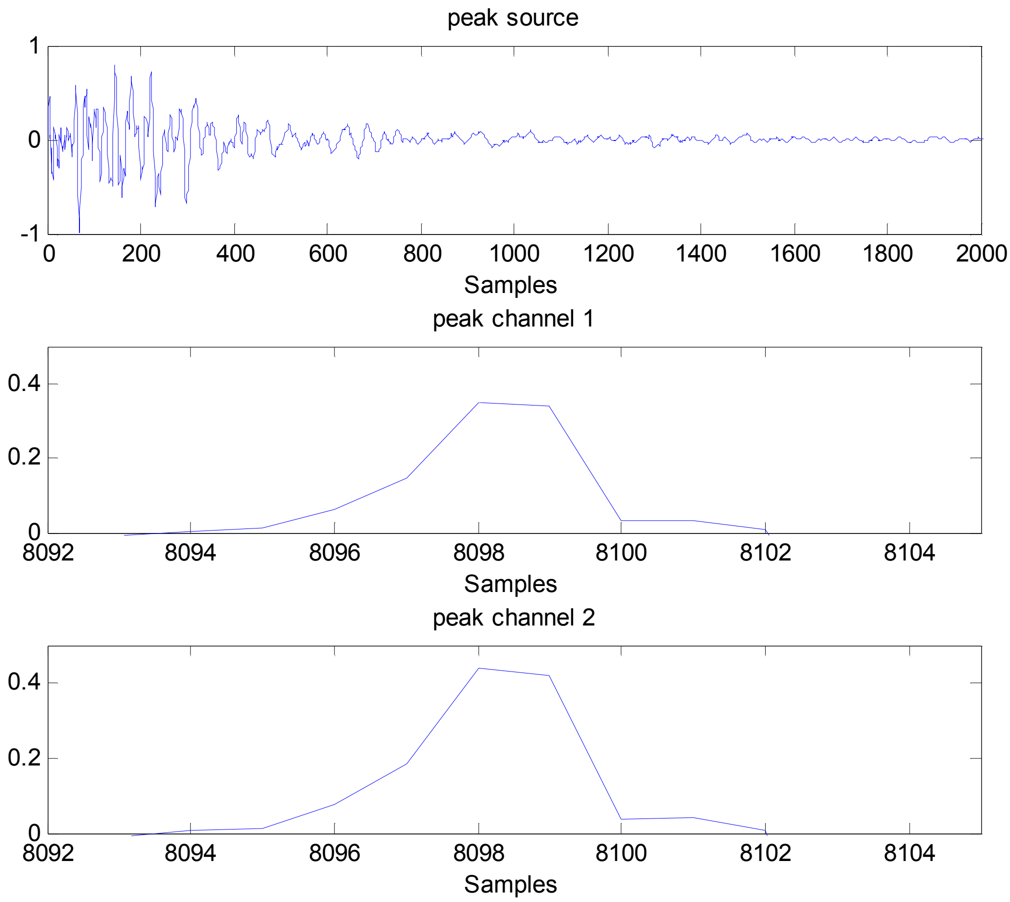



Fig. 4.3 Channel latency between channel 1 and channel 2

### 4.2.2 Sweep-based measurement of impulse response

The main subject during the calculations of the ideal filter curves is the measurement technique to get the right frequency and phase response of the DUT. There are many ways to achieve the required transfer function, like MLS (Maximum Length Sequence) or TDS (Time Delay Spectrometry). The choice of using an exponential sweep as excitation signal is based on several reasons. The main reasons are:

- Higher immunity against distortion and time invariance
- Implementation in time domain
- Pink spectrum (more energy at low frequencies)
- Short excitation signal

Detailed information about the exponential sweep method can be found in 2.4. Also, the underlying signal processing basics can be found in the theory chapter at 2.5.2.

#### 4.2.2.1 Signal paths

As mentioned above, the excitation signal for the impulse response measurements is an exponential sine sweep. The sweep will be generated in the program during the measurement process and feed output 1 and 3 as reference signal and output 2 and 4 as signal at the speaker and the IEM. Because MATLAB does not support ASIO, it is not possible to route the signals to each channel independently. Instead, it is only possible to use independent pairs of inputs and outputs. This constrains the routing as shown in fig. 4.4. This routing allows, on the one hand, separate processing of the measured impulse response of the measurement microphone and the ANC microphone and, on the other hand, the required routing for the output channels depending on the specific measurement under existing conditions.

The following tables will illustrate the routing for the 3 measurements which will be needed to get the ideal ANC filter curve (tab. 4.1, tab. 4.2), and the ANC performance curve (tab. 4.3, Tab. 4.4).

| | Measurement 1 | Measurement 2 | Measurement 3 |
|---|---|---|---|
| **Output** | 1/2 | 1/2 | 3/4 |
| **Input** | 1/2 | 3/4 | 1/2 |

Tab. 4.1 Soundcard I/O routing for ANC filter calculation

| | Type | Needs |
|---|---|---|
| **Measurement 1** | Passive attenuation | Speaker & measurement mic |
| **Measurement 2** | Noise 2 ANC mic | Speaker and ANC mic |
| **Measurement 3** | IEM 2 speaker | IEM & measurement mic |

Tab. 4.2 Measurement relations for ANC filter calculation

| | Measurement 1 | Measurement 2 | Measurement 3 |
|---|---|---|---|
| **Output** | 1/2 | 1/2 | 1/2 |
| **Input** | 1/2 | 1/2 | 1/2 |

Tab. 4.3 Soundcard I/O routing for ANC performance calculation

| | Type | Needs |
|---|---|---|
| **Measurement 1** | Coupler response | Speaker & measurement mic |
| **Measurement 2** | Passive Attenuation | Speaker & measurement mic |
| **Measurement 3** | Active Performance | Speaker & measurement mic |

Tab. 4.4 Measurement relations for ANC performance calculation

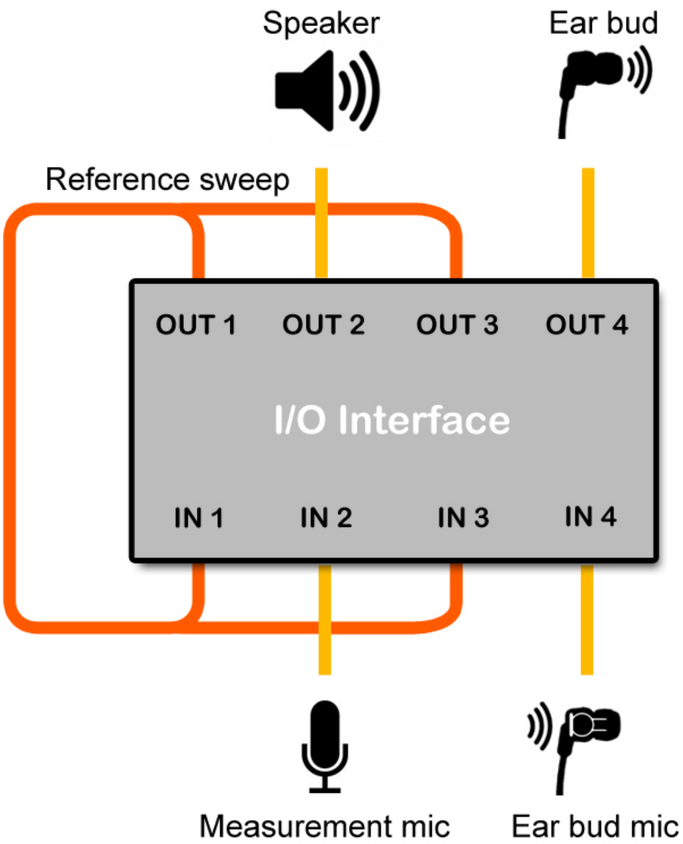



Fig. 4.4 I/O soundcard routing

This complex routing is also a product of the fact that every single measurement needs to be supplied with only one sweep. For example, in measurement 1, only the speaker has to be supplied with the sweep but, of course, NOT the IEM. Otherwise there will be 2 sweep sources simultaneously, and this will lead to the wrong impulse response. The same requirements apply to the other 2 measurements.

An accurate description of the measurement process and the filter calculation can be found in 2.3.3.

### 4.2.3 Data fitting

During the measurements, a huge amount of data was produced, mainly depending on the sampling frequency used. However, the sample rate has to be at least 44100 Hz, in order to achieve a good solution at low frequencies - the low frequency domain is especially important in ANC filter curves. To reduce datasets produced during measurement to useful value practicable for users, but simultaneously achieve high accuracy at low frequencies, it was necessary to interpolate and smooth the data received.

The interpolation is necessary because the calculated filter curves have to be saved as values in an excel sheet. To handle the data in a useful way, the frequency values are shown in a logarithmically-scaled style. To achieve uniformly distributed values over the whole logarithmic frequency range, two things have to be done. First, a frequency vector with uniformly logarithmic distributed values has to be created. Second, the corresponding values of the curve have to be found. To get these amplitude and phase values, corresponding to each of the logarithmic frequency values, it was necessary to interpolate between calculated values. Fig. 4.5 shows the principle of this process.

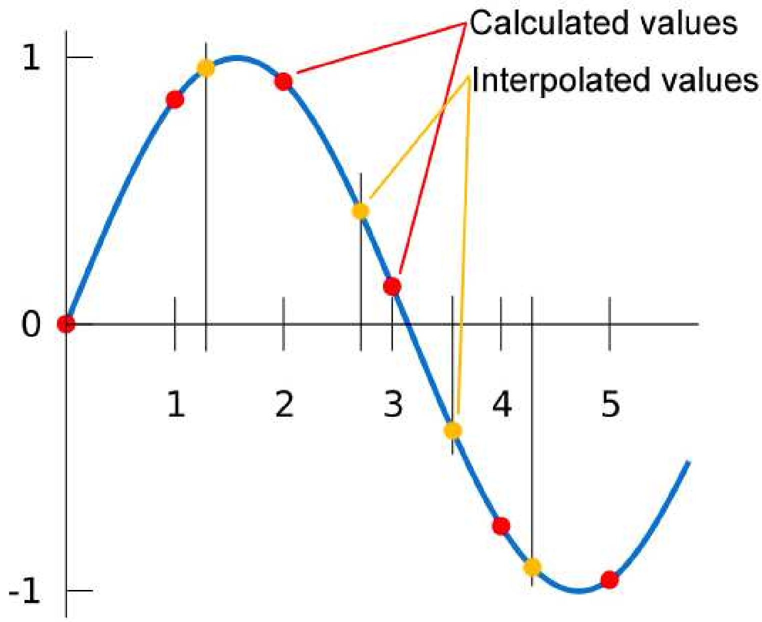



Fig. 4.5 Interpolation for logarithmic scaling

The last step, after interpolating, is the smoothing of the resulting curves. This step is founded in the further use of the ideal filter curves. The real implementation for the ANC filter will be done with analog chip design. The analog implementation leads to a limitation of possible filter characteristics. Therefore, the data smoothing approximates the real filter implementation. The smoothing algorithm is using a moving average filter. The span for the moving average is 5. This means, the algorithm uses 3 samples to calculate 1 averaged value.

The process for the phase calculations is similar. Additionally, phase shifts at 360° will be compensated for automatically and, thus, will not appear in the results.

## 4.2.4 Calculation of ideal filter curve and ANC performance

### 4.2.4.1 Ideal filter curve

The calculation of the ideal filter curve is a simple subtraction of the 3 performed measurements (tab. 4.2).

$$\text{Filter gain} = \text{Amplitude 1} - (\text{Amplitude 2} + \text{Amplitude 3}) \qquad [4.1]$$

$$\text{Filter phase} = 180° + \text{Phase 1} - (\text{Phase 2} + \text{Phase 3}) \qquad [4.2]$$

The filter gain curve represents the required filter curve to be implemented for best-possible ANC performance, depending on the DUT.

The filter phase curve represents the required phase curve, including a 180° shift (phase inversion for destructive interference), for the best ANC performance depending on the DUT.

#### 4.2.4.2 ANC performance

To calculate ANC performance, 3 measurements are necessary as well. (cf. tab. 4.4) The principle measurement process and the calculations for the single frequency responses are identical with the ANC ideal filter curve determination. In order to calculate and, thus, compare ANC performance against performance without ANC, the following simple subtractions were implemented:

$$\text{ANC ON} = \text{Amplitude 1} - \text{Amplitude 2} \qquad [4.3]$$

$$\text{ANC OFF} = \text{Amplitude 1} - \text{Amplitude 3} \qquad [4.4]$$

### 4.2.5 Evaluation process

In order to prove accuracy of achieved results, it was necessary to compare the output of the developed system with measurements processed by a reference system the results of which are definitely correct. In this case, a measurement system developed by a company named Audio Precision was used as reference. The two reasons for this choice were the usage of this system for all ANC measurements up until now and the good reputation enjoyed by this company, giving way to the assumption that we can trust the results gathered using this system.

Fig. 4.6 shows the first of three required measurements, the passive attenuation, once measured with Audio Precision (blue line) and once with the MATLAB-based, self-developed software (red line). The figure provides clear evidence of noticeable differences solely above 14 kHz in magnitude and around 10 kHz in phase. The main operating range of ANC headphones is located between 20 Hz and a maximum of 10 kHz because of the limited impact of the analog filter circuit and of the increasing phase shift at higher frequencies, which cannot be handled with simple analog filter implementations.

The minimal deviations from the reference are due to a smoothing algorithm used in MATLAB which is slightly different to the algorithm of Audio Precision. All in all, the measured curves are favorable and provide the results we need to arrive at the right ANC performance.

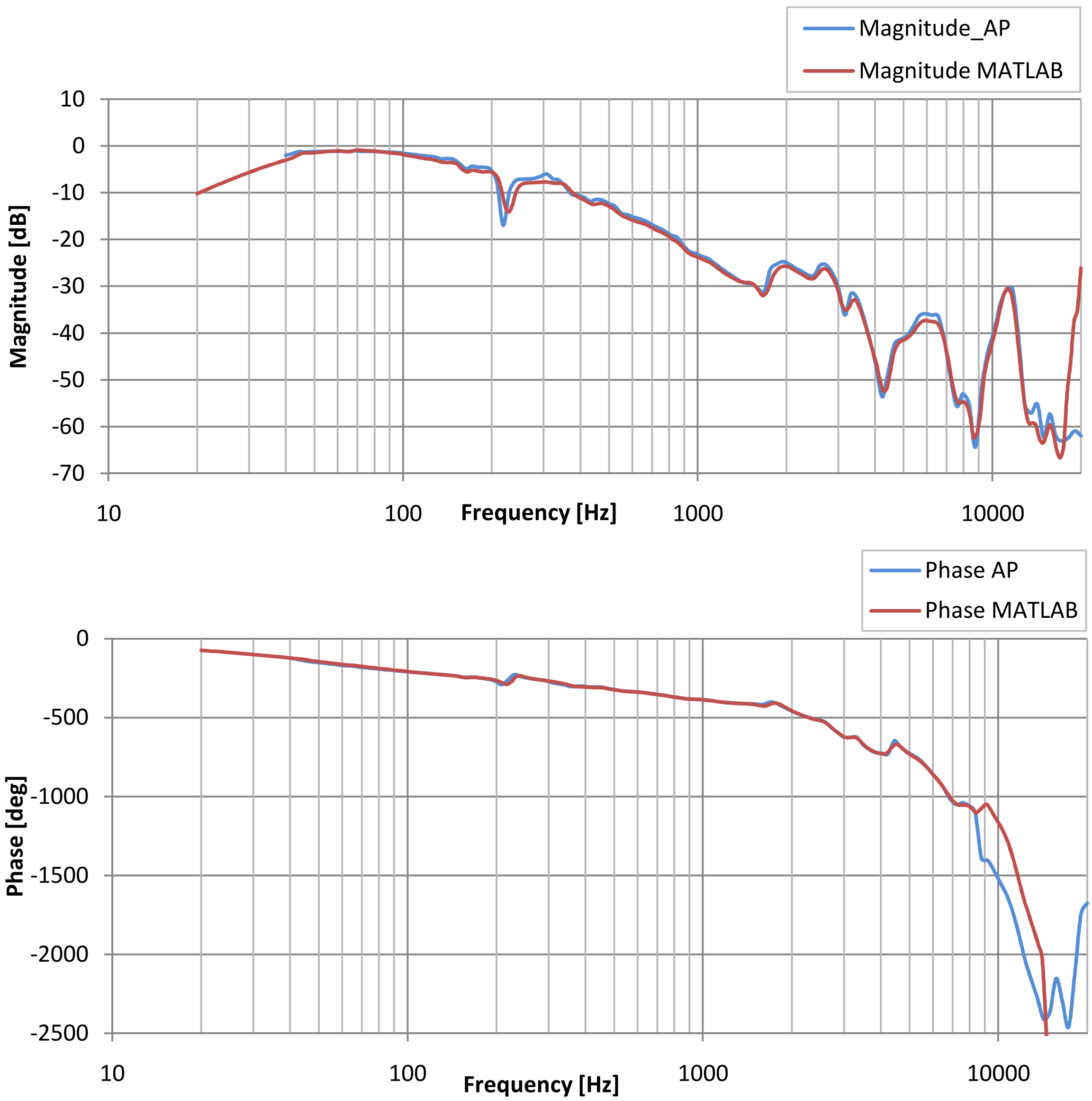



Fig. 4.6 Frequency and phase response, passive attenuation

# 4.3 GUI

The main goal of the GUI should be about supporting the user during the measurements and should guarantee simple and easy handling, in spite of awareness of correct solutions. In consideration of the matter of fact that there is no guarantee that end-users are experts in the area of acoustics, the decision was taken to design the GUI in a way allowing for adjusting only the core functions of the measurement system. On the one hand, this should prevent any instances of producing unintended errors; on the other hand, it is essential that the set parameters were accurately selected. The parameters have to be selected in a way that the measurement system works over the whole range of its application area with satisfying accuracy.

Another part of this thesis is about creating a user guide (Appendix A – User Manual). More information about the functions and the installation process can be found in there.

## 4.3.1 Installation process

Because of the MATLAB-based programming, it is necessary to install the MATLAB - runtime executable (exe) before running the program. The runtime exe is a free MATLAB product and will be included in the package of the measurement software. Alternatively, it can be downloaded on www.matlab.com.

To run the implemented executables FCT.exe or APT.exe, copy the folder into any directory on the computer and open the exe.

## 4.3.2 Layout

Because of 2 required analysis matters, the measurement software was separated into 2 applications. This should ease handling and keep users focused on the actual measurements. The filenames of the two applications are:

FCT.exe and APT.exe

To keep it simple and allow for as much user flexibility as possible, both GUI structures are based on a tab design. The user may switch between single measurement steps and retry every step as often as necessary. To prevent wrong calculations as a result of, for example, measurements not executed, some error dialogs have been implemented. Thus, it should not be possible to end up with a wrong result based on wrong software handling.

## 4.3.3 Filter Calculation Tool (FCT.exe)

### 4.3.3.1 Configuration

After opening the FCT.exe, an MS DOS window opens and initializes the program. The Configuration tab also serves as welcome page. This tab includes the property of switching between tabs, entering a filename, opening the user manual and taking notice of how to wire the measurement system. Also, a short description of the main functions and instructions is displayed. Fig. 4.7 shows a screenshot of the Configuration tab.

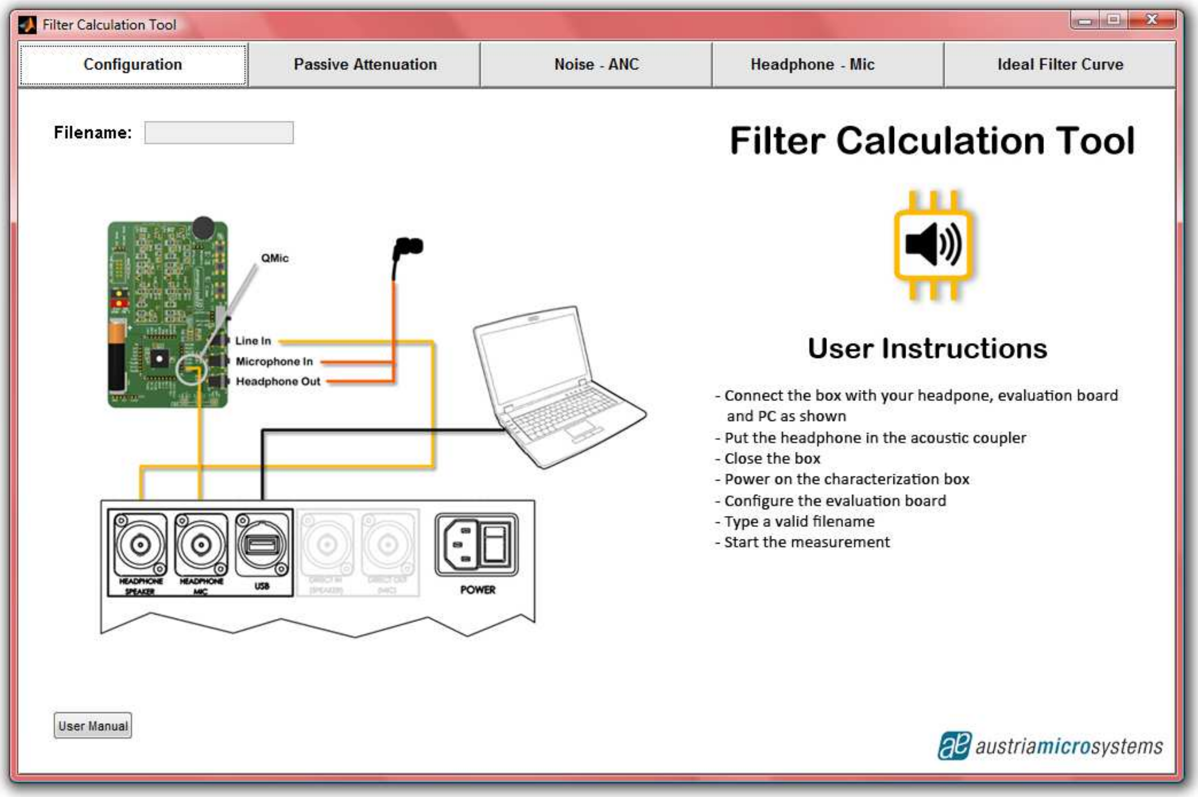



Fig. 4.7 Welcome page / Configuration, FCT.exe

### 4.3.3.2 Passive attenuation and Noise - ANC

The two tabs Passive Attenuation and Noise – ANC (fig. 4.8) are provided for measurements 1 and 2. The user may start the sweep and view the frequency and phase response of the DUT for the two measurements executed. The GUI also includes the possibility of a zoom function for all displayed figures, for viewing details of the resulting curves. The user is also given the choice of deleting the performed measurement and repeating it. Additionally, it is possible to save the data of every single measured curve in a user-defined folder. The following data formats are supported:

- .xls
- .csv

The data will be saved as amplitude and phase vectors in correlation to the corresponding frequency values.

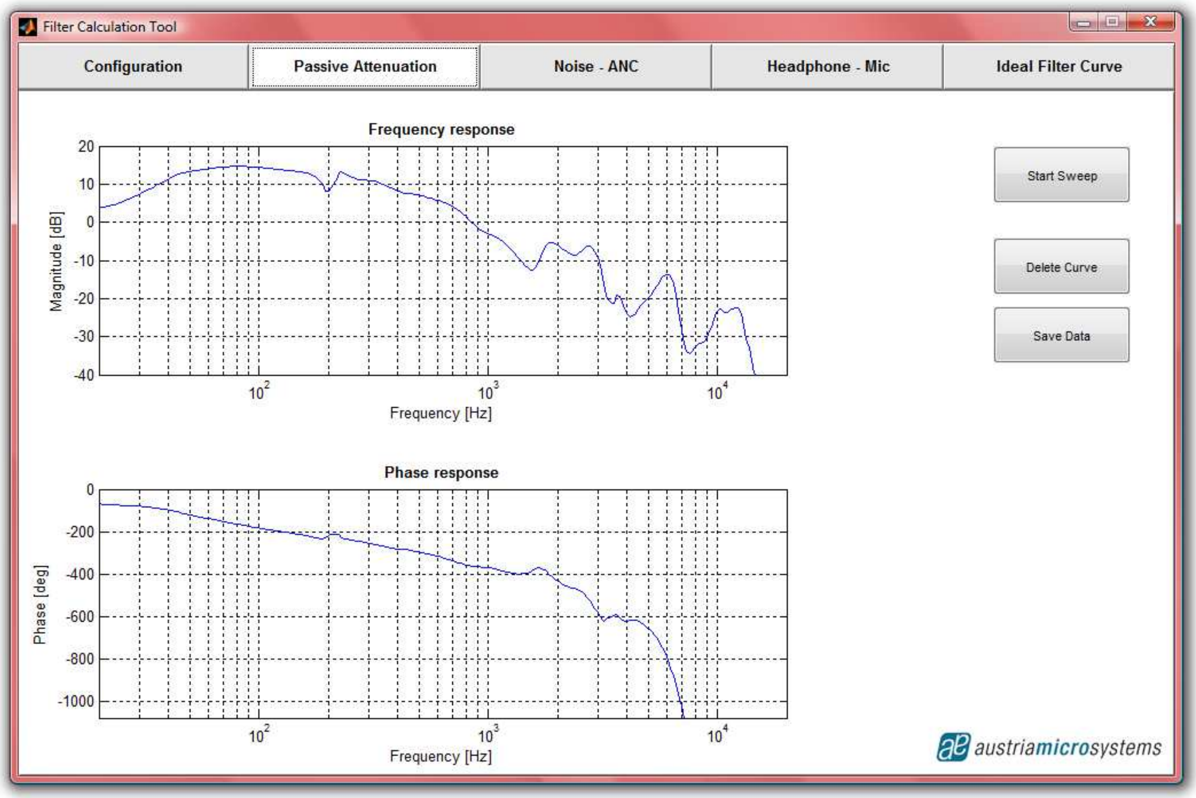



Fig. 4.8 Passive attenuation / Noise - ANC measurement, FCT

### 4.3.3.3 Headphone - Mic

The Headphone - Mic tab has exactly the same functions as the previous tabs, but additionally provides the option of attenuating the input signal at a rate of -10 dB or -20 dB. The reason for this feature is the variety of impedances of different IEMs. IEMs with lower impedance will lead to an increase in levels. At worst case, this could lead to distortions in the IEM speaker or in the measurement microphone and have a negative impact on the resulting curves. In order to get rid of this problem, a peak-detection was implemented. If the output level of the IEM increases at a defined level, a message alerts the user on attenuating the signal. (fig. 4.9)

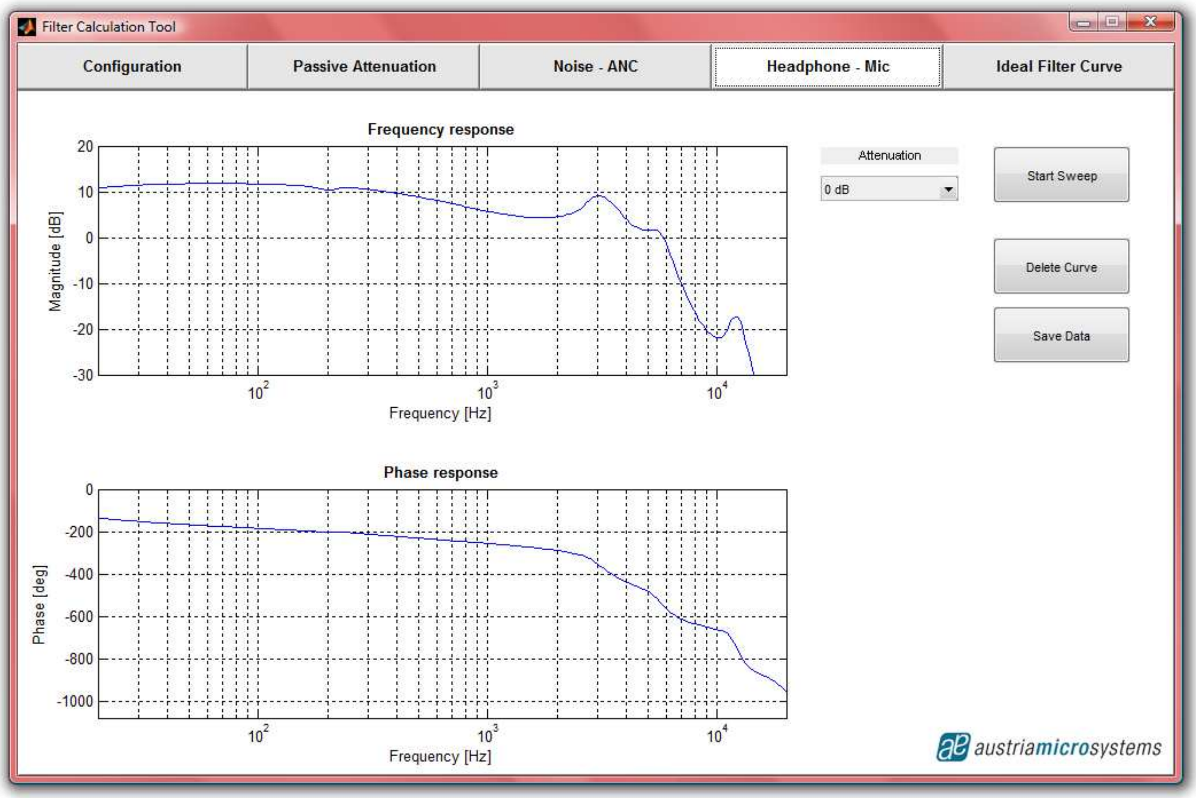



Fig. 4.9 IEM - Mic, FCT

### 4.3.3.4 Ideal filter curve

This tab (fig. 4.10) allows the user to calculate the ideal filter curve for the DUT, if all 3 measurements were done. The resulting filter curve follows the calculations described in 2.3.3.4. In case of any missing or corrupted measurement, an error dialog pops up and suggests to the user retrying the mentioned step. Furthermore, there are two possibilities for exporting the calculated data:

- Export as .xls or .csv file:
  The data, similar to the other tabs, will be saved as amplitude and phase vectors in correlation to the corresponding frequency values.

- Export as graphic file:
  The graphic export saves the calculated figures. It supports the following graphic formats:

  - .jpg
  - .emf
  - .tif
  - .png

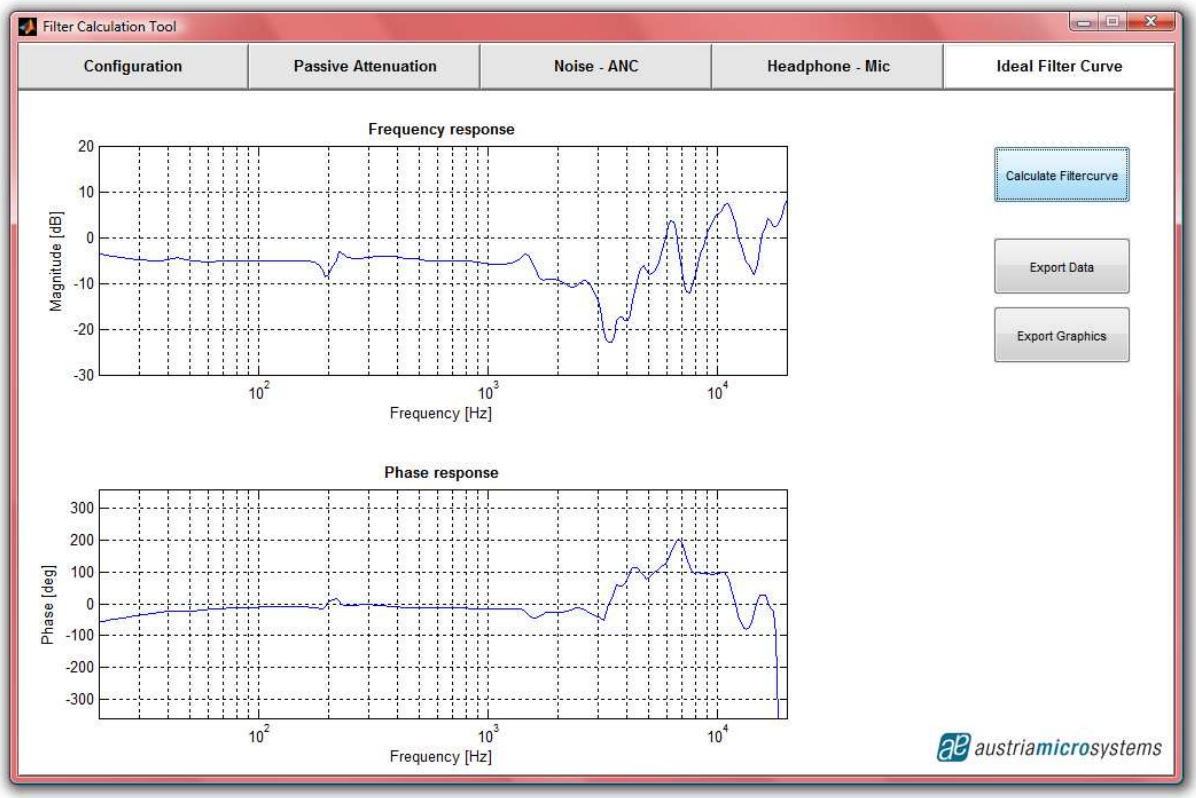



Fig. 4.10 Ideal filter curve, FCT

## 4.3.4 ANC Performance Tool *(APT.exe)*

The layout principles used for the APT.exe are exactly the same as those for the FCT.exe. The big difference is the functionality of the application. The APT.exe should help analyze the ANC performance by means of 3 measurements, including the coupler response itself and the IEM transfer function with and without ANC support. The next section gives a detailed description of the single functions of the mentioned GUI.

### 4.3.4.1 Configuration

The Configuration tab is similar to the FCT.exe

### 4.3.4.2 Coupler response

The measurement in this tab is the basis for the ANC performance calculations. In this step, the frequency response of the stand-alone coupler will be measured. This frequency response is needed for further calculations concerning ANC performance. For the calculation of ANC performance, it is sufficient to only measure the frequency response (fig. 4.11). Phase information is not relevant. The options of retrying the measurement and saving the data are similar to the FCT.exe. Further, support of the zoom in and zoom out function is available. It is very important to remove the IEM from the acoustic coupler during this measurement to only get the isolated coupler response.

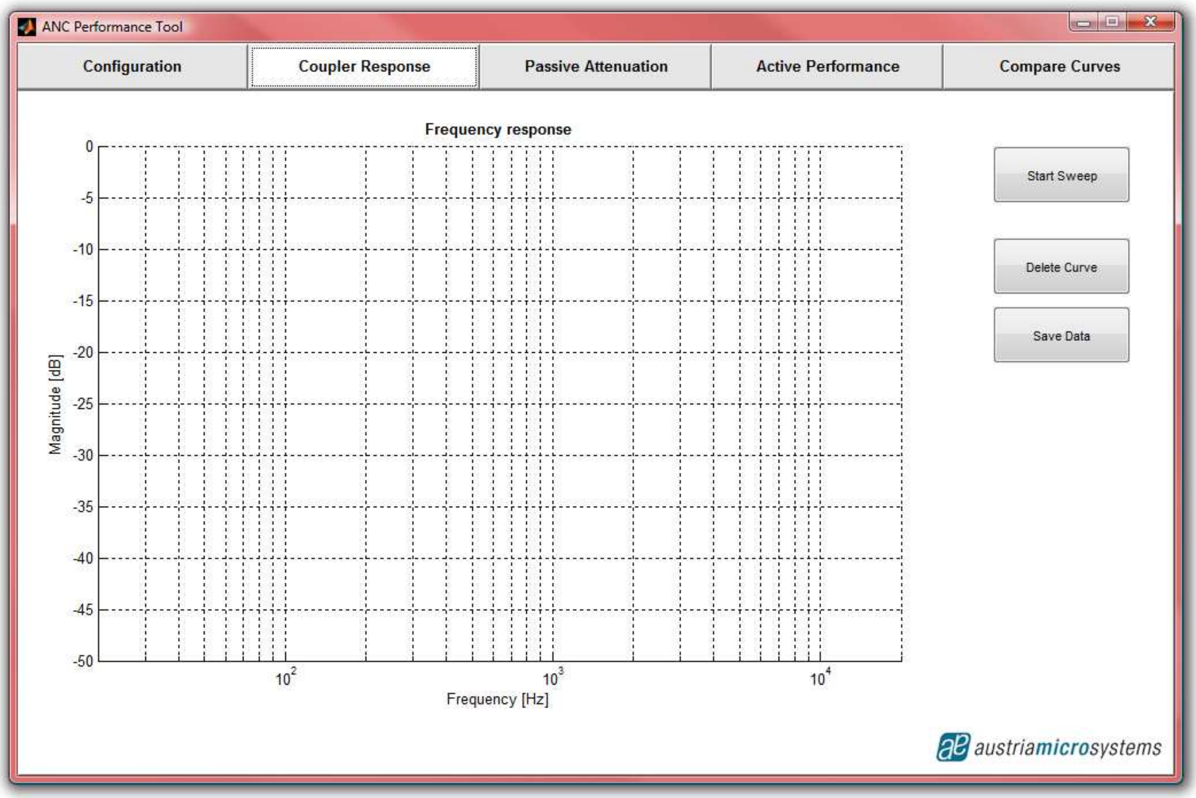



Fig. 4.11 Coupler response measurement, APT

### 4.3.4.3 Passive Attenuation

The view of the passive attenuation tab is similar to the coupler response (fig. 4.11), with the exception of the displayed curve being a result of measurement 1 and this measurement. Because of the dependence of both measurements, an error dialogue occurs if measurement 2 will be started and the first measurement was not performed at this stage.

$$\text{Passive attenuation} = \text{Measurement 1} - \text{Measurement 2} \qquad [4.5]$$

Because of the subtraction, the resulting curve shows the "real" part of passive attenuation, without the transfer function of the coupler. Additionally, the display area for this and the next tab was limited between -35 dB and +10 dB. Values outside this area should not be possible under common circumstances.

### 4.3.4.4 Active performance

The goal of the measurement tab Active Performance should be the visualization of the pure ANC performance - this means without the passive attenuation effect caused by the IEM. To achieve this result, it is necessary to activate the ANC system during measurement. To calculate the pure ANC performance, the following calculation was implemented:

$$\text{Active performance} = \text{Measurement 1} - \text{Measurement 3} \qquad [4.6]$$

An additional feature of this tab is the possibility of adjusting the filter gain for the DUT. The 3 gain adjust buttons, on the right bottom of the GUI allow the user to display several performed measurements in one figure. Each of those 3 measurements is rendered with its own color and can be used for recursive approximation to achieve the best possible filter gain for the DUT. In addition to the passive attenuation measurement, an error dialogue occurs if the necessary measurements were not completed during the start of this measurement.

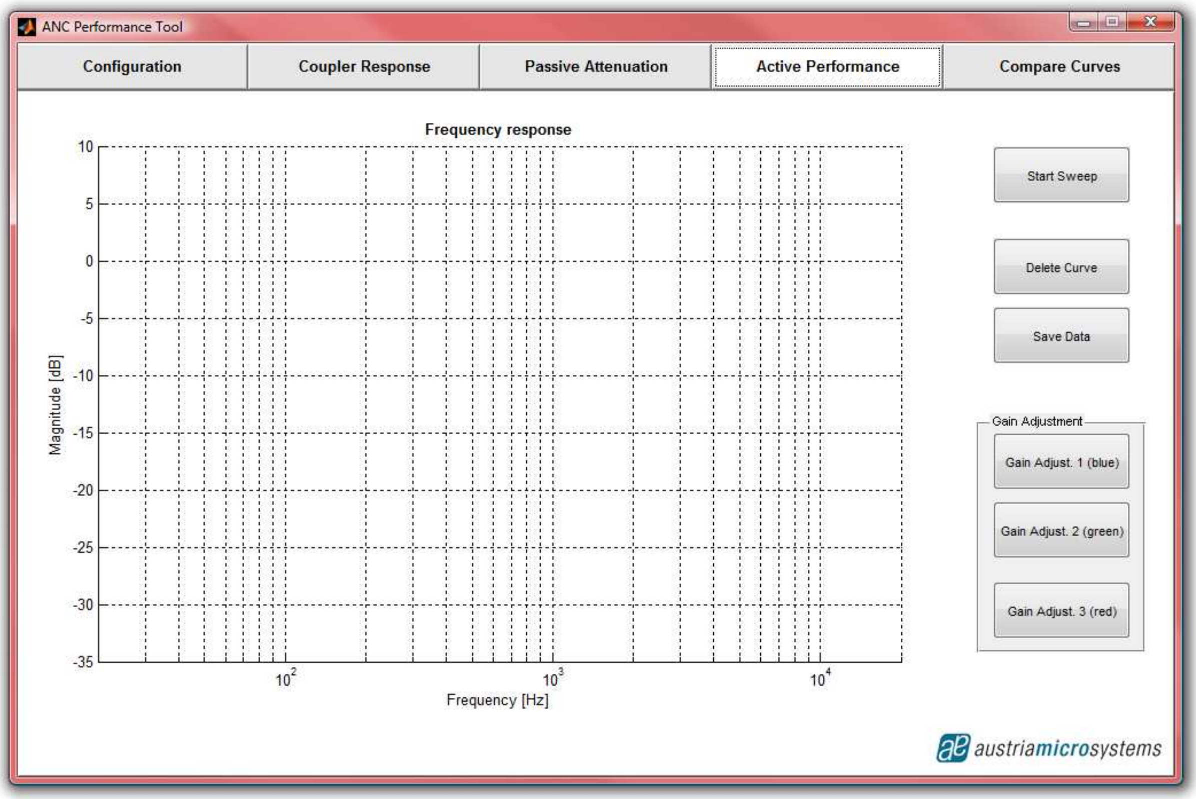



Fig. 4.12 Active performance, APT

### 4.3.4.5 Comparing curves

Finally, the last tab displays the completed ANC performance. The user is given the possibility of comparing the performances of the IEM between "ANC ON" and "ANC OFF". The visibility of the resulting filter curves can be chosen via two checkboxes on the right bottom of the GUI. Similar to the FCT.exe, data export as .xls or .csv and graphic export as .jpg, .png, .tif or .emf file is supported. Fig. 4.13 shows the Comparing curves tab.

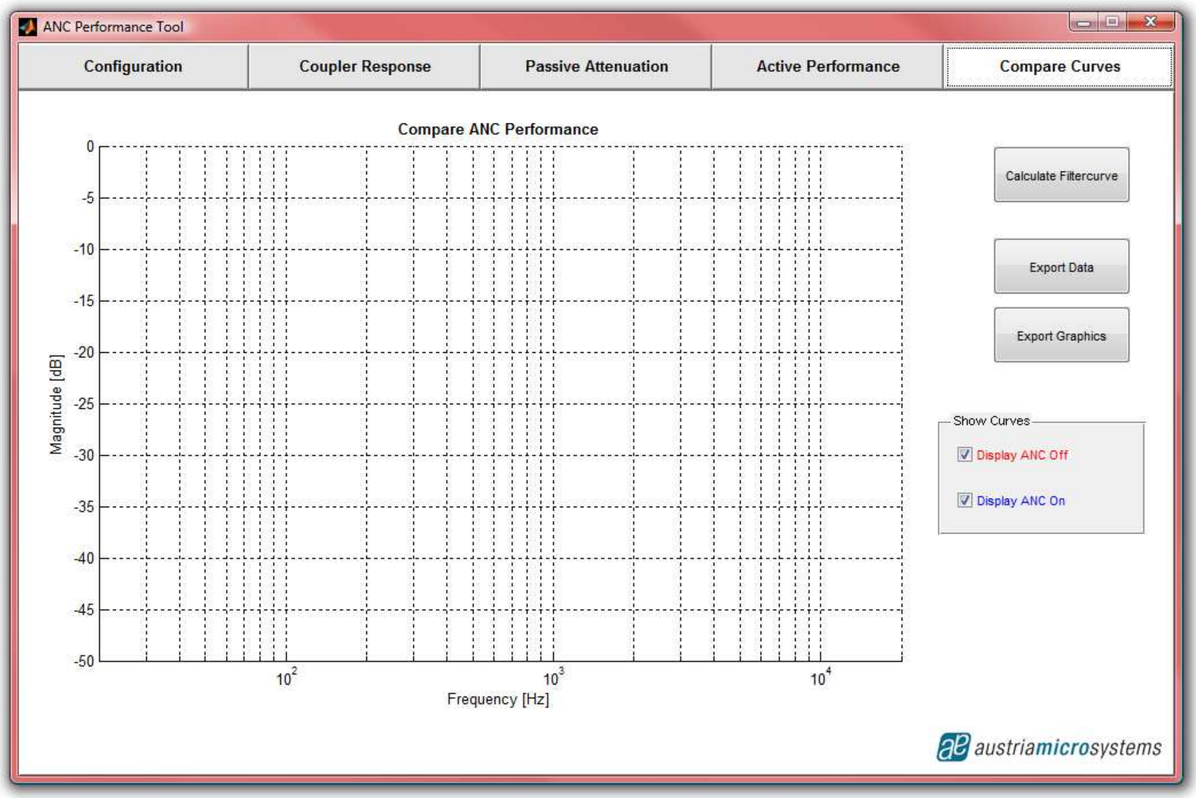



Fig. 4.13 Comparing curves, APT

# 5 The finalized measurement system

This chapter summarizes the finalized and completed characterization system including all developed hardware and software parts. I thought this to be useful for gaining an overview on all the different developments during this thesis to see how they work together. Another point should be presentation of the completed working system in the context of this thesis. Several details were developed on the way and it was not possible to mention every small development step in detail. Therefore, this chapter presents the final results, also including these developments mentioned. Fig. 5.1 shows a complete flow chart of the characterization system in use with the filter characterization software FCT.exe. The flow chart represents all hardware parts including the measurement box with acoustic coupler and peripherals like soundcard, microphone pre-amp, amplifier, speaker and microphone, as well as the software part with a schematic signal processing stage.

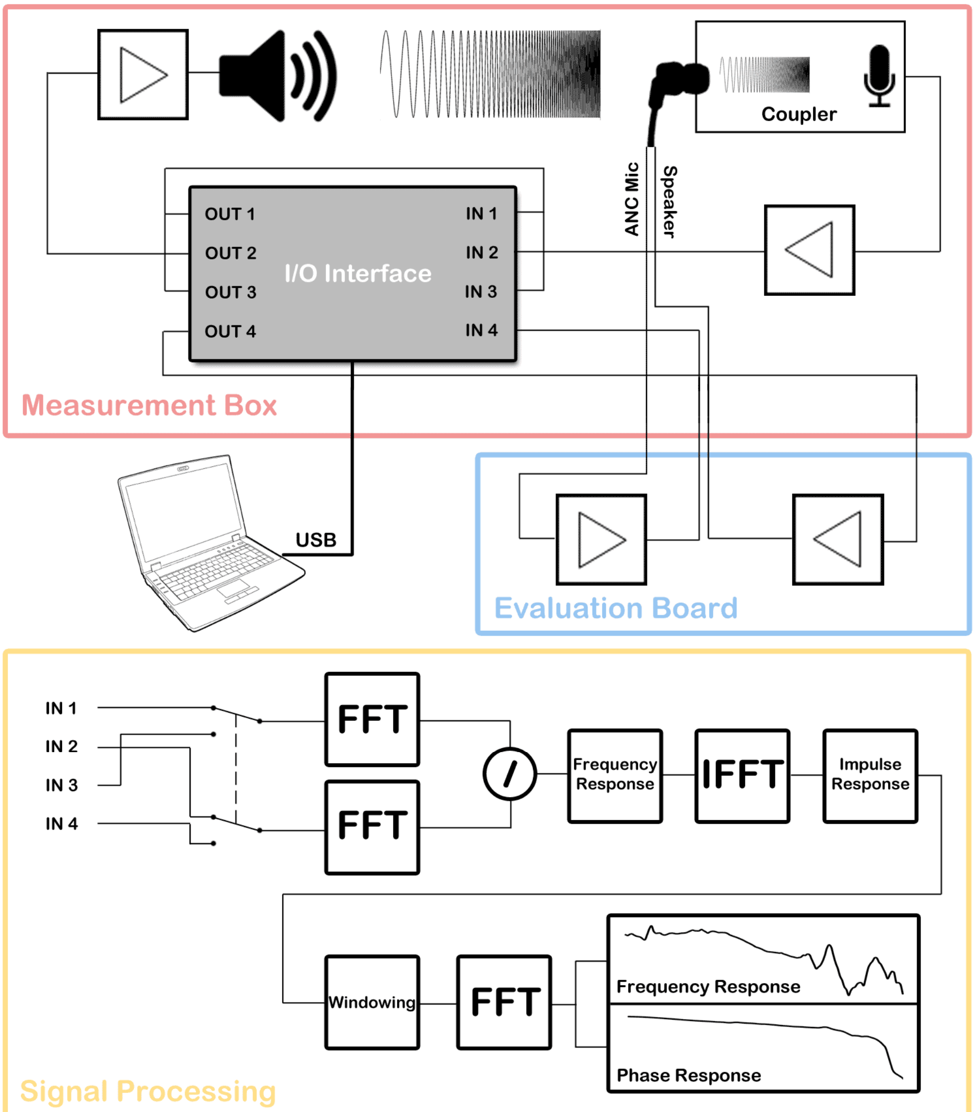



Fig. 5.1 Flowchart, complete measurement system in use with FCT.exe

# 5.1 Hardware

## 5.1.1 The measurement box

As described in 3.5, the box allows a location-independent characterization of IEMs. This behavior is a result of the requirements asked for at the beginning of this thesis. The big points in this case are especially the dimensions of the resulting system and the compact combination of all required parts in one system. This allows the user to work independently. The only additional equipment the user needs is a computer with USB connection and the IEMs, including a matched evaluation board to configure the headphones. All other required components are assembled in the box and have connectors at the backside of the characterization box.

During development process, the opening mechanism to place the IEM at the acoustic coupler was placed at the top of the box. But experience shows that this was not the optimal solution because the absorber material has to be moved during each IEM change. Because of the porosity of the absorber, this, unfortunately, leads to abrasion of the material after several changes. The upgraded final version of the box provides access on the side of the box. This allows changing the IEM without moving the absorber material and also supports easy change of the acoustic coupler (cf. fig. 5.2). Additionally, an anti-static rubber matte was placed at the top of the box. This supports storage placement of used evaluation boards during the measurement performance.

To avoid unwanted vibrations, 8 rubber feet were placed at the bottom of the box. The reason for shifting handles on the box was in box's balance point. Based on the collateral position of the amplifier and the soundcard, the balance point of the system is shifted sideways. To work against a tilt-over during positioning, the handles were also placed in the balance-point.

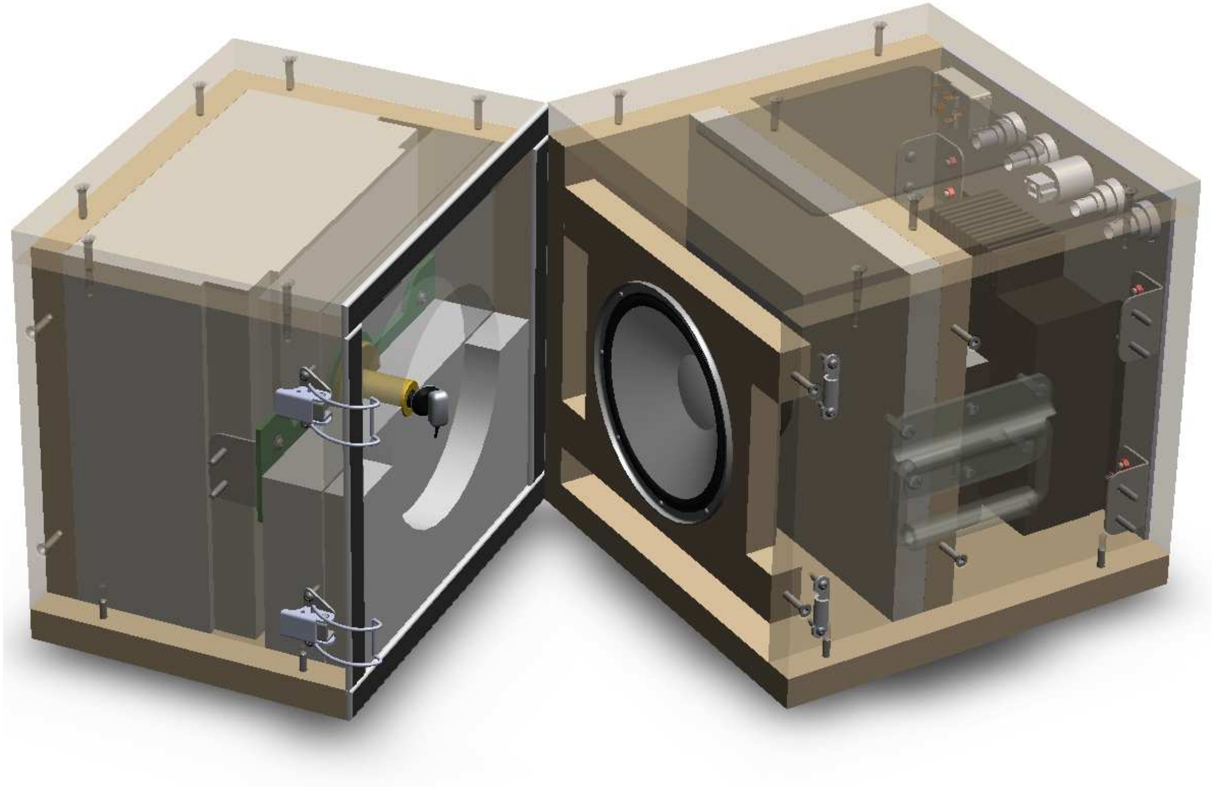


Fig. 5.2 3D Model opened characterization box

### 5.1.2 The acoustic couplers

During the evaluation measurements for the characterization system, many different IEMs run through the developed system. Because of the difference in the dimensions of the ear tips, sometimes the IEMs fall out of the coupler because the size of the tip does not match 100% with the acoustic coupler pick-up. Thus, 3 different acoustic couplers were developed, to guarantee the support for all well-established IEM models. A variety in the pick-up modeling seems to be the only way to adapt the coupler with different IEMs and safe them against dropping out. Design considerations for fixing the IEM from the backside with pressure or similar systems could not be used as this would effect the IEM's vibration properties. Furthermore, the ANC microphone is also placed on the backside of the IEM and does not leave any space for any kind of fixing construction. In addition to the first prototypes as mentioned in chapter 3.4, all finalized types have a second drill hole to lock the 0.5 mm conductor for the required pressure vent.

It became evident that acoustic coupler 2 supports 90 % of tested IEMs. But some of the IEMs use different materials for the ear tips. Some of these silicon-composites are more slippery. For these types, the cone at the pick-up is not useful. Therefore, the type-3 coupler is much better. The advantage of the type-3 coupler is the small cavity-circle at the entrance of the pick-up. During insertion of the IEM, the cavity-circle produces a little vacuum and prevents the IEM from sticking out. The type-1 coupler is dedicated to IEMs with bigger ear tips or slightly alternating design.

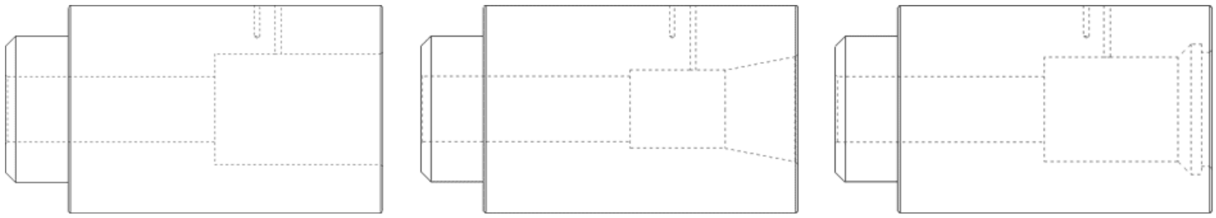


Fig. 5.3 Acoustic Couplers: Type1 (L), Type 2 (M), Type 3 (R)

### 5.1.3 The mounting plate

To fix the coupler in the measurement system, a plate of brass was designed (fig. 5.4). The mounting plate gives the possibility to change the 3 couplers if necessary. The connection between the coupler and the plate is a M14 x 1.5 ISO-metric fine pitch thread. Furthermore, it is possible change this without disassembling the complete measurement system (cf. Appendix A – User Manual). The plate is fixed with 3 screws directly on the front end of the PCB, which is where the pre-amp circuit is placed. Additionally, the mounting plate – coupler construction was prepared with 5 mm of absorber material on the backside, which leads to a separation of the acoustic path between Coupler and PCB. The connector for the measurement microphone, which is inside the coupler, is placed on the backside of the PCB. A concentric drill inside the plate and the PCB allows the cable connection between pre-amp and microphone.

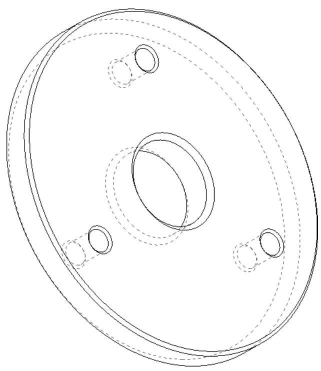


Fig. 5.4 Mounting plate for the acoustic coupler

### 5.1.4 The Absorber material

To reduce reflections and minimize on reverberation time, the measurement chamber was filled with absorber material. This also leads to good general attenuation of the characterization system from inside-out, as well as from outside-in (cf. 3.7.1). To enhance the frequency response of the speaker used and reduce unwanted resonances, the absorber material was also used at the speaker cabinet in conjunction with some loose sound sponge (c.f. 3.5.5)

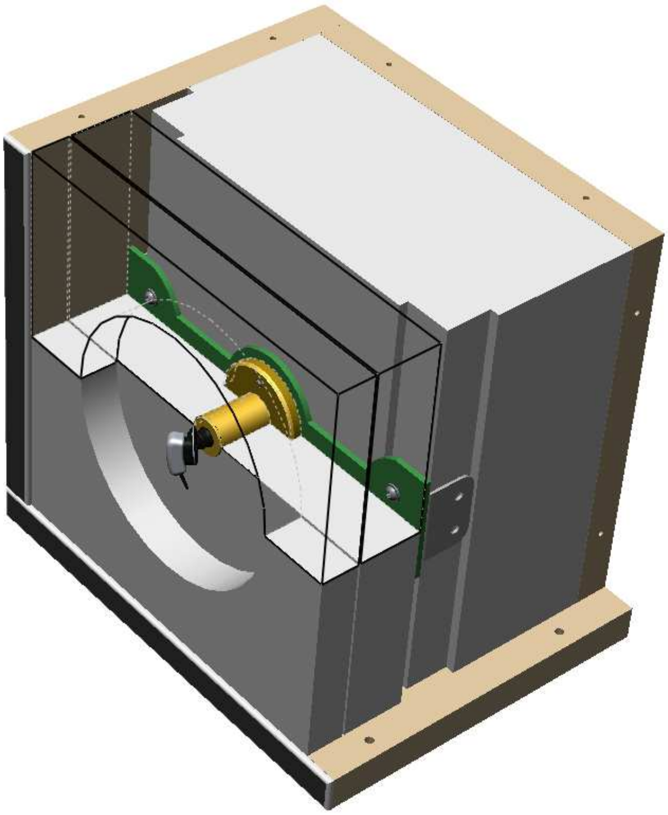


Fig. 5.5 Attenuation of the measurement chamber

# 5.2 Electronics

Fig. 5.6 illustrates a half-transparent view of the measurement box. It shows the electronic parts and the signal path inside the box. It starts at the right bottom with the green cross-styled PCB with the microphone pre-amplifier. The acoustic coupler was directly mounted on the circuit board. The IEM was mounted in the coupler. In front of it, centered, the speaker was placed, to support the excitation signal. Behind the speaker cabinet is a chamber which includes all necessary electronic parts like soundcard, amplifier, power supply and the whole wiring. At least on the backside the connectors for the outboard equipment like evaluation board, lap top including measurement program or external measurement systems was placed. A more detailed description measurement system can be found in Appendix A – User Manual.

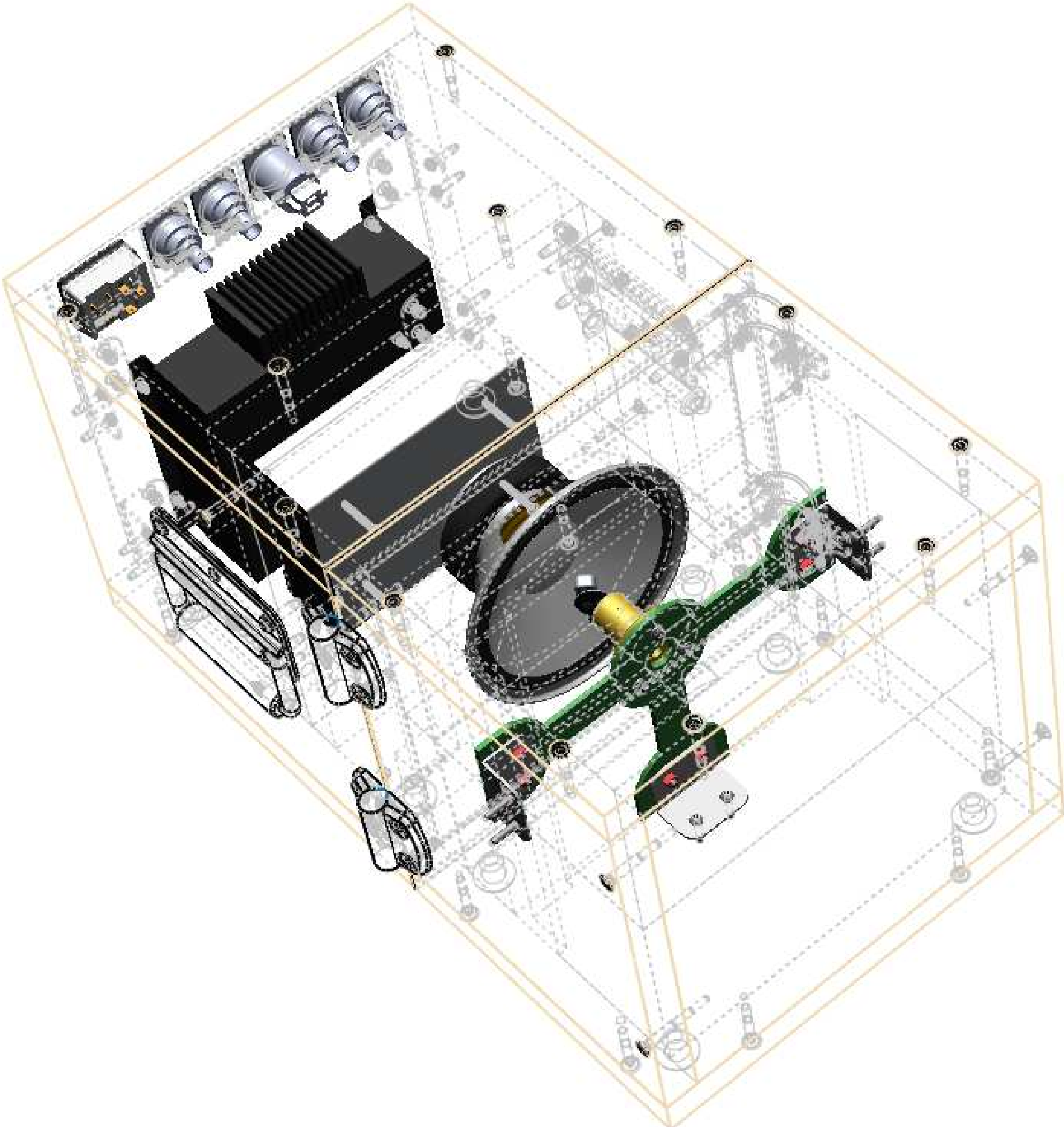


Fig. 5.6 3D Model - signal path, electric components

## 5.2.1 The Printed Circuit Board

As mentioned in chapter 3.2.3.2, the PCB was designed in a way to avoid reflections from the speaker during the performance of the measurement. This leads to the cross-style design the PCB displays in the end. The thickness of the board was chosen at 3 mm, this should prevent oscillation effects caused by the sound pressure during the measurement. Additionally, some rubber discs between PCB and the mounting angles should prevent vibration transfer via hardware parts.

The circuit for the pre-amplifier for the measurement microphone was implemented directly on the backside of the PCB (see also fig. 3.3). The connector for the acoustic coupler was also mounted there. Thus, it is possible to easily change between the 3 coupler types. Fig. 5.7 shows the PCB with the mounted acoustic coupler.

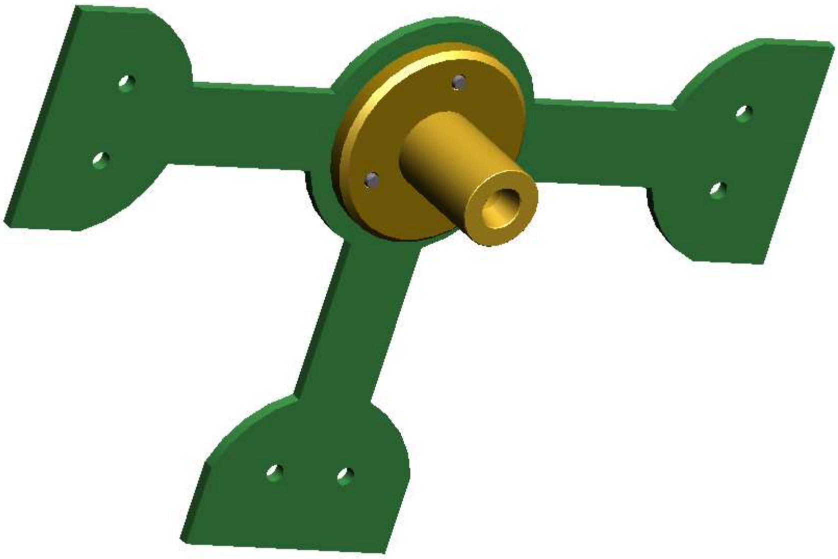


Fig. 5.7 PCB with mounted acoustic coupler

## 5.2.2 Amplifier and Soundcard

The supply for the speaker was accomplished using the the t-amp PM 40C, a 40 watts single channel power amplifier. Criteria for opting for this amplifier were in its manageable size, adequate power in relation to the speaker and the attribute of producing not too much heat while working. However, the amplifier was mounted at the backside of the aluminum plate to allow heat transmission in case of extremely long duty periods.

As I/O interface, a soundcard produced by the company Native Instruments called Traktor 6 was used. The advantages of this interface lie, in the first instance, also in size, but also simple handling and a stable service during operations. Other required features are sample rates up to 96 kHz and a resolution up to 24 Bit with low latency. An additional required feature is power supply via USB connection.

## 5.2.3 Speaker and Microphone

The fixed built-in speaker for the characterization system is the same as used during most of the development processes, a 5” 40 watts full range speaker with 8 ohms impedance, model type Ciare HX135. The reason for choosing this speaker was mentioned in 3.5.8.

The measurement microphone is the Panasonic WM-61A, an omni-directional back electret condenser microphone cartridge. This microphone is very good in its price-performance properties. Further considerations were made at 3.2.2.2.

## 5.2.4 The Connectors

All necessary connectors are mounted on the backside of the characterization box. The 3 connectors that are necessary for performing the measurement as required are placed at the left side of the plate and were marked with a black rectangle. Tab. 5.1 shows the signal paths of the mentioned connectors. On the right side of the plate are 2 additional connectors. These connectors allow for getting in touch with the direct signal from the measurement microphone and/or supports access directly to the speaker in the box (Tab. 5.1, gray highlighted).

| Connector | Signal Path |
|---|---|
| **Headphone Speaker** | Input signal to IEM speaker |
| **Headphone Mic** | Output Signal IEM microphone (ANC) |
| **USB** | Connection to the computer (Measurement software) |
| **Direct IN (Speaker)** | Direct connection to internal Speaker |
| **Direct OUT** | Pre-amplified signal from measurement microphone |

Tab. 5.1 I/O Connectors, Measurement Box

To support all system peripherals with the required power, a 250 volts power entry module was placed at the right top side of the back plate. The power entry is equipped with a 250 V fuse to protect the built-in equipment against overvoltage. Additionally, the power entry is connected to a Power on/off - switch. The switch is lighted during running mode and controls the power supply for the whole characterization system.

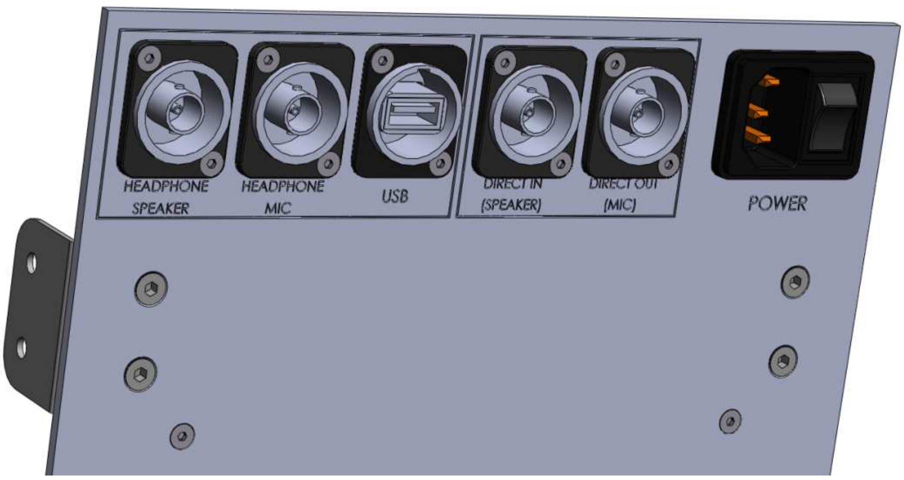



Fig. 5.8 Connector panel

# 6 Summary

The developed measurement system has to prove that a low-cost desktop ANC characterization system able to produce accurate results can be designed. It could be shown that the results of the developed system are in line with the results achieved during measurements with the “big system”. Positive behavior of the measurement box is about good attenuation which leads to negligible reverberation time and good noise separation from the outside. The employed FFT-based sweep measurement enables near-undisturbed measurement performance without any non-linearity caused by the speaker or the measurement microphone. The self-developed microphone pre-amp guarantees an ideal and constant supply for the measurement microphone, which rather helps with consistency of all performed measurements. Three different acoustic couplers support all common IEM types of different sizes. The design of the measurement box, especially the all-in-one concept, which includes all necessary hardware parts in one box, allows for easy handling. Therefore, the measurement system is easy to reproduce, which was also asked for at the beginning of the thesis.

Additionally, the developed system provides the opportunity of working more efficiently under real service conditions. On the one hand, this advantage is founded in the dimensions and the construction behavior of the characterization box and, on the other hand, on the tailor-made user interface (GUI) which was especially designed for users without any experience in acoustics or signal processing. A custom-made manual guides the user through the installation and measurement process and ensures the correct set-up process of the system.

The combination of the characterization system and the ANC performance measurement tool allows the user to directly prove the implemented filter for the DUT without changing the measurement setup.

Finally, the complete measurement system was built based on the aspect of low-cost design and ends up at a production fee of about 1,500 euros. This should also make it possible for smaller companies to perform ANC characterization autonomously.

During the end of the development phase of this measurement system, it has turned out that several companies had asked for the availability of the measurement system because of the very good performance and the mentioned above good price. This had led to the decision of producing a small batch series of the measurement system to satisfy the market demand, still before the finishing this thesis. The satisfaction of the company austriamicrosystems AG about the great success of the developed product let some space for the further development of the measurement system as mentioned in the section Outlook below. Furthermore, for all involved companies, it should keep open the possibilities for future cooperation.

## 6.1 Outlook

There are some parts of the measurement system development of which would go beyond the scope of this thesis. But development of these parts is still in progress and will be carried out in co-operation with austriamicrosystems AG outside the context of this thesis. The main open item on our plate is about enhancement of the software especially for chip designers. This means access to essential acoustic parameters of the measurement software which were – intentionally - not implemented at this stage of the system to avoid errors caused by not improper user handling. Additionally, a further development of the acoustic coupler to enhance the application area for other headphone types has been scheduled. Another possible enhancement will be the fully automatically gained adjustment feature for characterized ANC headphones. This would mark the last calibration step of the headphone before entry into the maturity phase. Including these mentioned additional features would make it possible to support the headphone - ANC related - through the entire development process, from design right up to serial production.

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# 8 Appendix A – User Manual

# ANC CHARACTERIZATION TOOLKIT

---

# USER MANUAL

austriamicrosystems

# Important Safety Instructions

- Read the manual carefully.
- Heed all warnings.
- Follow the instructions.
- Do not use the characterization box near water.
- Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus producing heat.
- Protect the power cable and all signal cables from being walked on or pinched, particularly at plugs, convenience receptacles, and the point they exit from the apparatus.
- Unplug the characterization box during lightning storms or when unused for long periods of time.
- Never break off the ground pin.
- This electrical apparatus should not be exposed to dripping or splashing and care should be taken not to place objects containing liquids, such as vases, upon the apparatus.
- Do not perform a measurement with the opened box. During the measurement high sound levels could be reached. Exposure to extremely high noise levels may cause permanent hearing loss.
- DO NOT OPEN the speaker-sided box part – risk of electric shock!
- Turn off the power switch when not in use

**Welcome to the ANC CHARACTERIZATION TOOLKIT including software!**

With this product you are able to perform a complete characterization of your feed-forward ANC headphones, including evaluation of the ANC-performance. The characterization toolkit enables you to perform all measurements simply on your desktop!

**The characterization toolkit includes the following parts:**

Characterization box

3 acoustic coupler types

Software (Filter Calculation Tool & ANC Performance Tool)

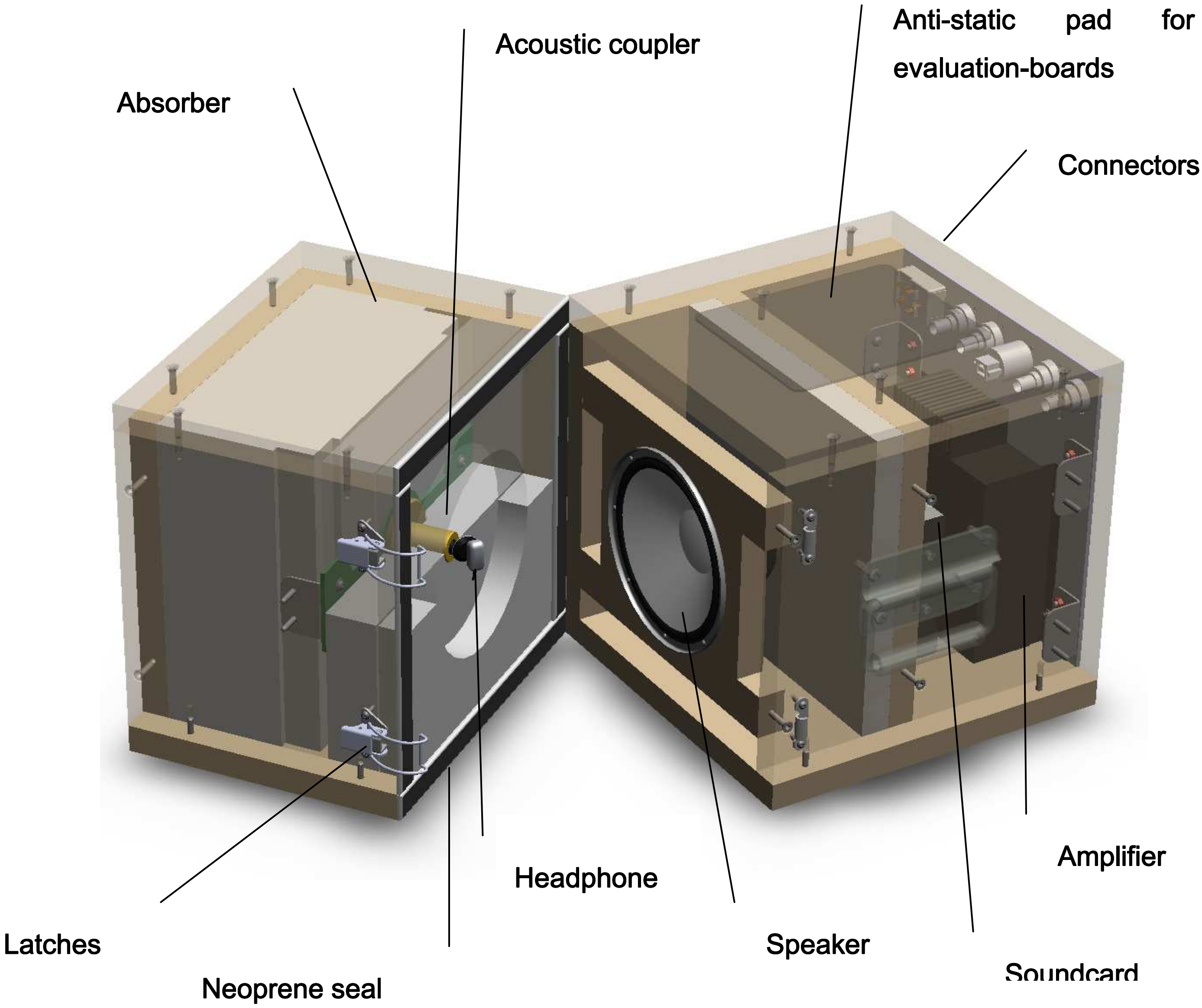
Connectors for signal paths

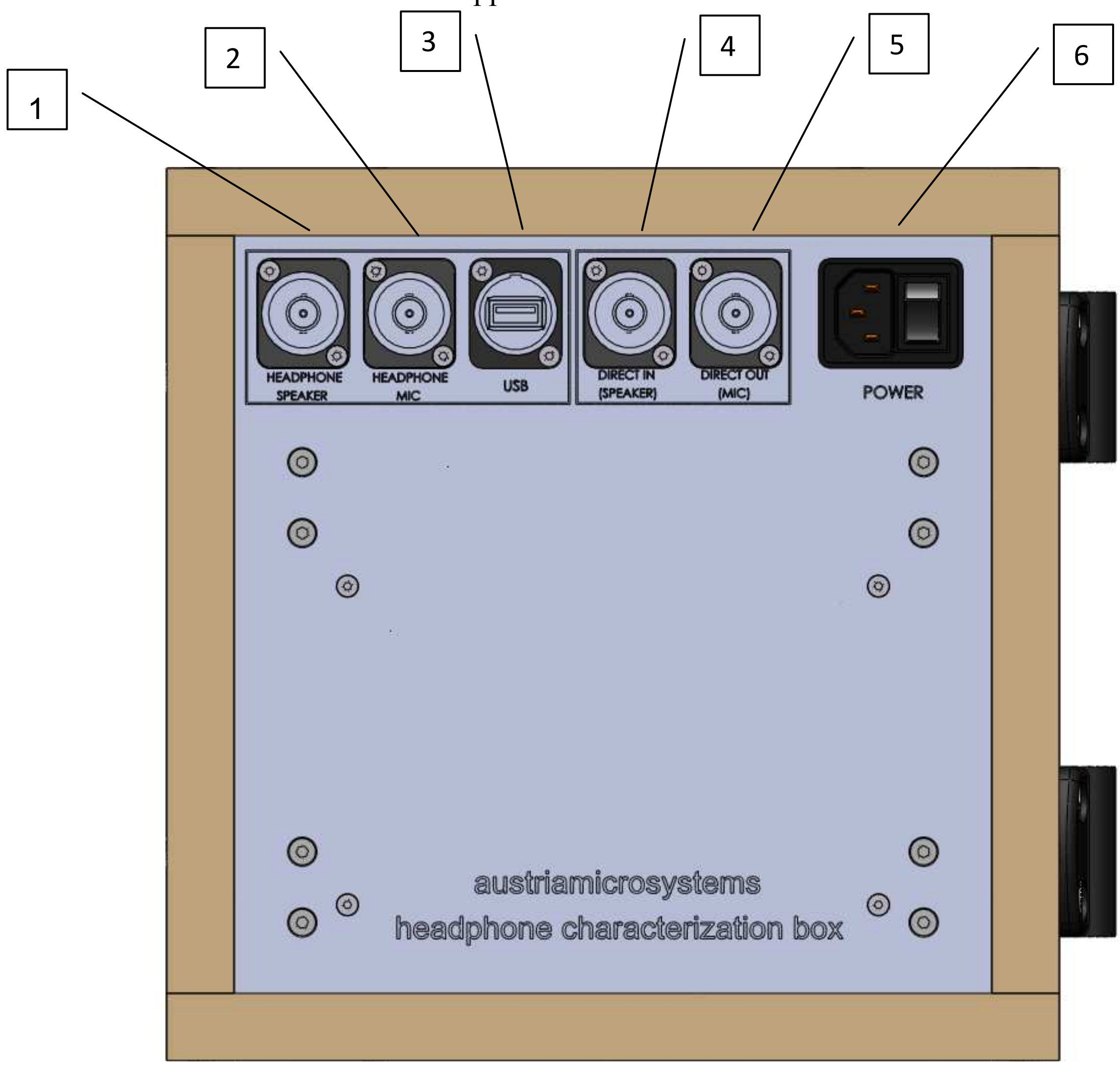
USB cable

Power cable

# Overview

This version of the characterization toolkit was specifically designed for evaluating intra-canal headphones. This limitation is due to the design structure of the provided acoustic couplers. However, the flexible design of the coupler-fixing mechanism allows for further development of coupler types to support other headphone types, as long the dimensions match with the box. The following figure illustrates the core parts of the hardware system.






**1** **Headphone Speaker**
Input signal for headphone

**2** **Headphone Mic**
Output Signal from ANC microphone

**3** **USB**
USB connection to PC

**4** **Direct In (Speaker)**
Input for direct speaker drive

**5** **Direct Out (Mic)**
Output of direct measurement microphone signal

**6** **Power**
Power supply connector with 250V fuse including On/Off-switch

# Preparing for Measurements

Make sure that the box is solidly placed with all feet of both parts on the ground. This avoids unnecessary pressure on the hinges. Open the box with the 2 latches on the front side. Squeeze the headphone with light pressure into the pickup of the acoustic coupler and be sure that the headphone does not stick out. Experience has shown that the tip of the headphone should rest with half of its length in the coupler. If the headphone does not solidly rest in the coupler, try one of the other coupler types. The connection cables of the headphone can pass through the opening surface. The flexible neoprene-sealing allows soundproof closing of the system without damaging the signal-cable from the headphone. Close the box and watch for the headphone cables not to be stretched. This could pull the headphone off the coupler and corrupt the measurement.

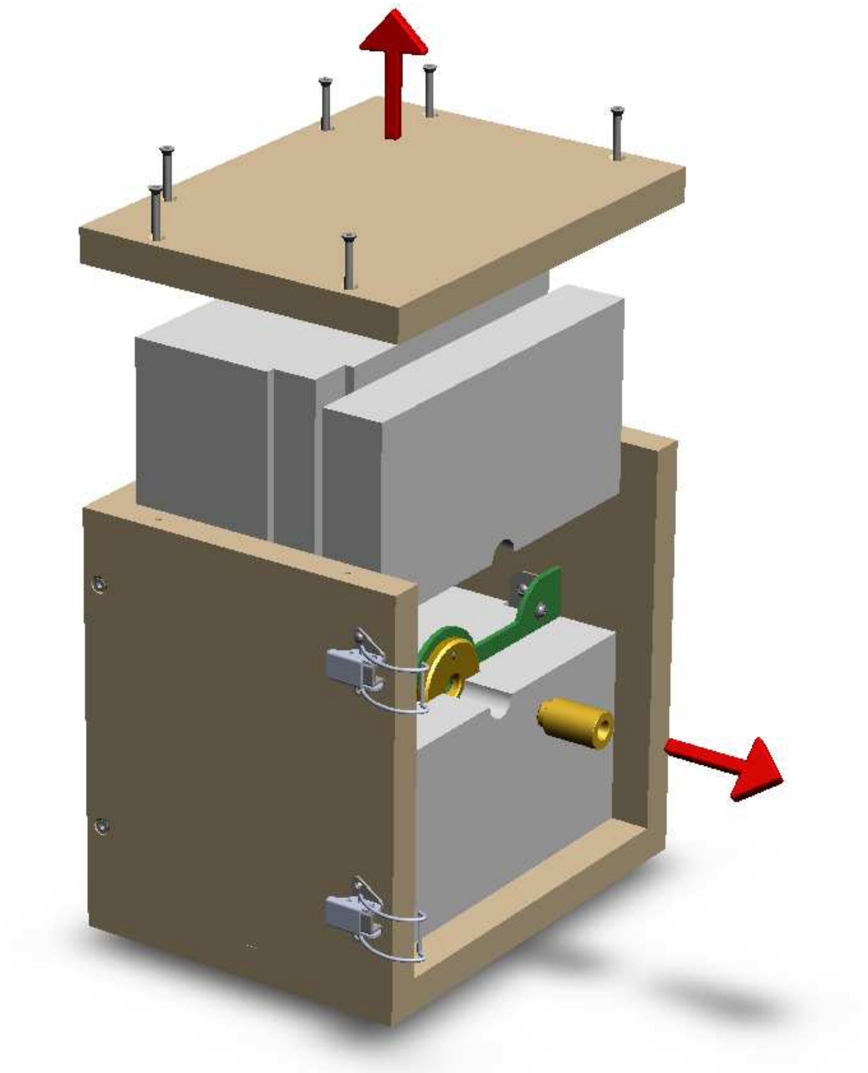
**Note: Do not lift the box when open – The hinges could come off!**

# Changing the Acoustic Coupler

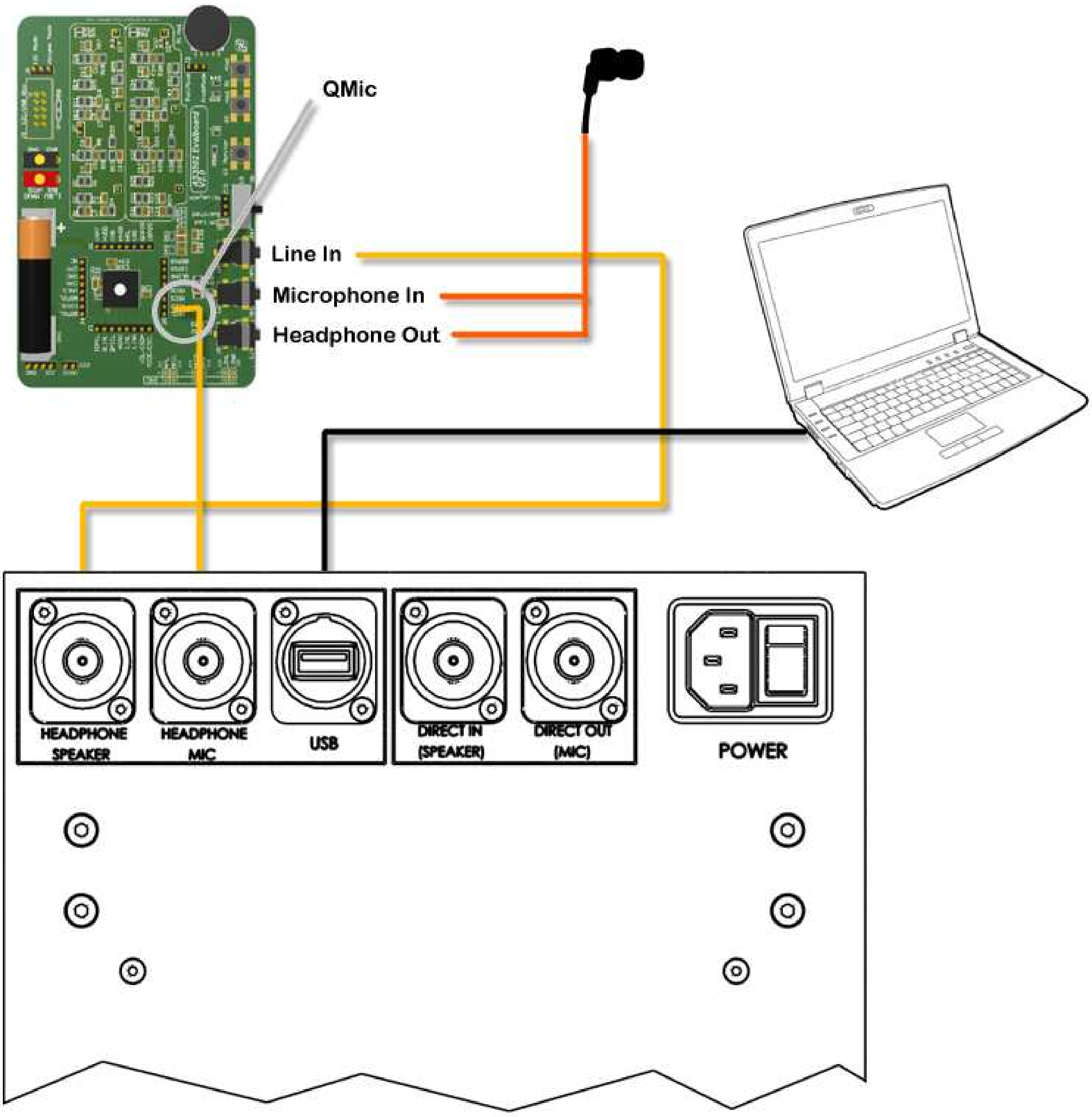
If the headphones used do not match with the built-in acoustic coupler, the coupler can be changed. The changing mechanism is quite simple. You just have to screw out the coupler from the cavity which is mounted on the PCB in the Box.

**Steps to change the coupler:**

1. Open the top plate of the measurement chamber (6 torx-screws)
2. Take out the top plates of the absorber material
3. Disconnect the microphone cable on the backside of the PCB
4. Screw out the coupler
5. Screw in the new coupler and connect it with the PCB
6. Put in the absorber material and close the cab



For final preparations for measurement, the characterization system has to be connected to the headphones, the evaluation board and the PC. Put the pre-prepared microphone and speaker connectors of the ANC-headphones in the associated jacks on your evaluation board. This provides the signal to the headphones speaker and supports the connection from the ANC microphone of the headphones to the evaluation board. To deliver the sweep signal to the headphone speaker, connect the **Headphone Speaker** jack of the box with the **Line In** jack on the evaluation board. To receive the ANC microphone signal, you have to connect the **Headphone Mic** jack with the **QMic-pins** on the evaluation board. This represents the pre-amplified output of the ANC microphone. The figure below should clarify these approaches. To connect the PC, simply plug a standard USB cable between PC and the **USB** - jack at on characterization box.




The jacks **Direct In (Speaker)** and **Direct Out (Mic)** provide direct access to the built-in speaker and the measurement microphone. The speaker signal has to be a line signal because the speaker input points at the amplifier. The microphone output provides a pre-amplified signal, too.

# The Software

**PC Minimum System Requirements:**

Windows® XP/Windows Vista®/Windows® 7 (Updated with the latest Service Packs)

2.2 GHz Pentium® IV or equivalent AMD Athlon® processor

1 GB available HD space

1 GB RAM

USB 2.0 Connector

1024 x 768 minimum display resolution

**Required software:**

Traktor Audio 6 Setup PC.exe (Audio interface driver)

MCRInstaller.exe (Matlab® component runtime installer)

AS34x0_EvalSW.exe or similar (Evaluation board software)

ezusb.sys & ezusbw2k.inf (USB interface driver – Evaluation board)

FCT.exe and APT.exe

**Optional requirements:**

Microsoft Excel

Adobe Acrobat Reader

## Installation

Before connecting the characterization box to the PC, above-mentioned required software parts have to be installed. After successful installation, the box can be connected via USB cable to the PC. Turn on the power switch of the box - it will automatically initiate a standard windows hardware dialog on your PC. Follow instructions.

## Windows configurations

During all measurements, the windows system volumes of all involved controllers have to be adjusted to maximum level. This is the only possible way of guaranteeing correct sound levels during measurements. Go to windows systems control and check the audio levels before performing your measurements!

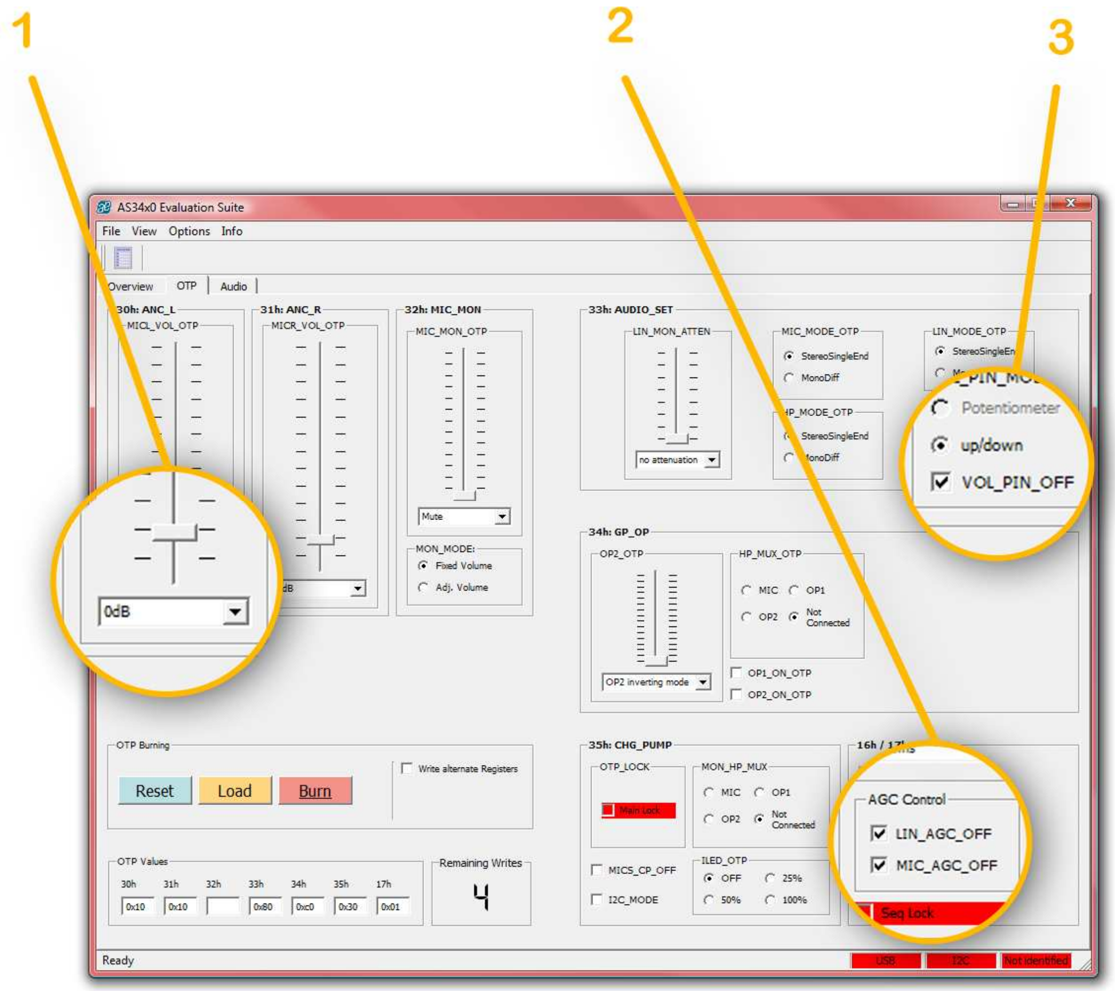
## Evaluation board configurations

A delicate part of ANC characterization is the sound level during measurement. If the level is too low, the SNR is bad. On the other hand, too high a sound level may produce non-linear distortions caused by the speaker and/or the microphone. All those effects result in inaccurate filter characterization. To avoid such a scenario, it is important to follow the proposed level adjustments.

[1] The ANC microphone gain should be 0 dB. Make sure to match the left gain with the left headphone and vice-versa!

[2] The AGC control has to be turned off. Otherwise, the internal compressors take effect and may influence the frequency response of the line signal and the ANC microphone signal!

[3] The volume pin should be turned off. This guarantees a constant level even if the volume pin is turned to different positions!

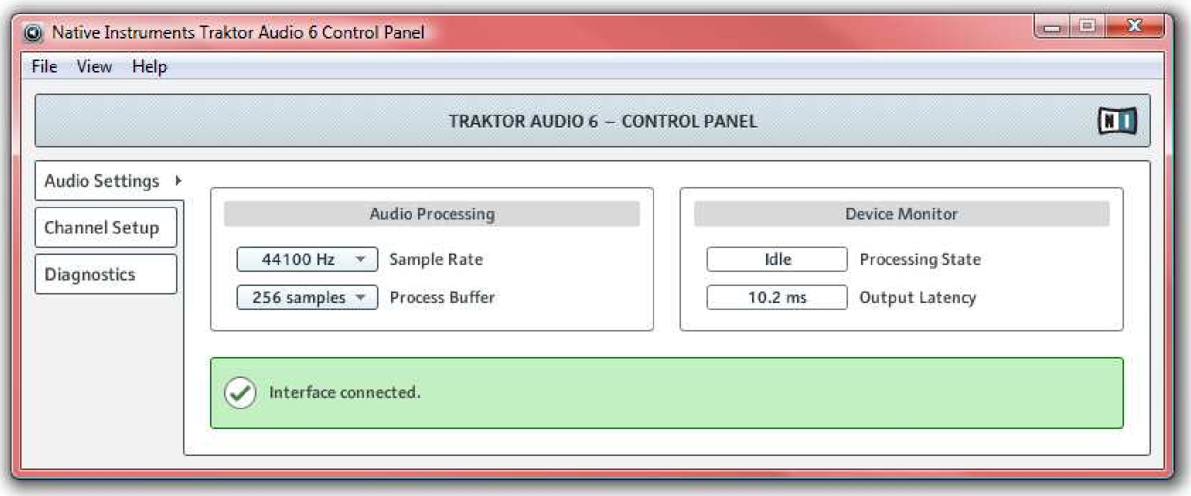


## Soundcard configurations

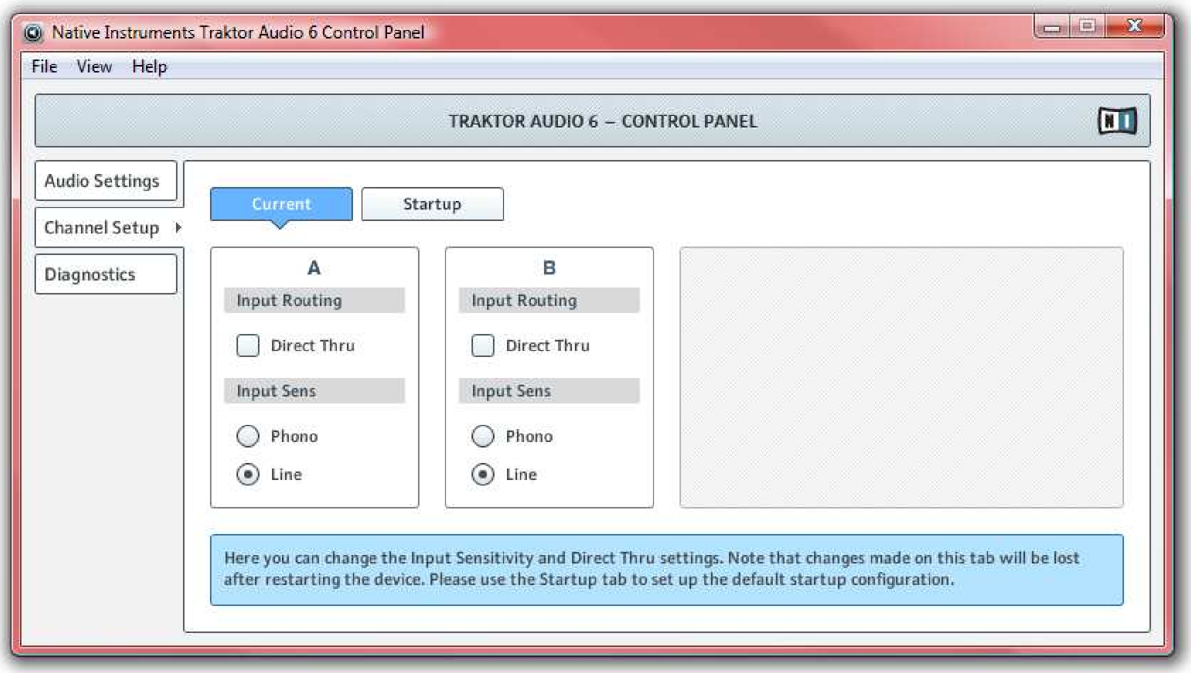
Typically, the soundcard will be delivered pre-configured. This includes all adjustments needed to run the measurement procedure without any problems. If this does not apply, you have to configure the soundcard manually in the following way:

Step 1 – Open the Traktor 6 control panel

Step 2 – Change the sample rate and process buffer size to the values below:

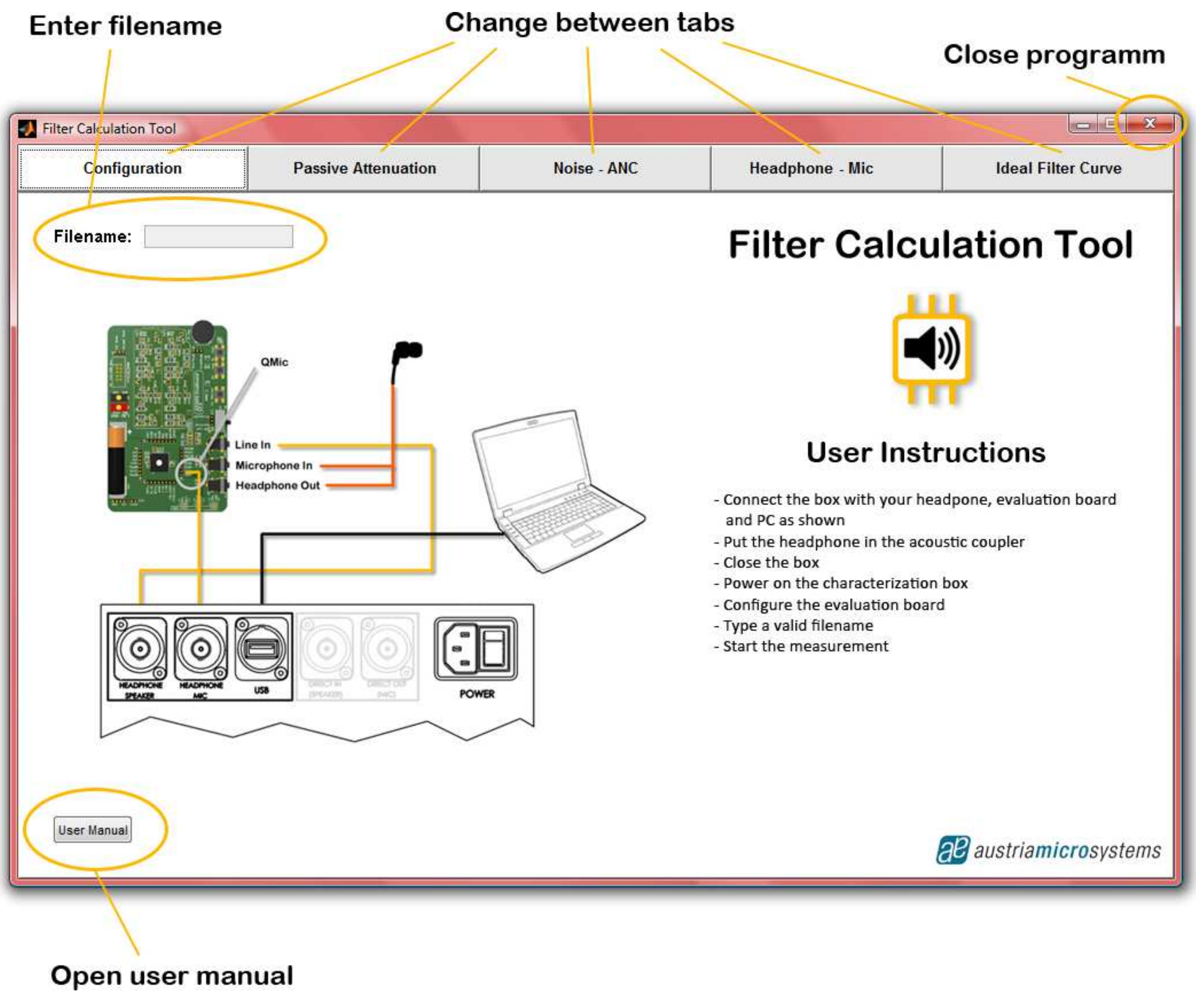



Step 3 – Configure the channel setup as shown below:



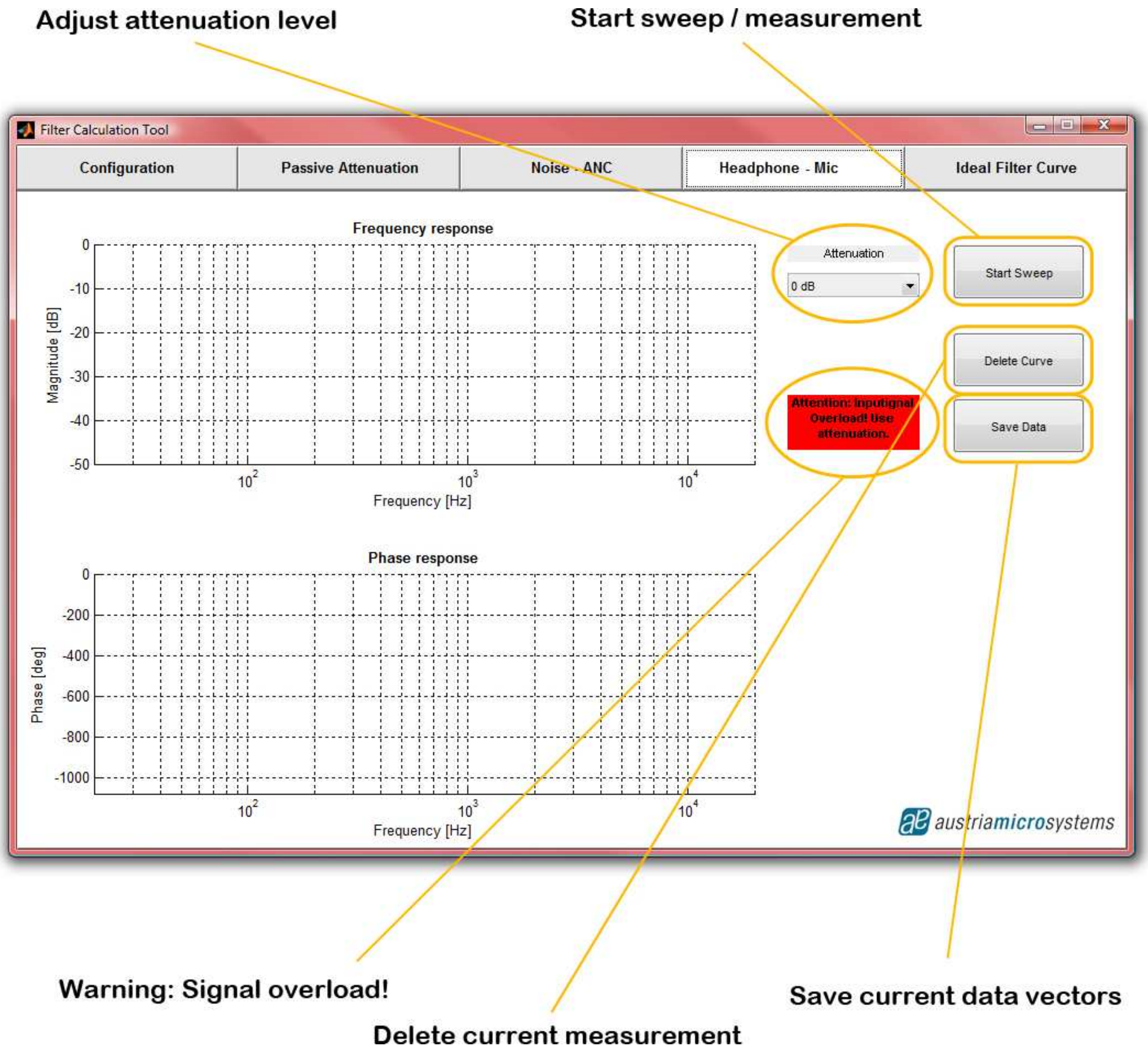
## Filter Calculation Tool

Use the *FCT.exe* for the characterization of your ANC - headphones. The program outputs are the filter gain and phase values for the DUT. After opening the *FCT.exe*, an MS DOS window opens and initializes the program. The **Configuration** tab also serves as welcome page. This tab includes the option of switching between other tabs, entering a filename, opening the user manual and taking notice of how to wire the measurement system. Also, a short description of the main functions and instructions is displayed.



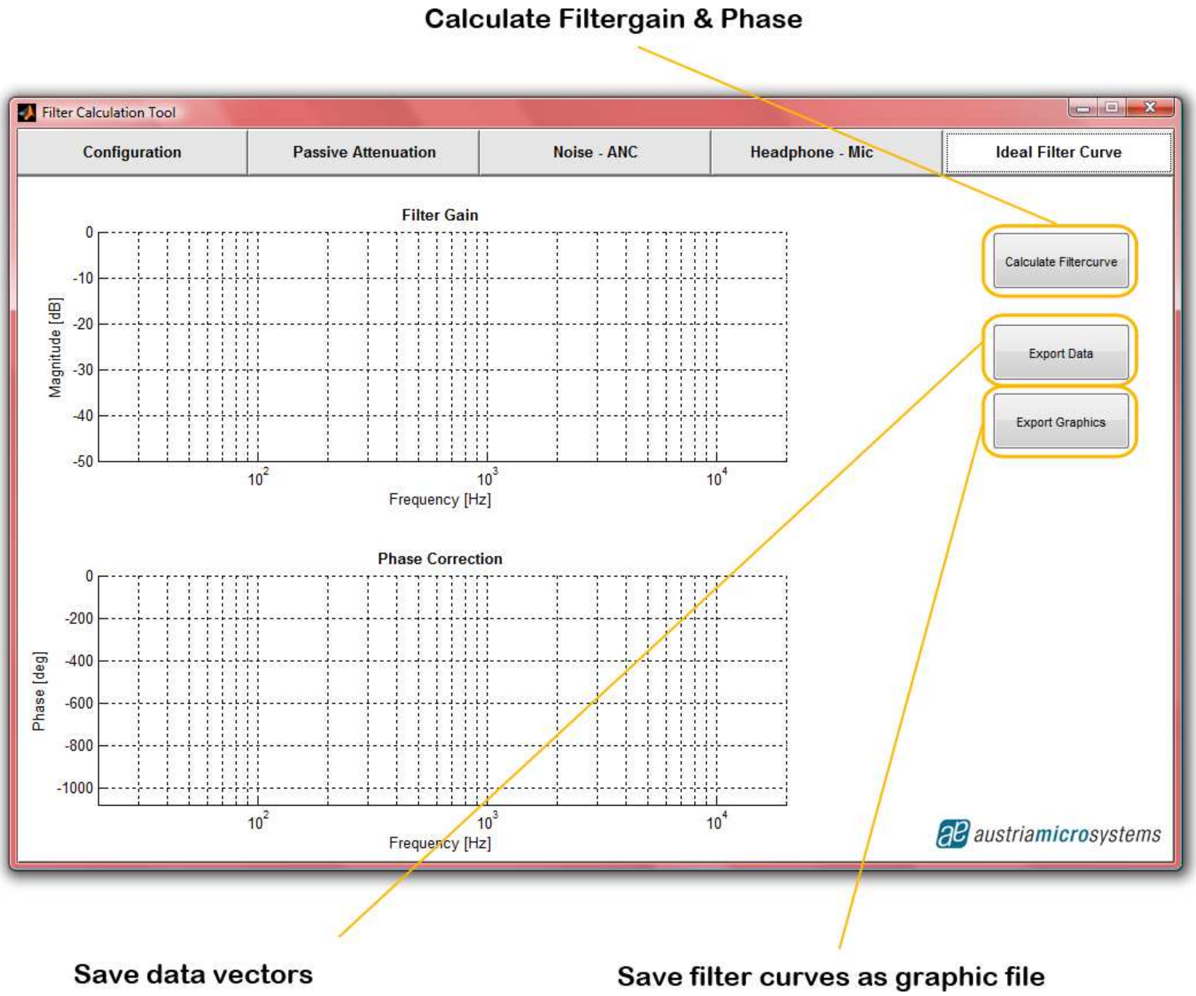
## Passive Attenuation, Noise - ANC and Headphone – Mic

The 3 mentioned tabs work similarly. You may start the sweep-measurement, save the data or delete the current measurement and repeat it, if necessary. Additionally, the Headphone – Mic measurement has an overload warning implemented. This informs you on the headphone signal level being too high. Thus, you are able to adjust the output level of the headphone with the attenuation drop – down box in 3 levels (0 dB, -10 dB and -20 dB). The measurement data can be exported as .xls or .csv file format. The program also includes the property of a zoom function for all displayed figures, for viewing details of the resulting curves.



The **Ideal Filter Curve** tab allows you to calculate the ideal filter curve for the DUT when all 3 measurements have been performed. In case of any missing or corrupted measurement, an error dialog pops up and suggests retrying the step in question. There are also two options for exporting the calculated data via *.xls* or *.csv* file. The data, similar to the other tabs, will be saved as amplitude and phase vectors in correlation to the corresponding frequency values. The graphic export saves the calculated figures and supports the following graphic formats:

- *.emf*
- *.jpg*
- *.tif*
- *.png*



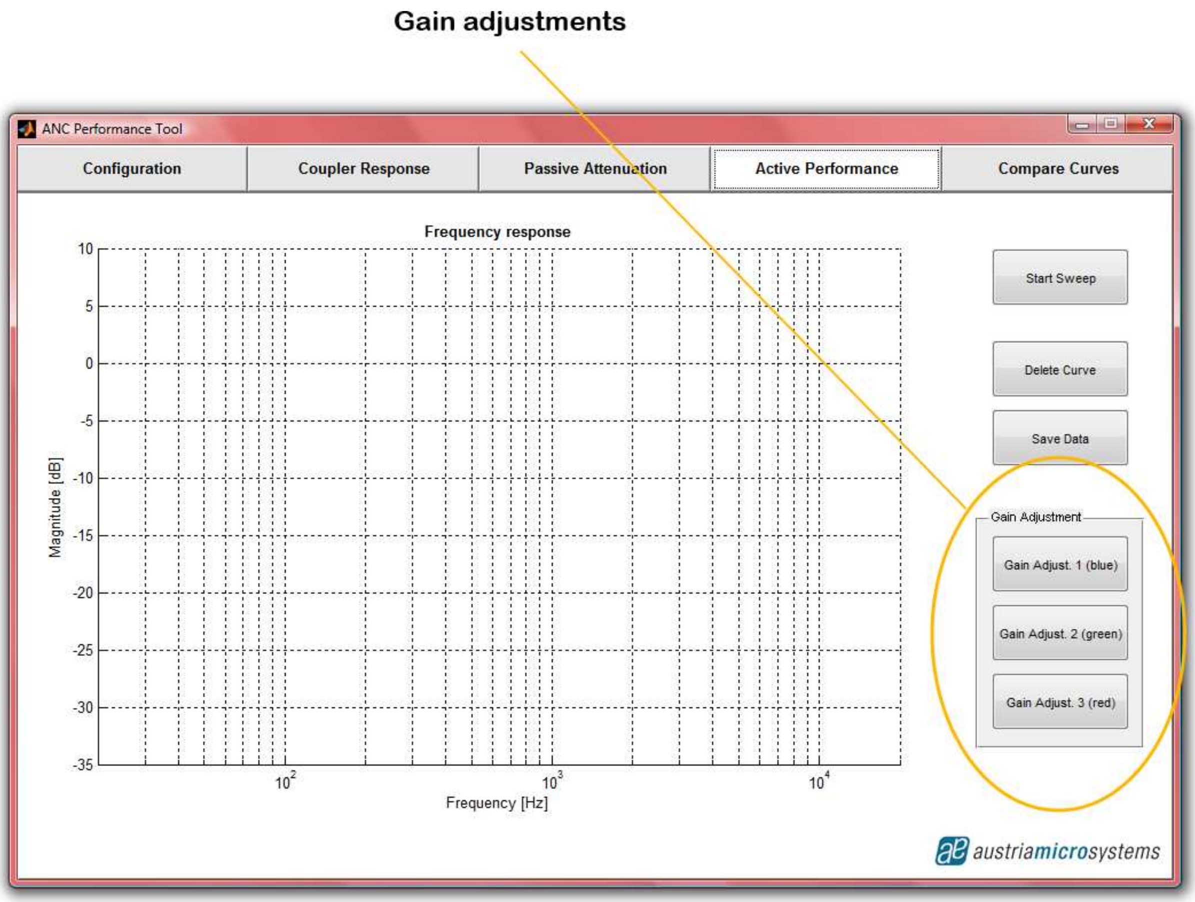
## ANC Performance Tool

Use the *ANC.exe* for evaluating the performance of your ANC - headphones. The **configuration** tab works similar to the *FCT.exe*. With *Start Sweep, Delete Curve* and *Save Data,* the **Coupler Response and Passive Attenuation** tabs render the same options as *FCT.exe* mentioned on page 14.

The **Coupler Response** tab measures the frequency response of the empty acoustic coupler. This frequency response is needed for further calculations regarding ANC performance. **For this measurement, it is important to remove the headphone from the acoustic coupler!!**

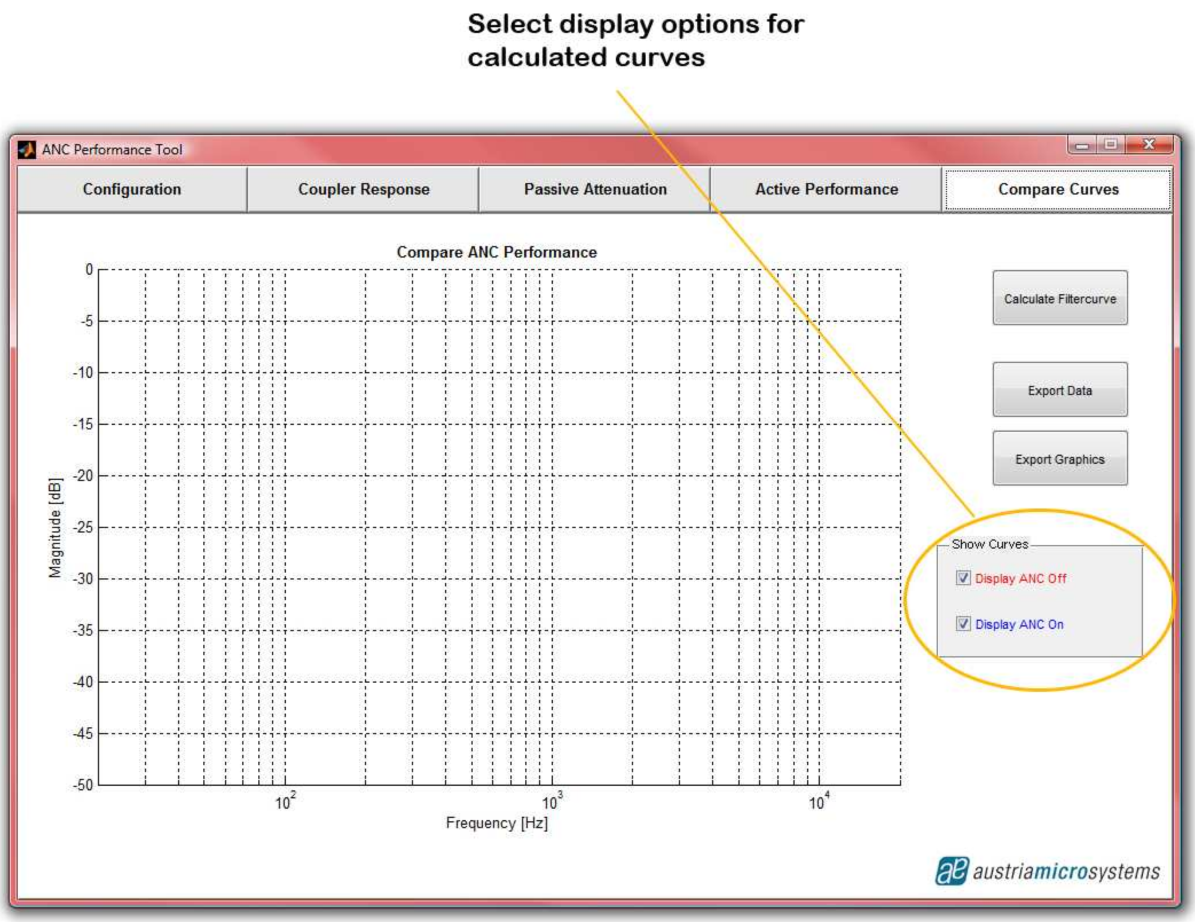
For the **Passive Attenuation** and **Active performance** measurements the headphone has to be put into the coupler again.

The goal of the measurement tab **Active performance** is in visualizing pure ANC performance - this means without the passive attenuation effect caused by the headphone. To achieve this result, it is necessary **to activate the ANC system during such measurements!!**



An additional feature of this tab is the option of adjusting the filter gain for the DUT. The 3 gain adjust buttons, on the right bottom of the GUI, allow you to display several performed measurements in one figure. Each of those 3 measurements is rendered with its own color and can be used for recursive approximation to achieve the best possible filter gain for the DUT. In addition to the passive attenuation measurement, an error dialogue occurs if the necessary measurements were not completed during the start of this measurement.

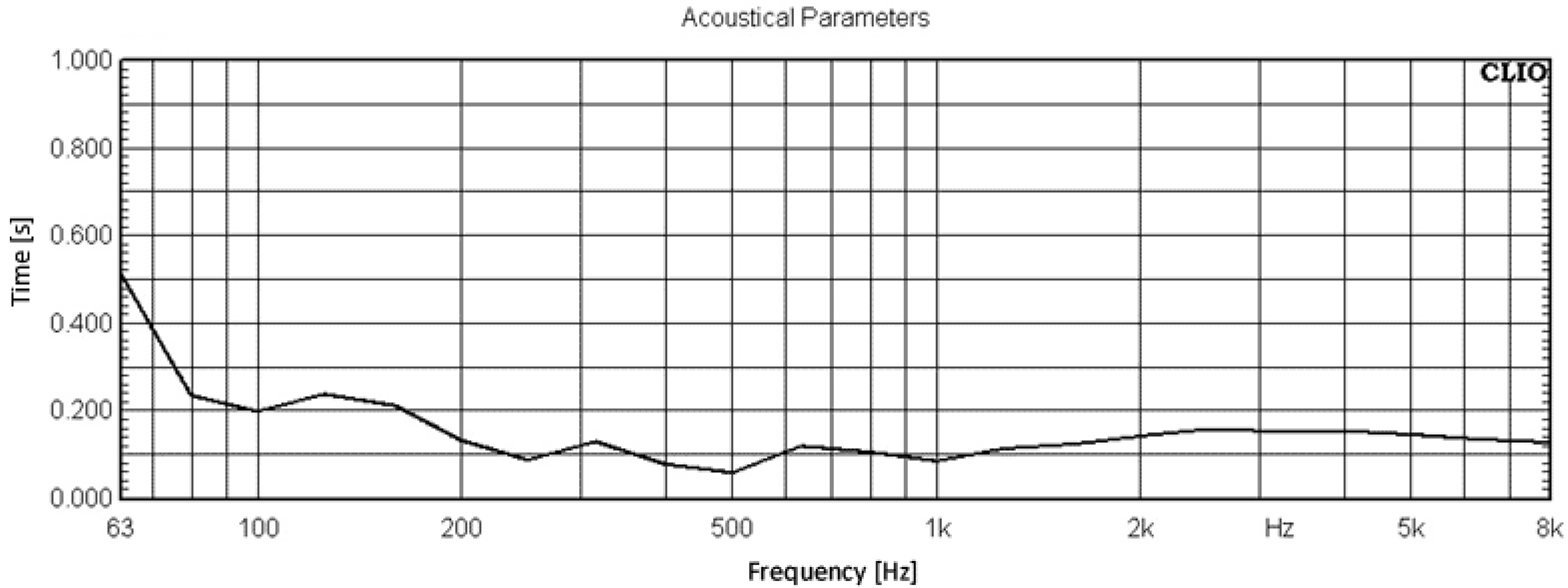
Finally, the last tab - **Compare Curves** - displays the completed ANC performance. There is the possibility of comparing the performances of the headphones between “ANC ON” and “ANC OFF”. The visibility of the resulting filter curves can be chosen via two checkboxes on the right bottom of the program. Similarly to the *FCT.exe,* data export as *.xls* or *.csv* and graphic export as *.jpg, .png, .tif or .emf* file is supported.



Appendix A – User Manual

# 9 Appendix B - Datasheets

Appendix B includes some of the most important Datasheets of hardware parts used during the development of this thesis. Because of the huge amount of different hardware used during development, a complete library of all datasheets used can be found on the enclosed CD.




| Frequency [Hz] | LIII | 63 | 80 | 100 | 125 | 160 | 200 | 250 | 315 | 400 | 500 | 630 | 800 | 1000 | 1250 | 1600 | 2000 | 2500 | 3150 | 4000 | 5000 | 6300 | 8000 | A |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Signal [dBSPL] | 75.1 | 46.8 | 45.8 | 49.9 | 50.3 | 51.4 | 54.4 | 52.9 | 53.1 | 52.5 | 59.9 | 58.3 | 56.7 | 60.9 | 60.7 | 61.8 | 62.8 | 63.4 | 64.0 | 64.7 | 64.4 | 64.4 | 65.1 | 75.1 |
| Noise [dBSPL] | 40.1 | -2.4 | 8.2 | 16.9 | 19.6 | 21.3 | 13.8 | 5.2 | 5.6 | 15.8 | 4.0 | -0.4 | 0.2 | 3.5 | 4.5 | 2.6 | 8.2 | 9.8 | 10.8 | 7.9 | 3.7 | 4.8 | 4.8 | 18.0 |
| C50 [dB] | 22.56 | -0.73 | 6.98 | 9.15 | 2.17 | 2.31 | 26.43 | 30.63 | 28.98 | 37.88 | 37.19 | 28.25 | 34.52 | 34.86 | 30.49 | 27.42 | 23.54 | 23.29 | 20.11 | 21.96 | 23.05 | 25.53 | 27.56 | 25.03 |
| C80 [dB] | 30.65 | 0.98 | 16.23 | 15.83 | 13.87 | 12.43 | 35.13 | 45.14 | 36.28 | 53.67 | 49.78 | 46.64 | 44.47 | 43.49 | 42.21 | 39.17 | 36.24 | 35.60 | 33.39 | 33.61 | 35.59 | 37.29 | 42.34 | 37.17 |
| D50 [%] | 99.4 | 45.8 | 83.3 | 89.2 | 62.2 | 63.0 | 99.8 | 99.9 | 99.9 | 100.0 | 100.0 | 99.9 | 100.0 | 100.0 | 99.9 | 99.8 | 99.6 | 99.5 | 99.0 | 99.4 | 99.5 | 99.7 | 99.8 | 99.7 |
| TS [ms] | 5.4 | 76.5 | 34.6 | 33.9 | 42.5 | 41.3 | 16.7 | 11.2 | 12.9 | 9.0 | 8.8 | 6.6 | 6.8 | 7.1 | 5.3 | 5.3 | 7.3 | 6.2 | 8.5 | 6.9 | 6.4 | 6.3 | 4.8 | 5.0 |
| EDT [s] | 0.103 | 0.880 | 0.401 | 0.365 | 0.507 | 0.477 | 0.197 | 0.121 | 0.141 | 0.103 | 0.116 | 0.064 | 0.064 | 0.079 | 0.069 | 0.089 | 0.124 | 0.092 | 0.185 | 0.153 | 0.113 | 0.120 | 0.105 | 0.105 |
| RT20 [s] | 0.159 | 0.546 | 0.242 | 0.220 | 0.228 | 0.170 | 0.095 | 0.093 | 0.079 | 0.083 | 0.049 | 0.101 | 0.095 | 0.109 | 0.137 | 0.125 | 0.139 | 0.152 | 0.155 | 0.147 | 0.147 | 0.125 | 0.114 | 0.138 |
| R (RT20) | -0.998 | -0.998 | -0.993 | -0.989 | -0.983 | -0.997 | -0.971 | -0.946 | -0.984 | -0.933 | -0.967 | -0.993 | -0.978 | -0.936 | -0.986 | -0.996 | -0.988 | -0.994 | -0.996 | -0.998 | -0.995 | -0.996 | -0.997 | -0.999 |
| RT30 [s] | 0.236 | 0.511 | 0.233 | 0.197 | 0.237 | 0.210 | 0.131 | 0.087 | 0.128 | 0.076 | 0.057 | 0.118 | 0.104 | 0.084 | 0.112 | 0.123 | 0.141 | 0.156 | 0.151 | 0.152 | 0.144 | 0.133 | 0.126 | 0.141 |
| R (RT30) | -0.973 | -0.998 | -0.992 | -0.995 | -0.992 | -0.990 | -0.971 | -0.978 | -0.959 | -0.985 | -0.982 | -0.986 | -0.992 | -0.979 | -0.986 | -0.998 | -0.996 | -0.997 | -0.998 | -0.998 | -0.997 | -0.998 | -0.996 | -0.999 |
| RTU [s] | 0.166 | 0.509 | 0.266 | 0.219 | 0.247 | 0.166 | 0.115 | 0.085 | 0.089 | 0.095 | 0.040 | 0.103 | 0.108 | 0.112 | 0.151 | 0.123 | 0.161 | 0.162 | 0.158 | 0.142 | 0.144 | 0.130 | 0.117 | 0.141 |
| R (RTU) | -0.998 | -0.999 | -0.994 | -0.980 | -0.975 | -0.994 | -0.966 | -0.930 | -0.985 | -0.902 | -0.963 | -0.991 | -0.992 | -0.909 | -0.976 | -0.991 | -0.990 | -0.995 | -0.990 | -0.996 | -0.993 | -0.992 | -0.995 | -0.999 |

Fig. 9.1 Measurement room response RT30

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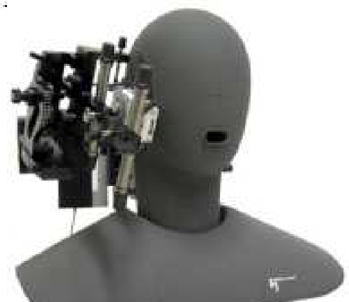
**DATENBLATT**

# HMS II.3 (Code 1230)

**HEAD Measurement System**

Kunstkopf-Messsystem mit Ohrsimulator und künstlichem Mund



## Überblick

HMS II.3 ist ein Kunstkopf-Messsystem zur Messung ohrnaher Schallquellen:

Kopfhörer, Handapparate, Headsets und Gehörschützer. Ohne Einschränkungen ermöglicht HMS II.3 darüber hinaus die Messung ohrferner Schallquellen wie beispielsweise Freisprecheinrichtungen.

HMS II.3 wurde entwickelt für akustische Messungen von Telefonendeinrichtungen und erfüllt die Forderungen der ITU-T-Empfehlungen P. 57 und P.58. Die auf die akustisch relevanten Strukturen reduzierte Nachbildung der menschlichen Anatomie ermöglicht eine Aufnahmetechnik, die den realen Einsatzbedingungen von Handapparaten und Freisprecheinrichtungen entspricht. Der Kunstkopf verfügt über einen Impedanz-Simulator im rechten Ohr und einen künstlichen Mund.

## BESCHREIBUNG

HMS II.3 eignet sich ideal für alle Messungen im Telekommunikationsbereich unter wirklichkeitsgetreuen Einsatzbedingungen. Beugungs- und Reflexionseigenschaften sind vergleichbar denen des menschlichen Zuhörers.

Das System HMS II.3 ist Aufnahme- und Sprachsimulationssystem zugleich. Es eignet sich deshalb für Messungen sowohl in Sende- als auch in Empfangsrichtung. HMS II.3 ist für Messungen an unterschiedlichen Endeinrichtungen vorbereitet, wie Handapparate, Hörgeräte oder Freisprecheinrichtungen, die auch sprachgesteuert sein können.

Als Aufnahmemedium dient das künstliche Außenohr (Pinna), das in den akustischen Eigenschaften dem menschlichen Außenohr nachempfunden ist. Innen schließt sich der Ohrsimulator nach IEC 711 an (rechtes Ohr).

Im Standlieferumfang ist HMS II.3 mit flexibler Pinna gemäß ITU-T P.57 Typ 3.4 ausgerüstet. Diese Pinna bildet neben den akustischen Eigenschaften das mechanische Verformungsverhalten der menschlichen Ohrmuschel nach. Die Verwendung dieser Pinna ist insbesondere bei Untersuchungen von Schallquellen (z.B. Handapparaten) sinnvoll, bei denen der Anpressdruck an das Ohr einen signifikanten Einfluss auf die Messung hat.

Alternativ kann bei der Erstbestellung von HMS II.3 statt der Pinna Typ 3.4 die anatomisch geformte Pinna gemäß ITU-T P.57 Typ 3.3 (auch als zusätzliche Option erhältlich) gewählt werden. Ihre Verwendung empfiehlt sich, wenn die Anatomie der menschlichen Ohrmuschel eine wichtige Rolle spielt (z.B. bei Intra-Concha Headsets).

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Das vom künstlichen Mund erzeugte, definierte Schallfeld ist in der Richtlinie ITU-T Empfehlung P.58 festgelegt. Das Freifeldübertragungsmaß des künstlichen Mundes, gemessen an den verschiedenen Positionen, entspricht ITU-T P.58. Es unterscheidet sich insbesondere im Fernfeld von den Messdaten, die ITU-T Empfehlung P.51 verlangt. Beugungen und Reflexionen an Schulter und Oberkörper, wie sie typisch auch an Versuchspersonen gemessen werden, sind die Ursache hierfür. Schulter- und Oberkörpernachbildung des HMS II.3 entsprechen in ihrem akustischen Verhalten menschlichen Versuchspersonen.

In seinen geometrischen Abmessungen entspricht HMS II.3 allen Forderungen nach ITU-T P.58. Darüber hinaus sind wesentliche Daten der Kopfkontur in unterschiedlichen Schnittebenen festgelegt. Für alle Schnittebenen liegt der HMS II.3 innerhalb der festgelegten Toleranzbereiche.

Der Torso des HMS II.3 kann weiteres Elektronikzubehör aufnehmen. Durch die kompakte Bauform lässt sich das gesamte System einfach handhaben und transportieren.

## SIGNALKONDITIONIERUNG

Bei der Messung von Telefonendgeräten im stationären Betrieb wird HMS II.3 über die Frontends MFE III - VIII an das Analysesystem ACQUA angeschlossen.

Darüber hinaus besteht die Möglichkeit, HMS II.3 mit dem digitalen Binauralen Equalizer BEQ II.1 (Option) auszurüsten. BEQ II.1 ist ein Messverstärker mit integrierter Signalkonditionierung für binaurale Messsysteme mit hohem Dynamikumfang. Er ermöglicht die präzise Entzerrung binauraler akustischer Signale und stellt verschiedene Entzerrungsvarianten zur Verfügung, wie sie beispielsweise in der ITU-T Empfehlung P.581 gefordert werden.

Die gemessenen Signale können direkt digital von einem Rechner oder DAT-Rekorder aufgezeichnet werden. Es ist möglich, die Aufnahmeeinstellungen mit dem Signal abzuspeichern und bei der Wiedergabe automatisch zu berücksichtigen.

Die Wiedergabe von Signalen mit dem künstlichen Mund erfolgt mit dem Messfrontend MFE VI.1 mit BEQ-Option, was neben der Signalverstärkung eine individuelle Entzerrung des Mundes ermöglicht.

Da MFE VI.1 netzunabhängig betrieben werden kann, ist HMS II.3 in Verbindung hiermit ein vollständiges, mobiles Messsystem.

## ERWEITERUNGEN

Der Kunstkopf HMS II.3 lässt sich mit dem Haltemechanismus HHP II.1/III ausstatten (Option), damit ein Handapparat in definierter und reproduzierbarer Position angebracht werden kann. Zur Kontrolle des Anpressdrucks verfügt dieser „künstliche Arm" über einen elektronischen Kraftaufnehmer.

Zur binauralen Messung von ohrnahen Schallquellen wie Kopfhörern und aktiven/passiven Gehörschutzsystemen kann das linke Ohr ebenfalls mit einem Ohrsimulator ausgestattet werden.

## ANWENDUNGEN

- Messung von Telefon-Endeinrichtungen
- Messung von Freisprecheinrichtungen jeglicher Art
- Test von aktiven und passiven Gehörschutz-Systemen industrieller Anwendung
- Test von Kopfhörern
- Messungen an Kombinationen von Gehörschutz und Kopfhörern
- Messungen an Hörsprech-Garnituren
- Einsatz in der Qualitätskontrolle, bei Sondermessaufgaben und bei Standardmessungen

## BESONDERE MERKMALE

- Entspricht ITU-T P.58
- Ohrsimulator nach IEC 711
- Künstlicher Mund nach ITU-T P.58
- Flexible Pinna entsprechend ITU-T P.57 Typ 3.4
- Anatomisch geformte Pinna gemäß ITU-T P.57 Typ 3.3 (Option)
- Handapparate-Halter HHP II.I/III (Option)
- Portables, netzunabhängiges System in Verbindung mit MFE VI.1 (Zubehör)
- Kompatibel zum menschlichen Gehör und zur konventionellen Messtechnik
- Kalibrierfähig
- Digitale Entzerrung in Verbindung mit Binauralem Equalizer BEQ II.1 (Option)
- Geringes Eigenrauschen
- Ausgezeichnetes Design

## LIEFERUMFANG

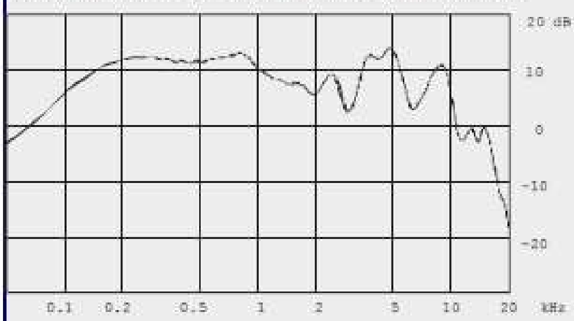
**HMS II.3 (Code 1230)**
umfasst folgende Komponenten:

- **Kunstkopf** bestehend aus:
  - **HHS II.3 (Code 1251):** Kopf- und Schulter-Nachbildung mit künstlichem Mund
  - **HER III.1 (Code 1249):** Pinna (rechts) entsprechend ITU-T P.57 Typ 3.4
    ***alternativ (bitte bei Bestellung angeben):*** *HER IV/IV.1/IV.2 (Code 1359/1372/1381): Pinna (rechts) entsprechend ITU-T P.57 Typ 3.3, anatomisch geformt*
  - **HIS R (Code 1232):** Impedanz-Simulation - rechtes Ohr
  - **ECS I.0 (Code 1350):** Ohrkanal-Simulation
- **Mikrofonhalter** mit 1/2"-Klemmadapter
- **Zubehörbox** (enthält MRP-Pointer, Lippenring, zwei verschiedene Kalibrieradapter)
- **Schraubendreher** Inbus 2,5 mm
- **Ohrkanalschlüssel**
- **Adapterkabel:** SPEAKON-Stecker männlich <-> 2x Laborstecker 4mm zur Verbindung mit MFE III
- **HTB V (Code 1374):** HEAD Torso Box für mobile Kunstkopfmessungen, erweiterbar für verschiedene Hardwarekonfigurationen
- **HSD II.2 (Code 1258):** Handbuch HMS II.3 / II.4 / II.5

## Technische Daten

| | |
|---|---|
| **Empfangsrichtung (Ohr)** | |
| Übertragungsbereich: | 3 Hz – 20 kHz |
| Dynamik: | > 110 dB |
| Frequenzgang: | Entspricht ITU-T P.58 |
| Richtcharakteristik: | Entspricht ITU-T P.58 |
| **Senderichtung (Mund)** | |
| Übertragungsbereich: | 100 Hz – 8 kHz |
| Belastungslimit: | 20 W (Sinus), 50 W (Musik) |
| Impedanz: | 4 Ω |
| Klirrfaktor: | Entspricht ITU-T P.58 |

Typischer (geglätteter) Frequenzgang des Mundes in Senderichtung:




| | |
|---|---|
| **Umgebungsbedingungen** | |
| Betriebstemperaturbereich: | 0°C - 65°C |
| Lagerungstemperaturbereich: | -40°C - 80°C |
| **Gehäuse (Kopf-Schulter-Simulation ohne Torso-Box)** | |
| Abmessung (BxHxT): | ca. 450 x 400 x 180 mm |
| Gewicht: | ca. 5,4 kg |

## OPTIONEN

- **HIS L (Code 1231):** Impedanz-Simulation - linkes Ohr, inklusive Mikrofon
- **HEL/HER IV (Code 1358/1359), HEL/HER IV.1 (Code 1371/1372), HEL/HER IV.2 (Code 1381/1382):** Pinna (links/rechts) entsprechend ITU-T P.57 Typ 3.3, anatomisch geformt
- **HHP II.1 (Code 1377):** Handapparate-Halter gemäß ITU-T P.64 mit Zuberhörkoffer (statt Zubehörbox)
- **HHP III (Code 1400):** Handapparate-Halter gemäß IEEE 269 und ITU-T P.64 mit Zuberhörkoffer (statt Zubehörbox)

## ZUBEHÖR

- **MFE III/VI/VII/VIII:** Analoge und digitale Frontends zur Datenerfassung und Signalkonditionierung
- **MFE VI.1 (Code 6462):** Mess frontend mit integriertem Mundverstärker und Pegelkonverter, optional mit binauraler Entzerrer-Funktionalität
- **BEQ II.1 (Code 1347):** Digitaler binauraler Entzerrer mit zwei Mikrofonverstärkern, individuell programmierbaren Filtern, Pulseingängen, analogen Ausgängen, USB
- **PEQ V (Code 2492):** Programmierbarer, digitaler Equalizer mit USB, AES/EBU, Line sowie Puls- und optischen Mehrkanalschnittstellen für die originalgetreue Wiedergabe gehörrichtiger Aufnahmen
- **HD IV.1 (Code 2380):** Dynamischer Kopfhörer für PEQ V
- **PVA IV.3 (Code 2486):** Leistungsendstufe für PEQ V für zwei HEADphones HA III
- **HA III (Code 2480):** Elektrostat-Kopfhörer für PVA IV.3
- **HWS (Code 1960):** Windschutz für Außenaufnahmen
- **HMT II (Code 1962):** Höhenverstellbares Stativ für HMS
- **HSC IV (Code 1524):** Tragekoffer für HMS

vertreten durch

**Panasonic** Microphone Cartridges

# Omnidirectional Back Electret Condenser Microphone Cartridge

Series: **WM-61A**
**WM-61 B** (pin type)

## ■ Features

- Small microphones for general use
- Back electret type designed for high resistance to vibrations, high signal-to-noise ratio
- High sensitivity type
- Microphone with pins for flexible PCB (WM-61B type)

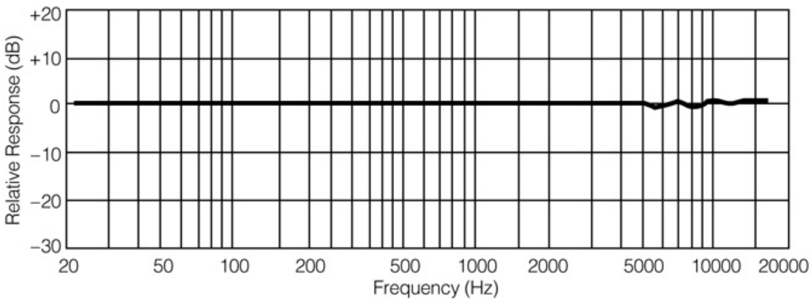
## ■ Sensitivity

| | |
|---|---|
| Vs = 2.0V<br>RL = 2.2kΩ | –35 ± 4dB |

## ■ Specifications

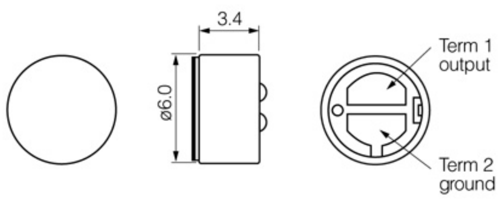
| | |
|---|---|
| Sensitivity | –35±4dB (0db = 1V/pa, 1kHz) |
| Impedance | Less than 2.2 kΩ |
| Directivity | Omnidirectional |
| Frequency | 20–20,000 Hz |
| Max. operation voltage | 10V |
| Standard operation voltage | 2V |
| Current consumption | Max. 0.5 mA |
| Sensitivity reduction | Within –3 dB at 1.5V |
| S/N ratio | More than 62 dB |

## ■ Typical Frequency Response Curve

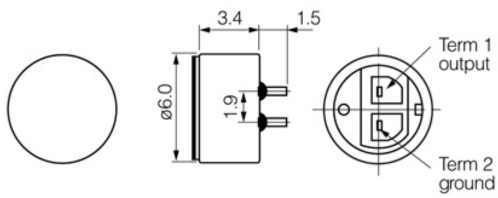



## ■ Dimensions in mm (not to scale)

WM-61A




WM-61B



# ½-inch Wide-frequency, Pressure Microphone Type 40AG

## *Product Data and Specifications*

***Typical applications***

- ***Precision sound pressure measurements***
- ***Very high frequency measurements***
- ***Laboratory reference measurements***
- ***In couplers and ear simulators***

The G.R.A.S. Microphone Type 40AG (Fig. 1) is a ½-inch precision reference microphone for laboratory use, e.g. in couplers, ear simulators, enclosures and at boundaries. It is an externally polarized pressure microphone with a large dynamic range and an extended frequency response.

As a pressure microphone, the Type 40AG measures the sound pressure at the location of its diaphragm. It has a flat pressure-frequency response over its entire working frequency range (see Fig. 2).

In an open sound field, a pressure microphone will also include the disturbing effects of its presence in the sound field. These are minimal at low frequencies (large wavelengths compared with microphone size).

At higher frequencies the effects of reflections and diffractions must be accounted for. Generally, they lead to an increase in the measured sound pressure and corrections have to be made. Fig. 3 shows what these corrections are in a free field for various angles of incidence.


*Fig. 1 ½-inch Wide-frequency, Pressure Microphone Type 40AG*

G.R.A.S. ½-inch preamplifiers (see data sheets for Types 26AG, 26AH, 26AJ, 26AK and 26AM) are also available for use with the Type 40AG. The mounting thread (11.7 mm - 60 UNS-2) is compatible with other available makes of similar microphone preamplifiers.

All G.R.A.S. microphones comply with the specifications of IEC 1094: *Measurement Microphones, Part 4: Specifications for working standard microphones*.

Non-corrosive, stainless materials are used in manufacturing these microphones to enable them to withstand rough handling and corrosive environments.

All G.R.A.S. microphones are guaranteed for 5 years and are individually checked and calibrated before leaving the factory. An individual calibration chart is supplied with each microphone.

## *Specifications*

| | | | |
|---|---|---|---|
| 3.15 Hz - 20 kHz | ±2.0 dB | Upper limit (3 % distortion): | 160 dB re. 20 μ Pa |
| 5 Hz - 12.5 kHz | ±1.0 dB | Microphone thermal noise: | 20 dBA re. 20 μ Pa |
| | 12,5 mV/Pa | | 20 pF |
| | 200 V | | -40 °C to +150 °C |

...continued overleaf

Vers.16-01-07

G.R.A.S.
Sound & Vibration

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e-mail: gras@gras.dk www.gras.dk

## ½-inch Wide-frequency, Pressure Microphone Type 40AG

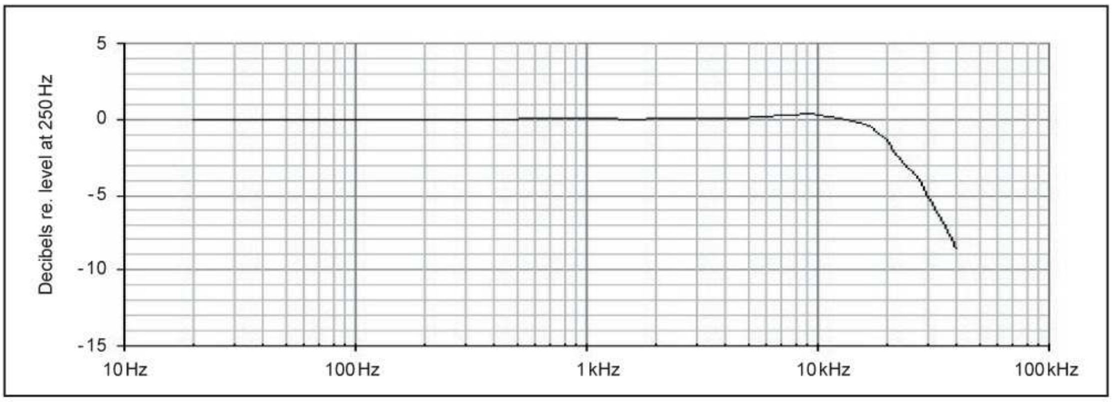


*Fig. 2 Typical frequency response for Type 40AG*

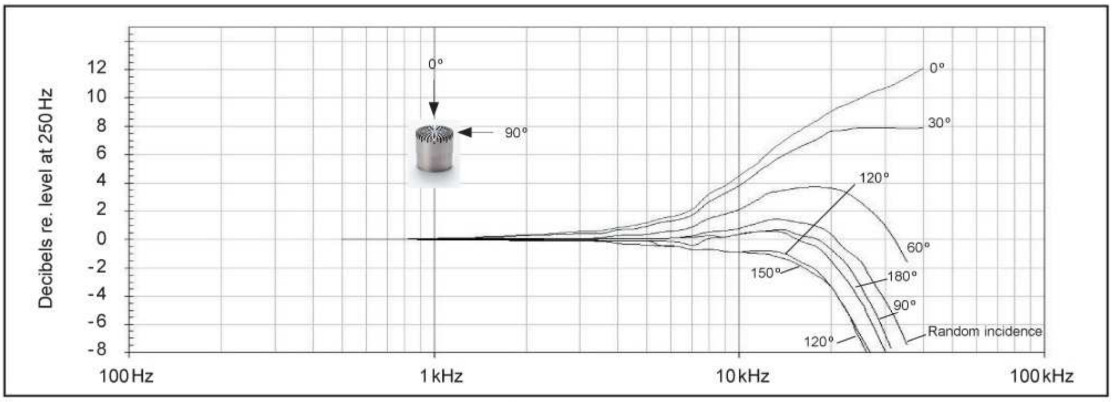


*Fig. 3 Free-field corrections for various angles of incidence*

## *Specifications (continued)*

| | |
|---|---|
| - 0.002 dB / °C | Length: . . . . . . . . . . 12.5 mm<br>Diameter: . . . . . . . . . . 13.2 mm |
| - 0.008 dB / k Pa | |
| 0 - 100 % (non-condensing) | Length: . . . . . . . . . . 11.6 mm<br>Diameter: . . . . . . . . . . 12.7 mm |
| <0.1 dB (0 - 100 % RH) | 12.1 mm |
| 66 dB re. 20 µ Pa | Protection Grid: . . . . . . . 12.7 mm - 60 UNS<br>Preamplifier Mounting: . . . . . . . 11.7 mm - 60 UNS |
| Rear vented | 7 g |
| WS2P | |



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# IEC 711 Ear Simulator Type RA0045

## *Product Data and Specifications*

***Typical applications***

- ***Insert-earphone measurements***
- ***Earphone-production tests***
- ***IEC 711 Standard measurements***
- ***ANSI S23.25 Measurements***
- ***ITU-T P.57 Type 2 Recommendation measure ments***
- ***Telephone testing with pinna simulators***

*Fig. 1 IEC Ear Simulator Type RA0045 Inset shows built-in Microphone Cartridge Type 40AG*

The IEC 711 Ear Simulator Type RA0045 (Fig. 1) is for making acoustic measurements on earphones coupled to the human ear by ear inserts such as tubes, ear moulds or ear tips. It is delivered with a built-in G.R.A.S. ½-inch pressure microphone Type 40AG and an individual calibration chart for the coupler-microphone combination.

Important! The microphone should not be removed from the coupler since this will jeopardise the factory calibration.

The RA0045 complies with the following international requirements:

- IEC 60711 Ed. 1.OB: *Occluded-ear simulator for the measurement of earphones coupled to the ear by ear inserts*.
- ITU-T Recommendations P.57 (08/96) Series P: *Telephone transmission quality, Objective measuring apparatus: Artificial ears*.

It is also part of the G.R.A.S. Artificial Ear Type 43AC.

The RA0045 can be used with a standard preamplifier, e.g. a ½-inch Preamplifier Type 26AK or a ¼-inch Preamplifier Type 26AC fitted with an adaptor. For a ¼-inch preamplifier, use either the straight Adaptor RA0003 or the right-angled Adaptor RA0001 (as in the case of the G.R.A.S. Artificial Ear Type 43AC).

The acoustic input impedance of the RA0045 closely resembles that of the human ear and, as a result, loads a sound source in very much the same way.

In accordance with ITU-T Recommendation P.57 (08/96): Series P: *Telephone transmission quality, Objective measuring apparatus: Artificial ears, Type 3.1-3.4,* the RA0045 can be used with the following G.R.A.S. pinna simulators for testing telephones:

- Low-leak Pinna Simulator Type RA0056
- High-leak Pinna Simulator Type RA0057

The RA0045 embodies a number of carefully designed volumes connected via well-defined and precisely tuned resistive grooves. In an equivalent electrical circuit, capacitors would represent the volumes, and inductance and resistance would represent respectively air mass and air flow within the resistive groves. Fig. 2 shows a typical coupler frequency response of the RA0045.

Vers.10-07-02

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## IEC 711 Ear Simulator Type RA0045

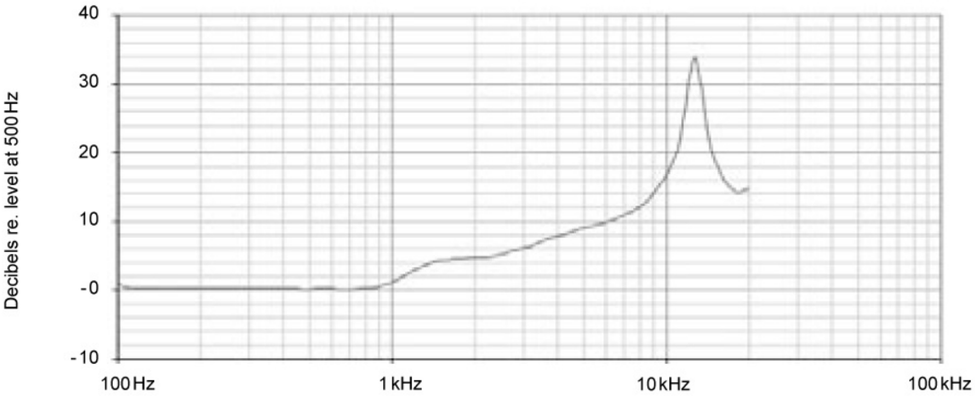



*Fig. 2 Type RA0045 - typical coupler frequency response re. 500 Hz*

The input impedance is measured using a special impedance probe as described in ITU-T Recommendations P.57 (08/96). This measures the impedance of the RA0045 as seen from the Ear Reference Point (ERP). The impedance is defined as the ratio of the sound pressure at the ERP to the corresponding particle velocity. The sound pressure is measured with a probe microphone while a constant particle velocity is maintained via a high acoustic impedance sound source.

## *Specifications*

**Standards:**

IEC 711 (1981): Occluded-ear simulators for the measurement of earphones coupled to the ear by ear inserts.

ITU-T Recommendation P.57 (08/96) "Series P: Telephone transmission quality, Objective measuring apparatus : Artificial ears"

**Resonant frequency:**

13.8 kHz ± 1 kHz

**Effective volume:**

1.28 cm³ ± 0.03 cm³

**Dimensions:**

Height: .............................. 23.0 mm
Diameter: .......................... 23.75 mm
Weight: ................................ 52 gm

**Environmental calibration conditions:**

Temperature:..................... 23 °C ± 3 °C
Relative humidity: ................ 60 % ± 20 %
Barometric pressure: .......... 101.3 kPa ± 3 kPa

**Accessories included:**

External-ear Simulator: ................ GR0408

**Accessories available:**

Calibration Simulator: ................. GR0433
Tube Adaptor:........................ GR0436
Stop Washer:......................... GR0434
Ear-mould Simulator:.................. GR0437
Retention Ring: ...................... GR0438
Tube Adaptor:........................ GR0440



G.R.A.S.
Sound & Vibration

Skovlytoften 33
2840 Holte, Denmark
Tel +45 45 66 40 46 Fax +45 45 66 40 47
e-mail: gras@gras.dk www.gras.dk

## Description:

The 26HG is a general purpose ¼" preamplifier with 3m cable terminated in a 7 pin LEMO 1B plug. It can also be used as a ½" preamplifier with G.R.A.S. ½" microphones, using the included ¼ to ½" adaptor GR0010.

**Noise floor (w. 20 pF input adapter):**

| | |
|---|---|
| A-Weighted: | 1.87 µV |
| Lin, 20-20kHz: | 3.79 µV |

Date: 18. maj 2010 Operator: PK

1/4" Preamplifier

Type 26HG

## Specifications:

**Frequency Range (±0.2 dB) (18 pF/small signal):**
2 Hz - 200 kHz

**Input Impedance:**
40 GΩ, 0.4 pF

**Output Impedance (Cs = 20 pF, f=1000Hz):**
55 Ω typical

**Output Voltage Swing (Peak):**
Min. ±50 V to Min. ±10 V

**Noise (measured with 20 pF ½" dummy mic.):**

| | |
|---|---|
| A-weighted: | < 2.5 µV |
| Linear (20Hz - 20kHz): | < 6 µV |
| Linear (20Hz - 200kHz): | < 8 µV |

**Gain:**
Typical -0.15 dB

**Power Supply:**
120V or ±60V, 2.5mA down to
28V or ±14V, 0.7mA

**Temperature:**

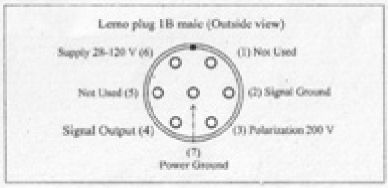
| | |
|---|---|
| Operation: | -20° - +60°C |
| Storage: | -25° - +70°C |

**Relative Humidity:**

| | |
|---|---|
| Operation: | 0 - 90% |
| Storage: | 0 - 95% |

**Dimensions:**

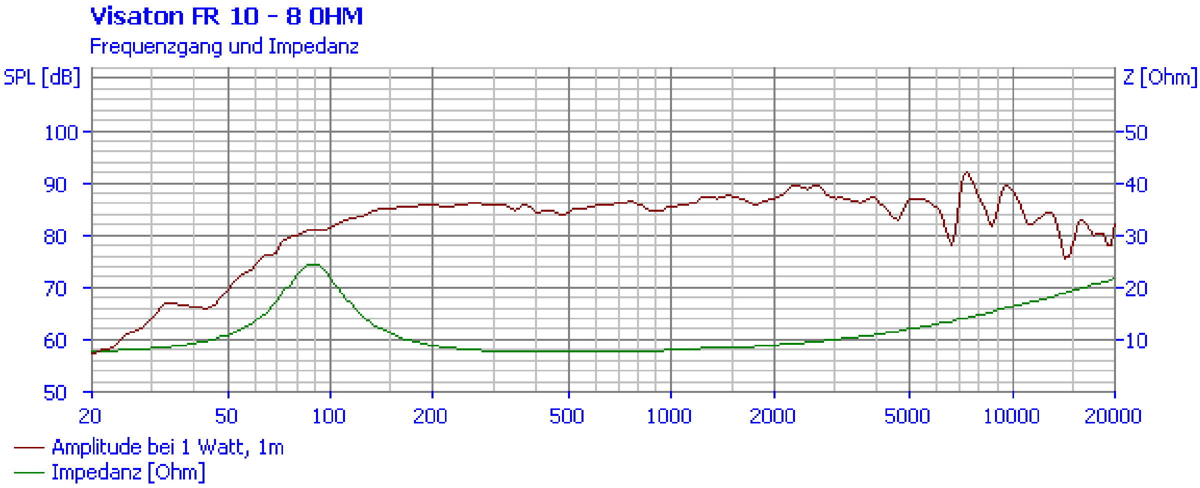
| | | |
|---|---|---|
| Diameter: | 6.35mm | |
| Length: | 43mm | |
| Weight: | 4g | (w.o. cable) |
| Weight: | 50g | (w. cable and plug) |

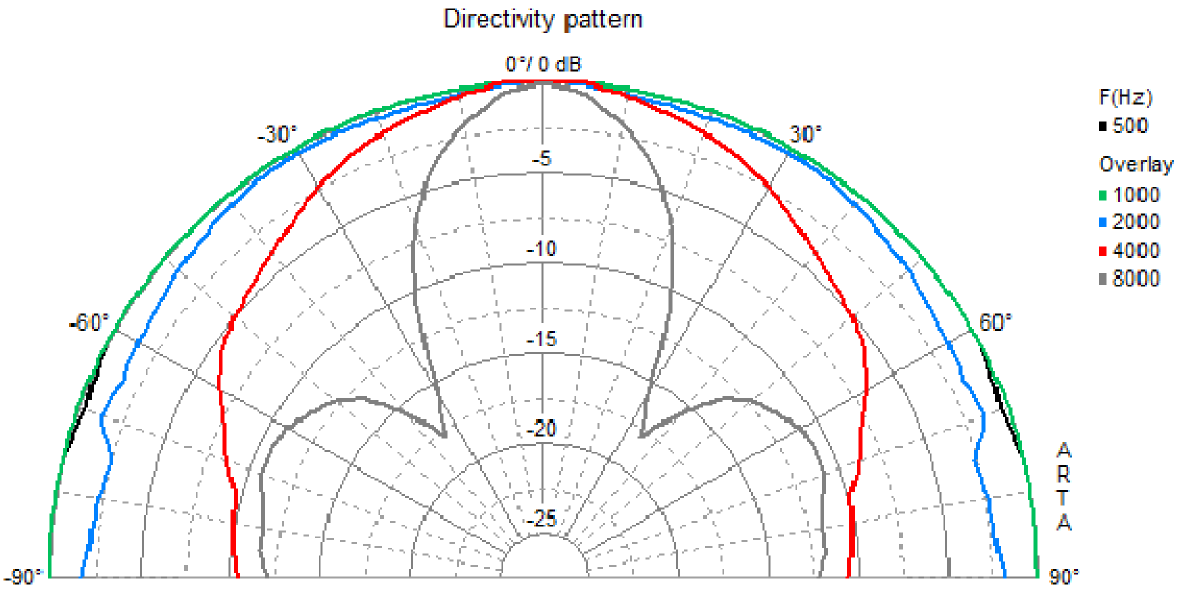


**VISATON FR 10**

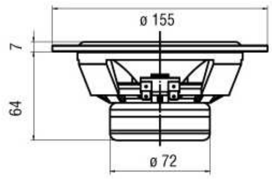
**Technical Data:**

| | |
|---|---|
| Rated Power | 30 W |
| Impedance | 8 Ohm |
| Frequencyrange (-10 dB) | 80–20000 Hz |
| Averaged Sound pressure Level | 86 dB (1 W/1 m) |
| Resonance Frequency | 88 Hz |

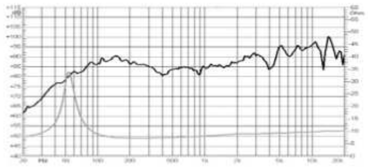


## HX 135 NEW — 80 Watt max | 8 Ω | Ø 130 mm / 5”




| | | | | | |
|---|---|---|---|---|---|
| Potenza nominale | 40 W | Re | 6,93 Ω | D | 103 mm |
| Impedenza nominale | 8 Ω | Fs | 54 Hz | Vas | 14,36 dm³ |
| Sensibilità (1W/1m) | 86 dB | Qms | 4,37 | B×l | 3,61 Wb/m |
| Ø bobina mobile | 25 mm | Qes | 1,07 | Xmax | 1,5 mm |
| Altezza traferro | 10 mm | Qts | 0,86 | $\eta_0$ | 0,20 % |
| Fori di fissaggio | 6 ø 4,5 / ø 141 | Mms | 5,94 g | Le | 0,06 mH |
| Foro pannello | 122 mm | Cms | 1,46 mm/N | Peso | 0,990 Kg |

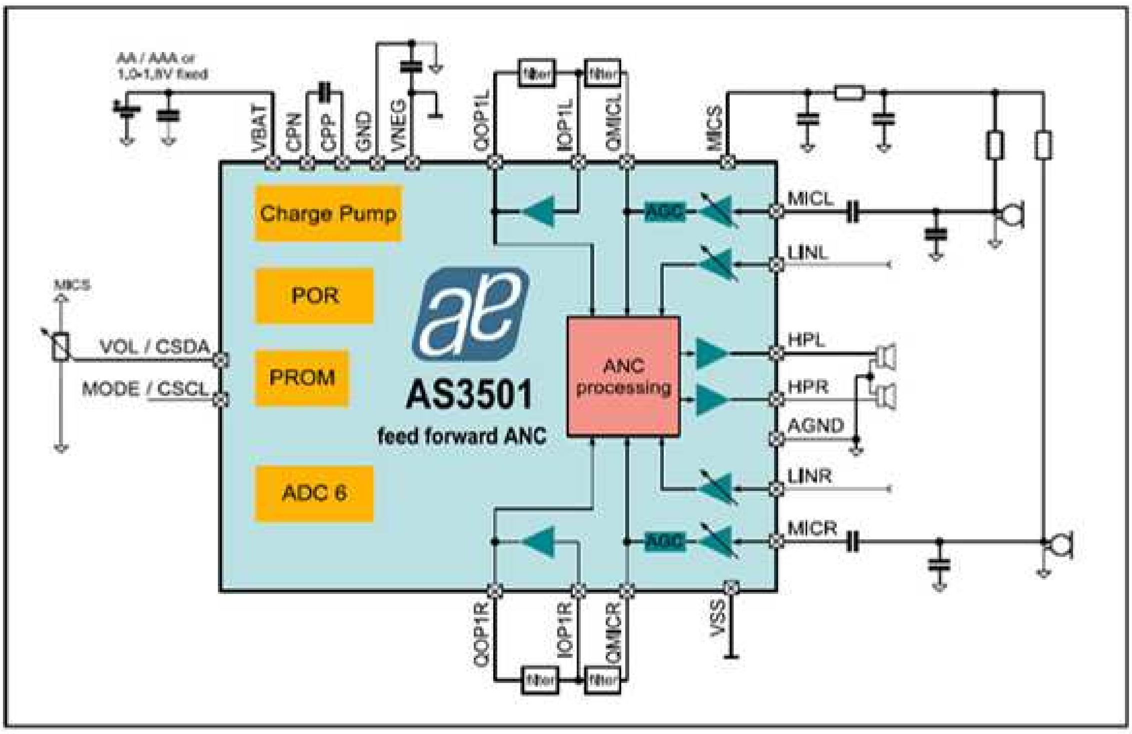


**FULL RANGE**

Membrana in cellulosa, sospensione in gomma, magnete in neodimio

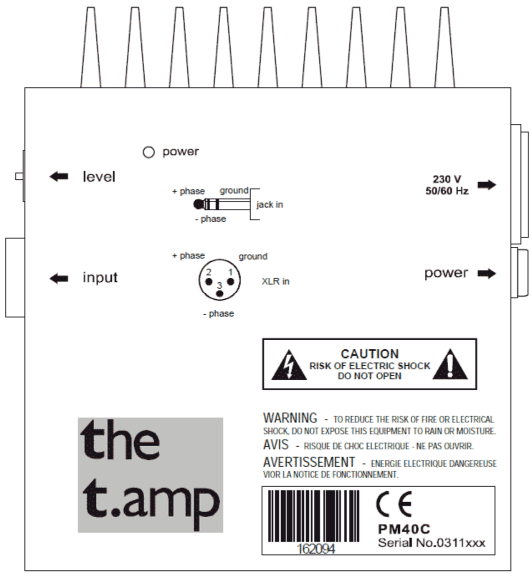
Paper cone, rubber surround, neodymium magnet

*AS3501 Feed Forward ANC Block Diagram*



**Technical Specifications**

| | |
|---|---|
| Output Power | |
| into 8 Ohms | 36W rms<br>55 musical |
| into 4 Ohms | 50W rms<br>75 musical |
| Frequency Response | 10Hz - 20 Khz / - 1dB |
| Voltage Gain | 26 dB |
| Input Impedance (active balanced) | 20 kOhm |
| Total Harmonic Distortion Plus Noise (THD+N) | 0.03% |
| Slew Rate | 19 V/µs |
| Signal-to-Noise Ratio | 92 dB |
| Power Consumption | 75 VA max. |
| Dimensions (WxHxD) | 155x166x55.5 |
| Weight | 1.8 kg |



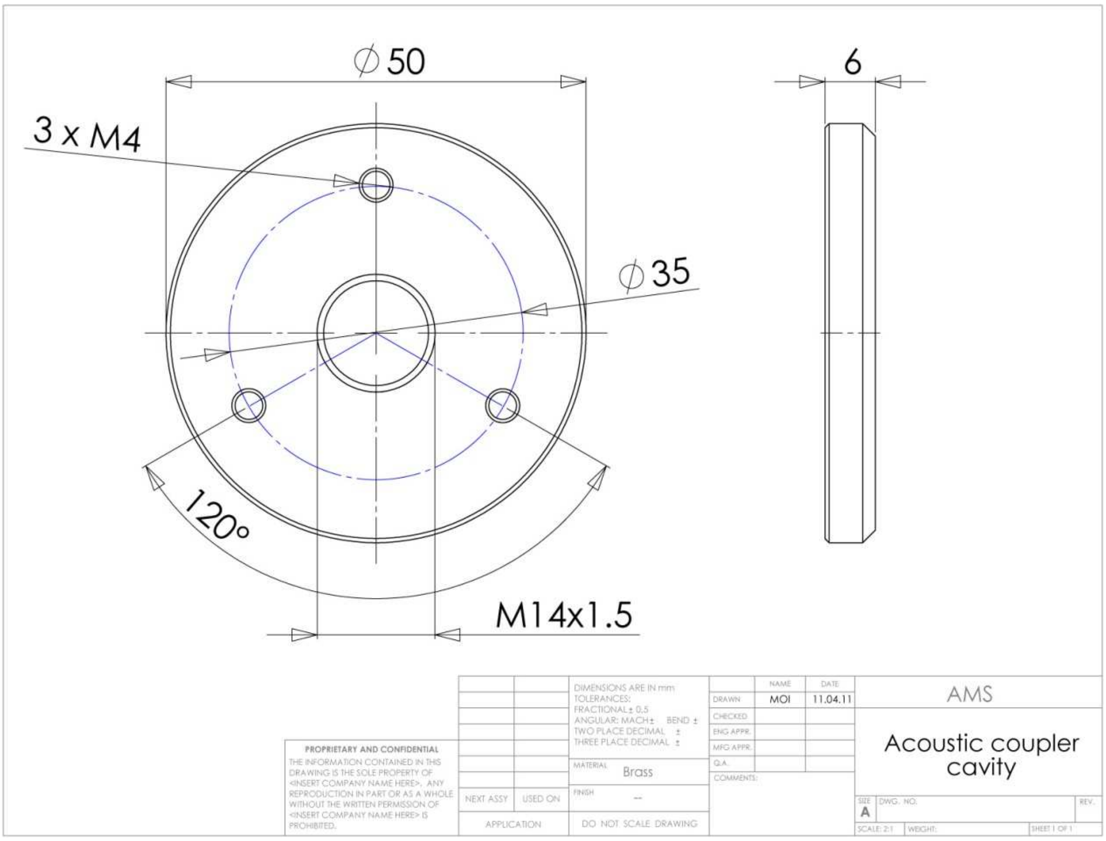

Power Amplifier Module

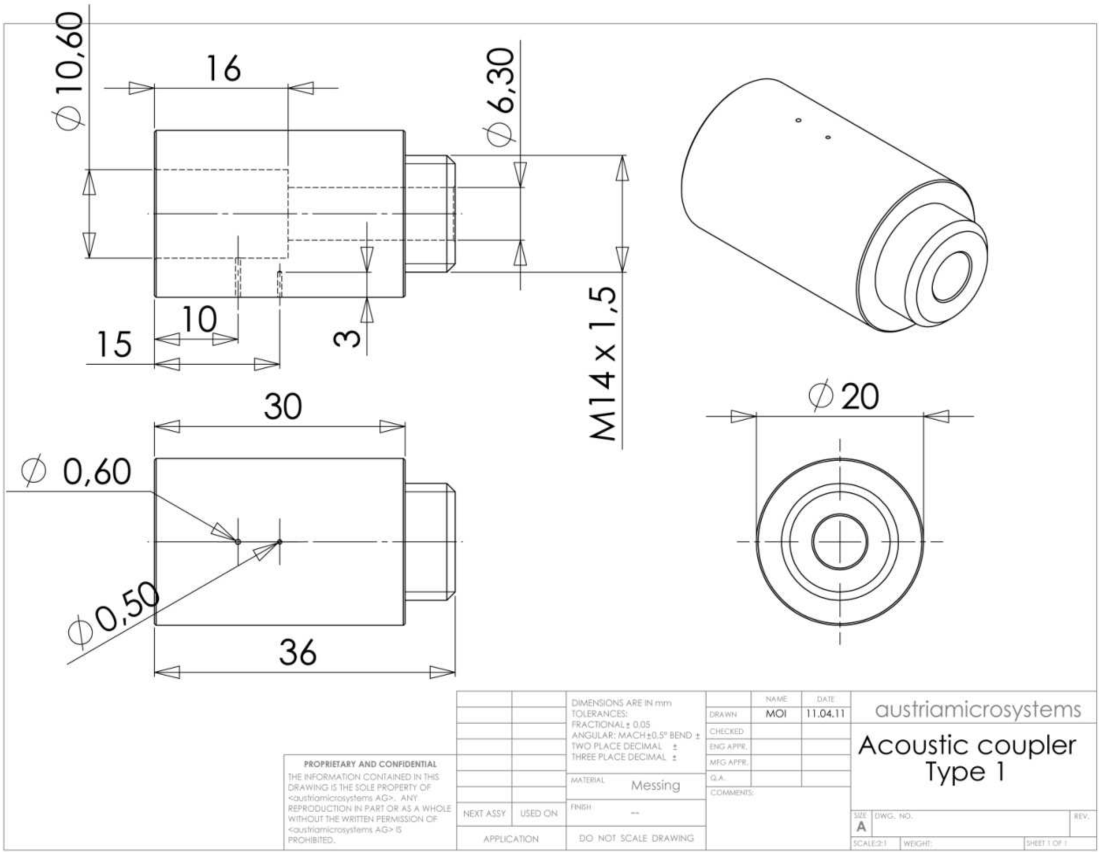
Appendix B - Datasheets

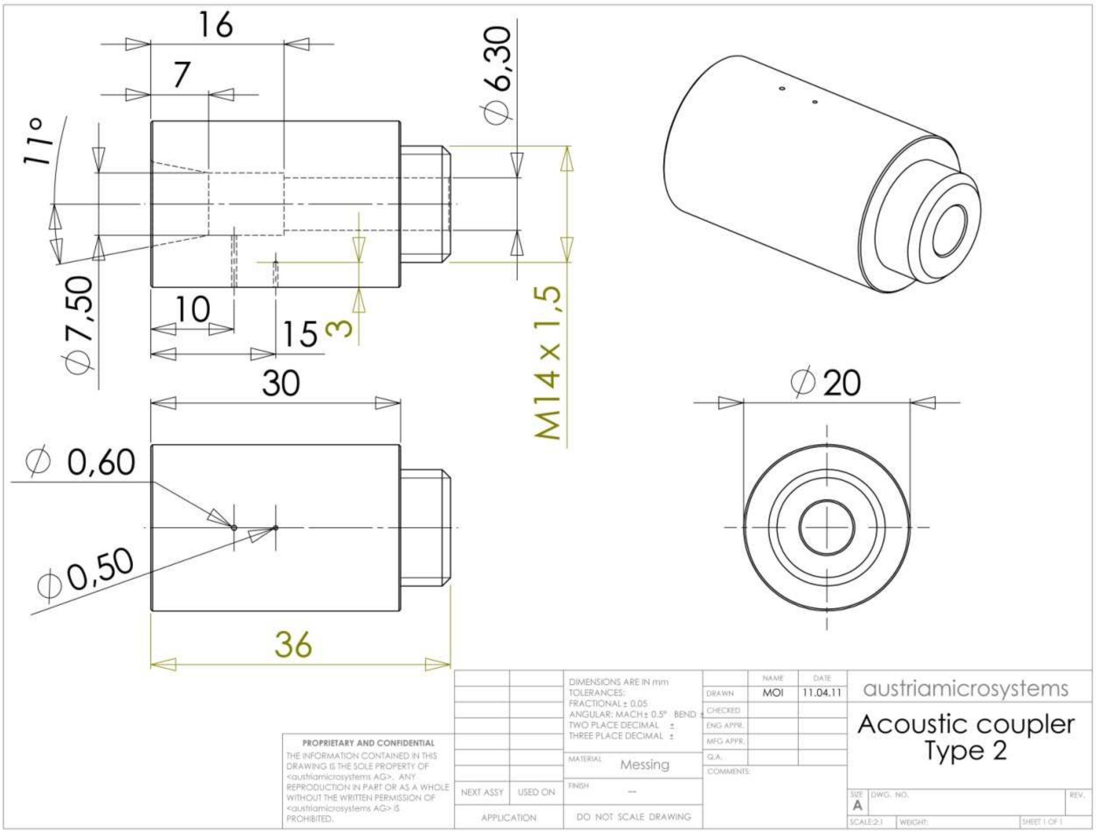
# 10 Appendix C – Design Drawings

Appendix C includes design drawings of the most important developed hardware parts. A complete and detailed library of all developed parts can be found on the enclosed CD. The enclosed CD also includes a detailed 3D Model of the complete measurement system.

# Appendix C – Design Drawings




16
7
11°
Ø7,50
10
15
3
Ø6,30
M14 x 1,5
30
Ø 0,60
Ø0,50
36
Ø20
DIMENSIONS ARE IN mm
TOLERANCES:
FRACTIONAL ± 0.05
ANGULAR: MACH ± 0.5° BEND ±
TWO PLACE DECIMAL ±
THREE PLACE DECIMAL ±
MATERIAL Messing
FINISH --
DO NOT SCALE DRAWING
NAME DATE
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CHECKED
ENG APPR.
MFG APPR.
Q.A.
COMMENTS:
austriamicrosystems
Acoustic coupler Type 2
SIZE A DWG. NO. REV.
SCALE:2:1 WEIGHT: SHEET 1 OF 1
NEXT ASSY USED ON
APPLICATION
PROPRIETARY AND CONFIDENTIAL
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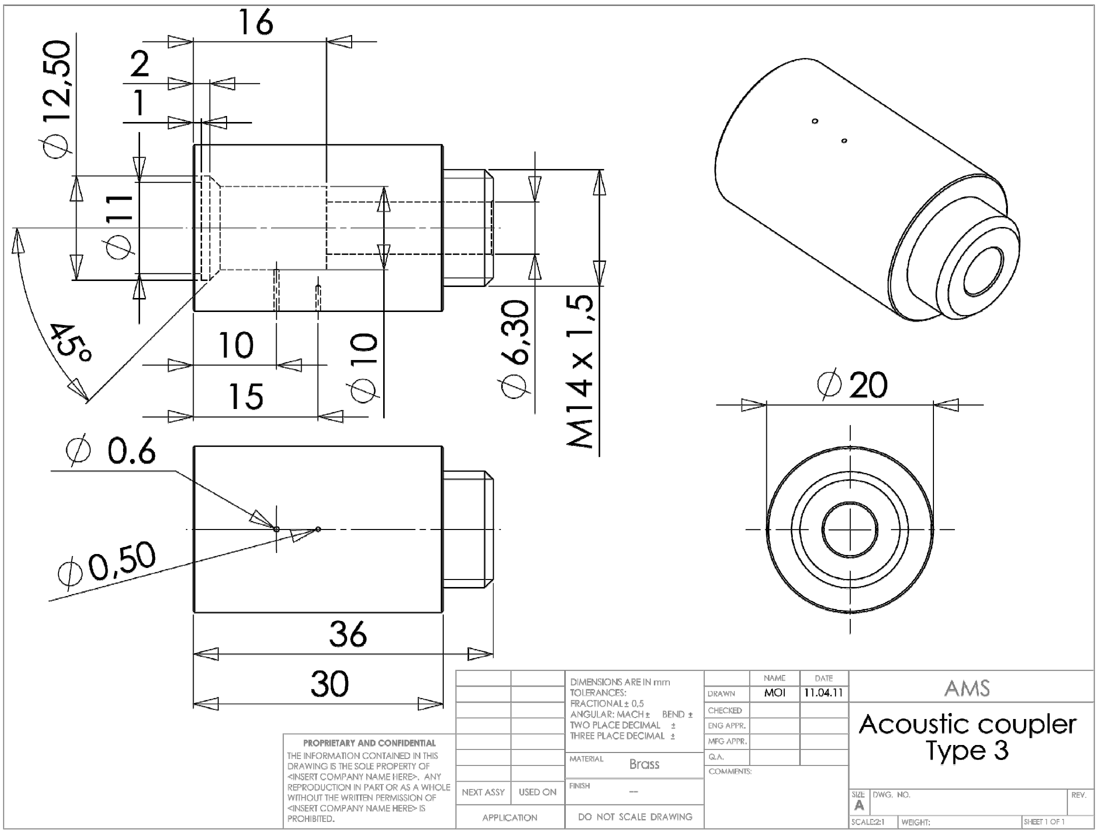
16
2
1
Ø12,50
Ø11
45°
10
15
Ø10
Ø6,30
M14 x 1,5
Ø 0.6
Ø0,50
36
30
Ø20
DIMENSIONS ARE IN mm
TOLERANCES:
FRACTIONAL ± 0.5
ANGULAR: MACH ± BEND ±
TWO PLACE DECIMAL ±
THREE PLACE DECIMAL ±
MATERIAL Brass
FINISH --
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MFG APPR.
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COMMENTS:
AMS
Acoustic coupler Type 3
SIZE A DWG. NO. REV.
SCALE:2:1 WEIGHT: SHEET 1 OF 1
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APPLICATION
PROPRIETARY AND CONFIDENTIAL
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Appendix C – Design Drawings