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List of papers

This thesis is based on the following papers, which are referred to in the text by their Roman numerals.

I **Annea Barkefors**, Simon Berthilsson and Mikael Sternad, "Extending the area silenced by active noise control using multiple loudspeakers," Presented at the *2012 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, Kyoto, Japan, April 2012

II Simon Berthilsson, **Annea Barkefors** and Mikael Sternad, "MIMO design of active noise controllers for car interiors: extending the silenced region at higher frequencies," Presented at the *American Control Conference (ACC)*, Montreal, Canada, June 2012

III **Annea Barkefors**, Simon Berthilsson and Mikael Sternad, "An investigation of a theoretical tool for predicting performance of an active noise control system," Presented at the *2012 International Congress on Sound and Vibration (ICSV)*, Vilnius, Lithuania, July 2012

IV **Annea Barkefors**, Mikael Sternad and Lars-Johan Brännmark, "Design and Analysis of Linear Quadratic Guassian Feedforward Controllers for Active Noise Control", submitted

V **Annea Barkefors**, "Adapting an MSE controller for Active Noise Control to nonstatic noise statistics", Technical Report

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List of work not included in this thesis

VI Tomas Olofsson, Martin Hansen Skjelvareid and **Annea Barkefors**, "Ultrasonic imaging of immersed objects using migration techniques," Presented at *2010 8th European Conference on Synthetic Aperture Radar*, Aachen, Germany, June 2010

VII Simon Berthilsson, **Annea Barkefors**, Lars-Johan Brännmark and Mikael Sternad, "Acoustical zone reproduction for car interiors using a MIMO MSE framework," Presented at the *2012 Audio Engineering Society 48th Conference on Automotive Audio*, Munich, Germany, September 2012

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I cannot stress enough how much my fellow PhD students mean to me. Simon, you stand out as my partner in crime (or, well, ANC) but Adrian, Mathias, Magnus, Rikke, Marcus and Fredrik are no less important to me. Thank you all for all the good discussions in the fika-room, all volleyball and ping-pong, and everything else. I know more will come. I also want to send out an equally big thank you to everyone else, both in the Signals and Systems group, and in CORE.

Finally I want to extend thanks to my family. Especially Irmeli, thank you for everything. You are a great support.

Cristobal, I love you, and I am very grateful for everything you do for me. Without your help I would not be finished by now. I hope that you know how much you mean to me. Last of all, thank you Simian Emilio for joining us. It might not have made my studies any easier but I love you deeply.

Annea Barkefors

Uppsala 2014-04-25

List of Symbols

$\|\| \cdot \|\|$	The L_2-norm.
∇^2	The Laplace operator, expressed in Cartesian coordinates as: $\nabla^2 = \frac{\partial^2}{\partial x_1^2} + \frac{\partial^2}{\partial x_2^2} + \frac{\partial^2}{\partial x_3^2}$.
$\frac{\partial^2}{\partial t^2}$	The second order partial differential operator.
c_0	The speed of sound.
$p(\boldsymbol{x}, t)$	Pressure of sound wave at spatial position $\boldsymbol{x}$ and time t.
$\boldsymbol{x}$	Vector of spatial coordinates.
$E\{\cdot\}$	Expectation with respect to stochastic signals.
J	Criterion function to be minimized.
$\boldsymbol{\varepsilon}(t)$	Vector with measurements of the total noise (remaining error) in the ROI.
$\boldsymbol{n}(t)$	Real-valued vector of feedforward primary noise signals.
$\boldsymbol{y}(t)$	Vector representing the contribution to the sound field in the ROI by the control signals.
$\boldsymbol{z}(t)$	Real-valued vector of primary noise signals in the ROI.
$\boldsymbol{\mathcal{R}}(q^{-1})$	Matrix of IIR filters describing the controller.
$\boldsymbol{D}(q^{-1})$	Polynomial matrix of FIR transfer functions describing the primary noise path from feedforward measurements to the ROI.
$\boldsymbol{B}(q^{-1})$	Polynomial matrix of FIR transfer functions describing the control path from the control loudspeakers to the ROI.
q^{-1}	Discrete-time shift operator, $q^{-1}\boldsymbol{y}(t) = \boldsymbol{y}(t-1)$.
t	Discrete time index

List of Abbreviations

ANC	Active Noise Control
dB	Decibels
GMV	General Minimal Variance
IMC	Internal Model Control
LMS	Least Mean Square
LQG	Linear Quadratic Gaussian
MIMO	Multiple Input Multiple Output
MSE	Mean Square Error
ROI	Region Of Interest
rpm	Revolutions Per Minute
SISO	Single Input Single Output

1. Introduction

Low frequency noise in car compartments is a problem that is easily recognized by anyone having had a ride in a poorly insulated car. Engine boom causes noise with spectra related to the rpm and the load on the engine, and road and wind noise adds a noise floor that, for high speeds, can make the journey quite uncomfortable. Spending a long time in such noisy surroundings is very tiring, why there is a high incentive to reduce the noise. To give driver and passengers a comfortable driving experience, the sound environment in a car cabin is carefully designed, especially in premium cars.

There are many approaches to reduce undesired noise within the cabin of a car. The engine mount is designed to reduce vibrations, as are the wheel suspensions. Passive means of damping, such as insulation, are used efficiently in many cars today. However, using insulating materials to reduce low frequency noise results in high material costs and heavy vehicles, with high fuel consumption. The possibility to use active control methods is therefore of great interest.

This licentiate thesis is a summary over my work since I started as an industrial PhD student in the Signals and Systems group at Uppsala University. The subject of my PhD is Electrical Engineering with speciality in Automatic Control. My project is to develop new and improved methods for Active Noise Control (ANC), with special focus on applications within the automotive industry. My industrial partner is Dirac Research AB, a research oriented company with quality-improving products for audio reproduction systems.

My work so far has had a heavy experimental focus. This is due to the limited validity of results that would be obtained from simulations studies that are based on assumptions of the system that might not even be close to reality, such as free-field propagation of the sound fields. Instead, all the simulations I have made have been based on real measured room impulse responses. Verification measurements and live attenuations of both broadband and narrowband signals have also been studied. Of this I am very proud, since it is so tempting to stay in the beautiful Matlab world were everything behaves the way you expect it to. Well, most of the time anyway.

The thesis is based on five papers. In this comprehensive summary, I will give an introduction to the area of research that is Active Noise Control.

1.1 Thesis Overview

The thesis is organized as follows. A basic introduction to active noise control is given in Chapter 2, along with some of the results from the papers. In Chapter 3 an overview of the research area is presented to give a context of what other researchers are and have historically been contributing to the field. The chapter is concluded with a section describing my contributions. Chapter 4 finally contains a summary of each of the papers that the thesis is based on.

2. Active Noise Control

2.1 The Basic Principle

Sound in gases, such as air, travel as waves of pressure changes propagating through the medium. The propagating sound wave will satisfy the wave equation

$$\nabla^2 p(\boldsymbol{x},t) - \frac{1}{c_0^2}\frac{\partial^2}{\partial t^2} p(\boldsymbol{x},t) = 0 \ , \tag{2.1}$$

where $p(\boldsymbol{x},t)$ is the pressure at spatial position $\boldsymbol{x}$ and time t, and c_0 is the speed of sound, which depends on the medium. The Laplace operator ∇^2 and the partial differential operator $\partial^2/\partial t^2$ are both linear and furthermore the pressure $p(\boldsymbol{x},t)$ appears in the equation linearly. Therefore, two solutions $p_1(\boldsymbol{x},t)$ and $p_2(\boldsymbol{x},t)$ that both satisfy the wave equation can be added to give a new solution $p_1(\boldsymbol{x},t) + p_2(\boldsymbol{x},t)$ that also satisfies the wave equation. This is called the *principle of superposition* in a linear system and is the basic principle that makes Active Noise Control (ANC) possible.

The linearity of the wave equation (2.1) depends on the changes in pressure being small in comparison to the nominal pressure. If too big, the pressure changes will cause changes in the speed of sound. With increasing pressure, the speed of sound will increase, to decrease again with decreasing pressure. The wave equation thus becomes nonlinear and the principle of superposition no longer holds. This phenomenon will become noticeable for sound pressures around and above 140 dB.

In an ANC application, the principle of superposition is used to attenuate an undesired sound wave. If the primary sound pressure $p_1(\boldsymbol{x},t)$ can be recreated with opposite sign throughout the volume to be controlled,

$$p_2(\boldsymbol{x},t) = -p_1(\boldsymbol{x},t) \ , \tag{2.2}$$

the resulting sound pressure becomes

$$p(\boldsymbol{x},t) = p_1(\boldsymbol{x},t) + p_2(\boldsymbol{x},t) = 0 \ . \tag{2.3}$$

The undesired sound is thus eliminated, as in the illustration in Figure 2.1. Although an easy concept to grasp, the recreation of a primary sound pressure throughout a volume is a difficult task.

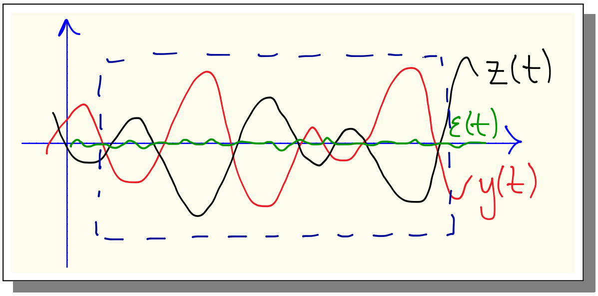

Figure 2.1. Destructive interference of two one-dimensional sound waves. The red line represent the primary noise $z(t)$, the black line the control signal $y(t)$ and the green line is the resulting error $\varepsilon(t)$.

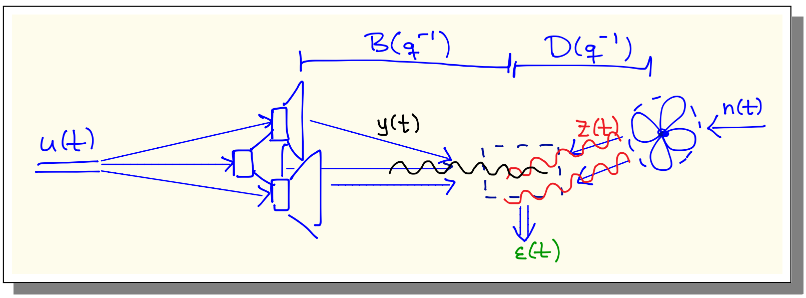

Figure 2.2. The acoustic system of primary and control sound field.

2.2 The Control Problem

In ANC, one or several loudspeakers (variously called secondary or control loudspeakers) are used to create a sound field that is equal in amplitude but opposite in phase to the noise to be controlled, called the primary sound field or primary noise. Except for some special cases, global noise control, where the noise in an entire room is cancelled, is not feasible, due to the high complexity of sound fields. Most of the time there is a trade-off between size of the controlled volume, for how high frequencies uniform damping can be achieved and the level of the attenuation. Therefore, the volume targeted for ANC needs to be defined. In the following, it will be called the Region Of Interest for control (ROI). In the experiments in the papers on which this thesis is based, the ROI is always centered around the head of a person, to create a *zone of silence* for that person.

The primary sound field, sampled in discrete time, can be described at M measurement positions in the ROI by

$$\boldsymbol{z}(t) = \boldsymbol{D}(q^{-1})\boldsymbol{n}(t) \ . \tag{2.4}$$

Here, $\boldsymbol{D}(q^{-1})$ is an $M \times L$ polynomial matrix in the discrete-time backward shift operator, containing transfer functions from L primary noise sources described by $\boldsymbol{n}(t)$ to the M measurement positions, as illustrated in Figure 2.2. The objective of ANC is to create a vector $\boldsymbol{u}(t)$ of control signals to N control loudspeakers so that the contribution to the sound field in the ROI becomes

$$\boldsymbol{y}(t) = \boldsymbol{B}(q^{-1})\boldsymbol{u}(t) \ , \tag{2.5}$$

where the $M \times N$ polynomial matrix $\boldsymbol{B}(q^{-1})$ describes the transfer functions from the control loudspeakers to the measurement positions. The sound field created by the control loudspeakers will be added to the primary sound field and in case of destructive interference the primary sound field will be attenuated. This is illustrated for one-dimensional sound waves in Figure 2.1.

Neglecting any noise other than the primary noise and the control noise, the remaining error in the measurement positions is simply the addition of the two sound fields

$$\boldsymbol{\varepsilon}(t) = \boldsymbol{z}(t) + \boldsymbol{y}(t) \ . \tag{2.6}$$

In order to be able to reproduce the primary sound field with a reasonable number of control loudspeakers, it has to be reasonably smooth. The higher the frequency, the more chaotic will the sound field be. At lower frequencies, with wavelengths longer than the dimensions of the room in which the sound field resides, the sound field can be shown to be dominated by a few so called acoustic modes [1]. An acoustic mode is a standing wave pattern in the room and the sound field is made up of the sum of the acoustic modes for each frequency. The number of acoustic modes grows rapidly with frequency, why the complexity of the sound field also grows rapidly.

For this reason, ANC works best for low frequency noise; a lucky coincidence considering that passive means of damping, such as insulation, are more efficient at higher frequencies.

To be able to actively control noise, a minimization criterion is needed. In early theoretical formulations of ANC, it was suggested in the case with an enclosed sound field[1] to minimize the total acoustic potential energy in the region of interest (ROI) for control [2]. This is approximated by sums of squares of error signals $\boldsymbol{\varepsilon}(t)$ from within the ROI, leading to Mean Square Error (MSE) criterions, the most basic of which is

$$J = E\{||\boldsymbol{\varepsilon}(t)||^2\} \ , \tag{2.7}$$

where $E\{\cdot\}$ denotes expectation value.

[1] An enclosed sound field is a sound field which is confined in some form of enclosure, such as a room. The sound field will then be built up of two parts: the direct sound wave from the source and reverberant sound due to reflections and refractions in the room.

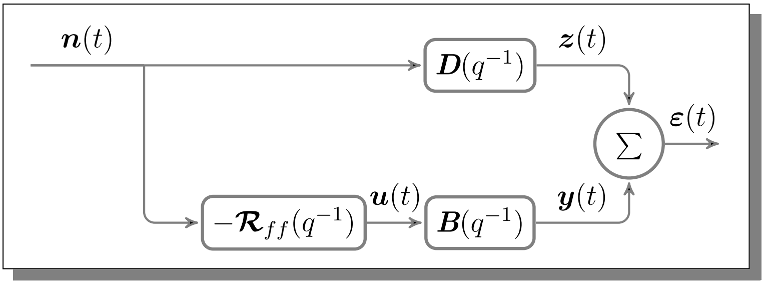

Figure 2.3. Block diagram describing a feedforward control system.

There are two main classes of controllers that can be designed to minimize a criterion such as (2.7). The first class consists of feedforward controllers and the second of feedback controllers.

2.3 Feedforward Control

Figure 2.3 shows a block diagram of a feedforward control system. L feedforward signals, $\boldsymbol{n}(t)$, are somehow picked up in advance, and fed to the controller $-\boldsymbol{\mathcal{R}}_{ff}(q^{-1})$, which represents an $L \times N$ matrix of rational transfer functions. The purpose of the controller then becomes to estimate how the noise will propagate to the ROI and how the primary sound field will behave within the ROI. From this estimate, N control signals $\boldsymbol{u}(t)$ are fed to the control loudspeakers in such a way that the sound waves reach the ROI out of phase from the primary sound field.

One main advantage of a feedforward control system is that the primary noise is picked up in advance. There are inherent time delays in the control system due to the time it takes the control sound to reach the ROI after sending it through the control loudspeakers. Anti-aliasing filters, buffers in sound cards and reconstruction filters etc further add to the delays. Every sample period that can be gained by picking up the feedforward signal further away from the ROI adds valuable computation time margin in the controller. Placing the control loudspeakers as close as possible to the ROI is another way of gaining computation time.

For a feedforward system, the primary path $\boldsymbol{D}(q^{-1})$ needs to be identified. If the primary path is known to be stationary, this can be done offline before calculating the controller. On the other hand, if the statistics of the primary path changes during control, it needs to be monitored and the models updated. As the models change, the controller will need updating as well. This leads to the need for adaptive control methods, which will be discussed further in Section 2.5 below.

From a system identification perspective, identifying the primary path $\boldsymbol{D}(q^{-1})$ can be tricky. In cases when the statistics are stationary and one can control the feedforward signal in the identification process, the identification is straightforward. For the experiments presented in the papers in this licentiate thesis, all stationary transfer functions have been identified prior to control using a method with sweeps of sinusoids [3].

When the primary path is nonstationary it needs to be identified online. For such a situation, or when the feedforward signal cannot be controlled in the identification process, there can be a problem with the input not being persistently exciting. An input signal to a system needs to be persistently exciting in order to be able to properly identify the system. This means that the signal has to excite all the properties of the system. It is for example not possiple to identify how a linear system will react to a frequency that is not present in the input signal during the identification. If the noise path is to be identified without control over the input signal, this has to be kept in mind.

There are several ways of acquiring feedforward signals, depending on the problem at hand. One way is to place one or several microphones close to the source of the primary noise to be cancelled. The primary path will then be the transfer function from the feedforward signal measurement positions to the ROI. Depending on how close these feedforward microphones are to the ROI, an unwanted side effect of this approach can be that the control noise sent out to attenuate the primary noise is also picked up by the feedforward microphones. The feedforward signals then becomes corrupt, a situation which has to be dealt with. A way of handling such a feedback situation is to use a model of the feedback path from the control loudspeakers to the feedforward microphones to subtract the contribution from the control signals before sending them to the controller.

In some particularly important applications for ANC, the primary sound field is produced by rotating machines. The sound field from such a machine is often narrowband, and the fundamental frequency of the resulting sound field can be deduced from the rotation speed. For such cases, feedforward signals can be obtained from measurements of the rotation, such as tachometer signals. Examples of noise generated by rotating machines that can be attenuated with ANC include fan noise in ducts, engine induced noise in cars and propeller induced aircraft cabin noise.

A potential problem with a pure feedforward controller such as the one illustrated in Figure 2.3 is the lack of supervision. There are no safety nets that take the actually obtained error measurements $\boldsymbol{\varepsilon}(t)$ into account to adjust the control signals. If the transfer paths of the system are estimated erroneously, or wrongly assumed to be stable over time, the performance will be bad. Instead of attenuating the primary noise, the controller could end up enhancing it. In order to avoid such a behaviour, it is important to ensure good modeling and make sure that the control method is robust to the type of model errors that can be expected.

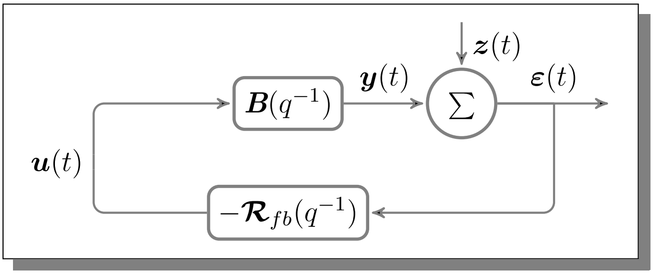

Figure 2.4. Block diagram describing a feedback control system

2.4 Feedback Control

In a feedback control system, microphones are placed in the ROI to measure the control error and measurements $\boldsymbol{\varepsilon}(t)$ from the error microphones are fed to the controller, as illustrated in Figure 2.4. No knowledge about the primary noise is required, as opposed to feedforward control. However, with feedback control it is even more important to keep down the delays in the controller filters to be able to catch up with the primary noise. In theory, a feedforward controller can catch up with the noise signal by reacting before the noise reaches the ROI. The feedback controller will not see the primary noise until it is already in the ROI. It is again a good idea to place the control loudspeakers as close as possible to the ROI.

A feedback control system will have an inherent robustness that is missing in the feedforward control system. Because the end result is always monitored through the error microphones, the controller will be made aware of the effects of its control signals. The price to pay for this is a slower response time and the risk of instability in the feedback loop.

Having error microphones within the ROI is a necessity in feedback control, a drawback compared to feedforward control. The volume of control is often quite small and for example centered around the head of someone occupying the volume. The need of error microphones within the volume can put severe limitations to the practical use of the methods. One way around this problem is to move the zones of silence away from the error microphones by use of virtual microphone control, see e.g. [4].

2.5 Adaptation

Controllers can be divided into feedforward and feedback controllers as described above. However, one can also classify them in terms of adaptive or nonadaptive controllers. When all transfer functions in the system are known to be stationary and it is possible to model them in advance, a nonadaptive controller can be implemented. The controller is then calculated offline, and is fixed during control.

If the statistics of one or several transfer functions are known or suspected to change during control, a controller calculated based on fixed transfer functions will soon be outdated and start to perform badly. Then the controller will need updating as the transfer functions change, which gives rise to adaptive control methods.

Many adaptive control methods still assume that the control path is stationary, such as the filtered-x LMS method discussed more in Chapter 3 and also the methods presented in this thesis. The primary path and the feedforward noise statistics are on the other hand more often allowed to be time-varying.

Even if the statistics of the transfer functions are stationary, it can be difficult or inconvenient to model them offline. Then adaptive methods are also needed, even if they are used more as self-tuners in such cases.

Gain scheduling is a middle way between adaptive and nonadaptive methods. It can be used for example in an ANC system in a passenger car to switch between different predefined controllers based on the number of passengers in the car.

It is an advantage to have as much as possible of the computations of the controller design done offline due to the computational complexity of the control methods which is often quite high. Even so, a lot can be gained by adapting to changes when they occur. Figure 2.5 shows the results from two different simulations, presented in Paper V. In the simulations, the primary noise $\boldsymbol{n}(t)$ was designed to contain the first two engine orders of a four-cylinder engine. The engine was reved up from 2000 rpm to 5000 rpm. For this scenario, four controllers were designed and evaluated, and the results from two of them are presented here.

For both controllers, the transfer functions of the noise path $\boldsymbol{D}(q^{-1})$ and the control path $\boldsymbol{B}(q^{-1})$ were modeled in advance and kept stationary during the simulations. The first controller did not assume any knowledge of the statistics of the feedforward noise signal, whereas the second controller had an adaptive scheme. A model was estimated repeatedly based on previous feedforward signals and the controller was updated according to the new model.

The results in Figure 2.5 clearly show how the adaptive method outperforms the nonadaptive. By following the changes in the primary noise statistics, more information was available to the controller. The tradeoff here is that one gets better performance but the adaptive method has a heavier computational burden during control. This places larger requirements on hardware in a real implementation.

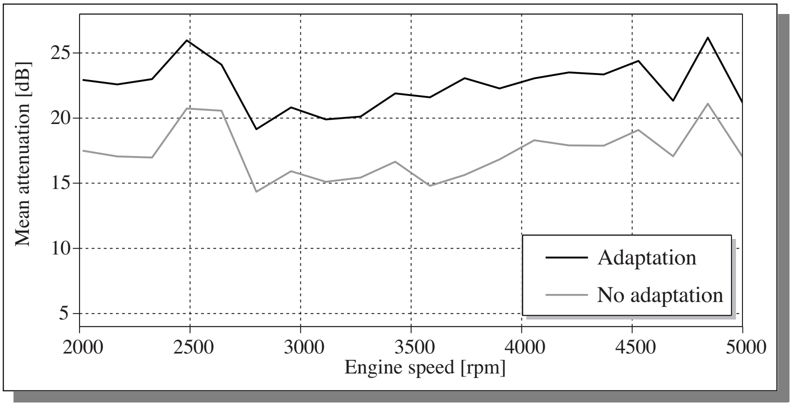

Figure 2.5. The improvement of using an adaptive controller to follow changes in the primary noise statistics compared to not assuming any knowledge about the primary noise.

2.6 Enhancement of Noise Outside the Region of Interest

As mentioned above, the basic underlying principle of ANC is the principle of superposition. A control wavefield out-of-phase with the primary wavefield will cause an attenuation of the latter. However, the control objective is in general only to match the phases of the soundfields in a limited region of control. The criterion to be minimized is in most problems formulated so that it does not regard what happens outside this region. Therefore the two wavefields may very well be in-phase somewhere outside the ROI, which will lead to an amplification of the primary noise there. It is actually most likely that the wavefields will be in-phase somewhere since global ANC is unreasonable in many real applications.

To illustrate how the attenuation within the ROI can transform to an amplicifation outside the region, predicted attenuations over a volume of dimensions $130 \times 30 \times 30$ cm are shown for four slices through the volume in Figures 2.6 and 2.7. These figures are presented in Paper IV. A feedforward controller was designed for 16 measurement positions distributed on the horizontal slice $z = 100$ mm, between the two vertical slices $x = 500$ mm and $x = 800$ mm. Using this controller, simulations were made with one 150 Hz and one 400 Hz primary noise signal.

The 150 Hz signal is attenuated throughout the volume shown in Figure 2.6 even though the attenuation falls off steadily outside the ROI. For the 400 Hz signal though, significant amplifications occur. At worst, there is a 10 dB amplification of the primary noise in the slices shown in Figure 2.7.

It is important to be aware of the risks of amplifying the primary noise outside the ROI. By placing the control loudspeakers close to the ROI, the effects can be limited. The closer the loudspeakers are to the ROI,

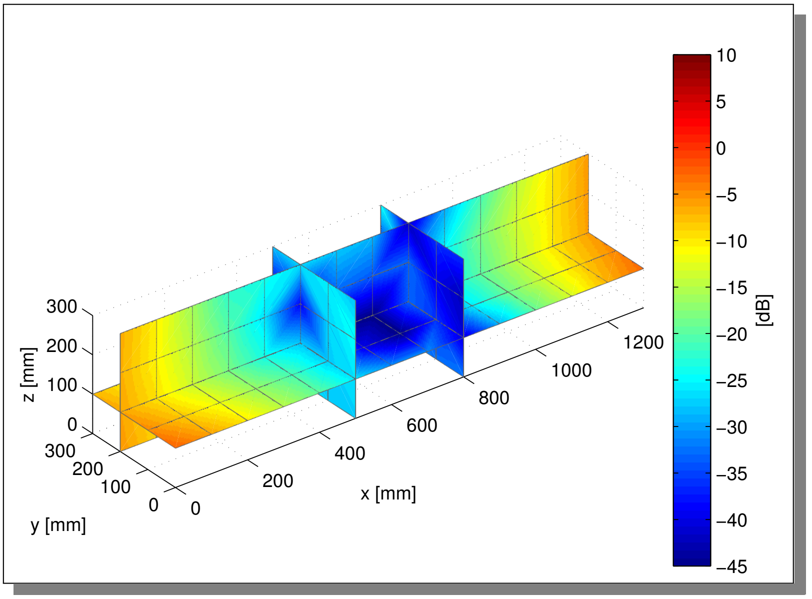

Figure 2.6. Volume plot of the resulting sound field in a 130 ×30×30 cm volume after control of a 150 Hz signal with a controller designed with a 30 × 30 cm ROI.

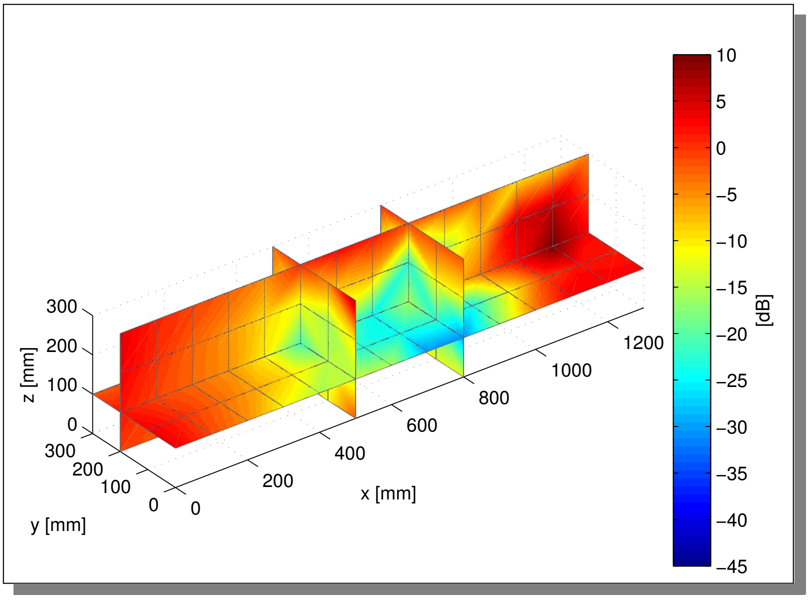

Figure 2.7. Volume plot of the resulting sound field in a 130 ×30×30 cm volume after control of a 400 Hz signal with a controller designed with a 30 × 30 cm ROI.

the lower output levels are needed. With lower output levels, the natural damping of the sound as it travels will limit the amplifications away from the ROI.

2.7 Nonlinearities

As mentioned in Section 2.1, the propagation of sound is a linear process. However, nonlinearities can be introduced to the system, the most significant source being the control loudspeakers. Harmonics introduced by the control loudspeakers would add frequencies to the sound field that are not present in the primary noise. These harmonics would contain higher frequencies than the primary noise and might very well become audible in which case it presents a problem.

A preventive action against introducing harmonics is to make sure that the control signals sent to the control loudspeakers will not saturate the loudspeakers. Saturation of a loudspeaker cause clipping, which apart from potentially being harmful to the loudspeaker causes higher frequency harmonics. Another important action is preventing the control loudspeakers from being used outside their operating frequency range.

Both keeping reasonable levels on the control signals and keeping them within the operating frequency ranges of the control loudspeakers can be done by adding a frequency-weighted control signal penalty term to the minimization criterion (2.7). Still, though, some safety net might be needed to monitor the control signals so that they will not cause saturation.

2.8 Reproducibility

This being said, in order to be able to achieve a satisfactory attenuation of the primary noise, the control loudspeakers must be able to generate sufficient sound pressure within the ROI. For this to be possible, loudspeaker placement is highly important. The control loudspeakers need to be placed so that they are able to couple well to the ROI, and to reach the subspaces in which the primary noise is dominant. A good tool to use to find out if the control loudspeakers are placed in a suitable way in consideration of the primary noise is the reproducibility measure. The reproducibility of the primary sound field by the control sound field is a measure that will give an indication of the possibilities for ANC in a specific setup.

The reproducibility measure is dealt with in paper III and paper IV. We show how it can be used as a prediction of achievable levels of attenuation for an ANC system. An example of this is shown in Figure 2.8. The

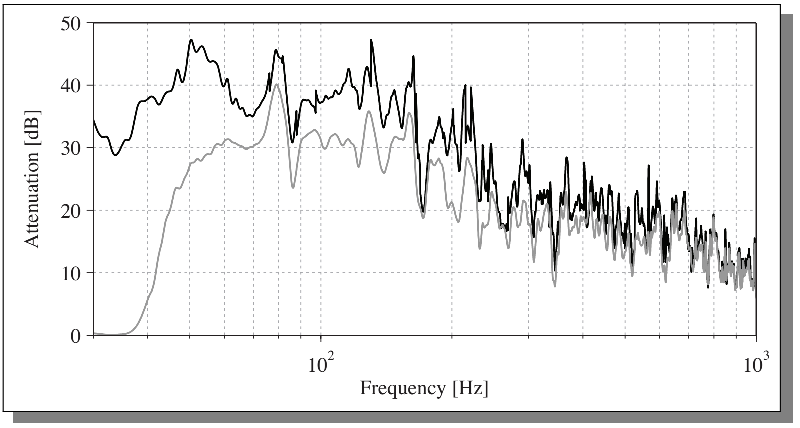

Figure 2.8. The predicted attenuation from the reproducibility of the primary sound field by the control sound field (black line) and the obtained attenuation from simulations (gray line).

upper, black, curve shows the reproducibility of the primary sound field in the ROI by the control sound field in a system with one primary noise signal and eleven control loudspeakers. The lower, gray, curve shows the resulting attenuation when using a feedforward controller with the assumption that the feedforward signal is perfectly predictable. However, the actual statistics of the feedforward signal has not been taken into account, as was discussed in Section 2.5. Had this been done, the resulting attenuation could be expected to be even closer to that predicted by the reproducibility curve.

3. Related Research

The ANC field is far from new. In 1936 the first patent [5] describing the principles of attenuating noise was published by Paul Lueg. The patent described a feedforward method for ANC in ducts which assumed knowledge about the primary path as well as the control path. At that time, implementations were limited to being analog which restricted the practical use of ANC. It is difficult to achieve the accuracy necessary for ANC in an analog implementation, why the field did not take off.[1]

Again in the 1950s there was some attention to the field. Olson and May [7] published a paper in 1953 that described a feedback ANC system that they called an electronic sound absorber, using no information at all about the transfer functions of the system. At around the same time, Conover [8] published his work on using reference signals made up of the same frequency components as that of the primary noise instead of using a microphone to pick it up as in the 1936 patent. Doing so eliminates the problem of secondary paths, where the noise picked up by the reference microphone not only contains the primary noise to be cancelled, but also the control signal.

Even though there was some attention to ANC before the 1970s it wasn't until then that the field really took off. In the early 70s a rapid development of digital signal processing techniques and devices led the way. Suddenly the tools needed to implement ANC algorithms were available. The first digital applications are attributed to Kido [9] in 1975 and Chaplin and Smith [10] in 1978. These papers has been considered the basis of the feedforward ANC research area [11].

Traditionally the most popular method for adaptive feedforward ANC is filtered-x LMS. It was introduced by Burgess [12] and Widrow [13] in the beginning of the 80s for the SISO case and extended by Elliot and Nelson [2] to the MIMO case. The method has been shown to work in cars, with implementations both for narrowband engine noise [14] and broadband road noise [15]. It has also been implemented in a fork-lift truck cabin,

[1] In 2001 though, an analog controller was indeed applied in a commercial car [6]. A station wagon had an unfortunate design leading to a drumming noise around 40 Hz to be generated in the front seat while driving. This problem was solved using feedback ANC implemented using analog circuitry combined with the existing audio system in the car. There are also noise canceling earphones in which the implementations are analog. However, even though there are analog implementations around, digital techniques are in no doubt needed for their development.

see e.g. [16] and in a slightly altered form in propeller aircraft [17]. In [18], the method is shown to be a special case of the more general Generalized Minimum Variance (GMV) adaptive controller. The adaptive feedforward filtered-x LMS algorithm has been presented in many more variations, recent examples include [19] with the filtered-x logLMS algorithm for impulsive noise and [20] in which a frequency estimator is added.

Adaptive methods are very popular for feedforward control, but have in general been used for self-tuning of fixed linear controllers, rather than true online adaptation when broadband noise is to be considered. Feedback control algorithms for ANC are traditionally focused on nonadaptive techniques. For example, the use of Internal Model Control (IMC) [21] for nonadaptive feedback ANC using filtered-x LMS for self-tuning of the controller has been investigated using simulations in [22, 23]. Another method that has been investigated is H_2/H_∞ control, see e.g. [24].

In the early 2000s adaptive feedback ANC started to attract more interest. An adaptive feedback ANC system using filtered-x LMS was implemented and successfully used to control vibratory bowl noise and welding power generator noise in 2003 [25]. Pawelczyk [4] proposed an adaptive virtual microphone control system based on IMC to move the zone of quiet away from the error microphones in an active headrest system.

Many adaptive ANC systems assume that the control path is known and stationary. An example of an adaptive feedback approach where no transfer paths are assumed to be stationary or even known is presented in [26, 27].

This short review of the research area is far from exhaustive. Over the years several overviews of the research area has been published, along with several books. For a deeper plunge, see e.g. [1, 11, 28–32].

3.1 Contributions

As is obvious from the research review above, ANC is not a new subject, and there is still much research going on today. This does not mean, however, that it is a saturated field. There are many aspects of a full ANC system that need attention and even though solutions have been proposed for most of these aspects, there is definitely room for improvements.

A full ANC system consists of many different parts. Apart from the obvious controller design stage, feedforward signals need to be acquired, through measurements or by synthesis or IMC. In the event of saturation of the control loudspeakers, this has to be detected and dealt with. In the case of feedback, it is reasonable to say that the use of virtual microphones to move the controlled region away from the error microphones is neccessary for the ROI not to be cluttered.

In this licentiate thesis, I have focused on a MIMO MMSE LQG controller. I have shown that this controller can be used to achieve uniform damping in an extended region in space and push the upper frequency that can be controlled.

Furthermore, I have investigated the influence of different design variables and tried to give guidance on how to choose them. In addition to this, I have investigated how the properties of the control path can be analyzed and compared to the properties of the primary path to indicate the achievable performance with for a given setup.

Finally, I have looked at adaptation of the controller to make it follow the statistics of the feedforward signals. I have shown that using estimates of the feedforward noise statistics to adapt the controller will give large gains on performance when the noise statistics is nonstationary; a likely scenario in many applications.

4. Summary of Papers

4.1 Paper I - *Extending the area silenced by active noise control using multiple loudspeakers*

This paper presents simulations and verification measurements from the first experiment we did while using the MSE feedforward controller for ANC. In the experiment, eight loudspeakers were set up around a sofa, seven of which were used as control loudspeakers to control the noise sent out from the remaining loudspeaker in a 3×3 dm area. Transfer functions were measured at 16 measurement points, positioned as a square pattern in the area. The results from the MIMO feedforward LQG controller is compared with results from a SISO LQG controller. The achievable attenuations for sinusoid signals at frequencies 200, 400 and 600 Hz are presented for each measurement position both for the MIMO and the SISO case, where the latter is optimized for one of the 16 measurement positions. This comparison shows that the limiting frequency for uniform damping in the examined area was increased from 200 Hz for the SISO case to around 600 Hz for the MIMO case.

There is a good correspondence between the simulations and the verification measurements. Also, a real-time broadband signal with energy in the frequency range 60-700 Hz was sent through the system. From this, an average attenuation over the controlled area of more than 10 dB was obtained for most frequencies in the range 70-500 Hz. For the frequency region 500-700 Hz, there was less attenuation, but it was still significant, especially considering the irregularity of the transfer functions between the control points in that frequency region.

The experiment was performed in a reverberant but rather well damped acoustic environment, and the noise path had the same general properties as the control paths, since the noise source was the same type of loudspeaker as the control loudspeakers. These two factors simplify the problem somewhat, which made for a good proof of concept.

The paper was presented at a poster session at the International Conference on Acoustics, Speech and Signal Processing (ICASSP) held in Kyoto, Japan in March 2012.

4.2 Paper II - *MIMO design of active noise controllers for car interiors: extending the silenced regions at higher frequencies*

In this paper we continue the investigation of the MSE feedforward controller for ANC applications. Here, the method is tested in the more difficult acoustic environment of a car cabin. The built in car sound system, consisting of nine low- and mid-range loudspeakers, were used as control loudspeakers and an external subwoofer was used as noise source. The external subwoofer was placed in the trunk of the car face down over a beam so as to couple well acoustically with the body of the car, simulating engine noise.

Experiments were performed to evaluate the performance of the controller, again in comparison with the SISO LQG controller. Both simultions and verification measurements were made for one narrowband design with the assumption of predictability of the feedforward noise signal, and one broadband design. The simulations show a considerable damping for both designs and the validation mesurements show good accordance with simulations except in the frequency range 150-180 Hz where the modeling errors are relatively high.

Validation measurements were made by attenuating sinusoidal signals of 100, 200, 300 and 350 Hz as well as a broadband signal with frequency content ranging from 30 Hz to 300 Hz. Attenuations of the 350 Hz signal in-between the original measurement positions and 1 dm outside of the area of control are also presented.

The results show that in comparison with the SISO controller, the evaluated method raises the level of attenuation and pushes the limiting frequency where uniform damping can be achieved up to around 450 Hz.

This paper was presented at the 2012 American Control Conference (ACC), held in Montreal, Canada in June 2012.

4.3 Paper III - *An investigation of a theoretical tool for predicting performance of an active noise control system*

The design process for the MIMO LQG controller is computationally demanding and involves making design choices that influence the results. This is a more theoretical paper that focuses on finding a way of predicting the achievable performance of an ANC system without having to go through the sometimes cumbersome controller design process. This is done by looking at the effective rank of the MIMO transfer function of the control path and the reproducibility of the noise path by the control path. It takes into account the signal directions in which the control system produce substantial control energy and projects the noise path onto this subspace.

The method was then investigated using estimates of real room impulse responses for the noise path and the control paths. It was demonstrated how the effective rank of the control path can be used to see if potentially there are superfluous loudspeakers. Also, the reproducibility of the noise path was compared to simulated attenuation curves for different controller designs. The results show that the reproducibility measure will give a good indication of achievable performance for the ANC system when the design is made for the case with feedforward signals that are highly predictable. For a design for a broadband noise signal when prediction is not feasible, the accordance of the theoretical prediction to the simulation results will be lower for lower frequencies.

This paper was presented at the 19th International Congress on Sound and Vibration (ICSV19), held in Vilnius, Lithuania in July 2012. I received the *Sir James Lighthill Award for Best Student Paper* for this contribution.

4.4 Paper IV - *Design and analysis of linear quadratic gaussian feedforward controllers for active noise control*

There are two main focus areas for this paper. First, the theoretical concepts of effective rank and reproducibility of a desired sound field for the narrowband case are evaluated in more depth than was room for in Paper III. Second, the influence of the control signal penalty term in the criterion is investigated, both in terms of performance in relation to the theoretical performance limits from the reproducibility measure, and in terms of control signal energy usage.

Two different ways of designing the control signal penalty matrix are evaluated. The first is a very simple diagonal matrix where each entry has a low gain in the operating frequency region of the respective control loudspeaker and a high gain outside of that region. This will prevent the control loudspeakers from being used outside of their operating region. The second is a more complex penalty matrix, designed based on the effective rank of the control path which will punish the output principal directions that are hard to reach with the available control loudspeaker setup. This latter penalty matrix has a close relationship with the reproducibility of the target (noise) path by the control path.

The results show that there is some control energy to gain by using the effective rank of the control path to find the possibility to remove control loudspeakers from the setup. It is also shown that there is no advantage to using the more complex penalty matrix investigated. The resulting attenuation is somewhat lower than the attenuation achieved when using a simple diagonal control penalty matrix. Furthermore, there is no gain in control energy. For the setup used, no reason to use the more complex control penalty matrix has been found.

This paper is submitted for publication.

4.5 Paper V - *Adapting an MSE controller for active noise control to nonstatic noise statistics*

For all papers above, all transfer functions were assumed to be stationary and the controllers were calculated offline. In paper V, the noise statistics are allowed to change and two partly adaptive methods are introduced and evaluated. Both are batch-methods that collect data during a time batch and calculate new signal models based in this data set. While this is done, the controller is kept fixed. A new controller is then calculated, which is used within the next time batch.

In the first method, a new controller is calculated for each time batch based on an estimation of the noise statistics from the previous time batch. The second method calculates a controller offline that only takes into account the predictability of the feedforward noise signal. A predictor is then built and updated each time batch, again based on an estimate of the noise statistics from the previous time batch.

The two methods are evaluated in simulations based on real measured room impulse responses and compared against a controller calculated offline with no consideration taken to the statistical properties of the feedforward noise signa. The feedforward noise model is in the simulations designed to consist of the frequencies corresponding to the first two engine orders of a four-cylinder car engine. The engine is then simulated to rev up quickly from 2000 rpm to 5000 rpm during a time interval of 2 seconds.

The results show that the first method has great promise, as it increases the attenuation by an additional 5 dB throughout the simulation when compared to the fixed offline-designed controller. The second method shows less advantage and is believed to have restricted fields of application.

This is a preliminary investigation which is included as a technical report in this licentiate thesis.

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