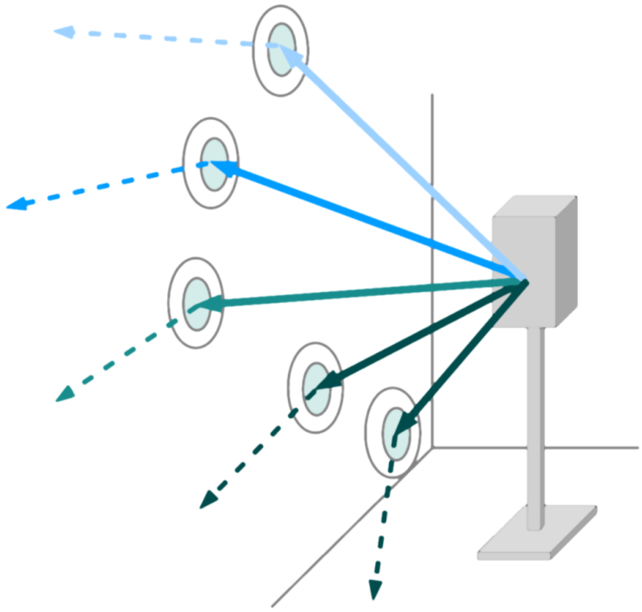
Estimation of Room Boundary Characteristics



Final project, Master of Acoustics, 4th Semester, Spring 2013

13gr1060:
Regnar Oxholm Bonde

Department of Electronics Systems

1 PREFACE

This report is written by group 13gr1060 at 4th semester on the Acoustics Master program at the department of electronics systems, Aalborg University, spring 2013. The report is a final project proposed by the company B&O.

Group member:

Regnar Oxholm Bonde

Internal supervisor:

Søren Krarup Olesen

External supervisor:

Martin Bo Møller , B&O

1.1 ACKNOWLEDGE

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Thanks to Peter Dissing and Claus Vestergaard Skipper for their help regarding equipment and measurements in the laboratory.

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1.2 READING GUIDE

The project documentation is divided into the following three parts:

- **Report:** is the main documentation for the project and is chronologically composed. To understand the project it is recommended to read this part. The report is divided into several smaller sections. A problem formulation section where the problem is described. An analysis section where theory and practical issues are discussed and analyzed. An implementation section where the development and implementation of the problem are described and finally a conclusion. If a fast overview is needed read the introduction, problem formulation and the conclusion.

- **Appendices:** include further and deeper information about the project. However the appendices are not mandatory for the project understanding. Measurement journals, references etc. are placed in the appendices, section 8.

- **DVD:** includes Matlab codes, measurement data, etc. Documents which have low importance for the project or data which are not printable. The DVD does also contain the report and the appendices as PDF.

References for used material are written in squared brackets with author surname and year of publication. The same is applicable for webpages but only the page name is in the brackets. A total list of references is available in section 8.5. References to codes and other files on the DVD are written in *italic* and a total list of DVD content is available in section 8.3.

The report is written in a manner that assumes the reader to have basics knowledge in the acoustic fields, room acoustics, loudspeakers, digital signal processing and psychology of hearing. If further and deeper information about these fields are needed, the referred books can be used to gain knowledge.

2 CONTENT

3 INTRODUCTION

Since Thomas Edison in 1877 developed the phonograph [Wiki1], sound reproduction has undergone a massive development. From sound stored and reproduced 100% analog and mechanical to digital stored sound, reproduced through multichannel loudspeaker systems with several amplifiers and digital filters. The reason behind this massive development is mainly based on our wish to reproduce sound as neutral and close to reality as possible. We want to feel that the concert, singer or other real sound source we are reproducing is real. However, the manner a lot of today's music is recorded and produced is far from a live performance and the natural sound and reality is therefore often what the recording engineer wants us to hear [Toole, 2008].

To achieve natural sound reproduction, not only the loudspeakers, amplifiers etc. are important but also the room where reproduction takes place. The room will affect the radiated sound from the loudspeakers and depending on the room characteristic, and particularly the room boundary characteristic, the sound at listener position will be colored. This coloration can e.g. be delays, reverberation and change of frequency response. To deal with this coloration, especially change of frequency response, different solutions already exist, e.g. the Audyssey room equalization system [Audyssey] and the Roomperfect system [Roomperfect]. These systems equalize the frequency response to a target curve at desired positions, based on measurements performed at the positions. However, this is not without issues. Room equalization is symptom treating which do not take care of the main problem, reflections, caused by room boundary characteristic. Therefore, the sound is only improved at the desired positions. Outside these positions the sound may be even worse than without the room equalization [Toole, 2008].

An improvement to sound reproduction will be to take care of the main problem in the rooms and in order to do that, investigation and estimation of room boundary characteristics are needed.

4 Problem formulation

4.1 Objective

The objective of this project is to investigate room boundary characteristics, in mind that the information about the room boundary characteristics is intended to be used for coloration compensation of sound, reproduced through a stereo loudspeaker system in a living room. Which room boundary characteristics that influences and add coloration to the sound, which of the room boundary characteristics that have the greatest influence on the sound and how the room boundary characteristics can be measured in-situ, is a part of the investigation. Combining the results from the investigation shall lead to a method to measure the needed room boundary characteristics, needed for sound coloration compensation, with minimum user interaction and equipment as possible. Sound coloration compensation may be equalization, change of loudspeaker radiation characteristics, feedback to user about loudspeaker and furniture positions etc. Actions which can attenuate or remove sound coloration caused by the room boundaries.

4.2 Limitations

Due to a limited time-period, man-hours, and for simplifying (easier analyzing of results), project limitations are introduced. The following points will be covered / not covered by the project and project report.

- The project and its investigations will be based on a standard stereo setup consisting of 2 full range (20Hz-20KHz) loudspeakers in a living room. This is a commonly used setup and very used reproduction format for music and audio material. Multichannel systems and subwoofers will not be used. Depending on loudspeaker type and technology, the low frequency limit at 20Hz may not be reached.

- The project will focus on mid and low frequencies (20Hz to 1KHz). High frequencies have shorter wavelength and higher precision for measurement positions etc. are needed. For convenience and simplification, only the mid and low frequencies will therefore be used.

- The investigation of room boundary conditions will mainly be based on the physical aspects.

- Loudspeakers, amplifiers etc. will not be designed unless special designs are needed. Existing equipment will be used much as possible.

- A sound coloration compensation system will not be designed. Only investigations and measurements of room boundary characteristics will be covered by this project.

5 ANALYSIS AND THEORY

5.1 INTRODUCTION

To obtain the results specified in problem formulation, theory are needed to support the ideas and solution. The analysis and theory section will therefore go through the theory needed to understand the behavior of sound in a room, loudspeakers and the stereo setup, and the behavior when the stereo setup is positioned in a room. Using the theory, different room boundary characteristics are then defined, discussed and the weighting between these are analyzed using a simple simulation. Finally, different measurement methods for the characteristics will be stated and analyzed.

5.2 THE ROOM

Rooms can have many shapes and sizes and the walls, which is obvious a part of the room boundary characteristics, can be made of many different materials and thereby affect the sound in a room. Also the furniture in rooms can be categories as room boundaries and furniture can likewise be made of different materials and have many shapes and sizes. Together these room boundaries will affect the behavior of sound in a room. In this section, about the room, the basic theory about this behavior is explained, but firstly the living room is defined.

5.2.1 DEFINITION OF A LIVING ROOM

Living rooms are the rooms where stereo setups are normally placed at private homes and are therefore the main room for enjoying music. The investigation of room boundary characteristics will therefore also be based on room boundaries which exist in living rooms. To define the living room, the definition from the standard for listening tests on loudspeakers [IEC 60268-13] is used because these definitions are intended to explain the average living room and its acoustical behavior. From this standard the room has to be between 25 to 40m^2, has a reverberation time, T60, between 0.3 and 0.6s, averaged for frequencies between 200Hz and 4KHz and the background noise should be below 25dB re. 20µPa A-weighted and should not contain tonal, impulsive or cyclical components.

5.2.2 TRANSITION FREQUENCY

The behavior of sound in a room can be divided into two different frequency ranges determined by the transition frequency, also known as the Schroeder frequency [Toole, 2008]. Below the transition frequency, room modes are the dominating phenomenon but the higher the frequency, the more room modes. The more room modes, the more will they smooth and cancel each other out and be less dominating. Above the transition frequency, the early reflections are therefore the dominating phenomenon. Early reflections are reflections delayed maximum 30ms from the direct sound and will be described further in section 5.4.2. The transition frequency, f_c[Hz], is calculated using equation (5.1), where T[s] is the reverberation time, T60, and V[m^3] the volume of the room.

$$f_c = 2000\sqrt{\frac{T}{V}} \tag{5.1}$$

Equation (5.1) is however an assumption and the exact transition frequency, for a particular room, can be found by measuring the room impulse response and by visually inspection of the frequency response.

5.2.3 ROOM MODES

The room modes are generated in rooms due to standing waves and occur when the path length of a reflected sound wave is equal an integer multiple of half wavelengths [Toole, 2008]. The standing waves and thereby room modes can be generated between two parallel room boundaries, between several room boundaries in one plane and between all room boundaries. These different mode types are respectively called axial, tangential and oblique modes and the axial modes are typical the most dominating. The room modes can be calculated using equation (5.2) where n is the mode order, l[m] is the room dimensions and c[m/s] is the speed of sound in air. x,y and z denotes the direction.

$$f_{n_x n_y n_z} = \frac{c}{2}\sqrt{\left(\frac{n_x}{l_x}\right)^2 + \left(\frac{n_y}{l_y}\right)^2 + \left(\frac{n_z}{l_z}\right)^2} \tag{5.2}$$

The equation assumes that the room is a perfect box with infinite-rigid room boundaries (walls) and therefore has perfect reflections at the boundaries. This is not true for real rooms. Real rooms are constructed using different materials like plasterboards, bricks, wood etc. and these materials cannot be seen as rigid at all frequencies. Depending on the material-type, thickness, stiffness, size etc. the boundaries will have different resonance frequencies and surfaces, and thereby reflect the incident sound wave differently depending on the frequency. The reflected sound wave may even be phase shifted. Also few living rooms are perfect boxes due to doors, windows, fireplaces and furniture. Equation (5.2) is therefore an assumption which can give a hint about the room modes. Real rooms are much more complex. In order to be able to explain better what happens in real rooms, information about the room boundaries and how they reflect sounds are needed. This information can be explained using the specific acoustic impedance.

5.2.4 SPECIFIC ACOUSTIC IMPEDANCES

The specific acoustic impedance, z[Pa·s/m], is one out of 3 typical used impedances in acoustics and is a characteristic property of the medium and of the type of wave that is being propagated [Kinsler, 1999]. When it comes to transmission and reflections this impedance is very useful. The two other impedances used in acoustic are the acoustic impedance, Z[Pa·s/m^3], and the radiation impedance, Z_r[Pa·s·m]. Acoustic impedance is useful in lumped acoustics and the radiation impedance is used for coupling between acoustic waves and a driving source or load. The specific acoustic impedance is described by equation (5.3), where p[Pa] is the pressure and u[m/s] is the particle velocity.

$$z = p/u \tag{5.3}$$

In words, the specific acoustic impedance is how much pressure which is needed to have unit velocity. E.g. an infinite rigid room boundary will have infinite specific acoustic impedance because the velocity at the room boundary will be 0m/s. No matter how big the pressure is, the room boundary will not move and the velocity will therefore be 0m/s. Combining the wave equation for a plane wave traveling along the x-axis, equation (5.4), with equation (5.3), newton's 2.law and thermodynamics leads to equation (5.5) where the specific acoustic impedance, z[Pa·s/m], can be calculated using the speed of sound, c[m/s], and density, ρ_0[Kg/m^3], for the specific medium.

$$\frac{\partial^2 p}{\partial x^2} = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} \tag{5.4}$$

$$z = \rho_0 c \tag{5.5}$$

The specific acoustic impedance will be a real quantity for the plane wave but this will not hold for standing waves or diverging waves. In general the specific acoustic impedance will be complex and is therefore of the form in equation (5.6) where r is the specific acoustic resistance and x is the specific acoustic reactance. The specific acoustic impedance will also be frequency depended.

$$z = r + jx \tag{5.6}$$

In a living room where a loudspeaker radiation sound, the sound waves from the loudspeaker cannot be seen as plane waves due the physics behind the loudspeaker and due to the short distances in the room compared with wavelengths. The radiation is closer to a spherical source which specific acoustic impedance is calculated using equation (5.7) and (5.8).

$$z = \rho_0 c \frac{(kr)^2}{1 + (kr)^2} + j\rho_0 c \frac{kr}{1 + (kr)^2} \tag{5.7}$$

$$kr = \frac{2\pi r}{\lambda} \tag{5.8}$$

r[m] is the distance from the sphere and λ[m] is the wavelength. If the distance, r[m], is long, the sound wave can be seen and calculated as a plane wave.

5.2.5 Reflection factor

When a sound wave is radiating from one medium to another medium, the ratio between specific acoustic impedances in the two mediums will decide how much of the incident sound wave will be reflected and transmitted [Kinsler, 1999]. If the two specific acoustic impedances are identical there will be a total transmission and therefore no reflected sound wave. If the specific acoustic impedances are very different the transmission will be low and the reflections high. In equation (5.9) the reflection factor, R, is calculated for a sound wave in air with incident angle, θ[rad], on a medium with the specific acoustic impedance, z[Pa·s/m]. The medium could be a wall or some furniture in a room, a room boundary. ρ_0c[Pa·s/m] is air density multiplied with speed of sound in air, the specific acoustic impedance for air.

$$R(\theta) = \frac{z \cdot \cos(\theta) - \rho_0 c}{z \cdot \cos(\theta) + \rho_0 c} \tag{5.9}$$

The reflection factor can also be explained as the ratio between the incident, p_i[Pa], and reflected sound wave, p_r[Pa], equation (5.10).

$$R = \frac{p_r}{p_i} \tag{5.10}$$

5.2.6 Absorption coefficient

When it comes to room acoustics, the reflection factor can be simplified to the absorption coefficient, α. Opposite the reflection factor, the absorption coefficient explains how much sound is absorbed instead of reflected. Any imaginary parts and thereby phase information is removed when the absorption coefficient is calculated. In equation (5.11) the absorption coefficient is calculated for specific angles, θ[rad], using the reflection factor [Kinsler, 1999].

$$\alpha(\theta) = 1 - |R(\theta)|^2 \tag{5.11}$$

When the absorption coefficient is stated for different materials it is typical for a random incident and in octave bands. This can be calculated using equation (5.12) [Kuttruff, 2009].

$$\alpha_{uni} = \int_0^{\pi/2} \alpha(\theta) \cdot \sin(2\theta)\, d\theta \tag{5.12}$$

Due to this manner of describing acoustic behavior of materials using only the abortion coefficient, it is not possible to tell or calculate anything about how the materials reflects sounds waves at different incident angles or if the materials provide some phase shift. In concert halls this is not a big issue since the sound field is close to diffuse field and the sound wave incident angles are therefore random. In living rooms the sound field behaves at bit differently.

5.2.7 THE SOUND FIELD

In concert halls the sound field is assumed to be diffuse, which means that the sound is uniformly distributed and radiated in all directions. This simplifies the calculations of different acoustic parameters and in general simplifies basic concert hall acoustics [Toole, 2008]. The diffuse field assumption is however not true since concert halls have different boundaries, seats and audience which reflect sounds differently and decrease the diffusivity, but the assumption is still very well used and give acceptable results. An example of a room with good diffuse field is the reverberation chamber, used to measure absorption coefficients of different materials, following the standard [ISO 354]. For small rooms like living rooms, the assumption of diffuse field will not apply due to the size of the rooms and types of room boundaries which exist in living rooms. Walls and ceiling are typically made of hard materials with high specific acoustic impedances and therefore reflect sounds very well. Furniture and an eventual carpet on the floor are typical made of soft materials with lower specific acoustic impedances and therefore absorb sounds. These different boundaries made of different materials are not uniformly distributed in living rooms and so the reflections. This gives overall a sound field which cannot be defined as diffuse and measurement and calculations that assumes diffusivity are therefore not appropriate in small rooms. If a sound source radiates a sound in a small room, the sound will typical hit a soft absorbing boundary or hit a hard reflecting boundary which reflect the sound into an absorbing boundary. The sound field will therefore be dominating by strong early reflections and highly damped late reflections. Not enough the lack of diffusivity increase the complexity of the sound field in a living room, the sound field may also change over time. [Toole, 2008] explains a measurement setup in a conference room with approximately 0.4s reverberation time, where the sound field was measured using a microphone array. Sound was generated by a loudspeaker radiating in the X direction in the room, seen from top view, and the reverberation sound field was measured using the microphone array. It was then discovered that the sound field changed from being dominated in the X direction at time 0s to be dominating in the Y direction. The field rotates 90 degrees but at some point, after approximately 50ms, the measured sound field seems to be diffuse or uniformly distributed before it becomes dominating in the Y direction.

5.2.8 WEATHER CONDITIONS

Since the speed of sound in air, c[m/s], and air density, ρ_0[Kg/m^3], depends on temperature, humidity and pressure, the weather conditions can influence the specific acoustic impedance for air and therefore also the reflection factor and absorption coefficient. The speed of sound can be calculated using equation (5.13) [Kinsler, 1999] where T[^{o}C] is the temperature and c_0[m/s] is the speed of sound in air a 0^{o}C which is 331.5m/s. This equation is a simplification and holds for temperatures between 0 and 100^{o}C and will not be exceeded in living rooms.

$$c = c_0\left(1 + \frac{T}{273}\right)^{\frac{1}{2}} \tag{5.13}$$

Air density can be calculated using equation (5.14) [Wiki2] where p_d[Pa] is partial pressure of dry air, p_v[Pa] is pressure of water vapor, T_k[^{o}K] is the temperature, R_d[J/Kg·^{o}K] is the specific gas constant for dry air (287.058J/(Kg·^{o}K)) and R_v[J/Kg·^{o}K] is the specific gas constant for water vapor (461.495 J/(Kg·^{o}K)).

$$\rho_0 = \frac{p_d}{R_d T_K} + \frac{p_v}{R_v T_K} \tag{5.14}$$

The pressure of water vapor, p_v[Pa], is calculated in equation (5.15) where ϕ[%] is the relative humidity and T[^{o}C] is the temperature. The partial pressure for dry air is calculated in equation (5.16) where p[Pa] is the static air pressure.

$$p_v = \phi \cdot 6.1078 \cdot 10^{\frac{7.5T}{T+237.3}} \tag{5.15}$$

$$p_d = p - p_v \tag{5.16}$$

5.3 THE LOUDSPEAKER AND STEREO SETUP

Since the objective is to investigate room boundary characteristics, in mind that the information is intended to be used for coloration compensation of sound, reproduced through a stereo loudspeaker system, some basic loudspeaker theory are needed. This basic theory address loudspeaker radiation characteristics since it may be important to know which frequencies are radiated in which directions, and the stereo setup. Different types of loudspeakers exist including different loudspeaker unit technologies and enclosure types but when nothing else is mentioned in this report, the referred loudspeaker type is a simple dynamic loudspeaker unit with moving coil motor mounted in an closed enclosure. See **Figure 5.1**.

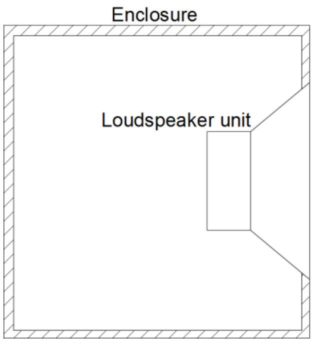


Figure 5.1 – Illustration of the loudspeaker consisting of a dynamic loudspeaker unit and a closed enclosure.

5.3.1 RADIATION CHARACTERISTICS

A sound source in its simplest form can be explained as a pulsating sphere where the radiated pressure field is uniform spherical distributed independent of frequency. This is called a simple source [Kinsler, 1999]. Loudspeakers are however far from this explanation and have much more complex radiation characteristics at some frequencies. If the sound source is non-spherical, which holds for loudspeakers, the sound source can only be equivalent a simple source if the radiated wavelength is longer than the dimensions of the sound source or, in loudspeaker cases, enclosure dimension. When the radiated wavelength is shorter than the sound source or enclosure dimensions, radiation characteristics depend on many different factors. One of them is the source dimensions or for the loudspeaker, the membrane dimensions.

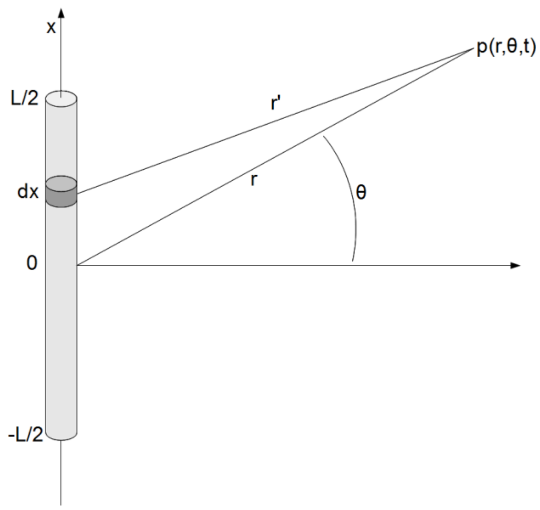


Figure 5.2 – Illustration of a continuous line source with length L and radius a. The pressure, p, in far field is calculated using the distances r and r', the angle, θ, and simple sources along the line source with length dx.

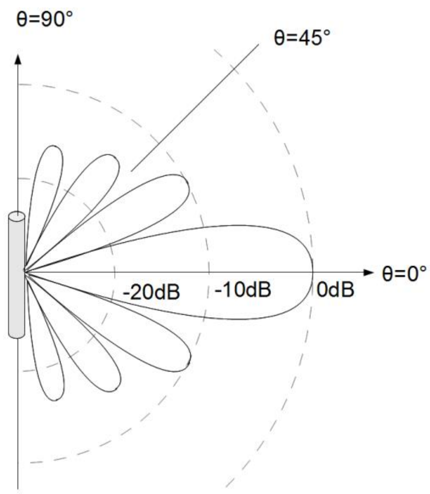


Figure 5.3 – Illustration of a beam pattern for a continuous line source radiation sound at a certain frequency. Depends on line source length and frequency, the beam pattern will look different.

Let’s first consider the continuous line source of simple sources, **Figure 5.2**, for which the pressure, p[Pa], at distance, r[m], and angle, θ[rad], can be calculated using equation (5.17) when the simple sources is vibrating radially with speed $U_0 e^{j\omega t}$[m/s].

$$p(r,\theta,t) = \frac{j}{2}\rho_0 c U_0 ka \int_{-L/2}^{L/2} \frac{1}{r'} e^{j(\omega t - kr')} dx \tag{5.17}$$

a[m] is the simple source diameter and L[m] is the continuously line source length. When the distance r[m] is much larger than the line source length, L[m] (r>>L), r’[m] is approximately equal r[m]. This approximation is however not good since r’[m] is used in the exponent in equation (5.1) and errors may occur. A better approximation is $r' \approx r - x \cdot sin\theta$ and equation (5.17) can then be simplified to obtain pressure amplitude at far field, equation (5.18) to (5.21).

$$p(r,\theta) = P_{ax}(r)H(\theta) \tag{5.18}$$

P_{ax}[Pa] is the amplitude of the far field axial pressure and H is the directional factor.

$$H(\theta) = \left|\frac{\sin(v)}{v}\right| \tag{5.19}$$

$$v = \frac{1}{2}kL \cdot \sin(\theta) \tag{5.20}$$

$$P_{ax}(r) = \frac{1}{2}\rho_0 c U_0 \frac{a}{r} kL \tag{5.21}$$

The pressure amplitude in far field of a plane circular piston, which better represent a loudspeaker membrane, can be derived using an array of continuous line sources of different lengths. The pressure amplitude in far field can then be calculated using equation (5.18) like for the continuous line source, but the directional factor, H, is instead calculated using equation (5.22) and (5.23). Far field for the circular piston is obtained when r[m] is much larger than the piston diameter, a[m], (r>>a).

$$H(\theta) = \left|\frac{2\mathrm{J}_1(v)}{v}\right| \tag{5.22}$$

$$v = ka \cdot \sin(\theta) \tag{5.23}$$

An illustration of a beam pattern for one frequency and source size is showed in **Figure 5.3**. Depends on radiation angle, the loudspeaker will therefore have different frequency responses. Membrane geometry, membrane resonances, enclosure geometry, enclosure resonances, edge diffraction etc. will also affect the radiation characteristics of a loudspeaker. Using only the pressure amplitude in far field calculation to obtain radiation characteristic for a loudspeaker is therefore not adequate. Adding an additional loudspeaker unit in the enclosure, e.g. a smaller unit to take care of the higher frequencies, and a crossover network, will increase the complexity of the radiation characteristics of the loudspeaker. Frequencies radiated from the loudspeaker around the crossover frequency, will be radiated from both loudspeaker units and due to the distance between them, the radiation characteristics is affected. Loudspeaker types as line sources, horns, bipolar, acoustic lenses etc. will also affect the radiation characteristics. E.g. the bipolar loudspeaker type radiates as a figure of 8 with positive pressure in one direction and negative in the other and with the line source or acoustic lens loudspeaker types, it is possible to limit radiation in some directions. In conclusion the loudspeaker characteristics are complex and will vary from loudspeaker to loudspeaker. See **Figure 5.4** and **Figure 5.5**, where different loudspeaker radiation characteristics are illustrated using polar plots.

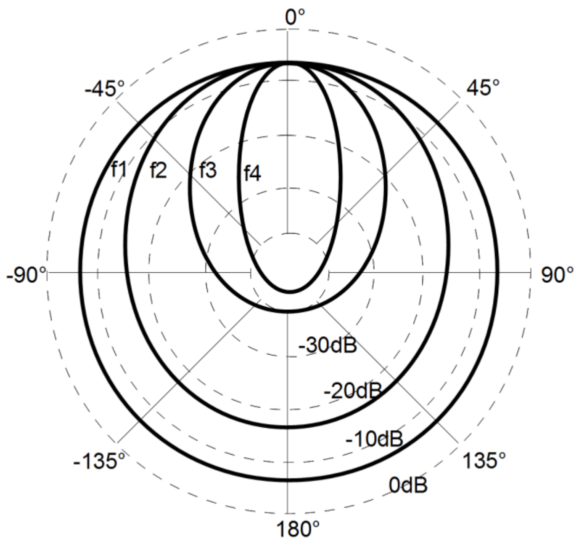


Figure 5.4 – Polar plot Illustration of a general monopole loudspeaker which may be a single loudspeaker unit mounted in a closed enclosure as in **Figure 5.1**. f1 is the lowest frequency and f4 is the highest. The higher frequency the narrower the radiation pattern.

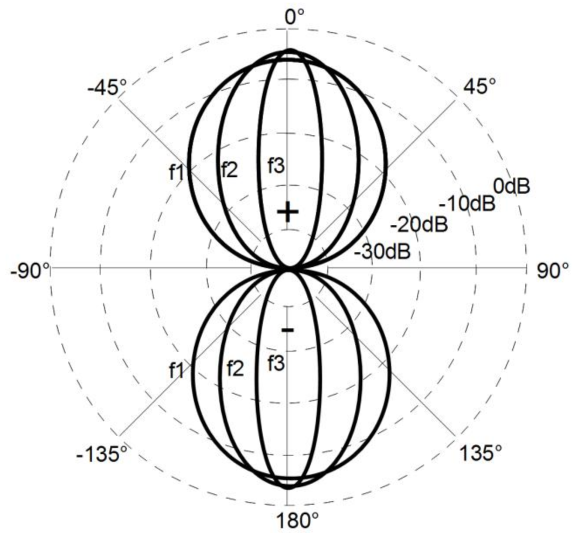


Figure 5.5 - Polar plot Illustration of a general bipolar loudspeaker which radiates as a figure of 8. f1 is the lowest frequency and f3 is the highest. The higher frequency the narrower the radiation pattern.

5.3.2 THE STEREO SETUP

Stereo reproduction of music is a commonly used reproduction technique (Cd's, web streaming services, mp3, etc.) and is intended to create the illusion of directionality between the two needed loudspeakers in a stereo setup. A sound radiated equally from both loudspeakers will generate a phantom image between the loudspeakers and by varying the amplitude of the signals to the right and left loudspeaker and applying delays, the phantom image can be moved in-between the left and right loudspeaker. In order to obtain the intended stereo image and illusion of directivity, the positions of the loudspeakers have to be within some specifications. From [IEC 60268-13], which is a standard for listening tests on loudspeakers, these specifications are illustrated in **Figure 5.6**. The left and right loudspeaker spacing should be between 2 and 3.5m and the distance between the listener position and line connecting the two loudspeakers shall be between 2.5 to 3m. The setup is ideally symmetrical and the listener position should be at least 1m from any significant room boundary.

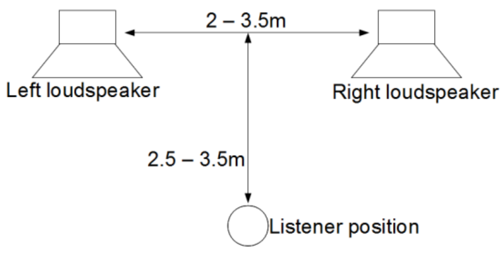


Figure 5.6 – The stereo setup and distances proposed by [IEC 60268-13].

Additionally the loudspeakers should be placed where they are intended to be placed, pointing towards the center of the listening position. E.g. floor loudspeakers on the floor and wall loudspeakers on the wall. Treble/mid loudspeaker units should also be at listener ear height.

5.4 THE LOUDSPEAKER IN THE ROOM

When a loudspeaker is positioned inside a room the interaction between the room and the loudspeaker will give rise to different phenomena. These phenomena can be described physical using the theory explained in the previous sections, section 5.2 and 5.3, but due to our sound perception some of these physical behaviors may not be important and can be downgraded in the further analysis of the room boundary characteristics. This section about the loudspeaker in the room is therefore divided into a physical part and a perceptual part. Firstly the physical behavior is explained.

5.4.1 PHYSICAL

The main physical phenomenon, which will affect the sound radiated from a loudspeaker in the room, are reflections. The loudspeaker will radiate the sound waves in many directions as explained in section 5.3 and the sound waves will therefore not only be radiated to the listener position but also towards room boundaries as walls and thereby generate reflections. Some of these reflections will be reflected towards the listener position and due to different travel lengths for the direct sound waves and the reflected sound waves, comb filtering will occur at the listener position. [Toole, 2008]. See **Figure 5.7**.

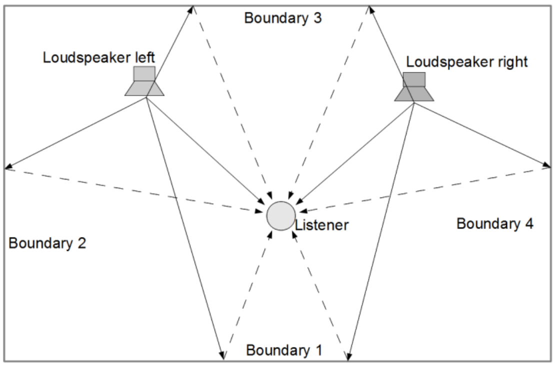


Figure 5.7 – Illustration of first order reflections in a rectangular room with a stereo setup. The solid lines are the direct sound waves and the dashed lines are reflected sound waves. 1. order reflections from left loudspeaker on bounday 4 and from right loudspeaker on boundary 2 will also exist but is leaved out for the sake of simplification.

The frequency responses of the reflected sound waves will not be equal the frequency response of the direct sound wave. As shown in **Figure 5.7** the reflected sound waves are radiated from different angles from loudspeaker and they will therefore have different frequency responses due to the radiation characteristics of the loudspeaker. E.g. the reflections from boundary 3 will only contain low frequencies since the radiation of high frequencies are typical limited to a narrow area in front of the loudspeaker. Also the room boundary specific acoustic impedances will affect the frequency response of the reflected sound wave and as the frequency response for the loudspeaker depends on radiation angle, the specific acoustic impedances depends on the incident angle. In **Figure 5.7** the incident angles on the room boundaries are equal the reflection angles because the room boundaries are assumed to be flat and without any kind of bumps. This does not hold in real life since room boundaries, and specific walls, can be made of bricks, covered with bumpy wallpapers, etc. For sound waves with wavelengths much larger than the room boundary texture or bump dimensions, the reflected sound wave angle will be equals the incident sound wave angle. But for sound waves with wavelengths much smaller than the room boundary texture or bump dimensions, the reflected angle depends on the texture or bump. If the incident sound waves wavelengths are equal the room boundary texture or bump dimensions, the reflected sound wave angle will be random. This behavior is called scattering [Kuttruff, 2009]. See **Figure 5.8**. In real life it is also possible to have sound waves traveling inside the room boundary as bending waves and then radiated somewhere else or have ventilation ducts which will affect the sound at listener position. However, it is believed that these behaviors has minor significance compared to the reflections and is therefore not discussed further in this project.

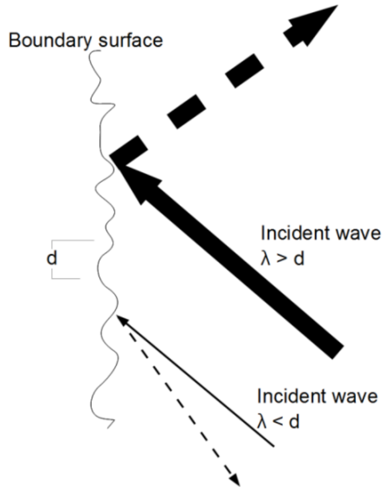


Figure 5.8 – Illustration of how sound waves, with short and long wavelength compared to the boundary surface, will be reflected. The solid lines are incident waves and the dashed lines are reflected waves.

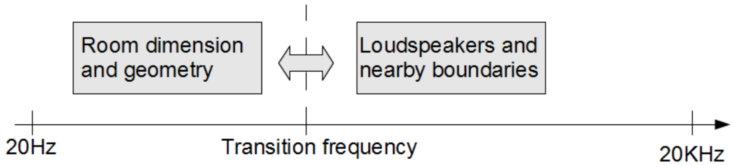


Figure 5.9 – Illustration of which factors that dominate the behavior of sound, radiated from loudspeakers, in a room, depends on frequency.

Since the room is a closed enclosure, standing waves will occur and give rise to resonant frequencies also known as room modes, explained in section 5.2.3. These modes depend on the room geometry and size but also the room boundary specific acoustic impedances. Placing the loudspeaker in an antinode which could be near a large rigid boundary like a wall, the mode frequency will be boosted, but placing the loudspeaker in a node it is opposite not possible for the loudspeaker to excite the mode frequency. The distributions of the room modes depend on the ratios between length, height and width of the room and the more evenly distributed the modes, the better because it will avoid frequency spans with peaks or dips cluttered together. From [IEC 60268-13] it is proposed that equation (5.24) and additional equation (5.25) and (5.26) are met to obtain evenly distributed room modes. L[m] is the length of the room, W[m] is the width and H[m] is the height.

$$\frac{W}{H} \leq \frac{L}{H} \leq 4.5\frac{W}{H} - 4 \tag{5.24}$$

$$\frac{L}{H} \leq 3 \tag{5.25}$$

$$\frac{W}{H} \leq 3 \tag{5.26}$$

The higher the frequency, the more room modes and they hereby cancel each other out. The behavior of sound in a room can therefore be divided into two frequency range divided by the transition frequency. Below the transition frequency, the room modes and hereby the room geometry and dimensions will be the dominating factors. Above the transition frequency, the reflections and hereby the loudspeaker radiation characteristics and the specific acoustic impedance of nearby boundaries will be dominating. See **Figure 5.9**. In summary, the following physical behaviors will affect the sound at listener position:

- Loudspeaker radiation characteristics.
- Boundary specific acoustic impedances (Radiation characteristics of reflected sound waves).
- Boundary texture (Scattering).
- Room geometry and size.
- Placement of loudspeakers and listener position.

5.4.2 PERCEPTUAL

When it comes to perception of sound in a room it is important to mention some properties of the human sound perception which can deal with some of the physical behavior of sound in a room. These properties are the ability to determine the direction of sound due to binaural hearing and the ability to distinguish between one or several sound events. The ability to distinguish between one or several sound events depends on the time interval between the events. This is approximately 30ms but varies among people. A sound presented twice with a delay longer than 30ms between the sounds will be perceived as two individual events but if the delay is less than 30ms the two presentations of the sound will be perceived as one sound event. If the two sounds are radiated from two different positions, the he first arriving sound will also dominate the impression of where the sound is coming from. This behavior is called the predance effect and applies to delays less than 30ms, also called early reflections [Toole, 2008] [Moore, 2012]. If the delay becomes smaller than 0.6-1ms a behavior called summing location will occur. The impression of where the sound is coming from will in this case be somewhere between the two sound sources, depends on the delay time.

Comparing this knowledge to the physical behavior of sound in a room radiated from a loudspeaker section 5.4.1 and the knowledge of the stereo setup section 5.3.2, the first order reflections will not be heard as individual events but perceived together with the direct sound due to the predance effect. This means that regardless the direction of the reflections, the impression of where the sound is coming from will be the loudspeaker positions. 30ms correspond to 10.3m when the speed of sound is 343m/s and the reflections therefore have to travel 10.3m longer than the direct sound to be perceived as individual events. If the reflected sound is less than 20.5 to 34.3cm longer than the direct sound, summing location will occur and the impression of where the sound is coming from will be somewhere between the direct sound sources and a mirror source for the reflected sound. 20.5 to 34.3cm corresponds to the delay of 0.6-1ms when the speed of sound is 343m/s. Even though the direct sounds and the reflected sounds are perceived as one event it is not without issues. Comp filtering will occur and this is perceived as timbre changes. Timbre is from the American standard association defined as [wiki3]: *"..that attribute of sensation in terms of which a listener can judge that two sounds having the same loudness and pitch are dissimilar".* To gain knowledge about timbre changes due to reflections, [Bech, 1994] and [Bech, 1996] investigated this by having subjects listening to noise and speech in an anechoic environment where 1. order reflections and the impression of a room was generated using several loudspeakers positioned around the listening position. The delays between the direct sound and the reflected sounds were adjusted from 1.64 to 14.98ms which correspond to the delays in a chosen room, and the loudspeakers positioned at the angles from where the reflections are radiated. Also the reflected sounds were filtered accordingly to the boundaries from where they are radiated using the absorption coefficients and incident angles. After 22ms, reverberation was generated to improve the impression of a real room. By varying the sound pressure levels (SPL) of the reflections, thresholds of detection (TD) and just noticeable differences (JND) were detected. It was found that the longer delay between the direct sound and reflected sound, the larger JND and the lower TD. When the delay is 1.49ms, the JND for the noise stimuli is approximately 2dB and the TD is approximately 9dB lower than the direct sound SPL and when the delay is 14.98ms the JND is approximately 6dB and the TD is approximately 5dB lower than the direct sound SPL. By fitting a linear curve, equation (5.27), to the TD results and comparing this curve with the distance law, which is 6dB damping when the distance is doubled, a critical distance can be calculated. This critical distance is the length of a reflected sound wave where the level of this wave is at threshold level. This means that reflected sound waves which travels longer than the critical distance will not influence the timbre. The linear curve is estimated using the TD values where the delays between the direct and reflected sounds are converted to distances. 1.49ms corresponds to 0.51m and 14.98ms corresponds to 5.13m when the speed of sound is 343m/s. However, some of the results from [Bech, 1996] will not fit perfectly to this curve and it is therefore just a rough estimation. TD is the threshold of detection at distance x_r[m] for the reflected sound.

$$TD = 0.86x_r - 9.4 \tag{5.27}$$

$$L = 20\log\left(\frac{x_d}{x_r}\right) \tag{5.28}$$

Setting equation (5.27), the linear estimation and equation (5.28), the distance law equal each other, equation (5.29), the distance, x_r[m], now becomes the critical distance and can be calculated when the distance for the direct sound, x_d[m] is known.

$$0.88x_r - 9.4 = 20\log\left(\frac{x_d}{x_r}\right) \tag{5.29}$$

The minimum distance for the direct sound following the guidelines from [IEC 60268-13] described in section 5.3.2, is 2.7m. At this distance the critical distance for the reflected sound become 5.81m. Using trigonometry, the guidelines from [IEC 60268-13] and the calculated critical distance, critical distances for nearby boundaries relative to the loudspeaker position can then be calculated. See **Figure 5.10**. Total reflection, R=1, and no scattering are assumed. The incident sound wave angles are therefore equal the reflected sound wave angles.

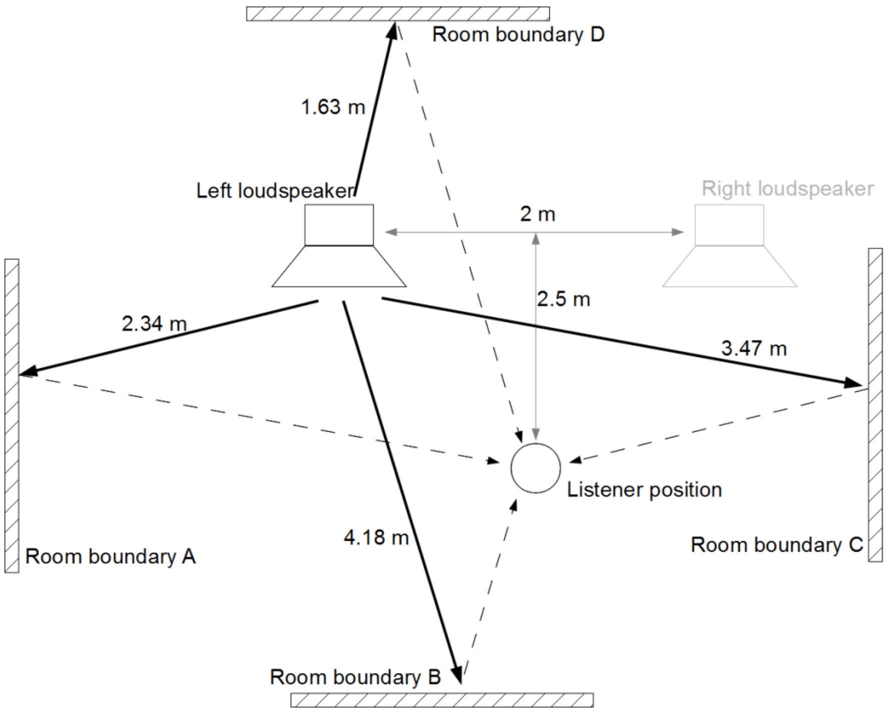


Figure 5.10 – Top view illustration of critical distances between left loudspeakers and boundaries following the stereo setup in [IEC 60268-13]. Boundaries within the distances will contribute to timbre changes. A longer distance between the loudspeaker and listener position will increase the critical distances.

In **Figure 5.10**, critical distances for only 4 room boundaries seen from top view are showed. However the critical distance will exist at any angles around the loudspeaker and can be explained using an ellipse with foci at the listener position and the loudspeaker position. In **Figure 5.11**, a critical distance is showed for a room boundary from side view. This room boundary can be the floor or ceiling and comparing with the guidelines from [IEC 60268-13], the floor and ceiling will always contribute timbre changes. This confirms with the results from [Bech, 1996]. When the distance between the loudspeaker and listener position is 2.7m, the listener and loudspeaker shall be positioned minimum 2.6m from the ceiling and floor to have reflections below the TD and therefore avoid change of timbre. This is not possible under normal circumstances.

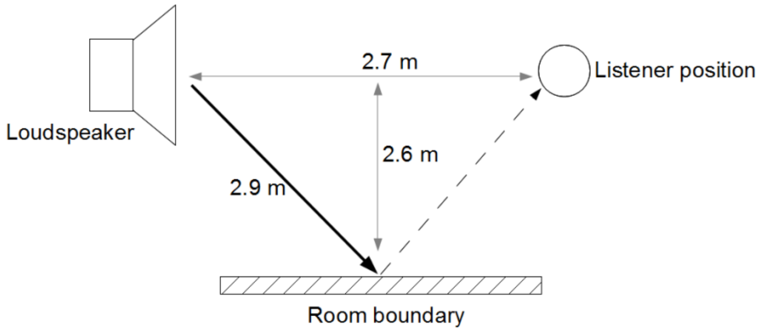


Figure 5.11 – Side view illustration of critical distances between loudspeakers and boundaries following the stereo setup in [IEC 60268-13]. It is assumed that the listener and loudspeaker height are equal. A longer distance between the loudspeaker and listener position will increase the critical distances.

The results and critical distances between room boundaries and loudspeaker obtained and showed in **Figure 5.10** and **Figure 5.11**, are mainly based on our ability to distinguish between sound events and the predance effect and is therefore not the total truth. Since our hearing is binaural, the direction, relative to the listener position, from where the reflected sounds are coming from, will also influence our perception of sound. [Toole, 2008] discovered, using a similar setup as [Bech, 1996], that we are less sensitive to reflections radiated from the same direction as the direct sound. However, when the reflected sound wave is radiated from the same direction as the direct sound, the overall impression of the perceived sound is degrading. This degrading was discovered while the impression of a sound with reflections radiated from respectively the same loudspeaker, a wall and another loudspeaker, was investigated [Toole, 2008]. The 3 different reflections were adjusted to have equal waterfall plots (time, frequency and relative level) as possible and then presented for different subjects who evaluated their impression of the sound. For the sound with reflections generated by the same loudspeaker, the overall impression was degrading, for the wall reflection, the change of impression was minimal and for the reflections generated with another loudspeaker, the change of impression was moderate to pleasing. This lead to that reflections actually can have positive effects on the impression of sound in a room. From different investigations it is discovered that the reflections in a room combined with the binaural hearing and higher cognitive processes will lead to a smoothing effect called spectral smoothing [Toole, 2008]. Due to this effect we become less sensitive to resonances, and timbre changes due to comp filtering will therefore be less problematic. Spectral smoothing occurs even when the delayed sound is 30 to 40 dB below the direct sound. Comparing the knowledge from this section lead to that reflections radiated nearby the loudspeakers are problematic but how problematic depends on other reflections in the room. In highly damped rooms with almost no reflections and low reverberation times, the spectral smoothing effect will be low and timbre changes due to reflections nearby the loudspeakers become audible. This is the situation in recording studios where critical listening to the audio material is needed. The low reverberation time and lack of reflections is however not preferred when listening to and enjoying music in e.g. a living room. This lead to that the radiation characteristics from loudspeakers combined with boundary characteristics will influence how the spectral smoothing will behave. The reflections are generated by sound radiated from all axes from the loudspeakers and the frequency response of those reflections will therefore depends on the radiation characteristics and boundary characteristics. [Evans, 2012] have an equal conclusion. The overall impression of sound from loudspeakers depends on the loudspeaker radiation characteristics and room boundary characteristics. In summary, the following perceptual effects will affect the perceived sound radiated from loudspeakers in a stereo setup in a living room:

- The predance effect. Reflections nearby loudspeakers will contribute to timbre changes.
- Spectral smoothing. Reflections far from the loudspeakers and reverberation will smooth resonances and timbre changes.

5.5 Analysis of characteristics

Combining the theory from section 5.2, 5.3 and 5.4, it can be concluded that the following characteristics can be seen as room boundary characteristics. Together these characteristics are able to explain how a boundary will influence the sound.

1. Specific acoustic impedances
2. Distances
3. Temperature
4. Humidity
5. Pressure

Assuming a standard stereo setup explained in section 5.3.2, the sound pressure at listener position will be the superposition of direct and reflected sound waves and the distances between loudspeakers, boundaries and listening position will therefore determine the frequencies which are affected. The temperature will also influence the frequencies since the speed of sound depends on that. The level at the different frequencies will be influenced by all characteristics. The specific acoustic impedances and weather conditions (temperature, humidity and pressure) will influence the reflection factor and thereby the level and the distance will influence the level due to the distance law. See Figure 5.12. Henceforth the specific acoustic impedance will be referred as impedance.

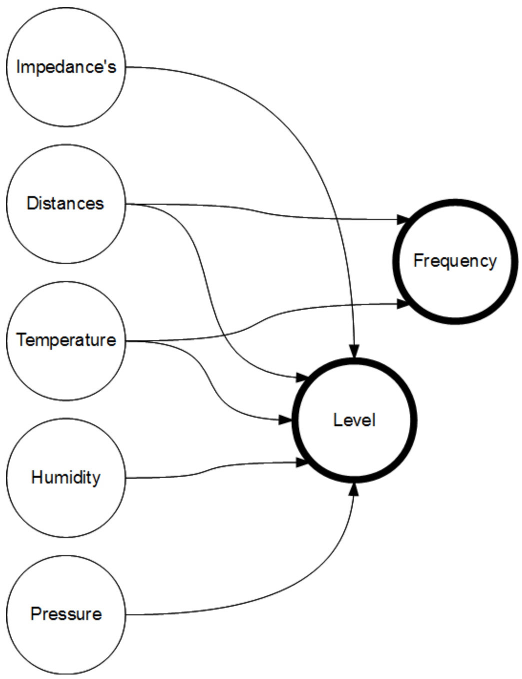


Figure 5.12 – Illustration showing which characteristics that influences the frequency and level using the theory from section 5.2.

Boundary surface texture can also be characterized as a boundary characteristic since it will cause scattering of the incident sound wave as described in section 5.4.1. The investigated frequencies are however below 1KHz which correspond to a wavelength of 2.9cm and since walls normally have texture or bumps less than this size, the boundary surface texture is not taken into account. The reflected sound wave angle will be equal the incident angle. The weather conditions are expected to have minor influences on the sound in the room, but how small are not known. Also it is not known if the distances or the impedances have greatest influence on the sound. Therefore, an analysis of the weightings between the characteristics is needed.

5.5.1 WEIGHTING BETWEEN THE CHARACTERISTICS

To analyze and get the weightings between the characteristics, the idea is to observe the frequency response of a sound wave reflected from a boundary while changing the different characteristics within their normal ranges in a living room. The chosen method to do that is simulation which allows the characteristics to be easily changed. If the analysis is based on measurements in a real room, especially the weather conditions will be hard to change and control. Also reflections from the boundaries in a real room will be uncontrollable and the obtained results will maybe not be generic enough. The simulation can however also be too simple and therefore not fit reality. In the simulation it is chosen to have three plane sound waves, an incident sound wave with normal incident angle to a boundary, a reflected sound wave and a direct sound wave. In other words it is a simulation of a 1.order reflection. See **Figure 5.14**. The incident and direct sound waves are generated by a sound source and are radiated respectively to a boundary and a receiver position. The incident sound wave will be reflected at the boundary and thereby it becomes the reflected sound wave which is reflected back to the receiver position. The receiver will therefore receive the reflected sound wave and the direct sound wave which summed will have some level variations depends on frequency, and the room boundary characteristics. The sound field in the simulation is free-field and the only reflecting surface is therefore the boundary. For the sake of convenience the boundary is assumed to be rigid, solid and flat which means that the mass law, bending waves, transversal shear waves, scattering and other physic phenomena for a real boundaries is ignored. Only the impedances and hereby the reflection factor, described in section 5.2.5, will influence how much of the incident sound wave is reflected. The sound wave transmitted through the boundary is also ignored since it will not influence the sound at receiver position. The distance L1[m] is the distance between boundary and sound source and the distance L2[m] is the distance between sound source and receiver. During the simulation, the distance L2[m] is fixed to 1m. Since the distances are short and the frequencies below 1KHz, the damping in air caused by humidity is ignored. Only the 6dB damping due to distance doubling is taking into account. Equation (5.30) explains the calculation of the pressure offset at receiver position, p_{offset}[Pa], with use of super position. The reflected sound wave, which is delayed with L_1[m]/c[m/s] seconds and multiplied by the reflection factor, R, and L_2[m]/L_1[m]due to the distance law, is added to the direct sound wave. A[Pa] is the amplitude for the direct sound wave at receiver position, t[s] is time, ω[rad/s] is angular frequency and c[m/s] is speed of sound in air.

$$p_{offset} = Ae^{j\omega t} + Ae^{j\omega(t-L_1/c)} \cdot R \cdot \frac{L_2}{L_1} \tag{5.30}$$

The reflection factor, R, and speed of sound in air, c[m/s], are determined by the impedances and weather conditions explained in section 5.2.8 and **Figure 5.13**. The pressure offset is converted to a level offset in dB using equation (5.31).

$$L_{offset} = 20\log(p_{offset}) \tag{5.31}$$

To find the highest possible level change for each of the characteristics, the simulation sweeps the characteristics within their normal range in the frequency range from 20Hz to 1KHz, one by one while the other characteristics are fixed. The normal ranges (maximum and minimum values) for the different characteristics used in the simulation are available in **Table 5.1**. These values are estimations of the possible ranges of the characteristics in a living room. The impedances varies from 415Pa·s/m to infinity which corresponds respectively to the impedance of air at 20^{o}C and an infinite-rigid, solid boundary. The air is chosen as minimum because it could correspond to an open window or door which will lead to total transmission of the incident sound wave and therefore no reflected sound wave. The distance L1[m] varies from 0 to 1m which is estimated to be the distance range where loudspeakers are normally positioned from a boundary. This distance range is also estimated to be important from a perceptual point of view section 5.4.2. The temperature, humidity and pressure are from [IEC 60268-13] and fit very well the weather conditions in Denmark [DMI].

Characteristic	Minimum value	Maximum value
Impedance boundary [Pa·m/s]	415	∞
Distance L1 [m]	0	1
Temperature [°C]	18	27
Humidity[%]	25	75
Pressure [Pa]	86e3	106e3

Table 5.1 – Minimum and maximum values for the characteristic used in the simulation.

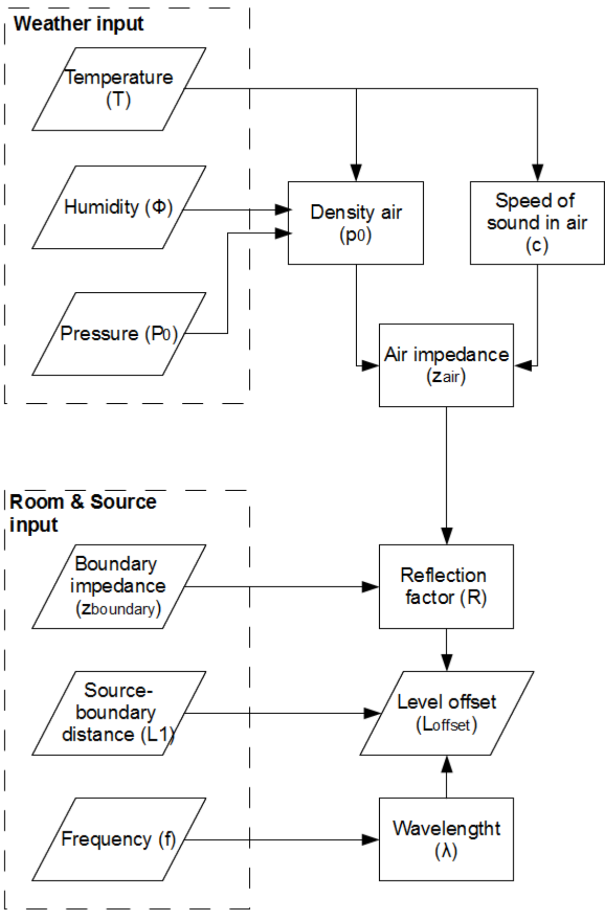


Figure 5.13 – Flowchart of the level offset calculations used in the simulation. The inputs are varied while change of level offset is observed.

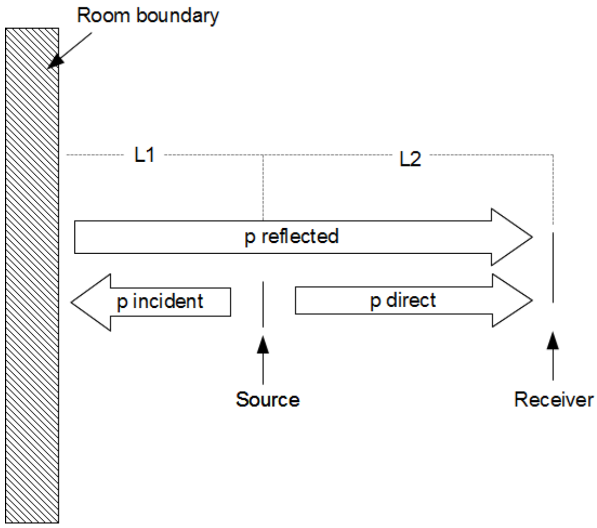


Figure 5.14 –Illustration of the simulated setup. The source is generating a direct and incident sound wave. The incident sound wave is reflected at the room boundary and becomes a reflected sound wave which is received by the receiver together with the direct sound wave.

In **Figure 5.15**, the result of the simulation *DVD\matlab code\SingleWallSimulation.m* is showed. For every characteristic, the maximum level change is plotted versus the frequency and it is obvious that the higher the frequency, the higher possible level change for the characteristics impedance and distance. When the frequency is increasing, the wavelength is decreasing and therefore the needed distance L1[m], to obtain a reflected sound wave with reverse phase compared to the direct sound wave, becomes smaller. The shorter distance L1[m], the less damping of the reflected sound wave due to distance and therefore greater damping at the receiver position. The knee point at 85Hz is due to the limitation of the distance, L1[m], to 1m. For frequencies below 85Hz the maximum level change will therefore occur at the limit 1m and above 85Hz the maximum level change will occur at distance, L1[m], smaller than 1m. The exact distance depends on the frequency. The higher the frequency, the smaller the distance. The impedance are equal the distance because the chosen maximum and minimum impedances for the simulation corresponds respectively to reflection factors of 1 and 0. When the reflection factor is equal 0, no reflections exist and the maximum level change caused by the distance is canceled out. The weather conditions have almost no influence on the sound and since they are below 1dB, which is the SPL JND [Moore, 2012] the level changes due to weather conditions are not audible.

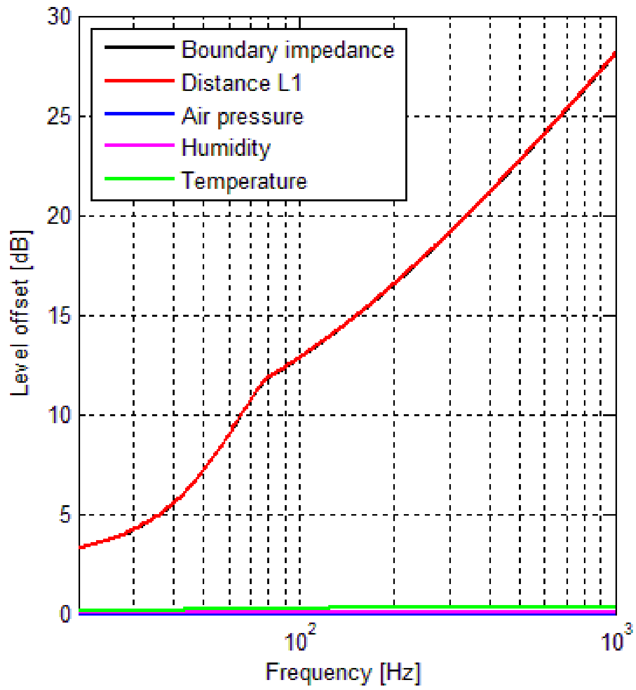


Figure 5.15 – Possible level offset in dB caused by the characteristics within their normal range. The boundary impedance, black line, is hidden behind the distance L1, red line.

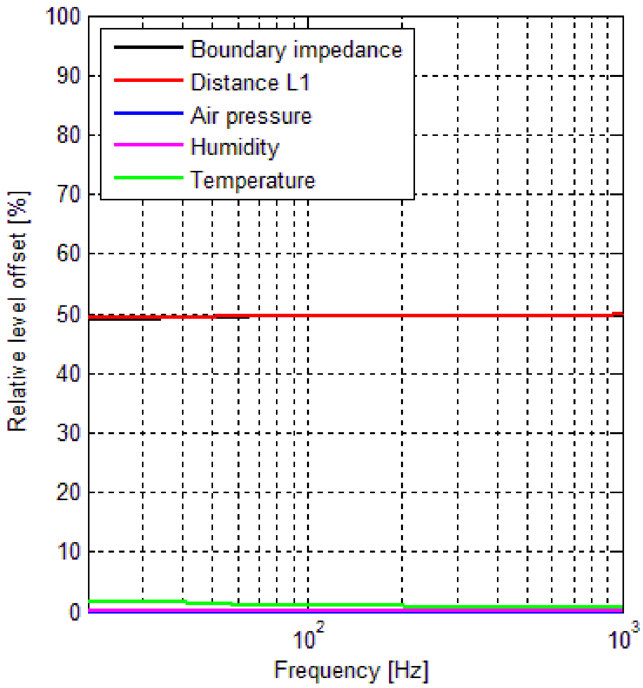


Figure 5.16 – Relative level offset in % caused by the characteristics within their normal range. The boundary impedance, black line, is hidden behind the distance L1, red line.

In **Figure 5.16**, the level influence is normalized to percent and the impedance and distance are showed to have equal weighting in the simulated frequency range. They are therefore both needed to predict how a boundary will influence the sound while the weather conditions can be ignored. From **Figure 5.15** it can also be concluded that the higher the frequency the bigger influence relative to sound pressure level. This fit very well the theory about transition frequency in the room, section 5.4.1, where the frequencies below the transition frequency is dominated by room modes and frequencies above the transition frequency is dominated by the reflections.

5.6 MEASUREMENTS OF THE CHARACTERISTICS

From section 5.5.1, the weather conditions, in their normal range in living rooms, was found to have almost no influence on the sound reflected from a room boundary. Measurements of the weather conditions are also straight forward and there is therefore no need to investigate and describe the weather conditions and how to measure them further. Measurement of distances can be performed using several methods. One method is echo localization but it is simpler to use a laser distance meter or a ruler. The measurements of distances are therefore not a problem and will likewise not be discussed further. What is left behind and harder to measure is the impedance.

5.6.1 MEASUREMENTS OF IMPEDANCES

A lot of different methods have been developed to measure impedances and common for those are that they can be divided into a laboratory group and an in-situ group. For the laboratory group the test specimen have to be brought to a laboratory to perform the measurement and for in-situ group the measurements can be performed in-situ without removing the specimen from its original position. The laboratory methods are good to compare different materials but since the laboratory methods using specific test specimen sizes, the measured impedances may not be equal the impedances for materials at their original position. E.g. a wooden door will have a resonance frequency determined by the door dimensions and material and therefore behave as a resonance absorber at some frequencies. This resonance will not exist if a part of the door is mounted in the impedance tube when using the standing wave or transfer function method and the impedance will therefore be different than if the impedance was measured for the entire door. In **Table 5.2** an overview of the investigated measurements method are given. These methods are described further in next sections, section 5.6.1.1 to 5.6.1.9. The interesting methods, project point of view, are the in-situ measurements but the laboratory methods are also investigated for completeness. Some of the laboratory and in-situ measurements are related too.

Method	Laboratory / in-situ	Complex Z	Incident angle
Reverberation room [ISO 354]	Laboratory	No	Random incident
Standing wave method in tube[ISO 10534-1]	Laboratory	Yes	Normal incident
Transfer function method in tube [ISO 10534-2]	Laboratory	Yes	Normal incident
Transfer function method	In-situ	Yes	Normal and oblique incident
Subtraction method	In-situ	Yes	Normal and oblique incident
Intensity method	In-situ	No	Normal and oblique incident
Two microphones and ambient noise method	In-situ	Yes	Normal and oblique incident
Inverse method	In-situ	Yes	Normal incident
Method using microflown sensor	In-situ	Yes	Normal and oblique incident

Table 5.2 – Overview of the investigated impedance measurement methods.

5.6.1.1 REVERBERATION ROOM

The reverberation room method is a laboratory method based on reverberation time measurements in a reverberation room and Sabine's equation for reverberation time equation (5.32). T[s] is the reverberation time, V[m^3] the volume of the room and A[m^2] is the total absorption [ISO 354] [Toole, 2008].

$$T = 0.161\frac{V}{A} \tag{5.32}$$

The total absorption, A[m^2], is calculated using equation (5.33) where S[m^2] is the surface area for surface i and α is the absorption coefficient for the surface.

$$A = \sum_i S_i \alpha_i \tag{5.33}$$

By measuring the reverberation times in the reverberation room with and without the test specimen and knowing the size of the room and specimen, it is possible to calculate the absorption coefficient, α, for the specimen at random incident. It is random because the reverberation room sound field is assumed to be diffuse. From section 5.2.6 it is proved that the absorption coefficient can be calculated from the impedance, however the phase of the impedance is lost in that calculation. It is therefore not possible to calculate the true impedance using the reverberation method. Other downsides are that the method needs quite big samples of the specimen (10-12m^2), the reverberation room is a big and highly specialized room and Sabine's reverberation equation assumes diffuse field. The bigger and more absorbent the test specimen is, the poorer diffuse field. The measurement conditions will therefore change depending on the test specimen.

5.6.1.2 Standing wave method in tube

The standing wave method in tube is a laboratory method where the test specimen is mounted at one of the tube openings and in the opposite opening a loudspeaker is mounted to generate sound waves [ISO 10534-1]. See **Figure 5.17**. A standing wave will, due to the reflections from the test specimen, be generated and by measuring the first pressure minima, the pressure maxima between the two first pressure minima and the distance between the test specimen and pressure minima, the impedance can be calculated. The method assumes that the sound waves in the tube are plane waves, therefore equation (5.34) shall be satisfied. This equation also ensures that the wavelengths are short enough to have two pressure minima inside the tube.

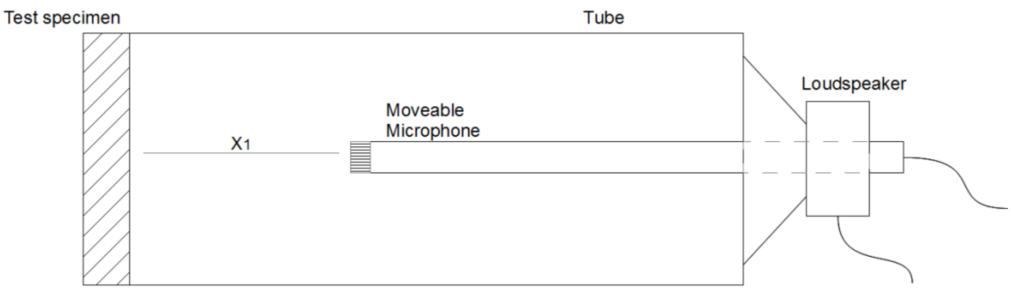


Figure 5.17 – Illustration of the standing wave method in tube measurement setup. The microphone is moveable to detect pressure minima and maxima at distance X1.

$$f_l < f < f_u \tag{5.34}$$

f[Hz] is the measurement frequency, f_l[Hz] is the lower frequency limit and f_u[Hz] is the upper frequency limit. The upper and lower limits are determined using equation (5.35) and (5.36) where d[m] is the tube diameter and l[m] is the tube length.

$$f_l = \frac{-250}{3 \cdot d \cdot l} \tag{5.35}$$

$$f_u = \frac{200}{d} \tag{5.36}$$

When the pressures and the distance between test specimen and pressure minima are known, the impedance can be calculated. First the standing wave ratio, s, is calculated using pressure minima, p_{min}[Pa], and pressure maxima, p_{max}[Pa].

$$s = \frac{|p_{max}|}{|p_{min}|} \tag{5.37}$$

Then the absolute reflection factor is calculated using the standing wave ratio.

$$|R| = \frac{s-1}{s+1} \tag{5.38}$$

The phase angle, θ[rad], is calculated using the distance between pressure minima and test specimen, x_1[m], and the wavelength, λ[m].

$$\theta = \pi\left(\frac{4X_1}{\lambda} - 1\right) \tag{5.39}$$

Using the absolute reflection factor and the phase angle, the complex reflection factor, R, is calculated.

$$R = |R| \cdot e^{j\theta} \tag{5.40}$$

The impedance can then be calculated using equation (5.9) in section 5.2.5. Downsides for the method are time-consuming measurements, limited frequency range and the only possible incident angle is normal incident. However the method is useful when comparing different materials.

5.6.1.3 Transfer function method in tube

The transfer function method in tube is a laboratory method and like the standing wave method, the test specimen is mounted in one of the tube openings and the sound waves are generated by a loudspeaker mounted in the opposite opening. The method using two microphones mounted with a certain distance and by measuring the transfer functions between the microphones it is possible to calculate the impedance [ISO 10534-2]. See Figure 5.18.

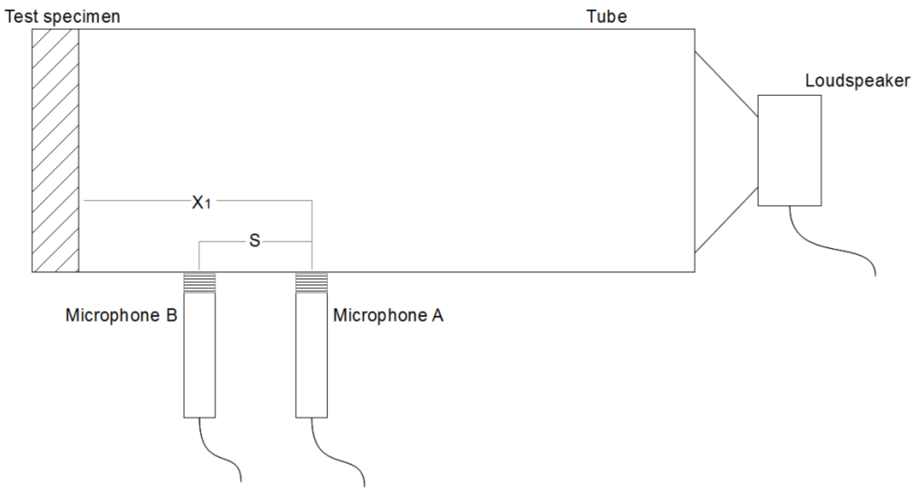


Figure 5.18 – Illustration of the transfer function method in tube measurement setup.

The upper frequency limit is limited by the tube diameter like the standing wave method, equation (5.36) and the spacing, S[m], between the microphones, equation (5.41). The lower frequency limit is limited by the accuracy of the signal processing equipment.

$$f_u < \frac{154}{S} \tag{5.41}$$

When the upper frequency limits are satisfied the sound wave inside the tube is assumed to be a plane wave. The pressure for the incident and the reflected sound wave can therefore be defined as in equation (5.42) and (5.43), where $\hat{p}$[Pa] is the magnitude, k is the wavenumber and x[m] is the distance between test specimen and measurement position.

$$p_I = \hat{p}_I e^{jkx} \tag{5.42}$$

$$p_R = \hat{p}_R e^{-jkx} \tag{5.43}$$

Using the definition for the incident and reflected pressure, the pressure at microphone position A, p_a[Pa], and B, p_b[Pa], are calculated using equation (5.44) and (5.45) where x_a[m] and x_b[m] are respectively the distance to microphone A and B.

$$p_a = \hat{p}_I e^{jkx_a} + \hat{p}_R e^{-jkx_a} \tag{5.44}$$

$$p_b = \hat{p}_I e^{jkx_b} + \hat{p}_R e^{-jkx_b} \tag{5.45}$$

The transfer functions for the incident and reflected pressure is calculated in equation (5.46) and (5.47) where s[m] is the spacing between microphone A and B.

$$H_I = \frac{p_{bI}}{p_{aI}} = e^{-jk(x_a - x_b)} = e^{-jks} \tag{5.46}$$

$$H_R = \frac{p_{bR}}{p_{aR}} = e^{jk(x_a - x_b)} = e^{jks} \tag{5.47}$$

Dividing equation (5.45) with (5.44) to get the transfer function from microphone A to B and combining with the reflection factor, R, equation (5.10), leads to equation (5.48).

$$H_{ab} = \frac{p_b}{p_a} = \frac{e^{jkx_b} + Re^{-jkx_b}}{e^{jkx_a} + Re^{-jkx_a}} \tag{5.48}$$

The reflection factor is isolated in equation (5.49) where x_1 is the distance between microphone A and test specimen as illustrated in Figure 5.18.

$$R = \frac{H_{ab} - H_I}{H_R - H_{ab}} e^{2jkx_1} \tag{5.49}$$

The impedance can then be calculated using equation (5.9) in section 5.2.5. Compared with the standing wave method, this method is faster and it is possible to measure impedances at lower frequencies, using the same tube dimensions. Downsides are that the method has heavier computations and the test specimen still needs to be bought to the laboratory in a specific sample size. An in-situ variation of this measurement method however exists, where the tube is modified to be transportable. This is described in [ISO 13472-2] for road impedance measurements.

5.6.1.4 Transfer function method

The transfer function method is an in-situ method where the transfer function is measured between a sound source and a receiver [Mellers & Nocke, 2004]. See Figure 5.19. By applying time windows to the measured signal, Figure 5.20, it is possible to isolate the direct sound and the reflected sound and hereby calculate the reflection factor from which the impedance can be calculated following the theory from section 5.2.5. If there are any parasite reflections they are also leaved out due to the time windowing. This method is used in the [ISO 13472-1] for road measurements and can be performed for both normal and oblique incident angles. However when the incident angle is oblique, the reflected and direct sound wave is radiated from two different angles from the loudspeaker. The frequency responses for these angles need to be equal to obtain valid results.

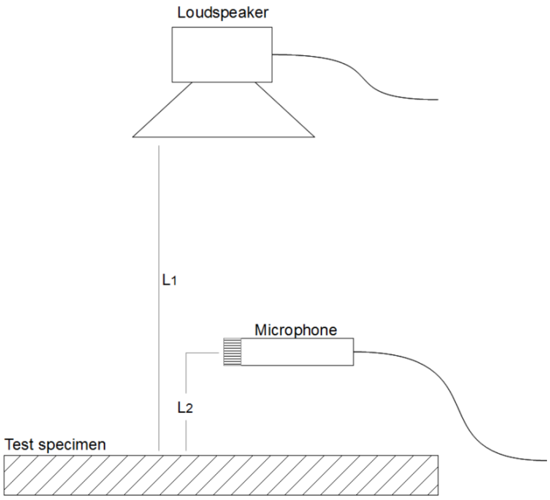


Figure 5.19 – Illustration of the in situ transfer function method measurement setup.

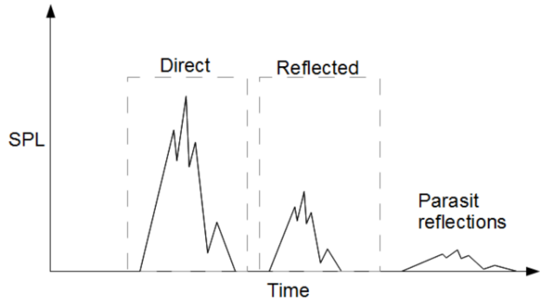


Figure 5.20 – Illustrated impulse response measured with the transfer function method with applied time windows on the direct and reflected sound.

5.6.1.5 Subtraction method

The subtraction method is an in-situ method using a sound source and a microphone. To obtain the impedance for the test specimen, two impulse response measurements are carried out. One with the microphone positioned close to the test specimen and one without the test specimen [Mellers & Nocke, 2004]. The last measurement serves as a reference and is subtracted from the first measurement. Thereby the complex reflection factor and the impedance for the test specimen can be calculated. If parasite reflections exist, they are removed using time windowing.

5.6.1.6 Intensity method

The intensity method is a method using concepts of sound intensity and energy density to calculate the absorption coefficient [Farina & Torelli, 1997]. Since it calculates the absorption coefficient it is not possible calculate the complex impedance with this method following the theory from section 5.2.6. The method is measuring the intensity using a 3d sound intensity probe and it is hereby possible to calculate the reflected and incident intensity which is needed to calculate the absorption coefficient, equation (5.50), where I_r[w/m^2] is the reflected intensity and I_i[w/m^2] is the incident intensity.

$$\alpha = 1 - \frac{|I_r|}{|I_i|} \tag{5.50}$$

Other intensity methods exist and rely on the use of the intensity probe as an impedance meter. In these cases the impedance is obtained by the sound pressure and particle velocity but should be less robust than the described intensity method.

5.6.1.7 Two microphones and ambient noise method

The two microphones and ambient noise method is an in-situ measurement method which calculate the impedance using the transfer functions measured between two microphone positions with help from ambient noise[Takahashi, 2004]. See Figure 5.21. The ambient noise is assumed to be EA-noise (environmental anonymous noise) which indicates that the noise is not exciting distinct room modes and the incident sound waves are assumed to be plane waves with random incident angles. These condition may be hard to achieve in practice [Nava,Yasuda, Sato & Sakamoto, 2009] but are however almost equal the assumptions for the sound wave in the transfer function method in tube. Due to this and the equal use of the two microphones in both the two microphones and ambient noise method and transfer function method in tube, the math behind the methods is equal. They are however derived differently.

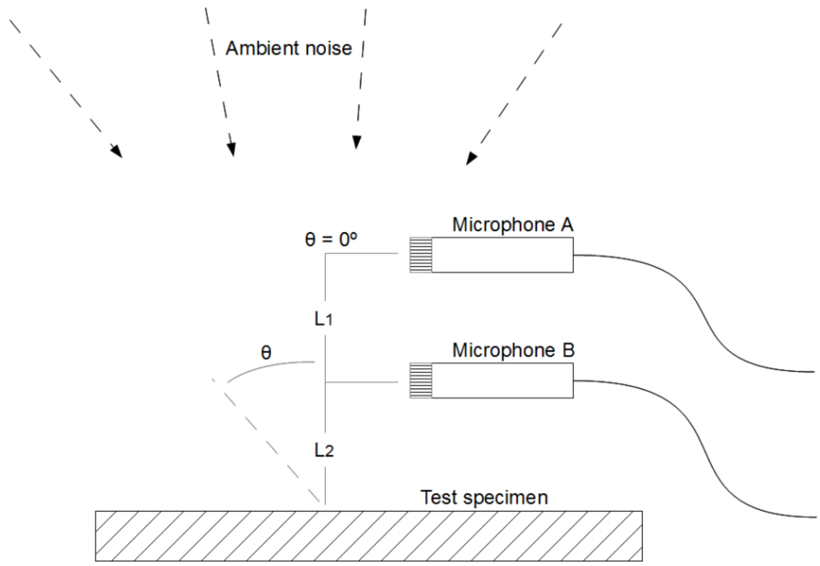


Figure 5.21 – Illustration of the two microphones and ambient noise method measurement setup.

Using the definition for incident and reflected plane waves, equation (5.42) and (5.43), the pressure at microphone A, p_a[Pa], and B, p_b[Pa], are calculated like the transfer function method in tube, in equation (5.51) and (5.52). L_1[m] and L_2[m] are the distances between microphones and test specimen, k is the wave number and θ[rad] is measurement angle. 0rad corresponds to normal incident.

$$p_a = p_i e^{jk(L_1+L_2)\cos\theta} + p_r e^{-jk(L_1+L_2)\cos\theta} \tag{5.51}$$

$$p_b = p_i e^{jkL_2\cos\theta} + p_r e^{-jkL_2\cos\theta} \tag{5.52}$$

Then the reflection factor is calculated using the pressure for the incident and reflected sound waves.

$$R = \frac{p_r}{p_i} = \frac{p_a - p_b e^{jkL_1 cos\theta}}{p_b e^{-jkL_1 cos\theta} - p_a} e^{2jkL_2\cos\theta} \tag{5.53}$$

From equation (5.9) in section 5.2.5, the impedance is then calculated.

$$z = \frac{\rho_0 c}{cos\theta} \frac{1+R}{1-R} \tag{5.54}$$

Having the transfer function from microphone A to B the impedance is calculated as in equation (5.55).

$$z = \frac{\rho_0 c}{cos\theta} \frac{H_{ab}\left(1 - e^{2jk(L_1+L_2)cos\theta}\right) - e^{jkL_1 cos\theta}\left(1 - e^{2jkL_2 cos\theta}\right)}{H_{ab}\left(1 + e^{2jk(L_1+L_2)cos\theta}\right) - e^{jkL_1 cos\theta}\left(1 + e^{2jkL_2 cos\theta}\right)} \tag{5.55}$$

And at normal incident the expression cosθ is equal 1 and is removed from equation (5.56).

$$z = \rho_0 c \frac{H_{ab}\left(1 - e^{2jk(L_1+L_2)}\right) - e^{jkL_1}\left(1 - e^{2jkL_2}\right)}{H_{ab}\left(1 + e^{2jk(L_1+L_2)}\right) - e^{jkL_1}\left(1 + e^{2jkL_2}\right)} \tag{5.56}$$

5.6.1.8 THE INVERSE METHOD

The inverse method is an in-situ measurement method able to measure the impedance of the boundaries in a room with use of the room geometry, sound pressure at different arbitrary points in the room and the sound source strength [Nava,Yasuda, Sato & Sakamoto, 2009][Piechowicz, 2011]. When these parameters are known, the impedances for the different boundaries can be calculated using inverse boundary element method (BEM) which is a computational method of solving linear partial differential equations. See Figure 5.22 where different microphone positions are used to measure the pressure at the positions and the loudspeaker is used as the sound source.

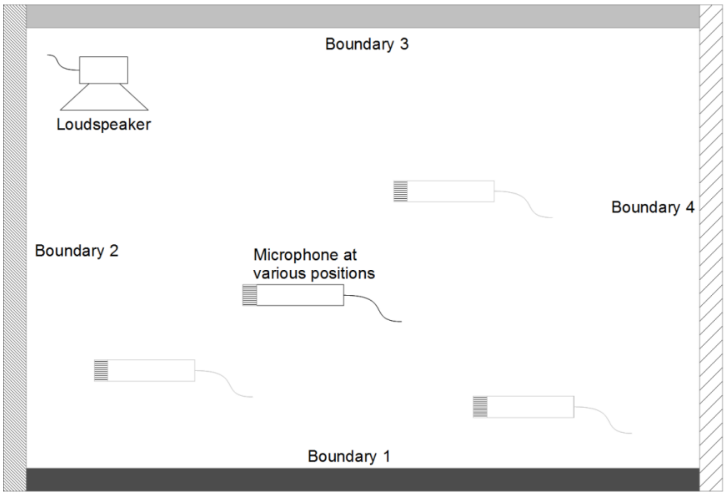


Figure 5.22 – Illustration of the inverse method measurement setup.

5.6.1.9 METHOD USING MICROFLOWN SENSOR

The method using a microflown sensor is based on a new sensor type called a microflown which measures the particle velocity [Tijs, Botts, Bree & Arato, 2009]. The sensor is based on two heated strings and the velocity is measured by measuring the temperature difference on those strings. When an air particle is traveling with a certain speed and passing the strings, the temperatures will change accordingly and the velocity can therefore be measured [Microflown]. Combining the sensor with a microphone it becomes a PU (pressure and velocity) probe and the impedance can therefore be measured and calculated using equation (5.3) from section 5.2.4. $z = p/u$. The total impedance measurement setup is almost equal the transfer function method in **Figure 5.19** with a sound source but the microphone replaced by the PU probe. **Figure 5.23** shows a handheld in situ measuring setup.

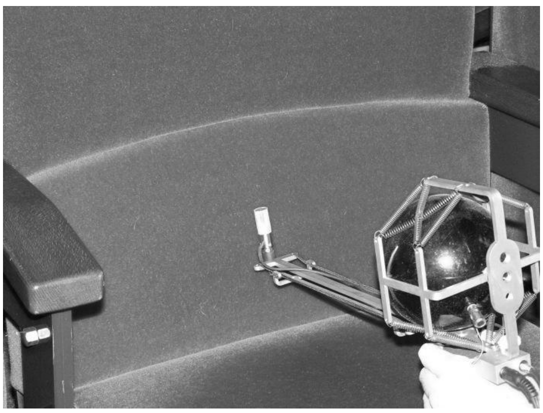


Figure 5.23 – Picture of a handheld measurement setup with the PU probe, measuring the impedance of a chair backrest. The black sphere contains a small loudspeaker used as sound source and the small cylinder, close to the backrest, is the PU probe [Microflown].

6 IMPLEMENTATION

6.1 INTRODUCTION

This section will take care of the second part from the problem formulation namely investigations of methods to measure the needed room boundary characteristics needed for sound coloration compensation with minimum user interaction and equipment as possible. From section 5.5.1, it was found that the impedances and distances are important room boundary characteristics needed to predict how sound is influenced by room boundaries. The distances are however easy to measure and this section will therefore only discuss and analyze measurements of impedances. Firstly a measurement method based on the analysis and theory from section 5.6.1, will be chosen and analyzed. The implementation of the method will be described and measurement using the method will be performed and compared with measurement using the transfer function method in tube. Finally the results will be analyzed and discussed with emphasis on usability for a sound coloration compensation system.

6.2 MEASUREMENT METHOD DISCUSSION

6.2.1 MEASUREMENT METHOD REQUIREMENTS

The measurement method will be based on one of the methods explained in section 5.6.1 and chosen on the basis of the ability to measure the impedance with a minimum user interaction and less equipment as possible seen from a listener/user point of view. The perfect situation will be a measurement system mounted inside the loudspeakers in the stereo setup and able to measure the needed impedances when a button is pushed. Also which room boundaries that are necessary to measure, will be taken into account. From section 5.4.2 it was found that the room boundaries, both horizontally and vertically, within approximately 3m from the loudspeaker, will add unwanted reflections and thereby unwanted coloration of the sound. This means that the floor and ceiling will always be a problem using a standard stereo setup in a living room. However floor and ceiling reflections can be avoided using loudspeaker technologies as line arrays or acoustic lenses, since they are able to change radiation angles and thereby minimize the radiation angle in altitude direction. Vertical room boundaries are therefore seen as the main problem to unwanted reflections. To avoid these reflections, the loudspeaker radiation angle in azimuth direction could be minimized but this will lead to minimized listening area (sweet spot) and removal of, in some cases, wanted sidewall reflections and will in general not be a good solution. It is therefore necessary to measure the impedance of the vertical room boundaries within approximately 1m from the loudspeaker to be able to compensate for the sound coloration using some kind of sound coloration system. In summary the requirements for the measurement method are:

No.	Requirement
1	Minimum user interaction.
2	Minimum use of equipment seen from listener/user point of view.
3	Measurement of vertical room boundaries within 1m from the loudspeaker.
4	Able to be perform the measurements in a living room environment.

Table 6.1 – Measurement method requirements.

6.2.2 CHOSEN MEASUREMENT METHOD

With basis on the requirements, **Table 6.1**, the chosen method is the two microphones and ambient noise method. The method does not require a direct sound source pointing towards the room boundary needed to be measured which is the case for transfer function, subtraction, intensity and microflown sensor method. Instead the method requires ambient noise which is easily obtained in a living room and may already exist due to ventilation systems, traffic noise radiated through windows and doors, noise from computers or other equipment, etc. One advantage using the ambient noise as measurement signal is that the ambient noise is desired and will therefore not decrease the signal to noise ratio as it would do for other measurement methods. The only unwanted noise is noise generated by the measurement system itself. Since the room boundaries, needed to measure, are within 1m from the loudspeaker, item 3, **Table 6.1**, the two microphones and ambient noise method seems promising because a proper position of the two microphones on the loudspeaker may be able to measure these boundaries. The inverse method is able to do the same and even get information about every single room boundary, but several microphone positions are needed, which violates with the requirements item 1 and/or 2, **Table 6.1**.

6.3 MEASUREMENTS INVESTIGATION DISCUSSION

To obtain more knowledge about the two microphones and ambient noise method and how well it fits the requirements stated for the measurement method, **Table 6.1**, some investigations are needed. These are:

No.	Investigation
1	How good are the measurements using the method compared with the transfer function method in tube?
2	What is the possible frequency range for the method?
3	How will changes of microphone spacing, d, and distance to the room boundary (test specimen), l, influence the measurements?
4	Are there any requirements for the ambient noise?
5	Can the ambient noise be generated using the loudspeakers in a stereo setup in case that the existing ambient noise is not sufficient?

Table 6.2 – Measurement method investigations.

It is obvious that the measurements have to be reliable to be used in a sound coloration compensation system and therefore the performance of the measurement method will be investigated. Item 1, **Table 6.2**. The performance will be investigated through comparison of measurement on two different materials with the two microphones and ambient noise method and the transfer function in tube method. The transfer function in tube method is a widely used and exact method and since the two microphones and ambient noise method is based on this method, it seems as a good method to compare the measurements with the two microphones and ambient noise method. The two chosen test specimen are a MDF (medium density fiber) board and a sponge mat. The MDF board is a hard material with a hard surface and will have higher impedance than the sponge mat which is a softer material. The two materials are also well representing materials in different furniture in living rooms.

The two microphones and ambient noise method is only useful if it covers the frequency range between the transition frequency and 20KHz(human hearing upper limit). Therefore the possible frequency range covered by the method will be investigated. Item 2, **Table 6.2**. The project upper frequency limit is however chosen to 1KHz and there will not be much effort in the frequency range between 1 and 20KHz.

Since the perfect mounting of the measurement setup is on the loudspeakers, it is necessary to investigate how the spacing between the 2 microphones and the distances between microphones and test specimen (room boundary) will influence the measurements. Item 3, **Table 6.2**. The results from [Takahashi, 2004], using the 2 microphones and ambient noise method, is obtained with a small distance, 1cm, between the microphones and test specimen and how the measurements will change due to change of distances are not known exactly.

From [Takahashi, 2004], the only explained requirement for the ambient noise is that is shall be diffuse. Requirements about SPL, frequency content and periodicity are not known exactly and it is therefore necessary to investigate the requirements for the ambient noise. Item 4, **Table 6.2**. In cases where the existing ambient noise is not sufficient, it may be generated or supported by noise radiated from the loudspeakers in the stereo setup. The possibility of that will also be investigated. Item 5, **Table 6.2**.

6.4 IMPLEMENTATION OF TWO MICROPHONES AND AMBIENT NOISE METHOD

6.4.1 THE STANDARD STEREO LISTENING ROOM

The room chosen for impedance measurements using the two microphones and ambient noise method is the standard stereo listening room [Listening room] at Aalborg University. This room confirms with the [IEC 60268-13] standard and therefore describes the room acoustically as an average living room which, from this project point of view, fits well the real measurement conditions for the room boundaries. From the problem formulation section 4, the room boundaries are defined to be in a living room and the standard stereo listening room will therefore be a realistic condition to measure the impedances. However the standard stereo listening room may not be the ideal room for the two microphone and ambient noise method since this method assumes diffuse field which is not the case in living rooms, section 5.2.7. The reverberation chamber may provide a better results since the sound field is assumed to be diffuser than the sound field in the stereo listening room, but in general the reverberation chamber is far from the living room. The standard stereo listening room at Aalborg University is approximately 85m^3 big with a floor area on approximately 32m^2 and it is possible to change, remove or add different panels to change the reverberation time. Drawing of the room is available in appendix section 8.2. In the standard condition, the reverberation time, T60, is measured to 0.35s using room impulse responses and Schroeder backwards integration and it is possible to have reverberation time up to 1.2s if all panels and removable damping materials are removed [Cervantes, Prepelita, Peyret & Bonde, 2012]. Combining the room volume and reverberation time, the transition frequency, as explain in section 5.2.2 equation (5.1), can be calculated. Equation (6.1) and (6.2).

$$f_c = 2000\sqrt{\frac{0.35[s]}{85[m^3]}} = 128[Hz] \tag{6.1}$$

$$f_c = 2000\sqrt{\frac{1.2[s]}{85[m^3]}} = 238[Hz] \tag{6.2}$$

Depending on the reverberation time, the transition frequency will vary from 128Hz to 238Hz and the room modes will be the dominating phenomena below these frequencies in the standard stereo listening room.

6.4.2 SETUP AND EQUIPMENT

From section 5.6.1.7, the two microphones and ambient noise method is based on measurements from two microphones positioned nearby the test specimen. Using the signals from these microphones, the transfer function from microphone A to B is calculated and combined with the distance between microphones and the test specimen and the spacing between the microphones, the impedance is calculated. See **Figure 5.21** which explains the setup and **Figure 6.1** which explains the signal processing. The signals from microphone A and B are firstly multiplied by calibration factors which are calculated using recordings of the 1KHz, 94dB re. 20µPa calibrator sound. Then windows are applied to the signals, the signals are converted to frequency domain using FFT and the transfer function from microphone A to B is calculated. Windowing, FFT and calculation of transfer function is performed by the matlab function tfestimate.m which is using overlapping welch windows as window function. The last step is to combine the transfer function from microphone A to B with the distances in order to get the impedance of the test specimen. This is performed in the Hab2Z block which is based on equation (5.56) section 5.6.1.7.

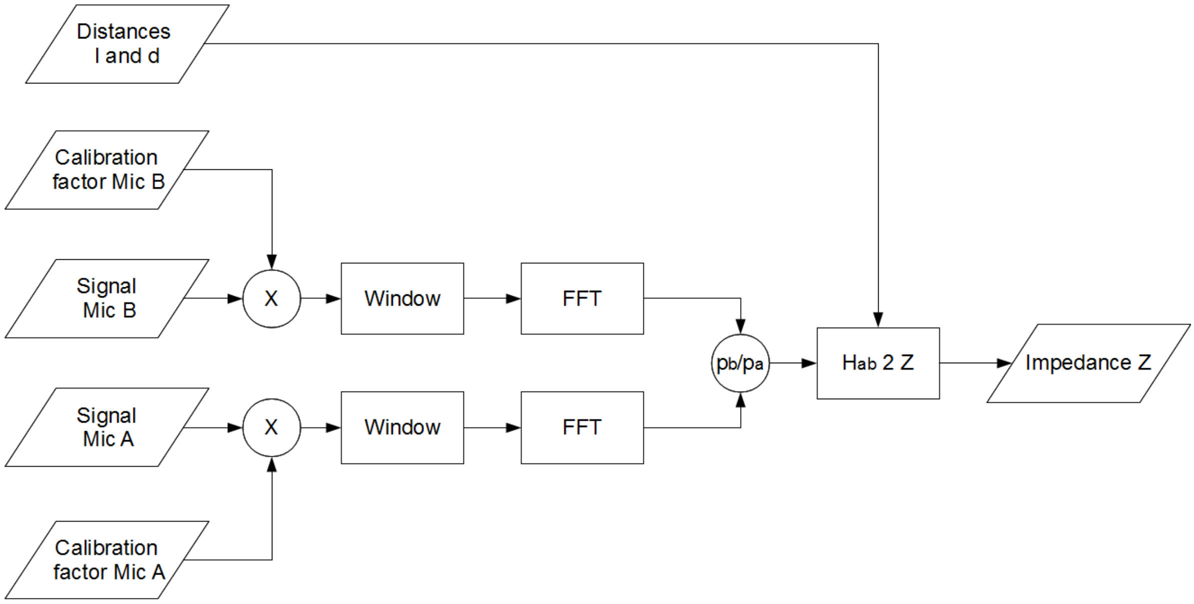


Figure 6.1 – Signal processing flowchart for the two microphones and ambient noise method. The inputs are the distances, measured signals in time domain for microphone A and B and calibration factors for microphone A and B. The output is the impedance. The microphone A and B signals are calibrated and the transfer function between them is then calculated. The window and FFT blocks converts a certain length of the microphone A and B signals to frequency domain. The Hab2Z block calculations the impedance with use of the transfer function between microphone A and B and the distances. Hab2Z is based on equation (5.56) section 5.6.1.7.

To measure the microphone A and B signals the measurement setup showed in **Figure 6.2** and **Figure 6.3**, is used. The microphone capsules are connected to preamps and phantom power supplies and the signals are then recorded using a soundcard connected to a laptop with DAW (digital audio workstation) software. The amplifier and loudspeaker connected to the soundcard are used to generate noise in case the ambient noise is not sufficient. The chosen microphone capsules are GRAS 40EN which are 1 inch pressure field microphones and are compatible with the used B&K preamplifiers and phantom power supply. The choice of 1 inch microphone capsules rather than more commonly used ½ inch types are due to better sensitivity and lower thermal noise which for the GRAS 40EN is respectively 50mV/Pa and 9.6dB re. 20µPa[Gras]. Disadvantages are bigger physical dimension which limit the distance between the microphones in the measurement setup to minimum 24mm(distance between microphone centers) and lower upper frequency limit which is 8KHz for the GRAS 40EN. This is however not a problem due to the project limitation at 1KHz. The used soundcard is Roland Quad capture UA-55, which is an external USB soundcard intended for music recording and production and when the internal automatic gain function, compressors and filters are disabled it can be used in the setup. The microphones are connected to input 1 and 2 and the gains are manually and equally adjusted. The maximum expected pressure to measure is the 94dB re. 20µPa 1KHz tone generated by the calibrator and the gains are therefore adjusted to this level. To record and save the signals on the laptop, the DAW software Flstudio 11 is used together with the ASIO drivers for the soundcard. The signals are then recorded in 24bit wav-files with a sample rate of 48KHz. The chosen loudspeaker is B&W DM601-S2 which is a 2-way, vented box, mid end hifi loudspeaker with frequency range 70Hz to 20KHz +-3dB [B&W]. The loudspeaker is believed to represent well the average loudspeaker in living rooms and is therefore chosen. Detailed descriptions of the setup and equipment settings are available in appendix section 8.1.3.

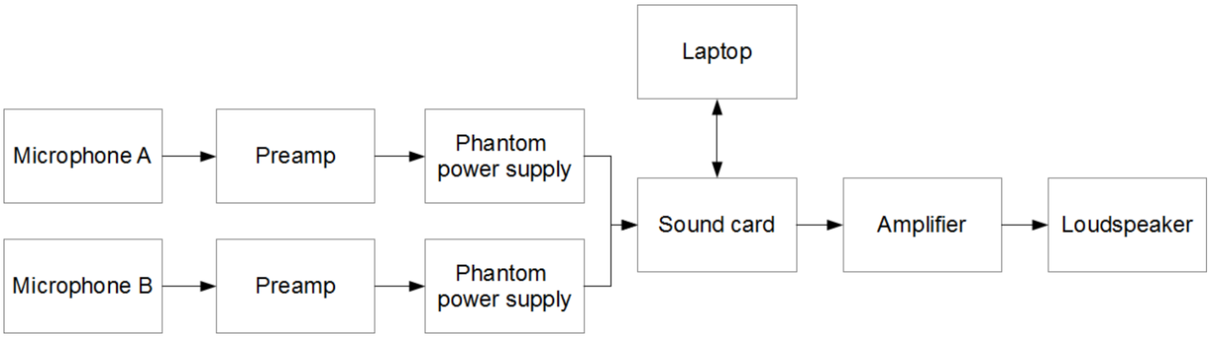


Figure 6.2 – Setup and connections between equipment for the two microphones and ambient noise method.

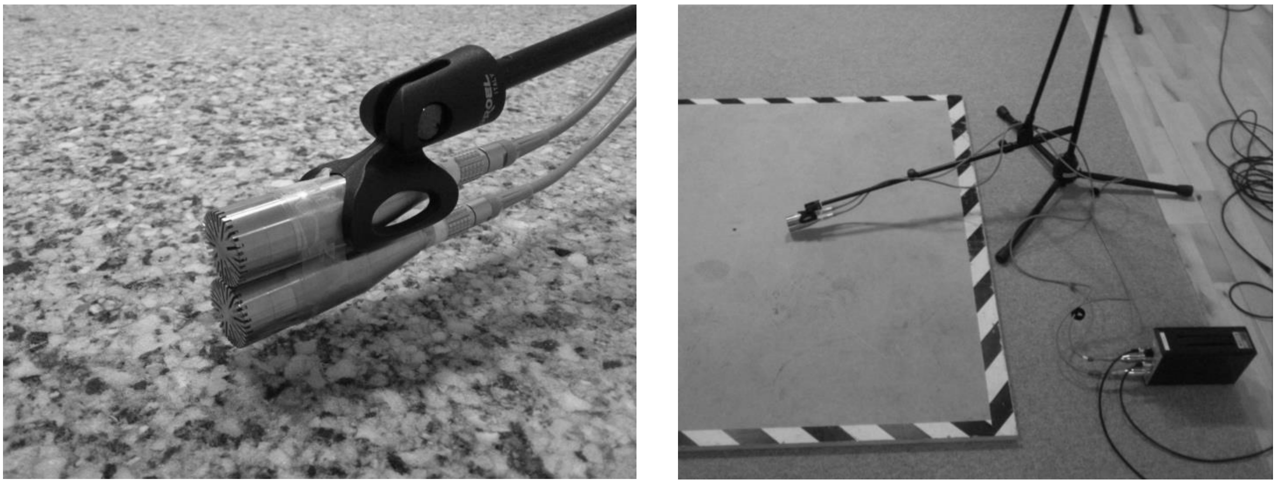


Figure 6.3 – Picture of the setup in the standard stereo listening room. Left: close up picture of the two microphones and the sponge mat. Right: picture of the microphone stand, phantom power supply, microphones and MDF board.

Since the measurement is based on the transfer function from microphone A to B it is important that the two microphones and signal chains have equal frequency and phase responses in order to avoid including the signal chain differences in the transfer function between microphone A and B. A solution is to match the microphones and signal chains by measuring their responses when the microphones are mounted in a coupler. The measured differences can then be taken into account in the transfer function from microphone A to B calculations. Another solution and also the chosen one, is to interchange the microphone positions during measurements and average the measured signals in frequency domain. The signal from microphone position A and B are hereby measured with both microphones and possible miss match will be canceled out. This method is also proposed by the standard for transfer function method in tube [ISO 10534-2]. Even though interchanging of microphone positions will cancel out miss match, phase and frequency responses for the soundcard inputs were verified. The soundcard outputs were looped to the soundcard inputs and impulse response measurements using [Holmimpulse] was performed in order to get frequency and phase responses. **Figure 6.4**. As expected the phase and frequency responses for both inputs were equal and almost flat. Exact description of the soundcard verification and results are available in appendix section 8.1.1.

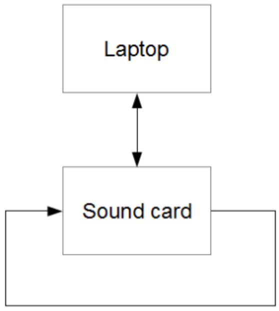


Figure 6.4 – Setup and connections between equipment for soundcard verification.

For the sake of convenience the test specimen is placed horizontally on the floor instead of vertically on the wall in the standard stereo listening room. From section 5.4.2 it was found that vertical boundaries nearby the loudspeaker shall be measured but since it is difficult to mount especially the sponge mat on the wall and since position is not expected to affect the results much, the specimen is placed on the floor.

6.4.3 Measurement procedure

To obtain result regarding the investigations stated in Table 6.2 section 6.3, measurements on the two test specimen will be performed with different noise sources, **Table 6.3**, and with different distances between the microphones and the test specimens.

Noise no.	Noise source	Noise type
1	Ambient	Ambient
2	Speaker	Pink noise. Full band
3	Speaker	Pink noise. Limited band
4	Speaker	Pink noise. Full band. Pulsing
5	Speaker	Pink noise. Limited band. Pulsing

Table 6.3 – Noise sources and types.

Noise no. 1 in **Table 6.3** is the ambient noise which exists in the standard stereo listening room. The two microphones and ambient noise method propose to use the existing noise and the existing noise in the standard stereo listening room is therefore an obvious first choice of noise source. The standard stereo listening room at Aalborg University is however a double wall construction (a box in a box) and the ambient noise may be at low SPL and maybe lower than the noise generated by the measurement equipment itself at some frequencies. If this is the case it is not possible to measure and calculate the needed transfer function from microphone A to B. To avoid this problem, noise no.2 and no.3 are pink noise radiated from a loudspeaker while measuring. Noise no. 2 is full band (20-20KHz) pink noise only limited by the loudspeaker frequency response and noise no.3 is also full band pink noise but high pass filtered at 128Hz. The pink noise is high pass filtered at 128Hz, which is the transition frequency in the standard stereo listening room at normal configuration, to avoid exciting distinct room modes which exist below the transition frequency. Low frequencies easily travel through walls and buildings and it is believed that even though the loudspeaker is not radiating noise below 128Hz, noise will still exist below this frequency in the standard stereo listening room. A problem with noise no.2 and no.3 is that the noise source is a loudspeaker which is directional. In other words, the noise will not be diffuse as specified in the two microphones and ambient noise method. Noise no. 4 and 5 may solve this problem. Instead of radiating the pink noise simultaneously while measuring, the noise is instead pulsed and measurement shall take place between the pulses. The measured noise is therefore the reverberation in the standard stereo listening room which is a more diffuse sound source than the loudspeaker. Noise no. 4 and no.5 are respectively full band and high pass filtered at 128Hz as noise no. 2 and no. 3. In **Figure 6.5**, the signal processing of the pulsed noise is explained. All the pink noise pulses are removed to have a pure reverberation signal and to improve the results, different window types and window sizes will be verified. The choice of window type and window size is a tradeoff between sound field and signal to noise ratio. The first part of a single reverberation event is typical dominated by 1.order reflections, then 2.order reflections, 3.order reflection and so forth. The longer the time period after the reverberation event start, the more will the reverberation be dominated by higher order reflections. This means that a single reverberation event becomes more diffuse over time but the reverberation does also decrease in SPL over time due to the physics behind reverberation. 60dB decrement pr. T60. See Figure 6.6. From section 5.2.7 it was found that the reverberation field is changing over time and the optimal window type and window size may also be affected by this behavior. The chosen distances between the microphones and test specimen are 2cm, 7cm and 20cm. The distance 2cm is expected to give useful result and will be used as a reference result to compare the result using the distances 7cm and 20cm. The temperature and humidity are measured and monitored during the measurement. A detailed description of the measurement procedure is available in appendix section 8.1.3 where the different noise generation methods are investigated, and appendix section 8.1.4 where the method using noise no. 4 is investigated further.

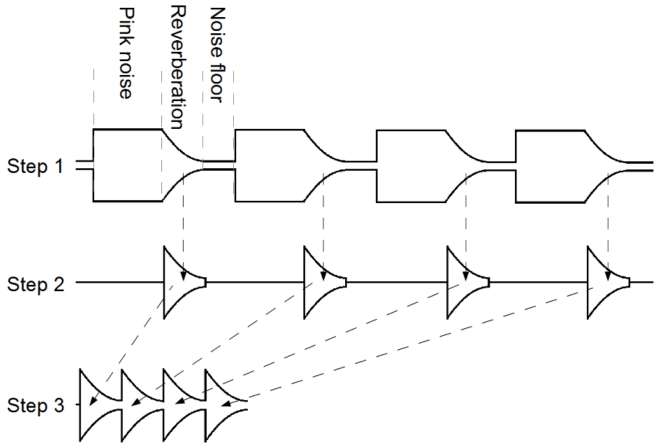


Figure 6.5 – Signal processing of the pulsed noise signal, noise no. 4 or 5. Step 1 is the raw signal with pulsed pink noise, reverberation tails and noise floor. Step 2 is the result when windows are applied around the reverberation tails to remove the pink noise and noise floor. In step 3 the zeroes between the reverberation tails are removed.

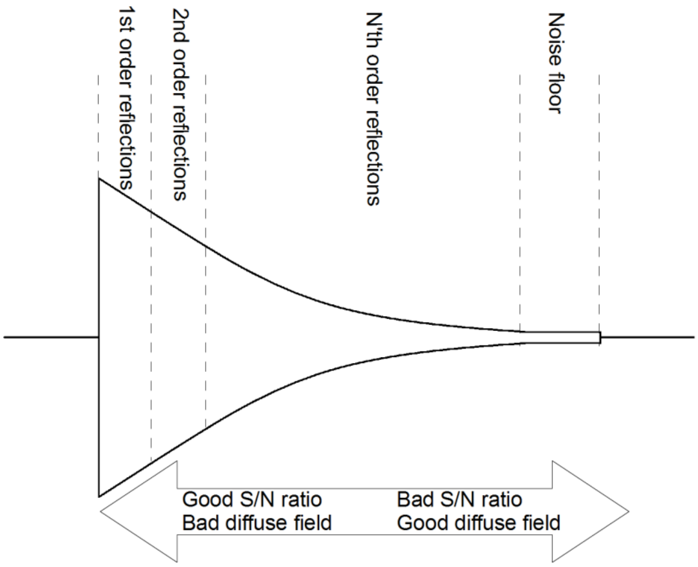


Figure 6.6 – Illustration of a single reverberation event. The first part is dominated by the 1st order reflections, 2nd order reflections and so forth. The longer the time period after the reverberation event start, the higher order reflections will dominate which lead to better diffuse field.

6.5 Implementation of the transfer function method using impedance tube

To be able to compare the results from the two microphones and ambient noise method another reliable measurement method is needed. The chosen method is the transfer function method in tube from [ISO 10534-2] which is explained in section 5.6.1.3. This method is a commonly used laboratory method and because the two microphones and ambient noise method is related to this method it seems as a good choice to create reference measurements. The available impedance tube in the acoustic laboratory at Aalborg University is the Brüel and Kjær type 4002, which is a tube with only one but moveable microphone. It is therefore necessary to perform 2 measurements each test specimen to obtain data from the two needed microphone positions. This method is called the one-microphone method and an advantage compared to the two-microphone method, is that it eliminates phase mismatch since the measurements at the two microphone positions are performed by the same microphone. However the one-microphone method require more time. The setup is equal the standing wave method in tube setup, **Figure 5.17**. From section 5.6.1.3, it is known that the needed data to calculate the impedance is the transfer function from microphone A to B, the distance x_1[m] and the spacing between the microphones, s[m]. To obtain the transfer function from microphone A to B, the transfer functions for respectively microphone A and B are measured and then divided with each other. See **Figure 6.7**.

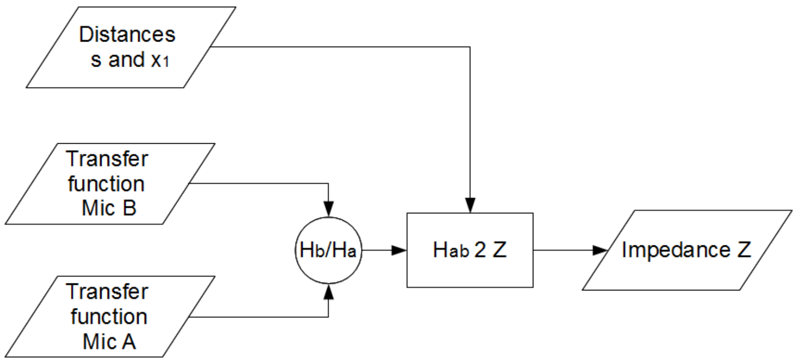


Figure 6.7 – Signal processing flowchart for the transfer function method in tube. The inputs are the distances and transfer functions for microphone A and B and the output is the impedance. The Hab2Z block calculates the impedance with use of the transfer function between microphone A and B and the distances. Hab2Z is based on equation (5.48) and (5.49) section 5.6.1.3.

To measure the transfer functions for microphone A and B, a computer based setup using the software [Holmimpulse] is chosen. Holmimpulse generates a signal which is amplified through the measurement amplifier for the impedance tube and then calculates the transfer function using the signal from the microphone. The D/A and A/D conversion of the signals are performed using the same approved soundcard as used in the two microphones and ambient noise method. The measurement setup is showed in **Figure 6.8**.

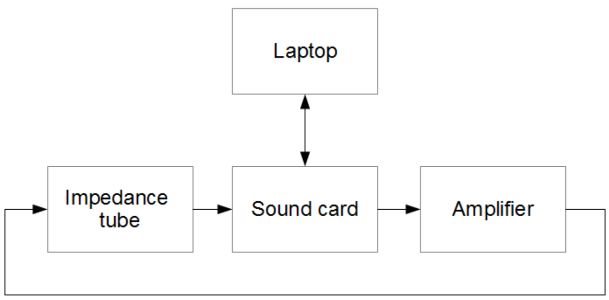


Figure 6.8 - Setup and connections between equipment for the transfer function method in tube.

The chosen signal is logarithmic sweep with length of 2^17 samples and is chosen over MLS signal for several reasons. Sweeps perform better when it comes to distortion and time variance and sweeps also improves the signal to noise ratio since the sweep signal can be played at higher levels than the MLS [Müller & Massarini, 2001]. However, according to [ISO 10534-2] the signal just has to be 10dB louder than background noise and the SPL is therefore not an issue. When the transfer functions are measured, they are imported to the developed matlab script *DVD\Matlab code\ImpedanceTube1.m*, which is based on the equations (5.48) and (5.49) section 5.6.1.3 and herby calculates the impedance. The upper frequency limit is determined by the tube diameter and spacing between the microphones as explained in section 5.6.1.3. For the used impedance tube, B&K 4002, it is possible to mount two different types of tubes with different length and diameter. One with a length of 1m and a diameter of 0.1m and one with a length of 0.28m and a diameter of 0.03m. For the tube which has a diameter of 0.1m, the upper frequency limit, from equation (5.36) and (6.3), is 2000Hz and since it is within the project limitations, section 4.2, this tube is therefore used. The upper frequency limit is applied to Holmimpulse.

$$f_u = \frac{200}{0.1[m]} = 2000[Hz] \tag{6.3}$$

When the upper frequency limit is 2000Hz the spacing, S[m], between the microphones shall be less than 0.077m. Equation (5.41) and (6.4).

$$S < \frac{154}{2000[Hz]} = 0.077[m] \tag{6.4}$$

The chosen spacing between the microphones are therefore 0.05m and the distance x_1[m] is chosen to 0.1m. At this distance the reflected sound waves from the test specimen are assumed to be a plane wave and the microphone position B is approximately within the first and second pressure minima. The [ISO 10534-2] proposes that the microphone position B should be close to the first or second pressure minima but since the measurements are performed using sweeps, this is not possible for all frequencies. To increase precision, the [ISO 10534-2] also propose to measure the acoustic center of the microphone since it may be different from the geometrical center of the microphone. This is however not performed because the measured impedances are only used for roughly comparison of the measured impedances from the two microphones and ambient noise method. The measurement setup, used equipment and exact procedure are available in appendix section 8.1.2.

6.6 Results and analysis

6.6.1 Transfer function method in tube

Following the measurement procedure in section 8.1.2, the transfer functions for microphone position A and B in the tube were measured with respectively the sponge mat and MDF board as test specimens. Loading these transfer functions into the matlab code *DVD\Matlab code\ImpedanceTube1.m,* which follows the signal processing explained in section 6.5, the impedances and reflection factors are calculated and plotted in the frequency range 100Hz to 2KHz and smoothed with 1/3 octave band resolution. **Figure 6.9**, **Figure 6.10** and **Figure 6.11**. The lower frequency limit at 100Hz is chosen due to ongoing construction work nearby the acoustic laboratory while the measurements were performed. The low frequency noise generated by the construction work obscured the measurements below 100Hz and the results were therefore not useful below this frequency. Averaging of several measurements would have improved the results but was not performed. The upper frequency limit at 2KHz is the upper frequency limit for the impedance tube due to tube diameter and is therefore chosen as the upper limit even though the project limitations, section 4.2, propose the upper frequency limit to 1KHz. There may be useful information between 1 and 2KHz and this frequency range is therefore included in the measurement results. The 1/3 octave band smoothing of the plotted data is chosen for better visualization and easier analysis of the results. The small fluctuations in the measured data, which are not hearable [Toole, 2008] are smoothed out. The raw data and data processing before plotting are kept in their original resolution without smoothing to avoid data loss. Even though the main purpose is to measure, calculate and analyze the impedances, the reflection factors are also calculated and plotted in order to be analyzed. This is done because the reflection factors are more palpable than the impedances and it is easier to imagine how a given reflection factor will affect reflections of sounds. The measured transfer function for microphone position A and B are available on the *DVD\Measurement data\Transfer function method in tube* as txt files.

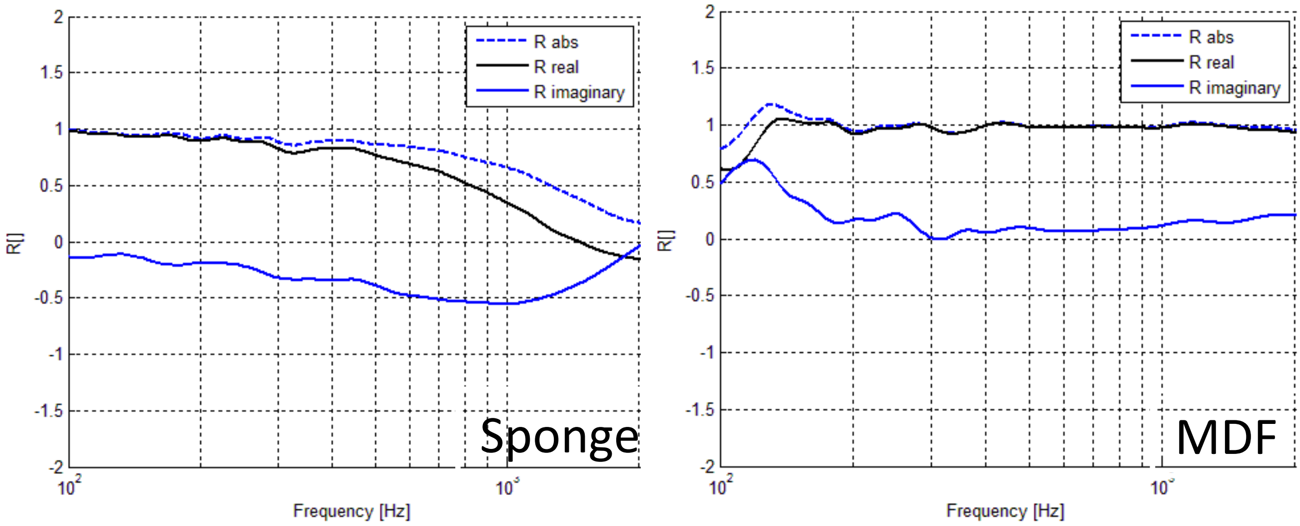


Figure 6.9 – Measured reflection factors for the sponge mat and MDF board using transfer function method in tube.

By examining the reflection factor results plotted in **Figure 6.9**, it is obvious that the sponge mat and MDF board reflects sounds differently as expected. The sponge mat absolute value of reflection factor decreases when the frequency increases while the MDF board absolute value of reflection factor is constant 1 for the plotted frequency range. The imaginary and real parts of the reflection factors also reveal that the materials have different phase behavior and the phase is therefore plotted in **Figure 6.10**. The MDF board has phase changes slightly above 0 degree while the sponge mat phase decreases from around 0 degree to -150 degree during the frequency range.

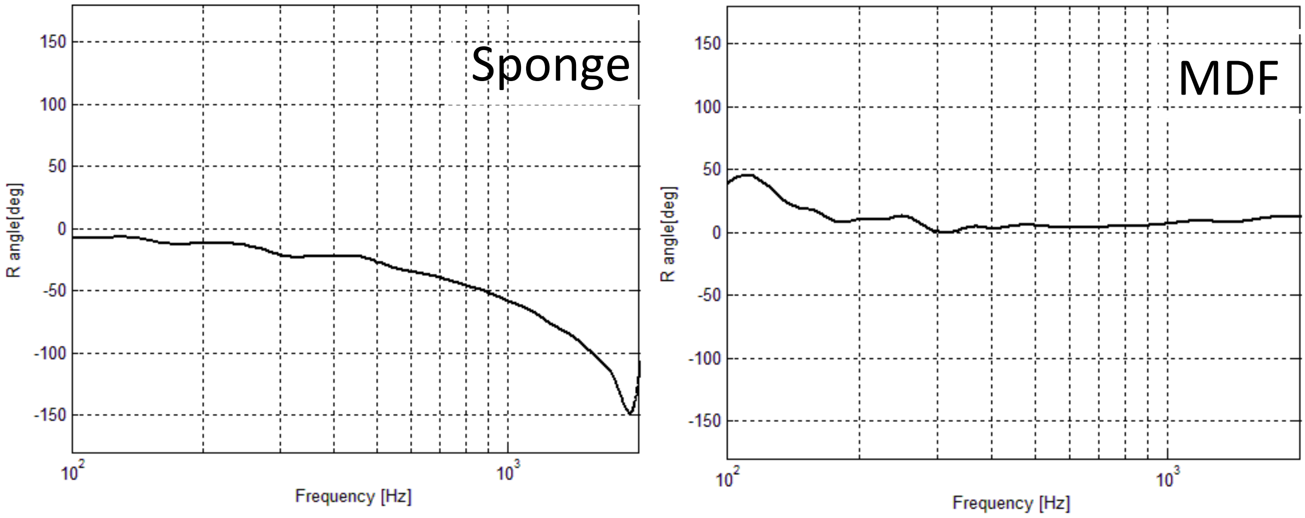


Figure 6.10 – Measured reflection factor phases for the sponge mat and MDF board using transfer function method in tube.

In **Figure 6.11** the impedances for the sponge mat and MDF board are plotted and as expected the MDF board has higher impedance than the sponge mat. Both impedances are also larger than impedance for air which is approximately 415Pa·s/m at 20°C and the results seems valid even though there are some fluctuations especially for the MDF board. The low impedance for the MDF board in the frequency range below 250Hz may be due to bad/loose fixing of the MDF sample in the impedance tube. If the MDF sample is able to move, it will behave as a resonance absorber and the resonance may be below 250Hz. This will explain the behavior of the measured impedance and also the roll off at around 150Hz in the reflection factor for the MDF board, **Figure 6.9**.

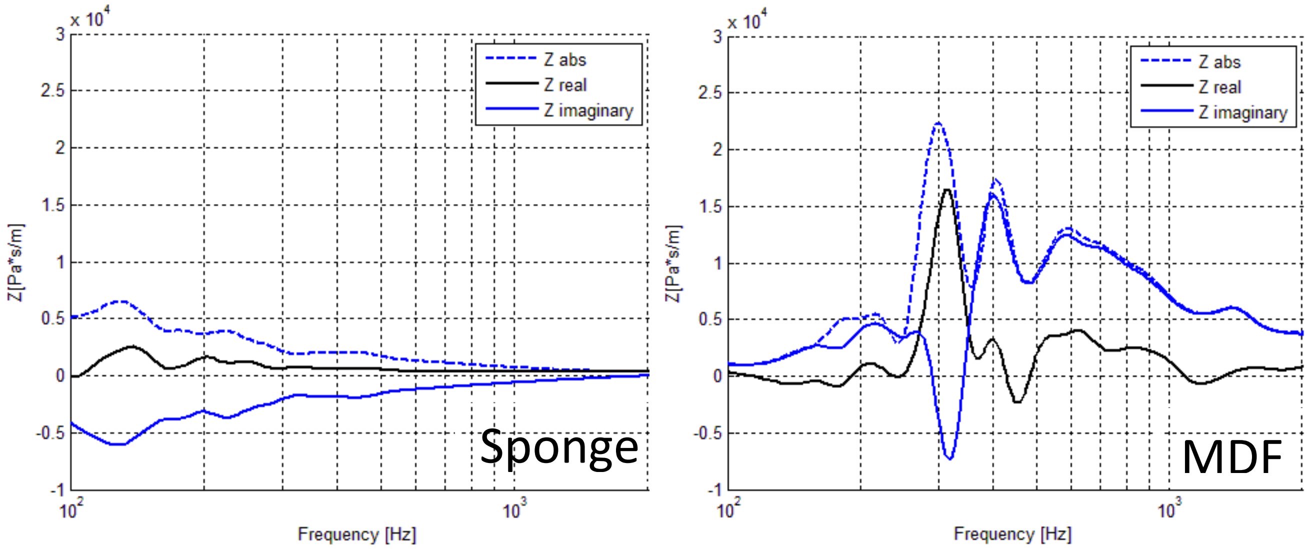


Figure 6.11 – Measured impedances for the sponge mat and MDF board using transfer function method in tube.

6.6.2 Two microphones and ambient noise method

To analyze the performance of the two microphones and ambient noise method and investigate the method according to Table 6.2 in section 6.3, the results for the two microphones and ambient noise method are compared with the results from the transfer function method in tube. An important aspect for the two microphones and ambient noise method is the ambient noise and an analysis of the different noise sources and types proposed in Table 6.3 in section 6.4.3 are therefore firstly carried out. Secondly the distance between microphones and test specimen and microphone spacing will be analyzed and finally an analysis of the overall performance will be carried out.

6.6.2.1 Noise source analysis

In order to measure the transfer function between the two microphones, needed for impedance calculations and in order to get useful data, it is important that the ambient noise fulfill some requirements. From the theory section 5.6.1.7 and implantation section 6.4, the following known requirements for the noise can be derived:

No.	Requirement
1	The ambient noise sound pressure level shall be louder than the noise generated by the measurement system.
2	The ambient noise shall cover the frequency range in which the measurements are performed.
3	The ambient noise shall be diffuse field.

Table 6.4 – Ambient noise requirements.

The first and second requirements are needed in order to measure the transfer function between the two microphones in the desired frequency range. If the ambient noise sound pressure level is less than the noise generated by the system, the transfer function will be based on system noise and not the ambient noise. The used GRAS 40EN microphone capsules have a thermal noise of 9.6dB re. 20µPa [Gras] and the ambient noise therefore needs to exceed that sound pressure level. How good signal to ratio is needed is not known. The third ambient noise requirement is due to the theoretically assumptions of diffuse field in the reflection factor and impedance calculations. To analyze the ambient noise requirement 1 and 2, the sound pressure levels for the ambient noise types Table 6.3 in section 6.4.3 have been measured in 1/3 octave band using *DVD\matlab code\SPLlevels.m* and the matlab toolbox Octave [Mathworks] which follows the standard for octave band filters [ANSI S1.11-1986]. Minor corrections and changes are however added to the toolbox which is available on the *DVD\matlab code* including the corrections. The measurements are performed following the procedure from section 8.1.3. In **Figure 6.12** the sound pressure level for the ambient noise which exist in the standard stereo listening, is shown. From Table 6.3 in section 6.4.3, this is noise no. 1. The low frequency noise is mainly generated by construction work nearby the acoustic laboratory and at the higher frequencies the noise is dominated by the microphone capsules thermal noise. Using the existing ambient noise in the standard stereo listening room as noise source for the two microphones and ambient noise method is therefore not sufficient.

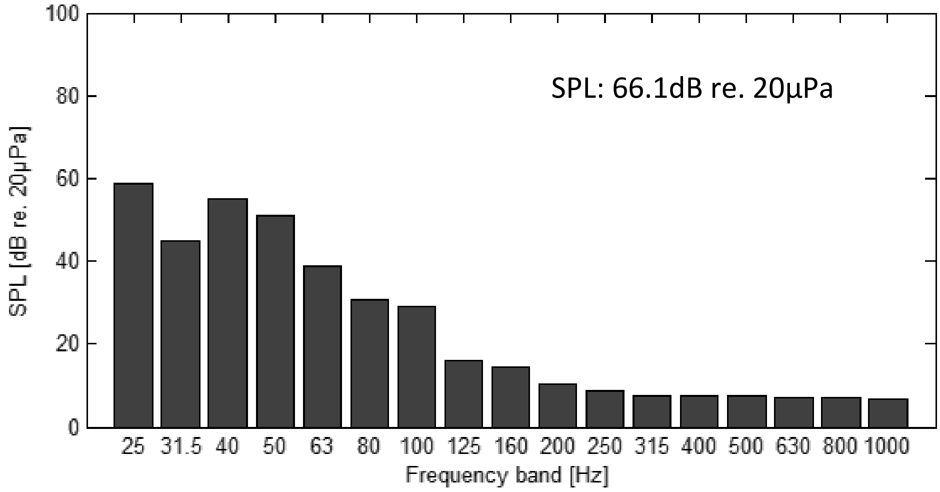


Figure 6.12 – SPL level in 1/3 octave band for existing ambient noise in the standard stereo listening room. Noise no. 1.

In **Figure 6.13** and **Figure 6.14** the sound pressure levels for respectively pink noise, noise no.2, and filtered pink noise, noise no. 3, generated using a loudspeaker, are shown. These sound pressure levels are also used for the pulsed version of both pink noises, noise no.4 and no.5. Different from noise no. 1, noise no. 2 to 5 covers the entire desired frequency range and the combination of noise from the construction work, nearby the acoustic laboratory, and the noise generated by the loudspeaker, gives a nearly linear pink noise response from 25Hz to 1KHz. From these results it can be concluded that noise no.2 to 5 fulfills the first and the second requirement for the ambient noise.

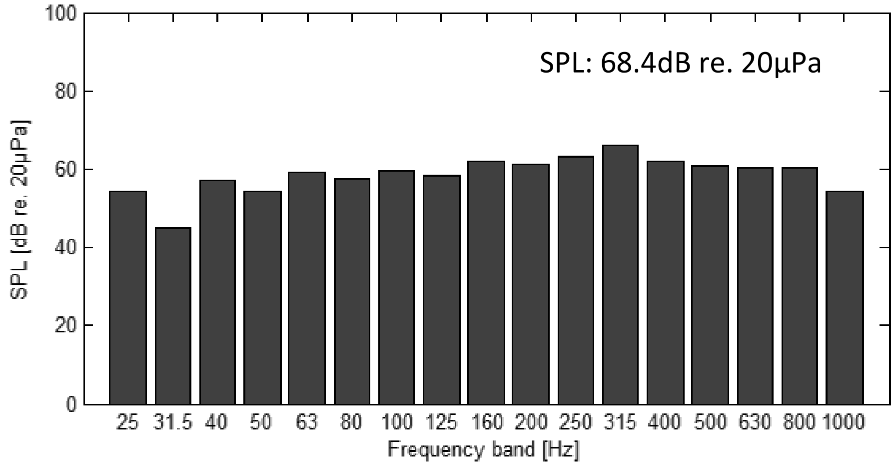


Figure 6.13 – SPL level in 1/3 octave band for pink noise and pulsed pink noise. Noise no. 2 and 4.

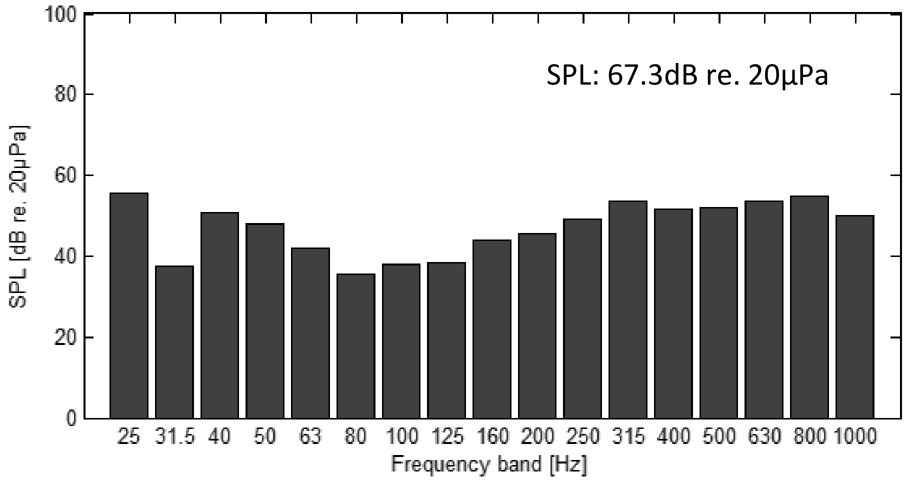


Figure 6.14 – SPL level in 1/3 octave band for pink noise and pulsed pink noise highpass filtered at 130Hz. Noise no. 3 and 5.

In order to have a useful noise source for the two microphones and ambient noise method, the third requirement shall also be fulfilled. Noise no. 2 and 3, which are noise generated continuously from a loudspeaker while measuring, are therefore not sufficient for the method. Their sound fields are far from diffuse due to the loudspeaker radiation characteristic. This is also confirmed by the measurements, section 8.1.3, and the result *DVD\Measurement data\Two microphones and ambient noise method. Iteration 1*, using noise no.2 and 3. The measured transfer function between the microphones, using noise no. 2 and 3, is affected by the direct sound radiated from the loudspeaker and becomes close to 1. What are left behind are the pulsed pink noises, noise no.4 and 5 which from the measurements section 8.1.3, and the result *DVD\Measurement data\Two microphones and ambient noise method. Iteration 1,* seems to have the best performance. The results using noise no. 5, are however very noise below 150Hz which may be due to the lower signal to noise ratio around this frequency. The best noise source, for this particular measurement setup, is therefore the noise no.4, the pulsed pink noise. For further investigations of the noise no.4 as noise source, measurements was performed using improved signal to noise ratio, better recording technique and more measurements in order to average the data, section 8.1.4. The noise no.4 sound pressure level, during the measurements, is shown in **Figure 6.15**. Since the measurements were performed on a Saturday with no ongoing construction work, the low frequency sound pressure levels are lower than earlier measurements.

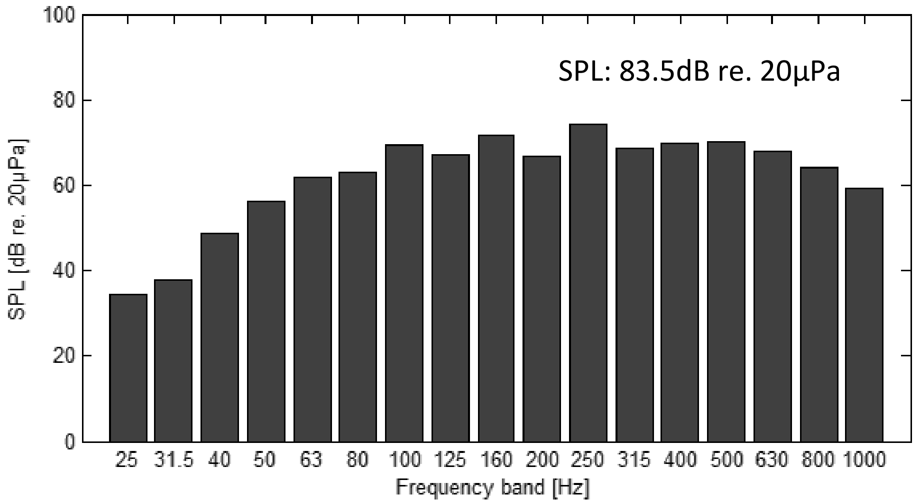


Figure 6.15 – SPL level in 1/3 octave band for pulsed pink noise with increased sound pressure level. Noise no. 4.

6.6.2.2 Analysis of distances

If a measurement setup is mounted on a loudspeaker in order to measure impedances of nearby boundaries using the two microphones and ambient noise method, the method needs to be reliable for different distances between the microphones and test specimen. The distance L2[m] in Figure 5.21 section 5.6.1.7. It is therefore necessary to investigate how different distances affect the measurements. In order to do that, the measurement results *DVD\Measurement data\Two microphones and ambient noise method. Iteration 2,* from section 8.1.4, are analyzed. These measurements results are impedance measurements using the sponge mat and the MDF board as test specimen, using noise no.4 as noise source and repeated using 3 different distances between microphones and test specimen. The distances are 2, 7 and 20cm. Since the two microphones and ambient noise method is based on the measured transfer function between the two microphones the measured transfer functions at the different distances is examined. See **Figure 6.16**, where the transfer functions for the different distances and using the sponge mat as test specimen, are showed.

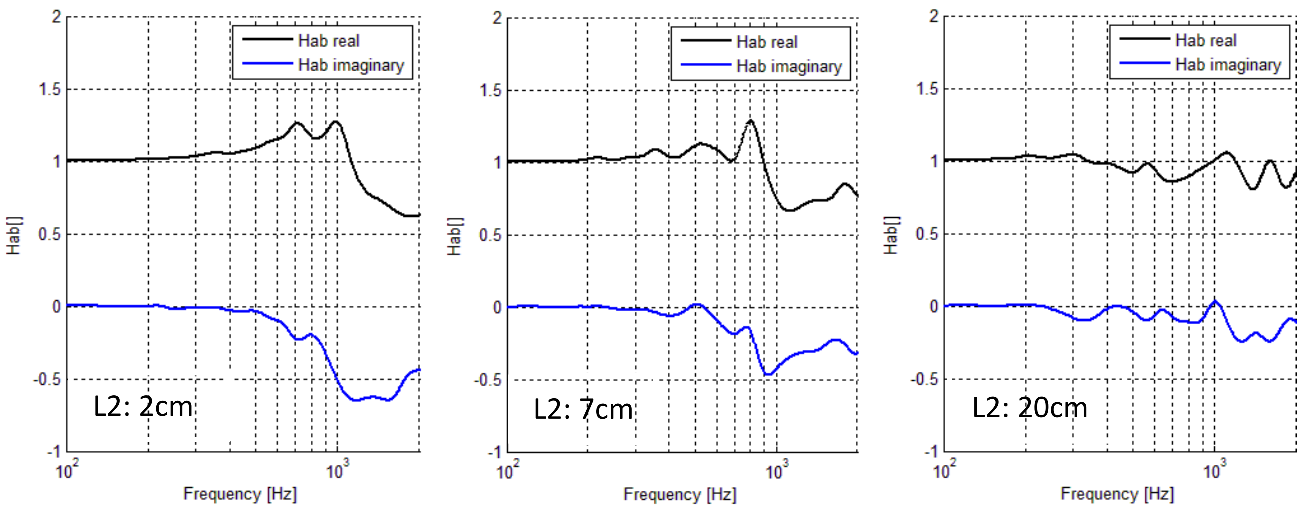


Figure 6.16 – The measured transfer function between microphone A and B for different distances between the microphones and test specimen. From left to right the distances are respectively 2, 7 and 20 cm and the transfer function becomes closer to 1 for the real part and 0 for the imaginary part. The sponge mat is used as the test specimen in all 3 cases.

Examining the transfer function in **Figure 6.16** it is obvious that the larger the distance, L2[m], the closer the real part approaching 1 and the imaginary part approaching 0. This means that the larger the distance, the less information about the boundary and the calculated reflection factor and impedances becomes noisier and less reliable. This is confirmed by comparing the measured reflection factors for the 3 different distances with the reflection factors measured using the transfer function method in tube and mean square error calculation, equation (6.5).

$$\mathrm{MSE} = \frac{1}{n}\sum_{i=1}^{n}(\widehat{R_i} - R_i)^2 \tag{6.5}$$

R is the reflection factor measured using transfer function in tube, R hat is the reflection factor measured using the two microphones and ambient noise method and n is number of samples. Since the measurements are only valid in the frequency range between 100Hz and 2KHz, the mean square error is limited to this area. Results are available in **Table 6.5**.

Distance L2 [cm]	|R| Sponge MSE	|R| MDF MSE
2	0.07	0.20
7	0.23	2.06
20	0.83	1.60

Table 6.5 – Mean square errors between absolute values of reflection factors measured with the two microphones and ambient noise method. The reflection factors, measured using the transfer function method in tube, are used as reference.

The distance L1[m], which is the spacing between the microphones, is like the transfer function method in tube, determined by the desired frequency range to measure. The distance has to be shorter than the wavelength of the desired upper frequency limit but the shorter the distance the more difficult it becomes to get useful data at low frequencies. When the wavelength is much larger than the distance, which is the case for low frequencies, the pressure difference between the microphones become small and maybe smaller than what the measurement equipment is capable to measure. The distance L1[m] is therefore a tradeoff between measurement of low and high frequencies.

6.6.2.3 Overall performance

From the noise analysis section 6.6.2.1 and the distance analysis section 6.6.2.2, the best performance was obtained using noise no.4, the pulsed pink noise, and the distance, 2cm, between the microphones and test specimen, L2[m]. In **Figure 6.17** to **Figure 6.20**, the measured reflections factors and impedances for the sponge mat and MDF board, using these settings, are shown together with the reflection factors and impedances measured using the transfer function method in tube. From Figure 6.5 in section 6.4.3 the signal processing of the pulsed pink noise is explained and by trial and error the best results was obtained using the first 0.1s of the reverberation tail windowed with a hann window. Better fits may exist but without the knowledge about the exact sound field at the particular measurement and measurement position, it is hard to find. In **Figure 6.17** and **Figure 6.18** the reflection factors are shown and except some fluctuations and deviant results for the MDF board, the measurements are equal the transfer function method in tube measurements. Using the two microphones and ambient noise method therefore gives at least the opportunity to measure and calculate the absorption coefficient. Examining the measured impedances in **Figure 6.19** and **Figure 6.20** reveals that the impedances are very fluctuating and at some point deviates a lot from the transfer function method in tube. It is however possible to deduce that the sponge mat have a lower impedance than the MDF board. During window fitting of the reverberation tail it was discovered that the impedance calculation is very sensitive to small changes and a better window fit will maybe improve the impedance results. All results and measurements including the 3 different distances and different noise sources are available on *DVD\Measurement data\Two microphones and ambient noise method. Iteration 1.* and *DVD\Measurement data\Two microphones and ambient noise method. Iteration 2.*

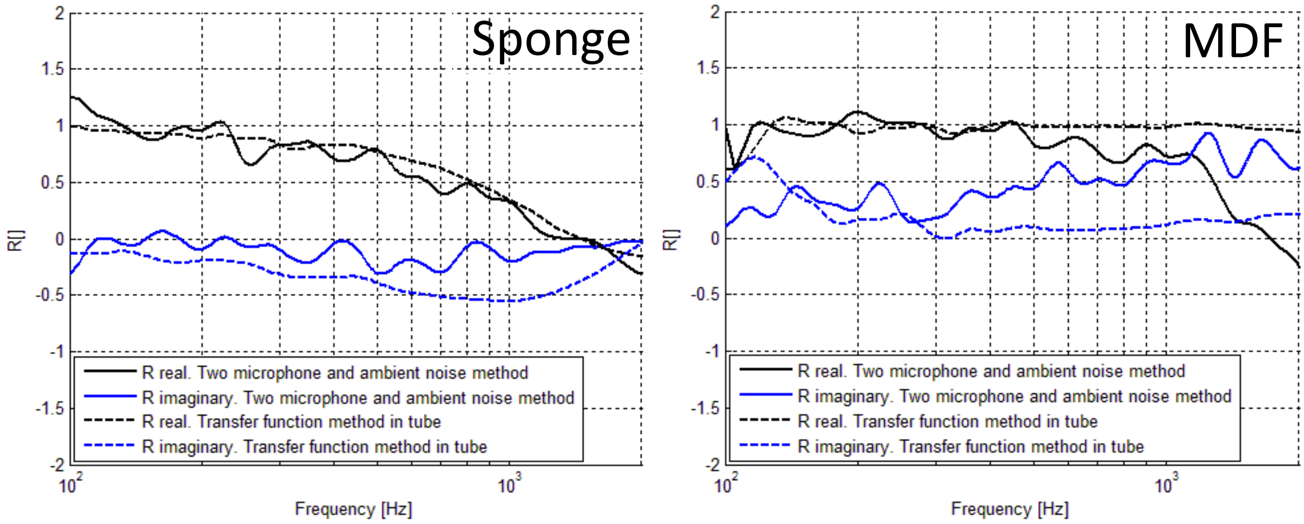


Figure 6.17 – Reflection factors measured using the two microphones and ambient noise method and the transfer function method in tube. Real and imaginary values.

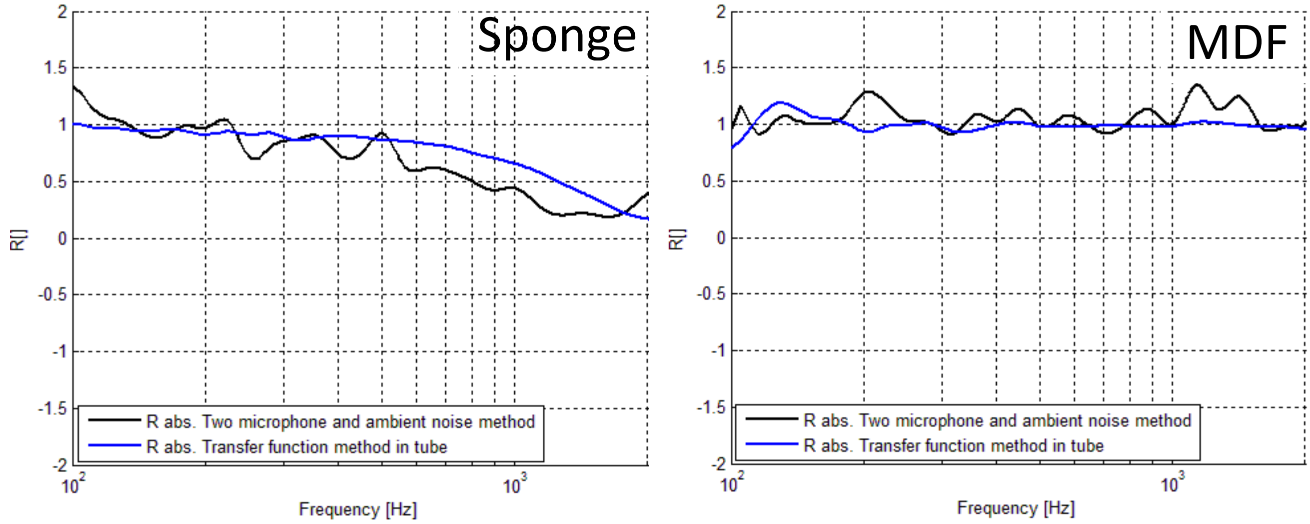


Figure 6.18 – Reflection factors measured using the two microphones and ambient noise method and the transfer function method in tube. absolute values.

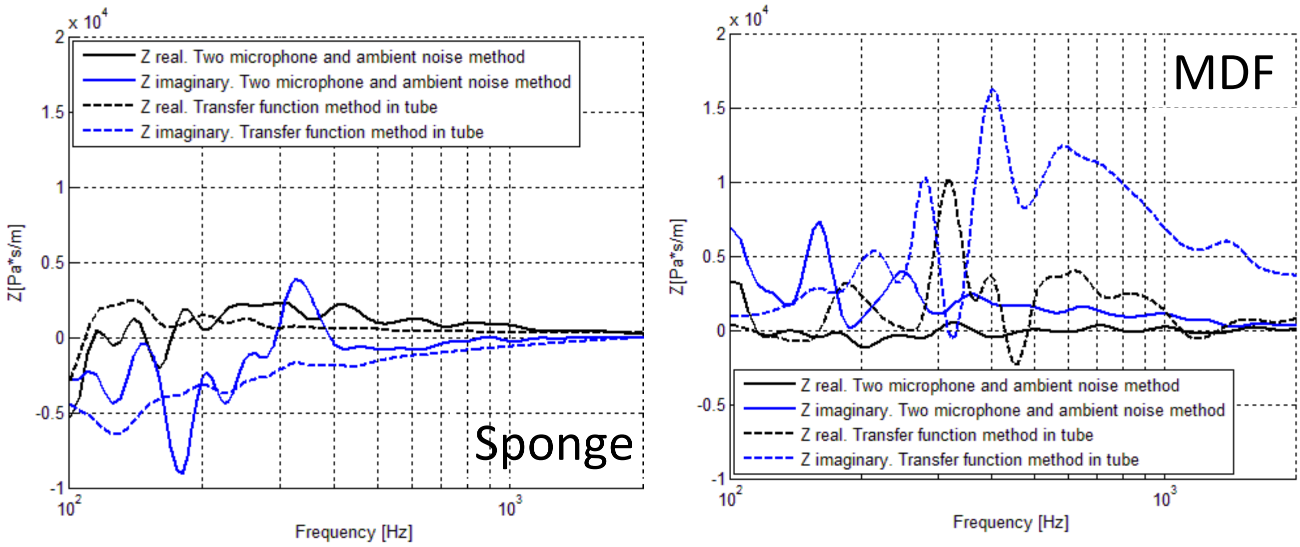


Figure 6.19 – Impedances measured using the two microphones and ambient noise method and the transfer function method in tube. Real and imaginary values

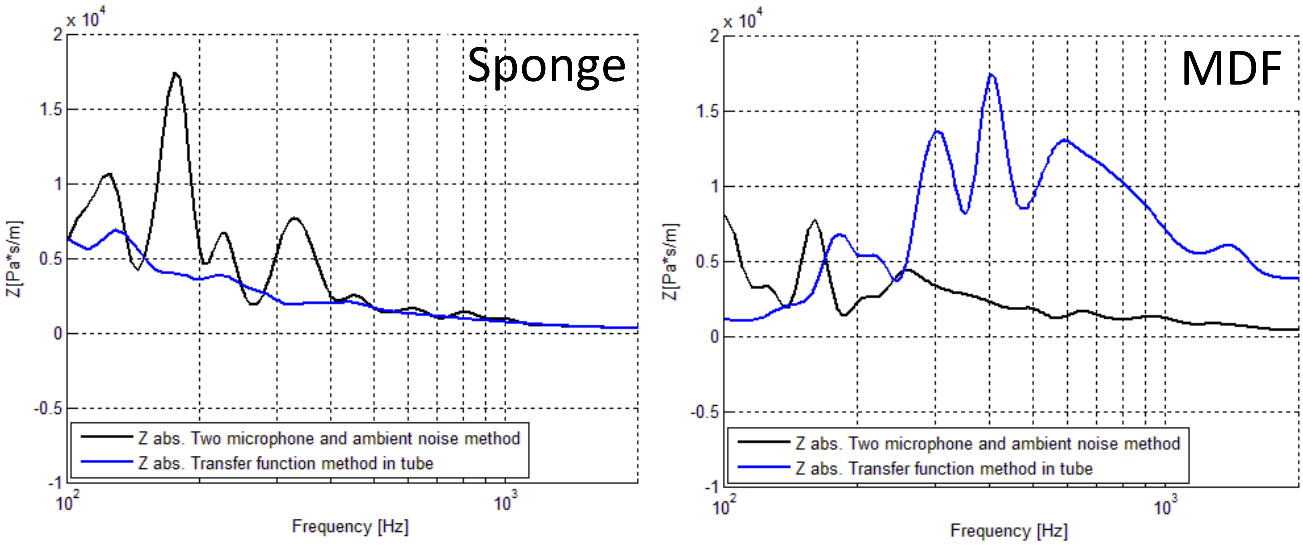


Figure 6.20 – Impedances measured using the two microphones and ambient noise method and the transfer function method in tube. Absolute values.

7 CONCLUSION

7.1 ANALYSIS OF ROOM BOUNDARY CHARACTERISTICS

When sound is reproduced in a living room through loudspeakers in a stereo setup, different acoustic phenomena will occur due to the physics behind sound radiation in rooms. These phenomena are divided into two frequency ranges divided by the transition frequency, also known as the Schroeder frequency. Below the transition frequency, room modes will be the dominating phenomenon due to standing waves and above the transition frequency early reflections will be the dominating phenomenon. Common for both phenomena are that their behavior can be explained using some well-known properties, namely impedances, distances, weather conditions and radiation characteristics for the sound sources, which in this case are loudspeakers. The room boundary characteristics are therefore defined as impedances, distances and weather conditions. The impedances explain how hard or soft boundaries are and thereby how sound waves will be reflected by the boundaries, the distances explain the size of the boundaries and their position in space and the weather conditions explain the air conditions which surround the boundaries. Even though boundary positions in space and weather conditions not explain the actual boundaries, they are included in the room boundary characteristics because they explain how the boundaries affect sound in space. Boundary texture can also characterized as a room boundary characteristic since it will cause scattering of the reflected sound waves but due to the project frequency limit at 1KHz, scattering and boundary textures is not taken into account. By means of a simple simulation with a single sound source and one boundary, the weighting between the defined room boundary characteristics were analyzed. From this analysis it was discovered that the weather conditions has minor influence on the sound while the impedances and distances are equally weighted. The possible changes of weather conditions in living rooms are not capable to change the sound so much it becomes audible. The distances, especially between a sound source and a room boundary, will lead to comb filtering and depends on impedances, comb filtering will be more or less dominating. From a perceptual point of view the comb filtering leads to timbre changes for the perceived sound but depends on the direction, from where the reflections are radiated in a room, the perceived change of timbre is affected due to our binaural hearing and spectral smoothing. Reflections radiated nearby the direct sound will degrade the perceived sound while the reflections from sides will downgrade this degrading effect. The room boundaries nearby the loudspeakers in a living room will therefore have the greatest influence on the perceived sound but how great is influenced by other boundaries in the room and the loudspeaker radiation characteristics.

7.2 MEASUREMENT OF IMPEDANCE

In order to compensate for the room boundary sound influence using e.g. some kind of sound coloration compensation system, measurements of the room boundary characteristics are needed. The optimal solution will be a measurement system mounted in or on the loudspeakers in the stereo setup and able to measure the room boundary characteristics with minimum user interaction as possible. Distances are relatively easy to measure using laser distance meter etc. but for impedances it is differently. Different impedance measurement methods exist including laboratory and in-situ methods but common for them all are that they do not seem to be useful for the desired measurement system. The most promising found measurement method is the two microphones and ambient noise method which is able to measure the impedances using two microphones, ambient noise and known distances between the microphones and test specimen or in this case room boundaries. In order to be useful for the desired measurement system, the method have to perform well with different distances between microphones and room boundaries, therefore impedance measurement on a MDF board and a sponge mat at different distances were performed and compared with impedances measured using the transfer function method in tube. The measurement was performed at 2, 7 and 20cm and the results revealed that the longer the distance, the less information about the boundaries in the measurement data and therefore less reliable measurement results. The measurements performed at 2cm are good and the deviations from the transfer function method in tube measurements are small. The two microphones and ambient noise method measurements are however noisy and have some fluctuations and is far from exact. In the theory about the two microphones and ambient noise method it is assumed that the ambient noise is diffuse field and present in the frequency range desired to measure. This is however not the case in all living rooms and an investigation on different noise sources was performed. The different noise source was the existing noise in the standard stereo listening room, pink noise and filtered pink noise radiated from a loudspeaker and pulsed pink noise radiated from a loudspeaker where the measurements was performed in the reverberation field in-between the pulses. The existing noise in the standard stereo listening room was not loud enough at frequencies above 100Hz and the pink noise and filtered pink noise was not diffuse enough. These noise sources are therefore not sufficient for the two microphones and ambient noise method. The pulsed pink noise gave the best results due to the diffuse nature of the reverberation field but since the reverberation decreases 60dB pr. reverberation time, T60, and becomes more diffuse over time, the measurement in the reverberation field is a tradeoff between good diffuse field and good signal to noise ratio. The sound field is not measured for the particular measurement and measurement position and best fit is therefore not known. In summary the two microphones and ambient noise method is able to measure the impedances of room boundaries using a loudspeaker and the reverberation field as noise source, but the distance between the microphones and test-specimen have to be short as possible. This disqualifies the two microphones and ambient noise method for a measurement system mounted in or on the loudspeakers in the stereo setup. The distances will be too long. The method may however be used in a handheld device.

7.3 FURTHER INVESTIGATIONS

In order to improve knowledge about room boundaries and how they affect sound at listener position and in order to improve the two microphones and ambient noise method, the following investigations could be performed:

- Investigate how room boundary texture and scattering affect the sound field in a room in the frequency range above 1KHz.

- Investigate how reflections radiated from sides will affect the reflections radiated nearby the loudspeaker, perceptually. Which influences will the frequency range, sound pressure level and radiation angles have?

- Investigate the living room sound field using e.g. an intensity probe or microphone array. To improve the two microphones and ambient noise method using pulsed pink noise, the sound field at measurement position can be measured to obtain the best diffuse field as possible in the reverberation tail.

8 APPENDICES

8.1 APPENDIX A. MEASUREMENT JOURNALS

8.1.1 SOUNDCARD VERIFICATION

8.1.1.1 PURPOSE

The purpose is to verify the frequency and phase response for the microphone inputs 1 and 2 on the soundcard. It is expected that the responses are flat and its necessary that the responses for both inputs are equal each other.

8.1.1.2 USED EQUIPMENT

Description	Manufacture and type	AAU serial number
Soundcard	Roland QUAD capture UA-55	2157-79
Laptop with Holmimpulse	Asus	N/A
Various cables	N/A	N/A

Table 8.1 – Used equipment.

8.1.1.3 MEASUREMENT SETUP

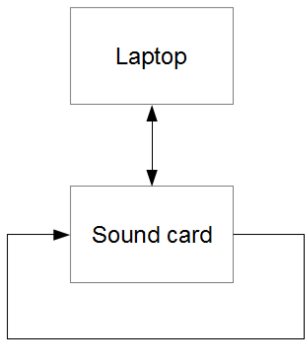


Figure 8.1 – Flowchart of the measurement setup. The soundcard outputs are looped back to the inputs.

8.1.1.4 EQUIPMENT SETTINGS

- Laptop:
    - ***Driver:*** Roland QUAD-capture driver v1.5.1.
    - ***ASIO buffer size:*** 256 samples.
    - ***Sample rate:*** 48KHz.
    - ***Compressor channel 1+2:*** Bypass on.
    - ***Sens channel 1+2:*** 12.
    - ***Phase channel 1+2:*** Off.
    - ***Lo-cut channel 1+2:*** Off.
- Holm impulse:
    - ***Version:*** 1.4.2.0.
    - ***PCM amplitude:*** 0.25.
    - ***Measurement signal :*** Logarithmic sine sweep.
    - ***Start frequency:*** 10Hz.
    - ***Fade in:*** 10ms.
    - ***Fade out:*** 1ms.
    - ***Signal length:*** 2^17 samples.

- Soundcard:
    - ***Sens 1L:*** Adjusted by driver software.
    - ***Sens 2L:*** Adjusted by driver software.
    - ***Auto-sens:*** Controlled by driver software.
    - ***Mono:*** Off.
    - ***Mix:*** Min (Playback).
    - ***Output:*** Max.
    - ***Ground lift:*** Off.
    - ***Phantom:*** Off.
    - ***Hi-Z input 1:*** Off.

8.1.1.5 Procedure

- Setup the equipment according to section 8.1.1.3.
- Adjust the settings according to section 8.1.1.4.
- Perform impulse response measurement for input 1 and save results.
- Perform impulse response measurement for input 2 and save results.

8.1.1.6 Results

The measured impulse, frequency and phase response for the soundcard input 1 and 2 are shown in **Figure 8.2** and **Figure 8.3**. As expected and necessary, the responses are equal each other and the phase and frequency responses are almost flat. The results are approved.

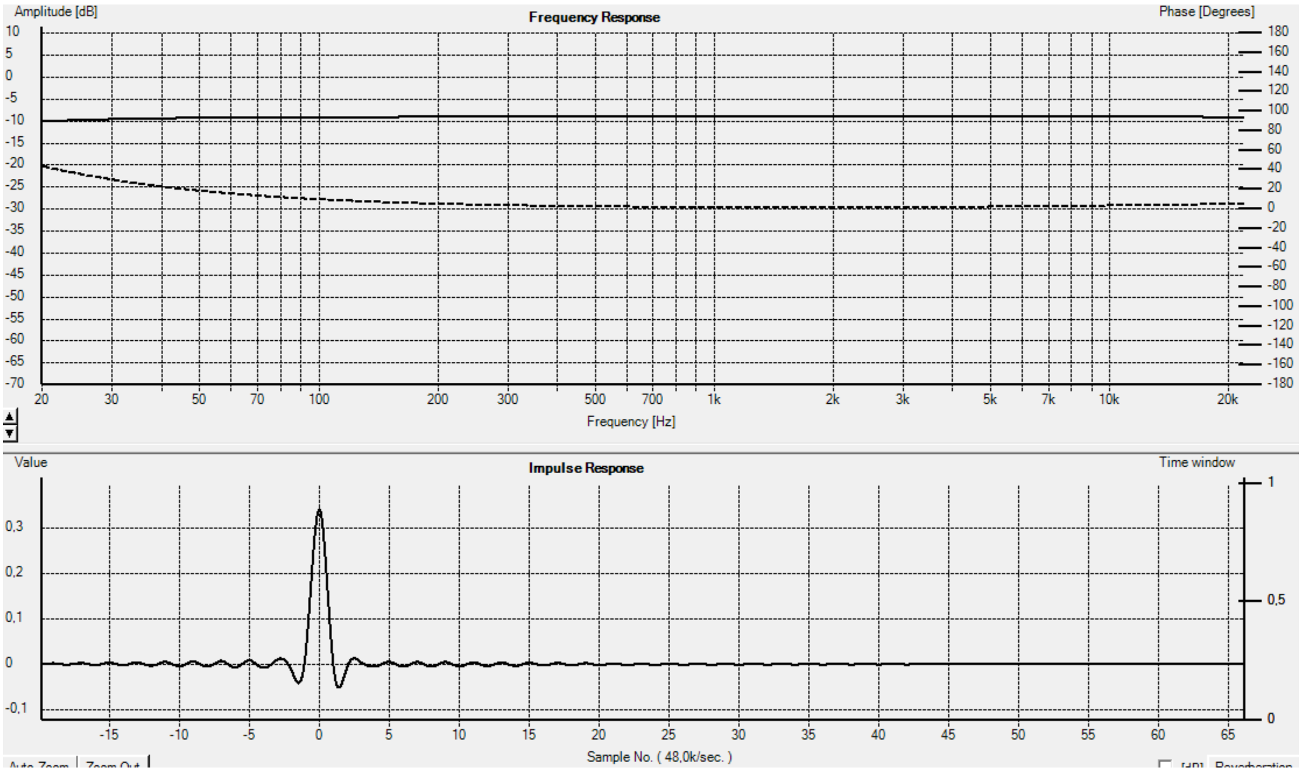


Figure 8.2 – The impulse, frequency and phase response for soundcard input channel 1. The frequency response is the solid line and the phase response is the dashed line.

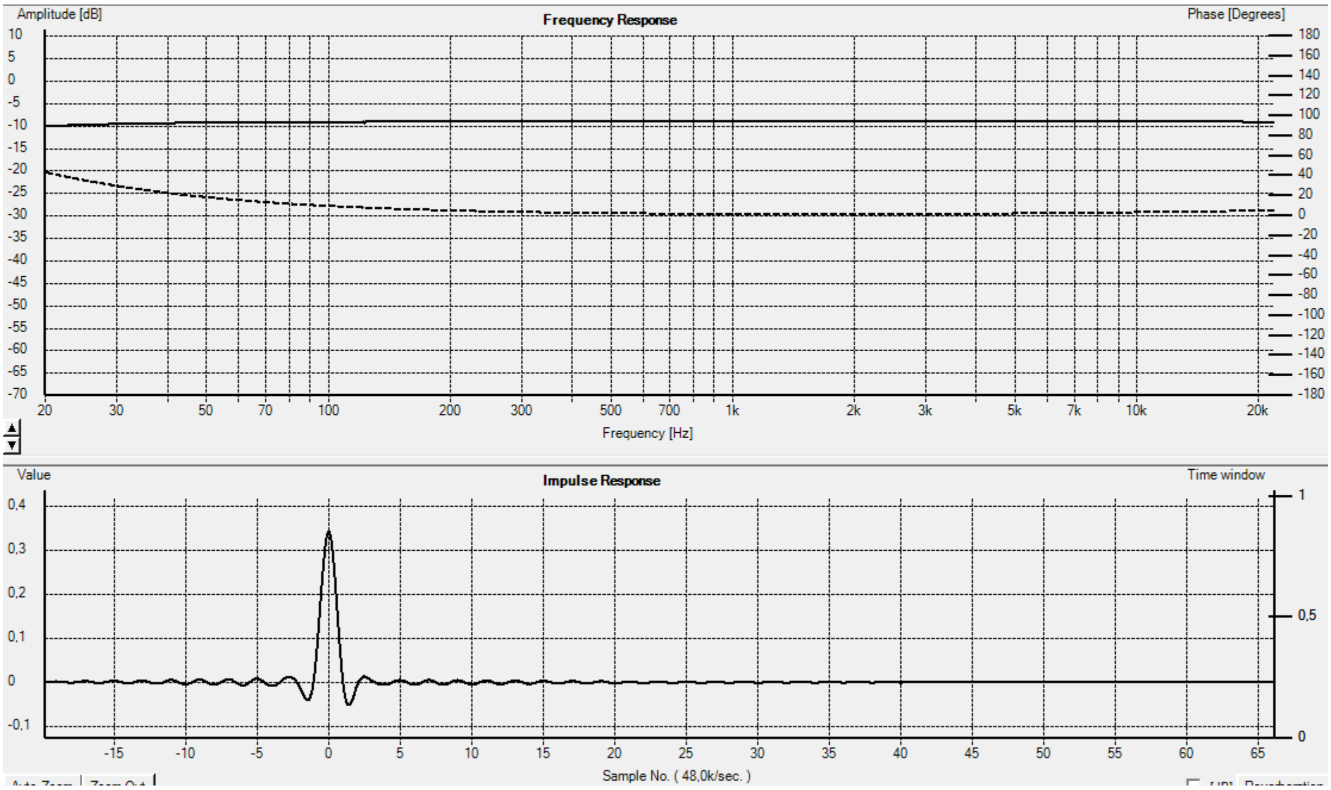


Figure 8.3 – The impulse, frequency and phase response for soundcard input channel 2. The frequency response is the solid line and the phase response is the dashed line.

8.1.2 Impedance measurement: Transfer function in tube method

8.1.2.1 Purpose

The purpose is to measure the impedance of a MDF (medium density fiber) board and a sponge mat using the transfer function method in the impedance tube described in [ISO 10534-2].

8.1.2.2 Test specimens

Specimen	Material	Dimensions [m]	Additional information
1	MDF	0.1 x 0.022	
2	Sponge	0.1 x 0.03	

Table 8.2 - Test specimens. The dimensions are diameter and thickness.

8.1.2.3 Used equipment

Description	Manufacture and type	AAU serial number
Amplifier	B&K 2706	08716-00
Impedance tube	B&K 4002	06619-00
Soundcard	Roland QUAD capture UA-55	2157-79
Laptop with Holmimpulse	Asus	N/A
Thermometer and hygrometer	KM 8004	33192
Various cables	N/A	N/A

Table 8.3 – Used equipment.

8.1.2.4 Measurement setup

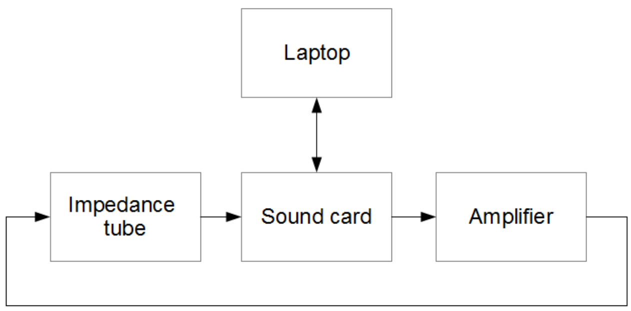


Figure 8.4 – Setup and connections between equipment.

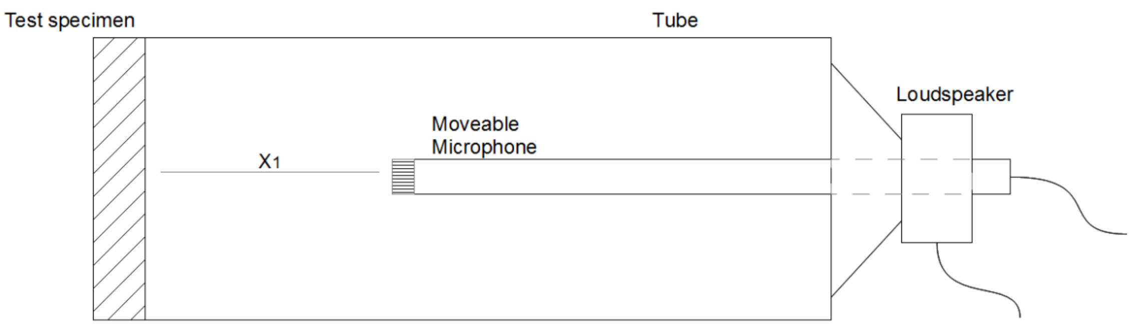


Figure 8.5 – Setup in the impedance tube.

8.1.2.5 EQUIPMENT SETTINGS

- Laptop:
    - ***Driver:*** Roland QUAD-capture driver v1.5.1.
    - ***ASIO buffer size:*** 256 samples.
    - ***Sample rate:*** 48KHz.
    - ***Compressor channel 1:*** Bypass on.
    - ***Sens channel 1:*** 50.
    - ***Phase channel 1:*** Off.
    - ***Lo-cut channel 1:*** Off.
- Holmimpulse:
    - ***Version:*** 1.4.2.0.
    - ***PCM amplitude:*** 0.25.
    - ***Measurement signal :*** Logarithmic sine sweep.
    - ***Start frequency:*** 10Hz.
    - ***Fade in:*** 10ms.
    - ***Fade out:*** 1ms.
    - ***Signal length:*** 2^17samples.
- Soundcard:
    - ***Sens 1L:*** Adjusted by driver software.
    - ***Auto-sens:*** Controlled by driver software.
    - ***Mono:*** Off.
    - ***Mix:*** Min (Playback).
    - ***Output:*** Max.
    - ***Ground lift:*** Off.
    - ***Phantom:*** Off.
    - ***Hi-Z input 1:*** Off.
- Amplifier:
    - ***Attenuation:*** 40dB.
    - ***Gain:*** Max.

8.1.2.6 PROCEDURE

- Setup the equipment according to section 8.1.2.4.
- Adjust the settings according to section 8.1.2.5.
- Mount specimen 1 in the impedance tube.
- Adjust distance X1 to 0.10m.
- Measure temperature and humidity.
- Measure the impulse response using Holmimpulse and export as ascii file.
- Adjust distance X1 to 0.15m.
- Measure the impulse response using Holmimpulse and export as ascii file.
- Repeat for specimen 2.

8.1.2.7 RESULTS

The temperature was measured to 24^{o}C and the humidity to 23%. All measured data are available on the *DVD\Measurement data\Transfer function method in tube* and using *DVD\Matlab code\ImpedanceTube.m* the reflection factors and impedances are calculated for the sponge mat and MDF board.

8.1.3 Impedance measurement: Two microphones and ambient noise method. Iteration 1

8.1.3.1 Purpose

The purpose is to measure the impedance of a MDF (medium density fiber) board and a sponge material using 2 microphones and ambient noise based on the paper by [Takahashi, 2004]. Different configurations are carried out in order to analyze the performance of the method.

8.1.3.2 Test specimen

Specimen	Material	Dimensions [m]	Additional information
1	MDF	1.18 x 1.22 x 0.022	
2	Sponge	1.8 x 1 x 0.03	

Table 8.4 – Test specimen.

8.1.3.3 Used equipment

Description	Manufacture and type	AAU serial number
Amplifier	Pioneer A-656	08700-00
Loudspeaker	B&W DM601 S2	2144-00
2 Microphones	GRAS 40EN	56539 & 56528
2 Preamps	B&K 2669	56510 & 56509
Phantom power supply	B&K 2804	07304-00
Soundcard	Roland QUAD capture UA-55	2157-79
Laptop with matlab	Asus	N/A
Calibrator	B&K	08373
Thermometer and hygrometer	KM 8004	33192
Various stands and cables	N/A	N/A
Room	Standard stereo listening room	B4-107

Table 8.5 – Used equipment.

8.1.3.4 Measurement setup

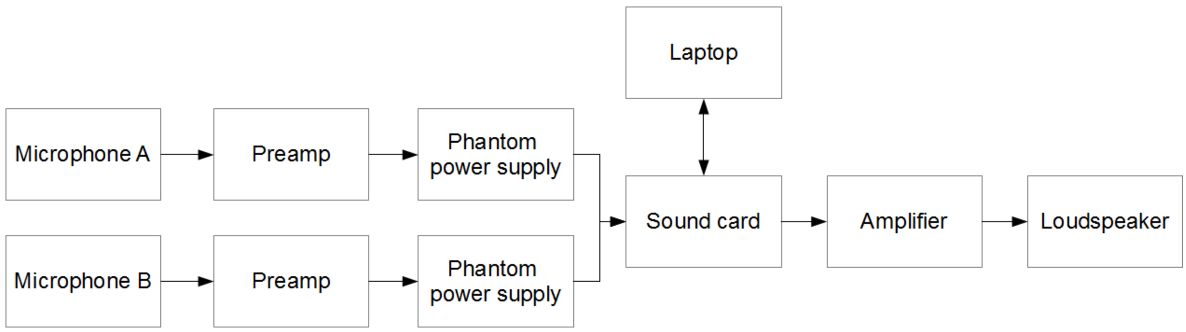


Figure 8.6 – Setup and connections between equipment.

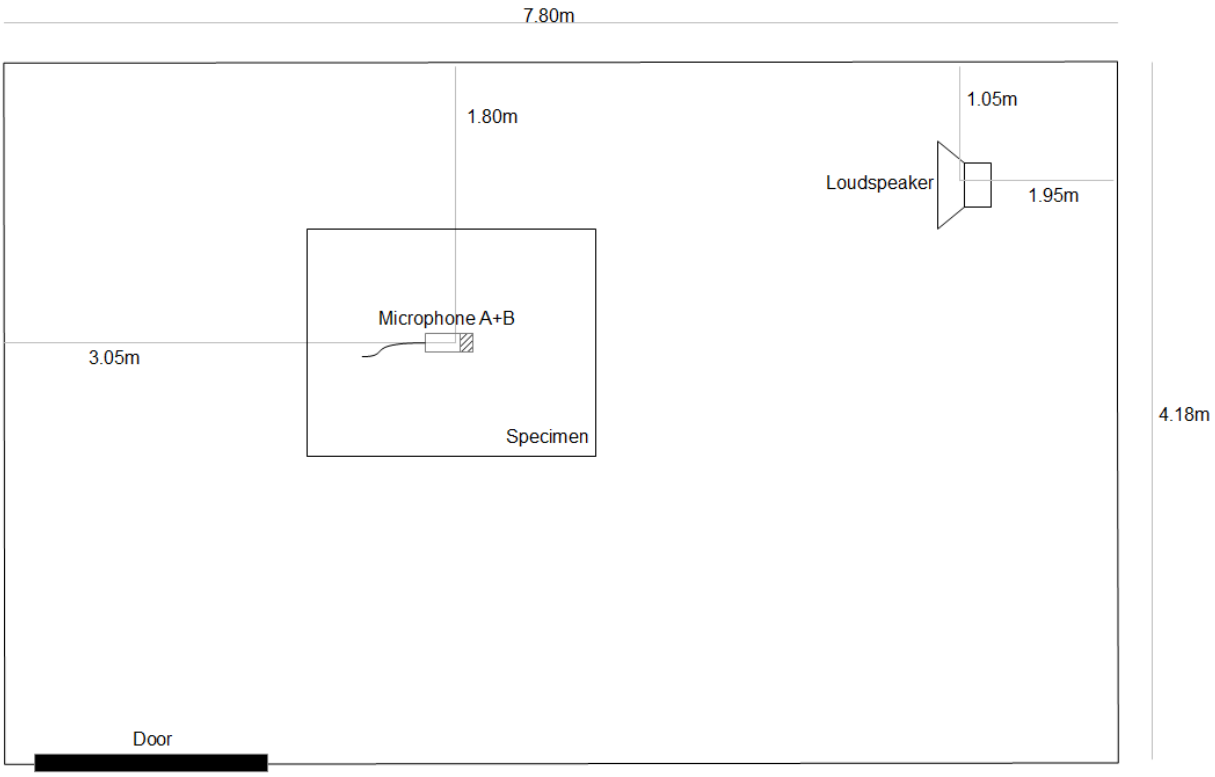


Figure 8.7 – Equipment and test specimen positions in the standard stereo listening room.

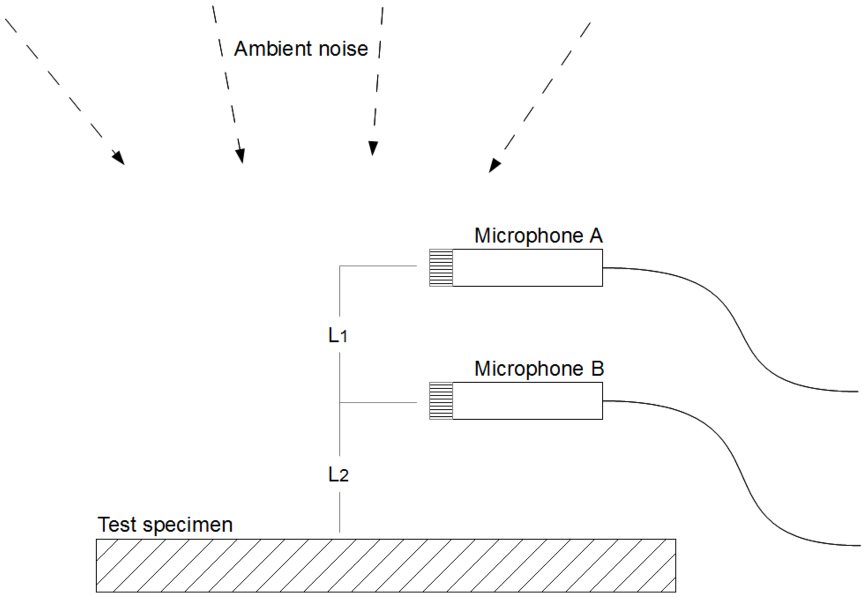


Figure 8.8 – Microphones and test specimen setup.

8.1.3.5 Equipment settings

- Laptop:
    - ***Driver:*** Roland QUAD-capture driver v1.5.1.
    - ***ASIO buffer size:*** 256 samples.
    - ***Sample rate:*** 48KHz.
    - ***Compressor channel 1+2:*** Bypass on.

- ◦ ***Sens channel 1+2:*** 34.
- ◦ ***Phase channel 1+2:*** Off.
- ◦ ***Lo-cut channel 1+2:*** Off.

- Fl studio:
  - ***Bit depth:*** 24bit.
- Soundcard:
  - ***Sens 1L:*** Adjusted by driver software.
  - ***Sens 2L:*** Adjusted by driver software.
  - ***Auto-sens:*** Controlled by driver software.
  - ***Mono:*** Off.
  - ***Mix:*** Min (Playback).
  - ***Output:*** Max.
  - ***Ground lift:*** Off.
  - ***Phantom:*** Off.
  - ***Hi-Z input 1:*** Off.
- Amplifier:
  - ***Output gain:*** 40.
  - ***Direct:*** On.

8.1.3.6 PROCEDURE

- Setup the equipment according to section 8.1.3.4.
- Adjust the settings according to section 8.1.3.5.
- Measure temperature, humidity and pressure.
- Apply the calibrator to microphone A and record 10s of calibrator sound. Repeat for microphone B.
- Record ambient noise for 1min.
- Mount specimen 1, use noise no. 1 explained in **Table 8.6** and distance configuration no. 1 explained in **Table 8.7**.
- Record 30s and repeat for interchanged microphone positions.
- Repeat for both specimen, noise no. and distance configurations.

Noise no.	Noise source	Noise type
1	Ambient	Ambient
2	Speaker	Pink noise. Full band
3	Speaker	Pink noise. Limited band
4	Speaker	Pink noise. Full band. Pulsing
5	Speaker	Pink noise. Limited band. Pulsing

Table 8.6 – Noise sources and types.

Distance configuration no.	L1 [m]	L2 [m]
1	0.023	0.02
2	0.023	0.07

Table 8.7 – Distance configutations.

8.1.3.7 RESULTS

The temperature was measured to 20^{o}C and the humidity to 23%. All measured data are available on the *DVD\Measurement data\Two microphones and ambient noise method. Iteration 1* and using *DVD\Matlab code\TwoMicsMethod3.m* the reflection factors and impedances are calculated for the sponge mat and MDF board.

8.1.4 IMPEDANCE MEASUREMENT: TWO MICROPHONES AND AMBIENT NOISE METHOD. ITERATION 2

8.1.4.1 PURPOSE

The purpose is to measure the impedance of a MDF (medium density fiber) board and a sponge material using 2 microphones and ambient noise based on the paper by [Takahashi, 2004] and generation of ambient noise using a loudspeaker and the reverberation field in a room.

8.1.4.2 TEST SPECIMEN

Specimen	Material	Dimensions [m]	Additional information
1	MDF	1.18 x 1.22 x 0.022	
2	Sponge	1.8 x 1 x 0.03	

Table 8.8 – Test specimen.

8.1.4.3 USED EQUIPMENT

Description	Manufacture and type	AAU serial number
Amplifier	Pioneer A-656	08700-00
Loudspeaker	B&W DM601 S2	2144-00
2 Microphones	GRAS 40EN	56539 & 56528
2 Preamps	B&K 2669	56510 & 56509
Phantom power supply	B&K 2804	07304-00
Soundcard	Roland QUAD capture UA-55	2157-79
Laptop with matlab	Asus	N/A
Calibrator	B&K	08373
Thermometer and hygrometer	KM 8004	33192
Various stands and cables	N/A	N/A
Room	Standard stereo listening room	B4-107

Table 8.9 – Used equipment.

8.1.4.4 MEASUREMENT SETUP

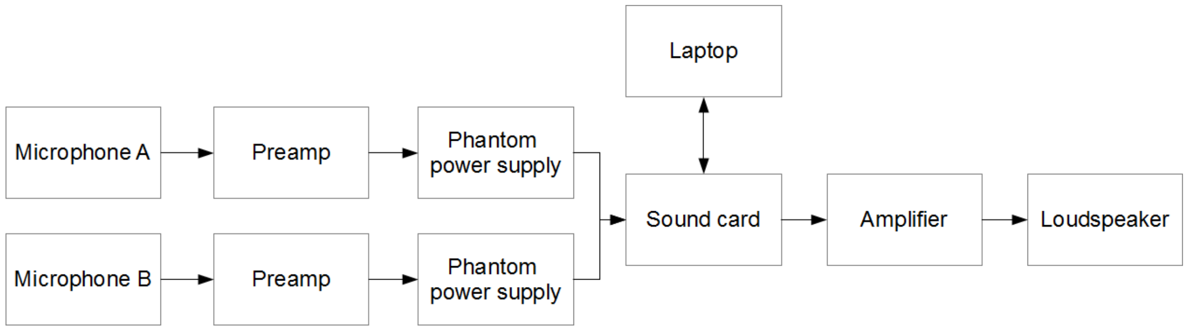


Figure 8.9 – Setup and connections between equipment.

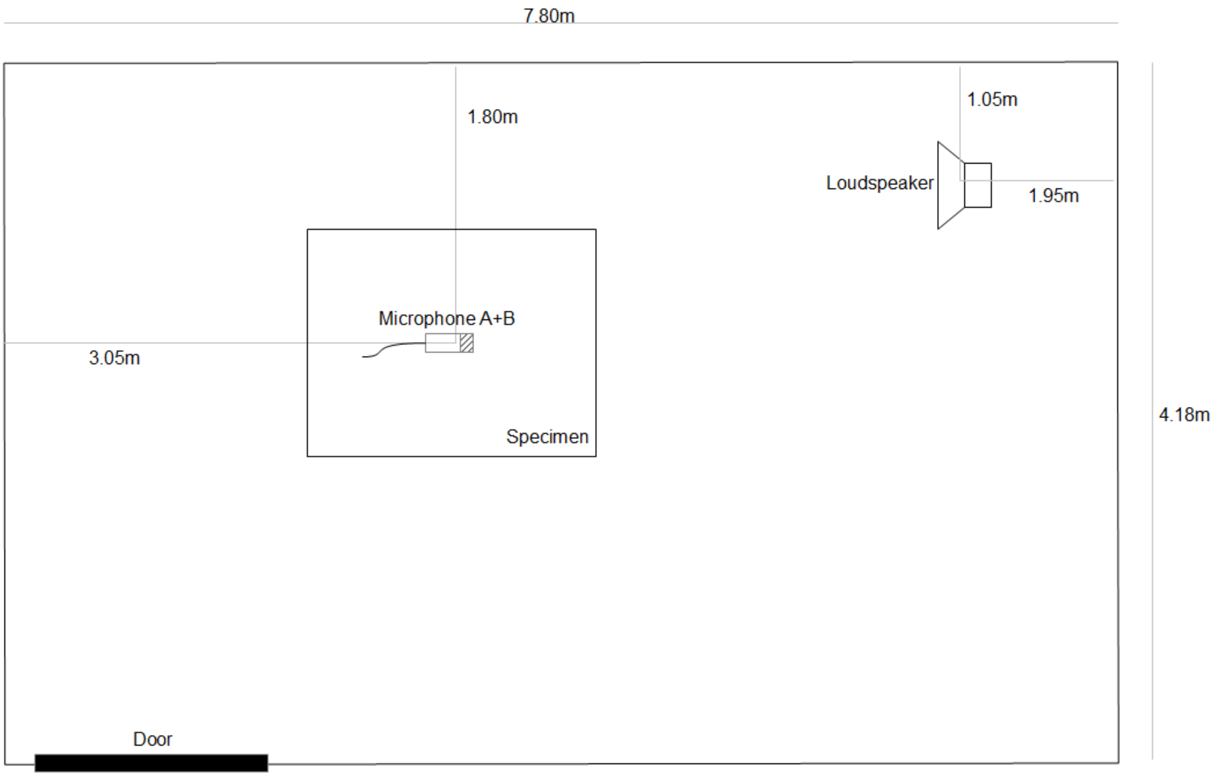


Figure 8.10 – Equipment and test specimen positions in the standard stereo listening room.

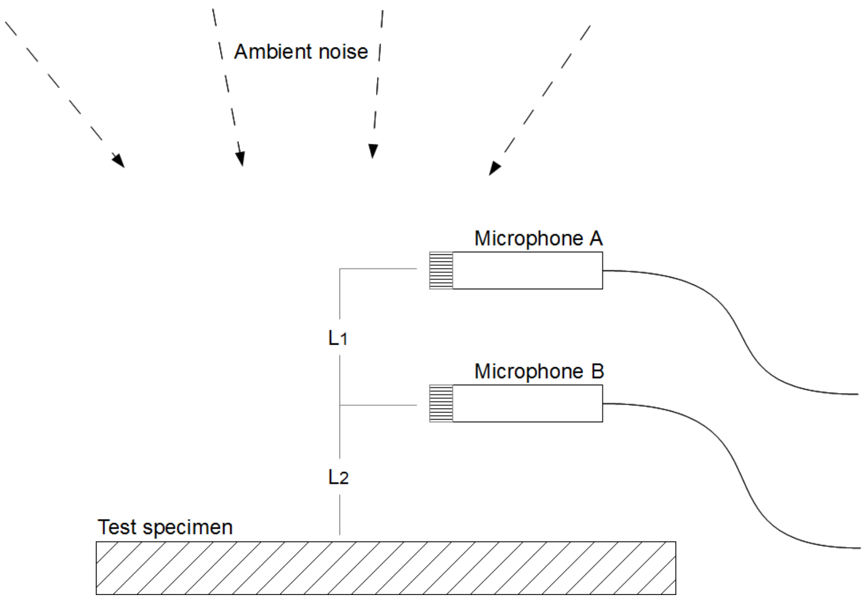


Figure 8.11 – Microphones and test specimen setup.

8.1.4.5 Equipment settings

- Laptop:
  - ***Driver:*** Roland QUAD-capture driver v1.5.1.
  - ***ASIO buffer size:*** 256 samples.
  - ***Sample rate:*** 48KHz.
  - ***Compressor channel 1+2:*** Bypass on.

  - ***Sens channel 1+2:*** 34.
  - ***Phase channel 1+2:*** Off.
  - ***Lo-cut channel 1+2:*** Off.
- Soundcard:
  - ***Sens 1L:*** Adjusted by driver software.
  - ***Sens 2L:*** Adjusted by driver software.
  - ***Auto-sens:*** Controlled by driver software.
  - ***Mono:*** Off.
  - ***Mix:*** Min (Playback).
  - ***Output:*** Max.
  - ***Ground lift:*** Off.
  - ***Phantom:*** Off.
  - ***Hi-Z input 1:*** Off.
- Amplifier:
  - ***Output gain:*** *60*.
  - ***Direct:*** On.

8.1.4.6 Procedure

- Setup the equipment according to section 8.1.3.4.
- Adjust the settings according to section 8.1.3.5.
- Measure temperature, humidity and pressure.
- Apply the calibrator to microphone A and use the *DVD\Matlab code\TwoMicsMethod5calibration.m* to obtain calibration factor. Repeat for microphone B and apply the calibration factors in *DVD\Matlab code\TwoMicsMethod5recording.m*.
- Mount specimen 1 and measure using *DVD\Matlab code\TwoMicsMethod5recording.m* and distance configuration no. 1 explained in **Table 8.7**.
- Interchange microphone positions and measure with distance configuration no. 1 again.
- Repeat for both specimen and distance configurations.

Distance configuration no.	L1 [m]	L2 [m]
1	0.023	0.02
2	0.023	0.07
3	0.023	0.20

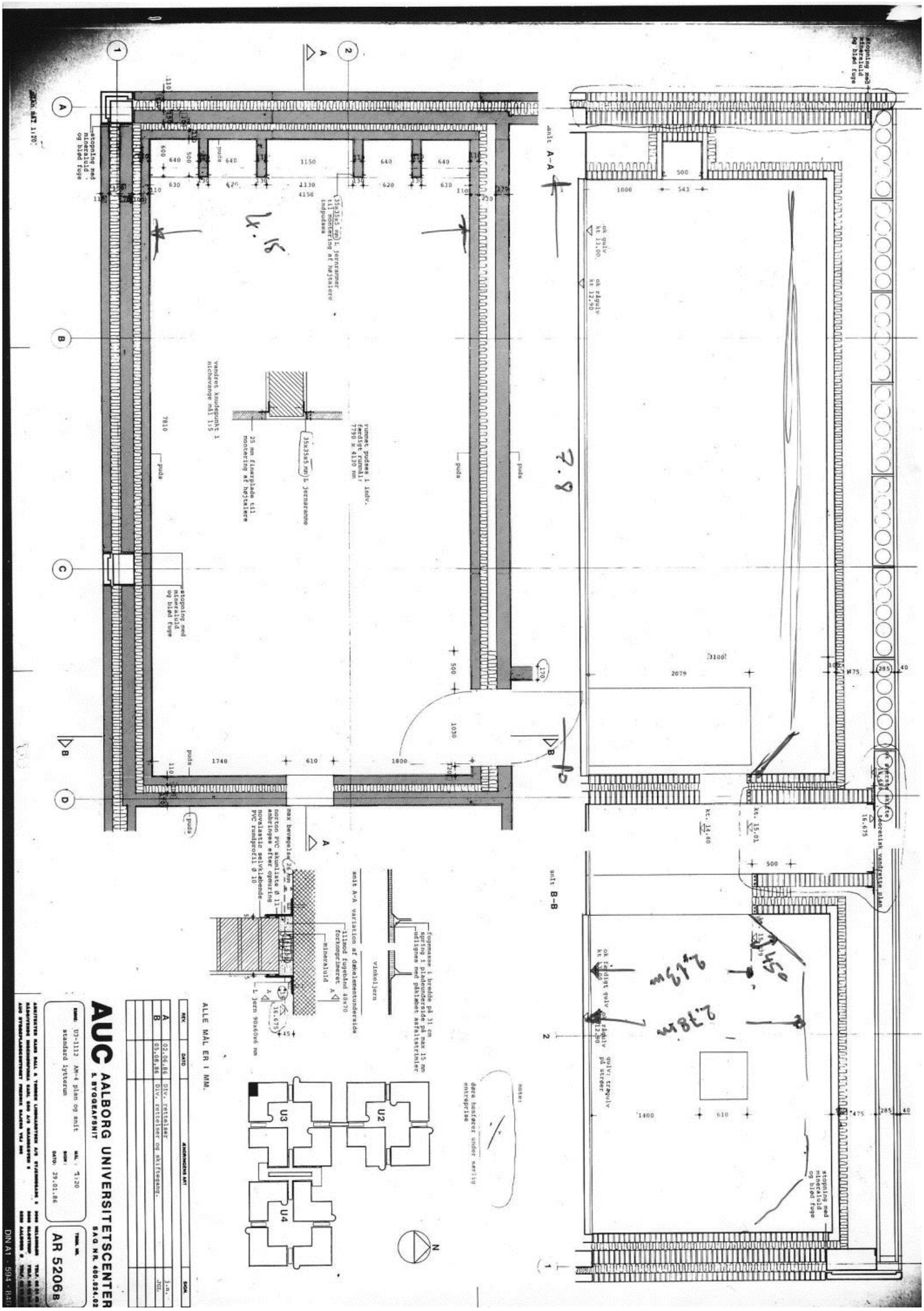
Table 8.10 – Distance configutations.

8.1.4.7 Results

The temperature was measured to 21°C and the humidity to 23%. All measured data are available on the *DVD\Measurement data\Two microphones and ambient noise method. Iteration 2* and using *DVD\Matlab code\TwoMicsMethod5.m* the reflection factors and impedances are calculated for the sponge mat and MDF board.

8.2 Appendix B. The standard stereo listening room drawing

[Listening room]



8.3 Appendix C. DVD content

- Matlab codes
- Measurement data
  - Transfer function method in tube
    - MDF
    - Sponge
    - Results
  - Two microphones and ambient noise method. Iteration 1
    - MDF
    - Sponge
    - Results
  - Two microphones and ambient noise method. Iteration 2
    - MDF
    - Sponge
    - Results
- Time schedules
- Report and appendices as PDF

8.4 Appendix D. Glossary of symbols

a	Diameter [m]
A	Total absorption [m^2]
c	Speed of sound [m/s]
d	Diameter [m]
f	Frequency [Hz]
H	Height [m]
	Directional factor
	Transfer function
I	Intensity [w/m^2]
k	Wavenumber
L	Length [m]
	Sound pressure level [dB re. 20µPa]
p	Pressure [Pa]
r	Radius [m]
R	Reflection factor
s	Standing wave ratio
S	Surface area [m^2]
t	Time [s]
T	Temperature [^oC]
	Reverberation time [s]
T_k	Temperature [^oK]
u	Particle velocity [m/s]
V	Volume [m^3]
W	Width [m]
x	Distance [m]
z	Specific acoustic impedance [Pa·s/m]
α	Absorption coefficient
θ	Angle [rad]
λ	Wavelength [m]
ρ	Density [Kg/m^3]
ρ_0	Density of air [Kg/m^3]
ω	Angular frequency [rad/s]

8.5 Appendix E. References

8.5.1 Standards

[ISO 354]: Acoustics. Measurement of sound absorption in a reverberation room.

[ISO 10534-1]: Acoustics. Determination of sound absorption coefficient and impedance in impedance tubes. Part 1: Method using standing wave ratio.

[ISO 10534-2]: Acoustics. Determination of sound absorption coefficient and impedance in impedance tubes. Part 2: Transfer-function method.

[ISO 13472-1]: Acoustics. Measurements of sound absorption properties of road surfaces in situ. Part 1: Extended surface method.

[ISO 13472-1]: Acoustics. Measurements of sound absorption properties of road surfaces in situ. Part 2: Spot method for reflective surfaces.

[IEC 60268-13]: Sound system equipment – part 13.Listening tests on loudspeakers.

[ANSI S1.11-1986]: American national standard. Specification for octave-band and fractional-octave-band analog and digital filters.

8.5.2 Books

[Toole, 2008]: Floyd E. Toole. Sound reproduction, loudspeakers and rooms. Focal press. 2008.

[Kinsler, 1999]: Lawrence E. Kinsler, Austin R. Frey, Alan B. Coppens, James v. Sander. Fundamentals of acoustics, 4th edition. Wiley. 1999.

[Kuttruff, 2009]: Heinrich Kuttruff. Room acoustics, 5th edition. Spon press. 2009.

[Moore, 2012]: Brian C J Moore. An introduction to psychology of hearing. 6th edition. Emerald. 2012

8.5.3 Papers

[Bech, 1994]: Søren Bech. Timbral aspects of reproduced sound in small rooms. I. 1994.

[Bech, 1996]: Søren Bech. Timbral aspects of reproduced sound in small rooms. II. 1996.

[Mellers & Nocke, 2004]: Volker Mellert & Christian Nocke. Applications of in-situ measurement techniques of absorption. Oldenburg university, institute for physics. 2004.

[Piechowicz, 2011]: Janusz Piechowicz. Estimating surface acoustic impedance with the inverse method. International Journal of Occupational Safety and Ergonomics (JOSE). 2011.

[Takahashi, 2004]: Y. Takahashi, T. Otsuru & R. Tomiku. In situ measurements of surface impedance and absorption coefficients of porous materials using two microphones and ambient noise. Elsevier. 2004.

[Farina & Torelli, 1997]: Angelo Farina & Anna Torelli. Measurement of the sound absorption coefficient of materials with a new sound intensity technique. 1997.

[Nava,Yasuda, Sato & Sakamoto, 2009]: Gabriel Pablo Nava, Yosuko Yasuda, Yoichi Sato and Shinichi Sakamoto. On the in situ estimation of surface acoustic impedance in interiors of arbitrary shape by acoustical inverse methods. The Acoustical Society of Japan. 2009.

[Tijs, Botts, Bree & Arato, 2009]: Emiel Tijs, Jonathan Botts, Hans-Elias de Bree and Eva Arato. Acoustic particle velocity enabled methods to assess room acoustics. EAA. 2009.

[Evans, 2012]: William Evans. The influence of loudspeaker directivity upon the perception of reproduced sound in domestic listening rooms. PhD thesis. University of Surrey. 2012.

[Cervantes, Prepelita, Peyret & Bonde, 2012]: Pablo Cervantes, Sebastian Prepelita, Paul Peyret and Regnar Oxholm Bonde. Small rooms – Investigation in the relationships between subjective judgments and acoustical properties. Semester project. 3rd semester. Master of acoustics. AAU. 2012.

[Müller & Massarini, 2001]: Sven Müller and Paulo Massarini. Transfer-Function measurement with sweeps. Director's cut including previously unreleased material and some corrections. 2001.

8.5.4 WEBPAGES

[Listening room]: Descriptions and information about the standard stereo listening room at Aalborg University. April 2013.
http://www.es.aau.dk/sections/acoustics/lab-facilities/
http://doc.es.aau.dk/labs/acoustics/facilities/listening_room_stereo/

[Playrec]: Playrec plugin for matlab. May 2013.
http://www.playrec.co.uk/index.php

[B&W]: Manual and specifications for B&W601-S2. May 2013.
http://www.scribd.com/doc/40401/BW-DM601and2-S2-User-Manual

[Gras]: Information and datasheet for the Gras 40EN microphone cartridges. May 2013.
http://www.gras.dk/products/measurement-microphone-cartridge/externally-polarized-cartridges-200-v/g-r-a-s-40en-1-pressure-microphone.html

[Mathworks]: Matlab octave band filter package. May 2013.
http://www.mathworks.com/matlabcentral/fileexchange/69-octave

[Soundcard]: Information, driver and user manual for the Roland quad-capture ua-55 soundcard. April 2013.
http://www.rolandus.com/products/details/1166/features/

[Holmimpulse]: Software for impulse response measurements. April 2013.
http://www.holmacoustics.com/holmimpulse.php

[Mircoflown]: Information about the particle velocity sensor. March 2013.
http://www.microflown.com/

[DMI]: Air pressure Denmark. March 2013.
http://www.dmi.dk/dmi/hoejeste_lufttryk_i_20_aar

[Wiki1]: Article about the phonograph. February 2013.
http://en.wikipedia.org/wiki/Phonograph

[Wiki2]: Density of air calculations. March 2013.
http://en.wikipedia.org/wiki/Density_of_air

[Wiki3]: Timbre definition. April 2012.
http://en.wikipedia.org/wiki/Timbre

[Audyssey]: Room equalization system. February 2013.
http://www.audyssey.com/audio-technology/multeq

[Roomperfect]: Room equalization system. February 2013.
http://steinwaylyngdorf.com/technology-and-innovation/roomperfect