

**Use of a rotating source for acoustic
measurements**

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M.Sc. Thesis

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Delft, June 2000

Abstract

At the Laboratory of Acoustic Imaging and Sound Control, much research is done in the field of room acoustics. The impulse response is widely regarded as the most meaningful characteristic in room acoustics. For a certain combination of source and receiver position, the impulse response is the sound pressure at the receiver position as a function of time generated by an acoustic pulse emitted at the source position. A major requirement for the source is omnidirectionality: the source must radiate acoustic energy equally in all directions. In 1980, Beentjes [2] has developed an acoustic source consisting of a 32-face source sphere and a separate bass loudspeaker. This source radiates omnidirectionally from 60 to 2800 Hz. Taking in account that the range of interest is from 50 to 5000 Hz, a successor is desirable. In the present research, the utility of a rotating loudspeaker for room response measurements is investigated.

In theory, reproducing a linear frequency sweep with a rotating source results in decomposition of the directivity of the source. The contributions of the monopole, dipole and higher order pole terms to the directivity pattern of the source are separated. A monopole is omnidirectional, thus the response to an omnidirectional source can be calculated.

Simulations confirm the theory, the directivity is decomposed.

Use of a 'normal' linear frequency sweep results in a required measuring time of at least 100 minutes, which is not acceptable. Use of parallel linear frequency sweeps reduces the required time to a couple of minutes. A disadvantage of parallel sweeps is that the position of the turntable at the beginning must be the same as the position at the ending of the measurement. This requires a very constant angular velocity of the turntable.

Doppler effects have a large influence on the impulse response. Doppler effects result in a complex directivity, resulting in many higher order pole contributions to the impulse response. In addition, the power spectrum gets the shape of a sinc-function, with a smaller central peak for a higher angular velocity.

Experiments confirm the results of the simulation. The impulse response of a rotating loudspeaker under anechoic conditions shows many peaks at the positions

corresponding to monopole, dipole and higher order poles. The S/N ratio in room measurements is 65 dB at best and 40 dB at worst, which is a very promising result. Using better equipment, the S/N ratio can become even higher.

The rotating source in combination with a frequency sweep is a promising tool for impulse response measurements in rooms. The idea of creating a monopole with one loudspeaker is elegant and the achieved S/N ratio is very good. A negative point is the high demand on the measuring equipment (a very constantly rotating turntable is necessary).

A problem that has not been investigated in this research, is the extension to 3D. A rotating speaker has only a monopole directivity component in a horizontal plane perpendicular to the rotation axis. Radiation in the direction of the rotation axis is not influenced by the rotation. Therefore, a loudspeaker rotating on one axis does not radiate isotropically.

Samenvatting

Bij de vakgroep Akoestische en Seismische Technieken (AK-ST) van de Faculteit Technische Natuurwetenschappen van de TU Delft, wordt veel onderzoek gedaan naar zaalakoestiek. De pulsresponsie wordt algemeen gezien als de meest zinvolle akoestische parameter in de zaalakoestiek. Voor een bepaalde combinatie van bron- en ontvangerpositie, is de pulsresponsie gelijk aan de geluidsdruk als functie van de tijd gemeten door de ontvanger.

Een belangrijke eis aan de bron voor het meten van pulsresponsies in een zaal is omnidirectionaliteit. Een omnidirectionele bron straalt naar alle richtingen uit evenveel vermogen uit. In 1980 heeft Beentjes [2] een geluidsbron ontwikkeld die bestaat uit een 32-vlaksbron en een lage-tonen-luidspreker. Deze bron is omnidirectioneel van 60 tot 2800 Hz. Aangezien het interessante gebied zich van 50 tot 5000 Hz uitstrekt, is een opvolger wenselijk. In dit onderzoek is gekeken naar de bruikbaarheid van een roterende bron voor het meten van pulsresponsies in een zaal.

Theoretisch leidt het weergeven van een lineaire frequentiesweep met een roterende bron tot ontbinding van de richtingskarakteristiek van de bron. De bijdragen van de monopool, dipool en hogere orden polen aan de richtingskarakteristiek zijn afzonderlijk zichtbaar in de pulsresponsie. Een monopool is omnidirectioneel, dus door deze scheiding kan de pulsresponsie van de zaal op een omnidirectionele bron verkregen worden.

Simulaties bevestigen de theorie dat de richtingskarakteristiek wordt ontbonden. Indien een 'gewone' lineaire frequentiesweep gebruikt wordt, is de meettijd onacceptabel lang, minstens 100 minuten. Door parallelle sweeps te gebruiken, wordt de meettijd gereduceerd tot een paar minuten. Een nadeel van parallelle sweeps is dat de positie van de draaitafel aan het begin van de meting gelijk moet zijn aan de positie aan het einde van de meting. Dit vereist een zeer constante rotatiesnelheid van de draaitafel.

Doppler effecten spelen ook een belangrijke rol in de pulsresponsie. Dopplereffecten resulteren in een complexe richtingskarakteristiek, wat resulteert in veel hogere orden-polen-bijdrage in de pulsresponsie. Daarnaast vervormen de

Dopplereffecten het vermogensdichtheidspectrum van de pulsresponsie, hogere frequenties worden sterk gedempt.

Metingen bevestigen de resultaten van de simulaties. De pulsresponsie van een roterende luidspreker in de dode kamer bevat vele pieken. Die pieken liggen precies op de plaatsen die horen bij een monopool, dipool of een hogere orde pool. Bij metingen in een gewone ruimte, is de signaal-ruis verhouding minstens 40 dB en in gunstige gevallen 65 dB. De resultaten van de roterende bron zijn beslist beter dan die van de 'Beentjes bol'.

Een roterende bron met een lineaire frequentiesweep is een veelbelovende methode om de pulsresponsie van een zaal te meten. De methode om met één luidspreker een monopool te maken is zeer elegant. De signaal-ruis verhouding is zeer goed. Een nadeel is de hoge eis die gesteld wordt aan de draaitafel. Een probleem wat nog niet onderzocht is de uitbreiding van een monopool in 2 dimensies (zoals in dit onderzoek), naar een monopool in 3 dimensies. Voor toepassing van de roterende luidspreker voor pulsresponsie metingen is dit wel een voorwaarde.

Acknowledgements

I would like to thank these people:

<i>Diemer de Vries</i>	For the full support and the patiently waiting for my arrival at (almost) every meeting
<i>Jan Baan</i>	For the idea and useful directions
<i>Donald E. Knuth</i>	For writing T _E X, the base of the solid writing language L ^A T _E X
<i>Wilfred van Rooijen</i>	For the L ^A T _E X-support
<i>Henry den Bok</i>	For the electro-technical advises and the charging of the battery packs
<i>My parents</i>	For the indispensable support throughout my study, but especially for the support on the moments I needed it most
<i>Nadine Vos</i>	For not being jealous of my computer, which got much more attention than she got.

List of Symbols

Δf	bandwidth of the sweep signal
Δt	time between pulses the n-poles in the impulse response, equal to $t_n - t_{n-2}$
Δt^*	slightly changed Δt to accomplish a full number of revolutions during the measuring time
ϕ	angle of the source compared to the reference angle of the source
ψ	angular velocity of the rotating source
$\rho(t)$	distance between the moving source and the stationary receiver
a	distance between the receiver and the rotation axis
c	velocity of sound in air
$d'(\phi)$	directivity as a function of the angle ϕ
$d(t)$	directivity as a function of time ($d(t) = d'(\psi \cdot t)$)
$d_{n,c}(t)$	directivity of a complex n-pole as a function of time
f_h	highest frequency of the sweep
f_l	lowest frequency of the sweep
f_s	sample frequency of the D/A and A/D controller
$h_c(t)$	anechoic impulse response of a rotating source with directivity $d_{n,c}$
i	number of strips the diaphragm is divided into
k	wave number, $k = \omega/c$
m	number of revolutions of the loudspeaker during the measurement
$m(t)$	measured signal, response of the system to $s(t)$

n	index variable for discrete time functions, for example $s[n]$ is the discrete sweep signal
$n(t)$	additional noise in measurements
p	number of sweeps in parallel in $s(t)$
r	distance between the source and the rotation axis
r_d	radius of the emitting diaphragm
$s(t)$	input signal for the system, a sweep
$s_c(t)$	complex sweep signal
T	revolution time of the rotation source, equal to $2\pi/\psi$
t	time
t_n	time of the shifted peak in the impulse response caused by a n-pole
T_{meas}	time of the measurement, equal to $m \cdot T$ and to T_{sweep}/p
T_{sweep}	duration of the sweep
$w[n]$	window function used for tapering the sweep with a Tukey Hanning shape at both edges
$x(t)$	monochromatic source signal of the rotating monopole

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Chapter 1

Introduction

Every room has its own sound. A cathedral sounds different from a concert hall, a lecture hall sounds different from a living room. Every choice for certain acoustics has its pros and cons. For example, a trumpet or violin sounds 'empty' if the reverberation time of the room is too short, the sounds dies immediately. But a speaker cannot make himself audible in a room with much reverberation. Reverberance is an important perceptual acoustic criterion, but not the only one: clarity, ensemble, balance, liveness, warmth and uniformity are just some of the attributes used for qualification of the acoustic properties of a room.

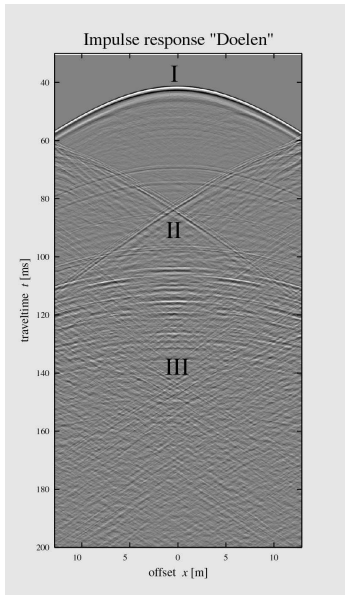
The acoustic properties of a room can be determined objectively or subjectively. For objective measurements, an artificial source and a microphone (array) are used. In subjective measurements, a test subject listens to the sound of an orchestra or speaker and qualifies his perception.

At the Laboratory of Acoustical Imaging and Sound Control, the focus is on objective measurements. The impulse response is widely regarded as the most meaningful characteristic in room acoustics. From the impulse response, most objective acoustical parameters can be derived, like for example the reverberation time.

The impulse response does not exist. For a certain combination of source and receiver position, the impulse response is the sound pressure at the receiver position as a function of time generated by an acoustic pulse emitted at the source position. Using array techniques (fixed source position and repeatedly shifted receiver position), the separate influence of the different reflectors in the room can be calculated. An example of an array measurement of a room is shown in Figure 1.1.

A important requirement for the source for impulse response measurements is omnidirectionality: the source must radiate acoustic energy isotropically, equally into all directions. If the source radiates no power into a certain direction, reflections from that direction will not be taken into account in the impulse response.

Omnidirectionality is hard to achieve, all loudspeakers have a clear directivity. In 1980, Beentjes [2] has developed an acoustic source consisting of a 32-face loud-

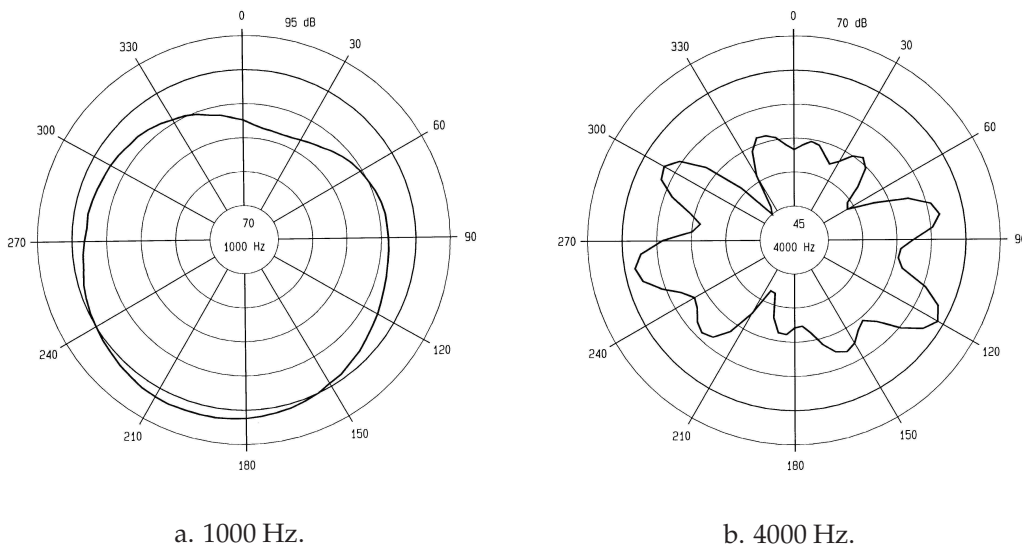


This plot is built up from many impulse responses placed in parallel, with an offset ranging from -15 to +15 meter. The horizontal axis represents the offset of the receiver position and on the vertical axis the travel time from source to receiver is set out.

In the impulse response, three phases can be distinguished. First, the direct sound arrives (I). Ten to hundred seconds after the direct sound (depending on the room), the early reflections arrive (II). The separate reflections of the different walls can be easily distinguished. If the separate reflections cannot be distinguished anymore, the sound field is diffuse (III), reverberation is dominating.

Figure 1.1 The impulse response of the 'Doelen' hall in Rotterdam.

speaker and a separate bass loudspeakerbox. This source radiates isotropic from 60 to 2800 Hz. In Figure 1.2, the directivity pattern of the 'Beentjes bol' is plotted for 2 frequencies: 1000 Hz and 4000 Hz.



a. 1000 Hz.

b. 4000 Hz.

Figure 1.2 The directivity of the 32-faces 'Beentjes bol' for 1000 Hz and 4000 Hz.

The figures show the 'Beentjes bol' is reasonable omnidirectional at 1000 Hz, but operates poorly at 4000 Hz. Taking in account that the range of interest is from 50 to 5000 Hz, a successor of the 'Beentjes bol' is desirable. A main problem of

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multi-face sources is the interference at higher frequencies, being is the cause of the sharp directivity of Figure 1.2b.

In the present research, the applicability of a rotating loudspeaker for impulse response measurements in a room is investigated. A single loudspeaker cannot suffer from interference, thus does not lose its omnidirectionality at higher frequencies. A rotating loudspeaker has, because of the rotation, no front or back, the directivity has axial symmetry around the rotation axis. The acoustic emission is not isotropic, because a loudspeaker radiates more energy to the front direction than to the top or bottom directions.

In this research, only measurements on a horizontal plane perpendicular to the rotation axis are considered. In this somewhat simplified situation, many problems and effects can be investigated properly. Extension to a 3-D rotating speaker or to a multiple-loudspeaker configuration did not fit into the scope of this research work.

In the present research, the effects of rotation on the impulse response are investigated in theory, in simulations and in experiments. The theory and simulations are treated under anechoic conditions, the experiments were done in anechoic as well as in reverberant situations. The anechoic impulse response is a pulse, the echoic impulse has a time decay.

Chapter 2

Theory

2.1 Introduction

For the determination of the acoustic properties of a room, the impulse response is of major importance. From the impulse response, most acoustical parameters can be derived, like for example the reverberation time. **The** impulse response does not exist. For a certain combination of source and receiver position, the impulse response is the sound pressure at the receiver position as a function of time generated by an acoustic pulse emitted at the source position. This acoustic pulse must be powerful and omnidirectional:

- **powerful** to assure a high signal-to-noise ratio (S/N ratio) in the main part of the impulse response;
- **omnidirectional** to radiate acoustic power equally into all directions.

A powerful pulse is hard to realize, instead a linear frequency sweep is used. A linear frequency sweep contains all frequencies of interest and has a longer duration and therefore can be more powerful than a pulse. The impulse response can be calculated by deconvolution of the response of the room to the sweep with the sweep itself. The principles of convolution and deconvolution will be explained in section 2.2.

Much more difficult is the design of an omnidirectional source, a monopole. In the present research, omnidirectionality is partly achieved by rotating the source. The source is isotropic in a plane perpendicular to the rotation axis, not into all directions. Rotation has a large influence on the impulse response, rotation results in a spherical decomposition of the directivity of the source. A thorough analysis of this decomposition can be found in section 2.3.

2.2 Deconvolution

2.2.1 The impulse response

A linear time invariant system (a LTI system) is fully characterized by its impulse response. All properties of a system can be derived from its impulse response.

Unfortunately, accurate direct measurement of the impulse response is very difficult. For an acceptable signal-to-noise ratio (S/N ratio) in the measurement of the impulse response, a certain average power is required. The pulse $\delta(t)$ is a signal that is equal to zero for $t \neq 0$ and infinite for $t = 0$. In real world, 'infinite' is of course not attainable. This results in the approximation of a pulse by a signal with the highest possible value for the shortest possible period of time and zero elsewhere. This approximation can mostly not fulfill the demands for a certain average power, because of the short period of time in which the energy must be provided. Direct measurement of the impulse response is therefore in most situations not possible.

Fortunately, the impulse response can also be measured indirectly. Not the response of a system to a pulse is measured, but the response to the signal $s(t)$. The result of the measurement, $m(t)$, must be 'compensated' for the signal $s(t)$. Also, for good results, the signal $s(t)$ must meet certain demands with respect to frequency spectrum and average power. The most used choices for $s(t)$ are: the linear frequency sweep, the logarithmic frequency sweep, the Maximum Length Sequence and white or pink noise.

2.2.2 Convolution

To determine the impulse response $h(t)$ of a system, one can measure the response $m(t)$ of the system to the signal $s(t)$. With $n(t)$ being the additional noise, $m(t)$ is given by the convolution of $h(t)$ and $s(t)$ (see Oppenheim [6]):

$$m(t) = h(t) * s(t) + n(t) = \int_{-\infty}^{+\infty} s(\tau) \cdot h(t - \tau) d\tau + n(t). \quad (2.1)$$

Convolution is, because of the integral that has to be calculated for every point, a very time consuming task. Of course, a computer can not calculate a continuous time convolution, but calculates a discrete time convolution. Discrete time is indicated with $[\]$ instead of $()$ and n instead of t . Using the same variable names as in continuous time, equation 2.1 becomes:

$$m[n] = h[n] * s[n] + n(t) = \sum_{k=-\infty}^{+\infty} h[k] \cdot h[n - k] + n[n]. \quad (2.2)$$

For N points, N summations over N points are required, a total of N^2 operations.

More efficient is the performance of the operations in the frequency domain. Convolution in the time domain is equivalent to multiplication in the frequency domain. With $M(\omega)$, $H(\omega)$, $S(\omega)$ and $N(\omega)$ being the Fourier transforms of $m(t)$, $h(t)$, $s(t)$ and $n(t)$ respectively, $M(\omega)$ can be calculated easily (and fast):

$$M(\omega) = H(\omega) \cdot S(\omega) + N(\omega). \quad (2.3)$$

The Fourier transform can be calculated very efficiently using the Fast Fourier Transform [7] algorithm under the condition that N is a power of 2. With $M(\Omega)$, $H(\Omega)$, $S(\Omega)$ and $N(\Omega)$ being the discrete Fourier transforms of $m[n]$, $h[n]$, $s[n]$ and $n[n]$ respectively, convolution can be calculated in the discrete frequency domain:

$$M(\Omega) = H(\Omega) \cdot S(\Omega) + N(\Omega) \quad (2.4)$$

Calculation of the convolution of $h[n]$ and $s[n]$ can be done directly or indirectly, i.e., via multiplication in the frequency domain. Assume $N = 2^p$. Direct calculation of the convolution requires N times N equals $N^2 = 2^{2p}$ operations.

Indirect calculation of the convolution by multiplication in the frequency domain is much more efficient than direct calculation in the time domain. Indirect calculation requires 3 Fourier transforms (2 normal and 1 inverse), each requiring $2pN$ operations and in addition, one multiplication is necessary (N operations). So opposite to the 2^{2p} operations for direct calculation, indirect calculation requires a total of $(6p + 1)N \approx 6p \cdot 2^p$ operations. For $p = 20$ ($\pm 10^6$ points), indirect calculation is approximately 10.000 times more efficient.

2.2.3 Deconvolution

Deconvolution is the inverse of convolution. Using the notation of the previous expressions, $h(t)$ is the deconvolution of $m(t)$ with $s(t)$. Similar to convolution, deconvolution is most efficient via the frequency domain. The inverse of multiplication is division, so intuitively, $H_{SQ}(\omega)$ (estimation of $H(\omega)$ with simple quotient) can be calculated with

$$H_{SQ}(\omega) = \frac{M(\omega)}{S(\omega)}. \quad (2.5)$$

This is not a robust method for deconvolution. If the denominator approaches zero, the quotient explodes. The noise $N(\Omega)$ in $S(\omega)$ can therefore have large influence.

A good and well-accepted solution is the use of the least square criterion, resulting in

$$H_{LS}(\omega) = \frac{M(\omega) \cdot S^*(\omega)}{|S(\omega)|^2 + \tilde{R}_{n,n}(\omega)}, \quad (2.6)$$

with $H_{LS}(\omega)$ as the least square approximation of $H(\omega)$ and $\tilde{R}_{n,n}(\omega)$ as the Fourier transform of the autocorrelation function of the noise $n(t)$.

If the power spectrum is equal to unity ($|S(\omega)| = 1$) in the region of interest and is approximating zero elsewhere, a more robust method can be used to calculate $H_R(\omega)$:

$$H_R(\omega) = M(\omega) \cdot S^*(\omega). \quad (2.7)$$

With this method, $H_R(\omega) = H_{LS}(\omega)$ in the region of interest and $H_R(\omega)$ approaches zero elsewhere. This method is often used in seismics and will be used for deconvolution in the present research.

2.3 The rotating source

2.3.1 Introduction

In the present research, a linear frequency sweep is used for the measurements. A linear frequency sweep fulfills all the demands in the fields of spectrum and power. A rotating loudspeaker reproduces this signal to excite the room that has to be examined. A microphone records the response of the room. Deconvolution of this recorded signal with the input signal results in the impulse response of a system consisting of the rotating source and the room.

The combination of a rotating source and a linear frequency sweep has some special properties. To understand these properties, a detailed look is necessary. In this theory chapter, the influence of the room is not taken into account.

To describe the model and understand the theory, some variables have to be introduced:

ϕ	angle of the source compared to the reference angle of the source
ψ	angular velocity of the rotating source
t	time
$d'(\phi)$	directivity as a function of the angle ϕ
$d(t)$ and $D(\omega)$	directivity as a function of time and its Fourier transform ($d(t) = d'(\psi \cdot t)$)
$s(t)$ and $S(\omega)$	input signal and its Fourier transform
$m(t)$ and $M(\omega)$	measured signal and its Fourier transform

In Figure 2.1 the signal path is illustrated.

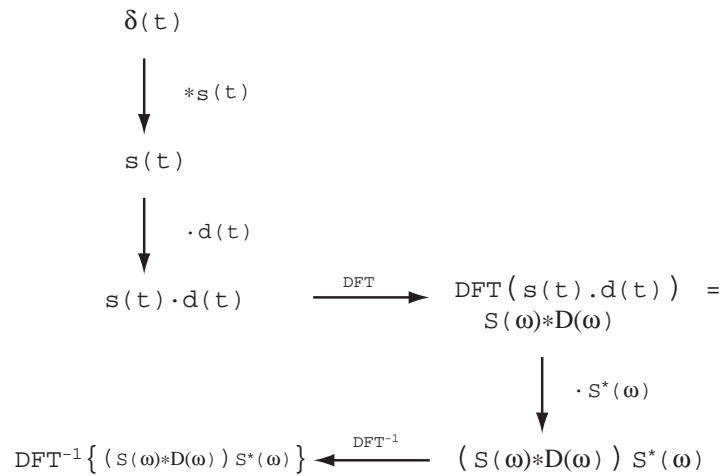


Figure 2.1 The signal path from input to impulse response.

2.3.2 Directivity

Every source has a directivity pattern. The power and phase of the radiated acoustic waves are not equal for all directions, but rather depend on the direction. This directivity pattern is frequency dependent. For most sources the acoustic emission is more bundled (more restricted to a certain direction) at higher frequencies.

For the modeling of stationary sources, standard sources are introduced: monopoles, dipoles and higher order poles. A monopole is a source that radiates acoustic energy isotropically. In 2 dimensions, a dipole has a directivity pattern equal to the cosine of the angle: 1 for $\phi = 0$, 0 for $\phi = \frac{\pi}{2}$ and $\phi = \frac{3\pi}{2}$ and -1 for $\phi = \pi$. In Figure 2.2, the directivity pattern is plotted for a monopole, dipole, quadrupole and a cardioid (summation of a monopole and dipole).

The directivity pattern of any source can be decomposed into monopole, dipole, quadrupole and higher order pole sources (spherical harmonics). This decomposition of the directivity pattern is analogous to Fourier transform of a signal into sines and cosines. Strictly spoken, the directivity pattern of a monopole, dipole or quadrupole is solely built of cosine terms; for the full description of an arbitrary directivity, sine terms will be necessary as well.

The directivity is normally expressed as a function of the angle, but in the case of a rotating source, directivity can also be expressed as a function of time. If the directivity is expressed as a function of time, it can easily be decomposed using the Fourier transform.

Set the angular velocity of the source to ψ , now $d(t)$ will be periodical with period $T = 2\pi/\psi$. The Fourier transform $D(\omega)$ of $d(t) = \cos \psi t$ consists of two delta pulses with magnitude 1/2 at $\omega = \psi$ and $\omega = -\psi$. The Fourier transforms of the

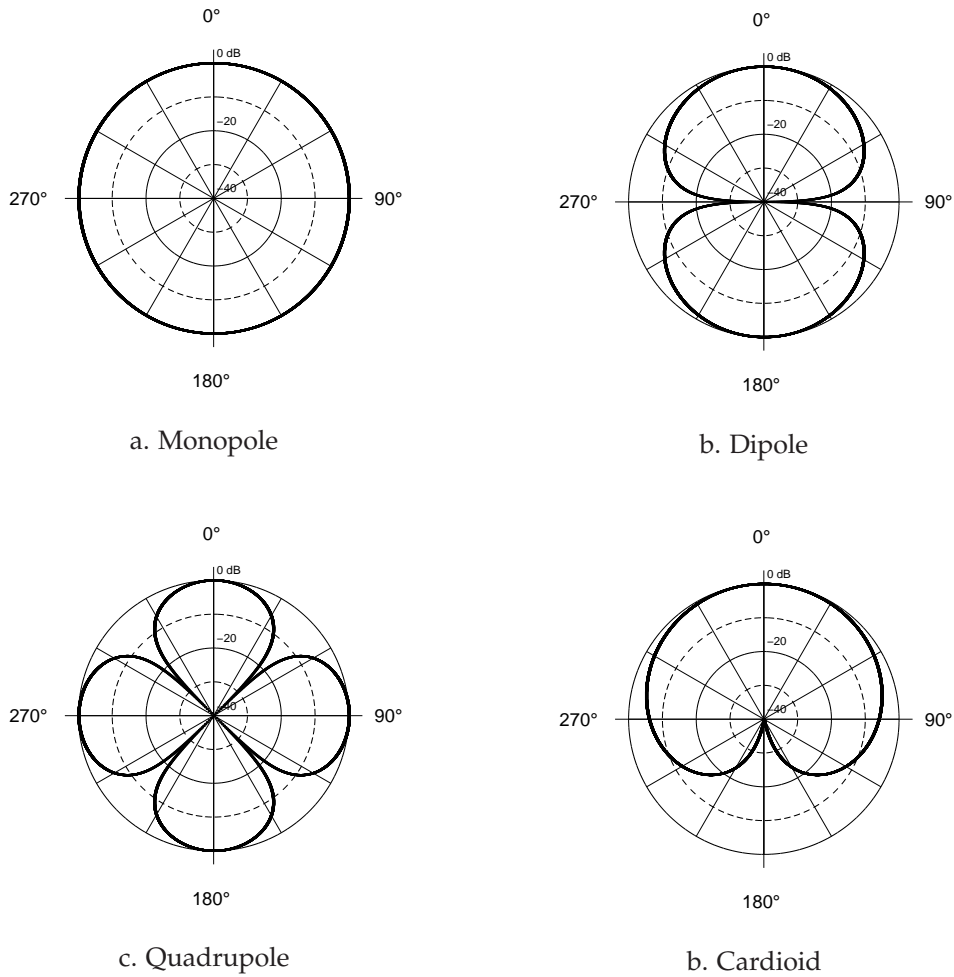


Figure 2.2 The directivity patterns of a monopole (a), a dipole (b), a quadrupole (c) and a cardioid (d).

directivity of different sources rotating with angular velocity ψ as function of time are illustrated in the next table:

Monopole	$d_{mono}(t)=1$	$D_{mono}(\omega)$ consists of 1 delta pulse with magnitude 1 at $\omega = 0$.
Dipole	$d_{di}(t)=\cos(\psi t)$	$D_{di}(\omega)$ consists of 2 delta pulses with magnitude 1/2 at $\omega = \psi$ and $\omega = -\psi$
Quadrupole	$d_{quad}(t)=\cos(2\psi t)$	$D_{quad}(\omega)$ consists of 2 delta with magnitude size 1/2 at $\omega = 2\psi$ and $\omega = -2\psi$
n-pole	$d_n=\cos(\frac{1}{2}n\psi t)$	$D_n(\omega)$ consists of 2 delta pulses with magnitude 1/2 at $\omega = \frac{1}{2}n\psi$ and $\omega = -\frac{1}{2}n\psi$

The general formula for the n-pole suggest that a zeropole would be a more consequent naming for a monopole. But monopole is the widely used term and therefore will be used in the present research.

The number of full revolutions does not influence the position of the delta pulses, the 'frequency' of the directivity remains the same. The position of the delta pulses are solely determined by the speed the cosines of the poles are 'passing by'.

2.3.3 From input to output, step by step

To accomplish a better understanding of the consequences of the successive steps described in Figure 2.1, a detailed look is necessary. To give an impression of the consequences, the intermediate results of the successive steps will be illustrated with figures.

A linear frequency sweep is a signal with a duration T_{sweep} and a frequency starting at the lowest frequency f_l and rising linearly to the highest frequency f_h . Mathematically, the sweep $s(t)$ is given by

$$s(t) = \cos\left(2\pi f_l t + \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}} t^2\right), \quad 0 \leq t \leq T_{sweep} \quad (2.8)$$

Differentiation of the argument of the sine shows that the frequency rises linearly from f_l to f_h .

The Fourier transform of a real signal always consists of a part with negative and a part with positive frequencies. To avoid the negative frequencies, analytical signals are used (see [8]). This results in a complex input signal and a complex n-pole. The complex sweep $s_c(t)$ is given by

$$s_c(t) = e^{j\left(2\pi f_l t + \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}} t^2\right)} \quad 0 \leq t \leq T_{sweep}. \quad (2.9)$$

The advantage of using analytical signals instead of real signals can be found in the simplicity of the Fourier transform. The Fourier transform of equation 2.9 is only non-zero for $f_l < f < f_h$.

Analogous to the complex sweep, the directivity of a complex n-pole $d_{n,c}(t)$ can be calculated:

$$d_{n,c}(t) = e^{j\frac{1}{2}n\psi t}, \quad (2.10)$$

with as the Fourier transform a pulse at $\omega = \frac{1}{2}n\psi$ only.

The Fourier transforms of equations 2.9 and 2.10 are plotted in Figure 2.3 and 2.4 respectively. The angular velocity is $\psi = 200 \cdot 2\pi$ rad/s: the source is rotating at an angular velocity of 200 rpm (not a realistic speed, only used by way of illustration). The other parameters are: $f_l = 50$ Hz, $f_h = 1000$ Hz and $T_{sweep} = 1$ s. For the directivity of the source in the figures, a complex dipole is chosen ($n = 2$).

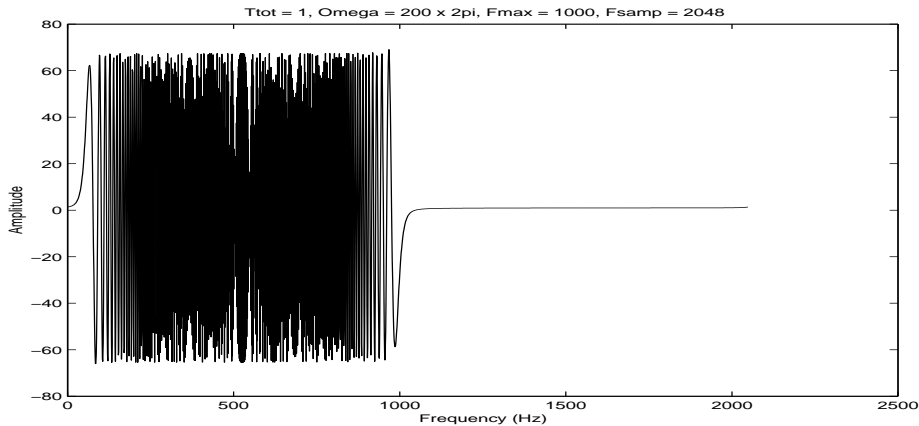


Figure 2.3 Fourier transform of the complex linear frequency sweep: $S_c(\omega)$.

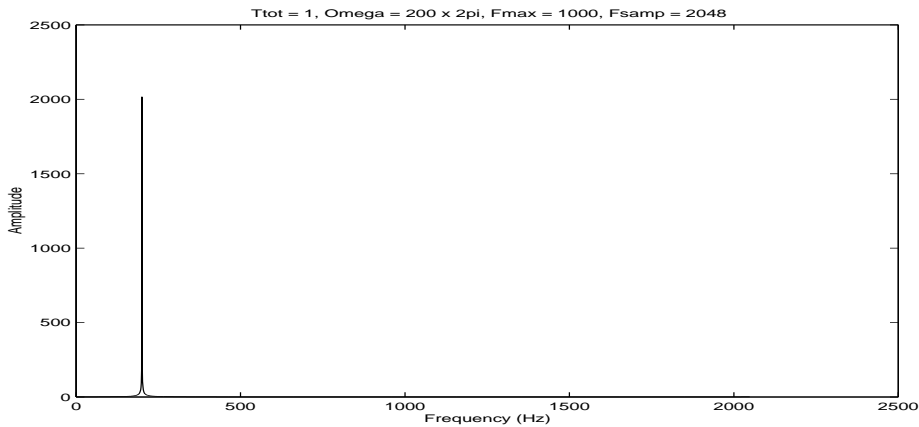


Figure 2.4 Fourier transform of the directivity of a complex dipole: $D_{2,c}(\omega)$.

Multiplication of the input signal $s_c(t)$ by the directivity $d_{n,c}(t)$ in the time domain (see Figure 2.1) is equivalent to convolution in the frequency domain. In the frequency domain, the result of the convolution of $D_{n,c}(\omega)$ and $S_c(\omega)$ is equal to $S_c(\omega)$ shifted over 200 Hz (the ‘rotational frequency’ of the source). In Figure 2.5, this shift is clearly visible.

To obtain the anechoic impulse response $h_c(t)$, the output signal $s_c(t) \cdot d_{n,c}(t)$ has to be deconvoluted with the input signal, $s_c(t)$. For deconvolution, equation 2.7 is used. Use of this equation is justified: the power spectrum is almost flat in the region of interest. Inverse Fourier transform of the result of the deconvolution yields the impulse response, shown in Figure 2.6.

At first sight, this seems a little strange. The impulse response of the system is still a pulse, as expected, but now shifted to $t = -0.21s$. To understand this shift, a more extensive mathematical foundation of the process is required.

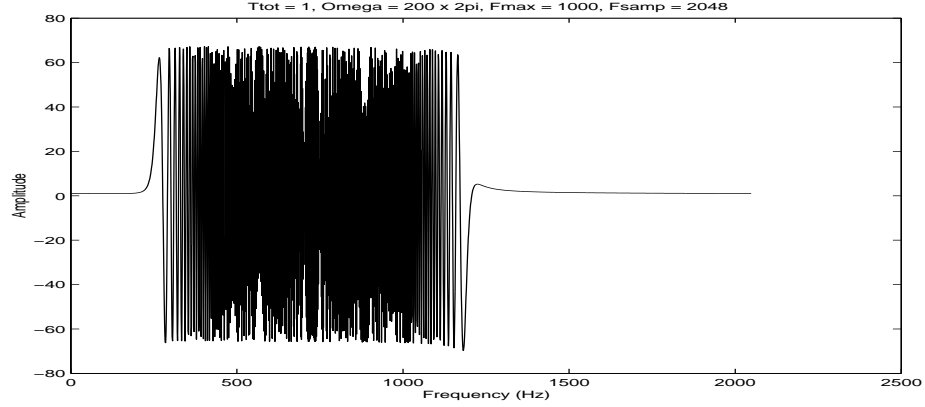


Figure 2.5 Fourier transform of the product of $s_c(t)$ and $d_{2,c}(t)$: $S_c(\omega) * D_{2,c}(\omega)$.

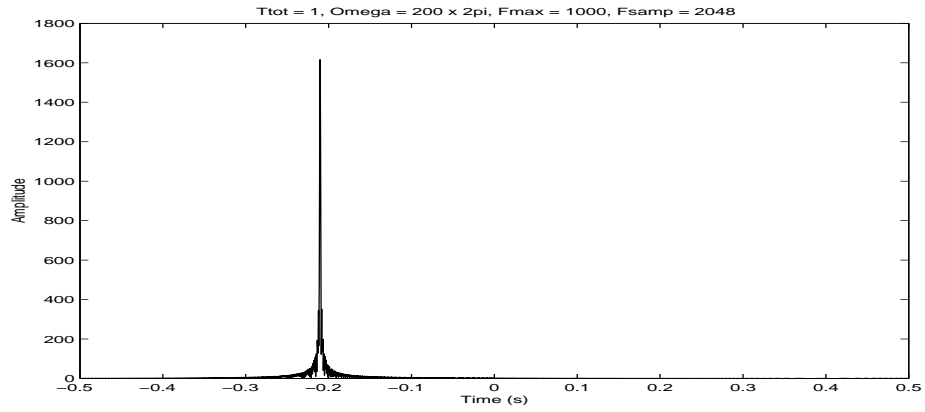


Figure 2.6 The dipole impulse response: Inverse Fourier transform of the multiplication of $S_c(\omega) * D_{2,c}(\omega)$ with $S_c^*(\omega)$.

2.3.4 Mathematical foundation

For better understanding of the response plotted in Figure 2.6, the shift of the pulse in the time domain, mathematical analysis is necessary.

The product of the complex sweep $s_c(t)$ (Eq. 2.9) and the directivity of a complex n -pole $d_{n,c}(t)$ (Eq. 2.10) is given by

$$s_c(t) \cdot d_{n,c}(t) = e^{j(2\pi f_l t + \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}} t^2)} \cdot e^{j\frac{1}{2}n\psi t} = e^{j(2\pi(f_l + \frac{1}{2}\frac{n\psi}{2\pi})t + \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}} t^2)}. \quad (2.11)$$

Comparison of equation 2.9 and the last part of equation 2.11 clearly shows that $s(t) \cdot d_{n,c}(t)$ is a complex linear frequency sweep, starting at frequency $f_l + \frac{1}{2}n\psi/2\pi$ and ending at frequency $f_h + \frac{1}{2}n\psi/2\pi$.

To calculate the impulse response, the product $s_c(t) \cdot d_{n,c}(t)$ must be deconvoluted

with the sweep signal. Or, the Fourier transform of this product must be multiplied with $S_c^*(\omega)$ (Eq. 2.7). Multiplication with $S_c^*(\omega)$ in the frequency domain is equal to convolution with $s_c^*(-t)$ in the time domain:

$$h_c(t) = (s_c(t) \cdot d_{n,c}(t)) * s_c^*(-t) = \int_{-\infty}^{+\infty} s_c(\tau) \cdot d_{n,c}(\tau) \cdot s_c^*(\tau - t) d\tau. \quad (2.12)$$

For convenience, the parameters α and β are introduced:

$$\alpha = 2\pi f_l \quad \beta = \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}}. \quad (2.13)$$

Using parameters α and β , equation 2.12 can be rewritten to:

$$\begin{aligned} h_c(t) &= \int_{-\infty}^{+\infty} s_c(\tau) \cdot d_{n,c}(\tau) \cdot s_c^*(\tau - t) d\tau \\ &= \int_{-\infty}^{+\infty} e^{j(\alpha\tau + \beta\tau^2)} \cdot d_{n,c}(\tau) \cdot e^{-j(\alpha\tau - \alpha t + \beta\tau^2 - 2\beta\tau t + \beta t^2)} d\tau \\ &= \int_{-\infty}^{+\infty} e^{j(\alpha\tau + \beta\tau^2)} \cdot e^{-j(\alpha\tau + \beta\tau^2)} \cdot e^{-j(-\alpha t + \beta t^2)} \cdot d_{n,c}(\tau) \cdot e^{j2\beta\tau t} d\tau \\ &= \int_{-\infty}^{+\infty} 1 \cdot e^{-j(\alpha(-t) + \beta(-t)^2)} \cdot d_{n,c}(\tau) \cdot e^{j2\beta\tau t} d\tau \\ h_c(t) &= s_c^*(-t) \int_{-\infty}^{+\infty} d_{n,c}(\tau) \cdot e^{j2\beta\tau t} d\tau. \end{aligned} \quad (2.14)$$

The last integral is the non-normalized correlation of $d_{n,c}(\tau)$ and $e^{j2\beta\tau t}$. If the two terms in the integral are not correlated, the integral will be approximately zero. Two sines with a different angular velocity are not correlated, therefore, $d_{n,c}(\tau)$ and $e^{j2\beta\tau t}$ must have the same angular velocity if $h_c(t)$ is non-zero. If $d_{n,c}(t)$ is chosen as in equation 2.10, $h_c(t)$ will only be non-zero at time t_n if

$$e^{j\frac{1}{2}n\psi\tau} \cdot e^{j2\beta t_n\tau} = 1 \quad \forall \tau, \text{ or } \frac{1}{2}n\psi + 2\beta t_n = 0. \quad (2.15)$$

Or formulated otherwise: the impulse response of a rotating source with a complex n -pole as directivity pattern, $h(t)$, will be non-zero only at time t_n , given by:

$$t_n = -\frac{\frac{1}{2}n\psi}{2\beta} = -\frac{\frac{1}{2}n\psi}{2 \cdot \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}}} = -\frac{\frac{1}{2}n\psi}{2\pi \cdot \frac{f_h - f_l}{T_{sweep}}}. \quad (2.16)$$

This result is deduced using a complex sweep with the directivity of a complex n-pole. Using the linearity of all the operations in the derivation, this result can be extended to real situations. A detailed extension can be found in Appendix A.

The result of this extension: a real sweep and a real directivity of the form $d(t) = \cos(\psi t)$ result in a calculated impulse response consisting of **two** peaks, one at $t = -t_n$ and one at $t = +t_n$.

Calculation of t_2 for the sweep and directivity used in Figure 2.3 to 2.6 results in:

$$t_n = -\frac{\frac{\frac{1}{2}n\psi}{2\pi}}{\frac{f_h - f_l}{T_{Sweep}}} = -\frac{\frac{\frac{1}{2} \cdot 2 \cdot 200 \cdot 2\pi}{2\pi}}{\frac{1000 - 50}{1}} = -\frac{200}{950} = -0.2105 \text{ s.} \quad (2.17)$$

This is in accordance with Figure 2.6

The directivity of a real source more complex than $d(t) = \cos(\frac{1}{2}n\psi t)$. The directivity pattern will consist of monopole, dipole, quadrupole and higher order terms. A linear frequency sweep and a rotating source rotating with angular velocity ψ together have the special property to decompose the directivity of the source into the contributions of the separate poles. To illustrate this, the impulse response of a more complex source will be shown.

As the input signal, a linear frequency sweep from 100 to 1000 Hz and duration 1 s is taken. The directivity is built by addition of a monopole, dipole and quadrupole, or mathematically:

$$d(t) = 1 + \cos(\psi t) + \cos(2\psi t). \quad (2.18)$$

The directivity pattern is shown in Figure 2.7.

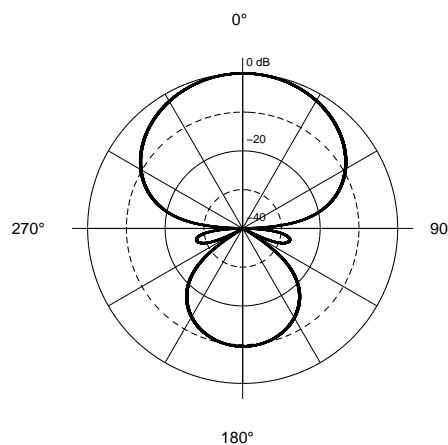


Figure 2.7 The directivity pattern build up from a monopole, dipole and a quadrupole of equal strength.

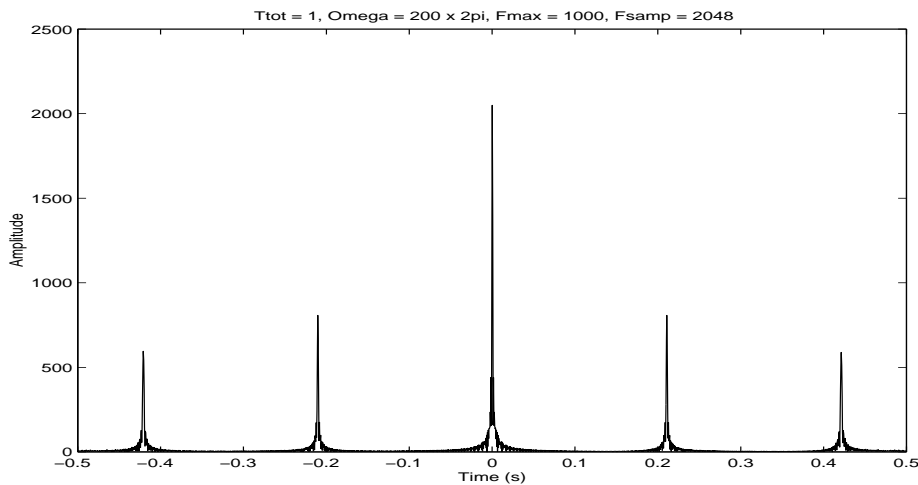


Figure 2.8 The impulse response of a linear frequency sweep from a rotating source with directivity given by equation 2.18.

In Figure 2.8, the impulse response is plotted. The five peaks are not equally high. The peak of the monopole at $t = 0$ is twice as high as the other peaks. This is logical, because the ‘powers’ of the dipole and quadrupole are distributed over 2 poles (one at $t < 0$ and one at $t > 0$).

The small difference in height between the dipole and quadrupole can be explained by the term $s^*(-t)$ preceding the integral in equation 2.14.

The impulse response of a rotating loudspeaker using a linear frequency sweep has some important properties:

- the impulse response of an arbitrary loudspeaker is decomposed in the n-poles separated in time. The rotating monopole impulse response (desired!) is separate available. The directivity of a monopole is isotropic, so a rotating monopole has the same directivity as a non-rotating monopole.
- the responses to the dipole, quadrupole and higher order poles are irrelevant. No physical meaning can be attributed to a rotating dipole response.
- in a room, the gaps between the pulses of the n-poles will be ‘filled’ with the room response.
- any required distance between the n-poles can be realized by proper choice of ψ and T_{sweep} .

Chapter 3

Simulations

3.1 Introduction

In Chapter 2, a special property of the combination of a linear frequency sweep and a rotating source was explained: spherical decomposition of the directivity pattern of the source. This decomposition into monopole, dipole and higher order pole characteristics can be used to measure the monopole response of a room. A monopole radiates acoustic power isotropically, exactly the requirement to the ideal source for most room acoustic measurements.

To accomplish a good impulse response, the input signal must meet some requirements. In this chapter, the influence of tapering, parallel sweeps, full revolutions and Doppler effects respectively, are analysed using simulations.

3.2 Tapering

The impulse response is calculated from the measured room response to the linear frequency sweep using a discrete Fourier transform. The discrete Fourier transform has as a side effect that the signal is made periodic. This can result in a bad connection of the beginning and the end of the signal, resulting in undesired high frequency components. To smooth the edges of the signal, tapering is necessary.

Only a signal that is zero at both edges can connect smooth under all conditions. To accomplish this, the signal is multiplied with a tapering window that is equal to one in the main part and drops gradually to zero at the edges. A requirement for 'gradually dropping' is a low value of the second derivative of the tapering window.

A tapering window that is often used is the Tukey-Hanning window (see Priestley [7]). With M being the truncation point (the starting point for the gradually dropping), the window function $\lambda[n]$ is given by:

$$\lambda[n] = \begin{cases} \frac{1}{2} + \frac{1}{2} \cos(\pi n/M) & |n| \leq M \\ 0 & |n| > M. \end{cases} \quad (3.1)$$

A disadvantage of this window function is the loss of power at the center, whereas the effect of the tapering is at the edges. To accomplish a flat power spectrum in the region of interest, a slightly different tapering window is used. The tapering window used in the present research is flat in the main part and uses a Tukey-Hanning shape at the edges. The tapering window of equation 3.1 is symmetrical around zero. Usually, the data ranges from $n = 0$ to $n = N-1$. With S being the width of the sections with the Tukey-Hanning shape, the window function $w[n]$ is given by:

$$w[n] = \begin{cases} \frac{1}{2} - \frac{1}{2} \cos \frac{\pi n}{S} & 0 \leq n < S \\ 1 & S < n \leq (N-1) - S \\ \frac{1}{2} + \frac{1}{2} \cos \frac{\pi(n-N-1+S)}{S} & (N-1) - S < n < N-1 \end{cases} \quad (3.2)$$

The shape of the tapering window function $w[n]$ is plotted in Figure 3.1.

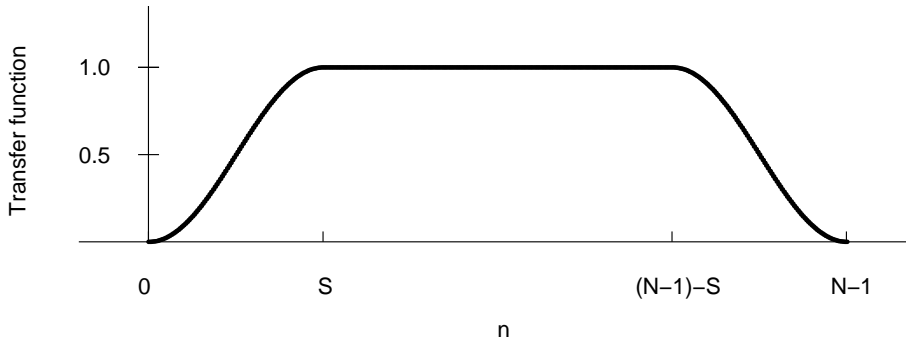


Figure 3.1 The window function $w[n]$ of equation 3.2

The tapering window is used in the time domain, but is also a tapering window in the frequency domain, as a result of the linear rising frequency of the sweep. Choosing S very large will result in loss of power at the starting and ending frequencies of the linear frequency sweep.

In Figure 3.2, the effect of tapering on the impulse response¹ and on the power spectrum is made clear for three different choices of the length of the tapering window. The power spectrum is calculated by squaring the absolute value of the Fourier transform of the impulse response in a window from $t = -0.1$ to $t = +0.1$. The input signal is a sweep from 100 to 4500 Hz with a duration of 88 seconds. The time between the n-pole responses is 0.2 second. The source is rotating at 600 rpm

¹Strictly spoken, this is not a plot of the impulse response, but a dB-plot of the absolute value of the impulse response.

and the directivity is the sum of a monopole, dipole and quadrupole response (see Figure 2.7).

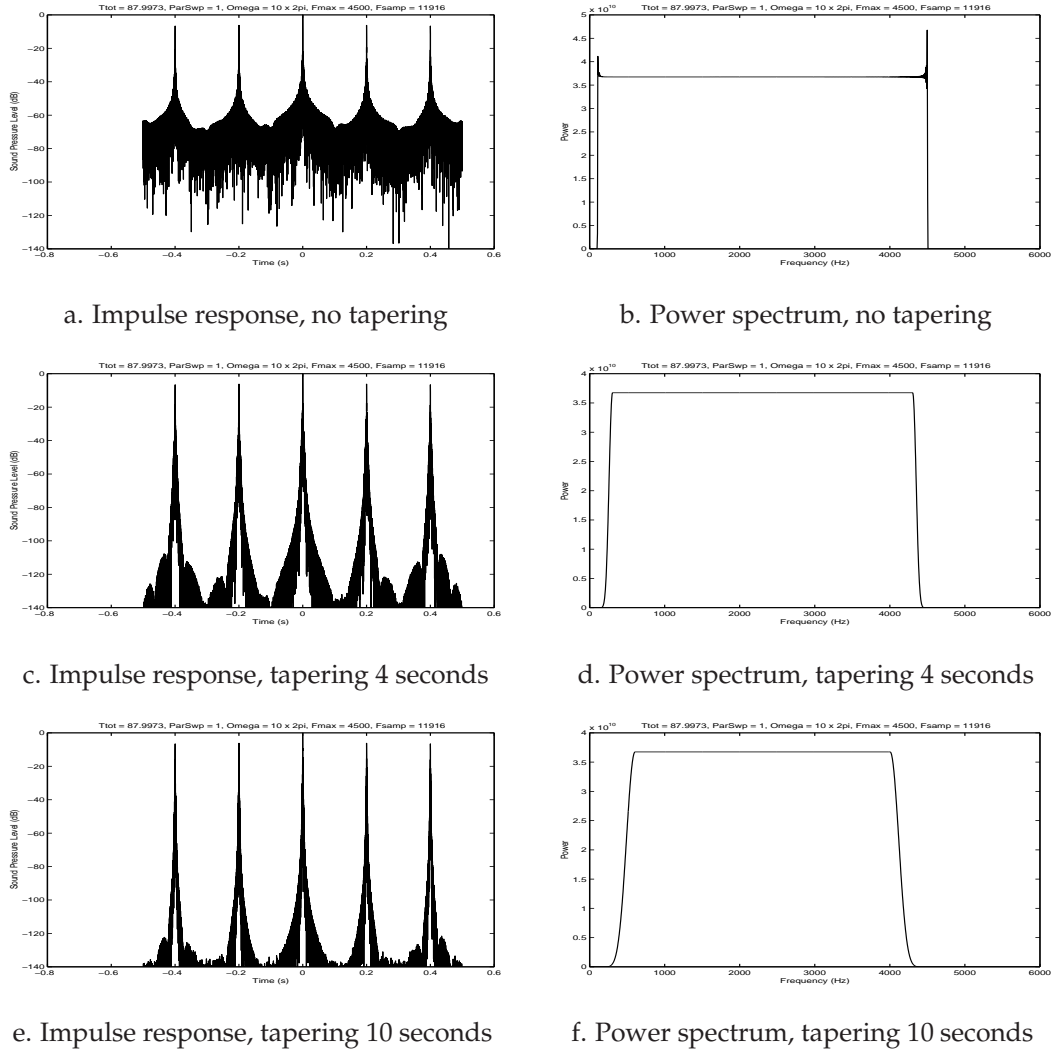


Figure 3.2 The influence of the size of the tapering window on the impulse response (dB scale) and on the power spectrum (linear scale).

Tapering clearly raises the S/N ratio². In this situation, there is no use in applying a tapering window larger than 4 seconds. The S/N ratio of 100 dB is more than sufficient for all practical situations. The signal is tapered in the time domain, but because of the linearly rising frequency, the signal has a tapering window in the frequency domain with the same shape. The disadvantage of a tapering window with a large section with the Tukey-Hanning shape is the loss of bandwidth. A

²Strictly spoken, S/N ratio is not a good description. The main part of the 'garbage' in Fig 3.2a is not (stochastic) noise, but is induced by the method of calculation. In the present research, S/N ratio is used as the term for the distance between the desired signal and 'garbage'.

tapering section of 4 seconds reduces the peaks at the beginning and ending sufficiently.

This result can be generalized³ : the tapering window must be at least $1/20^{\text{th}}$ of the total length of the sweep, or $S = N/20$.

3.3 Parallel sweeps

For room measurements, the distance between the n-pole responses must be twice the reverberation time. In real situations, the reverberation time will range from 0.5 s to 2.0 s. Using a turn table with a revolution speed of 1 cycle per second, the required measuring time for one measurement will be very large. The time between 2 n-poles can be calculated using equation 2.16. The time Δt between two n-pole responses is given by

$$\Delta t = t_n - t_{n-2} = \frac{\frac{\psi}{2\pi}}{\frac{f_h - f_l}{T_{\text{sweep}}}} = \frac{\frac{\psi}{2\pi} T_{\text{sweep}}}{f_h - f_l} = \frac{m}{\Delta f}, \quad (3.3)$$

introducing m as the number of revolutions and Δf as the bandwidth of the used sweep. Taking $\Delta t = 1$ s and $\Delta f = 4500$ Hz results in $m = 4500$ cycles. A regular turntable has a maximal revolution speed of 1 cycle/s, resulting in a required measuring time of 4500 s. This is not acceptable, nor the $2\frac{1}{4}$ hours time required for one measurement, nor the required computing power for processing afterwards.

A solution can be found in reproducing parts of the the sweep in parallel. The sweep is cut into smaller pieces with equal duration and these are reproduced simultaneously. If the sweep is split into p pieces which are reproduced in parallel, the time required for one measurement will be reduced by a factor p or mathematically

$$\Delta t = \frac{m}{p \Delta f}. \quad (3.4)$$

The construction of the parallel sweep is illustrated in Figure 3.3.

The basis of the parallel sweep is a linear frequency sweep (I). This signal has low frequencies at the beginning and high frequencies at the ending. In the processing afterwards, the sweep is made periodical by the Fourier Transform. For smooth connection of the successive sweep, tapering is necessary (II). To reduce the total length, the signal is cut into p equal pieces III. These pieces are placed in parallel in the time interval from t_0 to t_1 (IV). Addition of these parallel pieces results in the parallel sweep signal (V).

As mentioned before, a side effect of the discrete Fourier Transform is that the signal is made periodical. This results in multiple parallel sweeps, see Figure 3.4.

³This generalization is only valid for the signal used in the present research. Widely applicable generalizations may not be extended from the conclusions in the present research

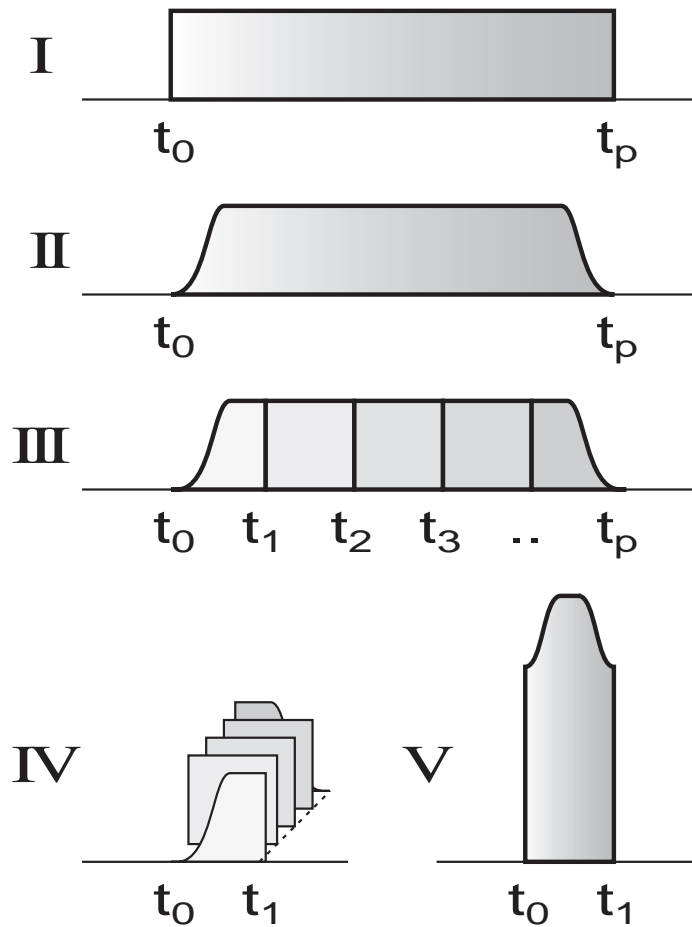


Figure 3.3 Split up from a linear frequency sweep into a parallel sweep. For further explanation see text.

At item (a), the parallel sweep is depicted, exactly as in Figure 3.3 IV. The parallel sweep is built up from the successive pieces of a sweep placed in parallel. Since the discrete Fourier Transform makes the signal periodical, the signal (a) is repeated. Item (b) is exact the same signal as item (a), but all 'plates' have shifted one place forwards except for the front one, which is placed at the back. The signal remains the same, it still is the sum of all the 'plates'. At (c), this is demonstrated once again.

If the parts (a), (b) and (c) are placed after each other, they connect fluently. At item (d), p parts have been placed after each other. It is shown that the total signal consist of p parallel sweeps, with a smooth beginning and ending.

Although the sweep of Figure 3.3 V is non-zero at the beginning and the ending, the signal connects fluently with itself when made periodically.

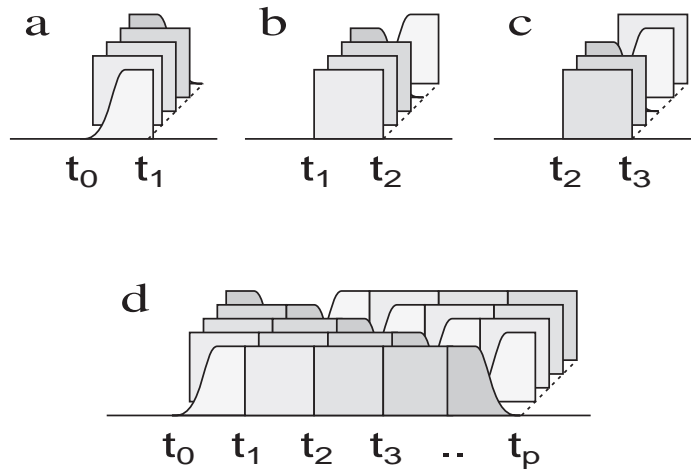


Figure 3.4 One parallel sweep signal periodically is equal to multiple full sweeps in parallel

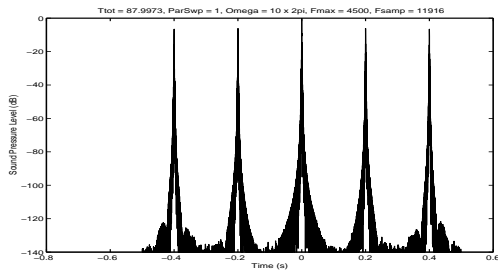
Building the parallel sweep in the way described above has some major advantages:

1. The signal has a smooth beginning and ending, no distortion is caused by abrupt transitions.
2. Tapering is only required at t_0 and t_p . The transitions in between don't need tapering, because of the seamless connection due to the periodicity of the discrete Fourier Transform.
3. The signal can be much shorter because several sweeps are reproduced simultaneously.

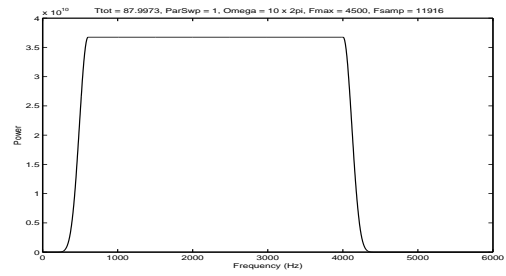
The effects of using a parallel sweep instead of a linear frequency sweep are clearly visible in the simulations. In Figure 3.5, the impulse response and the power spectrum of a parallel sweep are plotted for $p = 1, 2, 8, 32$ and 128 . For good comparison, the duration of the total signal is 88 seconds in all situations. By reducing the revolution speed for higher values of p , Δt is kept to 0.2 s in all simulations. The basis in all simulations is a linear frequency sweep from 100 to 4500 Hz.

The figures show that the S/N ratio decreases as more sweeps are reproduced in parallel. Placing two sweeps in parallel reduces the maximum distance between the top of the peaks and the 'background noise' from 120 dB to 80 dB. This is still more than sufficient for room measurements.

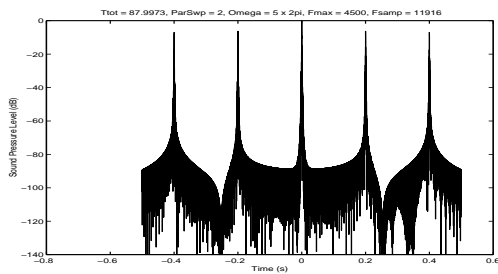
Besides reduction of the S/N ratio, the bottom of the sweep widens. This effect shows already at 2 sweeps in parallel, but gets significantly stronger for higher values of p . For useful measurements, the pulse must be much smaller than the



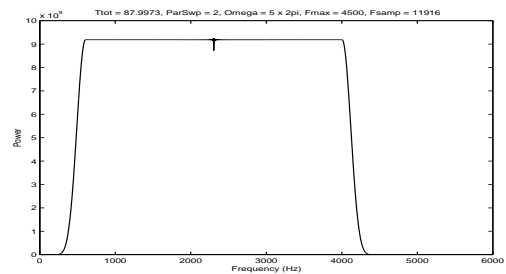
a. Impulse response, no parallel sweeps (p=1)



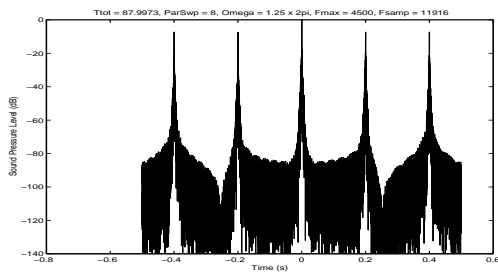
b. Power spectrum, no parallel sweep (p=1)



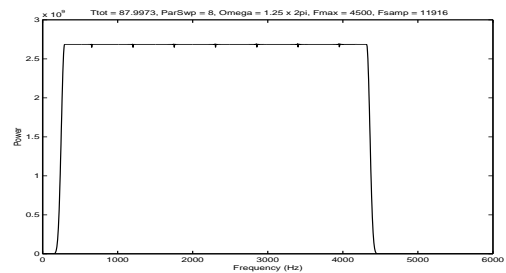
c. Impulse response, p=2



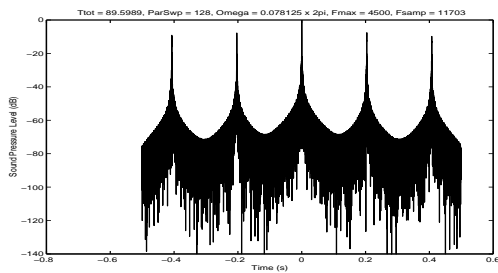
d. Power spectrum, p=2



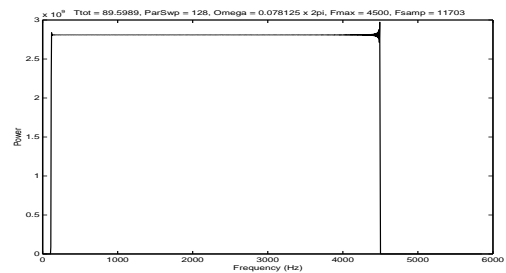
e. Impulse response, p=8



f. Power spectrum, p=8



g. Impulse response, p=128



h. Power spectrum, p=128

Figure 3.5 The influence of the number of sweeps in parallel on the impulse response (dB scale) and on the power spectrum (linear scale).

decay of the room. Larger rooms have larger reverberation times and therefore require more distance between the peaks. But a larger reverberation time also allows for a wider peak and thus more sweeps in parallel. For every application, a compromise between measuring time and S/N ratio is necessary.

Use of parallel sweeps in stead of a normal sweep reduces the energy of the signal. If a sweep is cut in two and the second part is added to the first part, the energy of the resulting signal is exactly equal to the original signal, the power is twice as high, the time has been halved. But the peak value of the resulting system is twice as high as the peak value of the original signal. To avoid clipping, the peak value must be equal to the peak value of the original signal, thus the signal must be decreased with factor 2, resulting in a energy loss of 6 dB.

Conclusion: Parallel sweeps reduce the required time for a room measurement to acceptable values. Every situation requires a optimal compromise between required measuring time and S/N ratio.

3.4 Full revolutions

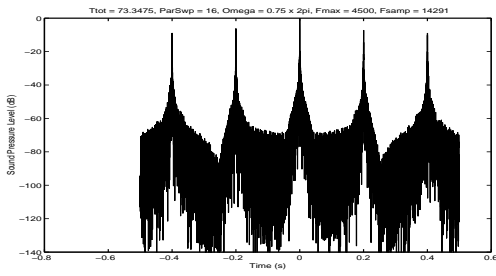
To achieve a high signal to noise ratio in the impulse response, it is of major importance that the beginning and ending of a measurement connect smoothly. By building up the measurement signal as described in section 3.2 and 3.3, this condition is fulfilled satisfactory. For a smooth connection, it is also important that the angle of the turntable at the beginning of the measurement is equal to the angle at ending of the measurement. If, for example, the position at the ending of the measurement differs half a cycle from the starting position, the dipole and higher order poles cause bad connection of the beginning and ending of the signal. The distortion shows clearly in the simulations.

The simulations to investigate the influence of rotating a not entire number of cycles cannot be done under exact equal circumstances. Equation 3.3 can be rewritten to:

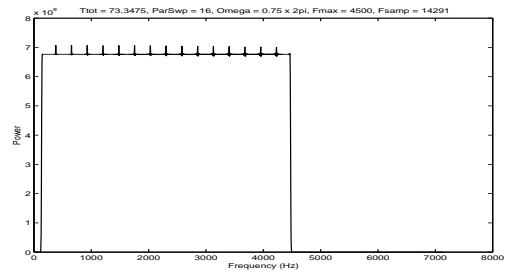
$$m = \Delta f \cdot \Delta t. \quad (3.5)$$

The number of revolutions m is fully determined by the required time interval between the n-pole responses and the required bandwidth. To change m , either Δt or Δf must be changed. Adaptation of Δt is preferable, because adaptation of Δf changes also the shape of the pulses. The adaptation of Δt will be very small. The maximum possible adaptation of half a cycle on 55 cycles in total, requires only a 1% change of Δt .

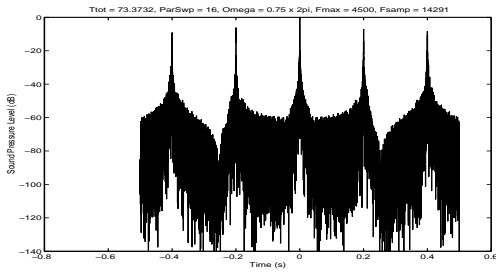
In Figure 3.6, the influence of m on the impulse response and the power spectrum is plotted. All sweeps are built using the following parameters: $f_l = 100$ Hz,



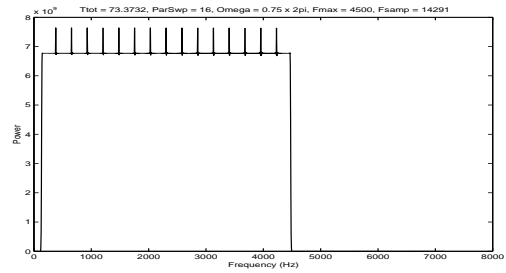
a. Impulse response, $m = 55.01$



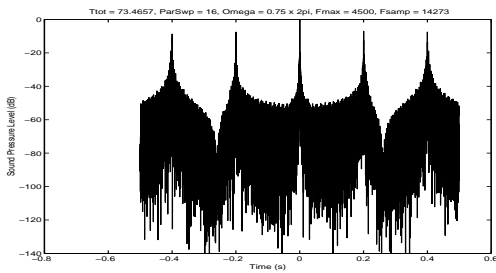
b. Power spectrum, $m = 55.01$



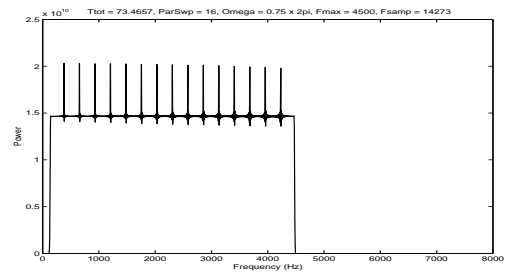
c. Impulse response, $m = 55.03$



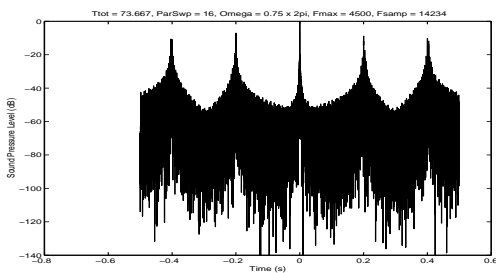
d. Power spectrum, $m = 55.03$



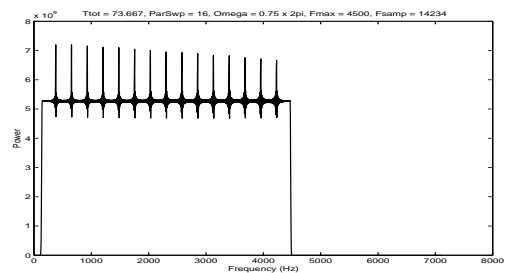
e. Impulse response, $m = 55.10$



f. Power spectrum, $m = 55.10$



g. Impulse response, $m = 55.25$



h. Power spectrum, $m = 55.25$

Figure 3.6 The influence of the number of revolutions on the impulse response (dB scale) and on the power spectrum (linear scale).

$f_h = 4500$ Hz, $\psi = 2\pi \cdot 0.75$ rad/s, number of n-poles: 3, number of parallel sweeps: 16.

The simulation clearly demonstrates that rotation over a not entire number of rounds reduces the S/N ratio strongly. If the required S/N ratio is 60 dB, the maximum allowed deviation is 0.03 circle. In this simulation $m \approx 55$, resulting in a maximum allowed deviation of the revolution speed ψ of 0.05%. This sets high demands on the steadiness of the revolution speed of the turntable.

If not parallel sweeps are used, but just one single frequency sweep, the system is much less sensitive to little variations in the revolution speed: a single frequency sweep is tapered at the beginning and ending of the signal, a parallel sweep is not (compare Figure 3.3 II with Figure 3.3 V).

3.5 Doppler effects

Doppler effects occur when either the source or the receiver moves. In the present research, the receiver does not move and the source rotates. The complete diaphragm of the loudspeaker radiates acoustic energy. If the loudspeaker is positioned on the rotation axis, the centre of the diaphragm has no velocity caused by the rotation. But the edges of the diaphragm do gain some speed. The acoustic power emitted by the centre of the diaphragm is not influenced by Doppler effects, but the power emitted by the edges is substantially influenced.

The Doppler effects for a moving source and a stationary receiver are well-known. Define $\rho(t)$ as the distance between the moving source S and the stationary receiver R. If the acoustic emission of the source is $x(t) = \cos(\omega t)$, the pressure $y(t)$ at the receiver is (see Jansen [5])

$$y(t) = \frac{1}{\rho(t)} \cos \left(\omega \left(t - \frac{\rho(t)}{c + \dot{\rho}(t)} \right) \right), \quad (3.6)$$

with c the velocity of sound and $\dot{\rho}(t)$ the first derivative of ρ to t . If the source is receding from the receiver, the perceived frequency at the receiver is lower than the emitted frequency. Equation 3.6 has the term $c + \dot{\rho}(t)$ instead of just the velocity of sound c , because the source is moving and therefore has nonconcentric wavefronts.

3.5.1 The rotating monopole

The most plain situation of a rotating source 'suffering' from Doppler effects, is a monopole source rotating with angular velocity ψ at a distance r of the rotation axis (Figure 3.7).

The distance a between the receiver and the rotation axis is much larger than the distance r between the source and the rotation axis ($a \gg r$). The distance $\rho(t)$ between the source and receiver, depends on the angle between the imaginary

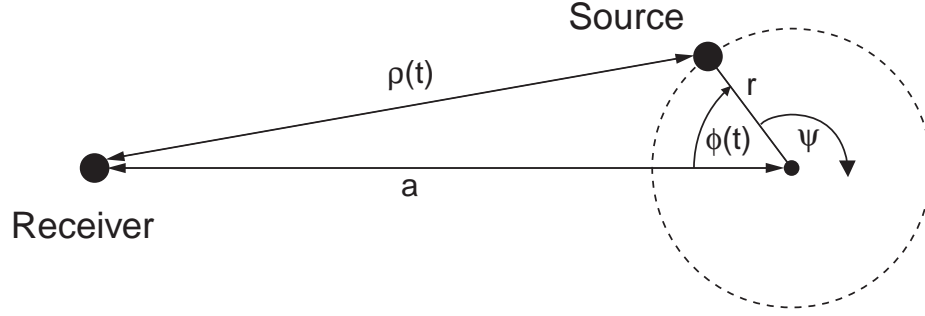


Figure 3.7 A monopole source rotating with angular velocity ψ at a distance $\rho(t)$ of the receiver

lines from the rotation axis to the receiver and to the source. If $\phi = 0$ at $t = 0$, ρ is given by:

$$\rho(t) = a - r \cos(\phi(t)) = a - r \cos(\psi t). \quad (3.7)$$

The source radiates a monochromatic signal with a frequency of $\frac{\omega}{2\pi}$: $x(t) = \cos(\omega t)$. The pressure $y(t)$ at the receiver is now given by

$$y(t) = \frac{1}{\rho(t)} \cos\left(\omega\left(t - \frac{\rho(t)}{c + \dot{\rho}(t)}\right)\right), \quad (3.8)$$

with c being the velocity of sound and $\dot{\rho}(t)$ the first derivative to the time of $\rho(t)$. In practical situations, $\dot{\rho}(t)$ will be much smaller than c and a will be much larger than r . Introducing these approximations and the wave number $k = \omega/c$, equation 3.8 becomes

$$y(t) \approx \frac{1}{a} \cos(\omega t - ka + kr \cos(\psi t)). \quad (3.9)$$

This can be interpreted as an axial rotating point source ($r=0$, so no Doppler effects) at a distance a of the receiver with a phase directivity $d'(\phi)$. The emitted power is equal for all directions. The phase directivity is

$$d'(\phi) = e^{kr \cos(\psi t)}. \quad (3.10)$$

The phase shift $\Delta\phi(t) = kr \cos(\psi t)$ as a result of the rotation is plotted in Figure 3.8 for 2 different choices of kr . The phase shift is larger for higher values of the product kr , thus for higher frequencies or for a larger distance between the source and the rotation axis.

Doppler effects clearly result in a directivity effect. Contrarily to the directivity of a single n-pole (see Fig. 2.2a), the directivity caused by Doppler effects consists of

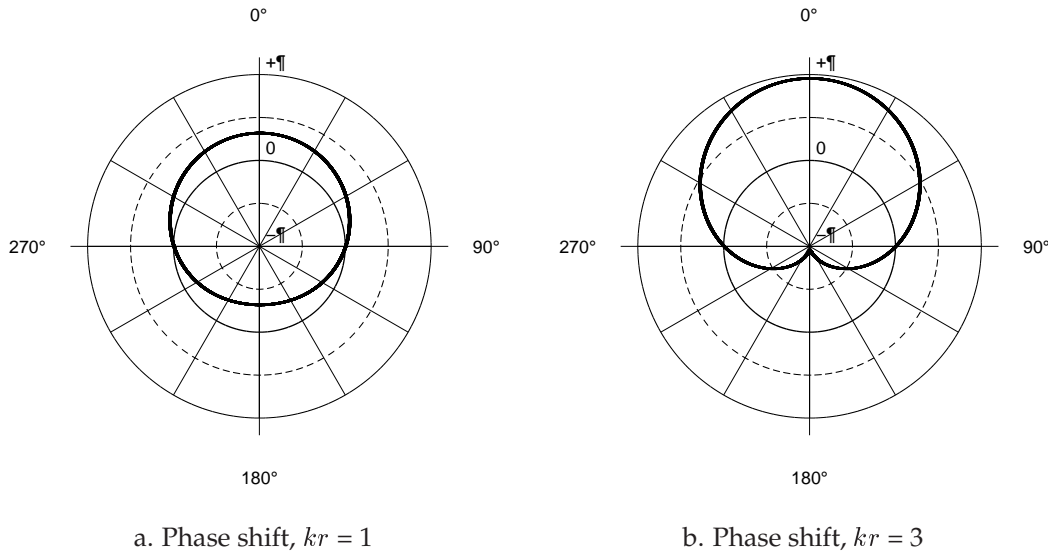


Figure 3.8 The phase shift in radians as a function of the angle ϕ of a monochromatic monopole source rotating at a radius r around a fixed point, perceived by a stationary receiver at a large distance (see Fig. 3.7). The phase shift increases as kr increases.

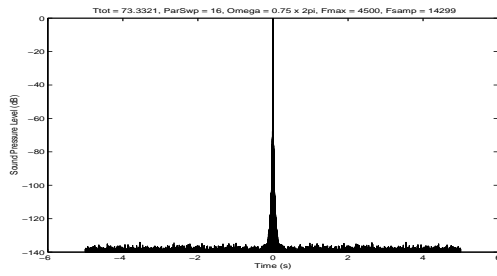
a phase shift only. The radiated acoustic energy is equal for all directions.

Simulations are necessary to investigate the influence of Doppler effects. In Figure 3.9 the impulse response is plotted for a rotating monopole. All simulations are made using the following parameters: $f_l = 100$ Hz, $f_h = 4500$ Hz, $\psi = 2\pi \cdot 0.75$ rad/s, number of parallel sweeps: 16.

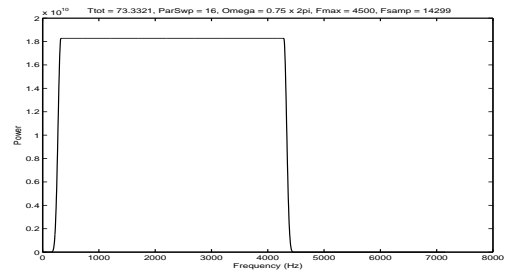
In Figure 3.9a and b, a nice and sharp pulse is visible, together with a flat power spectrum. Figure 3.9 c shows the Doppler effect results in many extra peaks in the pulse response, even when r is small. The Doppler pulses show up at the calculated positions of the n-pole responses. The distance between the peaks is 0.2 s, exactly equal to Δt . Apparently, the Doppler phase directivity $d'(\phi)$ of Eq. 3.10 is decomposed into monopole, dipole and higher order pole contributions. This is understandable, a linear frequency sweep decomposed the directivity of a rotating source and does not make an exception for a directivity pattern caused by Doppler.

The power spectrum of Figure 3.9d has a slight decay to the higher frequencies. The valleys between the peaks are still deep, 100 dB.

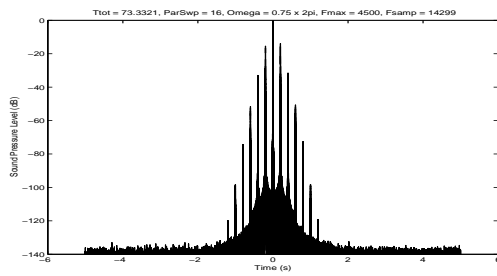
As r increases, the number of peaks increases. The depth of the valleys decreases to 80 dB. This is still more than sufficient for room measurements. More problematic is the power spectrum. The power spectrum gets a sinc-shape, resulting in a loss of power, especially at the higher frequencies. The Doppler effect is proportional



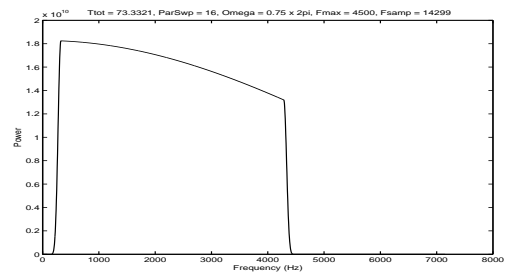
a. Impulse response, $r = 0.00\text{m}$



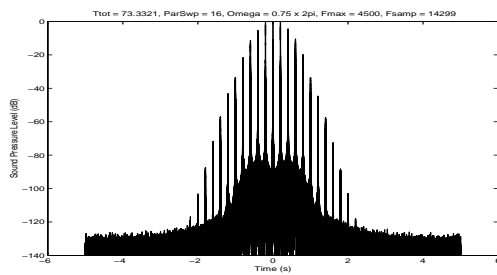
b. Power spectrum, $r = 0.00\text{m}$



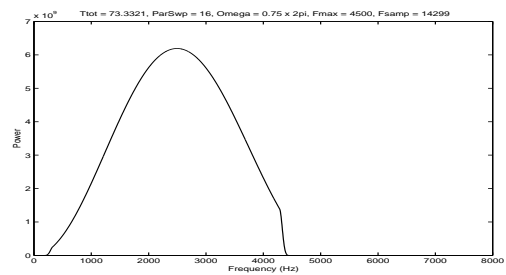
c. Impulse response, $r = 0.01\text{m}$



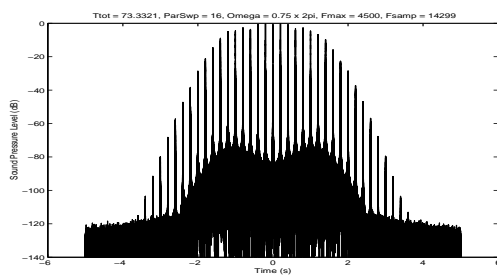
d. Power spectrum, $r = 0.01\text{m}$



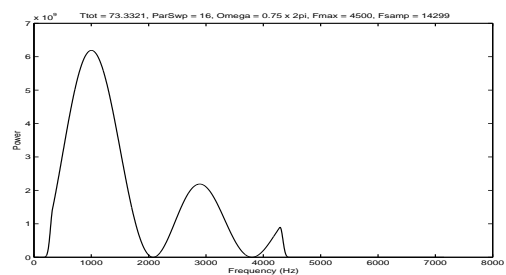
e. Impulse response, $r = 0.03\text{m}$



f. Power spectrum, $r = 0.03\text{m}$



g. Impulse response, $r = 0.10\text{m}$



h. Power spectrum, $r = 0.10\text{m}$

Figure 3.9 The influence of the distance r between the monopole and the rotation axis on the impulse response (dB scale) and on the power spectrum (linear scale).

to the product $r\psi$, so for good measurements, the speaker must be small and the angular velocity of the source cannot be large.

3.5.2 The rotating diaphragm

The experience on Doppler effects gained in the situation of a monopole rotating at a short distance of the rotation axis, must be extended to a rotating diaphragm. A diaphragm can be considered as a large collection of point sources. The distance between rotation axis of the diaphragm and the point source is decisive for the determination of the contribution of the separate point sources to the Doppler effects. Therefore, the diaphragm is not divided into separate point sources, but into i strips, see figure 3.10.

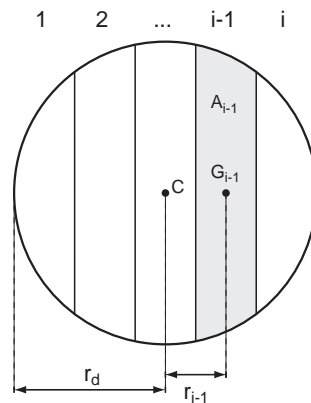


Figure 3.10 The diaphragm of the loudspeaker is divided into i strips with equal width, surface A_i and centre of gravity G_i . The radius of the diaphragm is r_d .

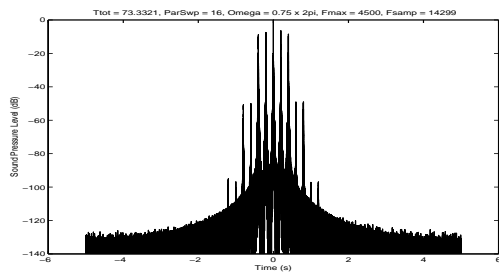
The choice of i is free, i can be odd or even. The centre of gravity G and the surface area A of every strip is calculated. Hereafter, the distance between the centre of gravity of the strip and the centre of the diaphragm (= radiation axis) is calculated. In the simulations, the loudspeaker diaphragm is considered to be composed of i monopoles with weight factor A_i and rotating at a distance r_i of the rotation axis. In the simulations, first the influence of the value i is investigated, to determine what a correct choice is for i in simulations. A low value of i reduces the required computing time, but may result in a bad simulation of the reality. It proved that only choice of a small value for i ($i < 3$), dramatically reduces the Doppler effects. This is comprehensible, low i reduces the maximum value of r_i and thus the diaphragm appears as a smaller one. The impulse response and power spectrum don't change significantly for $i > 8$. The optimal choice of i for simulations is $i = 8$. Simulations clearly show the large influence of the Doppler effects, especially on the power spectrum. In Figure 3.11, the impulse responses and power spectra are

plotted for a radius of the diaphragm r_d of 0.01, 0.03, 0.05 and 0.10m and the diaphragm is simulated with 8 strips. The input is a sweep from 100 to 4500 Hz, the angular velocity of the turntable is 45 rpm and 16 sweeps are reproduced in parallel, to gain a signal of 73 s with $\Delta t = 0.2$ s. For the directivity, the combination of a monopole, dipole and quadrupole is chosen (see Fig. 2.7).

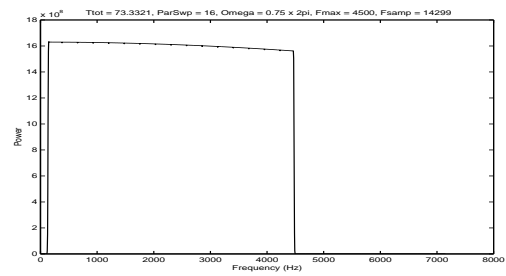
Doppler effects cause the appearance of more peaks in the anechoic impulse response. This is in accordance with the results of the rotating monopole. The rotating diaphragm is a weighted sum of rotating monopoles. The power spectrum for high values of r_d is not as bad as for high values of r in the simulations with the rotating monopole. This is caused by the contribution of the strips near the rotation axis, that do not suffer from Doppler effects.

The Doppler effects are proportional to ψr_d . A high angular velocity reduces the required measuring time, but raises the influence of Doppler effects. A large diaphragm can radiate more acoustic power, but also raises the influence of Doppler effects. The S/N ratio of the monopole impulse response is affected, but remains higher than 80 dB in these simulations. The power spectrum of the monopole impulse response is affected harmfully for higher values of ψr_d .

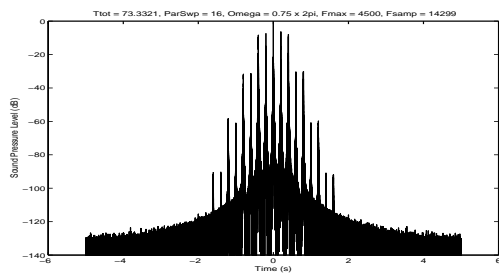
The conclusion of the simulations on Doppler effects is that a larger diaphragm and a higher angular velocity result in more Doppler effects. The S/N ratio of the monopole impulse response is not affected harmfully, but the higher frequencies in power spectrum of the monopole are weakened harmfully. The Doppler effects are proportional to ψr_d . This requires a compromise between on one side the flat power spectrum and on the other side the loudspeaker surface and angular velocity (and thus measuring time).



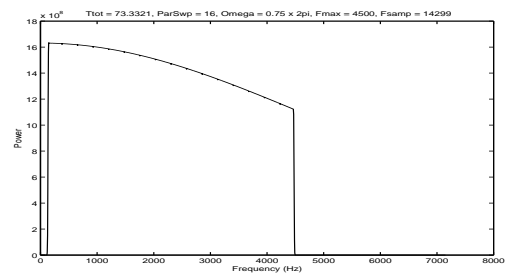
a. Impulse response, $r_d = 0.01\text{m}$



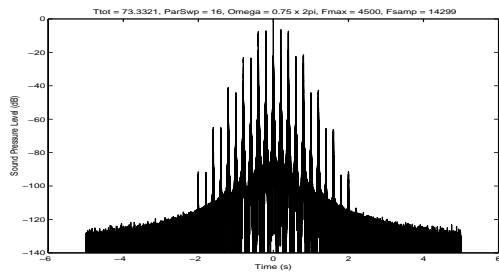
b. Power spectrum, $r_d = 0.01\text{m}$



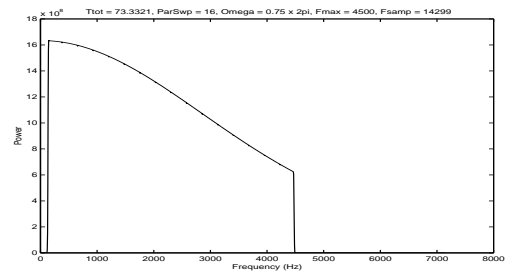
c. Impulse response, $r_d = 0.03\text{m}$



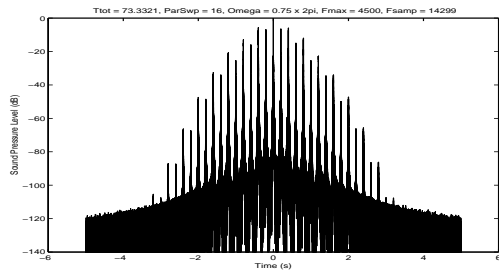
d. Power spectrum, $r_d = 0.03\text{m}$



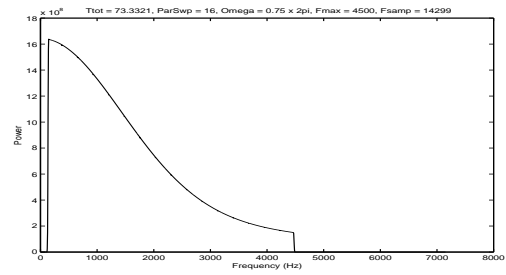
e. Impulse response, $r_d = 0.05\text{m}$



f. Power spectrum, $r_d = 0.05\text{m}$



g. Impulse response, $r_d = 0.10\text{m}$



h. Power spectrum, $r_d = 0.10\text{m}$

Figure 3.11 The influence of the radius of the diaphragm on the impulse response (dB scale) and on the power spectrum (linear scale).

Chapter 4

Experiments

4.1 Construction of the sweep signal

The simulations in chapter 3 have shown that to obtain useful results from the measurements, the input signal, the sweep $s(t)$, must be adapted to the angular velocity of the turntable. The sweep signal must be ‘constructed’ meticulously. First of all, some parameters have to be chosen:

- f_l the lowest frequency of the sweep
- f_h the highest frequency of the sweep
- Δt the desired time between the pulses of the n-poles in the impulse response
- p the number of parallel sweeps.

The value of Δt must be higher than the reverberation time of the room. Using equation 3.4, the required number of revolutions m is:

$$m = \frac{\Delta f \Delta t}{p}. \quad (4.1)$$

Taking for example $f_l = 100$ Hz, $f_h = 4500$ Hz, $p = 32$ and $\Delta t = 1$ s, the required number of revolutions is 137.53 revolution. In section 3.4, simulations have shown that m should be an integer. To accomplish this, either Δf or Δt must be altered. To preserve the bandwidth, Δt will be changed a little to Δt^* . The nearest integer for m is 138, resulting in $\Delta t^* = 138/137.53 \cdot \Delta t = 1.0034$ s. This change is hardly noticeable in measurements.

The total measuring time is the product of m and the revolution time of the turntable. After accurate measurement of the revolution time, the required time for the measurement T_{meas} must be calculated. In this example, an angular velocity of 45 rpm is used, resulting in $T_{meas} = 184$ s.

To avoid aliasing, the sample frequency f_s of the A/D and D/A converter must be at least twice the highest frequency f_h of the sweep. A small safety margin of 10 % is used. In this example, the sample frequency must be at least 9900 Hz.

The minimum number of data points is the product of T_{meas} and the lowest acceptable sample frequency. In this example, at least $184 \cdot 9900 = 1821600$ data points are necessary. For effective Fourier transformation, the number of data points must be a power of 2. The smallest power of 2 greater than 1821600 is 2^{21} , thus the number of data points is 2097152. Finally, the required sample frequency is calculated. The sample frequency is the quotient of the number of data points and the measuring time. In this example, $f_s = 2^{21}/184 = 11397$.

Since all the required parameters for the measuring signal are calculated, the sweep can be constructed. Using equation 2.8 one long continuous sweep is calculated. This sweep is tapered using the tapering window of equation 3.2. After tapering, the sweep is cut into p equal pieces, which are placed in parallel.

To accomplish that the complete response of the room to the measuring signal is taken in account correctly, a little trick is necessary. Before the actual measuring signal is reproduced (with simultaneously recording of the room response), the last 10 seconds of the measuring signal are reproduced. The additional reverberation that is recorded during the reproduction of the measuring signal is exactly the same as the reverberation of the measuring signal that is NOT recorded after the measuring signal, because the recording stops simultaneously with the reproduction. The periodical properties of the discrete Fourier Transform see to the correct placement of this additional reverberation.

The result of these calculations is a tapered, parallel, linear frequency sweep, impatiently waiting for a room measurement.

4.2 The measuring system

4.2.1 Overview

The goal of the present research is to determine the impulse response of a room. Instead of the impulse response, the response of the room to a linear frequency sweep is measured. From this response, the impulse response is calculated. The measuring system consists of an A/D and a D/A converter and analog devices (microphone, amplifier), see Figure 4.1.

The initial signal is a pulse $\delta(t)$. This signal is convoluted with a sweep, resulting in a linear frequency sweep $s(t)$. This sweep is converted to an analog signal. A radio aerial transmits this analog signal to a rotating radio-receiver on the turntable. The signal from the receiver is amplified and reproduced by the rotating loudspeaker. The response of the room to the signal of the loudspeaker is recorded using a microphone. After A/D conversion, the impulse response $h(t)$ is calculated.

The most important analog components of the measuring system are the D/A and the A/D converter, the turntable, the wireless transmitter, the loudspeaker and the microphone. These parts will be discussed separately.

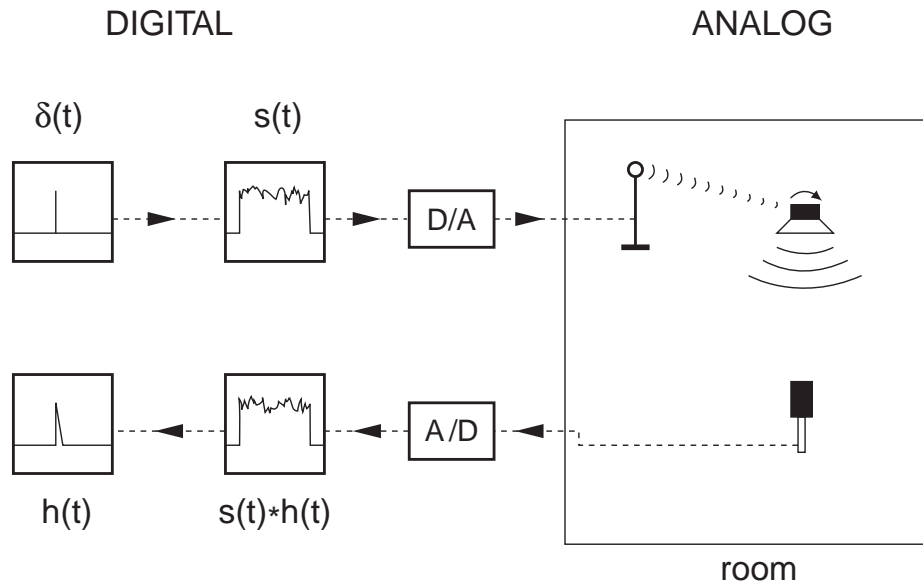


Figure 4.1 Overview of the signal path.

The influence of the components are determined using the same signal as for room measurements, the parallel sweep. The parameters of the sweep are: $f_l = 100$ Hz, $f_h = 4500$ Hz, $p = 32$, $T = 1.3330$ s and $\Delta t^* = 1.0034$ s, resulting in a measurement of 184 s.

4.2.2 The D/A and the A/D converter

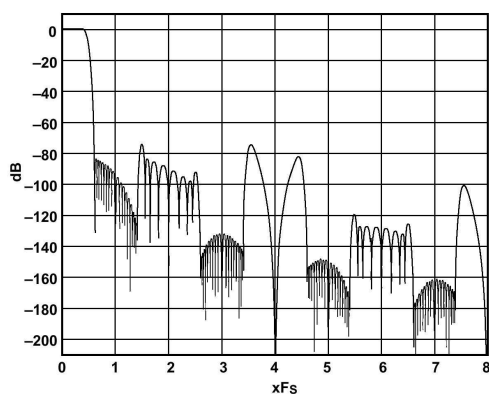
As shown in Chapter 2 and 3, there are some conditions the sweep signal must meet. The number of data points must be a power of 2, for effective Fast Fourier Transformation. The position of the turntable must be the same at the beginning and at the ending of the measurement, requiring an entire number of revolutions during the measurement. Using the sample frequency f_s of the D/A and A/D converter and the revolution time T of the turntable, the restriction can be formulated mathematically by

$$m \cdot T \cdot f_s = 2^k \quad m, k \in \mathbf{N}. \quad (4.2)$$

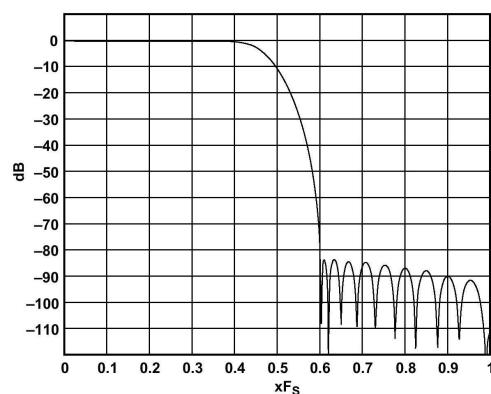
The number of data points (2^k) must be equal to the measuring time ($m \cdot T$) multiplied with the sample frequency. The value of m should be as close to an integer as possible. In section 3.4, simulations have shown that a deviation of 0.03 full circle is acceptable (less is better). This requires that the sample frequency can be adjusted in steps smaller than 0.05% of the sample frequency. Using a sample frequency of approximately 10 kHz, the frequency steps must be smaller than 5 Hz.

Smaller steps are preferred, because the rotational velocity of the turntable is not absolutely constant.

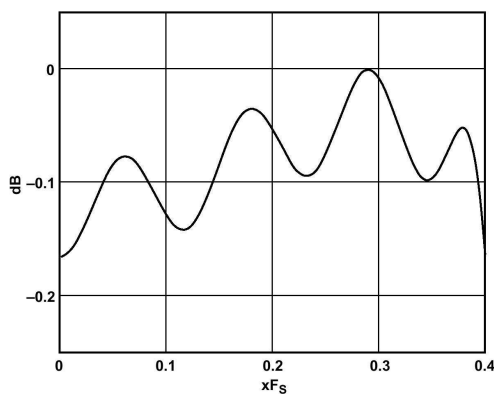
The data acquisition cards available at the department did not meet this requirement. Desirable is a card with a sample frequency adjustable in steps of 1 Hz. Analog Devices produces a chip for PC sound cards, the AD1816A chip, that satisfies the demands of the present research. The A/D converter and D/A converter sample frequencies can be adjusted in steps of 1 Hz independently. The chip has a good anti-aliasing filter with low ripples. The specified transfer functions of the A/D and D/A converter are plotted in Figure 4.2. These figures are taken from the AD1816A Soundport Controller Datasheet [1].



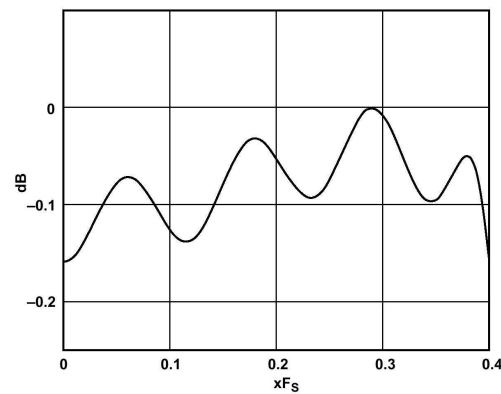
a. DAC transfer function



b. ADC transfer function



a. DAC transfer function - bandpass range



b. ADC transfer function - bandpass range

Figure 4.2 The specified transfer function of the AD1816A.

The figures show that the power spectrum is not completely flat, but the ripple of 0.1 dB is not objectionable. The AD1816A is implemented in several soundcards. For the present research, the Terratec Base 1 is chosen, a little outdated, but still high quality soundcard with the AD1816A chip.

By measuring a constant signal at different sample frequencies, it is clearly shown that the soundcard is capable of adjusting the sample frequency in steps of 1 Hz. Unfortunately, the full-duplex operation (simultaneous playback and recording) at arbitrary sample frequencies is not satisfactory. The input sample frequency and the output sample frequency differ slightly, approximately 0.02%. In a measurement of 2-3 minutes, this results in the loss of 30 data points. This is not acceptable, as well the bad connection in the FFT as the (small) frequency shift. Expensive and high quality soundcards simulate a precisely adjustable sample frequency by hardware sample rate conversion. Whether this is a suitable solution, is not investigated in the present research. The AD1816A does not lose data points at the 'standard' soundcard frequencies (6000, 8000, 11.025, 22.050, 44.100 and 48.000 Hz). The good overall performance and excellent documentation of the AD1816A, led to the decision to use the AD1816A in combination with software sample rate conversion. The advantage of software sample rate conversion is the available time for filtering and pre- and post-processing (for example anti-alias filtering). Measurements with the AD1816A and software sample rate conversion show good results. This is shown in Figure 4.3.

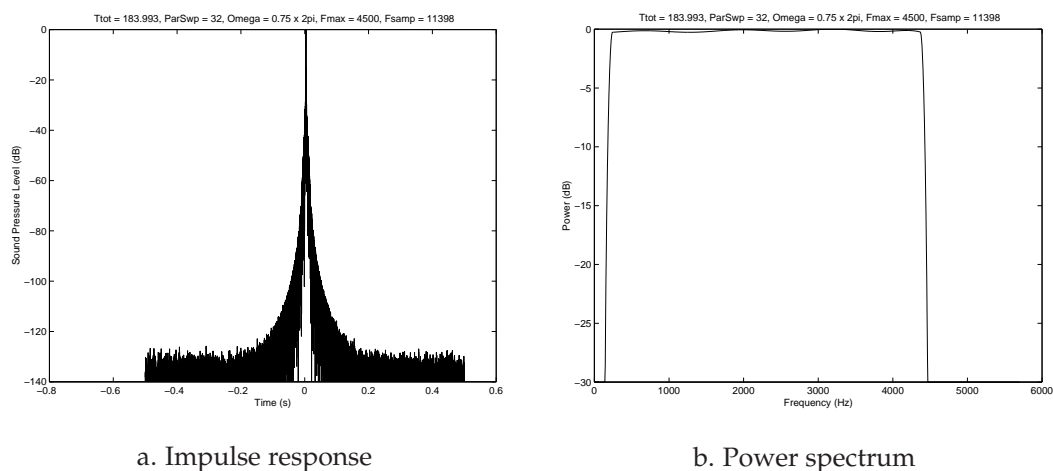


Figure 4.3 The impulse response and the power spectrum of the D/A and A/D converter. Both plots have a dB scale at the vertical axis

The quality of the soundcard and the software sample rate conversion is measured by short circuiting the output of the D/A converter to the input of the A/D converter. The original sweep signal is at a sample rate of 11398 Hz and is converted to 11025 Hz using the sample rate conversion function of CoolEdit Pro¹. After simultaneous recording and playback with CoolEdit Pro, the signal is converted back to the original sample frequency of 11398 Hz. In the forward and back conversion,

¹CoolEdit Pro is a multitrack recording program of Syntrillium Software Corporation. The program is user friendly and allows for meticulous editing of the signal. For more info, look at <http://www.syntrillium.com>.

one or at most two data points are lost in total. The S/N ratio of the impulse response is higher than 120 dB. The power spectrum shows the same small ripples as in the specified transfer function (see Fig. 4.2). The ripples are the result of a D/A and an A/D conversion and thus the combination of Figure 4.2c and 4.2d. The complicated way of down and upsampling instead of just 1 sample rate conversion after the measurement is necessary because the FFT is very sensitive to loss of just a few datapoints. Loss of 30 datapoints results in a distorted power spectrum. The total number of data points is 2^{21} , so the precision required is 0.0014% or 0.16 Hz at a sample frequency of 11.025 Hz. Simulations show that a conversion to a nearby frequency results in a better conversion, less data points are added or have disappeared.

4.2.3 The turntable

The turntable must rotate a high and constant rotational velocity. A low revolution time shortens the measuring time but raises the influence of Doppler effects. To accomplish a maximum measuring time of a few minutes, a rotational velocity of 30 revolutions per minute is required. At the Laboratory of Acoustical Imaging and Sound Control, no turntable with such a high rotational velocity is available. In the present research, a record-player is used. A record-player rotates at $33\frac{1}{3}$ or 45 revolutions per second. The rotational velocity is constant, the main problem is the maximum load the record-player can handle. The loudspeaker, receiver and battery pack together weight approximately 1 kg. A prehistoric record-player, the 'Dual 1228 turntable' proves to be capable of a constant rotational velocity with a load of 2 kg. This record-player has been produced from 1973 to 1974. The specified wow & flutter is 0.09%.

The revolution time must be determined very accurately. The use of a reflective sensor to measure the revolution time is illustrated in Figure 4.4.

The Honeywell HOA1405 [4] serves the purpose of reflective sensor very well. The HOA1405 consists of an infrared emitting diode and a NPN silicon phototransistor encased in a single package. The switching time of $5 \mu\text{s}$ of the detector is sufficient to measure the revolution time with high accuracy. The light of the IR emitter is reflected by the rotating platter with the loudspeaker. The amount of light received by the detector depends on the reflection coefficient of the platter. On the side of the platter, a small black, light absorbing strip is mounted. Passing by of the strip reduces the light received by the detector and thus the output of the detector. The output of the detector is recorded using an A/D converter, the right channel of the soundcard is used for this purpose. In Figure 4.5, the output of the reflective sensor is plotted.

In Figure 4.5 the passing by of the black strip is clearly visible, the output voltage

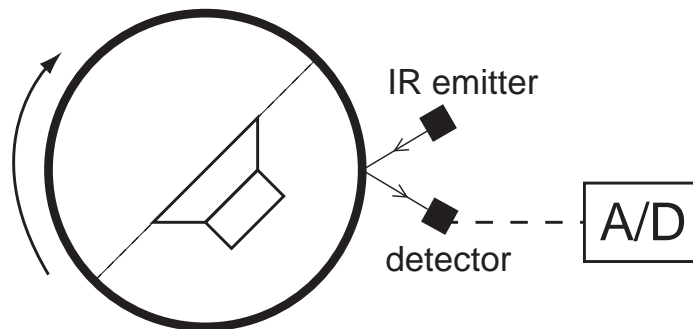


Figure 4.4 A reflective sensor is used to determine the revolution time of the turntable. The IR light of the IR emitter is reflected by the rotating platter with speaker. The amount of reflected light determines the output of the receiver. The A/D converter reads the output of the detector into a computer.

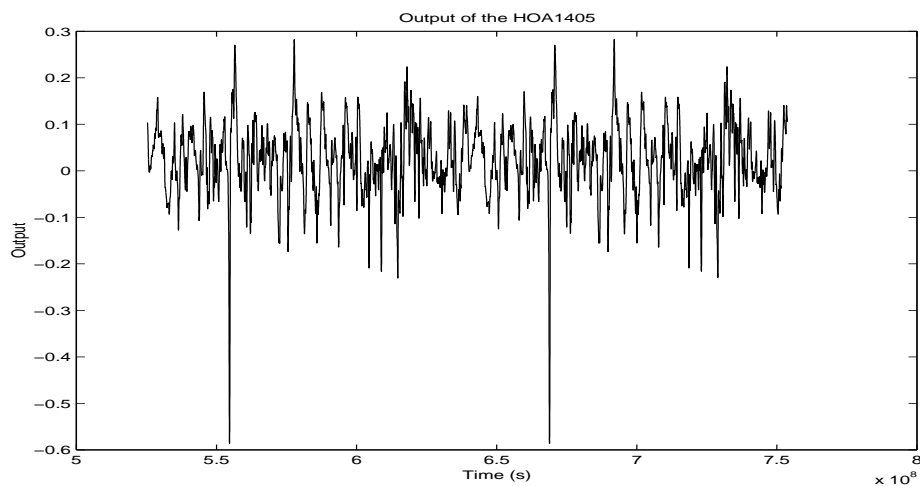


Figure 4.5 The output of the reflective sensor HOA1405. The record-player is rotating at a speed of 33 rpm.

decreases clearly. The revolution time can be calculated very accurately. In Figure 4.6, the calculated revolution time is plotted for a 1 minute measurement.

The angular velocity of the turntable is quite constant. The maximum deviation of the revolution time in 40 revolutions is sometimes only 0.0005 or 0.03 % compared to the average revolution time. For the Doppler effects, this will not have any influence.

More important than a slight variation in the angular velocity, is the trend in the angular velocity. In a period of an hour, the average revolution time of the turntable can change slowly from 1.3345 to 1.3328. This change can be caused by small expansion of parts in the turntable drive caused by heating. This requires a specific

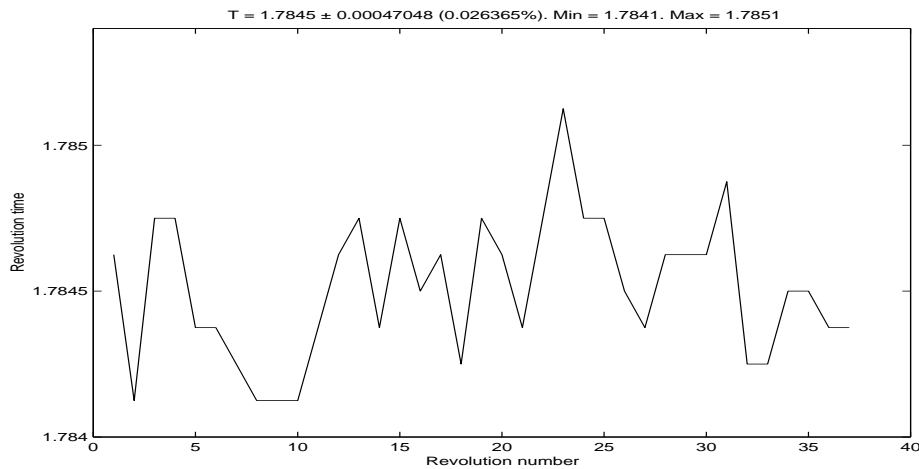


Figure 4.6 The calculated revolution time of the Dual 1228 turntable.

order for a measurement. First, the angular velocity is determined. Secondly, the sweep signal is calculated using the measured angular velocity. Finally, the impulse response is measured using the recently generated sweep signal. During the measurement of the impulse response, the angular velocity is monitored to check whether the adjusted angular velocity is equal to the actual angular velocity.

4.2.4 The wireless transmitter

The linear frequency sweep is reproduced using a loudspeaker placed on a turntable. It is not possible to connect the loudspeaker using a wire. The signal can be radio transmitted to the loudspeaker or transmitted using a sliding contact.

A sliding contact has some major disadvantages: noise caused by surface roughness and wearing. To reduce noise, use of a soft and less durable sliding contact is inevitable, resulting in high wear of the sliding contact. The sliding contact should be replaced approximately every 6 hours, resulting in a non-duplicative measuring system which is not acceptable.

In the present research, radio transmission is used to provide the speaker with the output signal. To guarantee a good transmission, the sender and receiver of a HiFi wireless headphone are used. The Sennheiser RS4 fulfills the demands, the S/N ratio is 70 dB, the audio frequency response range is 20 - 19.500 Hz. The signal is transmitted using a carrier frequency of 434 MHz. The output signal of the receiver is fed to a small amplifier, the Velleman K4001. This is a small amplifier with a S/N ratio of 86 dB and a low supply current. A low supply current is necessary because the voltage for the receiver and the amplifier must be supplied by a battery pack. The quality of the radio transmission can be measured by feeding the A/D converter with the output signal of the amplifier. In Figure 4.7 the impulse response

and the power spectrum are plotted.

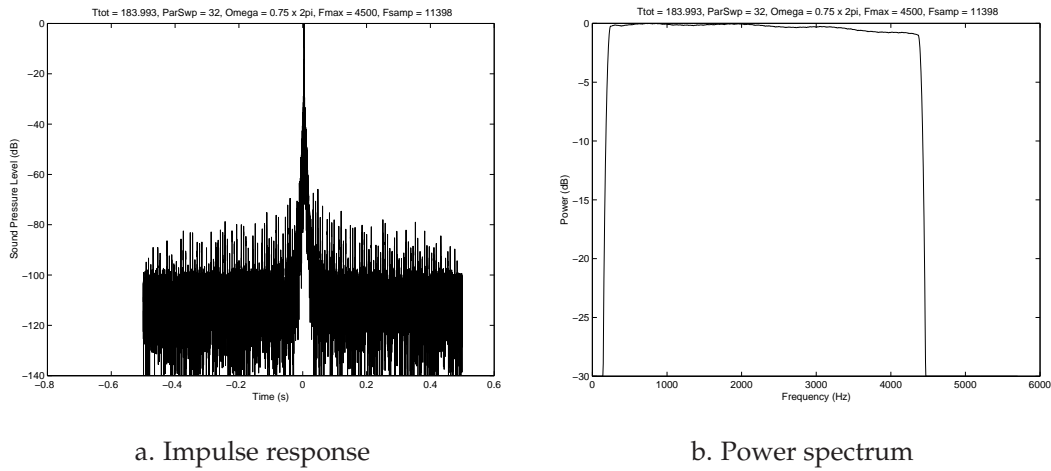


Figure 4.7 The impulse response and the power spectrum of the combination of the radio transmitter with the radio receiver. Both plots have a dB scale at the vertical axis

The impulse response has a S/N ratio of 100 dB, satisfactory for the present research. The power spectrum has a slight decay at the higher frequencies, resulting in -1 dB at 5000 Hz.

The radio transmitter and receiver reduce the signal quality, but not significantly.

4.2.5 The loudspeaker and the microphone

The major part of the measuring system is the loudspeaker. The loudspeaker box is placed 40 cm above the platter of the turntable to minimize the influence of reflections caused by the turntable. The loudspeaker housing has an internal volume of 1.5 dm³ and is filled with sound absorbing material. The loudspeaker used is a Vifa M-110, a compact 4" midrange unit.

The frequency response is omnidirectional to 2 kHz. Thus the dipole and higher order pole effects will be visible only for frequencies above 2000 Hz.

The microphone used is the Brüel & Kjær condenser microphone type 4125. The sensitivity is plotted in Figure 4.8.

The characteristics of the loudspeaker and microphone have been determined under anechoic conditions, in the anechoic room of the Laboratory of Acoustical Imaging and Sound Control. The loudspeaker is placed on top of the record-player, the record player being switched off, not rotating. The impulse response and the power spectrum are plot in Figure 4.9.

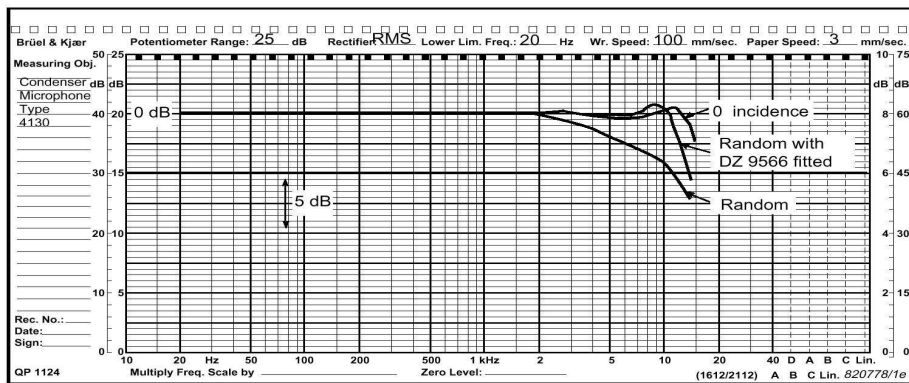


Figure 4.8 The frequency response characteristics of the B&K 4125 condenser microphone.

The impulse response of Figure 4.9 clearly shows that the room is not perfectly anechoic: the reverberation time, defined as the time needed for a 60 dB decrease of the sound pressure level, is of the order of 40 ms. The travel time from the loudspeaker to the microphone is 10 ms, which corresponds to 3.4 m.

The power spectrum is not flat. The small irregularities are caused by the fine resolution of the spectrum. The large deviations (-5 dB for $f < 1500$ Hz and $f > 3000$ Hz) are the result of the non-ideal behaviour of the loudspeaker.

The impulse response of the microphone-loudspeaker combination is very good, the power spectrum is acceptable. For proof of the theory and the simulations, this loudspeaker is usable. For precise measurements in rooms, a better loudspeaker is advisable.

4.3 Measurements

The impulse response can be measured under anechoic or under reverberant (or echoic) conditions.

Anechoic measurements are done in the anechoic room of the Laboratory of Acoustical Imaging and Sound Control. The reverberation time is very short in the anechoic room (see above), which makes this room very useful for measurements with reduced complexity. Due to the absence of reverberation, the impulse response is a sharp pulse, not a pulse with a decay.

Echoic measurements can be done in any arbitrary room. The reverberation results in an impulse response being a pulse train with a decay. The main goal of the present research is to develop an omnidirectional source for impulse response measurements in rooms. Theory and simulations showed the opportunities of the rotating source, these experiments have to prove them.

Due to the high sensitivity of the measurements to a slight change in the angular

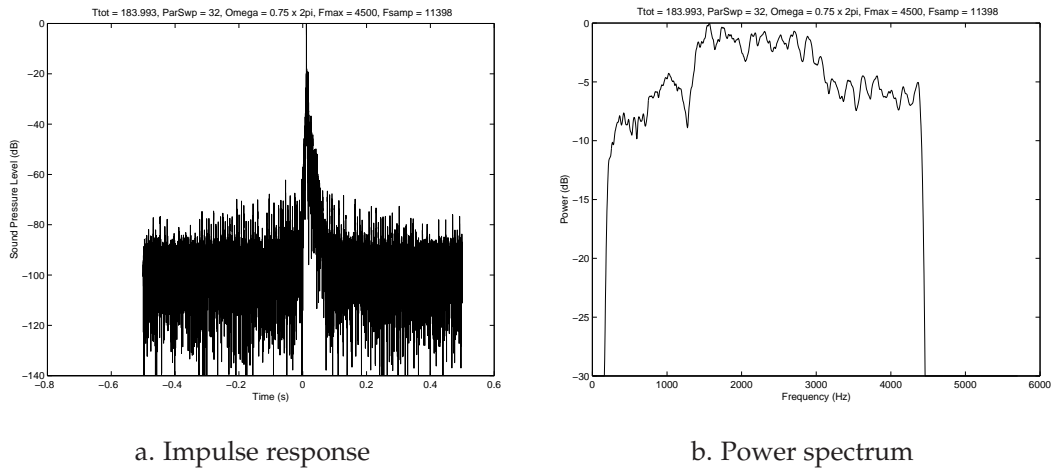


Figure 4.9 The anechoic impulse response and the power spectrum of the combination of the microphone and the loudspeaker. The loudspeaker is not rotating. Both plots have a dB scale at the vertical axis

velocity of the turntable, array measurements, being the trend nowadays in advanced room acoustic research, are impractical since they require many successful measurements. With the equipment used, about 50 % of the measurements led to a useful result. Taking in account the required measuring time of 3 minutes, 7 useful measurements in an hour is the maximum achievable.

The purpose of the measurements in the present research is to validate the results of the simulations. Building of a operational measuring system for array measurements would require more time than available in the present research.

4.3.1 Anechoic measurements

The anechoic measurements were done in the anechoic room of the Laboratory of Acoustical Imaging and Sound Control. The parameters are essentially the same as for the testing of the measuring equipment. The parameters used are: $f_l = 100$ Hz, $f_h = 4500$ Hz, $p = 32$ and $\Delta t = 1$ s. With a (pre-) measured revolution time of 1.3315, the duration of this sweep is 184 seconds. The angular velocity is recorded simultaneously with the impulse response, thus afterwards the actual revolution time can be calculated. The impulse response and power spectrum of three successful measurements are plotted in Figure 4.10.

The plots look very similar. In accordance with simulations, many peaks are visible. The peaks are located exactly at a time difference of 1 second, just as predicted by theory and simulations. The S/N ratio of more than 60 dB in the impulse response is an excellent result.

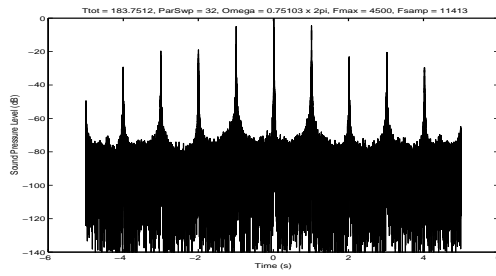
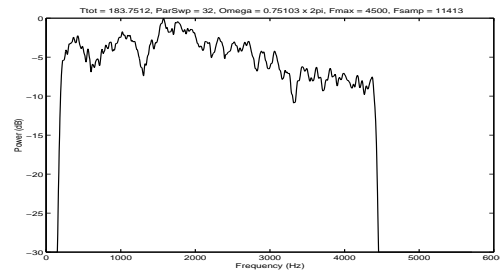
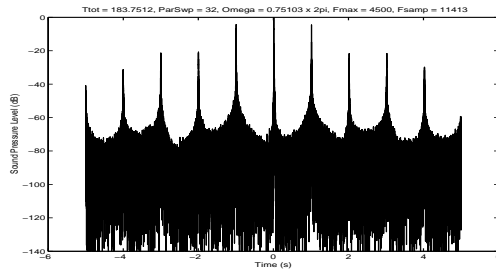
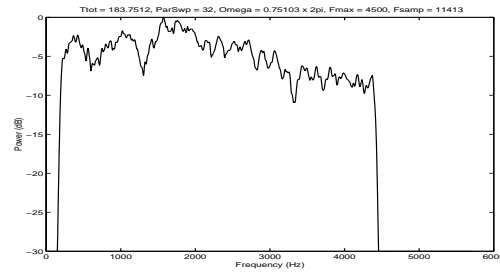
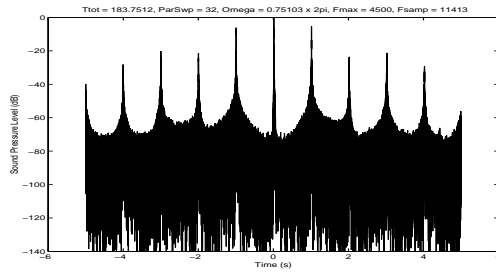
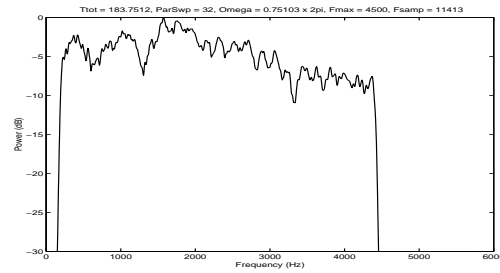
a. Impulse response, $m = 137.99$ b. Power spectrum, $m = 137.99$ c. Impulse response, $m = 137.97$ d. Power spectrum, $m = 137.97$ e. Impulse response, $m = 138.07$ f. Power spectrum, $m = 138.07$

Figure 4.10 The impulse response and power spectrum of three successful measurements (both dB scale).

Comparison of these plots with the power spectrum of the non-rotating loudspeaker (see Fig. 4.9) shows some interesting results. The power spectrum of the rotating loudspeaker has relatively more power at the lower frequencies. This is in accordance with decomposition in monopole, dipole and higher order pole characteristics. A single loudspeaker is better omnidirectional (thus has more 'monopole content') at lower frequencies. Secondly, a constant decline of power at frequencies higher than 2000 Hz is visible. This is in accordance with the plots for a rotating diaphragm (see Fig. 3.11f).

The power spectrum is calculated by Fourier transformation of the monopole impulse response. Of course, the power spectrum of the other peaks can be calculated

as well. In Figure 4.11, the power spectrum of the monopole pulse and the power spectrum of the dipole pulse are plot next to each other.

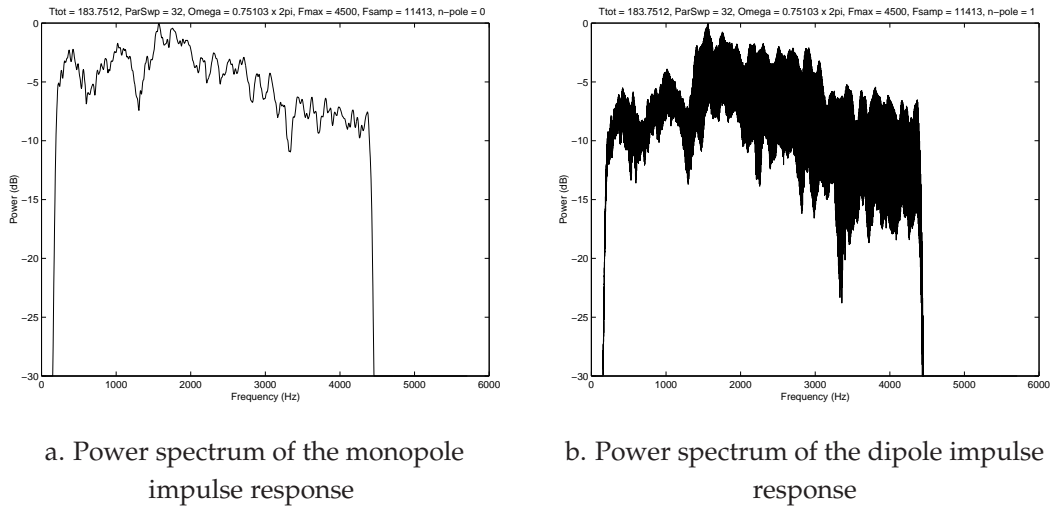


Figure 4.11 The power spectra of the monopole and the dipole impulse responses respectively for a measurement with a rotating loudspeaker and $m = 137.99$

The power spectrum of the dipole is more distorted, resulting in a 'surface' instead of a single line. The dipole power spectrum relatively has a little less power at lower frequencies and a little more power at higher frequencies. This is accordance of the directivity of a loudspeaker, more bundled ('dipole content') at higher frequencies.

The conclusions are clear: the experimental results are in perfect accordance with the results of the simulations. Under anechoic conditions, a S/N ratio of 75 dB is possible. It is necessary to compensate for the non-flat power spectrum of the Vifa M110, but not in the present research.

4.3.2 Echoic measurements

The anechoic measurements were done in a moderately small room, room D266 at the Laboratory of Acoustical Imaging and Sound Control. The size of the room is 5m x 5m x 3m. The walls are reflective, but much sound absorbing obstacles are present in the room. The estimated reverberation time is 0.5 s. The parameters of the measuring signal are the same as for the testing of the measuring equipment and for the anechoic measurements: $f_l = 100$ Hz, $f_h = 4500$ Hz, $p = 32$ and $\Delta t = 1$ s. In Figure 4.12 the same impulse response is plotted for three different time windows together with the power spectrum of the monopole impulse response.

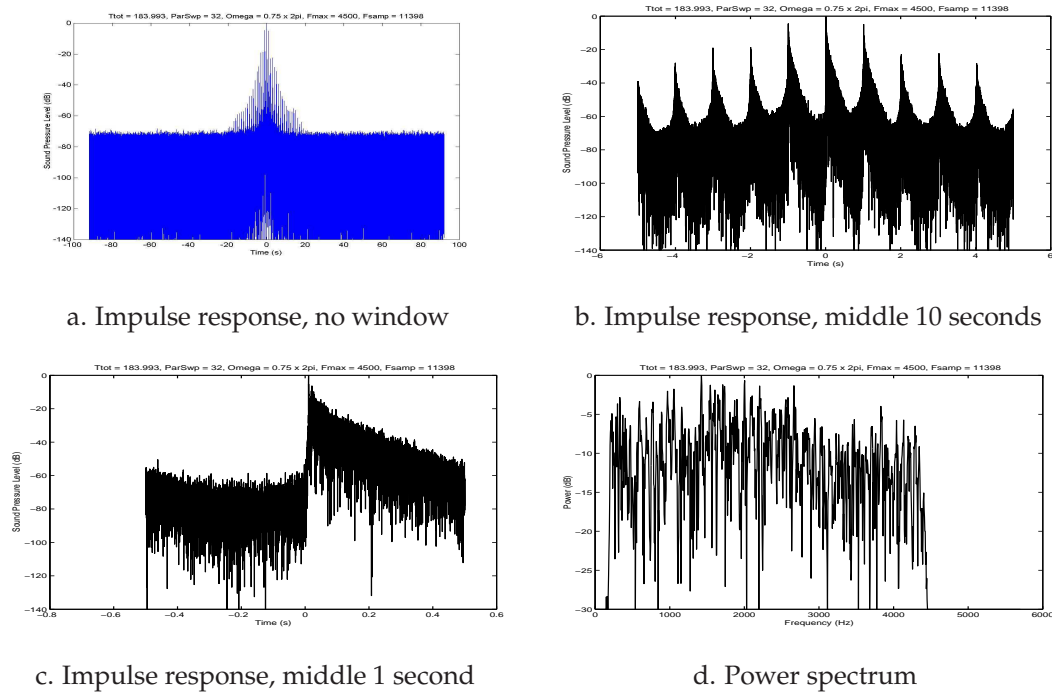


Figure 4.12 The echoic impulse response of the rotating loudspeaker. In plot a, b and c the windows are 184, 10 and 1 seconds respectively. In plot d the power spectrum of the monopole impulse response is plotted (all dB scale).

The impulse response is in accordance with the expectations. The reverberation time is approximately 0.5 s. In Figure 4.12c the short delay between the direct sound and the early reverberation is visible. The S/N ratio is 60 dB, which is a good result. The power spectrum looks very random, but vaguely the shape of Fig 4.11 can be recognized. The vicious pattern is likely the result of the used method for calculation of the power spectrum. Under echoic conditions, Fourier transformation of a small window around the monopole impulse response is no appropriate method for calculation, although this method served its purpose very well in all other situations.

To allow for a good comparison with the ‘Beentjes bol’, the echoic measurements are also done using the ‘Beentjes bol’ instead of the rotating loudspeaker. In Figure 4.13 the impulse response is plotted in a 1 second window around the highest pulse for the rotating loudspeaker as well as the ‘Beentjes Bol’.

The rotating loudspeaker clearly has a high S/N ratio. The impulse response of the rotating loudspeaker is superior over the impulse response of the ‘Beentjes bol’. The S/N ratio of 40 dB for the ‘Beentjes Bol’ using is signal of of 3 minutes is not satisfactory at all.

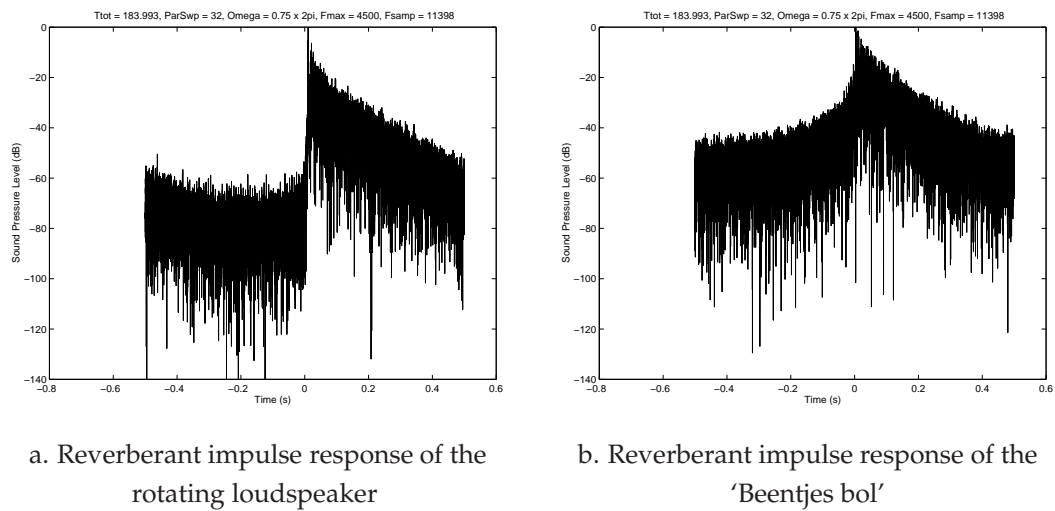


Figure 4.13 The impulse response of the rotating loudspeaker compared to the impulse response of the 'Beentjes bol'

In a horizontal plane, the the rotating loudspeaker is onnidirectional to 4500 Hz, opposed to the 2000 Hz of the 'Beentjes Bol'.

Chapter 5

Conclusions and recommendations

5.1 Conclusions

5.1.1 Theory and simulations

- Using a linear frequency sweep and a rotating source, it is possible to decompose the directivity pattern of the source in monopole, dipole and higher order pole contributions. A monopole radiates omnidirectional, thus is ideal for impulse response measurements.
- The required time for measurements is too long without use of a parallel sweep. The major disadvantage of a parallel sweep is a high sensitivity to a not entire number of rotations. The angular velocity must vary less than 0.05%.
- Doppler effects have a negative influence on the power spectrum. High frequencies are more suppressed for larger sources or higher angular velocity. A very high angular velocity is not a good solution to reduce the time required for measurements.
- A parallel sweep has a very high S/N ratio and is therefore applicable in all sort of situations to improve S/N ratio in broadband measurements.

5.1.2 Experiments

- The results of the measurements prove the validity of the theory and simulations.
- A S/N ratio of 60 dB in the impulse response of a room is possible using a parallel linear frequency sweep.

- The impulse response of the rotating monopole looks much better than the impulse response of the 'Beentjes Bol'. The impulse response of the 'Beentjes bol' is not satisfactory, especially considering the poor omnidirectionality.
- The quality of a decent (multi-channel) soundcard is satisfactory for most scientific experiments. Considering the quality of multitrack mixer programs and their ease in use (on screen tracking of measurements), soundcards are sometimes preferable over 'scientific data acquisition cards'. A condition for successful use of a soundcard is the availability of a good documentation.

5.2 Recommendations

- For further research, a solution for a successful extension from this 2-D monopole to a 3-D monopole deserves highest priority. A 2-D monopole serves no purpose in room acoustics. Use of a scattering surface is one of the possible solutions.
- Use of a ready-to-use transmitter-receiver-loudspeaker combination must be preferred over the separate element approach of the present research.
- The method used for deconvolution is robust but cannot compensate for a non-flat power spectrum. For compensation of Doppler effects in the power spectrum, an another deconvolution operator is necessary.
- A turntable rotating with a very constant angular velocity is indispensable. If the turntable rotates more constantly, more sweeps in parallel are possible.

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Appendix A

Extension from complex to real signals

In Chapter 2, an expression for the time shift t_n of the pulse in the impulse response is deduced. This expression is deduced using a complex sweep with the directivity of a complex n-pole. Using the linearity of all the operations in the derivation, this result can be extended to real situations.

The impulse response of a complex sweep and a rotating source with a complex directivity is only non-zero at the time t_n given by (Eq. 2.16) :

$$t_n = -\frac{\frac{1}{2}n\psi}{2\beta} = -\frac{\frac{1}{2}n\psi}{2 \cdot \frac{1}{2} \cdot 2\pi \cdot \frac{f_h - f_l}{T_{sweep}}} = -\frac{\frac{\frac{1}{2}n\psi}{2\pi}}{\frac{f_h - f_l}{T_{sweep}}} \quad (\text{A.1})$$

To extend this real situations, both the sweep and the directivity must be extended from a complex signals to a real signal.

Firstly, a real sweep with a complex directivity is analyzed.

The frequency spectrum of a complex sweep (Eq. 2.9) is only non-negative for $f > 0$. The frequency spectrum of a real sweep is non-zero for $-f_h < f < f_l$ and $f_l < f < f_h$. Using the linearity of all the operations, the effects of multiplication and deconvolution on the parts $f < 0$ and $f > 0$ can be regarded separately:

f > 0 The effects of the multiplication and deconvolution on the part $f > 0$ are described in figures 2.5 and 2.6 and in the theory above, resulting in a pulse at $t = t_n$

f < 0 The effects on the part $f < 0$ are different. Multiplication with the complex directivity causes a shift to the right in the frequency spectrum with $\frac{1}{2}n\psi/2\pi$ Hz. Contrary to the part $f > 0$, this is a shift towards the lower frequencies (less negative). Deconvolution of the shifted frequency spectrum $f < 0$ results in a pulse at $t = -t_n$.

Summarizing: deconvolution of a real sweep with a complex directivity results in

2 pulses: one at $t = t_n$ and one at $t = -t_n$.

Secondly, a complex sweep with a real directivity is analyzed.

The frequency spectrum $D(\omega)$ of a real directivity, for example $d(t) = \cos(\frac{1}{2}n\psi t)$, consists of two peaks: one at $\omega = -\frac{1}{2}n\psi$ and one at $\omega = +\frac{1}{2}n\psi$. The convolution in the frequency domain of $D(\omega)$ and $S(\omega)$ consists of 2 shifted and partly overlapping complex sweep spectra. One is shifted $\frac{1}{2}n\psi$ rad/s to the left (towards lower positive frequencies), the other $\frac{1}{2}n\psi$ rad/s to the right (towards higher positive frequencies). Multiplication with $S^*(\omega)$ (deconvolution with $s(t)$ in the time domain) again results in two peaks in the result in the time domain:

The 'left-shifted complex sweep' produces a pulse at $t = -t_n$, the 'right-shifted complex sweep' results in a pulse at $t = +t_n$.

Thirdly, a real sweep with a real directivity is analyzed.

The only part of $S(\omega) * D(\omega)$ that is not already discussed, is the convolution of $S(\omega)$ for $f < 0$ and $D(\omega)$ for $f < 0$. This is a complex sweep at only negative frequencies that has been shifted to the left, to the higher negative frequencies. After multiplication with $S^*(\omega)$, this results in the time domain in a peak at $t = +t_n$.

In the next table shows an overview of the position of the peaks as a result of the positive or negative frequency parts of $S(\omega)$ and $D(\omega)$:

		$S(\omega)$	
		$f < 0$	$f > 0$
$D(\omega)$	$f < 0$	t_n	$-t_n$
	$f > 0$	$-t_n$	t_n

Conclusion: a real sweep and a real directivity of the form $d_n(t) = \cos(\frac{1}{2}n\psi t)$ result in a calculated impulse response consisting of two peaks, one at $t = -t_n$ and one at $t = +t_n$ (t_n being given by Eq. A.1).

In reality, the directivity will never be as simple as $d(t) = \cos(\frac{1}{2}n\psi t)$. The directivity will consist of monopole, dipole, quadrupole and higher order terms. A linear frequency sweep and a rotating source rotating with angular velocity ψ together have the special property to decompose the directivity of the source into the separate poles.

Appendix B

Active Reflection

B.1 Abstract

In September 1998, O.M. van der Wiele started a M.Sc. research project on active reflection using a patent of P.A. Swarte. The goal was to develop a wall with variable acoustic properties using active transducers that operate as well as loudspeakers and as microphones.

During the research, some problems arose which led to the suspicion that the idea of P.A. Swarte was not realizable. In January 1999, O.M. van der Wiele wrote a intermediate report with the cautious conclusion the idea was not practical realizable. This report (in Dutch) is included as section B.3.

In Februari 1999, a theoretical obstruction for the idea of P.A. Swarte was discovered. Electro-dynamic loudspeakers are anti-reciprocal, which causes a non-applicability for the idea of P.A. Swarte. A detailed explanation can be found in section B.4 (also in Dutch).

B.2 Samenvatting

In september 1998 is O.M. van der Wiele begonnen aan een afstudeeronderzoek naar actieve reflectie zoals beschreven in een patent van P.A. Swarte. Het doel was een wand te ontwikkelen met variable akoestische eigenschappen door gebruikmaking van transducers die zowel als luidspreker en als microfoon dienst doen.

Gedurende het onderzoek leidden een aantal problemen tot het vermoeden dat het idee van P.A. Swarte niet uitvoerbaar was. In januari 1999 is door O.M. van der Wiele een tussenrapport geschreven met de voorzichtige conclusie dat het idee niet praktisch uitvoerbaar was. Dit verslag (in het Nederlands) staat in sectie B.3.

In februari 1999 werd een theoretisch bezwaar tegen het idee van P.A. Swarte ontdekt. The anti-reciprociteit van elektrodynamische luidsprekers maken die ongeschikt voor toepassing als in het idee van P.A. Swarte. Een gedetailleerde uitleg hiervan is in sectie B.4 te vinden (ook in het Nederlands).

B.3 Verslag januari 1999

B.3.1 Het onderzoek in vogelvlucht

Doel van het onderzoek

Op 8 december 1987 is door Peter Swarte (toen in Philips-dienst) een patent aangevraagd op een methode om actief de reflectiecoëfficiënt van een luidspreker te kunnen variëren. Dit betreft US Patent 4,712,247: "Electro-acoustic systems having a variable reflection/absorption characteristic." Bijzonder aan de methode beschreven in het patent is dat de luidspreker zowel als opnemer en als weergever gebruikt wordt. De methode is principieel anders dan anti-geluid. Bij anti-geluid wordt het directe geluidsveld zoveel mogelijk opgeheven door een veld in tegenfase te maken. Bij de methode van Peter Swarte wordt alleen de hoeveelheid reflectie gevarieerd.

Toepassing van zo'n systeem zal vooral in multi-purpose zalen plaatsvinden, waar verschillende akoestische eigenschappen gewenst zijn voor verschillende toepassingen van de zaal, bv voor een symposium (spraak), voor toneel of voor een concert. Het is de bedoeling om een systeem te ontwikkelen waarbij je met één druk op de knop een wand meer of minder reflecterend kan maken. Een andere toepassing kan gevonden worden in geluidsisolatie. Door een wand (bijna) volledig reflecterend te maken, zal er weinig vermogen door de wand opgenomen worden en dus weinig geluid door de wand heen gaan. Of dit idee correct is, zal later uitgeprobeerd moeten worden.

In de metingen wordt alleen gekeken naar frequenties onder de 1500 Hz. Boven deze frequentie gaat de konus van een breedbandluidspreker opbreken. De konus van de luidspreker is dan niet stijf genoeg en dit geeft problemen bij weergave. Tevens geldt dat de golflengte dan ongeveer 20 cm is en dus van dezelfde grootteorde is als diameter van de konus, waardoor er faseverschillen op de konus ontstaan bij niet loodrecht invallende golffronten. Dit maakt het onmogelijk om goed de fase op te nemen.

De resultaten in het kort

De opzet van het onderzoek was om aan eerst te tonen dat de methode van P.A. Swarte werkt in de praktijk in een meetopstelling met een luidspreker. Daarna moet het idee geïmplementeerd worden in een paneel met meerdere luidsprekers, waarvan de reflectiecoëfficiënt eenvoudig gevarieerd kan worden.

Tot nu toe is gebleken dat de methode in de praktijk niet goed werkt. De oorzaak hiervan ligt in de ongevoeligheid van de luidspreker als opnemer. Het signaal uit de luidspreker moet zo veel versterkt worden, dat het systeem instabiel wordt.

B.3.2 Het idee

Wat is nieuw aan dit idee

Er zijn al vele elektro-akoestische systemen waarmee de akoestische eigenschappen van een ruimte beïnvloed kunnen worden. Met een microfoon wordt een geluidssignaal opgenomen en na versterking en eventuele bewerking wordt dat weergegeven via een luidspreker. Bij de plaatsing van de microfoon en luidspreker moet rekening gehouden worden met knopen van de staande golven. Nadeel van deze systemen is dat de prestaties niet geweldig zijn en dat het heel duur is om een systeem te maken waarmee ook de absorptie van de wanden verhoogd kan worden.

In het systeem bedacht door Peter Swarte wordt niet eerst het geluid met een microfoon opgenomen, versterkt en daarna weergegeven via een luidspreker, maar kan van wanden met luidsprekers de absorptie (en dus ook de reflectie) variabel ingesteld worden. Hierin worden luidsprekers niet alleen gebruikt voor weergave, maar ook voor de opname van het geluid. Daarnaast zijn een of twee weerstanden en een of twee versterkers nodig per luidspreker. Hiermee kan het gedrag van de luidsprekers beïnvloed worden. Het systeem kan zo worden ingesteld dat de luidspreker haast zonder kracht van de lucht op de konus meebeweegt met de lucht. De lucht merkt dus weinig van de luidspreker en de luidspreker absorbeert het geluid bijna volledig. Daarentegen kan ook actief de konus zo stil mogelijk gehouden worden, zodat het geluid een zeer harde wand ziet en vrijwel volledig reflecteert.

De theorie

In de eenvoudige vorm bestaat het systeem uit een spanning-spanning-versterker, een impedantie Z_1 en een luidspreker (zie figuur B.1).

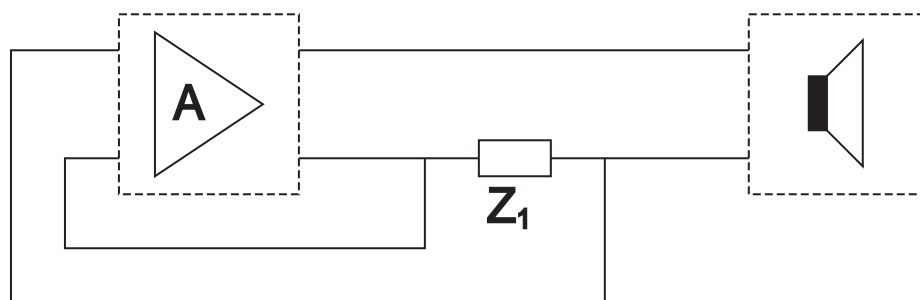


Figure B.1 Het systeem in eenvoudige vorm

De luidspreker en de impedantie staan in serie aan de uitgang van de versterker. De spanning over de impedantie is het ingangssignaal van de versterker (terugkoppeling). Luchtdrukverschillen ten gevolge van het aanwezige geluidsveld resul-

teren in beweging van de konus van de luidspreker en dus ook van de spreekspoel in het magneetveld. De luidspreker kan schematisch weergegeven worden door een spanningsbron u_1 en een inwendige impedantie Z_L (zie figuur B.2).

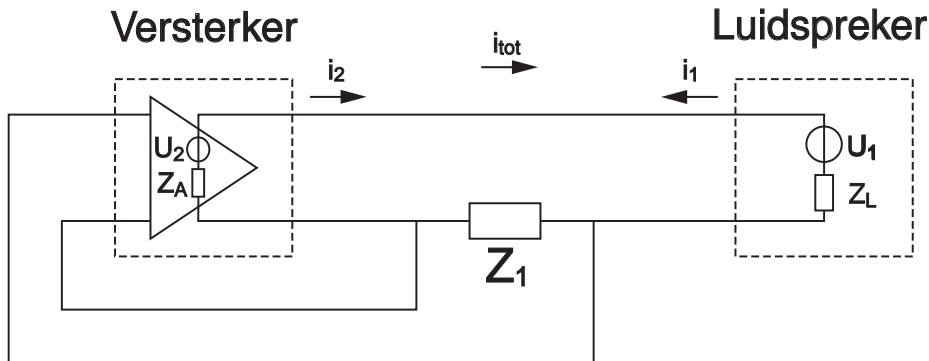


Figure B.2 Het systeem uitgewerkt

De versterker wordt schematisch weergegeven door een spanningsbron u_2 met versterking A en een inwendige weerstand Z_A . Zowel de versterking A als de impedantie Z_1 kunnen variabel of vast zijn. Als i_1 de stroom is ten gevolge van de bron in de luidspreker en i_2 die tengevolge van de bron in de versterker, kan voor de totale stroom i_{tot} (die rechtsoom wordt gekozen):

$$i_{tot} = i_2 - i_1 = \frac{u_2 - u_1}{Z_A + Z_L + Z_1}. \quad (\text{B.1})$$

De uitgangsspanning van de versterker kan als volgt geschreven worden:

$$u_2 = A \cdot Z_1 \cdot i_{tot}. \quad (\text{B.2})$$

Invullen van formule B.1 in formule B.2 geeft als resultaat:

$$\frac{i_{tot}}{u_1} = \frac{1}{-Z_{tot} + AZ_1}, \quad (\text{B.3})$$

met $Z_{tot} = Z_A + Z_L + Z_1$.

In figuur B.3 is de functie van i_{tot}/u_1 uitgezet tegen de versterking A .

De hyperbool heeft een pool bij $A = Z_{tot}/Z_1$. Voor $A > Z_{tot}/Z_1$ is het systeem instabiel. Zeer grote negatieve versterking leidt tot $i_{tot} = 0$, dus een stilstaande konus (veel reflectie). Als A net iets kleiner gekozen wordt dan Z_{tot}/Z_1 , is i_{tot} groot en negatief, dus de totale stroom heeft dezelfde richting als i_1 en dan beweegt de konus mee met het externe geluidsveld). Als de waarde van A van zeer negatief richting de pool loopt, zal de reflectie steeds minder worden en de absorptie toenemen. Oftewel: het akoestische gedrag van de luidspreker kan gevarieerd worden door de waarden van A en Z_1 te variëren.

Met behulp van een extra weerstand en een extra versterker, kan het systeem beter regelbaar worden, maar het basisprincipe blijft gelijk.

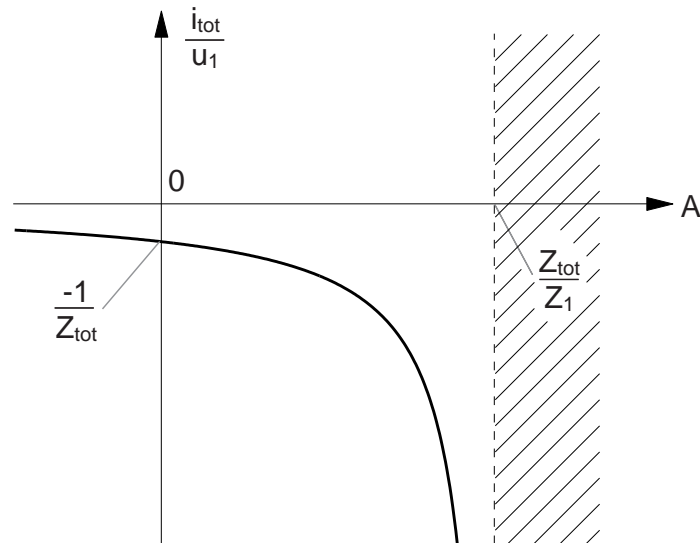


Figure B.3 De functie van $\frac{i_{tot}}{u_1}$ uitgezet tegen de versterking A .

B.3.3 De interferometer

Werking van de interferometer

Het doel van het onderzoek is om de reflectiecoëfficiënt van een luidspreker in te kunnen stellen. De reflectiecoëfficiënt wordt bepaald met een interferometer. De gebruikte interferometer ziet er schematisch als volgt uit:

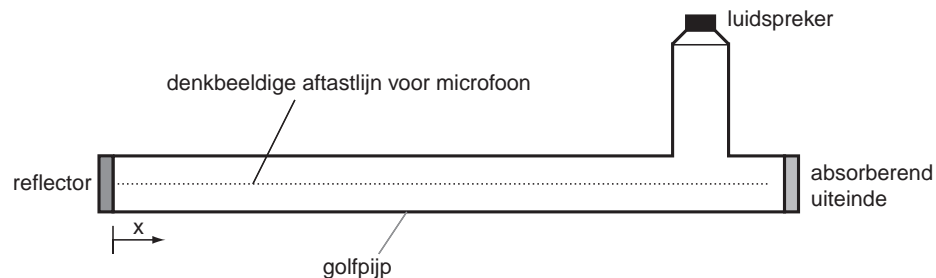


Figure B.4 De interferometer

Het hart van de interferometer is een golfpijp, een vierkante buis met een lengte van 2 meter en een doorsnede van 6×6 cm. De zijwanden zijn volledig hard, 100% reflecterend. Bij frequenties onder de 2800 Hz kunnen zich geen golven in de dwarsrichting voortplanten en kunnen er alleen vlakke golven lopen in de lengterichting van de pijp. Vanuit de zijpijp wordt met een luidspreker een geluidsgolf in de pijp opgewekt. Door het rechteruiteinde absorberend te maken, zal dat uiteinde niet reflecteren en wordt de golf daar gesmoord. Als linkeruiteinde wordt een te onderzoeken materiaal genomen, waarmee de pijp wordt afgesloten. De

reflectie-eigenschappen van dat materiaal bepalen het staande golfpatroon in de pijp. Met een microfoon wordt het interferentiepatroon in de pijp afgetast. Het verschil in intensiteit tussen de buiken en knopen van de staande golf bepaalt de absolute waarde van de reflectiecoëfficiënt R van de afsluiting. Hiervoor moeten de minimale en maximale waarde van de geluiddruk, $|P|_{min}$ respectievelijk $|P|_{max}$, gemeten worden. Allereerst wordt de staande golf verhouding S berekend met:

$$S = \frac{|P|_{max}}{|P|_{min}} = \frac{1 + |R|}{1 - |R|}, \quad (\text{B.4})$$

waarmee de modulus van de reflectiecoëfficiënt gegeven wordt door:

$$|R| = \frac{S - 1}{S + 1}. \quad (\text{B.5})$$

Uit de ligging van de buiken en de knopen kan de fasedraaiing van de afsluiting berekend worden. Als l_{min} de plaats is van het eerste minimum (een knoop dus), wordt de fasedraaiing ϕ gegeven door:

$$\phi = 2kl_{min} - \pi. \quad (\text{B.6})$$

Hierin is k het golfgetal van de gebruikte geluidsgolf. Deze kan bepaald worden uit de afstand tussen 2 minima, deze is namelijk gelijk aan een halve golflengte. Voor de berekening van de fout in $|R|$, moet eerst de fout in de staande golf verhouding berekend worden. De fout in het staande golf verhouding wordt gegeven door:

$$\frac{(\Delta S)^2}{S^2} = \frac{(\Delta P_{max})^2}{P_{max}^2} \frac{(\Delta P_{min})^2}{P_{min}^2}. \quad (\text{B.7})$$

Hiermee kan de fout $\Delta|R|$ in de absolute waarde van de reflectiecoëfficiënt berekend worden met:

$$\Delta|R| = \frac{2}{(1 + S)^2} \Delta S. \quad (\text{B.8})$$

De fout $\Delta\phi$ in de fasedraaiing wordt gegeven door:

$$(\Delta\phi)^2 = 4k^2(\Delta l_{min})^2 + 4l_{min}^2(\Delta k)^2. \quad (\text{B.9})$$

De opstelling met de interferometer

De geluiddruk in de interferometer wordt gemeten met een condensatormicrofoon. De meetopstelling voor de bepaling van de reflectiecoëfficiënt is afgebeeld in figuur B.5.

Een toongenerator levert het signaal waarmee de luidspreker van de interferometer wordt aangestuurd. Met behulp een elektretmicrofoon wordt de geluiddruk in

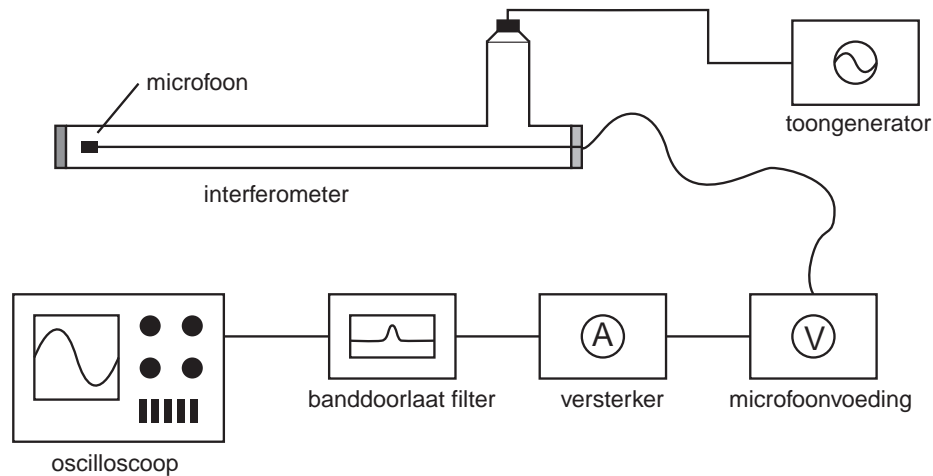


Figure B.5 Meetopstelling met de interferometer

de golfpijp gemeten. De uitgang van de microfoonvoeding wordt aangesloten op een versterker, die het signaal 40 dB versterkt. Het versterkte signaal wordt naar een banddoorlaatfilter geleid, dat is afgestemd op dezelfde frequentie als de toongenerator. Zonder banddoorlaatfilter bevat het signaal te veel verstoringen, vooral van lage frequenties (<200 Hz). Tenslotte wordt de uitgang van het bandfilter aangesloten op de oscilloscoop. Het signaal op de oscilloscoop is evenredig met de geluidsdruk ter plaatse van de microfoon. In de berekeningen wordt alleen de verhouding tussen de hoogste en laagste geluidsdruk gebruikt. De geluidsdruk wordt in Volt weergegeven, niet de officiële eenheid, maar de spanning is wel evenredig met de geluidsdruk.

Testen van de interferometer

Allereerst is er gemeten met open en hard afgesloten uiteinde, bij een frequentie van 1100 Hz. De golflengte was $\lambda = 31.80 \pm 0.05$ cm en dus is het golfgetal $k = 19.76 \pm 0.08$ m⁻¹.

Vervolgens is er gemeten met verschillende afsluitimpedanties. De resultaten hiervan zijn weergegeven in de onderstaande tabel. Gemeten worden de minimale geluidsdruk $|P|_{min}$, de maximale geluidsdruk en de plaats van het eerste minimum. De fout in $|P|_{max}$ is 0.01 V, de fout in λ_{min} is 0.1 cm.

	$ P _{max}$ (V)	$ P _{min}$ (mV)	$\Delta P _{min}$ (mV)	λ_{min} (cm)	$ R $	$\Delta R $	ϕ °	$\Delta\phi$ °
Hard uiteinde	0.33	3	0.5	8.04	0.98	0.003	0.04	0.4
Open uiteinde	0.53	80	3.0	16.66	0.74	0.02	130	0.4
Geïsoleerd uiteinde	0.53	125	5.0	16.16	0.62	0.02	118	0.4

Duidelijk is dat de interferometer goede resultaten geeft met een hard uiteinde. De reflectiecoëfficiënt hoort 1 te zijn, 0.98 is daar slechts een kleine afwijking van. De afwijking is groter dan de theoretisch berekende fout. Een verklaring hiervoor kan gevonden worden in het andere uiteinde. Indien dat niet volledig absorberend is, zal er een (kleine) extra golf lopen, die als duidelijkste gevolg heeft dat de minima niet nul zullen zijn. Indien de opening aan het uiteinde heel veel kleiner is dan de golflengte, is het open uiteinde een reflector met fasedraaiing van 180° . De berekende fout is zo klein, dat deze geen rol speelt bij de metingen. Voor de eenvoud zal de fout in de absolute waarde van R en in de fasedraaiing weggelaten worden.

B.3.4 Metingen met luidsprekers

Metingen met de Monacor luidspreker

Allereerst is er gemeten met een luidspreker van het type Monacor SP 45/4. De konus hiervan heeft een diameter van ongeveer 62 mm, dus sluit de vierkante pijp bijna volledig af. Er zit een relatief zware magneet op deze luidspreker, dit heeft als voordeel dat de luidspreker meer gevoelig is als microfoon. Om de gevoeligheid als luidspreker én microfoon te bepalen, zijn twee identieke luidsprekers op 5 cm afstand van elkaar geplaatst en is de overdracht van de ene naar de andere luidspreker bepaald, wederom met MLSSA. Het resultaat hiervan staat in figuur B.6. Deze overdracht is het product van het rendement van de Monacor als luidspreker en het rendement van de Monacor als microfoon. Het rendement is in het interessante gebied 40 tot -50 dB.

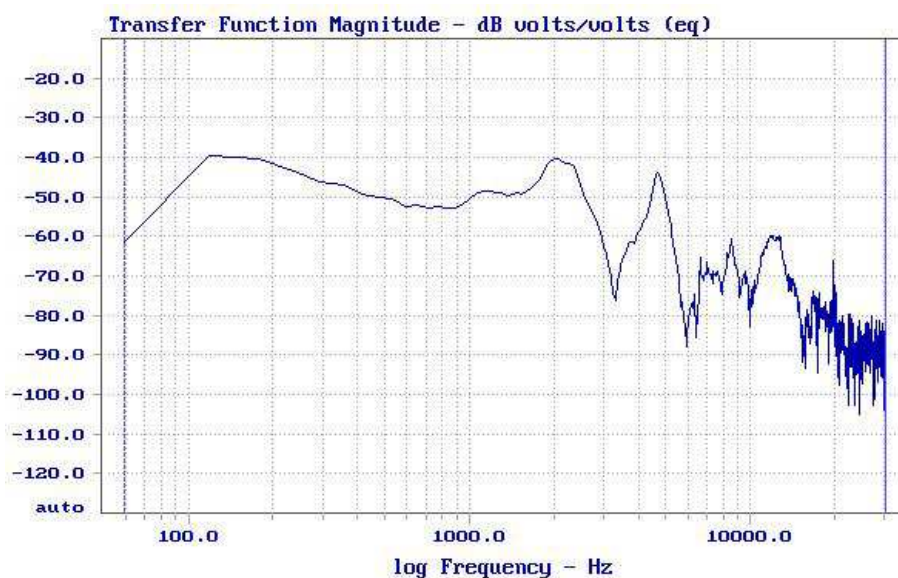


Figure B.6 De overdrachtsfunctie van Monacor- naar Monacorluidspreker

De Monacor is met een precies passend gemaakt plankje aan de interferometer bevestigd. De reflectiecoëfficiënt is bepaald met 3 verschillende afsluitingen: alleen plankje (dus geen luidspreker, maar wel een opening), luidspreker met open klemmen en tenslotte luidspreker met dichte klemmen. Volgens het model van Peter Swarte zou er verschil gemeten moeten worden tussen de luidspreker met open en met met gesloten klemmen. Met gesloten klemmen zal de spanning die door de beweging van de spreekspoel opgewekt worden resulteren in een klein stroompje. Hier zal de spreekspoel meer weerstand ondervinden tegen beweging en dus minder meebewegen met de invallende golf. Met gesloten klemmen moet dus een hogere reflectiecoëfficiënt gemeten worden, omdat de afsluiting harder is. De praktijk laat echter slechts een minimaal faseverschil zien zien tussen open en dichte klemmen:

	$ P _{max}$ (V)	$ P _{min}$ (mV)	λ_{min} (cm)	$ R $	ϕ °
Alleen plankje	0.42	50	12.06		93
Luidspreker met open klemmen	0.34	60	7.48	0.70	-11
Luidspreker met gesloten klemmen	0.34	60	7.56	0.70	-9

Aangezien de absolute waarde van de reflectiecoëfficiënt gelijk is voor open en gesloten klemmen en er slechts een minimaal faseverschil is, kan geconcludeerd worden dat het stroompje dat in de spreekspoel wordt opgewekt verwaarloosbaar klein is, want het beïnvloed de luidspreker niet.

Indien de konus passief vastgezet wordt daalt de reflectiecoëfficiënt iets: $R = 0.67$, $\phi = 6^\circ$. In de theorie van Peter Swarte wordt maximale reflectie verkregen door de konus actief stil te zetten. Maar actief kan nooit meer stil worden gezet dan passief. Een hogere reflectiecoëfficiënt dan 0.70 is dus niet realiseerbaar met de Monacorluidsprekers.

Het lage rendement was aanleiding om niet verder te meten met de Monacorluidspreker, maar op zoek te gaan naar een luidspreker met een hoger rendement. Voorlopig zijn alleen er twijfels geplaatst bij de grootte van het bereik waarover de reflectiecoëfficiënt ingesteld kan worden in het uiteindelijke systeem.

Metingen met de Philips luidsprekers

Peter Swarte heeft luidsprekers opgestuurd die volgens de specificaties ideaal zijn voor toepassing in het systeem met terugkoppeling. Deze luidsprekers zijn groter en passen daarom niet op het uiteinde van de interferometer. De luidspreker wordt gekoppeld aan het uiteinde van de golfpijp met een kartonnen trechter, zodat de binnenmaat van het inwendige van de interferometer geleidelijk toeneemt. Nadeel is wel dat er nu weinig zinnigs te zeggen is over de fase van de afsluiting, omdat de overgang naar een grotere diameter ook fasedraaiing geeft. Desondanks zal wel uit de metingen de fasedraaiing berekend worden waarbij verondersteld wordt dat

de afsluiting zich wel aan het uiteinde van de golfpijp bevindt. Hiermee wordt een indicatie van het verschil in fasedraaiingen tussen twee verschillende afluitingen op afstand verkregen. De fasedraaiing zal cursief weergegeven worden, om aan te geven dat deze geen absolute betekenis heeft. Allereerst is ook van de Philipsluidspreker de overdracht luidspreker-luidspreker B.7 bepaald:

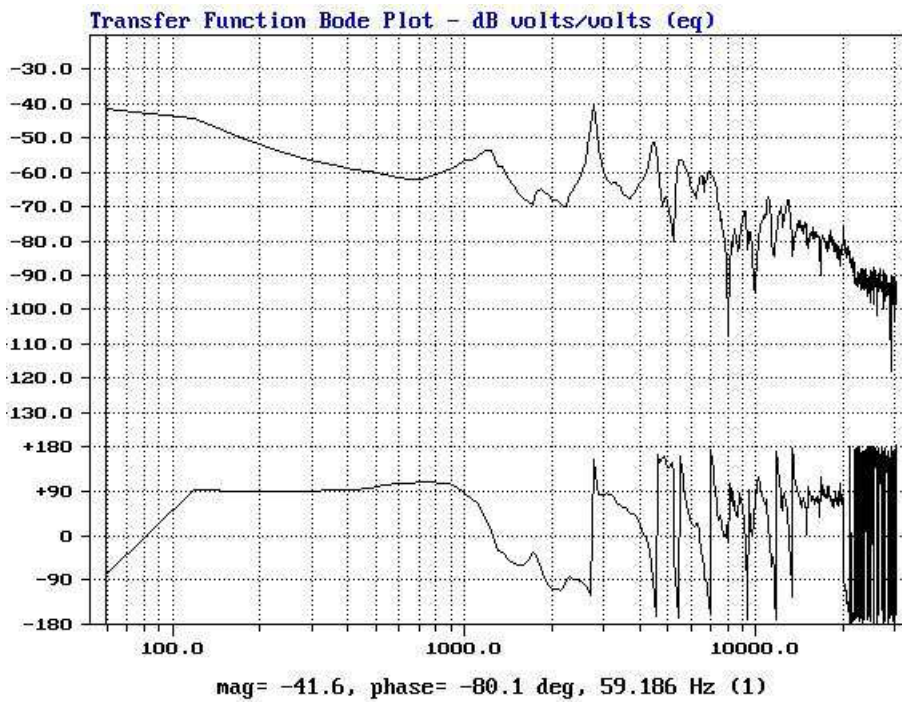


Figure B.7 Overdracht luidspreker-luidspreker van de Philipsluidspreker

Ook met deze luidsprekers is gekeken naar het verschil in reflectiecoëfficiënt met open en met dichte klemmen.

	$ P _{max}$ (V)	$ P _{min}$ (mV)	l_{min} (cm)	$ R $	ϕ °
Luidspreker met open klemmen	0.36	36	14.43	0.82	147
Luidspreker met gesloten klemmen	0.36	40	14.39	0.80	146

Ook hier een minimaal verschil tussen open en gesloten klemmen. Dit is niet onverwacht, want als het rendement zo laag is als duidelijk wordt uit figuur 9, dan is weinig invloed van de inductiespanning te verwachten.

Metingen met het terugkoppelnetwerk

Om meer inzicht te krijgen in eisen die aan de verschillende componenten in de schakeling van Peter Swarte gesteld worden, is de volgende meetopstelling

opgebouwd, zie figuur B.8

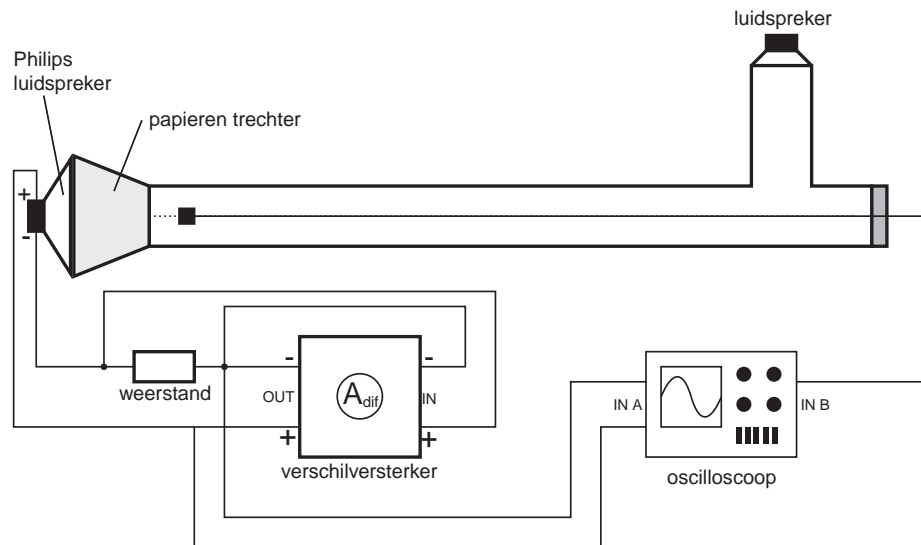


Figure B.8 Meetopstelling met eenvoudig terugkoppelnetwerk

De opstelling van de interferometer is net als in vorige metingen, voor de eenvoud zijn de toongenerator, versterker en bandfilter weggelaten. Extra is het terugkoppelnetwerk met de Philips luidspreker. In een gesloten kring staan de Philips luidspreker, een weerstand van 8.2 W en de uitgang van de verschilversterker. De spanning over de weerstand wordt teruggekoppeld via de verschilversterker. De verschilversterker heeft een ingebouwd bandfilter, dat op 1100 Herz is ingesteld. Door de ingangen van de verschilversterker om te wisselen, kan tussen tegen- en meekoppeling gekozen worden. De uitgang van de verschilversterker is ook op de eerste ingang (IN A) van de oscilloscoop aangesloten, om te kijken wat het signaal is bij verschillende versterkingsfactoren. Deze opstelling is identiek aan de opstelling in het idee van P.A. Swarte.

In experimenten is gekeken naar de invloed van de versterkingsfactor van de verschilversterker. De uitgangsspanning van de verschilversterker is haast onmeetbaar klein, totdat bij het geleidelijk opvoeren van de versterking opeens een sterke sinus zichtbaar is op de oscilloscoop. Er is dan ook een sterke fluittoon hoorbaar. Iets verkleinen van de versterkingsfactor heeft eerst heel weinig invloed, de sinus wordt iets minder sterk, tot opeens het uitgangssignaal van de verschilversterker weer ongeveer nul is. Het terugkoppelnetwerk doet dus weinig, totdat het ineens een heel sterk signaal levert en vertoont dan hysteresis. Boven het omslagpunt zorgt een iets hogere versterkingsfactor voor een steeds grotere sinus, totdat het signaal zo sterk wordt dat de beveiliging van de verschilversterker de versterker uitschakelt. Boven het omslagpunt is het systeem zeer instabiel. Er is niet eens een bron signaal in de interferometer nodig om het terugkoppelnetwerk een sinus

te laten produceren. De frequentie van de sinus uit de verschilversterker is ook onafhankelijk van de frequentie van de golf die door de luidspreker van de interferometer wordt opgewekt. Voor de volledigheid is de reflectiecoëfficiënt gemeten bij 4 verschillende waarden van de versterking: geen versterking, minimale versterking met fluittoon (dus eerst boven omslagpunt en dan net zover terugdraaien dat fluittoon net blijft), op het omslagpunt en maximale versterking (uiterste bereik van de verschilversterker). De resultaten hiervan staan in de volgende tabel:

	$ P _{max}$ (V)	$ P _{min}$ (mV)	l_{min} (cm)	$ R $	ϕ °
Geen versterking	0.4	42	14.39	0.81	146
Minimale versterking	3	37	12.78	0.98	109
op het omslagpunt	3	450	11.38	0.74	78
maximale versterking	16	1800	18.98	0.80	250

Aan deze metingen moet weinig betekenis toegekend worden. Inderdaad lijkt er zeer goede reflectie te zijn bij minimale versterking, maar zoals al eerder vermeld, is er geen relatie tussen het signaal uit de verschilversterker en het signaal waarmee de luidspreker van de interferometer aangestuurd wordt. Er is geen sprake van reflectie, maar van twee verschillende geluidsbronnen van ongeveer dezelfde frequentie en ongeveer dezelfde amplitude.

Deze metingen zijn gedaan met negatieve terugkoppeling. Positieve terugkoppeling gaf vergelijkbare resultaten, alleen bij lagere versterkingsfactoren (vanzelfsprekend).

Metingen met de fasedraaier: minimale en maximale reflectie

Om een indicatie te krijgen welke aanstuurspanningen nodig zijn voor minimale en maximale reflectie, is er een meetopstelling gebouwd met een interferometer en een fasedraaier, zie figuur B.9.

De uitgang van de toongenerator wordt aangesloten op de Philips luidspreker en op de ingang van een fasedraaier. Het uitgangssignaal van de fasedraaier heeft een instelbaar faseverschil met het ingangssignaal. Via een regelbare versterker wordt de uitgang van de fasedraaier aangesloten op de luidspreker van de interferometer. De microfoon van de interferometer is weer, zoals standaard, aangesloten op een oscilloscoop met tussenliggend de microfoonvoeding, een versterker en een banddoorlaatfilter.

Doel van deze opstelling is te bepalen wat de aanstuursignalen van de twee luidsprekers moeten zijn voor minimale en maximale reflectie. Strict genomen is het niet echte reflectie, want er wordt een apart bronsignaal toegevoegd met behulp van de Philipsluidspreker, maar de interferometer ziet geen verschil. Maximale reflectie vindt plaats indien de amplitude van de invallende golf gelijk is aan de golf

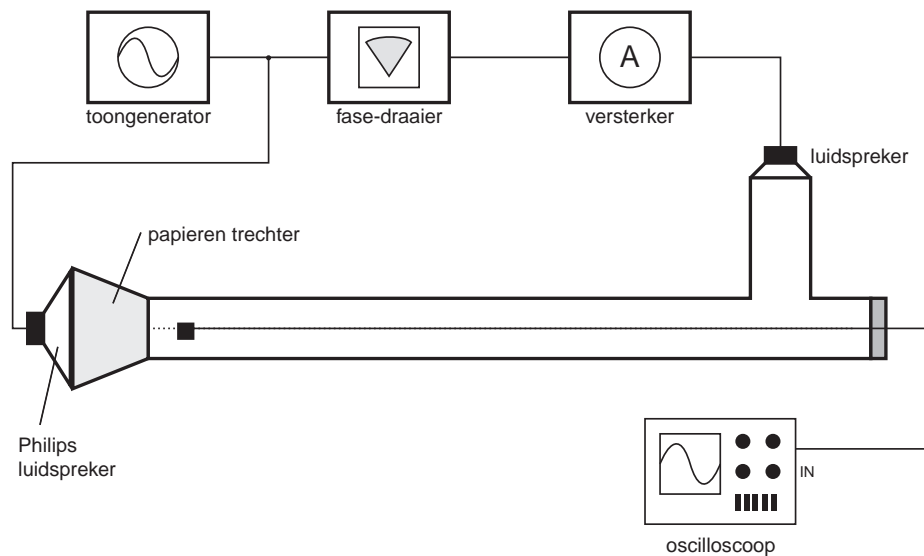


Figure B.9 Opstelling van de interferometer met fasedraaier.

die terugkomt van de Philips luidspreker. Deze teruggaande golf is opgebouwd uit reflectie van de invallende golf en de golf die actief wordt gegenereerd door de luidspreker. De gereflecteerde en gegenereerde golf moeten in fase zijn. De som van beide amplitudes moet gelijk zijn aan de amplitude van de invallende golf. Voor maximale absorptie geldt: geen teruggaande golf. Hiervoor moeten de gereflecteerde en gegenereerde golf elkaar opheffen: zelfde amplitude, maar in tegenfase. Metingen vonden plaats bij een frequentie van 999 Hz, dit was de maximale frequentie die de fasedraaier aankan.

Eerst zijn de fasedraaier en versterker zo ingesteld dat de geluidsdruk in de knopen zo laag mogelijk is, wat de toestand van maximale reflectie simuleert. De instellingen en meetresultaten daarvan staan in onderstaande tabel. Vervolgens is het verschil tussen de minima en maxima geminimaliseerd, wat de toestand van maximale absorptie simuleert.

	$V_{Philips}$ (mV)	V_{interf} (V)	$\phi_{interf} - \phi_{Philips}$ °	$ R $	ϕ °
Maximale reflectie	28	1	249	0.99	88
Maximale absorptie	72	1	75	0.01	-140

Dat minimale en maximale reflectie zo goed instelbaar zijn, is geen wonder. Absorptie komt neer op het opwekken van een anti-geluidsgolf, waarbij de frequentie van de golf al bekend is. In een eenvoudige omgeving als een golfpijp, is allang bewezen dat een golf uitdoven geen enkel probleem is. Bij maximale reflectie gaat het om de regelbaarheid van 2 luidsprekers. Die blijkt uitstekend te zijn. Het faseverschil tussen het signaal voor minimale en het signaal voor maximale reflectie

is 174° en dit komt heel aardig overeen met de theoretische waarde van 180° . De waarde van de schijnbare reflectiecoëfficiënt is dus goed regelbaar, maar dit zegt weinig over de haalbaarheid van de ideeën van Peter Swarte.

Het is heel interessant om te meten wat de uitgangsspanning is van de open klemmen van de Philipsluidspreker indien de luidspreker van de interferometer met 1 Volt wordt aangestuurd. Deze spanning is 0.25 mV. Deze waarde moet vergeleken worden met de aanstuurspanning van de Philips luidspreker ($V_{Philips}$) in de bovenstaande tabel. Dit laat zien dat er een versterking met ongeveer factor 100 nodig is van het bronsignaal van de Philips luidspreker om een aanstuurspanning te hebben die de reflectiecoëfficiënt beïnvloedt. (Die factor 100 is ongeveer gelijk aan -40 dB gevoeligheid in de overdracht luidspreker-luidspreker in figuur B.7). Met formule B.3 valt dan in te zien dat als de waarde van i_{tot}/u_1 ongeveer 100 is, de noemer ($-Z_{tot} + AZ_1$) gelijk is aan 0.01. Een verhoging van A (en dus van AZ_1) met 1% leidt dan al tot instabiliteit (noemer van de breuk gaat dan naar nul toe).

De conclusie is dat het idee van Peter Swarte met de gebruikte luidsprekers niet uit te voeren is: de gevoeligheid is zo laag, dat het systeem instabiel is voordat het interessant wordt.

B.3.5 Samenvatting en conclusie

Bij het bestuderen van de theorie van Peter Swarte zijn geen fouten geconstateerd. Bij de regeling moet gezorgd worden om voldoende uit de buurt te blijven van de asymptoten, want bij overschrijden daarvan is het systeem instabiel.

Allereerst is er een opstelling gemaakt met een interferometer. Een interferometer is een golfpijp waarin een staande golf wordt opgewekt waarvan de buiken en knopen bepaald kunnen worden. Het ene uiteinde van de golfpijp is absorberend, voor het andere uiteinde kan vrij een afsluiter gekozen worden. Dit kan zowel een passieve afsluiter zijn (een metalen plaat) of een actieve afsluiter (een luidspreker). De reflectie-eigenschappen van deze afsluiter bepalen de plaats en intensiteit van de buiken en knopen in de pijp. Omgekeerd geldt natuurlijk ook: uit de plaats van de buiken en knopen en hun intensiteit kunnen de reflectie-eigenschappen van de afsluiter berekend worden. Het testen van de interferometer liet zien deze goed functioneert, vooral bij harde afsluitingen.

Het model van Peter Swarte werkt in de praktijk niet goed. De eenvoudigste meting die de theorie kan ondersteunen of weerleggen is de bepaling van de reflectiecoëfficiënt van de luidspreker met open en met gesloten klemmen. Volgens de theorie zou de luidspreker met gesloten klemmen een hogere reflectiecoëfficiënt moeten hebben. In de praktijk is weinig verschil te meten, eerder een neiging naar een lagere reflectiecoëfficiënt bij gesloten klemmen. Vervolgens is er wel gemeten met versterking in de terugkoppellus, maar het enige resultaat was dat het systeem

of niets deed, of instabiel was. Het systeem is generlei wijze instelbaar.

De verklaring wordt gezocht in het intensiteitsverschil tussen de uitgangsspanning van de luidspreker als microfoon en de aanstuurspanning die nodig is voor de luidspreker als geluidsbron. Uit metingen is gebleken dat hier een factor 100 tussen zit. Aangezien de versterking van de terugkoppeling van de vorm is van:

$$\text{versterking} = \frac{1}{1 + A\beta}, \quad (\text{B.10})$$

met A de versterkingsfactor, de terugkoppelfactor en het product $A\beta$ de lusversterking. De versterking moet factor 100 zijn, daarvoor moet $A\beta = 0.99$ zijn (dus positieve terugkoppeling) en een kleine fluctuatie van 1% van A zorgt al dat de noemer naar nul gaat en dus instabiliteit van het syteem.

Oftewel: de theorie is prachtig, maar in de praktijk werkt het helemaal niet.

B.3.6 Artikelen

- Title:** Active Resonator for Noise Control
Authors: Dr. R. Maier, M.J. Pucher, Dr O. Bschorr
Source: ISMA 21. International Conference on Noise & Vibration Engineering. Leuven, Belgium, Sept. 18-20, 1996. Proceedings: p. 353-364
Institute: Daimler Benz AG, Research and development, Munich, Germany

De Active Resonator (AR) is luidspreker uitgerust met een terugkoppelsysteem. Met een sensor worden de konusbewegingen gemeten en het gemeten signaal wordt na bewerking op de klemmen van de luidspreker gezet. In vergelijking met passieve absorbeerdere heeft dit systeem als voordeel dat de resonantiefrequentie, de bandbreedte en de impedantie instelbaar zijn.

Het principe

Een luidspreker kan gemodelleerd worden als een lineair systeem met twee ingangen en een uitgang. Ingangsignalen zijn de geluidsdruk p_{em} op de konus en het teruggekoppelde signaal uit de controller, uitgangssignaal is de membraam snelheid v . De netto geluidsdruk p_{net} is de superpositie van de geluidsdruk p van een geluidsgolf en de geluidsdruk p_L opgewekt door de luidspreker zelf.

Het akoestische gedeelte van het model is alleen geldig indien de golflengte veel groter is dan de afmetingen van de konus en indien de propagatie van de externe geluidsgolf niet verstoord wordt door de AR. De impedantie Z_A van de AR is de verhouding tussen de netto geluidsdruk p_{net} en de konussnelheid v . Uit het blokschema volgt:

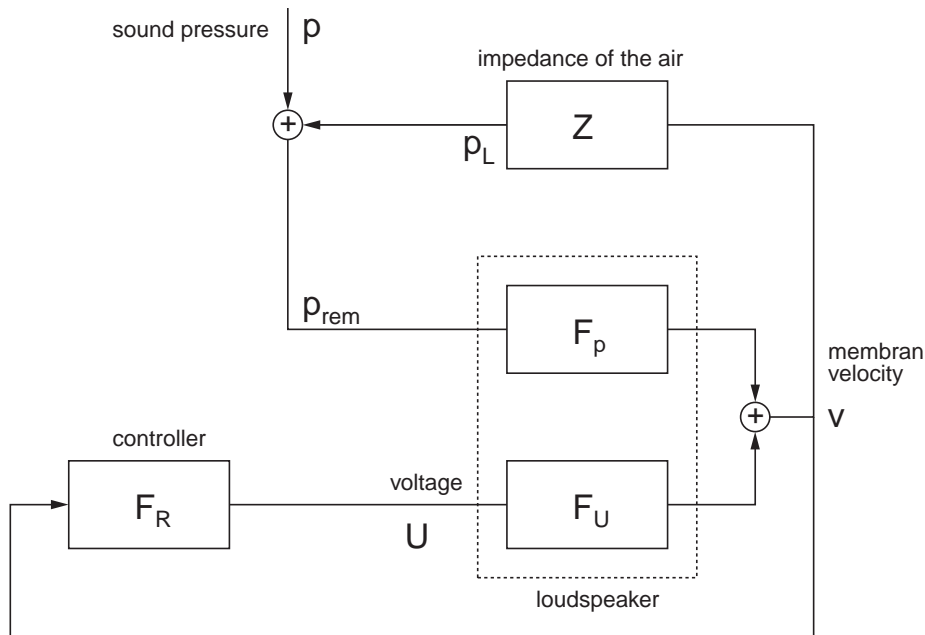


Figure B.10 Het blokdiagram van de Active Resonator.

$$Z_A = \frac{p_{rem}}{v} = \frac{1 - F_U F_R}{F_p}. \quad (\text{B.11})$$

De geluidsdrukreductie p_{rem}/p (transmissieverlies) kan worden geschreven als:

$$\frac{p_{rem}}{p} = \frac{1 - F_U F_R}{1 + F_p Z - F_U F_R} = \frac{Z_A}{Z + Z_A}. \quad (\text{B.12})$$

Vergelijking B.12 laat zien dat de impedantie Z_A van de actieve resonator kleiner moet zijn dan de impedantie Z van de lucht om een significante geluidsdrukreductie te bereiken. De optimale overdrachtsfunctie van de controller F_R voor het bereiken van de gewenste impedantie Z_A^* wordt gegeven door:

$$F_R^* = \frac{1}{F_U} (1 - F_p \cdot Z_A^*). \quad (\text{B.13})$$

Helaas levert dit een instabiel en niet-causaal systeem op. Het gedrag van de luidspreker is gemodelleerd. Het theoretische model komt zeer goed overeen met de metingen in de praktijk, pas boven de 8000 Herz is er duidelijke verschil te zien tussen de gemeten membraangedrag en het voorspelde gedrag. Om instabiliteit te voorkomen, is een bandfilter nodig. Het resultaat hiervan is vanzelfsprekend een bandbreedte waarin de speaker geluid absorbeert, net als geldt voor helmholzresonatoren. Het verschil ligt hierin dat met een AR de resonantiefrequentie vrij gekozen kan worden. Met behulp van het model van de luidspreker

en het gewenste gedrag kan er een zo goed mogelijke benadering gemaakt worden van ideale controllerfunctie. Stabiliteit wordt alleen bereikt door alleen in een beperkt frequentiegebied de ideale functie te benaderen.

Een voordeel van de AR is dat de akoestische terugkoppeling geen invloed heeft op de stabiliteit: indien het systeem stabiel is zonder akoestische terugkoppeling via de impedantie Z , is het ook daarmee stabiel.

Vervolgens is er gemeten aan de absorptie in de volgende opstelling

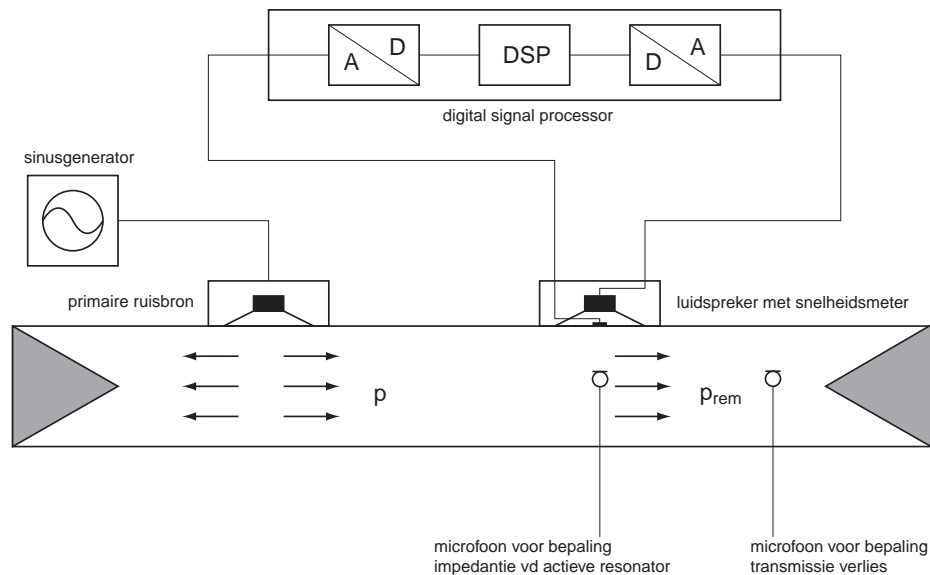


Figure B.11 De meetopstelling met de Active Resonator.

De frequentie van de AR is op 90 Hz ingesteld. De sinusgenerator heeft frequency sweeps gemaakt tussen 80 en 300 Hz. Binnen een bandbreedte van 30 Hz is de geluidsreductie meer dan 6 dB, de maximale reductie is 25 dB. De actieve resonator heeft een veel lagere impedantie dan een luidspreker zonder netwerk, rond de ingestelde frequentie is dit iets meer dan een factor 100.

B.4 Elektrodynamische luidsprekers

Er is aandachtig gekeken naar het model van een elektro-dynamische luidspreker, met als doel de meetresultaten te kunnen verklaren. Hiervoor is het nodig ook te kijken naar het werkingsprincipe van dit type luidspreker. De basis van een luidspreker is de spreekspoel geplaatst in het magnetische veld van de luidsprekermagneet. In onderstaande figuur is dat vereenvoudigd getekend. De draad van de spreekspoel begint bij a, gaat naar de spleet in de magneet (spleet is in werkelijkheid veel nauwer dan hier getekend), draait daar twee wikkelingen rechtsom (van bovenaf gezien) en gaat dan weer de magneet uit bij b.

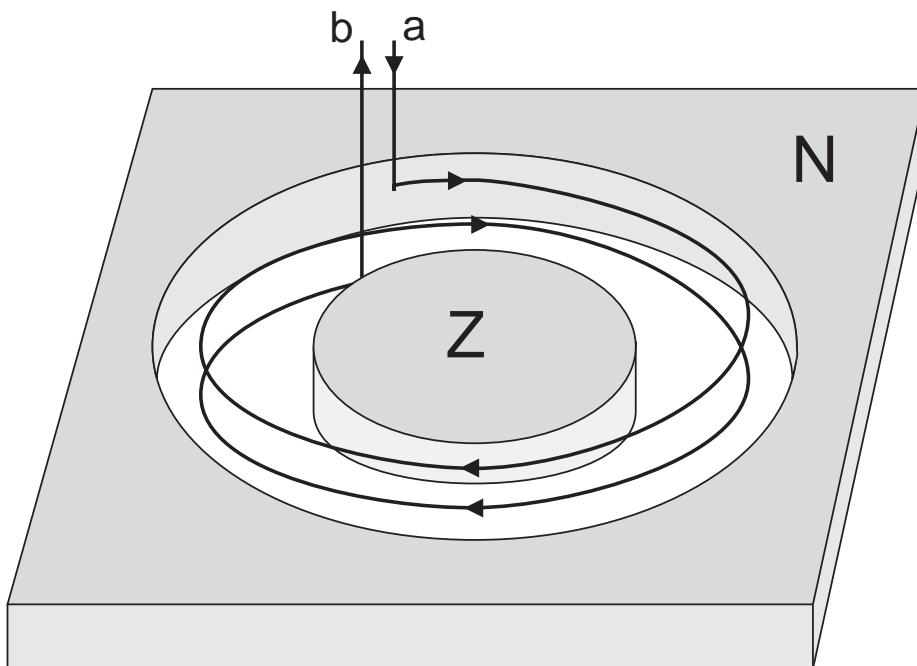


Figure B.12 De spreekspoel in het magnetische veld

Aan de buitenkant van de spreekspoel bevindt zich de noordpool van de magneet, aan de binnenkant de zuidpool. De veldlijnen van het magnetisch veld lopen van noord naar zuid, dus van buiten naar binnen. In de smalle spleet is de sterkte van het magneetveld homogeen. De drijvende kracht voor een luidspreker is de Lorentzkracht \vec{F}_L , die gegeven wordt door:

$$\vec{F}_L = q \vec{v} \times \vec{B}, \quad (\text{B.14})$$

waarin q de lading is van een deeltje, \vec{v} de snelheid van dat deeltje en \vec{B} het magnetische veld. Indien er gekeken wordt naar een lijnelementje $d\vec{l}$ waarvan de richting gelijk is aan de richting van de stroom I , dan is de Lorentzkracht $d\vec{F}_L$ die een stroomelementje in een magnetisch veld \vec{B} ondervindt gelijk aan:

$$\overrightarrow{dF_L} = I \overrightarrow{dl} \times \overrightarrow{B}, \quad (\text{B.15})$$

B.4.1 Werking als luidspreker

Voor werking als luidspreker kan gebruik gemaakt worden van formule B.14. We beschouwen een klein lijnelementje \overrightarrow{dl} , als in onderstaande figuur. In deze figuur is een bovenaanzicht getekend van een stukje van de spreekspoel uit figuur B.12.

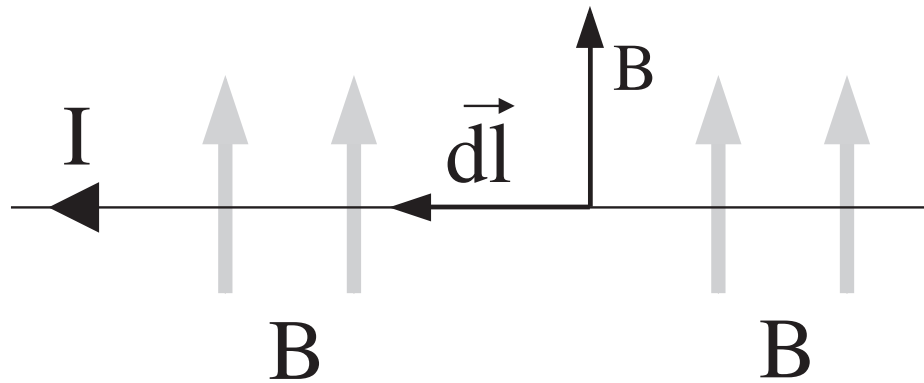


Figure B.13 Lijnelementje dl van de spreekspoel.

Indien de stroom loopt zoals aangegeven, levert toepassing van formule B.15 op dat de Lorenzkracht $\overrightarrow{dF_L}$ die werkt op het lijnelementje naar achteren wijst in het vlak van de tekening. Doordat \overrightarrow{B} overal loodrecht op dl staat in het horizontale vlak, geldt voor alle lijnelementjes dat de Lorenzkracht naar beneden gericht is. Voor de spreekspoel in figuur B.12 betekent dit dat een stroom van a naar b (dus volgens de pijltjes), zorgt voor een neerwaarts gerichte kracht. Deze zal dus voor een neerwaartse beweging zorgen. Integratie van formule B.15 levert op voor de totale neerwaartse kracht:

$$\overrightarrow{F_{L,tot}} = \int_a^b I \overrightarrow{dl} \times \overrightarrow{B} = BIl \quad (\text{B.16})$$

waarin l de totale lengte is van de draad van de spreekspoel in het magnetische veld

B.4.2 Werking als microfoon

Onder invloed van een invallend geluidsveld, beweegt de konus van een luidspreker en dus ook de spreekspoel. Stel nu dat de spreekspoel daardoor naar

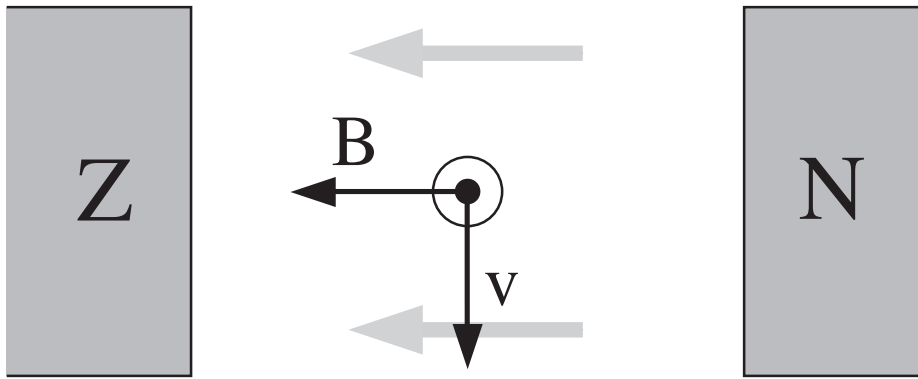


Figure B.14 Dwarsdoorsnede van een bewegende draad in het magneetveld.

beneden beweegt. In figuur B.14 is van die situatie een dwarsdoorsnede getekend.

De elektronen in de draad hebben ook een verticale snelheid \vec{v} . De Lorentzkracht op een elektron wordt gegeven door formule B.14. Het uitproduct van \vec{v} en \vec{B} wijst achteren. De Lorentzkracht daarentegen wijst naar voren (uit het vlak van tekening), omdat q negatief is. Indien dit meegenomen wordt naar figuur B.12, dan volgt dat een neerwaartse beweging van de spreekspoel de elektronen versnelt in de richting van de pijltjes in figuur B.12. Stroom loopt dan juist van b naar a . Een Lorentzkracht \vec{F}_L betekent voor de lading q een elektrisch veld $\vec{E} = \vec{F}_L/q$. Richting hiervan is tegengesteld aan de richting van de pijlen in figuur B.12. De totale spanning U_{ab} kan gevonden worden door het elektrische veld te integreren over de lengte l van de spreekspoel in het magnetische veld:

$$U_{ab} = \int_a^b \vec{E} \cdot d\vec{l} = \int_a^b \frac{\vec{F}_L}{q} \cdot d\vec{l} = \int_a^b (\vec{v} \times \vec{B}) \cdot d\vec{l} = -vBl \quad (\text{B.17})$$

Hierin is v naar beneden gericht in figuur B.12. Het minteken is het gevolg van de tegengestelde richting van \vec{E} en $d\vec{l}$.

B.4.3 Conclusie

Een elektrodynamische luidspreker is anti-reciprook. Voor toepassing in het systeem van Peter Swarte heeft dit grote consequenties. Als een invallend geluidsveld zorgt dat de konus naar beneden beweegt, geeft dit een negatieve spanning over de klemmen van de luidspreker. Om deze konus vervolgens mee te laten bewegen, is een positieve spanning over de klemmen nodig. De versterking i_{tot}/u_1 uit het model van Peter Swarte zou dus positief moeten zijn! Maar dat zou ook tot gevolg hebben dat er een positieve spanning over de klemmen komt te staan. De spanning ten gevolge van de beweging van de spreekspoel wordt gecompenseerd

en wel zo sterk, dat er een tegengestelde spanning komt te staan. De bronspanning (van de luidspreker als microfoon werkend) wordt overgecompenseerd. Dit kan niet werken.

Conclusie: anti-reciproke transducers zijn niet geschikt voor het systeem van Peter Swarte.

Daarnaast is het zo dat stroom door de spreekspoel aanleiding geeft tot een kracht, terwijl beweging van de spreekspoel aanleiding geeft tot een spanning. Geeft dit aanleiding tot een belangrijke fase draaiing? Beïnvloedt dit de stabiliteit?

B.4.4 Gevolgen voor het model van een elektrodynamische luidspreker

Het model van een elektrodynamische luidspreker, zoals staat in het dictaat Geluidsbeheersing / Sound Control [3], is dus niet correct!! Het gebruik van een transformator in dat model is niet correct, omdat een transformator reciprook is en een spreekspoel niet. Normaal gesproken merk je hier weinig van, als een transformatorrichting duidelijk de andere overheerst (dus maar bronnen slechts aan een kant) en verliezen in het systeem hoog genoeg zijn om te signalen te dempen als ze twee maal 'door de transformator' heengaan. Maar het is niet correct!