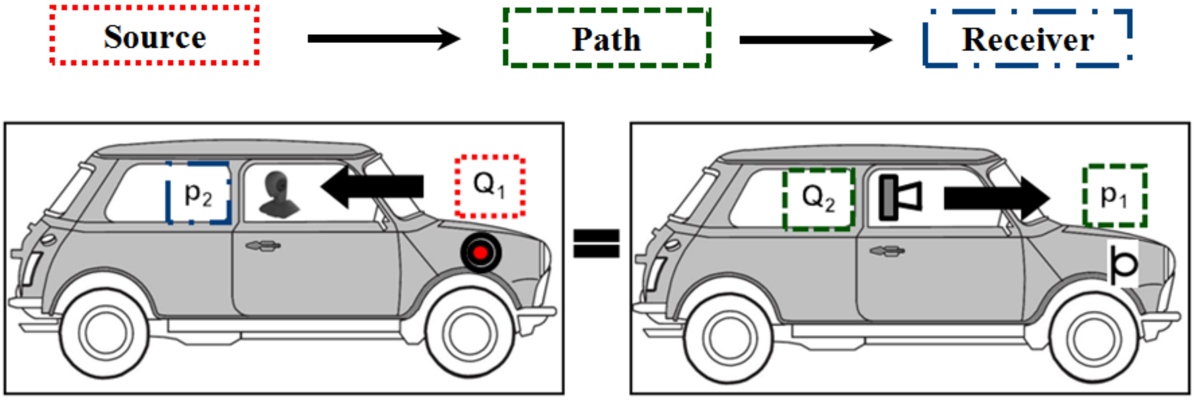

Applying the reciprocal Transfer Path Analysis (TPA) for the airborne sound of power train components

Master's Thesis in the International Master's programme Sound and Vibration

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Cover:
The principle of acoustic reciprocity with the supporting source-path-receiver model

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Abstract

The design and construction of power train components is increasingly characterized by shorter development periods and declining prices. Nowadays these entrepreneurial objectives basically affect almost every division in the automotive industry. As a result of this trend the acoustic engineers are relying more on numerical methods that are to some extent already implemented in conventional software applications. The usage of these approaches offers the potential to already describe the acoustical and structural behavior of certain component parts in the early stages of the development processes.

The objective of this thesis is to develop a procedure for predicting the airborne sound in the interior of a vehicle that results from the operation of a power train component. The functionality of this method should give the acoustic engineers the opportunity to only perform test bench measurements without the need of install the structural element in a vehicle. The central issues for solving this task are, on the one hand, the application of the reciprocal transfer path analysis and, on the other hand, the determination of the volume velocity of a power train component by sound pressure measurements in the near-field of a suitable test rig.

The usage of an interchangeable source-path-receiver model in principle requires a monopole sound source with a known volume velocity as an input parameter. This work therefore contains the description for the design and construction of a self-made point source. The determination of the volume velocity for such a self-made loudspeaker is then discussed. Hence the transfer paths from the interior to defined microphone positions in the engine bay can be built. The acting volume velocity on a test stand can be identified by sound pressure measurements in the near-field. Combining the measured transfer paths of a vehicle with the appointed volume velocity of test bench measurements finally enables the prediction of the sound pressure level in the interior that results from the respective component part.

Keywords: transfer path analysis (TPA), monopole sound source, acoustic reciprocity, volume velocity, airborne sound, vehicle acoustic, NVH

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... keep in touch!!!

Berlin, April 2011

Kai Rißler

List of notations

Roman upper case letters

A	*area*	$[m^2]$
C	*wave amplitude (cf. appendix I)*	
D	*wave amplitude (cf. appendix I)*	
F	*force*	[N]
H	*transfer function*	
I	*sound intensity*	$\left[\frac{W}{m^2}\right]$
P	*sound power*	[W]
Q	*volume velocity*	$\left[\frac{m^3}{s}\right]$
R	*radiation resistance factor*	
S	*cross section*	$[m^2]$
Z_0	*impedance of air (acoustic impedance)*	$\left[\frac{kg}{m^2 s}\right]$

Roman lower case letters

a	*distance (radius)*	[m]
c	*speed of sound*	$\left[\frac{m}{s}\right]$
d	*diameter*	[m]
d_i	*inner diameter*	[m]
f	*frequency*	$\left[\frac{1}{s}\right]$
f_0	*upper frequency limit*	$\left[\frac{1}{s}\right]$
k	*wavenumber*	$\left[\frac{1}{m}\right]$
l	*distance parameter (cf. appendix I)*	[m]
p	*sound pressure*	[Pa]
r	*reflection coefficient*	
s	*distance parameter (cf. appendix I)*	[m]
u	*particle velocity*	$\left[\frac{m}{s}\right]$

Greek lower case letters

ϱ_0	*density of air*	$\left[\frac{kg}{m^3}\right]$

List of abbreviations

BEM	Boundary Element Method
CVT	Continuously Variable Transmission
DMF	Dual Mass Flywheel
FEM	Finite Element Method
I-BEM	Inverse Boundary Element Method
LDV	Laser Doppler Vibrometer
NVH	Noise, Vibration and Harshness
PMV	Predicted Mean Vote
PPD	Predicted Percentage Dissatisfied
TPA	Transfer Path Analysis

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1 Introduction

Humans are strongly affected by their five senses. Everybody explores their surroundings using individually gained information and experiences. This means that everyone is capable of expressing their feeling about a situation, a room, a temperature or a sound. The feeling can vary from well-being to discomfort and depends on several factors.

Looking for example at the prediction of thermal comfort in an atmospheric environment and the related PMV and PPD indexes, one sees that there are, in the best case, still 5% of persons who are unhappy [30]. From these results one can conclude that it is generally impossible to find a suitable relation of temperature and humidity in a room that fulfills everyone's expectations. Transferring this approach to the sound quality of products definitely also be an interesting research question in the field of psychoacoustics. One would probably find similar results, that it is impossible to satisfy everbody's hopes, wishes and individual preferences.

Basically everyone is in an indefinite number of situations in life a customer of a product. For the producers, the customers is a black box, with a countless number of unknown likes and dislikes. Identifying these preferences is the great difficulty for engineers nowadays. The technical requirements of a product thereby fall more and more in to the background, as the so called "feel-good" aspects increasingly show more importance. The vibrational behavior and corresponding sound quality of a product are part of these "feel-good" comfort factors.

In the development of vehicles the acoustics and the sound quality also play an increasing role. The acoustic proporties are now one of the top arguments in the advertisement of automotive products. The NVH engineers therefore have to generate well-being and comfort for the end customer as well as the technical requirements. Since the technical components are an almost fully developed, the former clearly has increased in significance. Of particular importance is the perceived noise in the interior during the operation of a vehicle. A number of academics have therefore conducted numerous scientific investigations in order to, on the one hand, describe the noise in the interior and, on the other hand, develop potential measures for improving the sound quality.

One of the main procedure that is increasingly used in the field of vehicle acoustics is the introduction of a source-path-receiver model to describe a complete system. The conventional description of this method is called as the transfer path analysis in literature. The principle of a source-path-receiver model is based on the idea of considering the sound sources, the transfer paths and the receiver of a system separately from one another. The sound sources of a vehicle are primarily the combustion noise, the drive train noise, the rolling noise, the wind noise and the exhaust noise. The receiver is commonly standing for the passengers in the interior of the vehicle. The transfer path analysis basically includes the structure-borne sound and the airborne sound. The application of the transfer path analysis in the development processes of a vehicle provides the possibility to delineate the existing terms and conditions scientifically. Furthermore, the transfer path analysis facilitates the search for alternatives in the early stages of vehicle production.

The LuK GmbH & Co. KG in Bühl is one of the top manufacturers of gearbox systems, clutch systems, dampers and further assemblies. The acoustical optimization of the products is part of the intensive development processes, in order to remain market leader in the future.

1.1 Aim of the thesis

The task of this thesis is to develop a useful approach for predicting the receiving sound pressure level in the interior of a vehicle that is originally caused by a power train component part from LuK GmbH & Co. KG. Thus the existing development processes of the LuK GmbH & Co. KG should be improved significantly with the focus on the acoustical behavior of the products. The usage of this procedure should give the acoustic engineers the opportunity to consider their own components separately from the residual installed sound sources in an engine bay.

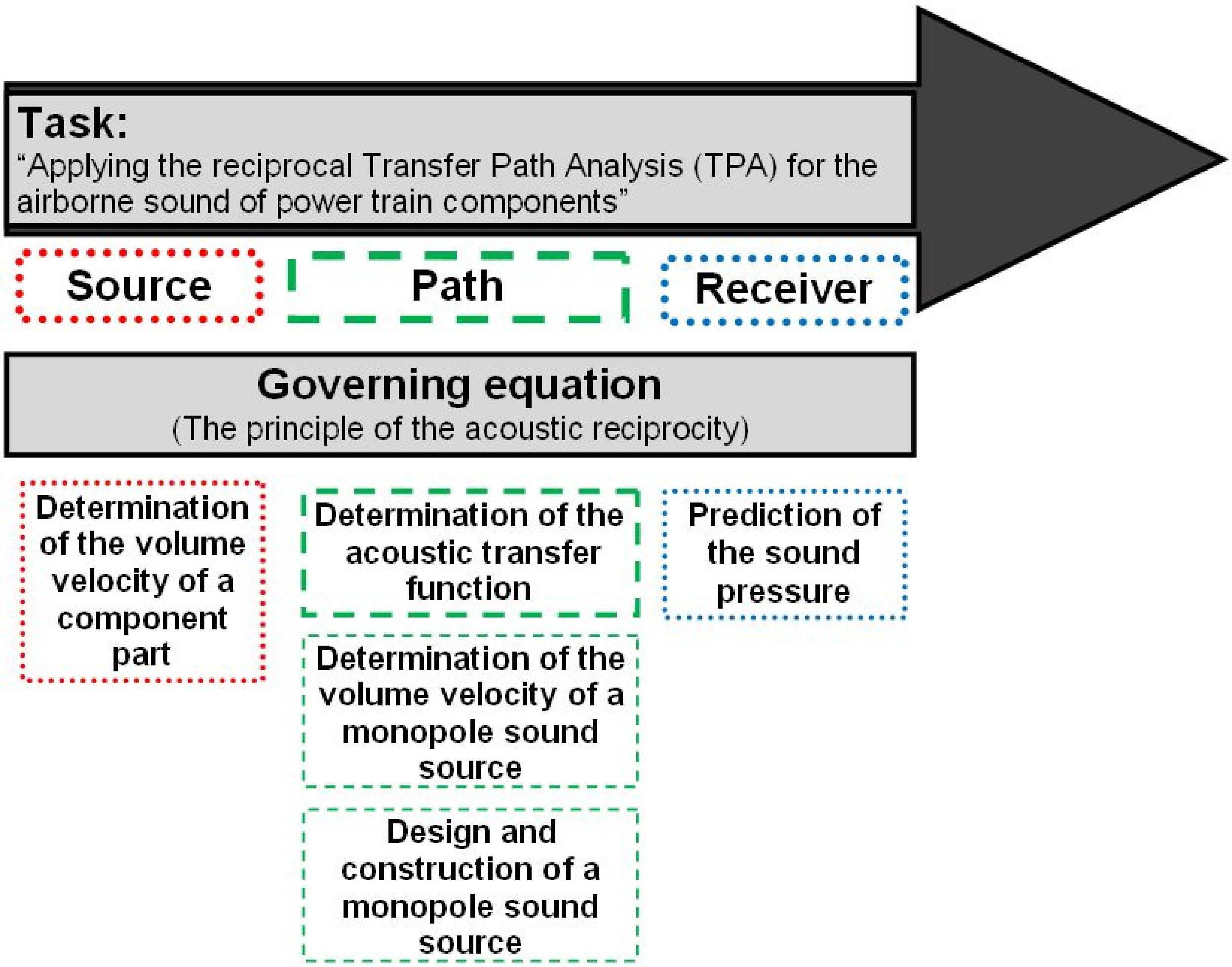

Figure 1-1: An overview of the tasks for this thesis

The tasks in **Figure 1-1** give an overview of the most important objectives in this thesis. The base of this work is the use of a source-path-receiver model that is grounded in the principle of acoustic reciprocity (governing equation). The source, path and receiver elements can be identified in **Figure 1-1**.

- The first central issue is the determination of the volume velocity of an examined structural element. The goal is to measure the radiated sound pressure in the near-field of a respective test bench consequently and to calculate the acting volume velocity.

- The second core problem in this thesis focuses on the application of the reciprocal transfer path analysis. The airborne sound paths from the motor compartment into the interior have to be measured reciprocally. Therefore is it necessary to place a loudspeaker with an omnidirectional radiation characteristic, in the passenger cabin and microphones on defined measurement positions in the engine bay. Since the LuK GmbH & Co. KG does not own such a volume sound source with an omnidirectional radiation characteristic, a secondary goal of this work is the design and construction of such a point source. Another problem is the determination of the acting volume velocity under operation of the self-made loudspeaker. Building the ratio of the received sound pressure to the acting volume velocity of the monopole sound source finally gives the respective transfer function.

- Allocating the volume velocity of test stand measurements for a considered component and the determined airborne sound transfer functions finally gives the predicted sound pressure in the interior. The governing equation therefore presents the principle of the acoustic reciprocity.

The next section presents the theory of this Master Thesis.

2 Theory

This chapter presents an overview of the theoretical background and the general application of the reciprocal transfer path analysis for the airborne sound in this work. At the beginning the essential basics of this approved method shall be discussed. The reader should get a fundamental explanation and an introduction to the issue. Simultaneously, this chapter covers the basics for further investigations that will be performed later in this thesis.

2.1 Introduction of a source-path-receiver model

For the performance of the reciprocal transfer path analysis it is helpful to separate a complete system into source, path and receiver elements. It is then possible to consider the three elements independently from each other [3,5,7,15]. This explains why the transfer path analysis is also called source-path-receiver model in literature [3,5]. Describing a system with three unaffiliated elements simplifies the procedure of the reciprocal transfer path analysis enormously, as will be shown subsequently in this section.

As was stated in the previous chapter the automotive sector is today the most important business area for the LuK GmbH & Co. KG. This part of the work will therefore consider a car as an exemplary system. Thus it should become easier for the reader to understand the theoretical application of the reciprocal transfer path analysis for a real object. A car generally has an amount of sound sources that generate structure-borne and airborne sound. The most important sound sources of a car are combustion noise, drive train noise, rolling noise, wind noise and exhaust noise [5].

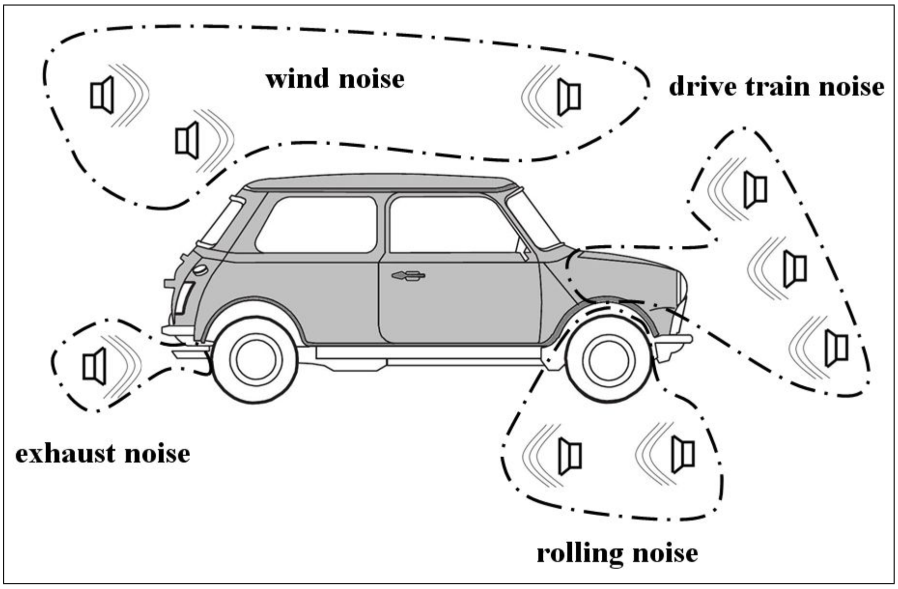

Figure 2-1: The most important sound sources of a car [5]

This thesis only looks at the airborne sound from the drive train components. It therefore focuses on the products of the LuK GmbH & Co. KG that are normally installed in an engine bay and thus part of the power train. The core products of the LuK GmbH & Co. KG are clutch systems, gear box systems, dampers and other assemblies (cf. section 1.3).

The clutch system of a car is an important element of the power train. As an example, one can consider this component part as the source element and the vehicle structure of a car as the path element; the receiver can be represented by a dummy head. The latter replaces the passengers in the interior. This example is graphically summarized in the illustration below.

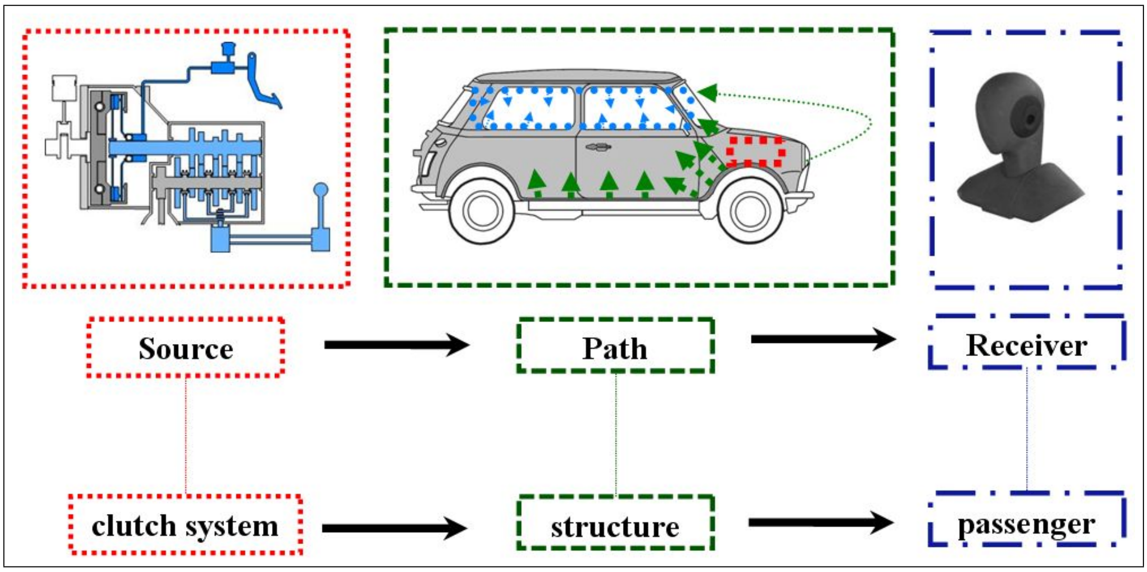

Figure 2-2: An example for a source-path-receiver model of a car

Generally an interaction exists between the source, path and receiver elements of a complex system like a car. The connection between these elements focuses more precisely on the correlation between the structure-borne sound, the airborne sound and the fluid-borne sound that occurs in the respective elements. The reason for this is the transformation of the sound and the corresponding impedance changes in the system. The functional principle of our ears provides the most common example for this procedure [2]. The radiating sound pressure (airborne sound) is bundled in the outer ear, transformed into structure-borne sound in the middle ear and finally converted to fluid-borne sound in the inner ear.

The clutch system (source element) in **Figure 2-2** introduces structure-borne and airborne sound into the vehicle structure (path element), whereas only the airborne sound is considered at the defined receiver position (dummy head). The next figure shows the complexity of this important issue.

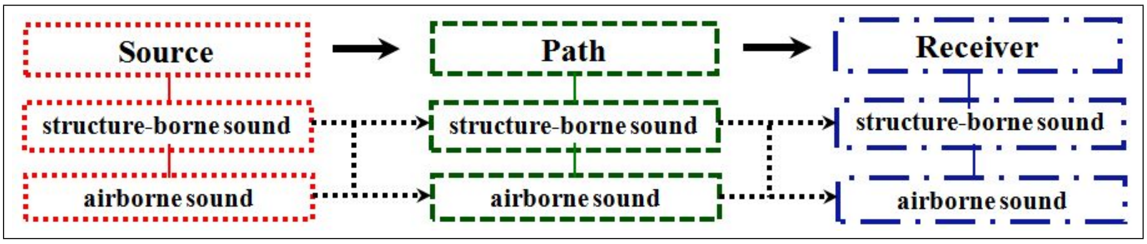

Figure 2-3: An example for a source-path-receiver model (including the existing interactions between the different forms of sound)

The interaction between the elements and the varying forms of sound can be regarded as arbitrary.

This thesis studies only the field of airborne sound of the respective elements (source, path and receiver) and neglects the residual forms of sound. Thus the relevant source-path-receiver model has to be established.

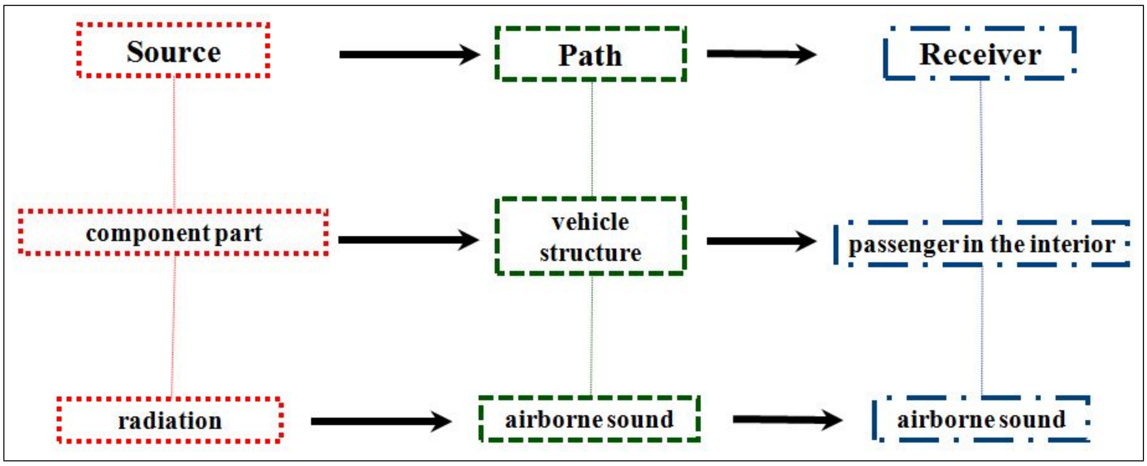

Figure 2-4: The fundamental source-path-receiver model for this thesis

This section will again examine the fundamental definition in **Figure 2-4** and explore the detailed elements. This includes the determination of the unknown parameters of the source, the path, as well as the receiver elements.

In the next section the governing equation for this thesis is presented. The above discussed investigation should then become mathematically solvable.

2.2 Governing equation

The elementary approach for performing the presented investigation in section 2.1 requires the use of the principle of acoustic reciprocity. This approach was successfully demonstrated in [3,5,7,10,12,13,15,18].

Assume interdependence between two points (excitation/response) provides the linearity and passivity of a system [3]. If the compliance of these two quantities is ensured, the positions of excitation and response are interchangeable. The first major step for applying a source-path-receiver model therefore focuses on determining the linearity of the respective system.

The acoustic reciprocity is generally defined as

$$\left.\frac{p_2}{\vec{Q}_1}\right|_{\vec{Q}_2=0} = \left.\frac{p_1}{\vec{Q}_2}\right|_{\vec{Q}_1=0} \qquad \textbf{Eq. 2-1}$$

The following figure the principle of acoustic reciprocity (cf. **Eq. 2-1**) is graphically represented using a car as exemplary system.

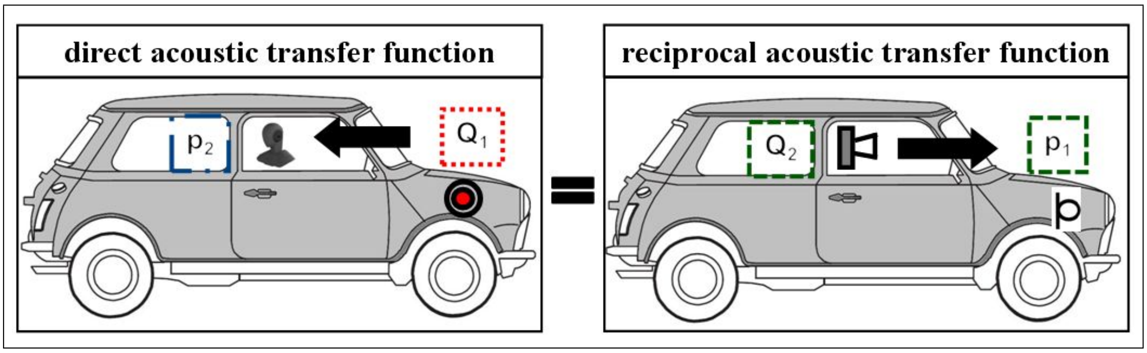

Figure 2-5: The application of the principle of acoustic reciprocity for a car [3,5,15]

The left drawing in **Figure 2-5** stands for the first part of **Eq. 2-1** (left term) and describes the direct acoustic transfer function. The input quantity represents the unknown volume velocity ($\vec{Q}_1$) of the considered component part in operation. Since the component part is not specified here in more detail, one can regard this volume velocity for the moment as a black box. The output parameter of the direct acoustic transfer function is given by the sound pressure (p_2) that is measured in the interior of the car during operation of the component part $\vec{Q}_1$. Usually this measurement is performed binaurally, with an artificial head in the passenger compartment at the level of driver's ears.

The second part of **Eq. 2-1** (right term) is depicted by the right drawing in **Figure 2-5** with plotting of the reciprocal acoustic transfer function. Compared to the left side of **Eq. 2-1** the positions of excitation and response are interchanged. A loudspeaker ($\vec{Q}_2$) is now installed in the interior, whereas a microphone (p_1) is positioned in the engine bay (close to the original position of the component part ($\vec{Q}_1$)).

According to the literature this procedure is only applicable in the field of acoustics when the microphone (p_2) and the loudspeaker ($\vec{Q}_2$) have the same directivity [3,7,10,12,18].

It can be noted that only microphones and loudspeakers with an omnidirectional radiation characteristic, in the frequency range of interest, are used in this work. This means that the basic prerequisite for the principle of acoustic reciprocity is given.

The combination of the source-path-receiver model of the previous section with the principle of acoustic reciprocity in **Figure 2-5** leads to the definition in the subsequent figure.

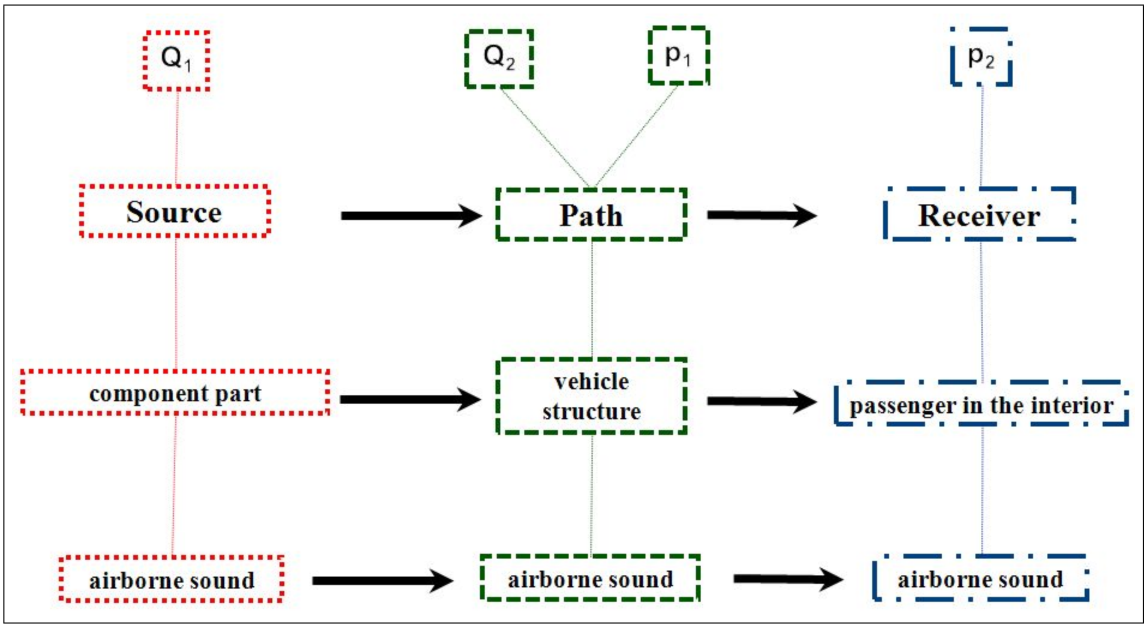

Figure 2-6: The combination of a source-path-receiver model (cf. section 2.1) and the principle of acoustic reciprocity (cf. **Eq. 2-1** and **Figure 2-5**)

In **Figure 2-6** one can see that the volume velocity $\vec{Q}_1$ is related to the source element of the source-path-receiver model. The second part of **Eq. 2-1** is pictured by the parameters $\vec{Q}_2$ and p_1 that are assigned to the path element, whereas the receiver element is given by the sound pressure p_2.

- The upcoming sections focus on the determination of the unknown parameters of the source-path-receiver model in **Figure 2-6**. The unknown volume velocity ($\vec{Q}_1$) of the component part will be estimated on the basis of sound pressure measurements in the near-field (cf. section 2.3).

- To achieve the volume velocity of a monopole sound source ($\vec{Q}_2$) and the resulting sound pressure (p_1), the acoustic transfer function is required (cf. section 2.4).

- The prediction of the sound pressure in the interior by investigating the principle of acoustic reciprocity will be shown later in section 2.5 (cf. **Eq. 2-1**).

2.3 The volume velocity of a component part (source)

As already described above, this part of the thesis deals with the estimation of the unknown volume velocity $\vec{Q}_1$ (cf. **Eq. 2-1**) for a considered component part. Basically this work studies a procedure that uses sound pressure measurements in the near-field of the structural element on a test bench. This procedure was already successful applied to an engine in literature [3,18].

There are a number of other approaches in literature for determining the volume velocity of a component part on a test stand. The experiments in [18] showed that a sound intensity probe [3,4,17] also achieves the volume velocity of a component part. The application of the Inverse Boundary Element Method (I-BEM) showed sufficient results in [3] sufficient results. For the performance of other numerical methods the acting surface velocity of the regarded source is generally required. The surface velocity stands for the input parameter of the Boundary Element Method (BEM) and can either be gained by measurements with accelerometers on the surface or by calculations of the Finite Element Method (FEM) [19]. If the assembly surface is accessible, one can also use an Laser Doppler Vibrometer (LDV) for achieving the acting surface velocity [1,2].

The determination of the volume velocity $\vec{Q}_1$ (cf. **Eq. 2-1**) and the analogous source element is the first step (cf. section 1.1 and **Figure 1-1**) for an efficient prediction of the sound pressure in the interior of a vehicle that results from the operation of a component part. However, the estimation of the volume velocity $\vec{Q}_1$ (cf. **Eq. 2-1**) of a component part by sound pressure measurements in the near-field can't be classified as an exact method [3]. For simplification, this fundamental process is now explained by introducing a black box as exemplary system for the source element. It is understood that this black box could thereby express any component part from the LuK GmbH & Co. KG.

First it is generally essential to subdivide each surface (A_i) of the black box into smaller subareas (A_{ij}) and to measure, for each partial area, the appropriate sound pressure (p_{ij}). In literature this procedure is also described as the discretization of the surface vibration field into a set of equivalent monopoles [3,18,19]. This explanation is later used again in this section and is therefore discussed in more detail.

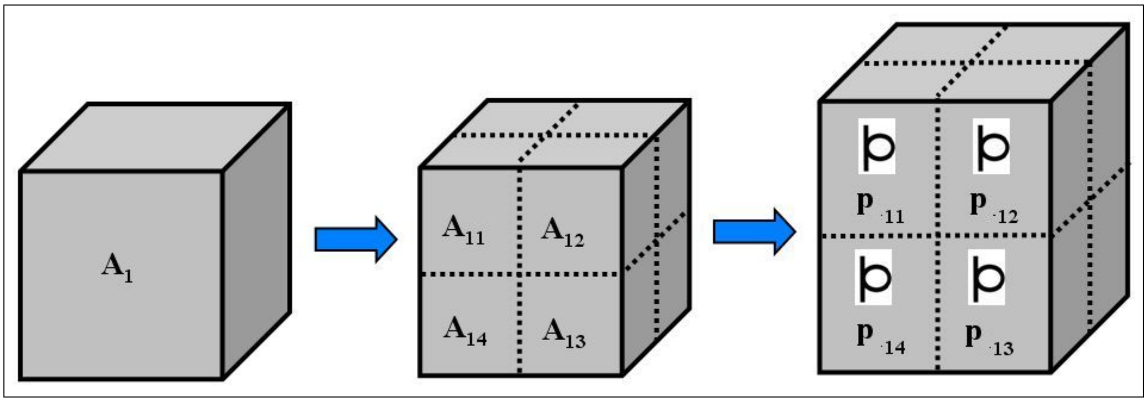

Figure 2-7: A procedure for subdividing the surfaces (A_i) of a black box (source element) in a number of various subareas (A_{ij}) and determining for each subarea the particular sound pressure (p_{ij})

The microphones in **Figure 2-7** have to be placed in the center of the respective subarea (A_{ij}). It is assumed that the higher the number of the subareas is, the more exact the results will be. However, the conclusions in [3] showed that dividing the side surfaces of an engine into respectively two areas is enough for the successful performance of this approach. In [18] was in contrast, a very large number of subareas was used, with the application of the sound intensity probe and the resulting contribution analysis of a car top. According to [3] it is useful to separate the subareas by a distance that is much less than an acoustic half-wavelength.

The above presented procedure in **Figure 2-7** is illustrated with four microphones for each surface. The remaining sections of this chapter and the corresponding examples will retain this procedure. Note that the number of microphones and the relating number of subareas was here freely chosen and is only used as an example.

The above described investigation requires that the radiated sound of the black box is unaffected by other sound sources and that almost free field conditions are met, in order to measure reasonable and undistorted results [3].

In a next step the theoretical foundations of the calculation of the volume velocity ($\vec{Q}_{1_{ij}}$) on the basis of sound pressure measurements for the regarded black box are discussed. Therefore is it helpful to allocate each subarea to a monopole sound source [3,19]. The sound power of such a point source, for the respective subarea can be expressed with

$$P_{ij} = \frac{\pi}{2c} \cdot f^2 \cdot \vec{Q}^2_{1_{ij}} \cdot R_{ij} \qquad \textbf{Eq. 2-2}$$

The radiation resistance factor R is defined as the ratio of radiated sound power by a monopole sound source on the body surface, to the amount that propagates in a free field [3,18].

The sound power P can be calculated with the sound intensity I and the respective subarea A_{ij} [3,18].

$$P_{ij} = \int_A \vec{I}_{ij} d\vec{A} \qquad \textbf{Eq. 2-3}$$

Assuming that the volume velocities $\vec{Q}_{1_{ij}}$ of the subareas A_{ij} are temporarily uncorrelated leads finally to

$$\vec{Q}_{1_{ij}} = \sqrt{\frac{2c \cdot Pij}{\pi \cdot \varrho_0 \cdot f^2 \cdot R_{ij}}} \qquad \textbf{Eq. 2-4}$$

Eq. 2-4 then gives the volume velocity $\vec{Q}_{1_{ij}}$ for the particular subareas (A_{ij}).

According to literature the radiation resistance factor R in most cases has a value of R_{ij}=2 [3,18]. Hence **Eq. 2-4** can be simplified to

$$\vec{Q}_{1_{ij}} = \sqrt{\frac{A_{ij}}{\pi}} \frac{p_{ij}}{\varrho_0 \cdot f} \qquad \textbf{Eq. 2-5}$$

Regarding **Eq. 2-5**, one sees that the calculation of the volume velocities $\vec{Q}_{1_{ij}}$ for the respective subareas requires only the parameter A_{ij} and the corresponding sound pressure p_{ij} of the respective subarea.

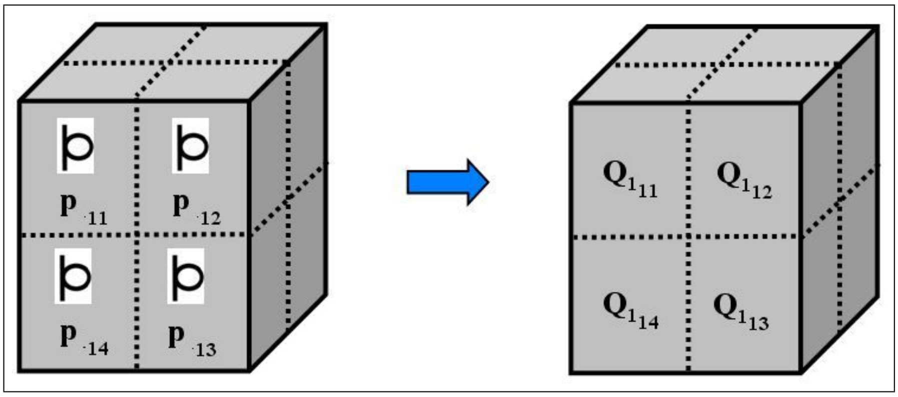

Figure 2-8: An illustration to show that the measured sound pressure p_{ij} leads to the calculated volume velocities $\vec{Q}_{1_{ij}}$ of the respective subareas A_{ij}

The limitations of the procedure set by the excitation and propagation of the relative structure-borne and airborne sound. Thus it can be assumed that this approach will not deliver useful results for a harmonic source vibration. This is the reason why the measurements results are fully correlated for this case over the whole surface and that the phases of the subarea velocities have to be identified [3,18].

In the **Figure 2-9** the insights from this section and the related principle of acoustic reciprocity (cf. section 2.2) are transferred to a car as exemplary system (cf. section 2.1).

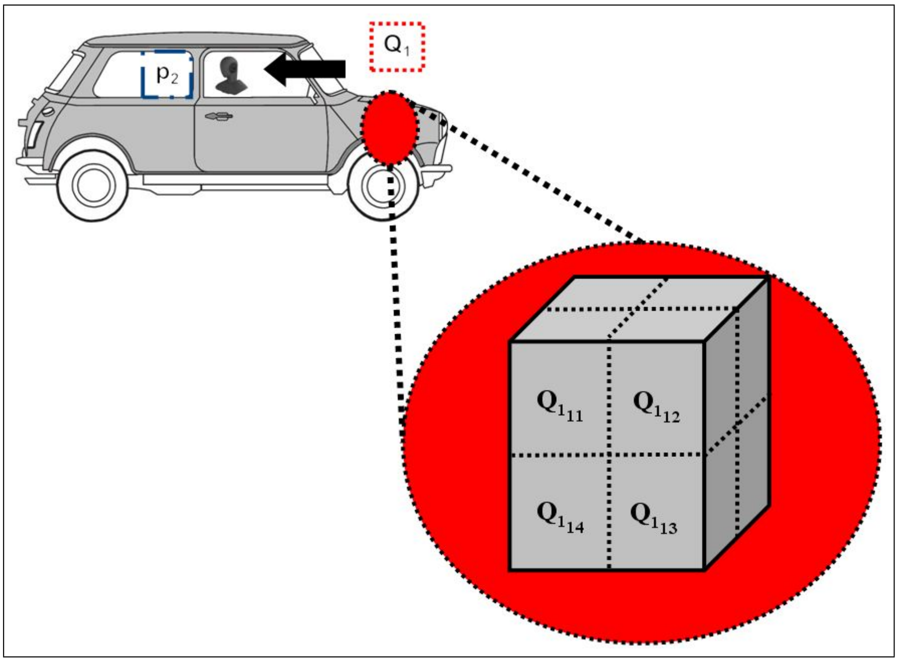

Figure 2-9: The determination of the volume velocity $\vec{Q}_{1_{ij}}$ for the respective sound source (black box) and the relation to the principle of acoustic reciprocity

The graphic in **Figure 2-9** describes the determination of the volume velocities $\vec{Q}_{1_{ij}}$ for each subarea of a component part. Since the component part is subdivided into several subareas A_{ij} (cf. **Figure 2-7**), the corresponding volume velocities $\vec{Q}_{1_{ij}}$ are summarized in a vector in **Eq. 2-1** (cf. section 2.1) and not just in a single value in the frequency range of interest.
The procedure depicted in **Figure 2-9** is normally performed on a test stand and not for the case, where the component part is applied to a car. One can therefore apply modifications and document their impact on the measurement results.

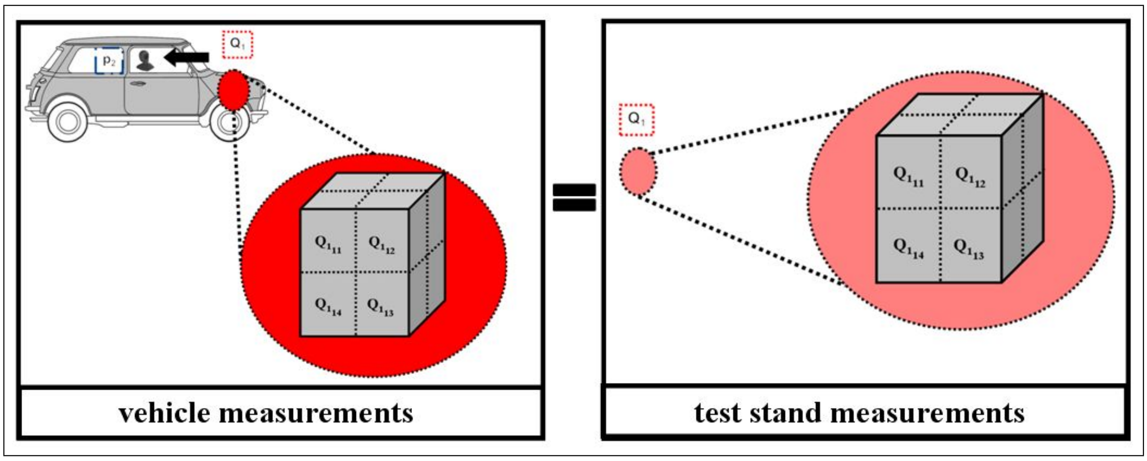

Figure 2-10: The determination of the volume velocity $\vec{Q}_1$ for a component part on a test stand and in a vehicle

Figure 2-10 shows that determining the volume velocity ($\vec{Q}_1$) of a component part on a test stand should give identical values to measurements in a vehicle. This relation is examined in more detail in section 6.3 of this work.

The larger the number of microphone positions for the black box (source element) is, the larger the vector $\vec{Q}_1$ will be.

In the following section, the theory for determining the acoustic transfer functions is presented. The measurement of the acoustic transfer functions stands for the path element of the defined source-path-receiver model (cf. section 2.1).

2.4 The acoustic transfer function (path)

As already mentioned in section 2.2, the acoustic transfer function is an essential input parameter, both for performing the principle of acoustic reciprocity (right term in **Eq. 2-1**) and for later determining the predicted sound pressure in the interior of a vehicle. Simultaneously, this investigation offers the potential to describe the acoustical and structural behavior of the system under study, with special focus on the corresponding linearity. From **Figure 1-1** (cf. section 1.1) one can conclude that identifying the acoustic transfer functions of a system constitutes the second step of this thesis.

The measurement and estimation of acoustic transfer functions is nowadays commonplace in a large number of applications. It is common in vehicle acoustics [3,7,10,15], as well as in many other sectors. The combination of acoustic transfer functions and parameters that are originally determined with the help of numerical methods is increasingly more routine. In the literature these investigations are classified as hybrid methods [3,16,19]. Reciprocally measuring the acoustic transfer functions in a car is considered to offer an important advantage in comparison to direct measurements [3,5,7,15]. This will be looked at later in this thesis in the course of finding a solution to the stated objectives.

To explain the determination of the acoustic transfer function, this part of the work again uses a car as an exemplary system. The demonstrated black box (source element) of the section 2.3 is also used here and can represent a random component part of the LuK GmbH & Co. KG.
As already discussed in section 2.1, the path element of the source-path-receiver model describes the vehicle structure of a car. The initial point for this procedure is therefore plotted in the following figure and described in more detail below.

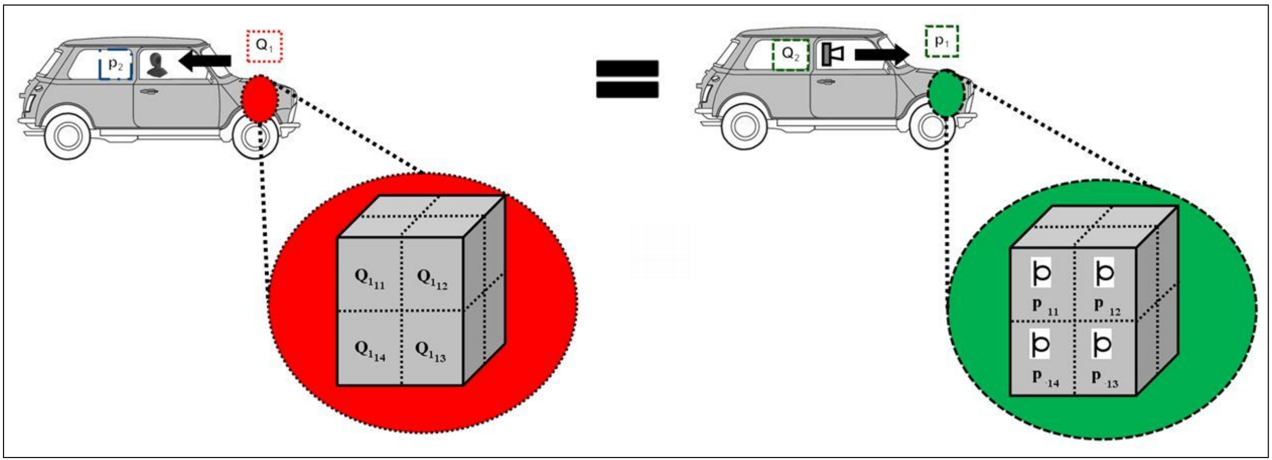

Figure 2-11: The determination of the acoustic transfer functions by using the definitions of section 2.3

In **Figure 2-11** it can be seen that the defined measurement positions for the volume velocities $\vec{Q}_{1_{ij}}$ in section 2.3 are additionally the guideline for the microphone positions p_{ij} of the acoustic transfer functions. This means that the appointed microphone positions for the sound pressure measurements of a black box (cf. section 2.3) simultaneously specify the measurement locations for the target microphones of the acoustic transfer functions. Hence one can conclude that the right term of **Eq. 2-1** has also to be a vector, plotting the related acoustic transfer functions from the loudspeaker in the interior ($\vec{Q}_2$) to the defined sound pressure measurement positions (p_{ij}) of the black box (engine bay).

The usage of acoustic reciprocity for achieving the acoustic transfer functions requires only one measurement. The time advantage of this method stands out instantly and the experiences of the literature can be verified.

In the last section of this chapter, the theory of this thesis is finalized. Therefore the course of action for predicting the sound pressure p_2 in the interior of a vehicle is explained in more detail.

2.5 The prediction of the sound pressure (receiver)

The theory behind this thesis will be completed in this part of the work. The explanations and definitions of the sections 2.2-2.4 are therefore linked together. The plotted tasks in **Figure 1-1** (cf. section 1.1) show that the prediction of the sound pressure p_2 for a component part is the third step of this thesis.

For the performance of the reciprocal transfer path analysis at the LuK GmbH & Co. KG, the radiated airborne sound from the component part in the engine bay to the receiving sound pressure at the driver's ear in the interior holds the most interest.

The results of the section 2.4 for the determination of the path element lead to the reasoning, that the sound pressure p_2 additionally has to be a vector. Thus the mathematical rules of the principle of acoustic reciprocity (cf. **Eq. 2-1**) can be ensured. The length of the vector $p_{2_{ij}}$ depends on the number of microphones p_{ij} that are used for calculating the volume velocity $\vec{Q}_{ij}$ of the component part. This means that the defined

number of microphone positions for the source element (cf. section 2.3) dominates consequently the path element (cf. section 2.4) as well as the receiver element.
Under normal circumstances it is assumed that the sound pressure $p_{2_{ij}}$ is predicted. This means that **Eq. 2-1** has to be transposed to the wanted parameter.
Another form for the principle of acoustic reciprocity in the **Eq. 2-1** is presented graphically below.

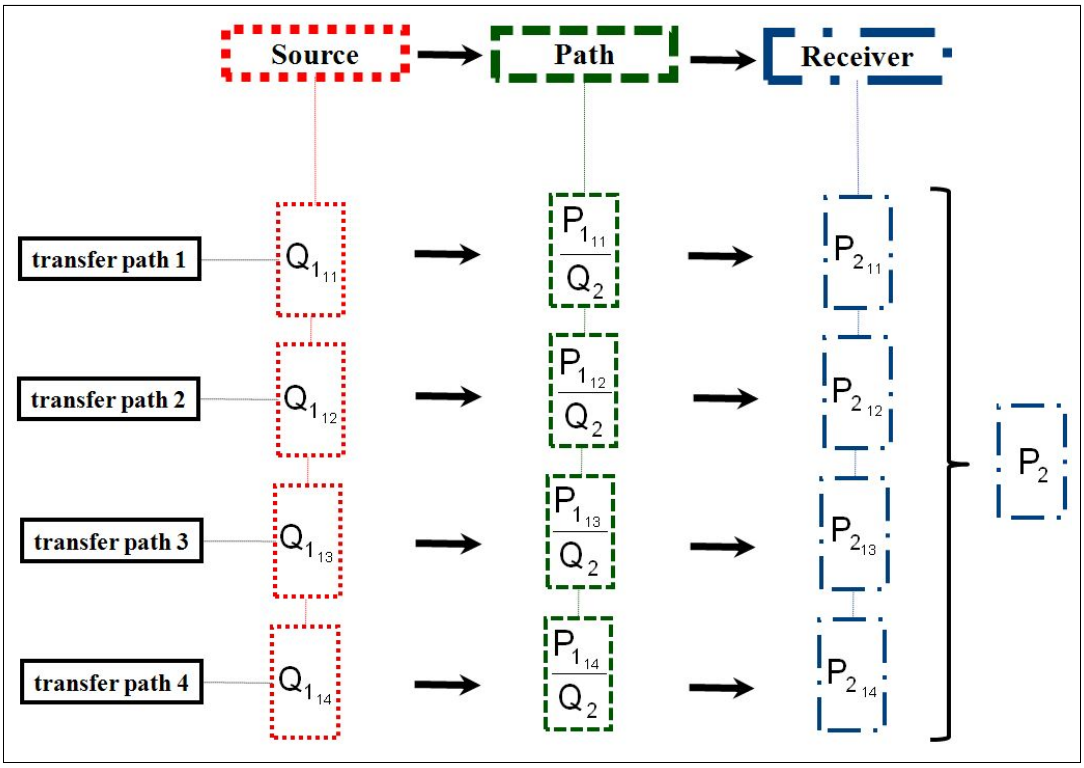

Figure 2-12: Another form of the principle of acoustic reciprocity

In **Figure 2-12** it turns out that the sound pressure $p_{2_{ij}}$ of the receiver element has to be expressed as a single value (p_2). Therefore the arithmetic mean of all sound pressure values $p_{2_{ij}}$ is required. Another important aspect that can be detected in **Figure 2-12** focuses on the explanation that the number of the defined microphone positions $p_{1_{ij}}$ (cf. section 2.3) represents additionally the number of transfer paths for the examined system.

In the next chapter the design and construction of a self-made monopole sound source is presented. Using this point source allow us to achieve the acoustic transfer functions of the path element (cf. section 2.4). The determination of the volume velocity $\vec{Q}_2$ for a self-made loudspeaker, represents an important challenge that will be faced in the chapter 4 of this thesis.

3 Design and construction of a volume sound source

As already discussed in section 2.4 the acoustic transfer functions are needed for the performance of the reciprocal transfer path analysis. According to literature is it preferable to use an omnidirectional loudspeaker to achieve the path element (cf. section 2.4) of a defined source-path-receiver model (cf. section 2.2). Unfortunately the LuK GmbH & Co. KG does not own such a monopole sound source.

This section therefore describes the design and construction of a self-made sound source, with an omnidirectional radiation characteristic in the frequency range of interest. This procedure was already performed in [7,8,9,11] and showed useful results.

3.1 Monopole sound source

In this part of the work the theoretical foundations for a monopole sound source are defined. This should enable the reader to better understand the subsequent steps in this section. Simultaneously this part of the work provides a fundamental background on this issue.

A monopole sound source is also known as a point source or an omnidirectional sound source in literature [1,2]. The "pulsating sphere" is another expression for a monopole sound source that is very often used in educational books [1,2].

The dimensions of a monopole sound source have to be small in comparison to the examined wavelength of the radiating sound [1,2]. A point sound source will basically tend to radiate sound equally in all directions. There are only some small sound sources that do not radiate the sound equally in all directions [2].

An equation for describing the sound pressure field of a monopole sound source in dependency of the distance a is basically given by

$$p(a) = ick\varrho_0 Q \frac{e^{ika}}{4\pi a} = i\frac{\varrho_0 f}{2a} Q e^{-ika} \qquad \textbf{Eq. 3-1}$$

The quotient $\frac{e^{-ika}}{4\pi a}$ of **Eq. 3-1** stands for the so called "free space" of Green's function [1,2]. The acoustic impedance of a monopole sound source can be derived by the ratio of the sound pressure p to the particle velocity u [1,2].

$$Z = \frac{p}{u} = \varrho_0 c \frac{ika}{ika+1} = \varrho_0 c \frac{1}{\frac{1}{ika}+1} \qquad \textbf{Eq. 3-2}$$

Note that **Eq. 3-2** represents the impedance of a spherical wave. The expression $\frac{1}{ika}$ in **Eq. 3-2** thereby explains a phase difference between the sound pressure and the particle velocity that exists in the near-field of a monopole sound source [2]. The spherical waves change to plane waves in the far-field. The most important parameters are the regarded distance a and the frequency f [2].

In the near-field of a monopole sound source the following condition is defined.

$$k \cdot a << 1 \qquad \textbf{Eq. 3-3}$$

The far-field is consequently described by

$$k \cdot a >> 1 \qquad \textbf{Eq. 3-4}$$

The combination of **Eq. 3-1** and **Eq. 3-2** leads to the particle velocity u.

$$u(a) = \frac{Q}{4\pi} \frac{ika + 1}{a^2} e^{-ika} \qquad \textbf{Eq. 3-5}$$

In literature it is shown that the particle velocity decreases faster in the near-field ($u \sim \frac{1}{a^2}$) than in the far-field ($u \sim \frac{1}{a}$). The sound pressure decreases constantly with $p \sim \frac{1}{a}$ in the near- and far-field of a point source [1,2]. At the same time, this procedure explains the phase shift between the sound pressure and the particle velocity in the near-field (the particle velocity in the near-field decreases faster than the sound pressure).

The sound power P of an omnidirectional sound source is given by

$$P = \frac{\rho c k^2 \left|Q^2\right|}{8\pi} \qquad \textbf{Eq. 3-6}$$

The literature releaved that a small pulsating sphere is not such an efficient radiator for smaller frequencies as for larger frequencies [1,2]. Regarding **Eq. 3-6** it can be seen that the sound power is proportional to the square of the frequency.

Looking at **Eq. 3-5** and **Eq. 3-6** it is assumed that the corresponding volume velocity of a point source, represents the major variable for achieving the above defined parameters. Chapter 4 presents a theoretical overview of the existing methods for solving this problem, as well as the practical investigations and related results.

3.2 Design of a volume sound source

This section presents the design process of a self-made volume sound source. The essential requirements as well as the theory of the existing standards are therefore discussed. This section also forms the theoretical foundation of the practical investigations in section 3.3.

To build a volume sound source a pressure chamber loudspeaker (cf. appendix II) and an attached impedance tube are generally needed [7,8,11]. The requirements of the impedance tube depend on the frequency range of interest. According to this the configuration of the dimensions and the material properties of the impedance tube can be adjusted.

The LuK GmbH & Co. KG empirically examines the complete frequency range when identifying a problem in the airborne sound and unwanted noise of the power train. Thus the impedance tube of the volume sound source should cover as broad and applicable frequency range as possible. In the following sections, the criteria that are of most importance for the calculation of the constructive dimensions of such an impedance tube are explained.

Note that the design of the volume sound source is based on the given procedure in [29].

The most important parameter for sound propagation is the upper frequency limit f_0. It is assumed that in the impedance tube only plane waves exist, as long as the upper frequency limit is not exceeded. At the end of the impedance tube the plane waves radiate as spherical waves. Thus the condition $f<f_0$ has to be fulfilled for the design of a volume sound source. According to [29] the upper frequency limit f_0 is also strongly related to the diameter d_i of the impedance tube

$$f_0 < \frac{0,585 \cdot c}{d_i} \qquad \textbf{Eq. 3-7}$$

For frequencies that are larger than the upper frequency limit ($f>f_0$), the vibration modes in the impedance tube can not be detected any longer as plane waves. These vibration modes are not of interest and will be neglected.

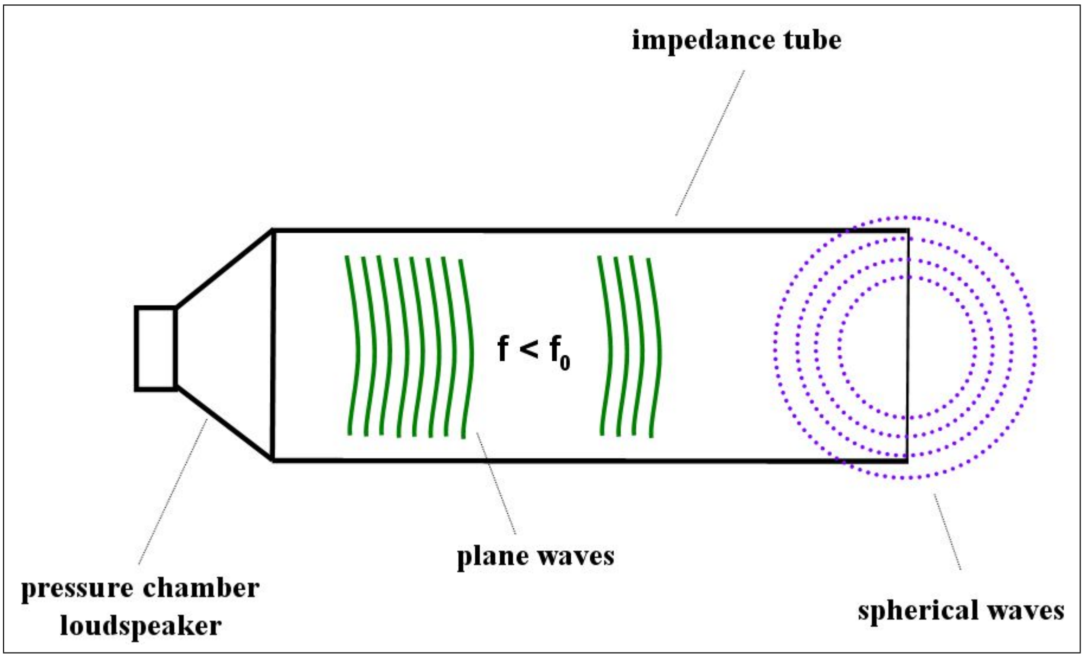

Figure 3-1: The principle of an operating volume sound source

Fulfilling the condition f$<f_0$ leads to plane waves in the impedance tube and spherical waves outside the volume sound source in the frequency range of interest (cf. **Figure 3-1**).

The transposition of **Eq. 3-7** to the inner diameter d_i of the impedance tube gives the maximal approved inner diameter in relation to the chosen upper frequency limit f_0.

$$d_i < \frac{0,585 \cdot c}{f_0} \qquad \textbf{Eq. 3-8}$$

The chosen inner diameter d_i of the impedance tube for this work was 2,2cm. Using this inner diameter and **Eq. 3-7** gives a theoretical upper frequency limit f_0 of 9 kHz for the impedance tube. Since the pressure chamber loudspeaker (cf. appendix II) only has a working frequency range up to 5,5kHz, the upper frequency limit f_0 will be not accessible.

In [29] further investigations for the design of a volume sound source are presented. Moreover, these parameters focus on the application of the two microphone method and the corresponding determination of the impedance inside. Knowing these quantities gives the possibility to achieve the volume velocity under operation of a volume sound source [7,8,6,20,21,22,23,24,29]. Since the two microphone method was not applied primarily for determining the volume velocity of a volume sound source (cf. Chapter 4), this procedure is discussed in more detail in appendix I.

According to the given requirements of this section, in the subsequent section the construction of the volume sound source and the relating impedance tube are presented.

3.3 Construction of a volume sound source

Here the construction of the self-made volume sound source is described and discussed. The reader is given an overview of the practical investigation of this objective. The usage of the calculated data from section 3.2 concerning the design of a volume sound source offers a helpful guideline for the further steps.

As already mentioned in section 3.2, the volume sound source consists of two major parts. First a driver is required, which is represented by a pressure chamber loudspeaker (cf. appendix II), second an attached impedance tube is needed. The impedance tube is made of aluminum and discussed more in detail in this section.

Note that the impedance tube was constructed by the author and was designed according to the given procedure in section 3.2. Thus an inner diameter of 2,2cm was chosen. The length of the impedance tube covers 45cm.

Since the pressure chamber loudspeaker has a special thread integrated, an adapter was constructed. Hence the pressure chamber loudspeaker and the impedance tube can be connected flush.
The following figure shows an illustration of the completed volume sound source.

Figure 3-2: The self-made volume sound source

To check the results of the directivity of the volume sound source, in the next section the corresponding values are pictured.

3.4 Directivity of a volume sound source

In this part of the thesis the directivity of the constructed volume sound source is studied. First the general procedure for determining the radiation pattern of a sound source is explained. Then, the corresponding results of the investigated measurements are presented and discussed. The main aim of this section is thereby to find the applicable frequency range of the constructed monopole sound source.

In this way a comparison between the theoretical design (cf. section 3.2), the manufacturer's data (cf. appendix II) and the existing practicability can be performed.

For the determination of the directivity of a loudspeaker is it in principle necessary to define a circular area around the examined sound source with a constant distance (diameter). In a next step the microphone positions are arranged on the circular surface. A larger number of microphone positions leads in principle to a smoother procedure, with a greater accuracy of the measurement results. Basically this effect is comparable to the sampling frequency in the field of signal processing. The larger the sampling frequency is, the more exact is the signal scanned.

The midpoint of the circular sphere corresponds to the acoustic center of the examined sound source. According to the standards given in [29], is it useful to drive the loudspeaker with a random noise during the measurements. Note that the directivity of a loudspeaker is generally measured three dimensionally. Hence the results for the defined xy plane, the xz plane and the yz plane have to be examined separately.

The above described procedure is explained more in detail in [27]. Additionally, permitted deviations for the radiation pattern of a volume sound source are listed in this literature.

frequency [kHz]	0,5	0,63	0,8	1,0	1,25	1,6	2,0	2,5	3,15	4,0	5,0	6,3
max. deviation [dB]	± 1	± 1	± 1	± 3	± 3	± 3	± 5	± 5	± 5	± 6	± 6	± 6

Figure 3-3: The permitted deviations for the radiation pattern of a volume sound source [27]

Using the given maximum deviations in **Figure 3-3** explains that the directivity of a monopole sound source is allowed to differ from the average of the respective octave band [27]. The values in **Figure 3-3** show that the allowed deviations increase for larger frequencies.

In the rest of this section the results of the directivity for the volume sound source are described. The examined frequency range for the radiation pattern starts at octave band mid-frequency 500Hz and moves up to 6,3kHz. The results are generally plotted in third octave bands. The measurement procedure was basically performed by following the above described course of action.
Since all defined planes in this work showed the same results, the xy plane is only pictured in the following.

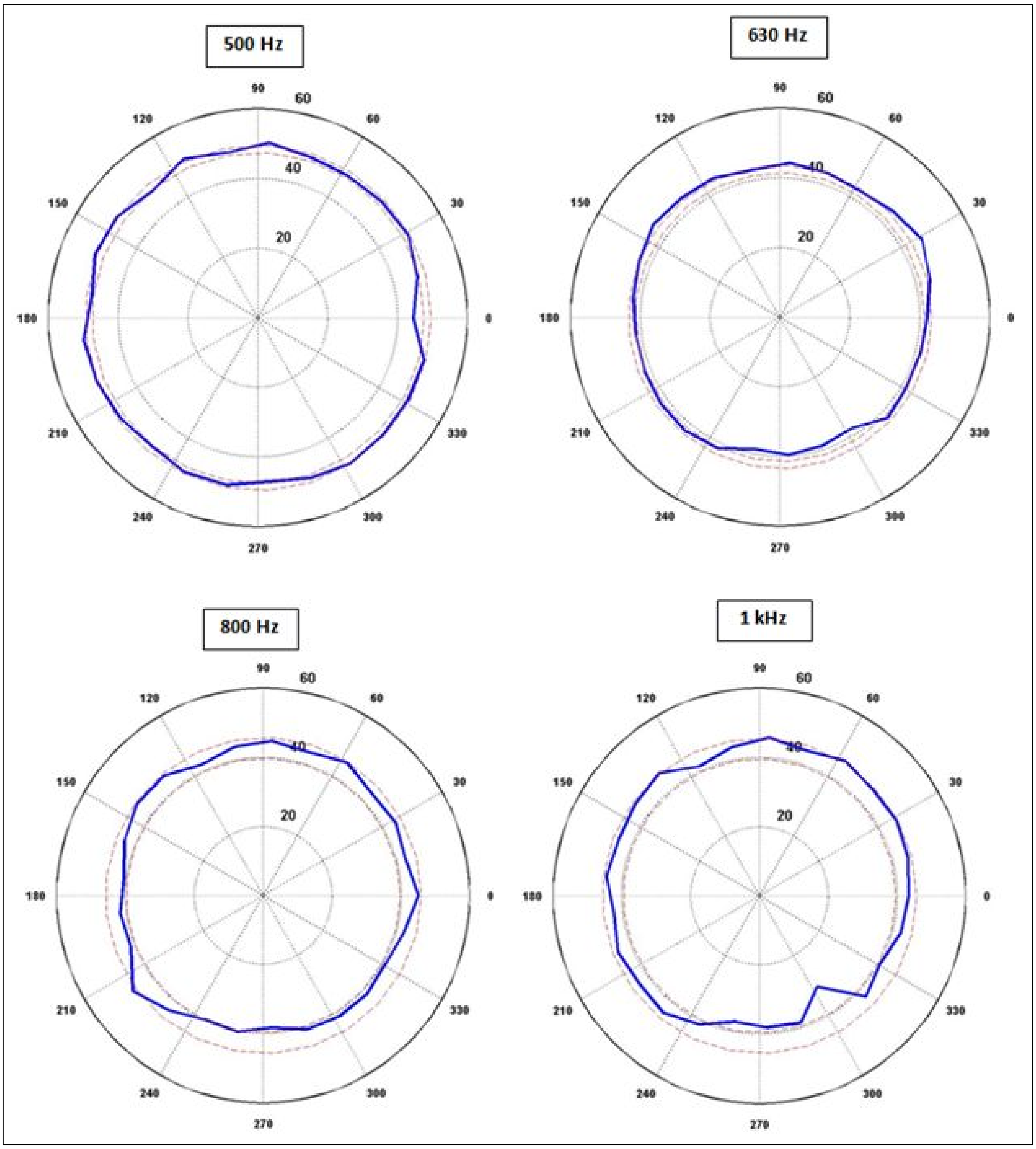

Figure 3-4: The radiation pattern of the self constructed volume sound source (0,5-1kHz)

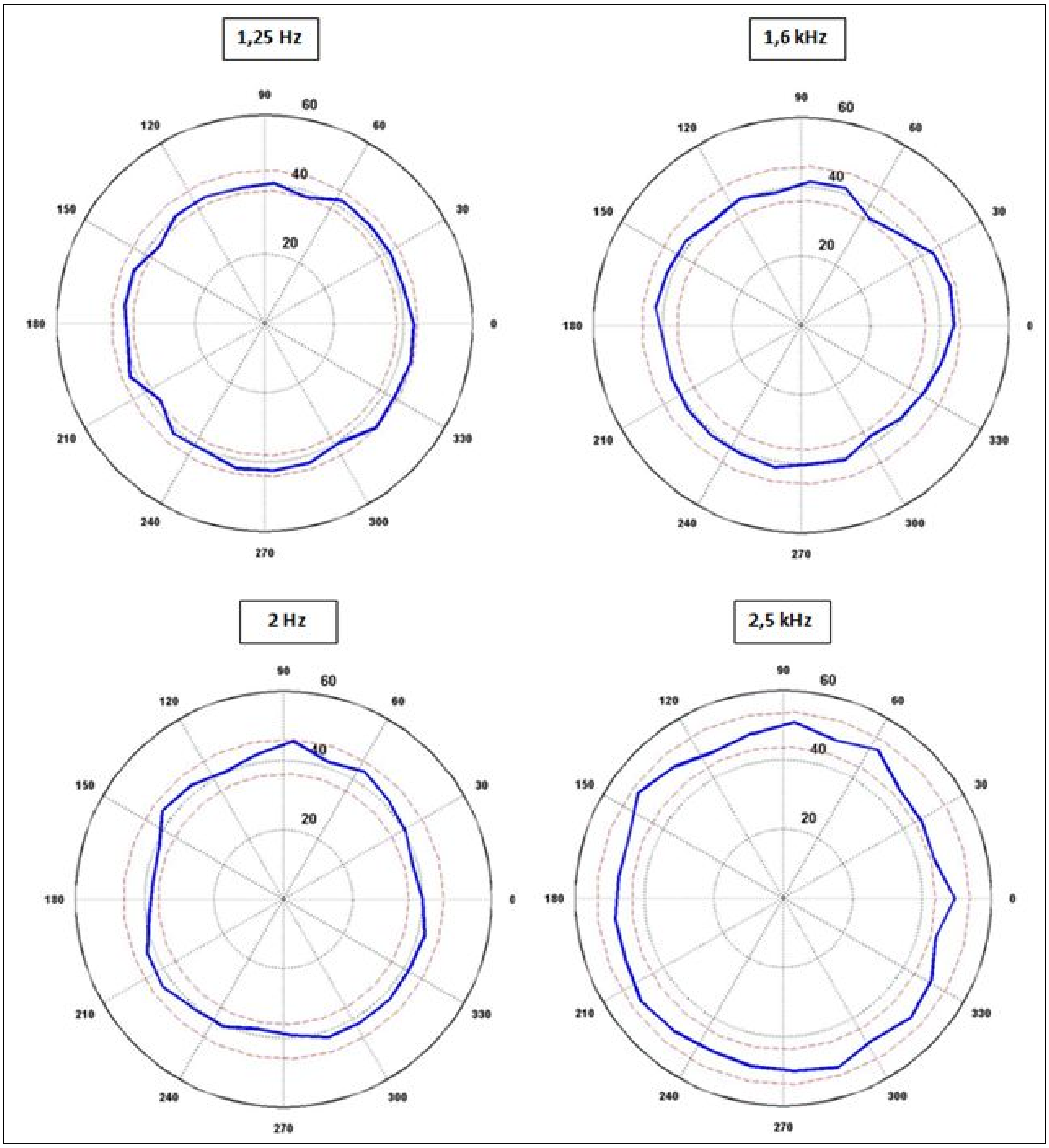

Figure 3-5: The radiation pattern of the self constructed volume sound source (1,25-2,5kHz)

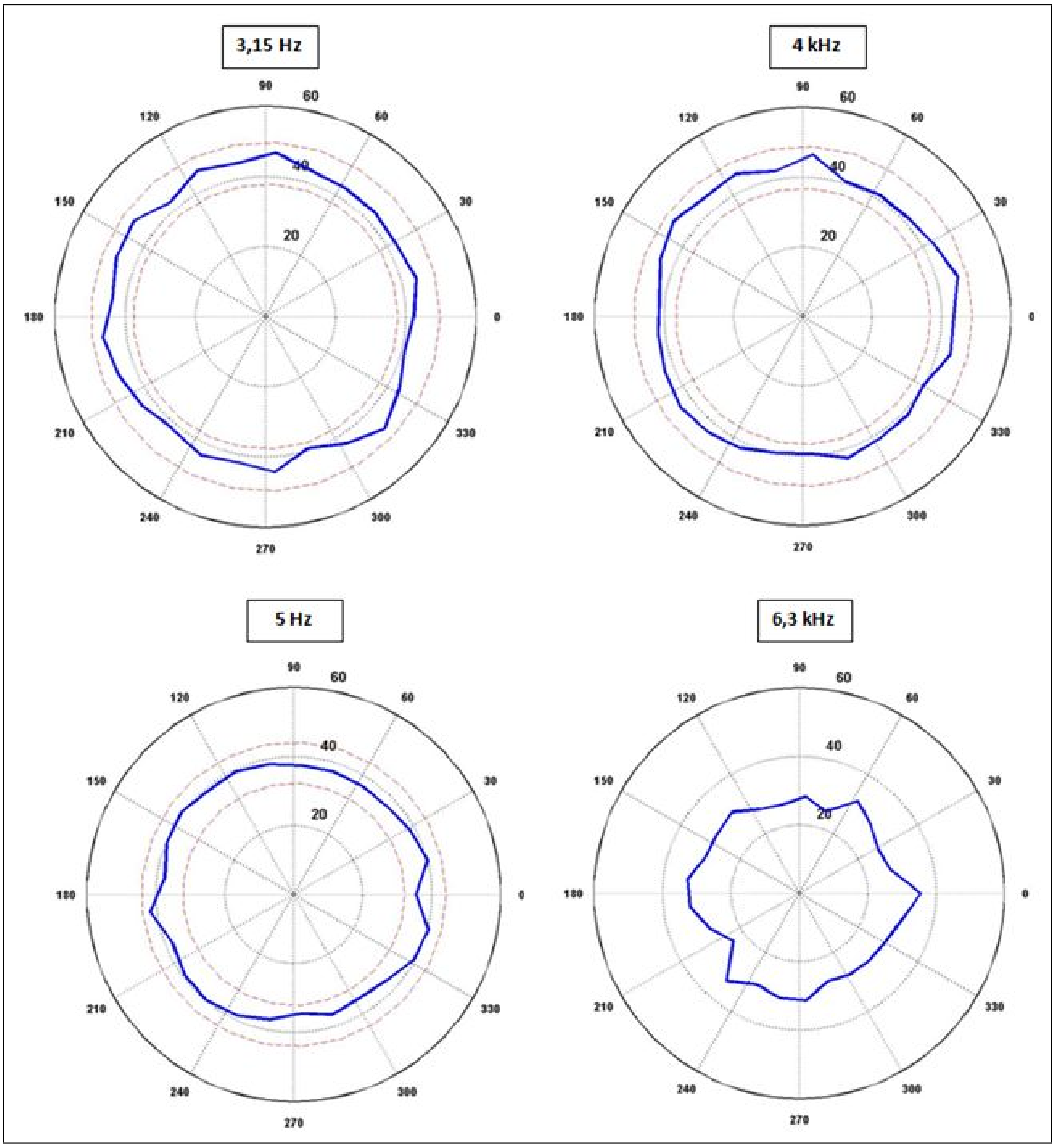

Figure 3-6: The radiation pattern of the self constructed volume sound source (3,15-6,3kHz)

The results in **Figure 3-4, 3-5 and 3-6** show that the volume sound source (cf. section 3.2 and section 3.3) has an omnidirectional radiation characteristic from 0,5 up to 5kHz. This conclusion is verified by the deviations given in **Figure 3-3**. Since the pressure chamber loudspeaker only works up to a frequency of 5,5kHz (cf. appendix II), the applicable frequency range of the constructed volume sound source is assumed to range from 0,5 to 5,5kHz.

However, comparing the results in **Figure 3-4, 3-5 and 3-6** with the plots of the radiation pattern of the LMS Q-Source (cf. appendix V), strong fluctuations can be detected.

The next chapter focuses on the problem of determining the operating volume velocity Q_2 of the volume sound source. Thus the input parameter for measuring the acoustic transfer functions (cf. section 2.4) should be detected.

4 Determination for the volume velocity of a volume sound source

As already mentioned in the previous sections the volume velocity Q_2 of a volume sound source is needed to perform the reciprocal transfer path analysis (cf. section 2.4). Using a point source and the corresponding volume velocity allows determining the acoustic transfer functions that represent simultaneously the path element of the defined source-path-receiver model in section 2.2 (cf. **Figure 2-5**).

The volume sound source (cf. chapter 3) that will be investigated in this work is basically not able to detect the acting volume velocity automatically under operation (cf. appendix IV). An overview of the most important approaches for this issue is provided, and four methods are applied practically.

4.1 Background

Detecting the volume velocity of a loudspeaker under load represents an important parameter for performing the reciprocal transfer path analysis. Investigating the volume velocity of a loudspeaker and the respective sound pressure gives the particular acoustic transfer function (cf. section 2.4). Currently there are a number of manufacturers that offer complete loudspeaker systems including adapters and integrated sensors, that can already determine the volume velocity under operation (cf. appendix IV). Since the volume velocity is in principle related to the particle velocity (cf. **Eq. 3-5** in section 3.1), the sensors of the manufacturers estimate in most cases the latter quantity.
In the following itemization the results of a literature study that solve the above described problem are presented.

- **PU sound intensity probe**

  The PU sound intensity probe is produced by Microflown and was originally developed for sound intensity measurements. There are a number of variations of the PU sound intensity probes that simultaneously enable a broad area of applications. The conclusions of a literature study showed that all PU sound intensity probes work according to a generalized principle [4,17].
  In the center of the PU sound intensity probe there is a sensor that measures the particle velocity. The functionality behind this technique relies on the technology of a heating wire. Therefore two thin wires are placed very close to each other. The particle velocity depends basically on the orientation and represents a quantity of how fast the molecules move around the position of rest. It can be concluded that the wire that is adjusted closer in the direction of the source will cool down faster,

because of the higher particle velocity. The temperature differences between the wires causes in turn a current flow between them [4,17].

Since the LuK GmbH & Co. KG does not own a PU sound intensity probe, this work will not contain any experiments or results with this approach.

- **Two microphone method**

  The two microphone method is an alternative technique for achieving the particle velocity and the corresponding volume velocity of a volume sound source under operation. The two microphone method is also called the standing wave ratio [20,21,22,23,24]. The principle of the two microphone method is presented in [20,21,22,23,24,29] where it is called the transfer function method. The estimation of a sound absorption coefficient in a Kundt's tube is strongly related to the two microphone method [20,21,22,24,29].

  The use of this technique requires an impedance tube attached to the pressure chamber loudspeaker (cf. appendix II), as well as at least one microphone. In appendix I the theory for this technique is sketched (cf. **Figure I-1**) and explained in more detail. Using the measured sound pressure at both microphones inside the impedance tube enables the calculation of the wanted particle velocity under operation (cf. appendix I).

  It turns out that the central problem of the two microphone method displays the impedance change at the end of the impedance tube [6,7,8,11]. The sensitivity of the applied microphones also plays an enormous role. However, there are a number of dissertations and papers that report the successful investigation of this approach [3,6,7,8,11,].

  Note that the two microphone method was partially studied in this thesis, but unfortunately it delviered insufficient results. As already mentioned above, the corresponding theory and conclusions of the practical application are therefore presented shortly in appendix I.

- **Electroacoustic reciprocity**

  For the appliance of the principle of electroacoustic reciprocity a second loudspeaker is usually needed (cf. **Figure 4-1**). When the first loudspeaker (Q_1) generates a sound pressure, the membrane of the second loudspeaker will vibrate. Measuring this vibration of the second loudspeaker leads to an open-circuit voltage U_2. In a next step the sensitivity of the first sound source to the second loudspeaker is required. Hence it is necessary to measure the sound pressure p_1 very close to the membrane of the first loudspeaker as well as the existing current I_2 of the second sound source.
  In the following figure the principle of electroacoustic reciprocity is sketched by introducing a quadrupole.

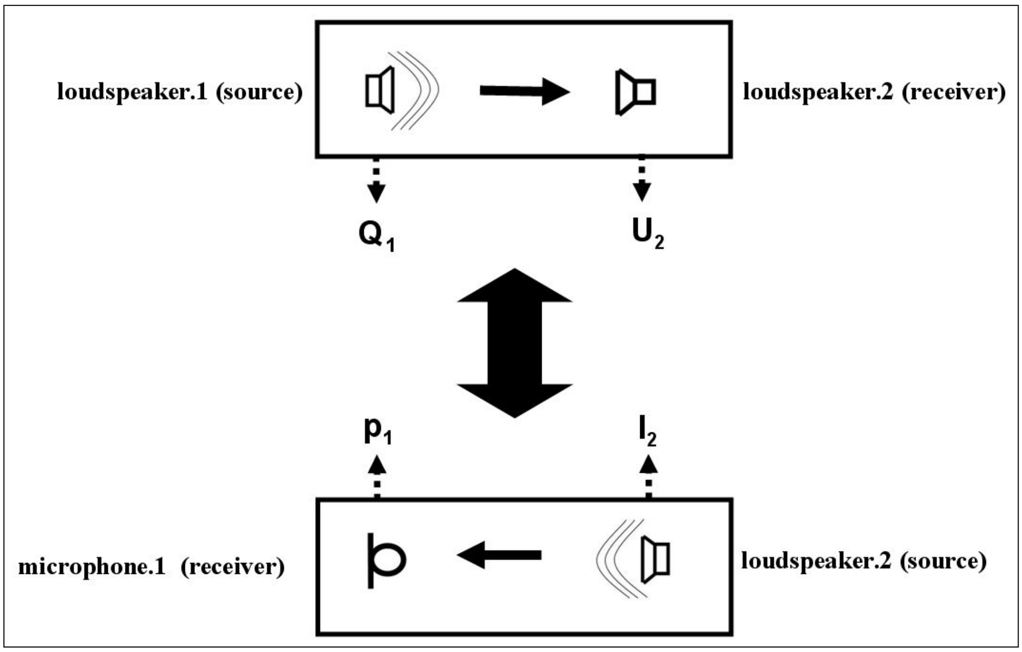

Figure 4-1: The principle of electroacoustic reciprocity

The principle of electroacoustic reciprocity was successfully applied in a number of publications. Normally this procedure is used for the calibration of microphones. Assuming that the volume velocity of the first loudspeaker (Q_1) displays the wanted parameter gives

$$Q_1 = \frac{U_2}{p_1/I_2} \qquad \textbf{Eq. 4-1}$$

When using the principle of electroacoustic reciprocity it is important to use loudspeakers and microphones that have the same radiation pattern.

The principle of electroacoustic reciprocity was not applied in this thesis.

- **Diffuse field method**

  For the diffuse field method a reverberation chamber or another spatial environment is required, in which a diffuse sound field can be built. With the use of the procedure described in [25,26] one can determine the sound power P of a volume sound source (cf. **Eq. 3-6** in section 3.1).

  The sound power is in turn linked to the volume velocity of a sound source by introducing

  $$Q = \sqrt{\frac{4\pi}{\rho c k^2} P} \qquad \textbf{Eq. 4-2}$$

  The sound power represents a useful parameter for achieving the volume velocity of a sound source. Assuming that the loudspeaker has an omnidirectional directivity means that the sound power is equal at every point around the sound source. For this reason the sound power does not decrease with distance, unlike the sound pressure or particle velocity (cf. section 3.1). It can be concluded that the usage of the diffuse field method requires only one single measurement.

  The LuK GmbH & Co. KG unfortunately does not have the testing environments to perform the diffuse field method. Thus this approach is not considered further in this work.

- **Scanning laser vibrometer**

  The scanning laser vibrometer (LDV) is a non contact optical measurement technique for studying the vibrational behavior of structures. The principle of this method originates in the Doppler shift of the laser beam frequency due to motion on the surface. It is generally true that measurements with an LDV deliver more precise results, comapred to the measurement results of an accelerometer. Since the laser does not interact with the structure, this assumption can be made.

  With the LDV one can measure the acting particle velocity under operation on the area of the loudspeakers membrane. The volume velocity can be moreover calculated with the respective cross section of the loudspeaker's membrane, as will be shown later in section 4.2 (cf. **Eq. 4-3**).

  Since in this work a pressure chamber loudspeaker is used as sound source, the LDV can't be applied (cf. appendix II). The pressure chamber shields the loudspeaker's membrane and is therefore not accessible for any measurements with a LDV. Thus the LDV will be not studied further in this thesis.

In the next section the theory of determining the volume velocity of a loudspeaker is presented. The core issues focus on the applied methods that solve this important subtask.

4.2 Theory

The volume velocity (Q) of a loudspeaker generally describes how fast a flowing gas propagates through a defined cross section S [1,2]. For a volume sound source the particle velocity u and the cross section S are of interest. Using these both parameters leads to

$$Q = \int_s u \, dS \qquad \textbf{Eq. 4-3}$$

Since the cross section of a volume sound source is normally measurable, the determination of the particle velocity represents the major problem for obtaining the volume velocity in the **Eq. 4-3**. As already mentioned in section 4.1, the manufacturers of loudspeaker systems focus more intensively on the determination of the particle velocity (cf. appendix IV).

In a next step four approaches that will be performed practically in the section 4.3 presented theoretically.

4.2.1 Monopole approach

The monopole approach can't be found with this name in literature. The reason is that the title of this principle was defined by the author in the course of this work. Introducing this declaration makes it easier later to refer to this method and to perform comparisons with other investigations. The monopole approach was already successfully performed in [6,18].

The background of the monopole approach relies on the procedures in [6,18] and requires that a monopole sound source is studied in the frequency range of interest. Hence the volume velocity of a point source can be determined with the usage of

$$Q = \frac{p(a) 2a}{i \varrho_0 f e^{ika}} \qquad \textbf{Eq. 4-4}$$

Regarding **Eq. 4-4** the sound pressure is generally linked to the volume velocity of a monopole sound source (cf. **Eq. 3-1** in section 3.1). Furthermore **Eq. 4-4** defines that measuring the acting sound pressure of a point source, in a known distance a, gives the respective volume velocity (Q).

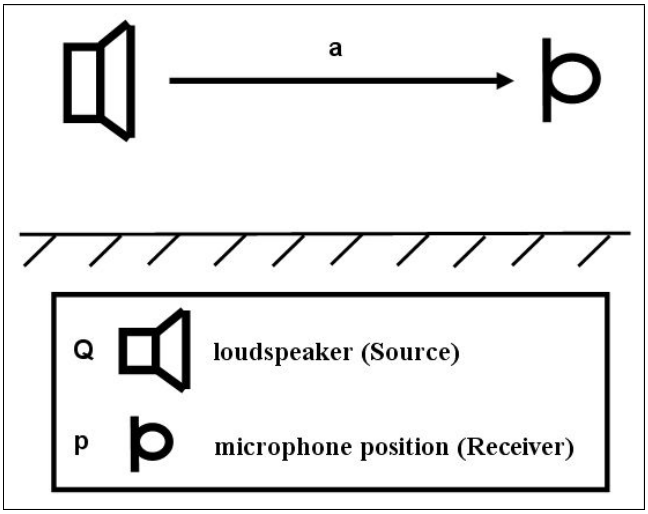

Figure 4-2: The procedure for the monopole approach

According to literature the sound power P stands for the strength of a loudspeaker and therefore represents an input parameter of a sound source [1,2]. The existing relations for the diffuse field method (cf. section 4.1 and **Eq. 4-2**) show that the volume velocity is also related to the sound power and likewise describes the source strength of a loudspeaker. Hence it can be assumed that the volume velocity is equal at every point around a sound source, as long as the loudspeaker behaves omnidirectional. However, regarding the existing terms in [6], it is shown that the appliance of the monopole approach also depends on the distance to the volume sound source and the frequency range of interest. This issue can be explained by regarding the near-field and far-field conditions of a monopole sound source (cf. section 3.1).
Note that the acoustical impedance Z_0 is generally linked with the sound pressure and the particle velocity

$$Z_0 = \frac{p}{u} \qquad \textbf{Eq. 4-5}$$

Measuring the sound pressure p in a known distance a, in the far-field of an acting point sound source, enables the calculation of the volume velocity as shown in **Eq. 4-4** or to investigate **Eq. 4-5**. Focusing on the latter procedure, it can be summarized that the particle velocity (cf. **Eq. 4-5**) will lead finally to the wanted volume velocity of a loudspeaker by additionally using the cross section S of the impedance tube (cf. **Eq. 4-3**).

The theory above presented can be regarded as very helpful for the performance of the monopole approach and the relating calculation of the volume velocity for a point source. Therefore it is essential to define first whether the measurement position of the sound pressure can be detected in the near-field or in the far-field region. Depending on this condition the necessary calculation procedures can be investigated.

In section 4.3.1 the monopole approach is carried out in practice. The distances between the volume sound source and the receiver microphone are therefore varied. Hence the distribution of the volume velocity for varying distances can be examined more carefully.

The next subsection presents the second approach for determining the wanted volume velocity of a loudspeaker.

4.2.2 Free-field method

The free-field method represents an alternative approach for the estimation of the volume velocity of a volume sound source. Note that the free-field method is strongly related to the monopole approach already presented (cf. section 4.2.1) and is based on the standard given in [1,2]. According to the literature the free-field method is normally used for determining the sound power P of a considered sound source.

For the performance of the free-field method it is assumed that the considered volume sound source has an omnidirectional radiation characteristic in the frequency range of interest. The free-field method requires a free-field room or at least a semi free-field room. Since the LuK GmbH & Co. KG has a semi free-field room (cf. appendix II), this investigation will be applied practically in the section 4.3.2.

For studying the free-field method 20 microphone positions are arranged with a constant distance a around the regarded volume sound source. The figure below therefore shows 20 microphone positions that have to be measured according to [27], for the application of the free-field method in a semi free-field room.

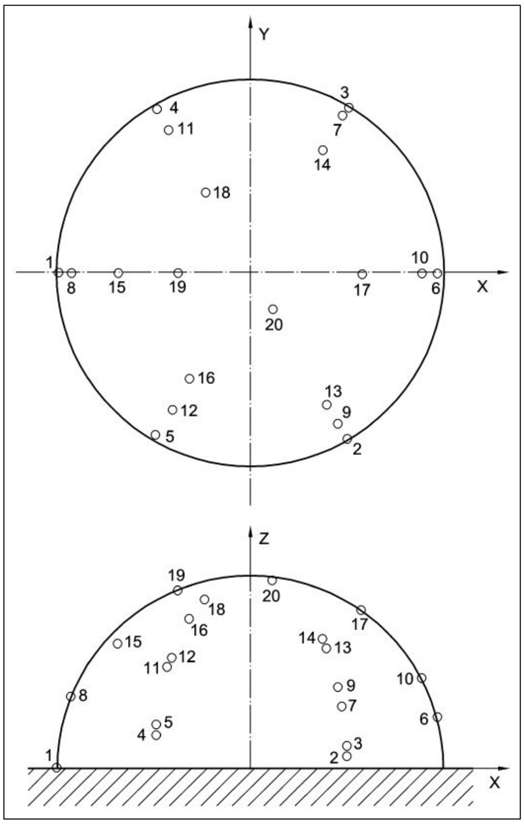

Figure 4-3: The microphone positions for the application of the free-field method in a semi free field room [27]

The configuration of these measurement positions depends only on the constant distance a between the sound source and the microphones (cf. appendix III). Using the coordinates in **Figure III-1** (cf. appendix III) it can be seen that the distance between each microphone position and the sound source is equal for all measurement positions. The origin of the coordinate system in appendix III represents the acoustic center of the investigated volume sound source.

Using the theory of the monopole approach (cf. section 4.2.1) leads to the relation

$$Q = \frac{\overline{p}(a)2a}{i\varrho_0 f e^{ika}} \qquad \textbf{Eq. 4-6}$$

Regarding **Eq. 4-6** it can be seen that the free-field method uses in general the same foundations as the monopole approach (cf. **Eq. 4-4** in section 4.2.1). The only difference between the monopole approach and the free-field method (cf. **Eq. 4-6**) is that the latter uses the averaged sound pressure of 20 microphone positions, instead of one microphone position.

Simultaneously this adjustment allows the assumption that the free-field method can be regarded as more time consuming in comparison to the monopole approach. However, it is also assumed that the volume velocity of the free-field method will give more precise results, because of the higher number of microphone positions, than the monopole approach.

The limitations of the free-field method focus especially on the distance a of the defined measurement positions [27]. Hence the distance between the microphones and the sound source should be

- at least one meter
- larger than a fourth of the wavelength for the lowest frequency of interest
- exceed more than the double of the largest dimension of the respective volume sound source

As mentioned above section 4.3.2 displays the corresponding results for the application of the free-field method.

In the next subsection the third investigation for estimating the volume velocity of a volume sound source is briefly discussed.

4.2.3 Transfer function method

The transfer function method is in addition a useful tool for determining the volume velocity of a volume sound source. The basic concept of this approach relies on the principle of vibroacoustic reciprocity which in the literature is also called the reciprocity method for the mechanical-acoustical transfer function [3,7,12,13]. The theory of the transfer function method is explained in the rest of this section by introducing a box as exemplary system.

A box is generally not a complex system like a car and therefore has a minimized number of possible sources of error.

For the investigation of the transfer function method two measurements are required. The subsequent figure pictures the procedure of the transfer function method schematically.

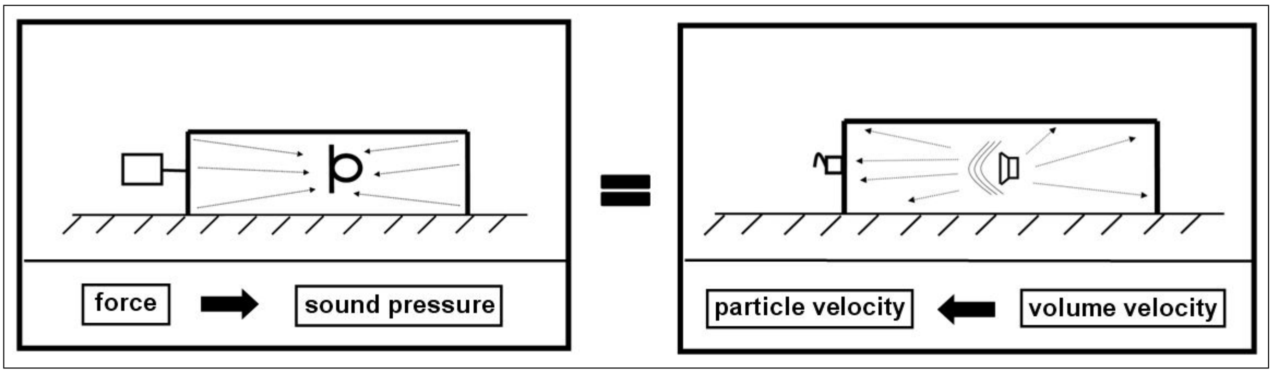

Figure 4-4: The appliance of the transfer function method

For the first step of the transfer function method a direct measurement is performed. Therefore, on the box's surface a force (e.g. with an impact hammer or a shaker) is introduced to the system at a defined excitation position and the sound pressure measured at a defined receiver position. Using this relation leads mathematically to the expression

$$H_{direct} = H_1 = \left.\frac{p}{F}\right|_{Q=0} \qquad \textbf{Eq. 4-7}$$

To determine the reciprocal transfer function, the positions of excitation and receiver of the first case (H_{direct}) have to be interchanged. Thus the examined volume sound source is placed in the box and driven by a chosen excitation signal, whereas an accelerometer records the receiving structural vibration at the original excitation position of the transfer function H_1. Investigating this relation gives

$$H_{reciprocal} = H_2 = \left.\frac{u}{Q}\right|_{F=0} \qquad \textbf{Eq. 4-8}$$

Combining **Eq. 4-7** and **Eq. 4-8** of both transfer functions by applying the principle of acoustic reciprocity (cf. section 4.2.2) yields

$$H_{direct} = H_{reciprocal} = H_1 = H_2 = \left.\frac{p}{F}\right|_{Q=0} = \left.\frac{u}{Q}\right|_{F=0} \qquad \textbf{Eq. 4-9}$$

Regarding **Eq. 4-9** it turns out that all parameters can be measured with the exception of the volume velocity Q of the volume sound source. Therefore **Eq. 4-9** can be transposed to

$$Q = \frac{u}{H_{direct}} = \frac{u}{H_2} = \frac{u}{\frac{p}{F}} \quad \textbf{Eq. 4-10}$$

With the help of **Eq. 4-10** the wanted volume velocity Q can be calculated. The transfer function method additionally represents an investigation that is studied practically in this work (cf. section 4.3.3).
The use of the transfer function method requires basically that the directivity pattern of the loudspeaker and microphone be identical [3,7,12,13]. Another important aspect focuses on the linearity of the considered system. It is assumed that the system behaves linearly when the transfer functions H_1 and H_2 show the same values.

Finally, the subsequent section investigates the principle of acoustic reciprocity.

4.2.4 Acoustic reciprocity

The principle of the acoustic reciprocity was already presented and discussed in detail in section 2.2 (cf. **Eq. 2-1**). This approach requires two loudspeakers and one microphone. The volume velocity of the second loudspeaker (Q_2) should be known or at least detectable under operation. The principle of acoustic reciprocity was successful investigated in [3,12,18] and is schematically pictured in the following.

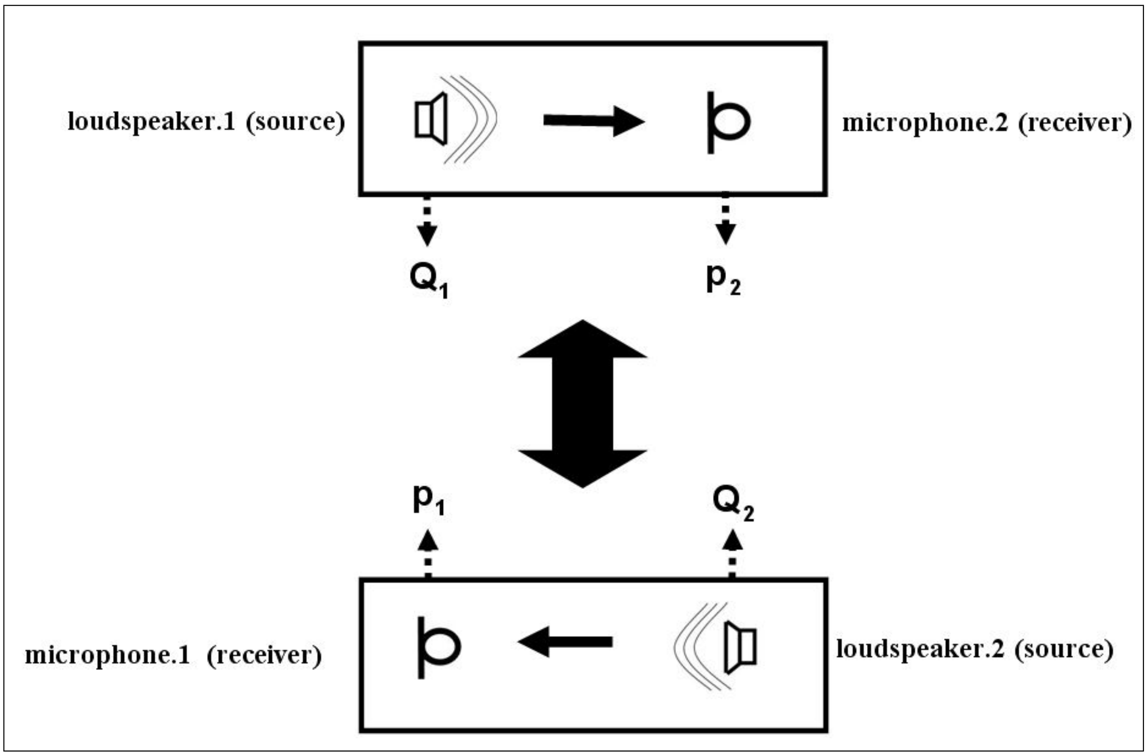

Figure 4-5: The principle of acoustic reciprocity

Regarding the quadrupole in **Figure 4-5** it can be seen that the principle of the acoustic reciprocity is based on two measurements. The first loudspeaker ($\vec{Q}_1$) is driven while a microphone records the sound pressure (p_2) at a defined receiver position.

$$H_{pQ_1} = \left.\frac{p_2}{\vec{Q}_1}\right|_{\vec{Q}_2=0} \quad \textbf{Eq. 4-11}$$

In a second measurement the positions of excitation and receiver are interchanged and a second loudspeaker ($\vec{Q}_2$) has to be introduced. According to this the reciprocal transfer function can be built with

$$H_{pQ_2} = \left.\frac{p_1}{\vec{Q}_2}\right|_{\vec{Q}_1=0} \qquad \textbf{Eq. 4-12}$$

Assuming that $\vec{Q}_1$ stands for the wanted volume velocity gives

$$\vec{Q}_1 = \frac{p_2}{H_{pQ_1}} = \frac{p_2}{p_1/\vec{Q}_2} \qquad \textbf{Eq. 4-13}$$

As already mentioned in section 2.2 it is required that the sound source $\vec{Q}_1$ and microphone p_1 give an identical directivity. The volume velocity of the principle of acoustic reciprocity was practically investigated in this work (cf. section 4.3.4).

The upcoming section presents the results of the measurements for determining the volume velocity of a constructed loudspeaker (cf. chapter 3). The procedures described in sections 4.2.1 - 4.2.4 are therefore applied and studied more intensively.

4.3 Measurements

In this section the above presented methods for determining the acting volume velocity of a volume sound source are applied practically. Hence the functionality of the volume sound source (cf. chapter 3) can be verified and the related volume velocity under operation should be achieved. At the same time, in this section the theories (cf. section 4.2) and their applicability are proven. Hence the most useful method should be found, to successfully investigate the reciprocal transfer path analysis later in this thesis.

As mentioned in section 2.5 the volume velocity is an important input parameter for later performing the reciprocal transfer path analysis. Here the practicality of the monopole approach (cf. section 4.2.1) is pointed out, and the free-field method (cf. section 4.2.2), the transfer function method (cf. section 4.2.3) and the acoustic reciprocity (cf. section 4.2.4) are studied. Thus a comparison between the existing approaches can be performed, focusing especially on the results, as well as on the usability and validity of the methods.

Note that all measurement results that will be regarded in this work are plotted in a frequency range from 0,5 up to 7 kHz. The volume sound source is, in all approaches, driven by a random noise.

The measurement equipment that is used for the determination of the volume velocity is listed briefly in appendix II.

4.3.1 Monopole approach

The monopole approach originates in the application of **Eq. 4-4** and was theoretically explained in the section 4.2.1. For the performance of the monopole approach only one microphone measurement is necessary. Knowing the distance between the volume sound source and the microphone position gives the possibility to perform this technique.

To study the behavior of this method, a concept test, with varying distances between the source (volume sound source) and the receiver (microphone) was arranged. The measurement setup is represented schematically in the figure below.

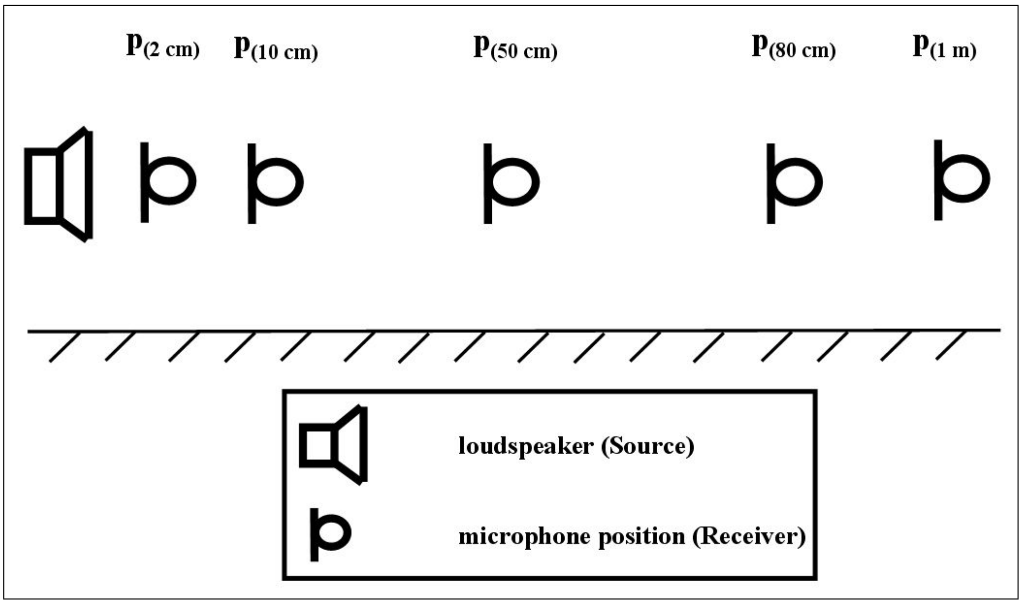

Figure 4-6: The measurement setup for the monopole approach

As can be seen in **Figure 4-6**, for this concept test the receiving sound pressure was recorded at five microphone positions. The only differences between the respective measurements are the varying distances between the sending volume sound source and the receiving microphone. Therefore the dependency of the distance and the relating sound pressure of the microphone positions, as well as the resulting.

The height for all measurement positions was equal and had the same orientation to the volume sound source.

The following figure pictures the derived volume velocities for the respective measurement positions of the concept test.

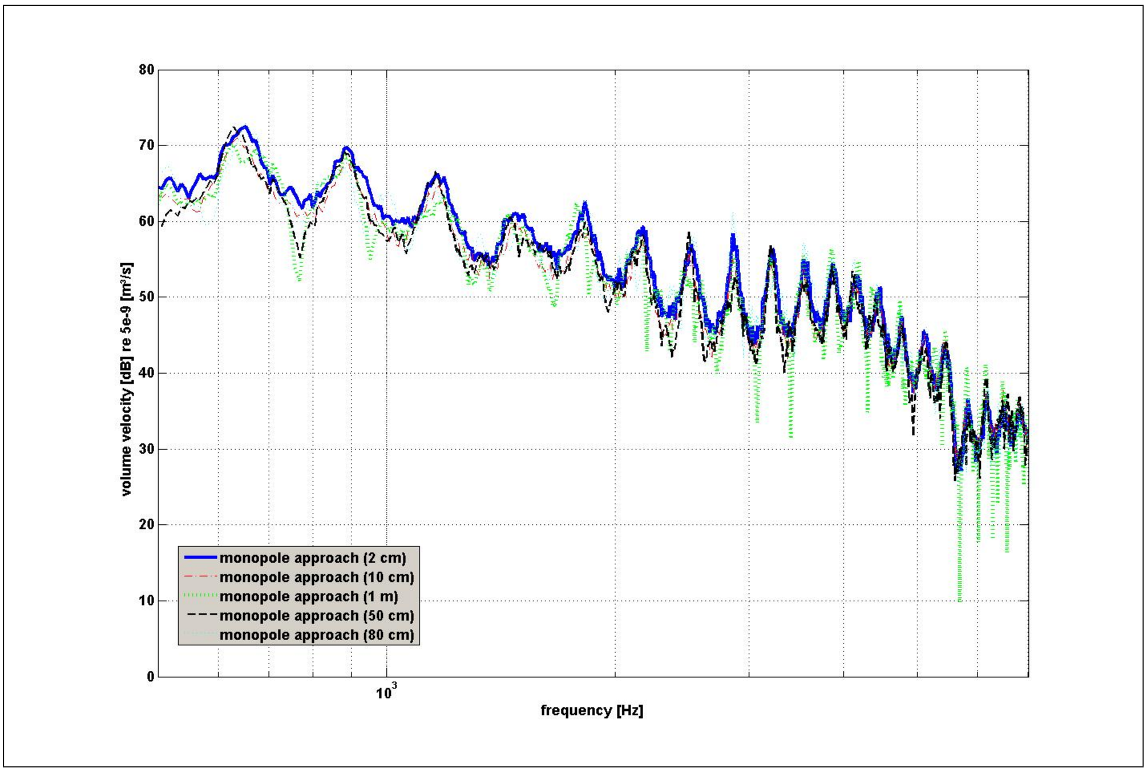

Figure 4-7: The volume velocities for the concept test (monopole approach)

The graphs of the volume velocities in the above presented figure show in principle an identical behavior. The fluctuations of the levels between the varying distances are at most 10dB. It is assumed that the peaks in **Figure 4-7** represent the existing resonances in the impedance tube of the volume sound source (cf. **Figure I-3** in appendix I). The resonance peaks are shifted for lower frequencies and different distances. Note that the resonance peaks generally increase with frequency. This could be related to an increasing modal density for larger frequencies. The bigger the gap between the volume sound source and the microphone position is, the more distinct are the drops in the results. All graphs show a clear drop at a frequency of 5,5kHz. According to the manufacturer's data of the pressure chamber loudspeaker (cf. appendix II), this frequency stands for the cutoff frequency of the volume sound source. Thus one can conclude that frequencies above 5,5kHz are not useful for the application of the volume sound source and the corresponding acoustic transfer function. The lower limitation can be detected at a frequency of 650Hz.

The results of the monopole approach in **Figure 4-7** imply that it is generally enough to place, in a distance of two centimeters from the tube opening, a microphone for determining the wanted volume velocity.

In a next step the other methods for determining the volume velocity of a volume sound source of section 4.2 are applied and studied practically.

4.3.2 Free-field method

In this section the free-field method (cf. section 4.2.2) is examined in a laboratory test. This is basically another approach for determining the volume velocity of a volume sound source. For the performance of the free-field method the sound pressure was measured at 20 defined measurement positions (cf. appendix III). Using the recorded data and **Eq. 4-6** (cf. section 4.2.2) allows us to calculate the wanted parameter. The distance a between the volume sound source and the 20 microphone positions was thereby constantly one meter (a=1m).

Since the volume velocity is averaged over 20 microphone positions, it is expected that the free-field method delivers the most precise results of all approaches.

Therefore the sound pressure levels of the free-field method are plotted.

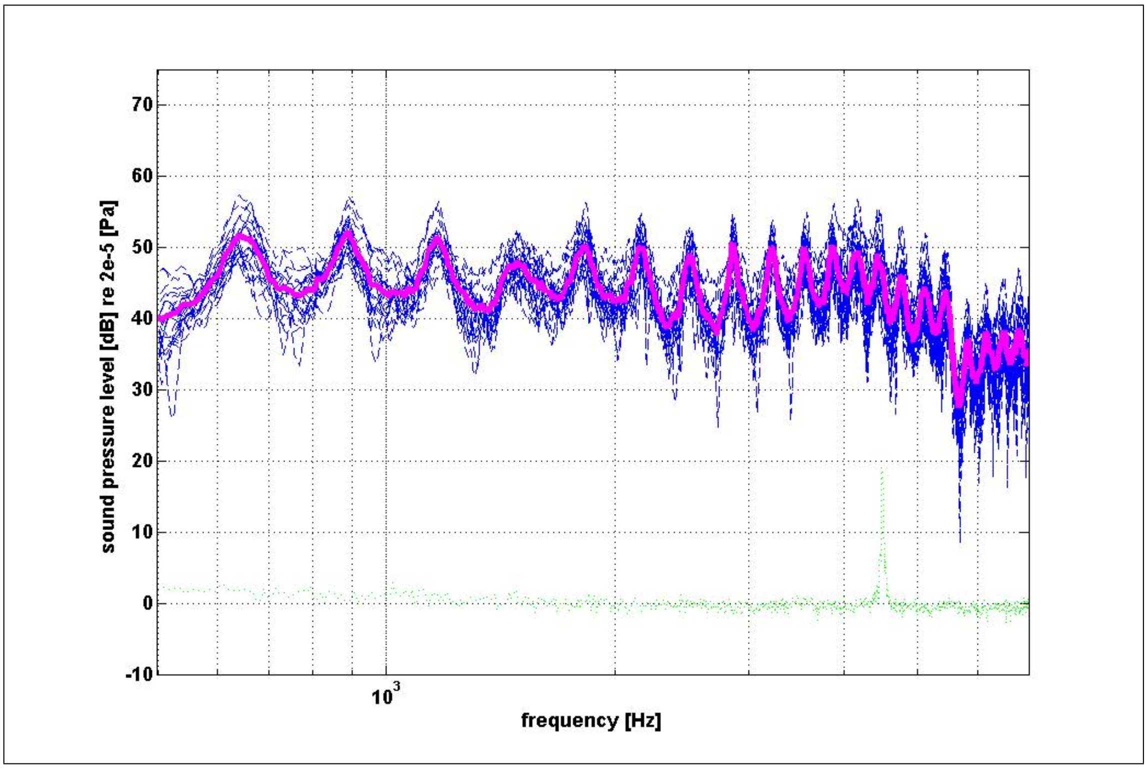

Figure 4-8: The sound pressure levels for the free-field method

The measurement results for the sound pressure levels of the free-field method show that the distance to the background noise level is sufficient for all measurements. The fluctuation of the levels between the respective measurement positions is up to 10dB and increases with frequency. The cutoff frequency of the volume sound source can be likewise detected at 5,5kHz. The curve shape in **Figure 4-8** is identical to the plot in **Figure 4-7** (cf. section 4.3.1), showing the existing resonances and dips in the results. The increasing differences at higher frequencies of the sound pressure levels in **Figure 4-8** indicate that the volume sound source does not have an ideal omnidirectional radiation characteristic for these frequencies (cf. section 3.4).

The next figure below analyzes the corresponding results of the volume velocity for the free-field method.

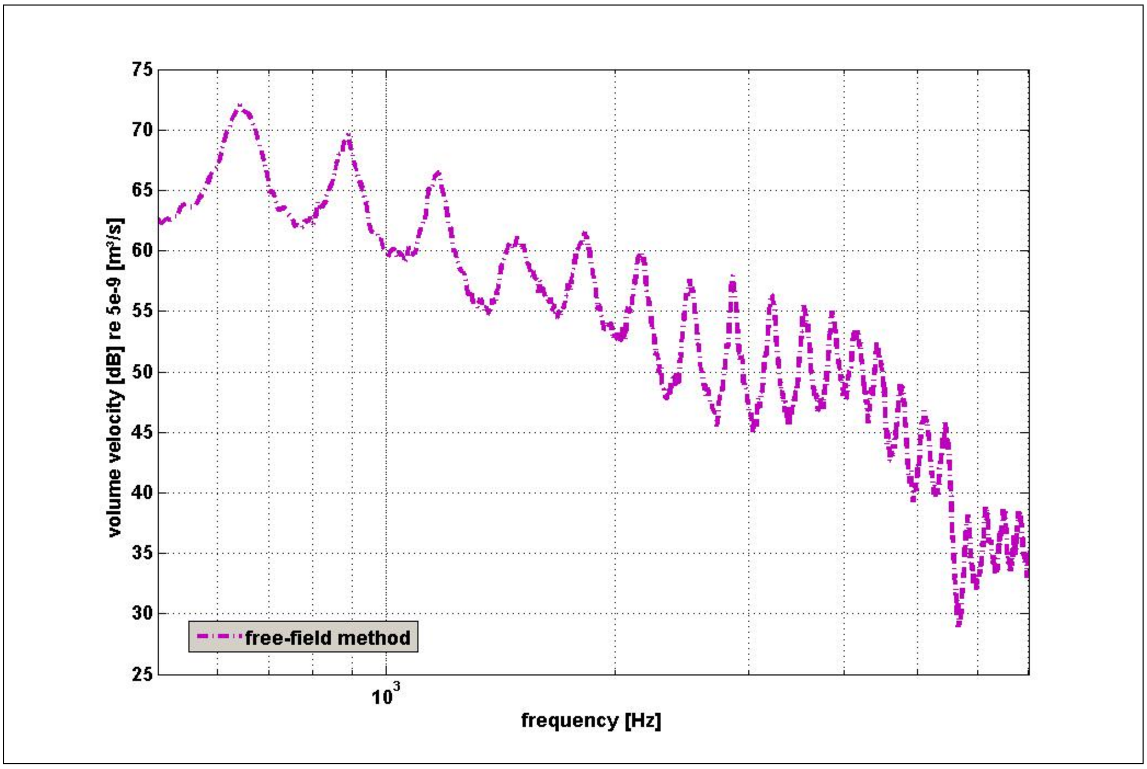

Figure 4-9: The volume velocity for the free-field method

Focusing on the pictured volume velocity in **Figure 4-9** shows that the volume velocity level generally decreases with frequency. The resonance peaks of the sound pressure levels in **Figure 4-8** are also visible for the volume velocity of the free-field method. Since the sound pressure represents the major input parameter for the volume velocity, this behavior was expected (cf. **Eq. 4-6** in section 4.2.2). Hence the cutoff frequency of the pressure chamber loudspeaker at 5,5kHz is again detectable.

Later in this section the usability of the free-field method, in comparison to the other approaches, is checked by examining the results of all investigations (cf. section 4.3.5).

The subsequent section focuses on the results of the volume velocity for the transfer function method.

4.3.3 Transfer function method

The procedure pictured in **Figure 4-4** (cf. section 4.2.3) represents the foundation for the investigation of the transfer function method. The figure below therefore shows therefore the direct measurement of the transfer function.

A shaker excites the structure of the box, and the receiving sound pressure is measured at a defined receiver position in the box.

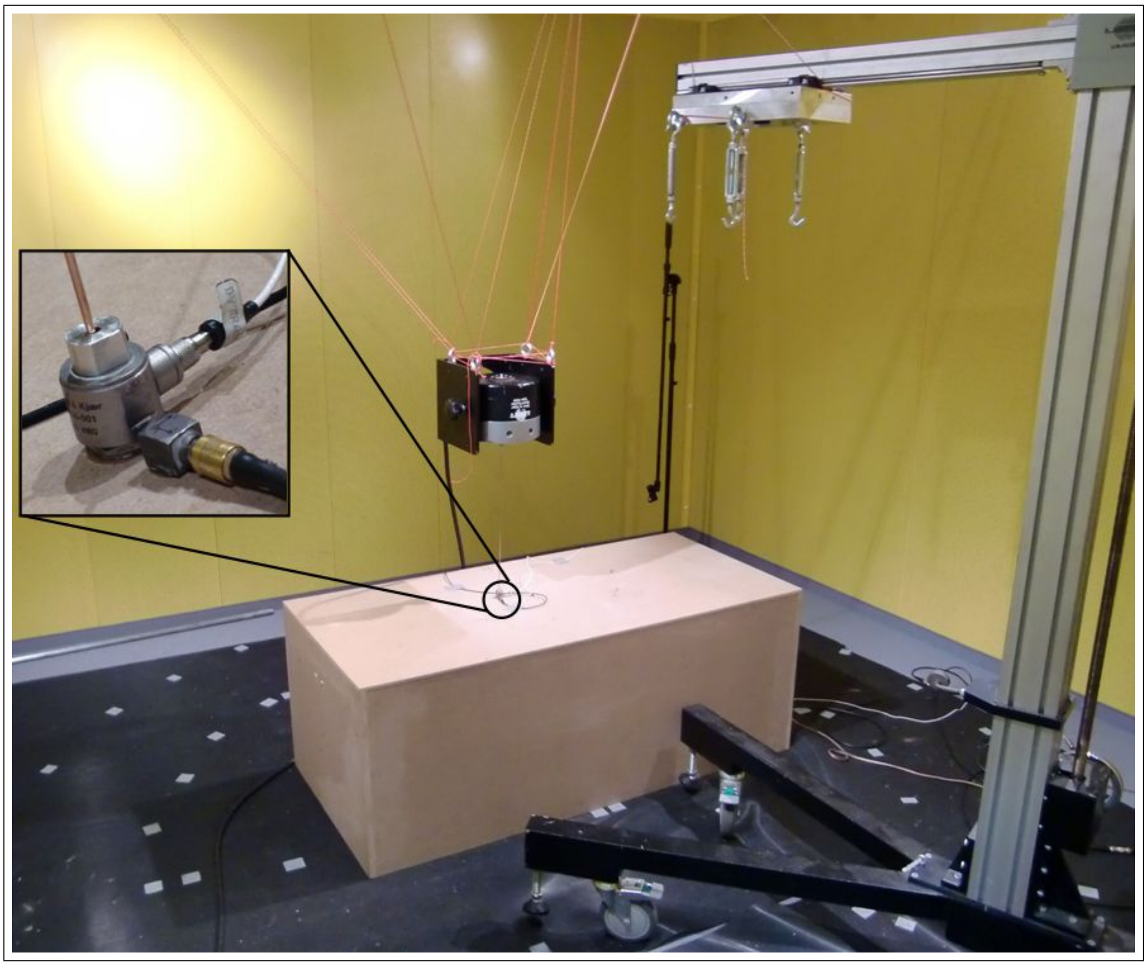

Figure 4-10: The measurement setup for the transfer function method (direct transfer function)

For the reciprocal measurement the self-made volume sound source is placed in the box and an accelerometer measures the receiving structural vibrations (cf. **Figure 4-10**).

Using **Eq. 4-10** (cf. section 4.3.3) for the transfer function method allows us to calculate the wanted parameter.

The next figure therefore plots the corresponding result of the volume velocity.

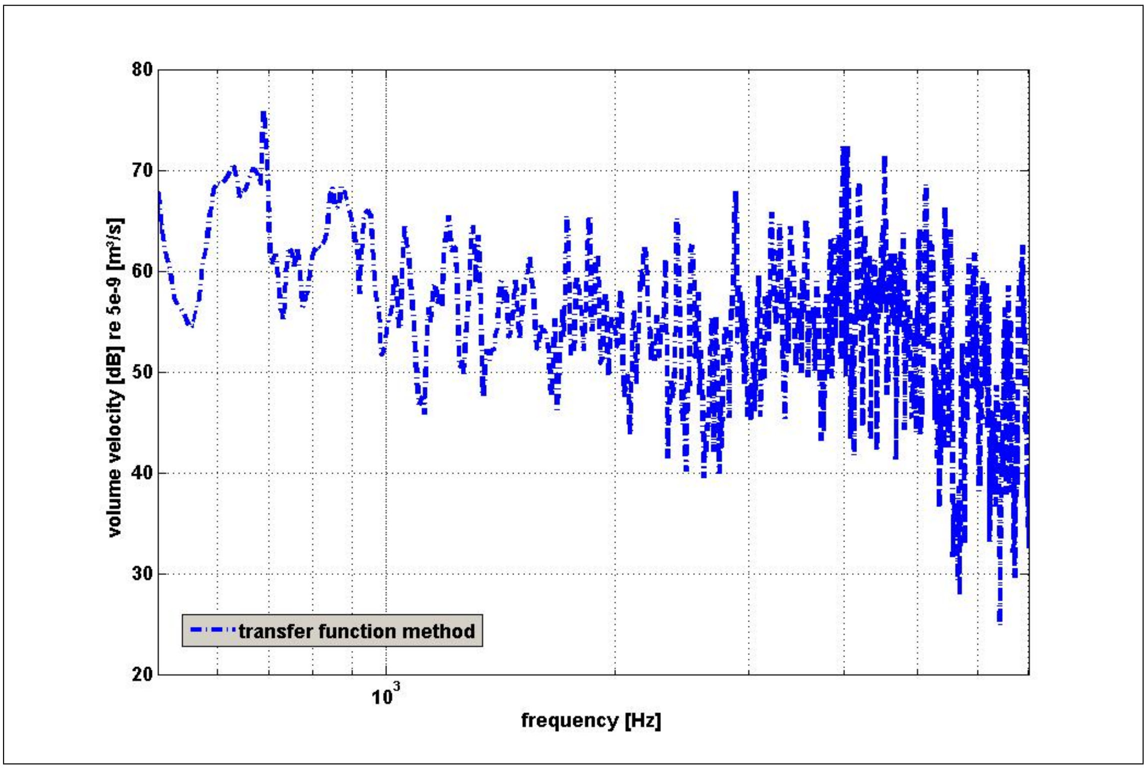

Figure 4-11: The volume velocity for the transfer function method

The volume velocity in **Figure 4-11** was determined through the appliance of the transfer function method and the related principle of mechanical-acoustical reciprocity (cf. **Eq. 4-10** in section 4.2.3). From the graph in **Figure 4-11** it can be seen that the volume velocity shows distinct fluctuations; it is assumed that the used structure has a strong influence on the results. Since the transfer function method originates in the interaction of the structure-borne and airborne sound, the relating results have to be considered carefully. The larger the frequency, the more visible is the divergence in the results.

In section 4.3.5 the transfer function method is compared with the residual investigations of this work and a classification can be done more easily.

Checking the linearity of the system (box) could represent a further step for studying the transfer function method in more detail; this, that is neglected at this stage of the work.

In the subsequent section the principle of acoustic reciprocity (cf. section 4.2.4) is investigated practically.

4.3.4 Acoustic reciprocity

To perform the principle of acoustic reciprocity (cf. section 4.2.4) a second monopole sound source ($\vec{Q}_2$) was required, as presented in appendix II. The measurement setup for investigating the principle of the acoustic reciprocity, as well as the corresponding theory, was already discussed in section 4.2.4. The distance between the first and the second measurement positions was constantly one meter and both sound sources were driven with the same excitation signal (random noise).

It is assumed that the sound source $\vec{Q}_1$ and the microphone p_1 have the same radiation pattern (omnidirectional) in the frequency range of interest.

In the next figure the respective volume velocity for the principle of the acoustic reciprocity is pictured.

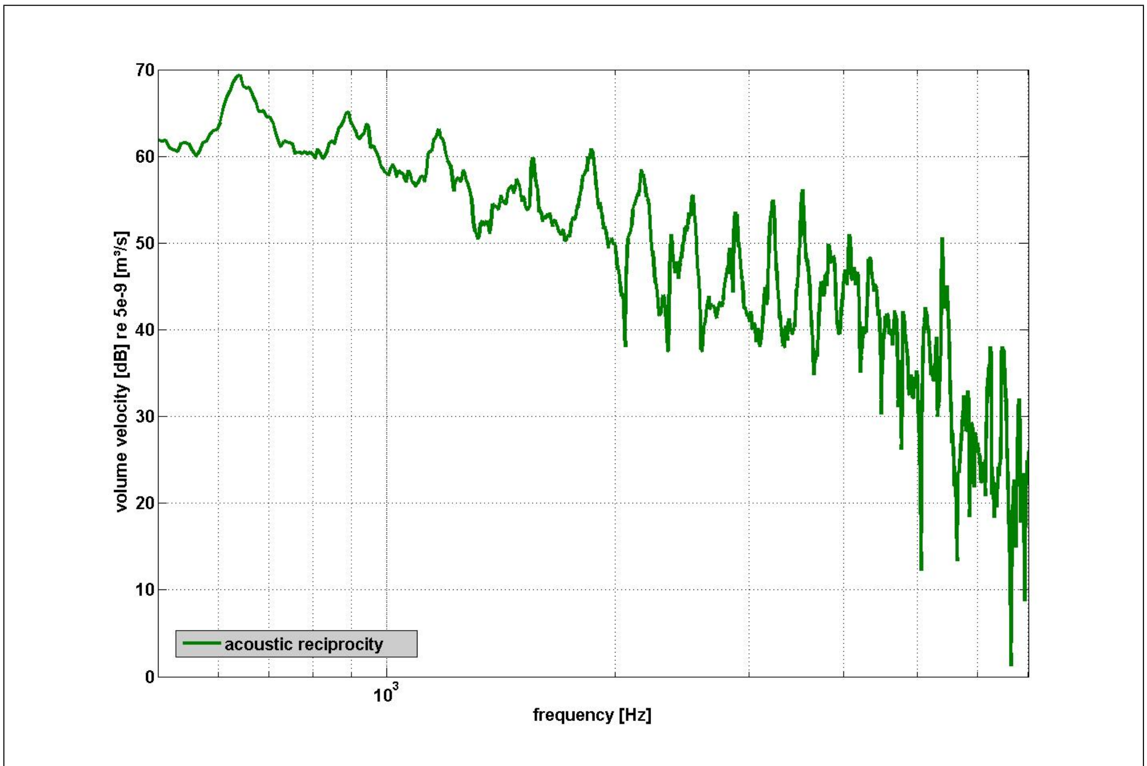

Figure 4-12: The volume velocity for the principle of acoustic reciprocity

The graph of the volume velocity for the investigation of the principle of the acoustic reciprocity in **Figure 4-12** gives a suitable result. Comparing the outcome in **Figure 4-12** to the residual results in the sections 4.3.1-4.3.3 shows good concordance. However, the larger the frequency, the more visible are the drops for the volume velocity of the principle of acoustic reciprocity in **Figure 4-12**.

The next section performs a summarized comparison between the applied methods of this section that focuses on the determination of the volume velocity for a constructed volume sound source (cf. chapter 3).

4.3.5 Comparison

As mentioned above, this section presents an overview of the investigated procedures of this thesis to achieve the volume velocity of a constructed volume sound source (cf. sections 4.3.1 - 4.3.4). Hence the most suitable method should be found for the further performance of the reciprocal transfer path analysis in this work (cf. chapter 5 and chapter 6).

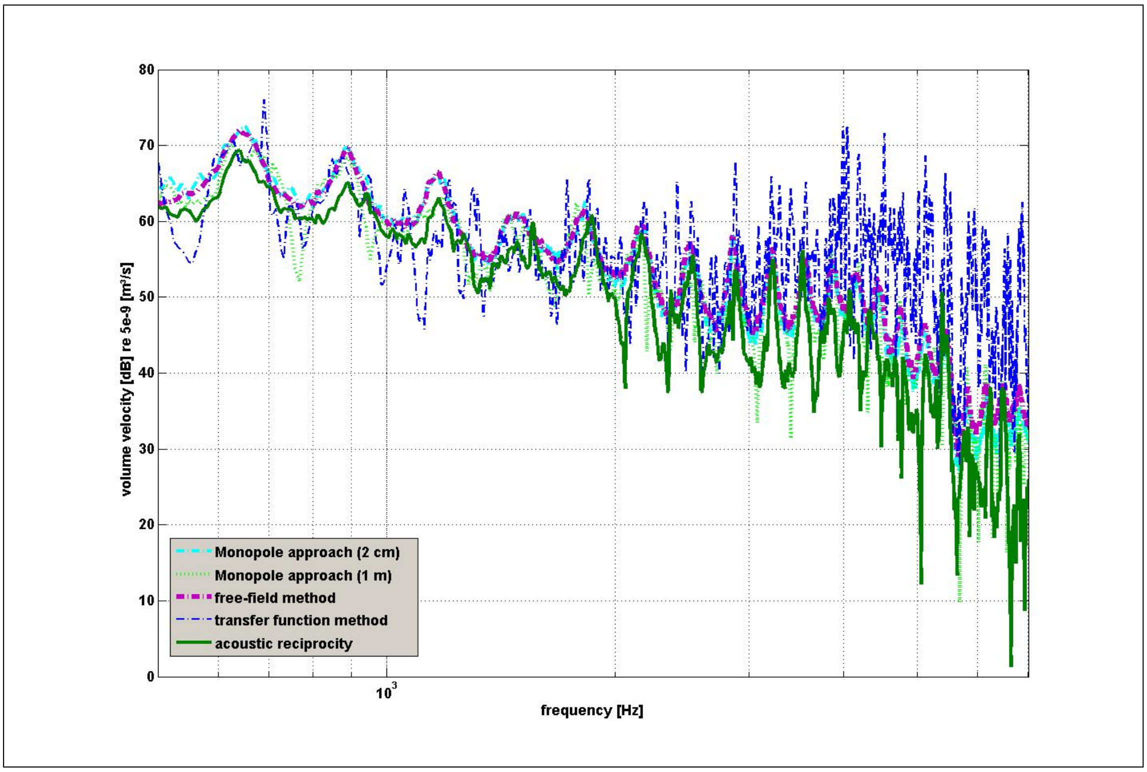

Figure 4-13: A comparison for the volume velocities of all investigated approaches

Regarding the curves in **Figure 4-13** it can be seen that the volume velocities all show the same behavior, with the existing resonances and antiresonances. All approaches lead to the same resonances that increase with frequency. That conclusion corresponds to an increasing modal density for larger frequencies. The level difference between the varying procedures is up to 10dB for lower frequencies. For larger frequencies the difference between the methods grow. Especially the approaches of acoustic reciprocity (cf. section 4.3.4) and the transfer function method (cf. section 4.3.3) differ dramatically in comparison to the values of the monopole approach (cf. section 4.3.1) and the free-field method (cf. section 4.3.2). The latter seems to be the most precise procedure, because of the 20 microphone positions. However, due to the large number of microphone positions, the free-field method is regarded as too time-consuming compared to the monopole approach. The benefit of the monopole approach is, for that reason, the time advantage and the functionality.

To conclude, the monopole approach provides the most practical results for determining the volume velocity of a constructed volume sound source with an omnidirectional directivity. Placing the microphone at a distance of 2cm from the tube's opening can be defined as sufficient. Hence the monopole approach is used later in this work for investigating the reciprocal transfer path analysis (cf. chapter 5 and chapter 6).

The subsequent section focus on the behavior of the volume velocity for another volume sound source.

4.4 The volume velocity of different volume sound sources (monopole approach)

To study the sensitivity and the related progression of the volume velocity for the monopole approach, a comparison of the results between two volume sound sources is performed in this part of the work. The volume velocity of the constructed volume sound source (cf. chapter 3) and the sound source of the LuK GmbH & co. KG (cf. appendix II) are investigated. It is noteworthy that the excitation signal was identical.

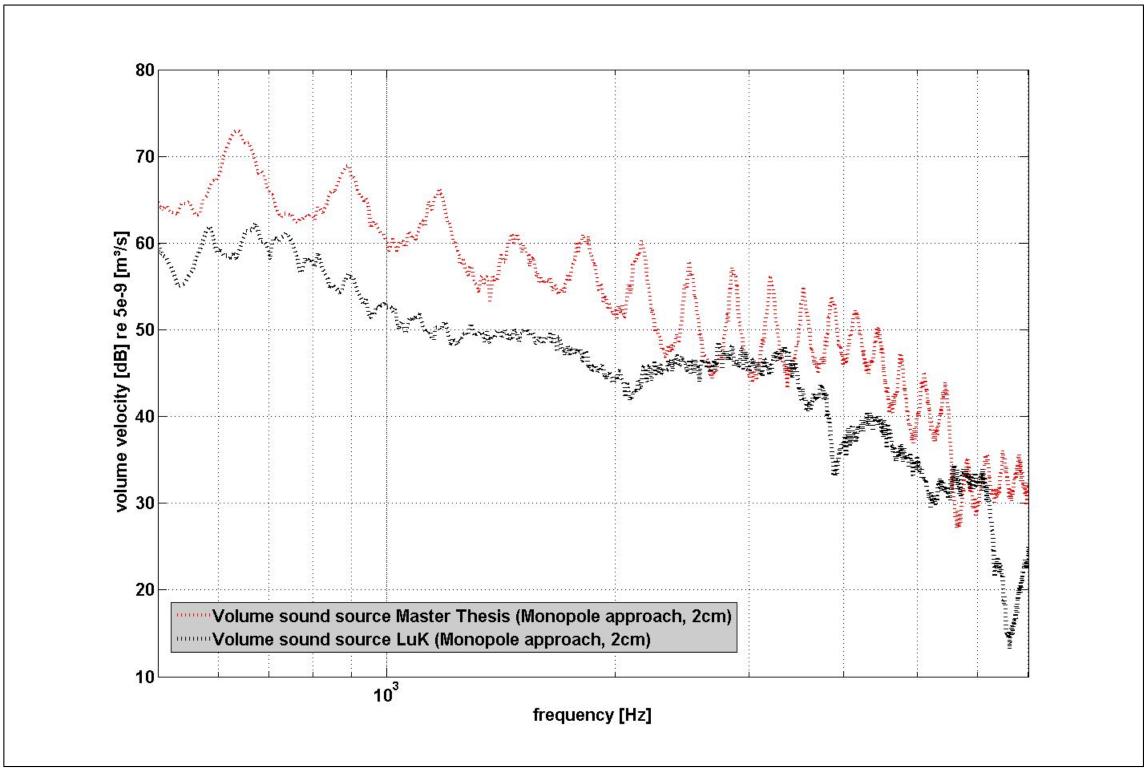

Figure 4-14: A comparison for the volume velocities of different volume sound sources (monopole approach)

The volume velocity for the sound source of the LuK GmbH & co. KG (cf. appendix II) shows a different behavior compared to the results of the constructed volume sound source (cf. chapter 3). Focusing on the loudspeaker of the LuK GmbH & Co. KG it turns out that the cut-off frequency appears at about 2kHz. This frequency is definitely smaller than the cut-off frequency of the pressure chamber loudspeaker (cf. appendix II) of this thesis (5,5kHz). However the sound source of the LuK GmbH & co. KG shows the same behavior at lower frequencies, as the constructed point source (cf. chapter 3), with the existing resonances and antiresonances. The levels of the monopole sound source (cf. chapter 3) are constantly larger for the considered frequency range, although the excitation signal was identical. The lower limitation of the volume sound source of the LuK GmbH & Co. KG can be detected at about 550Hz.

Since the sound source of LuK GmbH & Co. KG (cf. appendix II) has another inner diameter and a larger impedance tube than the constructed volume sound source (cf. chapter 3), the conclusions above can be confirmed.

The results of this section show that the designed and constructed volume sound source in chapter 3 can be regarded as more useful for the application of the reciprocal transfer path analysis than the sound source of the LuK GmbH & Co. KG (cf. appendix II).

Under normal circumstance the reciprocal transfer path analysis is applied for a vehicle. The subsequent section therefore studies the monopole approach and the corresponding volume velocity in more detail, focusing on the dependency of the location.

4.5 The volume velocity in dependency of the location (monopole approach)

In this part of the work the behavior of the volume velocity in dependency of the location is studied. Thus the sensitivity of the monopole approach and the respective surrounding area is examined. As one can see below in an exemplary illustration, the chosen testing environment represents on the one hand the interior and on the other hand the motor compartment of a vehicle.

To achieve the volume velocity here again the monopole approach was used by placing the microphone at a distance of 2cm from the tube's opening. The excitation signal in both cases was identical.

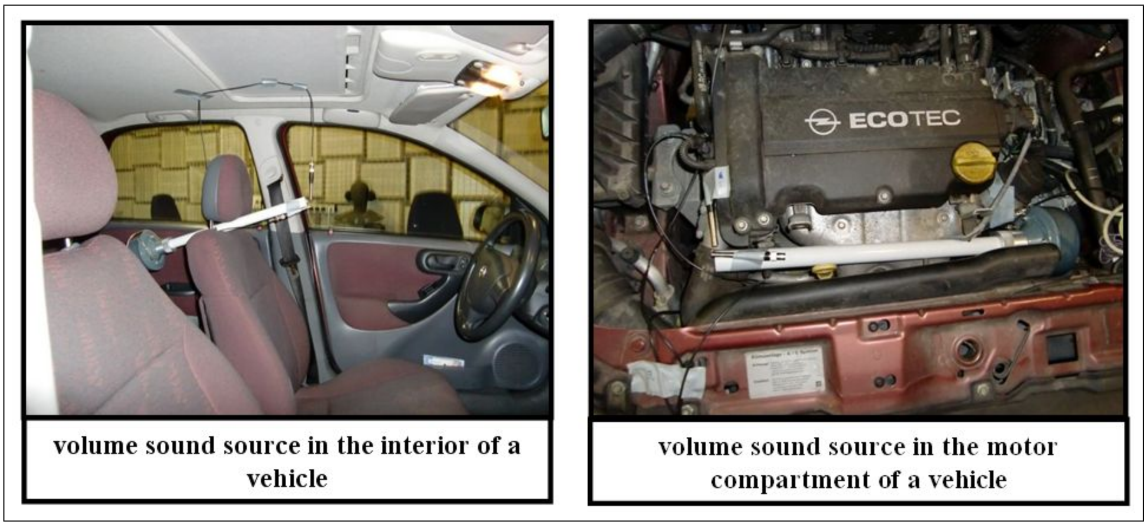

Figure 4-15: Placing the volume sound source in an interior and in an engine bay

The passenger compartment, with the cloth covers and sound-absorbing materials, presents an interesting ambiance with a "free-field" character. The engine bay on the other side is a strongly reflecting and small volume ("diffuse sound field" character). Hence two completely different locations are studied to focus on the behavior of the respective volume velocity of the monopole approach.

Note that the above described procedure was basically performed for two vehicles. The first vehicle is categorized in the small class, whereas the second vehicle is a large executive car.

- **vehicle.1 = underclass car**
- **vehicle.2 = large executive car**

The next figure depicts the corresponding results of the volume velocities for the respective vehicles and environments.

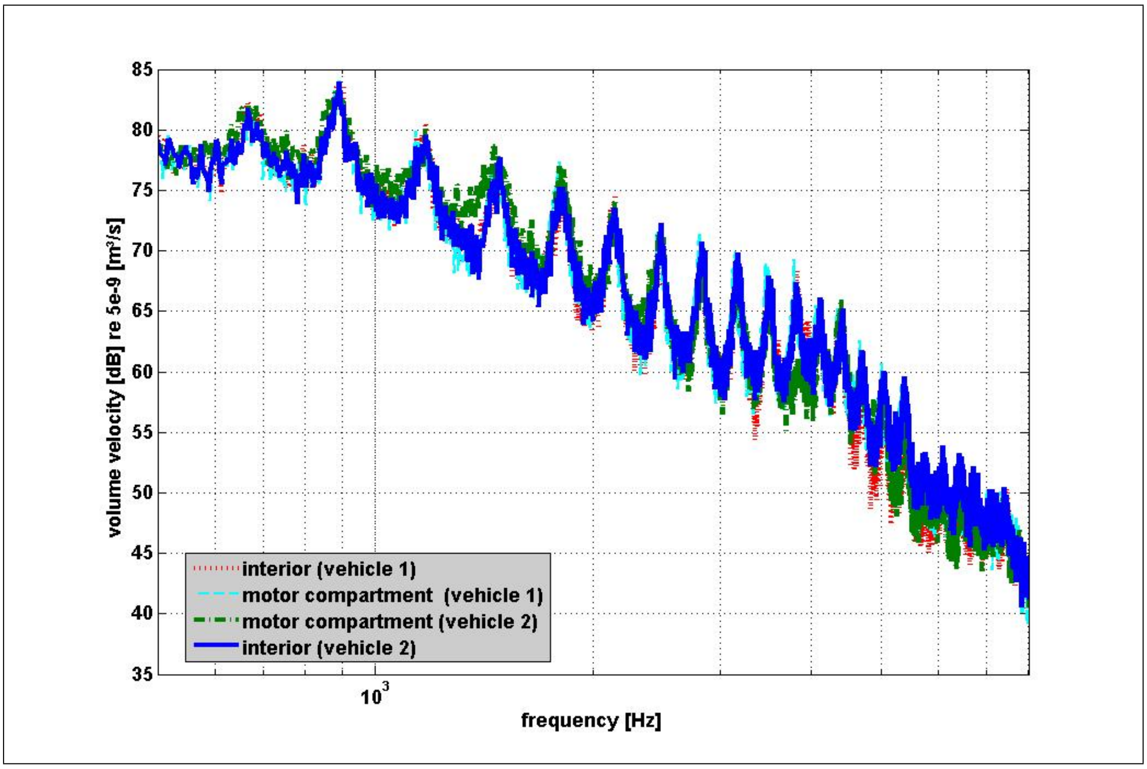

Figure 4-16: The volume velocities for different locations and vehicles (monopole approach)

The curves of the volume velocities in **Figure 4-16** show that the monopole approach can be classified as a robust investigation with a low sensitivity. The results of the motor compartment for the second vehicle deliver varying results that differ slightly in comparison to the residual measurement results.

Note that the above presented performance generally enables the possibility to perform the principle of acoustic reciprocity (cf. section 2.2) and to check the linearity of the regarded vehicle.

One can conclude that the monopole approach is not influenced by the surrounding environment and can be used for the reciprocal transfer function and the related determination of the acoustic transfer functions.

The volume velocity in connection to the monopole approach is a parameter that is completely independent of the environmental area and the type of vehicle. Thus the application of the monopole approach for a loudspeaker with an omnidirectional directivity represents a very robust investigation.

In the subsequent chapter the procedure of the reciprocal transfer path analysis is studied, with investigations in a laboratory test. Thus the principle of this method should become more understandable for the reader and at the same time it is possible to study the general behavior of this technique.

5 Laboratory test of the reciprocal transfer path analysis

In this chapter, the procedure and the results of a laboratory test conducted by the author for the reciprocal transfer path analysis are presented. The theory of chapter 2 is here applied in practice. A customized procedure gives the possibility to reduce the number of unknown parameters and to therefore test the theory under simplified boundary conditions.

The goals of the investigations in this chapter are, on the one hand, to verify the theory of the second chapter and, on the other hand, to prove the functionality as well as the sensitivity of the reciprocal transfer path analysis on the basis of a laboratory test.

The following section gives therefore an overview of the course of action.

5.1 Preparation

To perform the reciprocal transfer path analysis and the related prediction of the sound pressure for a sound source (e.g. component part), a source-path-receiver model is required (cf. section 2.1). The following figure therefore pictures the defined source-path-receiver model for the concept test of this section.

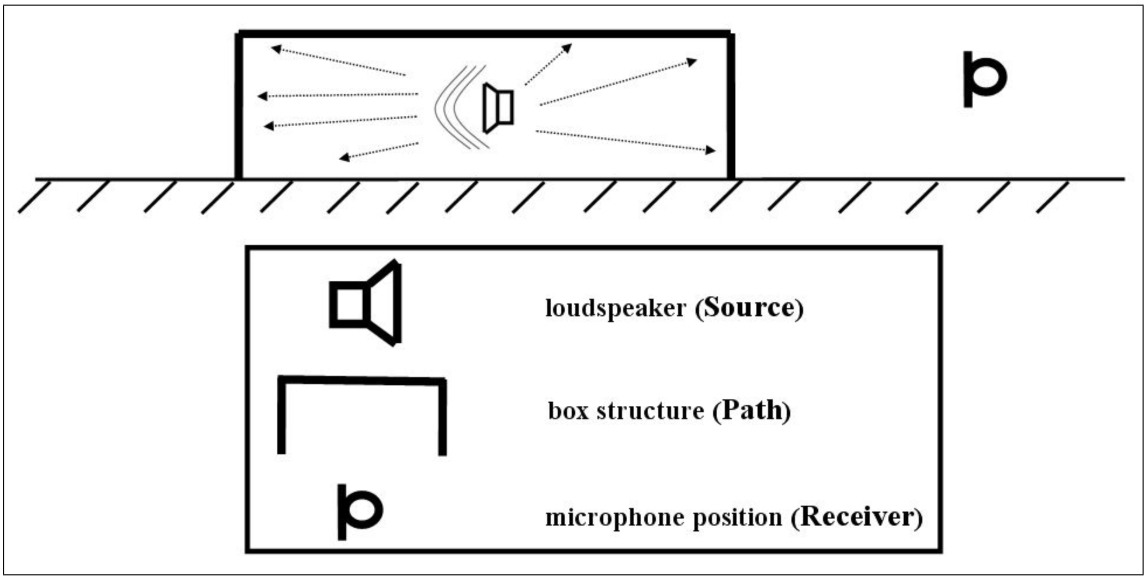

Figure 5-1: The conducted source-path-receiver model for the concept test

The plotted source-path-receiver model in **Figure 5-1** shows that

- the source element stands for the self-made loudspeaker (cf. chapter 3)
- the path element is represented by the structure of a box
- a microphone is located at a defined receiver position (receiver element)

The above pictured source-path-receiver model represents a highly simplified situation for the application of the reciprocal transfer path analysis. The volume velocity of the studied sound source is known by introducing the self-made volume sound source (cf. chapter 3), instead of focusing on a complex component part of the drive train of a vehicle. Using a box as path element is a lot simpler to understand than the complex architecture of a vehicle. The microphone at the appointed receiver position replaces the passengers in a car's interior.

The next sections focus on the determination of the respective elements of the defined source-path-receiver model in **Figure 5-1**. In a first step, the volume velocity of the source element is therefore examined.

5.2 Determination of the volume velocity (source)

The volume velocity of the defined source element for the source-path-receiver model in **Figure 5-1** is studied according to the definitions in section 5.1.
As mentioned above, the source element is represented by the self-made volume sound source from chapter 3 (cf. section 5.1). For the determination of the acting volume velocity the procedure given in section 2.3 is required. The application of the self-made loudspeaker from chapter 3, in relation to the monopole approach (cf. section 4.2.1), gives the possibility to verify the technique presented in section 2.3.

First it is necessary to subdivide the surfaces of the self-made volume sound source (cf. chapter 3) into a number of subareas (cf. **Figure 2-7** in section 2.3). In the next step it is assumed that each subarea corresponds to a monopole sound source [3,18,19]. Measuring the sound pressure of the respective subareas finally gives the volume velocity for each reference surface (cf. **Eq. 2-5 in section 2.3**).

The following figure plots the procedure described above schematically.

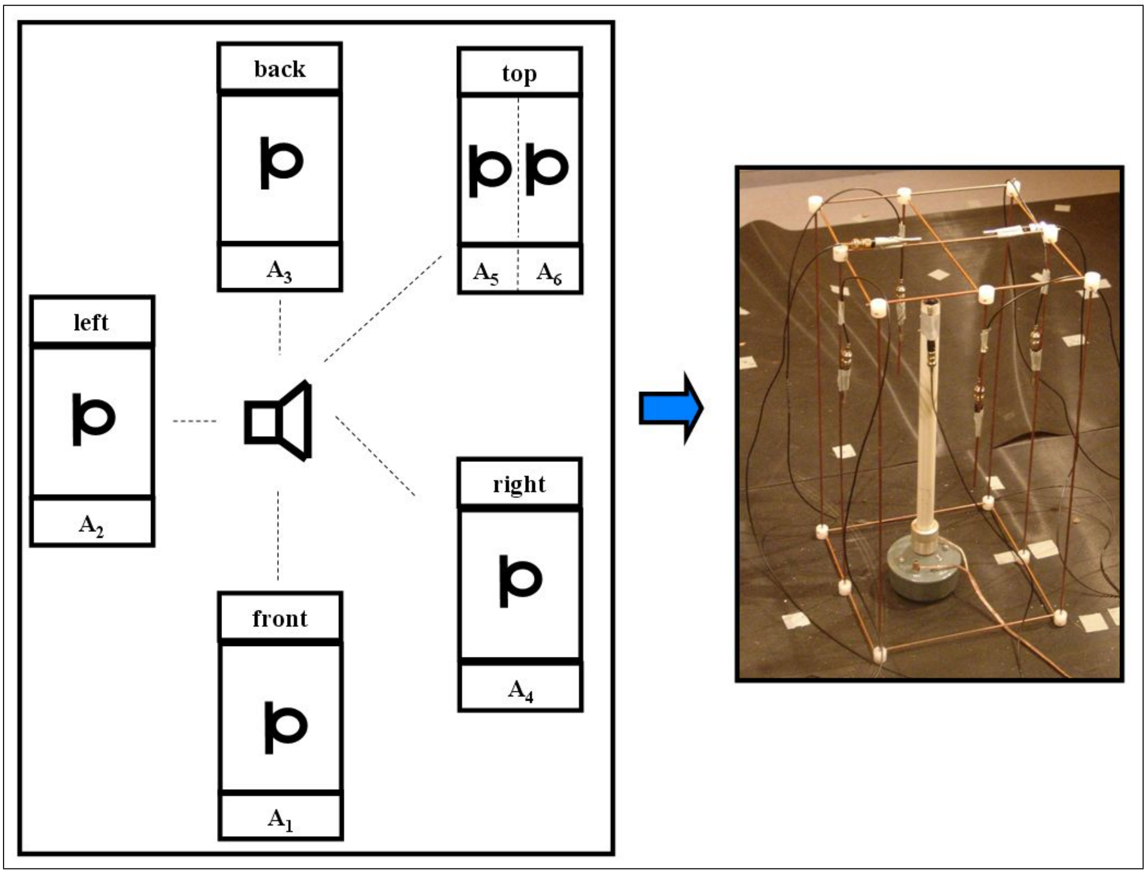

Figure 5-2: The microphone grid of the self-made volume sound source for the concept test

The schematic illustration in **Figure 5-2** shows that the surfaces (front, back, right, left and top) of the self-made volume sound source are subdivided into six subareas (A_1 up to A_6). The bottom surface of the volume sound source is neglected. The top surface is subdivided into two subareas, while the other surfaces represent one subarea each. A microphone is placed in the middle of each defined subarea to record the sound pressure. The arrangement of the microphones thereby corresponds to the particular subarea (cf. **Figure 5-3**). The distances between the tube's opening and the microphone positions on the surfaces (front, back, right and left) is identical (a=*20cm*), but differs from the microphones of the top side (a=*10cm*).

As can be seen in **Figure 5-2**, an additional microphone is placed at the tube's opening of the volume sound source. The previously calculated results can therefore be verified by additionally using the monopole approach (cf. section 4.2.1).

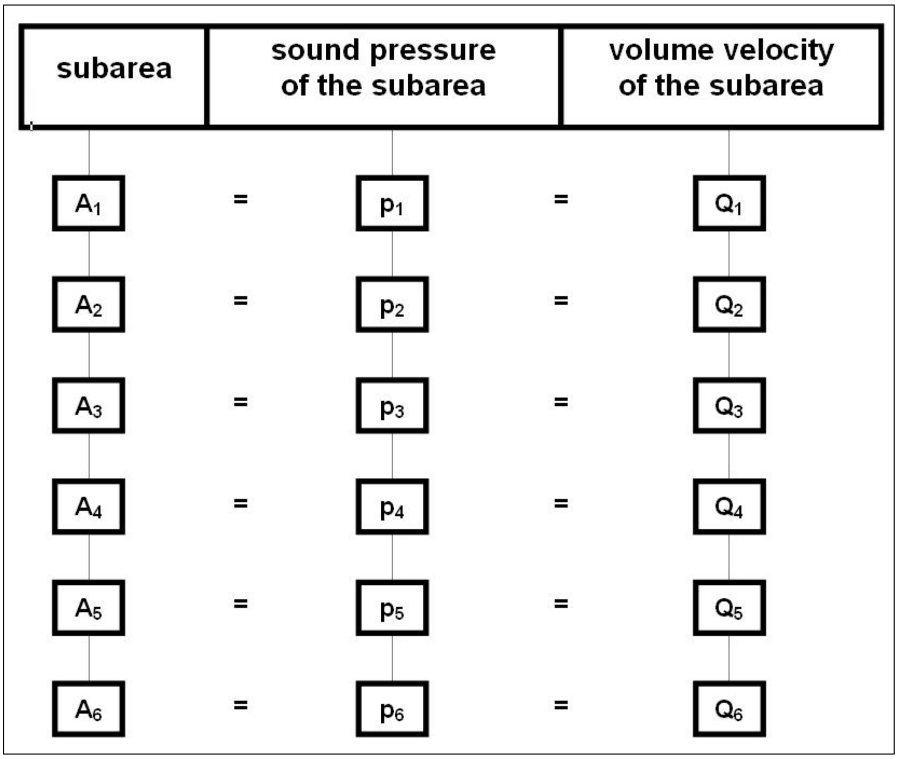

Figure 5-3: The existing relations for the procedure in this concept test (cf. **Figure 2-7** of section 2.3)

Assuming that each subarea relates to a monopole sound source allows reaching the volume velocities of the respective subareas (cf. **Eq. 2-5** in section 2.3).

From **Eq. 2-1** one can conclude that the subareas $A_1...A_6$ correspond to the respective sound pressures $p_1...p_6$ and additionally to the volume velocities $Q_1...Q_6$ plotted above.

Applying the above described procedure leads to an update of **Figure 5-1**, picturing the introduced microphone grid.

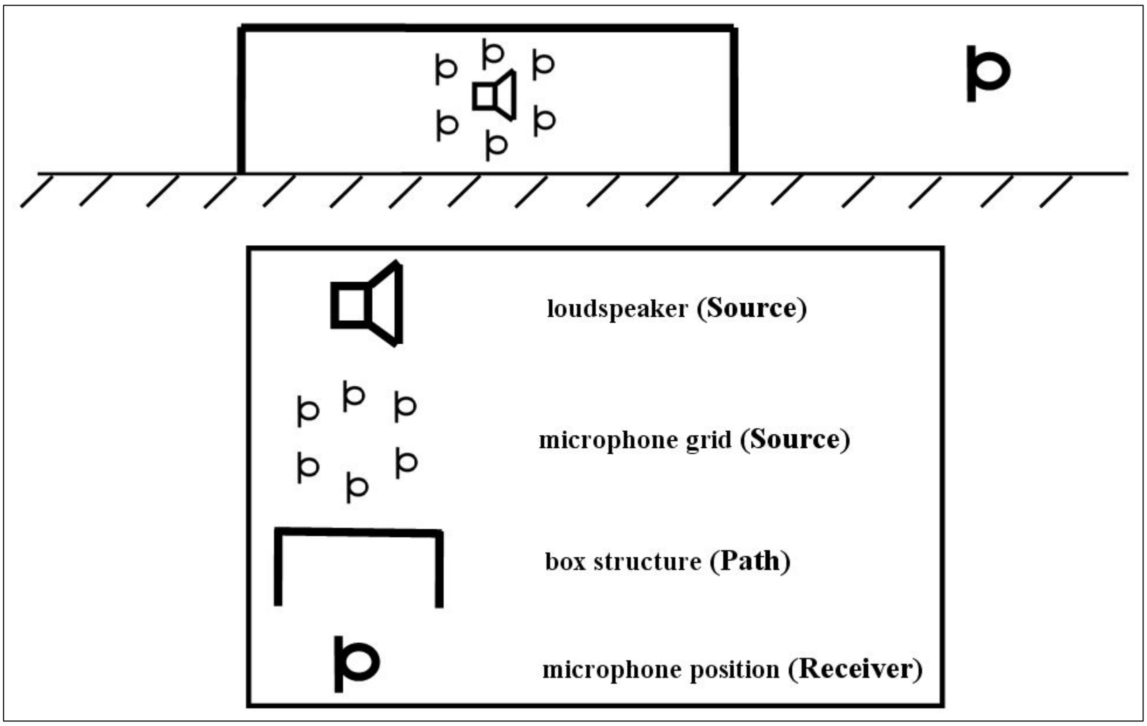

Figure 5-4: The updated source-path-receiver model for the laboratory test

Note that the positions of the microphones for the grid in **Figure 5-4** are drawn symbolically.

The reason for introducing a microphone grid for the performance of a reciprocal transfer path analysis was explained in section 2.3.

In the following parts of this section the corresponding results for the volume velocities of the respective subareas are plotted. Thus a comparison between the approach of **Eq. 2-5** (cf. section 2.3) and the monopole approach (cf. section 4.2.1) can be performed. Based on literature, it is here assumed that the radiation resistance factor R has a value of $R_{ij} = 2$ [3,18].

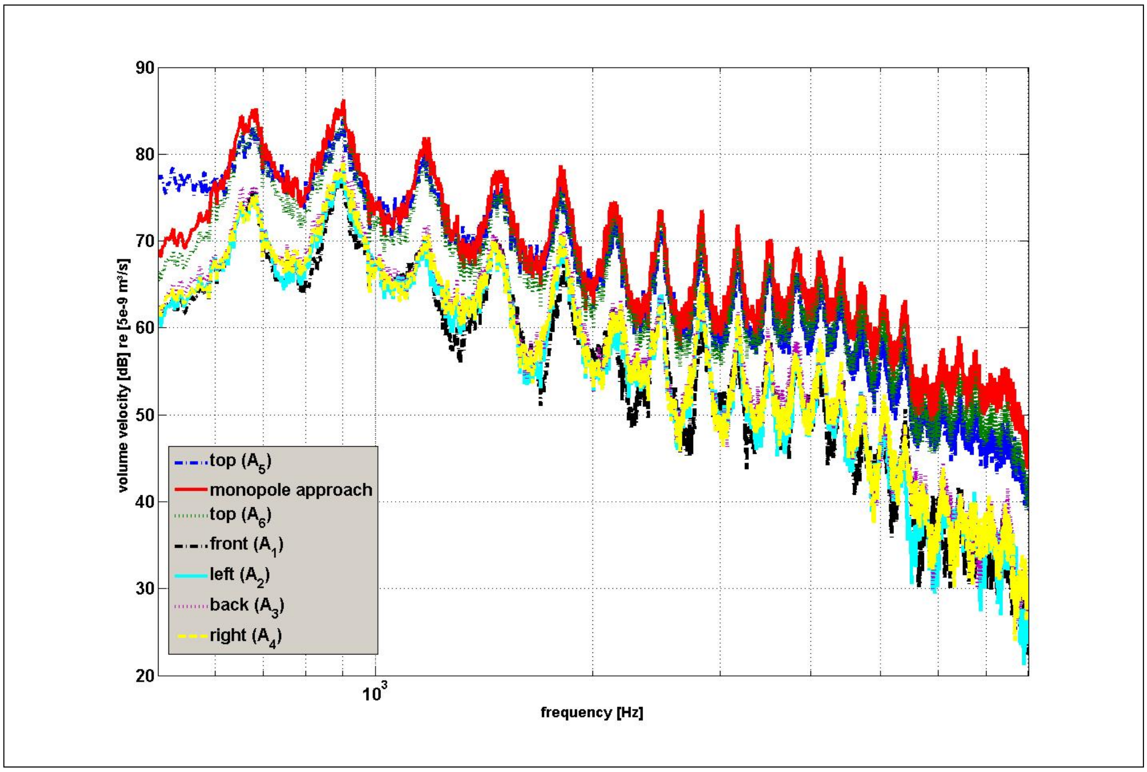

Figure 5-5: The results of the volume velocities for the concept test

Looking at the plotted volume velocities in **Figure 5-5**, one sees that the investigation of **Eq. 2-5** (cf. section 2.3) yields useful results. The characteristic of the graphs is comparable to the outcome of the monopole approach. The level difference of almost 10dB for the monopole approach and the top surfaces (p_5 and p_6) compared to the other surfaces (front, left, back and right) indicates that the distance to the tube's opening is too large for the latter investigations. Thus it can be concluded that the performance of **Eq. 2-5** (cf. section 2.3) better results the microphones are placed closer to the examined volume sound source. This assumption can be reached for an examined volume sound source with an omnidirectional directivity in the frequency range of interest.

It is noticeable that the excitation signal (random noise) of the volume sound source showed enough distance from the background noise level.

Measuring the acting sound pressure under operation at the defined receiver position represents another important parameter that has to be determined here. Therefore the target value for the predicted sound pressure in section 5.4 will be estimated. The following graph presents the receiving sound pressure.

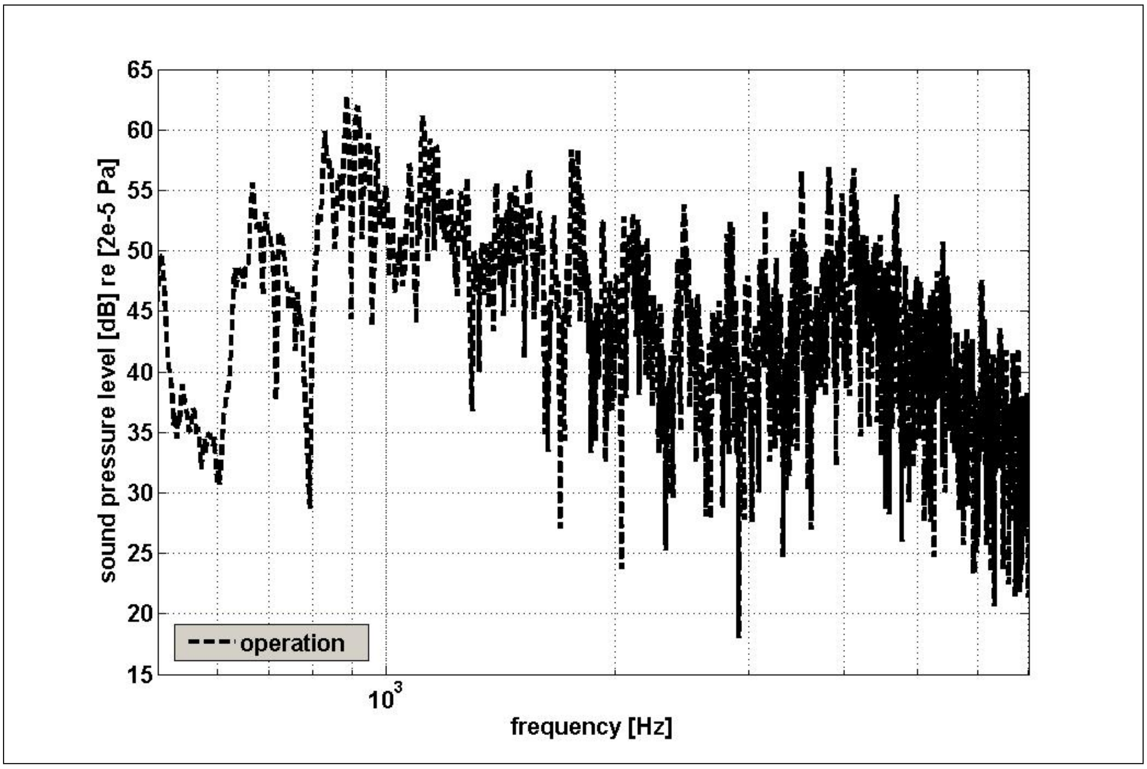

Figure 5-6: The sound pressure under operation for the laboratory test

Since in this concept test only the self-made monopole sound source (cf. chapter 3) is investigated, it is expected that the predicted sound pressure (cf. section 5.4) will give identical results to those plotted above.

In a next step, the acoustic transfer functions that represent the path element of the updated source-path-receiver model in **Figure 5-4** are determined.

5.3 Determination of the acoustic transfer functions (path)

This part of the work focuses on the measurement of the acoustic transfer functions for the presented laboratory test in **Figure 5-4**. As mentioned in section 2.2 and section 2.4, the acoustic transfer functions are required for the performance of the governing equation (cf. **Eq. 2-1** and cf. **Eq 2-7**).

To determine the acoustic transfer functions a volume sound source with an omnidirectional radiation characteristic is needed. As already presented in chapter 3 and chapter 4, the self-made loudspeaker has such a directivity in the frequency range from 0,5 to 5,5kHz.

Placing the monopole sound source at the originally defined receiver position (cf. **Figure 5-7**) gives the possibility to appoint the acoustic transfer functions to the respective microphones (subareas) of the specified measurement grid (cf. section 5.2). Hence the related path elements of the structure can be determined. The figure below presents this procedure graphically.

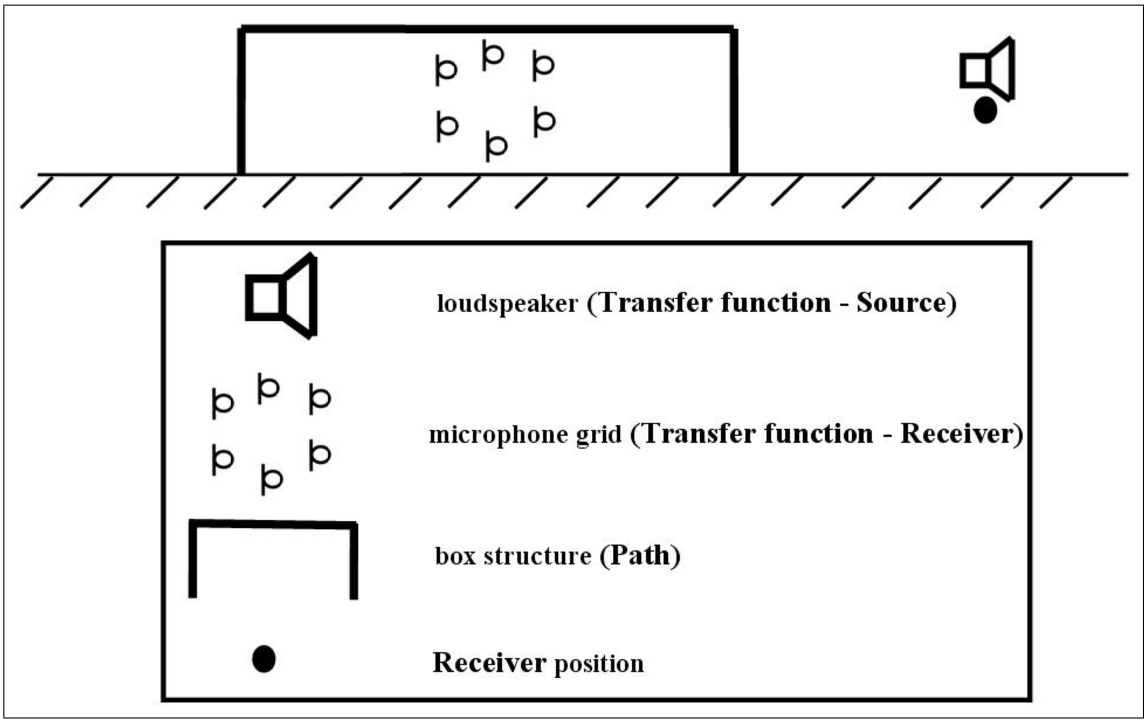

Figure 5-7: The procedure to measure the acoustic transfer functions for the concept test

Figure 5-7 represents a reciprocal measurement of the acoustic transfer functions. The advantage is that all frequency response functions can be determined simultaneously by one measurement.
The subsequent figure therefore plots the corresponding acoustic transfer functions of the concept test of this thesis. For the interpretation of the acoustic transfer functions is it helpful to understand that large values stand for a large transmission, whereas low levels mean an increasing sound insulation of the examined structure [1,2].

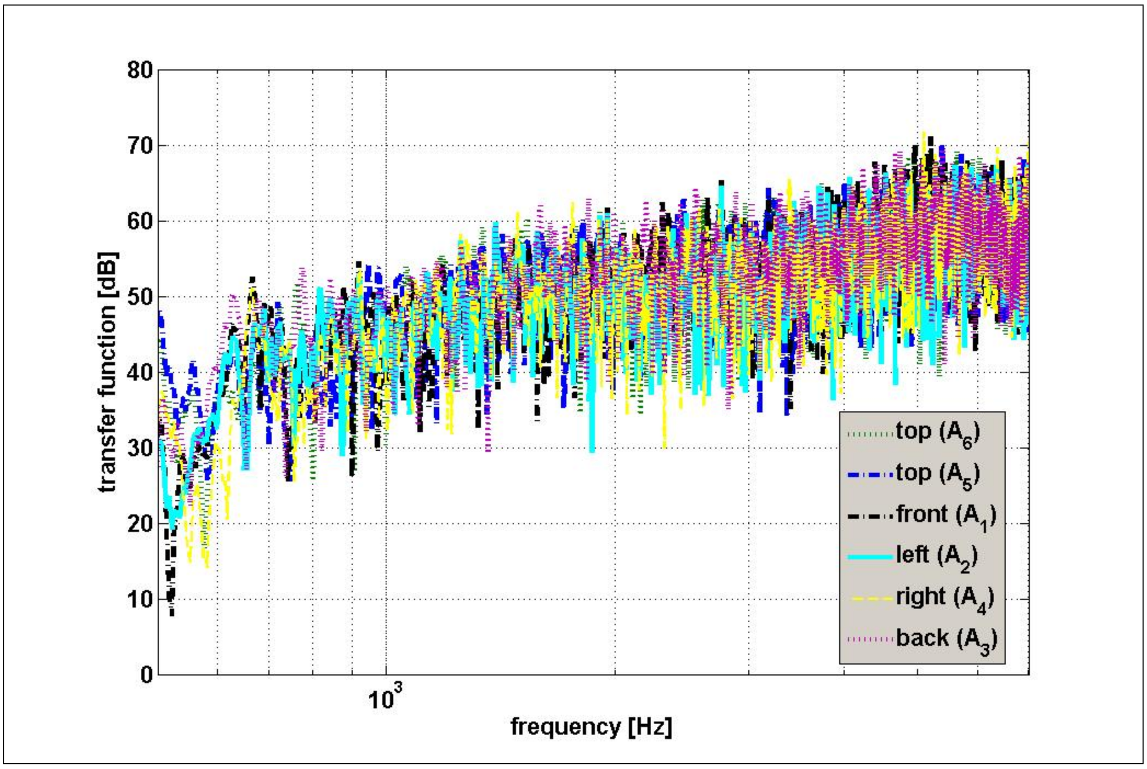

Figure 5-8: The acoustic transfer functions for the concept test

The acoustic transfer functions in **Figure 5-8** all show the same behavior. The smaller the frequency, the better the sound insulation of the path element (box). One sees that with an increasing frequency more sound is transmitted between the source (volume sound source) and the receiver (microphones of the grid).

Reversing the source and receiver positions would deliver the result of the linearity for the examined system that is neglected at this point of the work.

The resulting acoustic transfer functions of this part of the work and the determined volume velocities of section 5.2 finally offer the possibility to calculate the predicted sound pressure level for this concept test (cf. **Eq. 2.8** in section 2.5).

5.4 Predicting the sound pressure (Receiver)

As already mentioned above in this section the sound pressure level for the self conducted laboratory test is predicted. The principle of acoustic reciprocity (cf. **Eq. 2-1** in section 2.2) therefore is needed.

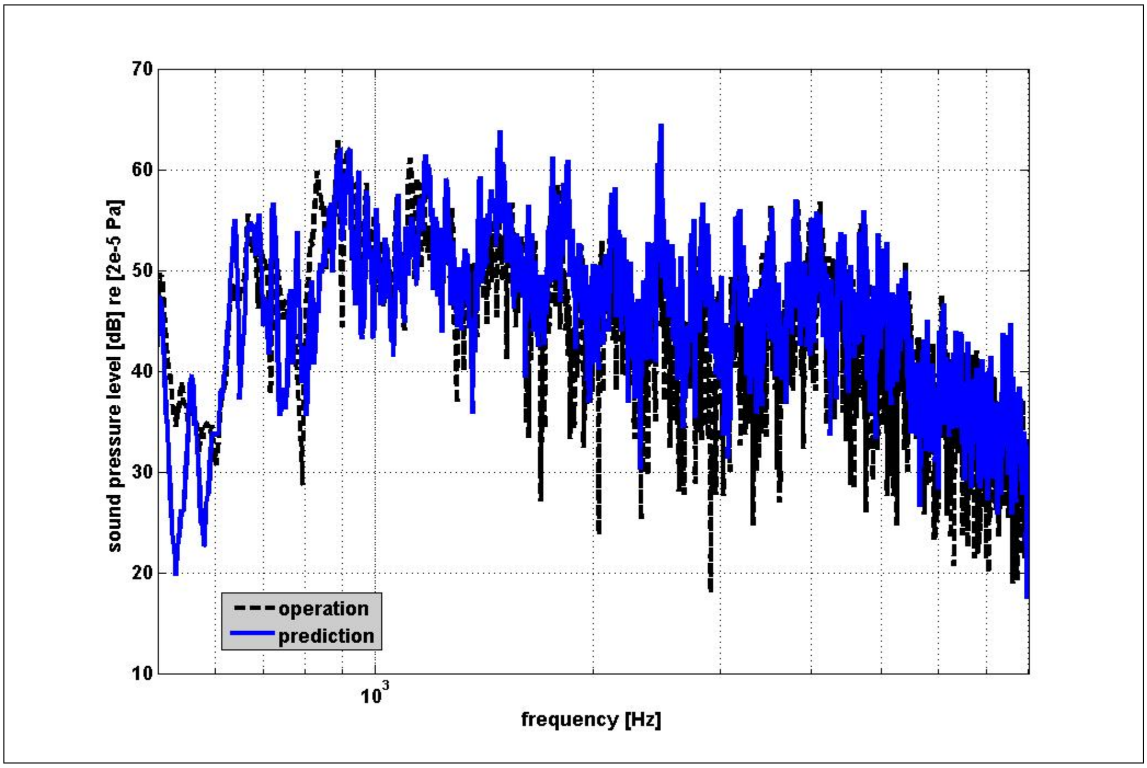

Figure 5-9: A comparison for the predicted and measured sound pressure at the defined receiver position for the laboratory test

Comparing both curves in **Figure 5-9** one can see that the results match for the complete frequency range. Note that this result was expected. Focusing more intensively on the existing drops in **Figure 5-9**, it turns out, that the operational data is more distinctive. The cut off frequency of the loudspeaker (5,5 kHz) is clearly detectable in the graphs. The differences between the calculated and the measured data are more visible for the antiresonances and at a maximum of 20dB. At existing resonance peaks both procedures show identical values.

One can conclude that the presented theory of performing the reciprocal transfer path analysis (cf. chapter 2) is verified by the results of the test in this chapter. At the same time, the applicability of the self-made volume sound source is verified. Hence the motivation arises to investigate this procedure for a real component from the LuK GmbH & Co. KG that is normally installed in a vehicle.

6 Application of the reciprocal transfer path analysis

The sixth chapter of this thesis presents to the reader the application of the reciprocal transfer path analysis for a component from the LuK GmbH & Co. KG. In the next sections the conducted procedure and the corresponding measurement results are therefore discussed. The objective of this chapter is to prove the functionality of the theory presented in chapter 2 for a component part of a vehicle.

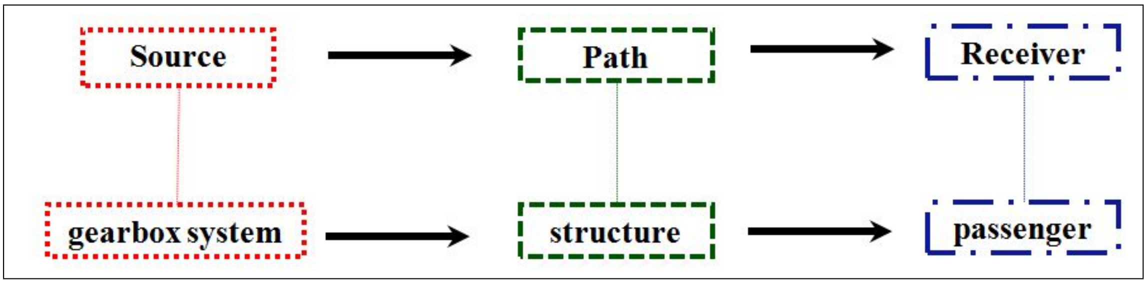

Figure 6-1: The source-path-receiver model for the application of the reciprocal transfer path analysis

As one can see in **Figure 6-1** in this chapter a gearbox system is examined as sound source. The path element is represented by the structure of the investigated vehicle, whereas the receiver is pictured by the passenger and therefore replaces the received sound pressure level in the interior.

The following sections focus on the procedure for studying this problem.

6.1 Preparation

As already shown in the previous sections, the procedure of the reciprocal transfer path analysis requires that the regarded sound source be subdivided into a number of subareas and that microphones be placed on the respective subareas of the examined sound source (cf. chapter 2 and chapter 5). Using the measured sound pressure of each microphone allows us to determine the wanted volume velocity of the particular subarea (cf. **Eq. 2-5**). First, the arrangement of the microphones for the investigated gearbox system is therefore schematically illustrated.

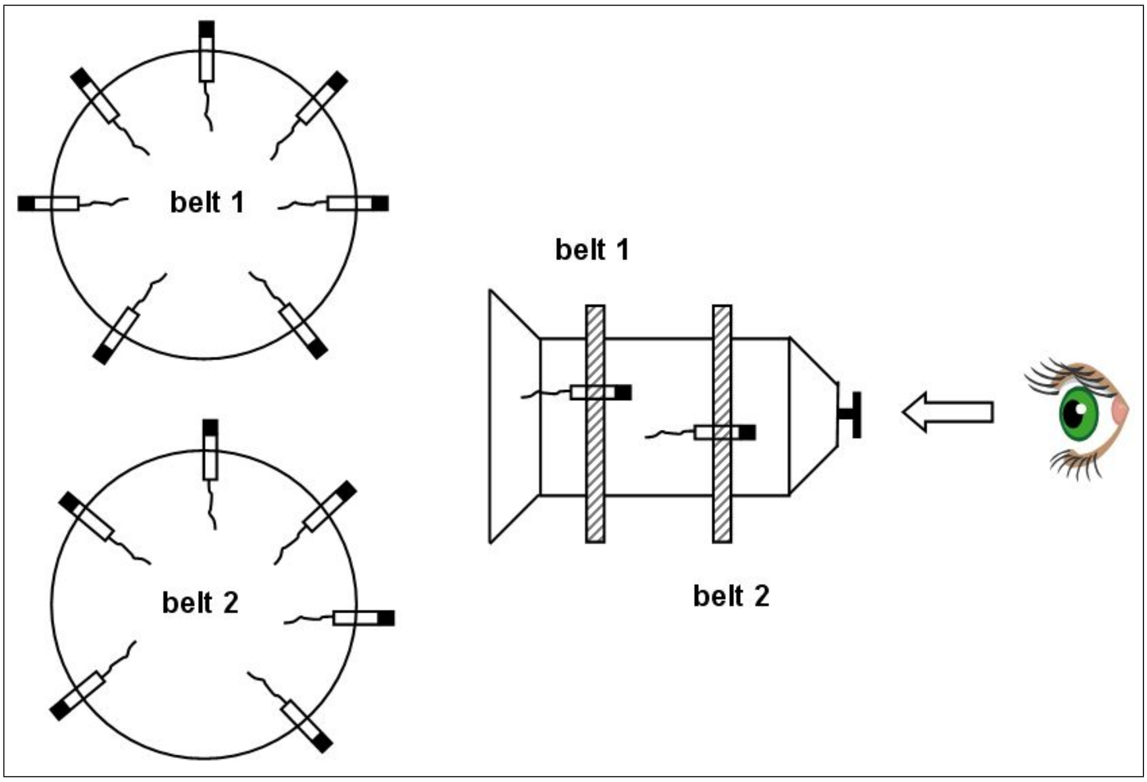

Figure 6-2: An illustration of the two microphone belts for the gearbox system

Basically, two microphone belts are applied to achieve the acting volume velocity of the examined gearbox system (cf. **Figure 6-2**). The first belt has six microphones, whereas the second belt contains five microphones. The first microphone belt is closer to the engine of the vehicle than the second microphone belt. The arrangement of the microphone belts was chosen according to the varying diameter of the gearbox system. Thus each microphone covers the same surface area A_{i_j} of the gearbox system. It is clear that the acoustic transfer functions have to contain eleven transfer paths from the gearbox system to the interior. appendix VI pictures the installed microphone belts for the gearbox system (cf. **Figure VI-1**).

The next section covers the determination of the acting volume velocity of the examined gearbox system.

6.2 Determination of the volume velocity (gearbox system)

In this part of the work the volume velocity for the investigated gearbox system is determined. The theory for achieving the volume velocity of a sound source was described in section 2.3 and discussed more in depth in section 5.2. The volume velocity of the examined gearbox system was studied both in test bench measurements and in the vehicle. The gearbox system was driven with an identical excitation signal for both cases. Hence a comparison of the respective volume velocity for both cases can be performed, with a focus on the sensitivity of the presented theory. According to the literature, the application of the reciprocal transfer path analysis delivers the time advantage that installing the regarded component part into the vehicle is not necessary for detecting the received sound pressure level in the interior. The procedure in this section therefore focus mainly on the validity of this statement.
The measurements on the test bench were performed under almost semi-free field conditions, whereas the measurements in the vehicle included the absorbing and reflecting materials of the engine compartment, as well as the residual sound sources. In appendix VI both cases are plotted in more detail with the corresponding photos (cf. **Figure VI-2** and **Figure VI-3**).

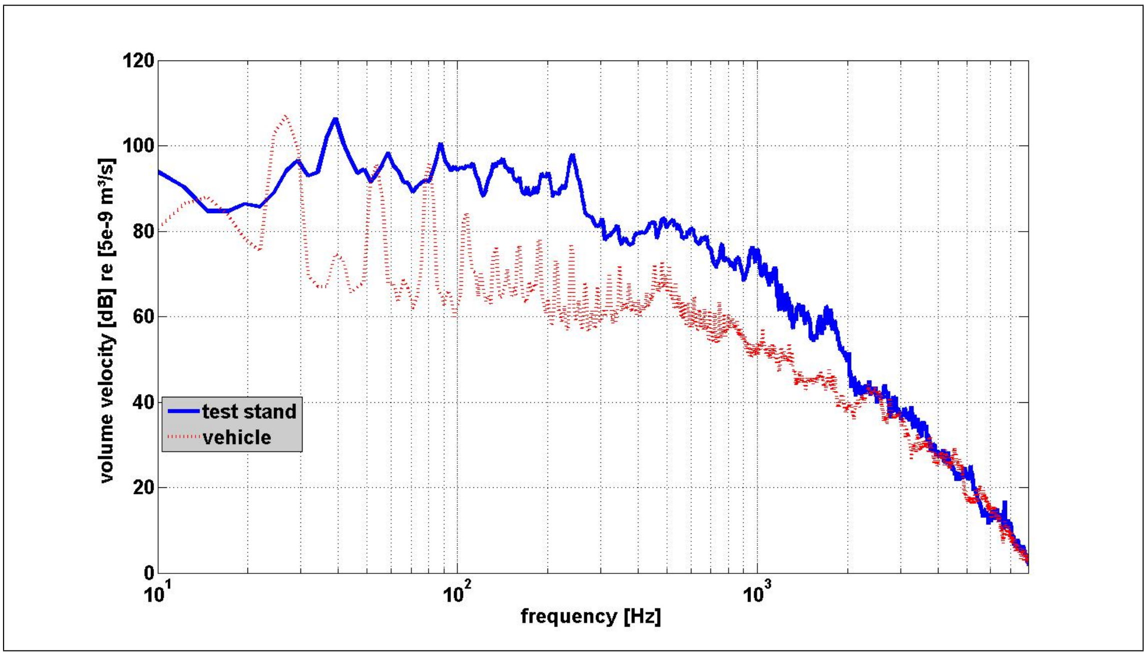

Figure 6-3: The volume velocities for the test stand and vehicle

The graphs in **Figure 6-3** imply that the vehicle case possesses significantly different characteristics from the test stand case below a frequency of 2,5kHz. For frequencies below 2,5kHz the test stand case gives larger values. This could indicate that the excitation of the gearbox system is not identical for both cases. The vehicle case shows significantly the engine orders at smaller frequencies.

The next section verifies the linearity of the vehicle that was used in this part of the work.

6.3 The linearity of the vehicle

To prove the linearity of the vehicle, the volume sound source is placed in the engine bay and driven by a random noise, while the sound pressure is recorded in the interior. Thus the transfer function for the acting volume velocity of the monopole sound source and the recorded sound pressure at the receiving microphone can be built. Reversing the positions of excitation (source) and system response (receiver) gives the possibility to examine the linearity of the regarded vehicle. This procedure simultaneously serves as a verification of the principle of acoustic reciprocity (cf. chapter 2). Fluctuations in the linearity represent an important indicator for the validity of the performance of the reciprocal transfer path analysis and the predicted sound pressure level that will be carried out later, in section 6.5.

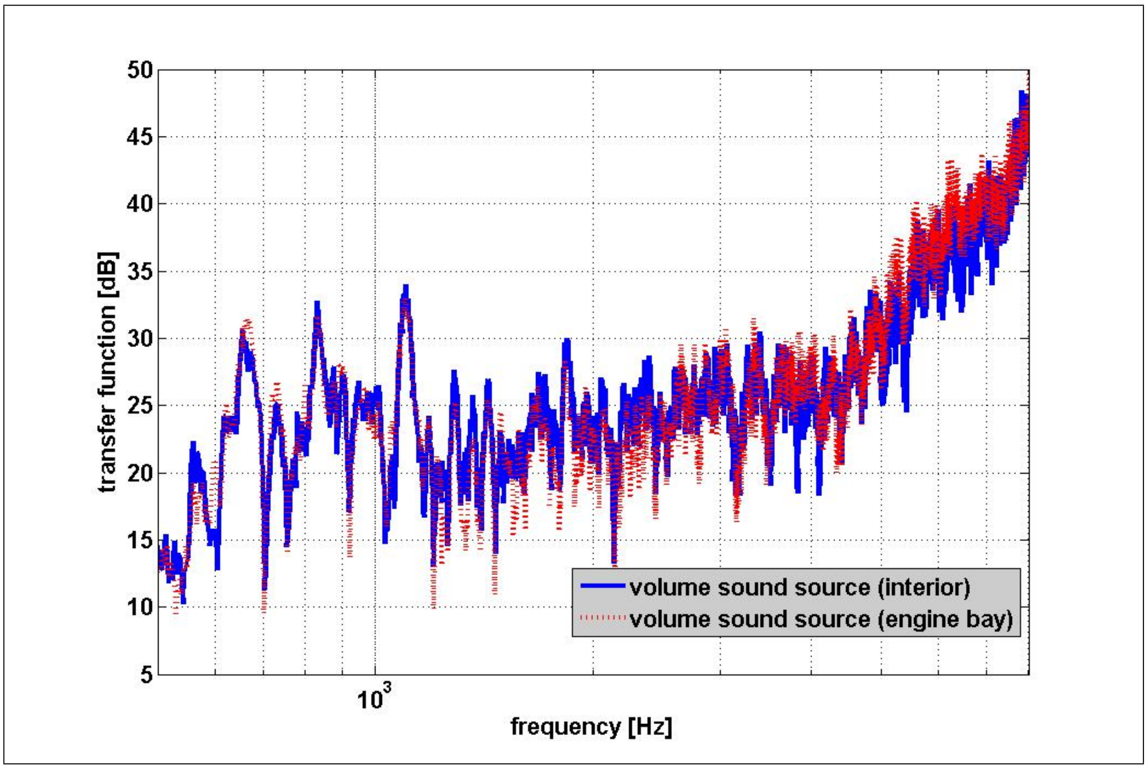

Figure 6-4: The linearity for the regarded vehicle

The differences between the graphs in **Figure 6-4** are at most 5dB. The larger the frequency, the stronger the fluctuations in the above plotted transfer functions. This is because of the shorter wavelengths for higher frequencies. The smaller the wavelengths are, the stronger is the influence of the surrounding environment, including reflecting and absorbing materials in the engine bay and interior. Generally both transfer functions in **Figure 6-4** show that the vehicle structure transmits the sound better for frequencies below 1,25kHz, looking on the working frequency range of the applied volume sound source. This means that the plotted results in **Figure 6-4** above 5,5kHz can be neglected. In the frequency range of interest the vehicle behaves as a linear system and the application of the reciprocal transfer path analysis can be regarded as useful.

The next section discusses the determination of the path element (acoustic transfer functions) for the examined vehicle.

6.4 Determination of the acoustic transfer functions (path)

To perform the reciprocal transfer path analysis according to the principle presented in chapter 2, it is necessary to determine the acoustic transfer functions of the vehicle. Note that these acoustic transfer functions stand for the path element, whereas the results in section 6.2 replace the source element of the defined source-path-receiver model.
To measure the acoustic transfer functions, the self-made volume sound source was driven in the interior of the vehicle (at the defined receiver position), while the acting sound pressure was measured at the microphones of the respective microphone belt (cf. section 6.1). Thus a transfer function between the volume velocity of the volume sound source and the sound pressure of the particular microphone can be built. Generally this procedure offers the possibility to focus on the residual acoustic transfer functions between the source and the receiver position.

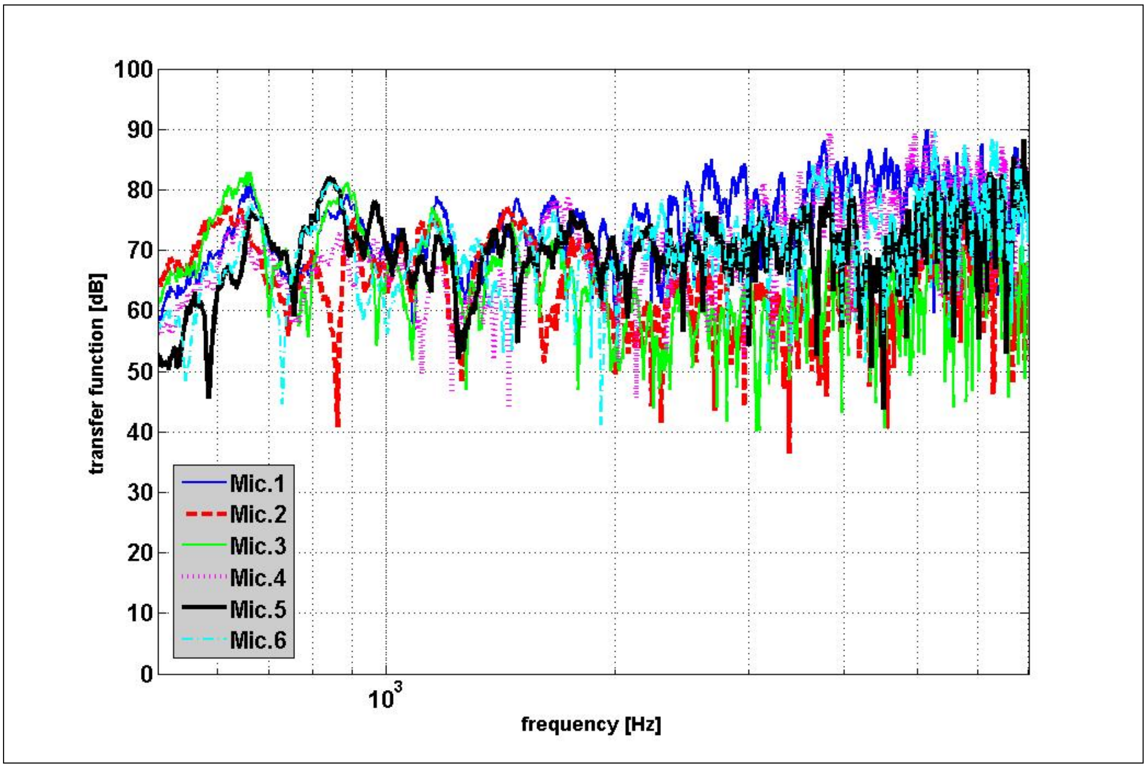

Figure 6-5: The acoustic transfer functions for the first belt

Regarding the results of the acoustic transfer functions of the first belt, it can be seen that the fluctuations between the graphs increases for larger frequencies. Below a frequency of 2kHz the values differ only slightly. The most distinctive drops are visible for the acoustic transfer functions of the second and the third microphones. The acoustic transfer functions for the first, fourth and sixth microphones have the largest magnitude.

Following the procedure depicted in section 6.1, the acoustic transfer functions for the second microphone belt are presented and discussed below.

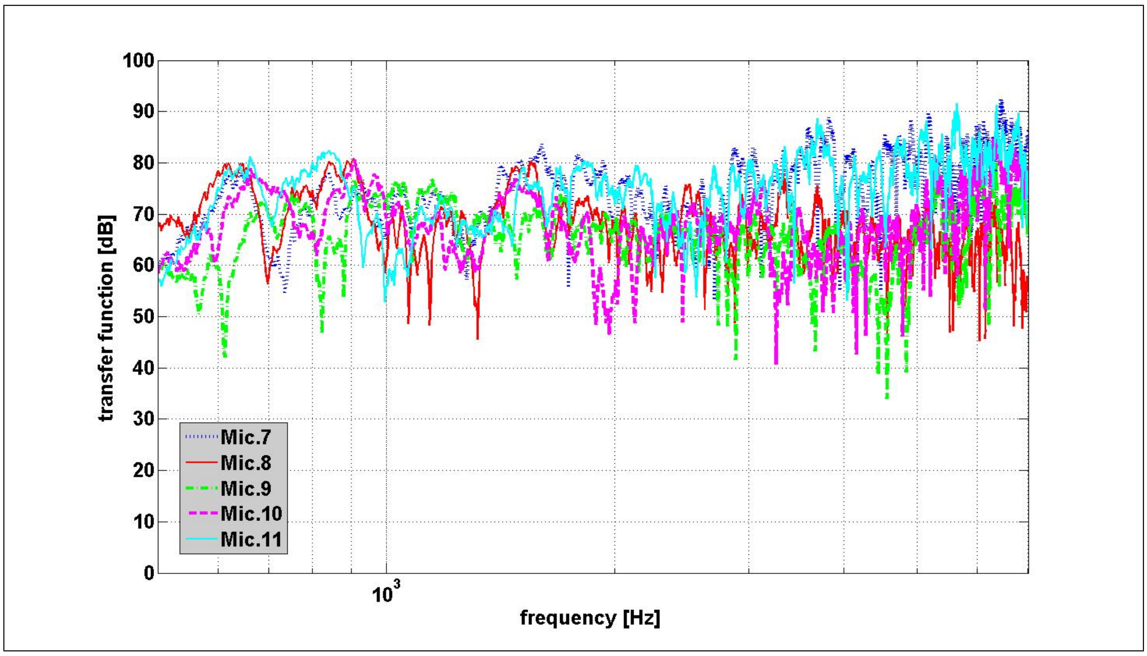

Figure 6-6: The acoustic transfer functions for the second belt

The graphs of the second microphone belt show a slight deviation at small frequencies. The acoustic transfer functions of the seventh and the eleventh microphones give distinctly larger values than the rest of the curves for an increasing frequency. The dominant drops of the acoustic transfer functions of microphones eight and nine are clearly visible in **Figure 6-6**.

Using the determined acoustic transfer functions of this part of the work and the volume velocities of section 6.2 allows us to achieve the predicted sound pressure level in the interior (cf. section 2.5).

6.5 Predicting the sound pressure level (receiver)

The calculation of the predicted sound pressure level of the examined gearbox system represents the finalization of the defined source-path-receiver model in this section. The figure below therefore plots the predicted sound pressure levels for the test stand and the vehicle. To prove the presented theory (cf. chapter 2) is it therefore essential to compare the predicted sound pressure level with the measured sound pressure level in the interior during operation.

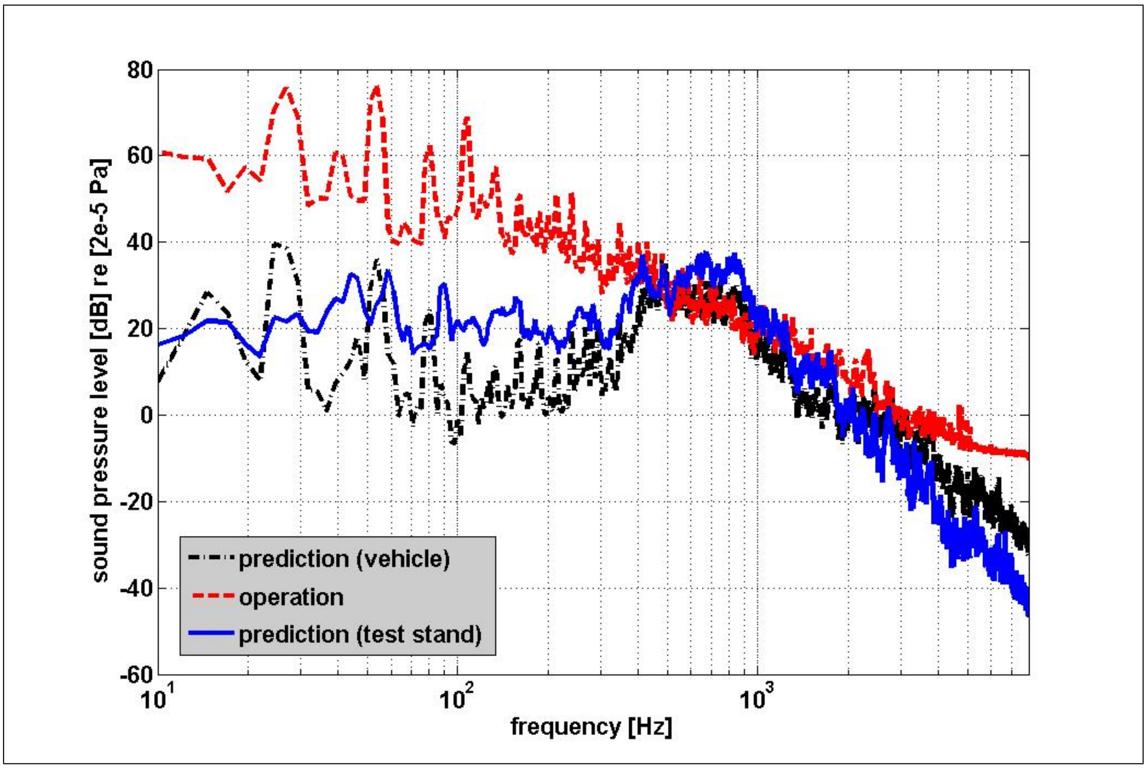

Figure 6-7: A comparison for the predicted sound pressure level

The results in **Figure 6-7** indicate that the predicted sound pressure level of both cases (test stand and vehicle) gives, in comparison to the operational data, insufficient values below 400Hz and above 1,25kHz. Between 0,4 and 1,25kHz the predicted sound pressure level for the vehicle case shows a good agreement with the sound pressure level under operation. The predicted sound pressure level of the test stand measurements differs slightly by a maximum of 5dB. The operational data (vehicle case) and the predicted sound pressure level of the vehicle significantly show the engine orders at lower frequencies. Since the volume sound source is applicable in a frequency range from 0,5 up to 5,5kHz (cf. chapter 3), the deviations for lower frequencies should be neglected. The slight differences between the excitation of the gearbox system for the test stand and vehicle measurements do not have an enormous influence on the predicted sound pressure level in the interior of the regarded vehicle.

Looking at the curves in **Figure 6-7**, one could argue that the examined gearbox system is in the frequency range from 0,4 to 1,25kHz, the sound source of interest. This means that the gearbox system is primarily responsible for the rising sound pressure level in the interior at these frequencies. For smaller and larger frequencies are other sound sources in the engine bay are dominant (e.g. the engine at small frequencies). Additionally scanning the engine with a microphone grid could be a helpful measure for improving the quality of the results for the predicted sound pressure level of the vehicle case.

It can be concluded that the application of the reciprocal transfer path analysis and the relating theory (cf. chapter 2) represents a useful tool for predicting the sound pressure level in the interior that results from the operation of a gearbox system. Using another volume sound source that covers a broader frequency range could be a measure for improving this approach. Determining the volume velocity of an examined component on a suitable test stand and combining this data with measured transfer functions of any vehicle gives the possibility to predict the sound pressure level in any interior, without the need to install it on the respective vehicle. It is important here that the excitation on the test stand be comparable to the real excitation of the installed case in the vehicle.

7 Conclusion and outlook

At the end of this thesis, the most important results and facts are summarized. After this review, issues to consider in the future are discussed.

Note that the conclusions of the individual sections are considered separately from each other.

Chapter 2

The theory presented in this part, with the introduction of a source-path-receiver model and the corresponding governing equation, is the foundation of this thesis. It was shown that the principle of acoustic reciprocity offers the possibility to solve the problem of the reciprocal transfer path analysis for the airborne sound of a component part. Therefore it is essential to subdivide the areas of the regarded sound source into subareas and to allocate each subarea to a microphone. The number of applied microphones concurrently implies the number of acoustic transfer functions that have to be measured reciprocally. The combination of the volume velocity for the examined sound source (component part) and the respective acoustic transfer functions offers the possibility to calculate the predicted sound pressure level in the interior.

Chapter 3

In this chapter, the construction of a self-made volume sound source was described. The reason is that the application of the reciprocal transfer path analysis requires, in principle, a monopole sound source. The self-made sound source has an existing omnidirectional directivity, in a frequency range from 0,5 up to 5,5kHz. At the same time, this chapter depicts a general procedure for the design and construction of such a point source, as well as the determination of the related radiation pattern.

Chapter 4

The fourth chapter covers the objective of achieving the volume velocity of a self-made volume sound source (cf. chapter 3) under operation. With this parameter the acoustic transfer functions can be measured reciprocally. Four investigations were presented and discussed theoretically. The functionality as well as the related values of the practical investigations for each approach was studied. From these conclusions it was shown that the free-field method and the monopole approach give the most useful results. The latter is more practical and was applied in the rest of this work.

Chapter 5

To investigate the theory of the reciprocal transfer path analysis (cf. chapter 2) under simplified requirements, a laboratory test was conducted in chapter 5. The self-made volume sound source of chapter 3 was used as noise source, because the related volume velocity of this sound source is already known from the results of the monopole approach and the free-field method (cf. chapter 4).
The results of the concept test for the reciprocal transfer path analysis verified that the theory of chapter 2 generally delivers useful results and that this method could be applied for further applications and component parts from the LuK GmbH & Co. KG.

Chapter 6

Finally the theory of chapter 2 was tested with a component part (gearbox system) from the LuK GmbH & Co. KG in chapter 6. The examined gearbox system was measured on a suitable test stand and was recorded in the in-situ situation. The operating cycle (excitation) was closely reproducible for both cases. A comparison between both situations gave a useful match above a frequency of 2,5kHz. The results for the predicted sound pressure level in the interior showed that both cases (test stand and vehicle) give almost identical values in a frequency range from 0,4 up to 1,25kHz, compared to operational sound pressure level. Hence it can be assumed that the examined gearbox system is, in this frequency range, responsible for the occurring sound pressure level under operation. At smaller frequencies than 0,4kHz the engine orders and other sound sources are more dominant than the gearbox system.

It can be concluded that the reciprocal transfer path analysis could be introduced in the everyday life of the LuK GmbH & Co. KG and serve as a helpful tool for the acoustic engineers.

Outlook

- The applied procedure of the reciprocal transfer path analysis (cf. chapter 6) should be generally investigated with further component parts.
- The introduction of a microphone grid (cf. chapter 5) should be studied more intensively, focusing on the particular parameters and the existing boundary conditions.
- The approach for the design and construction of a monopole sound source (cf. chapter 3) should be regarded additionally for further sound sources that cover another frequency range of interest.
- The determined volume velocity of a point source in chapter 4 should be verified with a commercial loudspeaker system (cf. appendix IV).
- For the performance of the two microphone method (cf. appendix I), one should use either more sensitive microphones or construct a smoother impedance change at the end of the impedance tube.
- There are companies that have already implemented the application of the reciprocal transfer path analysis in a commercial software. An alignment between the described procedure and these software should be performed.
- Measuring the time signal of the volume velocity (e.g. using a PU sound intensity probe), including phase information the sound of the examined component part could be auralized.
- A further investigation that focuses in a different way on the objective of this thesis, is the application of a virtual acoustic prototype [16]. Performing this approach also gives the possibility to auralize the sound of the examined component part in a vehicle, without the need for an installation.

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Appendix I - The two microphone method

As already mentioned in section 4.1 the two microphone method is a useful approach for determining the particle velocity u of a volume sound source under operation. Using the particle velocity and the corresponding cross section S of an impedance tube leads finally to the wanted volume velocity Q (cf. **Eq. 4-3** in section 4.2). The two microphone method was basically applied in this work to study the behavior and sensitivity of this approach. Ín this appendix the theory of the two microphone method and the corresponding results of the practical investigations are therefore briefly presented. Note that the derivation of the theory for the two microphone method focuses only on the relevant issues and can be found in more detail in [6,20,21,22,23,24,29].

The principle of the two microphone method generally requires an existing volume sound source (cf. chapter 3) with the components of a loudspeaker (sound pressure chamber) and an attached impedance tube, with at least one microphone integrated (cf. **Figure I-1**) [24].

The radiation of the loudspeaker generates plane waves in the impedance tube for the applicable frequency range. The considered frequency range depends on the working range of the sound pressure chamber and on the diameter d of the impedance tube (cf. section 3.2). The plane waves will be partly reflected at the end of the impedance tube, whereas another part will be transmitted and radiates as spherical waves. This explains the impedance change at the end of the impedance tube. A horn for example has a very smooth impedance change and is therefore able to transfer more energy to the surrounding area than a conventional loudspeaker or the volume sound source of this work [2].
The sound field in the impedance tube has another impedance than the sound field of the surrounding area of the volume sound source.

In the next figure the procedure of the two microphone method is schematically pictured.

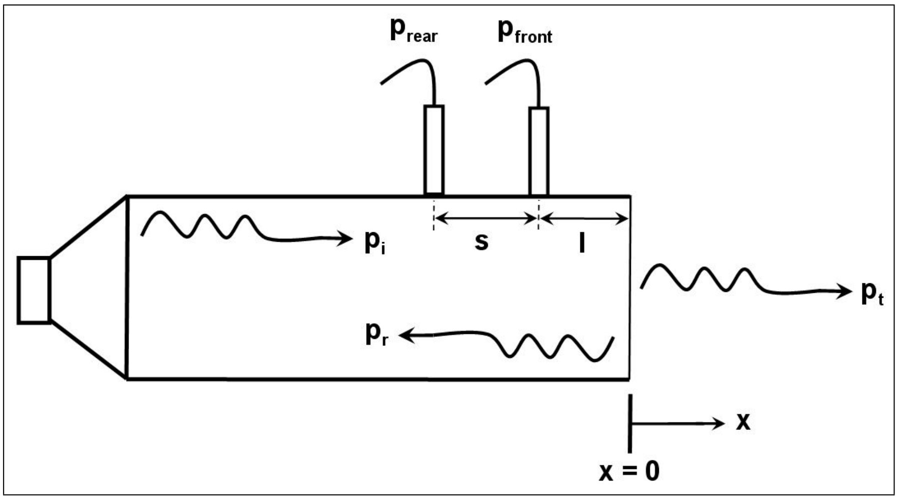

Figure I-1: An illustration for the relating theory of the two microphone method

In **Figure I-1** it can be seen that the most important parameters for describing the above pictured procedure are

- the incoming waves p_i that are generated from the operating pressure chamber loudspeaker
- the reflected waves p_r at the end of the impedance tube
- the transmitted waves p_t at the end of the impedance tube
- the sound pressure p_{rear} of the rear microphone
- the sound pressure p_{front} of the front microphone

The coefficient s describes the distance between both microphones, whereas the distance l gives the offset between the front microphone and the end of the impedance tube (x=0).

The incoming and reflected waves are superposed on each other and build a standing wave pattern in the impedance tube [1,2,20,21,22,23,24]. How strong the overlap between both waves is depends on the reflection coefficient r. This means that the radiated sound in form of the transmitted waves will also be related to the reflection coefficient.

According to the literature [29] the reflection coefficient inside an impedance tube can be calculated with

$$r = e^{2ikl}\frac{(e^{iks} - \underline{H}_{rear-front})}{(\underline{H}_{rear-front} - e^{-iks})} \qquad \textbf{Eq. I-1}$$

Eq. I-1 shows that the measured transfer function $\underline{H}_{rear-front}$ between both microphones in the impedance tube, as well as the corresponding distance parameters s and l are sufficient for calculating the reflection coefficient of a volume sound source.

Focusing on the determination of the volume velocity of a loudspeaker under operation, the literature presents two major investigations of the two microphone method [6,29].

The first version of the two microphone method is expressed by

$$Q = \frac{S}{\varrho_0 c} \frac{1 - (e^{2ikl} \frac{(e^{iks} - \underline{H}_{rear-front})}{(\underline{H}_{rear-front} - e^{-iks})})}{e^{ikl} + (e^{2ikl} \frac{(e^{iks} - \underline{H}_{rear-front})}{(\underline{H}_{rear-front} - e^{-iks})}) e^{-ikl}} \cdot p_{front} \qquad \textbf{Eq. I-2}$$

For the performance of the **Eq. I-2** are

- the measured transfer function $\underline{H}_{rear-front}$
- the cross section S of the impedance tube
- the distance parameters s and l
- the sound pressure p_{front} of the front microphone

needed.

The second version of the two microphone method relies on the usage of the unknown wave amplitudes C and D in the impedance tube [6]. Hence the subsequent relation can be built with

$$D = \frac{p_{front} e^{iks} - p_{rear}}{2isin(ks)} \qquad \textbf{Eq. I-3}$$

$$C = p_{front} - D = \frac{p_{rear} - p_{front} e^{-iks}}{2isin(ks)} \qquad \textbf{Eq. I-4}$$

One sees that **Eq. I-3** and **Eq. I-4** are additionally dependent only on the measured sound pressures (p_{rear} and p_{front}) inside the impedance tube and the distance parameters (s and l).

Introducing the impedance of air ($Z_0 = \varrho_0 \cdot c$) gives finally

$$Q = \frac{S}{\rho c} \left(C e^{-ikl} - D e^{ikl} \right) \Big|_{x=l} \qquad \textbf{Eq. I-5}$$

It has to be mentioned that the second version of the two microphone method has another point of origin (x) in comparison to the first version (cf. **Figure I-1**). Therefore the second case (cf. **Eq. I-5**) has to fulfill the boundary condition *x=l* instead of *x=0*, as it was for the first case (cf. **Eq. I-2**) of the two microphone method.

In contrast to the presented investigations, in chapter 3 a new impedance tube was designed and constructed for the performance of the two microphone method (cf. **Figure I-2**).

Combining the self-made impedance tube of this section and the sound pressure chamber (cf. chapter 3) gives a new volume sound source that is shown below.

Figure I-2: A new volume sound source for the two microphone method

In a next step the result of the application for the two microphone method is plotted. The measurements of the two microphone method were performed with two 1/4" ICP microphones (cf. appendix II) inside the impedance tube. The volume sound source was therefore driven by a white noise as excitation signal.

According to the results in section 4.4 the monopole approach gives the best investigation for determining the volume velocity of a monopole sound source under operation. Hence the following figure shows a comparison between the results of the detected volume velocity for the monopole approach and the two microphone method.
It is noticeable that the results of the monopole approach in this part of the work are not identical to the plots in section 4.3.1. This is because for the two microphone method was a new impedance tube was designed and constructed, as already discussed above. Thus the measurements of the monopole approach were investigated here additionally to the two microphone method of the new volume sound source (cf. **Figure I-2**).

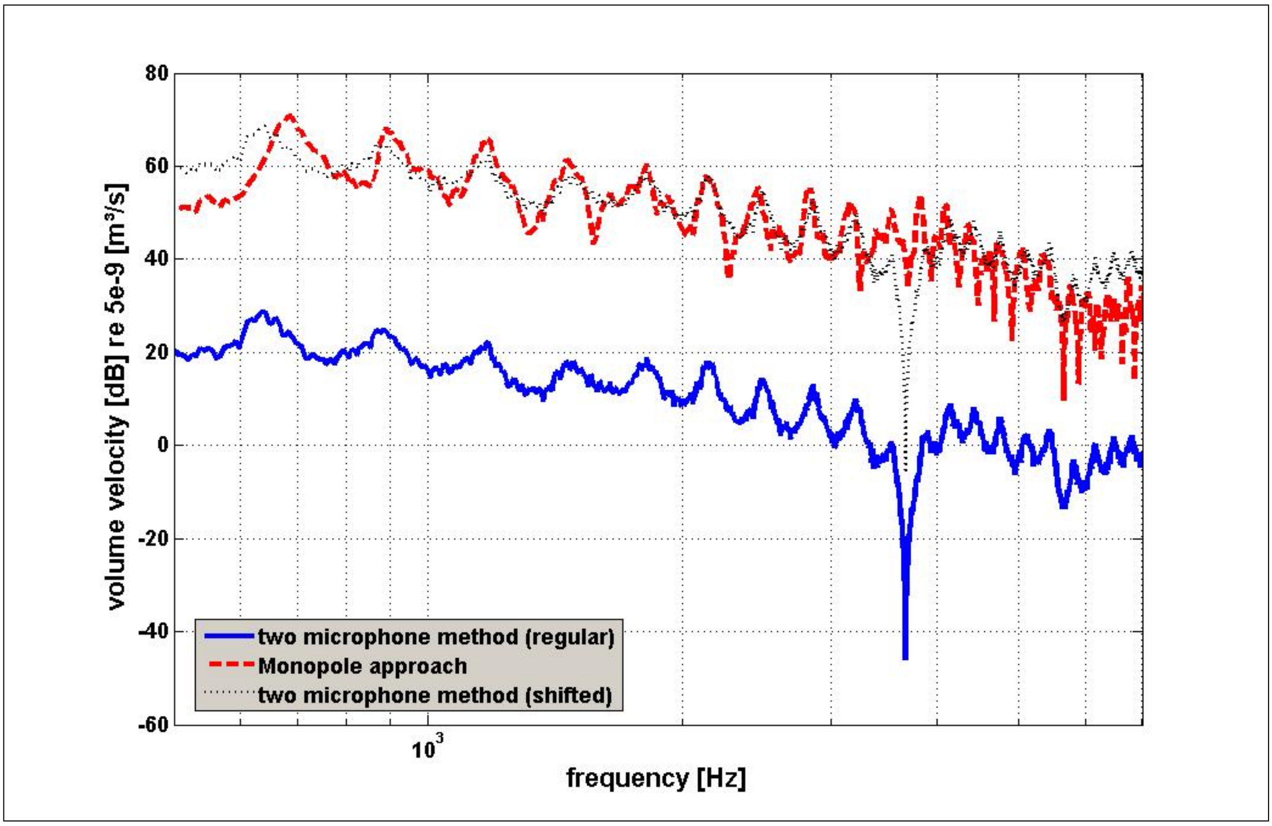

Figure I-3: A comparison between the two microphone method and the monopole approach

The comparison between the two microphone method (regular) and the monopole approach in **Figure I-3** shows that the curves are generally shifted by 40dB. The reason for this is that the measurements were performed with a different amplification. On the one hand, the sound pressure level inside the impedance tube gives too large values to apply the two microphone method, on the other hand, the sound pressure level outside the impedance tube is too small in relation to the background noise level for measurements with the monopole approach (cf. section 4.2.1).

One can conclude that the self-made volume sound source reflects, at the end of the impedance tube, too much of the incoming sound pressure. Increasing the amplification would imply that more sound pressure is transferred to the outside of the impedance tube, but simultaneously the microphones inside the impedance tube would be overloaded. So it turns out that the two microphone method is not usable for the application of the reciprocal transfer path analysis, since the volume velocity can't be detected correctly by overloaded microphones. The limitations of the literature [6,7] focusing on the sensitivity of the microphones when performing the two microphone method are confirmed by the results of this work.

Comparing the two microphone method (shifted) and the monopole approach in **Figure I-3** and concentrating on the behavior of the graphs, it can be seen that the measurement methods deliver nearly identical results. However, the two microphone method has a dip at about 3,5kHz and seems to be slightly shifted in the lower frequency region below 800Hz. The peaks in both graphs show the resonant behavior of the volume sound source that is originally caused by the standing wave ratio inside the impedance tube. Therefore it is interesting to see that the same resonant behavior can be detected outside the impedance tube when looking at the monopole approach.

In summary on the application of the two microphone method, it can be argued that this procedure was unfortunately not applicable for the reciprocal transfer path analysis in this work. However, the general functionality of the two microphone method has been verified in this section. Using more sensitive microphones inside the impedance tube or introducing a smoother impedance change at the end of the impedance tube could be helpful measures for the future.

The next appendix gives a short overview of the most important measurement device that was used in this thesis.

Appendix II - The measurement device

In this appendix the measurement device that was used in this thesis is presented. Thus the reader gets a look at the applied equipment and the corresponding properties.

The semi-free field room of the LuK GmbH & Co. KG

The acoustic measurement cabin of the LuK GmbH & Co. KG is built according to a room-in-room concept. The square base of the semi-free field room is 12 square meters. The walls are double walled and the semi-free field room is decoupled from the foundations of the building by a special bearing. According to the research in [32] it was shown that the average background noise level in this setting is 17dB(A).

The pressure chamber loudspeaker

Figure II-1: The pressure chamber loudspeaker for the self-made volume sound source

The pressure chamber loudspeaker is built of diecast, has a power output value of 50W (RMS) and a resistance of 16Ω. The thread of the pressure chamber loudspeaker is not isometric, so the introduction of an additional adapter was necessary for connecting the pressure chamber loudspeaker with the constructed impedance tube (cf. section 3.3). According to the manufacturer data the pressure chamber loudspeaker has an applicable frequency range from 0,16 to 5,5kHz.

The volume sound source of the LuK GmbH & Co. KG

As discussed in section 4.3.4 the engineers of the LuK GmbH & Co. KG had built their own volume sound source before this thesis. The sound source of the LuK GmbH & Co. KG was re-designed and constructed according to the principle given in [29]. It consists of a pressure chamber loudspeaker and an attached impedance tube.

Figure II-2: The self constructed volume sound source of the LuK GmbH & Co. KG

The diameter of the volume sound source in **Figure II-2** is 18mm, whereas the impedance tube has a length of 180cm. The impedance tube is basically constructed of metal that is additionally cased in a synthetic material.

The microphones

The microphones used in this thesis are exclusively 1/4" ICP microphones.

Figure II-3: The applied microphones in this work [33]

The 1/4" ICP microphones have an applicable frequency range from 0,02 up to 20kHz and can measure a maximum of about 130dB (SPL).

Appendix III - The free-field method

As mentioned in section 4.2.2, the free-field method is a useful tool for determining the wanted volume velocity of a volume sound source. This appendix therefore contains the coordinates of the 20 microphone positions for the application of the free-field method in a semi free-field room [27].

Note that the following figure displays the distance by the parameter r. However, in this work the distance is variabley defined by the abbreviation a (cf. List of notations). Hence the parameter r is identical to the variable a in this case.

Position	x/r	y/r	z/r
1	– 1,00	0	0,025
2	0,50	– 0,86	0,075
3	0,50	0,86	0,125
4	– 0,49	0,85	0,175
5	– 0,49	– 0,84	0,225
6	0,96	0	0,275
7	0,47	0,82	0,325
8	– 0,93	0	0,375
9	0,45	– 0,78	0,425
10	0,88	0	0,475
11	– 0,43	0,74	0,525
12	– 0,41	– 0,71	0,575
13	0,39	– 0,68	0,625
14	0,37	0,64	0,675
15	– 0,69	0	0,725
16	– 0,32	– 0,55	0,775
17	0,57	0	0,825
18	– 0,24	0,42	0,875
19	– 0,38	0	0,925
20	0,11	– 0,19	0,975

Figure III-1: The coordinates of the microphone positions for free-field method (cf. section 4.2.2 in dependency of the distance a) [27]

Appendix IV - A comparison of commercial loudspeaker systems

As already mentioned in section 3.3 this appendix provides a short comparison of commercial loudspeaker systems. Hence the reader gets an overview of the existing monopole sound sources that are available on the market. The shown point sound sources in **Figure IV-1** represent the actual products that are offered by the most well-known manufacturers.
The main aspects of the comparison below focus on the applicable frequency range and the corresponding price of the respective omnidirectional sound source.

	producer	name	freq. range	sound power	price
	LMS	Q - Source	0,2 – 8 kHz	-	~ 15.000 €
	Microflown	mid – high frequency monopole	0,1 – 7 kHz	-	~ 10.000 €
	Microflown	low frequency monopole	30 – 300 Hz	-	~ 10.000 €
	Brüel & Kjær	Dodecahedron Type 4292	0,05 – 6,3 kHz	105 dB	~ 15.000 €
	Brüel & Kjær	OmniSource Type 4295	0,05– 5 kHz	122 dB	~ 15.000 €

Figure IV-1: A comparison of commercial loudspeaker systems with an omnidirectional radiation pattern

Appendix V - The radiation pattern of the LMS Q-Source

The radiation pattern of the LMS Q-Source is presented in the frequency range of 0,63 up to 5kHz in this appendix.

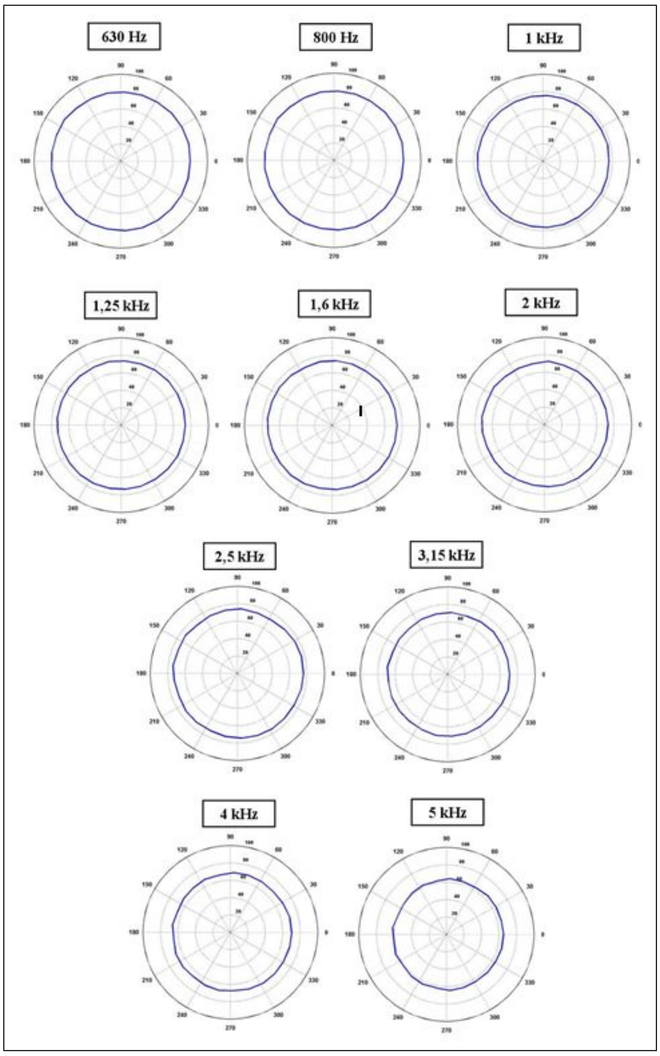

Figure V-1: The radiation pattern of the LMS Q-Source

The results in **Figure V-1** indicate that the LMS Q-Source has an omnidirectional radiation characteristic in the examined frequency range. Comparing the plots in **Figure V-1** with the determined radiation characteristic in **Figure 3-5** (cf. section 3.4) it can be concluded that the LMS Q-Source delivers more precise and constant results than the self-constructed volume sound source.

Appendix VI - Pictures

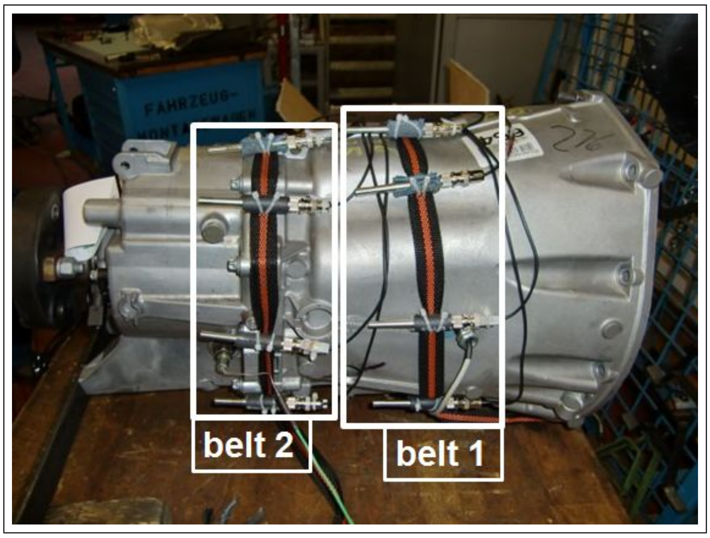

Figure VI-1: The arrangement of the microphone belts for the gearbox system in chapter 6

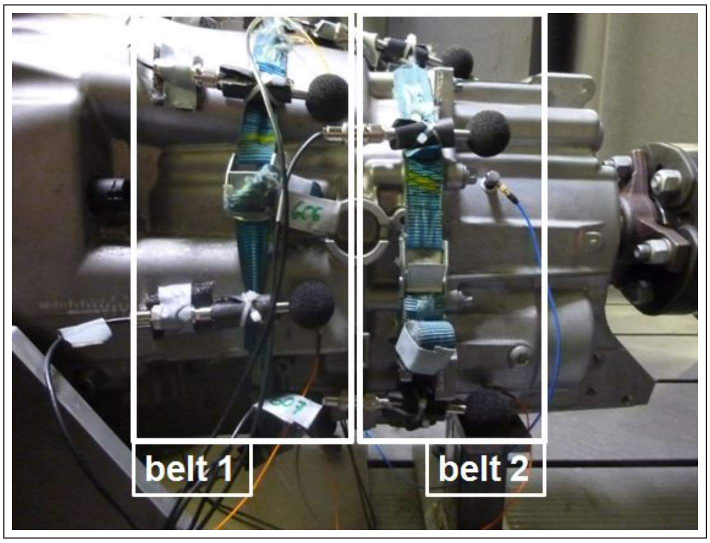

Figure VI-2: Measurements of the volume velocity on the test bench (cf. chapter 6)

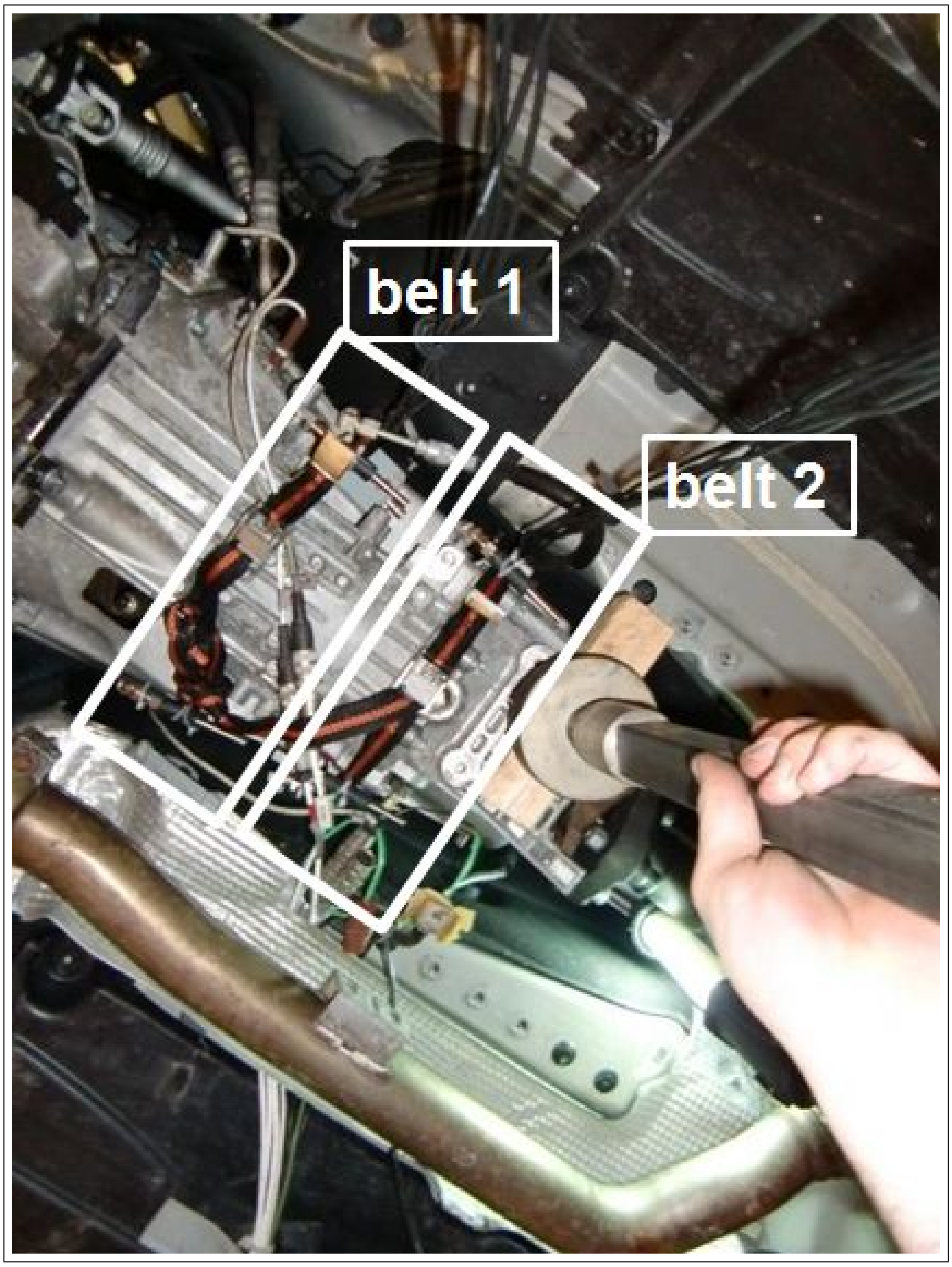

Figure VI-3: Measurements of the volume velocity in the vehicle (cf. chapter 6)