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A Loudspeaker Element as a Passive or Active Resonance Absorber

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Abstract

Absorbing low frequent sound is a challenge, but earlier studies, (Rivet, 2017), (R. Boulandet, 2014) and (Lissek et al., 2011), have shown that a loudspeaker can function well as an absorber with a broader frequency spectre than conventional resonance absorbers. It is claimed that the mechanical properties of the loudspeaker element, which is what makes it to a resonance absorber, can be cancelled by using the right feedback in an active chain. One of the methods, described in (Lissek et al., 2011), shows remarkable results using particle velocity and pressure in an analogue feedback circuit. This method is in this thesis further investigated together with the absorption capabilities of a standard loudspeaker as a semi-active or passive absorber, which means either connected to an amplifier or not.

The thesis firstly presents the theory needed to understand how a loudspeaker can function as an absorber. Further on several measurements are performed; finding the loudspeaker parameters, adjusting the impedance by connecting different resistances to the terminal of the loudspeaker element, adjusting the active system to achieve optimal absorption and finding the effective absorption area of an ordinary loudspeaker in a reverberant room.

An ordinary loudspeaker was found to have high absorption around its resonance frequency. An optimal shunt resistor value could be calculated by the loudspeaker element parameters, and by connecting the optimal shunt resistor, the absorption factor was measured to 1 at the resonance frequency. It is though some loudspeaker elements with parameters so that the optimal resistor value is negative. Those are not recommended to use as active absorbers, but if the negative value is low, they will be good passive absorbers. Overall, the result shows that a traditional loudspeaker has some affection to the acoustical properties of a room.

The active feedback system needs equipment where the sensitivity is not frequency dependent and the phase is unaffected for the frequency range of interest. It does also seem

like the parameters of the loudspeaker element must be carefully chosen to get good results. The results achieved, by using two different loudspeaker elements, indicates that it potentially might work, but was with the given loudspeaker elements far from as good as the results achieved by a previous study (Lissek et al., 2011). Suggestions for optimal loudspeaker parameters are given.

Sammendrag

Å absorbere lavfrekvent lyd på en praktisk måte er utfordrende, men tidligere studier, (Rivet, 2017), (R. Boulandet, 2014) og (Lissek et al., 2011), har vist at et høyttalerelement kan fungere godt som absorbent over et bredere frekvensområde enn tradisjonelle resonansabsorbenter. Det hevdes at de mekaniske egenskapene til høyttalerelementet kan kanselleres ved bruk av riktig feedback i et aktivt oppsett. En av metodene, beskrevet i (Lissek et al., 2011), har oppnådd forbausende resultater ved å bruke partikkelhastighet og trykk i en analog tilbakekoblingskrets. I oppgaven vil denne metoden undersøkes videre. I tillegg skal det ses på effekten av en standard høyttaler brukt som halv-aktiv eller passiv absorbent, noe som vil si koblet til forsterker eller ikke.

Oppgaven vil først presentere teorien som trengs for å skjønne hvordan en høyttaler fungerer som absorbent. Videre er flere målinger gjort; måling av høyttalerelementets parametere, endring av impedans ved innkobling av forskjellige motstander, justere det aktive systemet mot optimal absorbsjon, og måling av effektivt absorpsjonsareal i et klangrom.

Resultatene viser at et vanlig høyttalerelement har høy absorbsjon rundt sin resonansfrekvens. En optimal motstandsverdi kan bli kalkulert ut fra høyttalerens parametere. Ved innkobling av denne ble absorpsjonsfaktoren målt til 1 ved resonans. Noen høyttalerelement har parametere som vil gi negativ optimal motstandsverdi. Disse viser seg å være mindre egnet som aktive absorbenter, men om den negative verdien er lav kan de egne seg godt som passive. Alt i alt viser resultatene at en standard høyttaler vil utgjøre en endring i rommets akustikk.

Det aktive tilbakekoblingssytemet trenger utsyr der sensitiviteten er konstant for frekvensområdet man er interessert i. Fasen må heller ikke påvirkes. Det ser også ut som høyttalerelementet må velges omstendelig for å oppnå gode resultater. For de to høyttalerelementene brukt i denne oppgaven viser resultatene at metoden har potensiale,

men resultatene er langt fra så gode som vist tidligere i (Lissek et al., 2011). Anbefalte høyttalerparametere for bedre absorbsjon er diskutert.

Preface

This thesis is submitted as a part of the Master's Programme at the Department of Civil and Environmental Engineering, Norwegian University of Science and Technology (NTNU). The work has been done from February to July 2017 and is a continuation of a project done in preparation for the master thesis during the previous semester. Parts of two sub-chapters (2.3 and 2.4) in the theory and two figures (Figure 4.7 and Figure 4.14) in the results are taken from the project thesis.

Several people deserve gratitude for their contribution to this thesis. First, I would like to thank Anders Homb from the Department of Civil and Environmental Engineering, for being the responsible supervisor. Thanks to Peter Svensson from the Department of Electronic Systems, for being the primary supervisor with weekly guidance throughout the project. He has done a brilliant job taken me through the theory and come up with valuable advises. We have not had a minute to spare! Thanks also to Tim Cato Netland from Department of Electronic Systems for instructing me through a lot of equipment needed for the measurements, good advices and help when needed. Tore Landsem and Tore Berg from the Tele Engineering Workshop deserves gratitude for always putting me at the top of their list and by that saving me valuable time. I will also thank Tor Erik Vigran from Department of Electronic Systems for instructing me through the use of the impedance tube and letting me use it much longer than planned.

Last but not at least I want to thank my girlfriend for support and proofreading and my family for supporting me through all my years as a student.

Stian Sverdrup Lilleeng, Trondheim, July 2017

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Nomenclature

ρ_0	Equilibrium density
A	Equivalent absorption area
a	Effective radius
Bl	Force factor
c	Speed of sound
C_a	Acoustical compliance
C_{ms}	Mechanical compliance
d	Maximum side length
DF	Damping factor
e_G	Voltage
F	Force
f	Frequency
f_s	Resonance frequency
i	Current
j	Imaginary unit
k	Wave number
k_m	Mass correction factor
l	Length
L_{ec}	Electrical inductance of the coil
l_{eff}	Effective length
m	Air absorption coefficient
M_{md}	Mechanical moving mass of diaphragm without airload
M_{ms}	Total mechanical mass
p	Sound pressure
R	Reflection factor
R_{ec}	Electrical DC-resistance
R_{res}	Resistance at resonance
$R_{1,2}$	Chosen resistance value for finding frequency 1 and 2.
R_{ms}	Mechanical resistance
R_{opt}	Optimal value of a shunt resistance
R_{shunt}	Shunt resistance
s	Distance between microphones
S	Surface area
S_{lsp}	Cross section area of loudspeaker
S_{tube}	Cross section area of tube
T	Reverberation time
u	Particle velocity
U	Volume velocity
V	Volume
V_{cav}	Volume of cavity

$V_{lsp\ cav}$	Volume of loudspeaker cavity
x_1	Distance from specimen to the first microphone
Z	Impedance
Z_a	Acoustic impedance
Z_s	Specific acoustic impedance
Z_e	Electrical impedance
Z_{load}	Load impedance
Z_m	Mechanical impedance
$Z_{m,lsp}$	Mechanical impedance of loudspeaker
$Z_{mr'}$	Mechanical radiation impedance, front
$Z_{mr''}$	Mechanical radiation impedance, rear
Z_{out}	Output impedance
α	Absorption factor
λ	Wave length
ω	Angular frequency
Q_{ms}	Mechanical quality factor
Q_{es}	Electrical quality factor
Q_{ts}	Total quality factor

1 INTRODUCTION

1.1 Background

In a time with high urbanization rate, more noise sources (traffic, loudspeaker systems, ventilation systems, night-clubs, etc.) and more demanding people, it has never been more important to find good solutions to absorb sound. In the high frequency domain, several products can be used with good result for relatively low cost and little space required. On the other hand, in the low frequency domain it is not many solutions that do not require much space or money. Resonance absorbers is commonly used, but most of them are highly frequency dependent and have their effective range for a narrow frequency spectre (Cox and D'Antonio, 2009). For a wider frequency range, different traditional resonance absorbers must be combined to achieve good performance. Active sound absorbers may make it possible to extend the frequency bandwidth over which traditional resonance absorbers present a significant sound absorption capability (Rivet, 2017).

There are some active absorbers available in the market already. Bag End and Spatial Audio offer active bass traps that uses a microphone in front to measure the sound pressure and sends feedback to a loudspeaker, in which the phase is inverted (Bag End, 2014), (Spatial Audio, 2017). The E-Trap from Bag End is able to be tuned for two frequencies, while Spatial Audios Black Hole can only focus at one. The intentions are to damp one or two modes. As will be described later, the main problem using a loudspeaker as absorber is that the mechanical properties will cause a phase shift between the input pressure and the velocity of the diaphragm, which will set limitations to the absorbing capabilities above or below the resonance frequency. It is not known whether Bag End and Spatial Audio compensate for that or not. Bag End describes in the specifications that the Q-value can only be adjusted from narrow to narrower, which indicates absorption in a narrow frequency range. PSI Audio has just come up with an interesting solution using that is claimed to work for a broader frequency band and not just for one or two modes (PSI Audio, 2015b). The AVAA (Active Velocity Acoustic Absorber) as it is called, uses a microphone in front to measure the sound pressure, but as opposed to the E-Trap and the Black Hole, the loudspeaker has now a grid with acoustic resistance some distance in front. The loudspeaker element is used to increase the velocity at the grid by achieving zero pressure in front of the loudspeaker. High velocity through an acoustic resistance layer will cause damping to the incoming sound.

1.2 Motivation

An alternative to the above mention products is using a loudspeaker element with both pressure- and velocity feedback. Previous studies (Lissek et al., 2011), (R. Boulandet, 2014), (Rivet, 2017), has shown that using a loudspeaker in such an active way might achieve broad-banded absorption in the lower frequency specter. Lissek illustrates in (Lissek et al., 2011) that by use of a pressure- and velocity feedback, the mechanical properties of the loudspeaker can be cancelled, and in that way extend the frequency band where the absorber is efficient. It might seem like ordinary loudspeakers and amplifiers can be used and sensors added externally. If the system can tolerate sensors that do not need to be highly accurate, the solution might be cheap enough to be popular for every room struggling with too much low-frequent noise, or rooms for sound reproduction or studios that need to damp the lower room modes. The study by Lissek et al. show remarkable results, both simulated and for a few measurement setups. The implementation seems easy and straightforward.

1.3 Problem Description

The aim in this thesis is to confirm that the technique mentioned earlier in this chapter works, and see how easily implemented it really is. The previous studies of the topic explain the procedure briefly and details are left out. Limitations and uncertainties of the method will be tried identified and explained.

Another important aspect that the thesis will cover is the effect an ordinary loudspeaker has on the room. A standard procedure of measuring the acoustics of a room is to bring a loudspeaker and measure the reverberation time (ISO, 2008). If the loudspeaker affects the rooms acoustical properties, the measurement will also be affected and the properties will be different than without the loudspeaker.

1.4 Structure

Chapter 2 contains the relevant theoretical background needed to understand the measurements. It explains sound absorbers with focus on resonance absorbers and how a loudspeaker can be used, both in a passive and an active way.

Chapter 3 presents the measurement setup, how the measurements are performed and the equipment used. The loudspeaker parameters, the absorption in a waveguide and the absorption in a room are measured.

Chapter 4 contains the results of the measurements and discussions around it, including sources of error.

Chapter 5 presents the conclusion and suggestions for further work.

2 THEORY

2.1 Sound Absorbers

A sound absorber has the function of transmitting a part of the sound waves energy into other forms of energy. Sound absorbers can be divided into two groups, porous absorbers and resonance absorbers. The porous absorber consists of small pores, allowing air passing through. The pores are narrow and will absorb some energy from the sound wave mostly due to friction. Since the friction loss is highest at high particle velocities the porous absorber is most efficient where the particle velocity of the sound wave is highest. That is a quarter of the wavelength from a hard wall (Kinsler et al., 2000). Exemplified, when a sound wave with frequency of 100 Hz and a wavelength of 3.4 meters should be attenuated, the absorber needs to be placed almost a meter away from the wall, and is therefore no good solution to the low frequency domain considering the space required. Resonance absorbers on the other hand, can be tuned to absorb well at low frequencies without requiring too much space. A resonance absorber uses high pressure to excite a mass to high velocity and due to resistance in the system energy will be transferred to heat. A resonance absorber consists of a mass, a spring and a resistance, where the mass and the spring forms a resonator and the resistance cause the losses in the system.

The difference in impedance between the air and the absorber decides how much of the sound wave that is transmitted or reflected. The definition of mechanical impedance, Z_m, and acoustic impedance, Z_a, are given by eq. (2.1) and eq. (2.2):

$$Z_m = \frac{F}{u} \tag{2.1}$$

$$Z_a = \frac{p}{U} \tag{2.2}$$

Where F is force, u is particle velocity, p is sound pressure, and U is volume velocity (U=u*S, where S is the surface area).

2.2 Effective Absorption Area

The effective absorption area tells how big the absorber must be to have the required impact on the room. Large surface area, S, that have a high absorption factor, α, will give large equivalent absorption area, A, but for resonance absorbers the effective area might in fact be bigger than the surface area of the absorber. It can be shown that a perfect resonance absorber can achieve a maximum equivalent absorption area (Vigran, 2008):

$$A_{\max at\, f_0} = \frac{\lambda_0^2}{2\pi} \tag{2.3}$$

Where λ_0 is the wave length at resonance. This means that it is possible to get a large absorption area for low frequencies.

The effective absorption area is calculated by measuring the room's reverberation time, T. T is defined as the time it takes for the sound pressure level to decrease by 60 dB after the sound source is shut off. The relation between the reverberation time of the room, the volume of the room and the absorbing characteristics of the rooms surfaces are given in the equation below (Kinsler et al., 2000).

$$T = \frac{24\ln(10)V}{cA} \tag{2.4}$$

Where V is the volume of the room, A is the effective absorption area and c is the speed of sound. By measuring the reverberation time in a room with and without the absorber and solving eq. (2.4) for A, the effective area of the absorber A_{abs} will be known.

$$A_{abs} = \frac{24\ln(10)\,V}{c}\left(\frac{1}{T_{with\ absorber}} - \frac{1}{T_{without\ absorber}}\right) \tag{2.5}$$

The absorber should have a significant impact to the room for achieving reliable results. It is therefore recommended to do such measurement in a reverberant room. The equation also assumes diffuse field which is valid above the Schroeder frequency given by (Kinsler et al., 2000):

$$f_{Schroeder} \geq 2000\sqrt{\frac{T}{V}} \tag{2.6}$$

There are two common ways to define the effective absorption area, given by two well-known acousticians, Sabine and Eyring:

$$A_{Sabine} = 4mV + S\bar{\alpha} \tag{2.7}$$

$$A_{Eyring} = 4mV - Sln(1 - \bar{\alpha}) \tag{2.8}$$

Where m is the air absorption coefficient, V is the volume of the room, S is surface area and

$$\bar{\alpha} = \sum_i S_i \alpha_i \tag{2.9}$$

For low total absorption in the room, $A_{Sabine} \approx A_{Eyring}$, but for higher absorption the formula by Eyring is the most correct. The absorption factor, α, can be derived from the equations above by knowing the effective absorption area and the surface area of the absorber. However, since the effective absorption area can be higher than the surface area for resonators, the absorption factor might exceed 1. The absorption factor is therefore in such case preferred estimated using another method.

2.3 Absorption Factor

A standardized method (ISO, 1998) to find the absorption factor is using an impedance tube. A loudspeaker in one end is used as a source, and plane waves will propagate in the length direction of the tube. The test material is placed on the other side marked with dashed line in Figure 2.1. The incident and the reflected waves are compared with use of two microphones with a given distance from each other, s, and the test specimen, x_1. The pressure at each microphone can be expressed as a sum of the incident and the reflected wave, and the amplitudes of each of the two waves can be assumed not to change between the two microphone positions.

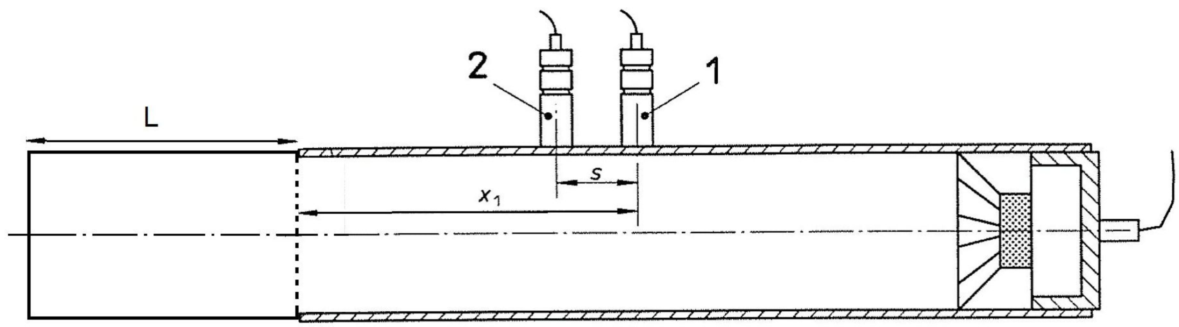

Figure 2.1 Impedance tube (ISO, 1998)

This method is only valid when the waves in the tube are plane. This is true up to the first transversal mode given by (ISO, 1998):

$$f = 0.5\frac{c}{d} \tag{2.10}$$

Where d is the maximum side length.

Also, when the microphones are λ/2 apart, the errors become very large.

For plane sound waves going from one media (with given impedance Z_1) to another (Z_2) the reflection factor R can be written like this (Kinsler et al., 2000):

$$R = \frac{Z_2 - Z_1}{Z_2 + Z_1} \tag{2.11}$$

By using this formula, the real part of the radiation impedance in front of the specimen should not be included in the expression for Z_2.

With known reflection factor the absorption factor is easy to calculate by eq. (2.12)

$$\alpha = 1 - |R|^2 \tag{2.12}$$

2.4 Loudspeaker as a Sound Absorber

A loudspeaker element placed in a box will act like a mass-spring system. The volume of air in the box, V, together with the stiffness, $1/C_{ms}$, of the loudspeaker element will act as the spring, and the mass of the diaphragm, M_{md}, plus air load will be the total mass. Additionally, there will be resistance in the coil, R_{ec}, when it is connected and some resistance in the membrane suspension, R_{ms}, as well. Those resistances together with potential friction against the walls in the box will cause losses to the reflecting sound waves and give the loudspeaker element potential to be an absorber. The driving force is the sound pressure multiplied with the surface area of the loudspeaker element. Since the sound pressure for low frequencies will be distributed around the absorber, the absorber will function for all incidence angles. The loudspeaker element and its mechanical and electrical components are illustrated in Figure 2.2. All notations are given in the nomenclature list.

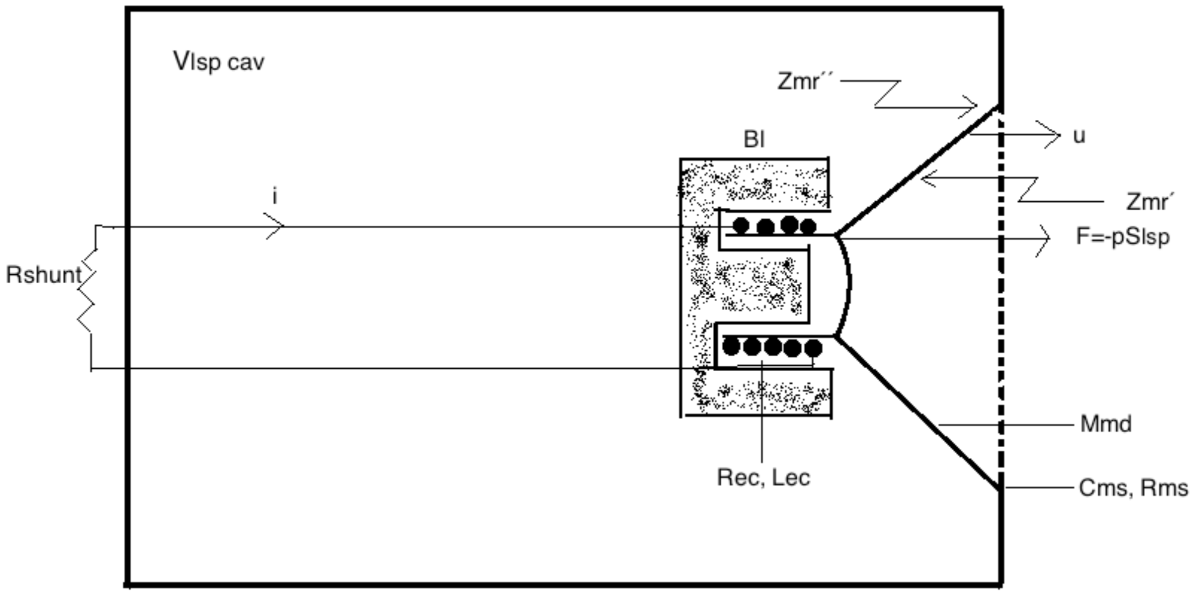

Figure 2.2 Loudspeaker element in a box

The loudspeaker can be used as an absorber in three ways: as a passive resonator, a semi-active resonator and as an active resonator. A shunt resistance is connected to the element in the drawing. By choosing the value of the resistance it can represent the three alternatives.

2.4.1 Passive and Semi-Active

As a passive resonator, the loudspeaker element is not connected to anything, that means $R_{shunt}=\infty$ and has therefore no circuit. No circuit will be the same as using a slave element, i.e the coil and the magnet has no contribution.

The semi-active term refers to having a positive electric resistance connected, i.e $0<R_{shunt}<\infty$. That implies a circuit through the coil and electrical current that generates a feedback force at the loudspeaker diaphragm, modifying its vibrating velocity which the sound pressure is causing. R_{shunt}=0 is the same as short-circuiting the loudspeaker element.

As long as the circuit is closed, the inductance, L_{ec}, and the resistance, R_{ec}, of the coil, together named Z_{ec} must be included. The loudspeaker is illustrated in Figure 2.3, showing the electrical and the mechanical side of the loudspeaker element. The electrical side is converted and the system is illustrated in a mechanical mobility analogy in Figure 2.4, and in a mechanical impedance analogy in Figure 2.5.

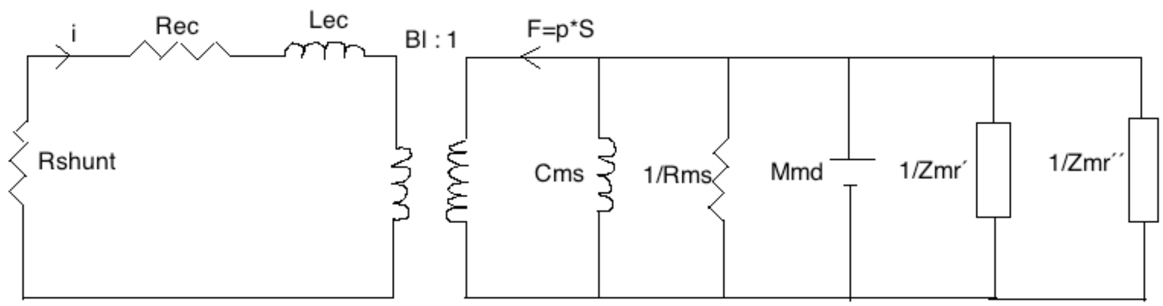

Figure 2.3 Electrical and mechanical parts of the loudspeaker element including a shunt resistance

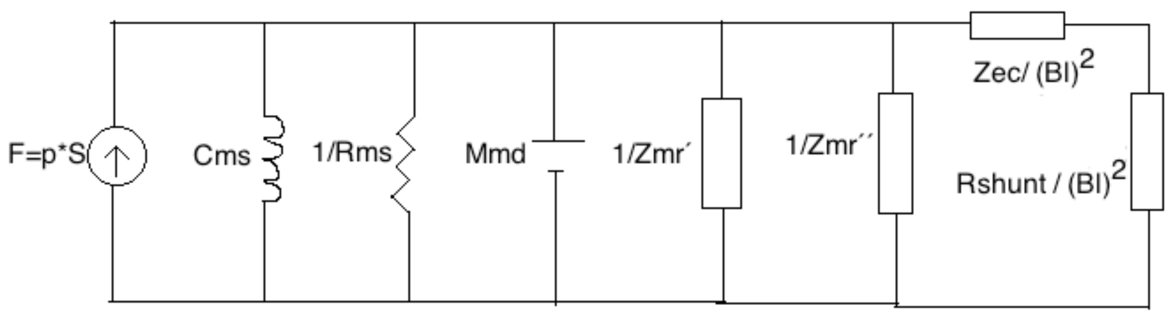

Figure 2.4 Mechanical mobility analogy for the loudspeaker element including a shunt resitance

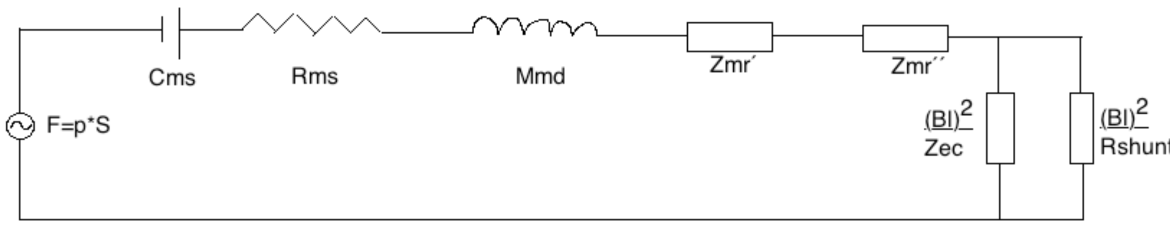

Figure 2.5 Mechanical impedance analogy for the loudspeaker element including a shunt resistance

From the impedance analogy in Figure 2.5 we can set up the expression of the total theoretical impedance of the loudspeaker element, $Z_{m,lsp}$.

$$Z_{m,lsp} = j\omega M_{md} + \frac{1}{j\omega C_{ms}} + R_{ms} + Z_{mr'} + Z_{mr''} + \frac{(Bl)^2}{Z_{ec} + R_{shunt}} \tag{2.13}$$

$$Z_{mr'} = \frac{\pi \rho_0 a^4 \omega^2}{2c} + j\frac{\omega B' \rho_0 S_{lsp}^2}{\pi a} \ (ka < 0.5) \tag{2.14}$$

$$Z_{mr''} = \frac{S_{lsp}^2}{j\omega C_a} + j\frac{\omega B'' \rho_0 S_{lsp}^2}{\pi a} \ (ka < 0.5) \tag{2.15}$$

Where B’ and B’’ is the end-correction factor for the reactance term of the radiation impedance of the loudspeaker diaphragm at front and rear side respectively (Beranek and Mellow, 2012). ρ_0 is the equilibrium density and a is the effective radius. The given expressions are valid for low frequencies as indicated, where k is the wave number. The last term of the radiation impedance is depending on the surface area of the loudspeaker element compared to the cross section of the volume it is radiating against. When the difference is big, it will be the same as for a piston in an infinite baffle, and when the cross-section area is the same as the surface area of the element, it will be as a closed tube. The closed volume behind the loudspeaker element, $V_{lsp\ cav}$, works like a spring for low frequencies which has the acoustic compliance:

$$C_a = \frac{V_{lsp\ cav}}{\rho_0 c^2} \tag{2.16}$$

From eq. (2.13) it can be seen that the total impedance will change by applying different shunt resistances. The impedance is at lowest for $R_{shunt}=\infty$, i.e. using the loudspeaker passive. It will then be no circuit and the last term will vanish. The highest impedance is given when $R_{shunt}=0$, which is the same as short-circuiting the loudspeaker.

2.4.2 Active

Active loudspeaker means that the loudspeaker is connected to an amplifier as illustrated in Figure 2.6 where it is shown an ordinary loudspeaker only subjected to the input voltage from the amplifier, e_G, and not incoming sound pressure.

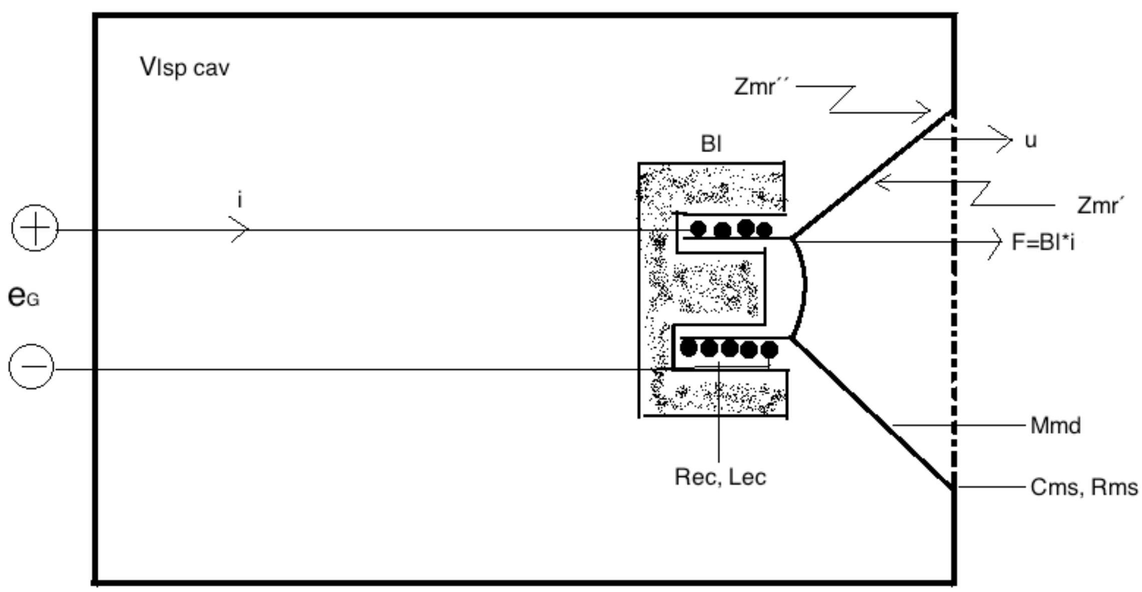

Figure 2.6 Loudspeaker element connected to an amplifier

According to (Self, 2013) the audio amplifiers will be constructed to have low output impedance so that the output is unaffected by loading. That is, the frequency variable impedance of the loudspeaker does not give an equally variable frequency response. The damping factor, DF, is defined as:

$$DF = \frac{Z_{load}}{Z_{out}} \tag{2.17}$$

Where Z_{load} in this case is the total impedance of the loudspeaker, and Z_{out} is the output impedance of the amplifier. We can see from the equation that DF will be high if Z_{out} is low, so if the amplifier is specified to have a high DF, which it normally is, the output resistance will be low. That means having an amplifier connected (power on) to the loudspeaker element will be the same as having the element short-circuited (R_{shunt}=0), but when the amplifier gets incoming signals, voltage and current from the amplifier will affect the impedance of the system as function of the force factor which can be seen in Figure 2.7.

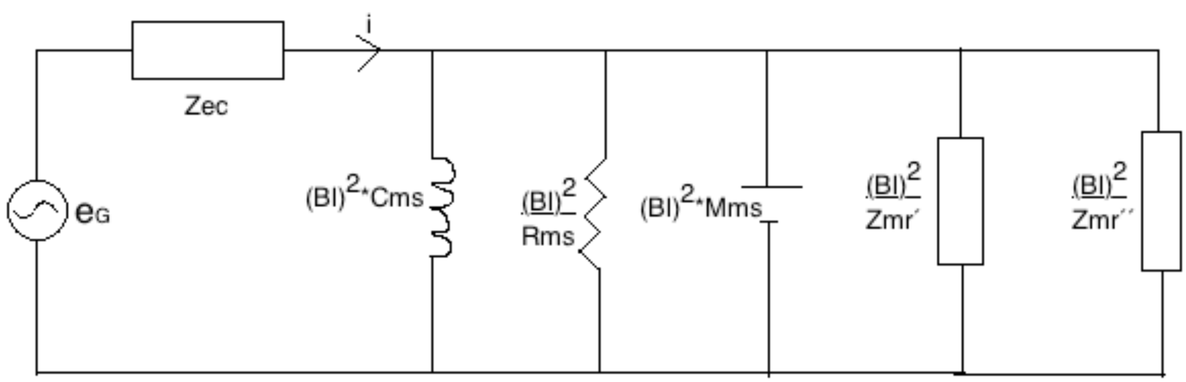

Figure 2.7 Electric impedance circuit representation

Input signals to the amplifier will change the force and the velocity at the element, and therefore also the impedance as shown in eq. (2.1). The goal is to match the impedance of the loudspeaker element to the impedance of the medium in front. A combination of using the right in-signals to the amplifier and having the right properties of the loudspeaker element might be a possible way to achieve that (Lissek et al., 2011).

2.5 Optimizing the Loudspeaker as a Sound Absorber

In this chapter, the optimization of the loudspeaker as an absorber in a plane wave tube will be presented. A plane wave condition can also be valid in a 3D room. Adrian Celestionos showed in his Ph.D thesis that by placing loudspeakers in the right positions, the first room modes can be supressed for rectangular rooms (Celestinos, 2006). The optimal positions for two loudspeakers are ¼ and ¾ in the width of the room and ½ in the height. Using more loudspeakers can supress more modes. The room will for frequencies below the first non-supressed mode act as a waveguide. Celestinos did also show that absorption of the low frequency sound can be achieved using two more loudspeakers in the rear wall of the room by giving them the same signal as the two acting as sound sources, but with opposite polarity, a time lag representing the length of the room and a reduced gain. This is relevant for music reproduction rooms, since a plane wave propagation with no or reduced reflection from the back wall, will increase the listening experience. It does also indicate that a loudspeaker used as absorber might work in 3D rooms.

2.5.1 Semi-Active

At resonance, the imaginary parts of the impedance cancel each other. The mechanical and acoustic impedance of the loudspeaker element can then be written like this

$$Z_{m,lsp\_at\_res} = R_{ms} + \frac{(Bl)^2}{R_{ec} + R_{shunt}} \tag{2.18}$$

$$Z_{a,lsp\_at\_res} = \frac{R_{ms} + \frac{(Bl)^2}{R_{ec} + R_{shunt}}}{S_{lsp}^2} \tag{2.19}$$

To achieve perfect absorption at resonance, the loudspeaker impedance must be equal to the impedance in front of the loudspeaker, which can be seen from eq. (2.11). In the standing wave tube, the acoustical impedance is

$$Z_{a,air} = \frac{\rho_0 c}{S_{tube}} \tag{2.20}$$

Where S_{tube} is the cross-sectional area of the tube.

By setting eq. (2.19) equal to eq. (2.20), we get

$$R_{ms} + \frac{(Bl)^2}{R_{ec} + R_{shunt}} = \rho_0 c \frac{S_{lsp}^2}{S_{tube}} \tag{2.21}$$

And end up with an expression for the optimal shunt resistance given by eq. (2.22)

$$R_{shunt\_optimal} = \frac{(Bl)^2}{\rho_0 c \frac{S_{lsp}^2}{S_{tube}} - R_{MS}} - R_{EC} \tag{2.22}$$

The ratio of the areas is important. When the loudspeaker is placed in a room, the ratio will be very small. As can be seen from eq. (2.21) the expression on the right side will be of low value. When the loudspeaker is connected to an amplifier (R_{shunt}=0) the R_{ms} value has to be very small to fulfil the expression for perfect absorption. The Bl value also has to be small and the R_{ec} value has to be large to get a low enough value.

2.5.2 Active

The definition of specific acoustic impedance is given by:

$$Z_S = \frac{p}{u} \tag{2.23}$$

And the characteristic specific impedance for air is $\rho_0 c$. By adjusting the feedback, the impedance of the loudspeaker element is supposed to match the impedance of the air in front. That means

$$\frac{p}{u} = \rho_0 c \tag{2.24}$$

Where p is the total pressure at the surface of the diaphragm and u is the total velocity of the diaphragm. As $\rho_0 c$ is constant, p/u must be constant too. Both mechanical reactance's (suspension compliance, C_{ms}, and dynamic mass, M_{ms}) in the loudspeaker impedance cause the diaphragm velocity to be out-of-phase with the input sound pressure by $\pi/2$ and $-\pi/2$, respectively, away from the loudspeaker mechanical resonance (R. Boulandet, 2014). At the resonance frequency, it is possible to adjust the total resistance of the loudspeaker to get the correct ratio of p and u, described in the previous subchapter. But away from the resonance frequency, the phase-shift is causing the ratio to be time variant. A system that cancel the mechanical properties of the loudspeaker element is therefore desired.

2.6 Finding the Loudspeaker Parameters

According to Ohms law the electrical impedance is defined as voltage divided by current. The electrical impedance of the loudspeaker element can be found as illustrated in Figure 2.8.

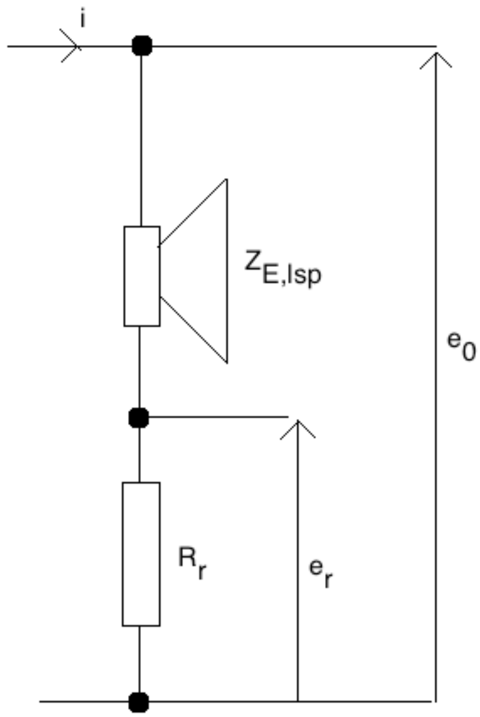

Figure 2.8 Determination of the electrical impedance

The current, i, is measured by inserting a small reference resistor, R_r, in series with the unknown electrical impedance, $Z_{e,lsp}$. The impedance is then found by using eq. (2.25) (WinMLS, 2004).

$$Z_{e,lsp} = \frac{e_0 * R_r}{e_r} - R_r \tag{2.25}$$

Where e_r is the voltage over the resistor R_r and e_0 is the voltage over the loudspeaker element and the resistor.

By finding the loudspeaker resonance frequency and the absolute value of the impedance at the peak, $R_{res} = |Z(f_s)|$, and in addition measure the DC voltage, R_{ec}, the mechanical quality factor, Q_{ms}, can be found using eq. (2.26) (W. Marshall Leach, 1999)

$$Q_{ms} = \frac{f_s}{f_2 - f_1}\sqrt{\frac{R_{res}}{R_{ec}}} \tag{2.26}$$

Where f_1 and f_2 is frequencies at a given resistance

$$R_{1,2} = \sqrt{R_{ec} * R_{res}} \tag{2.27}$$

The quality factor do also have the relation (Iversen et al., 2016)

$$Q_{ms} = \frac{2\pi f_s M_{ms}}{R_{ms}} \tag{2.28}$$

To find the other parameters, another measurement is needed. The probably most common method adds a mass to the loudspeaker diaphragm causing a lowered resonance frequency. Since the resonance frequency is given by

$$f_S = \frac{1}{2\pi}\sqrt{\frac{1}{C_{ms} * M_{ms}}} \tag{2.29}$$

Changing the mass and knowing the resonance frequencies of the two measurements will give us the mass of the diaphragm including air load, M_{ms}.

$$M_{ms} = \frac{Added\ mass}{\left(\frac{f_S}{f_{S,with\ added\ mass}}\right)^2 - 1} \tag{2.30}$$

$$M_{md} = M_{ms} - Airload \tag{2.31}$$

Where (Beranek and Mellow, 2012)

$$Air\ load = \begin{cases} \frac{16a^3\rho_0}{3} & for\ a\ piston\ in\ an\ infinite\ baffle\ \ radiating\ at\ two\ sides \\ \frac{8a^3\rho_0}{3} & for\ a\ piston\ radiating\ at\ two\ sides \end{cases} \tag{2.32}$$

C_{ms} can then be found from eq. (2.29) and R_{ms} from eq. (2.28).

The electrical quality factor, Q_{es}, and the total quality factor, Q_{ts}, is given by eq. (2.33) and (2.34) (W. Marshall Leach, 1999).

$$Q_{es} = Q_{ms} \frac{R_{ec}}{R_{res} - R_{ec}} \tag{2.33}$$

$$Q_{ts} = Q_{ms} \frac{R_{ec}}{R_{res}} \tag{2.34}$$

The force factor, Bl, can be found using eq. (2.35) (Rife, 1991).

$$Bl = \sqrt{\frac{R_{ec}}{2\pi * f_s * C_{ms} * Q_{es}}} \tag{2.35}$$

The parameters can be converted to infinite baffle parameters by dividing f_s by the mass correction factor, k_m, and multiplying Q_{ms} and Q_{es} by the same factor (W. Marshall Leach, 1999). Where

$$k_m = \sqrt{\frac{M_{ms\,(ib)}}{M_{ms\,(fa)}}} \tag{2.36}$$

Where $M_{ms(ib)}$ is the mass of the diaphragm plus air load in an infinite baffle and $M_{ms(fa)}$ is the mass of the diaphragm plus air load in free air.

3 MEASUREMENTS

3.1 Measurement of Loudspeaker Element Parameters

3.1.1 Setup

Using the WinMLS software and setup as shown in Figure 3.1, the electrical impedance as function of frequency was measured. The measurement was performed in an anechoic room to not get reflections disturbing the radiation impedance of the loudspeaker. The surfaces of the anechoic room are perfectly absorbing down to around 100 Hz. The contribution from the reflections below 100 Hz are assumed not to have a significant influence on the measurements.

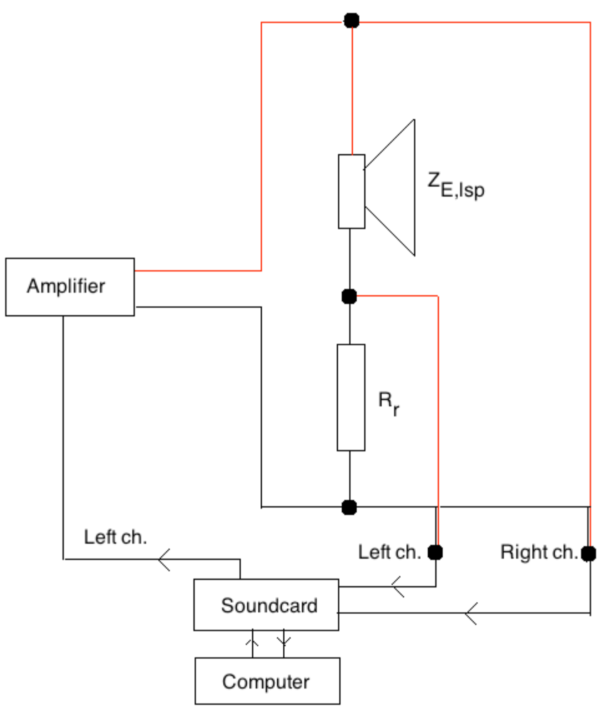

Figure 3.1 Connection diagram for electrical impedance measurement

3.1.2 Equipment

The equipment used to measure the loudspeaker element parameters are listed in Table 1.

Table 1 List of equipment for measuring the loudspeaker element parameters

Device	*Manufacturer*	*Model*
Multimeter	Fluke	77
Software	Morset Sound Development	WinMLS 2004
Computer	HP	Nc6400
Soundcard	AXYS	D-audio
Loudspeaker amplifier	QUAD	50E
Loudspeaker element	SEAS	H0489
Loudspeaker element	PRV	6MB200

3.1.3 Method

The loudspeaker parameters were measured using the software WinMLS and by use of the added mass method described in chapter 2.6. Two loudspeaker elements were measured. It was found best to have the loudspeaker element placed horizontally on the floor for the measurements. The weight of the diaphragm will cause some displacement from the neutral position when the element is vertical, but keeping the element vertical without causing reflections from other surfaces and adding an extra mass evenly distributed is hard with the element vertically. With the specified stiffness from the datasheets, the extra displacement due to horizontal placement was calculated and found to be (including added mass) 0.03 mm for the PVR element and 0.3 mm for the Seas element which is less than 1/20 of the maximum linear excursion for both elements. The measurement did either not require high sound level so the total excursion was assumed to be linear and not affected. After having tested different variants of extra masses, a metal ring was used with diameter approximate the same as the dust cap to avoid causing deformations to the diaphragm and drive the force directly to the voice coil. The metal ring was carefully weighed and several measurements were carried out by replacing the ring each time in case of misalignment to the centre.

3.2 Measurement of Sound Absorption Performance in a Waveguide

3.2.1 Semi-Active System

3.2.1.1 Setup

The setup is illustrated in Figure 3.2. It shows the waveguide tube used with a sound source in one end and the loudspeaker element of interest in the other.

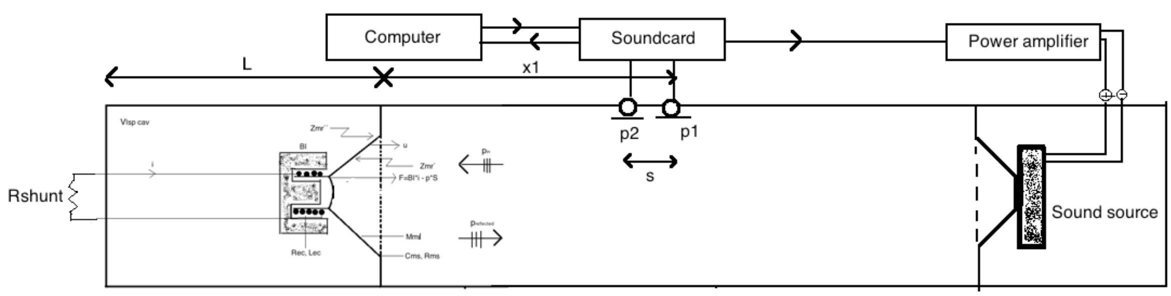

Figure 3.2 Setup for the semi-active system

The loudspeaker element is placed in a rectangular plate of plywood with a smooth surface coating and a rubber list along the perimeter not to let any air slip through. The plate is 20 mm thick, quadratic and has side lengths of 195 mm, while the loudspeaker element is circular and has an outer diameter of 166 mm, see Figure 3.3.

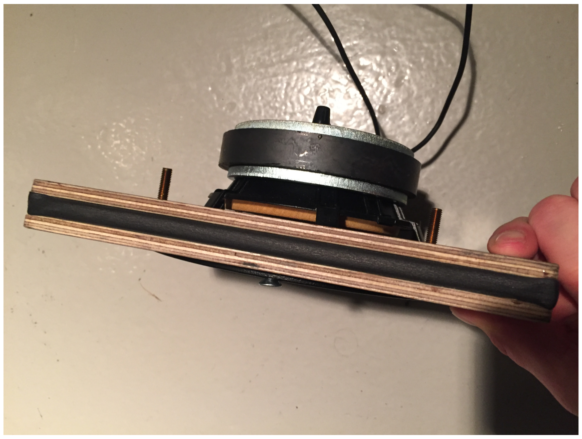

Figure 3.3 Loudspeaker element mounted in plate

The cables connected to the loudspeaker element for connecting different shunt resistances was drawn out of a small hole in the tube at the rear side of the loudspeaker element and sealed with putty. A variable resistance was used to add resistance, R_{shunt}, controlled by a multimeter. 0 ohm was measured by short-circuiting the wires to the element, and infinite resistance was measured by having no circuit. The cavity behind the element was chosen to a length of 150 mm which theoretically estimated the loudspeaker resonance frequency to 162 Hz.

From eq. (2.10) the first transversal mode in the tube is estimated to 860 Hz and the distance between the microphones used in the measurements is 150 mm, which means that the frequency is 1143 Hz when the microphones are $\lambda/2$ apart. Since we are aiming for the lower frequencies, the tube should give reasonable results.

3.2.1.2 Equipment

The equipment used for the measurement is listed in Table 2.

Table 2 Equipment used to measure the absorption performance in a waveguide

Device	*Manufacturer*	*Model*
Software	Morset Sound Development	WinMLS 2004
Computer	HP	Nc6400
Soundcard	AXYS	D-audio
LSP Amplifier	QUAD	50E
LSP element	PRV	6MB200
Microphone 1	BSWA	MPA201
Microphone 2	BSWA	MPA201
Impedance tube	Sintef/NTNU	N/A
Software	MatWorks	MATLAB R2014a

3.2.1.3 Method

Matlab and WinMLS were used to calculate the impedance and absorption factor using the transfer-function method described in (ISO, 1998). A sine sweep was used. Before start, it was checked that the signal to noise ratio was sufficiently high without overloading the loudspeaker or the amplifier.

3.2.2 Active System

3.2.2.1 Setup

The setup for the active system is illustrated in Figure 3.4. It is here added a feedback system using sound pressure, p, and velocity, u, to give feedback signals to the amplifier which will affect the loudspeaker element as explained in the theory chapter 2.4.2.

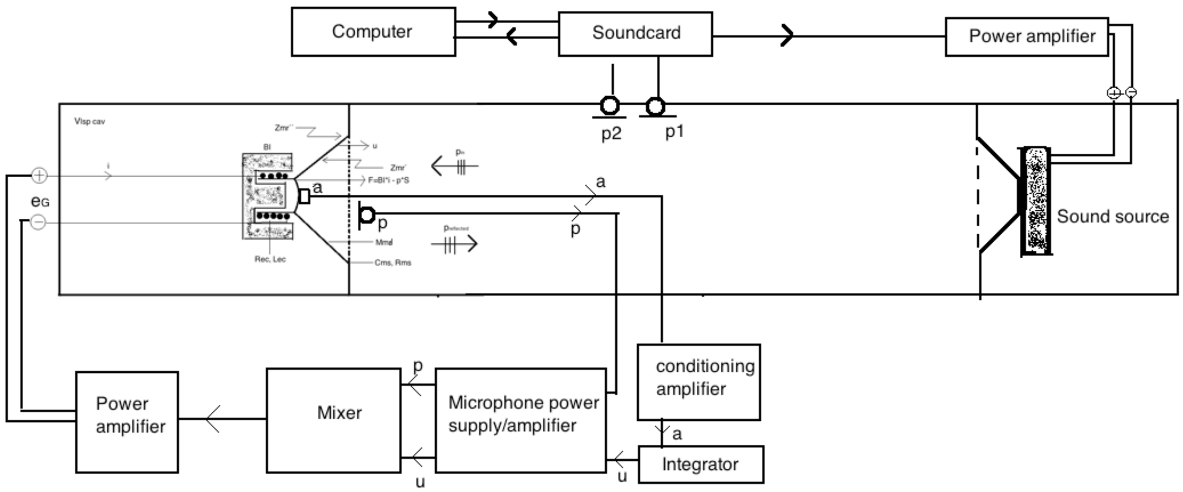

Figure 3.4 Setup for the active system

It is used a microphone for the pressure and an accelerometer for the velocity using an integrator. The accelerometer is attached at the centre of the loudspeaker element (at the dust cap) and the microphone a few centimetres in front as seen in Figure 3.5. The gain and the polarity of the feedback signals is adjusted by the mixer and mixed to a single channel output signal.

Figure 3.5 Picture of the loudspeaker element with accelerometer and microphone

3.2.2.2 Equipment

The equipment used for the feedback system are listed in Table 3. The equipment comes in addition to the equipment in Table 2. It was first used a Microflown UR-900843 probe that measured both pressure and velocity, but the sensitivity and phase were varying with the frequency which in this case made it useless.

Table 3 Additional equipment used for the active system

Device	*Manufacturer*	*Model*
Microphone	Bruel & Kjær	4190
Microphone preamplifier	Norsonic	1201
Microphone power supply/amplifier	Norsonic	336
Mixer	Yamaha	1608M
Oscilloscope	Rohde & Schwarz	RTE1022
Accelerometer	Bruel & Kjær	4393
Conditioning amplifier	Bruel & Kjær	2626
Integrator	Bruel & Kjær	ZR0020
Integrator ampflifier	Norsonic	1201
Power ampflifier	Rotel	RB-1552MKII

3.2.2.3 Method

The same method as for the semi-active system was used to measure the impedance and the absorption factor. The gains to the sound pressure feedback and the velocity feedback were adjusted stepwise with one measurement for each change. The polarities for the two input signals were also inverted. All the adjustments were adjusted by the mixer. The conditioning amplifier and the microphone power supply/amplifier was set to a gain that resulted in high signal to noise ratio and possibilities to adjust everything from the mixer. An oscilloscope was connected in parallel to the input of the mixer (not illustrated) to watch the changes and make it easier to tune the system right.

The total gain in each feedback chain were calculated by measuring the amplification using the oscilloscope and comparing with the specification from the manufacturer. The datasheet for the integrator was not found, but it was measured to amplify the signal by a factor of 27. The mixer was also just measured manually for each amplification step. The exact amplification was difficult to measure due to high sensitivity of the sensors and low amplification by the mixer.

3.3 Measurement of Sound Absorption Performance in a Room

3.3.1 Setup

The loudspeaker element was mounted in a closed box with a volume V_{box}=6135 cm^3, almost the same volume as the cavity behind the element in the tube (6000 cm^3). The loudspeaker element in box can be seen to the left in Figure 3.6. Two sound sources were used, one in each diagonal of the room, that is one in the ceiling and one on the floor. As it can be seen in the pictures, the room has flat hard concrete walls. The volume of the room is 120.7 m^3. It is approximately 6x5x4 m (l,w,h), but has a raised platform in the ceiling.

Figure 3.6 Left: The measured absorber. Right: One of the two external sound sources

A person is measured to have an effective absorption area from 0.1 to 0.2 m^2 between 125 Hz and 250 Hz (Ballou, 2015). Since the measured absorber is small and the contribution should be as large as possible, all the equipment and the person were placed outside the room when the measurements was ongoing. The door had a 1 cm opening to let the cables pass through. The opening was considered to have smaller impact than a person standing inside.

3.3.2 Equipment

Table 4 Equipment used to measure the absorption performance in a room

Device	*Manufacturer*	*Model*
LSP Amplifier	Fox	30
LSP element	PRV	6MB200
Soundcard	Roland	UA-1610
Microphone	Bruel & Kjær	4943
Microphone preamplifier	Norsonic	1201
Microphone power supply/amplifier	Norsonic	336
Computer	HP	Elitebook 850
Software	AFMG Technologies	EASERA
Software	MatWorks	MATLAB R2014a

3.3.2.1 Method

By doing several measurements with different microphone positions the reverberation time was measured as described in chapter 2.2. The following three configurations were measured:

1. The room without the absorber
2. The room with the absorber
3. The room with the absorber and using the absorbing loudspeaker as the source

For the two first, both of the two external source loudspeakers were used, one at the time.
A total of 60 measurements was carried out, 20 for each configuration. To avoid overload and not to cause non-linearities to the measured loudspeaker, it was assured that the source level was not set too high. That made the signal to noise ratio too low to use T60, so the more commonly used T30 value was used instead. A sine sweep was used as signal.
The absorbing loudspeaker was short-circuited when not used as sound source (configuration 1 and 2). As explained in chapter 2.4.2 this should be the most comparable way. The absorbing loudspeaker was placed in one corner of the room since the pressure is highest in the corners, and the pressure is the driving factor for the element to achieve high absorption. The two external loudspeakers were not moved during the measurements.

4 RESULTS

4.1 Loudspeaker Element Parameters

The measured parameters of the two loudspeaker elements are compared with the datasheets given by the producers in the table below (PRV, 2015), (Seas, 2003).

Table 5 The loudspeaker element parameters

Parameters	***PRV***		***SEAS***	
	Datasheet	Measured	Datasheet	Measured
f_s [Hz]	123.3	122.3	35	127.1
Q_{ms}	24.36	25.79	1.70	1.33
Q_{es}	1.32	1.46	0.29	0.86
Q_{ts}	1.25	1.38	0.25	0.52
M_{ms} [g]	11.97	11.5	14.5	11.5
M_{md} [g]	10.89	10.4	13.5	10.8
C_{ms} [mm/N]	0.139	0.148	1.4	0.136
Bl [Tm]	7.15	6.48	8.5	7.97
R_{ms} [Ns/m]	0.38	0.34	2.0	6.91
R_{ec} [Ohm]	7.3	6.95	6.1	5.95

The parameters are given for free air condition in accordance with the producers' specification. It can be seen that the measured parameters for the PRV element suites the datasheet very well and the differences are small. For the Seas element, this is definitely not the case. The compliance is a factor of 10 different from the datasheet and has a great impact on the resonance frequency, f_s, and the R_{ms} value, and smaller impact on the Q-values. The Seas loudspeaker element was therefore considered as inappropriate. Further investigation showed that the rubber surround was very sensible to temperature. This is interesting since the absorbing capabilities is highly depending on the correct parameters. The anechoic room was cold when the measurements took place, with a temperature below 10 degrees Celsius. A new measurement in 20 degrees, performed by Tim Cato Netland, resulted in a resonance frequency of 78 Hz. This means the rubber was now more flexible and thus the compliance was higher. It was checked another time, measuring f_s =120 Hz after having cooled the element, and 78 Hz again after warming it up. A last measurement was performed after having excited the loudspeaker element to high excursion at 15 Hz over a night, which gave a f_s right below 60 Hz. The Seas loudspeaker element is produced in 1995. It is natural that the rubber become stiffer by the years, but this was more than expected. An article by Kippel GmbH (Kippel, 2014) is telling that changing the temperature by 50 degrees might cause as much as 100% shift in resonance frequency of a loudspeaker element. Before making and tuning a loudspeaker element to an expensive absorber this should be taken into account. Using a loudspeaker element with foam surround or rubber that keeps its properties for different temperatures and years is recommended.

Since the anechoic room used for the measurements isn't perfectly absorbing below 100 Hz, deviations of the free air impedance at the lowest frequencies might also occur. If the Seas element was measured to have a resonance frequency of 35 Hz as in the datasheet this could have affected the parameters.

4.2 Sound Absorption Performance in a Waveguide

4.2.1 Loudspeaker Element as a Semi-Active Absorber

Figure 4.1 shows the theoretical calculated absorption factors for different shunt resistances connected to the loudspeaker, while Figure 4.2 shows the measured result.

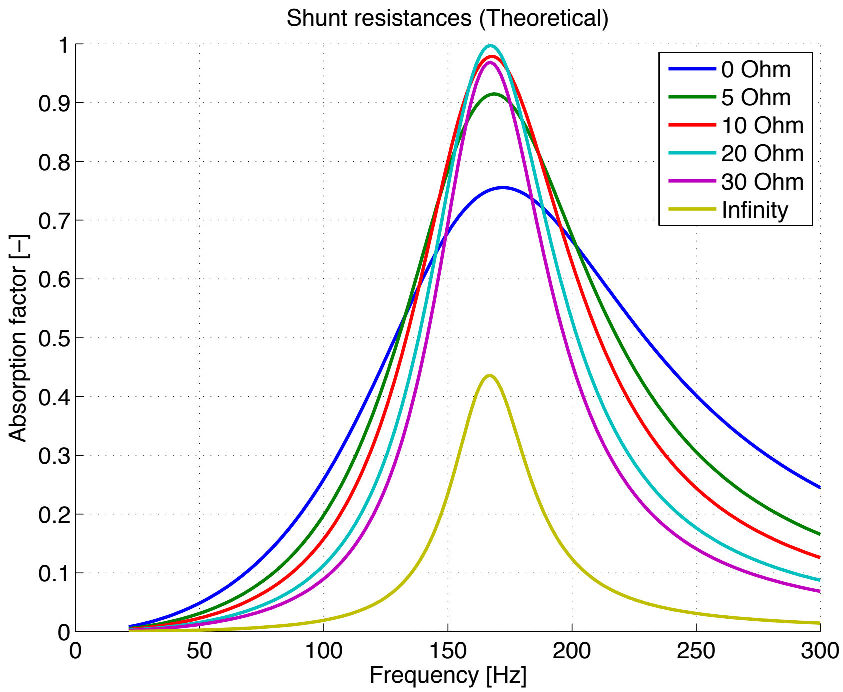

Figure 4.1 Calculated absorption curves for different shunt resistances

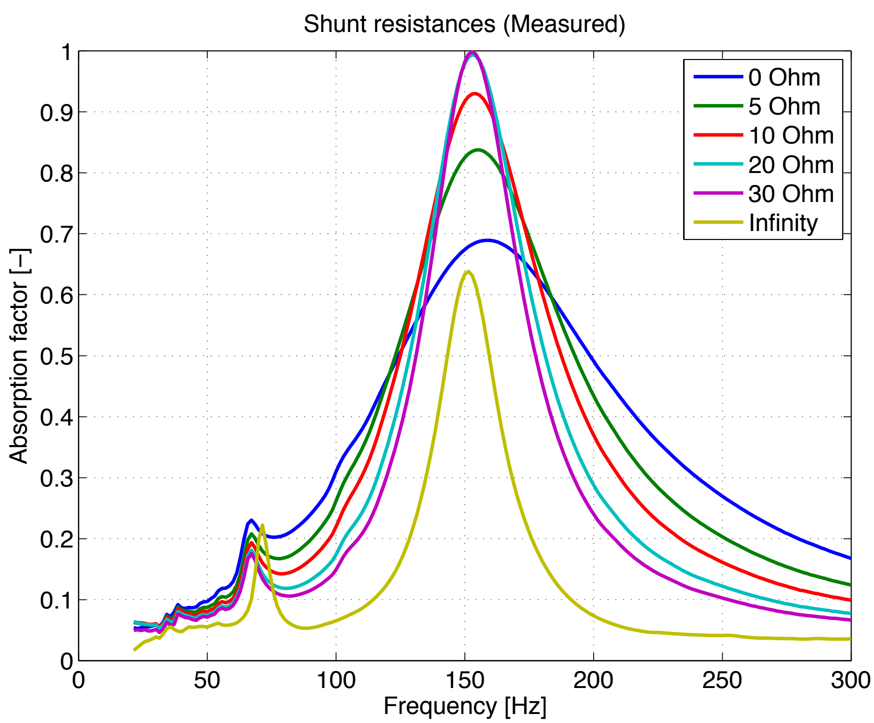

Figure 4.2 Measured absorption curves for different shunt resistances

Apart from a tiny frequency shift, wider curves and larger influence of the resistance value, it can be seen that the calculated curves correspond fairly well with the measured curves. The calculated optimal shunt value is according to eq. (2.22) calculated to be 13 Ohm using the measured loudspeaker parameters and 17 Ohm using the producer specifications. From the measurements, it looks like 17 Ohm is the best suited value. The resonance frequency is calculated to be 162 Hz, while the measured results show a resonance frequency that it is a bit lower. The deviations may be caused by wrong measured parameters, or expressions that do not coincide perfectly well with the setup.

4.2.2 Loudspeaker Element as an Active Absorber

By adjusting the feedback gain from the microphone called Γ_p, and the feedback gain from the accelerometer integrated to velocity, Γ_u, the following results were achieved. Figure 4.3 and Figure 4.4 is showing the average absorption factor and maximum absorption factor over a frequency range from 50 to 300 Hz for different gain settings. The different curves represent different Γ_p values while Γ_u is on the horizontal axis. Positive gain is defined as helping the diaphragm oscillate, while negative gain is doing the opposite. The change of polarity is what divide them. Negative Γ_p will increase the pressure in front of the diaphragm until the element is acting like a hard wall, but that case is not interesting and will not be displayed in the figures. Instability occurs when Γ_p>26 mV/Pa and Γ_u>86 Vs/m or Γ_u<-140 Vs/m when using the PRV loudspeaker element. Certain reservations must be taken to the values of the feedback gains as the amplification by both the mixer and the integrator was difficult to measure exact.

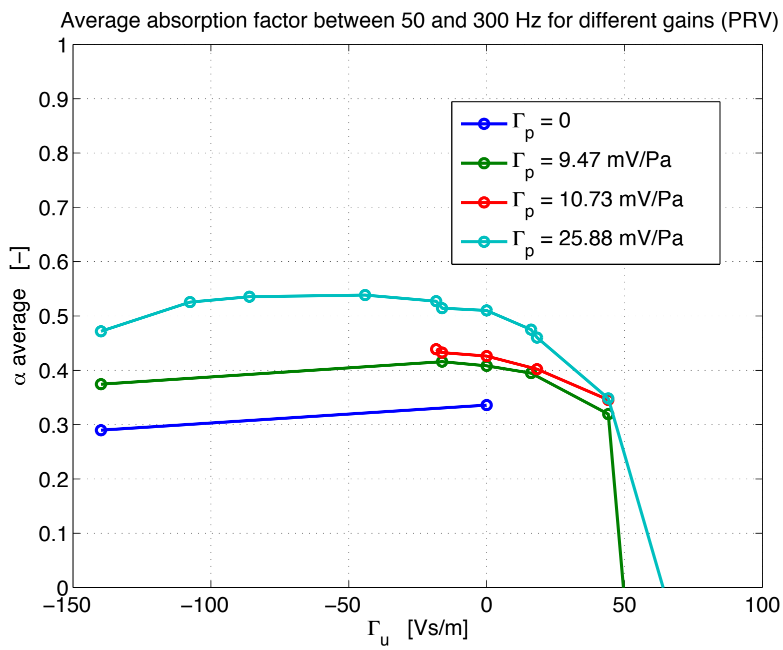

Figure 4.3 Average absorption factor between 50 and 300 Hz for different gain settings using the PRV loudspeaker element

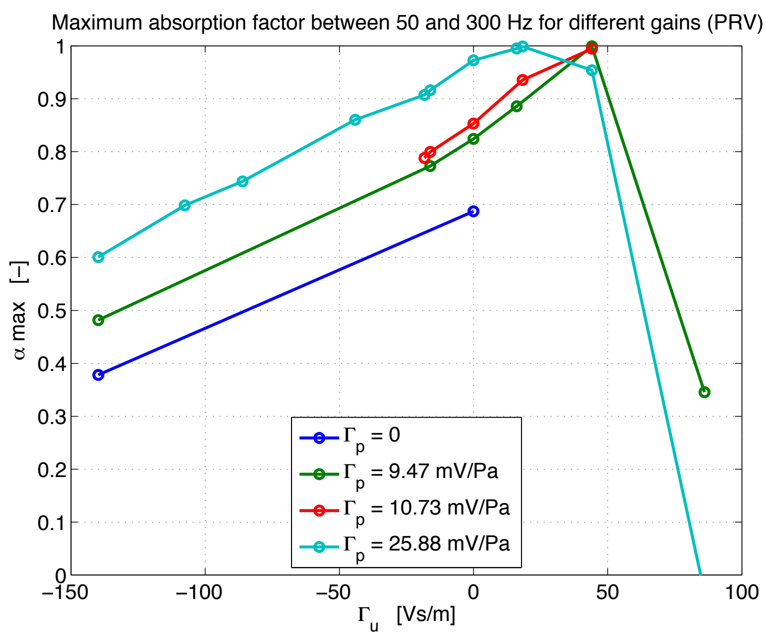

Figure 4.4 Maximum absorption factor between 50 and 300 Hz for different gain settings using the PRV loudspeaker element

It can be seen from the figures that the highest peaks of absorption are given for positive Γ_u values, while the average absorption over the frequency range is highest by negative Γ_u values. In both situations, high Γ_p is favourable. The following is observed by using the oscilloscope simultaneously as adjusting the gains:

At resonance (u and p is in phase):

- $\Gamma_u\uparrow \Rightarrow u\uparrow$ and $p\downarrow$ until $p=0$
- $\Gamma_u\downarrow \Rightarrow u\downarrow$ and $p\uparrow$
- $\Gamma_p\uparrow \Rightarrow u\uparrow$ and $p\downarrow$
- $\Gamma_p\downarrow \Rightarrow u\downarrow$ and $p\uparrow$ until $u=0$

Above resonance (u and p is not in phase):

- $\Gamma_u\uparrow \Rightarrow u\uparrow$ and $p\approx c$, u-phase shifts away from p-phase
- $\Gamma_u\downarrow \Rightarrow u\downarrow$ and $p\approx c$, u-phase shifts toward p-phase
- $\Gamma_p\uparrow \Rightarrow u\uparrow$ and $p\approx c$
- $\Gamma_p\downarrow \Rightarrow u\downarrow$ and $p\approx c$ until $u=0$

Below resonance (u and p is not in phase):

- $\Gamma_u\uparrow \Rightarrow u\approx c$ and $p\approx c$, u-phase shifts away from p-phase
- $\Gamma_u\downarrow \Rightarrow u\approx c$ and $p\approx c$, u-phase shifts toward p-phase
- $\Gamma_p\uparrow \Rightarrow u\uparrow$ and $p\approx c$
- $\Gamma_p\downarrow \Rightarrow u\downarrow$ and $p\approx c$ until $u=0$

Where p is the measured pressure in front of the loudspeaker element, u is the velocity at the diaphragm and c is a constant.

As described in chapter 2.4.2, the impedances must match to achieve high absorption. For a given sound pressure meeting the loudspeaker element, the particle velocity of the diaphragm should act as similar as possible as the air particles elsewhere in the tube. From the definition of the specific acoustic impedance in eq. (2.23), it can be seen that when the impedance is too high, the velocity will be too low for a given sound pressure coming in, to match the impedance of the air in the tube. There are two ways to solve that, increase the velocity or decrease the sound pressure. $\Gamma_p\uparrow$ is doing both and as seen in the figures, the effect on the absorption seems to agree. It is though more difficult to see the total effect of the velocity gain since the increase in u is counteracting with the phase-match. It seems beneficial to first achieve better phase-match by decreasing Γ_u, which is applying negative Γ_u values as seen in Figure 4.3, and secondly increase Γ_p, to the real value of the impedance is right. Unfortunately, instability occurs when increasing Γ_p more than 25.88 mV/Pa so the Γ_u must be of positive value to achieve absorption factor close to 1 for the loudspeaker used.

Even though the Seas loudspeaker element was measured to have uncertain parameters, it was performed some measurements using that element too. The results are shown in Figure 4.5 and Figure 4.6.

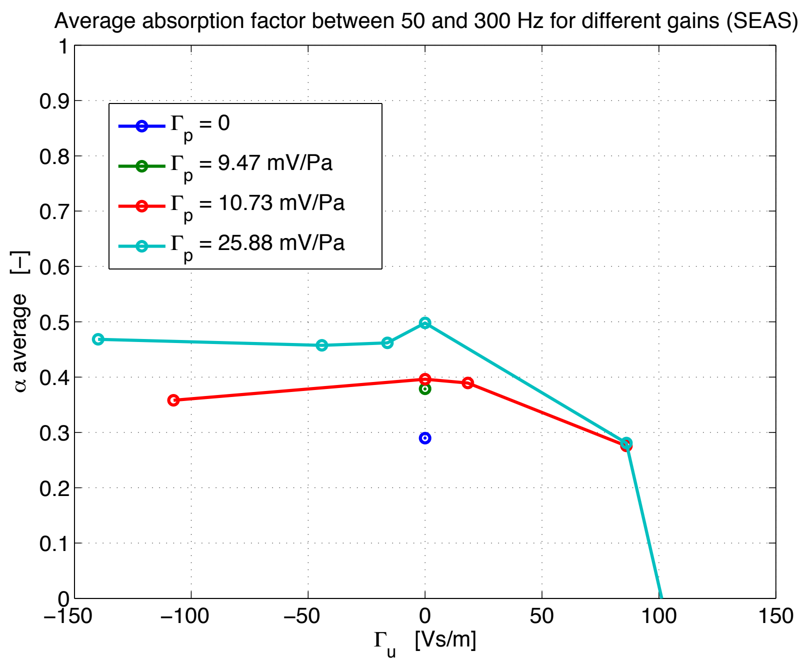

Figure 4.5 Average absorption factor between 50 and 300 Hz for different gain settings using the Seas loudspeaker element

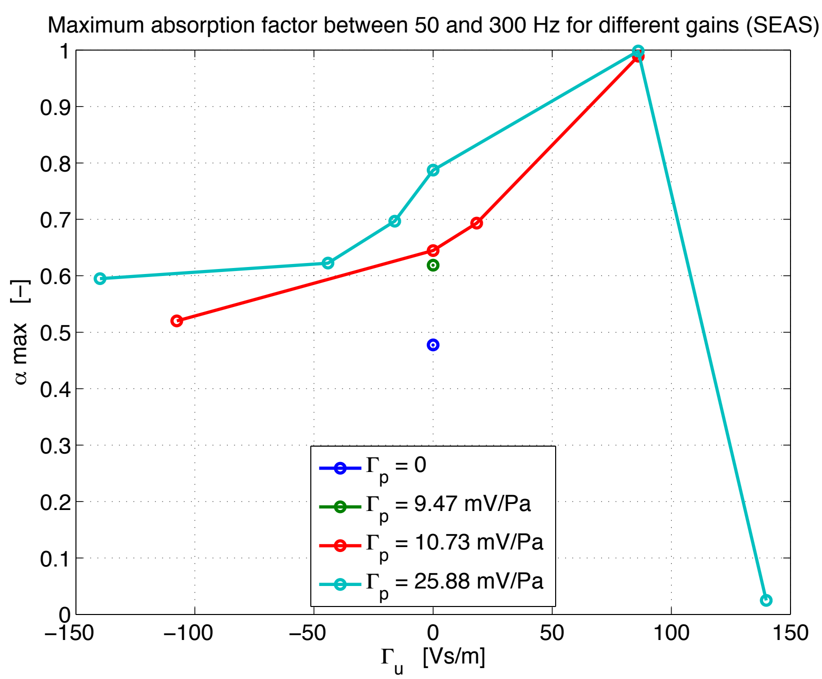

Figure 4.6 Maximum absorption factor between 50 and 300 Hz for different gain settings using the Seas loudspeaker element

The effects are the same as with the PRV element, but the Seas element needs even higher Γ_u to achieve absorption factor close to 1. That might indicate another loudspeaker element with better parameters than the PRV element could give better results. Figure 4.7, taken from an unpublished study by the same author, is showing the measured result of the Seas loudspeaker element with different shunt resistances connected. Even though the parameters of the Seas loudspeaker element are a bit uncertain, the measured results are assumed to be true, and confirm the observations.

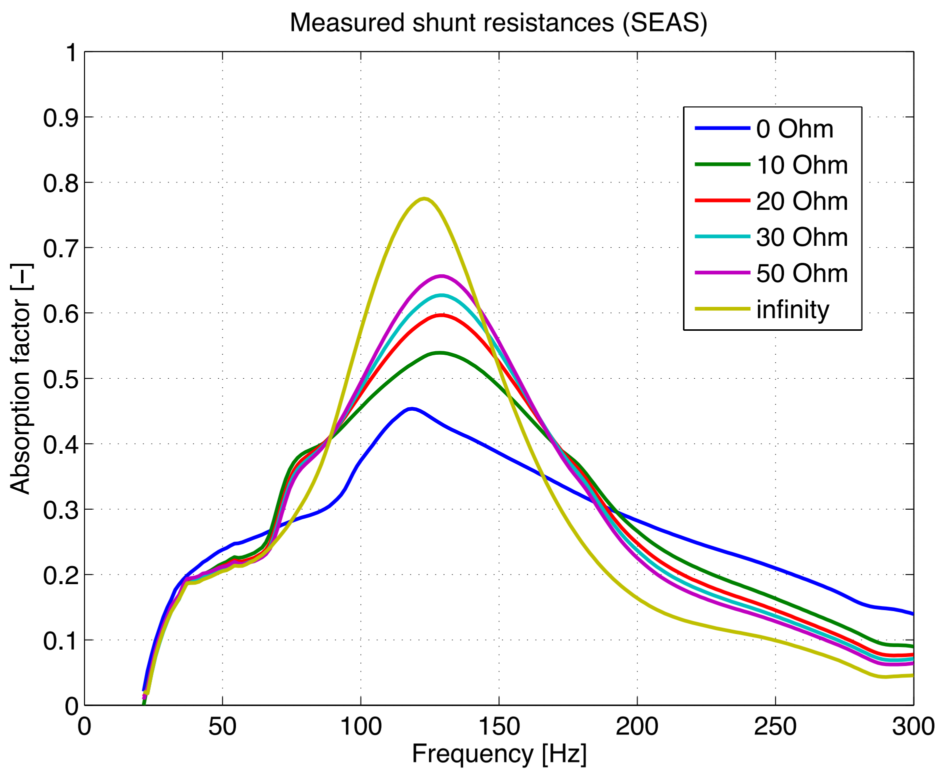

Figure 4.7 Absorption factor for different shunt resistances for the Seas loudspeaker element (Lilleeng, 2016)

The figure shows that the optimal shunt for the Seas element is of much higher value than for the PRV element. Using eq. (2.22) the optimal shunt value is calculated to be -18 Ohm with the measured parameters and -149 Ohm using the ones from the datasheet. That means a shunt value of infinity will make the total impedance of the loudspeaker element at resonance closest to the air impedance in the tube (see eq. (2.21)), unless negative resistances are implemented (Bobber, 1970).

Figure 4.8 is showing the absorption factor as function of frequency for some of the combinations that led to the best results. Γ_p is at maximum before instability in every case. The figure illustrates the effects discussed earlier in this chapter. The green curve has high absorption peak and is only subjected to Γ_p. Adding negative Γ_u to the feedback chain, the curve is widening, but the peak is lowered as seen in the blue curve. The same happens for the Seas element (light blue to red curve). The red curve is especially interesting, because the resonance peak is disappeared and the absorption factor is almost flat over a wide frequency range. It looks like the mechanical properties of the loudspeaker element are cancelled.

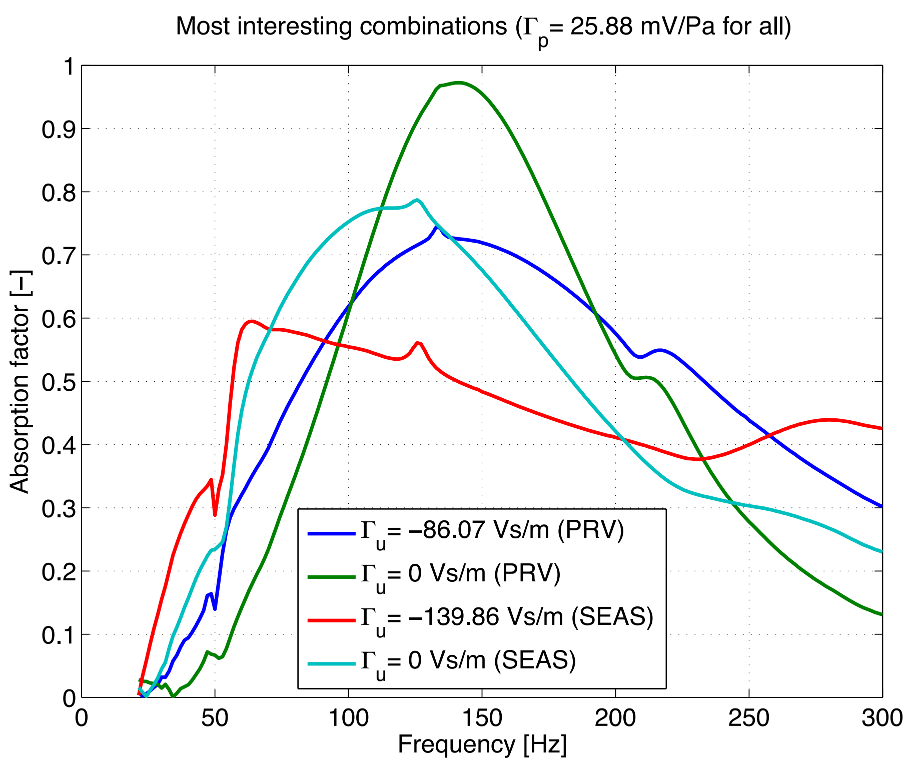

Figure 4.8 Absorption factor as function of frequency for some of the most interesting combinations

Another aspect that needs considering when choosing the loudspeaker element for an absorber is the Q-value. As known and seen from eq. (2.26), a high Q-value gives a narrow resonance peak, while a low Q-value gives a smoother peak. The Seas element has a much lower Q-value than the PRV element and might be one reason why the red curve is that flat. From eq. (2.28) we have

$$Q_{ms} * R_{ms} = \omega M_{ms} = M_{ms} * \sqrt{\frac{1}{C_{ms} * M_{ms}}} = \sqrt{\frac{M_{ms}}{C_{ms}}} \qquad (4.1)$$

When wanting low Q_{ms} and low R_{ms}, an element with low mass and low stiffness ($1/C_{ms}$) is preferable. Lissek had in his setup a loudspeaker element with an optimal shunt value lower than 5 Ohm (Lissek et al., 2011), and also a mechanical quality factor as low as 3.88 (Visation, 2015). From eq. (2.22) it can also be seen that high DC-resistance and low force factor may give even more potential to achieve higher absorption area at resonance frequency where the loudspeaker is small compared to the cross-section of air in front of it. The equation is valid for plane waves, but although the target impedance for a room will be more complicated, the effect is assumed to be valid for rooms as well.

4.3 Sound Absorption Performance in Room

The reverberation time with and without the semi-active loudspeaker in the reverberant room is displayed in Figure 4.9.

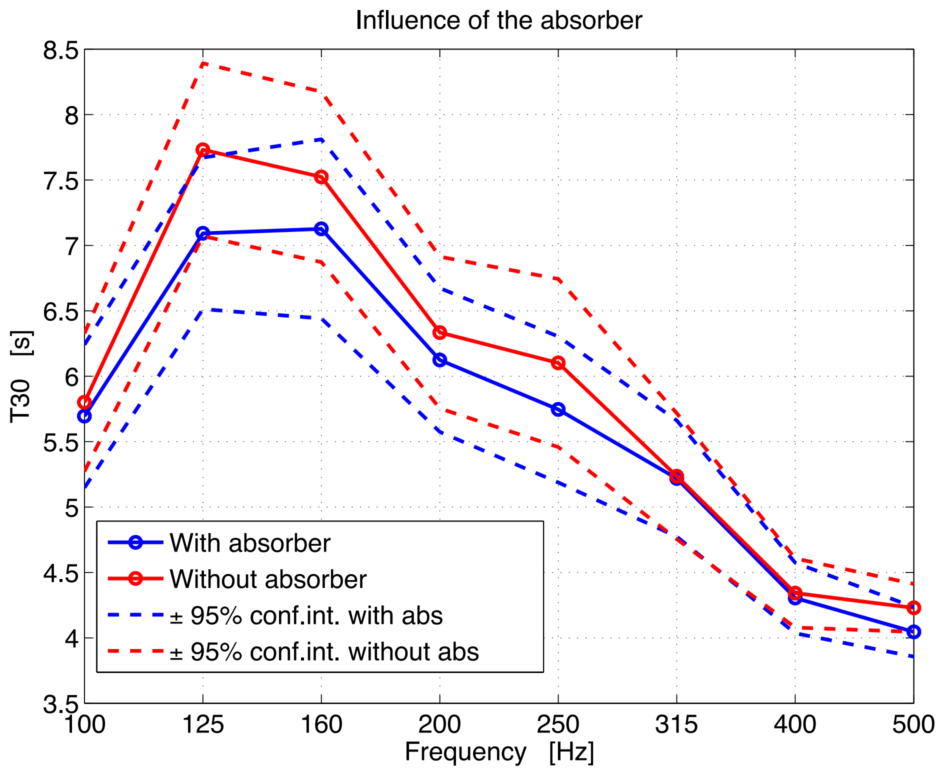

Figure 4.9 Influence of the absorber

It can be seen that the reverberation time is lowered with the loudspeaker in the room, but the confidence intervals overlap. Since the measurement is comparable, the difference between the measurements may nevertheless be significant. The normalized difference between the reverberation time in the room with and without the absorber is calculated and according to the paired sample t-test, significance is achieved (StatisticsSolutions, 2017). The results are shown in Figure 4.10.

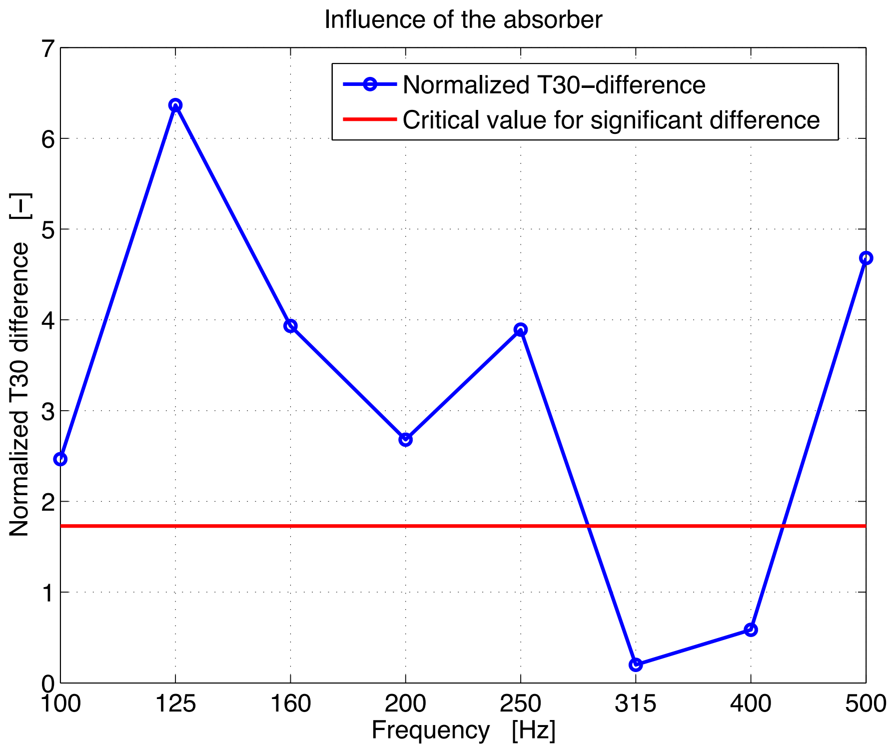

Figure 4.10 Significant influence of the absorber

The difference is significant below 280 Hz and it can be seen from both figures that the reverberation time is lowered in that frequency range. By theory, the resonance frequency of the loudspeaker in the box is calculated to be around 160 Hz. It is unknown why the largest difference is for the 125 Hz band. The resonance frequency in the tube was also a bit lower than calculated, but the difference here is bigger. The radiation impedance that is changing due to reflections is suggested as the reason for the deviation.

The effective absorption area is calculated from the reverberation times and displayed in Figure 4.11.

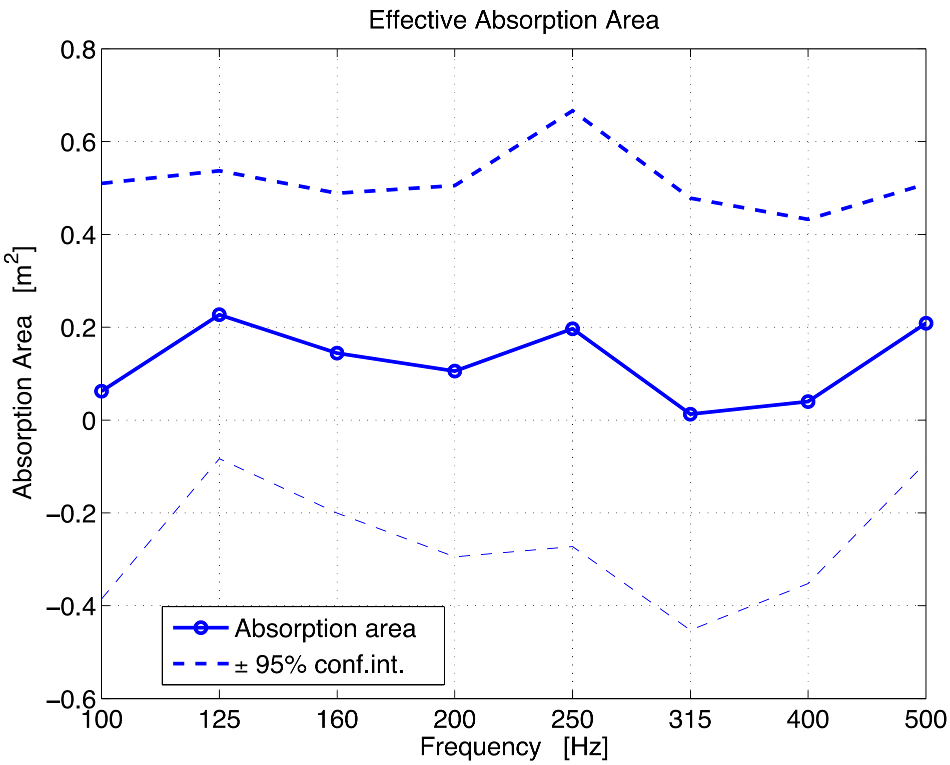

Figure 4.11 Effective absorption area

The absorption area is around 0.2 m^2 which is about 13 times the effective area of the loudspeaker element. In comparison, the AVAA from PSI Audio has a ratio between 5 and 20 (PSI Audio, 2015a). A larger loudspeaker element would as stated earlier have larger influence and bigger absorption area. The deviations of the reverberation measurement are causing a wide confidence interval, but since the difference in reverberation times is significant, we can also say that the absorption is significant. The large deviations between the measurement positions is probably caused by the room used for the measurements. The Schroeder frequency is for the room above 400 Hz for reverberation time measured between 0 and 350 Hz. The room will therefore for the frequency band of interest have dominating modes and the diffuse field assumption is strictly speaking not valid. Since the effective absorption is calculated by diffuse field assumption, it must be seen more as an indication of the absorption area than the exact value. The raised ceiling and the small opening in the door for the cables might also have an impact on the position-depending reverberation time difference.

Figure 4.12 is showing the influence of using the absorber as the source compared to using the external loudspeaker in the room as the source, and Figure 4.13 the significance using the paired sample t-test.

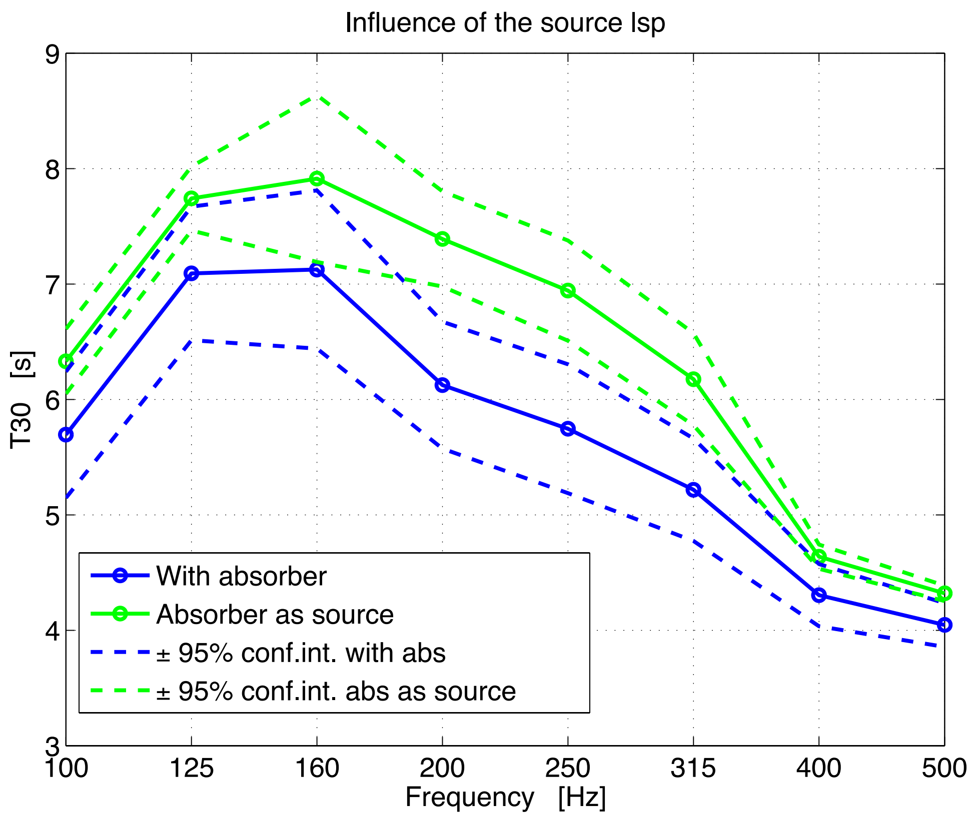

Figure 4.12 Influence of using the loudspeaker as source

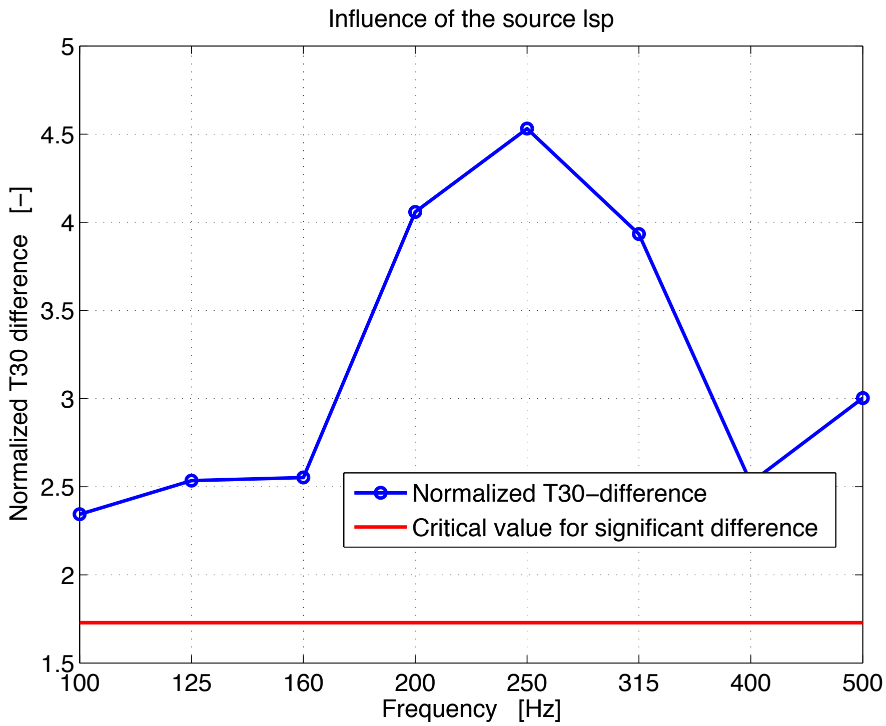

Figure 4.13 Significant influence of using the loudspeaker as source

The result is showing that using the absorber as source will increase the reverberation time compared with using the external loudspeaker as sound source. Measurement from the unpublished project thesis of the same topic (Lilleeng, 2016) is showing the Seas loudspeaker element in the tube playing music while the absorption was measured (Figure 4.14).

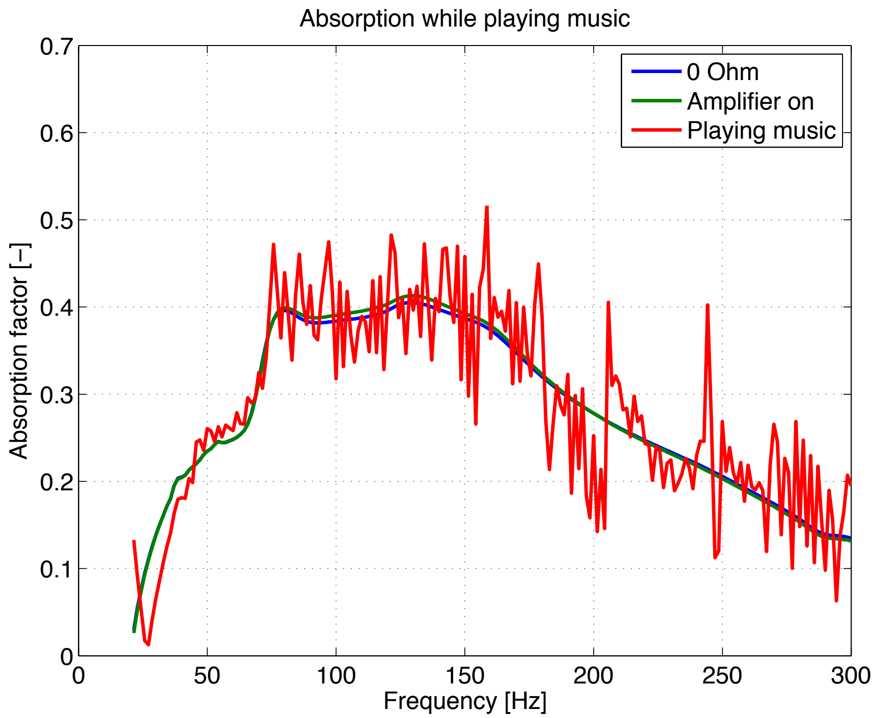

Figure 4.14 Loudspeaker playing music while measured (Lilleeng, 2016)

It can be seen from that measurement that the music signals only act as noise to the absorption capability and it was therefore naturally to expect the same when using the loudspeaker as source in Figure 4.12. Why the absorber as source increased the reverberation time significantly is not known and needs further investigation. It should be of great interest for all acousticians that brings a loudspeaker to measure the room properties.

A suggested explanation is that the sweep signal used was so long that the reflecting waves from the walls hit the loudspeaker element used as source before the frequency had changed too much. The system would be like the active system in the tube where the loudspeaker is subjected to both incoming sound pressure and input voltage from the amplifier. If the input voltage from the amplifier is corresponding well with the incoming sound wave, the absorption is higher and when it is mismatch, positive interference with the incoming sound wave can occur. The same thing will happen for the external loudspeaker when that is used as a source. If the resonance frequencies are different for the loudspeakers, difference in reverberation time might occur. Different parameters will also induce different effects of the input voltage regarding the absorption, as seen from adjusting Γ_p and Γ_u. The difference in surface area of the loudspeakers is of a factor 3.5 where the external loudspeaker is the biggest. That may cause differences in effective absorption area and hence the reverberation time.

Another factor that could have had influence is the difference in sound level. The amplifier and the soundcard was set to the same output level for both using the absorber as source and the external loudspeaker. Different sensitivities of the loudspeakers result in different sound levels. Although the signal to noise ratio seemed good during the measurements, it could have been too low when the absorbing loudspeaker was used as the source. If the sound level from the absorbing loudspeaker was too low, the Schroeder curve might have been flatter, and hence the reverberation time increased.

5 CONCLUSION AND FURTHER WORK

5.1 Conclusion

The loudspeaker parameters are subject to change by temperature and aging.

Bringing an ordinary loudspeaker to a room will have an influence of the acoustical properties of the room. It does seem like the loudspeaker absorption is changing by using it as the primary sound source for a sweep-measurement. Further investigation on that behaviour is needed.

The active system, which in Lissek's article (Lissek et al., 2011) looked easy and robust, was in this thesis found to have more limitations than expected. Firstly, it seems like the loudspeaker element cannot be randomly chosen, but needs to have the right parameters to function well. Secondly, the additional equipment needs to have a flat sensitivity and not affect the phase in the frequency range of interest.

Although electronic devices have become cheaper over the years, an active system like this will still be expensive. To be of interest it must be better than a regular Helmholtz absorber. From the results presented earlier in this thesis, it is not, but it seems like it has potential to be much better. If the surface area of the loudspeaker element had filled the tube perfectly, the relative absorption area would have been bigger. In addition, the optimal shunt value would have been close to zero, which as described earlier, probably could give more room for compensating the mechanical properties.

Although the system has its limitations, if a more suited loudspeaker element could increase the performance, the flexibility of the system and the possibility to go very low in frequency are giving advantages in comparison to the existing products and traditional Helmholtz absorbers.

5.2 Further Work

It is recommended to do further investigations of the loudspeaker's effect on a room, especially the difference by using it as a source and not. As the reverberation time of the room is shown to be highly varying for different microphone positions, it is recommended to use a larger loudspeaker to achieve better significance. High signal to noise ratio must be carefully checked for both cases. The effect of different source signals may also be interesting to measure.

For the active system, a more suited loudspeaker element is needed to conclude whether it has potential or not. That is, an element with low mass and low stiffness. High DC-resistance and low force factor may also give the loudspeaker higher potential. Using a laser velocimeter may reduce the sources of error due to fever components.

Electronic feedback controllers with phase compensation may also be the way to go.

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7 APPENDIX

Appendix contain Matlab scripts used for the calculations. The scripts that only contains plots are not included.

7.1 Finding the Loudspeaker Parameters

Thiele Small Measurement

Loading files

```
[freq, impM1] = textread('imp_prv_freeair_metalgitter.txt');
[freq2, impM2] = textread('imp_prv_freeair_metalgitter_ring.txt');
```

Input Parameters

```
Sd=154e-4;                                                    %Cone area (from
datasheet)
a=sqrt(Sd/pi);                                              %Equivalent radius
Rec=6.95;                                                     %Electrical DC
resistance [Ohm]
addedmass=9.8347e-3;                                          %Added mass [kg]
ro=1.21;                                                       %air density
[kg/m^3]
```

Calculating parameters

```
iv=find(freq>20 & freq<300);
[Rres,I]=max(impM1(iv));
fs=freq(I);
[Rres_addedmass,I2]=max(abs(impM2(iv)));
fs_addedmass=freq(I2);

Mms=addedmass/((fs/fs_addedmass)^2-1)   %Mms is the mass + air load in free air
Mmd=Mms-(8*a^3*ro/3)                   %if measured in free air
Cms=1/((2*pi*fs)^2*Mms)

iv1=find(freq>20 & freq<fs);
iv2=find(freq>fs & freq<300);
R12=sqrt(Rec*Rres);
[difference, index_At_impM1_Equals_R12] = min(abs(impM1(iv1)-R12));
f1 = freq(index_At_impM1_Equals_R12);
```

```
[difference, index_At_impM1_Equals_R12] = min(abs(impM1(iv2)-R12));
f2 = freq(index_At_impM1_Equals_R12+I);

Qms=(fs/(f2-f1))*sqrt(Rres/Rec)
Res=Rres-Rec;
Qes=Qms*Rec/Res
Qts=Qms*Rec/Rres
Bl= sqrt(Rec/(2*pi*fs*Cms*Qes))
Rms=2*pi*fs*Mms/Qms
```

From free air to infinite baffle parameters

```
Mmib=Mmd+(16*a^3*ro/3);
Mmfa=Mmd+(8*a^3*ro/3);
%k=sqrt(Mmib/Mmfa);        %Correction factor

%fs=fs/k
%Cms=1/((2*pi*fs)^2*Mmib);
%Qms=k*Qms;
%Qes=k*Qes;
%Bl= sqrt(Rec/(2*pi*fs*Cms*Qes));
```

Plot of electrical impedance

```
plot(freq(iv), impM1(iv), freq(iv), impM2(iv))
legend('ordinary','added mass')
title ('Electrical Impedance PRV element')
xlabel('Frequency [Hz]')
ylabel ('Impedance [Ohm]')
grid
```

Figure 7.1 and Figure 7.2 is showing the electrical impedance for the two loudspeaker elements.

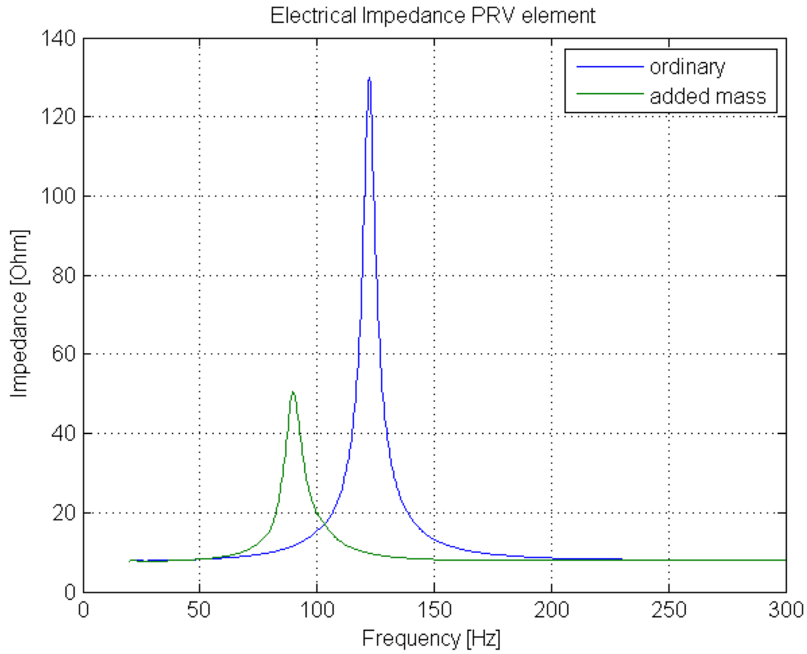

Figure 7.1 Electrical impedance for PRV element

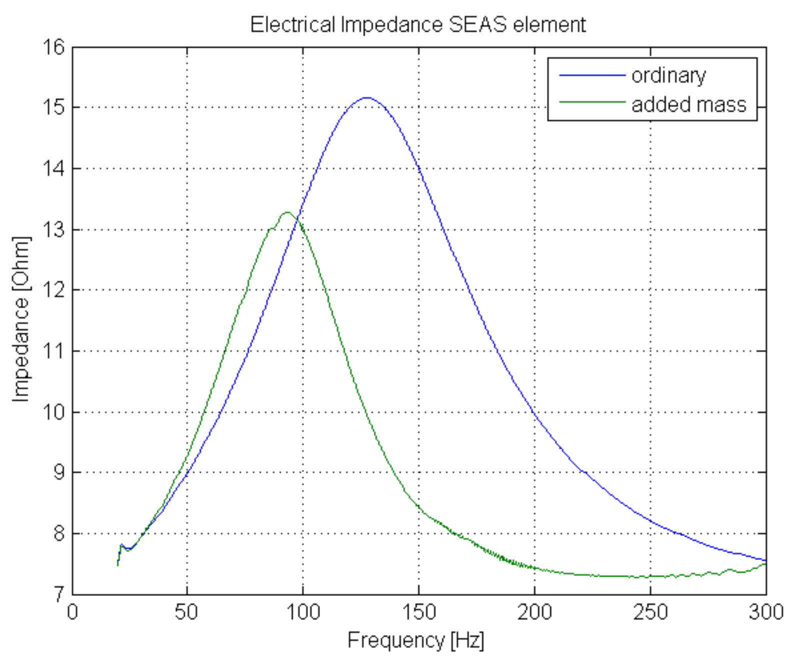

Figure 7.2 Electrical impedance of the Seas element

7.2 Sound Absorption Performance in a Waveguide

7.2.1 Absorption for Different Shunt Resistances

Loading data files

```
load IA781.mat
iv=find(frek>20 & frek<300);
alpha1=abs(alfa(iv));

load IA783.mat
alpha2=abs(alfa(iv));

load IA784.mat
alpha3=abs(alfa(iv));

load IA785.mat
alpha4=abs(alfa(iv));

load IA786.mat
alpha5=abs(alfa(iv));

load IA901.mat
alpha6=abs(alfa(iv));
```

Input data

```
Bl=7.15;                      %Force factor              [N/A]
Rec=7.3;                      %Voice coil resistance      [ohm]
Le=0.72e-3;                   %Voice coil inductance (eq) [H]
rho=1.21;                      %air density                [kg/m^3]
c=343;                       %speed of sound            [m/s]
w=2*pi.*frek;                %frek is frequency vector loaded from input data
Slsp=154*10^-4;              %effective piston area (from datasheet) [m^2]
a=sqrt(Slsp/pi);             %effective radius
Stube=0.2^2;                 %cross section of tube [m^2]
lrear=0.15;                  %length from lsp element to wall (rearside)
Mms=11.97e-3;                 %Moving mass
Mmd=10.89e-3;                   %moving mass without air load
Qms=24.36;
Cms=0.139e-3;                  %suspension compliance             [m/N]
Rms=0.38;                     %suspension mechanical resistance   [Ns/m]
Ca=lrear*Stube/(rho*c^2);     % acoustic cavity
jw=1j.*w;
Zec=Rec+jw*Le;
B_front=0.6;
```

```
B_rear=0.6;
Zmr_rear=Slsp^2./(jw.*Ca)+jw.*(B_rear*rho*Slsp^2/(pi*a));
Zmr_front=jw.*(B_front*rho*Slsp^2/(pi*a));
```

Calculation

```
Rshuntvalues=[0, 5, 10, 20, 30, exp(1000)];
alphamatrix=zeros(length(alfa),length(Rshuntvalues));

for s=1:length(Rshuntvalues)
    Rshunt=Rshuntvalues(s);                     %shunt resistance           [ohm]
    Zmtheo=jw*Mmd+(1./(jw*Cms))+Rms+Zmr_front+Zmr_rear+((Bl^2)./(Zec+Rshunt));
    Zatheo=Zmtheo./(Slsp.^2);
    Zaair=rho*c/Stube;
    Rtheo=(Zatheo-Zaair)./(Zatheo+Zaair);
    alphatheo=1-abs(Rtheo).^2;
    alphamatrix(:,s)=alphatheo(:);

end
```

Plot

```
%Theoretical:
figure(1)
j=plot(frek(iv), alphamatrix(iv,1), frek(iv), alphamatrix(iv,2), frek(iv), alphamatrix(iv,3),
frek(iv), alphamatrix(iv,4), frek(iv), alphamatrix(iv,5), frek(iv), alphamatrix(iv,6));
g = get(j(1),'Parent');
for ii = 1:6
    set(j(ii),'LineWidth',2);
end
set(g,'FontSize',14);
g = legend('0 Ohm','5 Ohm','10 Ohm','20 Ohm','30 Ohm','Infinity');
%set(g,'Location','best');
g= title ('Shunt resistances (Theoretical)');
g= xlabel('Frequency [Hz]');
g= ylabel ('Absorption factor [-]');
grid


%Measured:
figure(2)
j=plot(frek(iv), alpha1, frek(iv), alpha2, frek(iv), alpha3, frek(iv), alpha4, frek(iv), alpha5,
frek(iv), alpha6);
g = get(j(1),'Parent');
for ii = 1:6
    set(j(ii),'LineWidth',2);
end
```

```
set(g,'FontSize',14);
g = legend('0 Ohm','5 Ohm','10 Ohm','20 Ohm','30 Ohm','Infinity');
%set(g,'Location','best');
g= title ('Shunt resistances (Measured)');
g= xlabel('Frequency [Hz]');
g= ylabel ('Absorption factor [-]');
grid
```

7.2.2 Optimal Shunt Resistance

Input Data

```
rho=1.21;                      %air density                [kg/m^3]
c=343;                       %speed of sound             [m/s]
Slsp=154e-4;              %effective piston area (from datasheet) [m^2]
Stube=0.2^2;                 %cross section of tube [m^2]
```

Measured Parameters

```
Bl=6.48;                      %Force factor                [N/A]
Rec=6.95;                      %Voice coil resistance      [ohm]
Rms=0.34;                     %suspension mechanical resistance  [Ns/m]

Ropt_measured_input=((Bl)^2/(rho*c*(Slsp^2/Stube)-Rms))-Rec
```

Datasheet Parameters

```
Bl=7.15;                      %Force factor                [N/A]
Rec=7.3;                      %Voice coil resistance      [ohm]
Rms=0.38;                     %suspension mechanical resistance  [Ns/m]

Ropt_datasheet_input=((Bl)^2/(rho*c*(Slsp^2/Stube)-Rms))-Rec
```

7.2.3 Resonance frequency

With the parameters from measurement:

Loudspeaker in tube

```
Mmd=10.4e-3;             %measured
Mms=11.5e-3;             %measured
```

```
Cms=0.148e-3;           %measured


rho=1.21;
c=343;
Slsp=154e-4;
B1=0.6;
B2=0.6;     %for loudspeaker placed in tube

a=sqrt(Slsp/pi);
L=0.15;
Stube=0.2^2;
V=L*Stube;               %Cavity volume of the tube
Ca=V/(rho*c^2);     % acoustic cav

wo=sqrt(((1/Cms)+(Slsp^2/Ca))/((Mmd+B1*rho*Slsp^2/(pi*a))+(B2*rho*Slsp^2/(pi*a))));

fo_meas_in_tube=wo/(2*pi)
```

```
fo_meas_in_tube =

  161.9111
```

Loudspeaker in room

```
V=6135e-6;             %Volume of the closed box
Ca=V/(rho*c^2);          % acoustic cav
B2=0.85;                 %for loudspeaker placed in free air
wo=sqrt(((1/Cms)+(Slsp^2/Ca))/((Mmd+B1*rho*Slsp^2/(pi*a))+(B2*rho*Slsp^2/(pi*a))));

fo_meas_in_room=wo/(2*pi)
```

```
fo_meas_in_room =

  158.9479
```

Parameters from the datasheet:

Loudspeaker in tube

```
Mmd=10.89e-3;            %datasheet
Mms=11.97e-3;            %datasheet
Cms=0.139e-3;           %datasheet

B1=0.6;
B2=0.6;     %for loudspeaker placed in tube
```

```
a=sqrt(Slsp/pi);
L=0.15;
Stube=0.2^2;
V=6135e-6;              %Volume of the closed box
Ca=V/(rho*c^2);     % acoustic cav

wo=sqrt(((1/Cms)+(Slsp^2/Ca))/((Mmd+B1*rho*Slsp^2/(pi*a))+(B2*rho*Slsp^2/(pi*a))));

fo_datasheet_in_tube=wo/(2*pi)
```

```
fo_datasheet_in_tube =

  160.6916
```

Loudspeaker in room

```
V=L*Stube;                %for the cavity of the tube
Ca=V/(rho*c^2);           % acoustic cav
B2=0.85;                  %for loudspeaker placed in free air
wo=sqrt(((1/Cms)+(Slsp^2/Ca))/((Mmd+B1*rho*Slsp^2/(pi*a))+(B2*rho*Slsp^2/(pi*a))));

fo_datasheet_in_room=wo/(2*pi)
```

```
fo_datasheet_in_room =

  159.3997
```

7.3 Sound Absorption Performance in a Room

T30 input data

```
c = [
   7.9  7.54   8.19   7.22   6.27   5.64   4.5    4.32
5.93    7.1    7.31   8.89   9.6    8.28   4.62   4.42
7.18    8.29   8.42   6.68   6.57   5.81   4.68   4.13
5.49    7.49   7.38   5.94   8.69   8.23   5.17   4.12
7.61    7.41   10     9.18   6.61   6.7    4.98   4.64
6.83    7.51   8.14   8.7    6.42   5.68   4.73   4.33
5.96    7.49   7.91   6.42   6.14   6.27   4.92   4.3
6.17    6.97   7.29   6.25   7.99   6.7    4.21   4.27
6.01    8.93   8.66   7.71   6.61   5.08   4.65   4.29
6.15    7.97   2.43   7.58   6.65   5.66   4.47   4.18
6.27    8.78   7.98   6.65   6.4    5.63   4.59   4.56
5.84    6.43   6.37   6.58   7.81   6.85   4.54   4.39
6.1     8.16   7.82   7.05   6.72   5.76   4.87   4.45
5.89    8.09   8.04   7.17   6.58   6.08   4.71   4.37
6.24    8.08   8.71   8.02   6.29   5.51   4.45   4.34
6.06    7.88   8.22   7.24   6.43   5.5    4.64   4.27
6.28    7.47   7.62   7.03   7.4    6.4    4.68   4.26
6.17    8.16   9.61   7.6    6.07   5.59   4.54   4.19
6.23    7.56   9.28   7.96   7.24   6.38   4.32   4.1
6.29    7.5    8.88   7.93   6.36   5.76   4.48   4.47
5.78    9.1    8.85   6.66   6.36   5.58   4.72   4.3
5.91    6.27   5.68   5.78   5.59   5.08   4.45   4.43
3.96    7.39   6.86   5.59   5.54   5.11   4.7    4.24
4.63    5.26   5.56   5.03   4.83   4.24   3.56   3.72
7.11    7.61   8.4    5.52   4.75   5.49   4.7    4.22
5.49    7.29   8.45   5.95   5.03   5.2    4.01   4.16
6.22    7.55   6.14   6.42   6.94   6.71   4.72   4.26
6.14    6.5    6.53   3.75   3.14   3.26   2.79   2.71
8.12    7.88   9.01   7.03   6.44   4.7    4.57   4.24
8.22    8.65   8.03   6.01   5.82   5.02   4.39   4.09
6       9.74   9.19   6.79   6.58   5.2    4.74   4.42
5.91    6.92   6.13   6.98   5.33   4.73   4.45   4.64
4.2     8.6    7.01   5.86   5.43   5.07   4.62   4.39
4.77    4.96   5.53   4.77   4.98   4.45   3.81   3.84
7.43    9.03   8.59   6.06   5.1    5.34   4.91   4.24
5.23    8.71   8.65   6.28   5.97   5.56   4.31   4.35
6.29    8.06   7.57   6.64   7.88   6.16   4.56   4.38
6.01    6.74   6.38   3.85   3.04   3.12   2.94   3.25
8.27    8.75   9.17   7.24   6.4    4.68   3.64   4.29
8.17    8.74   8.19   6.42   5.62   4.82   4.44   4.29
5.97    6.92   8.5    7.41   5.76   5.21   4.63   3.8
4.49    4.63   6.02   4.95   4.83   3.88   3.27   3.54
5.73    7.11   4.1    5.8    5.51   4.84   3.99   4.15
```

```
4.94    5.5     7.49    7.01    8.69    7.61    4.17    3.87
5.09    8.48    8.45    7.31    5.32    5.46    4.93    3.95
6.6     7.5     7.61    8.71    6.01    5.71    4.85    4.32
4.42    7.36    7.91    6.57    6.15    5.58    4.74    4.54
5.51    6.61    7.37    5.92    7.5     6.07    4.44    4.28
5.19    8.84    7.4     7       6.39    5.23    4.67    4.26
4.36    5.38    4.15    4.05    4.31    4.37    3.78    3.83
6.09    7.03    8.27    7.85    6.03    5.43    4.8     4.32
4.63    5.36    6.57    4.99    5.41    4.06    3.23    3.49
5.73    8.2     5.26    5.61    5.7     4.54    4.13    4.22
5.16    6.22    7.73    6.77    9.27    7.45    4.3     3.95
5.46    9.15    9.69    7.49    5.97    5.13    4.53    4.35
6.26    7.84    7.99    8.62    7.18    6.64    4.93    4.74
4.65    8.21    8.23    6.65    6.26    5.47    4.67    4.66
5.89    6.82    7.43    5.91    8.27    7.08    4.87    4.48
5.37    9.67    8.26    7.78    6.93    5.5     4.92    4.56
4.47    5.87    4.62    4.11    4.66    4.28    4.05    3.72
];
```

Calculations

```
nbands = size(C,2);
nreps = size(C,1)/3;

T60w = mean(C([[21:30] [41:50]],:)).';                          %with abs
T60wo  = mean(C([[31:40] [51:60]],:)).';                         %without abs
% T60abssource_open=mean(C([1:10],:)).';                          %With abs as source
% T60abssource_short=mean(C([11:20],:)).';                          %With abs as source
T60abssource=mean(C([1:20],:)).';                          %With abs as source

stdw = std(C([[21:30] [41:50]],:)).';
stdwo  = std(C([[31:40] [51:60]],:)).';
% stdabssource_open=std(C([1:10],:)).';
% stdabssource_short=std(C([11:20],:)).';
stdabssource=std(C([1:20],:)).';

if nreps == 20
    tp = 2.093;
else
   error('Find the tp value for ',int2str(nreps),' repetitions');
end

ciwo = stdwo*tp/sqrt(nreps);
ciw  = stdw*tp/sqrt(nreps);
% ciabssource_open  = stdabssource_open*tp/sqrt(nreps/2);
% ciabssource_short  = stdabssource_short*tp/sqrt(nreps/2);
ciabssource  = stdabssource*tp/sqrt(nreps);
```

```
fvec = [1:nbands];
```

Plot

```
figure(1)
h =
plot(fvec,T60w,'-ob',fvec,T60wo,'-or',fvec,T60w+ciw,'b--',fvec,T60wo+ciwo,'r--',fvec,T60w-c
iw,'b--',fvec,T60wo-ciwo,'r--');
g = get(h(1),'Parent');
set(g,'FontSize',14);
for ii = 1:6
    set(h(ii),'LineWidth',2);
end
grid
labelvec = set(g,'XTickLabel');
labelvec{1} = '100';
labelvec{2} = '125';
labelvec{3} = '160';
labelvec{4} = '200';
labelvec{5} = '250';
labelvec{6} = '315';
labelvec{7} = '400';
labelvec{8} = '500';
set(g,'XTickLabel',labelvec);
g = legend('With absorber','Without absorber','\pm 95% conf.int. with abs','\pm 95% conf.int.
without abs');
set(g,'Location','best');
g = ylabel('T30    [s]');
set(g,'FontSize',14);
g = xlabel('Frequency    [Hz]');
g = title ('Influence of the absorber');
set(g,'FontSize',14);

%Influence of the sorce loudspeaker
figure(2)
j=plot(fvec,T60w,'-ob',fvec,T60abssource,'-og',...
     fvec,T60w+ciw,'b--',fvec,T60abssource+ciabssource,'g--',...
    fvec,T60w-ciw,'b--',fvec,T60abssource-ciabssource,'g--');
% j=plot(fvec,T60abssource_short,'-og',fvec,T60abssource_open,'-or',...
%
fvec,T60abssource_short+ciabssource_short,'g--',fvec,T60abssource_open+ciabssource_open,'r-
-',...
%
fvec,T60abssource_short-ciabssource_short,'g--',fvec,T60abssource_open-ciabssource_open,'r-
-');
g = get(j(1),'Parent');
set(g,'FontSize',14);
```

```
for ii = 1:6
    set(j(ii),'LineWidth',2);
end
grid
labelvec = set(g,'XTickLabel');
labelvec{1} = '100';
labelvec{2} = '125';
labelvec{3} = '160';
labelvec{4} = '200';
labelvec{5} = '250';
labelvec{6} = '315';
labelvec{7} = '400';
labelvec{8} = '500';
set(g,'XTickLabel',labelvec);
g = legend('With absorber','Absorber as source','\pm 95% conf.int. with abs','\pm 95% conf.int.
abs as source');
% g = legend('Absorber as source,short','Absorber as source,open','\pm 95% conf.int. with
abs','\pm 95% conf.int. abs as source');
set(g,'Location','best');
g = ylabel('T30   [s]');
set(g,'FontSize',14);
g = xlabel('Frequency   [Hz]');
g = title ('Influence of the source lsp');
set(g,'FontSize',14);

% Normalized difference between abs and no abs

T60diff = C([[31:40] [51:60]],:) - C([[21:30] [41:50]],:);

T60diffmean = mean(T60diff).';
T60diffstd = std(T60diff).';
tvalue = abs(T60diffmean./(T60diffstd/sqrt(nreps)));

tcritical = 1.7291;  % for n-1 = 19

%Influence of the absorber
figure(3)
h=plot(fvec,tvalue,'-ob',fvec,tcritical*ones(nbands,1),'r-');
g = get(h(1),'Parent');
set(g,'FontSize',14);
for ii = 1:2
    set(h(ii),'LineWidth',2);
end
grid
labelvec = set(g,'XTickLabel');
labelvec{1} = '100';
labelvec{2} = '125';
labelvec{3} = '160';
labelvec{4} = '200';
```

```
labelvec{5} = '250';
labelvec{6} = '315';
labelvec{7} = '400';
labelvec{8} = '500';
set(g,'XTickLabel',labelvec);
g = ylabel('Normalized T30 difference   [-]');
set(g,'FontSize',14);
g = xlabel('Frequency   [Hz]');
set(g,'FontSize',14);
g = legend('Normalized T30-difference','Critical value for significant difference');
set(g,'Location','best');
g = title ('Influence of the absorber');
set(g,'FontSize',14);

%Normalized difference between source loudspeakers

T60diff = C([[41:50] [21:30]],:) - C([1:20],:);

T60diffmean = mean(T60diff).';
T60diffstd = std(T60diff).';
tvalue = abs(T60diffmean./(T60diffstd/sqrt(nreps)));

tcritical = 1.7291;  % for n-1 = 19,

%influence of the source loudspeaker
figure(4)
h=plot(fvec,tvalue,'-og',fvec,tcritical*ones(nbands,1),'r-');
g = get(h(1),'Parent');
set(g,'FontSize',14);
for ii = 1:2
    set(h(ii),'LineWidth',2);
end
grid
labelvec = set(g,'XTickLabel');
labelvec{1} = '100';
labelvec{2} = '125';
labelvec{3} = '160';
labelvec{4} = '200';
labelvec{5} = '250';
labelvec{6} = '315';
labelvec{7} = '400';
labelvec{8} = '500';
set(g,'XTickLabel',labelvec);
g = ylabel('Normalized T30 difference   [-]');
set(g,'FontSize',14);
g = xlabel('Frequency   [Hz]');
set(g,'FontSize',14);
g = legend('Normalized T30-difference','Critical value for significant difference');
set(g,'Location','best');
```

```
g = title('Influence of the source lsp');
set(g,'FontSize',14);

%Absorption Area
V=120.7;                          %Volume of room
c=343;                            %speed of sound
A=(24*log(10)*V/c).*(1./T60w-1./T60wo);
ciA=sqrt(abs(0.161*V./-(T60w.^2).*ciw).^2 +abs(0.161*V./-(T60wo.^2).*ciwo).^2);

figure(5)
h = plot(fvec,A,'-ob',fvec,A+ciA,'b--',fvec,A-ciA,'b--');
g = get(h(1),'Parent');
set(g,'FontSize',14);
for ii = 1:2
    set(h(ii),'LineWidth',2);
end
grid
labelvec = set(g,'XTickLabel');
labelvec{1} = '100';
labelvec{2} = '125';
labelvec{3} = '160';
labelvec{4} = '200';
labelvec{5} = '250';
labelvec{6} = '315';
labelvec{7} = '400';
labelvec{8} = '500';
set(g,'XTickLabel',labelvec);
g = legend('Absorption area','\pm 95% conf.int.');
set(g,'Location','best');
g = ylabel('Absorption Area   [m^2]');
set(g,'FontSize',14);
g = xlabel('Frequency   [Hz]');
g = title ('Effective Absorption Area');
set(g,'FontSize',14);
```