Active control for adaptive sound zones in passenger train compartments

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Summary
The acoustics in train compartments is an important part of the comfort when travelling. To improve the acoustics, active noise control (ANC) can be used to create local quiet zones at the passenger ear position. In this thesis the fundamental ANC theory is explained and the possibilities and limitations with the technique are identified. An ANC system which uses the virtual microphone technique is implemented and quiet zones are realized at a chair in a train compartment studio. The performance of the ANC system is evaluated with a Styrofoam head model with built in microphones that is used to produce contour plots of the quiet zone shape and attenuation level. The results show that the 10 dB quiet zone is relatively large covering more than a square decimeter in low frequencies and decreases with frequency. Head movements effects are also evaluated and the results show that the system is very sensitive to head movements, especially in higher frequencies. In order to find an optimal positioning of the system components, several experiments have been made. From these it is concluded that varying noise incidence is problematic to some extent. Ways of handling the problems are presented and if the problems were properly dealt with ANC could be an effective way of reducing noise in passenger train compartments.

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1 Introduction

1.1 Project description
This master thesis focuses on improving the acoustic environment locally in train compartments with active noise control by creating quiet zones. The thesis is part of a project which with the combination of passive and active means aims on creating adaptive sound zones. Adaptive sound zones which the passenger can customize by choosing different sound modes will increase the level of comfort.

1.2 Problem formulation
The main question that this work focuses on answering is whether or not ANC is a suitable technique for controlling the noise in a train compartment. In order to answer this, knowledge regarding the ANC system performance and the quiet zone is needed. In what frequency range is ANC effective? How can one optimize the number and positions of actuators and microphones to get as good performance as possible? How does the quiet zone extend in space and how is the quiet zone affected by head movements? Can ANC control a diffuse sound field? Sections 1.3, 1.4 and 3 try to answer these questions. All information in section 1.3 is from reference [1].

1.3 Active Noise Control Theory [1]
Active noise control (ANC) is based on the principle of superposition. Noise is cancelled by a so-called anti noise of equal amplitude but with opposite phase than the original noise signal, see Figure 1. ANC systems consist of a plant, sensors, actuators and a controller. The plant is the environment in which the ANC system is implemented. It can for example be enclosed, as in a duct or be a free space. For sound, the sensors are typically microphones and the actuators are speakers. The ANC system can be single channel or multichannel referring to how many input signals and output signals the system uses. There are basically two types of ANC systems, namely feedback and feed forward systems which will be described for the single channel cases further down.

![Figure 1, Principal sketch of active noise control](image)

1.3.1 Feed forward active noise control
In a feed forward single channel ANC system a reference microphone measures the character of the noise to be cancelled. The original noise at the error microphone is called the primary noise and originates from the primary source. In Figure 2 a single channel feed forward ANC system in a duct is shown. The controller gets the input reference signal $x(n)$ to identify the primary noise and the error microphone signal $e(n)$ processes these microphone signals to produce the anti noise signal $y(n)$. The controller output signal $y(n)$ drives the speaker to play back the anti noise.
The anti noise will henceforth be called the secondary noise and is played back by the secondary source, the speaker. The secondary noise is designed to minimize the primary noise at the error microphone position. The error microphone measures the residual error and the error signal $e(n)$ is used to adapt the controller coefficients in order to minimize the error.

![Diagram of feed forward active noise control](image)

**Figure 2, Feed forward active noise control in a duct**

The coherence between the reference and error microphone needs to be good for good performance of the feed forward ANC system. The controller can only cancel out the part of the sound at the error microphone position that is correlated with the sound that is detected by the reference microphone. Hence a correlated reference signal from the primary noise source is needed for good performance. Also, only the noise that the error microphone is able to sense is possible to cancel out.

1.3.2 **Feedback active noise control**

In a feedback system for active noise control there is no reference microphone. The objective is, as for feed forward control, to cancel out the primary noise at the error microphone position. In Figure 3 a single channel feedback ANC system in a duct is shown. The controller processes the error signal $e(n)$ and reverses it with 180 degrees phase shift to produce the output signal $y(n)$. The output signal drives the secondary source to play back the secondary noise.

![Diagram of feedback active noise control](image)

**Figure 3, Feedback active noise control in a duct**

The performance of a feedback system depends predominantly on the system delay. The speaker response and the acoustic propagation time from the speaker to the error microphone introduce...
a phase shift and time delay. The phase shift increase with frequency and at some frequency the phase shift will turn the negative interference, which the ANC controller was designed for into positive and the system will become unstable.

1.3.3 Least mean square algorithms for adaptive systems

The noise to be cancelled is often time varying in a realistic implementation. Because of these reasons the frequency, phase and amplitude of the noise to be cancelled is non-stationary.

Hence adaptive filters are used. The adaptive filter algorithm modifies the controller filter coefficients in order to achieve some predefined objective. The most commonly used algorithm for ANC systems is the least mean square (LMS) algorithm which minimizes the squared error. In the LMS algorithm the filter coefficients \( w(n) \) are updated as

\[
    w(n + 1) = w(n) + \mu x(n)e(n)
\]

where \( \mu \) is the step size, \( x(n) \) the reference input vector and \( e(n) \) is the error signal at time \( n \).

There are many types of modified LMS algorithms. The modifications are motivated by practical reasons to optimize for example robustness, stability or convergence speed etc. The ANC system used in the experimental tests in this thesis uses the leaky filtered x LMS (FXLMS) algorithm. Hence a short description of the FXLMS algorithm and the leaky FXLMS follows.

The FXLMS algorithm takes in consideration the fact that the ANC system itself will have some effect on the signals required for the adaptive algorithm. The electrical devices that the system consists of will not have flat responses and will introduce time delays and so will the acoustic propagation paths as well. In order for the algorithm to converge the influence of the system has to be compensated for. Each component and propagation path in the system can be described by a transfer function. The influence of all the parts in the system can be described by a single transfer function which is a superposition of all the system transfer functions, the so called secondary path. In the FXLMS algorithm this secondary path is estimated and the reference signal \( x(n) \) is filtered by the estimated transfer function (secondary path) in the weight update path. The reference signal that the FXLMS algorithm uses is hence the filtered reference signal \( x' \) which is given as

\[
    x'(n) = s(n) * x(n)
\]

where \( s(n) \) is the impulse response of the estimated secondary path, \( x(n) \) is the reference microphone signal and * denotes linear convolution. The FXLMS algorithm is then

\[
    w(n + 1) = w(n) + \mu x'(n)e(n)
\]

where \( \mu \) is the step size, \( x'(n) \) the filtered reference input signal and \( e(n) \) is the error signal at time \( n \).

The FXLMS algorithm can be used in a system identification stage before the system is running. This is so called offline modeling and is frequently used in ANC systems where the secondary path can be considered as time invariant. With online modeling the FXLMS algorithm updates the secondary path filter coefficients continuously while the system is running. By doing so changes in the plant that will affect the secondary path can be compensated for and the performance of the system can be maintained.
Sometimes high primary noise levels due to low frequency resonances can make a direct implementation of the FXLMS unstable. Having an unconstrained output from the system may then overdrive the speakers causing nonlinear distortion. Introducing a leakage factor $\nu$ in the FXLMS algorithm limits the system output power and has a stabilizing effect on the adaptive algorithm. The leaky FXLMS algorithm is written as

$$w(n + 1) = \nu w(n) + \mu x'(n)e(n)$$

where $\nu$ is the leakage factor, $\mu$ is the step size, $x'(n)$ the filtered reference input signal and $e(n)$ is the error signal at time $n$. The leakage factor guarantees a unique solution for the algorithm which will be stable at the expense of introducing a bias in the converged solution.

1.4 Background

The acoustic environment is an important part of the perceived comfort when travelling. The demand for silent products is considerable and there are strict acoustic requirements on a large part of the products delivered in the EU [2]. Good acoustics indicates good product quality and is a product property of increasingly importance on the market [2]. The main train operator in Sweden is marketing trains as a clever and comfortable way of traveling and emphasizes the acoustics on trains compared to other ways of travelling. The continued work of developing and improving the acoustics is hence of great importance for the train industry in order to stay at the leading edge.

To improve the acoustics in train compartments passive means such as absorptive materials or reflective panels can be used. The characteristics of passive means are good for high and mid frequencies where large reductions of the sound pressure level (SPL) can be achieved. However, for low frequencies absorptive material and panel dimensions often gets impractical large or too heavy.

Active noise control (ANC) on the other hand works best at low frequencies and is consequently a promising technique for reducing noise. The noise in a train compartment could partly be cancelled by loudspeakers that are driven by a controller. A typical spectrum shape of the internal noise in a train compartment on a high speed train is shown in Figure 4 [3]. The low frequencies below 1000 Hz are the dominating contributions of the total SPL. The spectrum is shown for a train speed of 300 km/h. For lower train speeds the low frequencies will be even more prominent [3]. Since ANC is most efficient in low frequencies and much less space consuming in comparison to passive means in that frequency range it would be beneficial to implement it in train compartments for low frequency control.
The ideal case would of course be to get as large attenuation level as possible in the whole train compartment which is referred to as global reduction. In principle global reduction of the SPL is possible with ANC but in practice this is very difficult to achieve. To achieve global reduction all acoustic or coupled structural/acoustic modes in the primary noise must be observable and controllable by the ANC system [4]. This means that the reference microphones must be able to observe all modes that one wishes to cancel out, which places demands on the positioning of the microphones and the secondary sources must be placed so that they are able to control the primary noise.

ANC has successfully been implemented in ducts and reductions of typically 30 dB are possible for single tones [1]. In a duct the sound propagates as plane waves under the cut on frequency and also the sound incidence is well defined. Under the cut on frequency only the first mode is excited and the pressure is equal in any cross section. This makes it possible for a single channel ANC system to control the sound field globally downstream from the secondary source. Above the cut on frequency higher modes are excited and the sound no longer only propagates as plane waves. This complicates things for ANC more secondary sources are needed to control the sound field. In general one source is needed for each of the dominant modes in the medium [5].

For a more complex environment than a duct such as free field, a normal room or why not a train compartment a large number of modes build up the acoustic field and the noise sources are in general multiple. Then a multiple channel ANC system is required to effectively observe the noise. One realizes that for a practical ANC system, with finite number of sensor and actuators, global reduction will not be possible except at very low frequencies.

In more complex sound fields local quiet zones are still practically achievable and can be highly effective if the quiet zone is realized at the listener ears. A local quiet zone with 10 dB reduction can be achieved and the diameter of the quiet zone can be as big as a tenth of a wavelength for a monopole secondary source [6]. These results show that ANC works best at low frequencies where the quiet zone is large. To be practical effective the quiet zones needs to be sufficiently effective...
large allowing the listener to have a large active range for the head without moving out of the quiet zone.

Implementing ANC in a three dimensional space such as a train compartment with several noise sources and a complex sound field imposes many challenges. To get maximal performance careful considerations need to be made on what kind of ANC system to use. How many sensors and actuators should be used and where to put them? What type of controller algorithm is the best for controlling and how to set the parameters for this?

The sound field in the train compartment needs to be analyzed in order to choose what type of ANC system to use. The properties of the sound field will define the limits for a given ANC system. How high is the SPL in the train compartment? Is there some dominating sound propagation direction or can the compartment be modeled as a diffuse field built up by multiple modes? Is the sound field stationary or does it varies in time? All these questions and more needs to be answered to successfully implement ANC and hence follows some performance requirements and positioning guidelines for the different parts of an ANC system.

The performance and positioning of the speakers has a critical influence on the ANC system efficiency. To begin with, the speaker must be able to produce a sufficient level and be able to play back the frequencies in the primary noise field of interest to cancel out. Loudspeakers are usually not omnidirectional so the radiated pressure field varies spatially [6]. As the pressure radiated from the speaker gets more spatially uniform higher attenuation levels and bigger zones of quiet are achievable [6]. This indicates that the error microphone should be placed some distance from the loudspeakers. Generally larger loudspeakers produce larger zones of quiet [6].

The positioning of the secondary sources will determine the upper attenuation bounds for the ANC system [7] and is thus of great importance for the system performance. The number of secondary sources needed to cancel out the primary noise depends on their position relative to the primary source or relative to the quiet region [5]. If the sources are multiple generally more speakers are required to cancel out the primary noise field. Several loudspeakers in an array and microphones placed in an enclosure can create a quiet zone around the error microphones in a diffuse field [8]. This technique is the so called virtual sound barrier (VSB) and it can increase the quiet zone significantly [8]. The most important aspect of the VSB is that it can work in an environment where the noise comes from several directions such as a diffuse sound field [8]. Having several loudspeakers and reference microphones like the VSB could create large quiet zone in a train compartment but it would at the same time make the system a lot more complex and more difficult to implement due to practical reasons.

Implementing an ANC system in an enclosure such as a train compartment and play back a secondary noise will cause the total acoustic energy to increase. The controller is designed to interfere destructively at the cancellation points but there is nothing guaranteeing that positive interference will not occur at some other location in the enclosure [9]. By putting the speakers close to the error microphone the SPL well away from the loudspeakers will be largely unaffected by their output [9]. Then the speakers can control the sound field in their close environment without having too much effect on the far field. Placing the speakers near the control point also minimizes the power needed to drive them. Other ways to avoid positive
interference could be to use directive speakers or panels to try to screen away parts of the environment that should not be affected by the ANC system.

Regarding the reference microphones the most important aspect is to get a correlated signal from the noise source. If the noise sources are multiple and the primary noise field is complex several reference microphones will be needed. In order to get a correlated reference signal the microphone must not be placed in a node of any standing wave in which case it will not be able to observe it. Sometimes it is difficult to get a coherent reference signal with a reference microphone due to noise problems. Depending on the noise sources sometimes other sensors such as accelerometers or tachometers can be used to deal with such problems. In order to work the reference microphone in a feed forward system must be placed so that the acoustic delay is bigger than the electrical delay in the ANC system [10]. The acoustic delay is the time it takes for the wave to travel from the reference microphones to the secondary sources. This is the causality constraint for causal filters. If it is violated the controller response will be non causal and the controller will be required to predict the primary noise. This will reduce the performance for broadband noise reduction [10]. It is shown that the narrower the bandwidth of the primary noise spectrum peaks the better the ANC system will perform [10].

Practical considerations on where to put the error microphones also need to be taken into account. The error microphones should be placed at the location where the quiet zone is desired. This would most likely be the passenger ear position for a train compartment implementation. But placing the error microphones directly at the ear positions would be highly impractical and the appeal for the technique would be severely reduced. The passenger would most likely not want to wear microphones and or other technical equipment when travelling. To deal with this problem a virtual microphone technique [11] can be used. The transfer functions between the reference and error microphone positions are estimated. Then with only the information from the reference microphones and the estimated transfer functions the quiet zones can be projected at the error microphone positions without the information from the error microphones. If the transfer function does not change the error microphones can be removed but staying at their positions “virtually”. The technique is built on the assumption that the acoustic primary noise is the same at the reference microphone and the virtual microphone position, an assumption which holds for low frequencies depending on the separation distance between the physical and virtual microphones positions [11]. For higher frequencies and or for a more diffuse field the virtual microphone technique will perform poor since there will be larger phase differences between the physical microphone and the virtual microphone positions [11].

If the virtual microphone technique is not used the placement of the error microphones can at first seem obvious. The ANC algorithm is designed to cancel out the noise at the error microphone position and hence the error microphone should be placed at the ear location. However if sufficiently large local quiet zones could be realized, making it possible to enclose the entire passenger head inside the quiet zone the placement of the error microphones could be more deliberated. With multiple error microphones a larger quiet zone is possible to realize [12]. But there is a tradeoff between the quiet zone size and attenuation level [12]. Jing Yuan found that varying the distance between two error microphones affect the quiet zone. The quiet zone is smaller when the distance between the error microphones is small, but the attenuation level increase [12]. If it is possible to use several error microphones, larger quiet zones or higher attenuation levels are possible to achieve.
The performance of the ANC system will be affected by head movements. The passenger when travelling will move their head and it is important to examine how the movements affect the ANC performance. If the virtual microphone technique is used head movements may cause the ears to move outside the quiet zone. Also the estimated transfer functions will be affected by head movements and no longer estimate the transfer functions well. This can lead to that the quiet zone is projected at some other place than the ear positions. If the zone was large then head movements would not be a problem. The active range for the head would be large and even if the passenger moved their head it would still be inside of the quiet zone. Because of limited quiet zone sizes the problem of moving away from the quiet zone exists.

A way of dealing with the problem with head movements would be to track the head movements and project the quiet zone at the ear location. If the ANC system used online modeling movements could be tracked and the filter coefficients could be updated as the ear position was moved. This would unfortunately require that the passenger had the error microphone on them. Online system identification also imposes other difficulties which will not be discussed here.

Another way to avoid online modeling but still track the head movements is described by Tom Waite [13] and others. The main idea is that the ANC system have several pre-identified systems (plants) and a built in plant selector that with the information from the tracking system can choose which plant to use for best performance [13].

If an ANC system was to be implemented in a train compartment the system could be customized by the passenger. Since ANC is an active signal processing technique the passenger could be able to choose different modes in an implementation such as perhaps "cell phone mode" or maybe use it as an active voice control (AVC) system. AVC is a technique which by active means tries to cancel unwanted speech. One of the big challenges with AVC is to cancel out the unwanted speech for some listeners while still preserving it for an intended audience [14]. AVC could, if the useful frequency range was extended to higher frequencies, be a possible technique [14] and maybe an application in an implemented ANC system. The speakers in the ANC system could also if preferred be used as a playback system for example music or so called sound-scapes. By combining both passive and active techniques adaptive acoustic zones could be realizable and higher acoustic comfort could be achieved.
2 Methodology

2.1 Objectives with the experiments
In this project several experimental tests were performed with an ANC system built by Silentium [15]. In the experiments quiet zones were realized. The main objective with the experiments was to examine if ANC is suitable for reducing noise in train compartments. The experiments were to examine how large the quiet zone near the listener ears could be and how big the reduction is in frequencies between 200- 2000 Hz. Further it was also examined how movements of the head, away from the position for which the system identification was designed for, affect the ANC system performance. The experiments were also to examine how the positioning of speakers and microphones affected the performance of the ANC system.

2.2 The Silentium ANC system
The ANC system that was used for the experiments is a multi channel feed forward ANC system which uses the virtual microphone technique. It is built by the company Silentium. There are two reference microphones, two error microphones and also two speakers (secondary sources). The system uses the leaky FXLMS algorithm and the system identification is done offline before the system can run. There is no online modeling. Since the system uses the virtual microphone technique a conflict in notation arises when calling the microphones for error and reference. The error microphones as Silentium are calling them are used in the system identification and then only monitor the system performance as the system run. They are put at the location where the quiet zone is desired. The reference microphones serve as reference microphones in their true meaning in the system identification stage. Then when the system runs they are the only microphones used for control and would perhaps better be called error microphones. However, in this thesis it was decided to stick with the Silentium notation.

The system identification involves estimating four transfer functions which are called MTF, STF, EC and PF by Silentium. The ANC system makes use of the leaky FXLMS algorithm to identify these transfer functions. All components of the system are first placed at their locations. The primary noise source, which in the laboratory experiments is a speaker playing back the primary noise, is placed at its location for the intended setup. Then the system identification can start.

The transfer functions between the reference and error microphones are called MTF. These transfer functions are models of the acoustic medium between the reference and error microphone positions. In order to estimate the MTF the primary noise source plays back the noise to be cancelled and the reference and error microphones record the noise signal. From these signals the MTF can be estimated. The MTF enables to calculate an estimate of the noise at the error microphone positions by using the reference signals. When the system is running the virtual microphone technique enables projecting the quiet zones to the error microphones position using the reference microphone signals and the MTF. The error microphones are then so to say the virtual microphones since their signals are not being used for control.

The transfer functions between the speakers (secondary sources) input and the error microphones are called STF. They are estimated by letting the signal generator in the ANC system feed a white noise signal to the speakers while the error microphones records the playback from the speakers. By knowing the STF the cancelling noise from the speakers can be estimated at the error microphone positions from the input to the speakers. This is the off line modeling of the secondary path.
The transfer functions between the speakers and reference microphones are called EC. By letting the ANC signal generator feed a white noise signal to the speakers and record the played back signals with the reference microphones the EC can be estimated. The EC enables information on the influence of the secondary noise at the reference microphone positions. The secondary noise that propagates back to the reference microphones from the cancelling speakers can be subtracted from the reference signal if the EC is known yielding the true primary noise signal.

The prediction filter, PF, is computed using the MTF and STF and the signals recorded by the reference and error microphone when doing the MTF estimation. From the signals the statistics of the primary noise can be estimated. The PF is used to predict future sampling data from the reference microphone signals. By predicting the data the system delay can be shortened allowing the system size to be more compact.

2.3 The train compartment studio
A full scale train compartment studio was built in the hemi anechoic room at MWL, see Figure 5. The studio simulated a three meter long section of a train and had one set of chairs. The width was three meter and the ceiling height 2.1 meters. It was built in plywood and on the walls absorbing materials were mounted to model the acoustics of a real train. At one of the chairs the ANC system was implemented.

2.4 The Styrofoam head model
A Styrofoam head model was built to evaluate the performance of the ANC system. The head model had 16 microphones at the left side of the head and was used to examine the extension of the quiet zone and how head movements affect the quiet zone extension. The microphones were mounted close to the surface of the head model and the information was used to produce
contour plots of the quiet zone. In Figure 6 two pictures of the Styrofoam head model are shown and in Figure 7 an example of how the contour plots map on the head model is shown. The contour plots are derived through linear interpolation between the data from 16 microphones points using the contour command in MATLAB and show the reduction in dB with the ANC system. The reduction is defined as the difference between the primary noise level and the noise level when the ANC system is running

\[ L_{\text{reduction}} = L_{\text{primary}} - L_{\text{ANC}} \text{ [dB]} \]  

\( L_{\text{ANC}} \) is measured by letting the primary source play back the primary noise as the ANC system play back the secondary noise through the speakers. The spatial resolution between the microphones sets a high frequency limit for which the results are valid. The speed of sound at normal pressure and temperature is approximately 340 m/s and it is 0.04 meters between the microphones. This gives a high frequency limit of 8500 Hz which is well above the intended measurement range.

Figure 6, The Styrofoam head model with 16 microphones at the left side of the head
Figure 7. The Styrofoam head model with a mapped contour plot derived from the microphone signal data
3 Experimental Tests

3.1 Extension of the quiet zone

The extension of the quiet zone was measured with the Styrofoam head model. The ANC system identification was made when the Styrofoam head model was in center position i.e. the ears at the same height as the upper edge of the headrest and the back of the head 0.10 m in front of the headrest, see Figure 8. The speakers were symmetrically placed at each side of the head facing the ears at a distance of 0.15 m from each ear. The error microphone positions were at each ear of the model. The primary noise source was placed 2 meters behind the chair and the reference microphones were on a stand one meter behind the chair, see Figure 9. The size of the quiet zone at the left side of the head model was measured in third octave bands for broadband random noise between 200-2000Hz. It was also measured for pure tones. Each 100th frequency from 200 to 1600 Hz was measured.

![Figure 8, Head model in center position](image1)

![Figure 9, Setup for measuring the extension of the quiet zone](image2)

3.2 Head movements

For the same setup as when measuring the size of the quiet zone the effect of moving the head from the center position was examined. The system identification was made for center position
of the head model and then without redoing the system identification the head was moved in the vicinity of the center position. The head was moved 0.10 m forwards, 0.10 m backwards, 0.10 m downwards and 0.10 m to each side. The sizes of the deviations from the center position are reasonable for a person sitting in a chair. For each position the quiet zone was measured in third octave bands.

3.3 Placement of the ANC system components
To examine how to place the components of the ANC system several measurements were made. The questions to answer were how to place the reference microphones and speakers for optimal performance and how the noise incidence affects the ANC system performance. For each of these measurements described in section 3.3 the primary noise was broadband random noise 200-2000Hz played back by one speaker (except for the diffuse incidence setup where five speakers were used as primary noise sources) and the error microphones were placed at each ear of the Styrofoam head model.

3.3.1 Reference microphone position
The primary noise source was placed 2 meters behind the chair and the speakers were symmetrically placed at a distance of 0.15 meters away from the ear positions at each side. The performance of the system was tested when the reference microphones were on a stand 0.3 meters apart placed 1.5 meters, 1.0 meter and 0.5 meters behind the chair. See Figure 10.

![Diagram of reference microphone position measurement setup]

3.3.2 Speaker placement
The primary noise source was placed 2 meters behind the chair and the reference microphones were placed on a stand 0.3 meters apart 1 meter behind the chair. The speakers were placed on each side of the head and the performance of the system was tested for the speaker distances 0.05, 0.1, 0.15, 0.2 and 0.25 meters away from the ear position. See Figure 11.
3.3.3 Noise incidence direction

To examine the influence of primary noise incidence three setups were tested. For all of these setups the reference microphones were placed 0.3 meters apart on a stand one meter behind the chair and the speakers were symmetrically placed 0.15 meters from each side of the head model. The error microphones were placed at each ear of the head model.

In the first setup the primary noise source was placed 2 meters behind the chair. See Figure 12.

In the second setup the primary noise source was placed one meter in front of the chair facing the chair. See Figure 13.
In the third setup the primary noise came from several sources that were randomly placed in the train compartment studio. This was an attempt to imitate a diffuse field incidence. Figure 14 shows the principle setup for diffuse noise incidence.
4 Results

4.1 Extension of the quiet zone

The silent zone was measured when the primary noise source was broadband random noise between 200 and 2000Hz. The reduction in dB with the ANC system was measured in third octave bands. For low frequencies the 10 dB quiet zone is large. The quiet zone decreases with increasing frequency. In figures 15 to 22 the shape of the quiet zones are shown.

Figure 15, The quiet zone at 250 Hz
Figure 16, The quiet zone at 315 Hz
Figure 17, The quiet zone at 400 Hz
Figure 18, The quiet zone at 500 Hz
Figure 19, The quiet zone at 630 Hz
Figure 20, The quiet zone at 800 Hz
4.2 Single frequency reduction

The reduction and extension of the quiet zone for four pure tone cancellations can be seen in figures 23 to 26.

The measured attenuation levels for single frequency cancellation are higher than for broadband noise and larger quiet zones at higher frequencies are measured.
4.3 Head movements

The performance of the ANC system is highly affected by head movements. There were no measured reductions above the 630 Hz third octave band for small head movements. The quiet zones for the 400 Hz third octave band measured for the deviating positions are compared to the tuned center position in figures 27 to 32. In Appendix the corresponding figures for the third octave bands 250 to 630 Hz can be seen.
As can be seen the shape of the quiet zone changes when the head position is different. Higher frequencies with shorter wavelengths are more affected by head movements. For example, if the head is displaced 0.1 meter from the system identified position this corresponds to roughly 70 degrees phase error for a wave at frequency 630 Hz. A displacement that leads to 90 degrees phase error will have no attenuating effect and larger phase errors than 90 degrees will lead to elevation of the noise level. For lower frequencies the quiet zone is still large but the attenuation level decreases with head displacements.

4.4 Reference microphone position

The reductions for varying reference microphones positioning are shown in Table 1

<table>
<thead>
<tr>
<th>Ref. microphones position, meters behind chair [m]</th>
<th>Left ear total reduction [dB]</th>
<th>Right ear total reduction [dB]</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.5</td>
<td>17</td>
<td>17</td>
</tr>
<tr>
<td>1.0</td>
<td>14</td>
<td>17</td>
</tr>
<tr>
<td>1.5</td>
<td>7</td>
<td>7</td>
</tr>
</tbody>
</table>

Table 1, Reduction for varying reference microphone position

As the distance from the chair increases so does the distance between the error and reference microphones. A comparison of the coherences between the error and reference microphones can be seen in Figure 33 when the reference microphones are placed 0.5 and 1.5 meters away from the chair.

As can be seen the coherences between the reference and error microphones are higher when the reference microphones are placed 0.5 meters behind the chair and hence also further away from the primary noise source.
4.5 Speaker position

The reduction for varying speaker positions are shown in Table 2

<table>
<thead>
<tr>
<th>Speaker position, distance between speaker and error microphones [cm]</th>
<th>Left ear total reduction [dB]</th>
<th>Right ear total reduction [dB]</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>15</td>
<td>16</td>
</tr>
<tr>
<td>10</td>
<td>15</td>
<td>15</td>
</tr>
<tr>
<td>15</td>
<td>18</td>
<td>17</td>
</tr>
<tr>
<td>20</td>
<td>18</td>
<td>16</td>
</tr>
<tr>
<td>25</td>
<td>18</td>
<td>17</td>
</tr>
</tbody>
</table>

Table 2, Reduction for varying speaker distance

Varying the speaker position had a small effect on the level of attenuation.

4.6 Noise incidence

The reduction for varying noise incidence is shown in Table 3

<table>
<thead>
<tr>
<th>Noise incidence</th>
<th>Left ear total reduction [dB]</th>
<th>Right ear total reduction [dB]</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 meters behind chair</td>
<td>14</td>
<td>17</td>
</tr>
<tr>
<td>1 meter in front of chair</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Diffuse incidence</td>
<td>-17</td>
<td>-17</td>
</tr>
</tbody>
</table>

Table 3, Reduction for varying noise incidence

When the noise source stood two meter behind the chair a reduction of 17 dB was measured at the right ear. The reduction was zero when the noise source was in front of the chair and for the case of simulated diffuse incidence the primary noise field was elevated with 17 dB and the ANC system played back high level noise which could be heard in the whole train compartment studio.

It should be noted that the basic design of this ANC system does not allow controlling diffuse fields and noise from other directions than coming from behind the reference microphone and the above results in Table 3 are hence the expected ones for this type of ANC system.
5 Discussion and Conclusions

The main goal with this thesis was to find out more about ANC system in order to determine if it is a suitable technique for reducing noise in train passenger compartments. Another goal was to implement an ANC system and try to realize local quiet zones around a passenger head in a train compartment studio.

From the experiments on how to place the system components some conclusions could be made. However, since the ANC system is so complex it is difficult to keep all parameters fixed while varying only one. For example when varying the speaker position the transfer function between the speakers and error microphones will change. At the same time changing the speaker positions will affect the transfer function between the reference and error microphones. Hence it is difficult to draw any clear conclusions regarding only one parameter.

From the experiments when varying the reference microphone positions the general conclusion is that the coherence between the reference and error microphones needs to be high for good results. The system uses the virtual microphone technique and estimates the transfer function between the reference and error microphones in order to project the quiet zone at the error microphones positions using only the input from the reference microphones and the estimated transfer functions. If the coherence between the error and reference microphones is low the quality of the estimated transfer functions will be bad and the system performance will consequently be reduced.

By looking at the coherence plots in figure 32 the varying results can be explained. When the coherence is high the performance is better. The actual distance between the reference and error microphones will have some influence on the coherence between them but in this case the low coherence is probably due to low signal to noise ratio. Setting the controller parameters differently could help getting higher coherence between the reference and error microphones even at some distance apart. For all setups the reference signal should be highly correlated with the primary noise which probably was the case but the tested ANC system gives no such information.

The results from varying reference microphone positions show that the performance was best for the shortest distance between the error and reference microphones. The virtual microphone technique is based on the assumption that the primary pressure is the same the reference and error microphone position [11]. An assumption which will be more valid at low frequencies where the phase shift between the reference and error microphones will be smaller, depending on the distance between them. This can also explain why the performance was best for a small reference to error microphone distance.

The experiments for varying speaker positions indicated that the distance between speaker and error microphone had little effect on the system performance. One could see that the overall SPL was reduced with 15 dB or more even though the speaker distance to error microphone varied from 5 to 25 centimeters. Some trend of higher reduction for larger distance between speaker and error microphone could be recognized.

In the experiments for varying speaker positions the primary noise source was placed 2 meters behind the chair and when arriving to the chair the noise incidence can be approximated as plane wave propagation. The speakers were placed orthogonal to the noise incidence and
varying the distance relative to the error microphones had little effect. For this setup the relative position between the secondary sources and the primary noise field was nearly the same for all speaker positions. The reason why this setup was tested was due to an intended implementation. Further experiments are however planned to investigate how the positioning of the secondary sources relative to the primary noise field affects the system performance. To better interfere with the primary noise field in larger areas the orientation of the speakers then will be in the same direction as the primary noise field which could enable larger quiet zones.

Having larger quiet zones would make an ANC implementation a lot more practical. However in laboratory experiments the primary noise source position and incidence is know. This would of course not be the case in a real implantation and knowing how to orientate the speakers would not be obvious. If the noise incidence could be controlled in some way by for example screens it would help in both know how to place the speakers and how to place the reference microphones.

When testing different noise incidence the rest of the system was kept in the same positions for all three cases. As the noise source was 2 meters behind the chair and the reference microphones 1 meter behind the performance was rather good. When placing the noise source in front of the chair and keeping the reference microphones behind the chair the system could not perform at all. In that setup the noise passes the speakers before it reaches the reference microphones. This meant that the reference signal did not provide any advance information to the controller which is a necessary requirement for a causal filter. Since the causality constraint is violated the controller filter is forced to act like a prediction filter. The performance of the system is then dependent on the predictability of the noise. If the noise signal would have been periodic then the system still would have been able to perform well. Generally the performance is better the narrower the noise spectrum is but if the signal is random the performance will be severely reduced. In the experiment the noise signal was broadband random noise which could not be attenuated at all by the ANC system.

A test when trying to simulate a diffuse field was performed which lead to an elevation of the sound field in the whole train compartment studio. This is probably one of the big challenges for implementing ANC in train compartments. In order to control a diffuse sound field more reference microphones are needed. When implementing feed forward ANC in a duct one reference microphone is sufficient. An open environment with diffuse incidence would actually need a lot more than one microphone to be able to control the noise. But a lot more meaning infinitely microphones would be impractical. Instead one could try to identify the dominating sources in the environment and try to place a reference microphone near each. By doing so a correlated reference signal is withheld from each dominating noise source and ANC could be effective. The problem with doing so is that the sources could be very many and that the dominating sources can change both in time and space. At one speed of the train the noise from the floor can be dominating and in another speed maybe the windows radiates a lot of noise. The environment in the train compartment also change during the travel as passengers move around in the compartment maybe blocking the reference microphones or such. Another possibility is to place the reference microphones in close proximity to the chair at which the ANC system shall control at. Then the problem with changing environment and shifting sources could be solved. However, by putting the reference microphones close to the chair and hence close to the speakers there is risk for feedback.
Another way of solving the problem of many incidence directions could, as previously mentioned, be to screen away the incidence from some directions. By doing so the incidence to the chair is more deterministic and hence easier to control. But introducing screens in the train compartment would also reduce the space and maybe interfere with other comfort aspects. If screens are not to be used, more than two reference microphones will definitely be needed to control the sound field in the train compartment. To avoid feedback effects the reference microphones must not be placed too close to the speakers but still close to the chair to minimize the effects of environment changes and varying dominating noise sources.

The crucial aspect for determine whether or not ANC is a suitable technique for reducing noise locally at the passenger head position is the size of the quiet zones. If the zones are too small only small movements away from the pre-identified position will make the ear move outside the quiet zone. The ANC system is designed to cancel out the primary noise at the error microphones position but away from these positions there is nothing guaranteeing that positive instead of negative interference occurs. This means moving outside the quiet zone can result in moving into a zone where the noise level is higher than the original primary noise level. The quiet zone is as can be seen in figures 13-20 frequency dependent and it decreases with increasing frequency. This has to do with the wavelength of the sound which is shorter in higher frequencies. According to [6] the size of the 10 dB attenuation zone is approximately one tenth of the wavelength, which the results from the experiments support to some extent. This means that the 10 dB quiet zone only is a couple of centimeters wide or even smaller in frequencies above 1000 Hz. Still one could argue that a reduction of 5 dB also would be a noticeable difference in SPL and the 5 dB quiet zone is generally larger and also extent higher in frequency. Also, the dominating part of a typical train noise spectrum is in low frequencies below 1000 Hz [3] where the quiet zones still are relatively large.

The performance of the ANC system is highly affected by head movements. Head movement changes the acoustic environment and the transfer functions that the ANC system uses are hence no longer good estimates. This reduces the ANC system performance in particular higher frequencies. As can be seen in figures 27 to 31 the shape of the quiet zone changes when the head position is different. This has to do with that the head is moved away from the quiet zone but also as the head is moved the acoustic environment is changed which changes the transfer functions and ultimately affects the quiet zone.

It seems the system is more sensitive to sideward movements than to forward/ backward movements. As the head is moved sideward towards one speaker the ear closest to that speaker will experience an increase in SPL. The system tries to project the quiet zone at the pre-identified position and to do this the output level needs to be sufficiently high. Because of this when the ear moves closer to the speaker it will experience a higher noise level than the original. The other ear is still in its quiet zone but not in the ideal position. Another explanation why the system is sensitive to sideward head movements could be that the transfer functions are more affected to this kind of movements. The head for example has a blocking effect on the speakers which is more evident for sideward displacements. It should be noted that the Styrofoam head model only gives information on the left side of the head. In the experiments the setups for the speakers and microphones have been symmetrical so the right ear side should have experienced a similar reduction as on the left side.
To deal with the problem of head movements for the tested ANC system without expanding it with more reference microphones and/or secondary sources some sort of tracking system with an intrinsic plant selector would probably be the easiest way of improving the system performance.

The quiet zone size and problems with head movements is not a decoupled issue. Since if the size of the quiet zones were sufficiently large then head movements would not be a problem. To enlarge the quiet zones more secondary sources could be used to create something similar to a virtual sound barrier. Having for example four speakers to control instead of two could give a significant improvement in both quiet zone size and attenuation level and would still be practically possible to implement at a chair in a train compartment.

There are still issues to solve before an ANC system can control the interior noise in a train compartment. In this thesis the performance of a given ANC system was evaluated. The quiet zone size that the system succeeded to create would be too small for an implementation since the passenger would risk moving out of the quiet zone for even remote head movements. There are also problems with multiple noise incidences and diffuse field for the tested ANC system. Solutions to these problems and suggested improvements of the system have been presented. If the issues were dealt with soon ANC would be possible to implement and the passenger comfort would be taken to a new level.
References

[13] Tom Waite, Frequency domain active noise control with ultrasonic tracking, Iowa State University, 2010
[14] Christopher Rose, Active voice control: An implementation of active noise control for canceling speech, University Honors College, Alabama, 2007
Appendix

Head movements at 250 Hz

Picture 1, Center position 250 Hz

Picture 2, Downward position 250 Hz

Picture 3, Forward position 250 Hz

Picture 4, Backward position 250 Hz

Picture 5, Left position 250 Hz

Picture 6, Right position 250 Hz
Head Movements at 315 Hz

Picture 7, Center position 315 Hz

Picture 8, Downward position 315 Hz

Picture 9, Forward position 315 Hz

Picture 10, Backward position 315 Hz

Picture 11, Left position 315 Hz

Picture 12, Right position 315 Hz
Head movements at 400 Hz

Picture 13, Center position 400 Hz

Picture 14, Downward position 400 Hz

Picture 15, Forward position 400 Hz

Picture 16, Backward position 400 Hz

Picture 17, Left position 400 Hz

Picture 18, Right position 400 Hz
Head movements at 500 Hz

Picture 19, Center position 500 Hz

Picture 20, Downward position 500 Hz

Picture 21, Forward position 500 Hz

Picture 22, Backward position 500 Hz

Picture 23, Left position 500 Hz

Picture 24, Right position 500 Hz
Head movements at 630 Hz

Picture 25, Center position 630 Hz

Picture 26, Downward position 630 Hz

Picture 27, Forward position 630 Hz

Picture 28, Backward position 630 Hz

Picture 29, Left position 630 Hz

Picture 30, Right position 630 Hz